SC250
Computer Networking I

TCP: The Transmission Control Protocol

Prof. Matthias Grossglauser

School of Computer and Communication Sciences
EPFL

http://lcawww.epfl.ch

ÉCOLE POLYTECHNIQUE FÉDÉRALE DE LAUSANNE

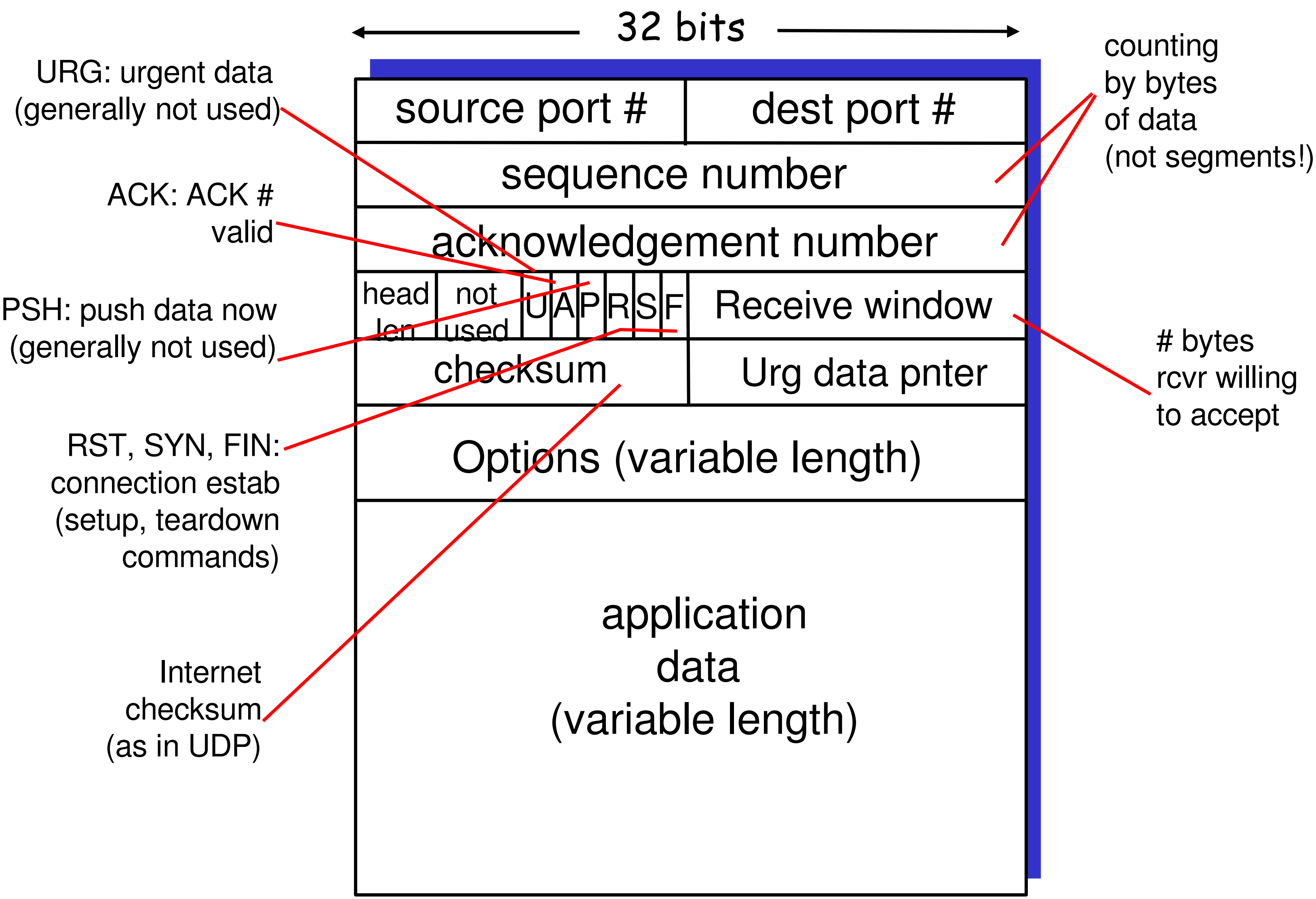
Objectives

- Connection-oriented, reliable transport: Transmission Control Protocol (TCP)
  - Segment structure
  - Reliable data transfer, based on principles seen last time
  - Connection management
- Congestion control
  - Principles
  - TCP

TCP: Overview

- RFCs:
  - 793, 1122, 1323, 2018, 2581
- Point-to-point:
  - One sender, one receiver
- Reliable, in-order byte steam:
  - No “message boundaries”
- Full duplex:
  - Bi-directional data flow in same connection
- Connection-oriented:
  - Handshaking (exchange of control msgs) init’s sender, receiver state before data exchange
- Flow controlled:
  - Sender will not overwhelm receiver
- Pipelined:
  - TCP congestion and flow control set window size
- Send & receive buffers

TCP Segment Structure



TCP Sequence and Acknowledgment Numbers

Seq. #'s:

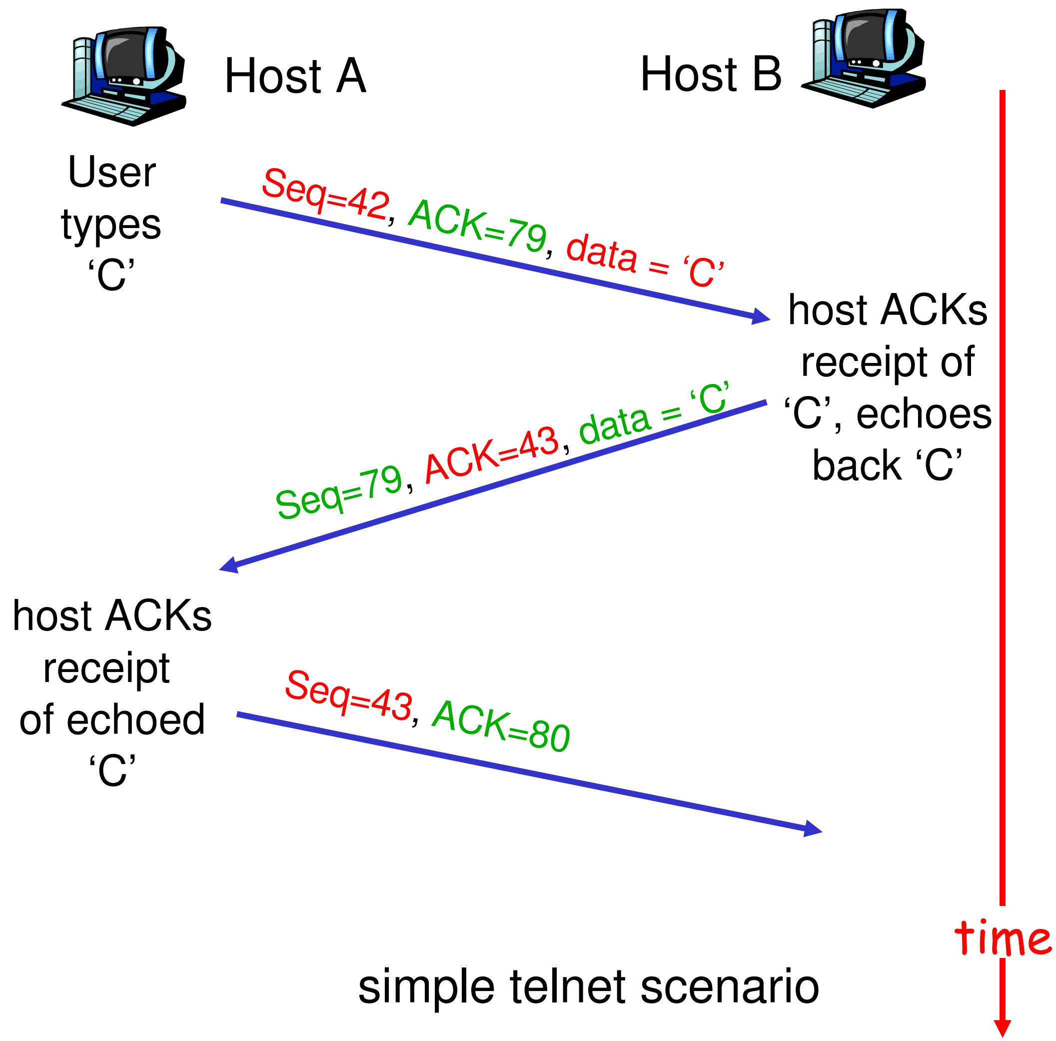
- byte stream "number" of first byte in segment's data

ACKs:

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

Q: how does receiver handle out-of-order segments

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



simple telnet scenario

TCP Round Trip Time (RTT) and Timeout

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?

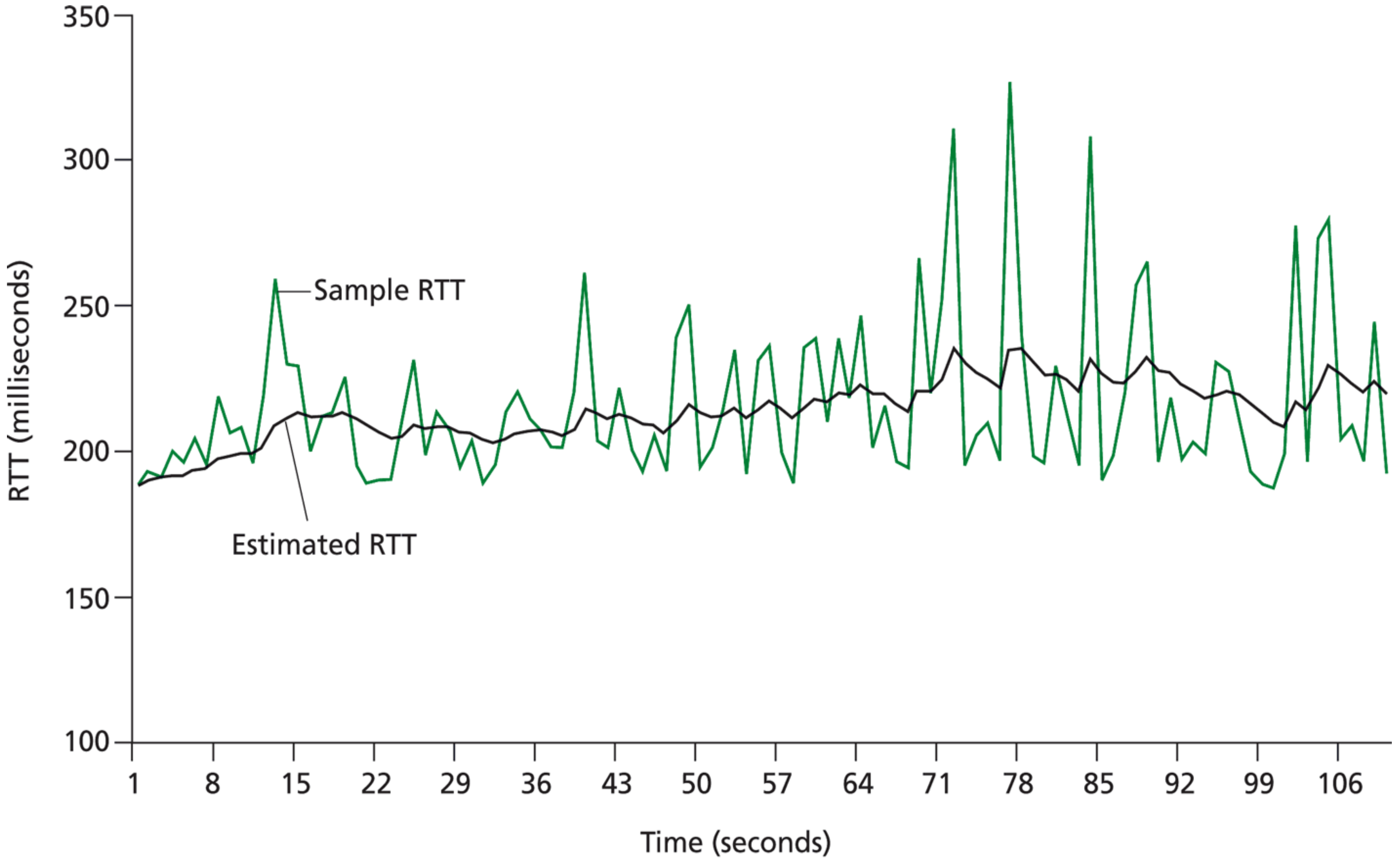
- `SampleRTT`: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- `SampleRTT` will vary, want estimated RTT "smoother"
  - average several recent measurements, not just current `SampleRTT`

TCP Round Trip Time and Timeout

$$\texttt{EstimatedRTT} =$$
$$(1-\alpha)*\texttt{EstimatedRTT} + \alpha*\texttt{SampleRTT}$$

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

Example RTT Estimation



TCP Round Trip Time and Timeout

Setting the timeout

- **`EstimatedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`** **->** larger safety margin
- first estimate of how much SampleRTT deviates from EstimatedRTT:

$$\texttt{DevRTT} = (1-\beta)\texttt{*DevRTT} + \beta\texttt{*|SampleRTT-EstimatedRTT|}$$

(typically, β = 0.25)

Then set timeout interval:

`TimeoutInterval = EstimatedRTT + 4*DevRTT`

TCP Reliable Data Transfer

- TCP creates reliable service on top of IP's unreliable service
- Pipelined segments
  - One segment encapsulated into one IP packet
  - MSS: Maximum segment size, approx. 1500 bytes
- Cumulative acks
- TCP uses single retransmission timer
- Retransmissions are triggered by:
  - timeout events
  - duplicate acks
- Initially consider simplified TCP sender:
  - ignore duplicate acks
  - ignore flow control, congestion control

TCP Sender Events:

- Data received from application:
  - 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 received:
  - If acknowledges previously unacked segments
    - update what is known to be acked
    - start timer if there are outstanding segments

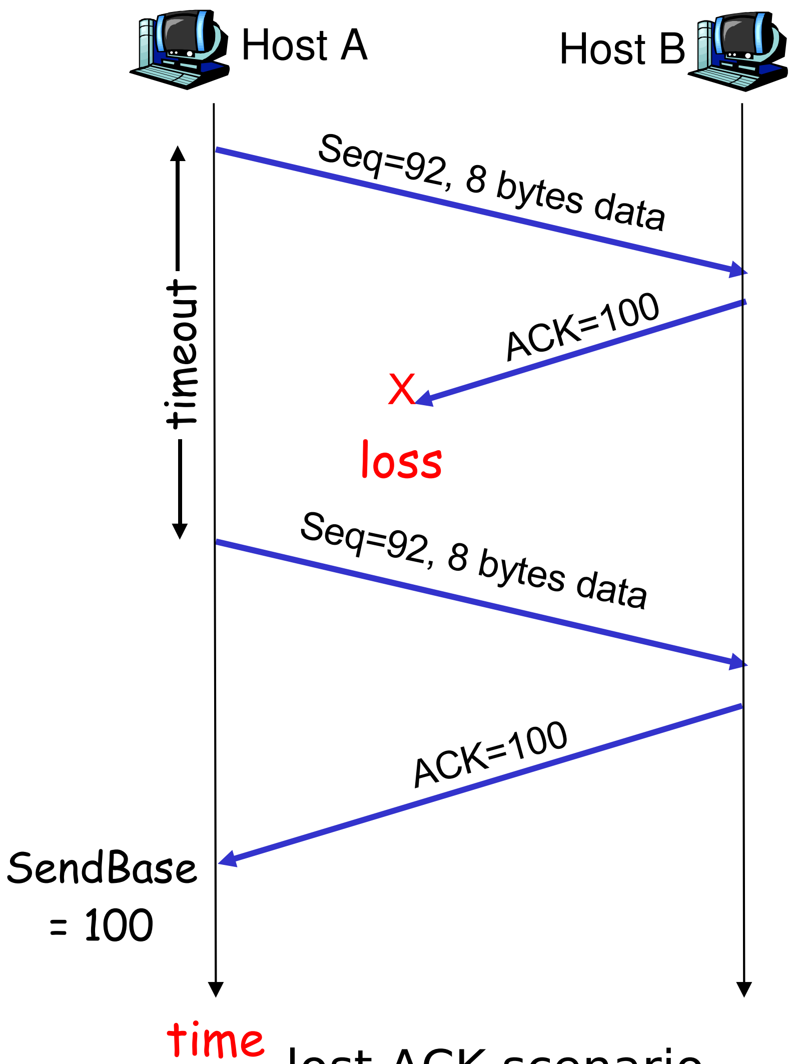
TCP Sender (Simplified)

```
NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum

 loop (forever) {
   switch(event)

   event: data received from application above
        create TCP segment with sequence number NextSeqNum
        if (timer currently not running)
              start timer
        pass segment to IP
        NextSeqNum = NextSeqNum + length(data)

    event: timer timeout
        retransmit not-yet-acknowledged segment with
                smallest sequence number
        start timer

    event: ACK received, with ACK field value of y
        if (y > SendBase) {
              SendBase = y
             if (there are currently not-yet-acknowledged segments)
                     start timer
             }
```

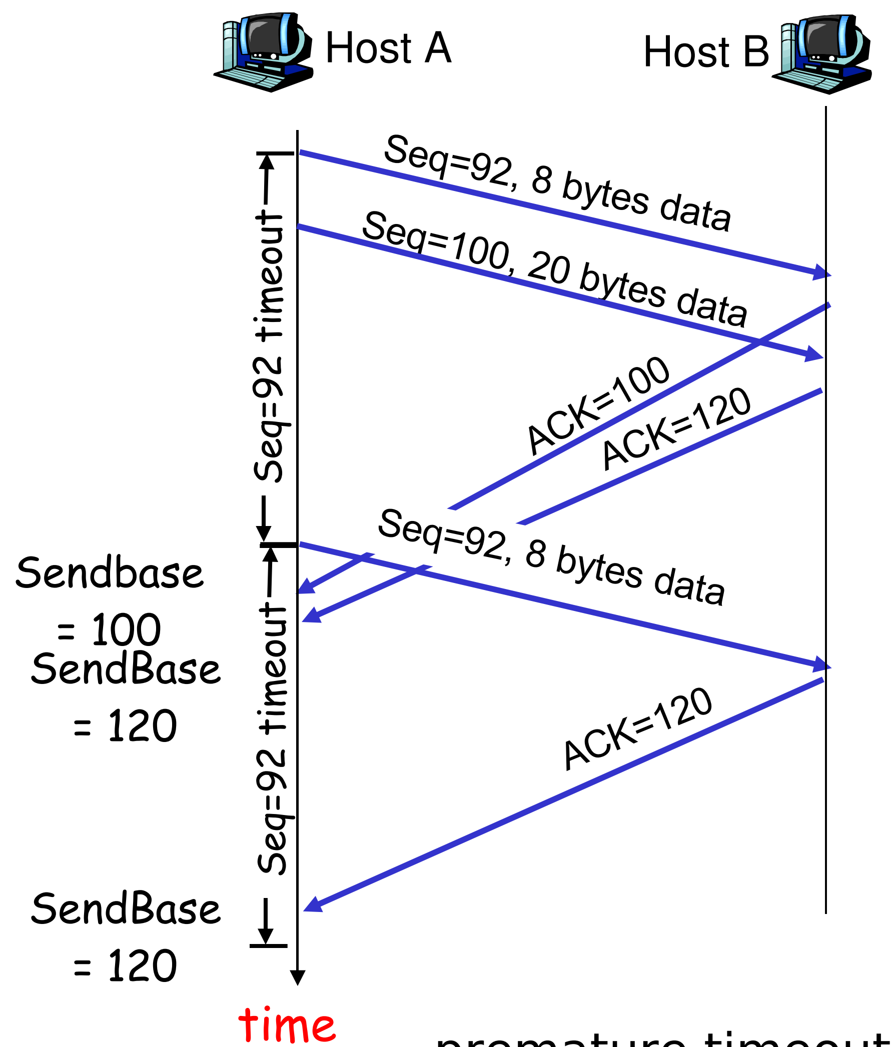
<u>Comment:</u>
- SendBase-1: last cumulatively ack'ed byte

<u>Example:</u>
- SendBase-1 = 71; y= 73, so the rcvr wants 73+ ; y > SendBase, so that new data is acked

TCP: Retransmission Scenarios

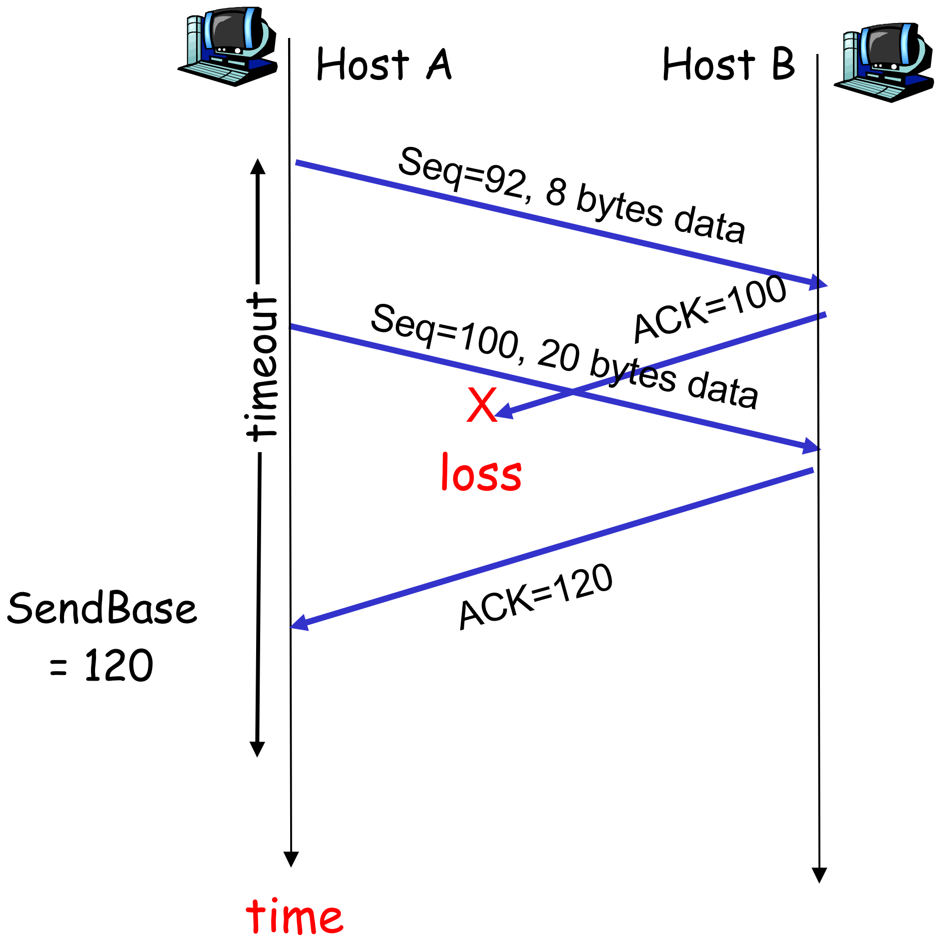


lost ACK scenario



premature timeout

TCP Retransmission Scenarios (more)



Cumulative ACK scenario

TCP ACK generation [RFC 1122, RFC 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

Fast Retransmit

- Time-out period often relatively long:
  - important to be conservative -> RTT estimation overestimates
  - 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
- If sender receives 3 ACKs for the same data, it supposes that segment after ACKed data was lost:
  - "ACK(i),ACK(i),ACK(i)": probably segment (i+1) was lost, afterwards at least two further segments (i+2,i+3) received and ackd by receiver
  - Fast retransmit: resend segment before timer expires

Fast Retransmit Algorithm

```
event: ACK received, with ACK field value of y :
          if (y > SendBase) {
             SendBase = y
             if (there are currently not-yet-ackd segments)
                start timer
           }
          else {
              increment count of dup ACKs received for y
              if (count of dup ACKs received for y = 3) {
                 resend segment with sequence number y
              }
```

a duplicate ACK for
already ACKed segment

fast retransmit

TCP Connection Management

Recall: TCP sender, receiver establish "connection" before exchanging data segments

- initialize TCP variables:
  - sequence #s
  - buffers, flow control info
- *client*: connection initiator

```
Socket clientSocket = new
 Socket("hostname","port

 number");
```

- *server*: contacted by client

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

Three way handshake:

Step 1: client host sends TCP SYN segment to server

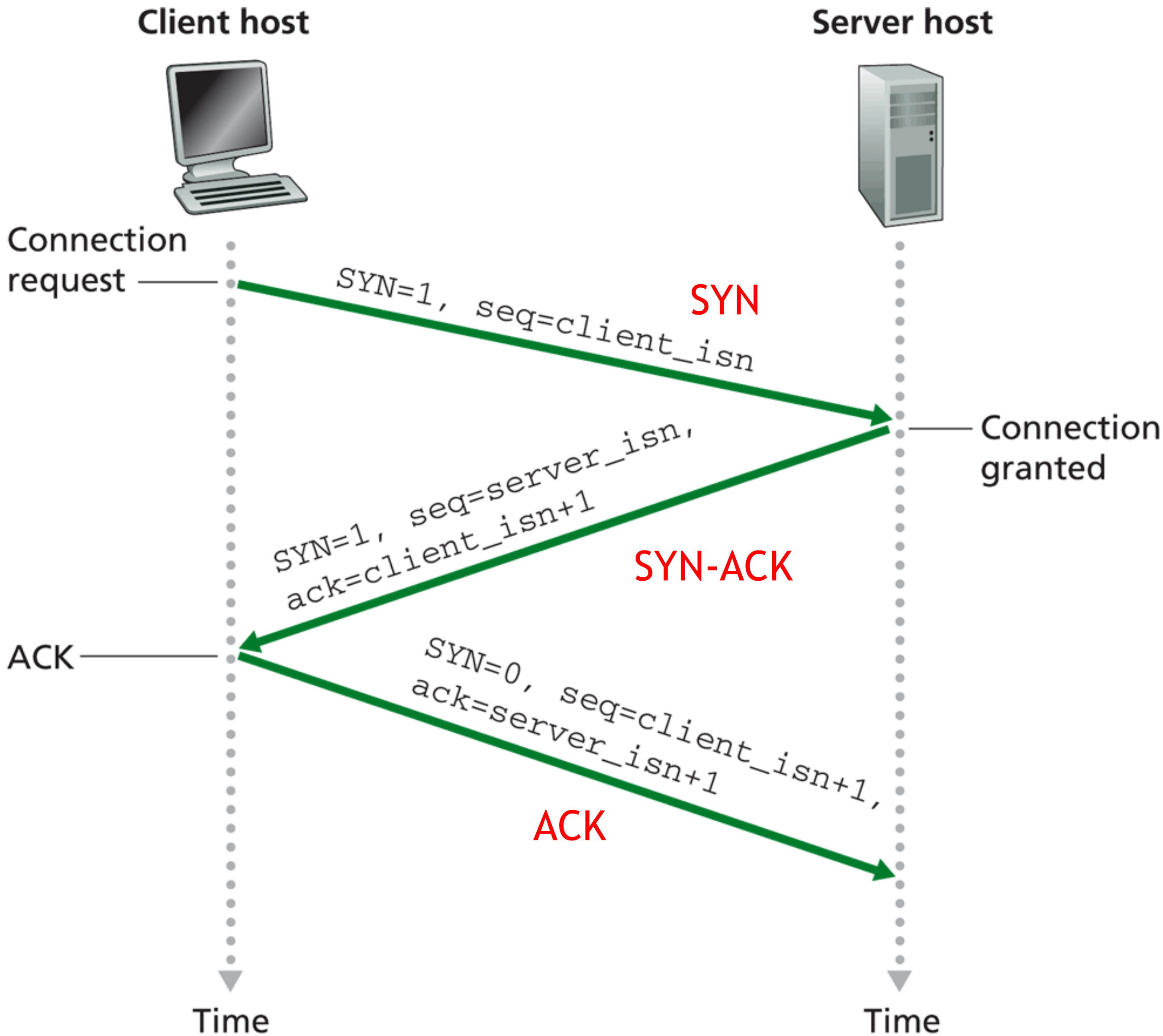
- specifies initial seq #
- no data

Step 2: server host receives SYN, replies with SYNACK segment

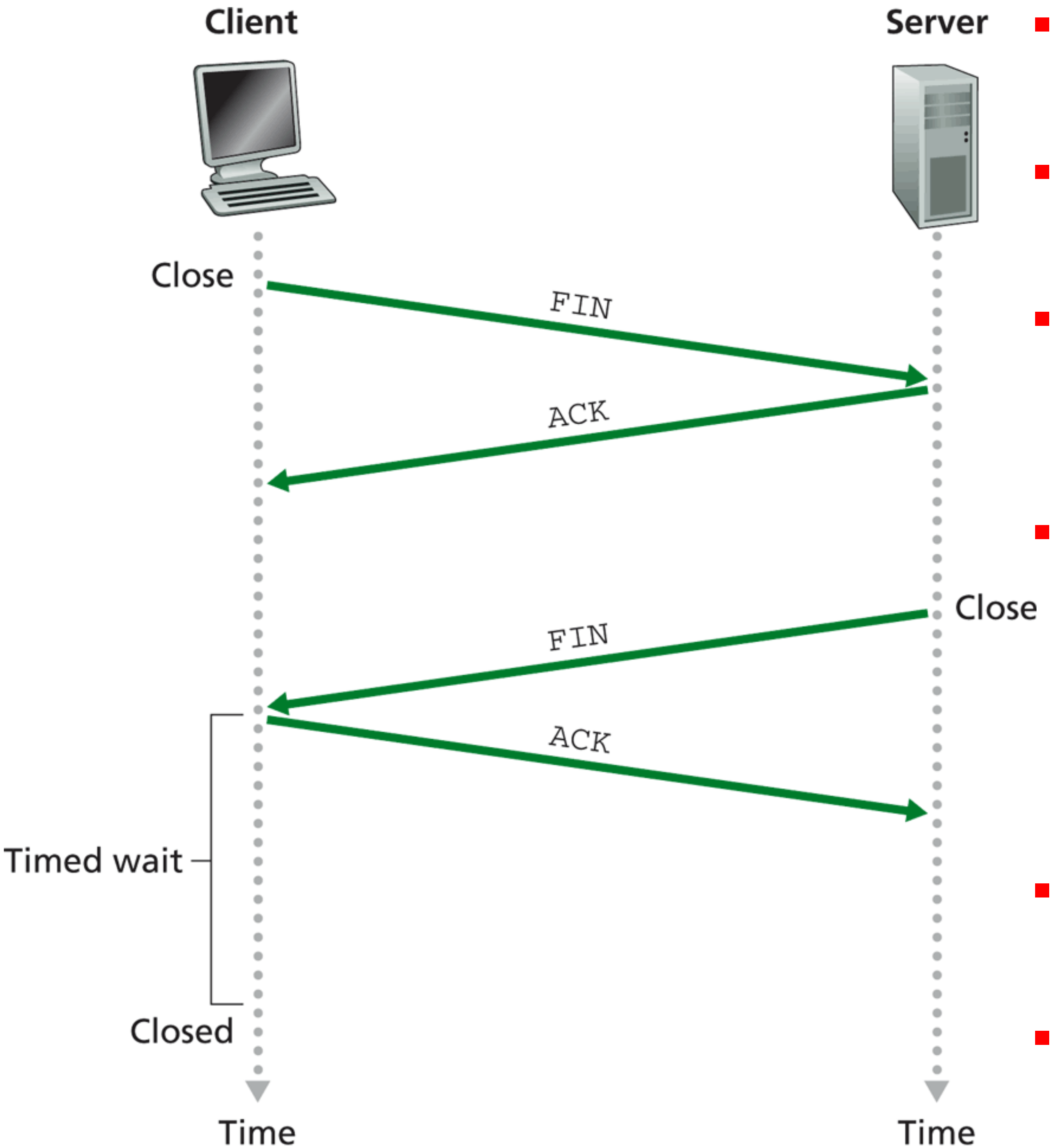
- server allocates buffers
- specifies server initial seq. #

Step 3: client receives SYNACK, replies with ACK segment, which may contain data

Three-Way Handshake for Connection Setup



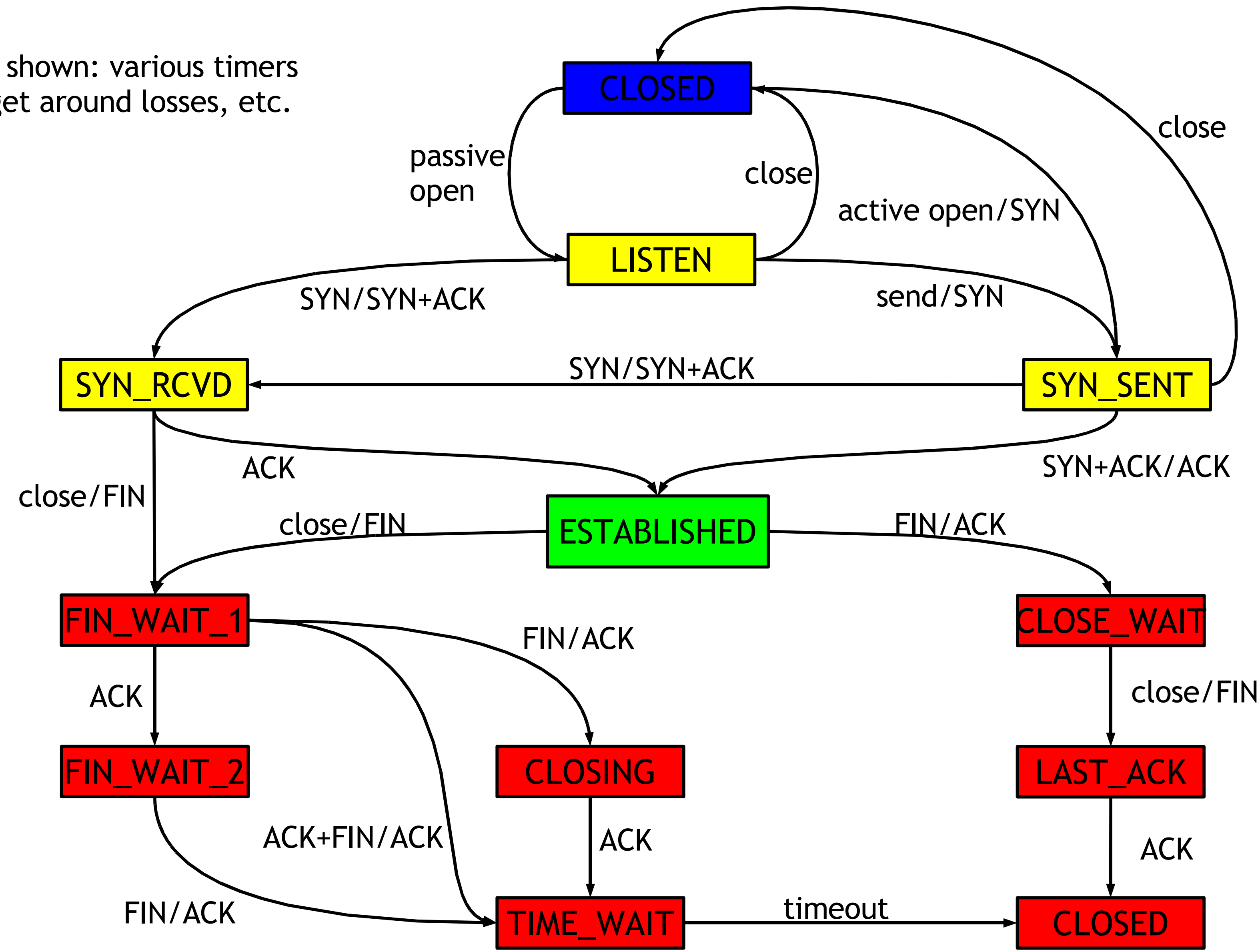
Connection Teardown



- Closing a connection:
  - client closes socket: clientSocket.close();
  - Step 1: client -> FIN ->server
  - Step 2: server receives FIN, replies with ACK. Closes connection, sends FIN.
  - 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: can handle simultaneous FINs.

TCP Connection FSM

Not shown: various timers
to get around losses, etc.



TCP Connection Management

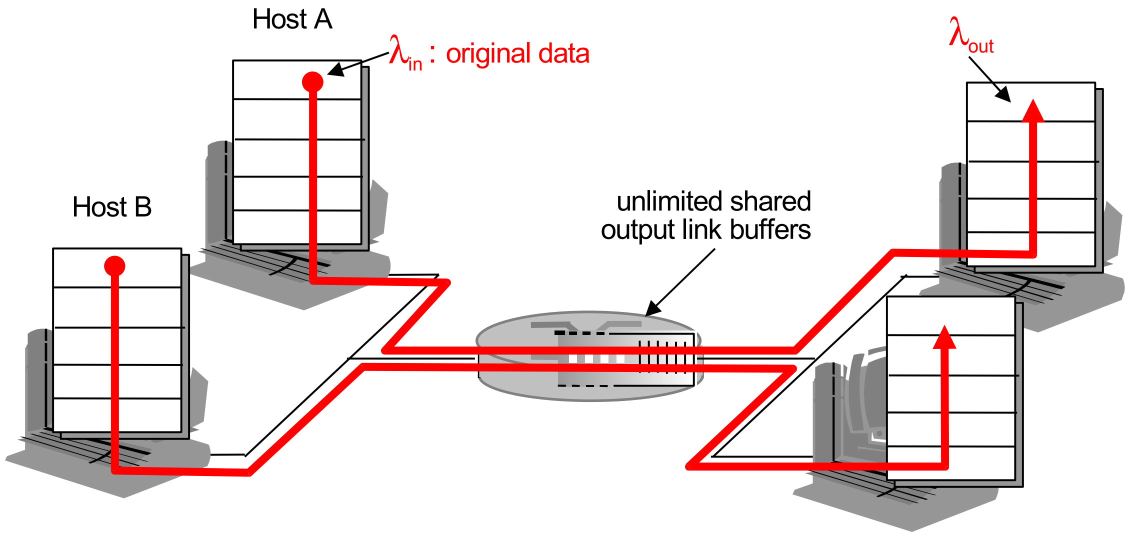
- Why so complex? Many challenges:
  - Handle simultaneous connection establishment and teardown requests from both ends
  - Packet reordering can lead to complex error conditions:
    - Connection (IP1,port1,IP2,port2) established, torn down, new connection (IP1,port1,IP2,port2) established: how to avoid leakage from first to second connection through old segments?
  - Teardown: connections can be half-open, i.e., able to receive, but not send
  - etc.

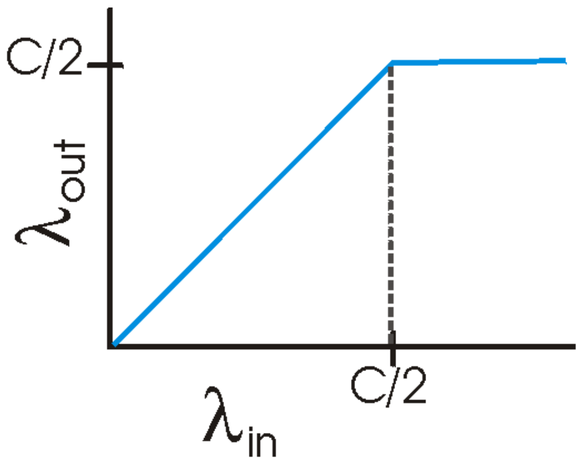
Principles of Congestion Control

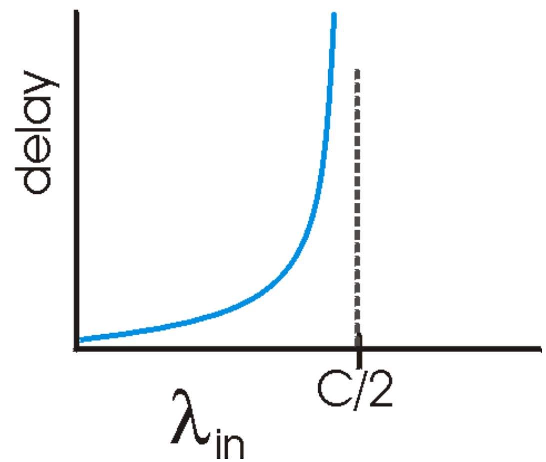
- Congestion:
  - informally: “too many sources sending too much data too fast for network to handle”
  - different from flow control!
  - manifestations:
    - lost packets (buffer overflow at routers)
    - long delays (queueing in router buffers)
  - a top-10 problem!

Causes/Costs of Congestion: Scenario 1

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



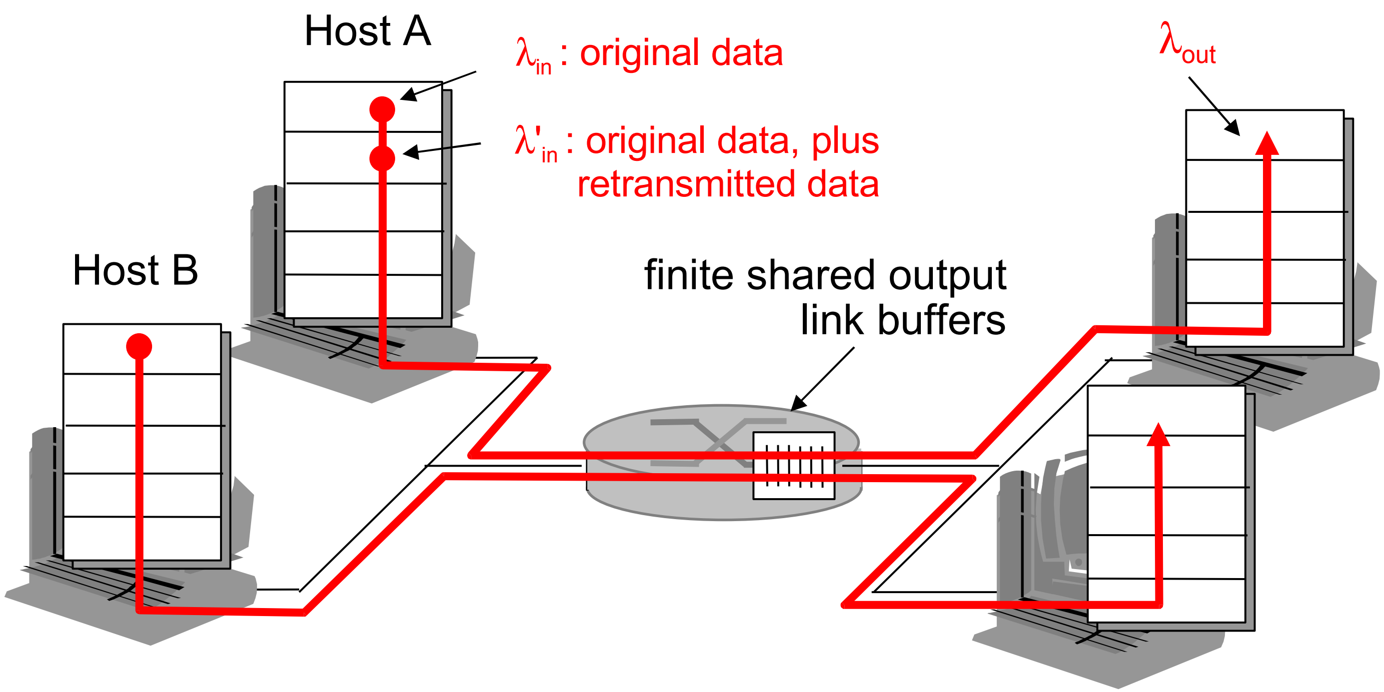




- large delays when congested
- maximum achievable throughput

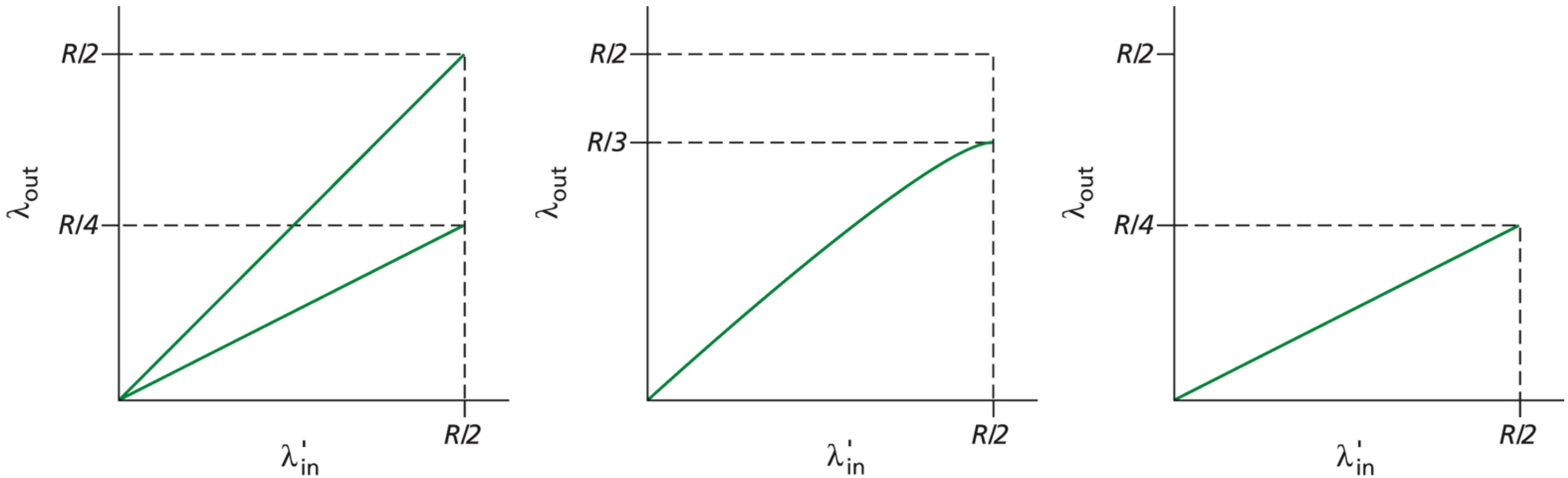
Causes/Costs of Congestion: Scenario 2

- 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}



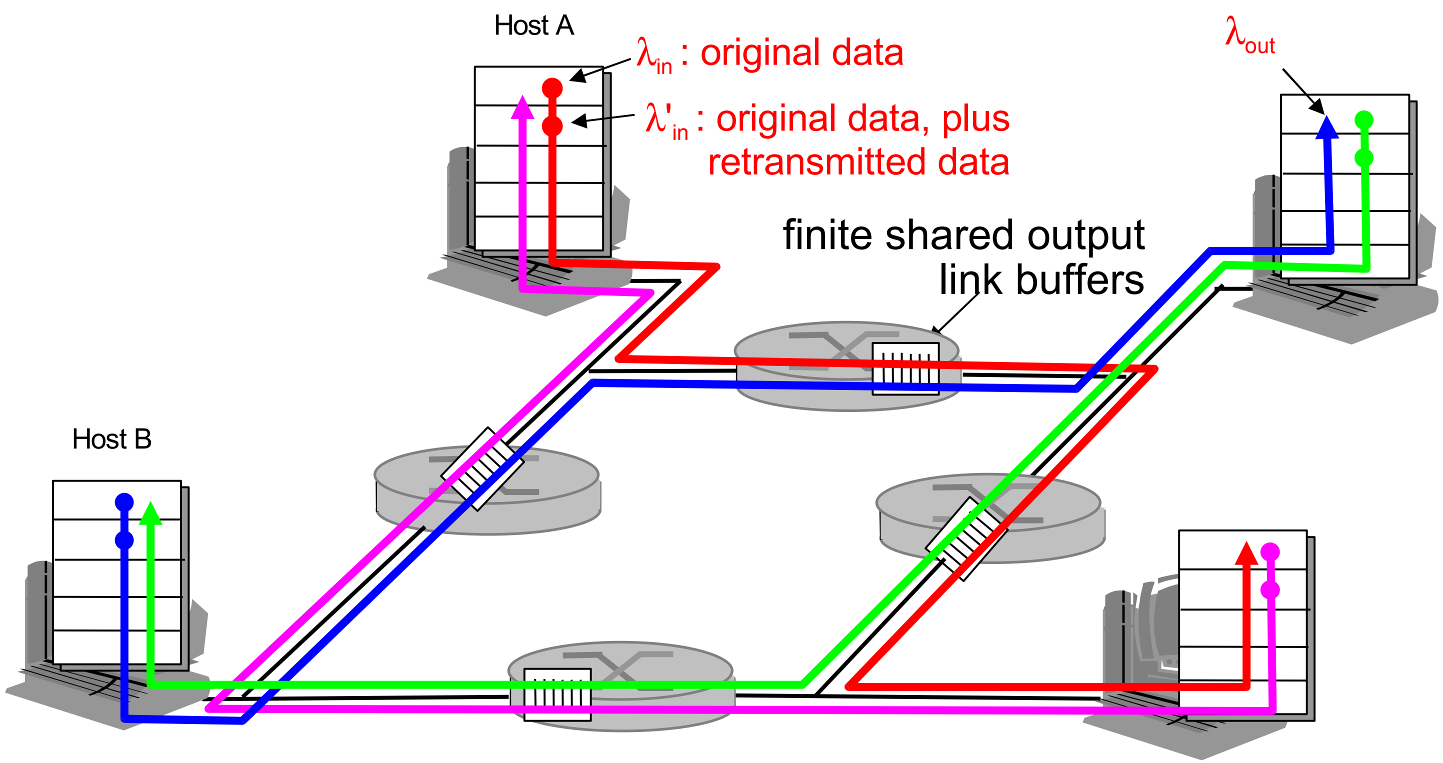
"costs" of congestion:

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

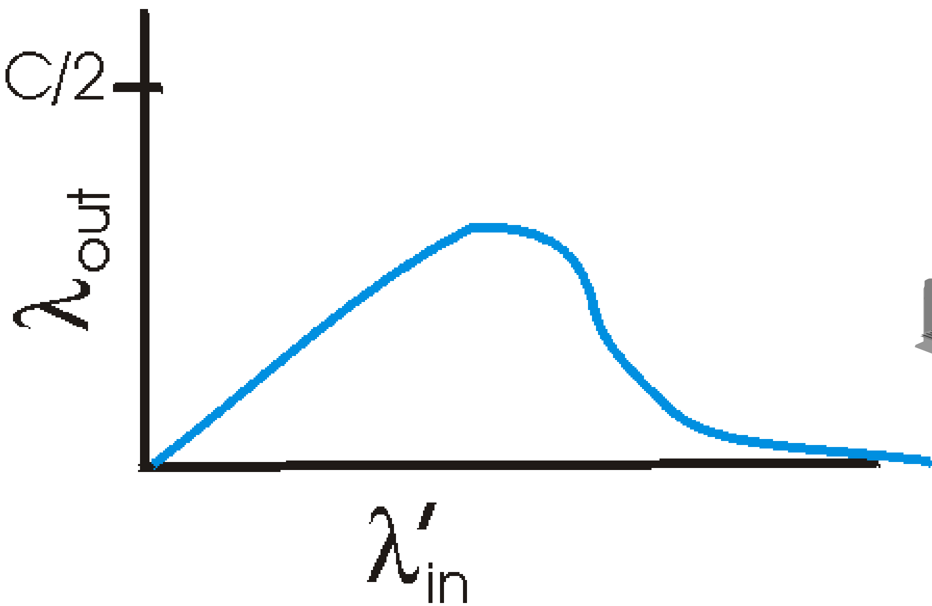
Causes/Costs of Congestion: Scenario 3

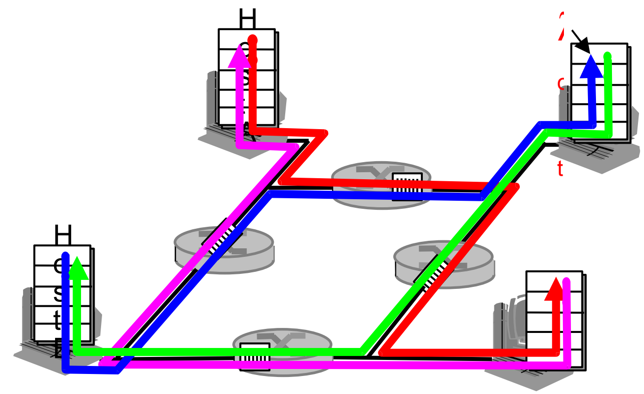
- four senders
- multihop paths
- timeout/retransmit

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

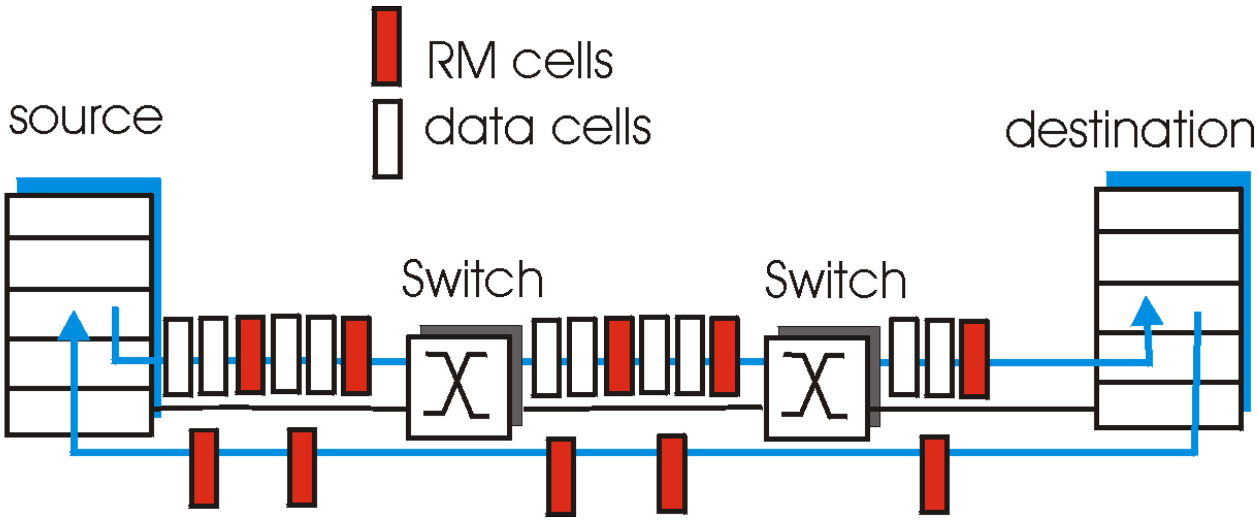
Network-assisted congestion control:

- 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, sender sets CI bit in returned RM cell

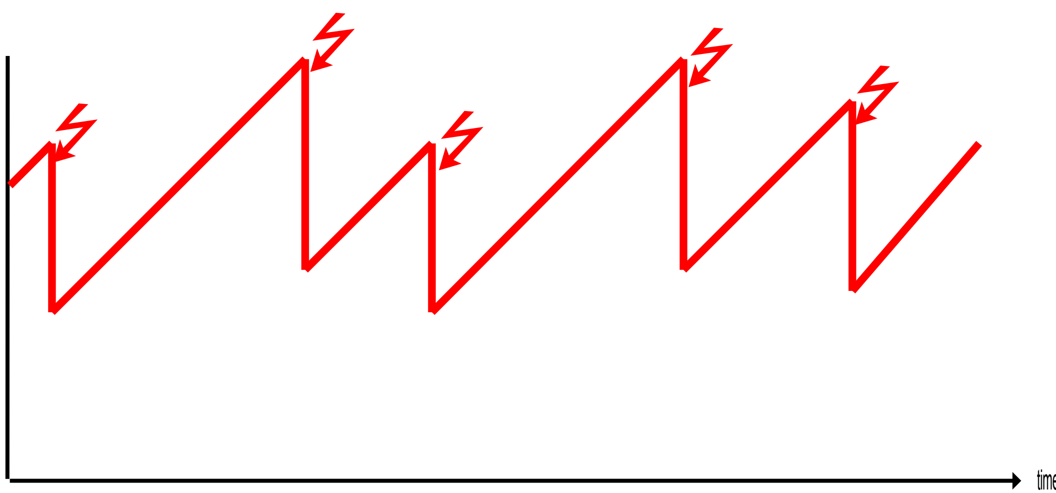
TCP Congestion Control

- End-end control (no network assistance)
- Sender limits transmission: `LastByteSent-LastByteAcked <= CongWin`
- rate R = $\frac{\text{CongWin}}{\text{RTT}}$ Bytes/sec
- Roughly,CongWin is dynamic, function of perceived network congestion

- How does sender perceive congestion?
  - Loss event = timeout or 3 duplicate acks
- TCP sender reduces rate (CongWin) after loss event
- Two mechanisms:
  - AIMD: Additive-Increase-Multiplicative-Decrease
  - Slow start: at beginning, after “really bad” congestion (timeout)

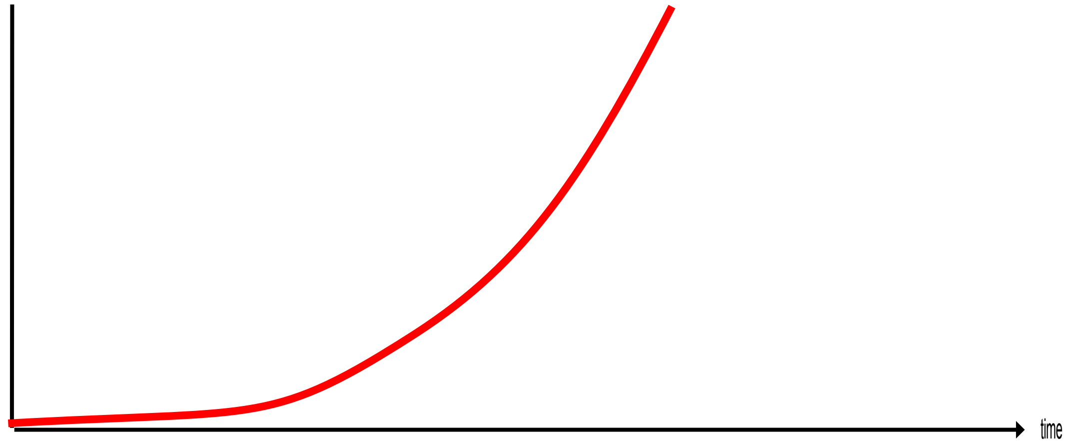
TCP AIMD in Congestion Avoidance (CA) State

- Approach:
  - Probe for available capacity by slowly increasing rate, back down when loss occurs
- Additive increase:
  - Increase CongWin by 1 MSS every RTT in the absence of loss events: probing
- Multiplicative decrease:
  - Cut CongWin in half after loss even



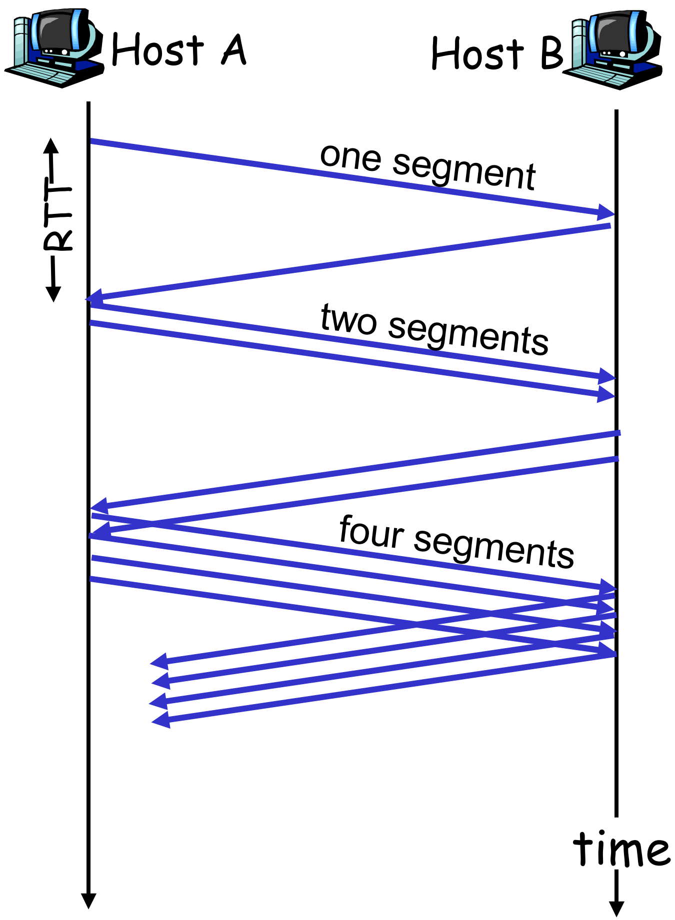
TCP Exponential Rampup in Slow Start State

- When connection begins: **`CongWin`** = 1 MSS
  - Example: MSS = 500 bytes & RTT = 200 msec
  - Initial rate R = 20 kbps
- Available bandwidth may be >> MSS/RTT
  - Desirable to quickly ramp up to respectable rate
- When connection begins, increase rate exponentially fast until first loss event



TCP Slow Start (more)

- When connection begins, increase rate exponentially until first loss event:
  - Double CongWin every RTT, by incrementing CongWin for every ACK received
- Summary:
  - Initial rate is slow
  - But ramps up exponentially fast!
  - (maybe it should be called FastStart...)



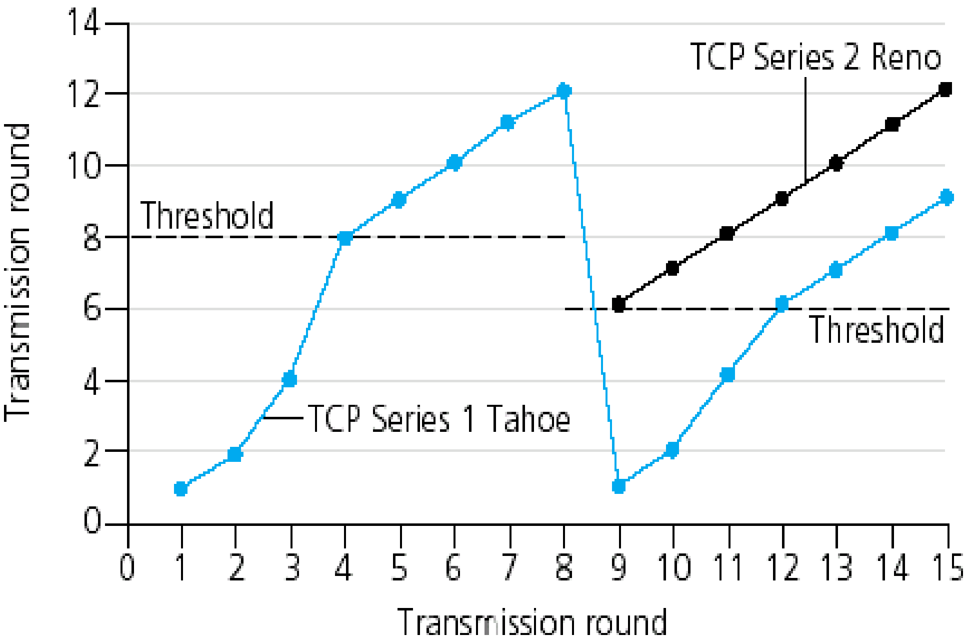
Refinement: SlowStart after Timeout

- After 3 dup ACKs, i.e., “ACK(i),ACK(i),ACK(i)”
  - CongWin is cut in half
  - Window then grows linearly
- But after timeout event: SlowStart again
  - CongWin instead set to 1 MSS;
  - Window then grows exponentially to a threshold,
  - Then grows linearly

- Philosophy:
  - 3 dup ACKs indicates network capable of delivering some segments -> keep going at lower rate
  - timeout before 3 dup ACKs is “more alarming” -> be more careful and slow down drastically

Refinement (more)

- Q: When should the exponential increase switch to linear?
  - When CongWin gets to 1/2 of its value before timeout
- Implementation:
  - Variable Threshold
  - At loss event, Threshold is set to 1/2 of CongWin just before loss event



TCP Sender Congestion Control

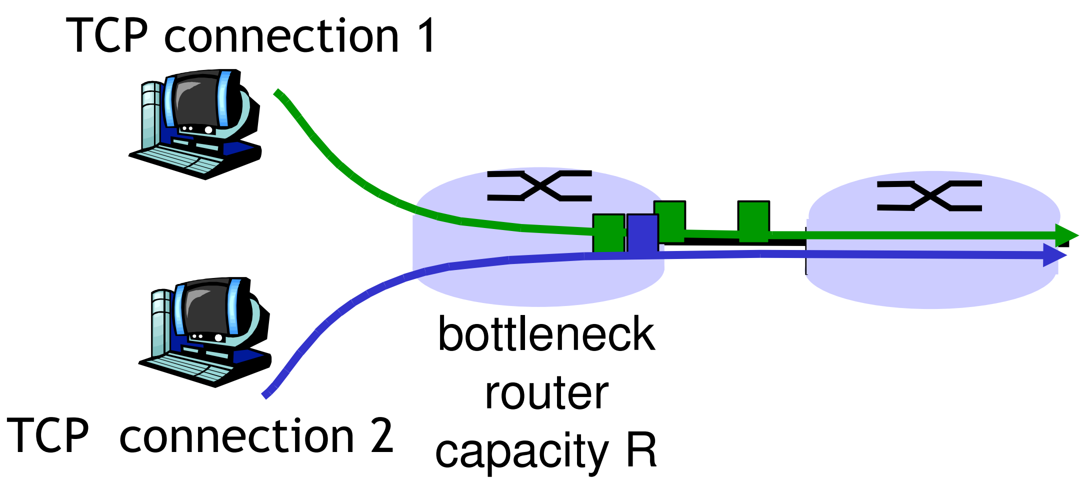
Event	State	TCP Sender Action	Commentary
ACK receipt for previously unacked data	Slow Start (SS)	CongWin = CongWin + MSS, If (CongWin > Threshold) set state to “Congestion Avoidance”	Resulting in a doubling of CongWin every RTT
ACK receipt for previously unacked data	Congestion Avoidance (CA)	CongWin = CongWin+MSS * (MSS/CongWin)	Additive increase, resulting in increase of CongWin by 1 MSS every RTT
Loss event detected by triple duplicate ACK	SS or CA	Threshold = CongWin/2, CongWin = Threshold, Set state to “Congestion Avoidance”	Fast recovery, implementing multiplicative decrease. CongWin will not drop below 1 MSS.
Timeout	SS or CA	Threshold = CongWin/2, CongWin = 1 MSS, Set state to “Slow Start”	Enter slow start
Duplicate ACK	SS or CA	Increment duplicate ACK count for segment being acked	CongWin and Threshold not changed

Summary: TCP Congestion Control

- When **`CongWin`** is below **`Threshold`**, sender in slow-start phase, window grows exponentially.
- When **`CongWin`** is above **`Threshold`**, sender is in congestion-avoidance phase, window grows linearly.
- When a triple duplicate ACK occurs, **`Threshold`** set to **`CongWin/2`** and **`CongWin`** set to **`Threshold`**.
- When timeout occurs, **`Threshold`** set to **`CongWin/2`** and **`CongWin`** is set to 1 MSS.

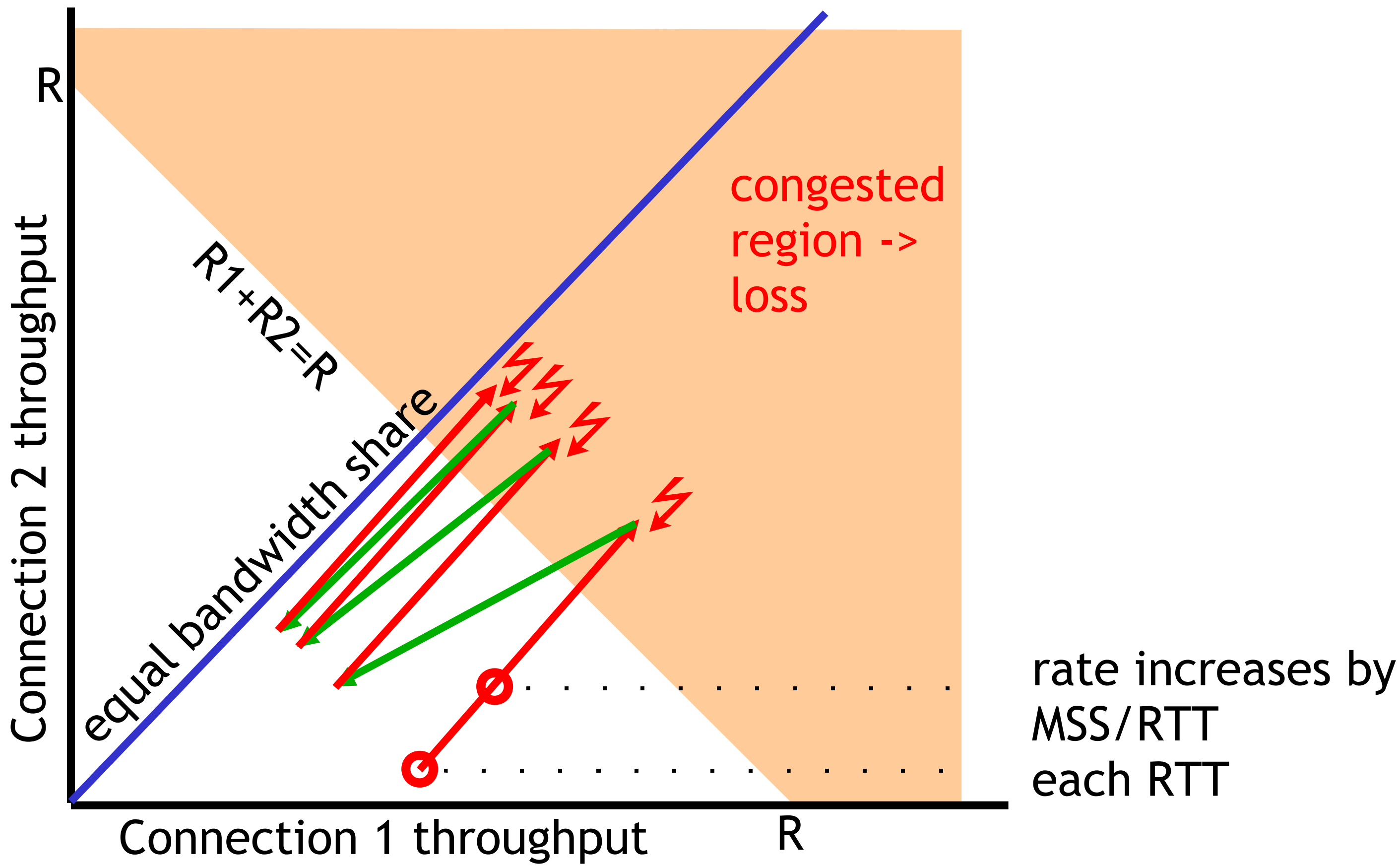
TCP Fairness

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?

- Two competing sessions:
  - Additive increase gives slope of 1, as throughput increases
  - multiplicative decrease decreases throughput proportionally

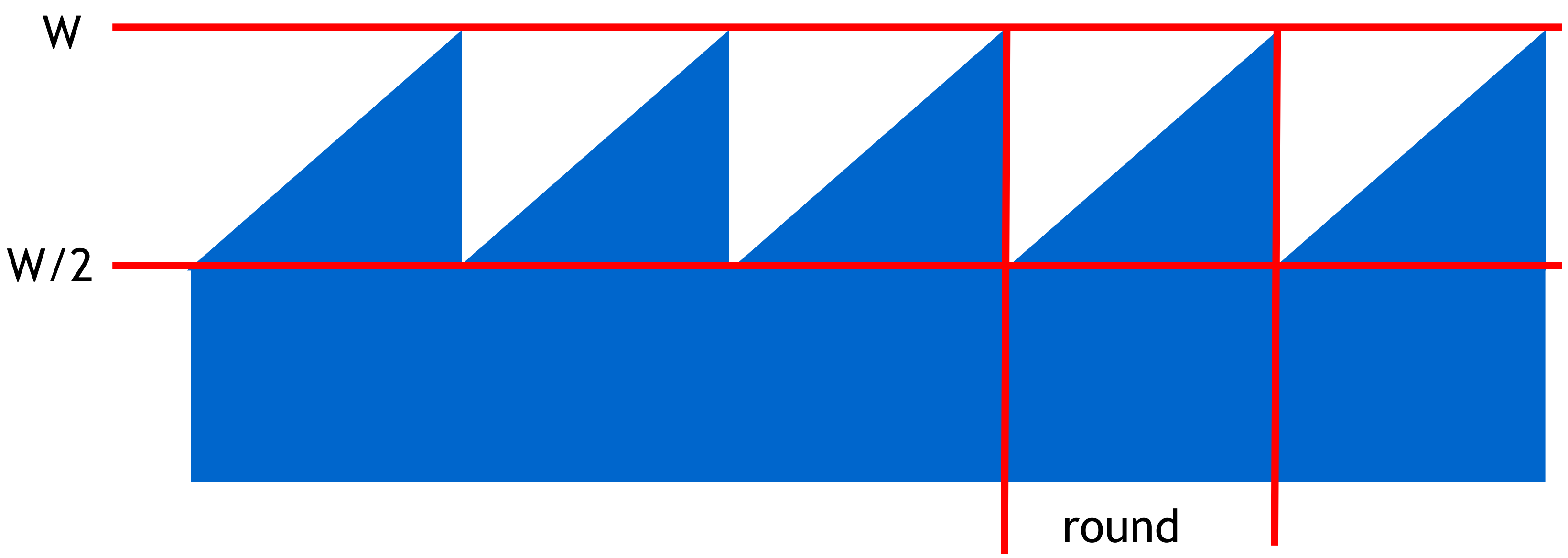


Fairness (more)

- Fairness and UDP
  - Multimedia apps often do not use TCP
    - do not want rate throttled by congestion control
  - Instead use UDP:
    - pump audio/video at constant rate, tolerate packet loss
  - Research area: TCP friendly
- Fairness and parallel TCP connections
  - nothing prevents app from opening parallel connections between 2 hosts.
  - Web browsers do this
  - Example: link of rate R supporting 9 existing connections
    - new app asks for 1 TCP, gets rate R/10
    - new app asks for 11 TCPs, gets R/2!

TCP Throughput

- Goal: compute average throughput R(L,RTT)
  - Ignore slow start
- Model: perfect sawtooth
  - Loss exactly periodic
  - No RTT fluctuation



TCP Throughput (more)

- Let W be the window size when loss occurs
  - When window is W, throughput is W/RTT
  - Just after loss, throughput drops to W/(2 RTT)
  - Average throughout: R=0.75 W/RTT
- In one round:
  - One packet lost
  - K=(W/2)+(W/2+1)+...+W packets sent
  - $K = 3/8\ W^2 + ¾\ W \approx 3/8\ W^2$ for W>>1
- Loss rate:
  - $L = 1/K = 8/3\ W^{-2}$
- Average rate:

$$R = \frac{1.22\ MSS}{RTT\sqrt{L}}$$

TCP Futures

- Example: MSS=1500 byte segments, 100ms RTT, want 10 Gbps throughput
  - Requires window size W = 83333 in-flight segments
  - Throughput in terms of loss rate:

$$R = \frac{1.22\, MSS}{RTT\sqrt{L}}$$

  - Solve for L ➜ **$L = 2 \cdot 10^{-10}$**
    - one segment in 5'000'000'000 lost: impossibly low!
    - bit error rate in optical fibers in same ballpark, wireless bit error rate many orders of magnitude higher
- Traditional TCP cannot operate in this regime
  - New versions of TCP for high-speed needed!

Summary

- Reliable transport in TCP:
  - cumulative ACKs
  - window, sequence numbers in bytes
  - bi-directional
  - optimizations: adaptive timer through RTT estimation, fast retransmit
- Window size is dynamic, control parameter for:
  - flow control
  - congestion control
- TCP congestion control:
  - goal: utilize network fully without overloading, fairness
  - fully distributed, end-to-end; no explicit network signal for congestion
  - "discovery" of fair share through slow start/congestion avoidance
- Next:
  - leaving the network "edge"
  - into the "core" -> routers