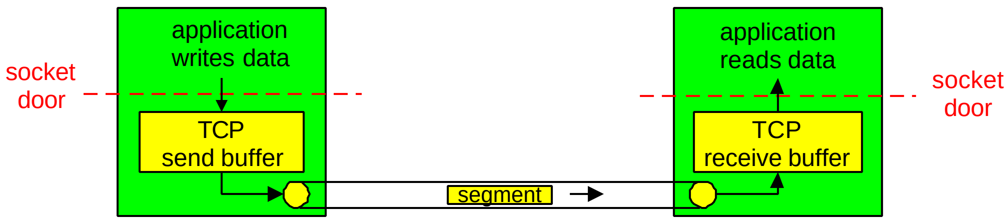
TCP: Overview

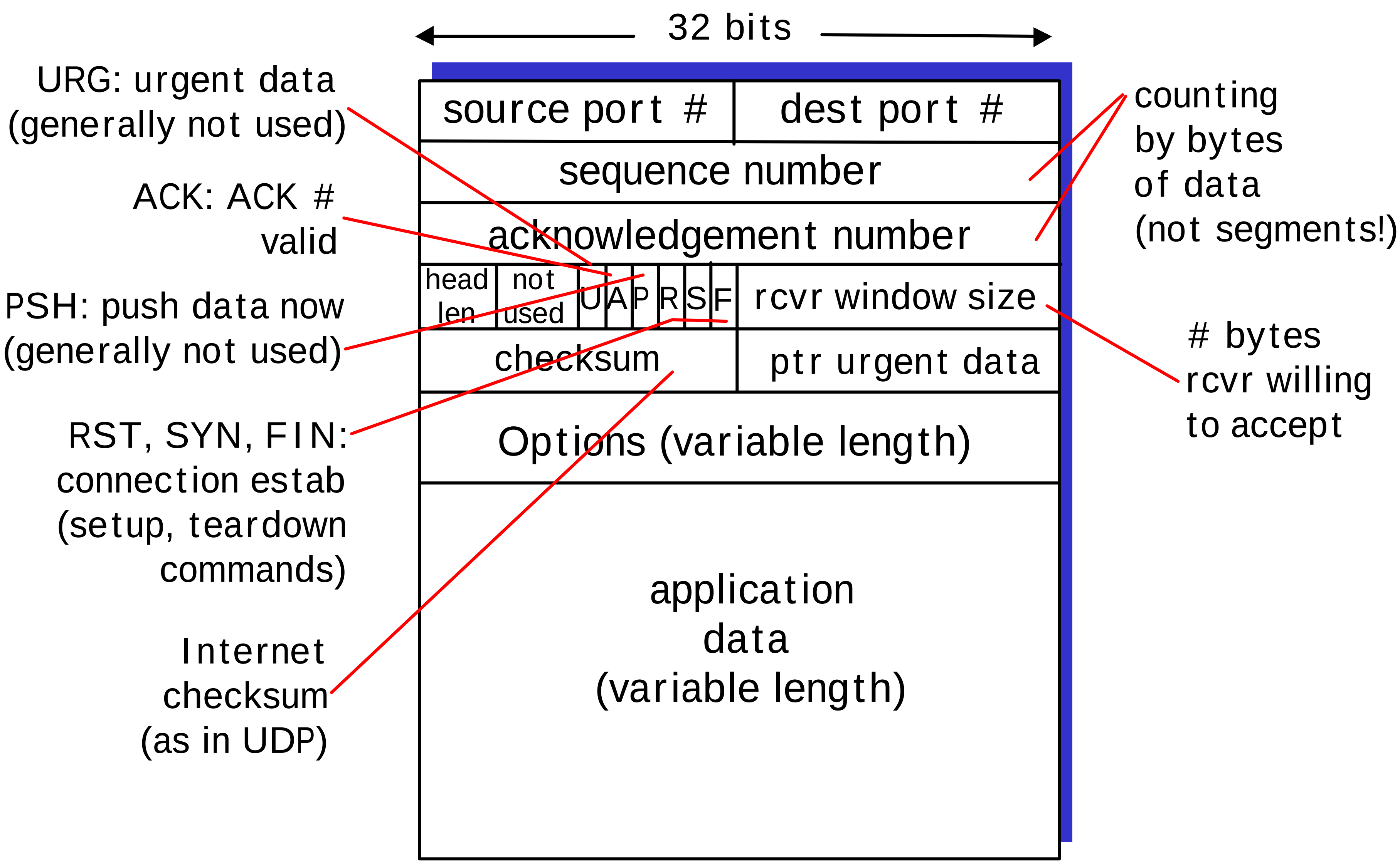
RFCs: 793, 1122, 1323, 2018, 2581

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - no “message boundaries”
- pipelined:
  - TCP congestion and flow control set window size
- *send & receive buffers*

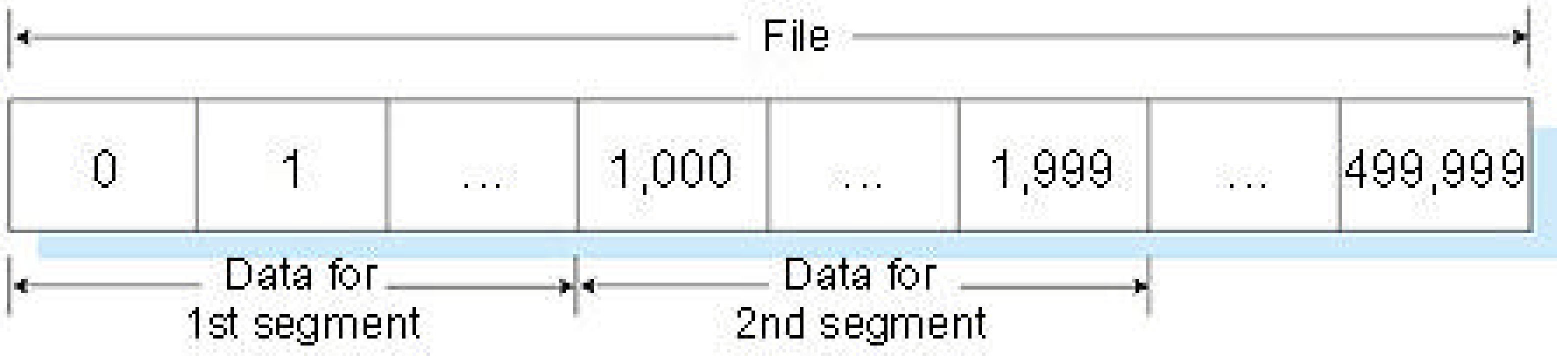


- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control msgs) init’s sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure



Dividing file data into TCP segments



TCP seq. #'s and ACKs

Seq. #'s:

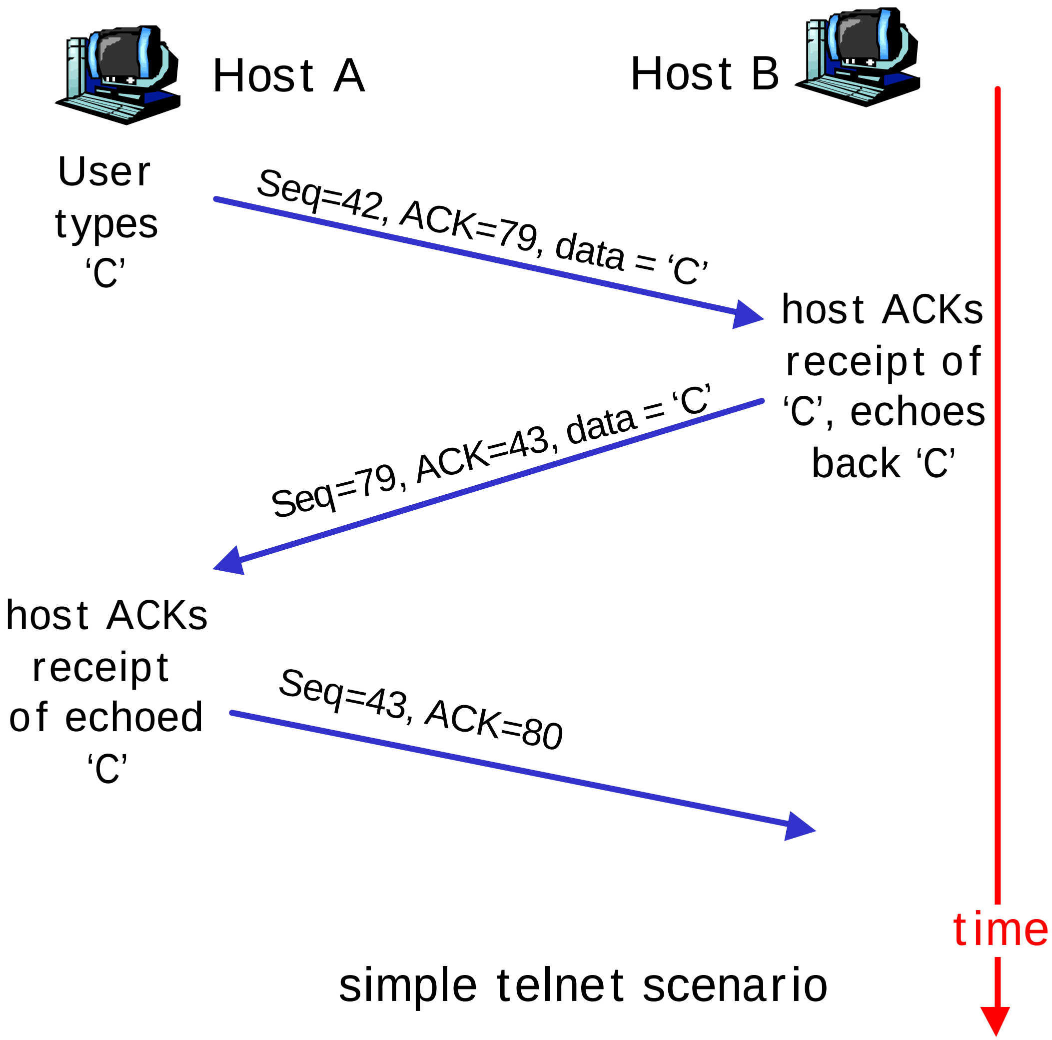
- byte stream "number" of first byte in segment's data

ACKs:

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

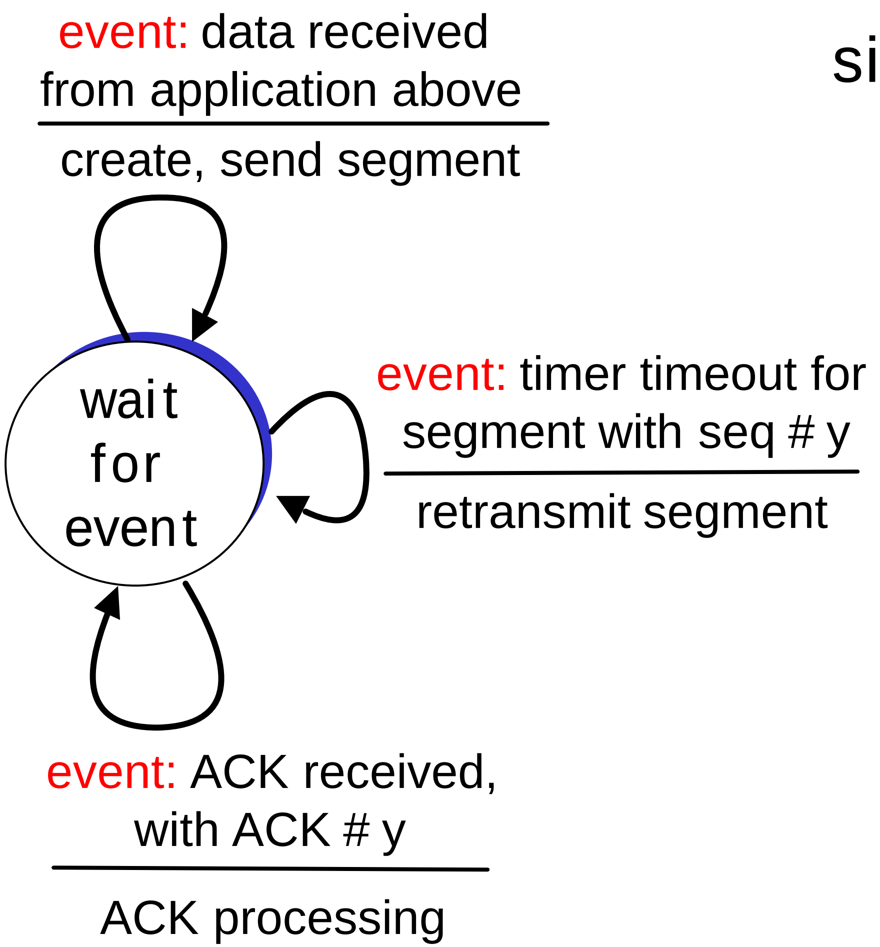
Q: how receiver handles out-of-order segments

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



simple telnet scenario

TCP: reliable data transfer



simplified sender, assuming

- one way data transfer
- no flow, congestion control

TCP: reliable data transfer

Simplified TCP sender

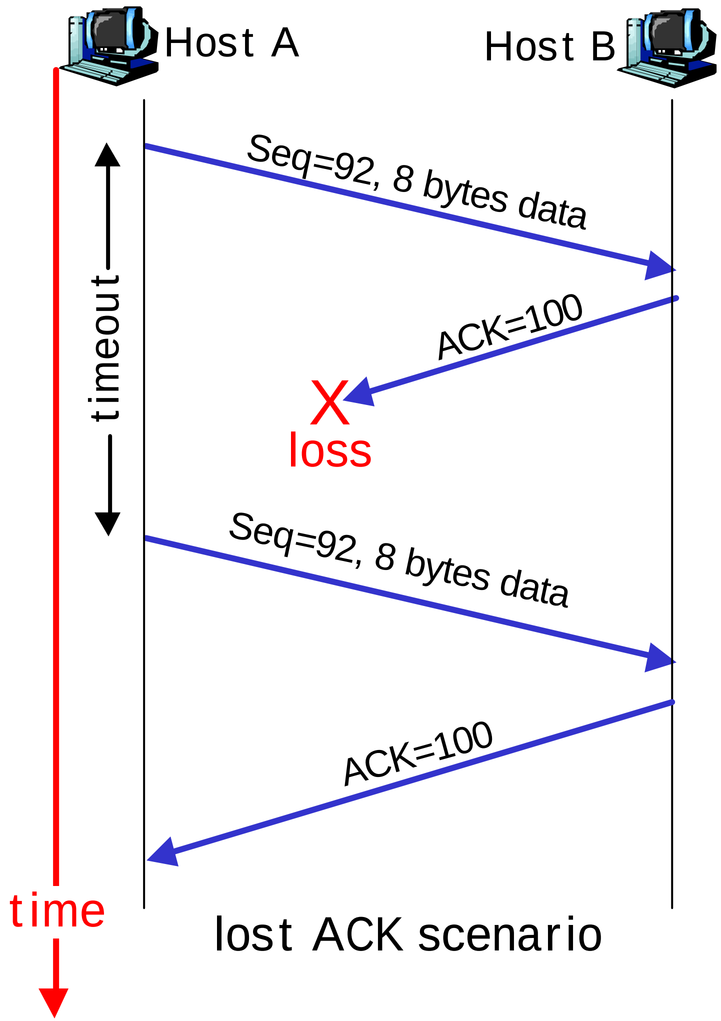
```
sendbase = initial_sequence number
nextseqnum = initial_sequence number

 loop (forever) {
   switch(event)
   event: data received from application above
         create TCP segment with sequence number nextseqnum
         start timer for segment nextseqnum
         pass segment to IP
         nextseqnum = nextseqnum + length(data)
    event: timer timeout for segment with sequence number y
         retransmit segment with sequence number y
         compue new timeout interval for segment y
         restart timer for sequence number y
    event: ACK received, with ACK field value of y
         if (y > sendbase) { /* cumulative ACK of all data up to y */
             cancel all timers for segments with sequence numbers < y
              sendbase = y
              }
         else { /* a duplicate ACK for already ACKed segment */
              increment number of duplicate ACKs received for y
              if (number of duplicate ACKS received for y == 3) {
                  /* TCP fast retransmit */
                 resend segment with sequence number y
                 restart timer for segment y
             }
   }  /* end of loop forever */
```

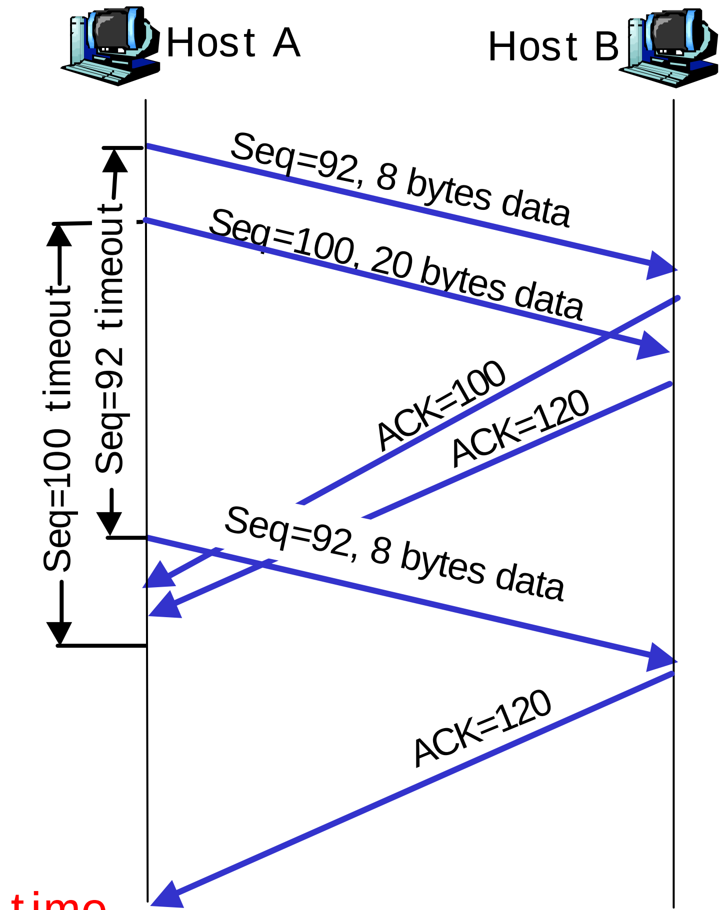
TCP ACK generation [RFC 1122, RFC 2581]

Event	TCP Receiver action
in-order segment arrival, no gaps, everything else already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
in-order segment arrival, no gaps, one delayed ACK pending	immediately send single cumulative ACK
out-of-order segment arrival higher-than-expect seq. # gap detected	send duplicate ACK, indicating seq. # of next expected byte
arrival of segment that partially or completely fills gap	immediate ACK if segment starts at lower end of gap

TCP: retransmission scenarios



lost ACK scenario



premature timeout,
cumulative ACKs

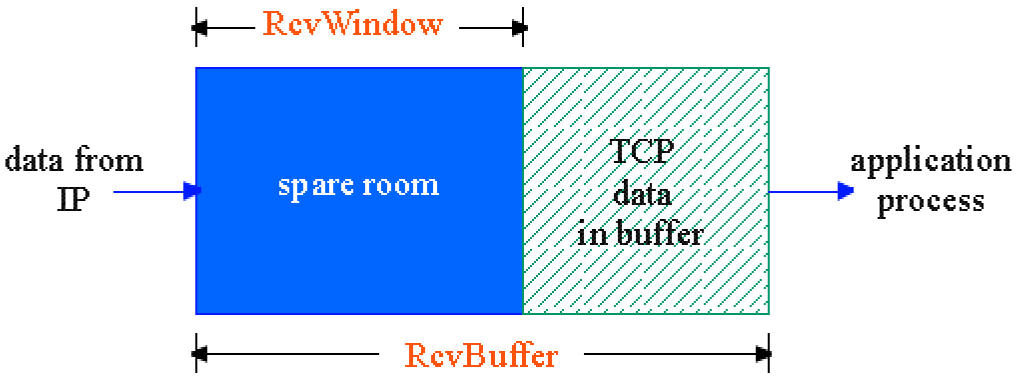
TCP Flow Control

flow control

sender won't overrun receiver's buffers by transmitting too much, too fast

RcvBuffer = size or TCP Receive Buffer

RcvWindow = amount of spare room in Buffer



receiver buffering

receiver: explicitly informs sender of (dynamically changing) amount of free buffer space

- **RcvWindow field** in TCP segment

sender: keeps the amount of transmitted, unACKed data less than most recently received **RcvWindow**

TCP Round Trip Time and Timeout

Q: how to set TCP timeout value?

- longer than RTT
  - note: RTT will vary
- 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, cumulatively ACKed segments
- **`SampleRTT`** will vary, want estimated RTT "smoother"
  - use several recent measurements, not just current **`SampleRTT`**

TCP Round Trip Time and Timeout

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

- Exponential weighted moving average
- influence of given sample decreases exponentially fast
- typical value of x: 0.1

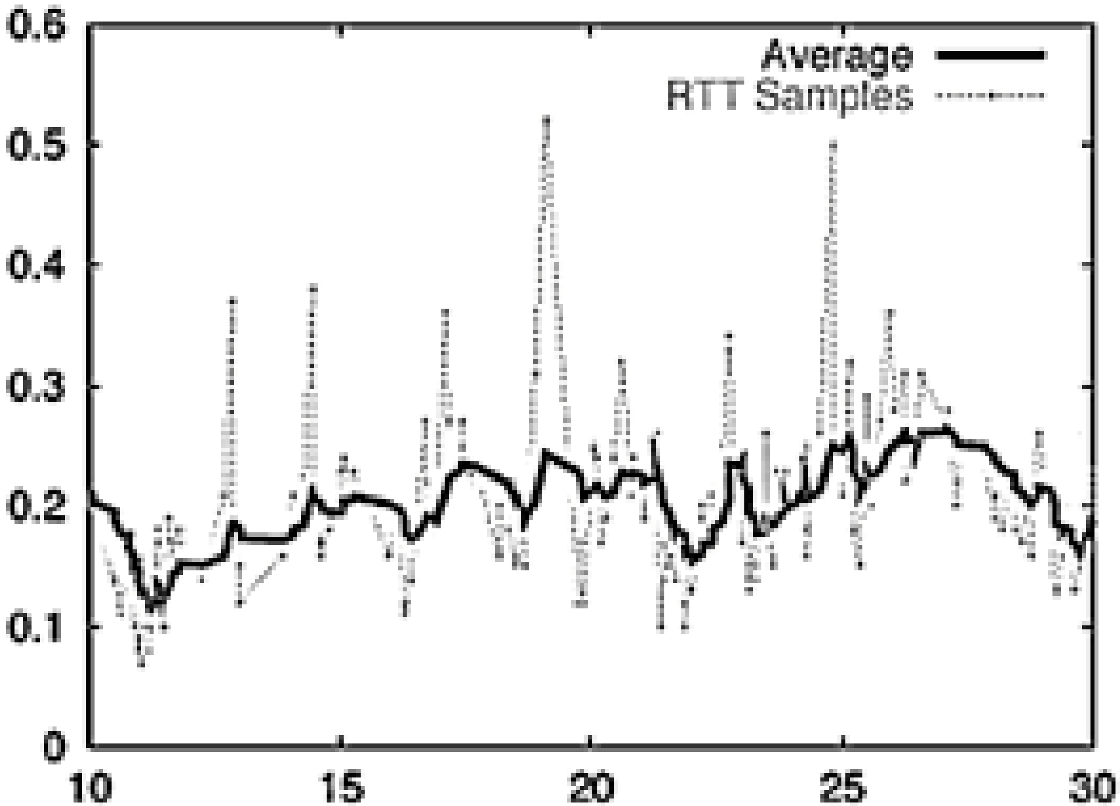
Setting the timeout

- **EstimtedRTT** plus "safety margin"
- large variation in **EstimatedRTT ->** larger safety margin

```
Timeout = EstimatedRTT + 4*Deviation

Deviation = (1-x)*Deviation +
            x*|SampleRTT-EstimatedRTT|
```

RTT samples/estimate



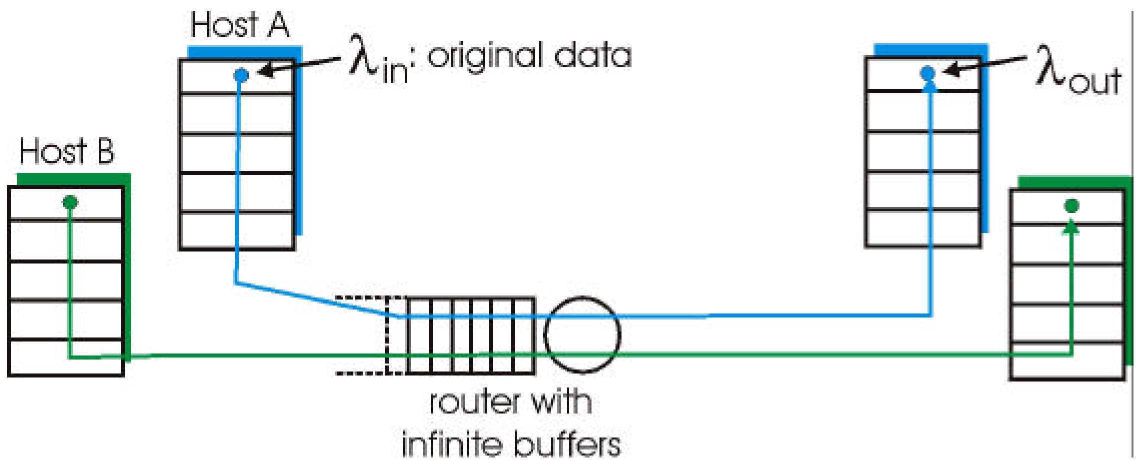
Principles of Congestion Control

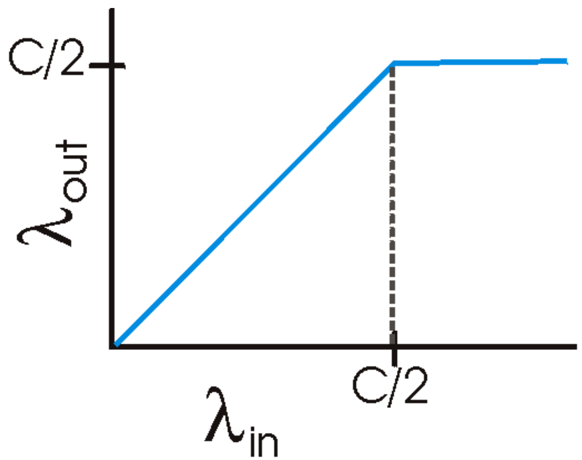
Congestion:

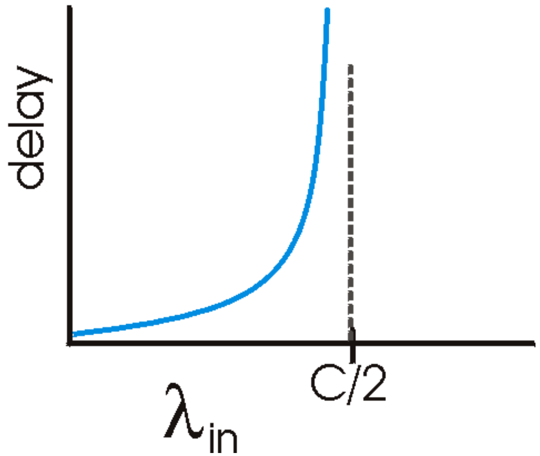
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- no retransmission



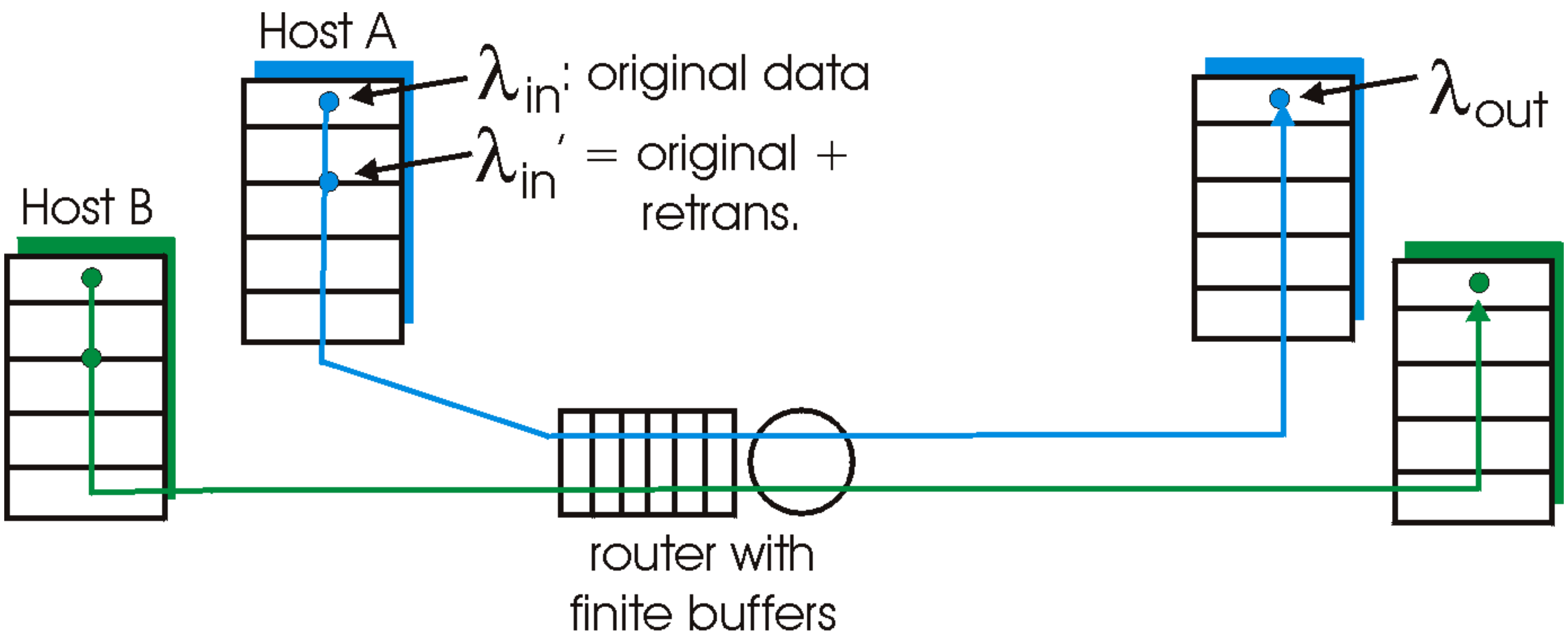




- large delays when congested
- maximum achievable throughput

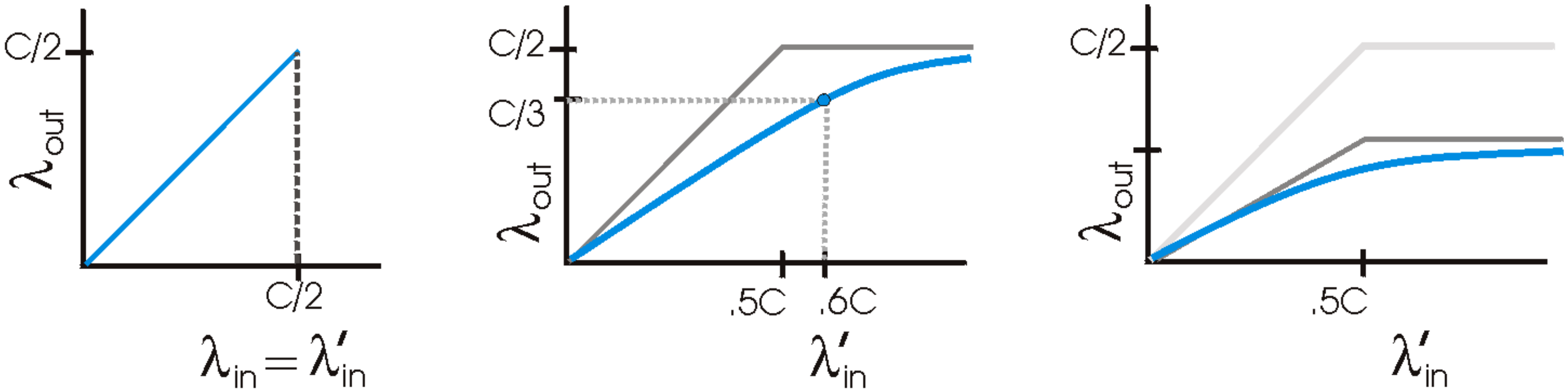
Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmission of lost packet



Causes/costs of congestion: scenario 2

- always: $\lambda_{in} = \lambda_{out}$ (goodput)
- "perfect" retransmission only when loss: $\lambda'_{in} > \lambda_{out}$
- retransmission of delayed (not lost) packet makes λ'_{in} larger (than perfect case) for same λ_{out}



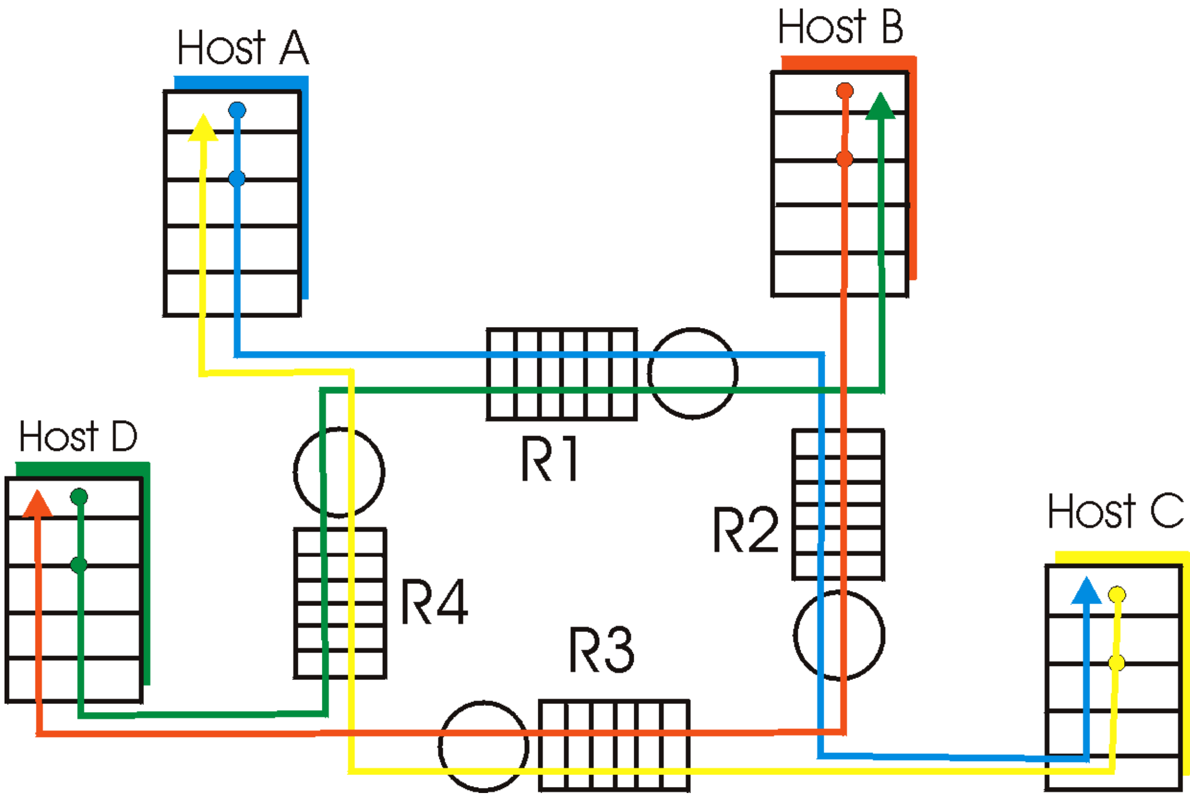
"costs" of congestion:

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

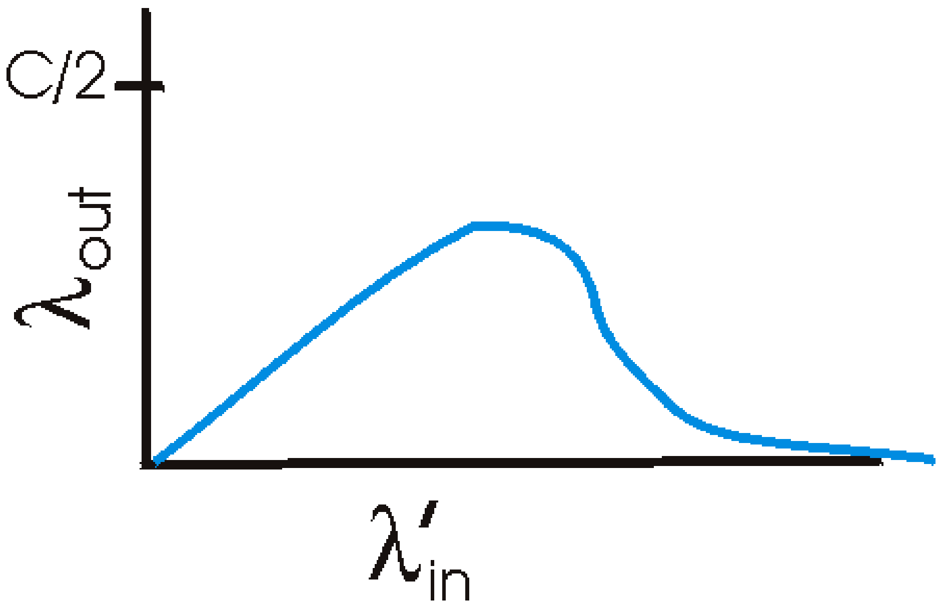
Causes/costs of congestion: scenario 3

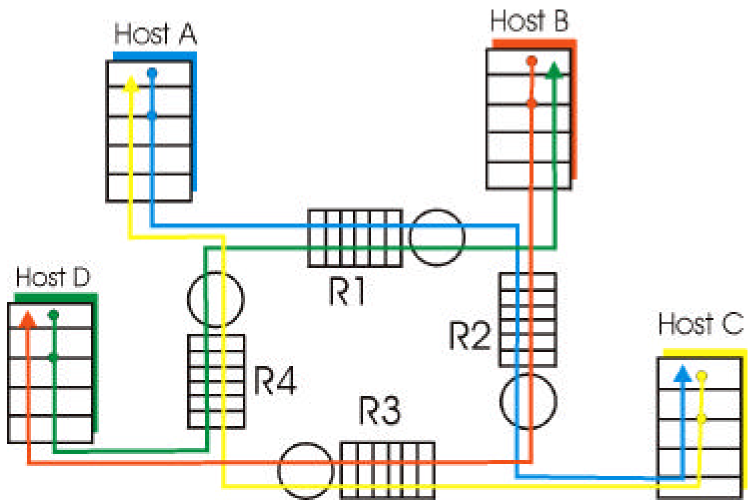
- four senders
- multihop paths
- timeout/retransmit

Q: what happens as λ_{in} and λ'_{in} increase ?



Causes/costs of congestion: scenario 3





Another "cost" of congestion:

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

Approaches towards congestion control

Two broad approaches towards congestion control:

End-end congestion control:

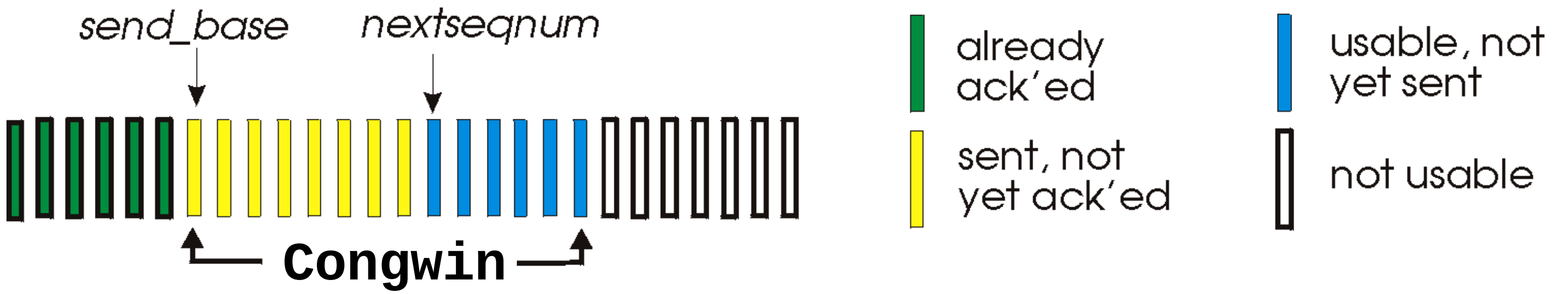
- no explicit feedback from network
- congestion inferred from end-system observed loss, delay
- approach taken by TCP

Network-assisted congestion control:

- routers provide feedback to end systems
  - single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
  - explicit rate sender should send at

TCP Congestion Control

- end-end control (no network assistance)
- transmission rate limited by congestion window size, **`Congwin`**, over segments:



- w segments, each with MSS bytes sent in one RTT:

$$\text{throughput} = \frac{w * MSS}{RTT} \text{ Bytes/sec}$$

TCP congestion control:

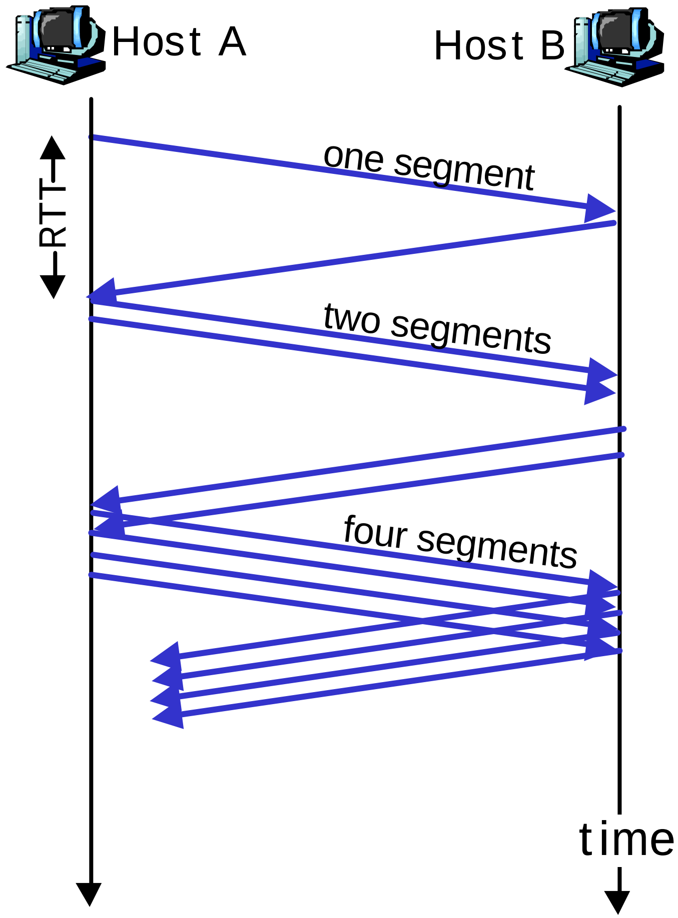
- "probing" for usable bandwidth:
  - ideally: transmit as fast as possible (**`Congwin`** as large as possible) without loss
  - *increase* **`Congwin`** until loss (congestion)
  - loss: *decrease* **`Congwin`**, then begin probing (increasing) again
- two "phases"
  - slow start
  - congestion avoidance
- important variables:
  - **`Congwin`**
  - **`threshold:`** defines threshold between two slow start phase, congestion control phase

TCP Slowstart

Slowstart algorithm

initialize: Congwin = 1
for (each segment ACKed)
    Congwin++
until (loss event OR
    CongWin > threshold)

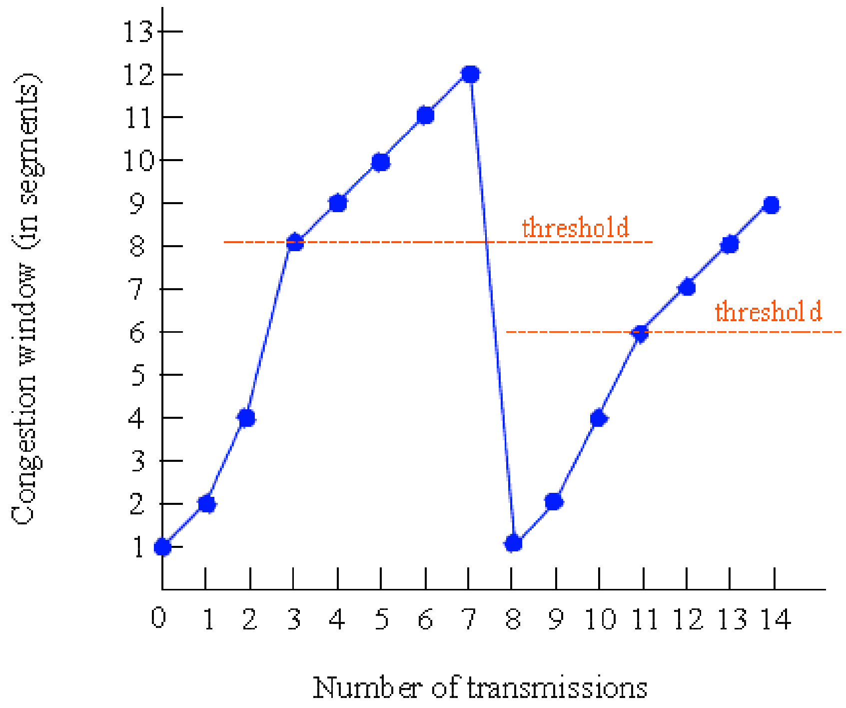
- exponential increase (per RTT) in window size (not so slow!)
- loss event: timeout (Tahoe TCP) and/or or three duplicate ACKs (Reno TCP)



TCP Congestion Avoidance

Congestion avoidance

```
/* slowstart is over       */
/* Congwin > threshold */
Until (loss event) {
  every w segments ACKed:
      Congwin++
  }
threshold = Congwin/2
Congwin = 1
perform slowstart[1]
```



1: TCP Reno skips slowstart (fast recovery) after three duplicate ACKs

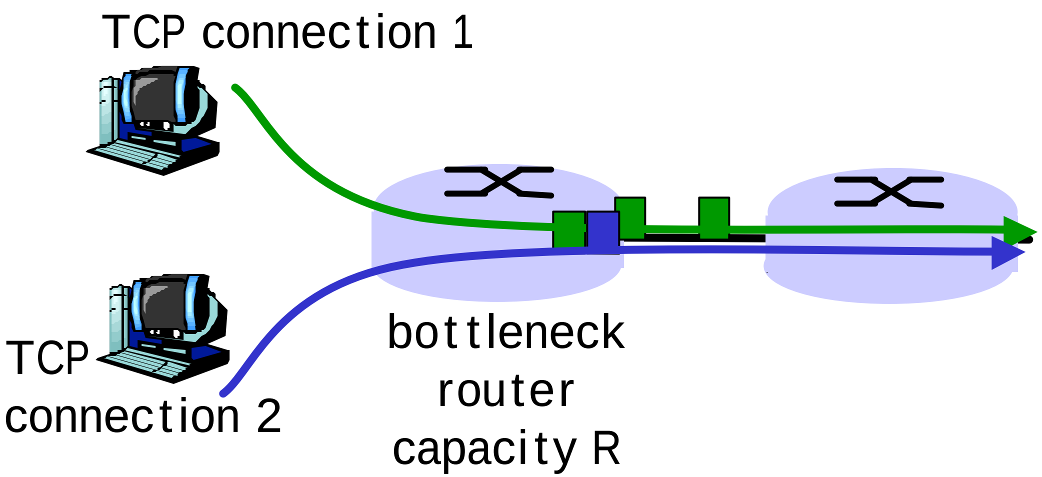
AIMD

TCP congestion avoidance:

- AIMD: *additive increase, multiplicative decrease*
  - increase window by 1 per RTT
  - decrease window by factor of 2 on loss event

TCP Fairness

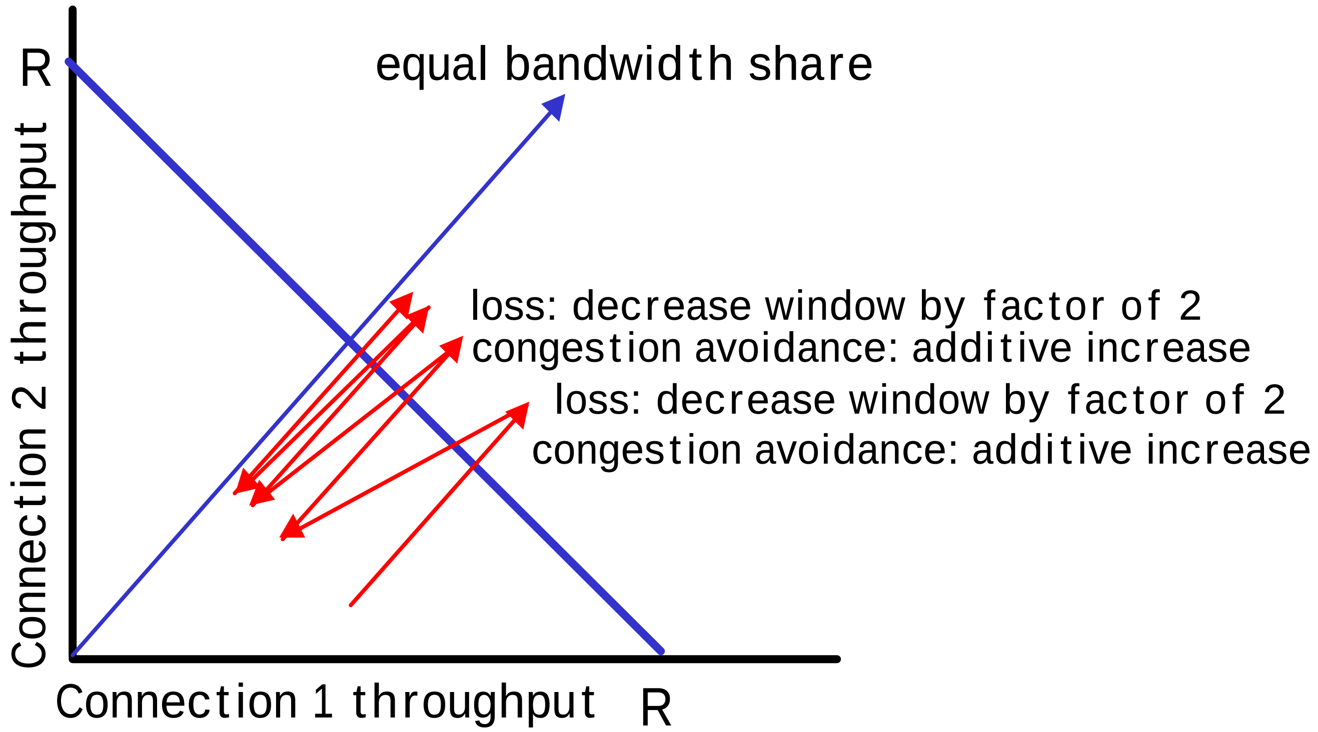
Fairness goal: if N TCP sessions share same bottleneck link, each should get 1/N of link capacity



Why is TCP fair?

Two competing sessions:

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



TCP latency modeling

Q: How long does it take to receive an object from a Web server after sending a request?

- TCP connection establishment
- data transfer delay

Notation, assumptions:

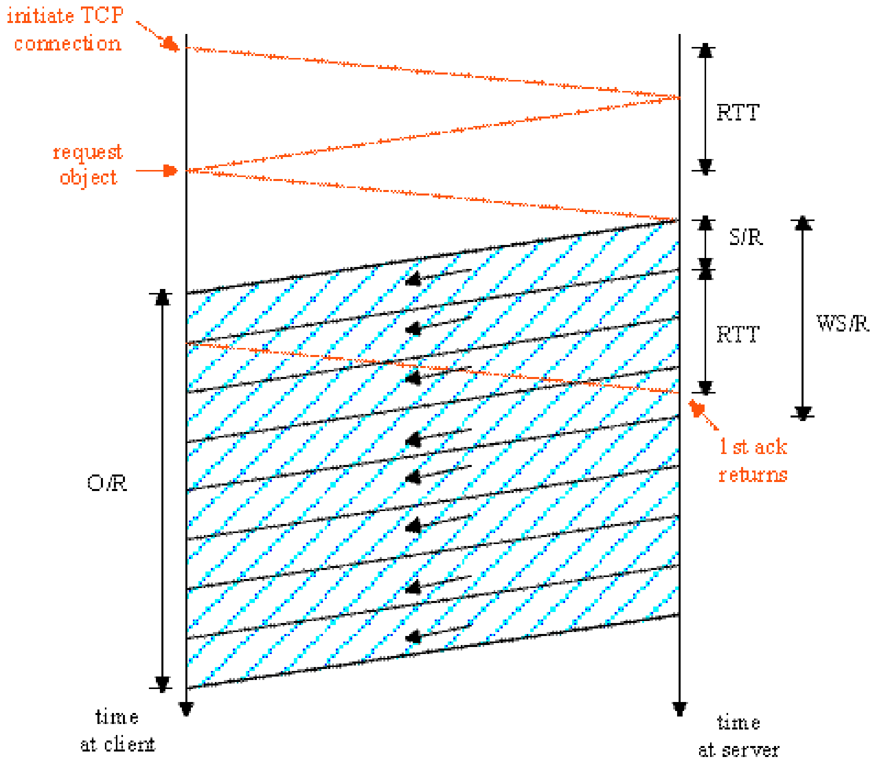
- Assume one link between client and server of rate R
- Assume: fixed congestion window, W segments
- S: MSS (bits)
- O: object size (bits)
- no retransmissions (no loss, no corruption)

Two cases to consider:

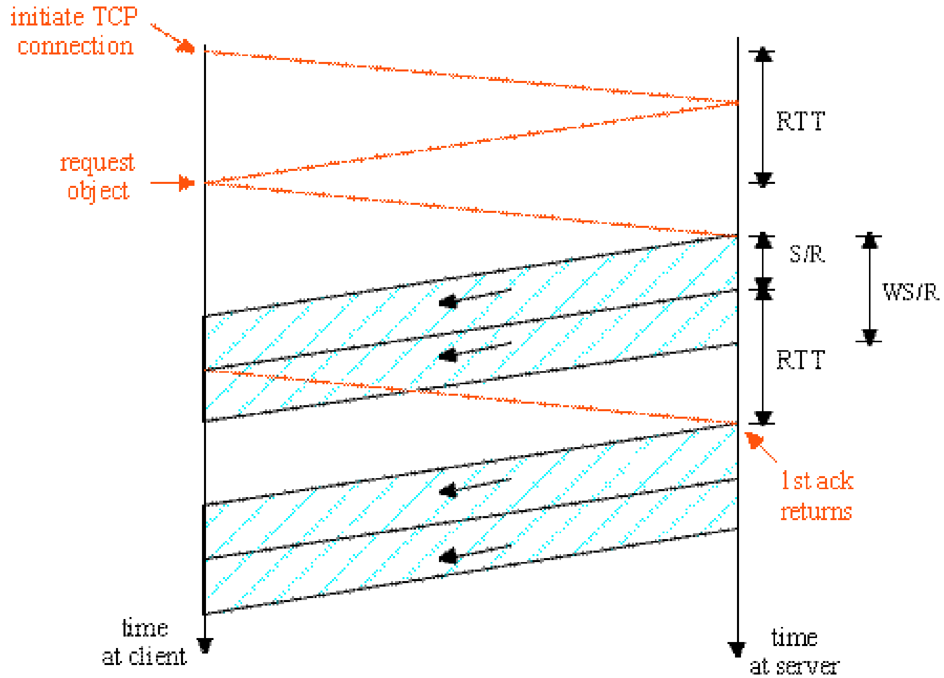
- WS/R > RTT + S/R: ACK for first segment in window returns before window's worth of data sent
- WS/R < RTT + S/R: wait for ACK after sending window's worth of data sent

TCP latency Modeling

K:= O/WS



Case 1: latency = 2RTT + O/R



Case 2: latency = 2RTT + O/R + (K-1)[S/R + RTT - WS/R]

TCP Latency Modeling: Slow Start

- Now suppose window grows according to slow start.
- Will show that the latency of one object of size O is:

$$Latency = 2RTT + \frac{O}{R} + P\left[RTT + \frac{S}{R}\right] - (2^P - 1)\frac{S}{R}$$

where *P* is the number of times TCP stalls at server:

$$P = \min\{Q, K-1\}$$

- where Q is the number of times the server would stall if the object were of infinite size.

- and K is the number of windows that cover the object.

TCP Latency Modeling: Slow Start (cont.)

Example:

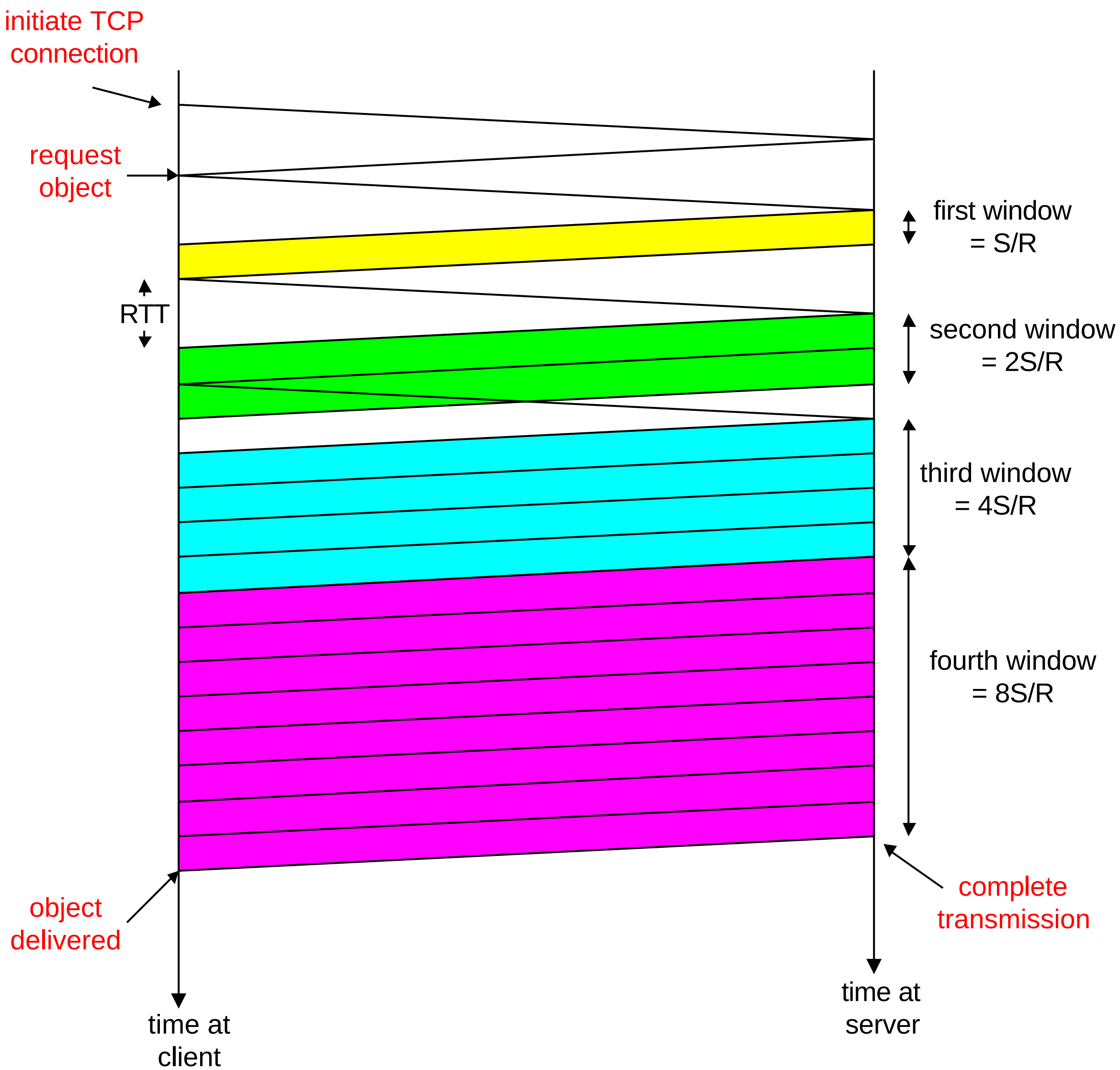
O/S = 15 segments

K = 4 windows

Q = 2

P = min{K-1,Q} = 2

Server stalls P=2 times.



TCP Latency Modeling: Slow Start (cont.)

$$\frac{S}{R} + RTT = \text{time from when server starts to send segment until server receives acknowledgement}$$

$$2^{k-1}\frac{S}{R} = \text{time to transmit the kth window}$$

$$\left[\frac{S}{R} + RTT - 2^{k-1}\frac{S}{R}\right]^{+} = \text{stall time after the } k\text{th window}$$

$$\text{latency} = \frac{O}{R} + 2RTT + \sum_{p=1}^{P} stallTime_p$$

$$= \frac{O}{R} + 2RTT + \sum_{k=1}^{P}\left[\frac{S}{R} + RTT - 2^{k-1}\frac{S}{R}\right]$$

$$= \frac{O}{R} + 2RTT + P\left[RTT + \frac{S}{R}\right] - (2^{P} - 1)\frac{S}{R}$$