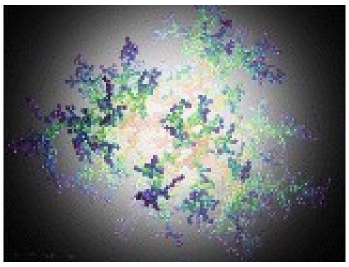
TCP/IP In More Detail



My previous page examined the Internet Protocol (IP) in considerable detail. The Internet Protocol handles the lower-layer functionality. Now you can look at the transport layer, where Transmission Control Protocol (TCP) comes into play.

TCP is one of the most widely used transport layer protocols, expanding from its original implementation on the ARPAnet to connect commercial sites all over the world. In my first web page, you saw the OSI seven-layer model, which bears a striking resemblance to TCP/IP's layered model; so it is not surprising that many of the features of the OSI transport layer were based on TCP.

In theory, a transport layer protocol could be a very simple software routine, but TCP cannot be called simple. Why use a transport layer that is as complex as TCP? The most important reason depends on IP's unreliability. As seen in my previous page, IP does not guarantee delivery of a datagram; it is a connectionless system. IP simply handles the routing of datagrams; and if problems occur, IP discards the packet without a second thought (generating an error message back to the sender in the process). The task of ascertaining the status of the datagrams sent over a network and handling the resending of information if parts have been discarded falls to TCP, which can be thought of as "*riding shotgun*" over IP.

Most users think of TCP and IP as a tightly knit pair, but TCP can (and frequently is) used with other transport protocols. For example, TCP or parts of it are used in the File Transfer Protocol (FTP) and the Simple Mail Transfer Protocol (SMTP), both of which do not use IP.

What Is TCP?

The Transmission Control Protocol provides a considerable number of services to the IP layer and the upper layers. Most importantly, it provides a connection-oriented protocol to the upper layers that enable an application to be sure that a datagram sent out over the network was received in its entirety. In this role, TCP acts as a message-validation protocol providing reliable communications. If a datagram is corrupted or lost, it is usually TCP (not the applications in the higher layers) that handles the retransmission.

TCP is not a piece of software. It is a communications protocol.

TCP manages the flow of datagrams from the higher layers, as well as incoming datagrams from the IP layer. It has to ensure that priorities and security are respected. TCP must be capable of handling the termination of an application above it that was expecting incoming datagrams, as well as failures in the lower layers. TCP also must maintain a state table of all data streams in and out of the TCP layer. The isolation of these services in a separate layer enables applications to be designed without regard to flow control or message reliability. Without the TCP layer, each application would have to implement the services themselves, which is a waste of resources.

TCP resides in the transport layer, positioned above IP but below the upper layers and their applications, as shown in Figure 1. TCP resides only on devices that actually process datagrams, ensuring that the datagram has gone from the source to target machines. It does not reside on a device that simply routes datagrams, so there is no TCP layer in a gateway. This makes sense, because on a gateway the datagram has no need to go higher in the layered model than the IP layer.

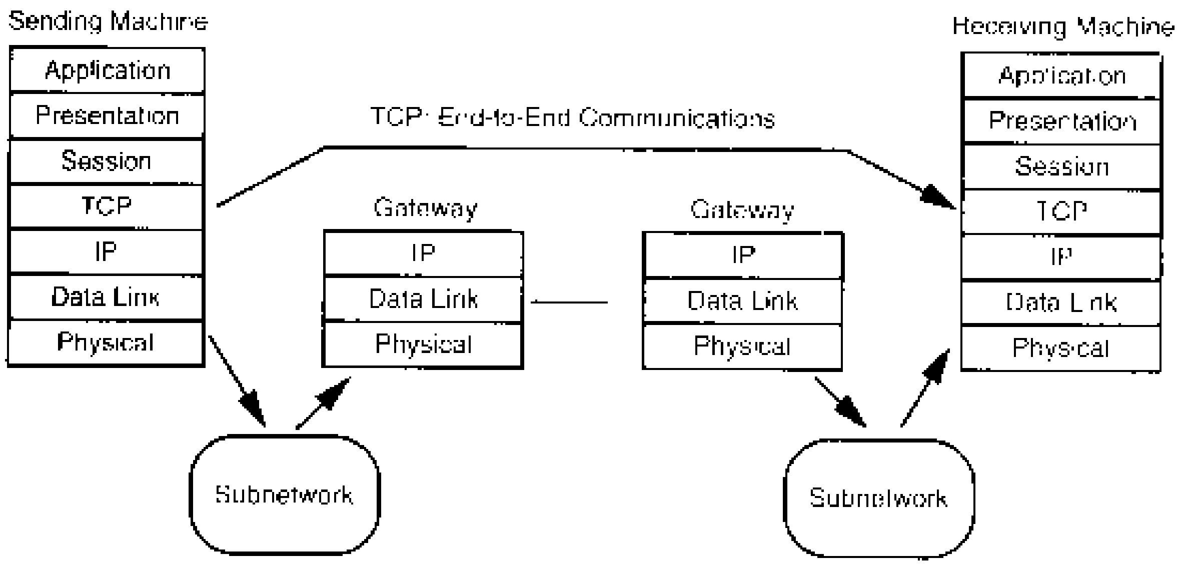


Figure 1 - TCP provides end-to-end communications

Because TCP is a connection-oriented protocol responsible for ensuring the transfer of a datagram from the source to destination machine (end-to-end communications), TCP must receive communications messages from the destination machine to acknowledge receipt of the datagram. The term *virtual circuit* is usually used to refer to the handshaking that goes on between the two end machines, most of which are simple acknowledgment messages (either confirmation of receipt or a failure code) and datagram sequence numbers.

Following a Message

To illustrate the role of TCP, it is instructive to follow a sample message between two machines. The processes are simplified at this stage, to be expanded upon later. The message originates from an application and is passed to TCP from the next higher layer in the architecture through some protocol (often referred to as an *upper-layer protocol*, or *ULP*, to indicate that it resides above TCP). The message is passed as a *stream* - a sequence of individual characters sent asynchronously. This is in contrast to most protocols, which use fixed blocks of data. This can pose some conversion problems with applications that handle only formally constructed blocks of data or insist on fixed-size messages.

TCP receives the stream of bytes and assembles them into *TCP segments*, or packets. In the process of assembling the segment, header information is attached. Each segment has a checksum calculated and embedded within the header, as well as a sequence number if there is more than one segment in the entire message. The length of the segment is usually determined by TCP or by a system value set by the system administrator.

If two-way communication is required (such as with Telnet or FTP), and prior to passing the segment to IP for routing, a connection (virtual circuit) between the sending and receiving machines is established. This process starts with the sending TCP software issuing a request for a TCP connection with the receiving machine. In the message is a unique number (called a *socket number*) that identifies the sending machine's connection. The receiving TCP software assigns its own unique socket number and sends it back to the original machine. The two unique numbers then define the connection between the two machines until the virtual circuit is terminated.

After the virtual circuit is established, TCP sends the segment to the IP software, which then issues the message over the network as a datagram. After winding its way over the network, the receiving machine's IP passes the received segment up to the recipient machine's TCP layer, where it is processed and passed up to the applications above it using an upper-layer protocol.

If the message was more than one segment long, the receiving TCP software reassembles the message using the sequence numbers contained in each segment's header. If a segment is missing or corrupt (which can be determined from the checksum), TCP returns a message with the faulty sequence number in the body. The originating TCP software can then resend the bad segment.

If only one segment is used for the entire message, after comparing the segment's checksum with a newly calculated value, the receiving TCP software can generate either a positive acknowledgment (ACK) or a request to resend the segment and route it back to the sending layer.

The receiving machine's TCP implementation can perform a simple flow control to prevent buffer overload. It does this by sending a *window value* to the sending machine, following which the sender can send only enough bytes to fill the window. After that, the sender must wait for another window value to be received. This provides a handshaking protocol between the two machines, although it does slow down the transmission time and slightly increase network traffic.

The use of a sliding window is more efficient than a single block send and acknowledgment scheme because of delays waiting for the acknowledgment. By implementing a sliding window, several blocks can be sent at once. A properly configured sliding window protocol will provide a much higher throughput.

As with most connection-based protocols, timers are an important aspect of TCP. The use of a timer ensures that an undue wait is not involved while waiting for an ACK or an error message. If the timers expire, an incomplete transmission is assumed. Usually an expiring timer before the sending of an acknowledgment message will cause a retransmission of the datagram from the originating machine.

Timers can cause some problems with TCP. The specifications for TCP provide for the acknowledgment of only the highest datagram number that has been received without error, but this cannot properly handle fragmentary reception. If a message is composed of several datagrams that arrive out of order, the specification states that TCP cannot acknowledge the reception of the message until all the datagrams have been received. So even if all but one datagram in the middle of the sequence has been successfully received, a timer may expire and cause all the datagrams to be resent. With large messages, this can cause an increase in network traffic.

If the receiving TCP software receives duplicate datagrams (as may occur with a retransmission after a timeout or due to a duplicate transmission from IP), the receiving version of TCP will discard any duplicate datagrams, without bothering with an error message. After all, the sending system cares only that the message was received - not how many copies were received.

TCP does not have a *negative acknowledgment* (NAK) function, relying on a timer to indicate lack of acknowledgment. If the timer has expired after sending the datagram without receiving an acknowledgment of receipt, the datagram is assumed to have been lost and is retransmitted. The sendingTCP software keeps copies of all unacknowledged datagrams in a buffer until they have been properly acknowledged. When this happens, the retransmission timer is stopped and the datagram is removed from the buffer.

TCP supports a *push* function from the upper-layer protocols. A push is used when an application wants to immediately send data and confirm that a message passed to TCP has been successfully transmitted. To do this, a *push flag* is set in the ULP connection, instructing TCP to forward any buffered information from the application to the destination as soon as possible (as opposed to holding it in the buffer until it is ready to transmit it).

Ports and Sockets

All upper-layer applications that use TCP (or UDP) have a port number that identifies the application. In theory, port numbers can be assigned on individual machines, or however the administrator desires, but some conventions have been adopted to allow better communication between TCP implementations. This enables the port number to identify the type of service that one TCP system is requesting from another. Port numbers can be changed, although this can cause difficulties. Most systems maintain a file of port numbers and their corresponding service.

Typically, port numbers above 255 are reserved for private use of the local machine, but numbers below 255 are used for frequently used processes. A list of frequently used port numbers is published by the Internet Assigned Numbers Authority. The commonly used port numbers on this list are shown in Table 1. The numbers 0 and 255 are reserved.

Port Number	Process Name	Description
1	TCPMUX	TCP Port Service Multiplexer
5	RJE	Remote Job Entry
7	ECHO	Echo
9	DISCARD	Discard
11	USERS	Active Users
13	DAYTIME	Daytime
17	Quote	Quotation of the Day
19	CHARGEN	Character generator
20	FTP-DATA	File Transfer Protocol - Data
21	FTP	File Transfer Protocol - Control
23	TELNET	Telnet
25	SMTP	Simple Mail Transfer Protocol
27	NSW-FE	NSW User System Front End
29	MSG-ICP	MSG-ICP
31	MSG-AUTH	MSG Authentication
33	DSP	Display Support Protocol
35		Private Print Servers
37	TIME	Time
39	RLP	Resource Location Protocol
41	GRAPHICS	Graphics
42	NAMESERV	Host Name Server
43	NICNAME	Who Is
49	LOGIN	Login Host Protocol
53	DOMAIN	Domain Name Server
67	BOOTPS	Bootstrap Protocol Server
68	BOOTPC	Bootstrap Protocol Client
69	TFTP	Trivial File Transfer Protocol
79	FINGER	Finger

101	HOSTNAME	NIC Host Name Server
102	ISO-TSAP	ISO TSAP
103	X400	X.400
104	X400SND	X.400 SND
105	CSNET-NS	CSNET Mailbox Name Server
109	POP2	Post Office Protocol v2
110	POP3	Post Office Protocol v3
111	RPC	Sun RPC Portmap
137	NETBIOS-NS	NETBIOS Name Server
138	NETBIOS-DG	NETBIOS Datagram Service
139	NETBIOS-SS	NETBIOS Session Service
146	ISO-TP0	ISO TP0
147	ISO-IP	ISO IP
150	SQL-NET	SQL NET
153	SGMP	SGMP
156	SQLSRV	SQL Service
160	SGMP-TRAPS	SGMP TRAPS
161	SNMP	SNMP
162	SNMPTRAP	SNMPTRAP
163	CMIP-MANAGE	CMIP/TCP Manager
164	CMIP-AGENT	CMIP/TCP Agent
165	XNS-Courier	Xerox
179	BGP	Border Gateway Protocol

Table 1 - Frequently used TCP port numbers

Each communication circuit in to and out of the TCP layer is uniquely identified by a combination of two numbers, which together are called a *socket*. The socket is composed of the IP address of the machine and the port number used by the TCP software. There is a socket on both the sending and receiving machines. Because the IP address is unique across the internetwork and the port numbers will be unique to the individual machine, the socket numbers will also be unique across the entire internetwork. This enables a process to talk to another process across the network, based entirely on the socket number.

TCP uses the connection (not the protocol port) as fundamental elements. A completed connection has two end points. This enables a protocol port to be used for several connections at the same time (multiplexing).

The last couple of paragraphs examined the process of establishing a message. During the process, the sending TCP requests a connection with the receiving TCP, using the unique socket numbers. This process is shown in Figure 2. If the sending TCP wants to establish a Telnet session from its port number 350, the socket number would be composed of the source machine's IP address and the port number (350), and the message would have a destination port number of 23 (Telnet's port number). The receiving TCP will have a source port of 23 (Telnet) and a destination port of 350 (the sending machine's port).

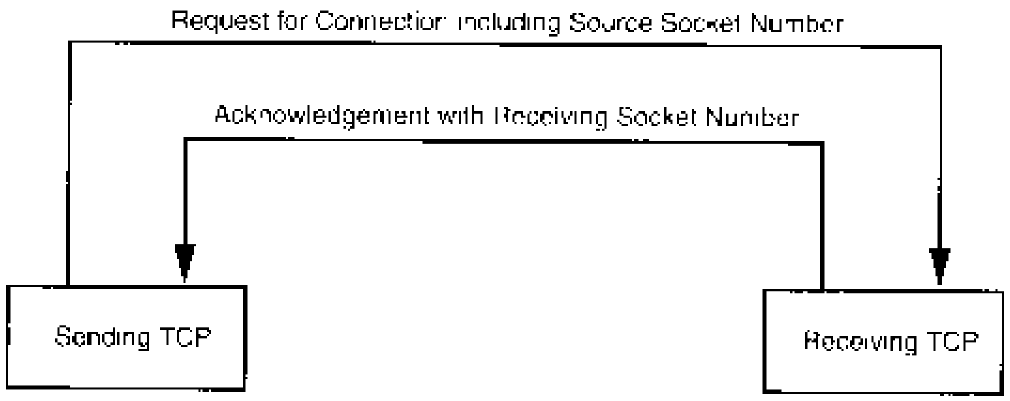

Request for Connection including Source Socket Number
Acknowledgement with Receiving Socket Number
Sending TCP
Receiving TCP

Figure 2 - Setting up a virtual circuit with socket numbers

The sending and receiving machines maintain a port table, which lists all active port numbers. The two machines involved will have reversed entries for each session between the two. This is called *binding* and is shown in Figure 3. The source and destination numbers are simply reversed for each connection in the port table. Of course, the IP addresses, and hence the socket number, will be different.

If the sending machine is requesting more than one connection, the source port numbers would be different, even though the destination port numbers might be the same. For example, if the sending machine were trying to establish three Telnet sessions simultaneously, the source machine port numbers might be 350, 351, and 352, while the destination port numbers would all be 23.

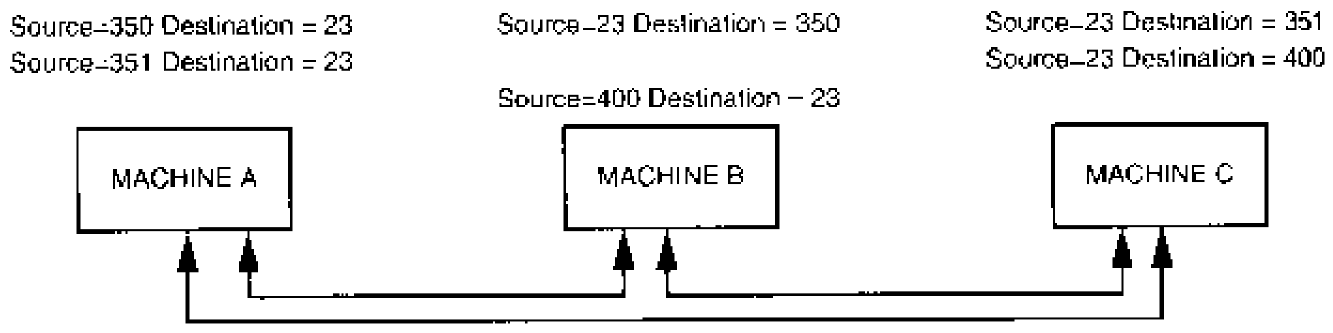

Source=350 Destination = 23
Source=351 Destination = 23
Source=23 Destination = 350
Source=400 Destination = 23
Source=23 Destination = 351
Source=23 Destination = 400
MACHINE A
MACHINE B
MACHINE C

Figure 3 - Binding entries in port tables

It is possible for more than one machine to share the same destination socket - a process called *multiplexing*. In Figure 4, three machines are establishing Telnet sessions with a destination. They all use destination port 23, which is port multiplexing. Because the datagrams emerging from the port have the full socket information (with unique IP addresses), there is no confusion as to which machine a datagram is destined.

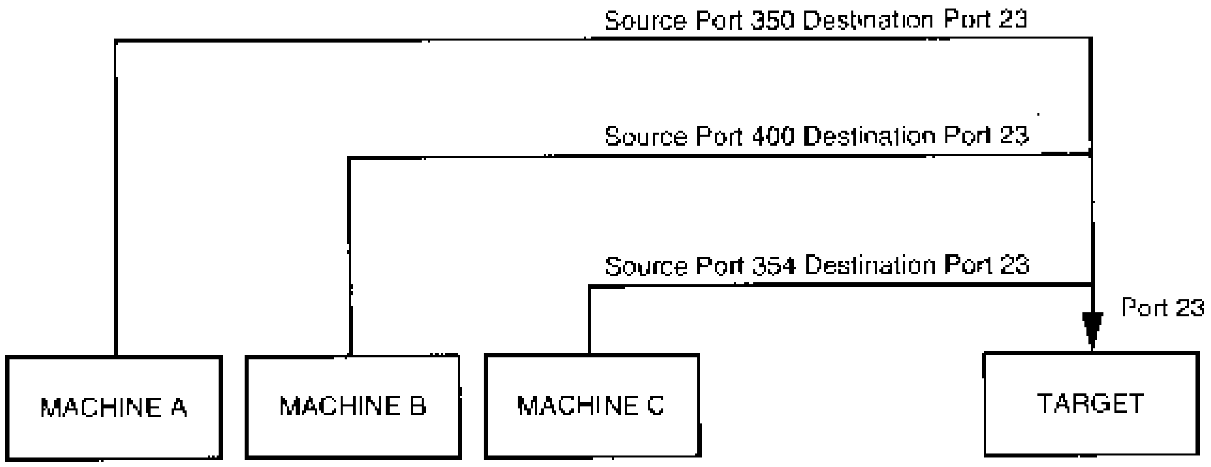

Source Port 350 Destination Port 23
Source Port 400 Destination Port 23
Source Port 354 Destination Port 23
Port 23
MACHINE A
MACHINE B
MACHINE C
TARGET

Figure 4 - Multiplexing one destination port

When multiple sockets are established, it is conceivable that more than one machine will send a connection request with the same source and destination ports. However, the IP addresses for the two machines will be different, so the sockets are still uniquely identified despite identical source and destination port numbers.

TCP Communications with the Upper Layers

TCP must communicate with applications in the upper layer and a network system in the layer below. There are several messages defined for the upper-layer protocol to TCP communications, but there is no defined method for TCP to talk to lower layers (usually, but not necessarily, IP). TCP expects the layer beneath it to define the communication method. It is usually assumed that TCP and the transport layer communicate asynchronously.

The TCP to upper-layer protocol (ULP) communication method is well-defined, consisting of a set of *service request primitives*. The primitives involved in ULP to TCP communications are shown below in Table 2.

Command	Parameters Expected
ABORT	Local connection name.
ACTIVE-OPEN	Local port, remote socket.
	Optional: ULP timeout, timeout action, precedence, security, options
ACTIVE-OPEN-WITH-DATA	Source port, destination socket, data, data length, push flag, urgent flag

	Optional: ULP timeout, timeout action, precedence, security
ALLOCATE	Local connection name, data length
CLOSE	Local connection name
FULL-PASSIVE-OPEN	Local port, destination socket
	Optional: ULP timeout, timeout action, precedence, security, options
RECEIVE	Local connection name, buffer address, byte count, push flag, urgent flag
SEND	Local connection name, buffer address, data length, push flag, urgent flag
	Optional: ULP timeout, timeout action
STATUS	Local connection name
UNSPECIFIED-PASSIVE-OPEN	Local port
	Optional: ULP timeout, timeout action, precedence, security, options
CLOSING	Local connection name
DELIVER	Local connection name, buffer address, data length, urgent flag
ERROR	Local connection name, error description
OPEN-FAILURE	Local connection name
OPEN-ID	Local connection name, remote socket, destination address
OPEN-SUCCESS	Local connection name
STATUS RESPONSE	Local connection name, source port, source address, remote socket, connection state, receive window, send window, amount waiting ACK, amount waiting receipt, urgent mode, precedence, security, timeout, timeout action
TERMINATE	Local connection name, description

Table 2 - ULP-TCP service primitives

Passive and Active Ports

TCP enables two methods to establish a connection: active and passive. An *active* connection establishment happens when TCP issues a request for the connection, based on an instruction from an upper-level protocol that provides the socket number. A *passive* approach happens when the upper-level protocol instructs TCP to wait for the arrival of connection requests from a remote system (usually from an active open instruction). When TCP receives the request, it assigns a port number. This enables a connection to proceed rapidly, without waiting for the active process.

There are two passive open primitives. A *specified passive open* creates a connection when the precedence level and security level are acceptable. An *unspecified passive open* opens the port to any request. The latter is used by servers that are waiting for clients of an unknown type to connect to it.

TCP has strict rules about the use of passive and active connection processes. Usually a passive open is performed on one machine, whereas an active open is performed on the other with specific information about the socket number, precedence (priority), and security levels.

Although most TCP connections are established by an active request to a passive port, it is possible to open a connection without a passive port waiting. In this case, the TCP that sends a request for a connection will include both a local socket number and the remote socket number. If the receiving TCP is configured to allow the request (based on the precedence and security settings, as well as application-based criteria), the connection can be opened.

TCP Timers

TCP uses several timers to ensure that excessive delays are not encountered during communications. Several of these timers are elegant, handling problems that are not immediately obvious at first analysis. Each of the timers used by TCP is examined in the following sections, which reveal its role in ensuring data is properly sent from one connection to another.

Retransmission Timer

The retransmission timer manages retransmission timeouts (RTOs), which occur when a preset interval between the sending of a datagram and the returning acknowledgment is exceeded. The value of the timeout tends to vary, depending on the network type, to compensate for speed differences. If the timer expires, the datagram is retransmitted with an adjusted RTO, which is usually increased exponentially to a maximum preset limit. If the maximum limit is exceeded, connection failure is assumed and error messages are passed back to the upper-layer application.

Values for the timeout are determined by measuring the average time data takes to be transmitted to another machine and the acknowledgment received back, which is called the *round trip time*, or RTT. From experiments, these RTTs are averaged by a formula that develops an expected value, called the *smoothed round trip time*, or SRTT. This value is then increased to account for unforeseen delays.

Quiet Timer

After a TCP connection is closed, it is possible for datagrams that are still making their way through the network to attempt to access the closed port. The quiet timer is intended to prevent the just-closed port from reopening again quickly and receiving these last datagrams.

The quiet timer is usually set to twice the maximum segment lifetime (the same value as the Time-To-Live field in an IP header), ensuring that all segments still heading for the port have been discarded. Typically, this can result in a port being unavailable for up to 30 seconds, prompting error messages when other applications attempt to access the port during this interval.

Persistence Timer

The persistence timer handles a fairly rare occurrence. It is conceivable that a receive window will have a value of 0, causing the sending machine to pause transmission. The message to restart sending may be lost, causing an infinite delay. The persistence timer waits a preset time and then sends a 1-byte segment at predetermined intervals to ensure that the receiving machine is still clogged.

The receiving machine will resend the zero window size message after receiving one of these status segments, if it is still backlogged. If the window is open, a message giving the new value is returned and communications resumed.

Keep-Alive Timer and Idle Timer

Both the keep-alive timer and the idle timer were added to the TCP specifications after their original definition. The keep-alive timer sends an empty packet at regular intervals to ensure that the connection to the other machine is still active. If no response has been received after sending the message, by the time the idle timer has expired, the connection is assumed to be broken. The keep-alive timer value is usually set by an application, with values ranging from 5 to 45 seconds. The idle timer is usually set to 360 seconds.

TCP uses adaptive timer algorithms to accommodate delays. The timers adjust themselves to the delays experienced over a connection, altering the timer values to reflect inherent problems.

Transmission Control Blocks and Flow Control

TCP has to keep track of a lot of information about each connection. It does this through a Transmission Control Block (TCB), which contains information about the local and remote socket numbers, the send and receive buffers, security and priority values, and the current segment in the queue. The TCB also manages send and receive sequence numbers.

The TCB uses a number of variables to keep track of the send and receive status and to control the flow of information. These variables are shown in Table 3.

Variable Name	Description
SND.UNA	Send Unacknowledged
SND.NXT	Send Next
SND.WND	Send Window
SND.UP	Sequence number of last urgent set
SND.WL1	Sequence number for last window update
SND.WL2	Acknowledgment number for last window update
SND.PUSH	Sequence number of last pushed set
ISS	Initial send sequence number
RCV.NXT	Sequence number of next received set
RCV.WND	Number of sets that can be received
RCV.UP	Sequence number of last urgent data
RCV.IRS	Initial receive sequence number

Table 3 - TCP send and receive variables

Using these variables, TCP controls the flow of information between two sockets. A sample connection session helps illustrate the use of the variables. It begins with Machine A wanting to send five blocks of data to Machine B. If the window limit is seven blocks, a maximum of seven blocks can be sent without acknowledgment. The SND.UNA variable on Machine A will indicate how many blocks have been sent but are unacknowledged (5), and the SND.NXT variable will have the value of the next block in sequence (6). The value of the SND.WND variable will be 2 (seven blocks possible, minus five sent), so only two more blocks could be sent without overloading the window. Machine B will return a message with the number of blocks received, and the window limit will be adjusted accordingly.

The passage of messages back and forth can become quite complex as the sending machine forwards blocks unacknowledged up to the window limit, waiting for acknowledgment of earlier blocks that have been removed from the incoming cue, and then sending more blocks to fill the window again. The tracking of the blocks becomes a matter of bookeeping, but with large window limits and traffic across internetworks that sometimes cause blocks to go astray, the process is, in many ways, remarkable.

TCP Protocol Data Units

As mentioned earlier, TCP must communicate with IP in the layer below (using an IP-defined method) and applications in the upper layer (using the TCP-ULP primitives). TCP also must communicate with other TCP implementations across networks. To do this, it uses Protocol Data Units (PDUs), which are called *segments* in TCP parlance. The layout of the TCP PDU (commonly called the header) is shown in Figure 5.

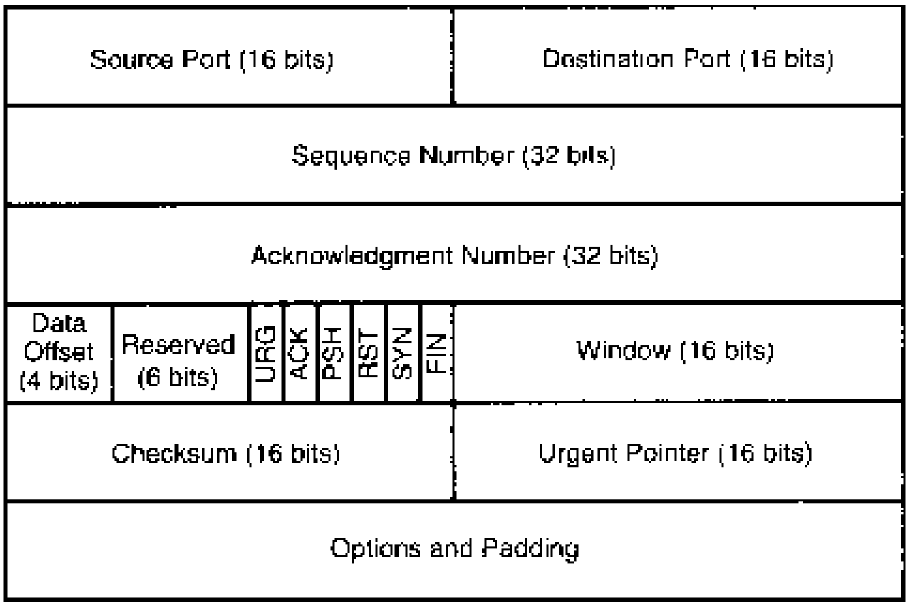


Figure 5 - The TCP Protocol Data Unit

The different fields are as follows:

Source port: A 16-bit field that identifies the local TCP user (usually an upper-layer application program).
Destination port: A 16-bit field that identifies the remote machine's TCP user.
Sequence number: A number indicating the position of the current block's position in the overall message. This number is also used between two TCP implementations to provide the initial send sequence (ISS) number.
Acknowledgment number: A number that indicates the next sequence number expected. In a backhanded manner, this also shows the sequence number of the last data received; it shows the last sequence number received plus 1.
Data offset: The number of 32-bit words that are in the TCP header. This field is used to identify the start of the data field.
Reserved: A 6-bit field reserved for future use. The 6 bits must be set to 0.
Urg flag: If on (a value of 1), indicates that the urgent pointer field is significant.
Ack flag: If on, indicates that the acknowledgment field is significant.
Psh flag: If on, indicates that the push function is to be performed.
rst flag: If on, indicates that the connection is to be reset.
Syn flag: If on, indicates that the sequence numbers are to be synchronised. This flag is used when a connection is being established.
Fin flag: If on, indicates that the sender has no more data to send. This is the equivalent of

an end-of-transmission marker.
Window: A number indicating how many blocks of data the receiving machine can accept.
Checksum: Calculated by taking the 16-bit one's complement of the one's complement sum of the 16-bit words in the header (including pseudo-header) and text together.
Urgent pointer: Used if the urg flag was set; it indicates the portion of the data message that is urgent by specifying the offset from the sequence number in the header. No specific action is taken by TCP with respect to urgent data: the action is determined by the application.
Options: Similar to the IP header option field, it is used for specifying TCP options. Each option consists of an option number (1 byte), the number of bytes in the option, and the option values. Only three options are currently defined for TCP:

0	End of option list
1	No operation
2	Maximum segment size

Padding: Filled to ensure that the header is filled to a 32-bit multiple.

Following the PDU or header is the data. The options field has one useful function: to specify the maximum buffer size a receiving TCP implementation can accommodate. Because TCP uses variable-length data areas, it is possible for a sending machine to create a segment that is longer than the receiving software can handle.

The checksum field calculates the checksum based on the entire segment size, induding a 96-bit pseudoheader that is prefixed to the TCP header during the calculation. The pseudoheader contains the source address, destination address, protocol identifier, and segment length. These are the parameters that are passed to IP when a send instruction is passed, and also the ones read by IP when delivery is attempted.

TCP and Connections

TCP has many rules imposed on how it communicates. These rules and the processes that TCP follows to establish a connection, transfer data, and terminate a connection are usually presented in state diagrams. (Because TCP is a state-driven protocol, its actions depend on the state of a flag or similar construct.) Avoiding overly complex state diagrams is difficult, so flow diagrams can be used as a useful method for understanding TCP.

Establishing a Connection

A connection can be established between two machines only if a connection between the two sockets does not exist, both machines agree to the connection, and both machines have adequate TCP resources to service the connection. If any of these conditions are not met, the connection cannot be made. The acceptance of connections can be triggered by an application or a system administration routine.

When a connection is established, it is given certain properties that are valid until the connection is closed. Typically, these will be a precedence value and a security value. These settings are agreed upon by the two applications when the connection is in the process of being established.

In most cases, a connection is expected by two applications, so they issue either active or passive open requests. Figure 6 shows a flow diagram for a TCP open. The process begins with Machine A's TCP receiving a request for a connection from its ULP, to which it sends an active open primitive to Machine B. (Refer back to Table 2 for the TCP primitives.) The segment that is constructed will have the SYN flag set on (set to 1) and will have a sequence number assigned. The diagram shows this with the notation *SYN SEQ 50* indicating that the SYN flag is on and the sequence number (Initial Send Sequence number or ISS) is 50. (Any number could have been chosen.)

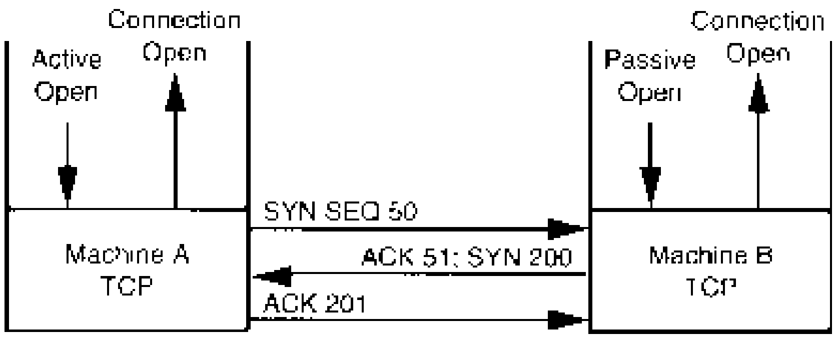


Figure 6 - Establishing a connection

The application on Machine B will have issued a passive open instruction to its TCP. When the SYN SEQ 50 segment is received, Machine B's TCP will send an acknowledgment back to Machine A with the sequence number of 51. Machine B will also set an Initial Send Sequence number of its own. The diagram shows this message as *ACK 51; SYN 200* indicating that the message is an acknowledgment with sequence number 51, it has the SYN flag set, and has an ISS of 200.

Upon receipt, Machine A sends back its own acknowledgment message with the sequence number set to 201. This is *ACK 201* in the diagram. Then, having opened and acknowledged the connection, Machine A and Machine B both send connection open messages through the ULP to the requesting applications.

It is not necessary for the remote machine to have a passive open instruction, as mentioned earlier. In this case, the sending machine provides both the sending and receiving socket numbers, as well as precedence, security, and timeout values. It is common for two applications to request an active open at the same time. This is resolved quite easily, although it does involve a little more network traffic.

Data Transfer

Transferring information is straightforward, as shown in Figure 7. For each block of data received by Machine A's TCP from the ULP, TCP encapsulates it and sends it to Machine B with an increasing sequence number. After Machine B receives the message, it acknowledges it with a segment acknowledgment that increments the next sequence number (and hence indicates that it received everything up to that sequence number). Figure 7 shows the transfer of only one segment of information - one each way.

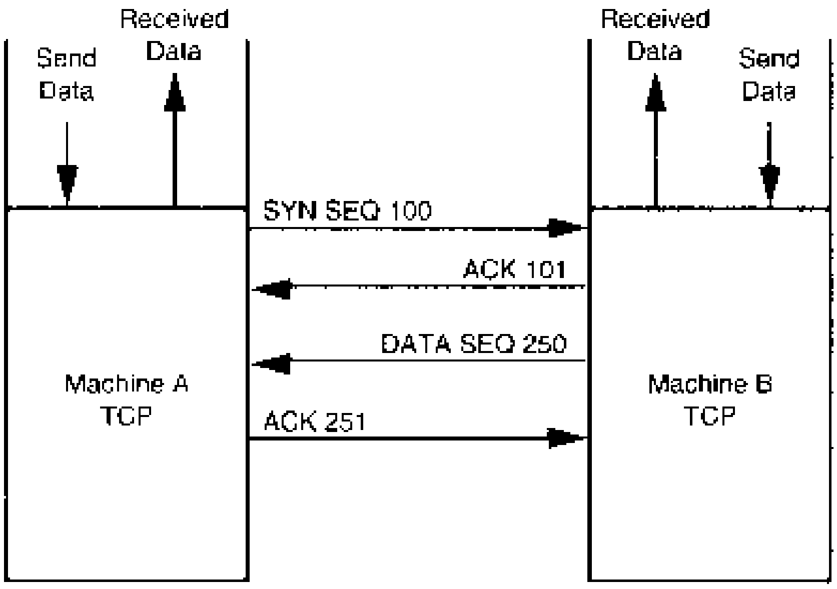


Figure 7 - Data transfers

The TCP data transport service actually embodies six different subservices:

Full duplex: Enables both ends of a connection to transmit at any time, even simultaneously.
Timeliness: The use of timers ensures that data is transmitted within a reasonable amount of time.
Ordered: Data sent from one application will be received in the same order at the other end. This occurs despite the fact that the datagrams may be received out of order through IP, as TCP reassembles the message in the correct order before passing it up to the higher layers.
Labeled: All connections have an agreed-upon precedence and security value.
Controlled flow: TCP can regulate the flow of information through the use of buffers and window limits.
Error correction: Checksums ensure that data is free of errors (within the checksum algorithm's limits).

Closing Connections

To close a connection, one of the TCPs receives a close primitive from the ULP and issues a message with the FIN flag set on. This is shown in Figure 8. In the figure, Machine A's TCP sends the request to close the connection to Machine B with the next sequence number. Machine B will then send back an acknowledgment of the request and its next sequence number. Following this, Machine B sends the close message through its ULP to the application and waits for the application to acknowledge the closure. This step is not strictly necessary; TCP can close the connection without the application's approval, but a well-behaved system would inform the application of the change in state.

After receiving approval to close the connection from the application (or after the request has timed out), Machine B's TCP sends a segment back to Machine A with the FIN flag set. Finally, Machine A acknowledges the closure and the connection is terminated.

An abrupt termination of a connection can happen when one side shuts down the socket. This can be done without any notice to the other machine and without regard to any information in transit between the two. Aside from sudden shutdowns caused by malfunctions or power outages, abrupt termination can be initiated by a user, an application, or a system monitoring routine that judges the connection worthy of termination. The other end of the connection may not realise an abrupt termination has occurred until it attempts to send a message and the timer expires.

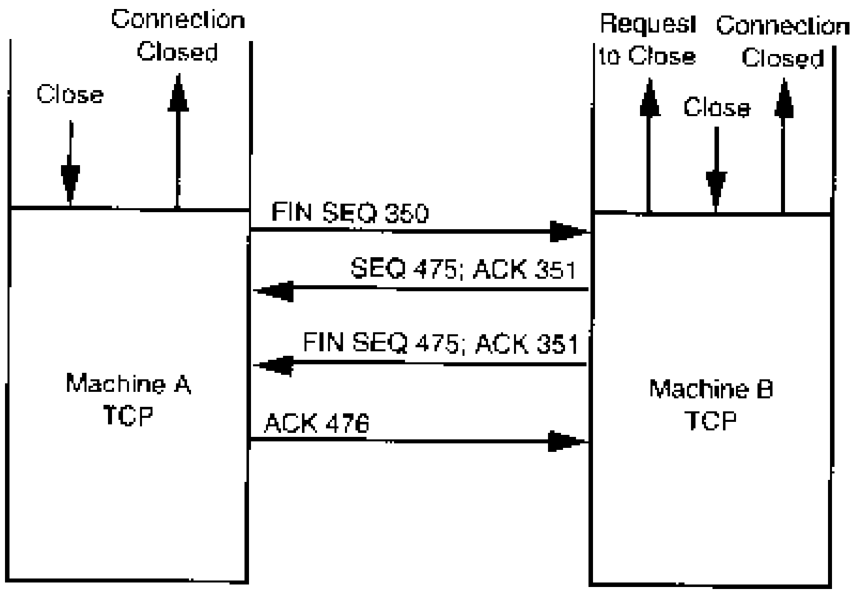
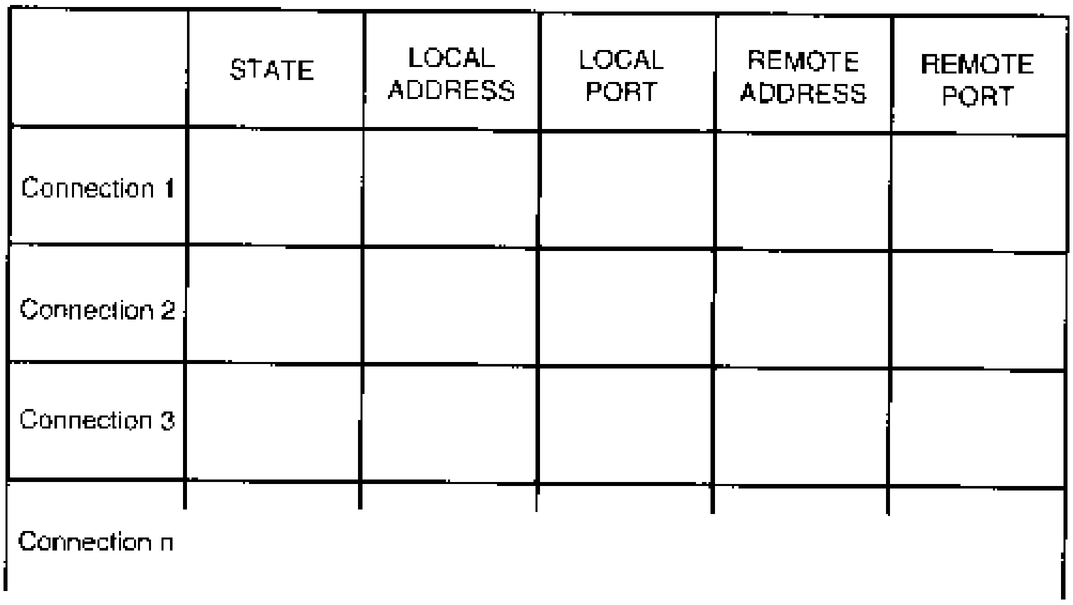


Figure 8 - Closing a connection

To keep track of all the connections, TCP uses a connection table. Each existing connection has an entry in the table that shows information about the end-to-end connection. The layout of the TCP connection table is shown in Figure 9



	STATE	LOCAL ADDRESS	LOCAL PORT	REMOTE ADDRESS	REMOTE PORT
Connection 1					
Connection 2					
Connection 3					
Connection n					

Figure 9 - The TCP connection table

The meaning of each column is as follows:

State: The state of the connection (closed, closing, listening, waiting, and so on).
Local address: The IP address for the connection. When in a listening state, this will set to 0.0.0.0.
Local port: The local port number.
Remote address: The remote's IP address.
Remote port: The port number of the remote connection.

User Datagram Protocol (UDP)

TCP is a connection-based protocol. There are times when a connectionless protocol is required, so UDP is used. UDP is used with both the Trivial File Transfer Protocol (TFTP) and the Remote Call Procedure (RCP). Connectionless communications don't provide reliability, meaning there is no indication to the sending device that a message has been received correctly. Connectionless protocols also do not offer error-recovery capabilities - which must be either ignored or provided in the higher or lower layers. UDP is much simpler than TCP. It interfaces to IP (or other protocols) without the bother of flow control or error-recovery mechanisms, acting simply as a sender and receiver of datagrams.

UDP is connectionless, TCP is based on connections.

The UDP message header is much simpler than TCP's. It is shown in Figure 10. Padding may be added to the datagram to ensure that the message is a multiple of 16 bits.

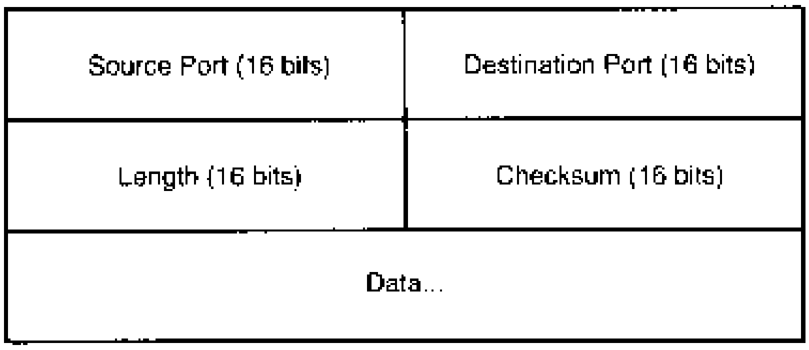


Figure 10 - The UDP header

The fields are as follows:

Source port: An optional field with the port number. If a port number is not specified, the field is set to 0.
Destination port: The port on the destination machine.
Length: The length of the datagram, including header and data.
Checksum: A 16-bit one's complement of the one's complement sum of the datagram, including a pseudoheader similar to that of TCP.

The UDP checksum field is optional; but if it isn't used, there is no checksum applied to the data segment because IP's checksum applies only to the IP header. If the checksum is not used, the field should be set to 0.

As well, there is an enormous amount of literature available on this topic. In particular the following books are worthwhile which can be bought from McGills here in Melbourne.

A good reference web site is as follows:-
http://www.lantronix.com/htmfiles/mrktg/catalog/et.htm

TCP/IP illustrated Volumes 1, 2 & 3, by W. Richard Stevens (Addison-Wesley), ISBN 0-201-63346-9

TCP/IP Network Administration, by Craig Hunt (O'Reilly & Associates), ISBN 0-937175-82-X

Teach Yourself TCP/IP, by Timothy Parker, Ph.D. (SAMS Publishing), ISBN 0-672-30549-6

Please E-mail me and tell me if you liked my TCP information, or even if you have any contributing sites on similar info that I can include here.

Click here to go back to my Technical Page

To go back to my previous page, please press or home to go back to my Home Page, or even to goto my next page.

Enter your email address to receive email when this page is updated.

Your Internet email address:

Press Here to Register

This page has been accessed times.

Last revised: Friday, 04 July 1997