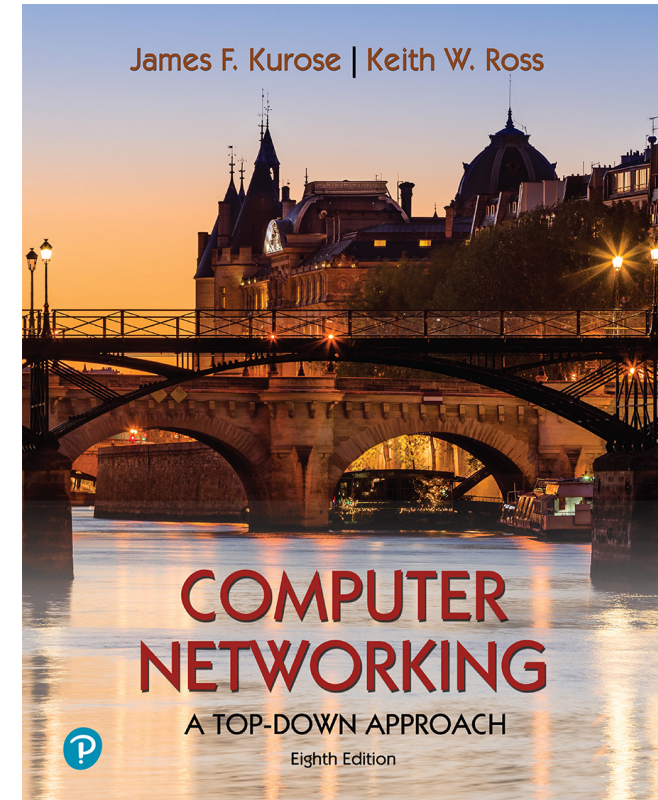


# Chapter 3

## Transport Layer



### *Computer Networking: A Top-Down Approach*

8<sup>th</sup> edition

Jim Kurose, Keith Ross  
Pearson, 2020

Acknowledgement: Based on the textbook's website:  
[https://gaia.cs.umass.edu/kurose\\_ross/index.php](https://gaia.cs.umass.edu/kurose_ross/index.php)

# Chapter 3: roadmap

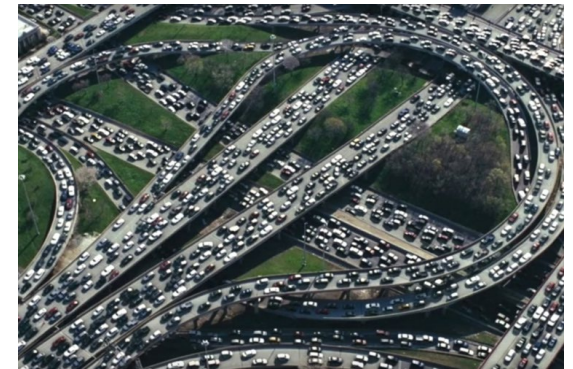
- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Connection-oriented transport: TCP
- **Principles of congestion control**
- TCP congestion control
- Evolution of transport-layer functionality



# 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

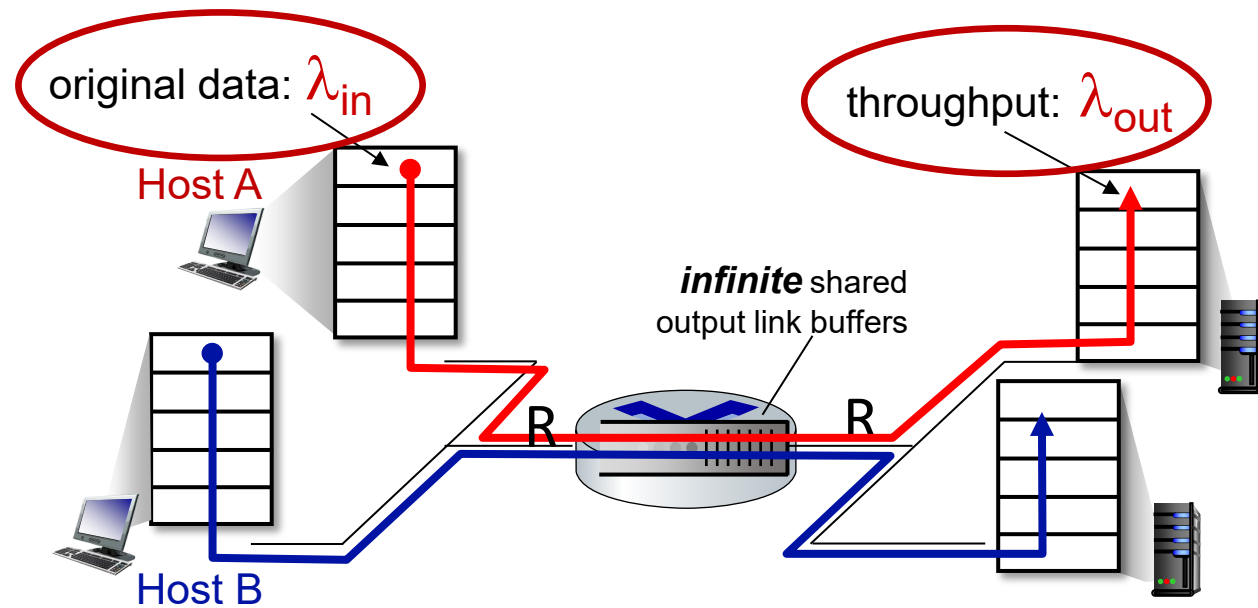


**flow control:** one sender  
too fast for one receiver

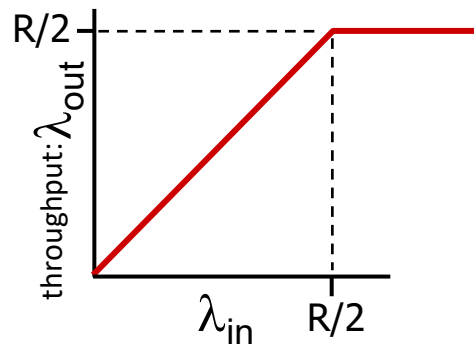
# Causes/costs of congestion: scenario 1

Simplest scenario:

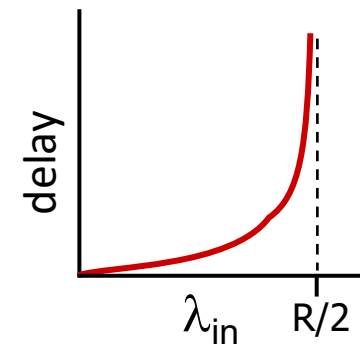
- one router, infinite buffers
- input, output link capacity:  $R$
- two flows
- no retransmissions needed



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



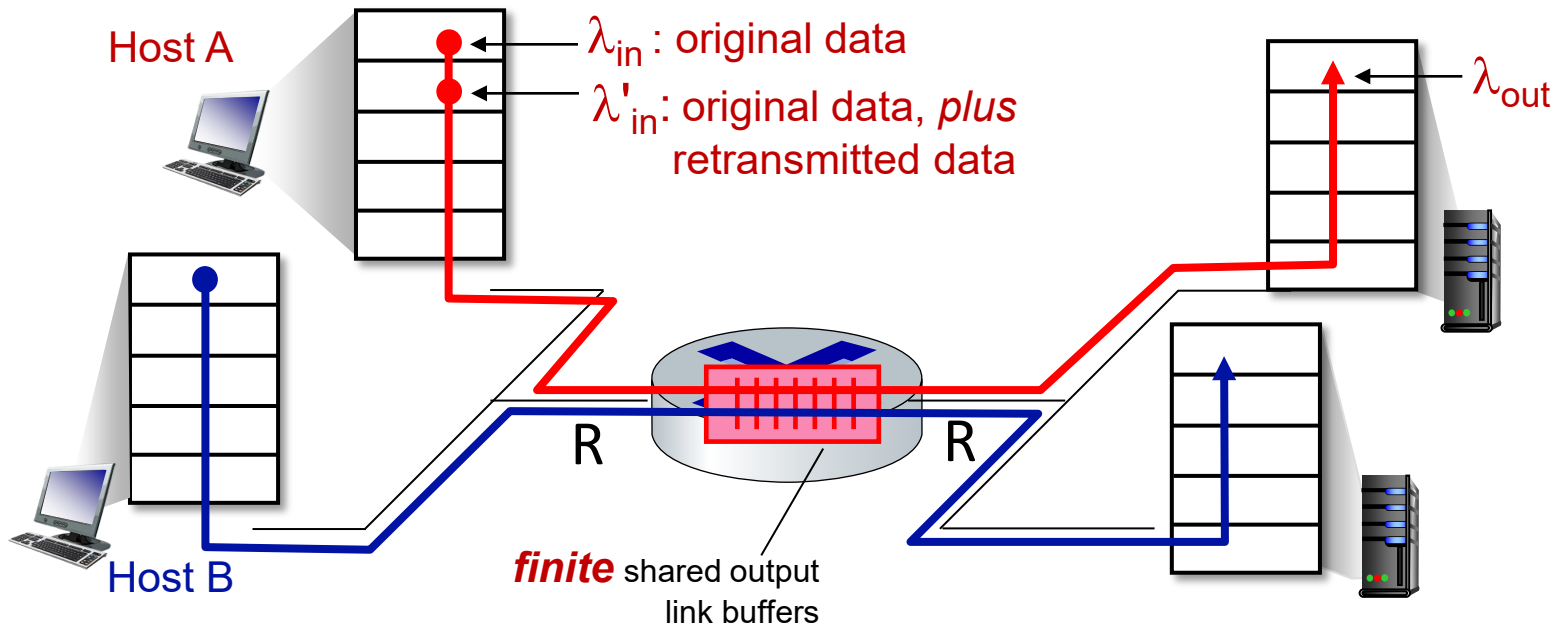
maximum per-connection throughput:  $R/2$



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

# Causes/costs of congestion: scenario 2

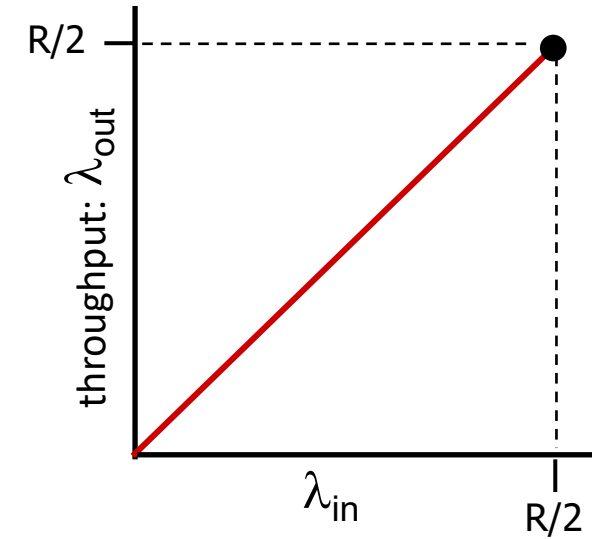
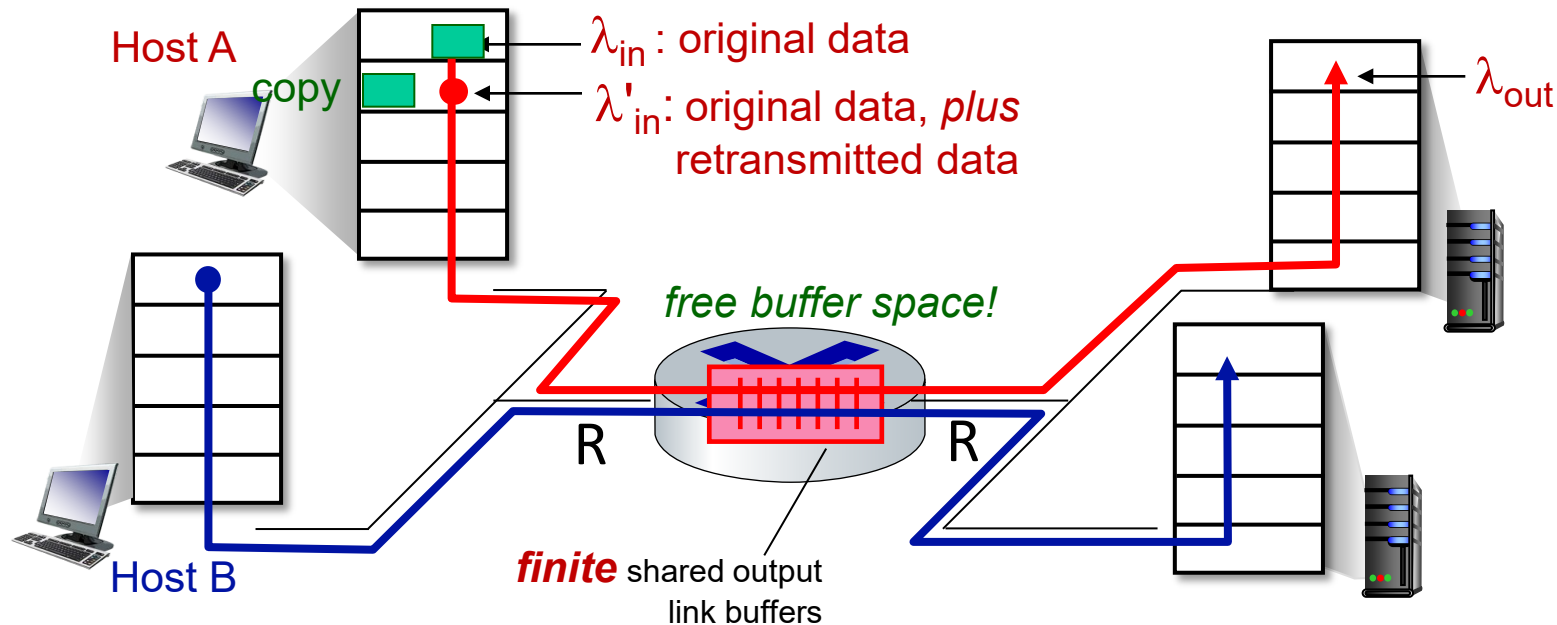
- one router, *finite* buffers
- sender retransmits lost, timed-out packet
  - application-layer input = application-layer output:  $\lambda_{in} = \lambda_{out}$
  - transport-layer input includes *retransmissions* :  $\lambda'_{in} \geq \lambda_{in}$



# Causes/costs of congestion: scenario 2

Idealization: perfect knowledge

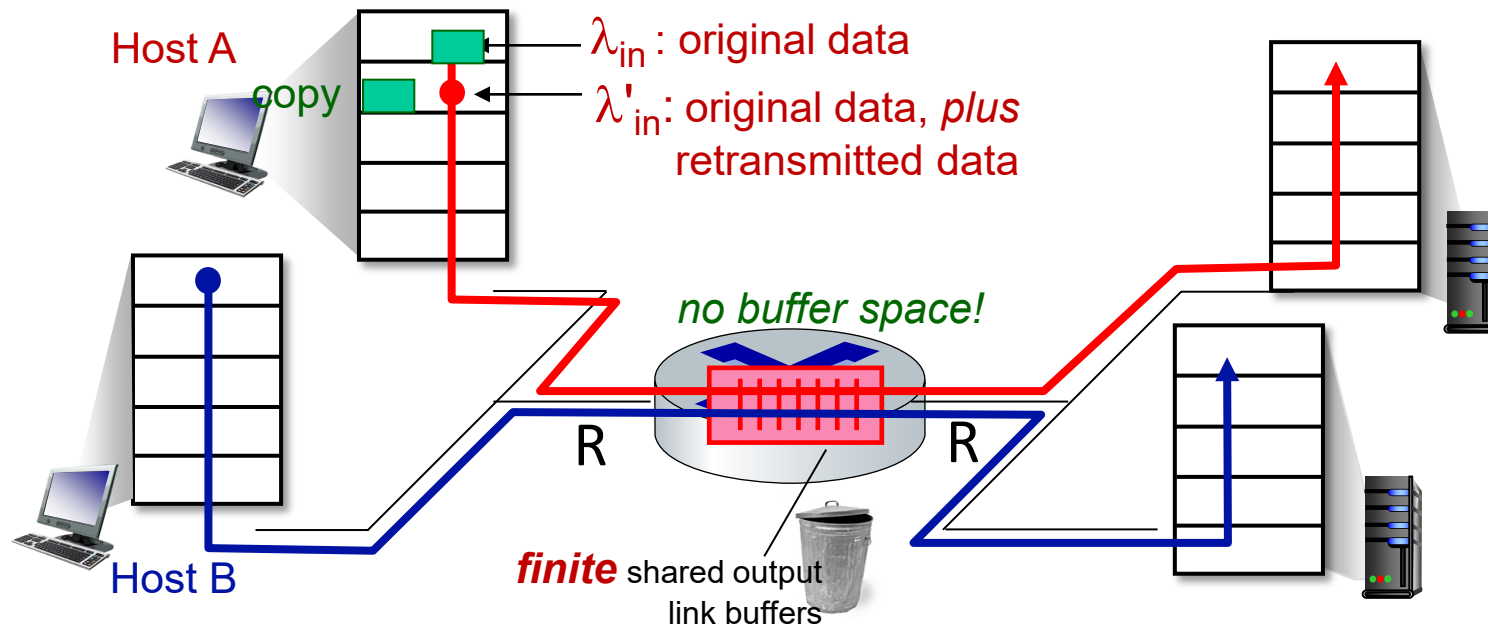
- sender sends only when router buffers available



# 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

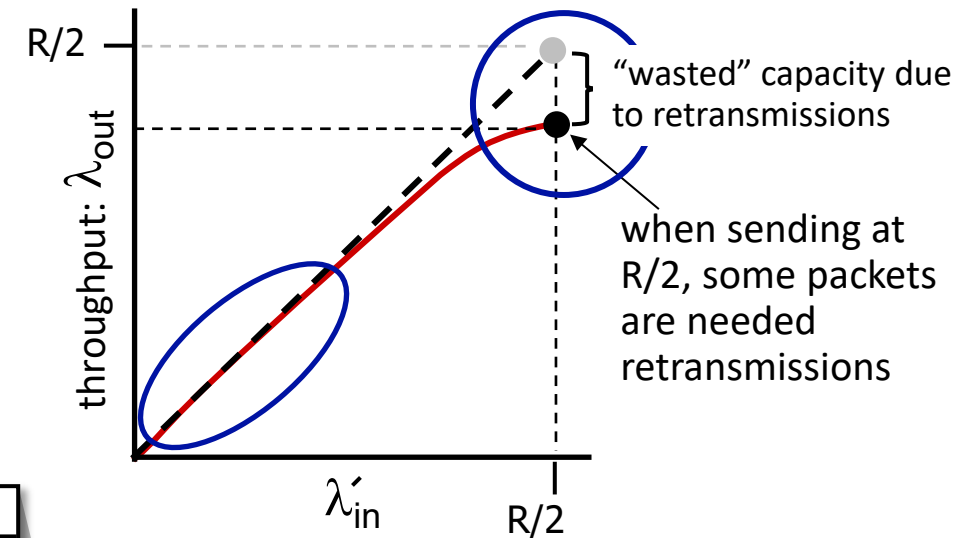
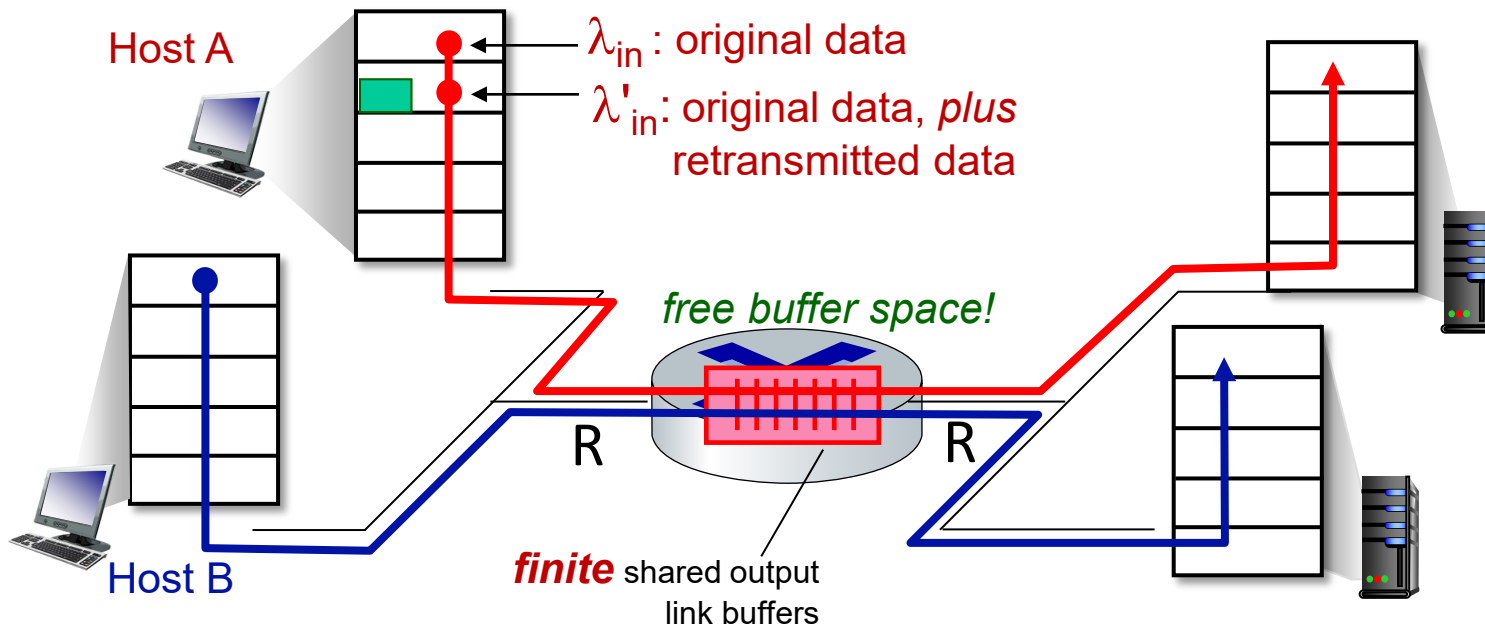




# Causes/costs of congestion: scenario 2

## 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

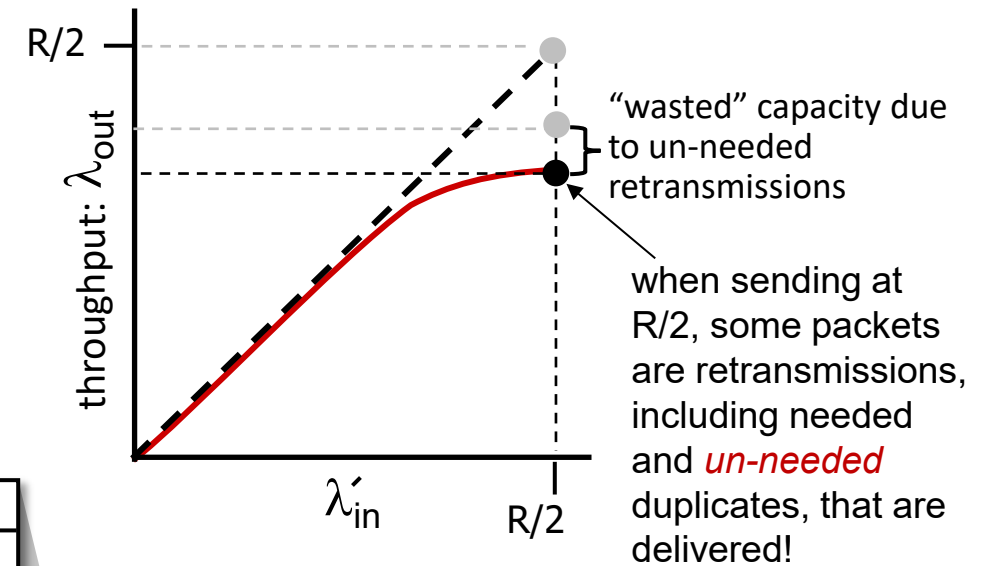
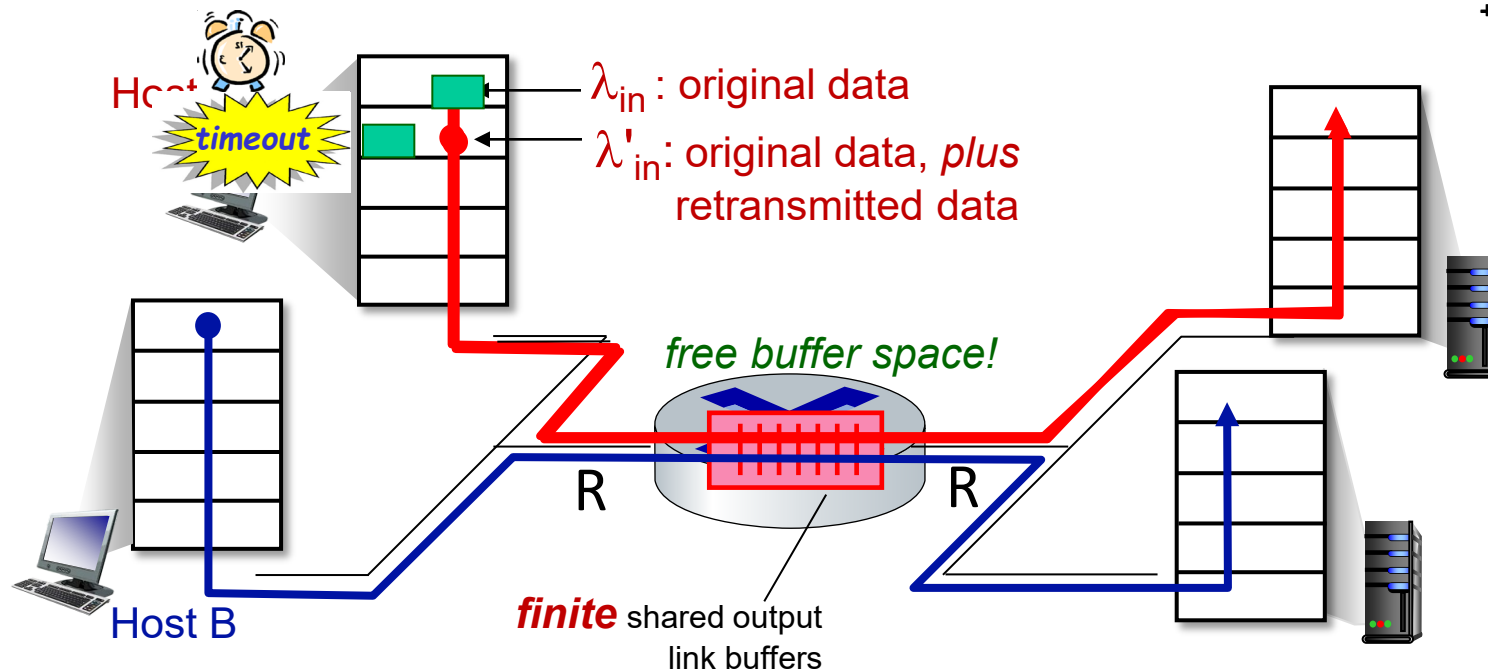




# Causes/costs of congestion: scenario 2

## Realistic scenario: *un-needed duplicates*

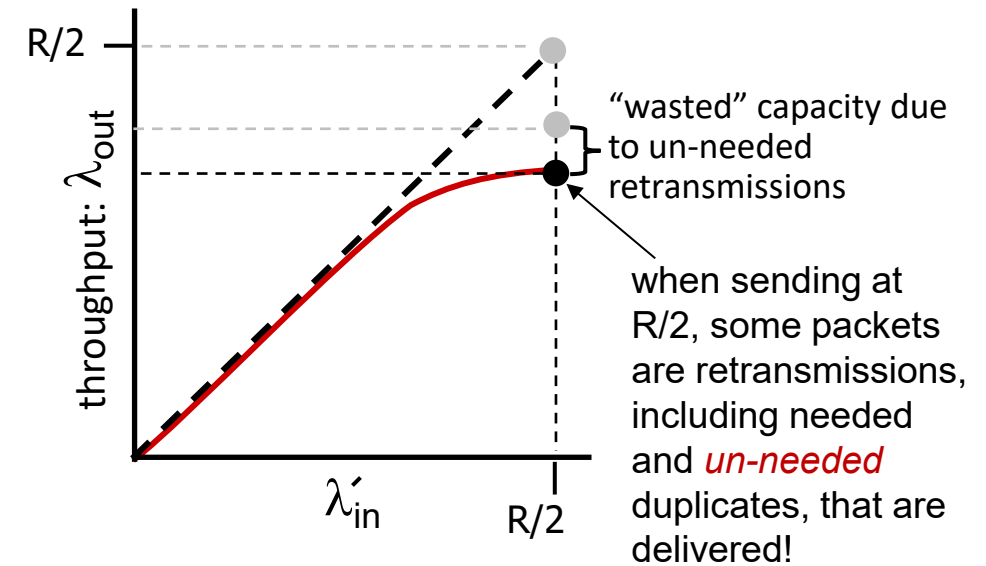
- packets can be lost, dropped at router due to full buffers – requiring retransmissions
- but sender times can time out prematurely, sending *two* copies, *both* of which are delivered



# 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:

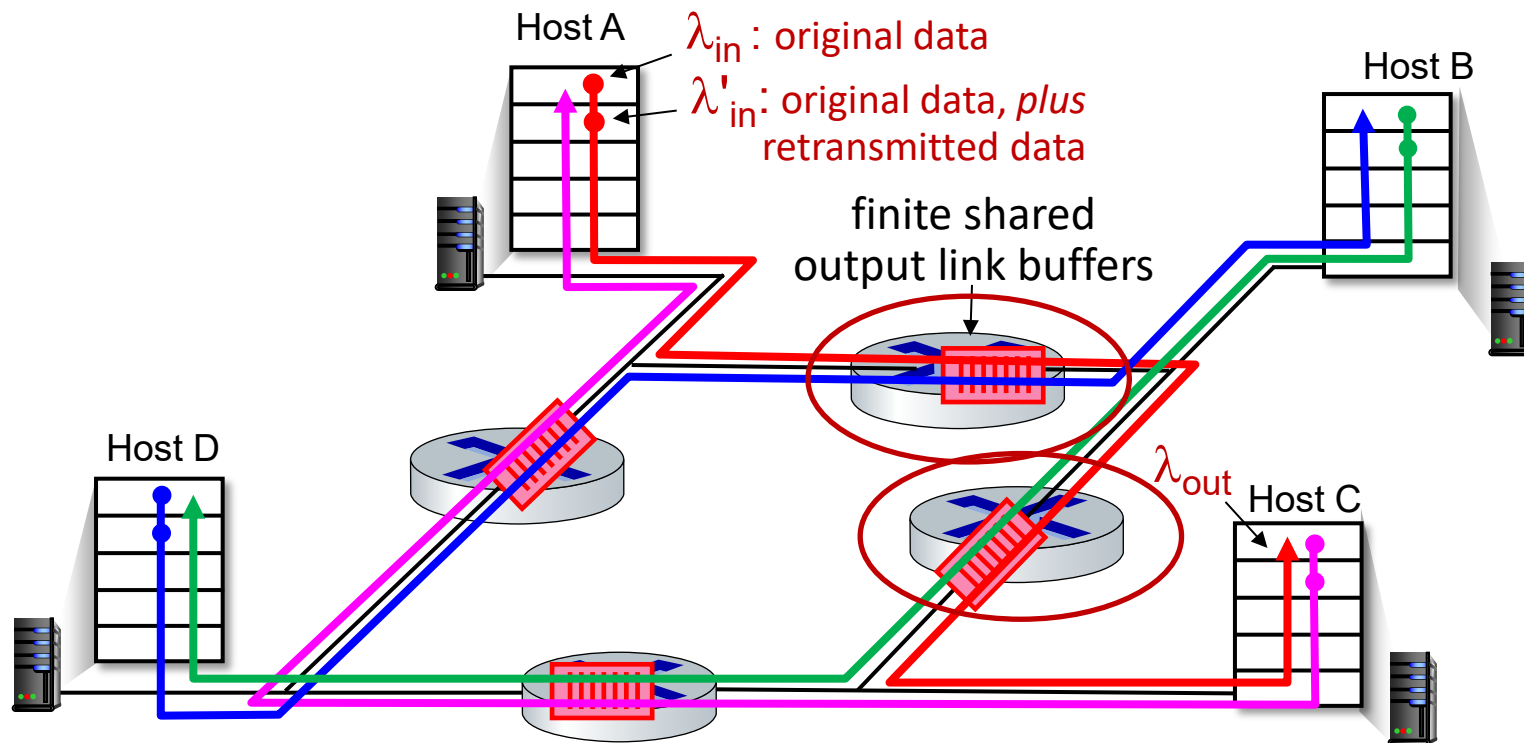
- more work (retransmission) for given receiver throughput
- unneeded retransmissions: link carries multiple copies of a packet
  - decreasing maximum achievable throughput

# Causes/costs of congestion: scenario 3

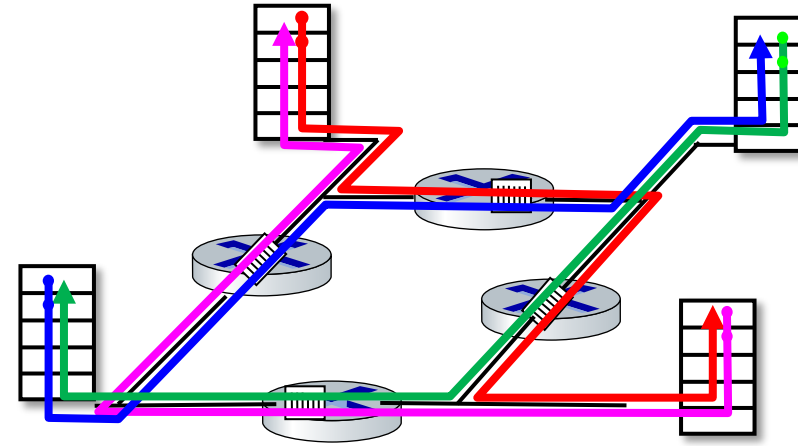
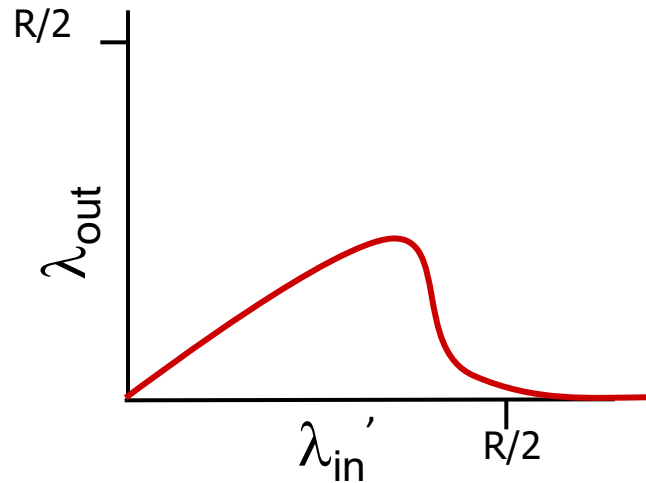
- *four* senders
- *multi-hop* paths
- timeout/retransmit

Q: what happens as  $\lambda_{in}$  and  $\lambda'_{in}$  increase ?

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



# Causes/costs of congestion: scenario 3

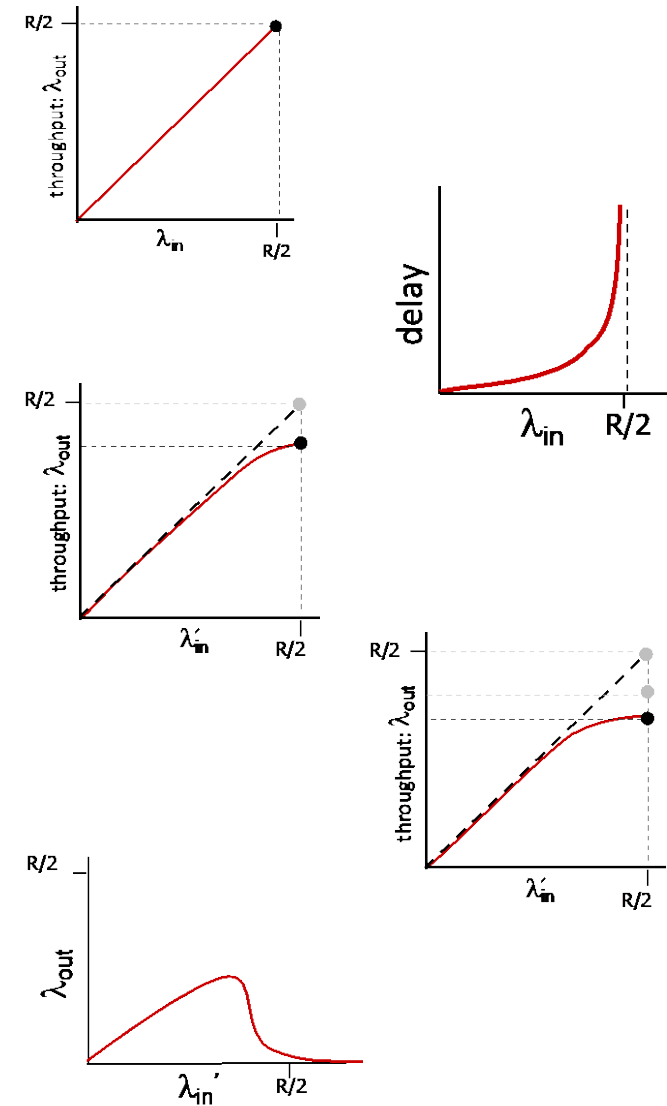


another “cost” of congestion:

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

# Causes/costs of congestion: insights

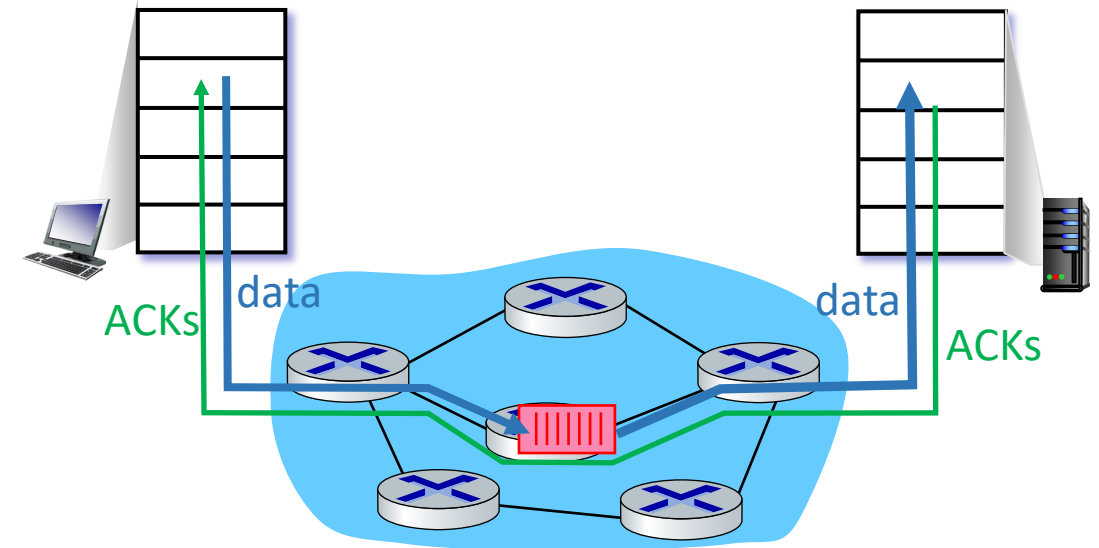
- throughput can never exceed capacity
- delay increases as capacity approached
- loss/retransmission decreases effective throughput
- un-needed duplicates further decreases effective throughput
- upstream transmission capacity / buffering wasted for packets lost downstream



# Approaches towards congestion control

## End-end congestion control:

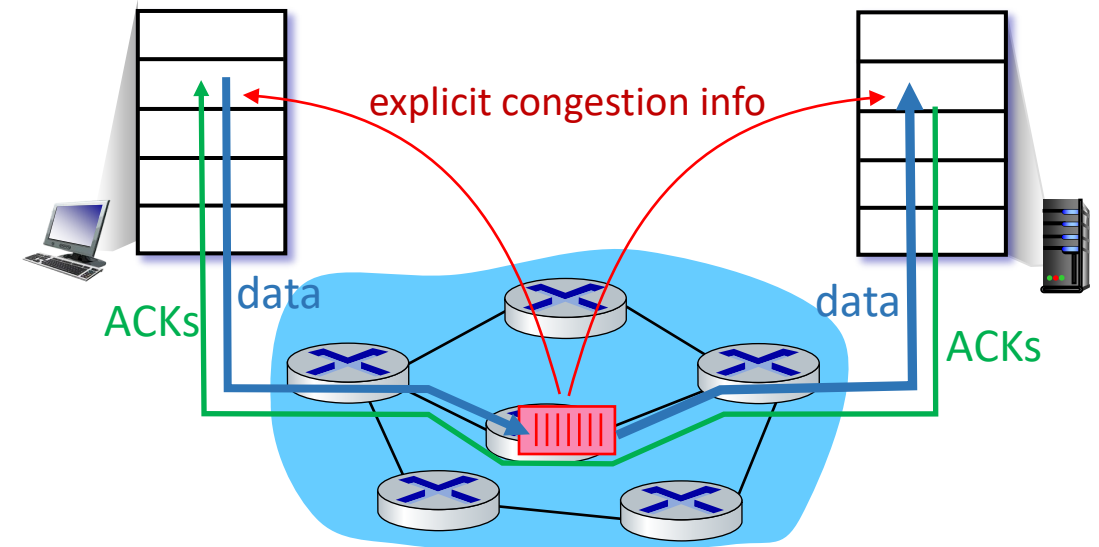
- no explicit feedback from network
- congestion *inferred* from observed loss, delay
- approach taken by TCP



# 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

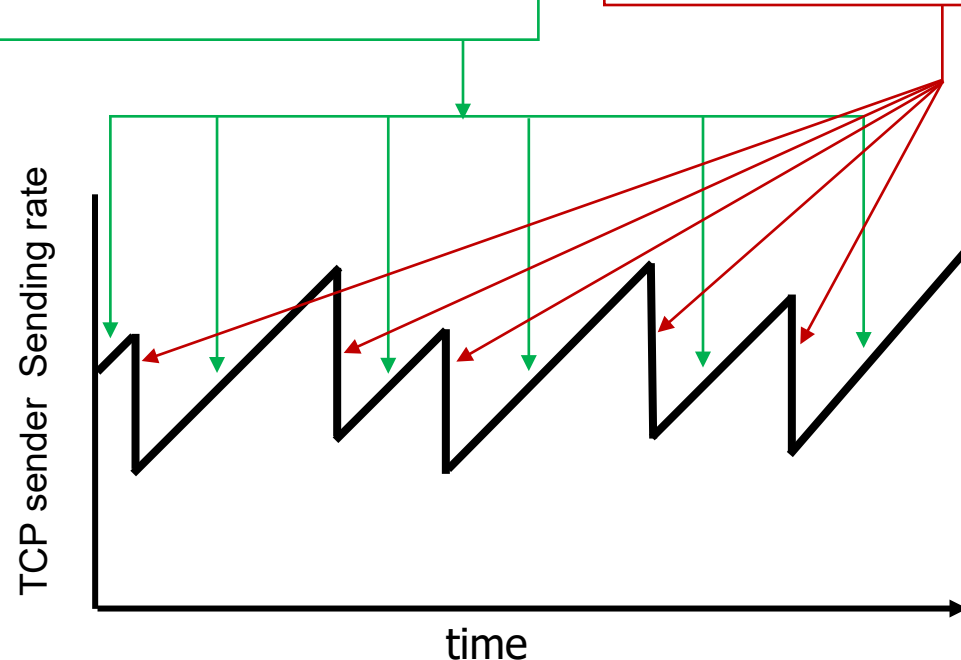
- *approach*: senders can increase sending rate until packet loss (congestion) occurs, then decrease sending rate on loss event

## 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

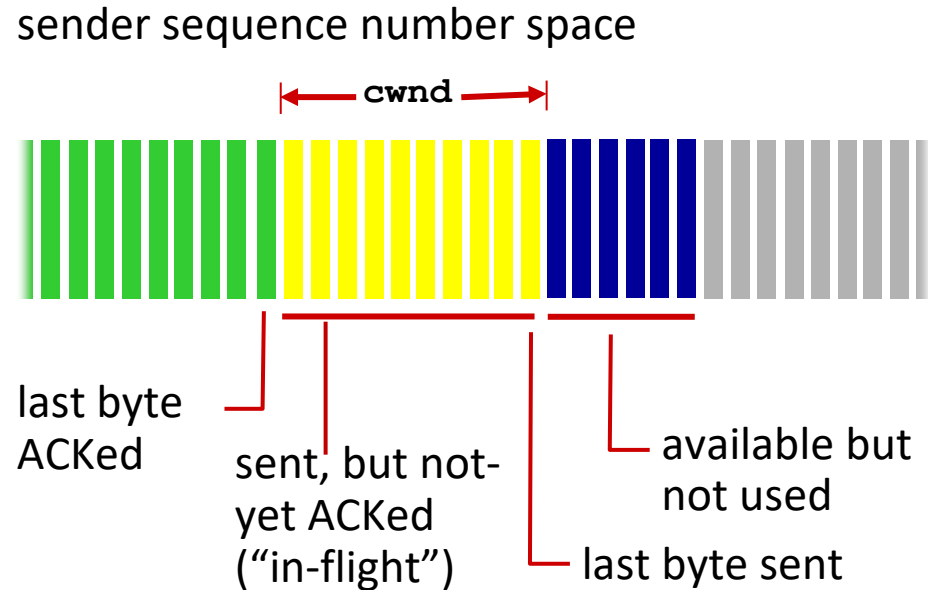
*Multiplicative decrease* detail: sending rate is

- 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:

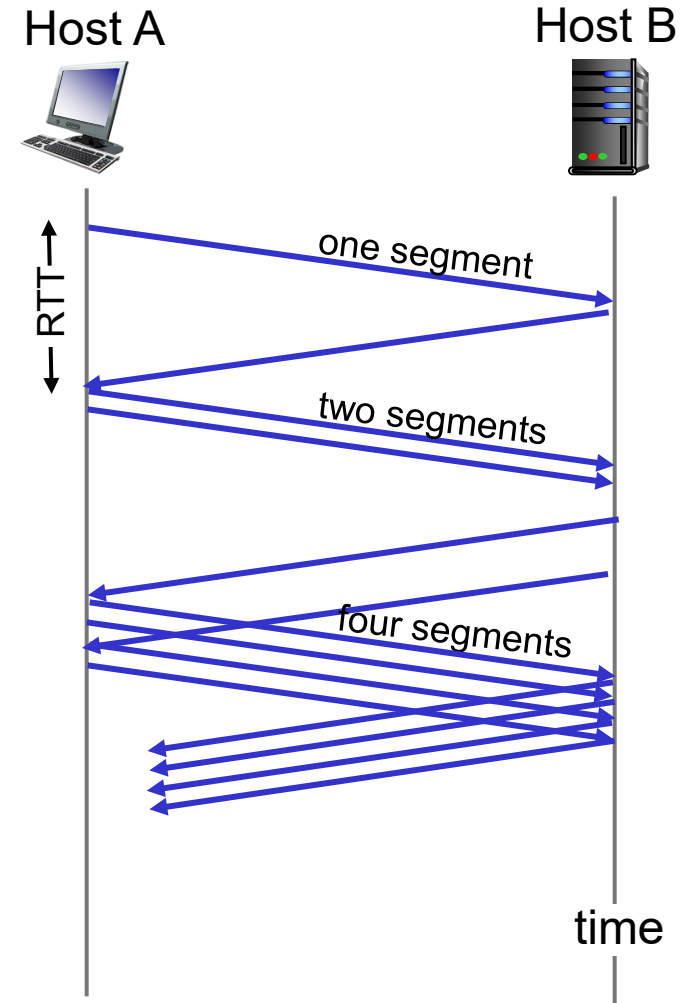
- *roughly*: send `cwnd` bytes, wait RTT for ACKS, then send more bytes

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

- 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



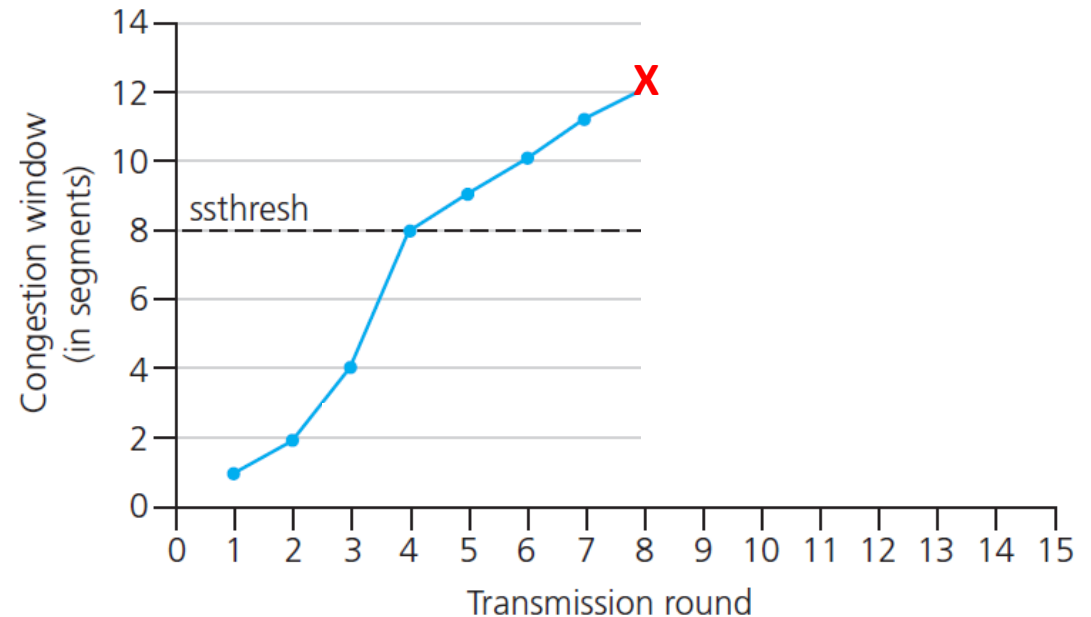
# TCP: from slow start to congestion avoidance

**Q:** when should the exponential increase switch to linear?

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

## Implementation:

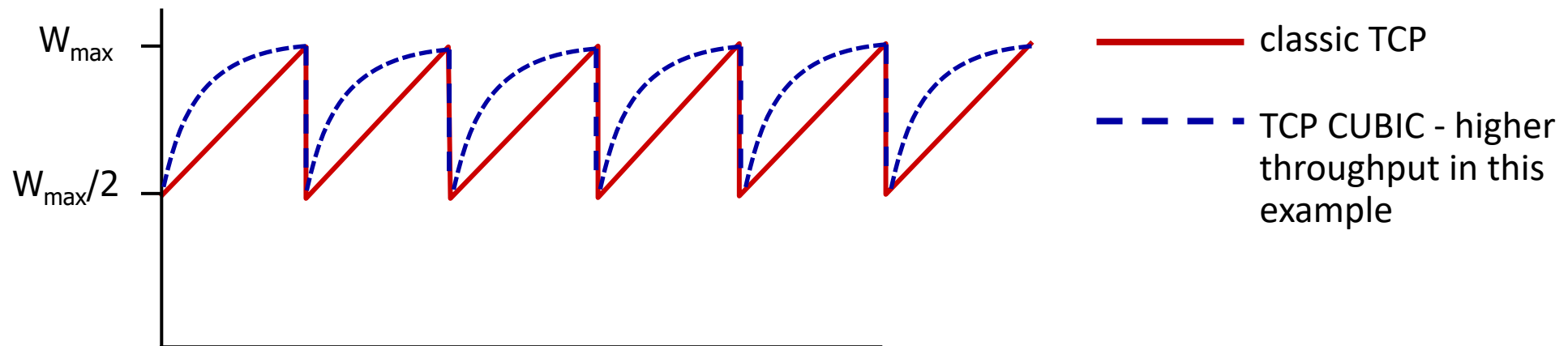
- variable **ssthresh**
- on loss event, **ssthresh** is set to 1/2 of **cwnd** just before loss event



\* Check out the online interactive exercises for more examples: [http://gaia.cs.umass.edu/kurose\\_ross/interactive/](http://gaia.cs.umass.edu/kurose_ross/interactive/)

# TCP CUBIC

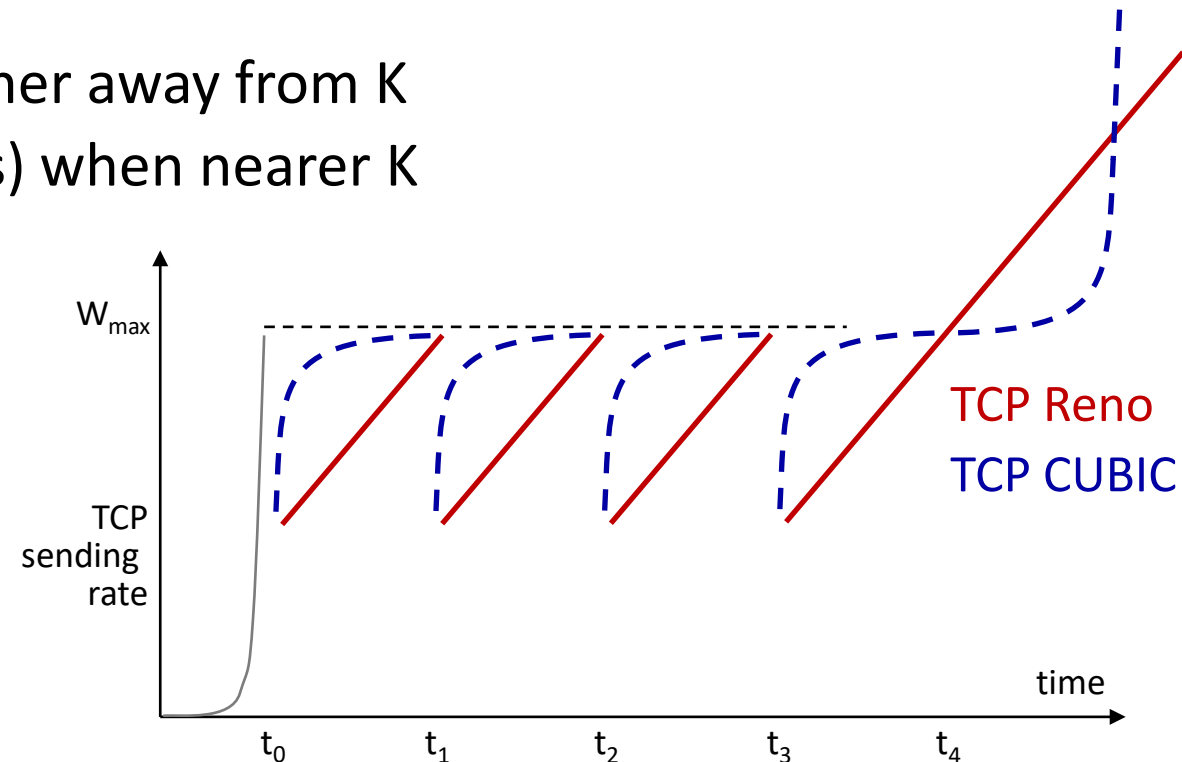
- Is there a better way than AIMD to “probe” for usable bandwidth?
- Insight/intuition:
  - $W_{\max}$ : sending rate at which congestion loss was detected
  - congestion state of bottleneck link probably (?) hasn't changed much
  - after cutting rate/window in half on loss, initially ramp to to  $W_{\max}$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





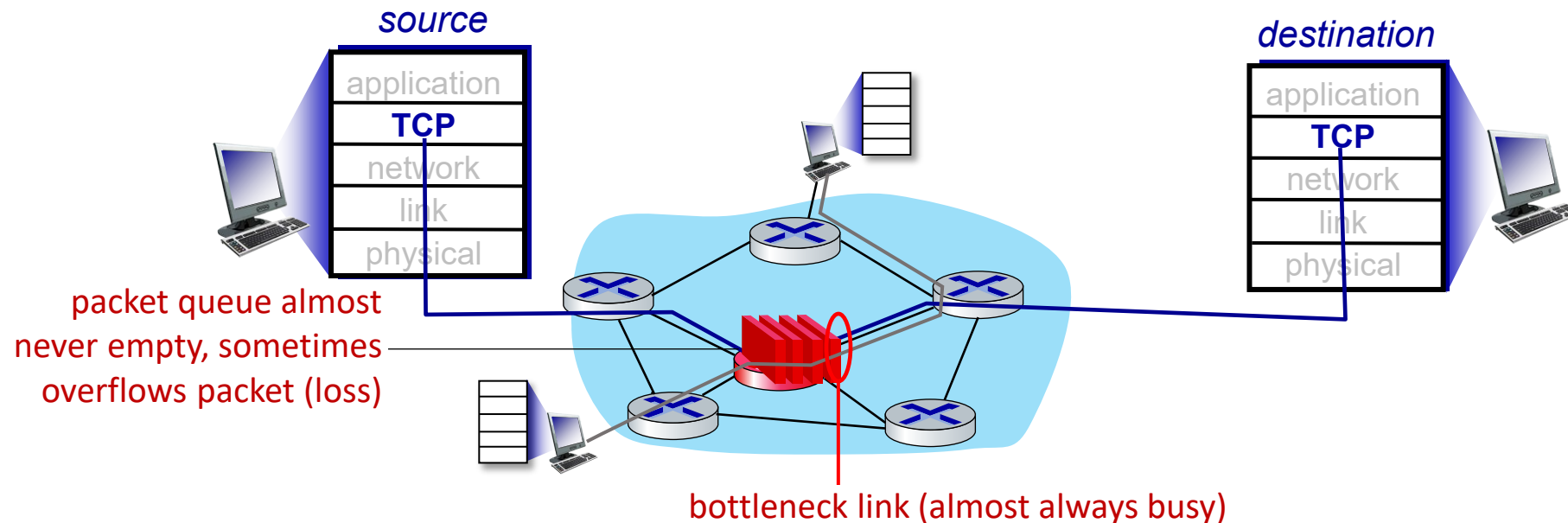
# TCP CUBIC

- 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



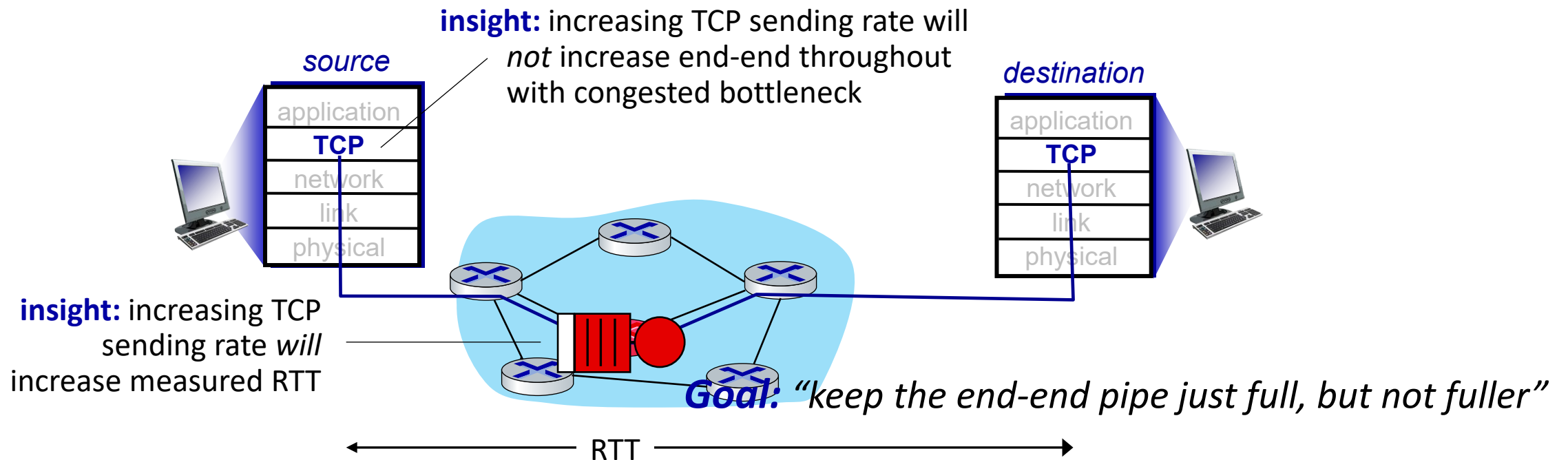
# TCP and the congested “bottleneck link”

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



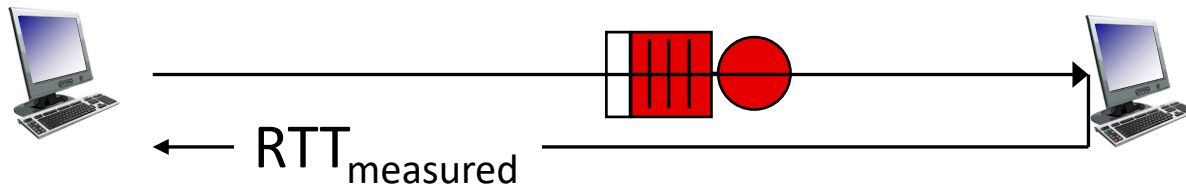
# TCP and the congested “bottleneck link”

- 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:

- $RTT_{\text{min}}$  - 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 \*/

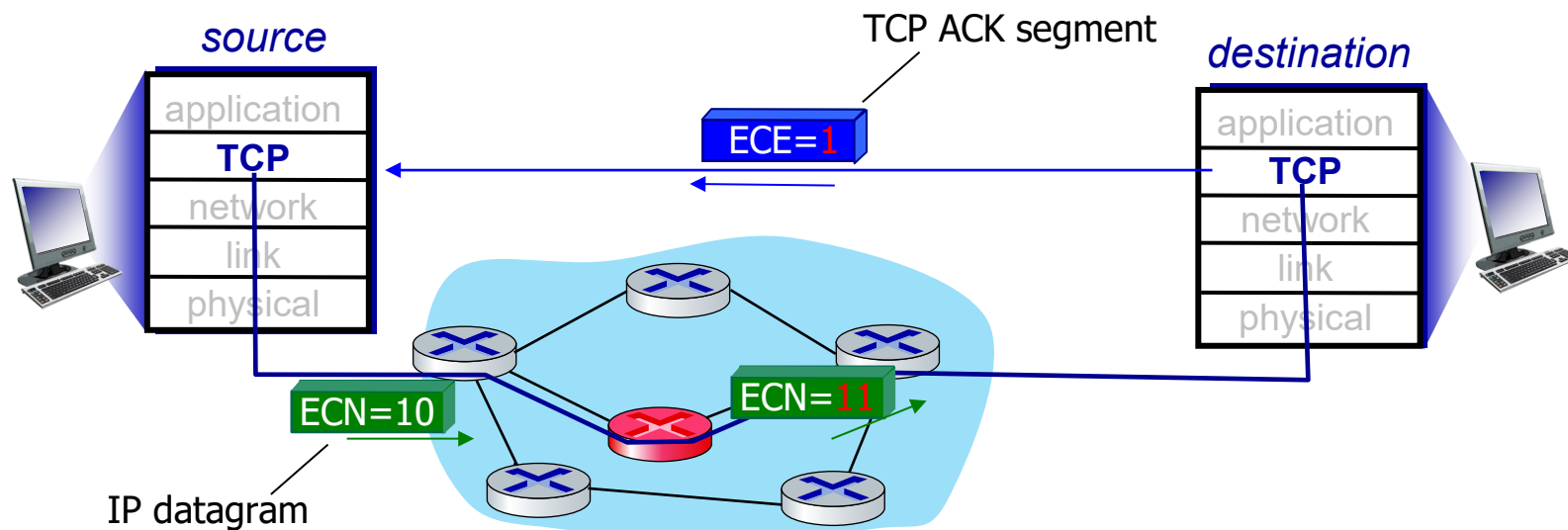
# Delay-based TCP congestion control

- 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)

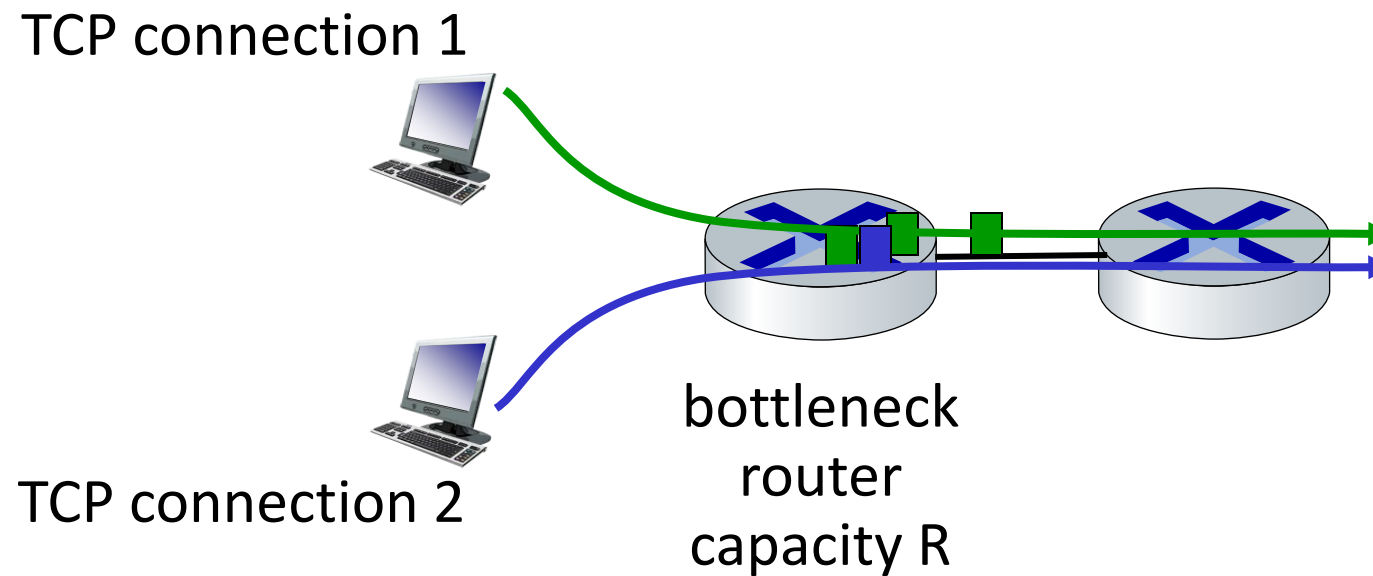
TCP deployments often implement *network-assisted* congestion control:

- 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  $R/K$

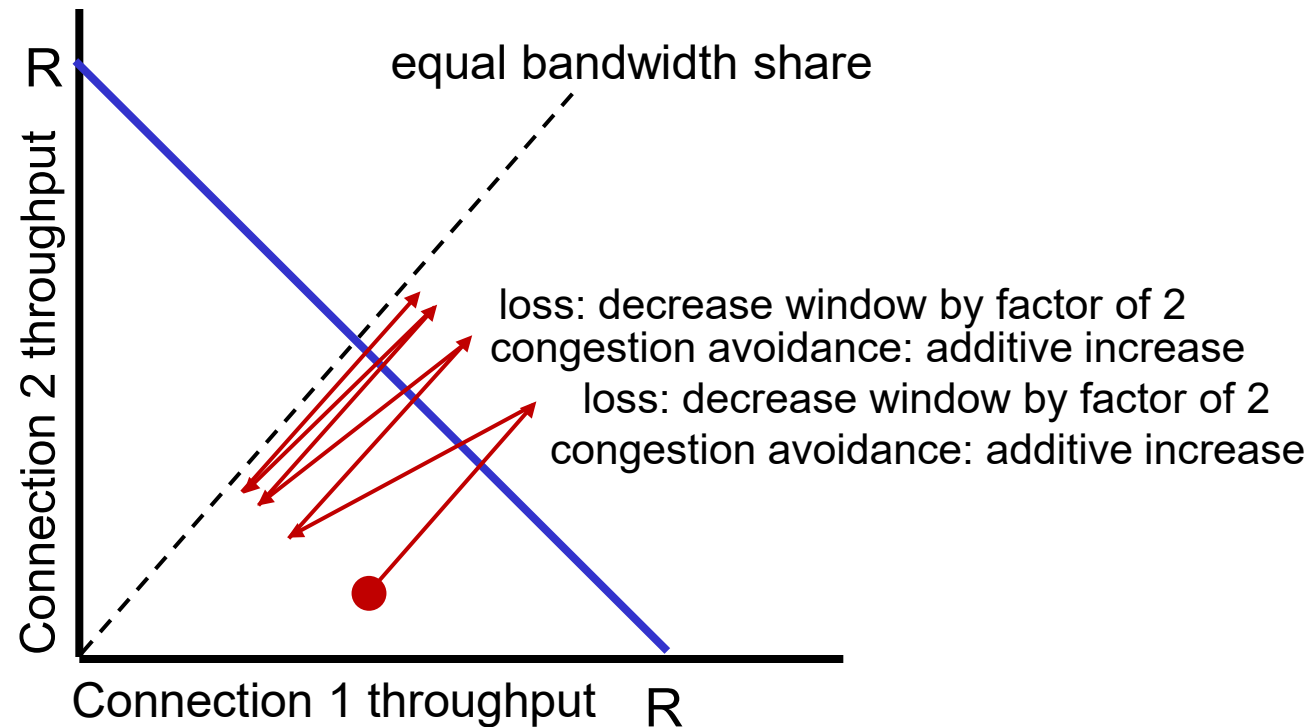




# Q: is TCP Fair?

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