COMP 6461

Computer Networks & Protocols

Winter 2024

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Week 1: Lecture 2

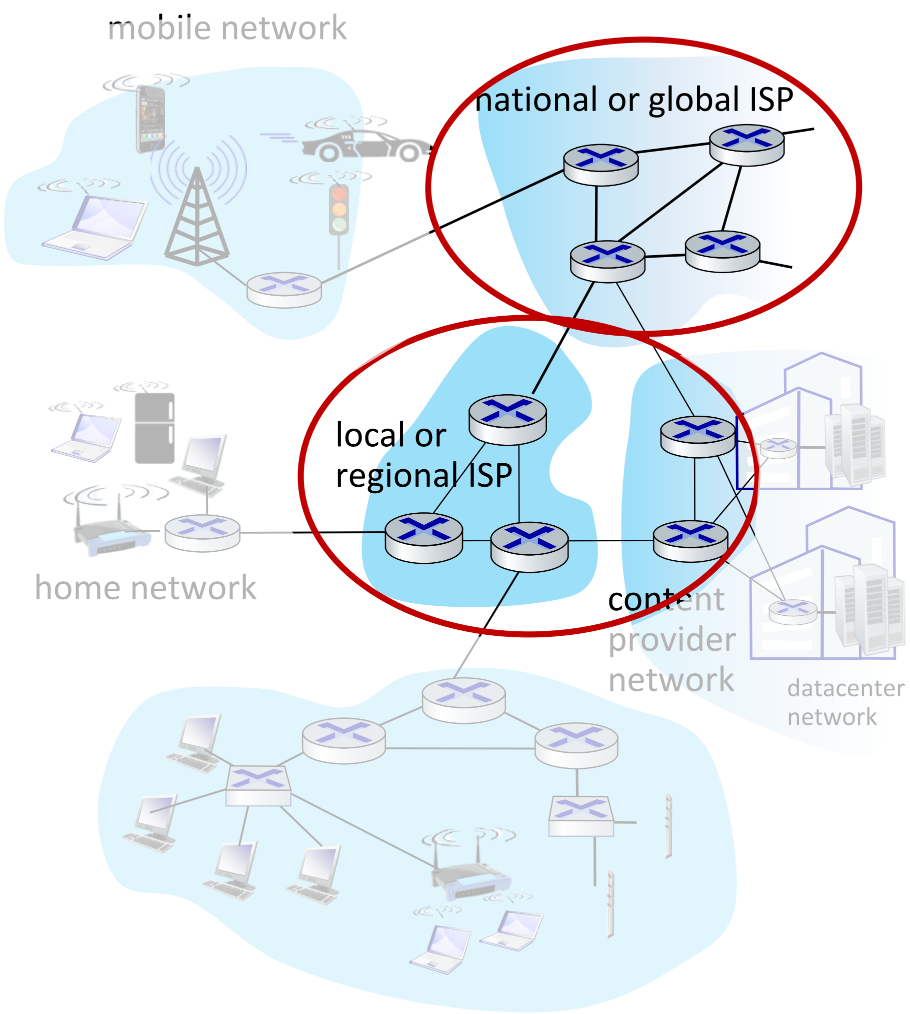
Introduction to Networking

Chapter 1: roadmap

- What *is* the Internet?
- What *is* a protocol?
- Network edge: hosts, access network, physical media
- Network core: packet/circuit switching, internet structure
- Performance: loss, delay, throughput
- Protocol layers and service models

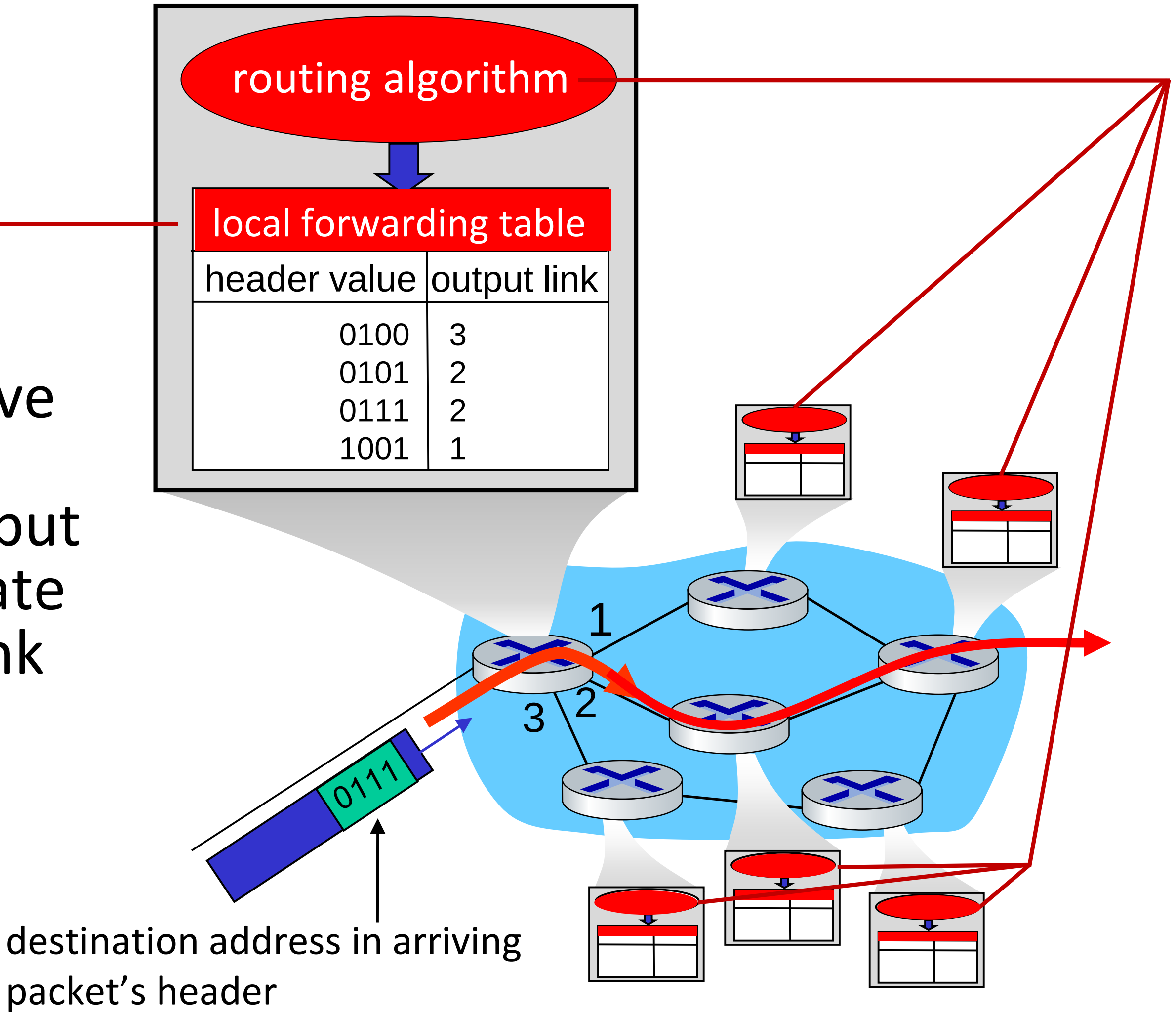
The network core

- mesh of interconnected routers
- packet-switching:
  - is a technique used in computer networks to transmit data in the form of packets, which are small units of data that are transmitted independently across the network.
  - hosts break application-layer messages into *packets*
  - Each packet contains a header, which includes information about the packet's source and destination, as well as the data payload.
  - network forwards packets from one router to the next, across links on path from source to destination



Two key network-core functions

Forwarding:
- aka "switching"
- *local* action: move arriving packets from router's input link to appropriate router output link



Routing:
- *global* action: determine source-destination paths taken by packets
- routing algorithms


45 h
3,053 miles
48 h
3,309 miles
San Jose
Northampton
United States
routing
Seattle
Portland
Sacramento
San Francisco
Las Vegas
Los Angeles
San Diego
Phoenix
Tucson
El Paso
Dallas
Austin
Houston
San Antonio
Kansas City
Minneapolis
Chicago
Detroit
Indianapolis
Nashville
Atlanta
Charlotte
New Orleans
Jacksonville
Orlando
Tampa
Toronto
Ottawa
Montreal
Quebec City
New York
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forwarding
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Las Vegas
Los Angeles
San Diego
ARIZONA
Phoenix
Tucson
El Paso
NEW MEXICO
COLORADO
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OKLAHOMA
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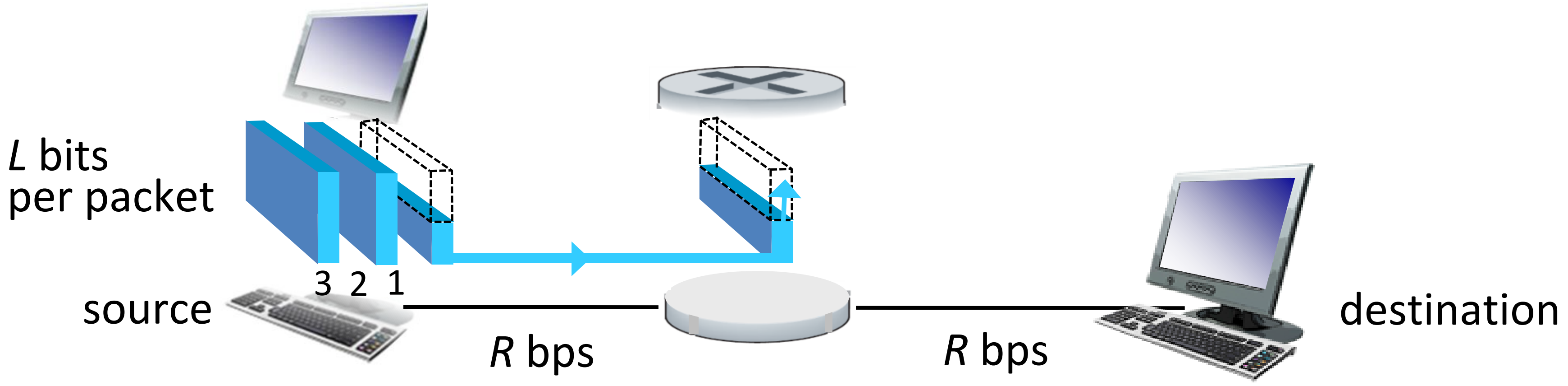
Austin
Houston
San Antonio
ARKANSAS
LOUISIANA
New Orleans
MISSISSIPPI
ALABAMA
Atlanta
TENNESSEE
Nashville
KENTUCKY
ILLINOIS
INDIANA
Indianapolis
Chicago
MICHIGAN
Detroit
OHIO
Toronto
Ottawa
Montreal
Quebec City
MAINE
VERMONT
NEW HAMPSHIRE
Northampton
New York
RI
CT
BAJA CALIFORNIA
SONORA
CHIHUAHUA
COAHUILA
Tampa
FLORIDA

Packet-switching: store-and-forward

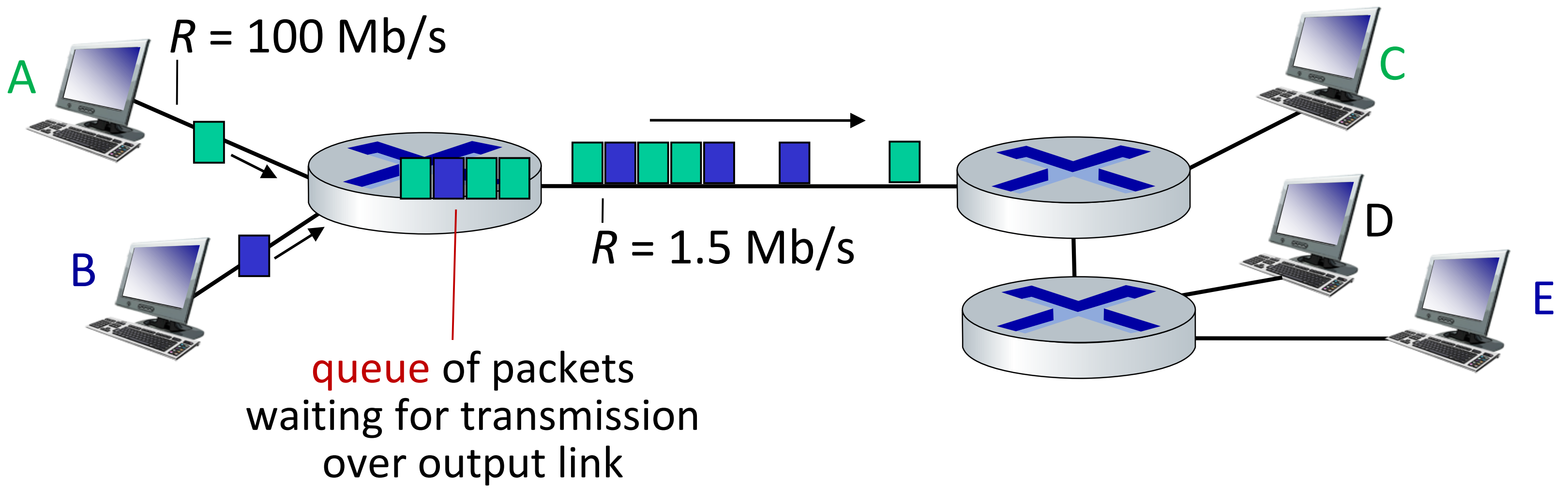


- packet transmission delay: takes *L*/*R* seconds to transmit (push out) *L*-bit packet into link at *R* bps
- *store and forward: entire* packet must arrive at router before it can be transmitted on next link

One-hop numerical example:

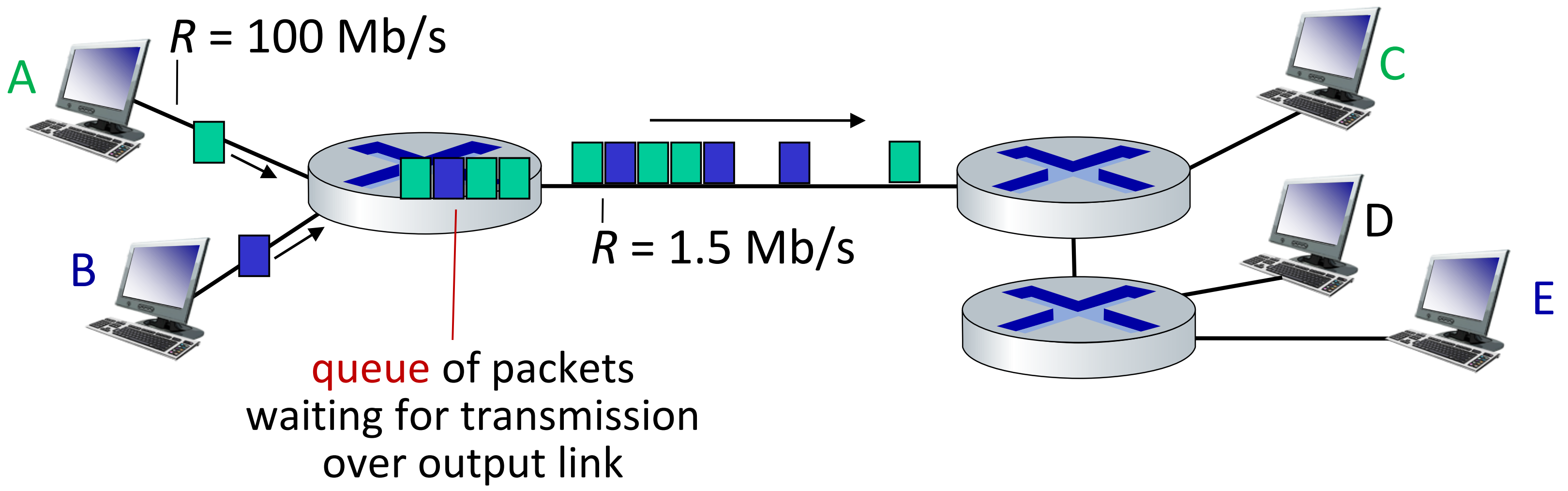
- *L* = 10 Kbits
- *R* = 100 Mbps
- one-hop transmission delay = 0.1 msec

Packet-switching: queueing



Queueing occurs when work arrives faster than it can be serviced:

Packet-switching: queueing



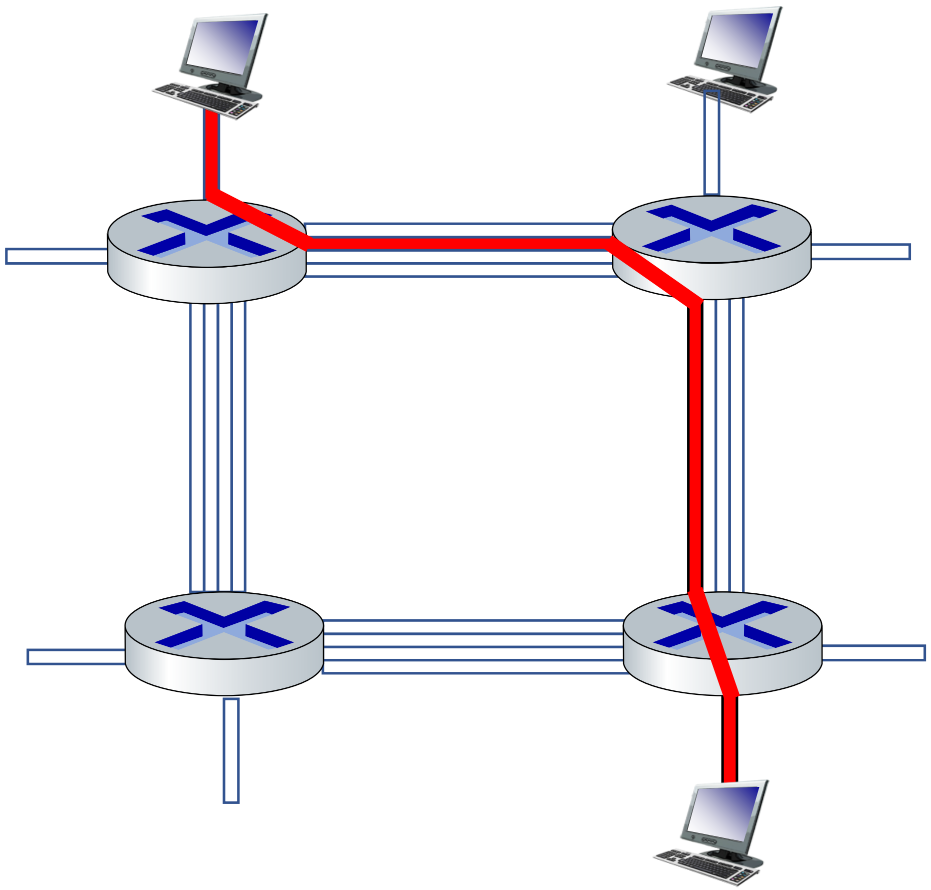
Packet queuing and loss: if arrival rate (in bps) to link exceeds transmission rate (bps) of link for some period of time:

- packets will queue, waiting to be transmitted on output link
- packets can be dropped (lost) if memory (buffer) in router fills up

Alternative to packet switching: circuit switching

end-end resources allocated to, reserved for "call" between source and destination

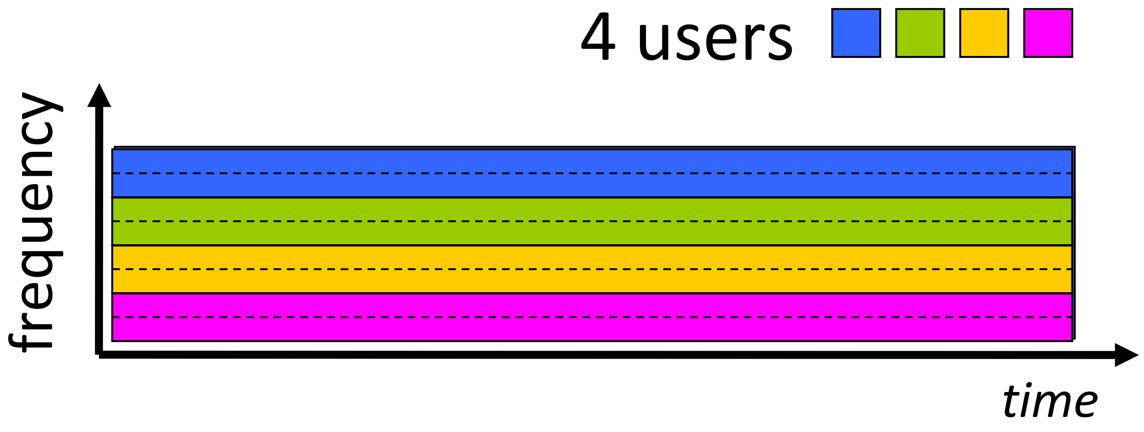
- commonly used in traditional telephone networks
- in diagram, each link has four circuits.
  - call gets 2nd circuit in top link and 1st circuit in right link.
- dedicated resources: no sharing
  - circuit-like (guaranteed) performance
- circuit segment idle if not used by call (no sharing)
- No delay and no packet loss (no congestion)



Circuit switching: FDM and TDM

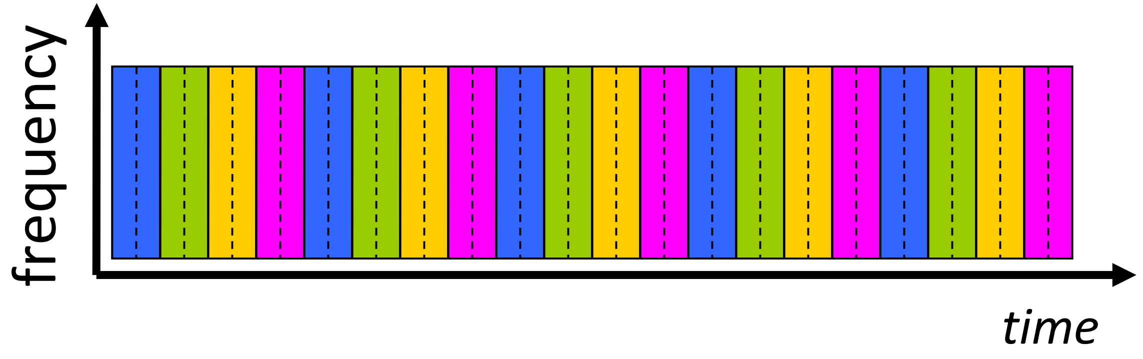
Frequency Division Multiplexing (FDM)

- optical, electromagnetic frequencies divided into (narrow) frequency bands
- each call allocated its own band, can transmit at max rate of that narrow band



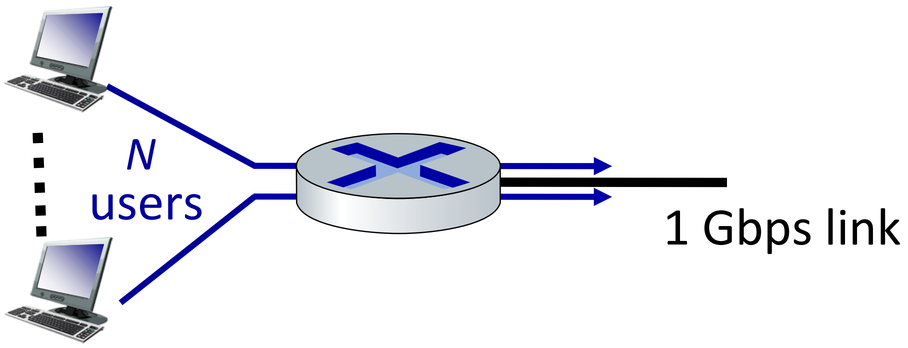
Time Division Multiplexing (TDM)

- time divided into slots
- each call allocated periodic slot(s), can transmit at maximum rate of (wider) frequency band (only) during its time slot(s)



Packet switching versus circuit switching

example:
- 1 Gb/s link
- each user:
  - 100 Mb/s when "active"
  - active 10% of time



Q: how many users can use this network under circuit-switching and packet switching?

- *circuit-switching: 10 users*

- *packet switching:* with 35 users, probability > 10 active at same time is less than .0004 *

Q: how did we get value 0.0004?

A: HW *(Hardy–Weinberg)* problem (for those with course in probability only) → *binomial probability formula*

Packet switching versus circuit switching

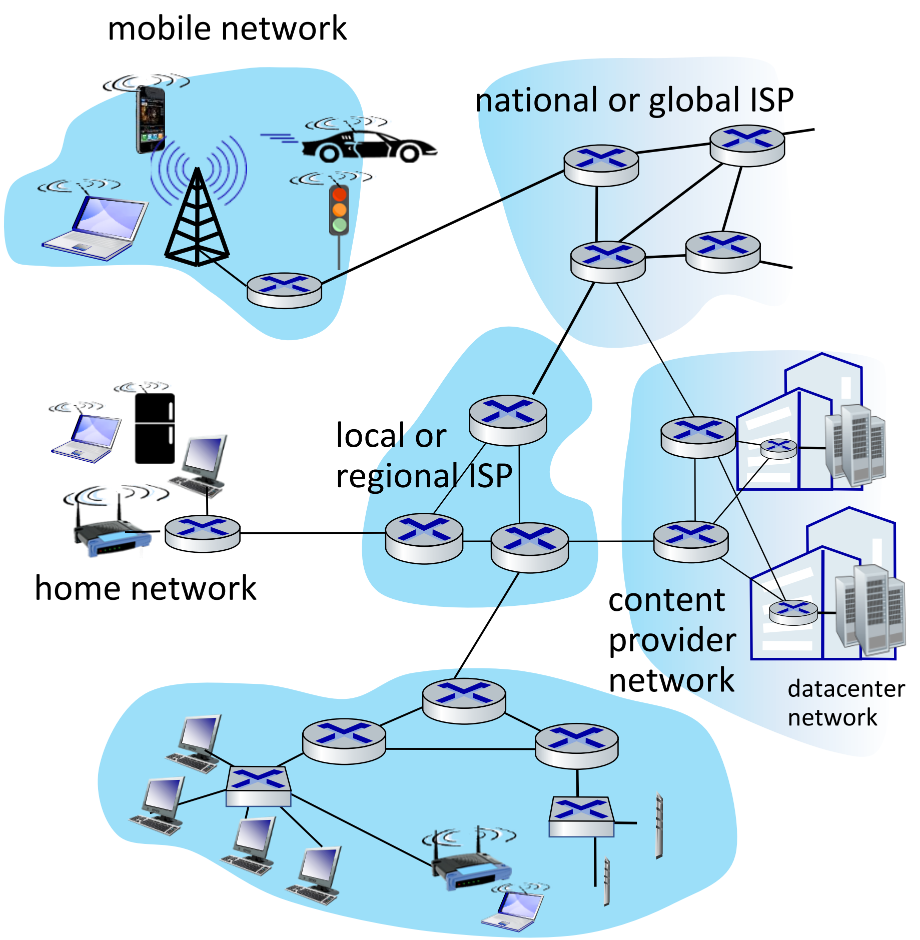
Is packet switching a "slam dunk winner"?

- great for "bursty" data – sometimes has data to send, but at other times not
  - resource sharing
  - simpler, no call setup
- excessive congestion possible: packet delay and loss due to buffer overflow
  - protocols needed for reliable data transfer, congestion control
- *Q:* How to provide circuit-like behavior with packet-switching?
  - "It's complicated." We'll study various techniques that try to make packet switching as "circuit-like" as possible.

Q: human analogies of reserved resources (circuit switching) versus on-demand allocation (packet switching)?

Internet structure: a "network of networks"

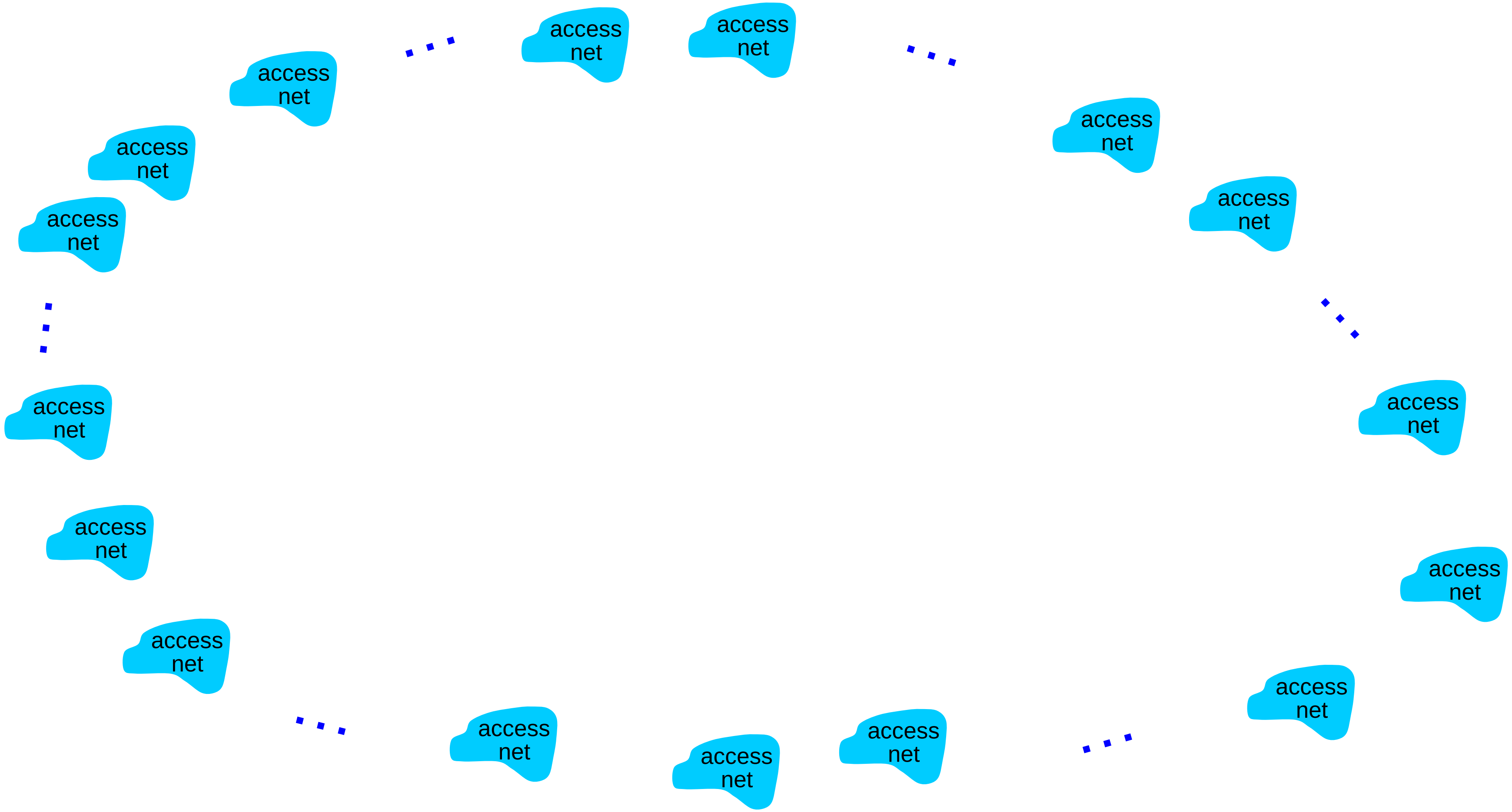
- hosts connect to Internet via access Internet Service Providers (ISPs)
- access ISPs in turn must be interconnected
  - so that *any* two hosts *(anywhere!)* can send packets to each other
- resulting network of networks is very complex
  - evolution driven by economics, national policies



Let's take a stepwise approach to describe current Internet structure

Internet structure: a "network of networks"

Question: given *millions* of access ISPs, how to connect them together?



Internet structure: a "network of networks"

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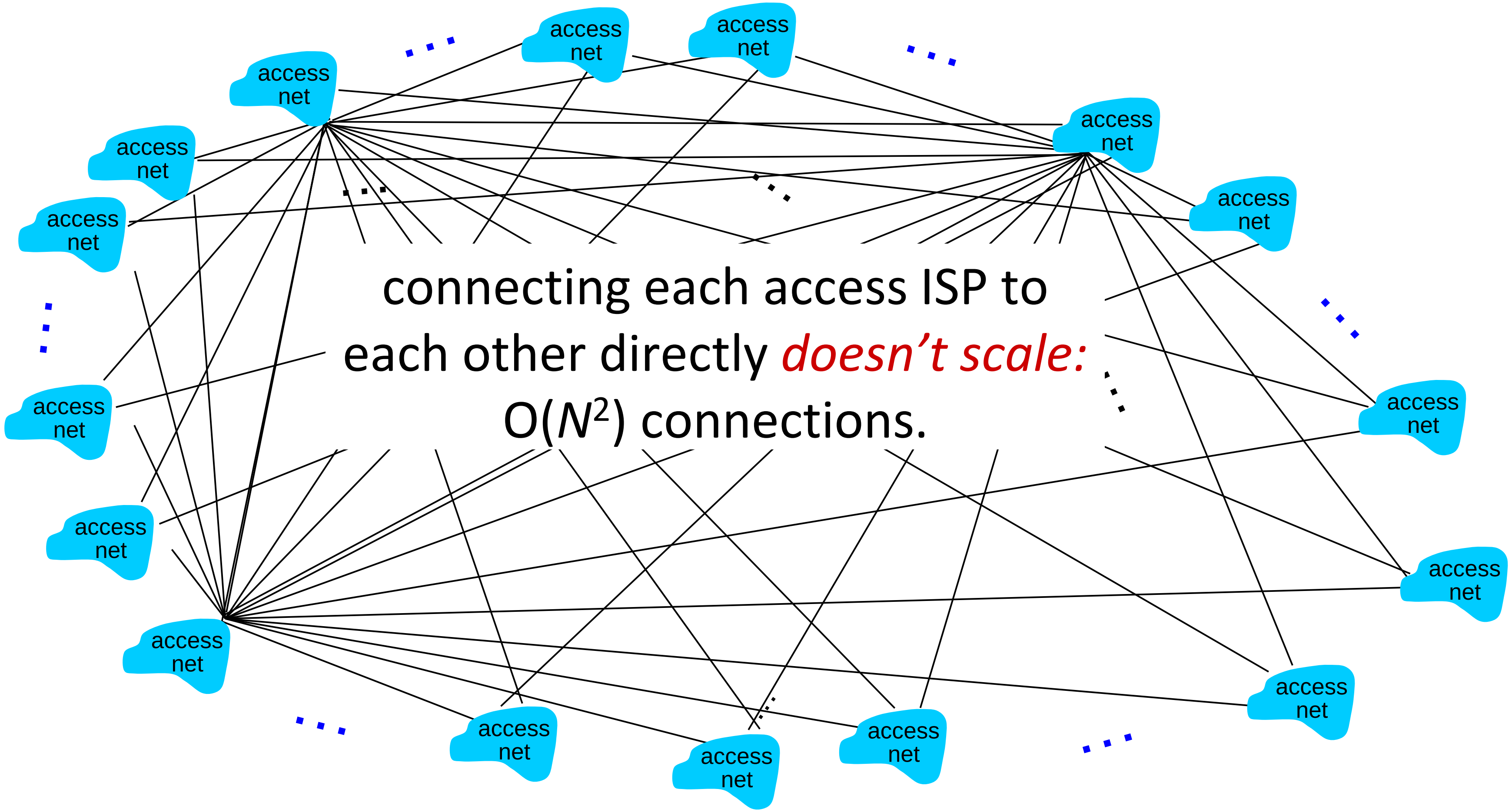
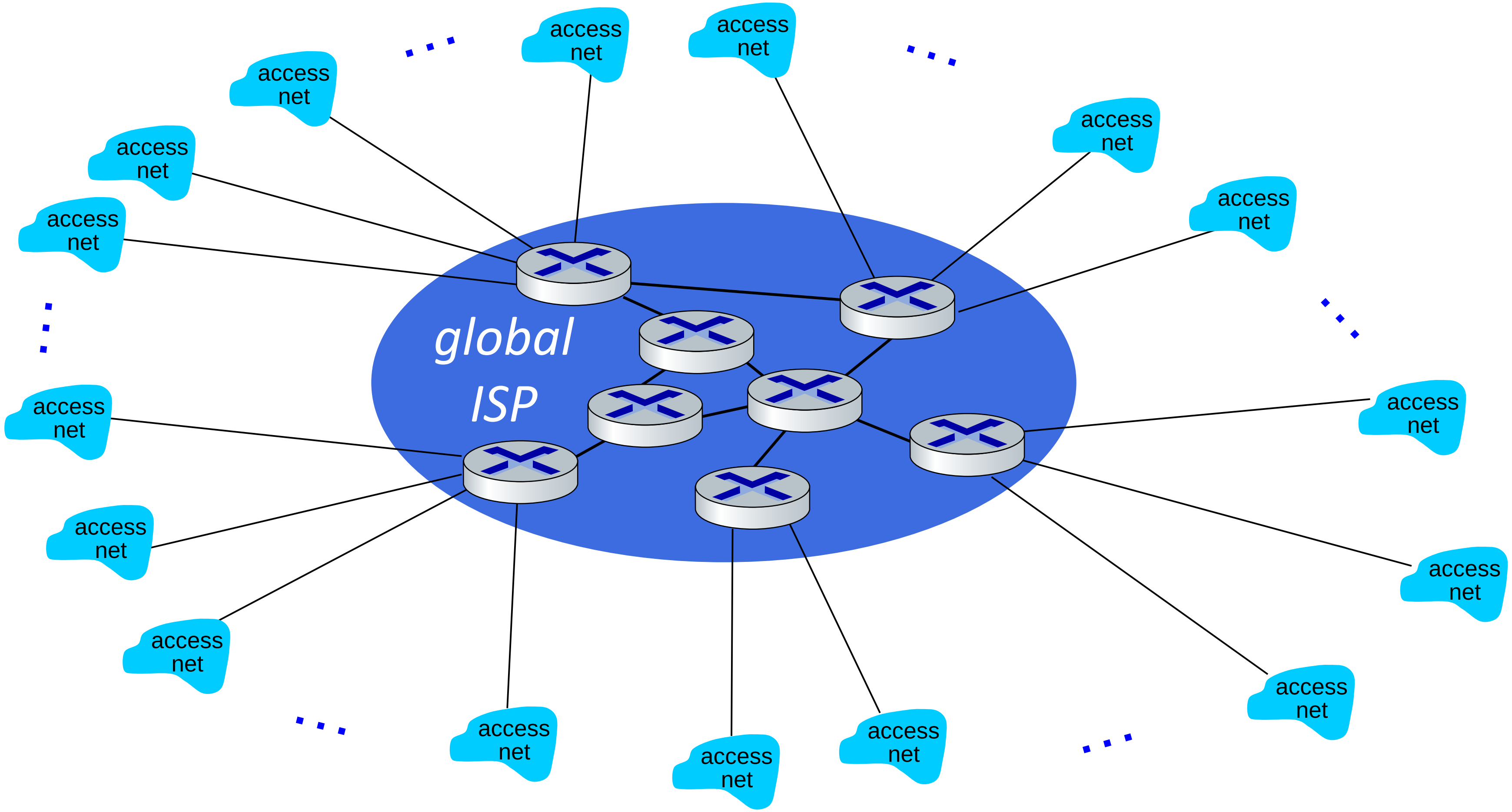


connecting each access ISP to each other directly *doesn't scale:* $O(N^2)$ connections.

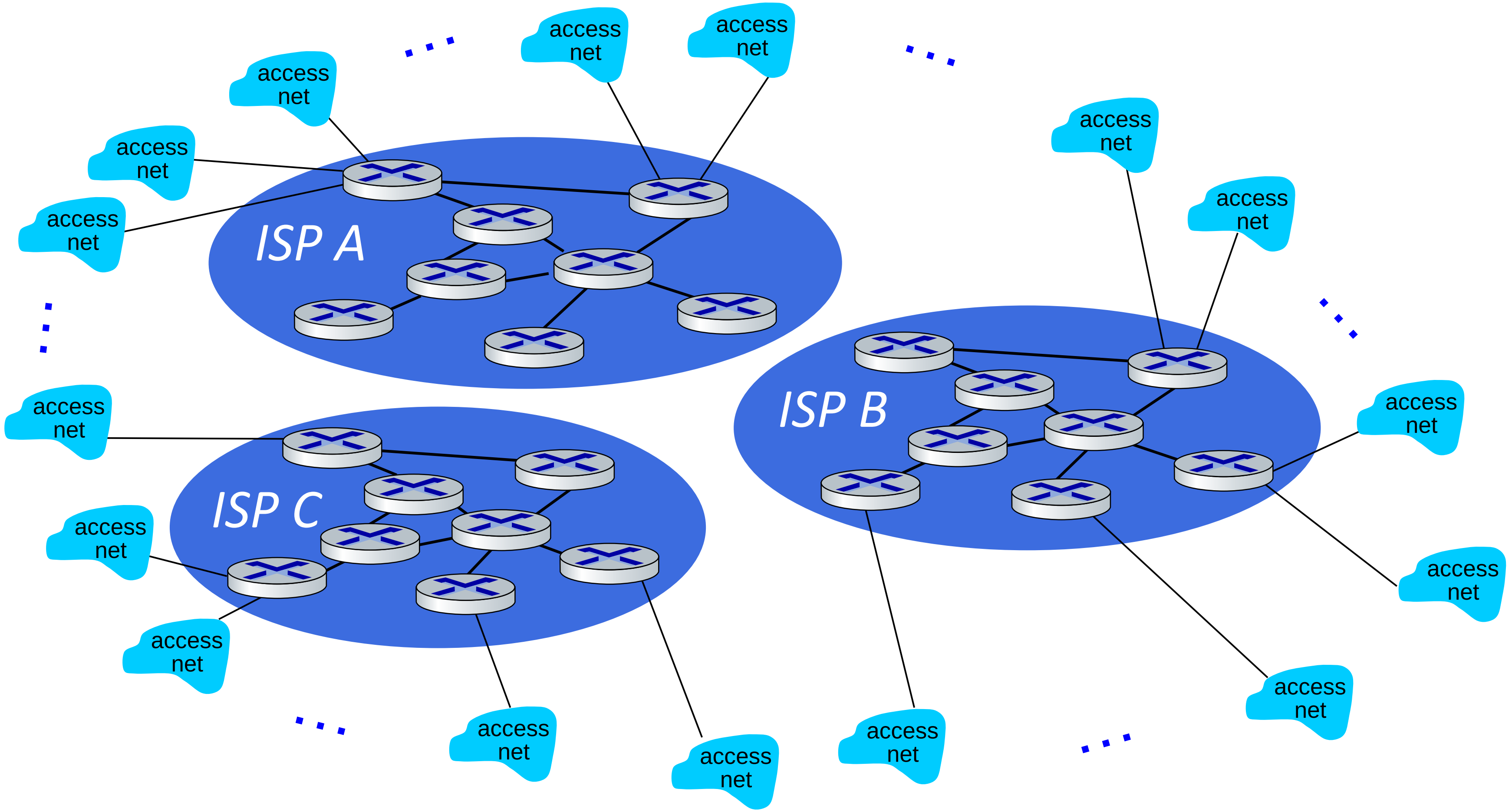
Internet structure: a "network of networks"

Option: connect each access ISP to one global transit ISP?
Customer and provider ISPs have economic agreement.



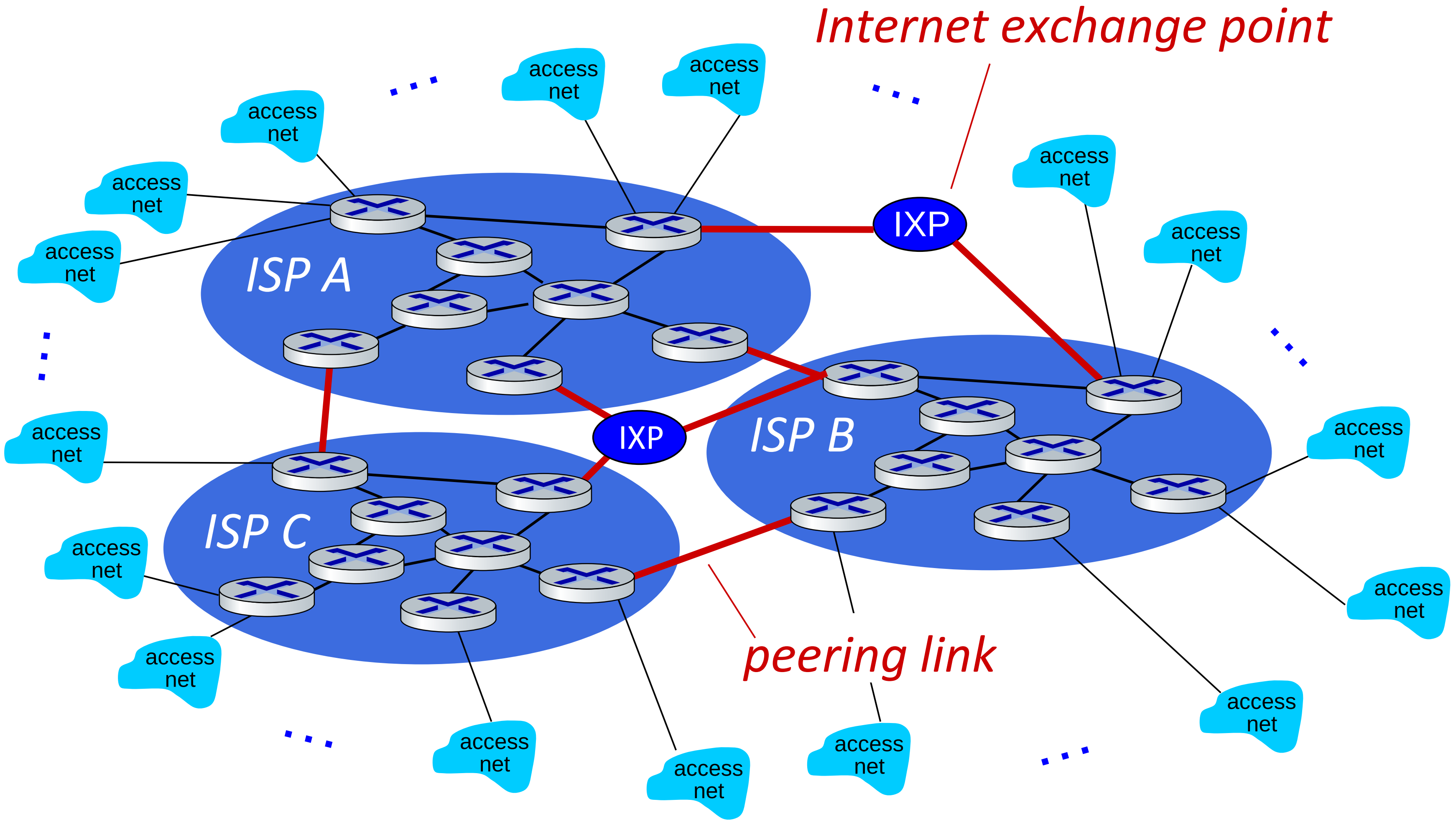
Internet structure: a "network of networks"

But if one global ISP is viable business, there will be competitors ….



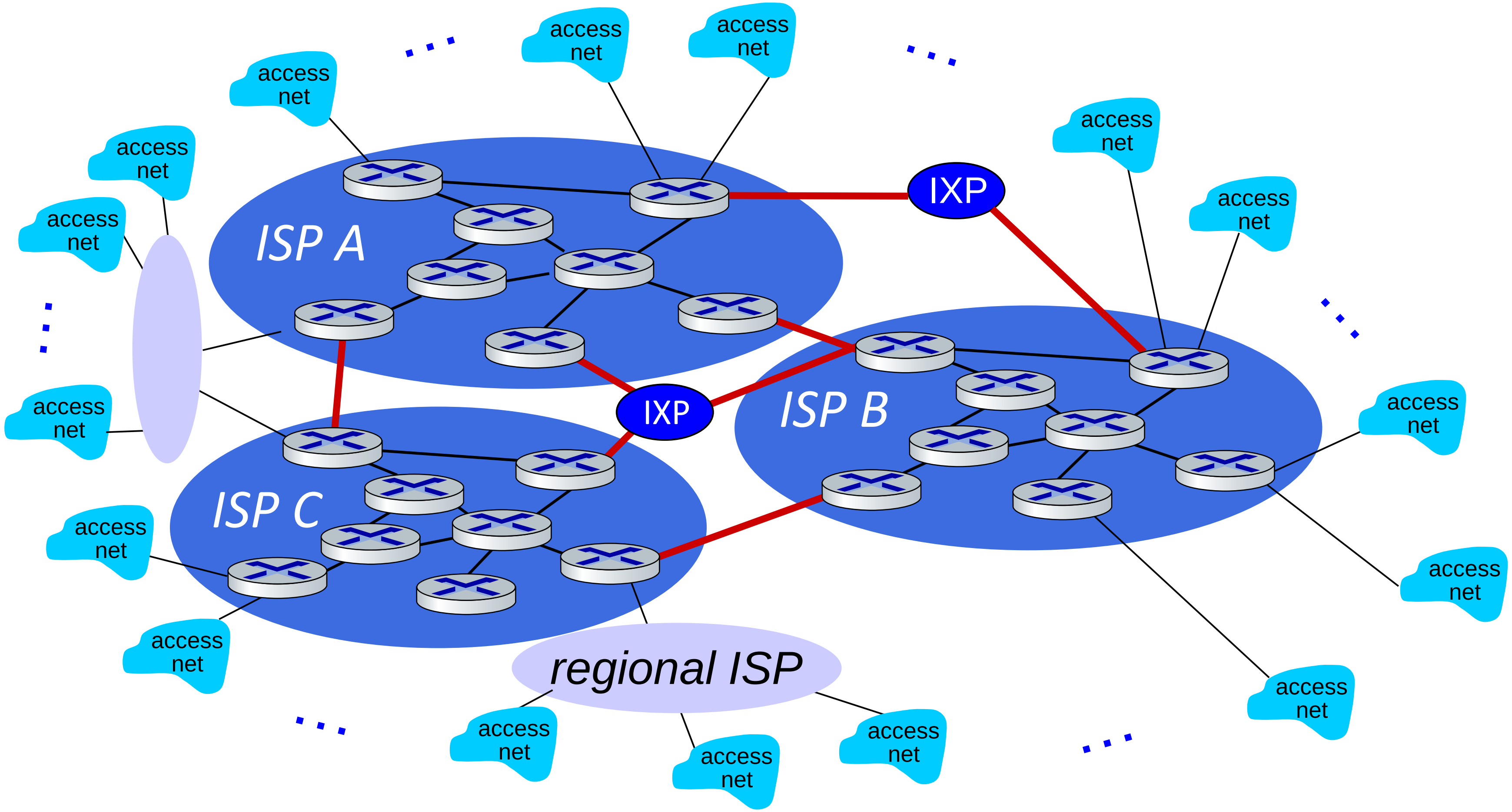
Internet structure: a “network of networks”

But if one global ISP is viable business, there will be competitors …. who will want to be connected



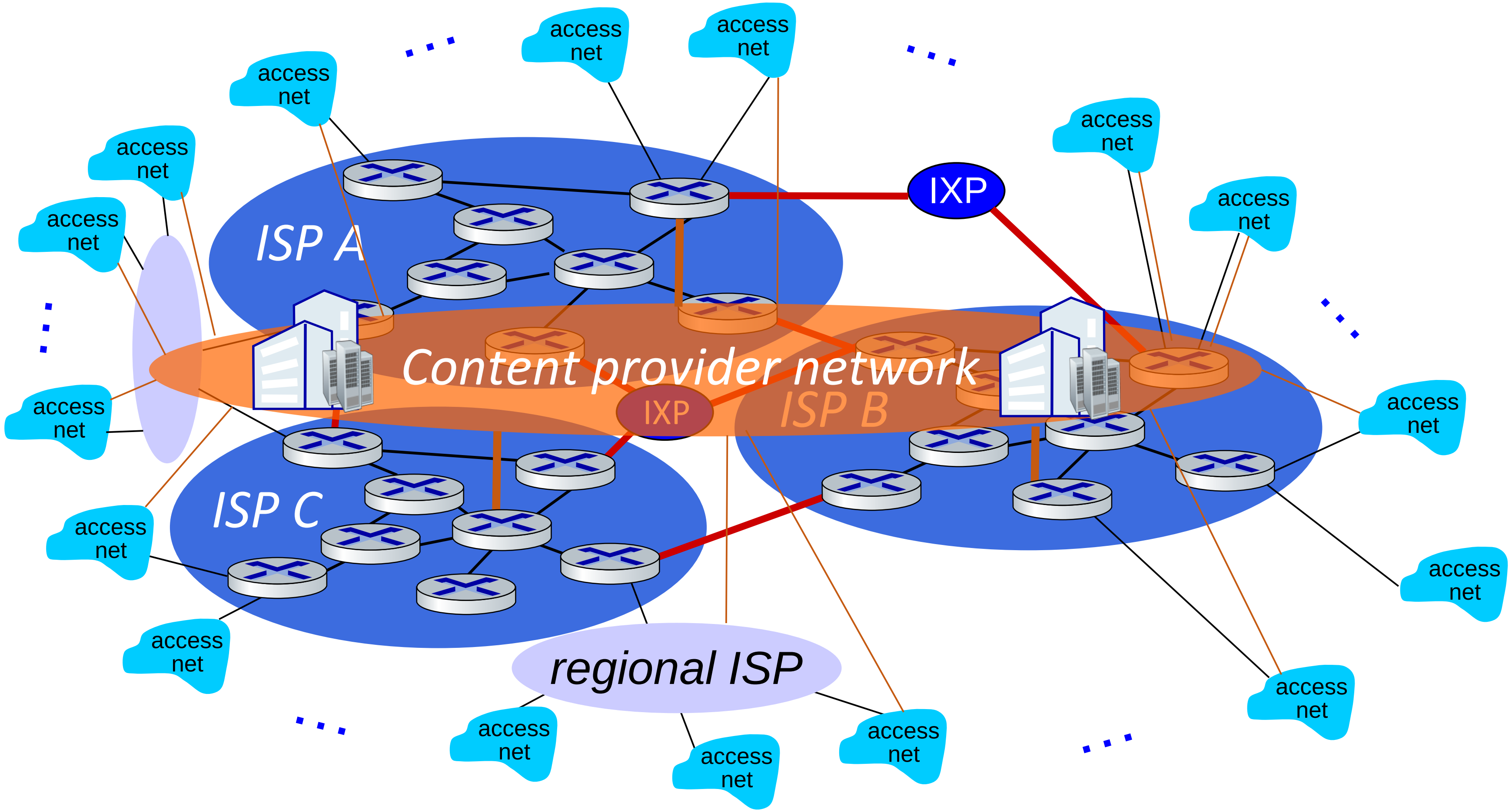
Internet structure: a “network of networks”

… and regional networks may arise to connect access nets to ISPs

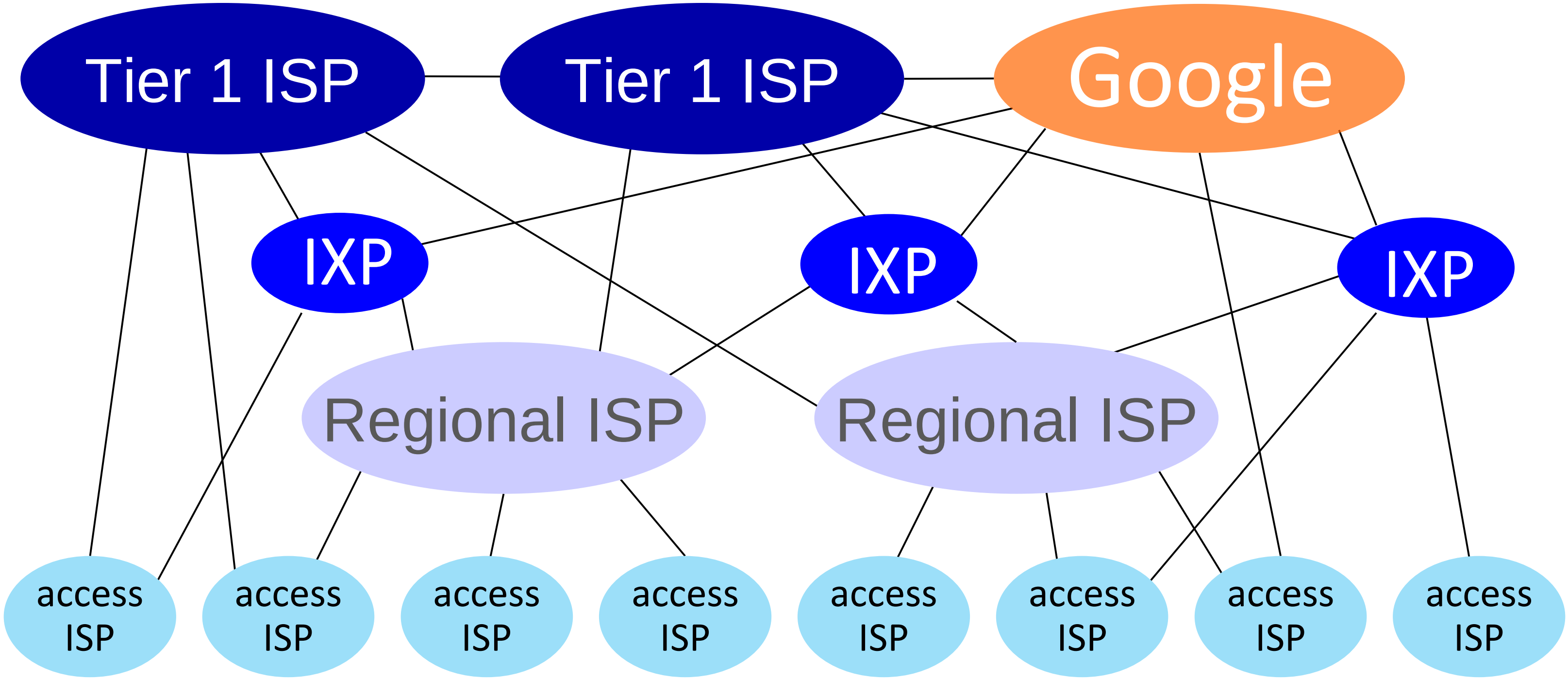


Internet structure: a "network of networks"

… and content provider networks (e.g., Google, Microsoft, Akamai) may run their own network, to bring services, content close to end users



Internet structure: a "network of networks"



At "center": small # of well-connected large networks

- "tier-1" commercial ISPs (e.g., Level 3, Sprint, AT&T, NTT), national & international coverage
- content provider networks (e.g., Google, Facebook): private network that connects its data centers to Internet, often bypassing tier-1 (their traffic not passing Tier 1), regional ISPs

Chapter 1: roadmap

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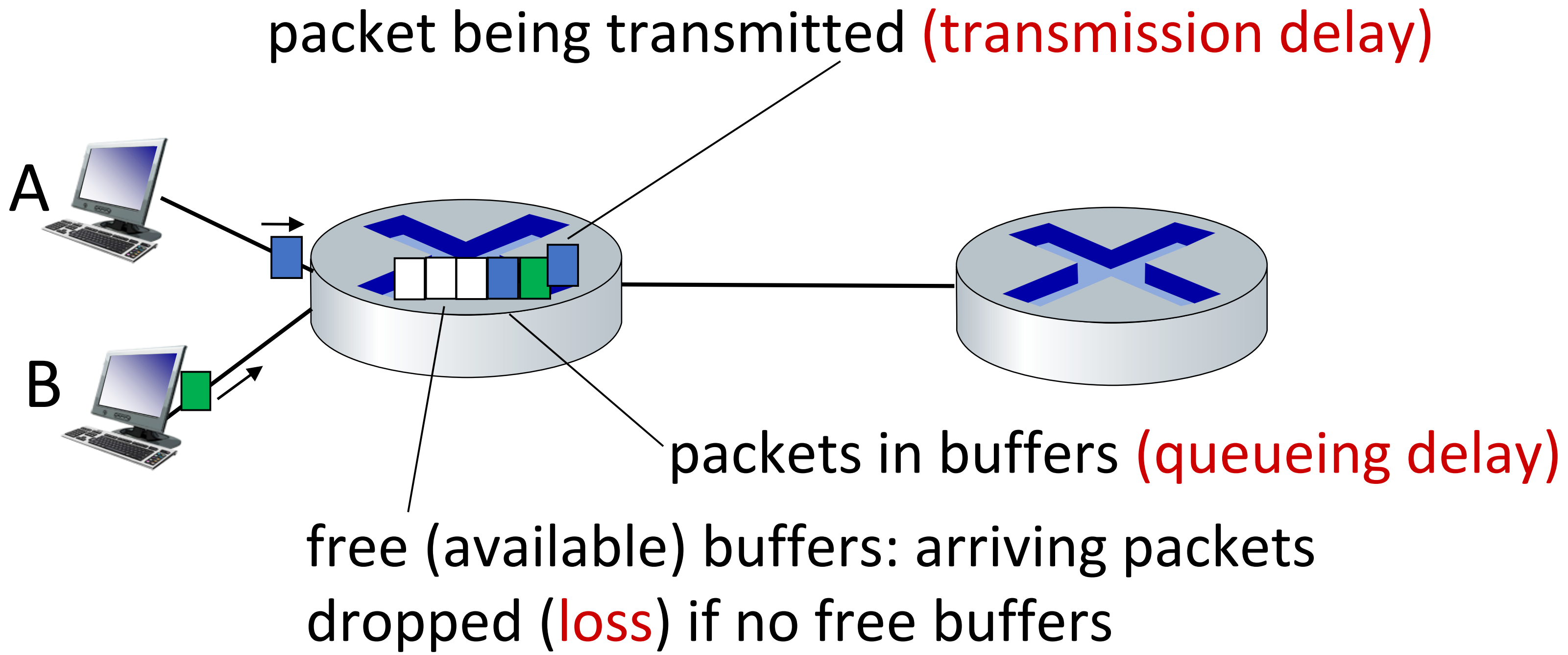
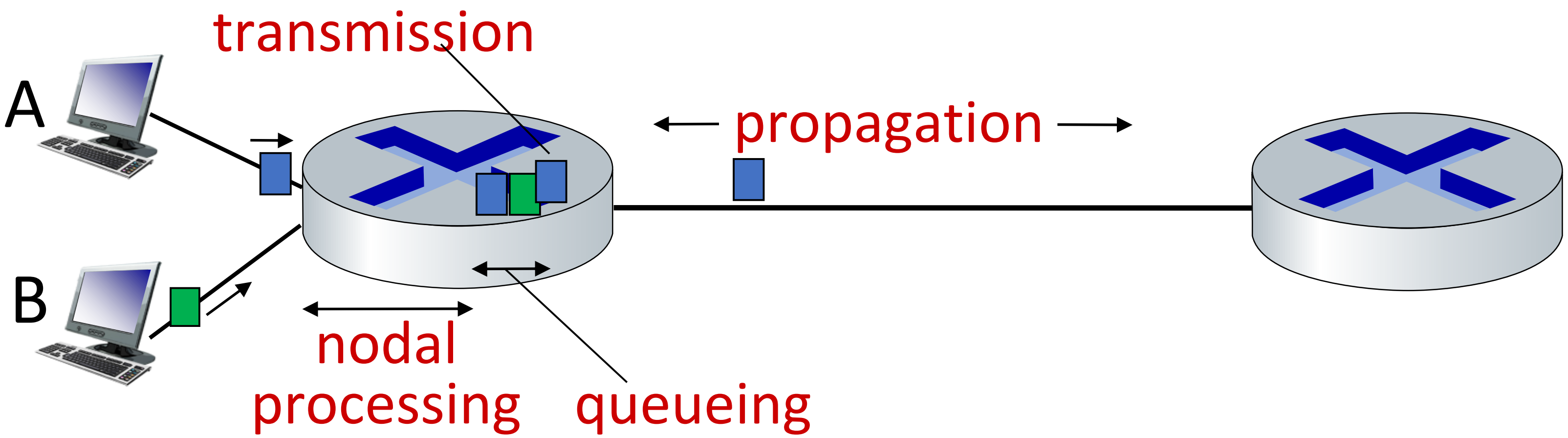
- What *is* a protocol?
- Network edge: hosts, access network, physical media
- Network core: packet/circuit switching, internet structure
- Performance: loss, delay, throughput
- Protocol layers, service models, security

How do packet delay and loss occur?

- packets *queue* in router buffers, waiting for turn for transmission
  - queue length grows when arrival rate to link (temporarily) exceeds output link capacity
- packet *loss* occurs when memory to hold queued packets fills up



Packet delay: four sources



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

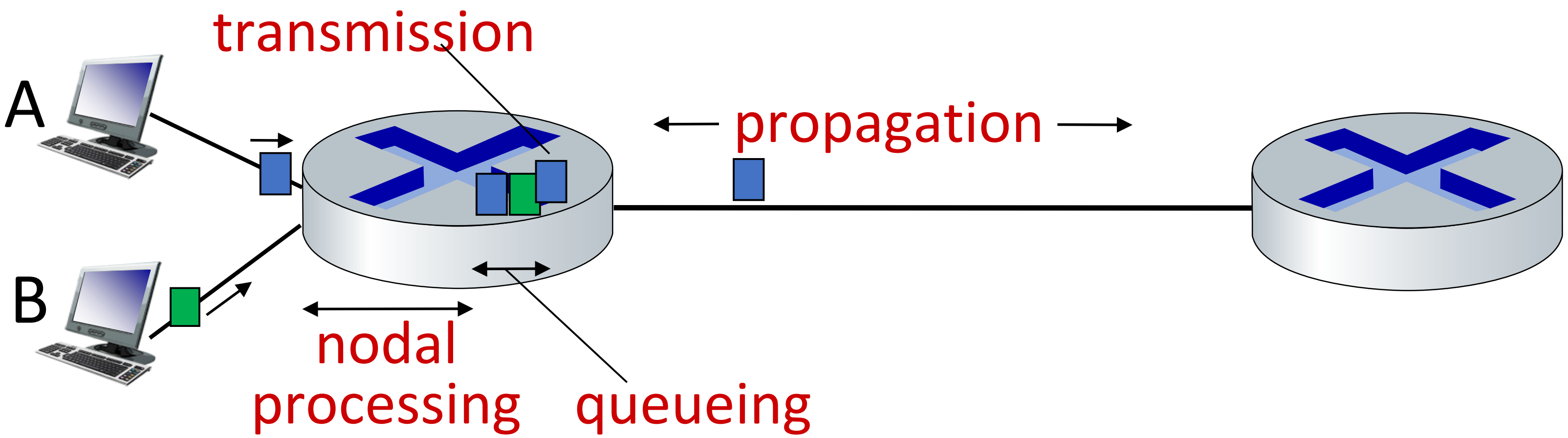
d_{proc}: nodal processing

- check bit errors
- determine output link
- typically < microsecs

d_{queue}: queueing delay

- time waiting at output link for transmission
- depends on congestion level of router

Packet delay: four sources



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

d_{trans}: **transmission delay:**

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

Transmission delay is the time taken to push all the packet bits on the transmission link.

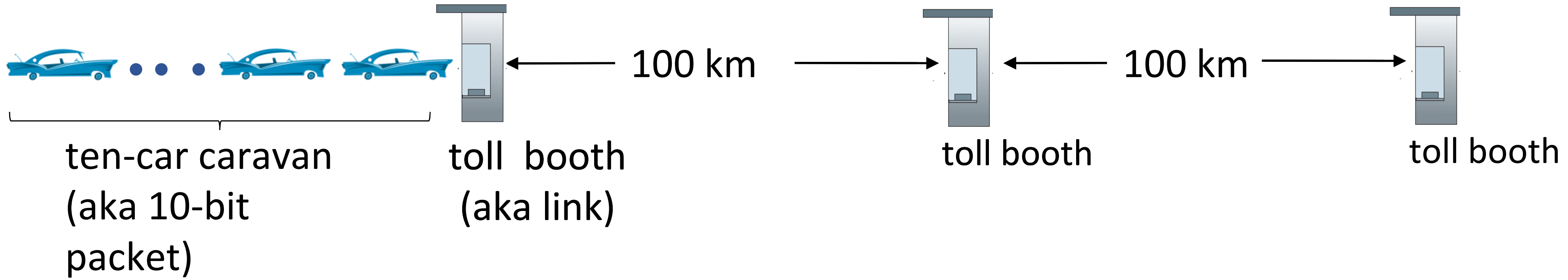
d_{trans} and d_{prop} *very* different

d_{prop}: **propagation delay:**

- d: length of physical link between end-points
- s: propagation speed (~2×10^8 m/sec)
- $d_{prop} = d/s$

Propagation delay is the time it takes for one bit to travel from one end of the link to the other.

Caravan analogy



- car ~ bit; caravan ~ packet; toll service ~ link transmission
- toll booth takes 12 sec to service car (bit transmission time)
- "propagate" at 100 km/hr
- *Q:* How long until caravan is lined up before 2nd toll booth?

- time to "push" entire caravan through toll booth onto highway = 12*10 = 120 sec
- time for last car to propagate from 1st to 2nd toll both: 100km/(100km/hr) = 1 hr
- *A:* 62 minutes

Numerical Example

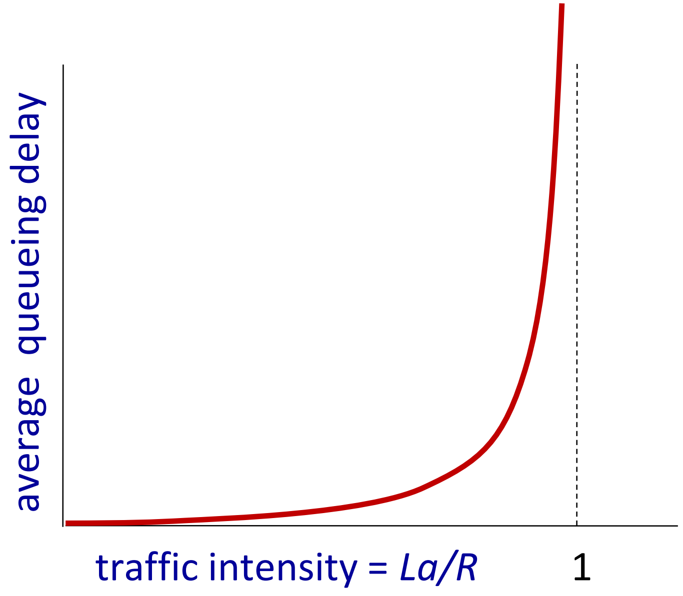
- Let’s say you want to send a message from your computer to a server located in another country.
  - The message is 1000 bytes long, and the link between your computer and the server has a bandwidth of 1 Mbps.
  - Transmission delay = Packet size / Link bandwidth = 1000 * 8 / 1,000,000 = 0.008 seconds
- Let’s assume that the distance between your computer and the server is 10,000 km, and the speed of light through the medium is 200,000 km/s. The propagation delay would be calculated as follows:
  - Propagation delay = Distance / Propagation sped = 10,000 / 200,000 = 0.05 seconds

Packet queueing delay (revisited)

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

$$\frac{L \cdot a}{R} : \quad \frac{\text{arrival rate of bits}}{\text{service rate of bits}}$$

"traffic intensity"



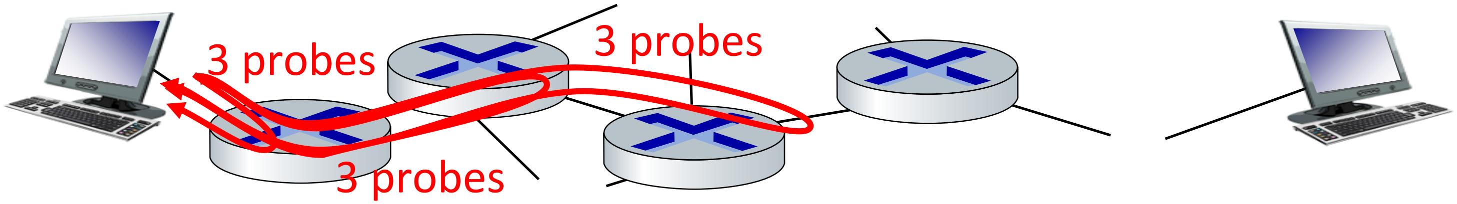
- *La/R* ~ 0: avg. queueing delay small
- *La/R* => 1: avg. queueing delay large
- *La/R* >> 1: more "work" arriving is more than can be serviced - average delay infinite!

La/R ~ 0

La/R -> 1

“Real” Internet delays and routes

- what do “real” Internet delay & loss look like?
- **traceroute** program: provides delay measurement from source to router along end-end Internet path towards destination. For all *i:*
  - sends three packets that will reach router *i* on path towards destination (with time-to-live field value of *i*)
  - router *i* will return packets to sender
  - sender measures time interval between transmission and reply



Real Internet delays and routes

traceroute: gaia.cs.umass.edu to www.eurecom.fr

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

3 delay measurements
to border1-rt-fa5-1-0.gw.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

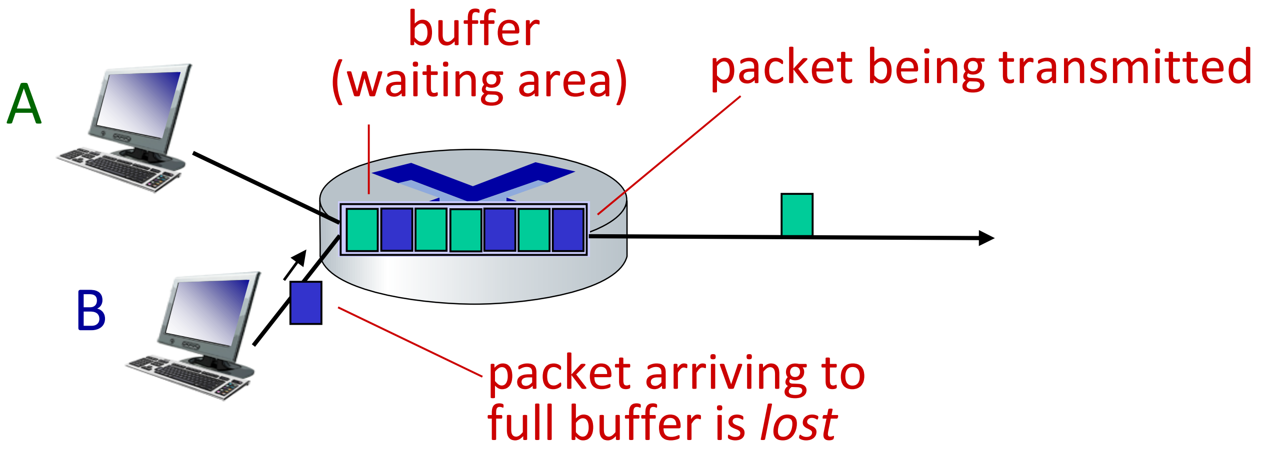
looks like delays *decrease*! Why?

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

* Do some traceroutes from exotic countries at www.traceroute.org

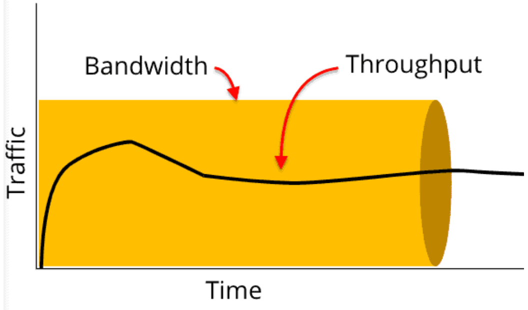
Packet loss

- 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 (e.g. TCP) end system, or not at all (UDP)



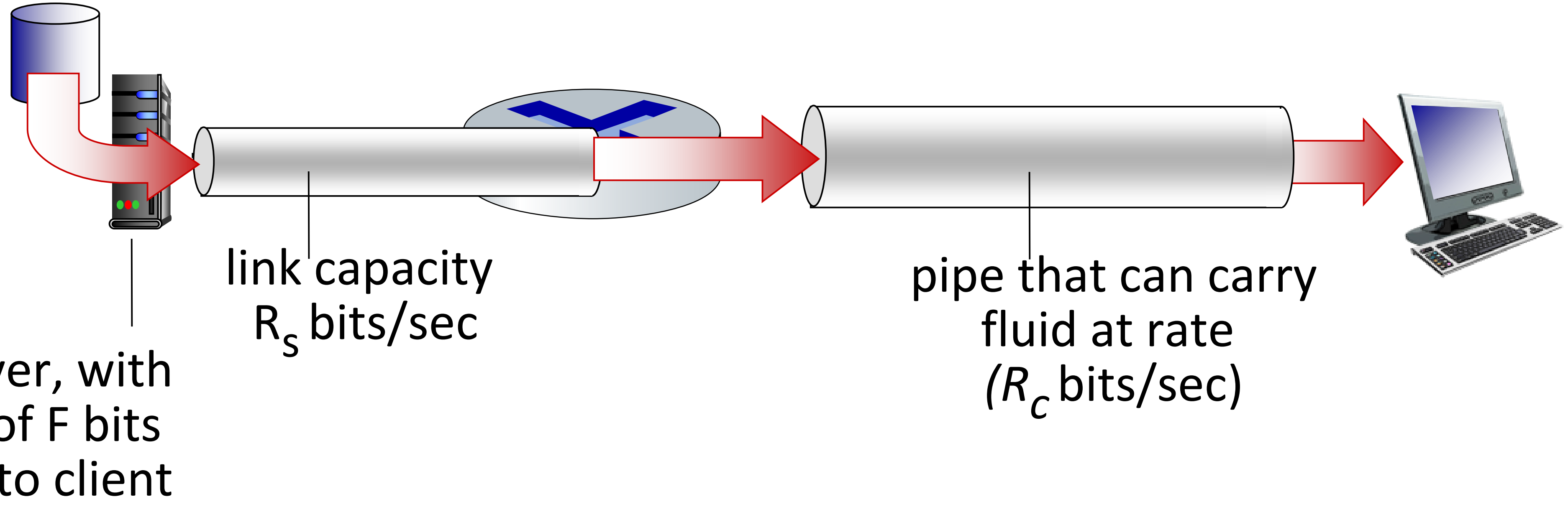
* Check out the Java applet for an interactive animation (on publisher's website) of queuing and loss

Throughput



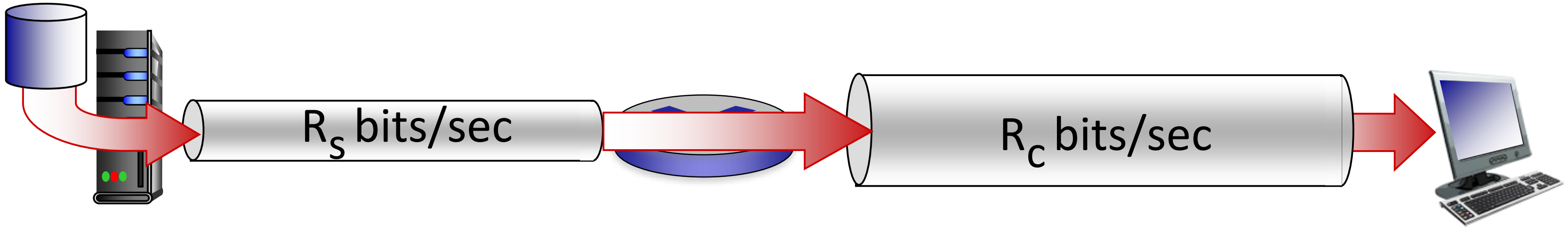
throughput:

- is the amount of data that can be transmitted for the sender to the receiver at a given period of time (how much data (max) the network can handle at given period of time)
  - *instantaneous:* rate at given point in time
  - *average:* rate over longer period of time
  - *In practice, Throughput never matches the bandwidth (total capacity of the link)*



Throughput

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



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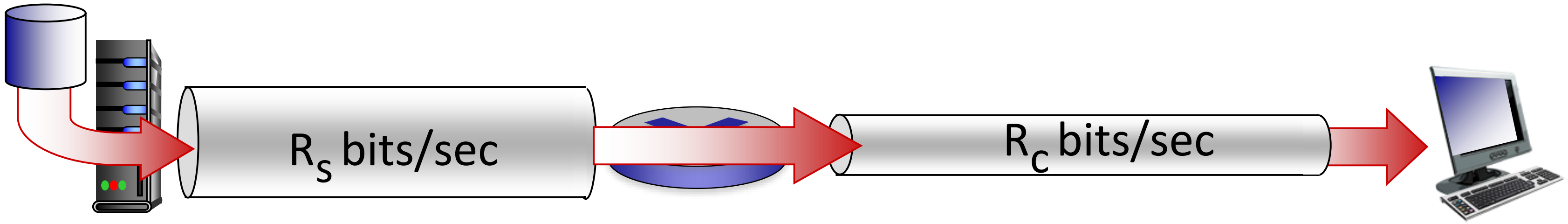
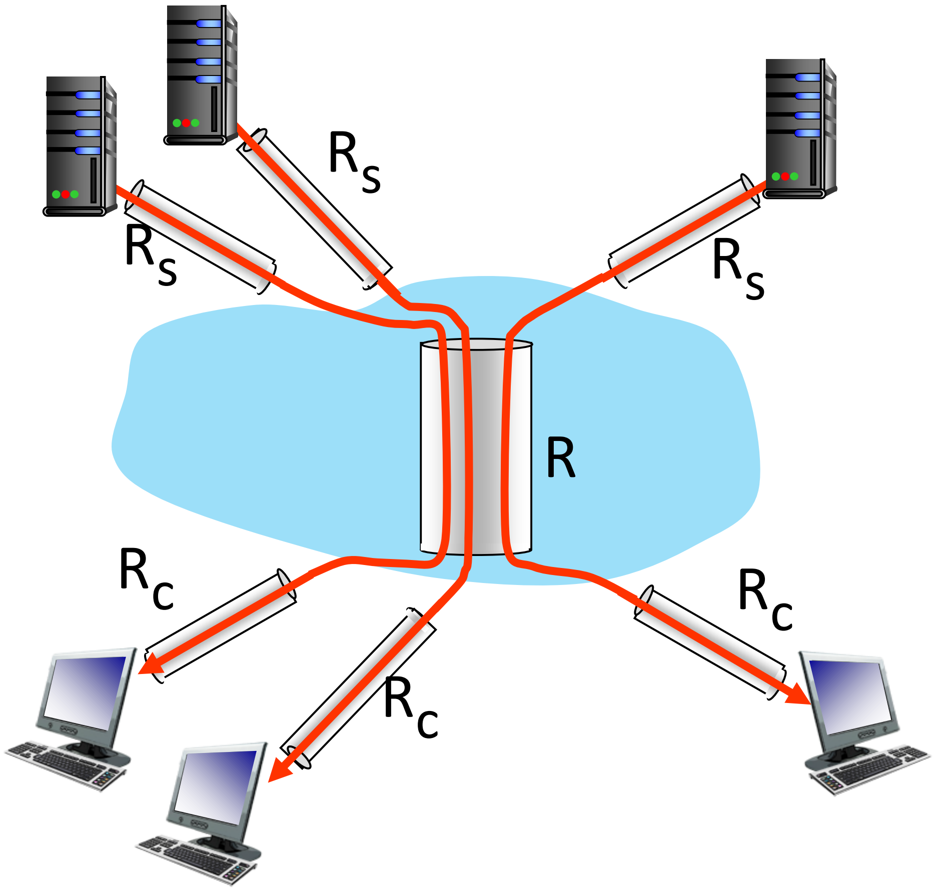


bottleneck link

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

Throughput: network scenario



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

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