CHALMERS UNIVERSITY OF TECHNOLOGY | UNIVERSITY OF GOTHENBURG

Course on Computer Communication and Networks

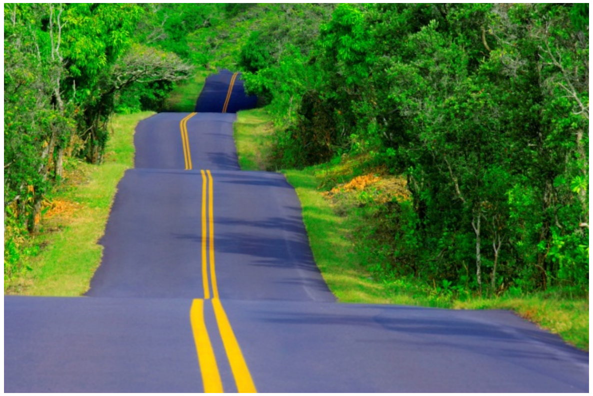
Lecture 5

Chapter 3; Transport Layer, Part B

EDA344/DIT 420, CTH/GU

Based on the book Computer Networking: A Top Down Approach, Jim Kurose, Keith Ross, Addison-Wesley.

Roadmap Transport Layer



- transport layer services
- multiplexing/demultiplexing
- connectionless transport: UDP
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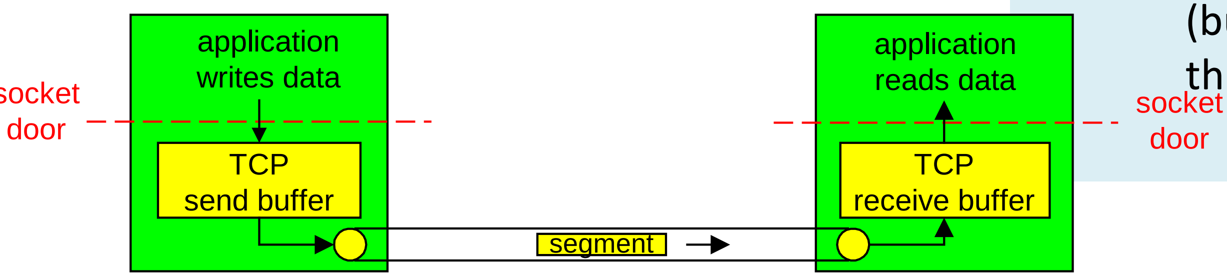
    - Retransmissions
    - Connection management
    - Flow control and buffer space
  - Congestion control
    - Principles
    - TCP congestion control

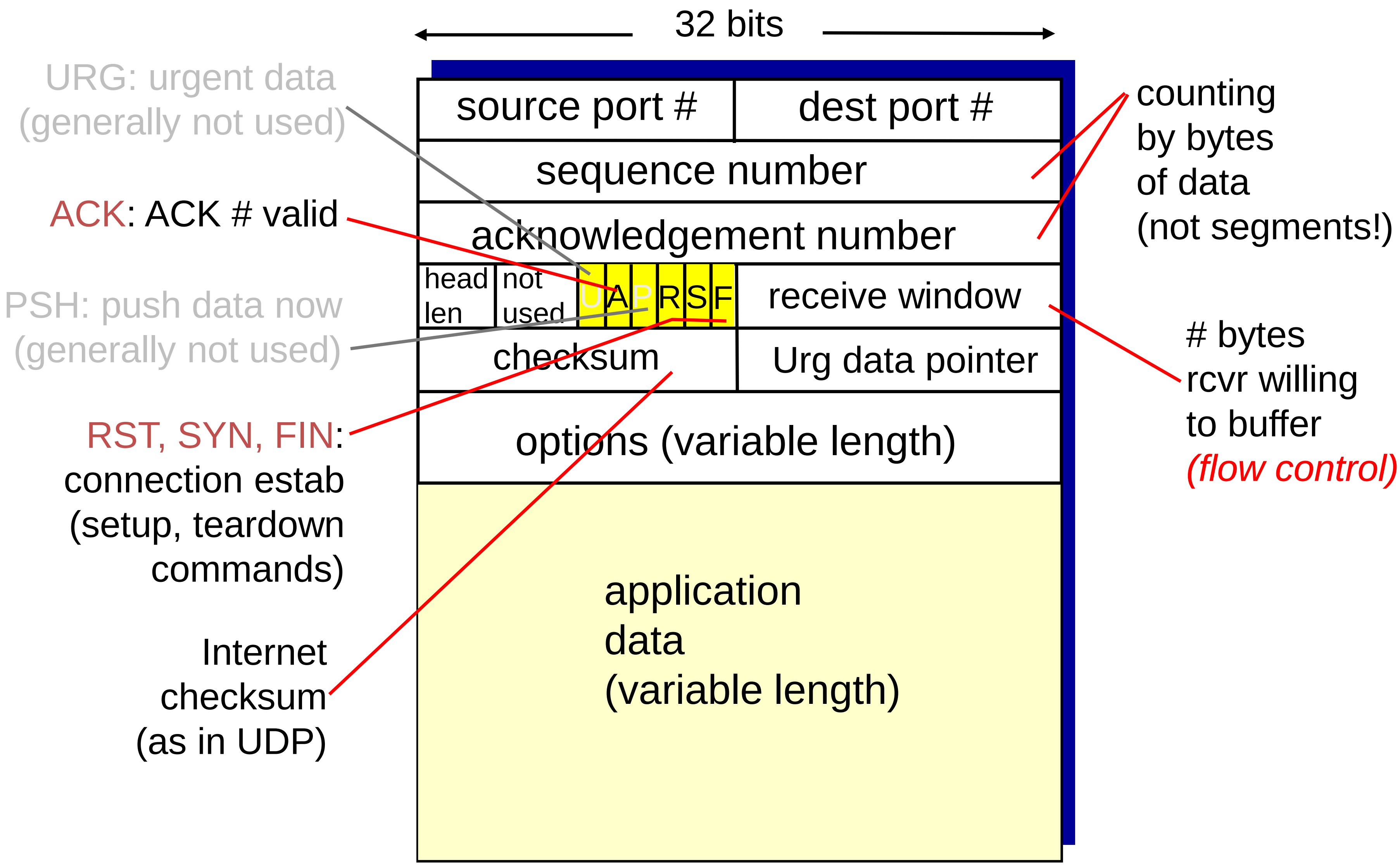
TCP: Overview RFCs: 793,1122,1323, 2018, 5681

- point-to-point:
  - one sender, one receiver
- reliable, in-order byte steam:
- 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 control:
  - sender will not overwhelm receiver
- congestion control:
  - sender will not flood network (but still try to maximize throughput)

TCP segment structure



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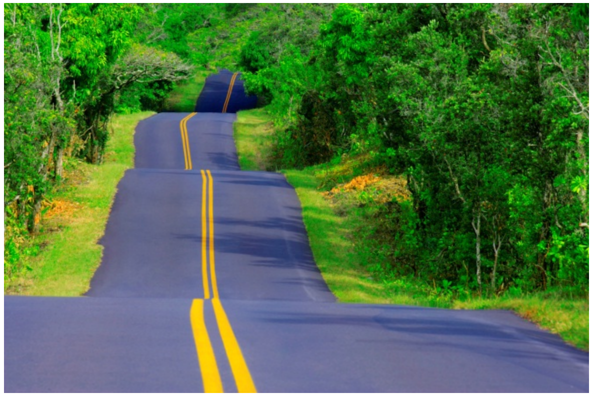


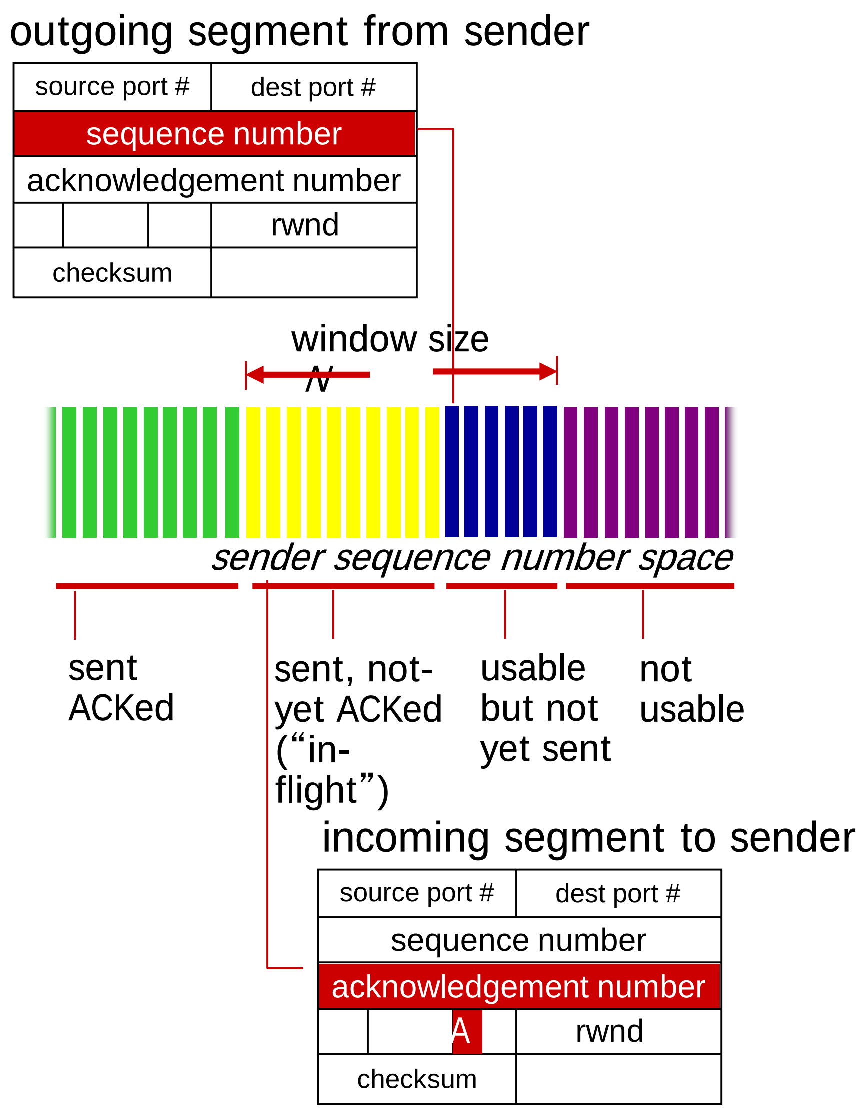
TCP seq. numbers, ACKs

sequence numbers:

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

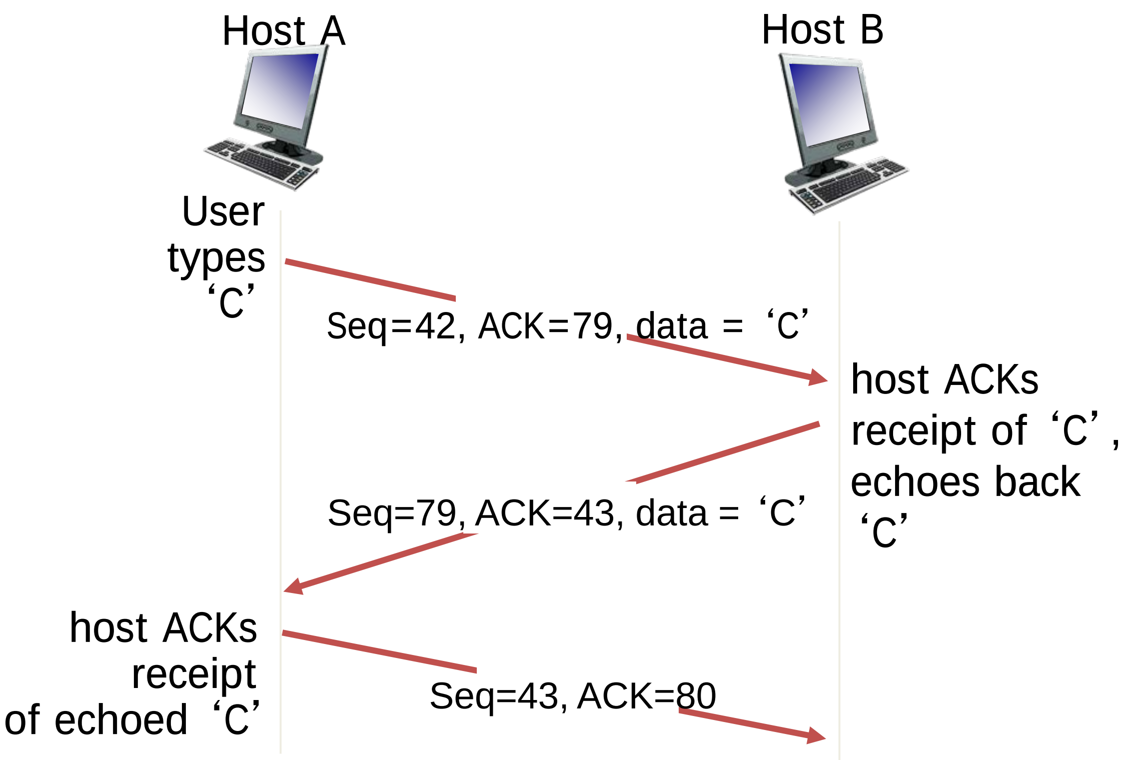
acknowledgements:

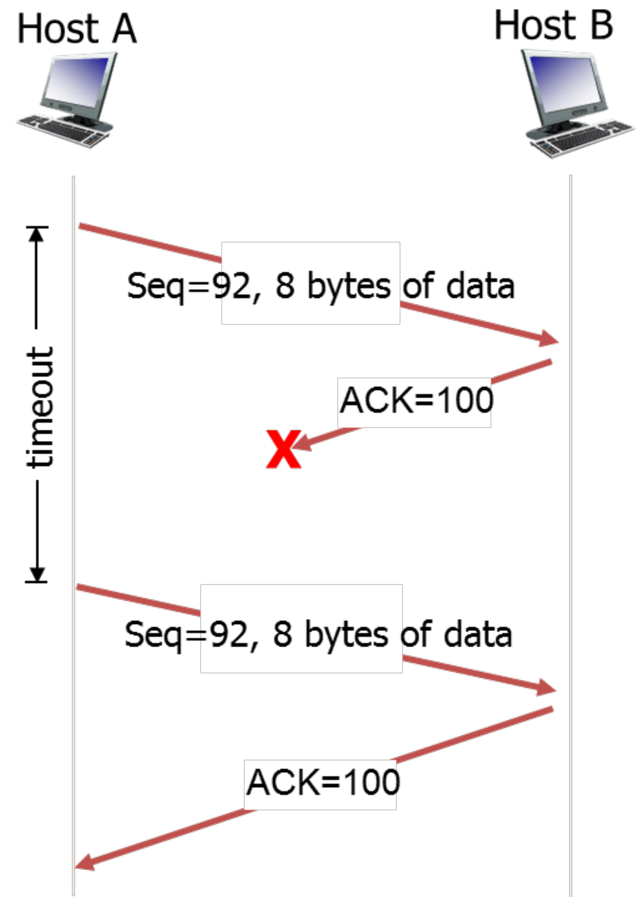
- seq # of next byte expected from other side
- **cumulative ACK**



TCP seq. numbers, ACKs

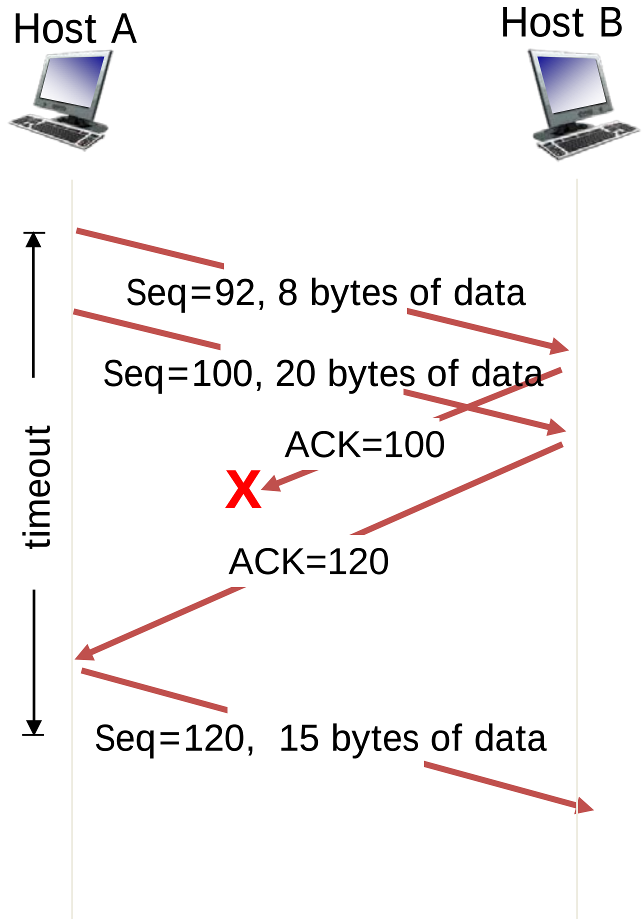
Always ack next in-order expected byte



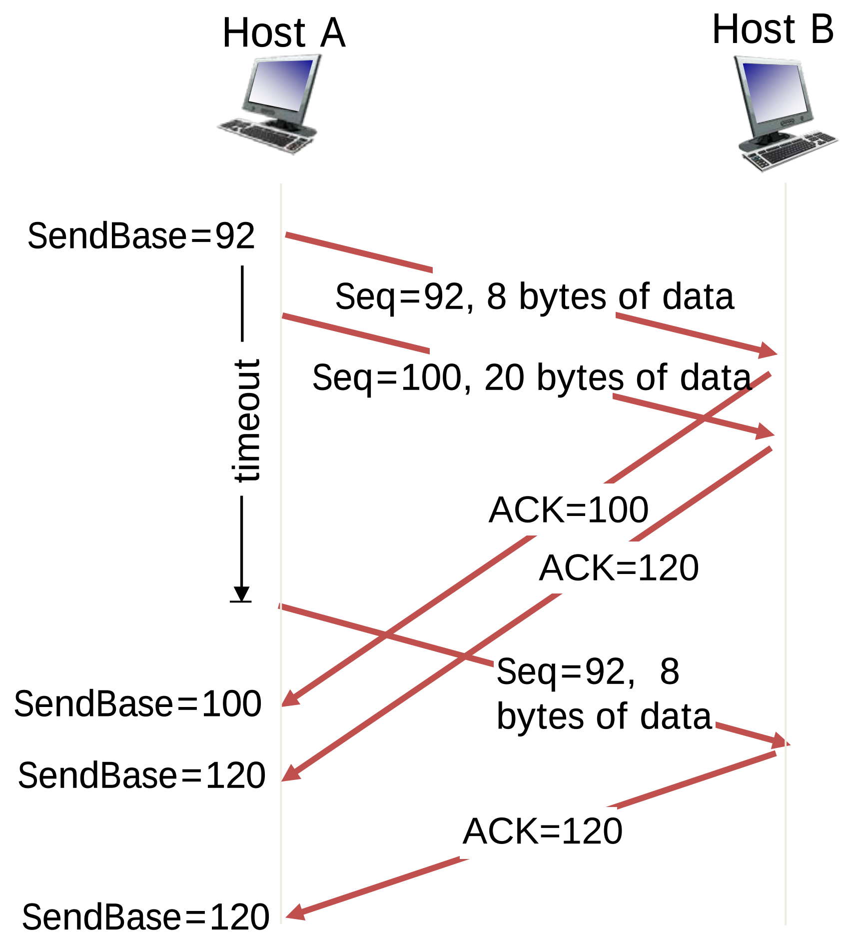


Simple example scenario
Based on telnet msg exchange

TCP:
cumulative Ack - retransmission scenarios



Cumulative ACK



(Premature) timeout

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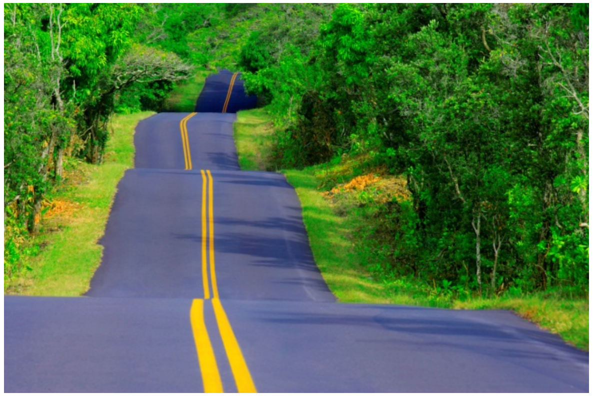
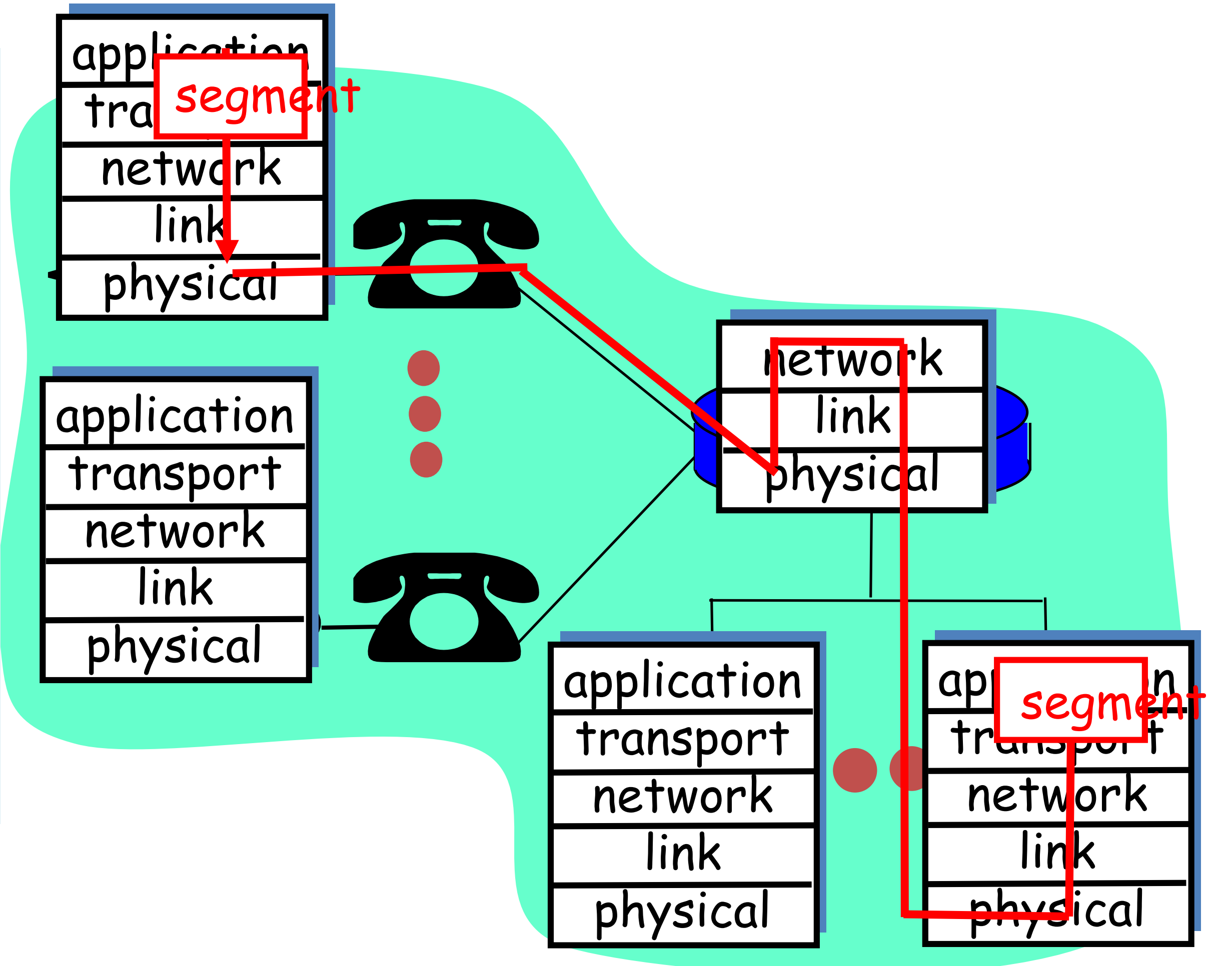


Q: how to set TCP timeout value?

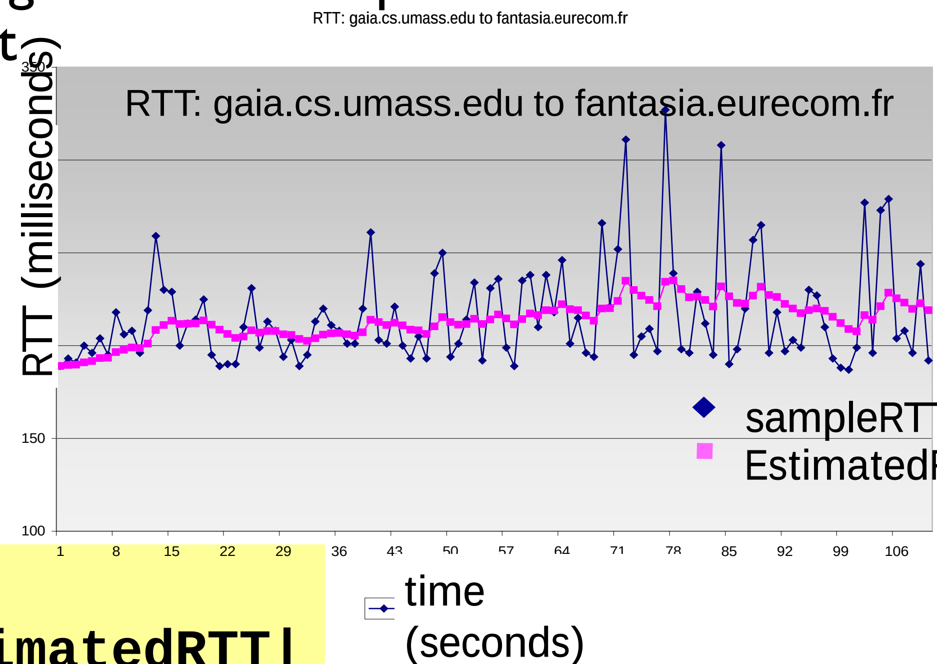
- longer than end-to-end RTT
  - but that varies!!!
- *too short* timeout*:* premature, unnecessary retransmissions
- *too long:* slow reaction to loss



TCP round trip time, timeout estimation

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

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



DevRTT = (1-β)*DevRTT +
β*|SampleRTT-EstimatedRTT|

(typically, β = 0.25)

TimeoutInterval = EstimatedRTT + 4*DevRTT

"safety margin"

TCP fast retransmit (RFC 5681)

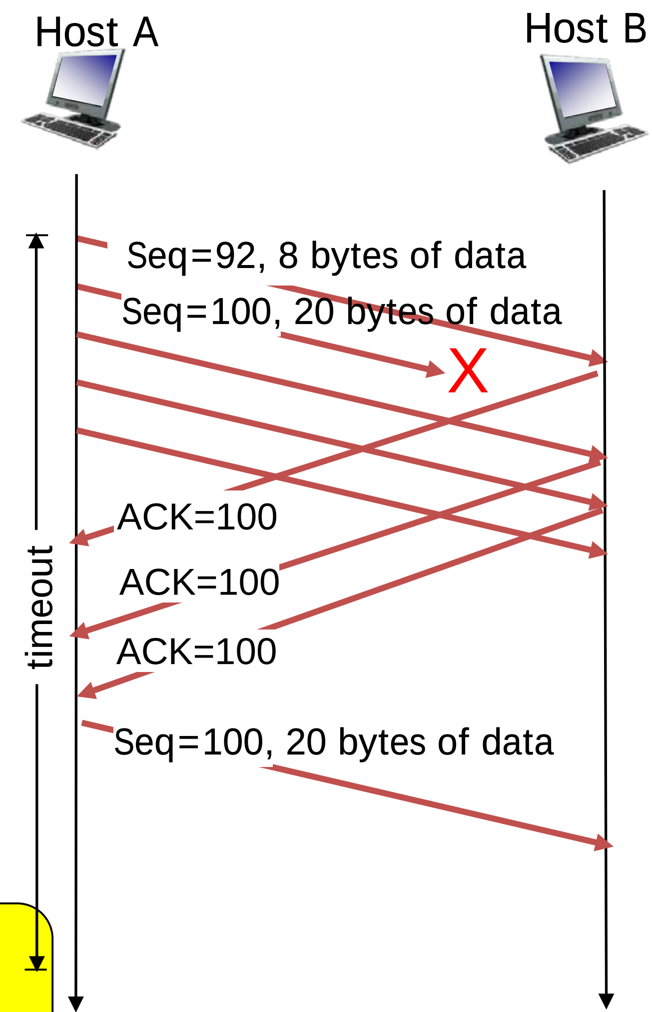
- time-out can be long:
  - long delay before resending lost packet
- IMPROVEMENT: detect lost segments via duplicate ACKs

TCP fast retransmit

if sender receives 3 duplicate ACKs for same data

- resend unacked segment with smallest seq #
  - likely that unacked segment lost, so don't wait for timeout

Implicit NAK!
Q: Why need at least 3?



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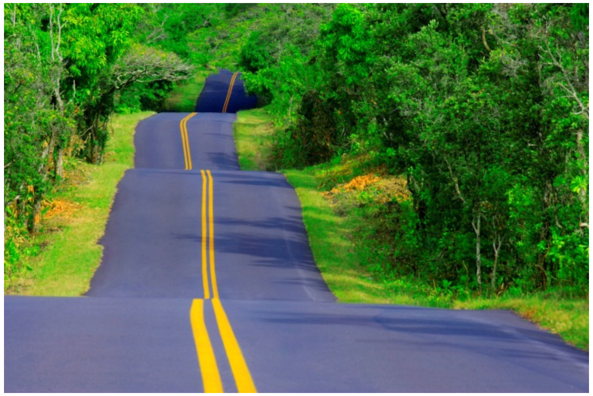
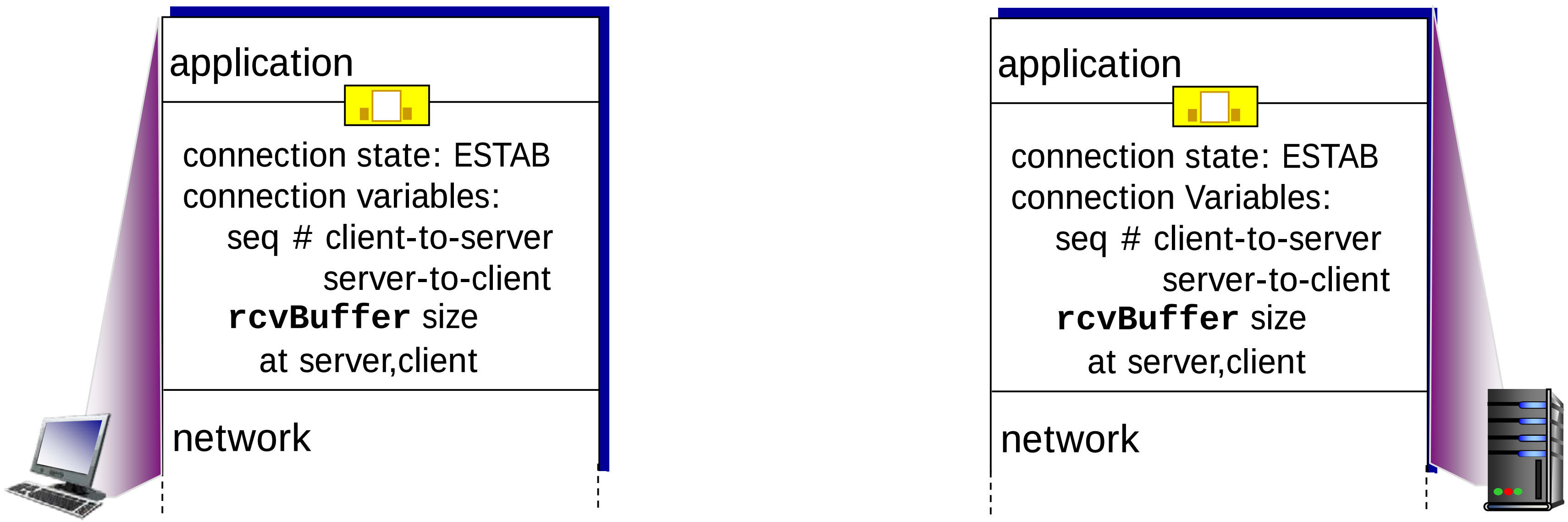


Connection Management

before exchanging data, sender/receiver “handshake”:

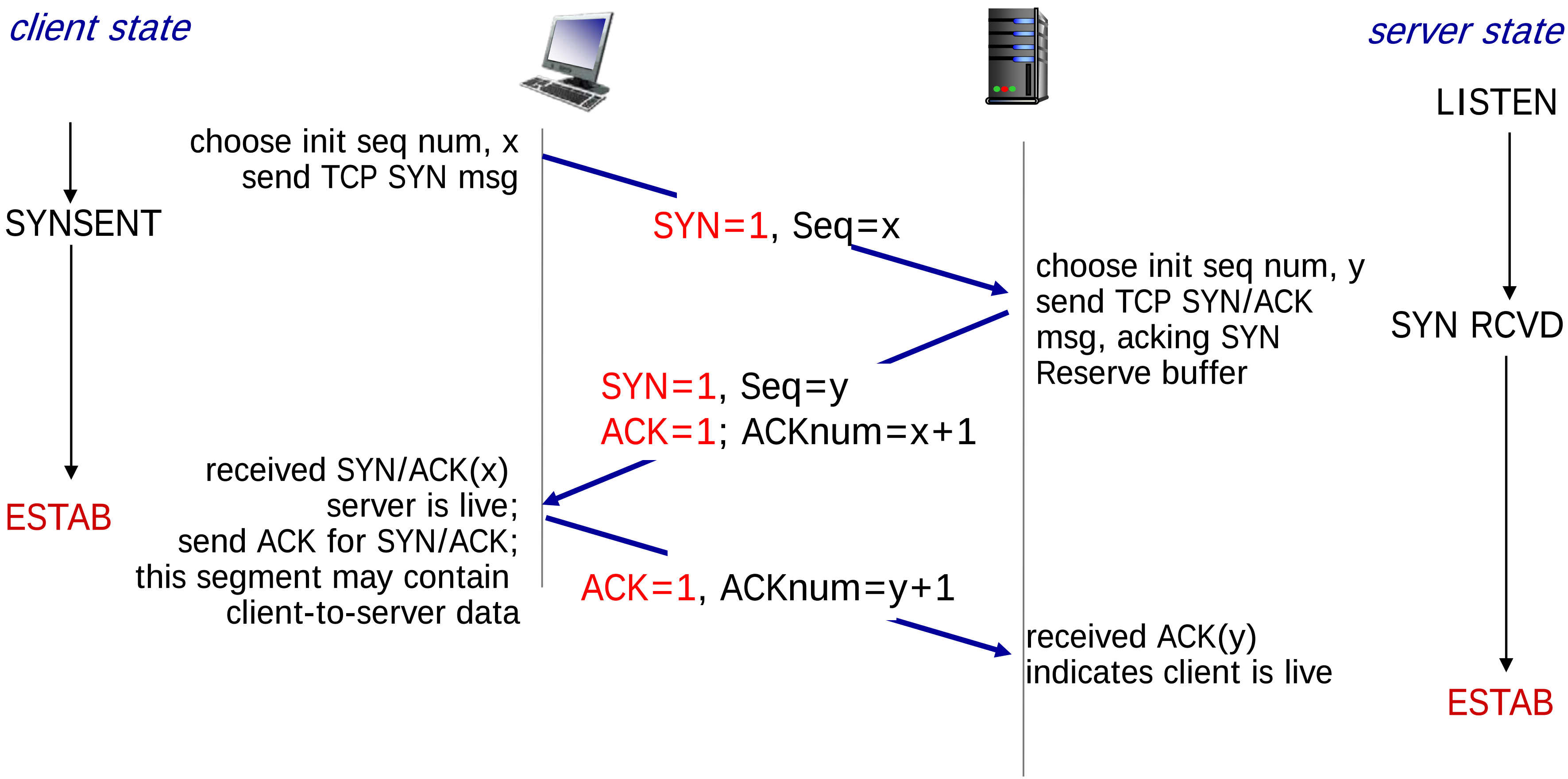
- agree to establish connection + connection parameters



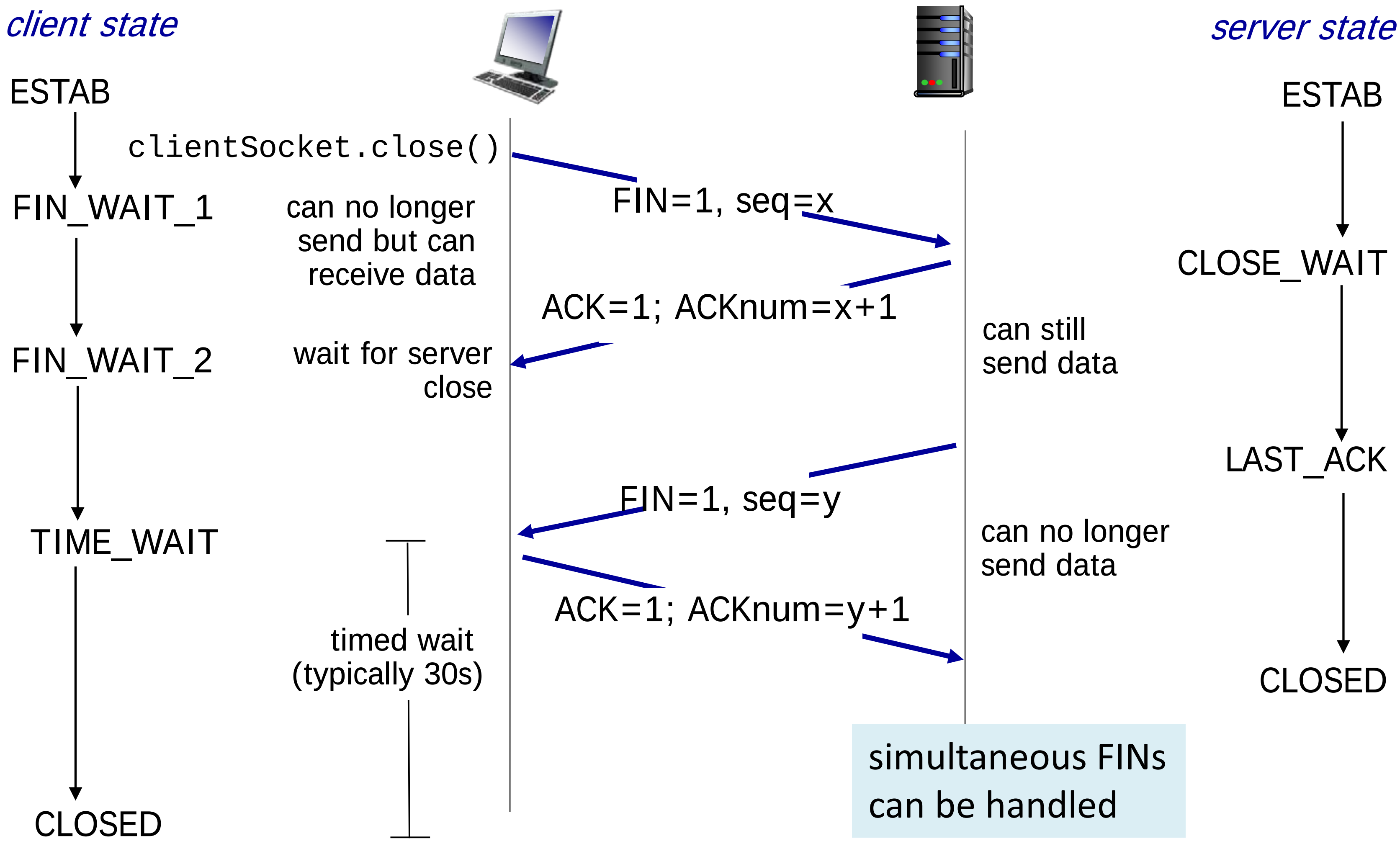

```
Socket clientSocket =
  newSocket("hostname","port
   number");
```

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

Setting up a connection: TCP 3-way handshake



TCP: closing a connection



simultaneous FINs can be handled

RST: alternative way to close connection immediately, when **error** occurs

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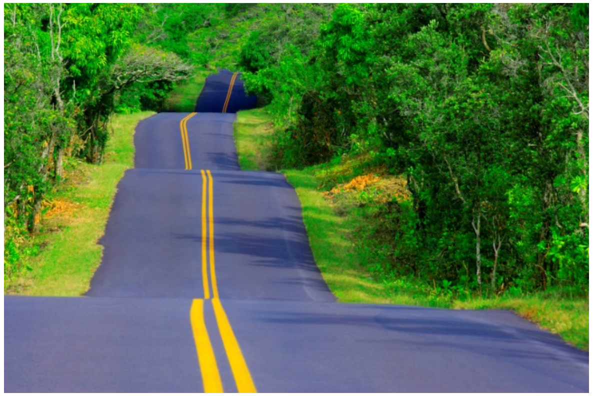
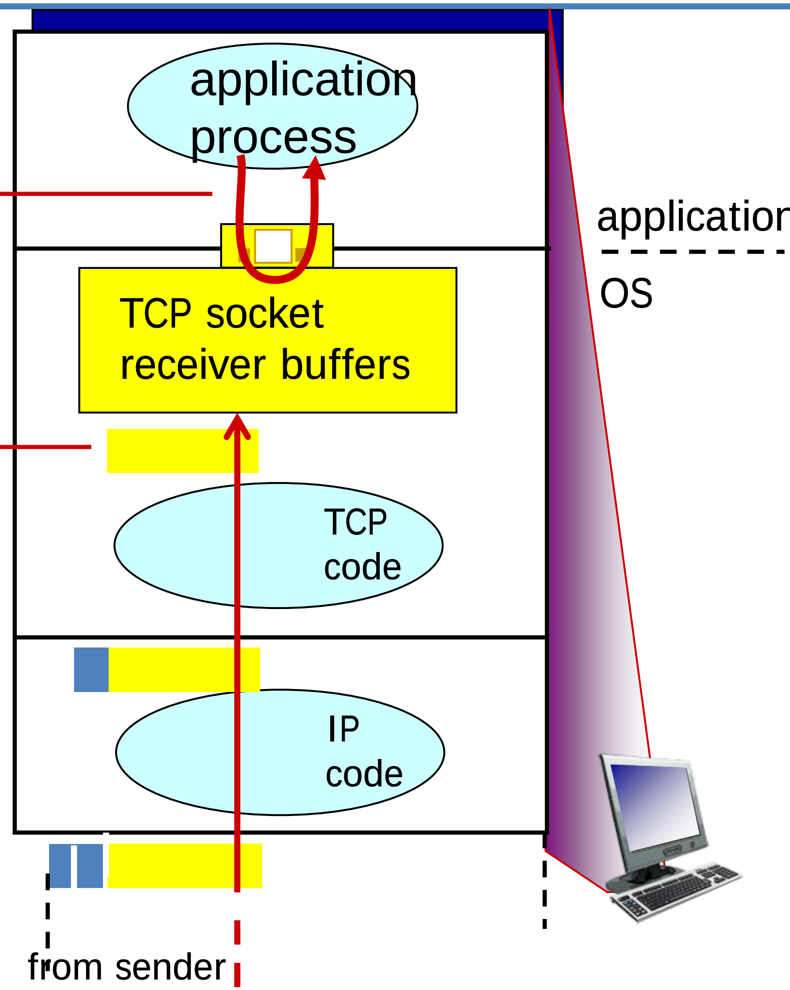


TCP flow control

application
might remove data from
TCP socket buffers ….

… slower than TCP
is delivering
(i.e. slower than
sender is sending)



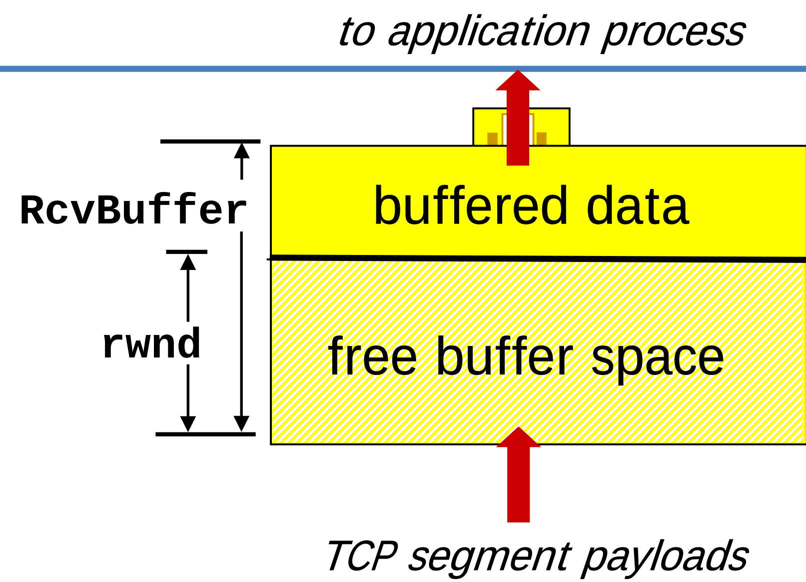
receiver protocol stack

flow control

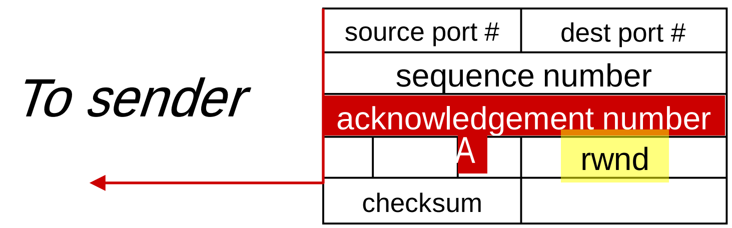
receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast

TCP flow control

- receiver "advertises" free buffer space through **`rwnd`** value in header
  - **`RcvBuffer`** size set via socket options (typical default 4 Kbytes)
  - OS can autoadjust **`RcvBuffer`**
- sender limits unacked ("in-flight") data to receiver's **`rwnd`** value
  - s.t. receiver's buffer will not overflow



receiver-side buffering



Q: Is TCP stateful or stateless?

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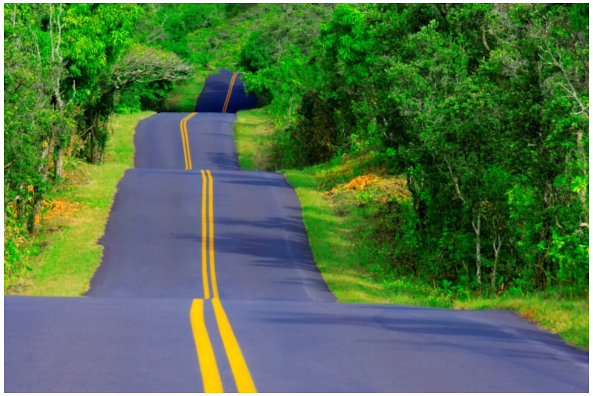


Principles of congestion control

congestion:

- informally: "many sources sending too much data too fast for *network* to handle"
- Manifestations?
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)

Distinction between flow control and congestion control

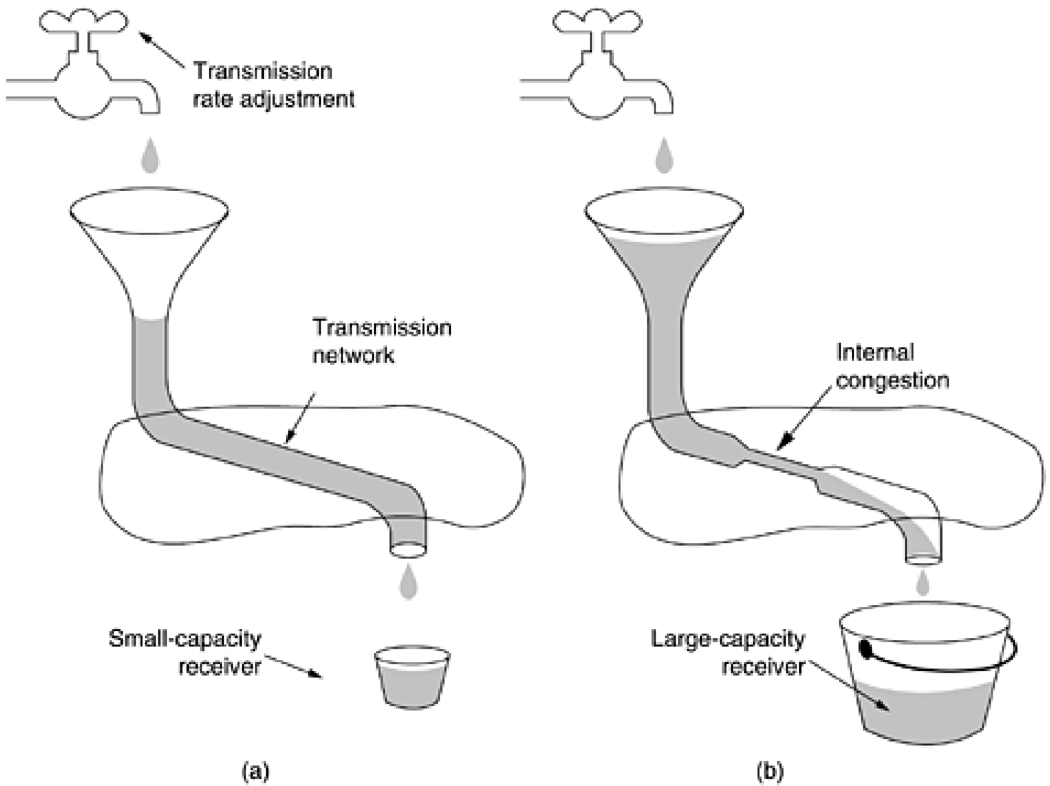
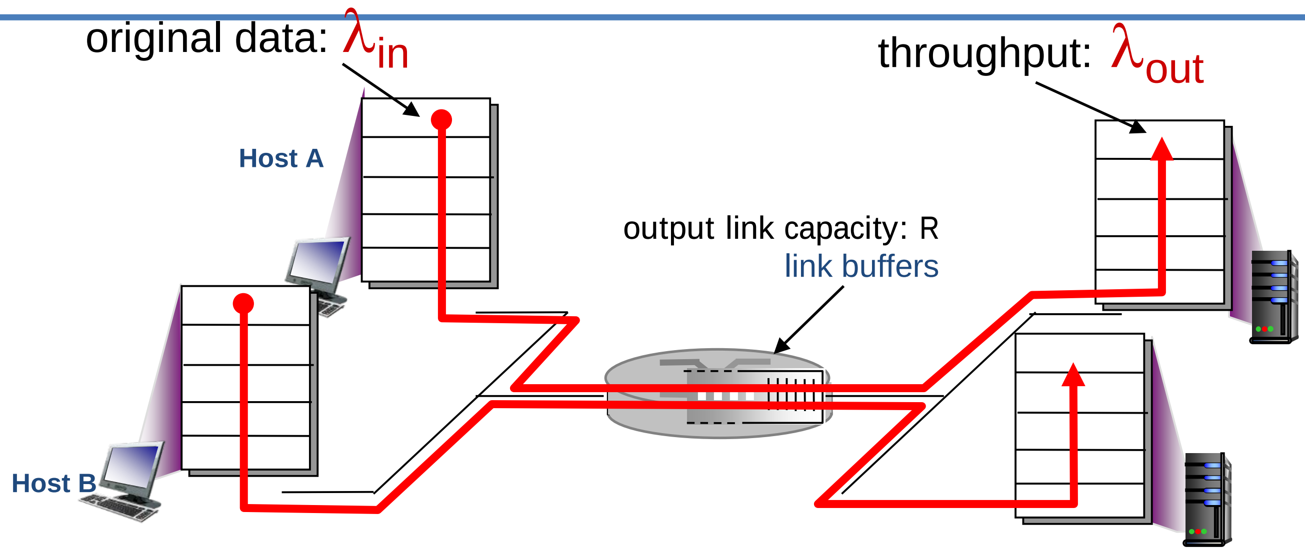


Fig. A. Tanenbaum
Computer Networks

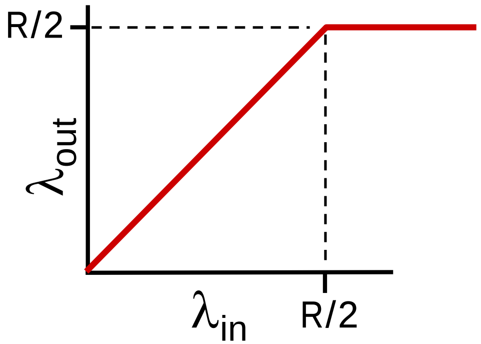
Need for flow control

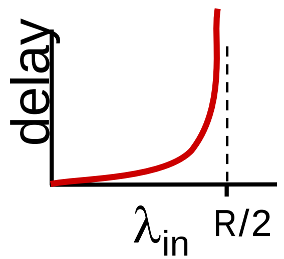
Need for congestion control

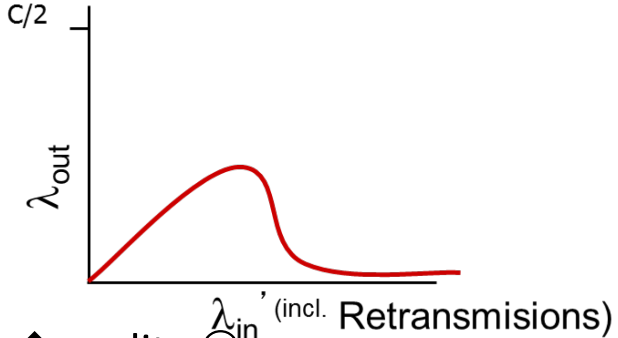
Causes/costs of congestion



- Recall queueing behaviour + losses
- Losses => retransmissions => even higher load…







- Ideal per-connection throughput: R/2 (if 2 connections)
- reality ☹

Approaches towards congestion control

approach taken by TCP

end-end congestion control:

- no explicit feedback from network
- congestion inferred from end-system observed loss, delay

Not present in Internet's network layer protocols

network-assisted congestion control:

- routers collaborate for optimal rates + provide feedback to end-systems eg.
  - a single bit indicating congestion
  - explicit rate for sender to send at

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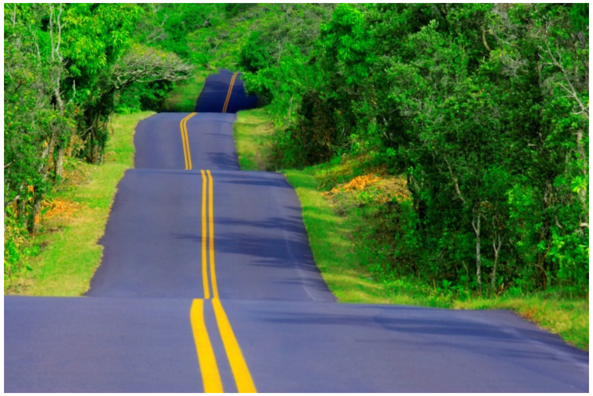


TCP congestion control:
additive increase multiplicative decrease

- end-end control (no network assistance), sender limits transmission

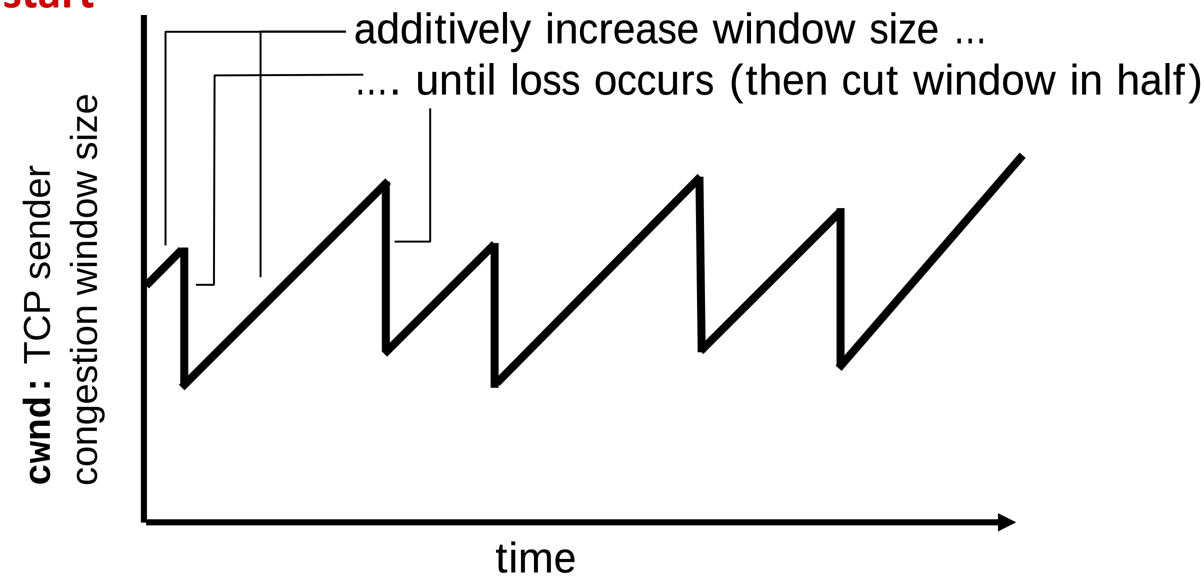
How does sender perceive congestion?

- loss = timeout *or* 3 duplicate acks
- TCP sender reduces rate (**Congestion Window**) then

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

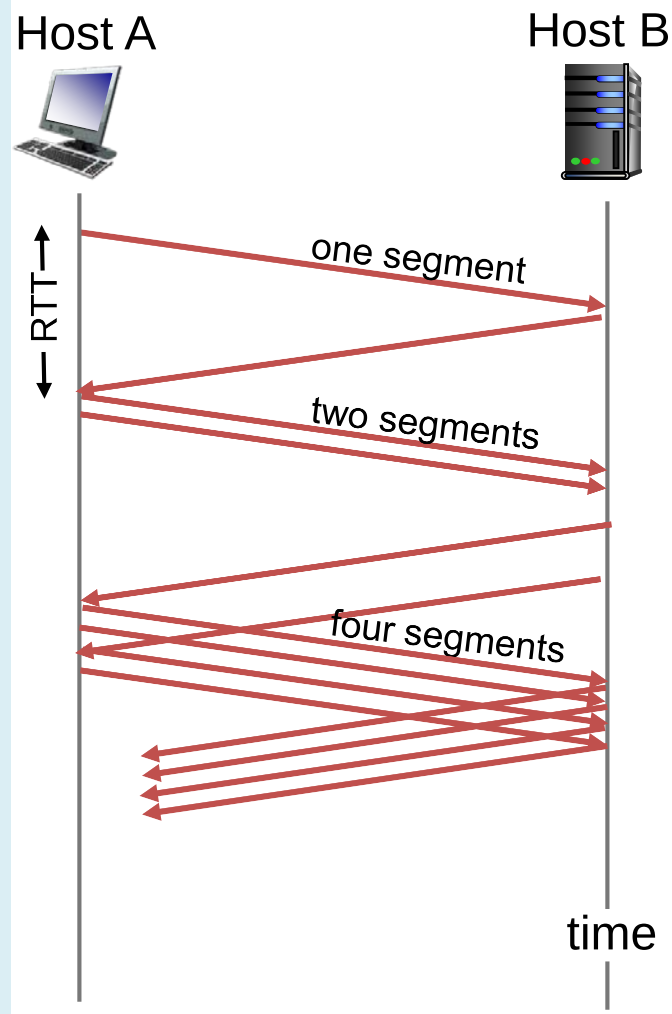
- *Additive Increase:* increase **cwnd** by 1 MSS every RTT until loss detected
- *Multiplicative Decrease*: cut **cwnd** in half after loss
- **To start with: slow start**

AIMD saw tooth behavior: probing for bandwidth

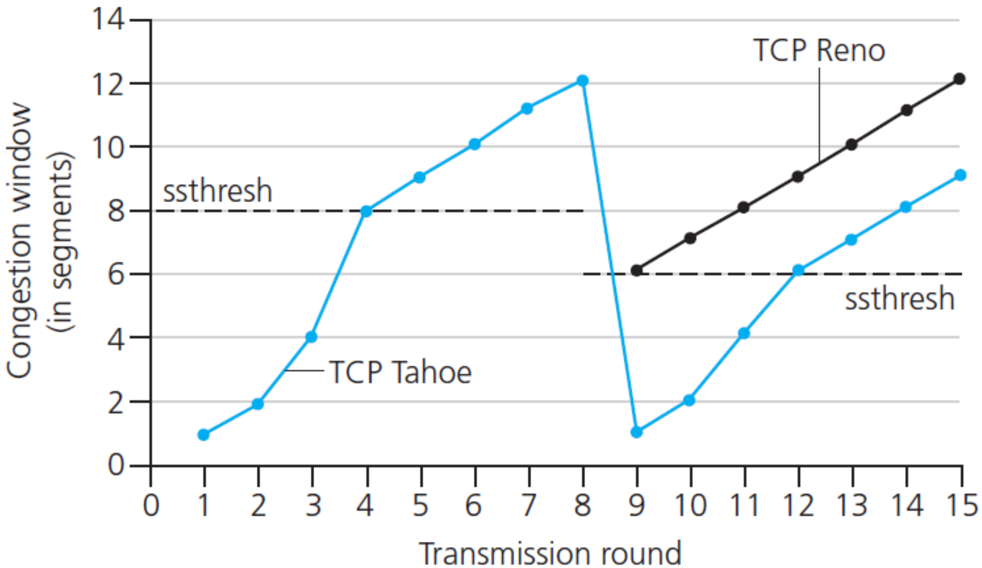


TCP Slow Start

- when connection begins, increase rate exponentially until first loss event:
  - initially **cwnd** = 1 MSS
  - double **cwnd** every ack of previous "batch"
  - done by incrementing **cwnd** for every ACK received
- *summary:* initial rate is slow but ramps up exponentially fast
  - then, saw-tooth



TCP cwnd: from exponential to linear growth + reacting to loss



Reno: loss indicated by timeout or 3 duplicate ACKs: cwnd is cut in half; then grows linearly

Non-optimized: loss indicated by timeout: cwnd set to 1 MSS; then window slow start to threshold, then grows linearly

Implementation:

- variable **ssthresh** (slow start threshold)
- on loss event, **ssthresh** = ½***cwnd**

Q: How many windows does a TCP's sender maintain?

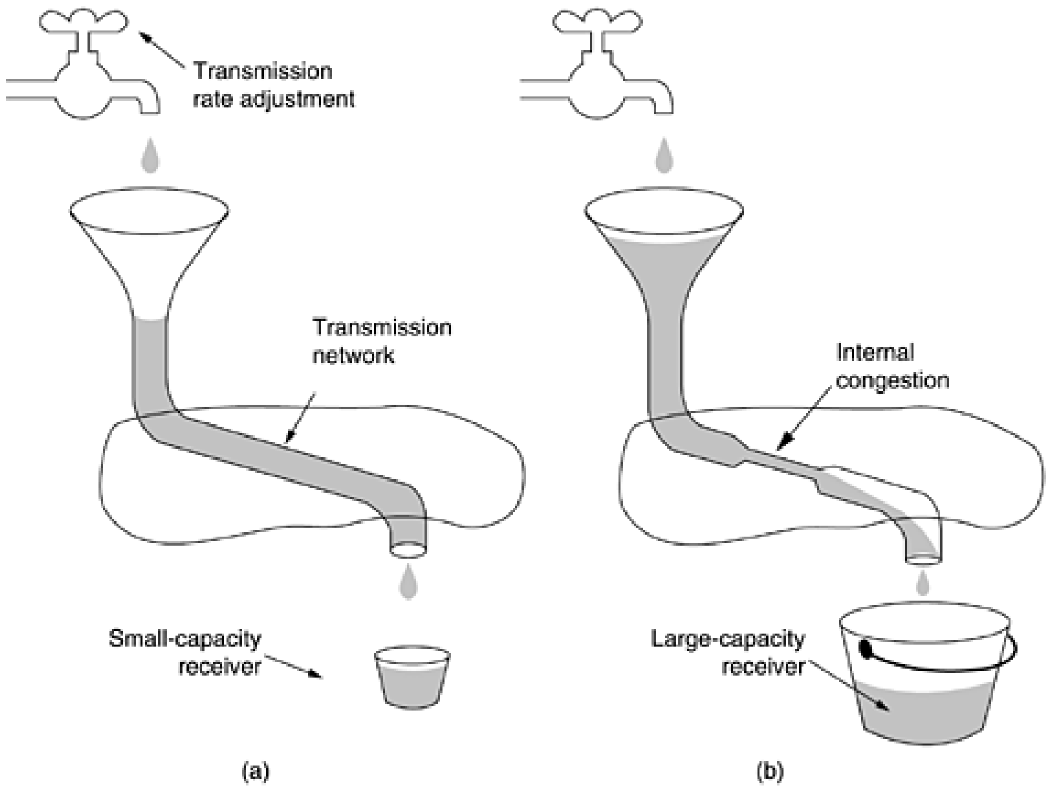


Fig. A. Tanenbaum
Computer Networks

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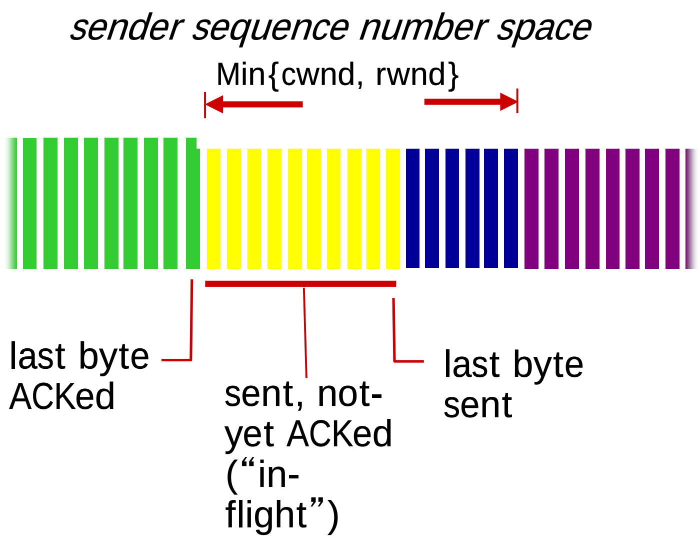
TCP combined flow-ctrl, congestion ctrl windows



TCP sending rate:

- send min {cwnd, rwnd} bytes, wait for ACKS, then send more

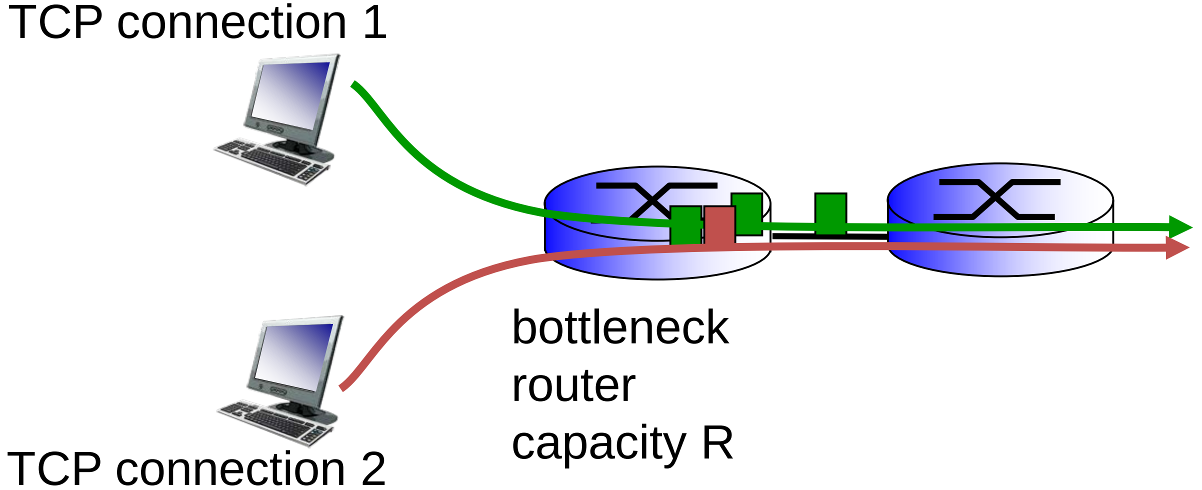
sender limits transmission:

$$\text{LastByteSent-LastByteAcked} \leq \text{Min\{cwnd, rwnd\}}$$

- **cwnd** is dynamic, function of perceived network congestion,
- **rwnd** dymanically limited by receiver's buffer space

TCP Fairness

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



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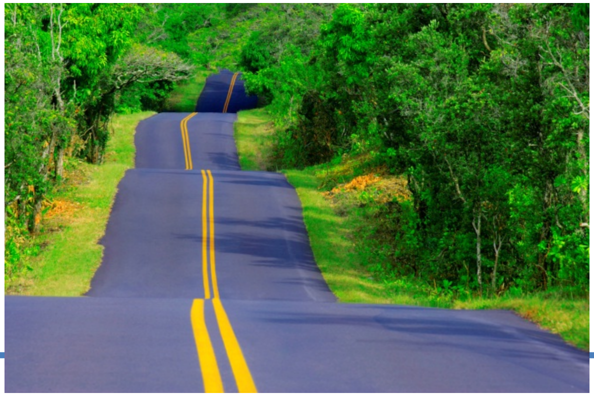


Chapter 3: summary

- principles behind transport layer services:
  - Addressing
  - reliable data transfer
  - flow control
  - congestion control
- instantiation, implementation in the Internet
  - UDP
  - TCP

next:

- leaving the network "edge" (application, transport layers)
- into the network "core"

Some review questions on this part

-
- t upon a 3rd ack and not a 2nd?
- ol: principle, method for detection
- : increase indefinitely?
- management?
- in the start and the end of
- data transfer if it uses UDP? How or

Reading instructions chapter 3

- **KuroseRoss book**

Careful	Quick
3.1, 3.2, 3.4-3.7	3.3

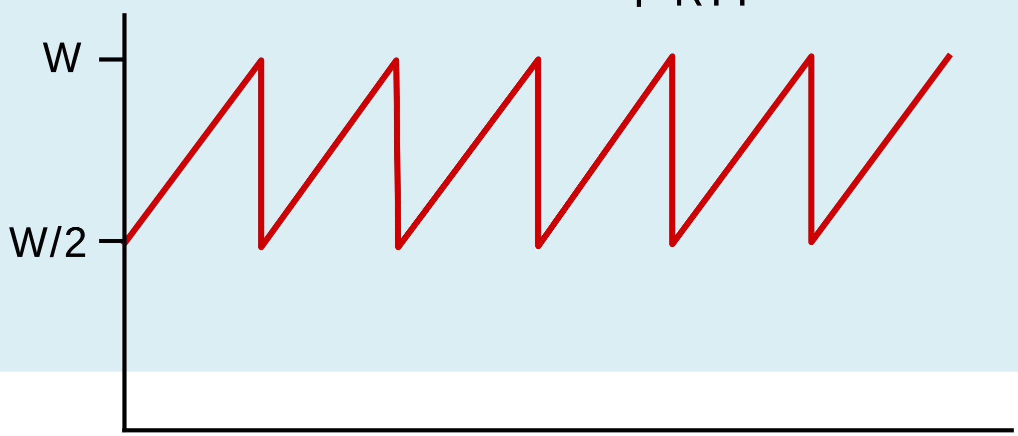
- **Other resources (further study)**
  - Eddie Kohler, Mark Handley, and Sally Floyd. 2006. Designing DCCP: congestion control without reliability. *SIGCOMM Comput. Commun. Rev.* 36, 4 (August 2006), 27-38. DOI=10.1145/1151659.1159918 http://doi.acm.org/10.1145/1151659.1159918
  - http://research.microsoft.com/apps/video/default.aspx?id=104005
  - Exercise/throughput analysis TCP in following slides

Extra slides, for further study

TCP throughput

- avg. TCP throughput as function of window size, RTT?
  - ignore 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. trhoughput is 3/4W per RTT

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



TCP Futures: TCP over "long, fat pipes"

- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- requires W = 83,333 in-flight segments
- throughput in terms of segment loss probability, L [Mathis 1997]:

$$\text{TCP throughput} = \frac{1.22 \cdot \text{MSS}}{\text{RTT}\sqrt{L}}$$

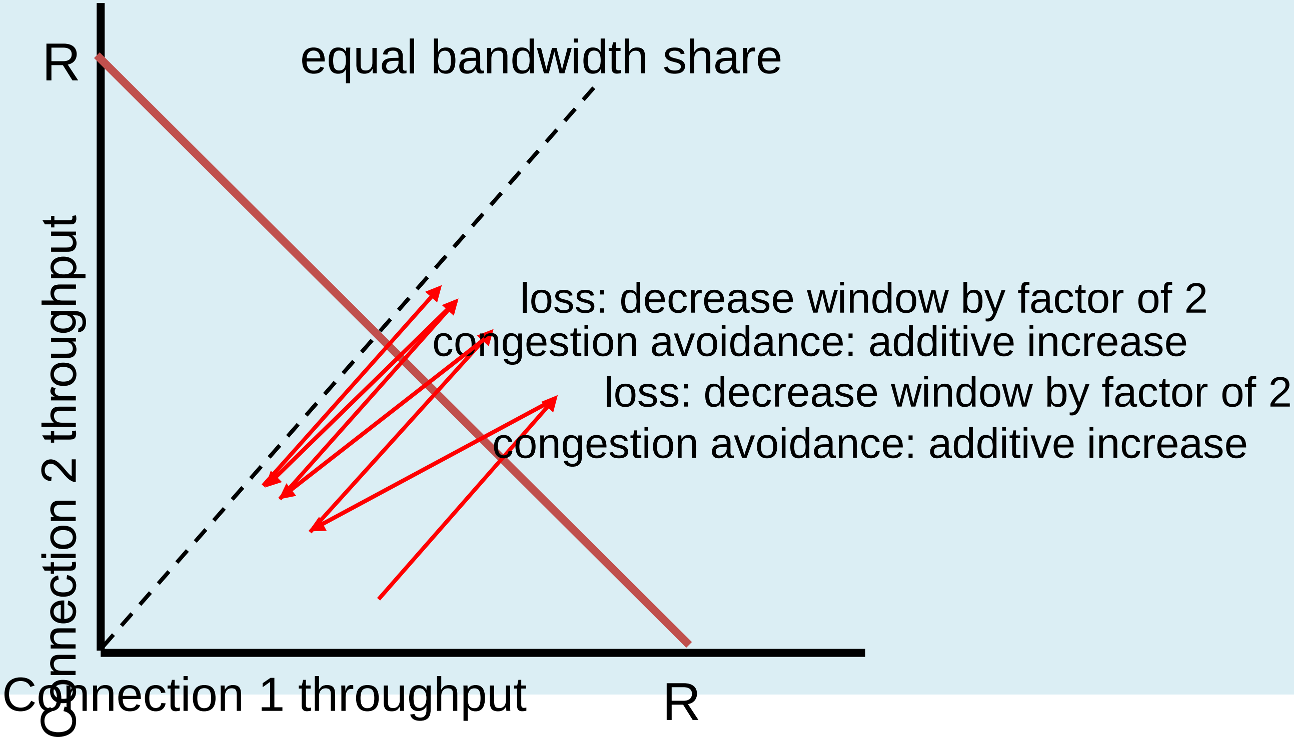
➜ to achieve 10 Gbps throughput, need a loss rate of L = $2 \cdot 10^{-10}$ *– a very small loss rate!*

- new versions of TCP for high-speed

Why is TCP fair?

two competing sessions:

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



Fairness (more)

Fairness and UDP

- multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- instead use UDP:
  - send audio/video at constant rate, tolerate packet loss

Fairness, parallel TCP connections

- application can open multiple parallel connections between two hosts
- web browsers do this
- e.g., link of rate R with 9 existing connections:
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets R/2

TCP delay modeling (slow start – related)

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

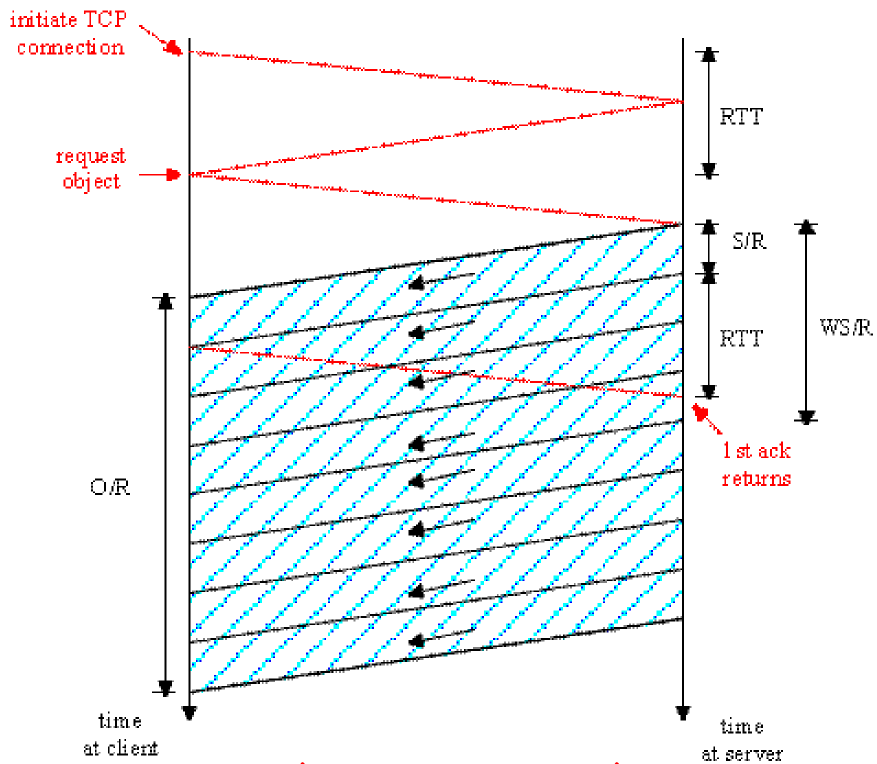
- TCP connection establishment
- data transfer delay

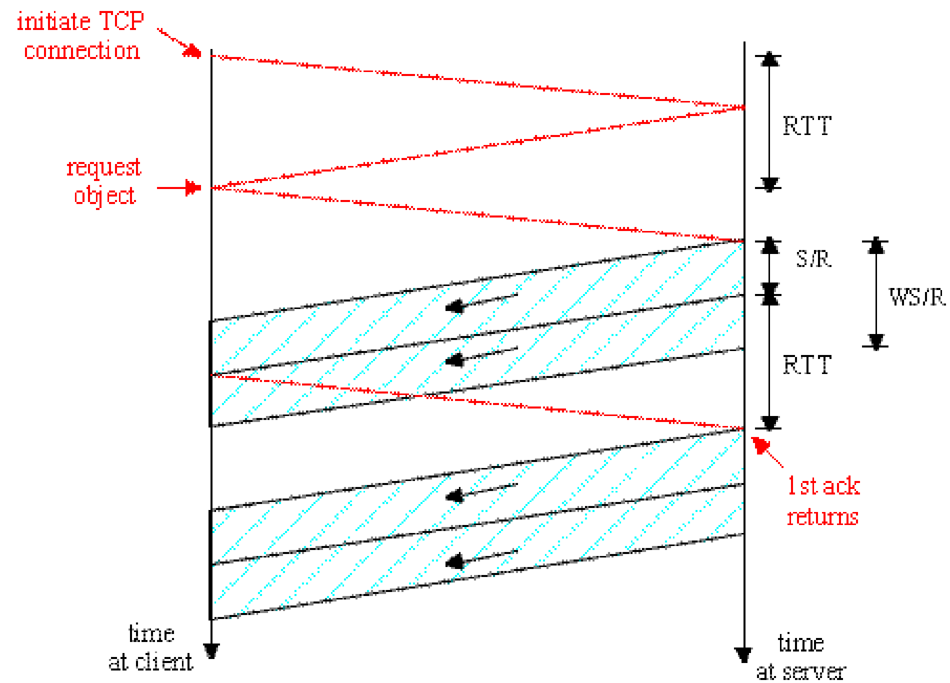
Notation, assumptions:

- 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)
- Receiver has unbounded buffer

TCP delay Modeling: simplified, fixed window

K:= O/WS



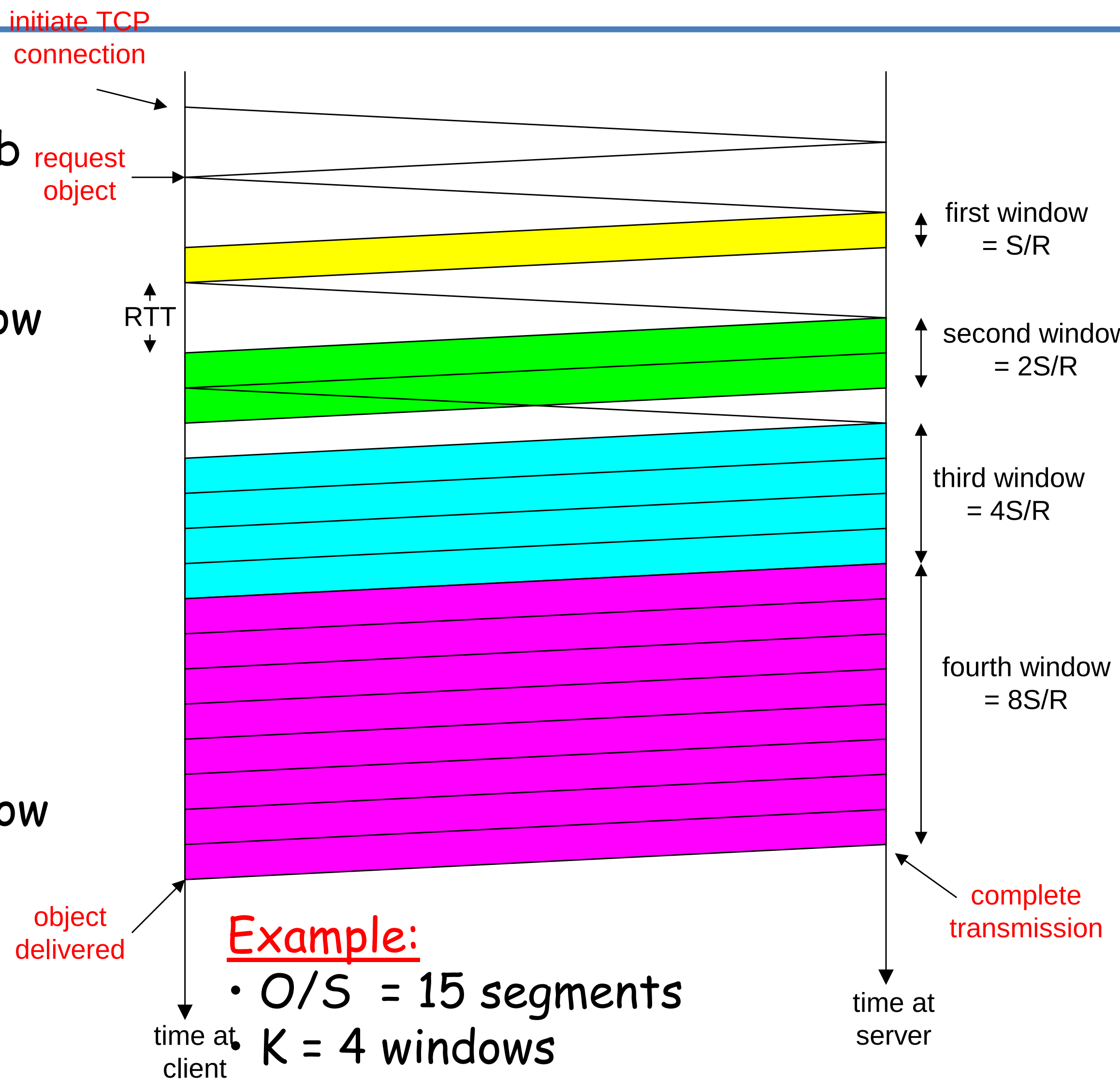


Case 1: WS/R > RTT + S/R:
ACK for first segment in window returns before window's worth of data nsent
delay = 2RTT + O/R

Case 2: WS/R < RTT + S/R:
wait for ACK after sending window's worth of data sent
delay = 2RTT + O/R
+ (K-1)[S/R + RTT - WS/R]

$$delay = \frac{O}{R} + 2RTT + \sum^{P} idleTime_p$$

TCP Delay Modeling: Slow Start



Delay components:

- 2 RTT for connection estab and request
- O/R to transmit object
- time server idles due to slow start

Server idles:
P = min{K-1,Q} times

where
- Q = #times server stalls until cong. window is larger than a "full-utilization" window (if the object were of unbounded size).

- K = #(incremental-sized) congestion-windows that "cover" the object.

Example:

- O/S = 15 segments
- K = 4 windows
- Q = 2
- Server idles P = min{K-1,Q} = 2 times

TCP Delay Modeling (slow start - cont)

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

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

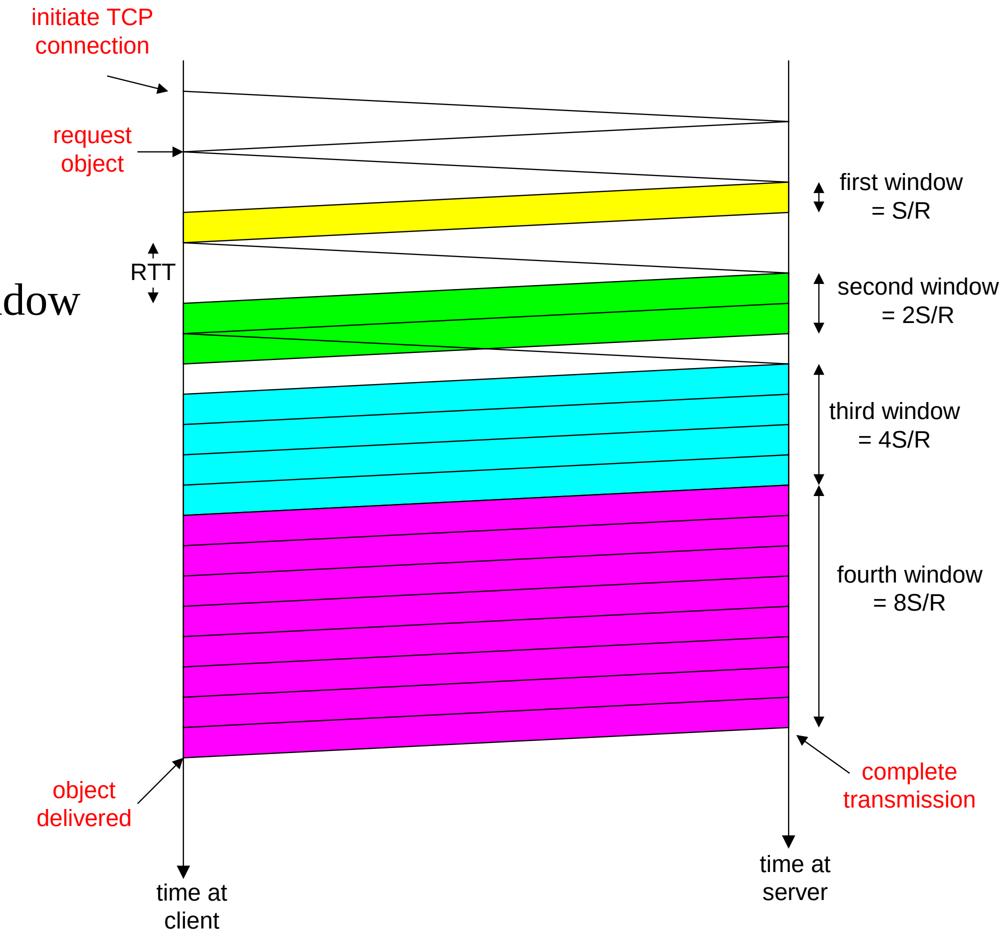
$$2^{k-1}\frac{S}{R} = \text{time to transmit the kth window}$$

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

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

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

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



TCP Delay Modeling

Recall K = number of windows that cover object

How do we calculate K ?

$$
\begin{aligned}
K &= \min\{k : 2^0 S + 2^1 S + \cdots + 2^{k-1} S \geq O\} \\
&= \min\{k : 2^0 + 2^1 + \cdots + 2^{k-1} \geq O/S\} \\
&= \min\{k : 2^k - 1 \geq \frac{O}{S}\} \\
&= \min\{k : k \geq \log_2(\frac{O}{S} + 1)\} \\
&= \left\lceil \log_2(\frac{O}{S} + 1) \right\rceil
\end{aligned}
$$

Calculation of Q, number of idles for infinite-size object, is similar.