School of Computing Science
Simon Fraser University

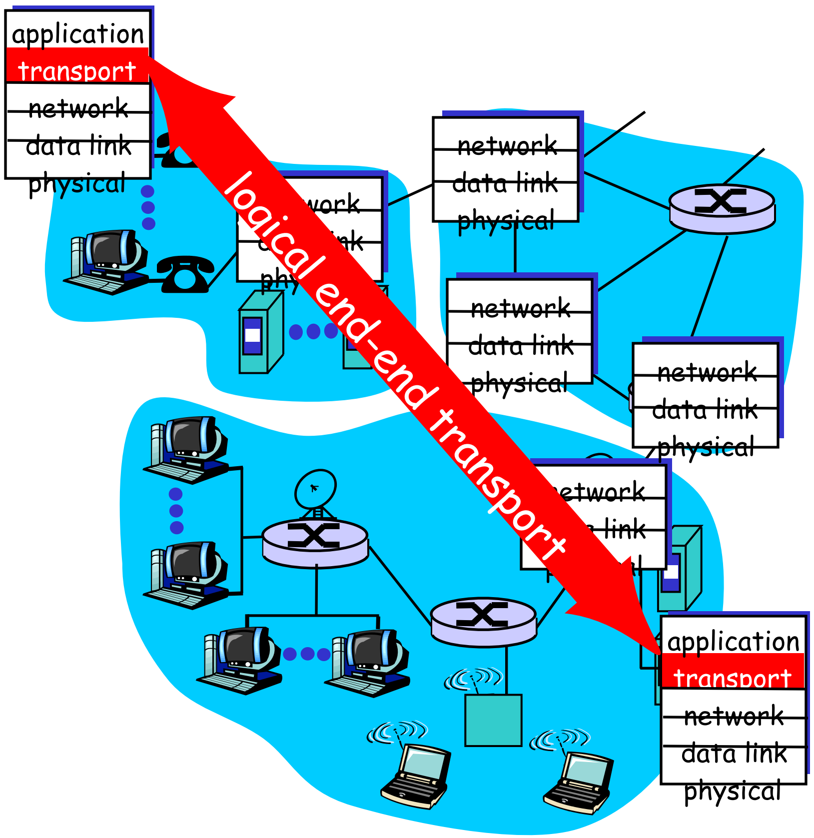
CMPT 471: Networking II

Transport Layer

Instructor: Mohamed Hefeeda

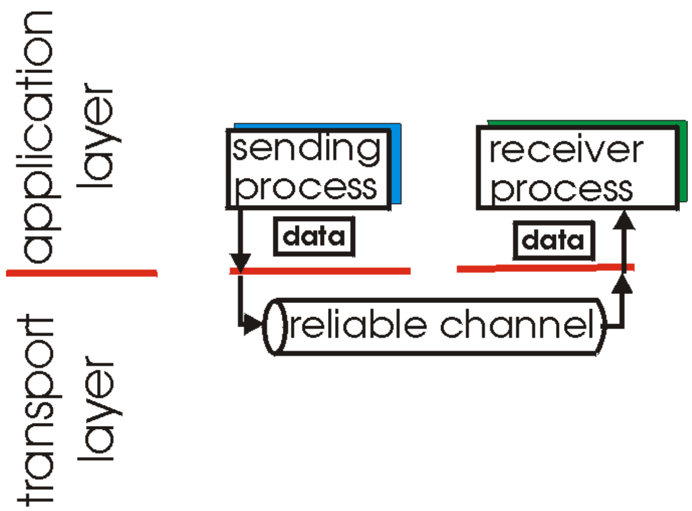
Transport services and protocols

- provide logical communication between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into segments, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP



Reliable data transfer

- important in application, transport, and link layers
- top-10 list of important networking topics!

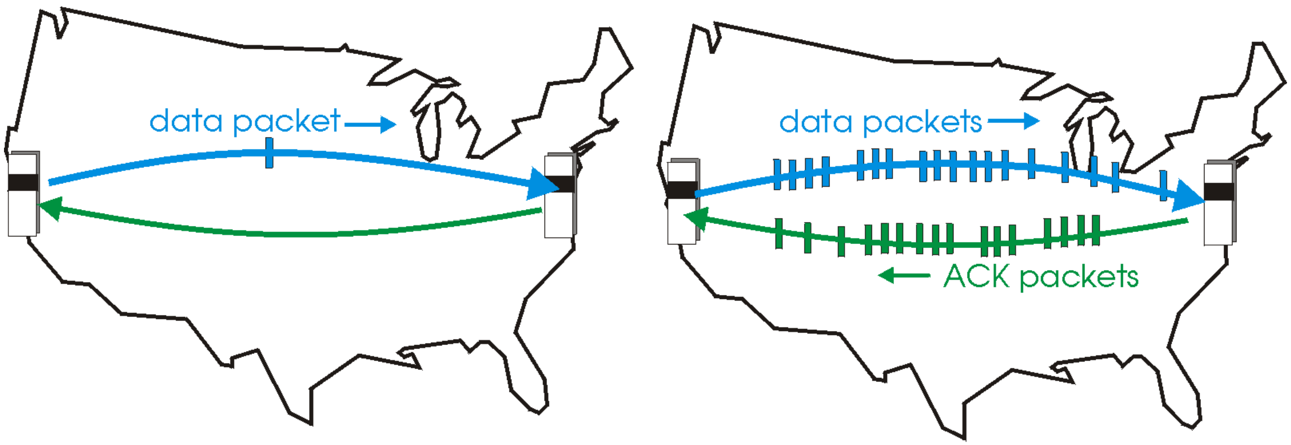


(a) provided service

Pipelined (Sliding Window) Protocols

Pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged pkts

- range of sequence numbers must be increased
- buffering at sender and/or receiver



(a) a stop-and-wait protocol in operation

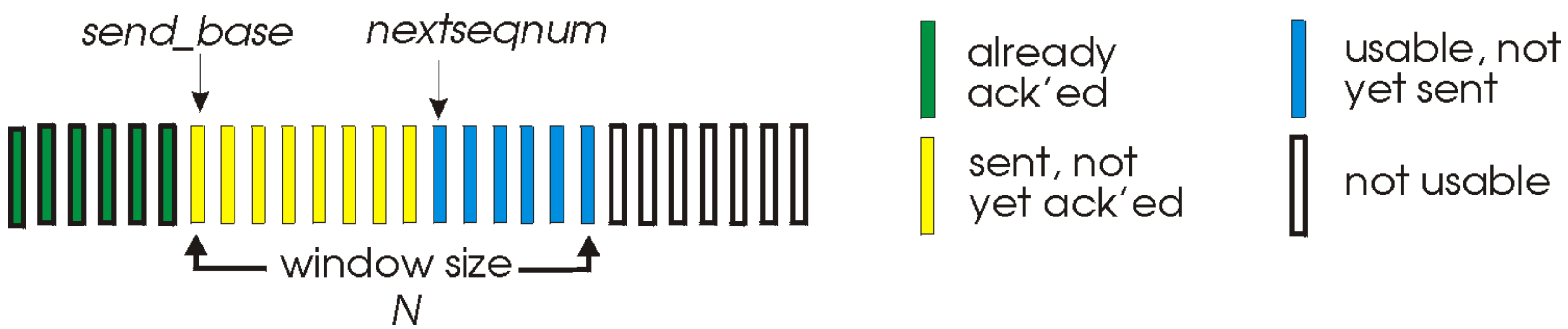
(b) a pipelined protocol in operation

- Two generic forms of pipelined protocols: go-Back-N, selective repeat

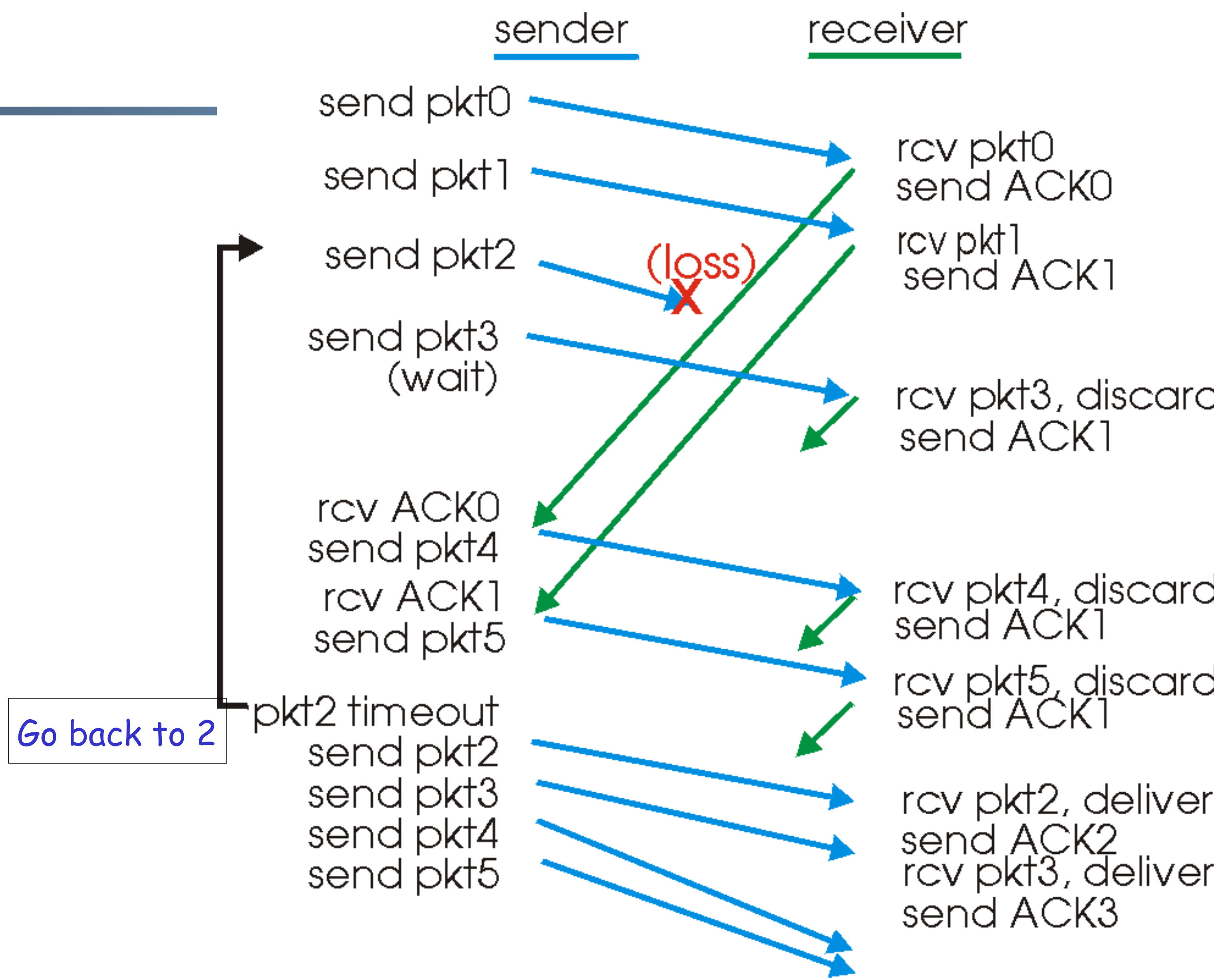
Go-Back-N

Sender:

- k-bit seq # in pkt header
- "window" of up to N consecutive unack'ed pkts allowed

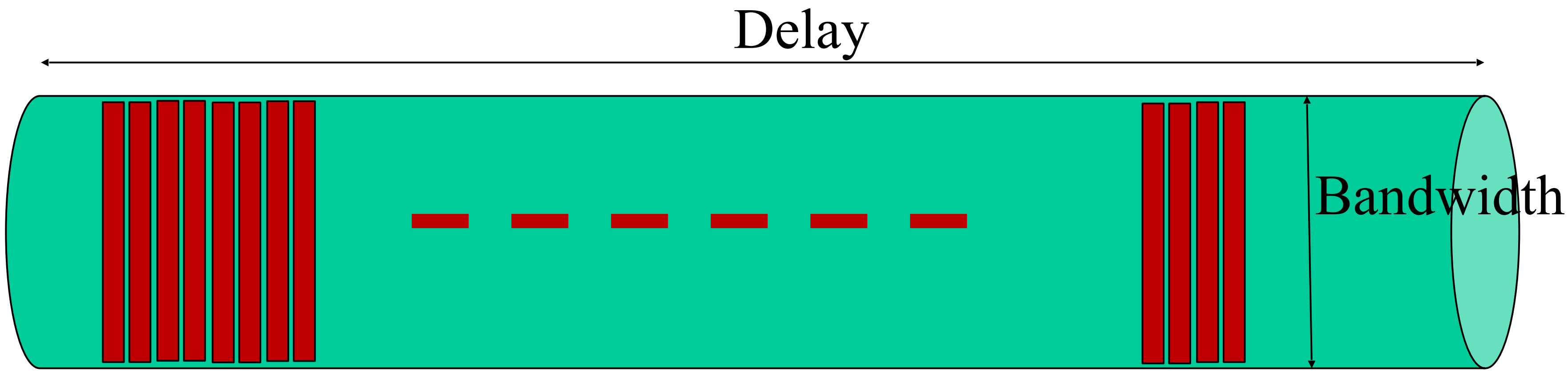


- ACK(n): ACKs all pkts up to and including seq # n -- cumulative ACK
  - may receive duplicate ACKs
- timer for each in-flight pkt
- timeout(n): retransmit pkt n and all higher seq # pkts in window
  - i.e., go back to n



Window size, N = 4

Go-Back-N: Problems?



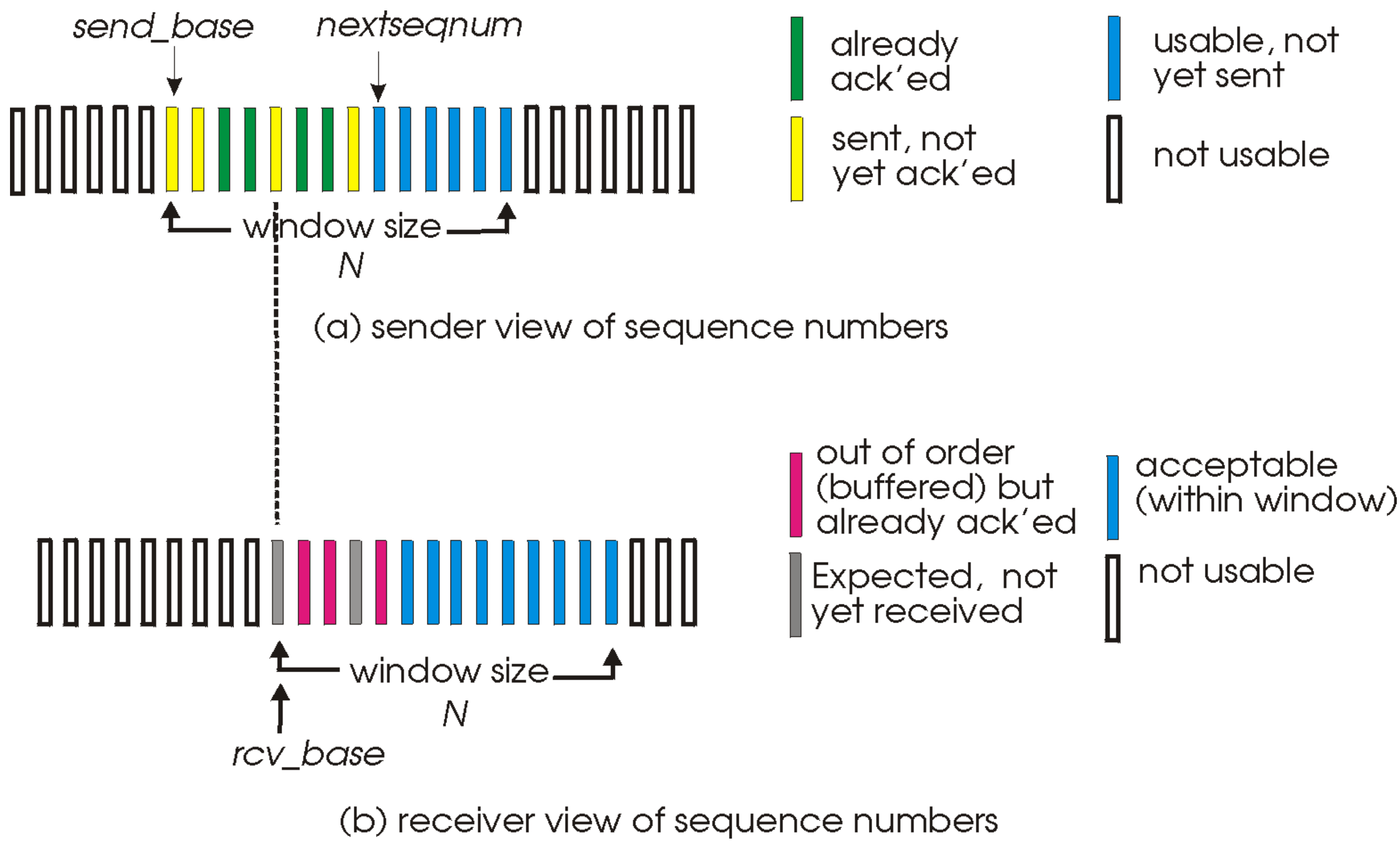
- consider high-speed links with long delays
  - (called large bandwidth-delay product pipes)
- GBN can fill that pipe by having large N →
- many unACKed pkts could be in the pipe
- A single lost pkt could cause re-transmission of huge number (up to N) of pkts → waste of bandwidth

- Solutions??

Selective Repeat

- receiver individually acknowledges correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
  - N consecutive seq #'s
  - again limits seq #s of sent, unACKed pkts

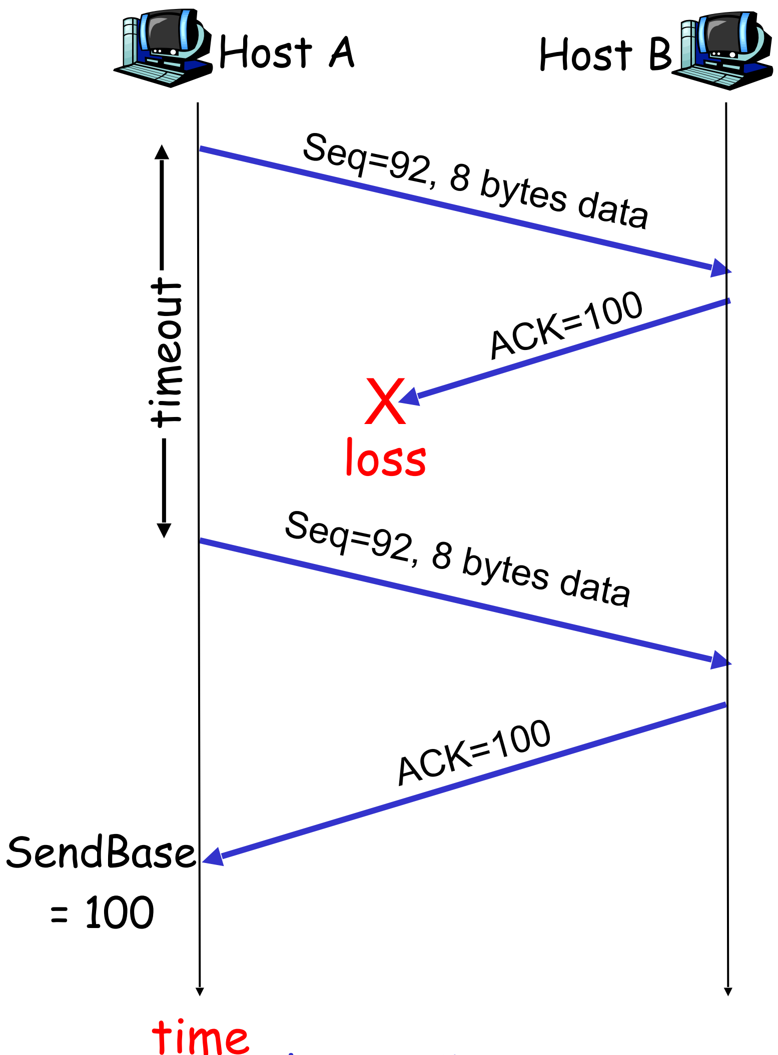
Selective repeat: sender, receiver windows



TCP reliable data transfer

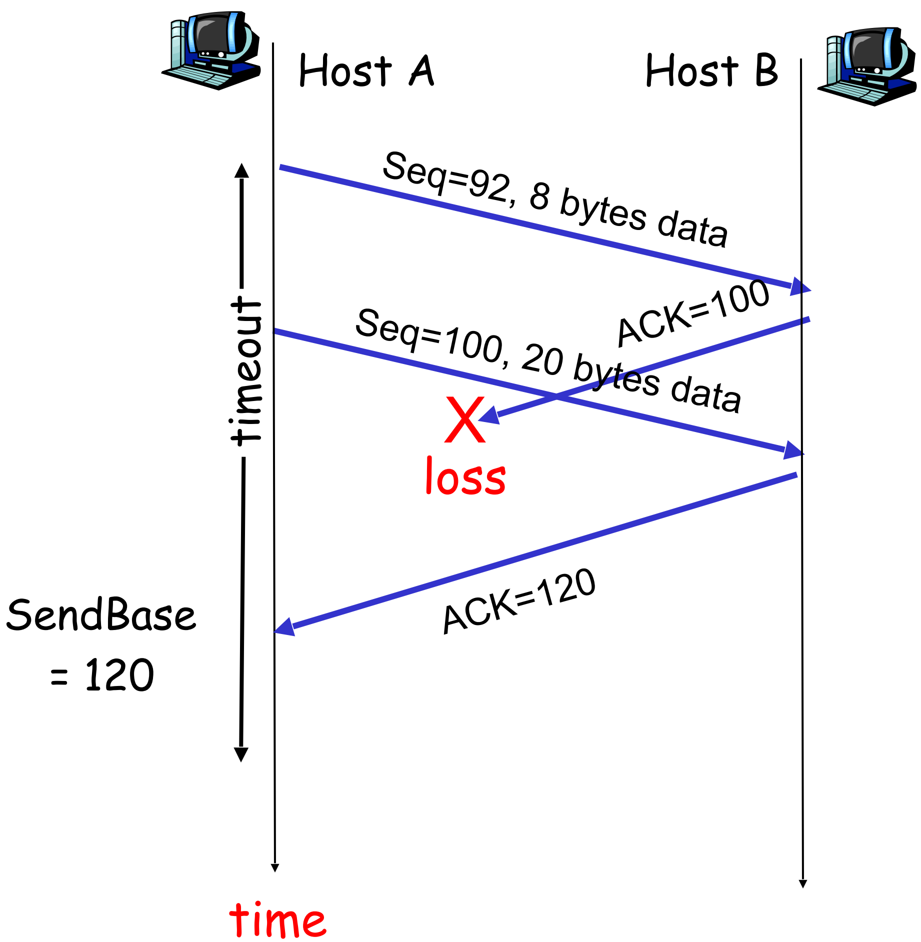
- TCP creates rdt service on top of IP's unreliable service
- Pipelined segments
- Cumulative acks
- TCP uses single retransmission timer
  - Why single timer?
- Retransmissions are triggered by:
  - timeout events
  - duplicate acks
- Initially consider simplified TCP sender:
  - ignore duplicate acks
  - ignore flow control, congestion control

TCP: retransmission scenarios



lost ACK scenario

TCP retransmission scenarios (more)



Cumulative ACK scenario

TCP Round Trip Time and Timeout

- If TCP timeout is
  - too short: premature timeout → unnecessary retransmissions
  - too long: slow reaction to segment loss

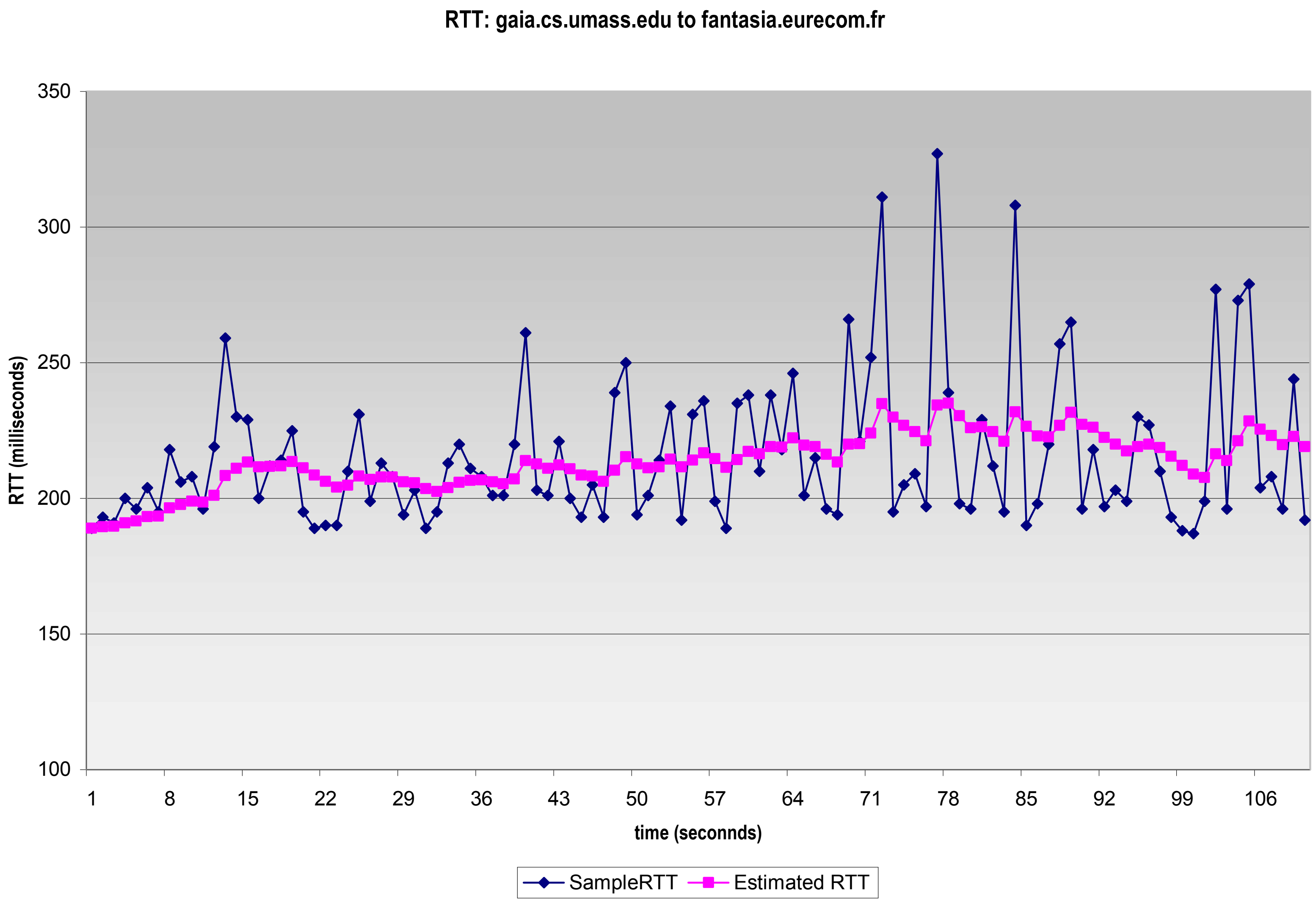
Q: how to set TCP timeout value?

- Based on Round Trip Time (RTT), but RTT varies with time!
- → We estimate current RTT

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

- Exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: $\alpha = 0.125$ (→ efficient computation why?)

Example RTT estimation:



TCP Round Trip Time and Timeout

Setting the timeout

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

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

(typically, β = 0.25)
```

Then set timeout interval:

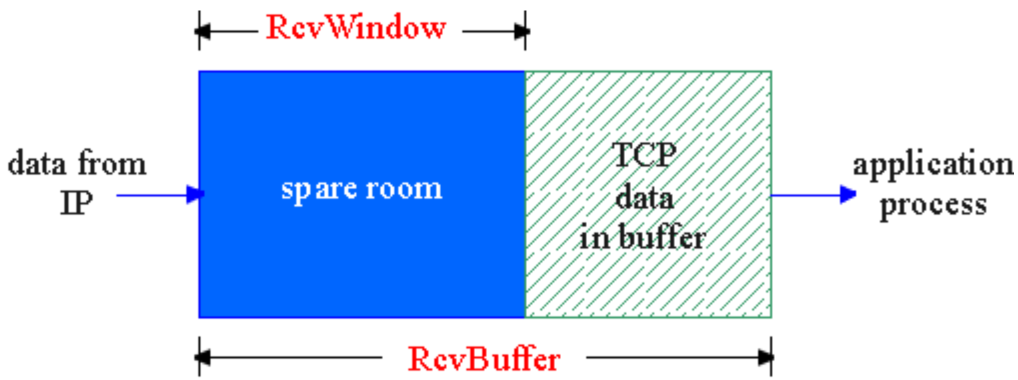
```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

Fast Retransmit

- Time-out period often relatively long:
  - 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:
  - <u>fast retransmit:</u> resend segment before timer expires

TCP Flow Control

- receive side of TCP connection has receive buffer:



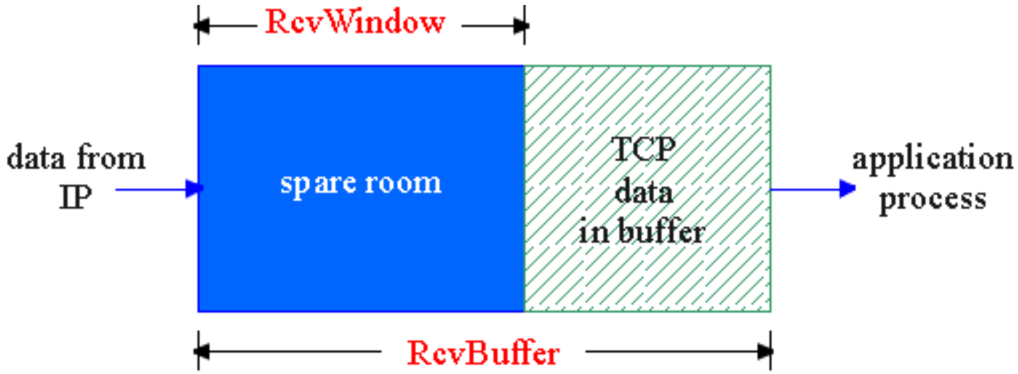
- app process may be slow at reading from buffer

flow control
sender won't overflow receiver's buffer by transmitting too much, too fast

- speed-matching service: matching send rate to receiving app's drain rate

- Flow control is end to end

TCP Flow control: how it works



(Suppose TCP receiver discards out-of-order segments)

- spare room in buffer

= `RcvWindow`

= `RcvBuffer-[LastByteRcvd - LastByteRead]`

- Rcvr advertises spare room by including value of `RcvWindow` in segments
- Sender limits unACKed data to `RcvWindow`
  - guarantees receive buffer doesn't overflow

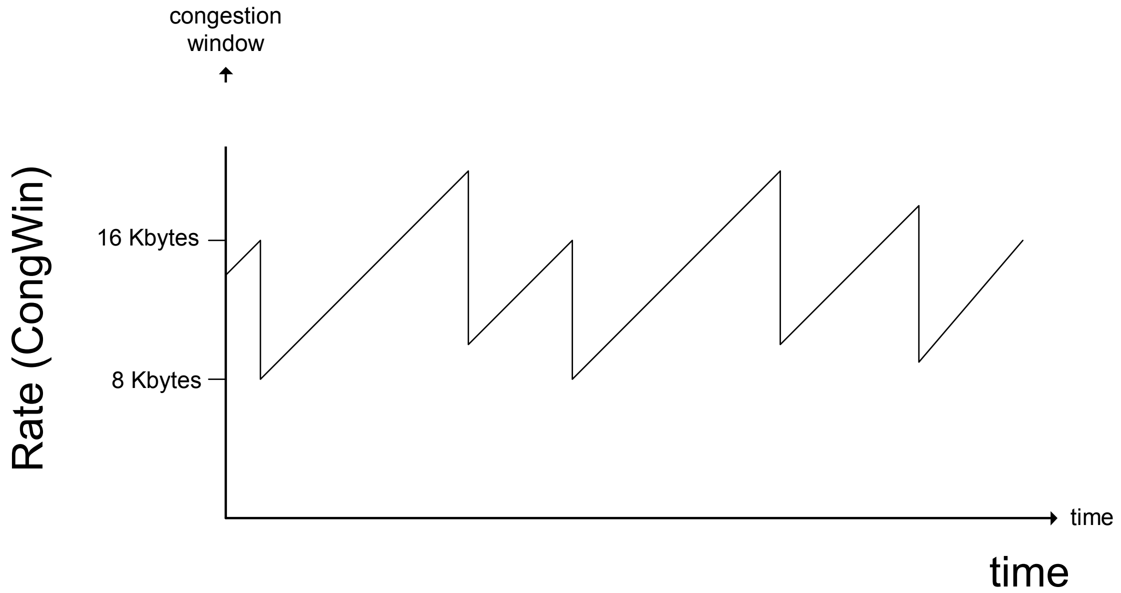
Congestion Control

- Congestion: sources send too much data for network to handle
  - different from flow control, which is e2e
- Congestion results in …
  - lost packets (buffer overflow at routers)
    - more work (retransmissions) for given "goodput"
    - waste of upstream links' capacity
      - Pkt traversed several links, then dropped at congested router
  - long delays (queuing in router buffers)
    - poor performance (less responsive app)
    - premature (unneeded) retransmissions

TCP congestion control: Approach

- **Approach: probe for usable bandwidth in network**
  - **increase transmission rate until loss occurs then decrease**
  - **Additive increase, multiplicative decrease (AIMD)**

Saw tooth behavior: probing for bandwidth



TCP Congestion Control

- Sender keeps new variable, Congestion Window (CongWin), and limits unacked bytes to:

```
LastByteSent - LastByteAcked ≤ min {CongWin, RcvWin}
```

  - For our discussion: assume RcvWin is large enough
  - Above equation achieves both flow and congestion control

- Roughly, what is the sending rate as a function of CongWin?
  - Ignore loss and transmission delay

- Rate = CongWin/RTT (bytes/sec)
  - So, rate and CongWin are somewhat synonymous

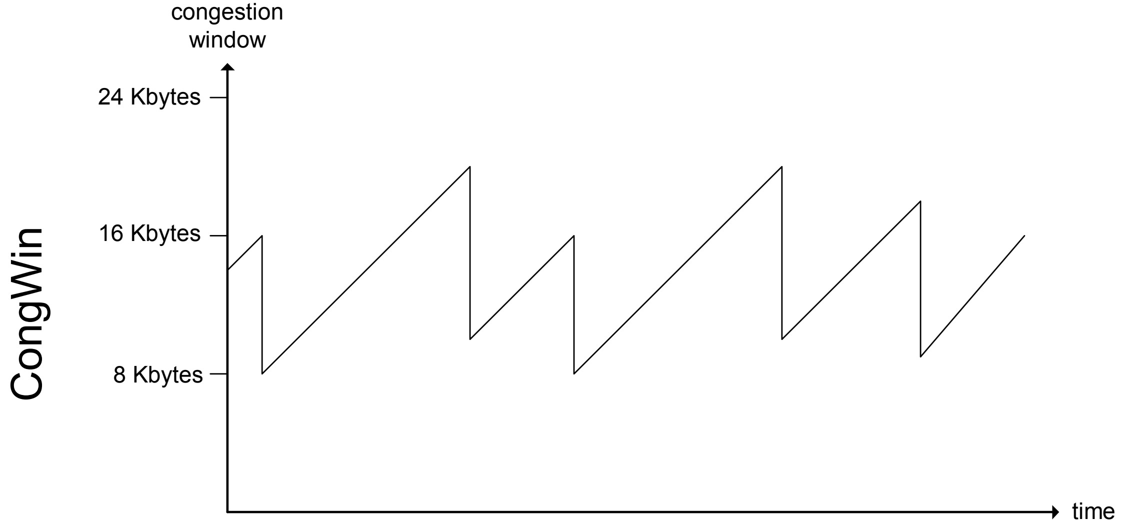
TCP Congestion Control

- Congestion occurs at routers (inside the network)
  - Routers do not provide any feedback to TCP
- How can TCP infer congestion?
  - From its symptoms: timeout or duplicate acks
  - Define loss event ≡ timeout or 3 duplicate acks
  - TCP decreases its CongWin (rate) after a loss event

- <u>TCP Congestion Control Algorithm: three components</u>
  - AIMD: additive increase, multiplicative decrease
  - slow start
  - Reaction to timeout events

AIMD

- additive increase: **(congestion avoidance phase)**
  - increase CongWin by 1 MSS every RTT until loss detected
  - TCP increases CongWin by: MSS x (MSS/CongWin) for every ACK received
  - Ex. MSS = 1,460 bytes and CongWin = 14,600 bytes
  - With every ACK, CongWin is increased by 146 bytes
- multiplicative decrease:
  - cut CongWin in half after loss



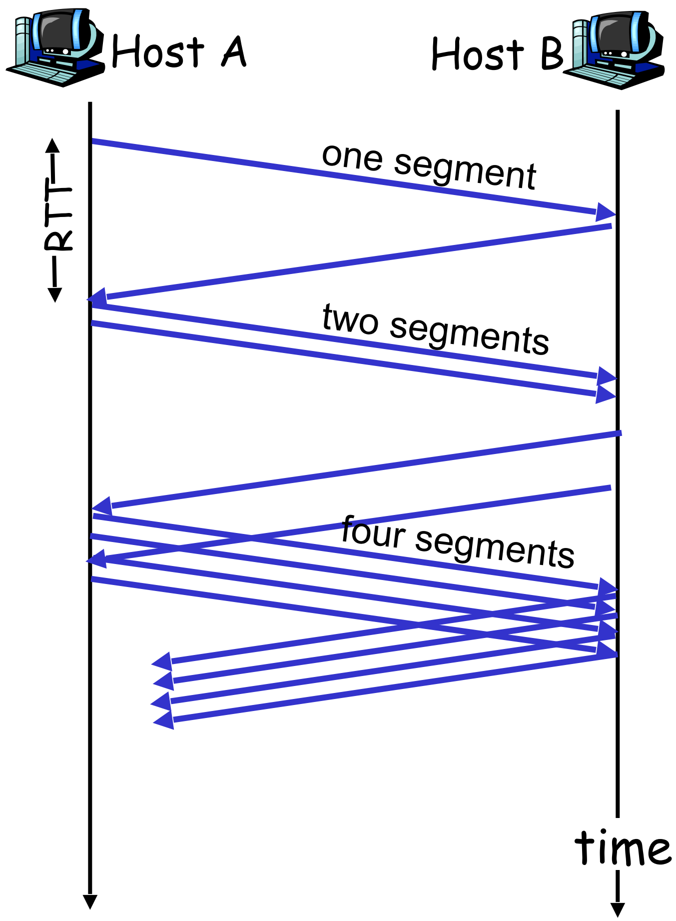
TCP Slow Start

- When connection begins, `CongWin` = 1 MSS
  - Example: MSS = 500 bytes & RTT = 200 msec
  - initial rate = CongWin/RTT = 20 kbps
- available bandwidth may be >> MSS/RTT
  - desirable to quickly ramp up to respectable rate
- Slow start:
  - When connection begins, increase rate exponentially fast until first loss event. How can we do that?
  - double `CongWin` every RTT. How?
  - Increment `CongWin` by 1 MSS for every ACK received

TCP Slow Start (cont'd)

- Increment CongWin by 1 MSS for every ACK

- Summary: initial rate is slow but ramps up exponentially fast



Reaction to a Loss event

- TCP Tahoe (Old)
  - Threshold = CongWin / 2
  - Set CongWin = 1 MSS
  - Slow start till threshold
  - Then Additive Increase // congestion avoidance
- TCP Reno (most current TCP implementations)
  - If 3 dup acks // fast retransmit
    - Threshold = CongWin / 2
    - Set CongWin = Threshold + 3 MSS // fast recovery
    - Additive Increase
  - Else // timeout
    - Same as TCP Tahoe

Reaction to a Loss event (cont'd)

Congestion
Window
(seg)

TCP Tahoe

TCP Reno

- Why differentiate between 3 dup acks and timeout?
- 3 dup ACKs indicate network capable of delivering some segments
- timeout indicates a "more alarming" congestion scenario

TCP Congestion Control: Summary

- Initially
  - `Threshold` is set to large value (65 Kbytes), has no effect
  - `CongWin` = 1 MSS
- Slow Start (SS): `CongWin` grows exponentially
  - till loss event occurs (timeout or 3 dup ACks) or reaches `Threshold`
- Congestion Avoidance (CA): `CongWin` grows linearly
- 3 duplicate ACK occurs:
  - `Threshold` = `CongWin/2`; `CongWin` = `Threshold +3 MSS;` CA
- Timeout occurs:
  - `Threshold` = `CongWin/2;` `CongWin` = 1 MSS; SS till `Threshold`

TCP Throughput Analysis

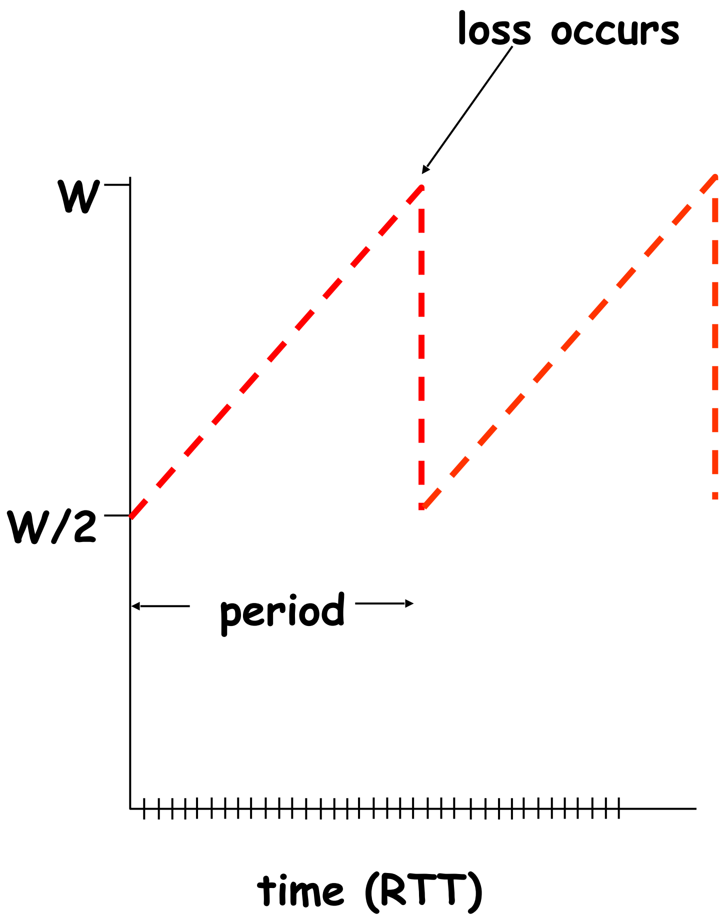
- Understand the fundamental relationship between
  - Packet loss probability,
  - RTT, and
  - TCP performance (throughput)

- We present simple model, with several assumptions
  - Yet it provides quite useful insights

TCP Throughput Analysis

- Any TCP model must capture
  - **Window Dynamics** (internal and deterministic)
    - Controlled internally by the TCP algorithm
    - Depends on the particular flavor of TCP
    - We assume TCP Reno (the most common)
  - **Packet Loss Process** (external and uncertain)
    - Models aggregate network conditions across all nodes on the TCP connection path
    - Typically modeled as Stochastic Process with probability p that a packet loss occurs
    - TCP responds by reducing the window size
- We usually analyze the steady state
  - Ignore the slow start phase (transient)
  - Although many connections finish within slow start, because they send only a few kilobytes

Simple (Periodic) Model



- Packet losses occur with constant probability p
- TCP window starts at W/2 grows to W, then halves, repeat forever … ➔
- TCP Throughput ranges between:
  - Min: MSS *(W/2) / RTT, and
  - Max: MSS* W / RTT ➔

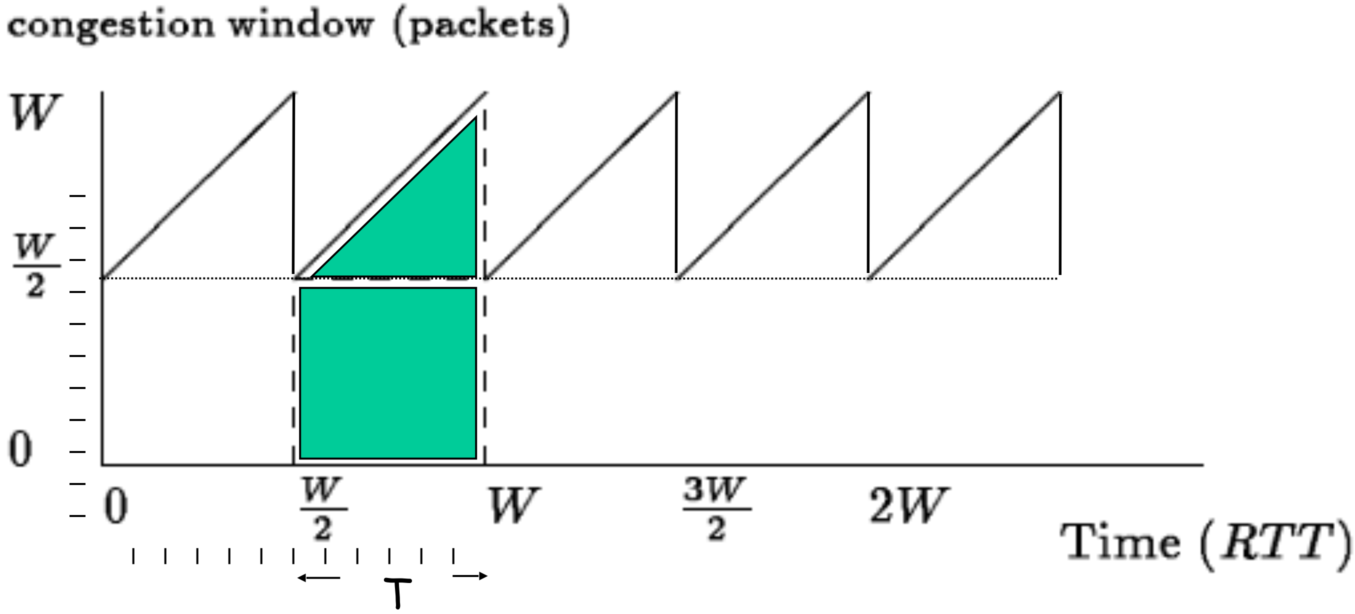
Avg throughput = MSS * (3/4 W) / RTT (1)

Simple (Periodic) Model

Avg throughput = MSS * (3/4 W) / RTT (1)

- Now, we want to relate W (max window size) with the packet loss rate in the network p
  - So that we have the TCP throughput as function of RTT and loss rate (both are external network parameters)

Simple (Periodic) Model



Throughput X(t) = green area (packets sent) / T

$$X(t) = \frac{\frac{W}{2} \times \frac{W}{2} + \frac{1}{2} \times \frac{W}{2} \times \frac{W}{2}}{T} = \frac{\frac{3}{8} W^2}{T} \qquad (2)$$

Simple (Periodic) Model

- On the other hand, we have average packet loss rate of p
  - How many packets we send until we observe a loss?
  - 1/p
- Throughput X(t) = number of packets sent / T ➔

$$X(t) = \frac{1/p}{T} \qquad (3)$$

- Solve (2) and (3) ➔

$$W = \sqrt{\frac{8}{3p}}$$

Simple (Periodic) Model

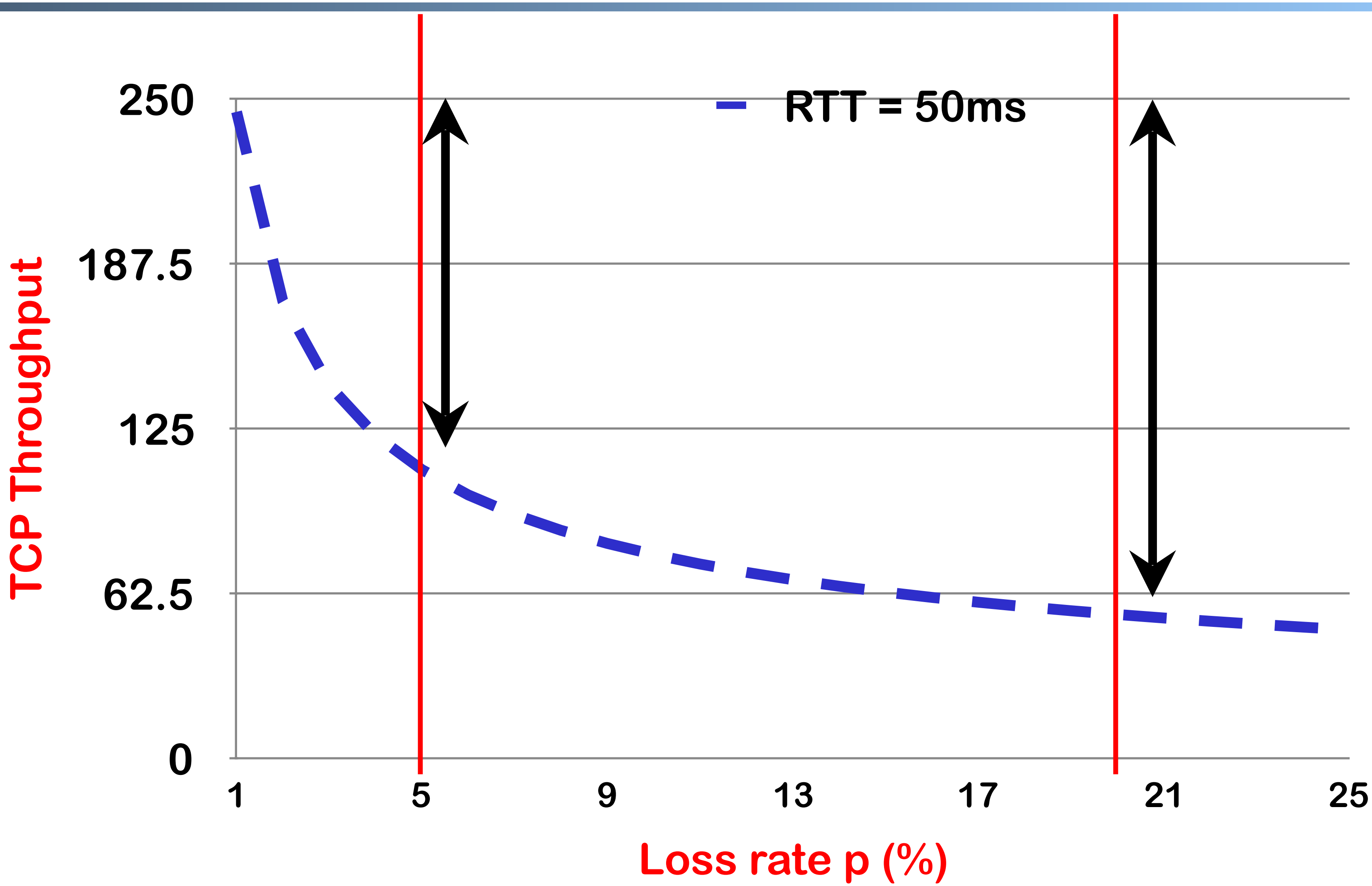
- Substitute w in (1):

$$X(p) = \frac{MSS}{RTT}\sqrt{\frac{3}{2p}}$$

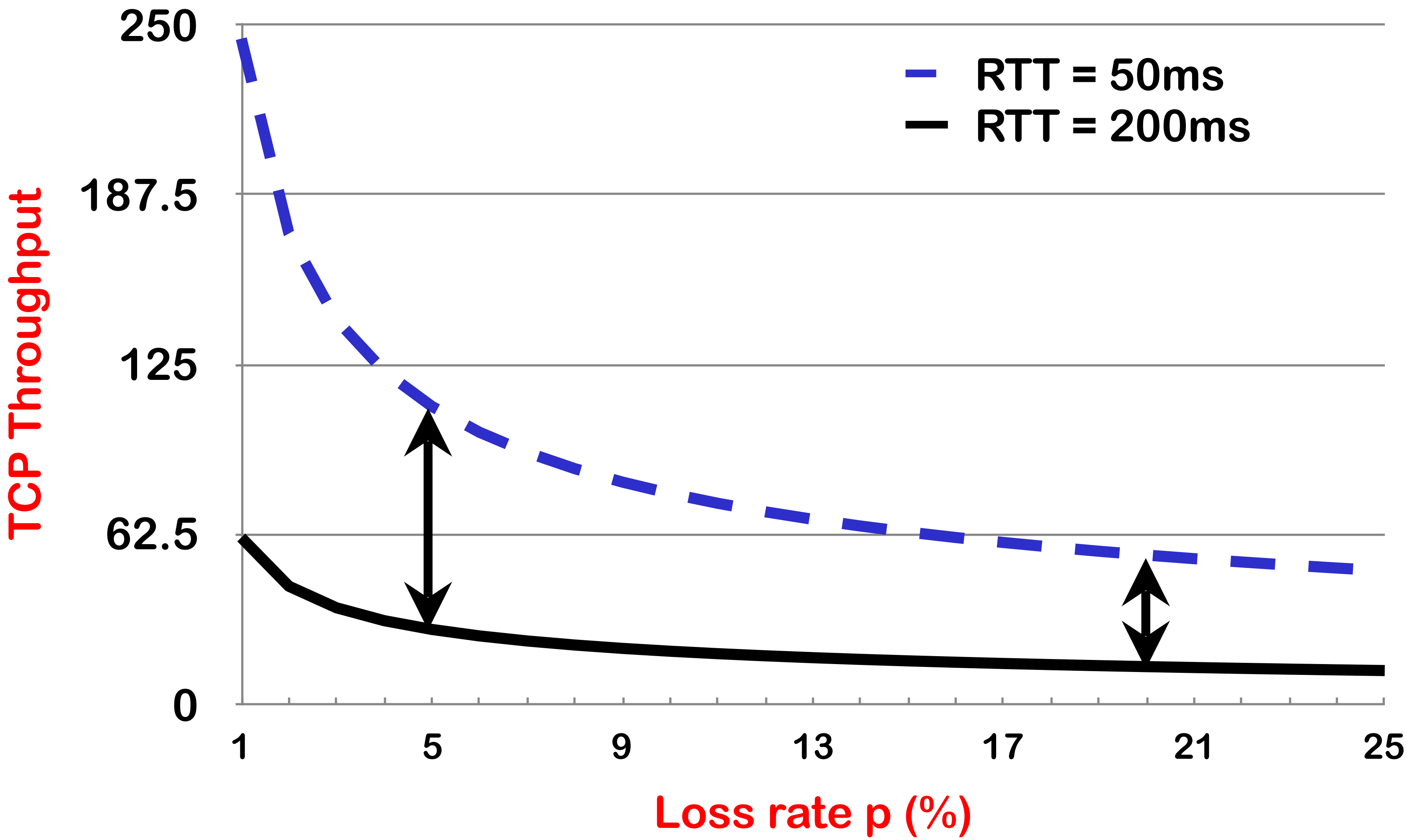
- Called inverse square-root-p law

- TCP throughput is inversely proportional to the square root of the packet loss probability

Impact of Loss Rate on TCP

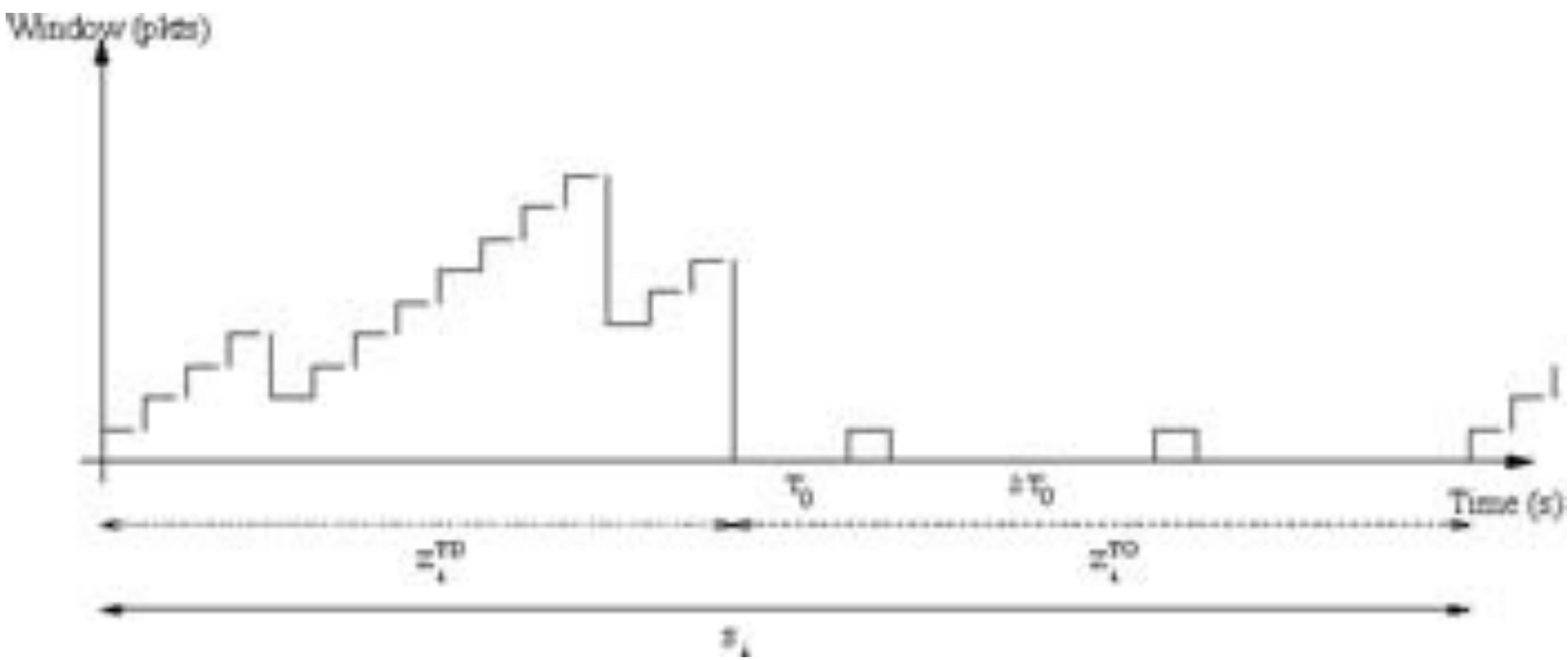


Impact of RTT & loss on TCP



In More Realistic Models …

- Packet loss probability is not constant and is bursty
- Consider effect of duplicate ACKs and Timeouts
- Consider receiver window limit



TCP Over High Speed Links

- Assume 1 Gbps link, RTT 100ms, what is the max packet loss rate for TCP to achieve 1 Gbps throughput? Assume MSS = 1000 bytes

$$X(p) = \frac{MSS}{RTT}\sqrt{\frac{3}{2p}}$$

$$10^9 = \frac{1000 * 8}{100 * 10^{-3}}\sqrt{\frac{3}{2p}}$$

- p = ~ 10^(-8)
  - At most one segment lost each 100 millions sent. That is way too low even for fiber optic links

High Speed TCP & other Flavors

- To support high bandwidth-delay product links
  - (detected when congWin gets very large)
  - Idea: Increase congWin by larger amount than standard TCP
  - E.g., in CUBIC (implemented in Linux), FAST TCP,…
- More flavors at:
  - https://en.wikipedia.org/wiki/TCP_congestion_control

TCP over Wireless Networks

- Performance of TCP suffers:
  - First recall that TCP interprets loss as congestion
  - But in wireless networks, packets can be lost because of bit-errors (usually high) and handoff (long delays)
  - Thus, TCP may un-necessarily decrease its sending rate (congestion window)
- Solutions?

Enhancing TCP Performance in Wireless Networks

- **Make TCP aware of the wireless link**
  - Usually uses probabilistic inference models
  - Distinguish loss due to congestion from others (e.g., wireless bit errors)
  - Decrease sending rate if congestion only

Enhancing TCP Performance in Wireless Networks

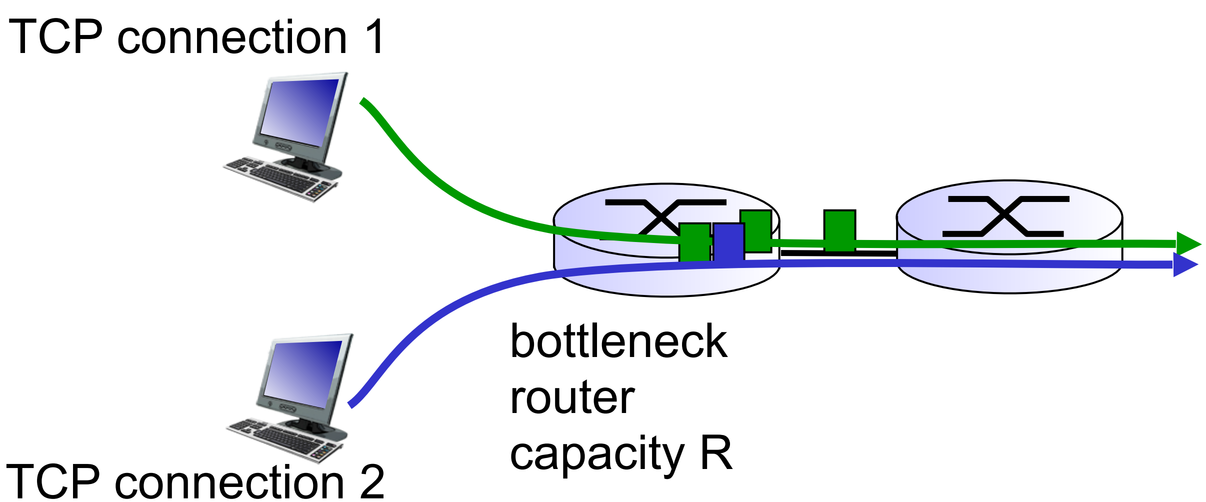
- **Split TCP connections**
  - One from mobile to base station, and another from base station to the other end
  - Over the wireless part, we can use either standard TCP (shorter delay now) or custom/new transport protocols, e.g. reliable UDP, TCP with selective repeat
  - Used in cellular networks, significant improvement in TCP

Enhancing TCP Performance in Wireless Networks

- **Local (link) Recovery**
  - ARQ: local retransmission (e.g., in 802.11)
  - FEC: for long-delay networks (e.g., cellular, satellite)
    - **Adds R redundant packets to N original packet such that any received N packets out of N+R packets can be used to recover the whole data.**

TCP Fairness

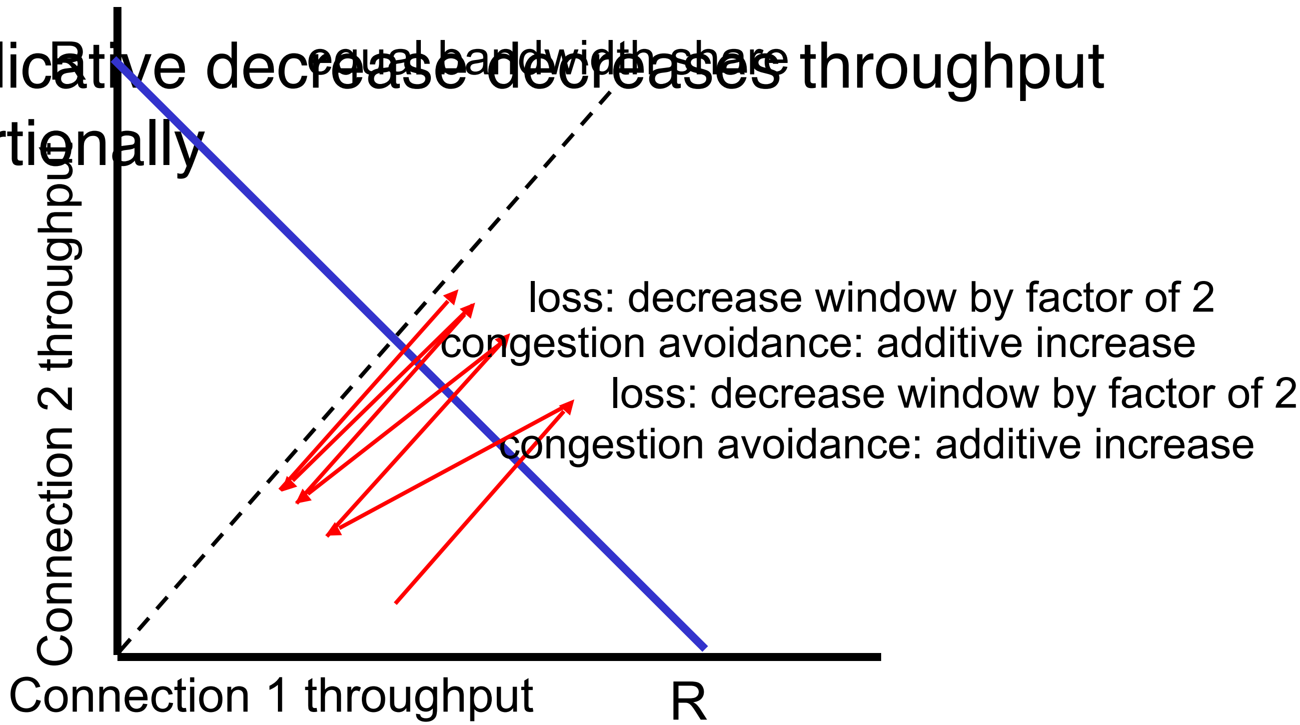
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Why is TCP fair?

two competing sessions:

- additive increase gives slope of 1, as 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

Network-Assisted Congestion Control

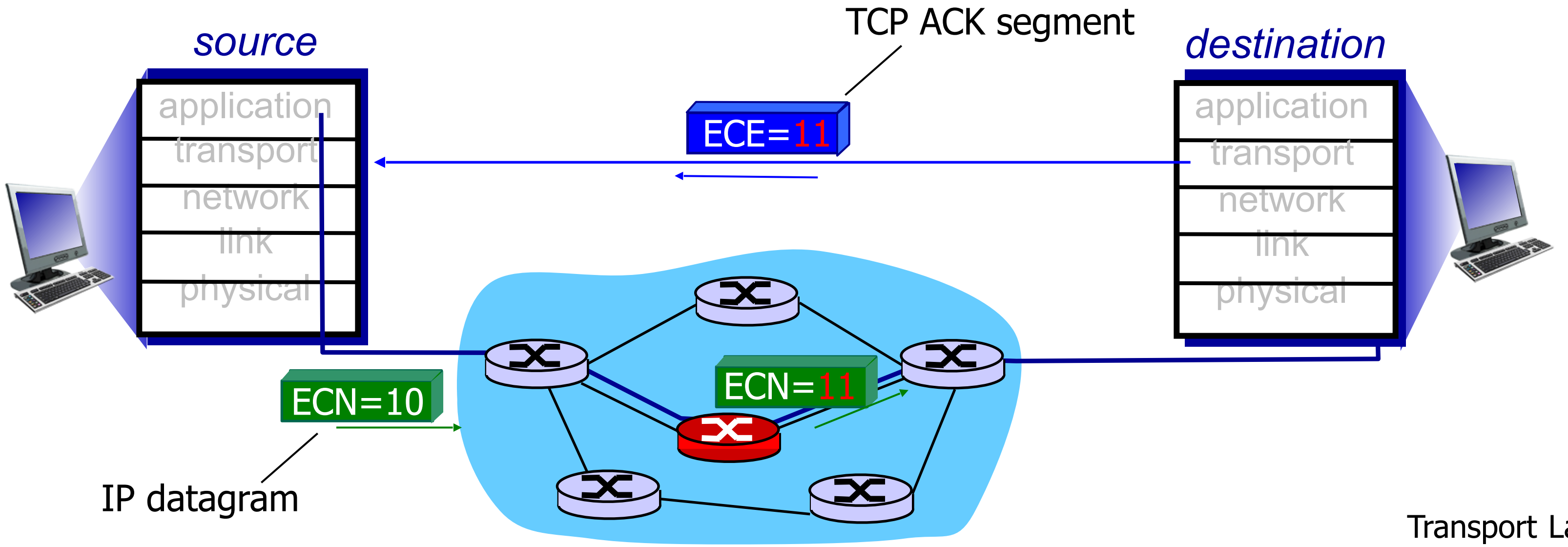
- Network (routers) help sources in detecting congestion

- Example: Explicit Congestion Notification (ECN)
  - Uses 2 bits in the IP header (ToS filed)
  - 1 bit to indicate to routers that source can use their help
  - The other bit is set by any router on the path that observes congestion (e.g., large queue length)

Explicit Congestion Notification ~~(ECN)~~

network-assisted congestion control:

- two bits in IP header (ToS field) marked *by network router* to indicate congestion
- congestion indication carried to receiving host
- receiver (seeing congestion indication in IP datagram)) sets ECN bit on receiver-to-sender ACK segment to notify sender of congestion



Summary

- Transport layer: logical channel between processes
- Main protocols: TCP and UDP
- TCP:
  - Reliable
  - Congestion control
  - Flow control
- Simple analytic model for TCP
  - Throughput inversely proportional with RTT and greatly affected by packet loss rate
  - TCP performance over wireless networks may suffer: performance mitigation