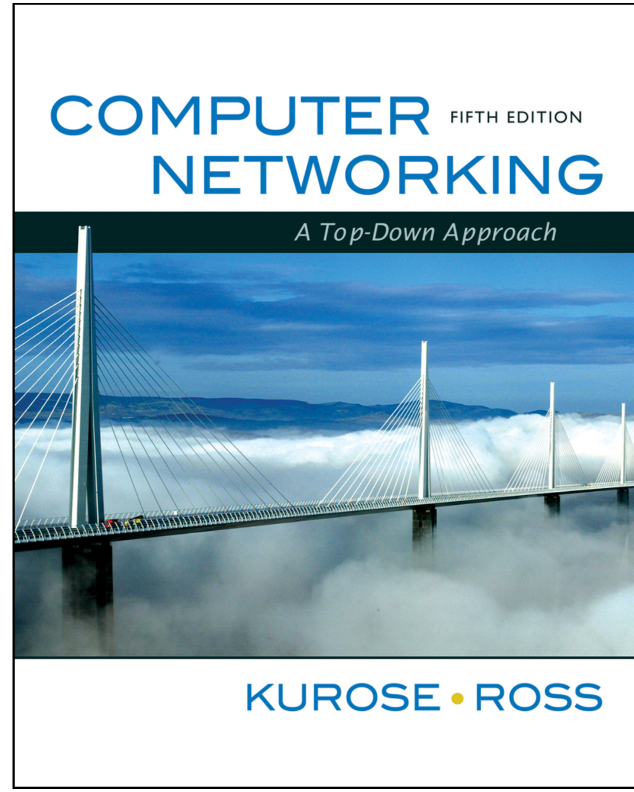
Transport Layer



Computer Networking: A Top Down Approach

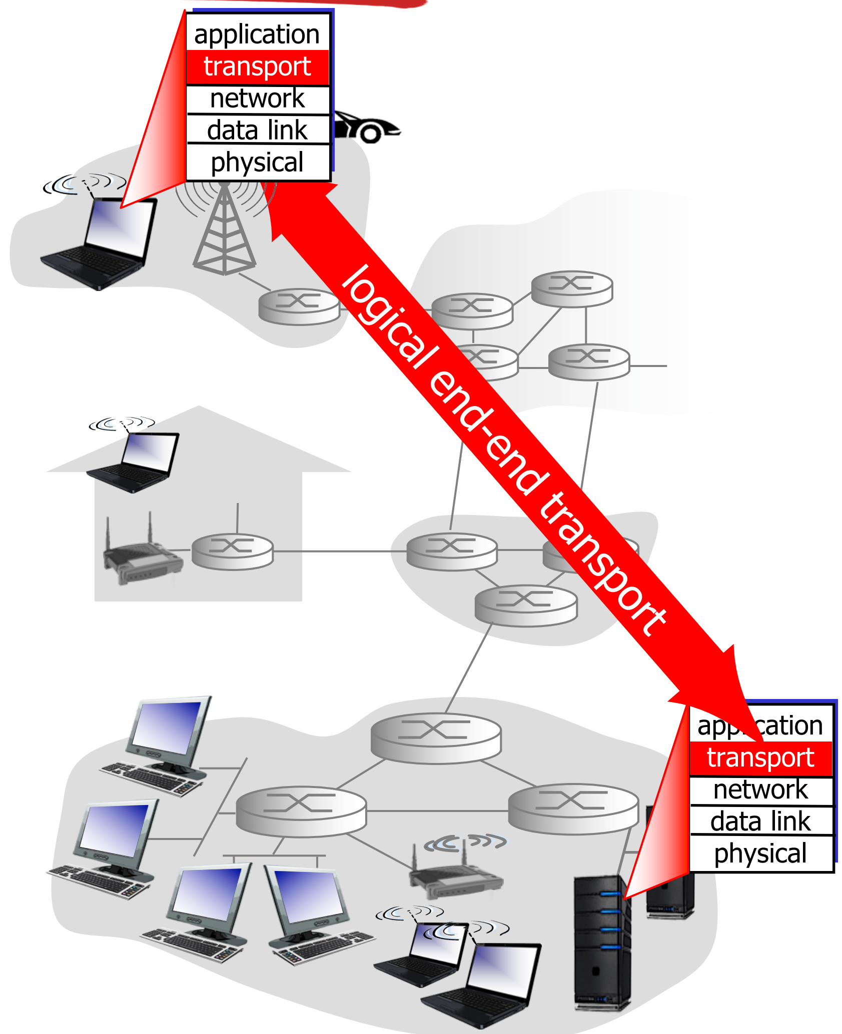
7th edition
Jim Kurose, Keith Ross
Pearson, 2017

Transport layer

- UDP: connectionless transport, not much to say
- TCP: connection-oriented reliable transport
- 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
- no delay/service guarantees



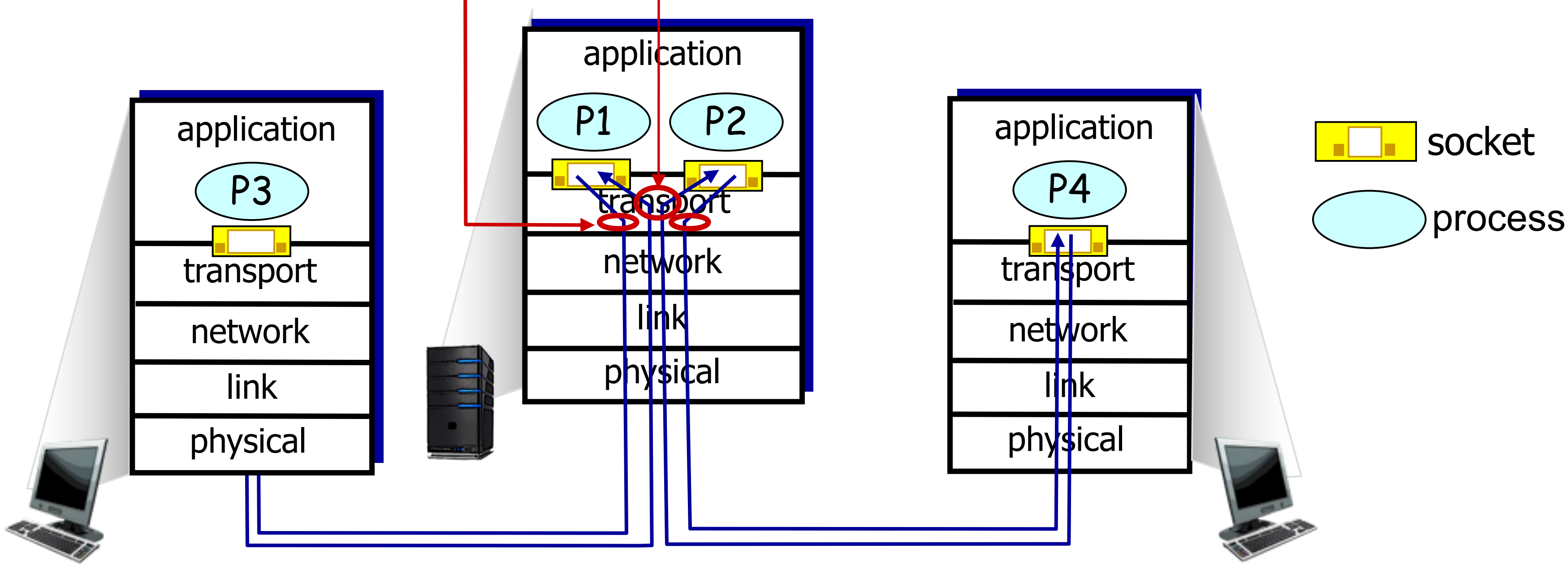
Multiplexing/demultiplexing

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

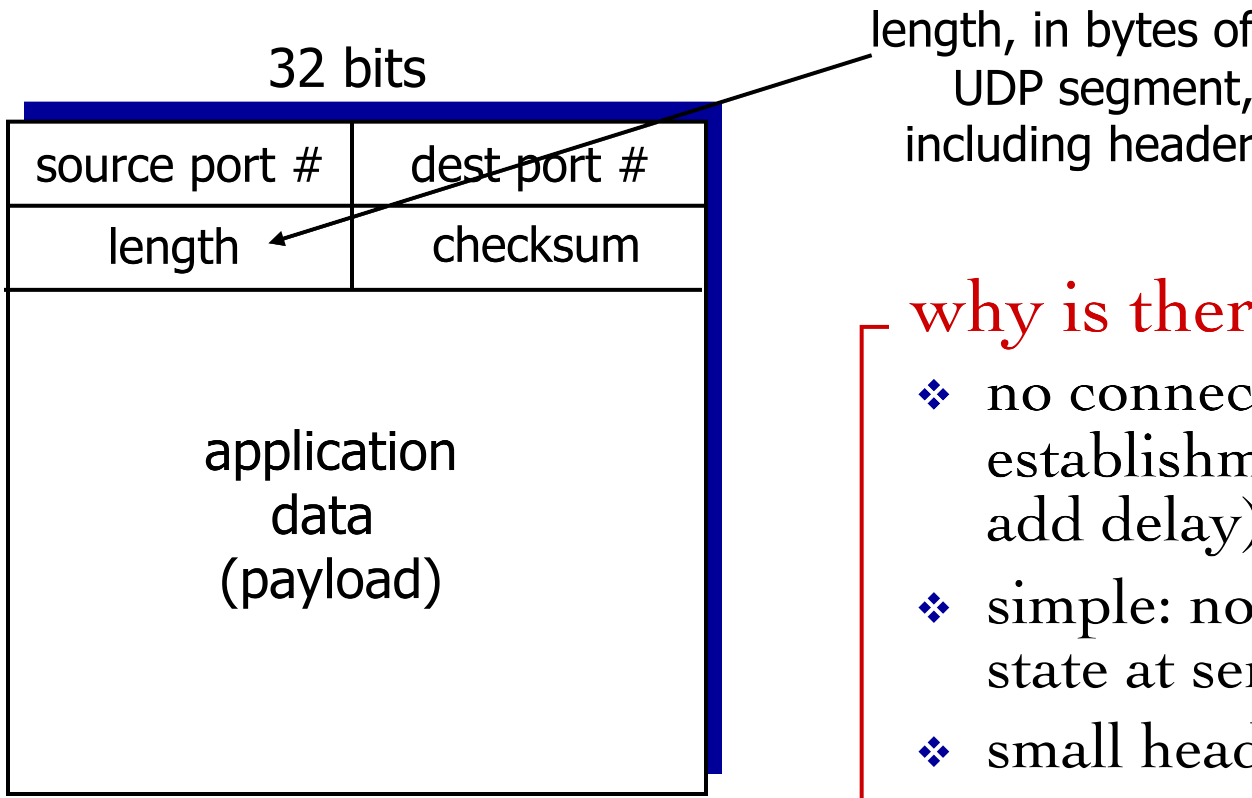


UDP: User Datagram Protocol [RFC 768]

- "no frills," "bare bones" 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?

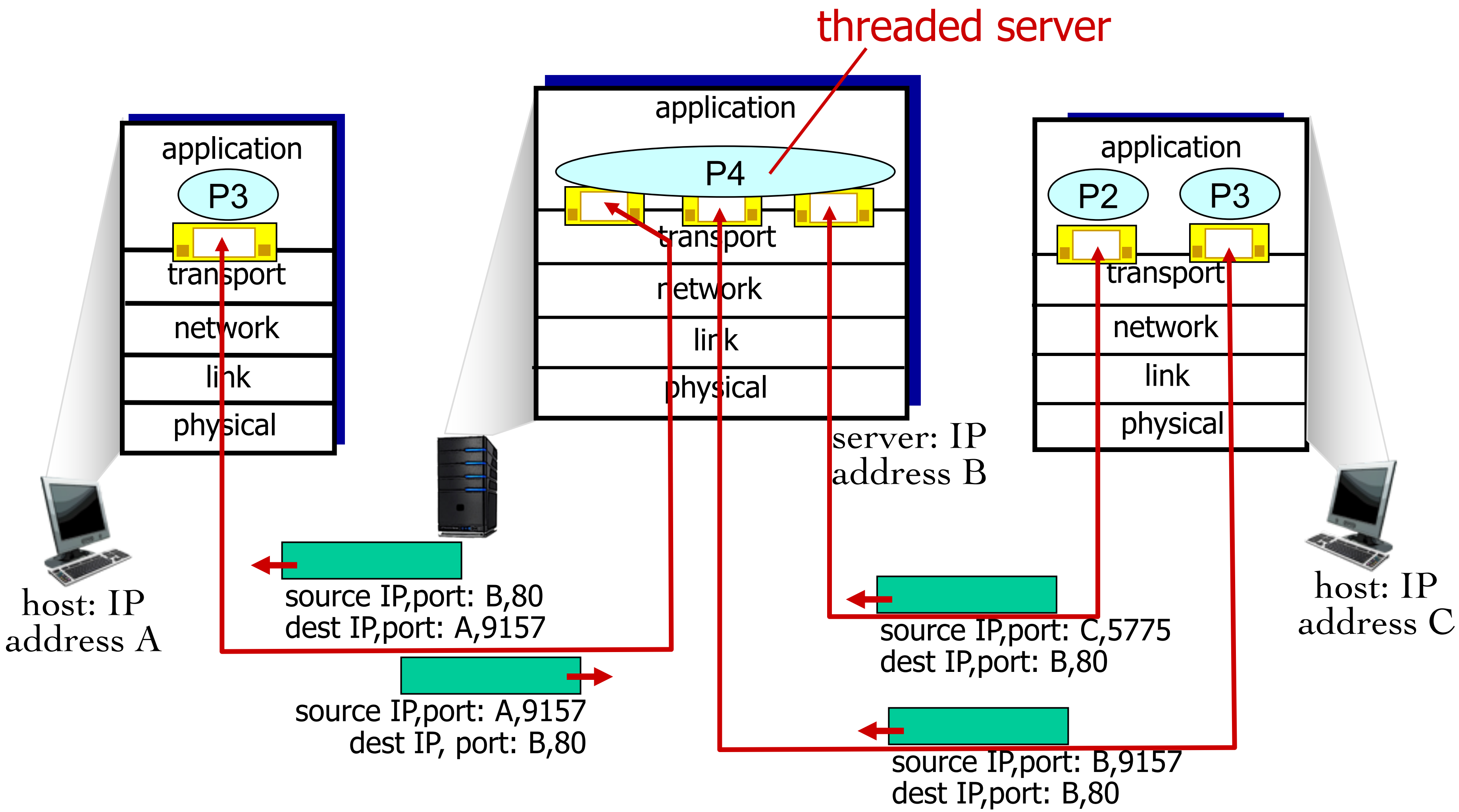
- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control: UDP can blast away as fast as desired

Connection-oriented demux

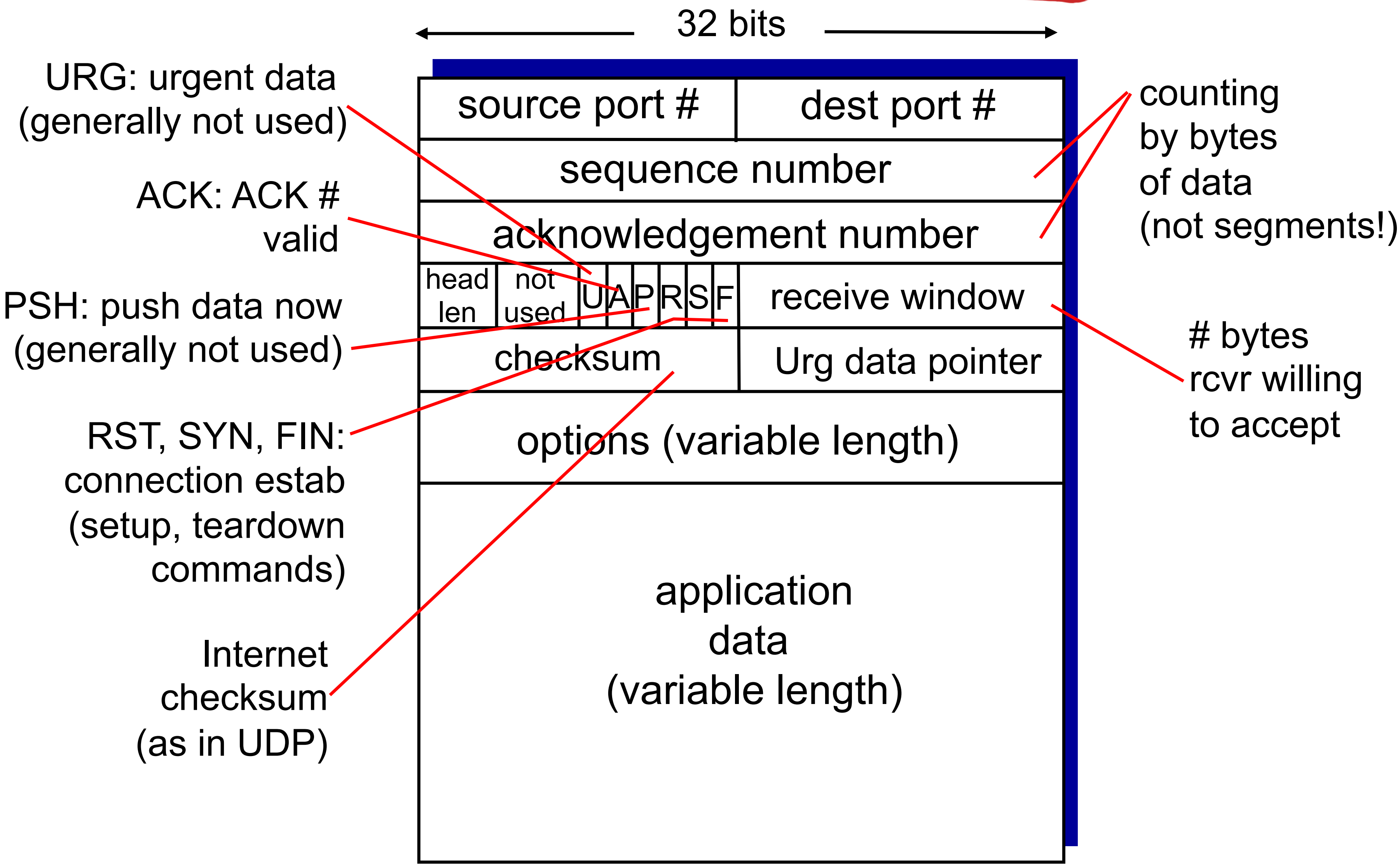
- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- demux: receiver uses all four values to direct segment to appropriate socket

- server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

Connection-oriented:demux



TCP segment structure



TCP overview [RFCs 793,1122,1323, 2018, 2581]

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte stream:*
  - no "message boundaries"
- pipelined:
  - TCP congestion and flow control set window size
  - Hybrid of go back n and selective repeat
- 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

Channels with errors and loss

assumption: underlying channel can lose packets (data, ACKs)

- checksum, seq. #, ACKs, retransmissions will be of help … but not enough

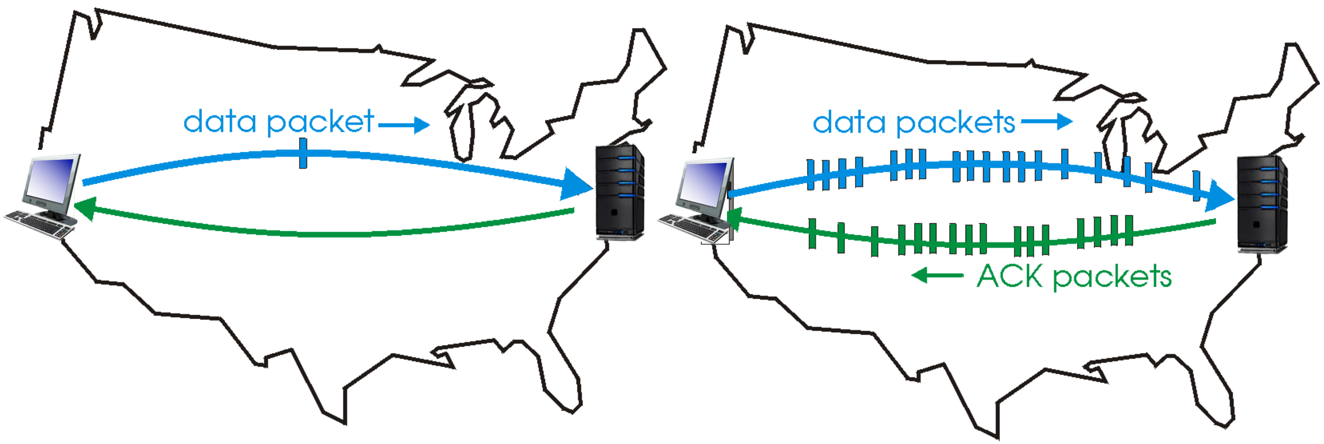
approach: sender waits "reasonable" amount of time for ACK

- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

Pipelined protocols

pipelining: sender allows multiple, "in-flight," yet-to-be-acknowledged packets

- range of sequence numbers must be increased
- buffering at sender and/or receiver



(a) a stop-and-wait protocol in operation (b) a pipelined protocol in operation

- two generic forms of pipelined protocols: *go-Back-N, selective repeat*

Pipelined protocols: concepts

Go-back-N:

- sender can have up to N unack'ed packets in pipeline
- receiver only sends *cumulative ack*
  - Doesn't ack packet if there's a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

Selective Repeat:

- sender can have up to N unack'ed packets in pipeline
- rcvr sends *individual ack* for each packet
- sender maintains timer for each unacked packet
  - when timer expires, retransmit only that unacked packet

TCP sequence numbers and ACKs

sequence numbers:

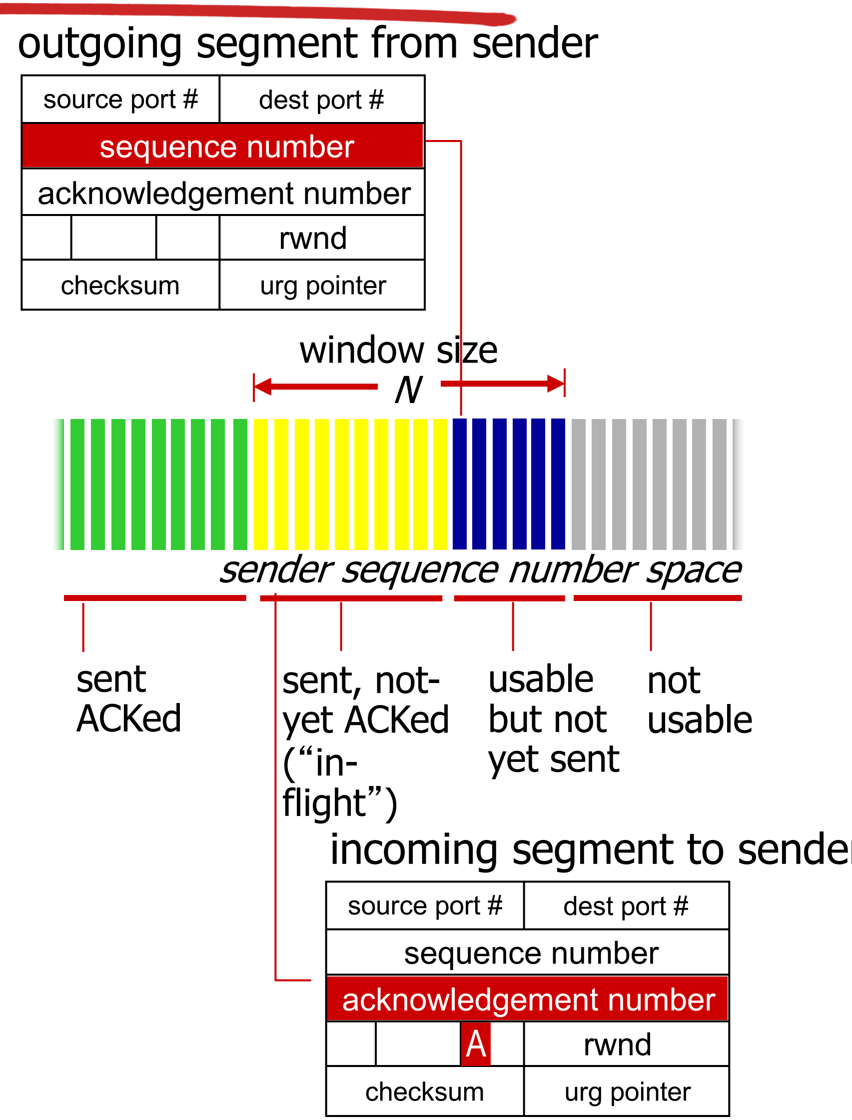
- byte stream "number" of first byte in segment's data

acknowledgements:

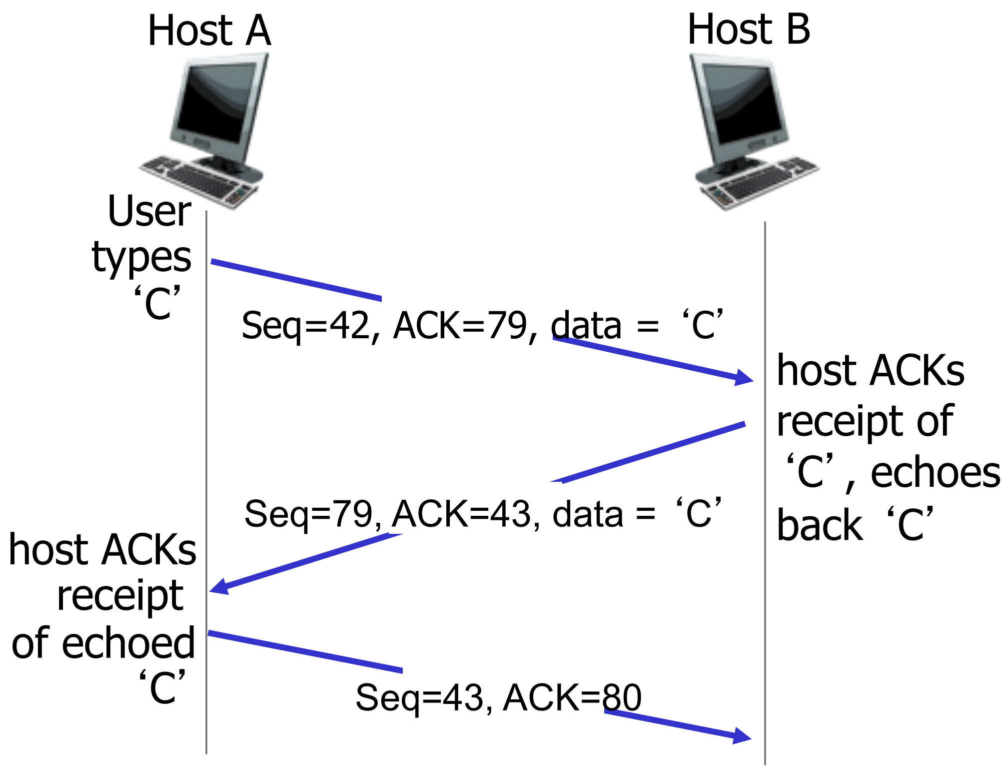
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

- A: TCP spec doesn't say, - up to implementor



TCP sequence numbers and acks



simple telnet scenario

TCP RTT estimation

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

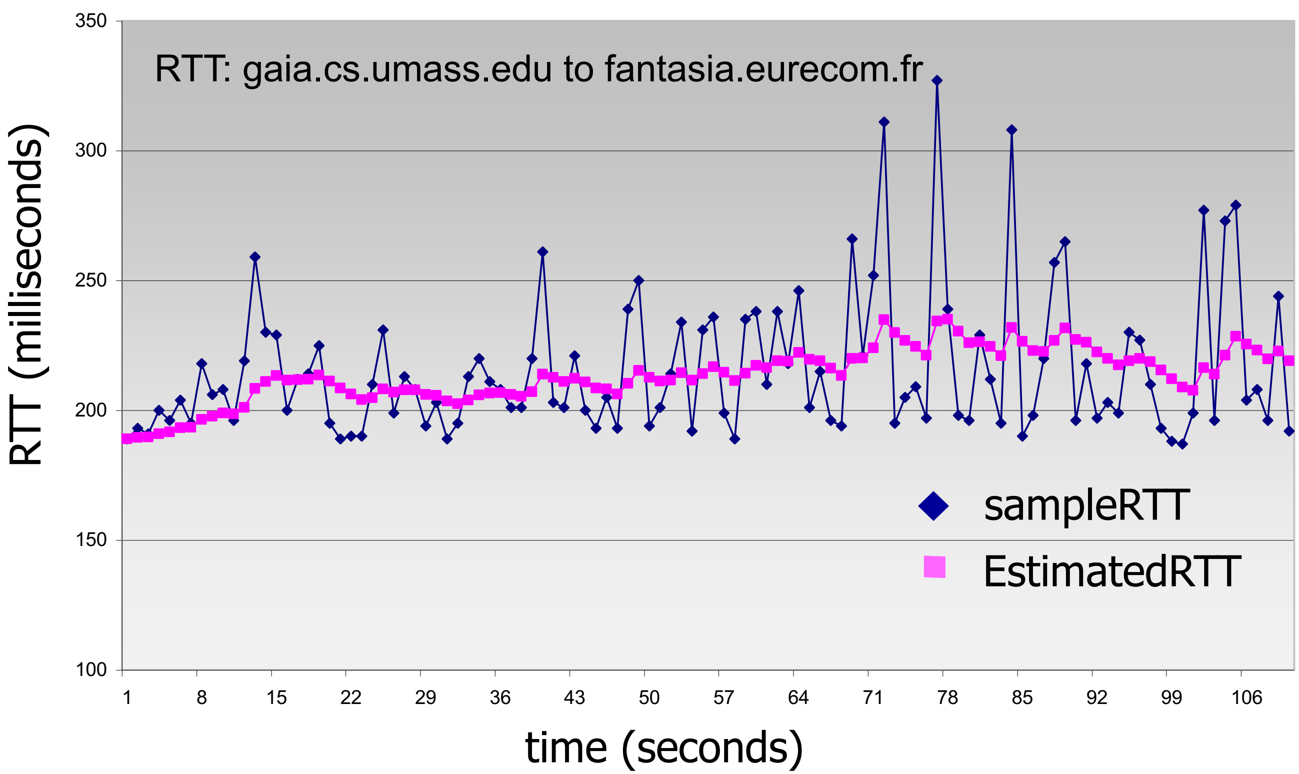
Q: how to estimate RTT?

- **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 RTT estimation

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

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



TCP 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 sender (no optimization, congestion control)

data rcvd from app:

- create segment with seq #
- seq # is byte-stream number of first data byte in segment
- start timer if not already running
  - think of timer as for oldest unacked segment
  - expiration interval: **`TimeOutInterval`**

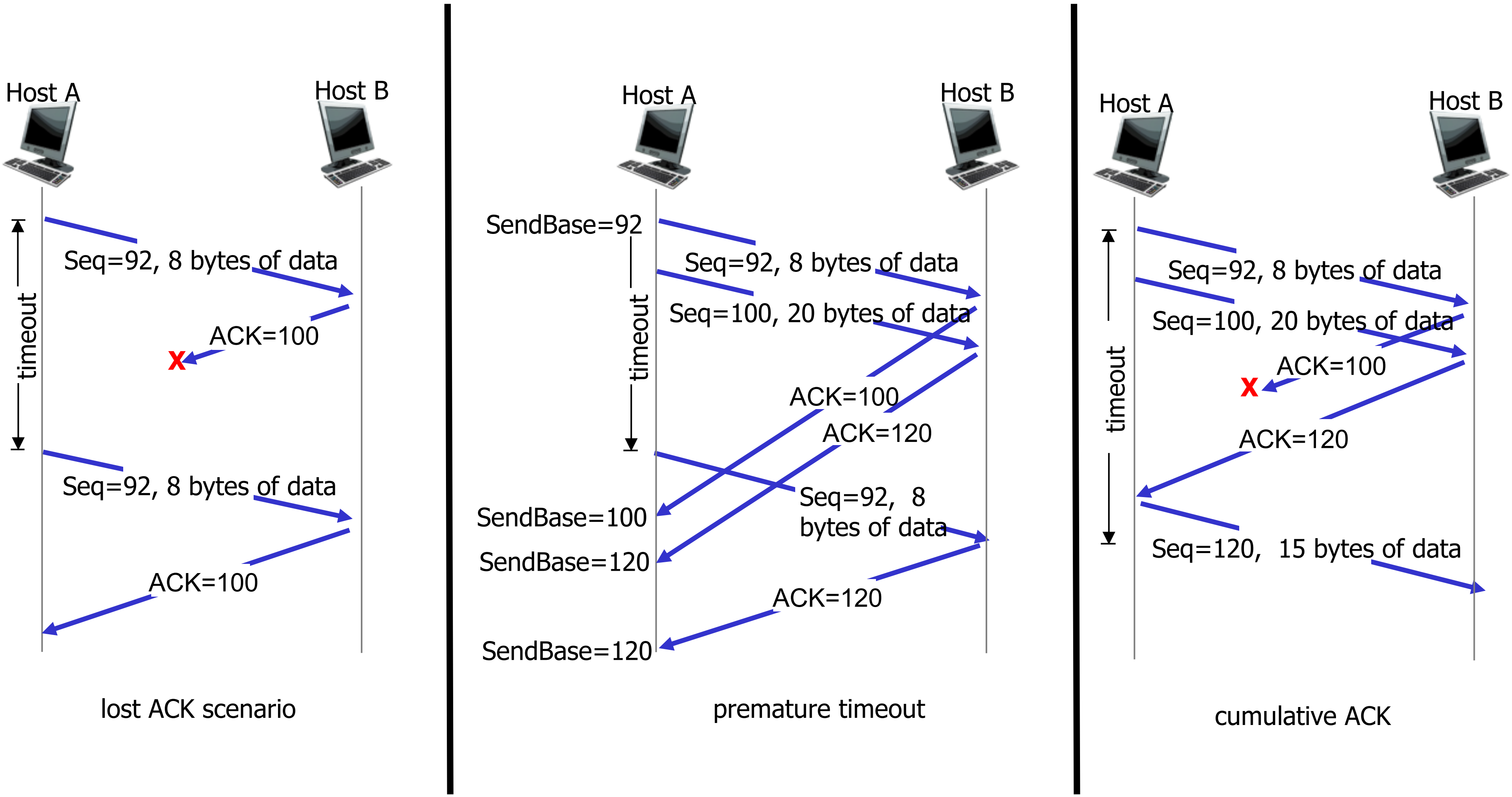
timeout:

- retransmit segment that caused timeout
- restart timer

ack rcvd:

- if ack acknowledges previously unacked segments
  - update what is known to be ACKed
  - start timer if there are still unacked segments

TCP retransmission scenarios



TCP ACK original generation [RFC 1122, 2581]

event at receiver	*TCP receiver action*
arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	delayed ACK: wait up to 500ms for next segment, if no next segment, send ACK
arrival of in-order segment with expected seq #. One other segment has ACK pending	immediately send single cumulative ACK, ACKing both in-order segments
arrival of out-of-order segment higher-than-expect seq. # . Gap detected	immediately send duplicate ACK, indicating seq. # of next expected byte
arrival of segment that partially or completely fills gap	immediate send ACK, provided that segment starts at lower end of gap

TCP fast retransmit

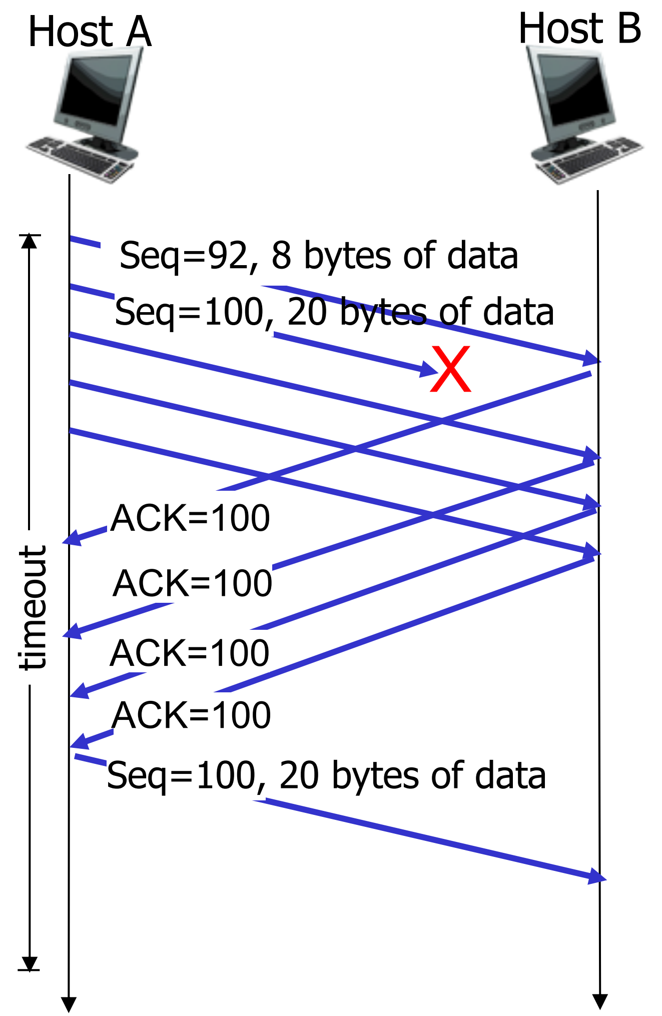
- time-out period often relatively long:
  - long delay before resending lost packet
- detect lost segments via duplicate ACKs.
  - sender often sends many segments back-to-back
  - if segment is lost, there will likely be many duplicate ACKs.

TCP fast retransmit

if sender receives 3 ACKs for same data ("triple duplicate ACKs"), resend unacked segment with smallest seq #

- likely that unacked segment lost, so don't wait for timeout

TCP fast retransmit

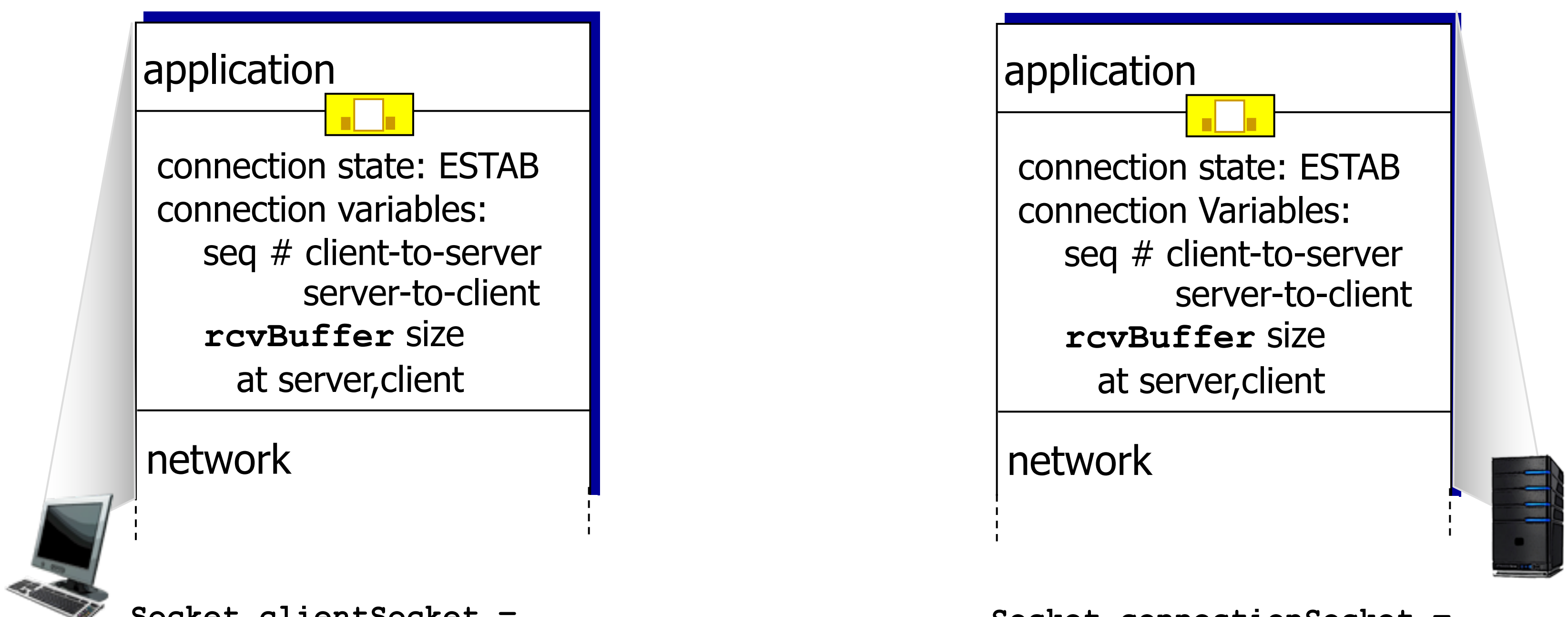


fast retransmit after sender
receipt of triple duplicate ACK

Connection management

before exchanging data, sender/receiver “handshake”:

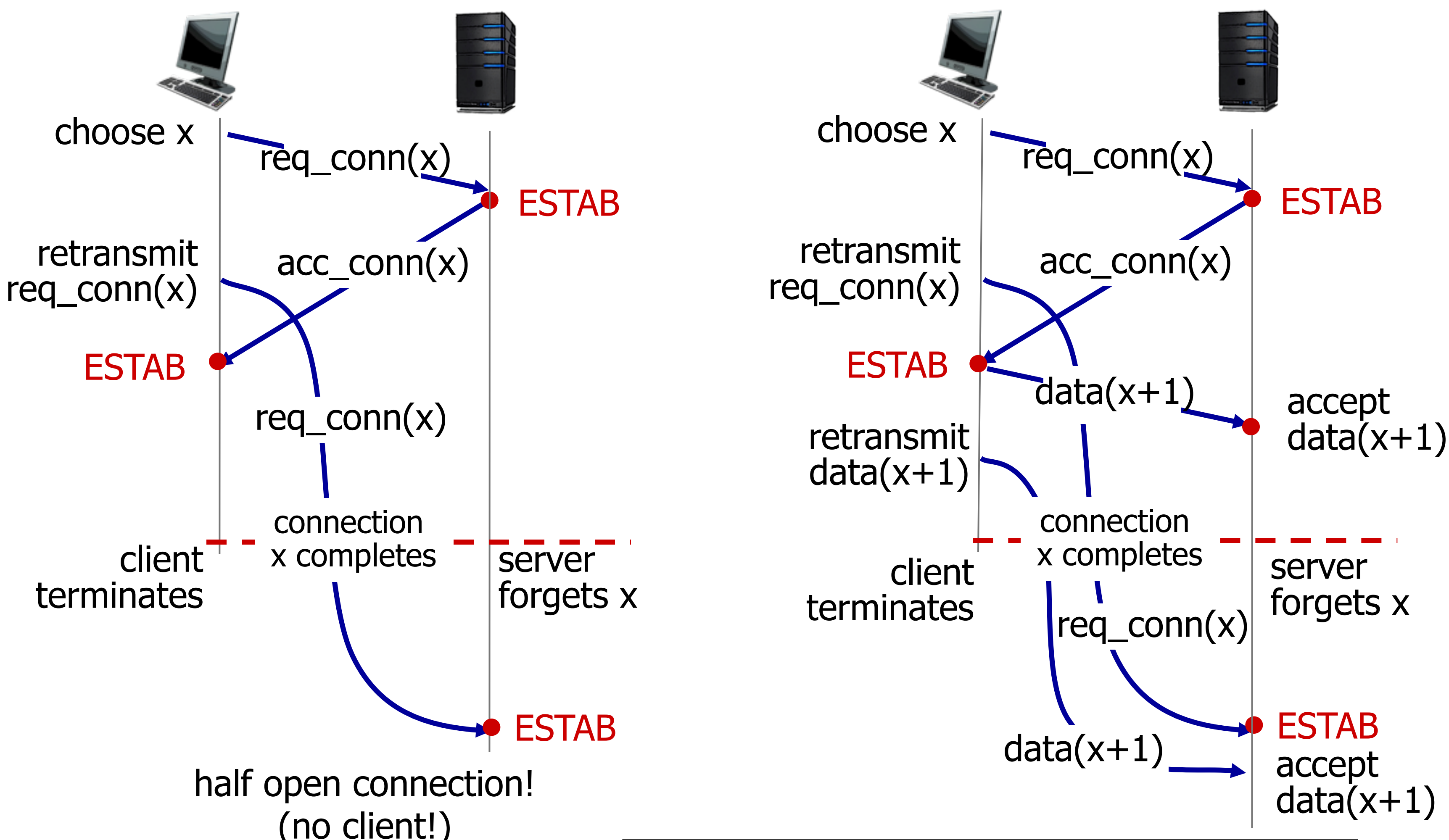
- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters – sequence number




```
Socket clientSocket =
  newSocket("hostname","port number");
```

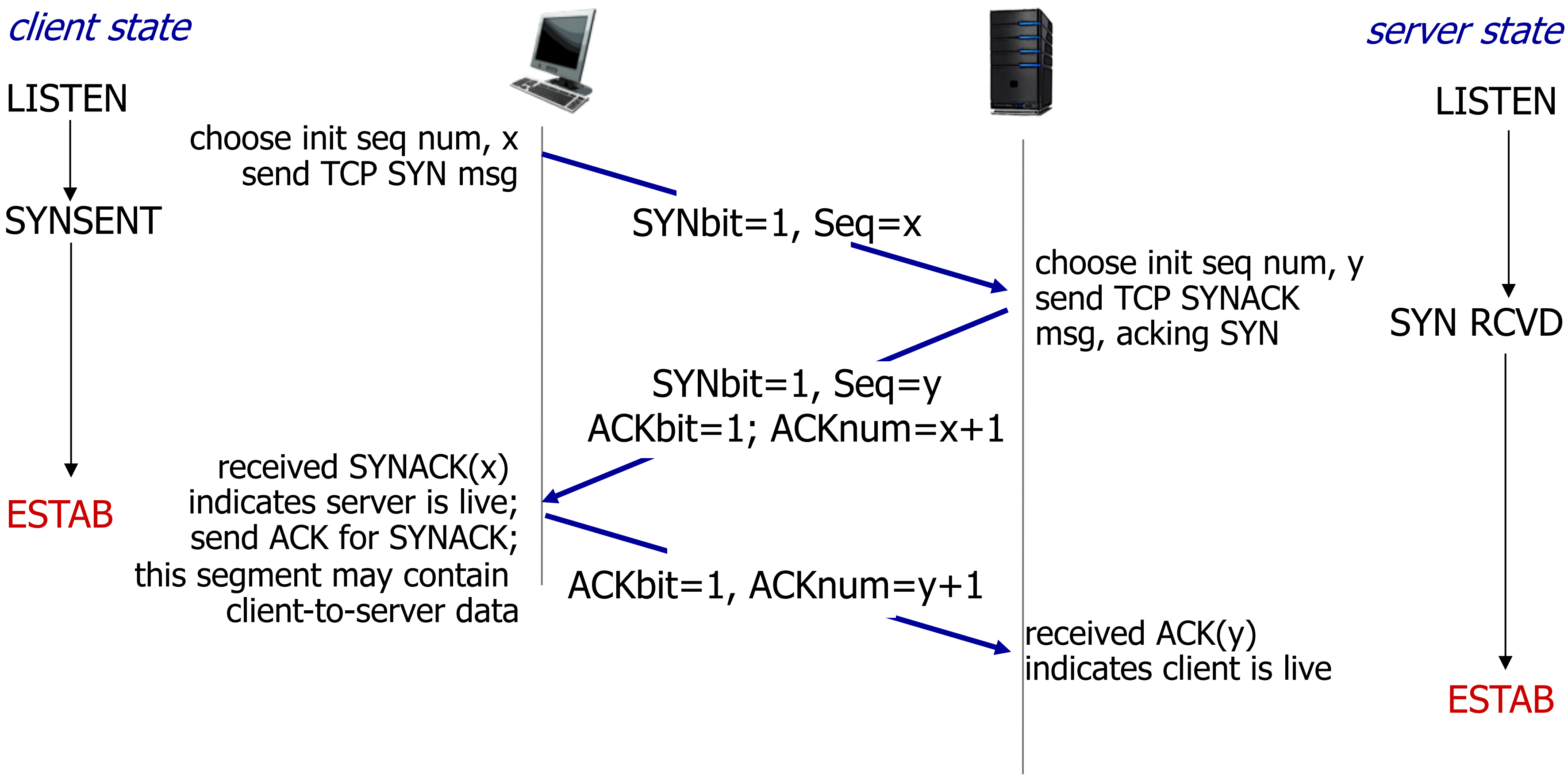
```
Socket connectionSocket =
  welcomeSocket.accept();
```

Agreeing to establish a connection



2-way handshake failure scenarios

TCP 3-way handshake



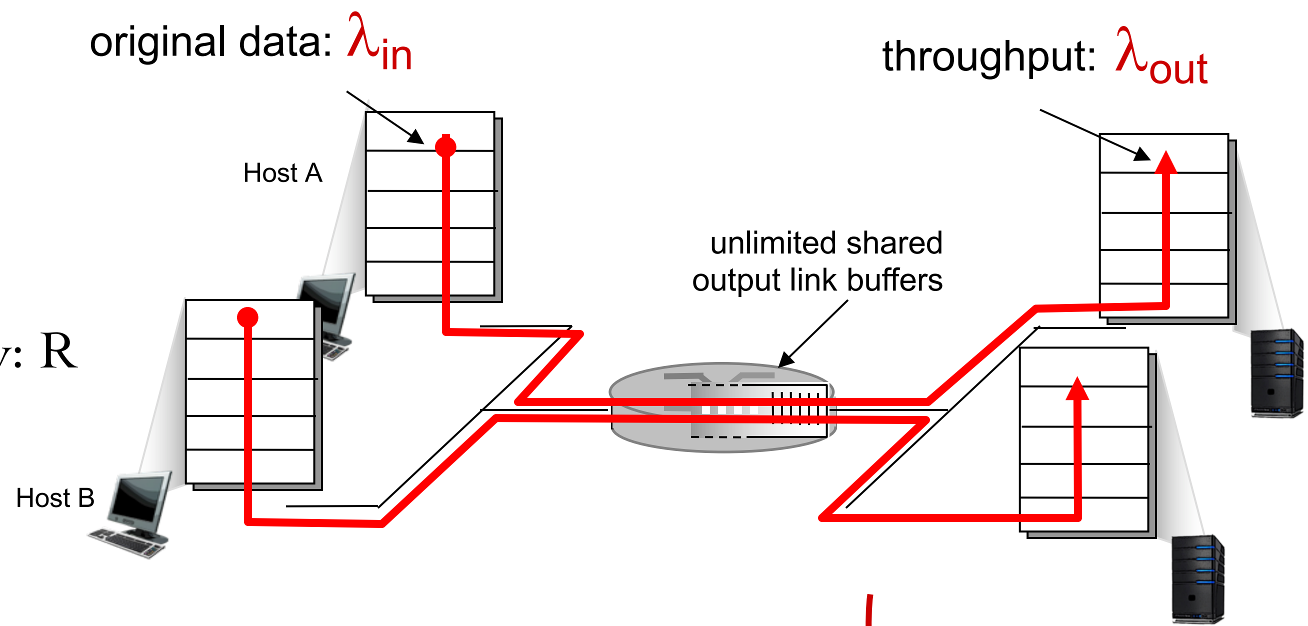
Principles of congestion control

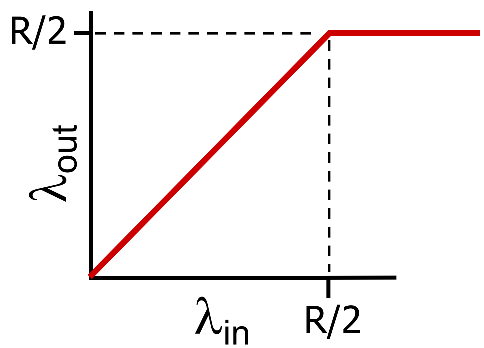
congestion:

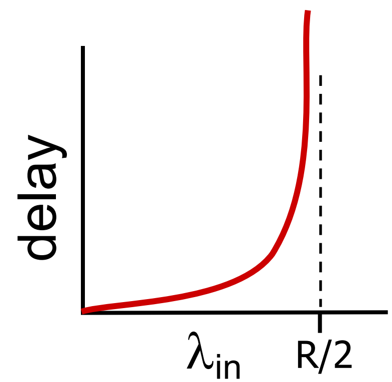
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: delay

- two senders, two receivers
- one router, infinite buffers
- output link capacity: R
- no retransmission



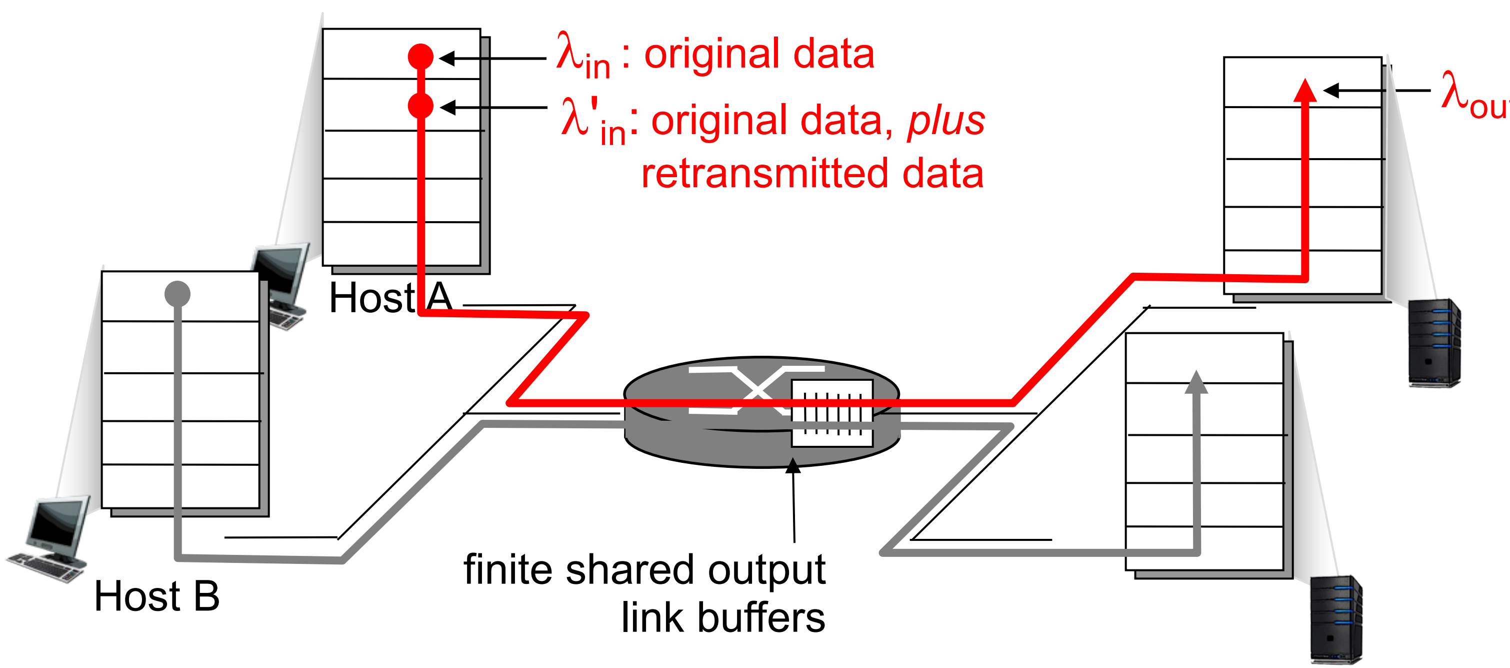




- maximum per-connection throughput: R/2
- large delays as arrival rate, λ_{in}, approaches capacity

Causes/costs of congestion: drop + duplicate

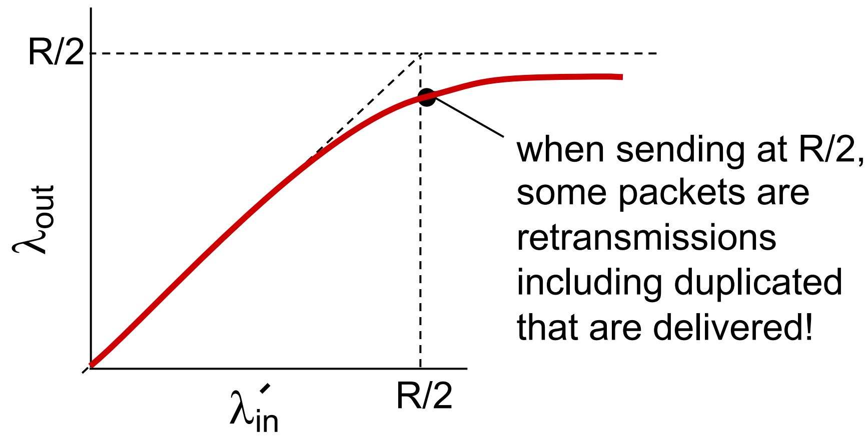
- one router, *finite* buffers
- sender retransmission of timed-out packet
  - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
  - transport-layer input includes ***retransmissions*** : $\lambda'_{in} \geq \lambda_{in}$



Causes/costs of congestion

duplicates

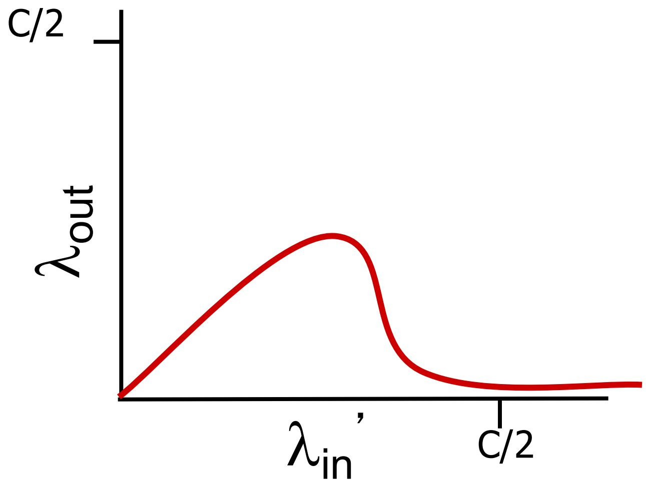
- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending two copies, both of which are delivered

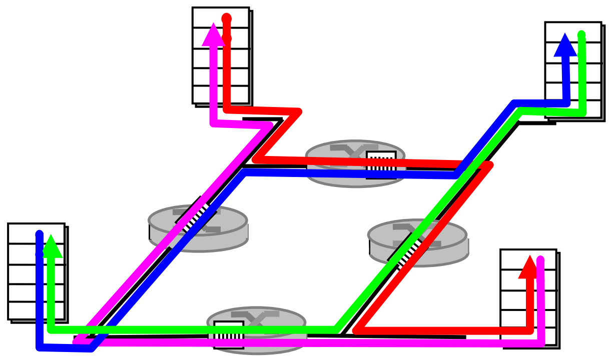


costs of congestion:

- more work (retransmissions) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkts
  - decreasing goodput

Causes/costs of congestion: multi-hop paths





another "cost" of congestion:

- when packet dropped, any "upstream" transmission capacity used for that packet was wasted!

Approaches to congestion control

two broad approaches towards congestion control:

end-end congestion control:

- no explicit feedback from network
- congestion inferred from end-system observed loss, delay, Ack signals from receiver
- approach taken by TCP

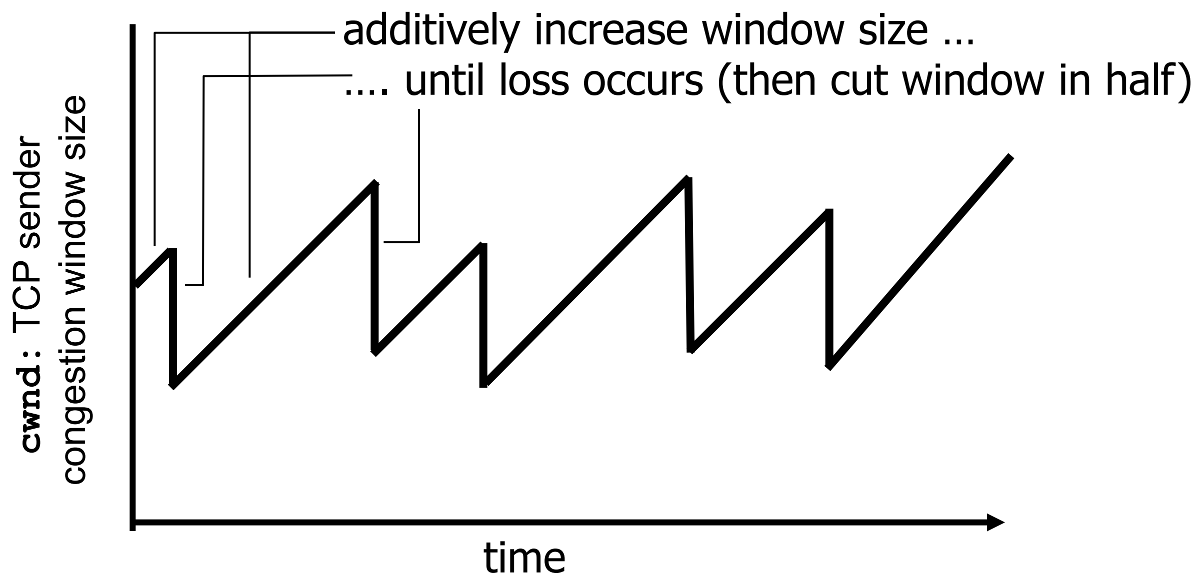
network-assisted congestion control:

- routers provide feedback to end systems
  - single bit indicating congestion as packet forwarded (SNA, DECbit, TCP/IP ECN)

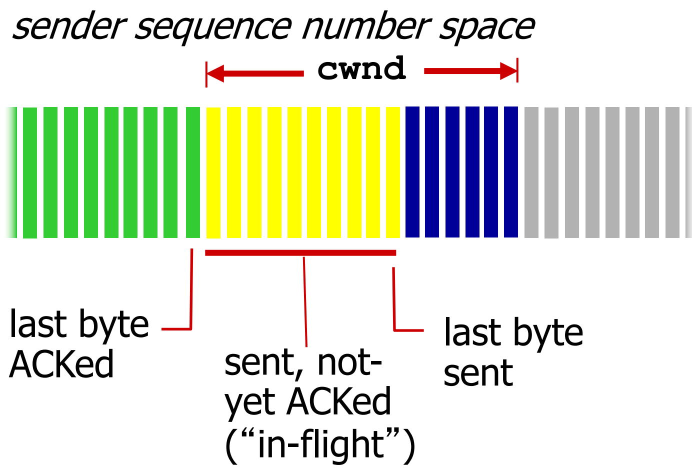
TCP congestion control basics

- approach: sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
  - additive increase: increase **`cwnd`** by 1 MSS (max segment size, default 536 bytes) every RTT until loss detected
  - multiplicative decrease: cut **`cwnd`** in half after loss

AIMD saw tooth behavior: probing for bandwidth



TCP congestion control window



- sender limits transmission:

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

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

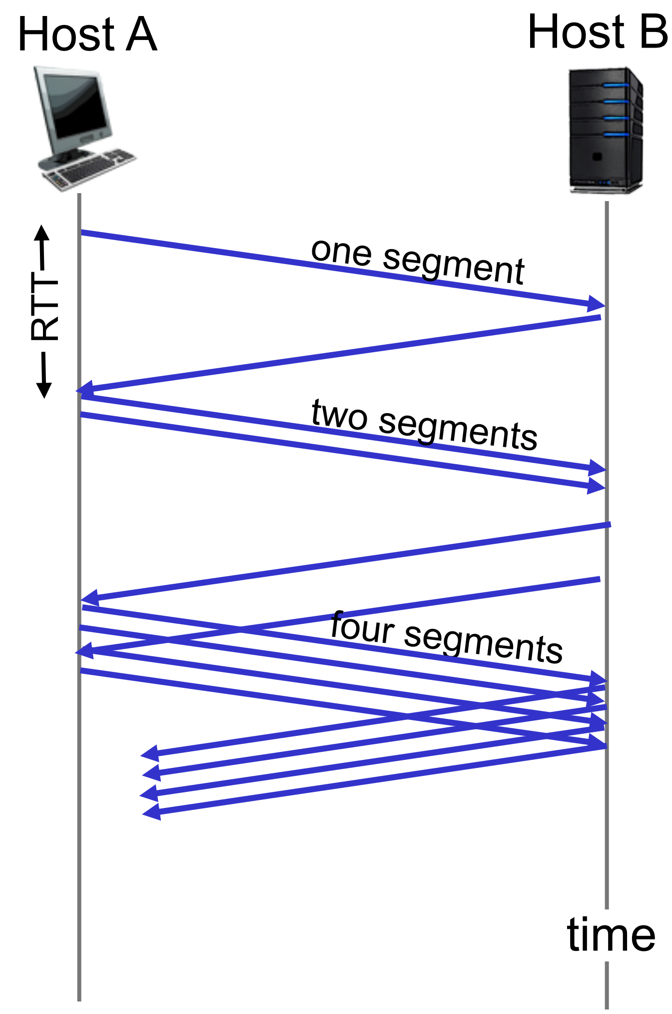
TCP sending rate:

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

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

TCP slow start

- when connection begins, rate (window) small but increase rate (window) 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 loss

- Timeout
- Duplicate ACKs from receiver
  - Receiver adjusted to send ACK each time packet received even if it's a duplicate (no advance in seq number)
  - Sender waits for 3 duplicate ACKs (not just one in case slightly out of order delivery)

TCP: responding to loss

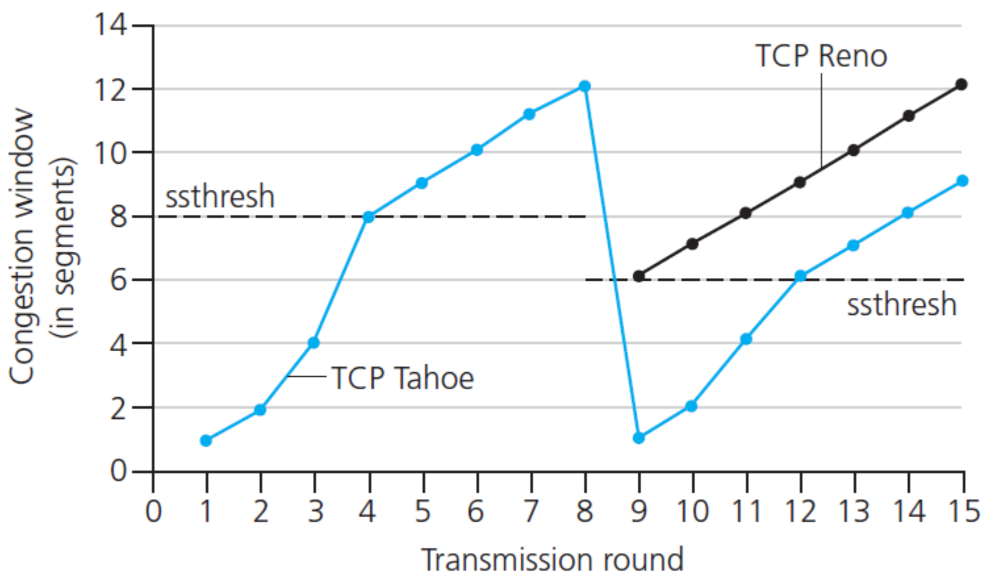
- If loss detected by timeout, **`cwnd`** set to 1 MSS
  - window initially grows exponentially to threshold (ssthresh), then grows linearly

- If lost detected by duplicate ACKS, can be less reactive since ACKs indicate network able to deliver packets to receiver
  - TCP Reno: **`cwnd`** is only cut in half (not set to 1); then window grows linearly
  - TCP New Reno: improves retransmit during fast recovery given wireless link loss is not usually due to network congestion
    - for every ACK that advances sequence space, send next packet beyond the ACKed sequence number as well

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.



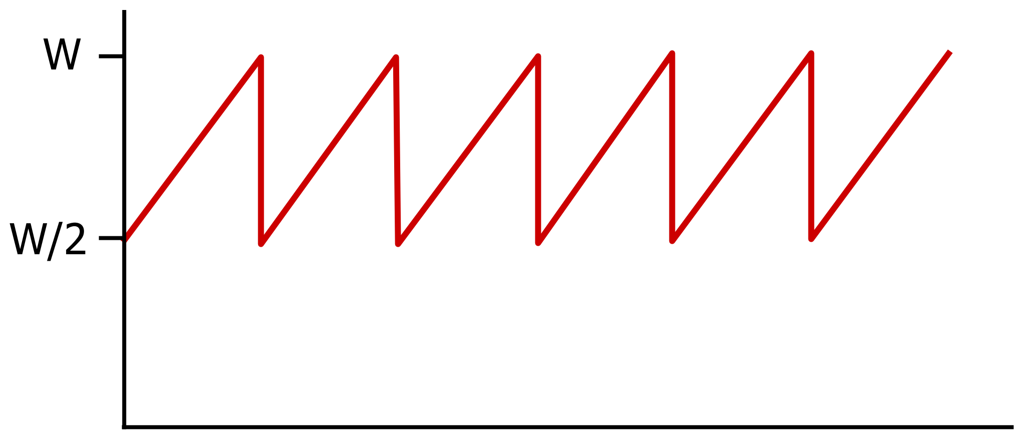
Implementation:

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

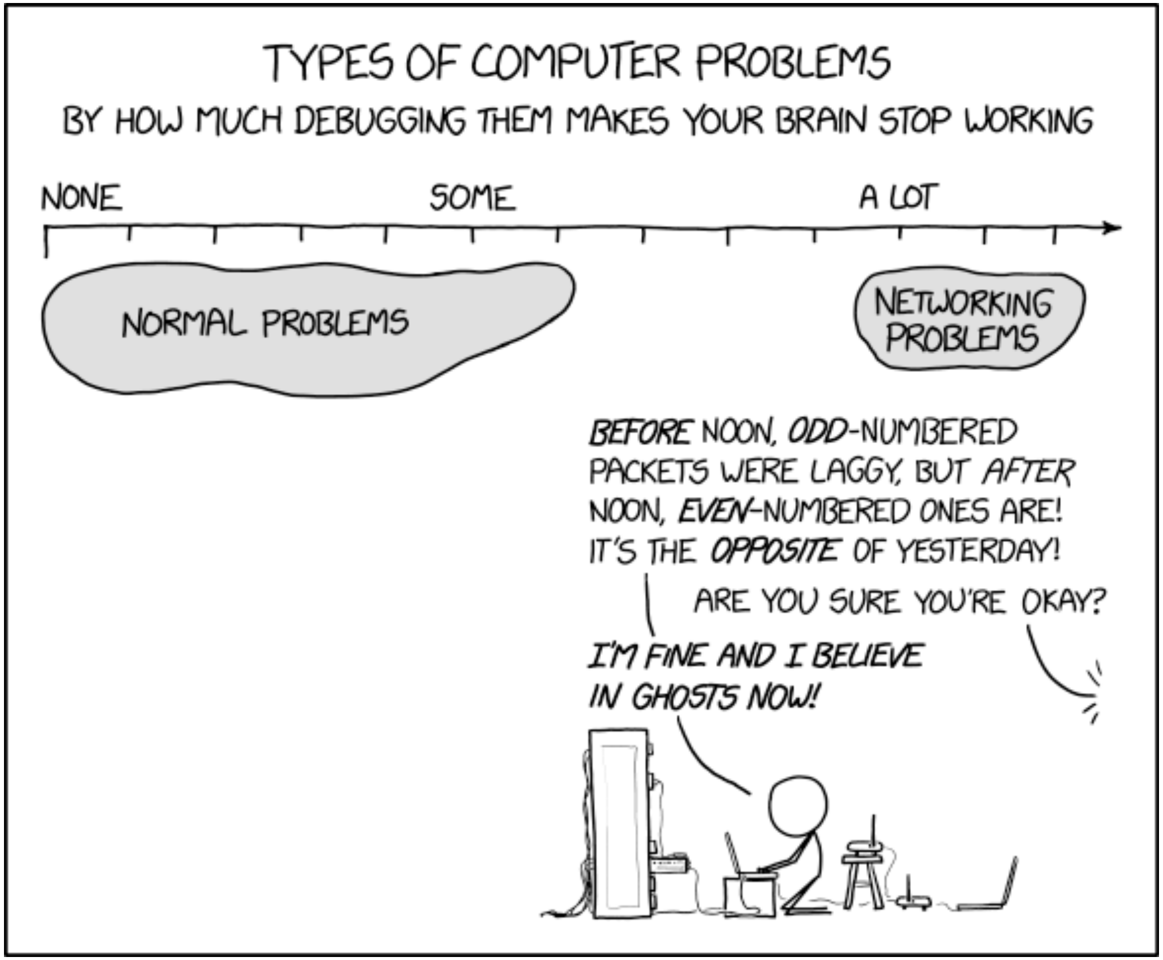
TCP throughput

- avg. TCP thruput as function of window size, RTT?
  - ignore transient of slow start, assume always data to send
- W: window size (measured in bytes) where loss occurs
  - avg. window size (# in-flight bytes) is ¾ W
  - avg. thruput is 3/4W per RTT

$$\text{avg TCP thruput} = \frac{3}{4}\frac{W}{RTT} \text{ bytes/sec}$$



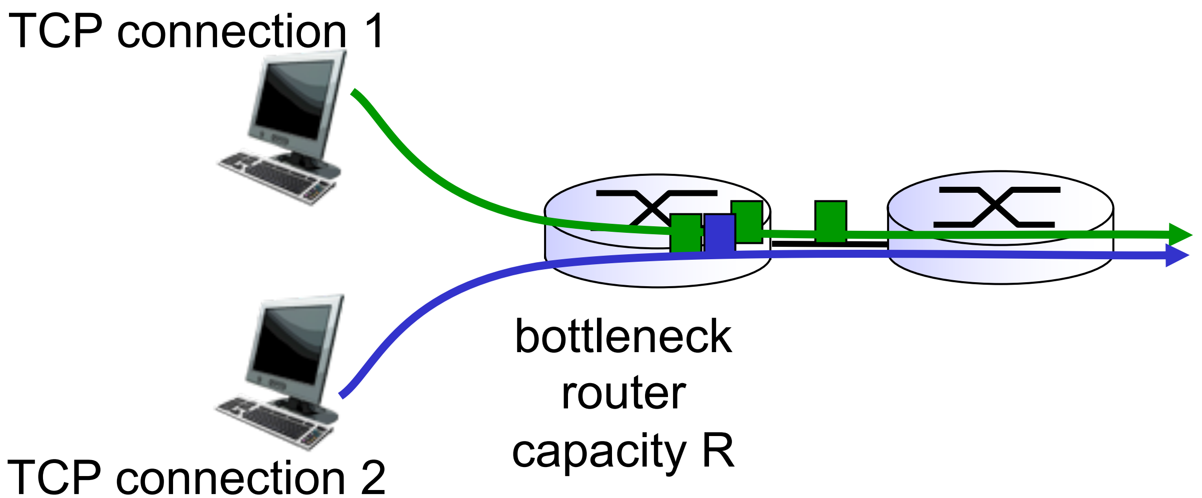
Understanding network behavior is hard



source: xkcd

TCP fairness

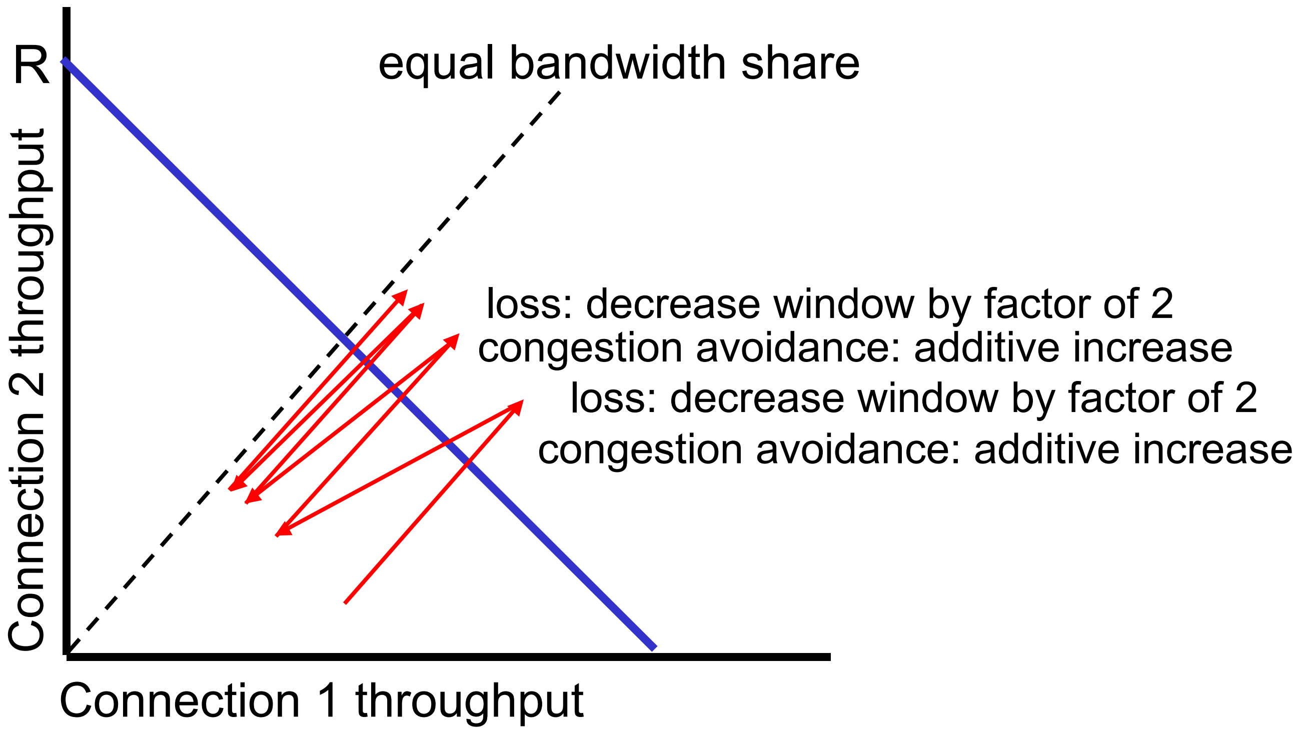
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Why is TCP fair, in concept

two competing sessions:

- additive increase gives slope of 1, as throughout increases
- multiplicative decrease decreases throughput proportionally



Side note: proportional fair scheduling in 3G

```
[~, b_user] = max(drc(i, :)/_avg_thruput(i, :));
avg_thruput(i+1, :) = (i/(i+1))*avg_thruput(i, :);
avg_thruput(i+1, b_user) = (i/(i+1))*avg_thruput(i, b_user) +
drc(i, b_user)/(i+1);
```

"Transmitter directed, multiple receiver system using path diversity to equitably maximize throughput," U.S. Patent No. 6449490, Sept. 10, 2002