Transport Layer

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

Layered architecture

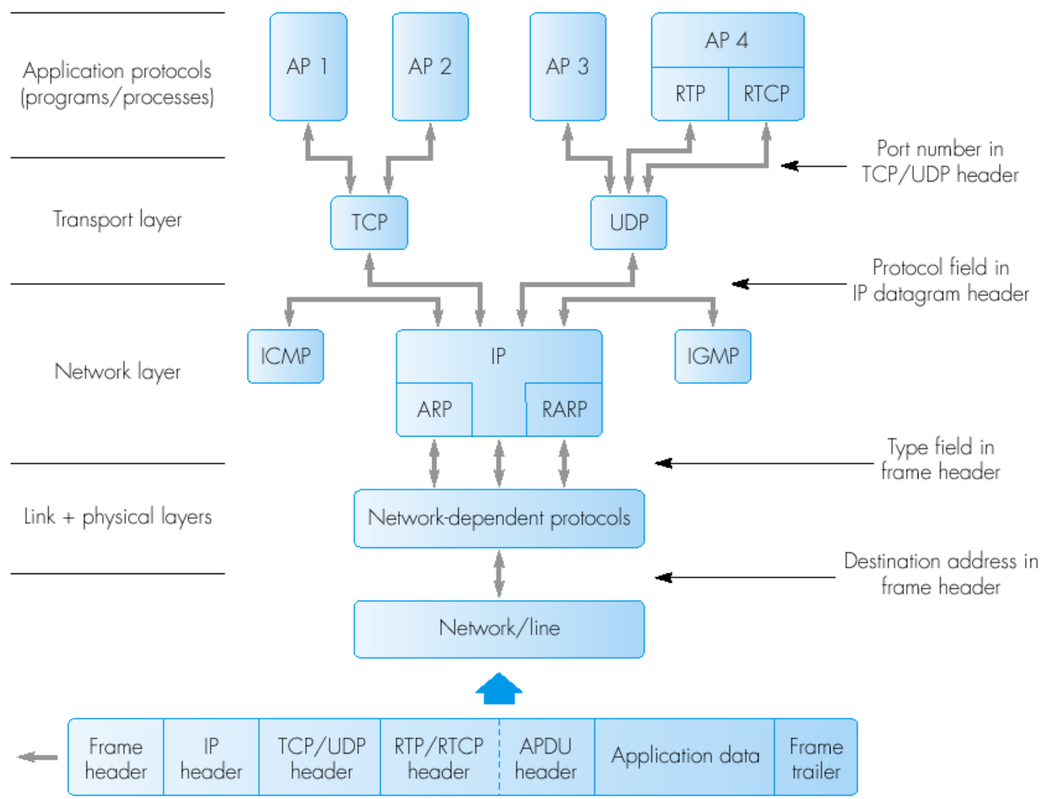
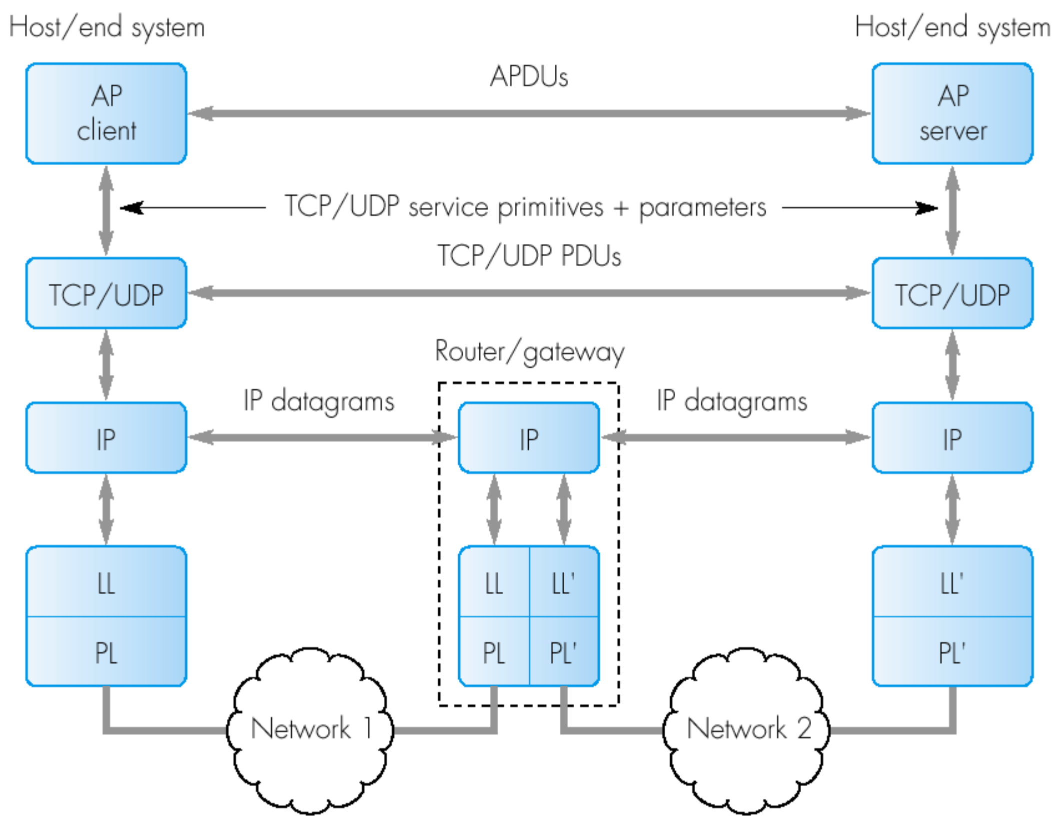


Figure 12.2 TCP/IP protocol suite interlayer communications.



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

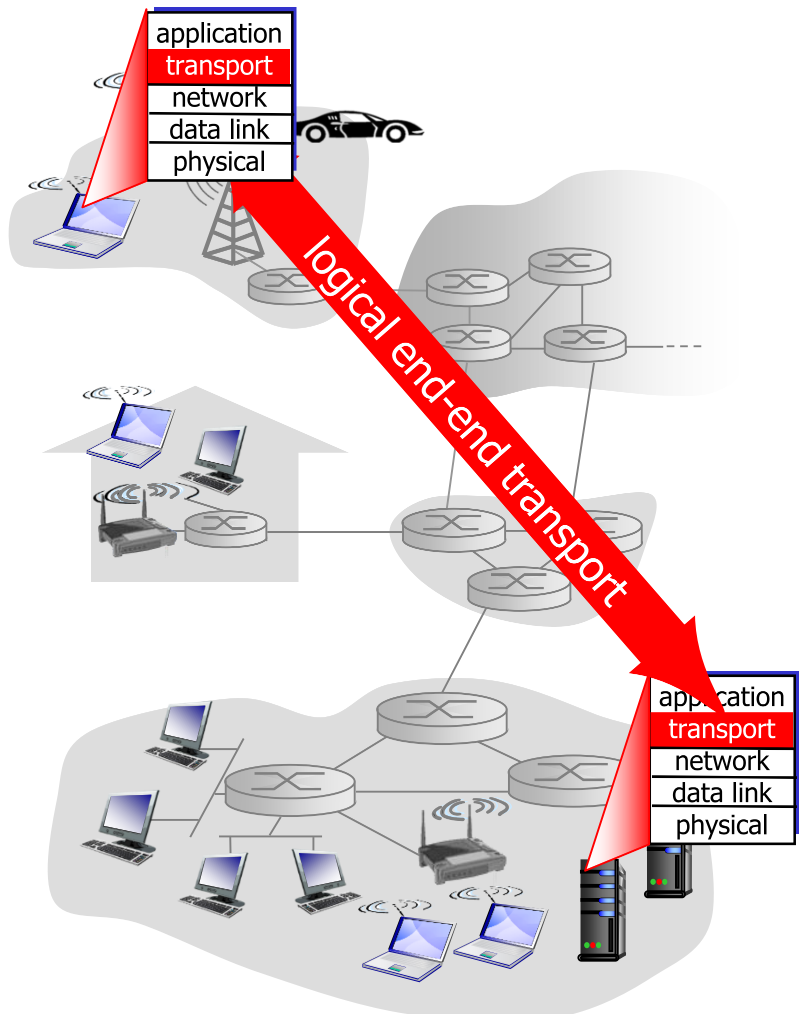
3.4 connection-oriented transport: TCP

- 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

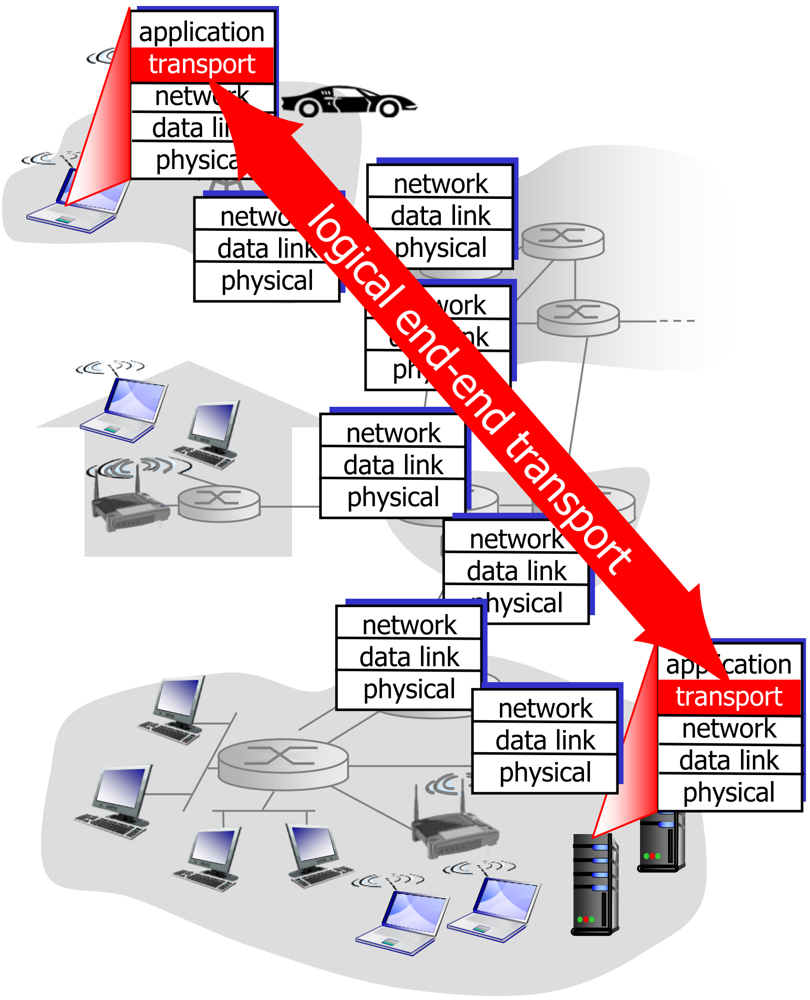
household analogy:

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

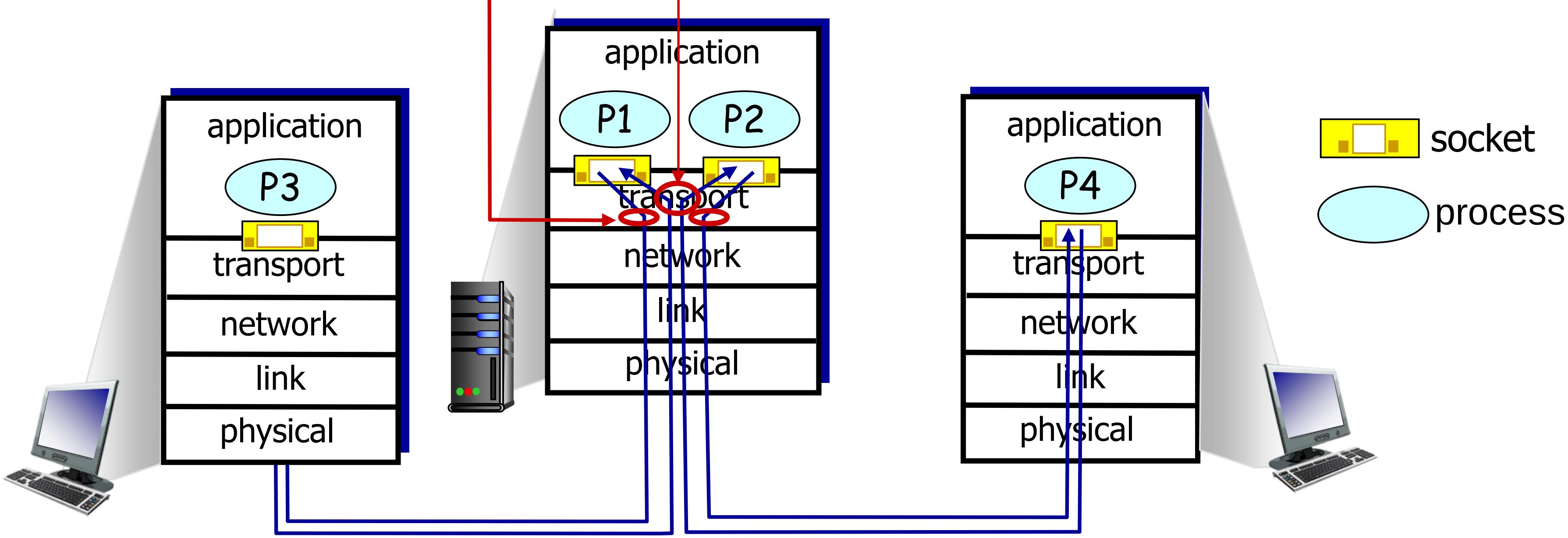
3.6 principles of congestion control

3.7 TCP congestion control

Multiplexing/demultiplexing

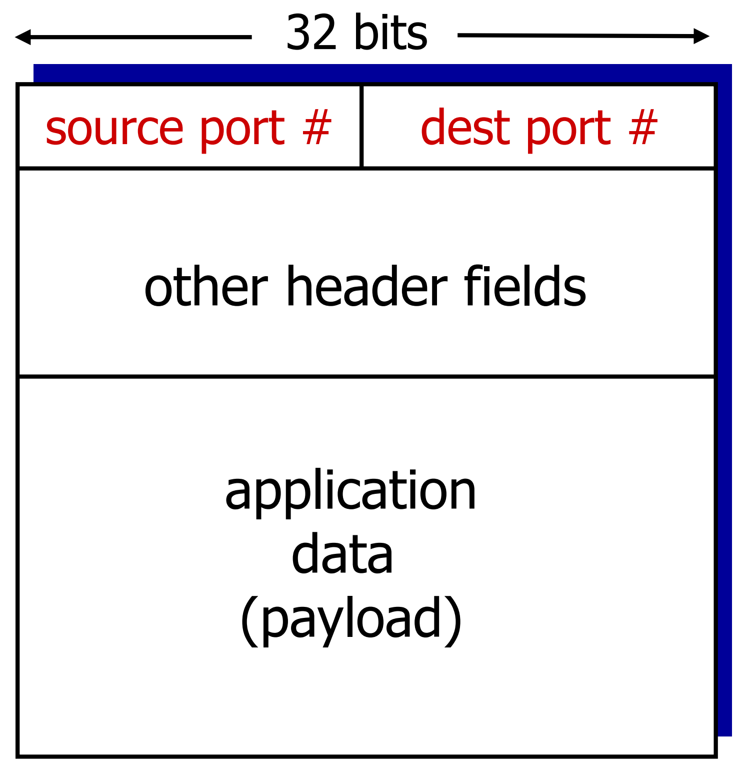
multiplexing at sender:
handle data from multiple sockets, add transport header (later used for demultiplexing)

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
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3.6 principles of congestion control

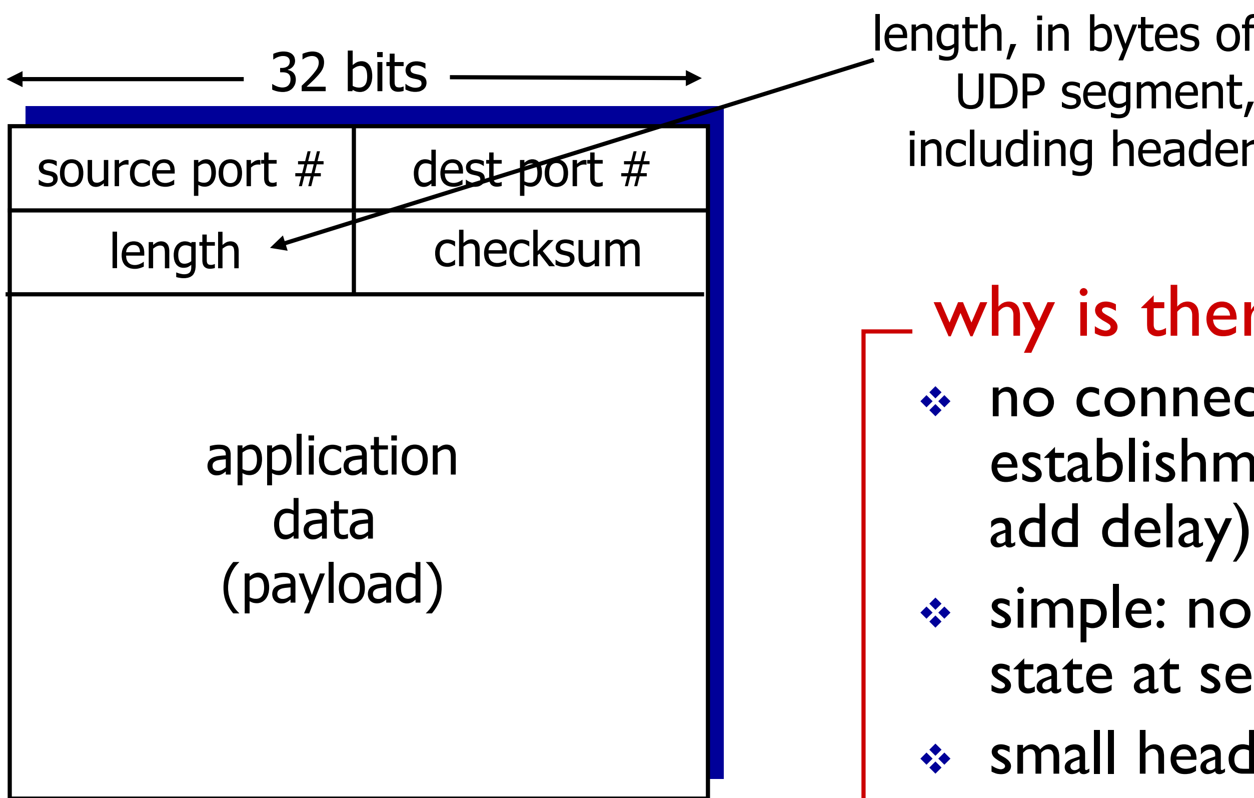
3.7 TCP congestion control

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
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- connection management

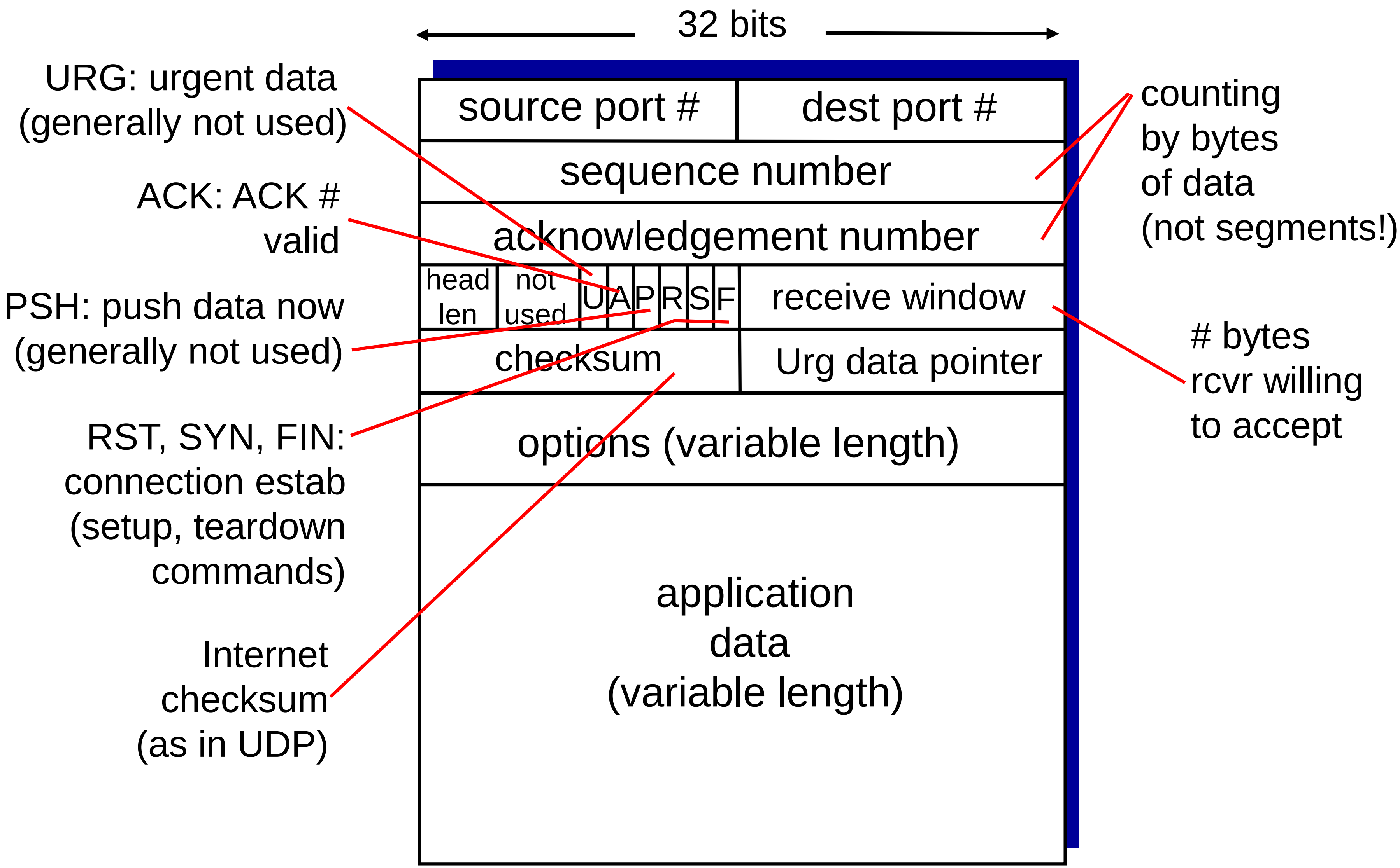
3.6 principles of congestion control

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:

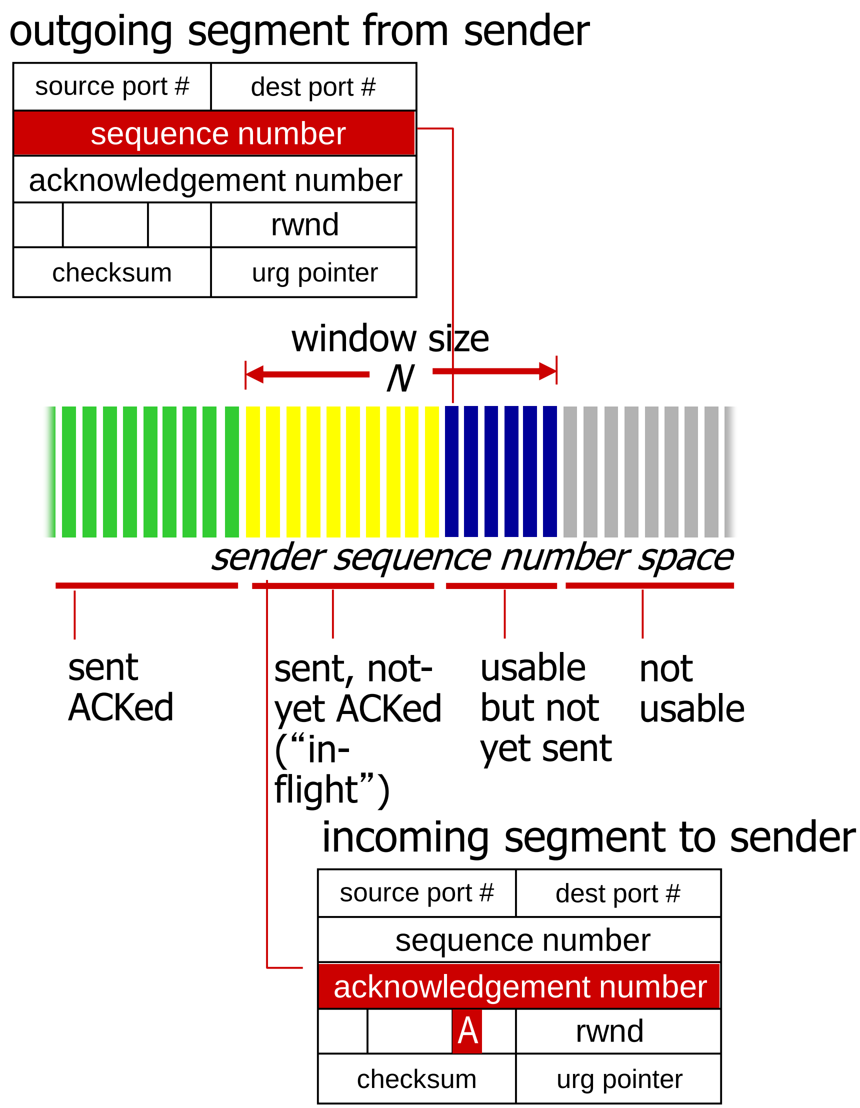
- byte stream "number" of first byte in segment's data

acknowledgements:

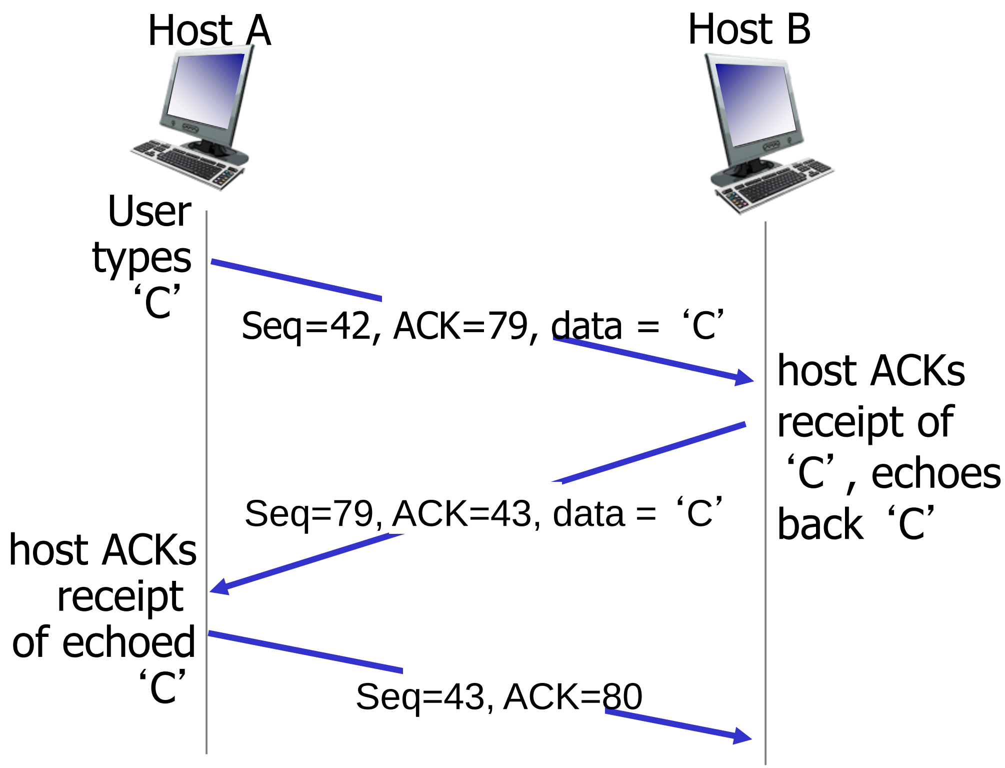
- seq # of next byte expected from other side
- cumulative ACK

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

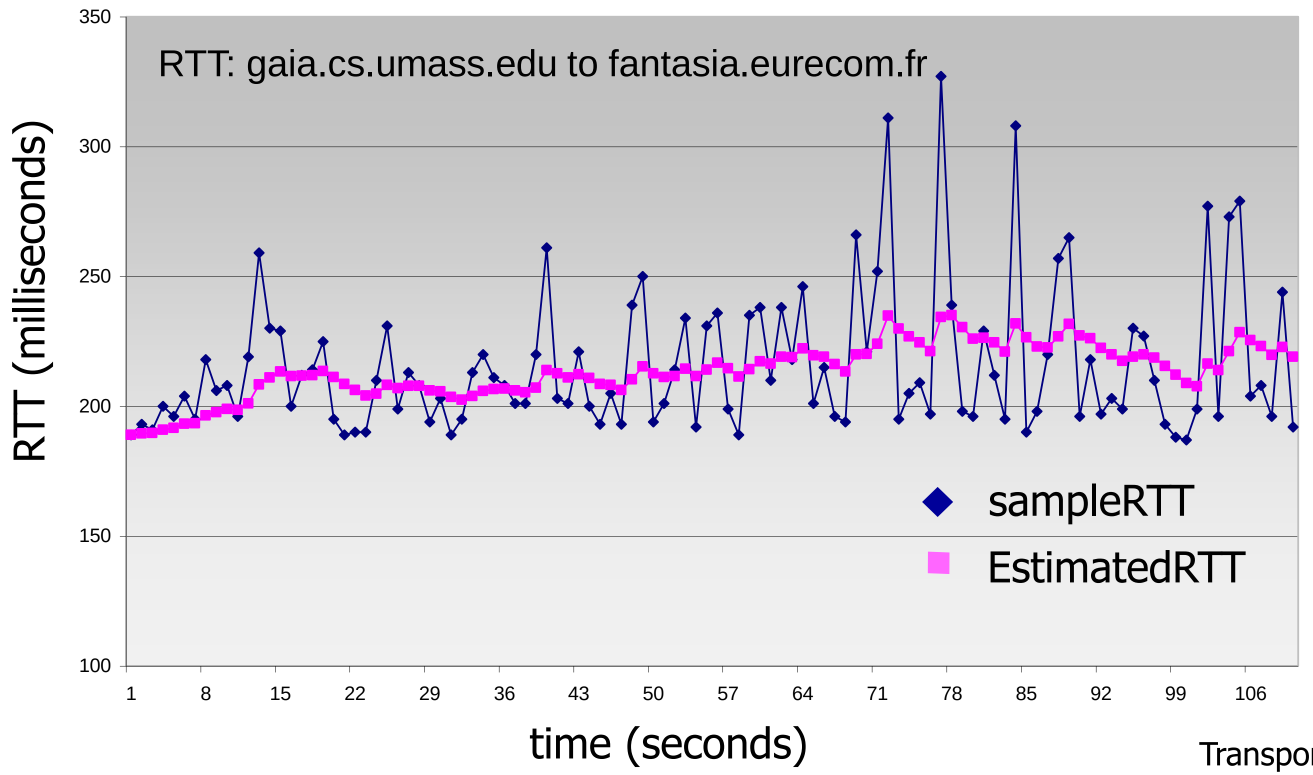
Q: how to estimate RTT?

- **`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
```

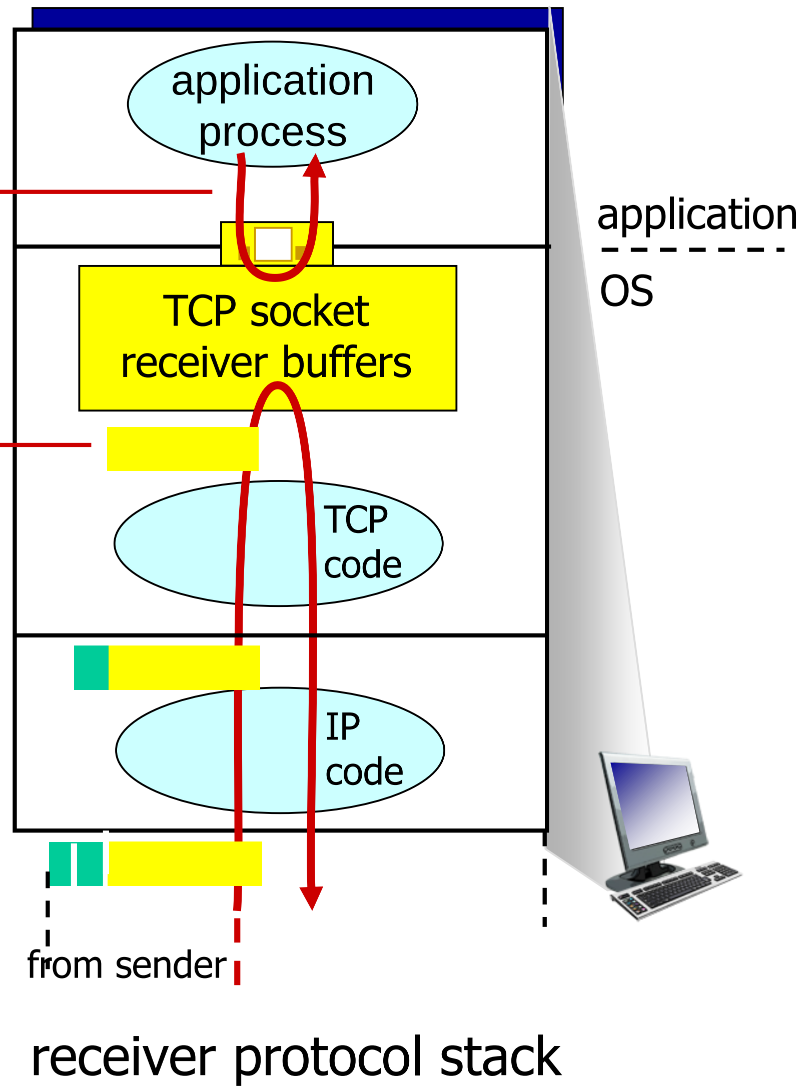
estimated RTT

"safety margin"

TCP flow control

application may remove data from TCP socket buffers ….

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

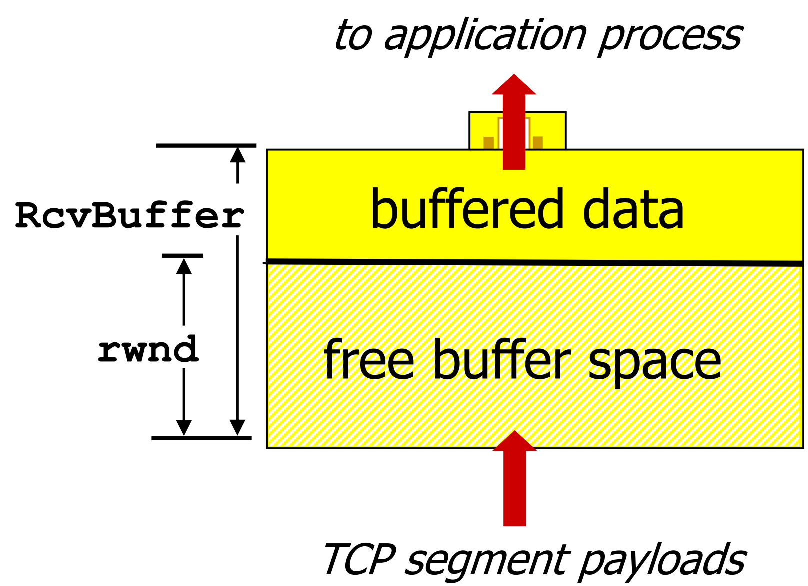


flow control

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

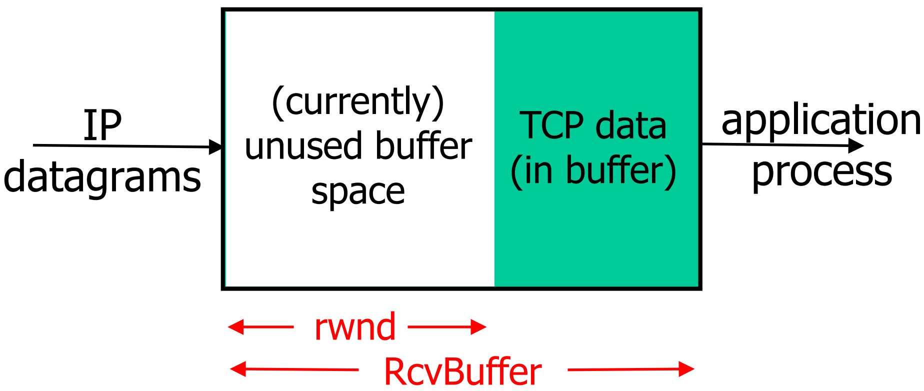
TCP flow control

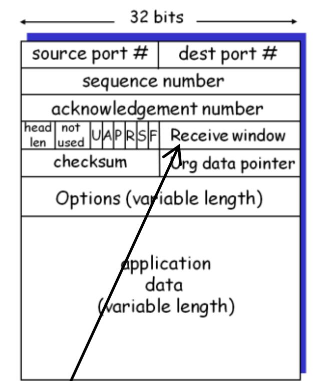
- receiver "advertises" free buffer space by including **rwnd** value in TCP header of receiver-to-sender segments
  - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **RcvBuffer**
- sender limits amount of unacked ("in-flight") data to 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`**

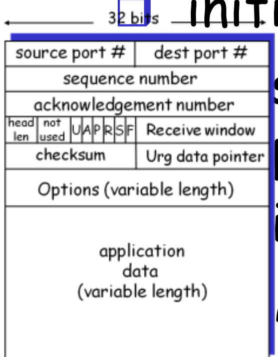
= **`RcvBuffer-[LastByteRcvd - LastByteRead]`**

- 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**)
- *nt*: connection initiator

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

 number");
```

- *server*: contacted by client

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

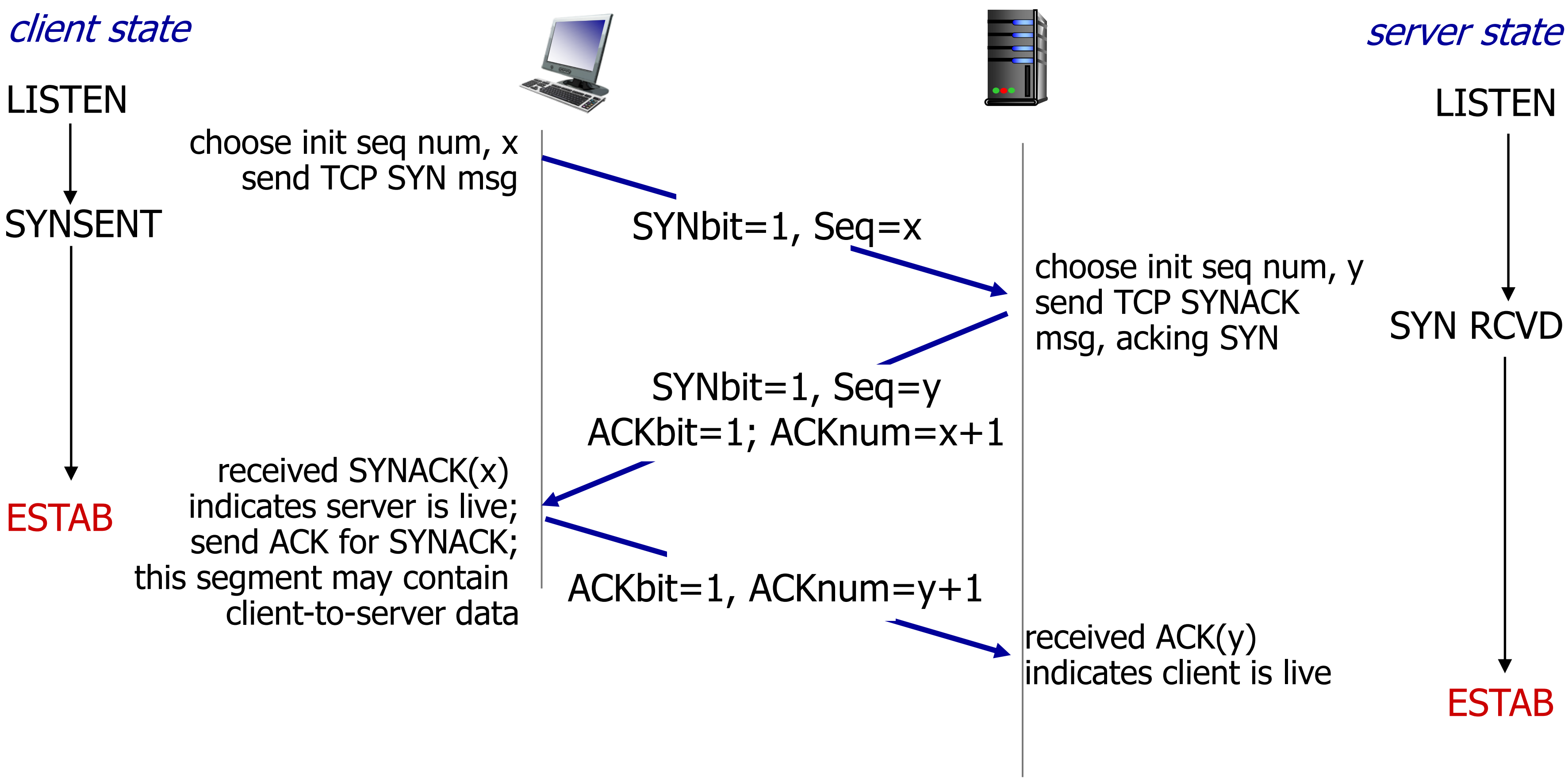
Three way handshake:

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 #

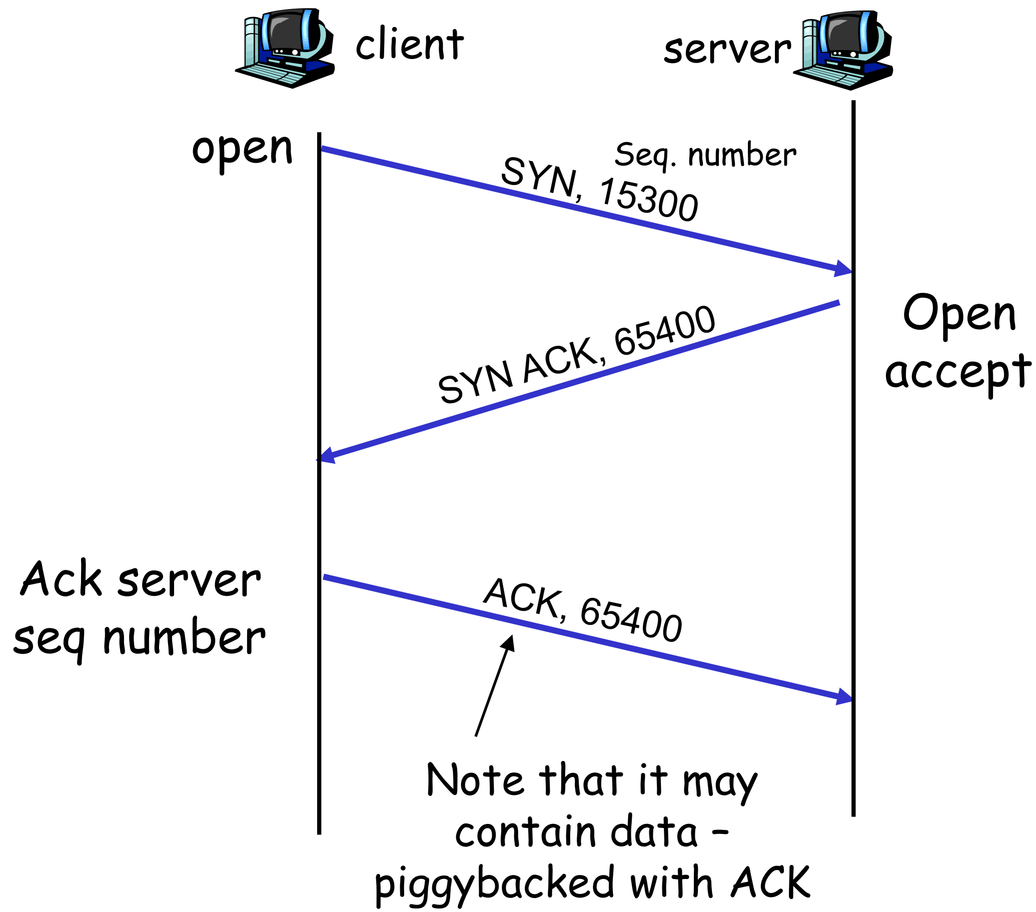
Step 3: client receives SYNACK, replies with ACK segment, which may contain data

TCP 3-way handshake

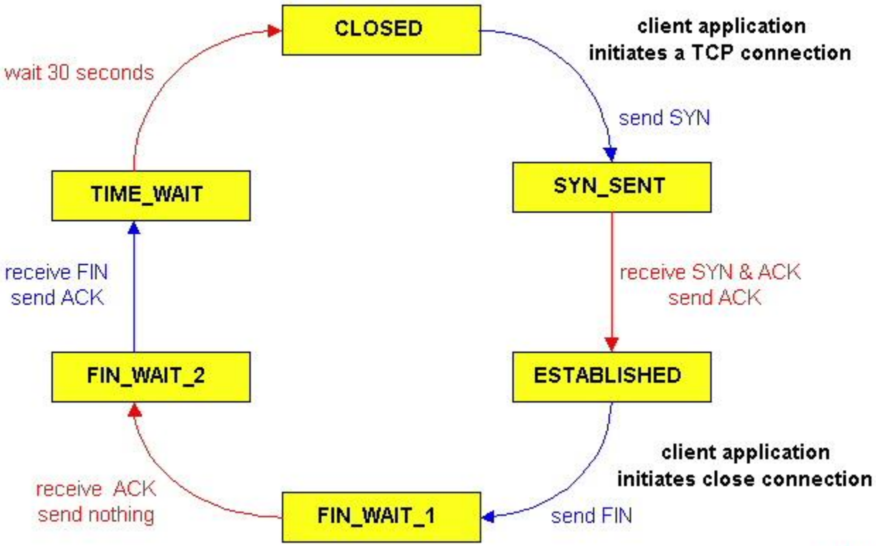


TCP Connection Management

- 3 way handshake

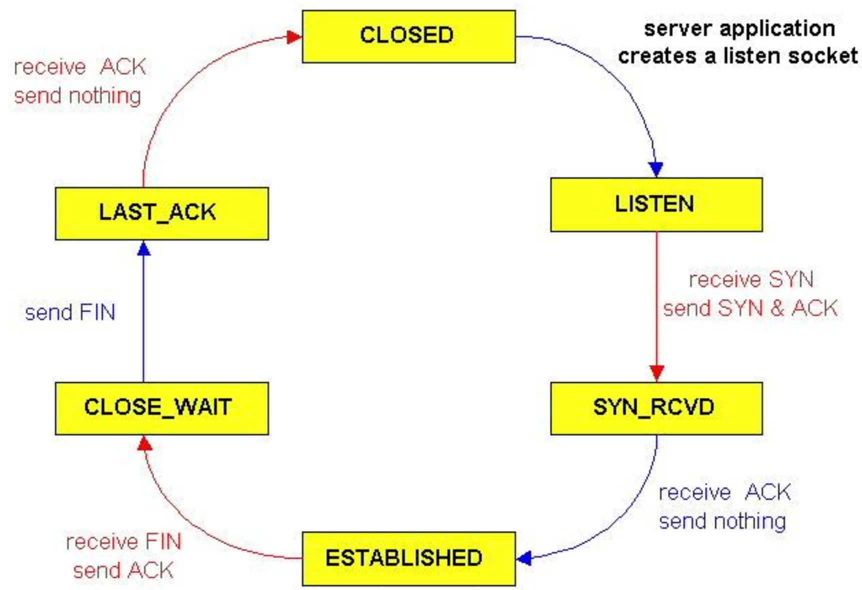


TCP Connection Management (cont)



TCP client lifecycle

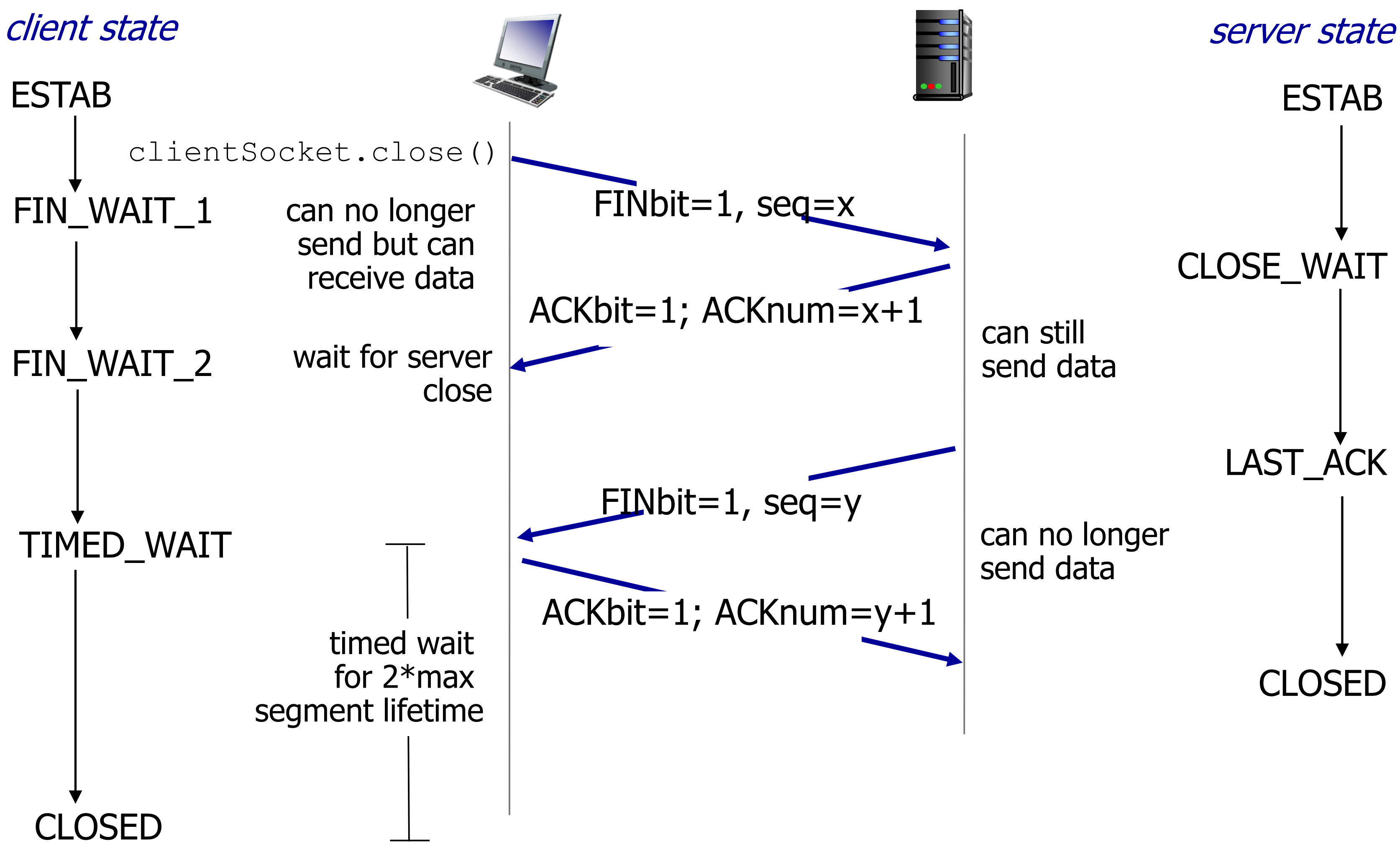
TCP server lifecycle



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();
```

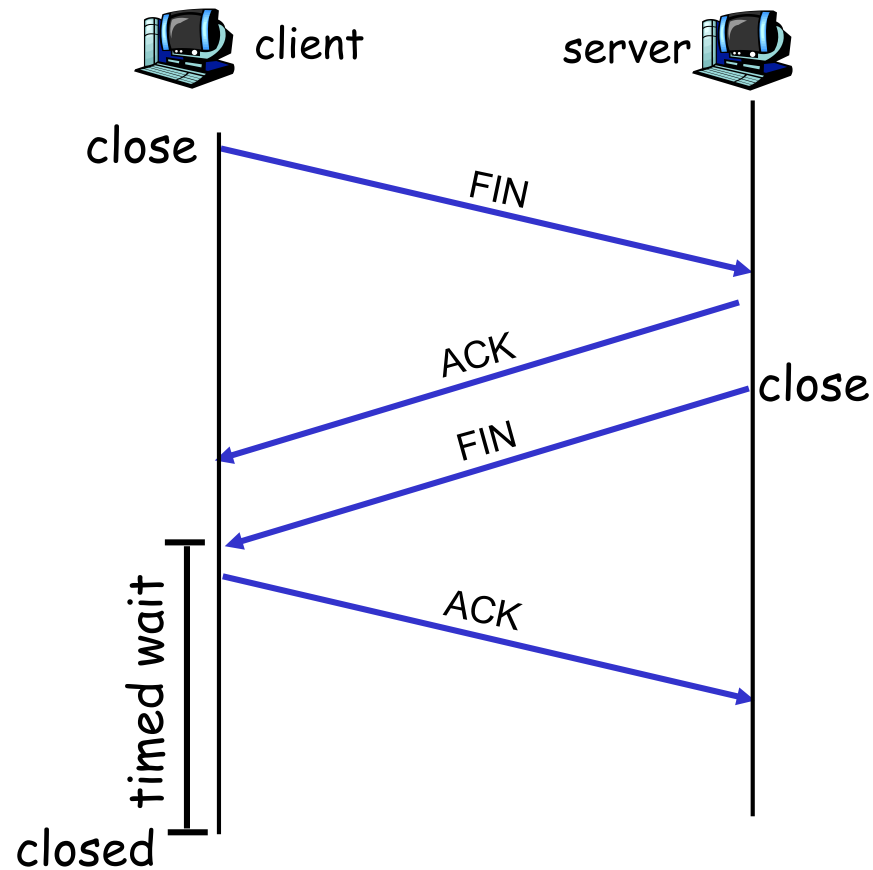
Step 1: client end system sends TCP FIN control segment to server

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:

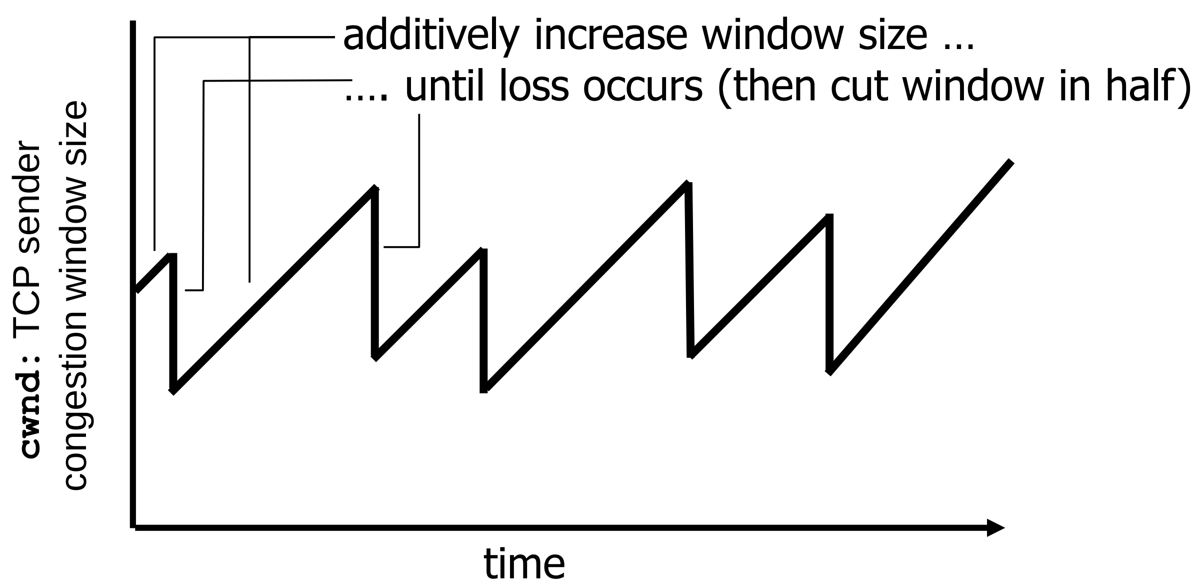
- informally: “too many sources sending too much data too fast for *network* to handle”
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

TCP CONGESTION CONTROL

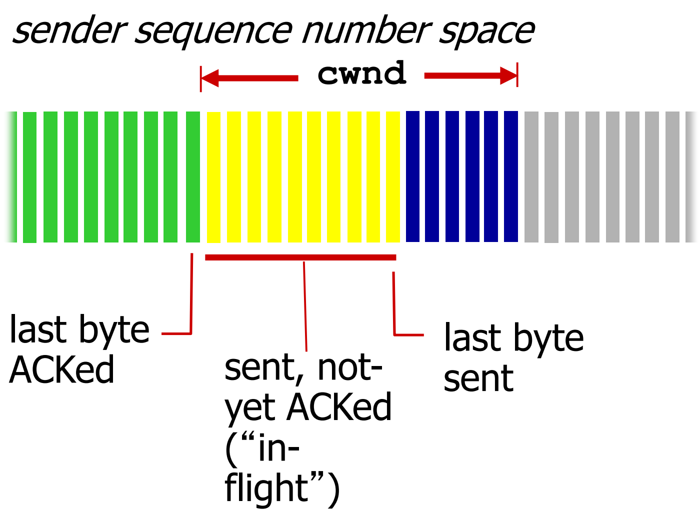
TCP congestion control: additive increase multiplicative decrease

- *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)

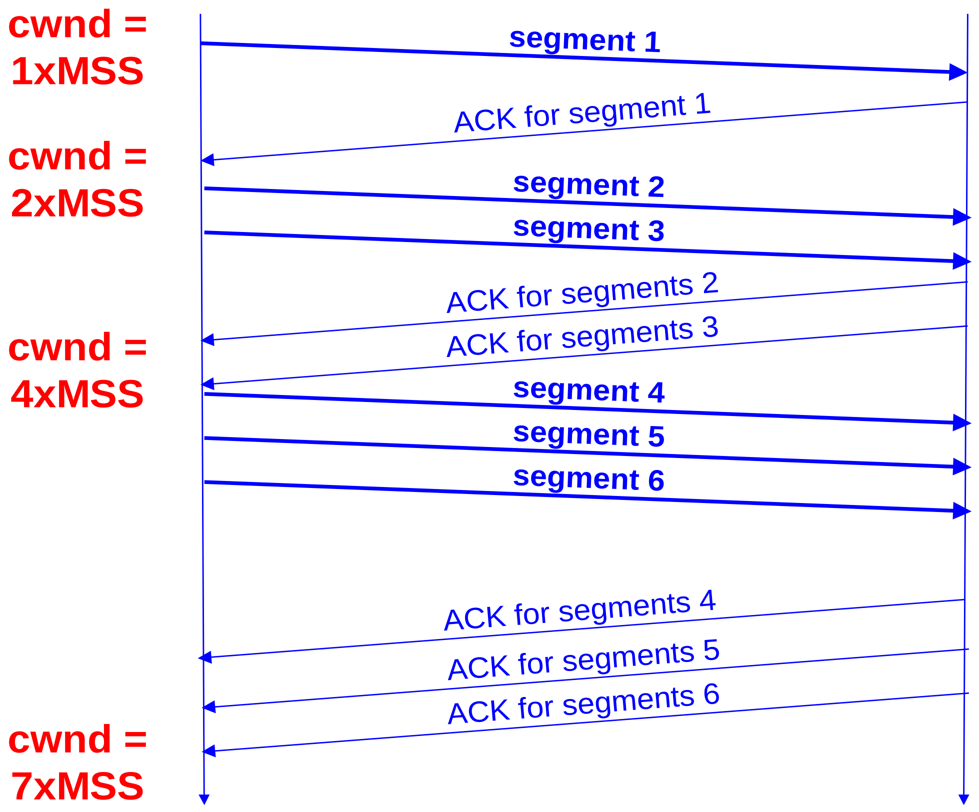
Slow Start

- 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* */*

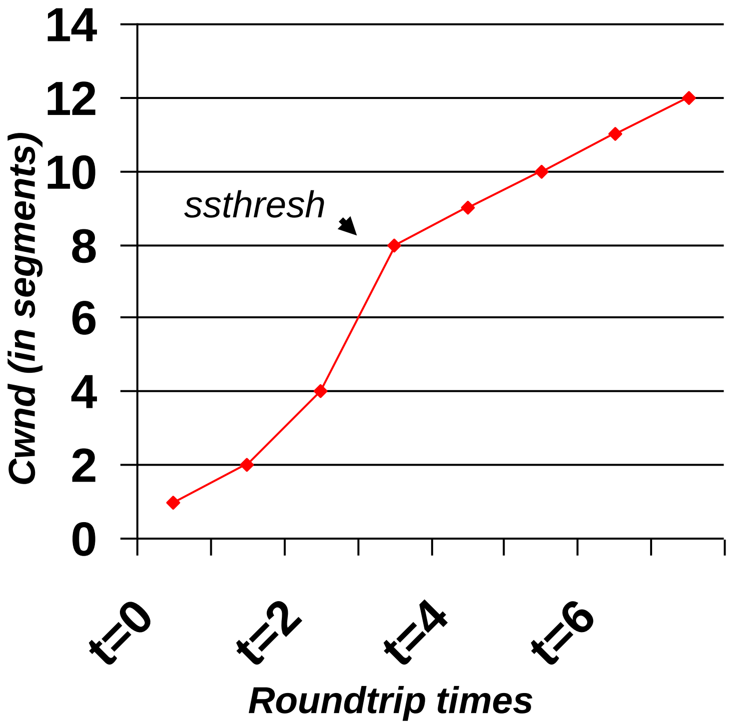
Each time an Ack is received :

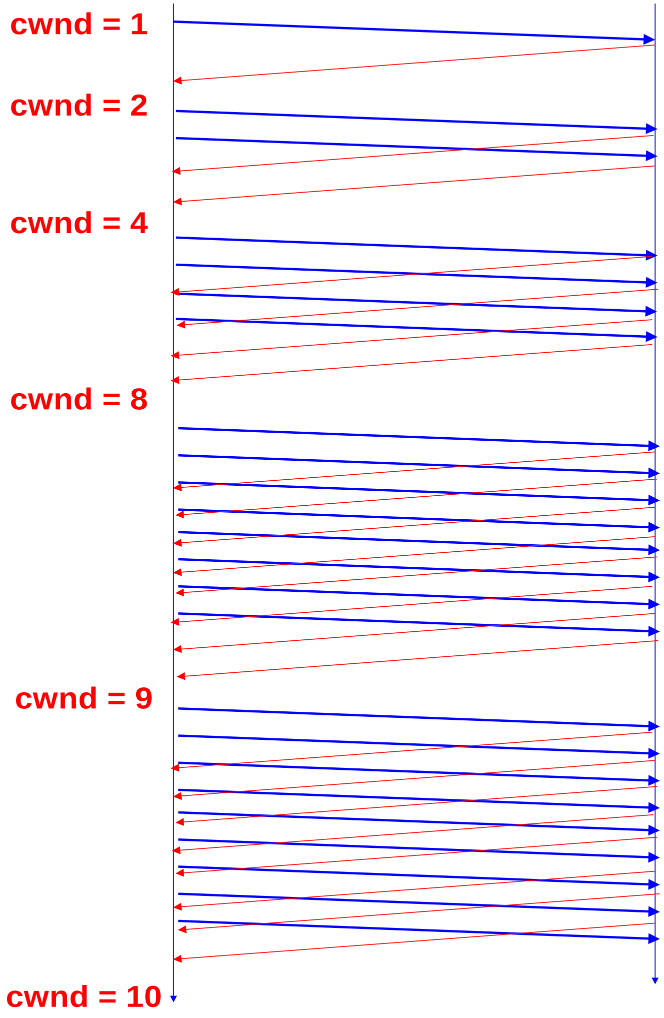
cwnd = cwnd + MSS. MSS / cwnd

endif

Example of Slow Start/Congestion Avoidance

Assume that *ssthresh = 8*





Responses to Congestion

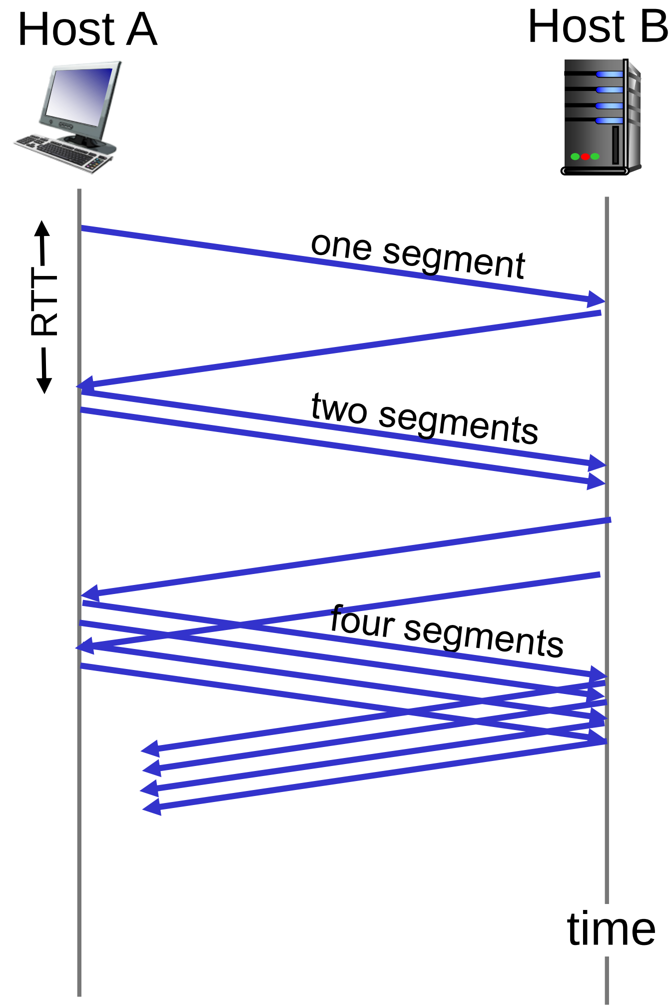
- Most often, a packet loss in a network is due to an overflow at a congested router (rather than due to a transmission error)

- 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



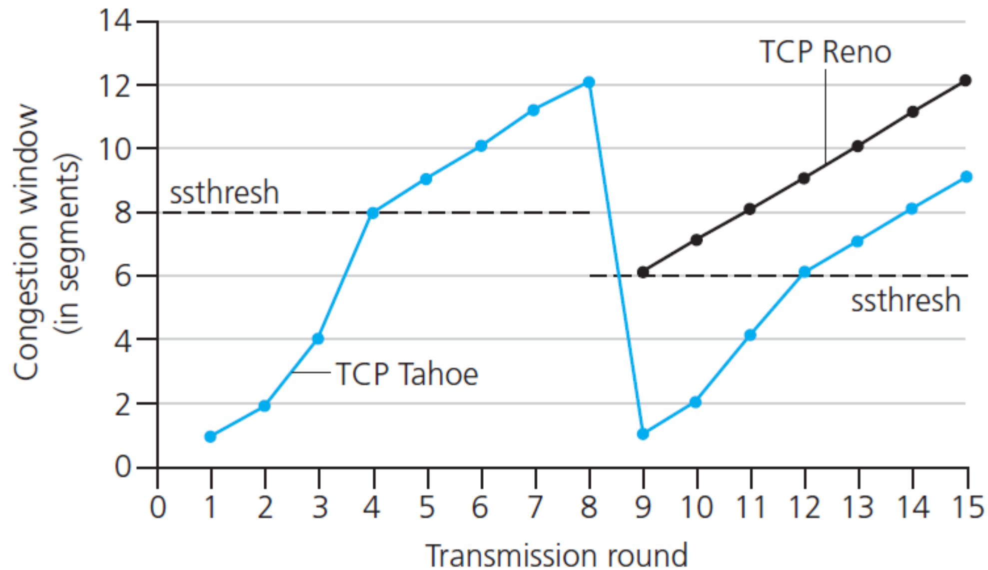
TCP: detecting, reacting to loss

- 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

NAT: network address translation