CS2105

An *Awesome* Introduction to Computer Networks

Lecture 1: Introduction

Department of Computer Science
School of Computing

Course Instructors

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What is CS2105 About?

- Discussion of fundamental **concepts** and **principles** behind computer networking
  - Using the Internet as a case study
- Introduction to networking tools and networked application programming
  - Programming with Python, Java, or C++

What you will NOT learn in CS2105

- How to configure hardware, e.g. router
  - This is covered in CS3103 Computer Networks Practice - perform hands-on experiments in subnetting, DHCP, DNS, RIP, OSPF, TCP handshaking and congestion mechanism
- Mobile and wireless networks
  - This is covered in CS4222 Wireless Networking

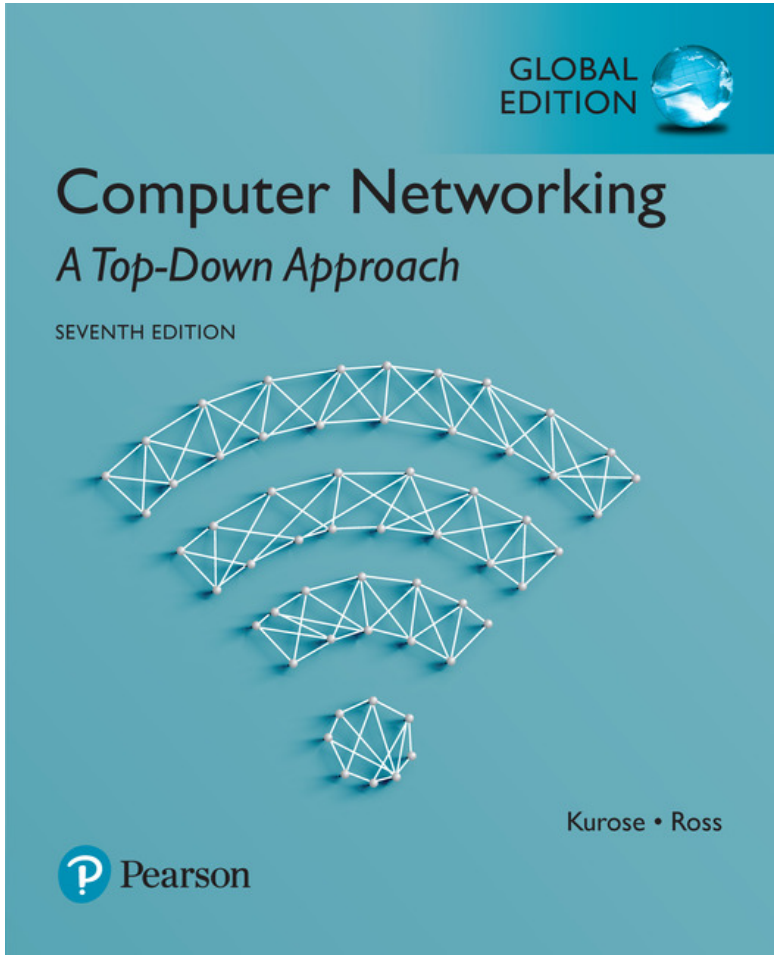
Textbook

Computer Networking: A Top-Down Approach: Global Edition, 7/E

Authors : Kurose
Ross

Publisher : Pearson

ISBN : 9781292153599



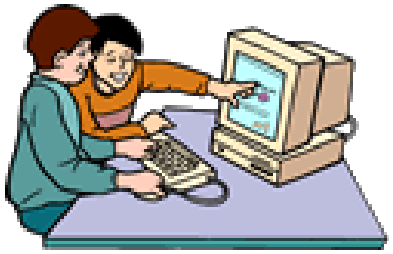
Acknowledgement:
Most of the lecture slides are adopted from slides of this textbook.

Available at **NUS Campus bookstore**

Contact Hours

- Lectures
  - Every Monday 2 - 4pm @ ICube Auditorium
  - 2 hours per session
  - week 8 lecture reserved for midterm test
- Tutorials
  - Start from week 3.
  - 1 hour per session
  - No tutorial in week 8 ☺
- Consultation
  - Email me or Wai Kay to make arrangement

Assessments



- CA (50%)
  - Tutorial attendance/participation - 5%
  - Individual programming assignments - 25%
  - Mid-term test (week 8 lecture time: Mon, 7 Oct 2019, 2:10 - 3:30pm) - 20%
- Final Exam (50%)
  - Mon, 2 Dec 2019, 5 - 7pm

Notes and Tips

- Why CS2105 can be **easy**
  - You use and interact with the Internet constantly
  - Many of the concepts are intuitive and based on very practical design considerations
  - There are very few equations!

- Why CS2105 can be **tough**
  - Many concepts are covered
  - Programming!

Lecture 1: Introduction

After this class, you are expected to:

- understand the basic terms, including host, packet, protocol, throughput, store-and-forward, and autonomous system.
- know about the logical (five protocol layers) and physical (a network of ASes) architecture of the Internet.
- understand the different components of end-to-end delay and their relations to bandwidth, packet size, distance, propagation speed, and queue size.

Lecture 1: Roadmap

1.1 What is the Internet?

1.2 Network Edge

1.3 Network Core

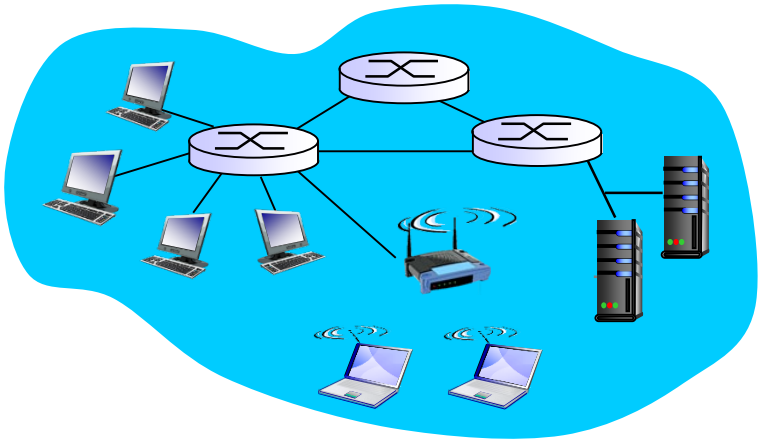
1.4 Delay, Loss and Throughput in Networks

1.5 Protocol Layers and Service Models

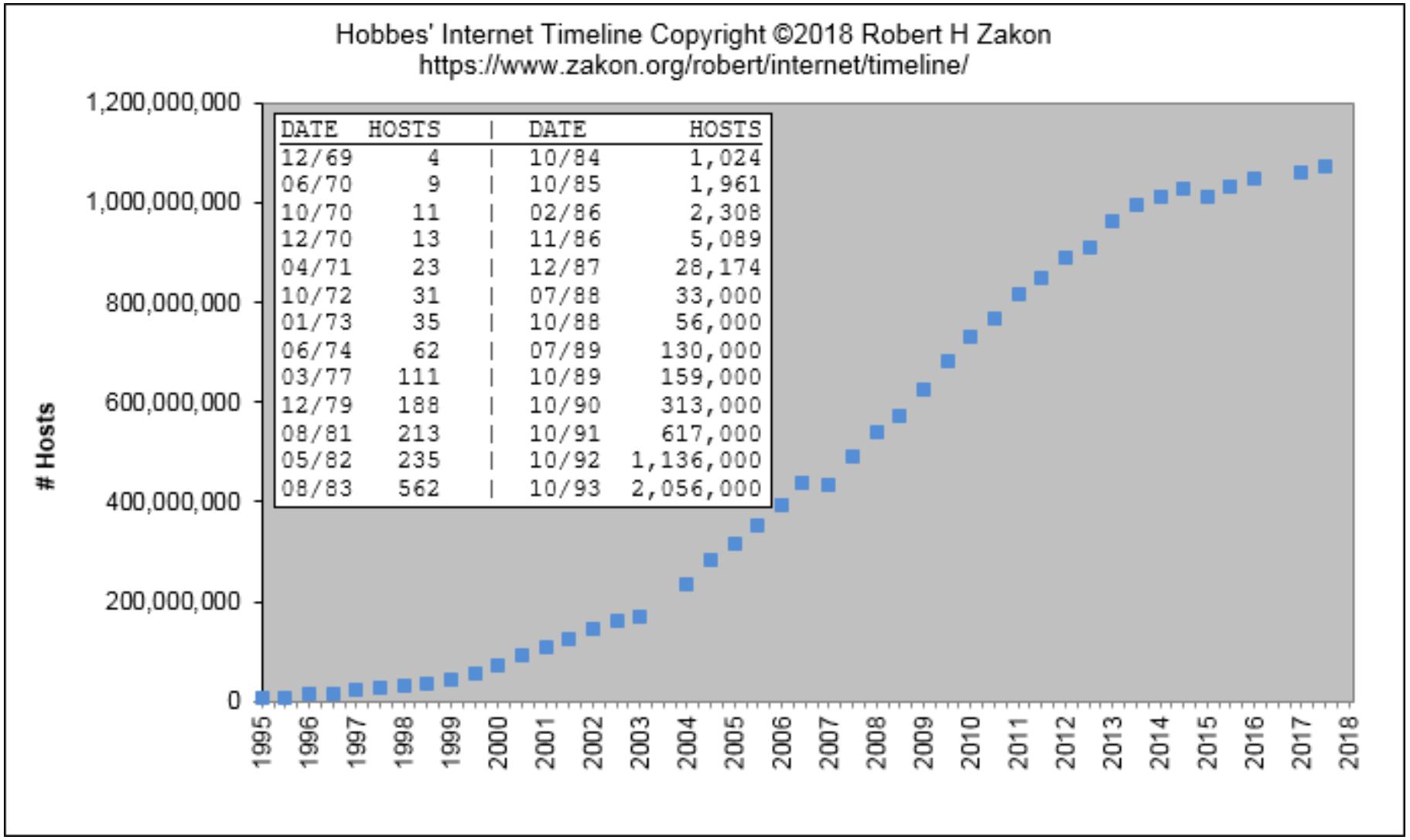
Kurose Textbook, Chapter 1
(Some slides are taken from the book)

Internet: "nuts and bolts" View

- The Internet is a network of connected computing devices (e.g. PC, server, laptop, smartphone)
  - such devices are known as *hosts* or *end systems*.
  - they run network applications (e.g. WhatsApp, browser).
  - they communicate over links.



Growth of Internet Hosts



DATE	HOSTS	DATE	HOSTS
12/69	4	10/84	1,024
06/70	9	10/85	1,961
10/70	11	02/86	2,308
12/70	13	11/86	5,089
04/71	23	12/87	28,174
10/72	31	07/88	33,000
01/73	35	10/88	56,000
06/74	62	07/89	130,000
03/77	111	10/89	159,000
12/79	188	10/90	313,000
08/81	213	10/91	617,000
05/82	235	10/92	1,136,000
08/83	562	10/93	2,056,000

"Fun" Internet-connected Devices

IP picture frame
http://www.ceiva.com/

Web-enabled toaster +
weather forecaster

Tweet-a-watt:
monitor energy use

Internet
refrigerator

Slingbox: watch,
control cable TV remotely

sensorized,
bed
mattress

Internet phones

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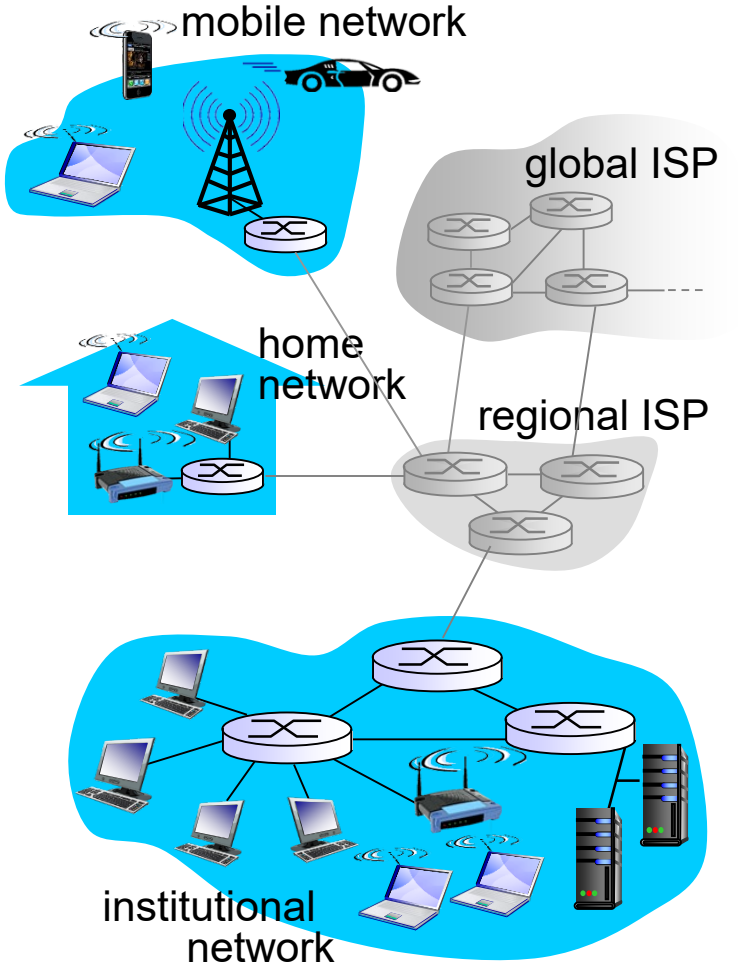
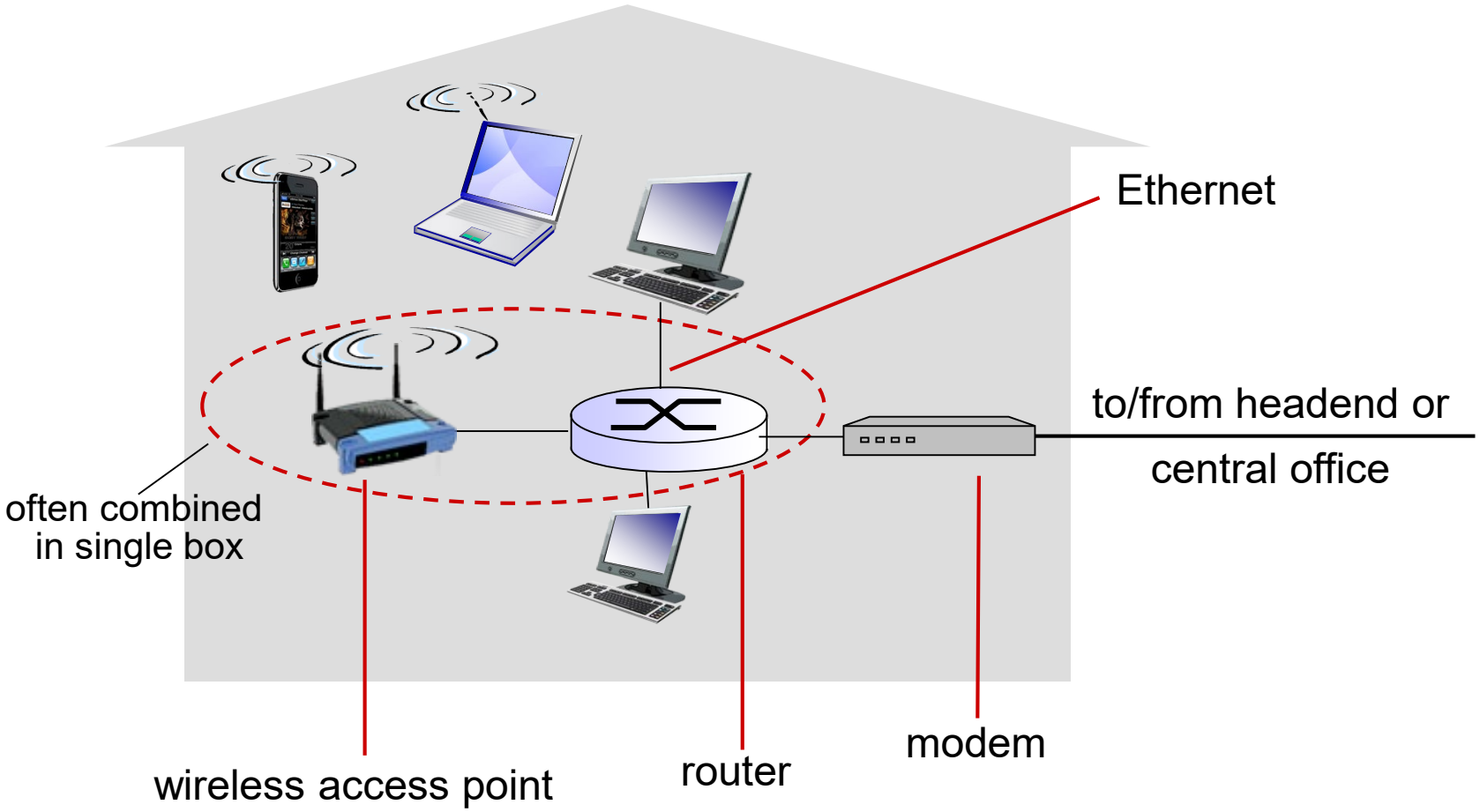
Access Networks

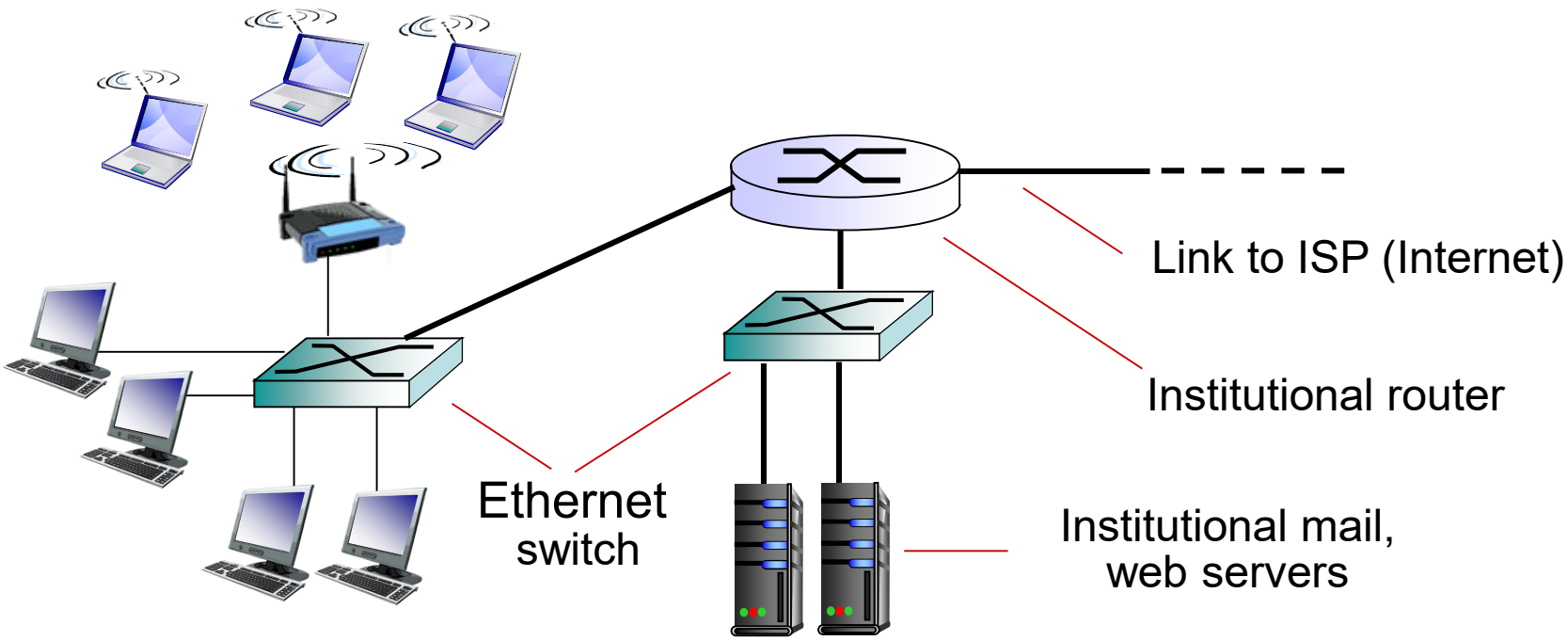
- Hosts access the Internet through *access network*.
  - Residential access networks
  - Institutional access networks (school, company)
  - Mobile access networks



Home Networks



Enterprise Access Networks (Ethernet)



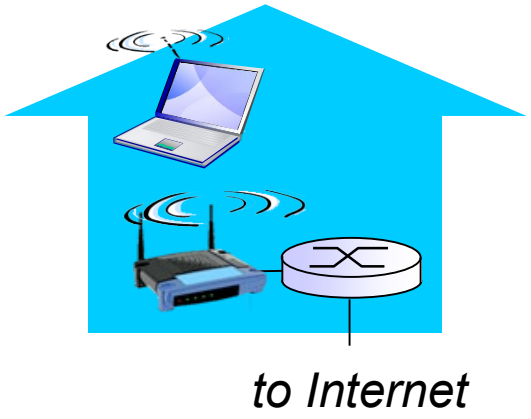
- Typically used in companies, universities, etc.
- 10 Mbps, 100Mbps, 1Gbps, 10Gbps transmission rates
- Today, hosts typically connect to Ethernet switch

Wireless Access Networks

- Wireless access network connects hosts to router
  - via base station aka "access point"

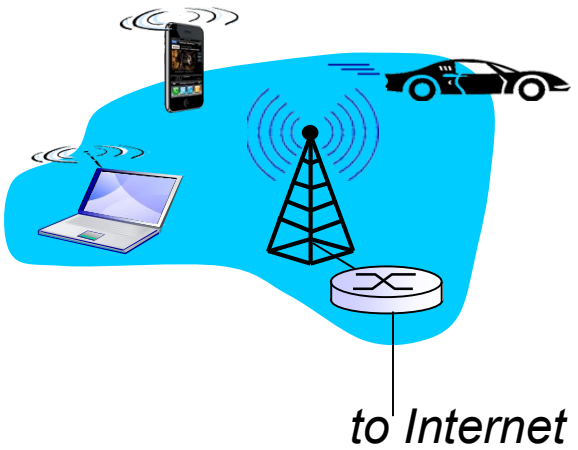
Wireless LANs:

- within building (100 ft)
- 802.11b/g/n/ac (Wi-Fi)



Wide-area wireless access

- 3G, 4G
- provided by telco (cellular) operator, 10's km



Physical Media

- Hosts connect to the access network over different physical media.
  - Guided media:
    - signals propagate in solid media

Twisted pair cable

Coaxial cable

Fiber optic cable

  - Unguided media:
    - signals propagate freely, e.g., radio

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- packet switching, circuit switching, network structure

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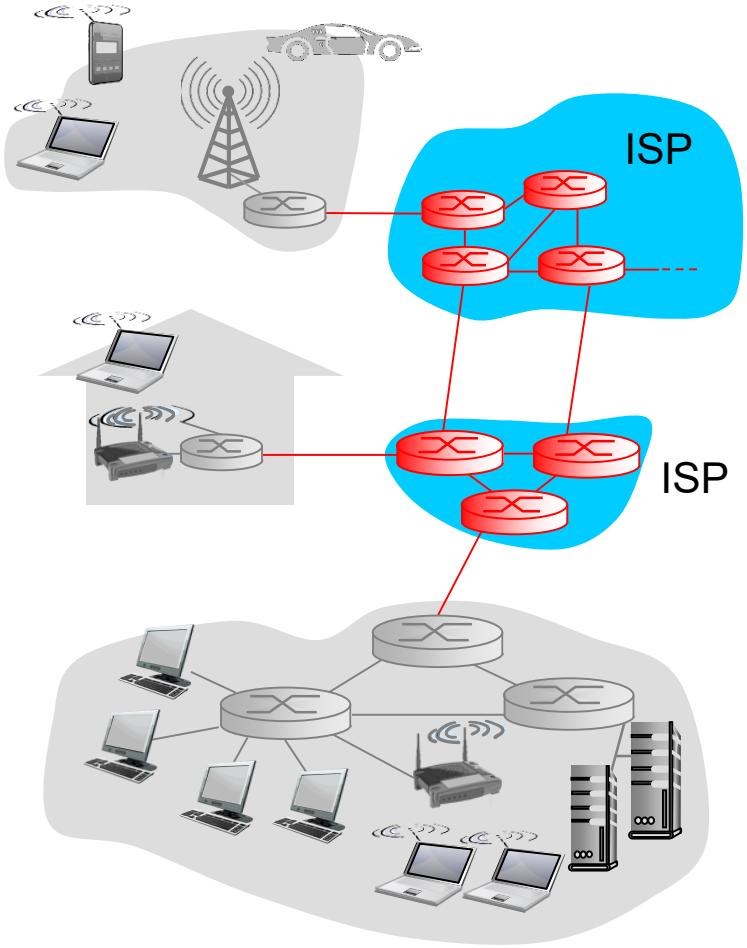
1.5 Protocol Layers and Service Models

The Network Core

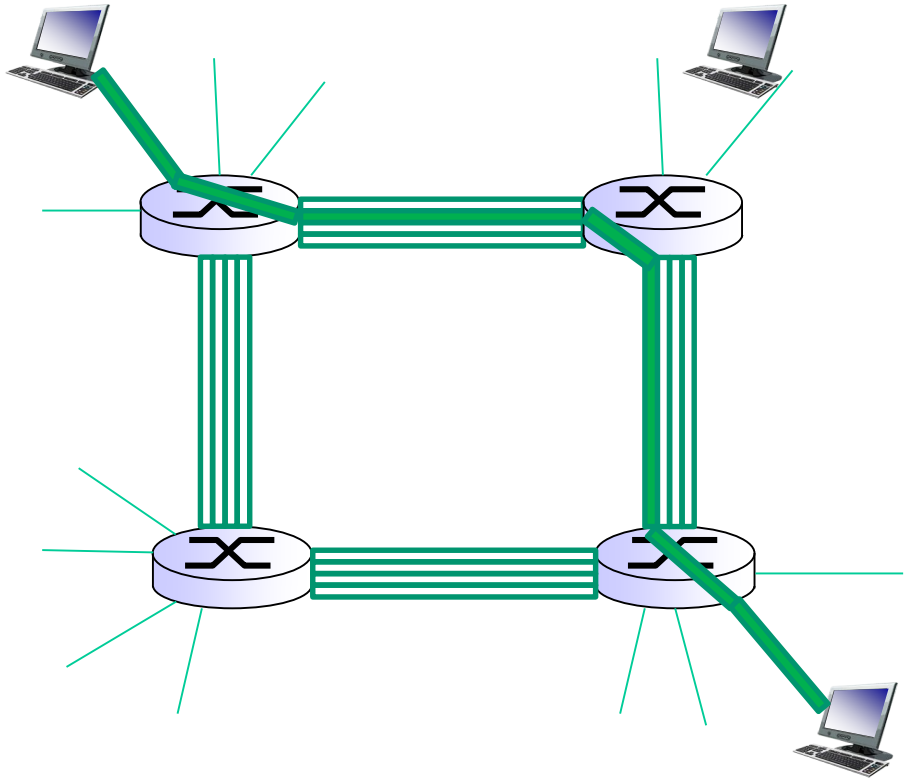
- A mesh of interconnected routers
- The fundamental question: how is data transmitted through net?
  - Circuit switching: dedicated circuit per call
  - Packet switching: data sent thru net in discrete “chunks”



Circuit Switching

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

- call setup required
- circuit-like (guaranteed) performance
- circuit segment idle if not used by call *(no sharing)*
- commonly used in traditional telephone networks
- divide link bandwidth into "pieces"
  - frequency division
  - time division

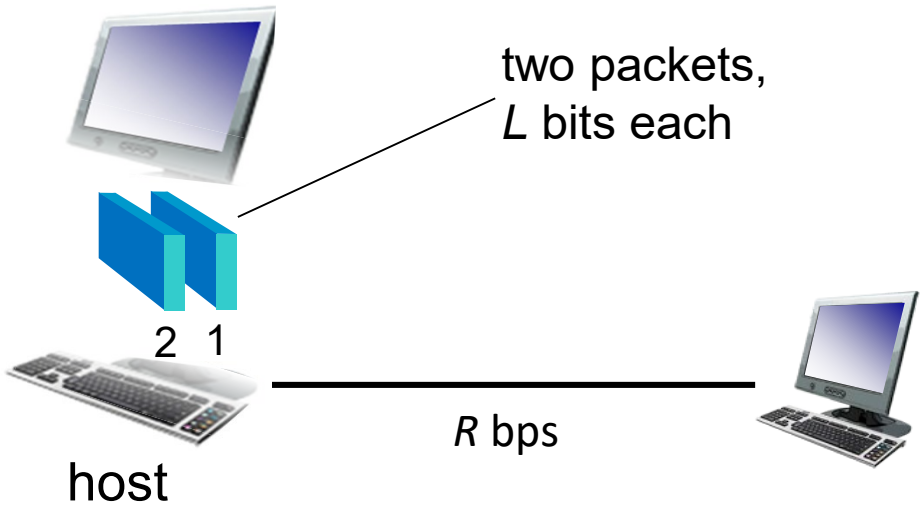


In above diagram, each link has four circuits. A "call" gets 2nd circuit in top link and 1st circuit in right link.

Packet Switching

Host sending function:

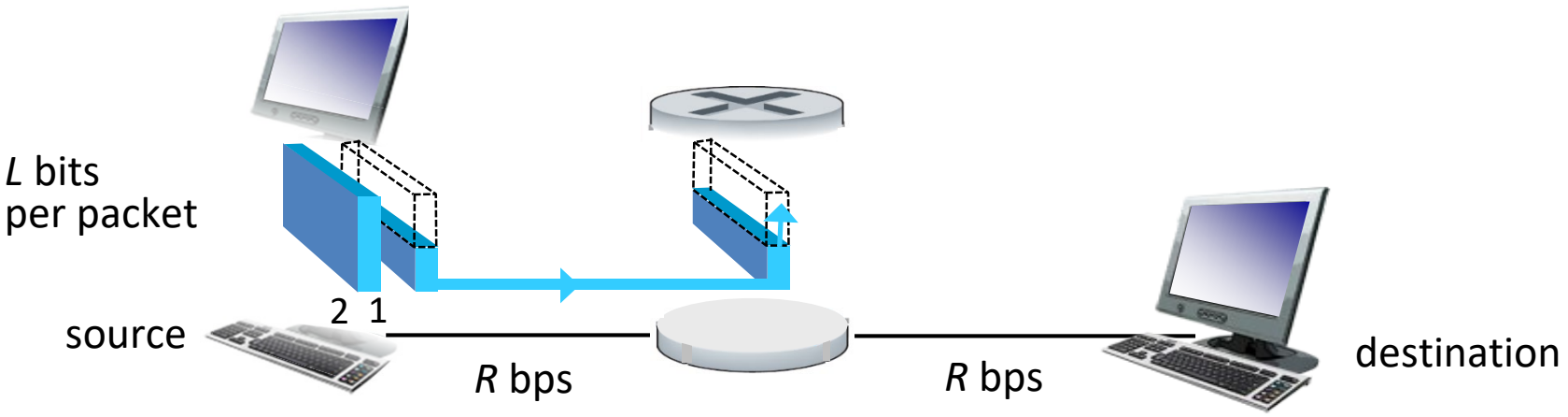
- breaks application message into smaller chunks, known as *packets*, of length *L* bits
- transmits packets onto the link at *transmission rate R*
  - link transmission rate is aka *link capacity* or *link bandwidth*



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

Packet-switching: store-and-forward

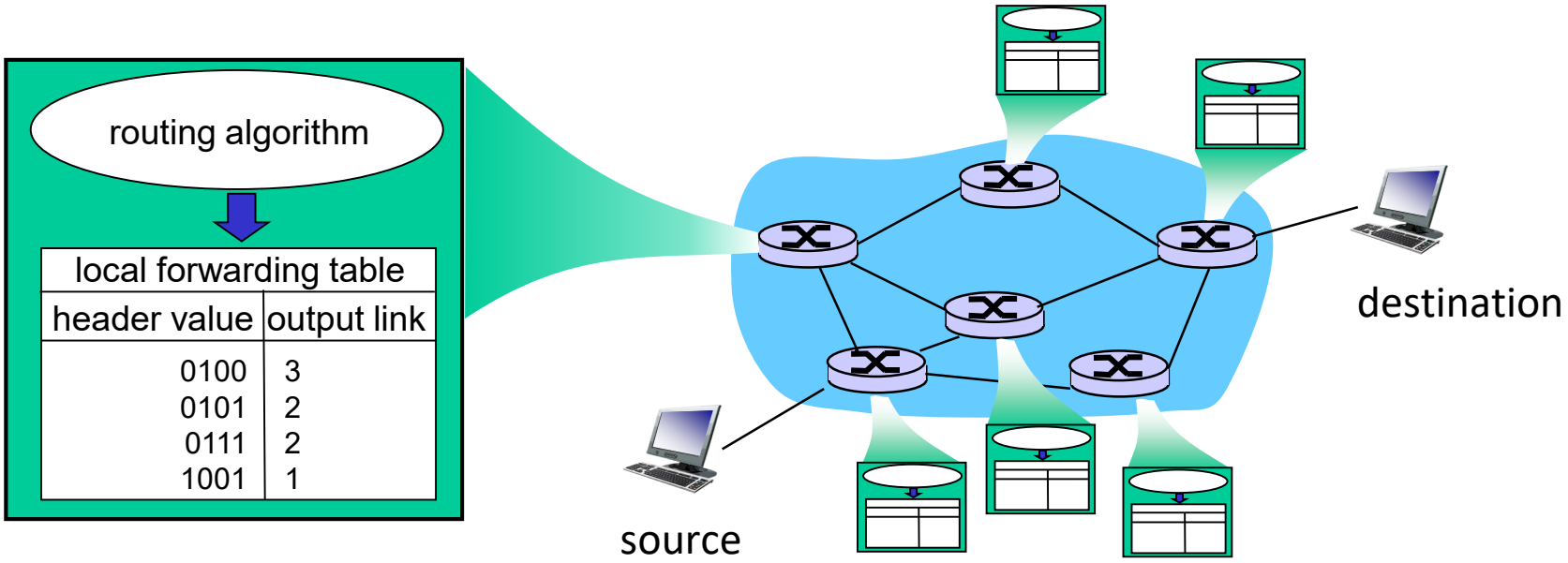
- Packets are passed from one router to the next, across links on path from source to destination.
- *Store and forward:* entire packet must arrive at a router before it can be transmitted on the next link.



End-to-end delay = 2*L/R (assuming no other delay)

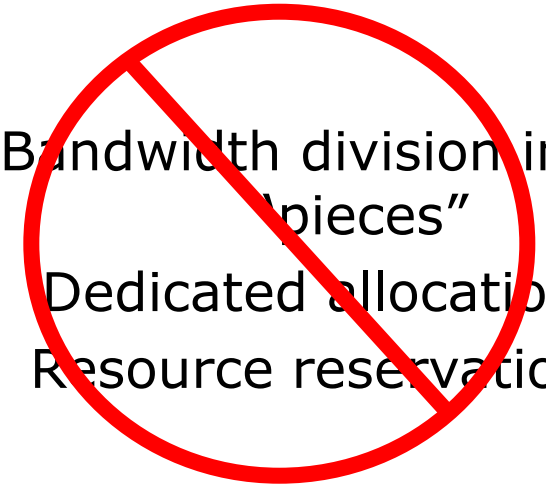
Routing and Addressing

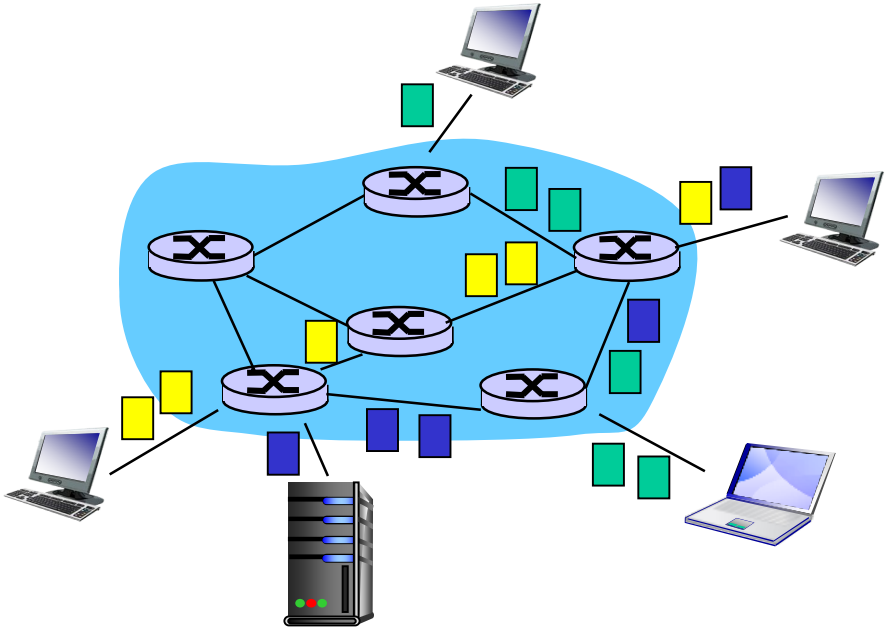
- Routers determine source-destination route taken by packets.
  - Routing algorithms
- Addressing: each packet needs to carry source and destination information



Packet Switching

- The Internet is a packet switching network
- User A, B … 's packets *share* network resources
- Resources are used on demand
- Excessive congestion is possible



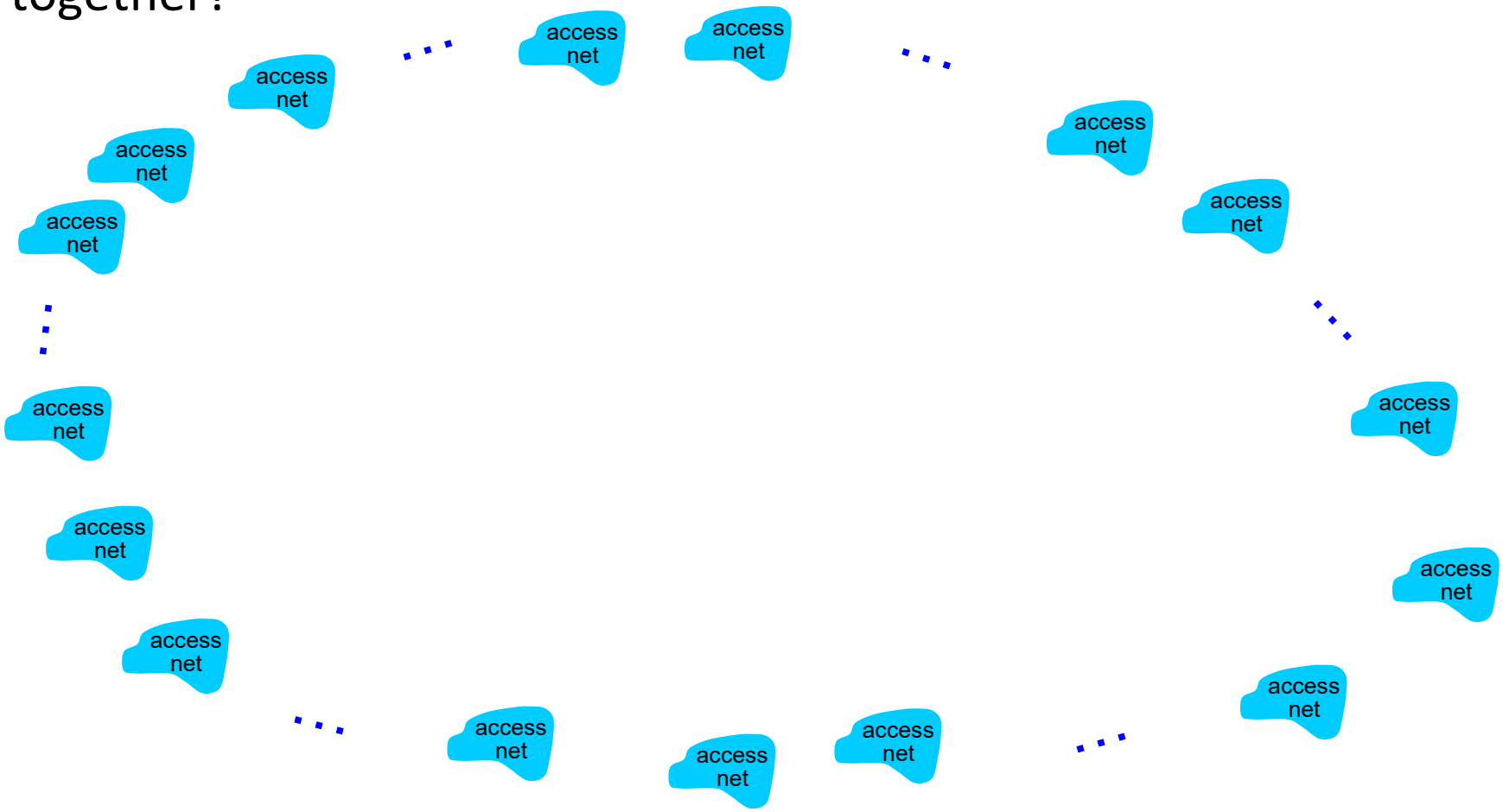


Internet Structure: Network of Networks

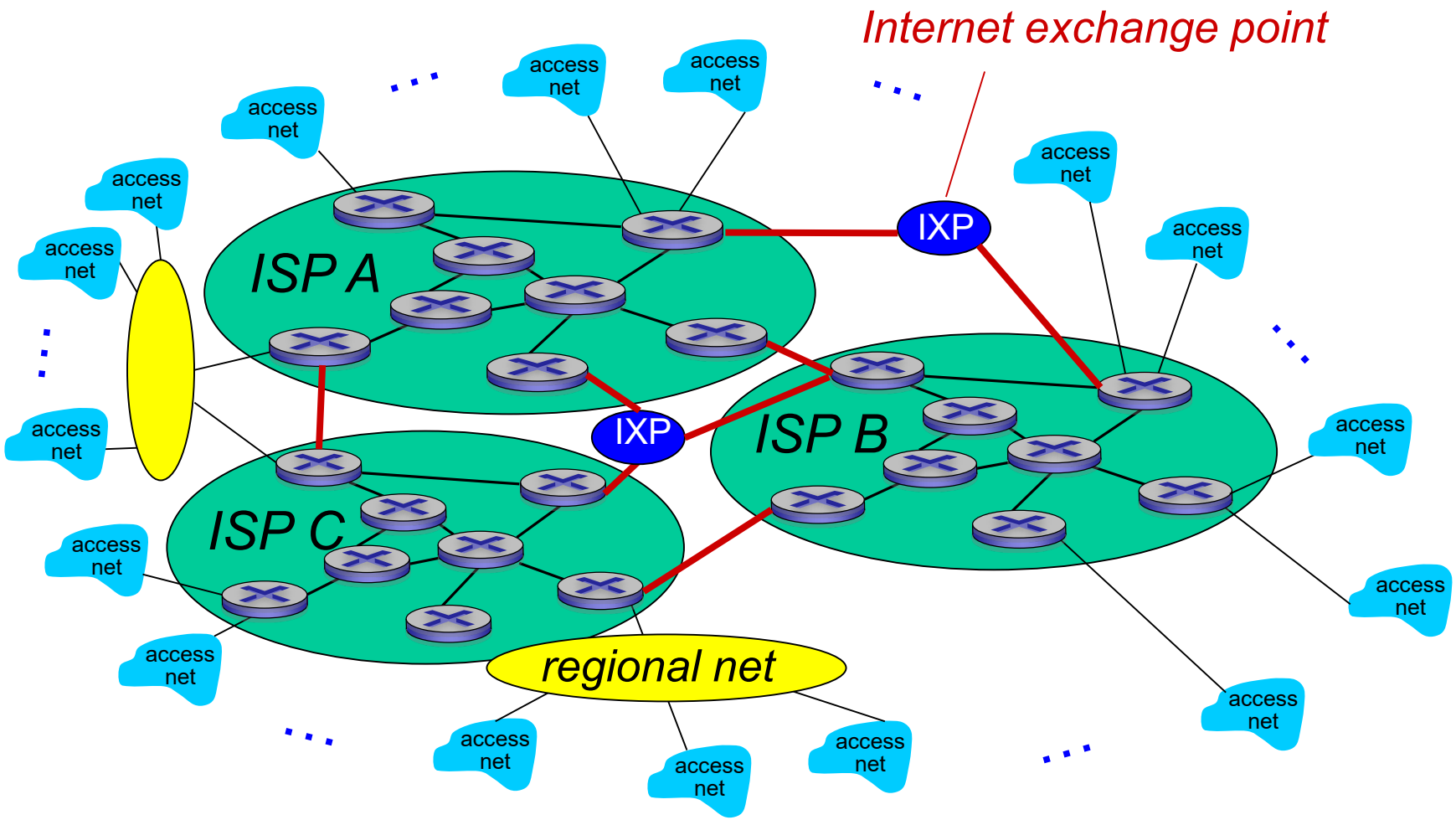
- Hosts connect to Internet via access ISPs (Internet Service Providers)
  - Residential, company and university ISPs
- Access ISPs in turn must be interconnected.
- Resulting network of networks is very complex
  - Evolution was driven by economics and national policies
- Therefore, the Internet is a "network-of-networks", organized into autonomous systems (AS), each is owned by an organization.

Internet Structure: Network of Networks

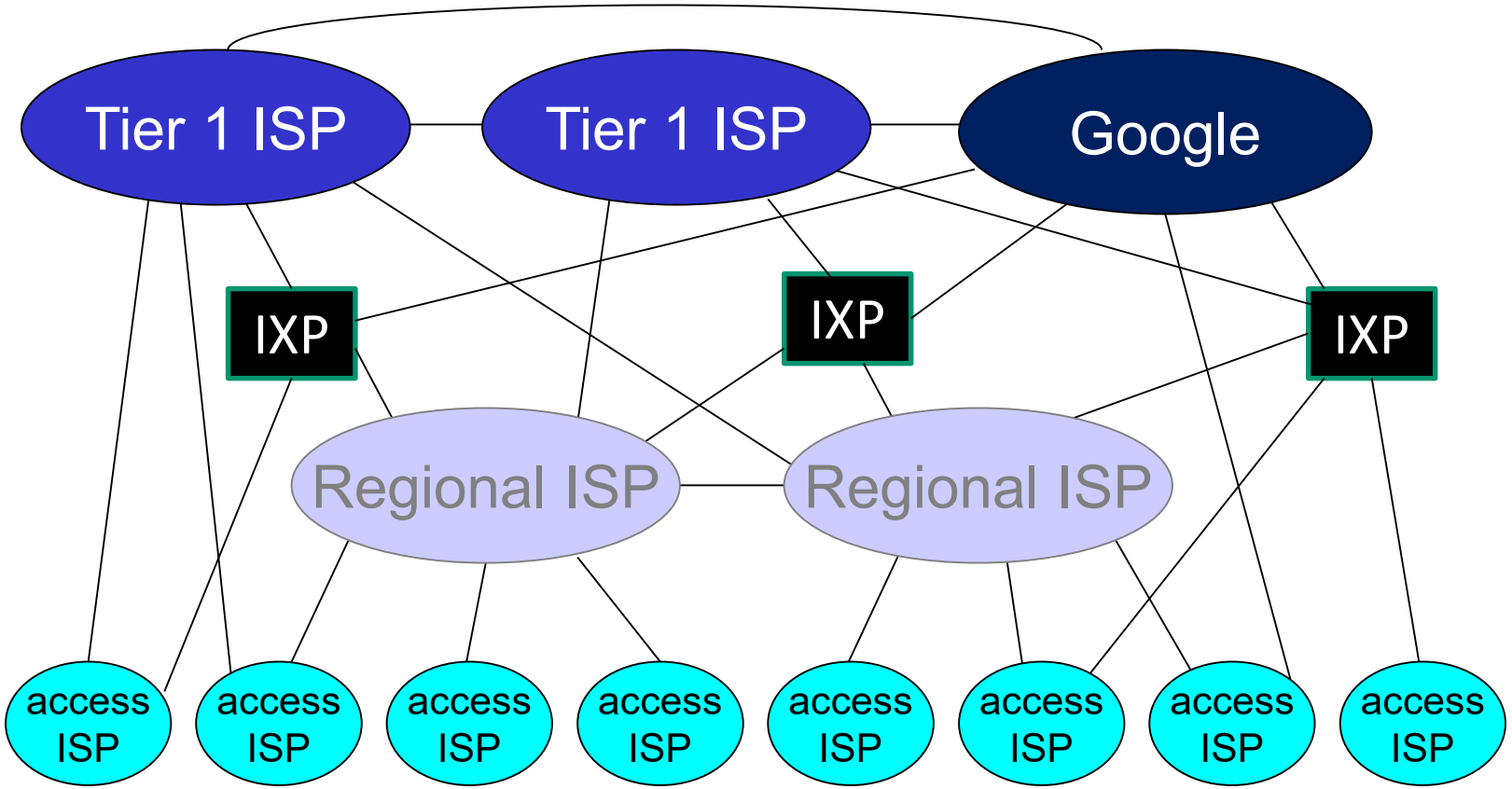
Question: given *millions* of access nets, how to connect them together?



Internet Structure: Network of Networks



Internet Structure: Network of Networks



Who Runs the Internet?

- IP address & Internet Naming administered by Network Information Centre (NIC)
  - Refer to: www.sgnic.net.sg; www.apnic.org
- The Internet Society (ISOC) - Provides leadership in Internet related standards, education, and policy around the world.
- The Internet Architecture Board (IAB) - Authority to issue and update technical standards regarding Internet protocols.
- Internet Engineering Task Force (IETF) - Protocol engineering, development and standardization arm of the IAB.
  - Internet standards are published as RFCs (Request For Comments)
    - Refer to: www.ietf.org; for RFCs: http://www.ietf.org/rfc.html

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Recall: Packet Switching Network

- To send a packet in a packet switching network,
  1. Sender transmit a packet onto the link as a sequence of bits.
  2. Bits are propagated to the next node (e.g. a router) on the link.
  3. Router stores, processes and forwards the packet to the next link.
  4. Steps 2 & 3 repeat till the packet arrives at the receiver.

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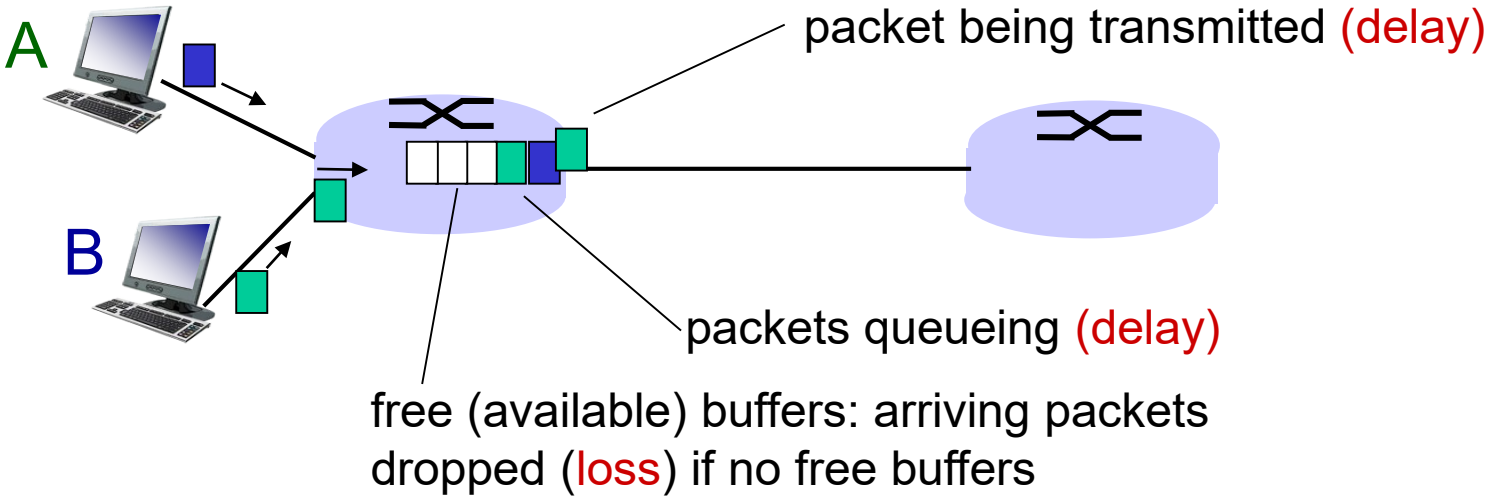
How do Delay and Loss Occur?

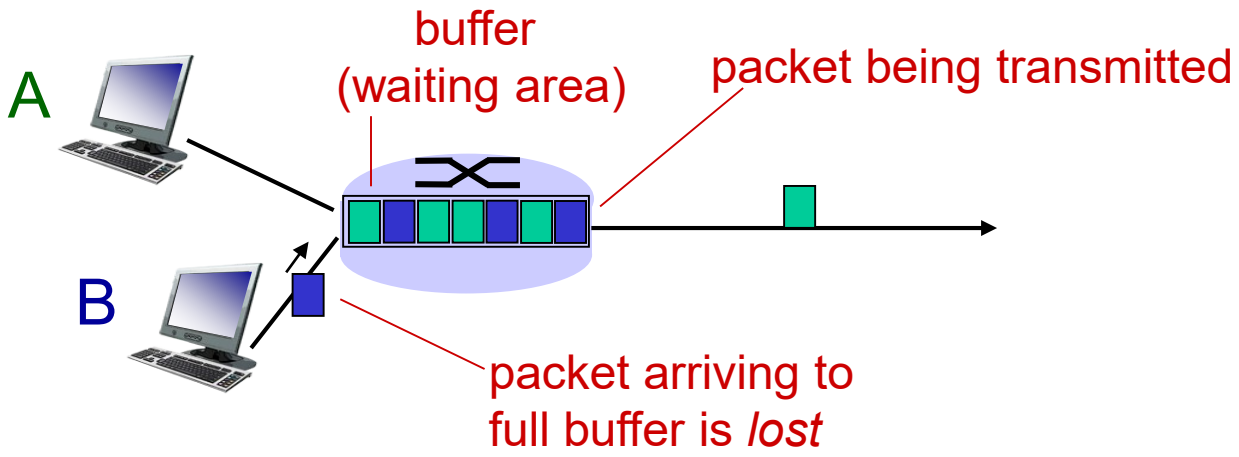
- Packets *queue* in router buffers
  - wait for turn to be sent out one by one



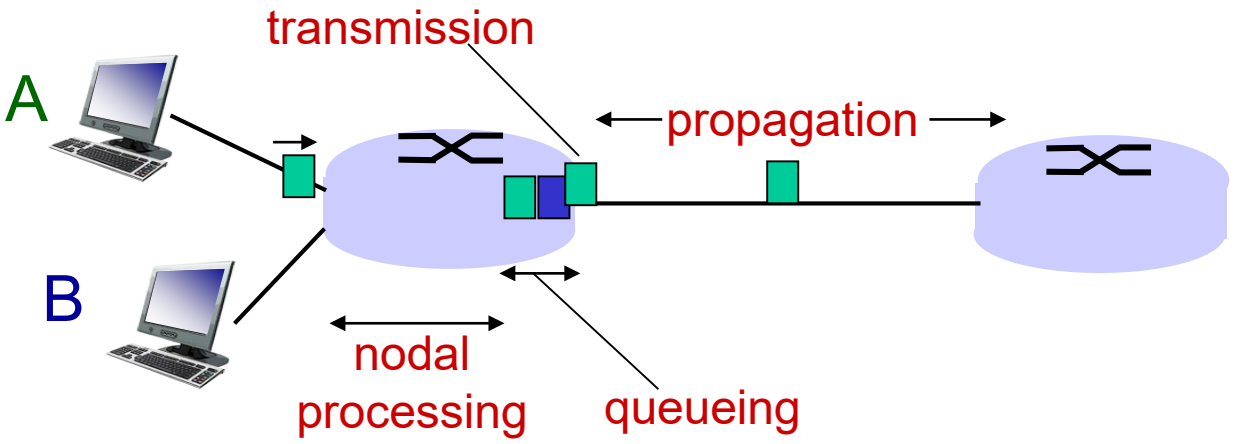
Q: What if packet arrival rate exceeds departure rate?

Packet Loss

- Queue (aka buffer) of a router has finite capacity.
- Packet arriving to full queue will be dropped (aka lost).
- Lost packet may be retransmitted by previous node, by source host, or not at all.



Four Sources of Packet Delay



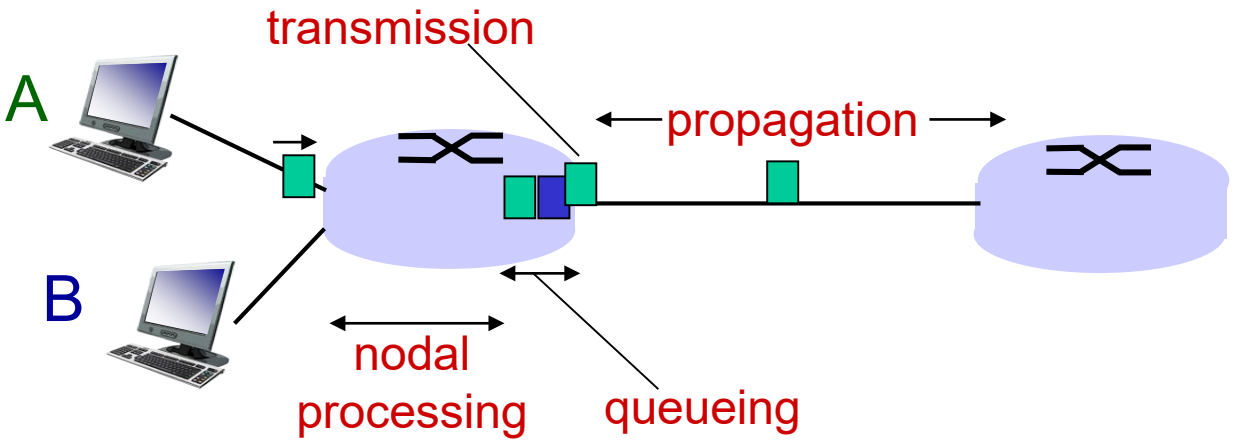
d_{proc}: nodal processing

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

d_{queue}: queuing delay

- time waiting in the queue for transmission
- depends on congestion level of router

Four Sources of Packet Delay



d_{trans}: transmission delay

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

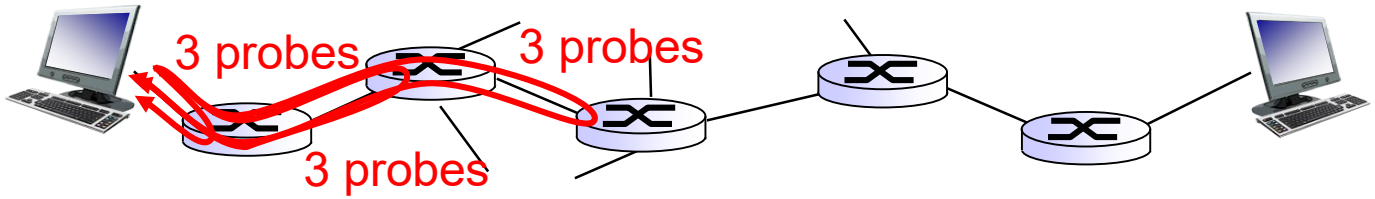
d_{prop}: propagation delay

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

End-to-end Packet Delay

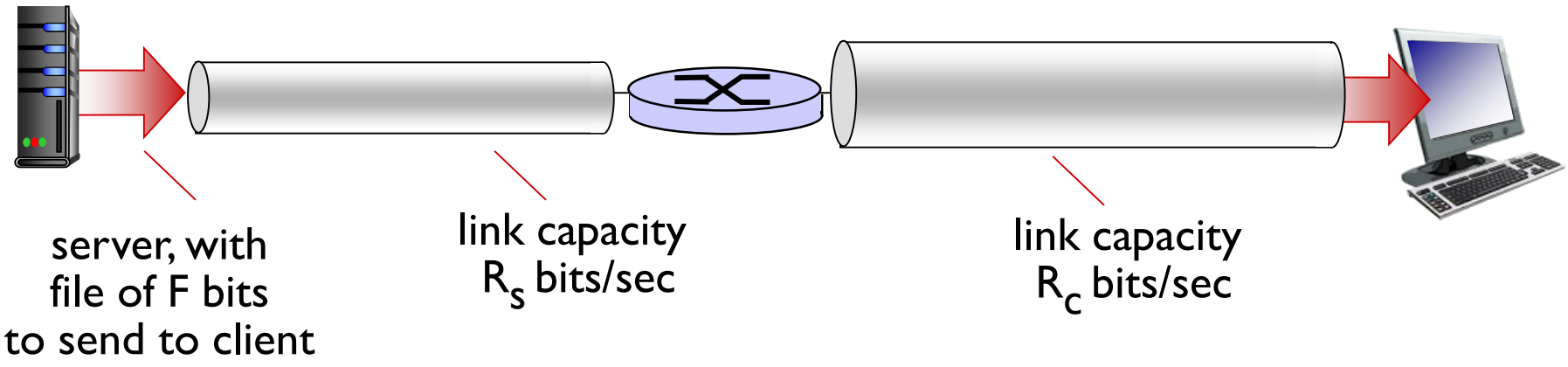
- End-to-end packet delay is the time taken for a packet to travel from source to destination. It consists of:
  - transmission delay
  - propagation delay
  - processing delay
  - queueing delay

traceroute program displays the route (path) from source to destination and measures the delay from source to each router along the end-end Internet path.



Throughput

- Throughput: how many bits can be transmitted per unit time.
  - Throughput is measured for end-to-end communication.
  - Link capacity (bandwidth) is meant for a specific link.



Metric Units

- 1 byte = 8 bits

Exp.	Explicit	Prefix	Exp.	Explicit	Prefix
10^{-3}	0.001	milli	10^{3}	1,000	Kilo
10^{-6}	0.000001	micro	10^{6}	1,000,000	Mega
10^{-9}	0.000000001	nano	10^{9}	1,000,000,000	Giga
10^{-12}	0.000000000001	pico	10^{12}	1,000,000,000,000	Tera
10^{-15}	0.000000000000001	femto	10^{15}	1,000,000,000,000,000	Peta
10^{-18}	0.000000000000000001	atto	10^{18}	1,000,000,000,000,000,000	Exa
10^{-21}	0.000000000000000000001	zepto	10^{21}	1,000,000,000,000,000,000,000	Zetta
10^{-24}	0.000000000000000000000001	yocto	10^{24}	1,000,000,000,000,000,000,000,000	Yotta

The principal metric prefixes

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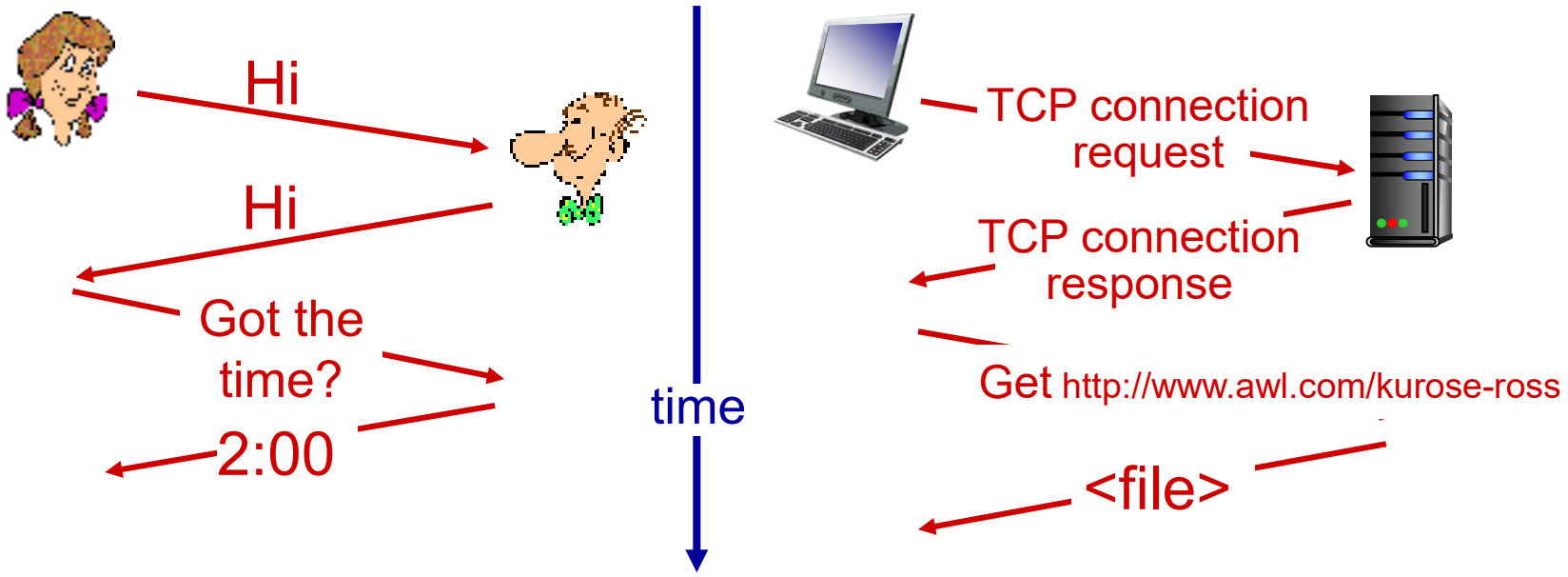
Internet: a Service View

- The Internet supports various kinds of network applications:
  - Web, VoIP, email, games, e-commerce, social nets, …
- Network applications exchange messages and communicate among peers according to protocols.

What's a Protocol?

a human protocol and a computer network protocol:



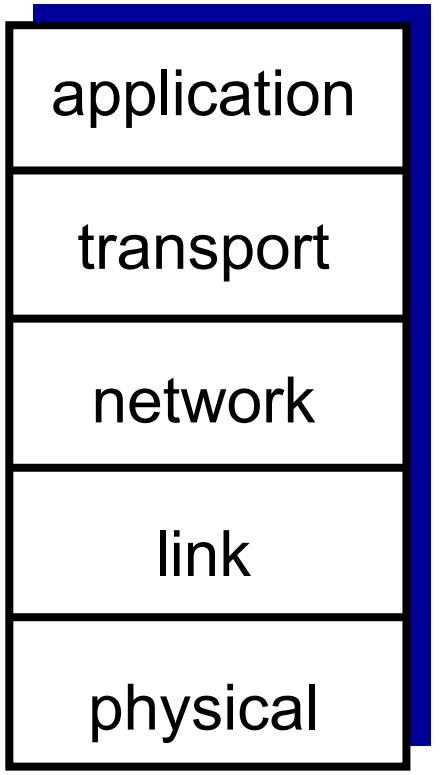
Protocols define format and order of messages exchanged and the actions taken after messages are sent or received.

Protocol "Layers"

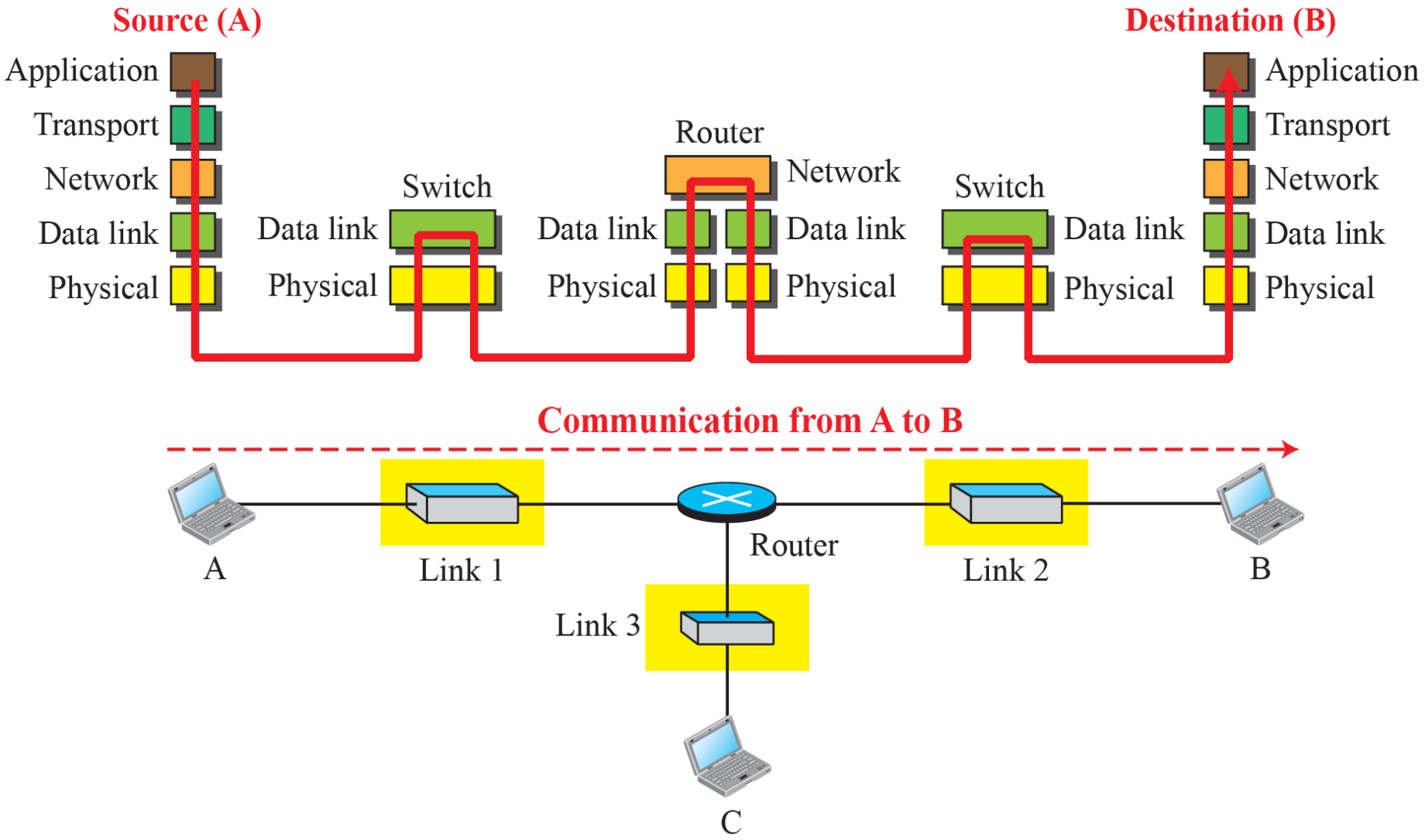
- Protocols in the Internet are logically organized into "layers" according to their purposes.
  - Each layer provides a service
  - Simple interfaces between layers
  - Hide details from each other
- Layering is a common CS trick to deal with large and complex systems.
  - Explicit structure allows identification, relationship of complex system's pieces
  - Modularization eases maintenance, updating of system
    - E.g. change of implementation of one layer's service is transparent to rest of system

Internet Protocol Stack

- *application:* supporting network applications
  - FTP, SMTP, HTTP
- *transport:* process-to-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"

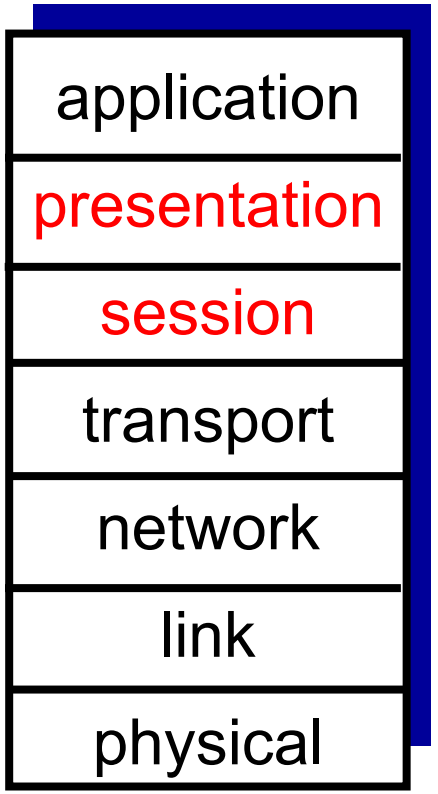


Example


Source (A)
Destination (B)
Application
Transport
Network
Data link
Physical
Switch
Router
Switch
Communication from A to B
A
Link 1
Router
Link 2
B
Link 3
C

ISO/OSI reference model (FYI)

- Theoretical model – not in use
- Two additional layers not present in Internet Protocol Stack
  - *presentation:* allow applications to interpret meaning of data, e.g., encryption, compression, machine-specific conventions
  - *session:* synchronization, checkpointing, recovery of data exchange



Lecture 1: Summary

covered a "ton" of material!

- Internet overview
- Network edge, core, access network
  - packet-switching versus circuit-switching
  - Internet structure
- Performance: loss, delay, throughput
- What's a protocol?
- Layering, service models

you now have:

- Context, overview, "feel" of networking
- More depth, detail *to follow!*

CS2105

An *Awesome* Introduction to Computer Networks

Lectures 2&3: The Application Layer

Department of Computer Science
School of Computing

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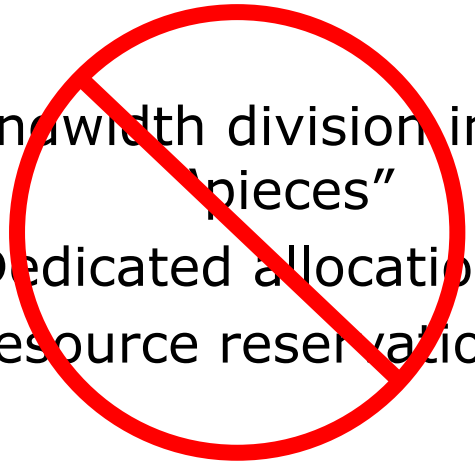
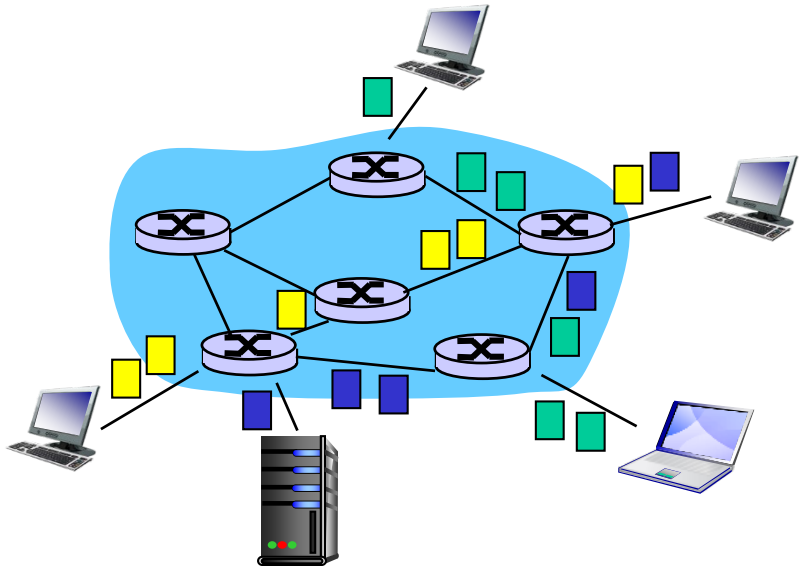
PREVIOUS LECTURE

Packet Switching

- The Internet is a packet switching network
  - Hosts share and contend network resources.
  - Application message is broken into a bunch of packets and sent onto the link one by one.
  - A router stores and forwards packets.
  - Receiver assembles all the packets to restore the application message.





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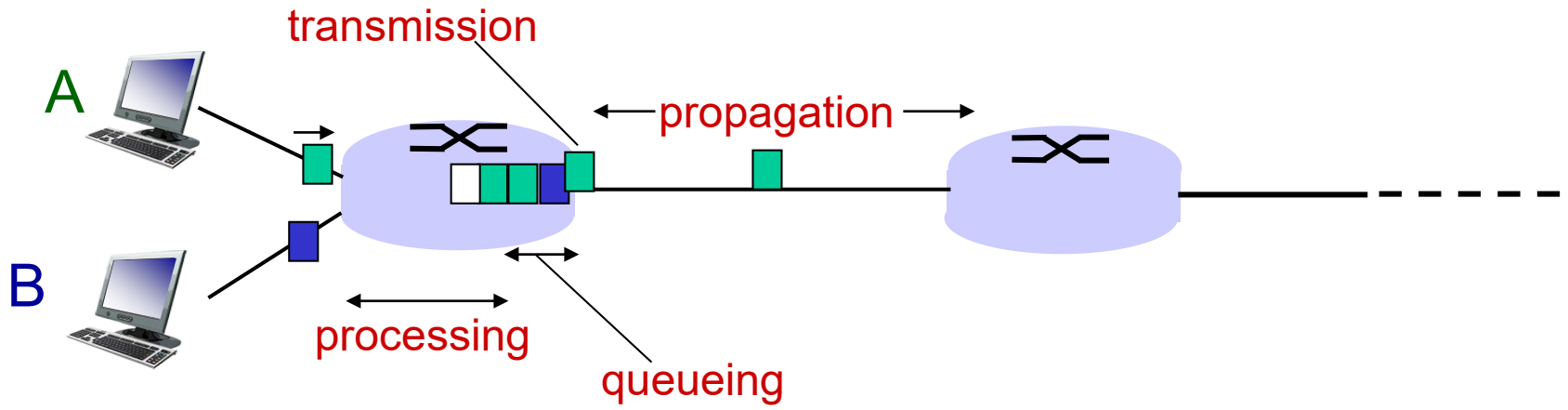
Packet Delay

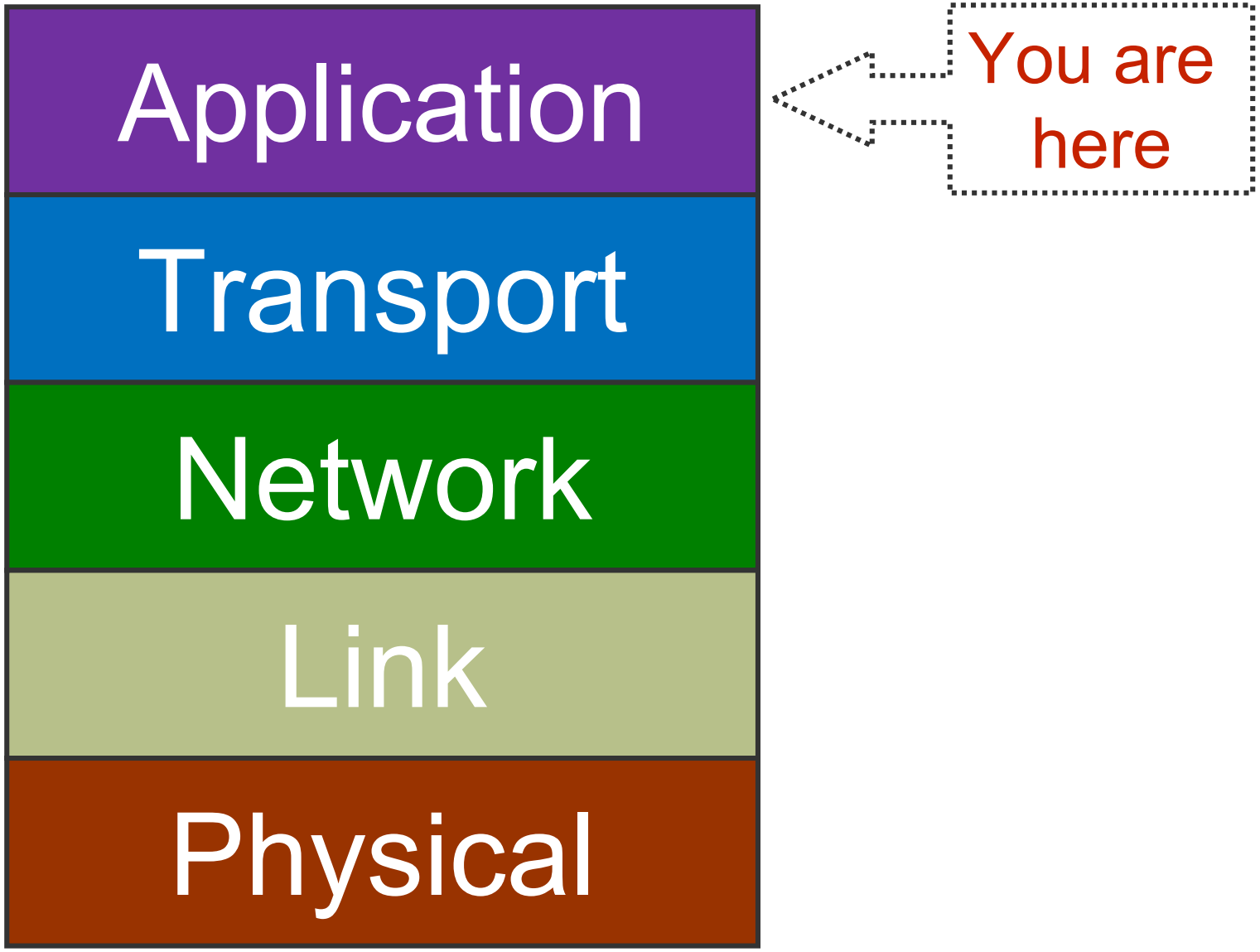
- End-to-end delay is the time taken for a packet to travel from source to destination. It consists of:
  - processing delay
  - queueing delay
  - transmission delay
  - propagation delay



Network Protocols

- Networks are complex. There are many issues to consider, to support different applications running on large number of hosts through different access technology and physical media.

- Protocols regulate communication activities in a network.
  - Define the *format* and *order* of messages exchanged between hosts for a specific purpose.


Application
Transport
Network
Link
Physical
You are here

Lectures 2&3: Application Layer

After this class, you are expected to:

- understand the basic HTTP interactions between the client and the server, the concepts of persistent and non-persistent connections.
- understand the services provided by DNS and how a DNS query is resolved.
- understand the concept of socket.
- be able to write simple client/server programs through socket programming.

Lectures 2&3: Roadmap

2.1 Principles of Network Applications

2.2 Web and HTTP

2.4 DNS

2.7 Socket programming

Kurose Textbook, Chapter 2
(Some slides are taken from the book)

Evolution of Network Applications

- Early days of Internet
  - Remote access (e.g. telnet, now ssh)
- 70s – 80s
  - Email, FTP
- 90s
  - Web
- 2000s – now
  - P2P file sharing
  - Online games
  - Instant Messaging, Skype
  - YouTube, Facebook

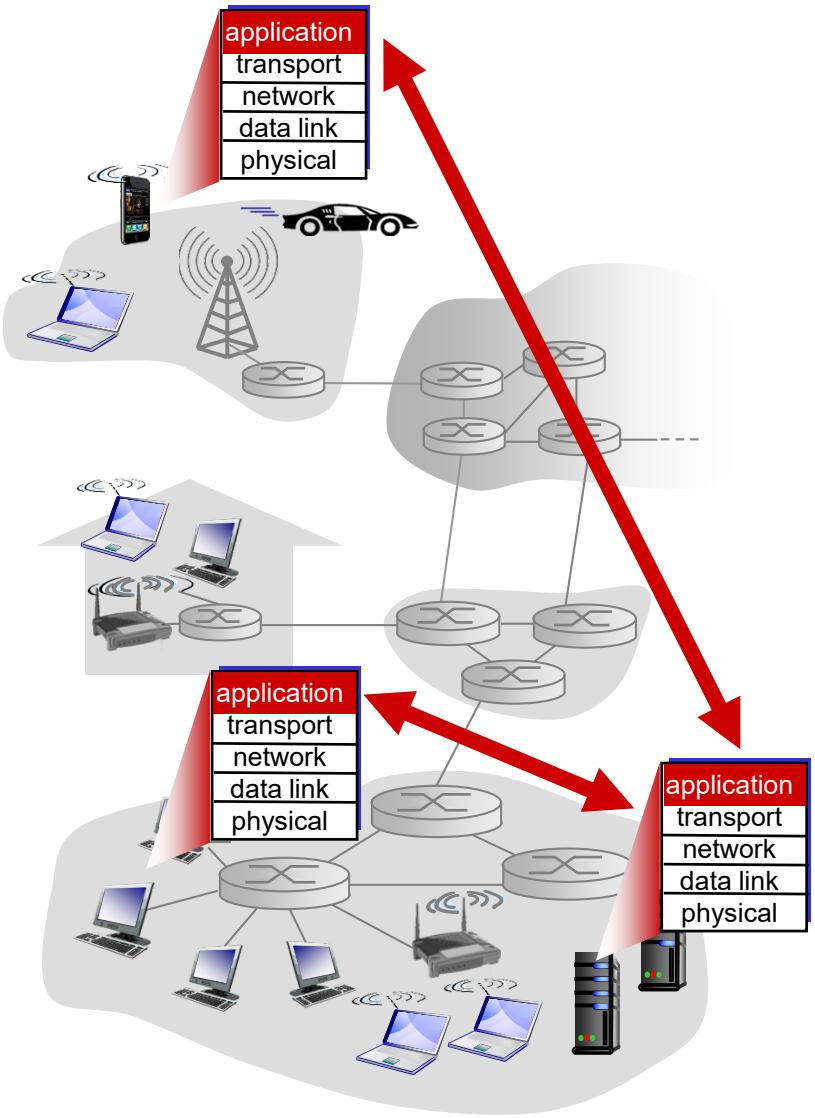
Creating Network Applications

write programs that:

- run on (different) *hosts*
- communicate over network
- e.g., web server software ⇔ browser software

classic structure of network applications:

- Client-server
- Peer-to-peer (P2P)



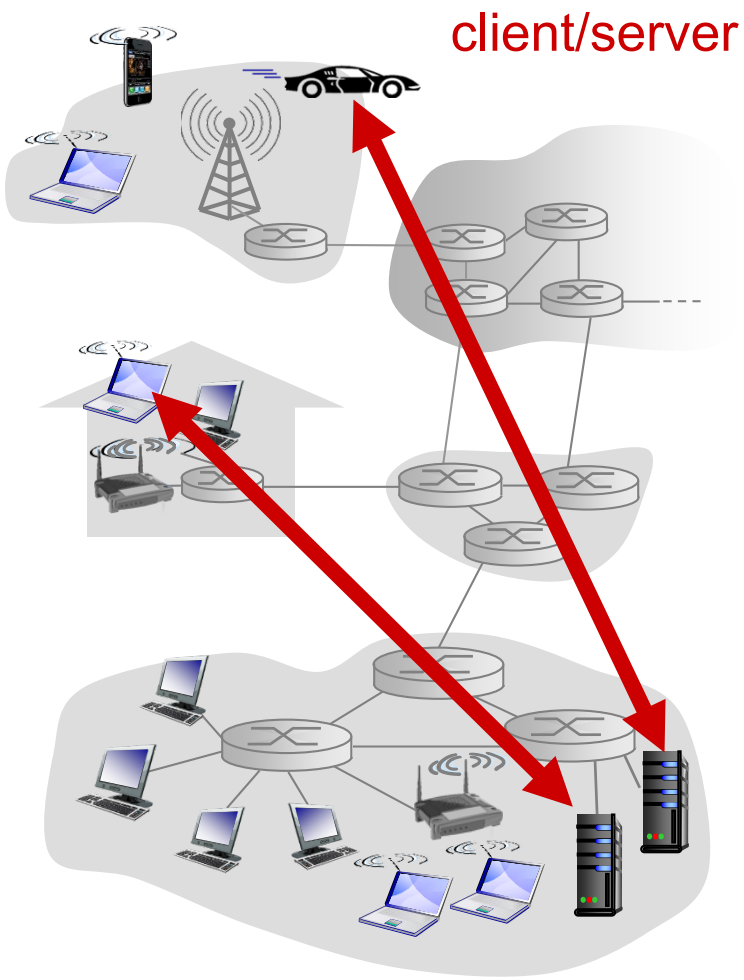
Client-Server Architecture

Server:

- Waits for incoming requests
- Provides requested service to client
- data centers for scaling

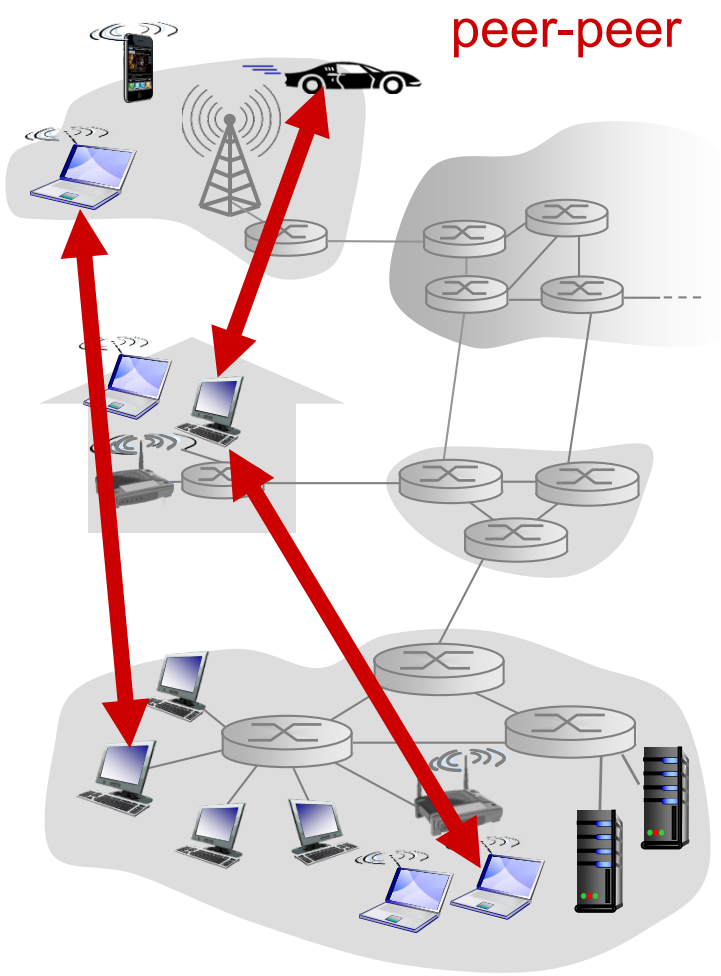
Client:

- Initiates contact with server ("speaks first")
- Typically requests service from server
- For Web, client is usually implemented in browser



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



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 streaming) can tolerate some data loss

Timing

- some apps (e.g., online interactive games) require low delay to be “effective”

Throughput

- some apps (e.g., multimedia) require minimum amount of bandwidth to be “effective”
- other apps (e.g., file transfer) make use of whatever throughput available

Security

- encryption, data integrity, authentication …

Requirements of Example Apps

Application	Data loss	Throughput	Time-sensitive
File transfer	No loss	Elastic	No
Electronic mail	No loss	Elastic	No
Web documents	No loss	Elastic	No
Real-time audio/video	Loss-tolerant	Audio: 5kbps-1Mbps Video:10kbps-5Mbps	Yes: 100s of msec
Stored audio/video	Loss-tolerant	Same as above	Yes: few seconds
Interactive games	Loss-tolerant	Few kbps – 10 kbps	Yes: 100s of msec
Text messaging	No loss	Elastic	Yes and no

App-layer Protocols Define…

- 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 applications send & respond to messages

- open protocols:
  - defined in RFCs
  - allows for interoperability
  - e.g., HTTP, SMTP
- proprietary protocols:
  - e.g., Skype

Internet Transport Protocols

TCP service:

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

UDP service:

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

Example App/Transport Protocols

Application	Application Layer Protocol	Underlying Transport Protocol
Electronic mail	SMTP [RFC 5321]	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)	TCP
Internet telephony	SIP [RFC 3261], RTP [RFC 3550], or proprietary (e.g., Skype)	TCP or UDP

Lectures 2&3: Roadmap

2.1 Principles of Network Applications

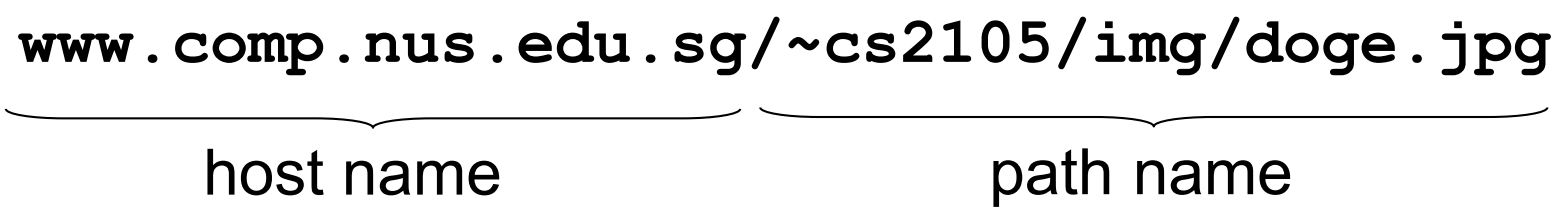
2.2 Web and HTTP

2.4 DNS

2.7 Socket programming

The Web: Some Jargon

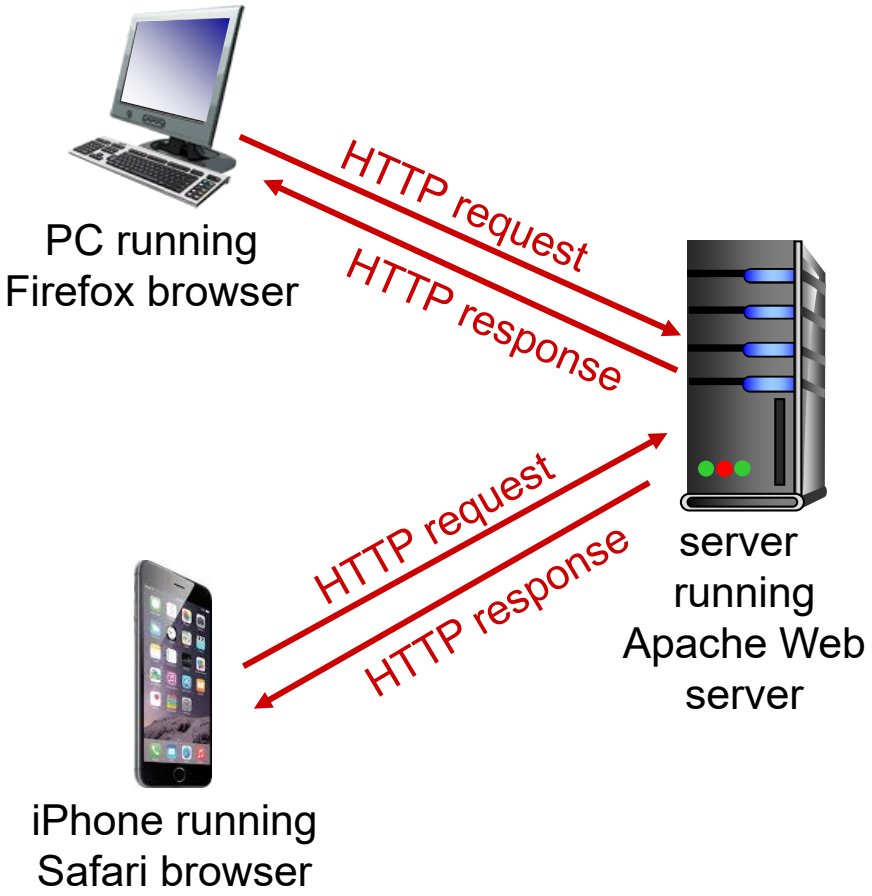
- A Web page typically consists of:
  - *base HTML file*, and
  - *several referenced objects*.
- An object can be HTML file, JPEG image, Java applet, audio file, …
- Each object is addressable by a *URL*, e.g.,



HTTP Overview

HTTP: Hypertext transfer protocol

- Web's application layer protocol
- Client/server model
  - *client:* usually is browser that requests, receives and displays Web objects
  - *server:* Web server sends objects in response to requests
- http 1.0: RFC 1945
- http 1.1: RFC 2616



HTTP Over TCP

HTTP uses TCP as transport service

- Client initiates TCP connection to server.
- Server accepts TCP connection request from client.
- HTTP messages are exchanged between browser (HTTP client) and Web server (HTTP server) over TCP connection.
- TCP connection closed.

Two Types of HTTP Connections

Non-persistent HTTP

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

Persistent HTTP

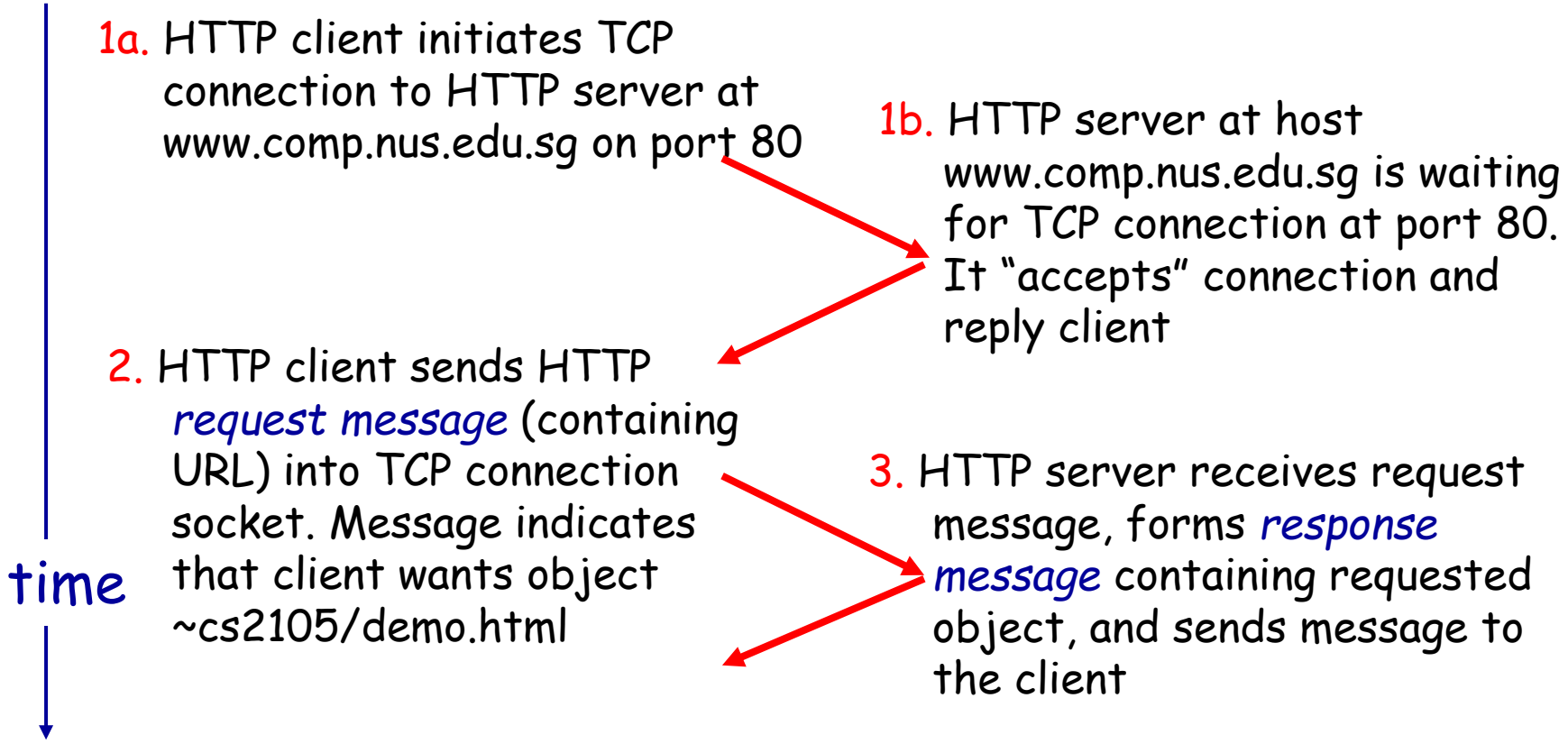
- multiple objects can be sent over single TCP connection between client, server

Non-persistent HTTP Example

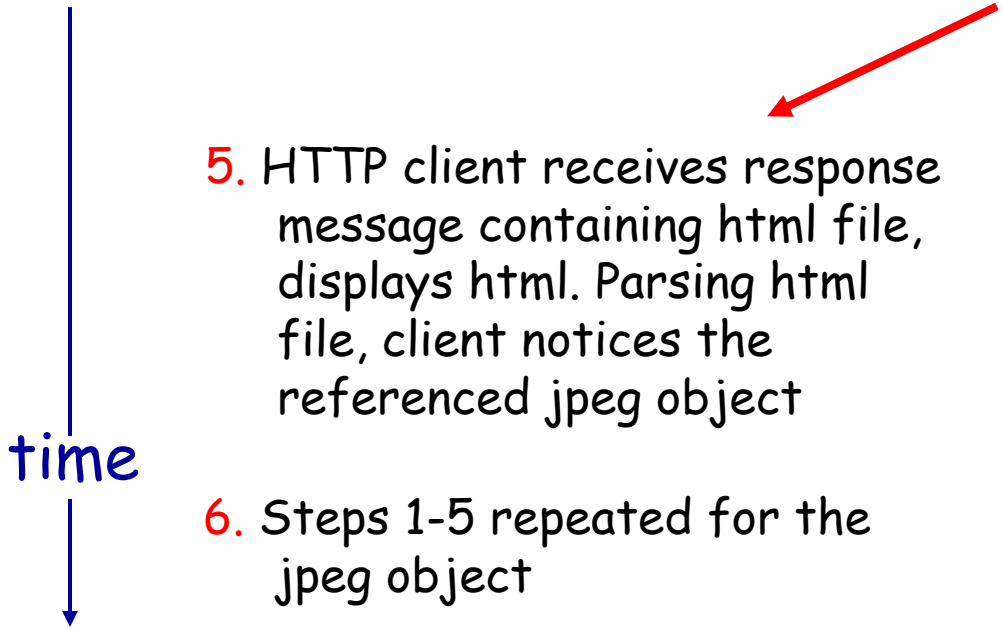
suppose user enters URL:

`www.comp.nus.edu.sg/~cs2105/demo.html`

(contains text, reference to a jpeg image)



Non-persistent HTTP Example



- This is an example of *non-persistent connection* (http 1.0).
  - One object per connection
- HTTP 1.1 allows *persistent connection* (to discuss).
  - Multiple objects per connection

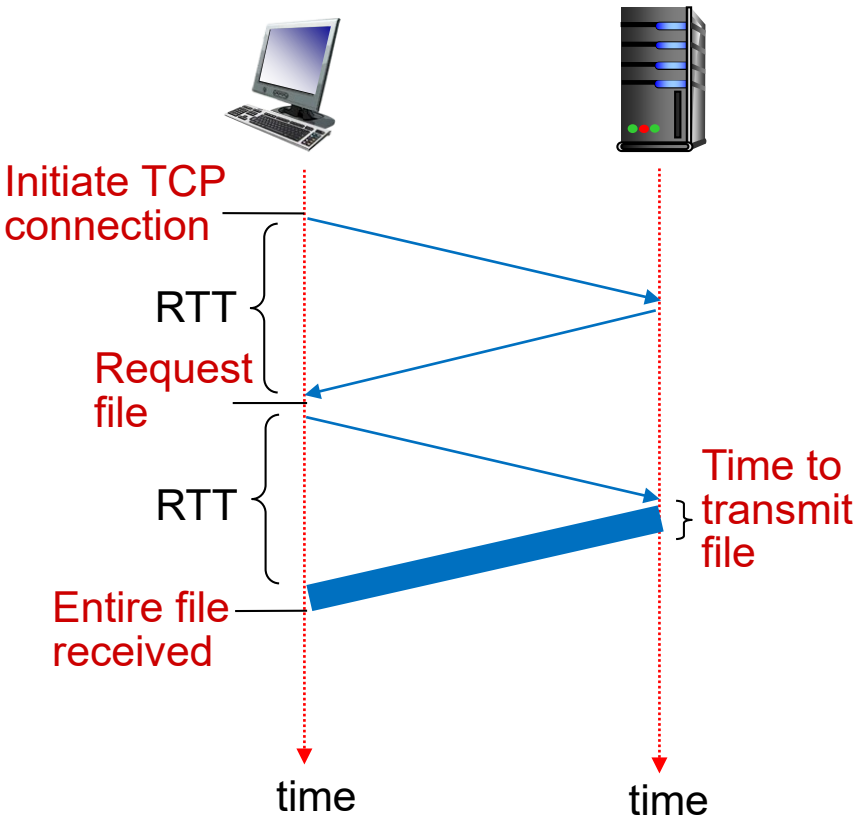
Non-persistent HTTP: Response Time

RTT: time for a packet to travel from client to server and go back

HTTP response time:

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

  2 * RTT+ file transmission time



Persistent HTTP

non-persistent HTTP issues:

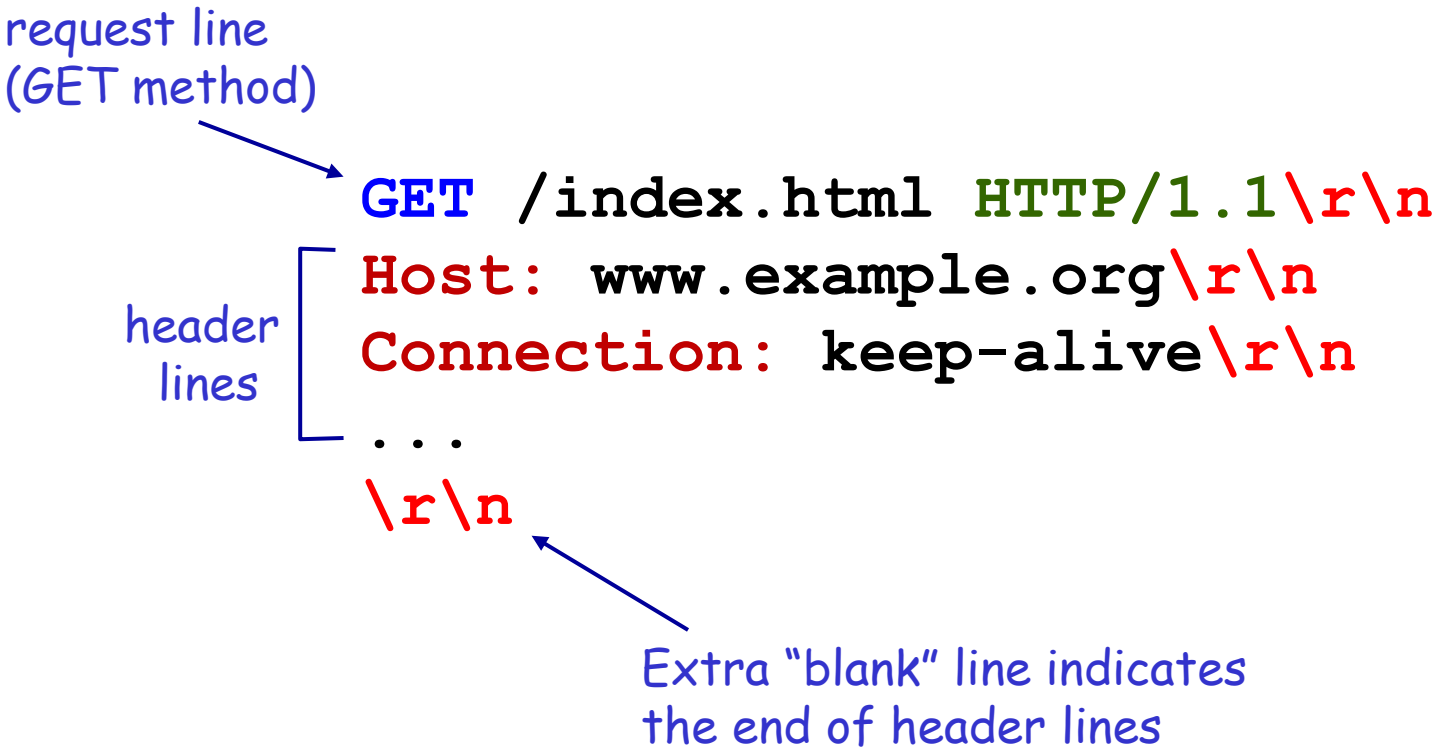
- requires 2 RTTs per object
- OS overhead for *each* TCP connection
- browsers often open parallel TCP connections to fetch referenced objects

persistent HTTP:

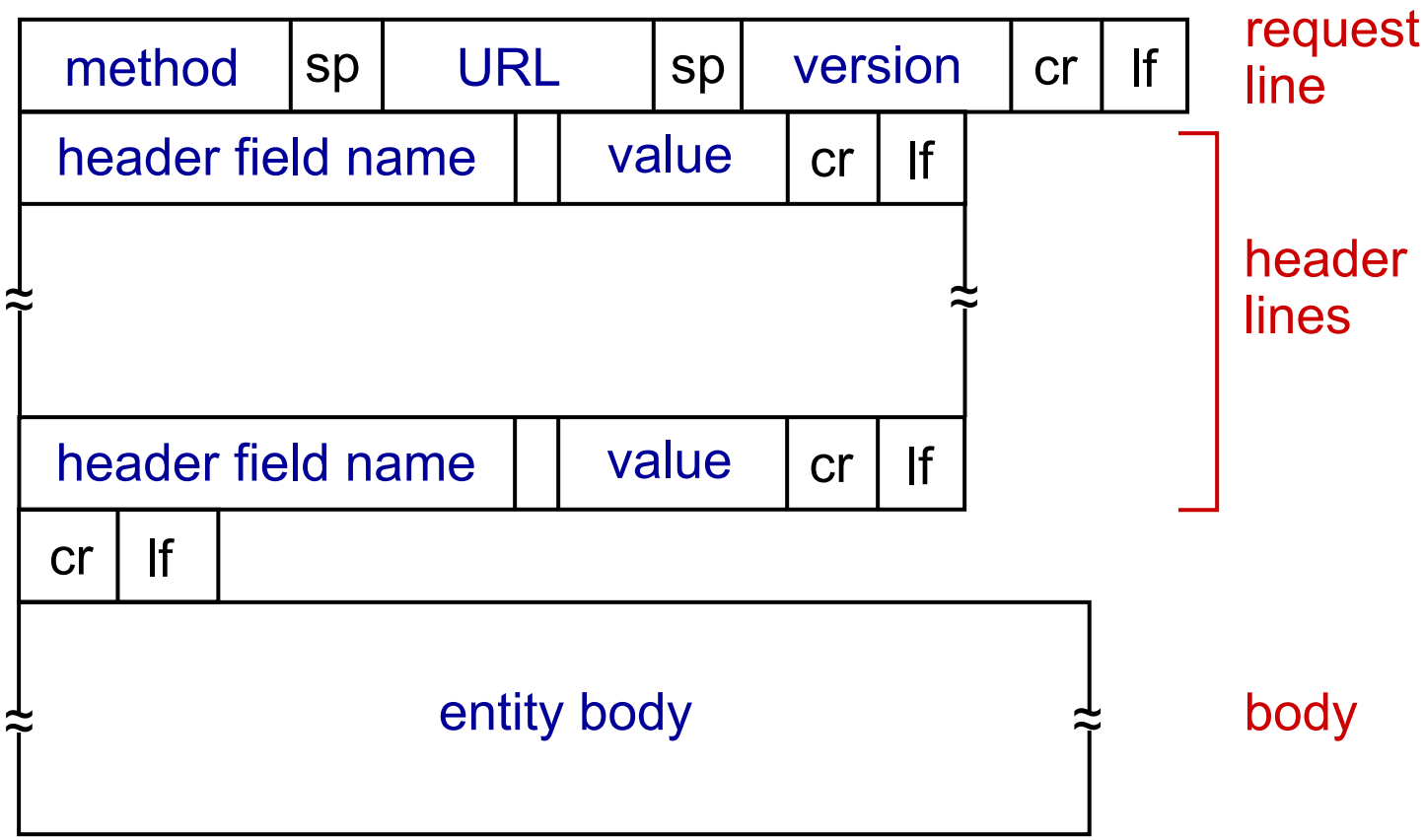
- server leaves connection open after sending response
- subsequent HTTP messages between same client/server sent over the same TCP connection
- moreover, client may send requests as soon as it encounters a referenced object (persistent with pipelining)
- as little as one RTT for all the referenced objects

Example HTTP Request Message

- Two types of HTTP messages: *request, response*
- HTTP request message:



HTTP Request Message: General Format



HTTP Request Method Types

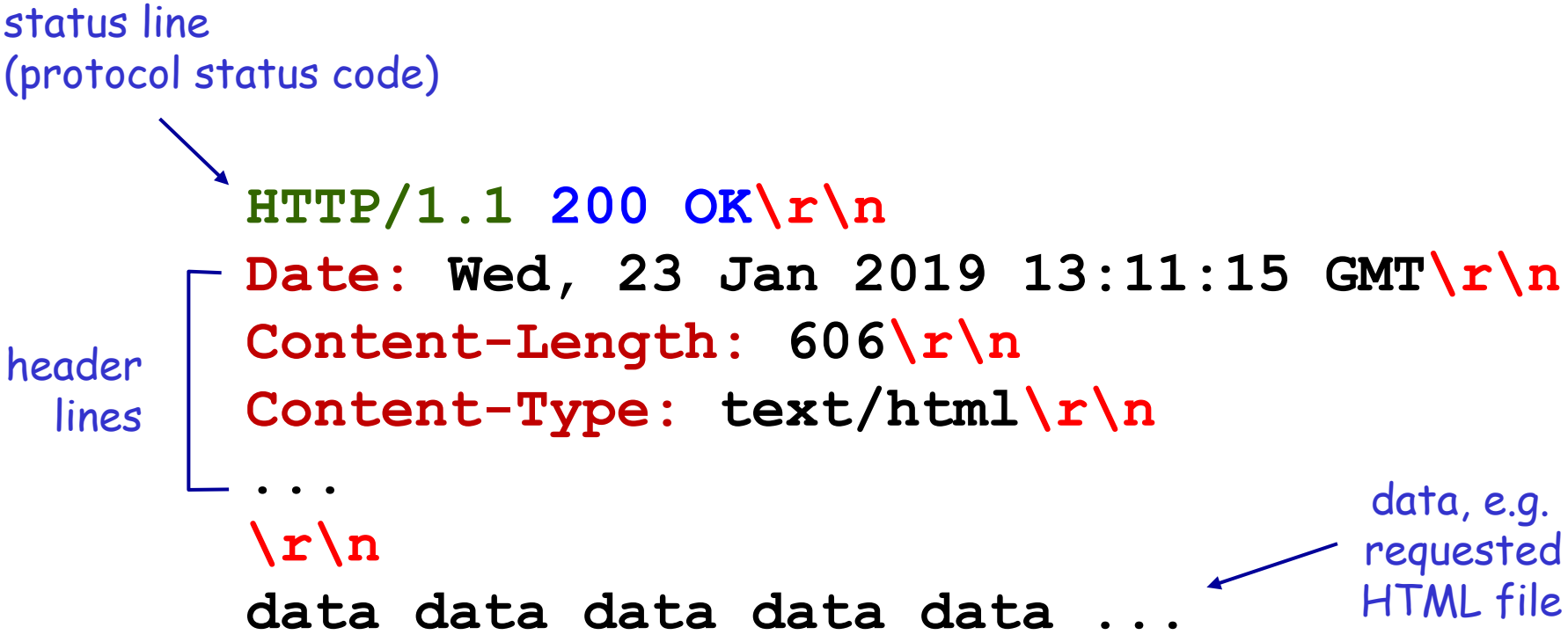
HTTP/1.0:

- GET
- POST
  - web page often includes form input
  - input is uploaded to server in entity body
- HEAD
  - asks server to leave requested object out of response

HTTP/1.1:

- GET, POST, HEAD
- PUT
  - uploads file in entity body to path specified in URL field
- DELETE
  - deletes file specified in the URL field

Example HTTP Response Message



For a full list of request/response header fields, check
http://www.w3.org/Protocols/rfc2616/rfc2616-sec14.html

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:)

403 Forbidden
- server declines to show the requested webpage

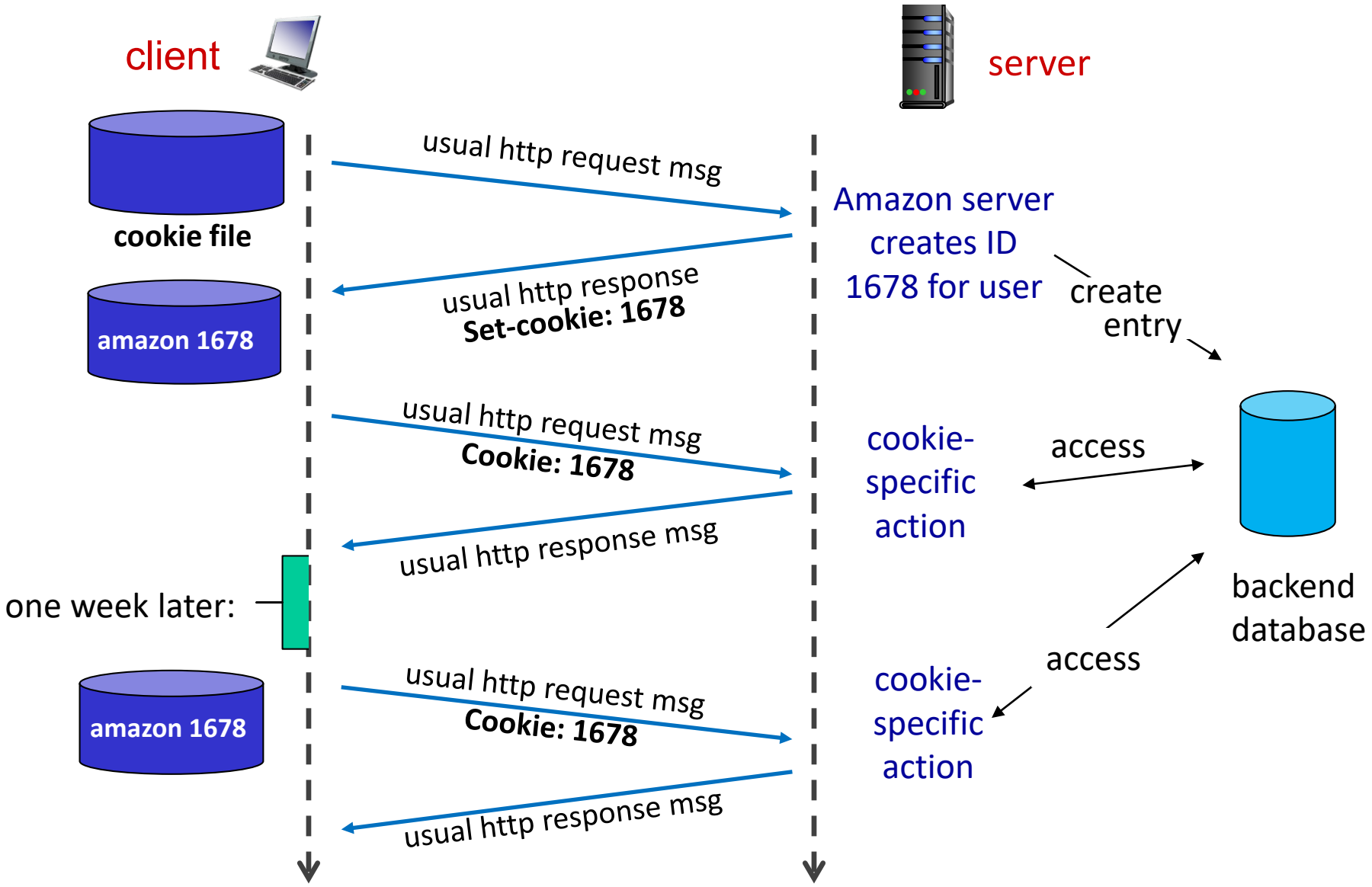
404 Not Found
- requested document not found on this server

For a full list of status codes, check
http://www.w3.org/Protocols/rfc2616/rfc2616-sec10.html

Cookies

- HTTP is designed to be “stateless”.
  - Server maintains no information about past client requests.
- Sometimes it’s good to maintain states (history) at server/client over multiple transactions.
  - E.g. shopping carts
- Cookie: http messages carry “state”
  1) cookie header field of HTTP *request* / *response* messages
  2) cookie file kept on user’s host, managed by user’s browser
  3) back-end database at Web site

Keep User States with Cookie



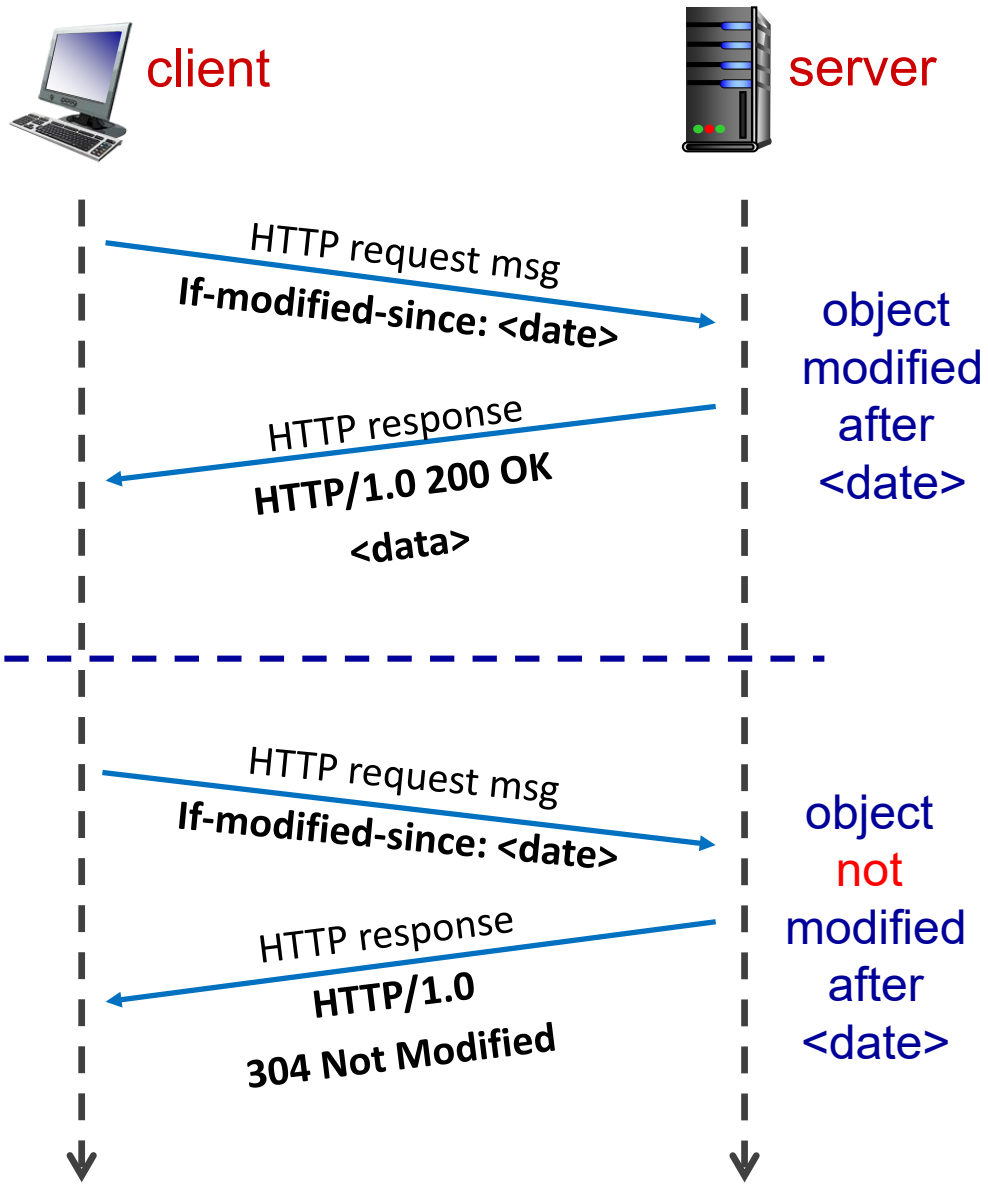
Conditional GET

- *Goal:* don't send object if (client) cache has up-to-date cached version
- *cache:* specify date of cached copy in HTTP request

```
If-modified-since:
  <date>
```

- *server:* response contains no object if cached copy is up-to-date:

```
HTTP/1.0 304 Not
  Modified
```



Lectures 2&3: Roadmap

2.1 Principles of Network Applications

2.2 Web and HTTP

2.4 DNS

2.7 Socket programming

Domain Name System [RFC 1034, 1035]

- Two ways to identify a host:
  - **Hostname**, e.g., www.example.org
  - **IP address**, e.g., 93.184.216.34

- DNS (Domain Name System) translates between the two.
  - A client must carry out a DNS query to determine the IP address corresponding to the server name (e.g., www.example.org) prior to the connection.

DNS: Resource Records (RR)

- Mapping between host names and IP addresses (and others) are stored as resource records (RR).

RR format: **`(name, value, type, ttl)`**

type = A

- **`name`** is hostname
- **`value`** is IP address

type = NS

- **`name`** is domain (e.g., **`nus.edu.sg`**)
- **`value`** is hostname of authoritative name server for this domain

type = CNAME

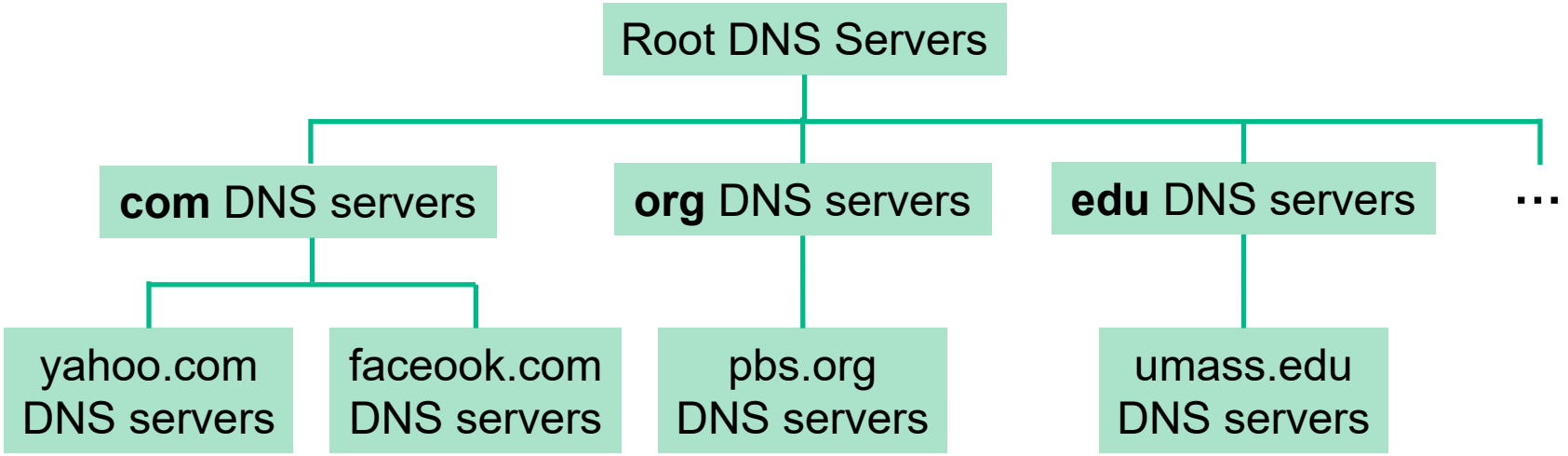
- **`name`** is alias name for some "canonical" (the real) name
- **`value`** is canonical name
- e.g. **`www.nus.edu.sg`** is really **`mgnzsqc.x.incapdns.net`**

type = MX

- **`value`** is name of mail server associated with **`name`**

Distributed, Hierarchical Database

- DNS stored RR in distributed databases implemented in hierarchy of many name servers.

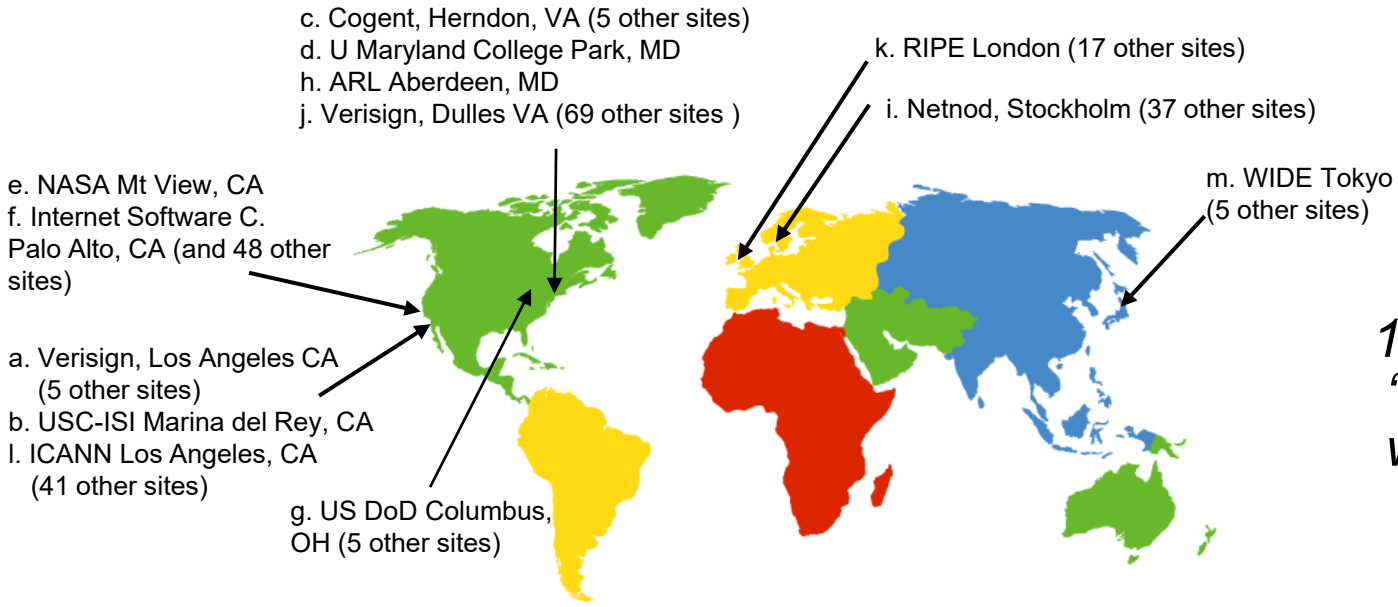


A client wants IP address for www.facebook.com:

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

Root Servers

- Answers requests for records in the root zone by returning a list of the authoritative name servers for the appropriate top-level domain (TLD).



13 root name "servers" worldwide

TLD and Authoritative Servers

Top-level domain (TLD) servers:

- responsible for com, org, net, edu, … and all top-level country domains, e.g., uk, sg, jp

Authoritative servers:

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

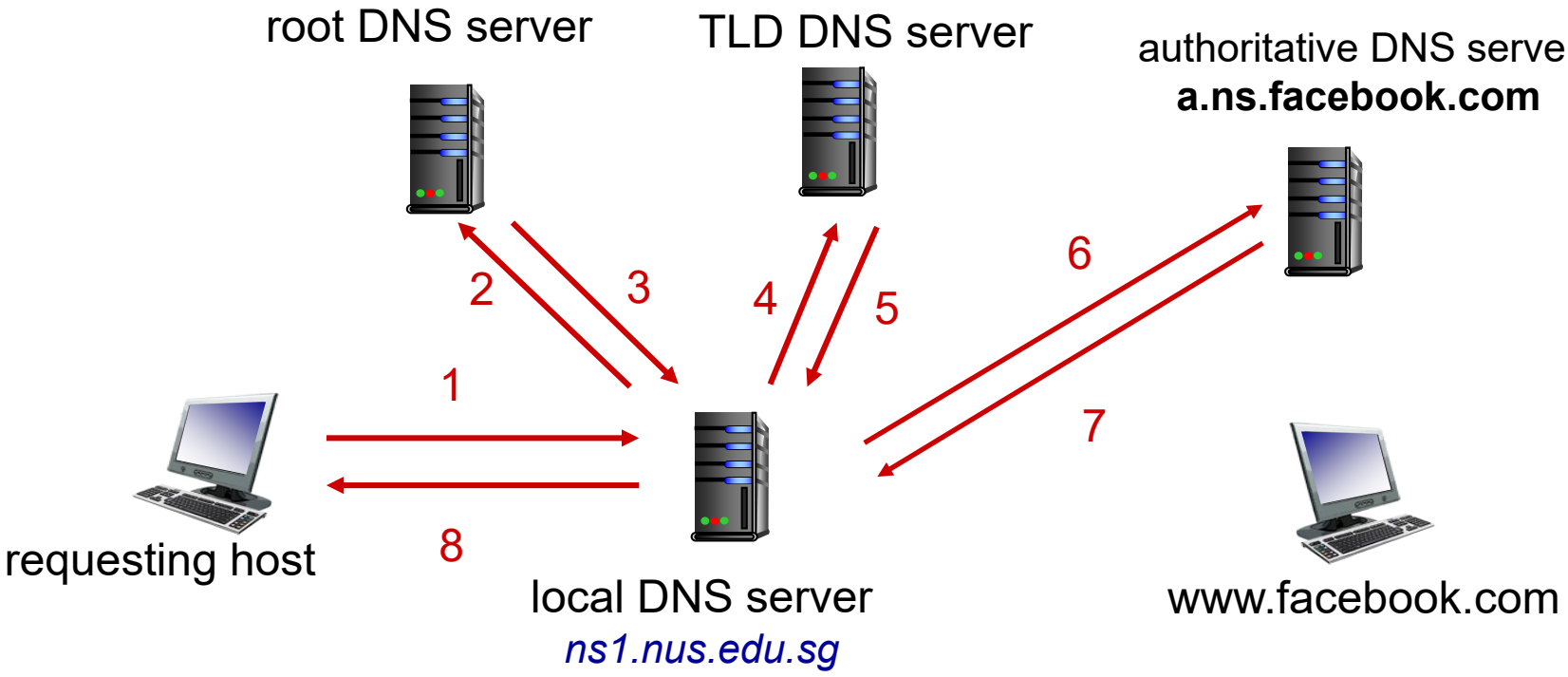
Local DNS Server

- Does not strictly belong to hierarchy
- Each ISP (residential ISP, company, university) has one local DNS server.
  - also called “default name server”
- When host makes a DNS query, query is sent to its local DNS server
  - Retrieve name-to-address translation from local cache
  - Local DNS server acts as proxy and forwards query into hierarchy if answer is not found locally

DNS Caching

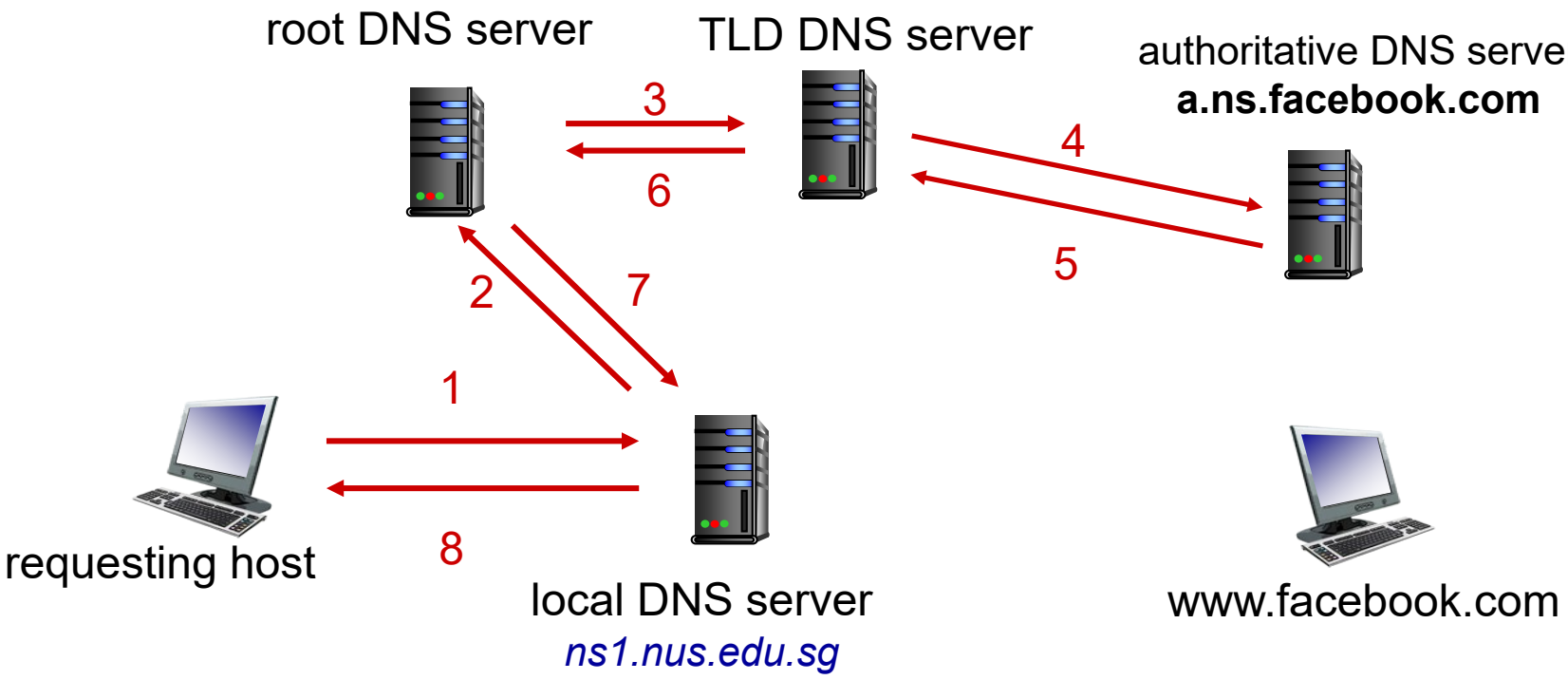
- Once a name server learns mapping, it *caches* mapping.
  - cached entries may be out-of-date (best effort name-to-address translation!)
  - cached entries expire after some time (TTL).
  - if name host changes IP address, may not be known Internet-wide until all TTLs expire
- Update/notify mechanisms proposed IETF standard
  - RFC 2136
- DNS runs over **UDP**.

DNS Name Resolution



- This is known as *iterative query*.

DNS Name Resolution



- This is known as *recursive query*.
  - rarely happens in practice

Lectures 2&3: Roadmap

2.1 Principles of Network Applications

2.2 Web and HTTP

2.4 DNS

2.7 Socket programming

Processes

- Process: program running within a host.
  - Within the same host, two processes communicate using inter-process communication (defined by OS).
  - Processes in different hosts communicate by exchanging messages (according to protocols).

In C/S model

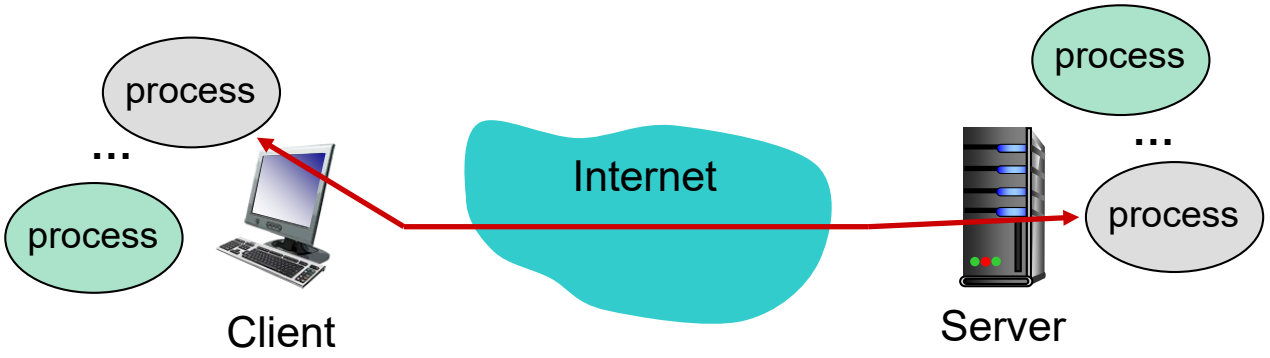
server process waits to be contacted

client process initiates the communication

Addressing Processes

- IP address is used to identify a host device
  - A 32-bit integer (e.g. 137.132.21.27)
- Question: is IP address of a host suffice to identify a process running inside that host?

A: no, many processes may run concurrently in a host.



Addressing Processes

- A process is identified by (IP address, port number).
  - Port number is 16-bit integer (1-1023 are reserved for standard use).
- Example port numbers
  - HTTP server: 80
  - SMTP server: 25
- IANA coordinates the assignment of port number:
  - http://www.iana.org/assignments/service-names-port-numbers/service-names-port-numbers.xhtml

Analogy

Postal service:

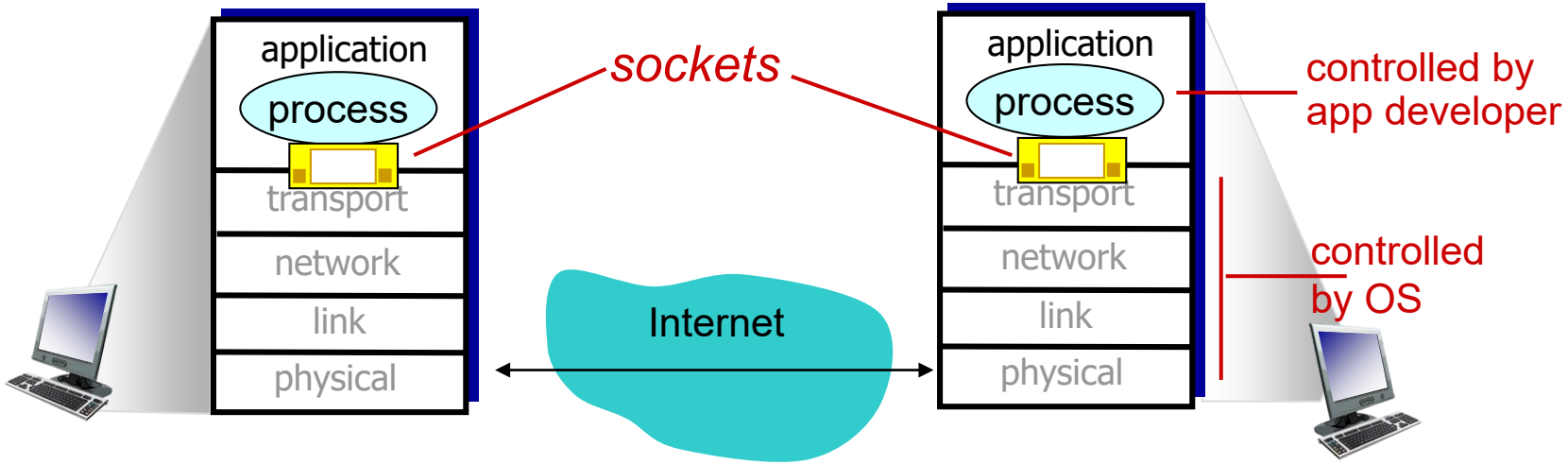
- *deliver letter to the doorstep:* home address
- *dispatch letter to the right person in the house:* name of the receiver as stated on the letter

Protocol service:

- *deliver packet to the right host:* IP address of the host
- *dispatch packet to the right process in the host:* port number of the process

Sockets

- Socket is the software interface between app processes and transport layer protocols.
  - Process sends/receives messages to/from its socket.
  - Programming-wise: a set of APIs



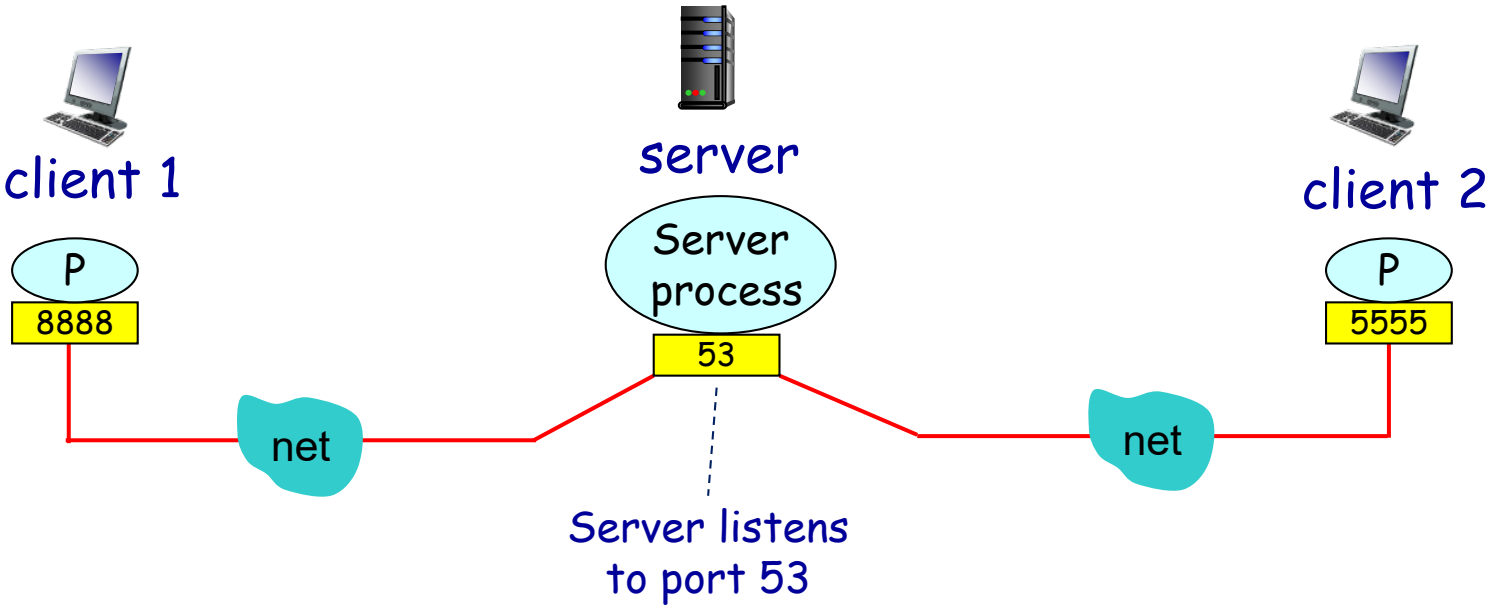
Socket Programming

- Applications (or processes) treat the Internet as a black box, sending and receiving messages through sockets.
- Two types of sockets
  - TCP: reliable, byte stream-oriented socket
  - UDP: unreliable datagram socket
- Now let's write a simple client/server application that client sends a line of text to server, and server echoes it.
  - We will demo both TCP socket version and UDP socket version.

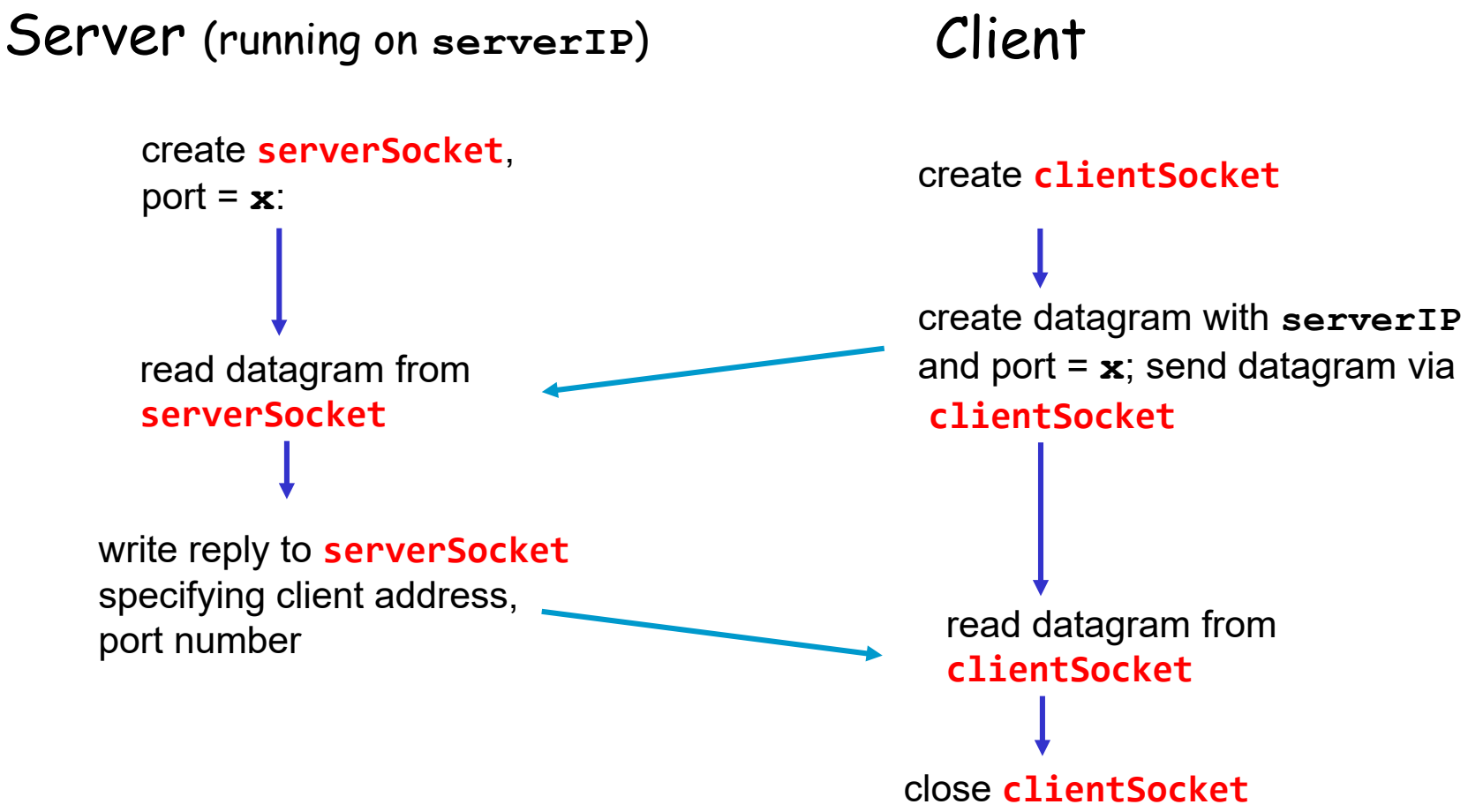
Socket Programming with *UDP*

UDP: no "connection" between client and server

- Sender (client) explicitly attaches destination IP address and port number to each packet.
- Receiver (server) extracts sender IP address and port number from the received packet.



UDP: Client/server Socket Interaction



Example: UDP Echo Server

```
from socket import *
serverPort = 2105

# create a socket
serverSocket = socket(AF_INET, SOCK_DGRAM)

# bind socket to local port number 2105
serverSocket.bind(('', serverPort))
print('Server is ready to receive message')

# extract client address from received packet
message, clientAddress = serverSocket.recvfrom(2048)

serverSocket.sendto(message, clientAddress)

serverSocket.close()
```

include Python's socket library

IPv4

UDP socket

receive datagram
buffer size: 2048B

Example: UDP Echo Client

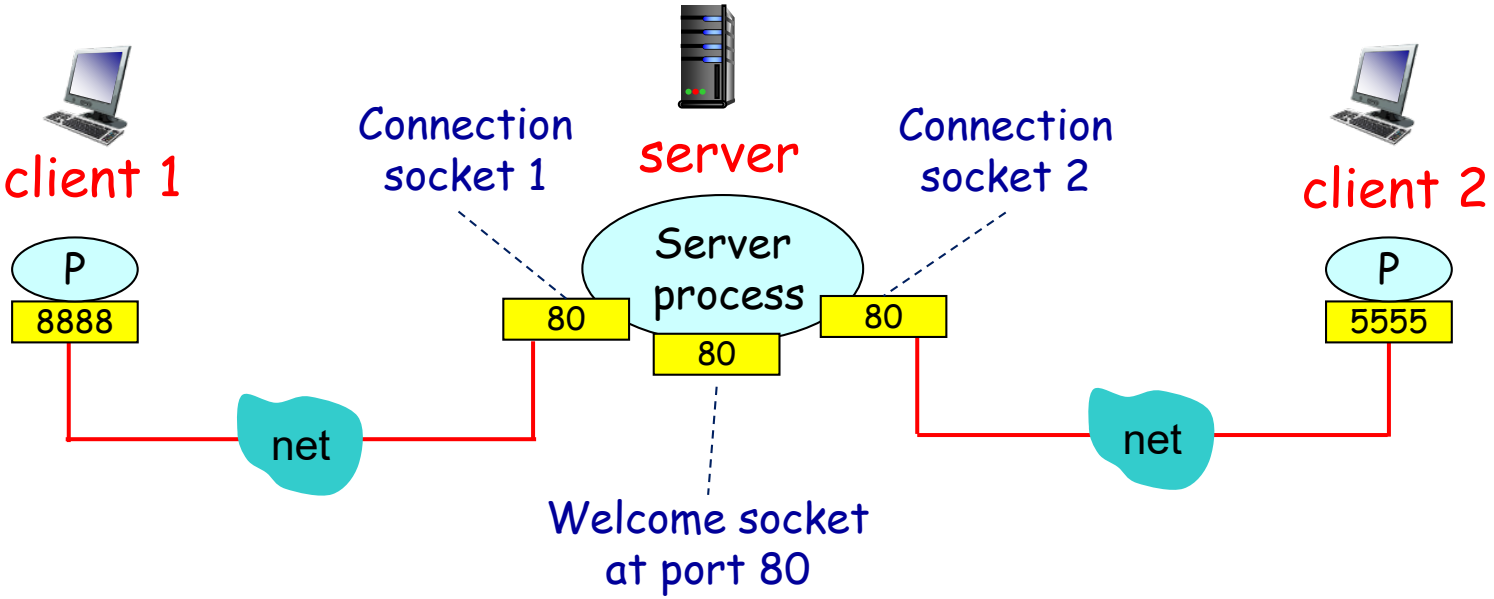
```
from socket import *

serverName = 'localhost' # client, server on the same host
serverPort = 2105

clientSocket = socket(AF_INET, SOCK_DGRAM)

message = input('Enter a message: ')

# send msg to server address
clientSocket.sendto(message.encode(), (serverName, serverPort))

receivedMsg, serverAddress = clientSocket.recvfrom(2048)

print('from server:', receivedMsg.decode())

clientSocket.close()
```

convert message from string to byte and send it

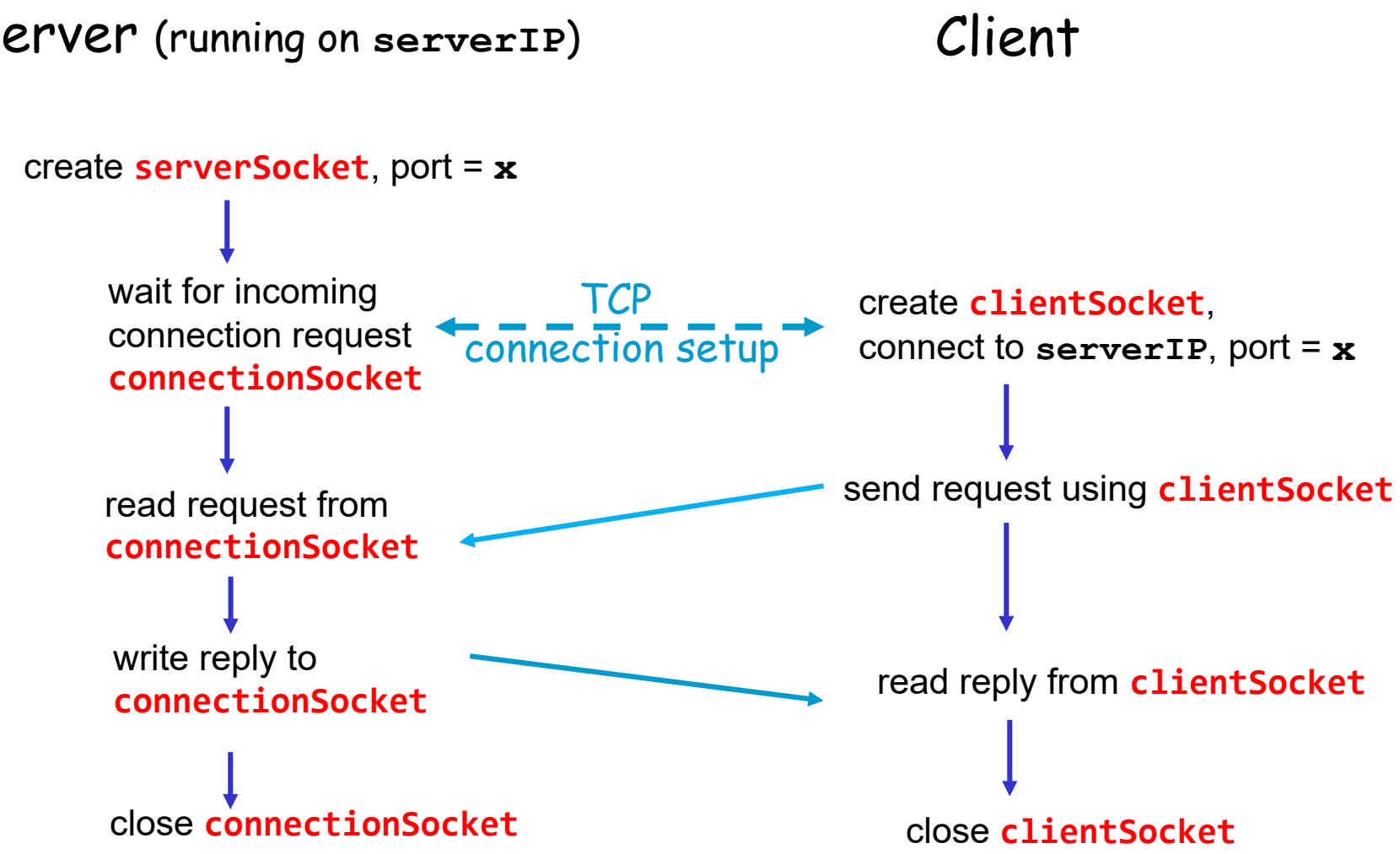
convert from byte to string

Socket Programming with *TCP*

- When client creates socket, client TCP establishes a connection to server TCP.
- When contacted by client, server TCP creates a new socket for server process to communicate with that client.
  - allows server to talk with multiple clients individually.



TCP: Client/server Socket Interaction



Example: TCP Echo Server

```
from socket import *

serverPort = 2105

serverSocket = socket(AF_INET, SOCK_STREAM)

serverSocket.bind(('', serverPort))

serverSocket.listen(1)
print('Server is ready to receive message')

connectionSocket, clientAddr = serverSocket.accept()
message = connectionSocket.recv(2048)

connectionSocket.send(message)

connectionSocket.close()
```

TCP socket

listens for incoming TCP request (not available in UDP socket)

returns a *new* socket to communicate with client socket

Example: TCP Echo Client

```
from socket import *

serverName = 'localhost'
serverPort = 2105

clientSocket = socket(AF_INET, SOCK_STREAM)
clientSocket.connect((serverName, serverPort))

message = input('Enter a message: ')

clientSocket.send(message.encode())

receivedMsg = clientSocket.recv(2048)

print('from server:', receivedMsg.decode())

clientSocket.close()
```

establish a connection

no need to attach server name, port

TCP Socket vs. UDP Socket

- In TCP, two processes communicate as if there is a pipe between them. The pipe remains in place until one of the two processes closes it.
  - When one of the processes wants to send more bytes to the other process, it simply writes data to that pipe.
  - The sending process doesn't need to attach a destination IP address and port number to the bytes in each sending attempt as the logical pipe has been established (which is also reliable).
- In UDP, programmers need to form UDP datagram packets explicitly and attach destination IP address / port number to every packet.

Lectures 2&3: Summary

- Application architectures
  - Client-server
  - P2P
- Application service requirements:
  - reliability, throughput, delay, security
- Specific protocols:
  - HTTP
  - DNS
- Internet transport service model
  - connection-oriented, reliable: TCP
  - Connection-less, unreliable: UDP

Lectures 2&3: Summary

- Socket programming
  - TCP socket
    - When contacted by client, server TCP creates new socket.
    - Server uses (client IP + port #) to distinguish clients.
    - When client creates its socket, client TCP establishes connection to server TCP.
  - UDP socket
    - Server uses one socket to serve all clients.
    - No connection is established before sending data.
    - Sender explicitly attaches destination IP address and port # to each packet.
    - Transmitted data may be lost or received out-of-order.

CS2105

An Awesome Introduction to Computer Networks

Lectures 4&5: The Transport Layer

NUS National University of Singapore
Department of Computer Science
School of Computing

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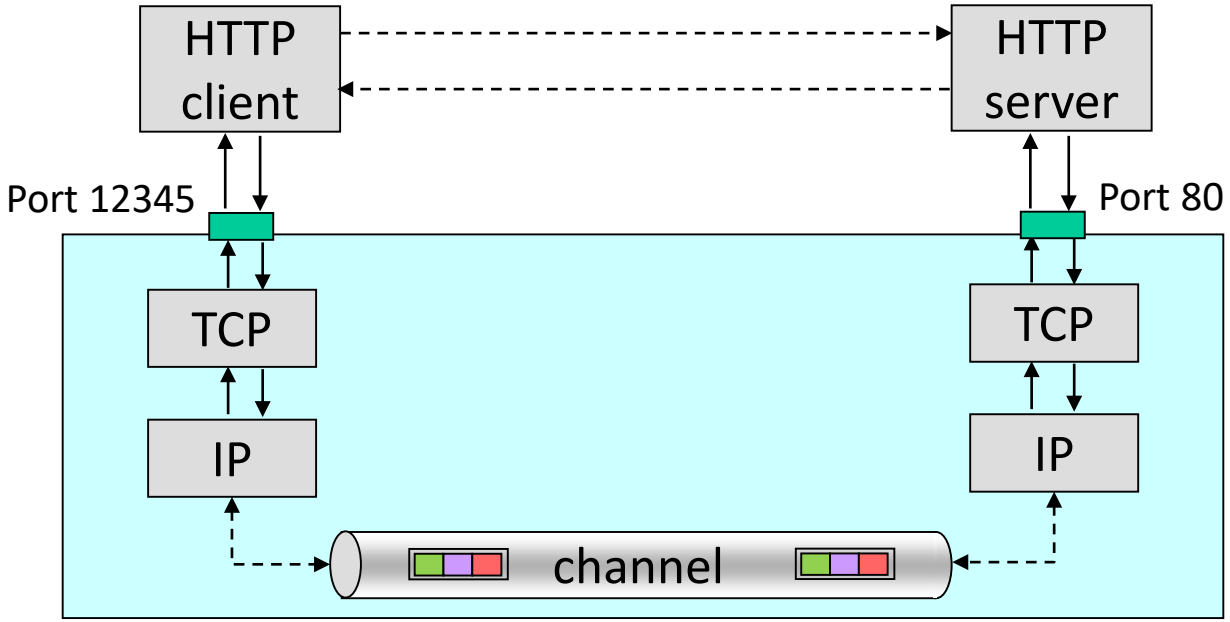
Web and HTTP

- A Web page consists of a *base HTML file* and *some other objects* referenced by the HTML file.
- HTTP uses TCP as transport service.
  - TCP, in turn, uses service provided by IP!



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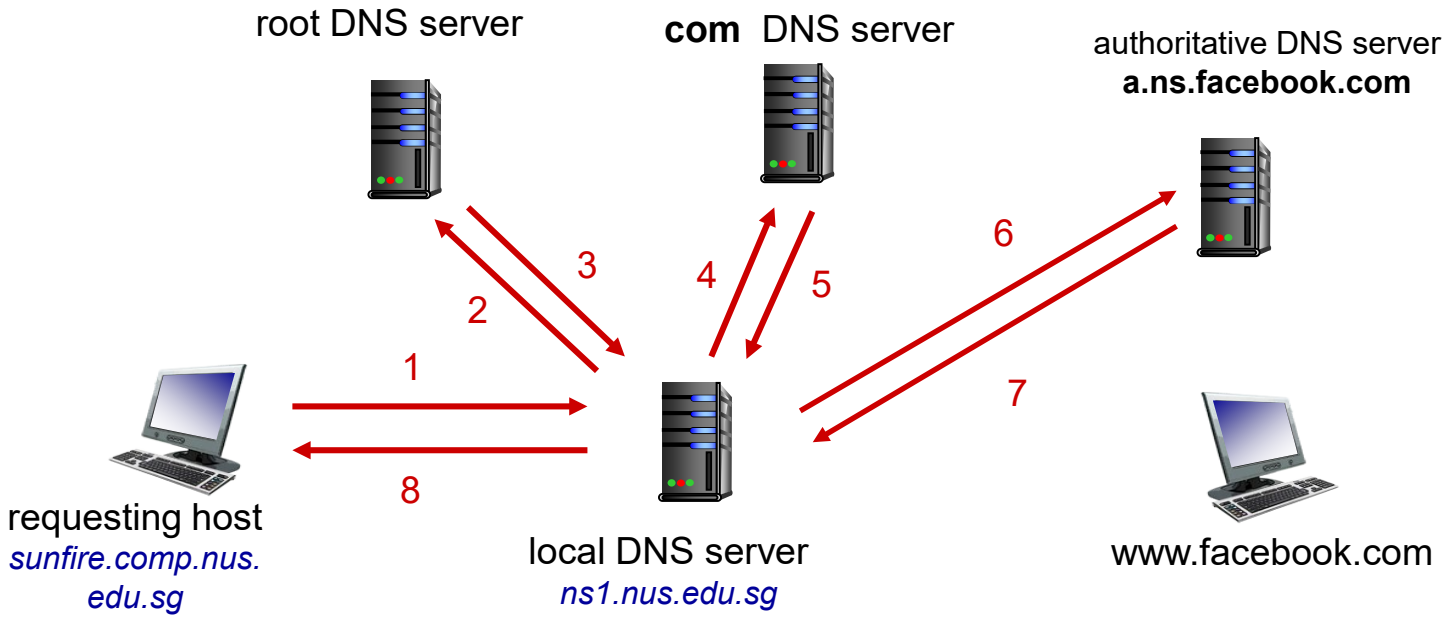
PREVIOUS LECTURE

Domain Name System

- DNS is the Internet's primary directory service.
  - It translates host names, which can be easily memorized by humans, to numerical IP addresses used by hosts for the purpose of communication.



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Socket

- Applications (processes) send messages over the network through sockets.
  - Conceptually, socket = IP address + port number
  - Programming wise, socket = a set of APIs

- UDP socket
  - Server uses one socket to serve all clients.
  - No connection is established before sending data.
  - Sender explicitly attaches destination IP address + port #.

- TCP socket
  - Server creates a new socket for each client.
  - Client establishes connection to server.
  - Server uses connection to identify client.

Lectures 4&5: The Transport Layer

After this class, you are expected to:

- appreciate the simplicity of UDP and the service it provides.
- know how to calculate the checksum of a packet.
- be able to design your own reliable protocols with *ACK*, *NAK*, *sequence number*, *timeout* and *retransmission*.
- understand the working of *Go-Back-N* and *Selective Repeat* protocols.
- understand the operations of TCP.

Lectures 4&5: Roadmap

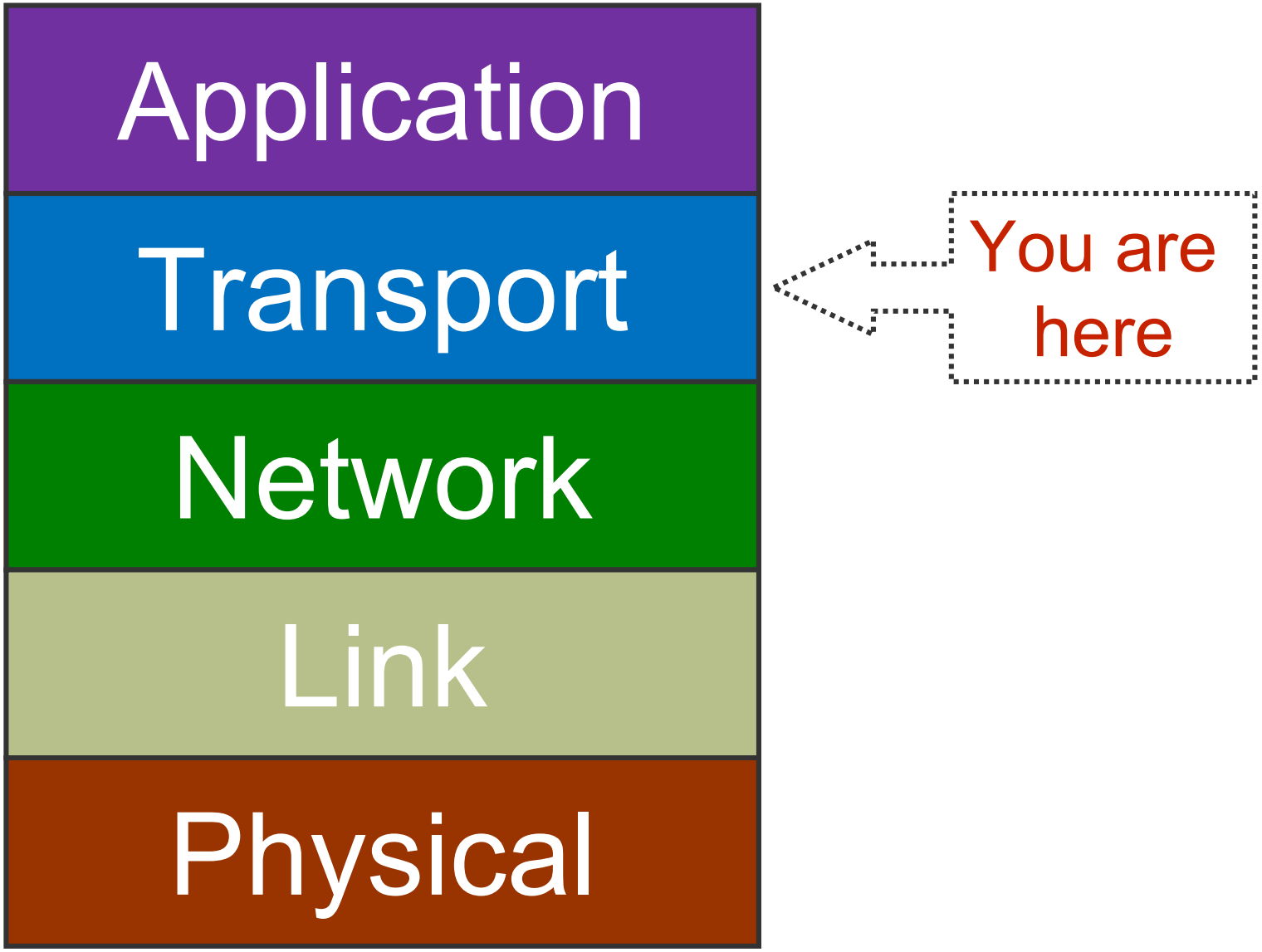
3.1 Transport-layer Services

3.3 Connectionless Transport: UDP

3.4 Principles of Reliable Data Transfer

3.5 Connection-oriented Transport: TCP

Kurose Textbook, Chapter 3
(Some slides are taken from the book)


Application
Transport
Network
Link
Physical
You are here

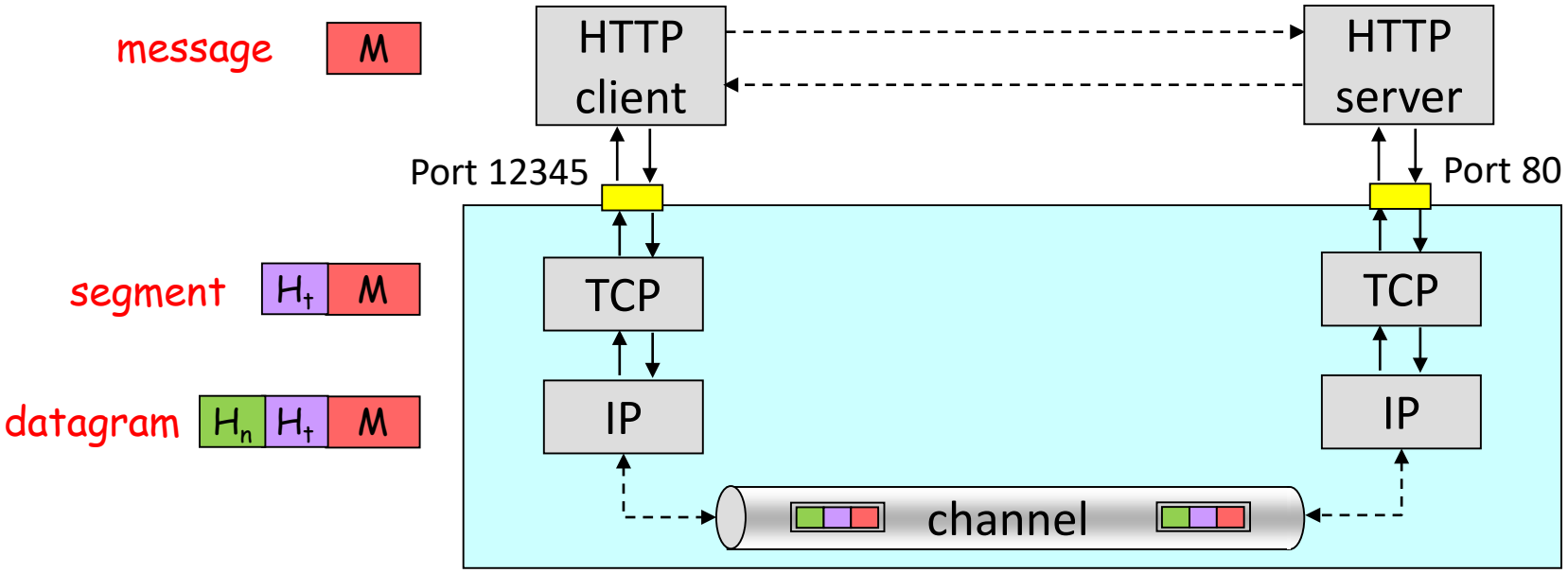
Transport Layer Services

- Deliver messages between application processes running on different hosts
  - Two popular protocols: TCP and UDP

- Transport layer protocols run in hosts.
  - Sender side: breaks app message into *segments* (as needed), passes them to network layer (aka IP layer).
  - Receiver side: reassembles segments into message, passes it to app layer.
  - Packet switches (routers) in between: only check destination IP address to decide routing.

Transport / Network Layers

- Each IP datagram contains source and dest IP addresses.
  - Receiving host is identified by dest IP address.
  - Each IP datagram carries one transport-layer segment.
  - Each segment contains source and dest port numbers.



Lectures 4&5: Roadmap

3.1 Transport-layer Services

3.3 Connectionless Transport: UDP

3.4 Principles of Reliable Data Transfer

3.5 Connection-oriented Transport: TCP

UDP: User Datagram Protocol [RFC 768]

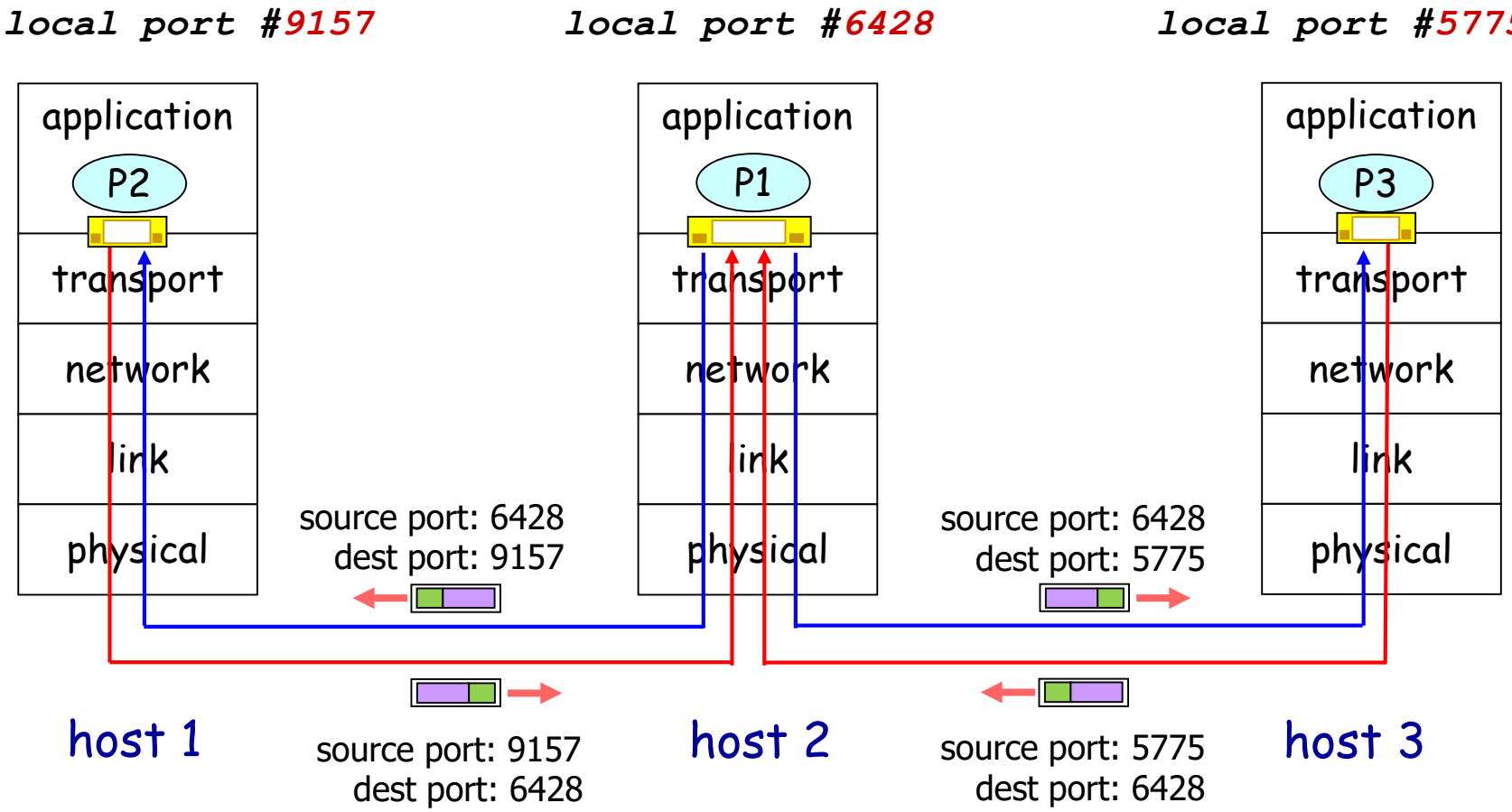
- UDP adds very little service on top of IP:
  - Connectionless multiplexing / de-multiplexing
  - Checksum
- UDP transmission is unreliable
  - Often used by streaming multimedia apps (loss tolerant & rate sensitive)
- To achieve reliable transmission over UDP
  - Application implements error detection and recovery mechanisms!

Connectionless De-multiplexing

- UDP sender:
  - Creates a socket with local port #.
  - When creating a datagram to send to UDP socket, sender must specify dest. IP address and port #.

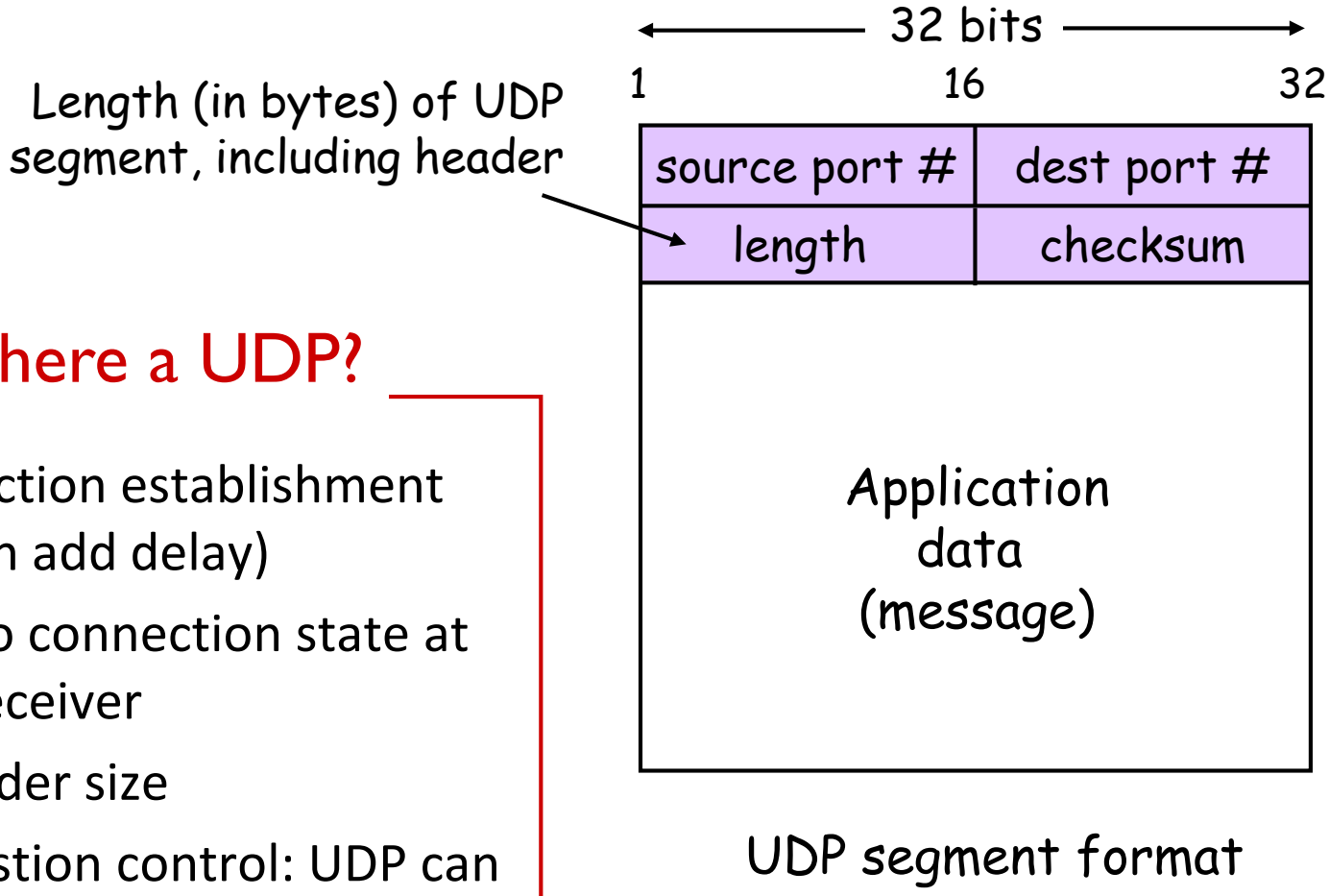
- When UDP receiver receives a UDP segment:
  - Checks destination port # in segment.
  - Directs UDP segment to the socket with that port #.
  - IP datagrams (from different sources) with the same destination port # will be directed to the same UDP socket at destination.

Connectionless De-multiplexing



IP datagrams (from different sources) with the same destination port # will be directed to the same UDP socket at destination.

UDP 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: to detect "errors" (i.e., flipped bits) in transmitted segment.

Sender:

- compute checksum value (next page)
- put 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 really no error?)

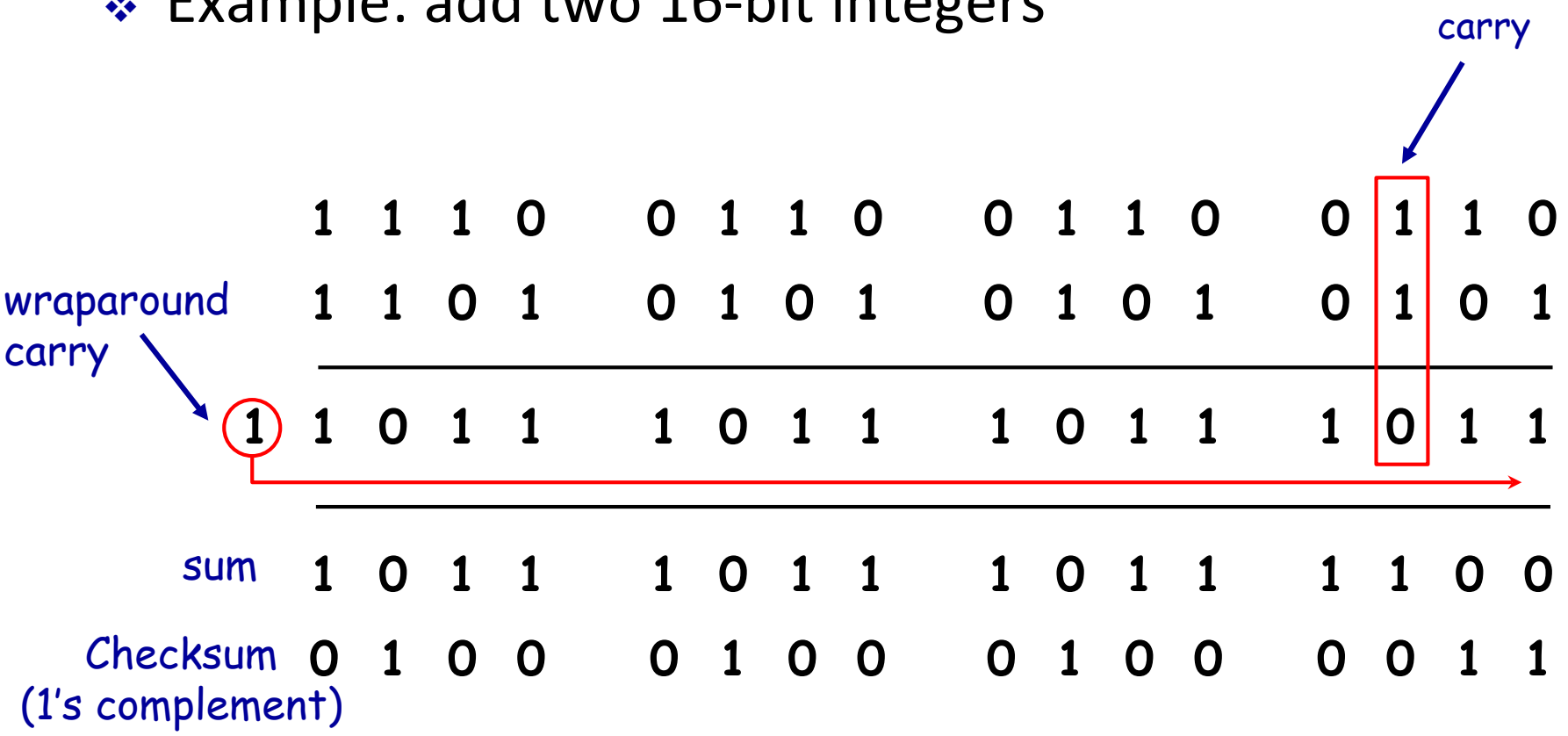
Checksum Computation

- How is UDP checksum computed?
  1. Treat UDP segment as a sequence of 16-bit integers.
  2. Apply binary addition on every 16-bit integer (checksum field is currently 0).
  3. Carry (if any) from the most significant bit will be added to the result.
  4. Compute 1's complement to get UDP checksum.

x	y	x ⊕ y	carry
0	0	0	-
0	1	1	-
1	0	1	-
1	1	0	1

Checksum Example

- Example: add two 16-bit integers



Lectures 4&5: Roadmap

3.1 Transport-layer Services

3.3 Connectionless Transport: UDP

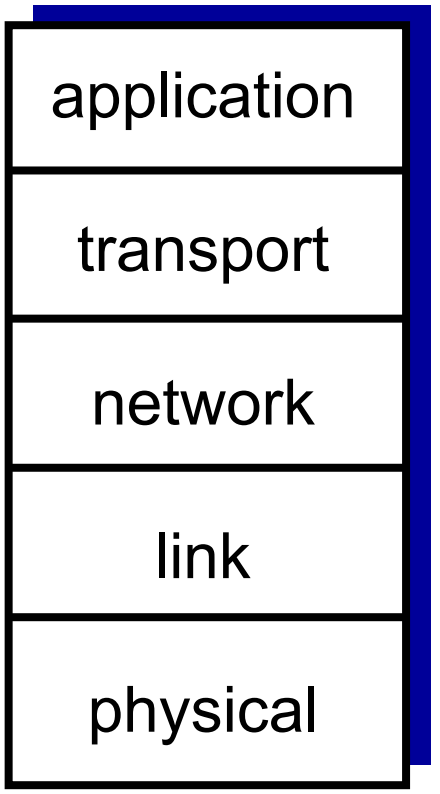
3.4 Principles of Reliable Data Transfer

3.5 Connection-oriented transport: TCP

"Sending Data Reliably Over the Internet is Much Harder Than You Think. The Intricacy Involved in Ensuring Reliability Will Make Your Head Explode."

Transport vs. Network Layer

- **Transport layer** resides on end hosts and provides process-to-process communication.
- **Network layer** provides host-to-host, best-effort and unreliable communication.



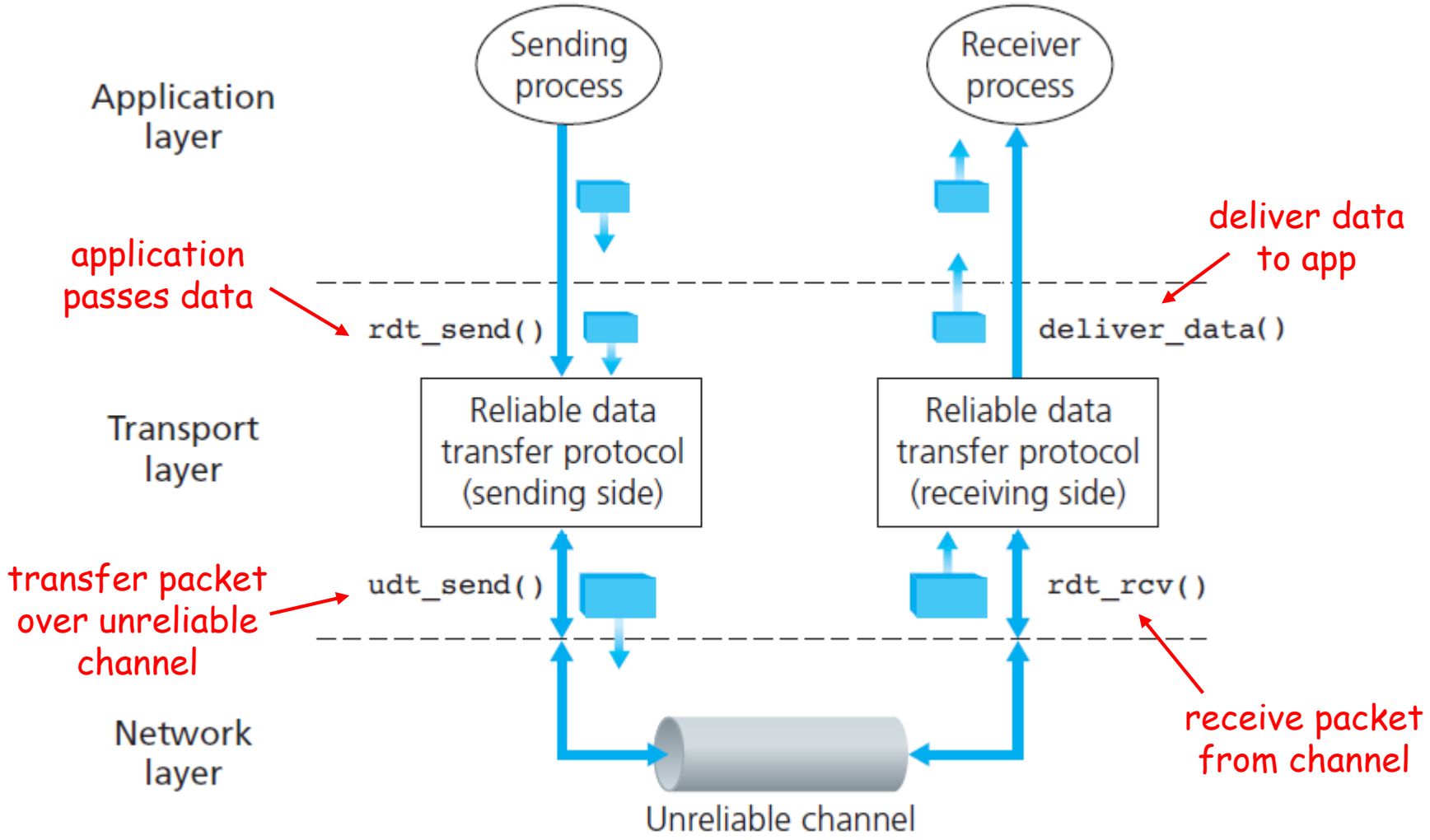
Question: How to build a **reliable transport layer protocol** on top of **unreliable communication**?

Reliable Transfer over Unreliable Channel

- Underlying network may
  - corrupt packets
  - drop packets
  - re-order packets (not considered in this lecture)
  - deliver packets after an arbitrarily long delay

- End-to-end reliable transport service should
  - guarantee packets delivery and correctness
  - deliver packets (to application) in the same order they are sent

Reliable Data Transfer: Service Model

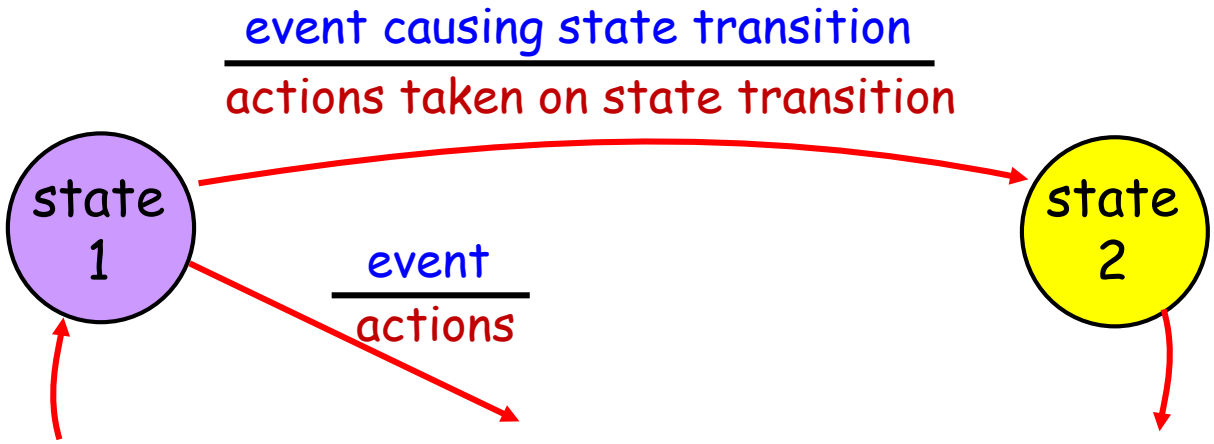


Reliable Data Transfer Protocols

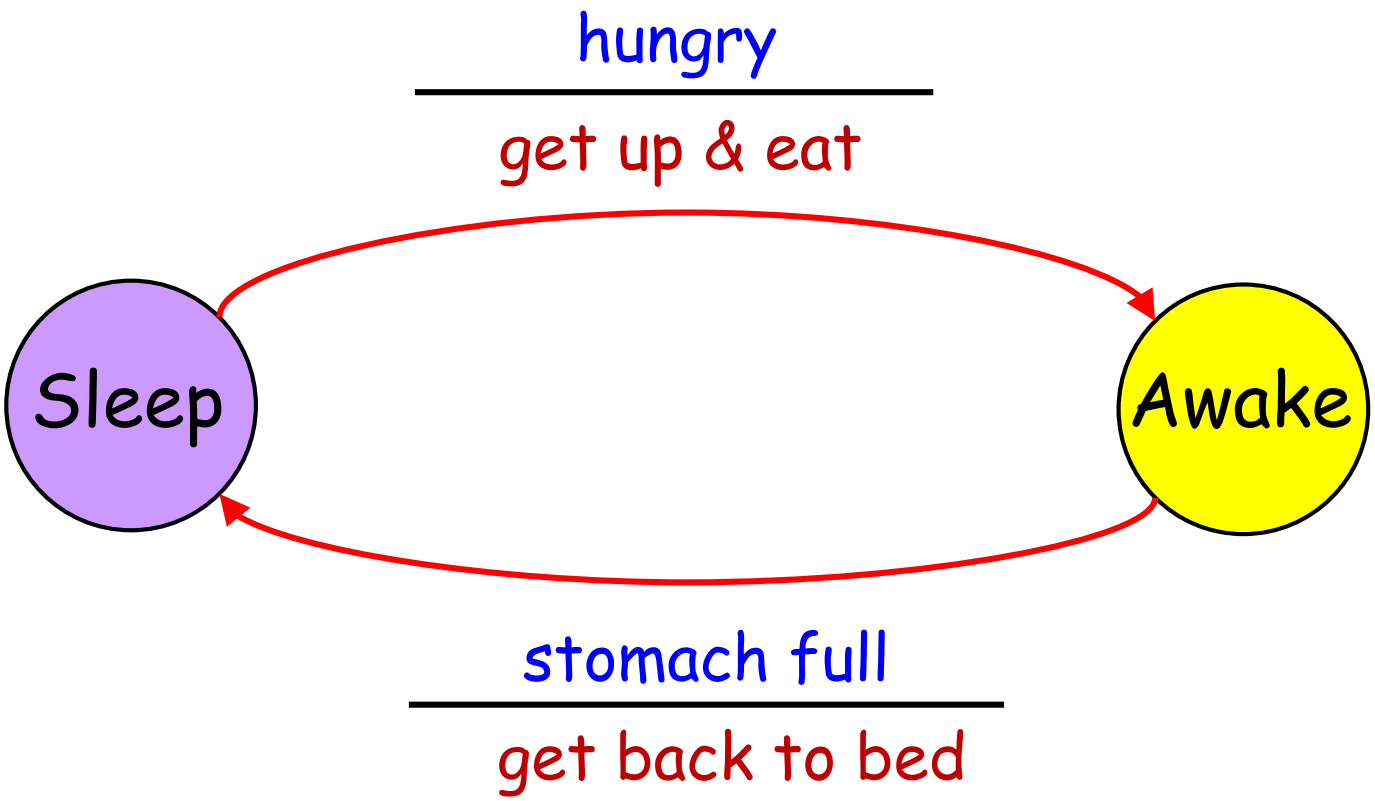
- Characteristics of unreliable channel will determine the complexity of reliable data transfer protocols (rdt).
- We will incrementally develop sender & receiver sides of rdt protocols, considering increasingly complex models of unreliable channel.
- We consider only unidirectional data transfer
  - but control info may flow in reverse direction!

Finite State Machine (FSM)

- We will use finite state machines (FSM) to describe sender and receiver of a protocol.
  - We will learn a protocol by examples, but FSM provides you the complete picture to refer to as necessary.



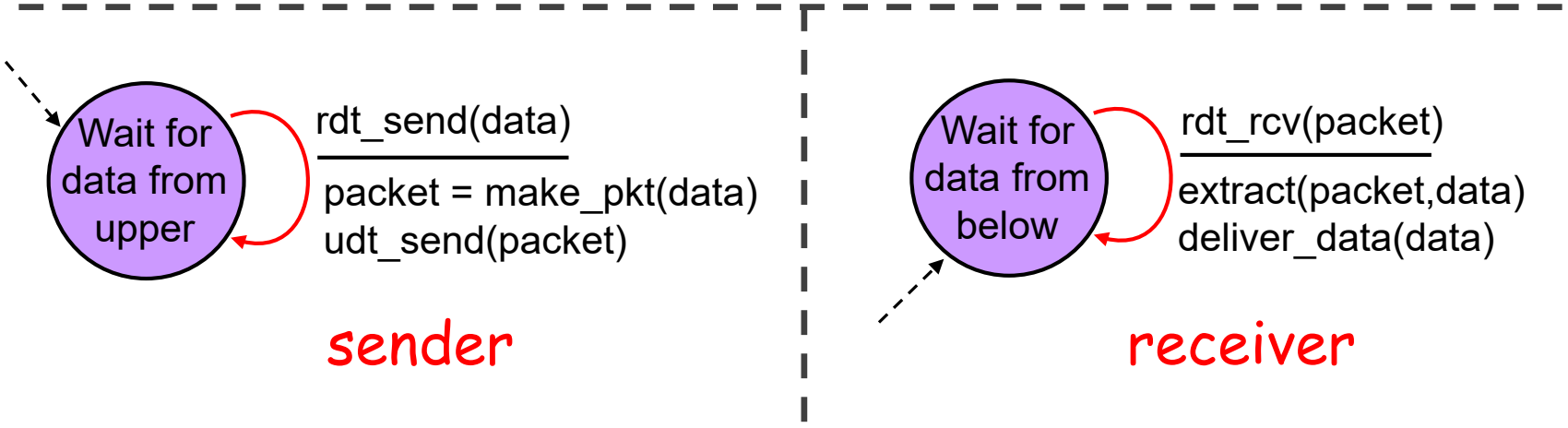
Example FSM



rdt 1.0: Perfectly Reliable Channel

- Assume underlying channel is perfectly reliable.

- Separate FSMs for sender, receiver:
  - Sender sends data into underlying (perfect) channel
  - Receiver reads data from underlying (perfect) channel

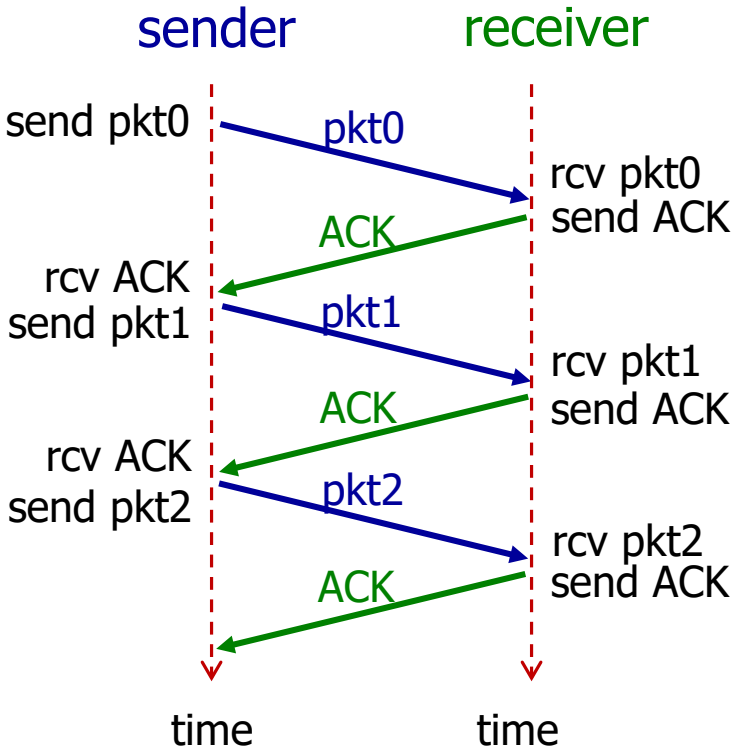


rdt 2.0: Channel with *Bit Errors*

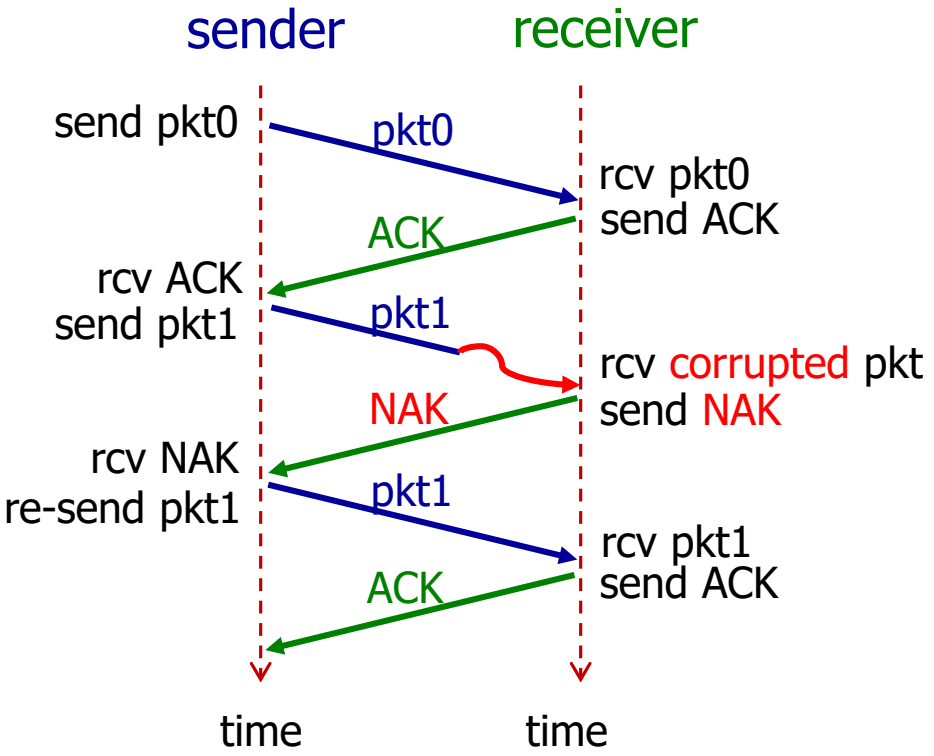
- Assumption:
  - underlying channel may flip bits in packets
  - other than that, the channel is perfect

- Receiver may use ***checksum*** to detect bit errors.
- Question: how to recover from bit errors?
  - *Acknowledgements (ACKs):* receiver explicitly tells sender that packet received is OK.
  - *Negative acknowledgements (NAKs):* receiver explicitly tells sender that packet has errors.
  - Sender retransmits packet on receipt of NAK.

rdt 2.0 In Action



(a) no bit error

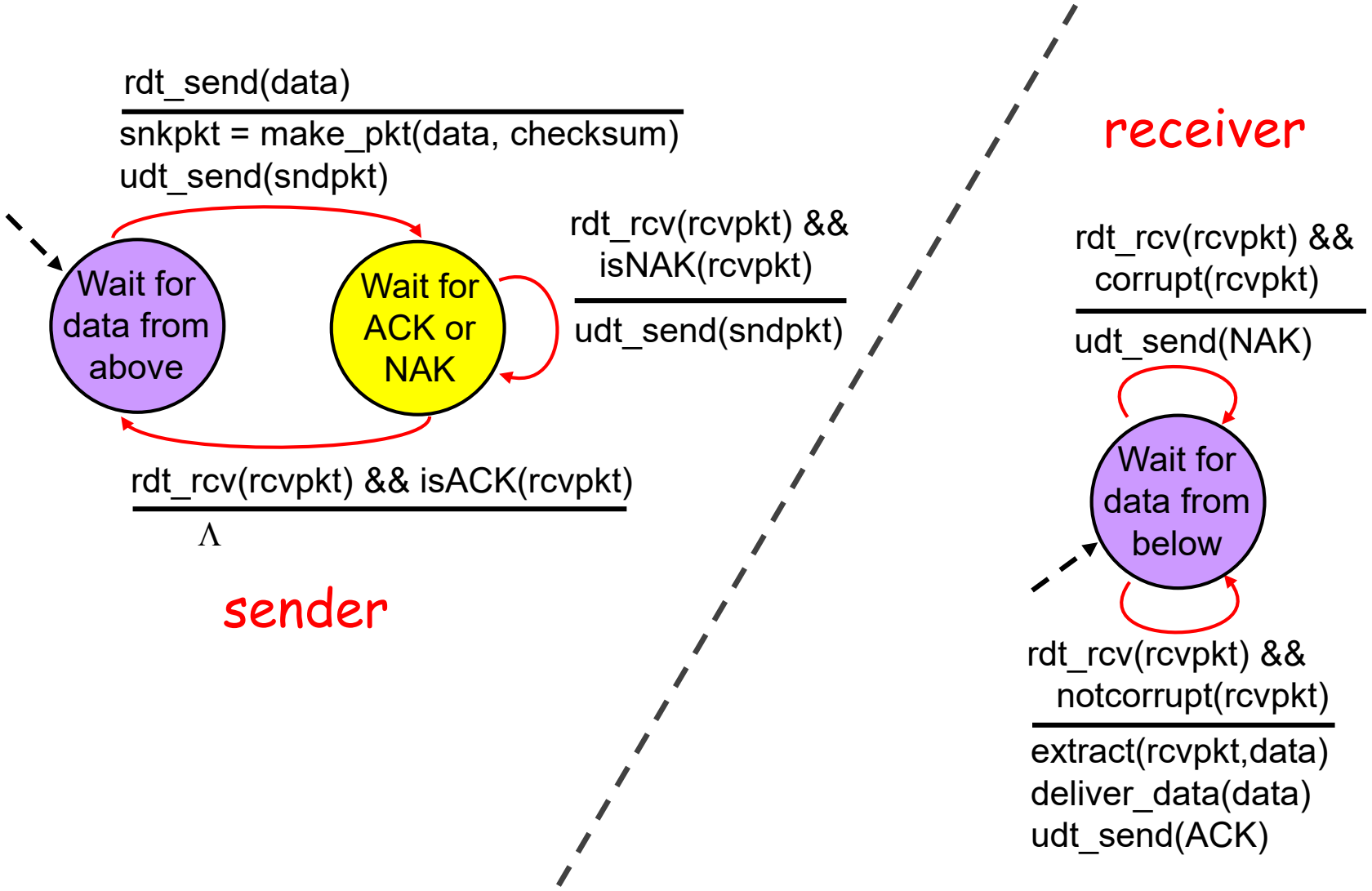


(b) with bit error

stop and wait protocol

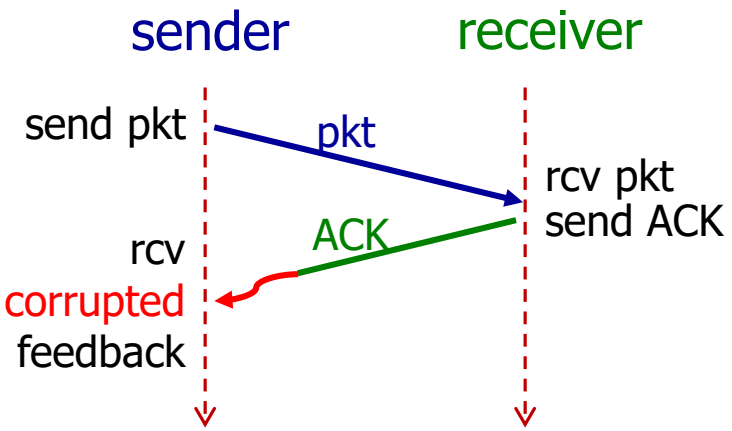
Sender sends one packet at a time, then waits for receiver response

rdt 2.0: FSM

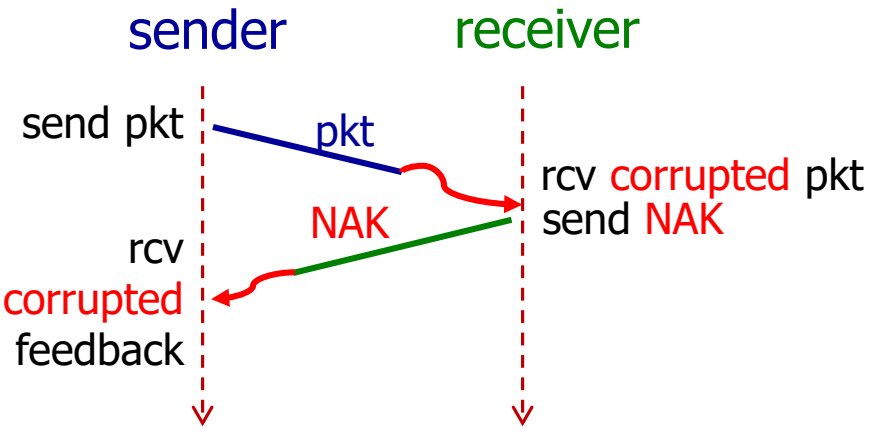

rdt_send(data)
snkpkt = make_pkt(data, checksum)
udt_send(sndpkt)
Wait for data from above
Wait for ACK or NAK
rdt_rcv(rcvpkt) && isNAK(rcvpkt)
udt_send(sndpkt)
rdt_rcv(rcvpkt) && isACK(rcvpkt)
Λ
sender
receiver
rdt_rcv(rcvpkt) && corrupt(rcvpkt)
udt_send(NAK)
Wait for data from below
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
extract(rcvpkt,data)
deliver_data(data)
udt_send(ACK)

rdt 2.0 has a Fatal Flaw!

- What happens if ACK/NAK is corrupted?
  - Sender doesn't know what happened at receiver!
- So what should the sender do?
  - Sender just retransmits when receives garbled ACK or NAK.
  - Questions: does this work?



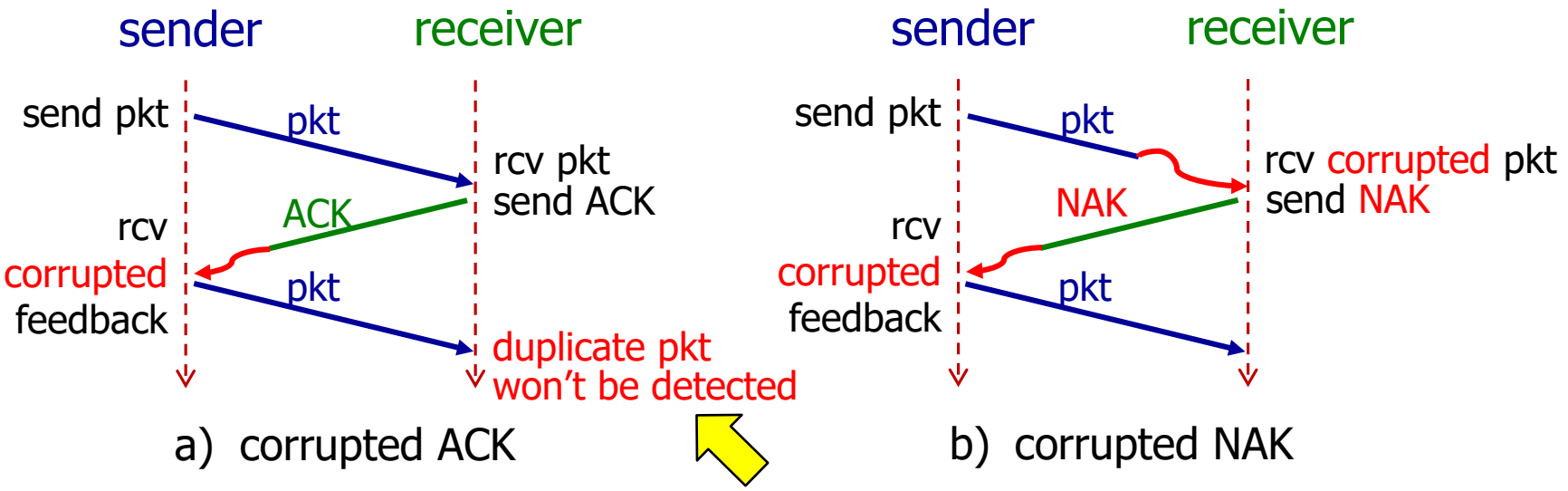
a) corrupted ACK



b) corrupted NAK

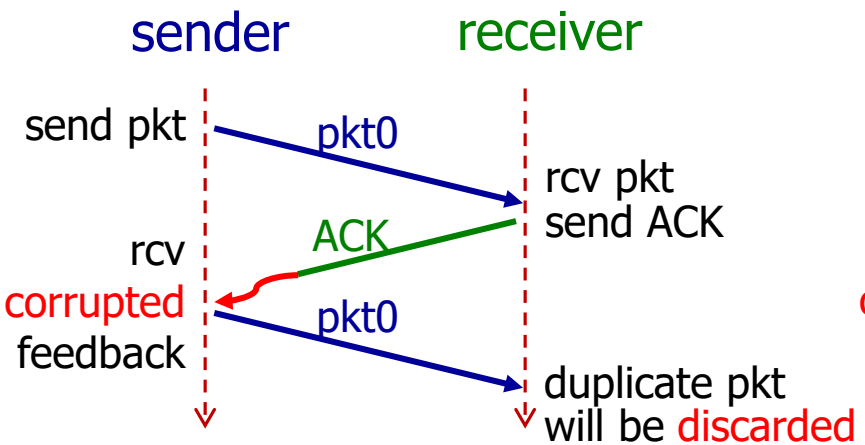
rdt 2.0 has a Fatal Flaw!

- Sender just retransmits when it receives garbled feedback.
  - This may cause retransmission of correctly received packet!
  - Question: how can receiver identify duplicate packet?

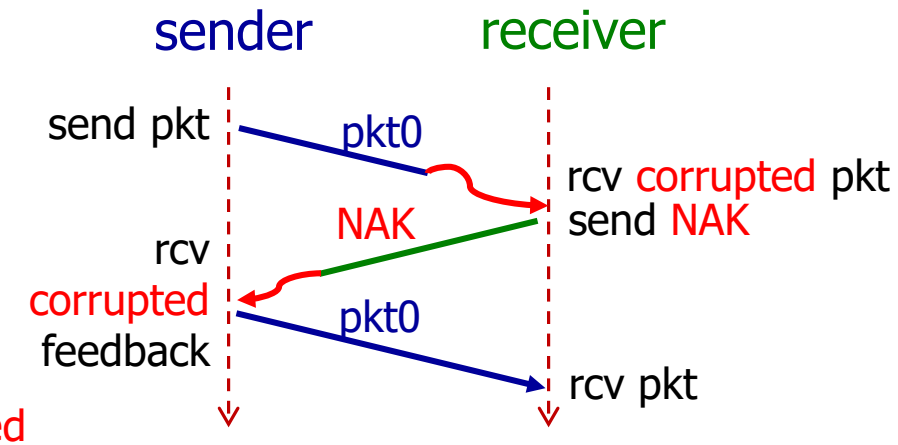


rdt 2.1: rdt 2.0 + Packet Seq.

- To handle duplicates:
  - Sender retransmits current packet if ACK/NAK is garbled.
  - Sender adds *sequence number* to each packet.
  - Receiver discards (doesn't deliver up) duplicate packet.
- This gives rise to protocol rdt 2.1.

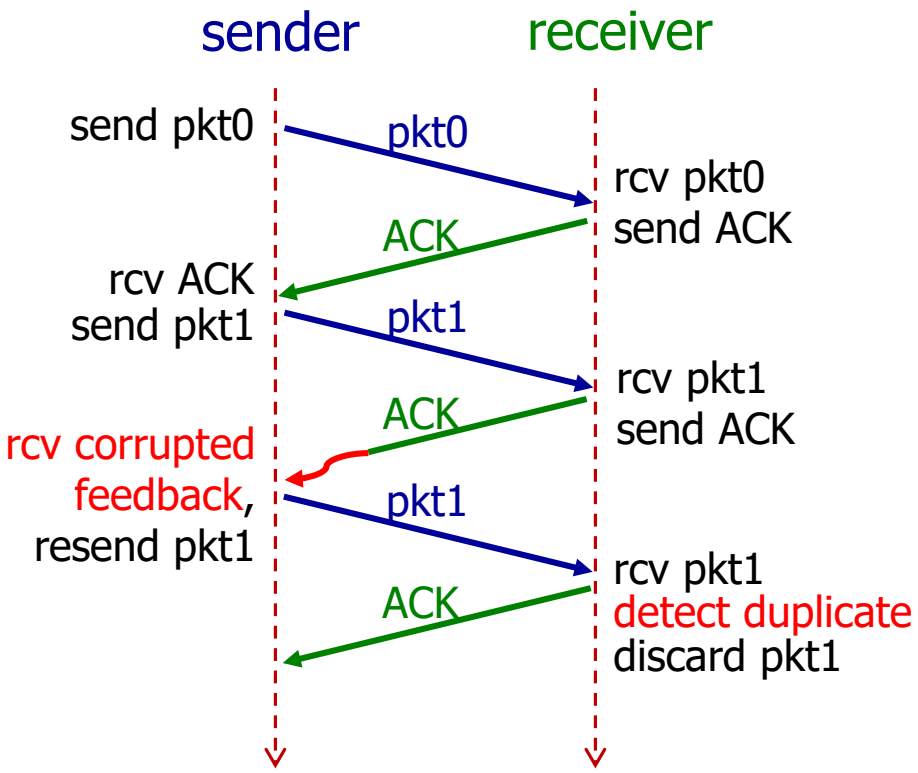


a) corrupted ACK

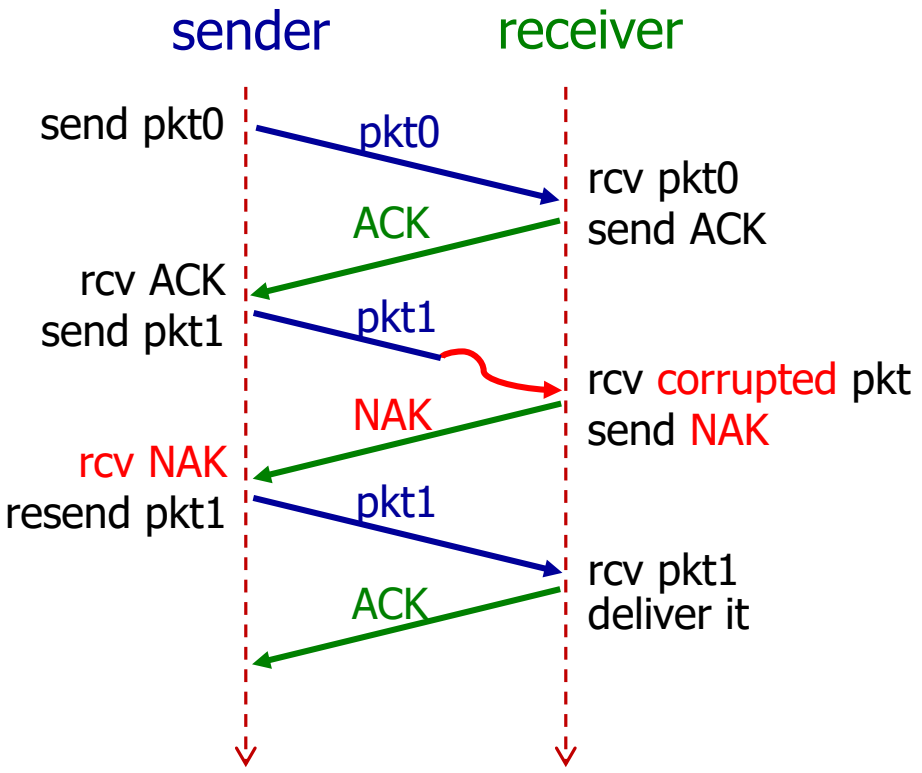


b) corrupted NAK

rdt 2.1 In Action

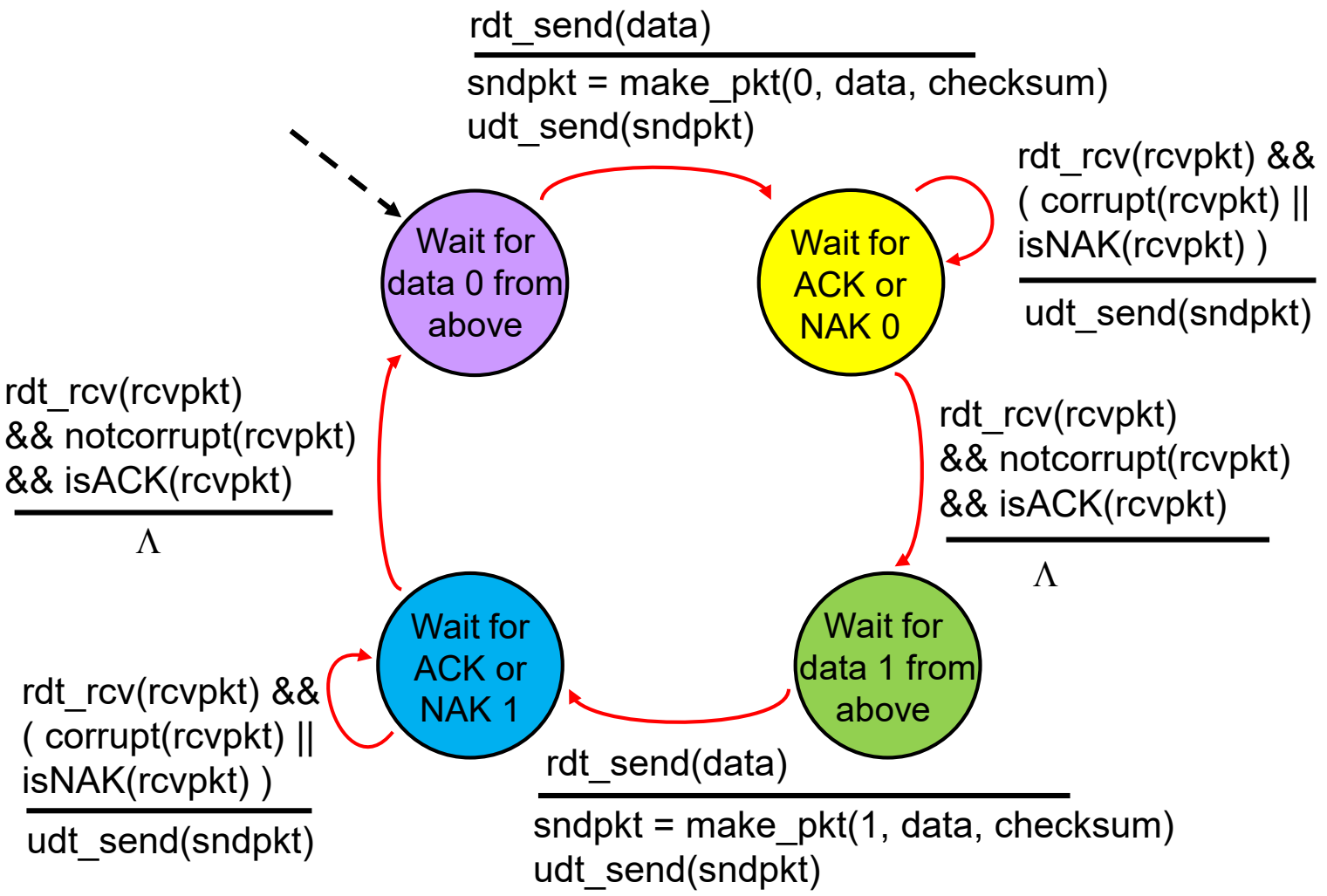


a) resend due to corrupted ACK

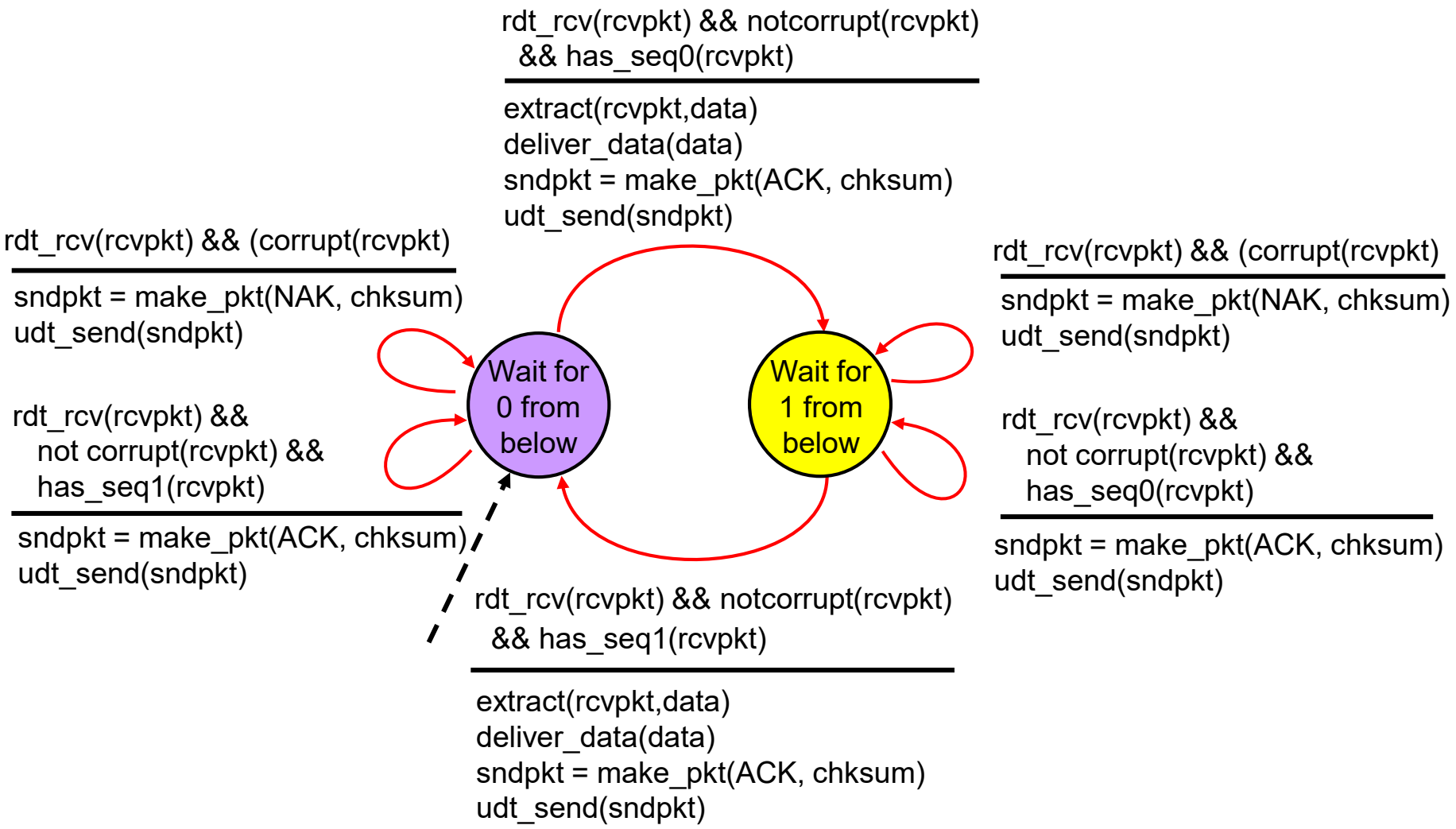


b) resend due to corrupted packet

rdt 2.1 Sender FSM



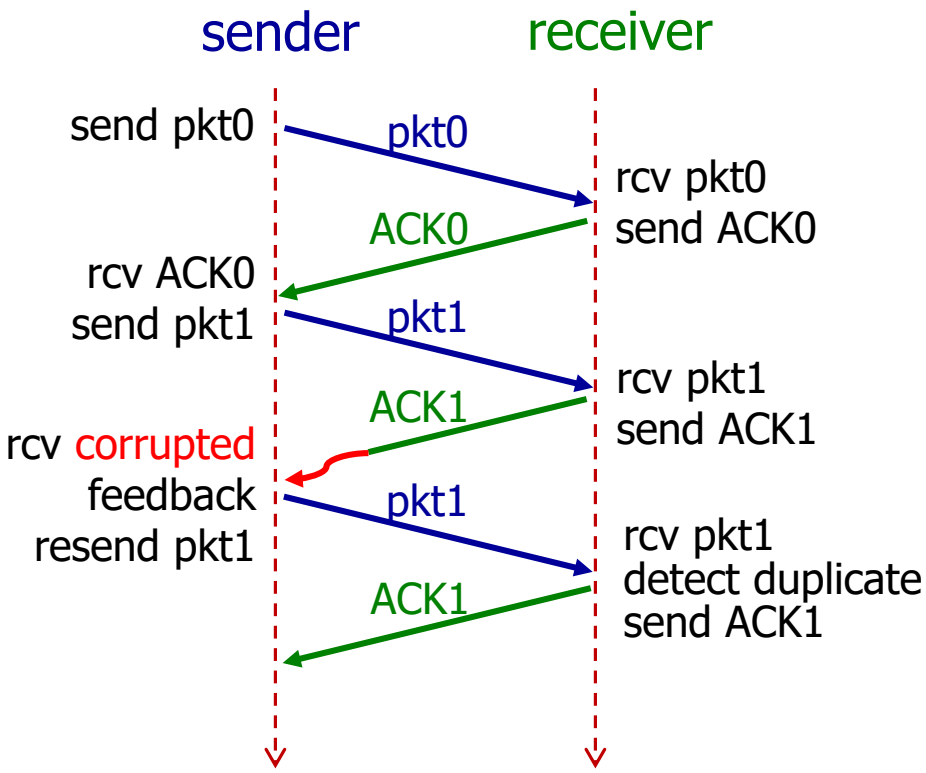
rdt 2.1 Receiver FSM



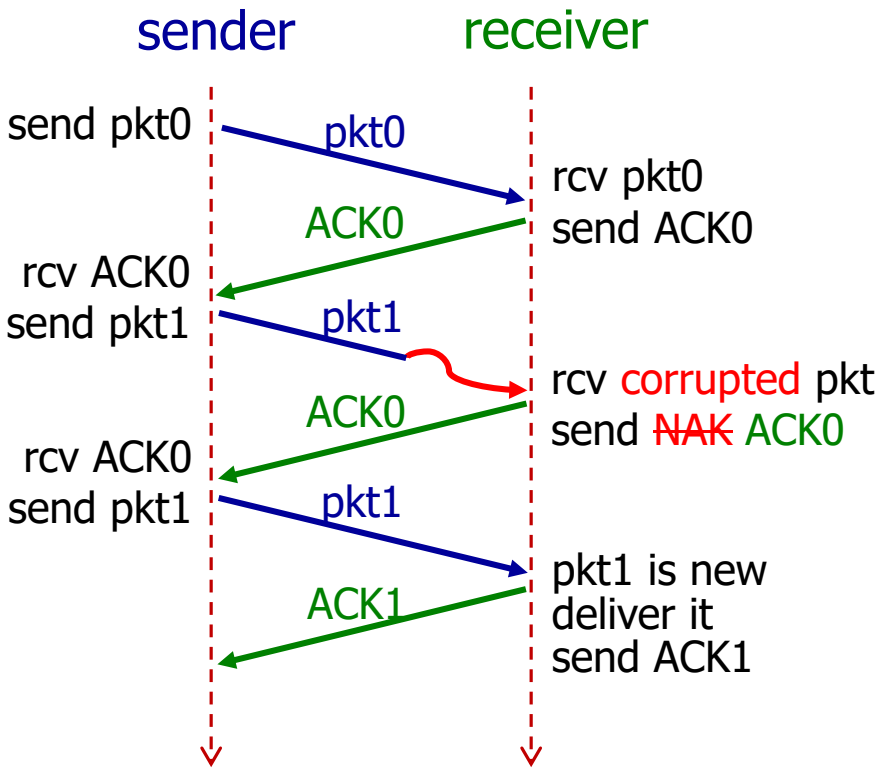
rdt 2.2: a NAK-free Protocol

- Same assumption and functionality as rdt 2.1, but use ACKs only.
- Instead of sending NAK, receiver sends ACK for the last packet received OK.
  - Now receiver must *explicitly* include seq. # of the packet being ACKed.
- Duplicate ACKs at sender results in same action as NAK: *retransmit current pkt*.

rdt 2.2 In Action



a) resend due to corrupted ACK



b) resend due to duplicate ACK

rdt 3.0: Channel with *Errors* and *Loss*

- Assumption: underlying channel
  - may flip bits in packets
  - may lose packets
  - may incur arbitrarily long packet delay
  - but won't re-order packets

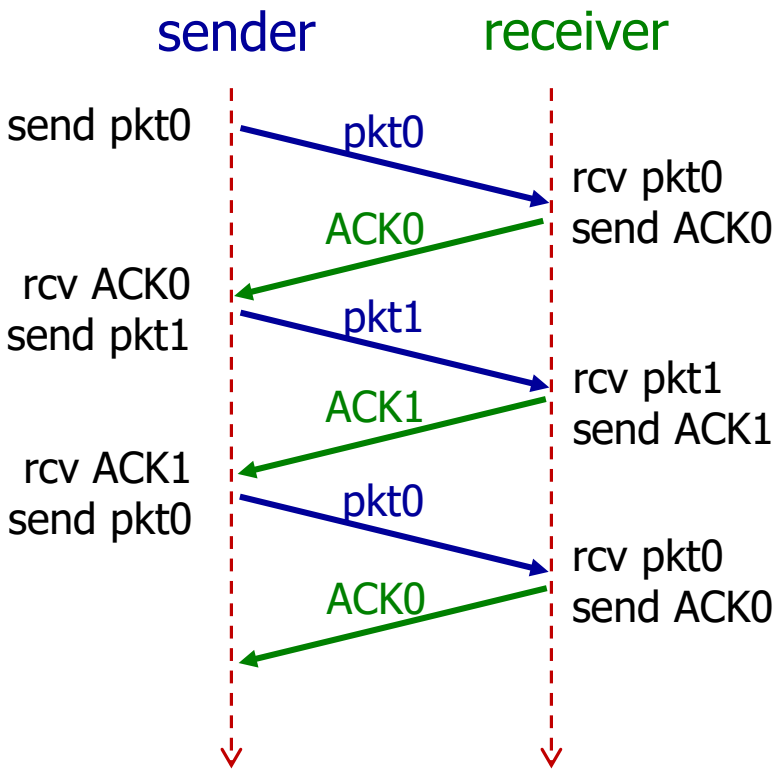
- Question: how to detect packet loss?
  - checksum, ACKs, seq. #, retransmissions will be of help… but not enough

rdt 3.0: Channel with *Errors* and *Loss*

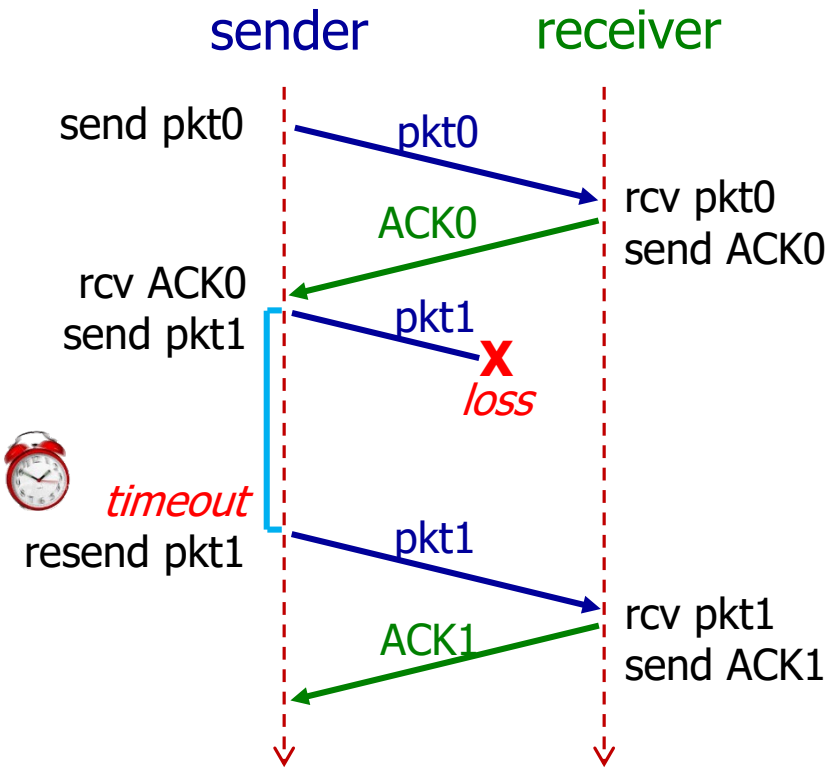
- To handle packet loss:
  - Sender waits "reasonable" amount of time for ACK.
  - Sender retransmits if no ACK is received till *timeout*.

- Question: what if packet (or ACK) is just delayed, but not lost?
  - Timeout will trigger retransmission.
  - Retransmission will generate duplicates in this case, but receiver may use seq. # to detect it.
  - Receiver must specify seq. # of the packet being ACKed (check scenario (d) two pages later).

rdt 3.0 In Action

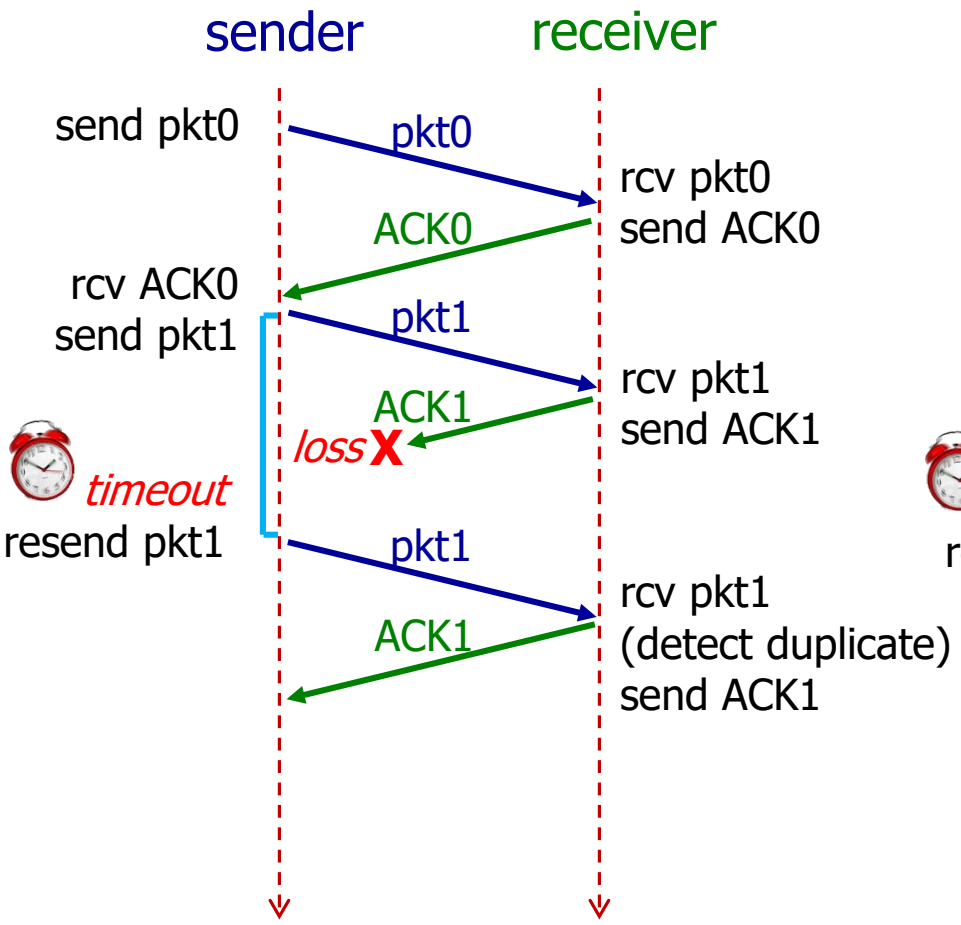


a) no packet loss

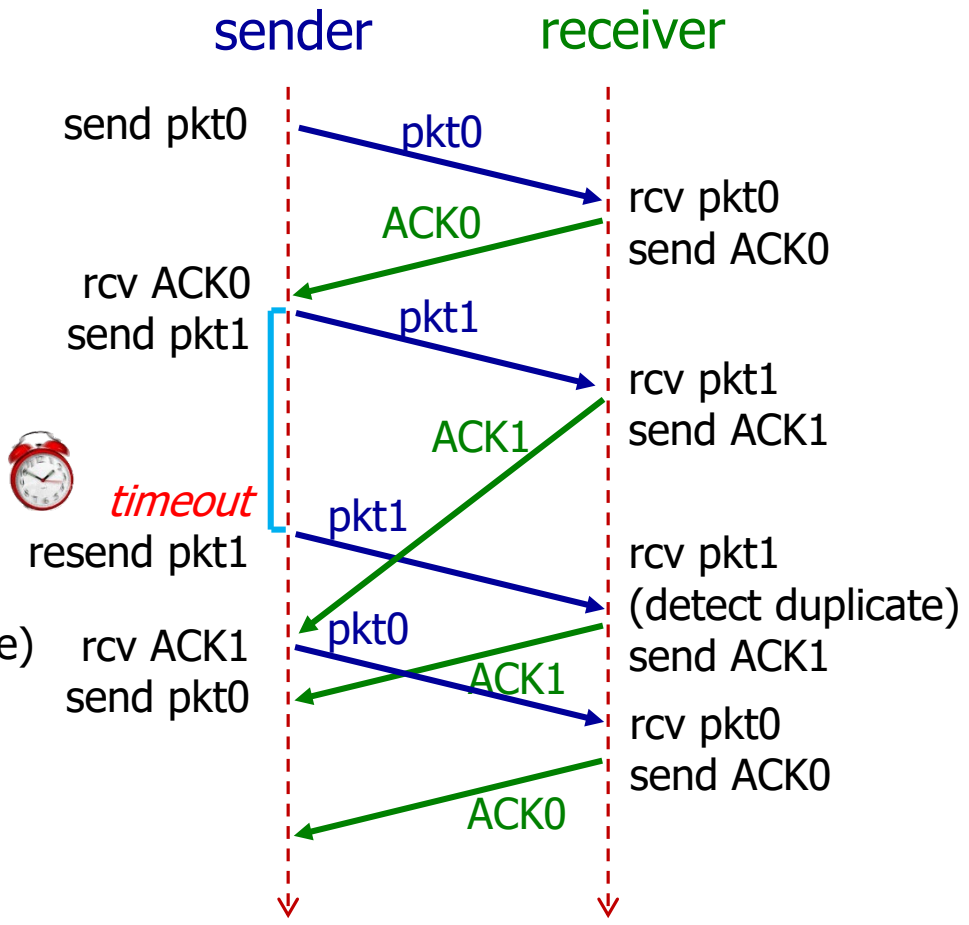


b) packet loss

rdt 3.0 In Action

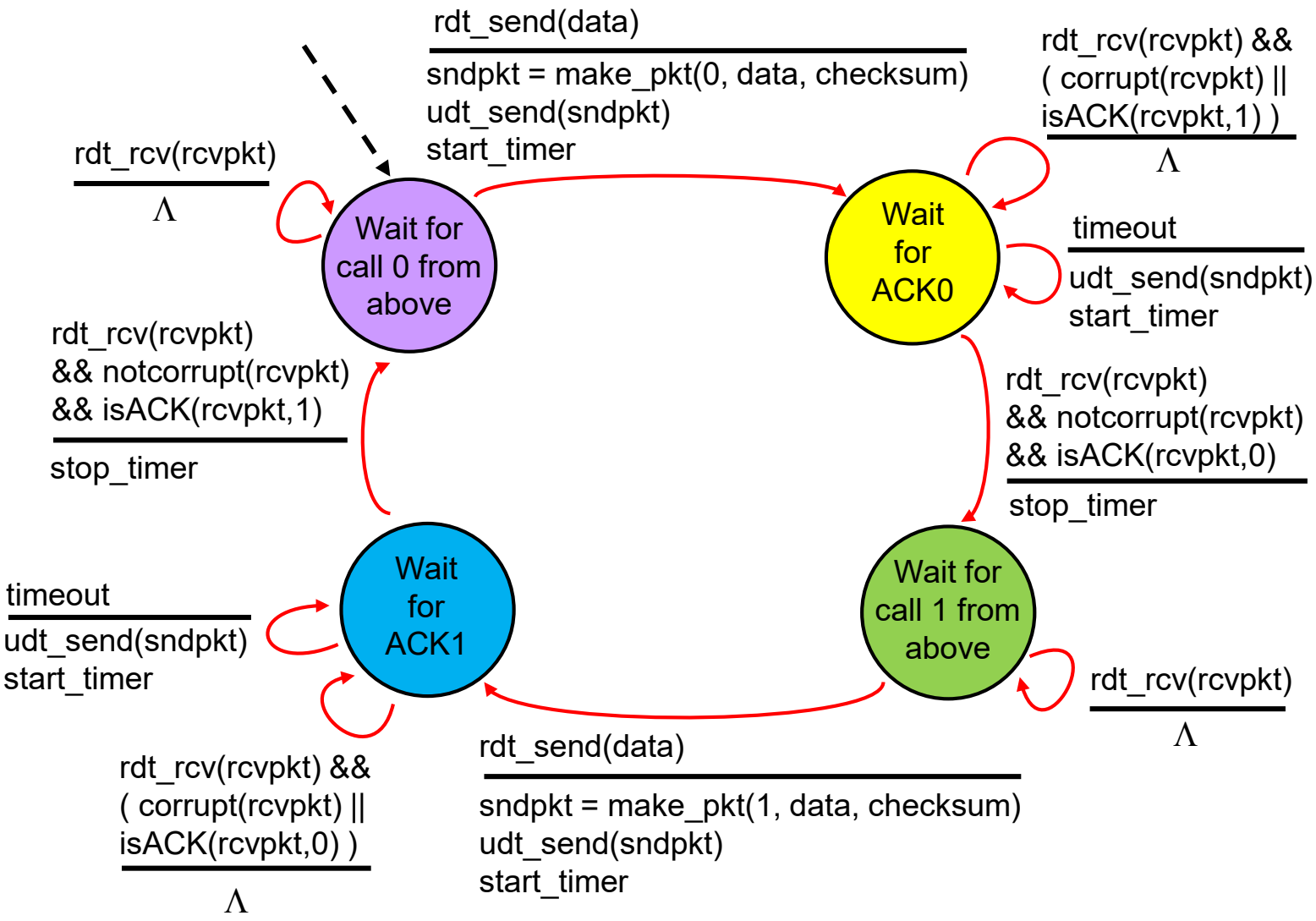


c) lost ACK



d) premature timeout / delayed ACK

rdt 3.0 Sender FSM



Performance of rdt 3.0

- rdt 3.0 works, but performance stinks.
- Example: packet size = 8000 bits, link rate = 1 Gbps:

$$d_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 0.008 \text{ msec}$$

  - If RTT = 30 msec, sender sends 8000 bits every 30.008 msec.

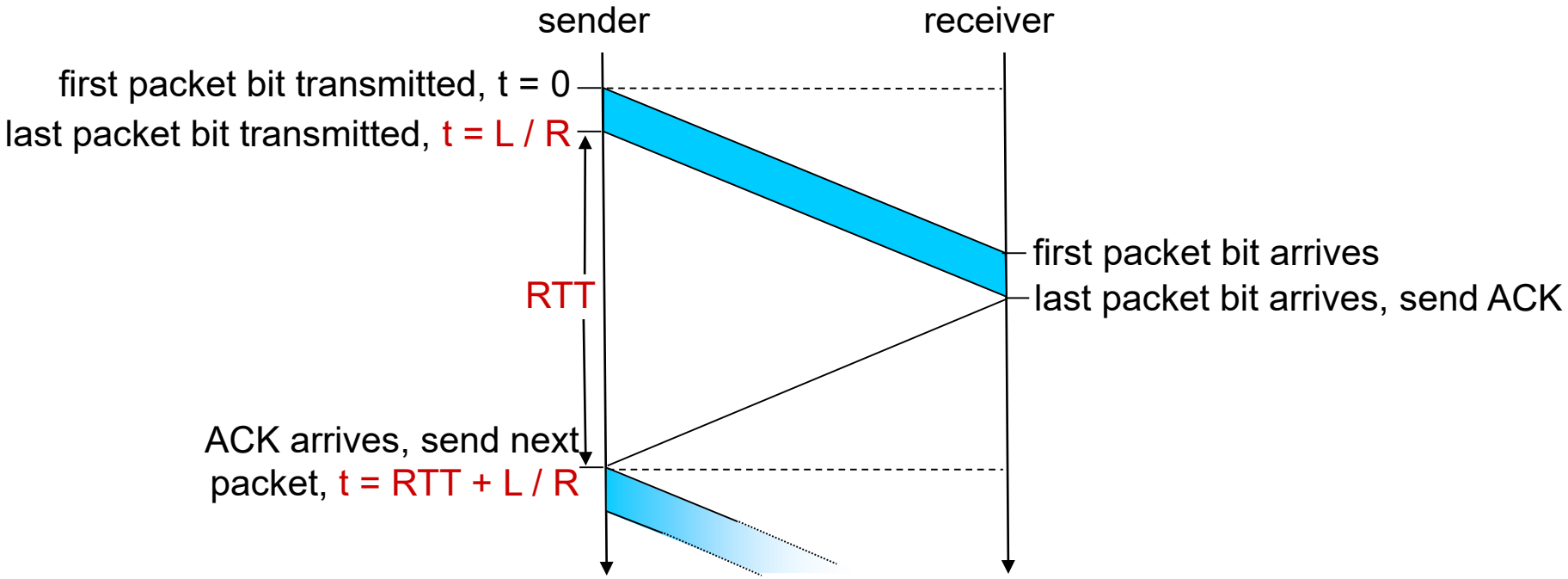
$$\text{throughput} = \frac{L}{RTT + d_{trans}} = \frac{8000}{30.008} = 267 \text{ kbps}$$

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

$$U_{sender} = \frac{d_{trans}}{RTT + d_{trans}} = \frac{0.008}{30 + 0.008} = 0.00027$$

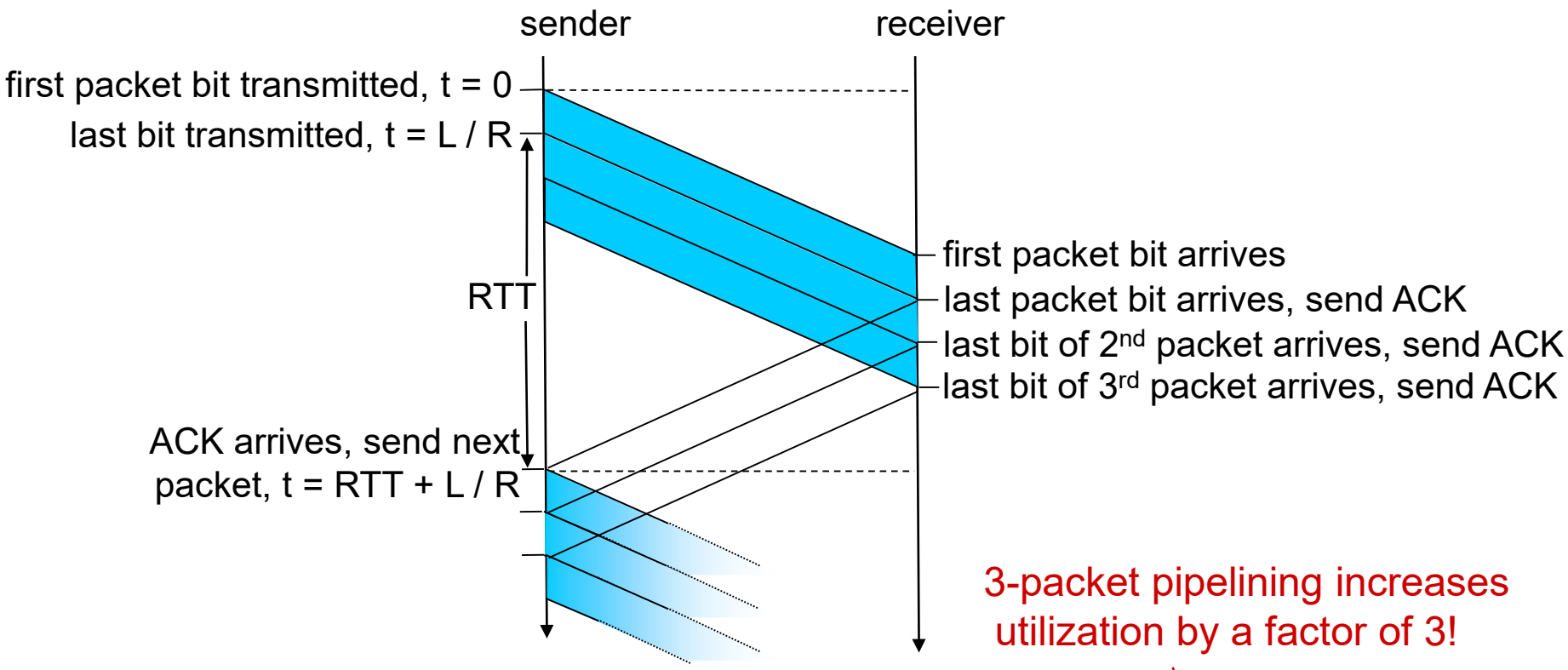
rdt 3.0: Stop-and-wait Operation

- Network protocol limits use of physical resources!



$$U_{sender} = \frac{L / R}{RTT + L/R} = \frac{0.008}{30 + 0.008} = 0.00027$$

Pipelining: Increased Utilization



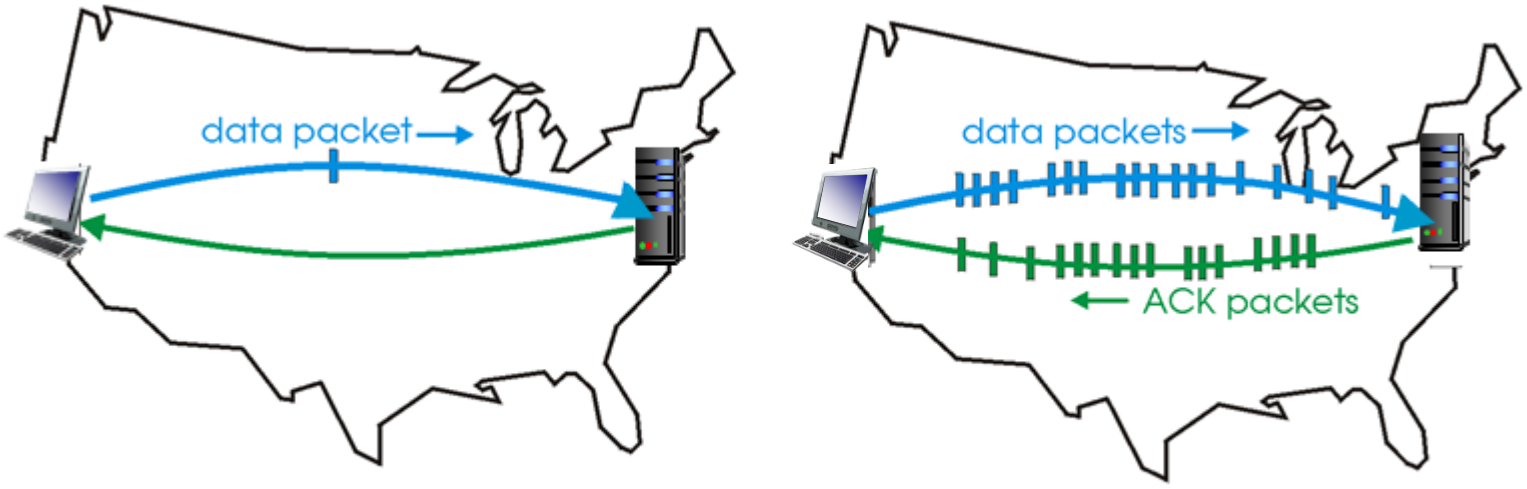
3-packet pipelining increases utilization by a factor of 3!

$$U_{sender} = \frac{3 * L / R}{RTT + L/R} = \frac{0.024}{30 + 0.008} = 0.00081$$

Pipelined Protocols

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

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



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

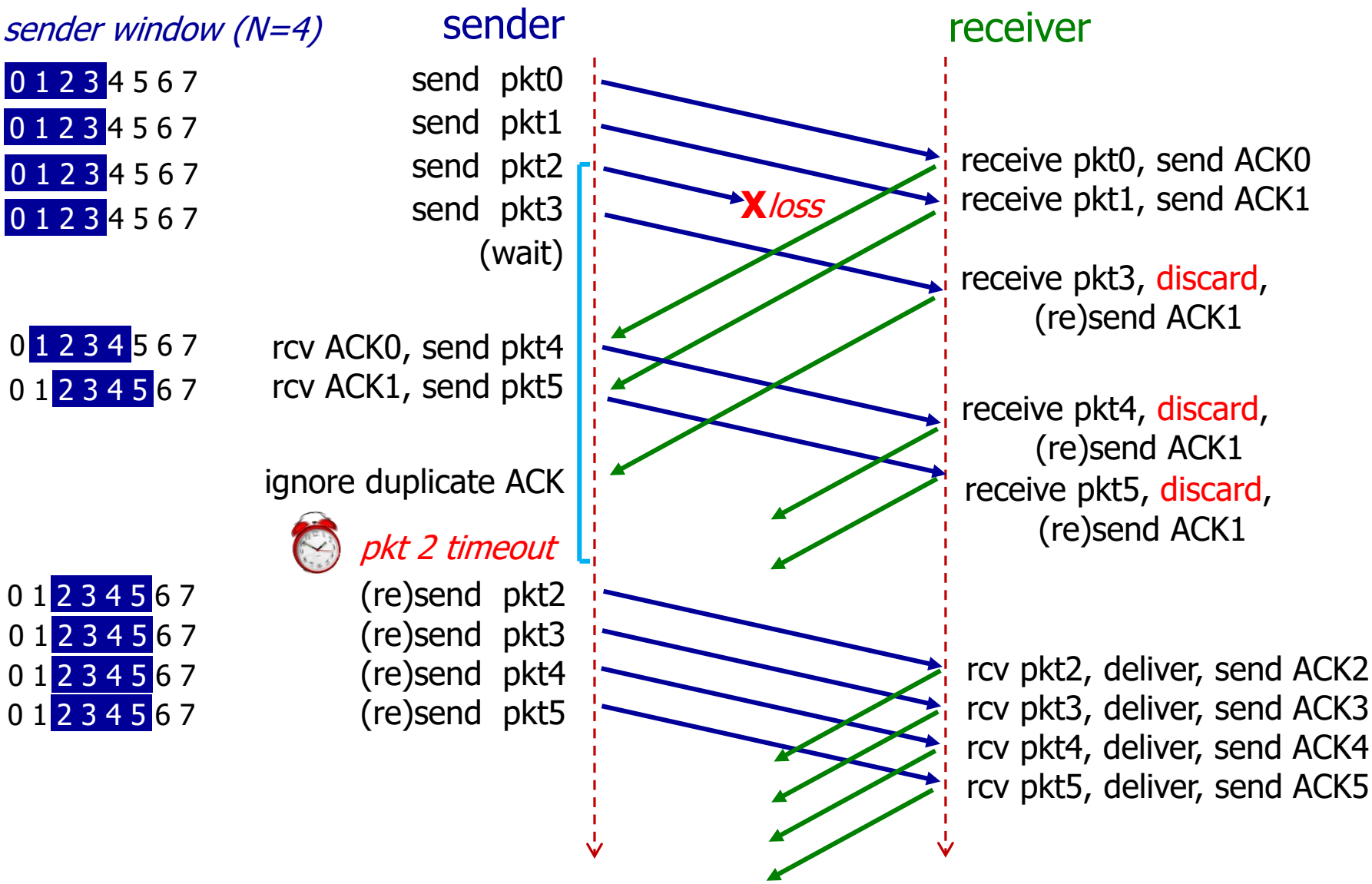
(b) a pipelined protocol in operation

Benchmark Pipelined Protocols

- Two generic forms of pipelined protocols:
  - *Go-Back-N*
  - *Selective repeat*

- Assumption (same as rdt 3.0): underlying channel
  - may flip bits in packets
  - may lose packets
  - may incur arbitrarily long packet delay
  - but won't re-order packets

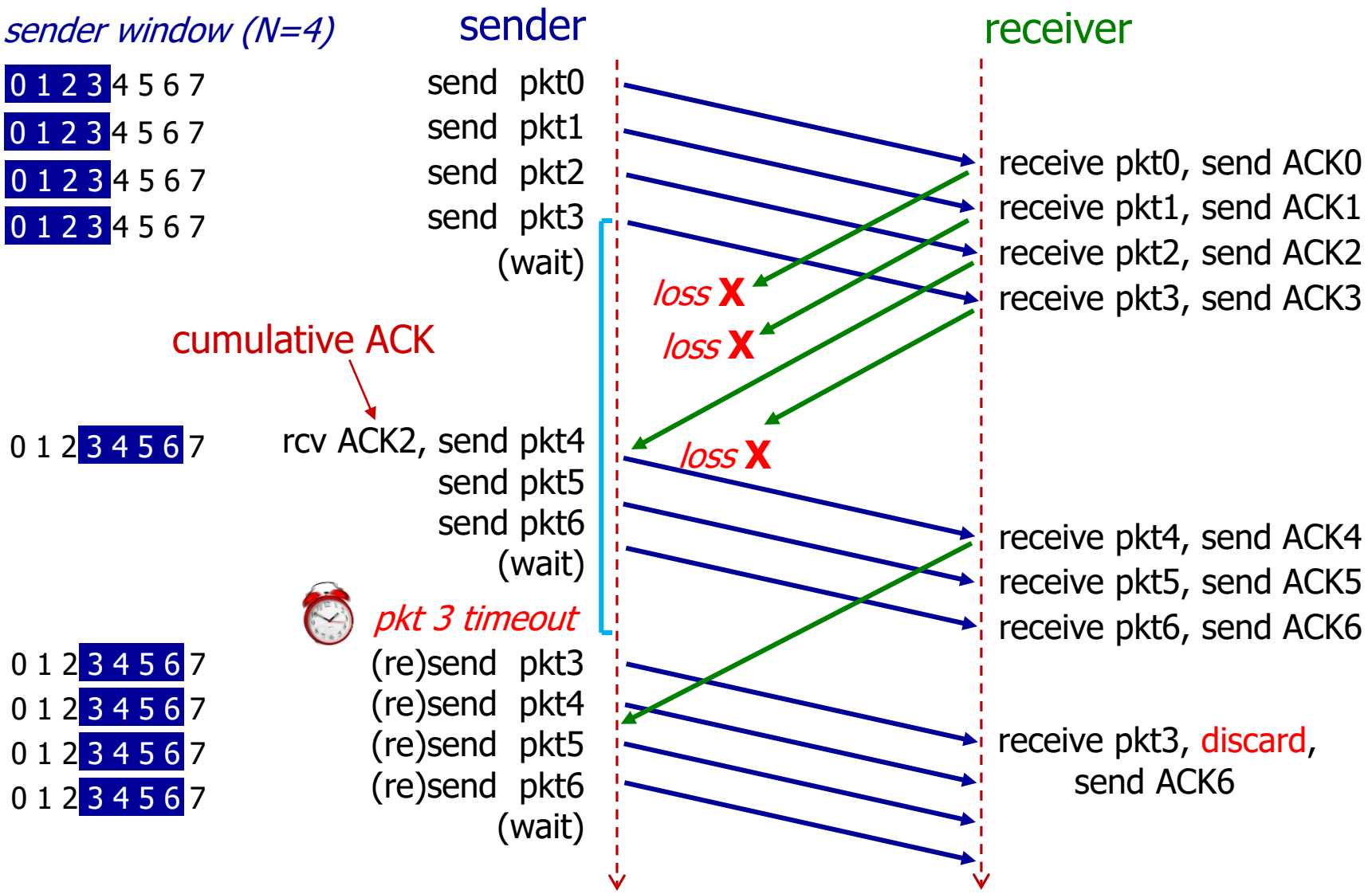
Go-back-N In Action



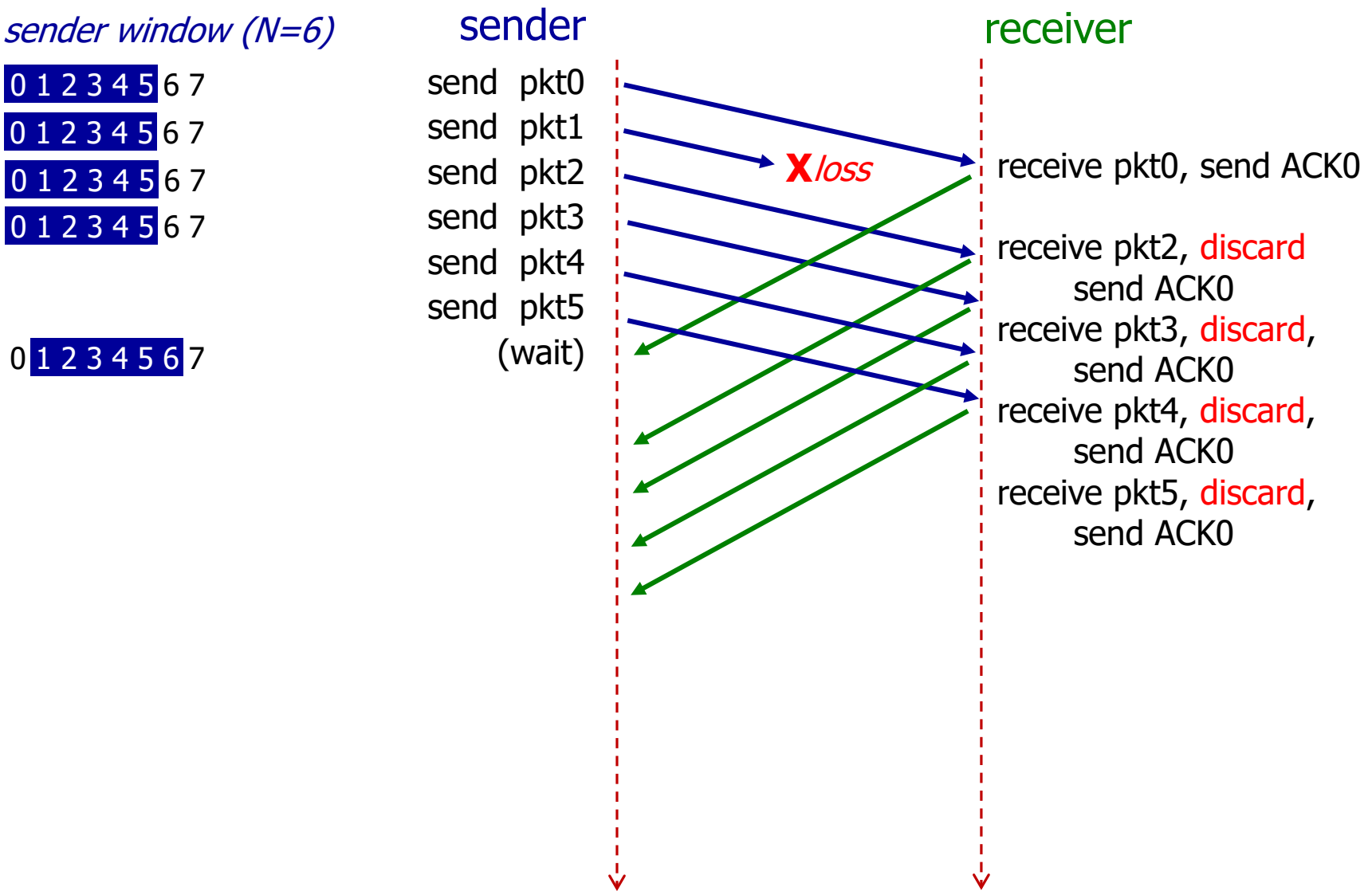
Go-back-N: Key Features

- GBN Sender
  - can have up to *N* unACKed packets in pipeline.
  - insert k-bits sequence number in packet header.
  - use a "sliding window" to keep track of unACKed packets.
  - keep a timer for the oldest unACKed packet.
  - *timeout(n):* retransmit packet *n* and all subsequent packets in the window.
- GBN Receiver
  - only ACK packets that arrive in order.
    - simple receiver: need only remember **`expectedSeqNum`**
  - discard out-of-order packets and ACK the last in-order seq. #.
    - *Cumulative ACK*: "*ACK m*" means all packets up to *m* are received.

Go-back-N In Action



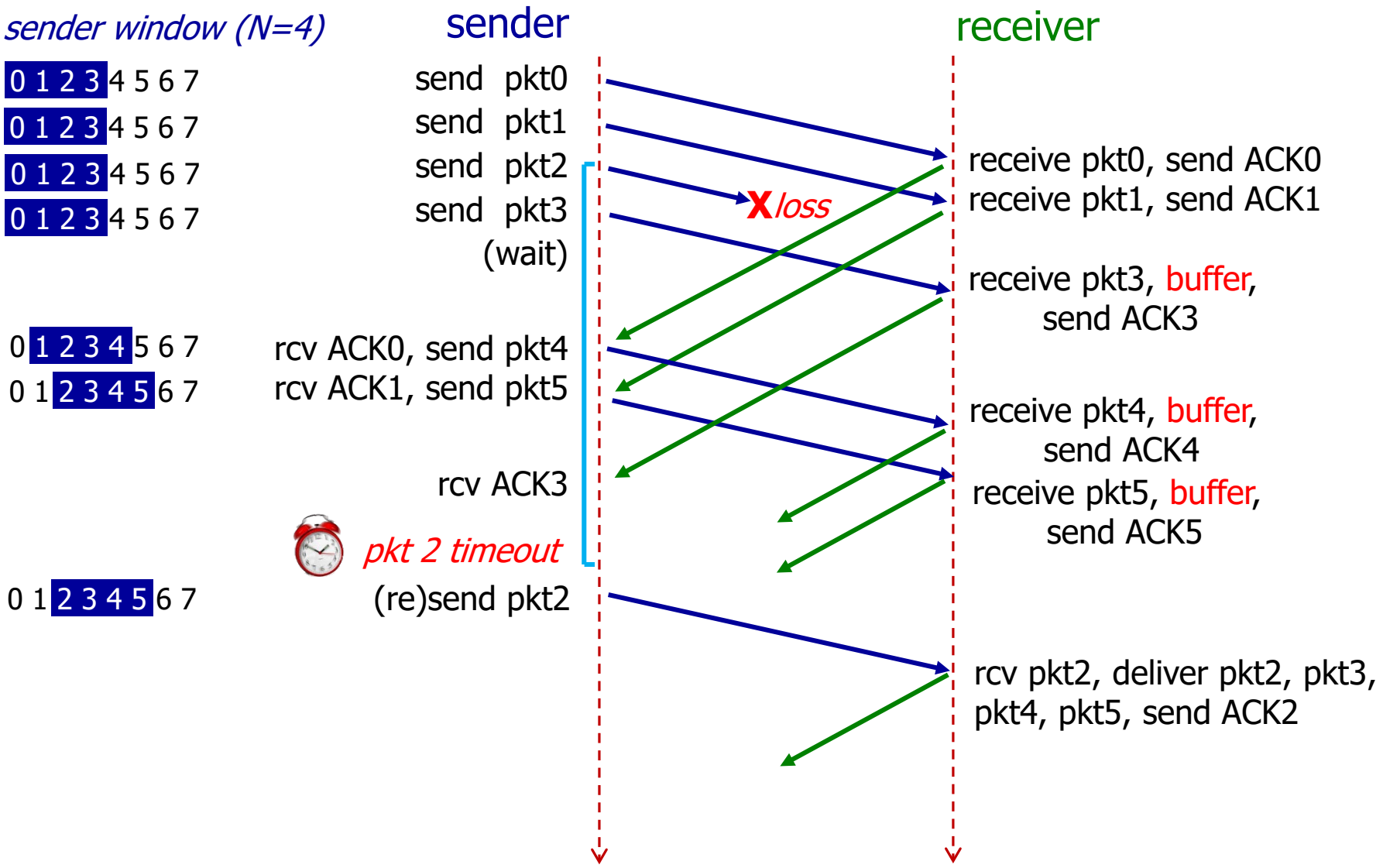
Go-back-N In Action



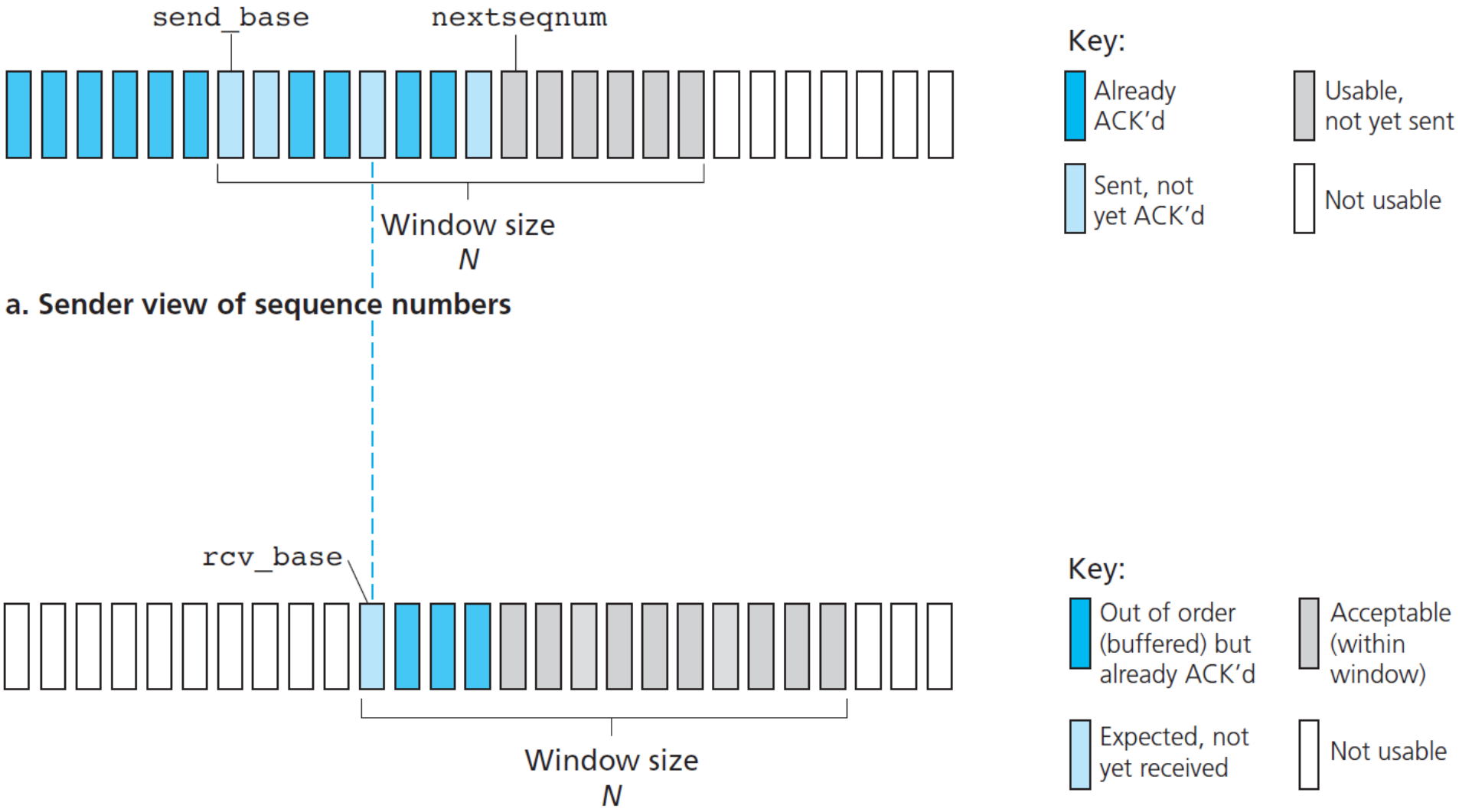
Selective Repeat: Key Features

- Receiver *individually* acknowledges all correctly received packets.
  - Buffers out-of-order packets, as needed, for eventual in-order delivery to upper layer.
- Sender maintains timer for each unACKed packet.
  - When timer expires, retransmit only that unACKed packet.

Selective Repeat In Action



SR Sender and Receiver Windows



Selective Repeat: Behaviors

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 is 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

Lectures 4&5: Roadmap

3.1 Transport-layer Services

3.2 Multiplexing and De-multiplexing

3.3 Connectionless Transport: UDP

3.4 Principles of Reliable Data Transfer

3.5 Connection-oriented transport: TCP

TCP: Transport Control Protocol

- In contrast to UDP, TCP is complex and is described in tens of RFCs, with new mechanisms or tweaks introduced throughout the years, resulting in many variants of TCP.

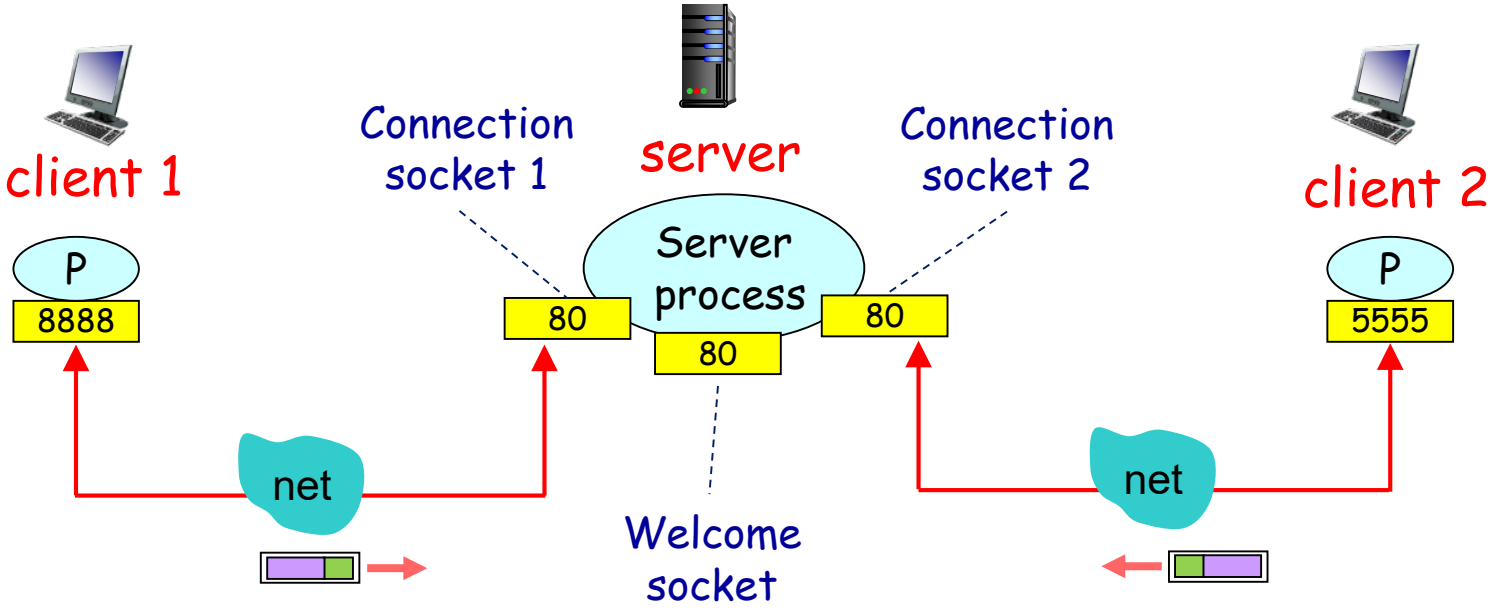
- We will only scratch the surface of TCP in CS2105.
  - More will be covered in CS3103.

TCP Overview

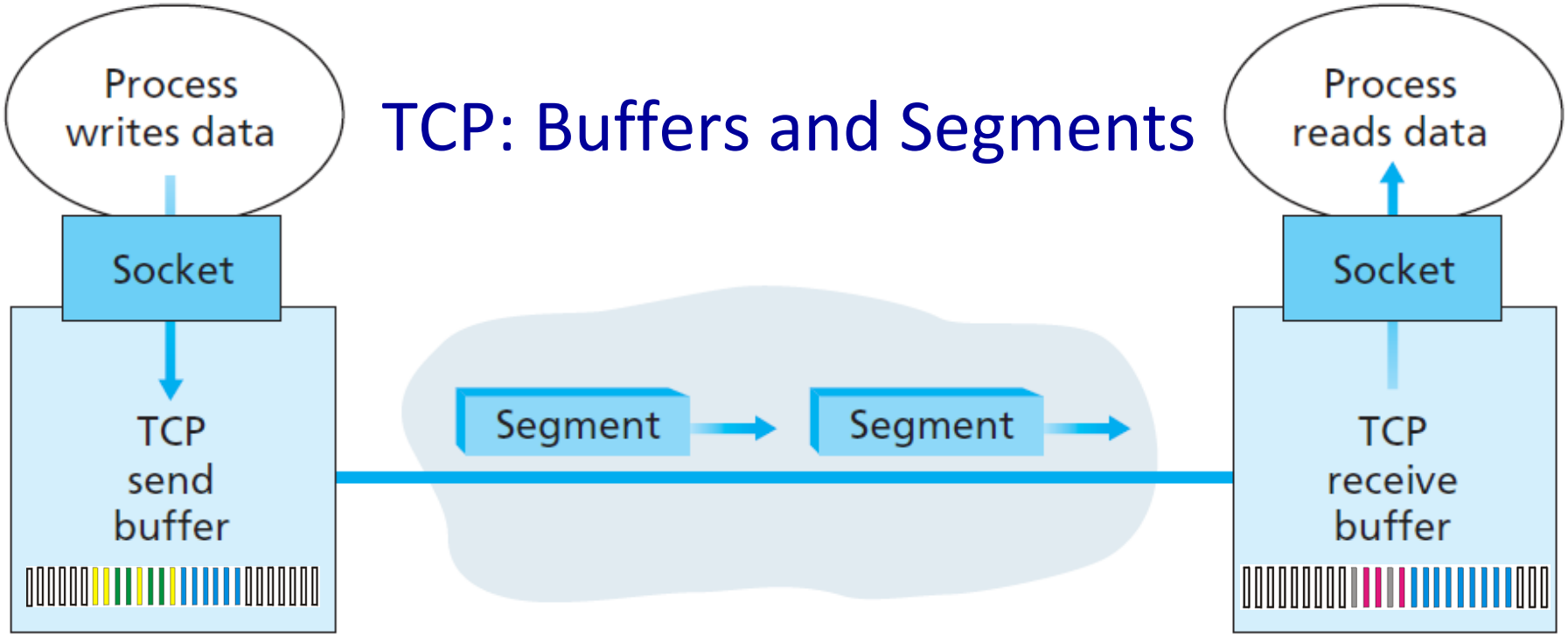
- Connection-oriented:
  - handshaking (exchange of control messages) before sending app data.
- Full duplex service:
  - bi-directional data flow in the same connection
- Reliable, in-order *byte steam:*
  - use sequence numbers to label bytes
- Flow control and congestion control
  - not in syllabus

Connection-oriented De-mux

- A TCP connection (socket) is identified by 4-tuple:
  - (srcIPAddr, srcPort, destIPAddr, destPort)
  - Receiver uses all four values to direct a segment to the appropriate socket.

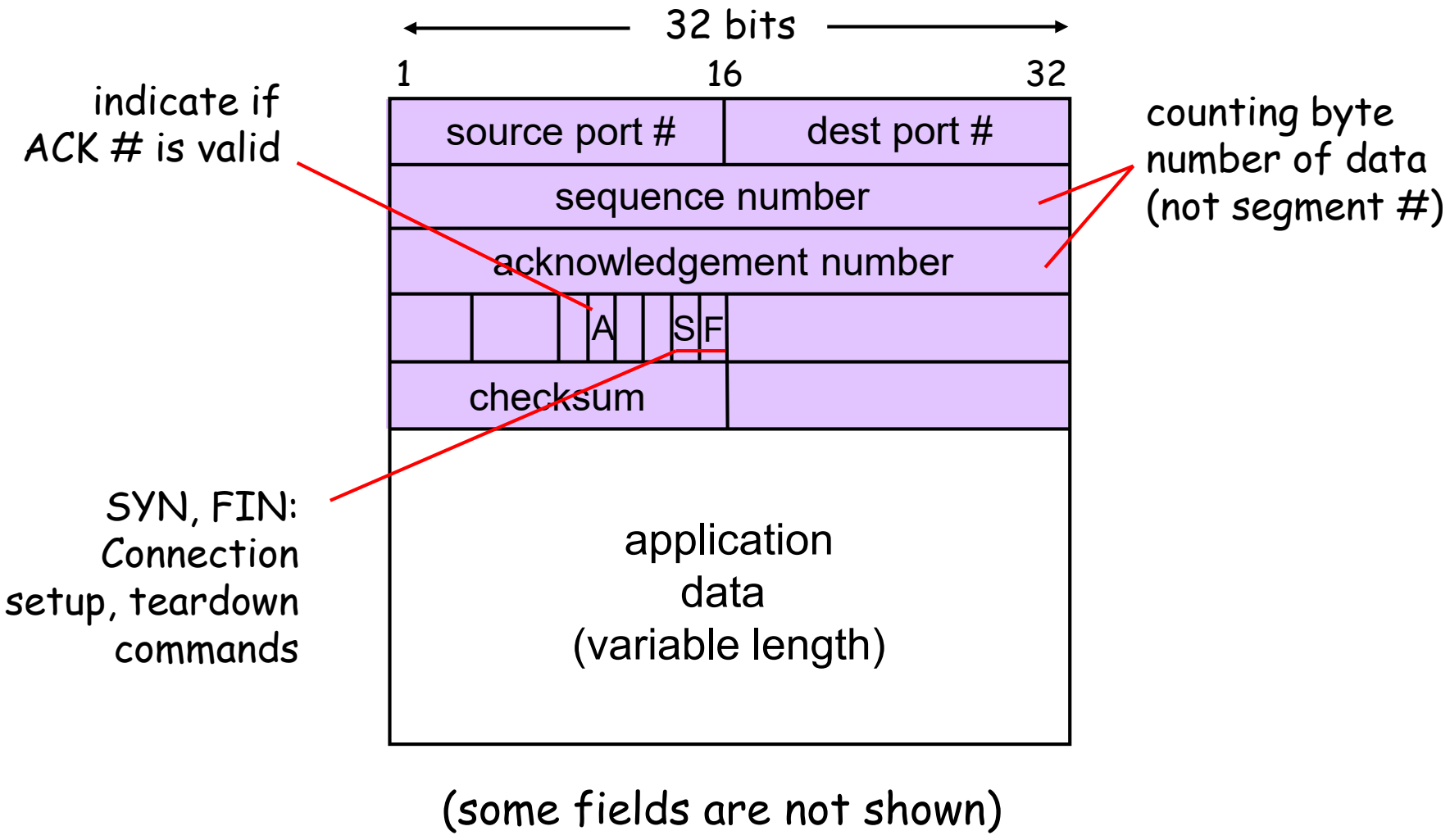


TCP: Buffers and Segments



- TCP send and receive buffers
  - two buffers created after handshaking at any side.
- How much app-layer data a TCP segment can carry?
  - maximum segment size (MSS), typically 1,460 bytes
  - app passes data to TCP and TCP forms packets in view of MSS.

TCP Header



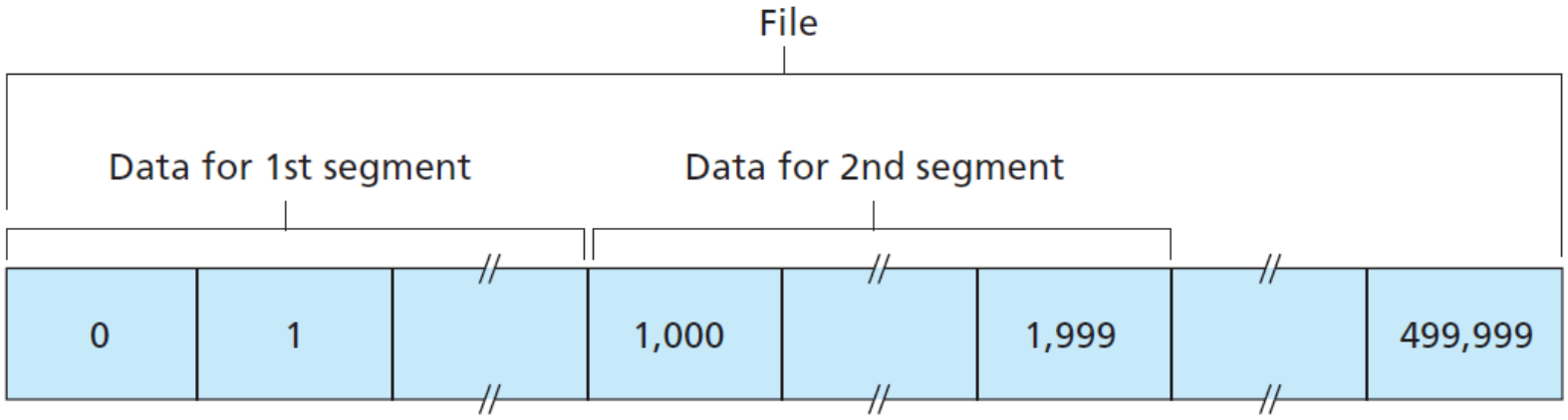
(some fields are not shown)

© CS2105

source port #	dest port #
sequence number	
ACK number	
checksum	

TCP Sequence Number

- "Byte number" of the first byte of data in a segment.
- Example: send a file of 500,000 bytes; MSS is 1,000 bytes.



Dividing file data into TCP segments

- Seq. # of 1st TCP segment: 0, 2nd TCP segment: 1,000, 3rd TCP segment: 2,000, 4th TCP segment: 3,000, etc.

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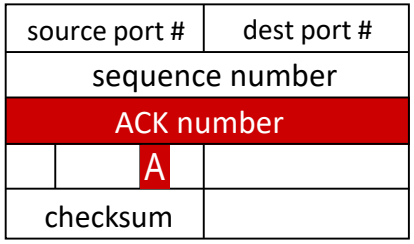


TCP ACK Number

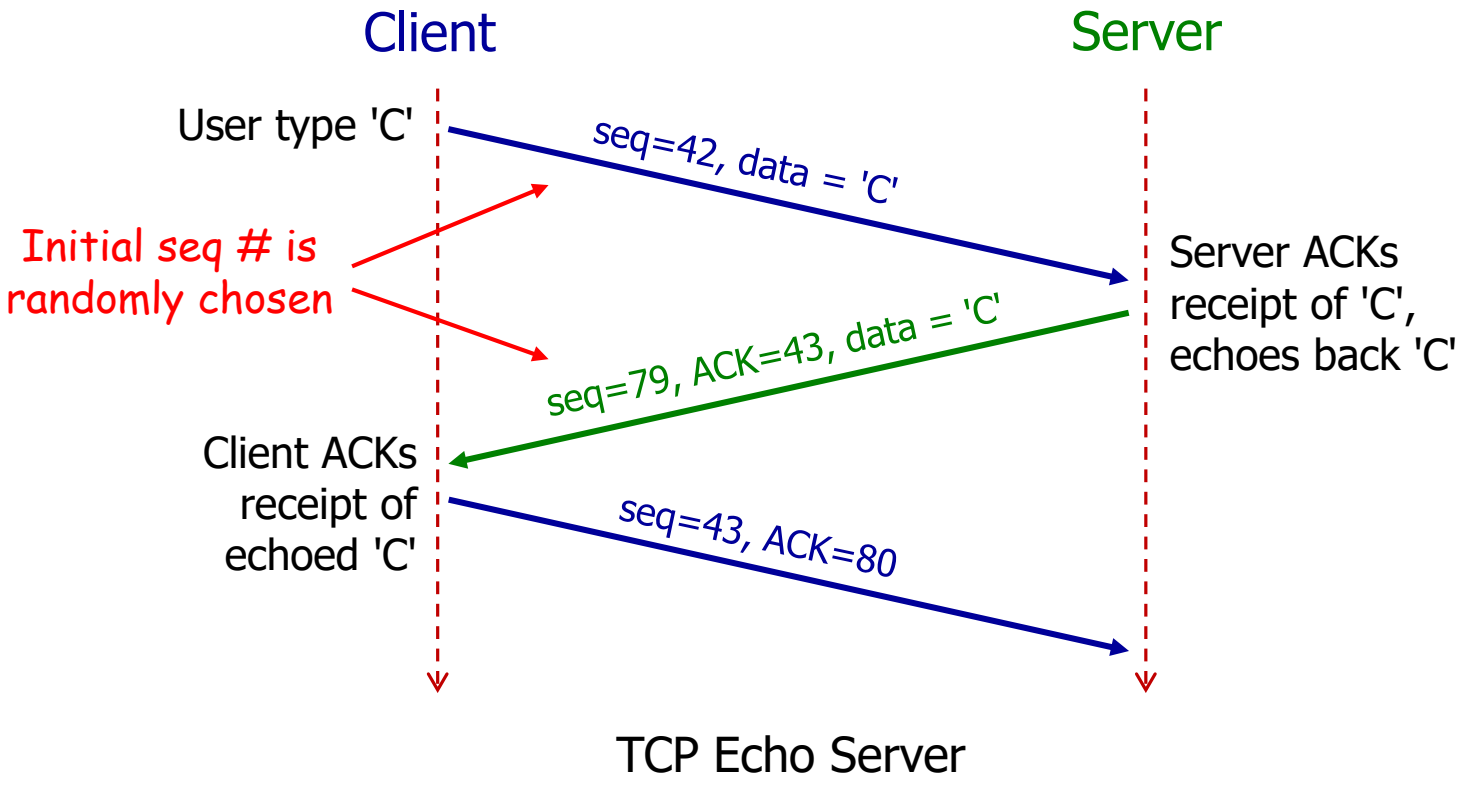
- Seq # of the next byte of data expected by receiver.

Sequence number of a segment	Amount of data carried	Corresponding ACK number
0	1,000	1,000
1,000	1,000	2,000
2,000	1,000	3,000
3,000	1,000	4,000
…	…	…

- TCP ACKs up to the first missing byte in the stream (cumulative ACK).
  - Note: TCP spec doesn't say how receiver should handle out-of-order segments - it's up to implementer.

Example: TCP Echo Server

- TCP (and also UDP) is a full duplex protocol
  - bi-directional data flow in the same TCP connection.
- Example:



TCP Sender Events (simplified)

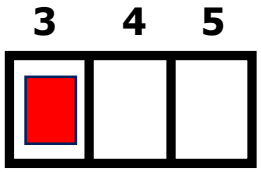
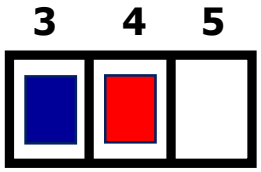
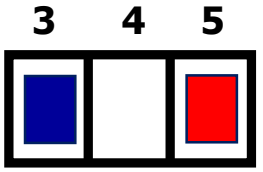
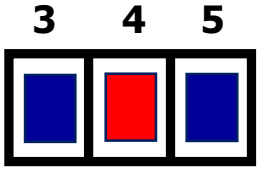
```
NextSeqNum=InitialSeqNumber
SendBase=InitialSeqNumber

loop (forever) {
    switch(event)

        event: data received from application above
            create TCP segment with sequence number NextSeqNum
            if (timer currently not running)
                start timer
            pass segment to IP
            NextSeqNum=NextSeqNum+length(data)
            break;

        event: timer timeout
            retransmit not-yet-acknowledged segment with
                smallest sequence number
            start timer
            break;

        event: ACK received, with ACK field value of y
            if (y > SendBase) {
                SendBase=y
                if (there are currently any not-yet-acknowledged segments)
                    start timer
                }
            break;

    } /* end of loop forever */
```

Sender keeps one timer only

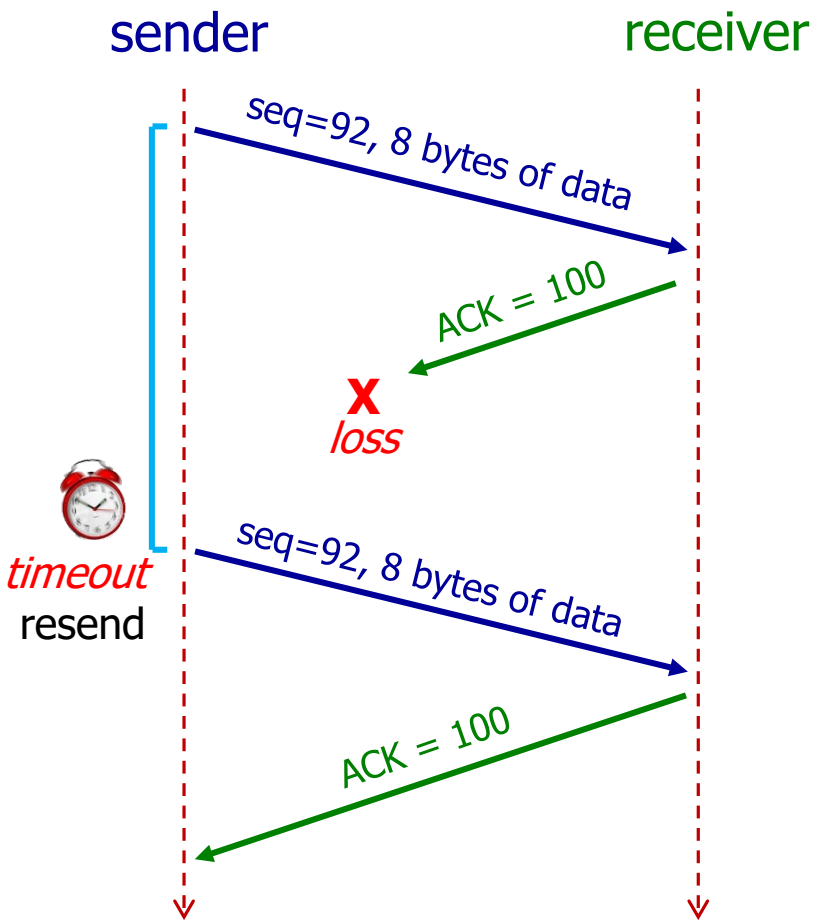
Retransmit only oldest unACKed packet

Cumulative ACK

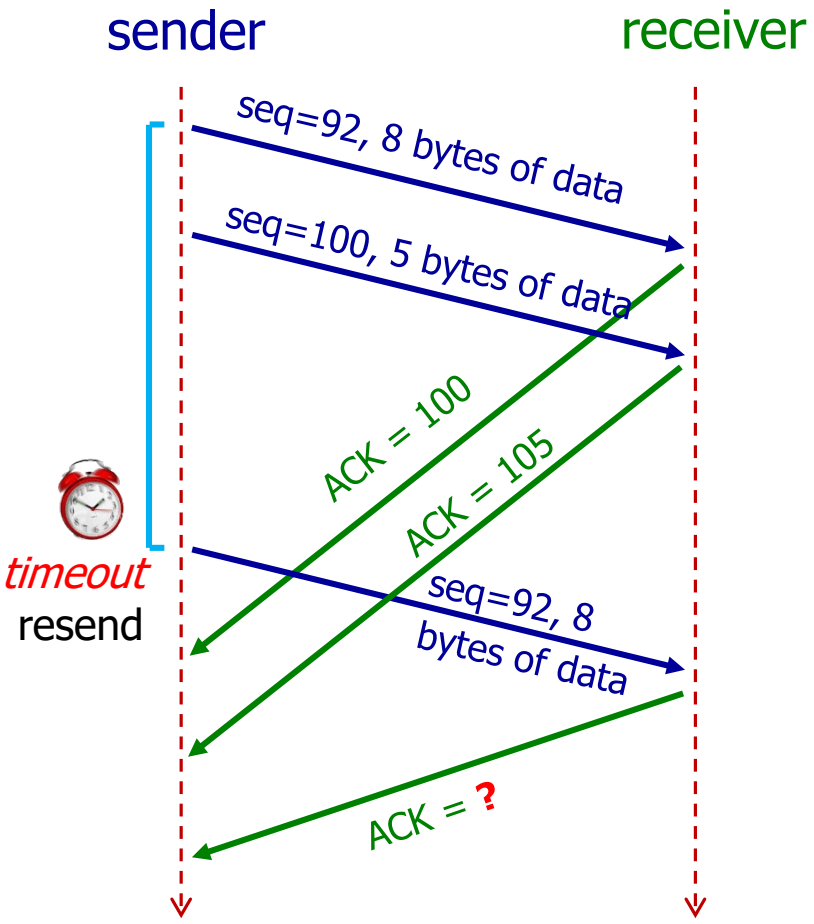
TCP ACK Generation [RFC 5681]

Event at TCP 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	Immediately send ACK, provided that segment starts at lower end of gap	

TCP Timeout / Retransmission



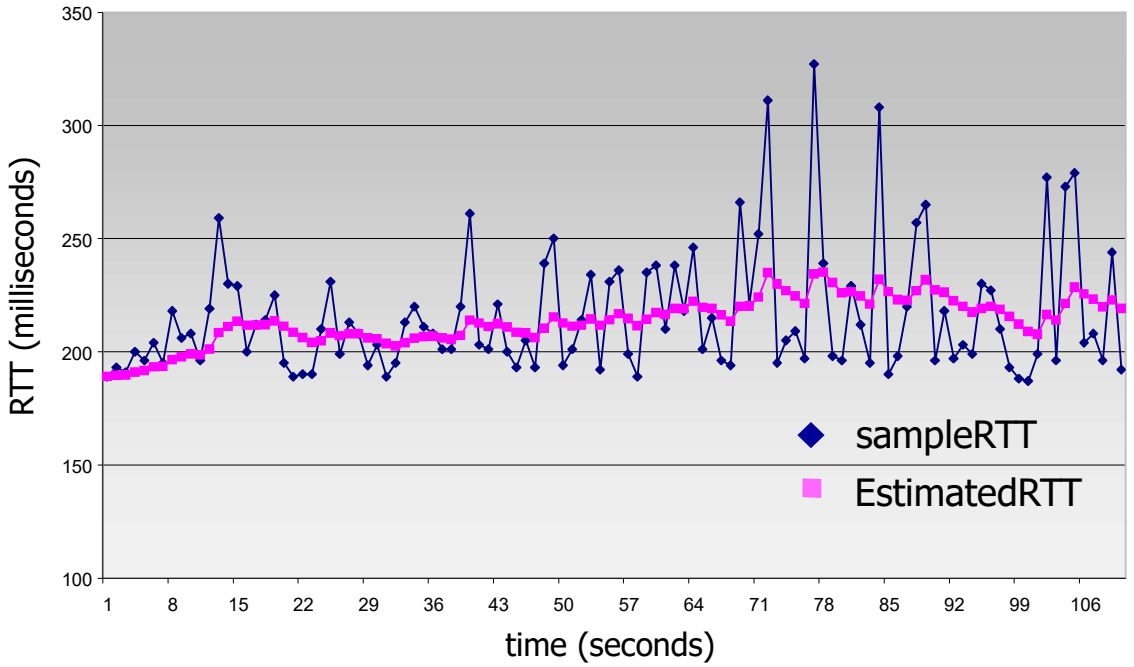
a) Lost ACK



b) premature timeout

TCP Timeout Value

- How does TCP set appropriate timeout value?
  - too short timeout: premature timeout and unnecessary retransmissions.
  - too long timeout: slow reaction to segment loss.
  - Timeout interval must be longer than RTT – but RTT varies!



TCP Timeout Value

- TCP computes (and keeps updating) timeout interval based on estimated RTT.

$$\texttt{EstimatedRTT} = (1-\alpha)*\texttt{EstimatedRTT} + \alpha*\texttt{SampleRTT}$$

(typical value of α : 0.125)

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

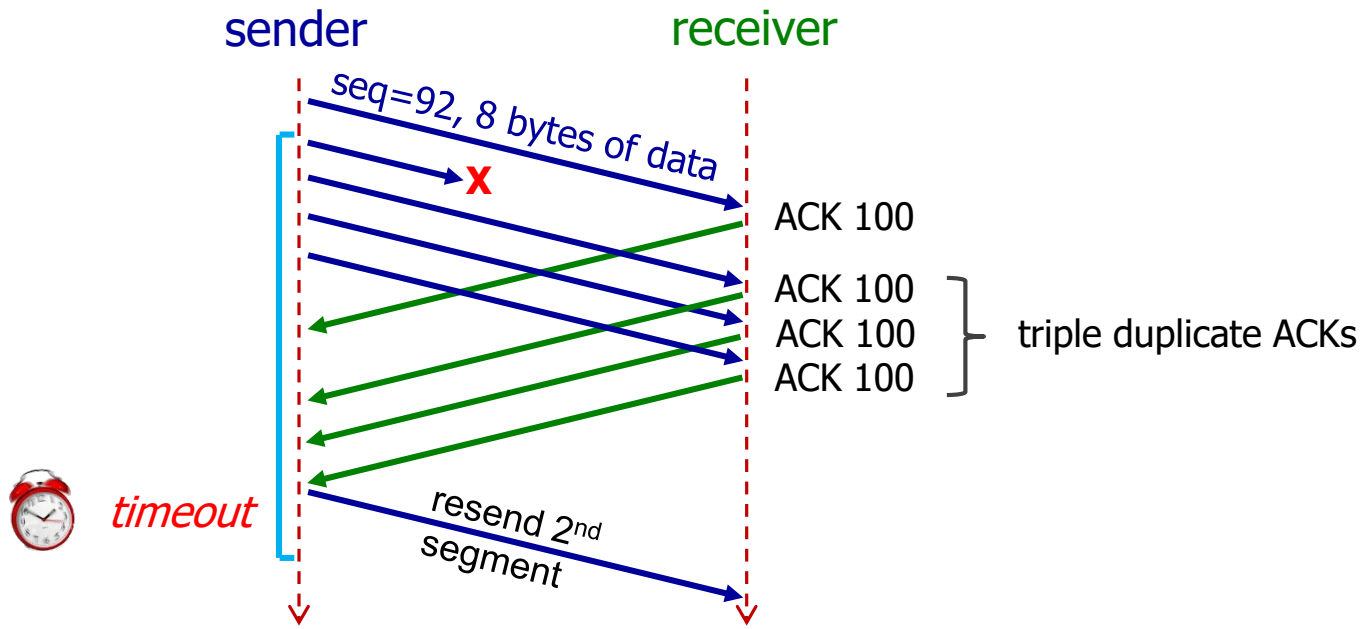
(typical value of β : 0.25)

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

"safety margin"

TCP Fast Retransmission [RFC 2001]

- Timeout period is often relatively long.
  - long delay before resending lost packet
- Fast retransmission:
  - **Event:** If sender receives 4 ACKs for the same segment, it supposes that segment is lost.
  - **Action:** resend segment (even before timer expires).



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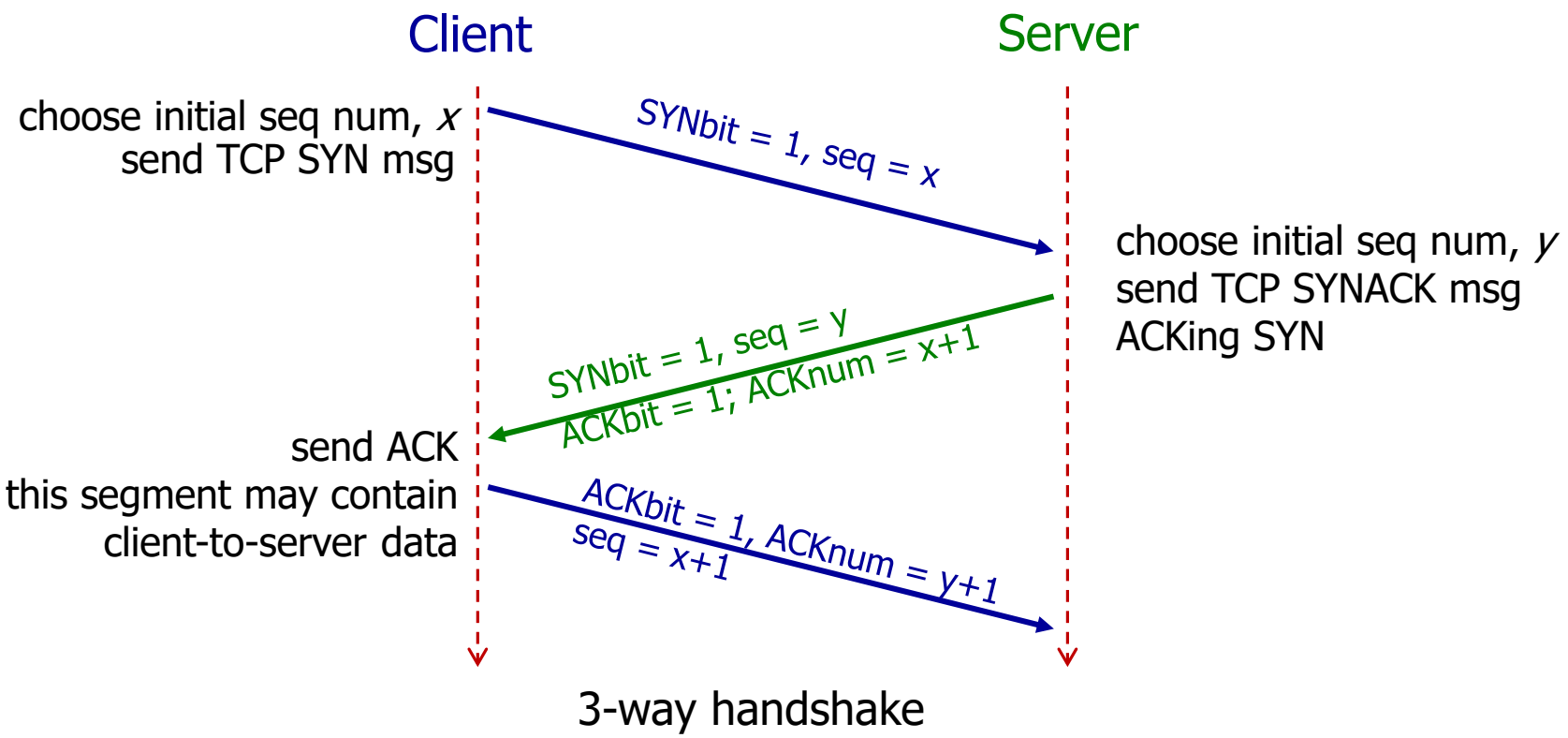
source port #	dest port #
sequence number	
ACK number	
A S	
checksum	

Establishing Connection

- Before exchanging data, TCP sender and receiver "shake hands".
  - Agree on connection and exchange connection parameters.



3-way handshake

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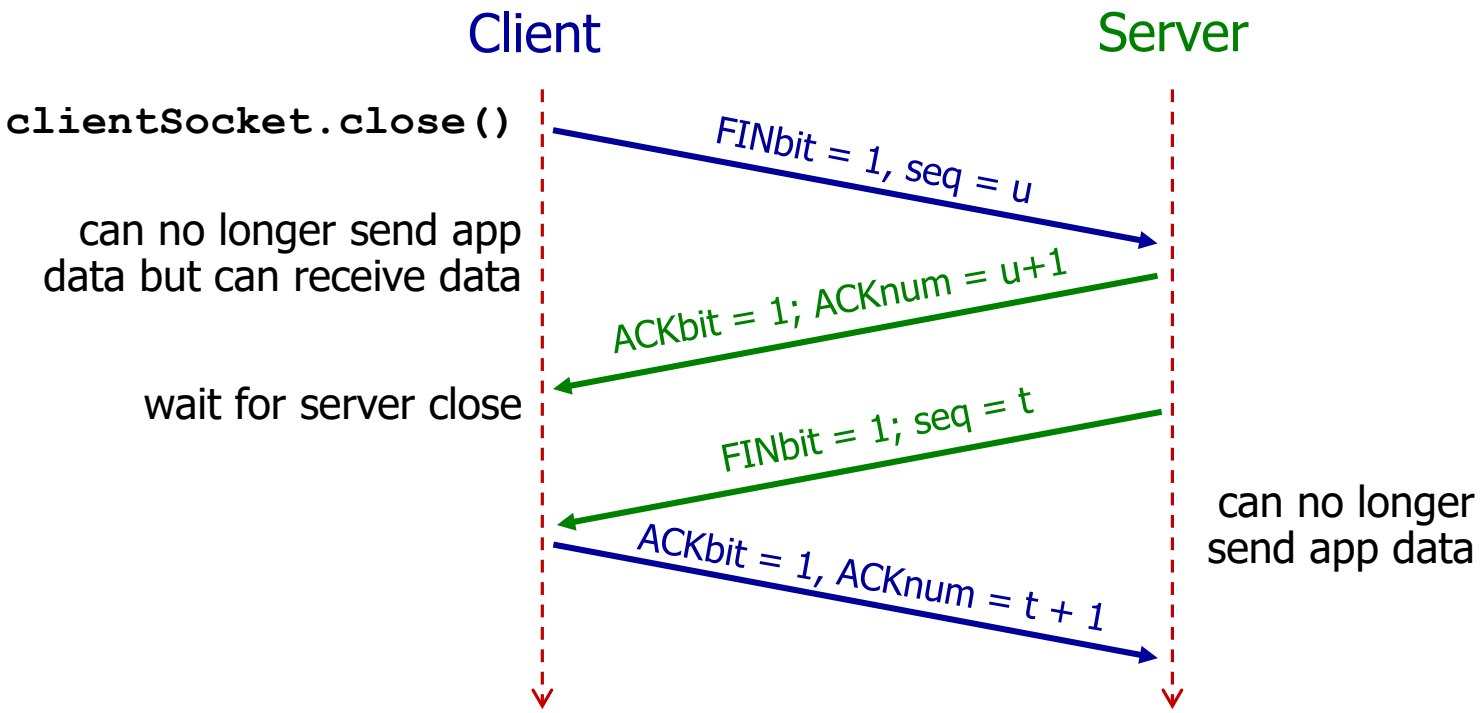
source port #	dest port #
sequence number	
ACK number	
A F	
checksum	

Closing Connection

- Client, server each close their side of connection.
  - send TCP segment with FIN bit = 1



What we did not cover….

- TCP flow control (Chapter 3.5.5)
  - Sender won't overflow receiver's buffer by sending too much or too fast.
  - Receiver feeds back to sender how many more bytes it is willing to accept.
- TCP congestion control (Chapter 3.6 & 3.7)
  - Be polite and send less if network is congested.
- They will be covered in the next course (CS3103)

Lectures 4&5: Summary

Go-back-N

- Sender can have up to *N* unACKed packets in pipeline
- Receiver only sends *cumulative ACKs*
  - Out-of-order packets discarded
- Sender sets timer for the oldest unACKed packet
  - when timer expires, retransmit ***all*** unACKed packets

Selective Repeat

- Sender can have up to *N* unACKed packets in pipeline
- Receiver sends *individual ACK* for each packet
  - Out-of-order packets buffered
- Sender maintains timer for ***each*** unACKed packet
  - when timer expires, retransmit only that unACKed packet

Lectures 4&5: Summary

- Connection-oriented transport: TCP
  - Segment structure
  - Reliable data transfer
  - Setting and updating retransmission time interval
  - 3-way handshake
  - Fast retransmission

CS2105

An Αωεσομε Introduction to Computer Networks

Lectures 6&7: The Network Layer

Department of Computer Science
School of Computing

Lectures 6&7: The Network Layer

After this class, you are expected to understand:

- the basic services network layer provides.
- the purpose of DHCP and how it works.
- IP address, subnet, subnet mask and address allocation.
- how longest prefix forwarding in a router works.
- the purpose of routing protocols on the Internet.
- the principle of Bellman-Ford equation.
- the workings of distance vector algorithm.
- the purpose of NAT and how it works.
- the Internet Protocol (IP) and how datagram fragmentation works.

Lectures 6&7: Roadmap

4.1 Overview of Network Layer

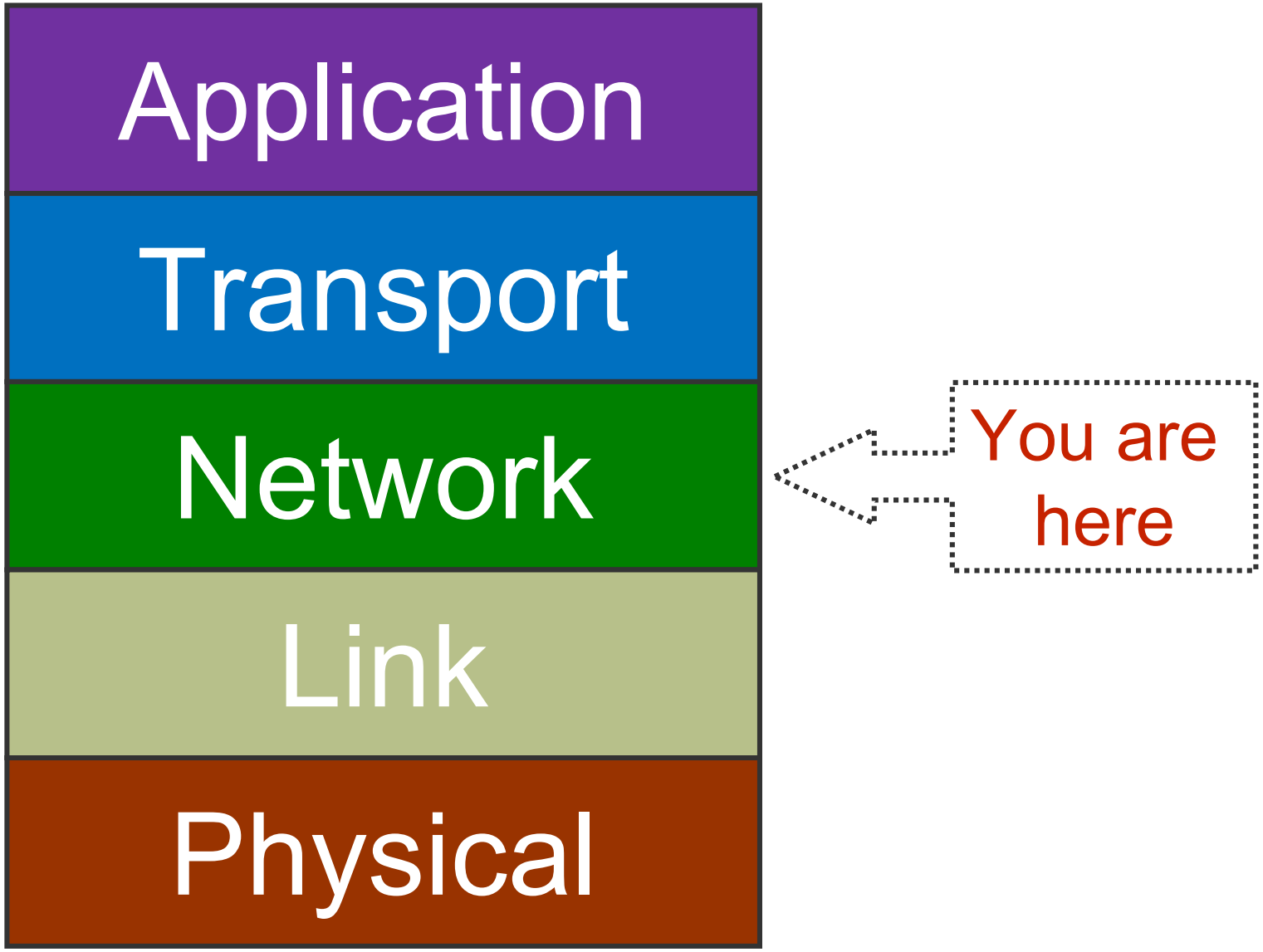
4.2 What's Inside a Router

4.3 The Internet Protocol (IP)

5.2 Routing Algorithms

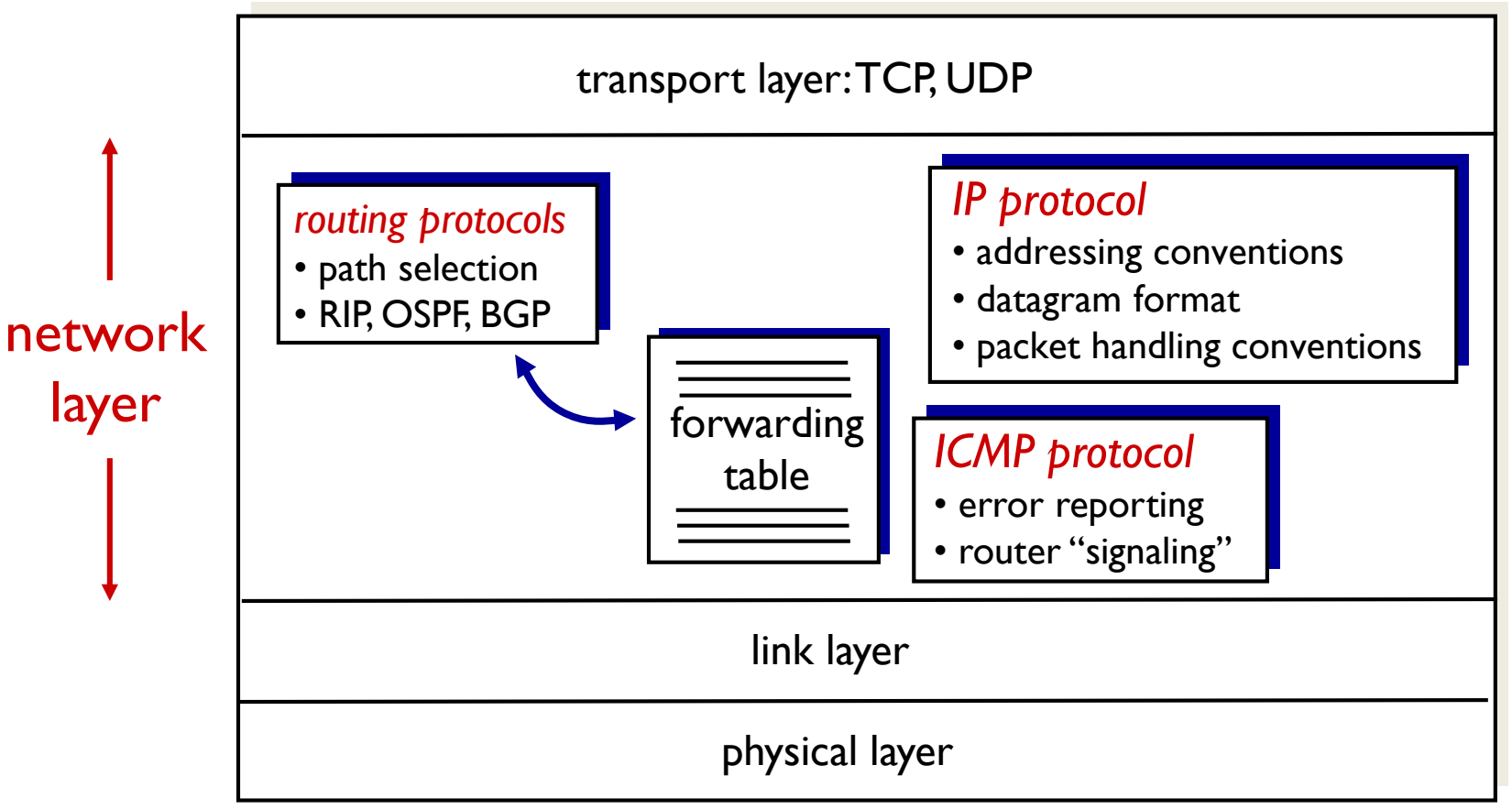
5.6 ICMP

Kurose Textbook, Chapters 4&5
(Some slides are taken from the book)


Application
Transport
Network
Link
Physical
You are here

Network Layer Services

- Network layer delivers packets to receiving hosts.
  - Routers examine header fields of IP datagrams passing it.



Lectures 6&7: Roadmap

4.1 Overview of Network Layer

4.2 What's Inside a Router

- 4.2.1 Destination-Based Forwarding

4.3 The Internet Protocol (IP)

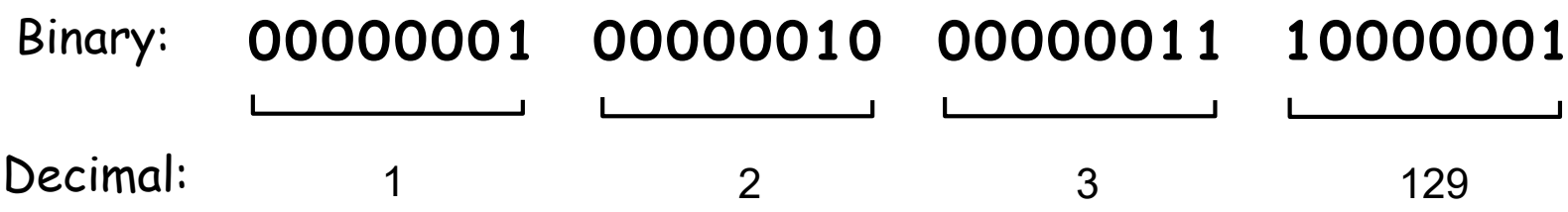
- 4.3.3 IPv4 Addressing

5.2 Routing Algorithms

5.6 ICMP

IP Address

- IP address is used to identify a host (or a router).
  - A 32-bit integer expressed in either binary or decimal

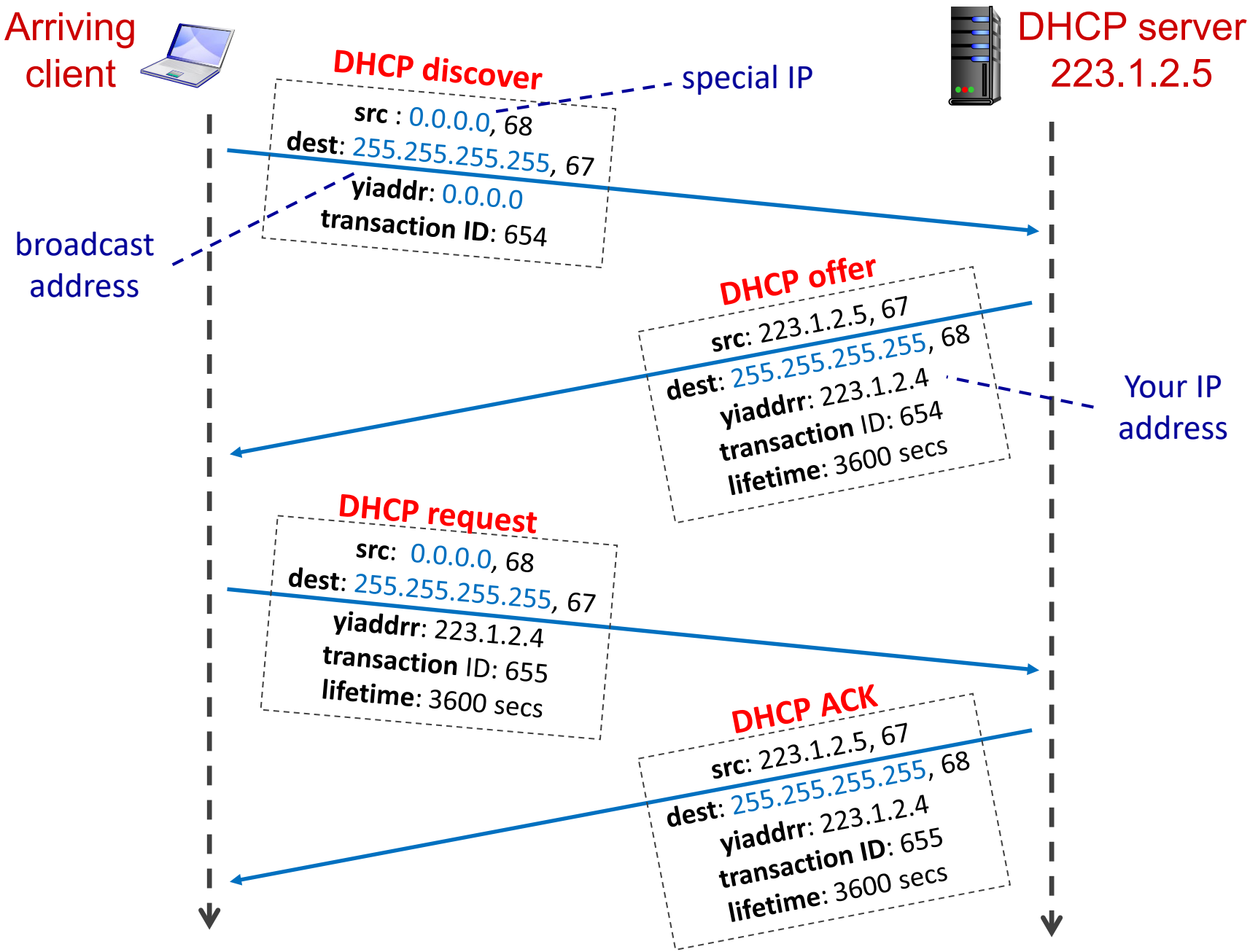


- How does a host get an IP address?
  - manually configured by system administrator, or
  - automatically assigned by a DHCP (Dynamic Host Configuration Protocol) server.

DHCP

- DHCP allows a host to dynamically obtain its IP address from DHCP server when it joins network.
  - IP address is renewable
  - allow reuse of addresses (only hold address while connected)
  - support mobile users who want to join network.

- DHCP: 4-step process:
  1) Host broadcasts "DHCP discover" message
  2) DHCP server responds with "DHCP offer" message
  3) Host requests IP address: "DHCP request" message
  4) DHCP server sends address: "DHCP ACK" message


Arriving client
DHCP server 223.1.2.5
DHCP discover
src : 0.0.0.0, 68
dest: 255.255.255.255, 67
yiaddr: 0.0.0.0
transaction ID: 654
special IP
broadcast address
DHCP offer
src: 223.1.2.5, 67
dest: 255.255.255.255, 68
yiaddrr: 223.1.2.4
transaction ID: 654
lifetime: 3600 secs
Your IP address
DHCP request
src: 0.0.0.0, 68
dest: 255.255.255.255, 67
yiaddrr: 223.1.2.4
transaction ID: 655
lifetime: 3600 secs
DHCP ACK
src: 223.1.2.5, 67
dest: 255.255.255.255, 68
yiaddrr: 223.1.2.4
transaction ID: 655
lifetime: 3600 secs

More on DHCP

- In addition to host IP address assignment, DHCP may also provide a host additional network information:
  - IP address of first-hop router
  - IP address of local DNS server
  - Network mask (indicating network prefix versus host ID of an IP address)

- DHCP runs over UDP
  - DHCP server port number: 67
  - DHCP client port number: 68

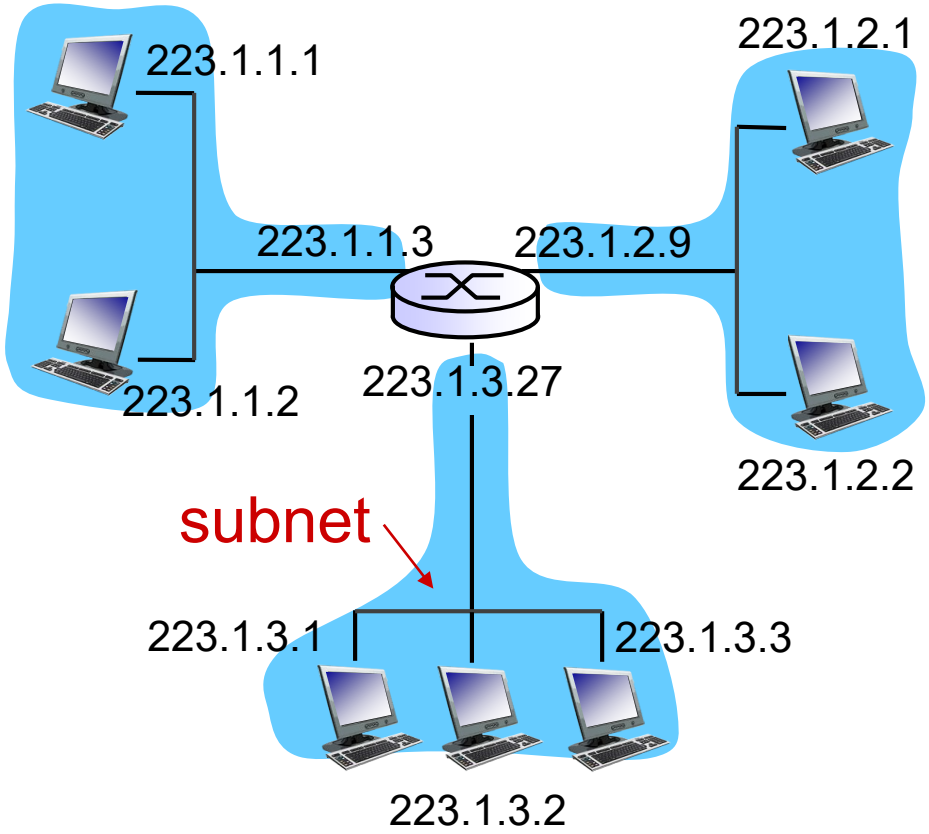
Some Special IP Addresses

Special Addresses	Present Use
0.0.0.0/8	Non-routable meta-address for special use
127.0.0.0/8	Loopack address. A datagram sent to an address within this block loops back inside the host. This is ordinarily implemented using only 127.0.0.1/32.
10.0.0.0/8 172.16.0.0/12 192.168.0.0/16	Private addresses, can be used without any coordination with IANA or an Internet registry.
255.255.255.255/32	Broadcast address. All hosts on the same subnet receive a datagram with such a destination address.

The full list of special IP addresses can be found in RFC5735:
https://tools.ietf.org/rfc/rfc5735.txt

IP Address and Network Interface

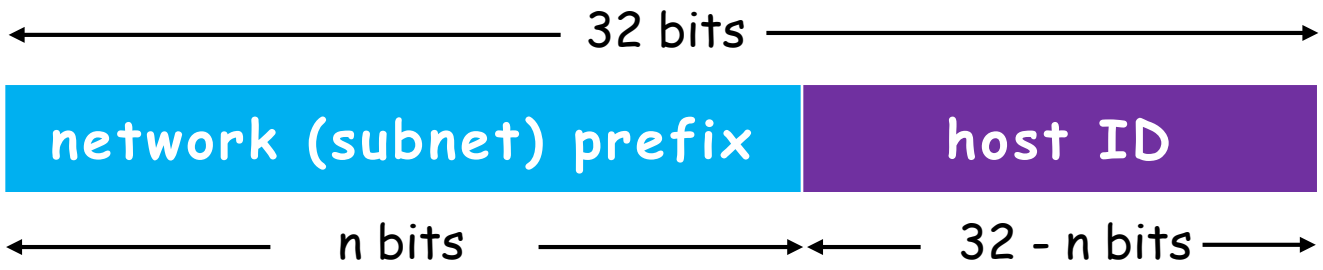
- An IP address is associated with a network interface.
  - A host usually has one or two network interfaces (e.g. wired Ethernet and WiFi).
  - A router typically has multiple interfaces.



A network consisting of 3 subnets
(first 24 bits of IP addr. are network prefix)

IP Address and Subnet

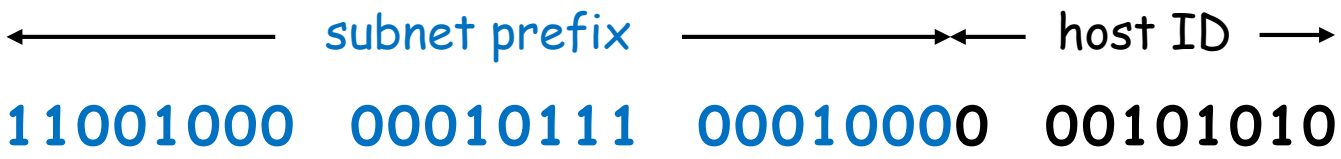
- An IP address logically comprises two parts:



- Subnet is a network formed by a group of "directly" interconnected hosts.
  - Hosts in the same subnet have the same network prefix of IP address.
  - Hosts in the same subnet can physically reach each other without intervening router.
  - They connect to the outside world through a router.

IP Address: CIDR

- The Internet's IP address assignment strategy is known as Classless Inter-domain Routing (CIDR).
  - Subnet prefix of IP address is of arbitrary length.
  - Address format: **a.b.c.d/x**, where **x** is the number of bits in subnet prefix of IP address.

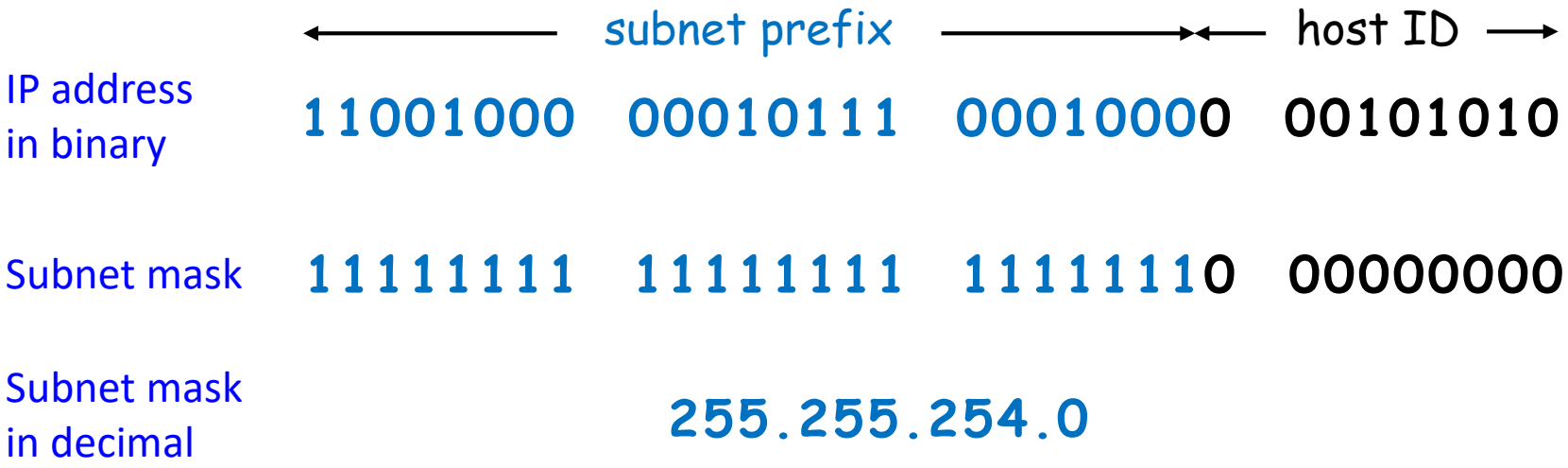


this subnet contains 2^9 IP addresses
subnet prefix: 200.23.16.42/23

/23 indicates the no. of bits of subnet prefix

Subnet Mask

- Subnet mask is used to determine which subnet an IP address belongs to.
  - made by setting all subnet prefix bits to "1"s and host ID bits to "0"s.
- Example: for IP address 200.23.16.42/23:



Quiz

- For the following 4 IP addresses, which one is in a different subnet from the rest 3?

a. 172.26.185.128/26

b. 172.26.185.130/26

c. 172.26.185.160/26

d. 172.26.185.192/26

IP Address Allocation

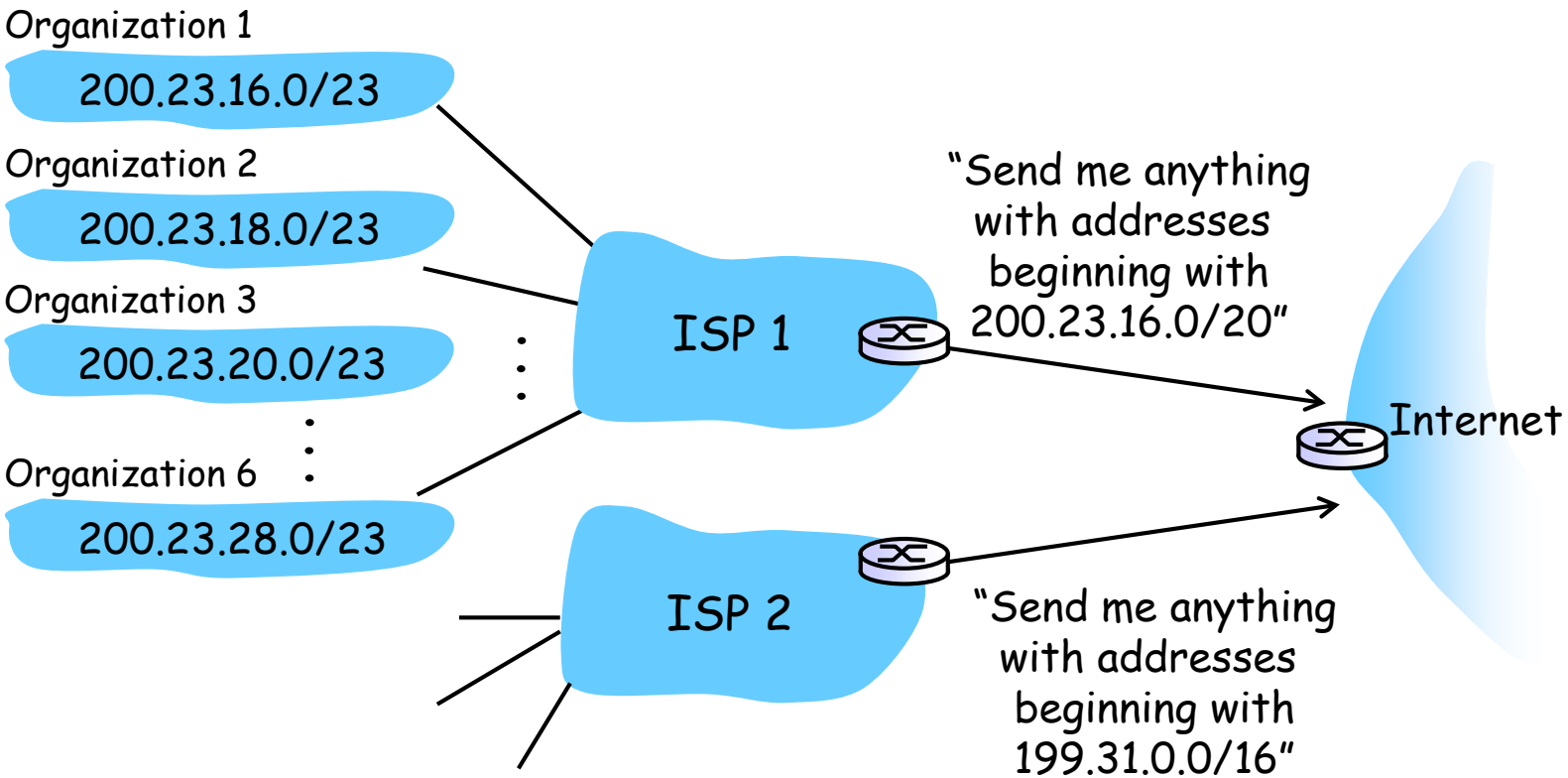
- Q: How does an organization obtain a block of IP addresses?
- A: Buy from registry or rent from ISP's address space.

	Binary Address	Decimal Address
ISP's block	11001000 00010111 0001 000 0 00000000	200.23.16.0/20
Organization 1	11001000 00010111 0001 000 0 00000000	200.23.16.0/23
Organization 2	11001000 00010111 0001 001 0 00000000	200.23.18.0/23
Organization 3	11001000 00010111 0001 010 0 00000000	200.23.20.0/23
…	…	…
Organization 6	11001000 00010111 0001 101 0 00000000	200.23.28.0/23

use 3 more bits to differentiate 6 organizations

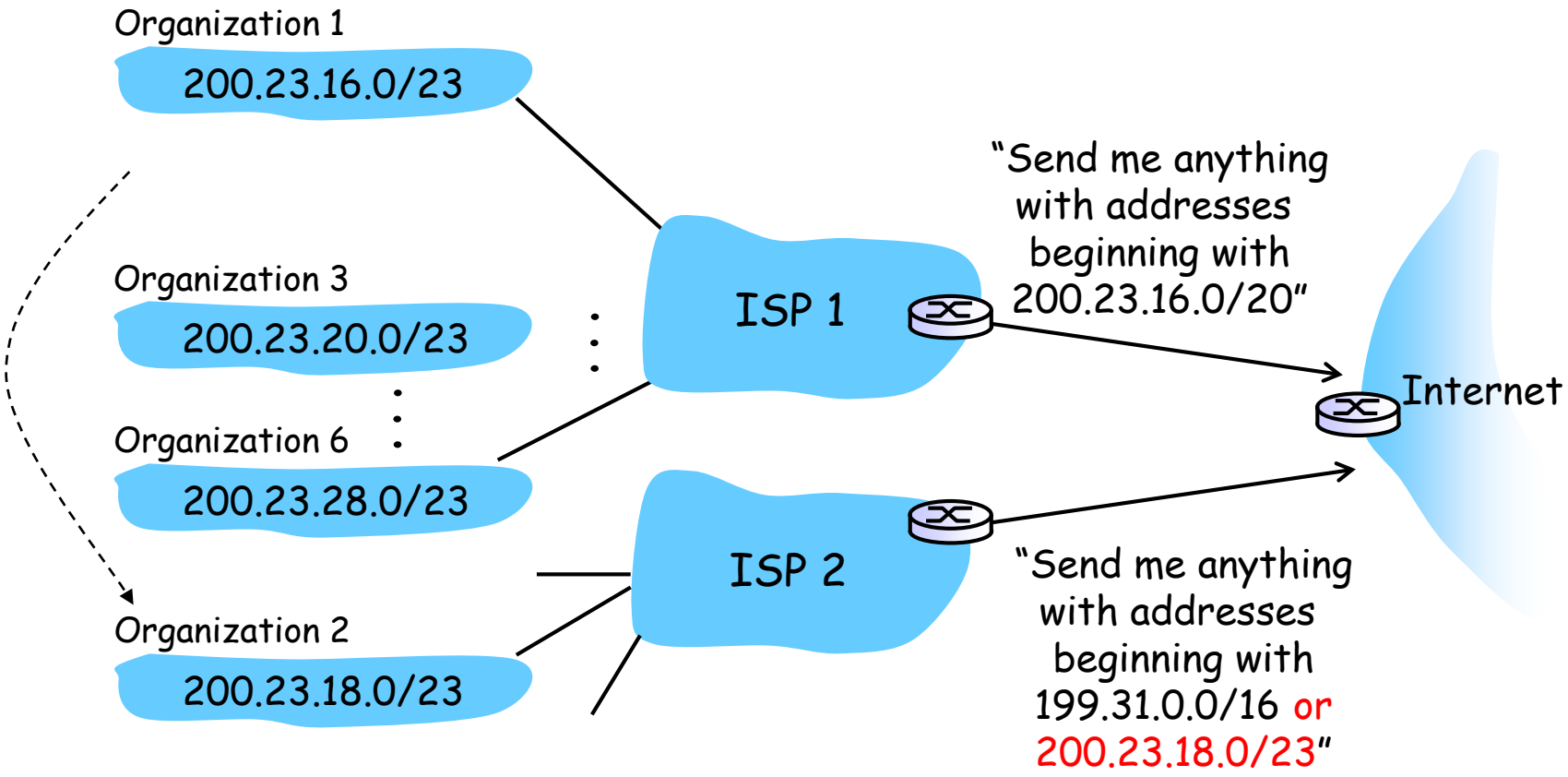
Hierarchical Addressing

Hierarchical addressing allows efficient advertisement of routing information:



Hierarchical Addressing

Suppose Organization 2 now switches to ISP 2, but doesn't want to renumber all of its routers and hosts.

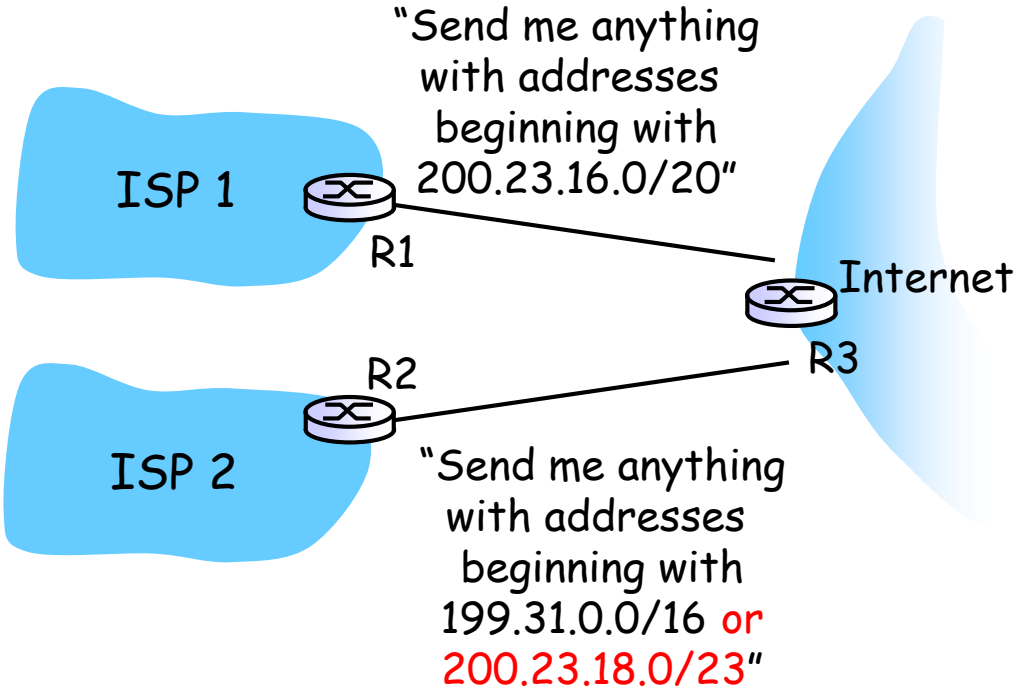


Longest Prefix Match (1/2)

- Question: which router to deliver to,
  - if a packet has destination IP 200.23.20.2?
  - if a packet has destination IP 200.23.19.3?

Forwarding Table at R3

Net mask	Next hop
200.23.16.0/20	R1
200.23.18.0/23	R2
199.31.0.0/16	R2
…	…



Longest Prefix Match (2/2)

- Packet with destination IP 200.23.20.2 ⇨ R1
  - (Binary: 11001000 00010111 00010100 00000010)
- Packet with destination IP 200.23.19.3 ⇨ R2
  - (Binary: 11001000 00010111 00010011 00000011)

match the longest prefix

Forwarding Table at R3

Net mask	Net mask in binary	Next hop
200.23.16.0/20	11001000 00010111 00010000 00000000	R1
200.23.18.0/23	11001000 00010111 00010010 00000000	R2
199.31.0.0/16	11000111 00011111 00000000 00000000	R2
…		…

IP Address Allocation

- Q: How does an ISP get a block of addresses?
- A: ICANN: Internet Corporation for Assigned Names and Numbers
  - Allocates addresses
  - Manages DNS
  - Assigns domain names, resolves disputes

Lectures 6&7: Roadmap

4.1 Overview of Network Layer

4.2 What's Inside a Router

4.3 The Internet Protocol (IP)

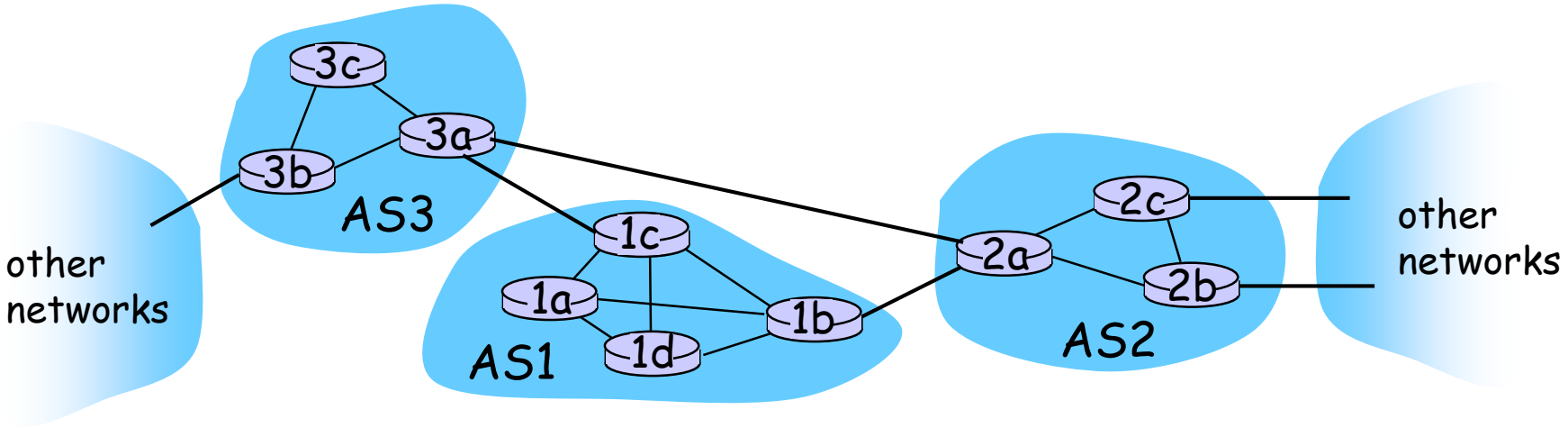
5.2 Routing Algorithms

- 5.2.2 The Distance Vector Routing Algorithm

5.6 ICMP

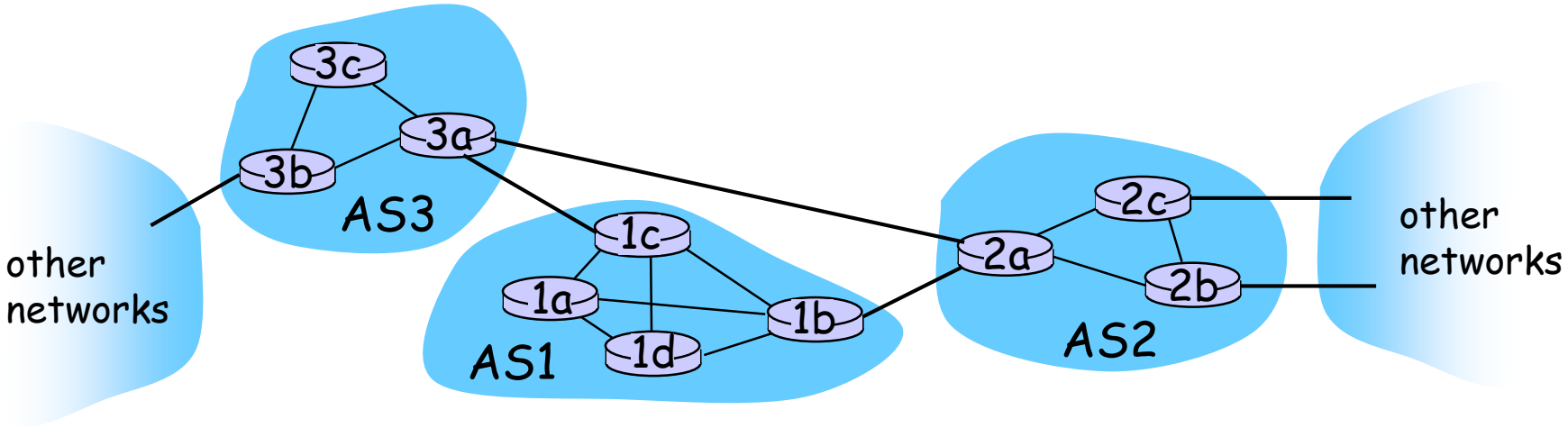
Internet: Network of Networks

- The Internet is a "network-of-networks".
  - A hierarchy of Autonomous Systems (AS), e.g., ISPs, each owns routers and links.
- Due to the size of the Internet and the decentralized administration of the Internet, routing on the Internet is done hierarchically.



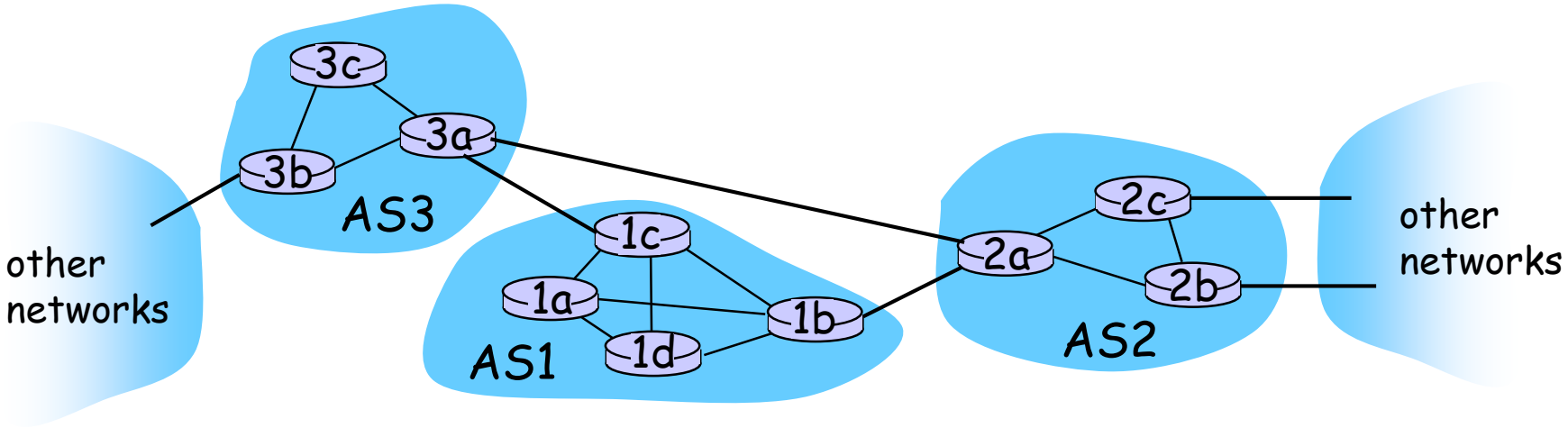
Routing in The Internet

- Intra-AS routing
  - Finds a good path between two routers within an AS.
  - Commonly used protocols: **RIP**, **OSPF**
- Inter-AS routing (not covered)
  - Handles the interfaces between ASs.
  - The de facto standard protocol: **BGP**

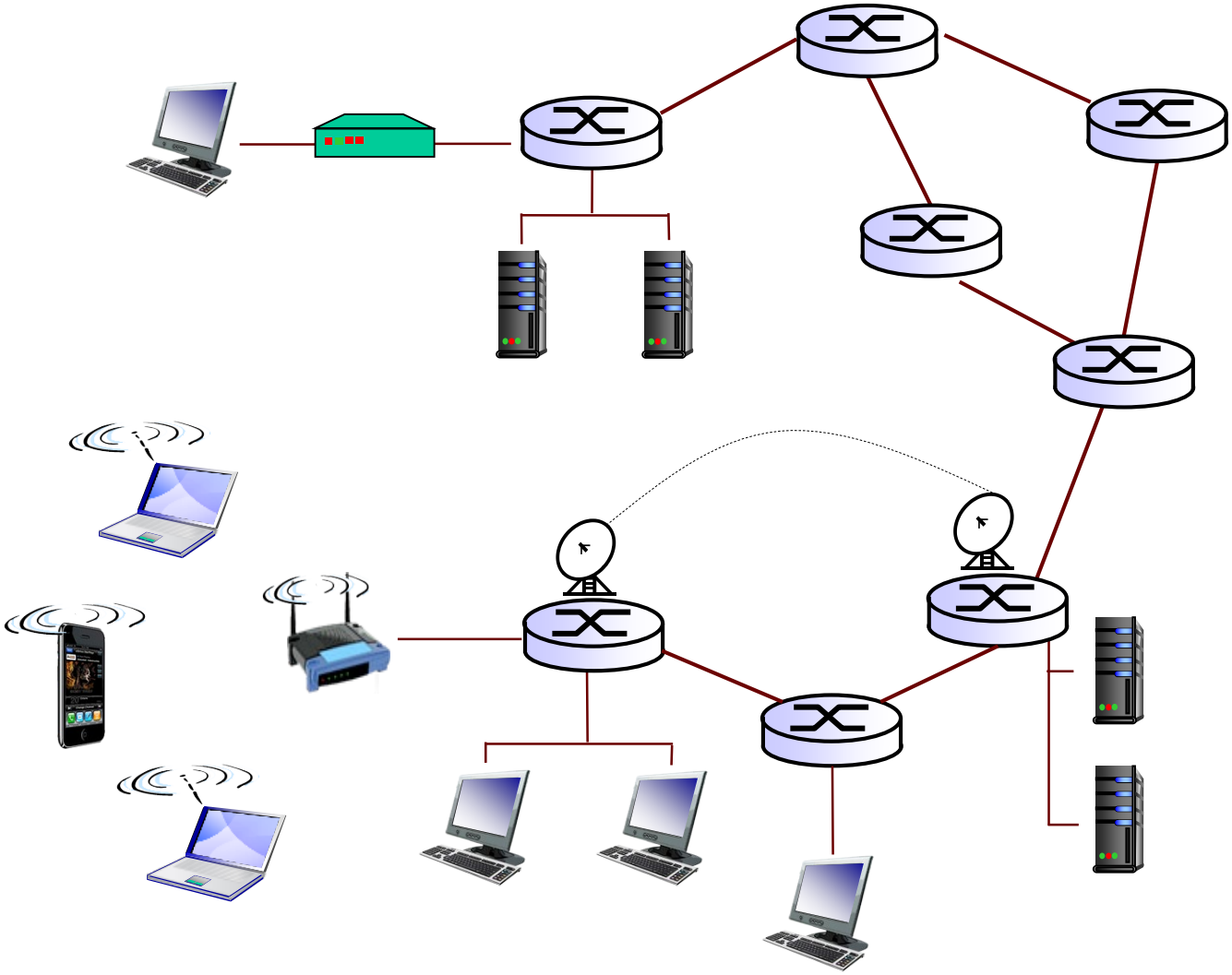


Routing in The Internet

- Intra-AS routing
  - Single admin, so no policy decisions are needed.
  - Routing mostly focus on performance.
- Inter-AS routing (not covered)
  - Admin often wants to control over how its traffic is routed, who routes through its net, etc.
  - Policy may dominate over performance.

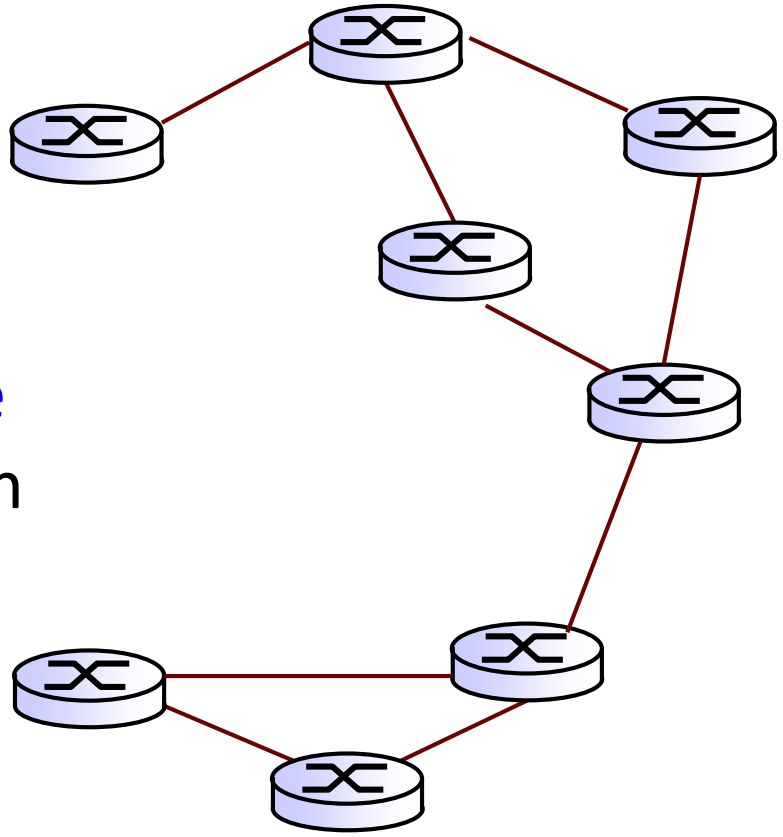


Abstract View of Intra-AS Routing



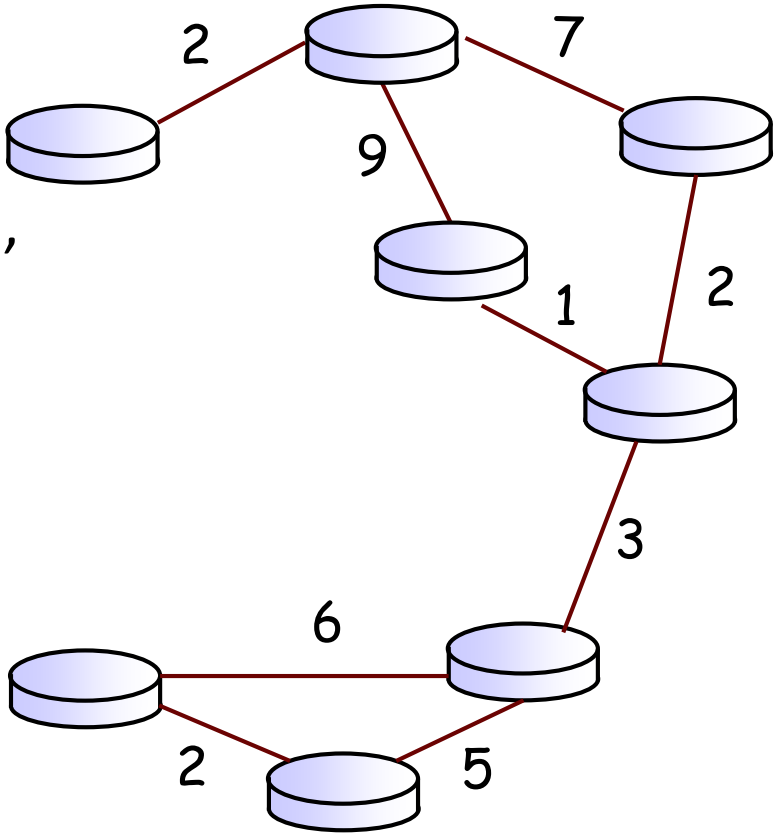
Abstract View of Intra-AS Routing

- We can abstractly view a network of routers as a graph, where vertices are routers and edges are physical links between routers.



Abstract View of Intra-AS Routing

- We can associate a cost to each link.
  - cost could always be 1, or inversely related to bandwidth, or related to congestion.

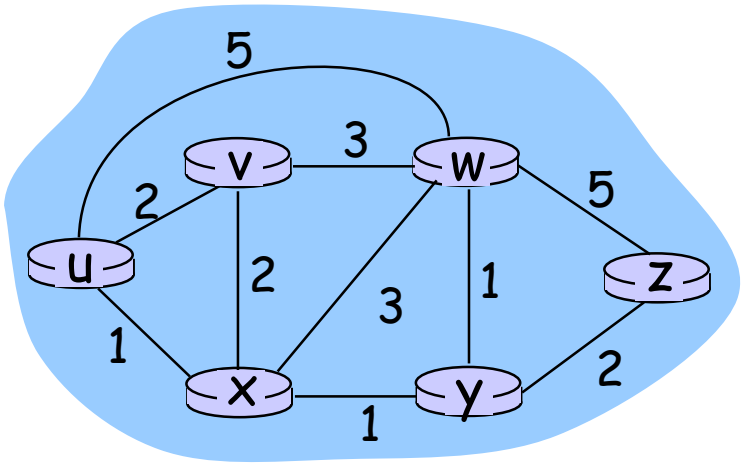


Routing: finding a least cost path between two vertices in a graph

Routing Algorithms Classification

"link state" algorithms

- All routers have the complete knowledge of network topology and link cost.
  - Routers periodically broadcast link costs to each other.
- Use Dijkstra algorithm to compute least cost path locally (using global map).
- Non-examinable ☺

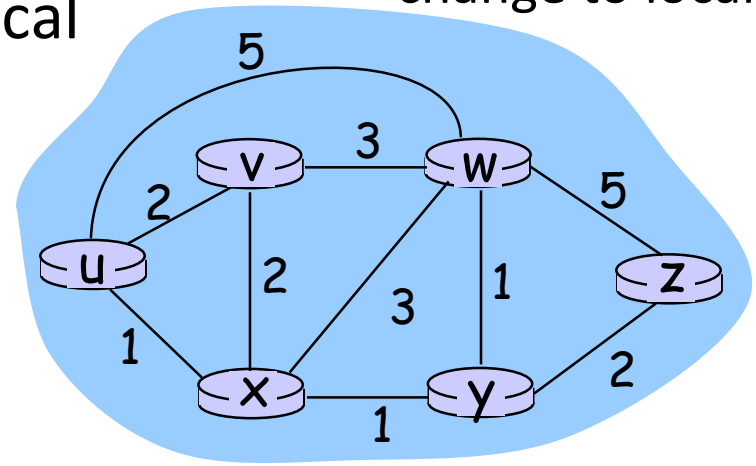


Routing Algorithms Classification

"distance vector" algorithms

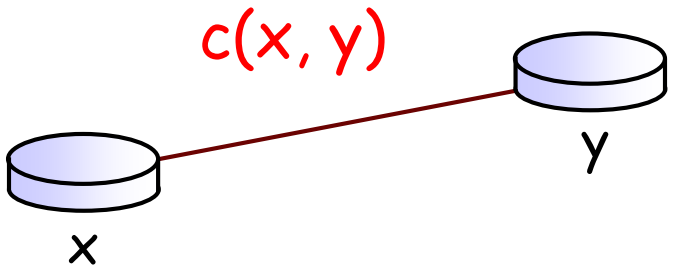
- Routers know physically-connected neighbors and link costs to neighbors.
- Routers exchange "local views" with neighbors and update own "local views" (based on neighbors' view).

- Iterative process of computation
  1. Swap local view with direct neighbours.
  2. Update own's local view.
  3. Repeat 1 - 2 till no more change to local view.

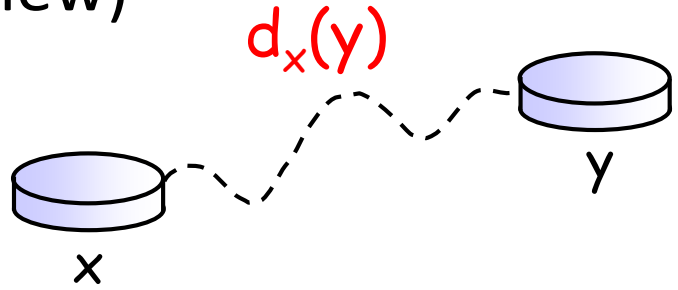


Some Graph Notations

- $c(x, y)$: the cost of link between routers x and y
  - = ∞ if x and y are not direct neighbours



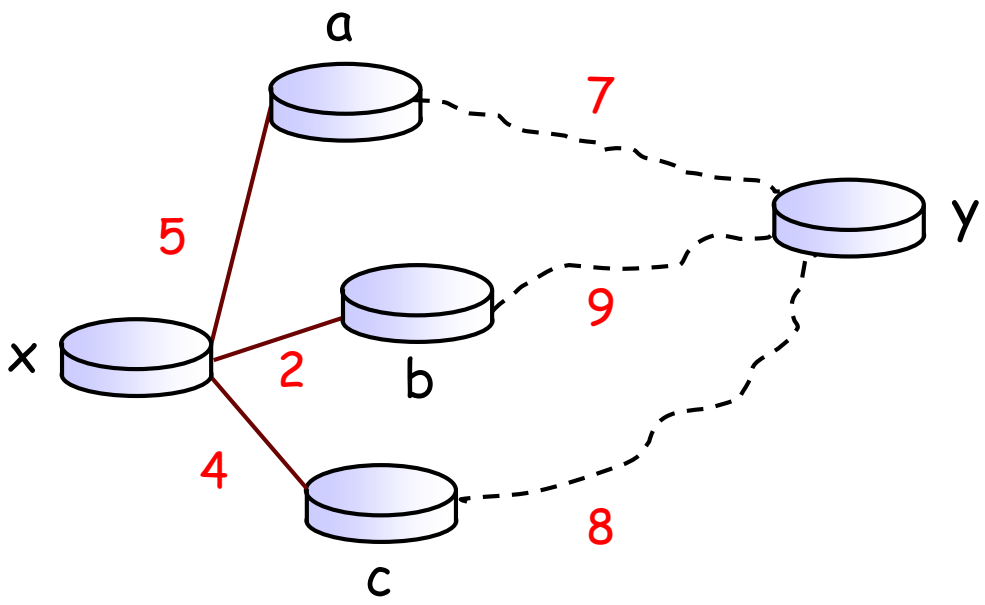
- $d_x(y)$: the cost of the least-cost path from x to y (from x's view)



Bellman-Ford Equation

$$d_x(y) = min_v \{c(x,v) + d_v(y)\}$$

where min is taken over all direct neighbors v of x

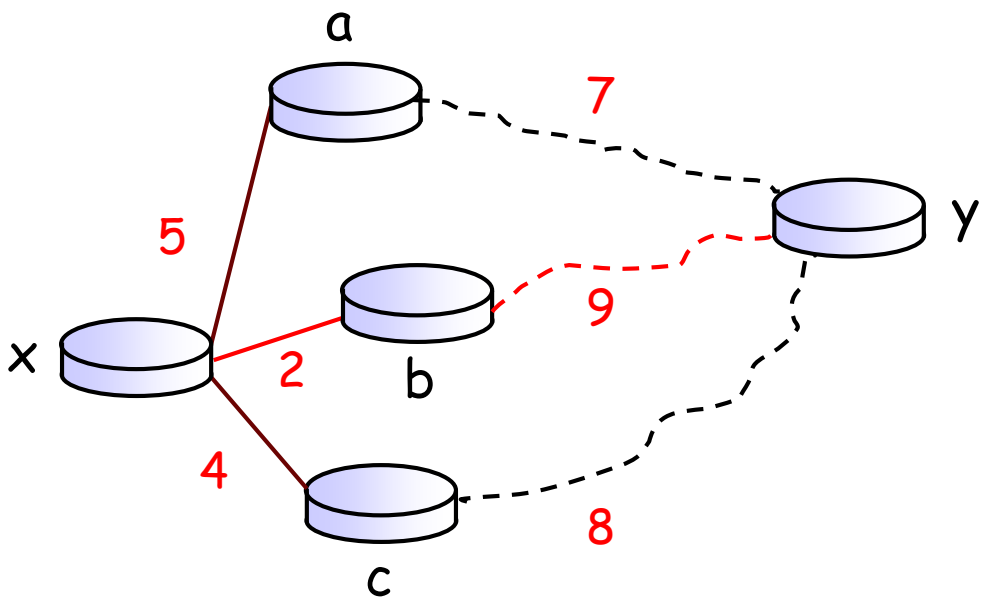


$$d_x(y) = min_v \{ c(x,a) + d_a(y),$$
$$c(x,b) + d_b(y),$$
$$c(x,c) + d_c(y) \}$$

$$= min \{12, 11, 12\} = 11$$

Bellman-Ford Equation

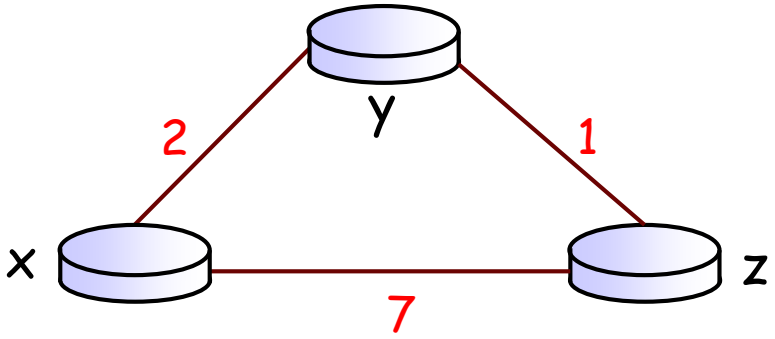
- To find the least cost path, x needs to know the cost from each of its direct neighbour to y.
- Each neighbour v sends its distance vector (y, k) to x, telling x that the cost from v to y is k.



Now x knows, to reach y, packet should be forward to b and the total cost would be 11.

Bellman-Ford Example

$$d_x(y) = min_v \{c(x,v) + d_v(y)\}$$



cost to

from	x	y	z
x			
y			
z			

x' view

cost to

from	x	y	z
x			
y			
z			

y' view

cost to

from	x	y	z
x			
y			
z			

z' view

Distance Vector Algorithm

- Every router, **x**, **y**, **z**, sends its distance vectors to its directly connected neighbors.
- When **x** finds out that **y** is advertising a path to **z** that is cheaper than **x** currently knows,
  - **x** will update its distance vector to **z** accordingly.
  - In addition, **x** will note down that all packets for **z** should be sent to **y**. This info will be used to create forwarding table of **x**.
- After every router has exchanged several rounds of updates with its direct neighbors, all routers will know the least-cost paths to all the other routers.

RIP

- RIP (Routing Information Protocol) implements the DV algorithm. It uses hop count as the cost metric (i.e., insensitive to network congestion).
- Entries in the routing table are aggregated subnet masks (so we are routing to destination subnet).
- Exchange routing table every 30 seconds over UDP port 520.
- "Self-repair": if no update from a neighbour router for 3 minutes, assume neighbour has failed.

Lectures 6&7: Roadmap

4.1 Overview of Network Layer

4.2 What's Inside a Router

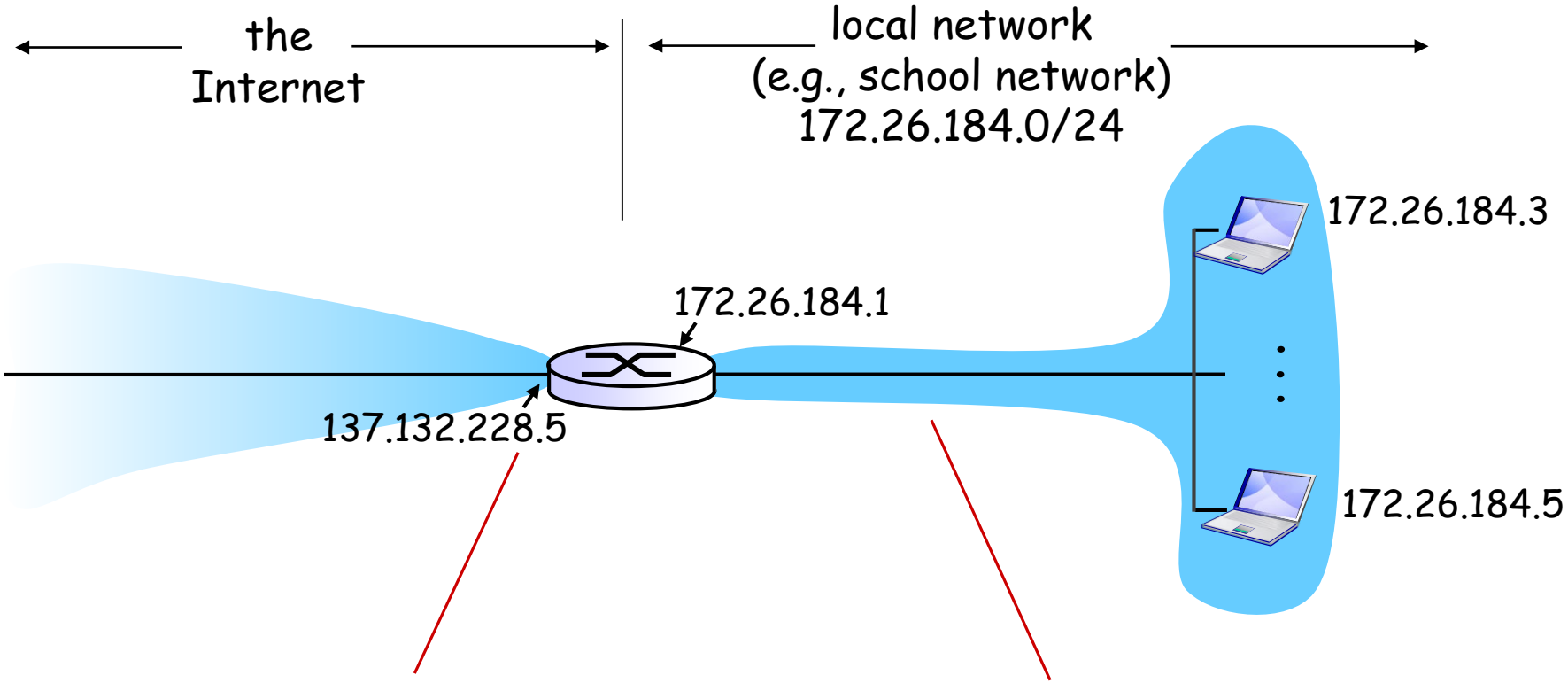
4.3 The Internet Protocol (IP)

- 4.3.4 Network Address Translation

5.2 Routing Algorithms

5.6 ICMP

NAT: Network Address Translation



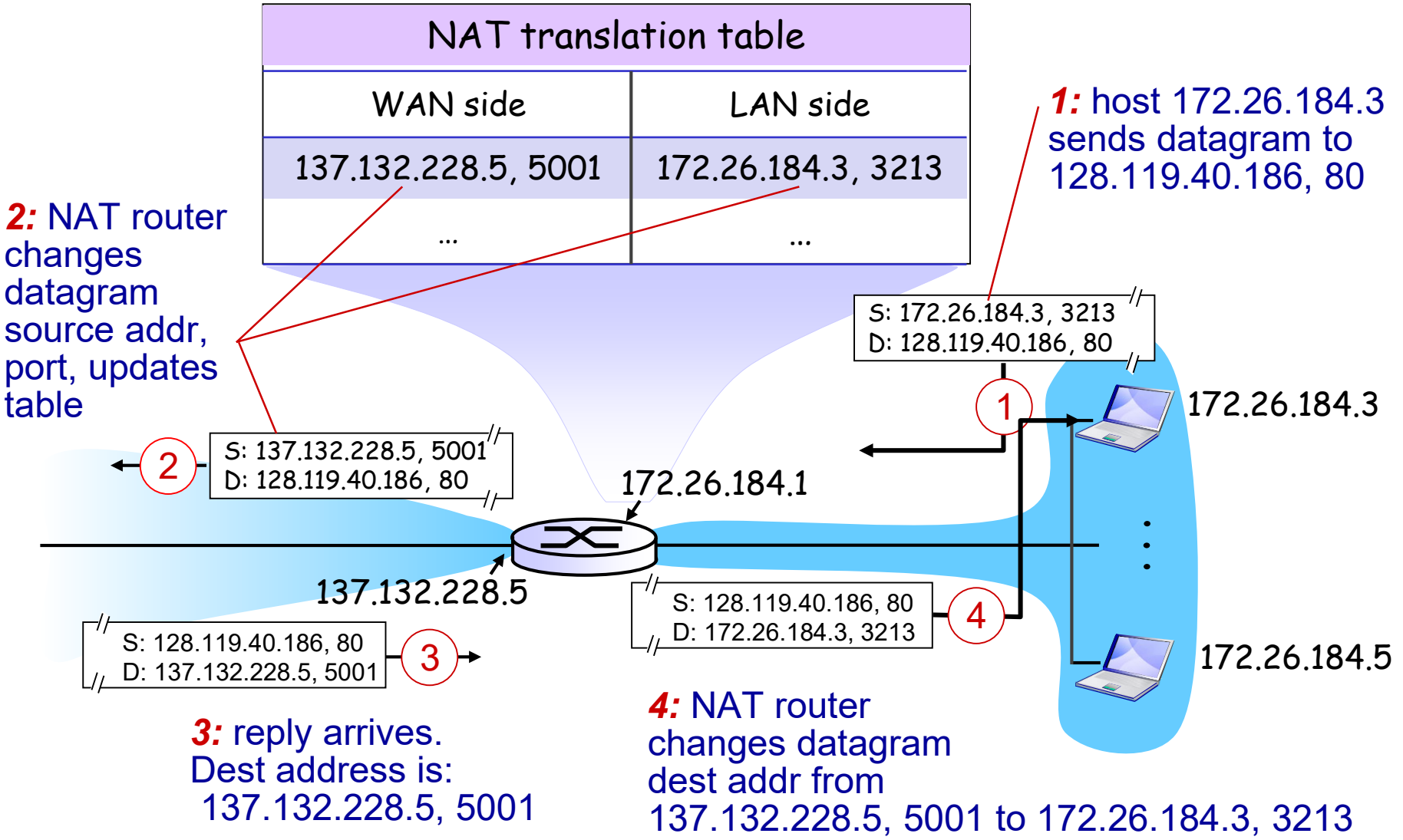
all datagrams *leaving* local network have the *same* source NAT IP address: 137.132.228.5

Within local network, hosts use private IP addresses 172.26.184.* for communication

NAT: Implementation

- NAT routers must:
  - Replace (source IP address, port #) of every outgoing datagram to (NAT IP address, new port #).
  - Remember (in NAT translation table) the mapping from (source IP address, port #) to (NAT IP address, new port #).
  - Replace (NAT IP address, new port #) in destination fields of every incoming datagram with corresponding (source IP address, port #) stored in NAT translation table.

NAT: Illustration



NAT: Motivation and Benefits

- No need to rent a range of public IP addresses from ISP: just one public IP for the NAT router.
- All hosts use private IP addresses. Can change addresses of hosts in local network without notifying the outside world.
- Can change ISP without changing addresses of hosts in local network.
- Hosts inside local network are not explicitly addressable and visible by outside world (a security plus).

Lectures 6&7: Roadmap

4.1 Overview of Network Layer

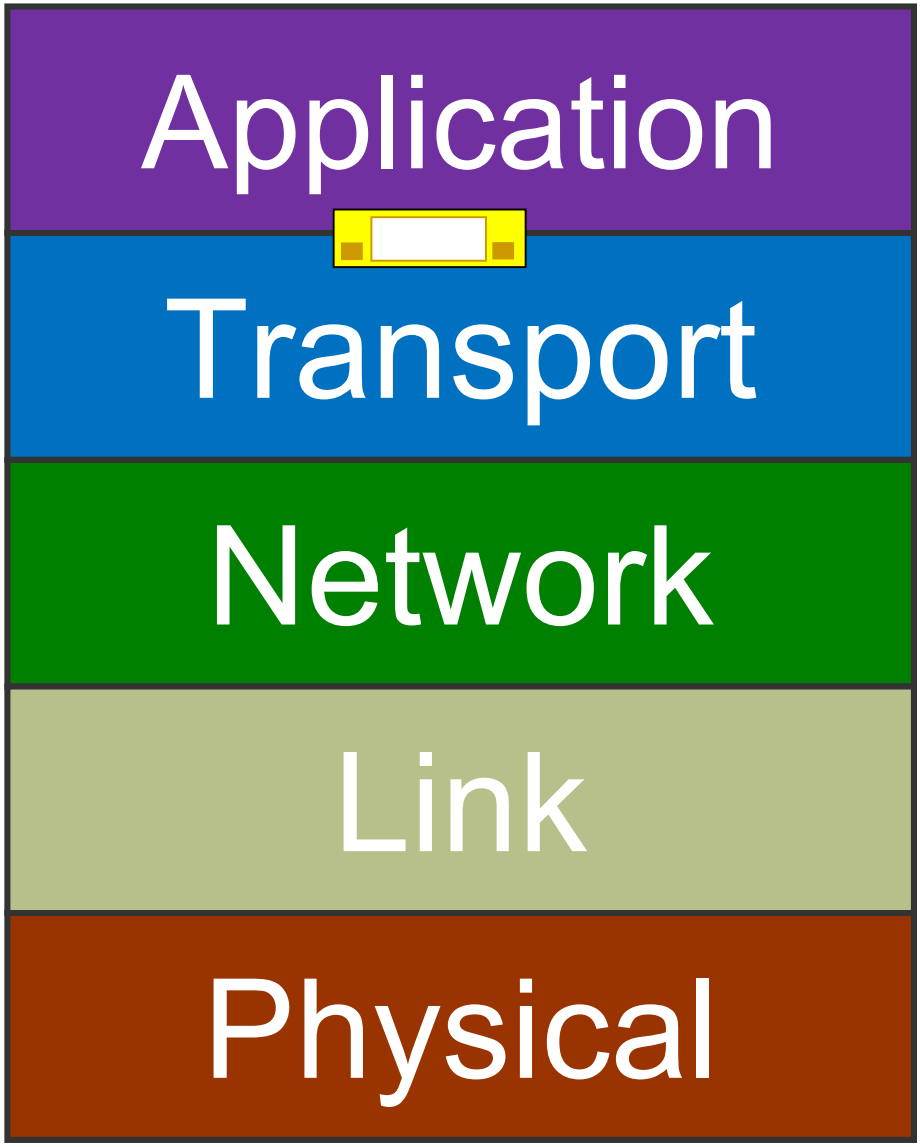
4.2 What's Inside a Router

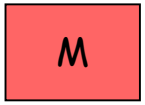
4.3 The Internet Protocol (IP)

- 4.3.1 IPv4 Datagram Format
- 4.3.2 IPv4 Datagram Fragmentation
- 4.3.5 IPv6 (non-examinable)

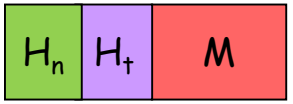
5.2 Routing Algorithms

5.6 ICMP


Application
Transport
Network
Link
Physical


M


H_t
M


H_n
H_t
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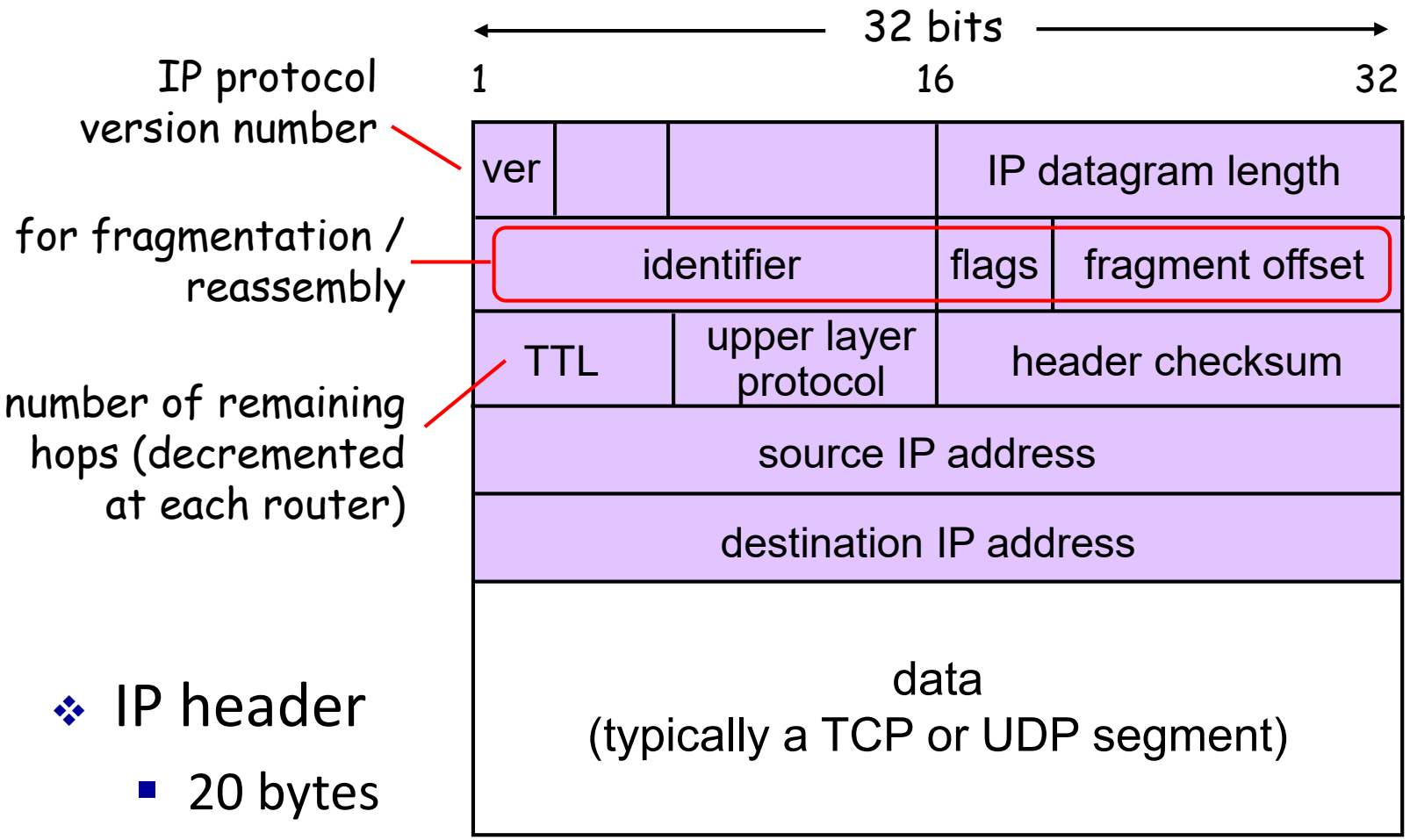
IPv4 Datagram Format

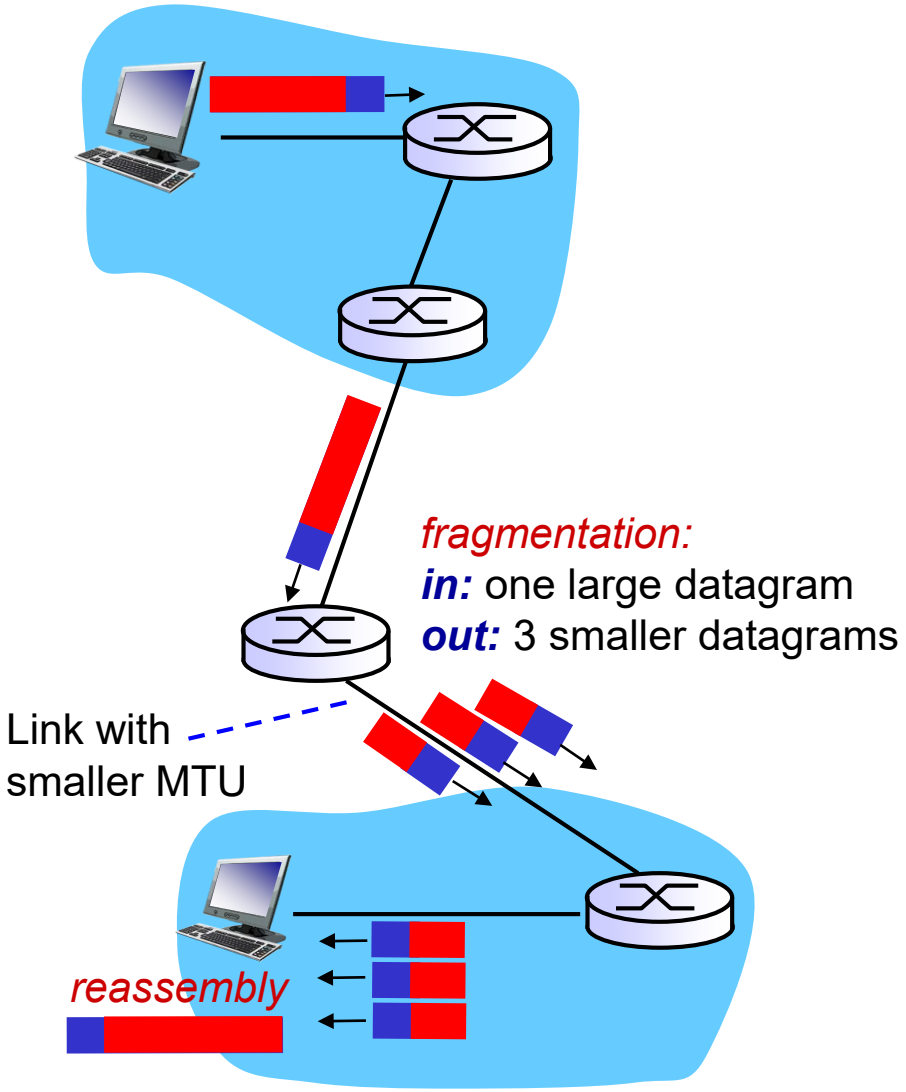


- IP header
  - 20 bytes

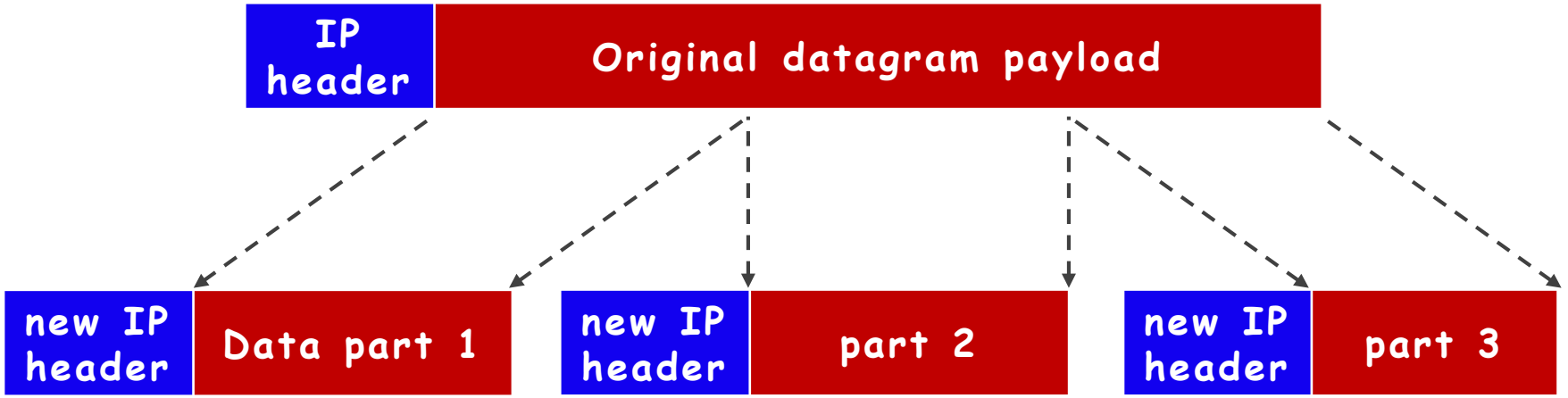
(some fields are not shown)

IP Fragmentation & Reassembly

- Different links may have different MTU (Max Transfer Unit) – the maximum amount of data a link-level frame can carry.
- "Too large" IP datagrams may be fragmented by routers.



IP Fragmentation Illustration



- Destination host will reassemble the packet.
- IP header fields are used to identify fragments and their relative order.

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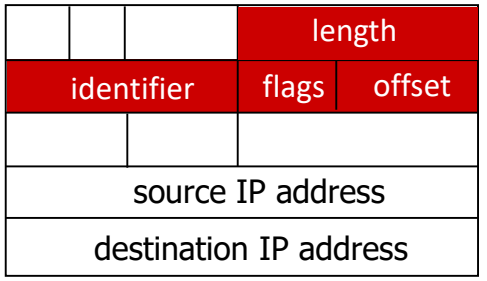
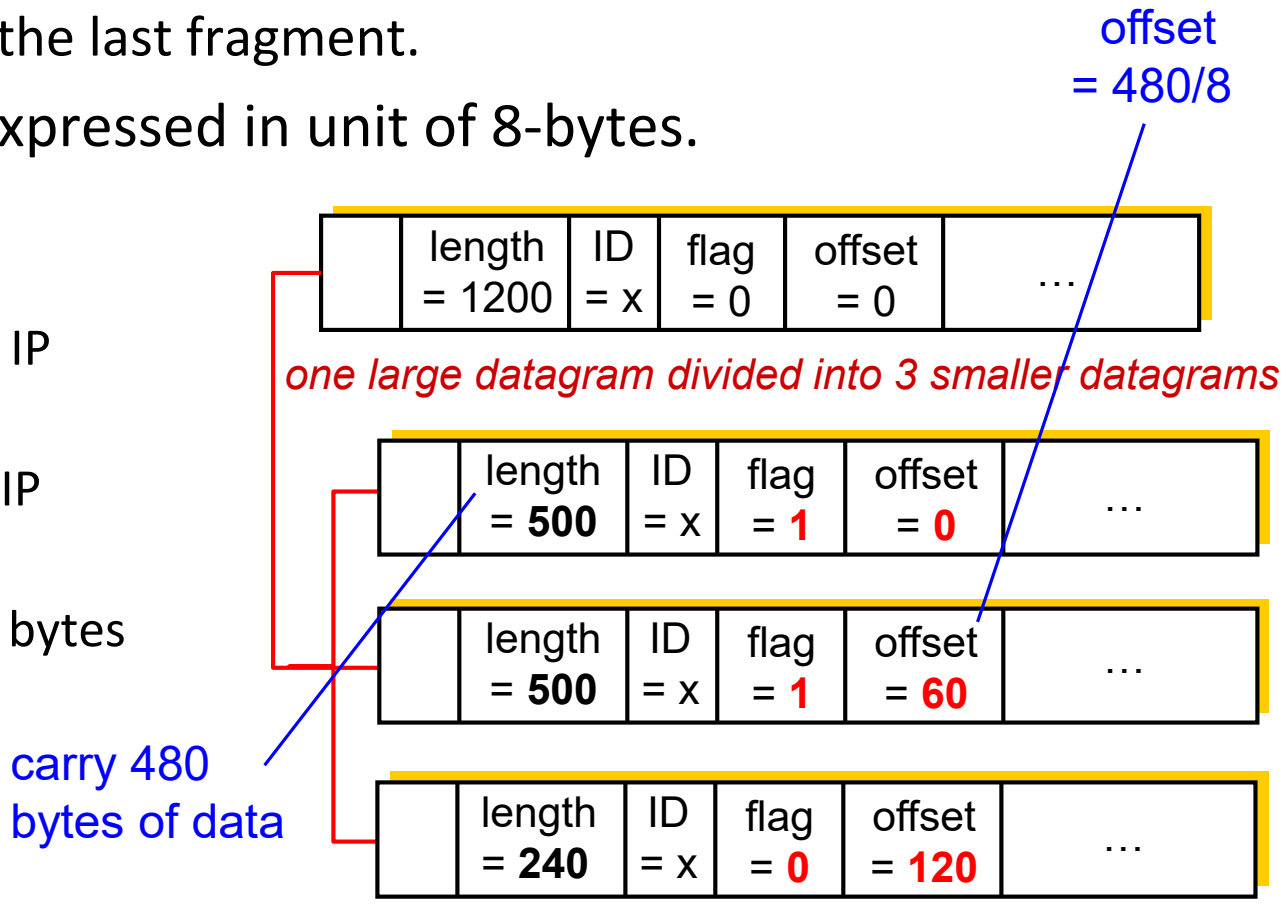


IP Fragmentation

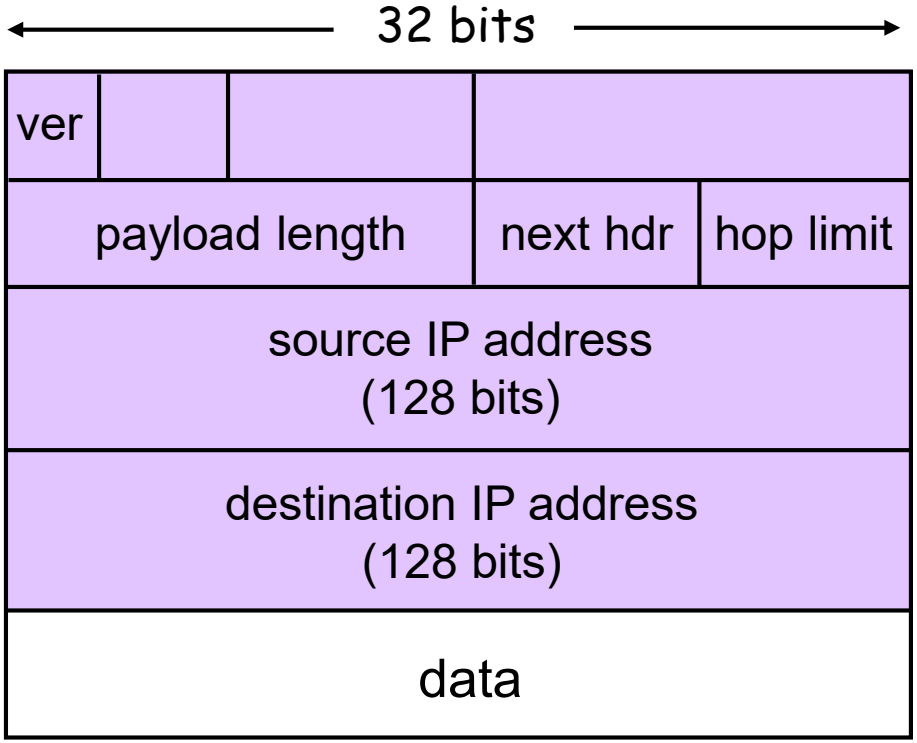
- Flag (frag flag) is set to
  - 1 if there is next fragment from the same segment.
  - 0 if this is the last fragment.
- Offset is in expressed in unit of 8-bytes.
- Example
  - 20 bytes of IP header
  - 1,200 byte IP datagram
  - MTU = 500 bytes



IPv6

Non-examinable

- IPv6 is designed to replace IPv4.
- Primary motivation: 32-bit IPv4 address space is soon to be completely allocated.
- IPv6 datagram:
  - 40 byte header



(some fields are not shown)

Example IPv6 address (in hexadecimal):

2001:0db8:85a3:0042:1000:8a2e:0370:7334

Lectures 6&7: Roadmap

4.1 Overview of Network Layer

4.2 What's Inside a Router

4.3 The Internet Protocol (IP)

5.2 Routing Algorithms

5.6 ICMP **Self-read**

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ICMP

Self-read

- ICMP (Internet Control Message Protocol) is used by hosts & routers to communicate network-level information.
  - Error reporting: unreachable host / network / port / protocol
  - Echo request/reply (used by ping)
- ICMP messages are carried in IP datagrams.
  - ICMP header starts after IP header.

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ICMP Type and Code

Self-read

- ICMP header: Type + Code + Checksum + others.

Type	Code	Description
8	0	echo request (ping)
0	0	echo reply (ping)
3	1	dest host unreachable
3	3	dest port unreachable
11	0	TTL expired
12	0	bad IP header

Selected ICMP Type and subtype (Code)

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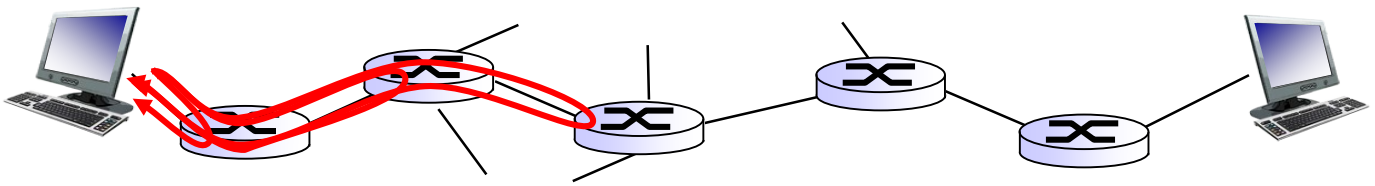
ping and *traceroute*

Self-read

- The command ping sees if a remote host will respond to us – do we have a connection?

- The command traceroute sends a series of small packets across a network, and attempts to display the route (or path) that the messages would take to get to a remote host.



Lectures 6&7: Summary

- An IP address is associated with a network interface. A device may have multiple network interfaces, thus multiple IP addresses.
- DHCP automates the assignment of IP addresses in an organization's network.
- On TCP/IP networks, subnets are defined as all devices whose IP addresses have the same network (subnet) prefix.
- Subnet mask is useful in checking if two hosts are on the same subnet.

Lectures 6&7: Summary

- Routing is the process of selecting best paths in a network.
- NAT maps one IP addresses space into another.
  - Commonly used to hide an entire private IP address space behind a single public IP address.
  - NAT router uses stateful translation tables to remember the mapping.
- ICMP is used by routers to send error messages.
  - E.g. when TTL is 0, a packet is discarded and an ICMP error message is sent to the datagram's source address.