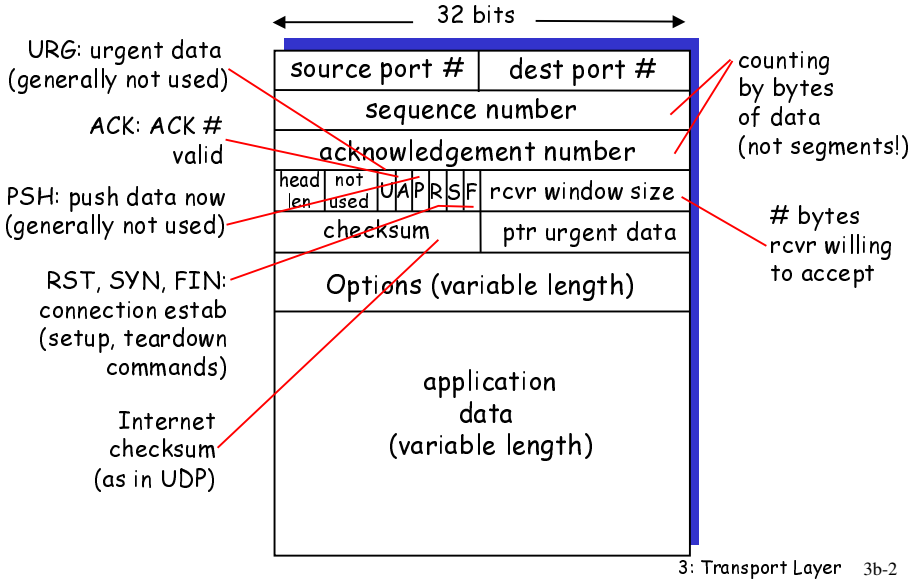
TCP: Overview

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) init's sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure

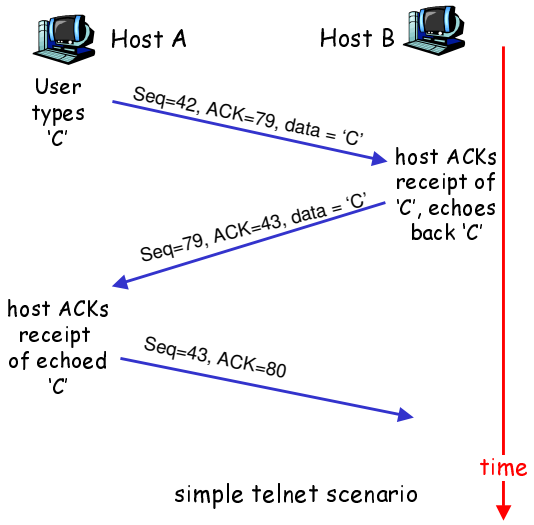


TCP seq. #'s and ACKs

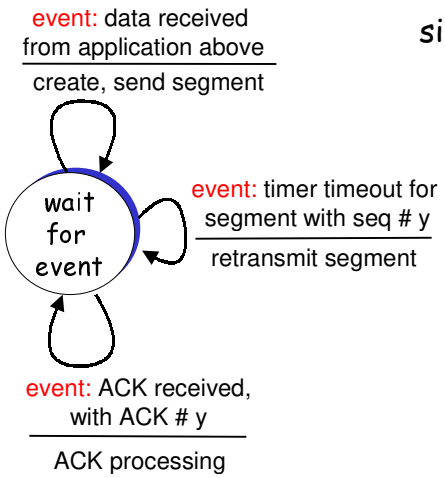
Seq. #'s:
- byte stream "number" of first byte in segment's data

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

Q: what receiver does w/ out-of-order segments
- TCP spec doesn't say, - up to TCP implementor



TCP: reliable data transfer



simplified sender, assuming
- one way data transfer
- no flow, congestion control

TCP: reliable data transfer

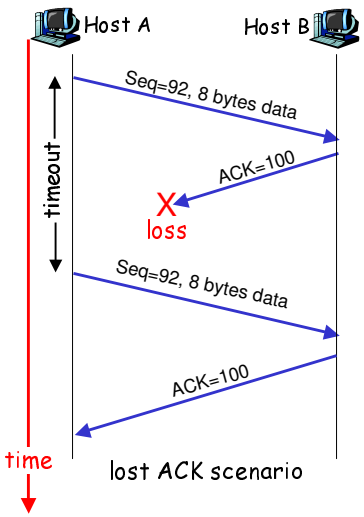
Simplified TCP sender

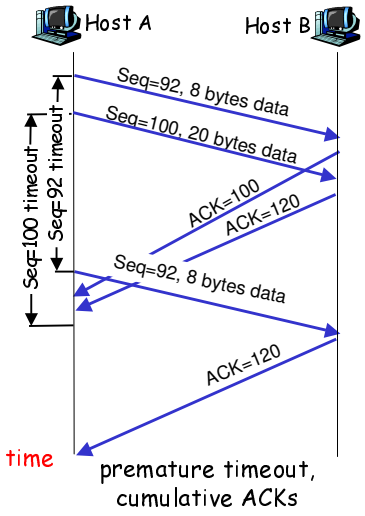
```
00  sendbase = initial_sequence number
01   nextseqnum = initial_sequence number
02
03    loop (forever) {
04      switch(event)
05      event: data received from application above
06           create TCP segment with sequence number nextseqnum
07           start timer for segment nextseqnum
08           pass segment to IP
09           nextseqnum = nextseqnum + length(data)
10       event: timer timeout for segment with sequence number y
11           retransmit segment with sequence number y
12           compute new timeout interval for segment y
13           restart timer for sequence number y
14       event: ACK received, with ACK field value of y
15           if (y > sendbase) { /* cumulative ACK of all data up to y */
16              cancel all timers for segments with sequence numbers < y
17               sendbase = y
18               }
19           else { /* a duplicate ACK for already ACKed segment */
20              increment number of duplicate ACKs received for y
21              if (number of duplicate ACKS received for y == 3) {
22                 /* TCP fast retransmit */
23                resend segment with sequence number y
24                restart timer for segment y
25              }
26      } /* 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 (missing) byte
arrival of segment that partially or completely fills gap	immediate ACK if segment starts at lower end of gap

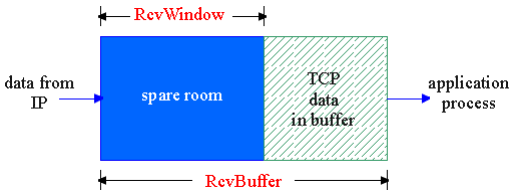
TCP: retransmission scenarios





TCP Flow Control

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



receiver buffering

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

- **`rcvr window size field`** in TCP segment

sender: amount of transmitted, unACKed data less than most recently-received **`rcvr window size`**

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
  - or timestamp pkts
- **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

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

```
Timeout = EstimatedRTT + 4*Deviation

Deviation = (1-x)*Deviation +
            x*abs(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. #

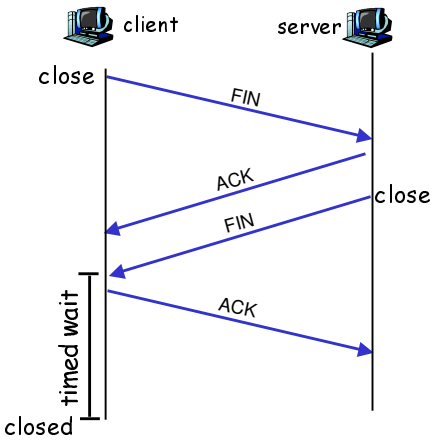
TCP Connection Management (cont.)

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.



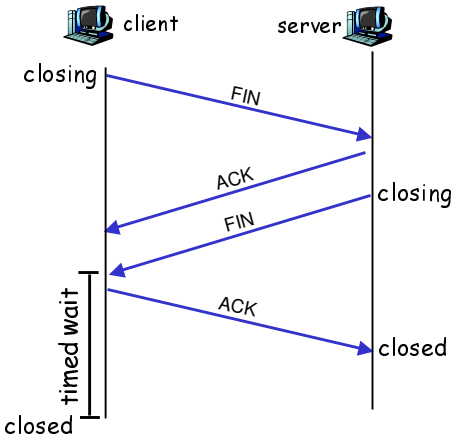
TCP Connection Management (cont.)

Step 3: client receives FIN, replies with ACK.

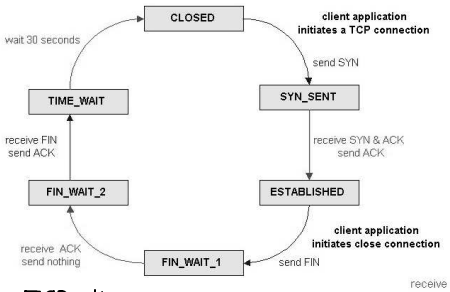
- Enters "timed wait" - will respond with ACK to received FINs

Step 4: server, receives ACK. Connection closed.

Note: with small modification, can handle simultaneous FINs.

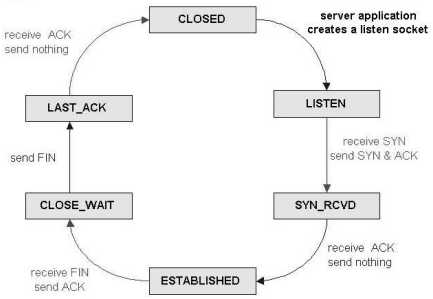


TCP Connection Management (cont)



TCP client lifecycle

TCP server lifecycle



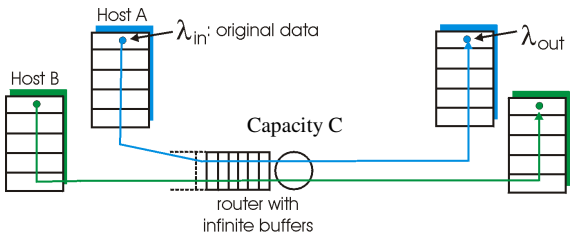
Principles of Congestion Control

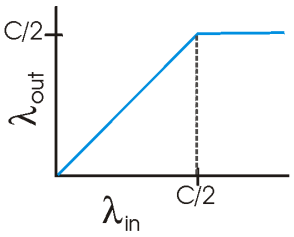
Congestion:

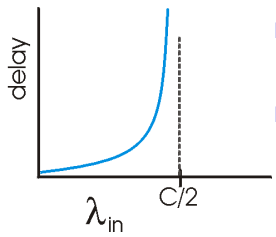
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

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



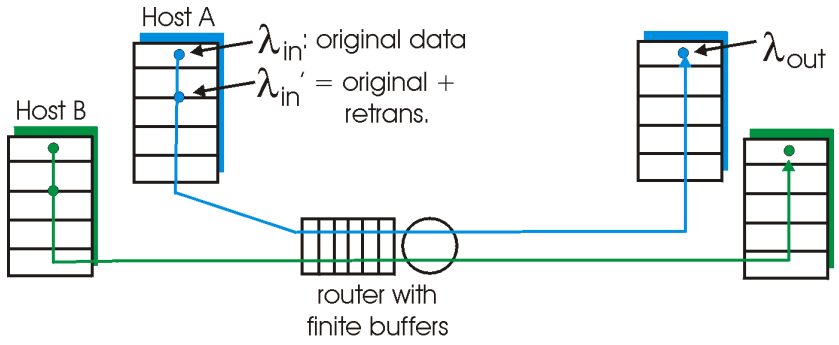




- large delays when congested
- maximum achievable throughput

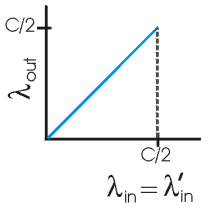
Causes/costs of congestion: scenario 2

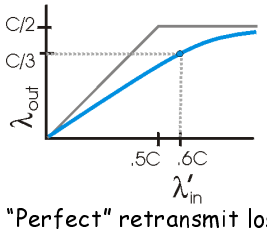
- one router, *finite* buffers
- sender retransmission of lost packet



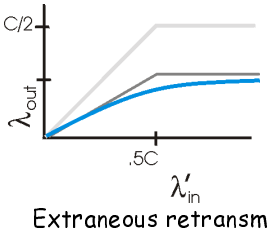
Causes/costs of congestion: scenario 2

- always: $\lambda_{in} = \lambda_{out}$ (goodput), but can have $\lambda_{in}' > \lambda_{out}$
- "perfect" when only retransmit lost pkts
- retransmission of delayed (not lost) packet makes λ_{in}' larger (than perfect case) for same λ_{out}





"Perfect" retransmit lost



Extraneous retransmit

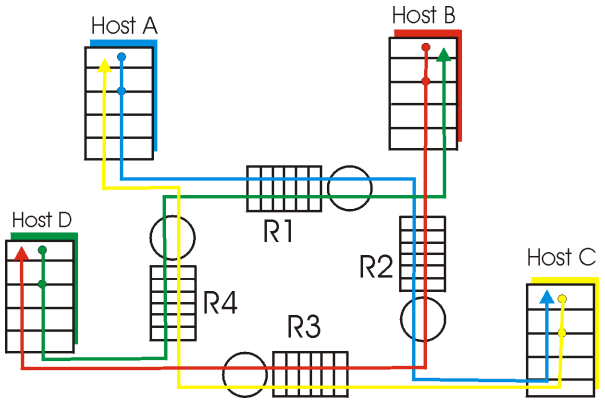
"costs" of congestion:

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

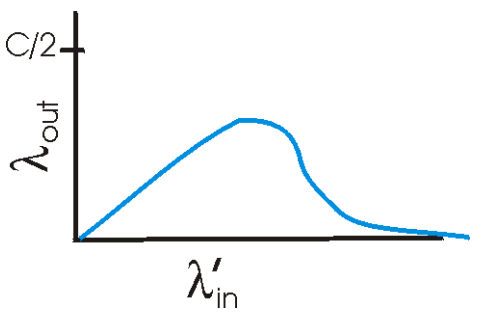
Causes/costs of congestion: scenario 3

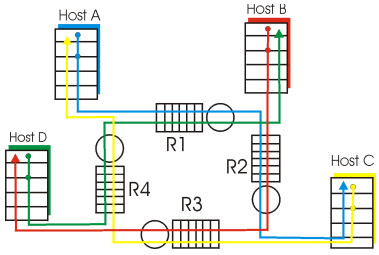
- four senders
- multihop paths
- timeout/retransmit

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

Case study: ATM ABR congestion control

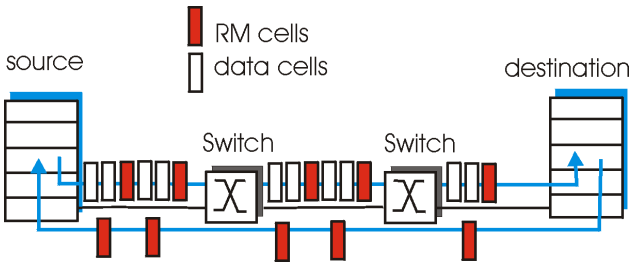
ABR: available bit rate:

- "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

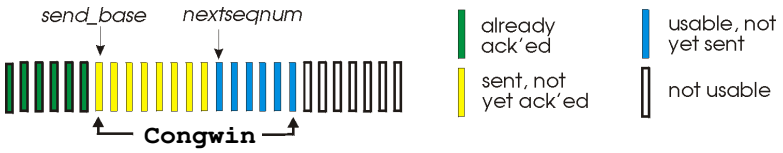
Case study: ATM ABR congestion control



- 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, CI bit set 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:

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

- Congestion Window never larger than rcvr-advertised window

TCP congestion control:

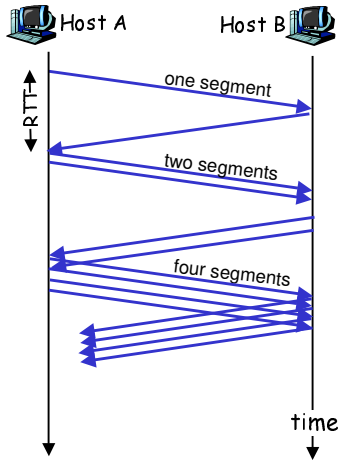
- "probing" for usable bandwidth:
  - ideally: transmit as fast as possible (**Congwin** as large as possible) without loss
  - *increase* **Congwin** until loss (congestion)
  - loss: *decrease* **Congwin**, then begin probing (increasing) again
- two "phases"
  - slow start
  - congestion avoidance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

TCP Slowstart

Slowstart algorithm

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

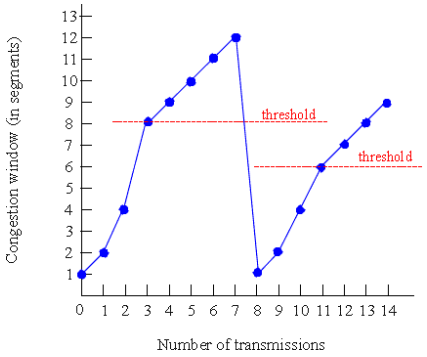
- exponential increase (per RTT) in window size (not so slow!)
- loss event: timeout (Tahoe TCP) and/or or three duplicate ACKs (Reno TCP)



TCP Congestion Avoidance (Tahoe)

Congestion avoidance

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



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

TCP Reno

- Most of today's TCPs are Reno
- Same behavior as Tahoe on timeout
- On Triple-duplicate ACK:
  - Tahoe does nothing (no window change)
  - Reno:
    - Threshold = congwin / 2
    - congwin = congwin / 2

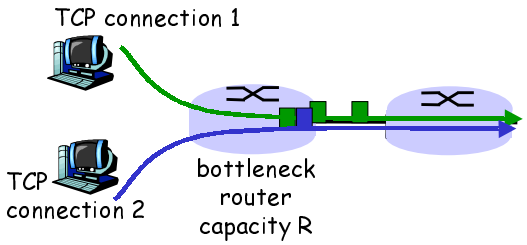
AIMD

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

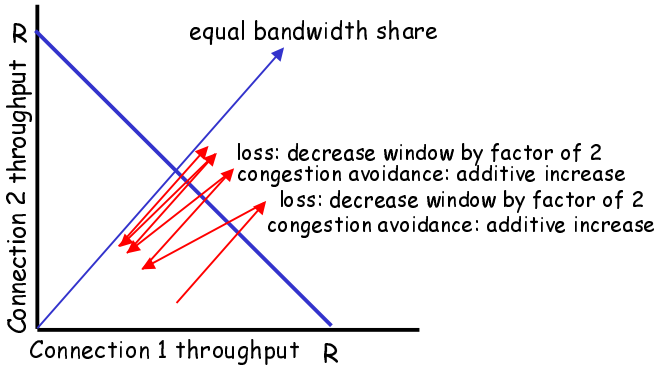
Fairness goal: if N TCP sessions share same bottleneck link, each should get 1/N of link capacity



Why is TCP fair?

Two competing sessions:

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



Effects of TCP latencies

Q: client latency from object request from WWW server to receipt?

- TCP connection establishment
- data transfer delay

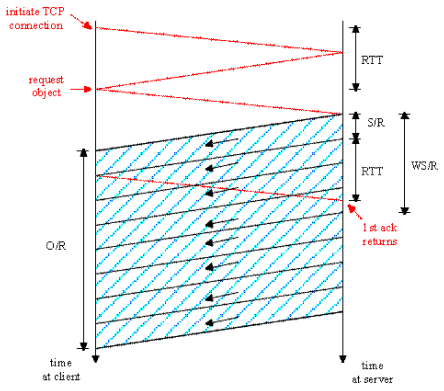
Notation, assumptions:

- Assume: fixed congestion window, W, giving throughput of R bps
- S: MSS (bits)
- O: object size (bits)
- no retransmissions (no loss, no corruption)

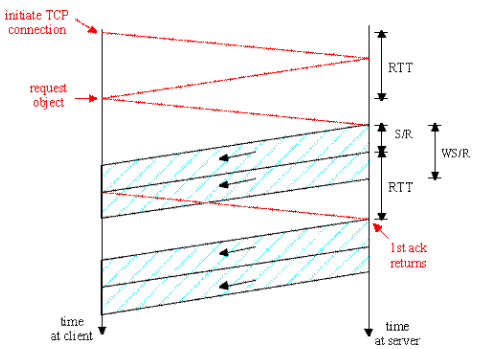
Two cases to consider:

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

Effects of TCP latencies



Case 1: latency = 2RTT + O/R



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

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"