CSE 150/L : Introduction to Computer Networks

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Chapter 1

What's in the Labs?

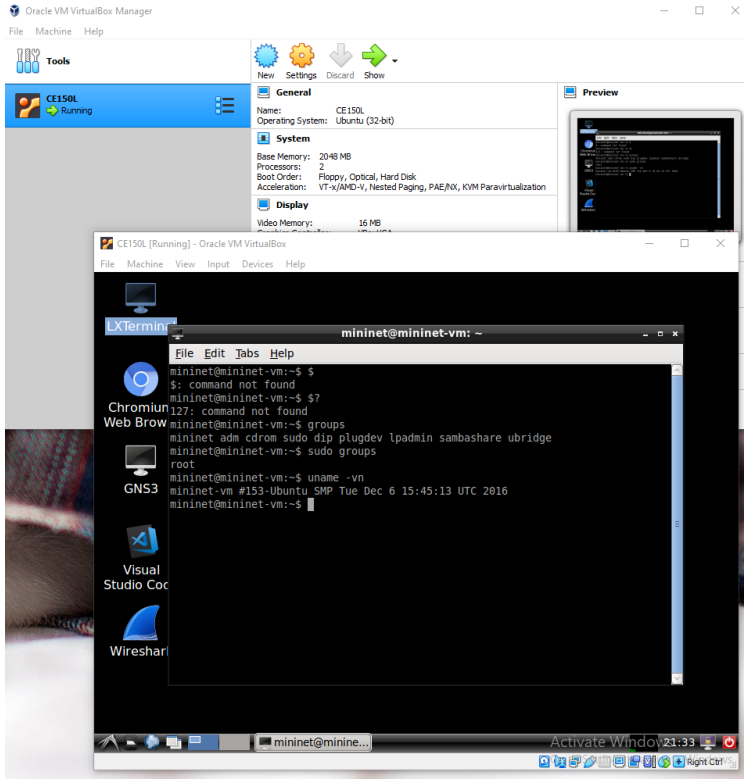
- Help you better understand how network works
- Help you get familiar with Mininet, Wireshark and Linux OS
- 3 prelabs+ 3 labs+ 1 final project
  - Lab 1: Installing and Using Mininet
  - Lab 2: HTTP, DNS, and TCP
  - Lab 3: Simple Firewall Using OpenFlow
  - Final project: build a network using Mininet

Prelab 1

- Tips:
  - Most of the questions can be solved via searching on Google
  - For Windows user: can also download Ubuntu subsystm in Microsoft store, but VirtualBox is more recommanded
  - For Mac Os user: Mac OS is a Unix OS and its command line is 99.9% the same as any Linux distribution
  - Install VirtualBox & Mininet (for the incoming lab 1)

Prerequisite

- Installing Virtualbox
  - Make sure you have downloaded the correct version(32-bit vs 64-bit)
  - Installing Mininet

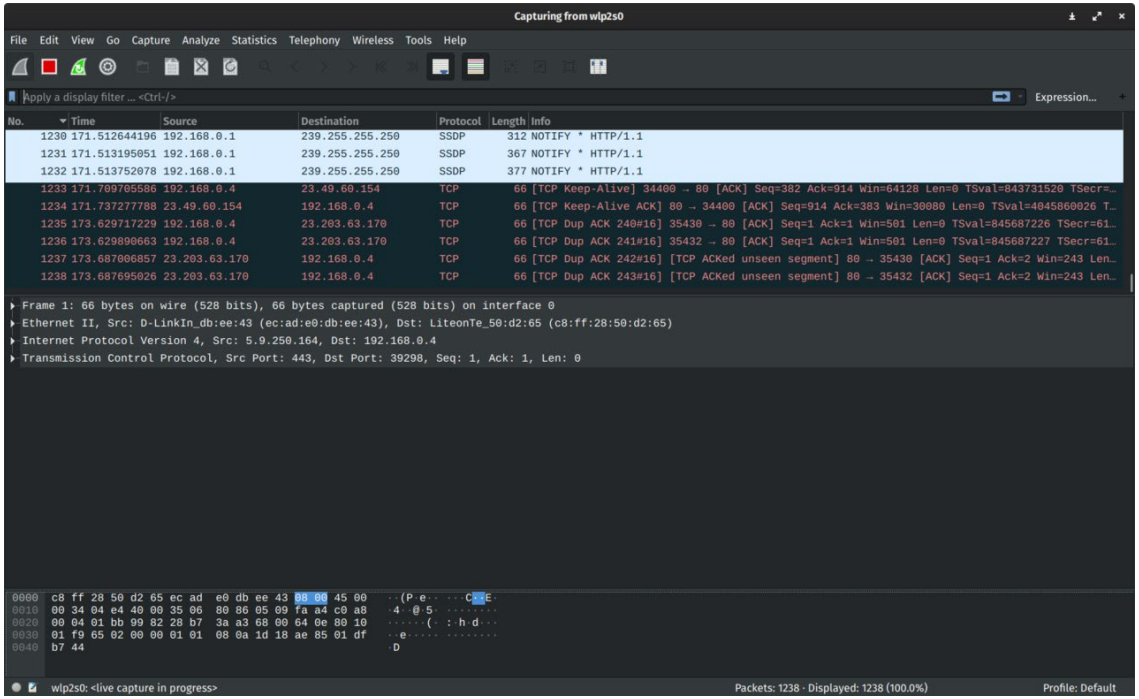


What is Mininet?

- A "network emulator" tool developed at Stanford in 2010.
- Designed to allow large scale networks to be emulated in software on a laptop
- Allow the user to quickly create, interact with, customize and share a software-defined network (SDN) prototype to simulate a network topology that uses **Openflow** switches.

What is WireShark?

- A free and open-source packet analyzer
- Used for network troubleshooting, analysis, software and communications protocol development, and education



Introduction

Fundamental concepts, terminology
(Chapter 1)

Chapter 1: roadmap

1.1 what *is* the Internet?

1.2 network edge

- end systems, access networks, links

1.3 network core

- packet switching, circuit switching, network structure

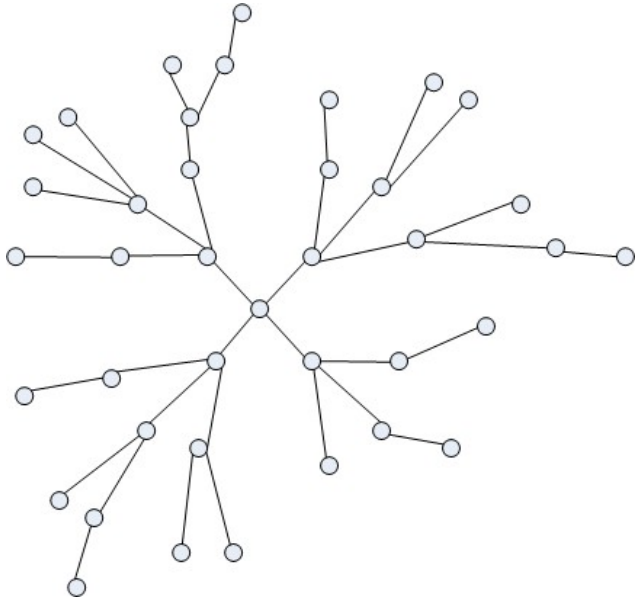
1.4 delay, loss, throughput in networks

1.5 protocol layers, service models

1.6 networks under attack: security

What is a computer network?

"A compute network is a group of two or more computer systems linked together."



What are the components of a computer (communication) network?

What are the components of a computer (communication) network?

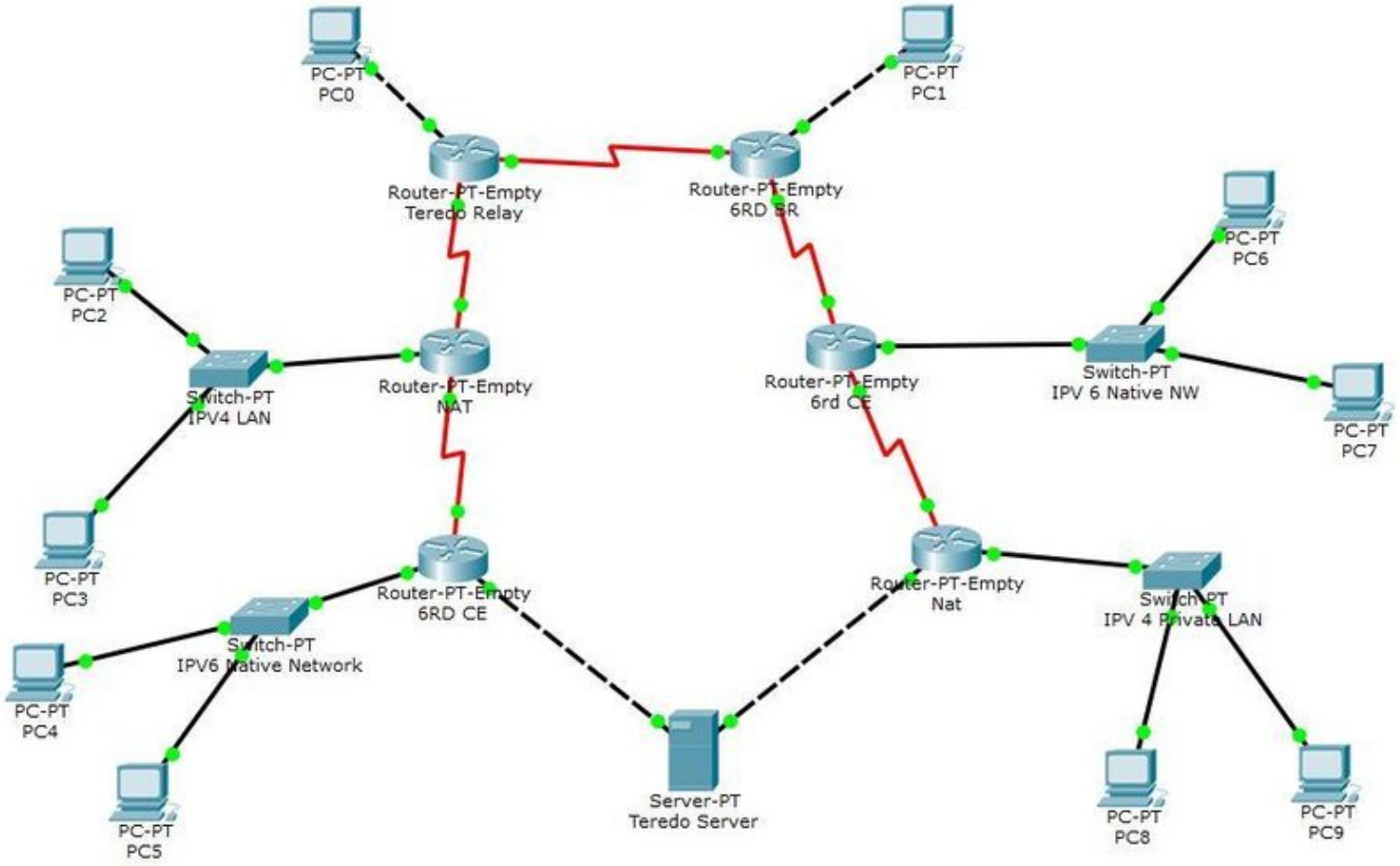
"How do you send text messages?"

What are the components of a computer (communication) network?

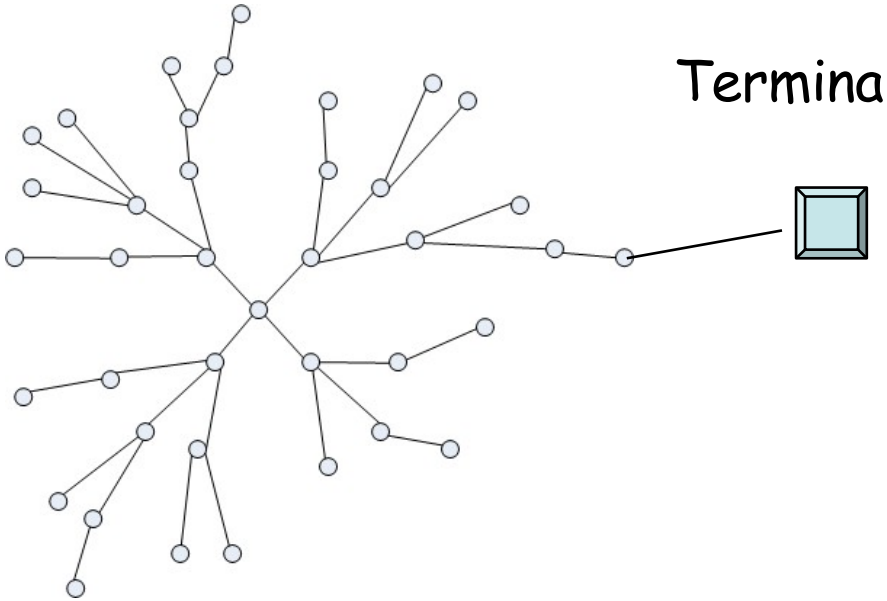
- Links, nodes, and
  - "terminals".
- What's the difference between "nodes" and "terminals"?

"How do you send text messages?"

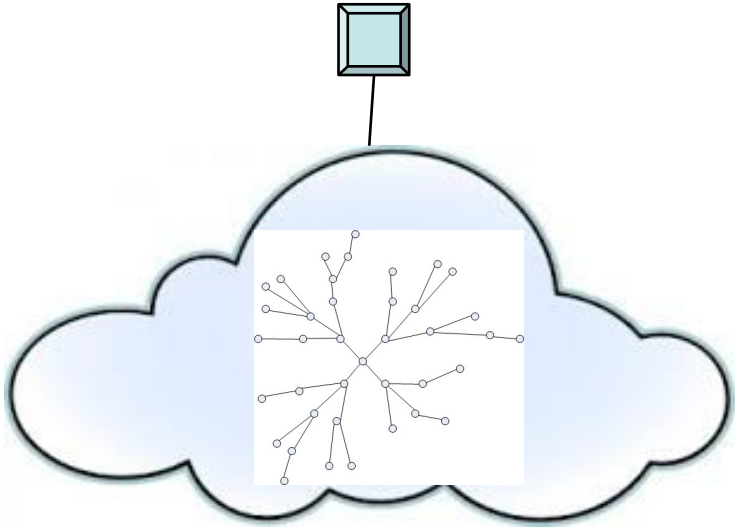
Nodes and Terminals



Nodes and Terminals



Terminals = Hosts, End-User Devices



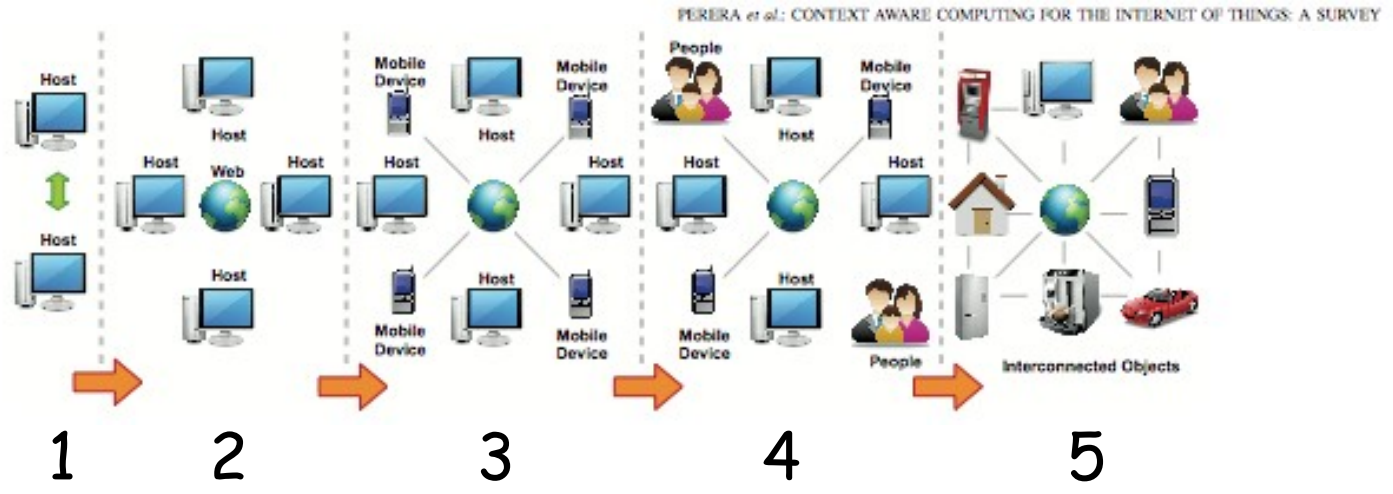
The Internet

- The Internet versus an internet?
- "internet" is an abbreviation of "internetwork".
  - Collection of interconnected networks, with no central administration or management.
  - A "network" has a single administrative authority.
- Intranetwork.

What made the Internet so popular?

What was the killer application ("killer app") of the Internet?
2nd killer application?
And more?

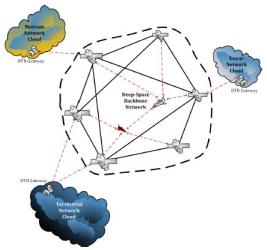
Internet Evolution



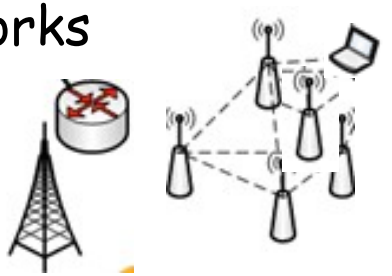
1: Connecting (few) computers: e-mail, file transfer, remote login.
2: Connecting larger number of computers: sharing information (WWW).
3: Connecting wireless and mobile devices.
4: Connecting people: social networks.
5: Connecting objects: Internet of Things (IoT), Context-Aware Networking.

Internets of the future: a vision

Smart home

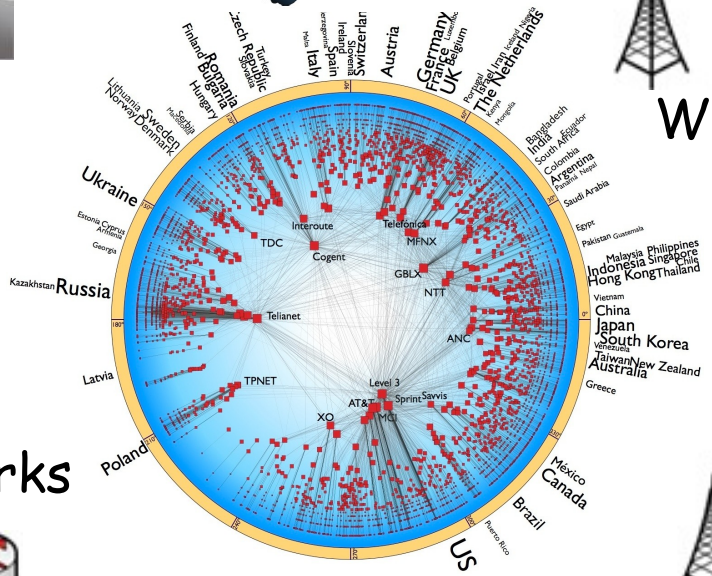


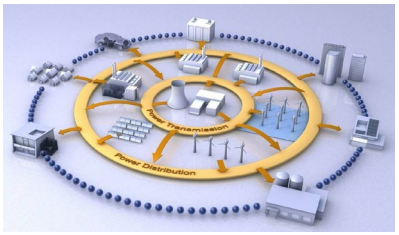
Interplanetary networks



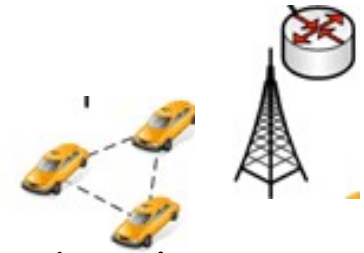
Wireless mesh network

Mobile ad-hoc networks

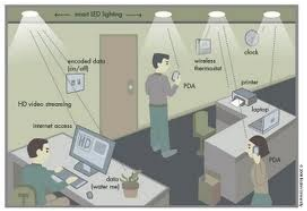




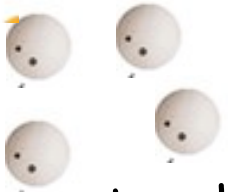
Smart grid



Vehicular networks

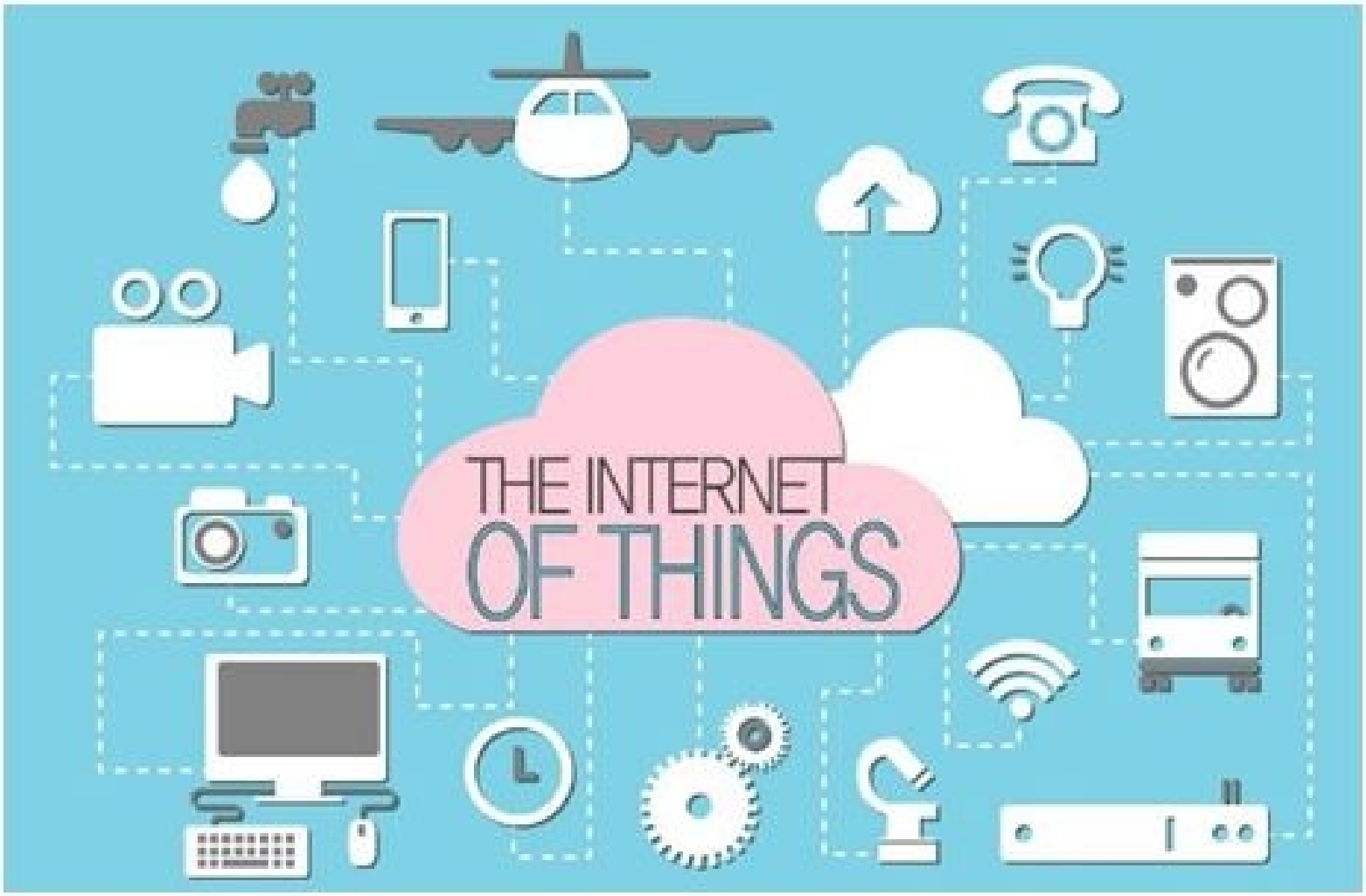


Smart office



Sensor networks

"The Internet of Everything"



Challenges

- Scalability
  - As of early 2013, ~1.5 billion connected PCs and ~1 billion Internet-enabled mobile phones.
  - By 2020, ~20-50 billion Internet-connected devices.
- Heterogeneity
  - Devices
  - Networks
  - Services
- Autonomy and administrative decentralization

What's the Internet?

What's the Internet: "Nuts and Bolts" View

PC

server

wireless laptop

cellular handheld

Millions of connected computing devices:
hosts = end systems
- Running *network apps*

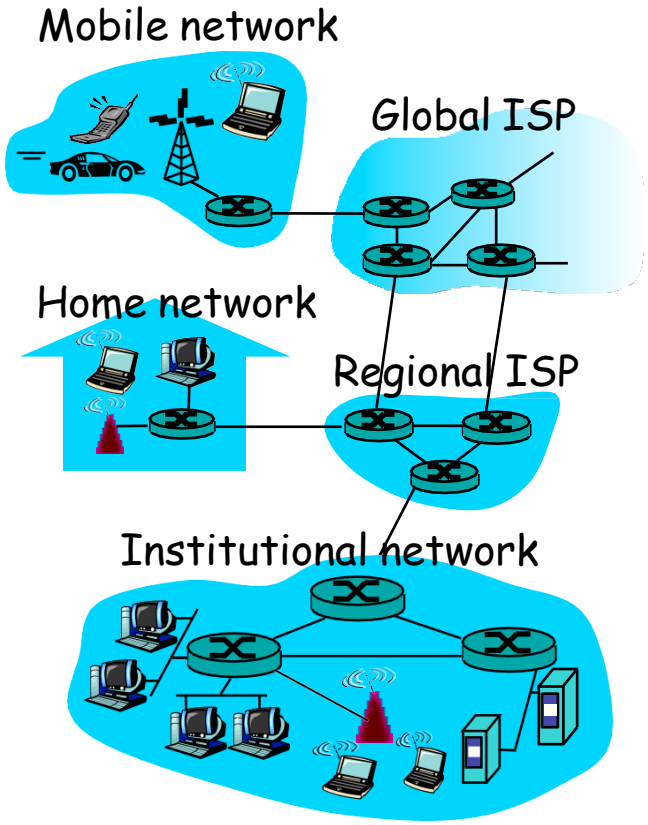
Access points

wired links

router

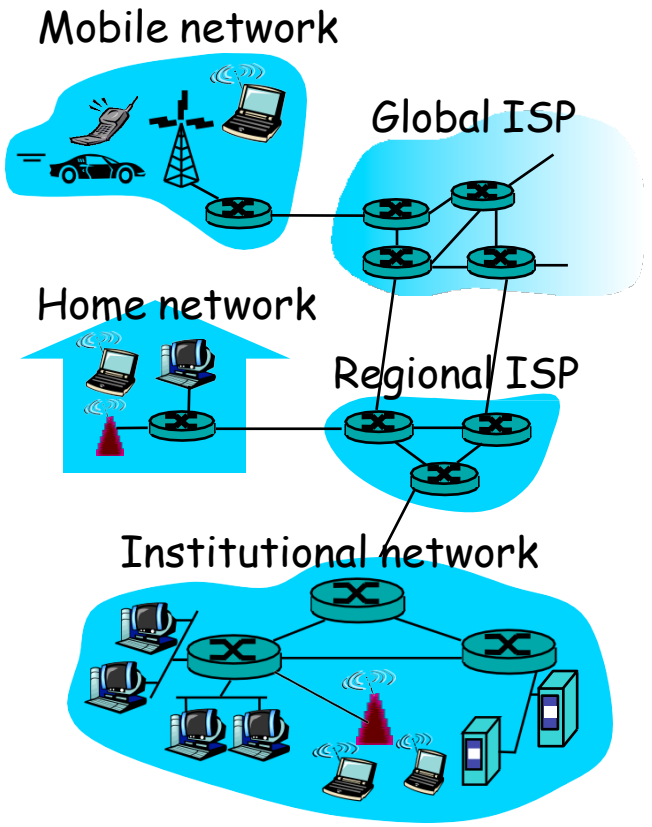
Connection links
Fiber, copper, radio, satellite

Routers:
Forward packets (chunks of data)



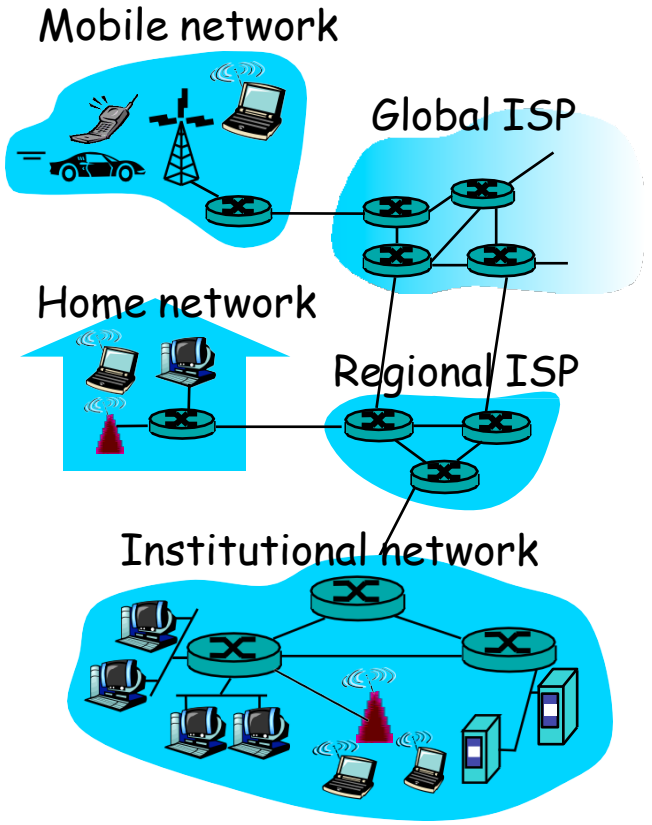
What's the Internet: "Nuts and Bolts" View

- *Internet:* "network of networks"
  - hierarchical



What's the Internet: "Service" View

- Communication *Infrastructure* enables distributed applications:
  - Web, VoIP, email, games, e-commerce, file sharing
- Communication services provided to apps:
  - reliable data delivery from source to destination
  - "best effort" (unreliable) data delivery



What's a protocol?

Human protocols:

- "What's the time?"
- "I have a question"
- Homeworks
- Exams

… specific messages sent

… specific actions taken when messages received, or other events

What's a protocol?

Human protocols:

- "What's the time?"
- "I have a question"

… specific messages sent

… specific actions taken when messages received, or other events

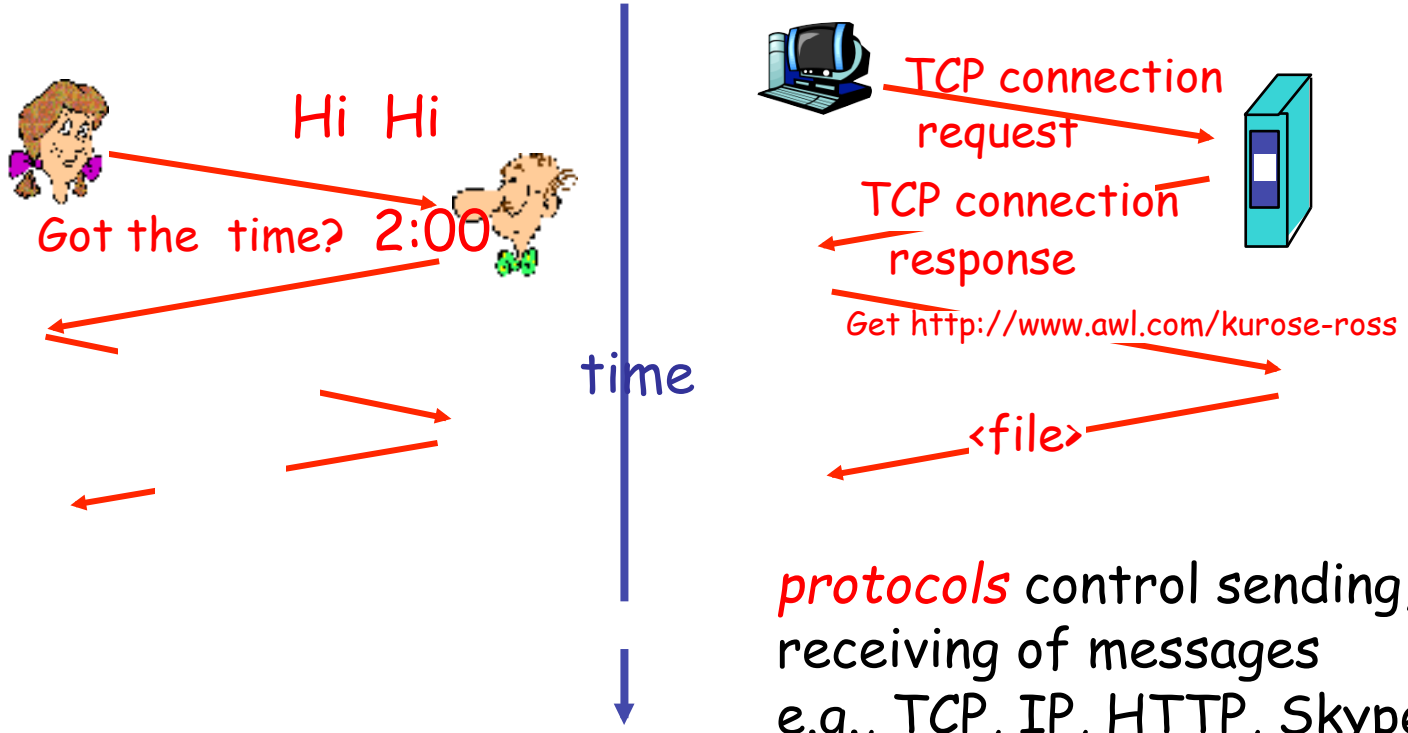
Network protocols:

- Machines rather than humans
- All communication activity in Internet governed by protocols

Protocols define format, order of messages sent and received among network entities, and actions taken on message transmission and receipt.

What's a protocol?

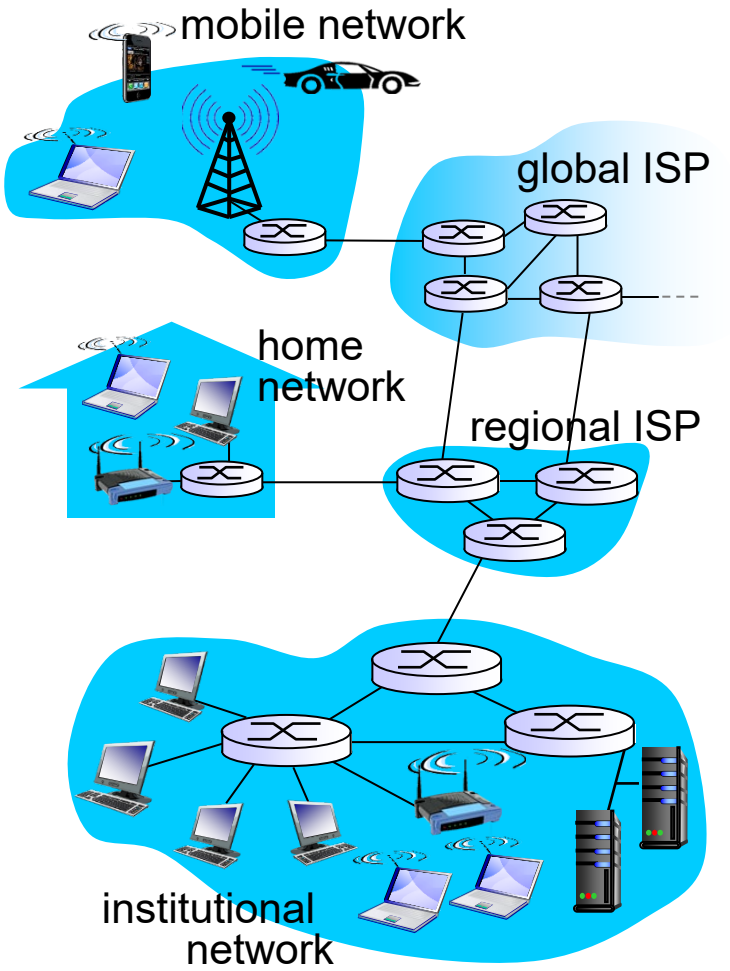
Human protocol and network protocol:



protocols control sending, receiving of messages
e.g., TCP, IP, HTTP, Skype, Ethernet

A closer look at network structure:

- *network edge:*
  - hosts: clients and servers
  - servers often in data centers
- *access networks, physical media:* wired, wireless communication links
- *network core:*
  - interconnected routers
  - network of networks



Chapter 1: roadmap

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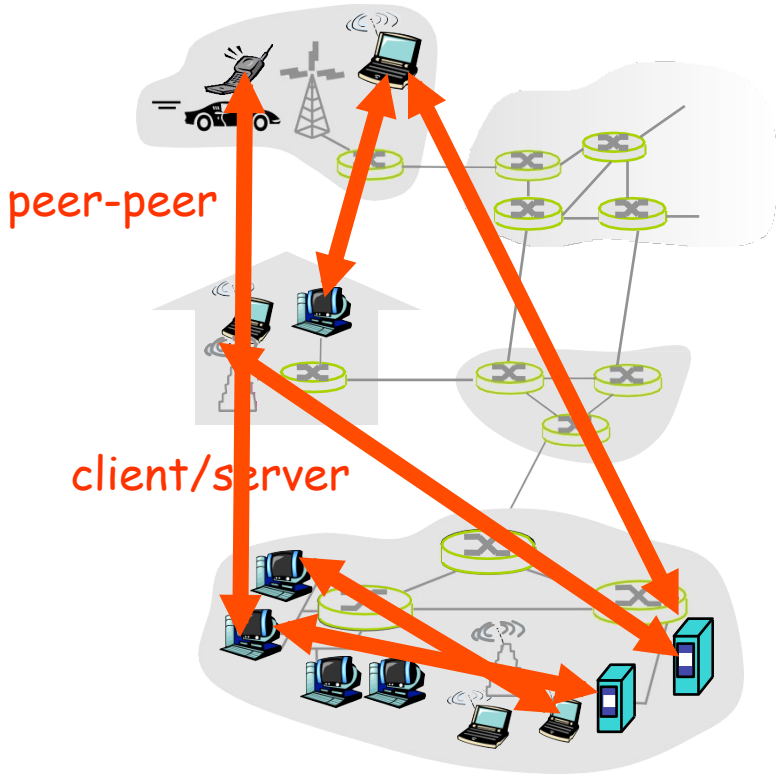
The Network Edge

- End systems (hosts):
  - run application programs
  - e.g. Web, email
  - at "edge of network"



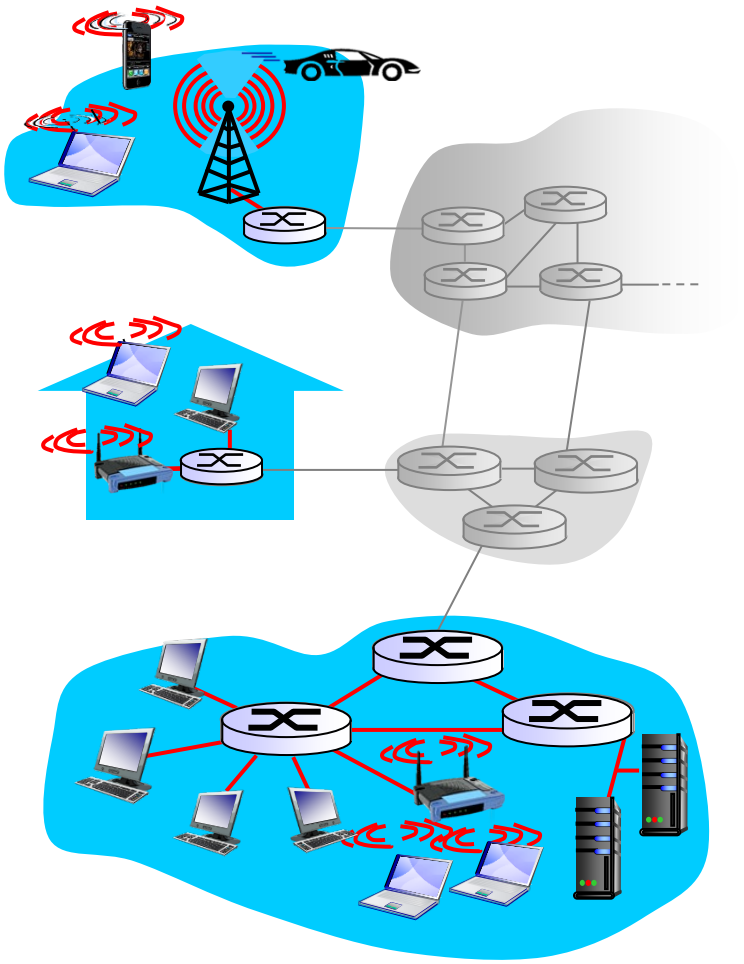
Access networks and physical media

Q: How to connect end systems to edge router?

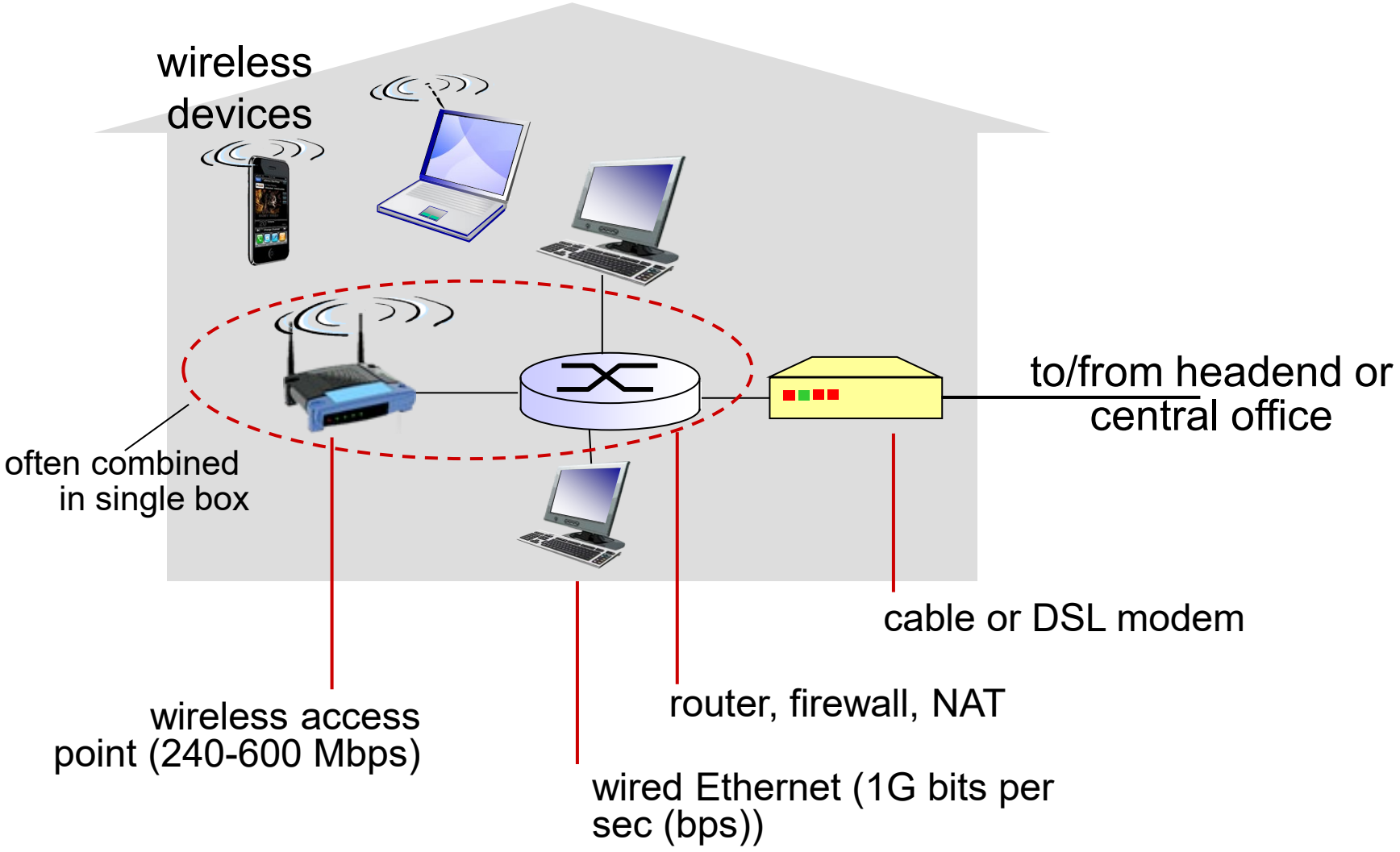
- residential access nets
- institutional access networks (school, company)
- mobile access networks

keep in mind:

- bandwidth (bits per second) of access network?
- shared or dedicated?
- Bandwidth cap



Access net: home network



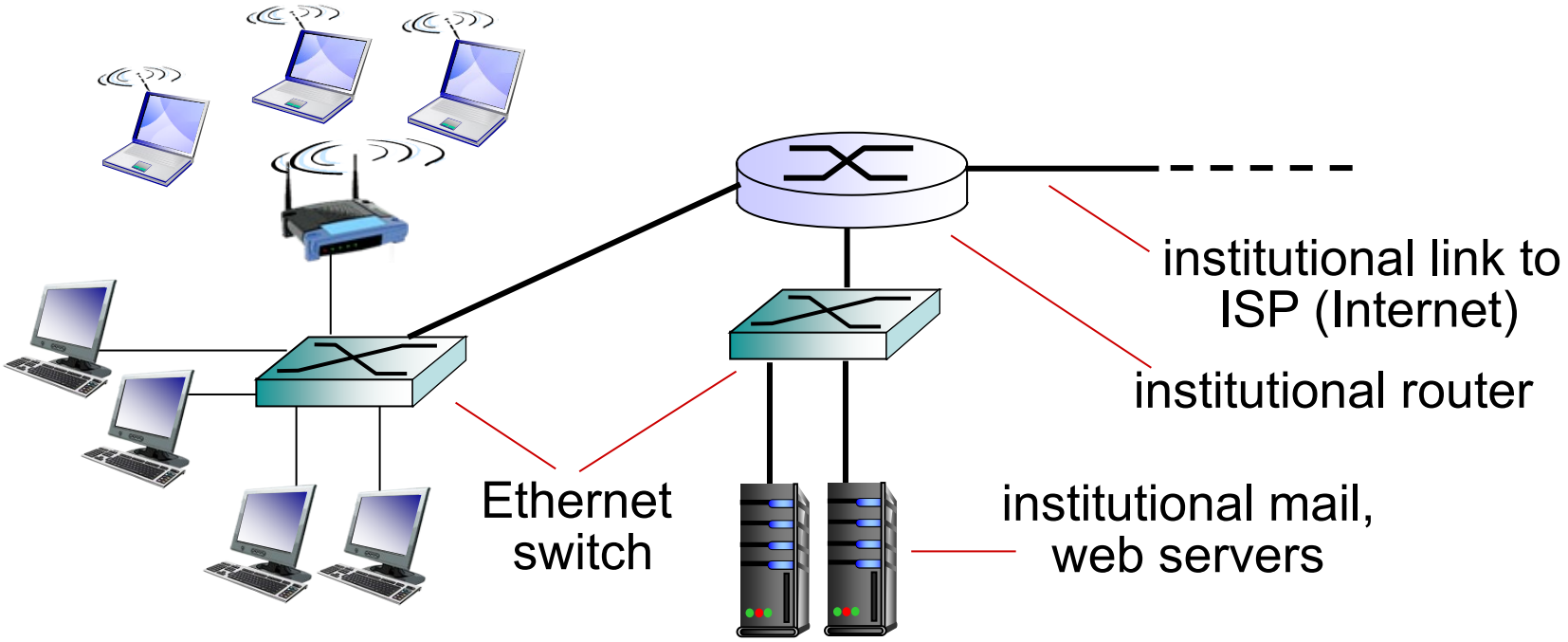
Tutoring session,

Wil Johnson <wijohnso@ucsc.edu>

Tuesday and Thursday 4-6pm

https://ucsc.zoom.us/j/81468111690?pwd=RVhCNmpnTW8vS1pMQmM4NGdvb01Ndz09

Enterprise access networks (Ethernet)



- typically used in companies, universities, etc
- 10 Mbps, 100Mbps, 1Gbps, 10Gbps, 40Gbps transmission rates
- today, end systems typically connect into Ethernet switch

Wireless access networks

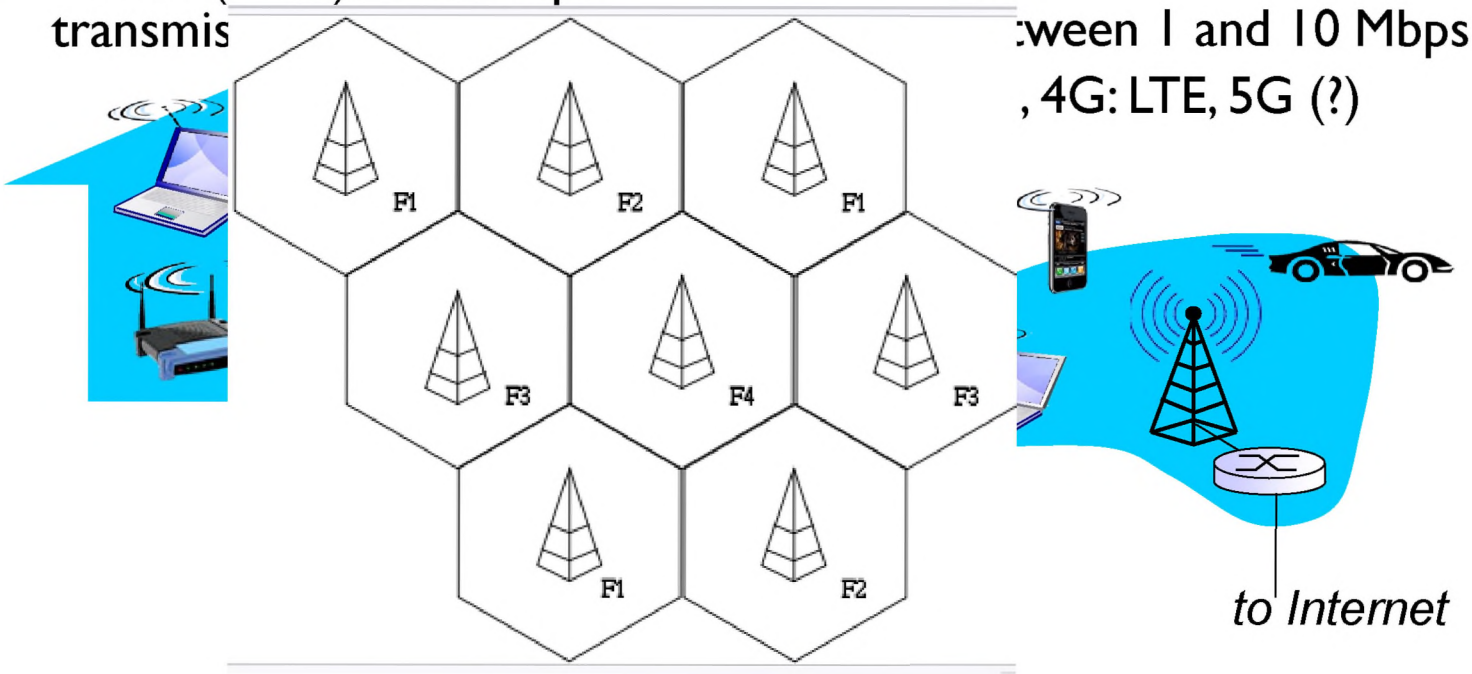
- shared *wireless* access network connects end system to router
  - via base station aka "access point"

wireless LANs:

- within building (100 ft)
- 802.11 (WiFi): >200Mbps transmis

wide-area wireless access

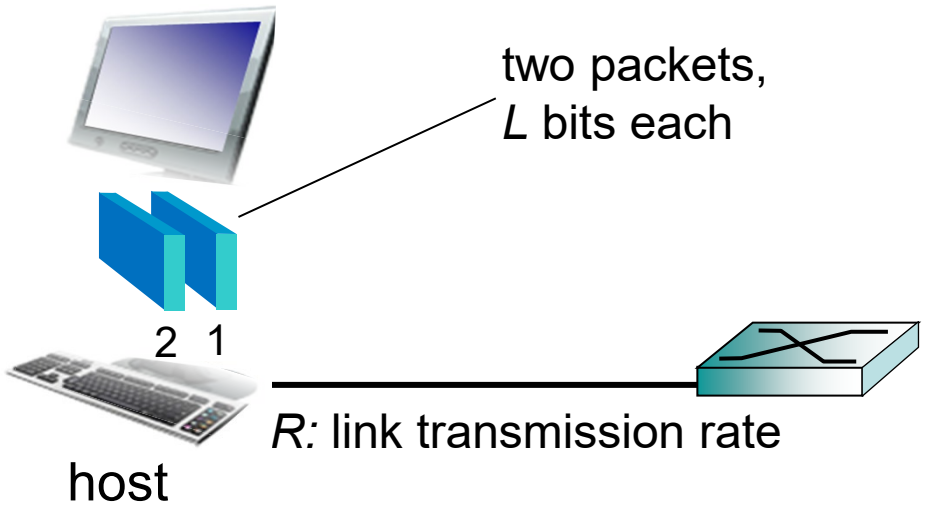
- provided by telco (cellular) operator, >10 km
- :ween 1 and 10 Mbps
- , 4G: LTE, 5G (?)



Host: sends *packets* of data

host sending function:

- takes application message
- breaks into smaller chunks, known as *packets*, of length *L* bits
- transmits packet into access network at *transmission rate R*
  - link transmission rate, aka link *capacity, aka link bandwidth*



$$\text{packet transmission delay} = \text{time needed to transmit } L\text{-bit packet into link} = \frac{L \text{ (bits)}}{R \text{ (bits/sec)}}$$

Chapter 1: roadmap

1.1 what *is* the Internet?

1.2 network edge

- end systems, access networks, links

1.3 network core

- packet switching, circuit switching, network structure

1.4 delay, loss, throughput in networks

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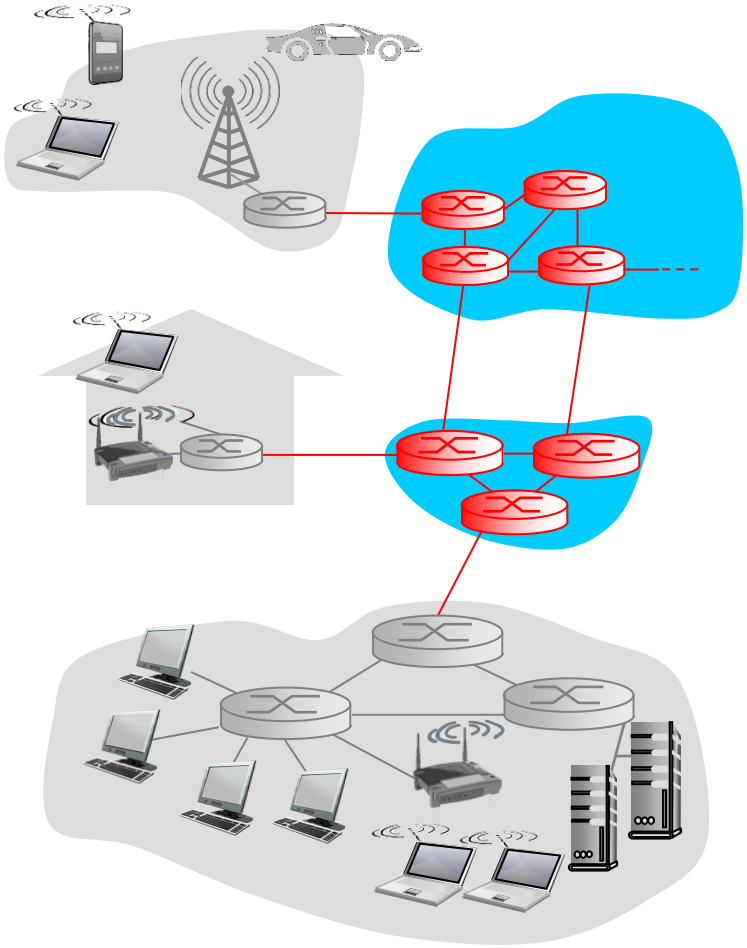
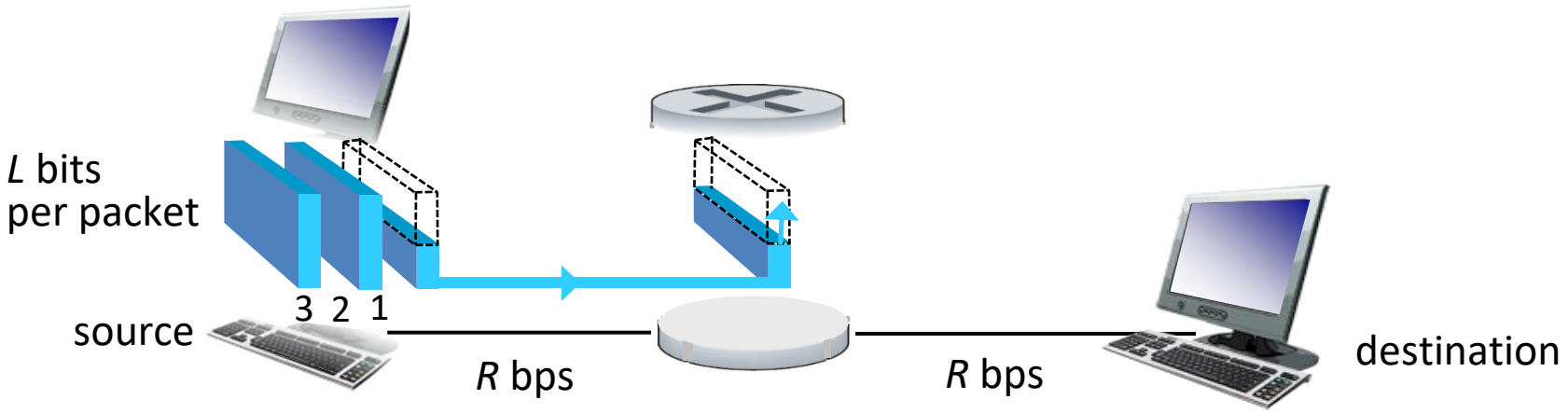
1.6 networks under attack: security

The network core

- mesh of interconnected routers
- https://www.youtube.com/watch?v=yU9oMOcRsuE
- 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



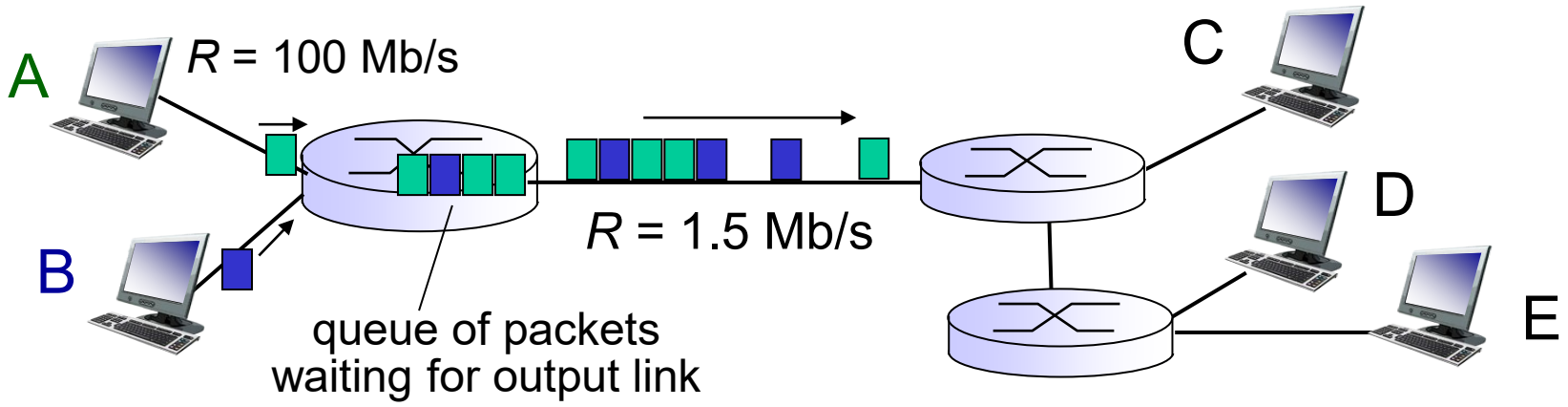
- takes *L*/*R* seconds to transmit (push out) *L*-bit packet into link at *R* bps
- *store and forward:* entire packet must arrive at router before it can be transmitted on next link
- end-end delay = 2*L*/*R* (assuming zero propagation delay)

one-hop numerical example:

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

more on delay shortly …

Packet Switching: queueing delay, loss

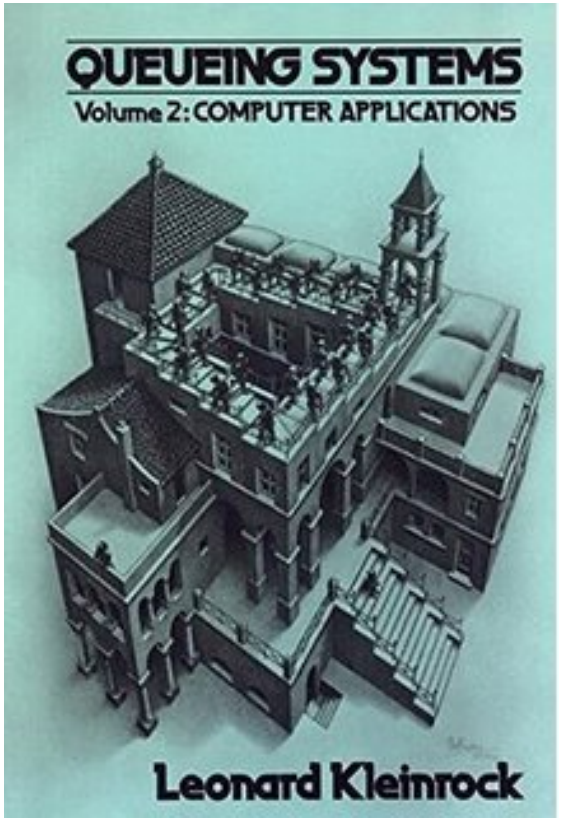


queuing and loss:

- 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

Mathematical background

Queuing theory:



Whenever $V(I) > 0$, then the system is said to be busy, and only when $V(I) = 0$ is the system said to be idle. The duration and location of these busy and idle periods are also quantities of interest.

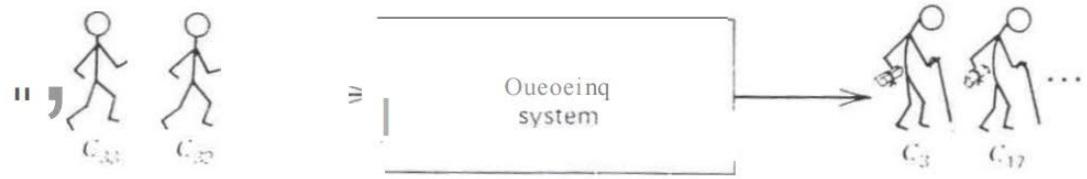


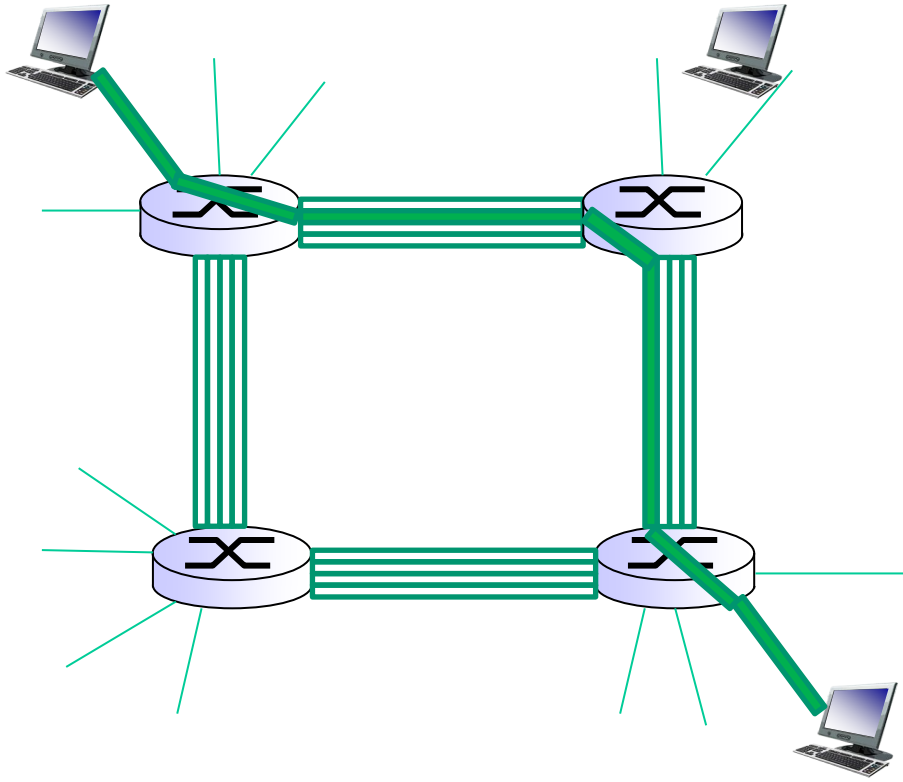
Figure 2.1 A general queueing system.

• The notation $\triangleq$ is to be read as "equals by definition."

Alternative core: circuit switching

end-end resources allocated to, reserved for "call" between source & dest:

- dedicated resources: no sharing
  - circuit-like (guaranteed) performance
- circuit segment idle if not used by call *(no sharing)*
- Commonly used in traditional telephone networks

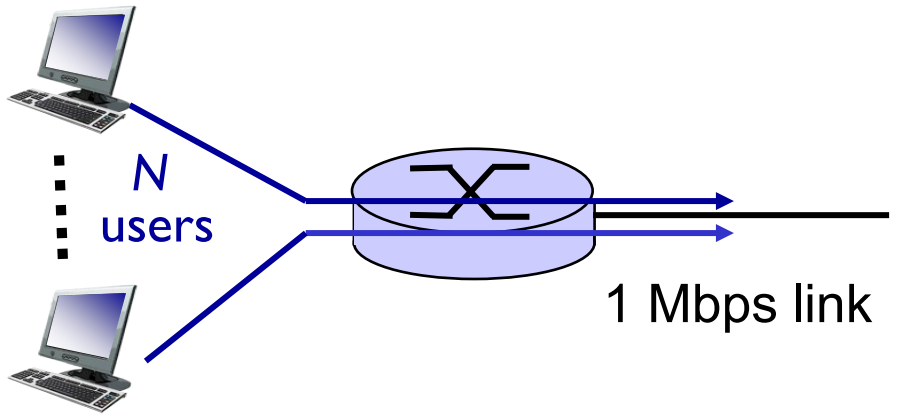


Packet switching versus circuit switching

packet switching allows more users to use network!

example:

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



- *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?

Q: what happens if > 35 users ?

Packet switching versus circuit switching

is packet switching a "slam dunk winner?"

- great for bursty data
  - resource sharing
  - simpler, no call setup
- excessive congestion possible: packet delay and loss
  - protocols needed for reliable data transfer, congestion control

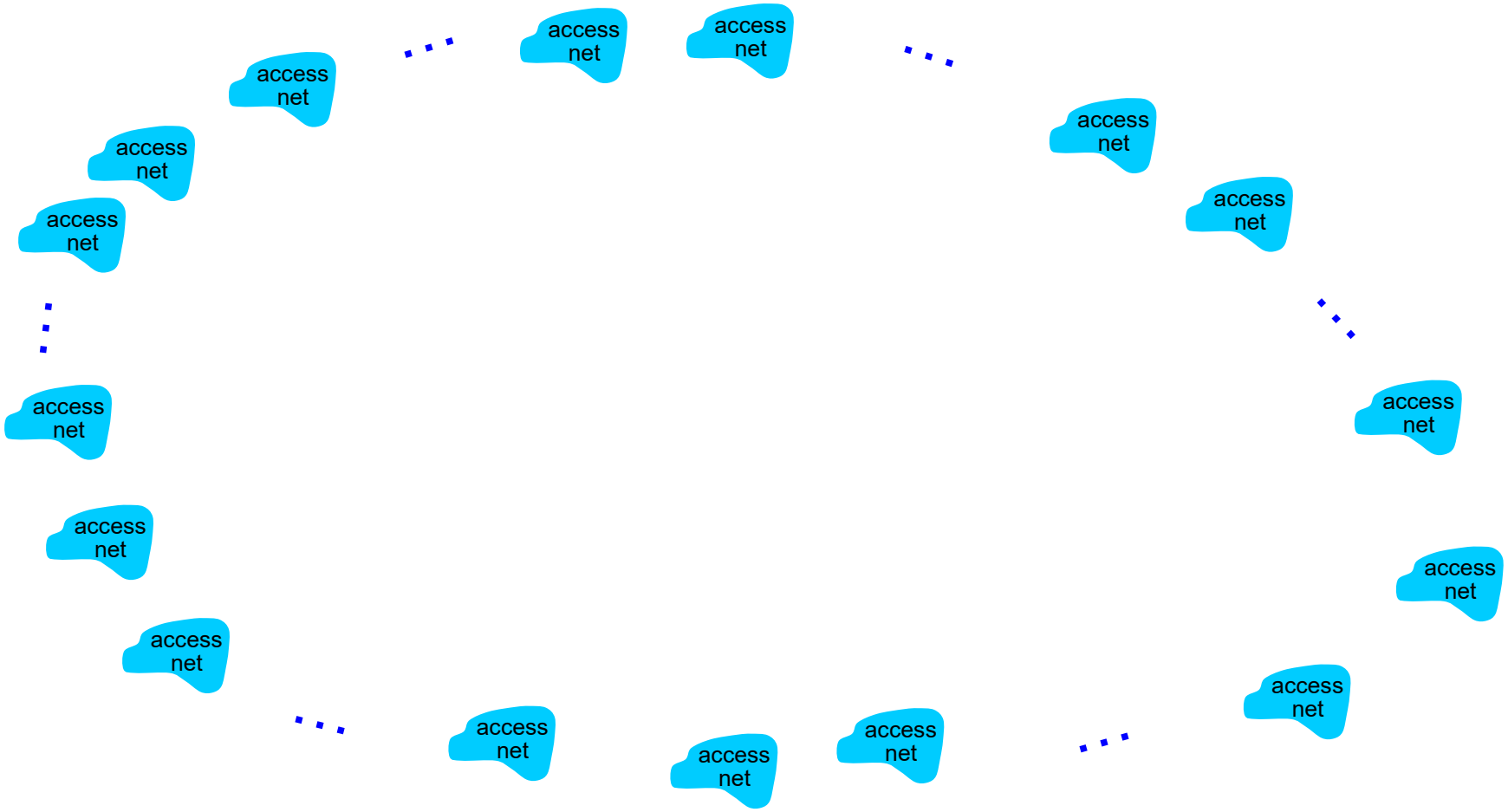
Q: human analogies of reserved resources (circuit switching) versus on-demand allocation (packet-switching)?

Internet structure: network of networks

- End systems connect to Internet via access ISPs (Internet Service Providers)
  - Residential, company and university ISPs
- Access ISPs in turn must be interconnected.
  - So that any two hosts can send packets to each other
- Resulting network of networks is very complex
  - Evolution was driven by economics and national policies

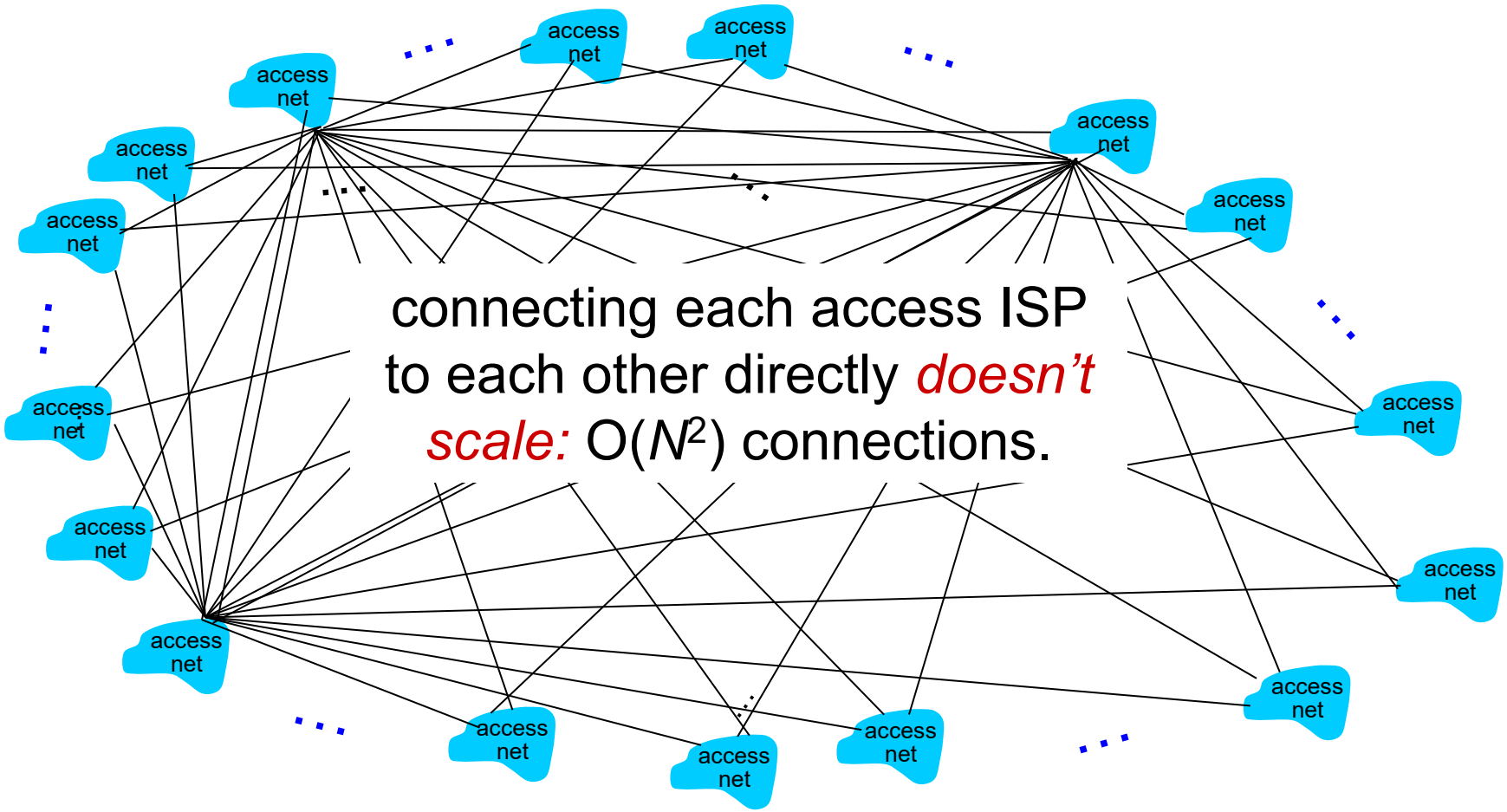
Internet structure: network of networks

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



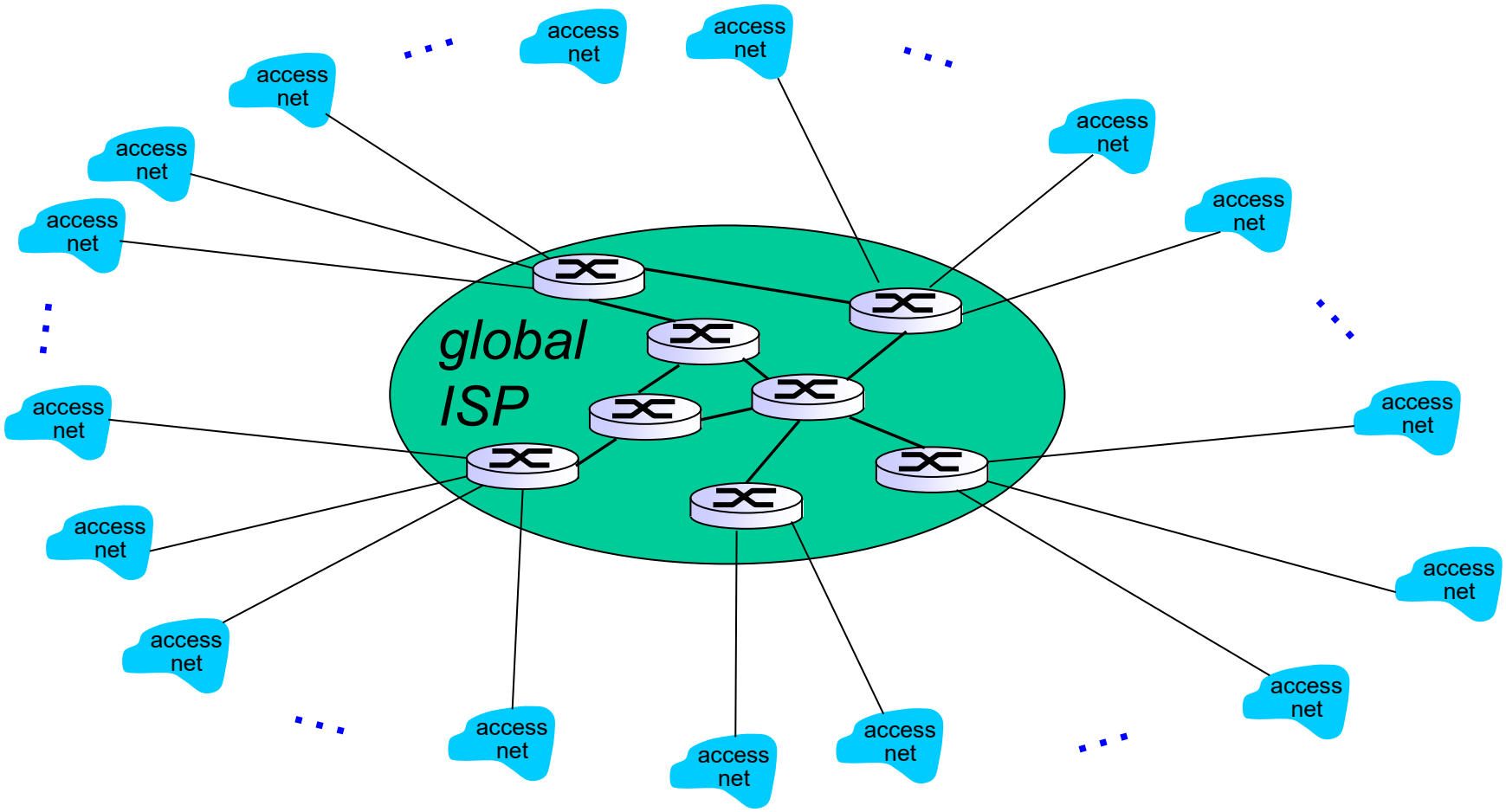
Internet structure: network of networks

Option: connect each access ISP to every other access ISP?



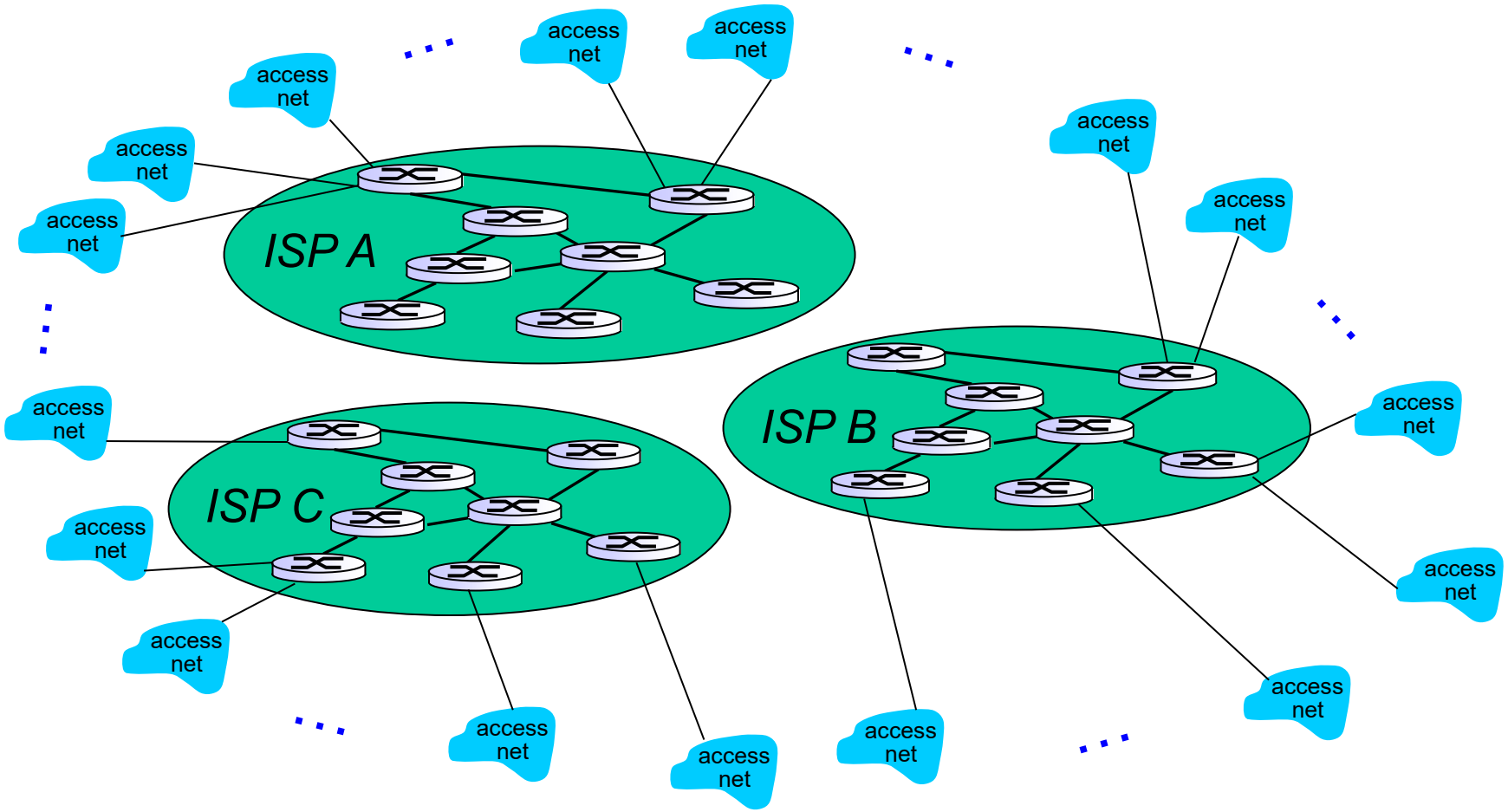
Internet structure: network of networks

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



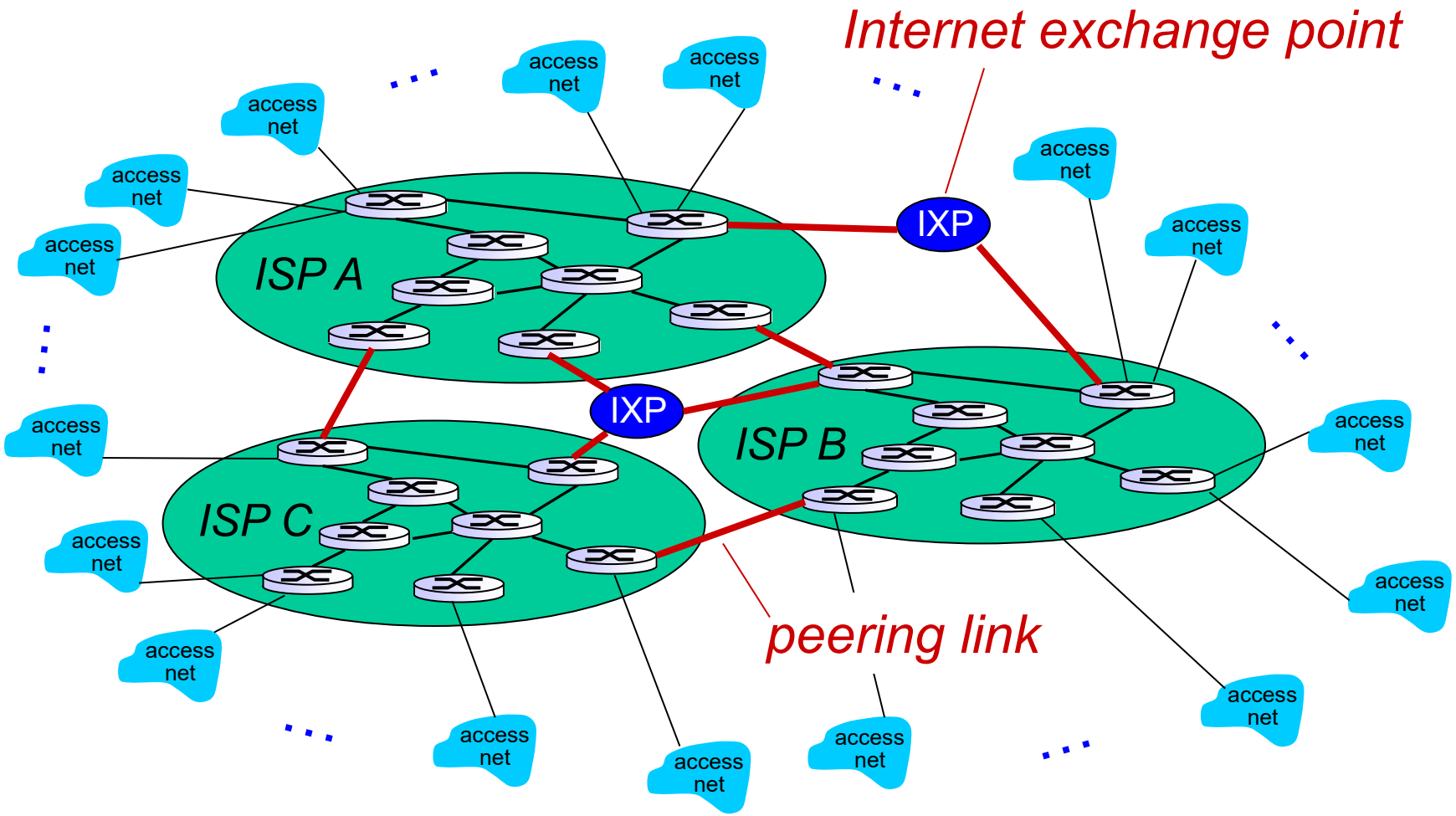
Internet structure: network of networks

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



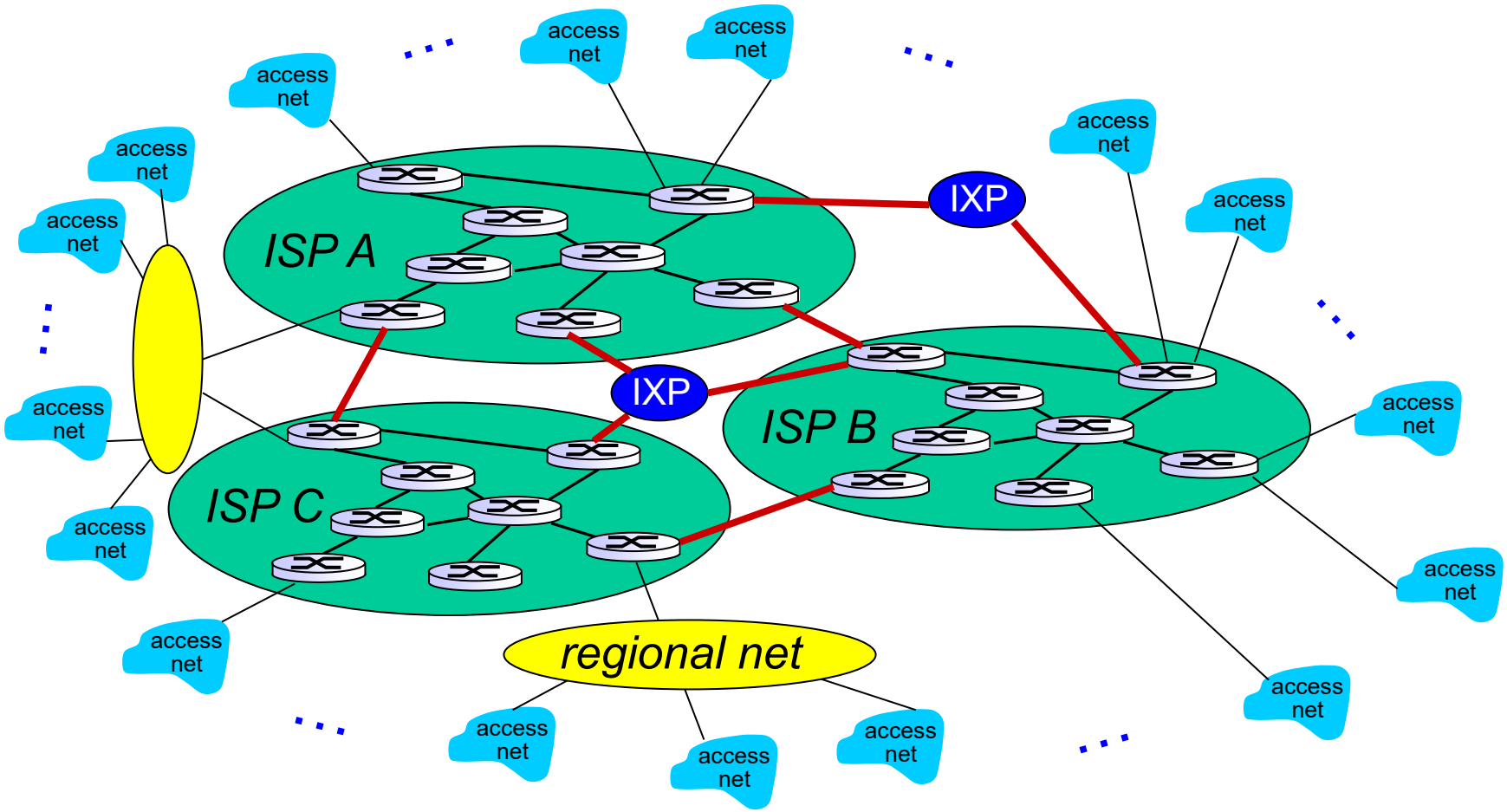
Internet structure: network of networks

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



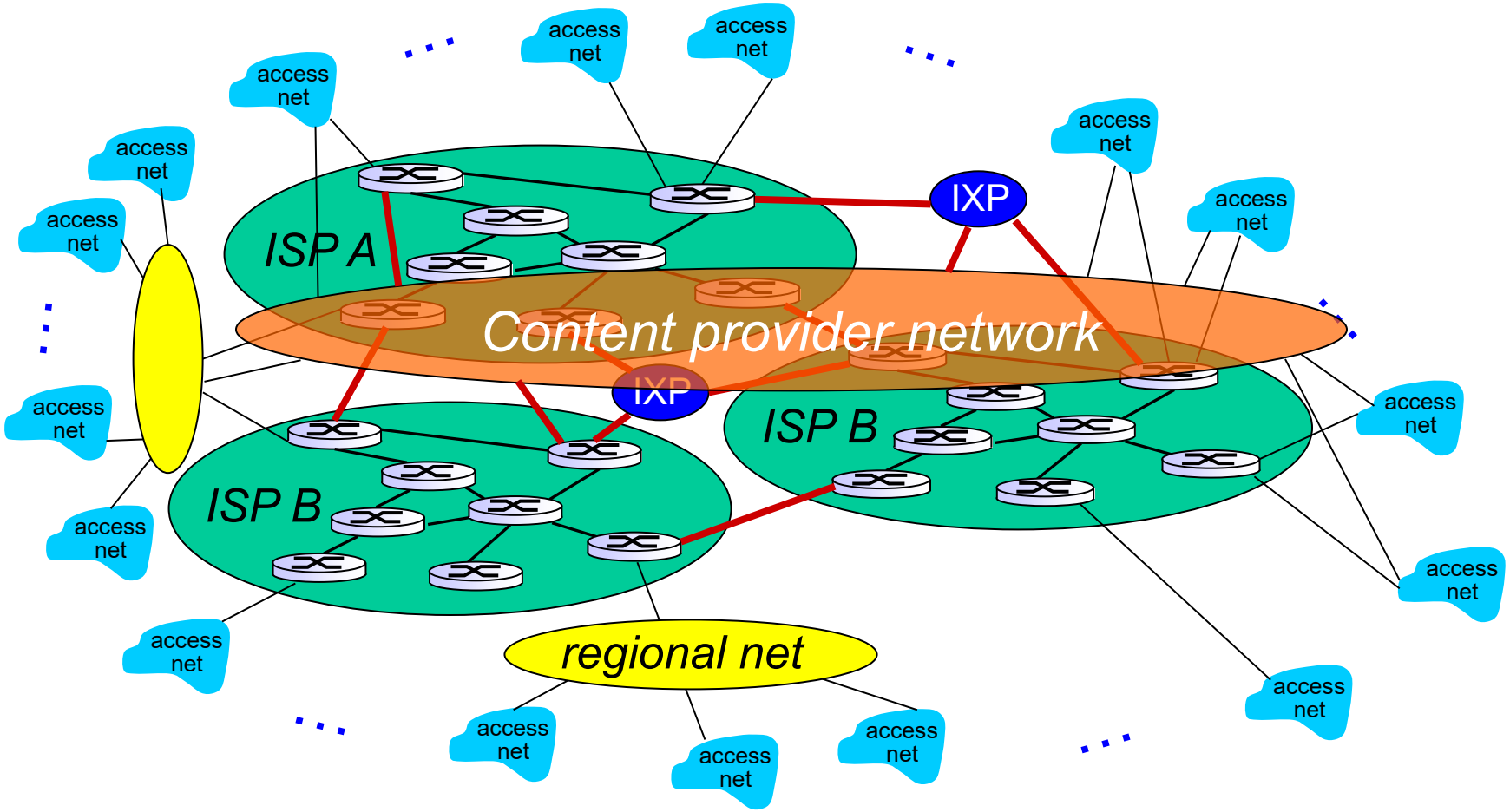
Internet structure: network of networks

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

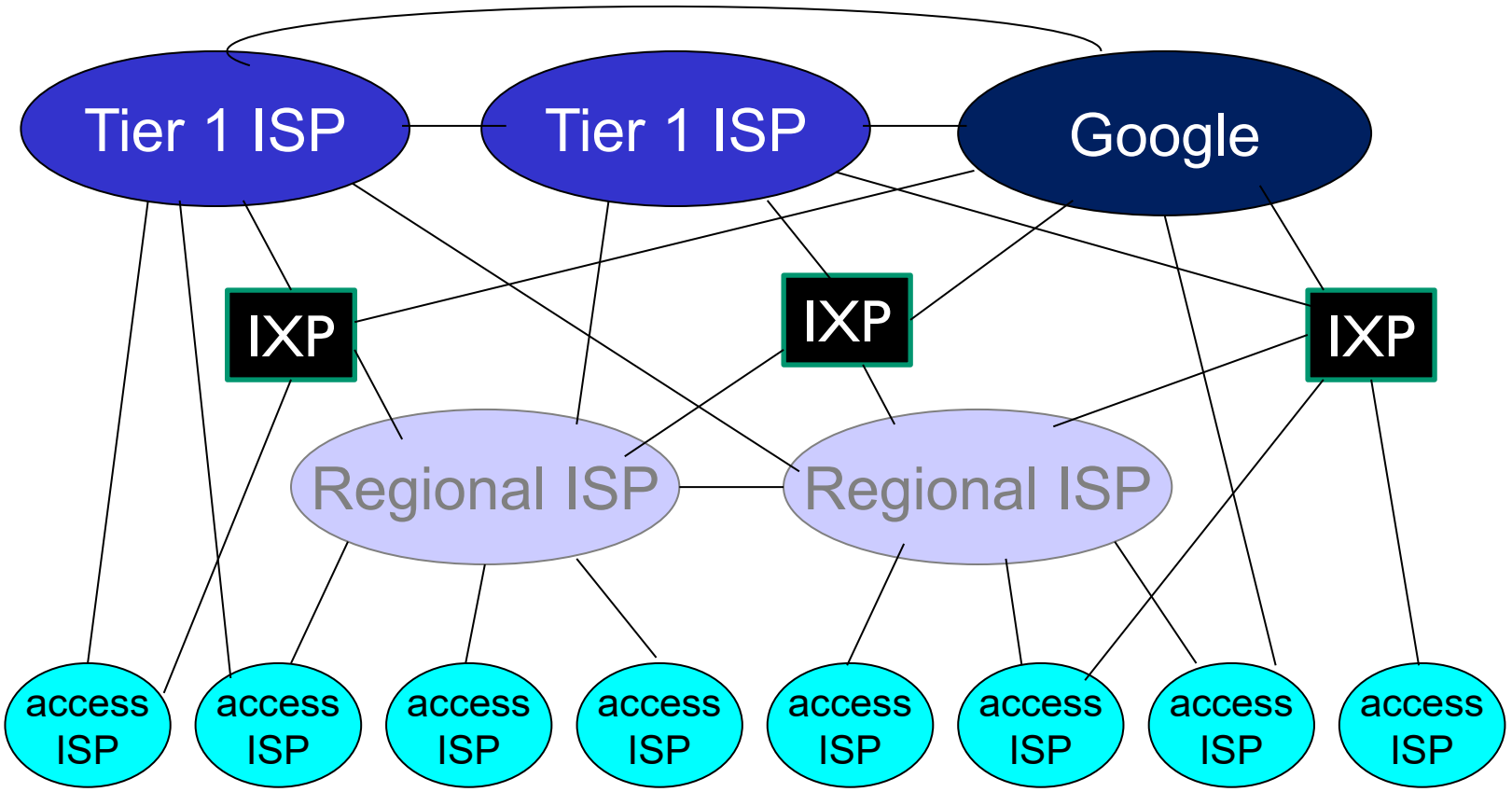


Internet structure: network of networks

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



Internet structure: 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 network (e.g, Google): private network that connects it data centers to Internet, often bypassing tier-1, regional ISPs

Top Internet Service Provider State-by-State

Source: 56 million web visits

WebpageFX
Webpage FX

Leonard Kleinrock talks about packet switch vs. circuit switching.

https://www.youtube.com/watch?v=rHHpwcZiEW4

Time: 2:55 – 7:00

- Units in networking (and computing!)
- 1b: 1 bit
- 1B: 1 byte
- 1 byte = 8 bits
- 1 Kbps = 1 kilobit per second =1000 bits per second
- 1 Mbps = 1,000,000 bits per second
  = 1000 Kbps

Chapter 1: roadmap

1.1 what *is* the Internet?

1.2 network edge

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1.4 delay, loss, throughput in networks

1.5 protocol layers, service models

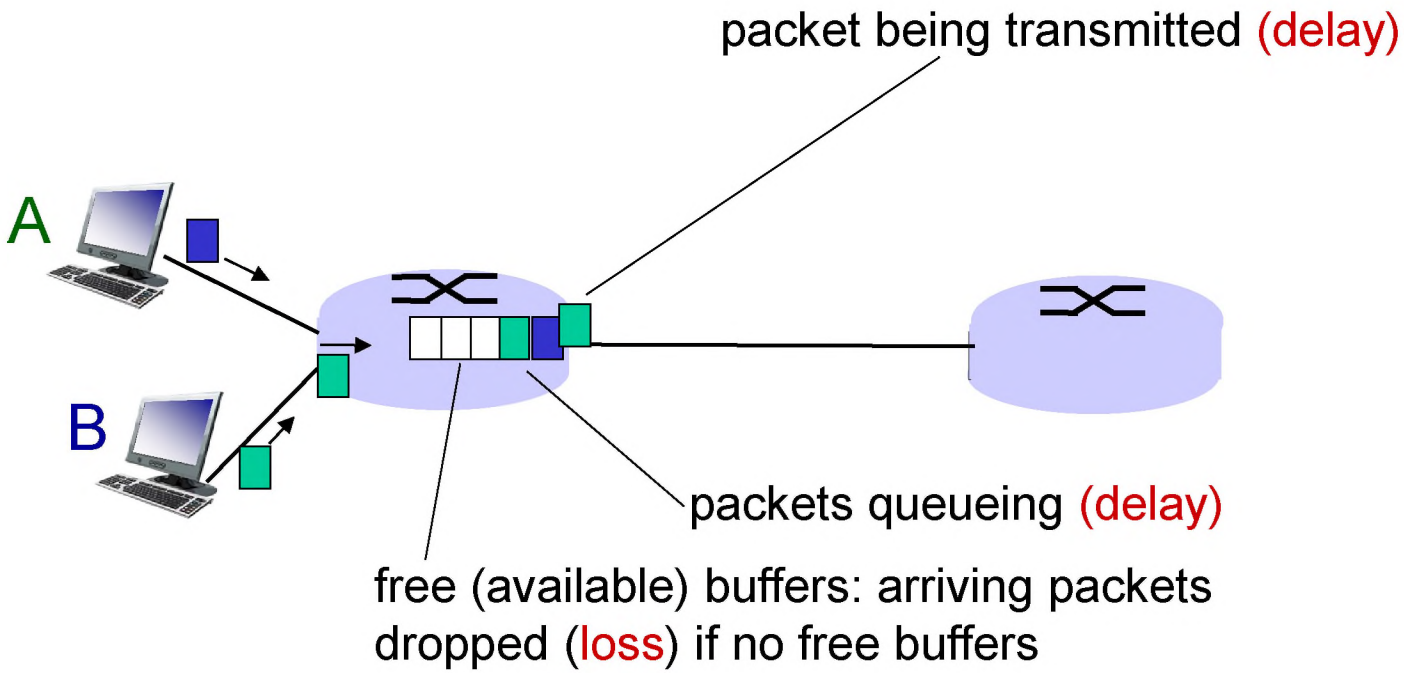
1.6 networks under attack: security

1.7 history

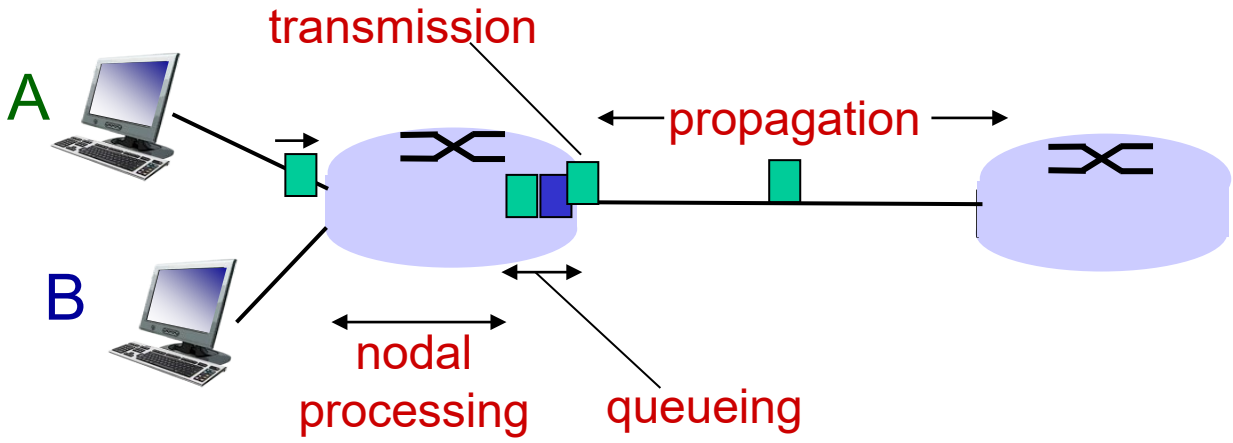
How do loss and delay occur?

packets *queue* in router buffers

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



Four sources of packet delay



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

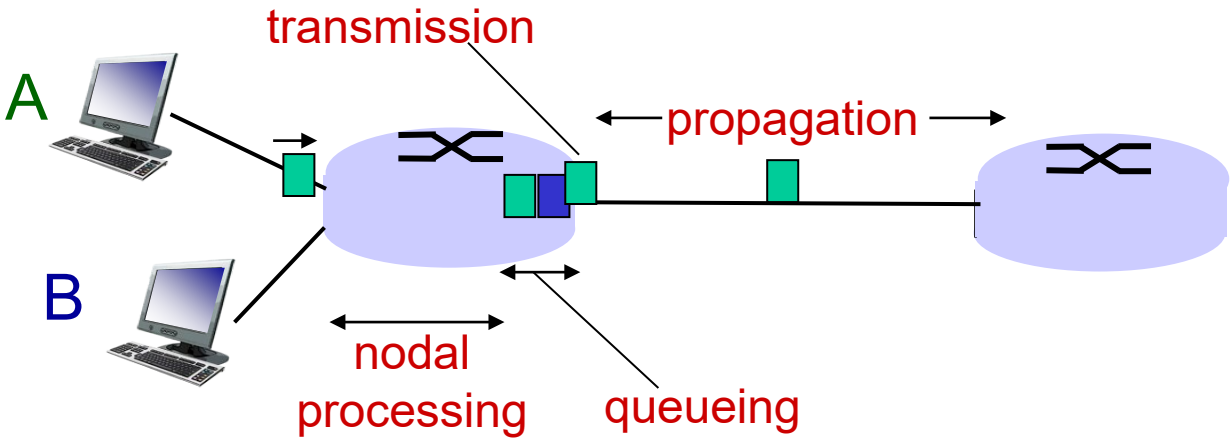
d_{proc}: nodal processing

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

d_{queue}: queueing delay

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

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_{prop}: propagation delay:

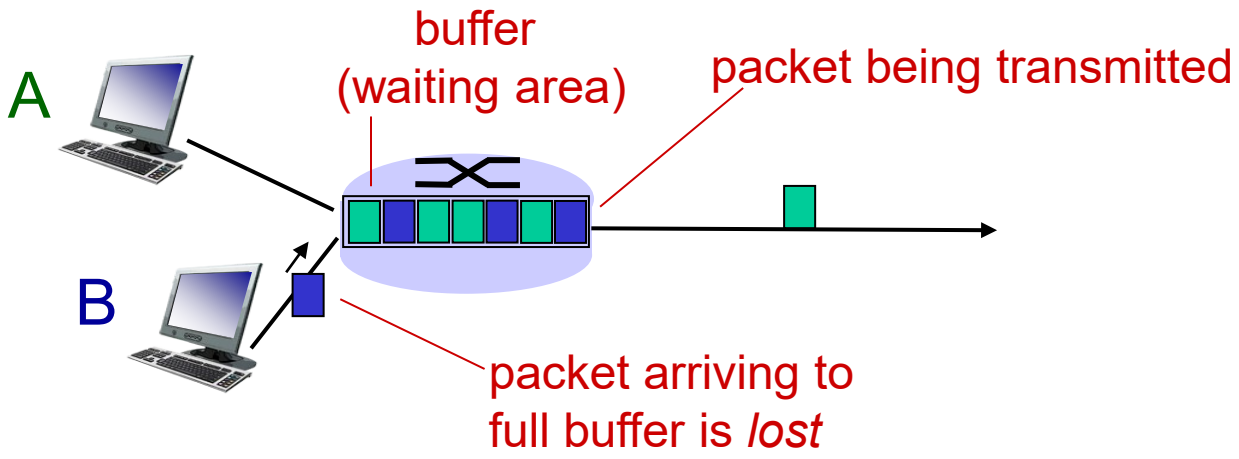
- *d*: length of physical link
- *s*: propagation speed in medium (~2×10^8 m/sec)
- $d_{prop} = d/s$

Analogy using Mailing network

- Transmission delay:
  - Time to write a mail
- Propagation delay
  - Post truck carrying the mail moves to the next post office
- Processing delay
  - Post officers distribute the mail to the right box for next station
- Queuing delay
  - Mails waiting in the post office for processing

Packet loss

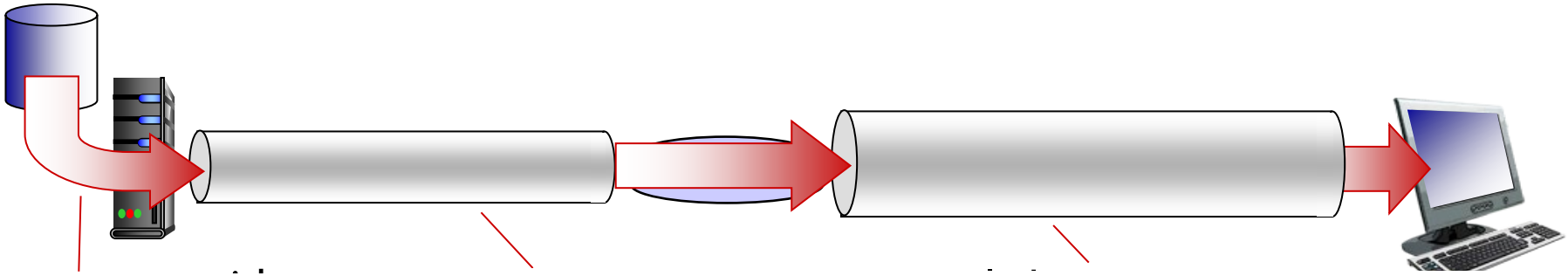
- queue (aka buffer) has finite capacity
- packet arriving to full queue dropped (aka lost)
- lost packet may be retransmitted by previous node, by source end system, or not at all



Animation of queuing

Throughput

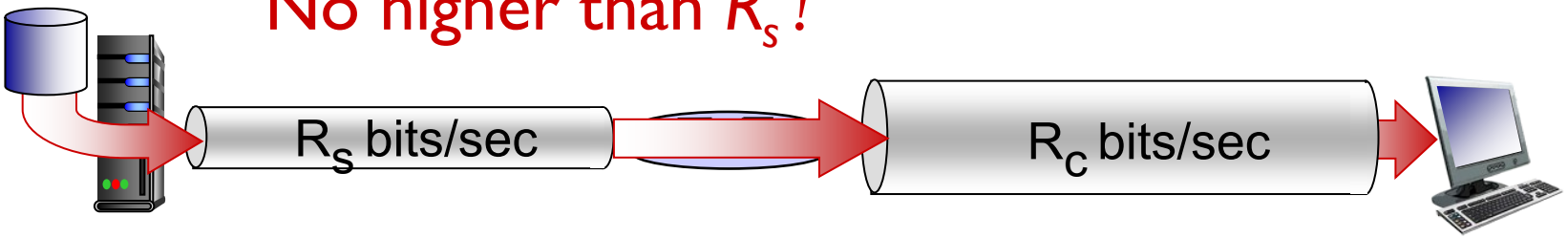
- *throughput:* rate (bits/time unit) at which bits transferred between sender/receiver
  - *instantaneous:* rate at given point in time
  - *average:* rate over longer period of time



Throughput (more)

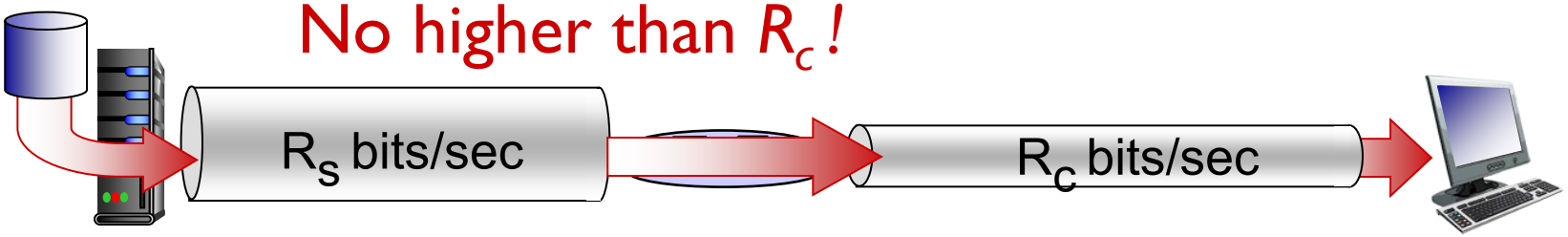
- $R_s < R_c$ What is average end-end throughput?

No higher than R_s !



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

No higher than R_c !

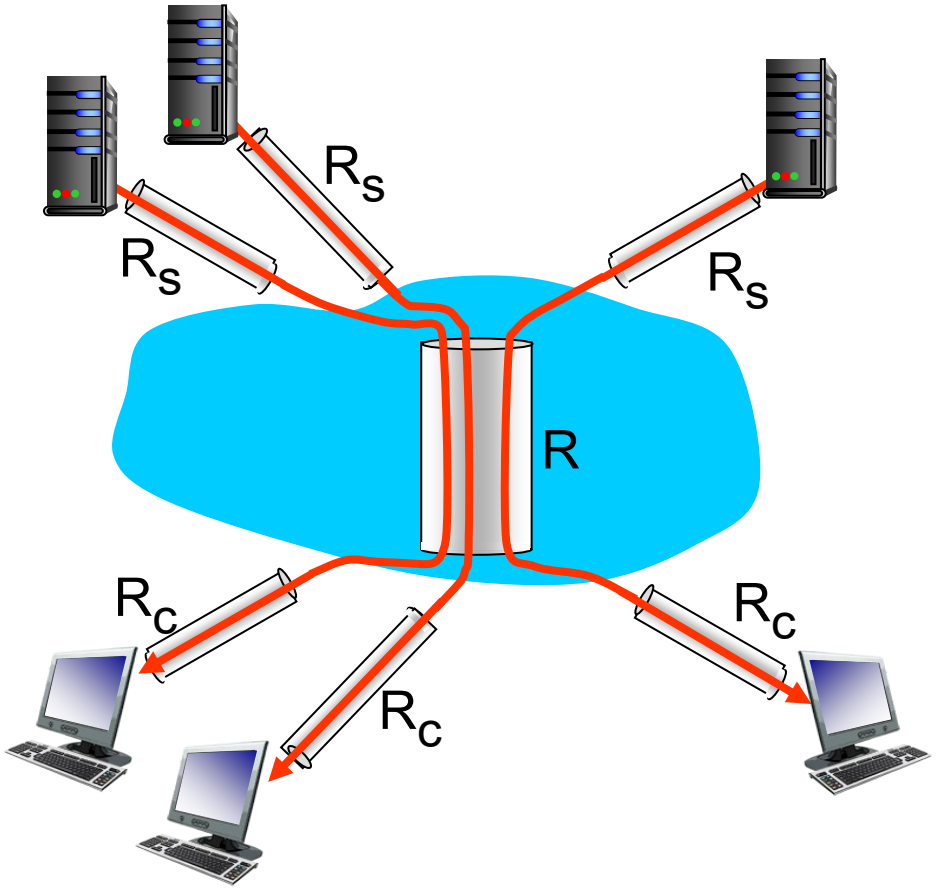


bottleneck link

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

Chapter 1: roadmap

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Protocol "layers"

Networks are complex, with many "pieces":

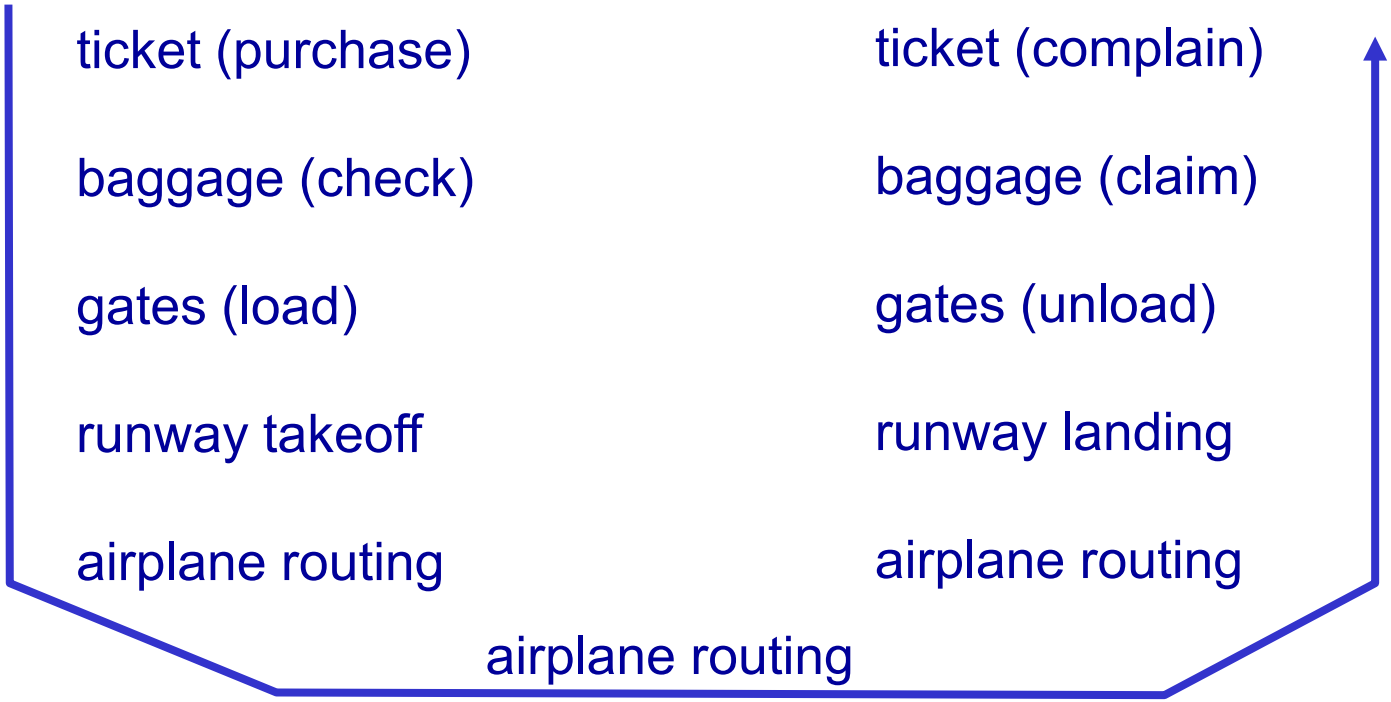
- hosts
- routers
- links of various media
- applications
- protocols
- hardware, software

Question:

is there any hope of *organizing* structure of network?

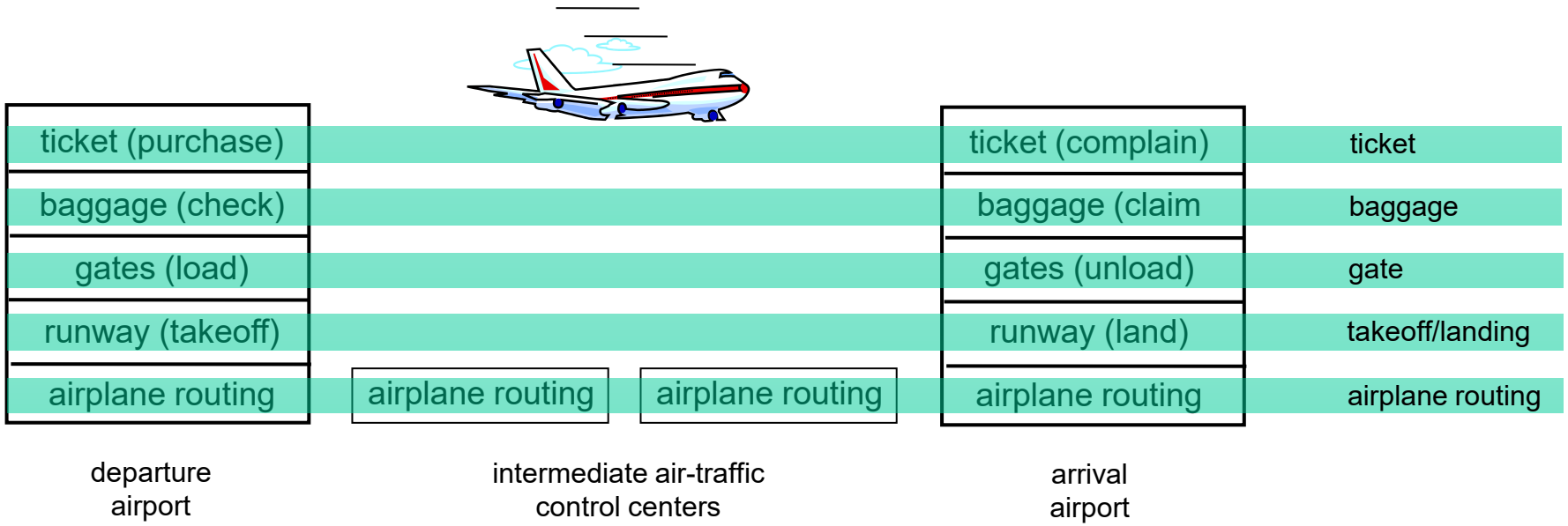
…. or at least our discussion of networks?

Organization of air travel



- a series of steps

Layering of airline functionality



layers: each layer implements a service

- via its own internal-layer actions
- relying on services provided by layer below

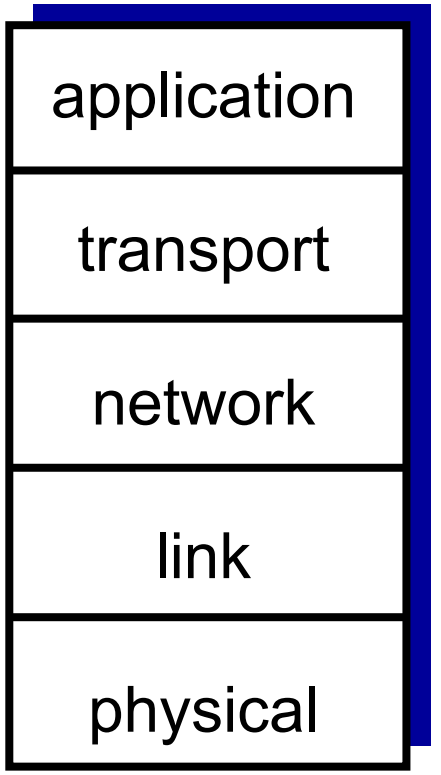
Why layering?

dealing with complex systems:

- explicit structure allows identification, relationship of complex system's pieces
  - layered *reference model* for discussion
- modularization eases maintenance, updating of system
  - change of implementation of layer's service transparent to rest of system
  - e.g., change in gate procedure doesn't affect rest of system
- layering considered harmful?

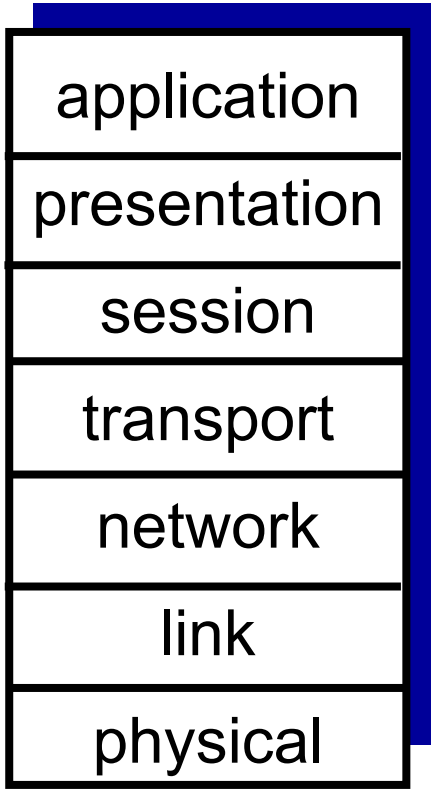
Internet protocol stack

- *application:* supporting network applications
  - FTP, SMTP, HTTP
- *transport:* process-process data transfer
  - TCP, UDP
- *network:* routing of datagrams from source to destination
  - IP, routing protocols
- *link:* data transfer between neighboring network elements
  - Ethernet, 802.11 (WiFi), PPP
- *physical:* bits "on the wire"

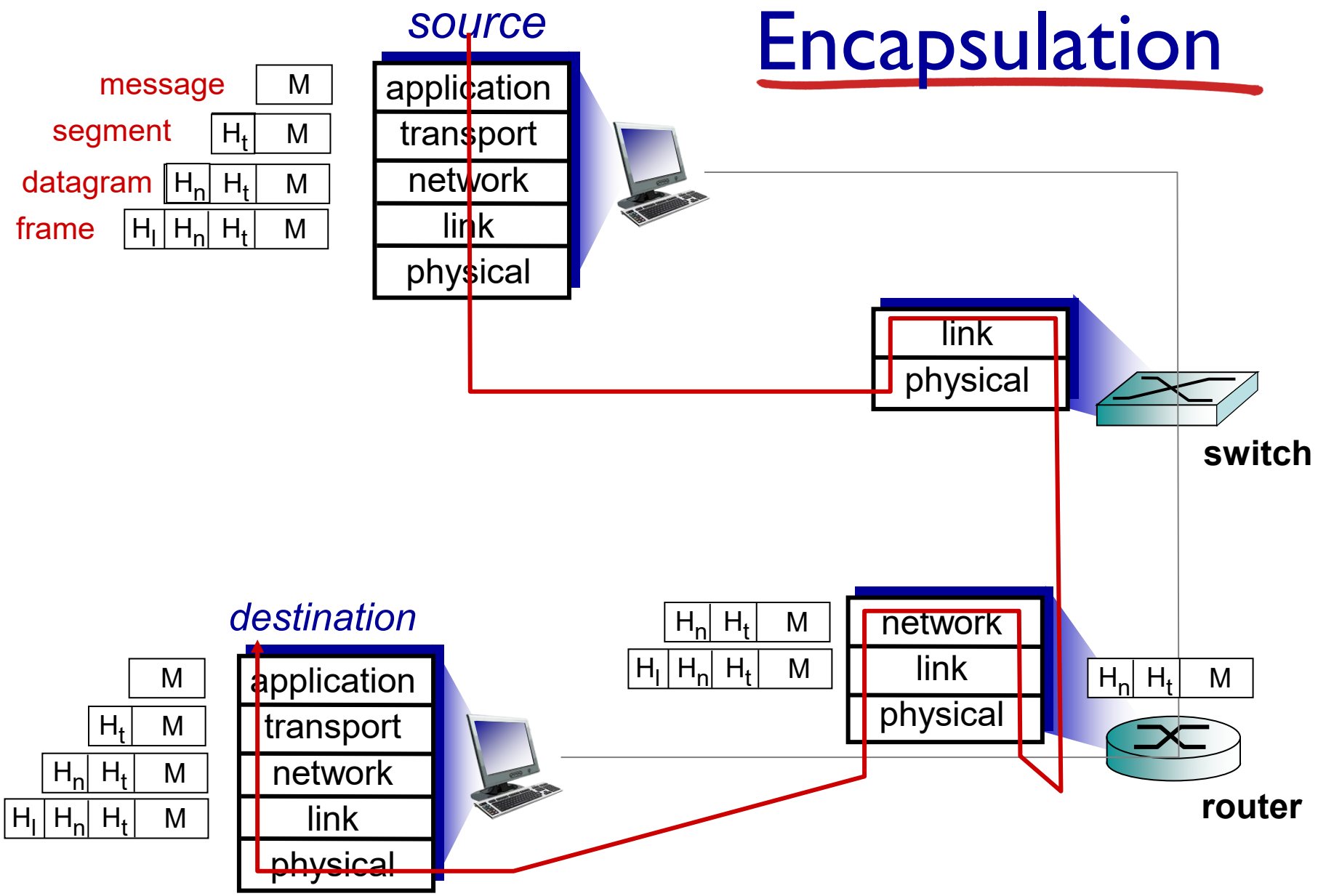


ISO/OSI reference model

- *presentation:* allow applications to interpret meaning of data, e.g., encryption, compression, machine-specific conventions
- *session:* synchronization, checkpointing, recovery of data exchange
- Internet stack "missing" these layers!
  - these services, *if needed*, must be implemented in application
  - needed?



Encapsulation



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Network security

- field of network security:
  - how bad guys can attack computer networks
  - how we can defend networks against attacks
  - how to design architectures that are immune to attacks
- Internet not originally designed with (much) security in mind
  - *original vision:* "a group of mutually trusting users attached to a transparent network" ☺
  - Internet protocol designers playing "catch-up"
  - security considerations in all layers!

Bad guys: put malware into hosts via Internet

- malware can get in host from:
  - *virus:* self-replicating infection by receiving/executing object (e.g., e-mail attachment)
  - *worm:* self-replicating infection by passively receiving object that gets itself executed
- spyware malware can record keystrokes, web sites visited, upload info to collection site
- infected host can be enrolled in botnet, used for spam. DDoS attacks

Robert Tappan Morris

- Son of Robert Morris Sr., chief scientist at National Security Agency (NSA)

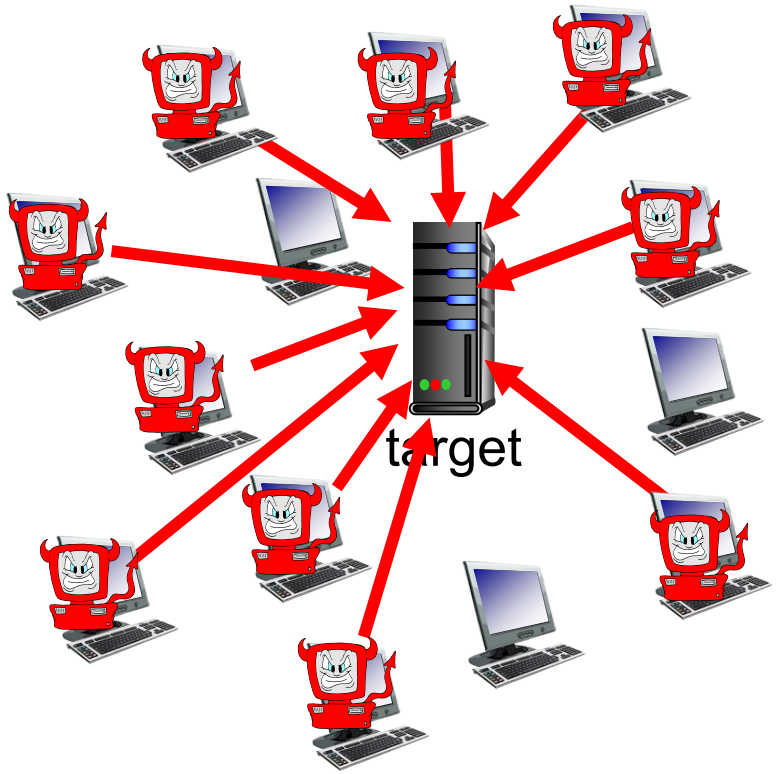
Robert Tappan Morris

- Developed the first worm on the Internet in1988, while he was a graduate student at Cornell University.
  - He said it was designed to gauge the size of the Internet.
  - released the worm from MIT, rather than Cornell.
  - Caused $10M - $100M loss
  - Sentenced to three years of probation, 400 hours of community service
- A Computer Science professor at MIT since 1999. And a member of National Academy of Engineering.
  - Not because of his worm!!

Bad guys: attack server, network infrastructure

Denial of Service (DoS): attackers make resources (server, bandwidth) unavailable to legitimate traffic by overwhelming resource with bogus traffic

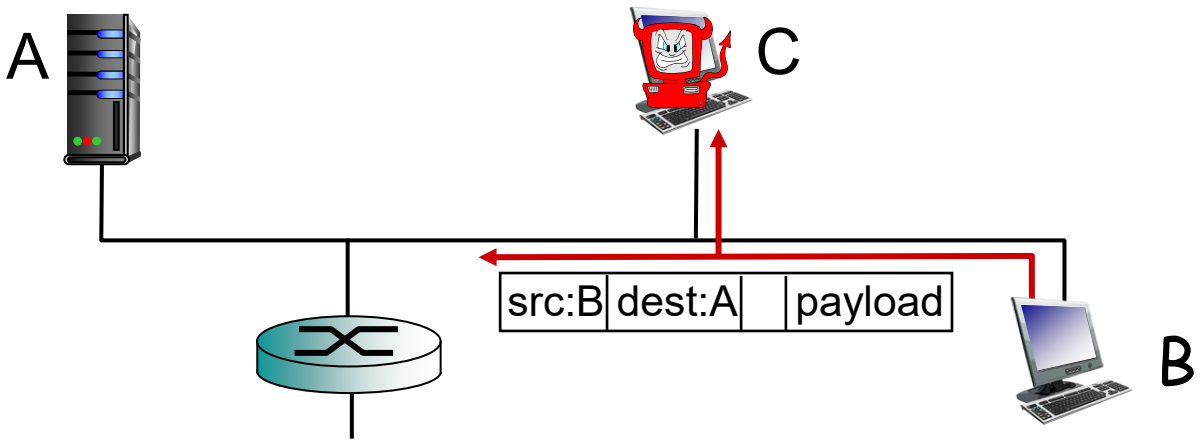
1. select target
2. break into hosts around the network (see botnet)
3. send packets to target from compromised hosts



Bad guys can sniff packets

packet "sniffing":

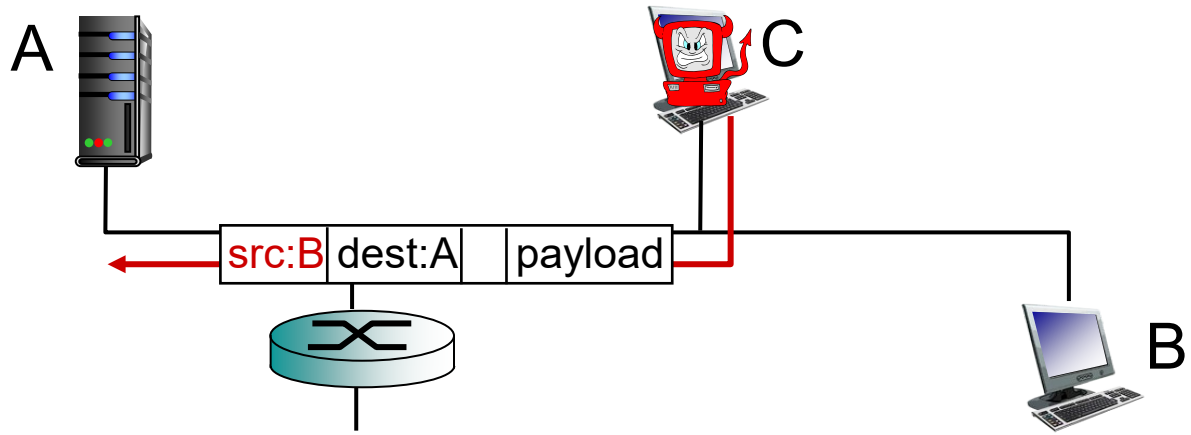
- broadcast media (shared ethernet, wireless)
- promiscuous network interface reads/records all packets (e.g., including passwords!) passing by



- wireshark software used for labs is a (free) packet-sniffer

Bad guys can use fake addresses

IP spoofing: send packet with false source address



… lots more on security (throughout, Chapter 8)

CSE 150 : Introduction to Computer Networks

Chen Qian
Computer Science and Engineering
UCSC Baskin Engineering

Chapter 2

Course evaluation

1 bonus points for your grade

Upload a screenshot showing you have completed the course and teaching evaluation before the final exam.

Don't include your responses and other private information in the screenshot.

Chapter 2 Application layer: outline

2.1 principles of network applications

2.2 Web and HTTP

2.4 electronic mail

- SMTP, POP3, IMAP

2.5 DNS

2.6 P2P applications

2.7 socket programming with UDP and TCP

Chapter 2: application layer

our goals:

- conceptual, implementation aspects of network application protocols
  - transport-layer service models
  - client-server paradigm
  - peer-to-peer paradigm
- learn about protocols by examining popular application-level protocols
  - HTTP
  - FTP
  - SMTP / POP3 / IMAP
  - DNS
- creating network applications
  - socket API

Some network apps

- e-mail
- web
- text messaging
- remote login
- P2P file sharing
- multi-user network games
- streaming stored video (YouTube, Hulu, Netflix)
- voice over IP (e.g., Skype)
- real-time video conferencing
- social networking
- search
- …
- …

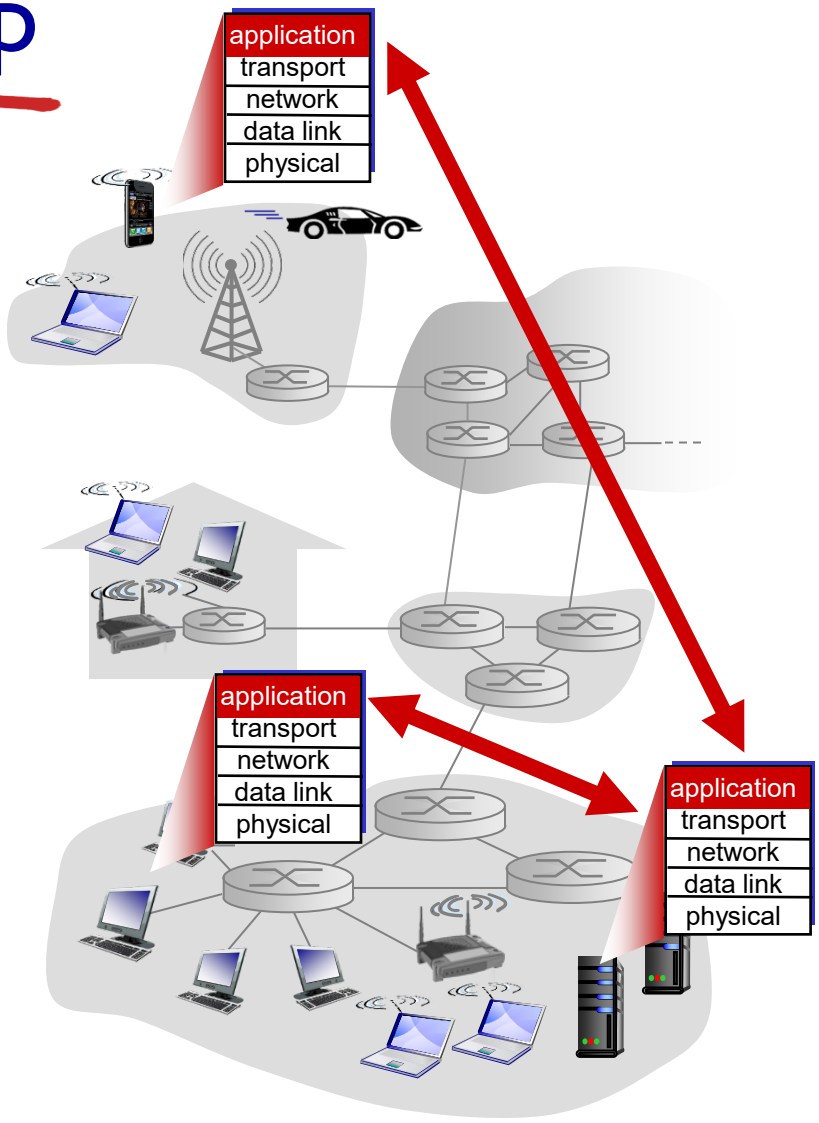
Creating a network app

write programs that:

- run on (different) *end systems*
- communicate over network
- e.g., web server software communicates with browser software

no need to write software for network-core devices

- network-core devices do not run user applications
- applications on end systems allows for rapid app development, propagation

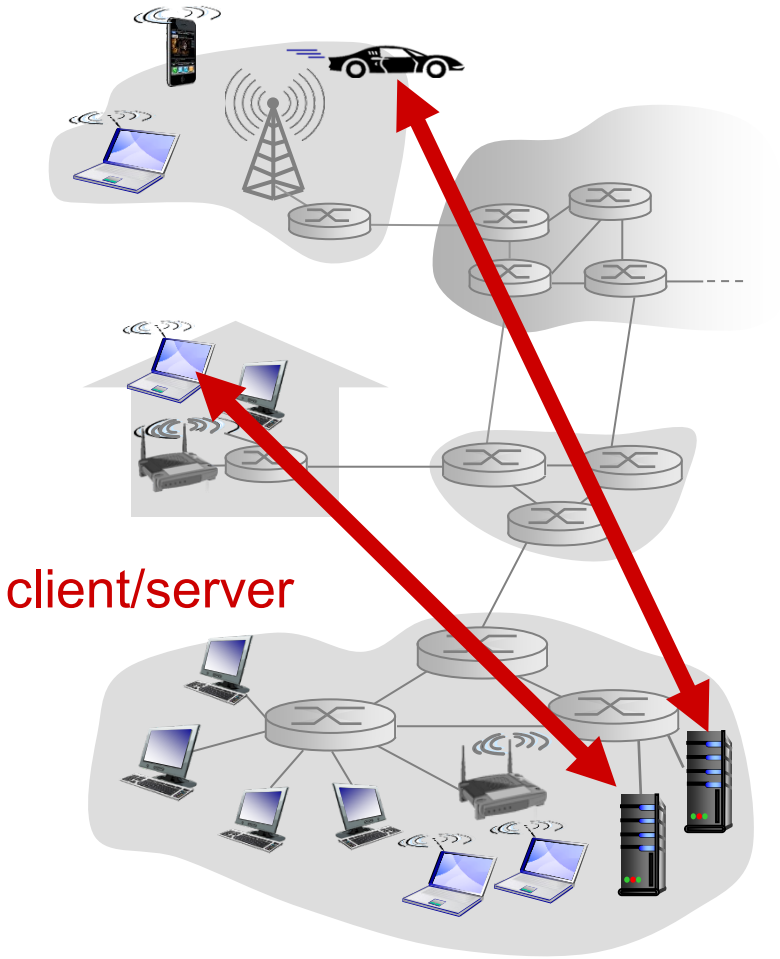


Application architectures

possible structure of applications:

- client-server
- peer-to-peer (P2P)

Client-server architecture



server:

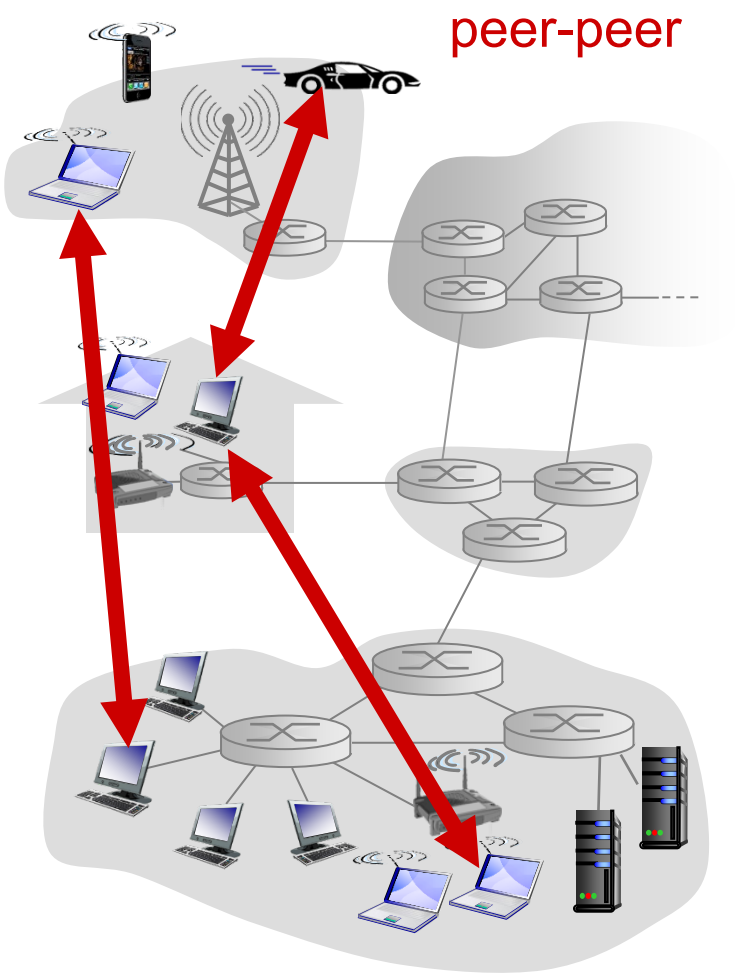
- always-on host
- permanent IP address
- data centers for scaling

clients:

- communicate with server
- may be intermittently connected
- may have dynamic IP addresses
- do not communicate directly with each other

P2P architecture

- *no* always-on server
- arbitrary end systems directly communicate
- peers request service from other peers, provide service in return to other peers
  - *self scalability* – new peers bring new service capacity, as well as new service demands
- peers are intermittently connected and change IP addresses
  - complex management



Client/server versus P2P

- Throughput and Scalability:
- P2P wins!
- Because a server can only serve limited number of clients
- P2P allows clients exchange data among them.
- That's why P2P became popular in early 2000

- Management
- C/S wins!
- Because users in P2P are HIGHLY unreliable.
- In the recent years, throughput are not a big problem, management became the main issue.
- That's why we now switch back to C/S

Hybrid of client-server and P2P

Skype

- voice-over-IP P2P application
- centralized server: finding address of remote party:
- client-client connection: direct (not through server)

Instant messaging

- chatting between two users is (can be) P2P
- centralized service: client presence detection/location
  - user registers its IP address with central server when it comes online
  - user contacts central server to find IP addresses of buddies

Processes communicating

process: program running within a host

- within same host, two processes communicate using inter-process communication (defined by OS)
- processes in different hosts communicate by exchanging messages

clients, servers

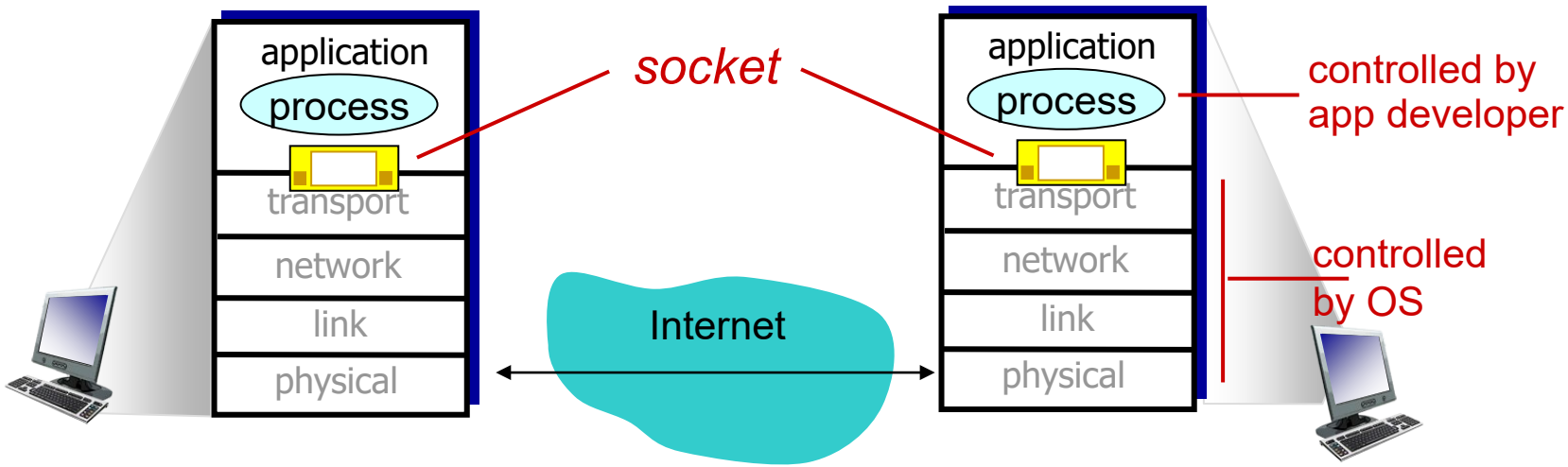
client process: process that initiates communication

server process: process that waits to be contacted

- aside: applications with P2P architectures have client processes & server processes

Sockets

- process sends/receives messages to/from its socket
- socket analogous to door
  - sending process shoves message out door
  - sending process relies on transport infrastructure on other side of door to deliver message to socket at receiving process



Addressing processes

- to receive messages, process must have *identifier*
- host device has unique 32-bit IP address
- *Q:* does IP address of host on which process runs suffice for identifying the process?
  - *A:* no, *many* processes can be running on same host

- *identifier* includes both IP address and port numbers associated with process on host.
- example port numbers:
  - HTTP server: 80
  - mail server: 25
- to send HTTP message to gaia.cs.umass.edu web server:
  - IP address: 128.119.245.12
  - port number: 80

App-layer protocol defines

- types of messages exchanged,
  - e.g., request, response
- message syntax:
  - what fields in messages & how fields are delineated
- message semantics
  - meaning of information in fields
- rules for when and how processes send & respond to messages

open protocols:

- defined in RFCs
- allows for interoperability
- e.g., HTTP, SMTP

proprietary protocols:

- e.g., Skype

What transport service does an app need?

data integrity

- some apps (e.g., file transfer, web transactions) require 100% reliable data transfer
- other apps (e.g., audio) can tolerate some loss

timing

- some apps (e.g., Internet telephony, interactive games) require low delay to be "effective"

throughput

- some apps (e.g., multimedia) require minimum amount of throughput to be "effective"
- other apps ("elastic apps") make use of whatever throughput they get

security

- encryption, data integrity, …

Transport service requirements: common apps

application	data loss	throughput	time sensitive
file transfer	no loss	elastic	no
e-mail	no loss	elastic	no
Web documents	no loss	elastic	no
real-time audio/video	loss-tolerant	audio: 5kbps-1Mbps video:10kbps-5Mbps	yes, 100's msec
stored audio/video	loss-tolerant	same as above	yes, few secs
interactive games	loss-tolerant	few kbps up	yes, 100's msec
text messaging	no loss	elastic	yes and no

Internet transport protocols services

TCP service:

- *reliable transport* between sending and receiving process
- *flow control:* sender won't overwhelm receiver
- *congestion control:* throttle sender when network overloaded
- *does not provide:* timing, minimum throughput guarantee, security
- *connection-oriented:* setup required between client and server processes

UDP service:

- *unreliable data transfer* between sending and receiving process
- *does not provide:* reliability, flow control, congestion control, timing, throughput guarantee, security, orconnection setup,

Q: why bother? Why is there a UDP?

Internet apps: application, transport protocols

application	application layer protocol	underlying transport protocol
e-mail	SMTP [RFC 2821]	TCP
remote terminal access	Telnet [RFC 854]	TCP
Web	HTTP [RFC 2616]	TCP
file transfer	FTP [RFC 959]	TCP
streaming multimedia	HTTP (e.g., YouTube), RTP [RFC 1889]	TCP or UDP
Internet telephony	SIP, RTP, proprietary (e.g., Skype)	TCP or UDP

Chapter 2: outline

2.1 principles of network applications
- app architectures
- app requirements

2.2 Web and HTTP

2.4 electronic mail
- SMTP, POP3, IMAP

2.5 DNS

2.6 P2P applications

2.7 socket programming with UDP and TCP

Web and HTTP

First, a review…

- *web page* consists of *objects*
- object can be HTML file, JPEG image, Java applet, audio file,…
- web page consists of *base HTML-file* which includes *several referenced objects*
- each object is addressable by a *URL,* e.g.,

`www.someschool.edu/someDept/pic.gif`

host name | path name

HTTP overview

HTTP: hypertext transfer protocol

- Web's application layer protocol
- client/server model
  - *client:* browser that requests, receives, (using HTTP protocol) and "displays" Web objects
  - *server:* Web server sends (using HTTP protocol) objects in response to requests

HTTP overview (continued)

uses TCP:

- client initiates TCP connection (creates socket) to server, port 80
- server accepts TCP connection from client
- HTTP messages (application-layer protocol messages) exchanged between browser (HTTP client) and Web server (HTTP server)
- TCP connection closed

HTTP connections

non-persistent HTTP

- at most one object sent over TCP connection
  - connection then closed
- downloading multiple objects required multiple connections

persistent HTTP

- multiple objects can be sent over single TCP connection between client, server

Non-persistent HTTP

suppose user enters URL:
`www.someSchool.edu/someDepartment/home.index`

(contains text, references to 10 jpeg images)

1a. HTTP client initiates TCP connection to HTTP server (process) at www.someSchool.edu on port 80

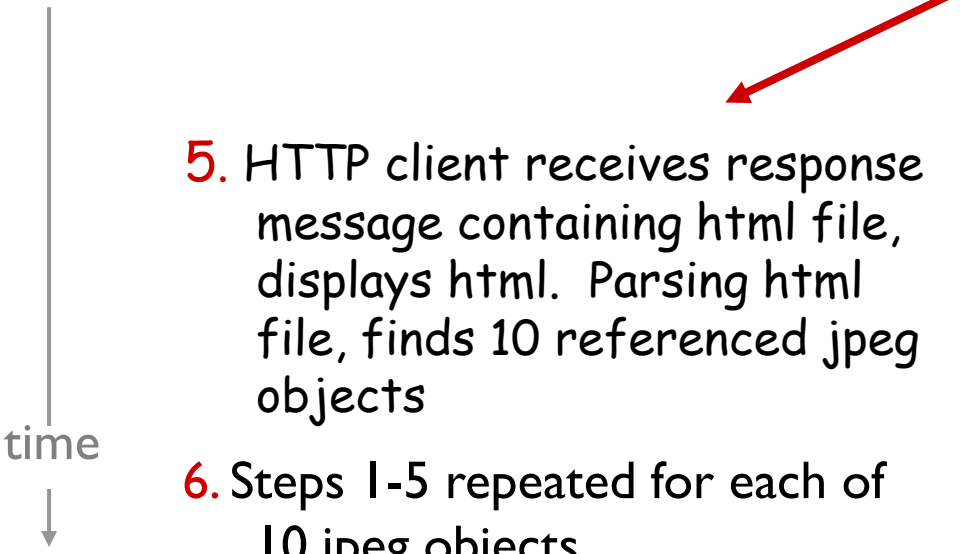
1b. HTTP server at host www.someSchool.edu waiting for TCP connection at port 80. "accepts" connection, notifying client

2. HTTP client sends HTTP *request message* (containing URL) into TCP connection socket. Message indicates that client wants object someDepartment/home.index

3. HTTP server receives request message, forms *response message* containing requested object, and sends message into its socket

time

Non-persistent HTTP (cont.)



4. HTTP server closes TCP connection.

5. HTTP client receives response message containing html file, displays html. Parsing html file, finds 10 referenced jpeg objects

6. Steps 1-5 repeated for each of 10 jpeg objects

Non-persistent HTTP: response time

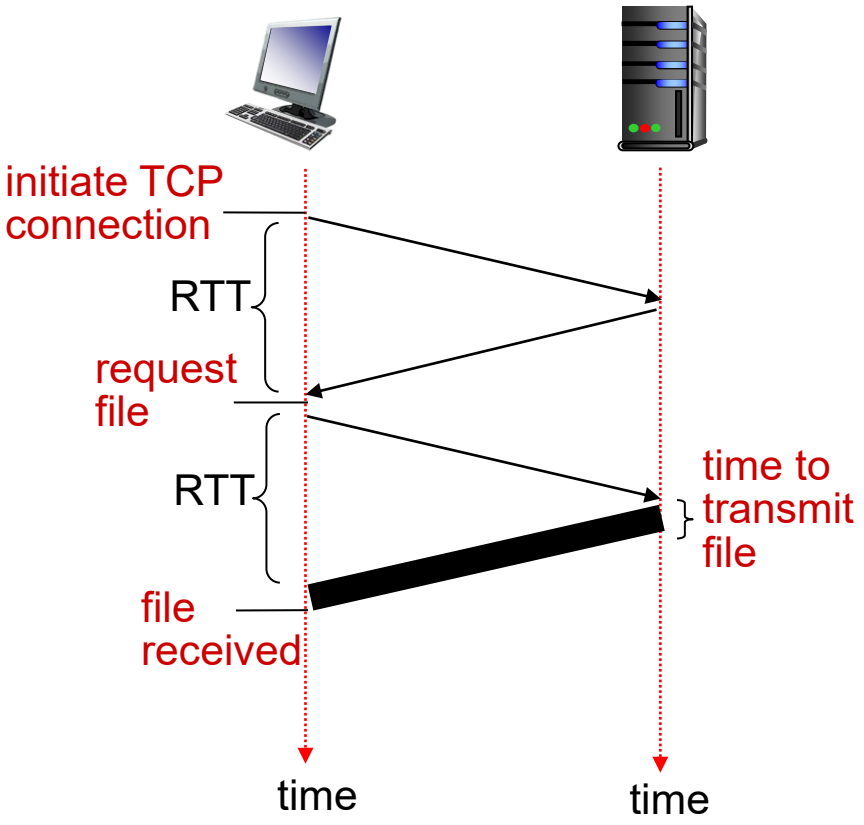
Round Trip Time (RTT)

definition: time for a small packet to travel from client to server and back

HTTP response time:

- one RTT to initiate TCP connection
- one RTT for HTTP request and first few bytes of HTTP response to return
- file transmission time
- non-persistent HTTP response time =

  2RTT+ file transmission time



Persistent HTTP

non-persistent HTTP issues:

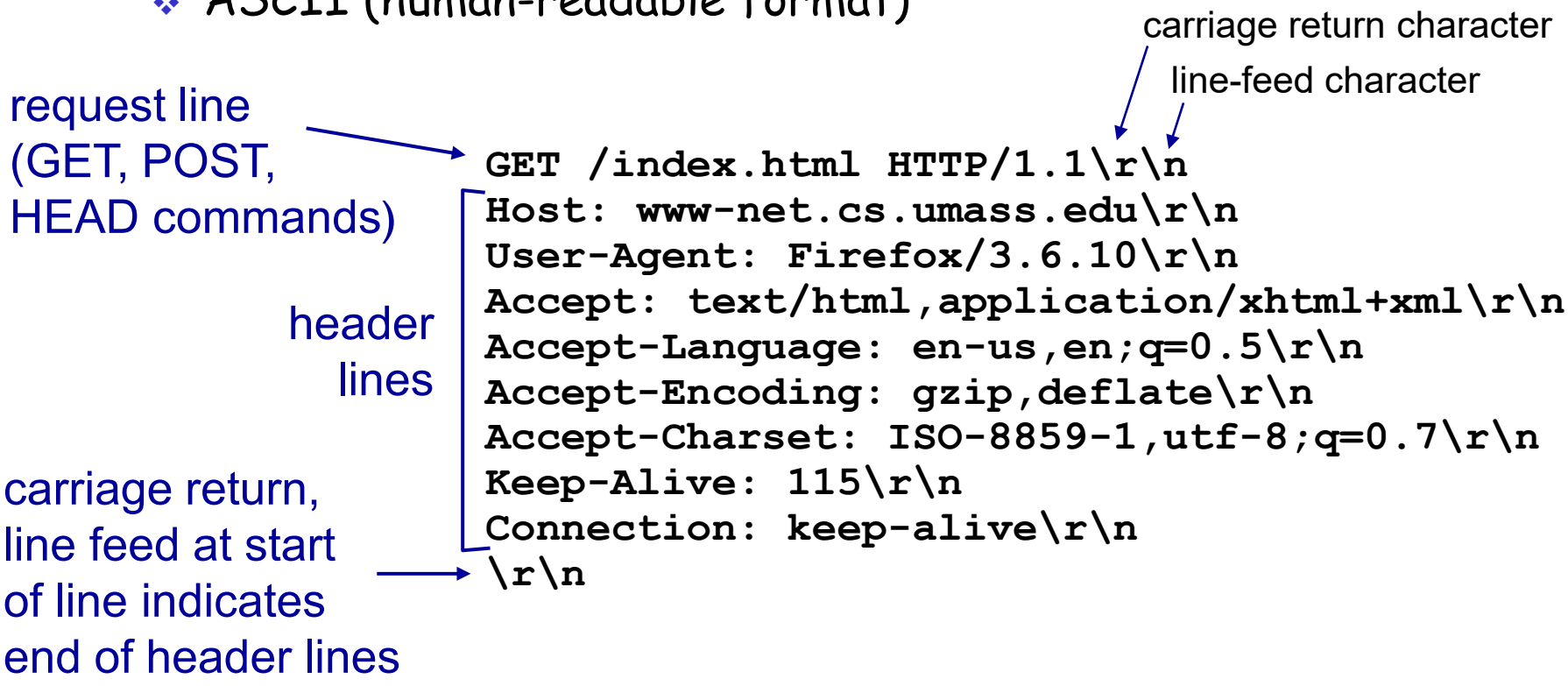
- requires 2 RTTs per object
- OS overhead for *each* TCP connection
- browsers often open parallel TCP connections to fetch referenced objects

persistent HTTP:

- server leaves connection open after sending response
- subsequent HTTP messages between same client/server sent over open connection
- client sends requests as soon as it encounters a referenced object
- as little as one RTT for all the referenced objects

HTTP request message

- two types of HTTP messages: *request, response*
- HTTP request message:
  - ASCII (human-readable format)



HTTP response message

status line (protocol status code status phrase)

header lines

data, e.g., requested HTML file

```
HTTP/1.1 200 OK\r\n
Date: Sun, 26 Sep 2010 20:09:20 GMT\r\n
Server: Apache/2.0.52 (CentOS)\r\n
Last-Modified: Tue, 30 Oct 2007 17:00:02
   GMT\r\n
ETag: "17dc6-a5c-bf716880"\r\n
Accept-Ranges: bytes\r\n
Content-Length: 2652\r\n
Keep-Alive: timeout=10, max=100\r\n
Connection: Keep-Alive\r\n
Content-Type: text/html; charset=ISO-8859-
   1\r\n
\r\n
data data data data data ...
```

HTTP response status codes

- status code appears in 1st line in server-to-client response message.
- some sample codes:

200 OK

- request succeeded, requested object later in this msg

301 Moved Permanently

- requested object moved, new location specified later in this msg (Location:)

400 Bad Request

- request msg not understood by server

404 Not Found

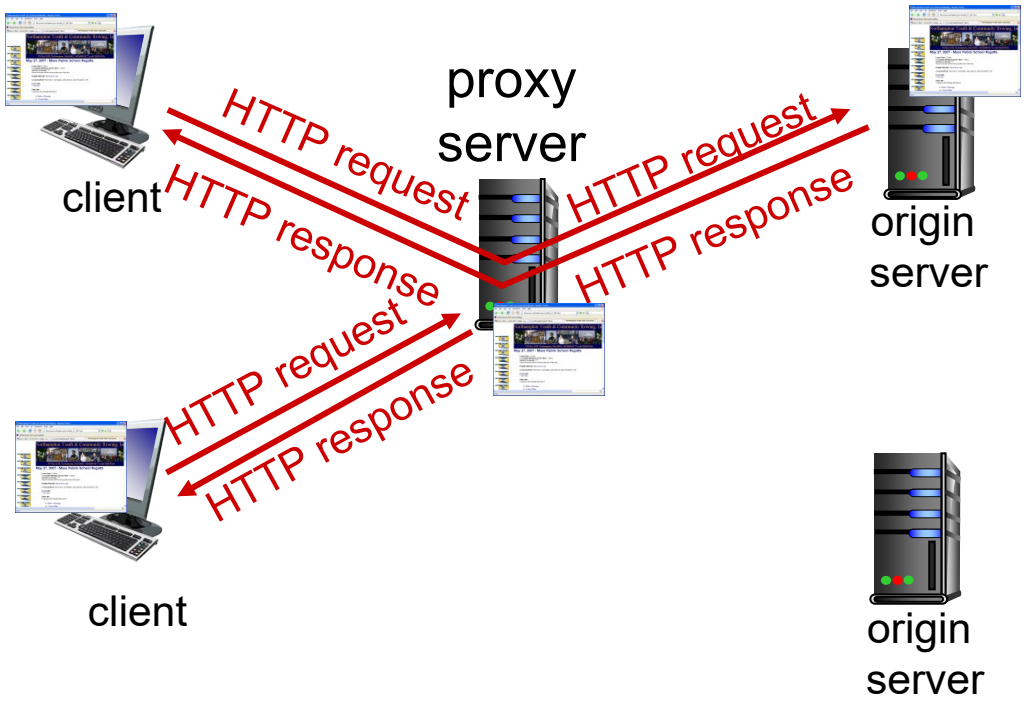
- requested document not found on this server

505 HTTP Version Not Supported

Web caches (proxy server)

goal: satisfy client request without involving origin server

- user sets browser: Web accesses via cache
- browser sends all HTTP requests to cache
  - object in cache: cache returns object
  - else cache requests object from origin server, then returns object to client



More about Web caching

- cache acts as both client and server
  - server for original requesting client
  - client to origin server
- typically cache is installed by ISP (university, company, residential ISP)

why Web caching?

- reduce response time for client request
- reduce traffic on an institution's access link

When is cache not good?

- Every client of the ISP requests different content.
  - Waste time on visiting cache server

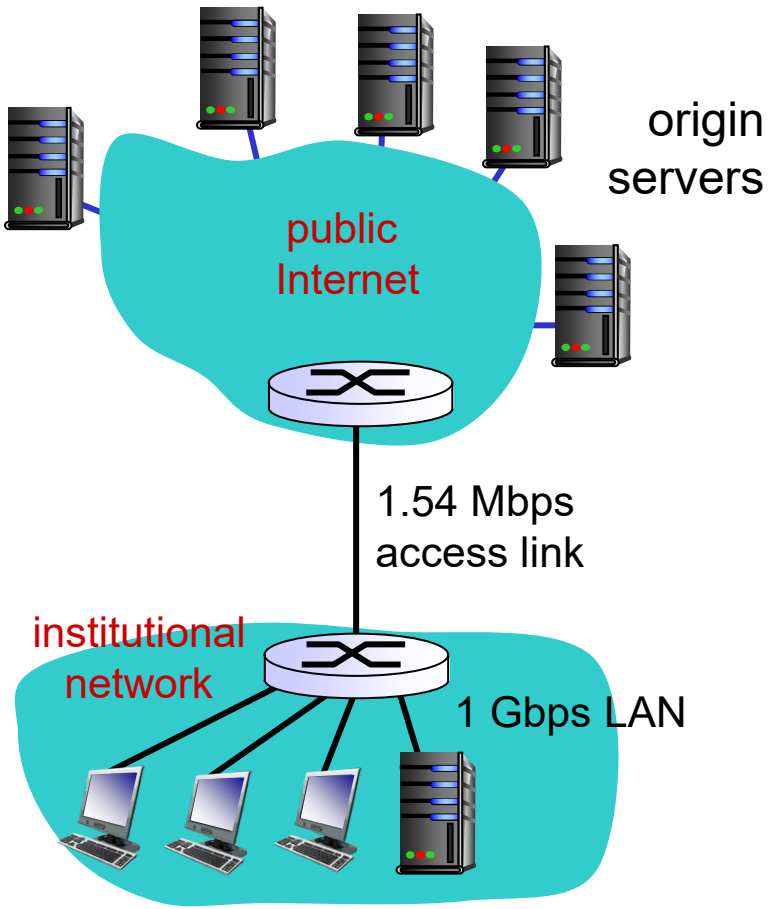
Caching example:

assumptions:

- avg object size: 100K bits
- avg request rate from browsers to origin servers:15/sec
- avg data rate to browsers: 1.50 Mbps
- RTT from institutional router to any origin server: 2 sec
- access link rate: 1.54 Mbps

consequences: *problem!*

- access link utilization = 99%
- total delay = Internet delay + access delay + LAN delay
  = 2 sec + minutes + usecs



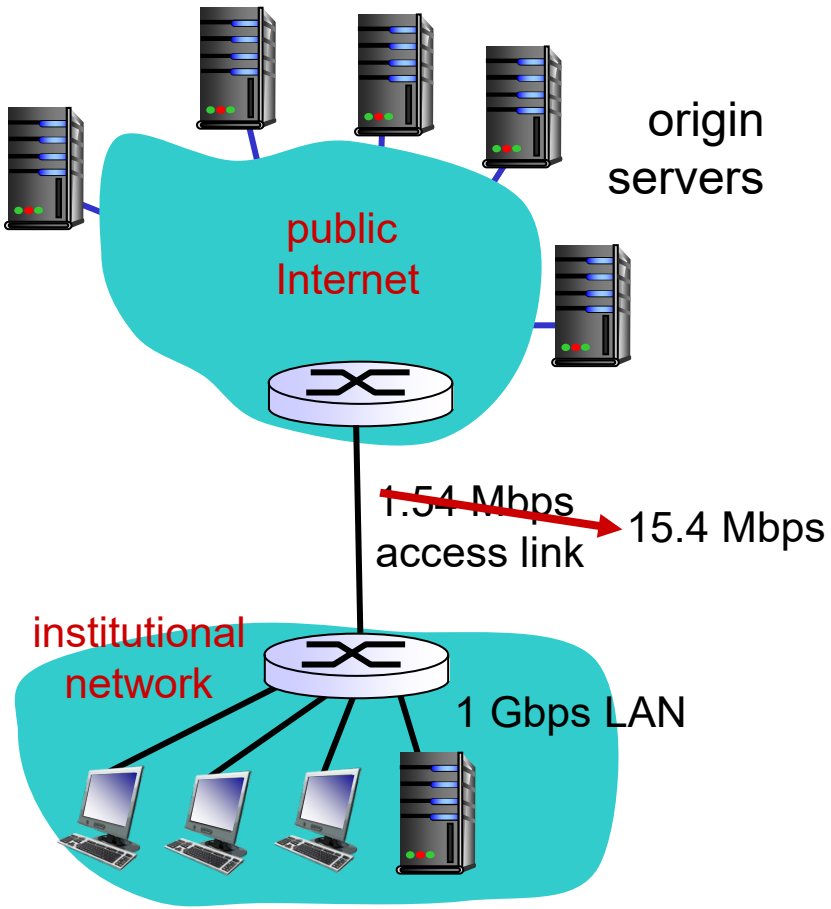
Caching example: fatter access link

assumptions:

- avg object size: 100K bits
- avg request rate from browsers to origin servers: 15/sec
- avg data rate to browsers: 1.50 Mbps
- RTT from institutional router to any origin server: 2 sec
- access link rate: ~~1.54 Mbps~~ 15.4 Mbps

consequences:

- access link utilization = ~~99%~~ 9.9%
- total delay = Internet delay + access delay + LAN delay
  = 2 sec + ~~minutes~~ + usecs
  msecs



Cost: increased access link speed (not cheap!)

Caching example: install local cache

assumptions:

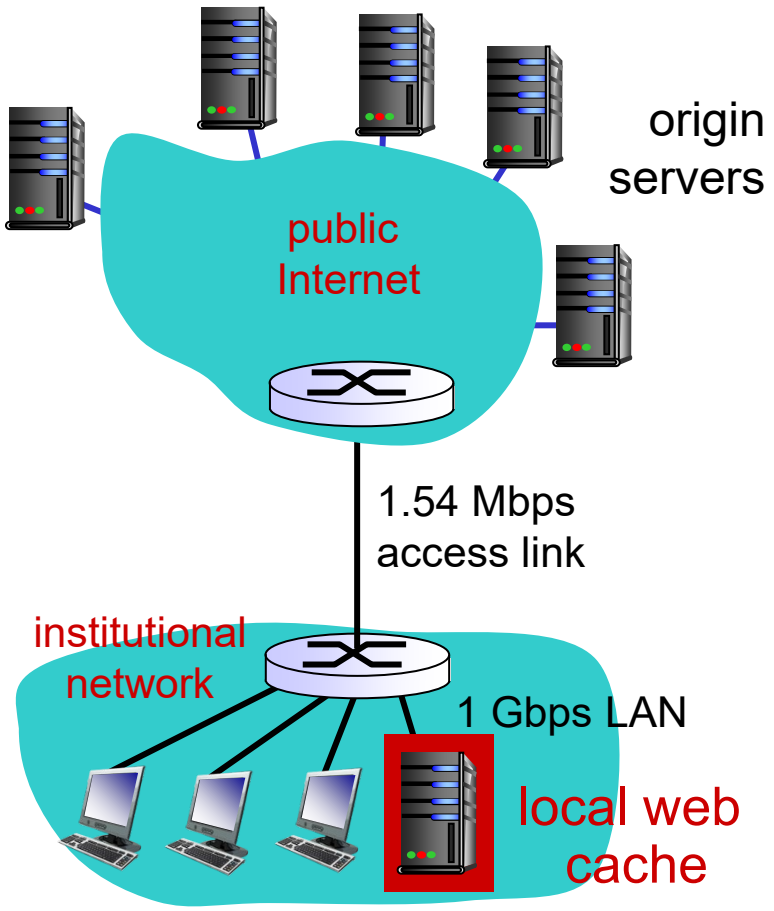
- avg object size: 100K bits
- avg request rate from browsers to origin servers:15/sec
- avg data rate to browsers: 1.50 Mbps
- RTT from institutional router to any origin server: 2 sec
- access link rate: 1.54 Mbps

consequences:

- access link utilization = ?
- total delay = ?

How to compute link utilization, delay?

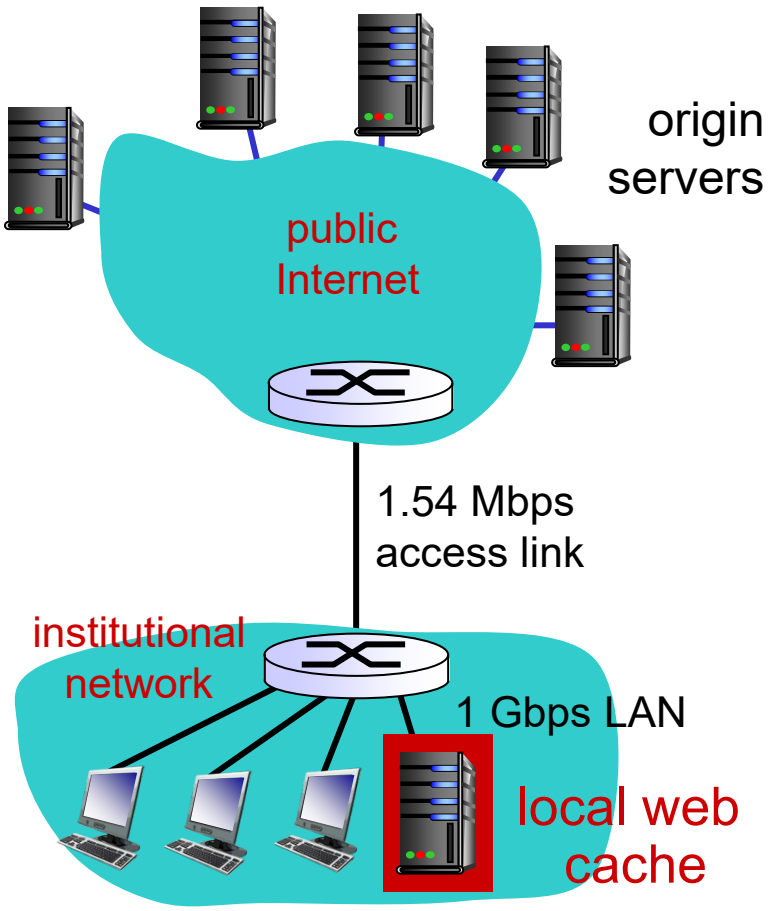
Cost: web cache (cheap!)



Caching example: install local cache

Calculating access link utilization, delay with cache:

- suppose cache hit rate is 0.4
  - 40% requests satisfied at cache, 60% requests satisfied at origin
- access link utilization:
  - 60% of requests use access link
- data rate to browsers over access link = 0.6*1.50 Mbps = .9 Mbps
  - utilization = 0.9/1.54 = .58
- total delay
  - = 0.6 * (delay from origin servers) +0.4 * (delay when satisfied at cache)
  - = 0.6 (2.01) + 0.4 (~msecs)
  - = ~ 1.2 secs
  - less than with 154 Mbps link (and cheaper too!)



Chapter 2: outline

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- app architectures
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2.2 Web and HTTP

2.4 electronic mail
- SMTP, POP3, IMAP

2.5 DNS

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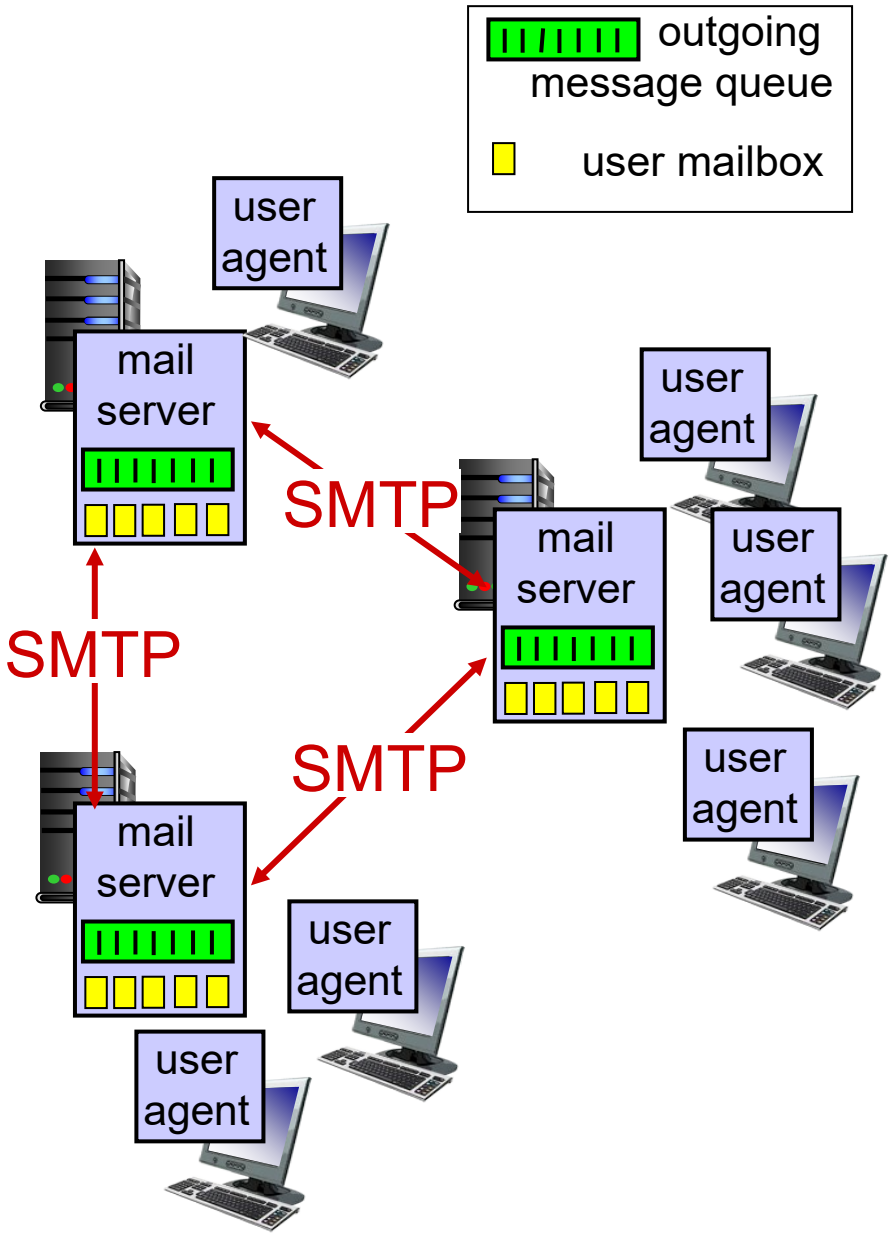
Electronic mail

Three major components:

- user agents
- mail servers
- simple mail transfer protocol: SMTP

User Agent

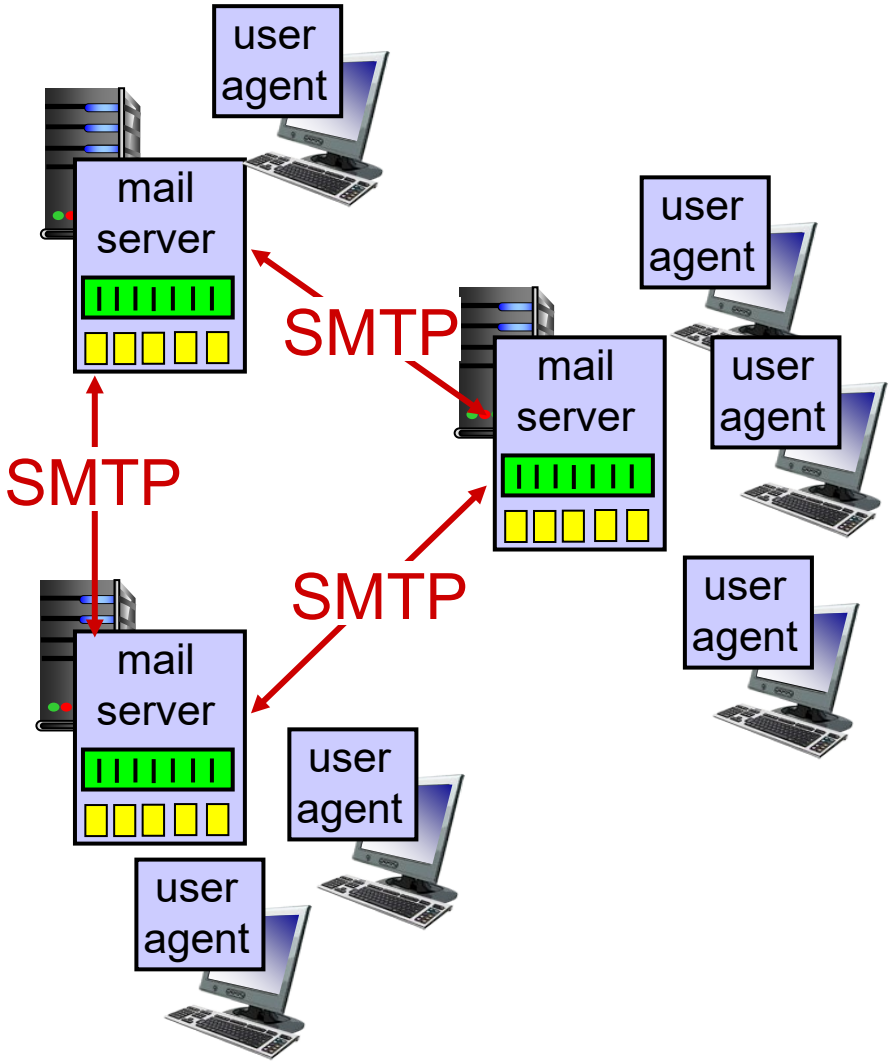
- a.k.a. "mail reader"
- composing, editing, reading mail messages
- e.g., Outlook, iPhone mail client
- outgoing, incoming messages stored on server



Electronic mail: mail servers

mail servers:

- *mailbox* contains incoming messages for user
- *message queue* of outgoing (to be sent) mail messages
- *SMTP protocol* between mail servers to send email messages
  - client: sending mail server
  - "server": receiving mail server

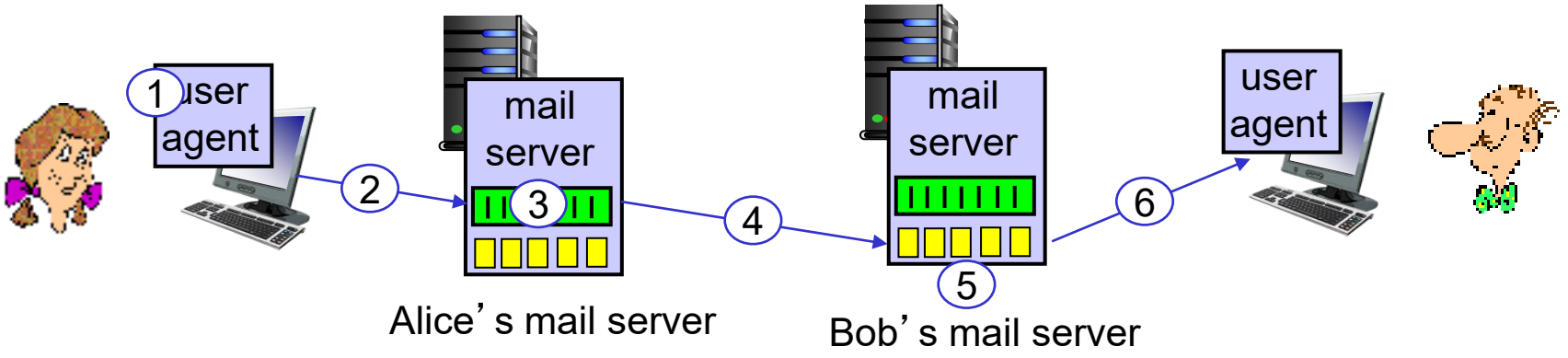


Electronic Mail: SMTP [RFC 2821]

- uses TCP to reliably transfer email message from client to server, port 25
- direct transfer: sending server to receiving server
- three phases of transfer
  - handshaking (greeting)
  - transfer of messages
  - closure

Scenario: Alice sends message to Bob

1) Alice uses UA to compose message "to" `bob@someschool.edu`
2) Alice's UA sends message to her mail server; message placed in message queue
3) client side of SMTP opens TCP connection with Bob's mail server
4) SMTP client sends Alice's message over the TCP connection
5) Bob's mail server places the message in Bob's mailbox
6) Bob invokes his user agent to read message



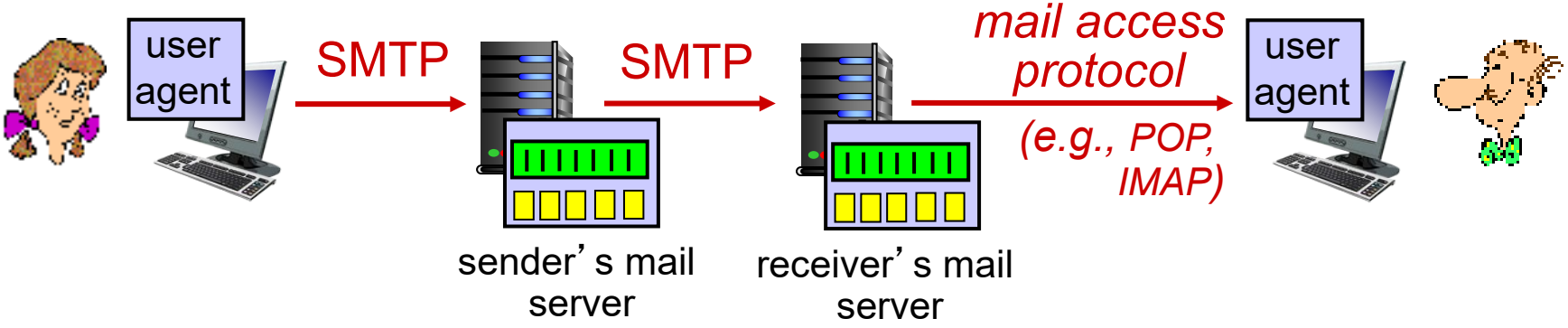
SMTP: final words

- SMTP uses persistent connections
- SMTP requires message (header & body) to be in 7-bit ASCII

comparison with HTTP:

- HTTP: pull
- SMTP: push
- both have ASCII command/response interaction, status codes

Mail access protocols



- SMTP: delivery/storage to receiver's server
- mail access protocol: retrieval from server
  - POP: Post Office Protocol [RFC 1939]: authorization, download
  - IMAP: Internet Mail Access Protocol [RFC 1730]: more features, including manipulation of stored msgs on server
  - HTTP: gmail, Hotmail, Yahoo! Mail, etc.

POP3 and IMAP

POP3

- POP3 "download and delete" mode
  - Bob cannot re-read e-mail if he changes client
- POP3 "download-and-keep": copies of messages on different clients
- POP3 is stateless across sessions

IMAP

- keeps all messages in one place: at server
- allows user to organize messages in folders
- keeps user state across sessions:
  - names of folders and mappings between message IDs and folder name

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DNS: domain name system

Internet hosts, routers:

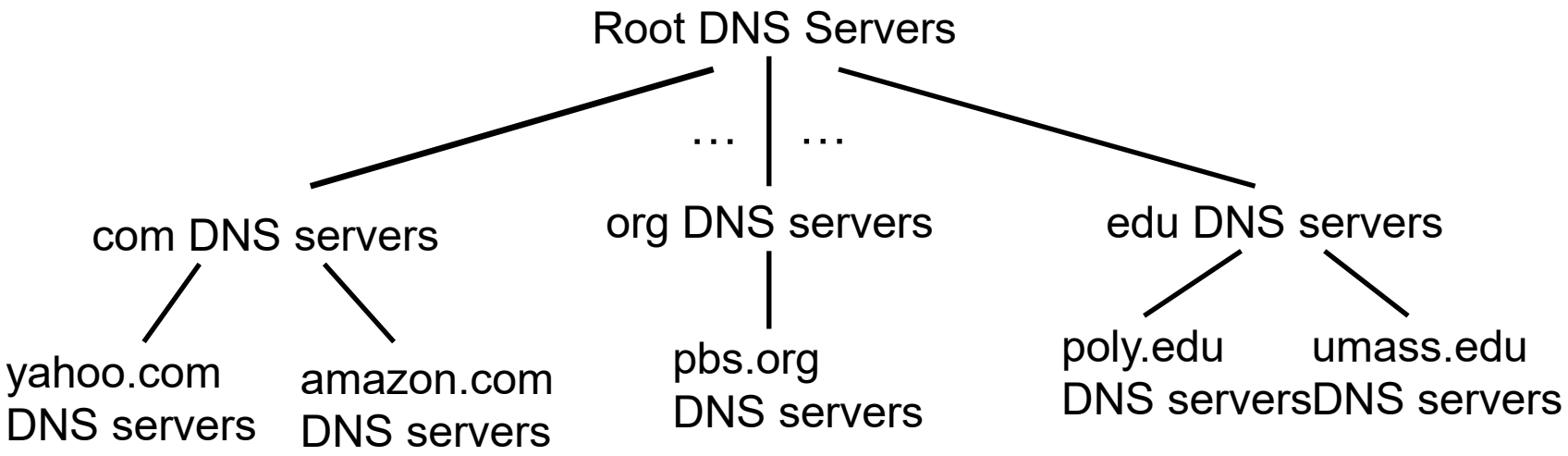
- IP address (32 bit) - used for addressing datagrams
- "name", e.g., www.yahoo.com - used by humans

Q: how to map between IP address and name, and vice versa ?

Domain Name System:

- *distributed database* implemented in hierarchy of many *name servers*
- *application-layer protocol:* hosts, name servers communicate to *resolve* names (address/name translation)

DNS: a distributed, hierarchical database



client wants IP for www.amazon.com; 1st approx:

- client queries root server to find com DNS server
- client queries .com DNS server to get amazon.com DNS server
- client queries amazon.com DNS server to get IP address for www.amazon.com

DNS: services, structure

DNS services

- hostname to IP address translation
- load distribution
  - replicated Web servers: many IP addresses correspond to one name

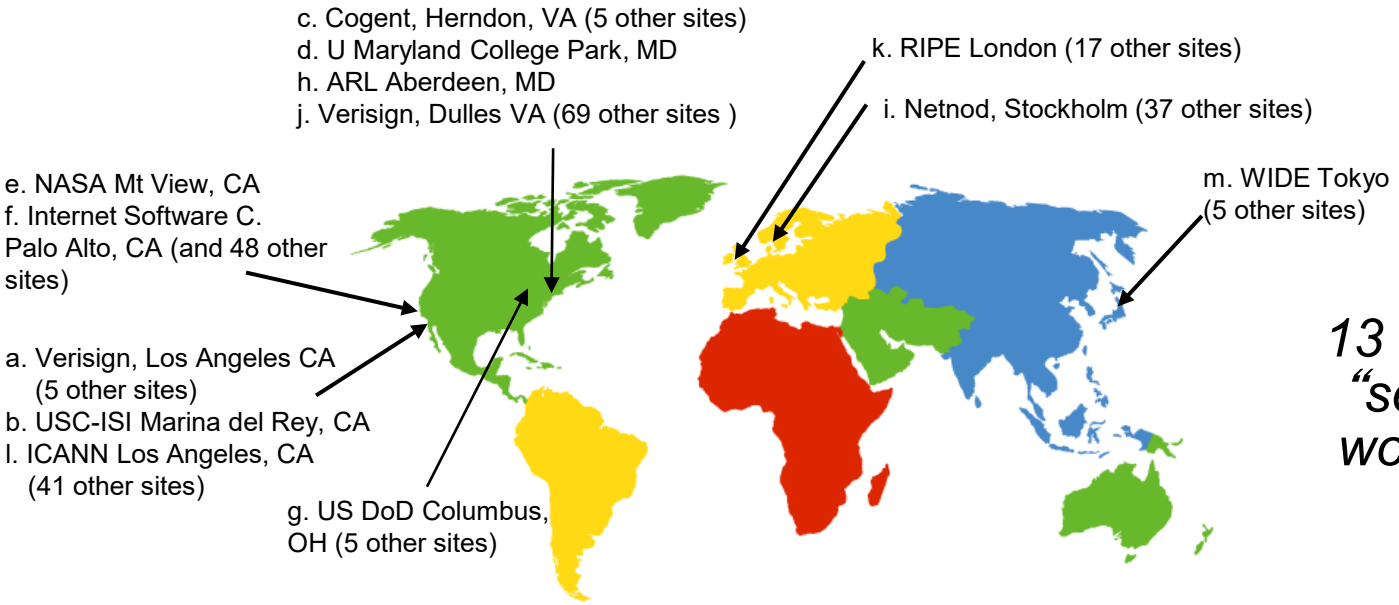
why not centralize DNS?

- single point of failure
- traffic volume
- distant centralized database
- maintenance

A: doesn't scale!

DNS: root name servers

- contacted by local name server that can not resolve name
- root name server:
  - contacts authoritative name server if name mapping not known
  - gets mapping
  - returns mapping to local name server



13 root name "servers" worldwide

TLD, authoritative servers

top-level domain (TLD) servers:

- responsible for com, org, net, edu, aero, jobs, museums, and all top-level country domains, e.g.: uk, fr, ca, jp
- Network Solutions maintains servers for .com TLD
- Educause for .edu TLD

authoritative DNS servers:

- organization's own DNS server(s), providing authoritative hostname to IP mappings for organization's named hosts
- can be maintained by organization or service provider

Local DNS name server

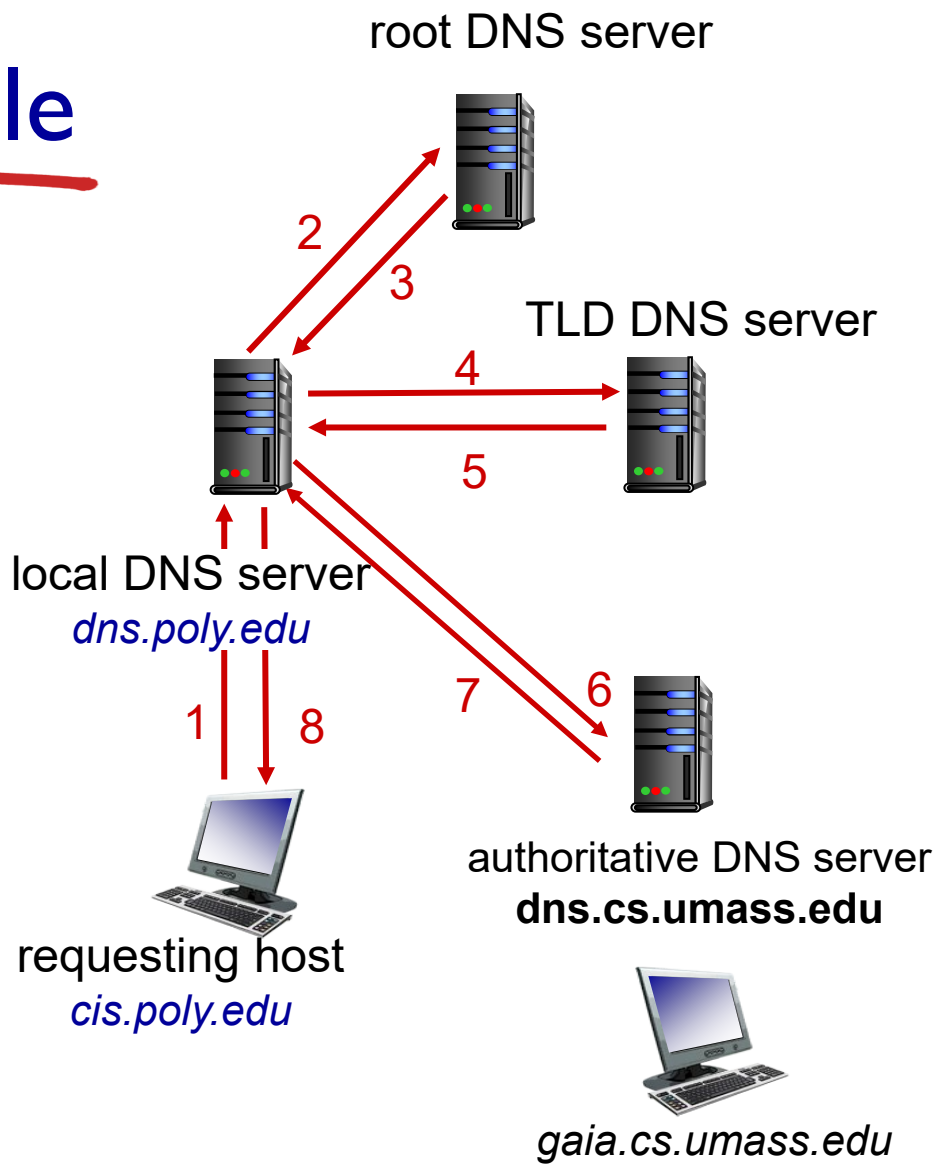
- each ISP (residential ISP, company, university) has one
  - also called "default name server"
- when host makes DNS query, query is sent to its local DNS server
  - has local cache of recent name-to-address translation pairs (but may be out of date!)
  - acts as proxy, forwards query into hierarchy

DNS name resolution example

- host at cis.poly.edu wants IP address for gaia.cs.umass.edu

iterated query:

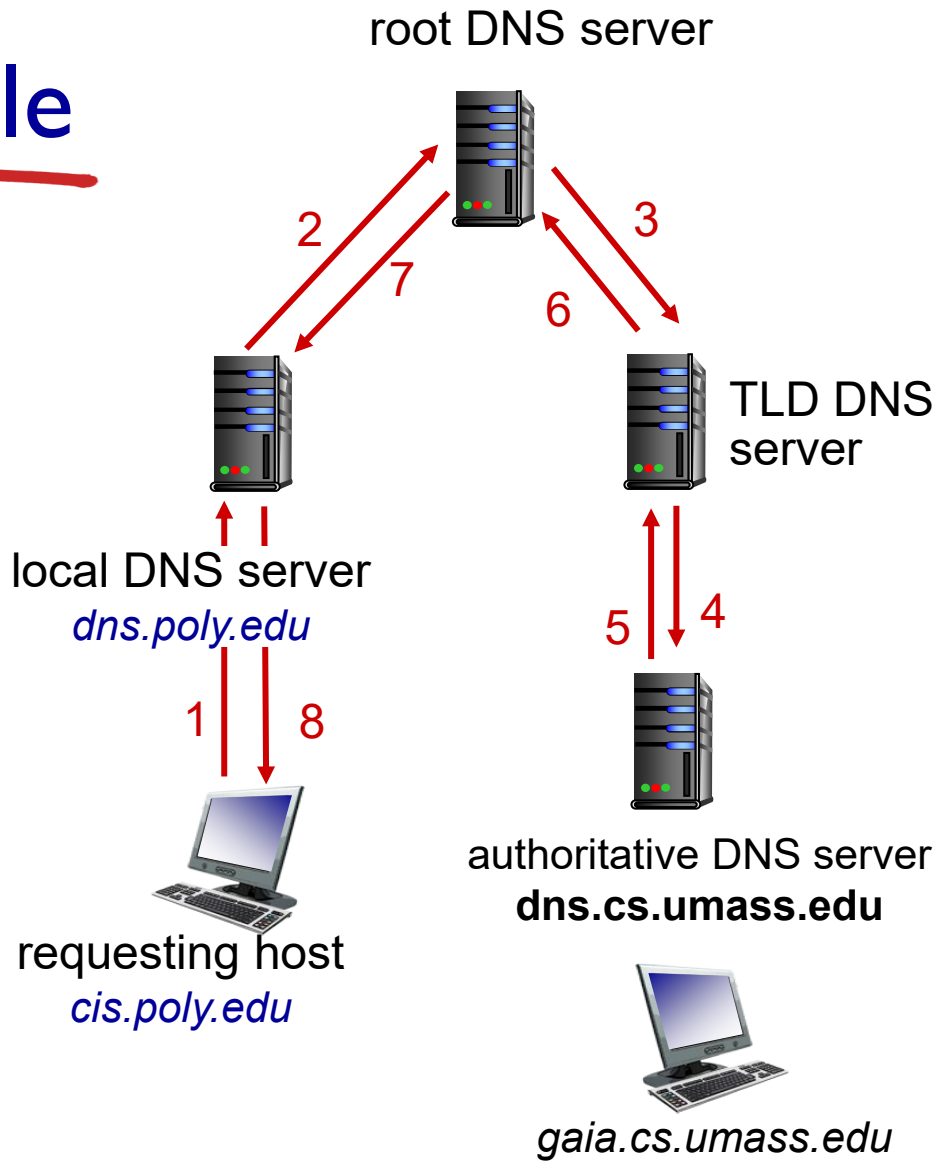
- contacted server replies with name of server to contact
- "I don't know this name, but ask this server"



DNS name resolution example

recursive query:

- puts burden of name resolution on contacted name server
- heavy load at upper levels of hierarchy?



DNS: caching, updating records

- once (any) name server learns mapping, it *caches* mapping
  - cache entries timeout (disappear) after some time (TTL)
  - TLD servers typically cached in local name servers
    - thus root name servers not often visited

- cached entries may be *out-of-date* (best effort name-to-address translation!)
  - if name host changes IP address, may not be known Internet-wide until all TTLs expire

Change of office hours

- From Wednesday after class to Monday after class.

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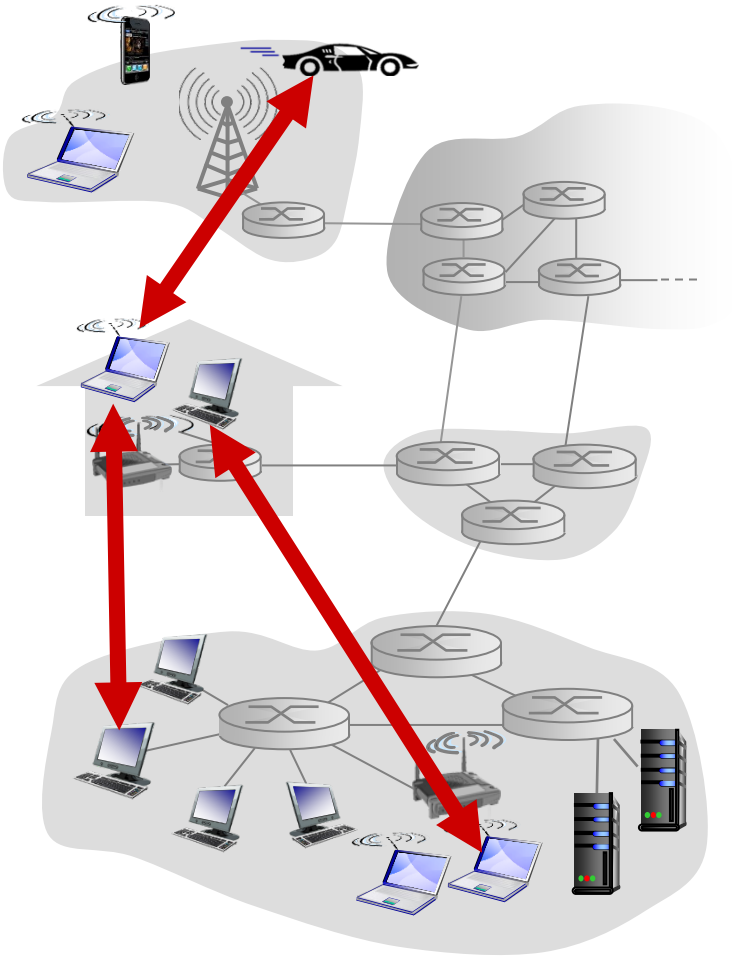
P2P architecture

- *no* always-on server
- arbitrary end systems directly communicate
- peers are intermittently connected and change IP addresses

examples:

- file distribution (BitTorrent)
- Streaming (KanKan)
- VoIP (Skype)

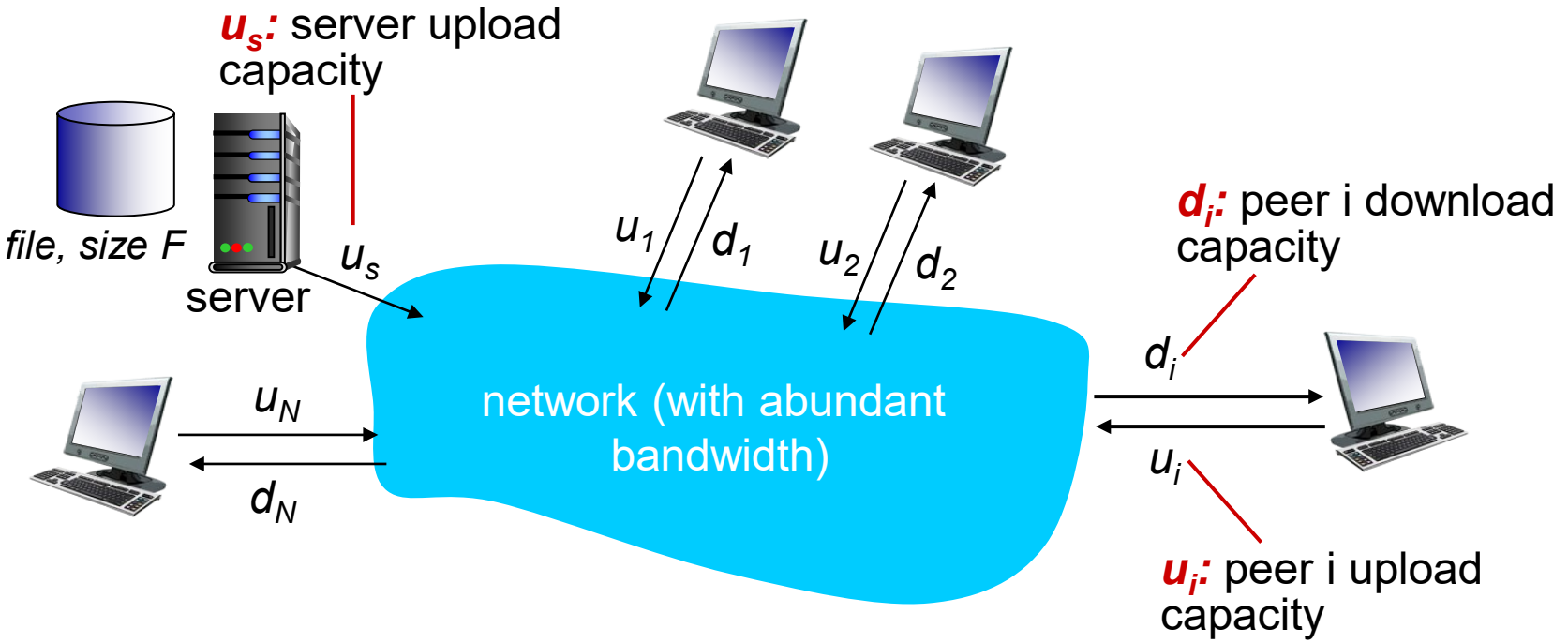


- However, most of them requires a central server to manage the peers

File distribution: client-server vs P2P

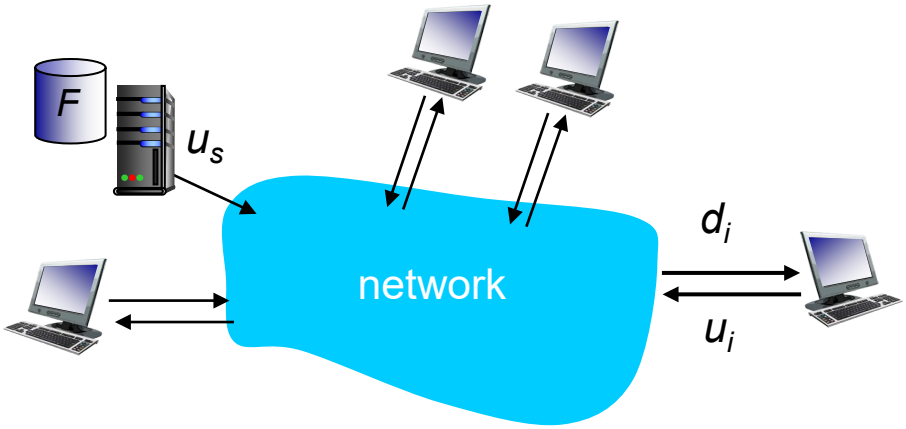
Question: how much time to distribute file (size *F*) from one server to *N* peers?

- peer upload/download capacity is limited resource



File distribution time: client-server

- *server transmission:* must sequentially send (upload) *N* file copies:
  - time to send one copy: F/u_s
  - time to send N copies: NF/u_s
- *client:* each client must download file copy
  - d_{min} = min client download rate
  - min client download time: F/d_{min}



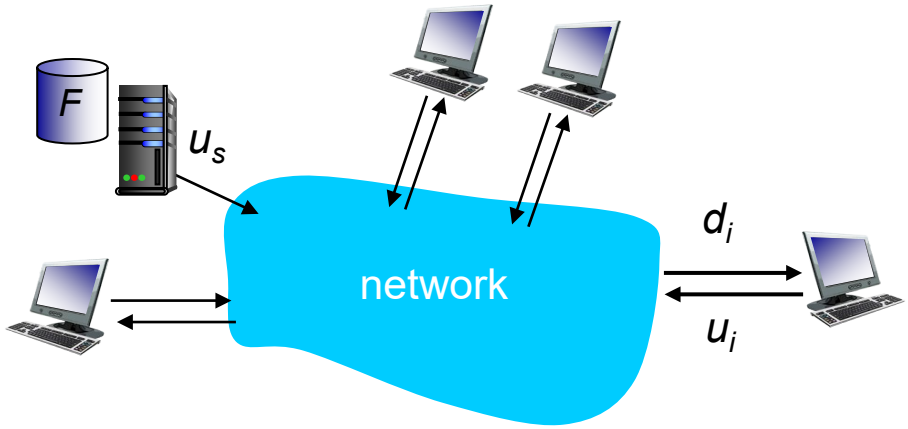
time to distribute F to N clients using client-server approach

$$D_{c\text{-}s} \geq max\{NF/u_s, F/d_{min}\}$$

increases linearly in N

File distribution time: P2P

- *server transmission:* must upload at least one copy
  - time to send one copy: F/u_s
- *client:* each client must download file copy
  - min client download time: F/d_{min}



- *clients:* as aggregate must download *NF* bits
  - max upload rate (limting max download rate) is $u_s + \Sigma u_i$

time to distribute F to N clients using P2P approach

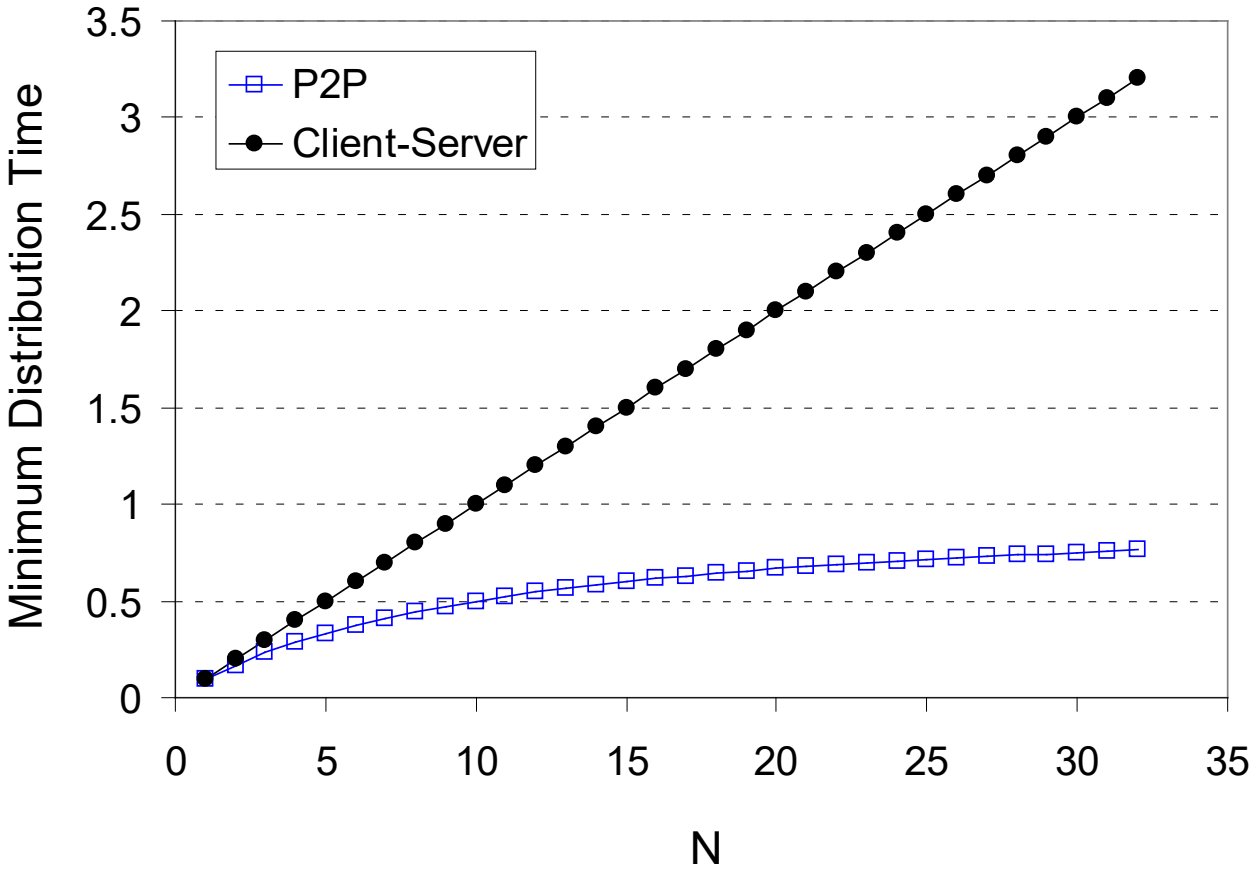
$$D_{P2P} \geq max\{F/u_s, F/d_{min}, NF/(u_s + \Sigma u_i)\}$$

increases linearly in *N* …

… but so does this, as each peer brings service capacity

Client-server vs. P2P: example

client upload rate = u, F/u = 1 hour, $u_s = 10u$, $d_{min} \geq u_s$

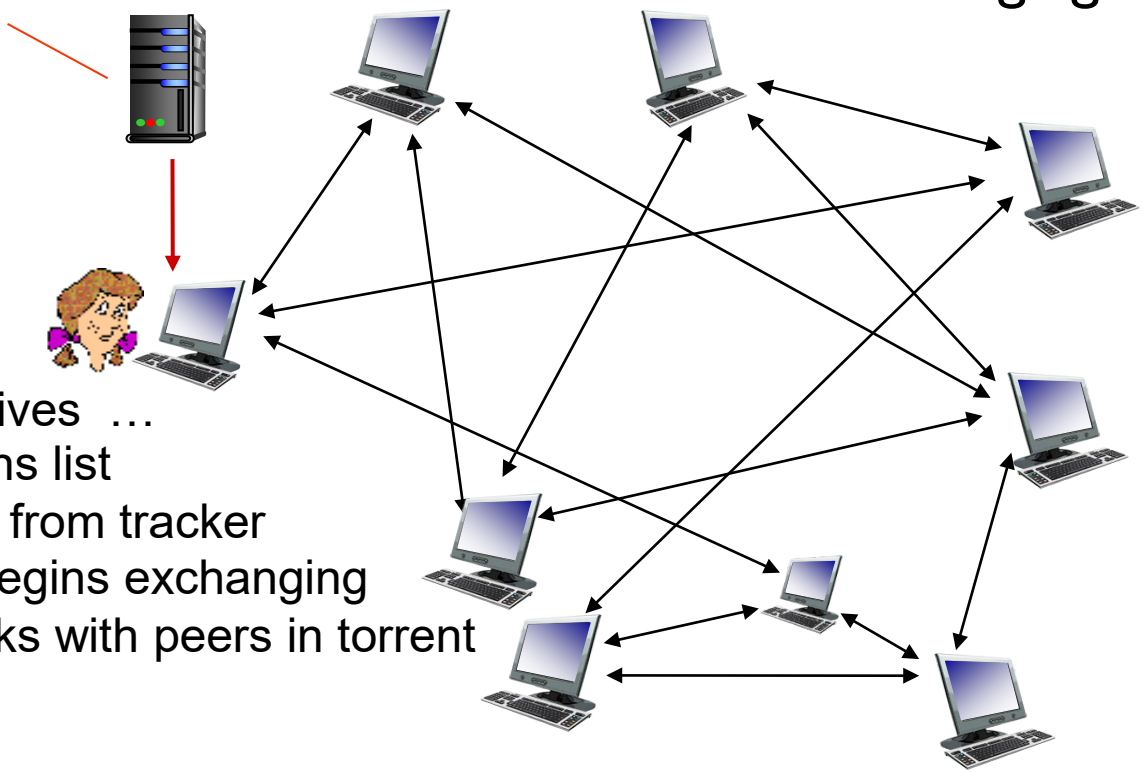


P2P file distribution: BitTorrent

- file divided into 256Kb chunks
- peers in torrent send/receive file chunks

tracker: tracks peers participating in torrent

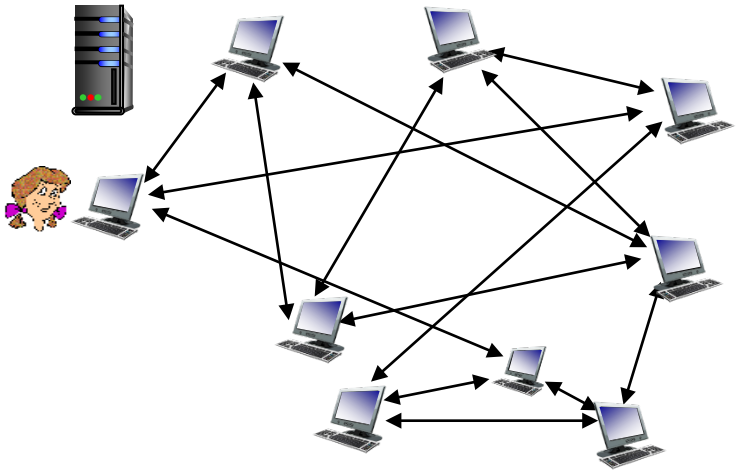
torrent: group of peers exchanging chunks of a file



Alice arrives …
… obtains list
of peers from tracker
… and begins exchanging
file chunks with peers in torrent

P2P file distribution: BitTorrent

- peer joining torrent:
  - has no chunks, but will accumulate them over time from other peers
  - registers with tracker to get list of peers, connects to subset of peers ("neighbors")



- while downloading, peer uploads chunks to other peers
- peer may change peers with whom it exchanges chunks
- *churn:* peers may come and go
- once peer has entire file, it may (selfishly) leave or (altruistically) remain in torrent

BitTorrent: requesting, sending file chunks

requesting chunks:

- at any given time, different peers have different subsets of file chunks
- periodically, Alice asks each peer for list of chunks that they have
- Alice requests missing chunks from peers, rarest first

sending chunks: tit-for-tat

- Alice sends chunks to those four peers currently sending her chunks *at highest rate*
  - other peers are choked by Alice (do not receive chunks from her)
  - re-evaluate top 4 every10 secs
- every 30 secs: randomly select another peer, starts sending chunks
  - "optimistically unchoke" this peer
  - newly chosen peer may join top 4

- Interview with Bram Cohen, inventor of BitTorrent
- https://www.youtube.com/watch?v=u0xngxfbKAE
- 2:25 – 6:25

Chapter 2: summary

our study of network apps now complete!

- application architectures
  - client-server
  - P2P
- application service requirements:
  - reliability, bandwidth, delay
- Internet transport service model
  - connection-oriented, reliable: TCP
  - unreliable, datagrams: UDP

- specific protocols:
  - HTTP
  - SMTP, POP, IMAP
  - DNS
  - P2P: BitTorrent, DHT

CSE 150 : Introduction to Computer Networks

Chen Qian
Computer Science and Engineering
UCSC Baskin Engineering

Chapter 3

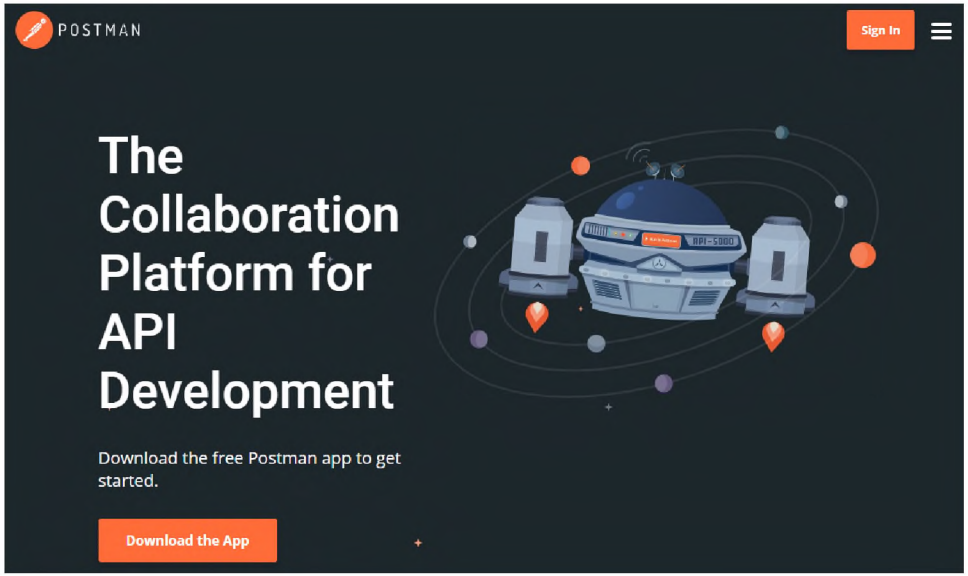
Lab 2 : HTTP,DNS, and TCP

HTTP

- Tips:
  - Status Code
  - Navigate to http://www.example.com (not https!)
  - Clear cache if not work

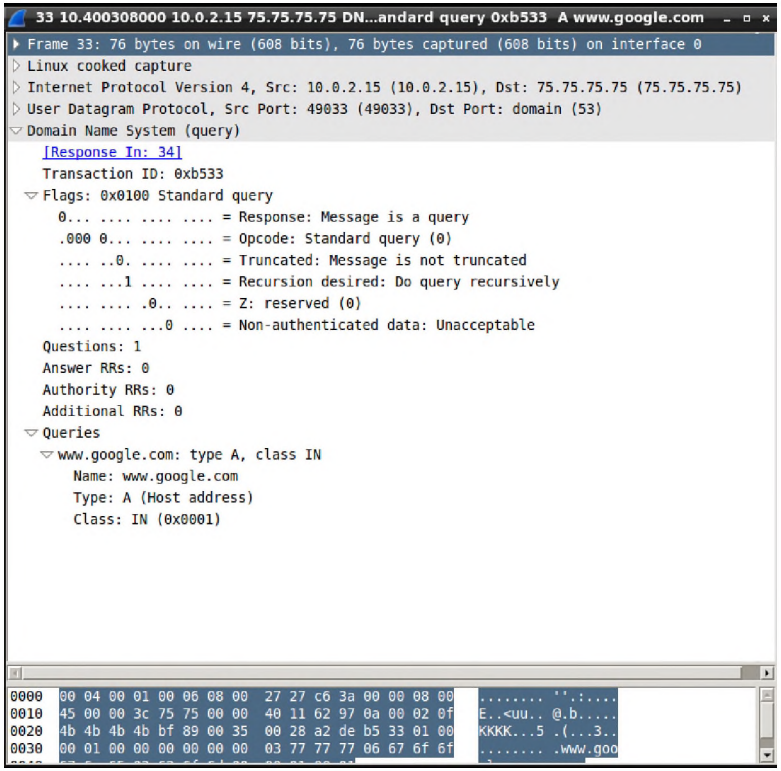
HTTP

- Tips:
  - Q4: Using Chromium (or any other Linux utility you are comfortable with), find a way to create an HTTP message using a method other than GET.
  - Download Postman app



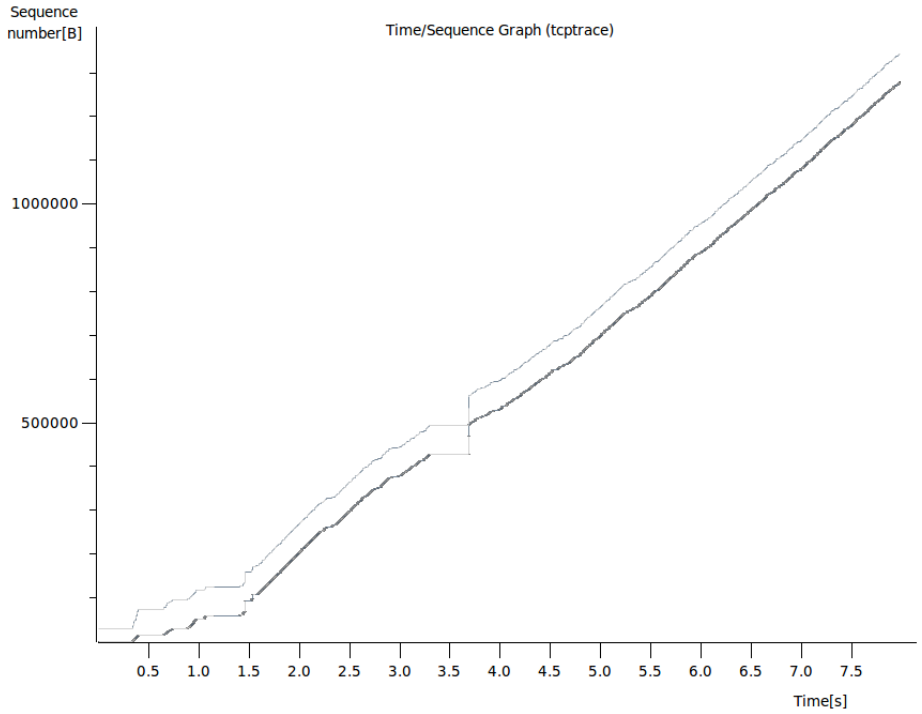
DNS

- Tips:
  - Q8: Did your computer want to complete the request recursively?
    - Look through the Flags section of the DNS request, and show "recursion desired".



TCP

- Tips:
  - Q12 : Create a tcptrace graph with TCP packet selected
    - Select the entry -> Statistics ->TCP StreamGraph ->Time Sequence Graph(tcptrace)



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

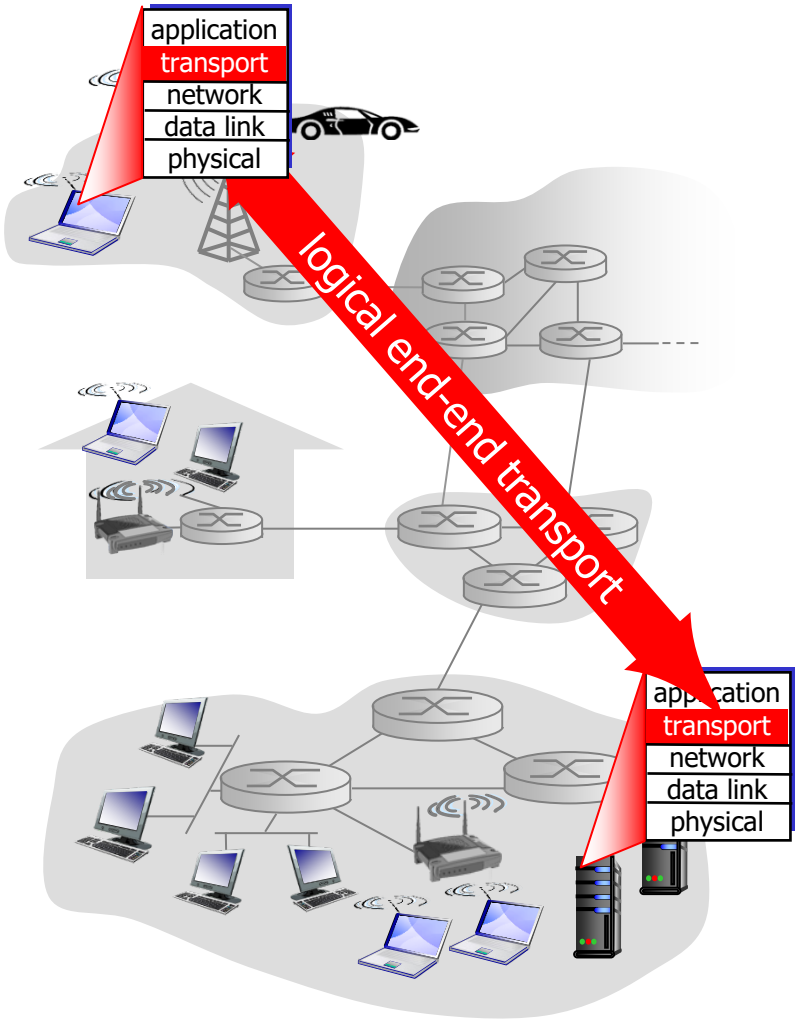
- segment structure
- reliable data transfer
- flow control
- connection management

3.6 principles of congestion control

3.7 TCP congestion control

Transport services and protocols

- provide *logical communication* between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into *segments*, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP



Transport vs. network layer

- *network layer:* logical communication between hosts
- *transport layer:* logical communication between processes
  - relies on, enhances, network layer services

household analogy:

3 kids in Ann's house sending letters to 3 kids in Bill's house:

- hosts = houses
- processes = kids
- app messages = letters in envelopes
- transport protocol = Ann and Bill who demux to in-house siblings
- network-layer protocol = postal service

Remember this

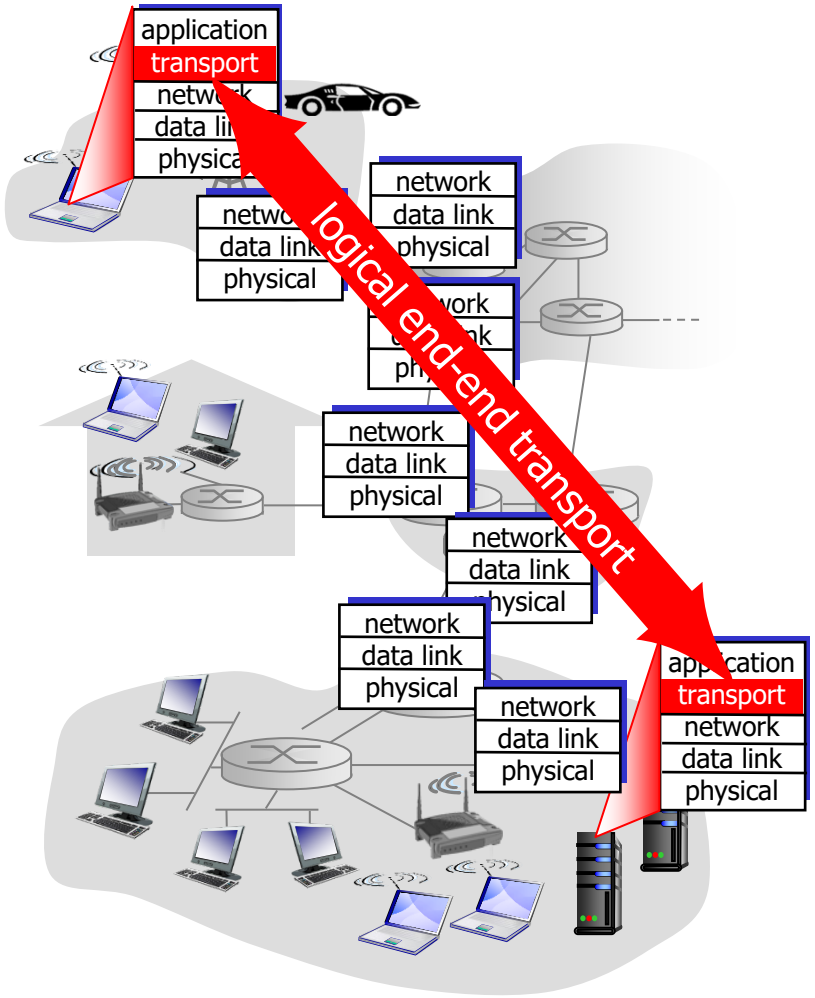
- Ann's home
  - Ann
  - Alice
  - Amy
- Home address:
  - 123, Santa Cruz

- Bill's home
  - Bill
  - Bob
  - Brent
- Home address:
  - 456, Scotts Valley

- *Application layer:* Bob wants to thank his friends Alice.
- *transport layer:* Bob mails a thank letter to Alice
- *network layer:* USPS delivers Bob's letter to Alice's main box

Internet transport-layer protocols

- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



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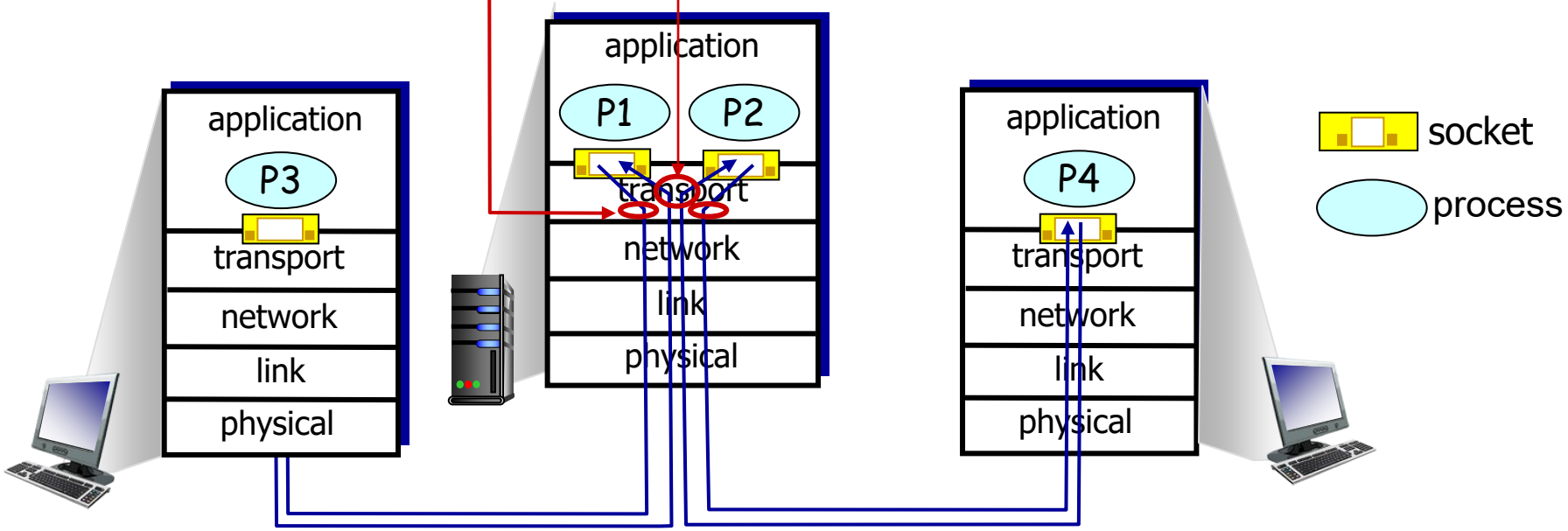
Multiplexing/demultiplexing

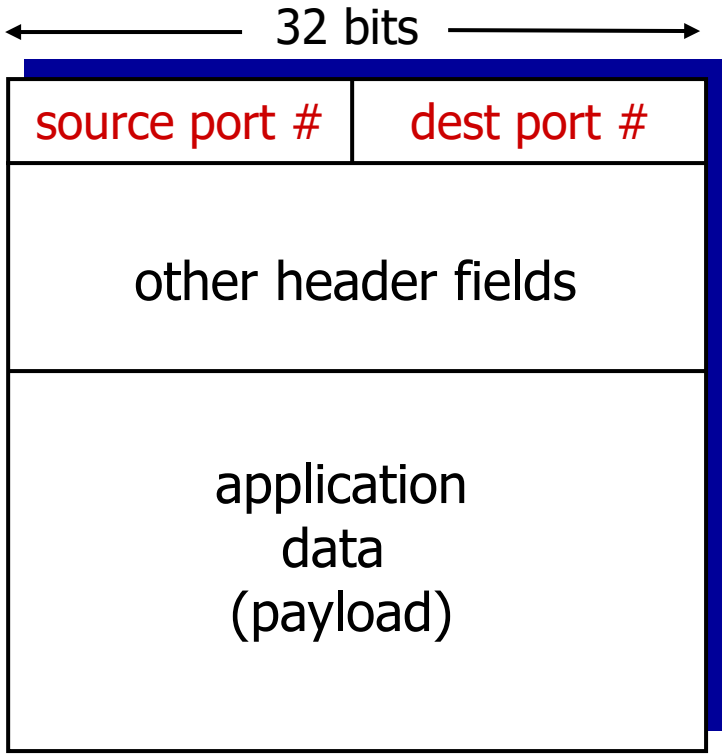
multiplexing at sender:
handle data from multiple sockets, add transport header (later used for demultiplexing)

demultiplexing at receiver:
use header info to deliver received segments to correct socket



How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries one transport-layer segment
  - each segment has source, destination port number
- host uses *IP addresses & port numbers* to direct segment to appropriate socket



TCP/UDP segment format

Connectionless demultiplexing

- *recall:* created socket has host-local port #:

```
DatagramSocket mySocket1
= new DatagramSocket(12534);
```

- *recall:* when creating datagram to send into UDP socket, must specify
  - destination IP address
  - destination port #

- when host receives UDP segment:
  - checks destination port # in segment
  - directs UDP segment to socket with that port #

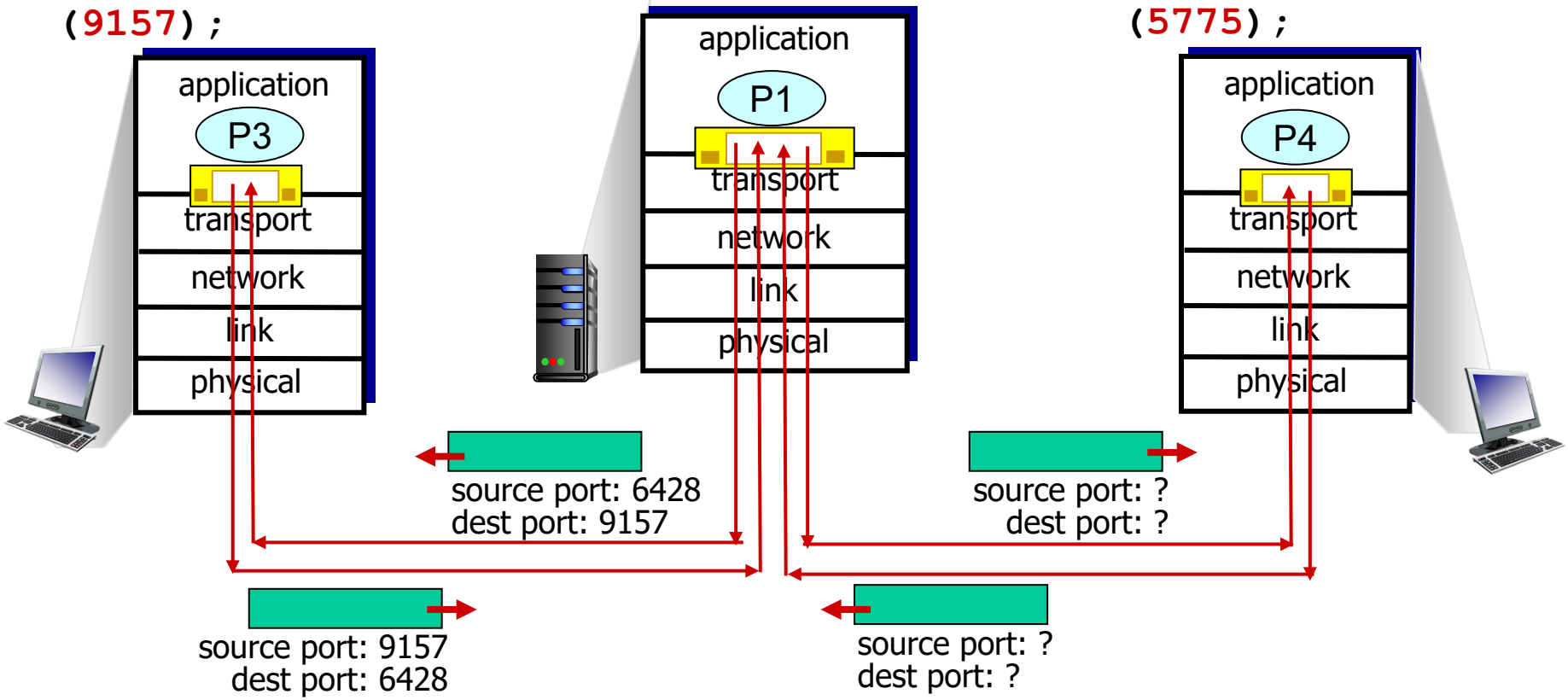
IP datagrams with *same dest. port #,* but different source IP addresses and/or source port numbers will be directed to *same socket* at dest

Connectionless demux: example

```
DatagramSocket
 mySocket2 = new
 DatagramSocket
 (9157);
```

```
DatagramSocket
 serverSocket = new
 DatagramSocket
 (6428);
```

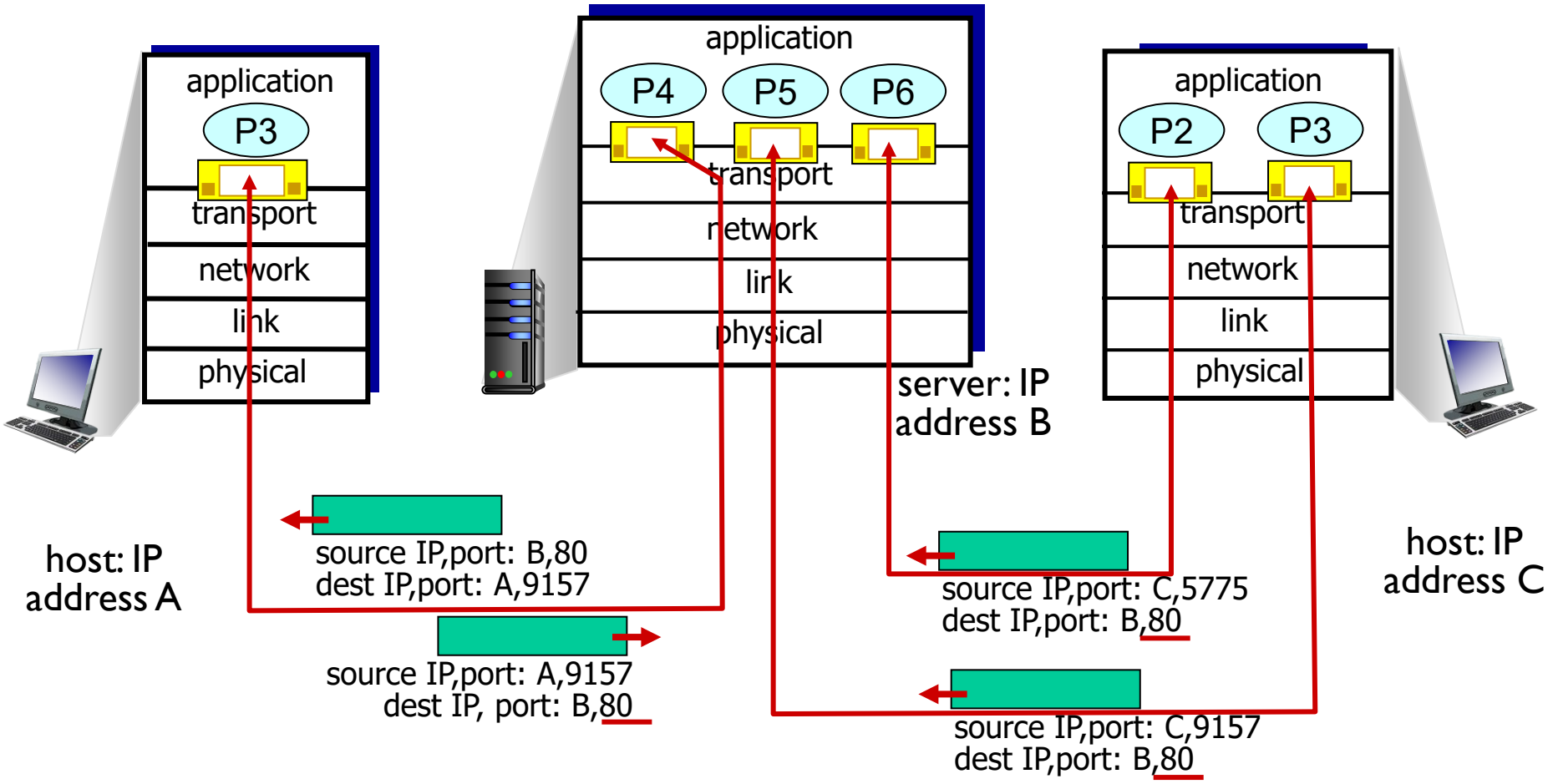
```
DatagramSocket
 mySocket1 = new
 DatagramSocket
 (5775);
```



Connection-oriented demux

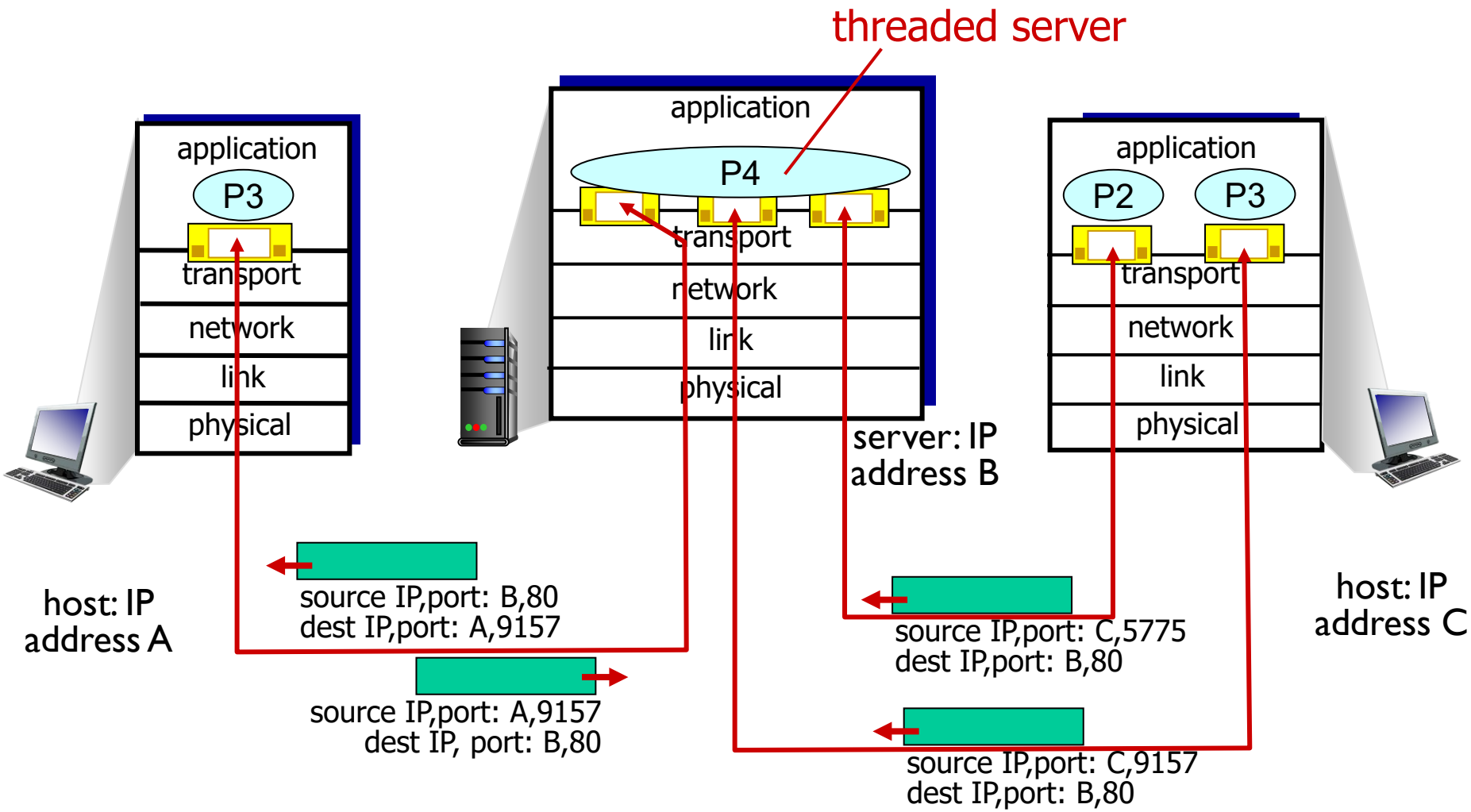
- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- demux: receiver uses all four values to direct segment to appropriate socket
- server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

Connection-oriented demux: example



three segments, all destined to IP address: B, dest port: 80 are demultiplexed to *different* sockets

Connection-oriented demux: example

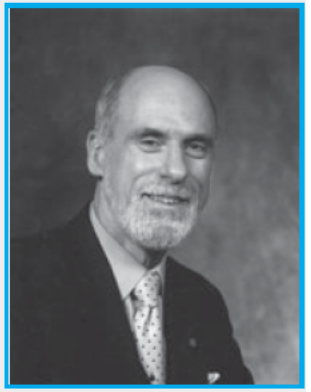


Interview with Vinton Cerf

- One of the inventors of the TCP and IP protocols.

- https://www.youtube.com/watch?v=ILiQnw0b-YQ 0:00-6:00

Vinton G. Cerf



Vinton G. Cerf is Vice President and Chief Internet Evangelist for Google. He served for over 16 years at MCI in various positions, ending up his tenure there as Senior Vice President for Technology Strategy. He is widely known as the co-designer of the TCP/IP protocols and the architecture of the Internet. During his time from 1976 to 1982 at the US Department of Defense Advanced Research Projects Agency (DARPA), he played a key role leading the

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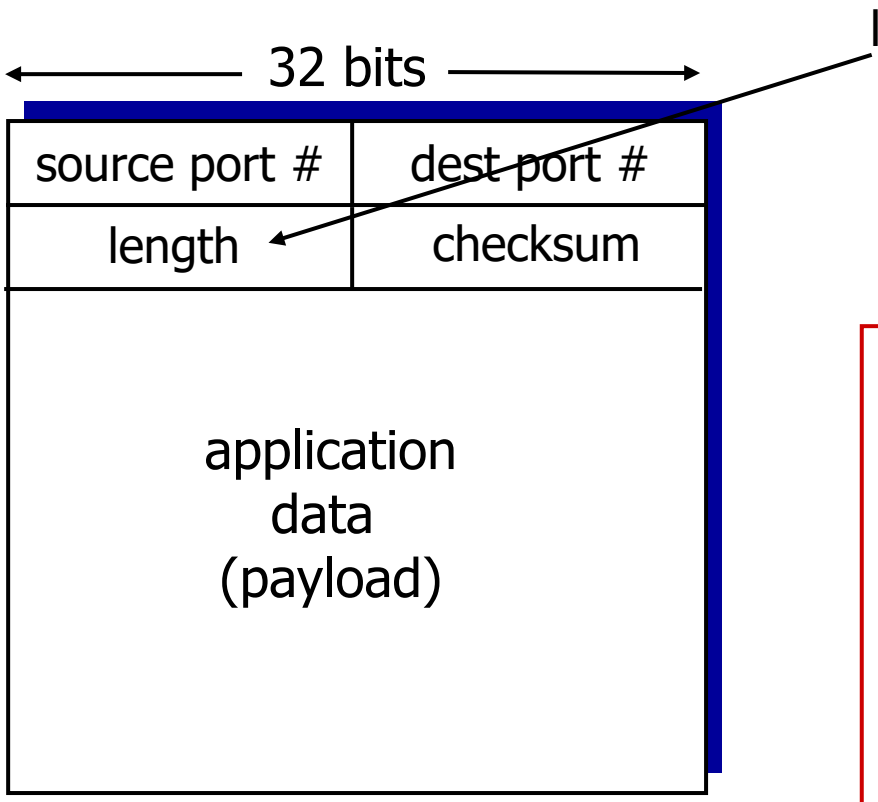
UDP: User Datagram Protocol [RFC 768]

- "best effort" service, UDP segments may be:
  - lost
  - delivered out-of-order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
  - Simple Network Management Protocol (SNMP)

UDP: segment header



UDP segment format

why is there a UDP?

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control: UDP can blast away as fast as desired

UDP checksum

Goal: detect "errors" (e.g., flipped bits) in transmitted segment

sender:

- treat segment contents, including header fields, as sequence of 16-bit integers
- checksum: addition (one's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonetheless?* More later ….

Internet checksum: example

example: add two 16-bit integers

	1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
	1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
wraparound	(1)	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

Note: when adding numbers, a carryout from the most significant bit needs to be added to the result

At the receiver, adding all words and checksum, the result should be all ones (also after wraparound). If there is a 0, some error must happen.

- The previous checksum algorithm may not be able to detect even-number bit errors.
  - Odd-number bit errors are guaranteed to detect

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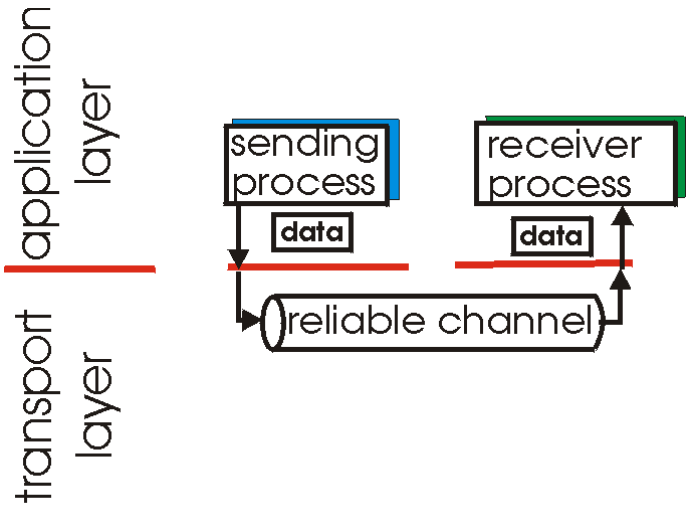
Principles of reliable data transfer

- important in application, transport, link layers
  - top-10 list of important networking topics!

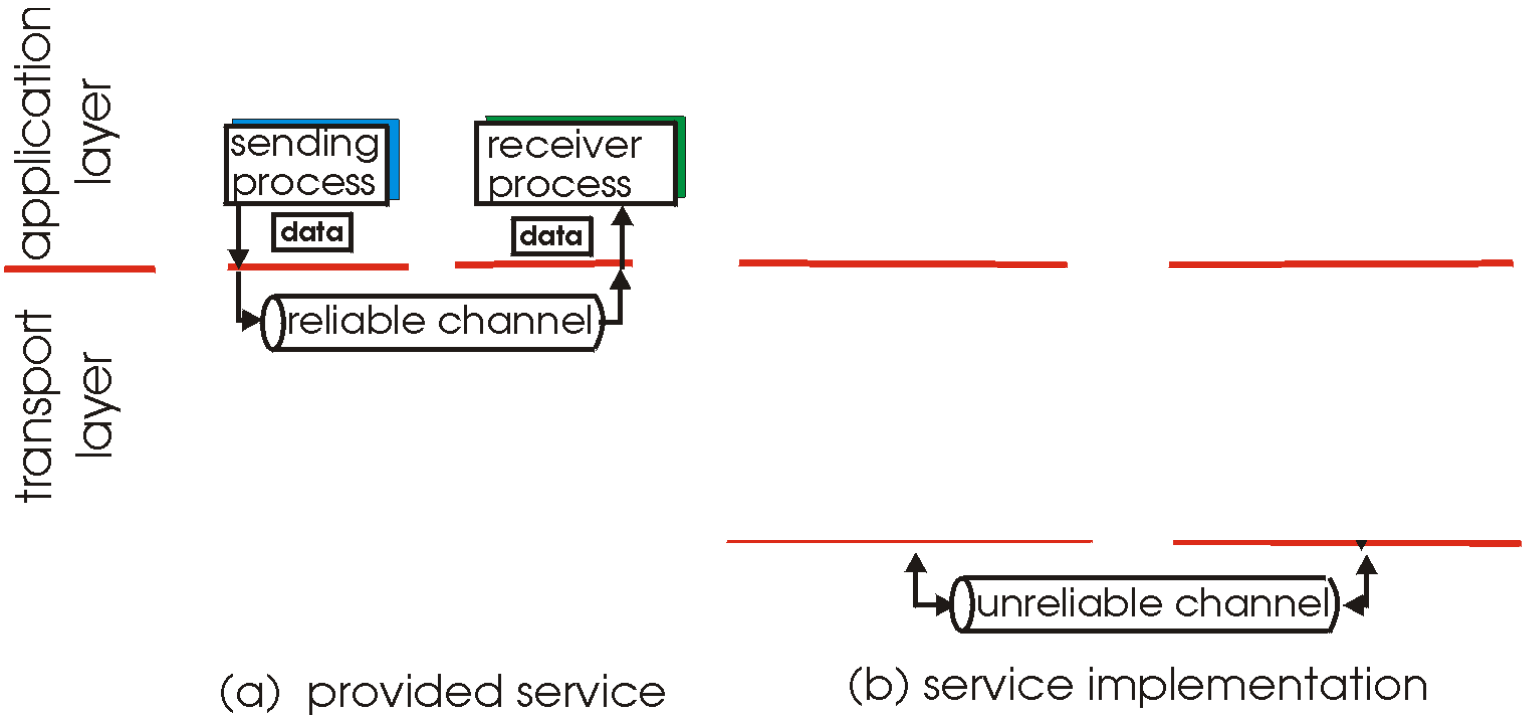


(a) provided service

- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Principles of reliable data transfer

- important in application, transport, link layers
  - top-10 list of important networking topics!



- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Principles of reliable data transfer

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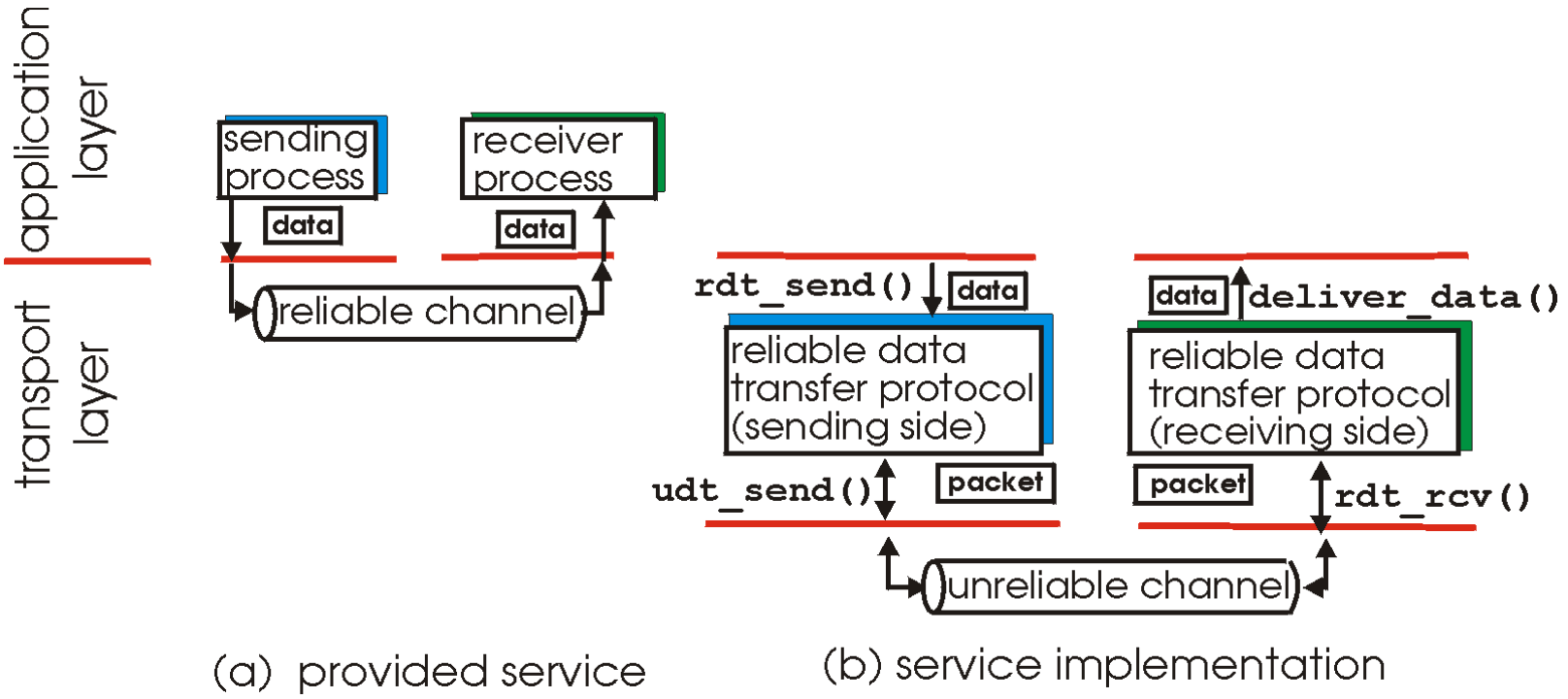


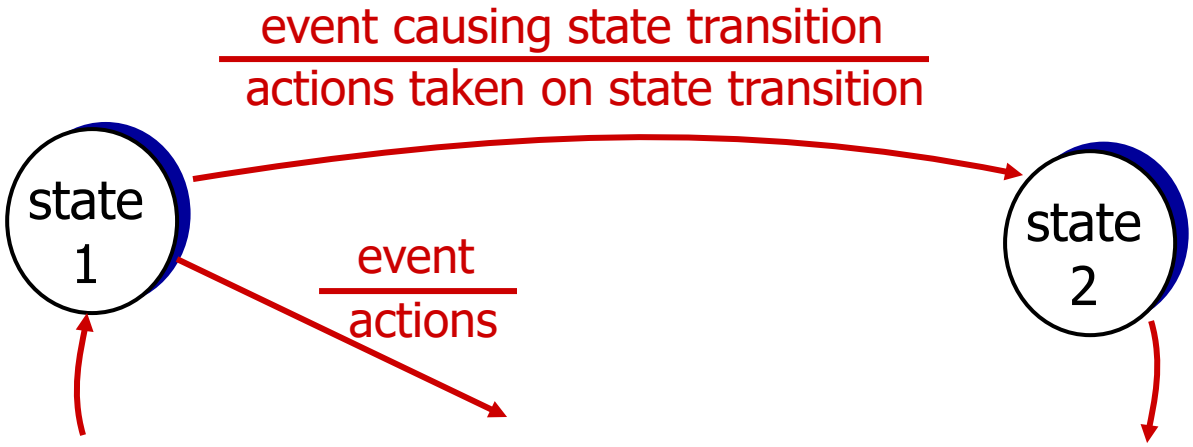
- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable data transfer: getting started

we'll:

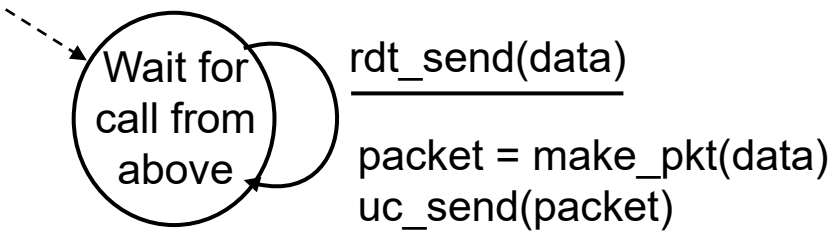
- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver

state: when in this "state" next state uniquely determined by next event

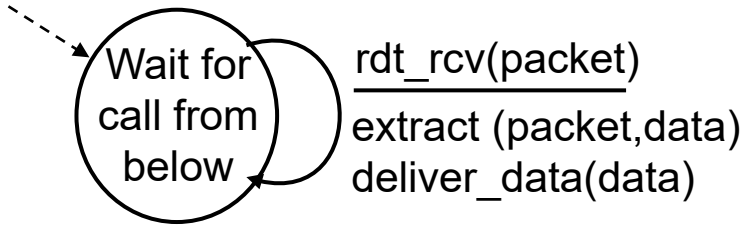


rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver reads data from underlying channel



sender



receiver

rdt2.0: channel with bit errors

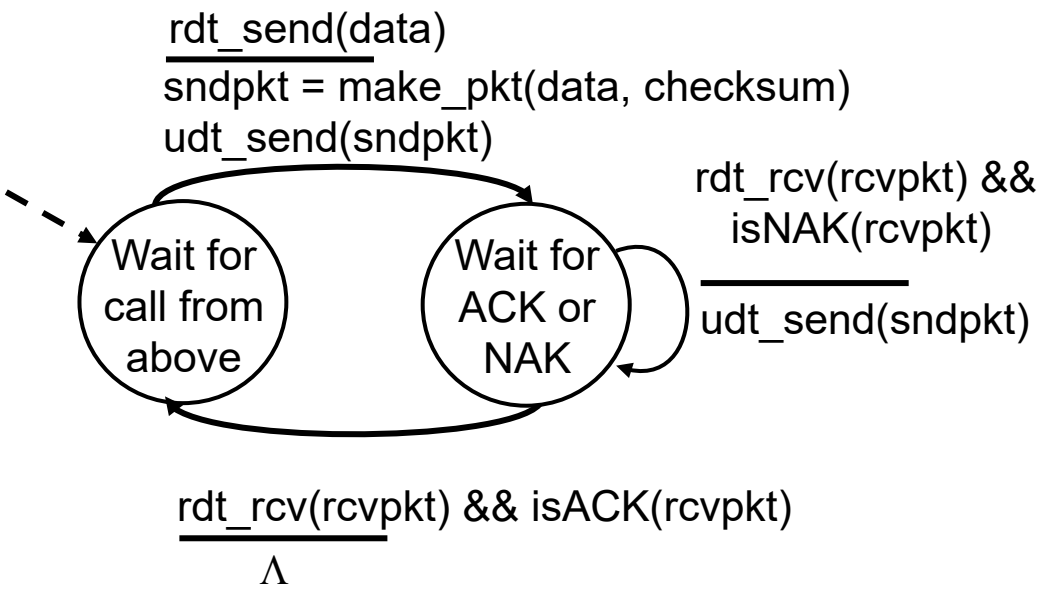
- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors:

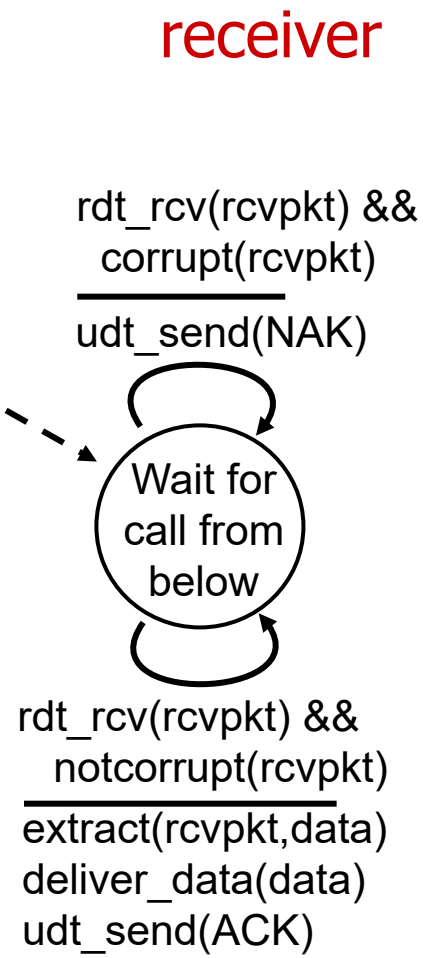
How do humans recover from "errors" during conversation?

rdt2.0: channel with bit errors

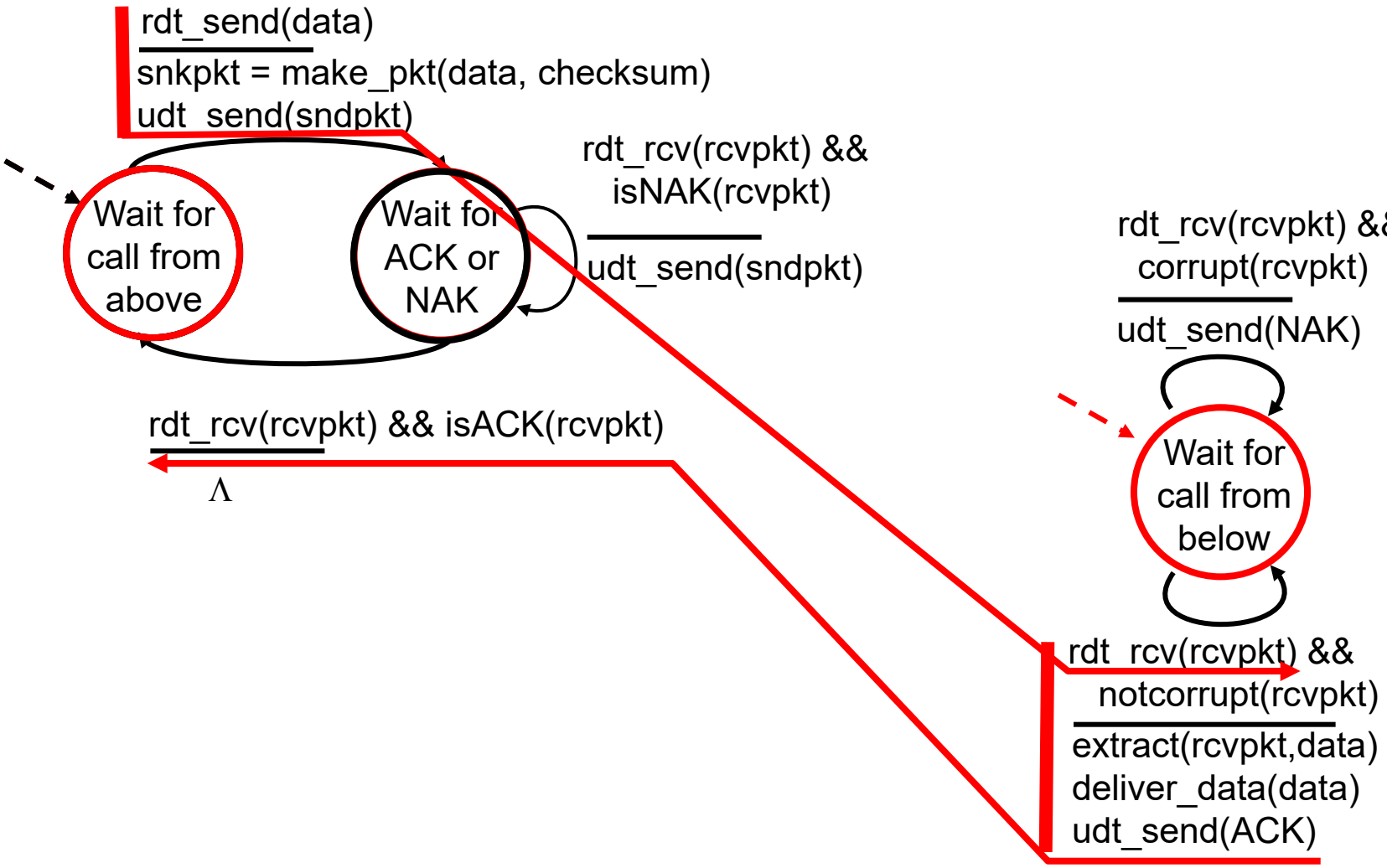
- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors:
  - *acknowledgements (ACKs):* receiver explicitly tells sender that pkt received OK
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors
  - sender retransmits pkt on receipt of NAK
- new mechanisms in `rdt2.0` (beyond `rdt1.0`):
  - error detection
  - feedback: control msgs (ACK,NAK) from receiver to sender

rdt2.0: FSM specification

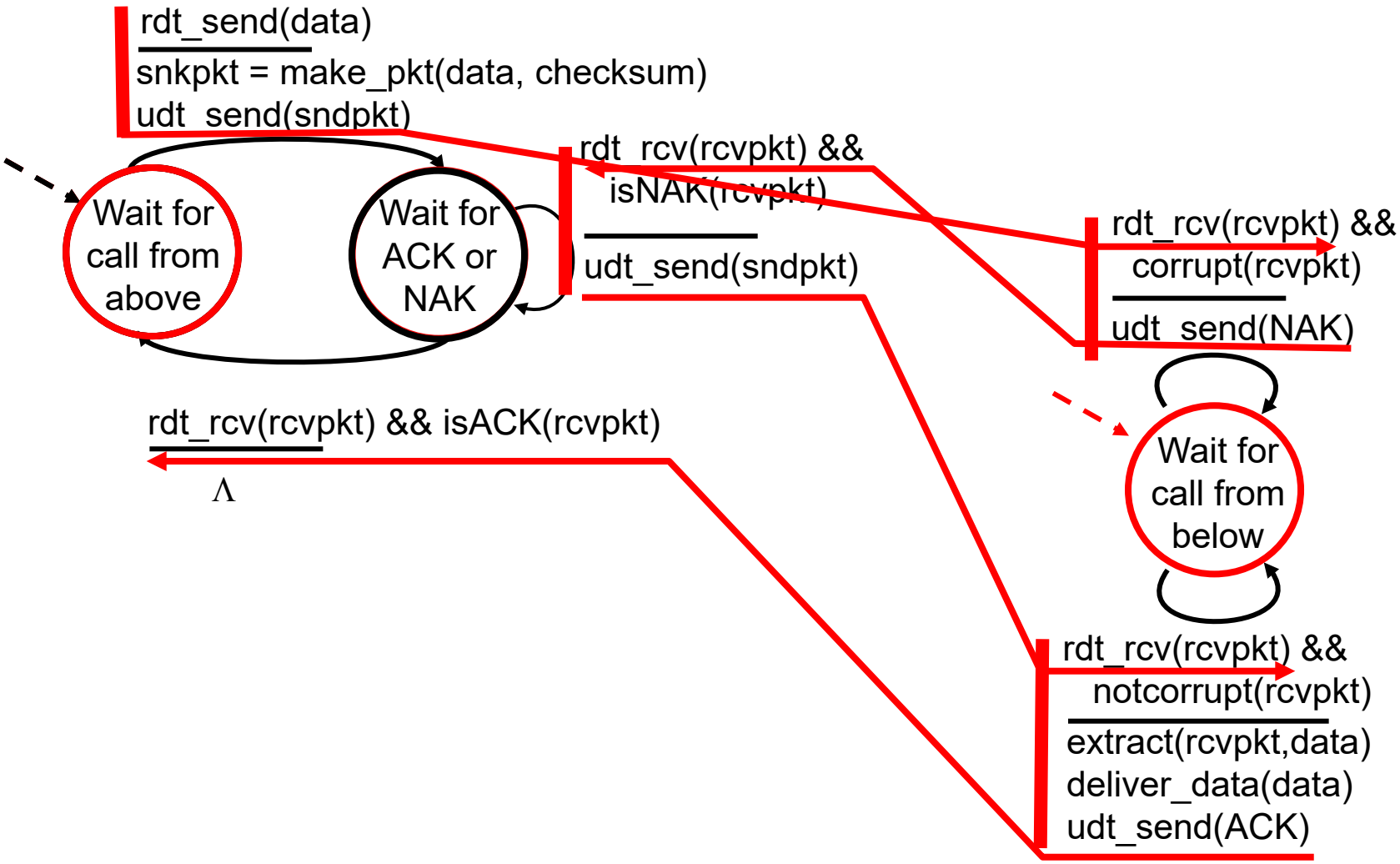




rdt2.0: operation with no errors



rdt2.0: error scenario



rdt2.0 has a fatal flaw!

what happens if ACK/NAK corrupted?

- sender doesn't know what happened at receiver!
- can't just retransmit: possible duplicate

handling duplicates:

- sender retransmits current pkt if ACK/NAK corrupted
- sender adds *sequence number* to each pkt
- receiver discards (doesn't deliver up) duplicate pkt

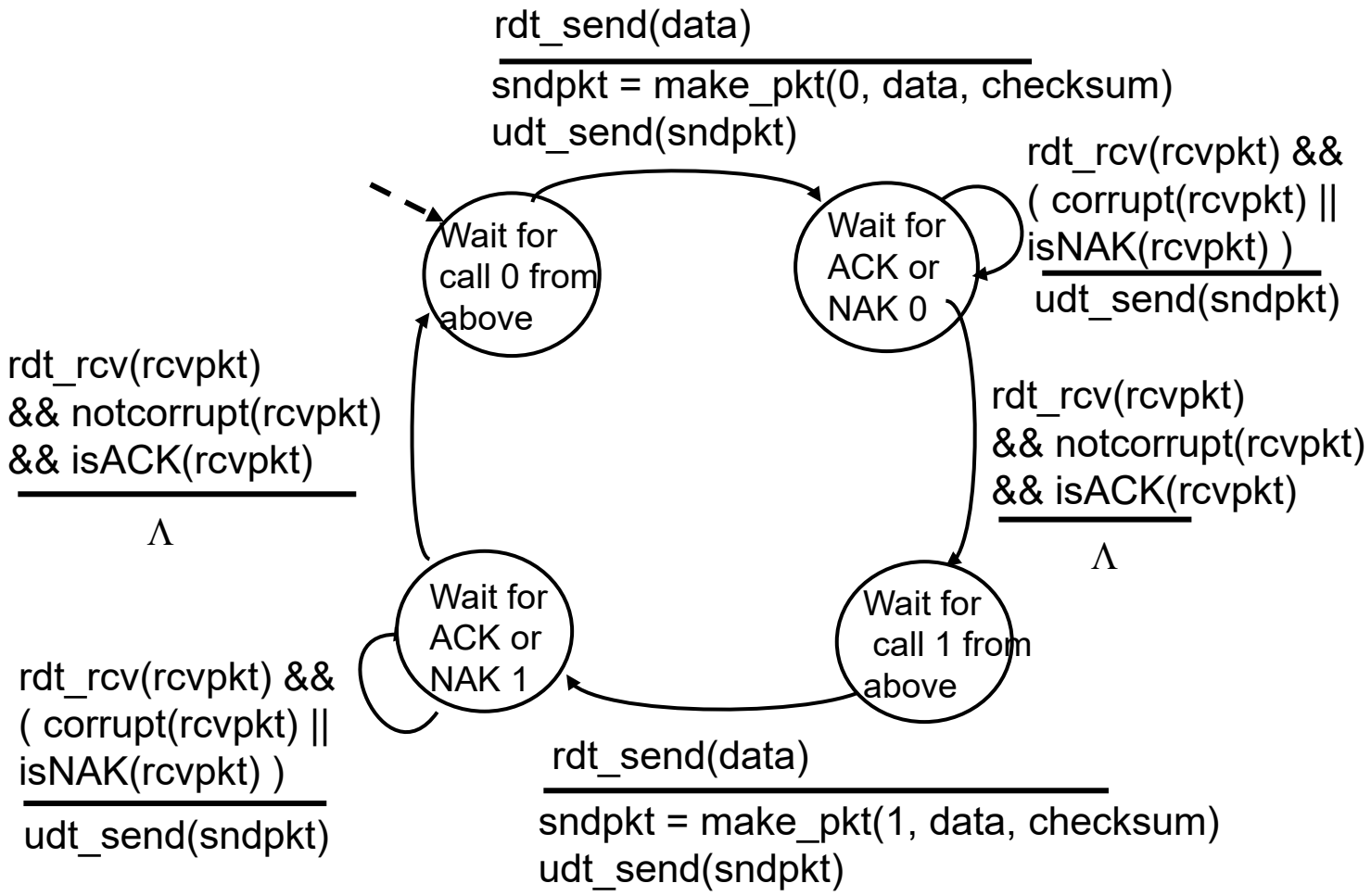
stop and wait

sender sends one packet, then waits for receiver response

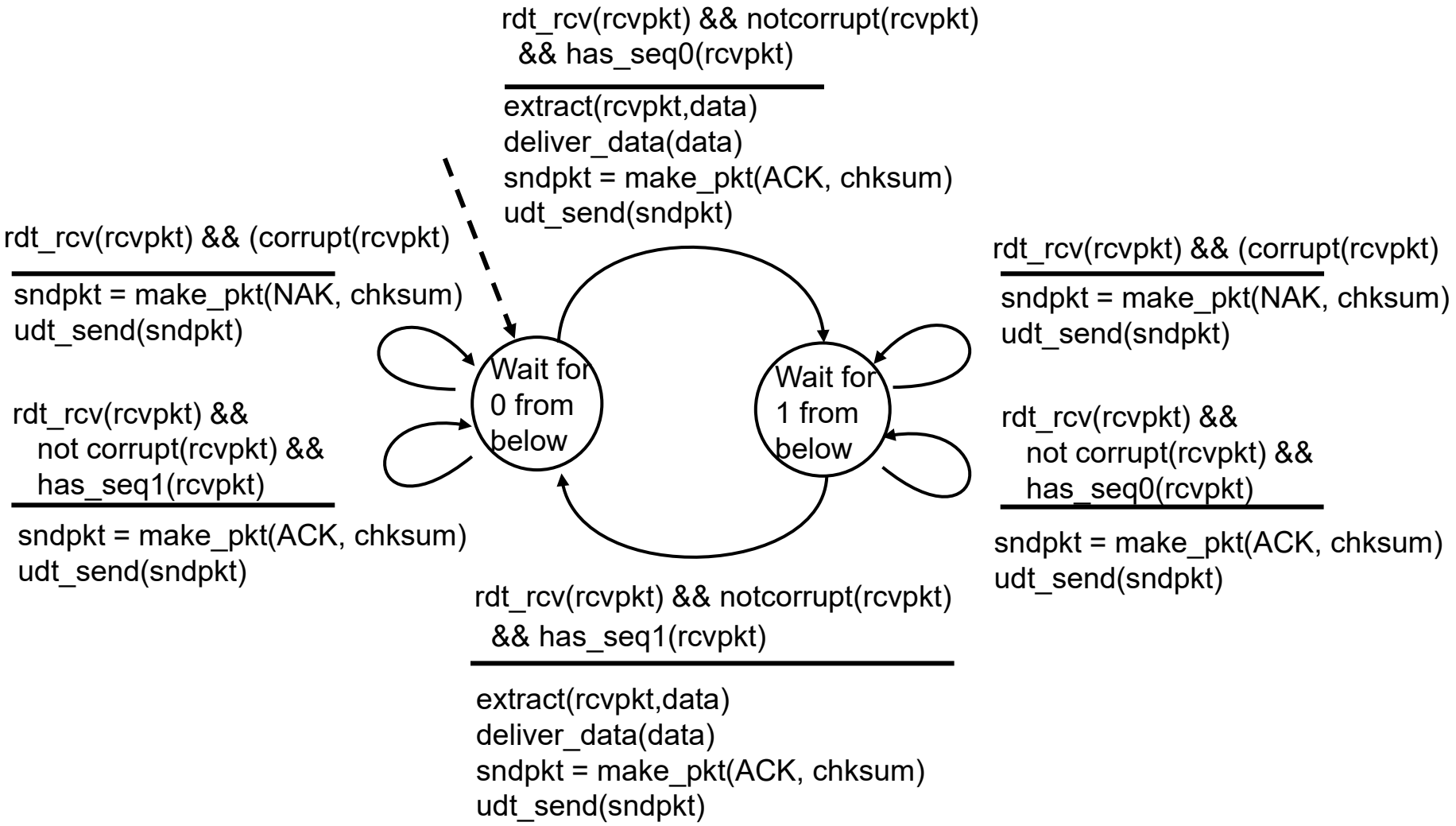
rdt2.1: idea

- Sender puts a seq num 0 or 1 to each segment.
- It sends a segment with 0 and then wait for an ACK.
- If receives ACK
  - Sends a segment with 1
- If receives NAK or corrupted ACK
  - Resends the segment with 0.
- Receiver receives a segment with 0.
  - Replies an ACK.
- Then if it receives a segment with 1.
  - The sender must received the ACK.
- If receives a segment with 0.
  - The sender did not receive the ACK.

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



rdt2.1: discussion

sender:

- seq # added to pkt
- two seq. #'s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "expected" pkt should have seq # of 0 or 1

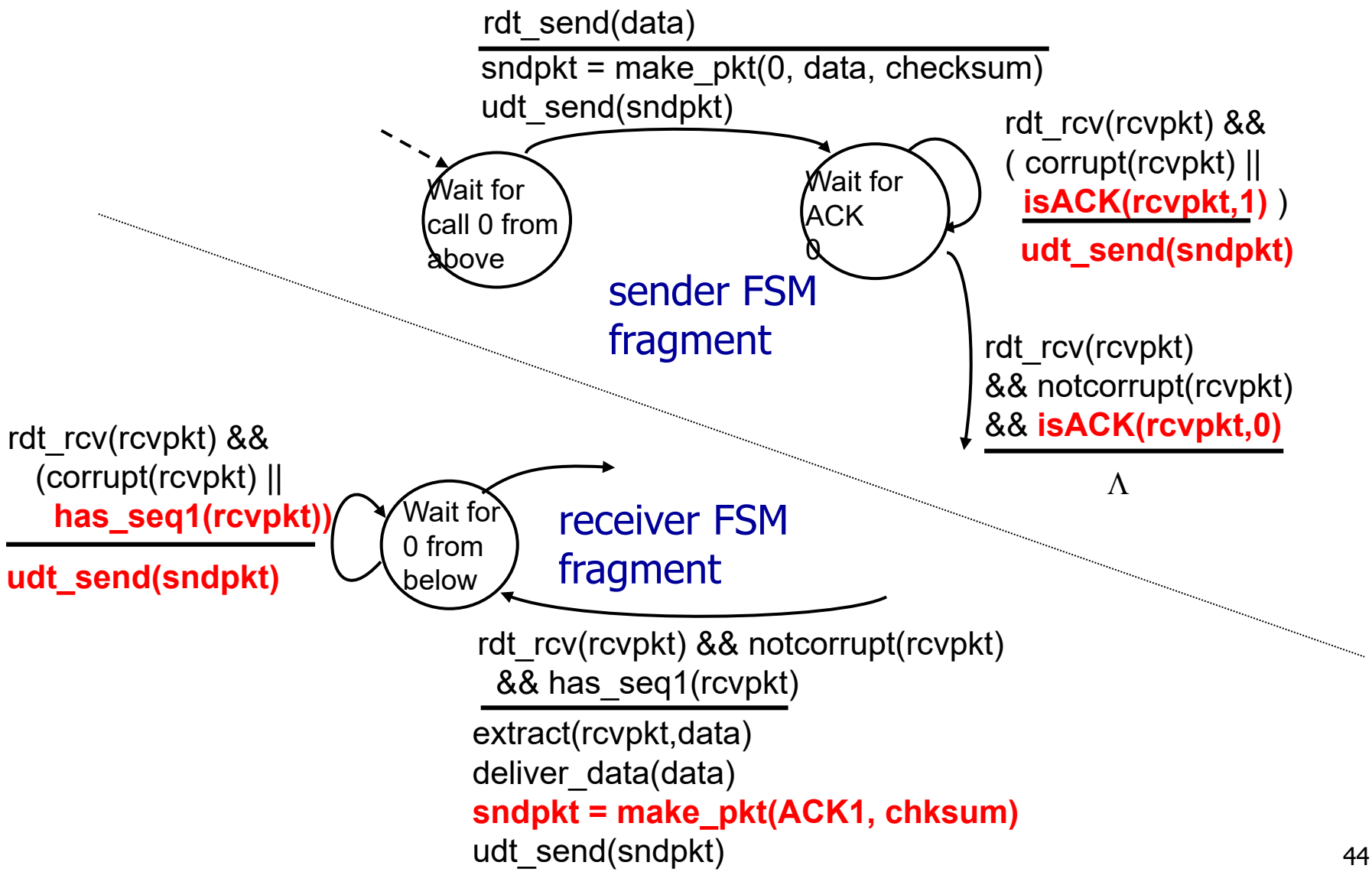
receiver:

- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

rdt2.2: a NAK-free protocol

- same functionality as rdt2.1, using ACKs only
- instead of NAK, receiver sends ACK for last pkt received OK
  - receiver must *explicitly* include seq # of pkt being ACKed
- duplicate ACK at sender results in same action as NAK: *retransmit current pkt*

rdt2.2: sender, receiver fragments



rdt3.0: channels with errors *and* loss

new assumption:

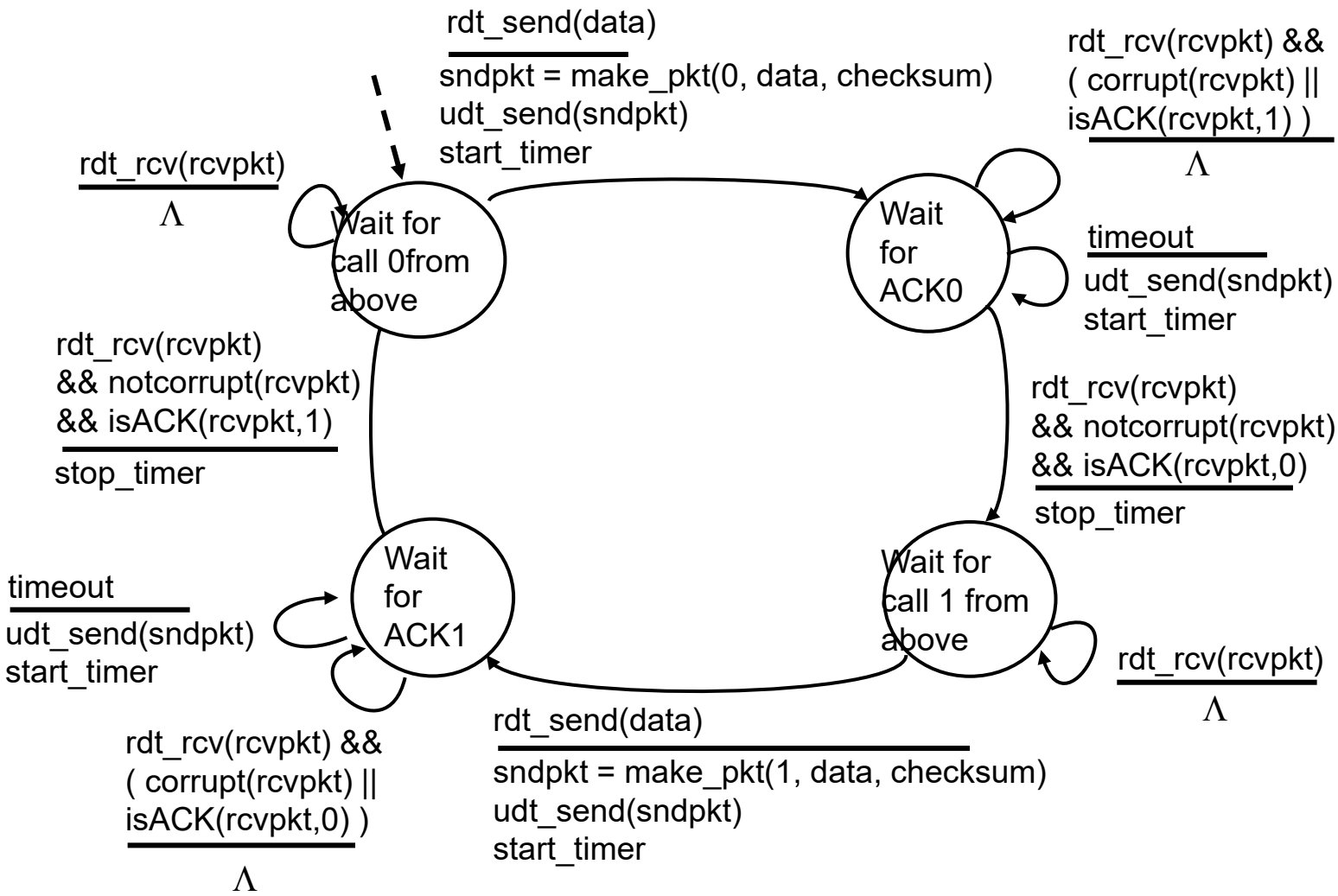
underlying channel can also lose packets (data, ACKs)

- checksum, seq. #, ACKs, retransmissions will be of help … but not enough

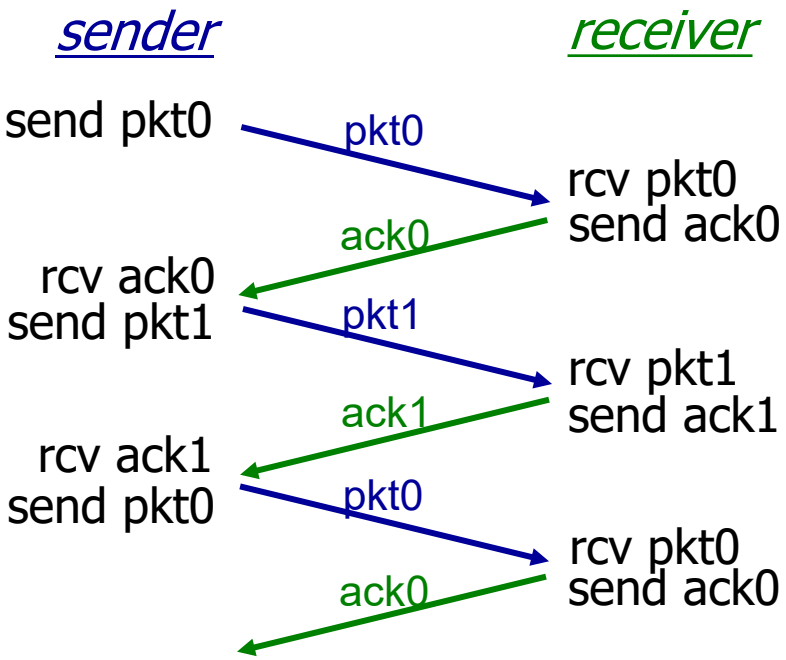
approach: sender waits "reasonable" amount of time for ACK

- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

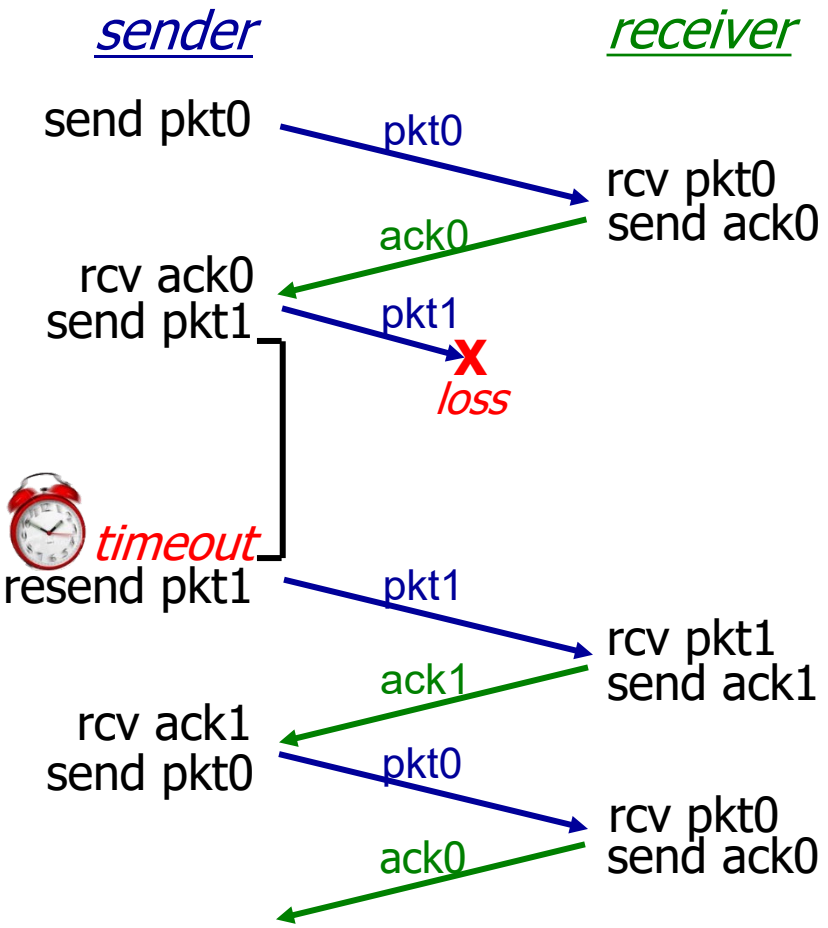
rdt3.0 sender



rdt3.0 in action

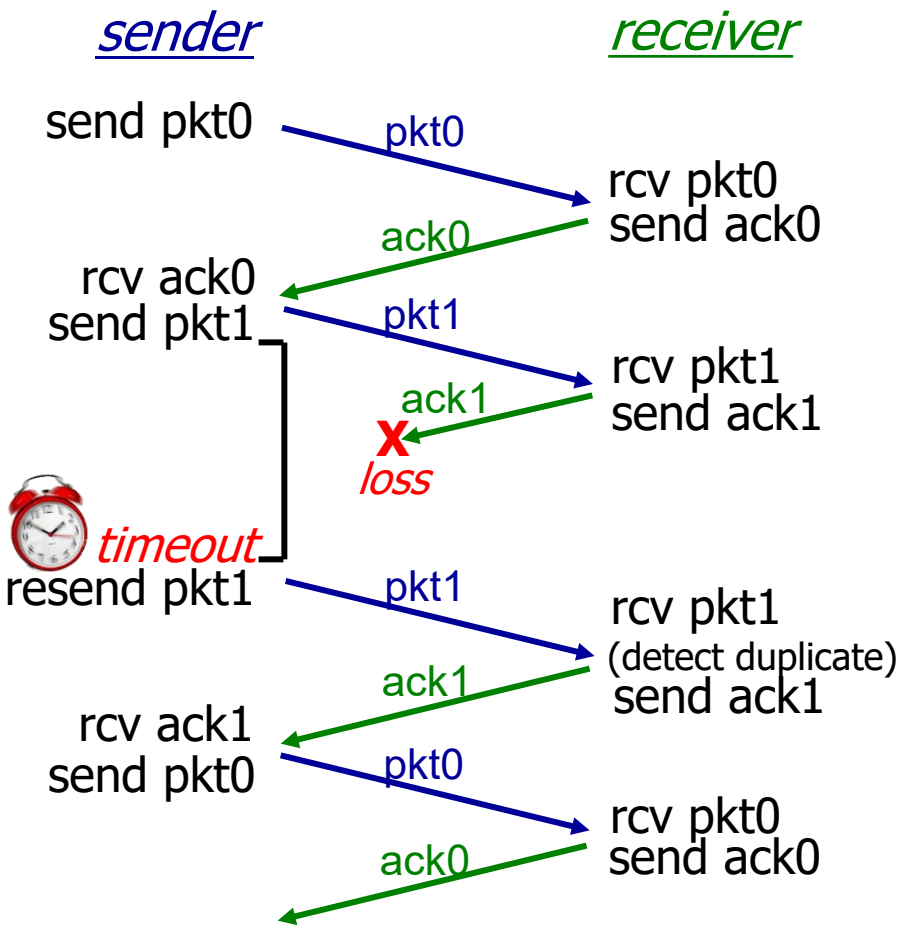


(a) no loss

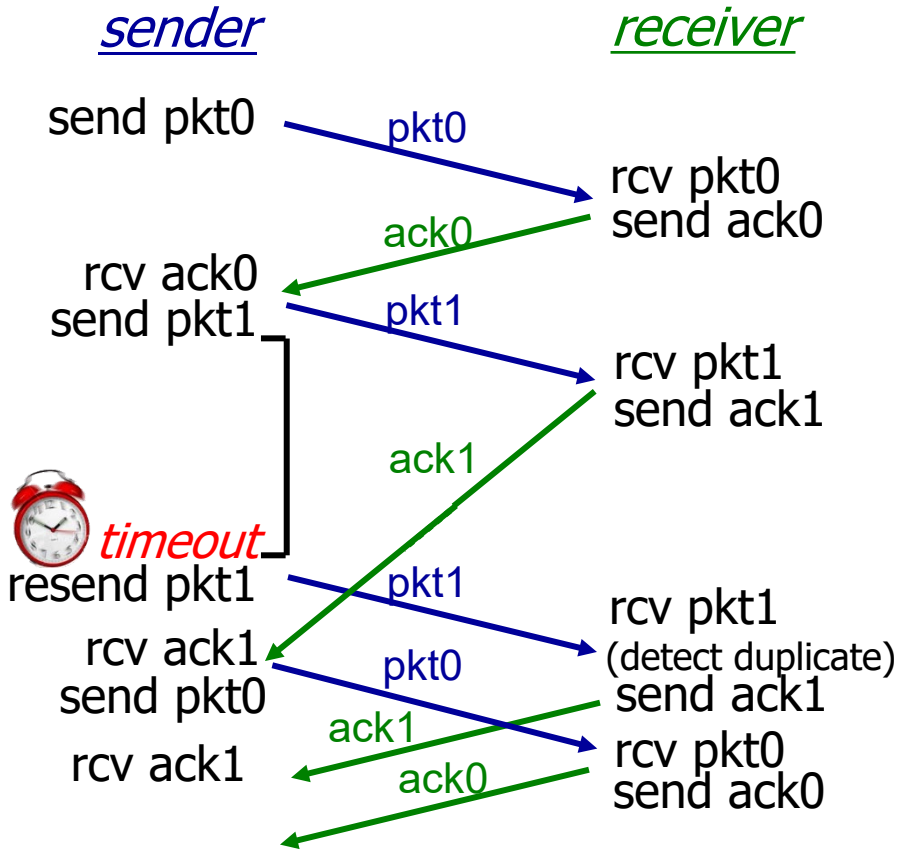


(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

Performance of rdt3.0

- rdt3.0 is correct, but performance stinks
- e.g.: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

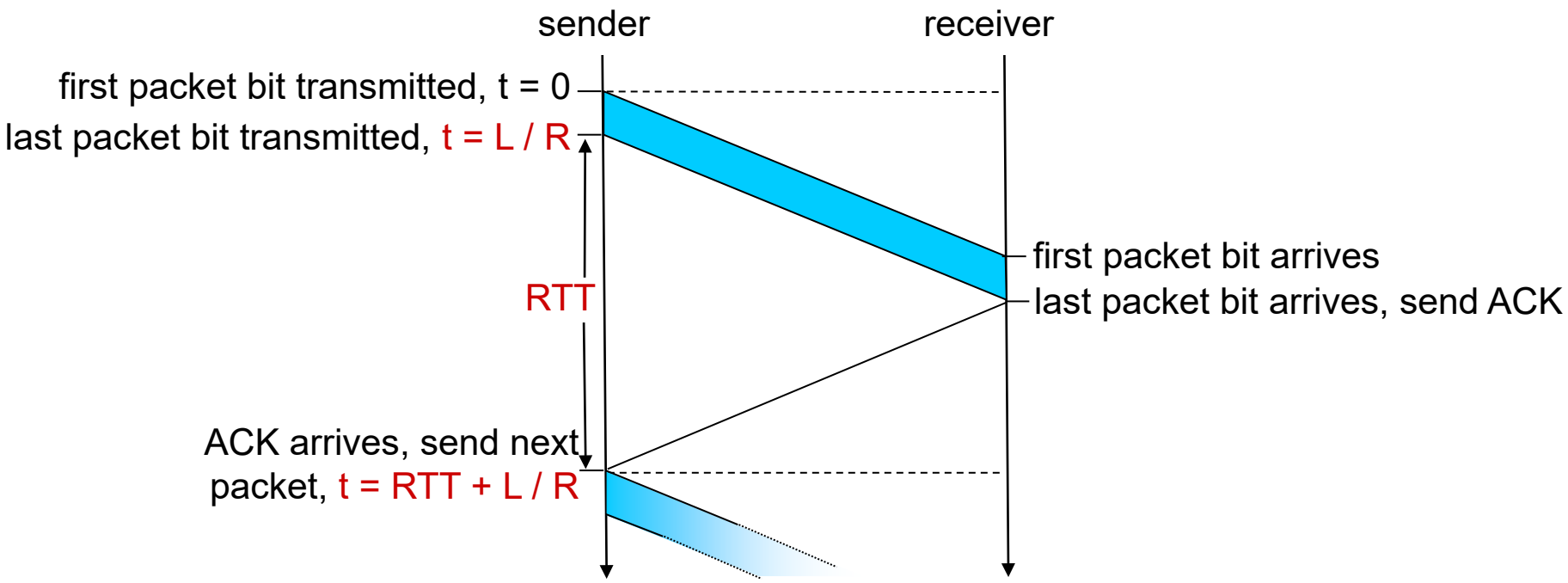
$$D_{trans} = \frac{L}{R} = \frac{8000\ bits}{10^9\ bits/sec} = 8\ microsecs$$

  - U_{sender}: *utilization* – fraction of time sender busy sending

$$U_{sender} = \frac{L/R}{RTT + L/R} = \frac{.008}{30.008} = 0.00027$$

  - if RTT=30 msec, 1KB pkt every 30 msec: 33kB/sec thruput over 1 Gbps link
- network protocol limits use of physical resources!

rdt3.0: stop-and-wait operation

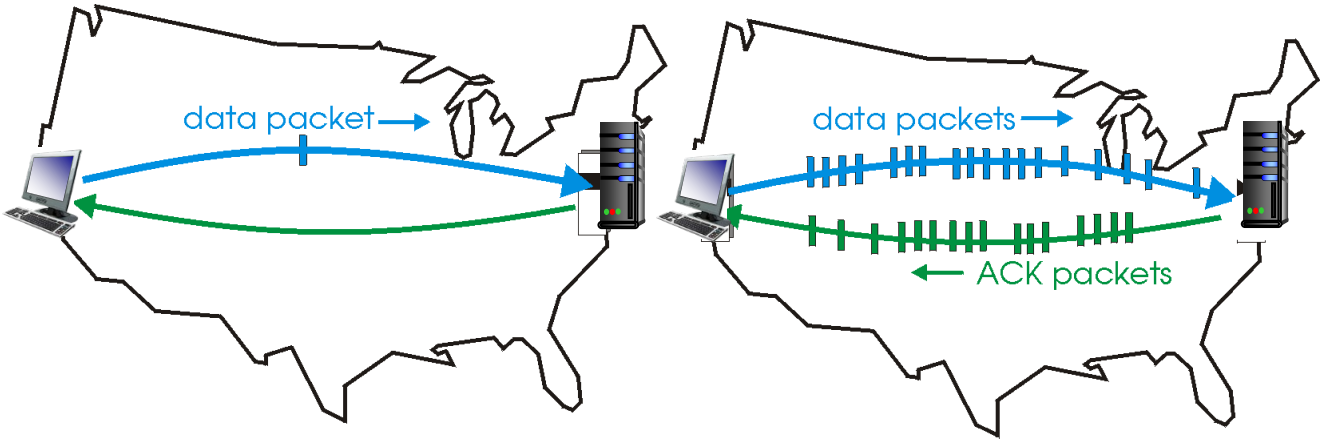


$$U_{sender} = \frac{L/R}{RTT + L/R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged pkts

- range of sequence numbers must be increased
- buffering at sender and/or receiver

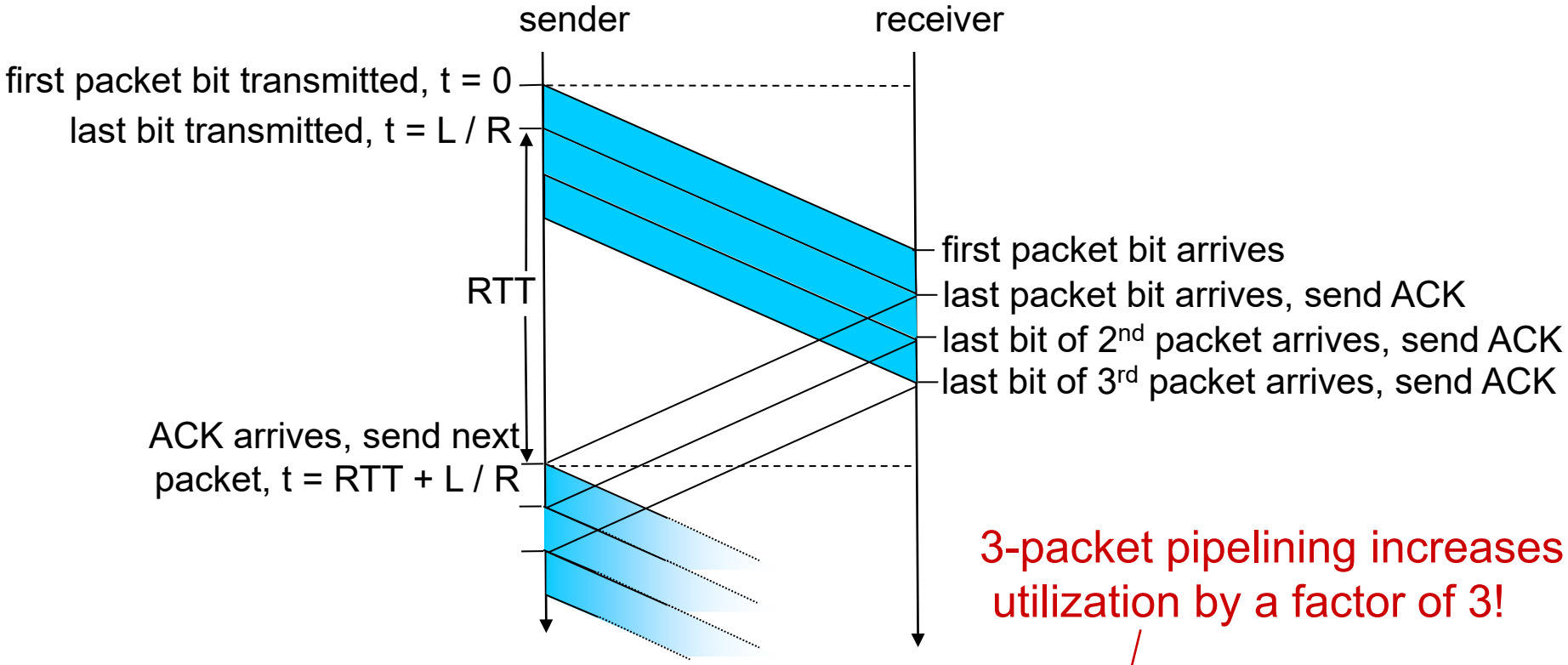


(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

- two generic forms of pipelined protocols: *go-Back-N, selective repeat*

Pipelining: increased utilization



3-packet pipelining increases utilization by a factor of 3!

$$U_{sender} = \frac{3L / R}{RTT + L / R} = \frac{.0024}{30.008} = 0.00081$$

Pipelined protocols: overview

Go-back-N:

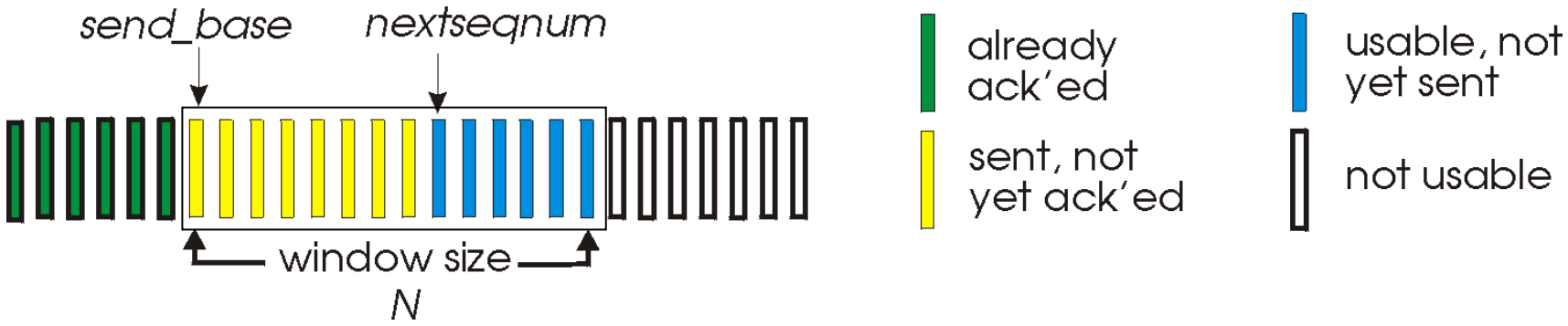
- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ack*
  - doesn't ack packet if there's a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

Selective Repeat:

- sender can have up to N unack'ed packets in pipeline
- rcvr sends *individual ack* for each packet
- sender maintains timer for each unacked packet
  - when timer expires, retransmit only that unacked packet

Go-Back-N: sender

- k-bit seq # in pkt header
- "window" of up to N, consecutive unack'ed pkts allowed

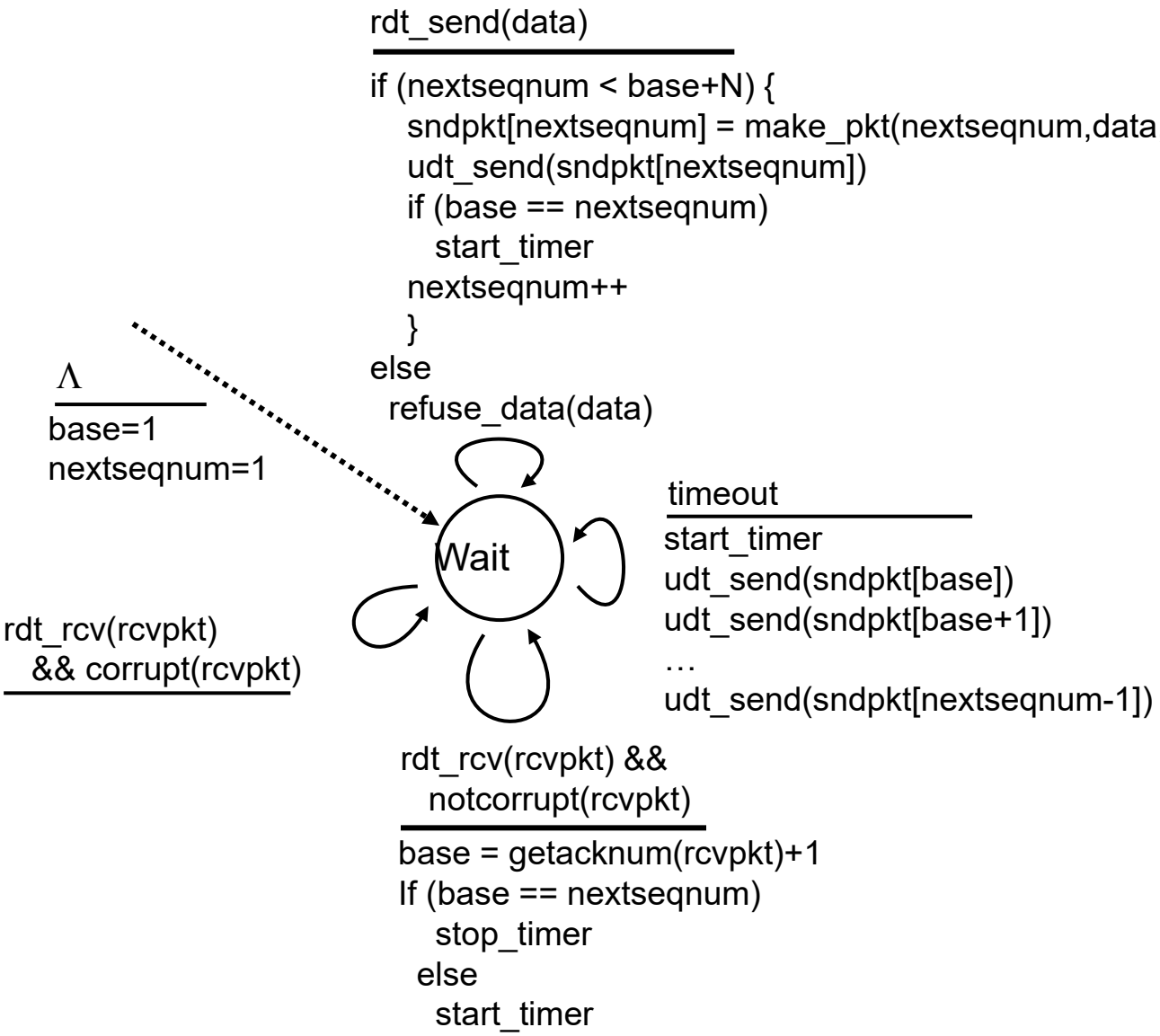


- ACK(n): ACKs all pkts up to, including seq # n - *"cumulative ACK"*
  - may receive duplicate ACKs (see receiver)
- timer for oldest in-flight pkt
- *timeout(n):* retransmit packet n and all higher seq # pkts in window

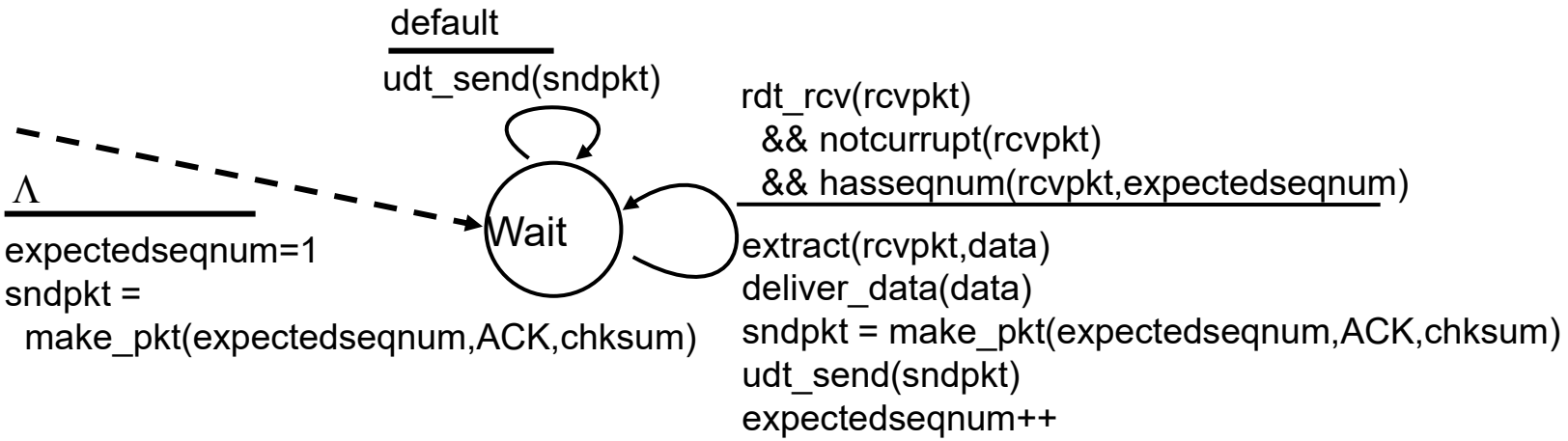
Animation

http://www.ccs-labs.org/teaching/rn/animations/gbn_sr/

GBN: sender extended FSM



GBN: receiver extended FSM

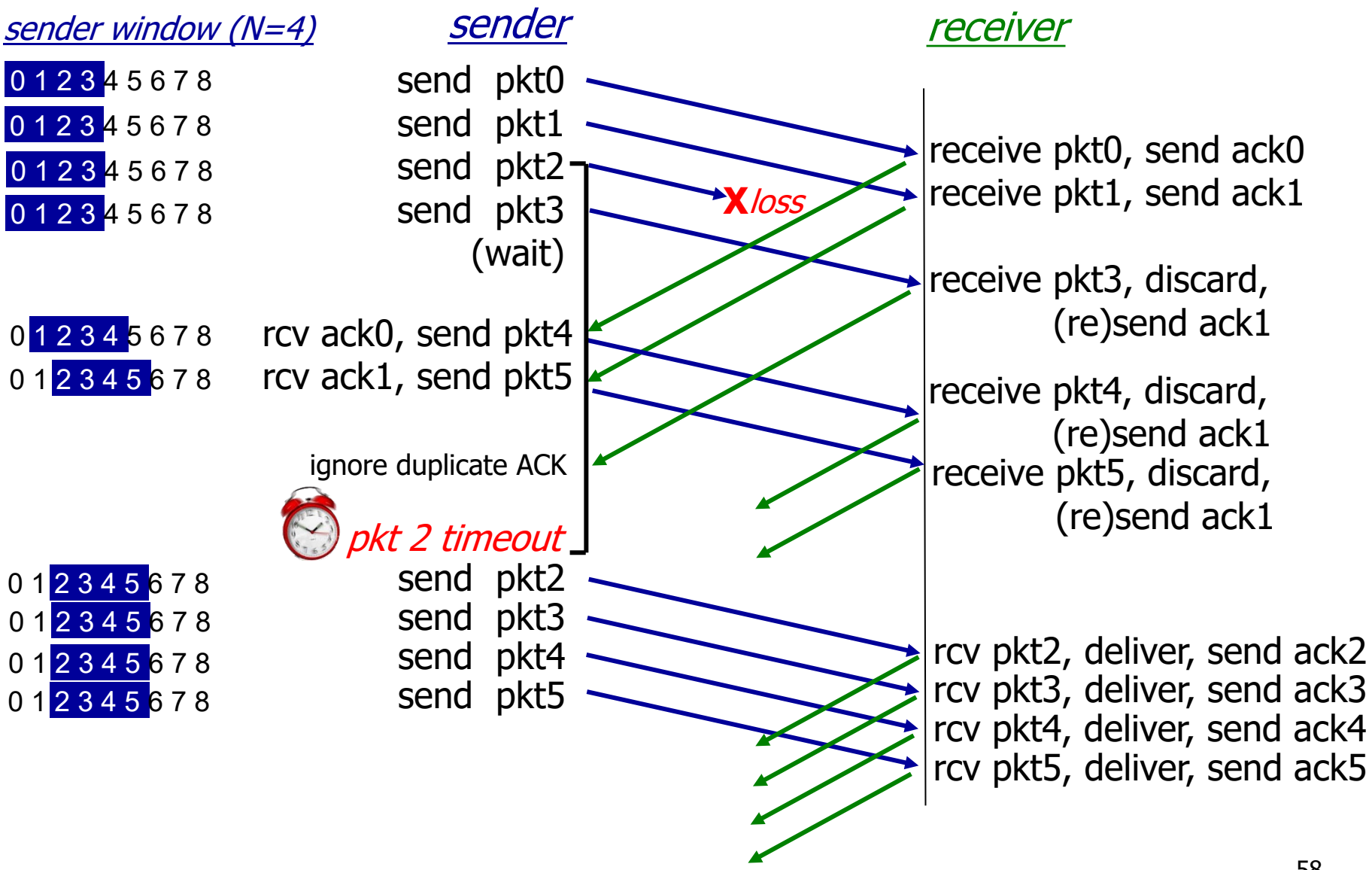


ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- may generate duplicate ACKs
- need only remember **`expectedseqnum`**

- out-of-order pkt:
  - discard (don't buffer): *no receiver buffering!*
  - re-ACK pkt with highest in-order seq #

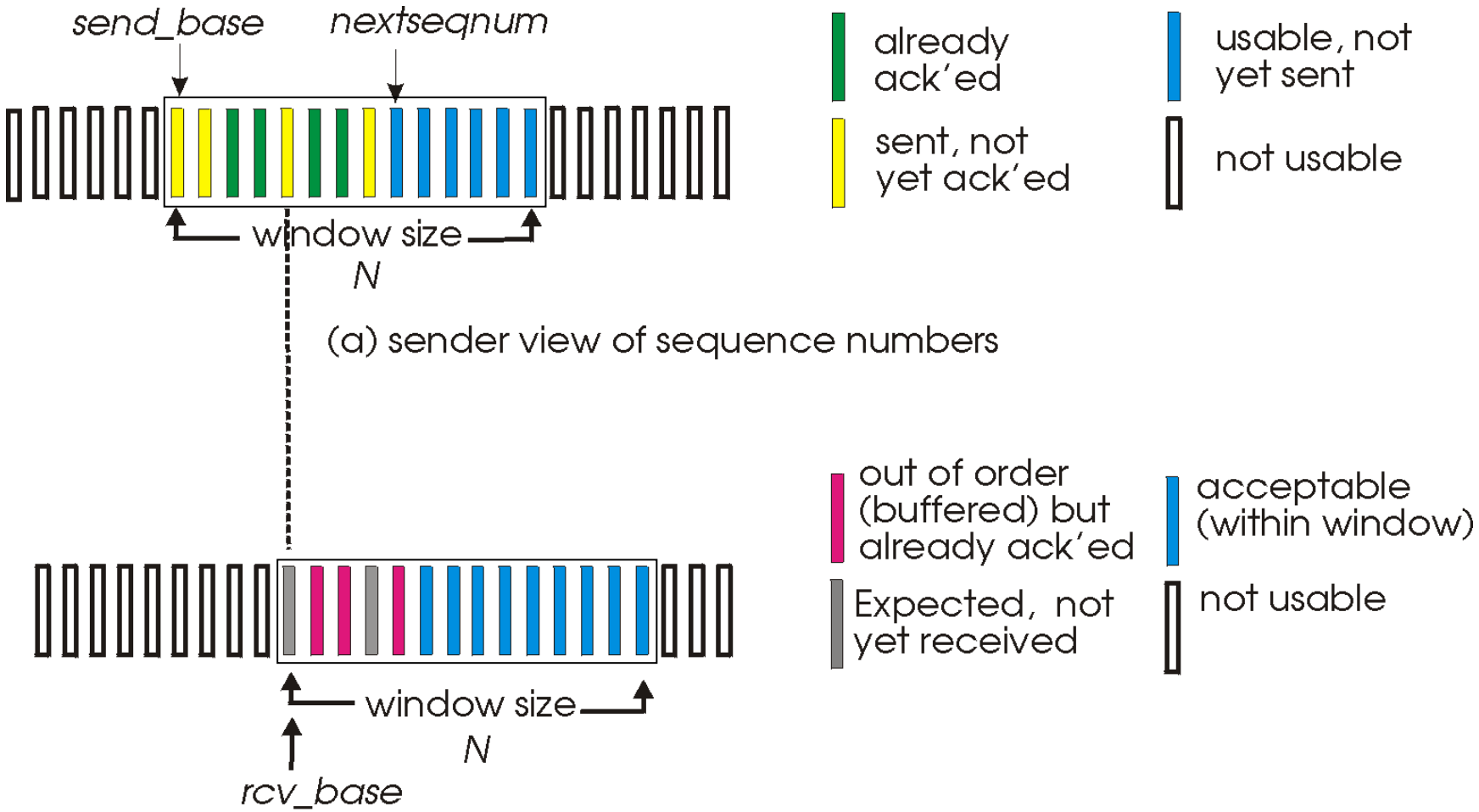
GBN in action



Selective repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
  - *N* consecutive seq #' s
  - limits seq #s of sent, unACKed pkts

Selective repeat: sender, receiver windows



Selective repeat

sender

data from above:

- if next available seq # in window, send pkt

timeout(n):

- resend pkt n, restart timer

ACK(n) in [sendbase,sendbase+N]:

- mark pkt n as received
- if n smallest unACKed pkt, advance window base to next unACKed seq #

receiver

pkt n in [rcvbase, rcvbase+N-1]

- send ACK(n)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

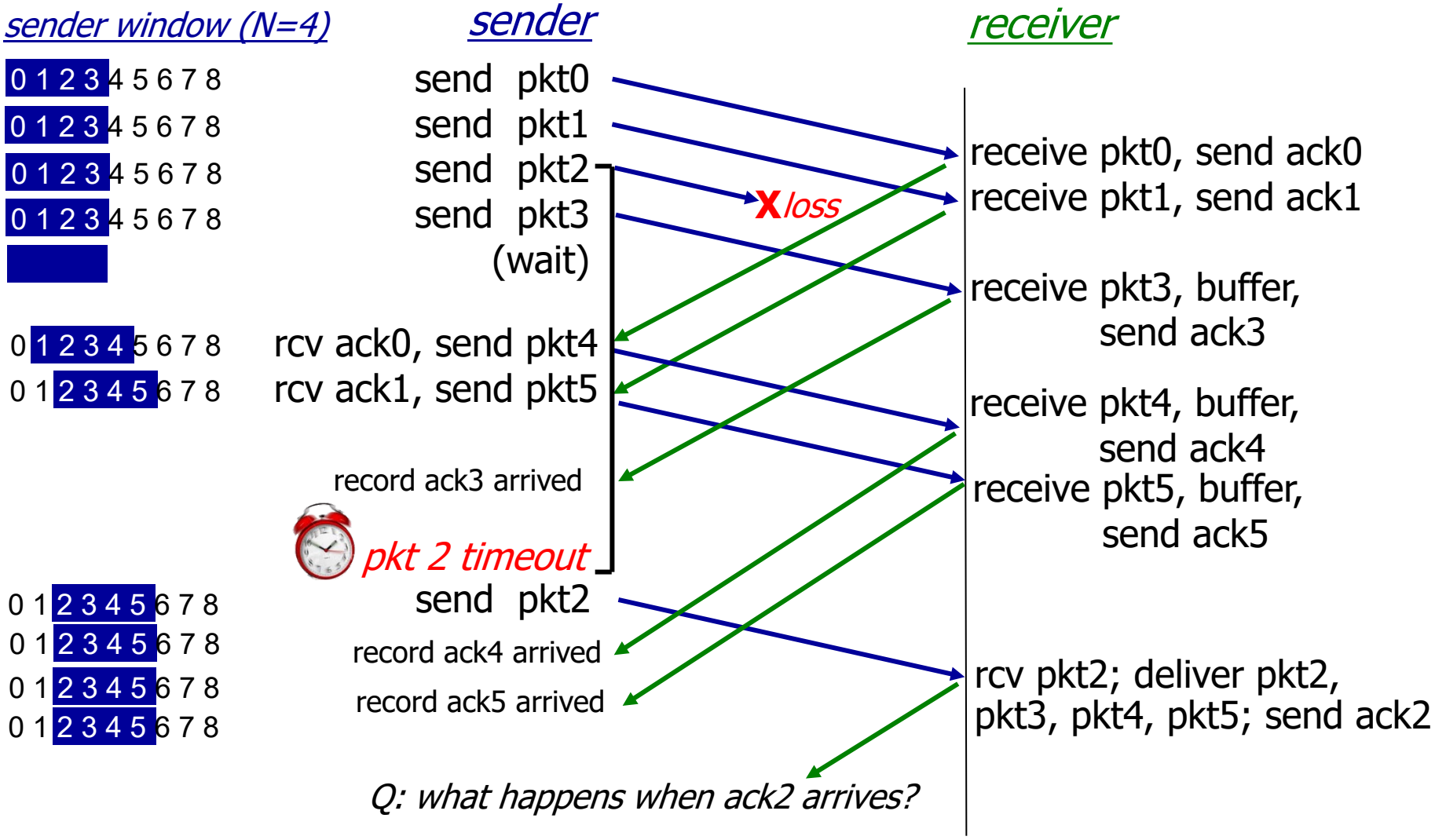
otherwise:

- ignore

Animation

http://www.ccs-labs.org/teaching/rn/animations/gbn_sr/

Selective repeat in action

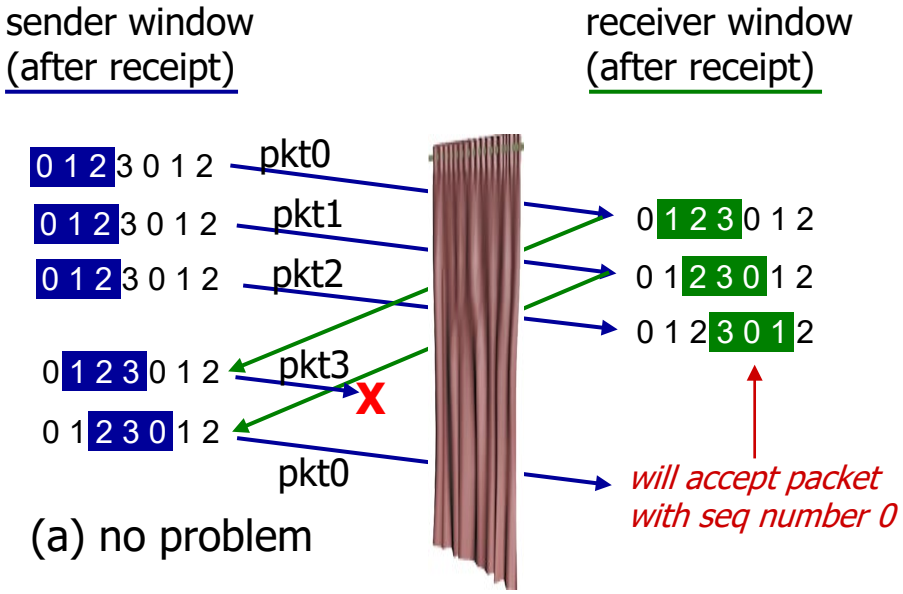


Selective repeat: dilemma

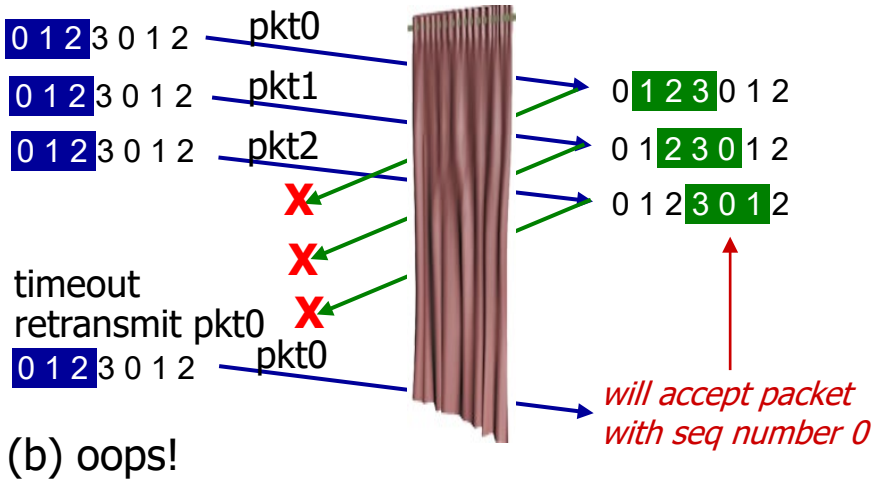
example:

- seq #'s: 0, 1, 2, 3
- window size=3
- receiver sees no difference in two scenarios!
- duplicate data accepted as new in (b)

Q: what relationship between seq # size and window size to avoid problem in (b)?



receiver can't see sender side.
receiver behavior identical in both cases!
something's (very) wrong!



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3.1 transport-layer services

3.2 multiplexing and demultiplexing

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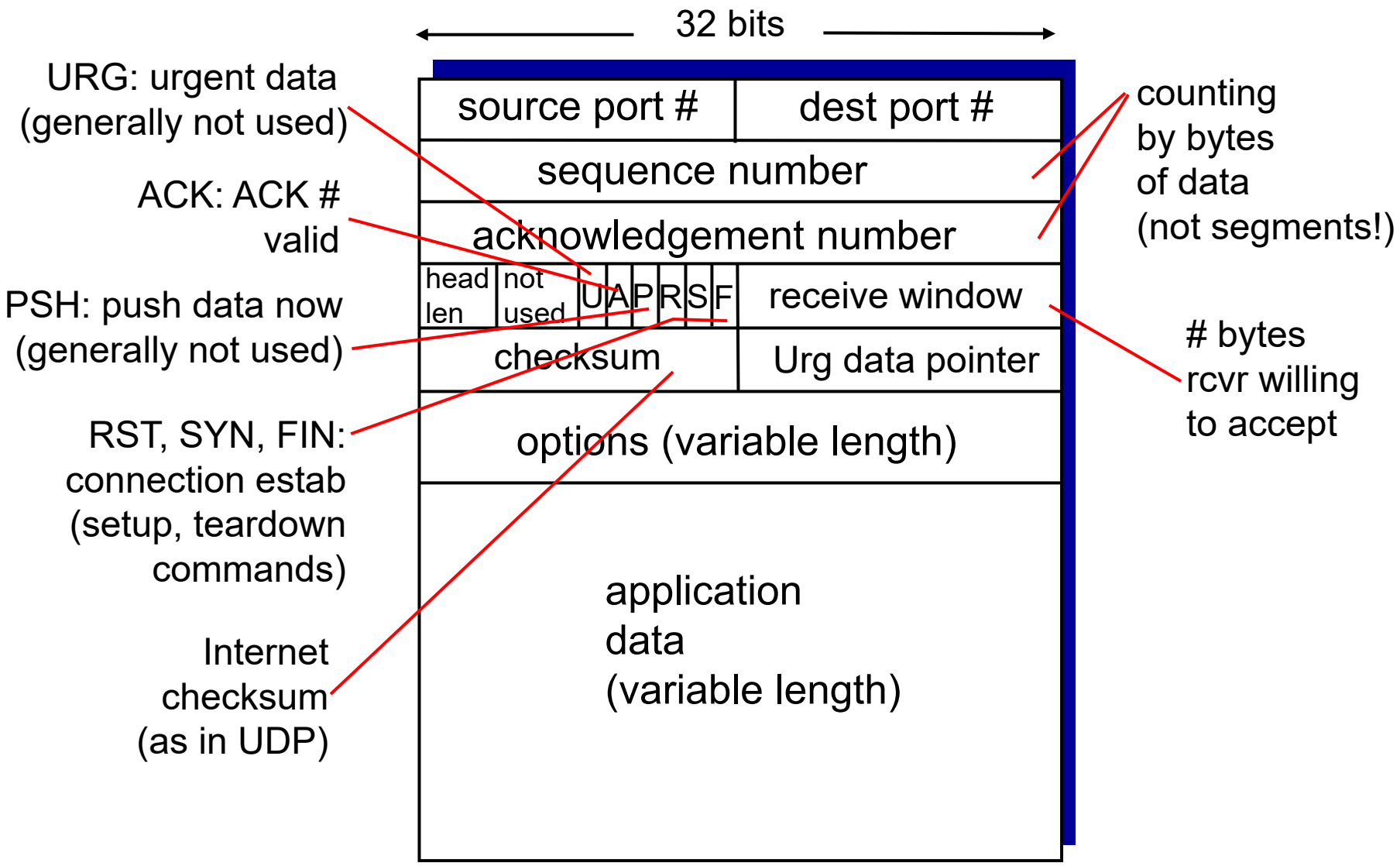
3.6 principles of congestion control

3.7 TCP congestion control

TCP: Overview RFCs: 793,1122,1323, 2018, 2581

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - no "message boundaries"
- pipelined:
  - TCP congestion and flow control set window size
- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control msgs) inits sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure



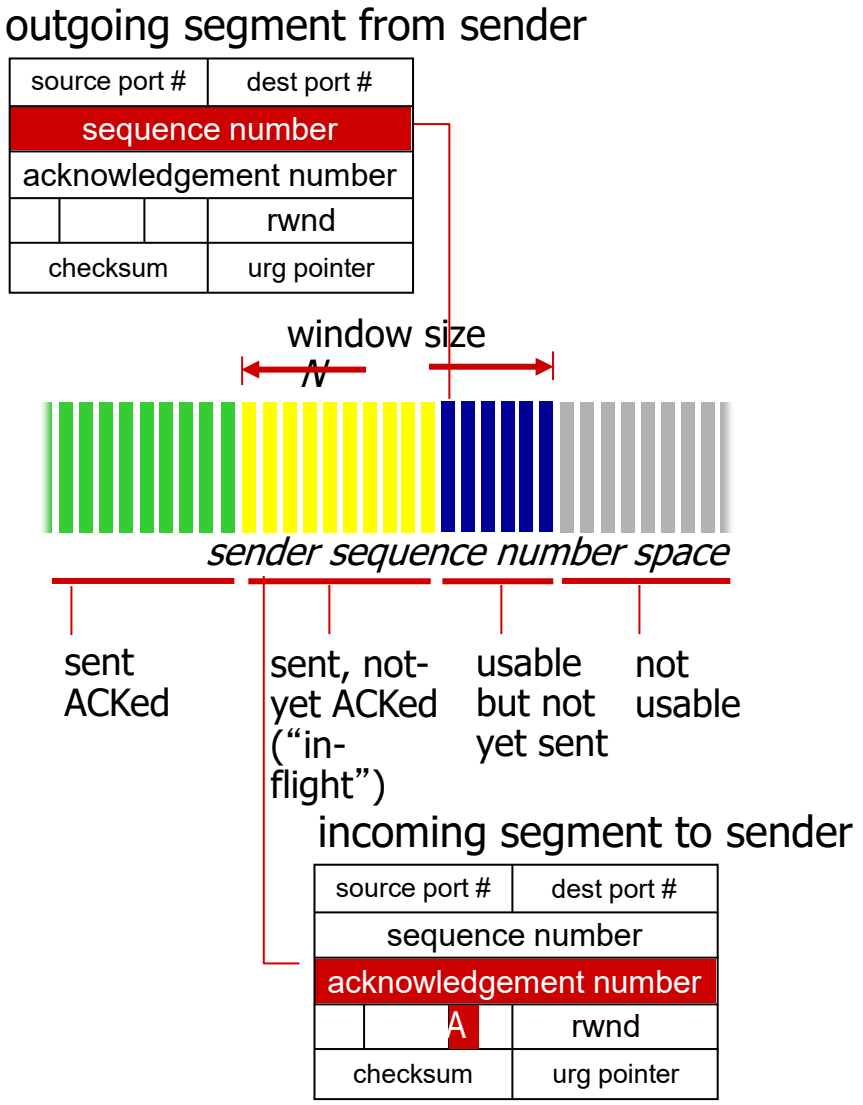
TCP seq. numbers, ACKs

sequence numbers:

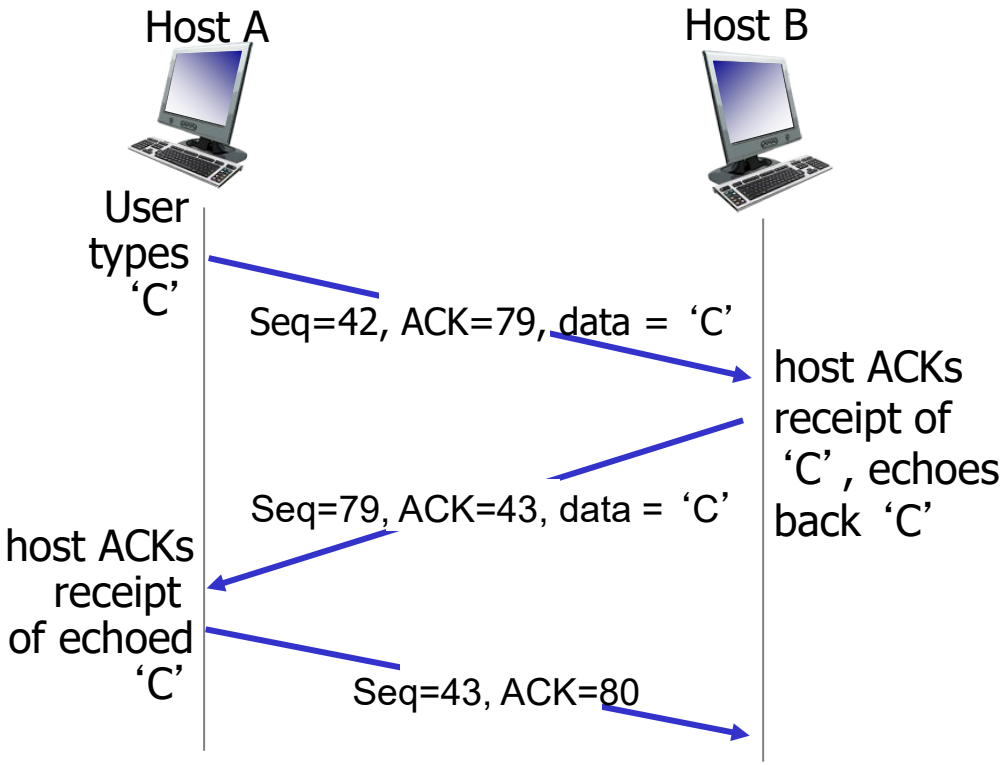
- byte stream "number" of first byte in segment's data

acknowledgements:

- seq # of next byte expected from other side
- cumulative ACK



TCP seq. numbers, ACKs



simple telnet scenario

TCP round trip time, timeout

Q: how to set TCP timeout value?

- longer than RTT
  - but RTT varies
- *too short:* premature timeout, unnecessary retransmissions
- *too long:* slow reaction to segment loss

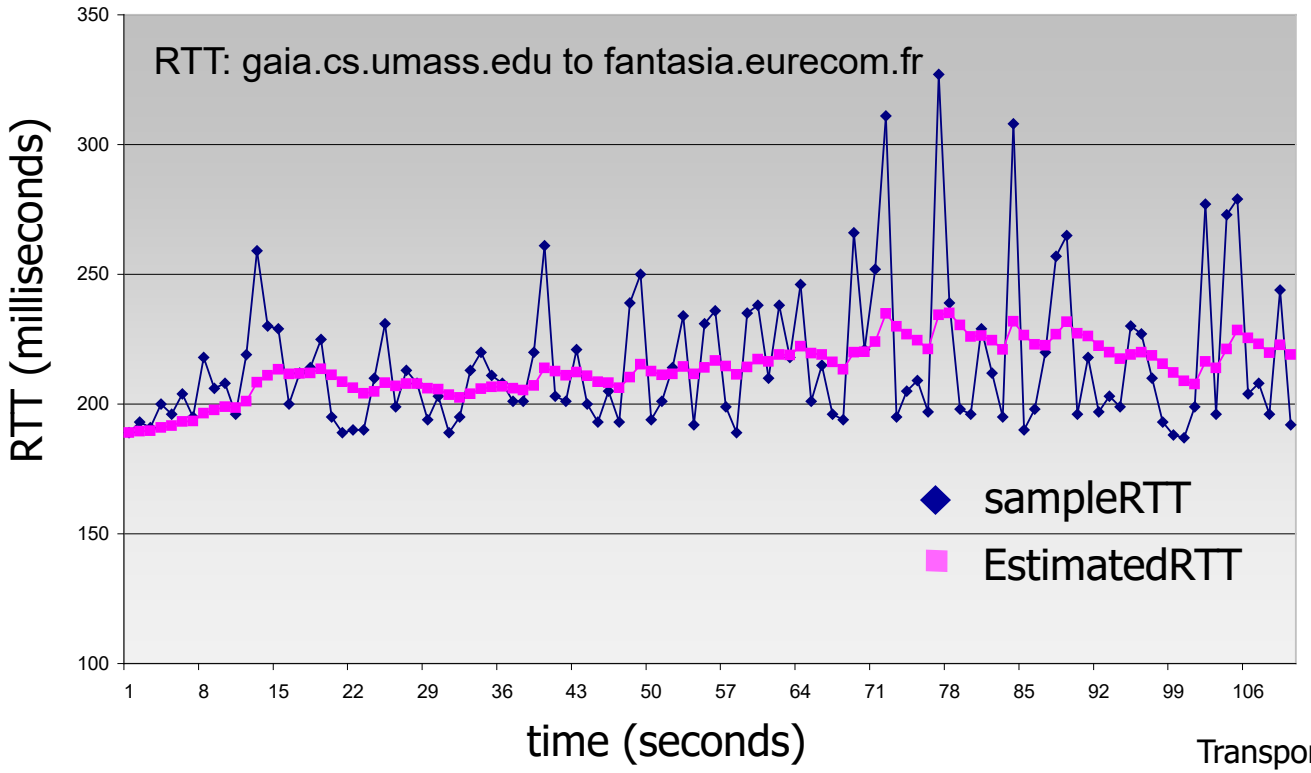
Q: how to estimate RTT?

- **`SampleRTT`**: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- **`SampleRTT`** will vary, want estimated RTT "smoother"
  - average several *recent* measurements, not just current **`SampleRTT`**

TCP round trip time, timeout

EstimatedRTT = (1- α)*EstimatedRTT + α*SampleRTT

- exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: α = 0.125



TCP round trip time, timeout

- timeout interval: **`EstimatedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`** -> larger safety margin
- estimate SampleRTT deviation from EstimatedRTT:

$$\texttt{DevRTT} = (1-\beta)*\texttt{DevRTT} + \beta*|\texttt{SampleRTT}-\texttt{EstimatedRTT}|$$

(typically, $\beta = 0.25$)

$$\texttt{TimeoutInterval} = \texttt{EstimatedRTT} + 4*\texttt{DevRTT}$$

estimated RTT

"safety margin"

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TCP reliable data transfer

- TCP creates rdt service on top of IP's unreliable service
  - pipelined segments
  - cumulative acks
  - single retransmission timer
- retransmissions triggered by:
  - timeout events
  - duplicate acks

let's initially consider simplified TCP sender:

- ignore duplicate acks
- ignore flow control, congestion control

TCP sender events:

data rcvd from app:

- create segment with seq #
- seq # is byte-stream number of first data byte in segment
- start timer if not already running
  - think of timer as for oldest unacked segment
  - expiration interval: `TimeOutInterval`

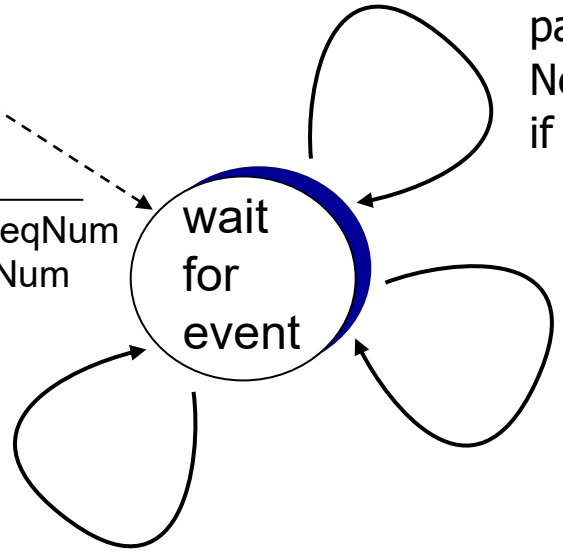
timeout:

- retransmit segment that caused timeout
- restart timer

ack rcvd:

- if ack acknowledges previously unacked segments
  - update what is known to be ACKed
  - start timer if there are still unacked segments

TCP sender (simplified)



Λ

NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum

data received from application above

create segment, seq. #: NextSeqNum
pass segment to IP (i.e., "send")
NextSeqNum = NextSeqNum + length(data)
if (timer currently not running)
    start timer

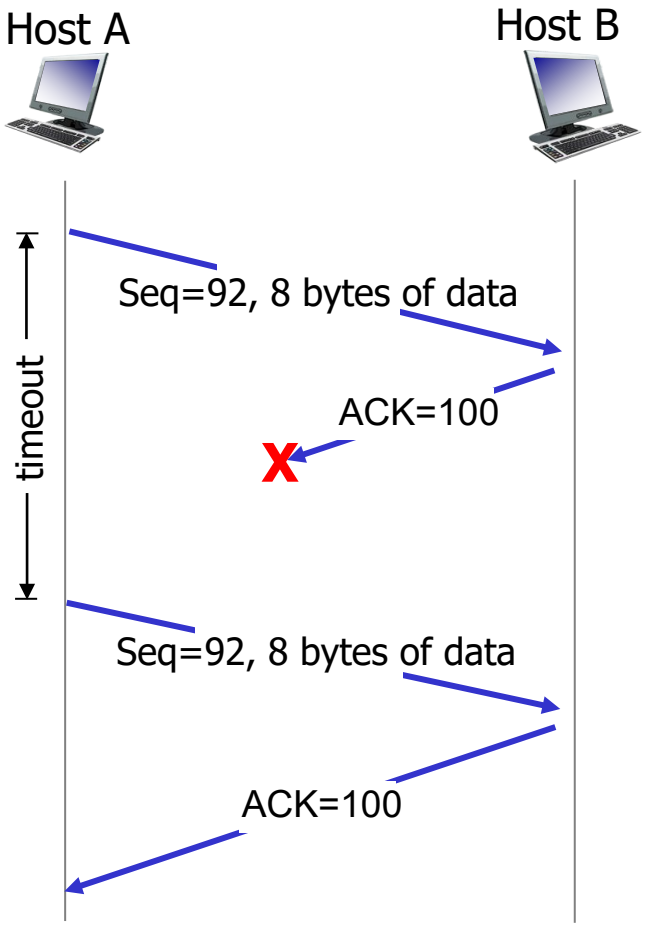
timeout

retransmit not-yet-acked segment
        with smallest seq. #
start timer

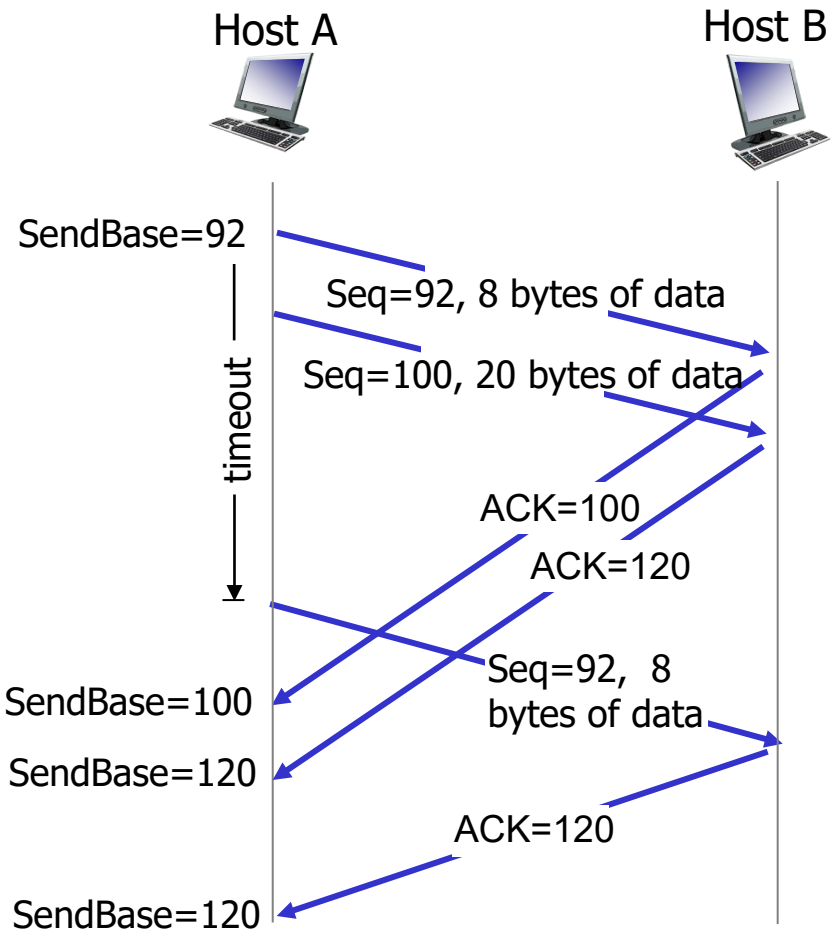
ACK received, with ACK field value y

```
if (y > SendBase) {
   SendBase = y
   /* SendBase–1: last cumulatively ACKed byte */
   if (there are currently not-yet-acked segments)
       start timer
     else stop timer
    }
```

TCP: retransmission scenarios

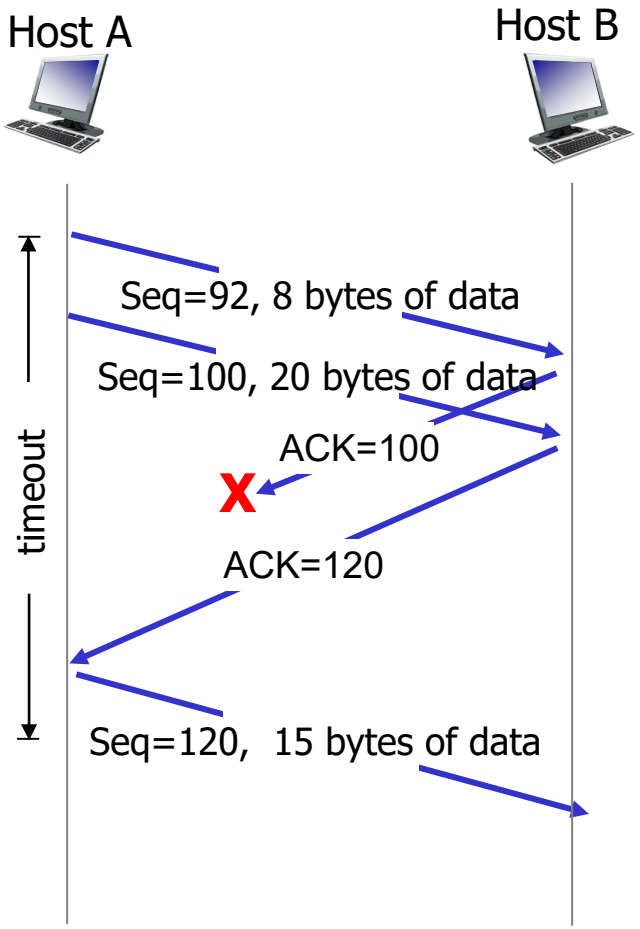


lost ACK scenario



premature timeout

TCP: retransmission scenarios



cumulative ACK

TCP ACK generation [RFC 1122, RFC 2581]

event at receiver	*TCP receiver action*
arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
arrival of in-order segment with expected seq #. One other segment has ACK pending	immediately send single cumulative ACK, ACKing both in-order segments
arrival of out-of-order segment higher-than-expect seq. # . Gap detected	immediately send *duplicate ACK*, indicating seq. # of next expected byte
arrival of segment that partially or completely fills gap	immediate send ACK, provided that segment starts at lower end of gap

TCP fast retransmit

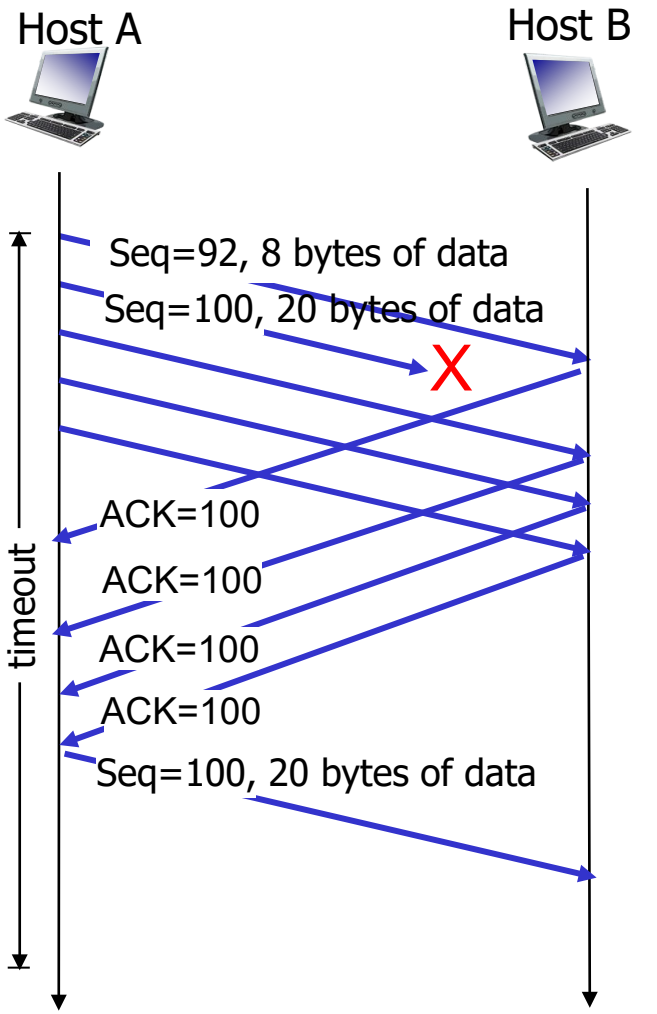
- time-out period often relatively long:
  - long delay before resending lost packet
- detect lost segments via duplicate ACKs.
  - sender often sends many segments back-to-back
  - if segment is lost, there will likely be many duplicate ACKs.

TCP fast retransmit

if sender receives 3 ACKs for same data ("triple duplicate ACKs"), resend unacked segment with smallest seq #

- likely that unacked segment lost, so don't wait for timeout

TCP fast retransmit



fast retransmit after sender receipt of triple duplicate ACK

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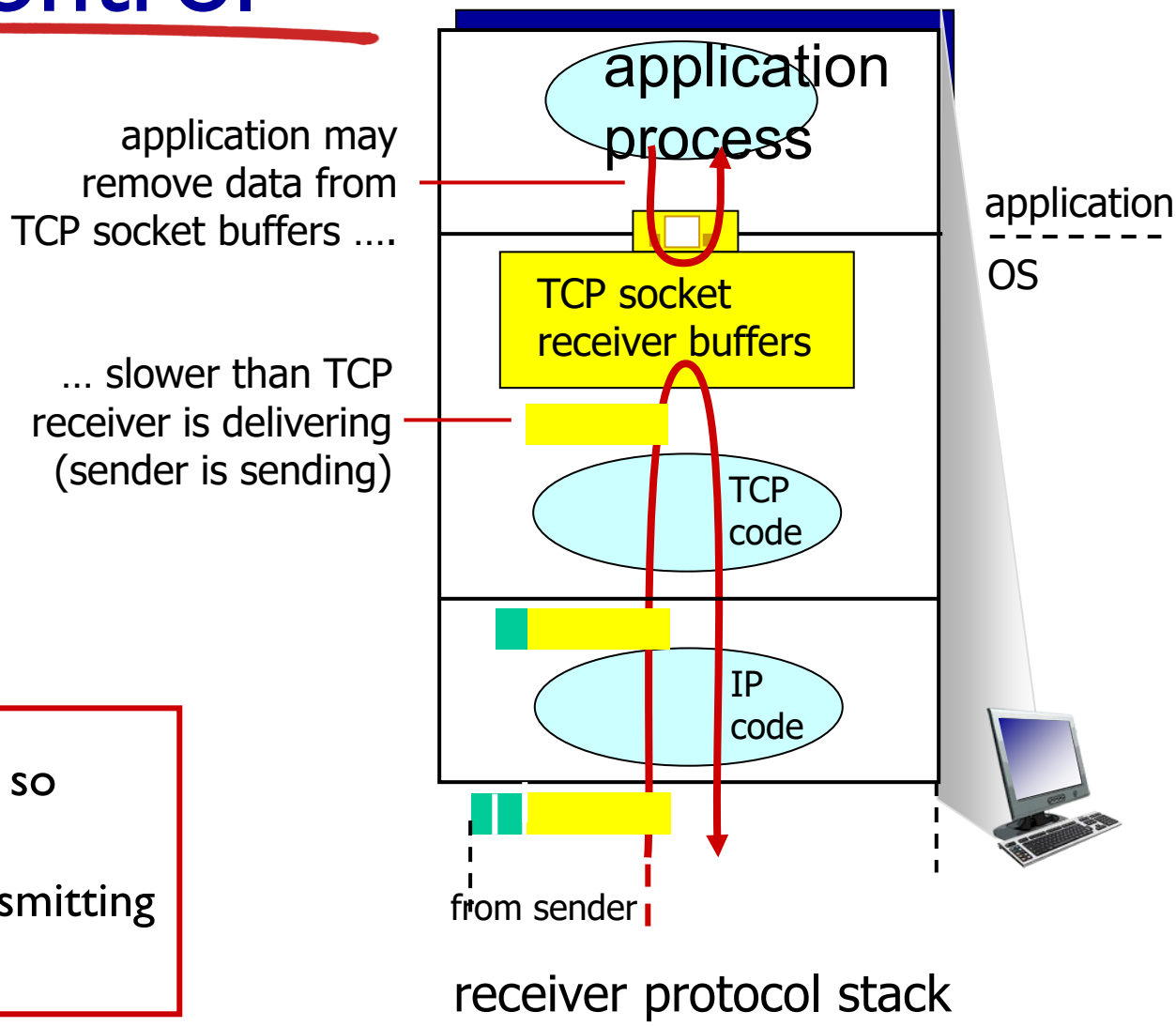
TCP flow control

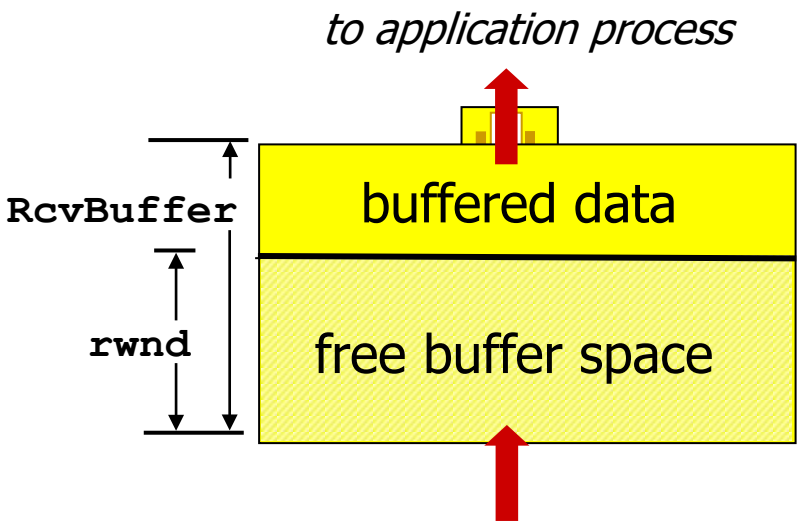


flow control

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

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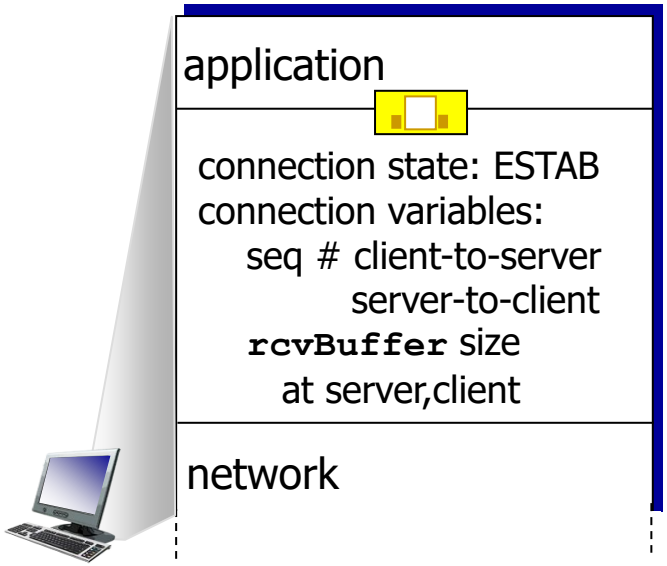
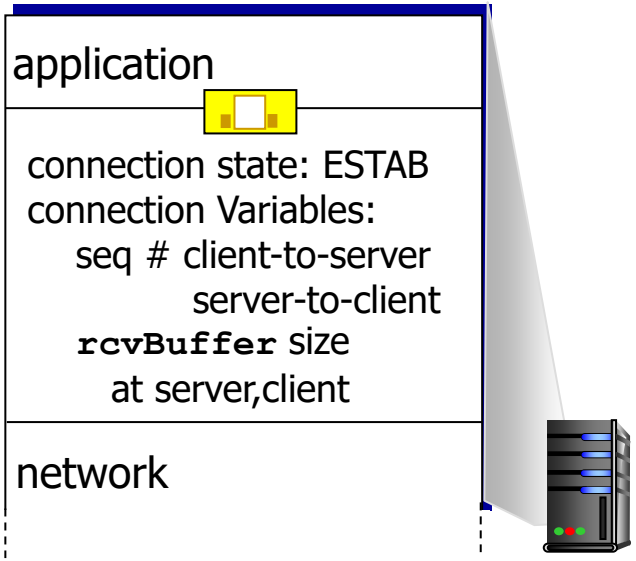
Connection Management

before exchanging data, sender/receiver "handshake":

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters



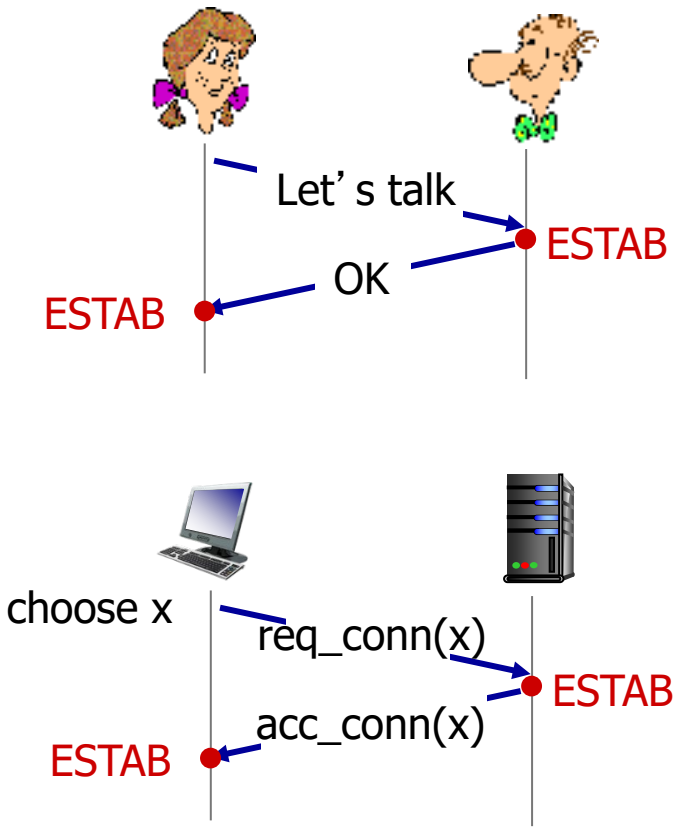



```
Socket clientSocket =
  newSocket("hostname","port
   number");
```

```
Socket connectionSocket =
   welcomeSocket.accept();
```

Agreeing to establish a connection

2-way handshake:

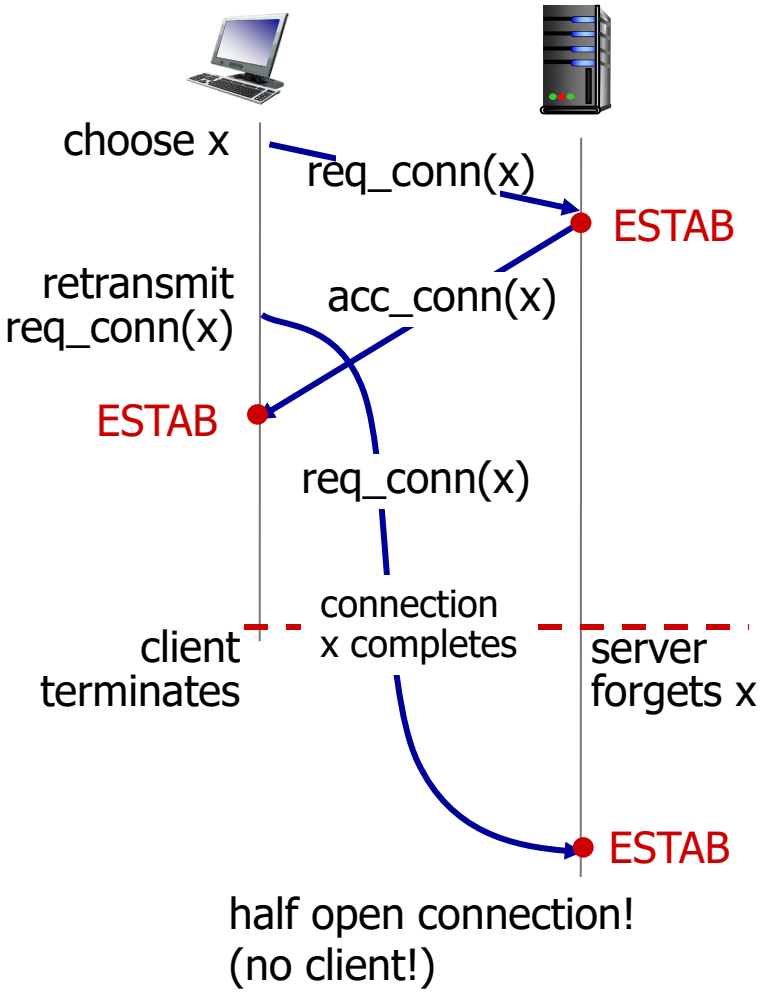


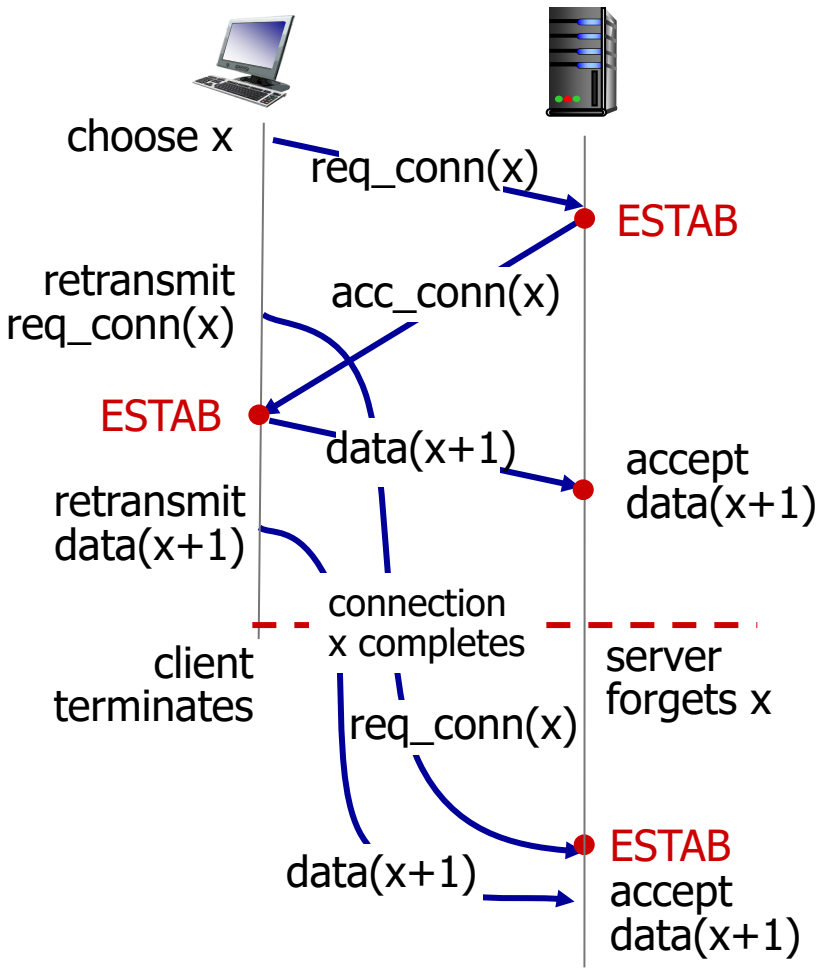
Q: will 2-way handshake always work in network?

- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side

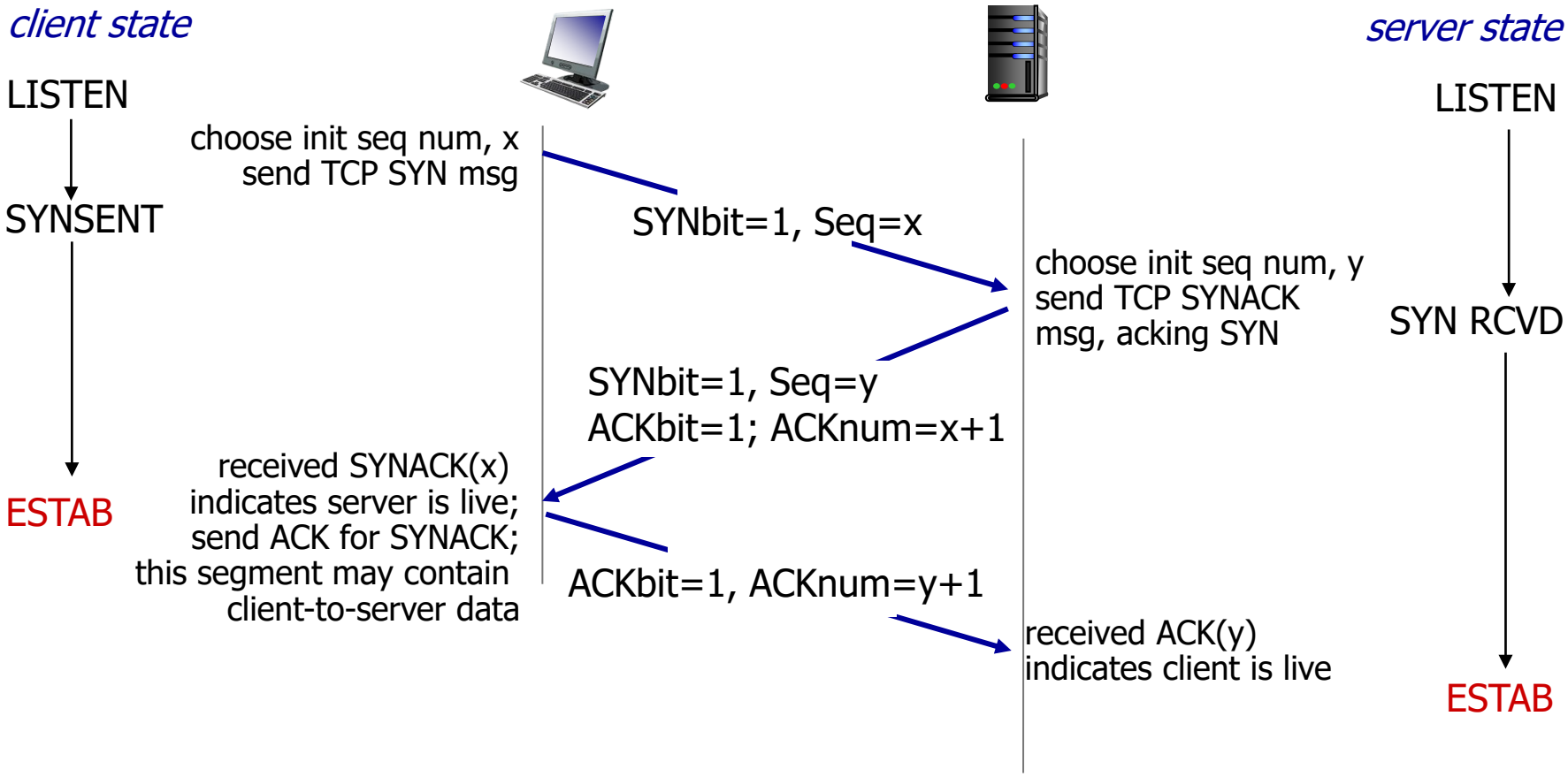
Agreeing to establish a connection

2-way handshake failure scenarios:





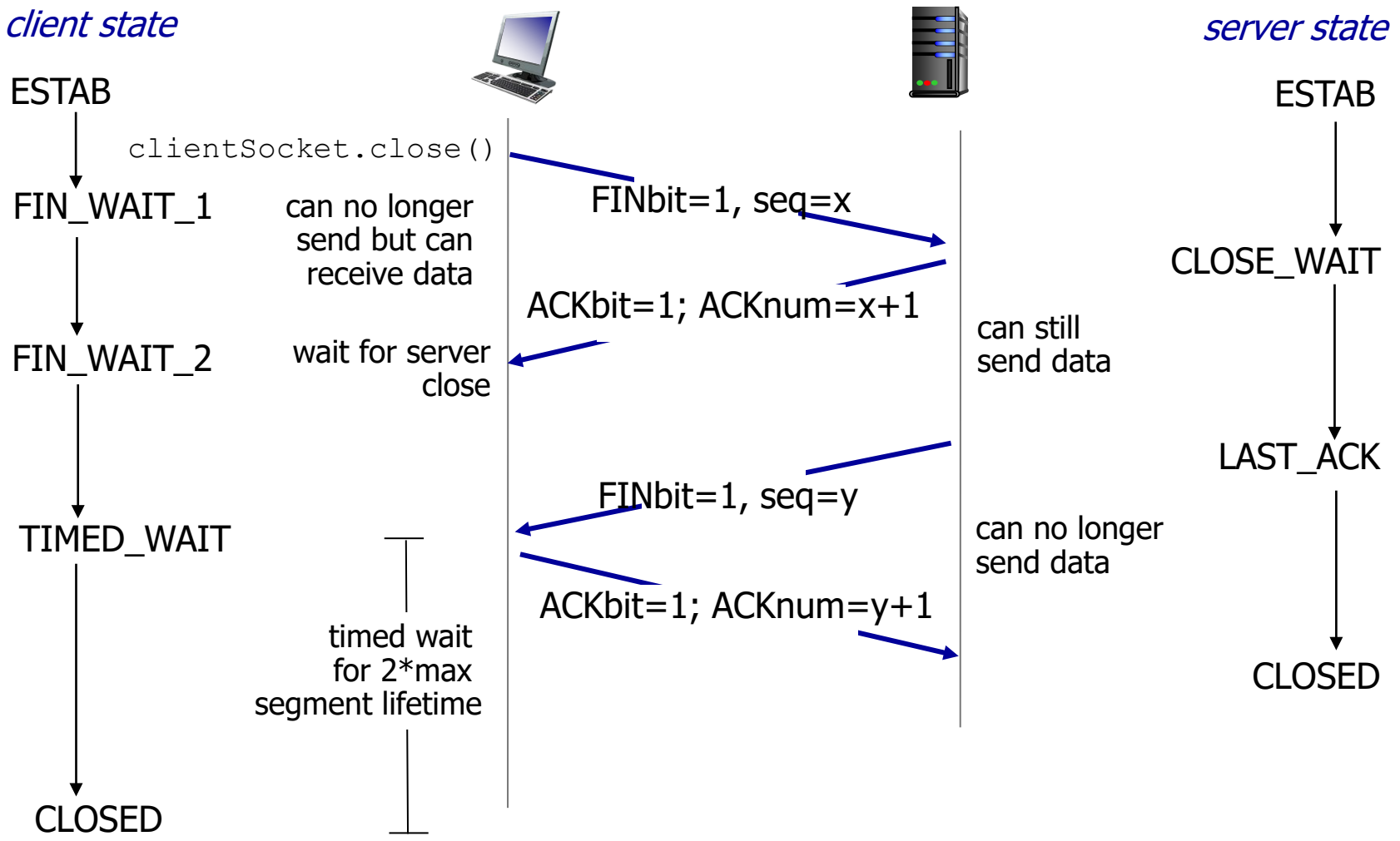
TCP 3-way handshake



TCP: closing a connection

- client, server each close their side of connection
  - send TCP segment with FIN bit = 1
- respond to received FIN with ACK
  - on receiving FIN, ACK can be combined with own FIN
- simultaneous FIN exchanges can be handled

TCP: closing a connection



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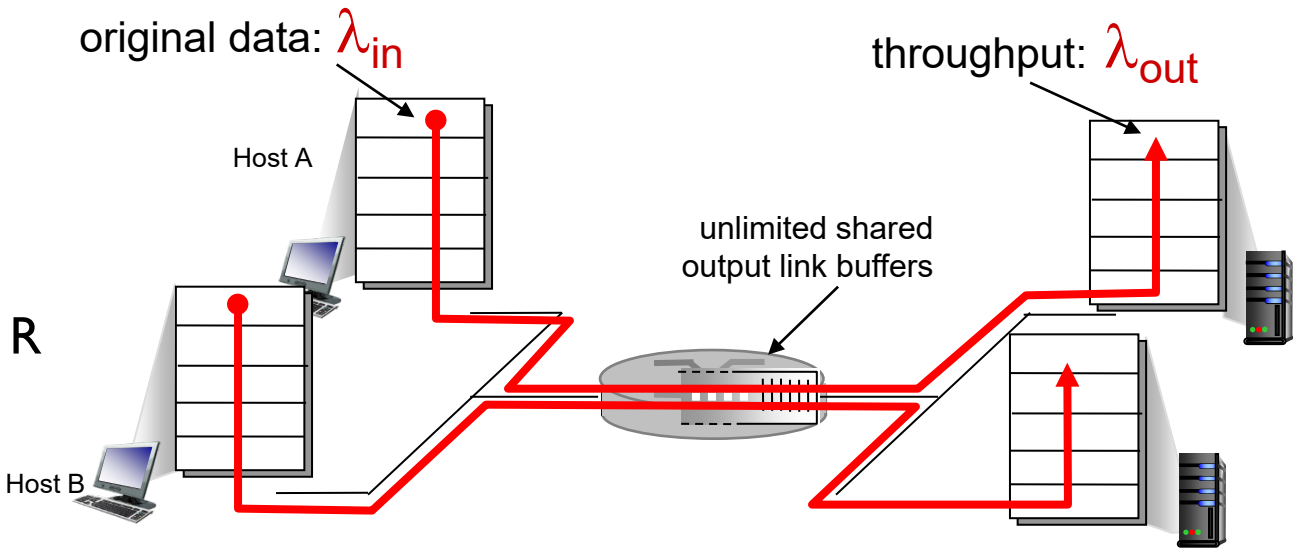
Principles of congestion control

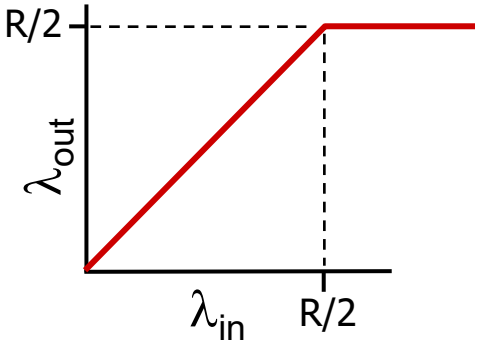
congestion:

- 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!

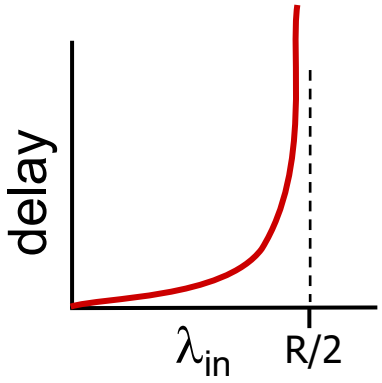
Causes/costs of congestion: scenario 1

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





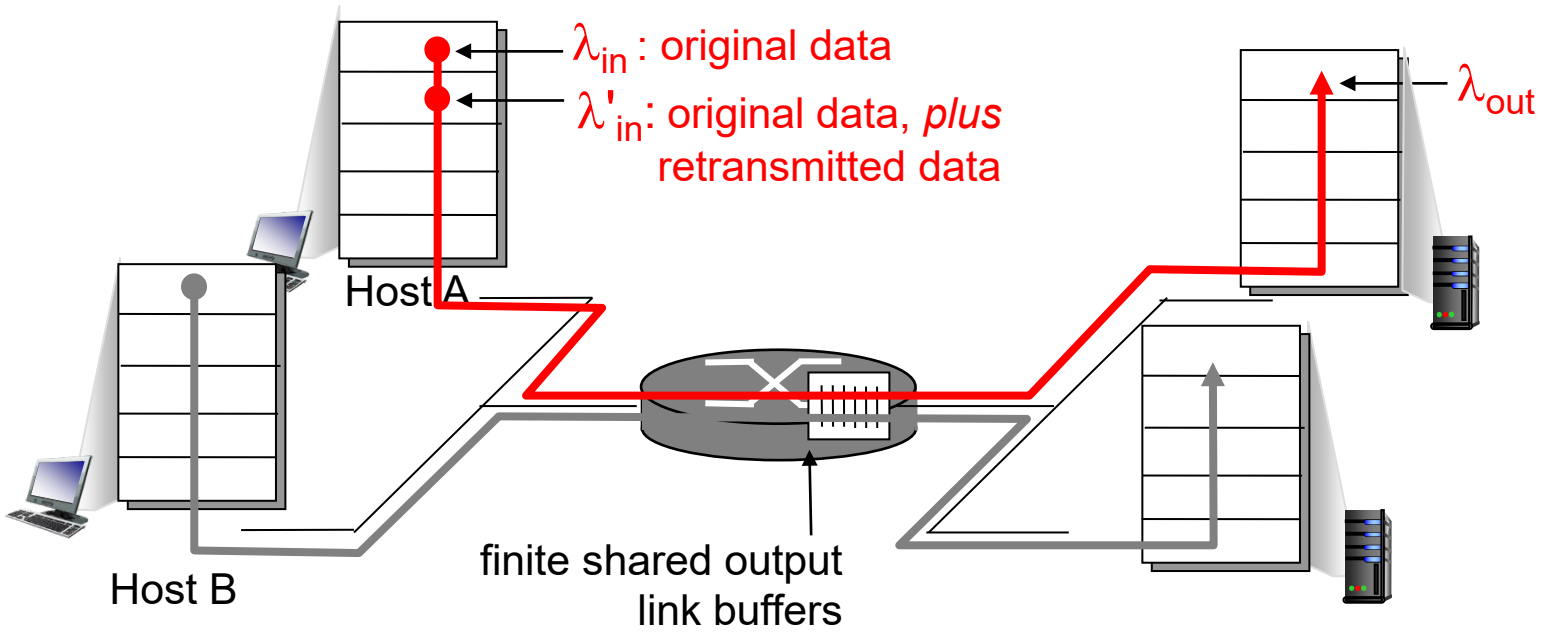
- maximum per-connection throughput: R/2



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

Causes/costs of congestion: scenario 2

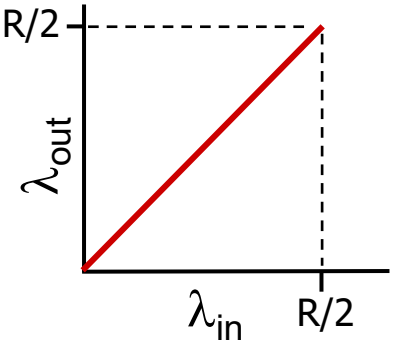
- one router, *finite* buffers
- sender retransmission of timed-out packet
  - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
  - transport-layer input includes *retransmissions* : $\lambda'_{in} \geq \lambda_{in}$

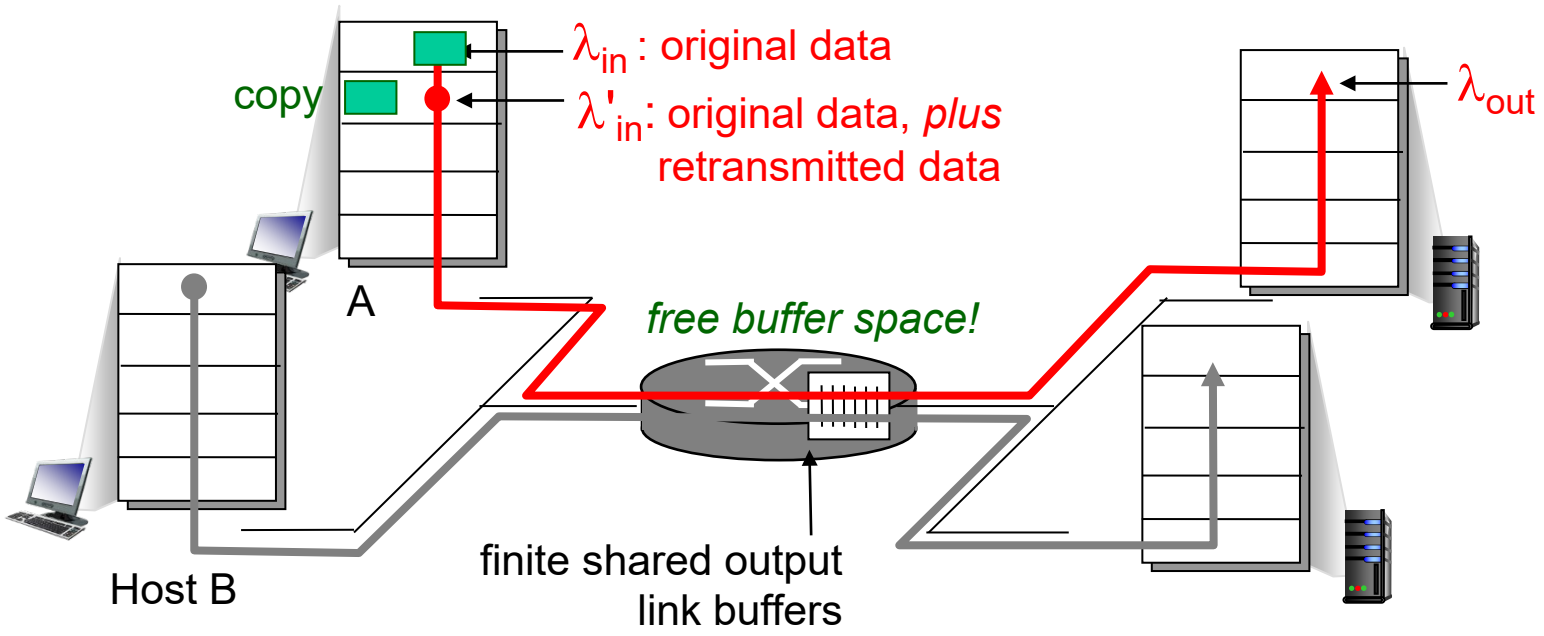


Causes/costs of congestion: scenario 2

idealization: perfect knowledge

- sender sends only when router buffers available



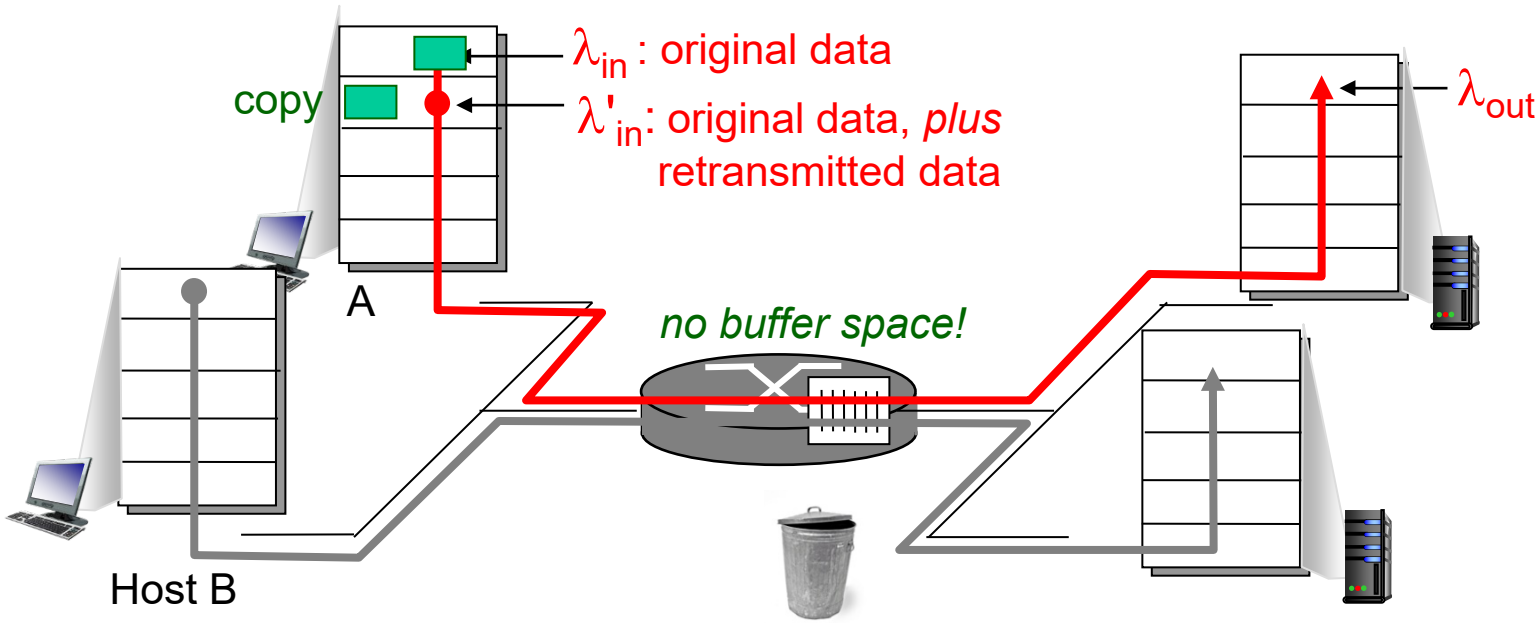


Causes/costs of congestion: scenario 2

Idealization: known loss

packets can be lost, dropped at router due to full buffers

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



Causes/costs of congestion: scenario 2

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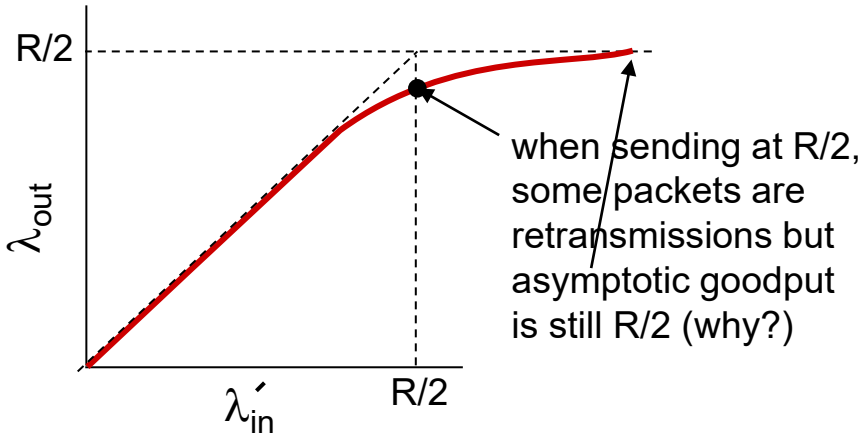
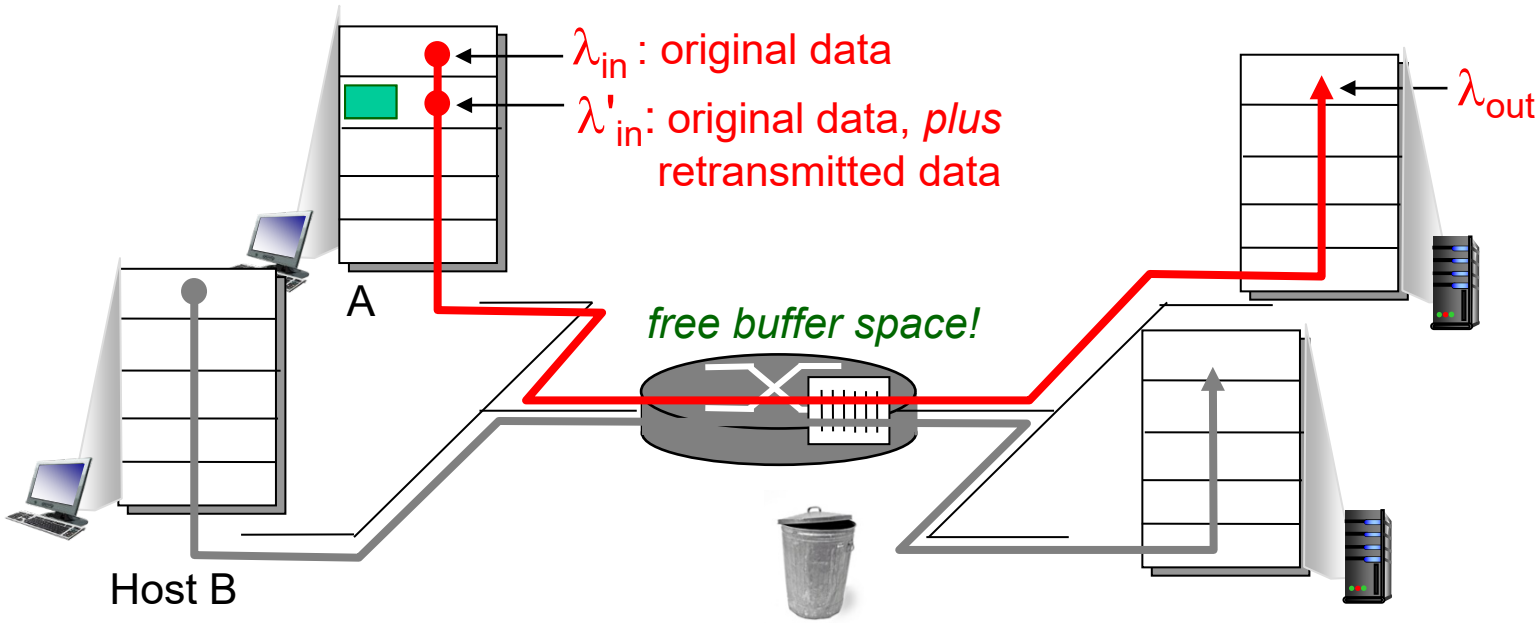
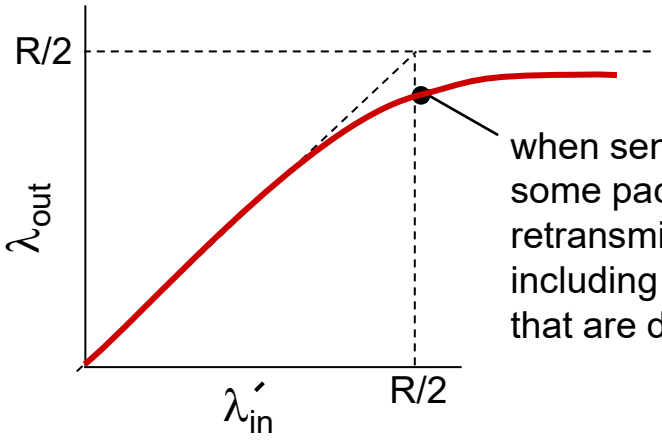


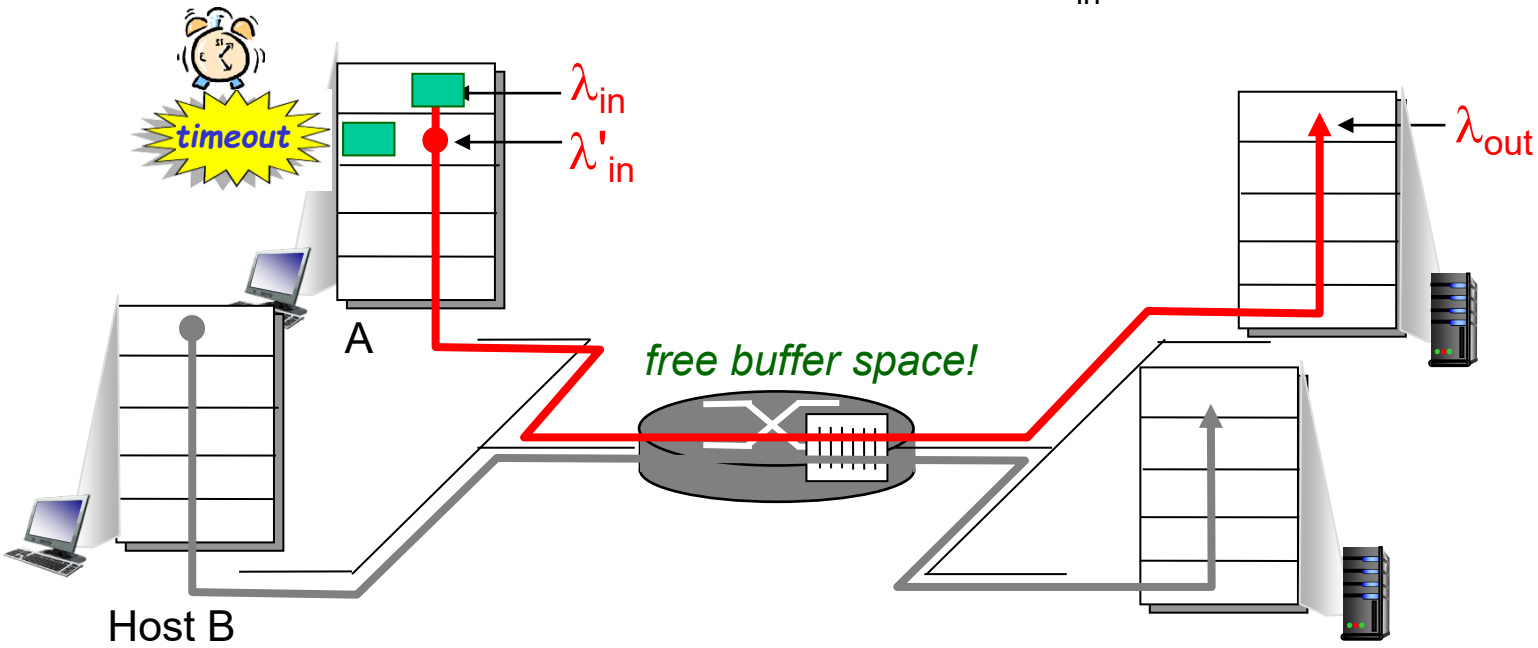


Causes/costs of congestion: scenario 2

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



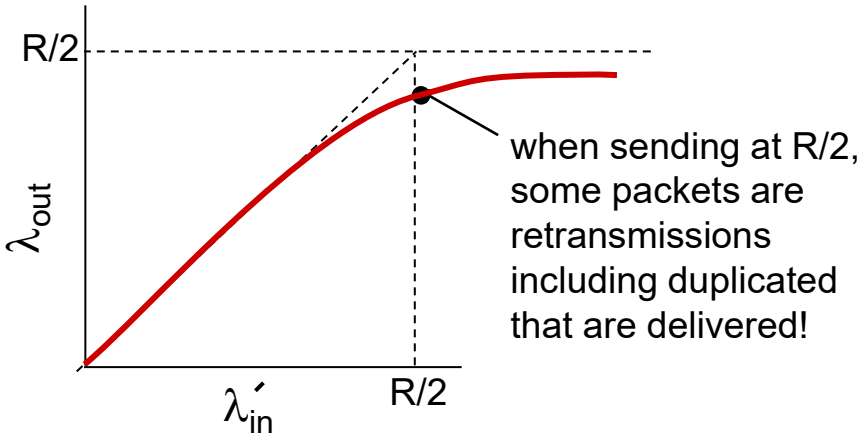


- Throughput:
  - Data rate at the receiver
- Goodput:
  - Rate at the receiver for data without duplicate!

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



"costs" of congestion:

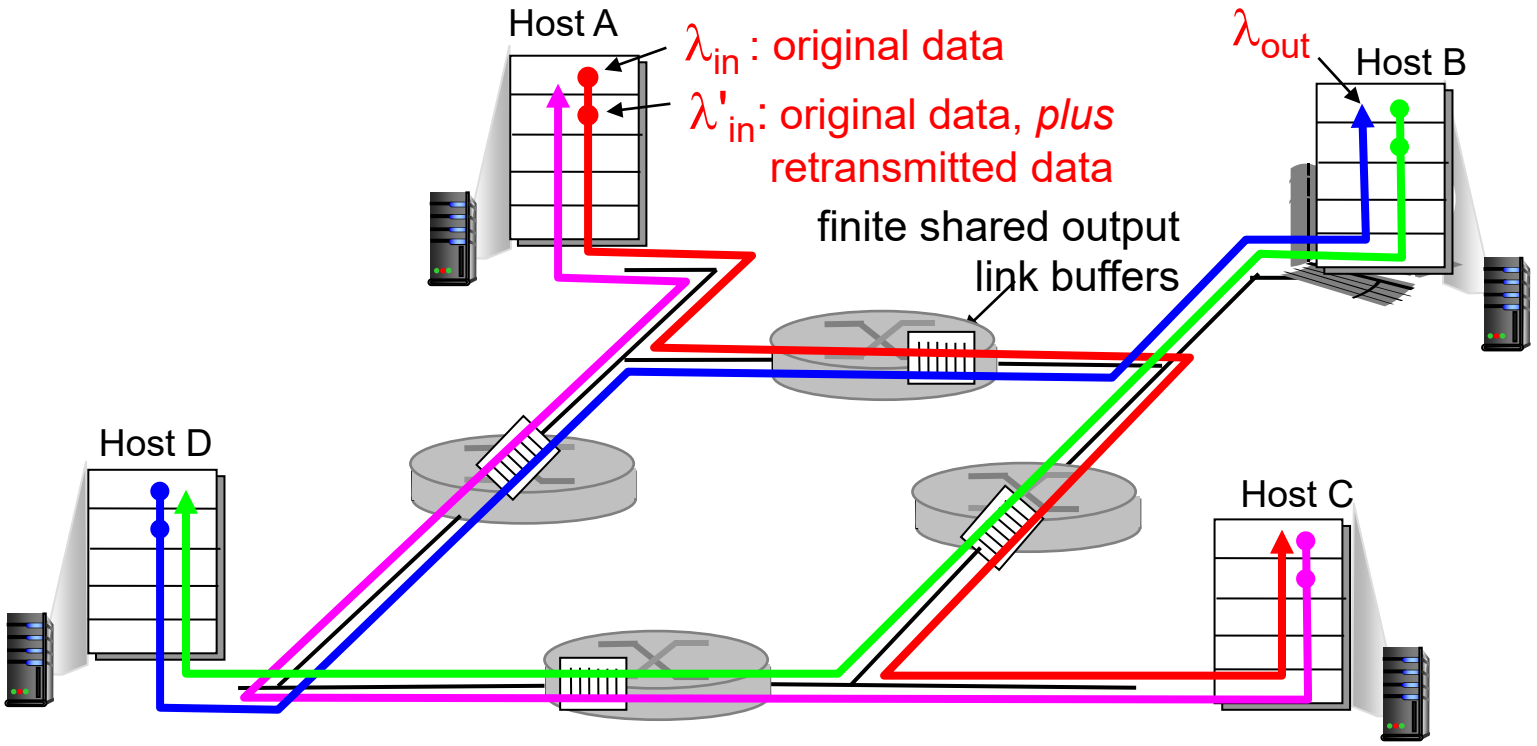
- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt
  - decreasing goodput

Causes/costs of congestion: scenario 3

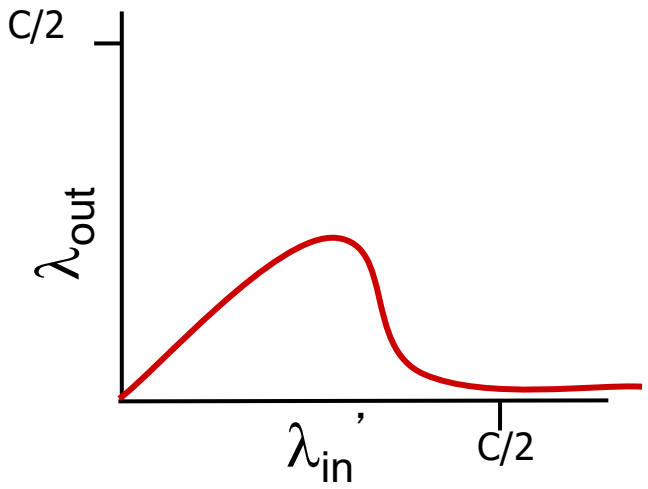
- four senders
- multihop paths
- timeout/retransmit

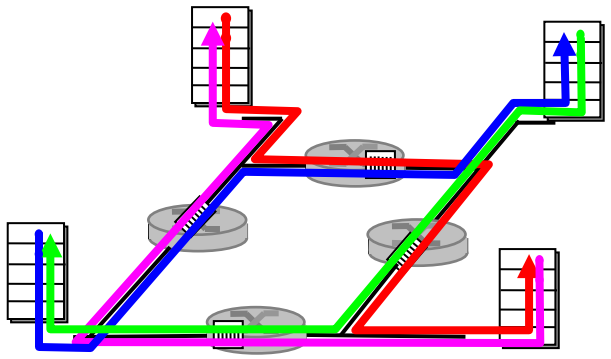
Q: what happens as λ_{in} and λ_{in}' increase ?

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!

Approaches towards congestion control

two broad approaches towards congestion control:

end-end congestion control:

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

network-assisted congestion control:

- routers provide feedback to end systems
  - single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
  - explicit rate for sender to send at

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

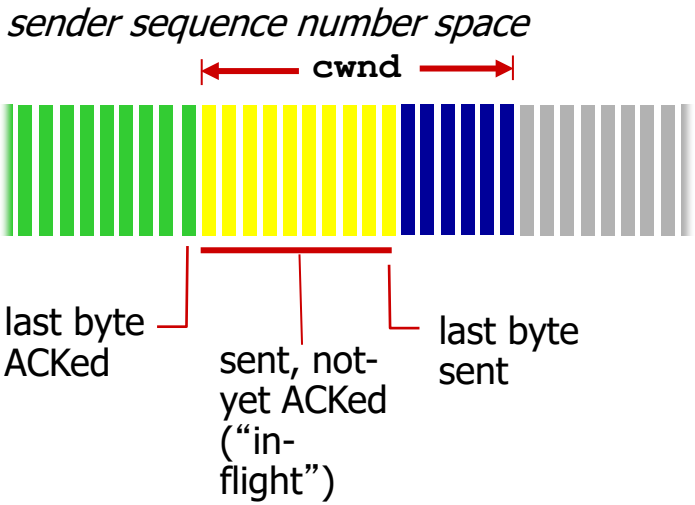
3.5 connection-oriented transport: TCP

- segment structure
- reliable data transfer
- flow control
- connection management

3.6 principles of congestion control

3.7 TCP congestion control

TCP Congestion Control: details



- sender limits transmission:

$$\texttt{LastByteSent} - \texttt{LastByteAcked} \leq \texttt{cwnd}$$

- **cwnd** is dynamic, function of perceived network congestion

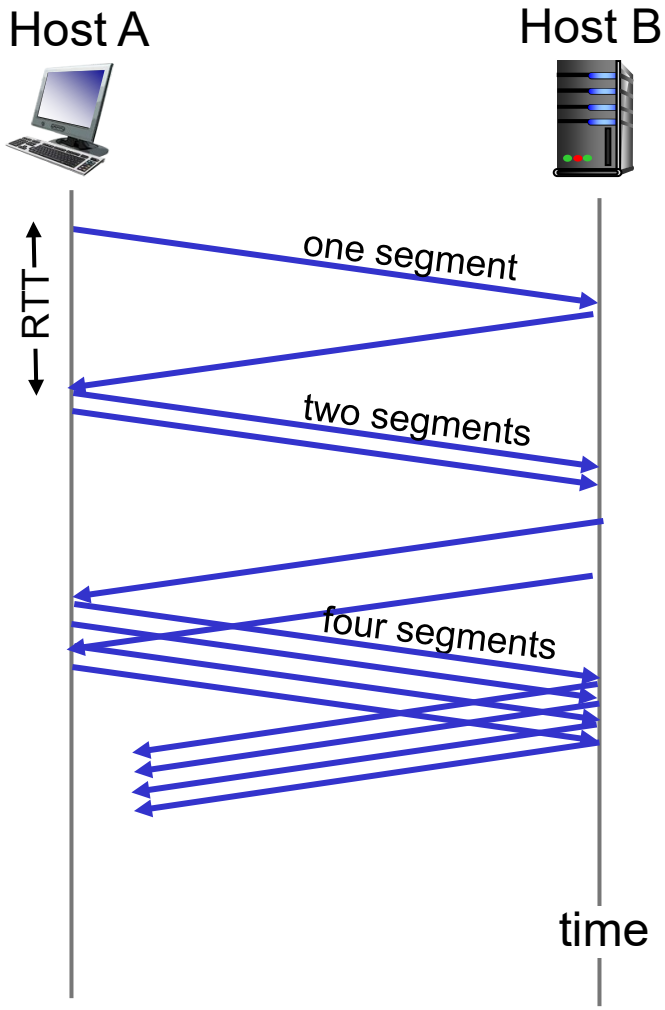
TCP sending rate:

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

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

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: detecting, reacting to loss

- loss indicated by timeout:
  - set a threshold **`ssthresh`** to half of the **`cwnd`**;
  - **`cwnd`** set to 1 MSS (by both TCP Tahoe and Reno);
  - window then grows exponentially (as in slow start) to threshold, then grows linearly
- TCP Tahoe always sets **`cwnd`** to 1 (timeout or 3 duplicate acks)
- TCP RENO: loss indicated by 3 duplicate ACKs
  - dup ACKs indicate network capable of delivering some segments
  - **`cwnd`** is cut in half window then grows linearly

After cwnd reaching the threshold

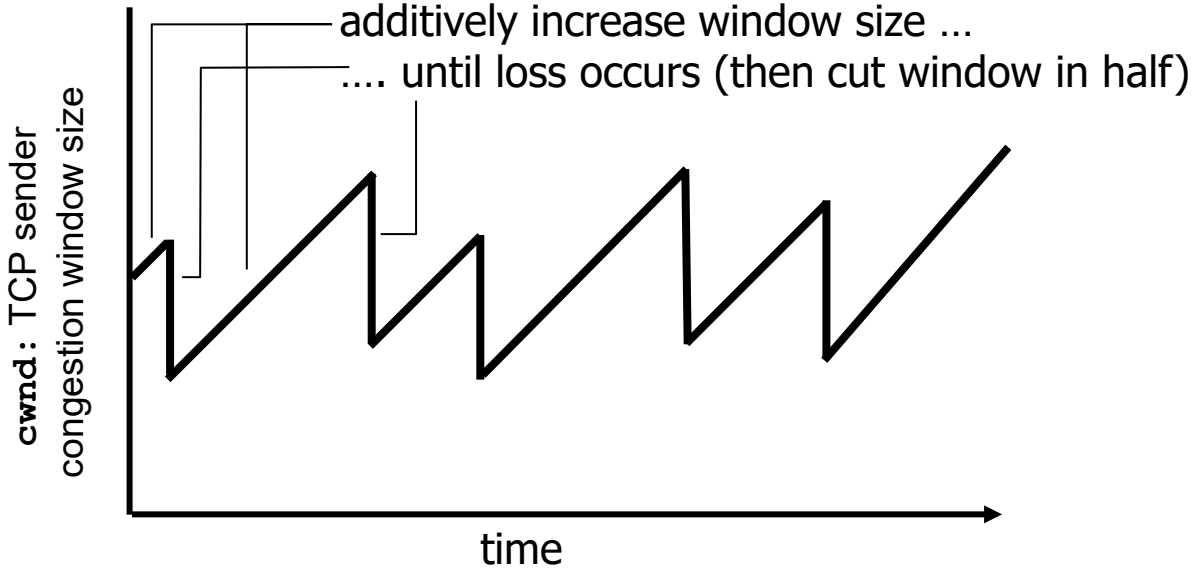
- Congestion avoidance algorithm:

- Additive increase multiplicative decrease (AIMD)

TCP congestion control: AIMD

- *approach:* sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
  - *additive increase:* increase **`cwnd`** by 1 MSS every RTT until loss detected
  - *multiplicative decrease:* cut **`cwnd`** in half after loss

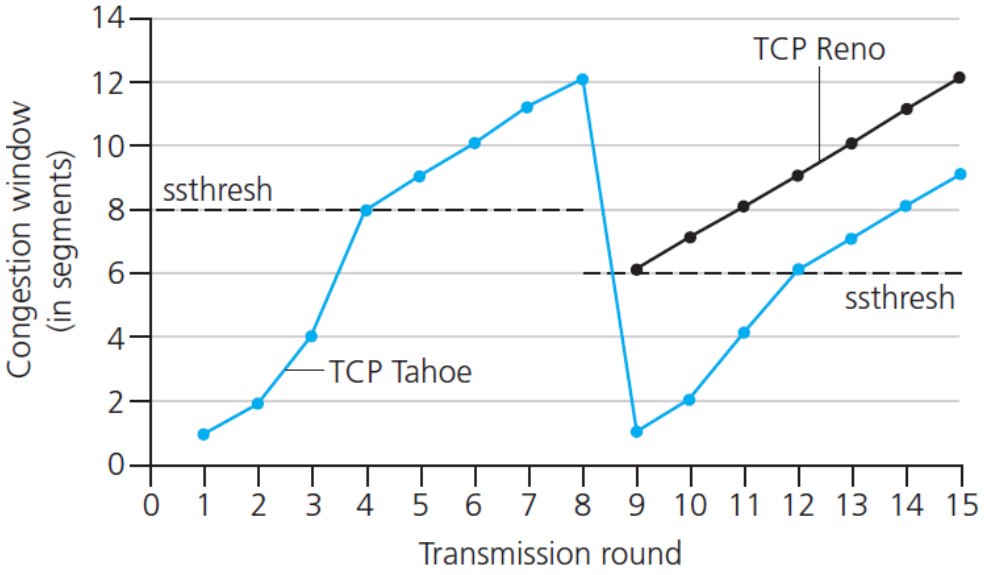
AIMD saw tooth behavior: probing for bandwidth



TCP: switching from slow start to CA

Q: when should the exponential increase switch to linear?

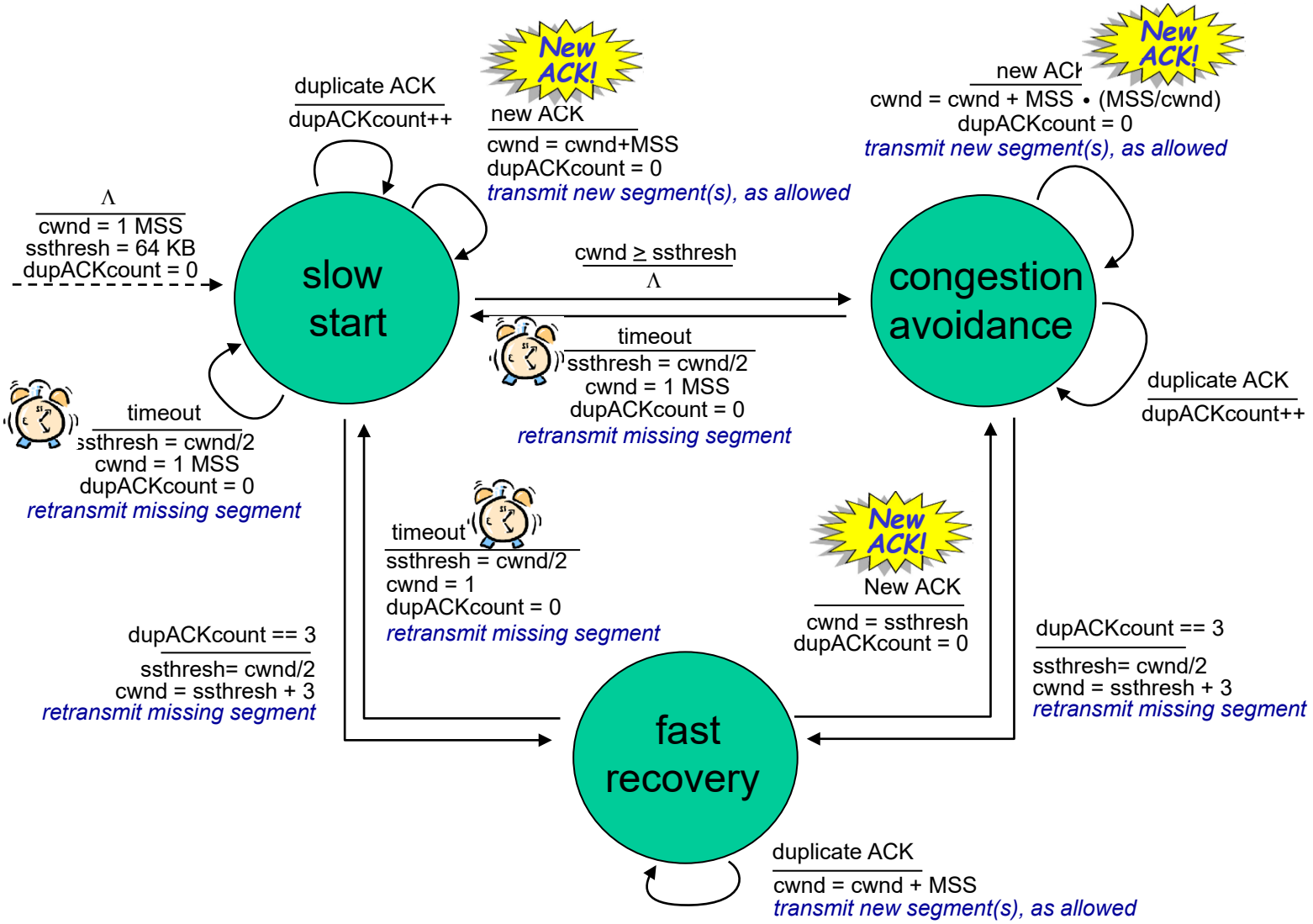
A: when **`cwnd`** gets to 1/2 of its value before timeout.



Implementation:

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

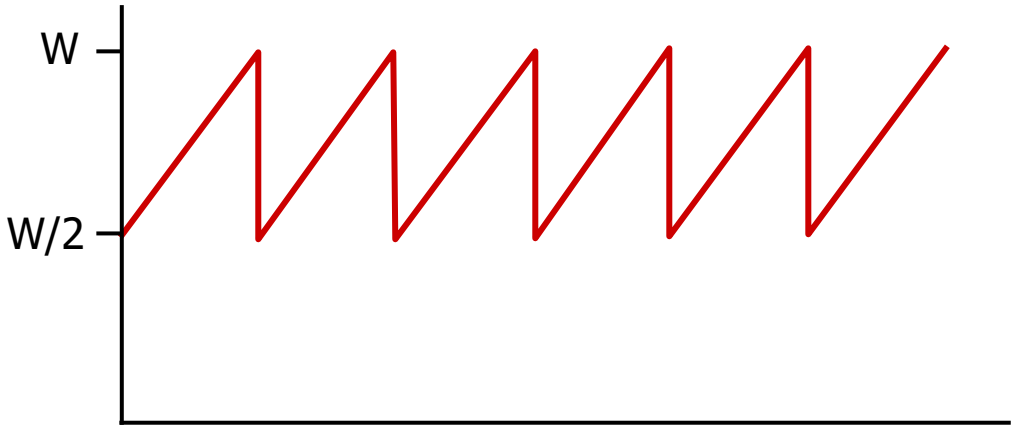
Summary: TCP Congestion Control



TCP throughput

- avg. TCP thruput as function of window size, RTT?
  - ignore slow start, assume always data to send
- W: window size (measured in bytes) where loss occurs
  - avg. window size (# in-flight bytes) is ¾ W
  - avg. thruput is 3/4W per RTT

$$\text{avg TCP thruput} = \frac{3}{4}\frac{W}{RTT} \text{ bytes/sec}$$



TCP Futures: TCP over "long, fat pipes"

- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- requires W = 83,333 in-flight segments
- throughput in terms of segment loss probability, L [Mathis 1997]:

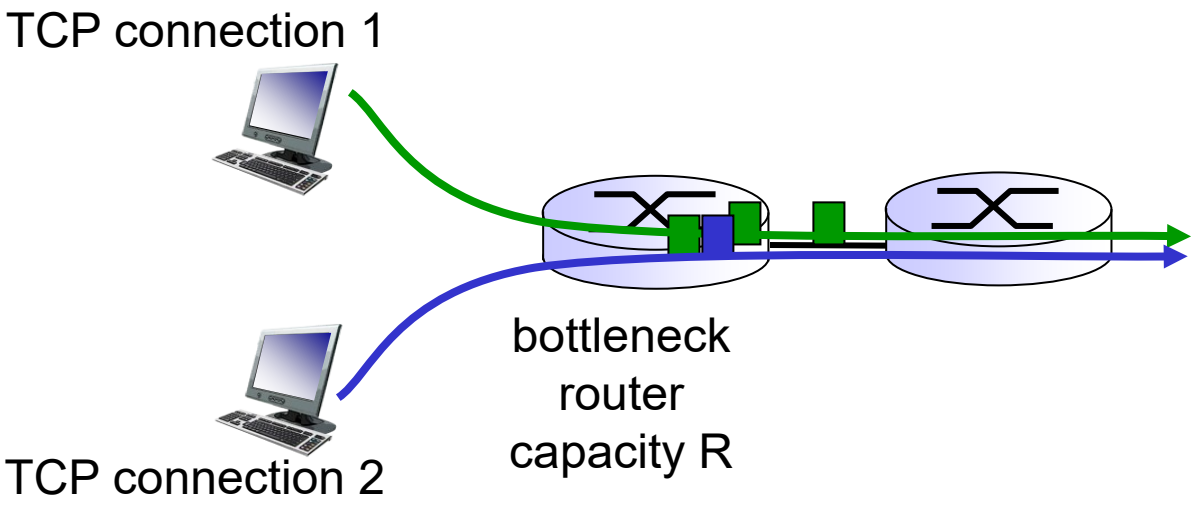
$$\text{TCP throughput} = \frac{1.22 \cdot MSS}{RTT\sqrt{L}}$$

➜ to achieve 10 Gbps throughput, need a loss rate of L = $2 \cdot 10^{-10}$ *– a very small loss rate!*

- new versions of TCP for high-speed

TCP Fairness

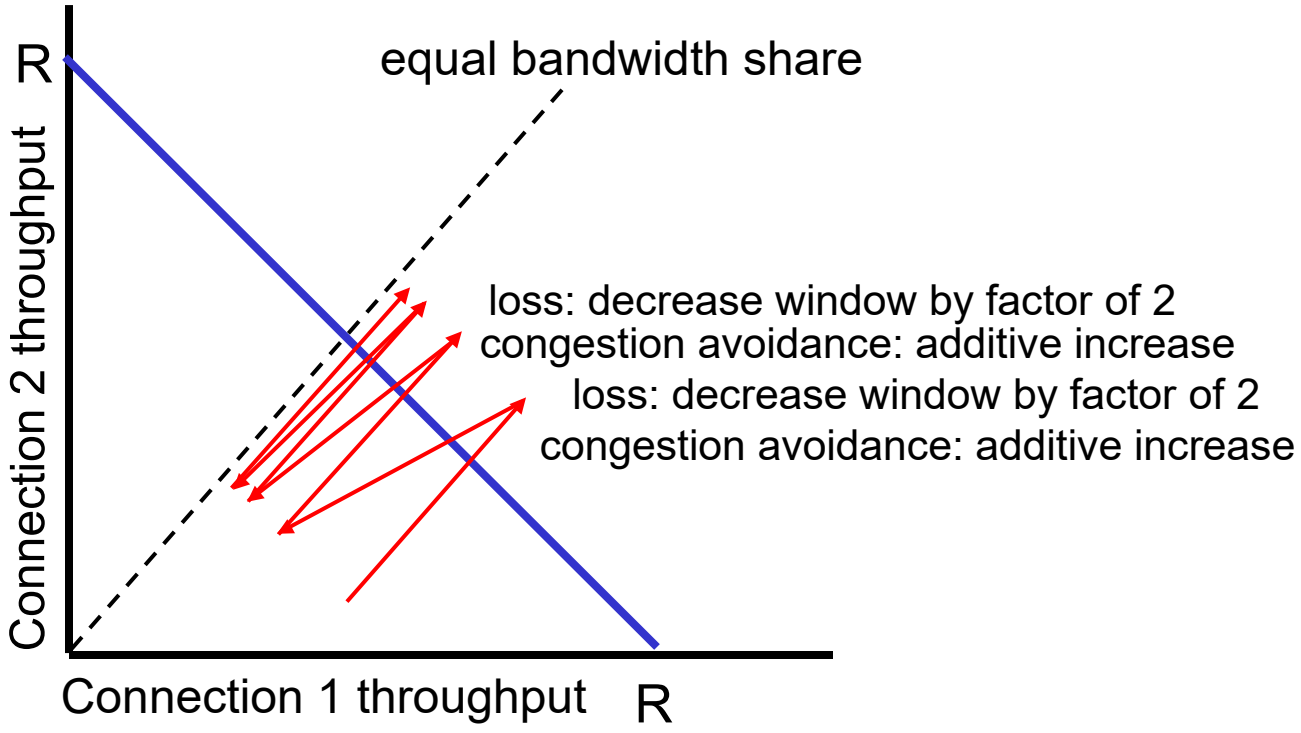
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Why is TCP fair?

two competing sessions:

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



Van Jacobson

- One of the key designers of TCP congestion control
- https://www.youtube.com/watch?v=QP4A6L7CEqA
- 1:40-9:20

Fairness (more)

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

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

Chapter 3: summary

- principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- instantiation, implementation in the Internet
  - UDP
  - TCP

next:

- leaving the network "edge" (application, transport layers)
- into the network "core"

- Midterm covers every slide until here.