

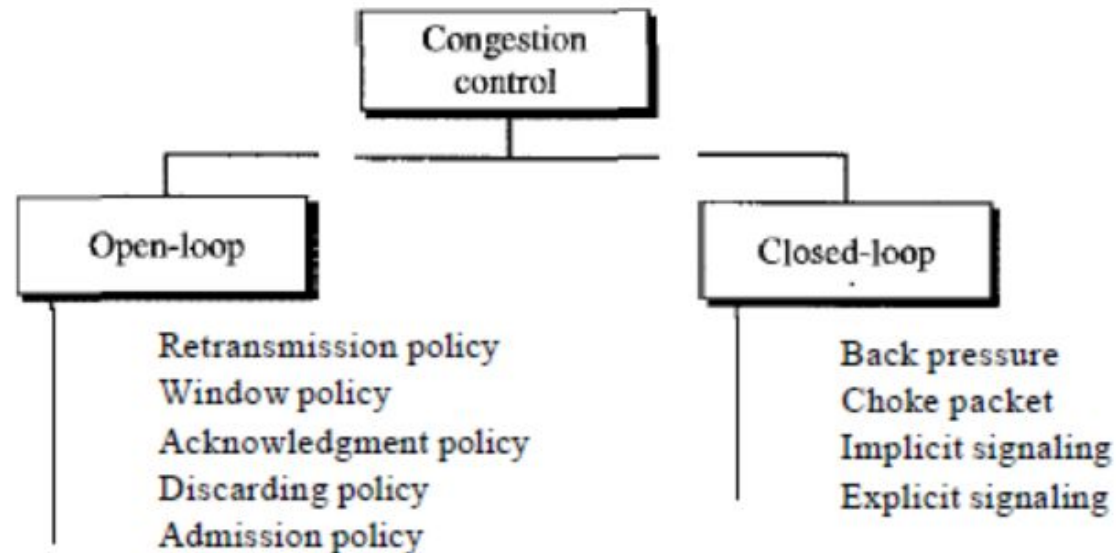
Module 6

Congestion control

Congestion Control

- An important issue in a packet-switched network is **congestion**. Congestion in a network may occur if the **load** on the network-the number of packets sent to the network-is greater than the *capacity* of the network-the number of packets a network can handle.
- **Congestion control** refers to the mechanisms and techniques to control the congestion and keep the load below the capacity.
- Congestion control involves two factors that measure the performance of a network: *delay* and *throughput*.
- **Congestion control** refers to techniques and mechanisms that can either prevent congestion, before it happens, or remove congestion, after it has happened. In general, we can divide congestion control mechanisms into two broad categories: open-loop congestion control (prevention) and closed-loop congestion control (removal).

Congestion control categories

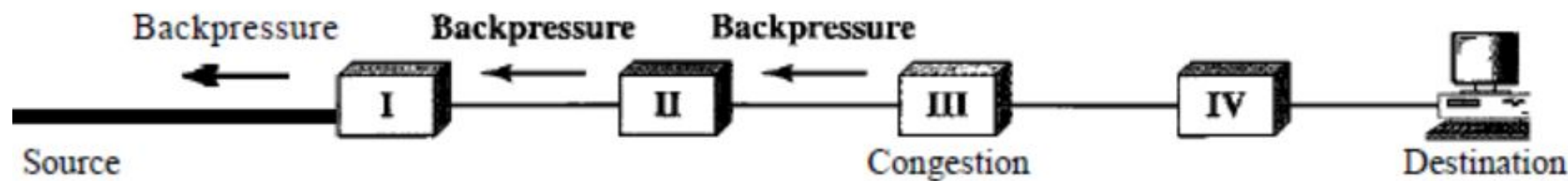


Open-Loop Congestion Control

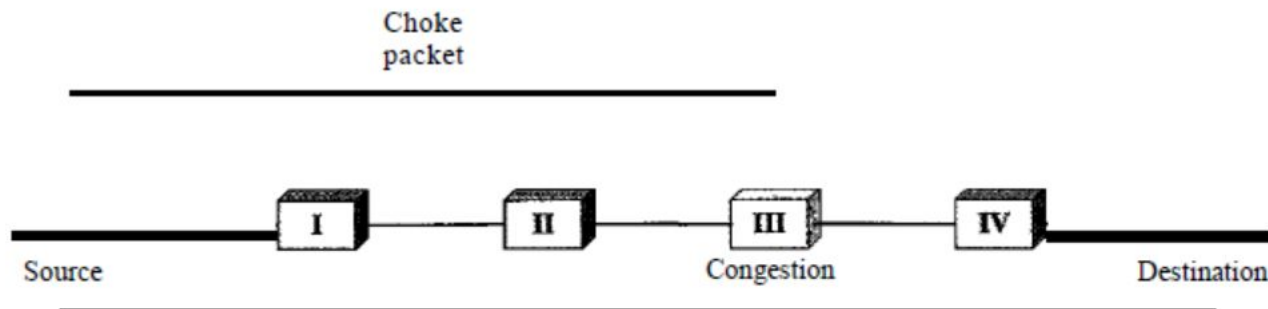
- In open-loop congestion control, policies are applied to prevent congestion before it happens. In these mechanisms, congestion control is handled by either the source or the destination.
- List of policies that can prevent congestion:
 - *Retransmission Policy*: If the sender feels that a sent packet is lost or corrupted, the packet needs to be retransmitted. Retransmission in general may increase congestion in the network. However, a good retransmission policy can prevent congestion. The retransmission policy and the retransmission timers must be designed to optimize efficiency and at the same time prevent congestion. For example, the retransmission policy used by TCP (explained later) is designed to prevent or alleviate congestion.
 - *Window Policy*: The type of window at the sender may also affect congestion. The Selective Repeat window is better than the Go-Back-N window for congestion control. In the *Go-Back-N* window, when the timer for a packet times out, several packets may be resent, although some may have arrived safe and sound at the receiver. This duplication may make the congestion worse. The Selective Repeat window, on the other hand, tries to send the specific packets that have been lost or corrupted.
 - *Acknowledgment Policy*: The acknowledgment policy imposed by the receiver may also affect congestion. If the receiver does not acknowledge every packet it receives, it may slow down the sender and help prevent congestion. A receiver may send an acknowledgment only if it has a packet to be sent or a special timer expires. A receiver may decide to acknowledge only N packets at a time. We need to know that the acknowledgments are also part of the load in a network. Sending fewer acknowledgments means imposing less load on the network.
 - *Discarding Policy*: A good discarding policy by the routers may prevent congestion and at the same time may not harm the integrity of the transmission. For example, in audio transmission, if the policy is to discard less sensitive packets when congestion is likely to happen, the quality of sound is still preserved and congestion is prevented or alleviated.
 - *Admission Policy*: An admission policy, which is a quality-of-service mechanism, can also prevent congestion in virtual-circuit networks. Switches in a flow first check the resource requirement of a flow before admitting it to the network. A router can deny establishing a virtual circuit connection if there is congestion in the network or if there is a possibility of future congestion.

Closed-Loop Congestion Control

- Closed-loop congestion control mechanisms try to alleviate congestion after it happens.
- *Backpressure*: The technique of *backpressure* refers to a congestion control mechanism in which a congested node stops receiving data from the immediate upstream node or nodes. This may cause the upstream node or nodes to become congested, and they, in turn, reject data from their upstream nodes or nodes. And so on. Backpressure is a node-to-node congestion control that starts with a node and propagates, in the opposite direction of data flow, to the source. The backpressure technique can be applied only to virtual circuit networks, in which each node knows the upstream node from which a flow of data is coming.



- *Choke Packet*: A choke packet is a packet sent by a node to the source to inform it of congestion. Note the difference between the backpressure and choke packet methods. In backpressure, the warning is from one node to its upstream node, although the warning may eventually reach the source station. In the choke packet method, the warning is from the router, which has encountered congestion, to the source station directly. The intermediate nodes through which the packet has traveled are not warned. We have seen an example of this type of control in ICMP.



- *Implicit Signaling*: In implicit signaling, there is no communication between the congested node or nodes and the source. The source guesses that there is a congestion somewhere in the network from other symptoms. For example, when a source sends several packets and there is no acknowledgment for a while, one assumption is that the network is congested. The delay in receiving an acknowledgment is interpreted as congestion in the network; the source should slow down.
- *Explicit Signaling*: The node that experiences congestion can explicitly send a signal to the source or destination. The explicit signaling method, however, is different from the choke packet method. In the choke packet method, a separate packet is used for this purpose; in the explicit signaling method, the signal is included in the packets that carry data. Explicit signaling, as we will see in Frame Relay congestion control.
 - Backward Signaling A bit can be set in a packet moving in the direction opposite to the congestion. This bit can warn the source that there is congestion and that it needs to slow down to avoid the discarding of packets.
 - Forward Signaling A bit can be set in a packet moving in the direction of the congestion. This bit can warn the destination that there is congestion. The receiver in this case can use policies, such as slowing down the acknowledgments, to alleviate the congestion.

UDP (User Datagram Protocol)

- The User Datagram Protocol (UDP) is called a connectionless, unreliable transport protocol.
- UDP is a very simple protocol using a minimum of overhead. If a process wants to send a small message and does not care much about reliability, it can use UDP. Sending a small message by using UDP takes much less interaction between the sender and receiver than using TCP or SCTP.

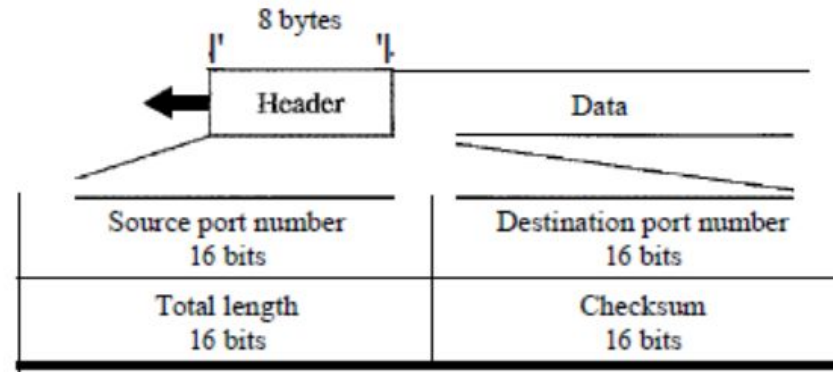
Table 23.1 *Well-known ports used with UDP*

<i>Port</i>	<i>Protocol</i>	<i>Description</i>
7	Echo	Echoes a received datagram back to the sender
9	Discard	Discards any datagram that is received
11	Users	Active users

<i>Port</i>	<i>Protocol</i>	<i>Description</i>
13	Daytime	Returns the date and the time
17	Quote	Returns a quote of the day
19	Chargen	Returns a string of characters
53	Nameserver	Domain Name Service
67	BOOTPs	Server port to download bootstrap information
68	BOOTPc	Client port to download bootstrap information
69	TFTP	Trivial File Transfer Protocol
III	RPC	Remote Procedure Call
123	NTP	Network Time Protocol
161	SNMP	Simple Network Management Protocol
162	SNMP	Simple Network Management Protocol (trap)

UDP format

User datagram format



- UDP packets, called user datagrams, have a fixed-size header of 8 bytes.
- **Source port number:** This is the port number used by the process running on the source host. It is 16 bits long, which means that the port number can range from 0 to 65,535. If the source host is the client (a client sending a request), the port number, in most cases, is an ephemeral port number requested by the process and chosen by the UDP software running on the source host. If the source host is the server (a server sending a response), the port number, in most cases, is a well-known port number.
- **Destination port number:** This is the port number used by the process running on the destination host. It is also 16 bits long. If the destination host is the server (a client sending a request), the port number, in most cases, is a well-known port number. If the destination host is the client (a server sending a response), the port number, in most cases, is an ephemeral port number.
- **Length:** This is a 16-bit field that defines the total length of the user datagram, header plus data. The 16 bits can define a total length of 0 to 65,535 bytes. $\text{UDP length} = \text{IP length} - \text{IP header's length}$
- **Checksum:** This field is used to detect errors over the entire user datagram (header plus data).

State transition diagram for TCP

- [Tcp State Transition Diagram - Digital Transmission \(fishercom.xyz\)](http://fishercom.xyz)

TCP: 3-way handshake

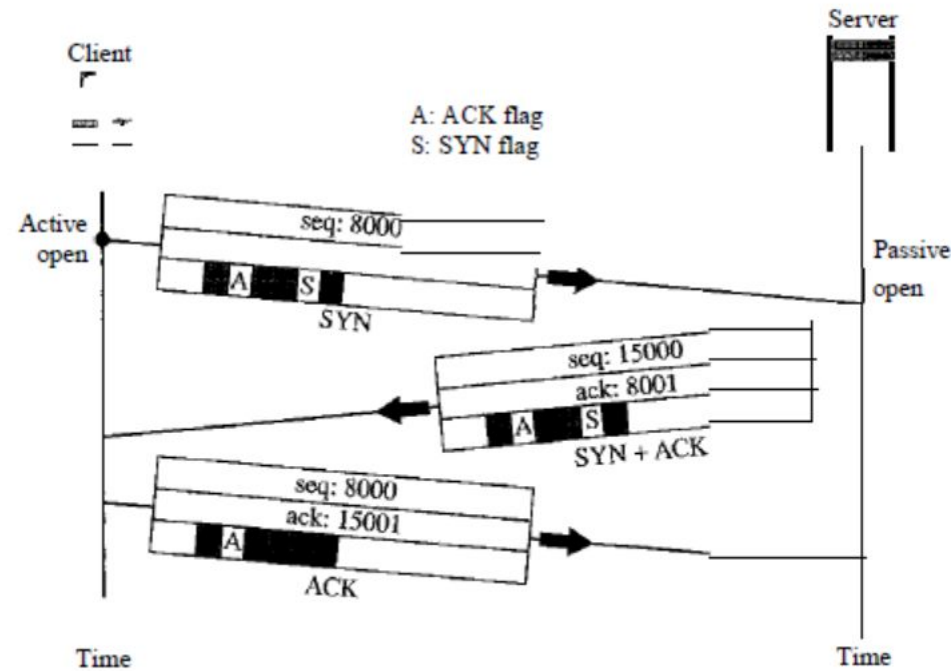
Three-Way Handshaking The connection establishment in TCP is called three-way handshaking. In our example, an application program, called the client, wants to make a connection with another application program, called the server, using TCP as the transport layer protocol.

The process starts with the server. The server program tells its TCP that it is ready to accept a connection. This is called a request for a *passive open*. Although the server TCP is ready to accept any 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 that it needs to be connected to that particular server. TCP can now start the three-way handshaking process as shown in Figure 23.18. To show the process, we use two time lines: one at each site. Each segment has values for all its header fields and perhaps for some of its option fields, too. However, we show only the few fields necessary to understand each phase. We show the sequence number,

3 steps

Figure 23.18 Connection establishment using three-way handshaking



Step 1: The client sends the first segment, a **SYN segment**, in which only the SYN flag is set. This segment is for synchronization of sequence numbers. It consumes one sequence number. When the data transfer starts, the sequence number is incremented by 1. We can say that the SYN segment carries no real data.

Step 2: The server sends the second segment, a **SYN + ACK segment**, with 2 flag bits set: SYN and ACK. This segment has a dual purpose. It is a SYN segment for communication in the other direction and serves as the acknowledgment for the SYN segment. It consumes one sequence number.

Step 3: The client sends the third segment. This is just an **ACK segment**. It acknowledges the receipt of the second segment with the ACK flag and acknowledgment number field. Note that the sequence number in this segment is the same as the one in the SYN segment;

HTTP

The Hypertext Transfer Protocol (HTTP) is a protocol used mainly to access data on the World Wide Web. HTTP functions as a combination of FTP and SMTP. It is similar to FTP because it transfers files and uses the services of TCP. However, it is much simpler than FTP because it uses only one TCP connection. There is no separate control connection; only data are transferred between the client and the server.

HTTP is like SMTP because the data transferred between the client and the server look like SMTP messages. In addition, the format of the messages is controlled by MIME-like headers. Unlike SMTP, the HTTP messages are not destined to be read by humans; they are read and interpreted by the HTTP server and HTTP client (browser). SMTP messages are stored and forwarded, but HTTP messages are delivered immediately. The commands from the client to the server are embedded in a request message. The contents of the requested file or other information are embedded in a response message. HTTP uses the services of TCP on well-known port 80.

HTTP Transaction

Figure 27.12 illustrates the HTTP transaction between the client and server. Although HTTP uses the services of TCP, HTTP itself is a stateless protocol. The client initializes the transaction by sending a request message. The server replies by sending a response.

Messages

The formats of the request and response messages are similar; both are shown in Figure 27.13. A request message consists of a request line, a header, and sometimes a body. A response message consists of a status line, a header, and sometimes a body.

Figure 27.12 HTTP transaction

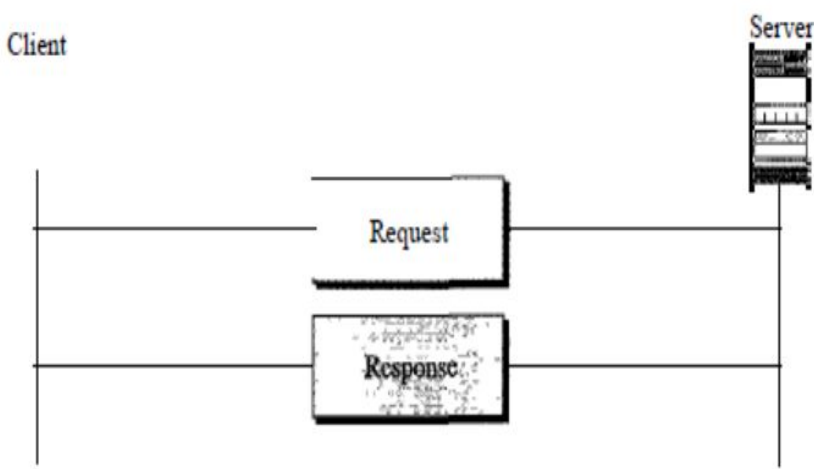
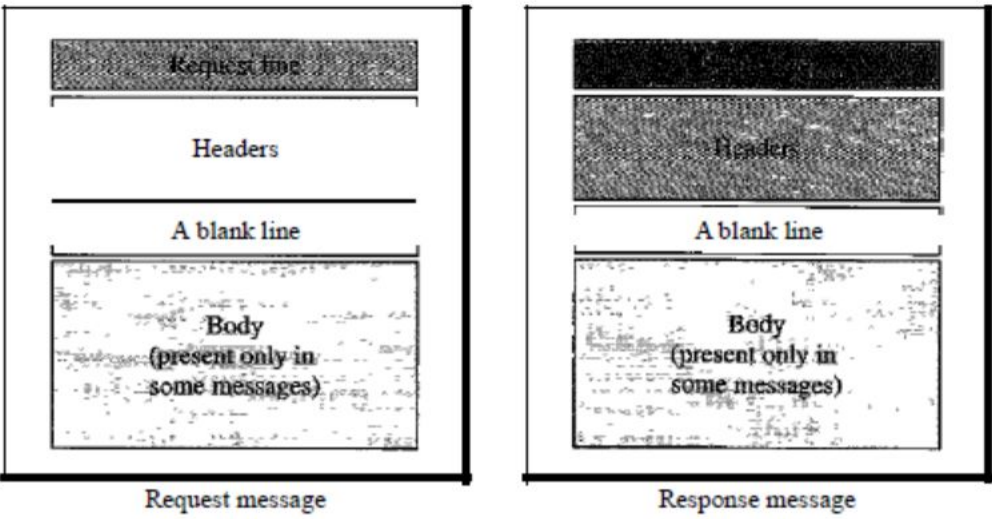
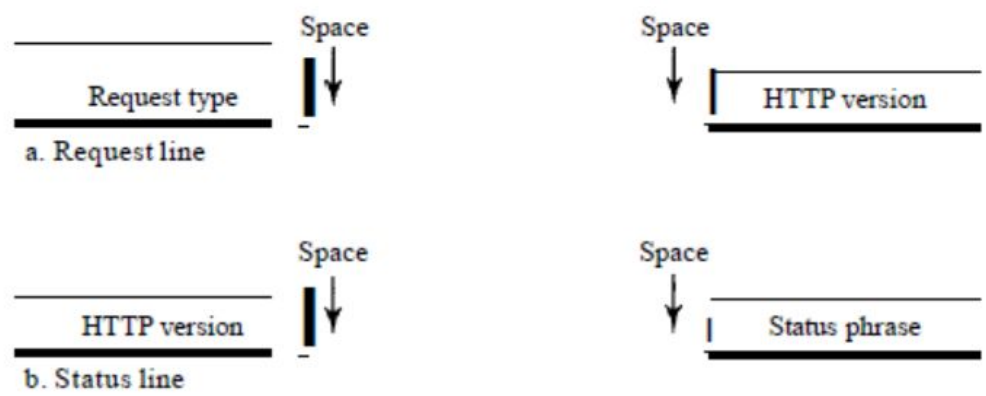


Figure 27.13 Request and response messages



Request and Status Lines The first line in a request message is called a request line; the first line in the response message is called the status line. There is one common field, as shown in Figure 27.14.

Figure 27.14 Request and status lines



Request type. This field is used in the request message. In version 1.1 of HTTP, several request types are defined. The request type is categorized into *methods* as defined in Table 27.1.

Table 27.1 *Methods*

<i>Method</i>	<i>Action</i>
GET	Requests a document from the server
HEAD	Requests information about a document but not the document itself
POST	Sends some information from the client to the server
PUT	Sends a document from the server to the client
TRACE	Echoes the incoming request
CONNECT	Reserved
OPTION	Inquires about available options