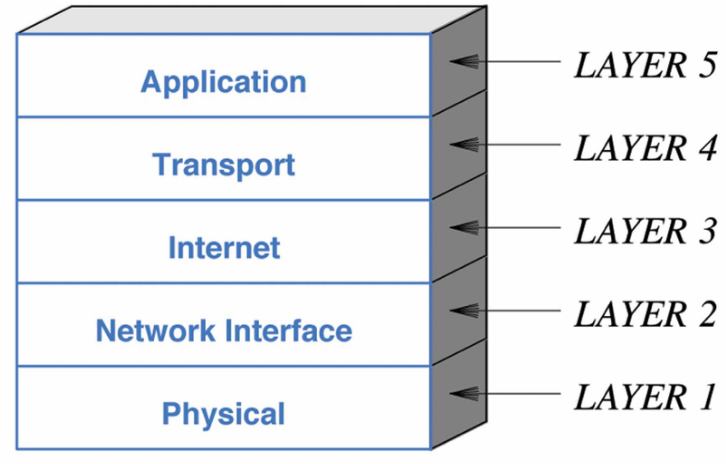
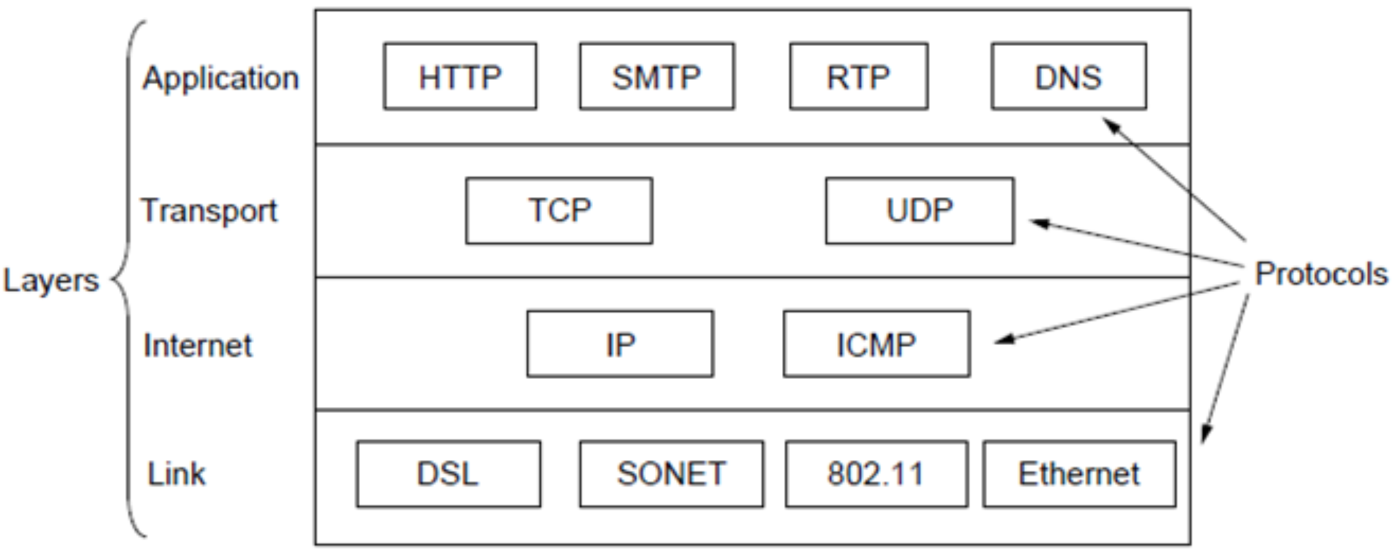
10. Transport Layer

10.1. TCP/IP layers



- recall the 5-layer model above
- the *network interface* layer is often called the *link* layer
- we use the generic term *packet* for each block of data transmitted
- recall that each layer adds its own header, so nature of "packet" varies
- so in fact the following terms are usually used for "packets" at each layer
    - *frames* at the link layer
    - *datagrams* at the internet layer
    - *segments* at the transport layer
- we focus on the transport layer in this section

10.2. TCP/IP layers with some protocols



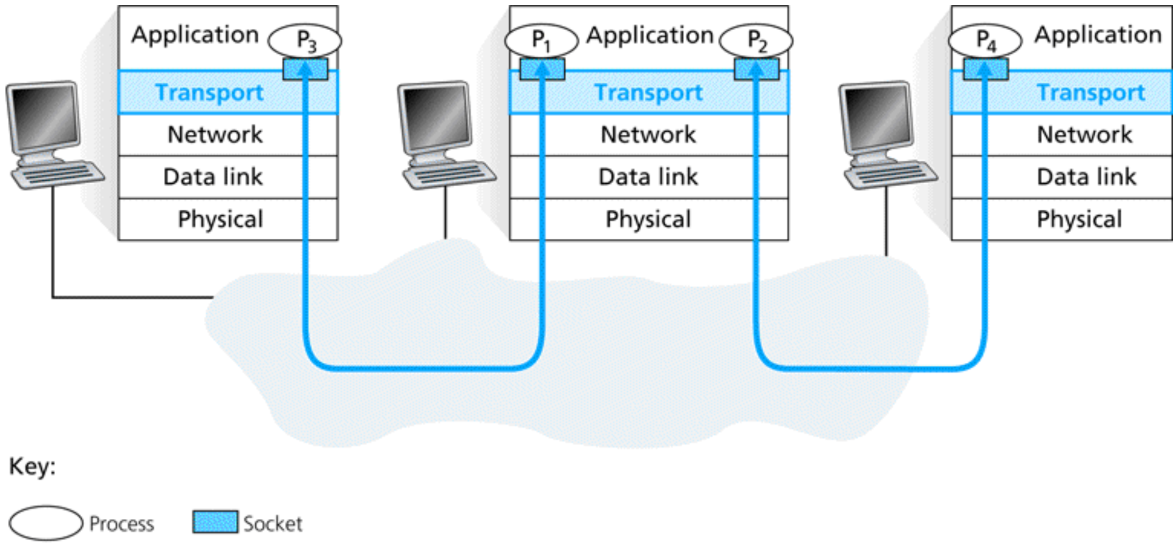
- we focus on the *UDP* and *TCP* in this section

10.3. Transport Protocols

- Internet Protocol (IP) provides a packet delivery service across an internet
- however, IP cannot distinguish between multiple *processes* (applications) running on the same computer
- fields in the IP datagram header identify only *computers*
- a protocol that allows an application to serve as an *end-point* of communication is known as a *transport protocol* or an *end-to-end protocol*
- the TCP/IP protocol suite provides two transport protocols:
    - the *User Datagram Protocol* (UDP)
    - the *Transmission Control Protocol* (TCP)

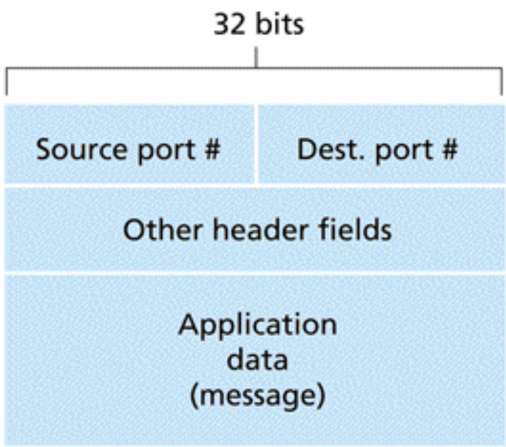
10.4. Multiplexing and Demultiplexing

- a *socket* is the interface through which a process (application) communicates with the transport layer
- each process can potentially use many sockets
- the transport layer in a receiving machine receives a sequence of segments from its network layer
- delivering segments to the correct socket is called *demultiplexing*
- assembling segments with the necessary information and passing them to the network layer is called *multiplexing*
- multiplexing and demultiplexing are need whenever a communications channel is *shared*



10.5. Endpoint Identification

- sockets must have unique identifiers
- each segment must include header fields identifying the socket
- these header fields are the *source port number field* and the *destination port number field*
- each port number is a 16-bit number: 0 to 65535



10.6. Well-known port numbers

- port numbers below 1024 are called *well-known ports* and are reserved for standard services, e.g.:

Port number	Application protocol	Description	Transport protocol
21	FTP	File transfer	TCP
23	Telnet	Remote login	TCP
25	SMTP	E-mail	TCP
53	DNS	Domain Name System	UDP
79	Finger	Lookup information about a user	TCP
80	HTTP	World wide web	TCP
110	POP-3	Remote e-mail access	TCP
119	NNTP	Usenet news	TCP
161	SNMP	Simple Network Management Protocol	UDP

- these pre-defined port numbers are registered with the Internet Assigned Numbers Authority (IANA)

10.7. The User Datagram Protocol

- UDP is less complex and easier to understand than TCP
- the characteristics of UDP are given below:
  - *end-to-end*: UDP can identify a specific process running on a computer
  - *connectionless*: UDP follows the connectionless paradigm (see below)
  - *message-oriented*: processes using UDP send and receive individual messages called *segments* or *user datagrams*
  - *best-effort*: UDP offers the same best-effort delivery as IP
  - *arbitrary interaction*: UDP allows processes to send to and receive from as many other processes as it chooses
  - *operating system independent*: UDP identifies processes independently of the local operating system

10.8. The Connectionless Paradigm

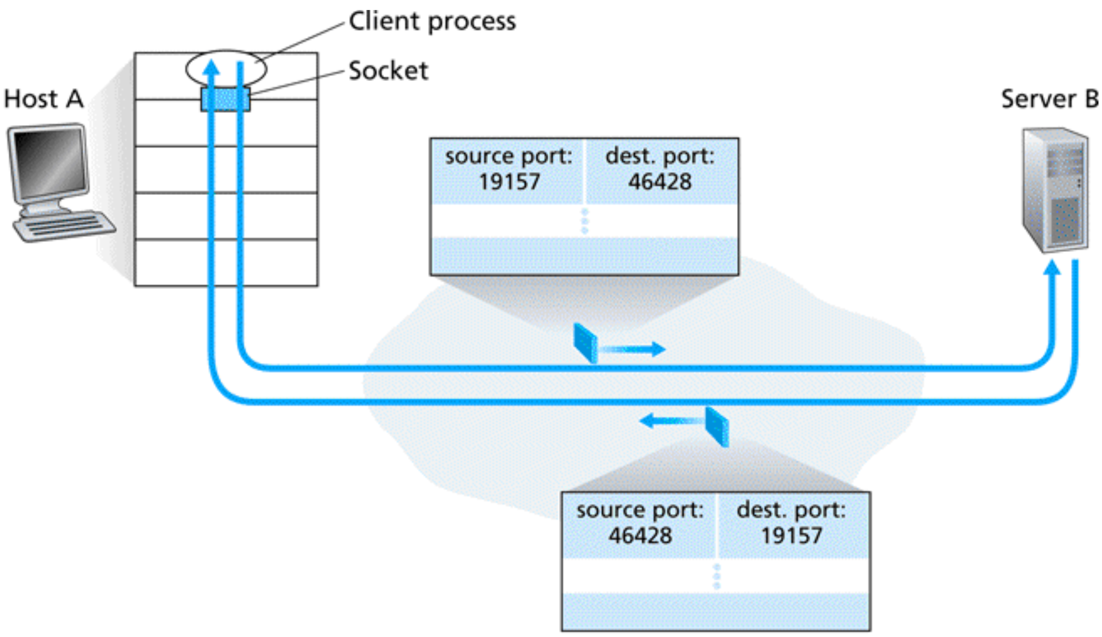
- UDP uses a *connectionless* communication setup
- a process using UDP does not need to establish a connection before sending data (unlike TCP)
- when two processes stop communicating there are no, additional, control messages (unlike TCP)
- communication consists only of the data segments themselves

10.9. Message-Oriented Interface

- UDP provides a *message-oriented* interface
- each message is sent as a single UDP segment
- however, this also means that the maximum size of a UDP message depends on the maximum size of an IP datagram
- allowing large UDP segments can cause problems
- sending large segments can result in IP fragmentation (see later)
- UDP offers the same best-effort delivery as IP
- this means that segments can be lost, duplicated, or corrupted in transit
- this is why UDP is suitable for applications such as voice or video that can tolerate delivery errors

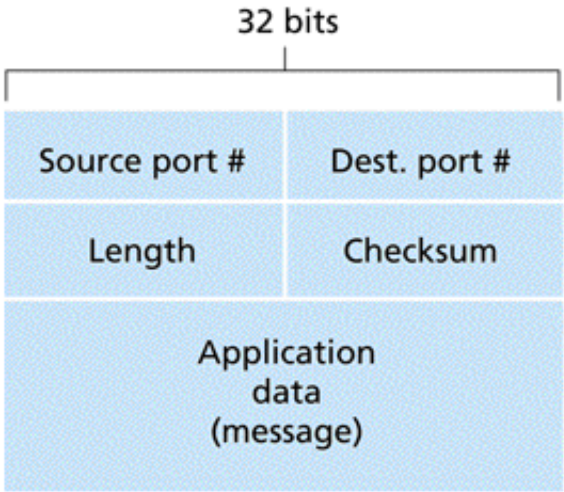
10.10. Connectionless Multiplexing and Demultiplexing

- say a process on Host A, with port number 19157, wants to send data to a process with UDP port 46428 on Host B
- transport layer in Host A creates a segment containing source port, destination port, and data
- passes it to the network layer in Host A
- transport layer in Host B examines destination port number and delivers segment to socket identified by port 46428
- note: a UDP socket is fully identified by a two-tuple consisting of
    - a destination IP address
    - a destination port number
- source port number from Host A is used at Host B as "return address":



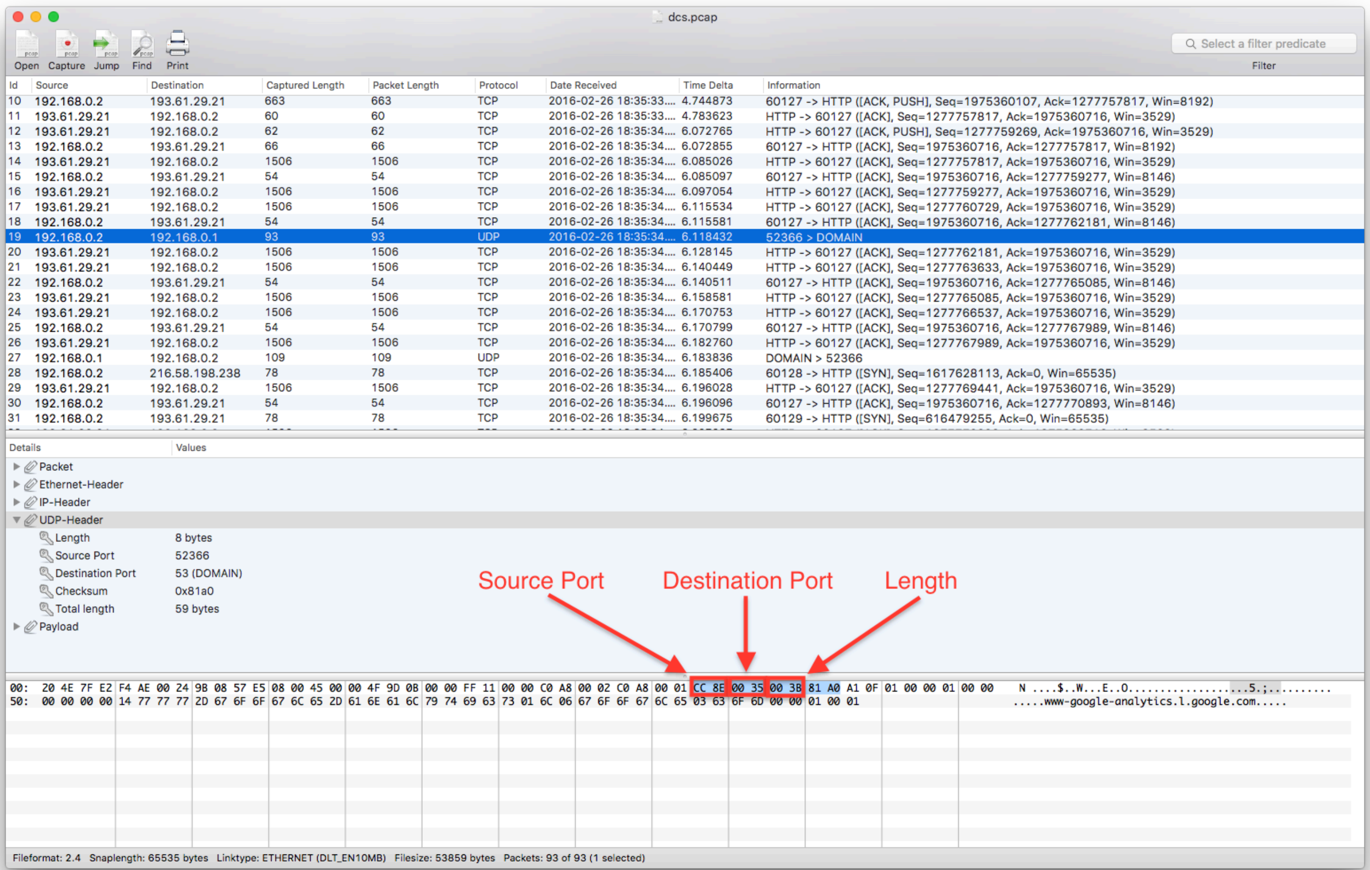
10.11. UDP Segment Structure

- UDP segment is sometimes called a *user datagram*
- it consists of an 8-byte header followed by the application data (sometimes called *payload*), as shown below

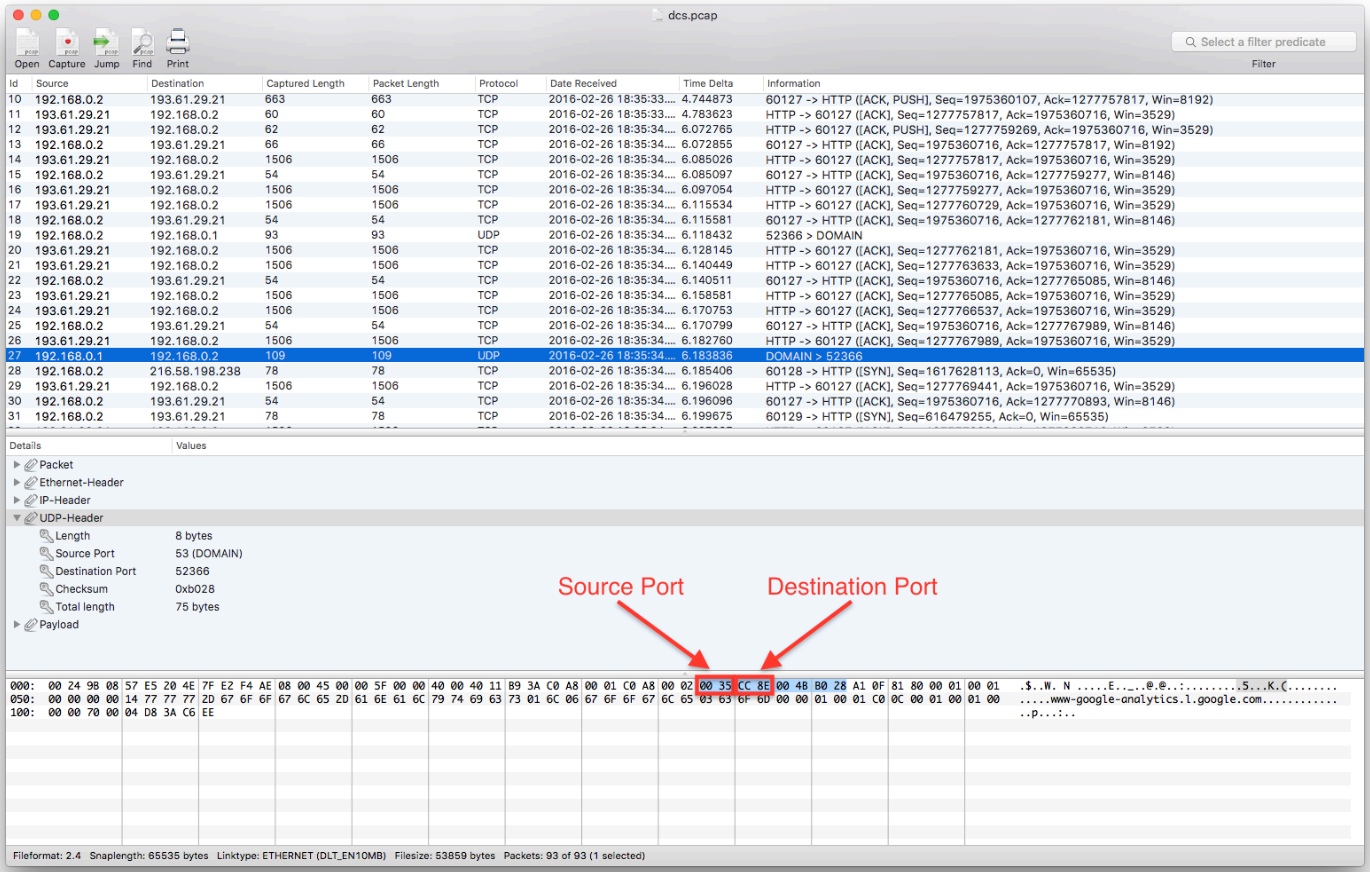


- *Source port #* identifies the *UDP process* which sent the segment
- *Dest port #* identifies the *UDP process* which will handle the application data
- *Length* specifies the length of the segment, including the header, in bytes
- *Checksum* is optional (see below)

10.12. UDP Header Example (DNS Request)



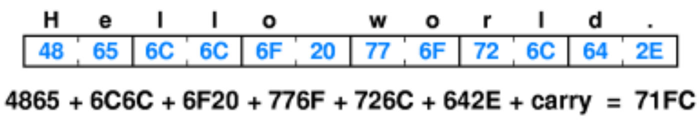
10.13. UDP Header Example (DNS Response)



10.14. Internet Checksum

- both UDP and TCP use a 16-bit *Checksum* field
- the sender can choose to compute a checksum or set the field to zero
- the receiver only verifies the checksum if the value is non-zero
- note that the checksum is computed using ones-complement arithmetic, so a computed zero value is stored as all-ones

10.15. Checksum Example



- to compute the checksum, the sender treats the data as a sequence of binary integers and computes their sum, as illustrated above
- each pair of characters is treated as a 16-bit integer
- if the sum overflows 16 bits, the carry bits are added to the total
- the advantage of such checksums is their size and ease of computation

- addition requires very little computation and the cost of sending an additional 16-bits is negligible

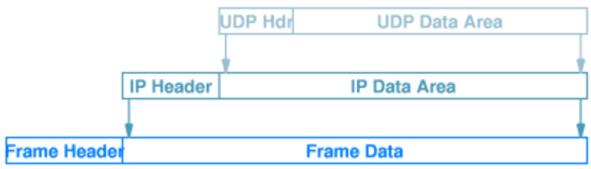
10.16. Example of Checksum Failure

Data Item In Binary	Checksum Value	Data Item In Binary	Checksum Value
0001	1	0011	3
0010	2	0000	0
0011	3	0001	1
0001	1	0011	3
totals	7		7

- checksums do not detect all common errors, as illustrated above
- a transmission error has inverted the second bit in each of the four data items, yet the checksums are identical

10.17. UDP Encapsulation

- recall that each layer in the protocol stack adds its own header
- each UDP segment is encapsulated in a network-layer (IP) datagram
- each IP datagram is encapsulated in a link-layer frame



10.18. Protocols Using UDP

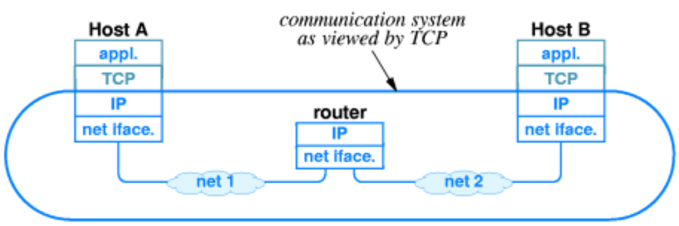
- UDP is especially useful in client-server situations, when a client sends a short request to the server and expects a short response
- if either the request or response is lost, the client times out and tries again
- if all is well, only two packets are required
- an example of an application that uses UDP in this way is the *Domain Name System* (DNS)

10.19. Transmission Control Protocol (TCP)

- the *Transmission Control Protocol* (TCP) is the transport level protocol that provides *reliability* in the TCP/IP protocol suite
- from an application program's perspective, TCP offers:
    - *connection-oriented*: an application requests a connection, and then uses it for data transfer
    - *point-to-point communication*: each TCP connection has exactly two end points
    - *reliability*: TCP guarantees that the data sent across the connection will be delivered exactly as sent, without missing or duplicate data
    - *full-duplex connection*: a TCP connection allows data to flow in both directions at any time
    - *stream interface*: TCP allows an application to send a continuous stream of bytes across the connection
    - *reliable startup*: TCP requires that two applications must agree to the new connection before it is established
    - *graceful shutdown*: TCP guarantees to deliver all the data reliably before closing the connection

10.20. End-To-End Service

- TCP uses IP to carry messages, known as *segments*
- each TCP segment is encapsulated in an IP datagram and sent across the Internet
- TCP treats IP as a packet communication system:



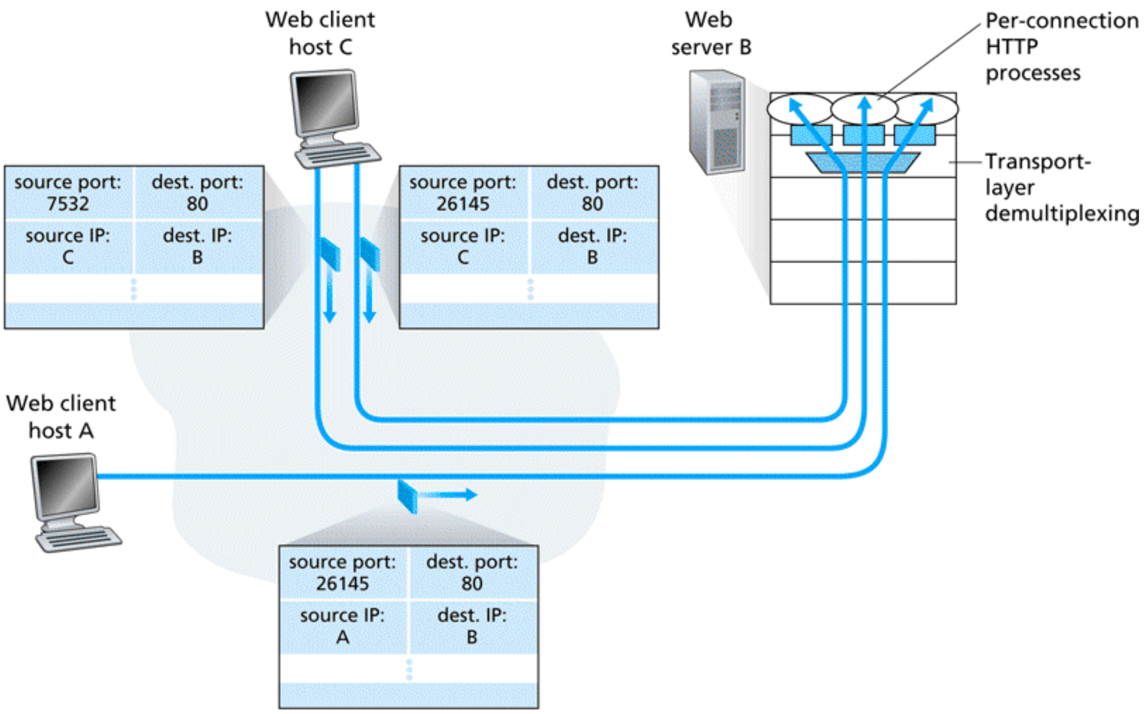
- as illustrated, TCP software is required at both ends of the virtual connection, but not on intermediate routers
- from TCP's point of view, the entire Internet is a communication system capable of accepting and delivering messages without changing their contents

10.21. Connection-Oriented Multiplexing and Demultiplexing

- each TCP connection has exactly two end-points
- this means that two arriving TCP segments with different source IP addresses or source port numbers will be directed to two *different* sockets, *even if they have the same destination port number*
- so a TCP socket is identified by a four-tuple:
  (source IP address, source port #, destination IP address, destination port #)
- recall UDP uses only (destination IP address, destination port #)

10.22. Multiplexing and Demultiplexing Example

- an example where clients A and C both communicate with B on port 80:



10.23. Reliable Data Transfer

- TCP is a *reliable* data transfer protocol
- implemented on top of an *unreliable* network layer (IP)
- some problems:
  - bits in a packet may be *corrupted*
  - packets can be *lost* by the underlying network
- some solutions:
  - *acknowledgements* (ACKs) can be used to indicate packet received correctly
  - a *countdown timer* can be used to detect packet loss
  - packet *retransmission* can be used for lost packets

10.24. Simple Reliable Data Transfer

- a simple reliable data transfer protocol might
  - send a packet
  - wait until it is sure the receiver has received it correctly
- such a protocol is known as a *stop-and-wait* protocol
- performance of such a protocol on the Internet would be poor

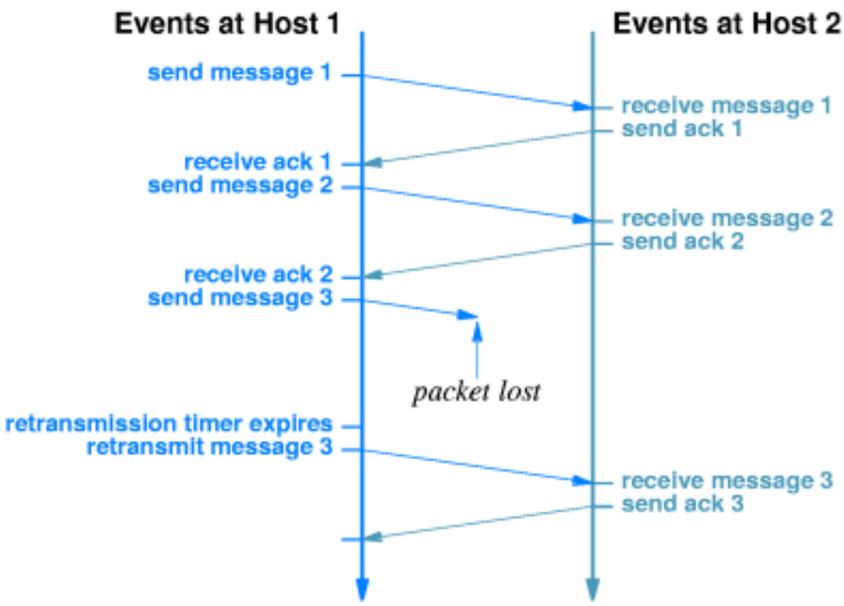
10.25. Pipelined Reliable Data Transfer

- a *pipelined* protocol allows for multiple data packets to be sent while waiting for acknowledgements
- this results in better network utilisation
- sender and receiver now need buffers to hold multiple packets
- packets need *sequence numbers* in order to identify them
- an acknowledgement needs to refer to corresponding sequence number
- retransmission can give rise to duplicate packets
- sequence numbers in packets allow receiver to detect duplicates

10.26. Packet Loss and Retransmission

- TCP copes with the loss of packets using *retransmission*
- when TCP data arrives, an *acknowledgement* is sent back to the sender
- when TCP data is sent, a timer is started
- if the timer expires before an acknowledgement arrives, TCP retransmits the data

10.27. Packet Loss and Retransmission - Example



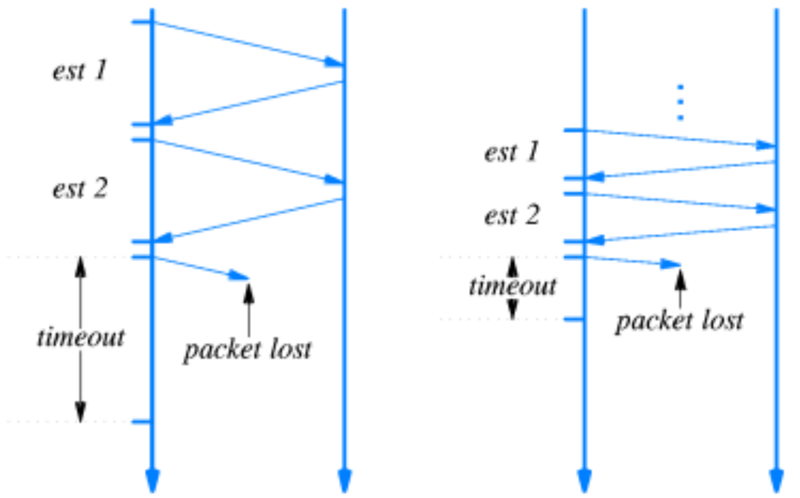
- host on the left is sending data; host on the right is receiving it
- TCP must be ready to retransmit any packet that is lost

- how long should TCP wait?
- the TCP software does not know whether it is using
  - a local area network (acknowledgements within a few milliseconds) or
  - a long-distance satellite connection (acknowledgements within a few seconds)

10.28. Adaptive Retransmission

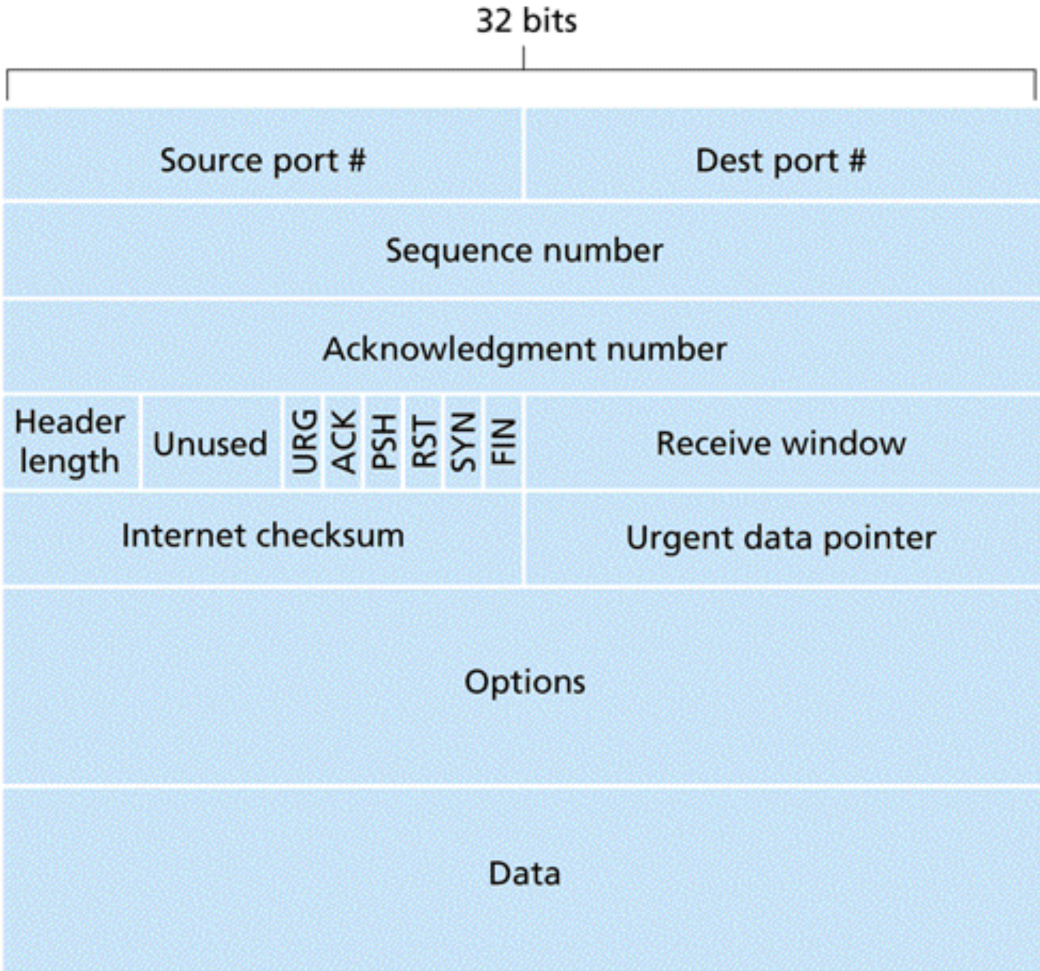
- TCP estimates the *round-trip delay* for each active connection
- for each connection, TCP generates a sequence of round-trip estimates and produces a weighted average (mean)
- it also maintains an estimate of the variance
- it then uses a linear combination of the estimated mean and variance as the value of the timeout

10.29. Adaptive Retransmission - Example



- the connection on the left above has a relatively long round-trip delay
- the connection on the right above has a shorter round-trip delay
- the goal is to wait long enough to decide that a packet was lost, without waiting longer than necessary
- when delays start to vary, TCP adjusts the timeout accordingly

10.30. TCP Segment Structure

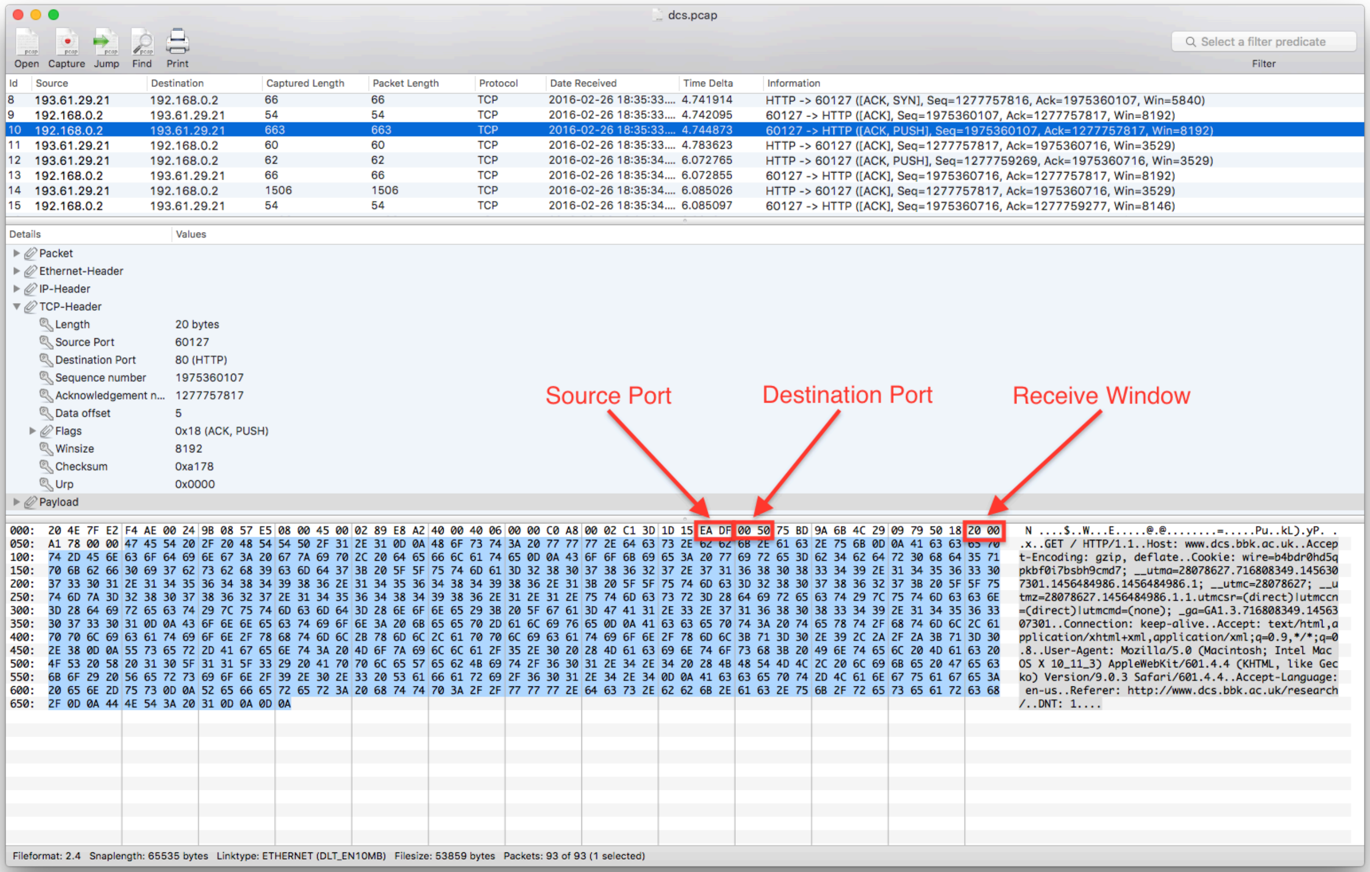


- *Source port #*, *Dest port #* and *Internet checksum* are as for UDP
- *Sequence number* (32 bits) and *Acknowledgement number* (32 bits) are used to implement reliable transfer (see below)
- *Header length* (4 bits) is the header length (including possible options) in 32-bit words
- the *flag field* contains 6 1-bit flags (see below)
- *Receive window* identifies how much buffer space is available for incoming data (used for *flow control*)

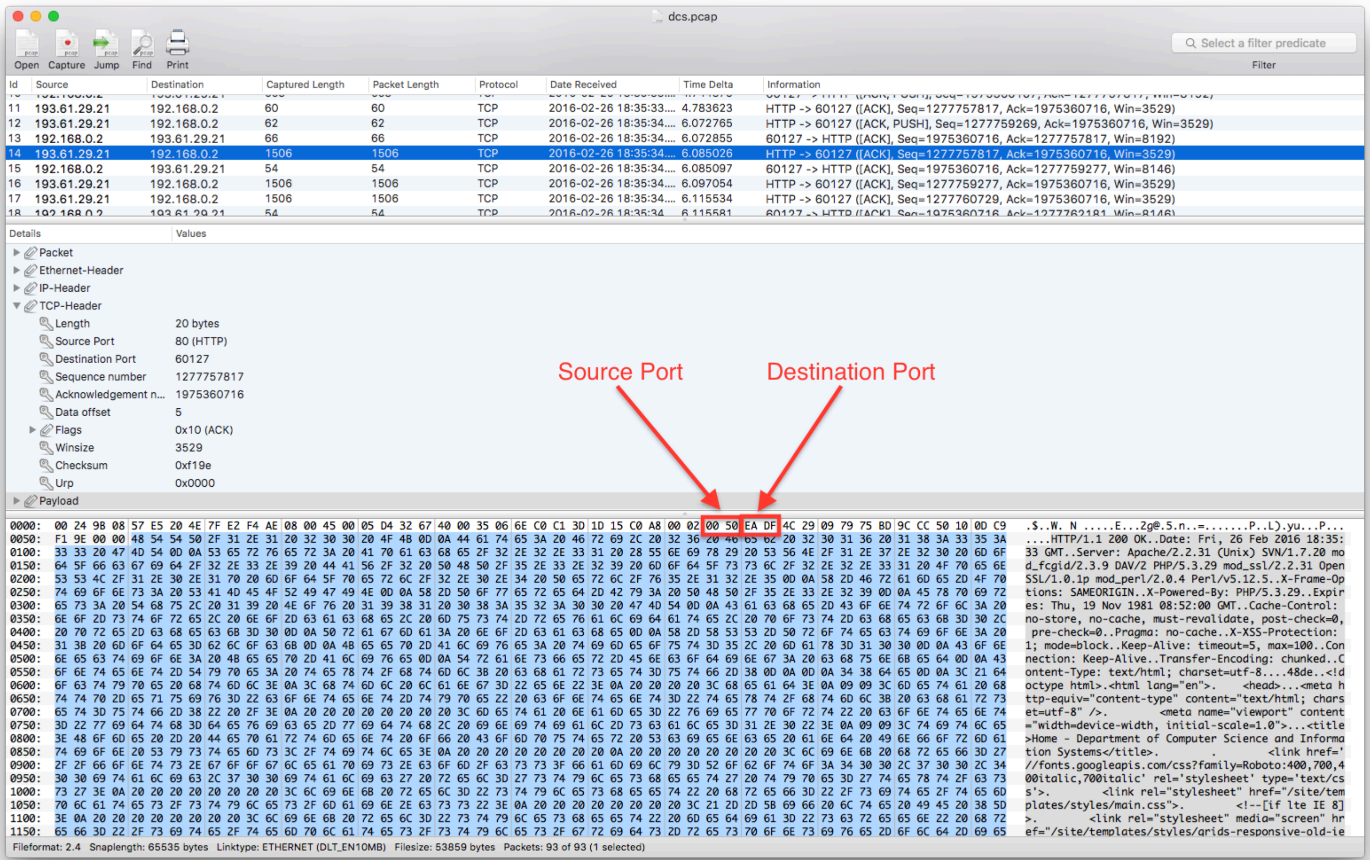
10.31. TCP Flags

- *URG* flag indicates that the sender has marked some data as urgent
- in this case, the *Urgent data pointer* contains an offset into the TCP data stream marking the last byte of urgent data
- *ACK* flag indicates that the acknowledgement number field is valid (i.e. the segment is an acknowledgement)
- *PSH* flag indicates that should be delivered immediately (PUSHed) and not buffered
- *RST* flag is used to reset a connection, i.e. a confused or refused connection
- *SYN* flag is used to establish a connection (see below)
- *FIN* flag is used to terminate a connection (see below)

10.32. TCP Example (HTTP Request)



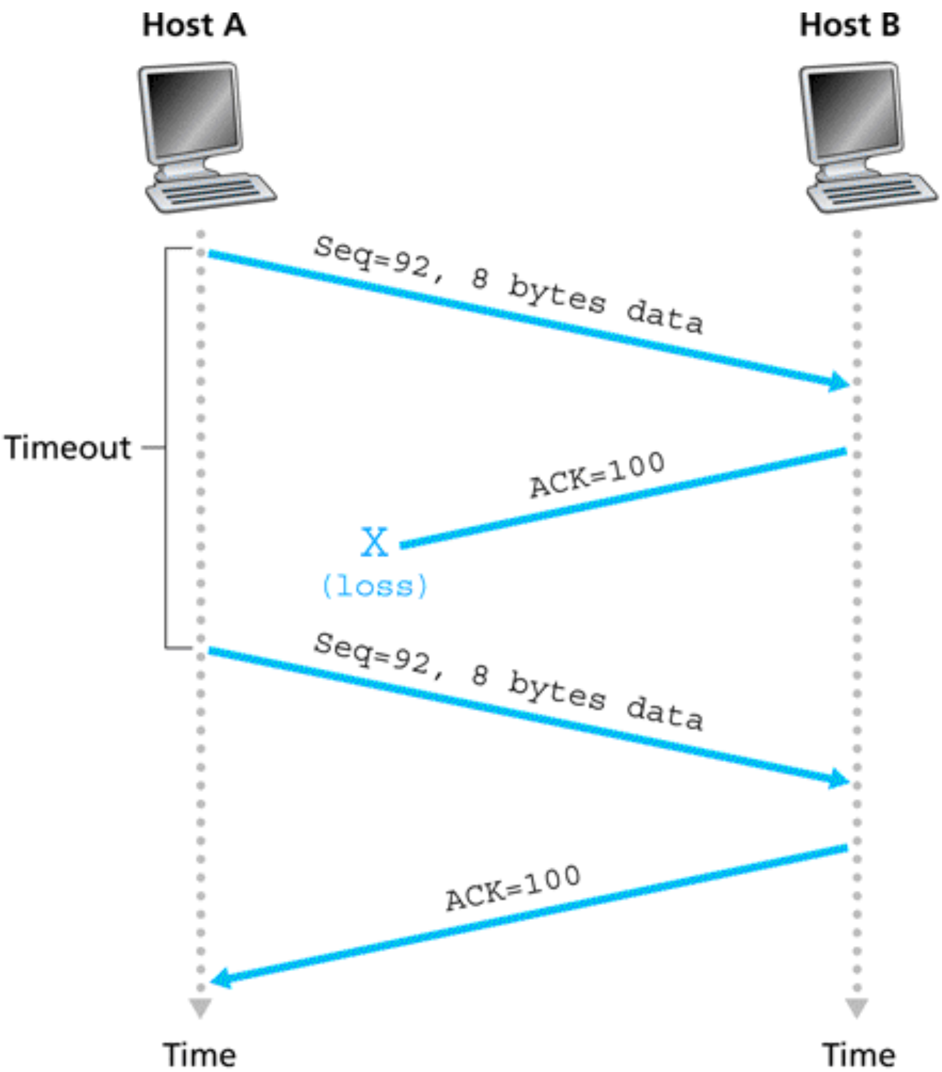
10.33. TCP Example (HTTP Response)



10.34. Sequence and Acknowledgement Numbers

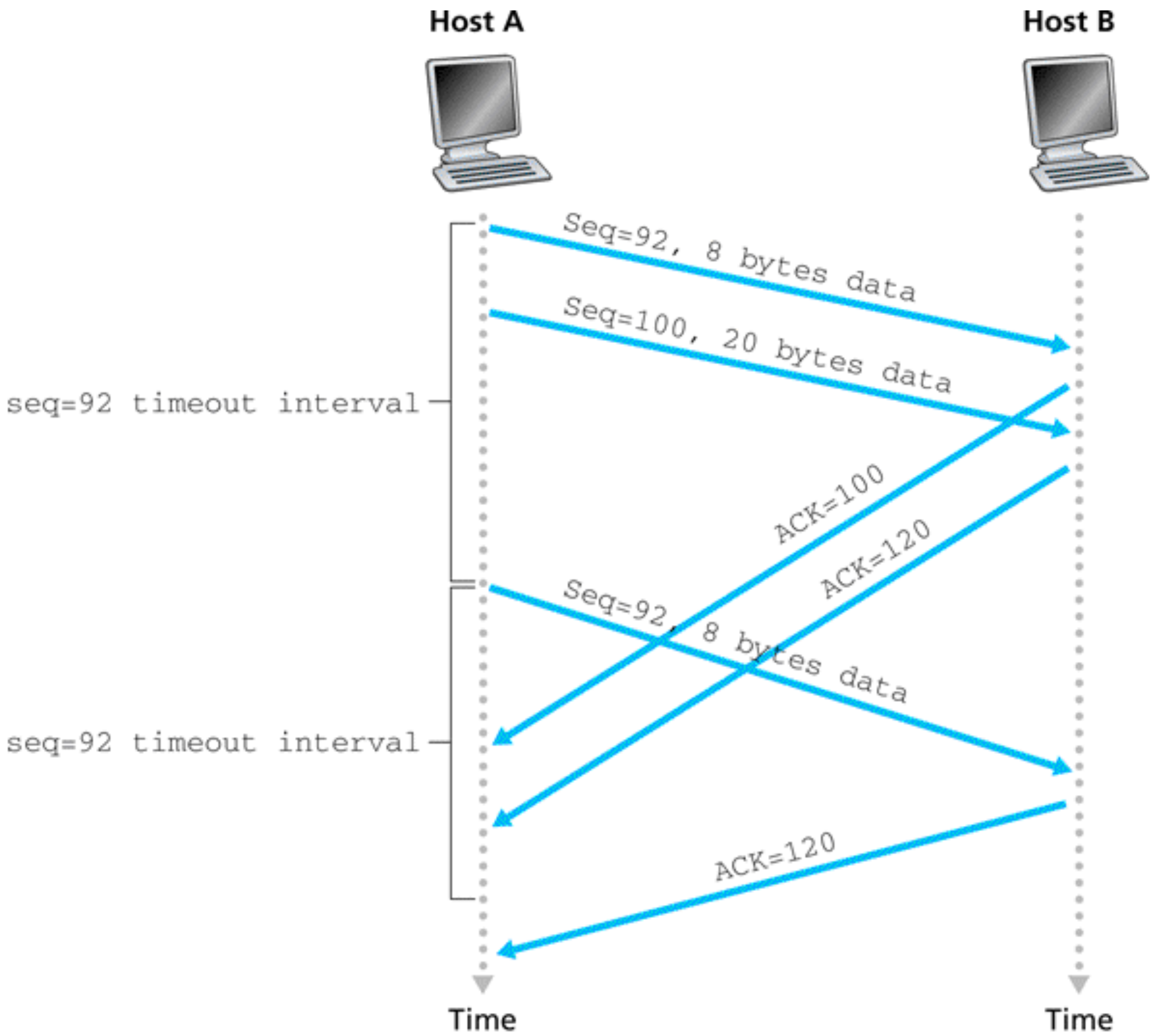
- TCP views data as an ordered stream of bytes
- sequence numbers are with respect to the stream of transmitted bytes
- the *sequence number* for a segment is therefore the byte-stream number of the first data byte in the segment
- the receiver uses the sequence number to re-order segments arriving out of order and to compute an acknowledgement number
- an *acknowledgement number* identifies the sequence number of the incoming data that the receiver expects next
- suppose Host A has received bytes 0 through 535 and 900 through 1000 from Host B, but not bytes 536 through 899
- A's next segment to B will contain 536 in the acknowledgement number field
- TCP only acknowledges bytes up to the first missing byte in the stream
- TCP is said to provide *cumulative acknowledgements*

10.35. Example: Lost Acknowledgement



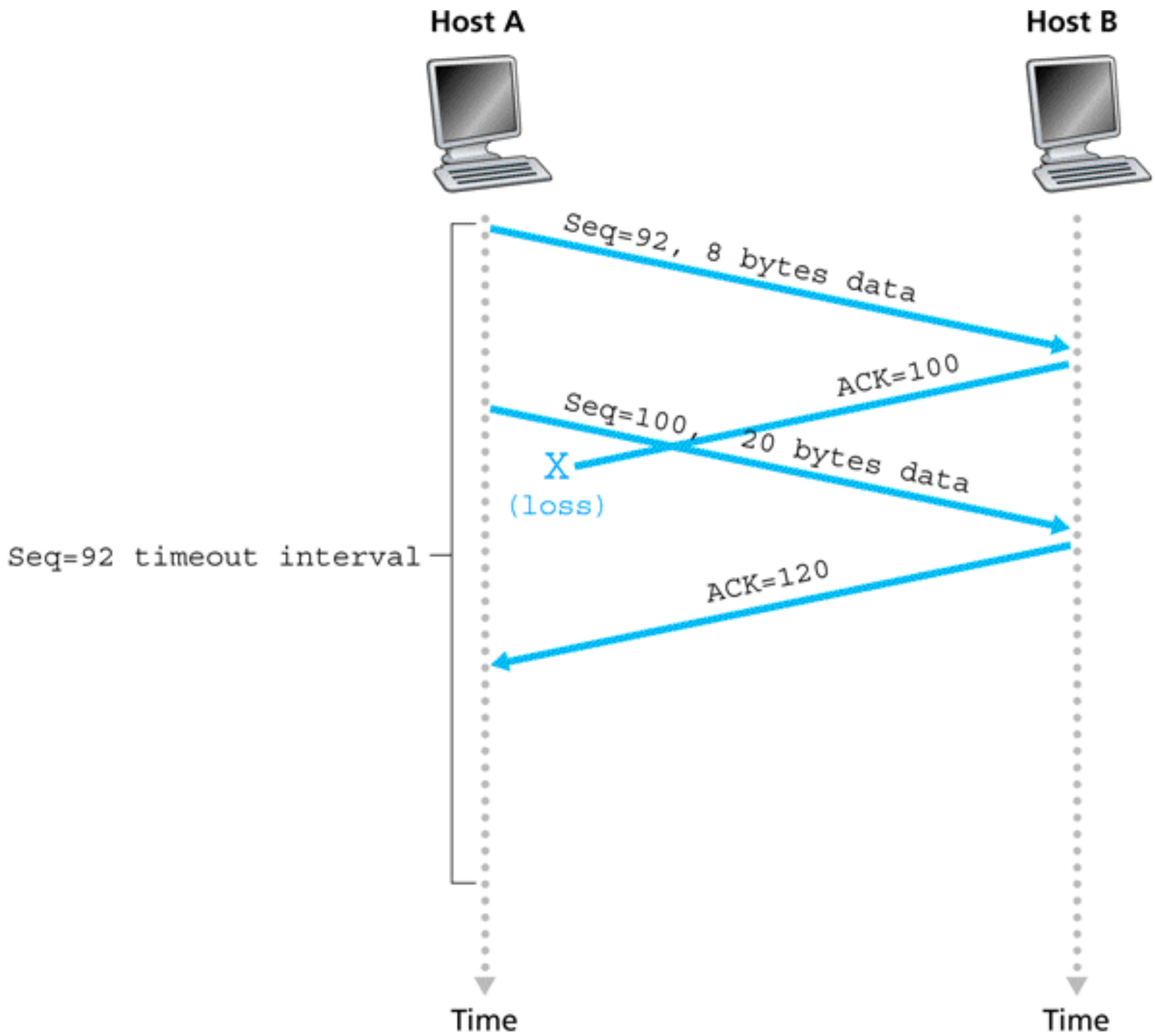
- Host A sends one segment to Host B
- this segment has sequence number 92 and contains 8 bytes of data
- the acknowledgement from B is lost
- A retransmits after its timer expires

10.36. Example: Single Retransmission



- Host A sends two segments back to back to Host B
- acknowledgements from B arrive only after timeout
- if acknowledgement for second segment arrives before the new timeout, the second segment will not be retransmitted

10.37. Example: No Retransmission Necessary

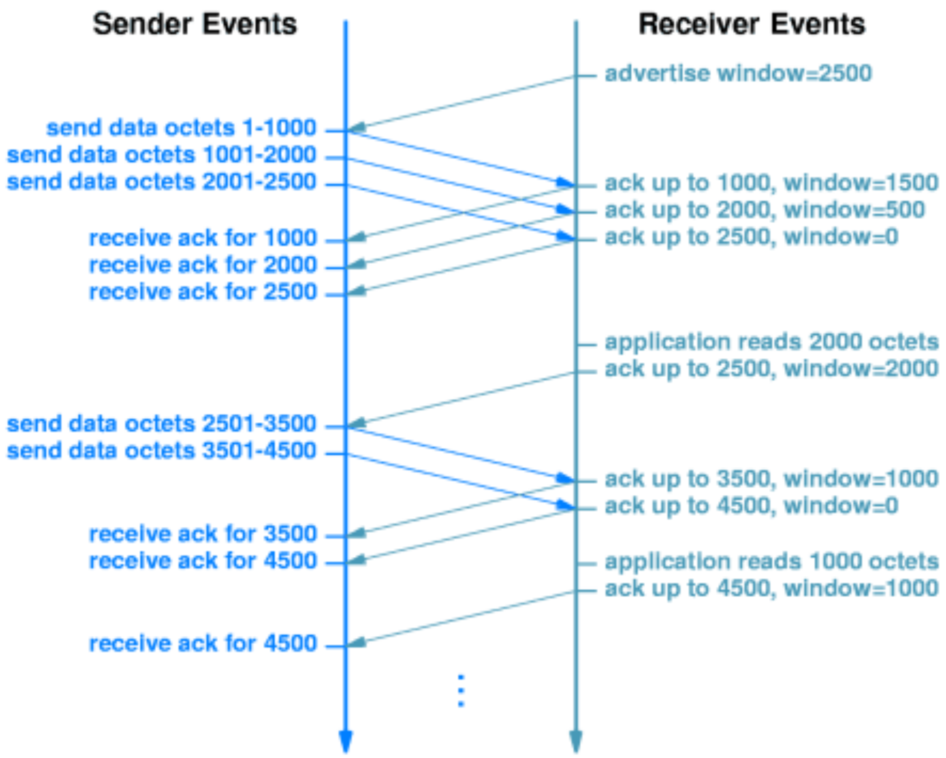


- Host A sends two segments back to back to Host B (as in previous example)
- suppose the acknowledgement for the first segment is lost
- if second acknowledgement arrives before timeout, A does not retransmit either segment

10.38. Flow Control

- TCP uses a *window* mechanism to control the flow of data
- when a connection is established, each end of the connection allocates a buffer to hold incoming data, and sends the size of the buffer to the other end
- as data arrives, the receiver sends acknowledgements together with the amount of buffer space available called a *window advertisement*
- if the receiving application can read data as quickly as it arrives, the receiver will send a positive window advertisement with each acknowledgement
- however, if the sender is faster than the receiver, incoming data will eventually fill the receiver's buffer, causing the receiver to advertise a *zero window*
- a sender that receives a zero window advertisement must stop sending until it receives a positive window advertisement

10.39. Flow Control Example

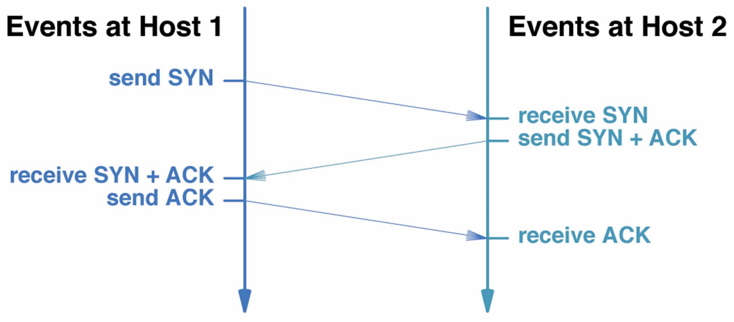


- sender is using a maximum segment size of 1000 bytes
- receiver advertises an initial window size of 2500 bytes
- sender transmits three segments (two containing 1000 bytes and one containing 500 bytes); then waits for an acknowledgement
- the first three segments fill the receiver's buffer faster than the receiving application can consume the data, so the advertised window reaches zero
- after the application reads 2000 bytes, the receiving TCP sends an additional acknowledgement advertising a window of 2000 bytes
- sender responds by sending two 1000-byte segments resulting in another zero window
- application reads 1000 bytes, so the receiving TCP sends an acknowledgement with a positive window size

10.40. TCP Connection Establishment

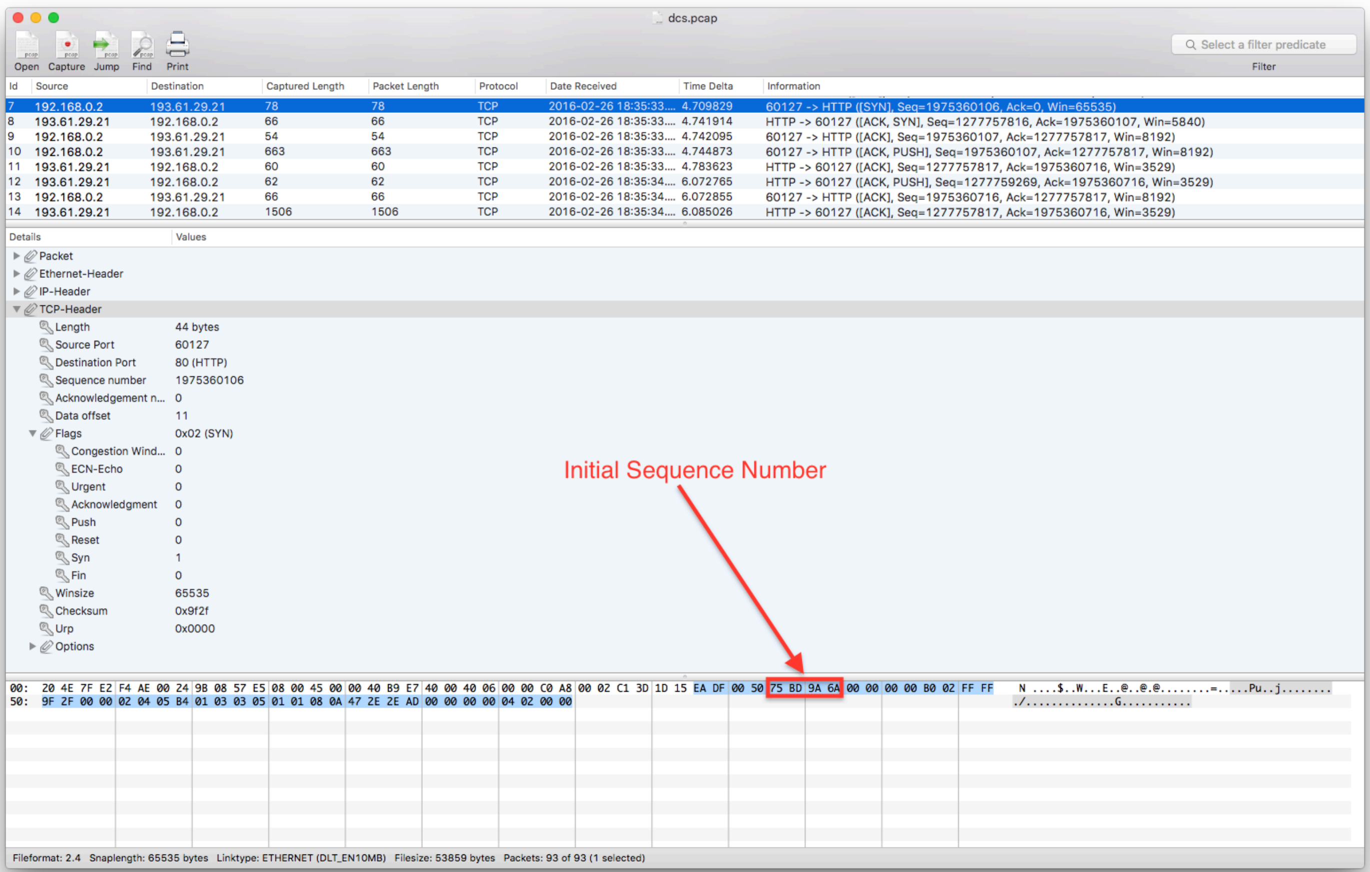
- connections are established by means of a *three-way handshake*
- each side sends a control message, specifying window size and *Initial Sequence Number* (ISN) which is randomly chosen
- a random ISN reduces the chance of a "lost" segment from an already-terminated connection being considered part of this connection
- the three steps are:
  - the sender sends a TCP segment (including window size and ISN) with the SYN flag on
  - the recipient sends a segment (including window size and ISN) with both SYN and ACK flags on
  - the sender replies with ACK

10.41. Example: Connection Establishment

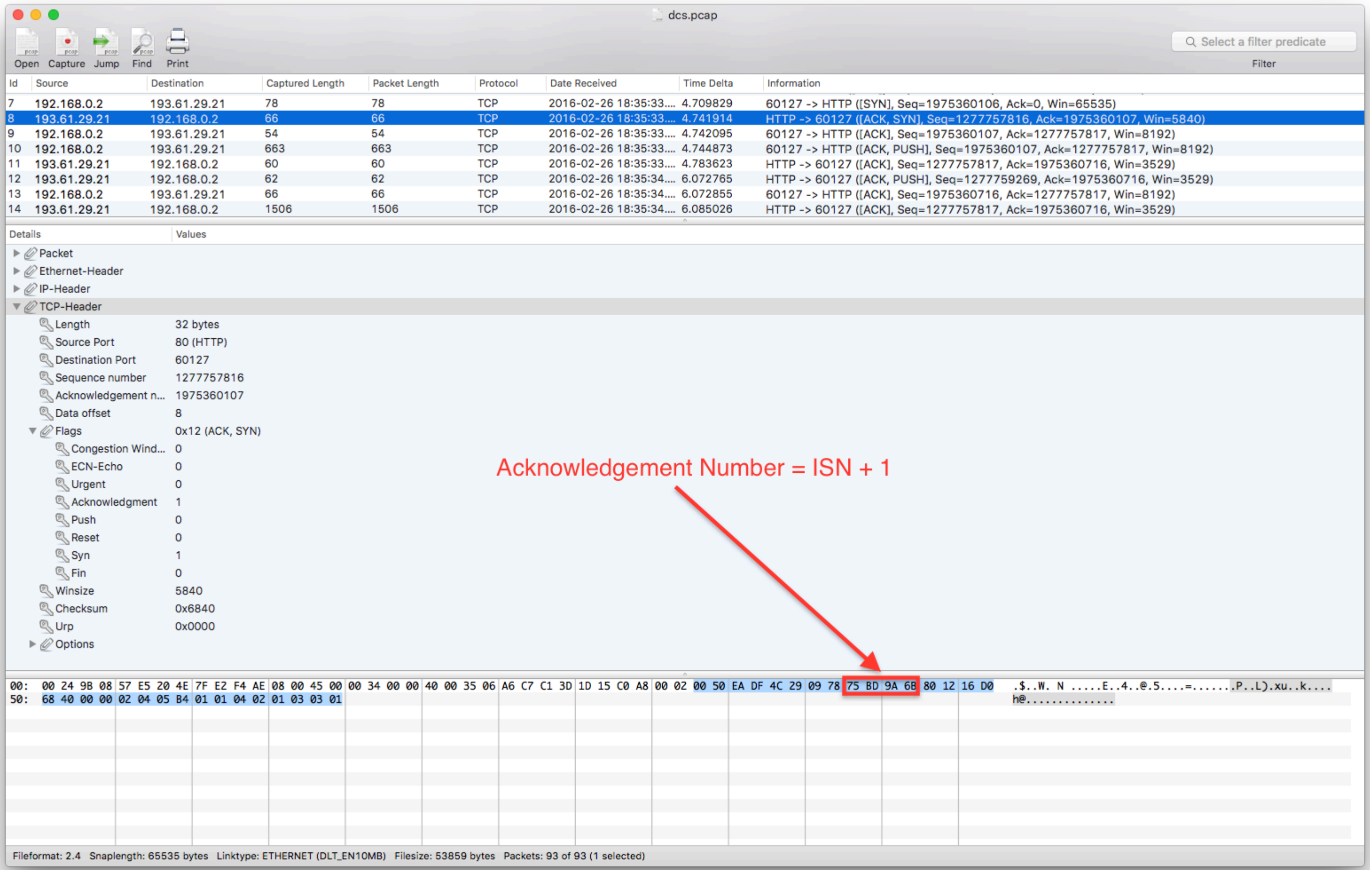


- host 1 opens the connection with an ISN
- host 2 accepts the connect request by sending a TCP segment which
  - acknowledges host 1's request (ACK flag on)
  - sets acknowledgement number to ISN+1
  - makes its own connection request (SYN flag on) with an ISN
- host 1 acknowledges this request
- note that the SYN flag "consumes" one byte of sequence space so that it can be acknowledged unambiguously

10.42. TCP Example SYN



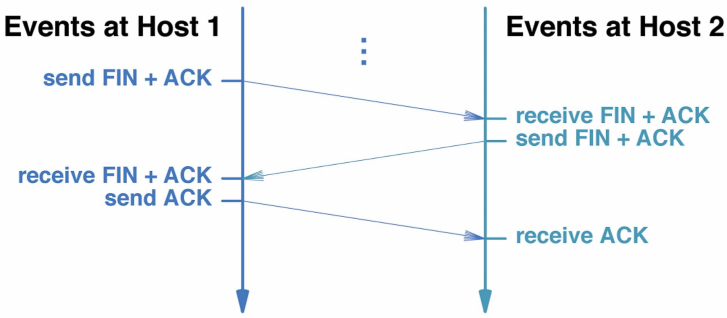
10.43. TCP Example ACK+SYN



10.44. SYN Flood Attack

- *SYN Flood Attack* is a type of Denial of Service (DoS) attack
- attacker sends a large number of TCP SYN segments without completing the third handshake step
- server sets up buffer space etc. for all SYN requests and so consumes all its resources
- solution is for server to choose as ISN a hash function of
    - source and destination IP addresses
    - source and destination port numbers
    - secret number known only to the server
- not to allocate resources until third handshake step
- nor to remember ISN
- if an ACK comes back, it can compute the hash value and check it against the ACK value (minus one)
- if no ACK, no resources have been allocated

10.45. TCP Connection Release



- a three-way handshake is also used to terminate a connection
- in this example, host 1 terminates the connection by transmitting a segment with the FIN flag set containing optional data
- host 2 acknowledges this (the FIN flag also consumes one byte of sequence space) and sets its own FIN flag
- the third and last segment contains host 1's acknowledgement of host 2's FIN flag

10.46. Congestion Control

- packet loss typically results from buffer overflow in routers as the network becomes *congested*
- congestion results from too many senders trying to send data at too high a rate
- packet retransmission treats a symptom of congestion, but not the cause
- to treat the cause, senders must be "throttled" (reduce their rate)
- TCP implements a congestion control algorithm based on perceived congestion by the sender:
    - if it perceives little congestion, it increases its send rate
    - if it perceives there is congestion, it reduces its send rate
- we will not cover the details of how TCP does this

10.47. Links to more information

- The companion web site for Tanenbaum's book, Chapter 6.

See Chapter 3 of [Kurose and Ross], Chapters 25 and 26 of [Comer] and parts of Chapter 6 of [Tanenbaum].