Transport Layer

Figure 12.1 TCP/IP protocol suite and interlayer address selectors.

Layered architecture

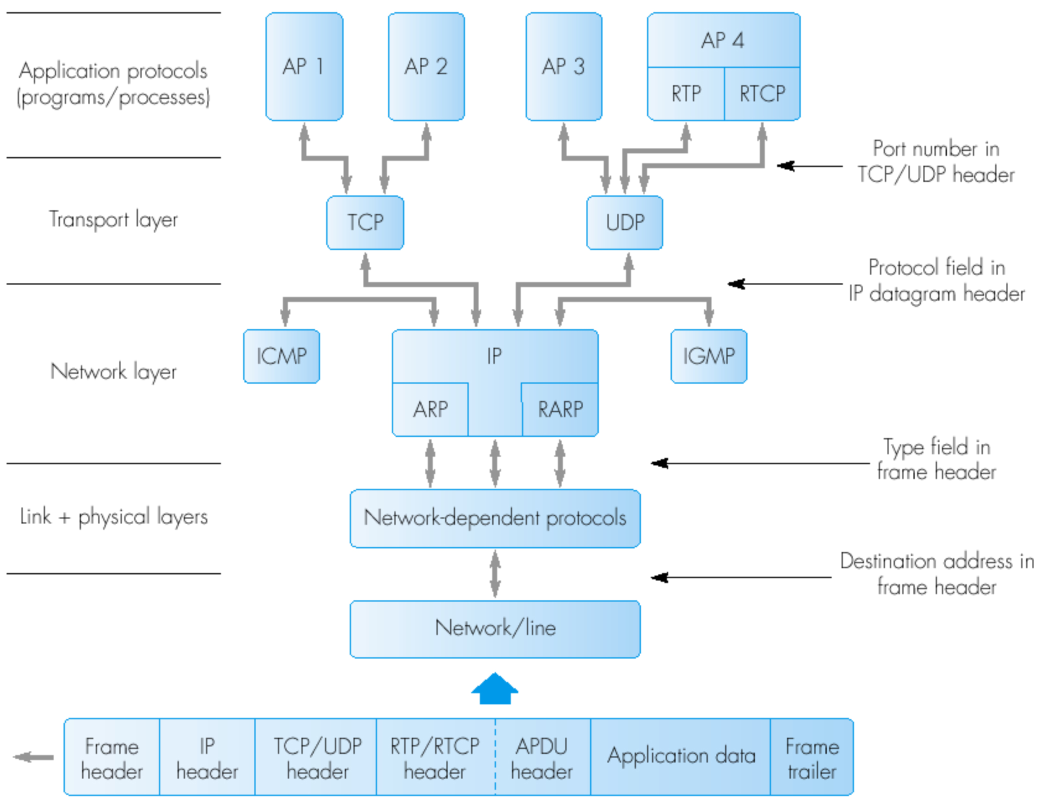
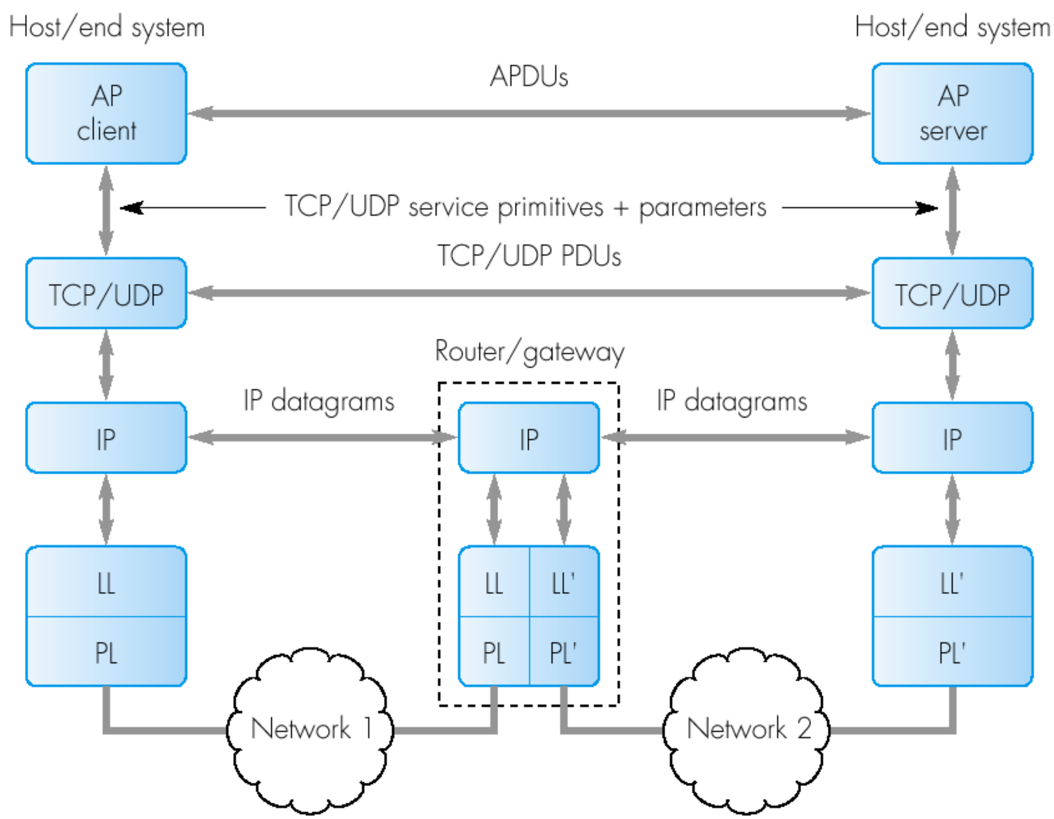


Figure 12.2 TCP/IP protocol suite interlayer communications.



Chapter : Transport Layer

our goals:

- understand principles behind transport layer services:
  - multiplexing, demultiplexing
  - flow control
  - congestion control
- learn about Internet transport layer protocols:
  - UDP: connectionless transport
  - TCP: connection-oriented reliable transport
  - TCP congestion control

Chapter outline

3.1 transport-layer services

3.2 multiplexing and demultiplexing

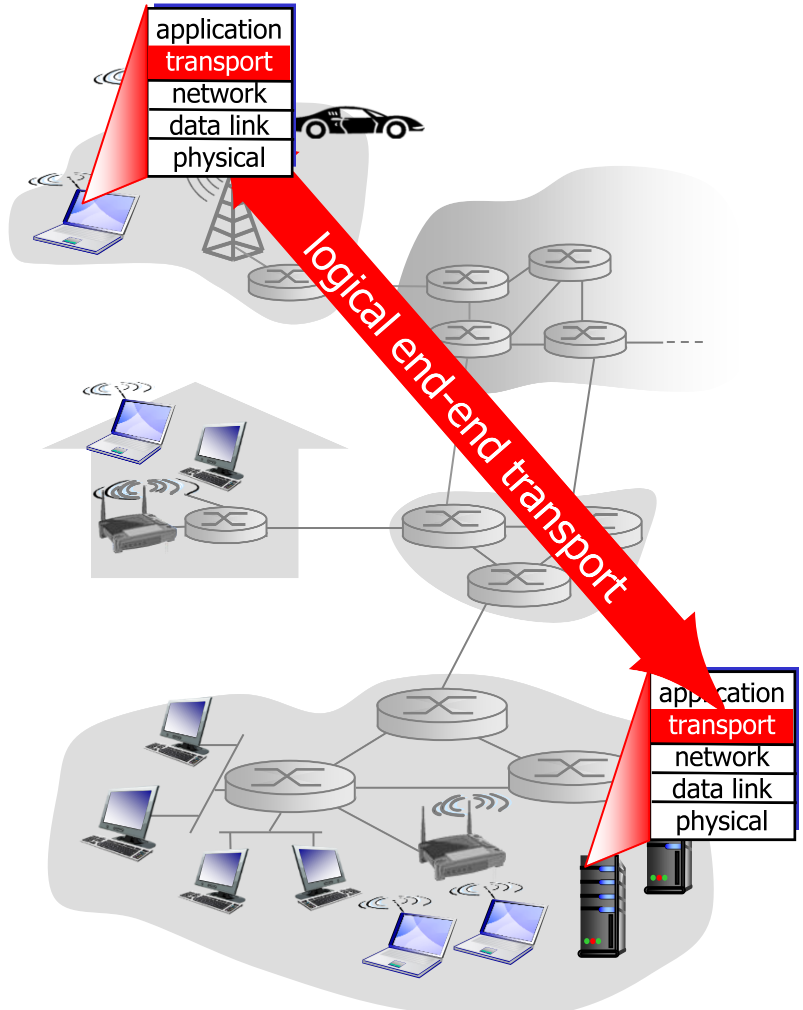
3.3 connectionless transport: UDP

3.4 connection-oriented transport: TCP
- segment structure
- reliable data transfer
- flow control
- connection management

3.6 TCP congestion control

Transport services and protocols

- provide *logical communication* between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into *segments*, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP



Transport vs. network layer

- *network layer:* logical communication between hosts
- *transport layer:* logical communication between processes
  - relies on, enhances, network layer services

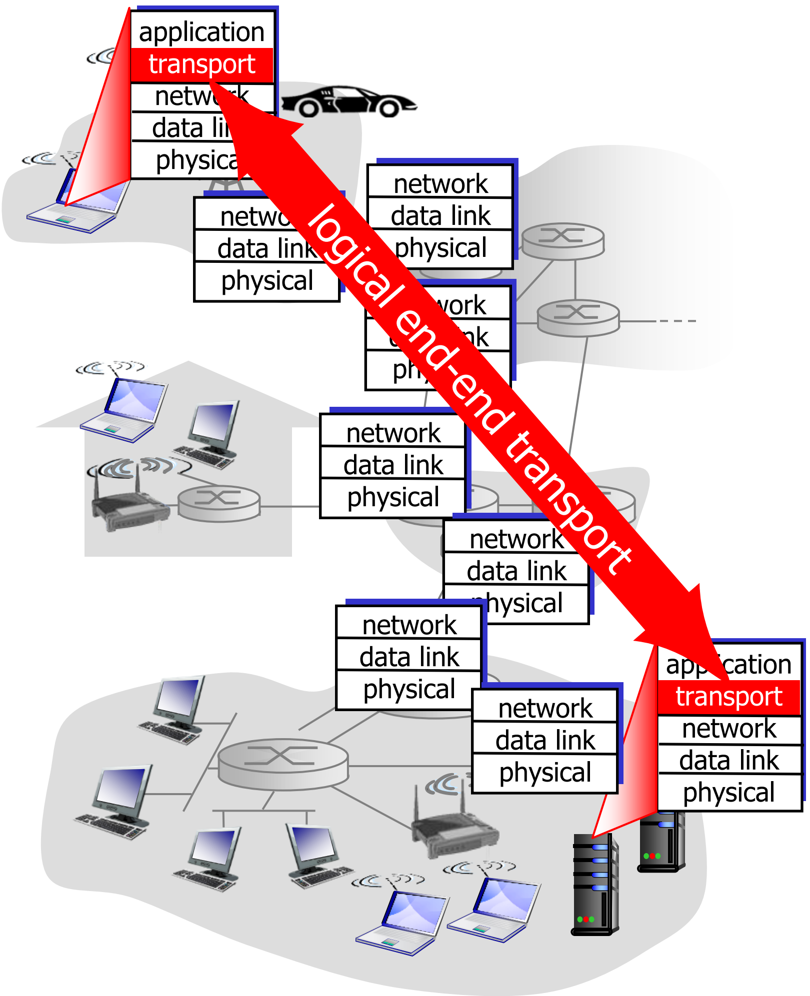
household analogy:

12 kids in Ann's house sending letters to 12 kids in Bill's house:

- hosts = houses
- processes = kids
- app messages = letters in envelopes
- transport protocol = Ann and Bill who demux to in-house siblings
- network-layer protocol = postal service

Internet transport-layer protocols

- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



Chapter 3 outline

3.1 transport-layer services

3.2 multiplexing and demultiplexing

3.3 connectionless transport: UDP

3.4 principles of reliable data transfer

3.5 connection-oriented transport: TCP
- segment structure
- reliable data transfer
- flow control
- connection management

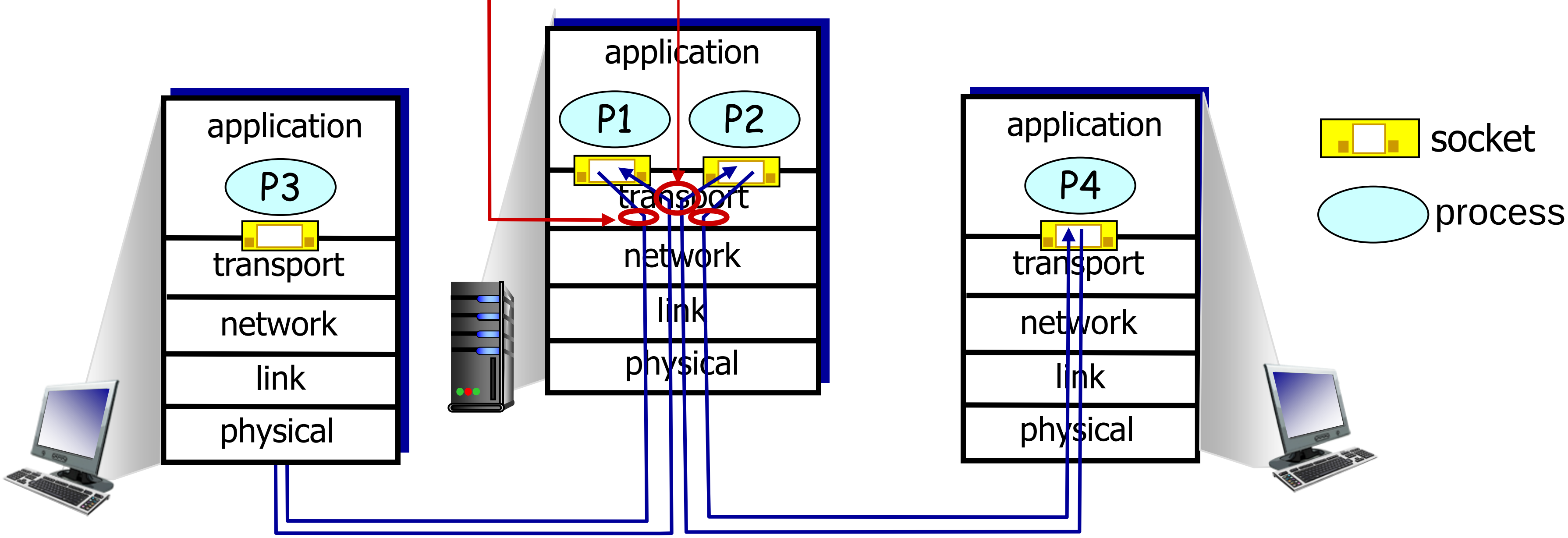
3.6 principles of congestion control

3.7 TCP congestion control

Multiplexing/demultiplexing

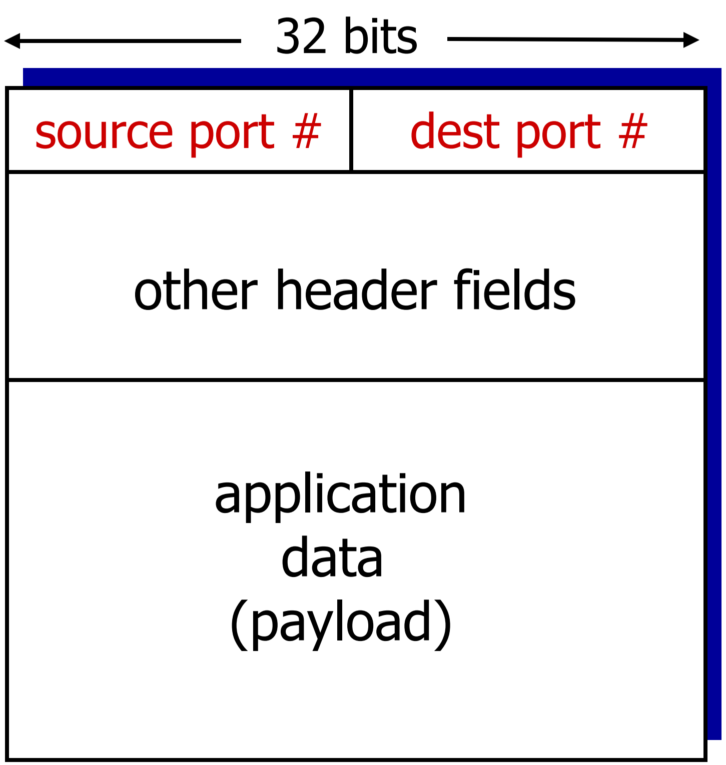
multiplexing at sender:
handle data from multiple sockets, add transport header (later used for demultiplexing)

demultiplexing at receiver:
use header info to deliver received segments to correct socket



How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries one transport-layer segment
  - each segment has source, destination port number
- host uses *IP addresses & port numbers* to direct segment to appropriate socket



TCP/UDP segment format

Outline

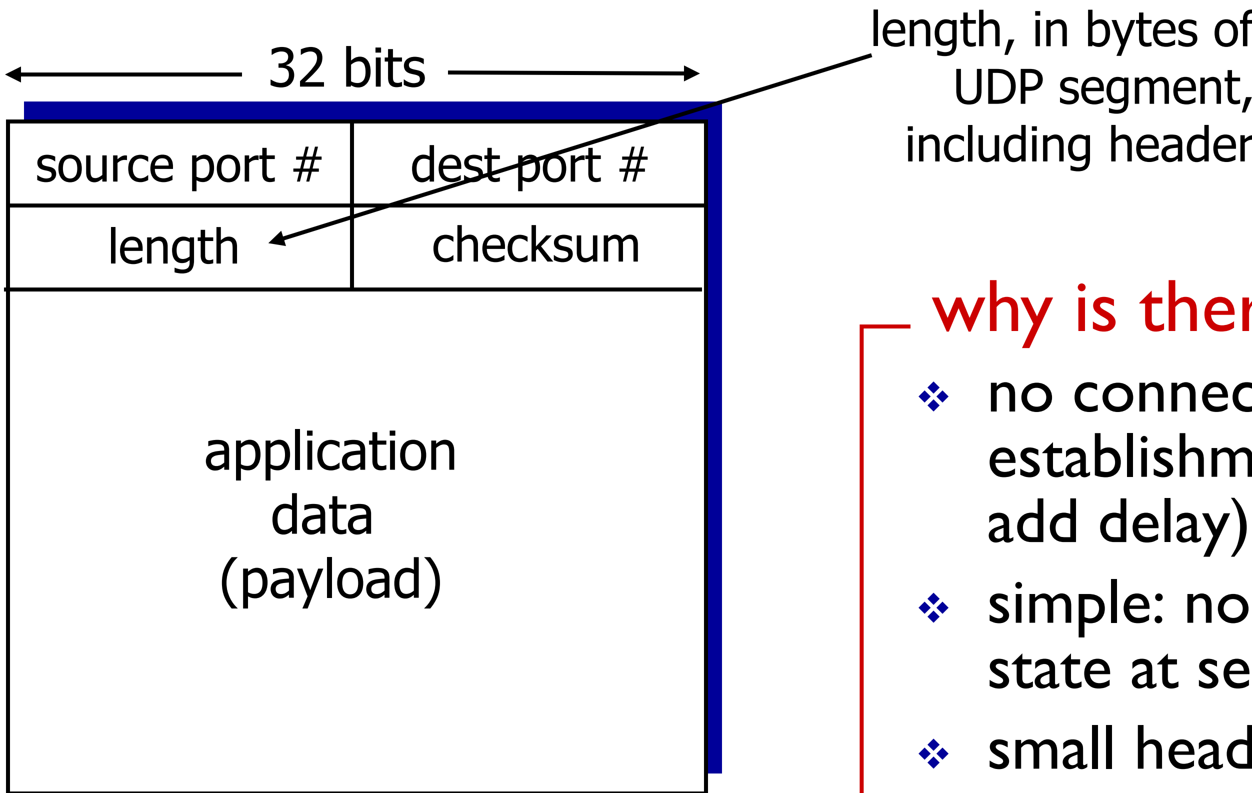
- 3.1 transport-layer services
- 3.2 multiplexing and demultiplexing
- 3.3 connectionless transport: UDP
- 3.4 principles of reliable data transfer
- 3.5 connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- 3.6 principles of congestion control
- 3.7 TCP congestion control

UDP: User Datagram Protocol [RFC 768]

- "no frills," Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out-of-order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
  - SNMP
- reliable transfer over UDP:
  - add reliability at application layer
  - application-specific error recovery!

UDP: segment header



UDP segment format

why is there a UDP?

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control: UDP can blast away as fast as desired

UDP checksum

Goal: detect "errors" (e.g., flipped bits) in transmitted segment

sender:

- treat segment contents, including header fields, as sequence of 16-bit integers
- checksum: addition (one's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonetheless?* More later ….

Internet checksum: example

example: add two 16-bit integers

		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

Note: when adding numbers, a carryout from the most significant bit needs to be added to the result

outline

3.1 transport-layer services

3.2 multiplexing and demultiplexing

3.3 connectionless transport: UDP

3.5 connection-oriented transport: TCP

- segment structure
- reliable data transfer
- flow control
- connection management

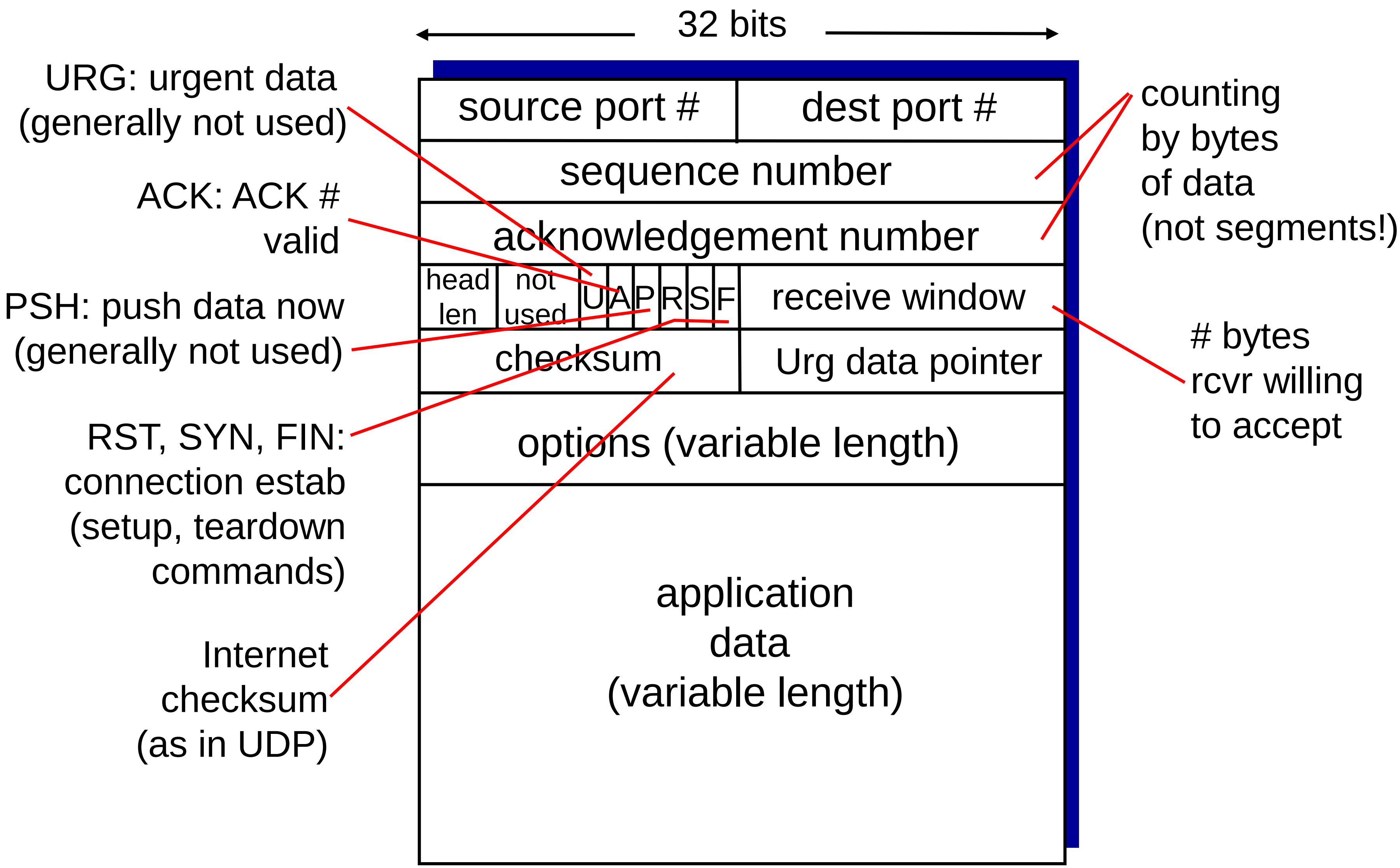
3.6 principles of congestion control

3.7 TCP congestion control

TCP: Overview RFCs: 793,1122,1323, 2018, 2581

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - no "message boundaries"
- pipelined:
  - TCP congestion and flow control set window size
- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control msgs) inits sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure

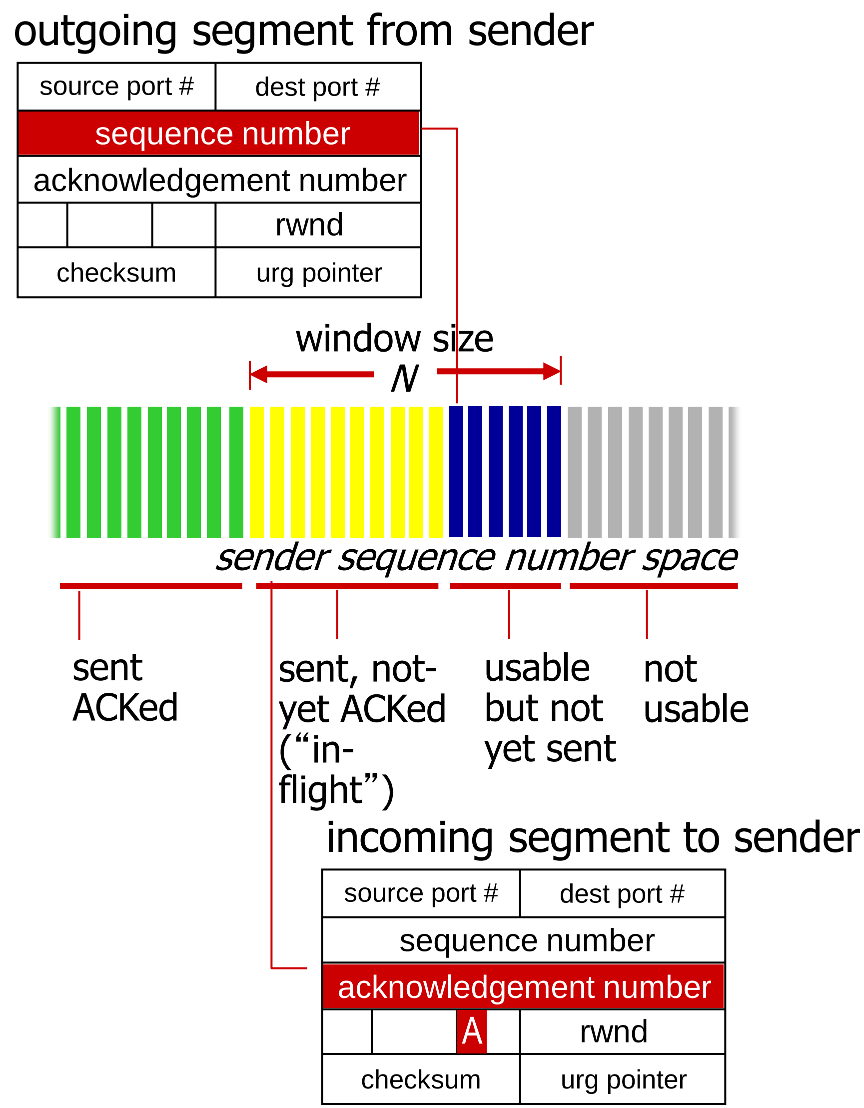


TCP seq. numbers, ACKs

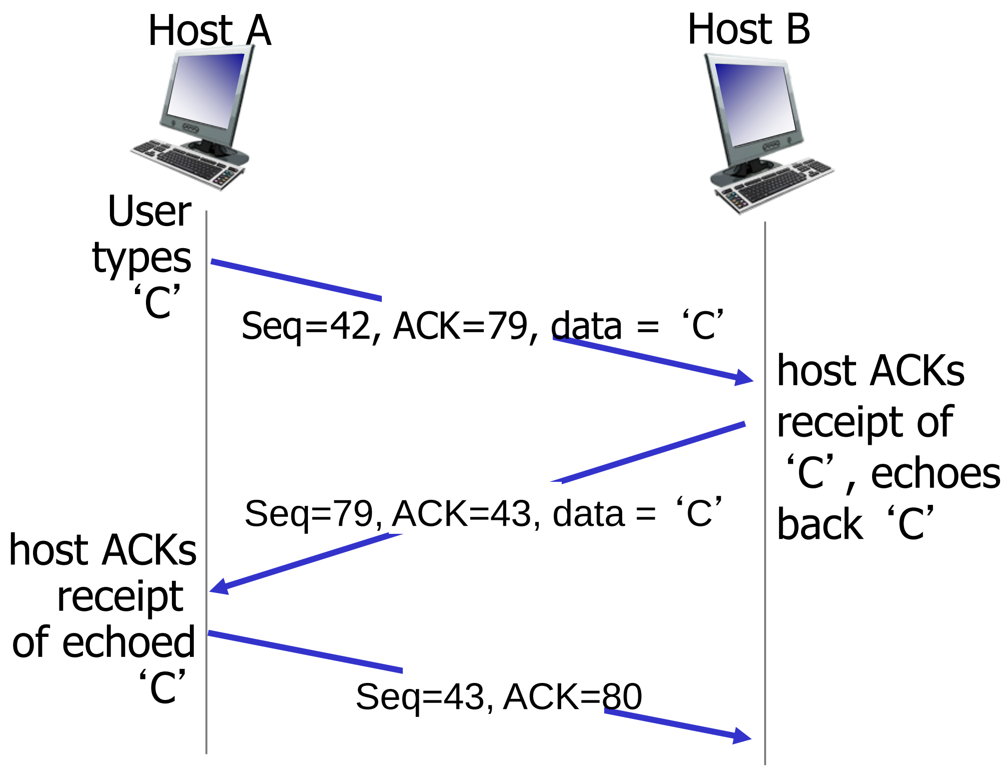
sequence numbers:
- byte stream "number" of first byte in segment's data

acknowledgements:
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments
- A: TCP spec doesn't say, - up to implementor



TCP seq. numbers, ACKs



simple telnet scenario

TCP round trip time, timeout

Q: how to set TCP timeout value?

- longer than RTT
  - but RTT varies
- *too short:* premature timeout, unnecessary retransmissions
- *too long:* slow reaction to segment loss

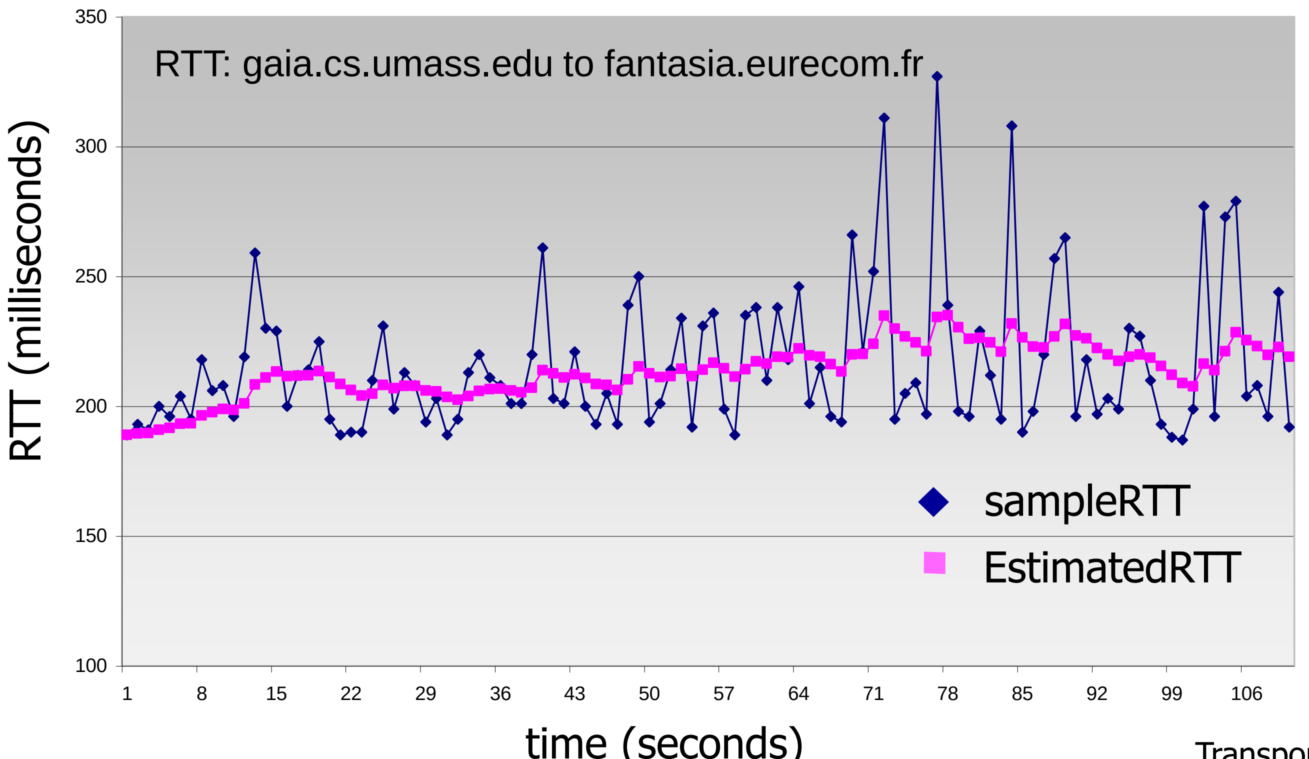
Q: how to estimate RTT?

- **`SampleRTT`**: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- **`SampleRTT`** will vary, want estimated RTT "smoother"
  - average several *recent* measurements, not just current **`SampleRTT`**

TCP round trip time, timeout

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

- exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: $\alpha = 0.125$



TCP round trip time, timeout

- timeout interval: **`EstimatedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`** -> larger safety margin
- estimate SampleRTT deviation from EstimatedRTT:

```
DevRTT = (1-β)*DevRTT +
              β*|SampleRTT-EstimatedRTT|
              (typically, β = 0.25)
```

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

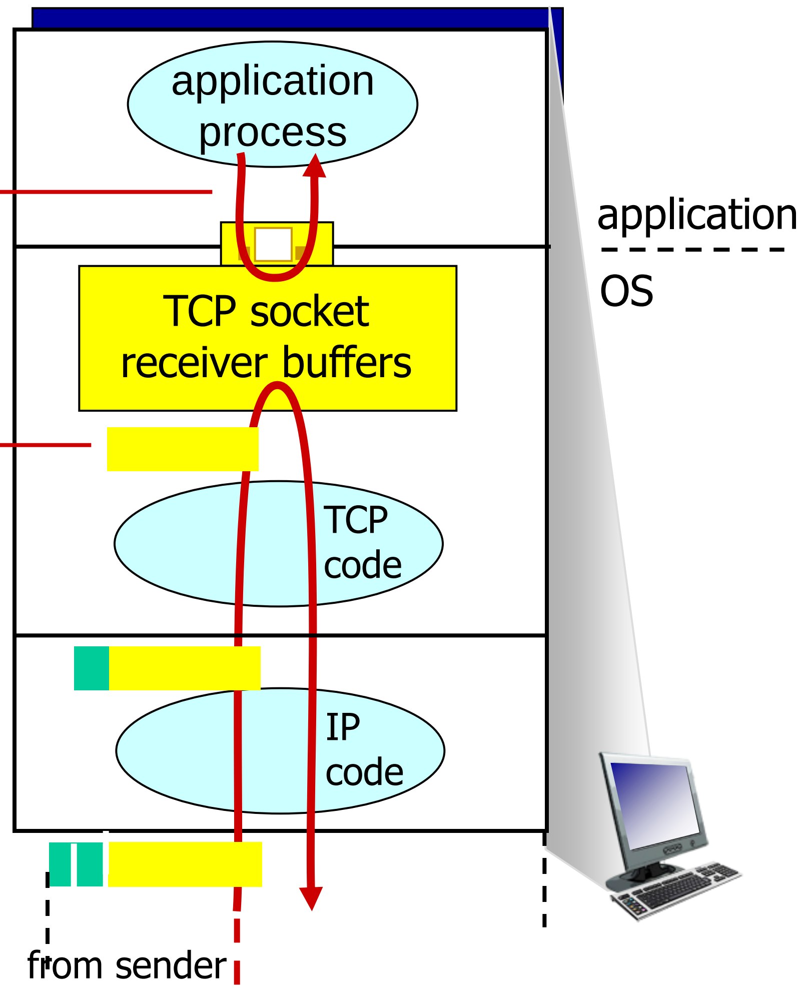
estimated RTT

"safety margin"

TCP flow control

application may remove data from TCP socket buffers ….

… slower than TCP receiver is delivering (sender is sending)



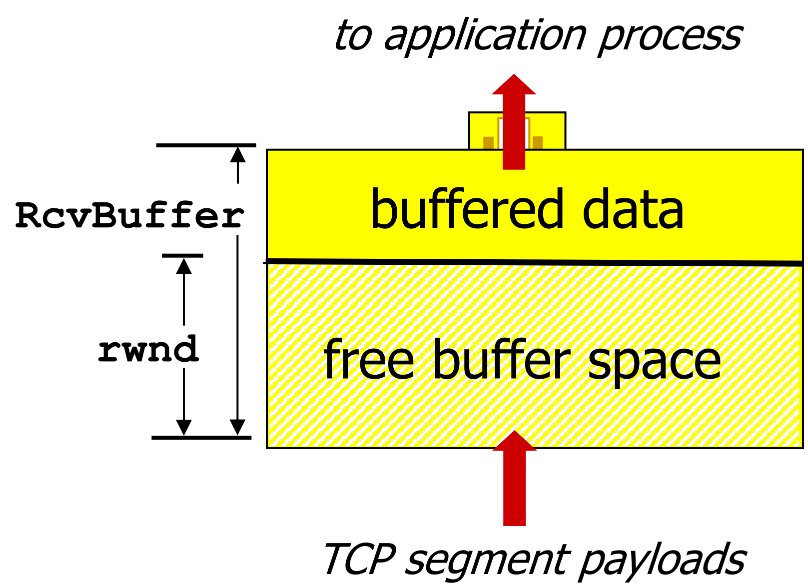
receiver protocol stack

flow control

receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast

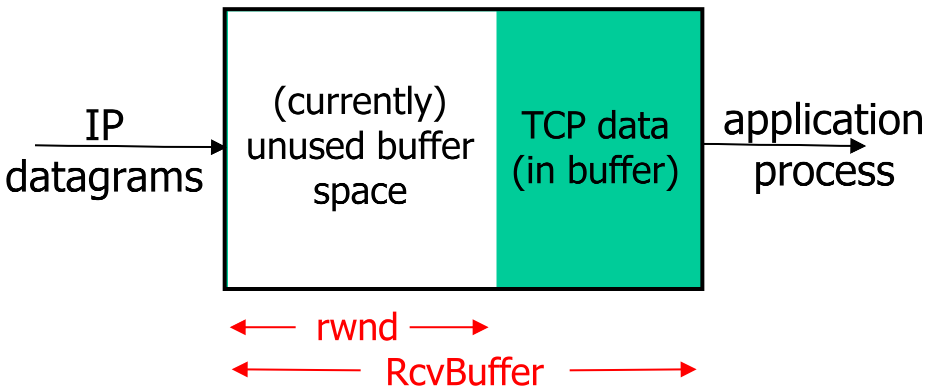
TCP flow control

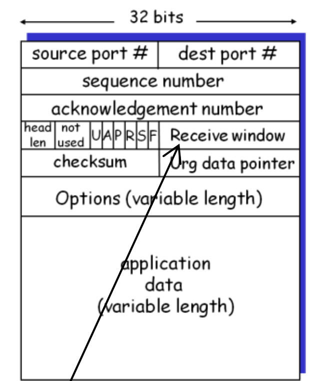
- receiver "advertises" free buffer space by including **`rwnd`** value in TCP header of receiver-to-sender segments
  - **`RcvBuffer`** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **`RcvBuffer`**
- sender limits amount of unacked ("in-flight") data to receiver's **`rwnd`** value
- guarantees receive buffer will not overflow



receiver-side buffering

TCP Flow control: how it works





(suppose TCP receiver discards out-of-order segments)

- unused buffer space:

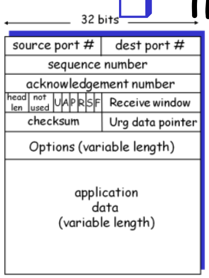
= **`rwnd`**

= **`RcvBuffer-[LastByteRcvd - LastByteRead]`**

- receiver: advertises unused buffer space by including rwnd value in segment header
- sender: limits # of unACKed bytes to `rwnd`
  - guarantees receiver's buffer doesn't overflow

TCP Connection Management

Recall: TCP sender, receiver establish "connection" before exchanging data segments



- initialize TCP variables:
  - seq. #s
  - buffers, flow control info (e.g. `RcvWindow`)
- *client:* connection initiator

```
Socket clientSocket = new
 Socket("hostname","port

 number");
```

- *server:* contacted by client

```
Socket connectionSocket =
 welcomeSocket.accept();
```

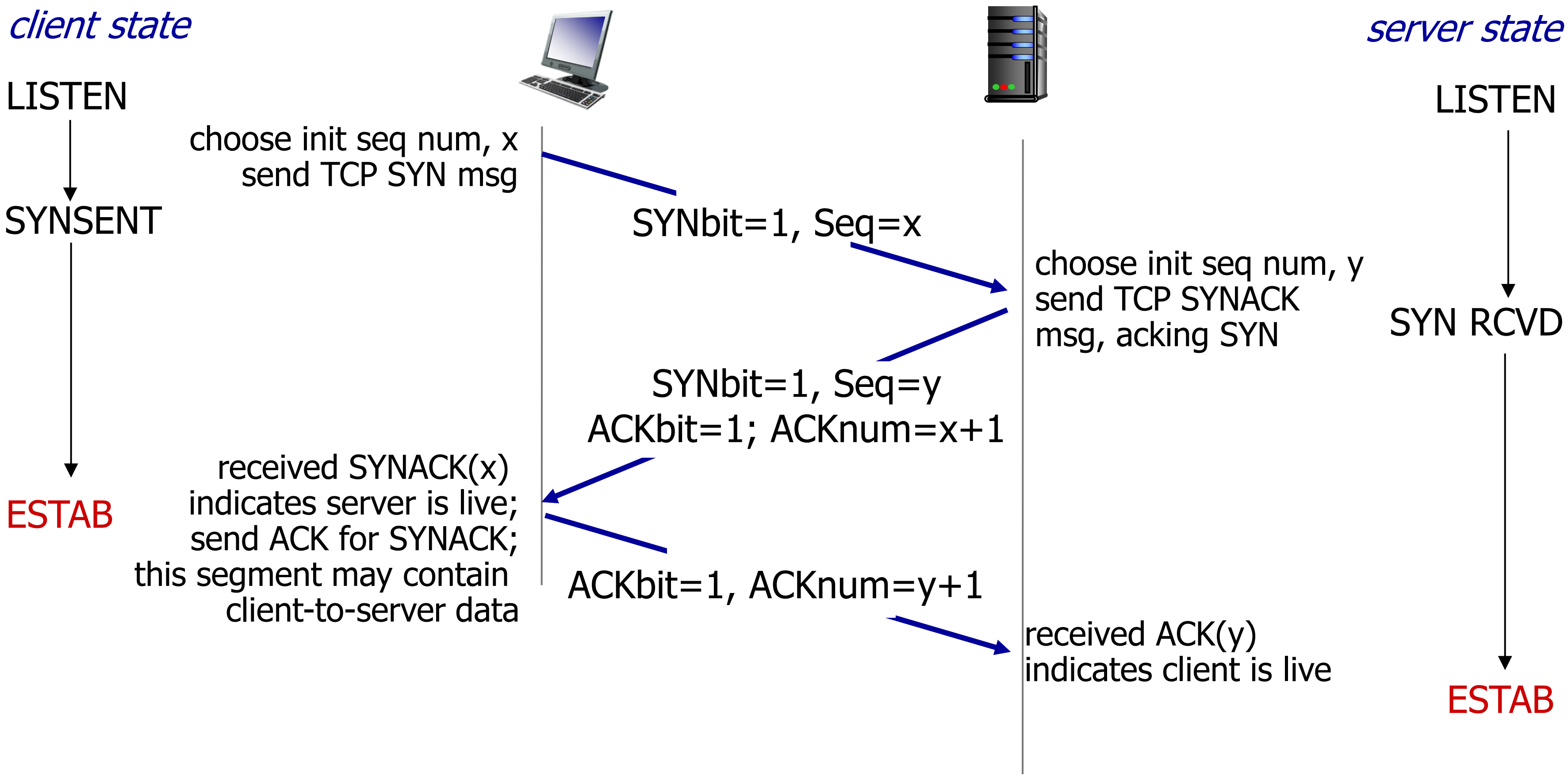
Three way handshake:

Step 1: client host sends TCP SYN segment to server
- specifies initial seq #
- no data

Step 2: server host receives SYN, replies with SYNACK segment
- server allocates buffers
- specifies server initial seq #

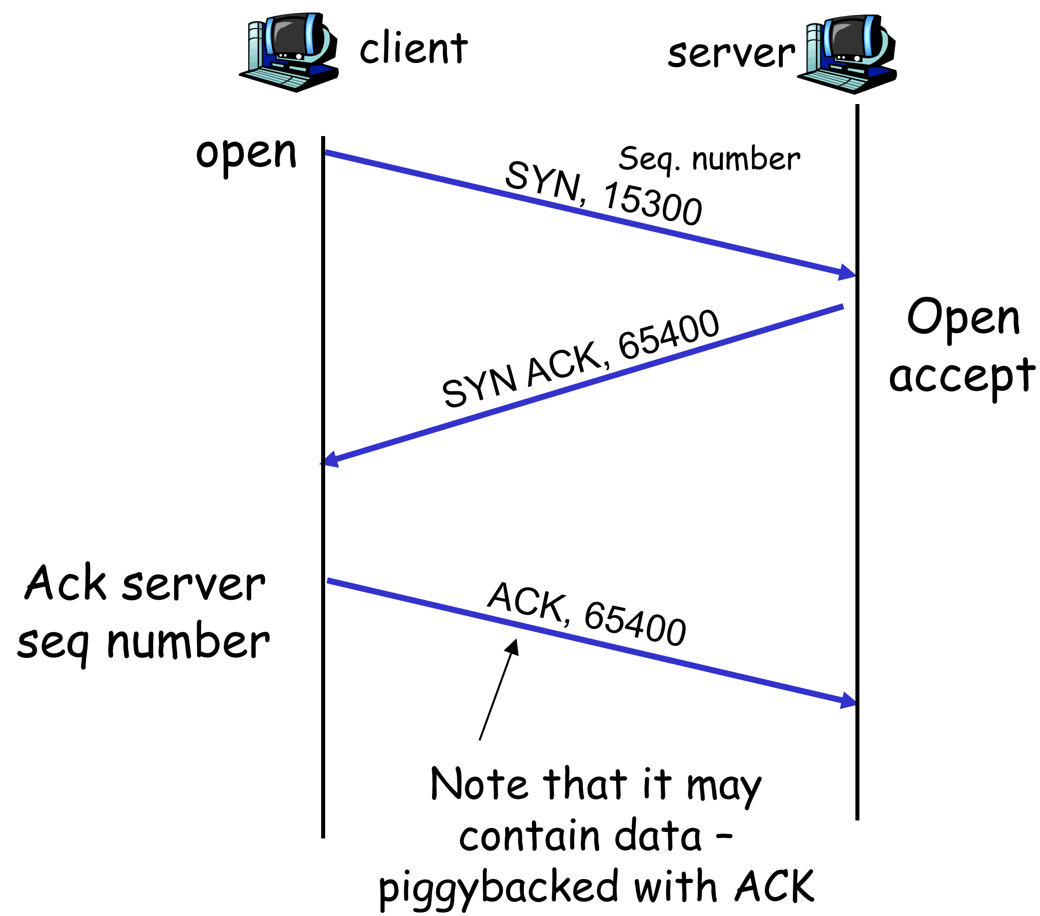
Step 3: client receives SYNACK, replies with ACK segment, which may contain data

TCP 3-way handshake

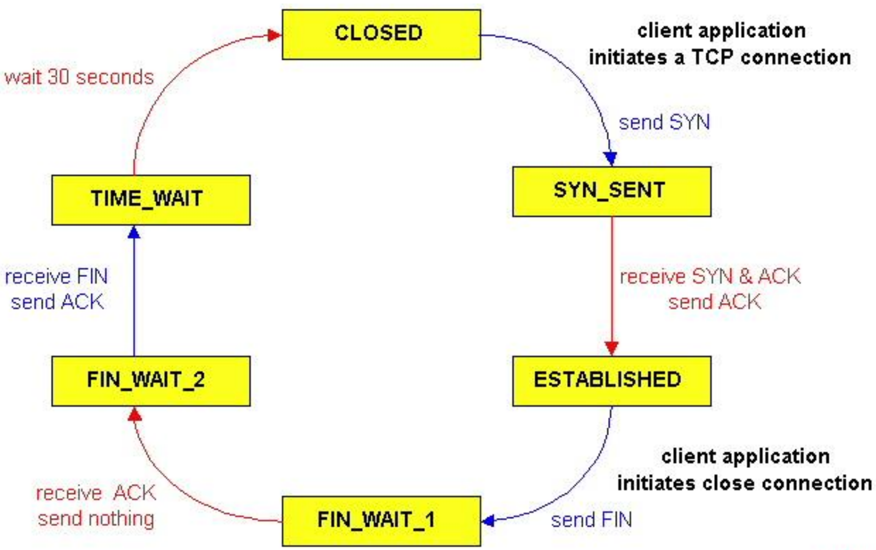


TCP Connection Management

- 3 way handshake

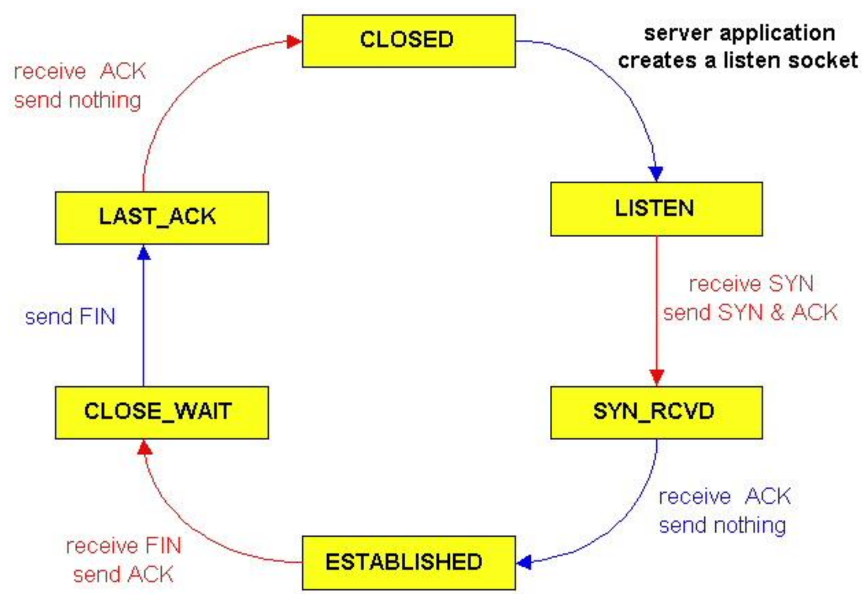


TCP Connection Management (cont)



TCP client lifecycle

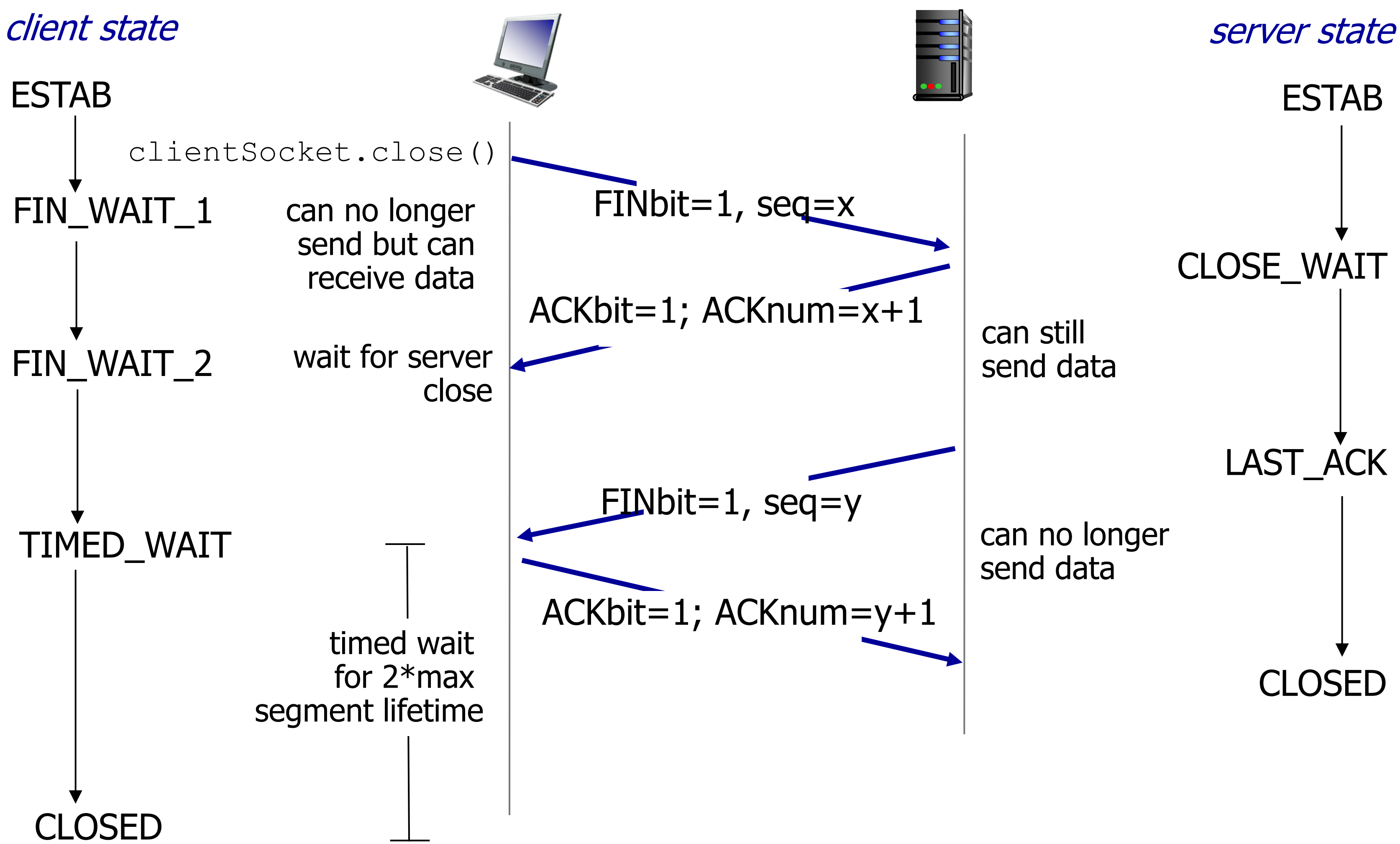
TCP server lifecycle



TCP: closing a connection

- client, server each close their side of connection
  - send TCP segment with FIN bit = 1
- respond to received FIN with ACK
  - on receiving FIN, ACK can be combined with own FIN
- simultaneous FIN exchanges can be handled

TCP: closing a connection



TCP Connection Management (cont.)

Closing a connection:

client closes socket:
    `clientSocket.close();`

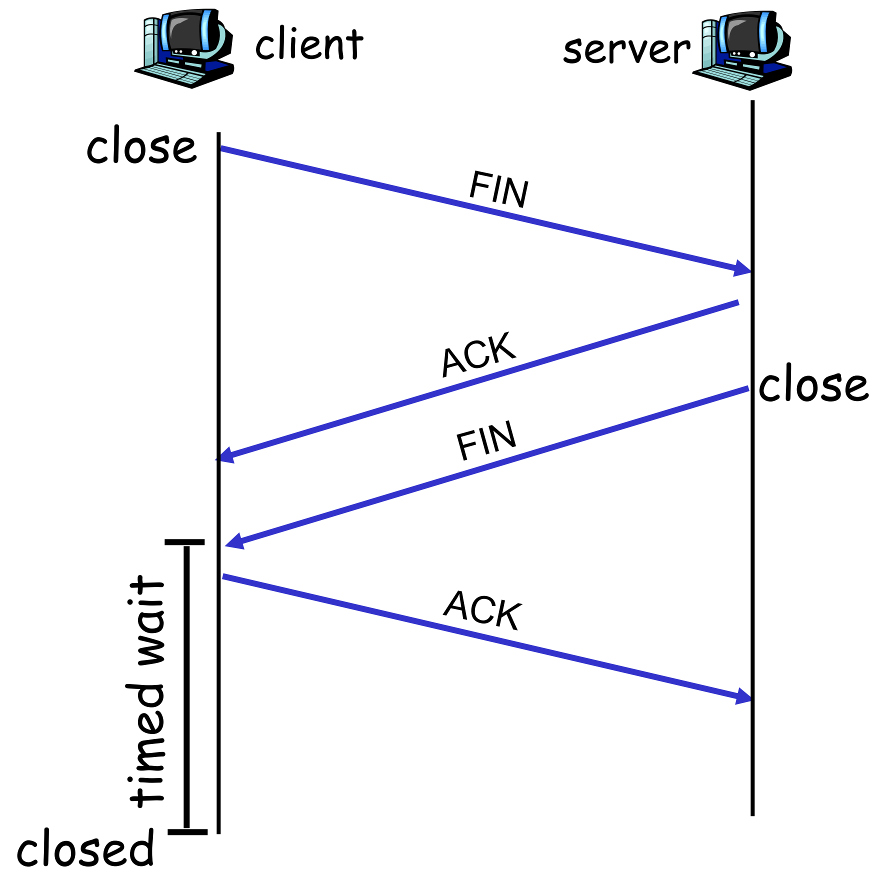
Step 1: client end system sends TCP FIN control segment to server

Step 2: server receives FIN, replies with ACK. Closes connection, sends FIN.

Step 3: client receives FIN, replies with ACK.

- Enters "timed wait" - will respond with ACK to received FINs

Step 4: server, receives ACK. Connection closed.



Principles of congestion control

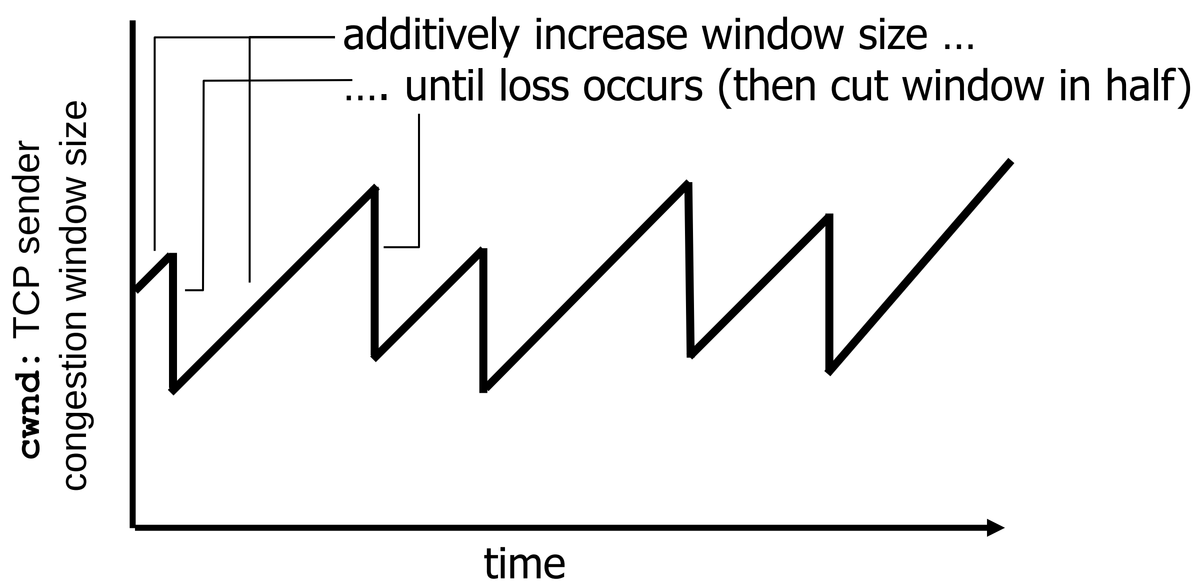
congestion:

- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

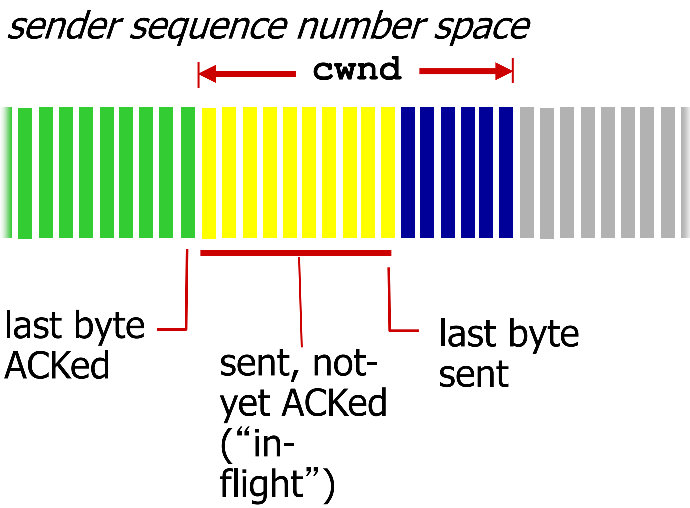
TCP congestion control: additive increase multiplicative decrease

- *approach:* sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
  - *additive increase:* increase **`cwnd`** by 1 MSS every RTT until loss detected
  - *multiplicative decrease:* cut **`cwnd`** in half after loss

AIMD saw tooth behavior: probing for bandwidth



TCP Congestion Control: details



- sender limits transmission:

$$\texttt{LastByteSent-LastByteAcked} \leq \texttt{cwnd}$$

- **`cwnd`** is dynamic, function of perceived network congestion

TCP sending rate:

- *roughly:* send cwnd bytes, wait RTT for ACKS, then send more bytes

$$\text{rate} \approx \frac{\text{cwnd}}{\text{RTT}} \text{ bytes/sec}$$

TCP Congestion Control

- TCP has a mechanism for congestion control. The mechanism is implemented at the sender

- The window size at the sender is set as follows:

•Send Window = MIN (flow control window, congestion window)

where

- flow control window is advertised by the receiver
- congestion window is adjusted based on feedback from the network

TCP Congestion Control

- The sender has two additional parameters:
  - **Congestion Window** (**cwnd**)
    Initial value is 1 MSS (=maximum segment size) counted as bytes
  - **Slow-start threshold Value** (**ssthresh)**
    Initial value is the advertised window size)

- Congestion control works in two modes:
  - **slow start** (cwnd < ssthresh)
  - **congestion avoidance** (cwnd >= ssthresh)

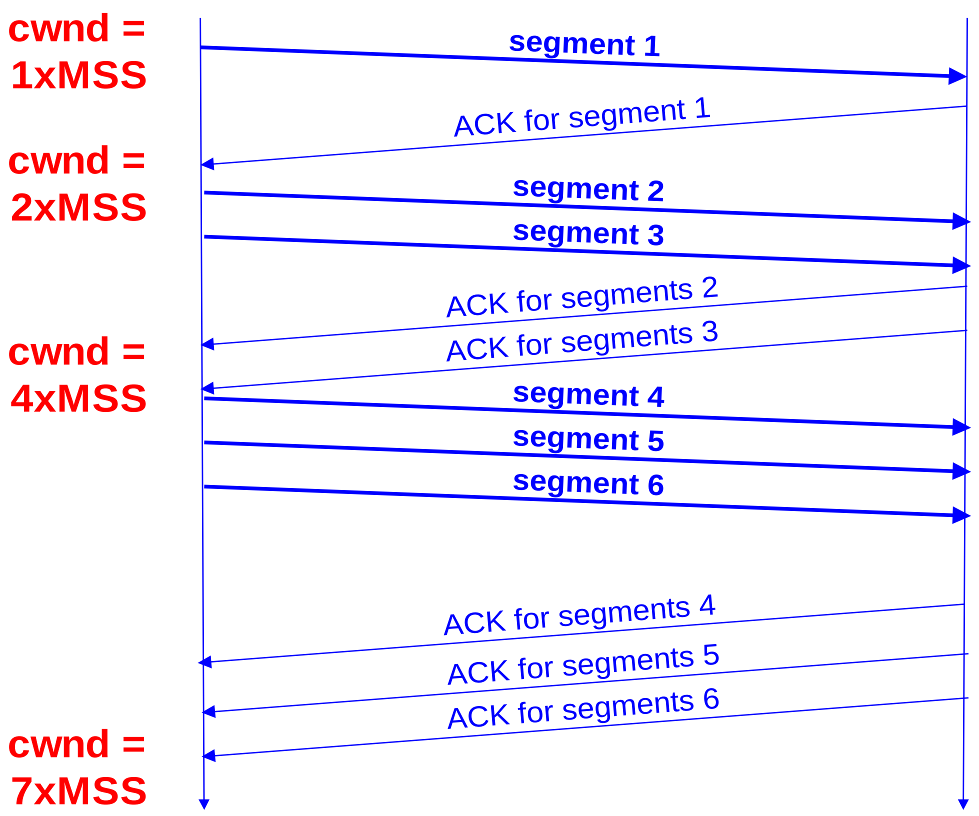
Slow Start

- Initial value:
  - **cwnd = 1 segment**
- **Note: cwnd is actually measured in bytes:**
  1 segment = MSS bytes
- Each time an ACK is received, the congestion window is increased by MSS bytes.
  - **cwnd = cwnd + MSS**
  - If an ACK acknowledges two segments, cwnd is still increased by only 1 segment.
  - Even if ACK acknowledges a segment that is smaller than MSS bytes long, cwnd is increased by MSS.

- Does Slow Start increment slowly? Not really.
  In fact, the increase of cwnd can be exponential

Slow Start Example

- The congestion window size grows very rapidly
  - For every ACK, we increase cwnd by 1 irrespective of the number of segments ACK'ed
- TCP slows down the increase of *cwnd* when
  cwnd > ssthresh



Congestion Avoidance

- Congestion avoidance phase is started if cwnd has reached the slow-start threshold value

- If cwnd >= ssthresh then each time an ACK is received, increment cwnd as follows:
  - cwnd = cwnd + MSS(MSS/ cwnd)

- So *cwnd* is increased by one segment (=MSS bytes) only if all segments have been acknowledged.

Slow Start / Congestion Avoidance

- Here we give a more accurate version than in our earlier discussion of Slow Start:

If *cwnd* < *ssthresh* **then**

Each time an Ack is received:

cwnd = cwnd *+ MSS*

else ***/**** *cwnd >= ssthresh */*

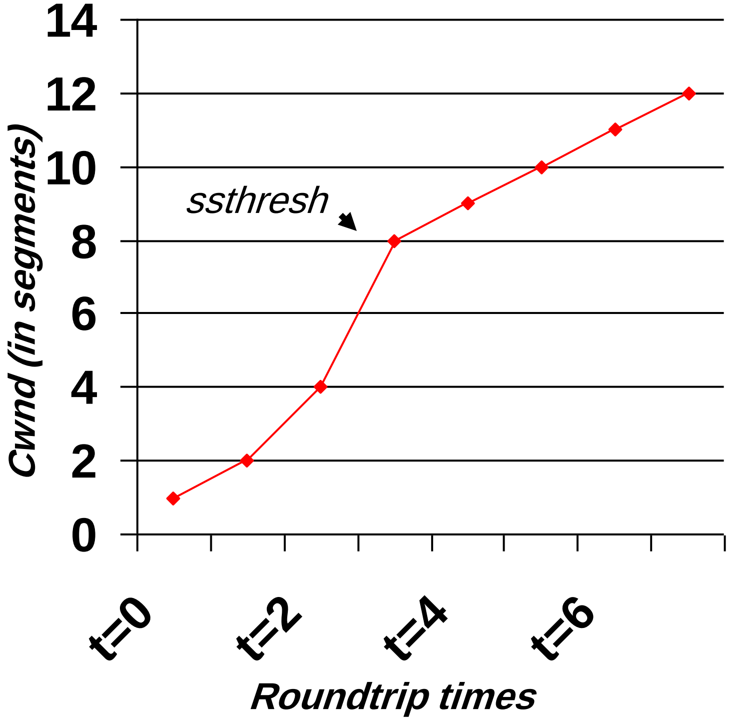
Each time an Ack is received :

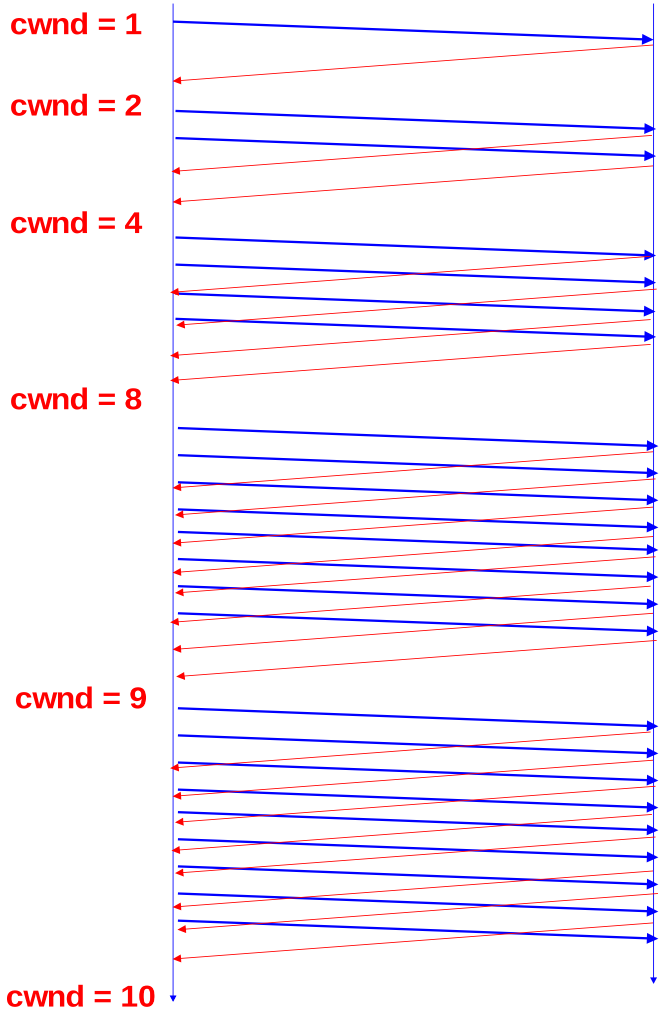
cwnd = cwnd + MSS. MSS / cwnd

endif

Example of
Slow Start/Congestion Avoidance

Assume that *ssthresh = 8*





Responses to Congestion

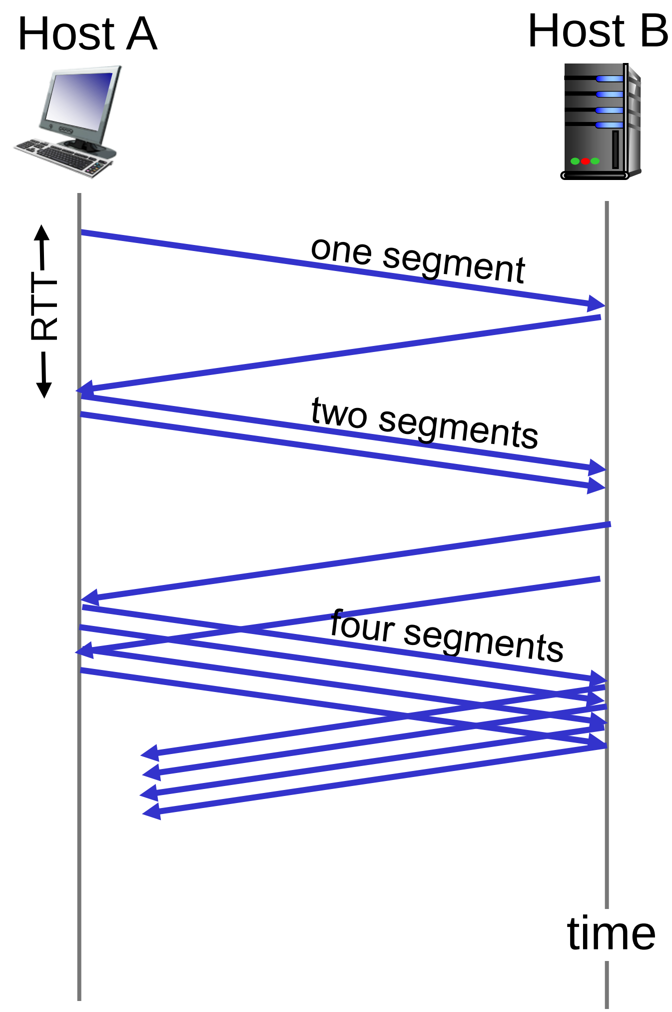
- Most often, a packet loss in a network is due to an overflow at a congested router (rather than due to a transmission error)

- So, TCP assumes there is congestion if it detects a packet loss
- A TCP sender can detect lost packets via:
  - Timeout of a retransmission timer
  - Receipt of a duplicate ACK

- When TCP assumes that a packet loss is caused by congestion it reduces the size of the sending window

TCP Slow Start

- when connection begins, increase rate exponentially until first loss event:
  - initially **cwnd** = 1 MSS
  - double **cwnd** every RTT
  - done by incrementing **cwnd** for every ACK received
- *summary:* initial rate is slow but ramps up exponentially fast



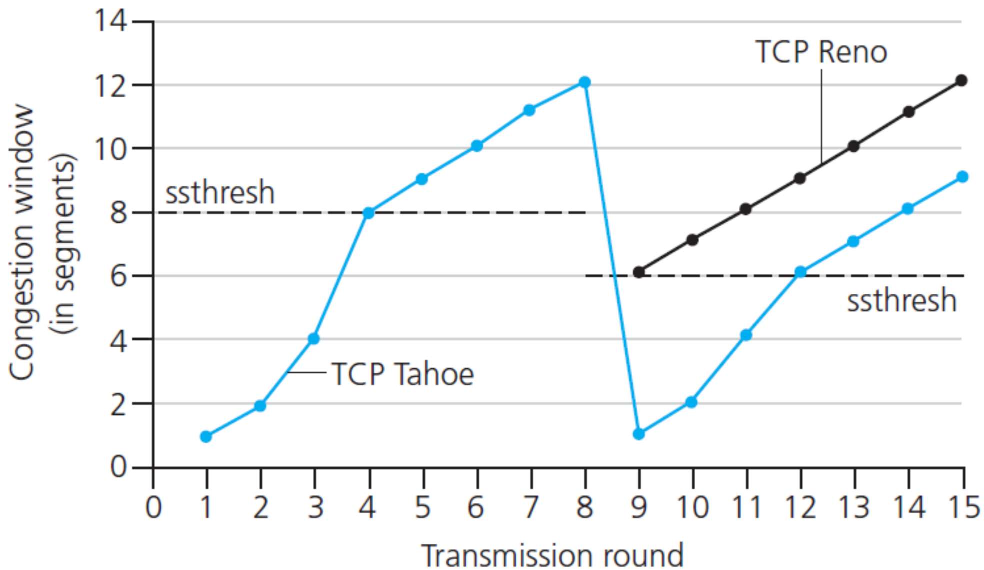
TCP: detecting, reacting to loss

- loss indicated by timeout:
  - **`cwnd`** set to 1 MSS;
  - window then grows exponentially (as in slow start) to threshold, then grows linearly
- loss indicated by 3 duplicate ACKs: TCP RENO
  - dup ACKs indicate network capable of delivering some segments
  - **`cwnd`** is cut in half window then grows linearly
- TCP Tahoe always sets **`cwnd`** to 1 (timeout or 3 duplicate acks)

TCP: switching from slow start to CA

Q: when should the exponential increase switch to linear?

A: when **cwnd** gets to 1/2 of its value before timeout.



Implementation:

- variable **ssthresh**
- on loss event, **ssthresh** is set to 1/2 of **cwnd** just before loss event