Network Performance

Objectives: What are the basic aspects (delay, throughput, loss) of end-to-end network performance? Why do they matter for different applications? How are they defined? Introduction of network tools: ping(8) and traceroute(8).

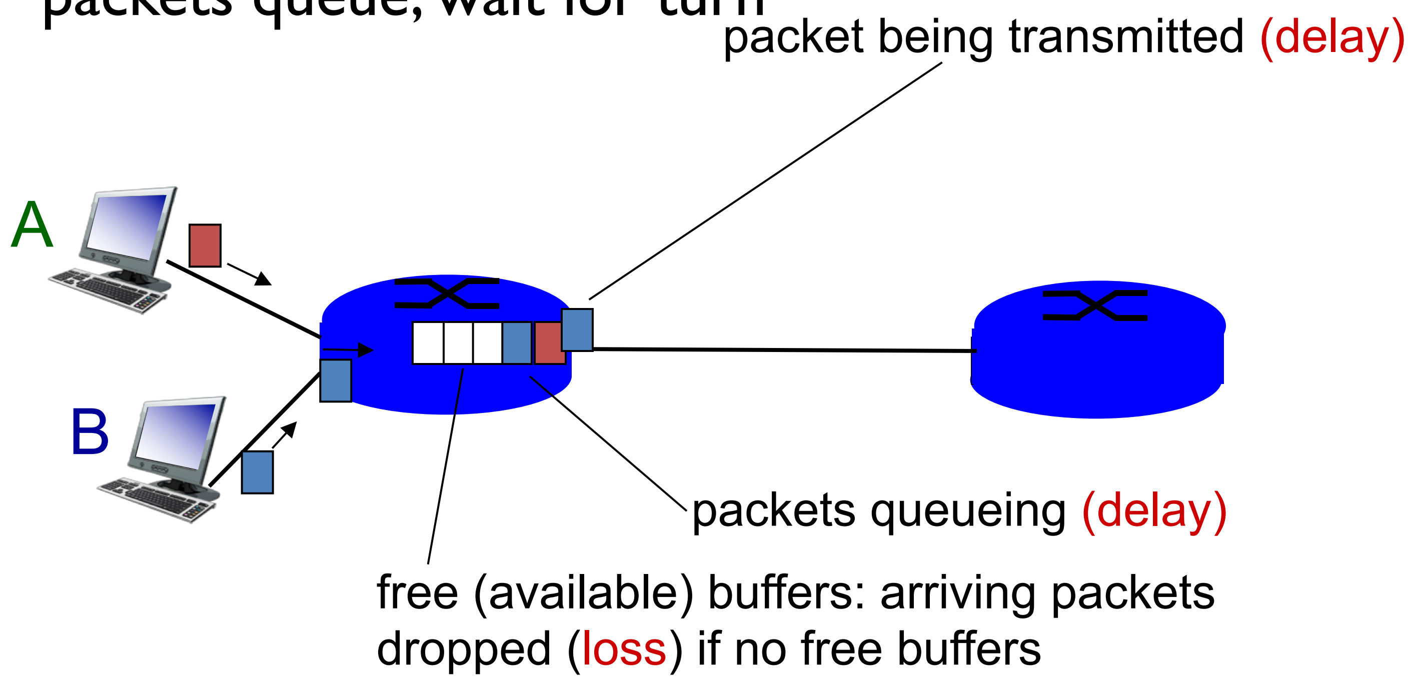
NS2: March, 2019

Textbook (K&R): Section 1.4

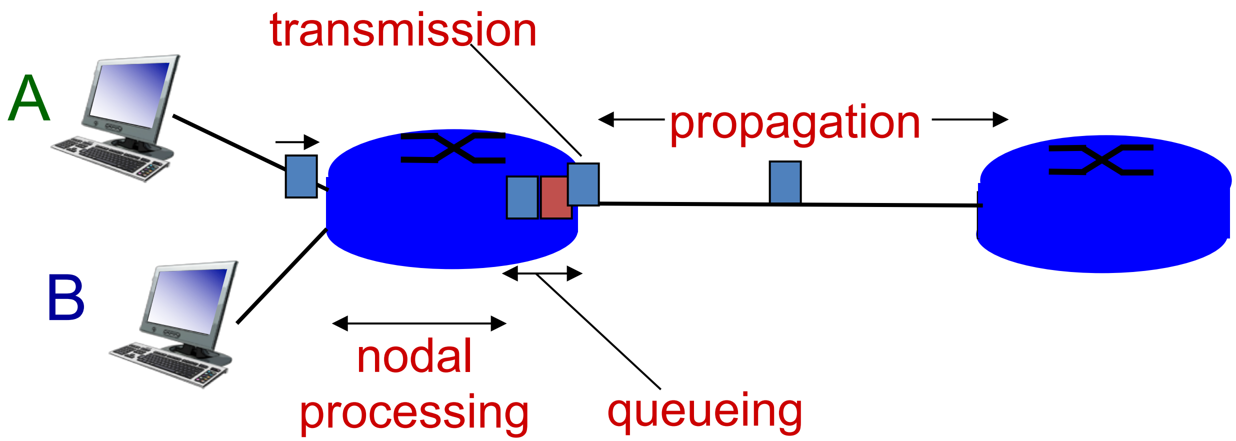
What happens when packet arrives at router?

packets *queue* in router buffers

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



Four sources of packet delay



Single "hop" nodal delay (i.e., delay from one node to *immediate* next one):

$$d_{nodal} = d_{proc} + d_{queue} + d_{trans} + d_{prop}$$

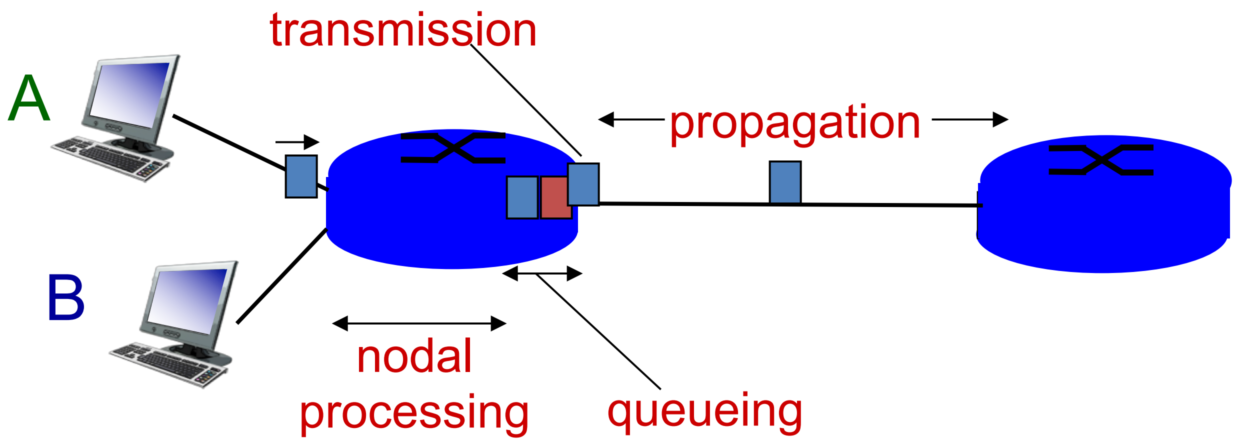
d_{proc}: nodal processing

- check for bit errors (by checksum in packet header)
- determine output link (by destination IP address in packet header)
- typically < msec

d_{queue}: queueing delay

- waiting time for packet to get to front of the queue for the output link
- depends on congestion level of router (i.e., how much *other* users are also sending data)

Four sources of packet delay



$$d_{nodal} = d_{proc} + d_{queue} + d_{trans} + d_{prop}$$

d_{trans}: transmission delay:

- L: packet length (bits)
- R: link *bandwidth (bps)*
- $d_{trans} = L/R$

d_{trans} and d_{prop}
very different

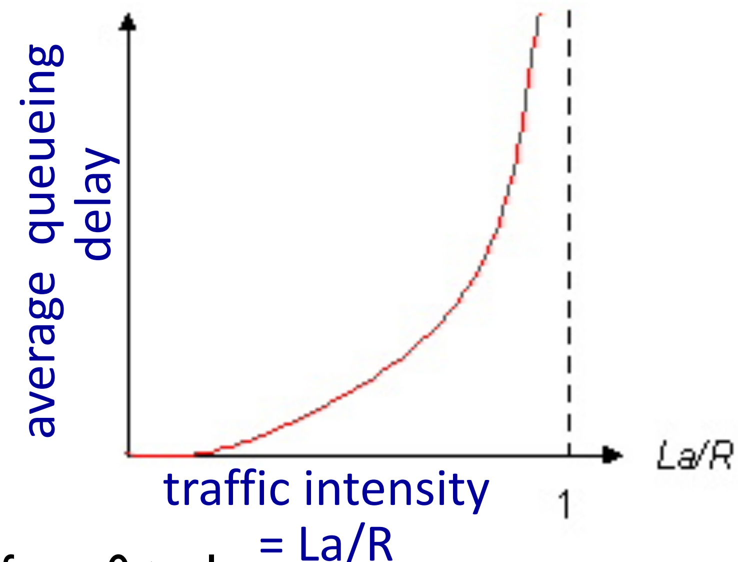
d_{prop}: propagation delay:

- d: length of physical link
- s: propagation speed in medium (~2x10^8 m/sec – 2/3 speed of light in vacuum)
- $d_{prop} = d/s$

1. Transmission delay: time to push whole packet (all the bits) from router to (beginning of) link – how quickly we can do this depends on the link technology, specifically its bndwidth (e.g., Ethernet has 10 Mbps bandwidth)
2. Propagation delay: time for packet to move from beginning to end of the link

Queueing delay (revisited)

- *R:* link bandwidth (bps)
- *L:* packet length (bits)
- a: average packet arrival rate



- *La*/*R* is also average link *utilization*, ranges from 0 to 1
- *La*/*R* ~ 0: avg. queueing delay small
- *La*/*R* -> 1: avg. queueing delay large
- *La*/*R* > 1: more “work” arriving than can be serviced, average delay infinite!

La/R ~ 0

La/R -> 1

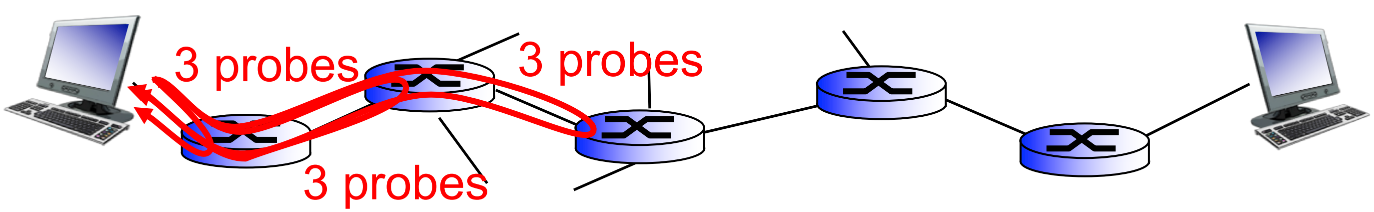
Note that queueing delay is *convex* increasing function of utilization

This assumes random bursty packet arrivals (natural for internet); traffic arrivals in special cases may have different results

You should not be too greedy and try to use every bit of the link capacity (e.g., 0.95 utilization) – why?

"Real" Internet delays and routes

- What do "real" Internet delay & loss look like?
  - "End-to-end" delay (i.e., from source to destination) has *multiple* hops – how to compose per-hop nodal delays into end-to-end delay?
- `Traceroute` program: provides delay measurement from source to router along end-to-end Internet path towards destination. For all *i:*
  - sends three packets that will reach router *i* on path towards destination
  - router *i* will return packets to sender
  - sender times interval between transmission and reply
- You will find out how traceroute works in NS Lab 1



Each probe gives one measurement of RTT (i.e., *round trip time* – a basic quantity that we'll use a lot) from the source to the router in question

Ping: Hi, are you there?

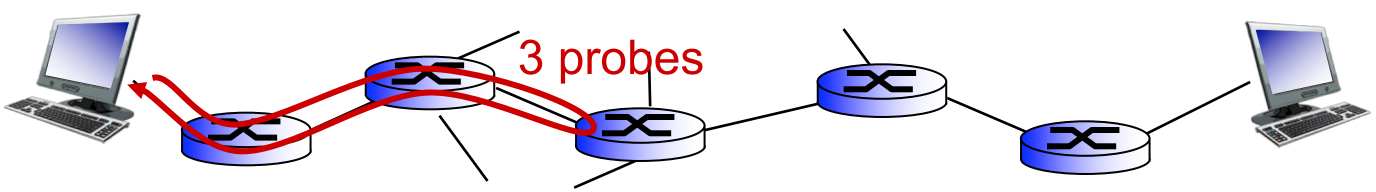
- Ping: another useful program besides traceroute
- % ping aranjuez.cs.purdue.edu
- Has answer (target is alive): Yes, I'm here!
- No answer ≠ I'm dead
  - Why? Because it's based on ICMP! (see Slide 10)
  - % ping www.microsoft.com
- For more details, as always
  - % man ping
  - note arguments + command line options
  - You'll try it out too in NS Lab 1

Traceroute and ICMP

- source sends series of UDP segments to destination
  - first set (of 3 packets) has TTL =1
  - second set has TTL=2, etc.
  - unlikely port number
- when *n*th set of packets arrives to *n*th router:
  - router discards datagrams
  - and sends source ICMP messages (type 11, code 0)
  - ICMP messages includes name of router & IP address
- when ICMP messages arrives, source records RTTs

stopping criteria:

- UDP segment eventually arrives at destination host
- destination returns ICMP "port unreachable" message (type 3, code 3)
- source stops



NB: TTL is *time-to-live* field in IP packet header; it defines maximum number of times the packet can be forwarded before the packet is dropped (considered not deliverable)
Each router decrements TTL – if decremented TTL still +ve, packet is forwarded; otherwise packet is dropped

"Real" Internet delays, routes

traceroute: gaia.cs.umass.edu (USA) to www.eurecom.fr (France)

3 delay measurements from gaia.cs.umass.edu to cs-gw.cs.umass.edu

```
1  cs-gw (128.119.240.254)  1 ms  1 ms  2 ms
2  border1-rt-fa5-1-0.gw.umass.edu (128.119.3.145)  1 ms  1 ms  2 ms
3  cht-vbns.gw.umass.edu (128.119.3.130)  6 ms 5 ms 5 ms
4  jn1-at1-0-0-19.wor.vbns.net (204.147.132.129)  16 ms 11 ms 13 ms
5  jn1-so7-0-0-0.wae.vbns.net (204.147.136.136)  21 ms 18 ms 18 ms
6  abilene-vbns.abilene.ucaid.edu (198.32.11.9)  22 ms  18 ms  22 ms
7  nycm-wash.abilene.ucaid.edu (198.32.8.46)  22 ms  22 ms  22 ms
8  62.40.103.253 (62.40.103.253)  104 ms 109 ms 106 ms
9  de2-1.de1.de.geant.net (62.40.96.129)  109 ms 102 ms 104 ms
10  de.fr1.fr.geant.net (62.40.96.50)  113 ms 121 ms 114 ms
11  renater-gw.fr1.fr.geant.net (62.40.103.54)  112 ms  114 ms  112 ms
12  nio-n2.cssi.renater.fr (193.51.206.13)  111 ms  114 ms  116 ms
13  nice.cssi.renater.fr (195.220.98.102)  123 ms  125 ms  124 ms
14  r3t2-nice.cssi.renater.fr (195.220.98.110)  126 ms  126 ms  124 ms
15  eurecom-valbonne.r3t2.ft.net (193.48.50.54)  135 ms  128 ms  133 ms
16  194.214.211.25 (194.214.211.25)  126 ms  128 ms  126 ms
17  * * *
18  * * *
19  fantasia.eurecom.fr (193.55.113.142)  132 ms  128 ms  136 ms
```

trans-oceanic link

* means no response (probe/reply lost, or router not replying)

- Do some traceroutes yourself from exotic countries at www.traceroute.org

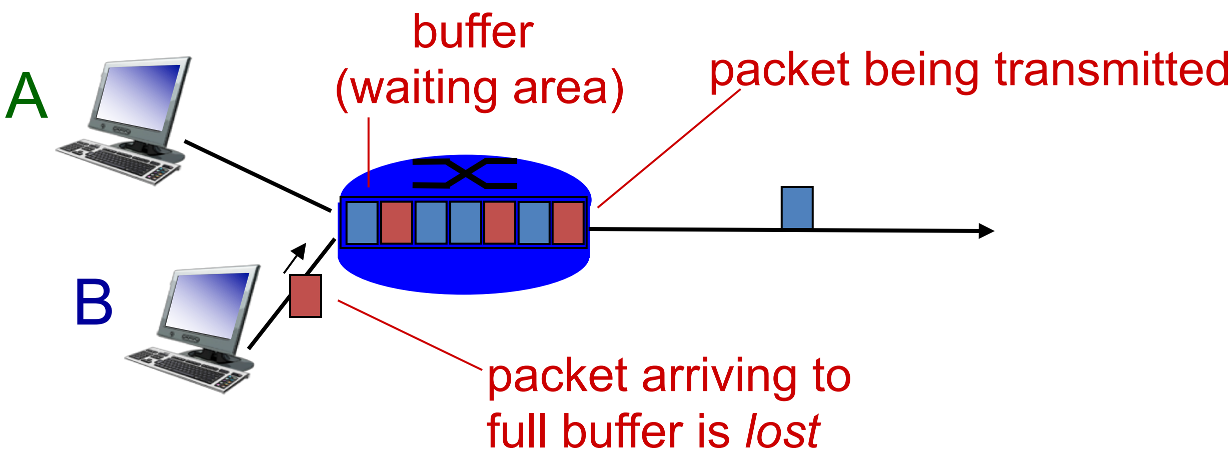
Observations: Are the 3 RTT samples for the same hop always the same? Does larger hop distance always give higher RTTs? Why or why not? How to deal with the problem?

More on ICMP

- Runs at both end hosts and routers
  - Traceroute uses (i) ICMP TTL Exceeded, (ii) ICMP port unreachable messages [port unreachable means no app is interested in the (destination port of the) packet]
  - Ping uses ICMP echo request/reply messages
- Mostly FYI: information about interesting events in the network
  - Not critical for normal operation
- Hence, not unusual for it to be
  - Disabled (hence * on previous slide)
  - Misconfigured
  - Buggy (see “man –s 8 traceroute” for examples)

Packet loss

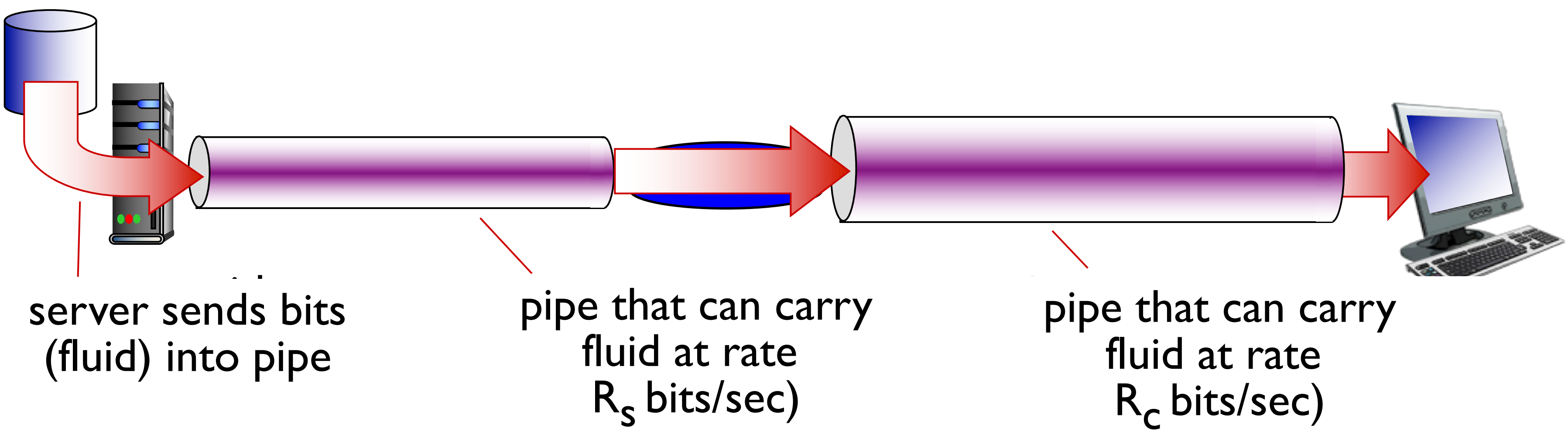
- queue (aka buffer) preceding link in buffer has finite capacity
- packet arriving to full queue dropped (aka lost)
- lost packet may be retransmitted by previous node, by source end system, or not at all



- Loss is *random* – each link has average *loss rate/probability*
- How do per-link loss rates compose into end-to-end loss rate?
- Impact of loss on applications: Email? Banking? Images? Video?

Throughput

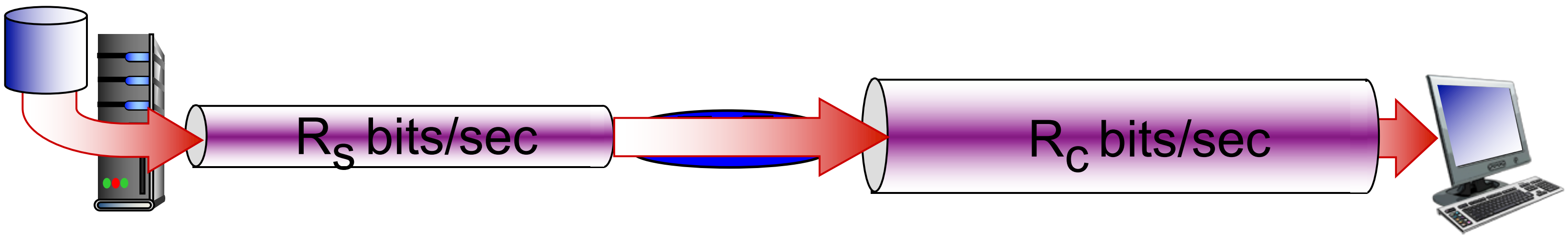
- *Throughput:* rate (bits/time unit) at which bits can be transferred between sender/receiver
  - *instantaneous:* rate at given point in time
  - *average:* rate over longer period of time
- Some link "speeds" (throughput): T1 line (1.5Mbps); Fast Ethernet (100Mb/s); T3 line (43Mbps); Gigabit Ethernet (1Gb/s); OC48 (2.5Gbps); OC192 (9.95Gbps)



NB: Not considering congestion, link throughput determined by *width* of the link (i.e., link bandwidth)

End-to-end throughput (how do per-hop throughputs compose)

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



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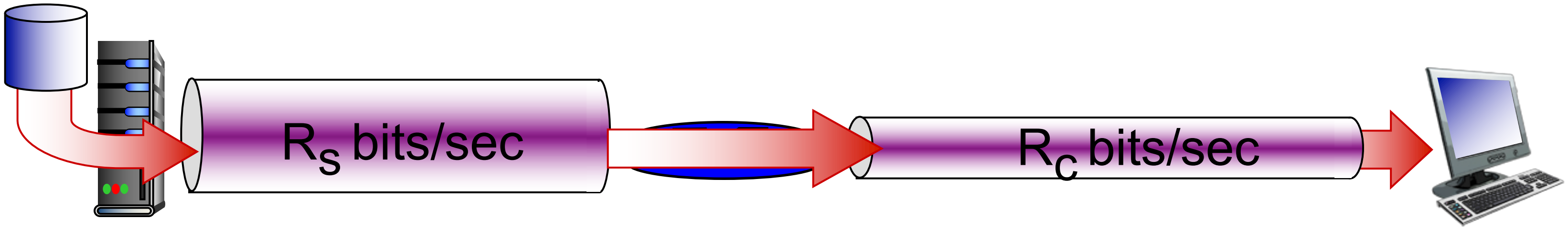


bottleneck link

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

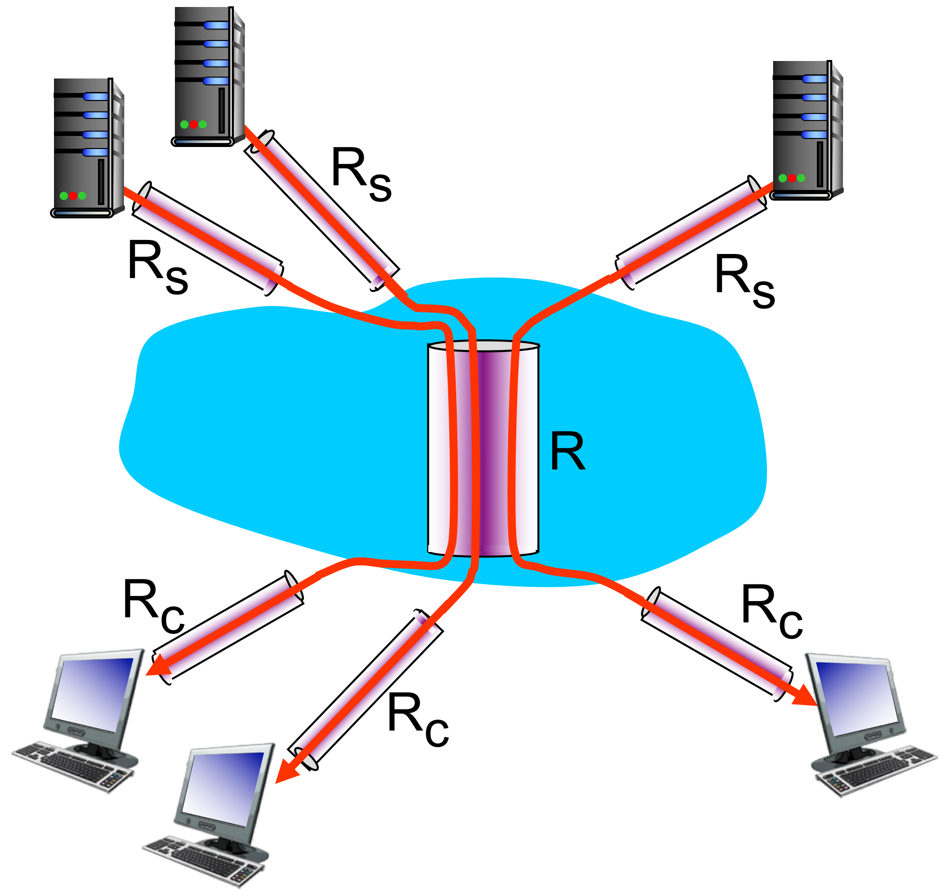
Activity 2.1

- An end-to-end network path *P* consists of three links 1, 2, 3.
- Link 1: delay 2ms, throughput 100Mb/s, loss rate: 5%; Link 2: 60ms, 1Gb/s, 10%; Link 3: 5ms, 10Mb/s, 10%
- What are *P*'s delay, throughput, and loss rate?
  - Assume that losses are independent

NB: Strictly speaking, 1k=1024, 1M=1024^2, etc. **But** in your calculations, use **1k=1000, 1M=10^6**, etc, whenever doing so simplifies your life.

Throughput: Internet scenario

- Middle link shared by 10 connections; each connection gets equal share of the throughput R
- 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

Activity 2.2

- Give an example A of an Internet application that transfers lots of data (bulk data) in one direction.
- Give an example B of an Internet application that does mainly a sequence smaller message transfers over long distances in both directions.
- Ignoring resource contention due to sharing, what performance metrics (i.e., delay, throughput, or both) mainly impact A vs. B?
- If we upgrade the network core from T3 links to OC48 links, which application (A or B) will likely benefit more?
- What will limit the performance gain for B fundamentally?
- How does loss rate interact with delay (e.g., tradeoff?)?
  - Without retransmission?
  - With retransmission?

Visualize by Space-Time Diagram

U.S. coast-to-coast transfer (5000 km) of L=10 Mbits on a link of ”speed” (throughput) R Mb/s

Gigabit Ethernet: R=1000 (i.e., 1Gb/s)
E2e transfer time: 0.035s

Fast Ethernet link: R=100 (Mb/s)
E2e transfer time: 0.125s