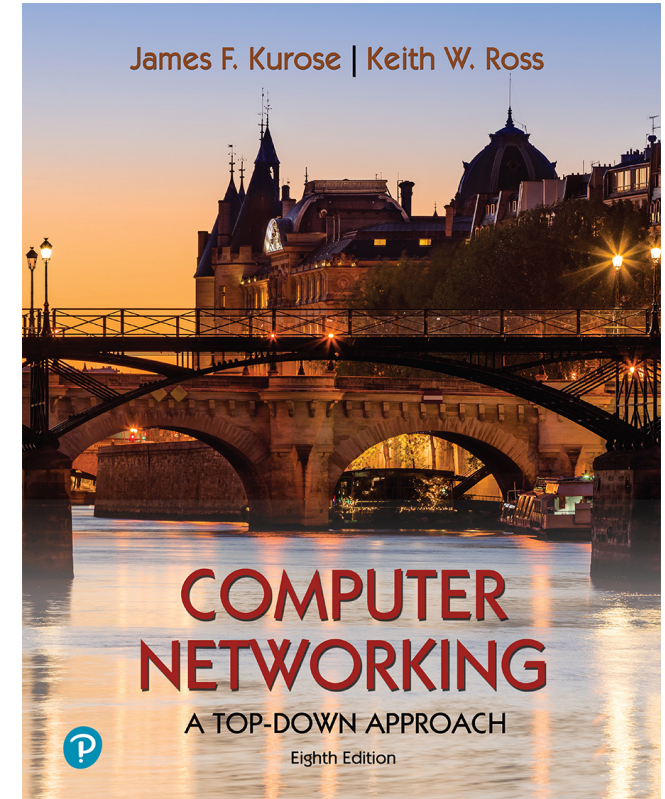


Chapter 3

Transport Layer



Computer Networking: A Top-Down Approach

8th edition

Jim Kurose, Keith Ross
Pearson, 2020

Acknowledgement: Based on the textbook's website:
https://gaia.cs.umass.edu/kurose_ross/index.php

Chapter 3: roadmap

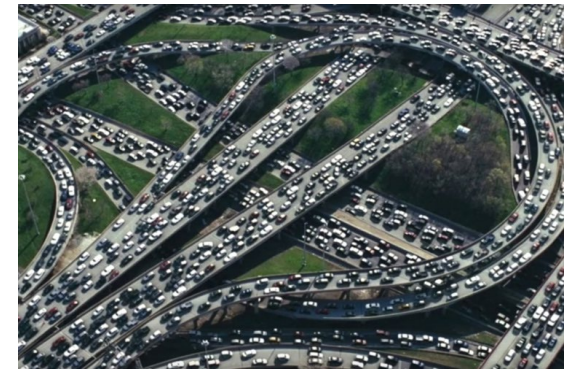
- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Connection-oriented transport: TCP
- **Principles of congestion control**
- TCP congestion control
- Evolution of transport-layer functionality



Principles of congestion control

Congestion:

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



congestion control:

too many senders,
sending too fast

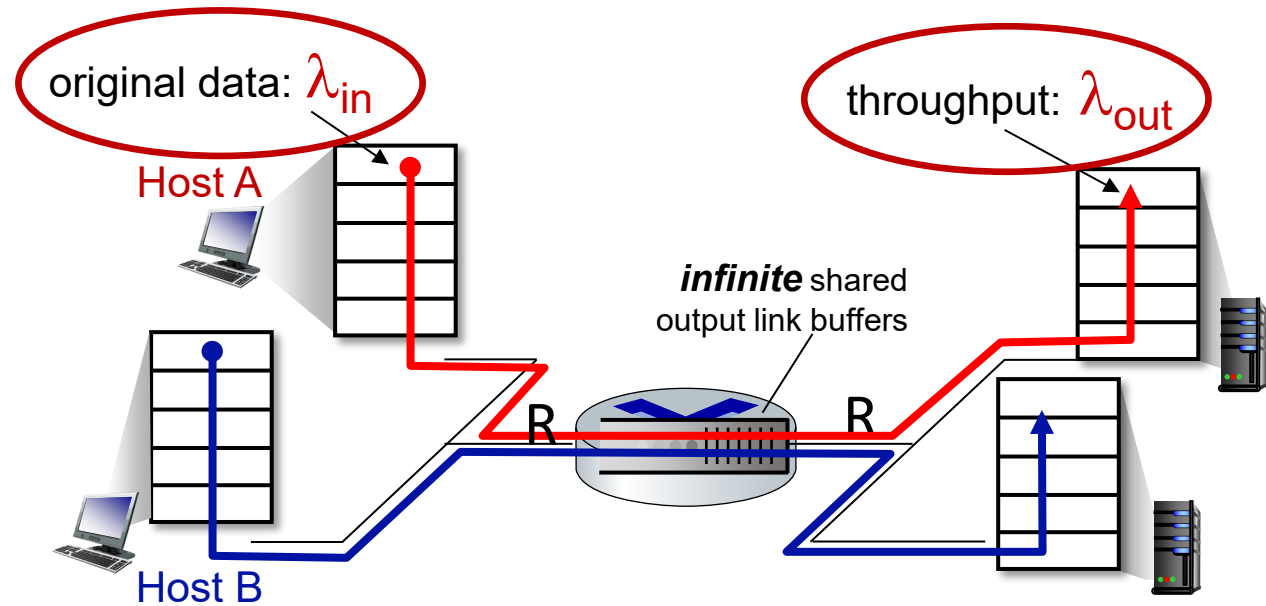


flow control: one sender
too fast for one receiver

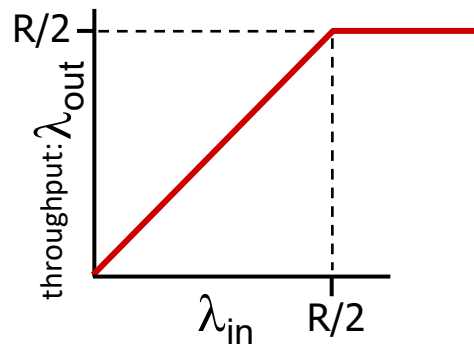
Causes/costs of congestion: scenario 1

Simplest scenario:

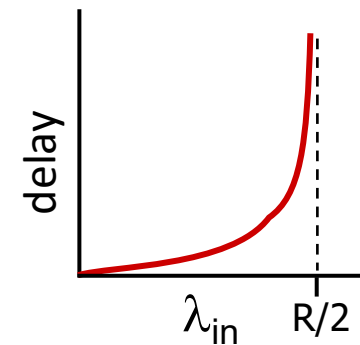
- one router, infinite buffers
- input, output link capacity: R
- two flows
- no retransmissions needed



Q: What happens as arrival rate λ_{in} approaches $R/2$?



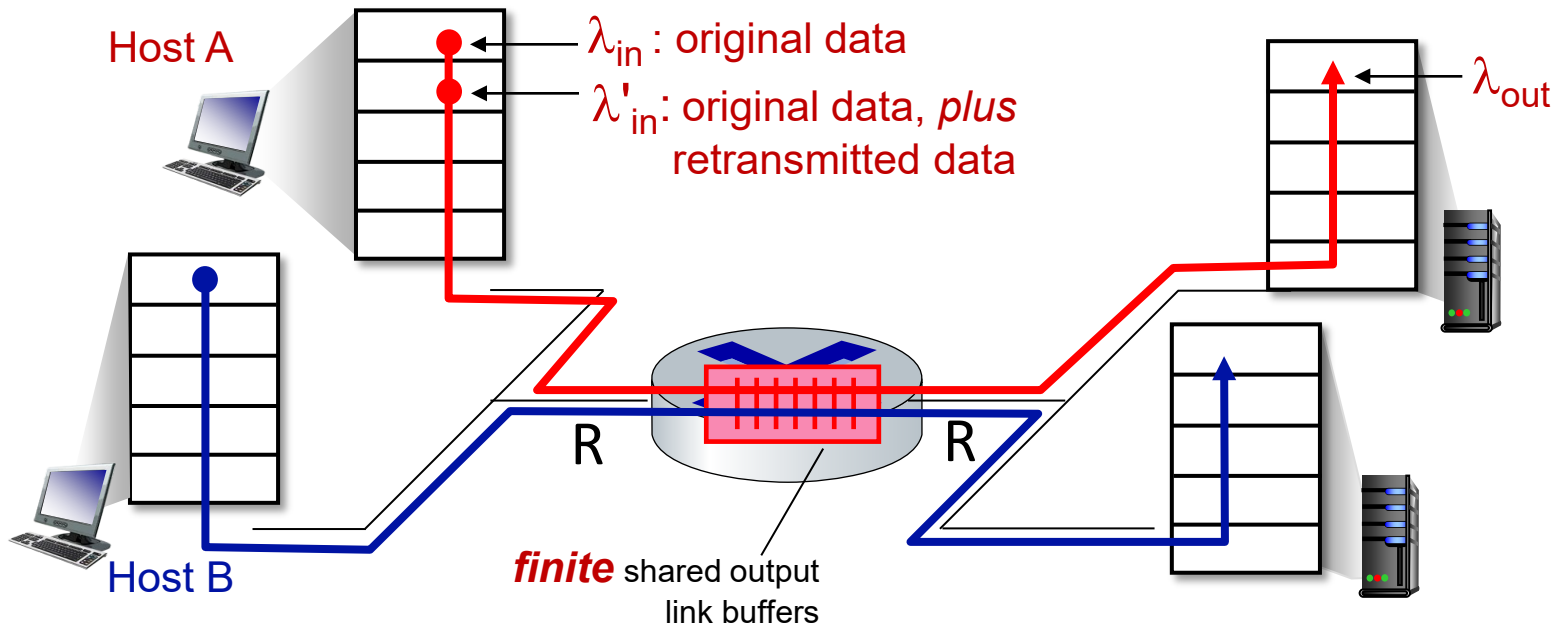
maximum per-connection throughput: $R/2$



large delays as arrival rate $\lambda_{in} \rightarrow R/2$

Causes/costs of congestion: scenario 2

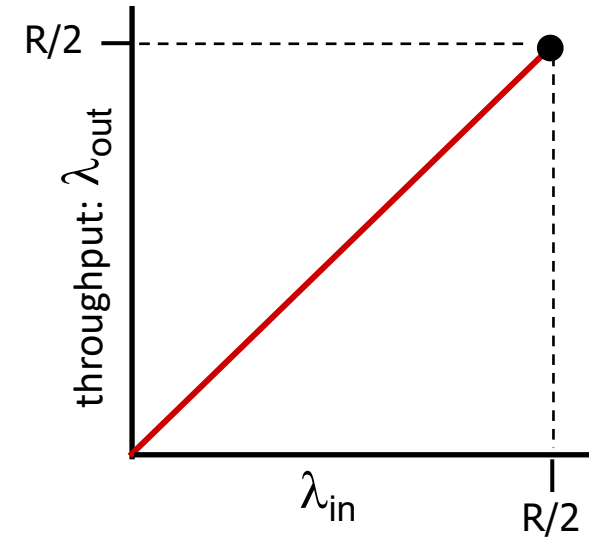
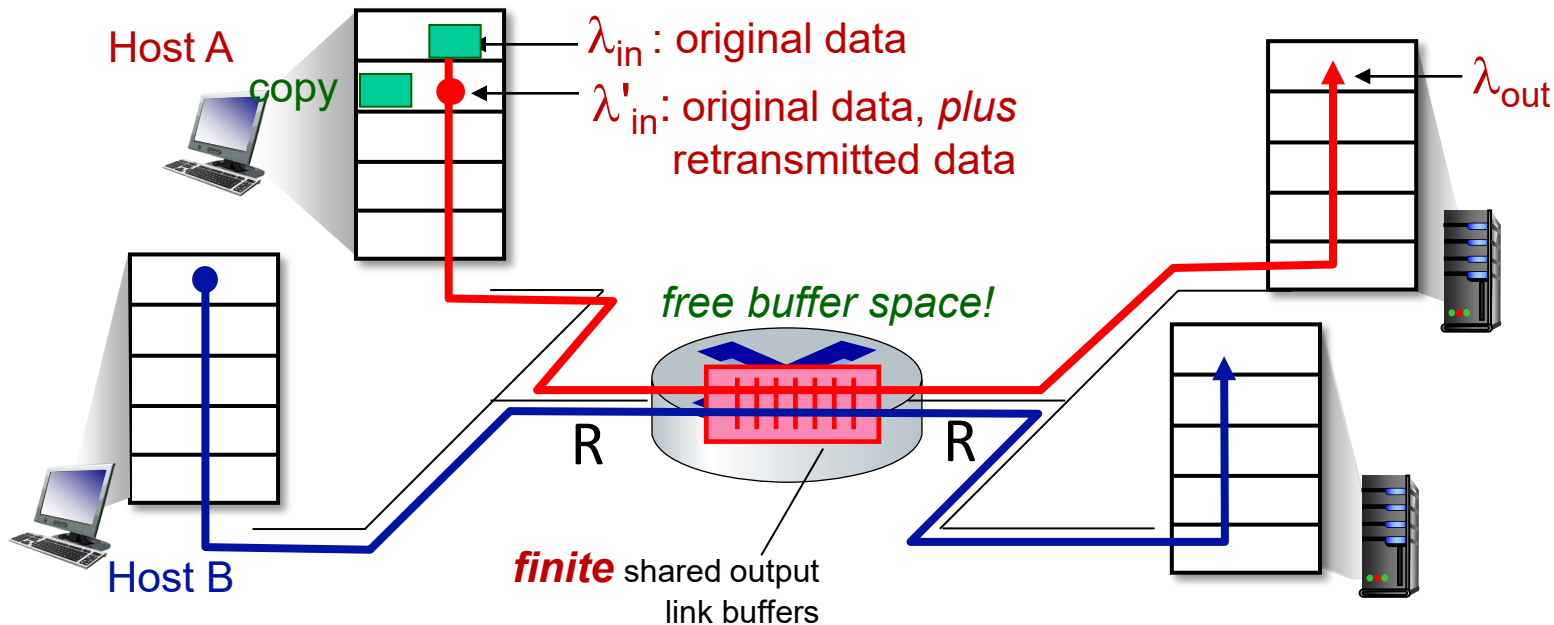
- one router, *finite* buffers
- sender retransmits lost, timed-out packet
 - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
 - transport-layer input includes *retransmissions* : $\lambda'_{in} \geq \lambda_{in}$



Causes/costs of congestion: scenario 2

Idealization: perfect knowledge

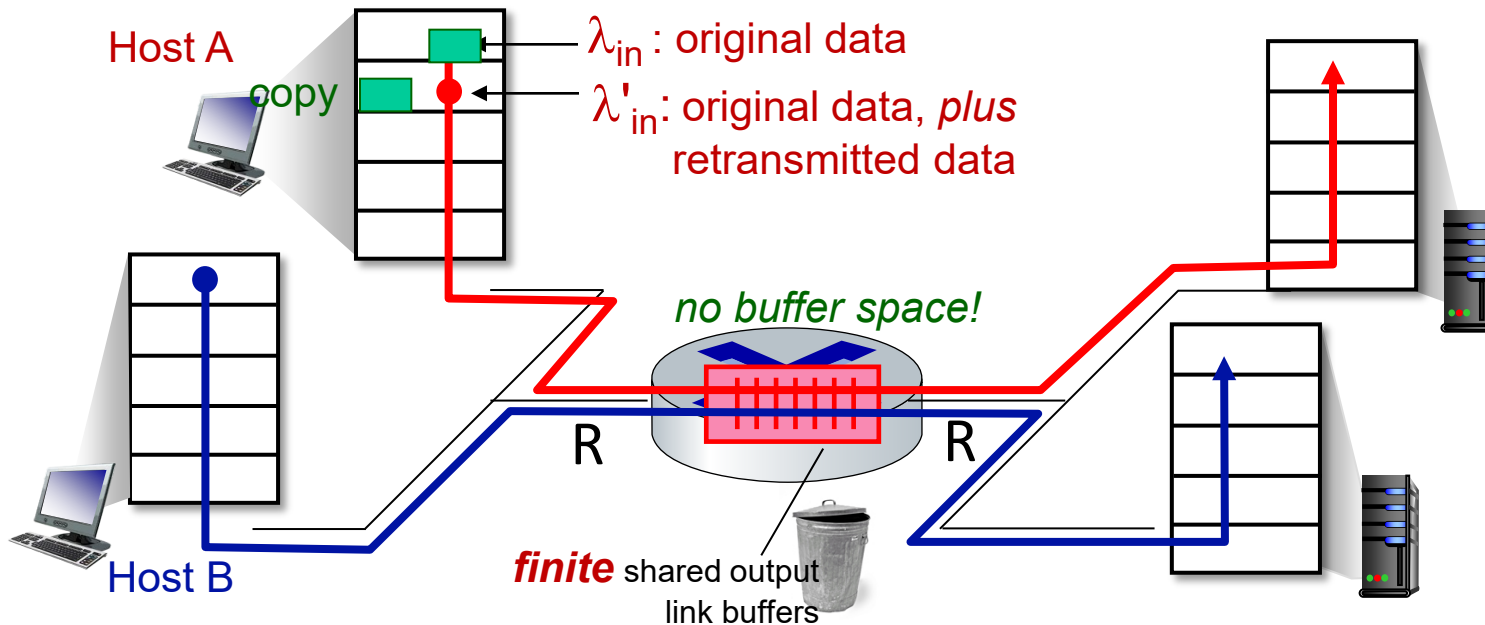
- sender sends only when router buffers available



Causes/costs of congestion: scenario 2

Idealization: *some* perfect knowledge

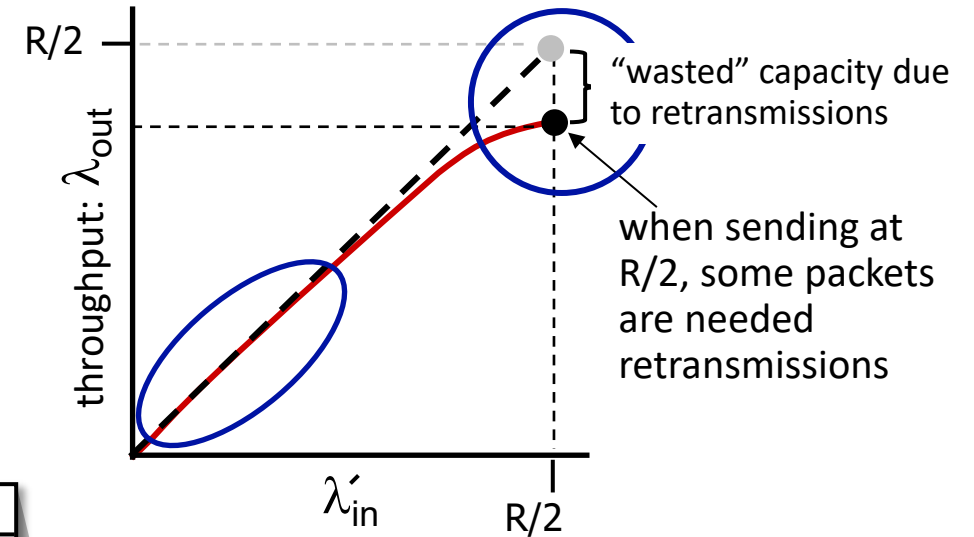
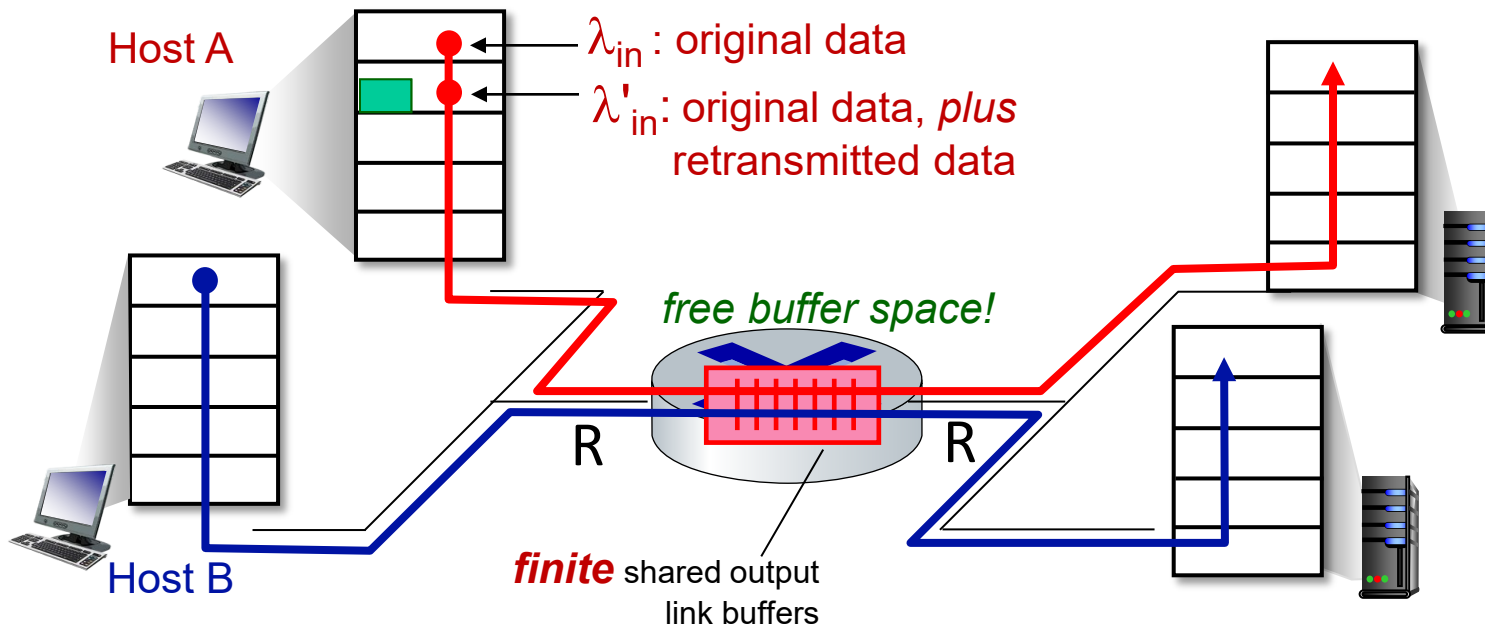
- packets can be lost (dropped at router) due to full buffers
- sender knows when packet has been dropped: only resends if packet *known* to be lost



Causes/costs of congestion: scenario 2

Idealization: *some* perfect knowledge

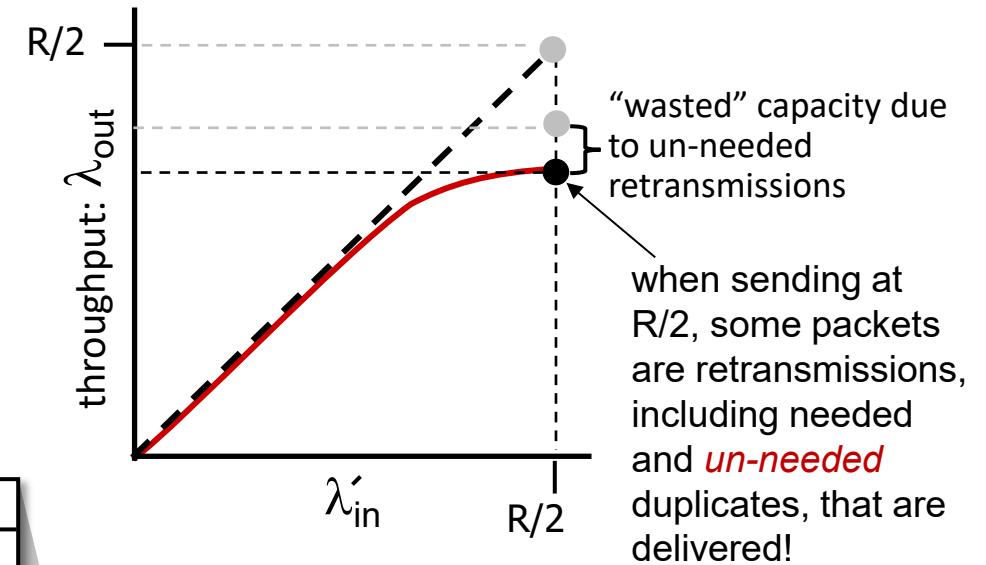
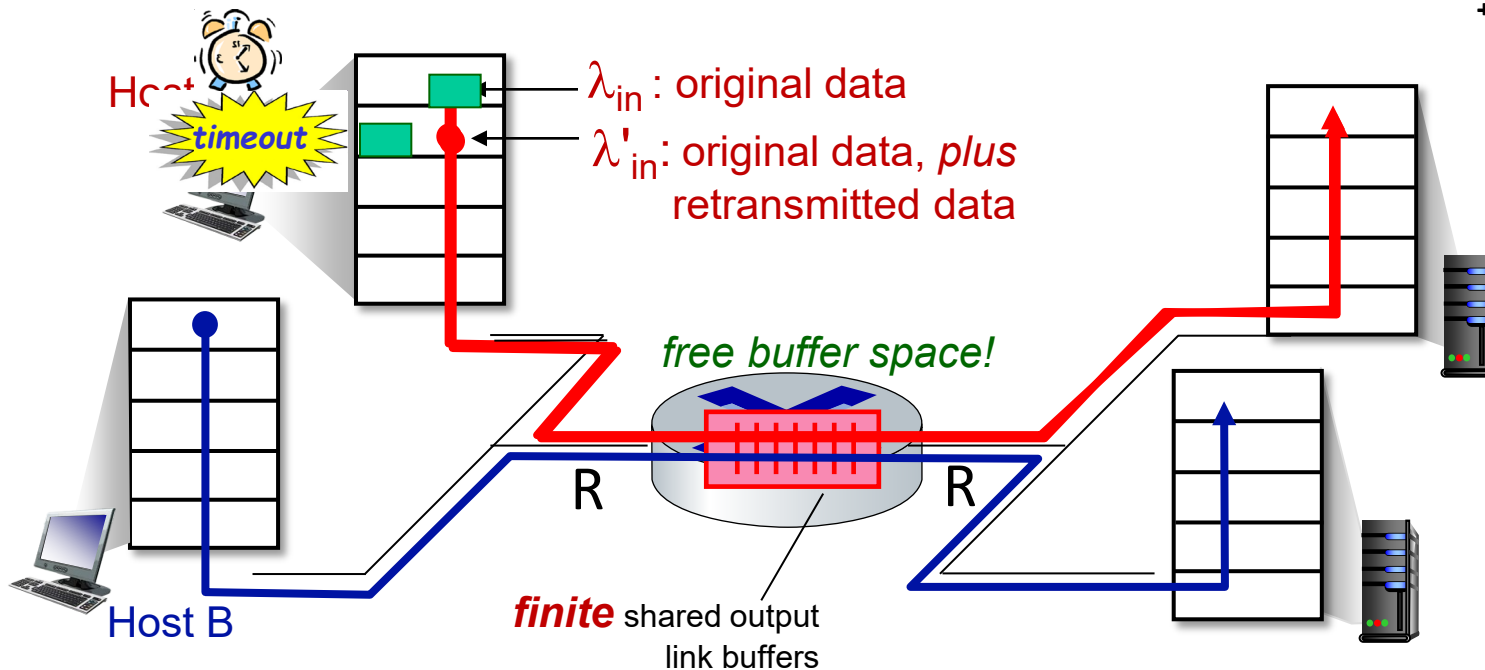
- packets can be lost (dropped at router) due to full buffers
- sender knows when packet has been dropped: only resends if packet *known* to be lost



Causes/costs of congestion: scenario 2

Realistic scenario: *un-needed duplicates*

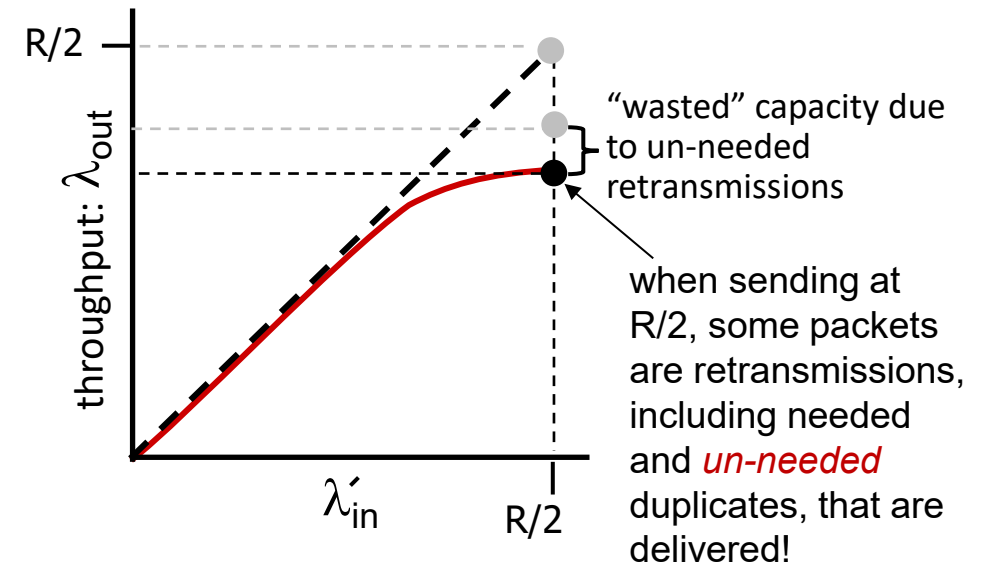
- packets can be lost, dropped at router due to full buffers – requiring retransmissions
- but sender times can time out prematurely, sending *two* copies, *both* of which are delivered



Causes/costs of congestion: scenario 2

Realistic scenario: *un-needed duplicates*

- packets can be lost, dropped at router due to full buffers – requiring retransmissions
- but sender times can time out prematurely, sending *two* copies, *both* of which are delivered



"costs" of congestion:

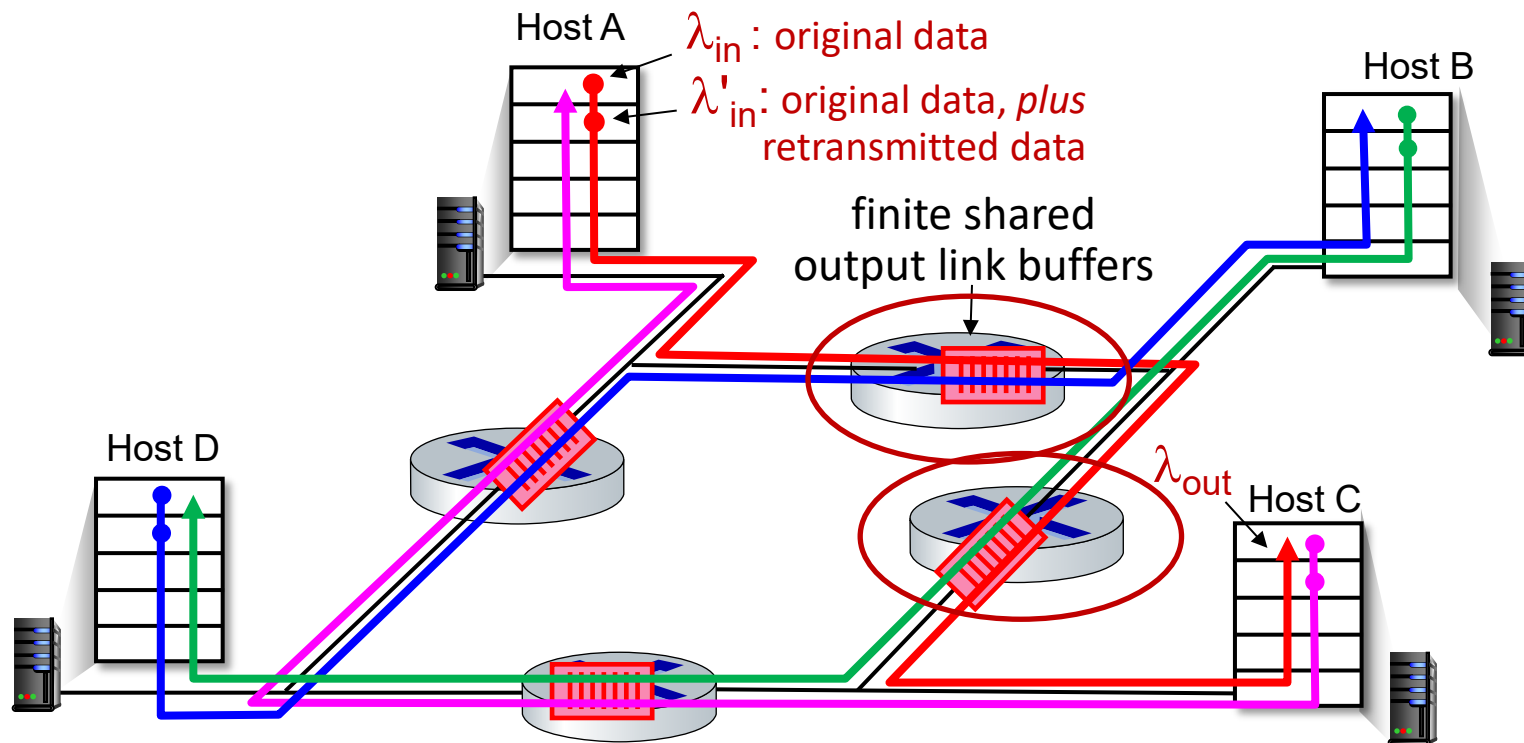
- more work (retransmission) for given receiver throughput
- unneeded retransmissions: link carries multiple copies of a packet
 - decreasing maximum achievable throughput

Causes/costs of congestion: scenario 3

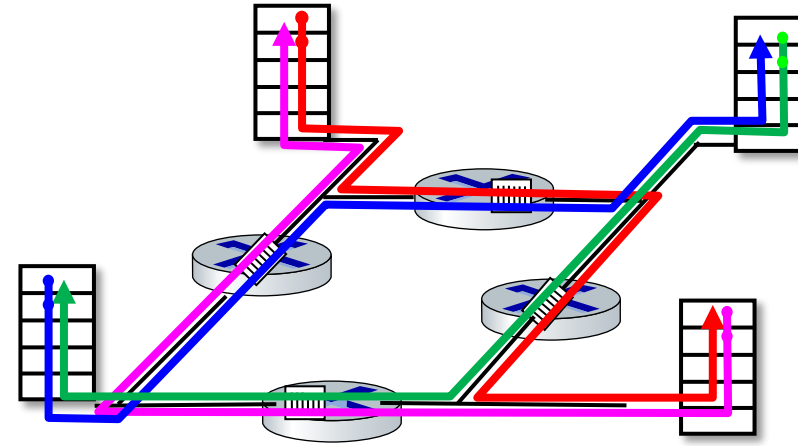
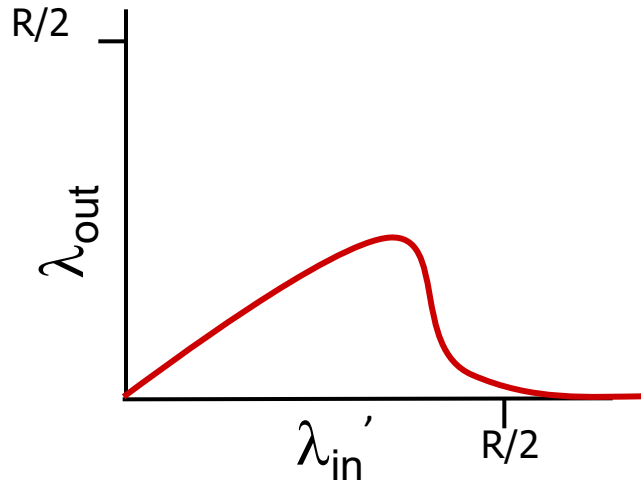
- *four* senders
- *multi-hop* paths
- timeout/retransmit

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

A: as red λ'_{in} increases, all arriving blue pkts at upper queue are dropped, blue throughput $\rightarrow 0$



Causes/costs of congestion: scenario 3

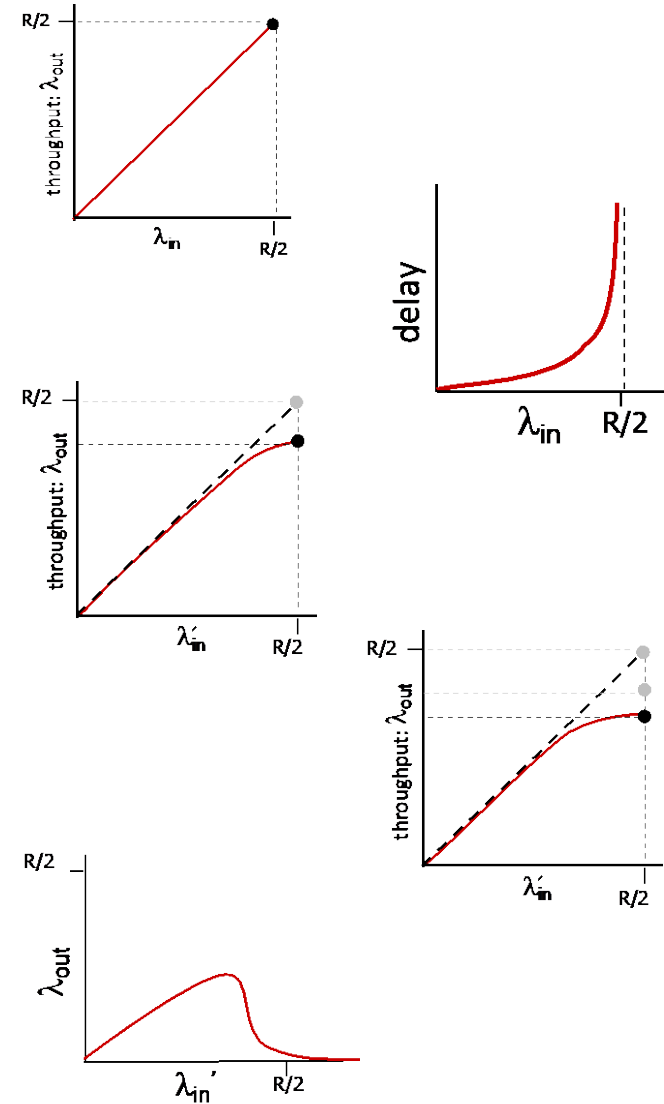


another “cost” of congestion:

- when packet dropped, any upstream transmission capacity and buffering used for that packet was wasted!

Causes/costs of congestion: insights

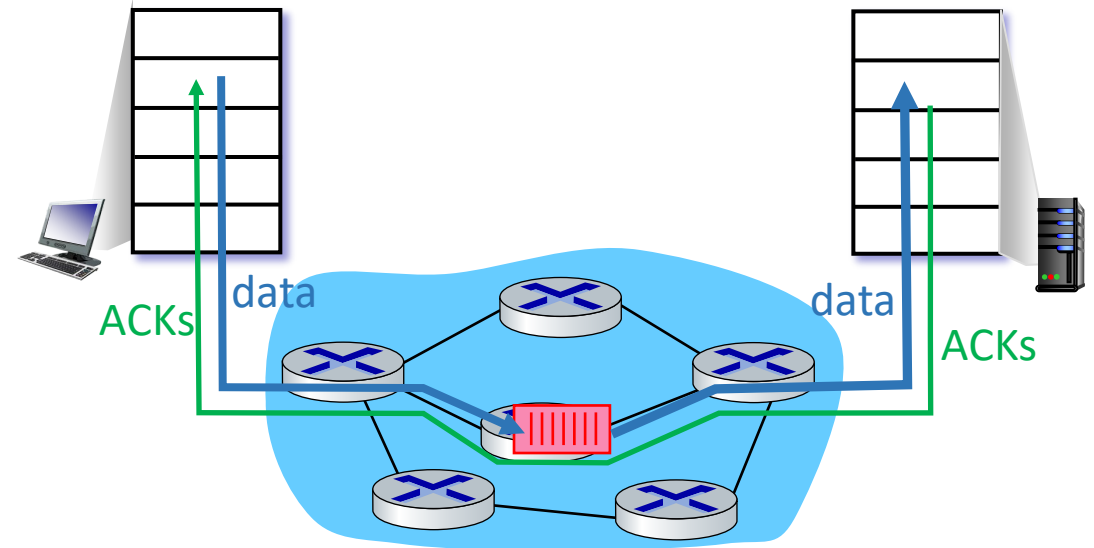
- throughput can never exceed capacity
- delay increases as capacity approached
- loss/retransmission decreases effective throughput
- un-needed duplicates further decreases effective throughput
- upstream transmission capacity / buffering wasted for packets lost downstream



Approaches towards congestion control

End-end congestion control:

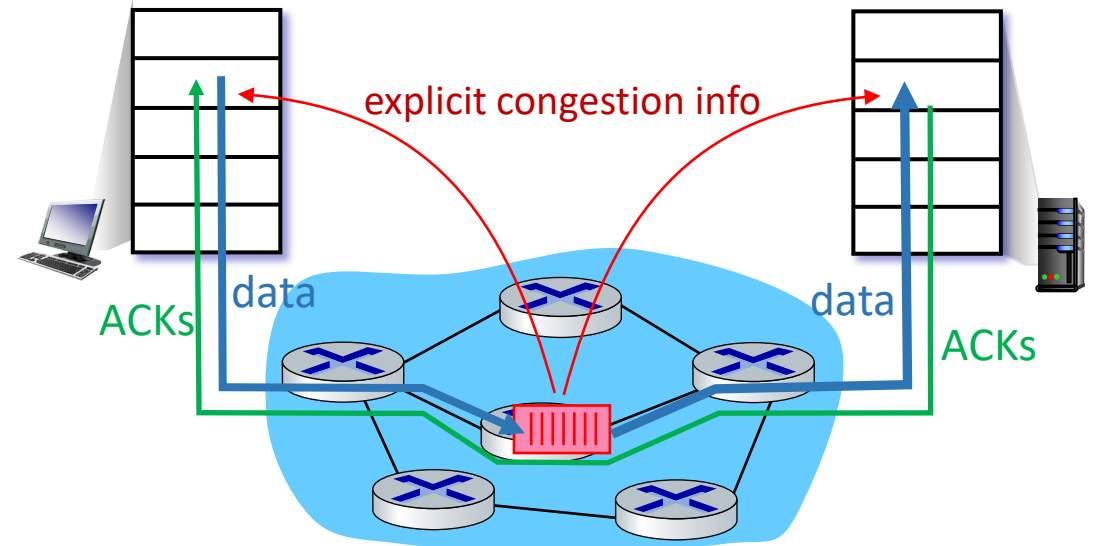
- no explicit feedback from network
- congestion *inferred* from observed loss, delay
- approach taken by TCP



Approaches towards congestion control

Network-assisted congestion control:

- routers provide *direct* feedback to sending/receiving hosts with flows passing through congested router
- may indicate congestion level or explicitly set sending rate
- TCP ECN, ATM, DECbit protocols



Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
- Principles of congestion control
- **TCP congestion control**
- Evolution of transport-layer functionality



TCP congestion control: AIMD

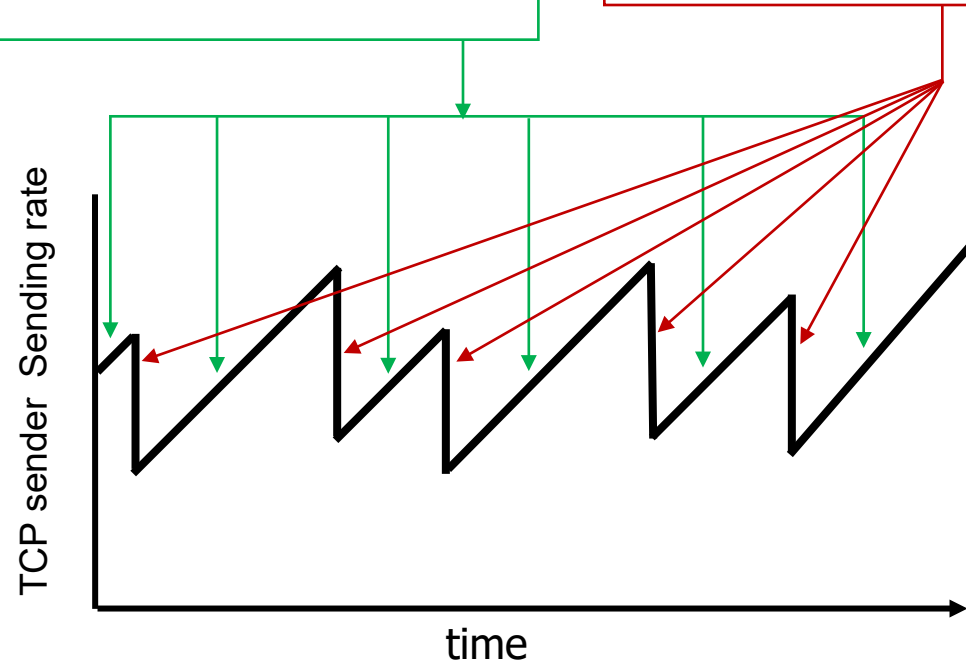
- *approach*: senders can increase sending rate until packet loss (congestion) occurs, then decrease sending rate on loss event

Additive Increase

increase sending rate by 1 maximum segment size every RTT until loss detected

Multiplicative Decrease

cut sending rate in half at each loss event



AIMD sawtooth behavior: *probing* for bandwidth

TCP AIMD: more

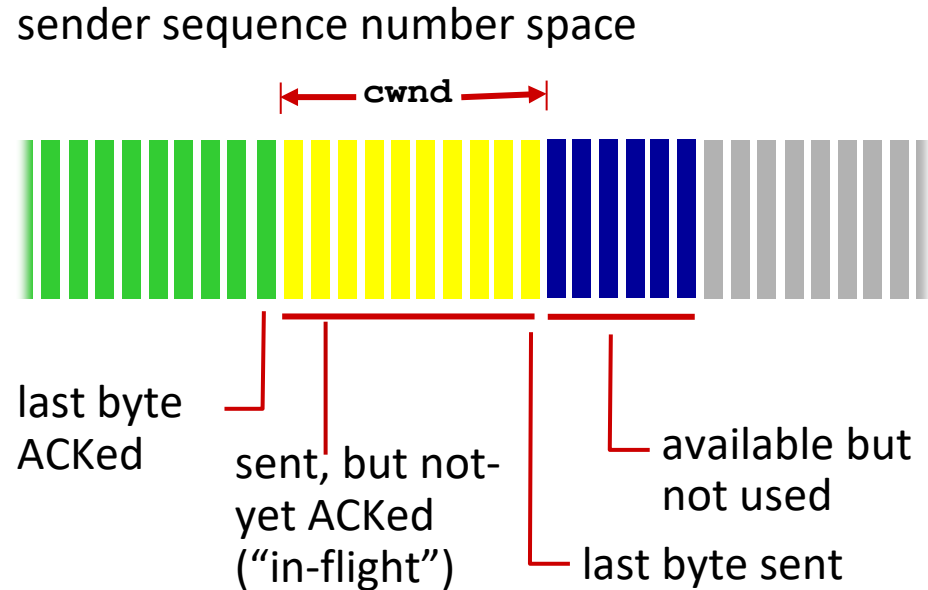
Multiplicative decrease detail: sending rate is

- Cut in half on loss detected by triple duplicate ACK (TCP Reno)
- Cut to 1 MSS (maximum segment size) when loss detected by timeout (TCP Tahoe)

Why AIMD?

- AIMD – a distributed, asynchronous algorithm – has been shown to:
 - optimize congested flow rates network wide!
 - have desirable stability properties

TCP congestion control: details



TCP sending behavior:

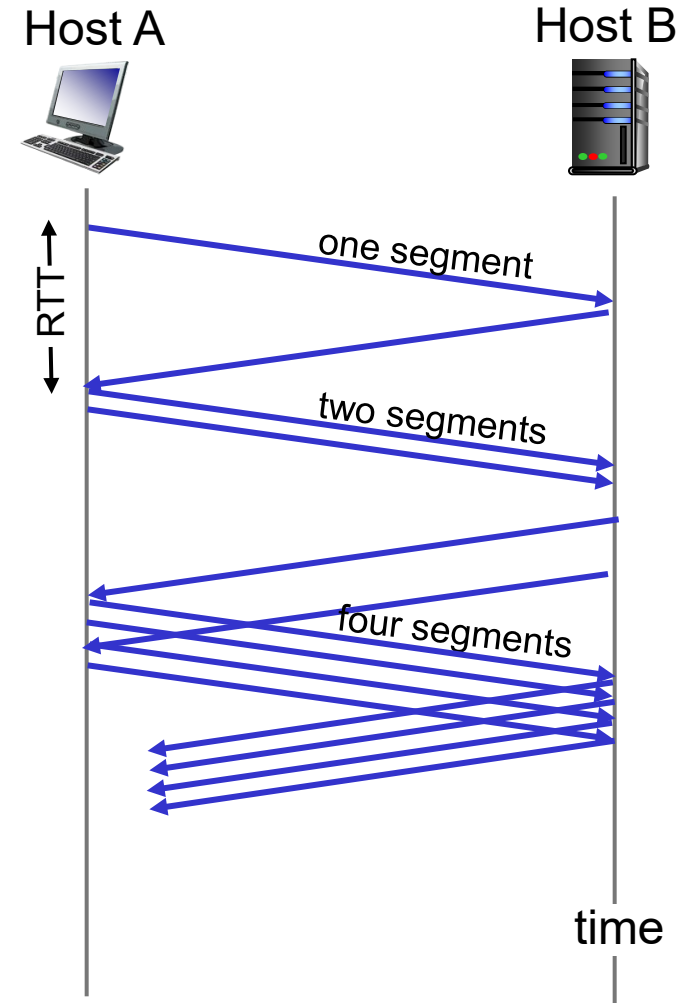
- *roughly*: send `cwnd` bytes, wait RTT for ACKS, then send more bytes

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

- TCP sender limits transmission: $\text{LastByteSent} - \text{LastByteAcked} \leq \text{cwnd}$
- `cwnd` is dynamically adjusted in response to observed network congestion (implementing TCP congestion control)

TCP slow start

- when connection begins, increase rate exponentially until first loss event:
 - initially **cwnd** = 1 MSS
 - double **cwnd** every RTT
 - done by incrementing **cwnd** for every ACK received
- *summary*: initial rate is slow, but ramps up exponentially fast



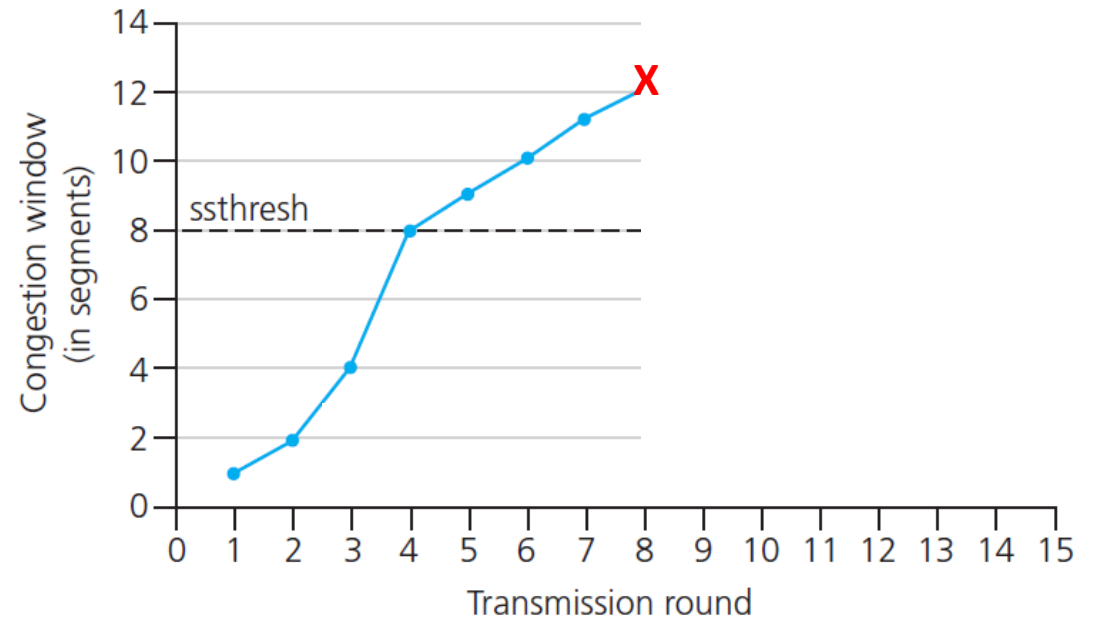
TCP: from slow start to congestion avoidance

Q: when should the exponential increase switch to linear?

A: when **cwnd** gets to 1/2 of its value before timeout.

Implementation:

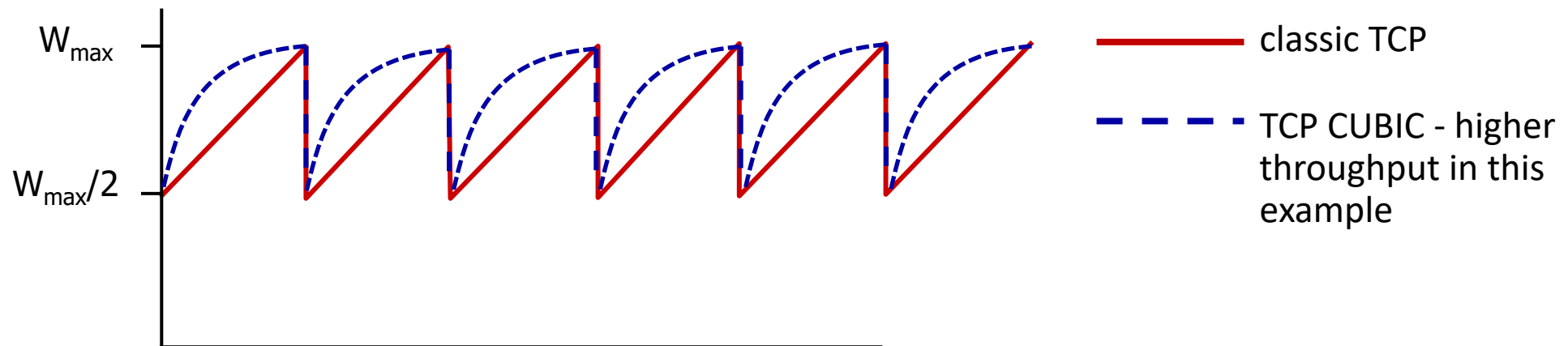
- variable **ssthresh**
- on loss event, **ssthresh** is set to 1/2 of **cwnd** just before loss event



* Check out the online interactive exercises for more examples: http://gaia.cs.umass.edu/kurose_ross/interactive/

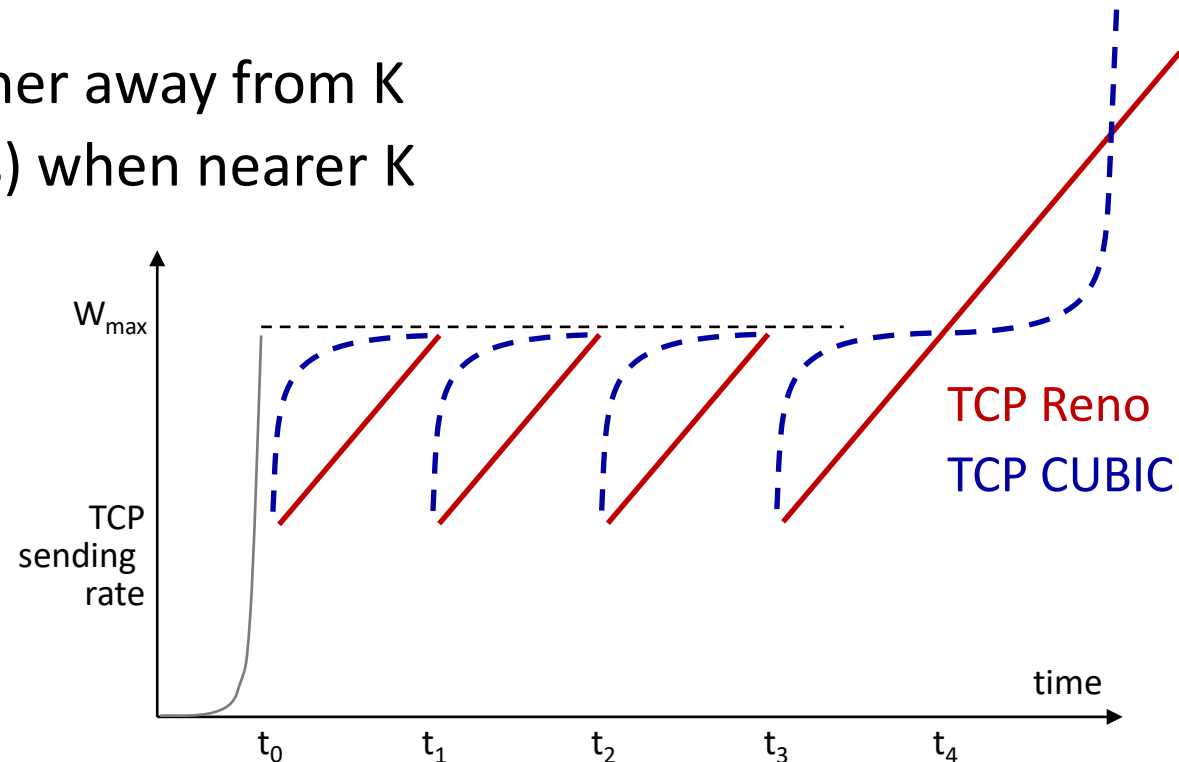
TCP CUBIC

- Is there a better way than AIMD to “probe” for usable bandwidth?
- Insight/intuition:
 - W_{\max} : sending rate at which congestion loss was detected
 - congestion state of bottleneck link probably (?) hasn't changed much
 - after cutting rate/window in half on loss, initially ramp to to W_{\max} *faster*, but then approach W_{\max} more *slowly*



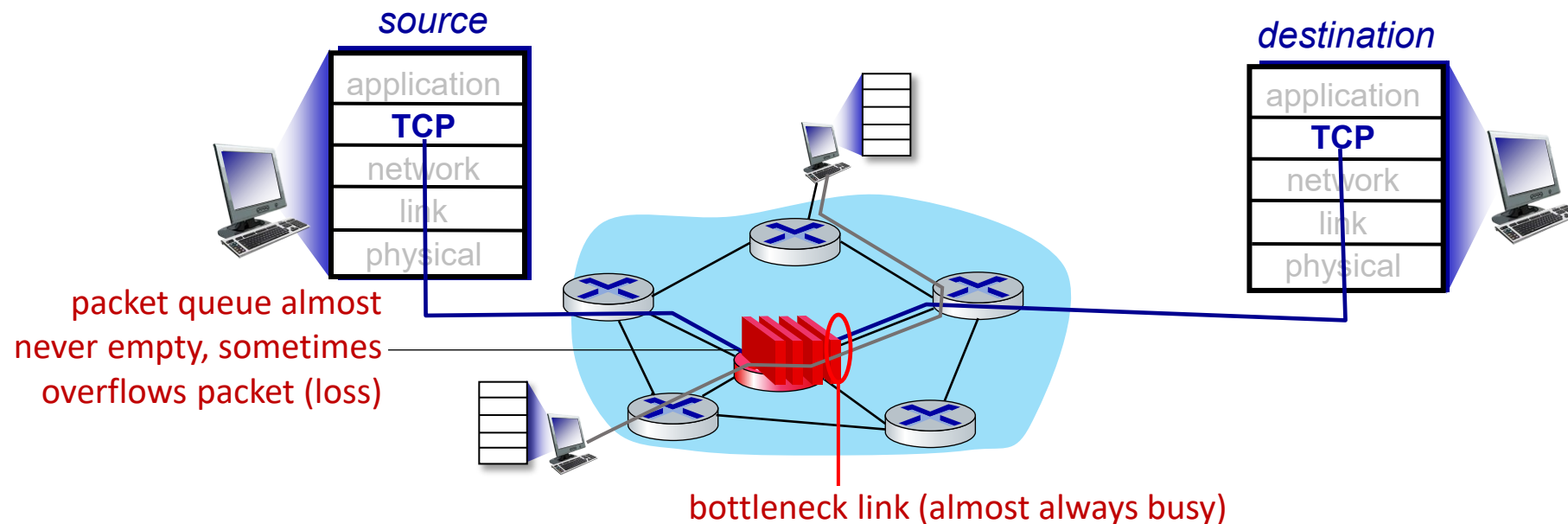
TCP CUBIC

- K: point in time when TCP window size will reach W_{\max}
 - K itself is tunable
- increase W as a function of the *cube* of the distance between current time and K
 - larger increases when further away from K
 - smaller increases (cautious) when nearer K
- TCP CUBIC default in Linux, most popular TCP for popular Web servers



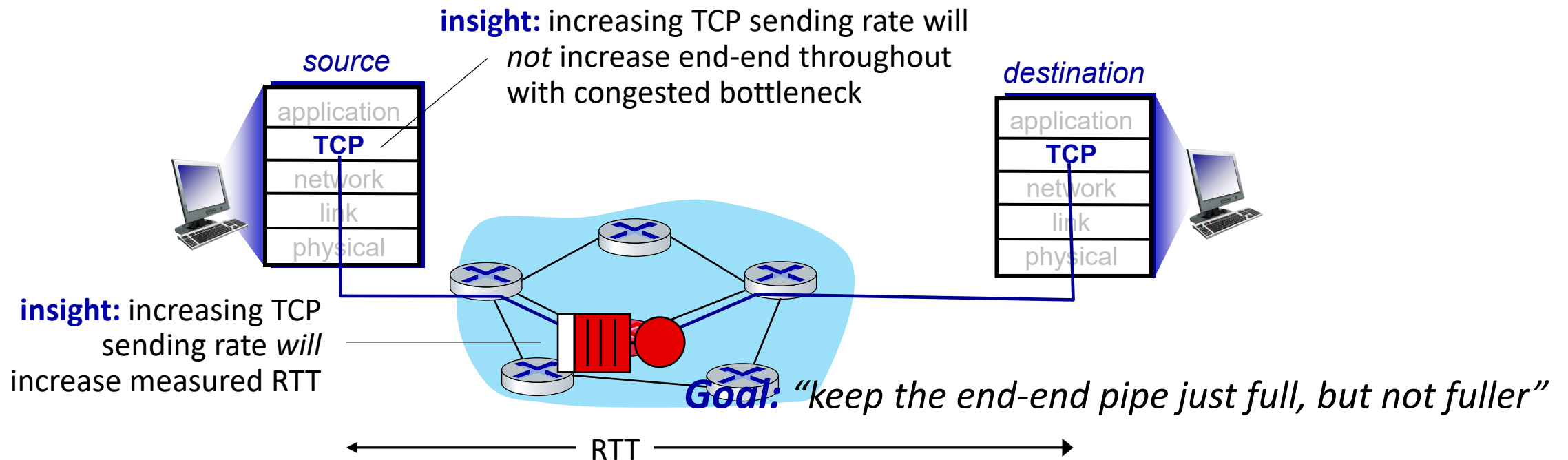
TCP and the congested “bottleneck link”

- TCP (classic, CUBIC) increase TCP’s sending rate until packet loss occurs at some router’s output: the *bottleneck link*



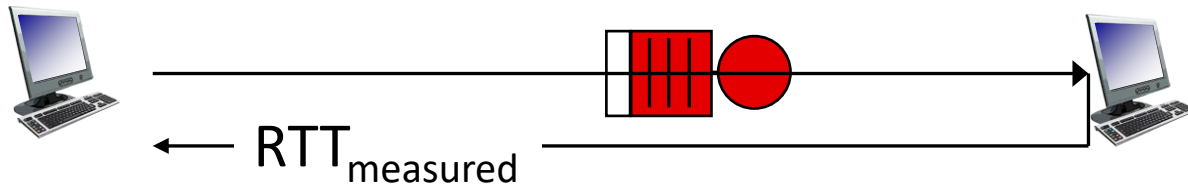
TCP and the congested “bottleneck link”

- TCP (classic, CUBIC) increase TCP’s sending rate until packet loss occurs at some router’s output: the *bottleneck link*
- understanding congestion: useful to focus on congested bottleneck link



Delay-based TCP congestion control

Keeping sender-to-receiver pipe “just full enough, but no fuller”: keep bottleneck link busy transmitting, but avoid high delays/buffering



$$\text{measured throughput} = \frac{\text{\# bytes sent in last RTT interval}}{RTT_{\text{measured}}}$$

Delay-based approach:

- RTT_{min} - minimum observed RTT (uncongested path)
- uncongested throughput with congestion window $cwnd$ is $cwnd/RTT_{\text{min}}$

if measured throughput “very close” to uncongested throughput
increase $cwnd$ linearly /* since path not congested */
else if measured throughput “far below” uncongested throughput
decrease $cwnd$ linearly /* since path is congested */

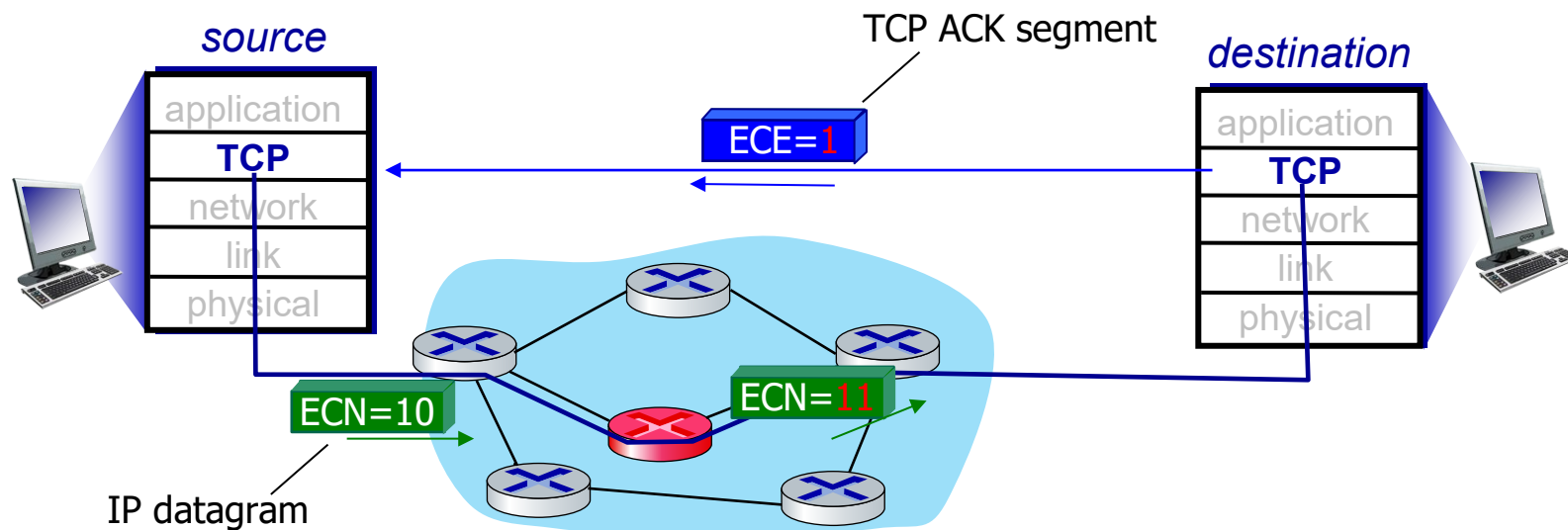
Delay-based TCP congestion control

- congestion control without inducing/forcing loss
- maximizing throughput (“keeping the just pipe full... ”) while keeping delay low (“...but not fuller”)
- a number of deployed TCPs take a delay-based approach
 - BBR deployed on Google’s (internal) backbone network

Explicit congestion notification (ECN)

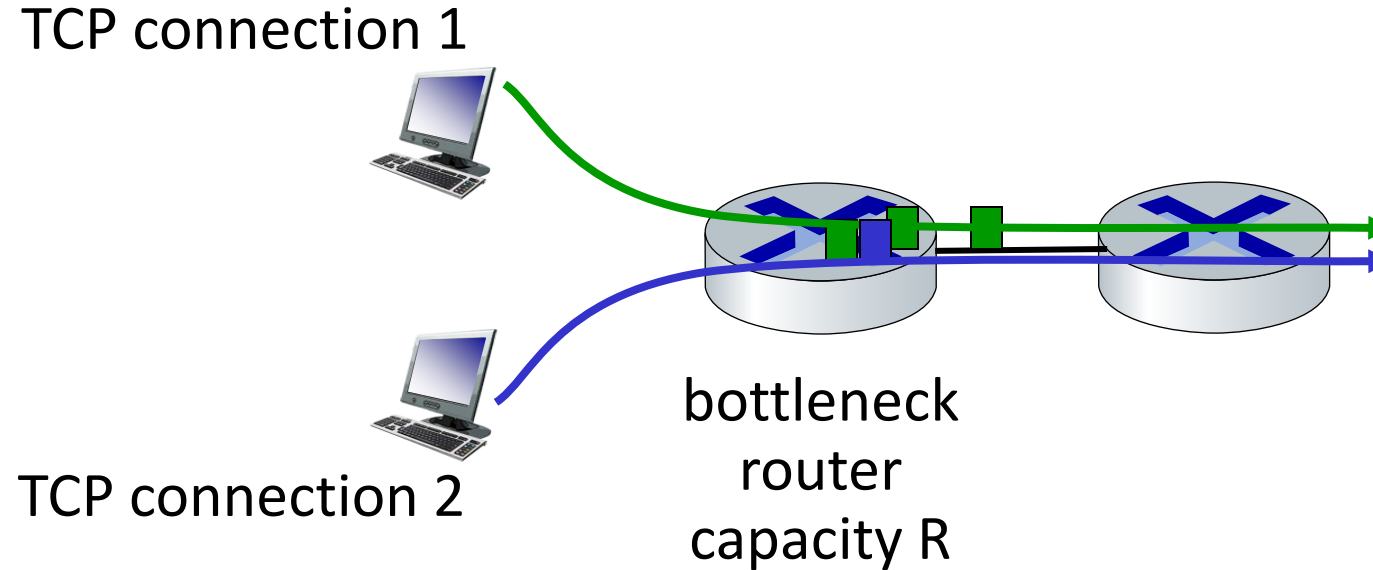
TCP deployments often implement *network-assisted* congestion control:

- two bits in IP header (ToS field) marked *by network router* to indicate congestion
 - *policy* to determine marking chosen by network operator
- congestion indication carried to destination
- destination sets ECE bit on ACK segment to notify sender of congestion
- involves both IP (IP header ECN bit marking) and TCP (TCP header C,E bit marking)



TCP fairness

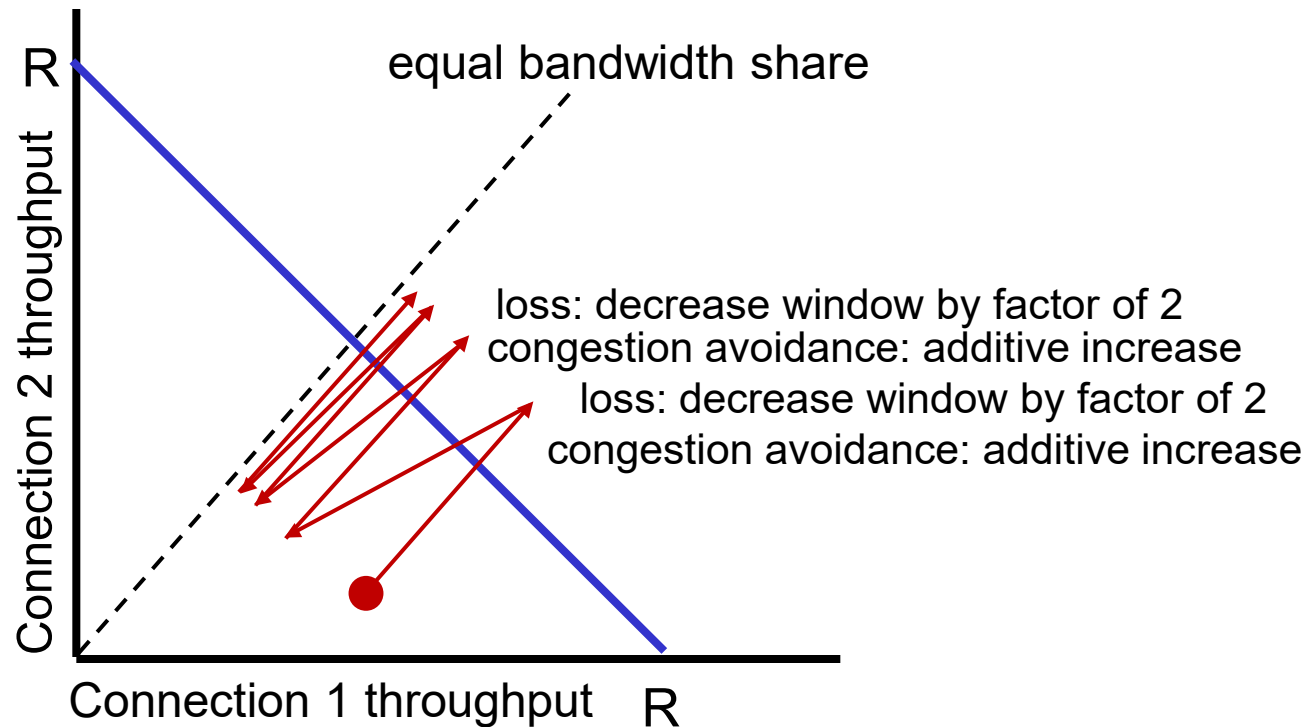
Fairness goal: if K TCP sessions share same bottleneck link of bandwidth R , each should have average rate of R/K



Q: is TCP Fair?

Example: two competing TCP sessions:

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



Is TCP fair?

A: Yes, under idealized assumptions:

- same RTT
- fixed number of sessions only in congestion avoidance

Fairness: must all network apps be “fair”?

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
- there is no “Internet police” policing use of congestion control
 - UDP is the “bully”, and TCP is the “nice guy”
- Interactive applications such as Zoom use UDP for transmitting real-time interactive media such as audio and video; Streaming video applications such as YouTube or Netflix use TCP for delivering high-quality video content that is more delay-tolerant

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 $R/2$ ($R * 11/20$)