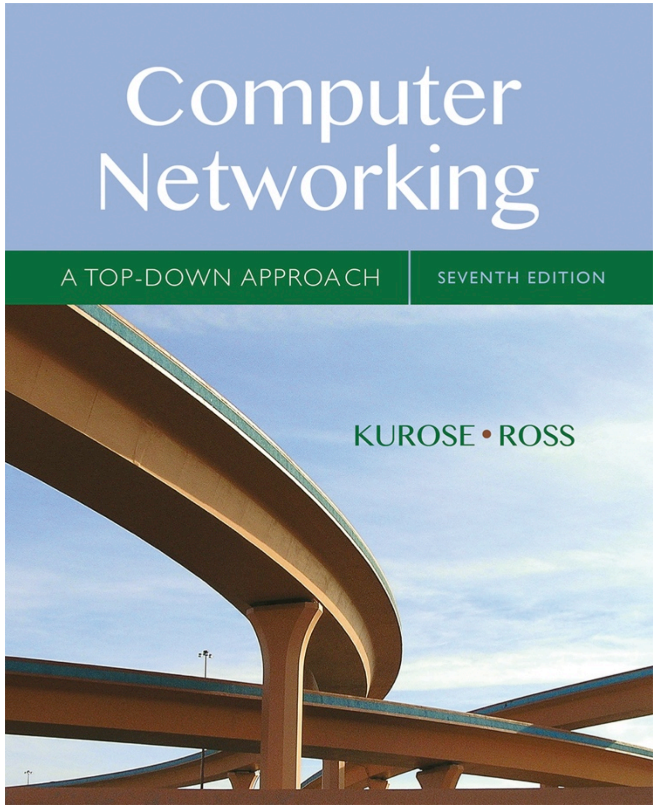
شبکه های کامپیوتری ۲

درس ۹ فصل ۳

Congestion

دانشگاه صنعتی اصفهان
دانشکده مهندسی برق و کامپیوتر

Chapter 1
Introduction



Computer Networking: A Top Down Approach

7th edition
Jim Kurose, Keith Ross
Pearson/Addison Wesley
April 2016

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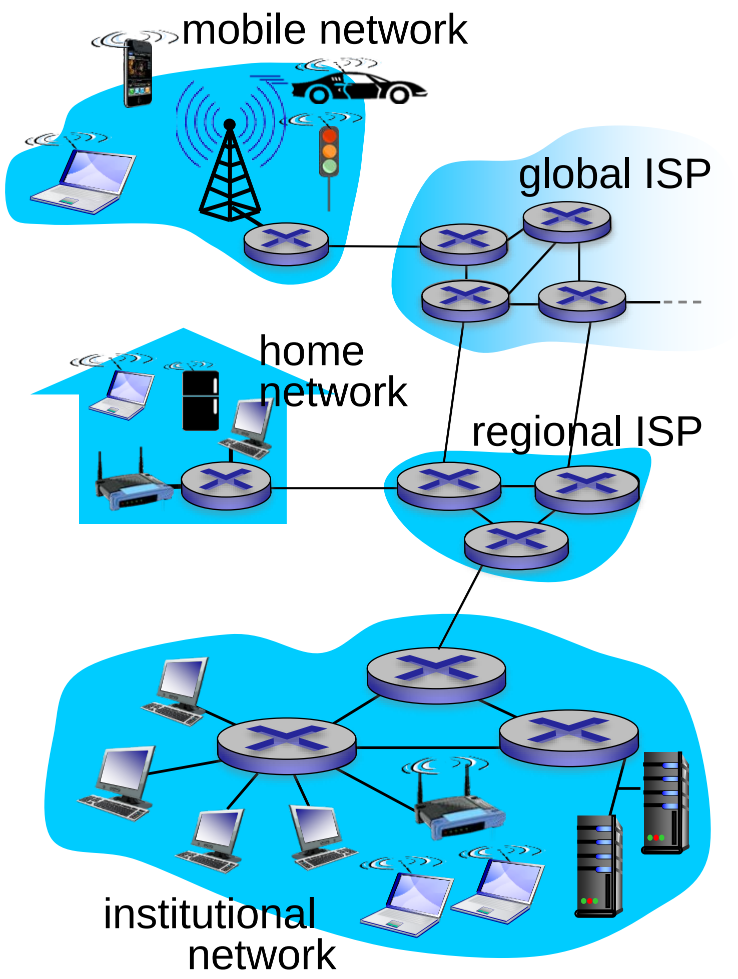
Thanks and enjoy! JFK/KWR

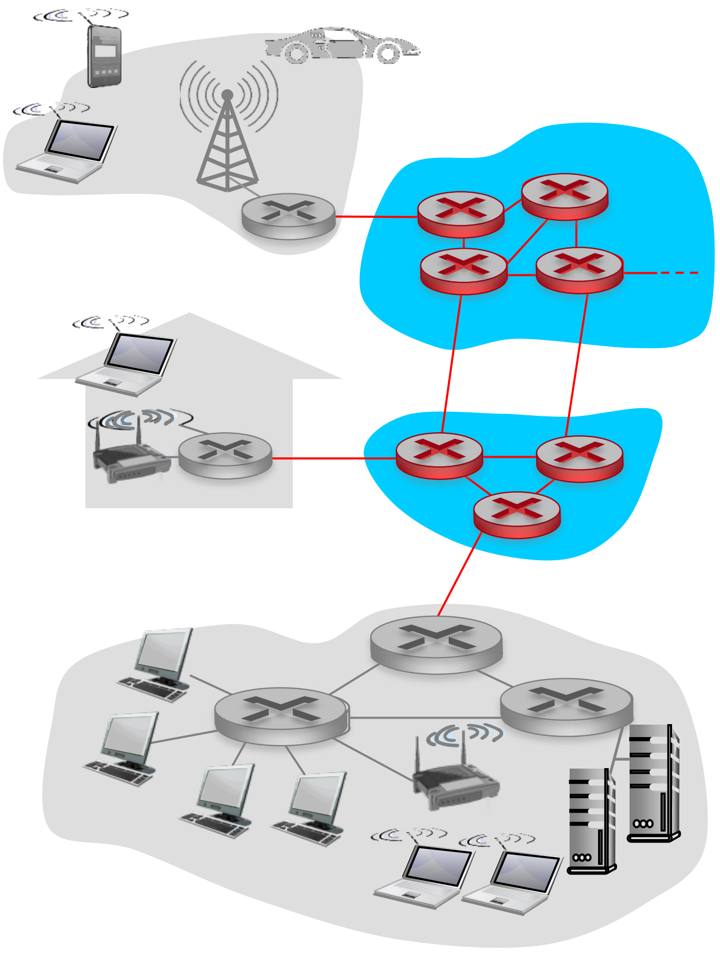
What's the Internet: a service view

- *infrastructure that provides services to applications:*
  - Web, VoIP, email, games, e-commerce, social nets, …
- *provides programming interface to apps*
  - hooks that allow sending and receiving app programs to "connect" to Internet
  - provides service options, analogous to postal service

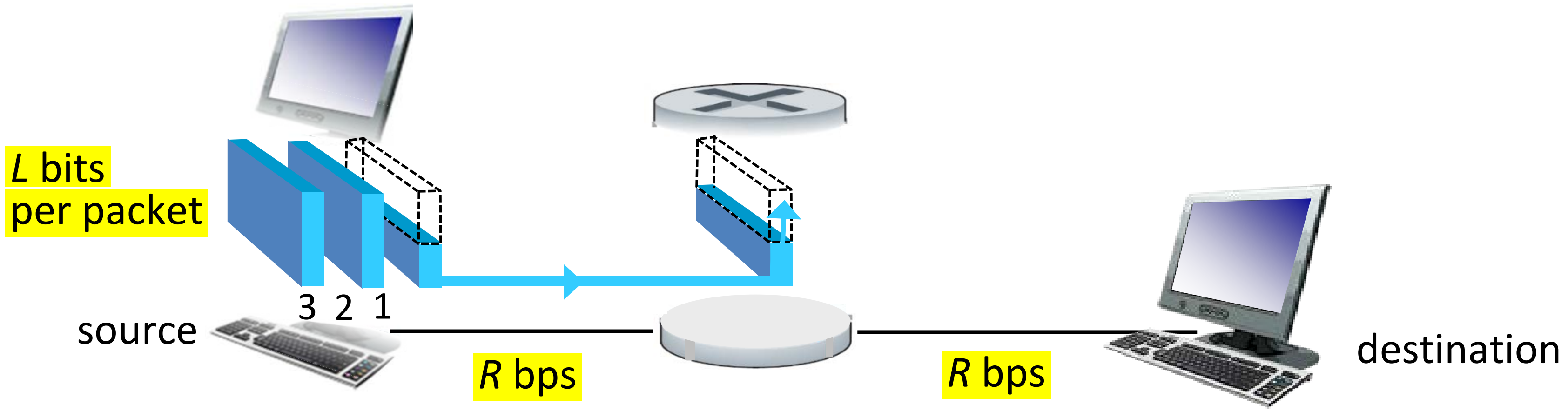


The network core

- mesh of interconnected routers
- packet-switching: hosts break application-layer messages into *packets*
  - forward packets from one router to the next, across links on path from source to destination
  - each packet transmitted at full link capacity



Packet-switching: store-and-forward



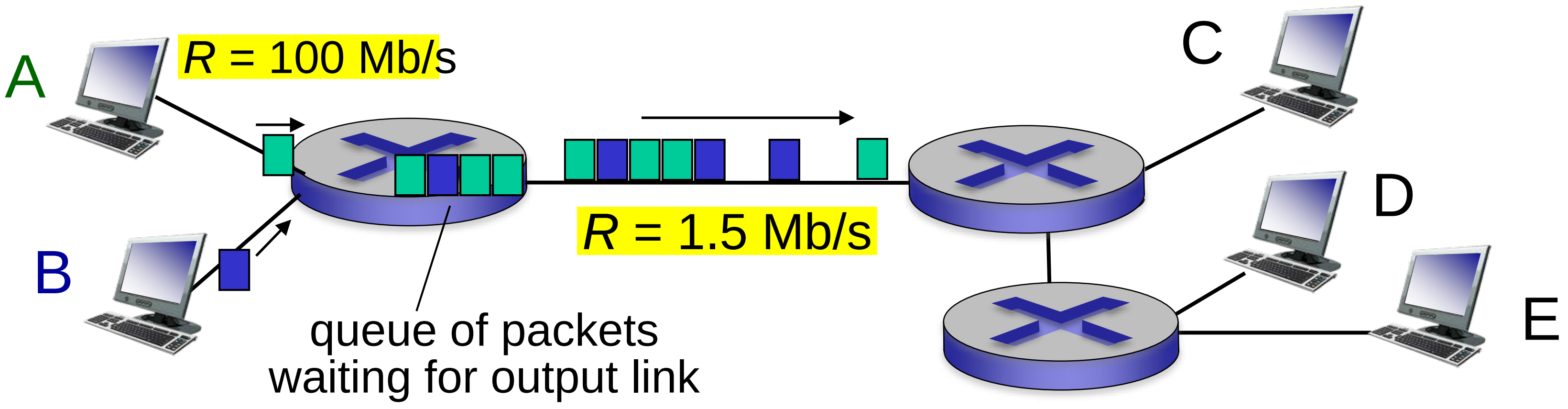
- takes L/R seconds to transmit (push out) L-bit packet into link at R bps
- *store and forward:* entire packet must arrive at router before it can be transmitted on next link
- end-end delay = $2L/R$ (assuming zero propagation delay)

more on delay shortly …

one-hop numerical example:

- L = 7.5 Mbits
- R = 1.5 Mbps
- one-hop transmission delay = 5 sec

Packet Switching: queueing delay, loss



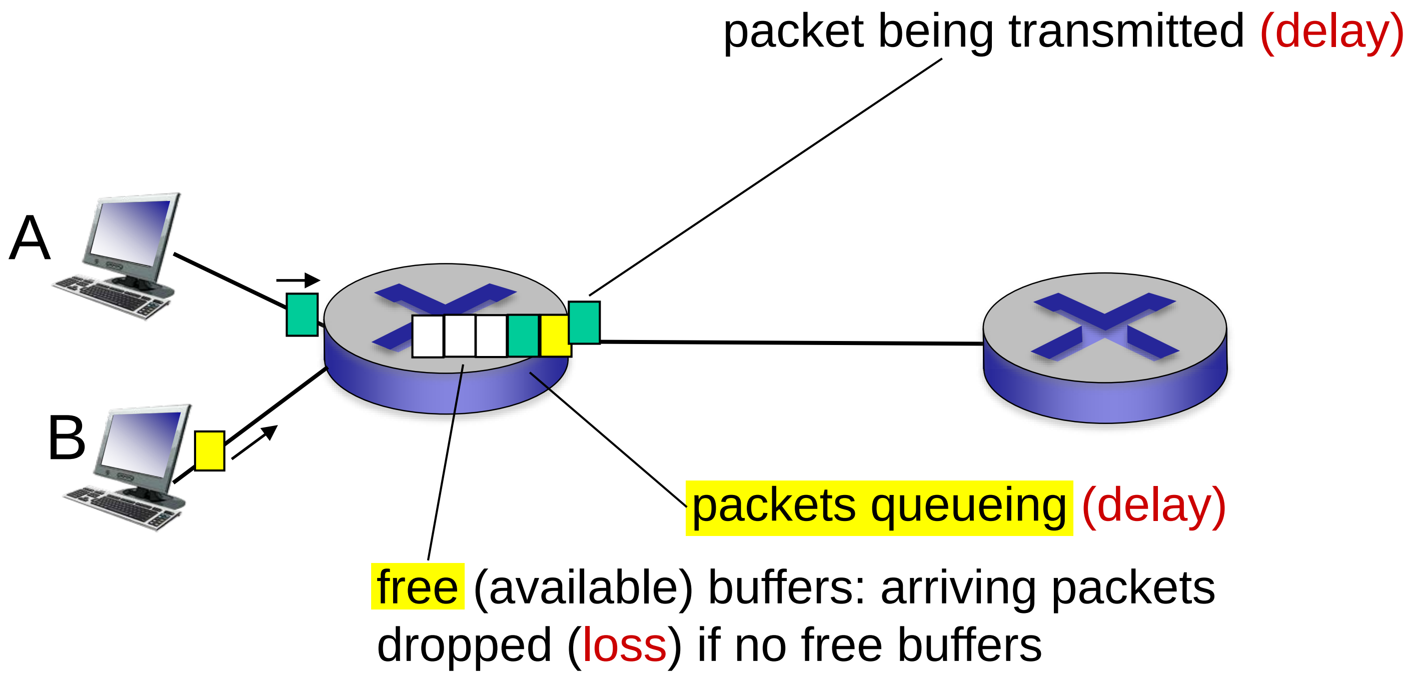
queuing and loss:

- if arrival rate (in bits) to link exceeds transmission rate of link for a period of time:
  - packets will queue, wait to be transmitted on link
  - packets can be dropped (lost) if memory (buffer) fills up

How do loss and delay occur?

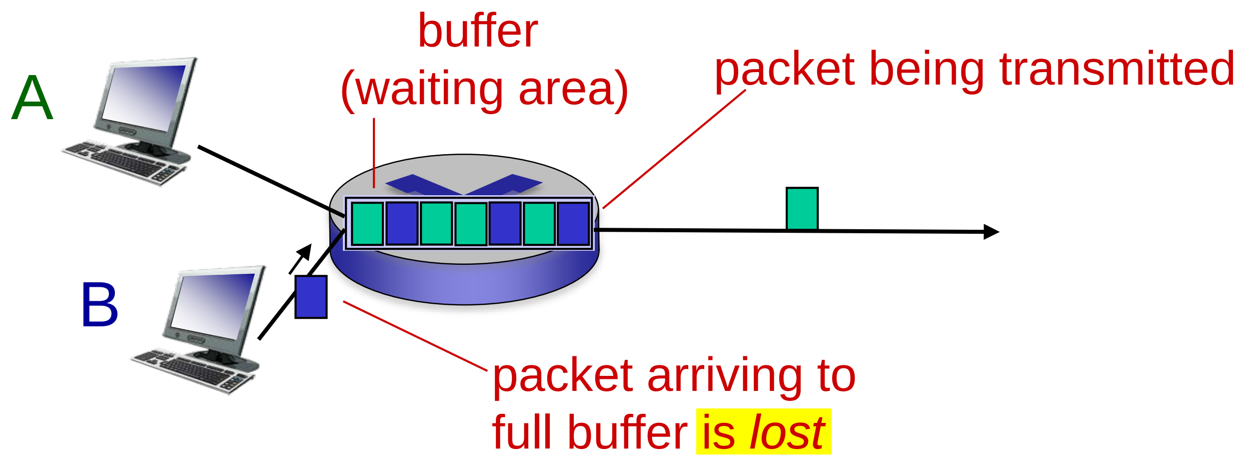
packets *queue* in router buffers

- packet arrival rate to link (temporarily) exceeds output link capacity
- packets queue, wait for turn



Packet loss

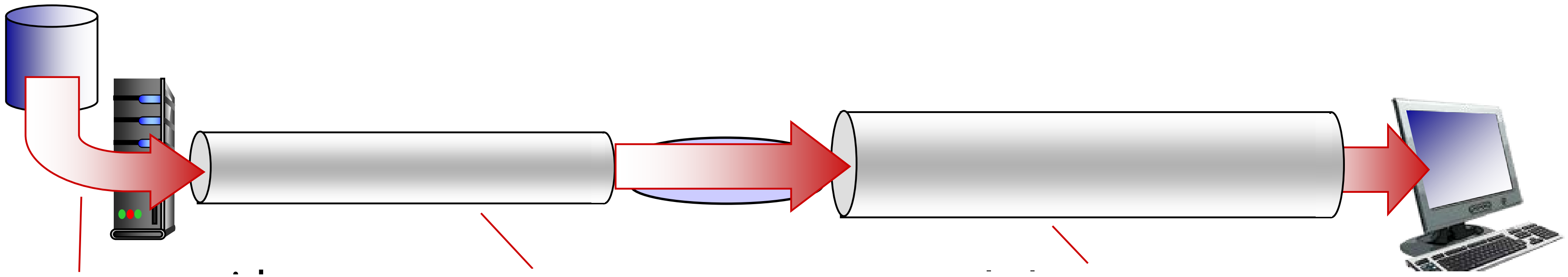
- queue (aka buffer) preceding link in buffer has finite capacity
- packet arriving to full queue dropped (aka lost)
- lost packet may be retransmitted by previous node, by source end system, or not at all



* Check out the Java applet for an interactive animation on queuing and loss

Throughput

- *throughput:* rate (bits/time unit) at which bits transferred between sender/receiver
  - *instantaneous:* rate at given point in time
  - *average:* rate over longer period of time



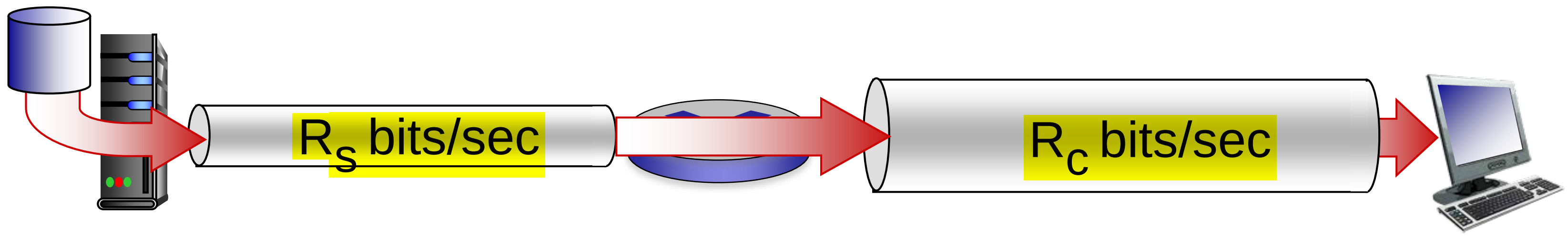
server sends bits (fluid) into pipe

pipe that can carry fluid at rate R_s bits/sec)

pipe that can carry fluid at rate R_c bits/sec)

Throughput (more)

- $R_s < R_c$ What is average end-end throughput?



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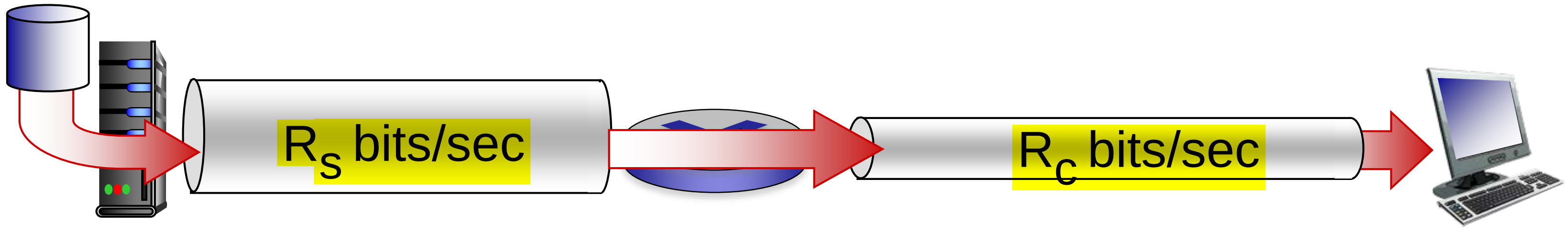
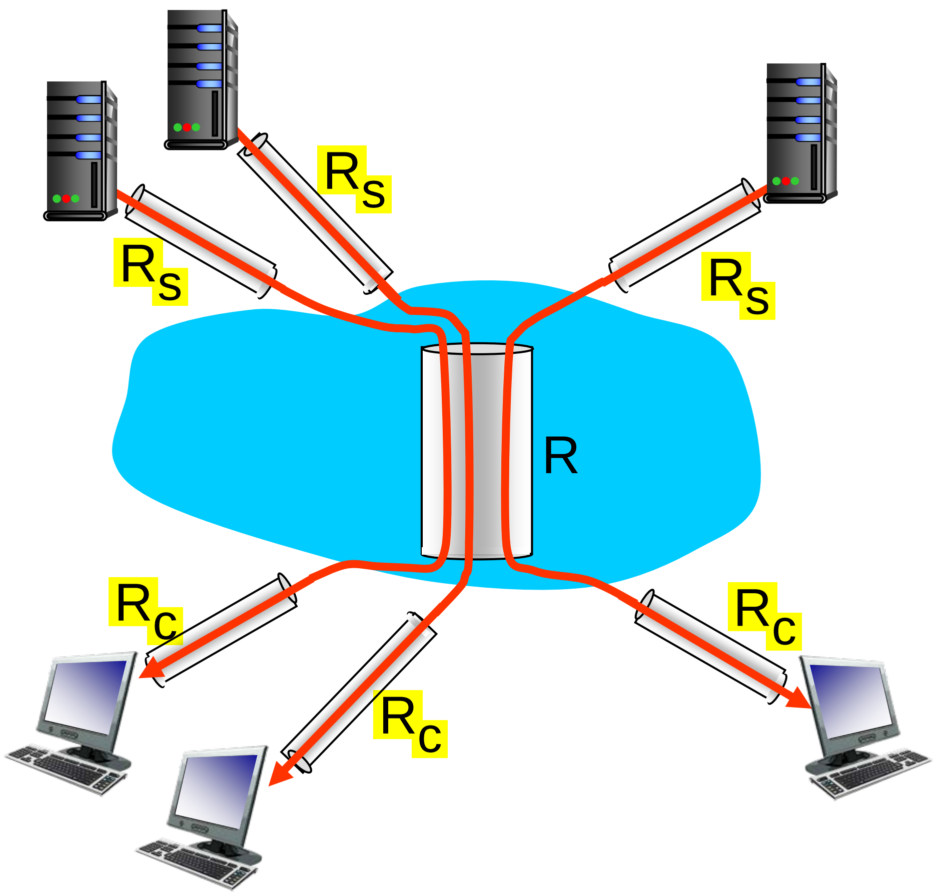


bottleneck link

link on end-end path that constrains end-end throughput

Throughput: Internet scenario

- per-connection end-end throughput: $min(R_c,R_s,R/10)$
- in practice: R_c or R_s is often bottleneck



10 connections (fairly) share backbone bottleneck link *R* bits/sec

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

Chapter 3
Transport Layer

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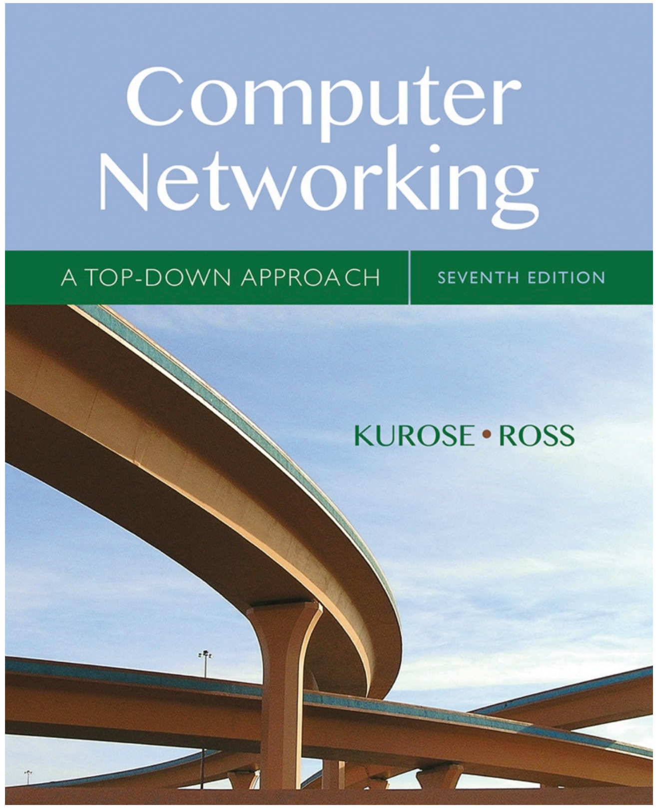

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Chapter 3 outline

3.1 transport-layer services

3.2 multiplexing and demultiplexing

3.3 connectionless transport: UDP

3.4 principles of reliable data transfer

3.5 connection-oriented transport: TCP
- segment structure
- reliable data transfer
- flow control
- connection management

3.6 principles of congestion control

3.7 TCP congestion control

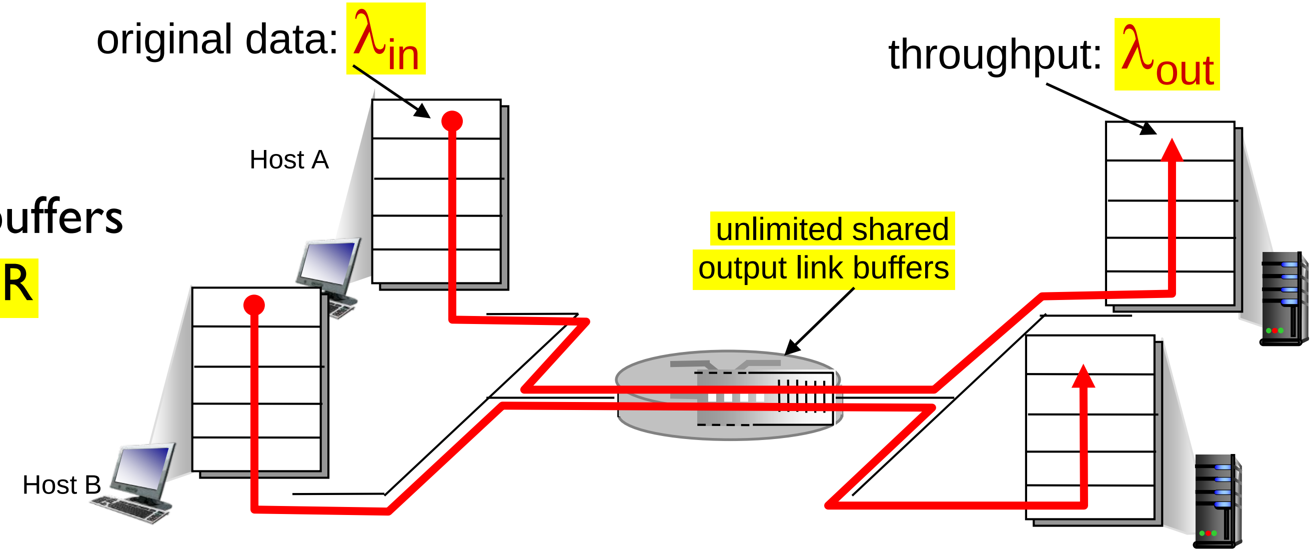
Principles of congestion control

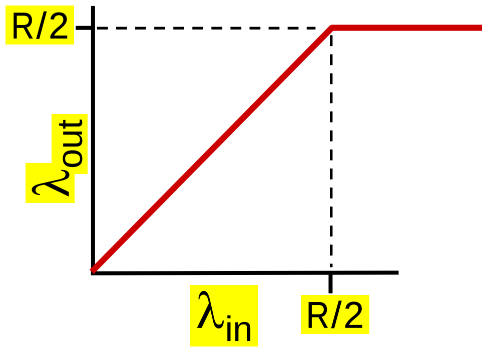
congestion:

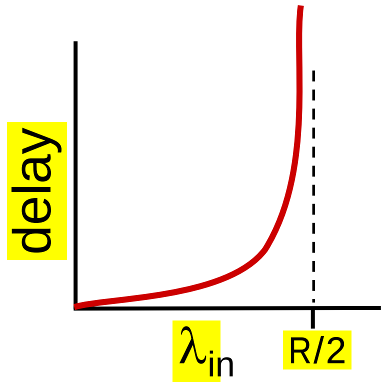
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- output link capacity: R
- no retransmission





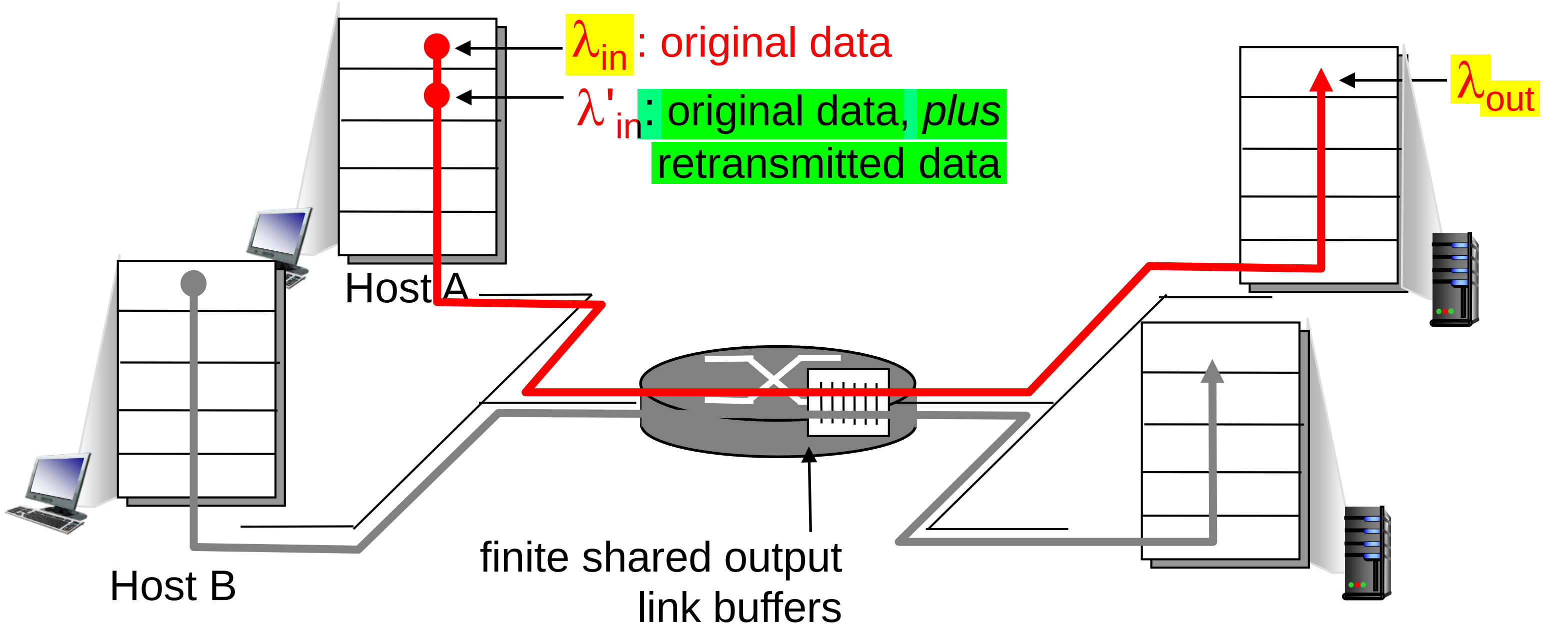


- maximum per-connection throughput: R/2

- large delays as arrival rate, λ_{in}, approaches capacity

Causes/costs of congestion: scenario 2

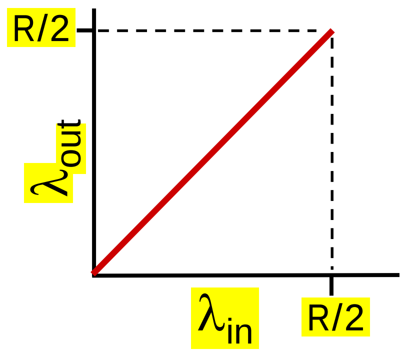
- one router, *finite* buffers
- sender retransmission of timed-out packet
  - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
  - transport-layer input includes *retransmissions* : $\lambda'_{in} \geq \lambda_{in}$

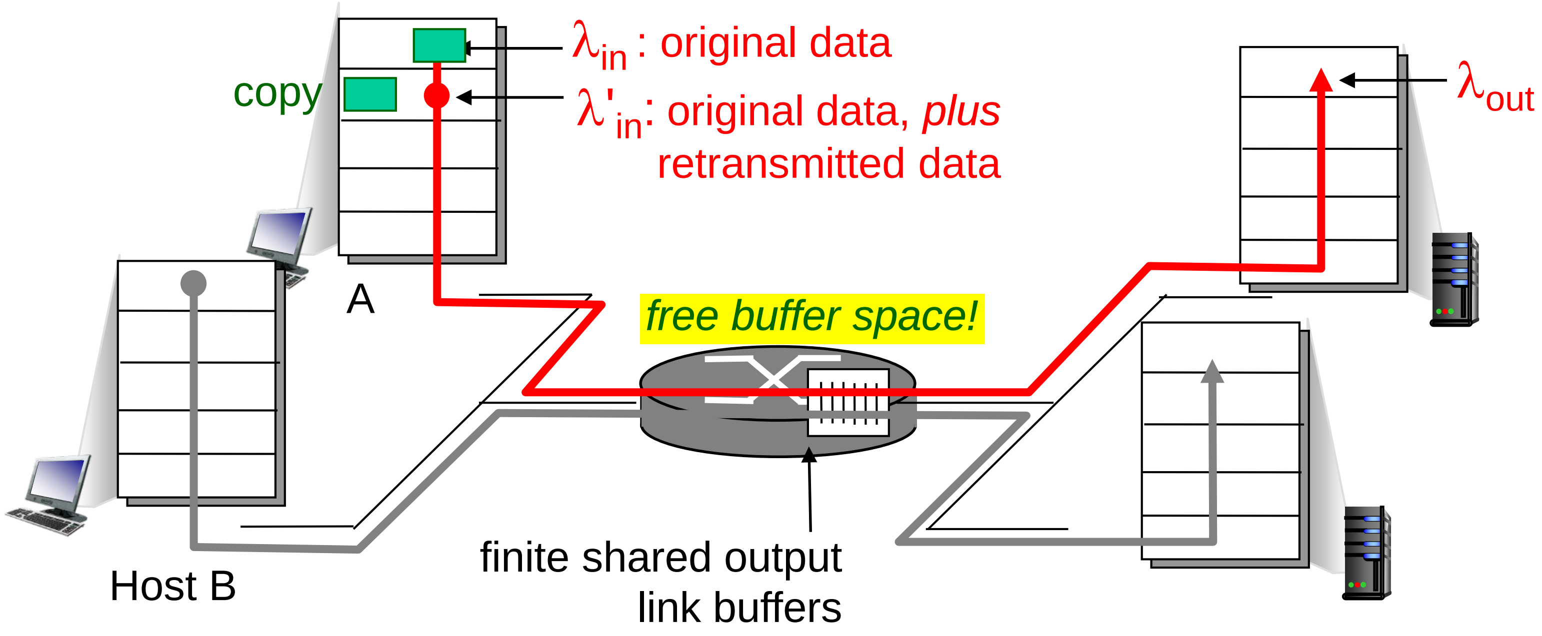


Causes/costs of congestion: scenario 2

idealization: perfect knowledge

- sender sends only when router buffers available

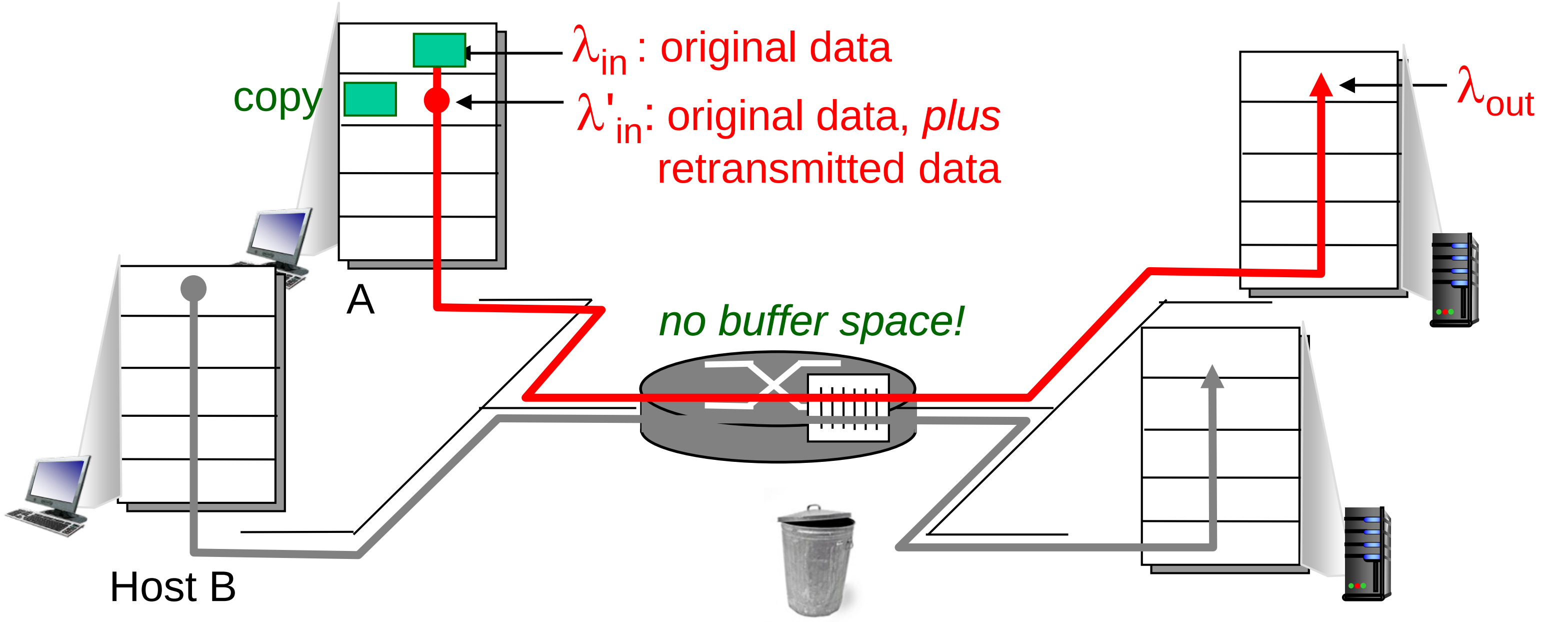




Causes/costs of congestion: scenario 2

Idealization: known loss
packets can be lost, dropped at router due to full buffers

- sender only resends if packet *known* to be lost



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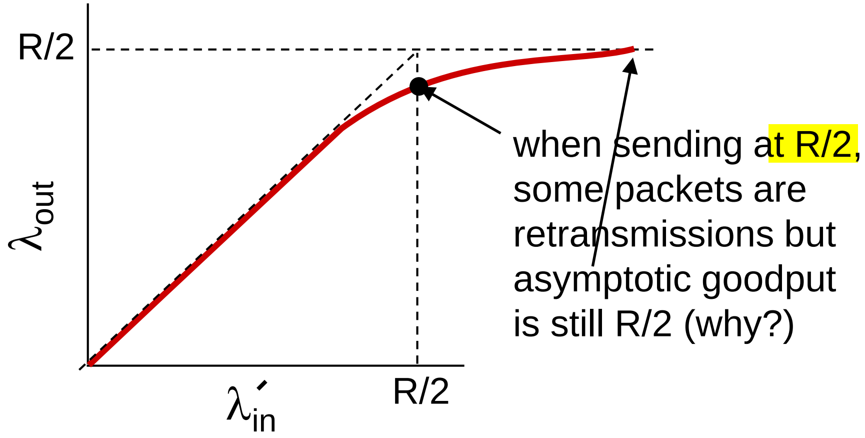
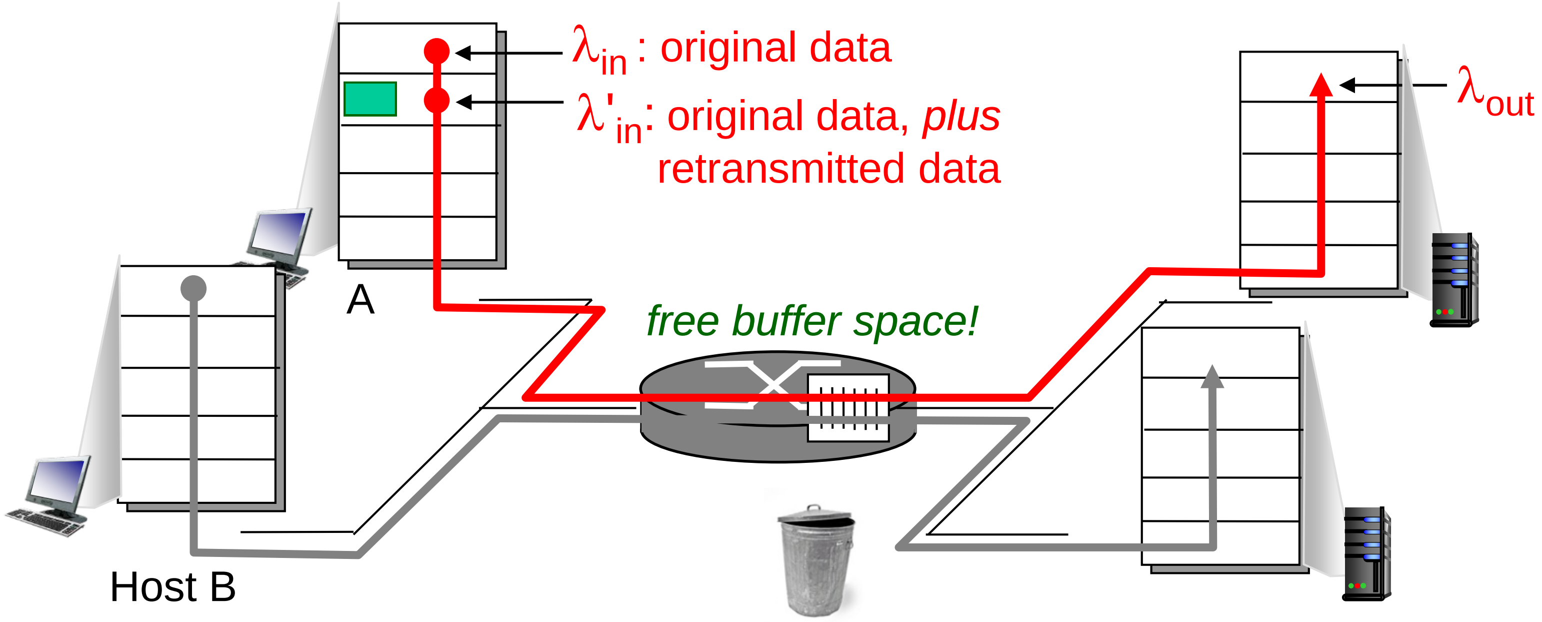
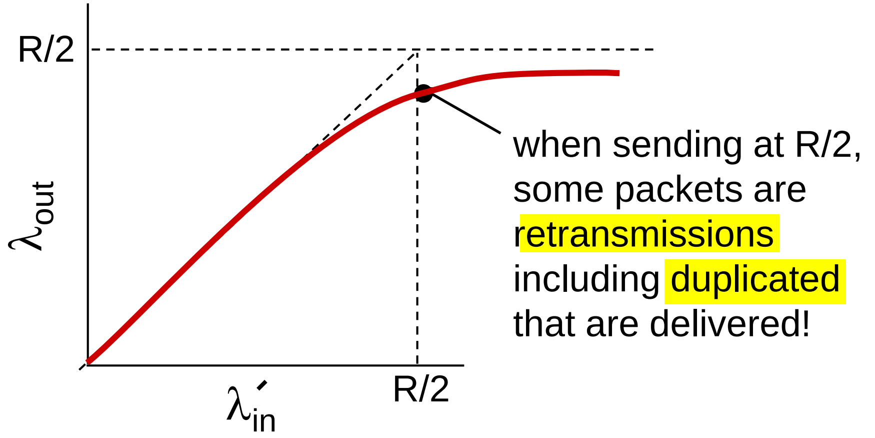


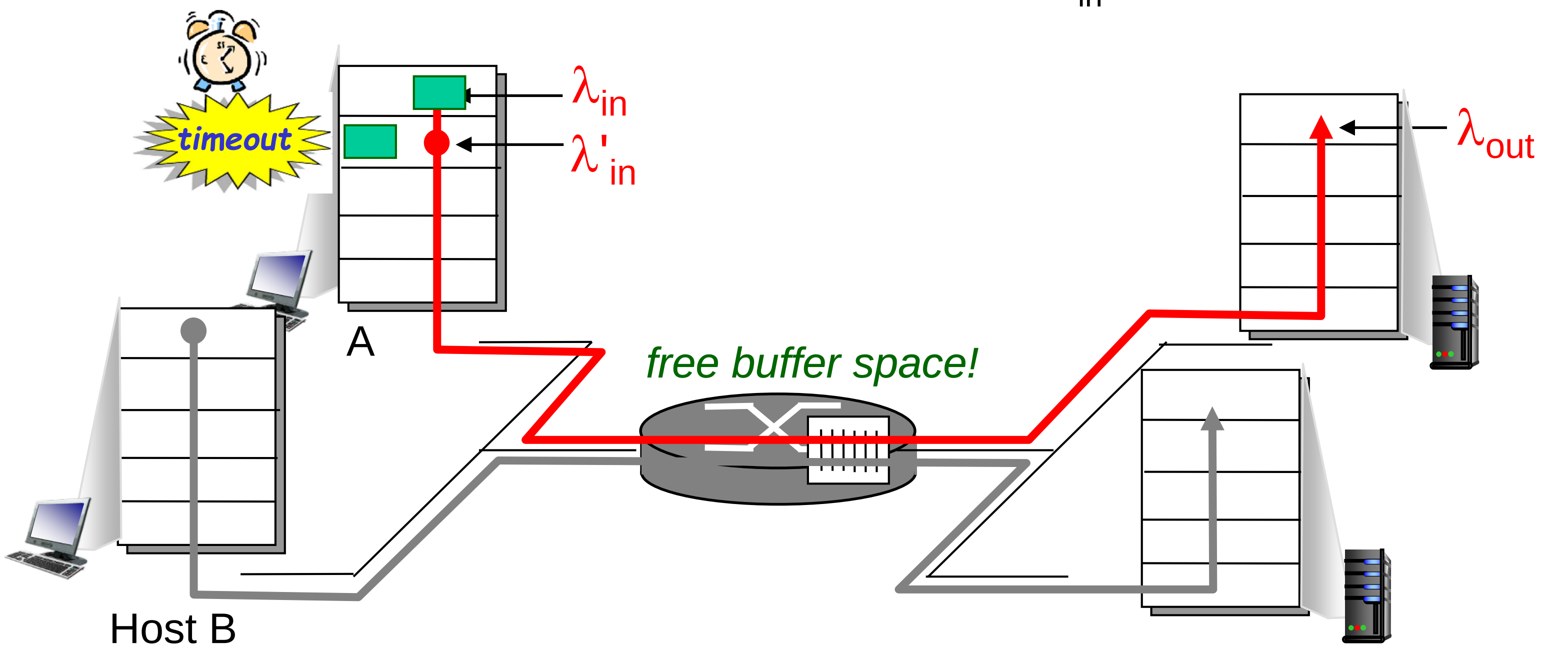


Causes/costs of congestion: scenario 2

Realistic: *duplicates*

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





Causes/costs of congestion: scenario 2

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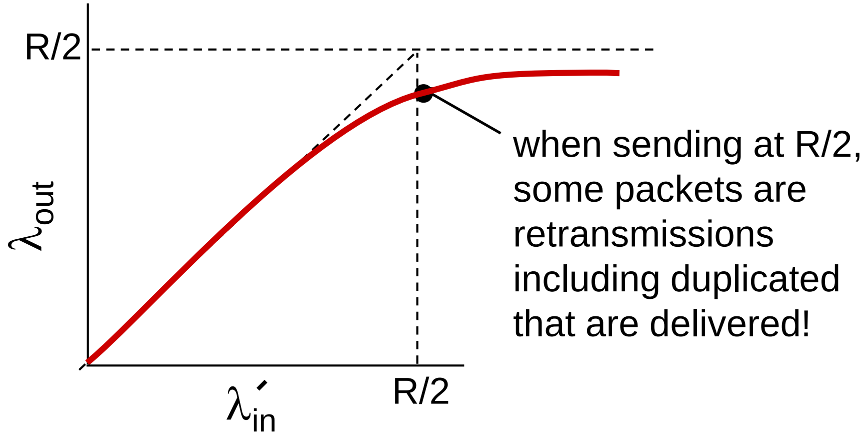


"costs" of congestion:

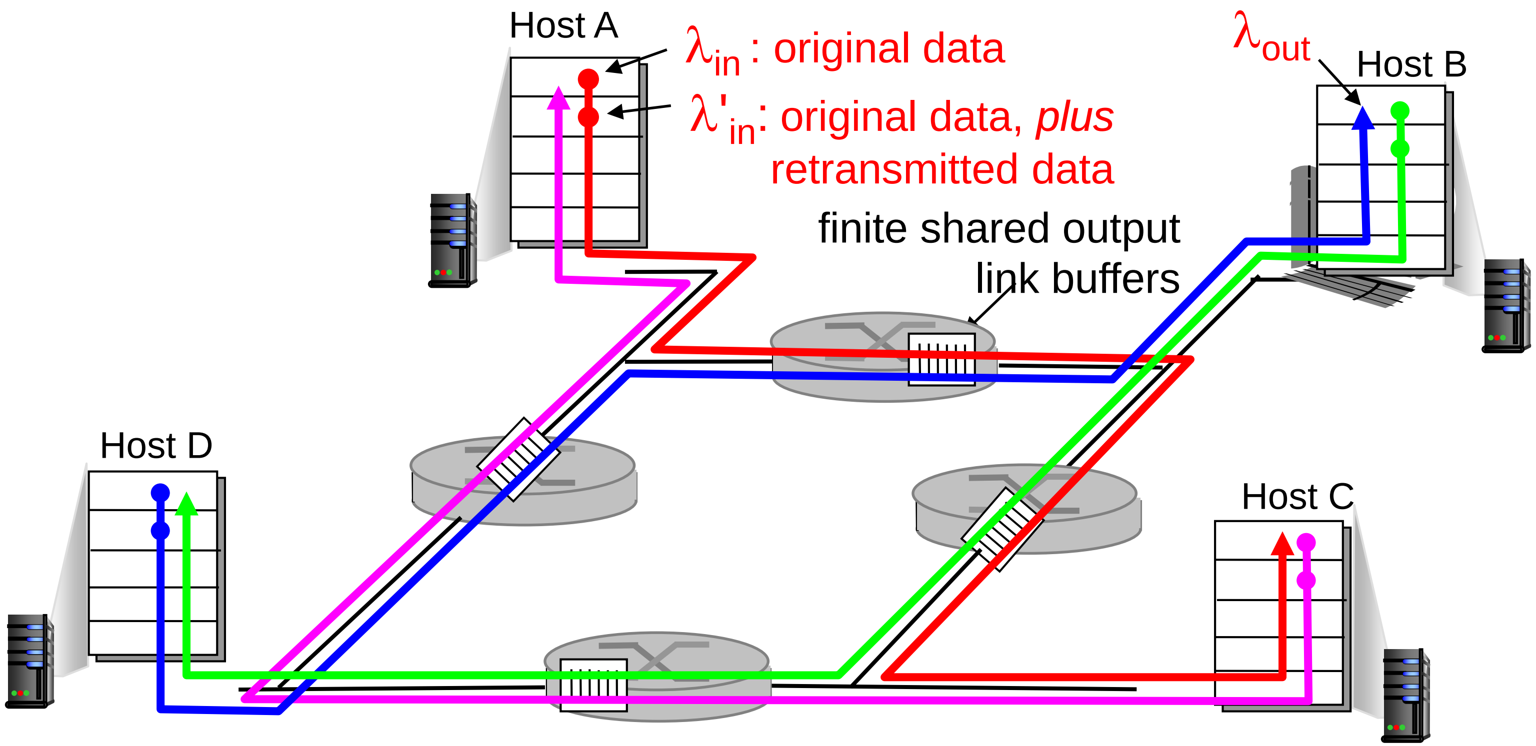
- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt
  - decreasing goodput

Causes/costs of congestion: scenario 3

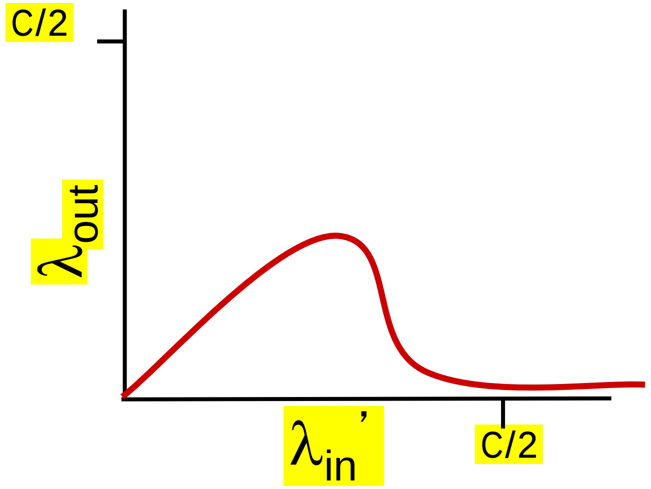
- four senders
- multihop paths
- timeout/retransmit

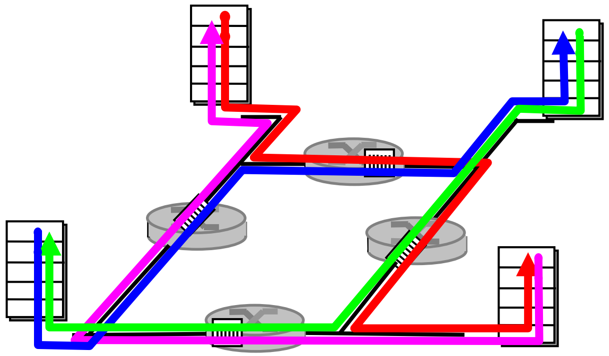
Q: what happens as λ_{in} and λ_{in}' increase ?

A: as red λ_{in}' increases, all arriving blue pkts at upper queue are dropped, blue throughput → 0



Causes/costs of congestion: scenario 3





another "cost" of congestion:

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

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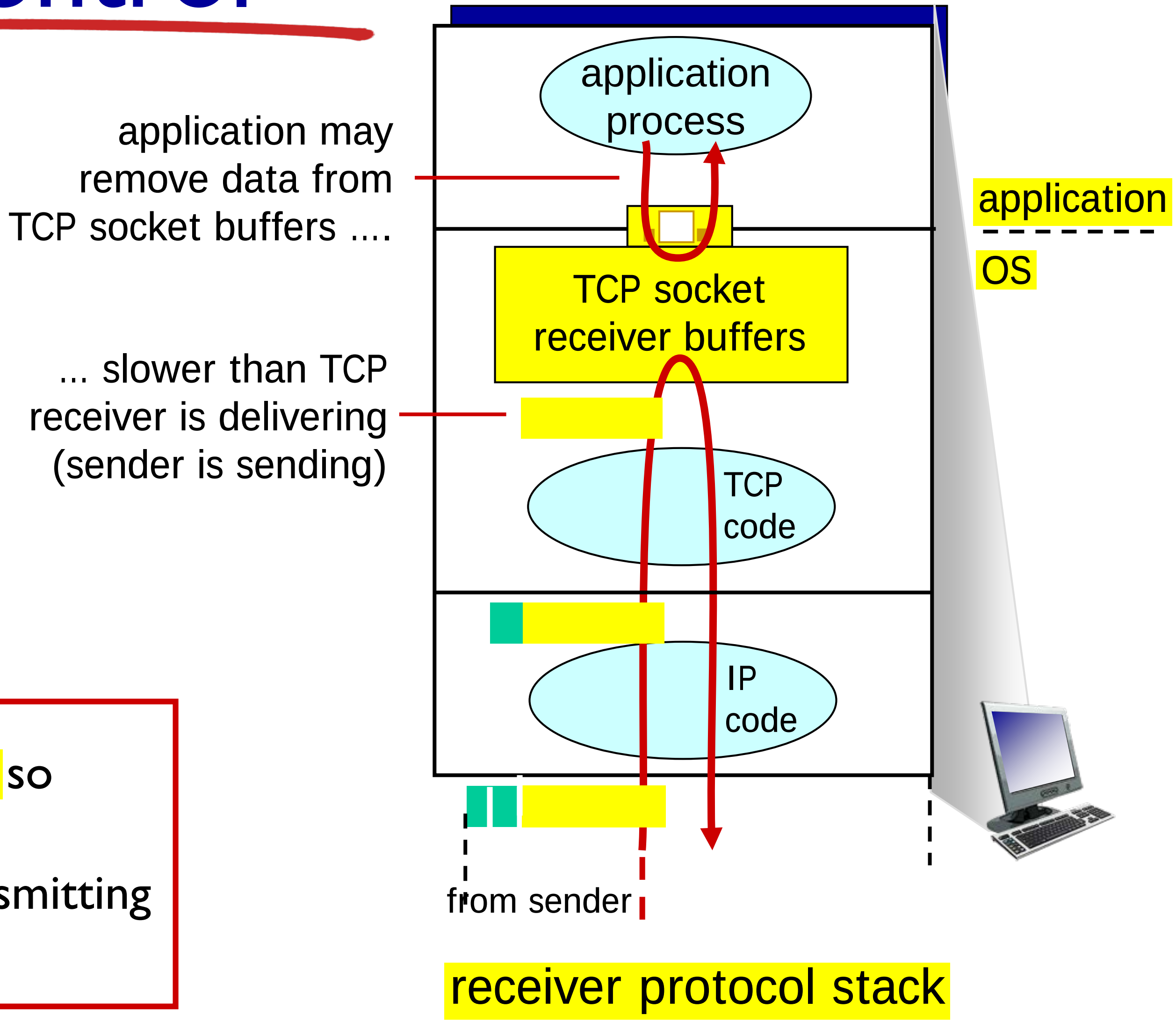
TCP flow control

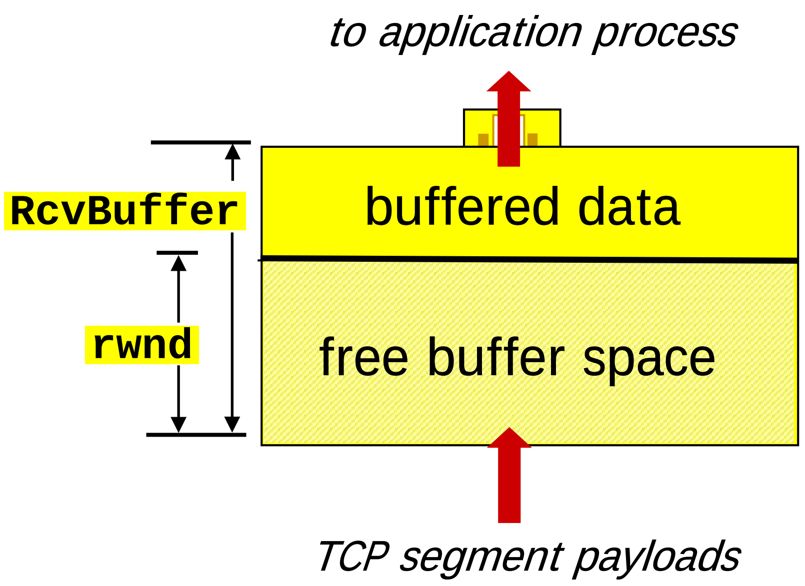


flow control

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 by including **rwnd** value in TCP header of receiver-to-sender segments
  - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **RcvBuffer**
- sender limits amount of unacked ("in-flight") data to receiver's **rwnd** value
- guarantees receive buffer will not overflow



receiver-side buffering