THE TRANSPORT LAYER

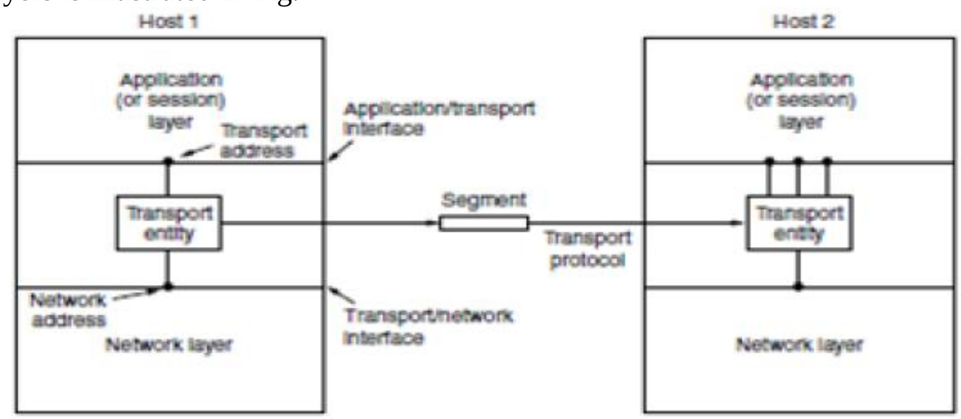
UNIT IV

The Transport Layer: The Transport Service, Elements of Transport Protocols, Congestion Control,The internet transport protocols: UDP, TCP, Performance problems in computer networks, Networkperformance measurement.

THE TRANSPORT SERVICE

Services Provided to the Upper Layers

The ultimate goal of the transport layer is to provide efficient, reliable, and cost-effective data transmission service to its users, normally processes in the application layer. To achieve this, the transport layer makes use of the services provided by the network layer. The software and/or hardware 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 library package bound into network applications, in a separate user process, or even on the network interface card. The first two options are most common on the Internet. The (logical) relationship of the network, transport, and application layers is illustrated in Fig.



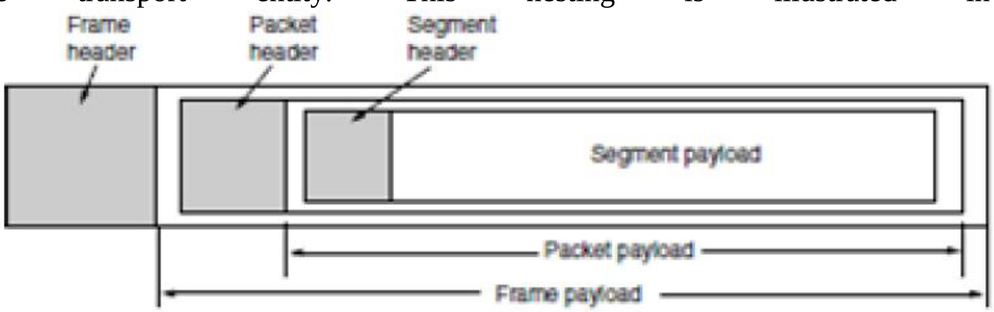
there are also two types of transport service. The connection-oriented transport service is similar to the connection-oriented network service in many ways. In both cases, connections have three phases: establishment, data transfer, and release. Addressing and flow control are also similar in both layers. Furthermore, the connectionless transport service is also very similar to the connectionless network service. However, note that it can be difficult to provide a connectionless transport service on top of a connection-oriented network service, since it is inefficient to set up a connection to send a single packet and then tear it down immediately afterwards. The obvious question is this: if the transport layer service is so similar to the network layer service, why are there two distinct layers? Why is one layer not Problems occur, that's what? The users have no real control over the network layer, so they cannot solve the problem of poor service by using better routers or putting more error handling in the data link layer because they don't own the routers. The only possibility is to put on top of the network layer another layer that improves the quality of the service. If, in a connectionless network, packets are lost or mangled, the transport entity can detect the problem and compensate for it by using retransmissions. If, in a connection-oriented network, a transport entity is informed halfway through a long transmission that its network connection has been abruptly terminated, with no indication of what has happened to the data currently in transit, it can set up a new network connection to the remote transport entity. Using this new network connection, it can send a query to its peer asking which data arrived and which did not, and knowing where it was, pick up from where it left off.

Transport Service Primitives

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. In this section, we will first examine a simple (hypothetical) transport service and its interface to see the bare essentials. In the following section, we will look at a real example. 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. Real networks can lose packets, so the network service is generally unreliable.

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.	Request a release of the connection

A quick note on terminology is now in order. For lack of a better term, we will use the term **segment** for messages sent from transport entity to transport entity.TCP, UDP and other Internet protocols use this term. Some older protocols used the ungainly name **TPDU** (**Transport Protocol Data Unit**). That term is not used much anymore now but you may see it in older papers and books the network entity similarly processes the packet header and then passes the contents of the packet payload up to the transport entity. This nesting is illustrated in Fig. 6-3



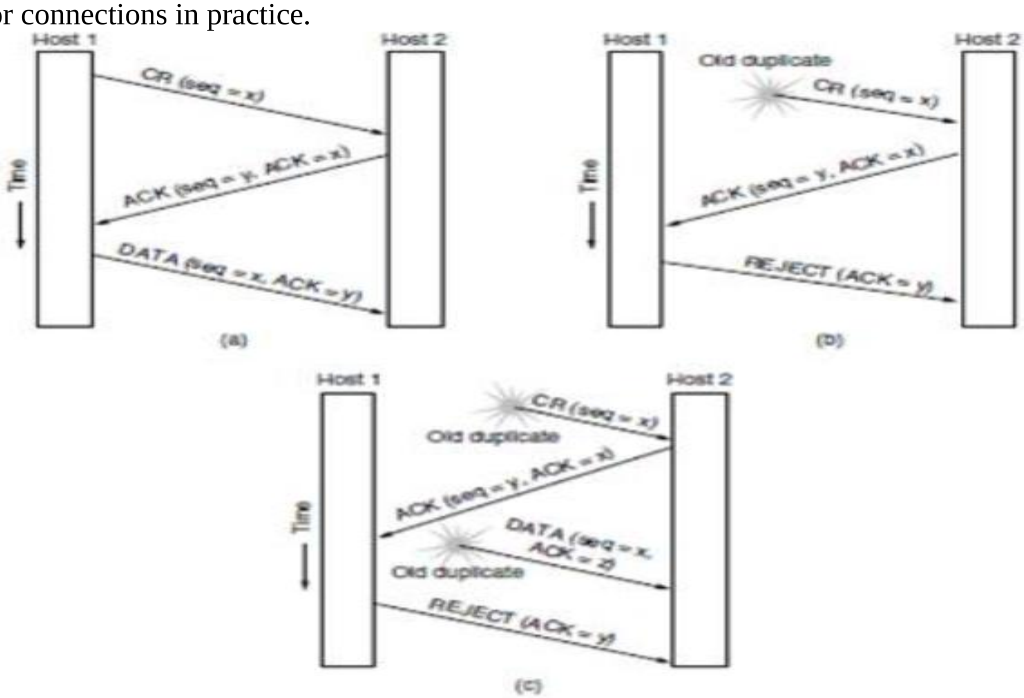
Connection Establishment

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 behaviour 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. That is, they may be delayed in the network and pop out much later, when the sender thought that they had been lost. The worst possible nightmare is as follows. A user establishes a connection with a bank, sends messages telling the bank to transfer a large amount of money to the account of a not-entirely-trustworthy person. Unfortunately, the packets decide to take the scenic route to the destination and go off exploring a remote corner of the network. The sender then times out and sends them all again. This time the packets take the shortest route and are delivered quickly so the sender releases 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. Time stamping each packet.
The first technique includes any method that prevents packets from looping, combined with some way of bounding delay including congestion over the (now known) longest possible path. It is difficult, given that internets may range from a single city to international in scope. The second method consists of having the hop count initialized to some appropriate value and decremented each time the packet is forwarded. The network protocol simply discards any packet whose hop counter becomes zero. The third method requires each packet to bear the time it was created, with the routers agreeing to discard any packet older than some agreed-upon time. This latter method requires the router clocks to be synchronized, which itself is a nontrivial task, and in practice a hop counter is a close enough approximation to age.
TCP uses this three-way handshake to establish connections. Within a connection, a timestamp is used to extend the 32-bit sequence number so that it will not wrap within the maximum packet lifetime, even for gigabit-per-second connections. This mechanism is a fix to TCP that was needed as it was used on faster and faster links. It is described in RFC 1323 and called **PAWS** (**Protection Against Wrapped Sequence numbers**). Across connections, for the initial sequence numbers and before PAWS can come into play, TCP originally use the clock-based scheme just described. However, this turned out to have security vulnerability. The clock made it easy for an attacker to predict the next initial sequence number and send packets that tricked the three-way handshake and established a forged connection. To close this hole, pseudorandom initial sequence numbers are used for connections in practice.

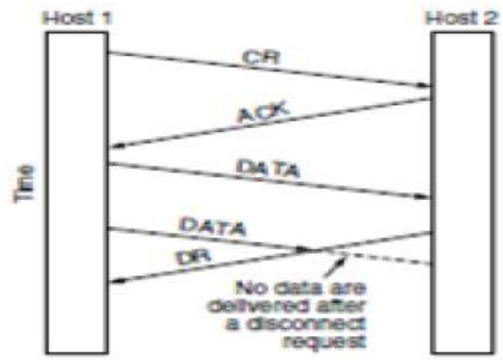


the initial sequence numbers not repeat for an interval even though they appear random to an observer. Otherwise, delayed duplicates can wreak havoc.

Connection Release

Releasing a connection is easier than establishing one. Nevertheless, there are more pitfalls than one might expect here. As we mentioned earlier, 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.

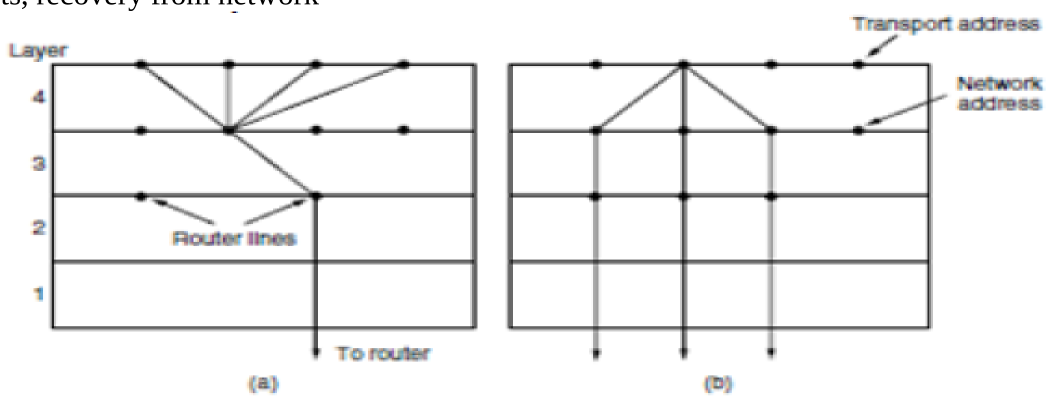
Asymmetric release is abrupt and may result in data loss. Consider the scenario of Fig. After the connection is established, host 1 sends a segment that arrives properly at host 2. Then host 1 sends another segment. Unfortunately, host 2 issues a DISCONNECT before the second segment arrives. The result is that the connection is released and data are lost.



Abrupt disconnection with loss of data.

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



(a) Multiplexing. (b) Inverse multiplexing.

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. To illustrate the difficulty, let us assume that one host, the client, is sending a long file to another host, the file server, using a simple Stop-and-wait protocol. The transport layer on the server just passes the incoming segments to the transport user, one by one. Partway through the transmission, the server crashes. When it comes back up, its tables are reinitialized, so it no longer knows precisely where it was. 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.

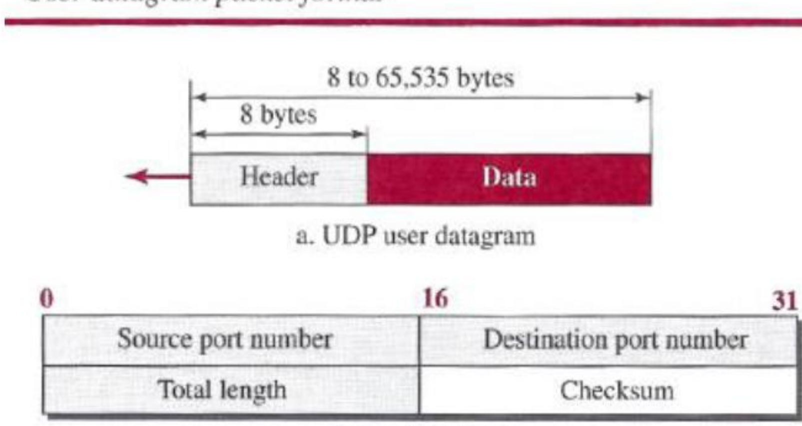
UDP Protocol

UDP provides connectionless, unreliable, datagram service. Connectionless service means that there is no logical connection between the two ends exchanging messages. Each message is an independent entity encapsulated in a datagram.

UDP does not see any relation (connection) between consequent datagram coming from the same source and going to the same destination.
UDP has an advantage: it is message-oriented. It gives boundaries to the messages exchanged. An application program may be designed to use UDP if it is sending small messages and the simplicity and speed is more important for the application than reliability.

User Datagram
UDP packets, called *user datagram*, have a fixed-size header of 8 bytes made of four fields, each of 2 bytes (16 bits).
. The 16 bits can define a total length of 0 to 65,535 bytes. However, the total length needs to be less because a UDP user datagram is stored in an IP datagram with the total length of 65,535 bytes. The last field can carry the optional checksum

User datagram packet format

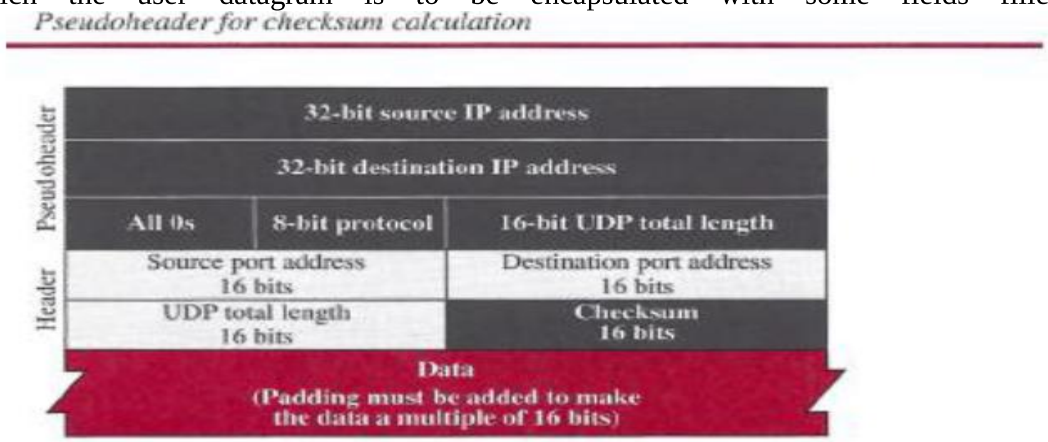


a. UDP user datagram

b. Header format

UDP Services
Process-to-Process Communication
UDP provides process-to-process communication using **socket addresses,** a combination of **IP** addresses and port numbers.
Connectionless Services
As mentioned previously, UDP provides a *connection less service*. This means that each user datagram sent by UDP is an independent datagram. There is no relationship between the different user data grams even if they are coming from the same source process and going to the same destination program.
Flow Control
UDP is a very simple protocol. There is no *flow control*, and hence no window mechanism. The receiver may overflow with incoming messages.
Error Control
There is no *error control* mechanism in UDP except for the checksum. This means that the sender does not know if a message has been lost or duplicated.
Checksum
UDP checksum calculation includes three sections: a pseudo header, the UDP header, and the data coming from the application layer. The *pseudo header* is the part of the header of the IP packet in

which the user datagram is to be encapsulated with some fields filled with 0s



UDP Applications

UDP Features

Connectionless Service

As we mentioned previously,

- UDP is a connectionless protocol. Each UDP packet is independent from other packets sent by the same application program. This feature can be considered as an advantage or disadvanta8e depending on the application requirements.
- UDP does not provide error control; it provides an unreliable service. Most applications expect reliable service from a transport-layer protocol. Although a reliable service is desirable.

Typical Applications

The following shows some typical applications that can benefit more from the services of UDP

- UDP is suitable for a process that requires simple request-response communication with little concern for flow and error control
- UDP is suitable for a process with internal flow- and error-control mechanisms. For example, the Trivial File Transfer Protocol (TFIP)
- UDP is a suitable transport protocol for multicasting. Multicasting capability is embedded in the UDP software
- UDP is used for management processes such as SNMP
- UDP is used for some route updating protocols such as Routing Information Protocol (RIP)
- UDP is normally used for interactive real-time applications that cannot tolerate uneven delay between sections of a received message

TRANSMISSION CONTROL PROTOCOL

Transmission Control Protocol (TCP) is a connection-oriented, reliable protocol. TCP explicitly defines connection establishment, data transfer, and connection teardown phases to provide a connection-oriented service.

TCP Services

Process-to-Process Communication

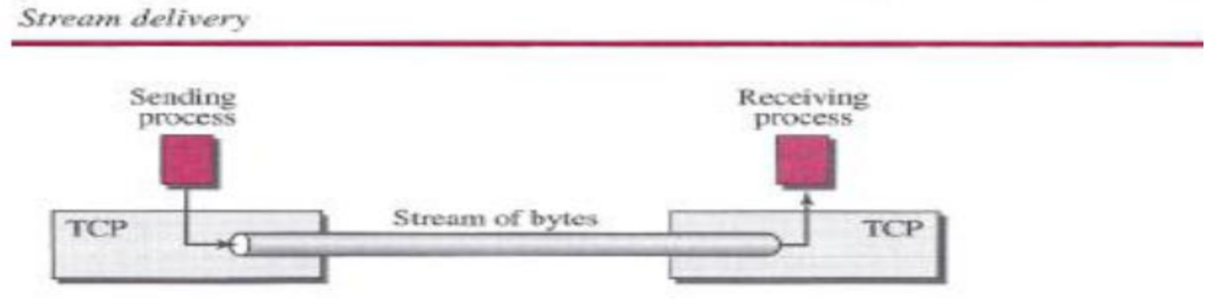
As with UDP, TCP provides process-to-process communication using port numbers. We have already given some of the port numbers used by TCP.

Stream Delivery Service

In UDP, a process sends messages with predefined boundaries to UDP for delivery. UDP adds its own header to each of these messages and delivers it to IP for transmission.

TCP, on the other hand, allows the sending process to deliver data as a stream of bytes and allows the receiving process to obtain data as a stream of bytes.

TCP creates an environment in which the two processes seem to be connected by an imaginary "tube" that carries their bytes across the Internet.



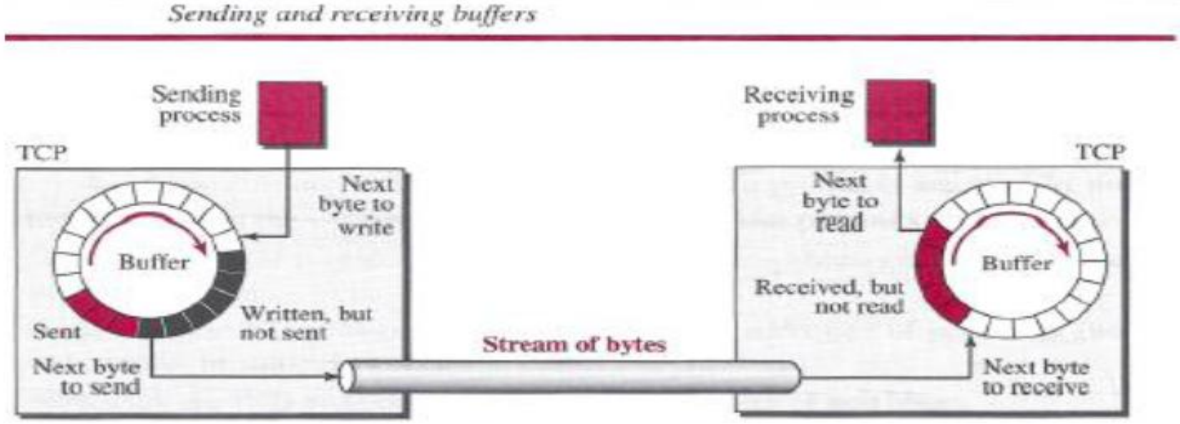
Sending and Receiving Buffers

Because the sending and the receiving processes may not necessarily write or read data at the same rate, TCP needs buffers for storage.

There are two buffers, the sending buffer and the receiving buffer, one for each direction.

- At the sender, the buffer has three types of chambers. The white section contains empty chambers that can be filled by the sending process (producer).
- The colored area holds bytes that have been sent but not yet acknowledged.
- The TCP sender keeps these bytes in the buffer until it receives an acknowledgment. The shaded area contains bytes to be sent by the sending TCP.
- The operation of the buffer at the receiver is simpler. The circular buffer is divided into two areas (shown as white and colored).
- The white area contains empty chambers to be filled by bytes received from the network.
- The colored sections contain received bytes that can be read by the receiving process. When a byte is read by the receiving process, the chamber is recycled and added to the pool of empty chambers.

Sending and receiving buffers

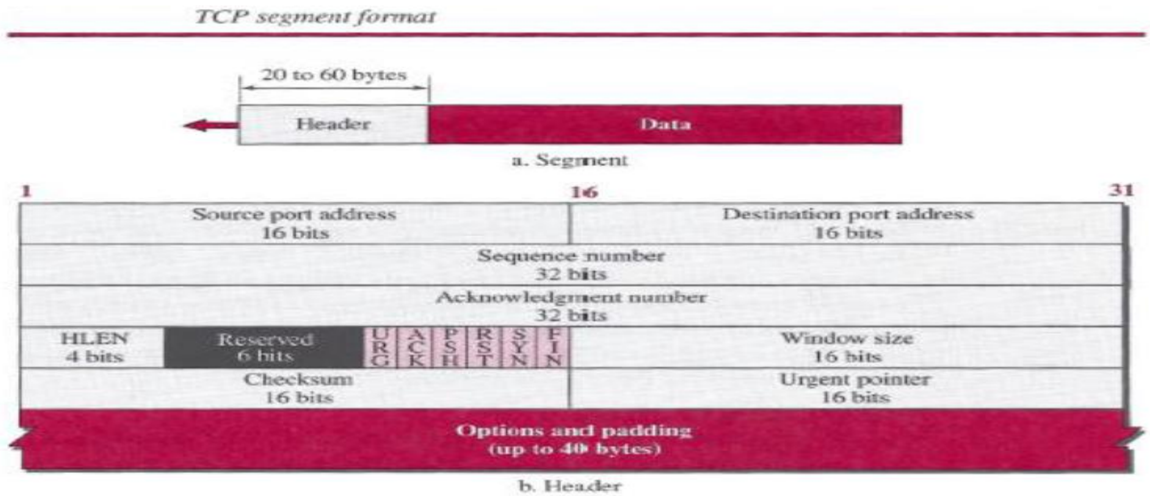


Segments

- Although buffering handles the disparity between the speed of the producing and consuming Processes, we need one more step before we can send data.
- The network layer, as a service provider for TCP, needs to send data in packets, not as a stream of bytes. At the transport layer, TCP groups a number of bytes together into a packet called a *segment.*
- The segments are encapsulated in an IP datagram and transmitted. This entire operation is transparent to the receiving process.

Format

The segment consists of a header of 20 to 60 bytes, followed by data from the application program.The header is 20 bytes if there are no options and up to 60 bytes if it contains options.

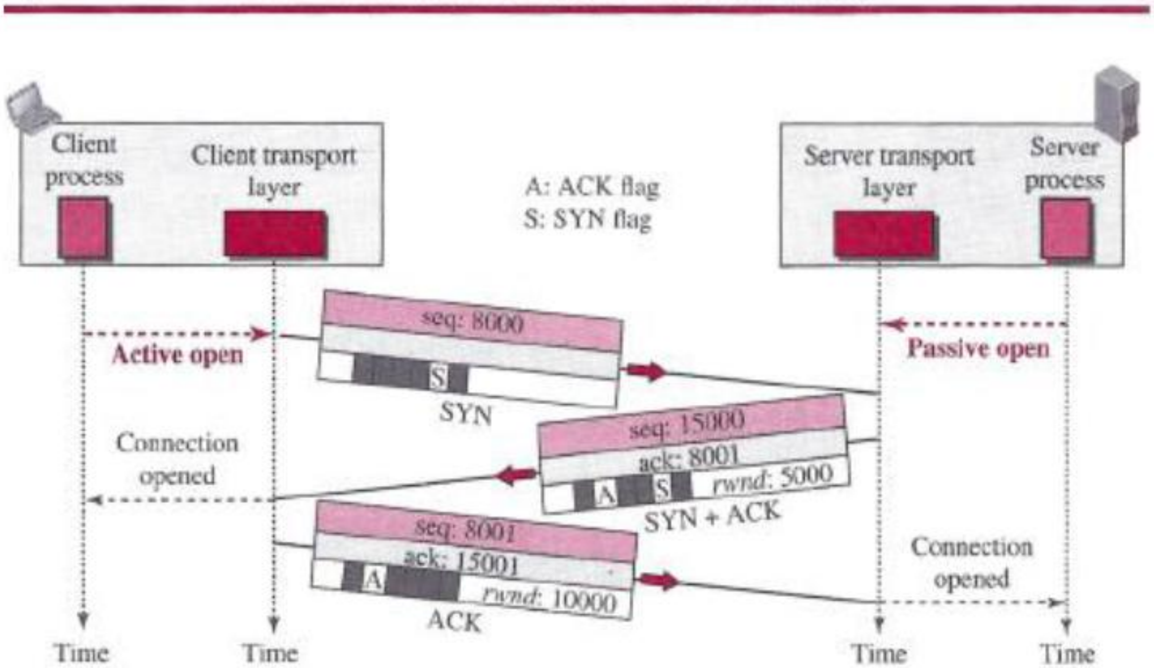


Source port address This is a 16-bit field that defines the port number of the application program in the host that is sending the segment.
Destination port address This is a 16-bit field that defines the port number of the application program in the host that is receiving the segment.
Sequence number This 32-bit field defines the number assigned to the first byte of data contained in this segment.
Acknowledgment number This 32-bit field defines the byte number that the receiver of the segment is expecting to receive from the other party.
Header length This 4-bit field indicates the number of 4-byte words in the TCP header. The length of the header can be between 20 and 60 bytes.

A TCP Connection

- TCP is connection-oriented. a connection-oriented transport protocol establishes a logical path between the source and destination.
- All of the segments belonging to a message are then sent over this logical path.
- TCP operates at a higher level. TCP uses the services of IP to deliver individual segments to the receiver, but it controls the connection itself.
- In TCP, connection-oriented transmission requires three phases: connection establishment, data transfer, and connection termination.

Connection establishment using three-way handshaking



Connection Establishment
TCP transmits data in full-duplex mode. When two TCPs in two machines are connected, they are able to send segments to each other simultaneously.

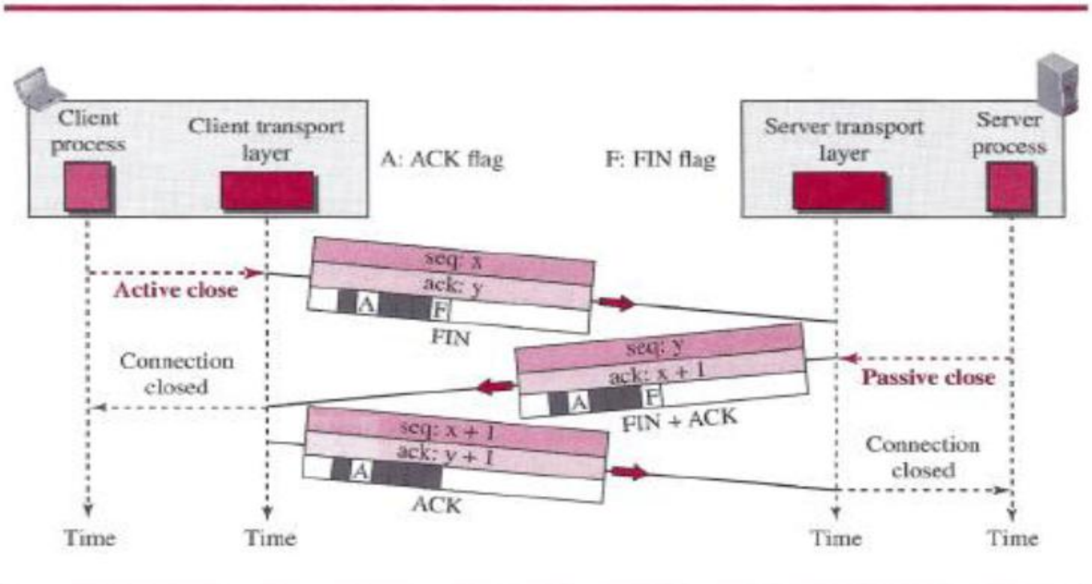
Three- Way Handshaking
The connection establishment in TCP is called *three-way handshaking.* an application program, called the *client*, wants to make a connection with another application program, called the *server*,

using TCP as the transport-layer protocol The process starts with the server. The server program tells its TCP that it is ready to accept a connection. This request is called a *passive open*.
Although the server TCP is ready to accept a connection from any machine in the world, it cannot make the connection itself.
The client program issues a request for an *active open.* A client that wishes to connect to an open server tells its TCP to connect to a particular server.

- A SYN segment cannot carry data, but it consumes one sequence number.
- A SYN + ACK segment cannot carry data, but it does consume one sequence number.

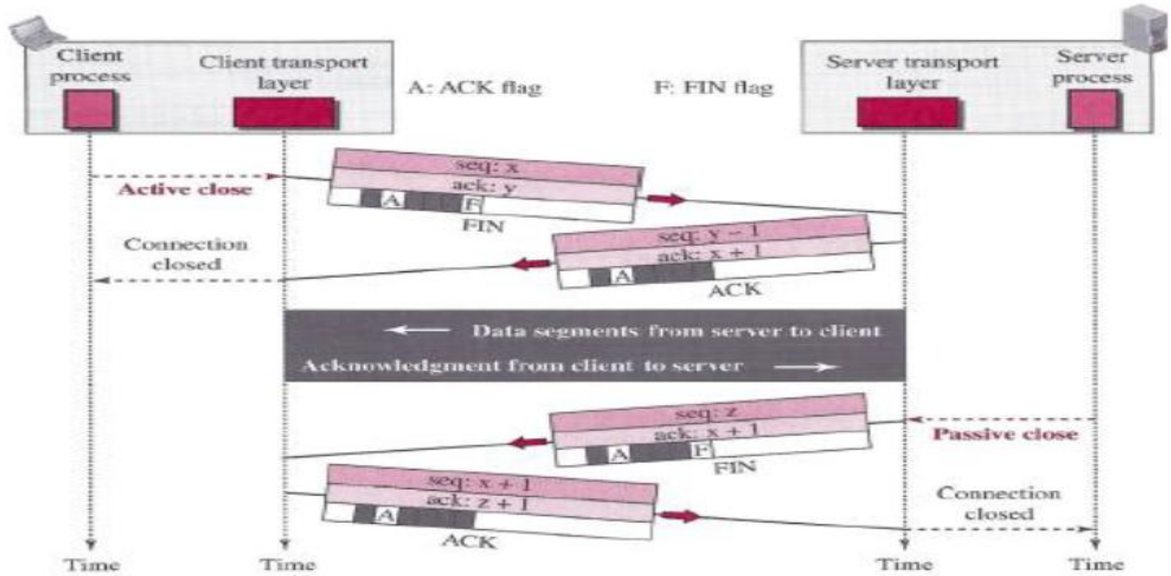
- An ACK segment, if carrying no data, consumes no sequence number.

Connection termination using three-way handshaking



The FIN + ACK segment consumes only one sequence number if it does not carry data.

Half-close



Client process
Client transport layer
A: ACK flag
F: FIN flag
Server transport layer
Server process
seq: x
ack: y
Active close
FIN
Connection closed
seq: y - 1
ack: x + 1
ACK
Data segments from server to client
Acknowledgment from client to server
seq: z
ack: x + 1
Passive close
FIN
seq: x + 1
ack: z + 1
Connection closed
ACK
Time
Time
Time
Time