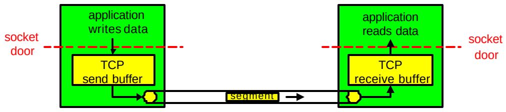
TCP: Overview

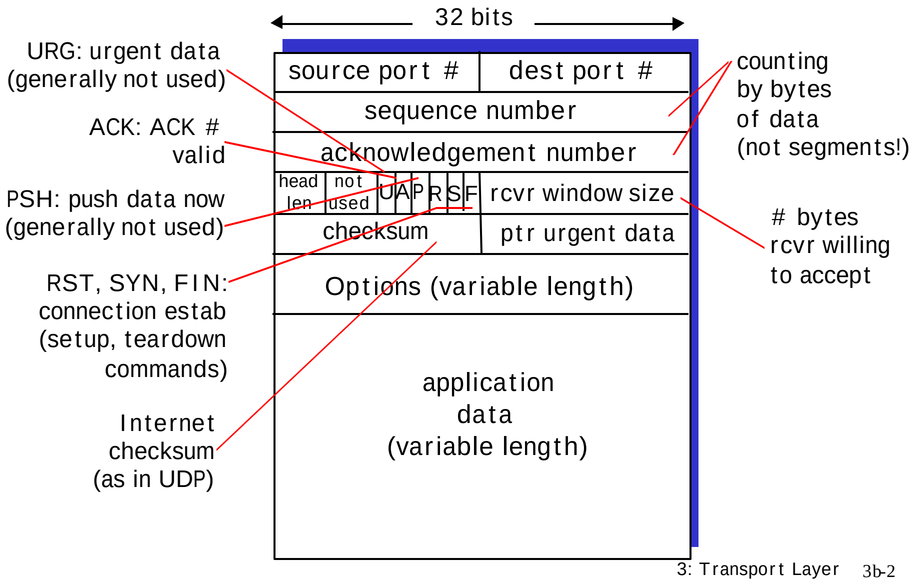
RFCs: 793, 1122, 1323, 2018, 2581

- 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



URG: urgent data (generally not used)

ACK: ACK # valid

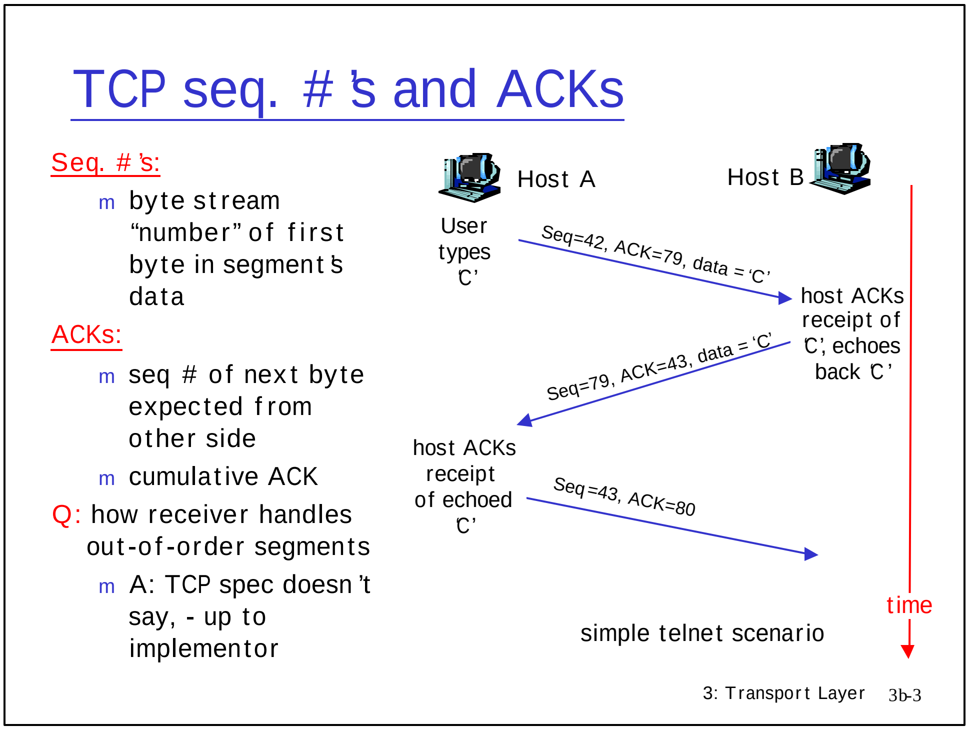
PSH: push data now (generally not used)

RST, SYN, FIN: connection estab (setup, teardown commands)

Internet checksum (as in UDP)

counting by bytes of data (not segments!)

bytes rcvr willing to accept



TCP seq. #’s and ACKs

Seq. #’s:

- byte stream “number” of first byte in segment’s data

ACKs:

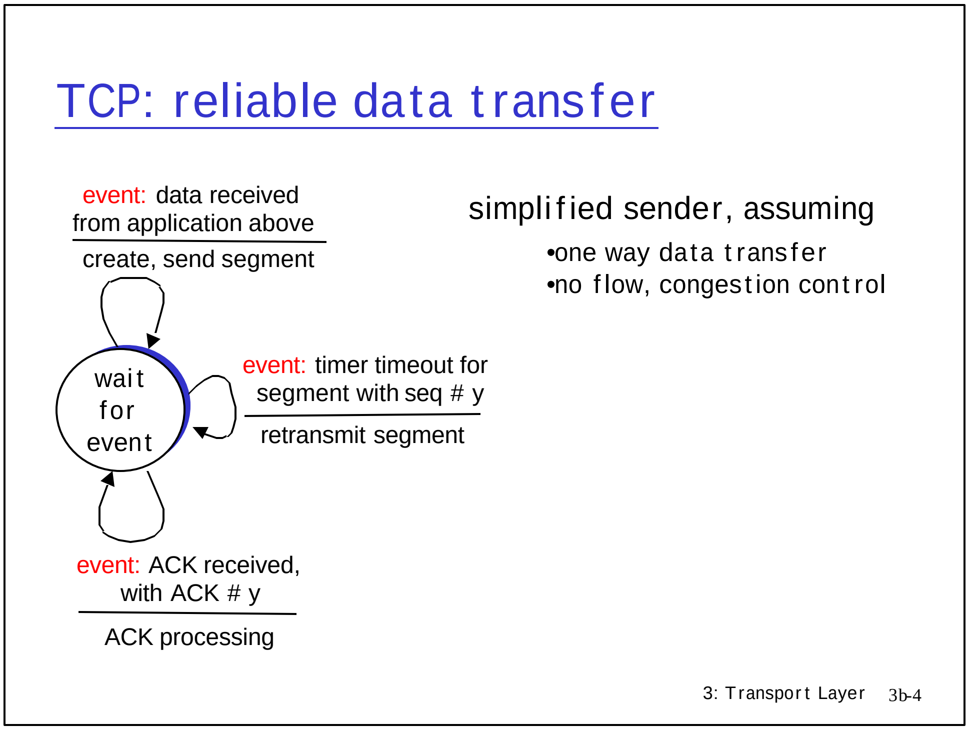
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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

Host A
Host B
User types ‘C’
Seq=42, ACK=79, data = ‘C’
host ACKs receipt of ‘C’, echoes back ‘C’
Seq=79, ACK=43, data = ‘C’
host ACKs receipt of echoed ‘C’
Seq=43, ACK=80
time

simple telnet scenario

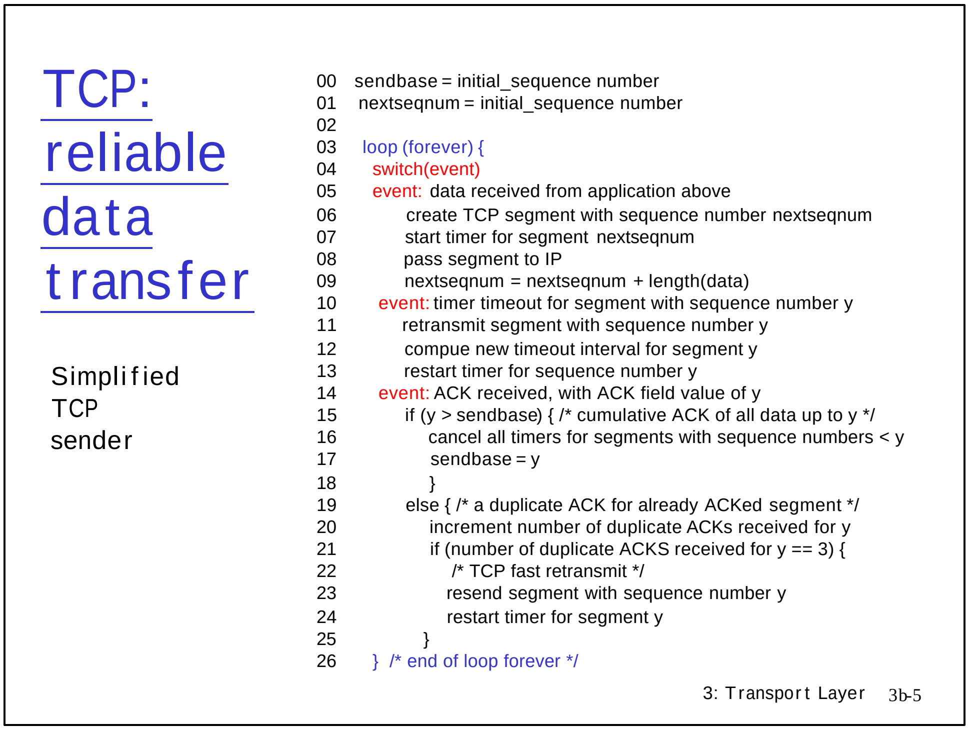


TCP: reliable data transfer

simplified sender, assuming

- one way data transfer
- no flow, congestion control

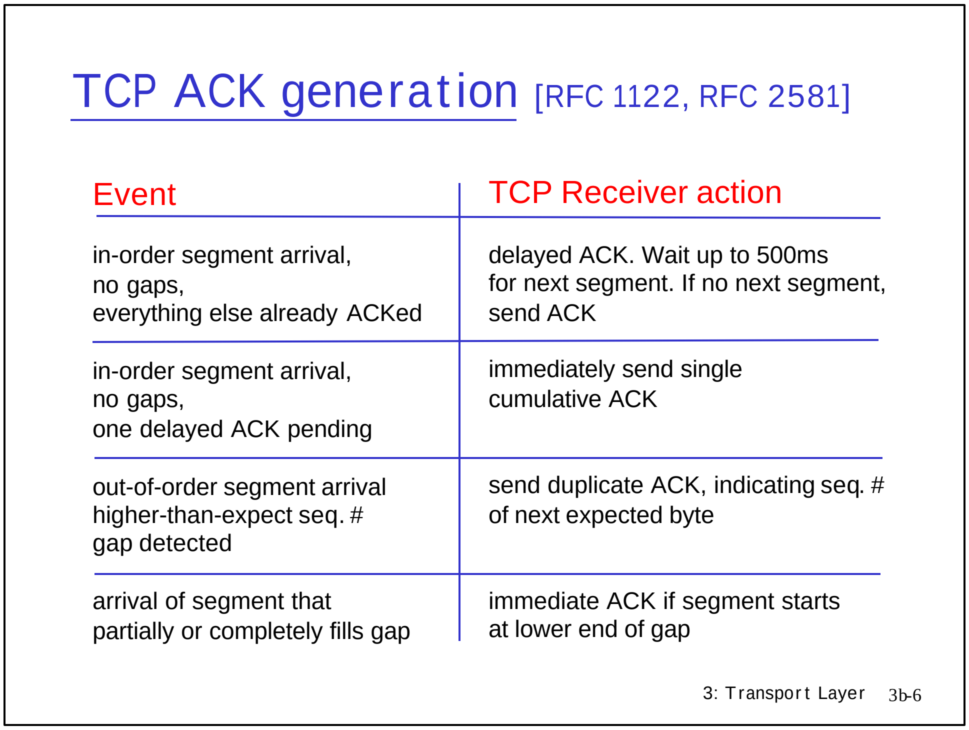
event: data received from application above
create, send segment
wait for event
event: timer timeout for segment with seq # y
retransmit segment
event: ACK received, with ACK # y
ACK processing



TCP: reliable data transfer

Simplified TCP sender

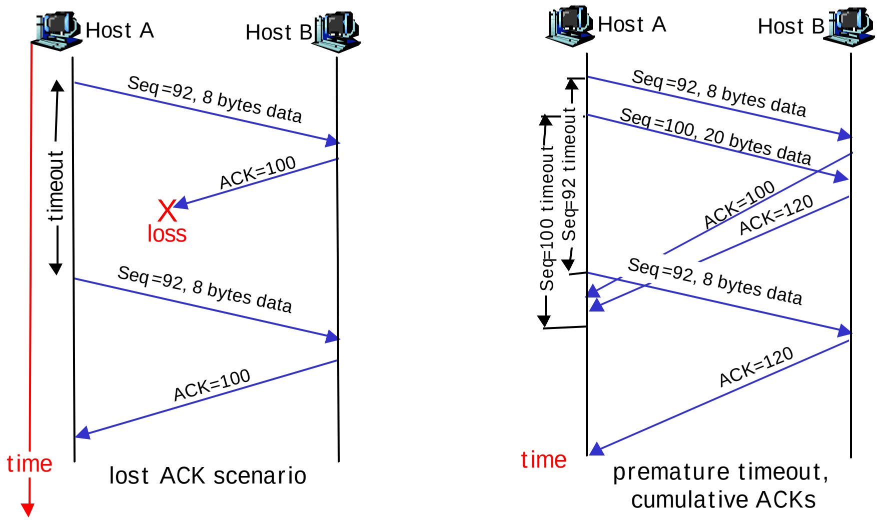
```
sendbase = initial_sequence number
 nextseqnum = initial_sequence number

 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

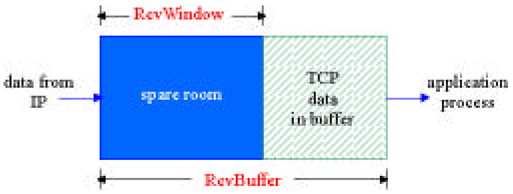
TCP Flow Control

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

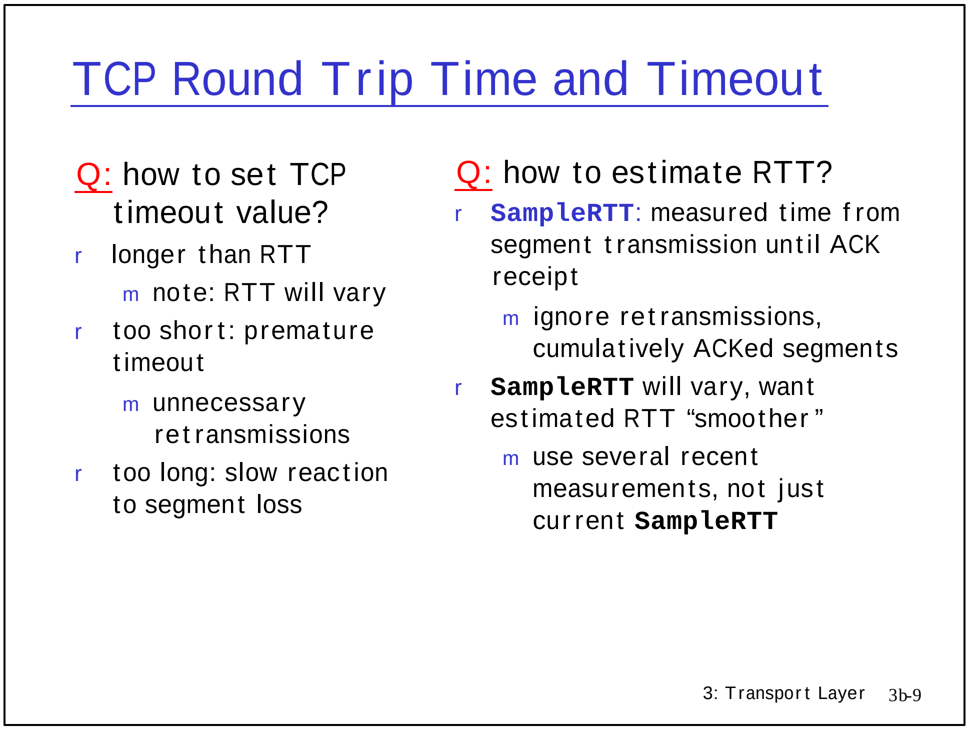


receiver buffering

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`**



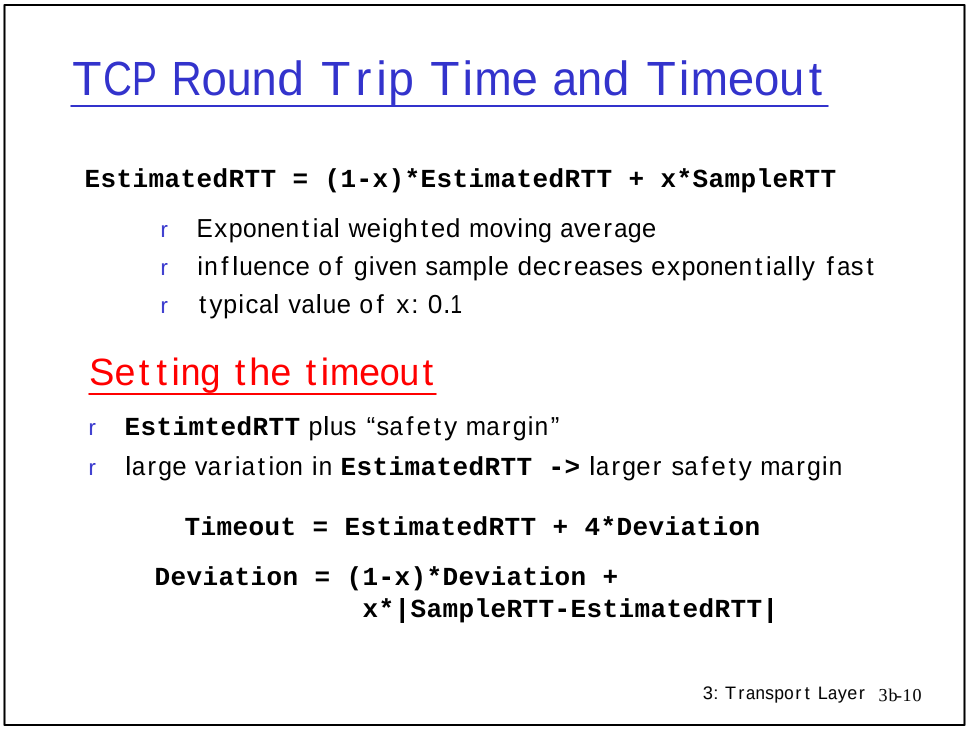
TCP Round Trip Time and Timeout

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|
```

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 end system sends TCP SYN control segment to server

- specifies initial seq #

Step 2: server end system receives SYN, replies with SYNACK control segment

- ACKs received SYN
- allocates buffers
- specifies server-> receiver initial seq. #

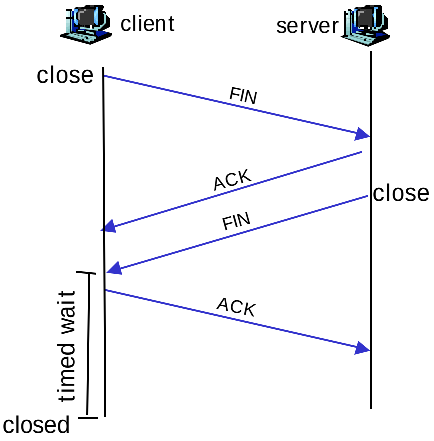
TCP Connection Management (cont.)

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.



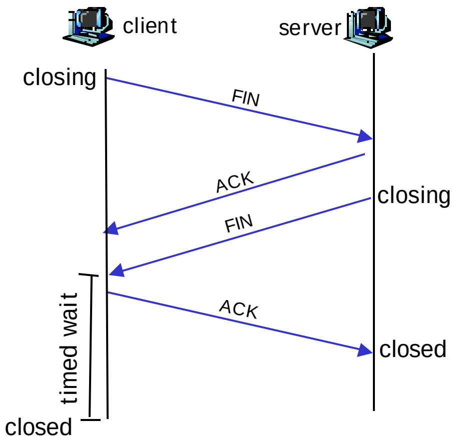
TCP Connection Management (cont.)

Step 3: client receives FIN, replies with ACK.

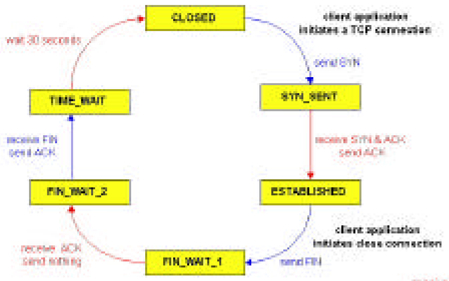
- Enters "timed wait" - will respond with ACK to received FINs

Step 4: server, receives ACK. Connection closed.

Note: with small modification, can handly simultaneous FINs.

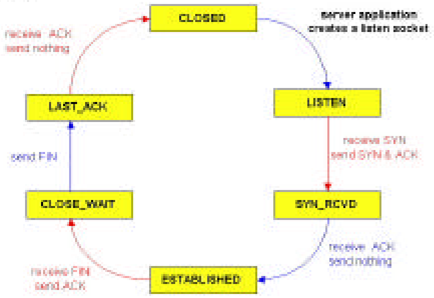


TCP Connection Management (cont)



TCP client lifecycle

TCP server lifecycle



Principles of Congestion Control

Congestion:

r informally: "too many sources sending too much data too fast for *network* to handle"

r different from flow control!

r manifestations:

  m lost packets (buffer overflow at routers)

  m long delays (queueing in router buffers)

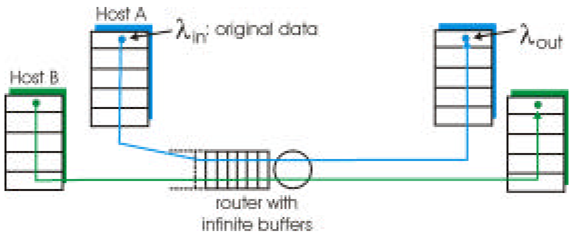
r a top-10 problem!

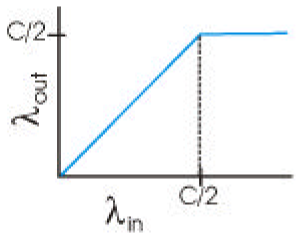
Causes/costs of congestion: scenario 1

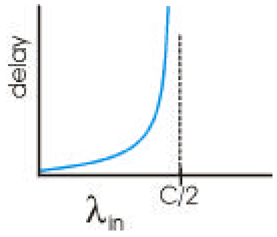
r two senders, two receivers

r one router, infinite buffers

r no retransmission

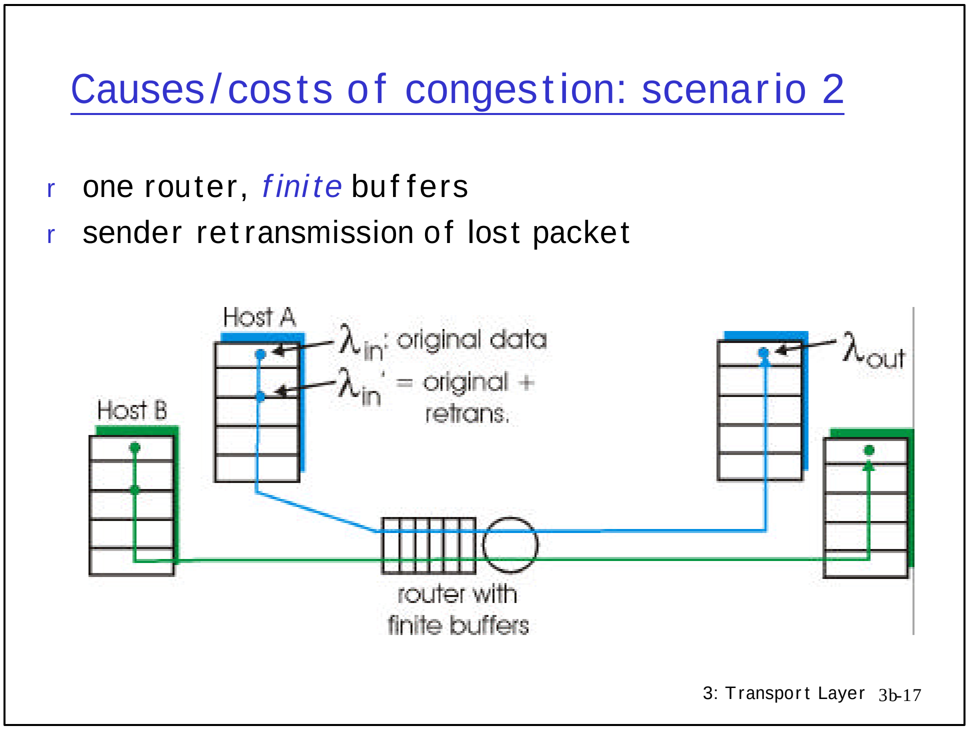






r large delays when congested

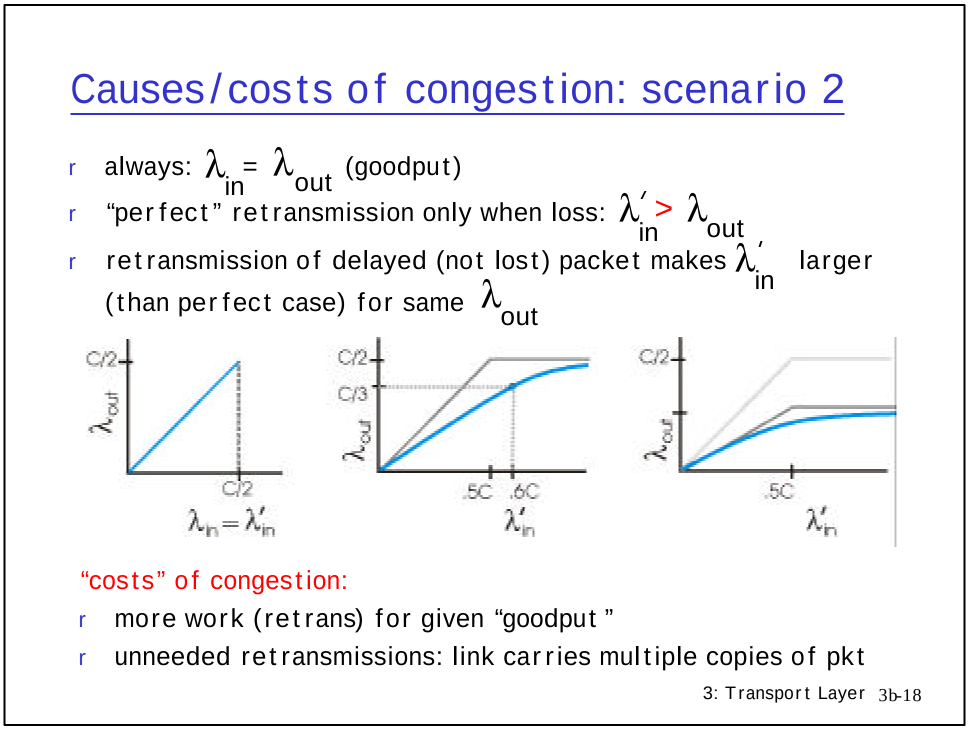
r maximum achievable throughput



Causes/costs of congestion: scenario 2

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

Host A
λ_{in}: original data
λ'_{in} = original + retrans.
λ_{out}
Host B
router with finite buffers

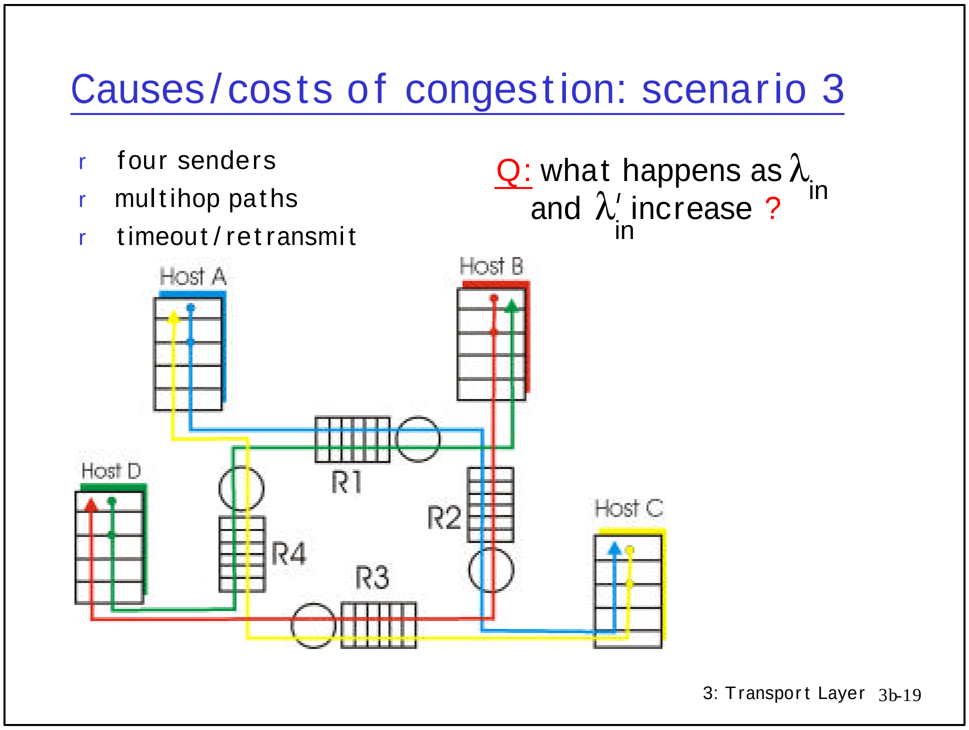


Causes/costs of congestion: scenario 2

- 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



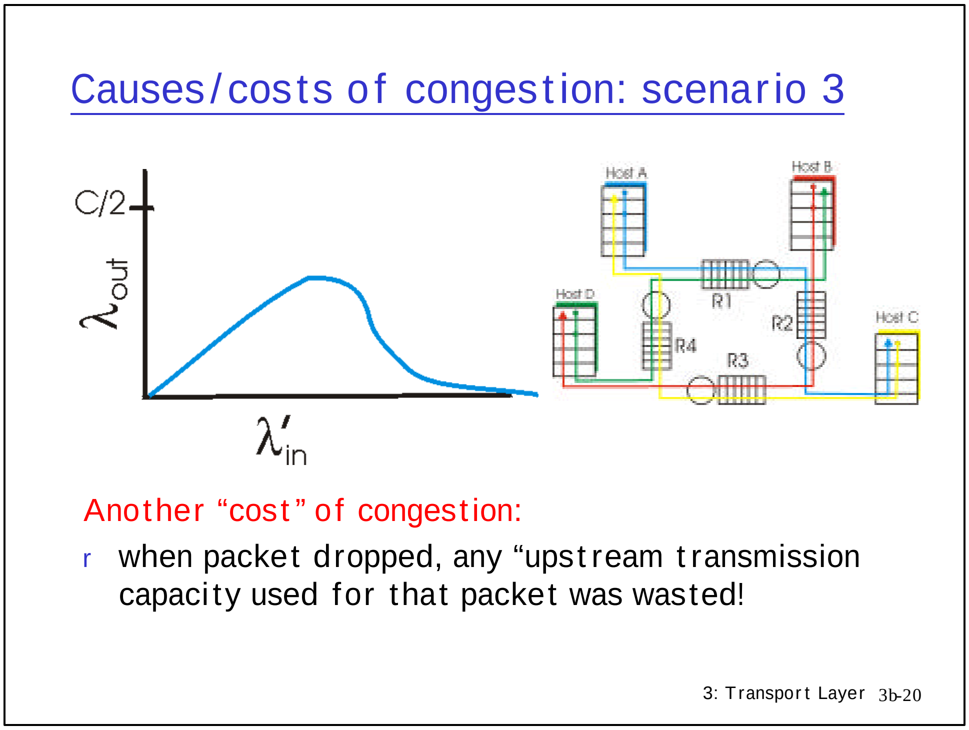
Causes/costs of congestion: scenario 3

r four senders

r multihop paths

r timeout/retransmit

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



Causes/costs of congestion: scenario 3

Another “cost” of congestion:

r 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

Case study: ATM ABR congestion control

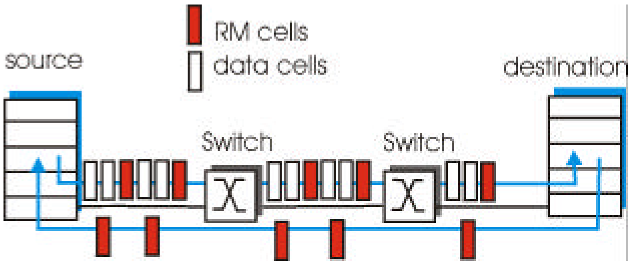
ABR: available bit rate:

- "elastic service"
- if sender's path "underloaded":
  - sender should use available bandwidth
- if sender's path congested:
  - sender throttled to minimum guaranteed rate

RM (resource management) cells:

- sent by sender, interspersed with data cells
- bits in RM cell set by switches ("*network-assisted*")
  - NI bit: no increase in rate (mild congestion)
  - CI bit: congestion indication
- RM cells returned to sender by receiver, with bits intact

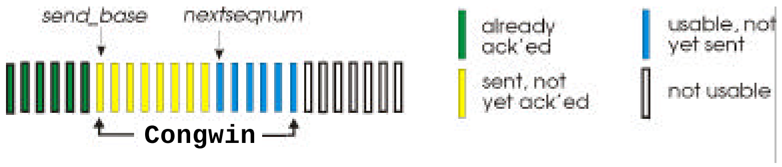
Case study: ATM ABR congestion control



- two-byte ER (explicit rate) field in RM cell
  - congested switch may lower ER value in cell
  - sender' send rate thus minimum supportable rate on path
- EFCI bit in data cells: set to 1 in congested switch
  - if data cell preceding RM cell has EFCI set, sender sets CI bit in returned RM cell

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:

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

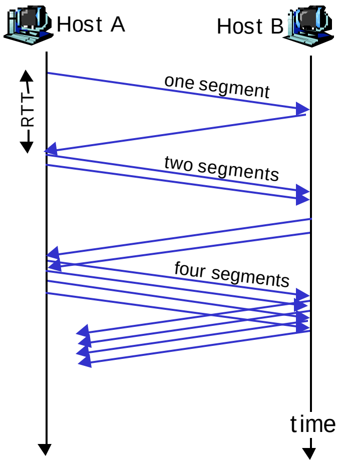
TCP congestion control:

- “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)
```

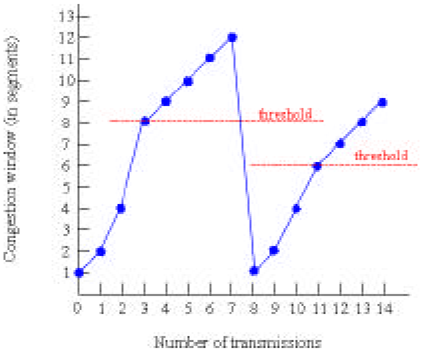


- exponential increase (per RTT) in window size (not so slow!)
- loss event: timeout (Tahoe TCP) and/or or three duplicate ACKs (Reno TCP)

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

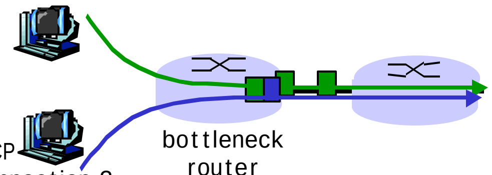
TCP congestion avoidance:

- r AIMD: *additive increase, multiplicative decrease*
  - m increase window by 1 per RTT
  - m 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

TCP connection 1



TCP connection 2

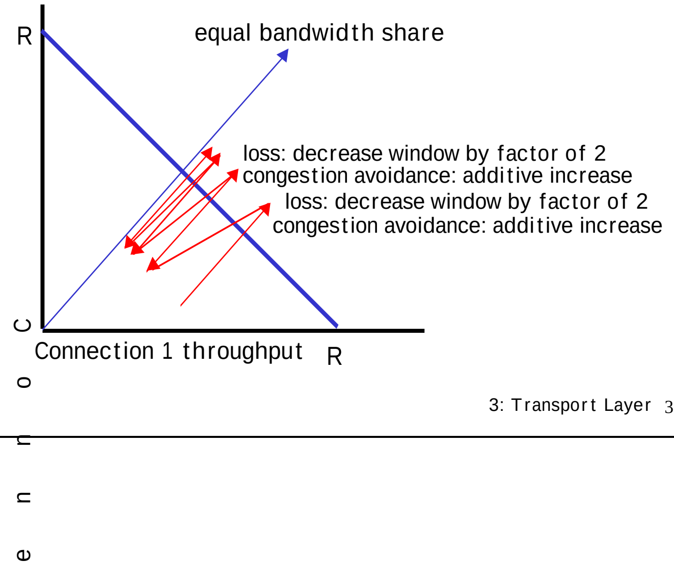
bottleneck router capacity R

Why is TCP fair?

Two competing sessions:

r Additive increase gives slope of 1, as throughout increases

r multiplicative decrease decreases throughput proportionally


R
equal bandwidth share
loss: decrease window by factor of 2
congestion avoidance: additive increase
loss: decrease window by factor of 2
congestion avoidance: additive increase
C o n n e
Connection 1 throughput
R

TCP latency modeling

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

r TCP connection establishment

r data transfer delay

Notation, assumptions:

r Assume one link between client and server of rate R

r Assume: fixed congestion window, W segments

r S: MSS (bits)

r O: object size (bits)

r no retransmissions (no loss, no corruption)

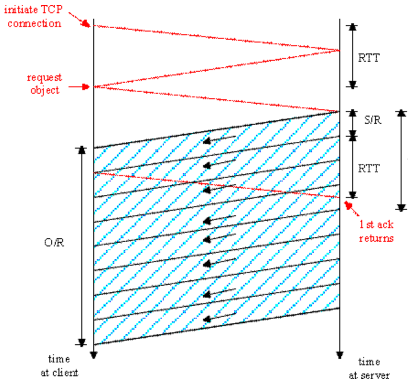
Two cases to consider:

r WS/R > RTT + S/R: ACK for first segment in window returns before window's worth of data sent

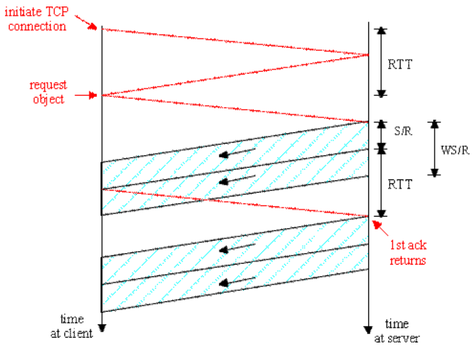
r WS/R < RTT + S/R: wait for ACK after sending window's worth of data sent

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:

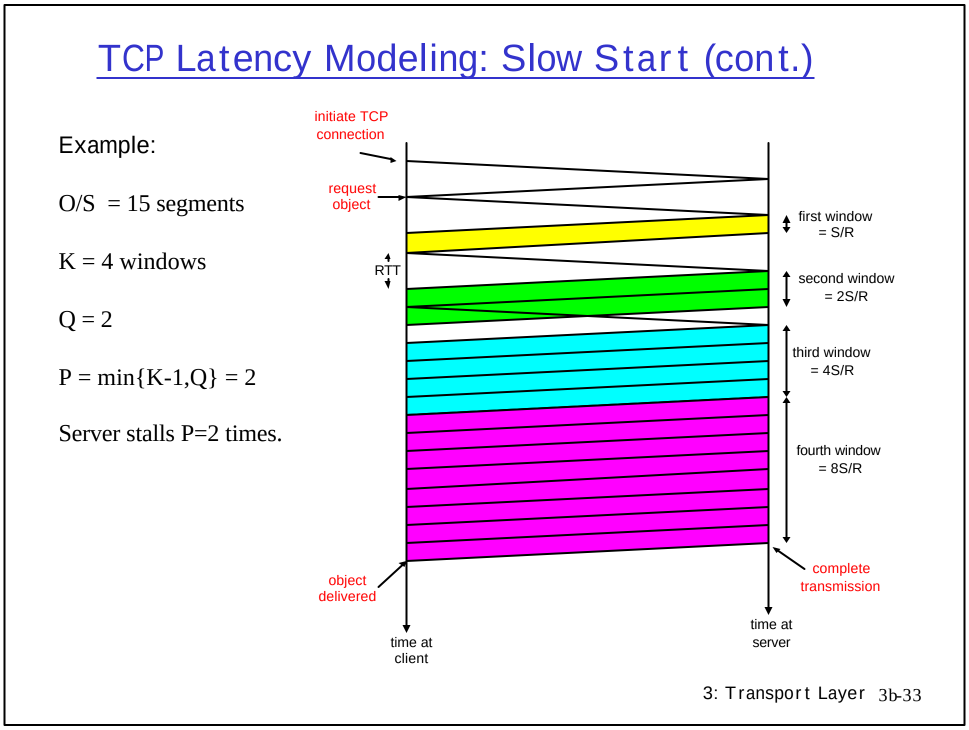
$$Latency = 2RTT + \frac{O}{R} + P\left[RTT + \frac{S}{R}\right] - (2^P - 1)\frac{S}{R}$$

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:

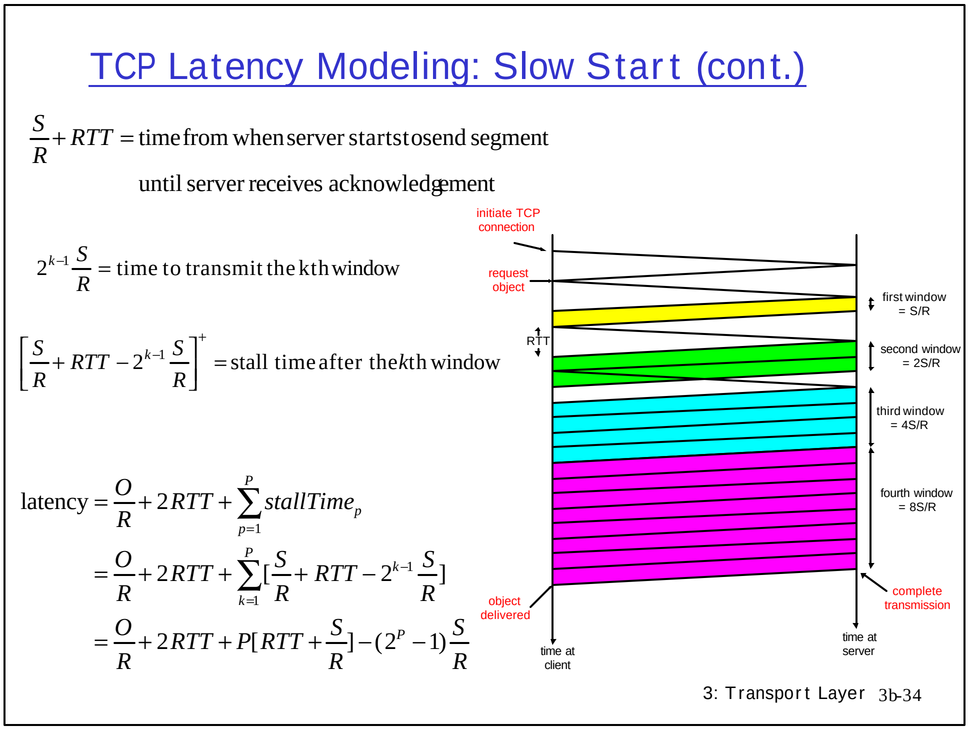
O/S = 15 segments

K = 4 windows

Q = 2

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

Server stalls P=2 times.



TCP Latency Modeling: Slow Start (cont.)

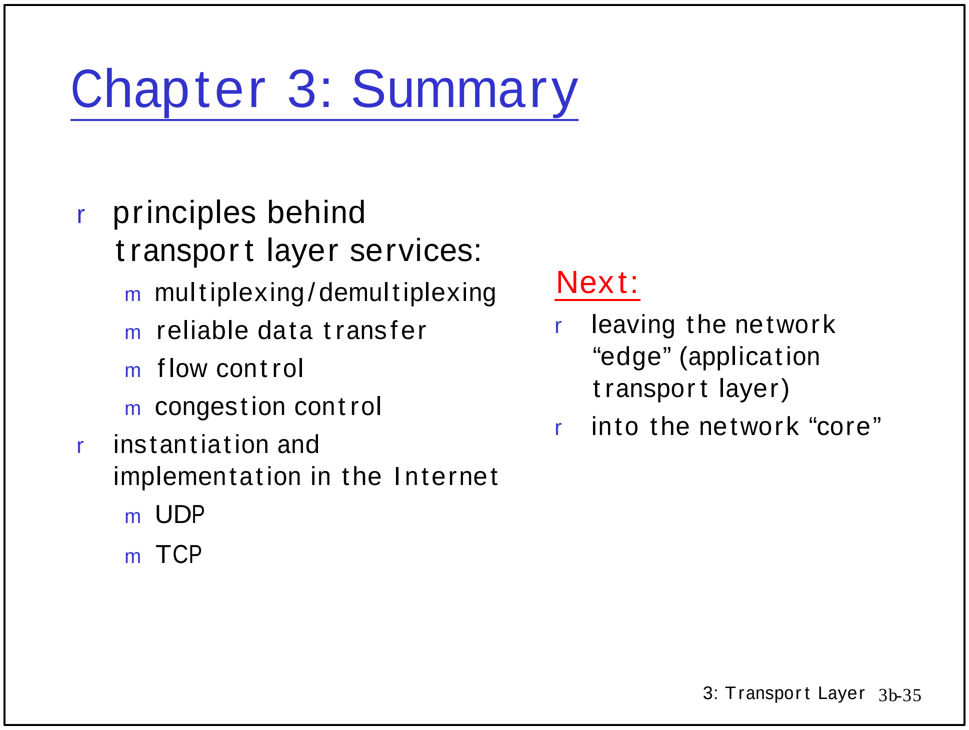
$$\frac{S}{R} + RTT = \text{time from when server starts to send segment}$$

$$\text{until server receives acknowledgement}$$

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

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

$$\begin{aligned}\text{latency} &= \frac{O}{R} + 2RTT + \sum_{p=1}^{P} stallTime_p \\ &= \frac{O}{R} + 2RTT + \sum_{k=1}^{P}[\frac{S}{R} + RTT - 2^{k-1}\frac{S}{R}] \\ &= \frac{O}{R} + 2RTT + P[RTT + \frac{S}{R}] - (2^P - 1)\frac{S}{R}\end{aligned}$$



Chapter 3: Summary

- principles behind transport layer services:
  - multiplexing/demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- instantiation and implementation in the Internet
  - UDP
  - TCP

Next:

- leaving the network “edge” (application transport layer)
- into the network “core”