# CS118 Discussion 1B, Week 4

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HAINES A2 / Friday / 12:00pm-1:50pm

# Outline

- Lecture review: Transport layer
- Homework 3 review

# Principles of Reliable Data Transfer

- How to deal with bit errors?
  - Error detection (e.g. checksum)
  - Receiver feedback
  - Retransmission
  - Why not error correction?
- How to deal with duplicate packets due to retransmission?
- How can the sender detect that ACK or data is lost‘

# Stop and Wait Protocol

- Main Issue: **limited performance**

- Consider two hosts that are directly connected by a 50 Kbps satellite link that has a 250 milliseconds propagation delay. If these hosts send 1000 bits segments, what is the maximum throughput in stop-and-wait protocol if we ignore the transmission time of ACK?

# Stop and Wait Protocol

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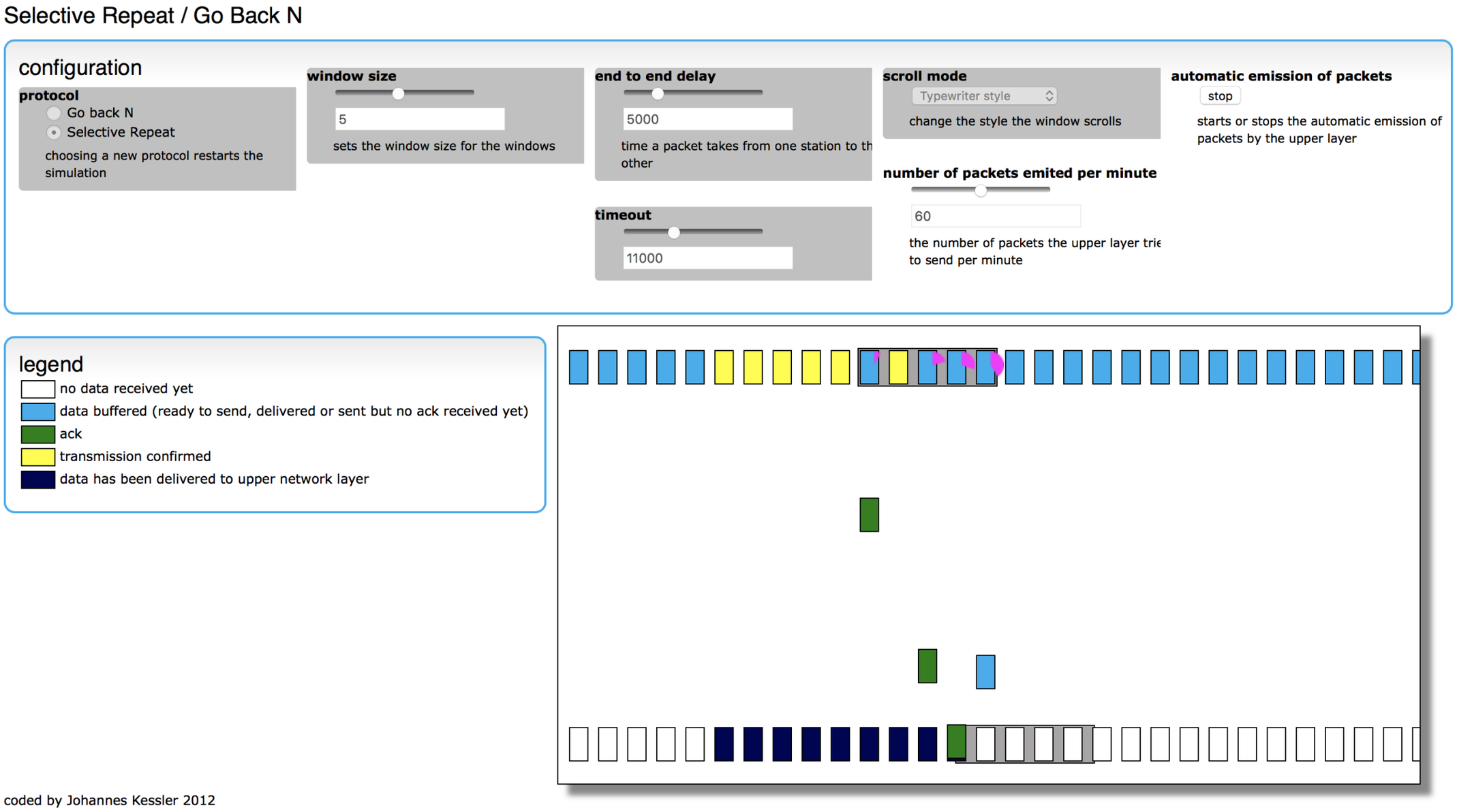
- 1000/(1000/50000+0.25+0.25) = 1923 bps < **2 Kbps << 50 Kbps!**

# Pipelined Protocols

- Go-back-N: receiver only sends **cumulative** ACKs
  - Drop out-of-order segments
  - reACK packet with highest in-order sequence number
  - Timer for oldest unACKed packet only, retransmit all unACKed packets
- Selective repeat: receiver ACKs **individual** packets
  - Buffer out of order segments
  - Timer for each individual unACKed packet, retransmit any unACKed packet

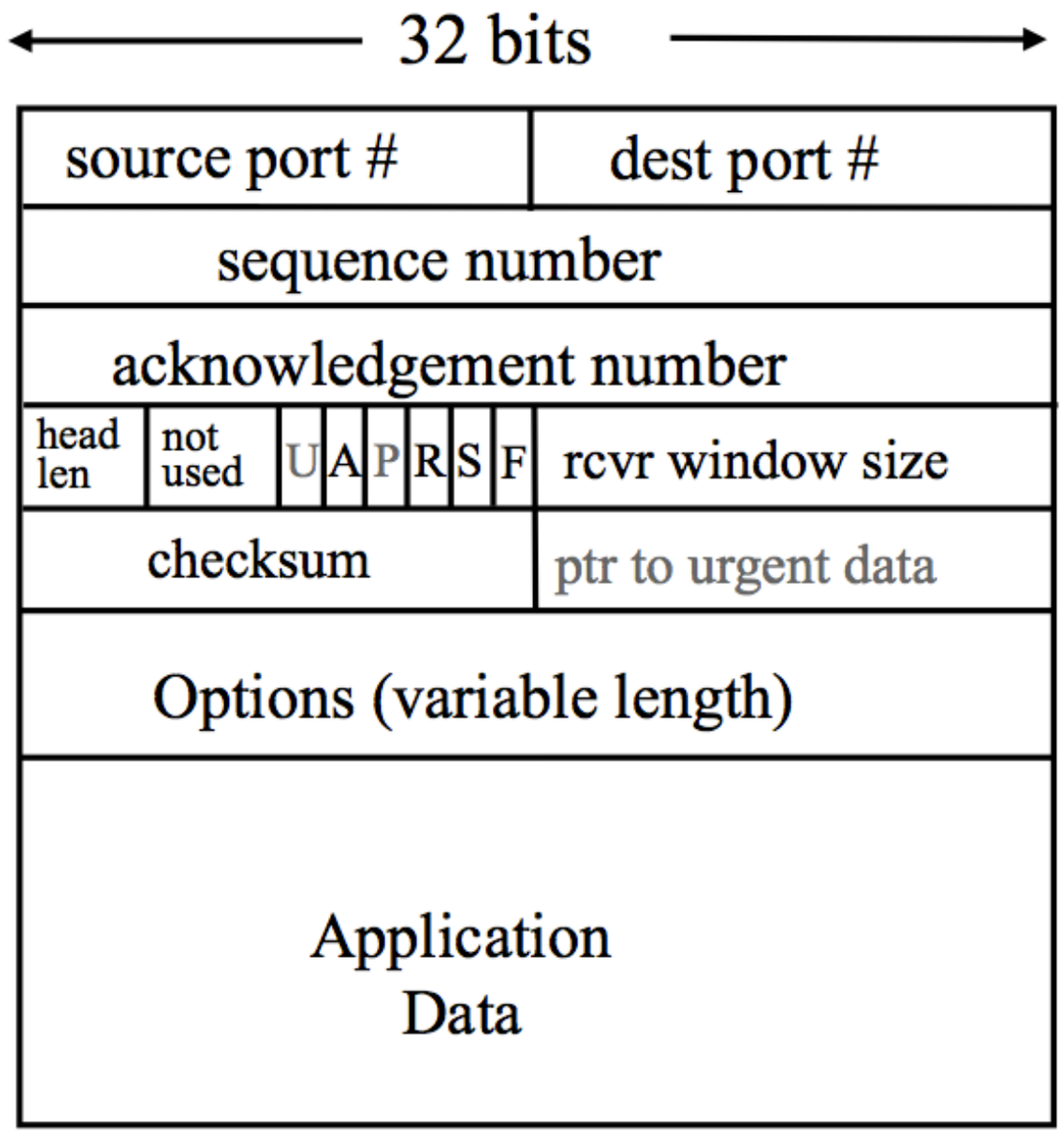
# Demo: Selective Repeat/Go Back N

- http://www.ccs-labs.org/teaching/rn/animations/gbn_sr/



# TCP

- Point-to-point, byte-stream reliable transport protocol
  - **Multiplexing/de-multiplexing**: Source/Dest port
  - **Reliable data transfer**: sequence number, ack, checksum, RTT estimation
  - **Connection setup**: sequence number, SYN, receive window
  - **Connection teardown**: sequence number, FIN



# Comparison of Reliable Transport Protocol

| Protocol | Buffer at sender | Buffer at receiver | ACK | Timeout/Retransmission |
|---|---|---|---|---|
| Stop & Wait | No | No | No out-of-order | Retransmit timeout packet |
| Go-Back-N | Yes | No | Accumulative Seq# | Retransmit all packets in window |
| Selective Repeat | Yes | Yes | Received Seq# | Retransmit timeout packet |
| TCP | Yes | Yes | Next expected Seq# | Retransmit timeout packet |

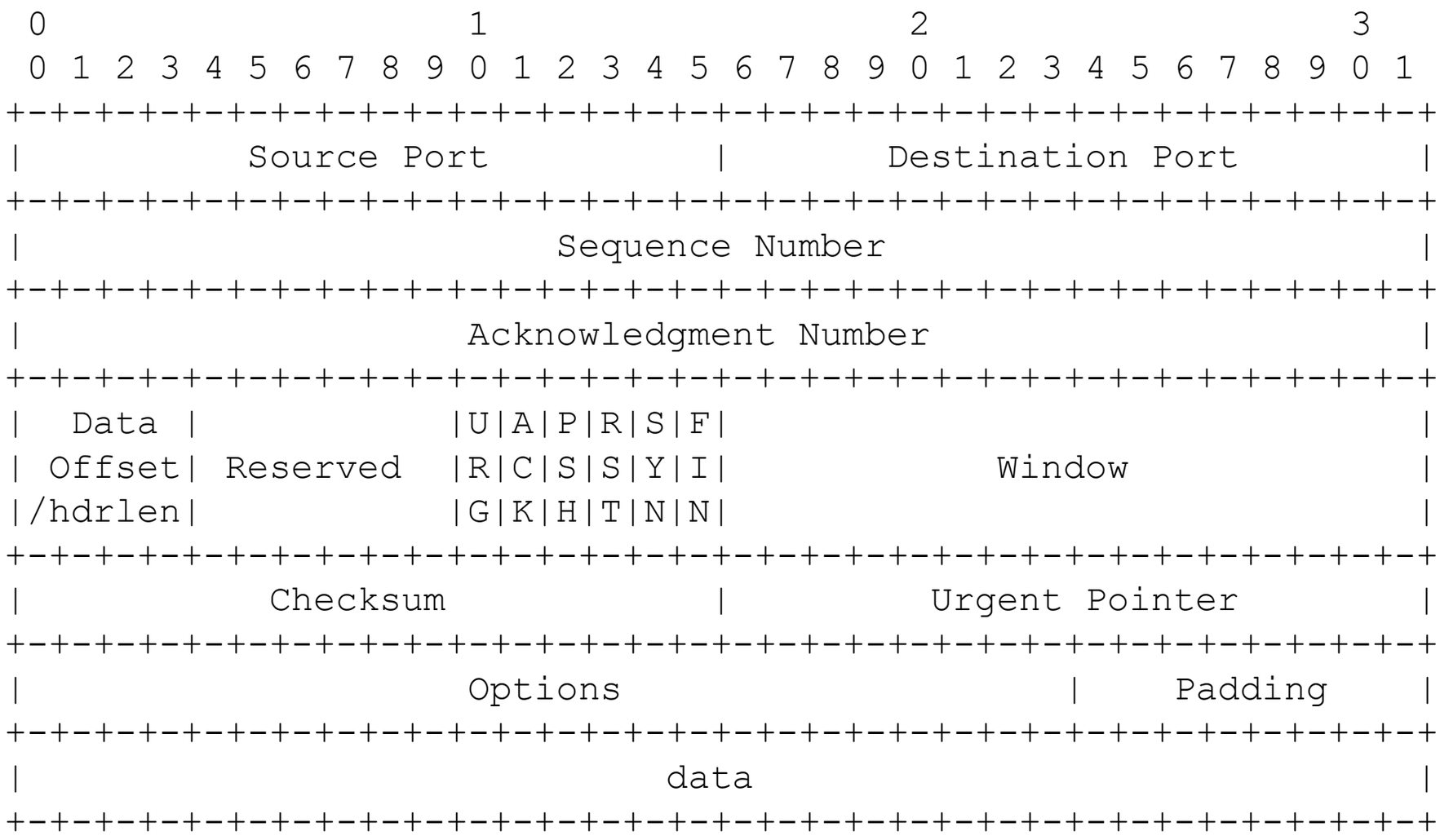
# TCP: overview

- The Transmission Control Protocol (TCP), documented in RFC 793, makes up for IP's deficiencies by providing reliable, stream-oriented connections that hide most of IP's shortcomings. The protocol suite gets its name because most TCP/IP protocols are based on TCP, which is in turn based on IP. TCP and IP are the twin pillars of TCP/IP.

# TCP: key functionalities

- TCP adds a great deal of functionality to the IP service it is layered over:
  - **Streams.** TCP data is organized as a stream of bytes, much like a file. The datagram nature of the network is concealed. A mechanism (the Urgent Pointer) exists to let out-of-band data be specially flagged.
  - **Reliable delivery.** Sequence numbers are used to coordinate which data has been transmitted and received. TCP will arrange for retransmission if it determines that data has been lost.
  - **Network adaptation.** TCP will dynamically learn the delay characteristics of a network and adjust its operation to maximize throughput without overloading the network.
  - **Flow control.** TCP manages data buffers, and coordinates traffic so its buffers will never overflow. Fast senders will be stopped periodically to keep up with slower receivers.

# TCP: header format




TCP Header Format

Note that one tick mark represents one bit position.

# TCP: header explained

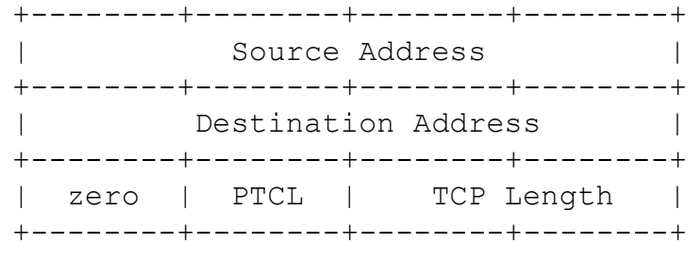
- Source Port: 16 bits
  - The source port number.
- Destination Port: 16 bits
  - The destination port number.
- Sequence Number: 32 bits
  - The sequence number of the first data octet in this segment (except when SYN is present). If SYN is present the sequence number is the initial sequence number (ISN) and the first data octet is ISN+1.
- Acknowledgment Number: 32 bits
  - If the ACK control bit is set this field contains the value of the next sequence number the sender of the segment is expecting to receive.  Once a connection is established this is always sent.
- Data Offset: 4 bits
  - The number of 32 bit words in the TCP Header.  This indicates where the data begins. The TCP header (even one including options) is an integral number of 32 bits long.

# TCP: header explained (cont'd)

- Reserved: 6 bits
  - Reserved for future use.  Must be zero.
- Control Bits: 6 bits (from left to right):
  - URG:  Urgent Pointer field significant
  - ACK:  Acknowledgment field significant
  - PSH:  Push Function
  - RST:  Reset the connection
  - SYN:  Synchronize sequence numbers
  - FIN:  No more data from sender
- Window: 16 bits
  - The number of data octets beginning with the one indicated in the acknowledgment field which the sender of this segment is willing to accept.

# TCP: header explained (cont'd)

- Checksum: 16 bits
  - The checksum field is the 16 bit one's complement of the one's complement sum of all 16 bit words in the header and text. If a segment contains an odd number of header and text octets to be checksummed, the last octet is padded on the right with zeros to form a 16 bit word for checksum purposes. The pad is not transmitted as part of the segment. While computing the checksum, the checksum field itself is replaced with zeros.
  - The checksum also covers a 96 bit pseudo header conceptually prefixed to the TCP header. This pseudo header contains the Source Address, the Destination Address, the Protocol, and TCP length. This gives the TCP protection against misrouted segments. This information is carried in the Internet Protocol and is transferred across the TCP/Network interface in the arguments or results of calls by the TCP on the IP. The TCP Length is the TCP header length plus the data length in octets (this is not an explicitly transmitted quantity, but is computed), and it does not count the 12 octets of the pseudo header.



# TCP: connection setup

- Connection setup: three-way handshaking
  - 1st round: SYN+initial sequence number
  - 2nd round: SYN+SYNACK+server's initial sequence number
  - 3rd round: SYNACK+(optional) data

# TCP: connection setup

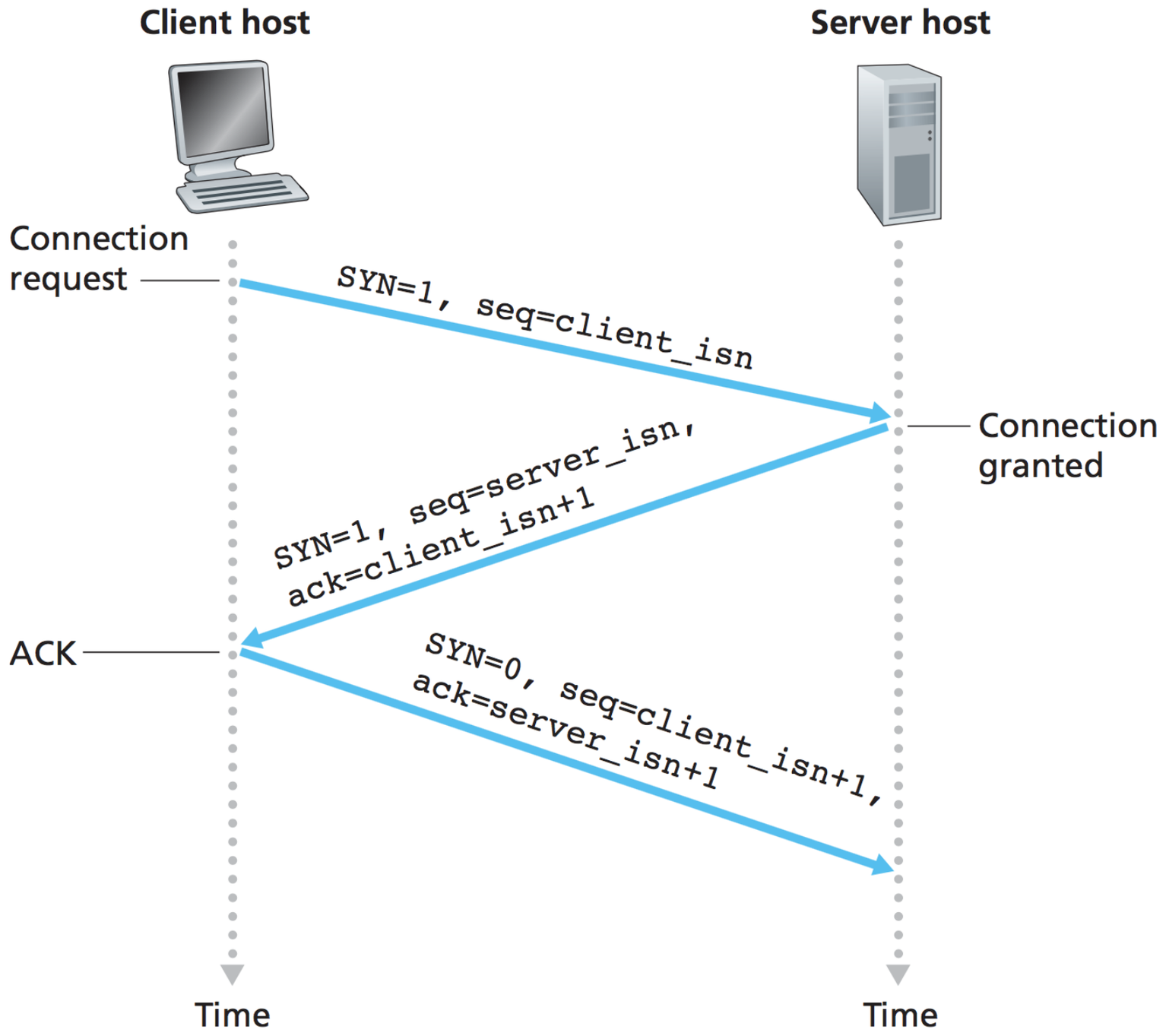
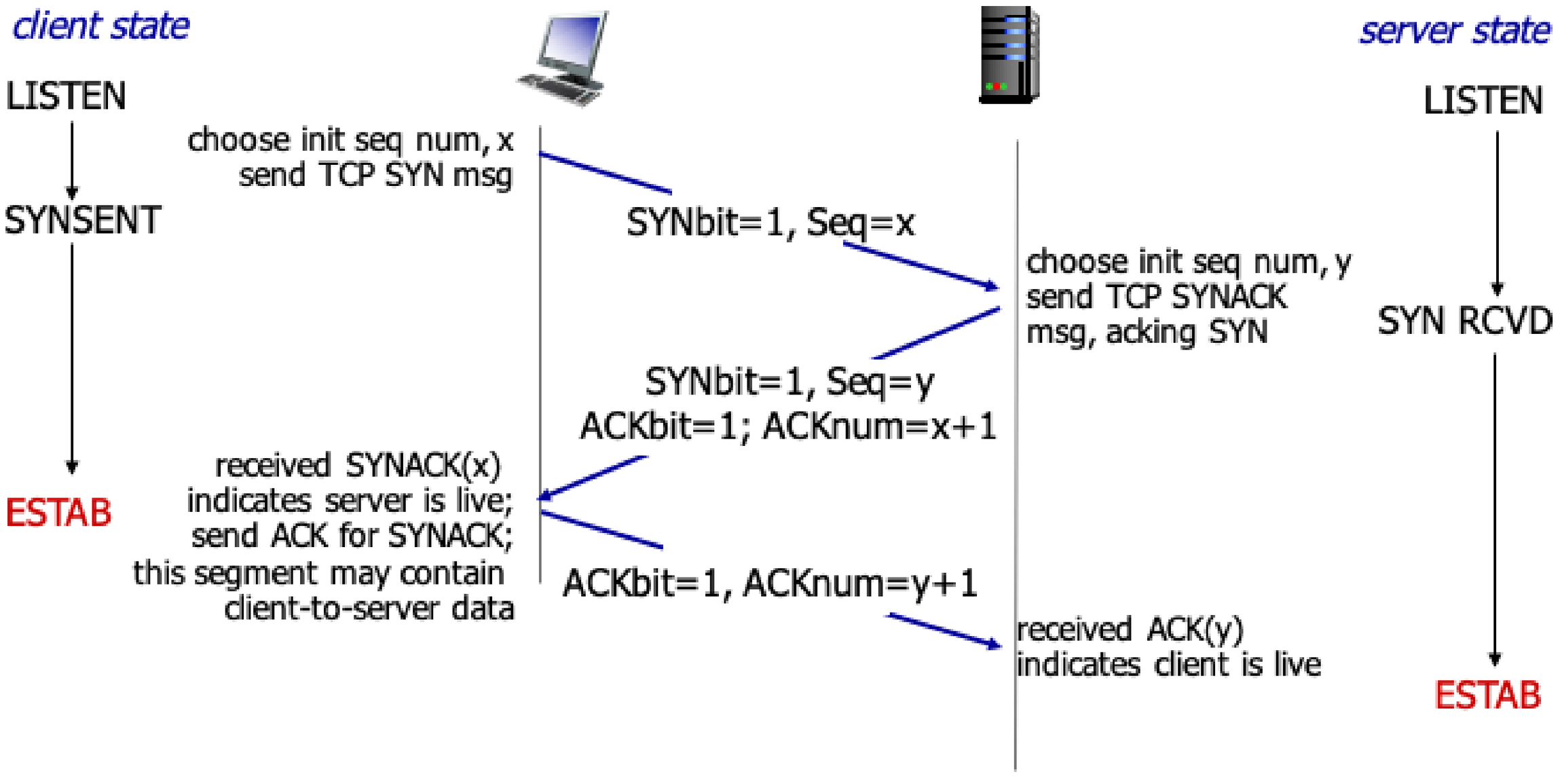



**Figure 3.39** ♦ TCP three-way handshake: segment exchange

# TCP: connection setup (cont'd)



# On TCP handshake sequence numbers

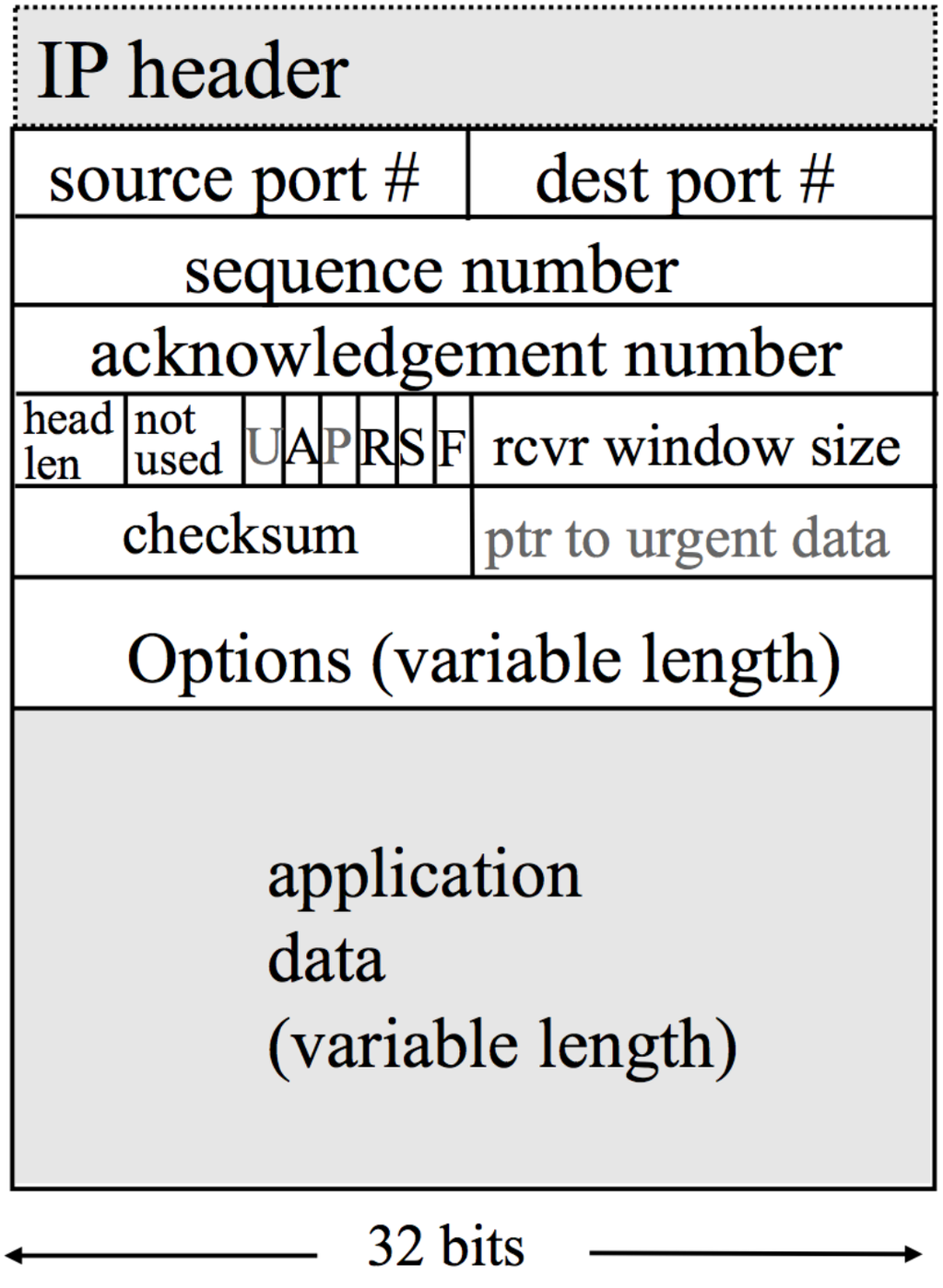
```
1) A --> B  SYN my sequence number is X
2) A <-- B  ACK your sequence number is X
3) A <-- B  SYN my sequence number is Y
4) A --> B  ACK your sequence number is Y
```

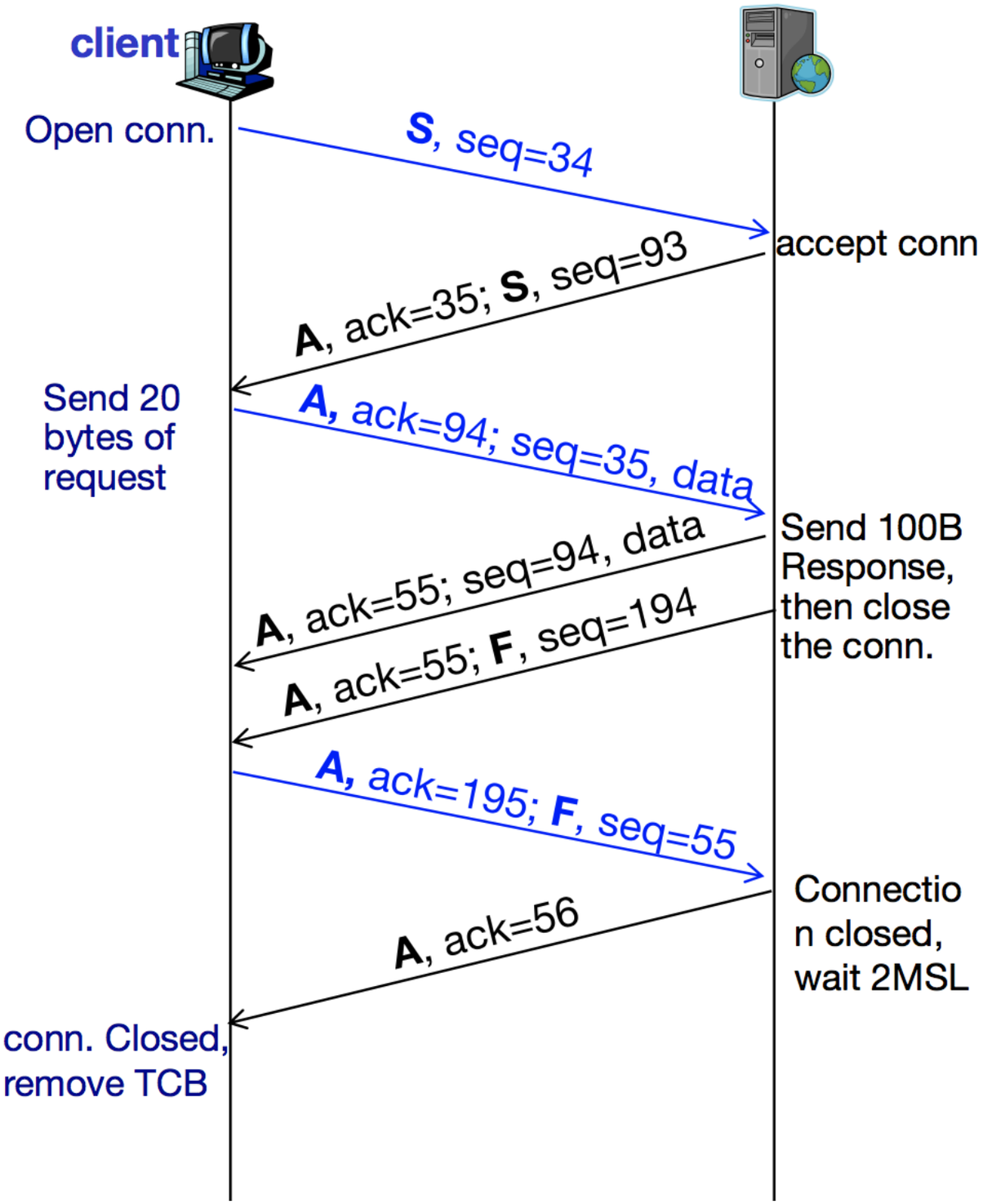
- Because steps 2 and 3 can be combined in a single message this is called the **three way (or three message) handshake**.

- How X and Y are chosen? Not specified; could be random numbers (using clock), as this is more secure. [RFC 793]

https://tools.ietf.org/html/rfc793#section-3.3
https://support.microsoft.com/en-us/help/172983/

# An HTTP 1.0 connection example



# TCP: connection teardown

- Normal termination
  - allow unilateral close
  - avoid sequence number overlapping
- TCP must continue to receive data even after closing
  - “Half-open/close” connection: cannot close connection immediately: what if a new connection restarts and uses same sequence number?

# TCP: connection teardown

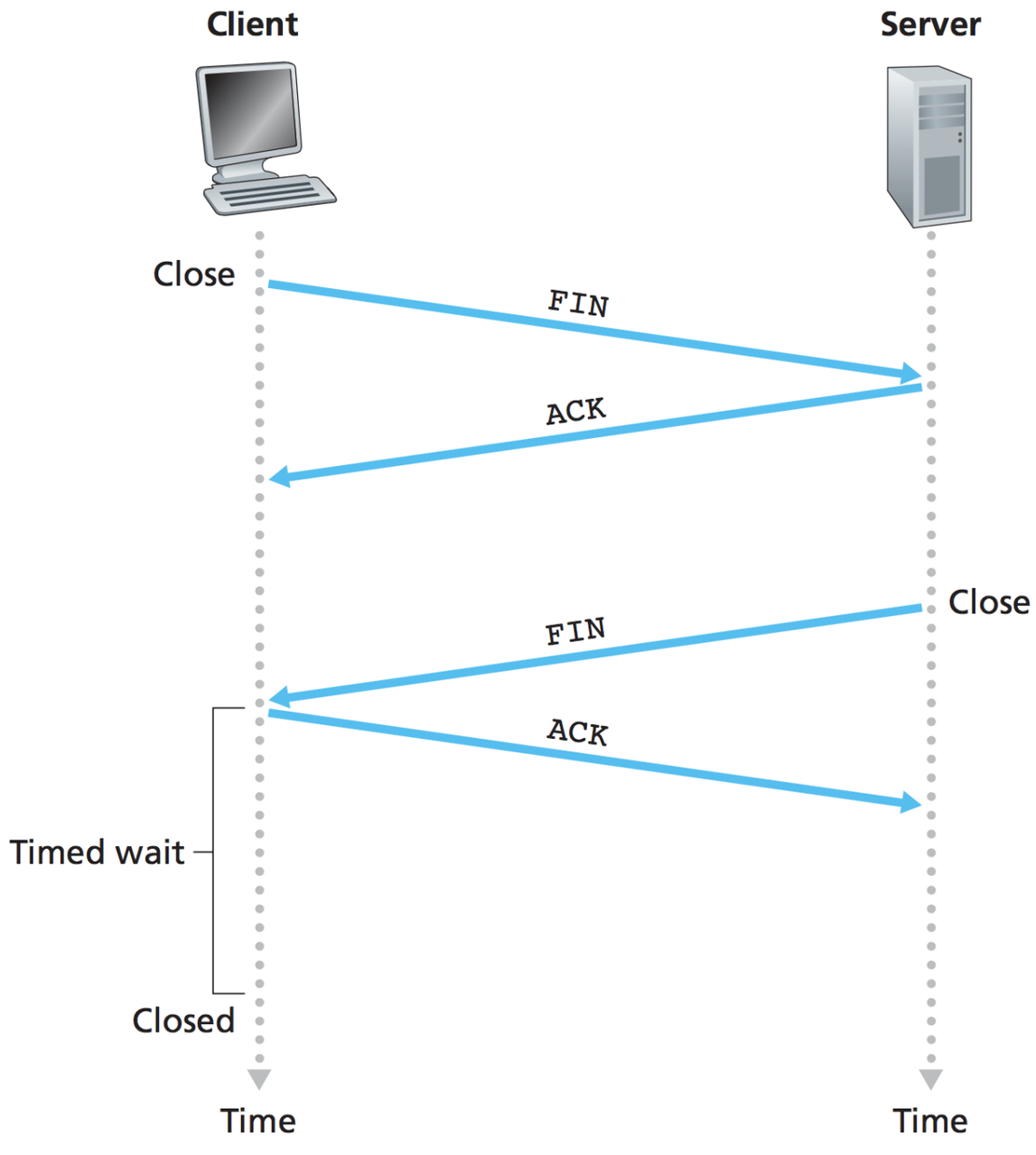



**Figure 3.40** ♦ Closing a TCP connection

# TCP: flow control

- Limits the rate a sender transfers data
- Avoid having the sender send data too fast
- Avoid exceeding the capacity of the receiver to process data
- Receiver specify the receive window
- The window size announce the number of bytes still free in the receiver buffer

# TCP: timeout

- EstimatedRTT = (1 - α) × EstimatedRTT + α × SampleRTT
- DevRTT = (1- β) x DevRTT + β × |SampleRTT - Estimated RTT|
- Retransmission Timer (RTO) = Estimated RTT + 4 × DevRTT

# Question

P31. Suppose that the five measured *SampleRTT* values (see **Section 3.5.3** ) are 106 ms, 120 ms, 140 ms, 90 ms, and 115 ms. Compute the *EstimatedRTT* after each of these SampleRTT values is obtained, using a value of $\alpha=0.125$ and assuming that the value of *EstimatedRTT* was 100 ms just before the first of these five samples were obtained. Compute also the *DevRTT* after each sample is obtained, assuming a value of $\beta=0.25$ and assuming the value of *DevRTT* was 5 ms just before the first of these five samples was obtained. Last, compute the TCP *TimeoutInterval* after each of these samples is obtained.

# Question

**Calculate the EstimatedRTT after obtaining the first sample RTT=106*ms*,**

*EstimatedRTT* = *α* * *SampleRTT*+(1- *α*) * *EstimatedRTT*

*EstimatedRTT* =0.125 * 106 + (1-0.125) * 100

=0.125* 106 + 0.875 * 100

=13.25 + 87.5

=100.75*ms*

**Calculate the DevRTT after obtaining the first sample RTT:**

*DevRTT* = *β* * | *SampleRTT- EstimatedRTT*|+(1- *β*)* *DevRTT*

=0.25 * |106-100.75| + (1-0.25) *5

=0.25 *5.25 + 0.75 * 5

=1.3125 + 3.75

=5.0625*ms*

**Calculate the Timeout Interval after obtaining the first sample RTT:**

*TimeoutInterval* = *EstimatedRTT* +4* *DevRTT*

= 100.75 + 4 *5.0625

=121*ms*

# TCP: timeout

- Karn’s algorithm — in case of retransmission
  - do not take the RTT sample (i.e. do not update SRTT or DevRTT)
  - double the retransmission timer value (RTO) after each timeout
  - Take RTT measure again upon next data transmission (that did not get retransmitted)

# References

- MIT 18.996: Topic in TCS: Internet Research Problems
- Princeton ELE539A: Optimization of Communication Systems
- http://www.freesoft.org/CIE/Course/Section4/