Data Communications Network

- Chapter01: Computer Networks and the Internet
- Chapter02: Application Layer
- Chapter03: Transport Layer
- Chapter04: The Network Layer
- Chapter05: The Link Layer: Links, Access Networks, and LANs
- Chapter06: The physical layer

Chapter 1: introduction

Chapter goal:

- Get "feel," "big picture," introduction to terminology
  - more depth, detail *later* in course

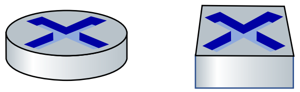
Overview/roadmap:

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

The Internet: a "nuts and bolts" view

Billions of connected computing *devices*:

- *hosts* = *end systems*
- running *network apps* at Internet's "edge"



Packet switches: forward packets (chunks of data)

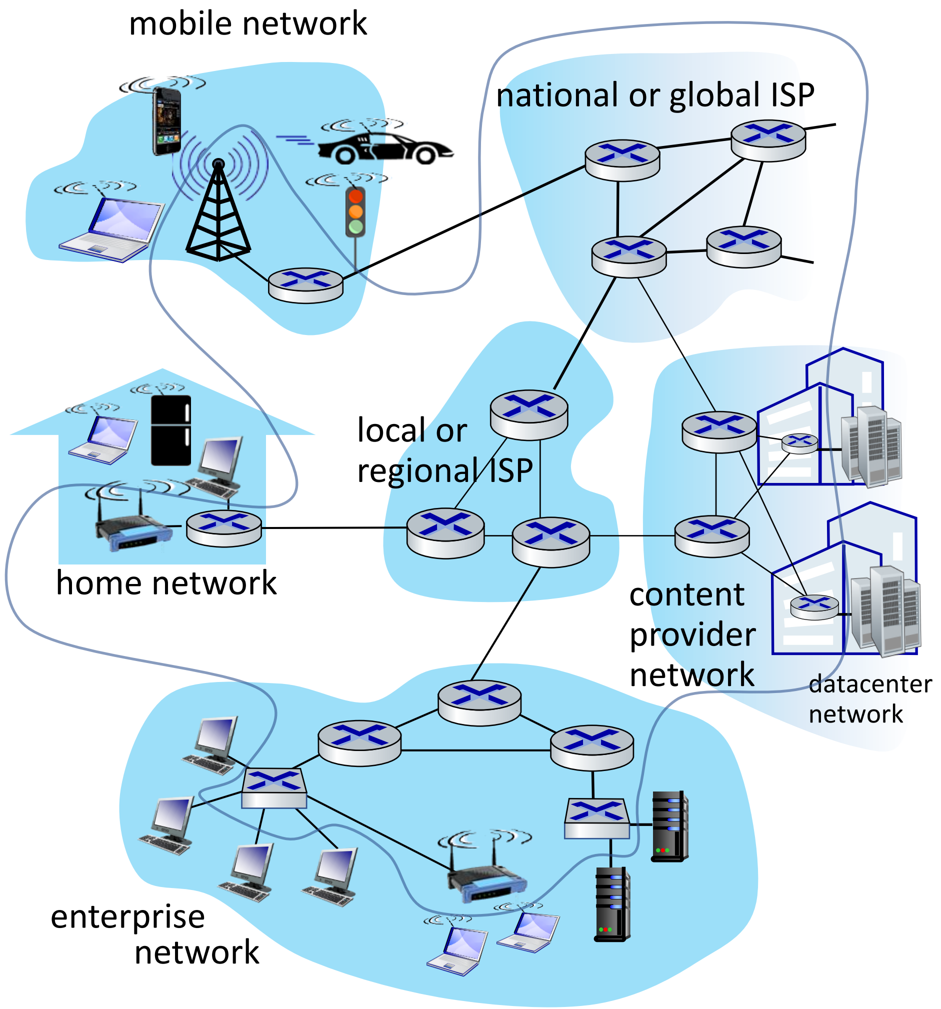
- *routers, switches*

Communication links

- fiber, copper, radio, satellite
- transmission rate: *bandwidth*

Networks

- collection of devices, routers, links: managed by an organization



"Fun" Internet-connected devices

Amazon Echo

Internet refrigerator

IP picture frame

Pacemaker & Monitor

Tweet-a-watt: monitor energy use

bikes

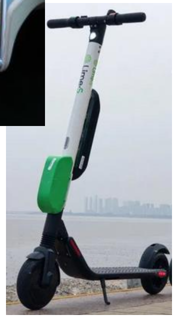
Security Camera

Slingbox: remote control cable TV

Web-enabled toaster + weather forecaster

cars

AR devices



scooters

Internet phones

Gaming devices

sensorized, bed mattress

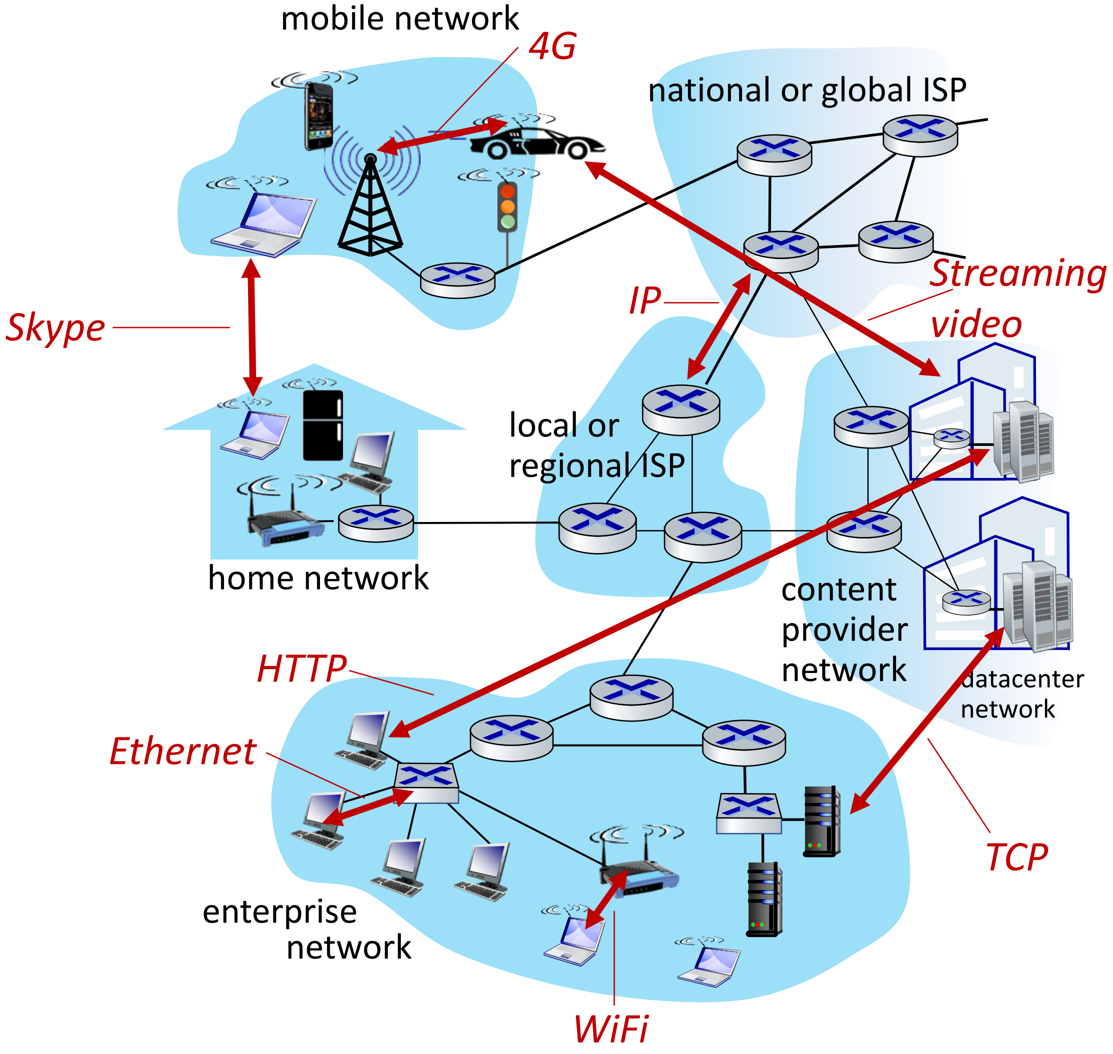
Fitbit

diapers

Others?

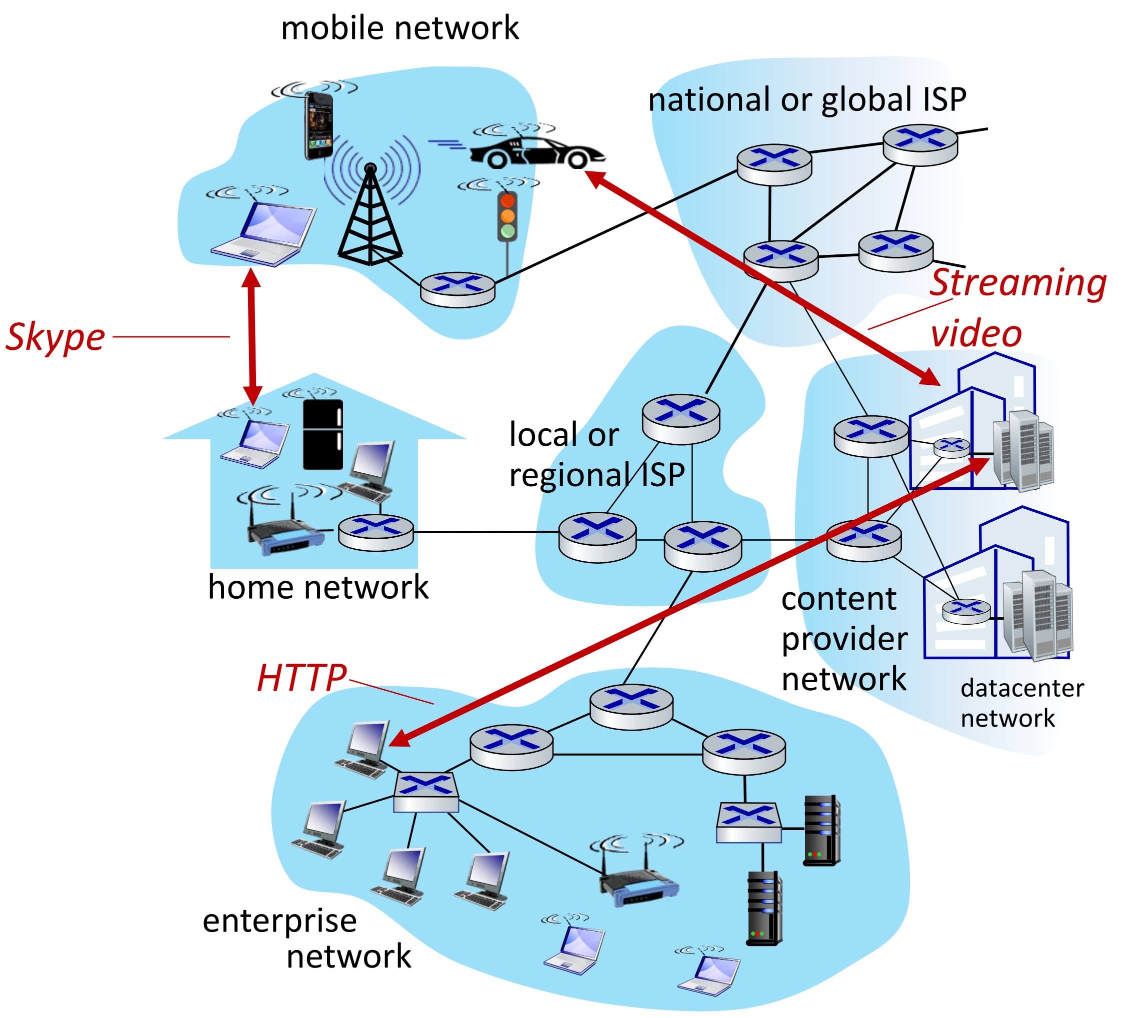
The Internet: a "nuts and bolts" view

- *Internet: "network of networks"*
  - Interconnected ISPs
- *protocols* are *everywhere*
  - control sending, receiving of messages
  - e.g., HTTP (Web), streaming video, Skype, TCP, IP, WiFi, 4/5G, Ethernet
- *Internet standards*
  - RFC: Request for Comments
  - IETF: Internet Engineering Task Force



The Internet: a "services" view

- *Infrastructure* that provides services to applications:
  - Web, streaming video, multimedia teleconferencing, email, games, e-commerce, social media, inter-connected appliances, …
- provides *programming interface* to distributed applications:
  - "hooks" allowing sending/receiving apps to "connect" to, use Internet transport service
  - provides service options, analogous to postal service



What's a protocol?

Human protocols:

- "what's the time?"
- "I have a question"
- introductions

Rules for:

… specific messages sent

… specific actions taken when message received, or other events

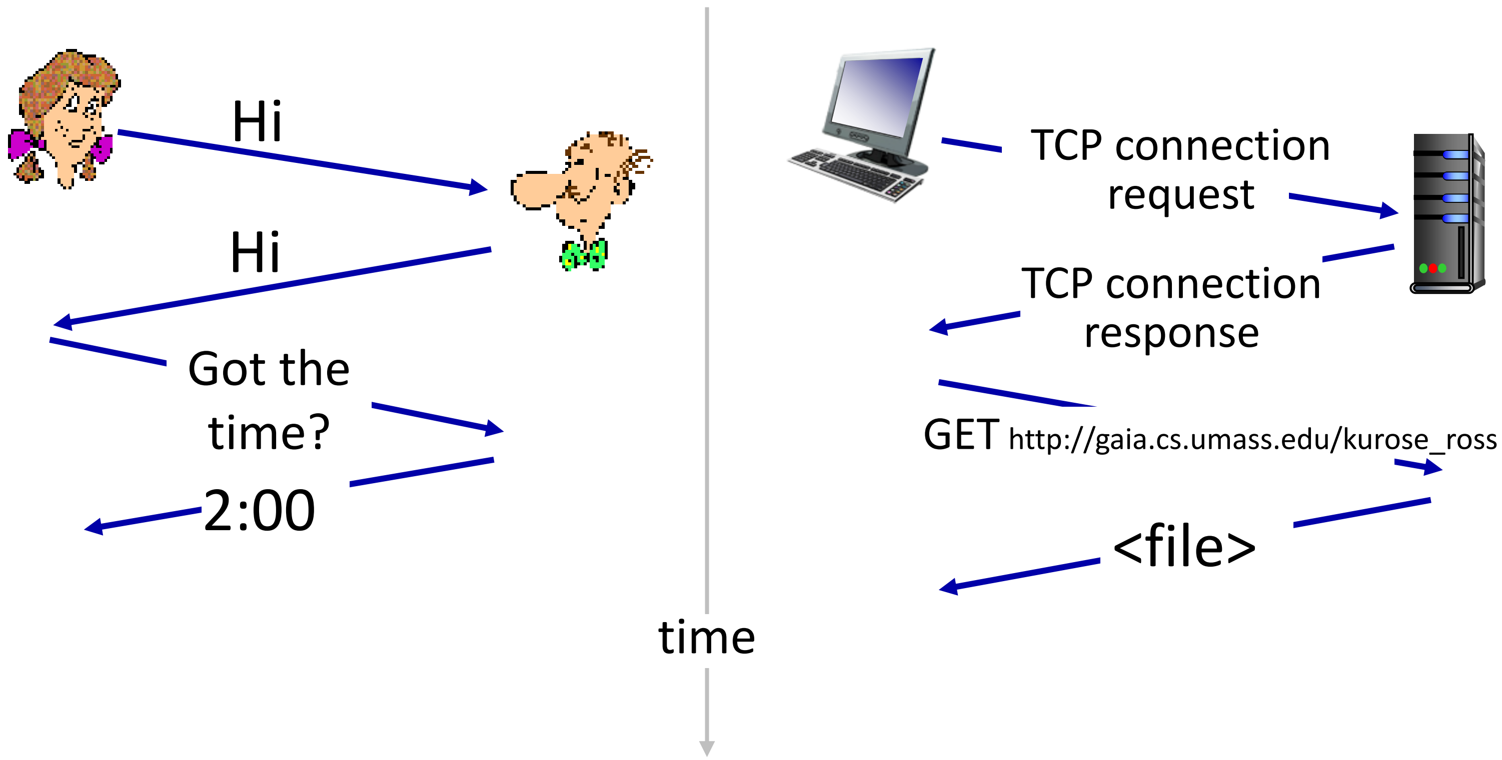
Network protocols:

- computers (devices) rather than humans
- all communication activity in Internet governed by protocols

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

What’s a protocol?

A human protocol and a computer network protocol:



Q: other human protocols?

Chapter 1: roadmap

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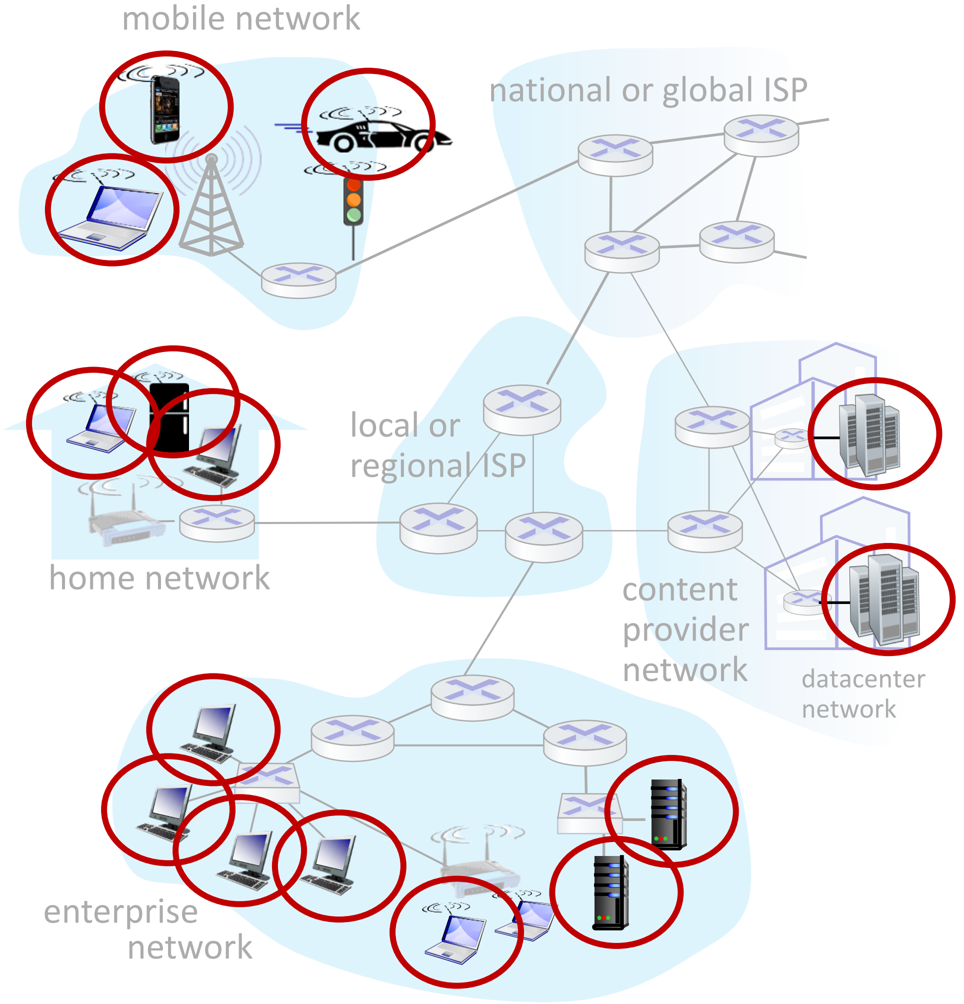
A closer look at Internet structure

Network edge:

- hosts: clients and servers
- servers often in data centers



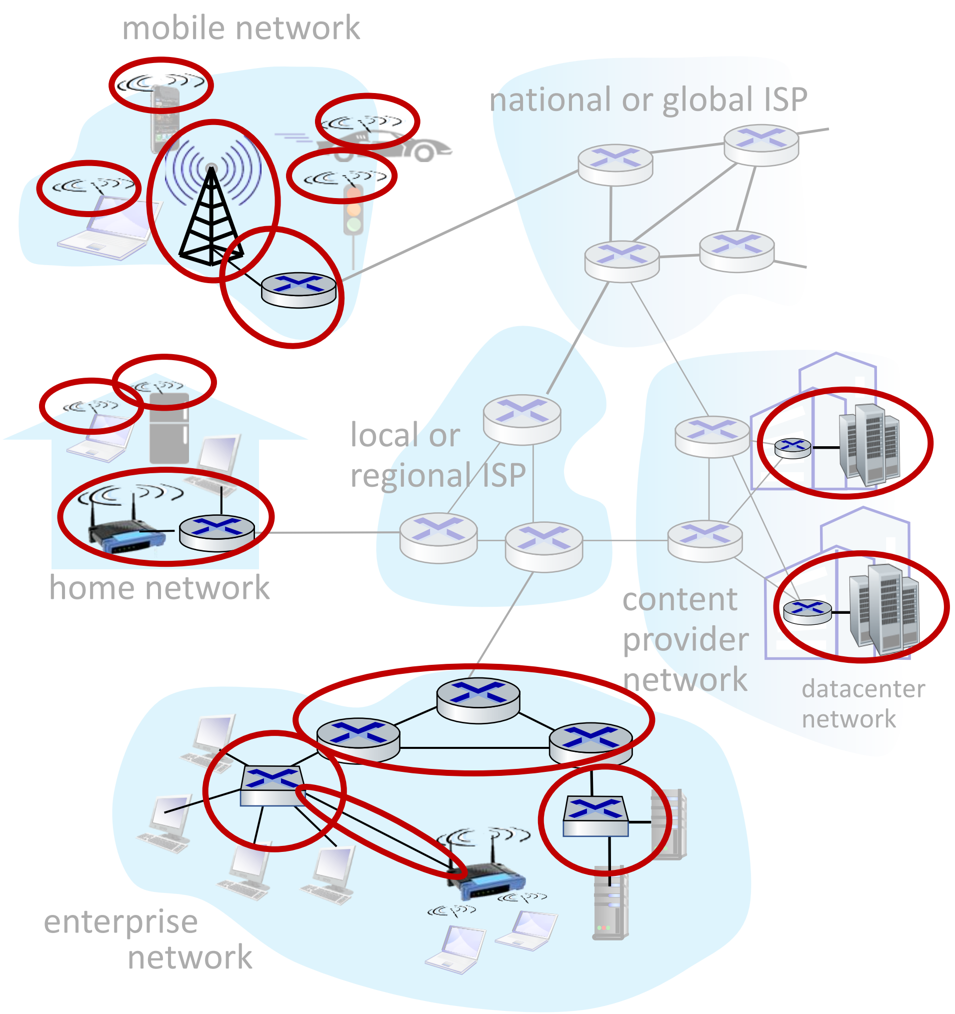
A closer look at Internet structure

Network edge:

- hosts: clients and servers
- servers often in data centers

Access networks, physical media:

- wired, wireless communication links



A closer look at Internet structure

Network edge:
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Access networks, physical media:
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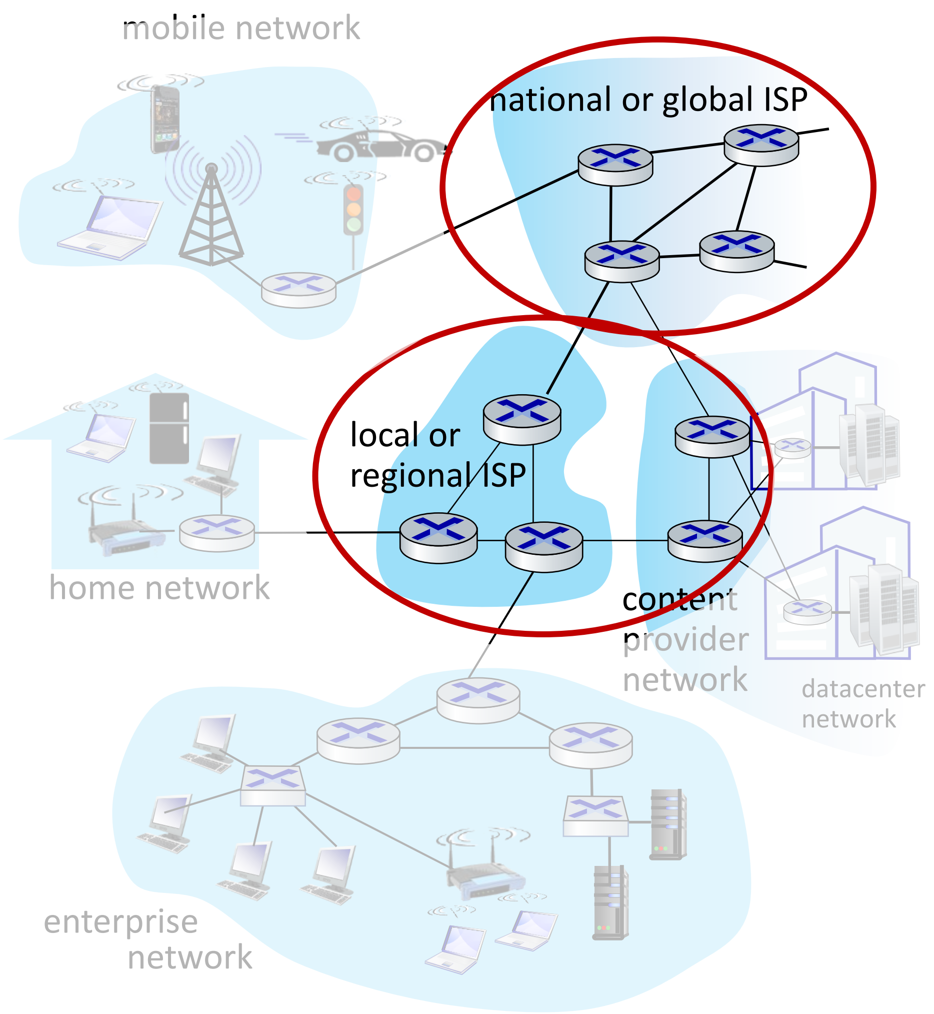
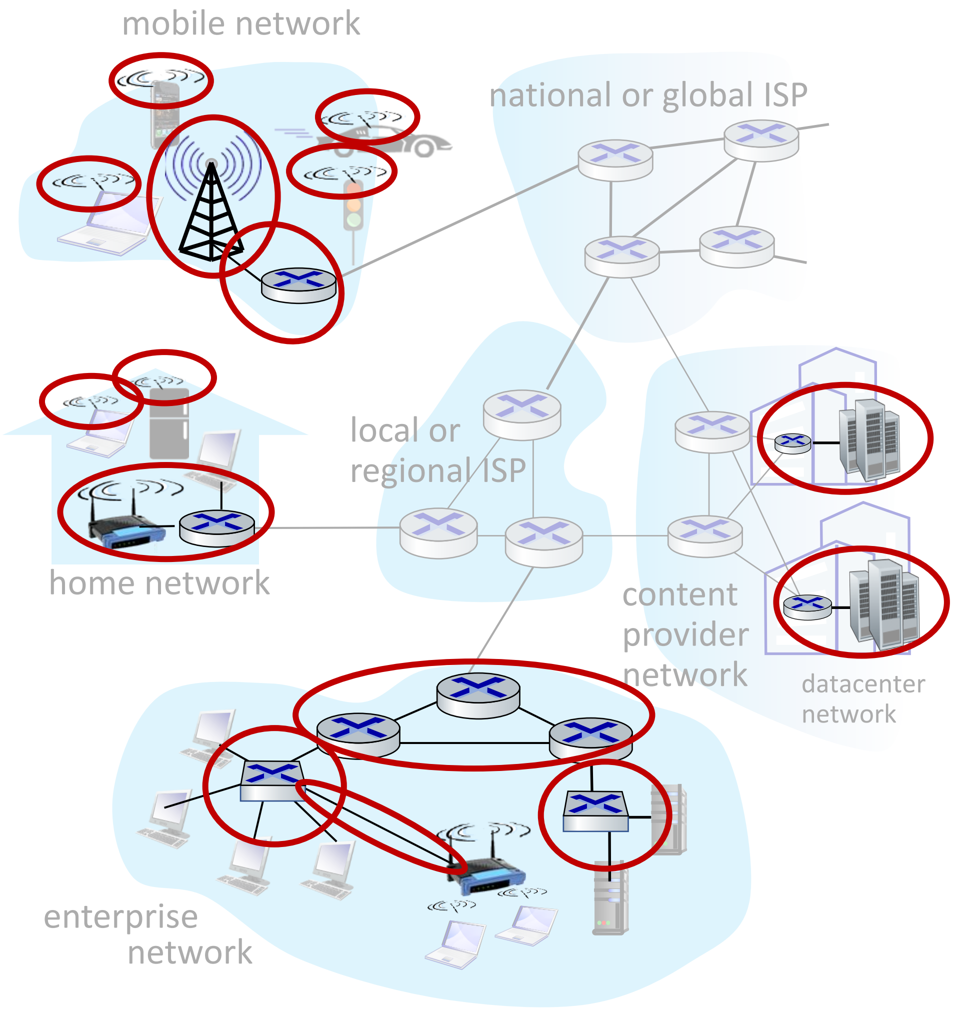
Network core:
- interconnected routers
- network of networks



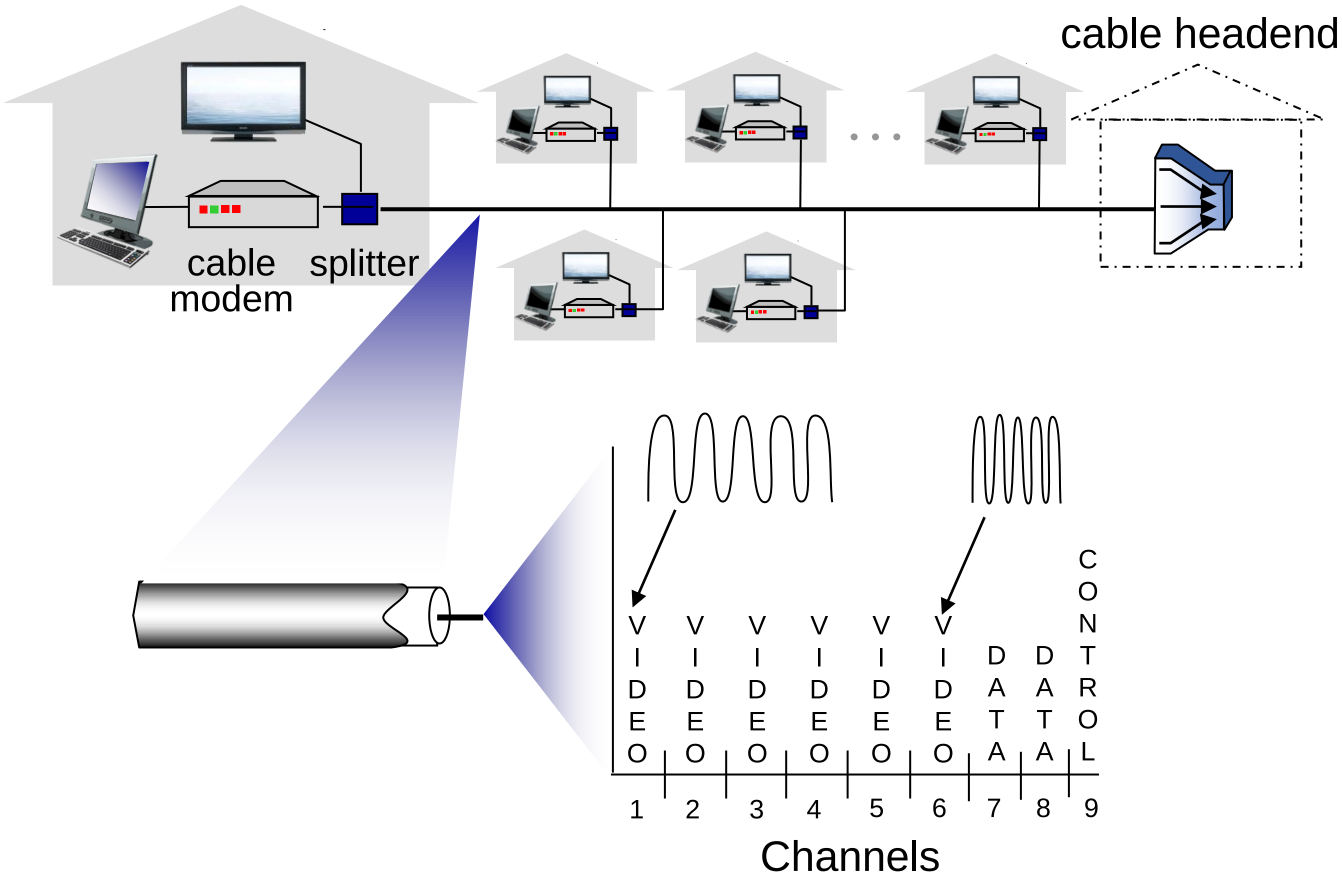
Access networks and physical media

Q: How to connect end systems to edge router?

- residential access nets
- institutional access networks (school, company)
- mobile access networks (WiFi, 4G/5G)

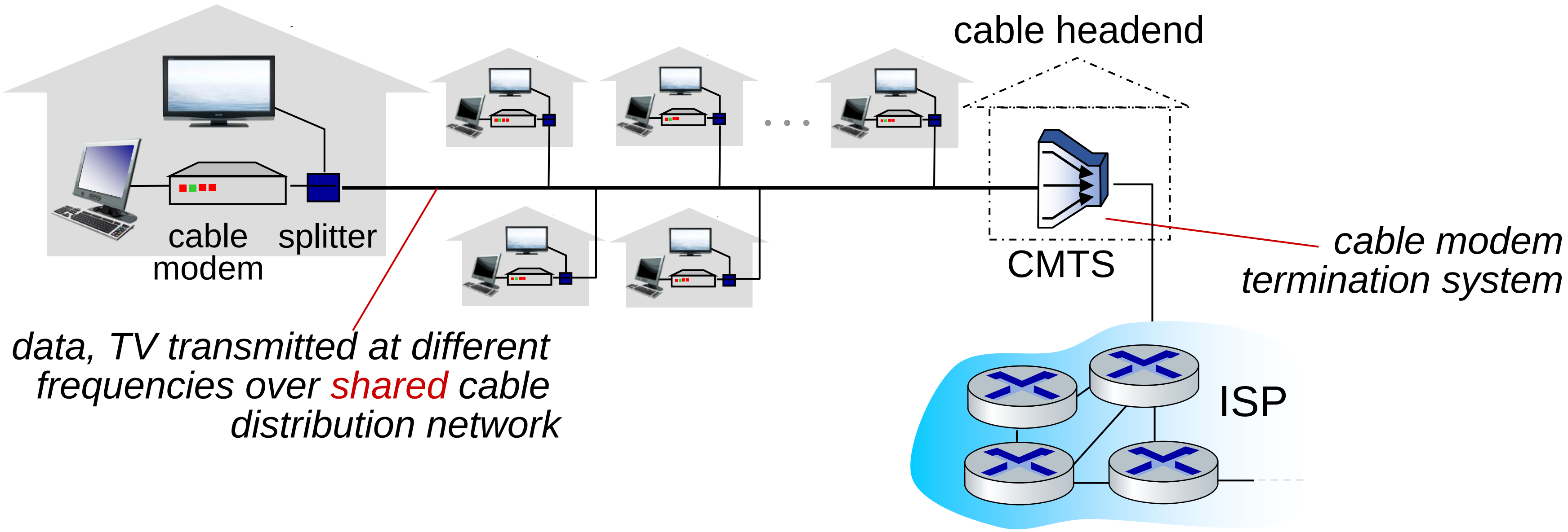


Access networks: cable-based access



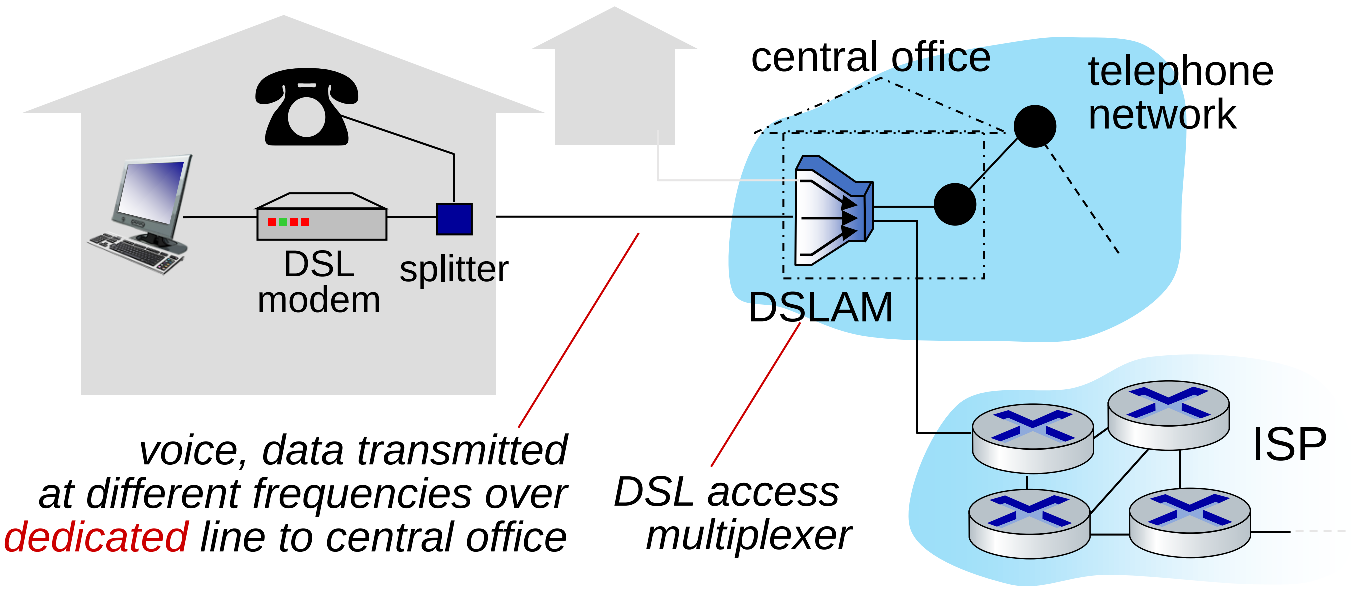
frequency division multiplexing (FDM): different channels transmitted in different frequency bands

Access networks: cable-based access



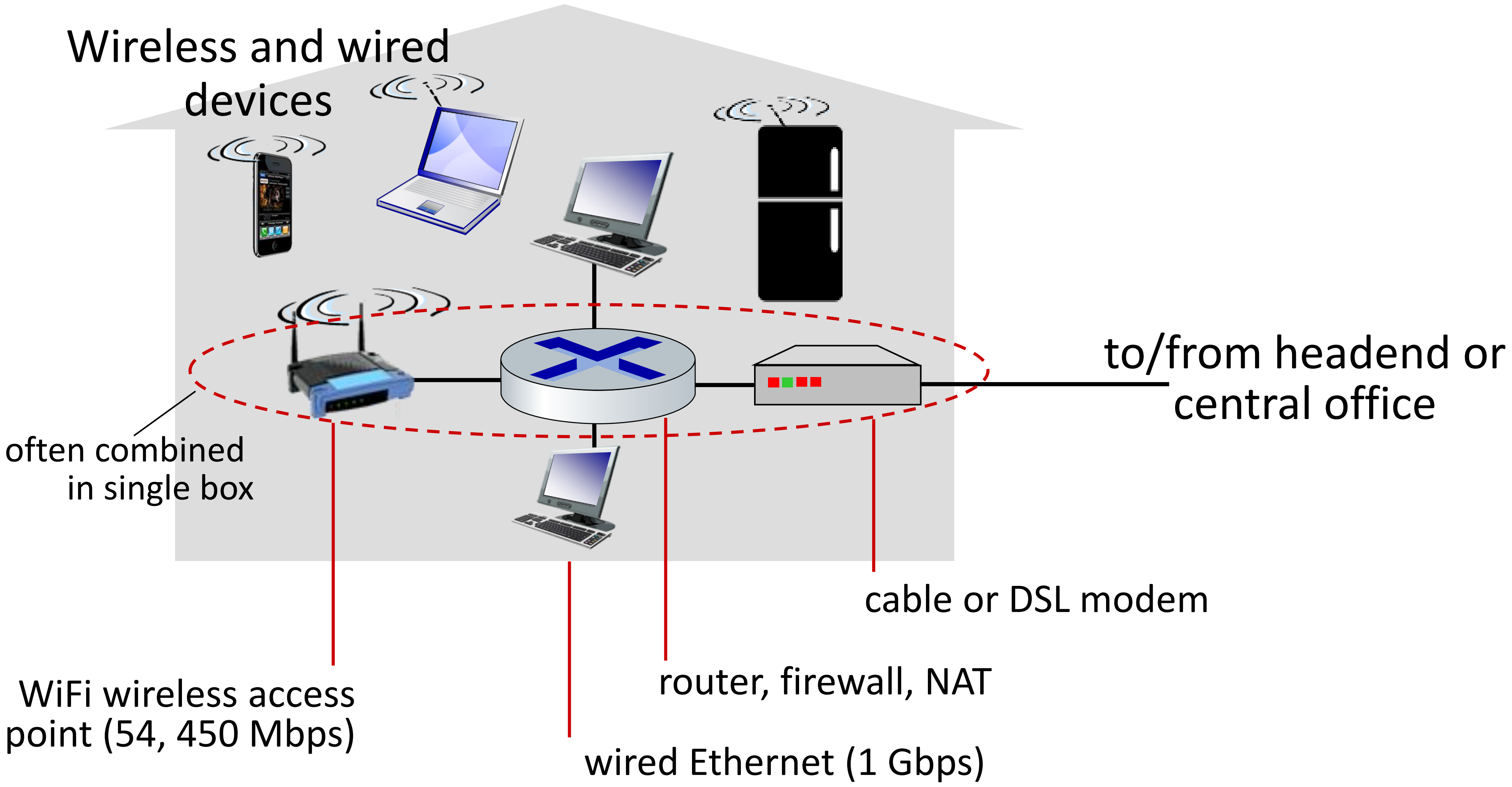
- HFC: hybrid fiber coax
  - asymmetric: up to 40 Mbps – 1.2 Gbps downstream transmission rate, 30-100 Mbps upstream transmission rate
- network of cable, fiber attaches homes to ISP router
  - homes *share access network* to cable headend

Access networks: digital subscriber line (DSL)



- use *existing* telephone line to central office DSLAM
  - data over DSL phone line goes to Internet
  - voice over DSL phone line goes to telephone net
- 24-52 Mbps dedicated downstream transmission rate
- 3.5-16 Mbps dedicated upstream transmission rate

Access networks: home networks



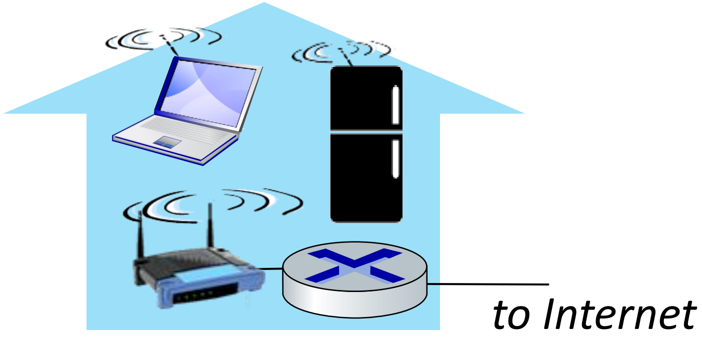
Wireless access networks

Shared *wireless* access network connects end system to router

- via base station aka “access point”

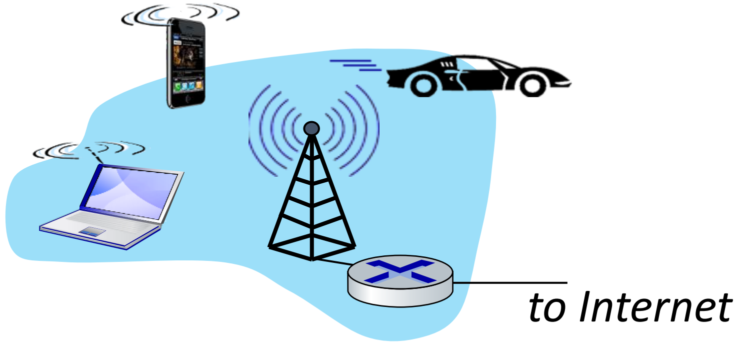
Wireless local area networks (WLANs)

- typically within or around building (~100 ft)
- 802.11b/g/n (WiFi): 11, 54, 450 Mbps transmission rate

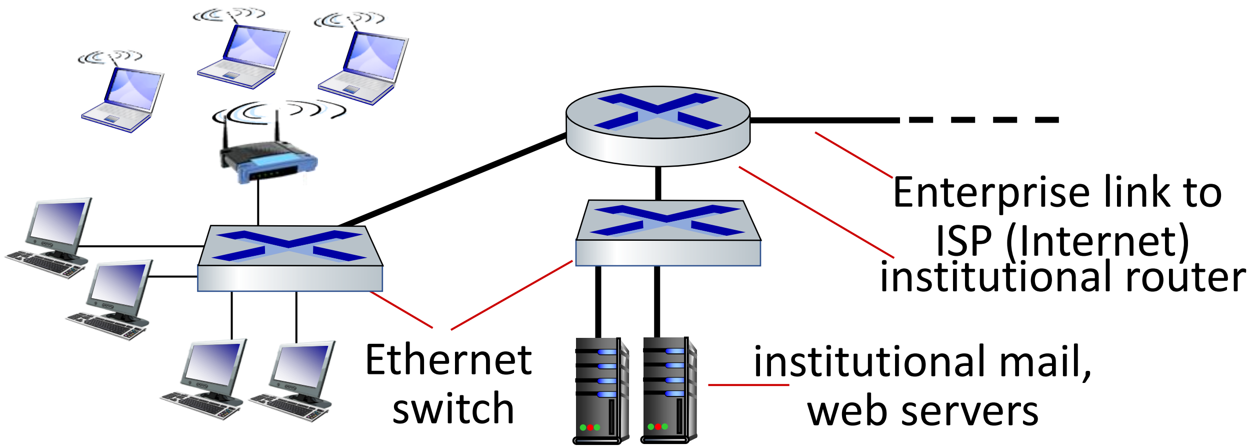


Wide-area cellular access networks

- provided by mobile, cellular network operator (10’s km)
- 10’s Mbps
- 4G/5G cellular networks



Access networks: enterprise networks

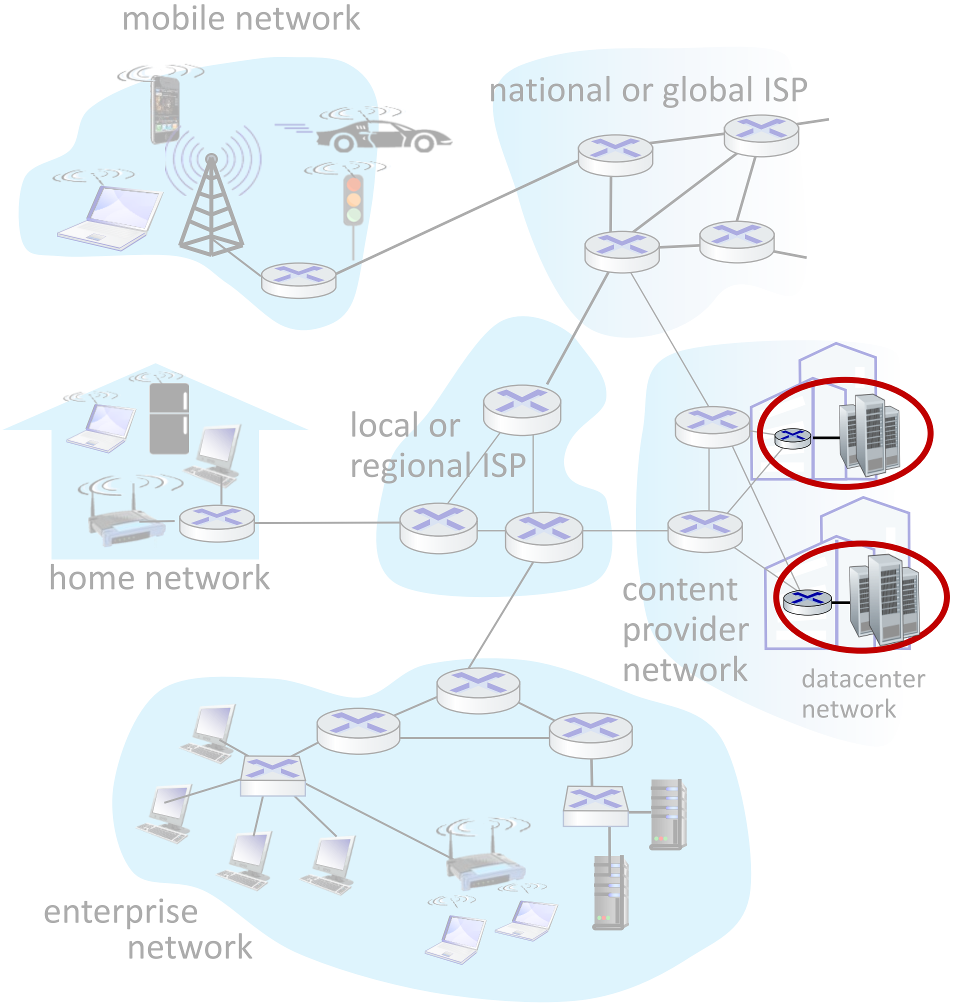


- companies, universities, etc.
- mix of wired, wireless link technologies, connecting a mix of switches and routers (we'll cover differences shortly)
  - Ethernet: wired access at 100Mbps, 1Gbps, 10Gbps
  - WiFi: wireless access points at 11, 54, 450 Mbps

Access networks: data center networks

- high-bandwidth links (10s to 100s Gbps) connect hundreds to thousands of servers together, and to Internet

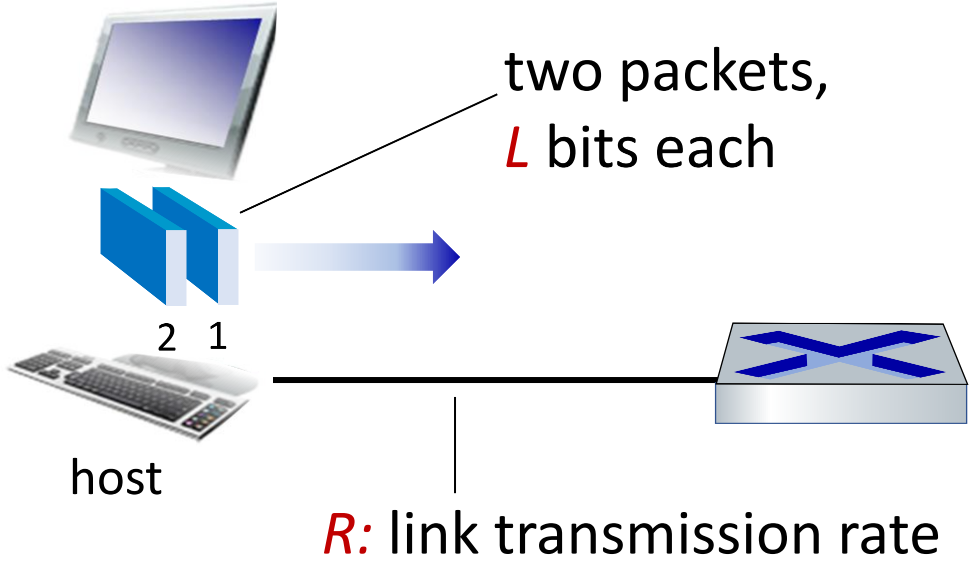
Courtesy: Massachusetts Green High Performance Computing Center (mghpcc.org)



Host: sends *packets* of data

host sending function:

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



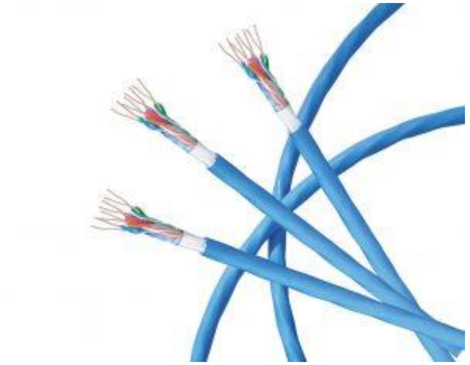
$$\text{packet transmission delay} = \text{time needed to transmit } L\text{-bit packet into link} = \frac{L \text{ (bits)}}{R \text{ (bits/sec)}}$$

Links: physical media

- bit: propagates between transmitter/receiver pairs
- physical link: what lies between transmitter & receiver
- guided media:
  - signals propagate in solid media: copper, fiber, coax
- unguided media:
  - signals propagate freely, e.g., radio

Twisted pair (TP)

- two insulated copper wires
  - Category 5: 100 Mbps, 1 Gbps Ethernet
  - Category 6: 10Gbps Ethernet



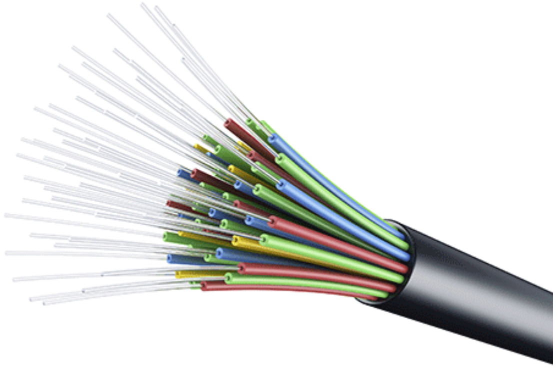
Links: physical media

Coaxial cable:

- two concentric copper conductors
- bidirectional
- broadband:
  - multiple frequency channels on cable
  - 100's Mbps per channel

Fiber optic cable:

- glass fiber carrying light pulses, each pulse a bit
- high-speed operation:
  - high-speed point-to-point transmission (10's-100's Gbps)
- low error rate:
  - repeaters spaced far apart
  - immune to electromagnetic noise



Links: physical media

Wireless radio

- signal carried in various "bands" in electromagnetic spectrum
- no physical "wire"
- broadcast, "half-duplex" (sender to receiver)
- propagation environment effects:
  - reflection
  - obstruction by objects
  - Interference/noise

Radio link types:

- Wireless LAN (WiFi)
  - 10-100's Mbps; 10's of meters
- wide-area (e.g., 4G/5G cellular)
  - 10's Mbps (4G) over ~10 Km
- Bluetooth: cable replacement
  - short distances, limited rates
- terrestrial microwave
  - point-to-point; 45 Mbps channels
- satellite
  - up to < 100 Mbps (Starlink) downlink
  - 270 msec end-end delay (geostationary)

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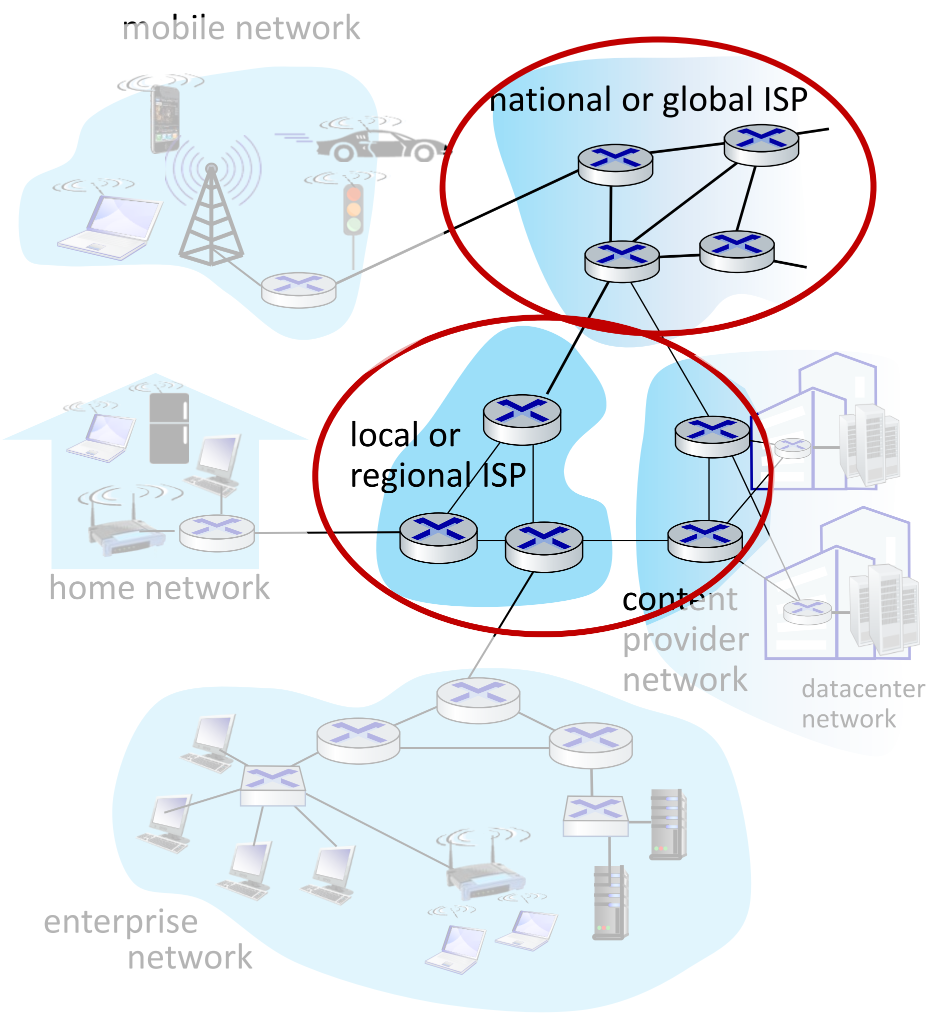
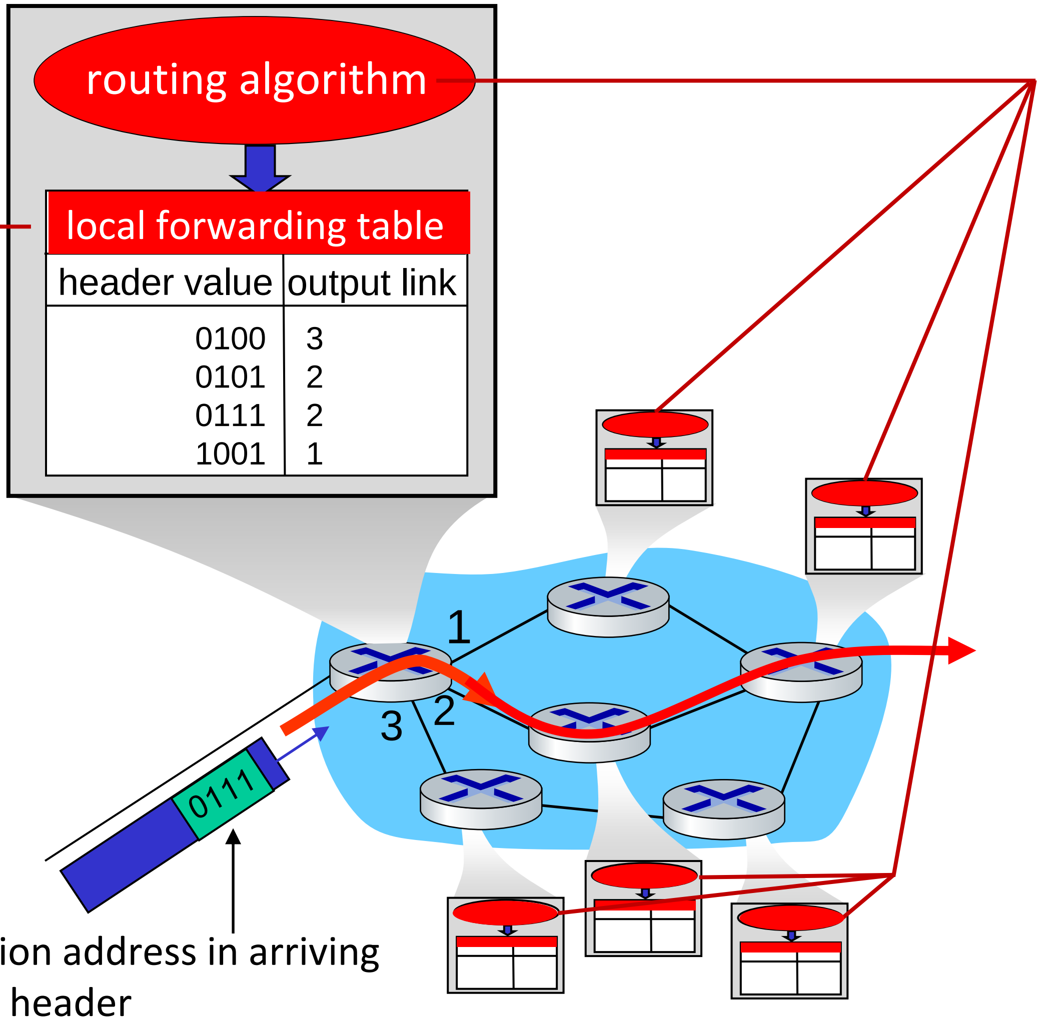
The network core

- mesh of interconnected routers
- packet-switching: hosts break application-layer messages into *packets*
  - network forwards packets from one router to the next, across links on path from source to destination



Two key network-core functions

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



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


routing
45 h
3,053 miles
48 h
3,309 miles
United States
San Jose
Northampton
Victoria
Seattle
WASHINGTON
MONTANA
NORTH DAKOTA
MINNESOTA
Minneapolis
WISCONSIN
SOUTH DAKOTA
Portland
OREGON
IDAHO
WYOMING
IOWA
Chicago
MICHIGAN
Detroit
Toronto
Ottawa
Montreal
Quebec City
MAINE
VERMONT
NEW HAMPSHIRE
NEVADA
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COLORADO
Kansas City
ILLINOIS
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Indianapolis
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New York
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MARYLAND
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WEST VIRGINIA
VIRGINIA
KENTUCKY
Nashville
TENNESSEE
NORTH CAROLINA
Charlotte
SOUTH CAROLINA
Atlanta
GEORGIA
ALABAMA
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OKLAHOMA
San Francisco
CALIFORNIA
Las Vegas
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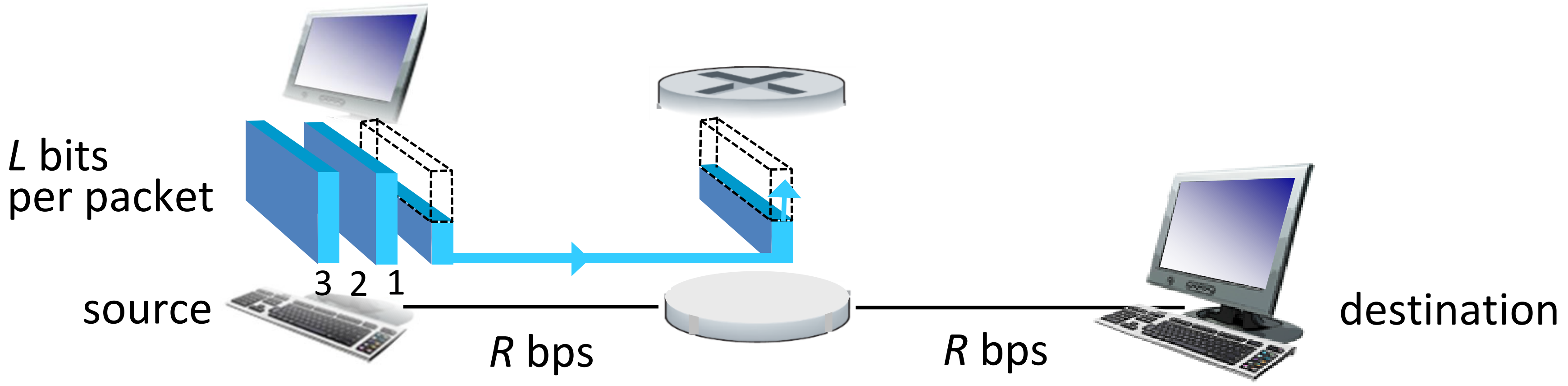
San Diego
ARIZONA
Phoenix
Tucson
NEW MEXICO
El Paso
Dallas
TEXAS
Austin
Houston
San Antonio
LOUISIANA
New Orleans
Jacksonville
Orlando
Tampa
FLORIDA
BAJA CALIFORNIA
SONORA
CHIHUAHUA
COAHUILA


forwarding
forwarding
45 h
3,053 miles
48 h
3,309 miles
United States
San Jose
Northampton
Sacramento
Cleveland

Packet-switching: store-and-forward

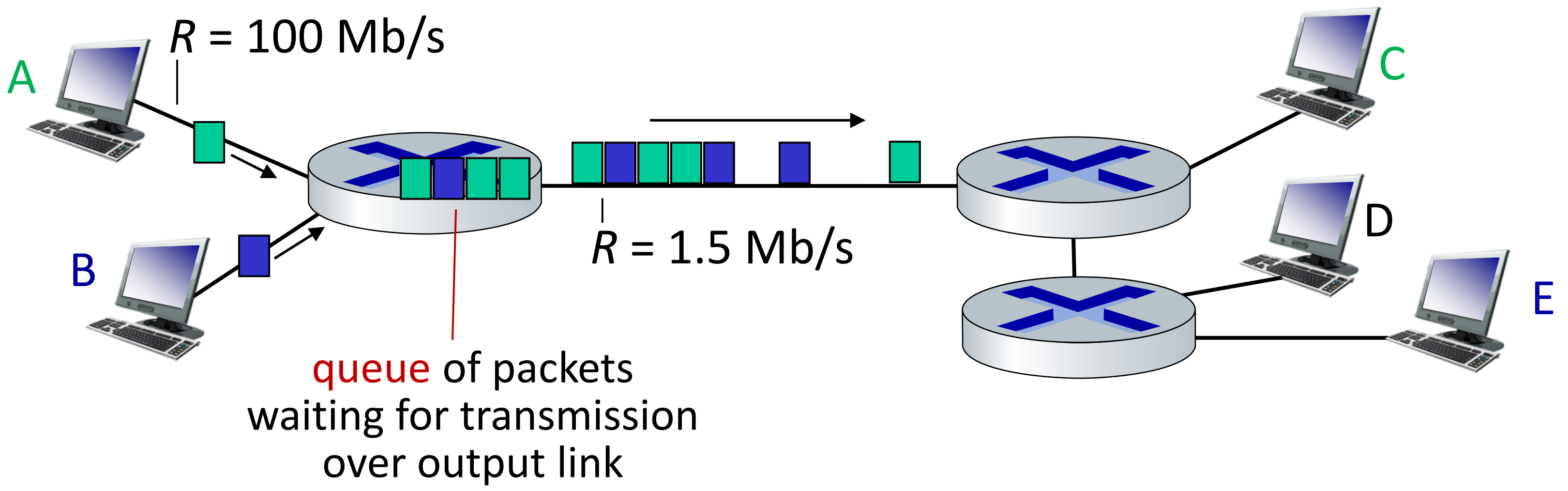


- packet transmission delay: takes L/R seconds to transmit (push out) L-bit packet into link at R bps
- *store and forward: entire* packet must arrive at router before it can be transmitted on next link

One-hop numerical example:

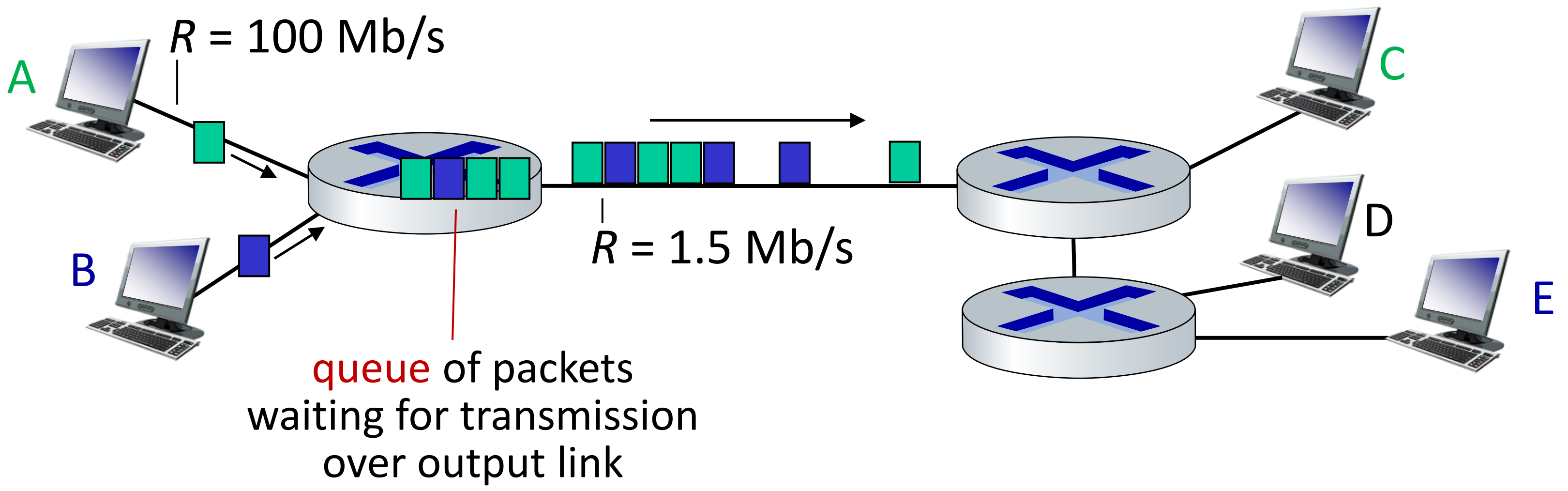
- L = 10 Kbits
- R = 100 Mbps
- one-hop transmission delay = 0.1 msec

Packet-switching: queueing



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

Packet-switching: queueing



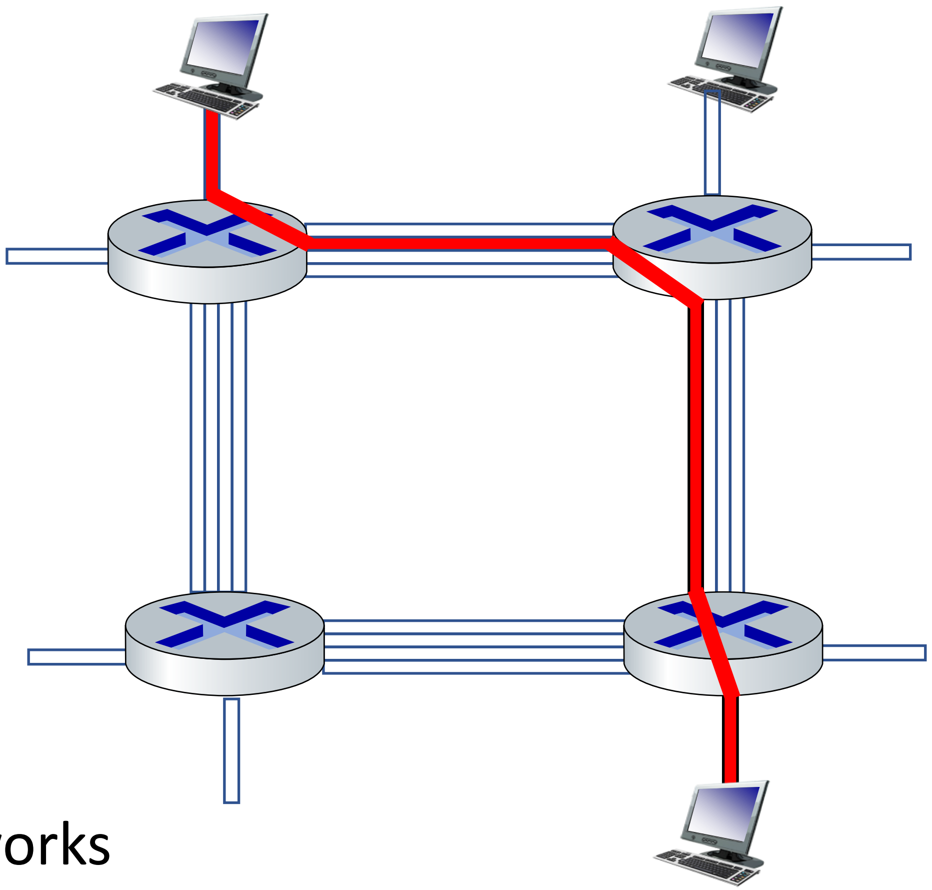
Packet queuing and loss: if arrival rate (in bps) to link exceeds transmission rate (bps) of link for some period of time:

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

Alternative to packet switching: circuit switching

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

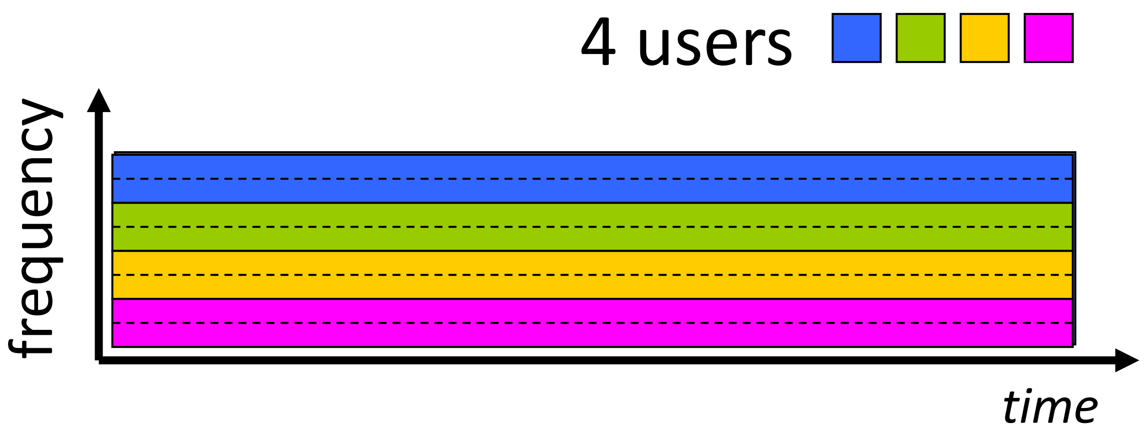
- in diagram, each link has four circuits.
  - call gets 2nd circuit in top link and 1st circuit in right link.
- dedicated resources: no sharing
  - circuit-like (guaranteed) performance
- circuit segment idle if not used by call (no sharing)
- commonly used in traditional telephone networks



Circuit switching: FDM and TDM

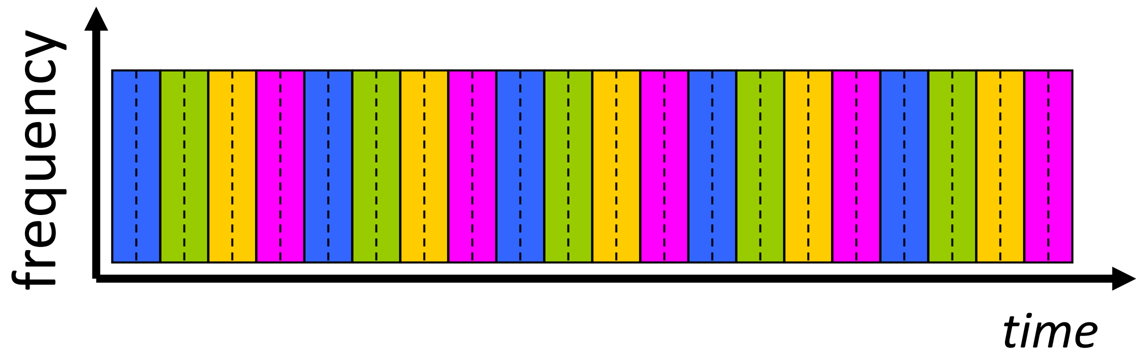
Frequency Division Multiplexing (FDM)

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



Time Division Multiplexing (TDM)

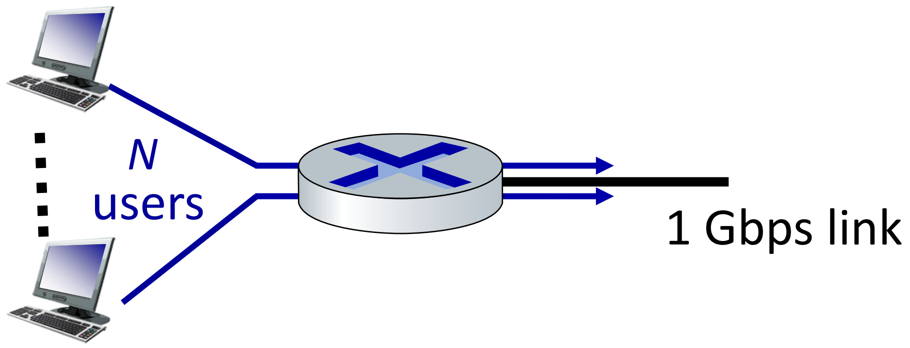
- time divided into slots
- each call allocated periodic slot(s), can transmit at maximum rate of (wider) frequency band (only) during its time slot(s)



Packet switching versus circuit switching

example:

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



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

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

Q: how did we get value 0.0004?

A: HW problem (for those with course in probability only)

Packet switching versus circuit switching

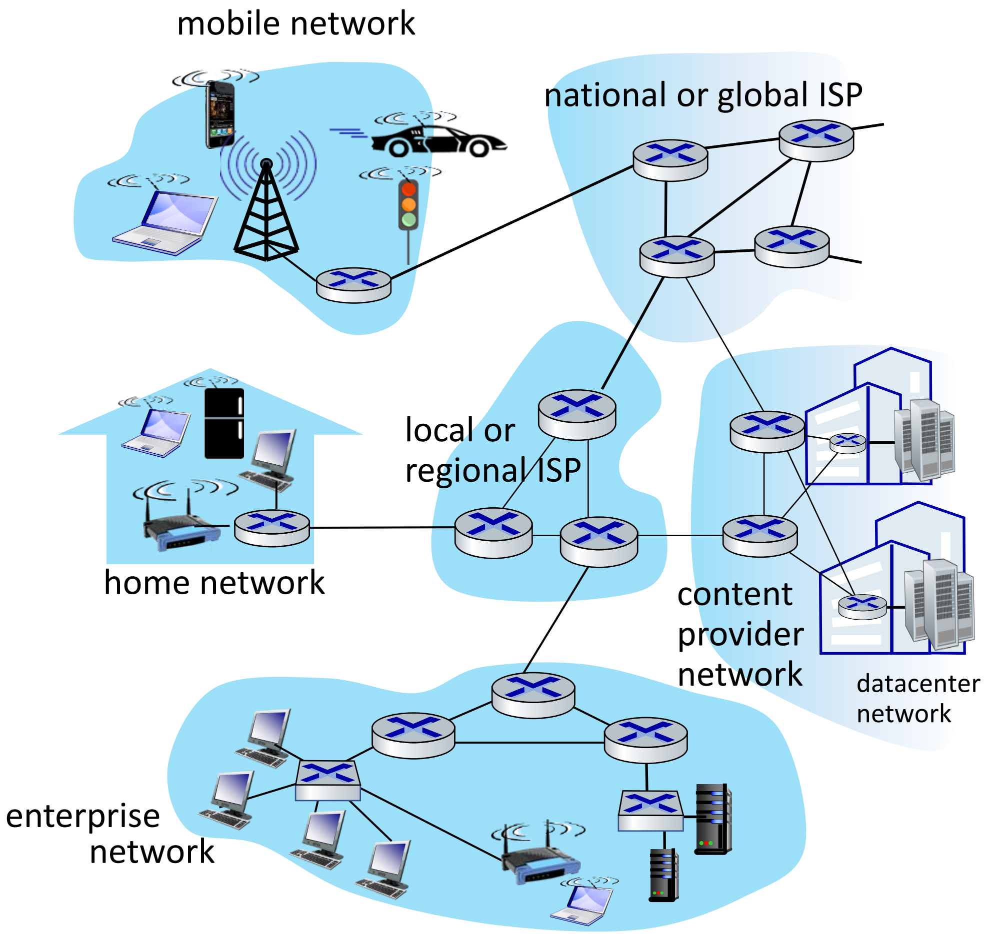
Is packet switching a "slam dunk winner"?

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

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

Internet structure: a "network of networks"

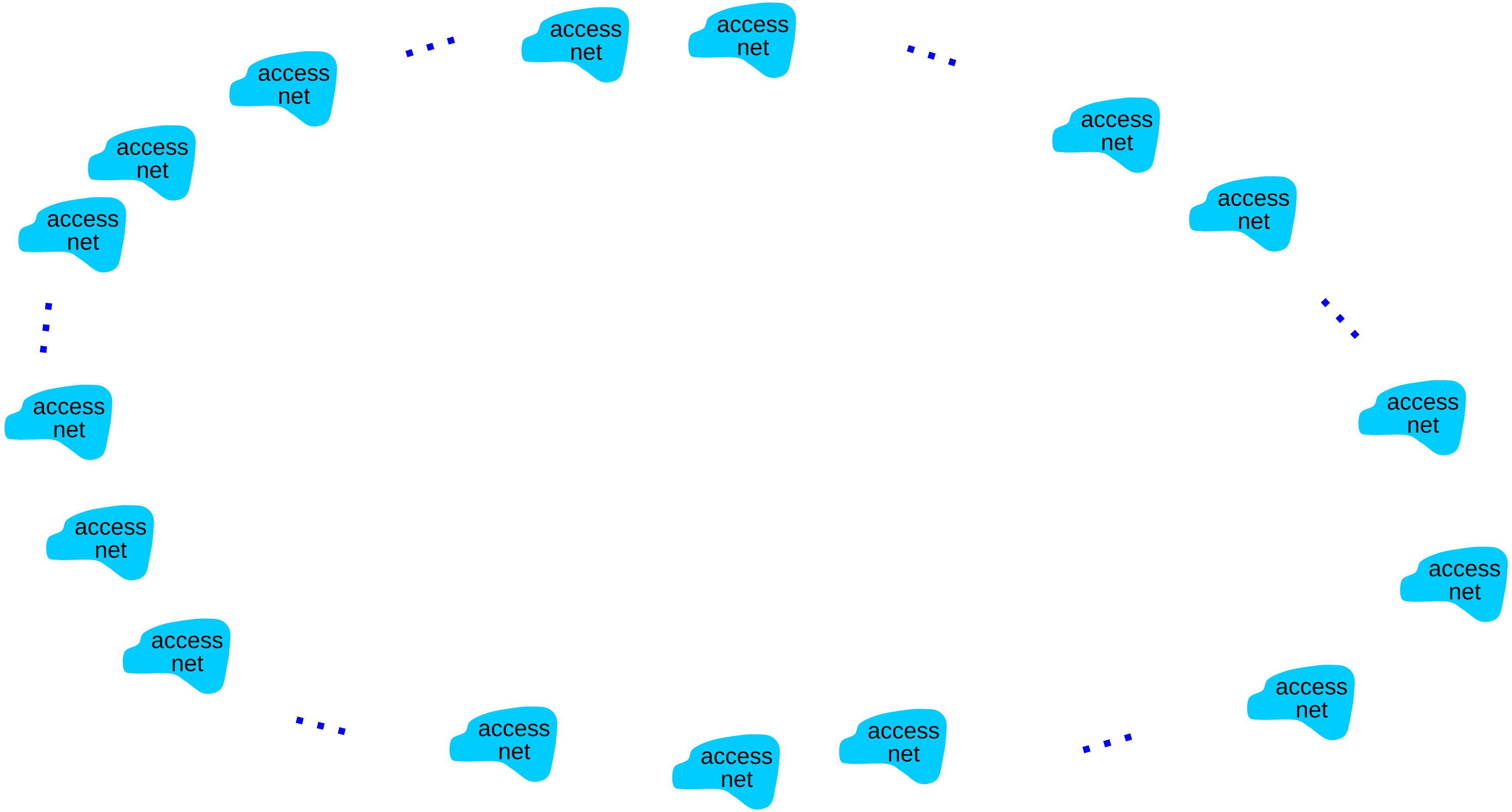
- hosts connect to Internet via access Internet Service Providers (ISPs)
- access ISPs in turn must be interconnected
  - so that *any* two hosts *(anywhere!)* can send packets to each other
- resulting network of networks is very complex
  - evolution driven by economics, national policies



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

Internet structure: a "network of networks"

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



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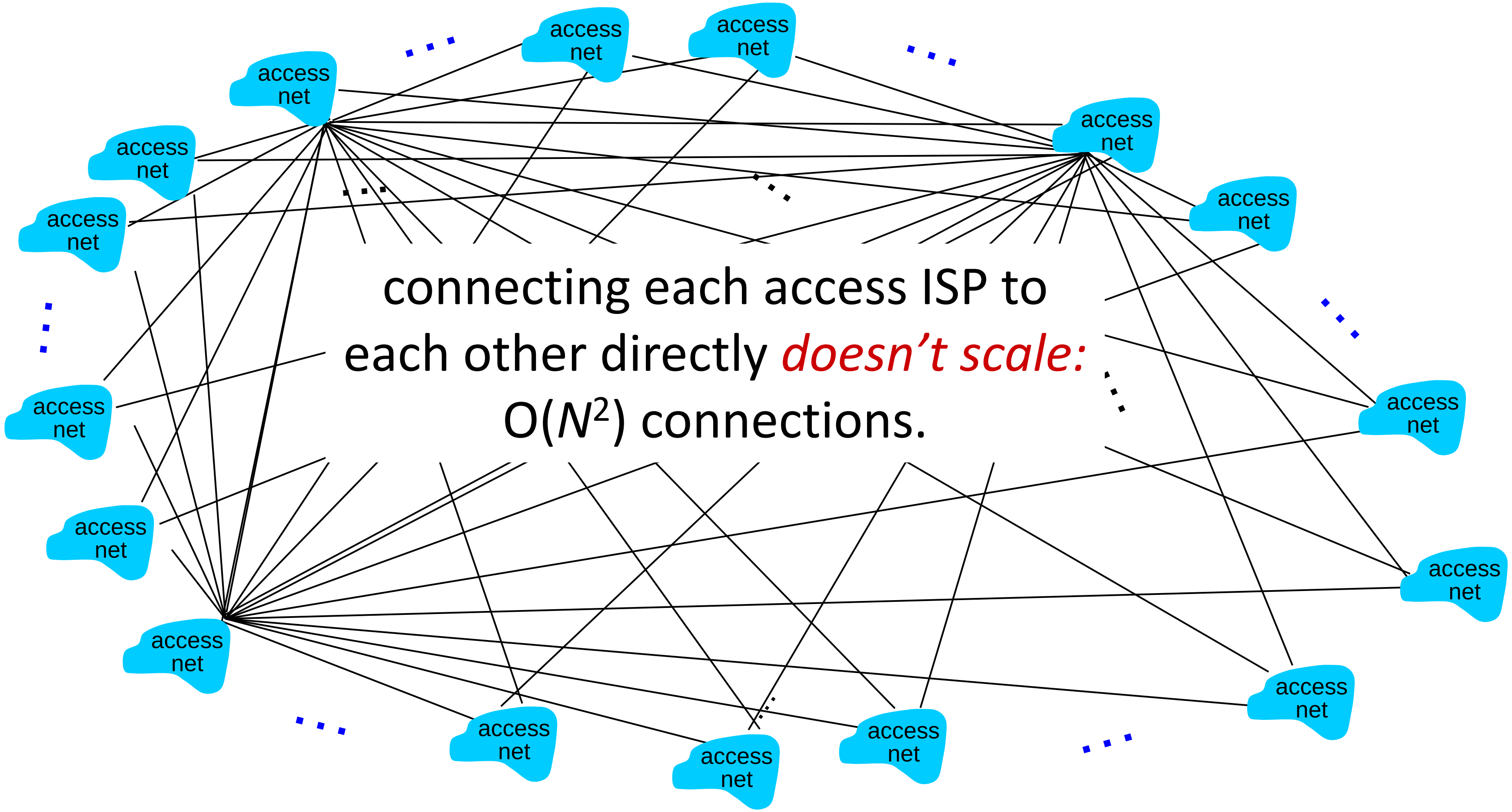
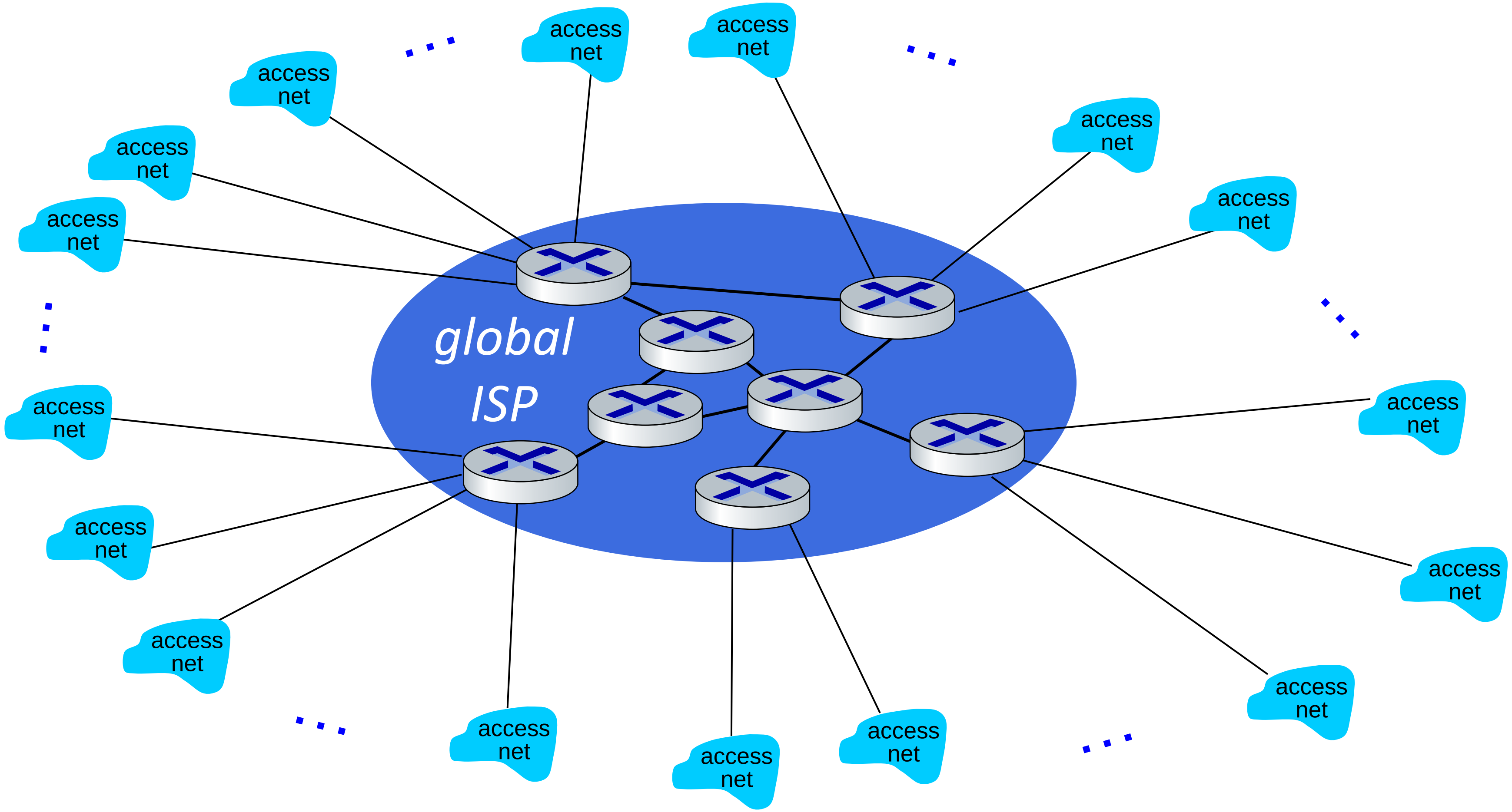


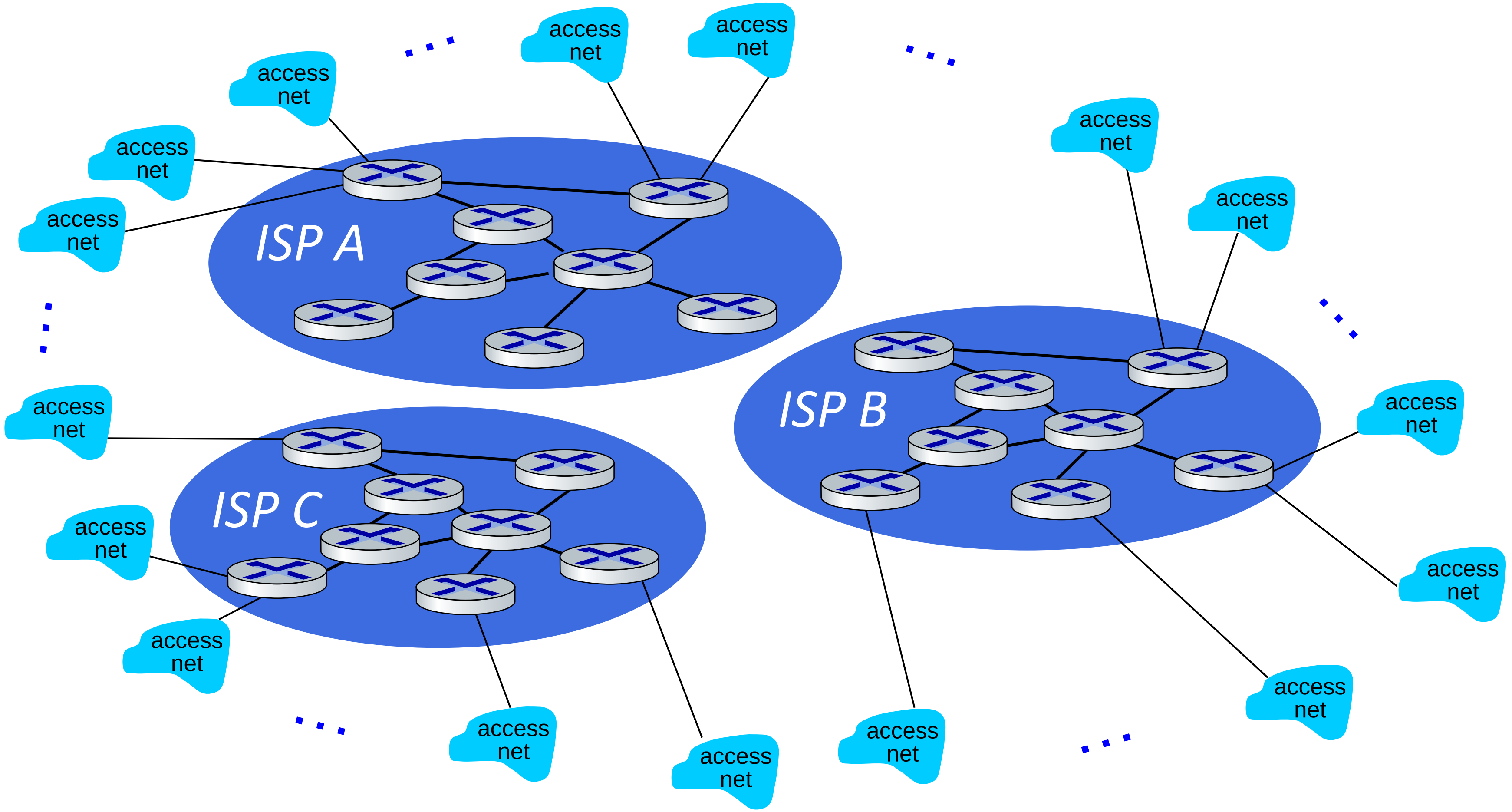
Internet structure: a “network of networks”

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



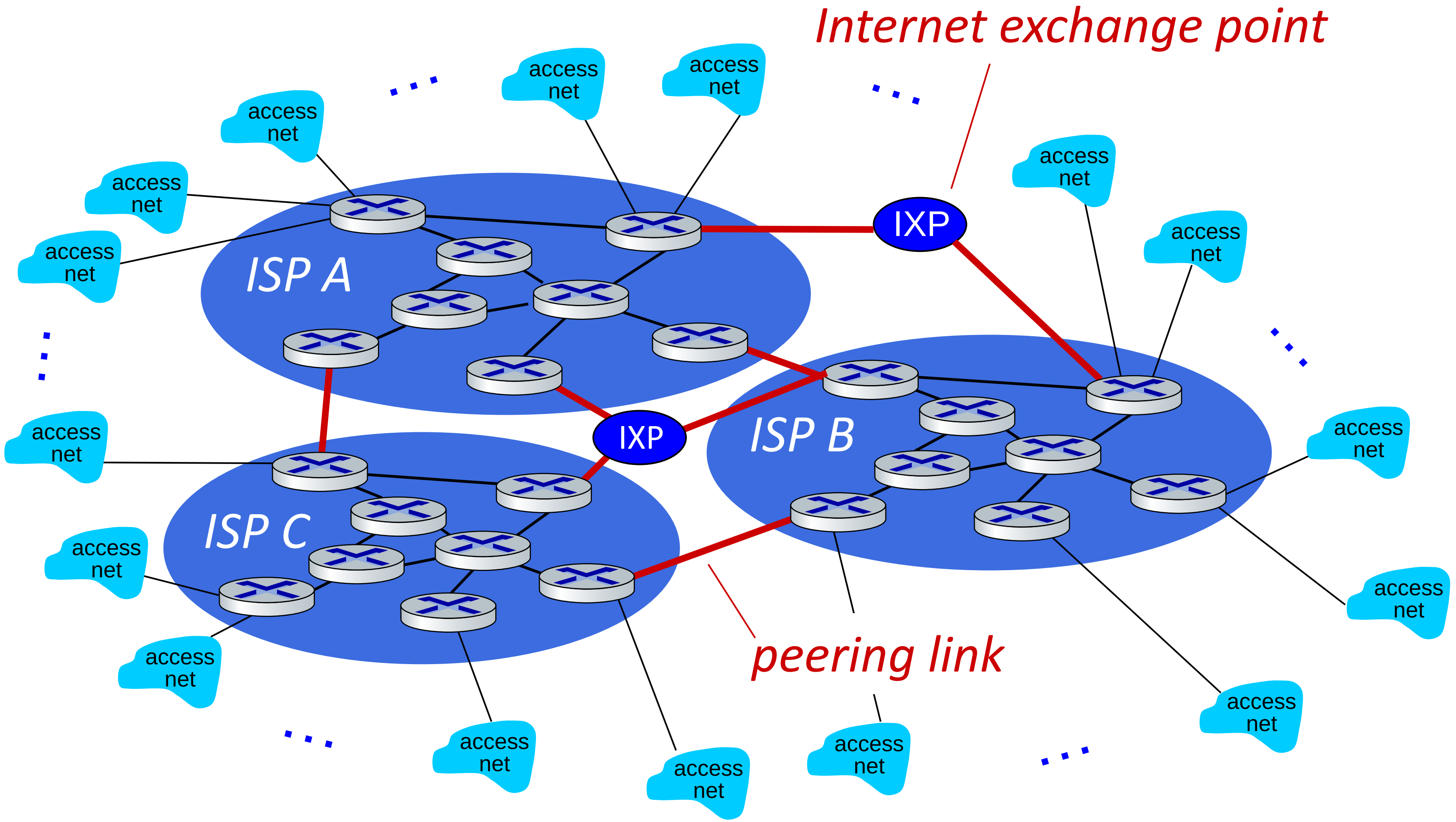
Internet structure: a “network of networks”

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



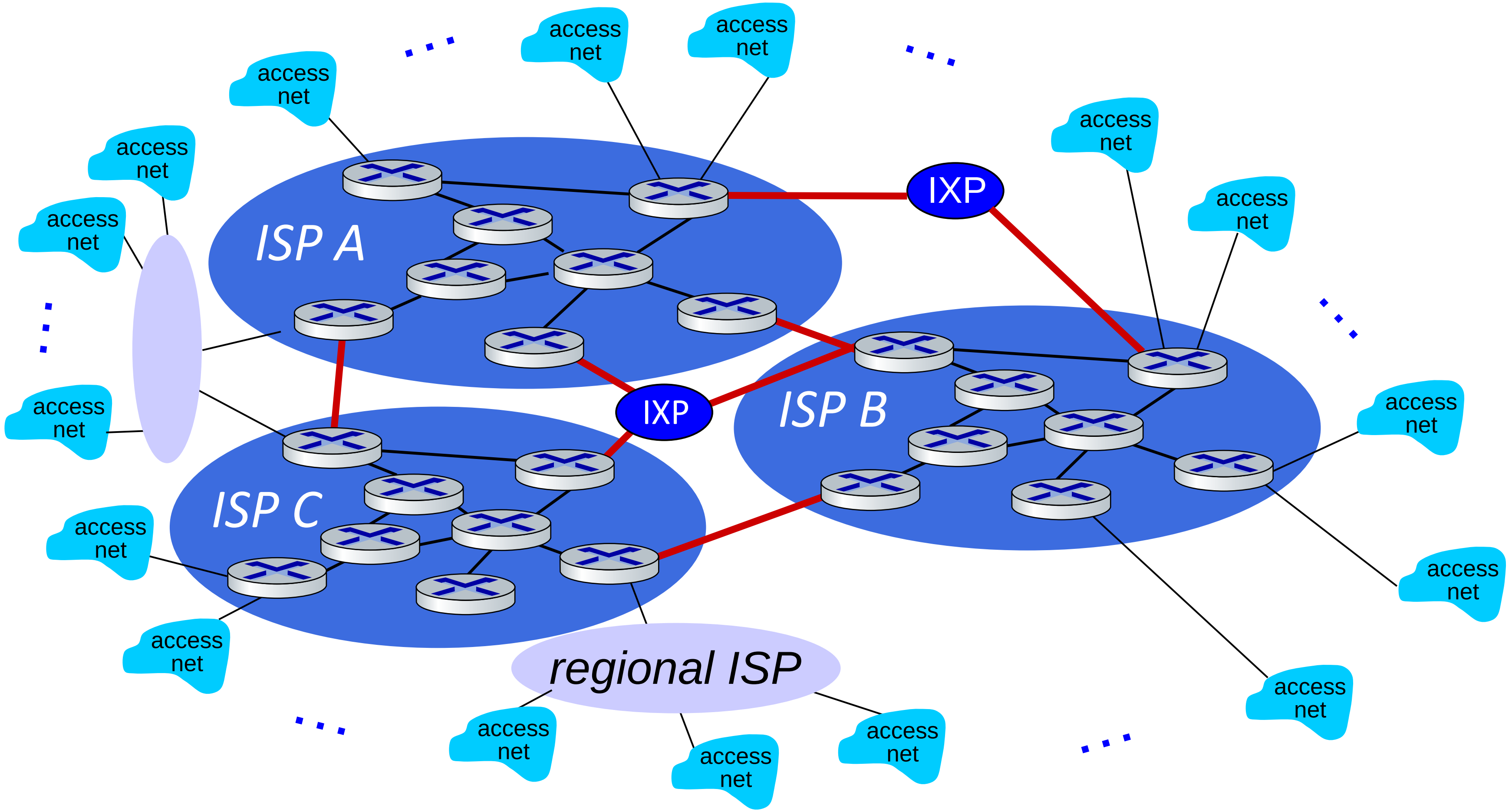
Internet structure: a "network of networks"

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



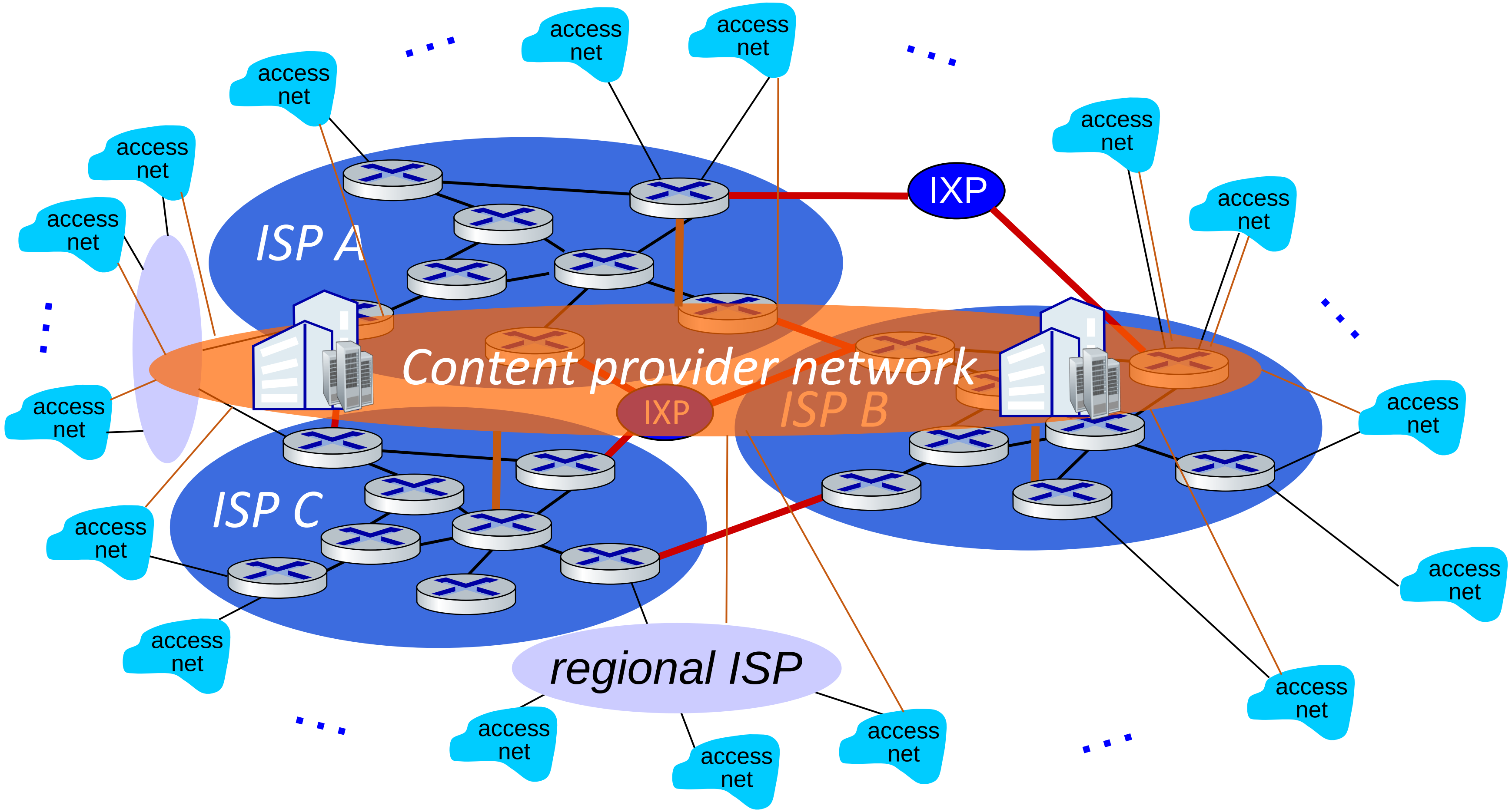
Internet structure: a "network of networks"

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

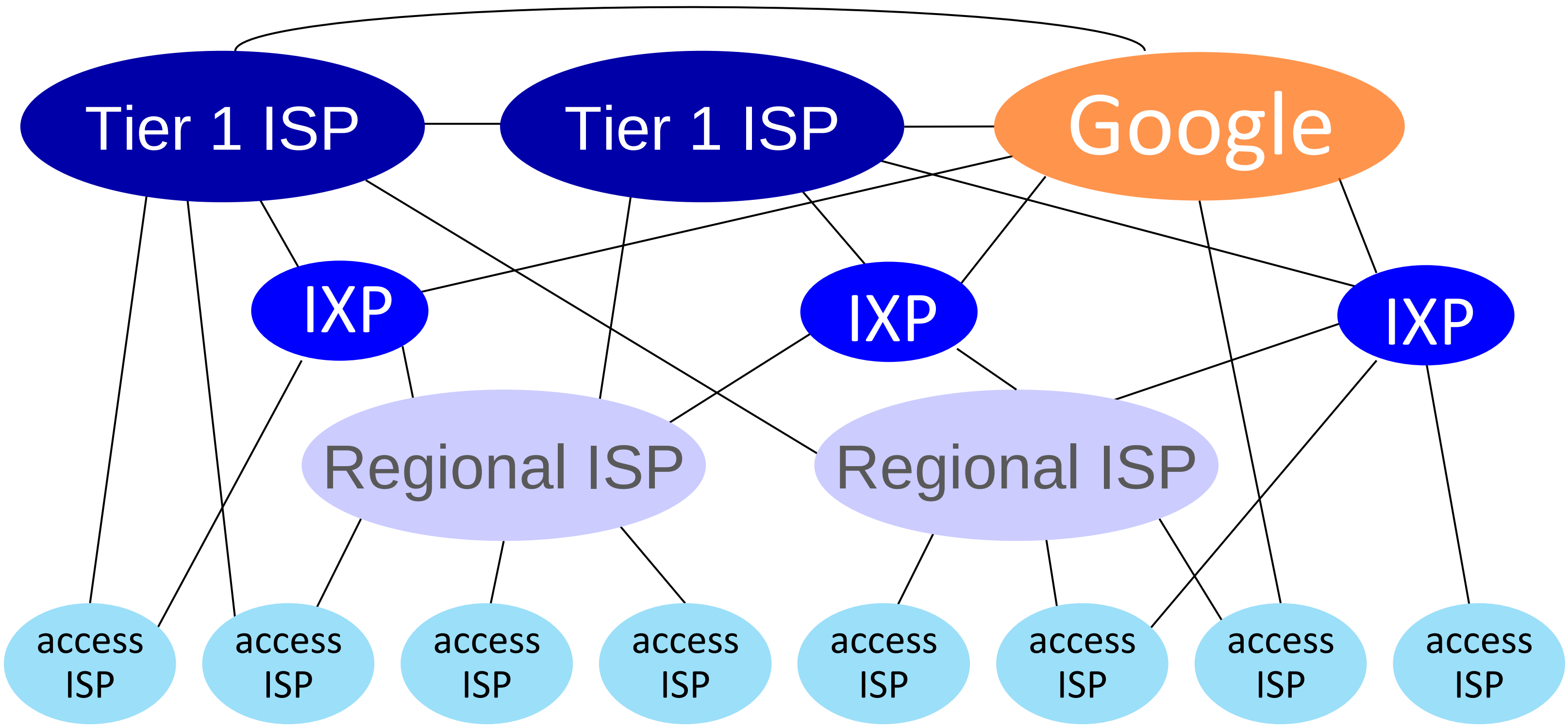


Internet structure: a "network of networks"

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



Internet structure: a “network of networks”



At “center”: small # of well-connected large networks

- “tier-1” commercial ISPs (e.g., Level 3, Sprint, AT&T, NTT), national & international coverage
- content provider networks (e.g., Google, Facebook): private network that connects its data centers to Internet, often bypassing tier-1, regional ISPs

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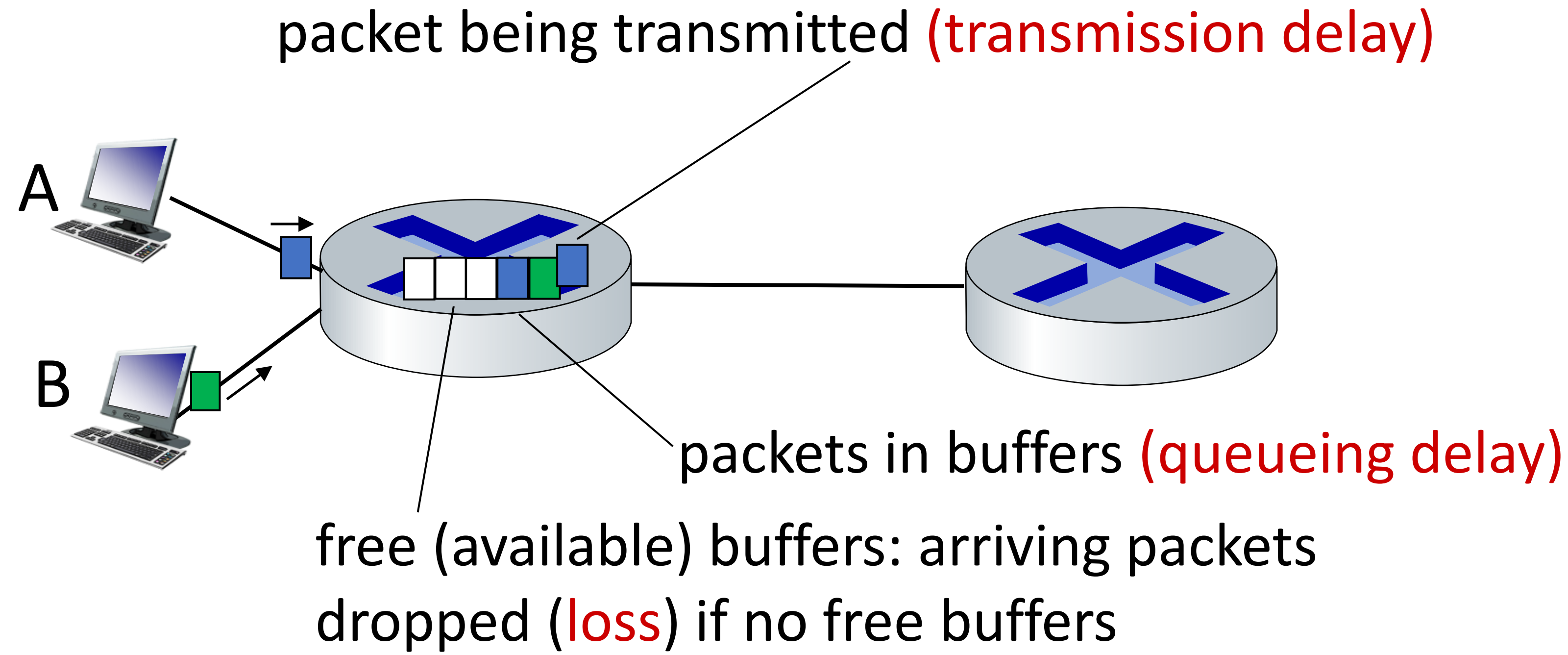
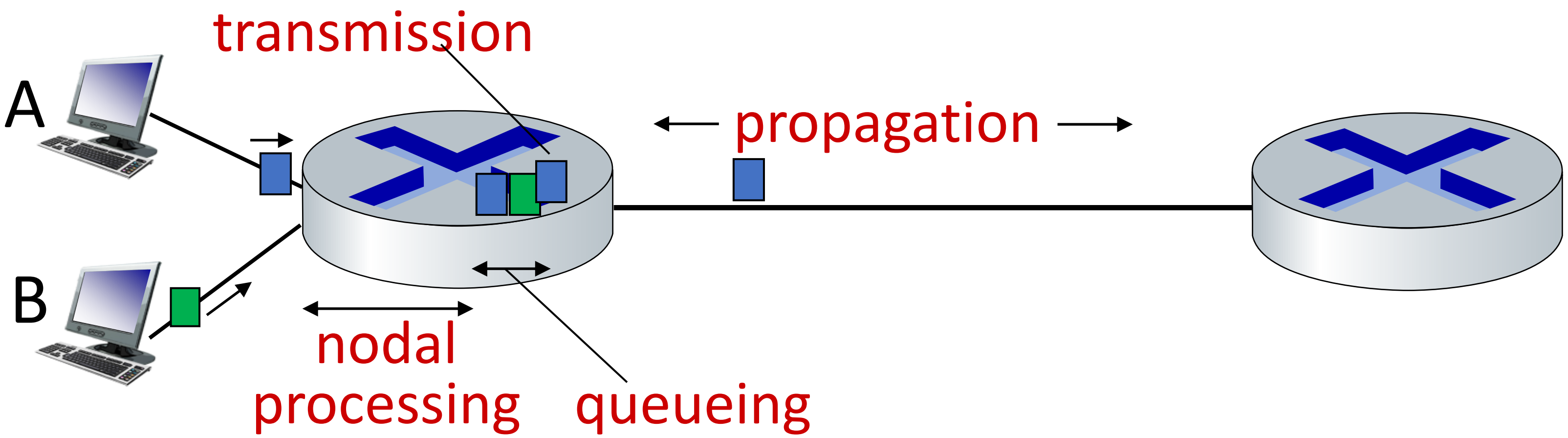
How do packet delay and loss occur?

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



Packet delay: four sources



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

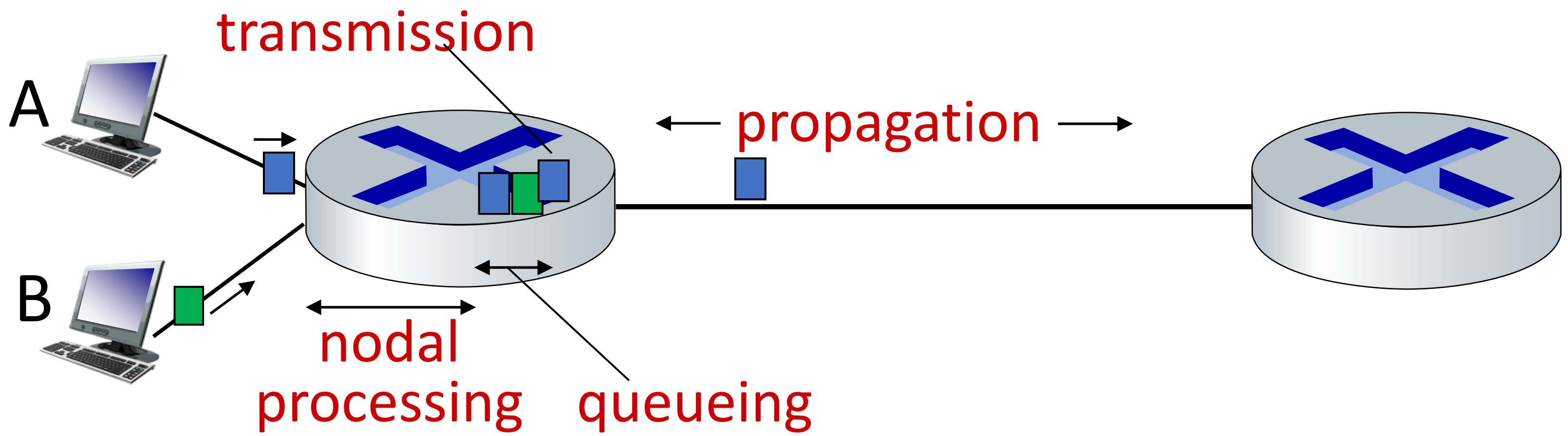
d_{proc}: nodal processing

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

d_{queue}: queueing delay

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

Packet delay: four sources



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

d_{trans}: transmission delay:

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

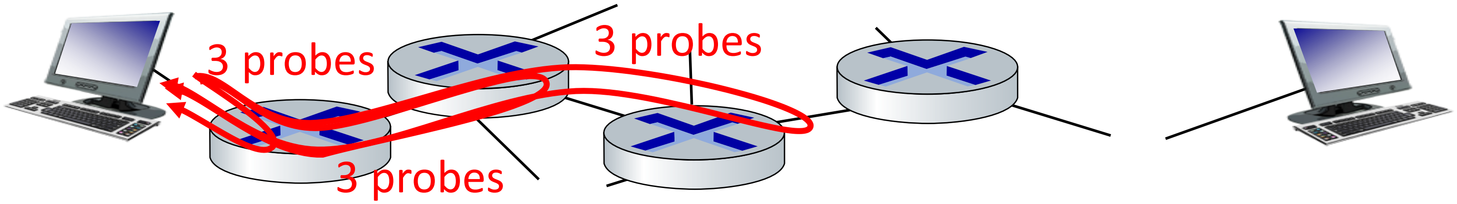
d_{prop}: propagation delay:

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

d_{trans} and d_{prop}
very different

“Real” Internet delays and routes

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



Real Internet delays and routes

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

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

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

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

trans-oceanic link

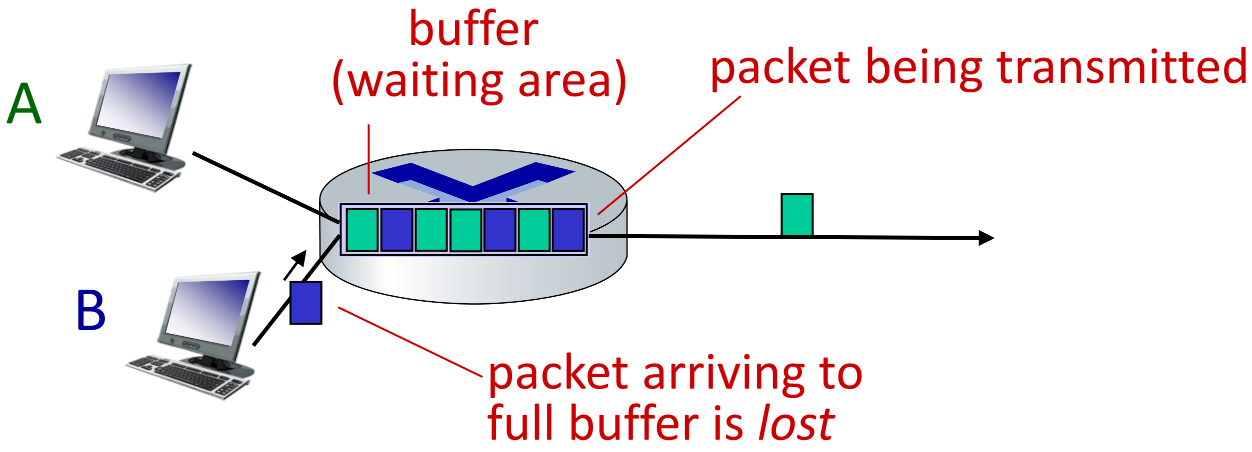
looks like delays *decrease*! Why?

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

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

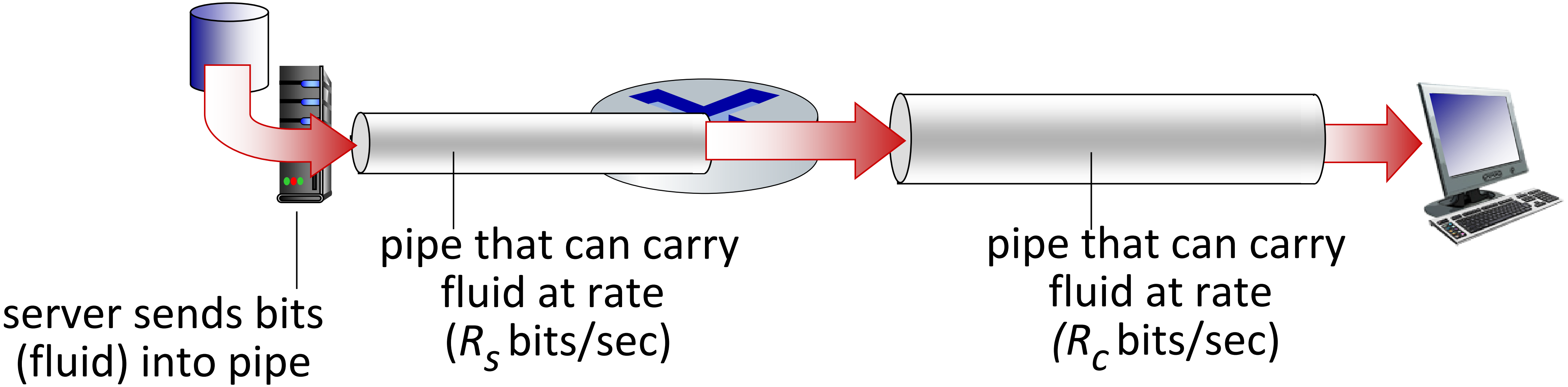
Packet loss

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



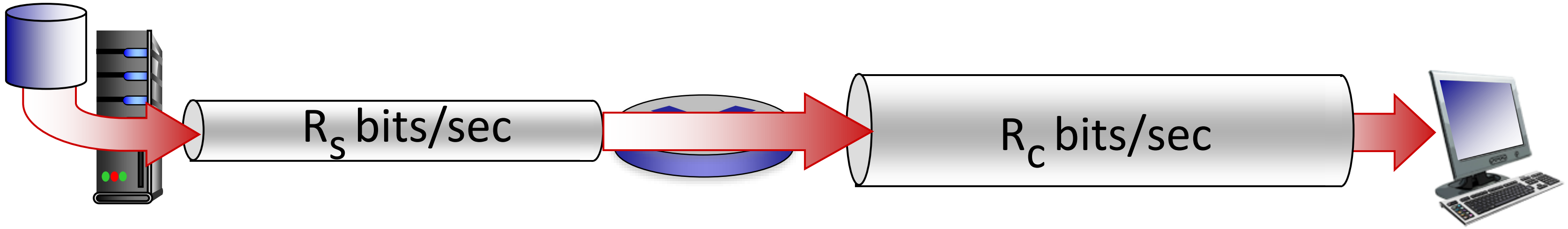
Throughput

- *throughput:* rate (bits/time unit) at which bits are being sent from sender to receiver
  - *instantaneous:* rate at given point in time
  - *average:* rate over longer period of time

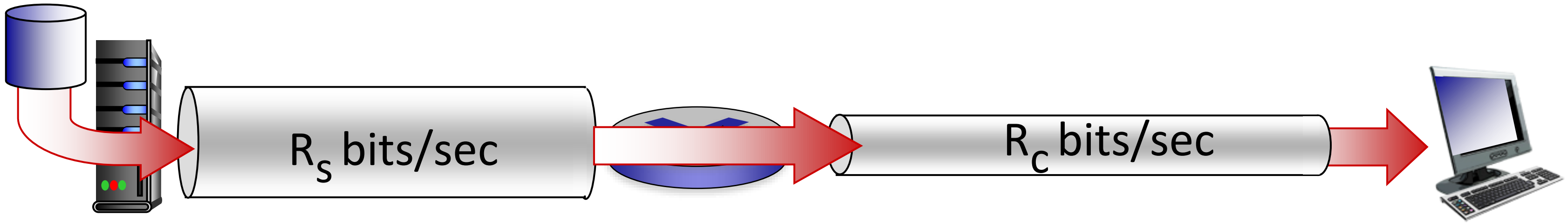


Throughput

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



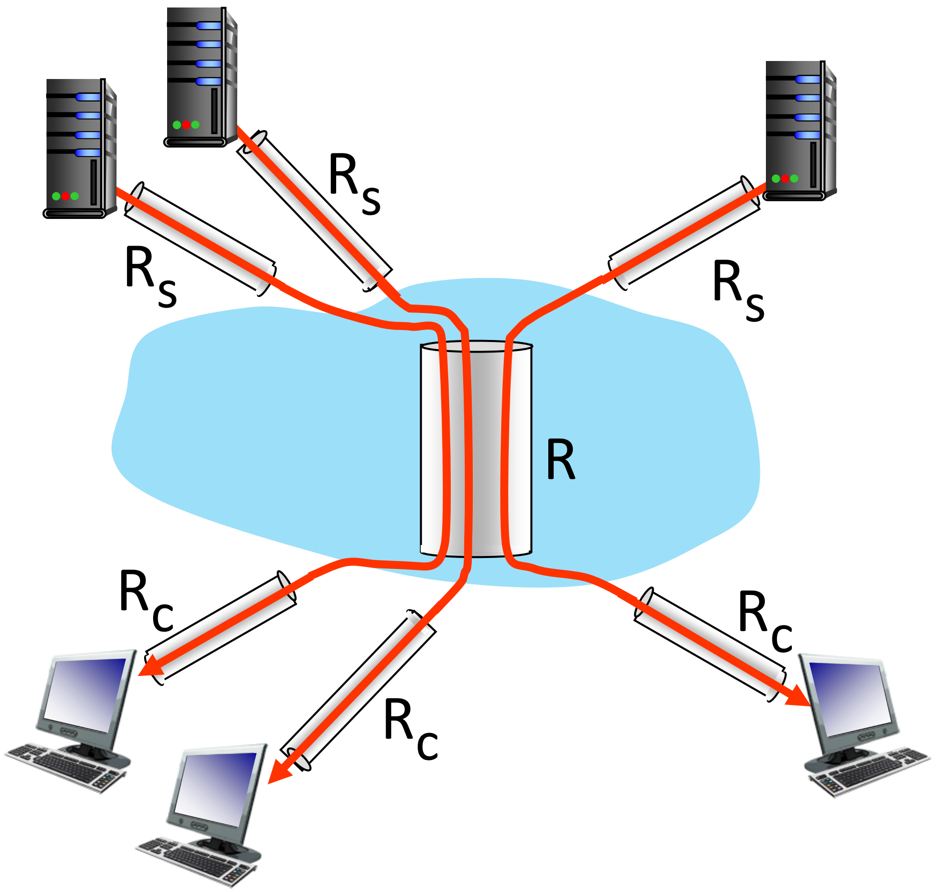
$R_s > R_c$ What is average end-end throughput?



bottleneck link

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

Throughput: network scenario



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

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

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

Chapter 1: roadmap

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

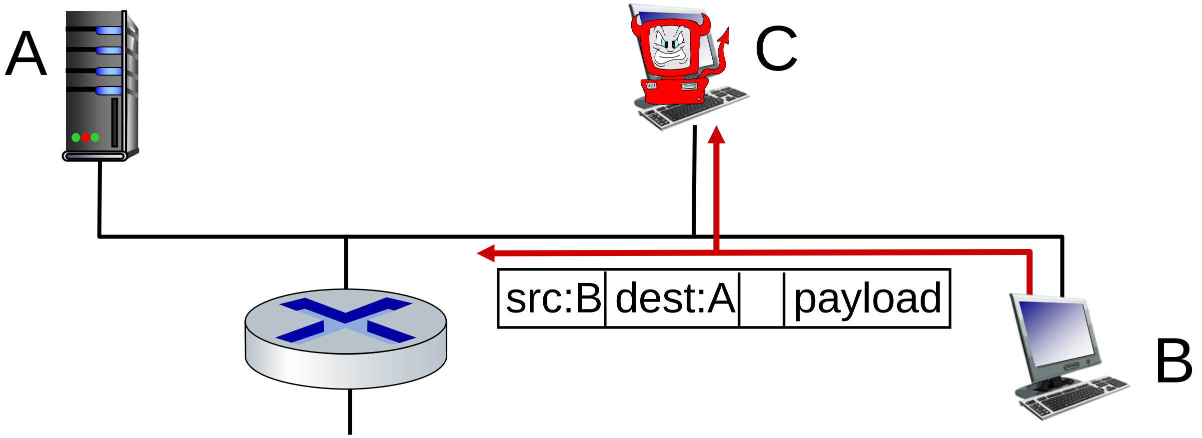
Network security

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

Bad guys: packet interception

packet "sniffing":

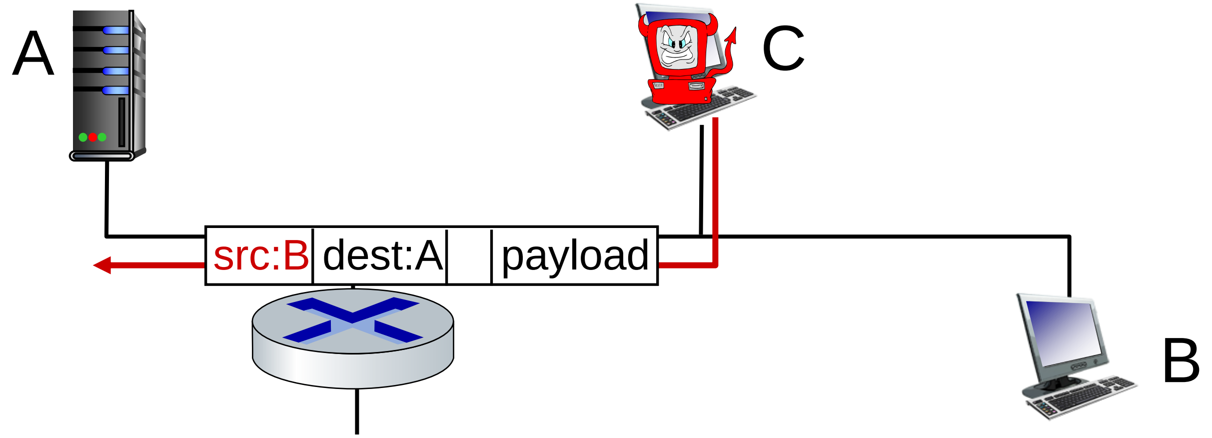
- broadcast media (shared Ethernet, wireless)
- promiscuous network interface reads/records all packets (e.g., including passwords!) passing by



Wireshark software used for our end-of-chapter labs is a (free) packet-sniffer

Bad guys: fake identity

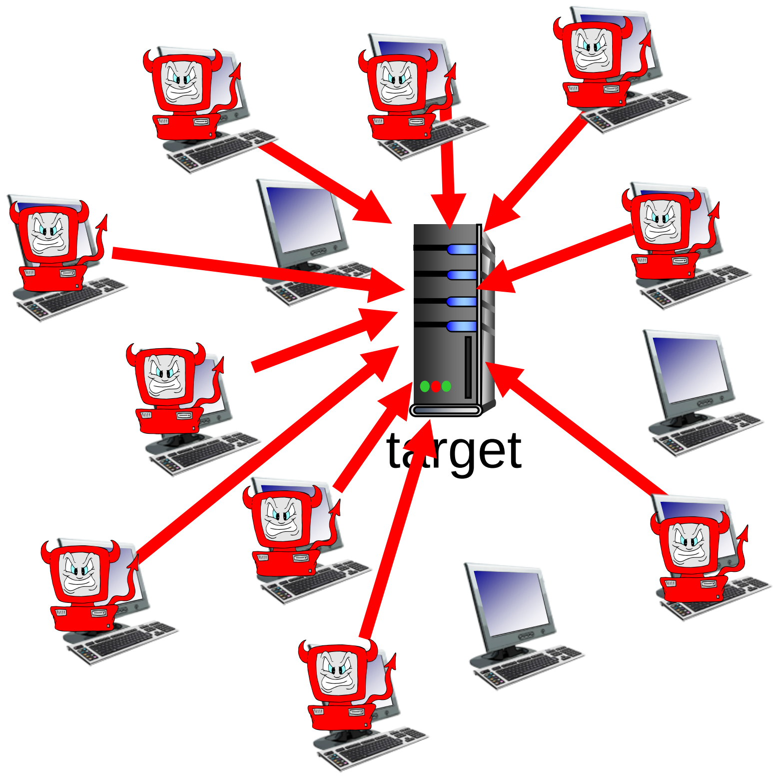
IP spoofing: injection of packet with false source address



Bad guys: denial of service

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

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



Lines of defense:

- authentication: proving you are who you say you are
  - cellular networks provides hardware identity via SIM card; no such hardware assist in traditional Internet
- confidentiality: via encryption
- integrity checks: digital signatures prevent/detect tampering
- access restrictions: password-protected VPNs
- firewalls: specialized "middleboxes" in access and core networks:
  - off-by-default: filter incoming packets to restrict senders, receivers, applications
  - detecting/reacting to DOS attacks

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Protocol "layers" and reference models

Networks are complex, with many "pieces":

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

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

- and/or our *discussion* of networks?

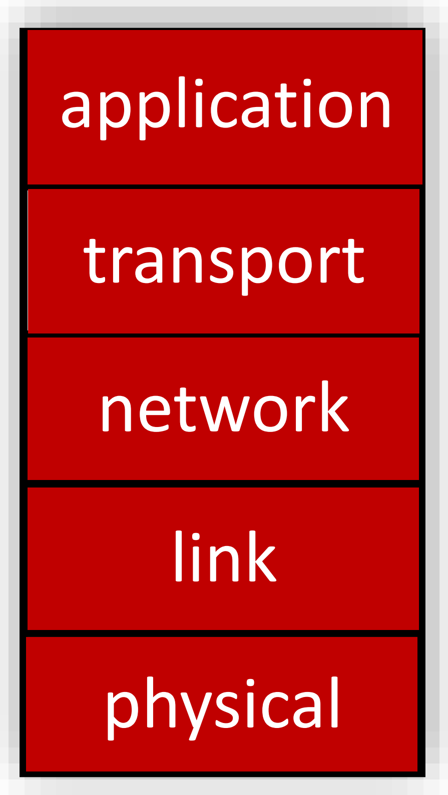
Why layering?

Approach to designing/discussing complex systems:

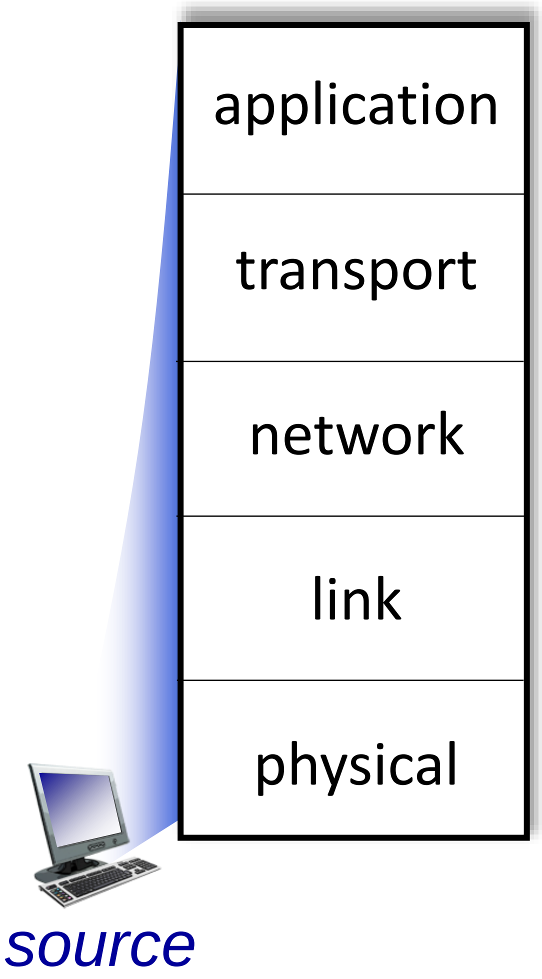
- explicit structure allows identification, relationship of system's pieces
  - layered *reference model* for discussion
- modularization eases maintenance, updating of system
  - change in layer's service *implementation*: transparent to rest of system
  - e.g., change in gate procedure doesn't affect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

Layered Internet protocol stack

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



Services, Layering and Encapsulation



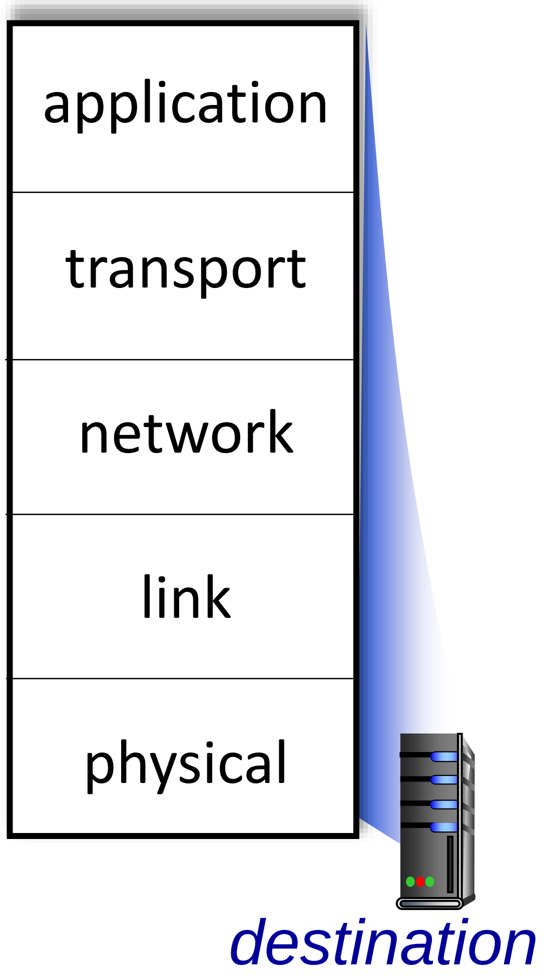
M

Application exchanges messages to implement some application service using *services* of transport layer

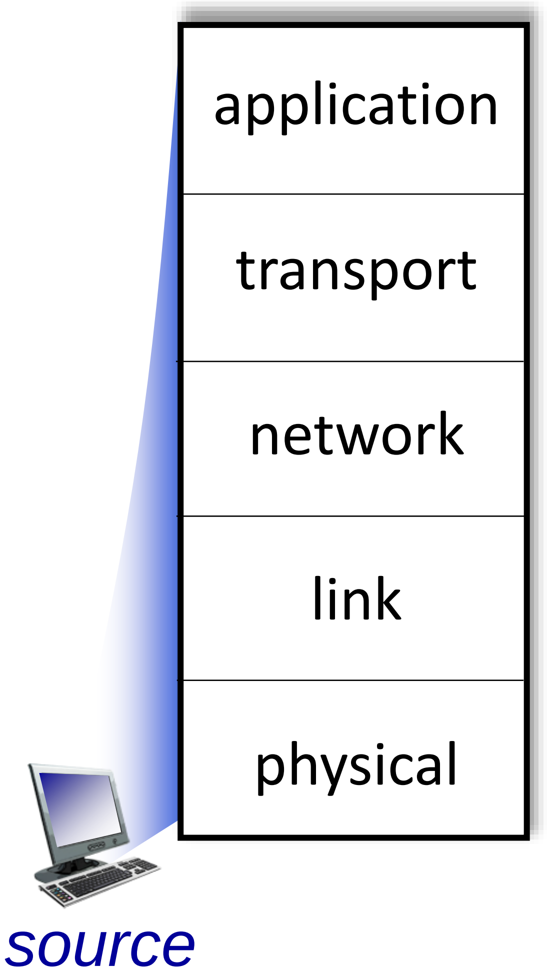
H_t M

Transport-layer protocol transfers M (e.g., reliably) from one *process* to another, using services of network layer

- transport-layer protocol encapsulates application-layer message, M, with *transport* layer-layer header H_t to create a transport-layer segment
  - H_t used by transport layer protocol to implement its service



Services, Layering and Encapsulation



M

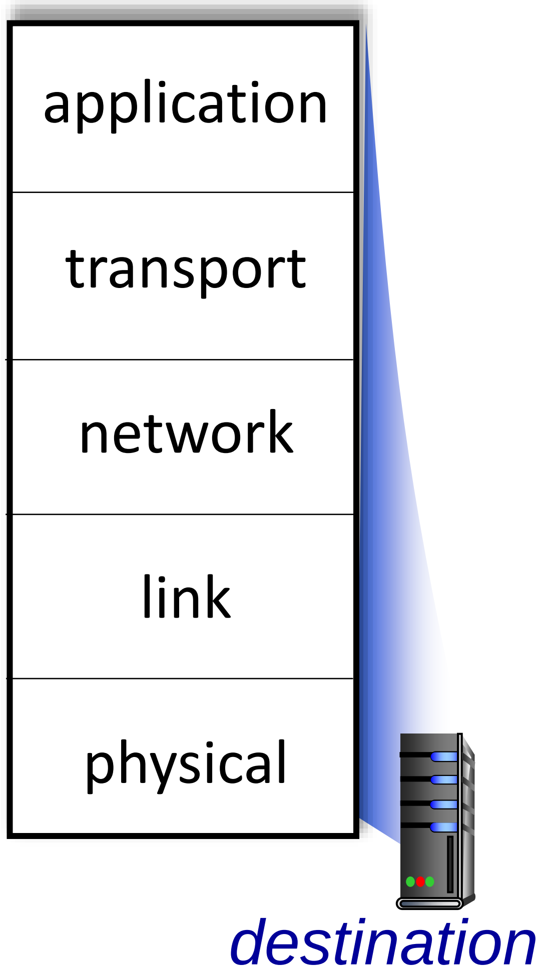
H_t M

Transport-layer protocol transfers M (e.g., reliably) from one *process* to another, using services of network layer

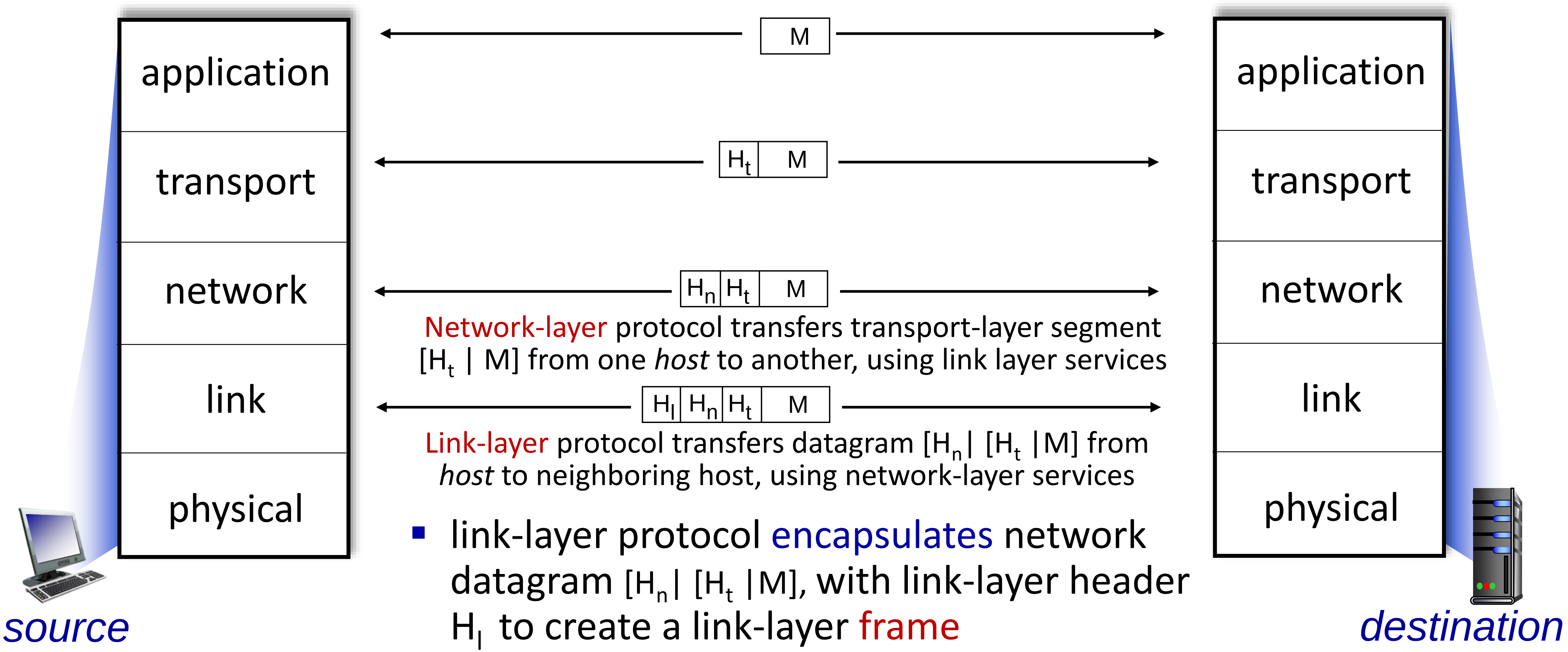
H_n H_t M

Network-layer protocol transfers transport-layer segment [H_t | M] from one *host* to another, using link layer services

- network-layer protocol encapsulates transport-layer segment [H_t | M] with network layer-layer header H_n to create a network-layer datagram
  - H_n used by network layer protocol to implement its service

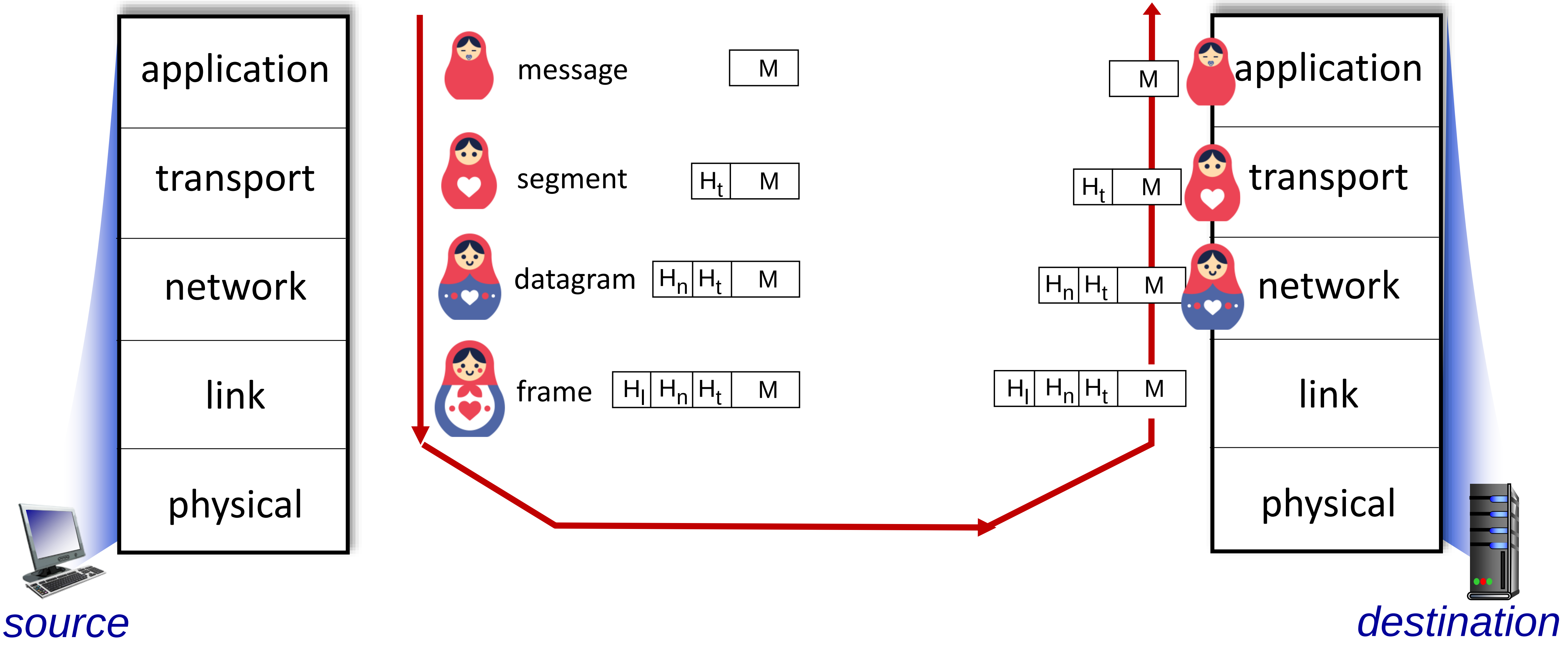


Services, Layering and Encapsulation

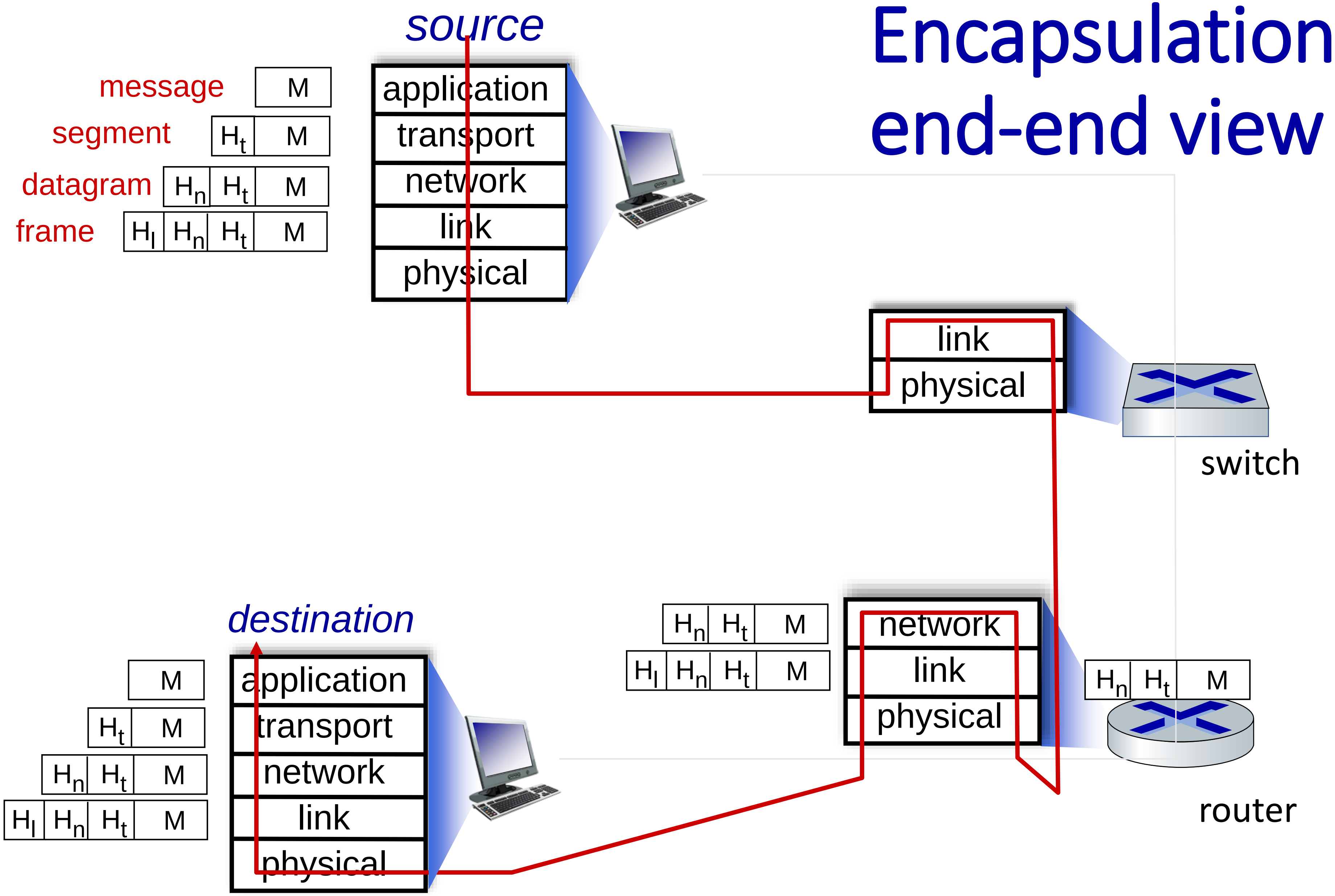


- link-layer protocol encapsulates network datagram [H_n| [H_t |M], with link-layer header H_l to create a link-layer frame

Services, Layering and Encapsulation



Encapsulation: an end-end view



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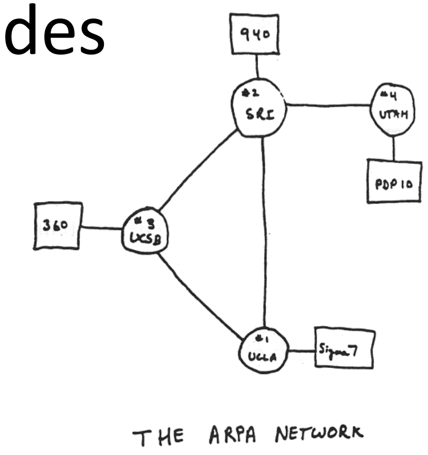
Internet history

1961-1972: Early packet-switching principles

- 1961: Kleinrock - queueing theory shows effectiveness of packet-switching
- 1964: Baran - packet-switching in military nets
- 1967: ARPAnet conceived by Advanced Research Projects Agency
- 1969: first ARPAnet node operational

- 1972:
  - ARPAnet public demo
  - NCP (Network Control Protocol) first host-host protocol
  - first e-mail program
  - ARPAnet has 15 nodes



Internet history

1972-1980: Internetworking, new and proprietary networks

- 1970: ALOHAnet satellite network in Hawaii
- 1974: Cerf and Kahn - architecture for interconnecting networks
- 1976: Ethernet at Xerox PARC
- late70's: proprietary architectures: DECnet, SNA, XNA
- 1979: ARPAnet has 200 nodes

Cerf and Kahn's internetworking principles:

- minimalism, autonomy - no internal changes required to interconnect networks
- best-effort service model
- stateless routing
- decentralized control

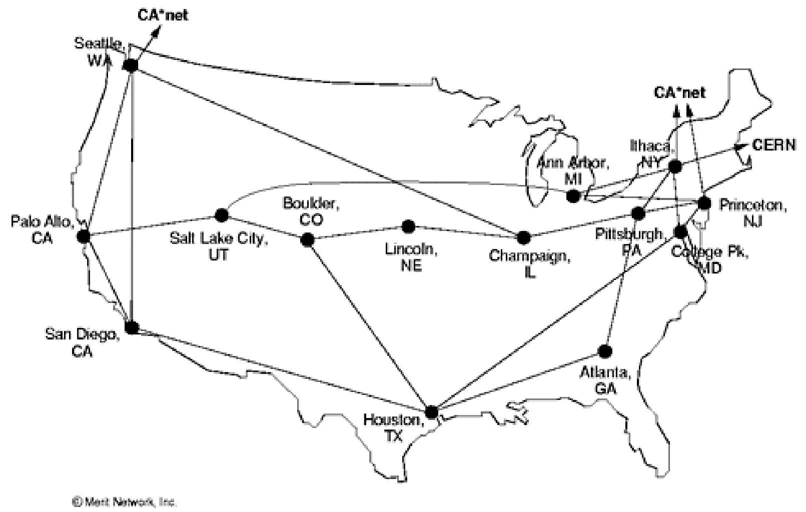
define today's Internet architecture

Internet history

1980-1990: new protocols, a proliferation of networks

- 1983: deployment of TCP/IP
- 1982: smtp e-mail protocol defined
- 1983: DNS defined for name-to-IP-address translation
- 1985: ftp protocol defined
- 1988: TCP congestion control
- new national networks: CSnet, BITnet, NSFnet, Minitel
- 100,000 hosts connected to confederation of networks

NSFNET T1 Network 1991



Internet history

1990, 2000s: commercialization, the Web, new applications

- early 1990s: ARPAnet decommissioned
- 1991: NSF lifts restrictions on commercial use of NSFnet (decommissioned, 1995)
- early 1990s: Web
  - hypertext [Bush 1945, Nelson 1960's]
  - HTML, HTTP: Berners-Lee
  - 1994: Mosaic, later Netscape
  - late 1990s: commercialization of the Web

late 1990s – 2000s:

- more killer apps: instant messaging, P2P file sharing
- network security to forefront
- est. 50 million host, 100 million+ users
- backbone links running at Gbps

Internet history

2005-present: scale, SDN, mobility, cloud

- aggressive deployment of broadband home access (10-100's Mbps)
- 2008: software-defined networking (SDN)
- increasing ubiquity of high-speed wireless access: 4G/5G, WiFi
- service providers (Google, FB, Microsoft) create their own networks
  - bypass commercial Internet to connect "close" to end user, providing "instantaneous" access to social media, search, video content, …
- enterprises run their services in "cloud" (e.g., Amazon Web Services, Microsoft Azure)
- rise of smartphones: more mobile than fixed devices on Internet (2017)
- ~15B devices attached to Internet (2023, statista.com)

Chapter 1: summary

We've covered a "ton" of material!

- Internet overview
- what's a protocol?
- network edge, access network, core
  - packet-switching versus circuit-switching
  - Internet structure
- performance: loss, delay, throughput
- layering, service models
- security
- history

You now have:

- context, overview, vocabulary, "feel" of networking
- more depth, detail, *and fun* to follow*!*

Chapter 2
Application Layer

Application layer: overview

- Principles of network applications
- Web and HTTP
- E-mail, SMTP, IMAP
- The Domain Name System DNS
- P2P applications
- video streaming and content distribution networks
- socket programming with UDP and TCP

Application layer: overview

Our goals:

- conceptual *and* implementation aspects of application-layer protocols
  - transport-layer service models
  - client-server paradigm
  - peer-to-peer paradigm
- learn about protocols by examining popular application-layer protocols and infrastructure
  - HTTP
  - SMTP, IMAP
  - DNS
  - video streaming systems, CDNs
- programming network applications
  - socket API

Some network apps

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

Q: *your* favorites?

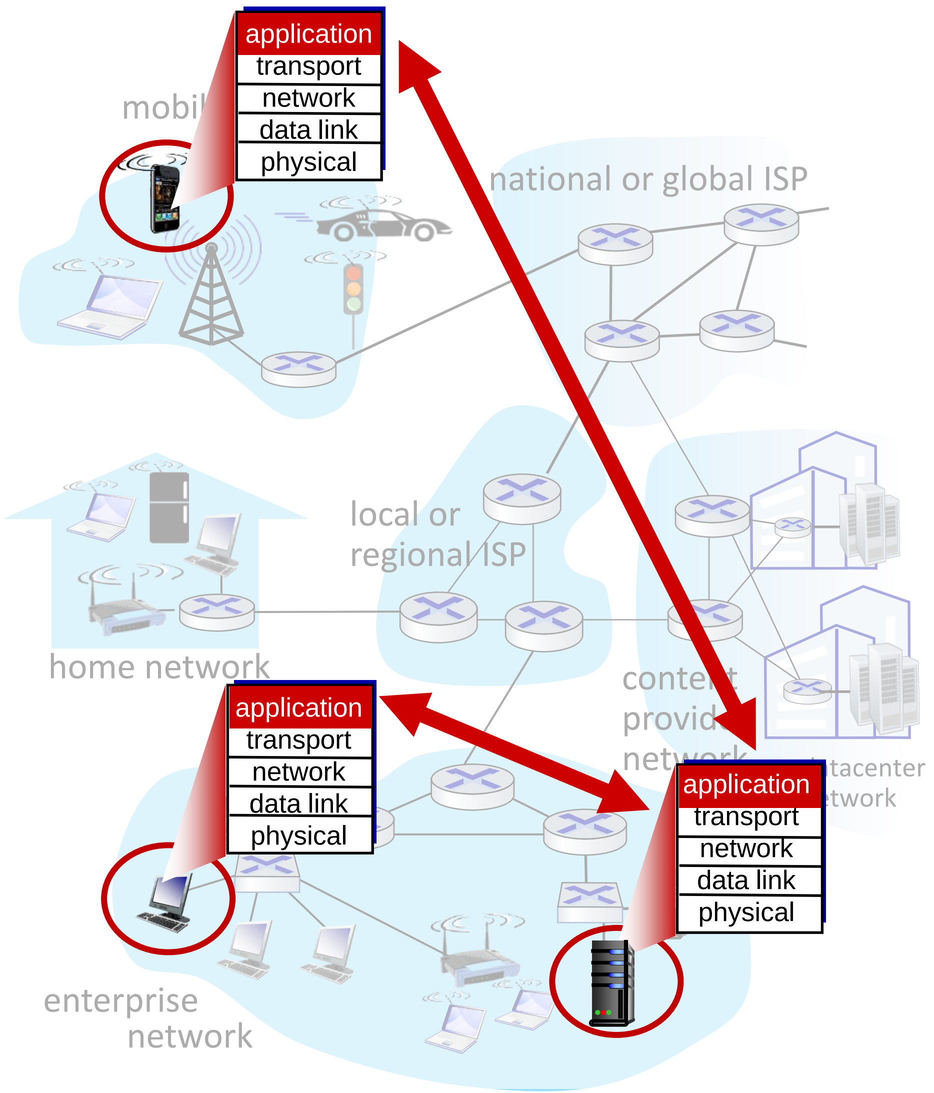
Creating a network app

write programs that:

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

no need to write software for network-core devices

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



Client-server paradigm

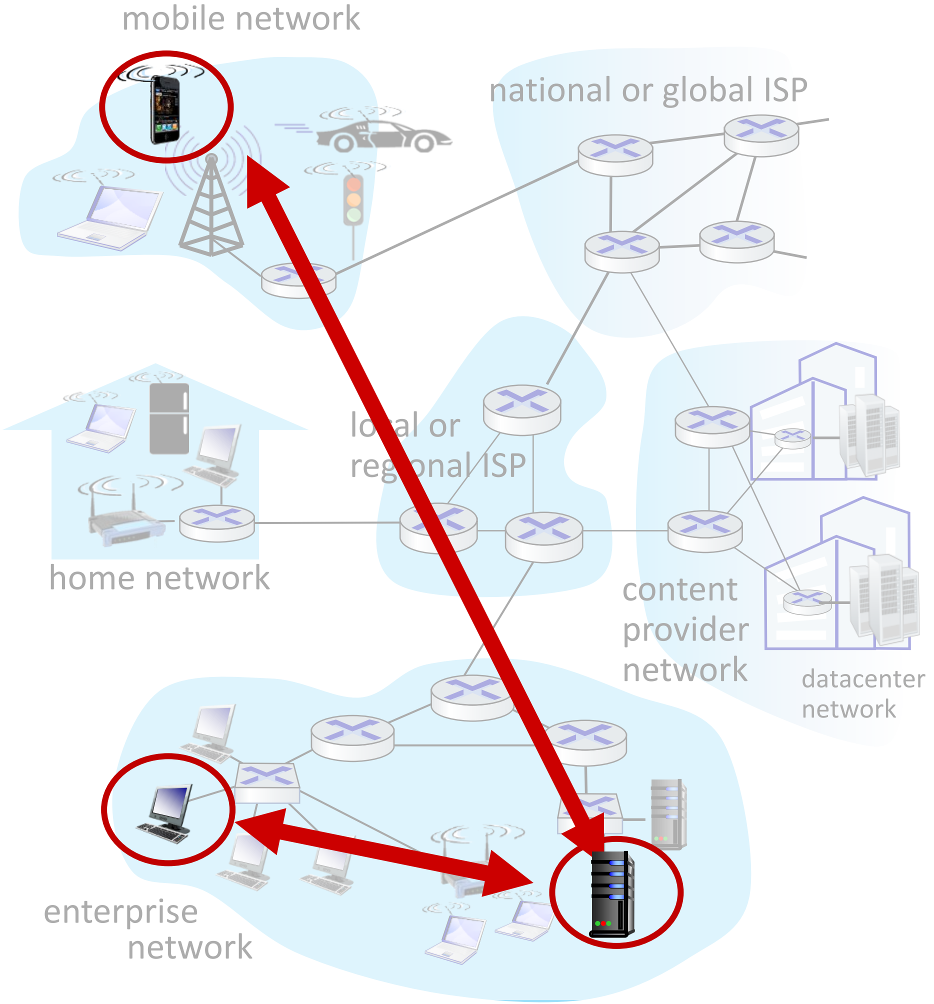
server:

- always-on host
- permanent IP address
- often in data centers, for scaling

clients:

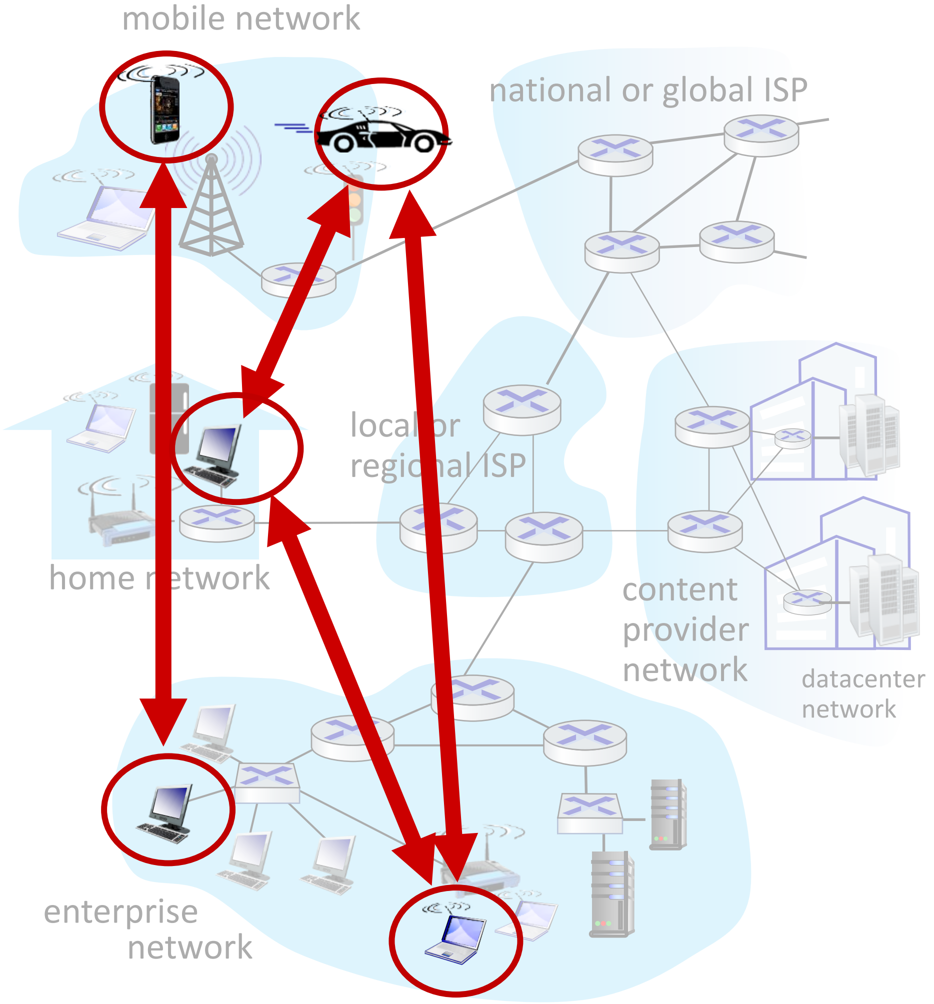
- contact, communicate with server
- may be intermittently connected
- may have dynamic IP addresses
- do *not* communicate directly with each other

- examples: HTTP, IMAP, FTP



Peer-peer 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
- example: P2P file sharing [BitTorrent]



Processes communicating

process: program running within a host

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

clients, servers

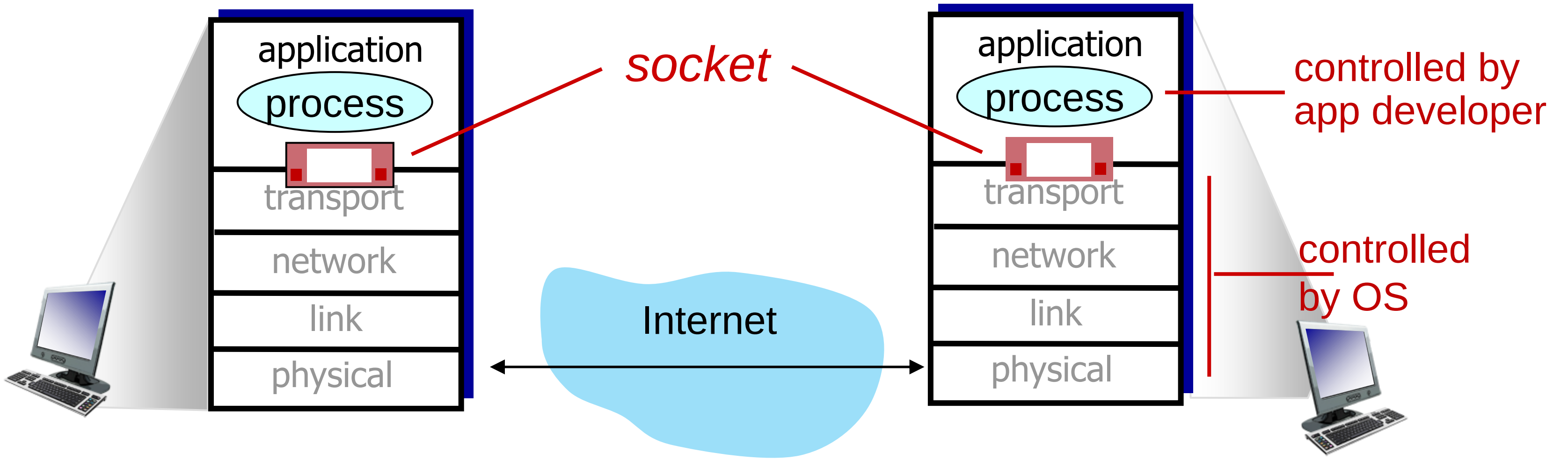
client process: process that initiates communication

server process: process that waits to be contacted

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

Sockets

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



Addressing processes

- to receive messages, process must have *identifier*
- host device has unique 32-bit IP address
- *Q:* does IP address of host on which process runs suffice for identifying the process?
  - *A:* no, *many* processes can be running on same host
- *identifier* includes both IP address and port numbers associated with process on host.
- example port numbers:
  - HTTP server: 80
  - mail server: 25
- to send HTTP message to gaia.cs.umass.edu web server:
  - IP address: 128.119.245.12
  - port number: 80
- more shortly…

An application-layer protocol defines:

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

open protocols:

- defined in RFCs, everyone has access to protocol definition
- allows for interoperability
- e.g., HTTP, SMTP

proprietary protocols:

- e.g., Skype, Zoom

What transport service does an app need?

data integrity

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

timing

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

throughput

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

security

- encryption, data integrity, …

Transport service requirements: common apps

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

Internet transport protocols services

TCP service:

- *reliable transport* between sending and receiving process
- *flow control:* sender won't overwhelm receiver
- *congestion control:* throttle sender when network overloaded
- *connection-oriented:* setup required between client and server processes
- *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, security, or connection setup.

Q: why bother? *Why* is there a UDP?

Internet applications, and transport protocols

application	application layer protocol	transport protocol
file transfer/download	FTP [RFC 959]	TCP
e-mail	SMTP [RFC 5321]	TCP
Web documents	HTTP [RFC 7230, 9110]	TCP
Internet telephony	SIP [RFC 3261], RTP [RFC 3550], or proprietary	TCP or UDP
streaming audio/video	HTTP [RFC 7230], DASH	TCP
interactive games	WOW, FPS (proprietary)	UDP or TCP

Securing TCP

Vanilla TCP & UDP sockets:

- no encryption
- cleartext passwords sent into socket traverse Internet in cleartext (!)

Transport Layer Security (TLS)

- provides encrypted TCP connections
- data integrity
- end-point authentication

TLS implemented in application layer

- apps use TLS libraries, that use TCP in turn
- cleartext sent into "socket" traverse Internet *encrypted*
- more: Chapter 8

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Web and HTTP

First, a quick review…

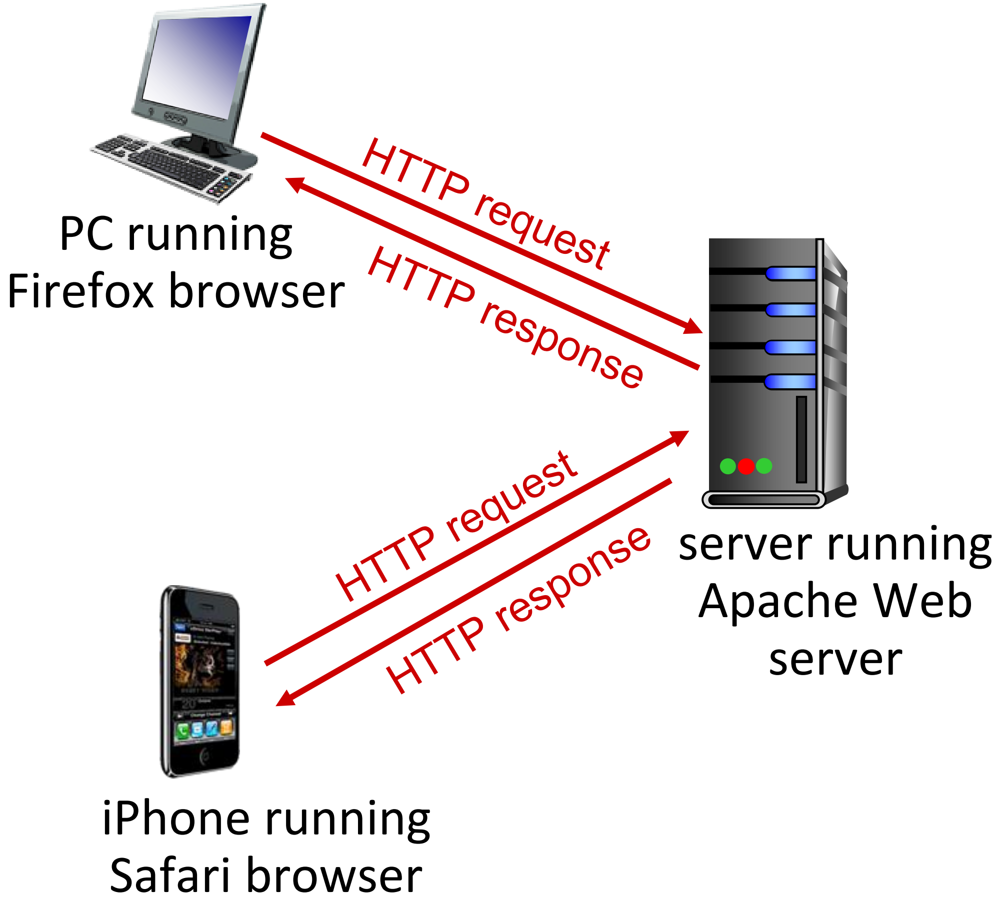
- web page consists of *objects,* each of which can be stored on different Web servers
- object can be HTML file, JPEG image, Java applet, audio file,…
- web page consists of *base HTML-file* which includes *several referenced objects, each* addressable by a *URL,* e.g.,

$$\underbrace{\texttt{www.someschool.edu}}_{\text{host name}}\underbrace{\texttt{/someDept/pic.gif}}_{\text{path name}}$$

HTTP overview

HTTP: hypertext transfer protocol

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



HTTP overview (continued)

HTTP uses TCP:

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

HTTP is "stateless"

- server maintains *no* information about past client requests

aside

protocols that maintain "state" are complex!

- past history (state) must be maintained
- if server/client crashes, their views of "state" may be inconsistent, must be reconciled

HTTP connections: two types

Non-persistent HTTP

1. TCP connection opened
2. at most one object sent over TCP connection
3. TCP connection closed

downloading multiple objects required multiple connections

Persistent HTTP

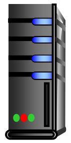
- TCP connection opened to a server
- multiple objects can be sent over *single* TCP connection between client, and that server
- TCP connection closed

Non-persistent HTTP: example

User enters URL: **`www.someSchool.edu/someDepartment/home.index`**

(containing text, references to 10 jpeg images)

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



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

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

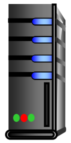
3. HTTP server receives request message, forms *response message* containing requested object, and sends message into its socket

time

Non-persistent HTTP: example (cont.)

User enters URL: **`www.someSchool.edu/someDepartment/home.index`**

(containing text, references to 10 jpeg images)



4. HTTP server closes TCP connection.

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

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

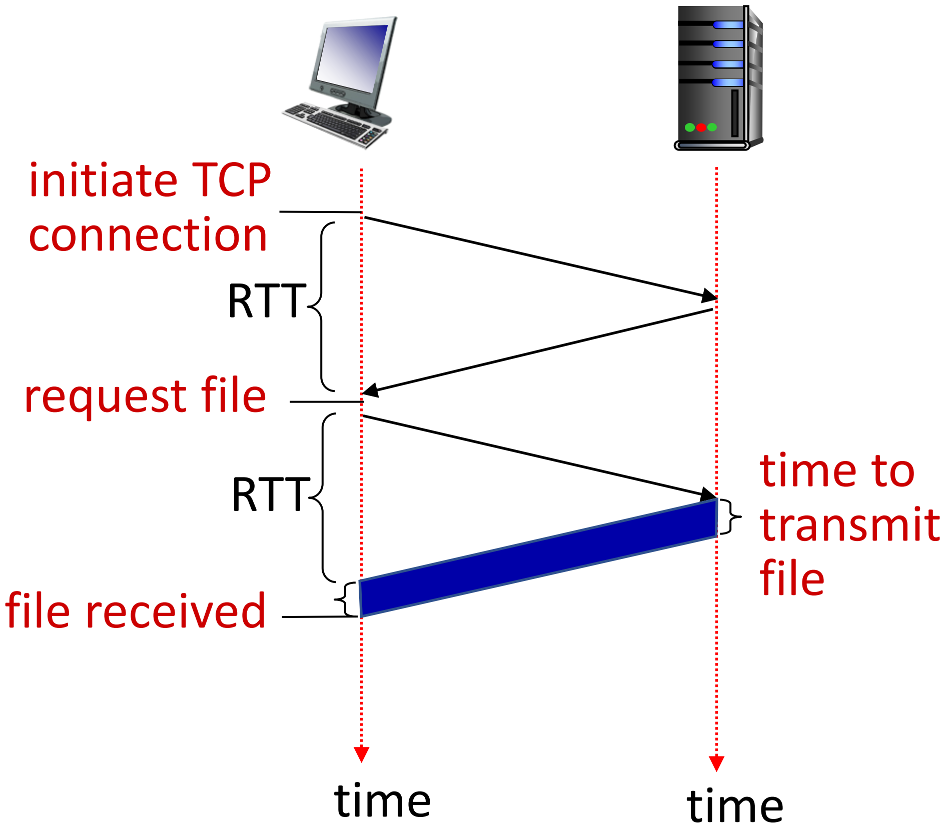
time

Non-persistent HTTP: response time

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

HTTP response time (per object):

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



Non-persistent HTTP response time = 2RTT+ file transmission time

Persistent HTTP (HTTP 1.1)

Non-persistent HTTP issues:

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

Persistent HTTP (HTTP1.1):

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

HTTP request message

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

carriage return character

line-feed character

request line (GET, POST, HEAD commands)

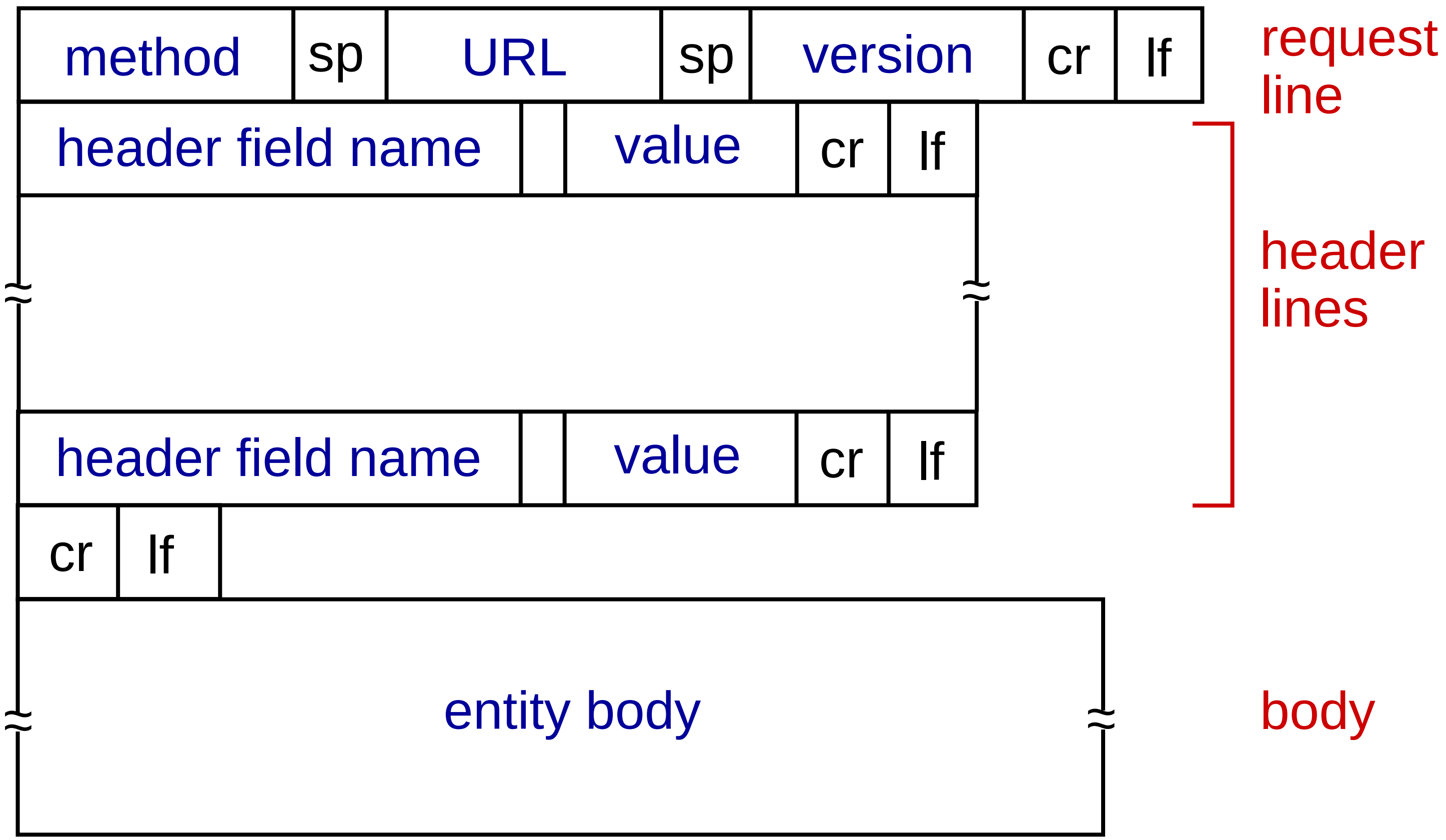
header lines

```
Host: www-net.cs.umass.edu\r\n
User-Agent: Mozilla/5.0 (Macintosh; Intel Mac OS X
   10.15; rv:80.0) Gecko/20100101 Firefox/80.0 \r\n
Accept: text/html,application/xhtml+xml\r\n
Accept-Language: en-us,en;q=0.5\r\n
Accept-Encoding: gzip,deflate\r\n
Connection: keep-alive\r\n
\r\n
```

carriage return, line feed at start of line indicates end of header lines

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

HTTP request message: general format



Other HTTP request messages

POST method:

- web page often includes form input
- user input sent from client to server in entity body of HTTP POST request message

GET method (for sending data to server):

- include user data in URL field of HTTP GET request message (following a '?'):

HEAD method:

- requests headers (only) that would be returned *if* specified URL were requested with an HTTP GET method.

PUT method:

- uploads new file (object) to server
- completely replaces file that exists at specified URL with content in entity body of POST HTTP request message

HTTP response message

status line (protocol status code status phrase)

header lines

data, e.g., requested HTML file

```
HTTP/1.1 200 OK
Date: Tue, 08 Sep 2020 00:53:20 GMT
Server: Apache/2.4.6 (CentOS)
   OpenSSL/1.0.2k-fips PHP/7.4.9
   mod_perl/2.0.11 Perl/v5.16.3
Last-Modified: Tue, 01 Mar 2016 18:57:50 GMT
ETag: "a5b-52d015789ee9e"
Accept-Ranges: bytes
Content-Length: 2651
Content-Type: text/html; charset=UTF-8
\r\n
data data data data data ...
```

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

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 message

  301 Moved Permanently
  - requested object moved, new location specified later in this message (in Location: field)

  400 Bad Request
  - request msg not understood by server

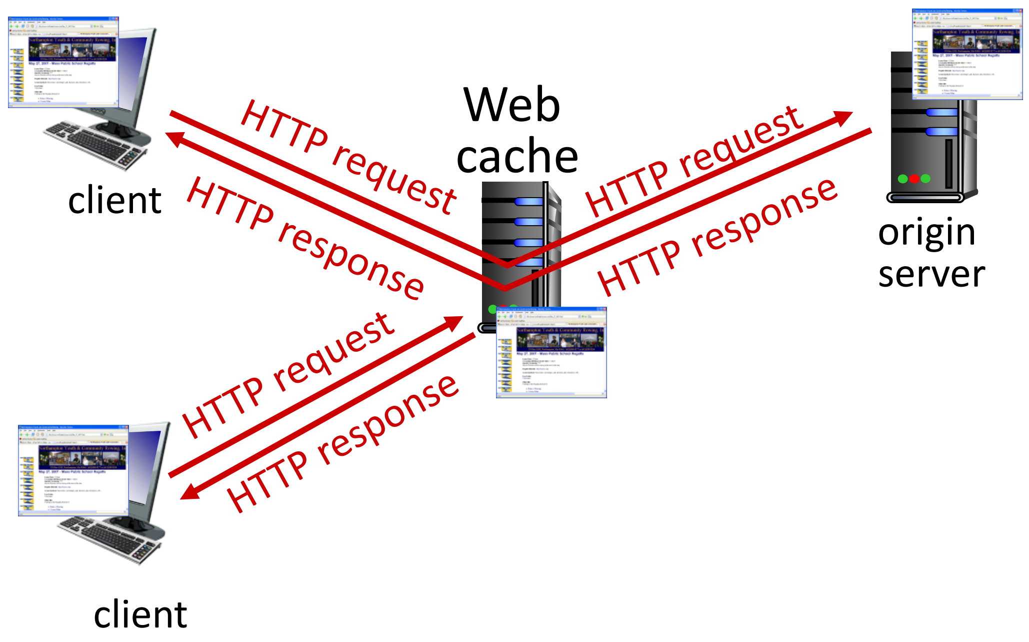
  404 Not Found
  - requested document not found on this server

  505 HTTP Version Not Supported

Web caches

Goal: satisfy client requests without involving origin server

- user configures browser to point to a (local) *Web cache*
- browser sends all HTTP requests to cache
  - *if* object in cache: cache returns object to client
  - *else* cache requests object from origin server, caches received object, then returns object to client



Web caches (aka proxy servers)

- Web cache acts as both client and server
  - server for original requesting client
  - client to origin server
- server tells cache about object's allowable caching in response header:

```
Cache-Control: max-age=<seconds>
```

```
Cache-Control: no-cache
```

Why Web caching?

- reduce response time for client request
  - cache is closer to client
- reduce traffic on an institution's access link
- Internet is dense with caches
  - enables "poor" content providers to more effectively deliver content

Caching example

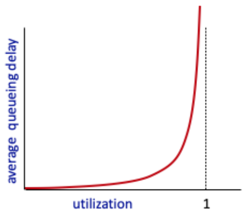
Scenario:

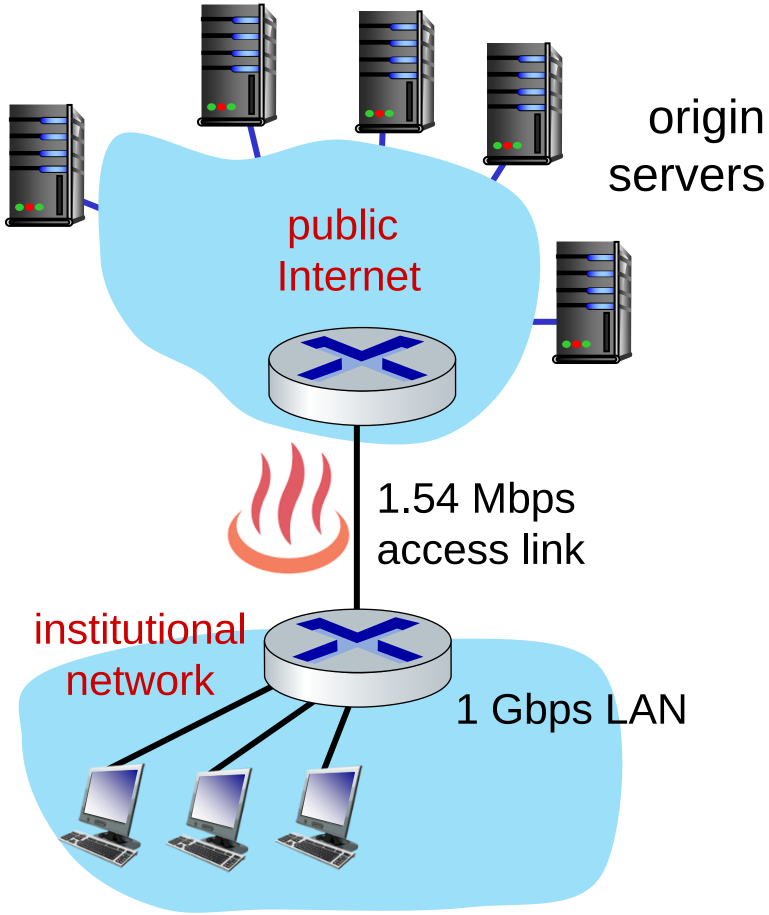
- access link rate: 1.54 Mbps
- RTT from institutional router to server: 2 sec
- web object size: 100K bits
- average request rate from browsers to origin servers: 15/sec
  - avg data rate to browsers: 1.50 Mbps

Performance:

- access link utilization = .97
- LAN utilization: .0015
- end-end delay = Internet delay + access link delay + LAN delay
  = 2 sec + minutes + usecs

problem: large queueing delays at high utilization!





Option 1: buy a faster access link

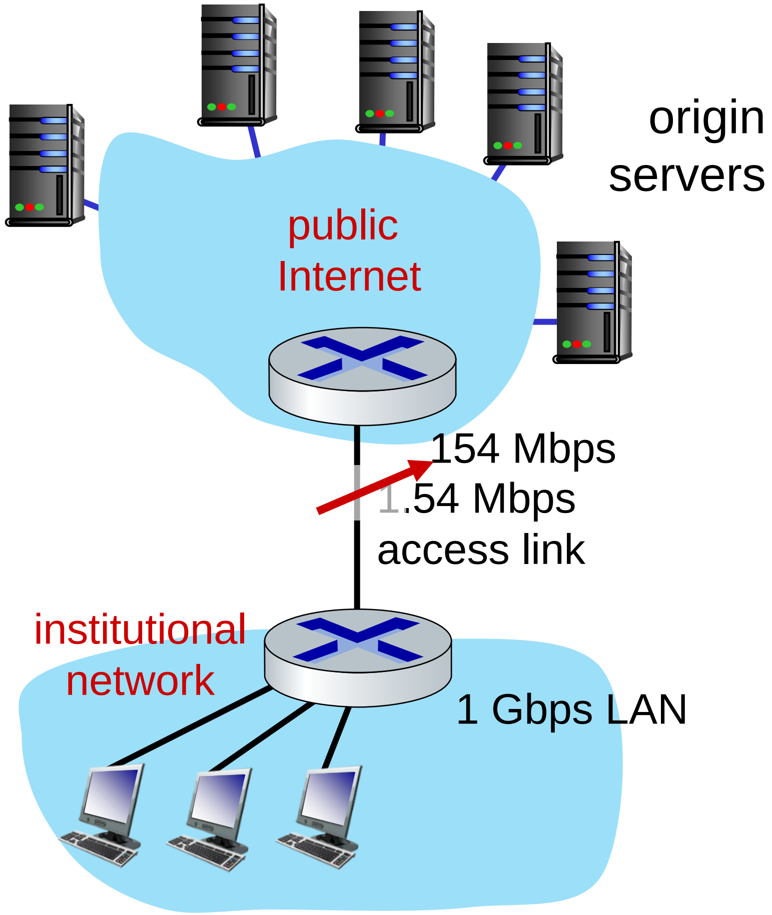
Scenario:

- access link rate: ~~1.54~~ Mbps 154 Mbps
- RTT from institutional router to server: 2 sec
- web object size: 100K bits
- average request rate from browsers to origin servers: 15/sec
  - avg data rate to browsers: 1.50 Mbps

Performance:

- access link utilization = ~~.97~~ .0097
- LAN utilization: .0015
- end-end delay = Internet delay + access link delay + LAN delay
  = 2 sec + ~~minutes~~ + usecs msecs

Cost: faster access link (expensive!)



Option 2: install a web cache

Scenario:

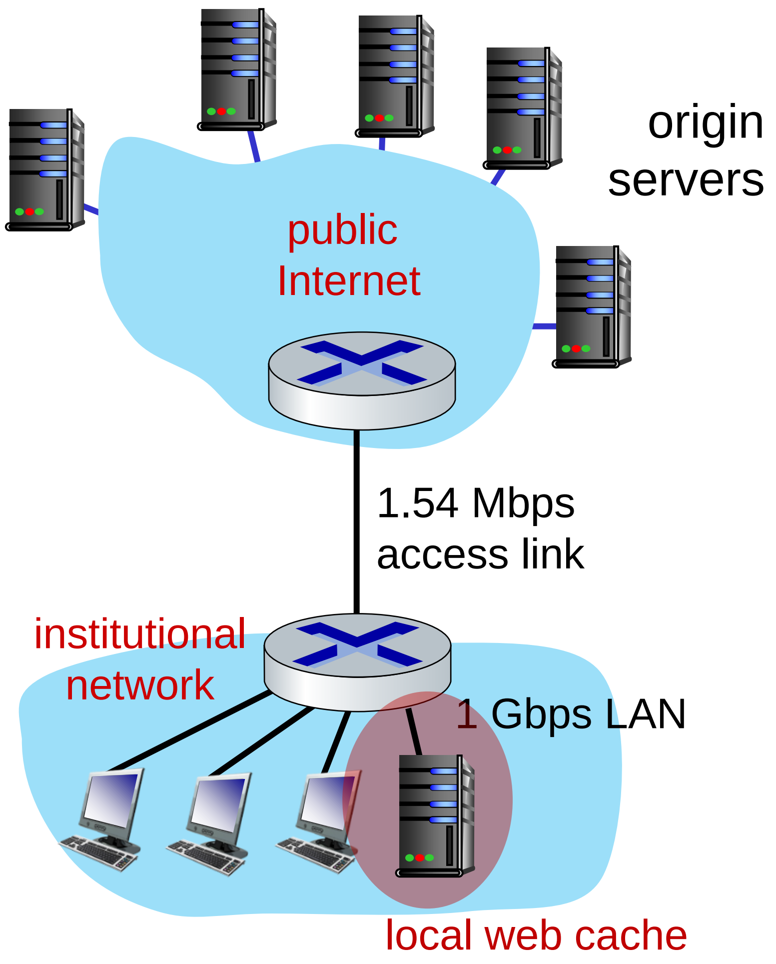
- access link rate: 1.54 Mbps
- RTT from institutional router to server: 2 sec
- web object size: 100K bits
- average request rate from browsers to origin servers: 15/sec
  - avg data rate to browsers: 1.50 Mbps

Cost: web cache (cheap!)

Performance:

- LAN utilization: .?
- access link utilization = ?
- average end-end delay = ?

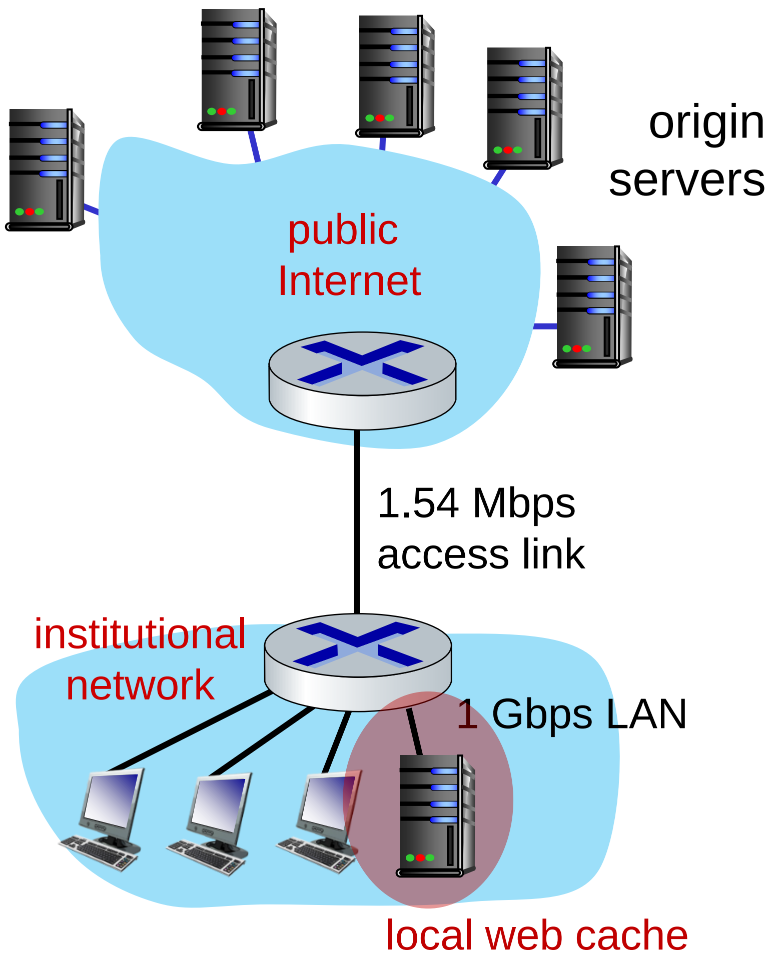
How to compute link utilization, delay?



Calculating access link utilization, end-end delay with cache:

suppose cache hit rate is 0.4:

- 40% requests served by cache, with low (msec) delay
- 60% requests satisfied at origin
  - rate to browsers over access link
    = 0.6 * 1.50 Mbps = .9 Mbps
  - access link utilization = 0.9/1.54 = .58 means low (msec) queueing delay at access link
- average end-end delay:
  = 0.6 * (delay from origin servers)
  + 0.4 * (delay when satisfied at cache)
  = 0.6 (2.01) + 0.4 (~msecs) = ~ 1.2 secs



lower average end-end delay than with 154 Mbps link (and cheaper too!)

Browser caching: Conditional GET

Goal: don't send object if browser has up-to-date cached version

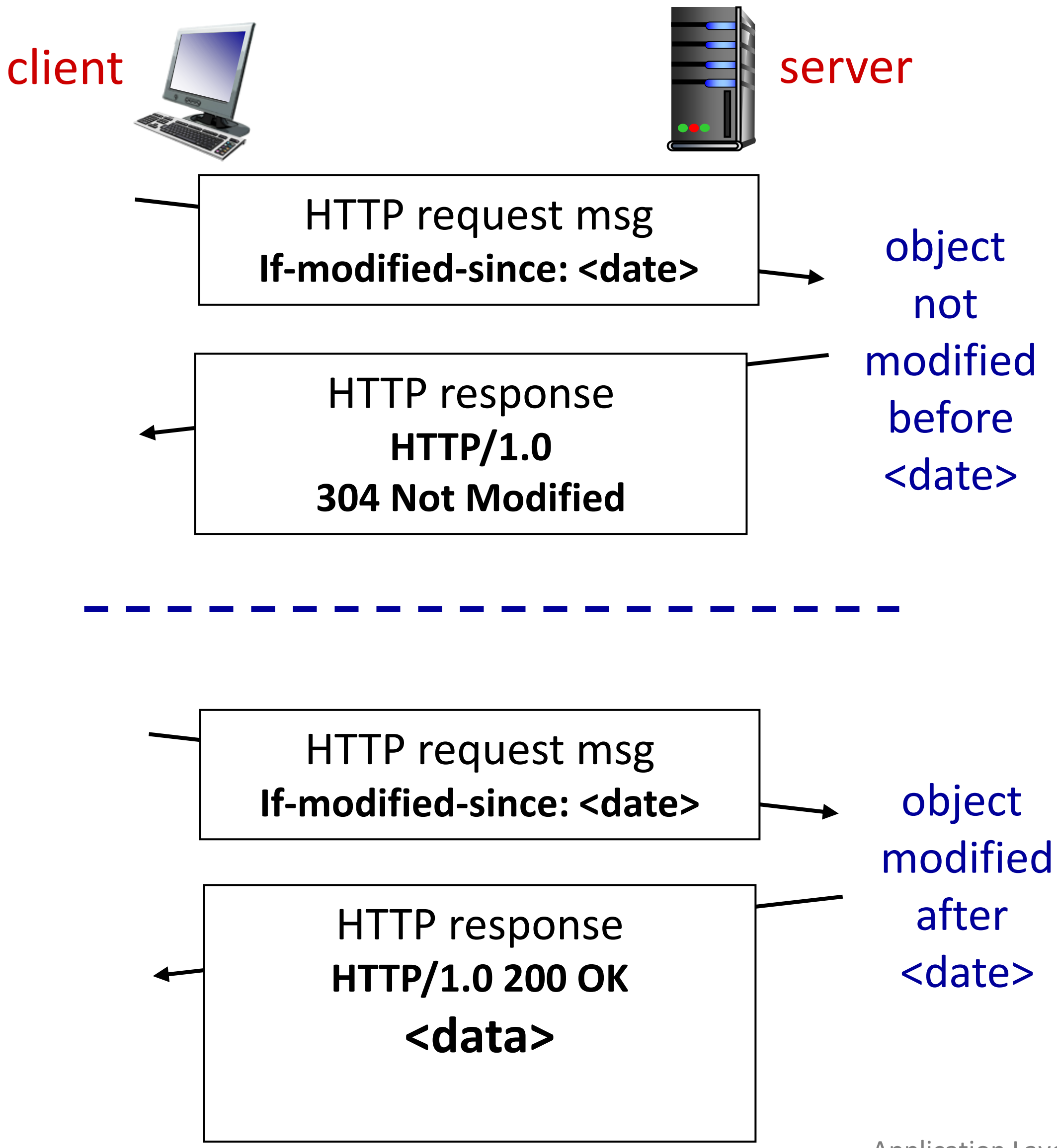
- no object transmission delay (or use of network resources)

- *client:* specify date of browser-cached copy in HTTP request

  If-modified-since: <date>

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

  HTTP/1.0 304 Not Modified



HTTP/2

Key goal: decreased delay in multi-object HTTP requests

HTTP1.1: introduced multiple, pipelined GETs over single TCP connection

- server responds *in-order* (FCFS: first-come-first-served scheduling) to GET requests
- with FCFS, small object may have to wait for transmission (head-of-line (HOL) blocking) behind large object(s)
- loss recovery (retransmitting lost TCP segments) stalls object transmission

HTTP/2

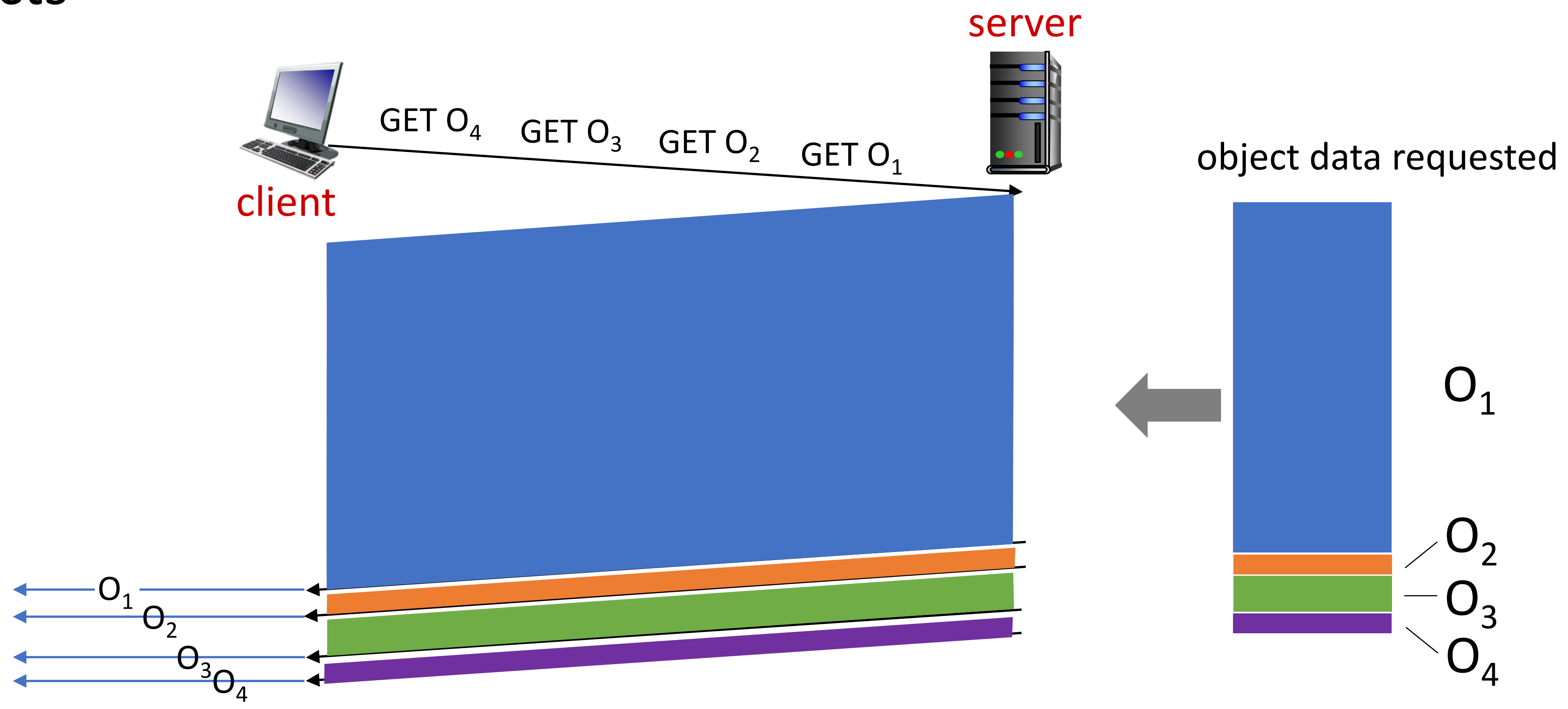
Key goal: decreased delay in multi-object HTTP requests

HTTP/2: [RFC 7540, 2015] increased flexibility at *server* in sending objects to client:

- methods, status codes, most header fields unchanged from HTTP 1.1
- transmission order of requested objects based on client-specified object priority (not necessarily FCFS)
- *push* unrequested objects to client
- divide objects into frames, schedule frames to mitigate HOL blocking

HTTP/2: mitigating HOL blocking

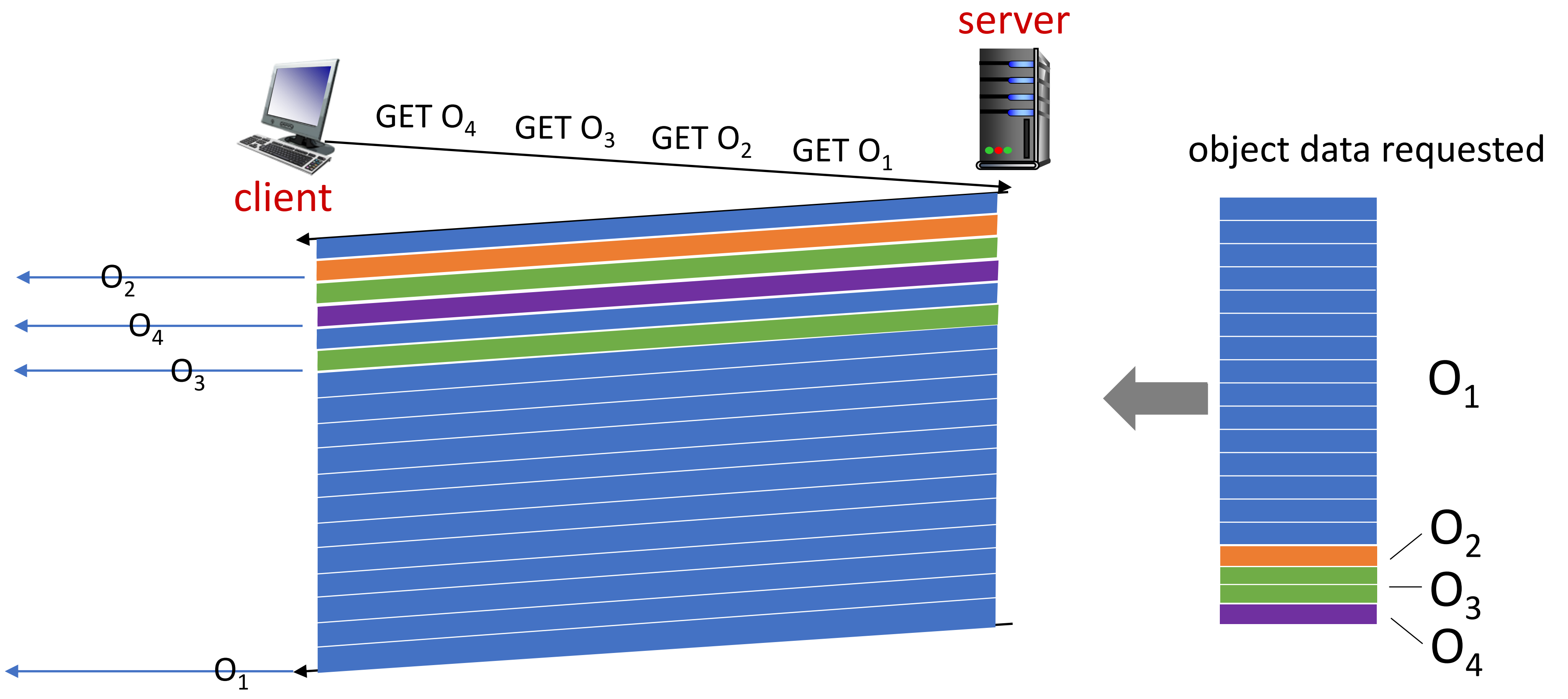
HTTP 1.1: client requests 1 large object (e.g., video file) and 3 smaller objects



objects delivered in order requested: O_2, O_3, O_4 wait behind O_1

HTTP/2: mitigating HOL blocking

HTTP/2: objects divided into frames, frame transmission interleaved



O_2, O_3, O_4 delivered quickly, O_1 slightly delayed

HTTP/2 to HTTP/3

HTTP/2 over single TCP connection means:

- recovery from packet loss still stalls all object transmissions
  - as in HTTP 1.1, browsers have incentive to open multiple parallel TCP connections to reduce stalling, increase overall throughput
- no security over vanilla TCP connection
- HTTP/3: adds security, per object error- and congestion-control (more pipelining) over UDP
  - more on HTTP/3 in transport layer

Application layer: overview

- Principles of network applications
- Web and HTTP
- E-mail, SMTP, IMAP
- The Domain Name System DNS
- P2P applications
- video streaming and content distribution networks
- socket programming with UDP and TCP

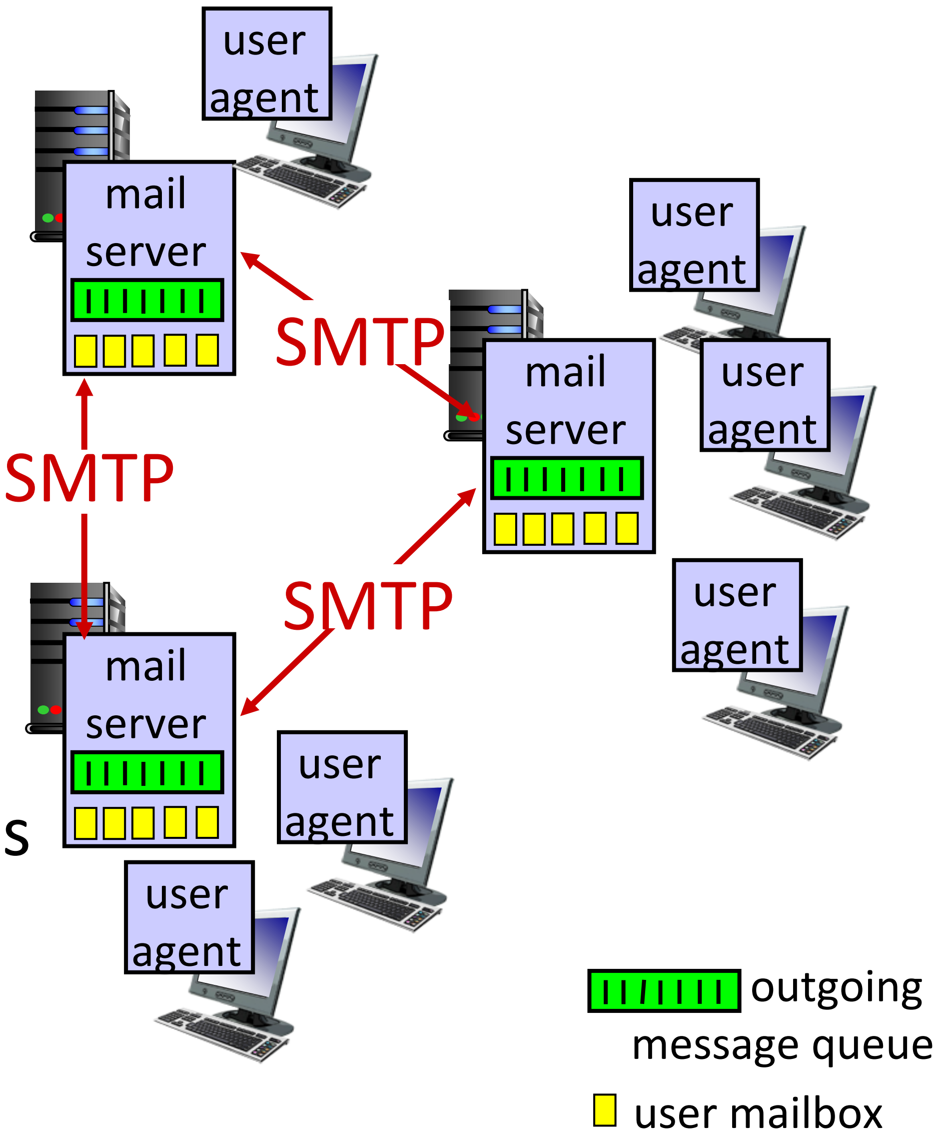
E-mail

Three major components:

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

User Agent

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



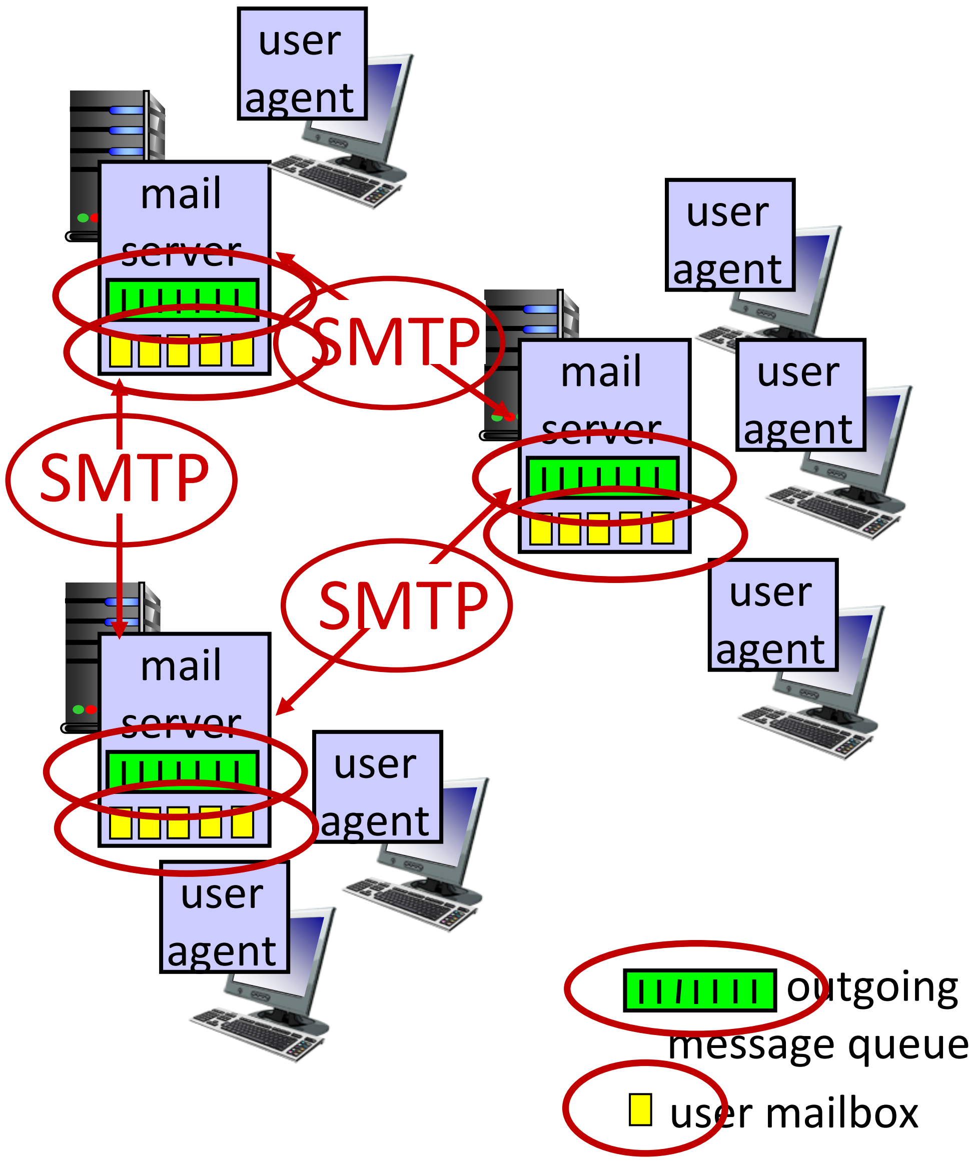
E-mail: mail servers

mail servers:

- *mailbox* contains incoming messages for user
- *message queue* of outgoing (to be sent) mail messages

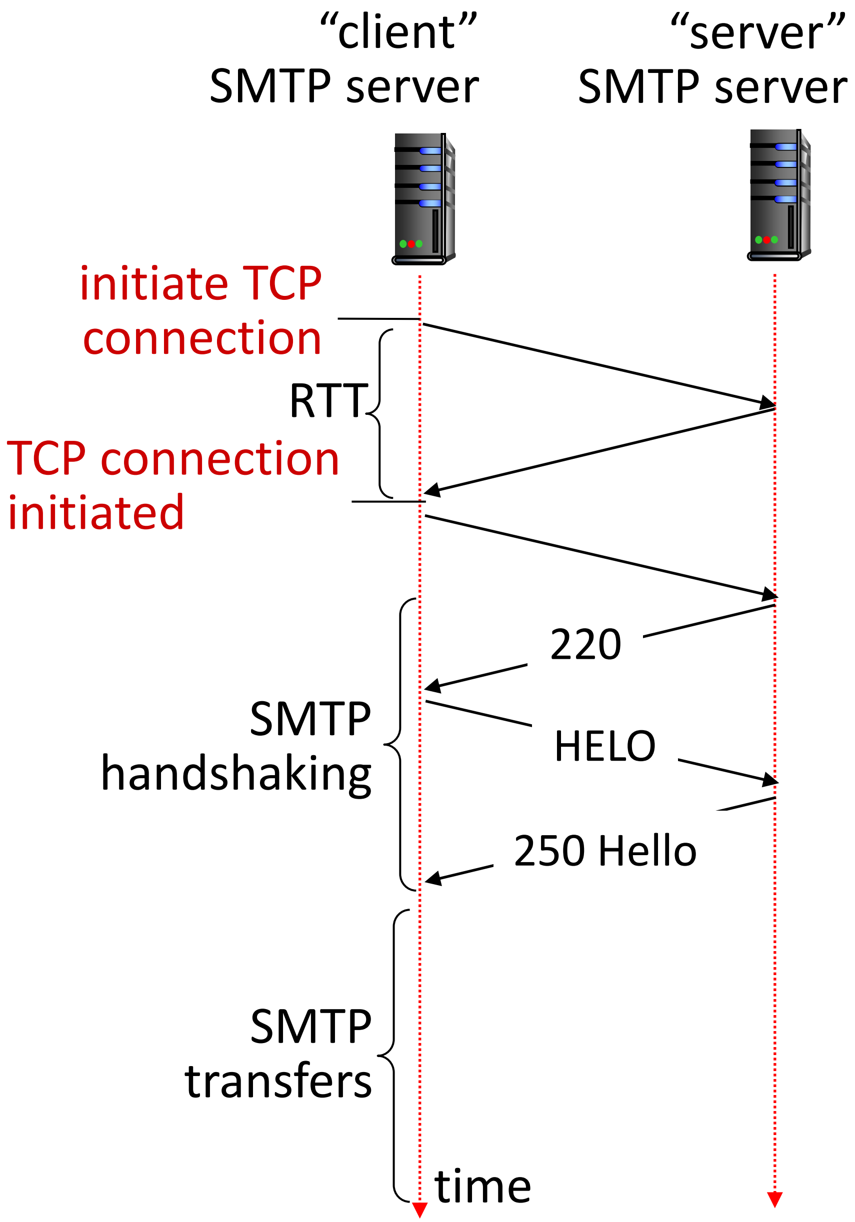
SMTP protocol between mail servers to send email messages

- client: sending mail server
- "server": receiving mail server



SMTP RFC (5321)

- uses TCP to reliably transfer email message from client (mail server initiating connection) to server, port 25
  - direct transfer: sending server (acting like client) to receiving server
- three phases of transfer
  - SMTP handshaking (greeting)
  - SMTP transfer of messages
  - SMTP closure
- command/response interaction (like HTTP)
  - commands: ASCII text
  - response: status code and phrase



Scenario: Alice sends e-mail to Bob

1) Alice uses UA to compose e-mail message "to" bob@someschool.edu

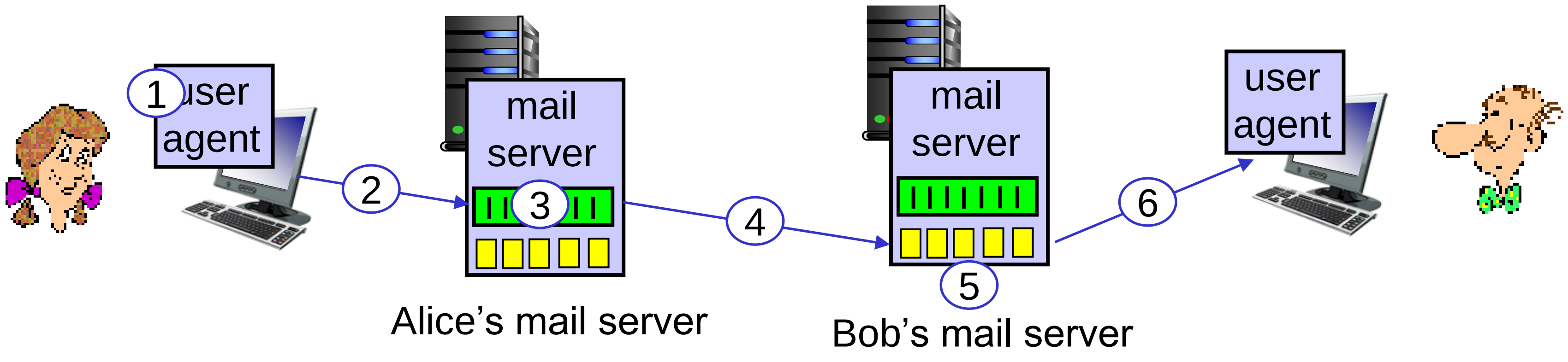
2) Alice's UA sends message to her mail server using SMTP; message placed in message queue

3) client side of SMTP at mail server opens TCP connection with Bob's mail server

4) SMTP client sends Alice's message over the TCP connection

5) Bob's mail server places the message in Bob's mailbox

6) Bob invokes his user agent to read message



Sample SMTP interaction

```
S: 220 hamburger.edu
C: HELO crepes.fr
S: 250  Hello crepes.fr, pleased to meet you
C: MAIL FROM: <alice@crepes.fr>
S: 250 alice@crepes.fr... Sender ok
C: RCPT TO: <bob@hamburger.edu>
S: 250 bob@hamburger.edu ... Recipient ok
C: DATA
S: 354 Enter mail, end with "." on a line by itself
C: Do you like ketchup?
C: How about pickles?
C: .
S: 250 Message accepted for delivery
C: QUIT
S: 221 hamburger.edu closing connection
```

SMTP: observations

comparison with HTTP:

- HTTP: client pull
- SMTP: client push
- both have ASCII command/response interaction, status codes
- HTTP: each object encapsulated in its own response message
- SMTP: multiple objects sent in multipart message

- SMTP uses persistent connections
- SMTP requires message (header & body) to be in 7-bit ASCII
- SMTP server uses CRLF.CRLF to determine end of message

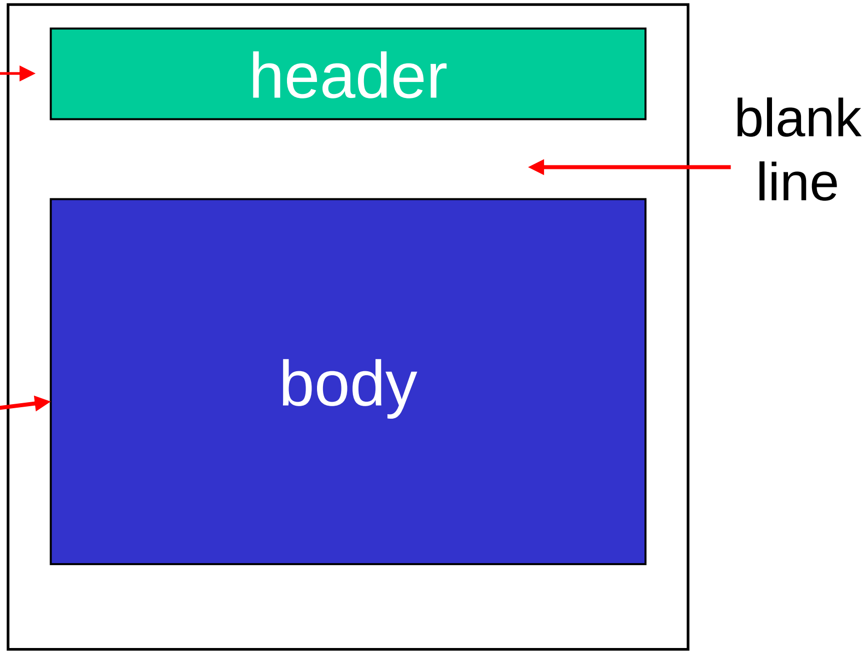
Mail message format

SMTP: protocol for exchanging e-mail messages, defined in RFC 5321 (like RFC 7231 defines HTTP)

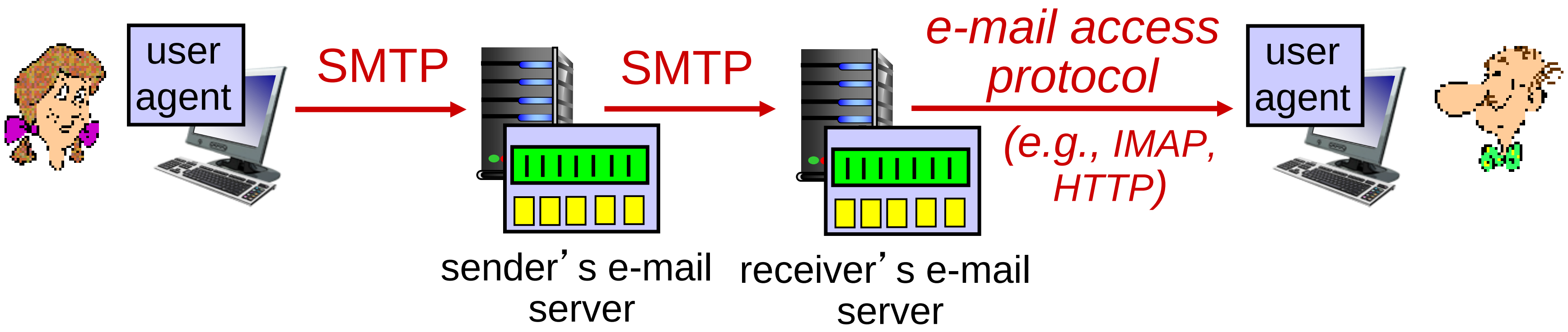
RFC 2822 defines *syntax* for e-mail message itself (like HTML defines syntax for web documents)

- header lines, e.g.,
  - To:
  - From:
  - Subject:

  these lines, within the body of the email message area different from SMTP MAIL FROM:, RCPT TO: commands!
- Body: the "message" , ASCII characters only



Retrieving email: mail access protocols



- SMTP: delivery/storage of e-mail messages to receiver's server
- mail access protocol: retrieval from server
  - IMAP: Internet Mail Access Protocol [RFC 3501]: messages stored on server, IMAP provides retrieval, deletion, folders of stored messages on server
- HTTP: gmail, Hotmail, Yahoo!Mail, etc. provides web-based interface on top of STMP (to send), IMAP (or POP) to retrieve e-mail messages

Application Layer: Overview

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DNS: Domain Name System

people: many identifiers:

- SSN, name, passport #

Internet hosts, routers:

- IP address (32 bit) - used for addressing datagrams
- "name", e.g., cs.umass.edu - used by humans

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

Domain Name System (DNS):

- *distributed database* implemented in hierarchy of many *name servers*
- *application-layer protocol:* hosts, DNS servers communicate to *resolve* names (address/name translation)
  - *note:* core Internet function, implemented as application-layer protocol
  - complexity at network's "edge"

DNS: services, structure

DNS services:

- hostname-to-IP-address translation
- host aliasing
  - canonical, alias names
- mail server aliasing
- load distribution
  - replicated Web servers: many IP addresses correspond to one name

Q: Why not centralize DNS?

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

A: doesn't scale!

- Comcast DNS servers alone: 600B DNS queries/day
- Akamai DNS servers alone: 2.2T DNS queries/day

Thinking about the DNS

humongous distributed database:

- ~ billion records, each simple

handles many *trillions* of queries/day:

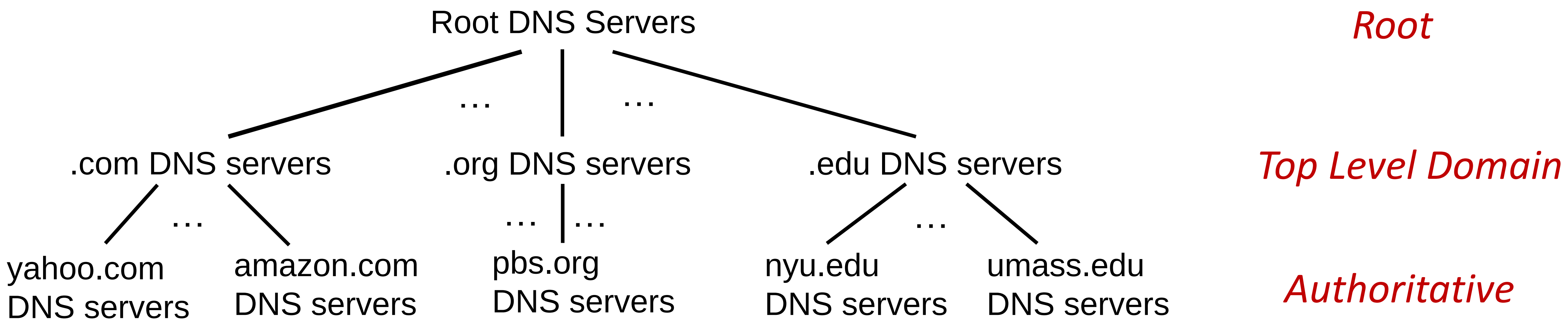
- *many* more reads than writes
- *performance matters:* almost every Internet transaction interacts with DNS - msecs count!

organizationally, physically decentralized:

- millions of different organizations responsible for their records

"bulletproof": reliability, security

DNS: a distributed, hierarchical database

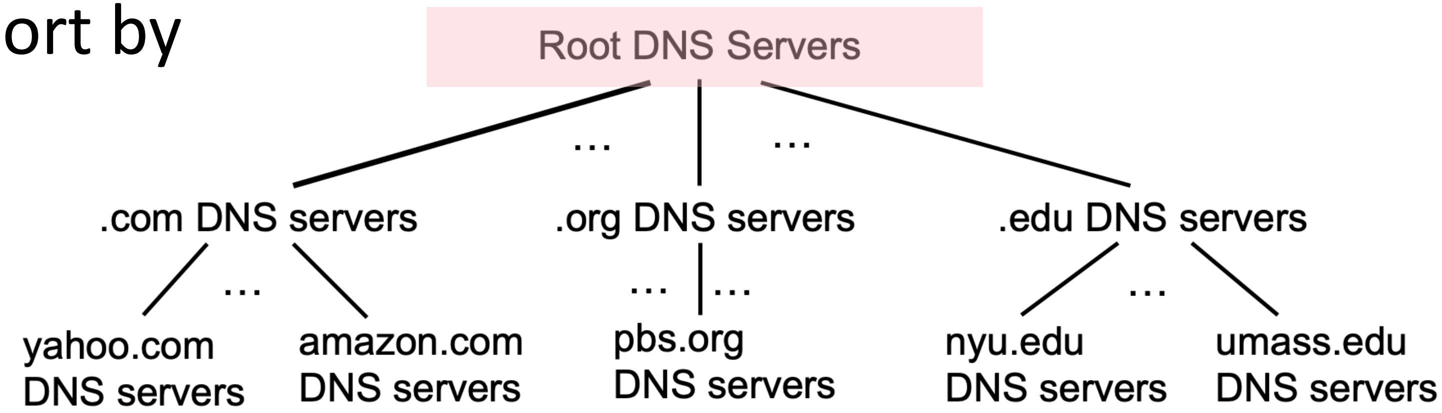


Client wants IP address for www.amazon.com; 1st approximation:

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

DNS: root name servers

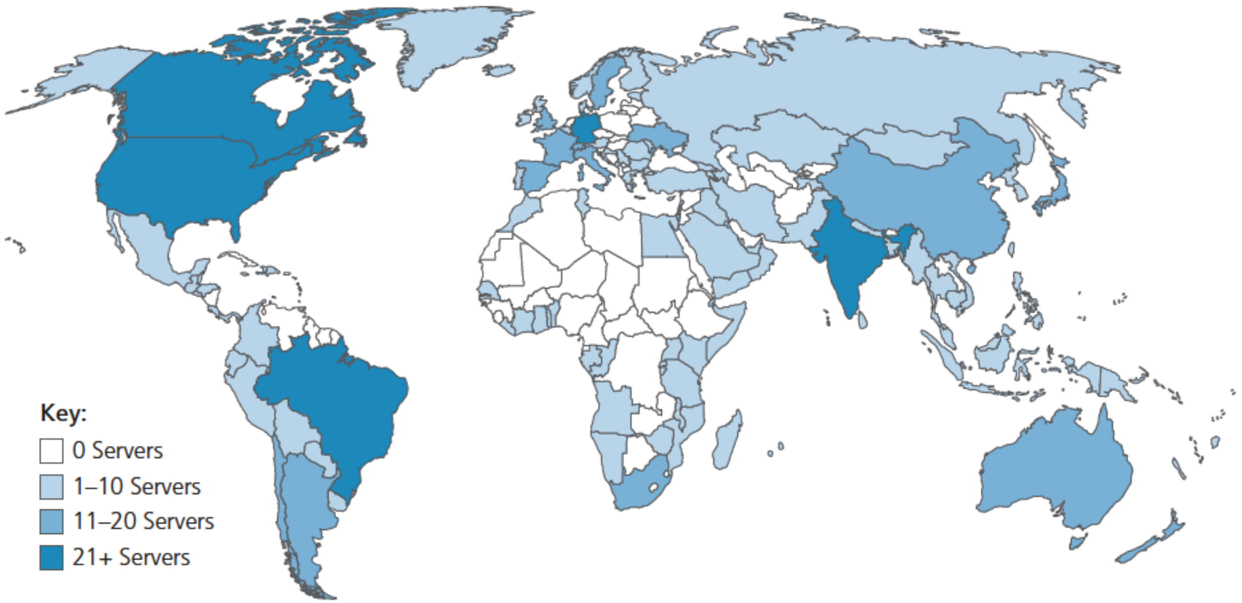
- official, contact-of-last-resort by name servers that can not resolve name



DNS: root name servers

- official, contact-of-last-resort by name servers that can not resolve name
- *incredibly important* Internet function
  - Internet couldn't function without it!
  - DNSSEC – provides security (authentication, message integrity)
- ICANN (Internet Corporation for Assigned Names and Numbers) manages root DNS domain

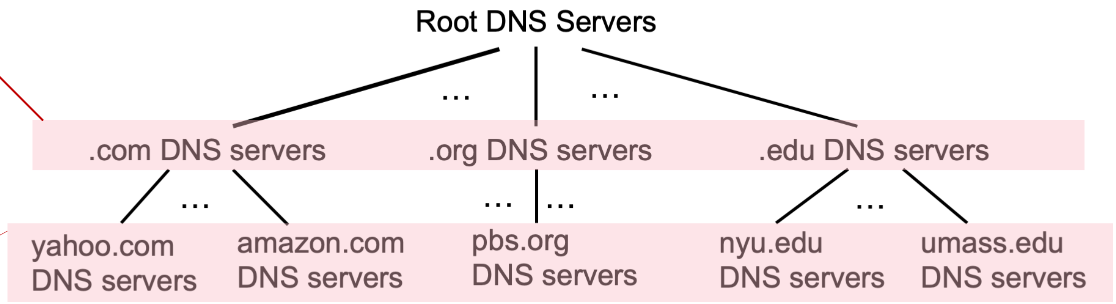
13 logical root name "servers" worldwide each "server" replicated many times (~200 servers in US)



Top-Level Domain, and authoritative servers

Top-Level Domain (TLD) servers:

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



authoritative DNS servers:

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

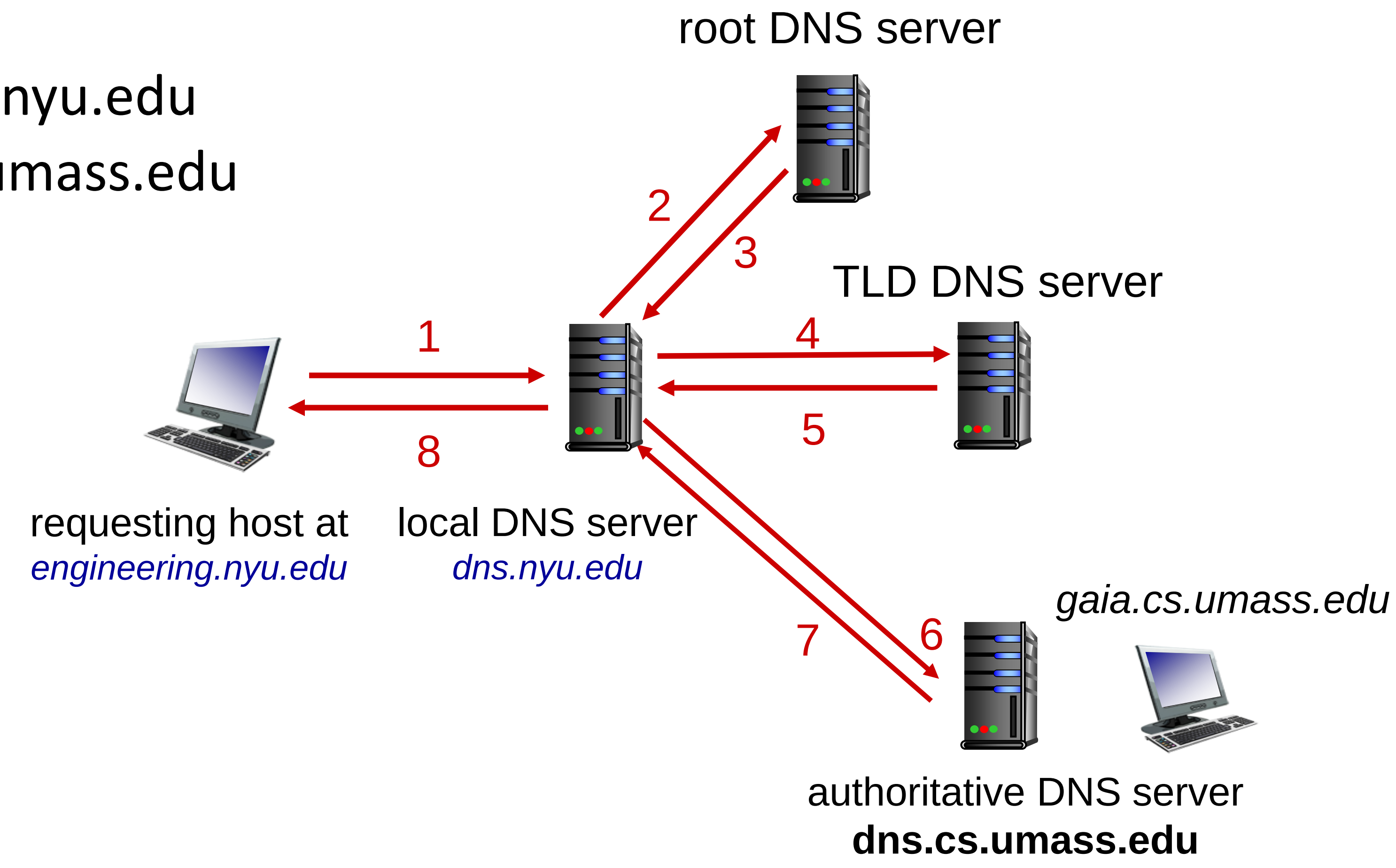
Local DNS name servers

- when host makes DNS query, it is sent to its *local* DNS server
  - Local DNS server returns reply, answering:
    - from its local cache of recent name-to-address translation pairs (possibly out of date!)
    - forwarding request into DNS hierarchy for resolution
  - each ISP has local DNS name server; to find yours:
    - MacOS: `% scutil --dns`
    - Windows: `>ipconfig /all`
- local DNS server doesn't strictly belong to hierarchy

DNS name resolution: iterated query

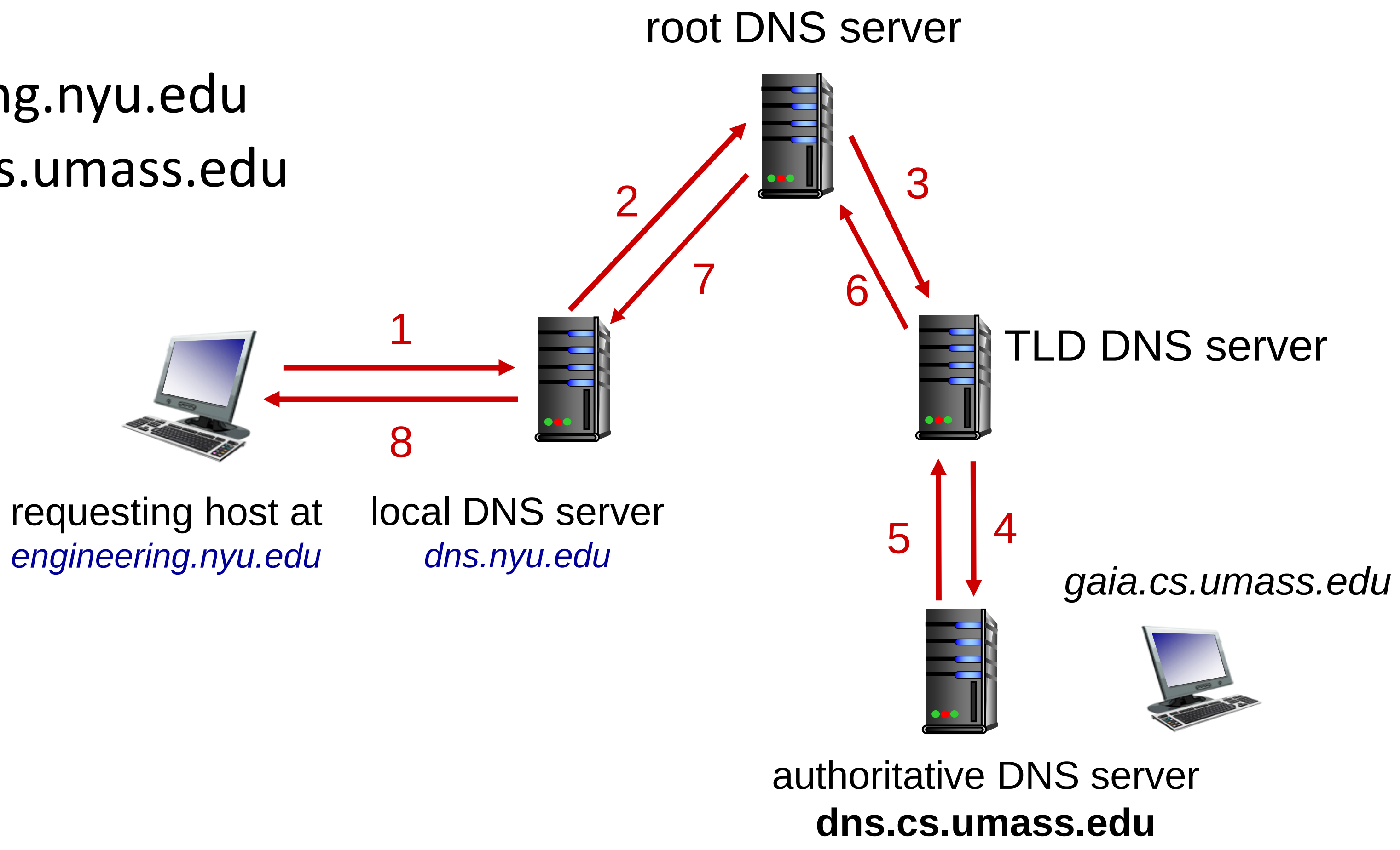
Example: host at engineering.nyu.edu wants IP address for gaia.cs.umass.edu

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



DNS name resolution: recursive query

Example: host at engineering.nyu.edu wants IP address for gaia.cs.umass.edu



Recursive query:

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

Caching DNS Information

- once (any) name server learns mapping, it *caches* mapping, and i*mmediately* returns a cached mapping in response to a query
  - caching improves response time
  - cache entries timeout (disappear) after some time (TTL)
  - TLD servers typically cached in local name servers
- cached entries may be *out-of-date*
  - if named host changes IP address, may not be known Internet-wide until all TTLs expire!
  - *best-effort name-to-address translation!*

DNS records

DNS: distributed database storing 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., foo.com)
- `value` is hostname of authoritative name server for this domain

type=CNAME

- `name` is alias name for some "canonical" (the real) name
- www.ibm.com is really servereast.backup2.ibm.com
- `value` is canonical name

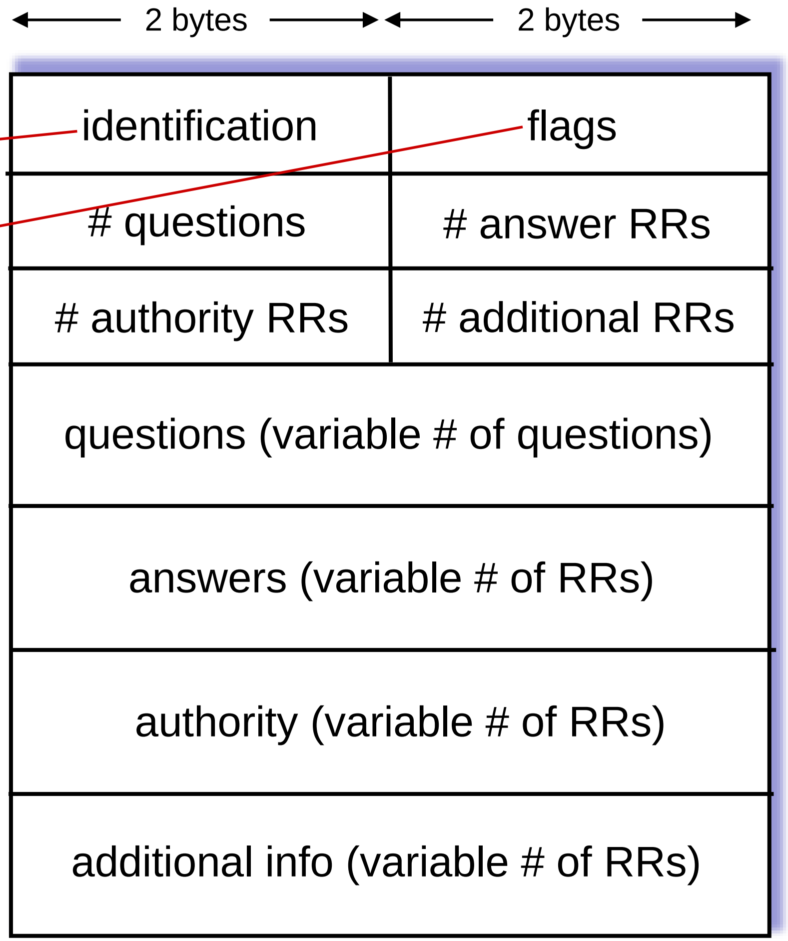
type=MX

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

DNS protocol messages

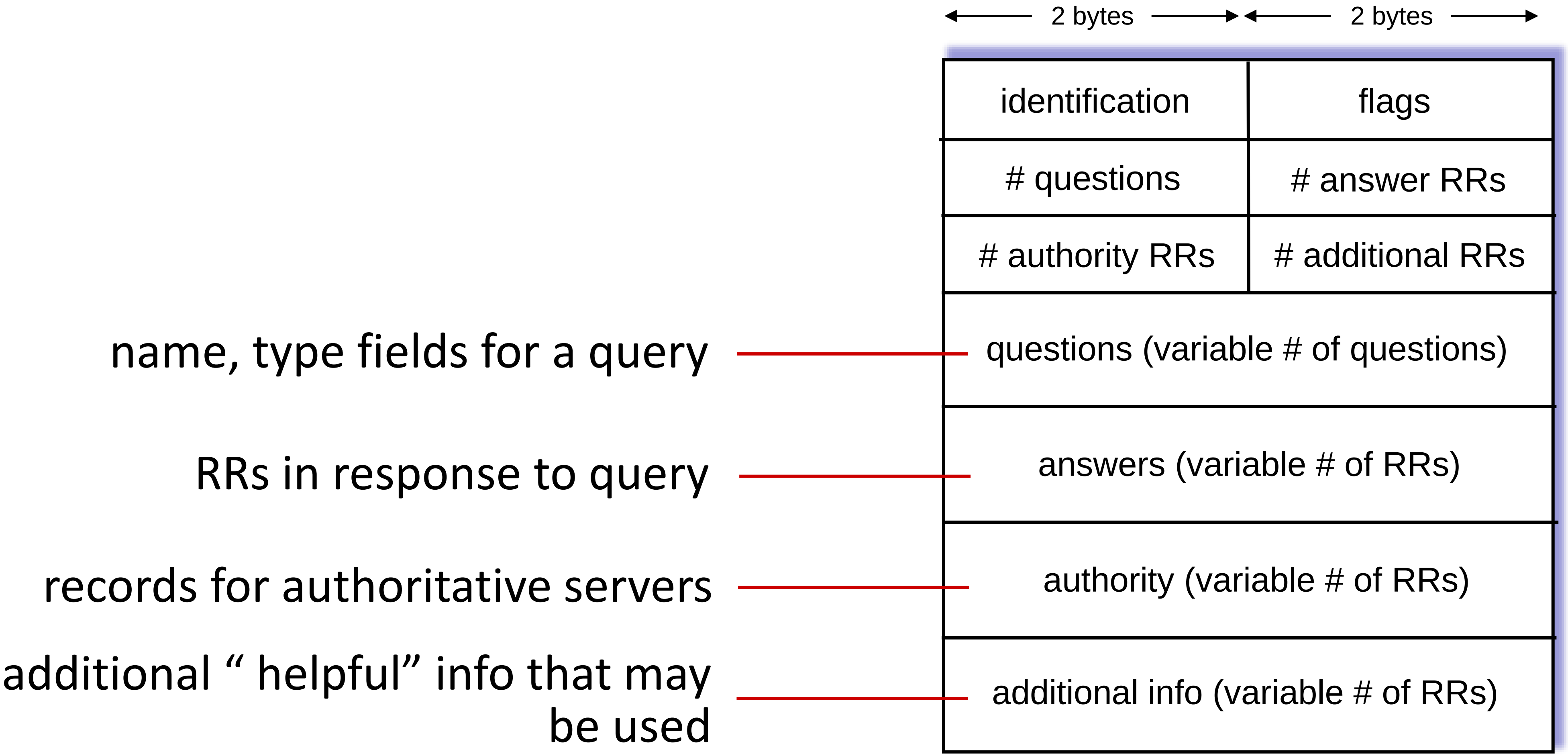
DNS *query* and *reply* messages, both have same *format:*

message header:
- identification: 16 bit # for query, reply to query uses same #
- flags:
  - query or reply
  - recursion desired
  - recursion available
  - reply is authoritative



DNS protocol messages

DNS *query* and *reply* messages, both have same *format:*



Getting your info into the DNS

example: new startup "Network Utopia"

- register name networkuptopia.com at *DNS registrar* (e.g., Network Solutions)
  - provide names, IP addresses of authoritative name server (primary and secondary)
  - registrar inserts NS, A RRs into .com TLD server:

    `(networkutopia.com, dns1.networkutopia.com, NS)`

    `(dns1.networkutopia.com, 212.212.212.1, A)`
- create authoritative server locally with IP address `212.212.212.1`
  - type A record for www.networkuptopia.com
  - type MX record for networkutopia.com

DNS security

DDoS attacks

- bombard root servers with traffic
  - not successful to date
  - traffic filtering
  - local DNS servers cache IPs of TLD servers, allowing root server bypass
- bombard TLD servers
  - potentially more dangerous

Spoofing attacks

- intercept DNS queries, returning bogus replies
  - DNS cache poisoning
  - RFC 4033: DNSSEC authentication services

Chapter 3
Transport Layer

Transport layer: overview

Our goal:

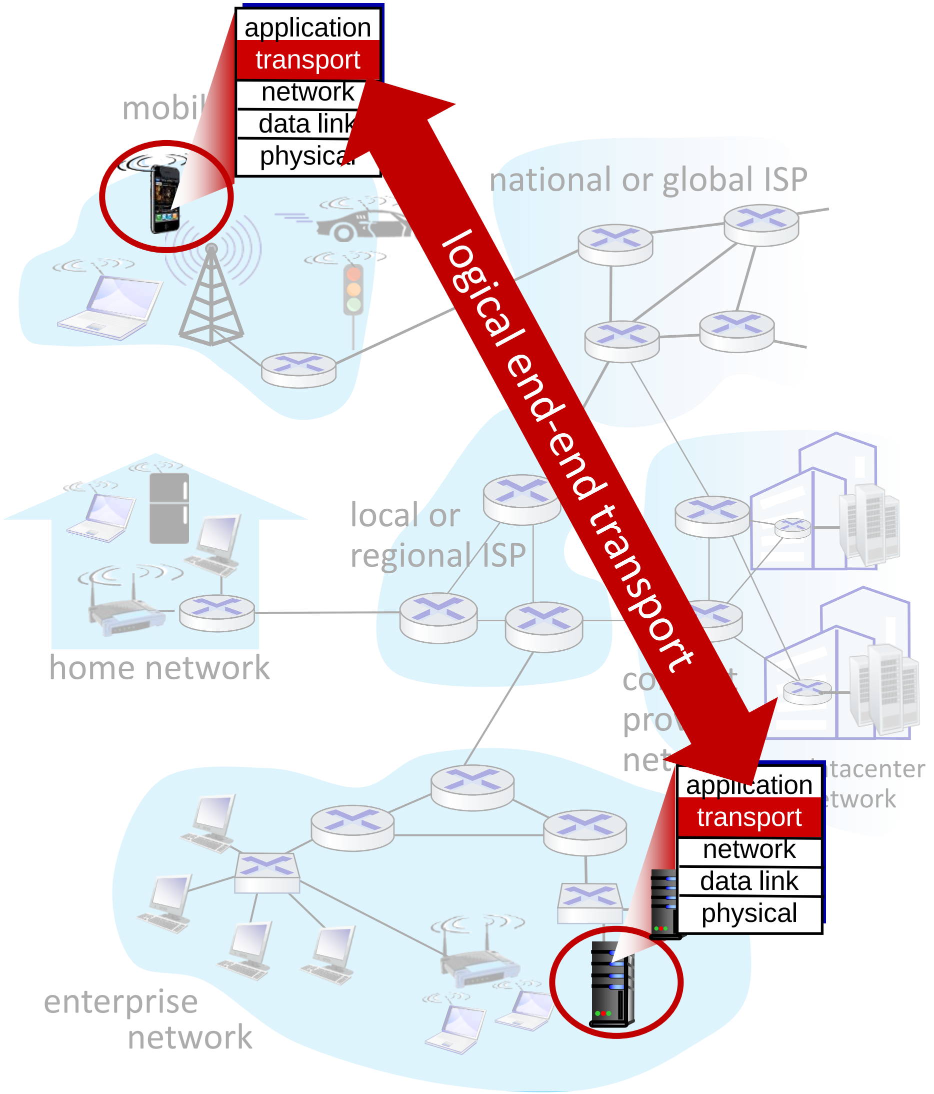
- understand principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about Internet transport layer protocols:
  - UDP: connectionless transport
  - TCP: connection-oriented reliable transport
  - TCP congestion control

Transport layer: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
- Principles of congestion control
- TCP congestion control
- Evolution of transport-layer functionality

Transport services and protocols

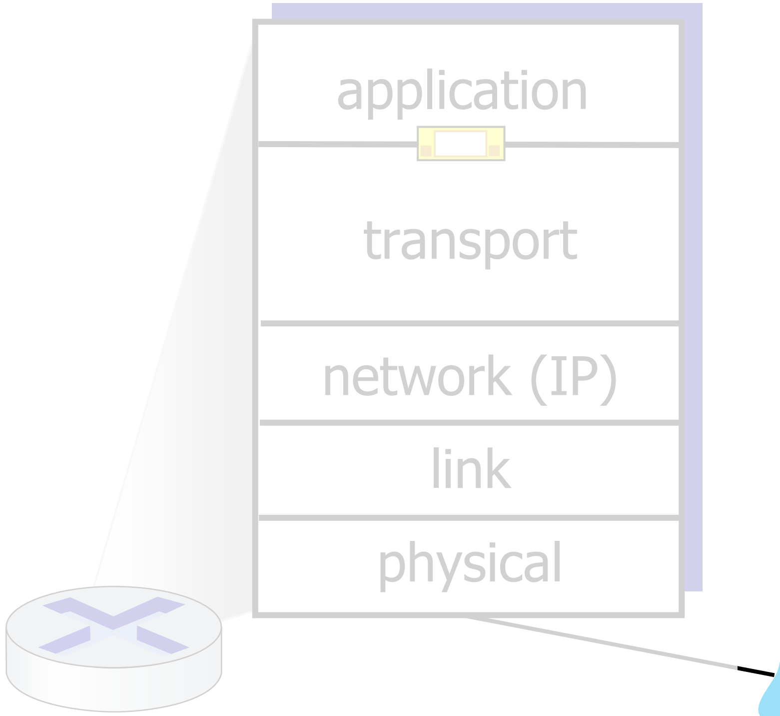
- provide *logical communication* between application processes running on different hosts
- transport protocols actions in end systems:
  - sender: breaks application messages into *segments*, passes to network layer
  - receiver: reassembles segments into messages, passes to application layer
- two transport protocols available to Internet applications
  - TCP, UDP



Transport vs. network layer services and protocols

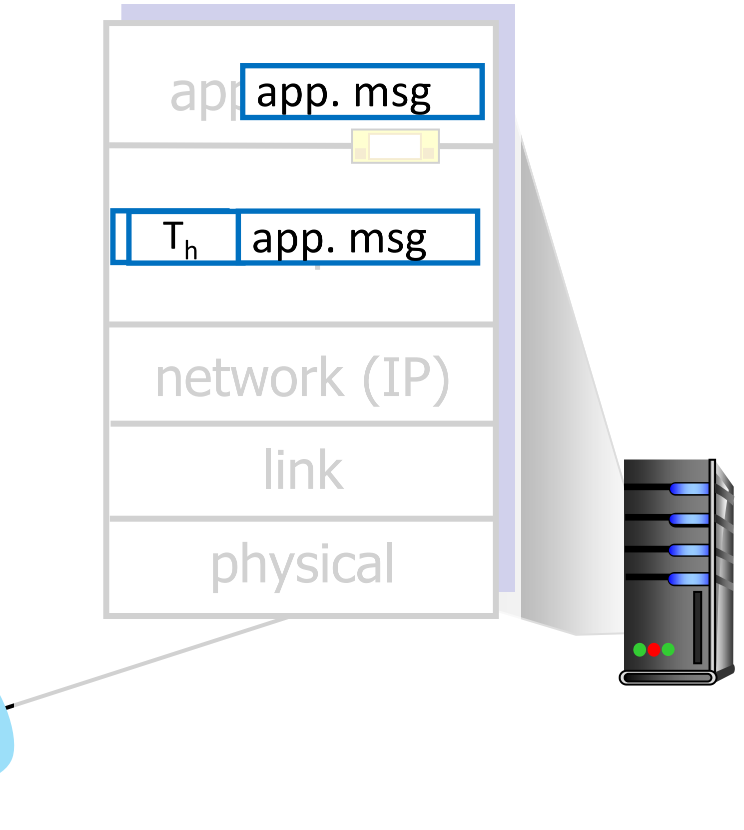
- **transport layer:** communication between *processes*
  - relies on, enhances, network layer services
- **network layer:** communication between *hosts*

Transport Layer Actions

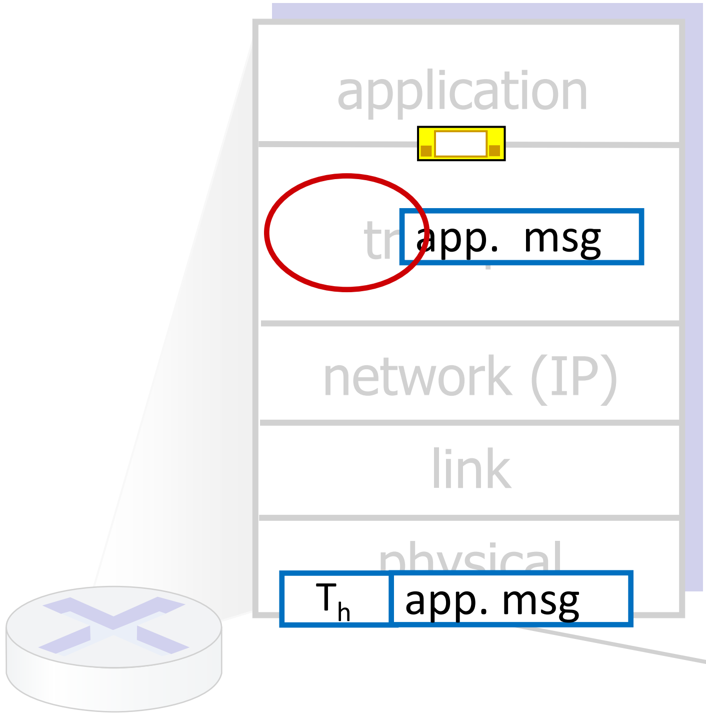


Sender:

- is passed an application-layer message
- determines segment header fields values
- creates segment
- passes segment to IP

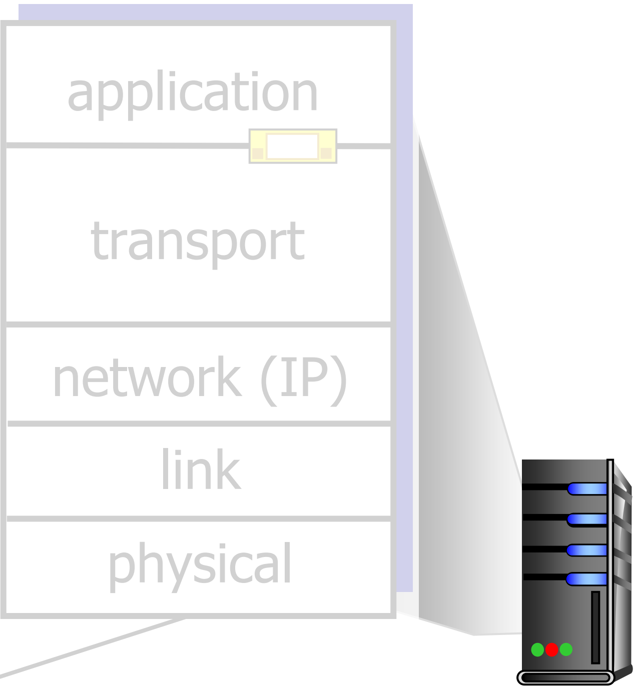


Transport Layer Actions



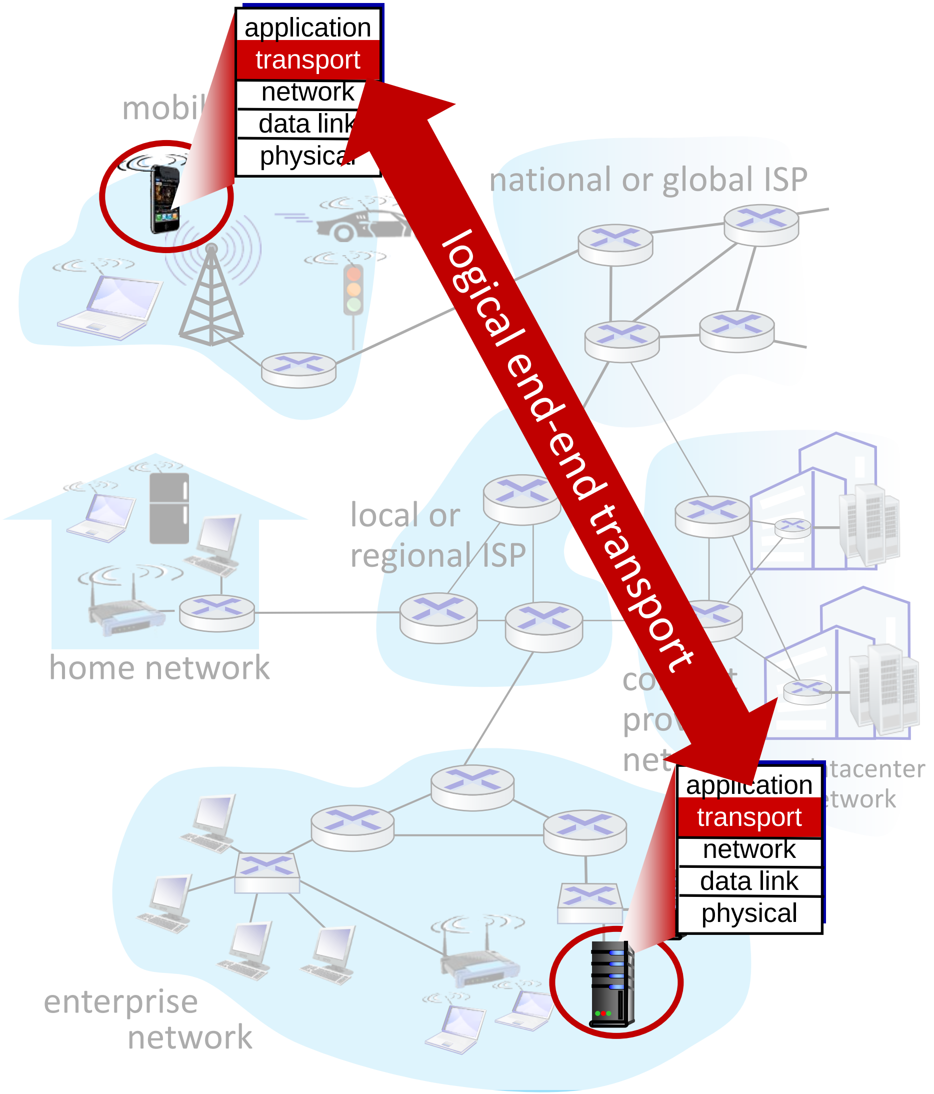
Receiver:

- receives segment from IP
- checks header values
- extracts application-layer message
- demultiplexes message up to application via socket



Two principal Internet transport protocols

- **TCP:** Transmission Control Protocol
  - reliable, in-order delivery
  - congestion control
  - flow control
  - connection setup
- **UDP:** User Datagram Protocol
  - unreliable, unordered delivery
  - no-frills extension of "best-effort" IP
- services *not* available:
  - delay guarantees
  - bandwidth guarantees



Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
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- Evolution of transport-layer functionality

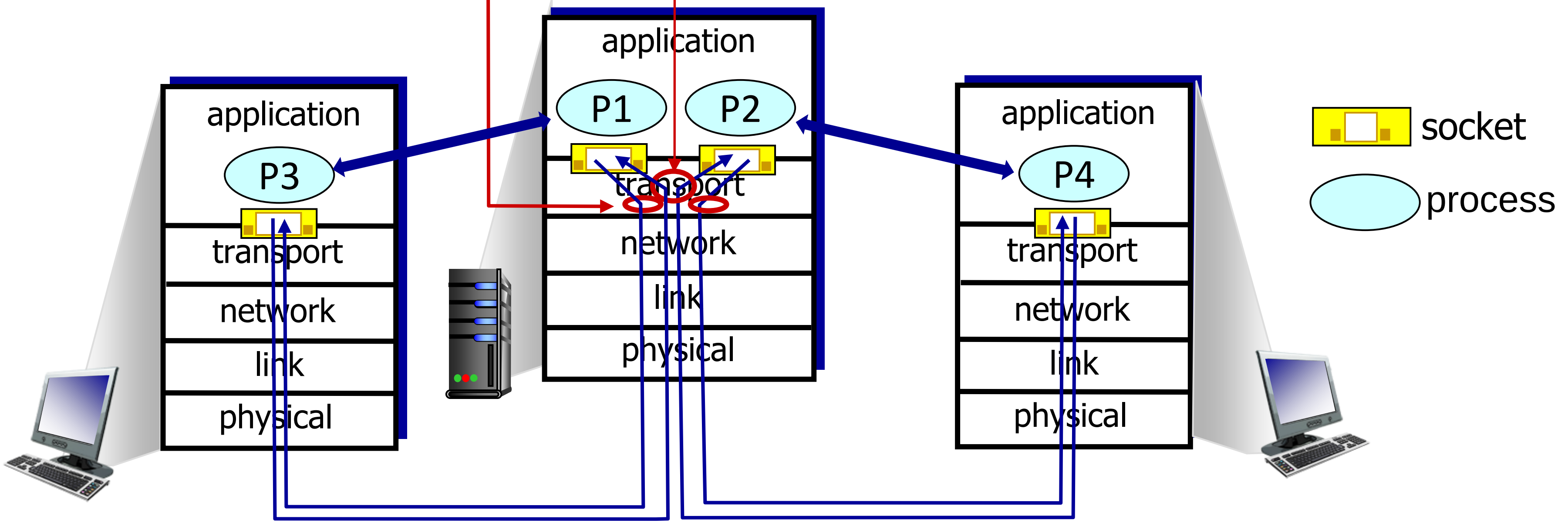
Multiplexing/demultiplexing

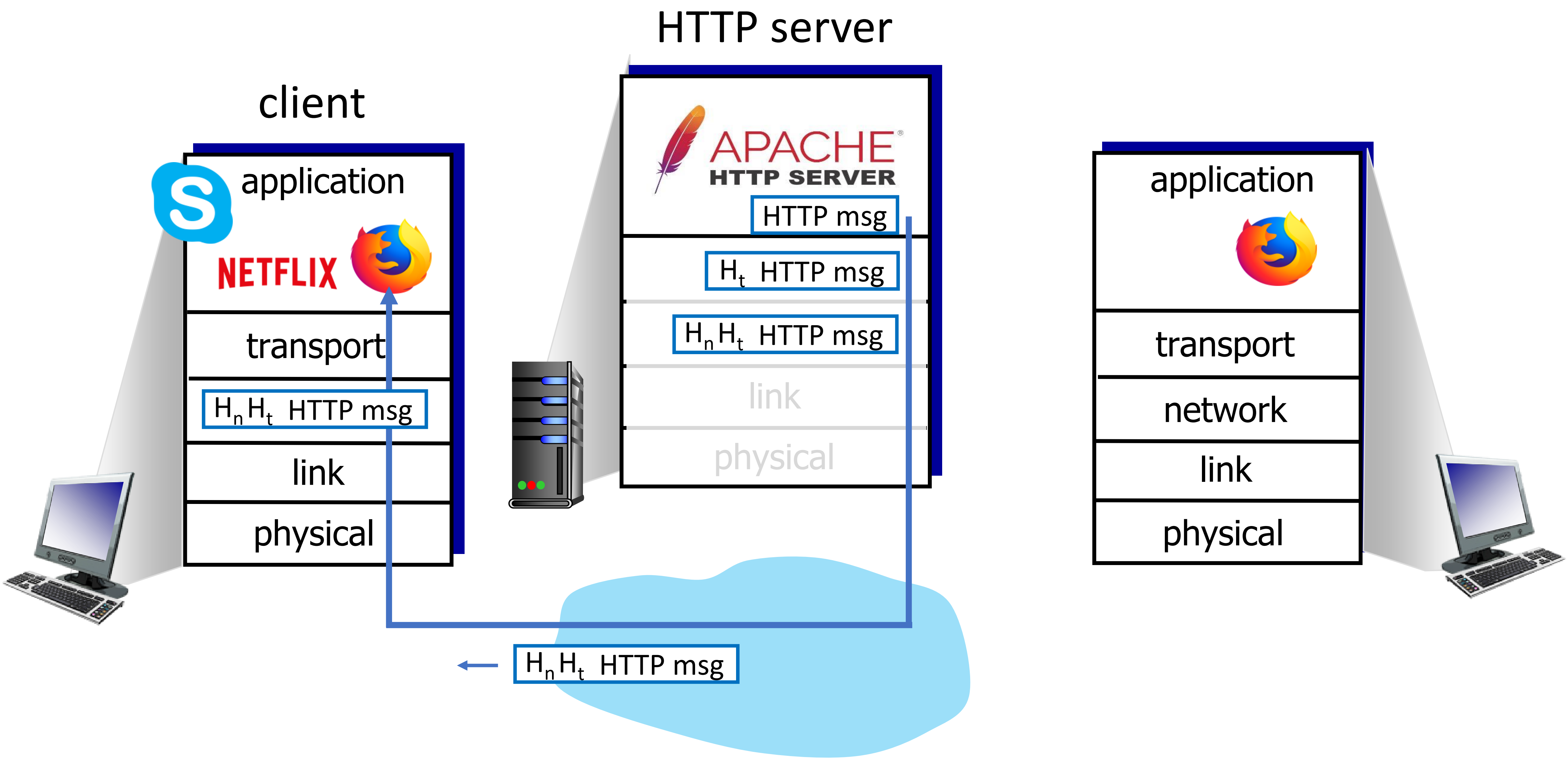
multiplexing as sender:

handle data from multiple sockets, add transport header (later used for demultiplexing)

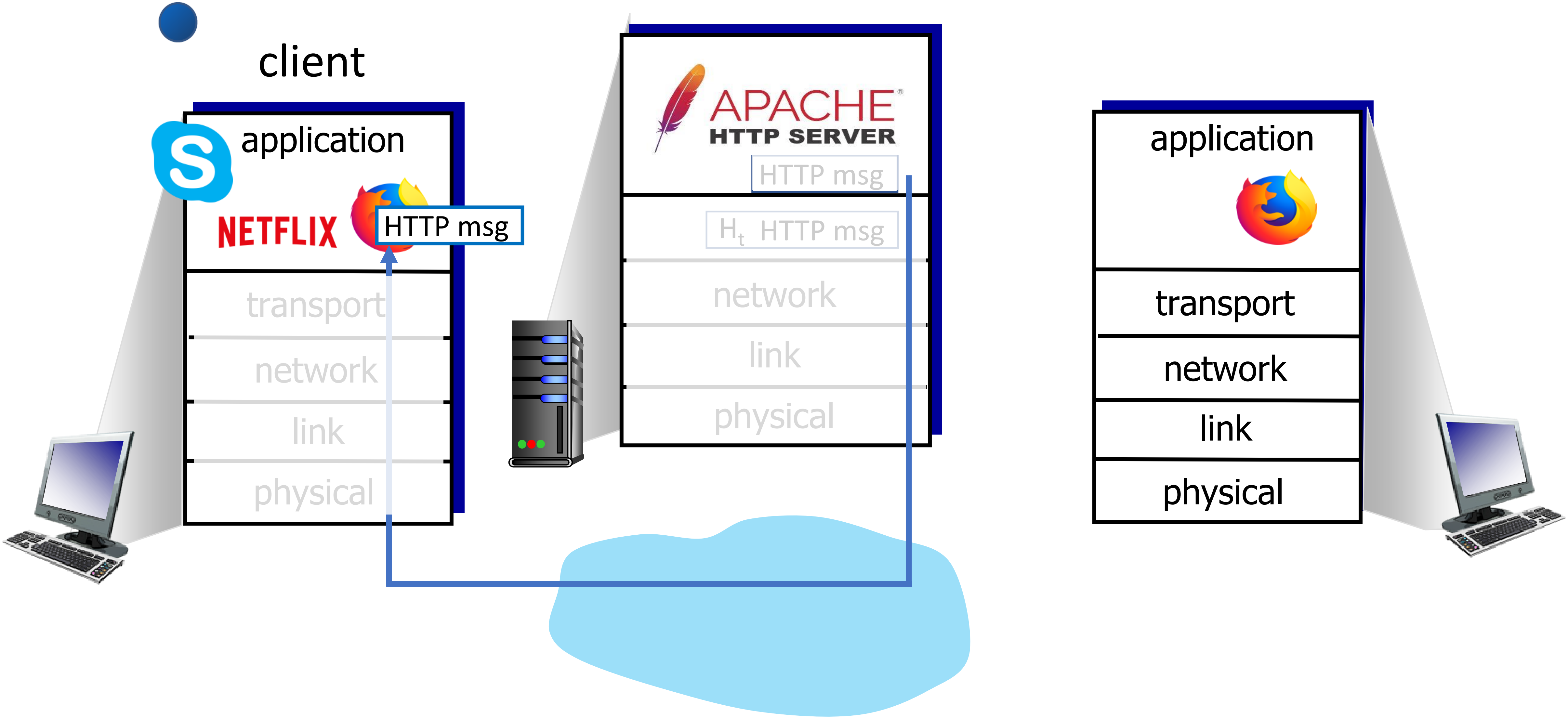
demultiplexing as receiver:

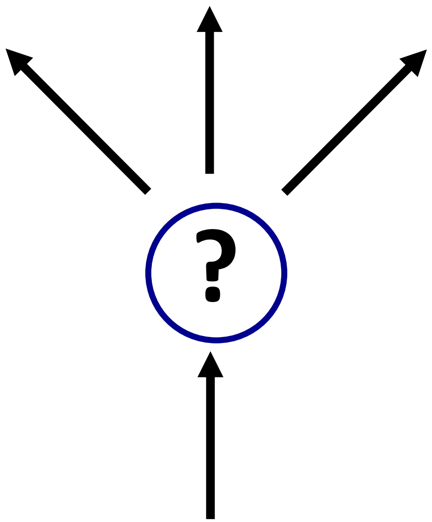
use header info to deliver received segments to correct socket



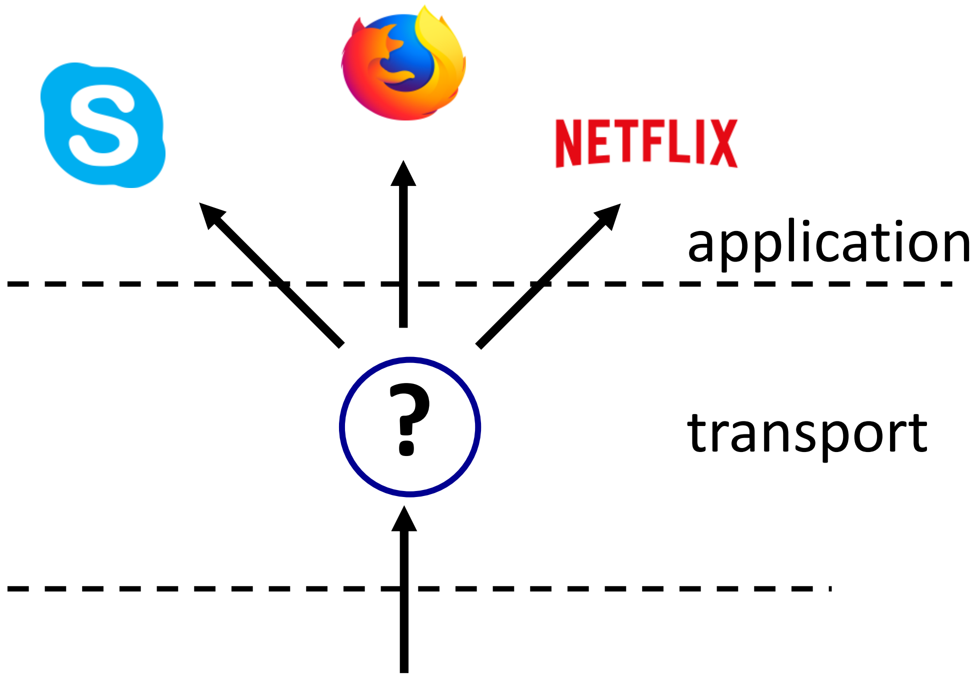

HTTP server
client
application
NETFLIX
transport
$H_n H_t$ HTTP msg
link
physical
APACHE
HTTP SERVER
HTTP msg
H_t HTTP msg
$H_n H_t$ HTTP msg
link
physical
$H_n H_t$ HTTP msg
application
transport
network
link
physical

Q: how did transport layer know to deliver message to Firefox browser process rather then Netflix process or Skype process?

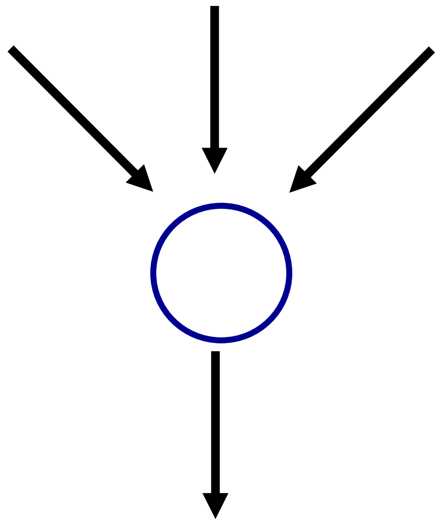




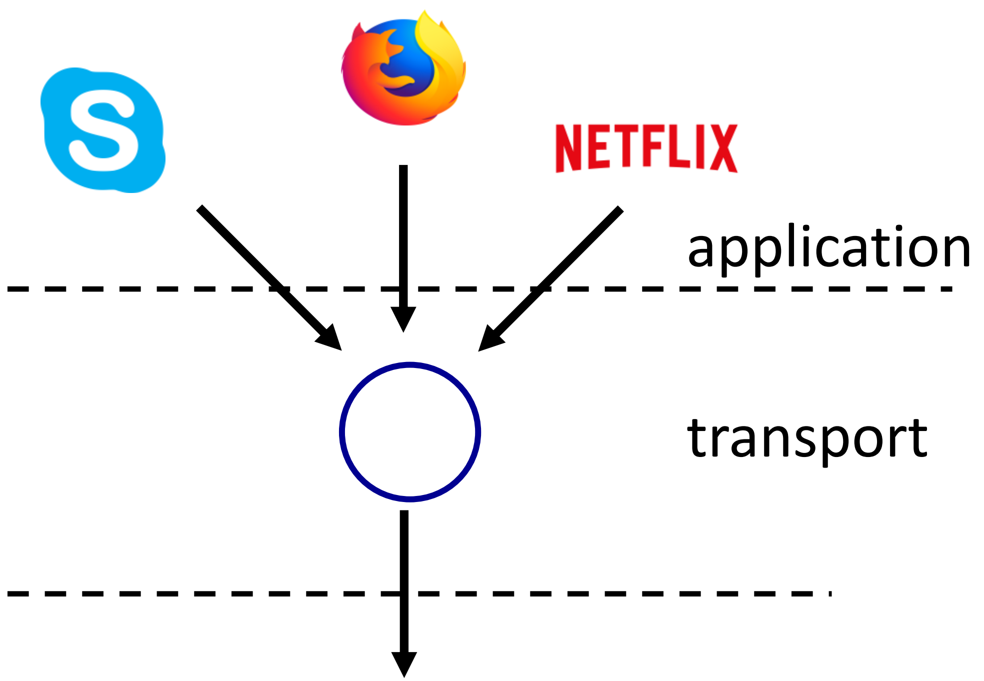
de-multiplexing



de-multiplexing



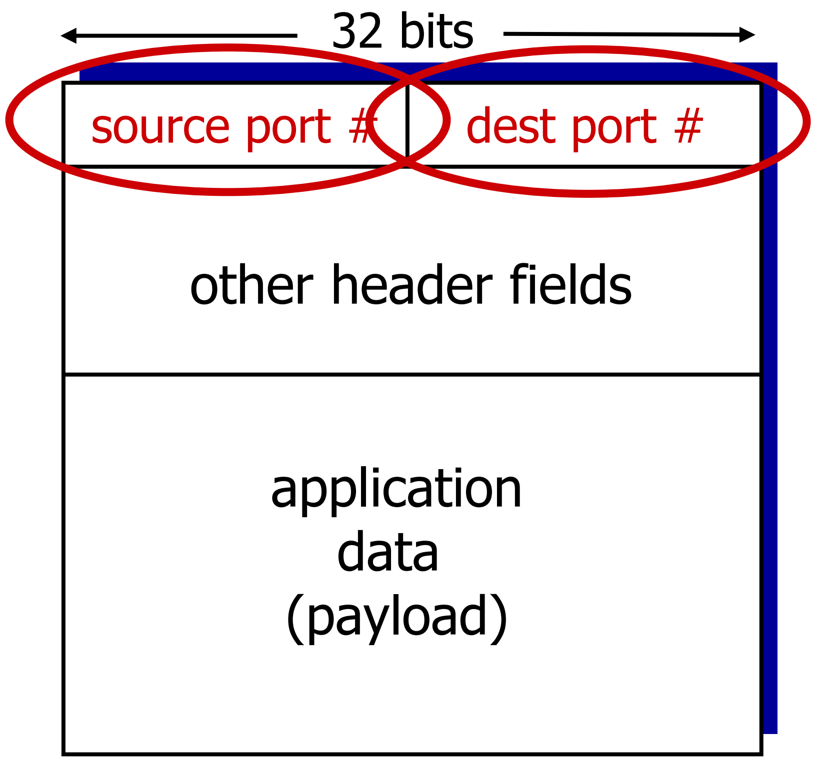
multiplexing



multiplexing

How demultiplexing works

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



TCP/UDP segment format

Connectionless demultiplexing

Recall:

- when creating socket, must specify *host-local* port #:

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

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

when receiving host receives *UDP* segment:

- checks destination port # in segment
- directs UDP segment to socket with that port #

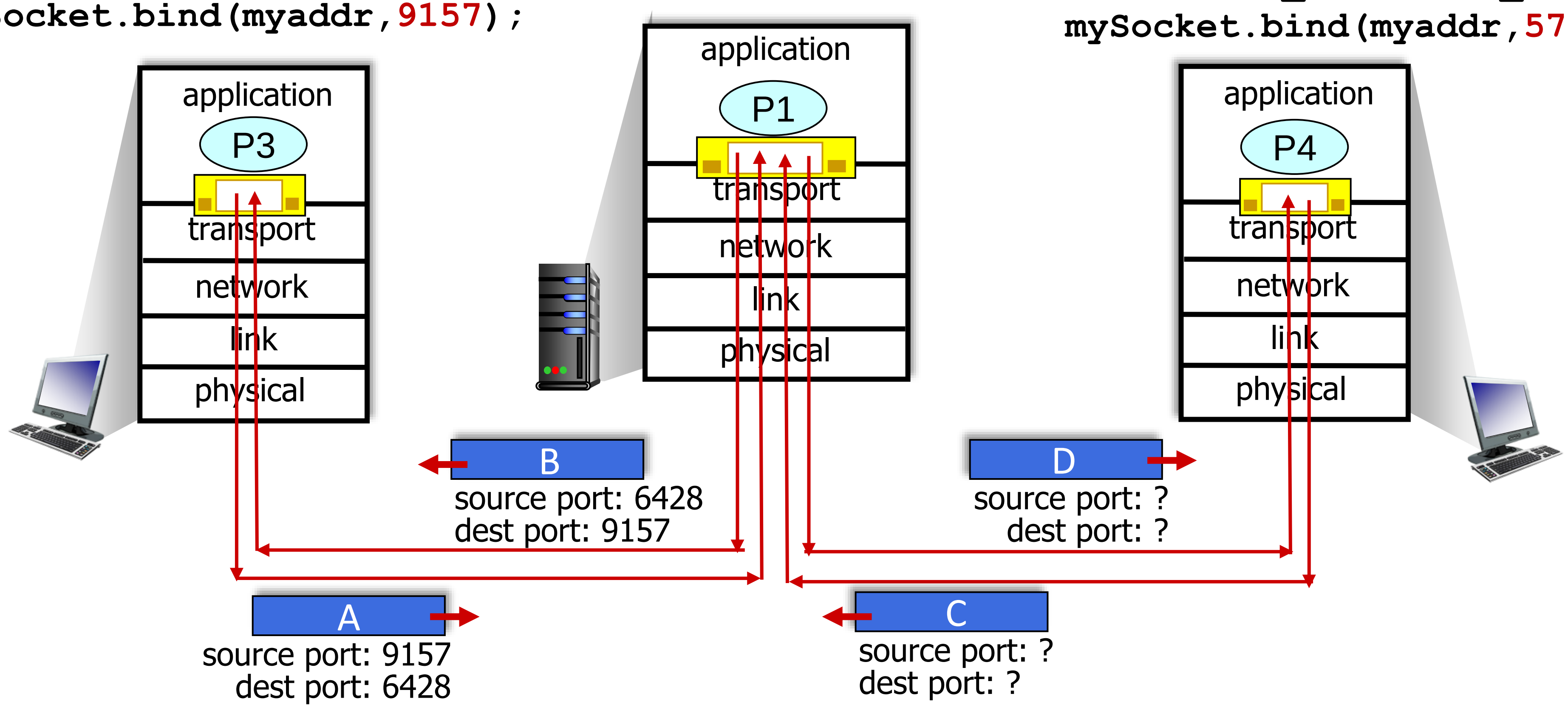
IP/UDP datagrams with *same dest. port #,* but different source IP addresses and/or source port numbers will be directed to *same socket* at receiving host

Connectionless demultiplexing: an example

```
mySocket =
 socket(AF_INET,SOCK_DGRAM)
mySocket.bind(myaddr,6428);
```

```
mySocket =
 socket(AF_INET,SOCK_STREAM)
mySocket.bind(myaddr,9157);
```

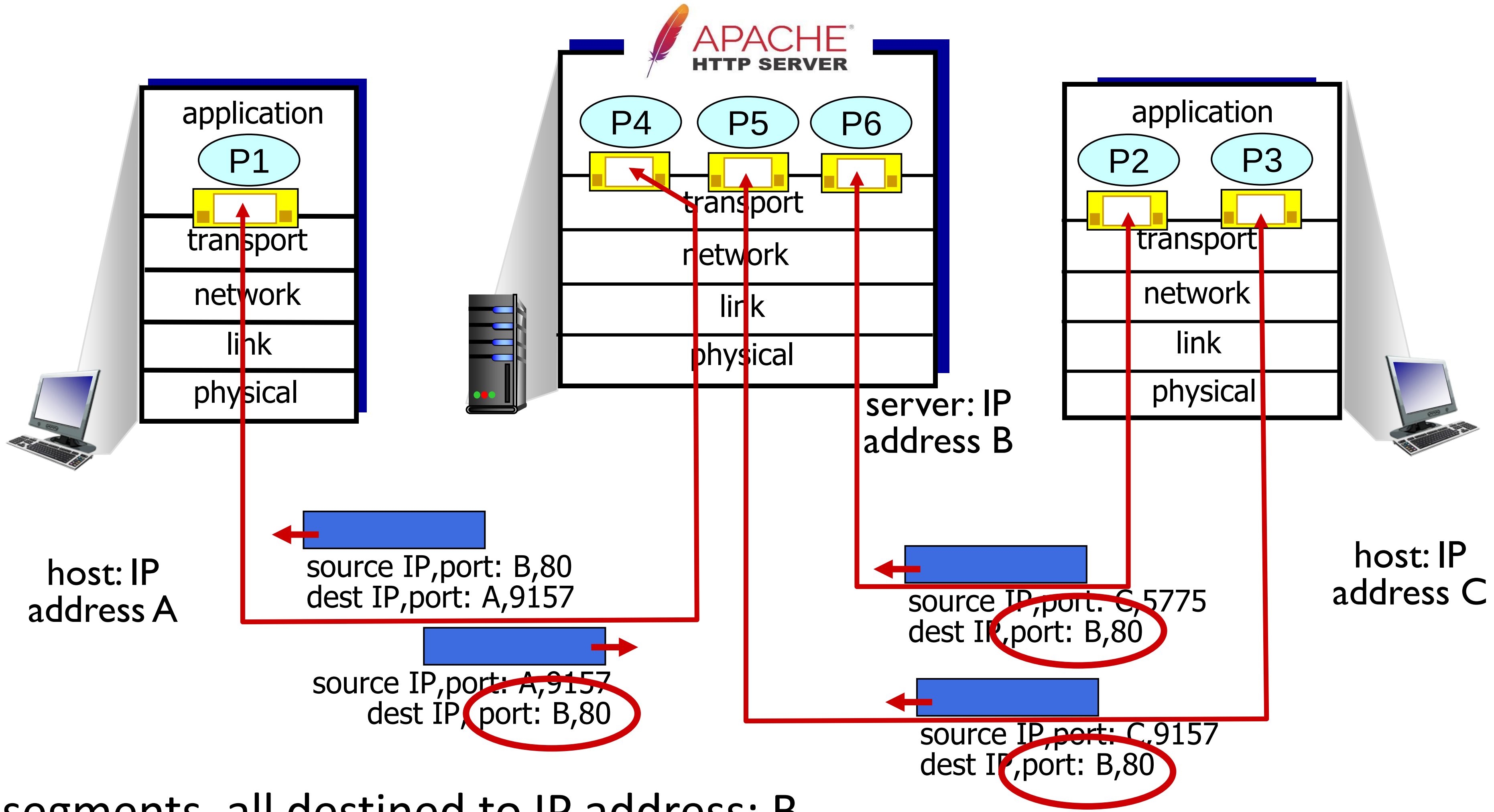
```
mySocket =
 socket(AF_INET,SOCK_STREAM)
mySocket.bind(myaddr,5775);
```



Connection-oriented demultiplexing

- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- demux: receiver uses *all four values (4-tuple)* to direct segment to appropriate socket
- server may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
  - each socket associated with a different connecting client

Connection-oriented demultiplexing: example



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

Summary

- Multiplexing, demultiplexing: based on segment, datagram header field values
- **UDP:** demultiplexing using destination port number (only)
- **TCP:** demultiplexing using 4-tuple: source and destination IP addresses, and port numbers
- Multiplexing/demultiplexing happen at *all* layers

Chapter 3: roadmap

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UDP: User Datagram Protocol

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

Why is there a UDP?

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

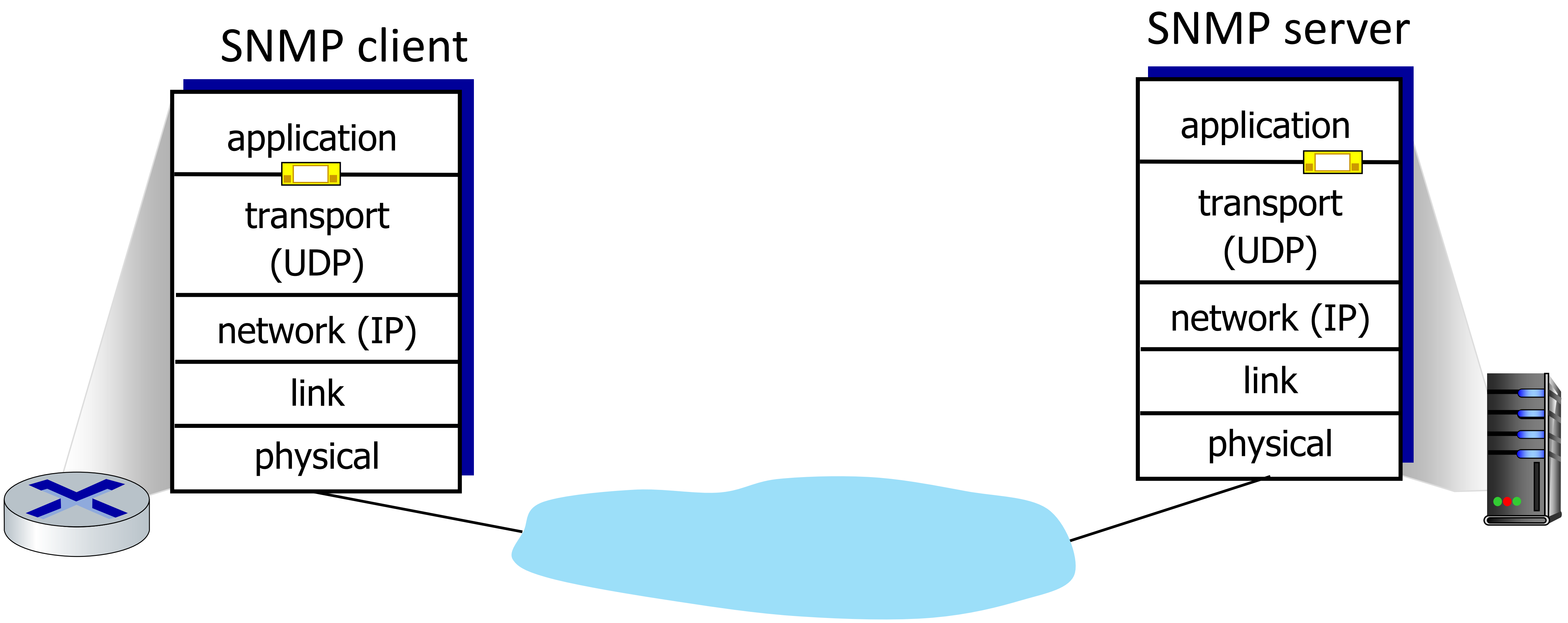
UDP: User Datagram Protocol

- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
  - SNMP
  - HTTP/3
- if reliable transfer needed over UDP (e.g., HTTP/3):
  - add needed reliability at application layer
  - add congestion control at application layer

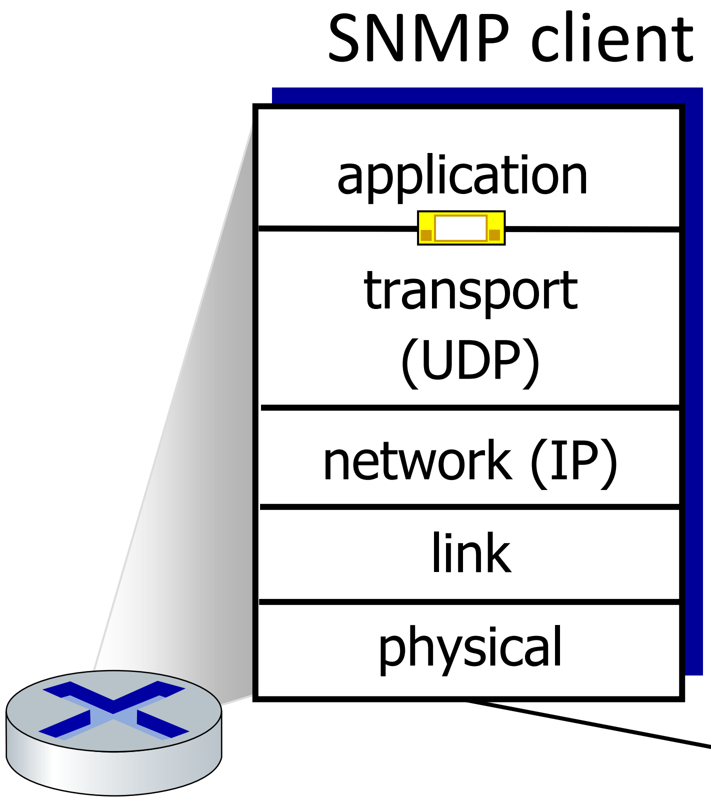
UDP: User Datagram Protocol [RFC 768]

```
                                                      INTERNET STANDARD
RFC  768                                                      J. Postel
                                                                    ISI
                                                         28 August 1980



                         User Datagram Protocol
                         ----------------------

Introduction
------------

This User Datagram  Protocol  (UDP)  is  defined  to  make  available  a
datagram   mode  of  packet-switched   computer   communication  in  the
environment  of  an  interconnected  set  of  computer  networks.   This
protocol  assumes  that the Internet  Protocol  (IP)  [1] is used as the
underlying protocol.

This protocol  provides  a procedure  for application  programs  to send
messages  to other programs  with a minimum  of protocol mechanism.  The
protocol  is transaction oriented, and delivery and duplicate protection
are not guaranteed.  Applications requiring ordered reliable delivery of
streams of data should use the Transmission Control Protocol (TCP) [2].

Format
------


                  0      7 8     15 16    23 24    31
                 +--------+--------+--------+--------+
                 |     Source      |   Destination   |
                 |      Port       |      Port       |
                 +--------+--------+--------+--------+
                 |                 |                 |
                 |     Length      |    Checksum     |
                 +--------+--------+--------+--------+
                 |
                 |          data octets ...
                 +---------------- ...
```

UDP: Transport Layer Actions

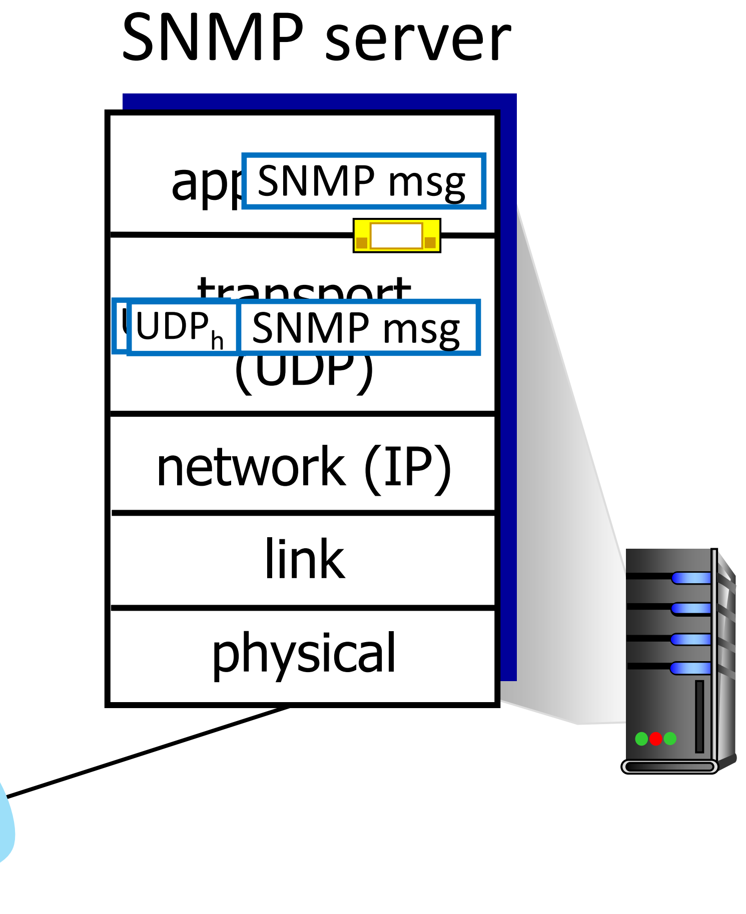


UDP: Transport Layer Actions

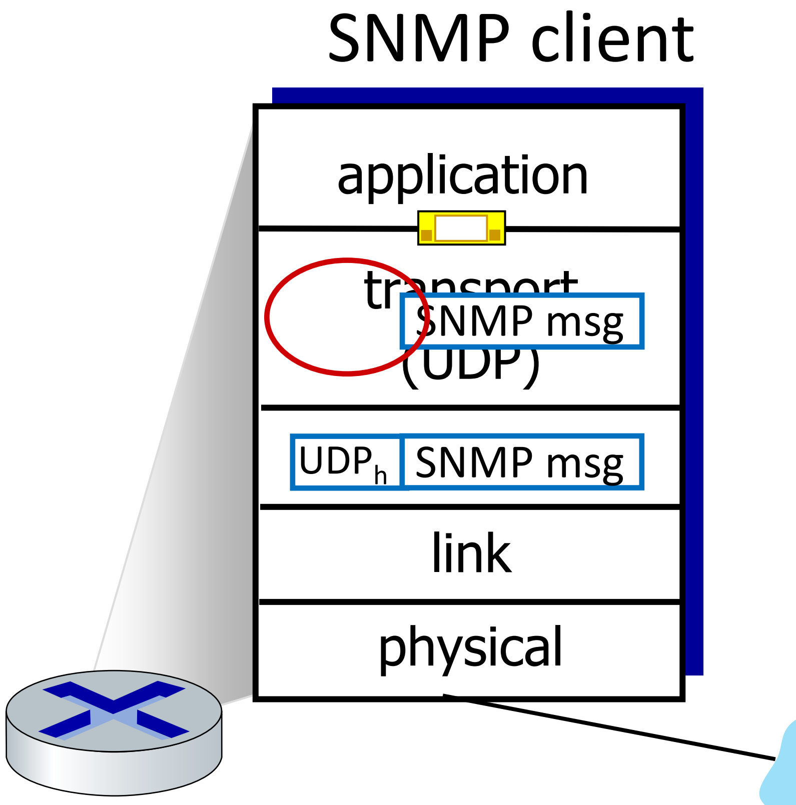


UDP sender actions:

- is passed an application-layer message
- determines UDP segment header fields values
- creates UDP segment
- passes segment to IP

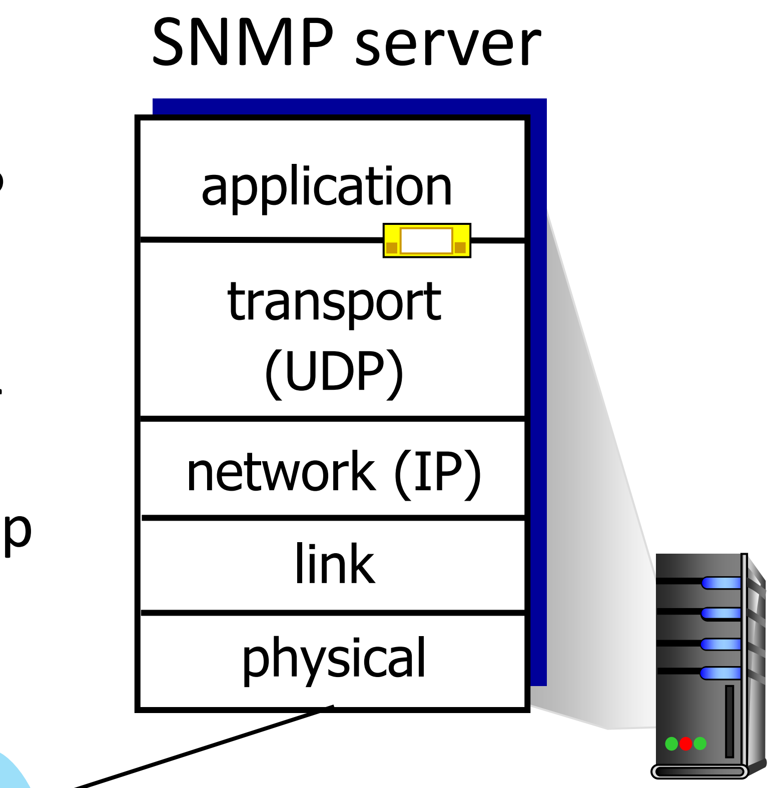


UDP: Transport Layer Actions

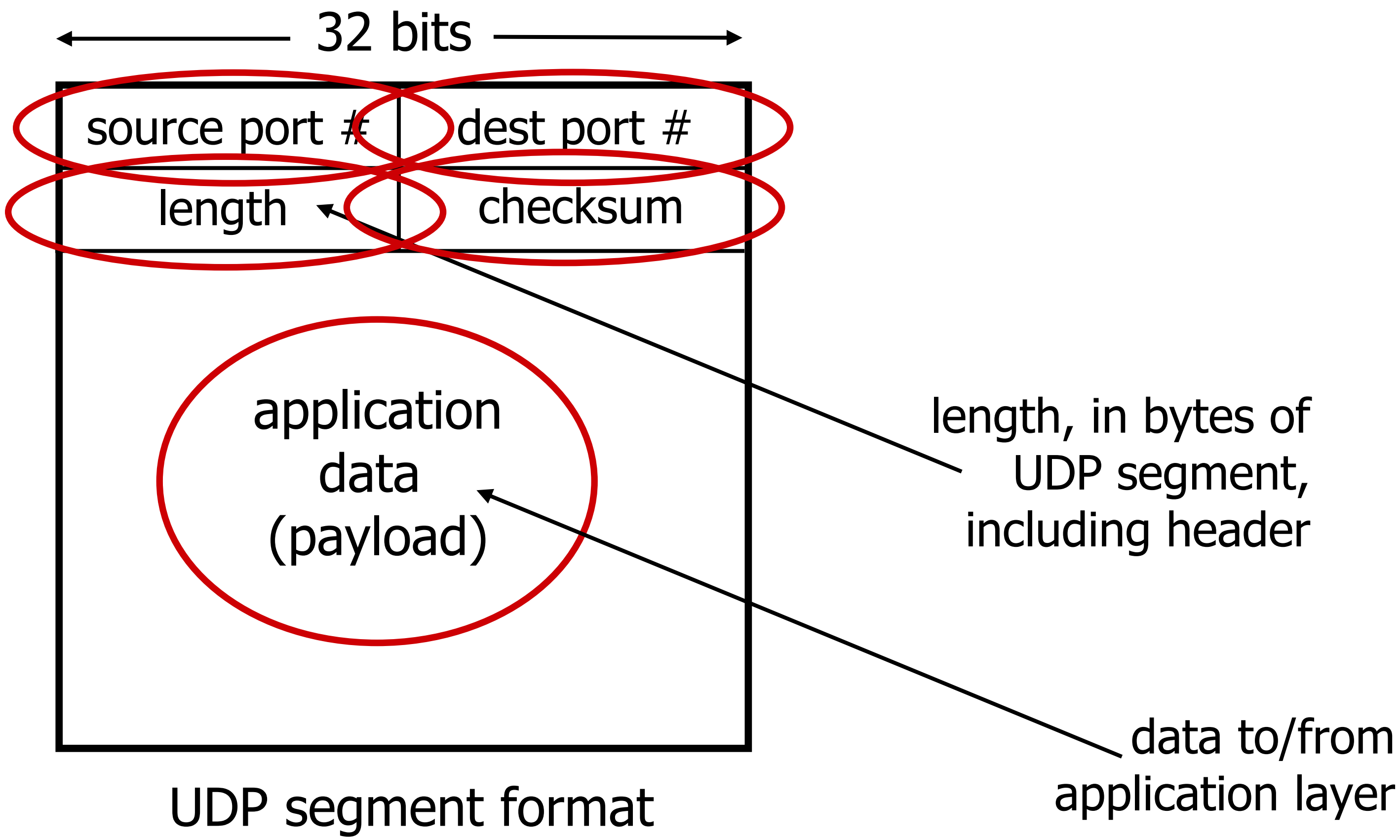


UDP receiver actions:

- receives segment from IP
- checks UDP checksum header value
- extracts application-layer message
- demultiplexes message up to application via socket



UDP segment header



UDP segment format

UDP checksum

Goal: detect errors (*i.e.*, flipped bits) in transmitted segment

	1st number	2nd number	sum
Transmitted:	5	6	11
Received:	4	6	11

receiver-computed checksum ≠ sender-computed checksum (as received)

Internet checksum

Goal: detect errors (*i.e.,* flipped bits) in transmitted segment

sender:

- treat contents of UDP segment (including UDP header fields and IP addresses) as sequence of 16-bit integers
- checksum: addition (one's complement sum) of segment content
- checksum value put into UDP checksum field

receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - not equal - error detected
  - equal - no error detected. *But maybe errors nonetheless?* More later ….

Internet checksum: an example

example: add two 16-bit integers

		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

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

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

Internet checksum: weak protection!

example: add two 16-bit integers

		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

0 1

1 0

Even though numbers have changed (bit flips), *no* change in checksum!

Summary: UDP

- "no frills" protocol:
  - segments may be lost, delivered out of order
  - best effort service: "send and hope for the best"
- UDP has its plusses:
  - no setup/handshaking needed (no RTT incurred)
  - can function when network service is compromised
  - helps with reliability (checksum)
- build additional functionality on top of UDP in application layer (e.g., HTTP/3)

Chapter 3: roadmap

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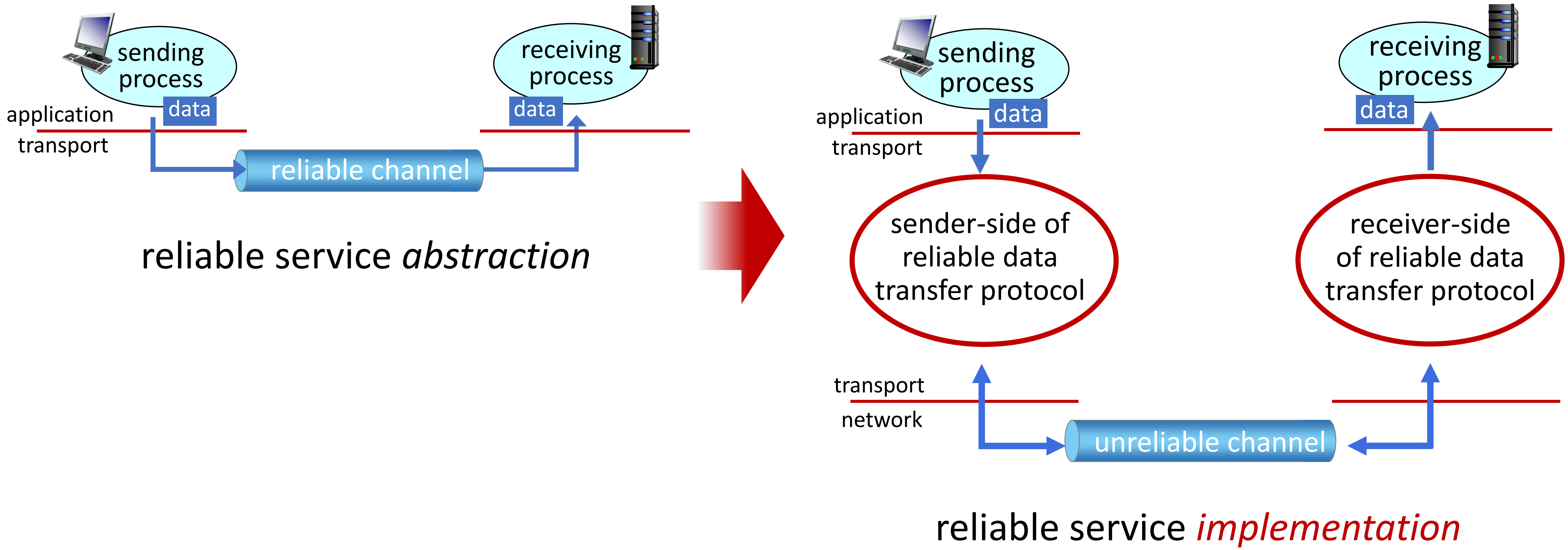
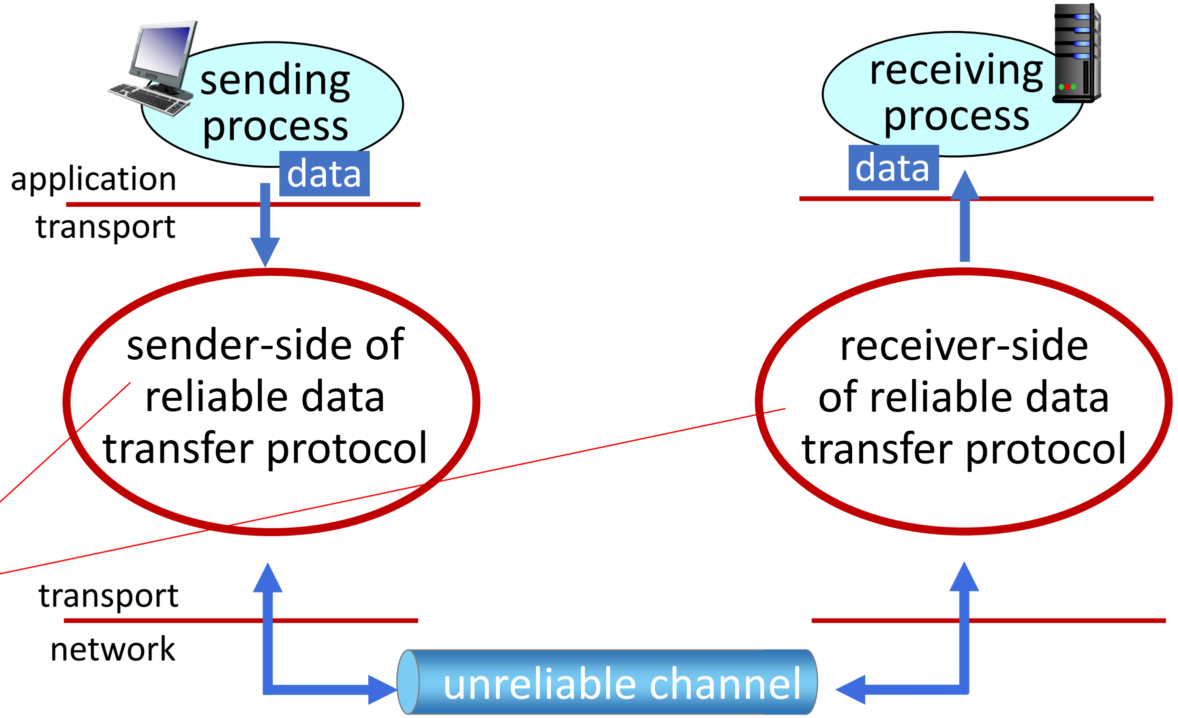
Principles of reliable data transfer

reliable service *abstraction*

Principles of reliable data transfer



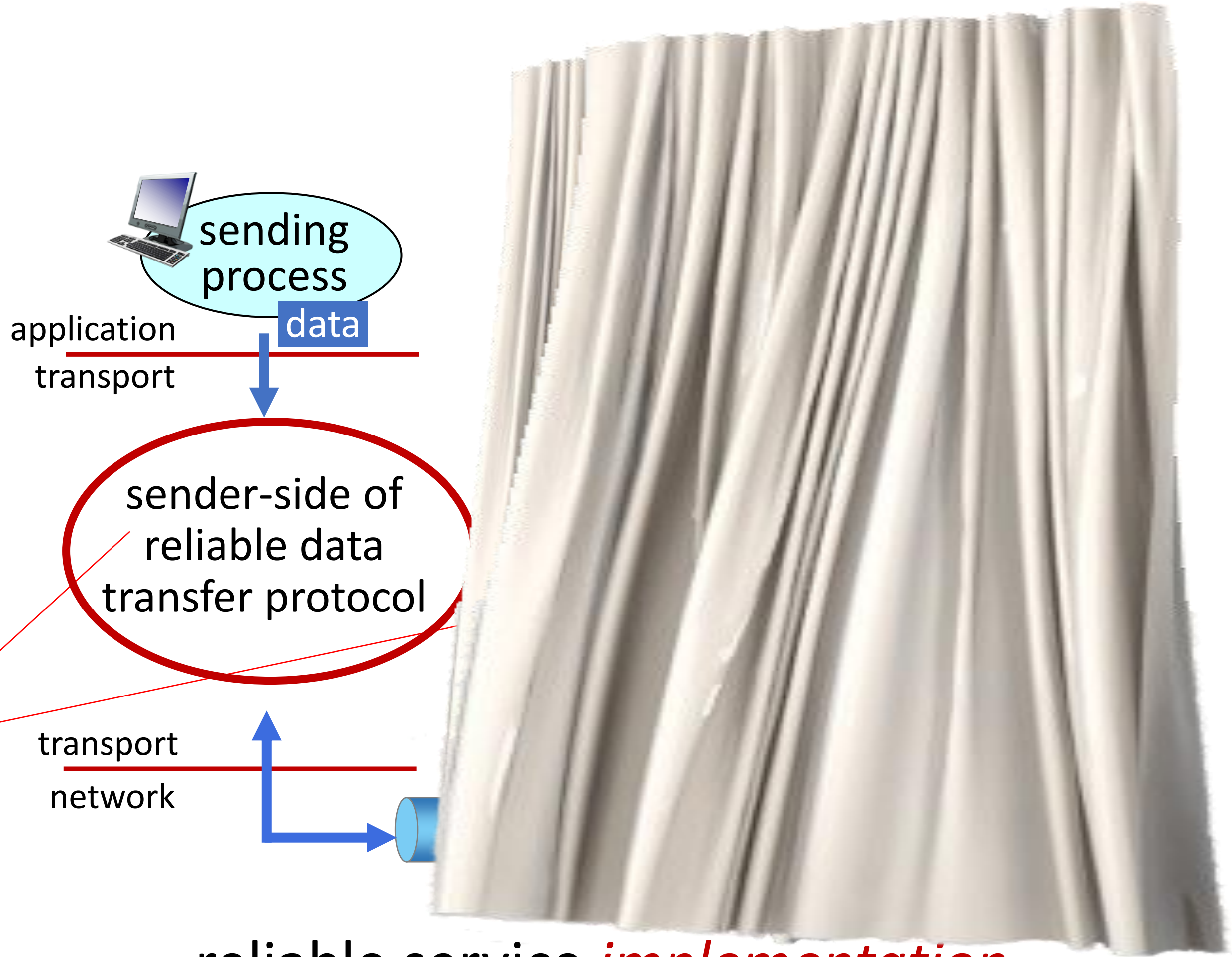
Principles of reliable data transfer



Complexity of reliable data transfer protocol will depend (strongly) on characteristics of unreliable channel (lose, corrupt, reorder data?)

reliable service *implementation*

Principles of reliable data transfer

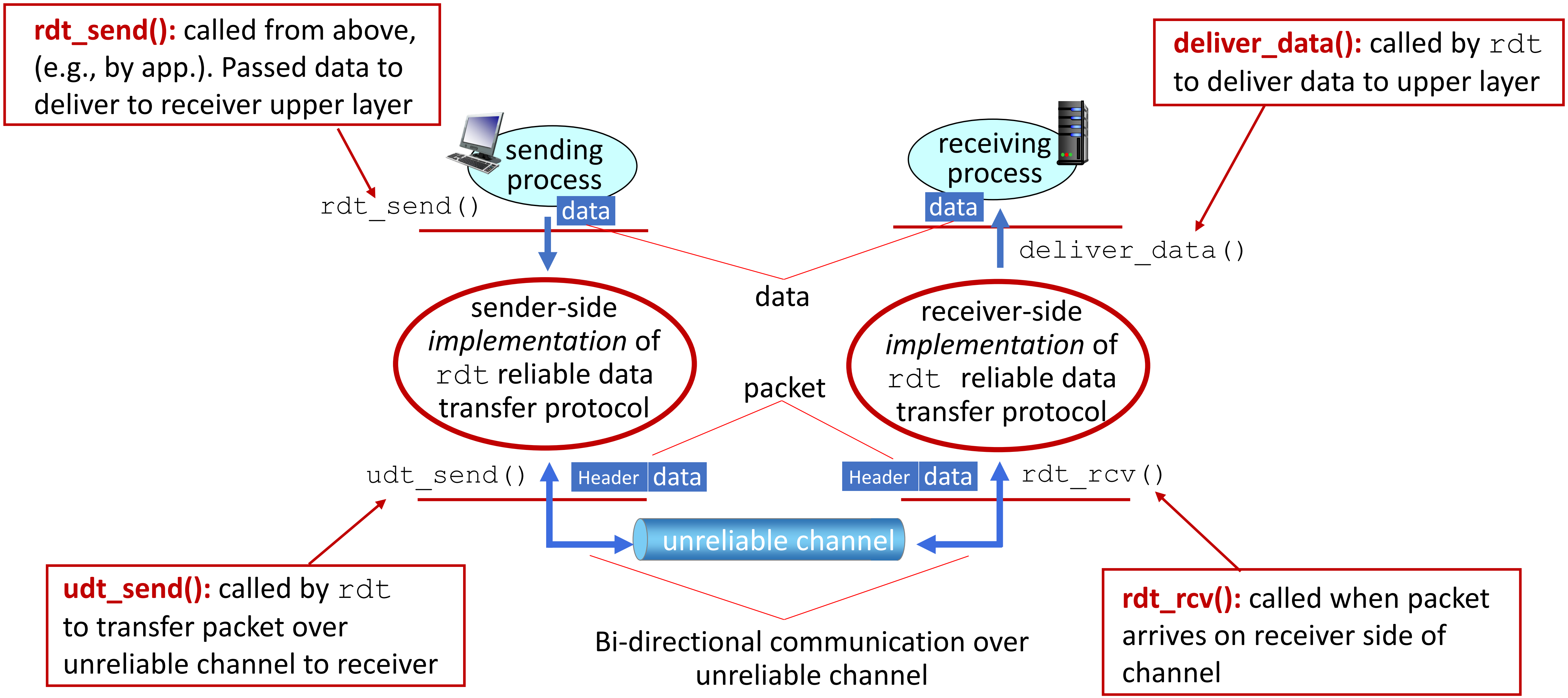


reliable service *implementation*

Sender, receiver do *not* know the "state" of each other, e.g., was a message received?

- unless communicated via a message

Reliable data transfer protocol (rdt): interfaces

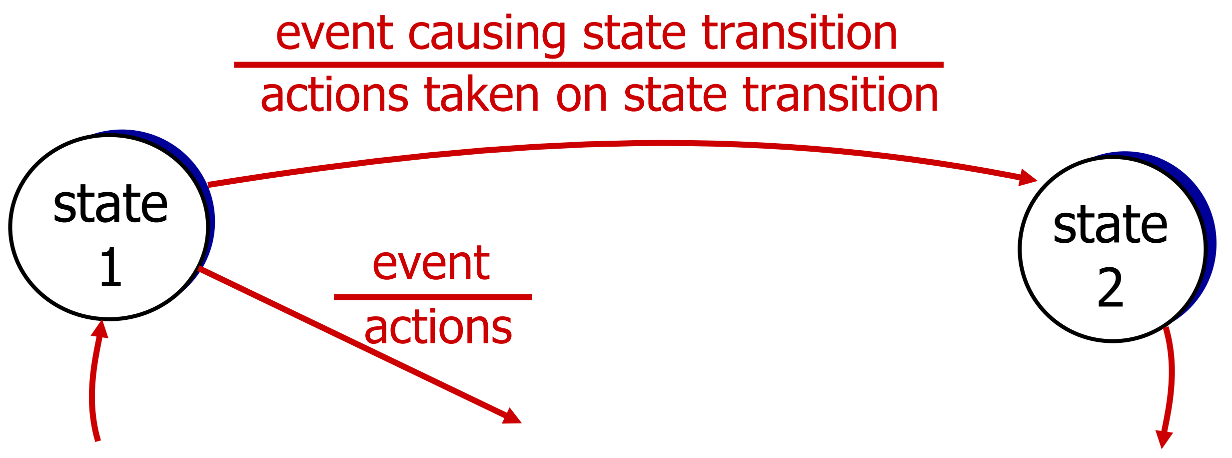


Reliable data transfer: getting started

We will:

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

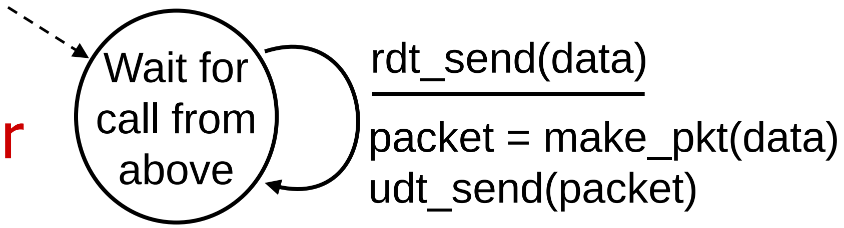
state: when in this "state" next state uniquely determined by next event



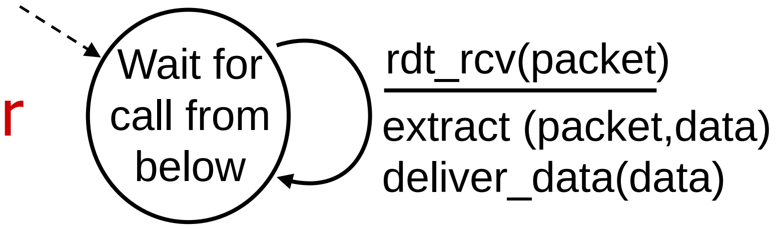
rdt1.0: reliable transfer over a reliable channel

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

sender



receiver



rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - checksum (e.g., Internet checksum) to detect bit errors
- *the* question: how to recover from errors?

How do humans recover from "errors" during conversation?

rdt2.0: channel with bit errors

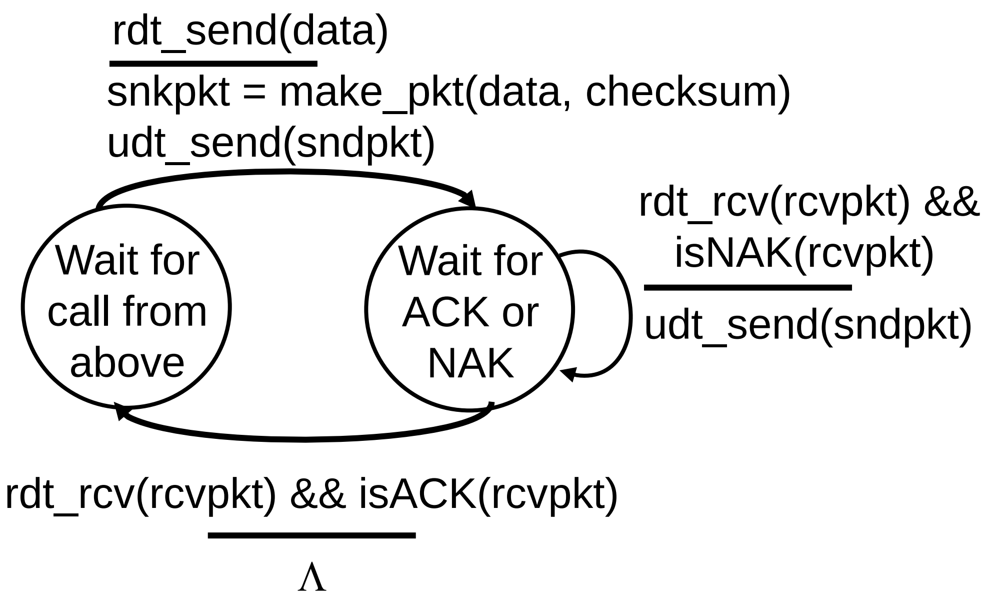
- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors?
  - *acknowledgements (ACKs):* receiver explicitly tells sender that pkt received OK
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors
  - sender *retransmits* pkt on receipt of NAK

stop and wait

sender sends one packet, then waits for receiver response

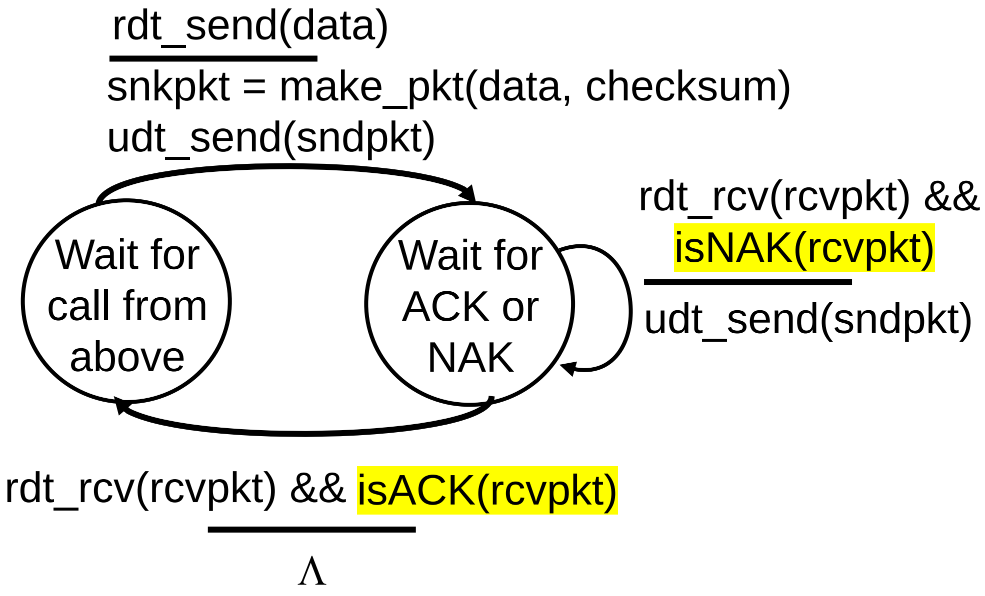
rdt2.0: FSM specifications

sender



rdt2.0: FSM specification

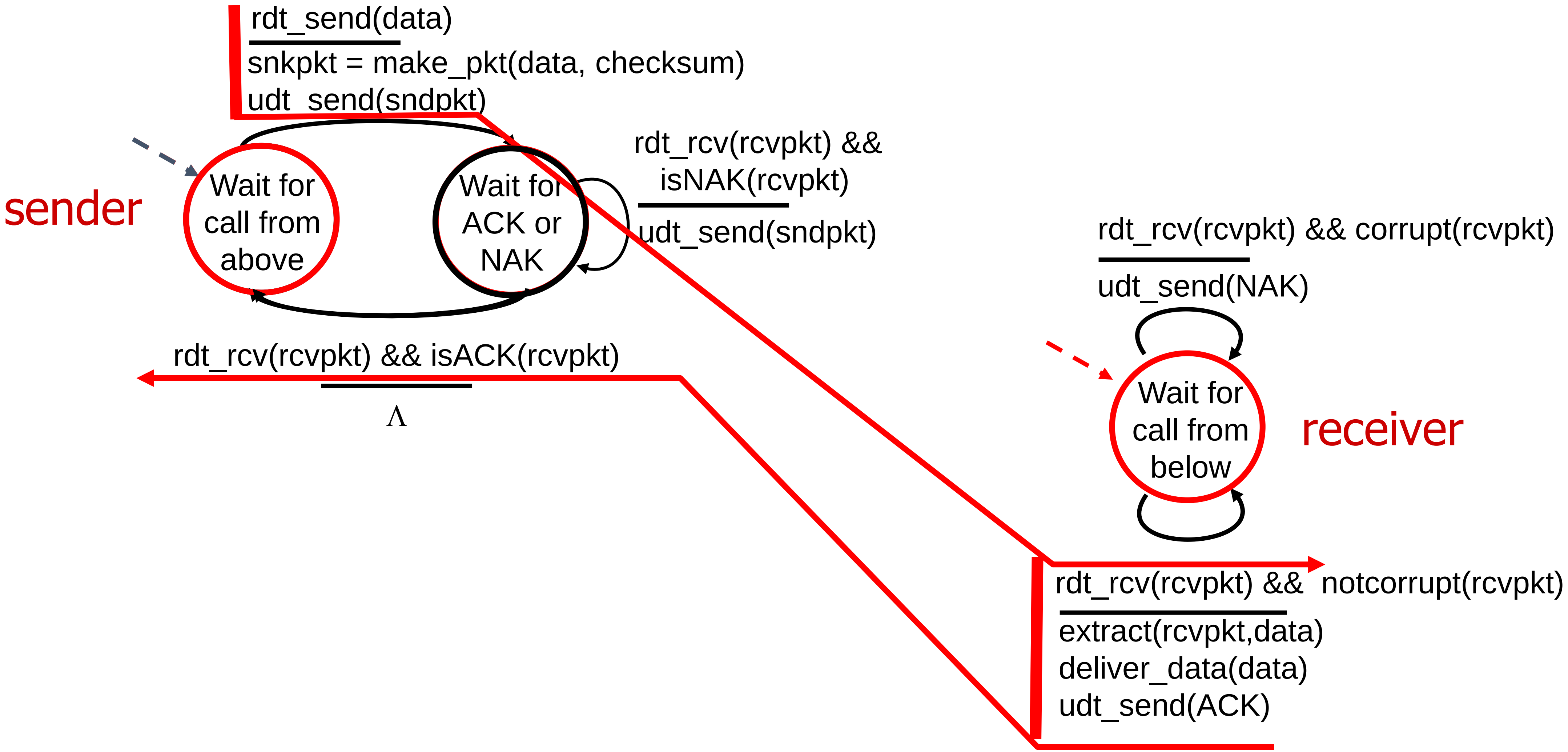
sender



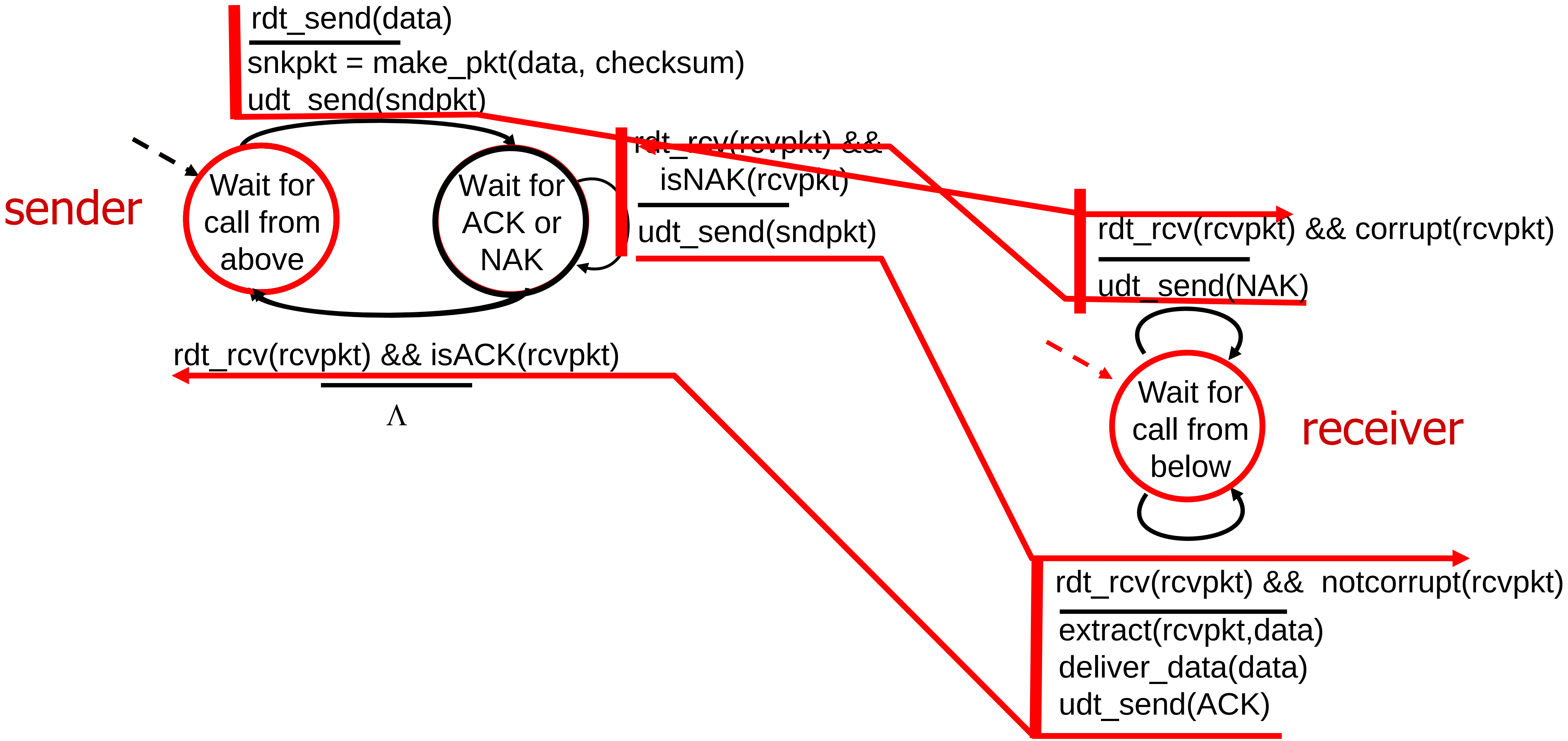
Note: “state” of receiver (did the receiver get my message correctly?) isn’t known to sender unless somehow communicated from receiver to sender

- that’s why we need a protocol!

rdt2.0: operation with no errors



rdt2.0: corrupted packet scenario



rdt2.0 has a fatal flaw!

what happens if ACK/NAK corrupted?

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

handling duplicates:

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

stop and wait

sender sends one packet, then waits for receiver response

rdt3.0: channels with errors *and* loss

New channel assumption: underlying channel can also *lose* packets (data, ACKs)

- checksum, sequence #s, ACKs, retransmissions will be of help … but not quite enough

Q: How do *humans* handle lost sender-to-receiver words in conversation?

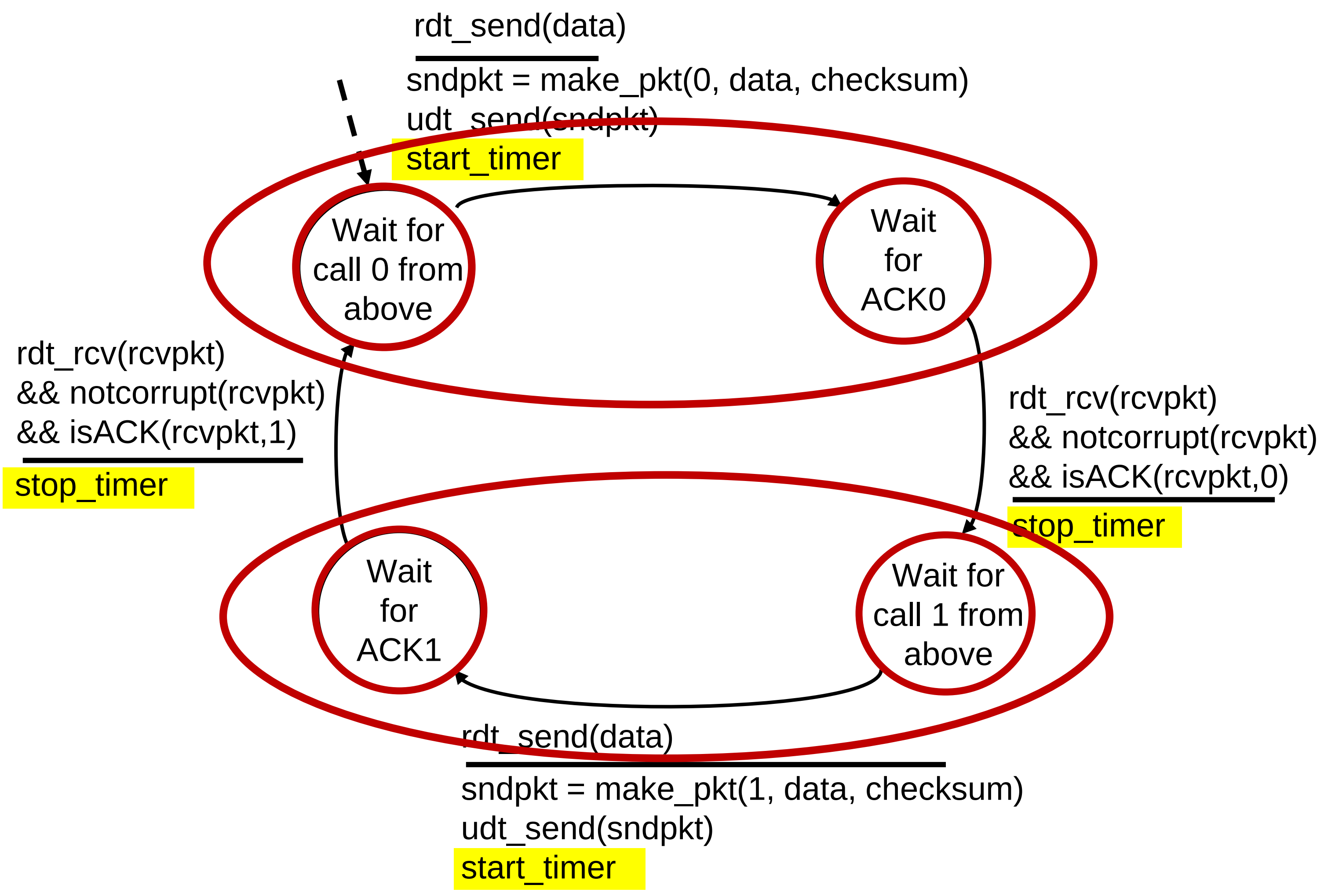
rdt3.0: channels with errors *and* loss

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

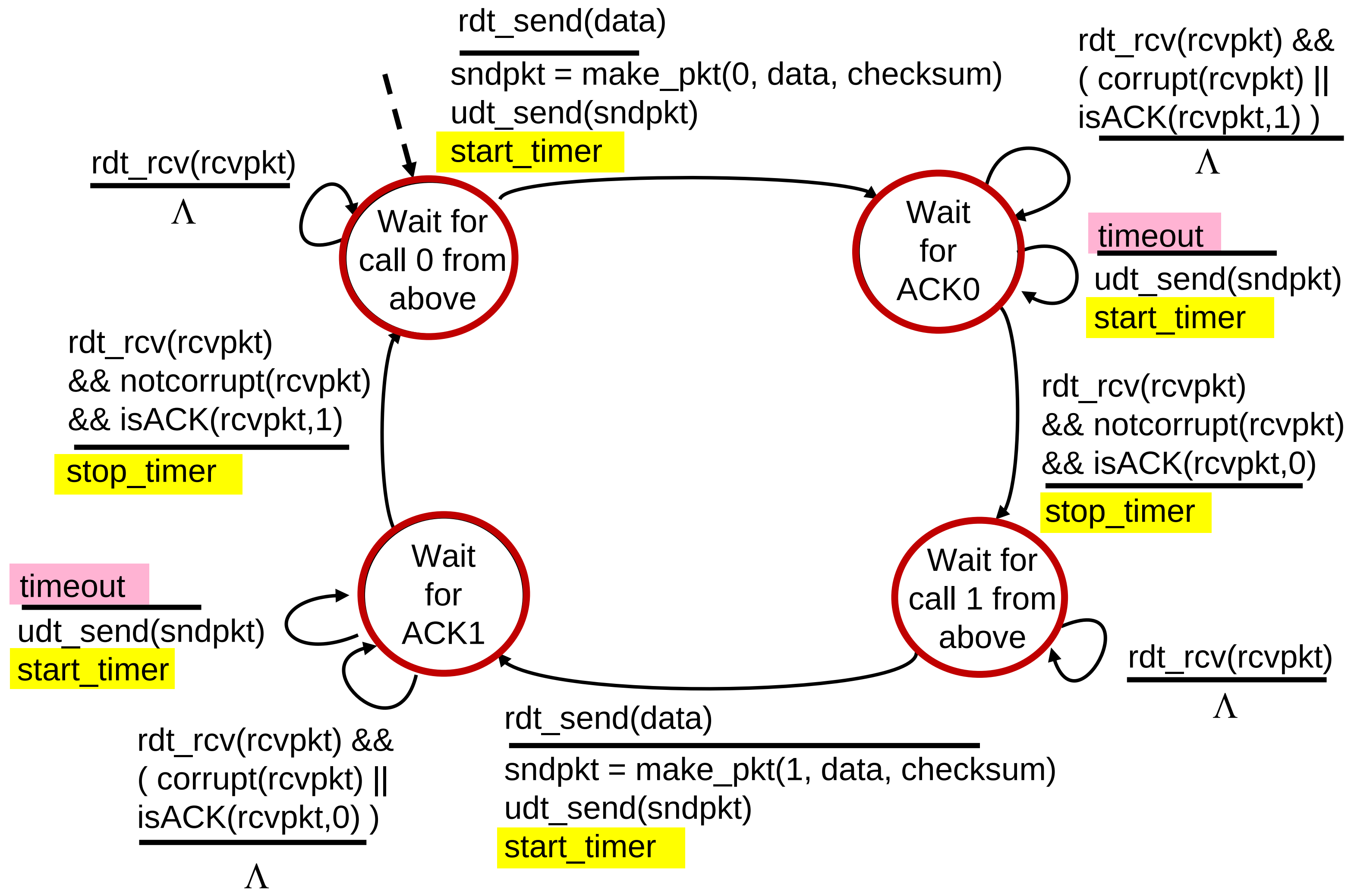
- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq #s already handles this!
  - receiver must specify seq # of packet being ACKed
- use countdown timer to interrupt after "reasonable" amount of time

timeout

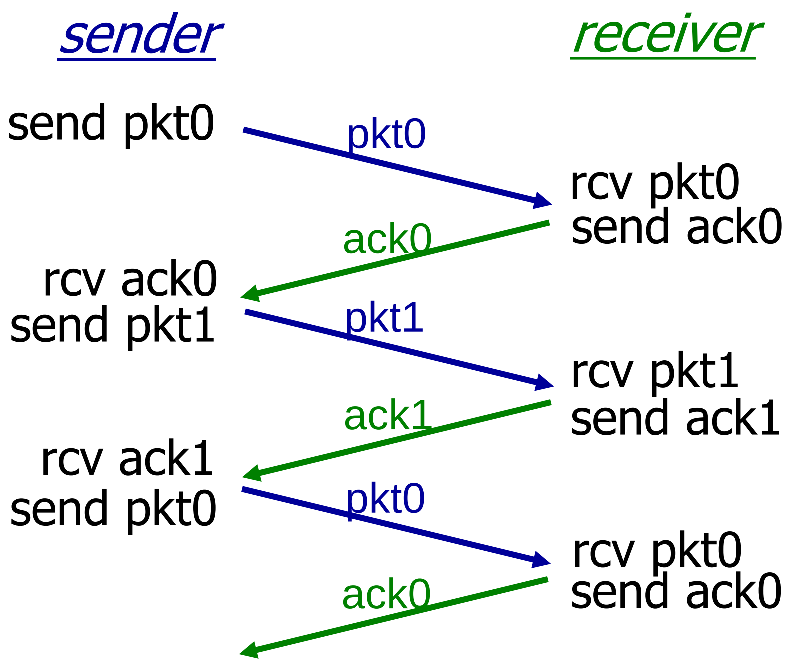
rdt3.0 sender



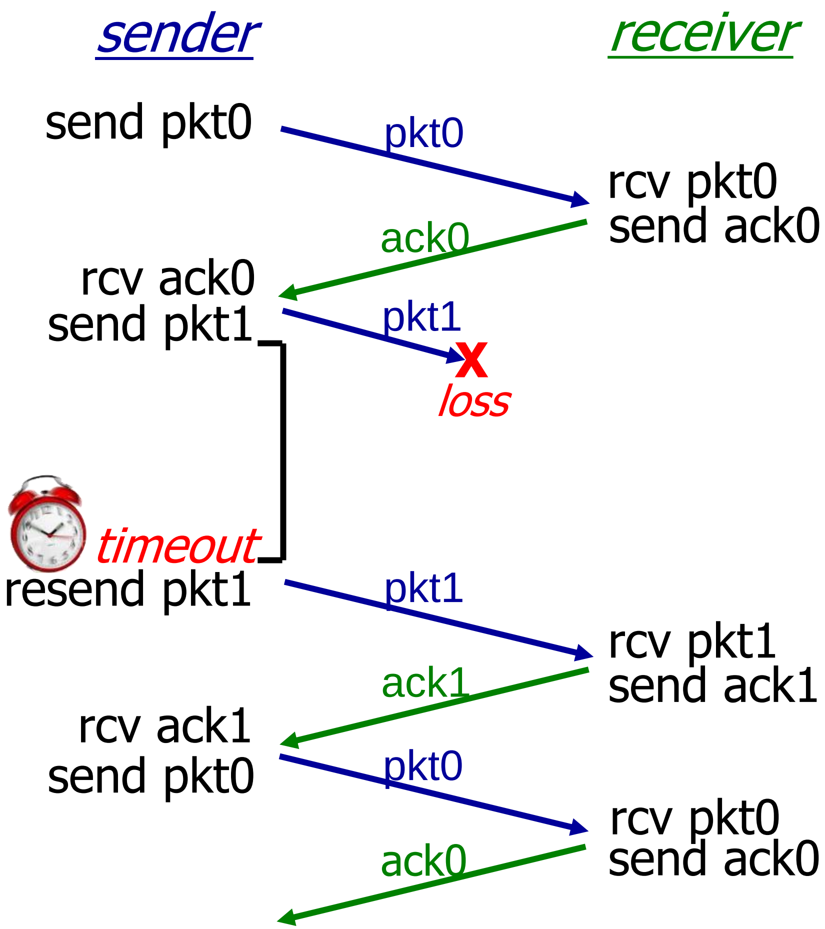
rdt3.0 sender



rdt3.0 in action

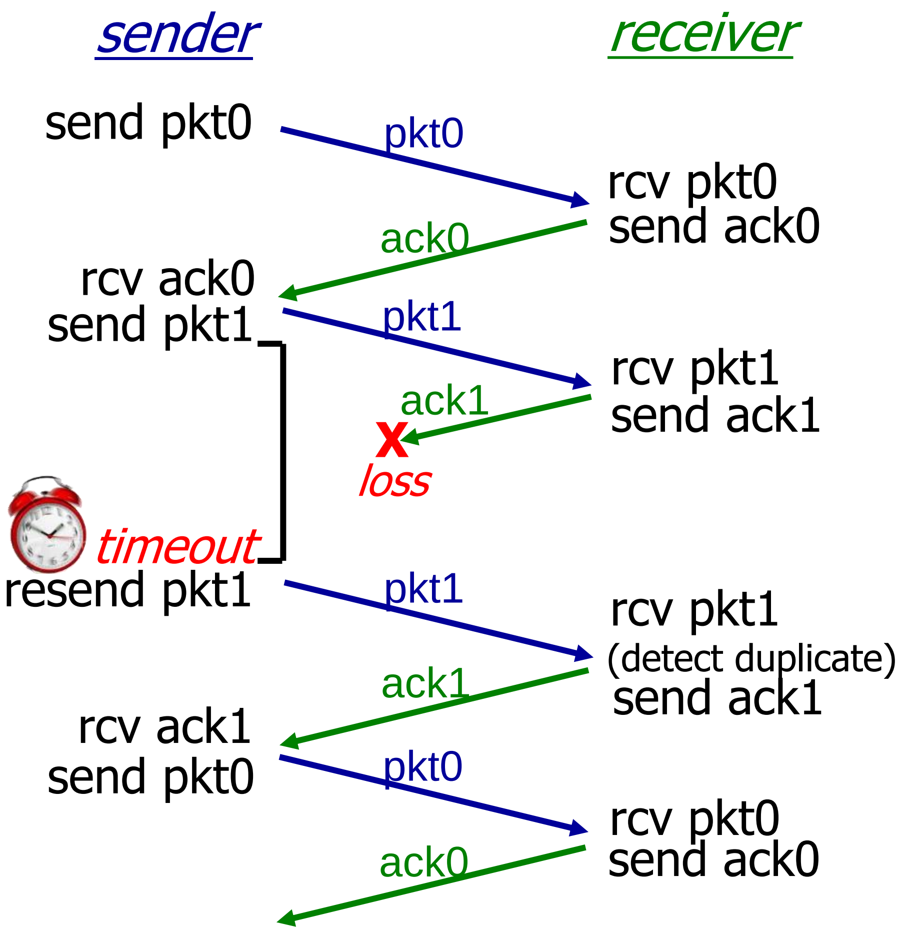


(a) no loss

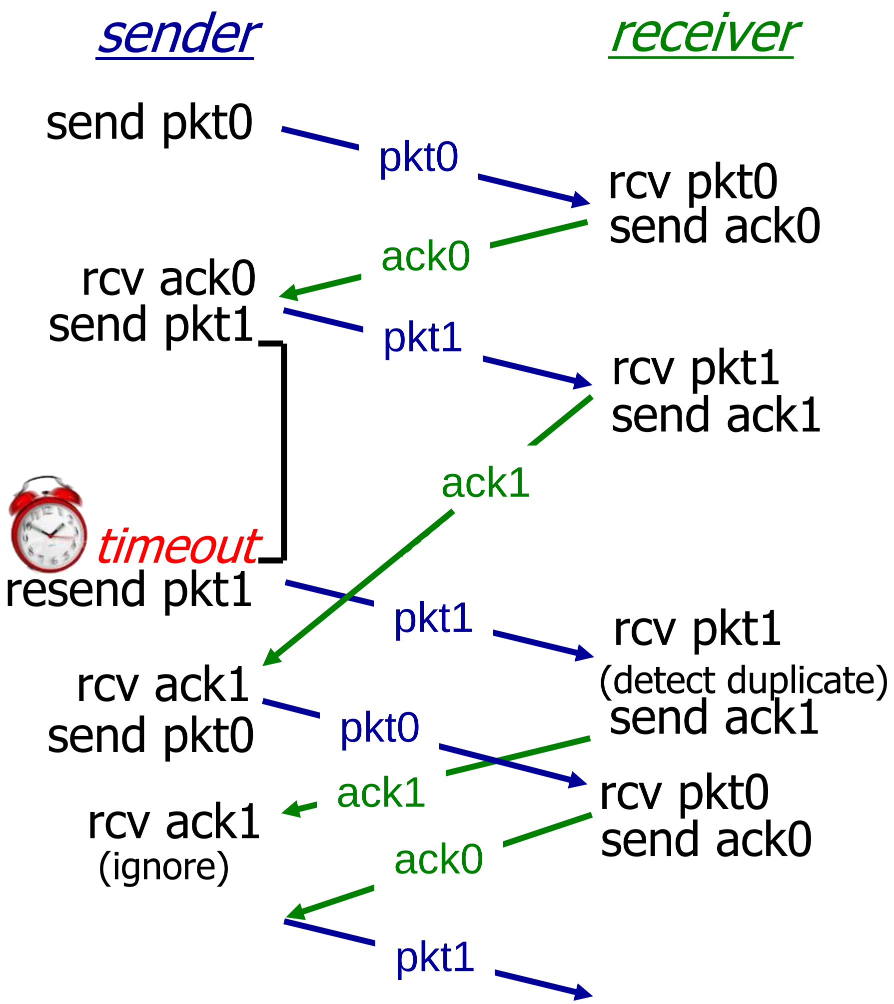


(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

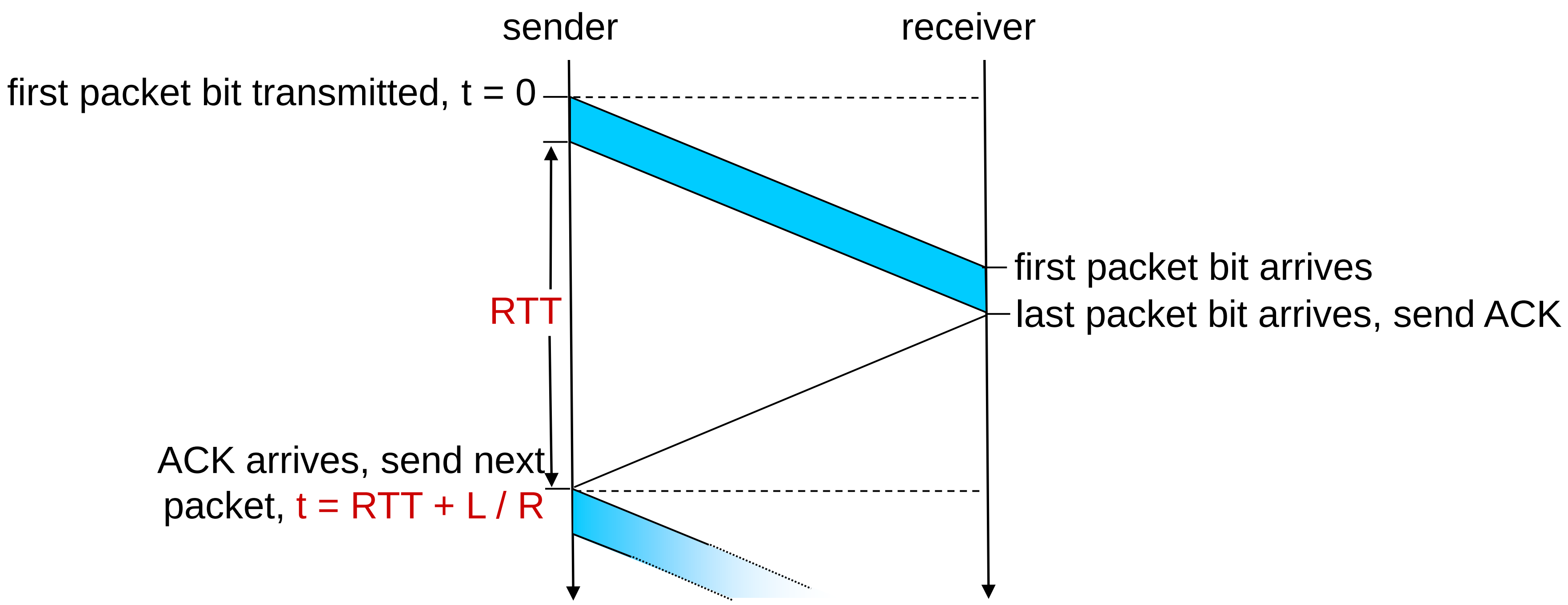
Performance of rdt3.0 (stop-and-wait)

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

- example: 1 Gbps link, 15 ms prop. delay, 8000 bit packet
  - time to transmit packet into channel:

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

rdt3.0: stop-and-wait operation

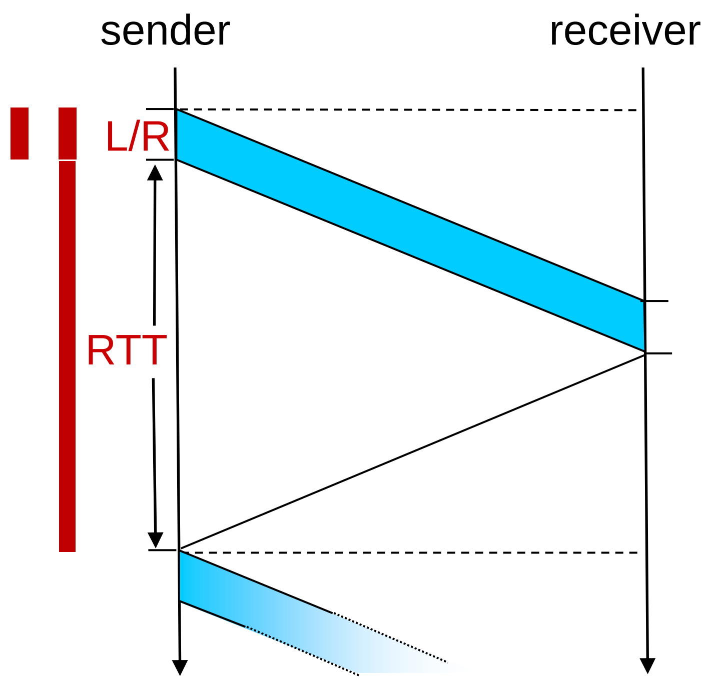


rdt3.0: stop-and-wait operation

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

$$= \frac{.008}{30.008}$$

$$= 0.00027$$

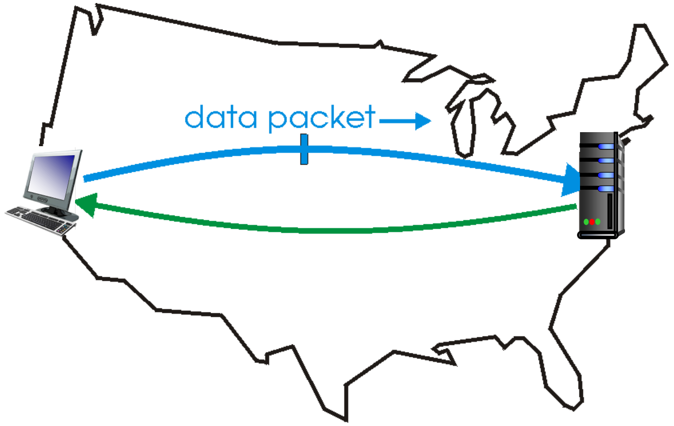


- rdt 3.0 protocol performance stinks!
- Protocol limits performance of underlying infrastructure (channel)

rdt3.0: pipelined protocols operation

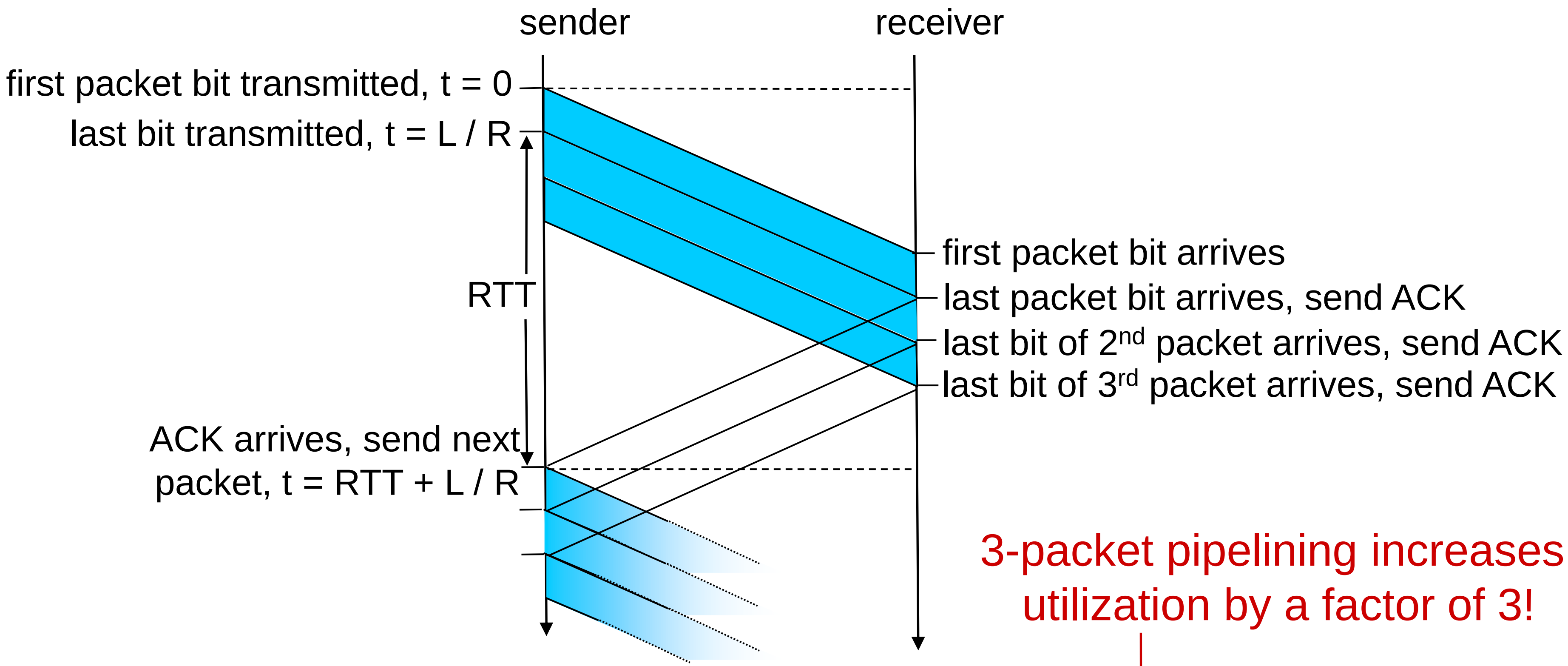
pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged packets

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



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

Pipelining: increased utilization

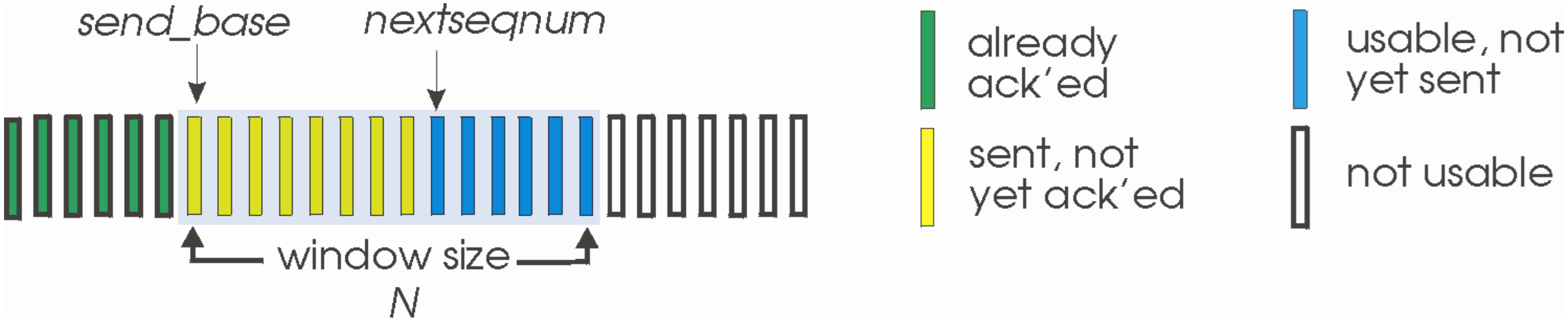


3-packet pipelining increases utilization by a factor of 3!

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

Go-Back-N: sender

- sender: "window" of up to N, consecutive transmitted but unACKed pkts
  - k-bit seq # in pkt header

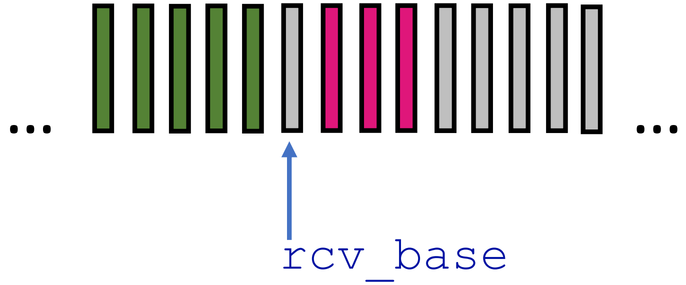


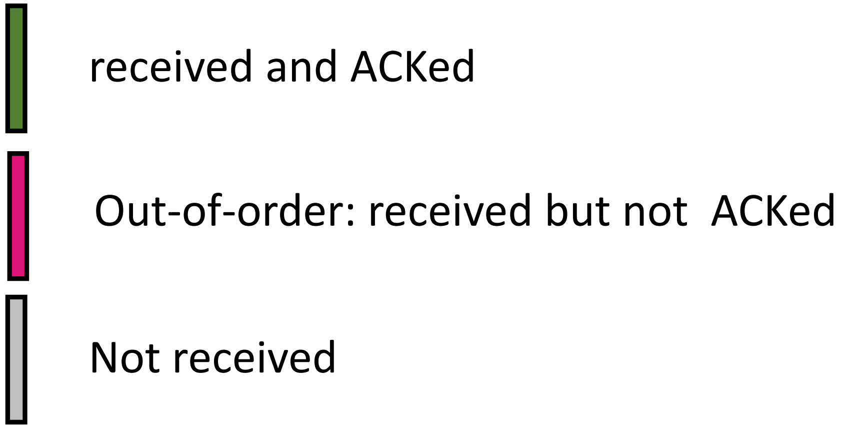
- *cumulative ACK:* ACK(*n*): ACKs all packets up to, including seq # *n*
  - on receiving ACK(*n*): move window forward to begin at *n+1*
- timer for oldest in-flight packet
- *timeout(n):* retransmit packet n and all higher seq # packets in window

Go-Back-N: receiver

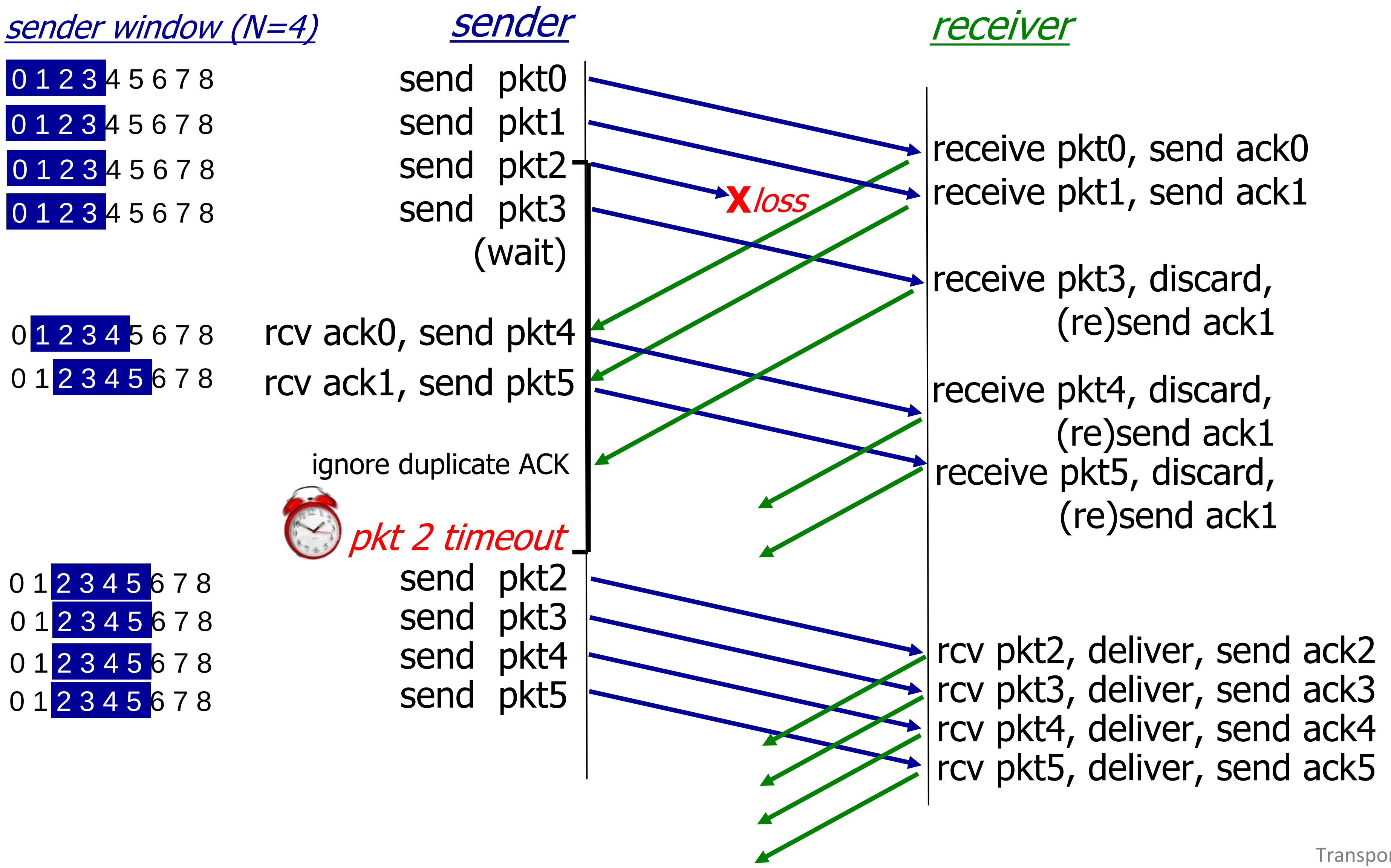
- ACK-only: always send ACK for correctly-received packet so far, with highest *in-order* seq #
  - may generate duplicate ACKs
  - need only remember `rcv_base`
- on receipt of out-of-order packet:
  - can discard (don't buffer) or buffer: an implementation decision
  - re-ACK pkt with highest in-order seq #

Receiver view of sequence number space:





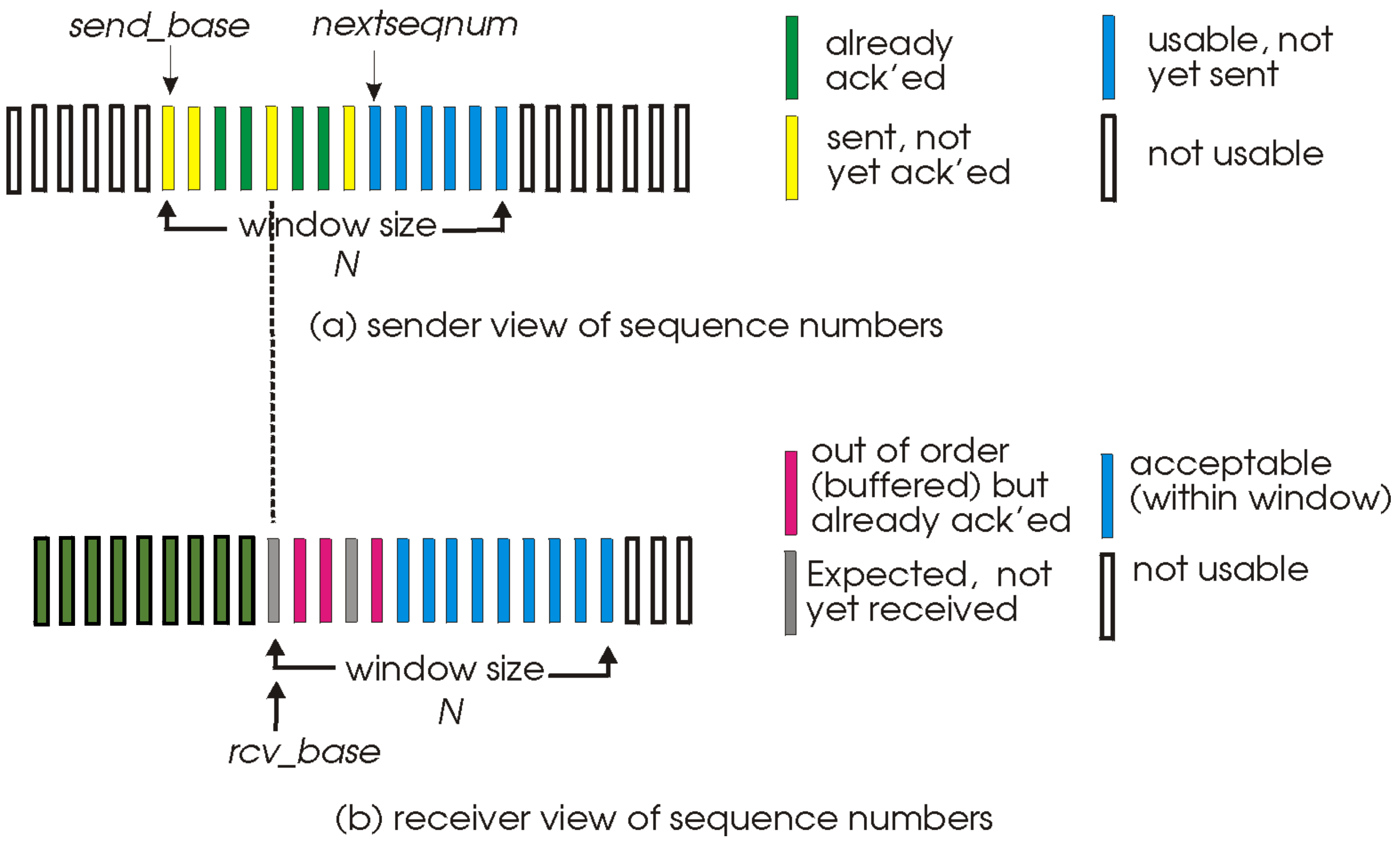
Go-Back-N in action



Selective repeat: the approach

- *pipelining*: *multiple* packets in flight
- *receiver individually ACKs* all correctly received packets
  - buffers packets, as needed, for in-order delivery to upper layer
- sender:
  - maintains (conceptually) a timer for each unACKed pkt
    - timeout: retransmits single unACKed packet associated with timeout
  - maintains (conceptually) "window" over *N* consecutive seq #s
    - limits pipelined, "in flight" packets to be within this window

Selective repeat: sender, receiver windows



Selective repeat: sender and receiver

sender

data from above:

- if next available seq # in window, send packet

timeout(*n*):

- resend packet *n*, restart timer

ACK(*n*) in [sendbase,sendbase+N-1]:

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

receiver

packet *n* in [rcvbase, rcvbase+N-1]

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

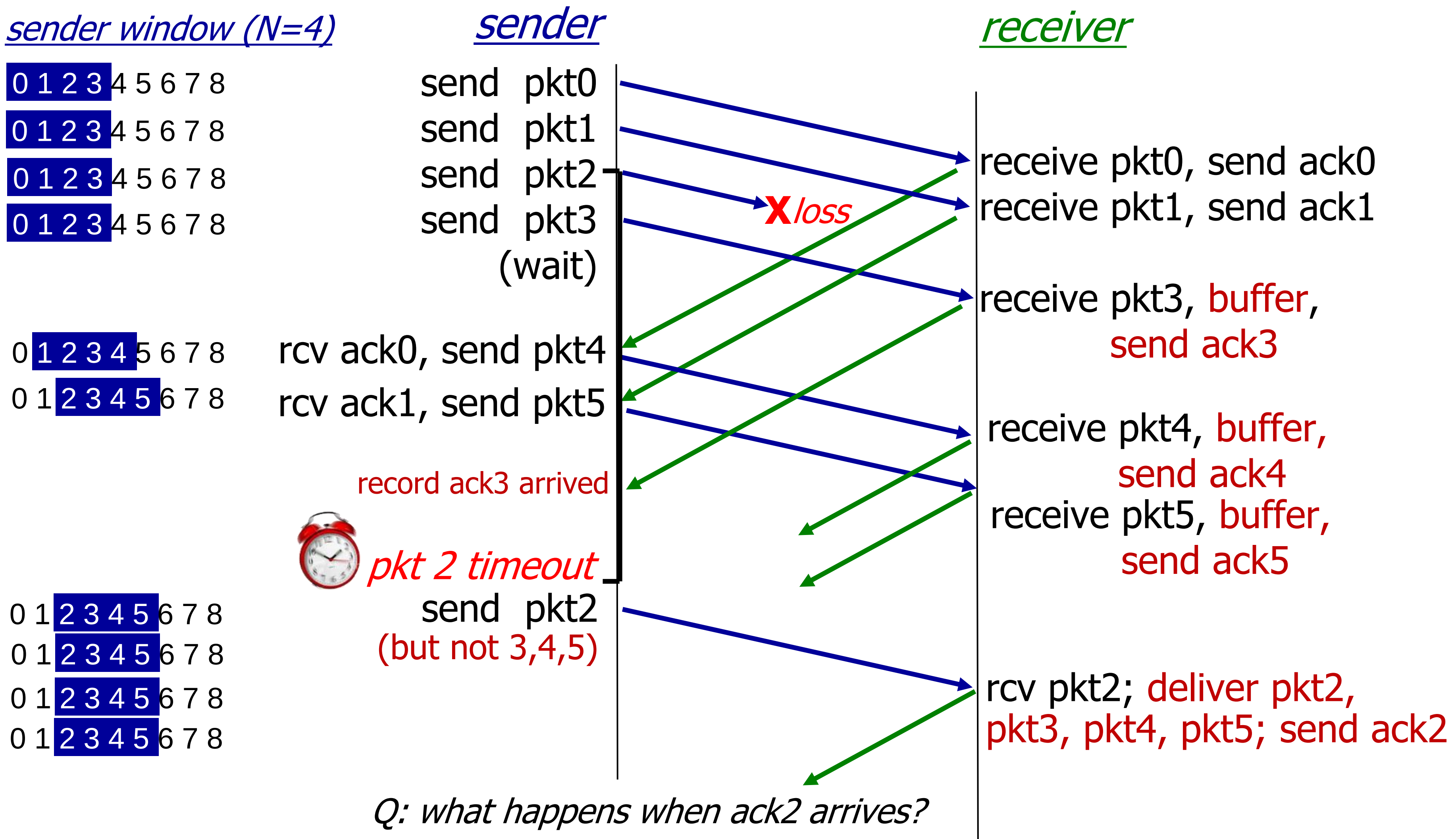
packet *n* in [rcvbase-N,rcvbase-1]

- ACK(*n*)

otherwise:

- ignore

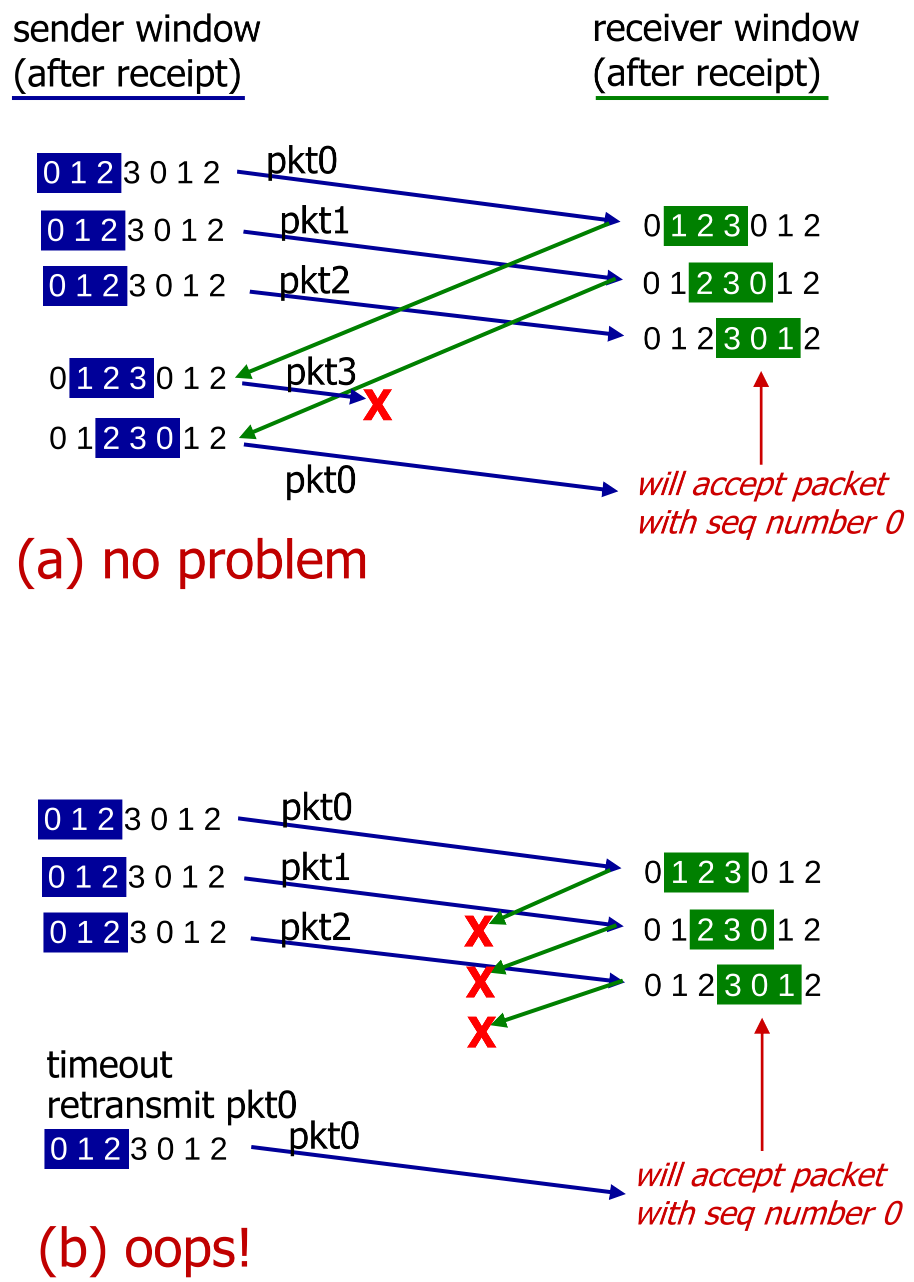
Selective Repeat in action



Selective repeat: a dilemma!

example:

- seq #s: 0, 1, 2, 3 (base 4 counting)
- window size=3



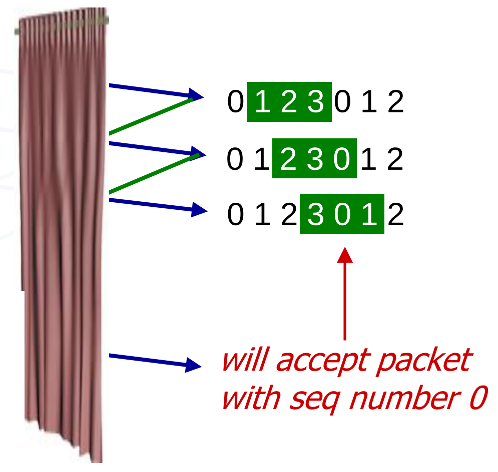
Selective repeat: a dilemma!

example:

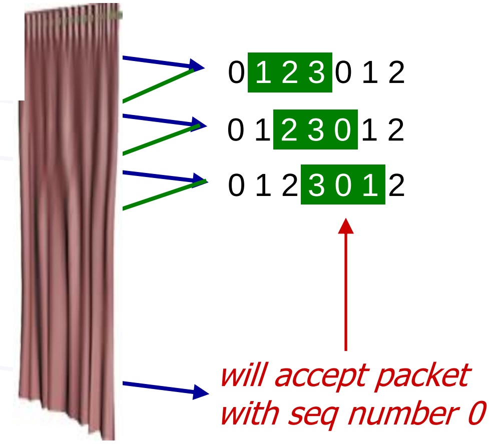
- seq #s: 0, 1, 2, 3 (base 4 counting)
- window size=3

sender window (after receipt)

receiver window (after receipt)



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



Q: what relationship is needed between sequence # size and window size to avoid problem in scenario (b)?

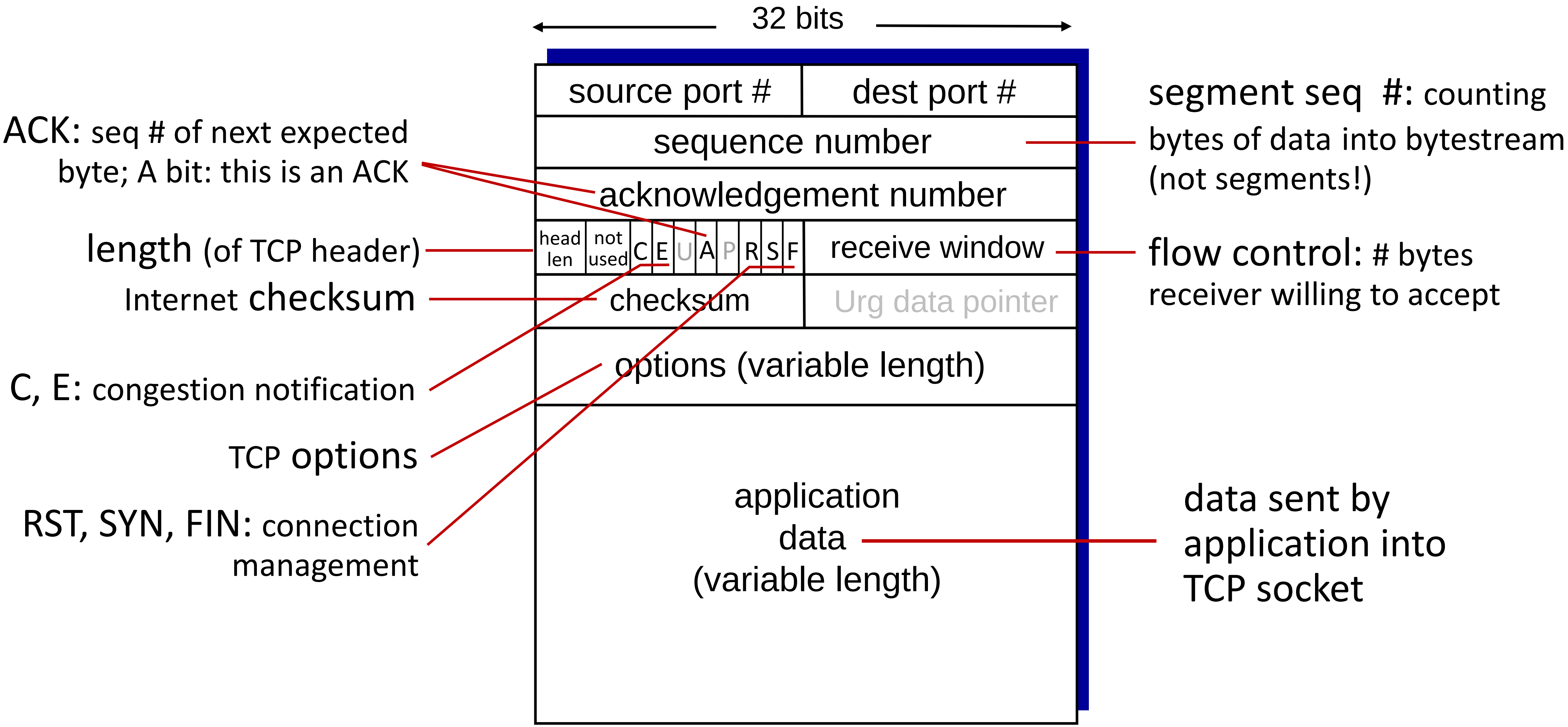
Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- Principles of congestion control
- TCP congestion control

TCP: overview RFCs: 793,1122, 2018, 5681, 7323

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

TCP segment structure



TCP sequence numbers, ACKs

Sequence numbers:

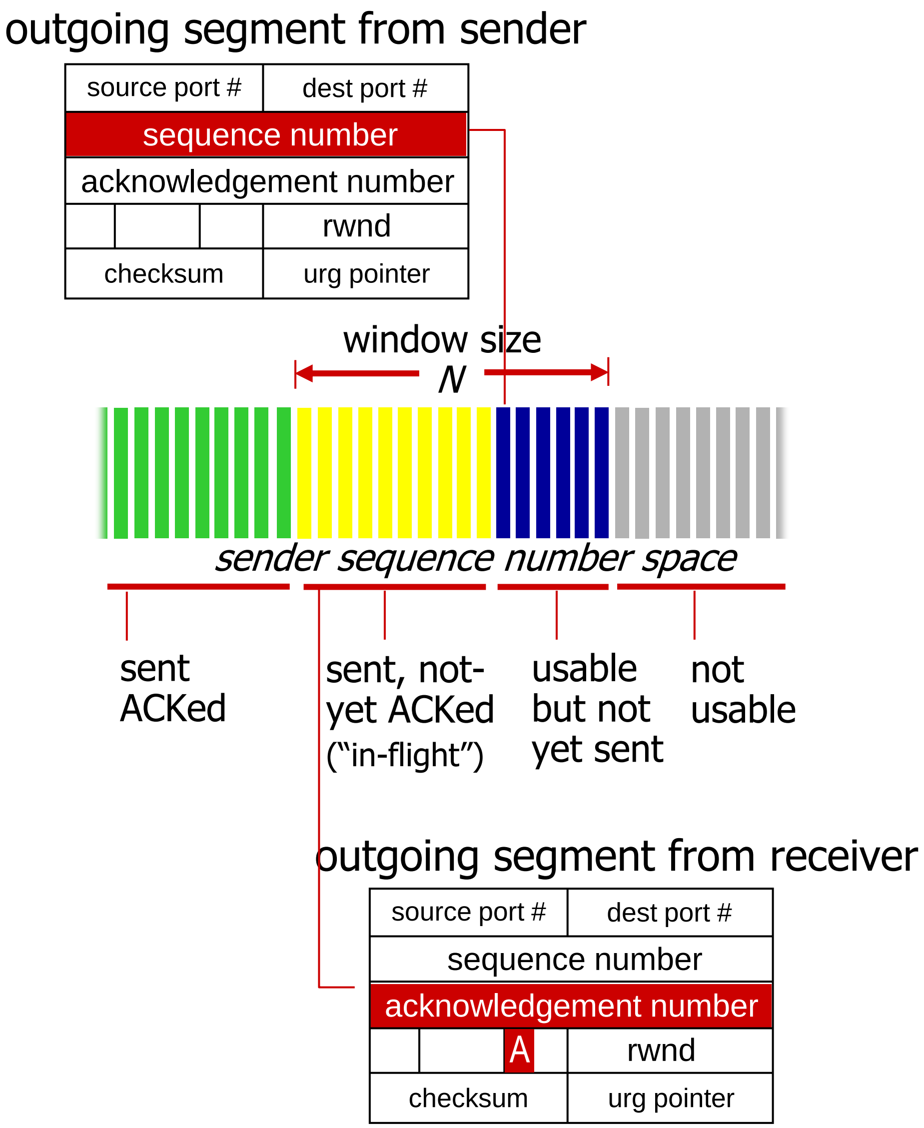
- byte stream "number" of first byte in segment's data

Acknowledgements:

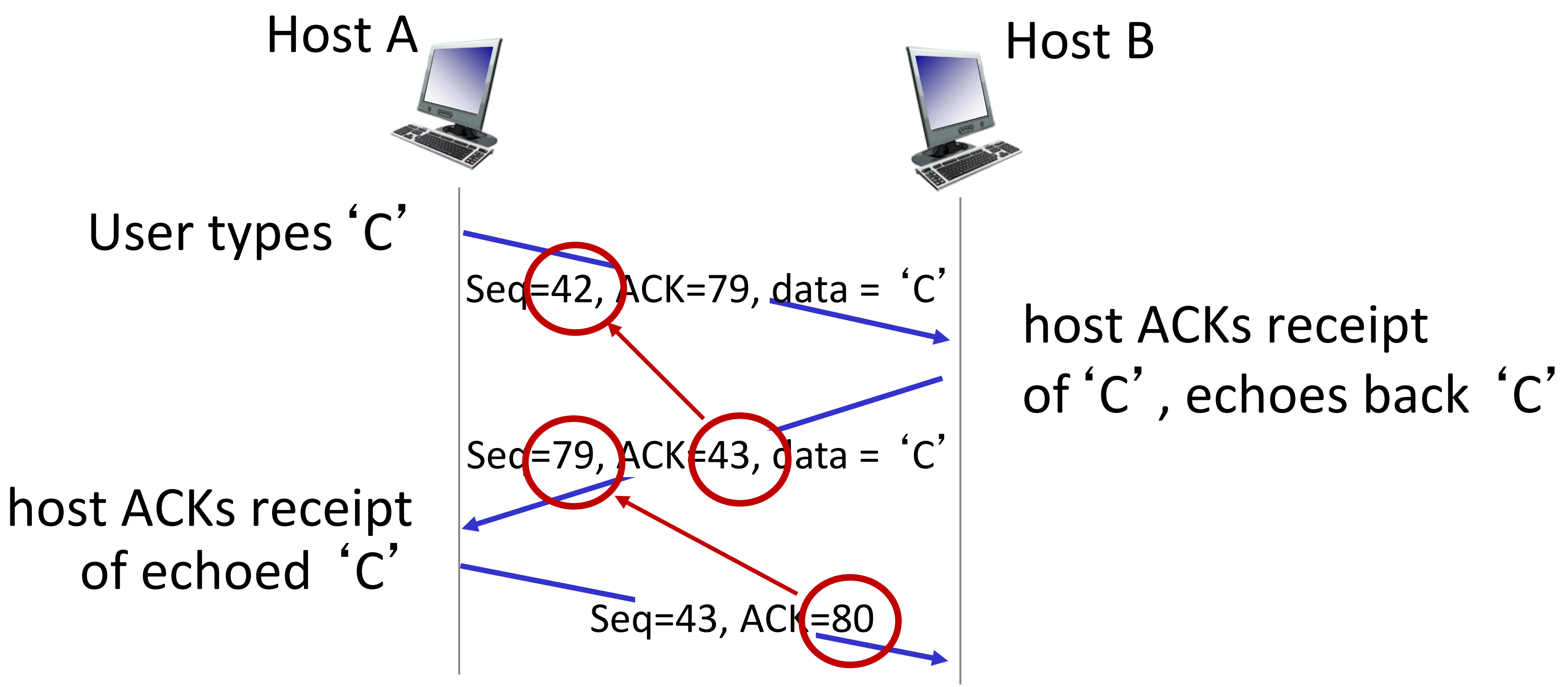
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

- *A:* TCP spec doesn't say, - up to implementor



TCP sequence numbers, ACKs



simple telnet scenario

Chapter 3: roadmap

- Transport-layer services
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- Connectionless transport: UDP
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  - segment structure
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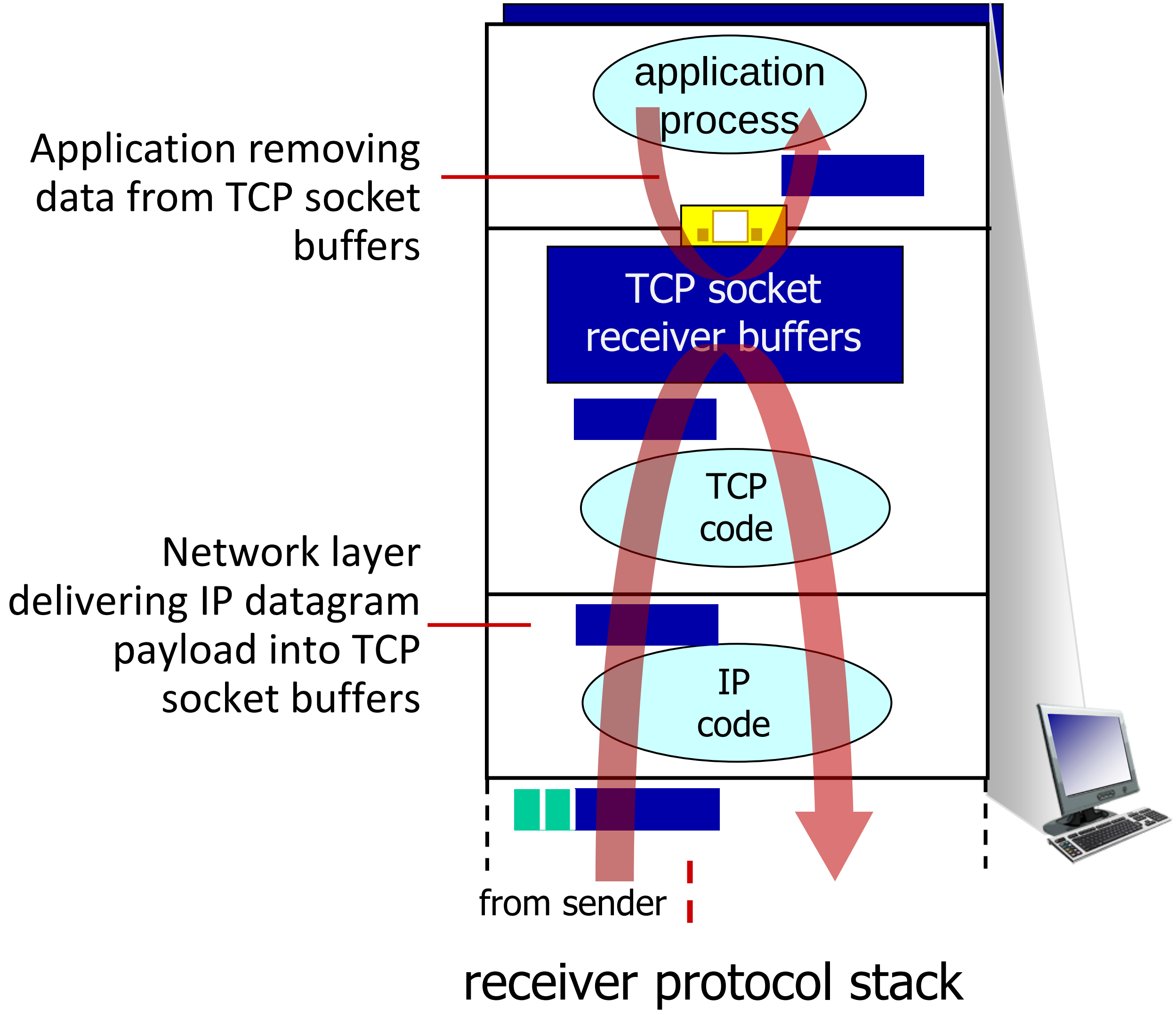
- TCP congestion control

TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?



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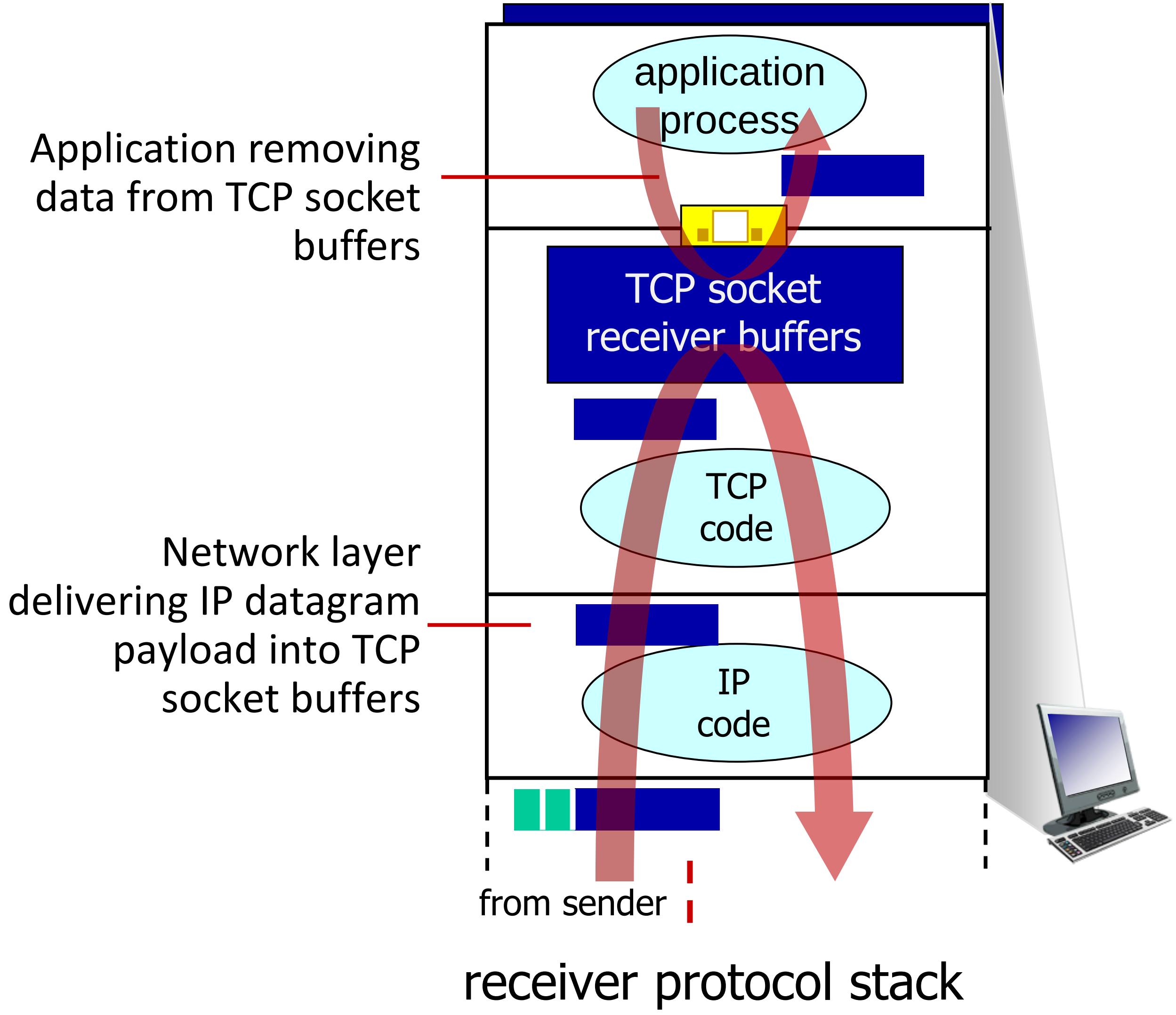

TCP flow control

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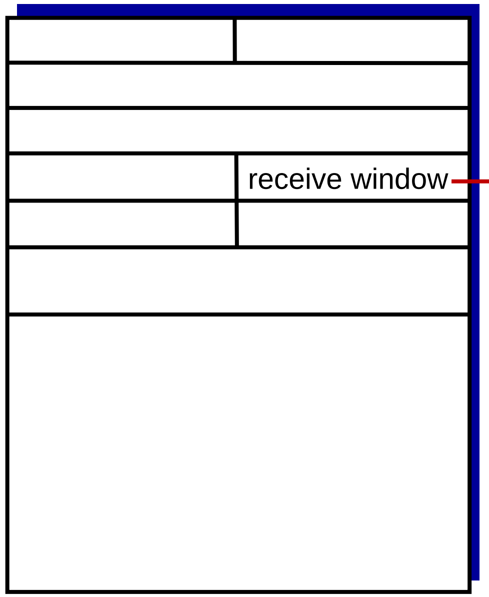
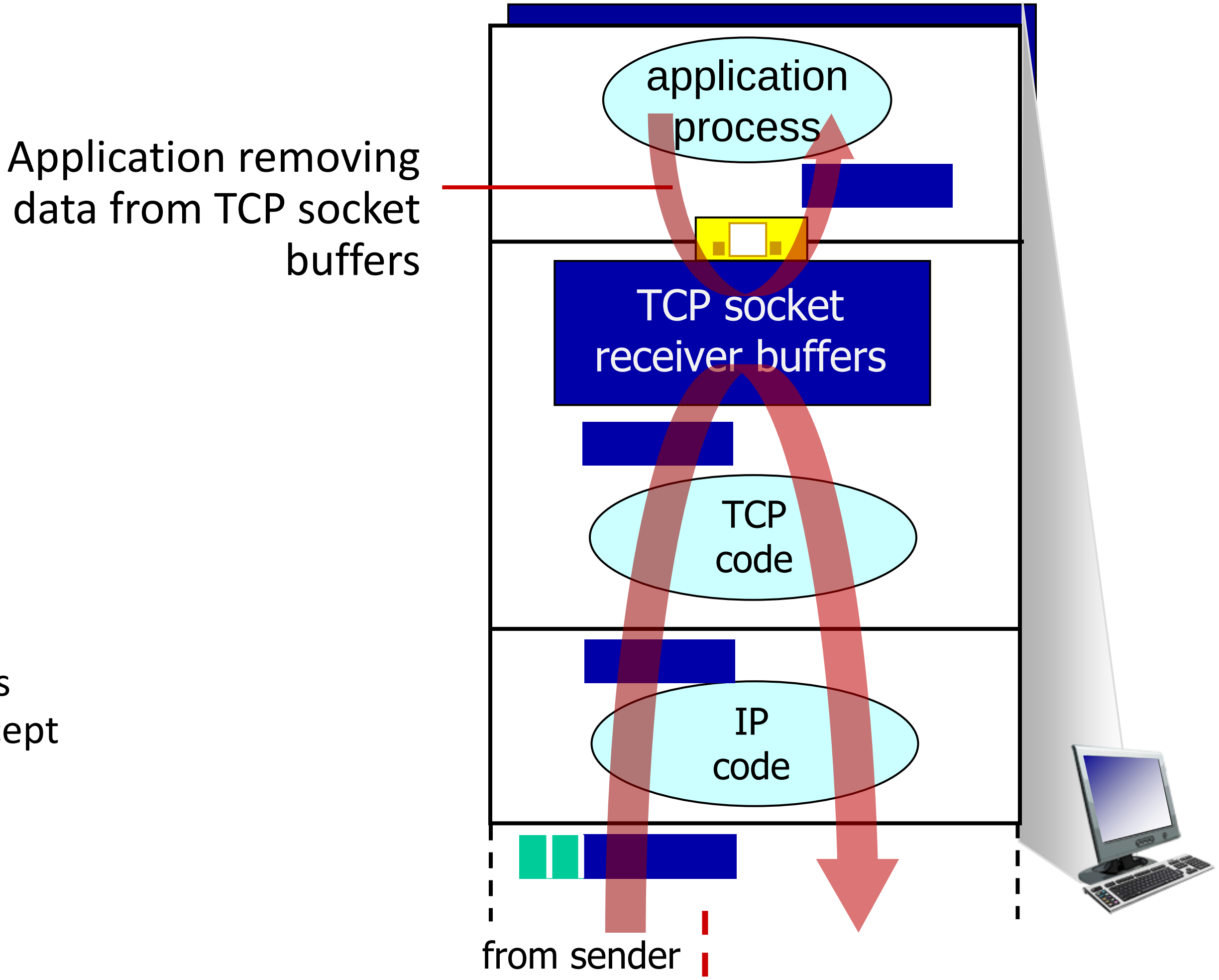


flow control: # bytes receiver willing to accept



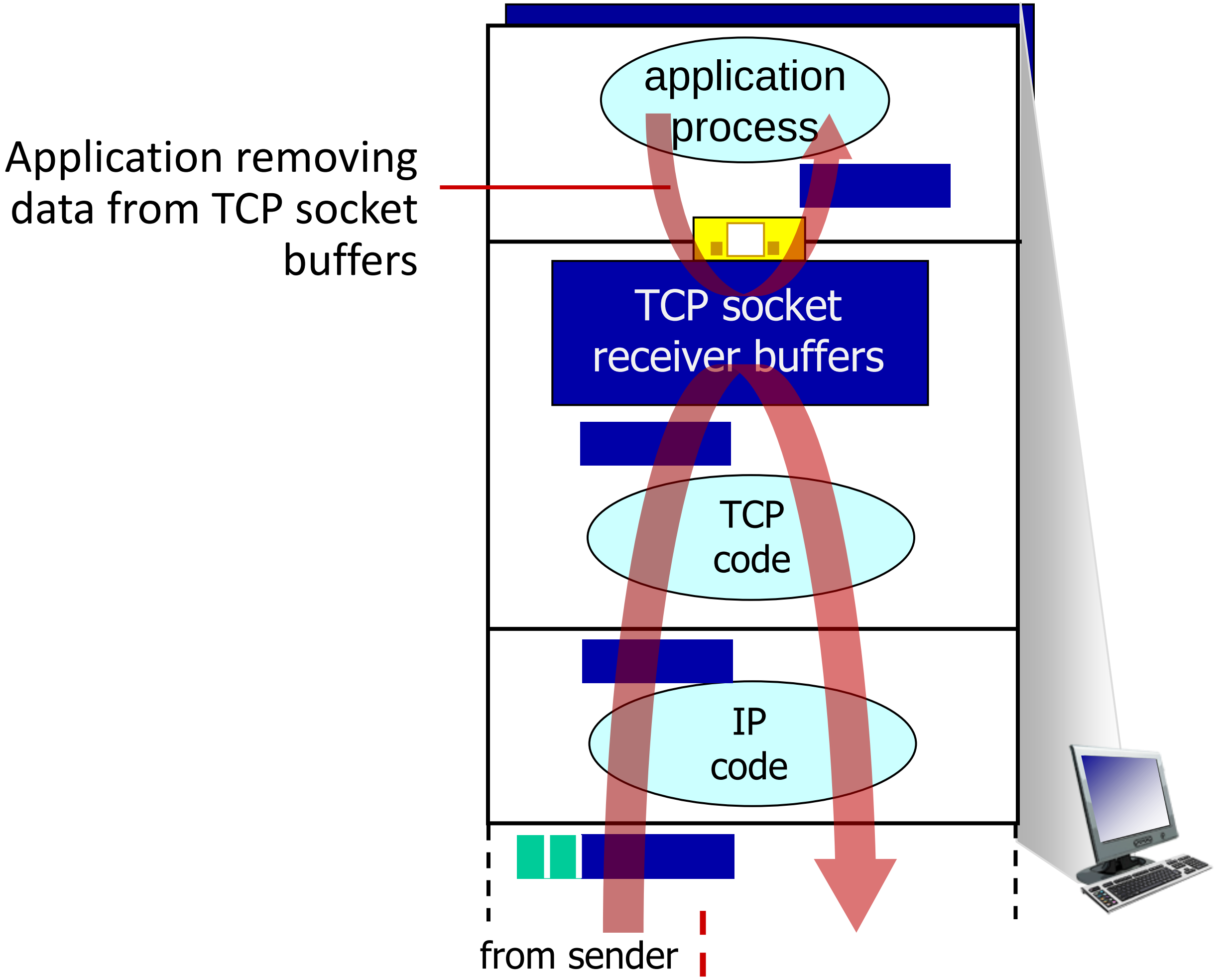
receiver protocol stack

TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?

flow control

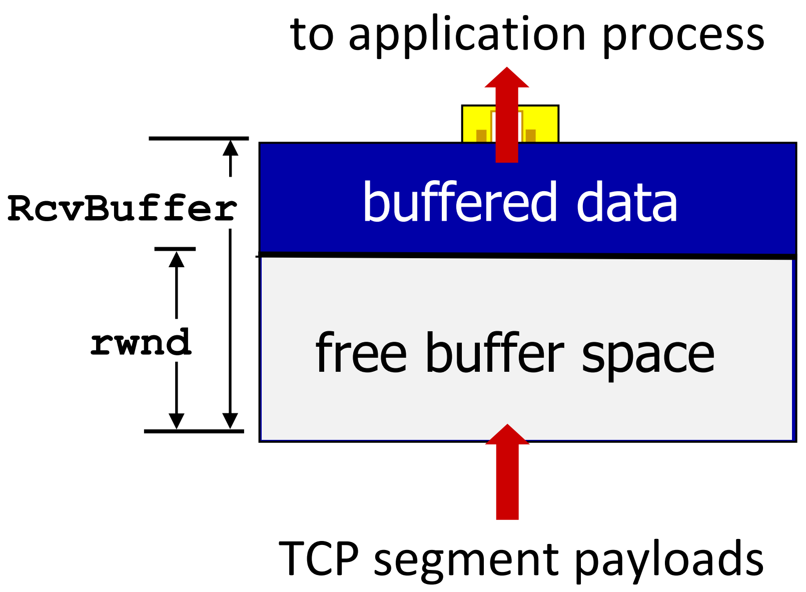
receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast



receiver protocol stack

TCP flow control

- TCP receiver "advertises" free buffer space in **`rwnd`** field in TCP header
  - **`RcvBuffer`** size set via socket options (typical default is 4096 bytes)
  - many operating systems auto-adjust **`RcvBuffer`**
- sender limits amount of unACKed ("in-flight") data to received **`rwnd`**
- guarantees receive buffer will not overflow



TCP receiver-side buffering

TCP flow control

- TCP receiver "advertises" free buffer space in **`rwnd`** field in TCP header
  - **`RcvBuffer`** size set via socket options (typical default is 4096 bytes)
  - many operating systems auto-adjust **`RcvBuffer`**
- sender limits amount of unACKed ("in-flight") data to received **`rwnd`**
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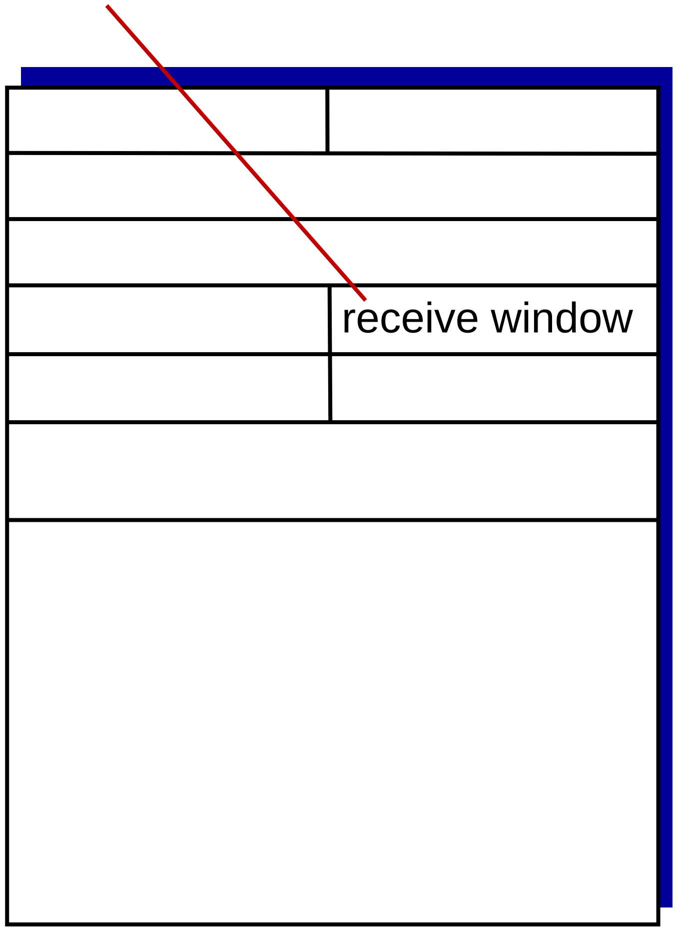
flow control: # bytes receiver willing to accept

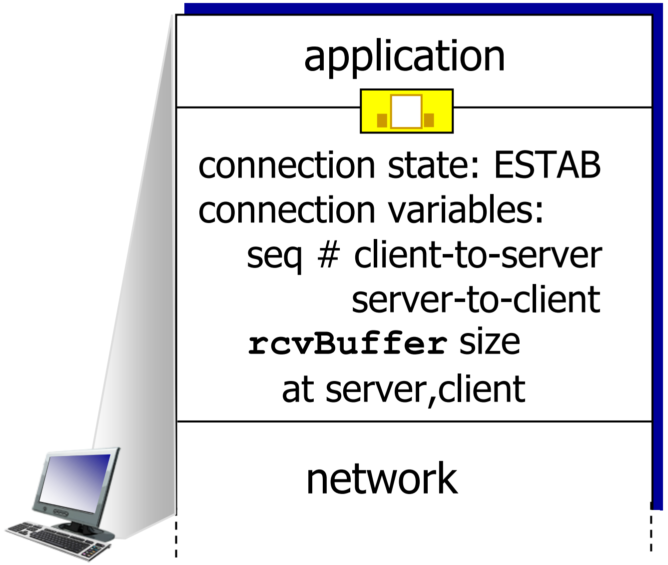


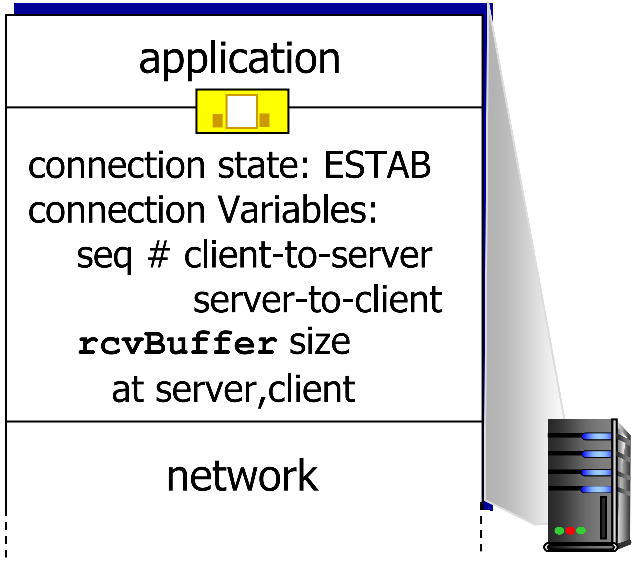
TCP segment format

TCP connection management

before exchanging data, sender/receiver "handshake":

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters (e.g., starting seq #s)



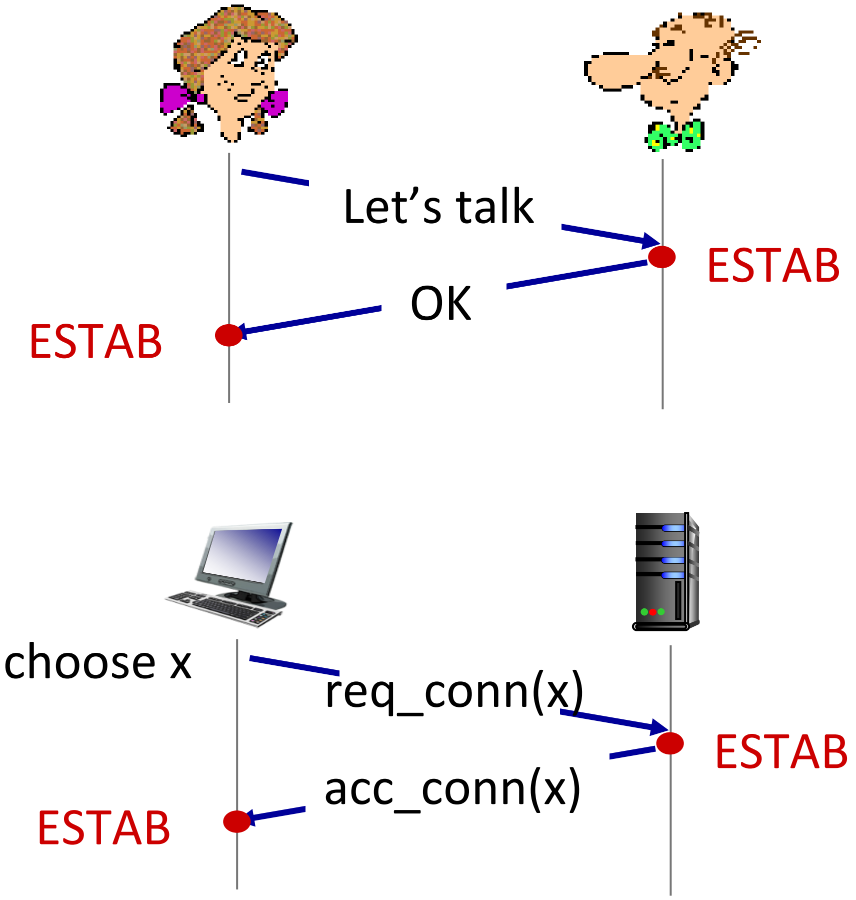



```
Socket clientSocket =
  newSocket("hostname","port number");
```

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

Agreeing to establish a connection

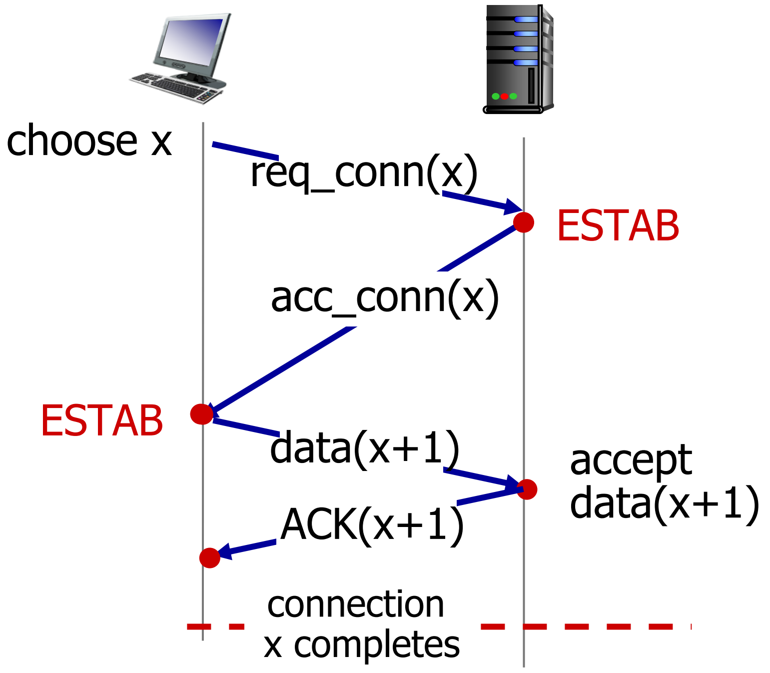
2-way handshake:



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

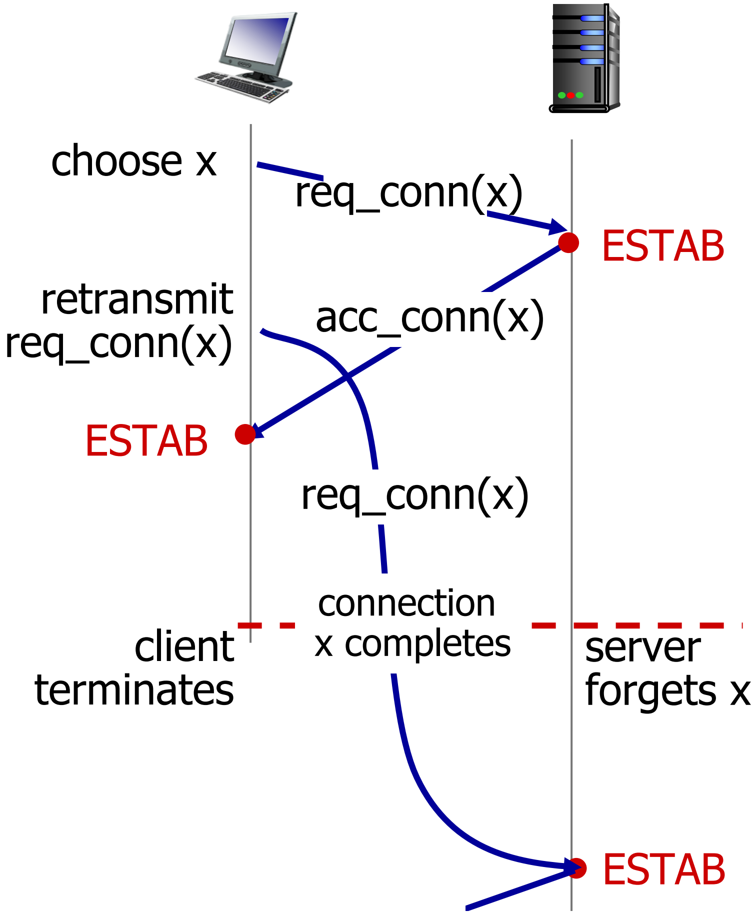
- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side

2-way handshake scenarios



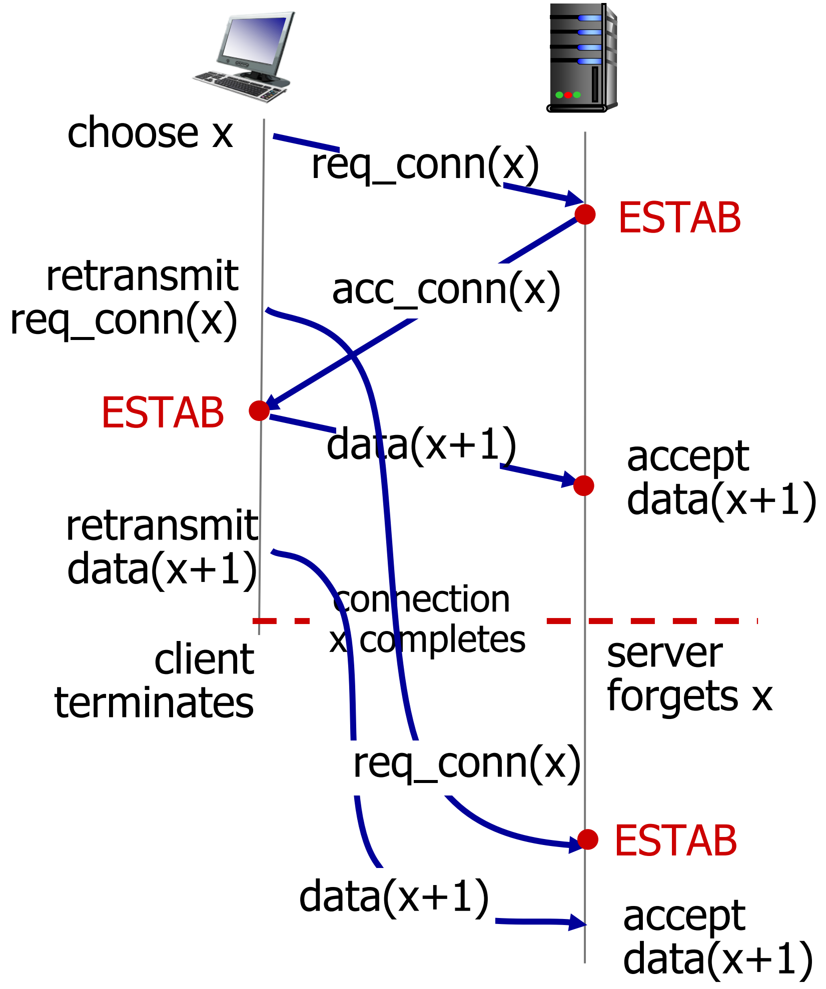
No problem!

2-way handshake scenarios



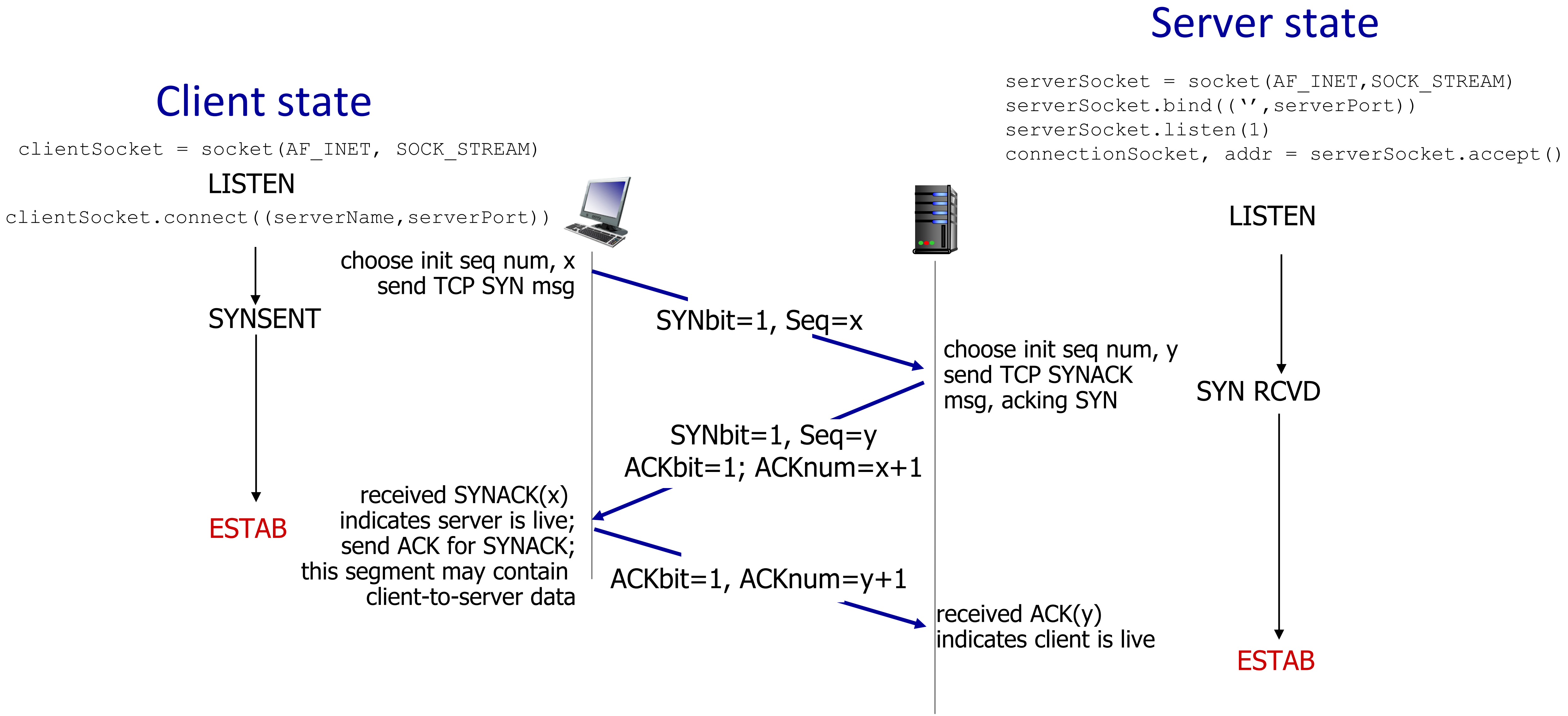
Problem: half open connection! (no client)

2-way handshake scenarios



Problem: dup data accepted!

TCP 3-way handshake



Closing a TCP connection

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

Chapter 4
Network Layer:
Data Plane

Network layer: our goals

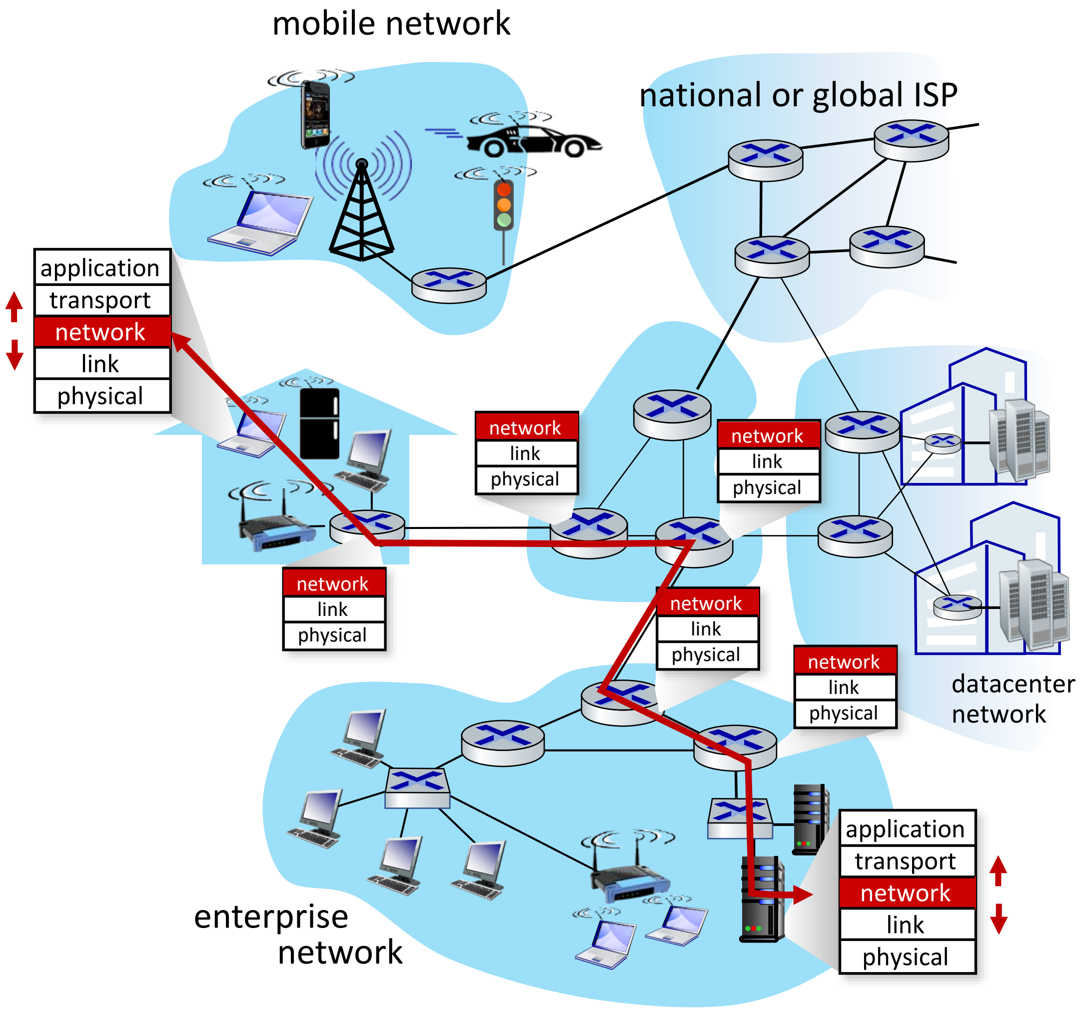
- understand principles behind network layer services, focusing on data plane:
  - network layer service models
  - forwarding versus routing
  - how a router works
  - addressing
  - generalized forwarding
  - Internet architecture
- instantiation, implementation in the Internet
  - IP protocol
  - NAT, middleboxes

Network layer: "data plane" roadmap

- Network layer: overview
  - data plane
  - control plane
- What's inside a router
  - input ports, switching, output ports
  - buffer management, scheduling
- IP: the Internet Protocol
  - datagram format
  - addressing
  - network address translation
  - IPv6
- Generalized Forwarding, SDN
  - Match+action
  - OpenFlow: match+action in action
- Middleboxes

Network-layer services and protocols

- transport segment from sending to receiving host
  - sender: encapsulates segments into datagrams, passes to link layer
  - receiver: delivers segments to transport layer protocol
- network layer protocols in *every Internet device*: hosts, routers
- routers:
  - examines header fields in all IP datagrams passing through it
  - moves datagrams from input ports to output ports to transfer datagrams along end-end path



Two key network-layer functions

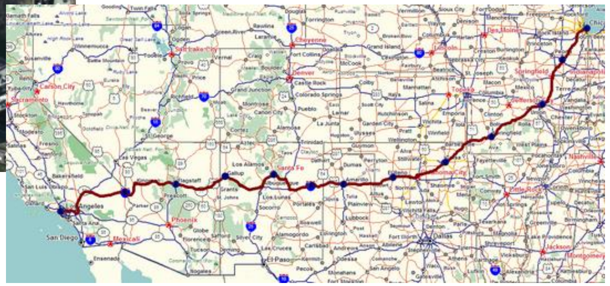
network-layer functions:

- *forwarding:* move packets from a router's input link to appropriate router output link
- *routing:* determine route taken by packets from source to destination
  - *routing algorithms*

analogy: taking a trip

- *forwarding:* process of getting through single interchange
- *routing:* process of planning trip from source to destination

forwarding

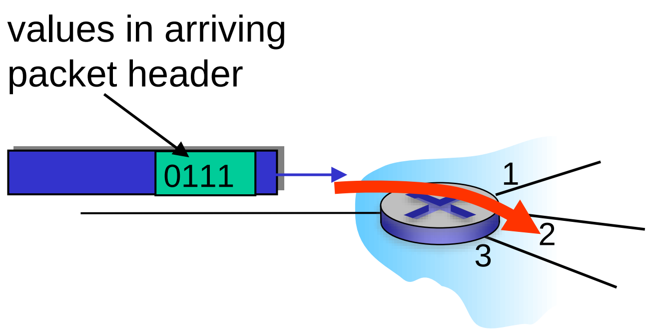


routing

Network layer: data plane, control plane

Data plane:

- *local*, per-router function
- determines how datagram arriving on router input port is forwarded to router output port

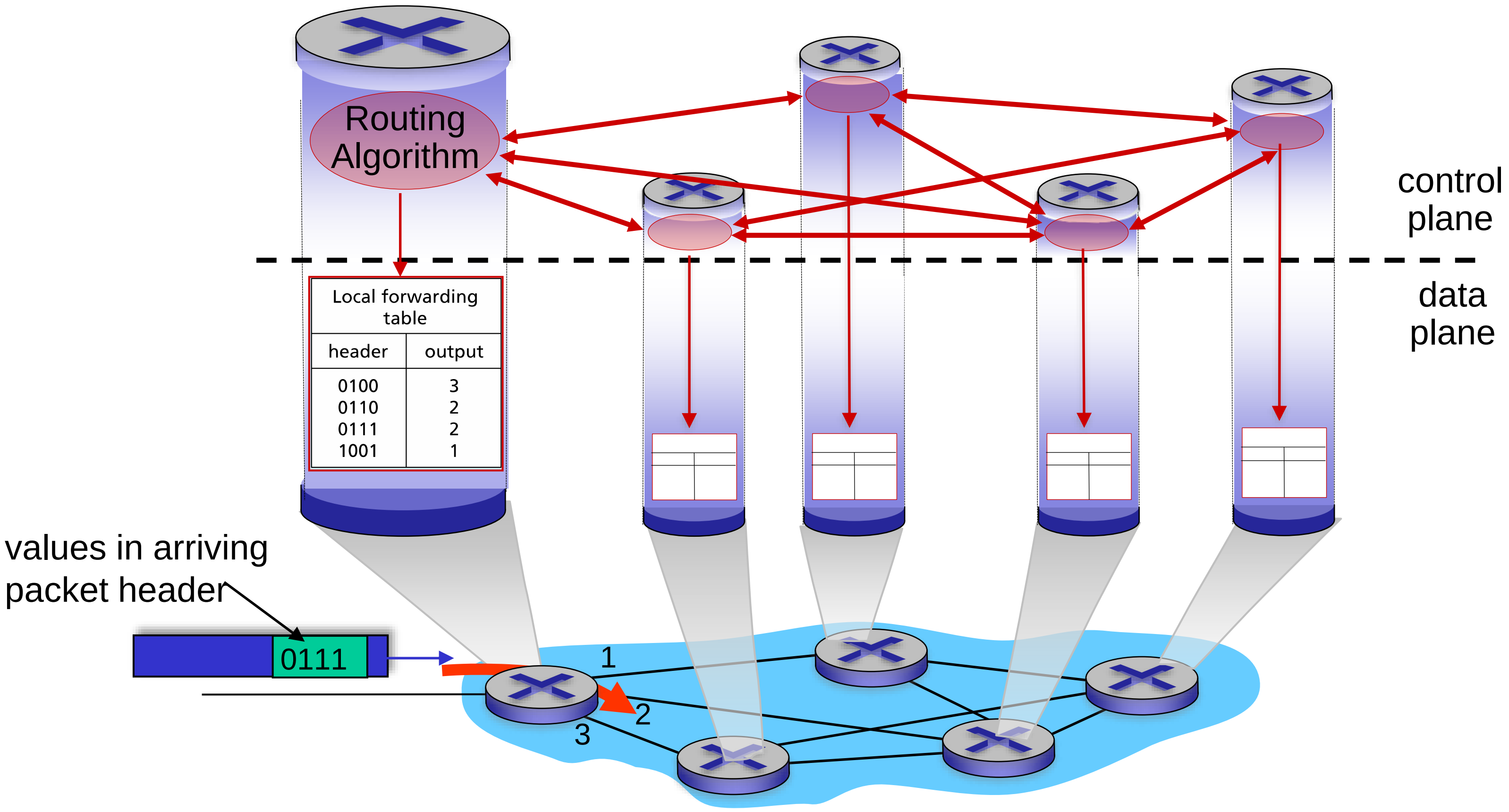


Control plane

- *network-wide* logic
- determines how datagram is routed among routers along end-end path from source host to destination host
- two control-plane approaches:
  - *traditional routing algorithms:* implemented in routers
  - *software-defined networking (SDN)*: implemented in (remote) servers

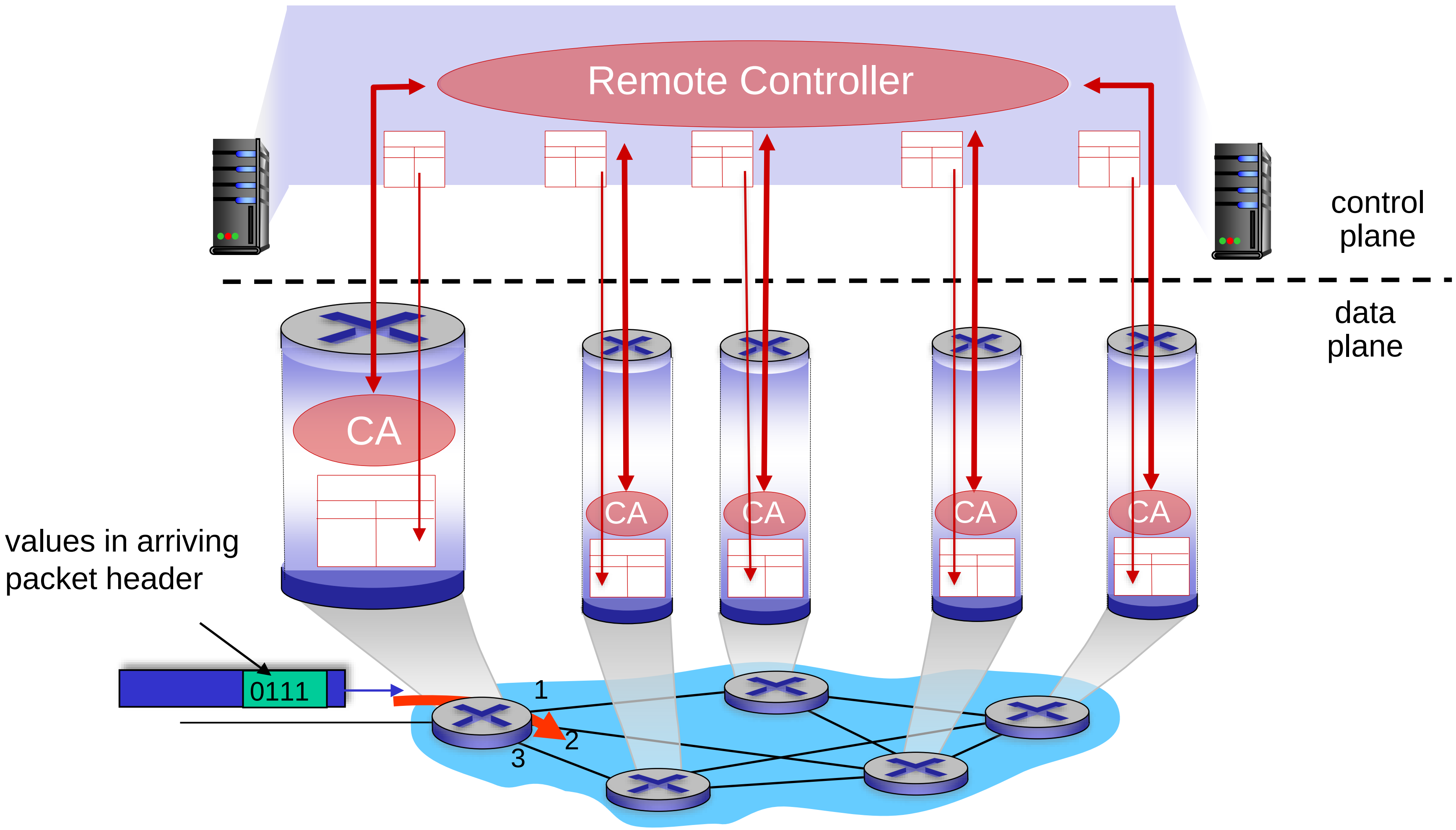
Per-router control plane

Individual routing algorithm components *in each and every router* interact in the control plane



Software-Defined Networking (SDN) control plane

Remote controller computes, installs forwarding tables in routers



Network service model

Q: What *service model* for "channel" transporting datagrams from sender to receiver?

example services for *individual* datagrams:

- guaranteed delivery
- guaranteed delivery with less than 40 msec delay

example services for a *flow* of datagrams:

- in-order datagram delivery
- guaranteed minimum bandwidth to flow
- restrictions on changes in inter-packet spacing

Network-layer service model

Network Architecture	Service Model	Quality of Service (QoS) Guarantees ? Bandwidth	Loss	Order	Timing
Internet	best effort	none	no	no	no

Internet "best effort" service model

No guarantees on*:*

i. successful datagram delivery to destination
ii. timing or order of delivery
iii. bandwidth available to end-end flow

Network-layer service model

Network Architecture	Service Model	Quality of Service (QoS) Guarantees ?			
		Bandwidth	Loss	Order	Timing
Internet	best effort	none	no	no	no
ATM	Constant Bit Rate	Constant rate	yes	yes	yes
ATM	Available Bit Rate	Guaranteed min	no	yes	no
Internet	Intserv Guaranteed (RFC 1633)	yes	yes	yes	yes
Internet	Diffserv (RFC 2475)	possible	possibly	possibly	no

Reflections on best-effort service:

- simplicity of mechanism has allowed Internet to be widely deployed adopted
- sufficient provisioning of bandwidth allows performance of real-time applications (e.g., interactive voice, video) to be "good enough" for "most of the time"
- replicated, application-layer distributed services (datacenters, content distribution networks) connecting close to clients' networks, allow services to be provided from multiple locations
- congestion control of "elastic" services helps

It's hard to argue with success of best-effort service model

Network layer: "data plane" roadmap

- Network layer: overview
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  - control plane
- What's inside a router
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  - buffer management, scheduling
- IP: the Internet Protocol
  - datagram format
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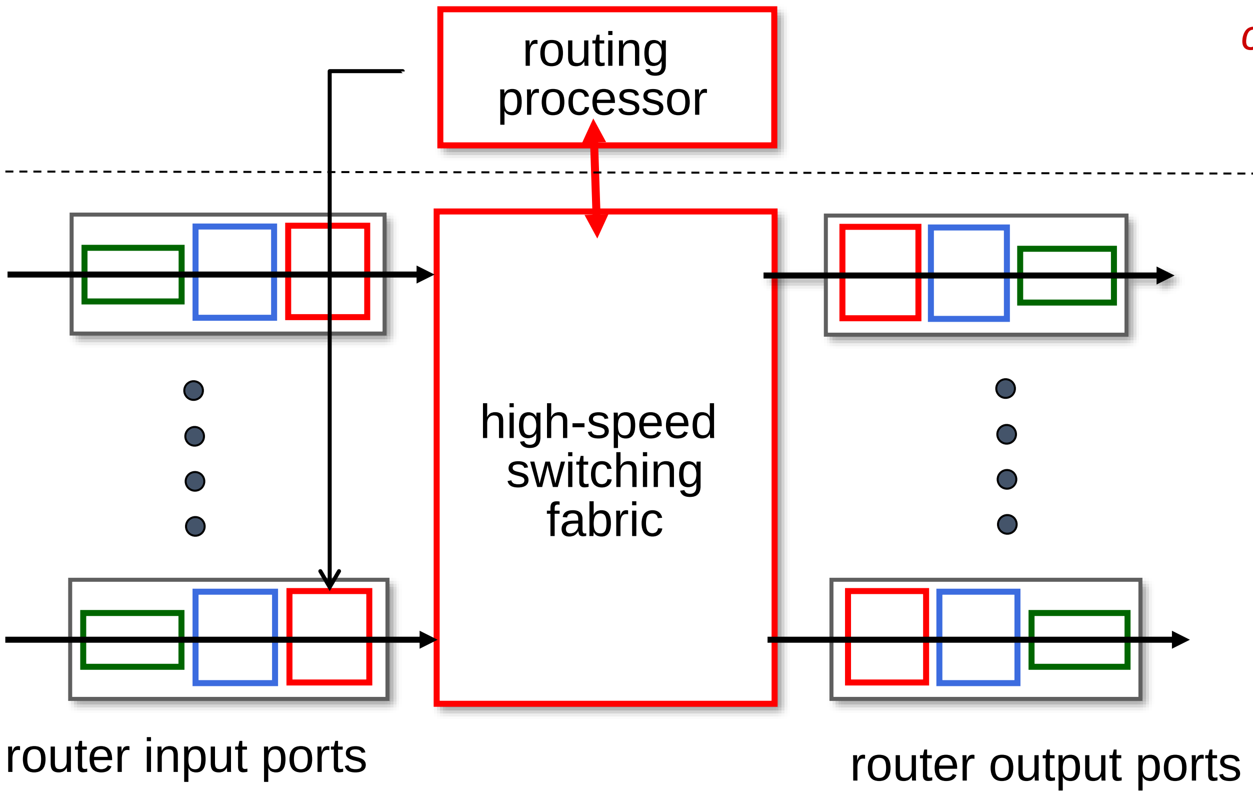
  - network address translation
  - IPv6
- Generalized Forwarding, SDN
  - Match+action
  - OpenFlow: match+action in action
- Middleboxes

Router architecture overview

high-level view of generic router architecture:

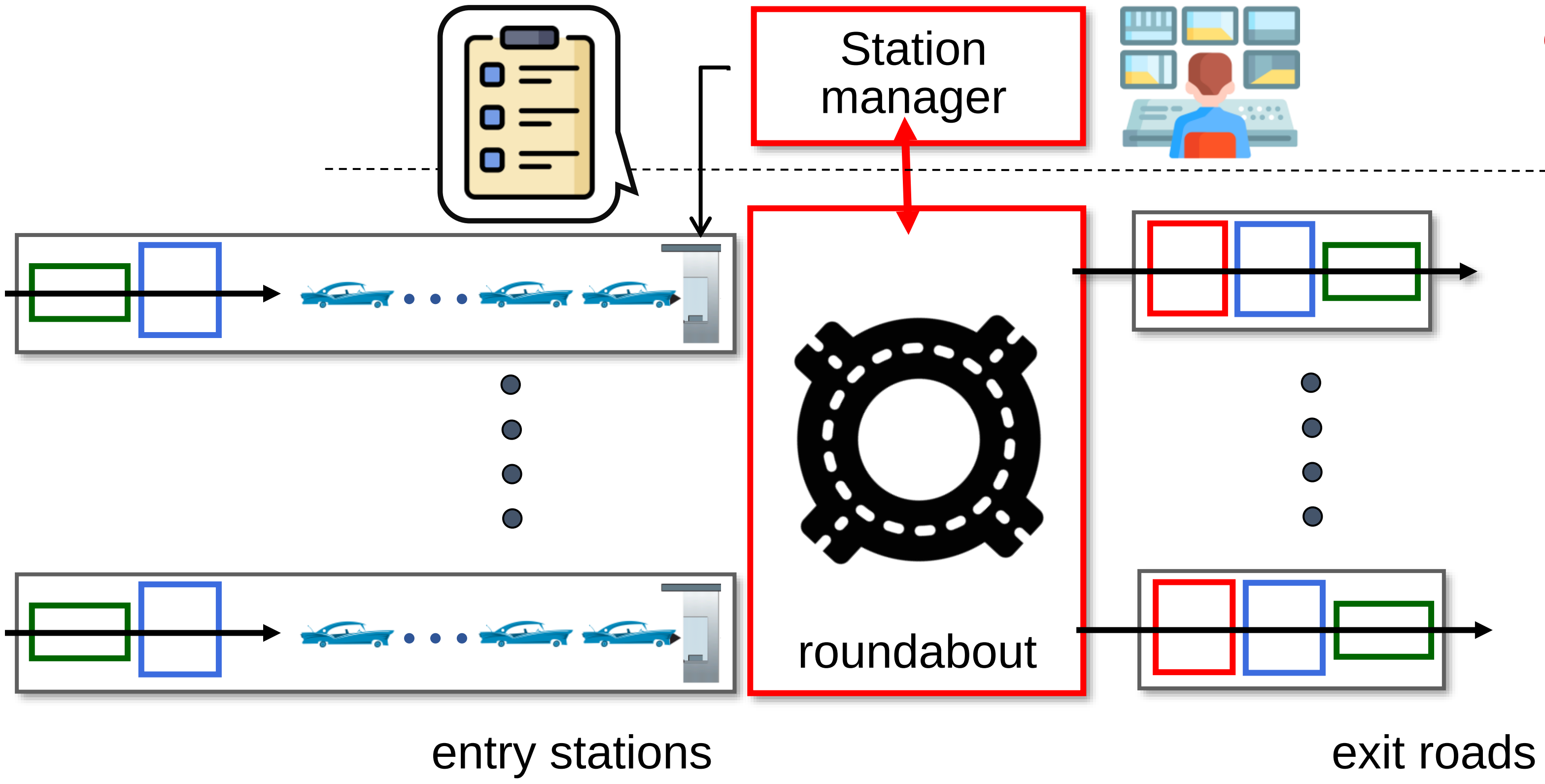


routing, management control plane (software) operates in millisecond time frame

forwarding data plane (hardware) operates in nanosecond timeframe

Router architecture overview

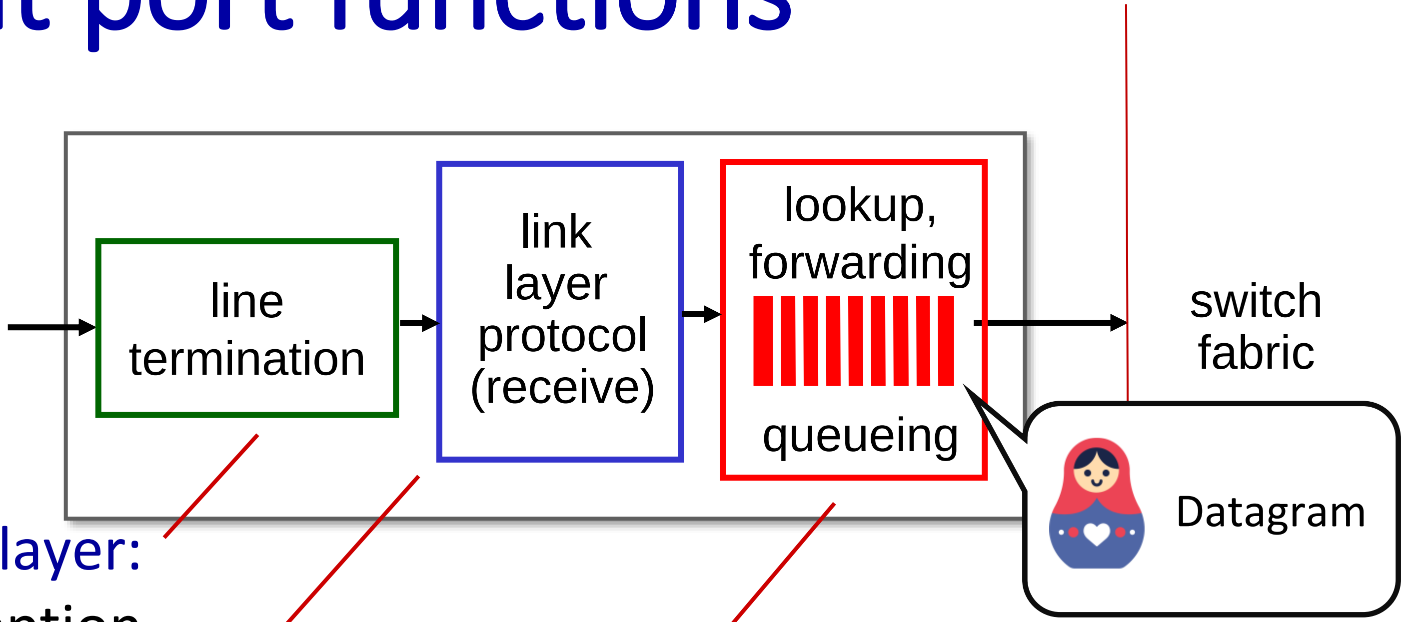
analogy view of generic router architecture:



routing, management control plane (software) operates in millisecond time frame

forwarding data plane (hardware) operates in nanosecond timeframe

Input port functions



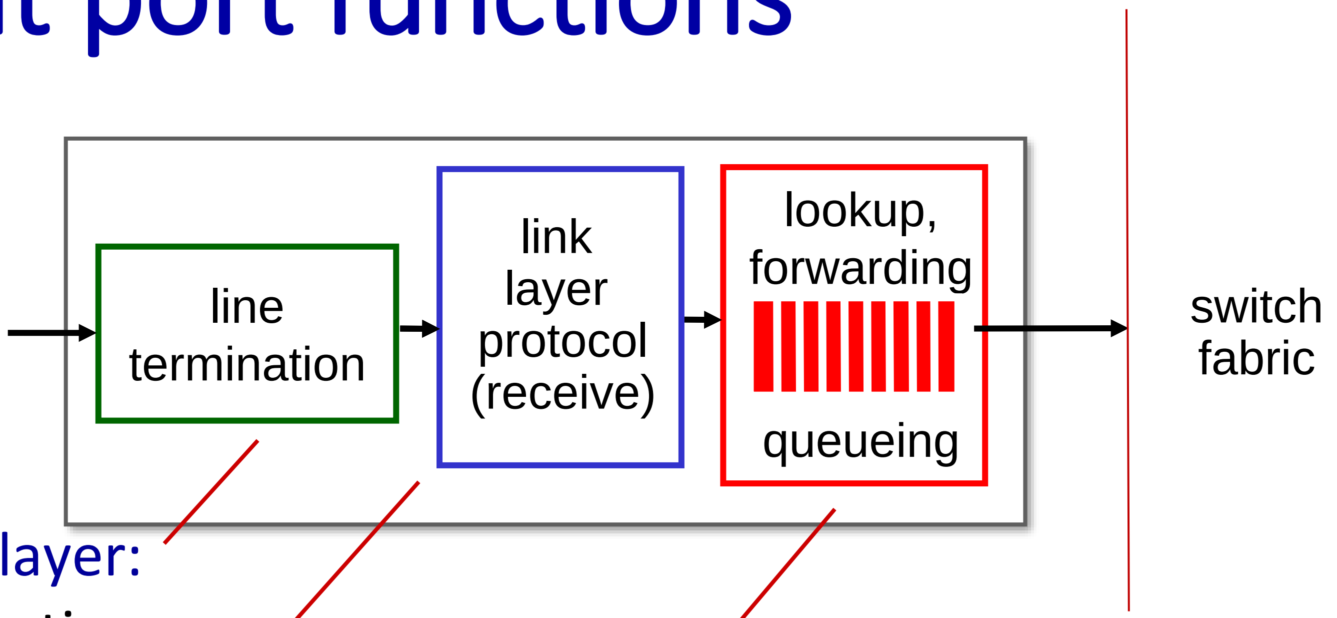
physical layer:
bit-level reception

link layer:
e.g., Ethernet (chapter 6)

decentralized switching:

- using header field values, lookup output port using forwarding table in input port memory *("match plus action")*
- goal: complete input port processing at 'line speed'
- input port queuing: if datagrams arrive faster than forwarding rate into switch fabric

Input port functions



physical layer:
bit-level reception

link layer:
e.g., Ethernet (chapter 6)

decentralized switching:

- using header field values, lookup output port using forwarding table in input port memory *("match plus action")*
- destination-based forwarding: forward based only on destination IP address (traditional)
- generalized forwarding: forward based on any set of header field values

Destination-based forwarding

forwarding table

Destination Address Range	Link Interface
11001000 00010111 00010000 00000000 through 11001000 00010111 00011000 11111111	0
11001000 00010111 00010000 00000100 through 11001000 00010111 00010000 00000111	3
11001000 00010111 00011001 00000000 through 11001000 00010111 00011111 11111111	2
otherwise	3

Q: but what happens if ranges don't divide up so nicely?

Longest prefix matching

longest prefix match

when looking for forwarding table entry for given destination address, use *longest* address prefix that matches destination address.

Destination Address Range	Link interface
11001000 00010111 00010*** ********	0
11001000 00010111 00011000 ********	1
11001000 00010111 00011*** ********	2
otherwise	3

examples:

11001000 00010111 00010110 10100001 which interface?

11001000 00010111 00011000 10101010 which interface?

Longest prefix matching

longest prefix match

when looking for forwarding table entry for given destination address, use *longest* address prefix that matches destination address.

Destination Address Range	Link interface
11001000 00010111 00010*** ********	0
11001000 000[illegible]0111 00011000 ********	1
11001000 [illegible]1 00011*** ********	2
otherwise	3

match!

examples:

11001000 00010111 00010110 10100001 which interface?

11001000 00010111 00011000 10101010 which interface?

Longest prefix matching

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match!

examples:

11001000 00010111 00010110 10100001	which interface?
11001000 00010111 00011000 10101010	which interface?

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otherwise	3

match!

examples:

11001000 00010111 00010110 10100001 which interface?

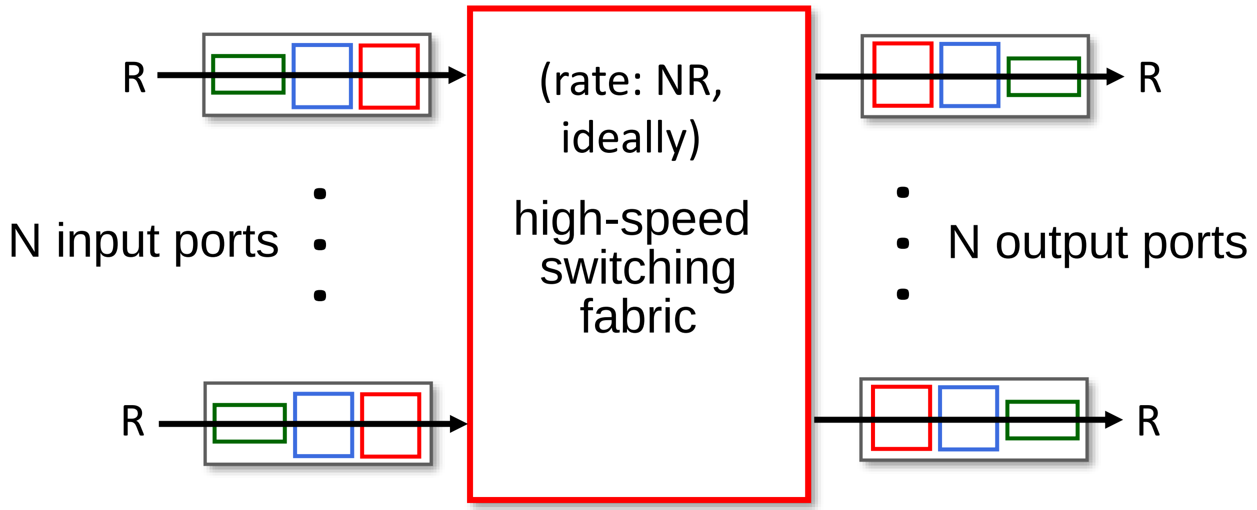
11001000 00010111 00011000 10101010 which interface?

Longest prefix matching

- we'll see *why* longest prefix matching is used shortly, when we study addressing
- longest prefix matching: often performed using ternary content addressable memories (TCAMs)
  - *content addressable:* present address to TCAM: retrieve address in one clock cycle, regardless of table size
  - Cisco Catalyst: ~1M routing table entries in TCAM

Switching fabrics

- transfer packet from input link to appropriate output link
- switching rate: rate at which packets can be transfer from inputs to outputs
  - often measured as multiple of input/output line rate
  - N inputs: switching rate N times line rate desirable



Switching fabrics

- transfer packet from input link to appropriate output link
- switching rate: rate at which packets can be transfer from inputs to outputs
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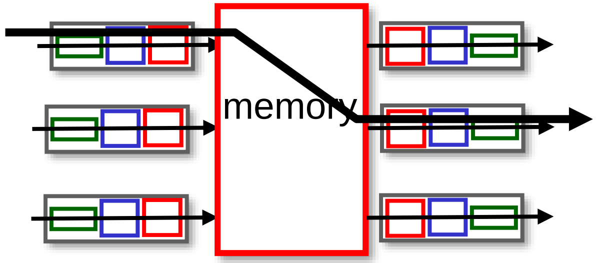
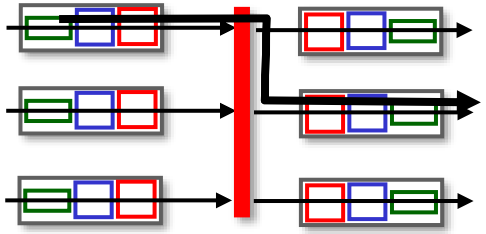
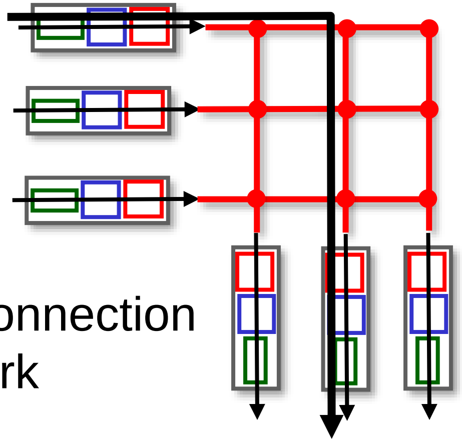
- three major types of switching fabrics:



memory



bus

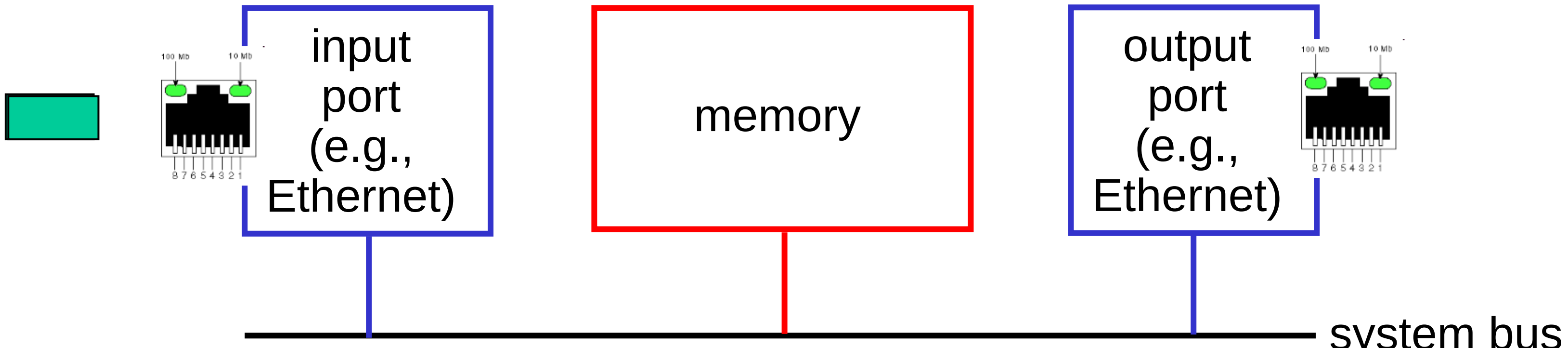


interconnection network

Switching via memory

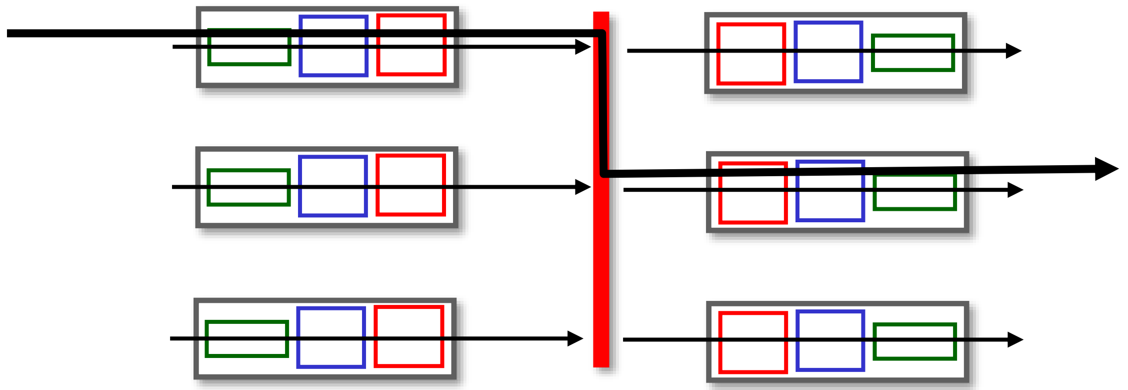
first generation routers:

- traditional computers with switching under direct control of CPU
- packet copied to system's memory
- speed limited by memory bandwidth (2 bus crossings per datagram)



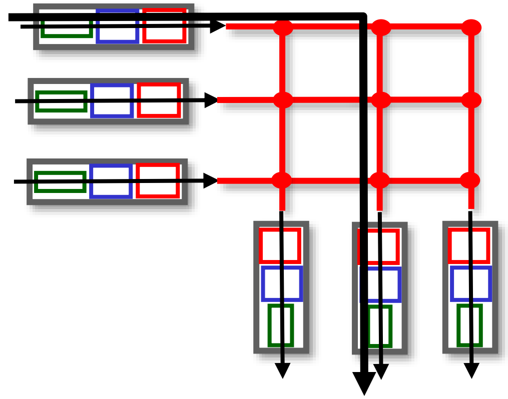
Switching via a bus

- datagram from input port memory to output port memory via a shared bus
- *bus contention:* switching speed limited by bus bandwidth
- 32 Gbps bus, Cisco 5600: sufficient speed for access routers

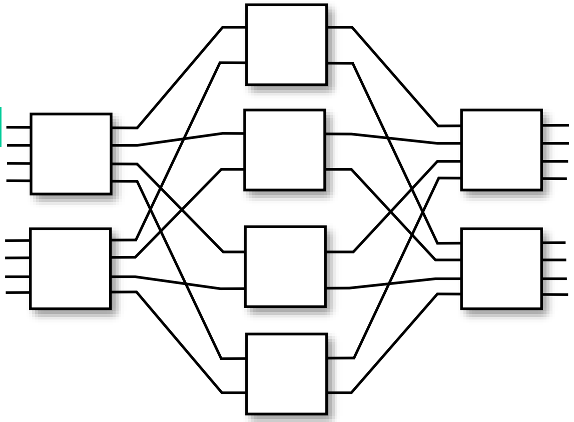


Switching via interconnection network

- Crossbar, Clos networks, other interconnection nets initially developed to connect processors in multiprocessor
- multistage switch: *nxn* switch from multiple stages of smaller switches
- exploiting parallelism:
  - fragment datagram into fixed length cells on entry
  - switch cells through the fabric, reassemble datagram at exit



3x3 crossbar

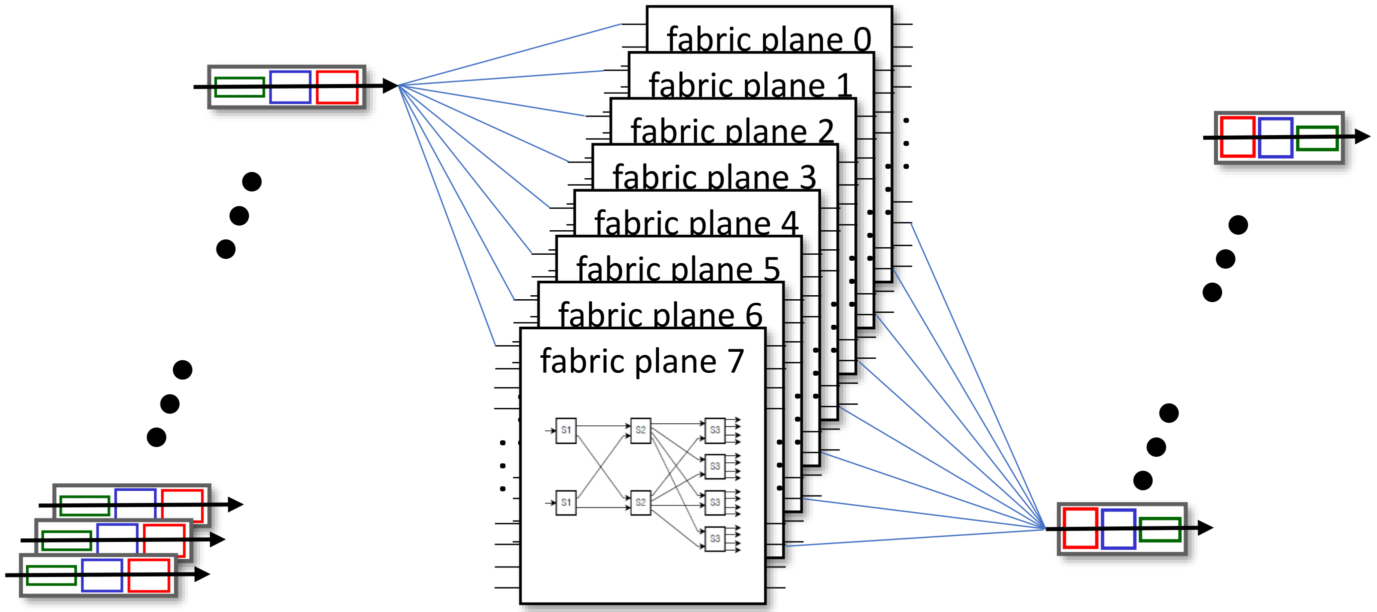


8x8 multistage switch
built from smaller-sized switches

Switching via interconnection network

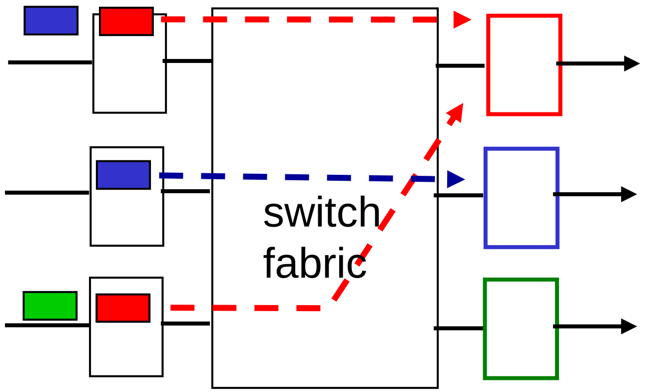
- scaling, using multiple switching “planes” in parallel:
  - speedup, scaleup via parallelism

- Cisco CRS router:
  - basic unit: 8 switching planes
  - each plane: 3-stage interconnection network
  - up to 100’s Tbps switching capacity

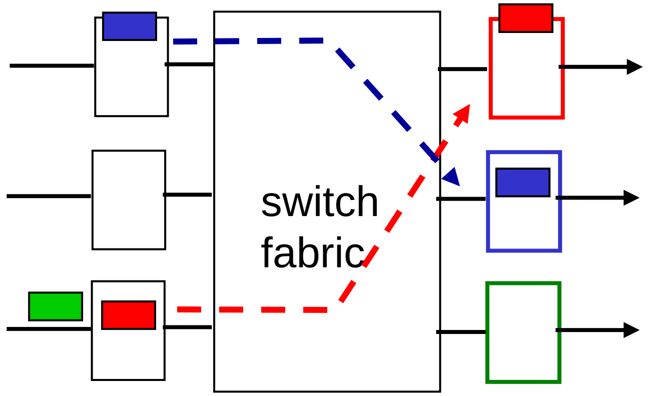


Input port queuing

- If switch fabric slower than input ports combined -> queueing may occur at input queues
  - queueing delay and loss due to input buffer overflow!
- Head-of-the-Line (HOL) blocking: queued datagram at front of queue prevents others in queue from moving forward

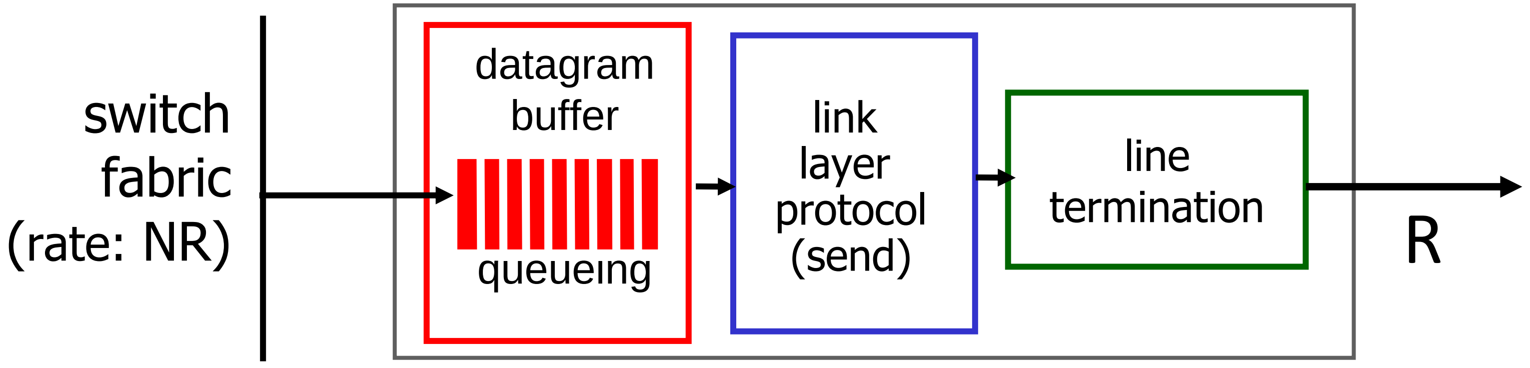


output port contention: only one red datagram can be transferred. lower red packet is *blocked*



one packet time later: green packet experiences HOL blocking

Output port queuing



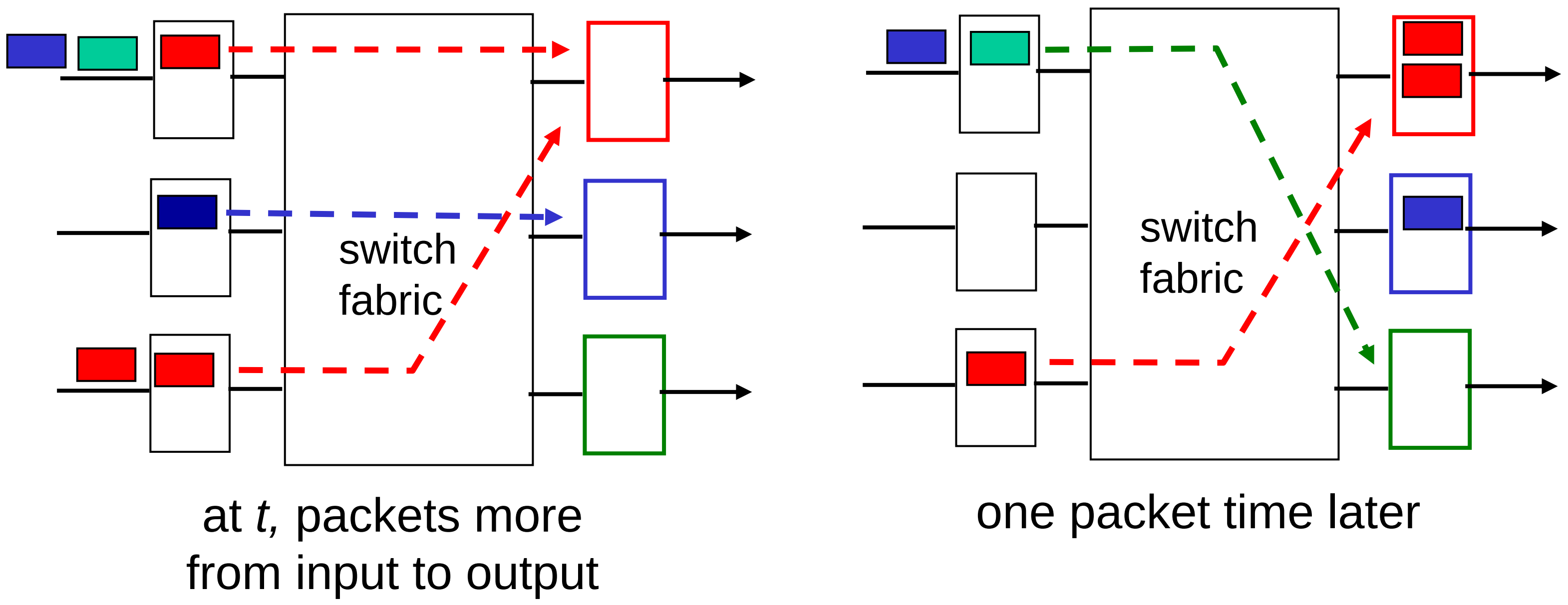
- *Buffering* required when datagrams arrive from fabric faster than link transmission rate. *Drop policy:* which datagrams to drop if no free buffers?

Datagrams can be lost due to congestion, lack of buffers

- *Scheduling discipline* chooses among queued datagrams for transmission

Priority scheduling – who gets best performance, network neutrality

Output port queuing



- buffering when arrival rate via switch exceeds output line speed
- *queueing (delay) and loss due to output port buffer overflow!*

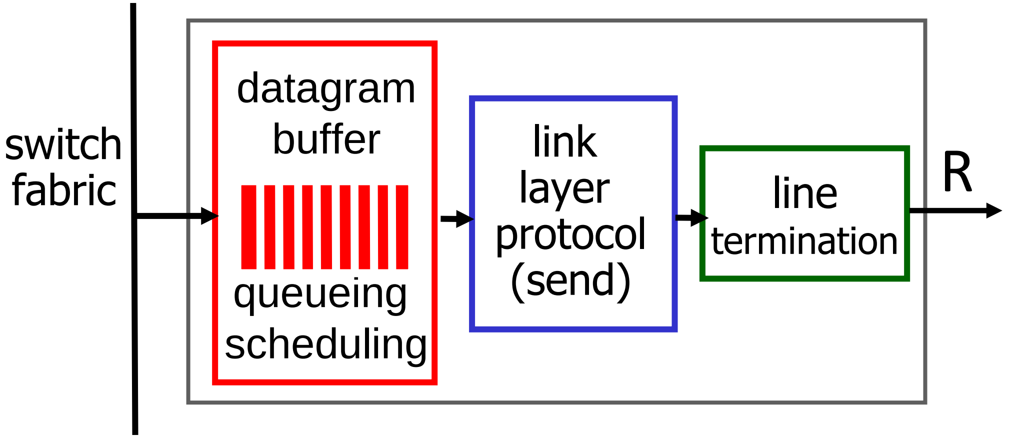
How much buffering?

- RFC 3439 rule of thumb: average buffering equal to "typical" RTT (say 250 msec) times link capacity C
  - e.g., C = 10 Gbps link: 2.5 Gbit buffer
- more recent recommendation: with *N* flows, buffering equal to

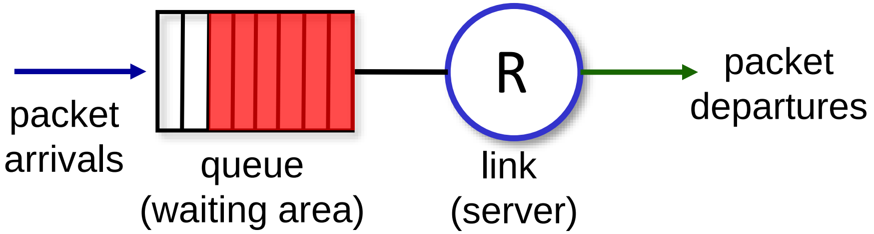
$$\frac{RTT \cdot C}{\sqrt{N}}$$

- but *too* much buffering can increase delays (particularly in home routers)
  - long RTTs: poor performance for real-time apps, sluggish TCP response
  - recall delay-based congestion control: "keep bottleneck link just full enough (busy) but no fuller"

Buffer Management



Abstraction: queue



buffer management:

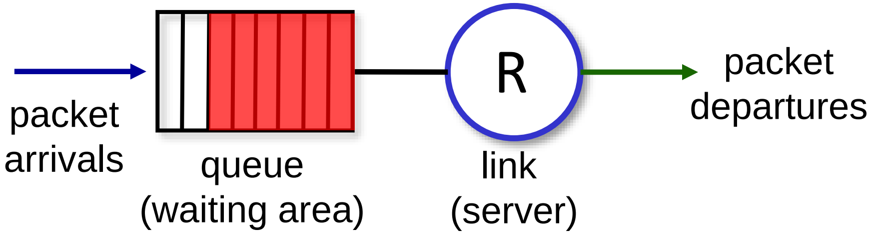
- drop: which packet to add, drop when buffers are full
  - tail drop: drop arriving packet
  - priority: drop/remove on priority basis
- marking: which packets to mark to signal congestion (ECN, RED)

Packet Scheduling: FCFS

packet scheduling: deciding which packet to send next on link

- first come, first served
- priority
- round robin
- weighted fair queueing

Abstraction: queue



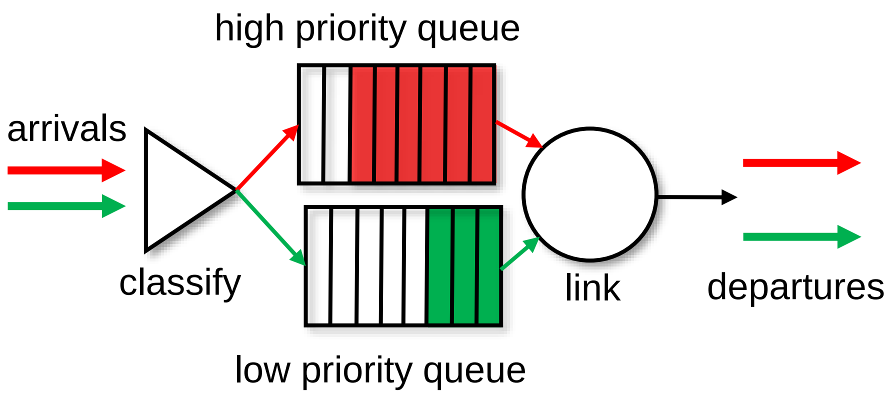
FCFS: packets transmitted in order of arrival to output port

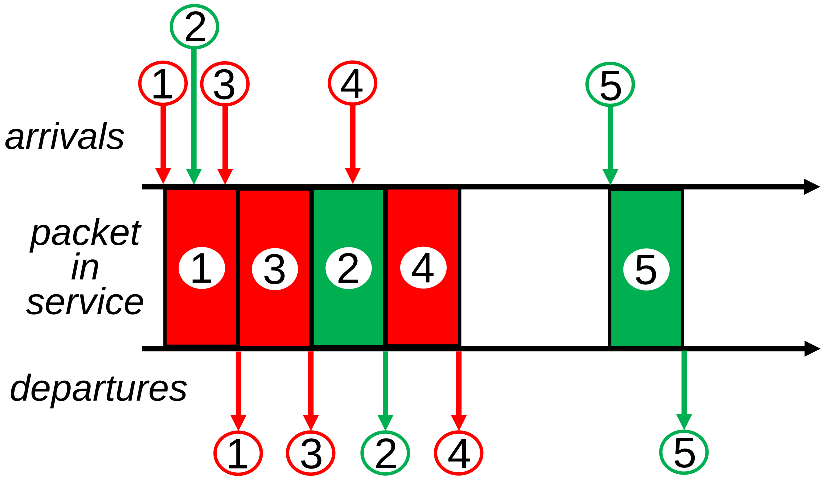
- also known as: First-in-first-out (FIFO)
- real world examples?

Scheduling policies: priority

Priority scheduling:

- arriving traffic classified, queued by class
  - any header fields can be used for classification
- send packet from highest priority queue that has buffered packets
  - FCFS within priority class

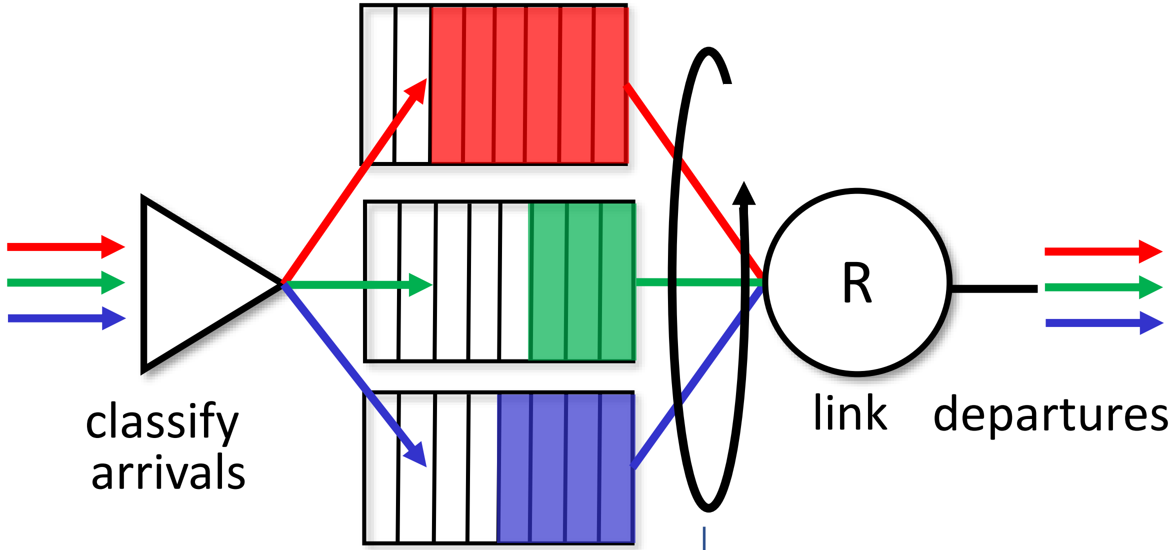




Scheduling policies: round robin

Round Robin (RR) scheduling:

- arriving traffic classified, queued by class
  - any header fields can be used for classification
- server cyclically, repeatedly scans class queues, sending one complete packet from each class (if available) in turn



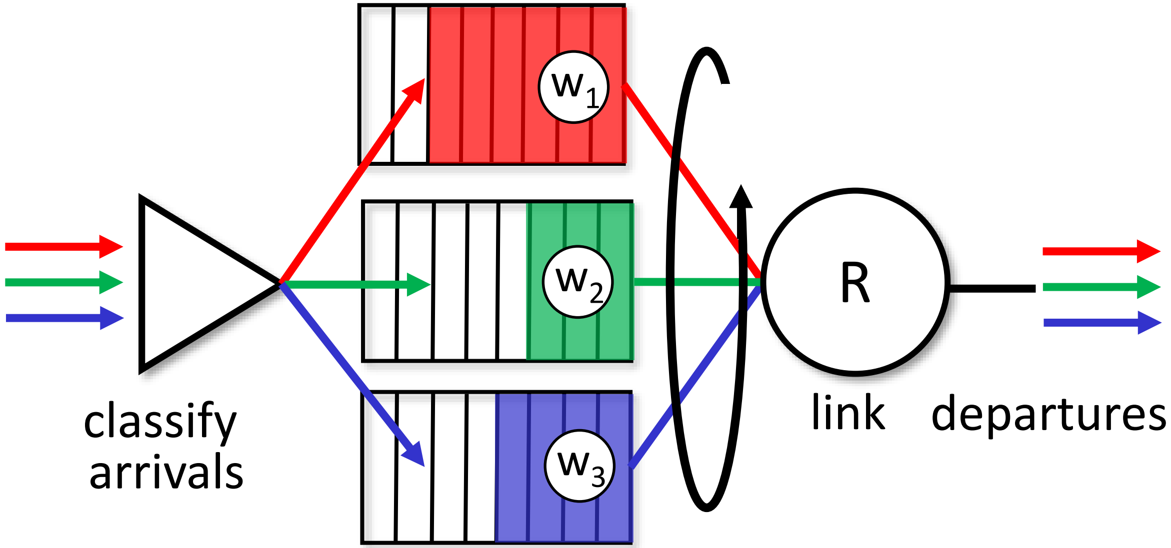
Scheduling policies: weighted fair queueing

Weighted Fair Queuing (WFQ):

- generalized Round Robin
- each class, i, has weight, w_i, and gets weighted amount of service in each cycle:

$$\frac{w_i}{\Sigma_j w_j}$$

- minimum bandwidth guarantee (per-traffic-class)



Sidebar: Network Neutrality

What is network neutrality?

- *technical:* how an ISP should share/allocation its resources
  - packet scheduling, buffer management are the *mechanisms*
- *social, economic* principles
  - protecting free speech
  - encouraging innovation, competition
- enforced *legal* rules and policies

Different countries have different "takes" on network neutrality

Sidebar: Network Neutrality

2015 US FCC *Order on Protecting and Promoting an Open Internet:* three "clear, bright line" rules:

- no blocking … "shall not block lawful content, applications, services, or non-harmful devices, subject to reasonable network management."
- no throttling … "shall not impair or degrade lawful Internet traffic on the basis of Internet content, application, or service, or use of a non-harmful device, subject to reasonable network management."
- no paid prioritization. … "shall not engage in paid prioritization"

ISP: telecommunications or information service?

Is an ISP a "telecommunications service" or an "information service" provider?

- the answer *really* matters from a regulatory standpoint!

US Telecommunication Act of 1934 and 1996:

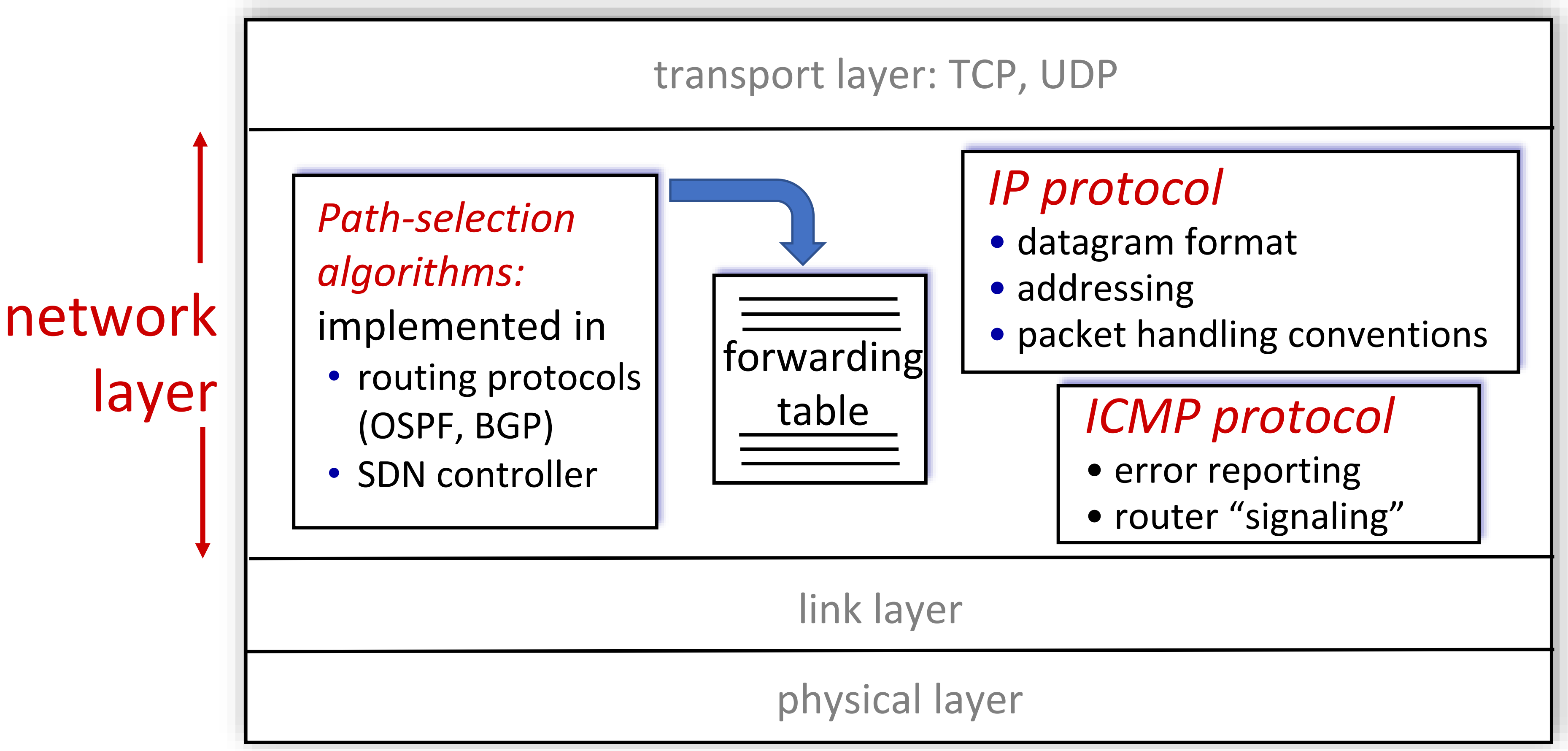
- *Title II:* imposes "common carrier duties" on *telecommunications services*: reasonable rates, non-discrimination and *requires regulation*
- *Title I:* applies to *information services:*
  - no common carrier duties (*not regulated*)
  - but grants FCC authority "… as may be necessary in the execution of its functions".[4]

Network layer: "data plane" roadmap

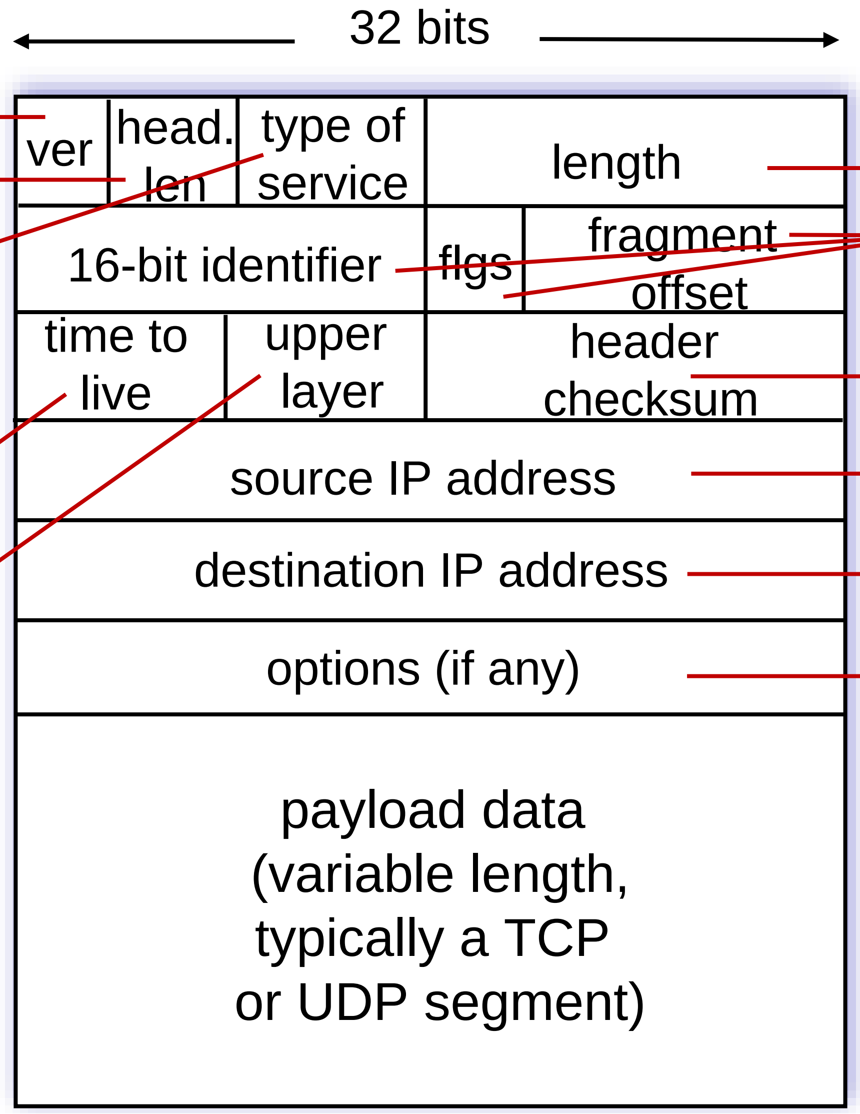
- Network layer: overview
  - data plane
  - control plane
- What's inside a router
  - input ports, switching, output ports
  - buffer management, scheduling
- IP: the Internet Protocol
  - datagram format
  - addressing
  - network address translation
  - IPv6
- Generalized Forwarding, SDN
  - match+action
  - OpenFlow: match+action in action
- Middleboxes

Network Layer: Internet

host, router network layer functions:



IP Datagram format



IP protocol version number

header length(bytes)

"type" of service:
- diffserv (0:5)
- ECN (6:7)

TTL: remaining max hops
(decremented at each router)

upper layer protocol (e.g., TCP or UDP)

total datagram length (bytes)

fragmentation/ reassembly

header checksum

32-bit source IP address

3

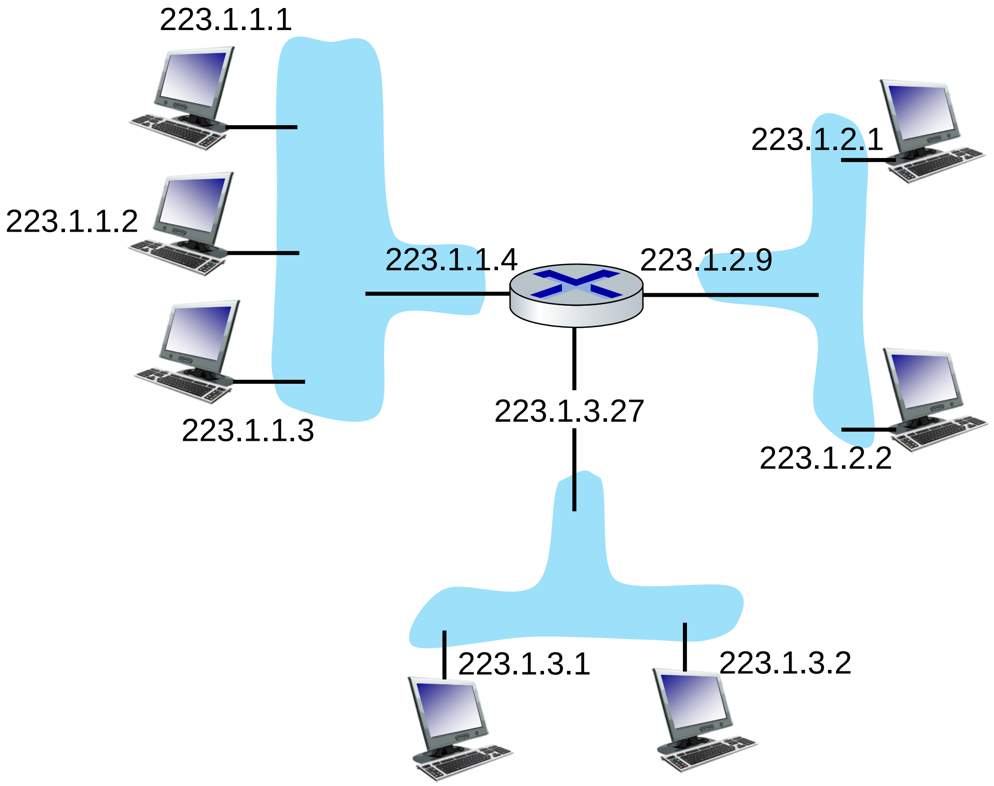
Maximum length: 64K bytes
Typically: 1500 bytes or less

e.g., timestamp, record route taken

overhead
- 20 bytes of TCP
- 20 bytes of IP
- = 40 bytes + app layer overhead for TCP+IP

IP addressing: introduction

- IP address: 32-bit identifier associated with each host or router *interface*
- interface: connection between host/router and physical link
  - router's typically have multiple interfaces
  - host typically has one or two interfaces (e.g., wired Ethernet, wireless 802.11)



dotted-decimal IP address notation:

$$223.1.1.1 = \underbrace{11011111}_{223}\ \underbrace{00000001}_{1}\ \underbrace{00000001}_{1}\ \underbrace{00000001}_{1}$$

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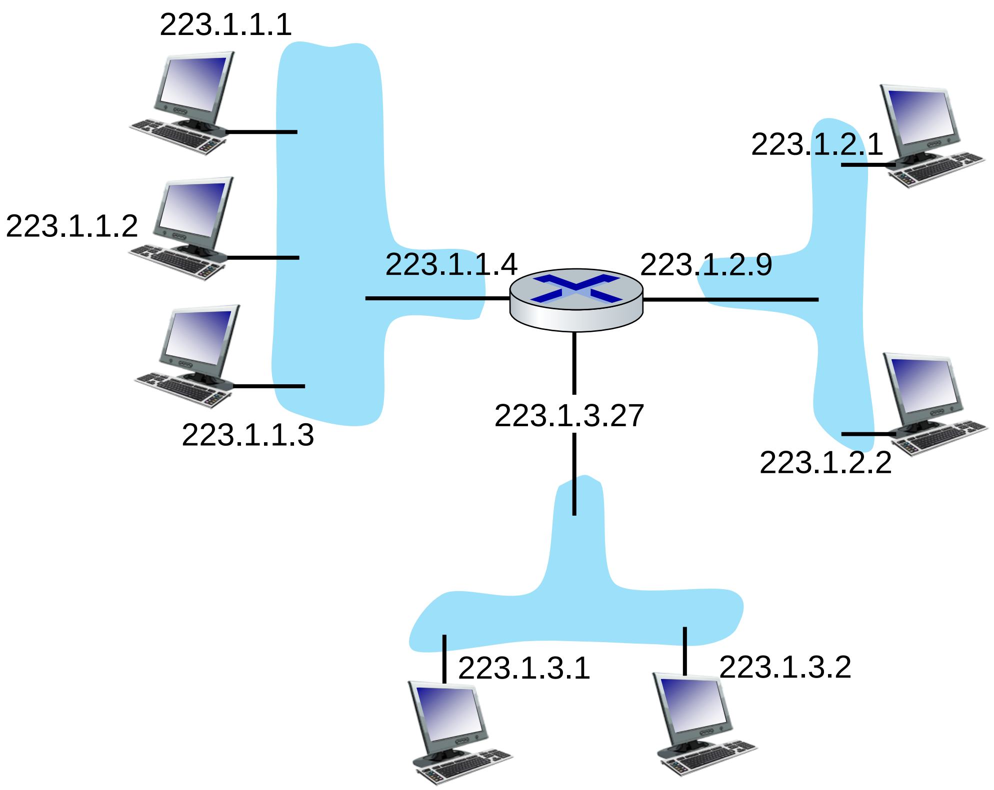


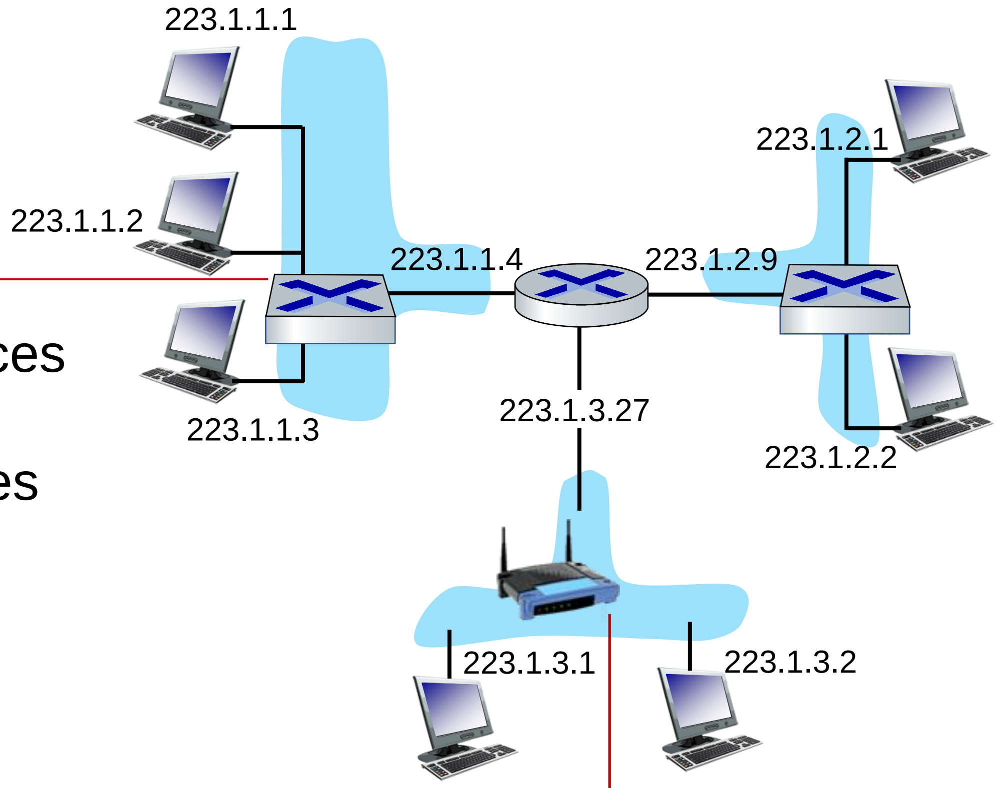
dotted-decimal IP address notation:

$$223.1.1.1 = \underbrace{11011111}_{223}\ \underbrace{00000001}_{1}\ \underbrace{00000001}_{1}\ \underbrace{00000001}_{1}$$

IP addressing: introduction

Q: how are interfaces actually connected?

A: we'll learn about that in chapters 6, 7



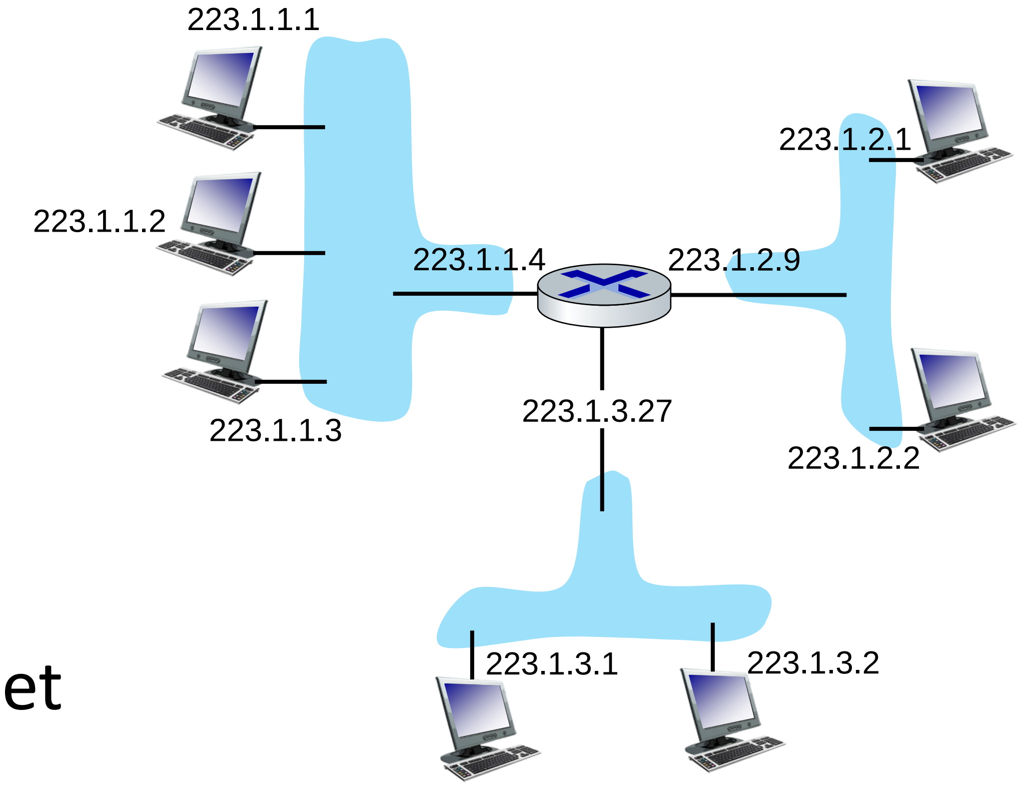
A: wired Ethernet interfaces connected by Ethernet switches

A: wireless WiFi interfaces connected by WiFi base station

For now: don't need to worry about how one interface is connected to another (with no intervening router)

Subnets

- *What's a subnet ?*
  - device interfaces that can physically reach each other without passing through an intervening router
- IP addresses have structure:
  - subnet part: devices in same subnet have common high order bits
  - host part: remaining low order bits

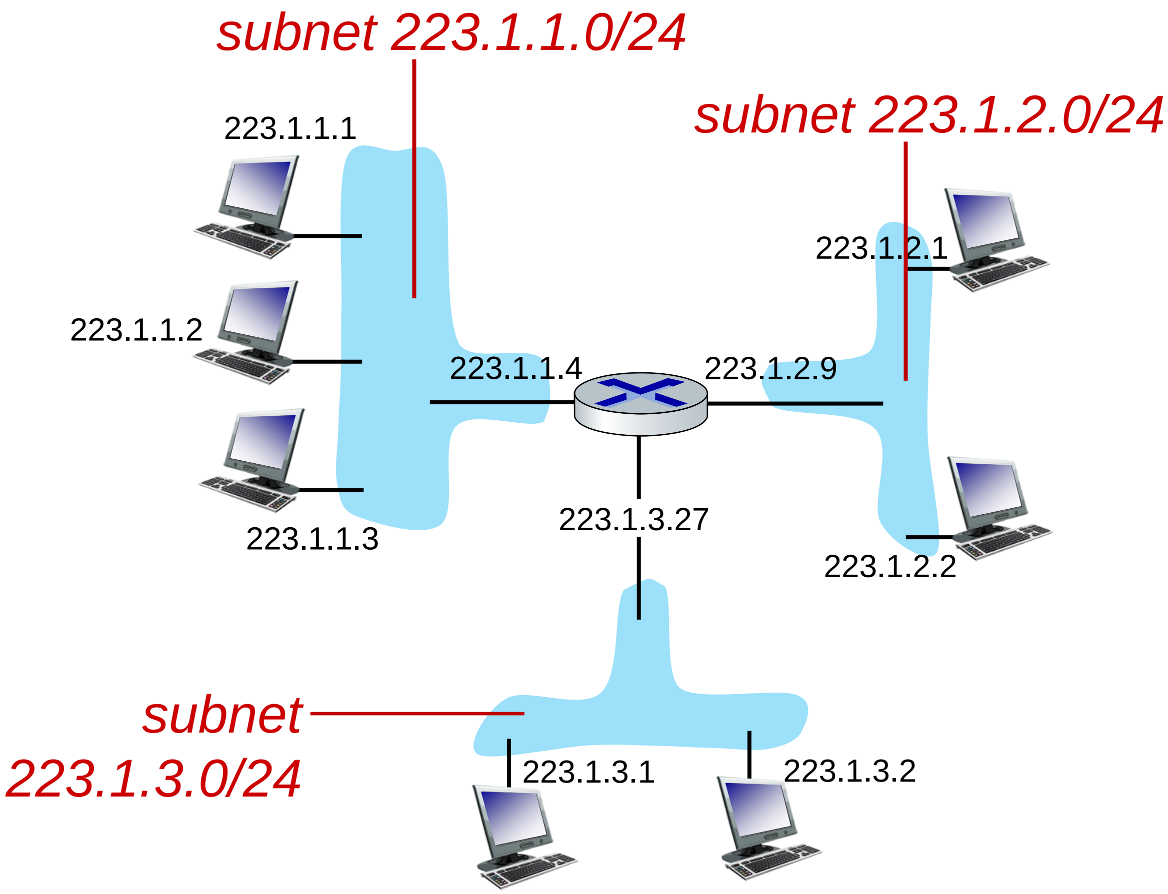


network consisting of 3 subnets

Subnets

Recipe for defining subnets:

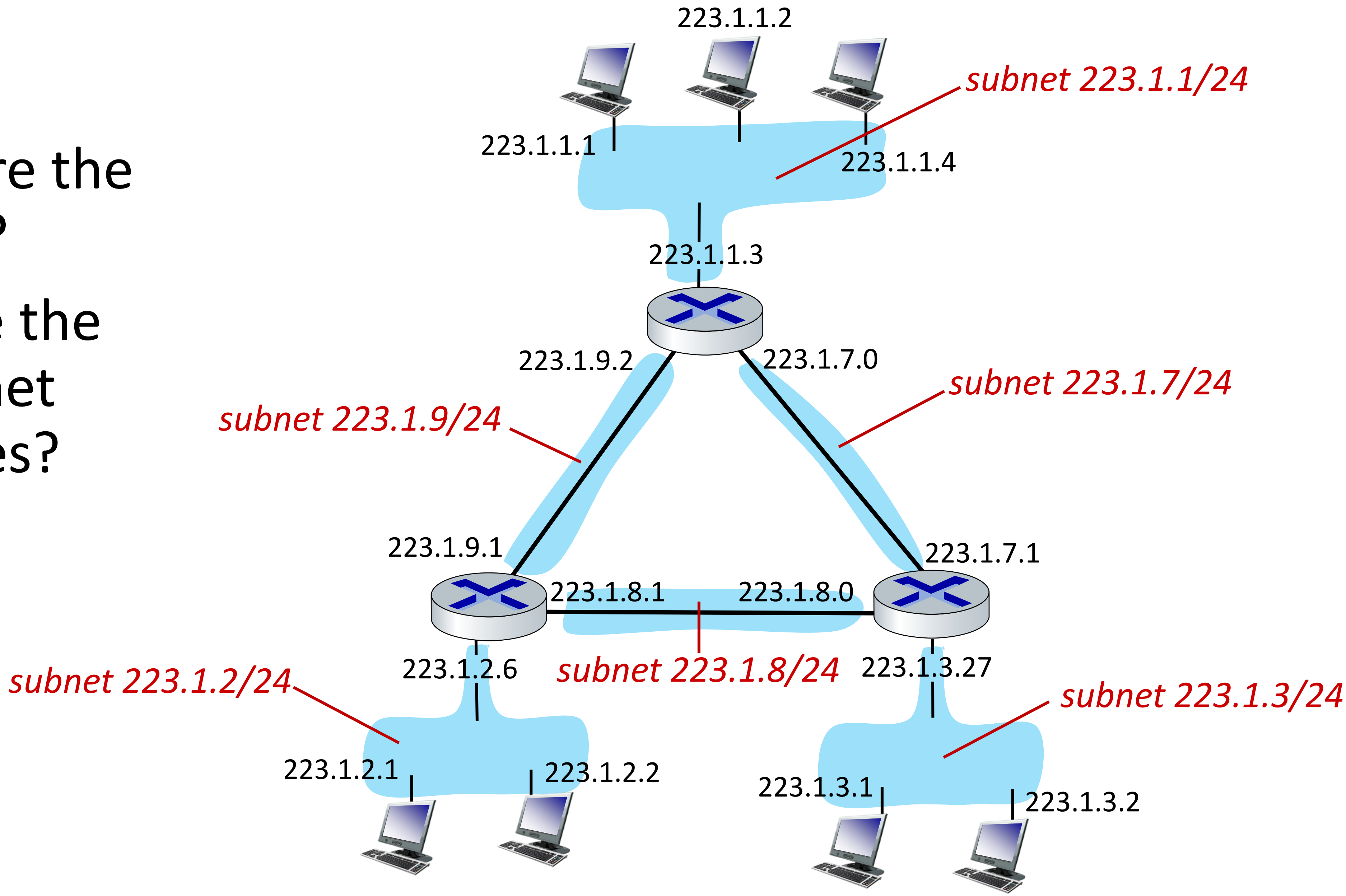
- detach each interface from its host or router, creating "islands" of isolated networks
- each isolated network is called a *subnet*



subnet mask: /24
(high-order 24 bits: subnet part of IP address)

Subnets

- where are the subnets?
- what are the /24 subnet addresses?



IP addressing: CIDR

CIDR: Classless InterDomain Routing (pronounced “cider”)

- subnet portion of address of arbitrary length
- address format: a.b.c.d/x, where x is # bits in subnet portion of address

subnet part | host part

11001000 00010111 00010000 00000000

200.23.16.0/23

IP addresses: how to get one?

That's actually two questions:

1. Q: How does a *host* get IP address within its network (host part of address)?
2. Q: How does a *network* get IP address for itself (network part of address)

How does *host* get IP address?

- hard-coded by sysadmin in config file (e.g., /etc/rc.config in UNIX)
- DHCP: Dynamic Host Configuration Protocol: dynamically get address from as server
  - "plug-and-play"

DHCP: Dynamic Host Configuration Protocol

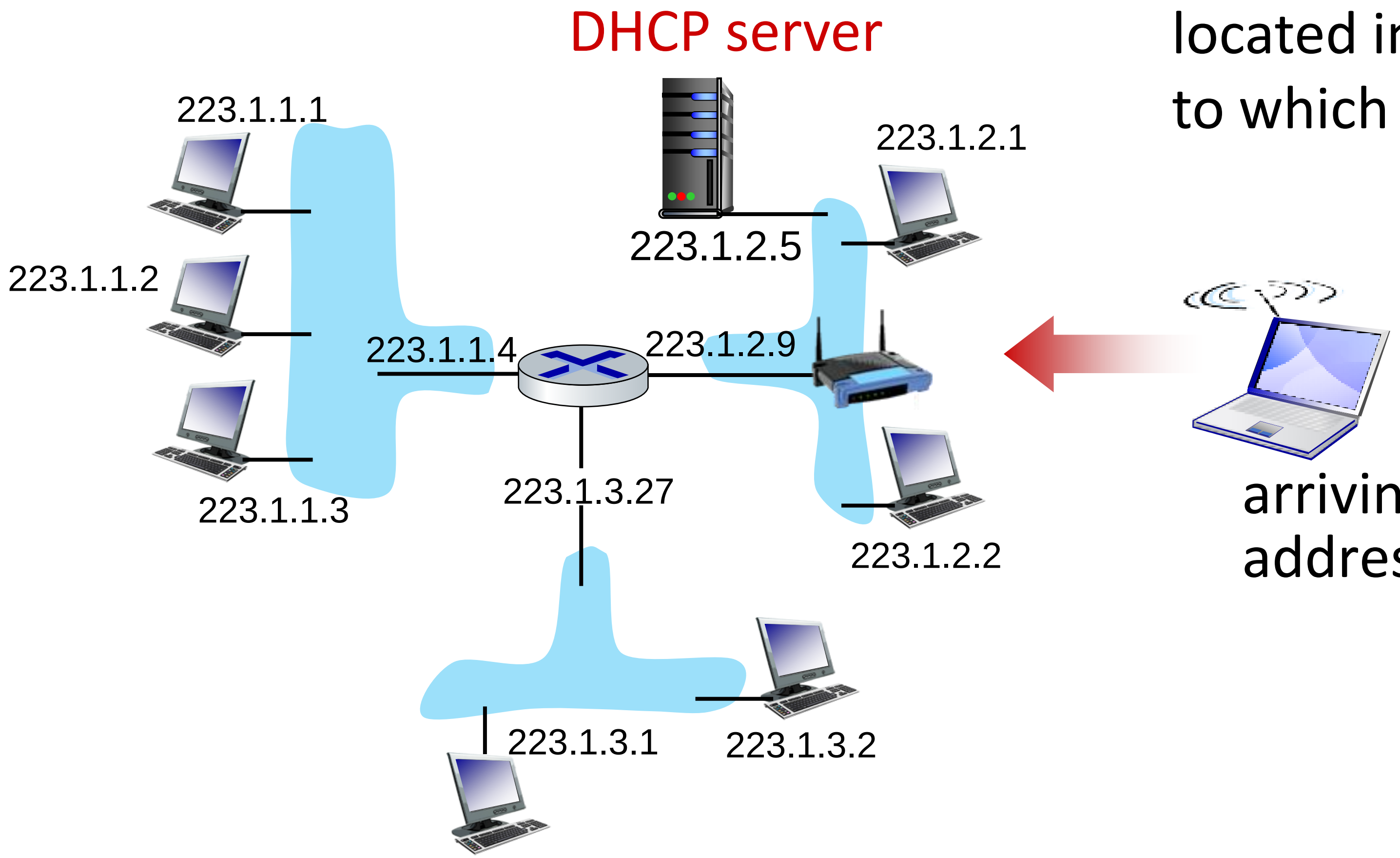
goal: host *dynamically* obtains IP address from network server when it "joins" network

- can renew its lease on address in use
- allows reuse of addresses (only hold address while connected/on)
- support for mobile users who join/leave network

DHCP overview:

- host broadcasts DHCP discover msg [optional]
- DHCP server responds with DHCP offer msg [optional]
- host requests IP address: DHCP request msg
- DHCP server sends address: DHCP ack msg

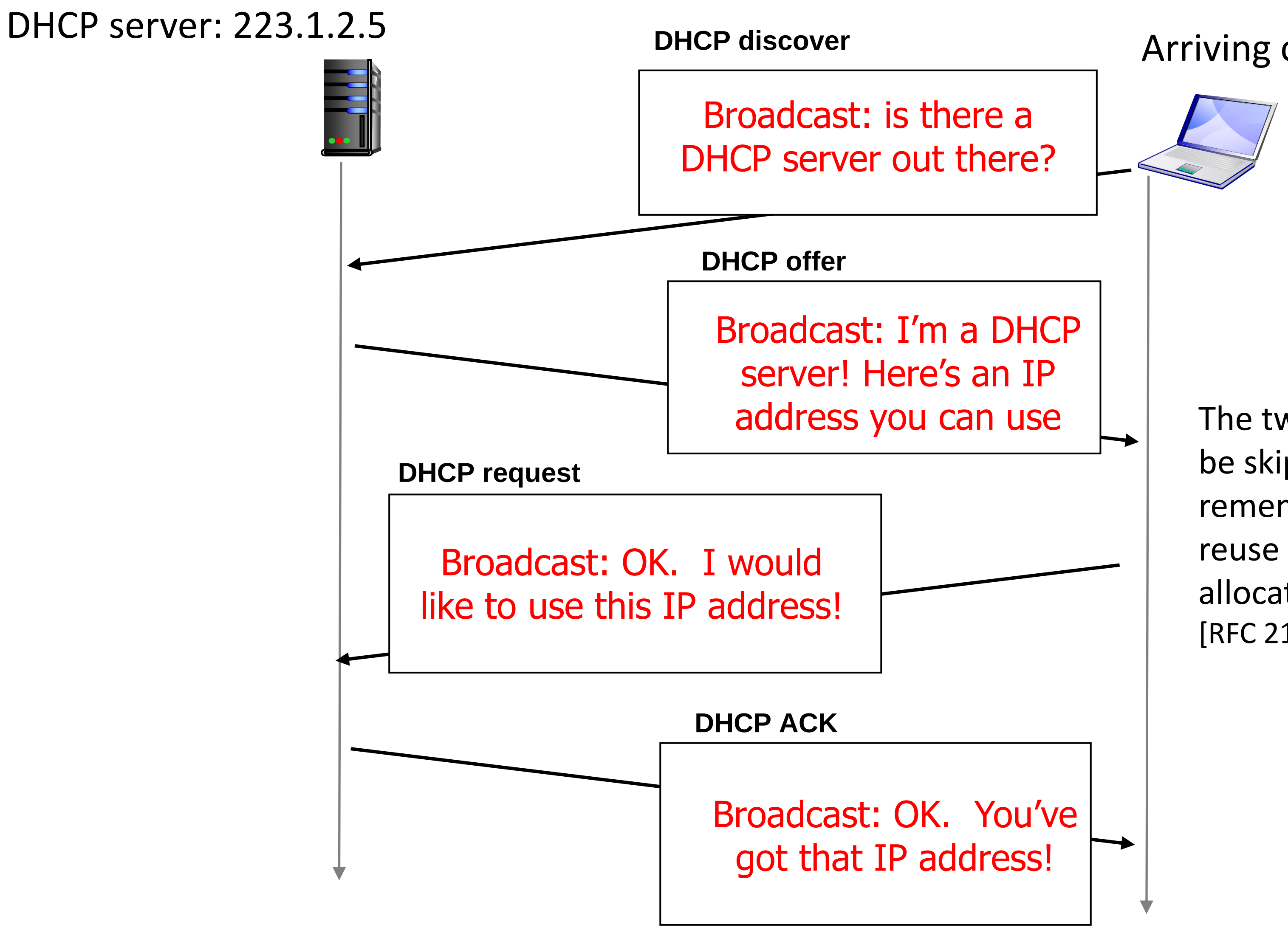
DHCP client-server scenario



Typically, DHCP server will be co-located in router, serving all subnets to which router is attached

arriving DHCP client needs address in this network

DHCP client-server scenario



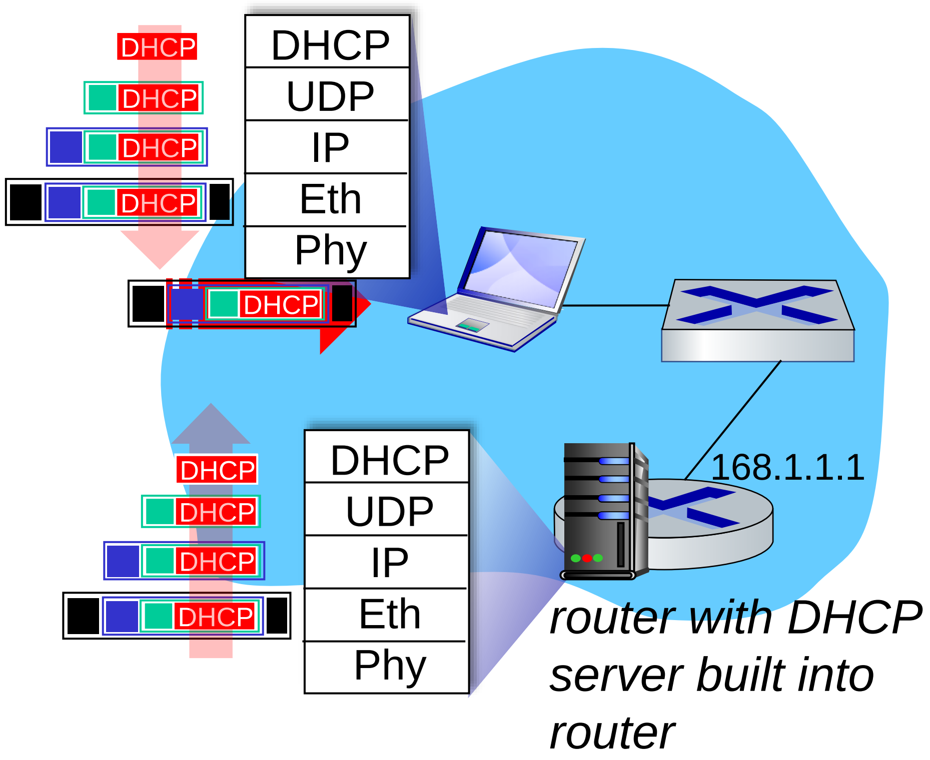
The two steps above can be skipped "if a client remembers and wishes to reuse a previously allocated network address" [RFC 2131]

DHCP: more than IP addresses

DHCP can return more than just allocated IP address on subnet:

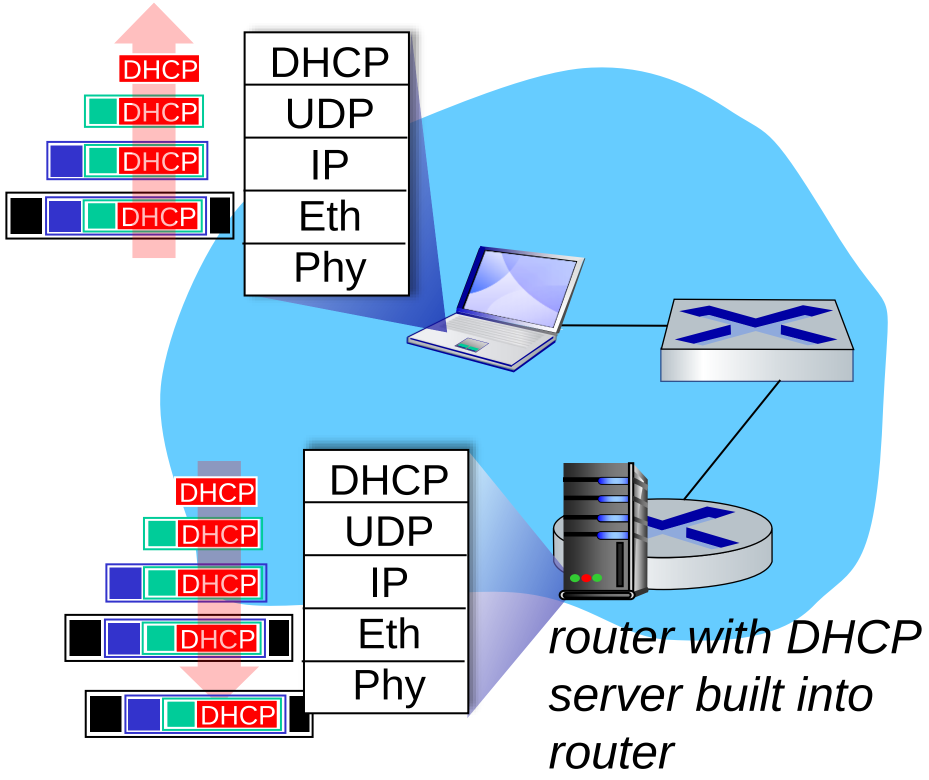
- address of first-hop router for client
- name and IP address of DNS sever
- network mask (indicating network versus host portion of address)

DHCP: example



- Connecting laptop will use DHCP to get IP address, address of first-hop router, address of DNS server.
- DHCP REQUEST message encapsulated in UDP, encapsulated in IP, encapsulated in Ethernet
- Ethernet frame broadcast (dest: FFFFFFFFFFFF) on LAN, received at router running DHCP server
- Ethernet de-mux'ed to IP de-mux'ed, UDP de-mux'ed to DHCP

DHCP: example



- DCP server formulates DHCP ACK containing client's IP address, IP address of first-hop router for client, name & IP address of DNS server
- encapsulated DHCP server reply forwarded to client, de-muxing up to DHCP at client
- client now knows its IP address, name and IP address of DNS server, IP address of its first-hop router

IP addresses: how to get one?

Q: how does *network* get subnet part of IP address?

A: gets allocated portion of its provider ISP's address space

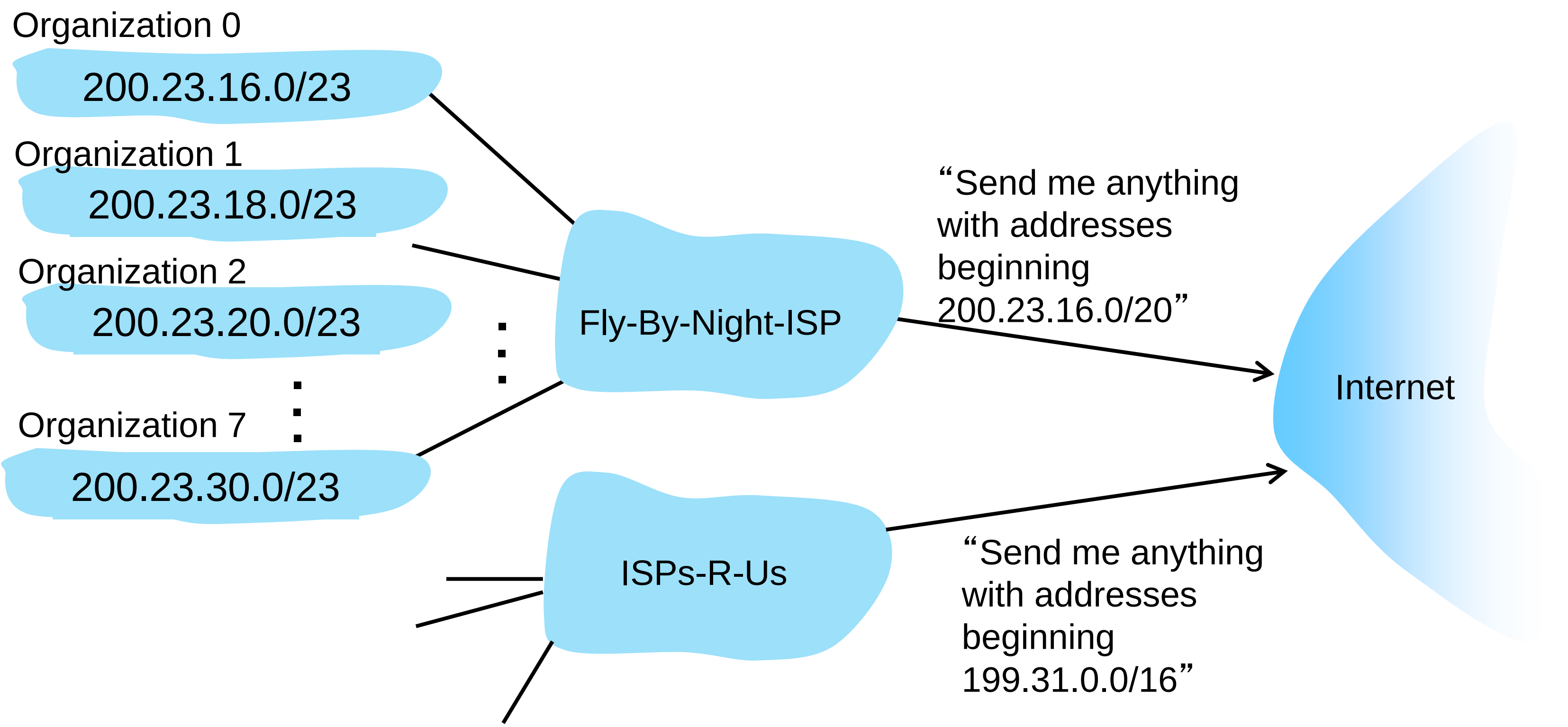
ISP's block	11001000 00010111 00010000 00000000	200.23.16.0/20

ISP can then allocate out its address space in 8 blocks:

Organization 0	11001000 00010111 00010000 00000000	200.23.16.0/23
Organization 1	11001000 00010111 00010010 00000000	200.23.18.0/23
Organization 2	11001000 00010111 00010100 00000000	200.23.20.0/23
...	....	....
Organization 7	11001000 00010111 00011110 00000000	200.23.30.0/23

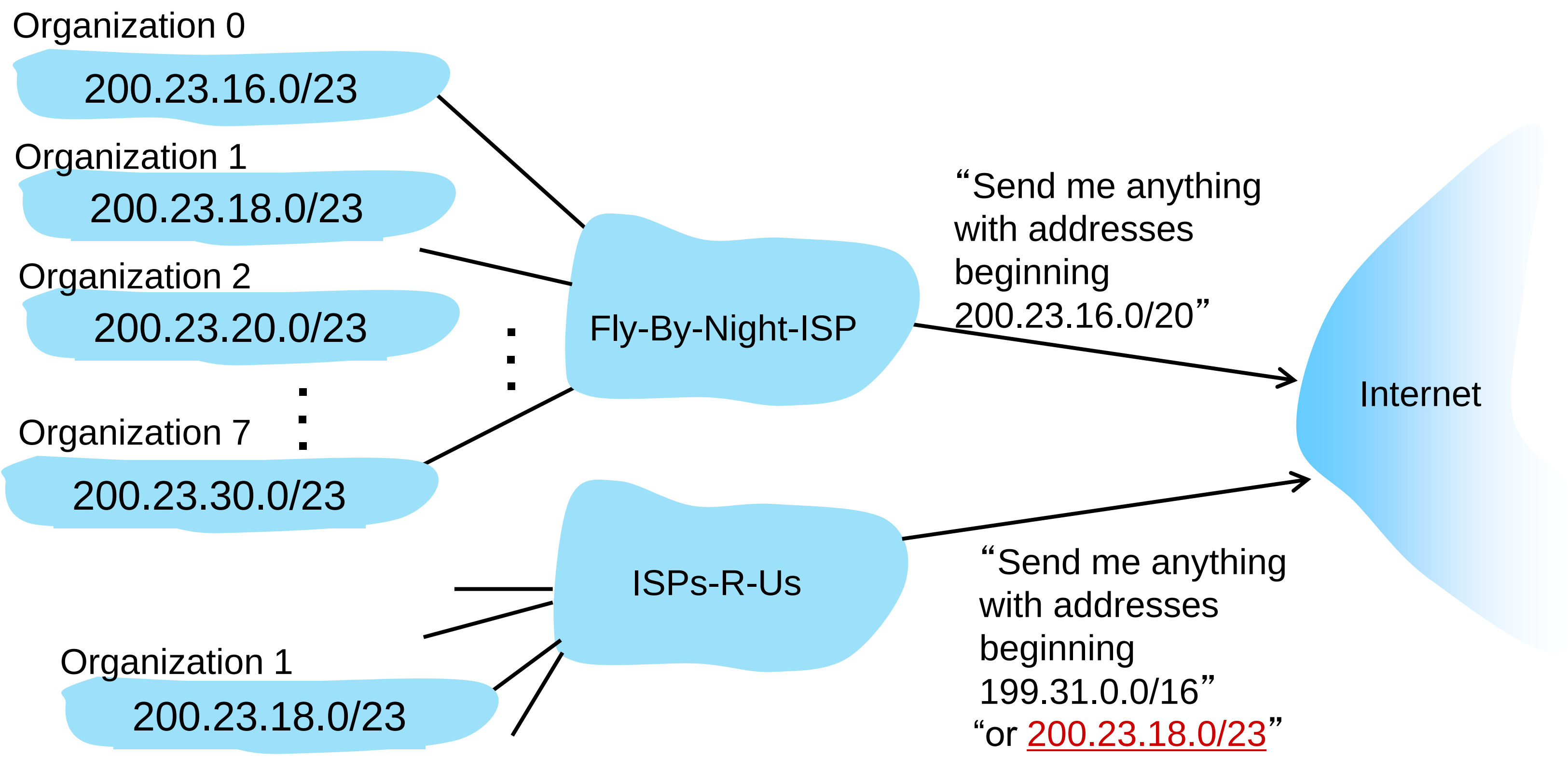
Hierarchical addressing: route aggregation

hierarchical addressing allows efficient advertisement of routing information:



Hierarchical addressing: more specific routes

- Organization 1 moves from Fly-By-Night-ISP to ISPs-R-Us
- ISPs-R-Us now advertises a more specific route to Organization 1



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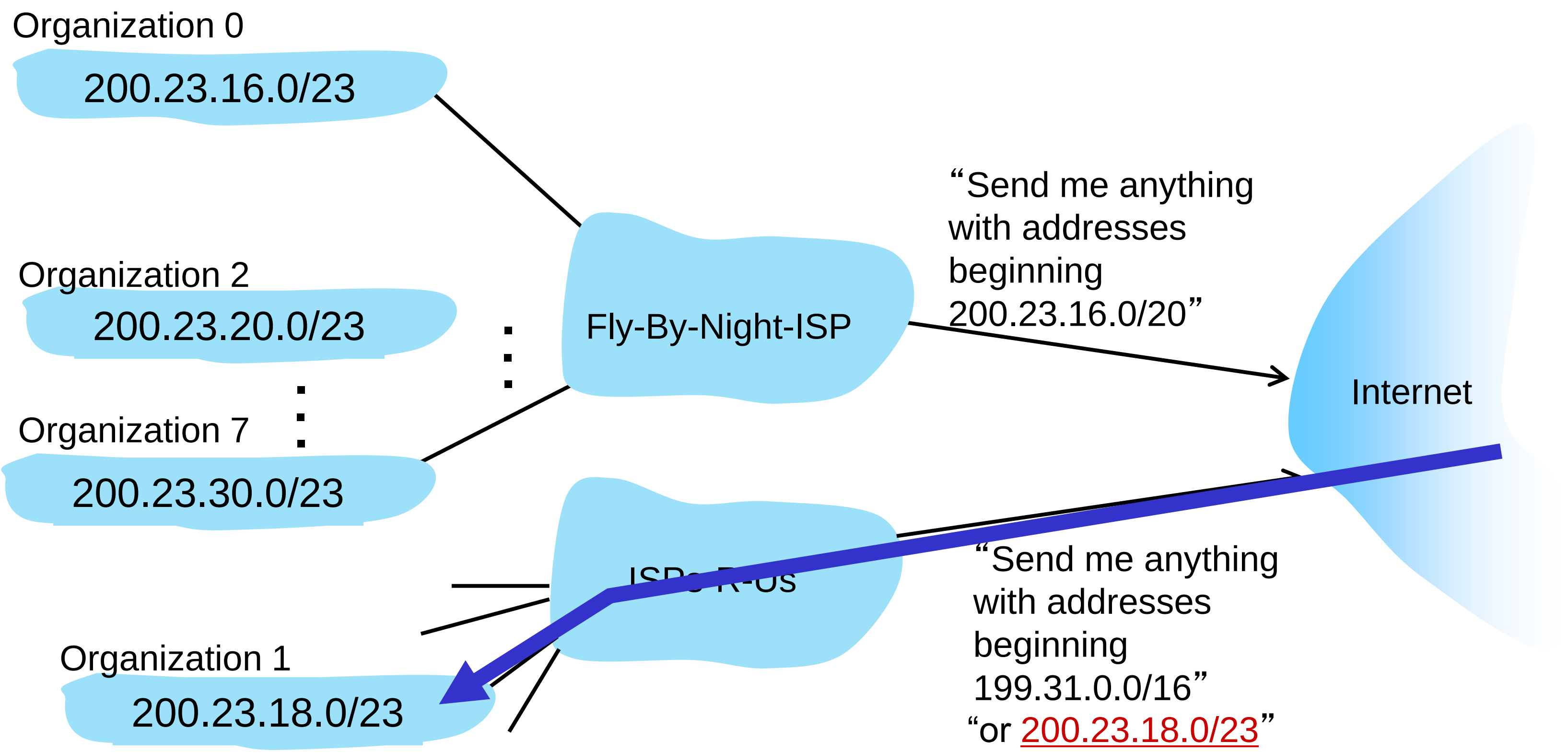

IP addressing: last words ...

Q: how does an ISP get block of addresses?

A: ICANN: Internet Corporation for Assigned Names and Numbers
http://www.icann.org/

- allocates IP addresses, through 5 regional registries (RRs) (who may then allocate to local registries)
- manages DNS root zone, including delegation of individual TLD (.com, .edu , ...) management

Q: are there enough 32-bit IP addresses?

- ICANN allocated last chunk of IPv4 addresses to RRs in 2011
- NAT (next) helps IPv4 address space exhaustion
- IPv6 has 128-bit address space

"Who the hell knew how much address space we needed?" Vint Cerf (reflecting on decision to make IPv4 address 32 bits long)

Network layer: "data plane" roadmap

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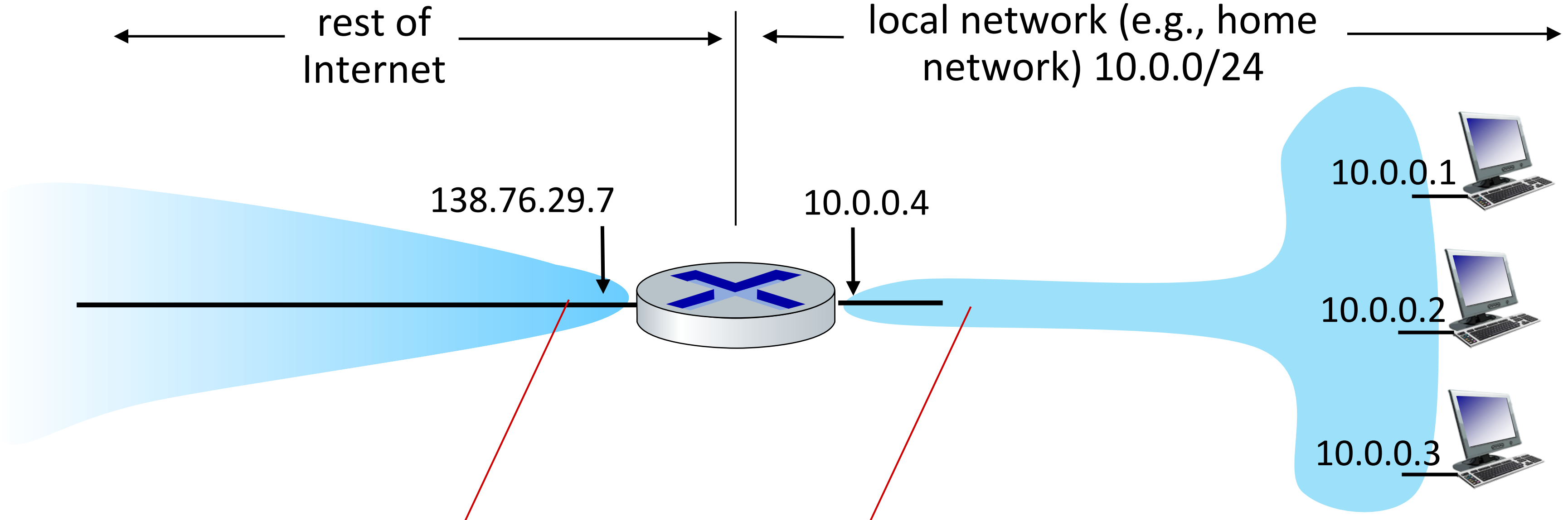
  - IPv6
- Generalized Forwarding, SDN
  - match+action
  - OpenFlow: match+action in action
- Middleboxes

NAT: network address translation

NAT: all devices in local network share just one IPv4 address as far as outside world is concerned



all datagrams *leaving* local network have *same* source NAT IP address: 138.76.29.7, but *different* source port numbers

datagrams with source or destination in this network have 10.0.0/24 address for source, destination (as usual)

NAT: network address translation

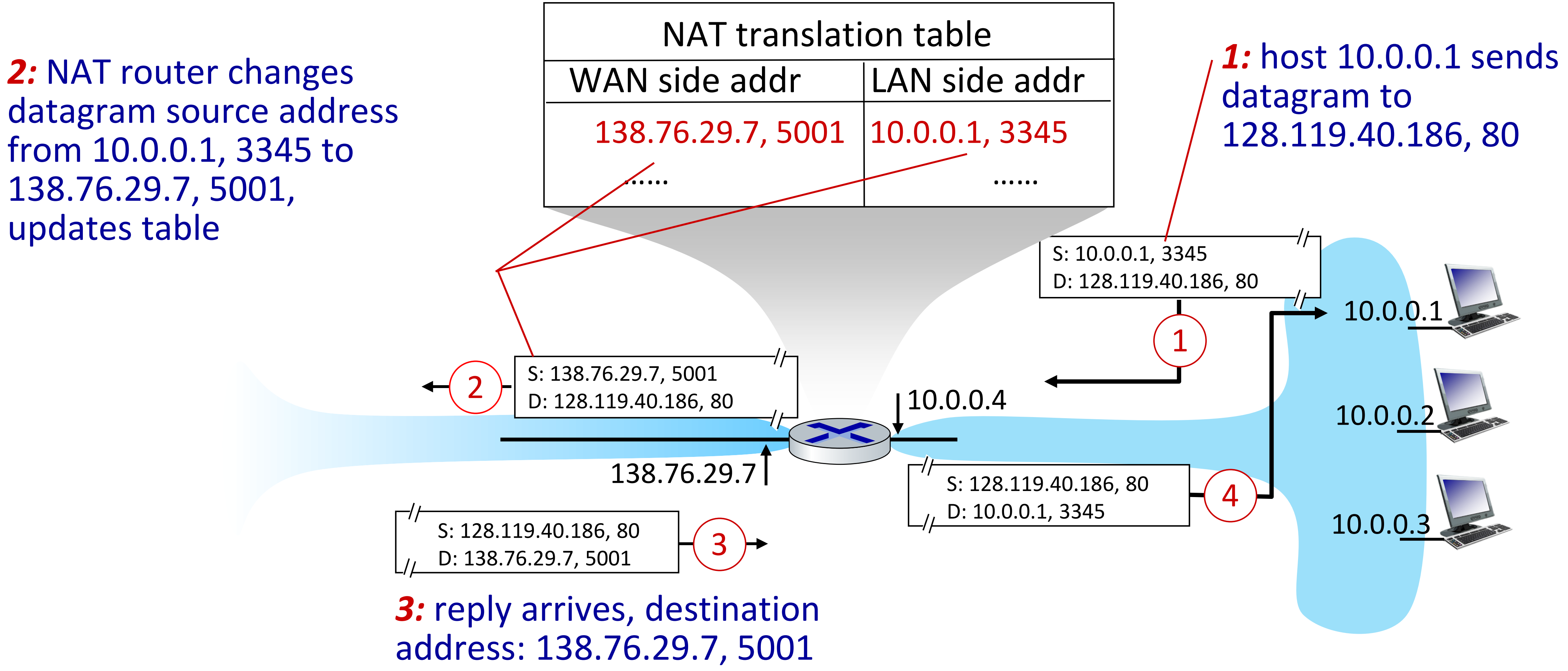
- all devices in local network have 32-bit addresses in a "private" IP address space (10/8, 172.16/12, 192.168/16 prefixes) that can only be used in local network
- advantages:
  - just one IP address needed from provider ISP for *all* devices
  - can change addresses of host in local network without notifying outside world
  - can change ISP without changing addresses of devices in local network
  - security: devices inside local net not directly addressable, visible by outside world

NAT: network address translation

implementation: NAT router must (transparently):

- outgoing datagrams: replace (source IP address, port #) of every outgoing datagram to (NAT IP address, new port #)
  - remote clients/servers will respond using (NAT IP address, new port #) as destination address
- remember (in NAT translation table) every (source IP address, port #) to (NAT IP address, new port #) translation pair
- incoming datagrams: replace (NAT IP address, new port #) in destination fields of every incoming datagram with corresponding (source IP address, port #) stored in NAT table

NAT: network address translation



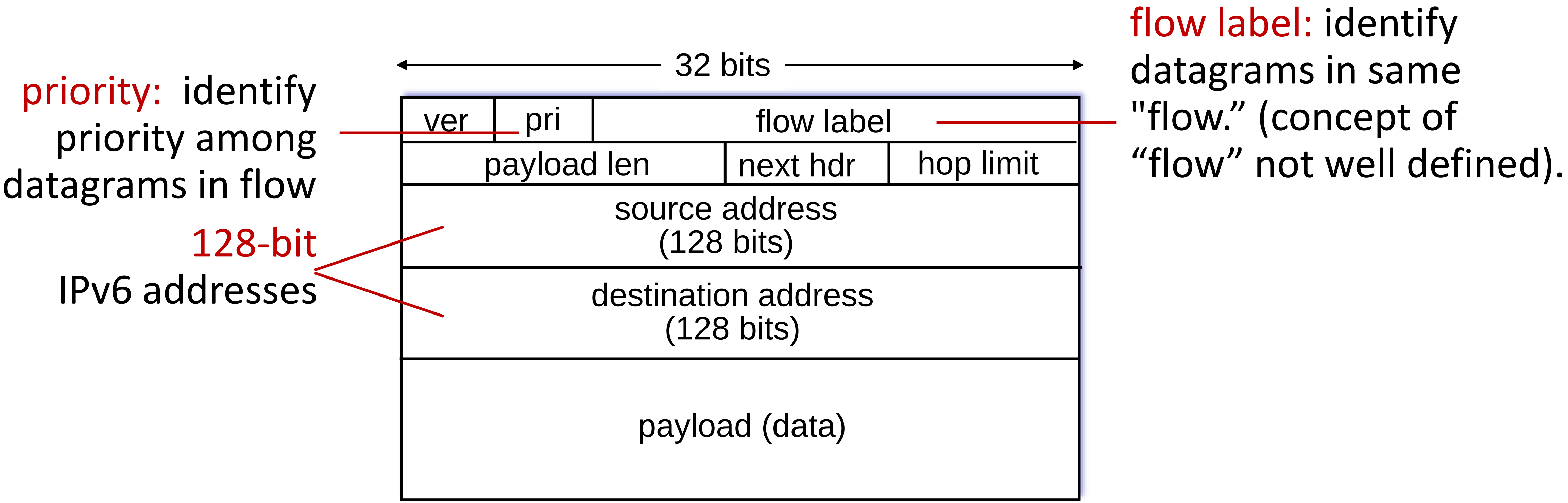
NAT: network address translation

- NAT has been controversial:
  - routers "should" only process up to layer 3
  - address "shortage" should be solved by IPv6
  - violates end-to-end argument (port # manipulation by network-layer device)
  - NAT traversal: what if client wants to connect to server behind NAT?
- but NAT is here to stay:
  - extensively used in home and institutional nets, 4G/5G cellular nets

IPv6: motivation

- initial motivation: 32-bit IPv4 address space would be completely allocated
- additional motivation:
  - speed processing/forwarding: 40-byte fixed length header
  - enable different network-layer treatment of “flows”

IPv6 datagram format



What's missing (compared with IPv4):

- no checksum (to speed processing at routers)
- no fragmentation/reassembly
- no options (available as upper-layer, next-header protocol at router)

Chapter 5
Network Layer:
Control Plane

Network layer control plane: our goals

- understand principles behind network control plane:
  - traditional routing algorithms
  - SDN controllers
  - network management, configuration

- instantiation, implementation in the Internet:
  - OSPF, BGP
  - OpenFlow, ODL and ONOS controllers
  - Internet Control Message Protocol: ICMP
  - SNMP, YANG/NETCONF

Network layer: “control plane” roadmap

- introduction
- routing protocols
  - link state
  - distance vector
- intra-ISP routing: OSPF
- routing among ISPs: BGP
- SDN control plane
- Internet Control Message Protocol
- network management, configuration
  - SNMP
  - NETCONF/YANG

Network-layer functions

- forwarding: move packets from router's input to appropriate router output

data plane

- routing: determine route taken by packets from source to destination

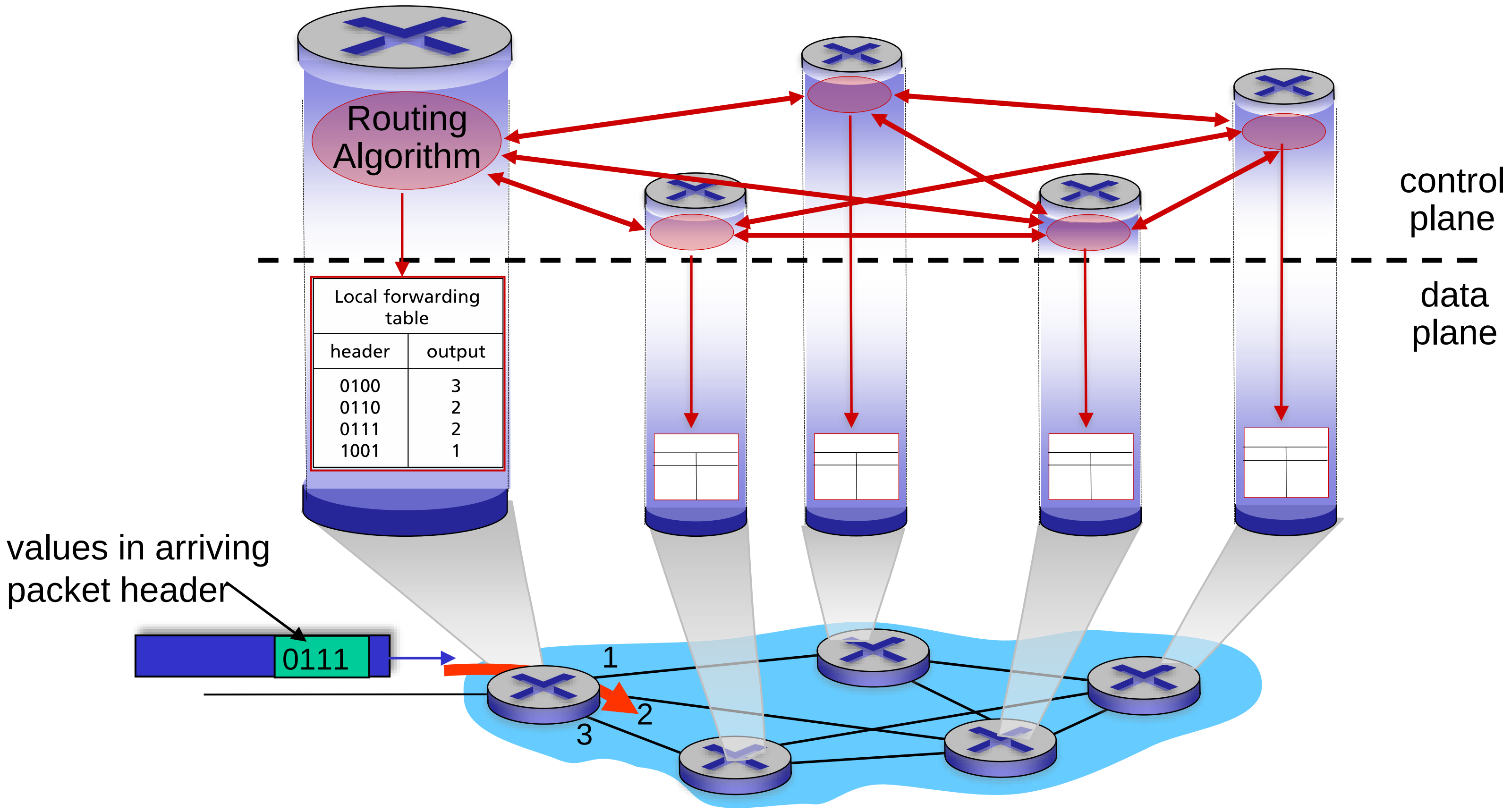
control plane

Two approaches to structuring network control plane:

- per-router control (traditional)
- logically centralized control (software defined networking)

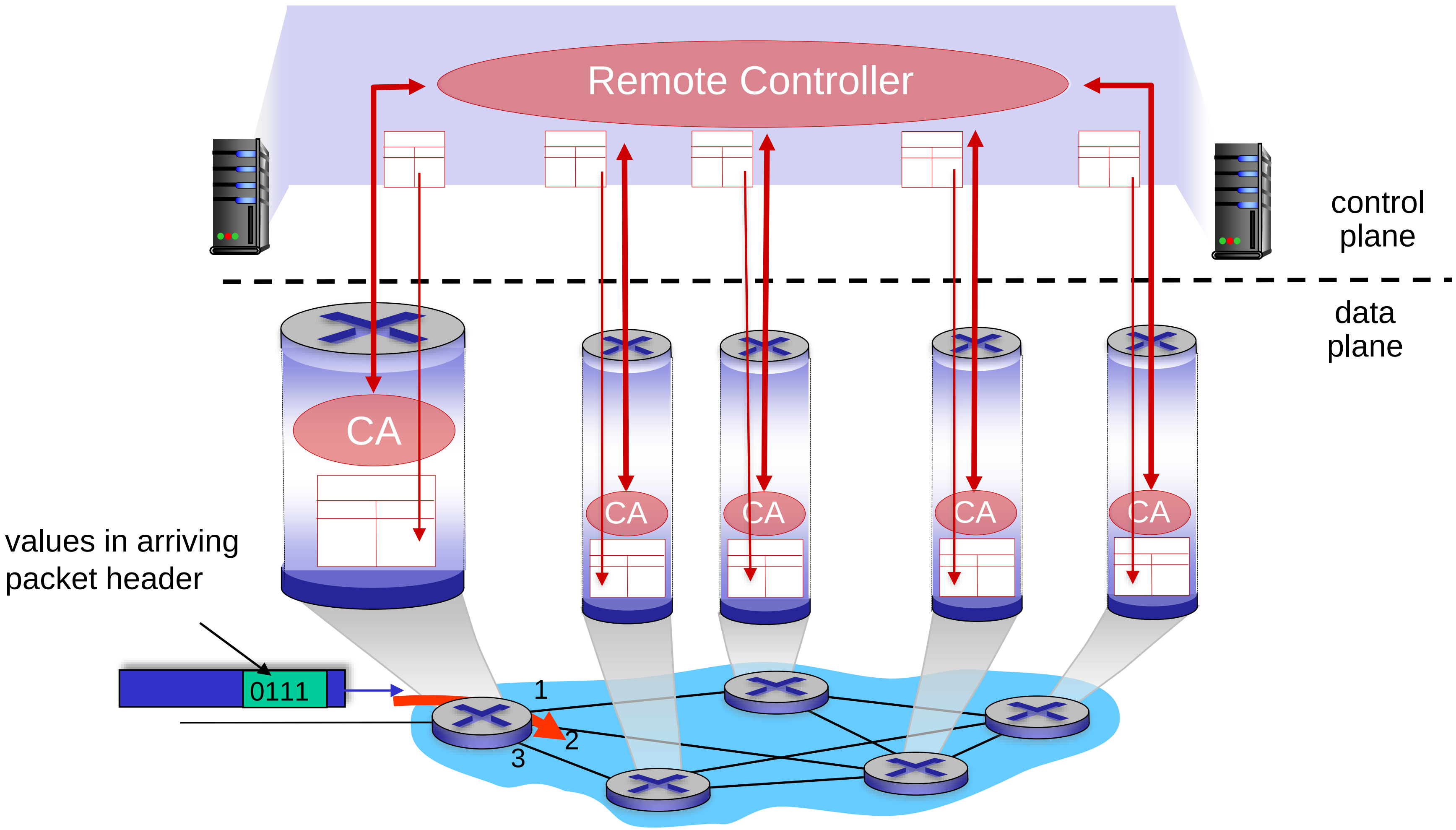
Per-router control plane

Individual routing algorithm components *in each and every router* interact in the control plane

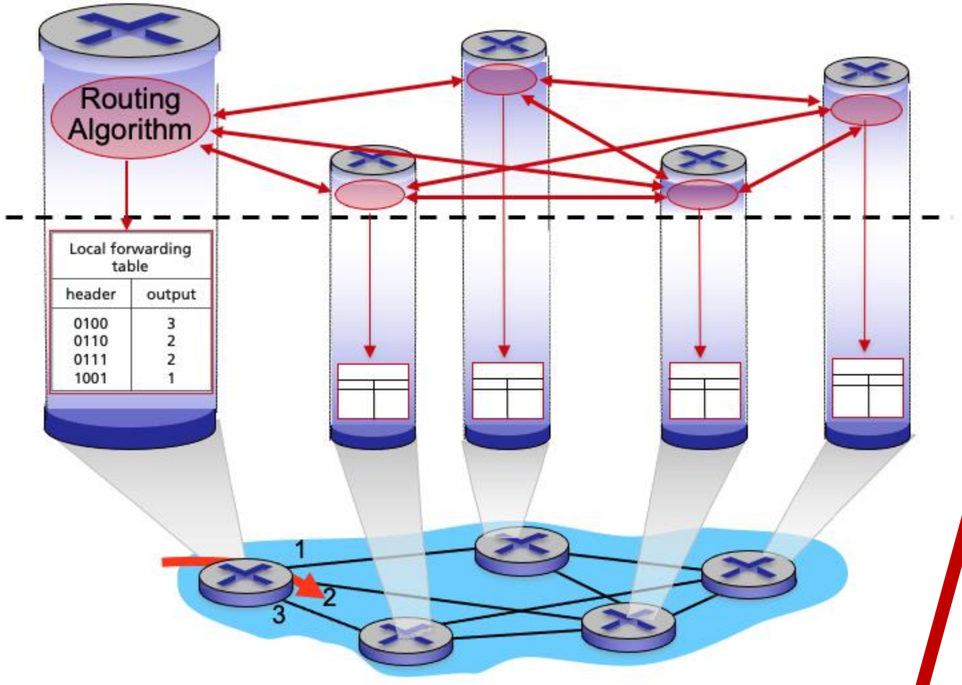


Software-Defined Networking (SDN) control plane

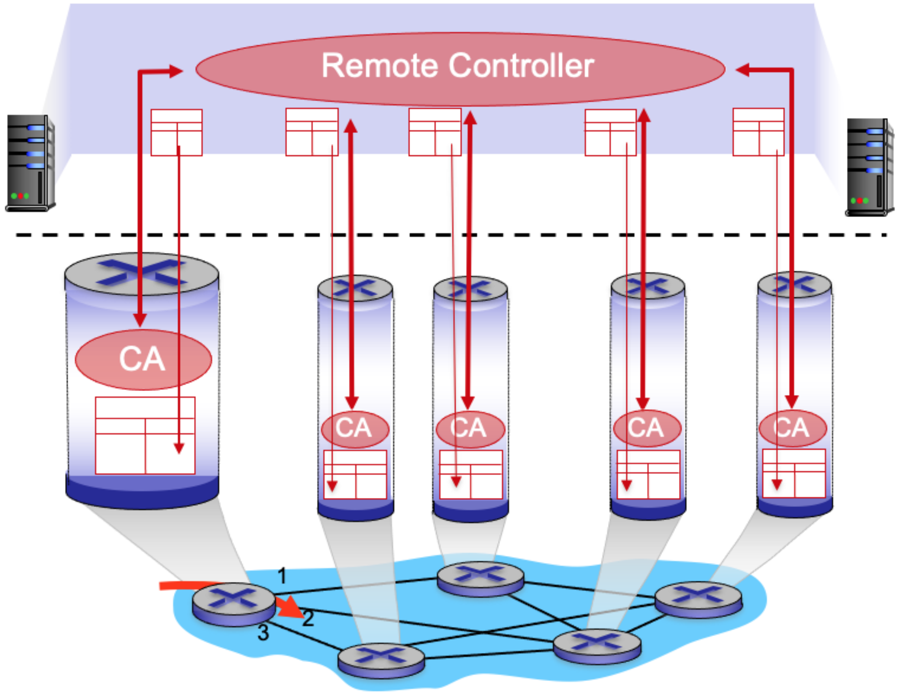
Remote controller computes, installs forwarding tables in routers



Per-router control plane



SDN control plane



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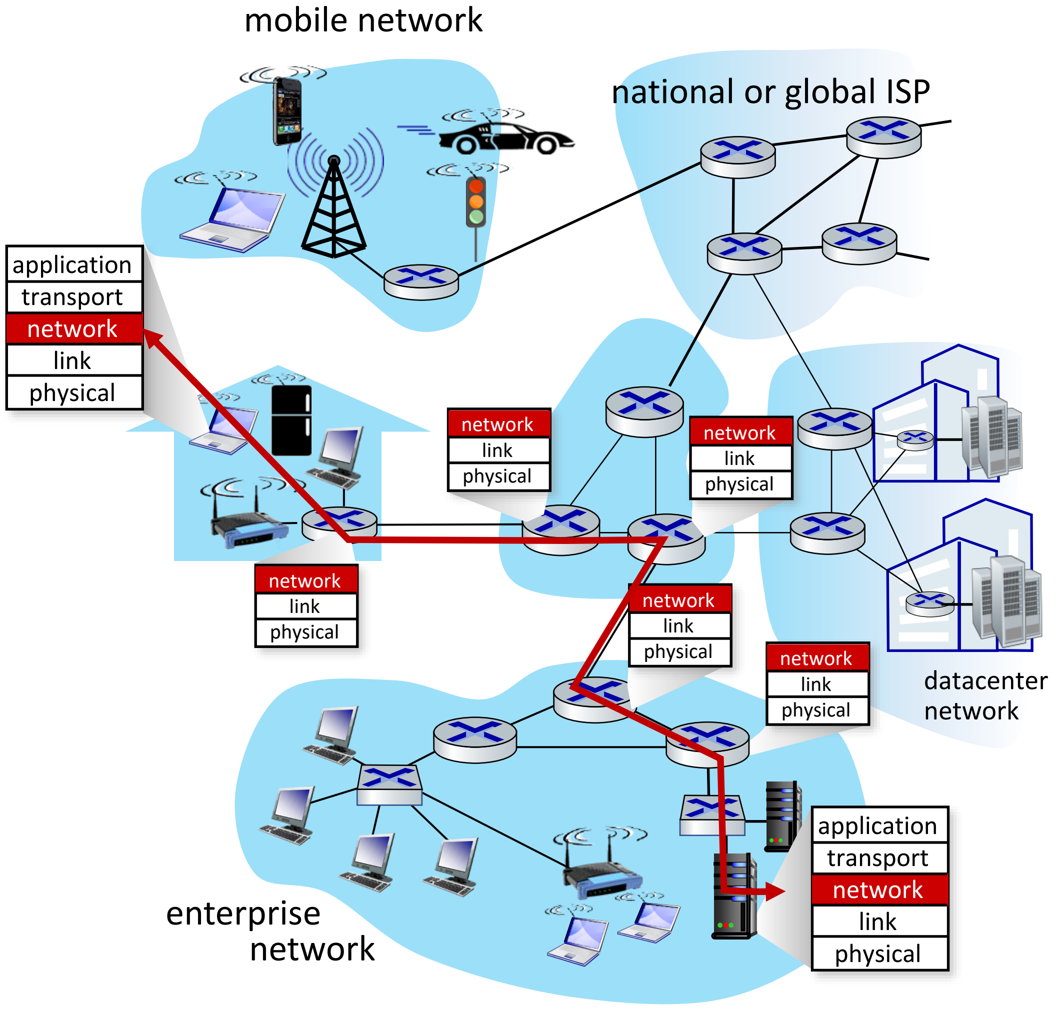
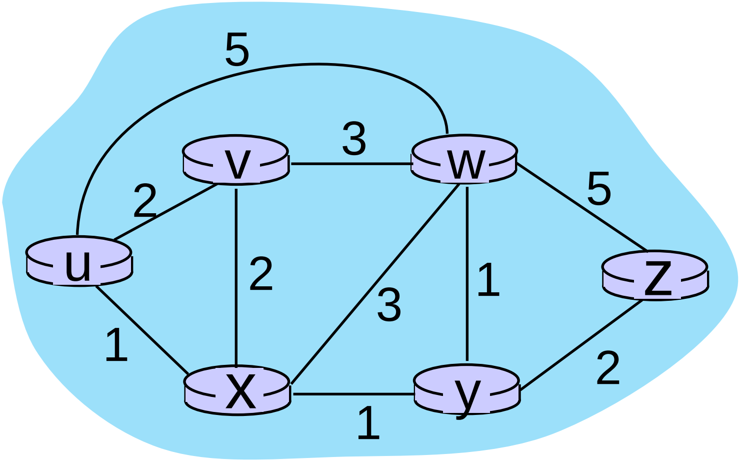
Routing protocols

Routing protocol goal: determine "good" paths (equivalently, routes), from sending hosts to receiving host, through network of routers

- path: sequence of routers packets traverse from given initial source host to final destination host
- "good": least "cost", "fastest", "least congested"
- routing: a "top-10" networking challenge!



Graph abstraction: link costs



$c_{a,b}$: cost of *direct* link connecting *a* and *b*

e.g., $c_{w,z} = 5$, $c_{u,z} = \infty$

cost defined by network operator: could always be 1, or inversely related to bandwidth, or inversely related to congestion

graph: $G = (N,E)$

N: set of routers = { *u, v, w, x, y, z* }

E: set of links ={ *(u,v), (u,x), (v,x), (v,w), (x,w), (x,y), (w,y), (w,z), (y,z)* }

Routing algorithm classification

global: all routers have *complete* topology, link cost info

- "link state" algorithms

How fast do routes change?

static: routes change slowly over time

dynamic: routes change more quickly

- periodic updates or in response to link cost changes

decentralized: iterative process of computation, exchange of info with neighbors

- routers initially only know link costs to attached neighbors
- "distance vector" algorithms

global or decentralized information?

Network layer: “control plane” roadmap

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Dijkstra’s link-state routing algorithm

- centralized: network topology, link costs known to *all* nodes
  - accomplished via “link state broadcast”
  - all nodes have same info
- computes least cost paths from one node (“source”) to all other nodes
  - gives *forwarding table* for that node
- iterative: after k iterations, know least cost path to k destinations

notation

- $c_{x,y}$: direct link cost from node x to y; $= \infty$ if not direct neighbors
- $D(v)$: *current* estimate of cost of least-cost-path from source to destination v
- $p(v)$: predecessor node along path from source to v
- N': set of nodes whose least-cost-path *definitively* known

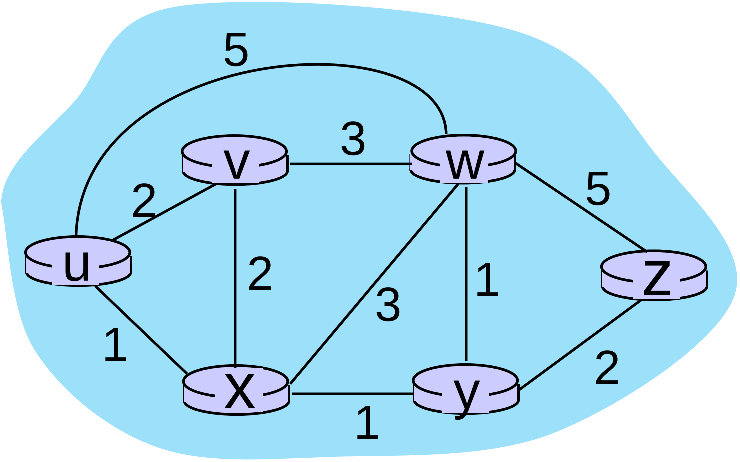
Dijkstra's link-state routing algorithm

Initialization:
  $N' = \{u\}$ /* compute least cost path from u to all other nodes */
  for all nodes v
    if v adjacent to u /* u initially knows direct-path-cost only to direct neighbors */
      then $D(v) = c_{u,v}$ /* but may not be *minimum* cost! */
    else $D(v) = \infty$

Loop
  find w not in N' such that $D(w)$ is a minimum
  add w to N'
  update $D(v)$ for all v adjacent to w and not in N' :
    $D(v) = \min(D(v), D(w) + c_{w,v})$
  /* new least-path-cost to v is either old least-cost-path to v or known
  least-cost-path to w plus direct-cost from w to v */
until all nodes in N'

Dijkstra's algorithm: an example

		v	w	x	y	z
Step	N'	D(v),p(v)	D(w),p(w)	D(x),p(x)	D(y),p(y)	D(z),p(z)
0	u	2,u	5,u	1,u	∞	∞
1						
2						
3						
4						
5						

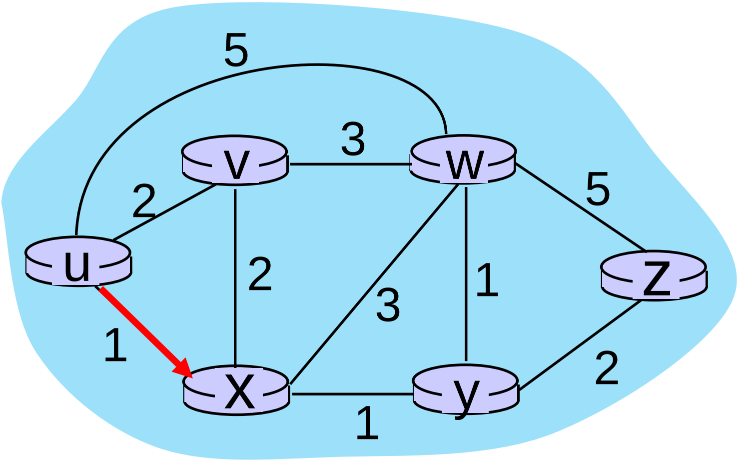


Initialization (step 0):

For all *a*: if *a* adjacent to *u* then $D(a) = c_{u,a}$

Dijkstra's algorithm: an example

Step	N'	v D(v),p(v)	w D(w),p(w)	x D(x),p(x)	y D(y),p(y)	z D(z),p(z)
0	u	2,u	5,u	1,u	∞	∞
1	ux					
2						
3						
4						
5						



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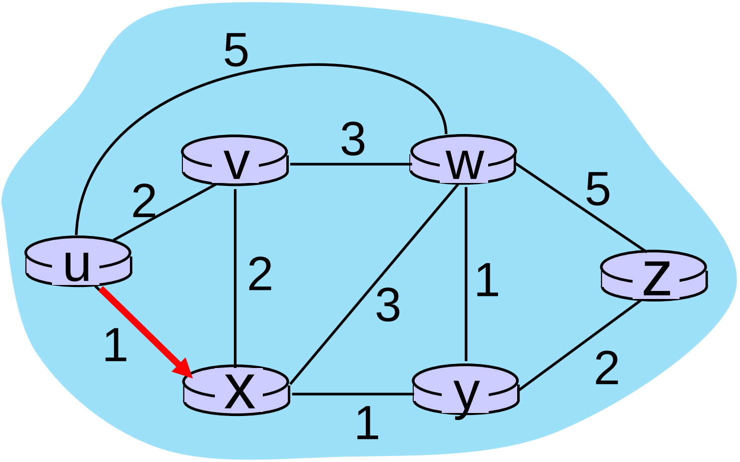
    add *a* to *N'*

Dijkstra’s algorithm: an example

Step	N'	v D(v),p(v)	w D(w),p(w)	x D(x),p(x)	y D(y),p(y)	z D(z),p(z)
0	u	2,u	5,u	1,u	∞	∞
1	ux	2,u	4,x		2,x	∞
2						
3						
4						
5						



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add *a* to *N'*

update *D(b)* for all *b* adjacent to *a* and not in *N'* :

$D(b) = \min(D(b), D(a) + c_{a,b})$

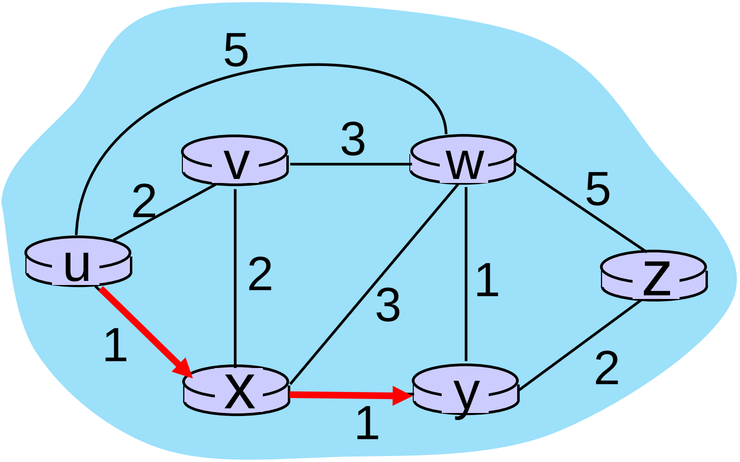
$D(v) = \min(D(v), D(x) + c_{x,v}) = \min(2, 1+2) = 2$

$D(w) = \min(D(w), D(x) + c_{x,w}) = \min(5, 1+3) = 4$

$D(y) = \min(D(y), D(x) + c_{x,y}) = \min(inf, 1+1) = 2$

Dijkstra's algorithm: an example

Step	N'	v D(v),p(v)	w D(w),p(w)	x D(x),p(x)	y D(y),p(y)	z D(z),p(z)
0	u	2,u	5,u	1,u	∞	∞
1	ux	2,u	4,x		2,x	∞
2	uxy					
3						
4						
5						



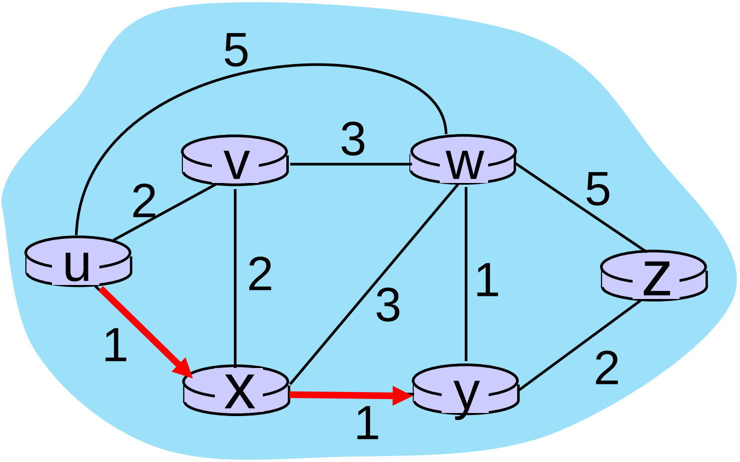
8 *Loop*

9 find *a* not in *N'* such that *D(a)* is a minimum

10 add *a* to *N'*

Dijkstra's algorithm: an example

Step	N'	v D(v),p(v)	w D(w),p(w)	x D(x),p(x)	y D(y),p(y)	z D(z),p(z)
0	u	2,u	5,u	1,u	∞	∞
1	ux	2,u	4,x		2,x	∞
2	uxy	2,u	3,y			4,y
3						
4						
5						



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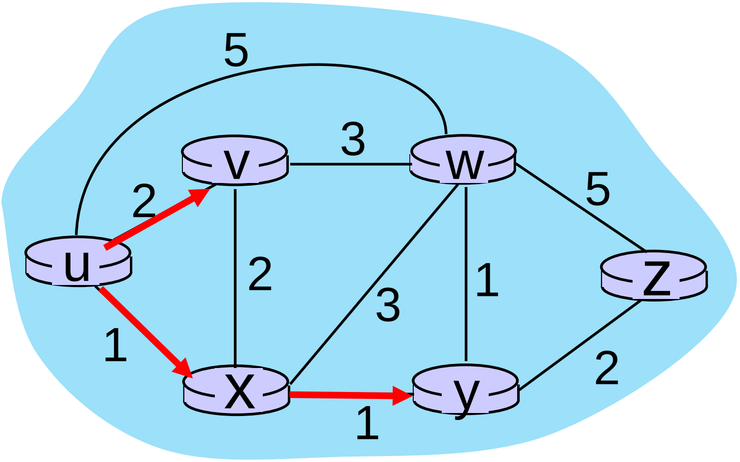
$$D(b) = \min (D(b), D(a) + c_{a,b})$$

$D(w) = \min (D(w), D(y) + c_{y,w}) = \min (4, 2+1) = 3$ NEW!

$D(z) = \min (D(z), D(y) + c_{y,z}) = \min(inf,2+2) = 4$ NEW!

Dijkstra's algorithm: an example

Step	N'	v D(v),p(v)	w D(w),p(w)	x D(x),p(x)	y D(y),p(y)	z D(z),p(z)
0	u	2,u	5,u	1,u	∞	∞
1	ux	2,u	4,x		2,x	∞
2	uxy	2,u	3,y			4,y
3	uxyv					
4						
5						



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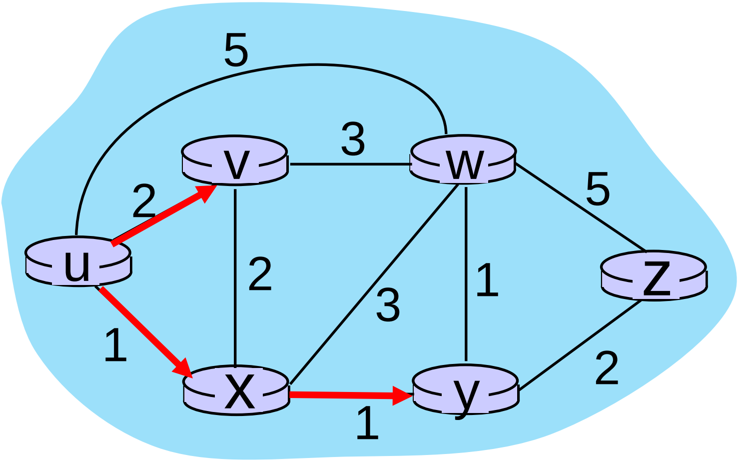
0	u	2,u	5,u	1,u	∞	∞
1	ux	2,u	4,x		2,x	∞
2	uxy	2,u	3,y			4,y
3	uxyv		3,y			4,y
4						
5						



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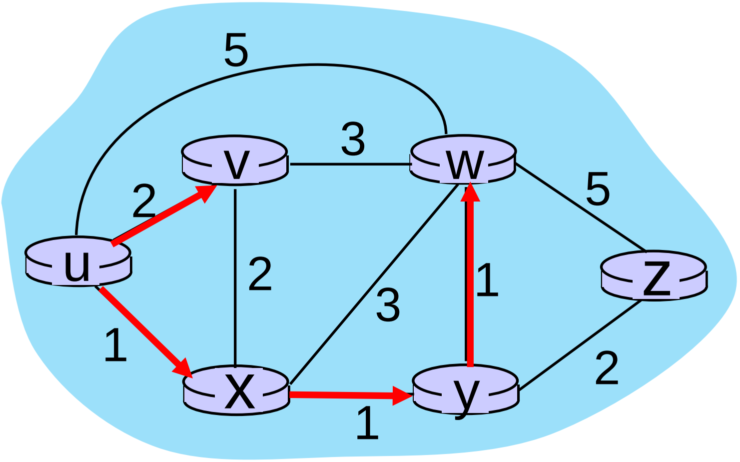
0	u	2,u	5,u	1,u	∞	∞
1	ux	2,u	4,x		2,x	∞
2	uxy	2,u	3,y			4,y
3	uxyv		3,y			4,y
4	uxyvw					
5						



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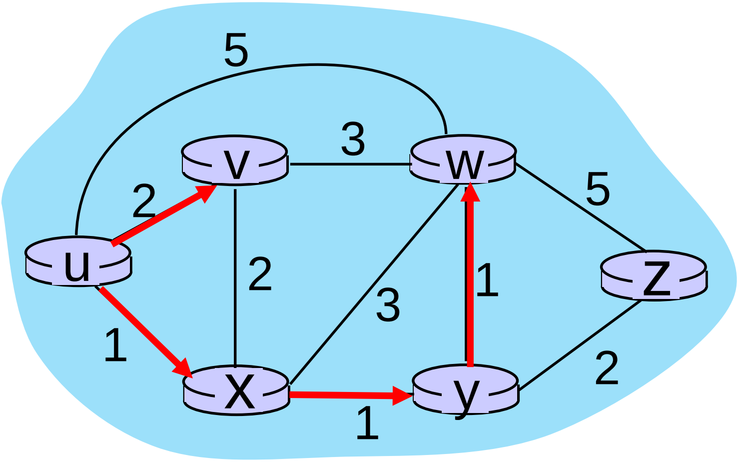
Dijkstra's algorithm: an example

		v	w	x	y	z
Step	N'	D(v),p(v)	D(w),p(w)	D(x),p(x)	D(y),p(y)	D(z),p(z)
0	u	2,u	5,u	1,u	∞	∞
1	ux	2,u	4,x		2,x	∞
2	uxy	2,u	3,y			4,y
3	uxyv		3,y			4,y
4	uxyvw					4,y
5						



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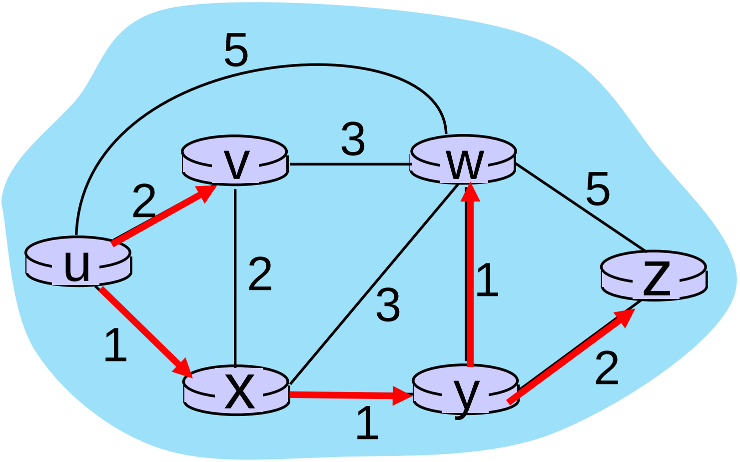
1	ux	2,u	4,x		2,x	∞
2	uxy	2,u	3,y			4,y
3	uxyv		3,y			4,y
4	uxyvw					4,y
5	uxyvwz					



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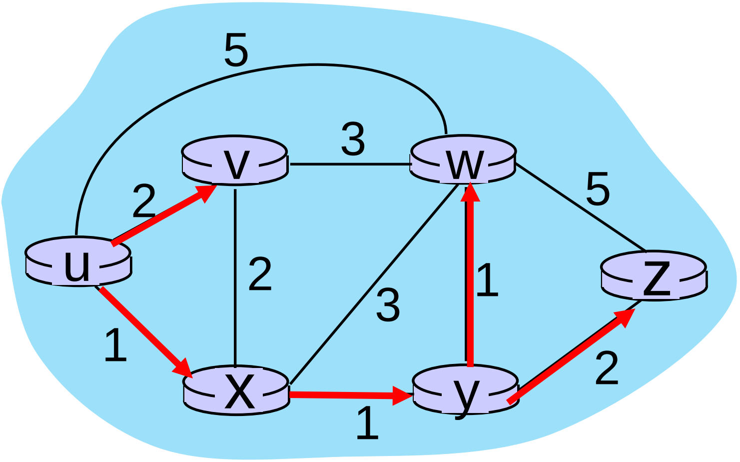
Dijkstra’s algorithm: an example

Step	N'	v D(v),p(v)	w D(w),p(w)	x D(x),p(x)	y D(y),p(y)	z D(z),p(z)
0	u	2,u	5,u	1,u	∞	∞
1	ux	2,u	4,x		2,x	∞
2	uxy	2,u	3,y			4,y
3	uxyv		3,y			4,y
4	uxyvw					4,y
5	uxyvwz					



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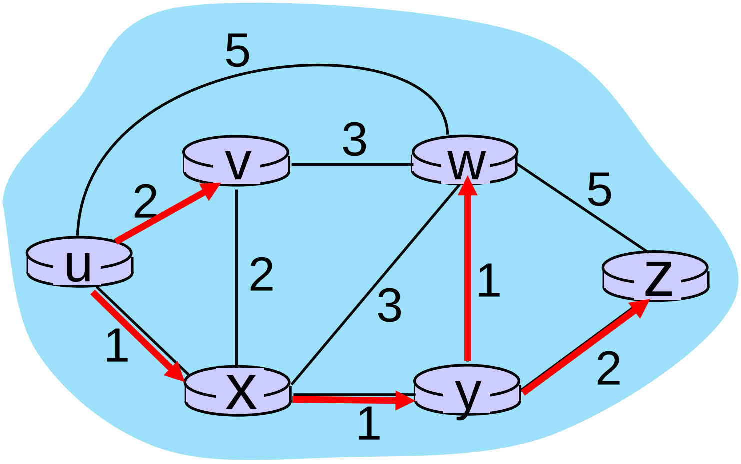
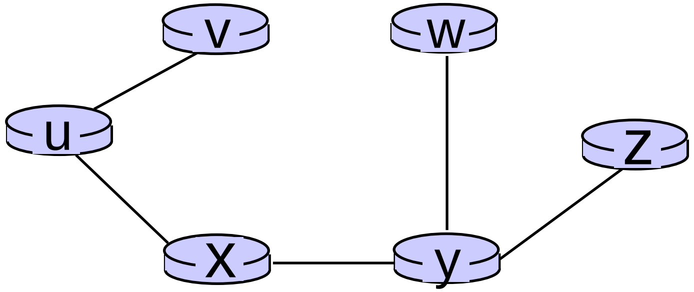
Dijkstra's algorithm: an example



resulting least-cost-path tree from u:



resulting forwarding table in u:

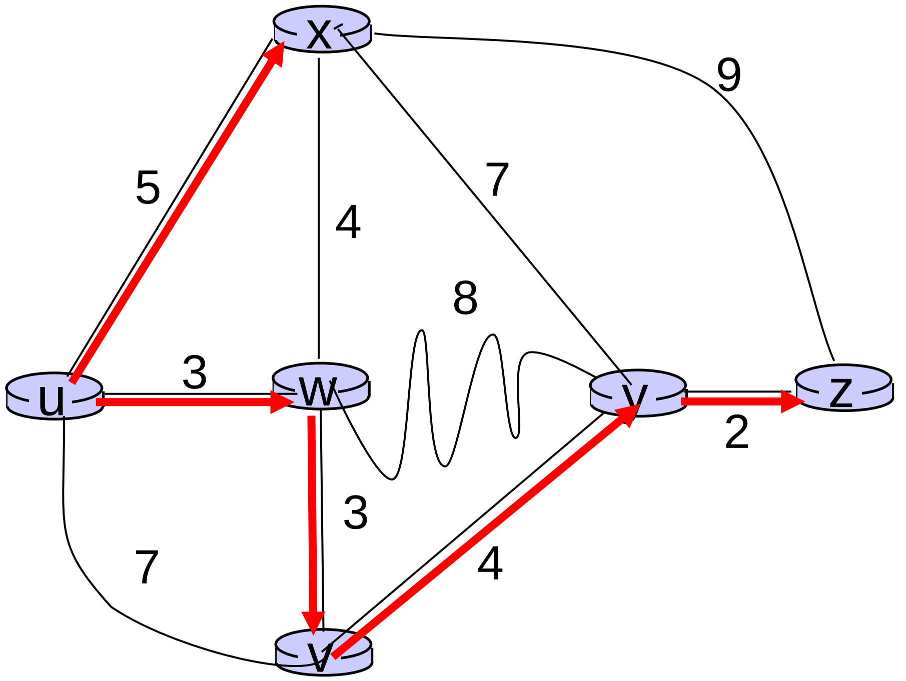
destination	outgoing link
v	(u,v)
x	(u,x)
y	(u,x)
w	(u,x)
x	(u,x)

route from *u* to *v* directly

route from u to all other destinations via *x*

Dijkstra’s algorithm: another example

Step	N'	D(v), p(v)	D(w), p(w)	D(x), p(x)	D(y), p(y)	D(z), p(z)
0	u	7,u	3,u	5,u	∞	∞
1	uw	6,w		5,u	11,w	∞
2	uwx	6,w			11,w	14,x
3	uwxv				10,v	14,x
4	uwxvy					12,y
5	uwxvyz					



notes:

- construct least-cost-path tree by tracing predecessor nodes
- ties can exist (can be broken arbitrarily)

Dijkstra's algorithm: discussion

algorithm complexity: *n* nodes

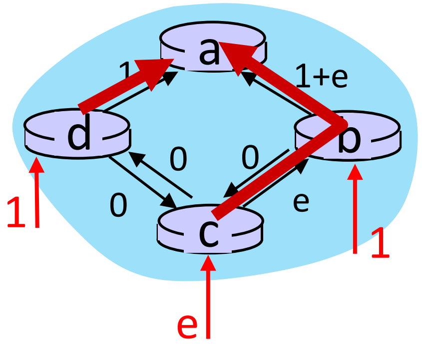
- each of *n* iteration: need to check all nodes, *w*, not in *N*
- $n(n+1)/2$ comparisons: $O(n^2)$ complexity
- more efficient implementations possible: $O(n \log n)$

message complexity:

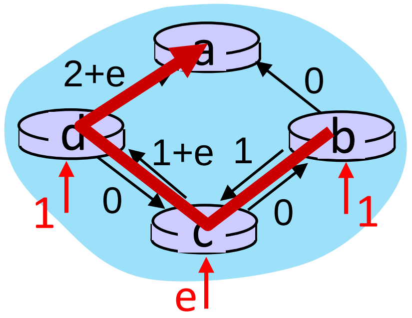
- each router must *broadcast* its link state information to other *n* routers
- efficient (and interesting!) broadcast algorithms: $O(n)$ link crossings to disseminate a broadcast message from one source
- each router's message crosses $O(n)$ links: overall message complexity: $O(n^2)$

Dijkstra's algorithm: oscillations possible

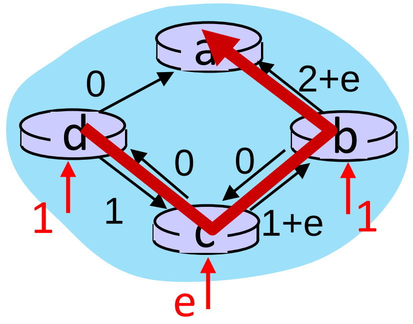
- when link costs depend on traffic volume, route oscillations possible
- sample scenario:
  - routing to destination a, traffic entering at d, c, e with rates 1, e (<1), 1
  - link costs are directional, and volume-dependent



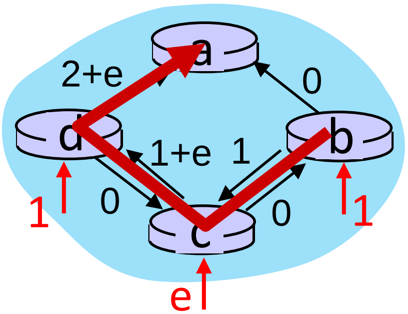
initially



given these costs, find new routing.... resulting in new costs



given these costs, find new routing.... resulting in new costs



given these costs, find new routing.... resulting in new costs

Network layer: “control plane” roadmap

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- Internet Control Message Protocol
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  - NETCONF/YANG

Distance vector algorithm

Based on *Bellman-Ford* (BF) equation (dynamic programming):

Bellman-Ford equation

Let $D_x(y)$: cost of least-cost path from x to y.
Then:

$$D_x(y) = \min_v \{ c_{x,v} + D_v(y) \}$$

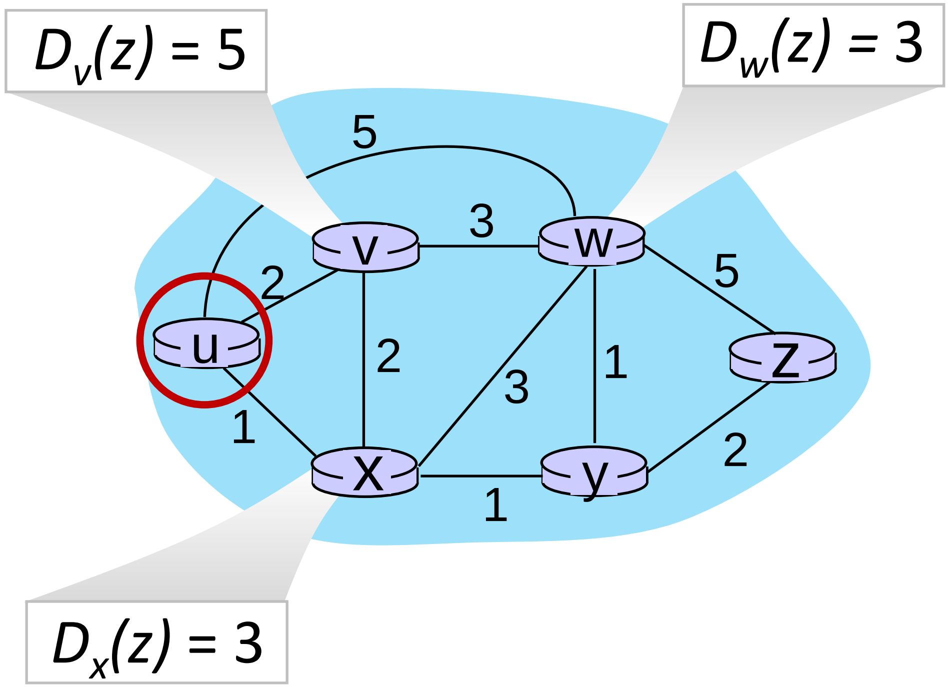
v's estimated least-cost-path cost to y

min taken over all neighbors v of x

direct cost of link from x to v

Bellman-Ford Example

Suppose that *u*'s neighboring nodes, *x,v,w,* know that for destination *z*:



Bellman-Ford equation says:

$$
\begin{aligned}
D_u(z) = \min \{ & c_{u,v} + D_v(z), \\
& c_{u,x} + D_x(z), \\
& c_{u,w} + D_w(z) \} \\
= \min \{ & 2 + 5, \\
& 1 + 3, \\
& 5 + 3\} = 4
\end{aligned}
$$

node achieving minimum (x) is next hop on estimated least-cost path to destination (z)

Distance vector algorithm

key idea:

- from time-to-time, each node sends its own distance vector estimate to neighbors
- when x receives new DV estimate from any neighbor, it updates its own DV using B-F equation:

$$D_x(y) \leftarrow \min_v\{c_{x,v} + D_v(y)\} \text{ for each node } y \in N$$

- under minor, natural conditions, the estimate *$D_x(y)$ converge to the actual least cost* $d_x(y)$

Distance vector algorithm:

each node:

wait for (change in local link cost or msg from neighbor)

↓

recompute DV estimates using DV received from neighbor

↓

if DV to any destination has changed, *notify* neighbors

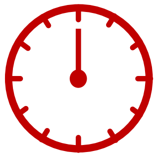
iterative, asynchronous: each local iteration caused by:

- local link cost change
- DV update message from neighbor

distributed, self-stopping: each node notifies neighbors *only* when its DV changes

- neighbors then notify their neighbors – *only if necessary*
- no notification received, no actions taken!

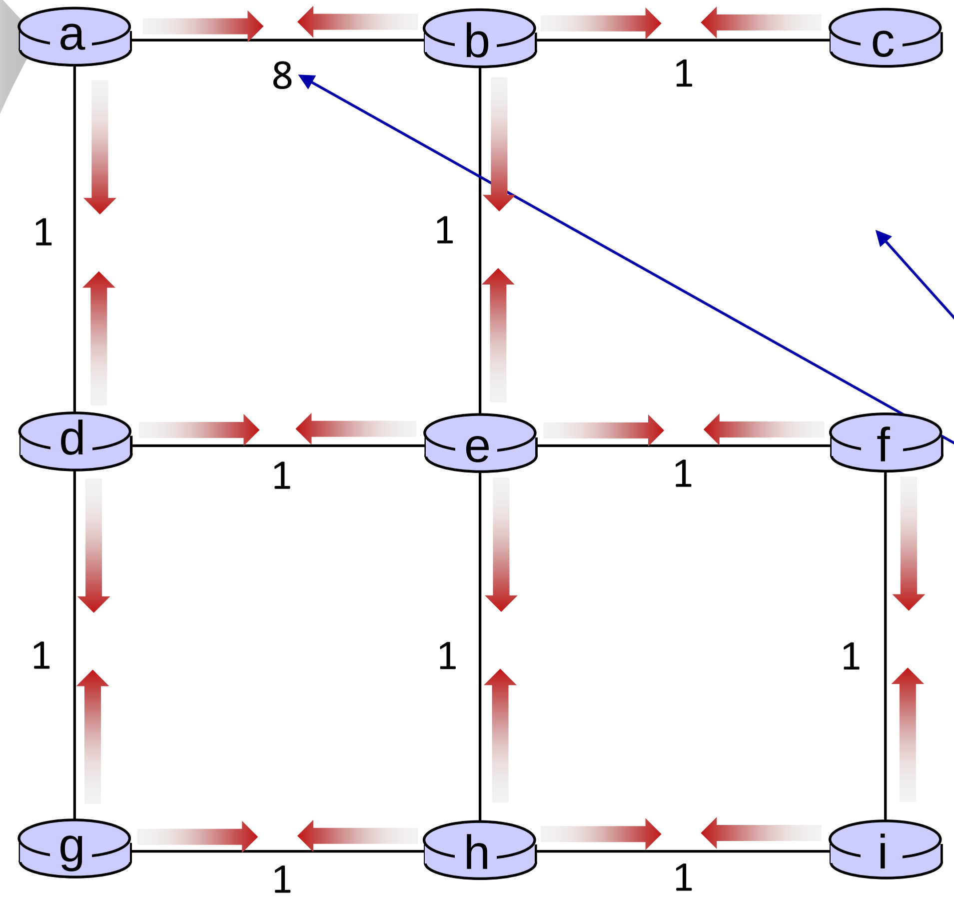
Distance vector: example



t=0

- All nodes have distance estimates to nearest neighbors (only)
- All nodes send their local distance vector to their neighbors

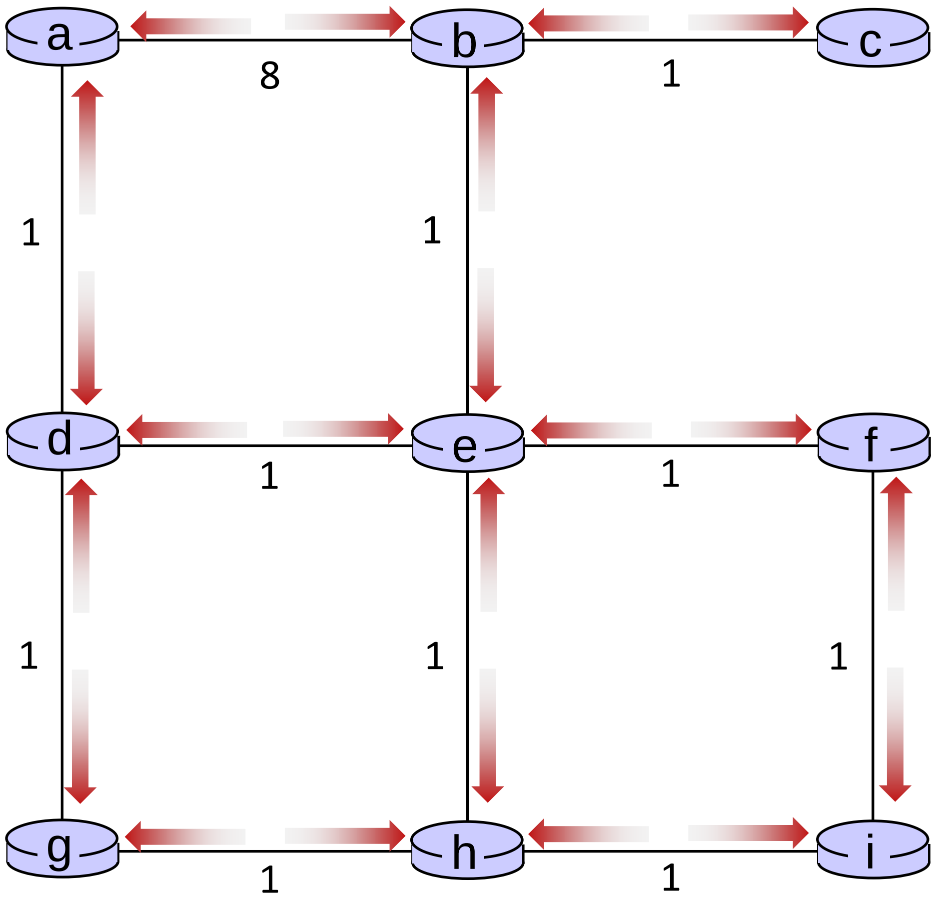
DV in a:
$D_a(a)=0$
$D_a(b) = 8$
$D_a(c) = \infty$
$D_a(d) = 1$
$D_a(e) = \infty$
$D_a(f) = \infty$
$D_a(g) = \infty$
$D_a(h) = \infty$
$D_a(i) = \infty$



A few asymmetries:
- missing link
- larger cost

Distance vector example: iteration

All nodes:
- receive distance vectors from neighbors
- compute their new local distance vector
- send their new local distance vector to neighbors

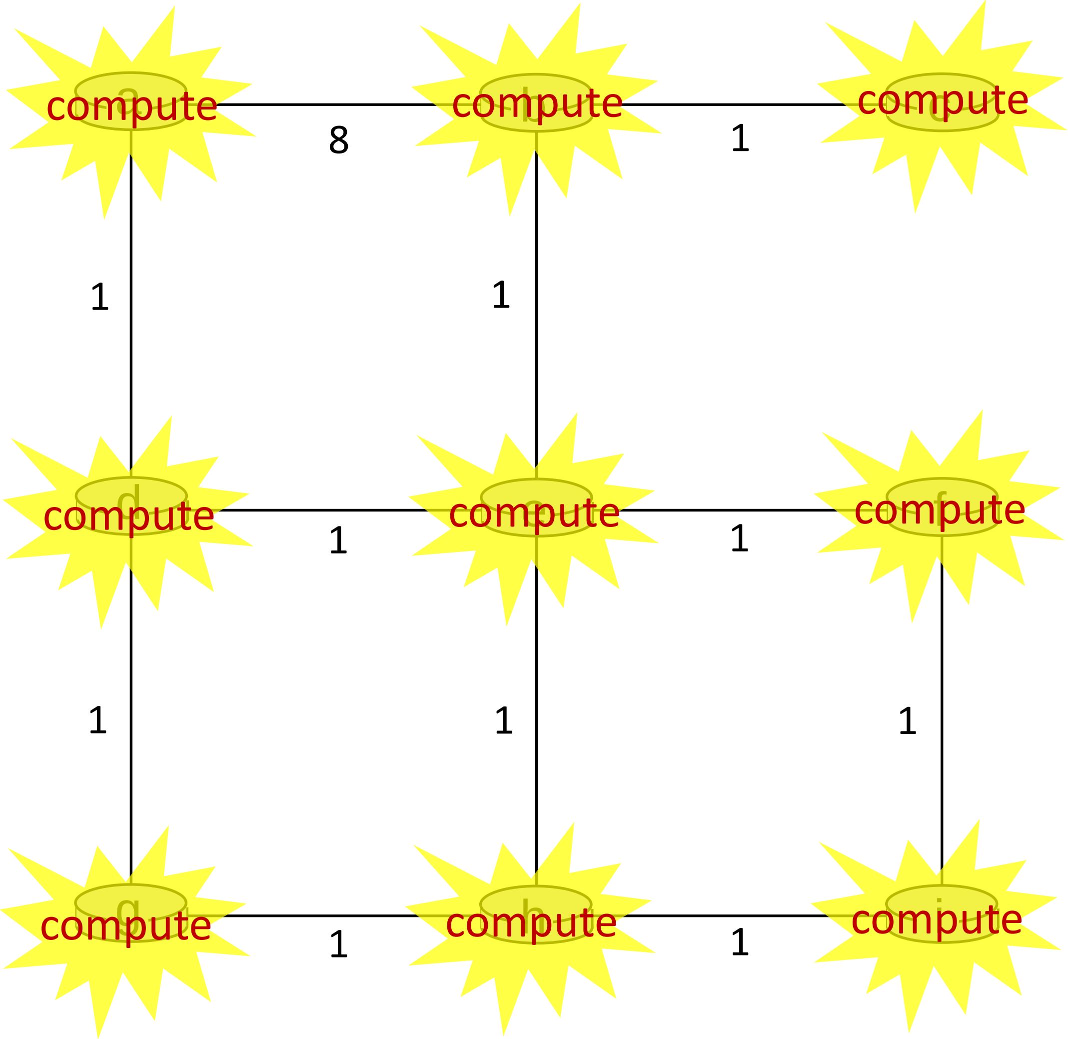


Distance vector example: iteration

t=1

All nodes:

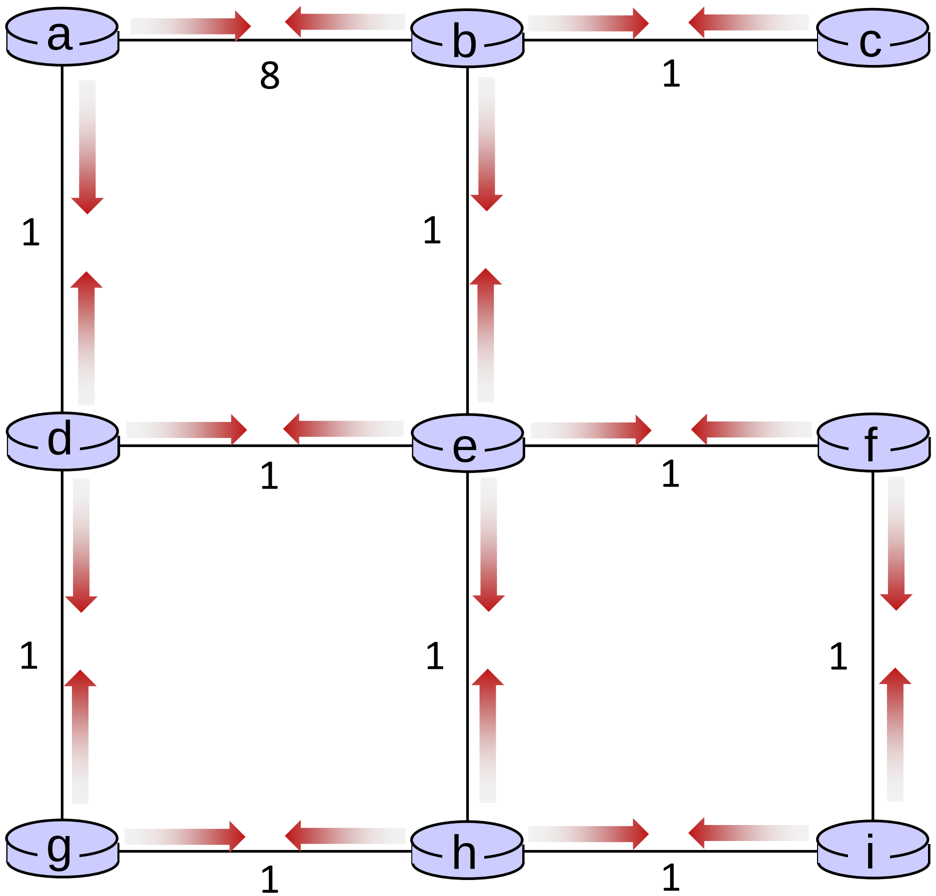
- receive distance vectors from neighbors
- compute their new local distance vector
- send their new local distance vector to neighbors



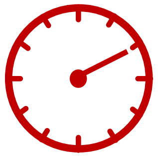
Distance vector example: iteration

All nodes:

- receive distance vectors from neighbors
- compute their new local distance vector
- send their new local distance vector to neighbors

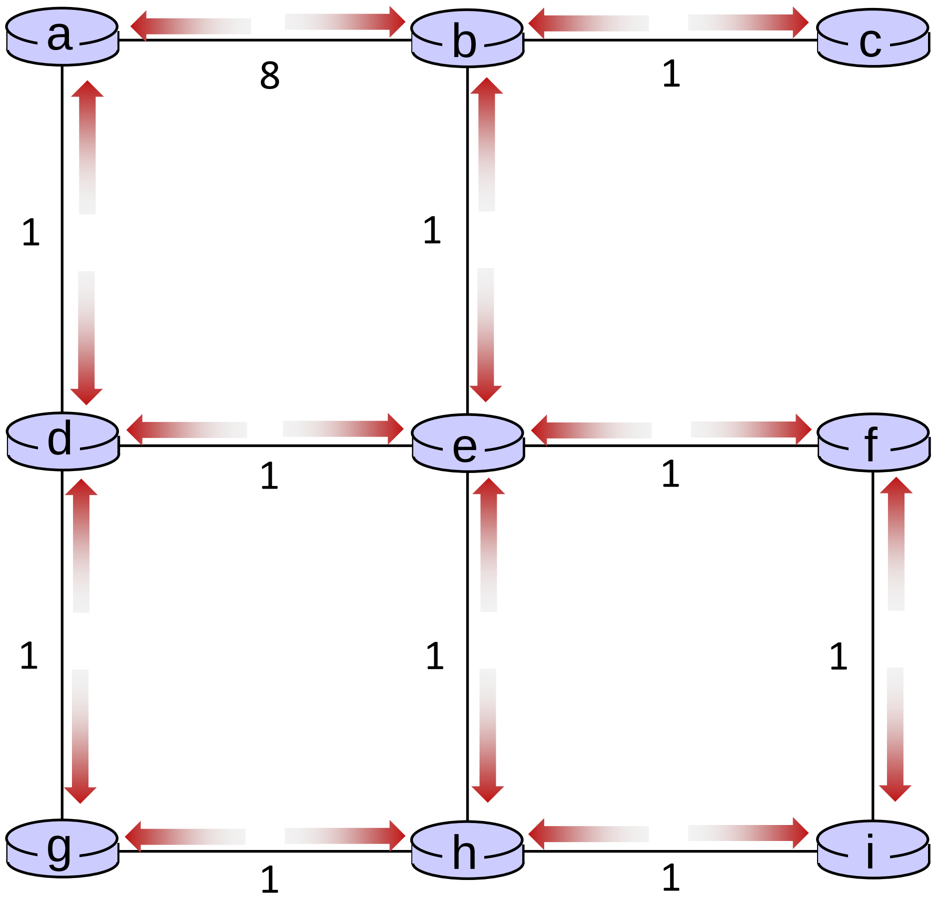


Distance vector example: iteration



t=2

All nodes:
- receive distance vectors from neighbors
- compute their new local distance vector
- send their new local distance vector to neighbors

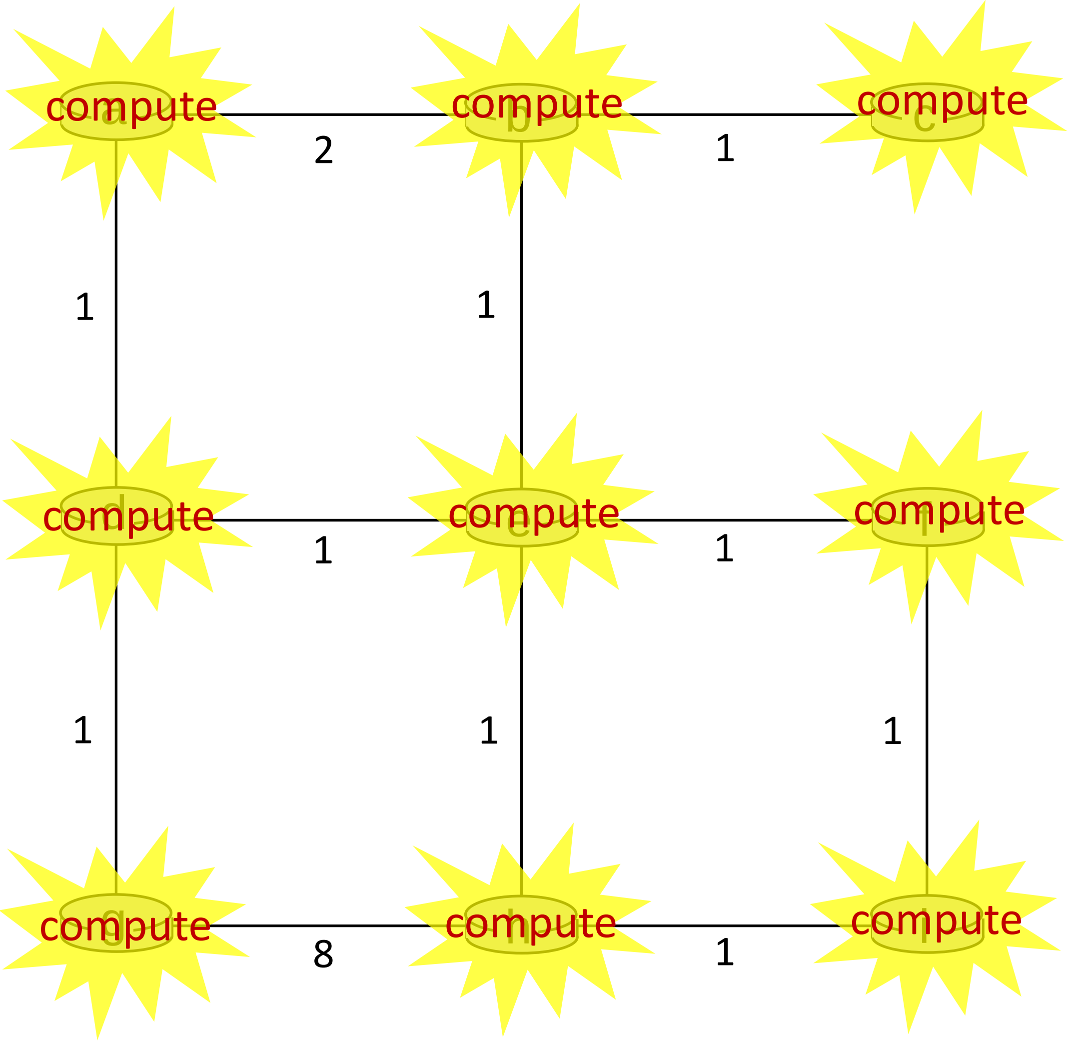


Distance vector example: iteration

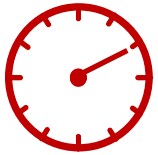
t=2

All nodes:

- receive distance vectors from neighbors
- compute their new local distance vector
- send their new local distance vector to neighbors

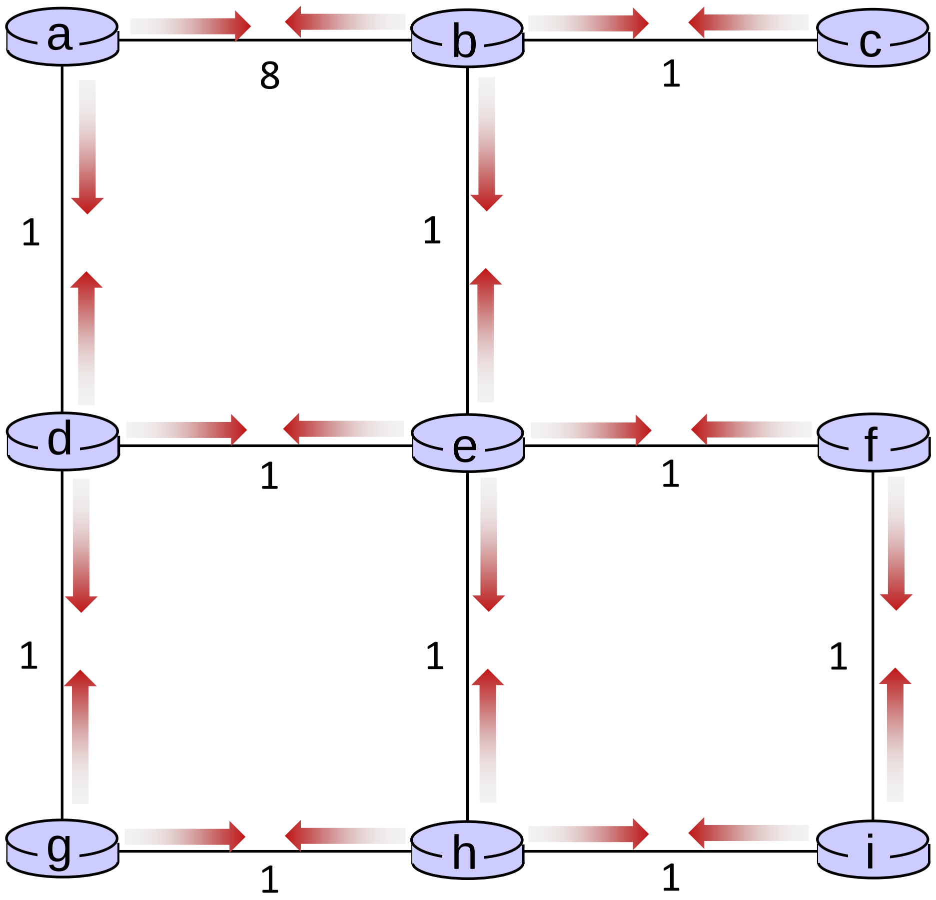


Distance vector example: iteration



t=2

All nodes:
- receive distance vectors from neighbors
- compute their new local distance vector
- send their new local distance vector to neighbors



Distance vector example: iteration

…. and so on

Let's next take a look at the iterative *computations* at nodes

Distance vector example: (

t=1

- b receives DVs from a, c, e

DV in a:

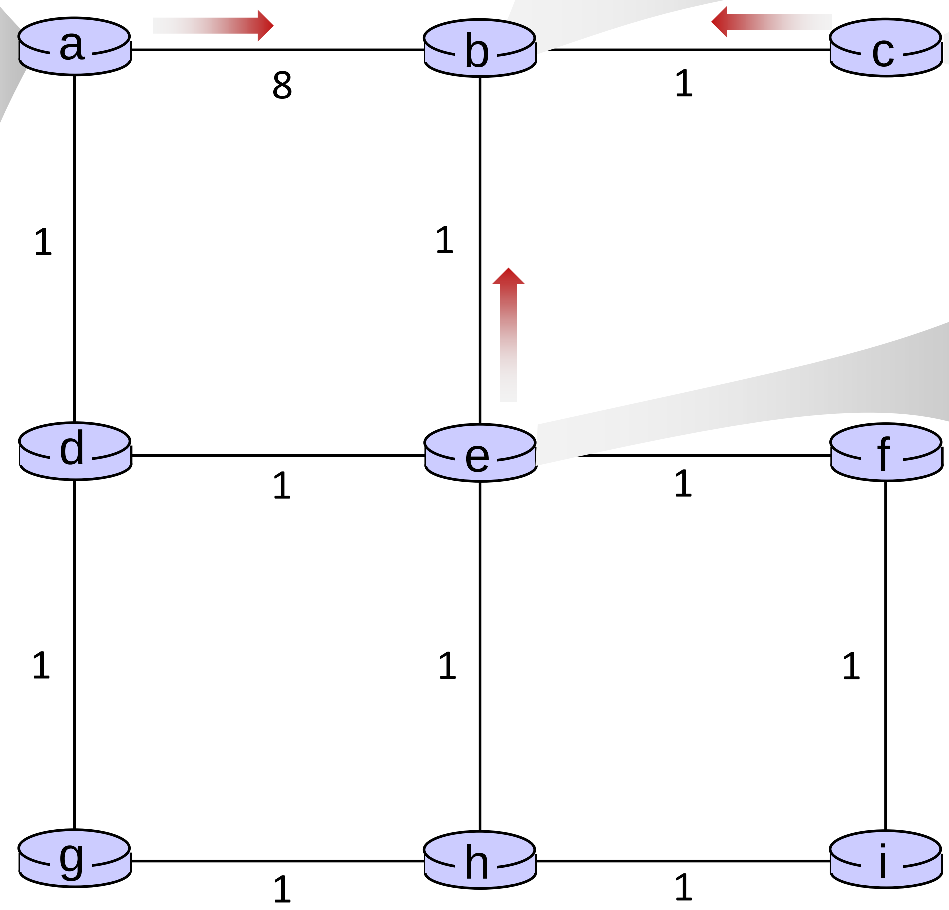
$D_a(a)=0$
$D_a(b) = 8$
$D_a(c) = \infty$
$D_a(d) = 1$
$D_a(e) = \infty$
$D_a(f) = \infty$
$D_a(g) = \infty$
$D_a(h) = \infty$
$D_a(i) = \infty$

DV in b:

$D_b(a) = 8$	$D_b(f) = \infty$
$D_b(c) = 1$	$D_b(g) = \infty$
$D_b(d) = \infty$	$D_b(h) = \infty$
$D_b(e) = 1$	$D_b(i) = \infty$

DV in c:

$D_c(a) = \infty$
$D_c(b) = 1$
$D_c(c) = 0$
$D_c(d) = \infty$
$D_c(e) = \infty$
$D_c(f) = \infty$
$D_c(g) = \infty$
$D_c(h) = \infty$
$D_c(i) = \infty$



DV in e:

$D_e(a) = \infty$
$D_e(b) = 1$
$D_e(c) = \infty$
$D_e(d) = 1$
$D_e(e) = 0$
$D_e(f) = 1$
$D_e(g) = \infty$
$D_e(h) = 1$
$D_e(i) = \infty$

Distance vector example: (

DV in b:

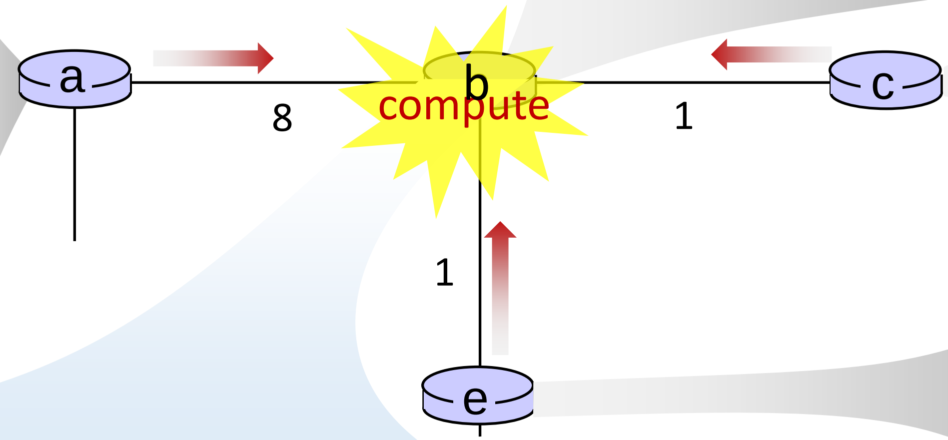
$D_b(a) = 8$ $D_b(f) = \infty$
$D_b(c) = 1$ $D_b(g) = \infty$
$D_b(d) = \infty$ $D_b(h) = \infty$
$D_b(e) = 1$ $D_b(i) = \infty$

DV in c:

$D_c(a) = \infty$
$D_c(b) = 1$
$D_c(c) = 0$
$D_c(d) = \infty$
$D_c(e) = \infty$
$D_c(f) = \infty$
$D_c(g) = \infty$
$D_c(h) = \infty$
$D_c(i) = \infty$

DV in a:

$D_a(a)=0$
$D_a(b) = 8$
$D_a(c) = \infty$
$D_a(d) = 1$
$D_a(e) = \infty$
$D_a(f) = \infty$
$D_a(g) = \infty$
$D_a(h) = \infty$
$D_a(i) = \infty$



t=1

- b receives DVs from a, c, e, computes:

DV in e:

$D_e(a) = \infty$
$D_e(b) = 1$
$D_e(c) = \infty$
$D_e(d) = 1$
$D_e(e) = 0$
$D_e(f) = 1$
$D_e(g) = \infty$
$D_e(h) = 1$
$D_e(i) = \infty$

$D_b(a) = \min\{c_{b,a}+D_a(a),\ c_{b,c}+D_c(a),\ c_{b,e}+D_e(a)\} = \min\{8,\infty,\infty\} = 8$

$D_b(c) = \min\{c_{b,a}+D_a(c),\ c_{b,c}+D_c(c),\ c_{b,e}+D_e(c)\} = \min\{\infty,1,\infty\} = 1$

$D_b(d) = \min\{c_{b,a}+D_a(d),\ c_{b,c}+D_c(d),\ c_{b,e}+D_e(d)\} = \min\{9,2,\infty\} = 2$

$D_b(e) = \min\{c_{b,a}+D_a(e),\ c_{b,c}+D_c(e),\ c_{b,e}+D_e(e)\} = \min\{\infty,\infty,1\} = 1$

$D_b(f) = \min\{c_{b,a}+D_a(f),\ c_{b,c}+D_c(f),\ c_{b,e}+D_e(f)\} = \min\{\infty,\infty,2\} = 2$

$D_b(g) = \min\{c_{b,a}+D_a(g),\ c_{b,c}+D_c(g),\ c_{b,e}+D_e(g)\} = \min\{\infty, \infty, \infty\} = \infty$

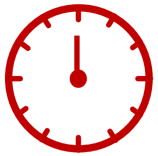
$D_b(h) = \min\{c_{b,a}+D_a(h),\ c_{b,c}+D_c(h),\ c_{b,e}+D_e(h)\} = \min\{\infty, \infty, 2\} = 2$

$D_b(i) = \min\{c_{b,a}+D_a(i),\ c_{b,c}+D_c(i),\ c_{b,e}+D_e(i)\} = \min\{\infty, \infty, \infty\} = \infty$

DV in b:

$D_b(a) = 8$ $D_b(f) =2$
$D_b(c) = 1$ $D_b(g) = \infty$
$D_b(d) = 2$ $D_b(h) = 2$
$D_b(e) = 1$ $D_b(i) = \infty$

Distance vector example:



t=1

- c receives DVs from b

DV in a:

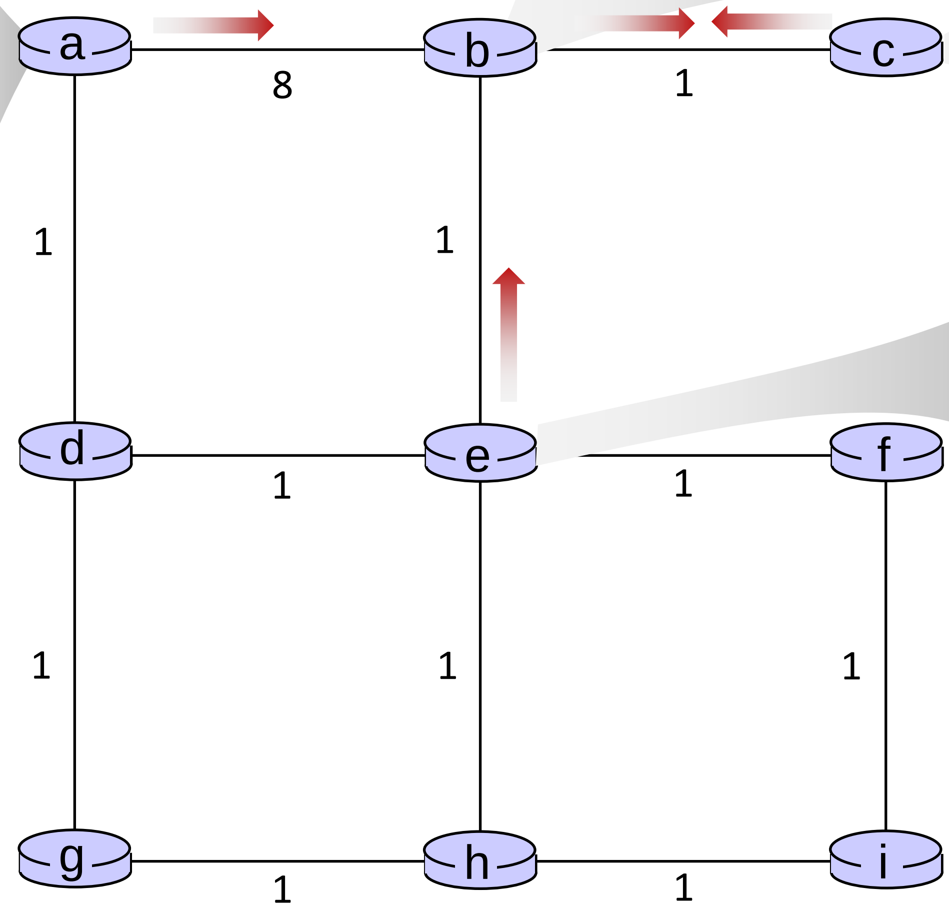
$D_a(a)=0$
$D_a(b) = 8$
$D_a(c) = \infty$
$D_a(d) = 1$
$D_a(e) = \infty$
$D_a(f) = \infty$
$D_a(g) = \infty$
$D_a(h) = \infty$
$D_a(i) = \infty$

DV in b:

$D_b(a) = 8$	$D_b(f) = \infty$
$D_b(c) = 1$	$D_b(g) = \infty$
$D_b(d) = \infty$	$D_b(h) = \infty$
$D_b(e) = 1$	$D_b(i) = \infty$

DV in c:

$D_c(a) = \infty$
$D_c(b) = 1$
$D_c(c) = 0$
$D_c(d) = \infty$
$D_c(e) = \infty$
$D_c(f) = \infty$
$D_c(g) = \infty$
$D_c(h) = \infty$
$D_c(i) = \infty$



DV in e:

$D_e(a) = \infty$
$D_e(b) = 1$
$D_e(c) = \infty$
$D_e(d) = 1$
$D_e(e) = 0$
$D_e(f) = 1$
$D_e(g) = \infty$
$D_e(h) = 1$
$D_e(i) = \infty$

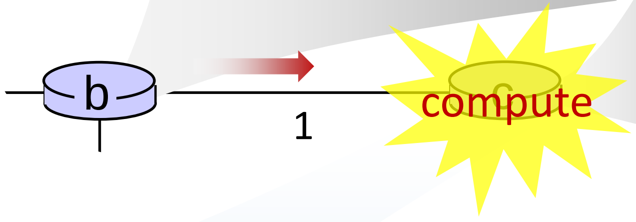
Distance vector example:

DV in b:

$D_b(a) = 8$	$D_b(f) = \infty$
$D_b(c) = 1$	$D_b(g) = \infty$
$D_b(d) = \infty$	$D_b(h) = \infty$
$D_b(e) = 1$	$D_b(i) = \infty$

DV in c:

$D_c(a) = \infty$
$D_c(b) = 1$
$D_c(c) = 0$
$D_c(d) = \infty$
$D_c(e) = \infty$
$D_c(f) = \infty$
$D_c(g) = \infty$
$D_c(h) = \infty$
$D_c(i) = \infty$



t=1

- c receives DVs from b computes:

$D_c(a) = \min\{c_{c,b}+D_b(a)\} = 1 + 8 = 9$

$D_c(b) = \min\{c_{c,b}+D_b(b)\} = 1 + 0 = 1$

$D_c(d) = \min\{c_{c,b}+D_b(d)\} = 1+ \infty = \infty$

$D_c(e) = \min\{c_{c,b}+D_b(e)\} = 1 + 1 = 2$

$D_c(f) = \min\{c_{c,b}+D_b(f)\} = 1+ \infty = \infty$

$D_c(g) = \min\{c_{c,b}+D_b(g)\} = 1+ \infty = \infty$

$D_c(h) = \min\{c_{bc,b}+D_b(h)\} = 1+ \infty = \infty$

$D_c(i) = \min\{c_{c,b}+D_b(i)\} = 1+ \infty = \infty$

DV in c:

$D_c(a) = 9$
$D_c(b) = 1$
$D_c(c) = 0$
$D_c(d) = 2$
$D_c(e) = \infty$
$D_c(f) = \infty$
$D_c(g) = \infty$
$D_c(h) = \infty$
$D_c(i) = \infty$

Distance vector example:

t=1

- e receives DVs from b, d, f, h

DV in d:

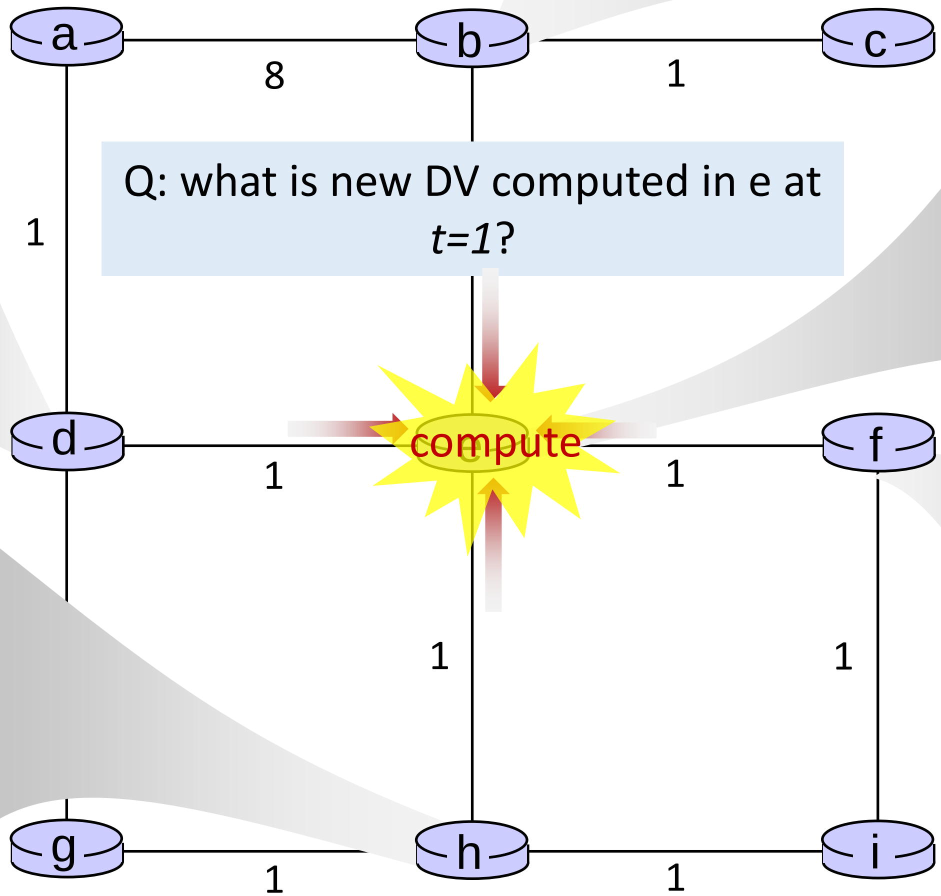
$D_c(a) = 1$
$D_c(b) = \infty$
$D_c(c) = \infty$
$D_c(d) = 0$
$D_c(e) = 1$
$D_c(f) = \infty$
$D_c(g) = 1$
$D_c(h) = \infty$
$D_c(i) = \infty$

DV in h:

$D_c(a) = \infty$
$D_c(b) = \infty$
$D_c(c) = \infty$
$D_c(d) = \infty$
$D_c(e) = 1$
$D_c(f) = \infty$
$D_c(g) = 1$
$D_c(h) = 0$
$D_c(i) = 1$

DV in b:

$D_b(a) = 8$	$D_b(f) = \infty$
$D_b(c) = 1$	$D_b(g) = \infty$
$D_b(d) = \infty$	$D_b(h) = \infty$
$D_b(e) = 1$	$D_b(i) = \infty$



DV in e:

$D_e(a) = \infty$
$D_e(b) = 1$
$D_e(c) = \infty$
$D_e(d) = 1$
$D_e(e) = 0$
$D_e(f) = 1$
$D_e(g) = \infty$
$D_e(h) = 1$
$D_e(i) = \infty$

DV in f:

$D_c(a) = \infty$
$D_c(b) = \infty$
$D_c(c) = \infty$
$D_c(d) = \infty$
$D_c(e) = 1$
$D_c(f) = 0$
$D_c(g) = \infty$
$D_c(h) = \infty$
$D_c(i) = 1$

Distance vector: state information diffusion

Iterative communication, computation steps diffuses information through network:

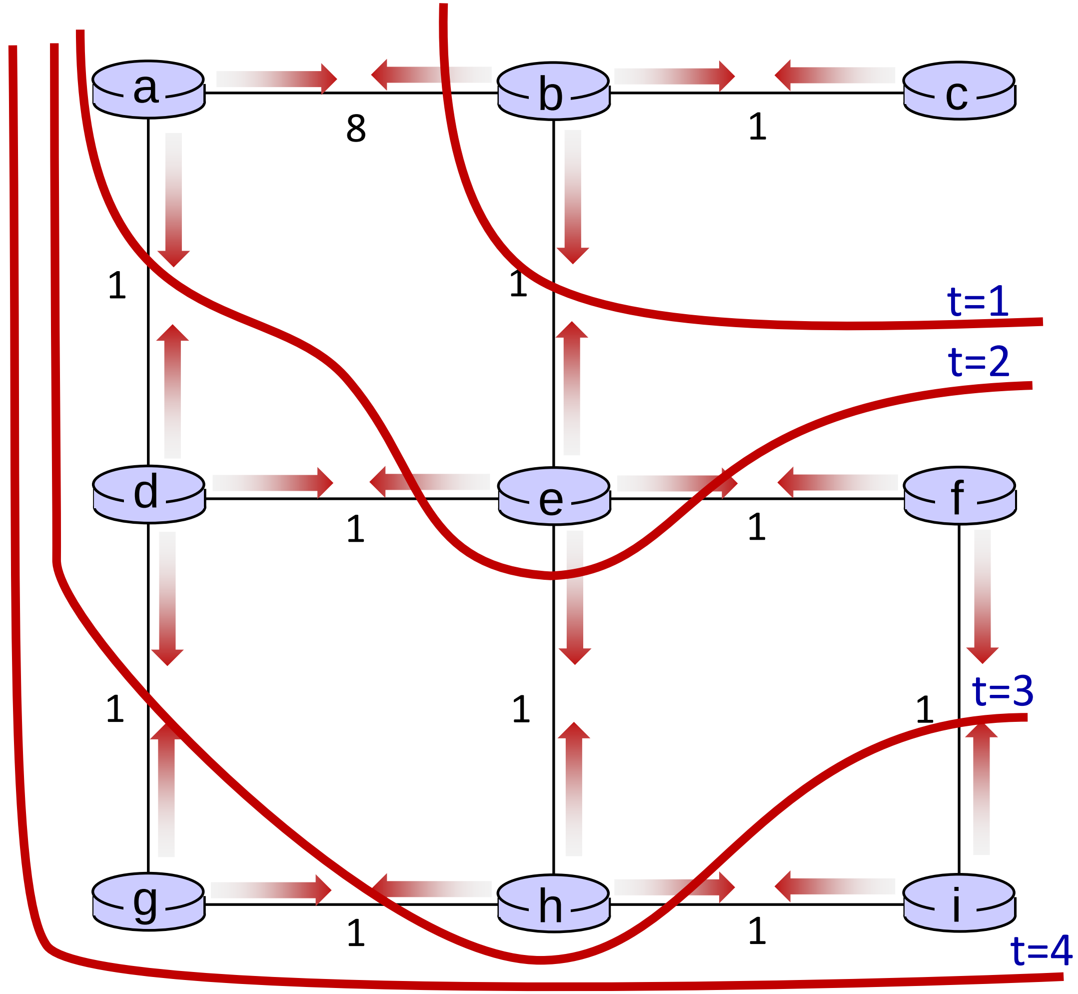
t=0 c's state at t=0 is at c only

t=1 c's state at t=0 has propagated to b, and may influence distance vector computations up to **1** hop away, i.e., at b

t=2 c's state at t=0 may now influence distance vector computations up to **2** hops away, i.e., at b and now at a, e as well

t=3 c's state at t=0 may influence distance vector computations up to **3** hops away, i.e., at d, f, h

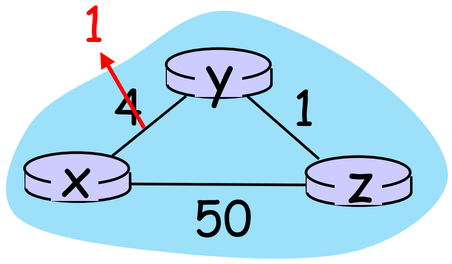
t=4 c's state at t=0 may influence distance vector computations up to **4** hops away, i.e., at g, i



Distance vector: link cost changes

link cost changes:

- node detects local link cost change
- updates routing info, recalculates local DV
- if DV changes, notify neighbors



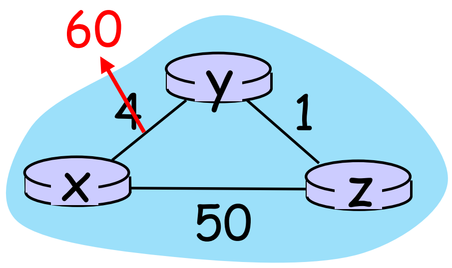
"good news travels fast"

t_0 : *y* detects link-cost change, updates its DV, informs its neighbors.

t_1 : *z* receives update from *y*, updates its DV, computes new least cost to *x* , sends its neighbors its DV.

t_2 : *y* receives *z*'s update, updates its DV. *y*'s least costs do *not* change, so *y* does *not* send a message to *z*.

Distance vector: link cost changes



link cost changes:

- node detects local link cost change
- "bad news travels slow" – count-to-infinity problem:
  - *y* sees direct link to *x* has new cost 60, but z has said it has a path at cost of 5. So *y* computes "my new cost to x will be 6, via z); notifies *z* of new cost of 6 to *x*.
  - *z* learns that path to *x* via *y* has new cost 6, so *z* computes "my new cost to *x* will be 7 via y), notifies *y* of new cost of 7 to *x*.
  - *y* learns that path to *x* via *z* has new cost 7, so *y* computes "my new cost to *x* will be 8 via y), notifies *z* of new cost of 8 to *x*.
  - *z* learns that path to *x* via *y* has new cost 8, so *z* computes "my new cost to *x* will be 9 via y), notifies *y* of new cost of 9 to *x*.

    …
- see text for solutions. *Distributed algorithms are tricky!*

Comparison of LS and DV algorithms

message complexity

LS: n routers, $O(n^2)$ messages sent

DV: exchange between neighbors; convergence time varies

speed of convergence

LS: $O(n^2)$ algorithm, $O(n^2)$ messages

- may have oscillations

DV: convergence time varies

- may have routing loops
- count-to-infinity problem

robustness: what happens if router malfunctions, or is compromised?

LS:

- router can advertise incorrect *link* cost
- each router computes only its *own* table

DV:

- DV router can advertise incorrect *path* cost ("I have a *really* low-cost path to everywhere"): *black-holing*
- each router's DV is used by others: error propagate thru network

Network layer: “control plane” roadmap

- introduction
- routing protocols
- intra-ISP routing: OSPF
- routing among ISPs: BGP
- SDN control plane
- Internet Control Message Protocol
- network management, configuration
  - SNMP
  - NETCONF/YANG

Making routing scalable

our routing study thus far - idealized

- all routers identical
- network "flat"

… not true in practice

scale: billions of destinations:

- can't store all destinations in routing tables!
- routing table exchange would swamp links!

administrative autonomy:

- Internet: a network of networks
- each network admin may want to control routing in its own network

Internet approach to scalable routing

aggregate routers into regions known as "autonomous systems" (AS) (a.k.a. "domains")

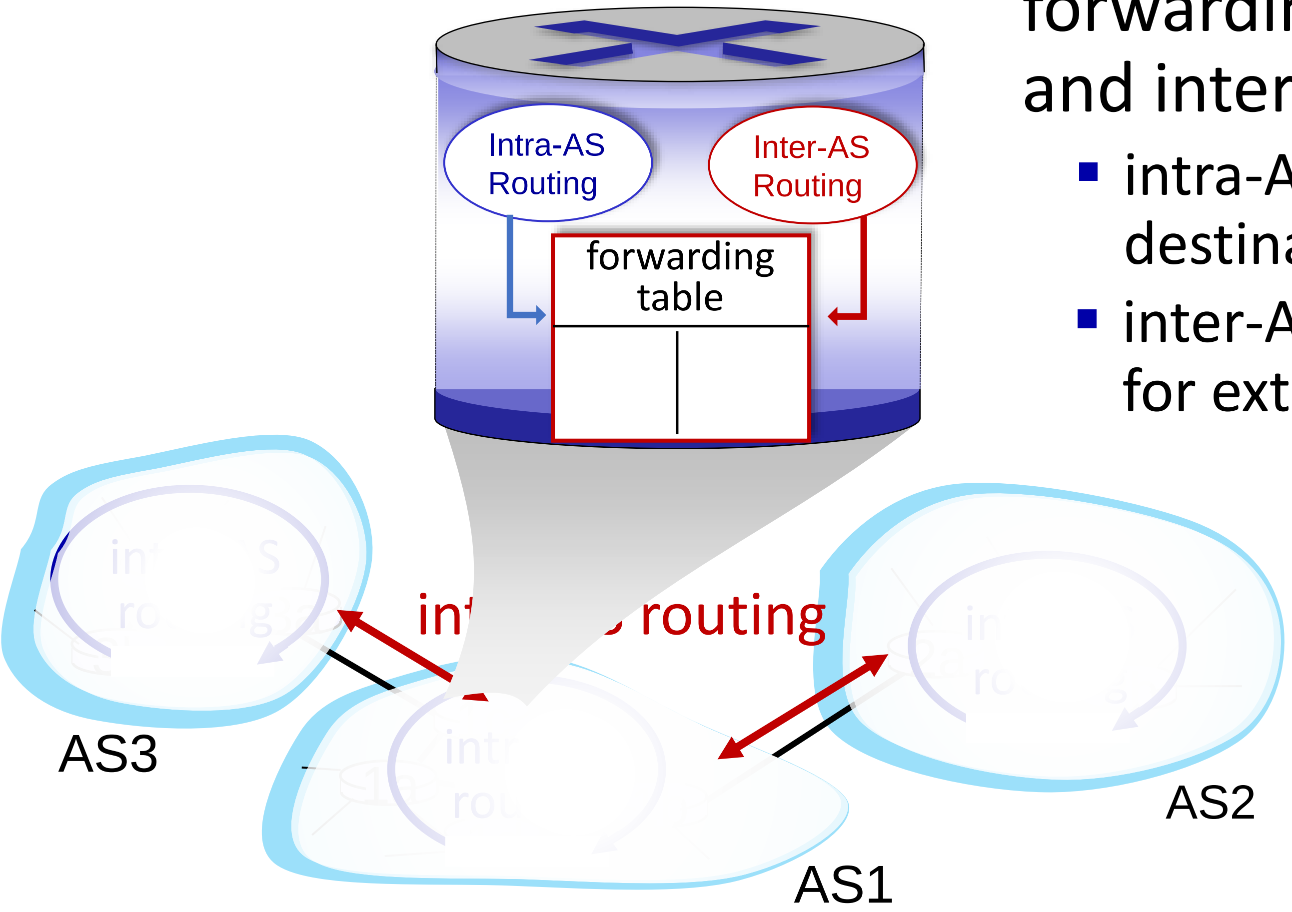
intra-AS (aka "intra-domain"):
routing among routers *within same AS ("network")*

- all routers in AS must run same intra-domain protocol
- routers in different AS can run different intra-domain routing protocols
- gateway router: at "edge" of its own AS, has link(s) to router(s) in other AS'es

inter-AS (aka "inter-domain"):
routing *among* AS'es

- gateways perform inter-domain routing (as well as intra-domain routing)

Interconnected ASes



forwarding table configured by intra- and inter-AS routing algorithms

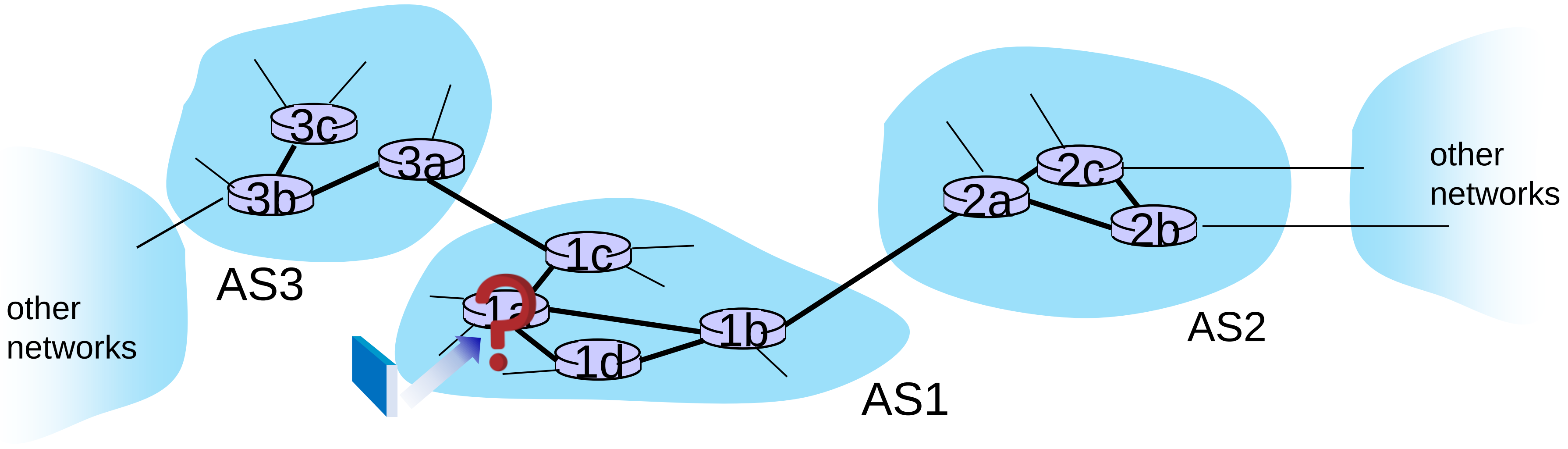
- intra-AS routing determine entries for destinations within AS
- inter-AS & intra-AS determine entries for external destinations

Inter-AS routing: a role in intradomain forwarding

- suppose router in AS1 receives datagram destined outside of AS1:
  - router should forward packet to gateway router in AS1, but which one?

AS1 inter-domain routing must:

1. learn which destinations reachable through AS2, which through AS3
2. propagate this reachability info to all routers in AS1



Intra-AS routing: routing within an AS

most common intra-AS routing protocols:

- RIP: Routing Information Protocol [RFC 1723]
  - classic DV: DVs exchanged every 30 secs
  - no longer widely used
- EIGRP: Enhanced Interior Gateway Routing Protocol
  - DV based
  - formerly Cisco-proprietary for decades (became open in 2013 [RFC 7868])
- OSPF: Open Shortest Path First [RFC 2328]
  - link-state routing
  - IS-IS protocol (ISO standard, not RFC standard) essentially same as OSPF

OSPF (Open Shortest Path First) routing

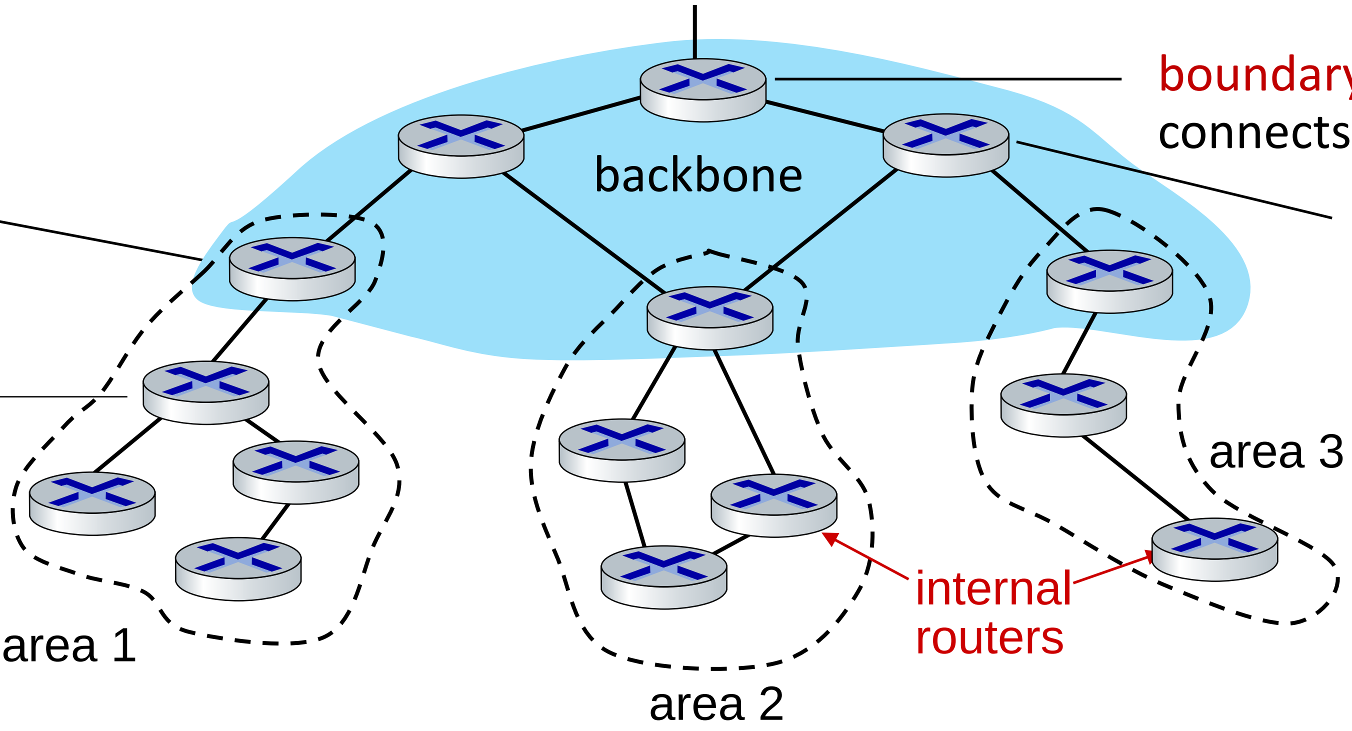
- “open”: publicly available
- classic link-state
  - each router floods OSPF link-state advertisements (directly over IP rather than using TCP/UDP) to all other routers in entire AS
  - multiple link costs metrics possible: bandwidth, delay
  - each router has full topology, uses Dijkstra’s algorithm to compute forwarding table
- *security:* all OSPF messages authenticated (to prevent malicious intrusion)

Hierarchical OSPF

- two-level hierarchy: local area, backbone.
  - link-state advertisements flooded only in area, or backbone
  - each node has detailed area topology; only knows direction to reach other destinations

area border routers:
"summarize" distances to destinations in own area, advertise in backbone

local routers:
- flood LS in area only
- compute routing within area
- forward packets to outside via area border router



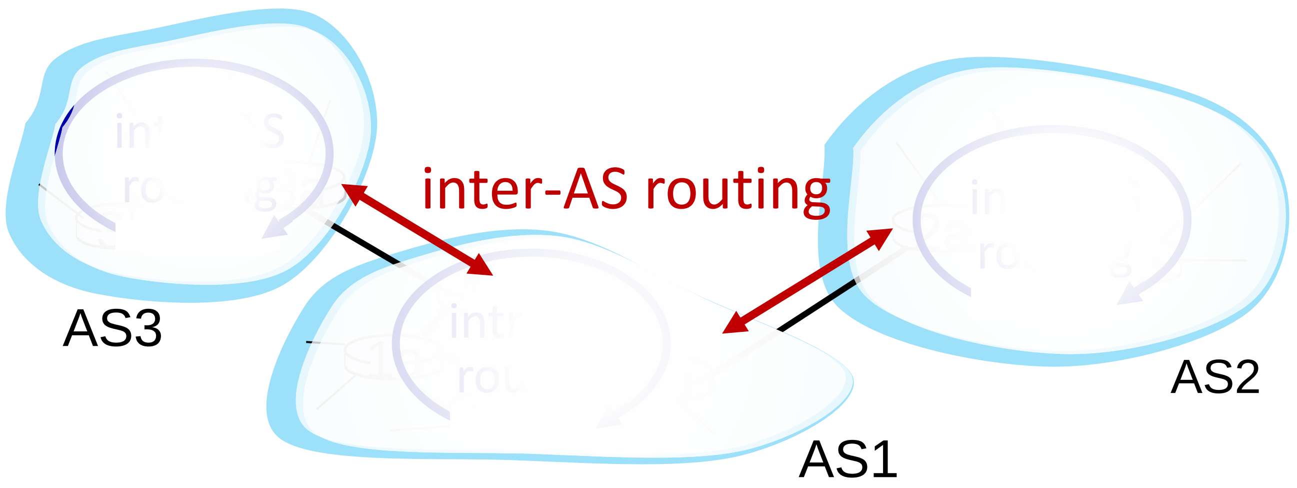
boundary router:
connects to other ASes

backbone router:
runs OSPF limited to backbone

Network layer: “control plane” roadmap

- introduction
- routing protocols
- intra-ISP routing: OSPF
- routing among ISPs: BGP
- SDN control plane
- Internet Control Message Protocol
- network management, configuration
  - SNMP
  - NETCONF/YANG

Interconnected ASes

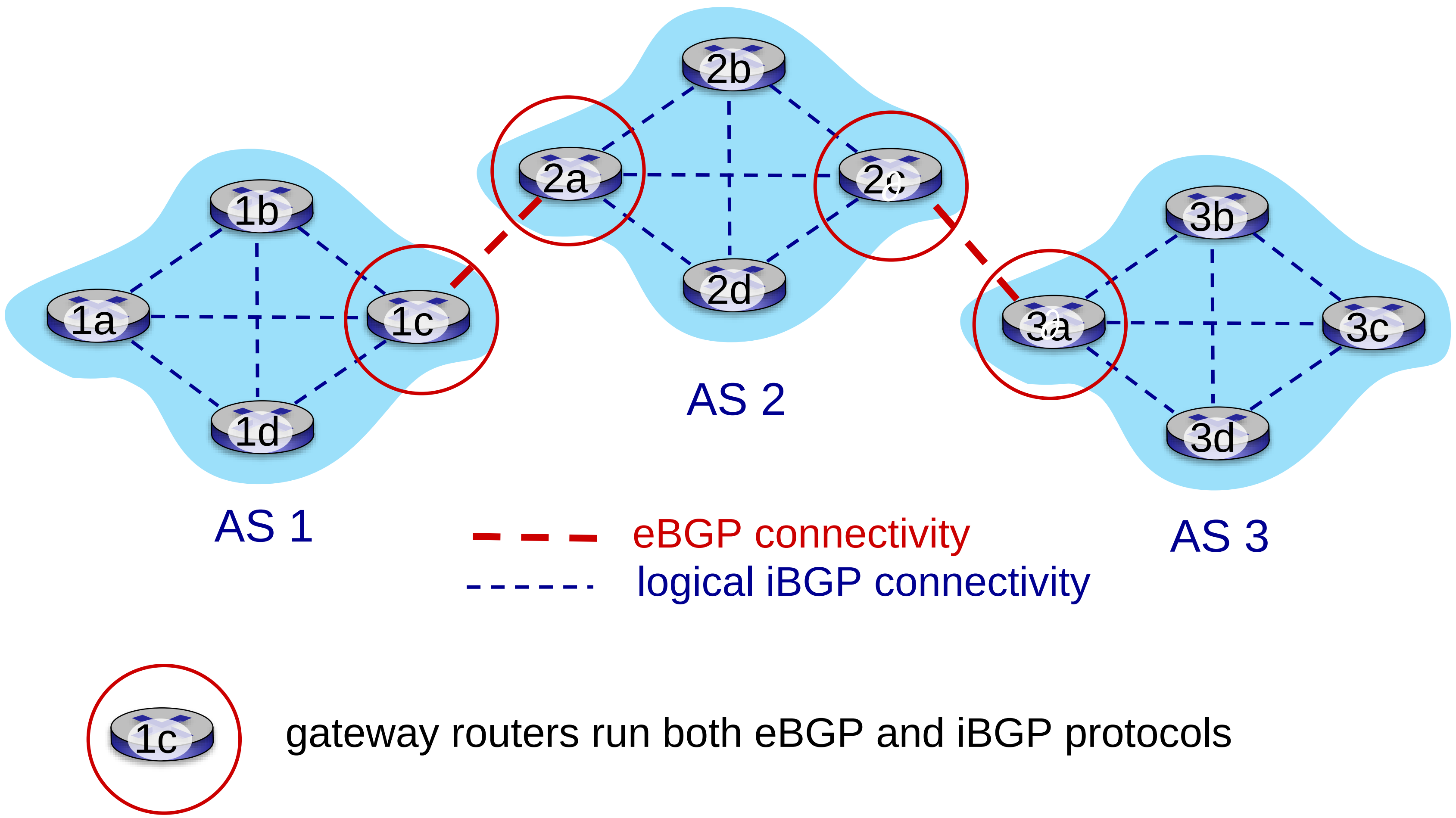


- intra-AS (aka "intra-domain"): routing among routers *within same AS ("network")*
- inter-AS (aka "inter-domain"): routing *among* AS'es

Internet inter-AS routing: BGP

- BGP (Border Gateway Protocol): *the* de facto inter-domain routing protocol
  - "glue that holds the Internet together"
- allows subnet to advertise its existence, and the destinations it can reach, to rest of Internet: *"I am here, here is who I can reach, and how"*
- BGP provides each AS a means to:
  - obtain destination network reachability info from neighboring ASes (eBGP)
  - determine routes to other networks based on reachability information and *policy*
  - propagate reachability information to all AS-internal routers (iBGP)
  - advertise (to neighboring networks) destination reachability info

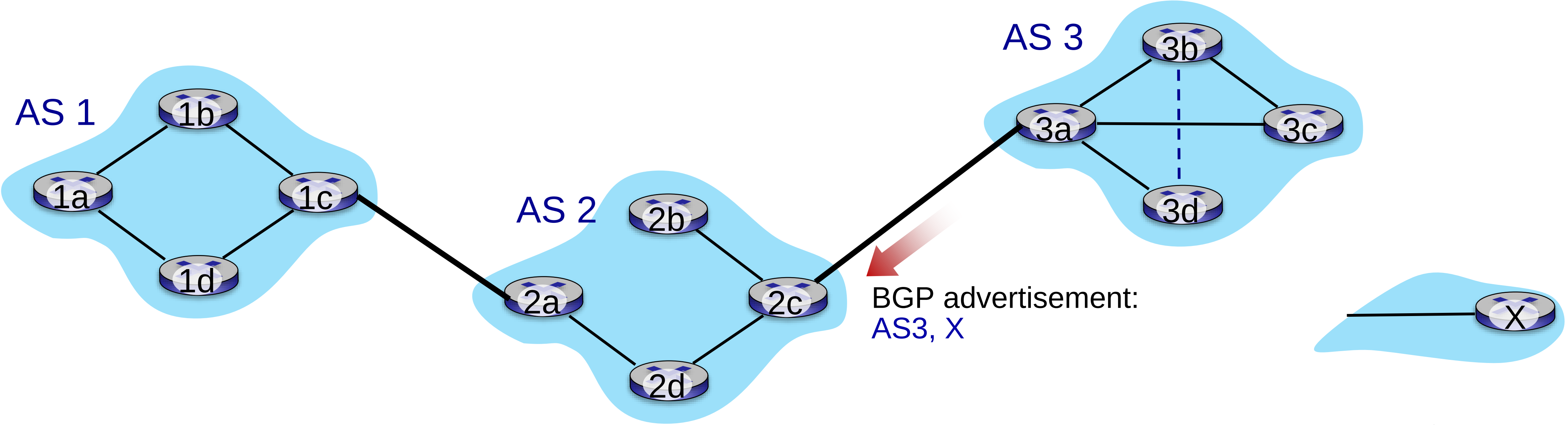
eBGP, iBGP connections



gateway routers run both eBGP and iBGP protocols

BGP basics

- BGP session: two BGP routers ("peers") exchange BGP messages over semi-permanent TCP connection:
  - advertising *paths* to different destination network prefixes (BGP is a "path vector" protocol)
- when AS3 gateway 3a advertises path AS3,X to AS2 gateway 2c:
  - AS3 *promises* to AS2 it will forward datagrams towards X



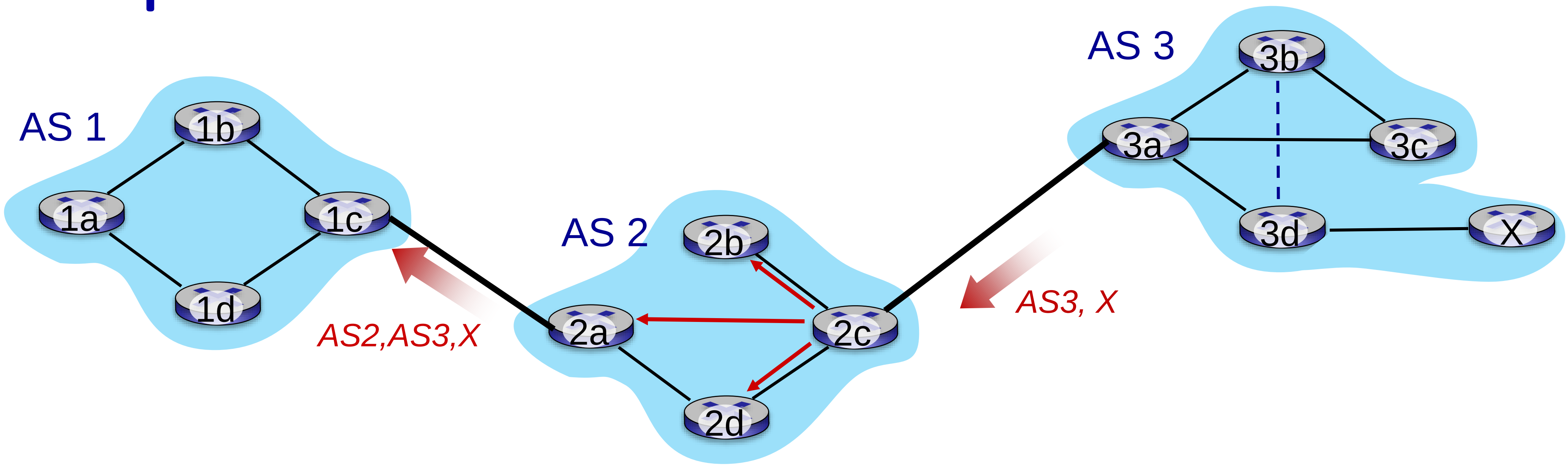
BGP protocol messages

- BGP messages exchanged between peers over TCP connection
- BGP messages [RFC 4371]:
  - OPEN: opens TCP connection to remote BGP peer and authenticates sending BGP peer
  - **UPDATE:** advertises new path (or withdraws old)
  - KEEPALIVE: keeps connection alive in absence of UPDATES; also ACKs OPEN request
  - NOTIFICATION: reports errors in previous msg; also used to close connection

Path attributes and BGP routes

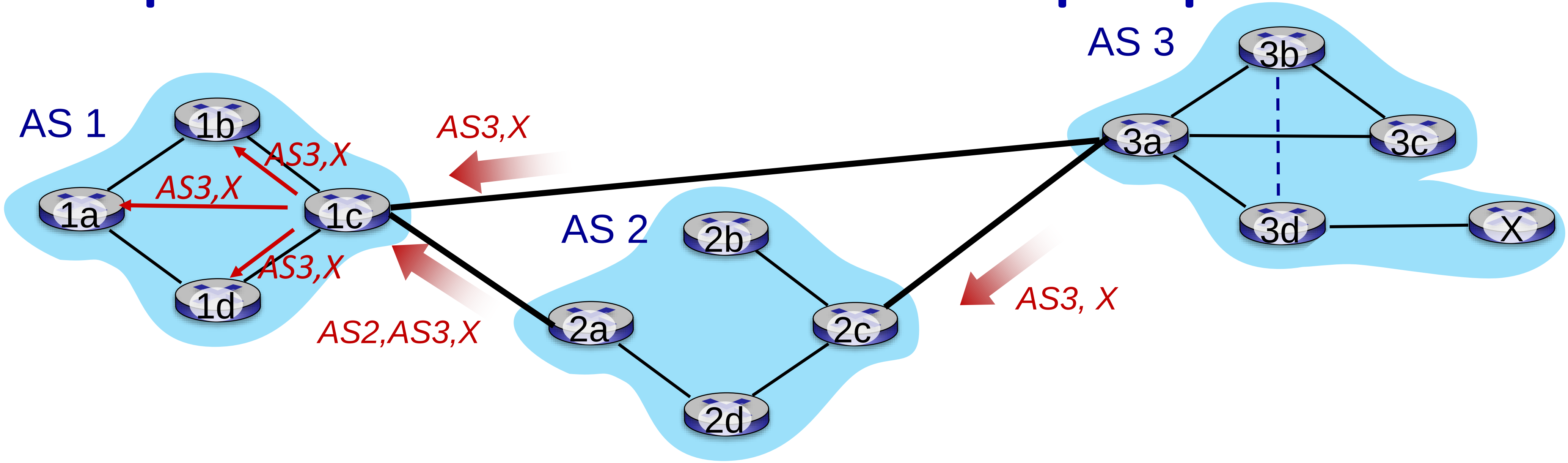
- BGP advertised route: prefix + attributes
  - prefix: destination being advertised
  - two important attributes:
    - AS-PATH: list of ASes through which prefix advertisement has passed
    - NEXT-HOP: indicates specific internal-AS router to next-hop AS
- policy-based routing:
  - gateway receiving route advertisement uses *import policy* to accept/decline path (e.g., never route through AS Y).
  - AS policy also determines whether to *advertise* path to other other neighboring ASes

BGP path advertisement



- AS2 router 2c receives path advertisement AS3,X (via eBGP) from AS3 router 3a
- based on AS2 policy, AS2 router 2c accepts path AS3,X, propagates (via iBGP) to all AS2 routers
- based on AS2 policy, AS2 router 2a advertises (via eBGP) path AS2, AS3, X to AS1 router 1c

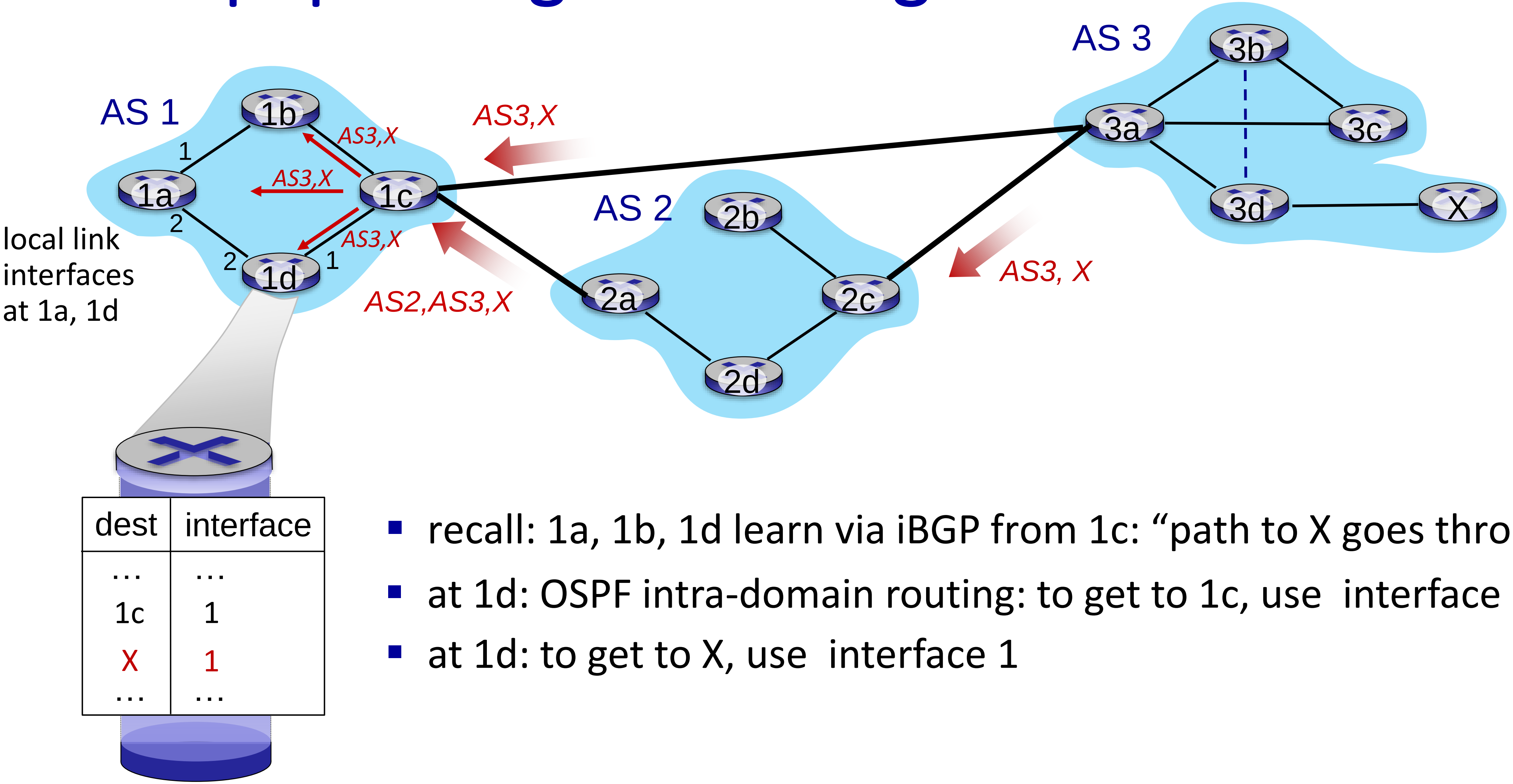
BGP path advertisement: multiple paths



gateway router may learn about multiple paths to destination:

- AS1 gateway router 1c learns path *AS2,AS3,X* from 2a
- AS1 gateway router 1c learns path *AS3,X* from 3a
- based on *policy,* AS1 gateway router 1c chooses path *AS3,X* and advertises path within AS1 via iBGP

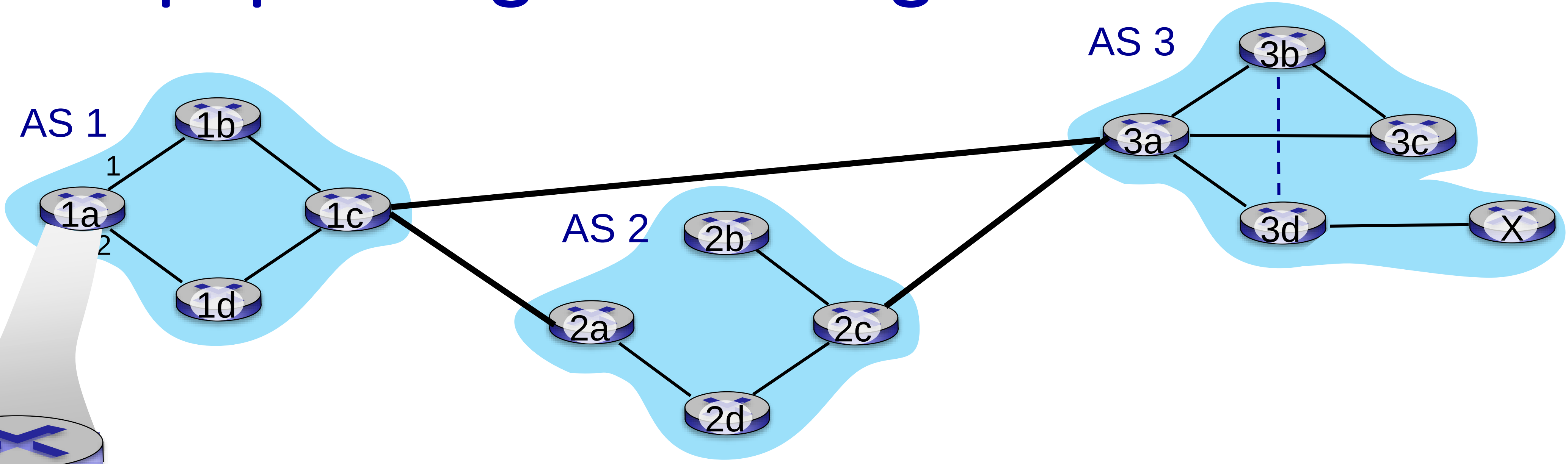
BGP: populating forwarding tables



dest	interface
…	…
1c	1
X	1
…	…

- recall: 1a, 1b, 1d learn via iBGP from 1c: "path to X goes through 1c"
- at 1d: OSPF intra-domain routing: to get to 1c, use interface 1
- at 1d: to get to X, use interface 1

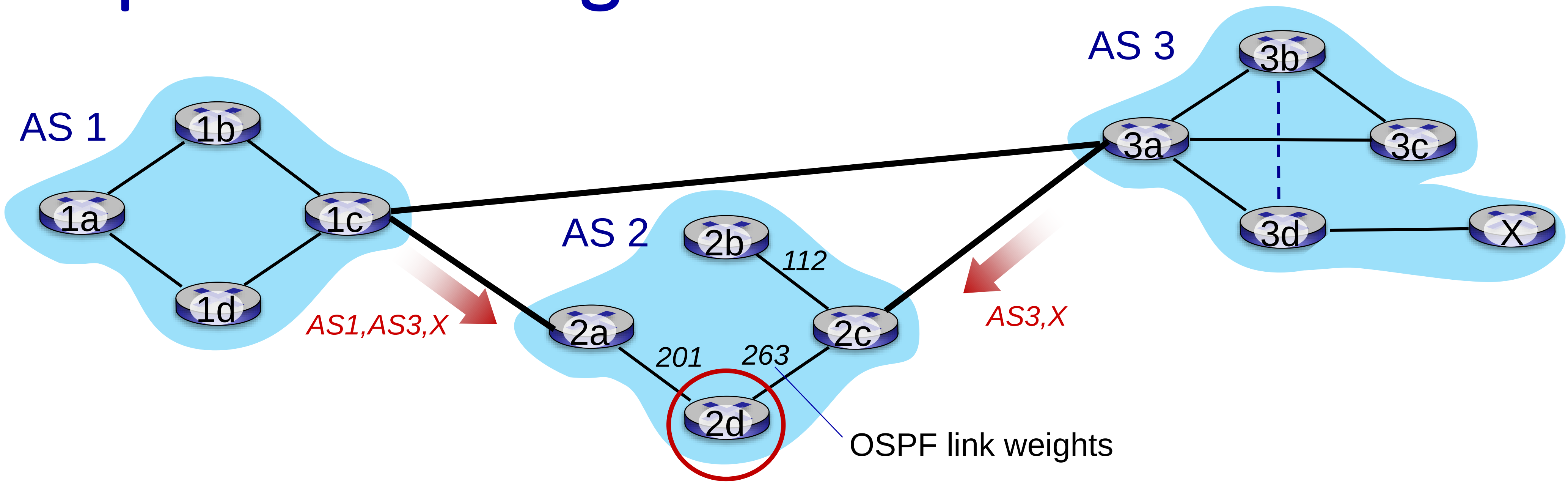
BGP: populating forwarding tables



dest	interface
…	…
1c	2
X	2
…	…

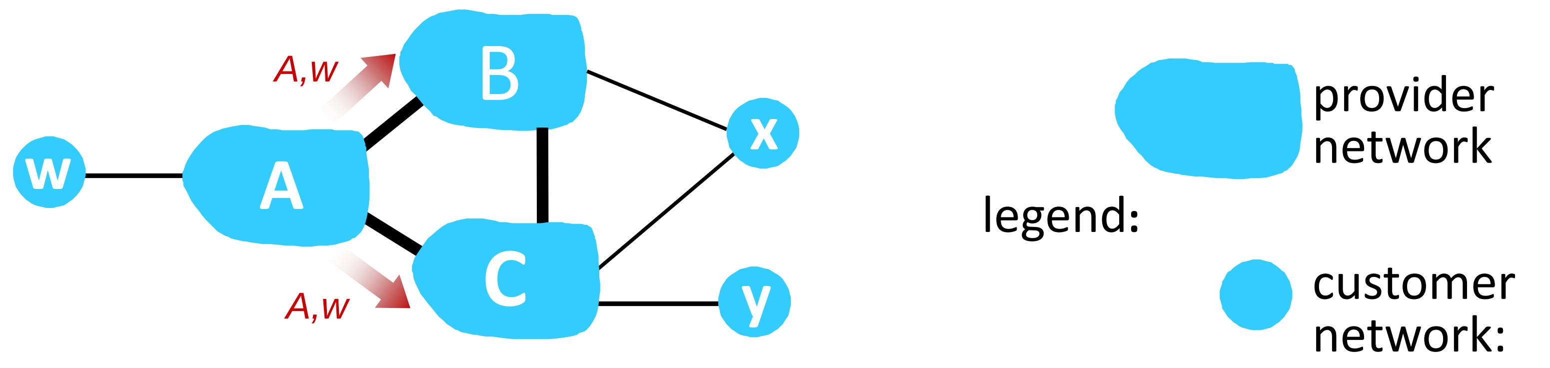
- recall: 1a, 1b, 1d learn via iBGP from 1c: “path to X goes through 1c”
- at 1d: OSPF intra-domain routing: to get to 1c, use interface 1
- at 1d: to get to X, use interface 1
- at 1a: OSPF intra-domain routing: to get to 1c, use interface 2
- at 1a: to get to X, use interface 2

Hot potato routing



- 2d learns (via iBGP) it can route to X via 2a or 2c
- hot potato routing: choose local gateway that has least *intra-domain* cost (e.g., 2d chooses 2a, even though more AS hops to *X*): don't worry about inter-domain cost!

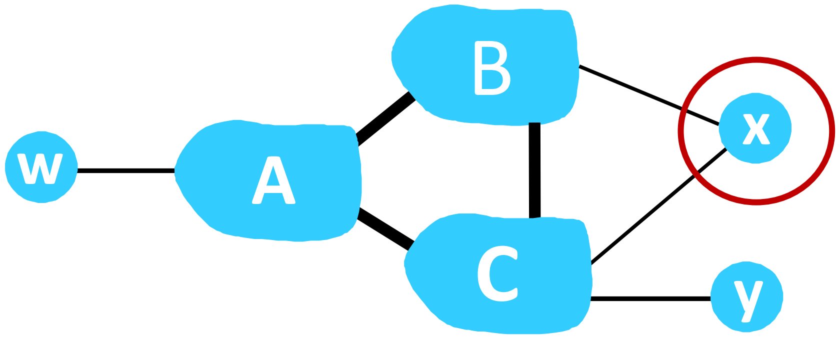
BGP: achieving policy via advertisements

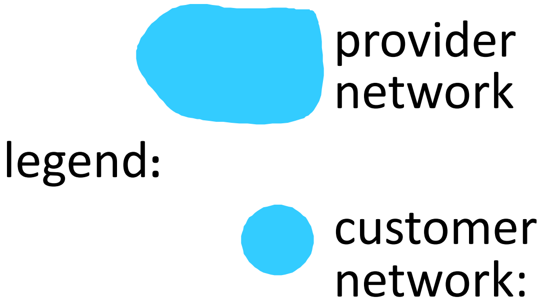


ISP only wants to route traffic to/from its customer networks (does not want to carry transit traffic between other ISPs – a typical "real world" policy)

- A advertises path Aw to B and to C
- B *chooses not to advertise* BAw to C!
  - B gets no "revenue" for routing CBAw, since none of C, A, w are B's customers
  - C does *not* learn about CBAw path
- C will route CAw (not using B) to get to w

BGP: achieving policy via advertisements (more)





ISP only wants to route traffic to/from its customer networks (does not want to carry transit traffic between other ISPs – a typical "real world" policy)

- A,B,C are provider networks
- x,w,y are customer (of provider networks)
- x is dual-homed: attached to two networks
- *policy to enforce:* x does not want to route from B to C via x
  - .. so x will not advertise to B a route to C

BGP route selection

- router may learn about more than one route to destination AS, selects route based on:
  1. local preference value attribute: policy decision
  2. shortest AS-PATH
  3. closest NEXT-HOP router: hot potato routing
  4. additional criteria

Why different Intra-, Inter-AS routing ?

policy:

- inter-AS: admin wants control over how its traffic routed, who routes through its network
- intra-AS: single admin, so policy less of an issue

scale:

- hierarchical routing saves table size, reduced update traffic

performance:

- intra-AS: can focus on performance
- inter-AS: policy dominates over performance

Network layer: “control plane” roadmap

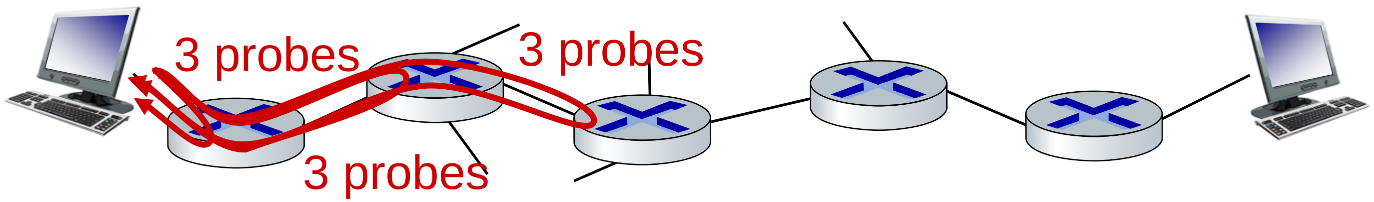
- introduction
- routing protocols
- intra-ISP routing: OSPF
- routing among ISPs: BGP
- SDN control plane
- **Internet Control Message Protocol**
- network management, configuration
  - SNMP
  - NETCONF/YANG

ICMP: internet control message protocol

- used by hosts and routers to communicate network-level information
  - error reporting: unreachable host, network, port, protocol
  - echo request/reply (used by ping)
- network-layer "above" IP:
  - ICMP messages carried in IP datagrams
- *ICMP message:* type, code plus first 8 bytes of IP datagram causing error

Type	Code	description
0	0	echo reply (ping)
3	0	dest. network unreachable
3	1	dest host unreachable
3	2	dest protocol unreachable
3	3	dest port unreachable
3	6	dest network unknown
3	7	dest host unknown
4	0	source quench (congestion control - not used)
8	0	echo request (ping)
9	0	route advertisement
10	0	router discovery
11	0	TTL expired
12	0	bad IP header

Traceroute and ICMP



- source sends sets of UDP segments to destination
  - 1st set has TTL =1, 2nd set has TTL=2, etc.
- datagram in *n*th set arrives to nth router:
  - router discards datagram and sends source ICMP message (type 11, code 0)
  - ICMP message possibly includes name of router & IP address
- when ICMP message arrives at source: record RTTs

stopping criteria:

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

Chapter 6
The Link Layer and LANs

Link layer and LANs: our goals

- understand principles behind link layer services:
  - error detection, correction
  - sharing a broadcast channel: multiple access
  - link layer addressing
  - local area networks: Ethernet, VLANs
- datacenter networks

- instantiation, implementation of various link layer technologies

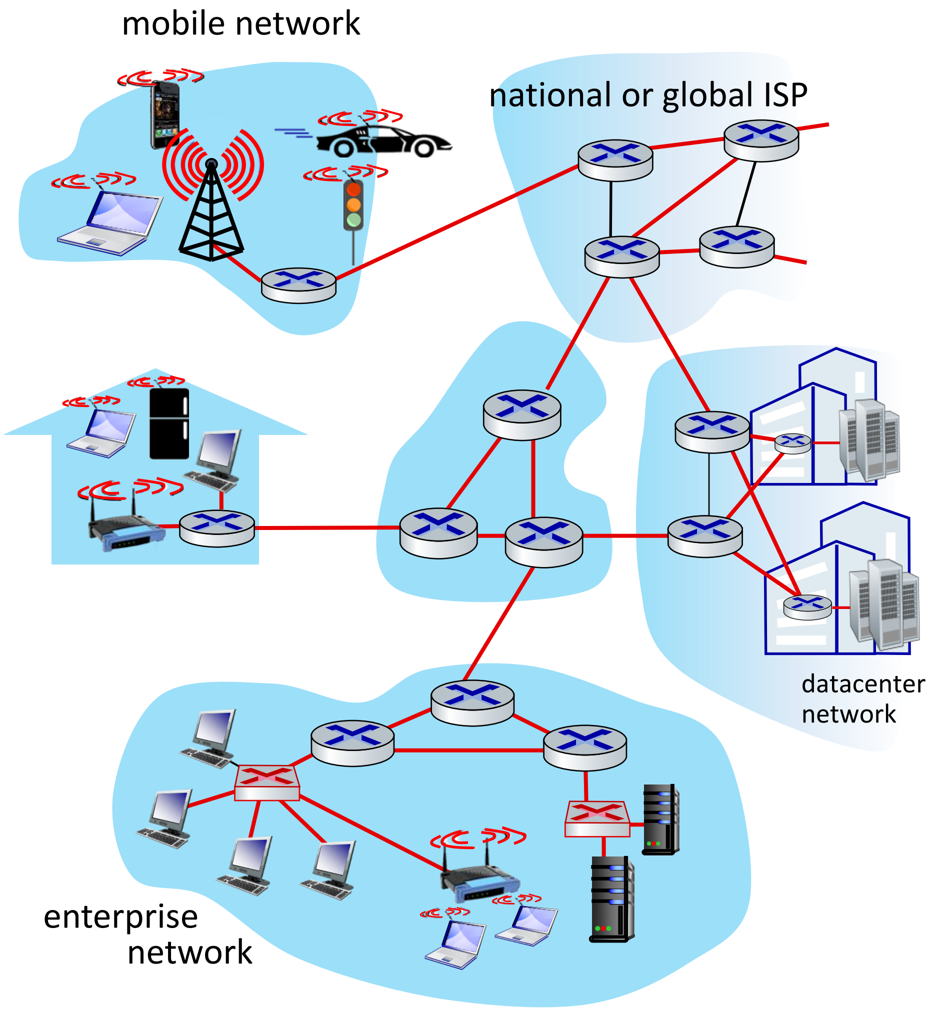
Link layer, LANs: roadmap

- introduction
- error detection, correction
- multiple access protocols
- LANs
  - addressing, ARP
  - Ethernet
  - switches
  - VLANs
- link virtualization: MPLS
- data center networking
- a day in the life of a web request

Link layer: introduction

terminology:

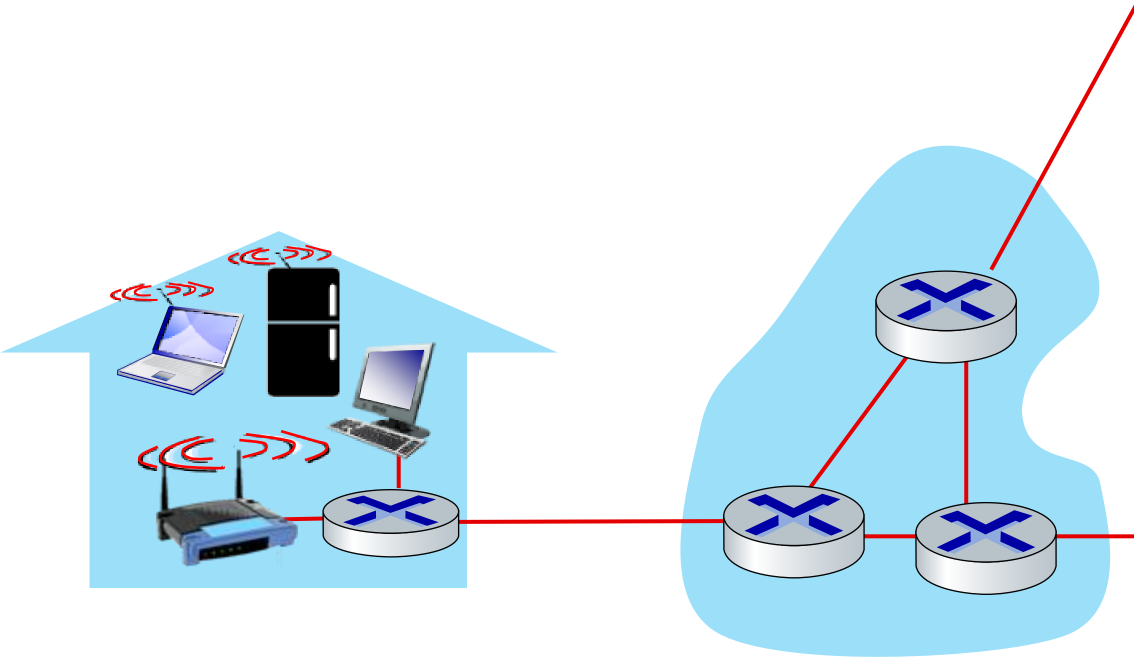
- hosts, routers: nodes
- communication channels that connect *adjacent* nodes along communication path: links
  - wired , wireless
  - LANs
- layer-2 packet: *frame*, encapsulates datagram



link layer has responsibility of transferring datagram from one node to physically adjacent node over a link

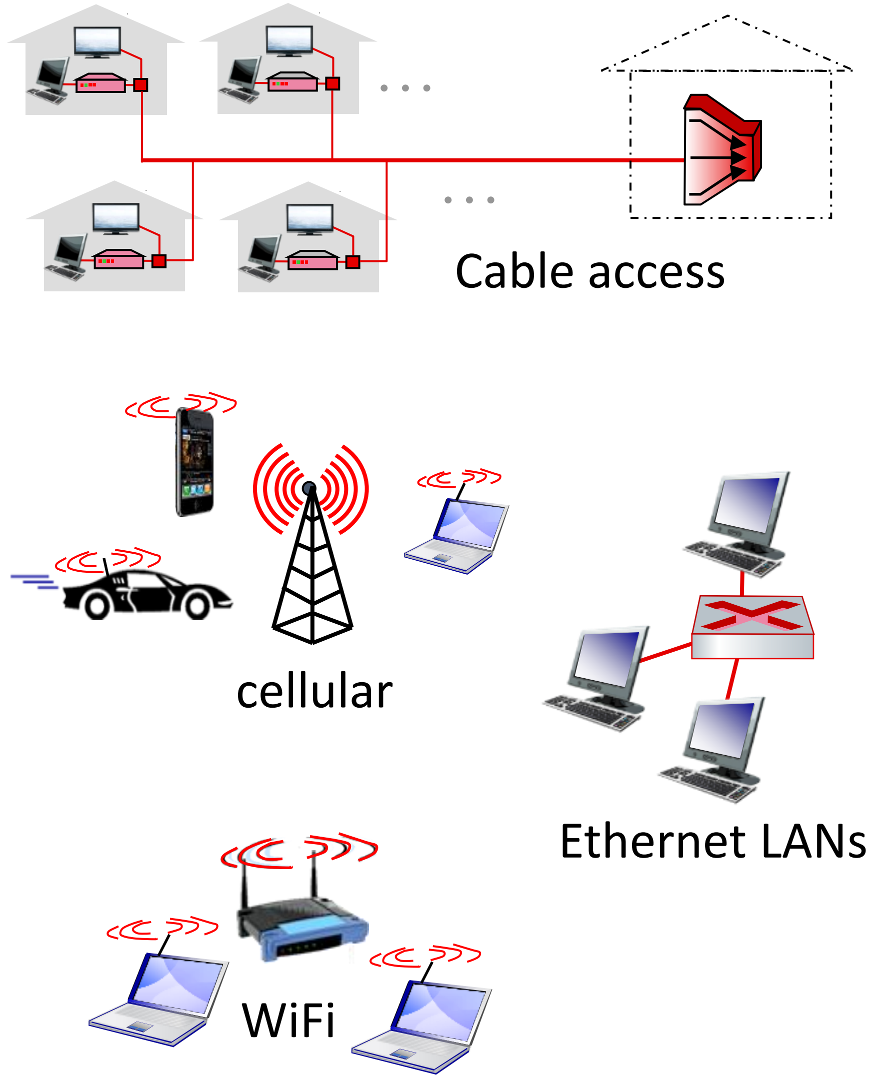
Link layer: context

- datagram transferred by different link protocols over different links:
  - e.g., WiFi on first link, Ethernet on next link
- each link protocol provides different services
  - e.g., may or may not provide reliable data transfer over link



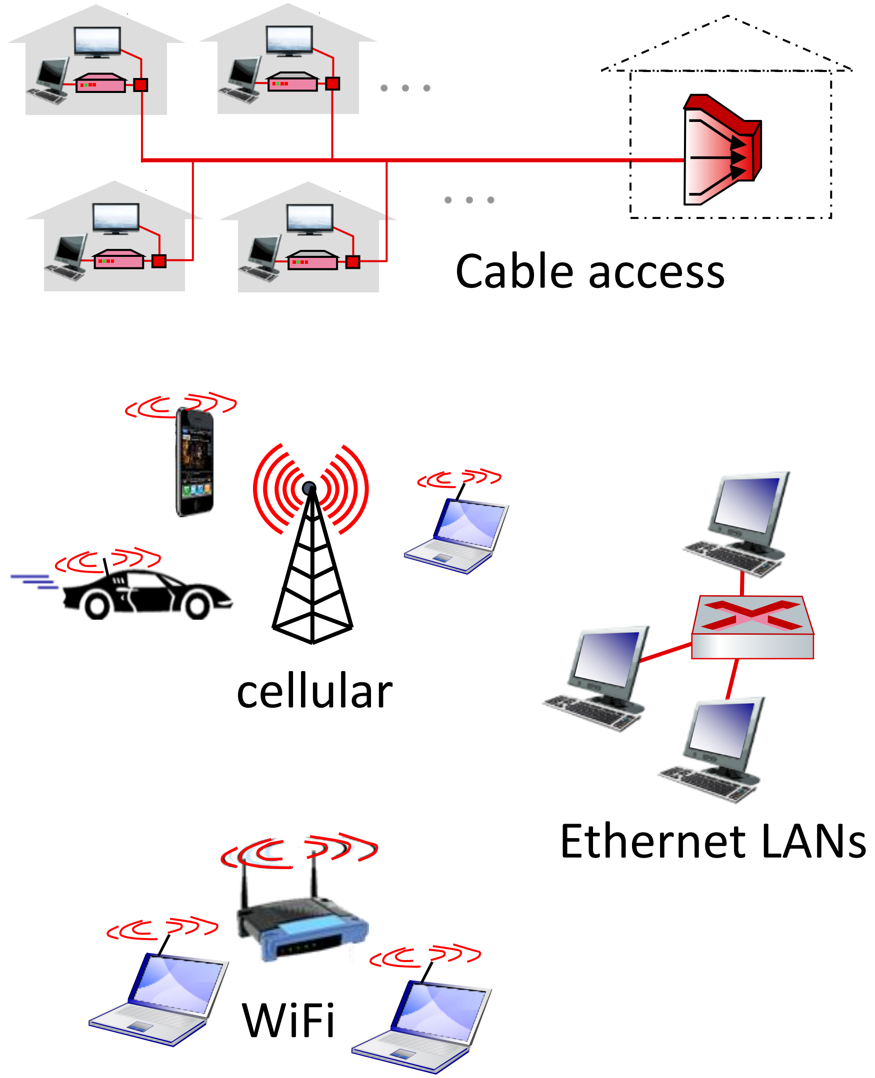
Link layer: services

- framing, link access:
  - encapsulate datagram into frame, adding header, trailer
  - channel access if shared medium
  - “MAC” addresses in frame headers identify source, destination (different from IP address!)
- reliable delivery between adjacent nodes
  - we already know how to do this!
  - seldom used on low bit-error links
  - wireless links: high error rates
    - ***Q:*** **why both link-level and end-end reliability?**



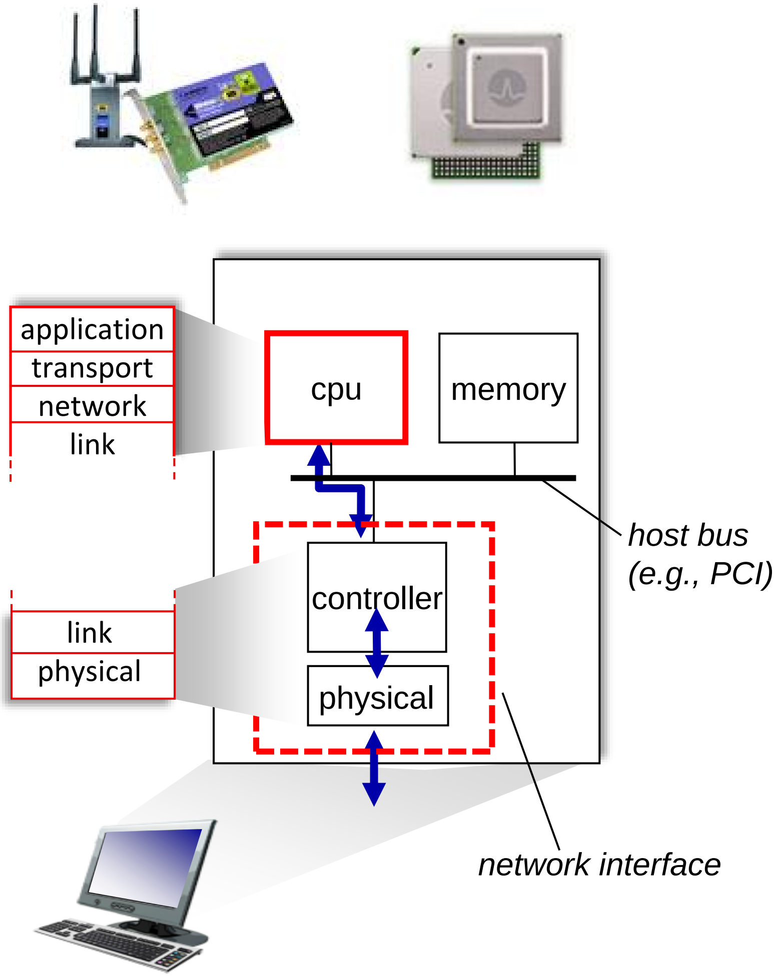
Link layer: services (more)

- flow control:
  - pacing between adjacent sending and receiving nodes
- error detection:
  - errors caused by signal attenuation, noise.
  - receiver detects errors, signals retransmission, or drops frame
- error correction:
  - receiver identifies *and corrects* bit error(s) without retransmission
- half-duplex and full-duplex:
  - with half duplex, nodes at both ends of link can transmit, but not at same time

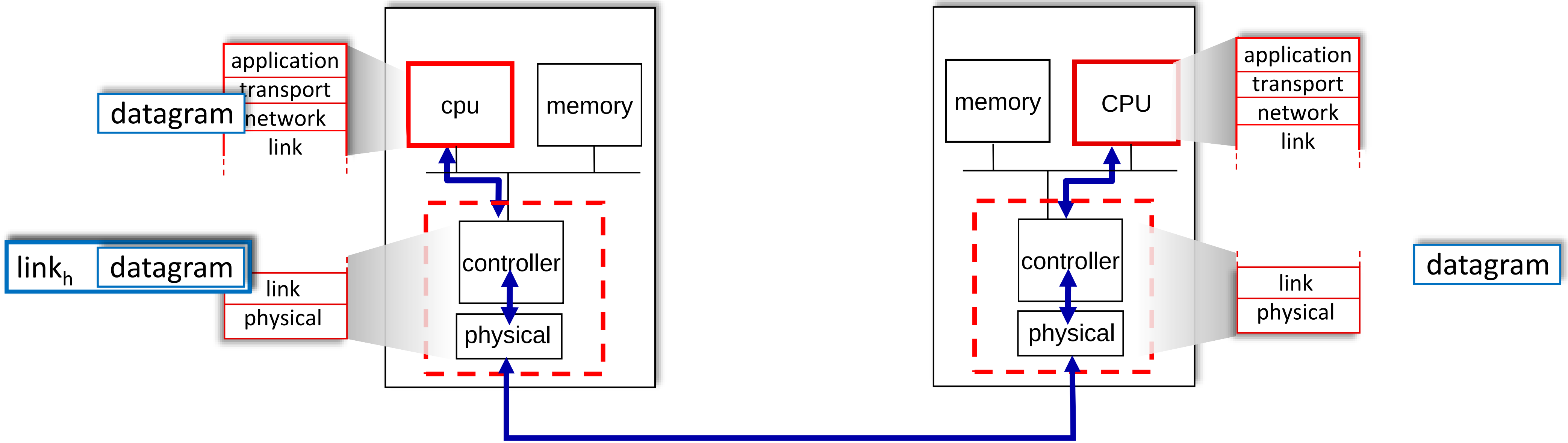


Host link-layer implementation

- in each-and-every host
- link layer implemented on-chip or in network interface card (NIC)
  - implements link, physical layer
- attaches into host's system buses
- combination of hardware, software, firmware



Interfaces communicating



sending side:

- encapsulates datagram in frame
- adds error checking bits, reliable data transfer, flow control, etc.

receiving side:

- looks for errors, reliable data transfer, flow control, etc.
- extracts datagram, passes to upper layer at receiving side

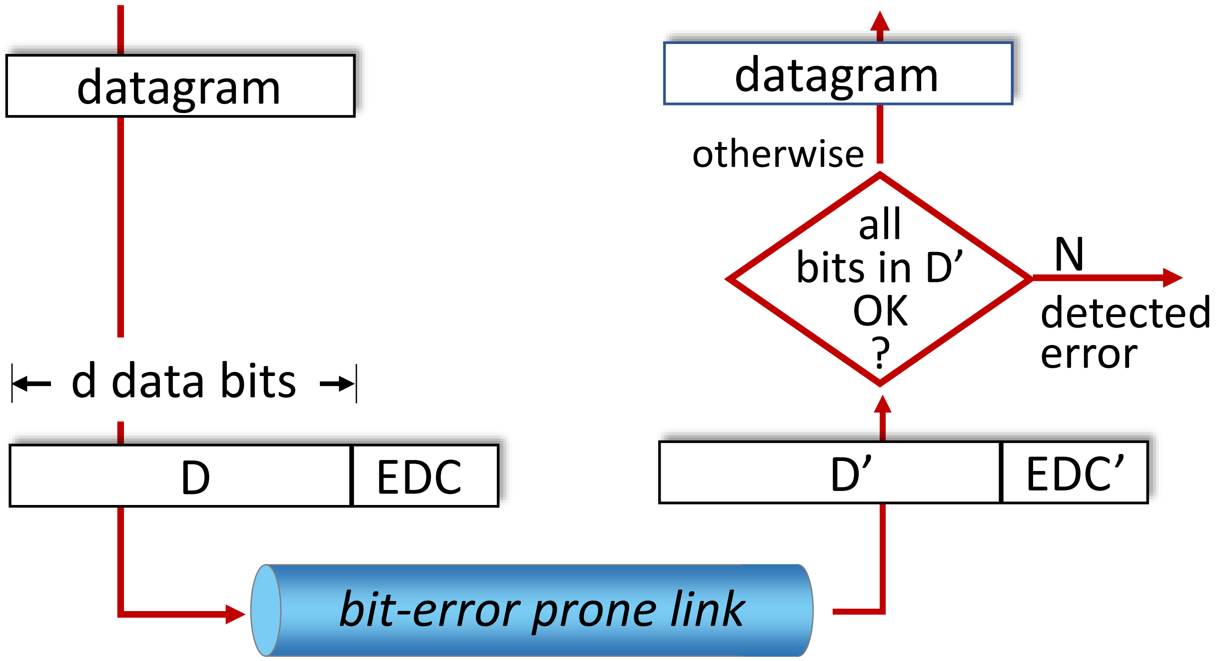
Link layer, LANs: roadmap

- introduction
- error detection, correction
- multiple access protocols
- LANs
  - addressing, ARP
  - Ethernet
  - switches
  - VLANs
- link virtualization: MPLS
- data center networking
- a day in the life of a web request

Error detection

EDC: error detection and correction bits (e.g., redundancy)

D: data protected by error checking, may include header fields



Error detection not 100% reliable!

- protocol may miss some errors, but rarely
- larger EDC field yields better detection and correction

Parity checking

single bit parity:

- detect single bit errors

0111000110101011	1

← *d* data bits → | parity bit

Even/odd parity: set parity bit so there is an even/odd number of 1's

At receiver:

- compute parity of *d* received bits
- compare with received parity bit – if different than error detected

Can detect *and* correct errors (without retransmission!)

- two-dimensional parity: detect *and correct* single bit errors

row parity →

column parity ↓

$$\begin{array}{cccc|c} d_{1,1} & \cdots & d_{1,j} & & d_{1,j+1} \\ d_{2,1} & \cdots & d_{2,j} & & d_{2,j+1} \\ \cdots & \cdots & \cdots & & \cdots \\ d_{i,1} & \cdots & d_{i,j} & & d_{i,j+1} \\ \hline d_{i+1,1} & \cdots & d_{i+1,j} & & d_{i+1,j+1} \end{array}$$

no errors:

$$\begin{array}{ccccc|c} 1 & 0 & 1 & 0 & 1 & 1 \\ 1 & 1 & 1 & 1 & 0 & 0 \\ 0 & 1 & 1 & 1 & 0 & 1 \\ \hline 1 & 0 & 1 & 0 & 1 & 0 \end{array}$$

detected and correctable single-bit error:

$$\begin{array}{ccccc|c} 1 & 0 & 1 & 0 & 1 & 1 \\ 1 & 0 & 1 & 1 & 0 & 0 \\ 0 & 1 & 1 & 1 & 0 & 1 \\ \hline 1 & 0 & 1 & 0 & 1 & 0 \end{array}$$

parity error

parity error

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

Internet checksum (review, see section 3.3)

Goal: detect errors (*i.e.,* flipped bits) in transmitted segment

sender:

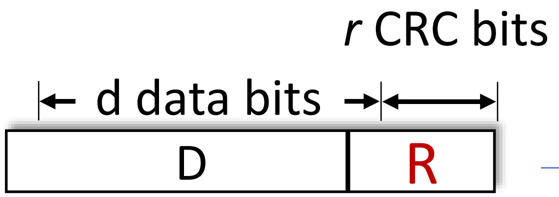
- treat contents of UDP segment (including UDP header fields and IP addresses) as sequence of 16-bit integers
- checksum: addition (one's complement sum) of segment content
- checksum value put into UDP checksum field

receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - not equal - error detected
  - equal - no error detected. *But maybe errors nonetheless?* More later ….

Cyclic Redundancy Check (CRC)

- more powerful error-detection coding
- D: data bits (given, think of these as a binary number)
- G: bit pattern (generator), of *r+1* bits (given, specified in CRC standard)



bits to send

<D,R> = $D*2^r$ XOR R — formula for these bits

sender: compute *r* CRC bits, R, such that <D,R> *exactly* divisible by G (mod 2)

- receiver knows G, divides <D,R> by G. If non-zero remainder: error detected!
- can detect all burst errors less than r+1 bits
- widely used in practice (Ethernet, 802.11 WiFi)

Cyclic Redundancy Check (CRC): example

Sender wants to compute R such that:

$D \cdot 2^r \ XOR \ R = nG$

... or equivalently (XOR R both sides):

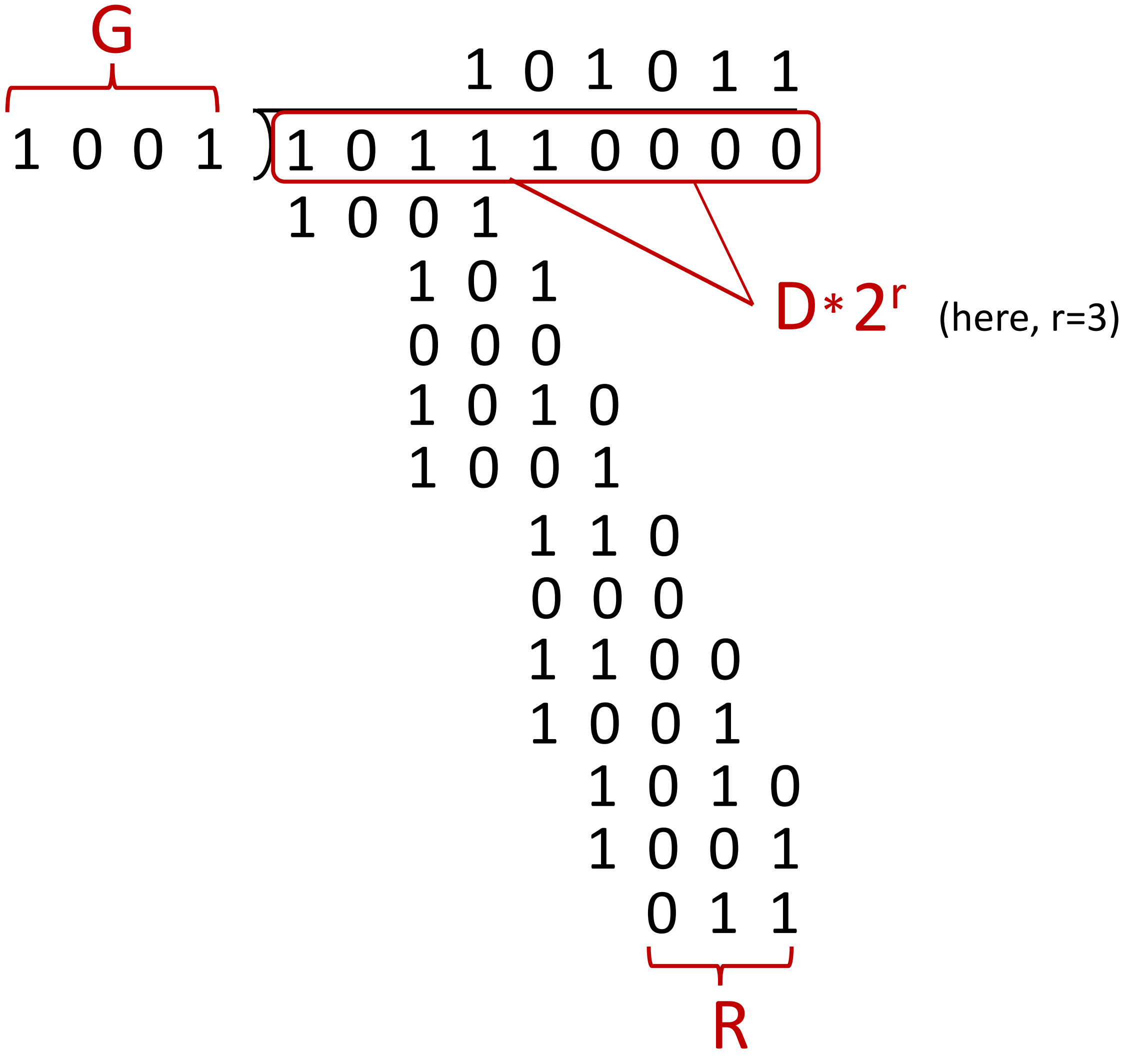
$D \cdot 2^r = nG \ XOR \ R$

... which says:

if we divide $D \cdot 2^r$ by G, we want remainder R to satisfy:

$$R = remainder\left[\frac{D \cdot 2^r}{G}\right]$$

algorithm for computing R



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Link layer, LANs: roadmap

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Multiple access links, protocols

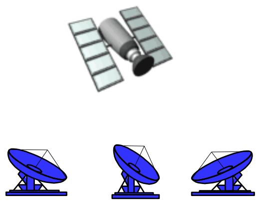
two types of "links":

- point-to-point
  - point-to-point link between Ethernet switch, host
  - PPP for dial-up access
- broadcast (shared wire or medium)
  - old-school Ethernet
  - upstream HFC in cable-based access network
  - 802.11 wireless LAN, 4G/4G. satellite

shared wire (e.g., cabled Ethernet)

shared radio: 4G/5G

shared radio: WiFi



shared radio: satellite

humans at a cocktail party (shared air, acoustical)

Multiple access protocols

- single shared broadcast channel
- two or more simultaneous transmissions by nodes: interference
  - *collision* if node receives two or more signals at the same time

multiple access protocol

- distributed algorithm that determines how nodes share channel, i.e., determine when node can transmit
- communication about channel sharing must use channel itself!
  - no out-of-band channel for coordination

An ideal multiple access protocol

given: multiple access channel (MAC) of rate R bps

desiderata:

1. when one node wants to transmit, it can send at rate R.
2. when M nodes want to transmit, each can send at average rate R/M
3. fully decentralized:
   - no special node to coordinate transmissions
   - no synchronization of clocks, slots
4. simple

MAC protocols: taxonomy

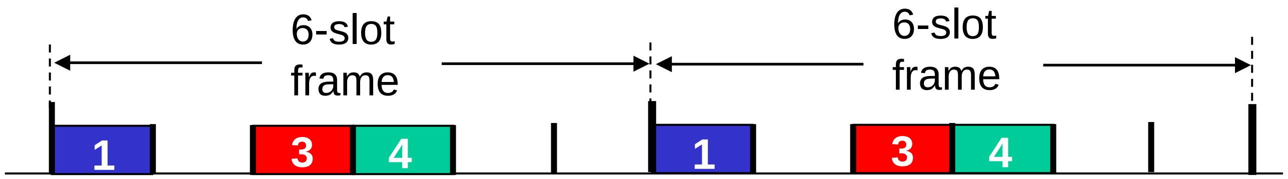
three broad classes:

- channel partitioning
  - divide channel into smaller “pieces” (time slots, frequency, code)
  - allocate piece to node for exclusive use
- random access
  - channel not divided, allow collisions
  - “recover” from collisions
- “taking turns”
  - nodes take turns, but nodes with more to send can take longer turns

Channel partitioning MAC protocols: TDMA

TDMA: time division multiple access

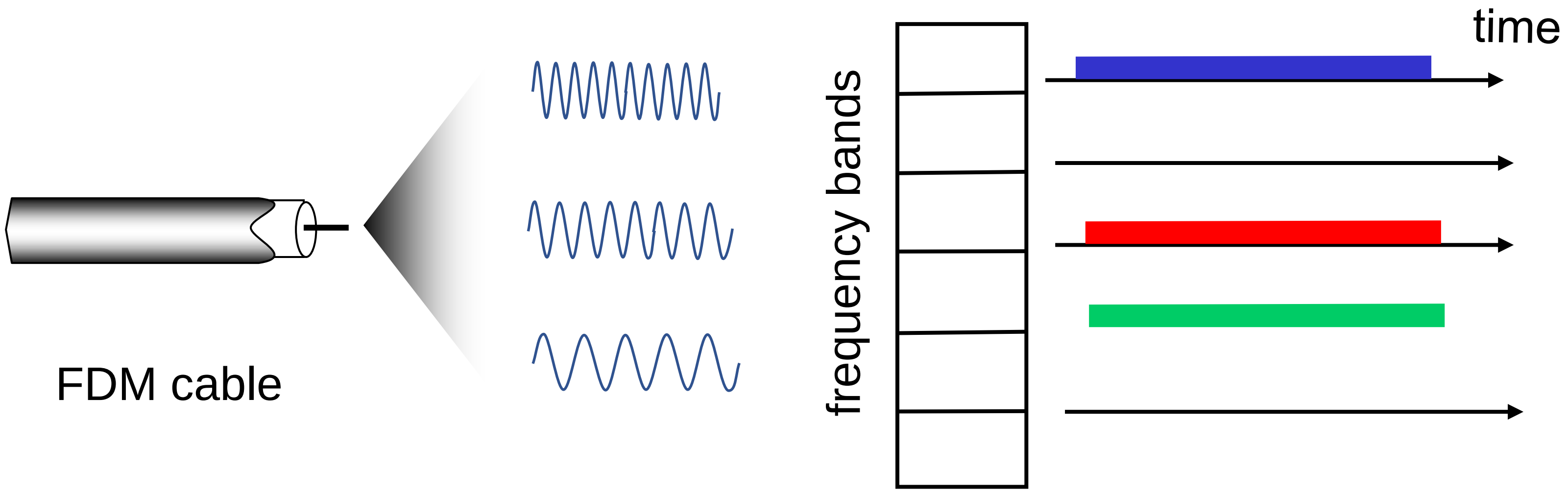
- access to channel in "rounds"
- each station gets fixed length slot (length = packet transmission time) in each round
- unused slots go idle
- example: 6-station LAN, 1,3,4 have packets to send, slots 2,5,6 idle



Channel partitioning MAC protocols: FDMA

FDMA: frequency division multiple access

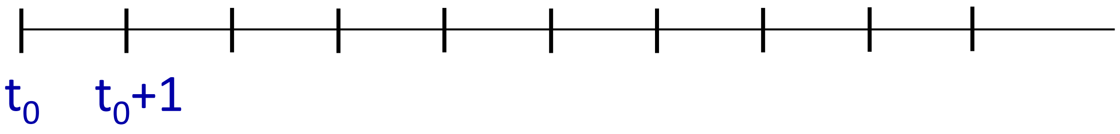
- channel spectrum divided into frequency bands
- each station assigned fixed frequency band
- unused transmission time in frequency bands go idle
- example: 6-station LAN, 1,3,4 have packet to send, frequency bands 2,5,6 idle



Random access protocols

- when node has packet to send
  - transmit at full channel data rate R
  - no *a priori* coordination among nodes
- two or more transmitting nodes: "collision"
- random access protocol specifies:
  - how to detect collisions
  - how to recover from collisions (e.g., via delayed retransmissions)
- examples of random access MAC protocols:
  - ALOHA, slotted ALOHA
  - CSMA, CSMA/CD, CSMA/CA

Slotted ALOHA



assumptions:

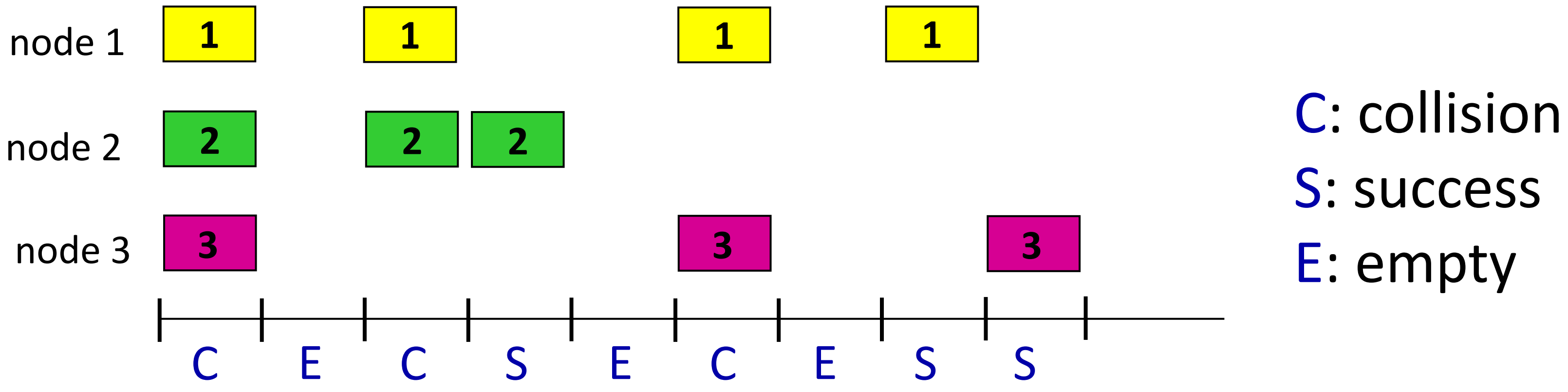
- all frames same size
- time divided into equal size slots (time to transmit 1 frame)
- nodes start to transmit only slot beginning
- nodes are synchronized
- if 2 or more nodes transmit in slot, all nodes detect collision

operation:

- when node obtains fresh frame, transmits in next slot
  - *if no collision:* node can send new frame in next slot
  - *if collision:* node retransmits frame in each subsequent slot with probability *p* until success

randomization – *why*?

Slotted ALOHA



Pros:

- single active node can continuously transmit at full rate of channel
- highly decentralized: only slots in nodes need to be in sync
- simple

Cons:

- collisions, wasting slots
- idle slots
- nodes may be able to detect collision in less than time to transmit packet
- clock synchronization

Slotted ALOHA: efficiency

efficiency: long-run fraction of successful slots (many nodes, all with many frames to send)

- *suppose: N* nodes with many frames to send, each transmits in slot with probability *p*
  - prob that given node has success in a slot = $p(1-p)^{N-1}$
  - prob that *any* node has a success = $Np(1-p)^{N-1}$
  - max efficiency: find p^* that maximizes $Np(1-p)^{N-1}$
  - for many nodes, take limit of $Np^*(1-p^*)^{N-1}$ as N goes to infinity, gives:

  max efficiency = 1/e = .37

- ***at best:*** channel used for useful transmissions 37% of time!

CSMA (carrier sense multiple access)

simple CSMA: listen before transmit:

- if channel sensed idle: transmit entire frame
- if channel sensed busy: defer transmission

- human analogy: <u>don't interrupt others!</u>

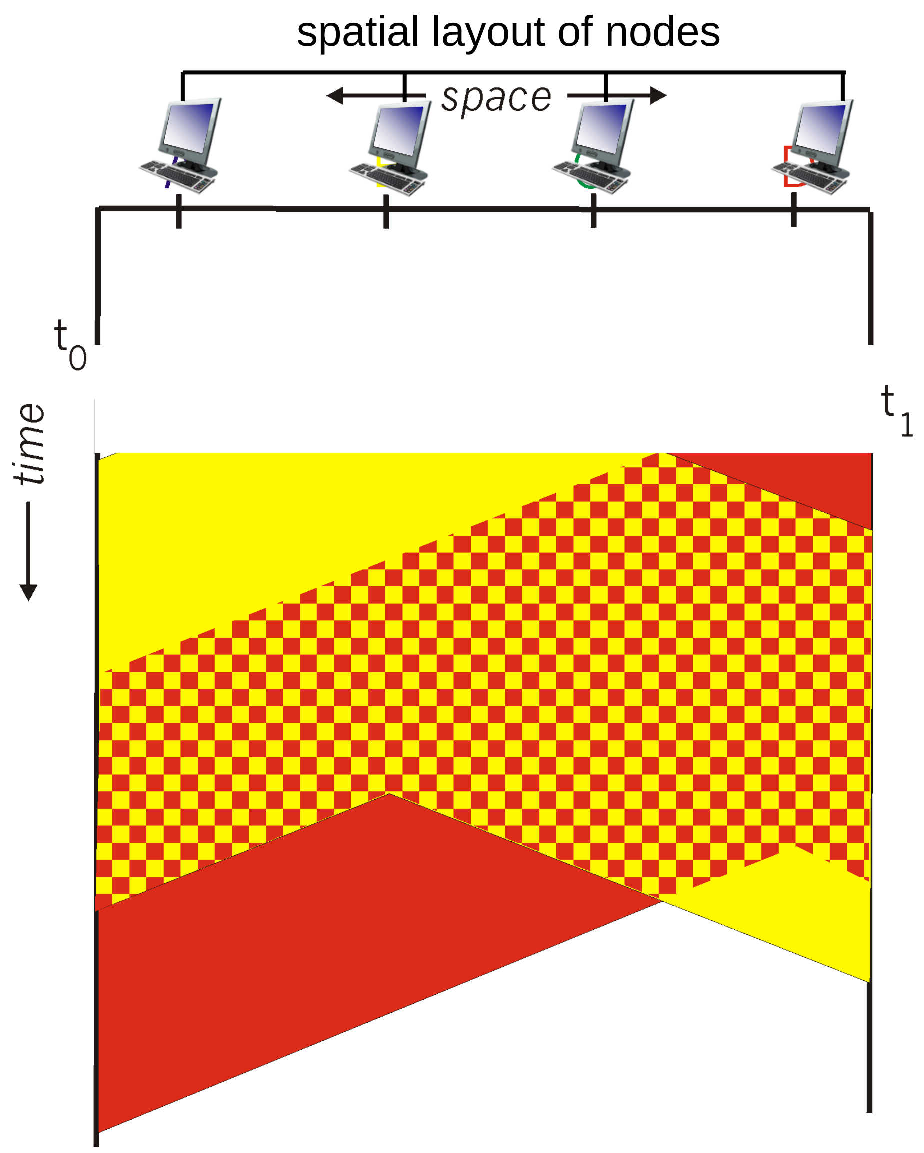
CSMA/CD: CSMA with *collision detection*

- collisions *detected* within short time
- colliding transmissions aborted, reducing channel wastage
- collision detection easy in wired, difficult with wireless

- human analogy: <u>the polite conversationalist</u>

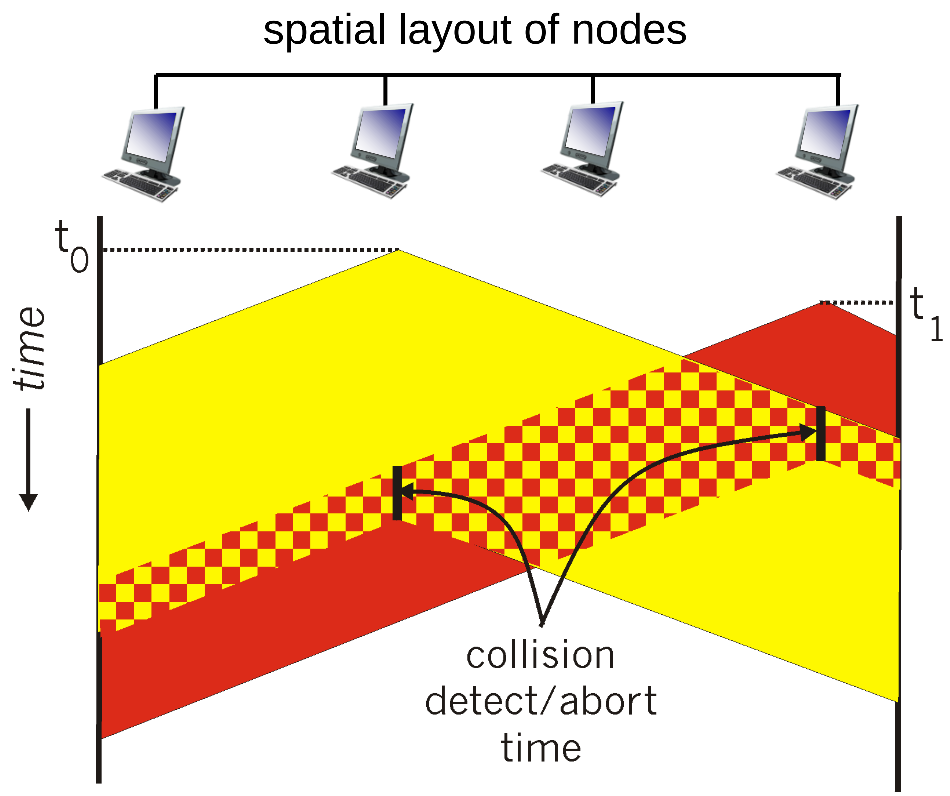
CSMA: collisions

- collisions can *still* occur with carrier sensing:
  - propagation delay means two nodes may not hear each other's just-started transmission
- collision: entire packet transmission time wasted
  - distance & propagation delay play role in in determining collision probability



CSMA/CD:

- CSMA/CD reduces the amount of time wasted in collisions
  - transmission aborted on collision detection



Ethernet CSMA/CD algorithm

1. Ethernet receives datagram from network layer, creates frame
2. If Ethernet senses channel:
   if idle: start frame transmission.
   if busy: wait until channel idle, then transmit
3. If entire frame transmitted without collision - done!
4. If another transmission detected while sending: abort, send jam signal
5. After aborting, enter *binary (exponential) backoff:*
   - after *m*th collision, chooses *K* at random from $\{0,1,2, \ldots, 2^m-1\}$. Ethernet waits $K \cdot 512$ bit times, returns to Step 2
   - more collisions: longer backoff interval

“Taking turns” MAC protocols

channel partitioning MAC protocols:

- share channel *efficiently* and *fairly* at high load
- inefficient at low load: delay in channel access, 1/N bandwidth allocated even if only 1 active node!

random access MAC protocols

- efficient at low load: single node can fully utilize channel
- high load: collision overhead

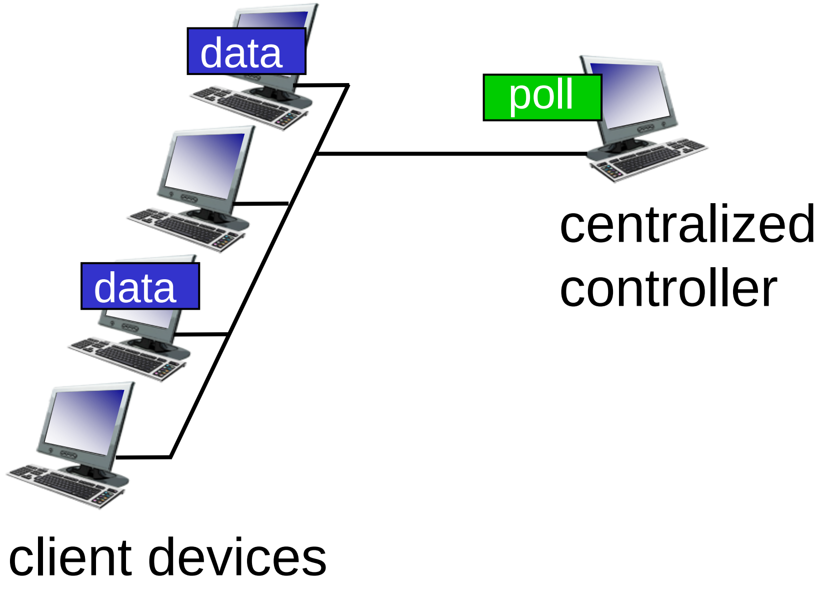
“taking turns” protocols

- look for best of both worlds!

“Taking turns” MAC protocols

polling:

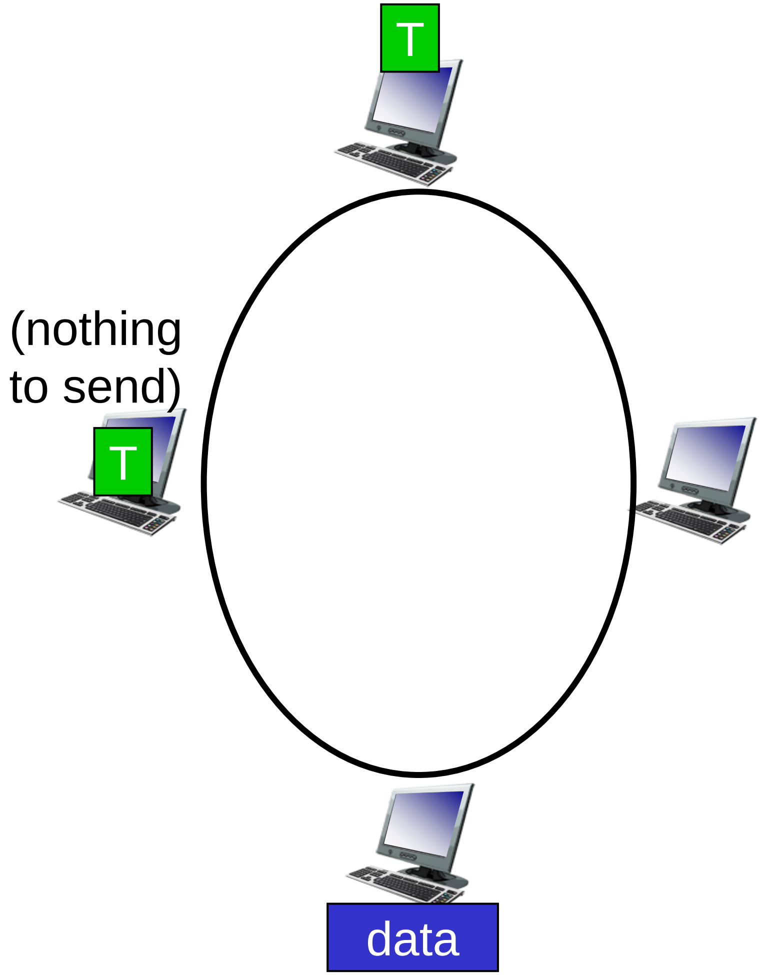
- centralized controller “invites” other nodes to transmit in turn
- typically used with “dumb” devices
- concerns:
  - polling overhead
  - latency
  - single point of failure (master)
- Bluetooth uses polling



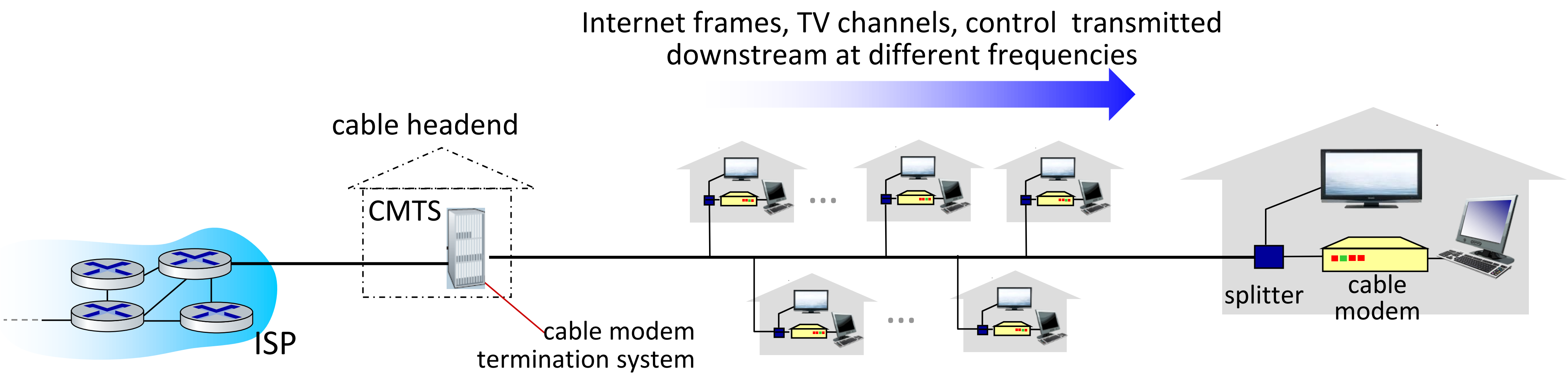
“Taking turns” MAC protocols

token passing:

- control *token* message explicitly passed from one node to next, sequentially
  - transmit while holding token
- concerns:
  - token overhead
  - latency
  - single point of failure (token)

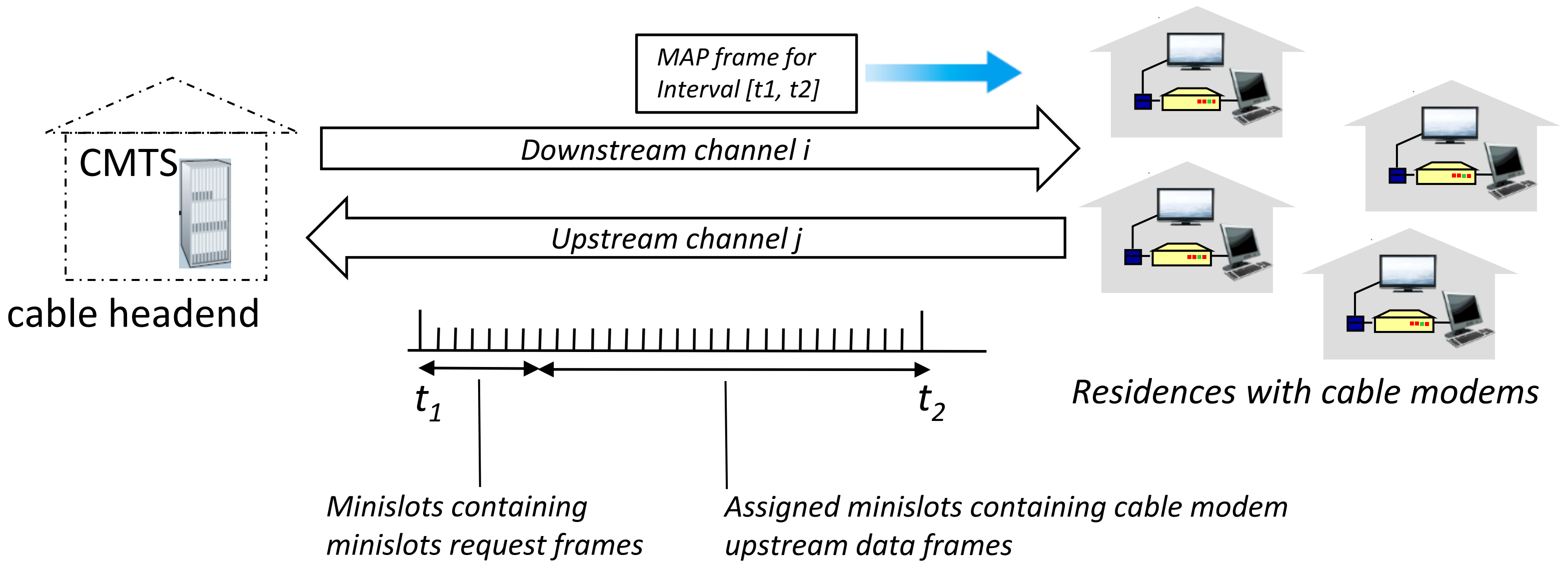


Cable access network: FDM, TDM *and* random access!



- multiple downstream (broadcast) FDM channels: up to 1.6 Gbps/channel
  - single CMTS transmits into channels
- multiple upstream channels (up to 1 Gbps/channel)
  - multiple access: all users contend (random access) for certain upstream channel time slots; others assigned TDM

Cable access network:



DOCSIS: data over cable service interface specification

- FDM over upstream, downstream frequency channels
- TDM upstream: some slots assigned, some have contention
  - downstream MAP frame: assigns upstream slots
  - request for upstream slots (and data) transmitted random access (binary backoff) in selected slots

Summary of MAC protocols

- channel partitioning, by time, frequency or code
  - Time Division, Frequency Division
- random access (dynamic),
  - ALOHA, S-ALOHA, CSMA, CSMA/CD
  - carrier sensing: easy in some technologies (wire), hard in others (wireless)
  - CSMA/CD used in Ethernet
  - CSMA/CA used in 802.11
- taking turns
  - polling from central site, token passing
  - Bluetooth, FDDI, token ring

Link layer, LANs: roadmap

- introduction
- error detection, correction
- multiple access protocols
- LANs
  - addressing, ARP
  - Ethernet
  - switches
  - VLANs
- link virtualization: MPLS
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- a day in the life of a web request

MAC addresses

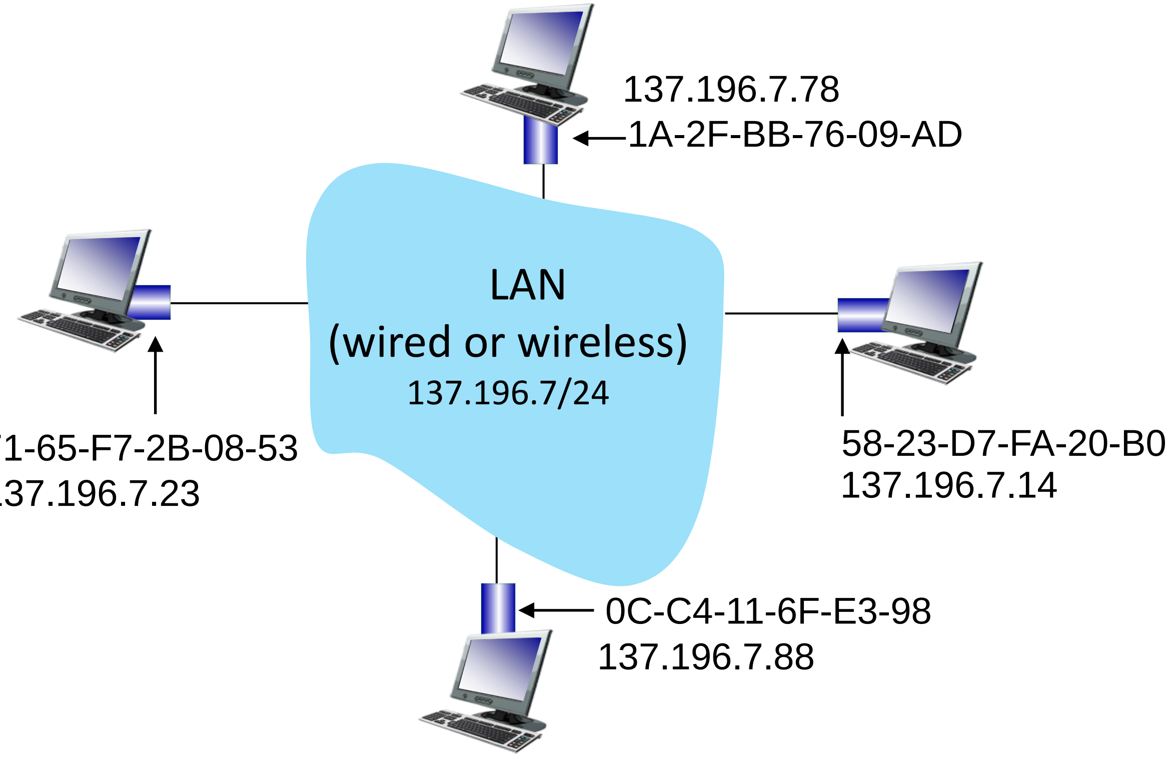
- 32-bit IP address:
  - *network-layer* address for interface
  - used for layer 3 (network layer) forwarding
  - e.g.: 128.119.40.136
- MAC (or LAN or physical or Ethernet) address:
  - function: used "locally" to get frame from one interface to another physically-connected interface (same subnet, in IP-addressing sense)
  - 48-bit MAC address (for most LANs) burned in NIC ROM, also sometimes software settable
  - e.g.: 1A-2F-BB-76-09-AD

hexadecimal (base 16) notation
(each "numeral" represents 4 bits)

MAC addresses

each interface on LAN

- has unique 48-bit MAC address
- has a locally unique 32-bit IP address (as we've seen)

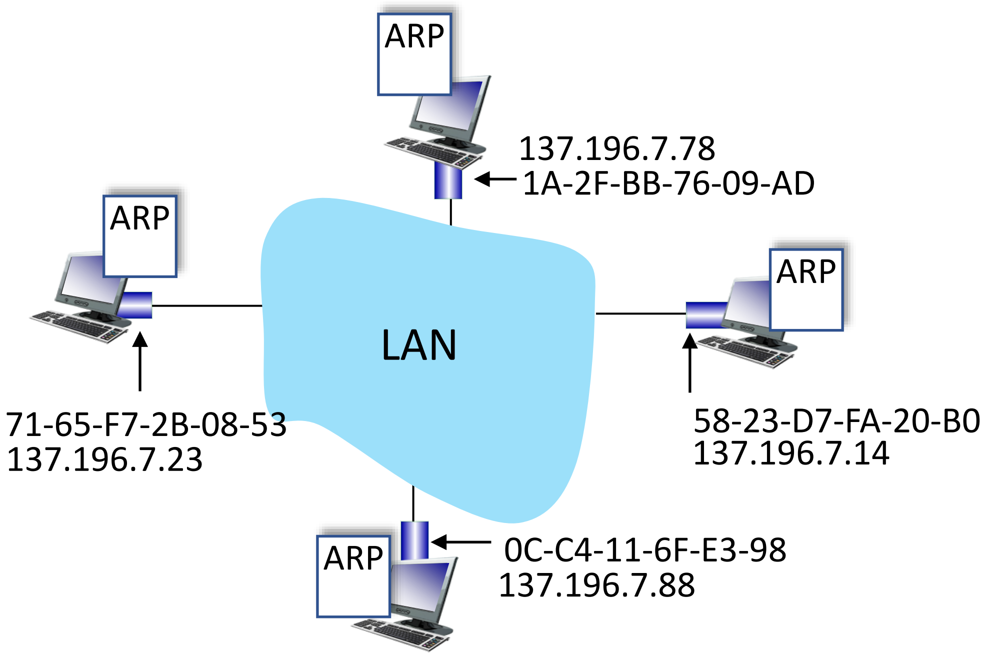


MAC addresses

- MAC address allocation administered by IEEE
- manufacturer buys portion of MAC address space (to assure uniqueness)
- analogy:
  - MAC address: like Social Security Number
  - IP address: like postal address
- MAC flat address: portability
  - can move interface from one LAN to another
  - recall IP address *not* portable: depends on IP subnet to which node is attached

ARP: address resolution protocol

Question: how to determine interface's MAC address, knowing its IP address?



ARP table: each IP node (host, router) on LAN has table

- IP/MAC address mappings for some LAN nodes:

  < IP address; MAC address; TTL>

- TTL (Time To Live): time after which address mapping will be forgotten (typically 20 min)

ARP protocol in action

example: A wants to send datagram to B

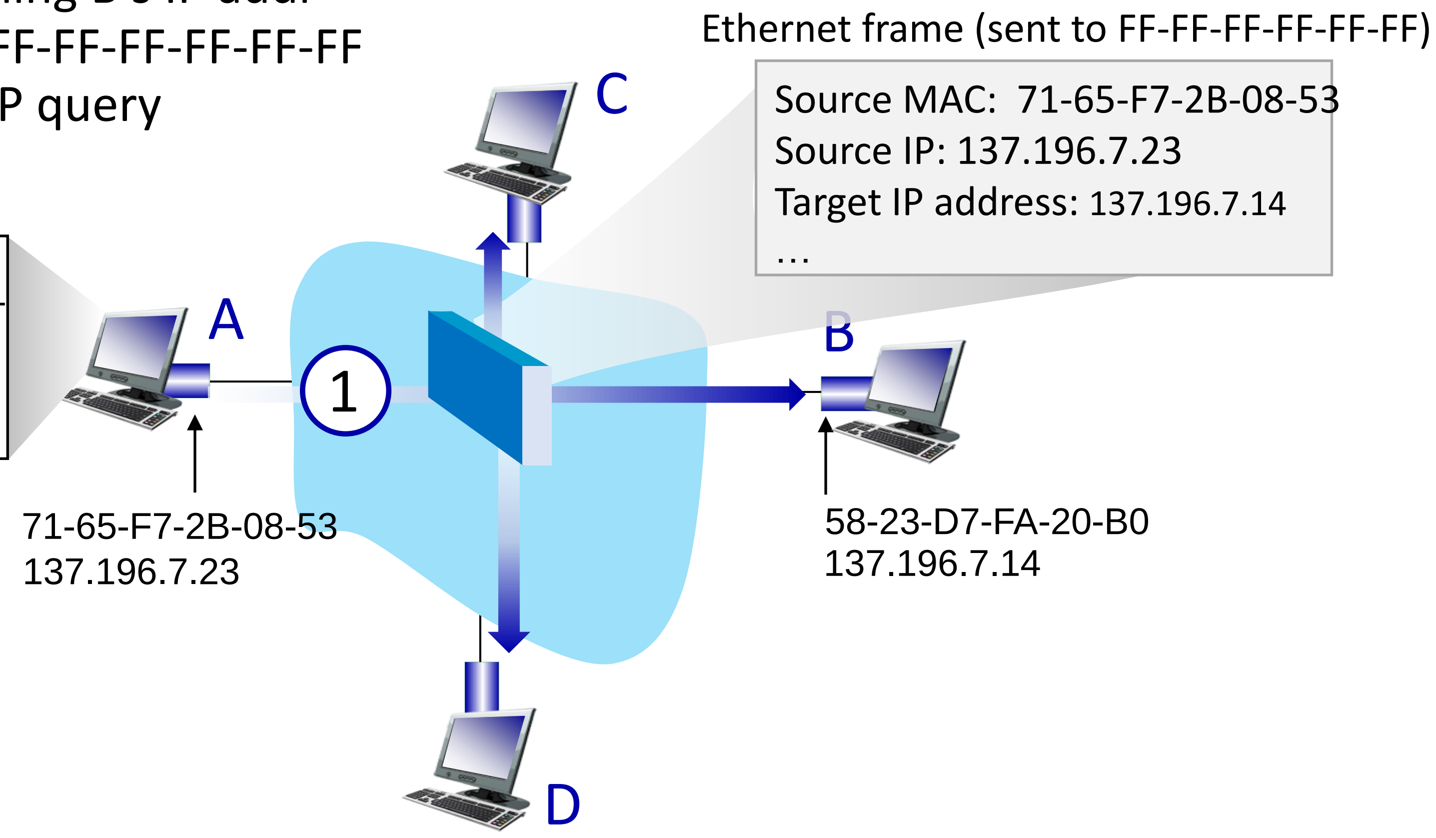
- B's MAC address not in A's ARP table, so A uses ARP to find B's MAC address

1 A broadcasts ARP query, containing B's IP addr

- destination MAC address = FF-FF-FF-FF-FF-FF
- all nodes on LAN receive ARP query

ARP table in A

IP addr	MAC addr	TTL



ARP protocol in action

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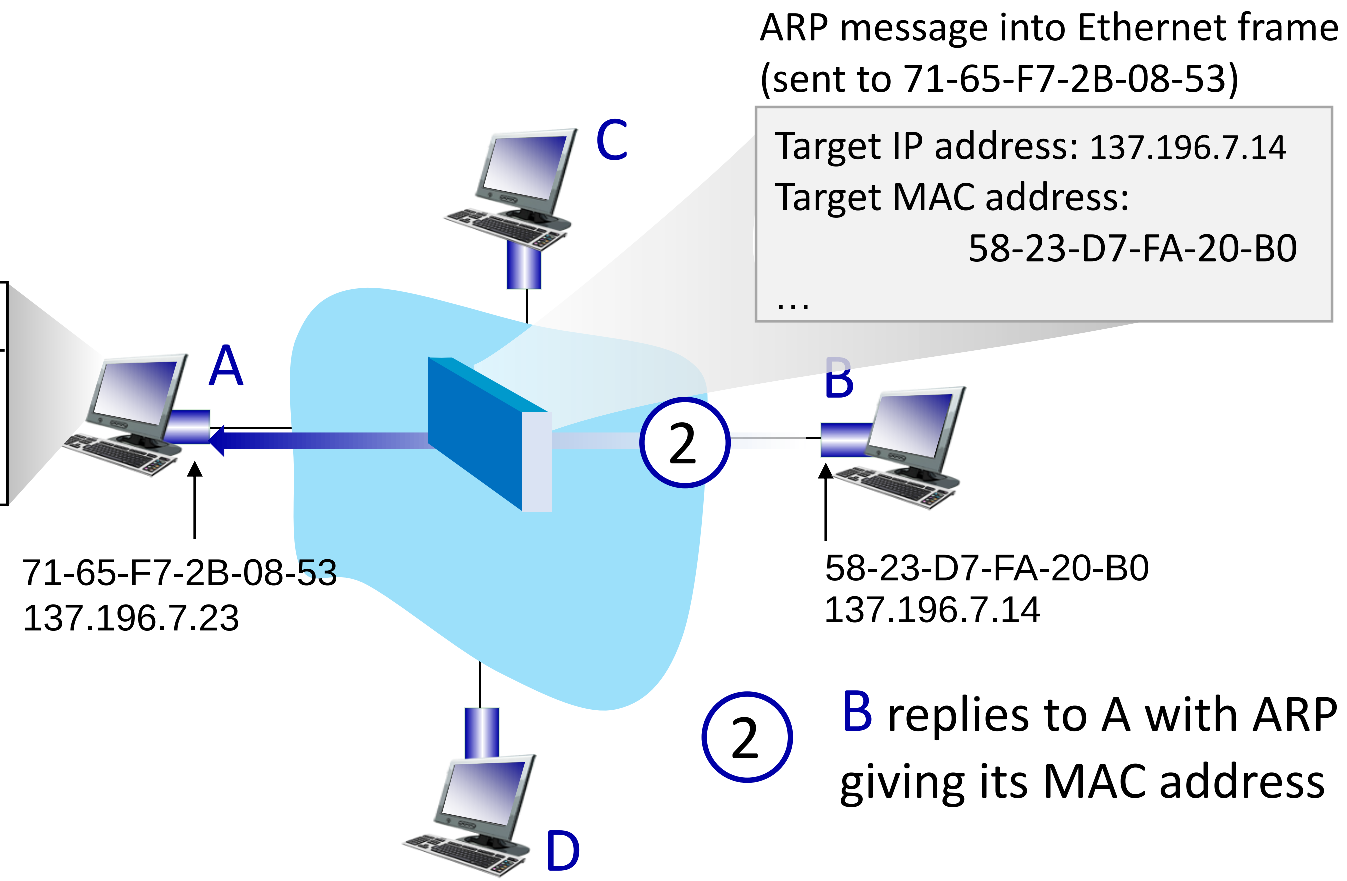


2 B replies to A with ARP response, giving its MAC address

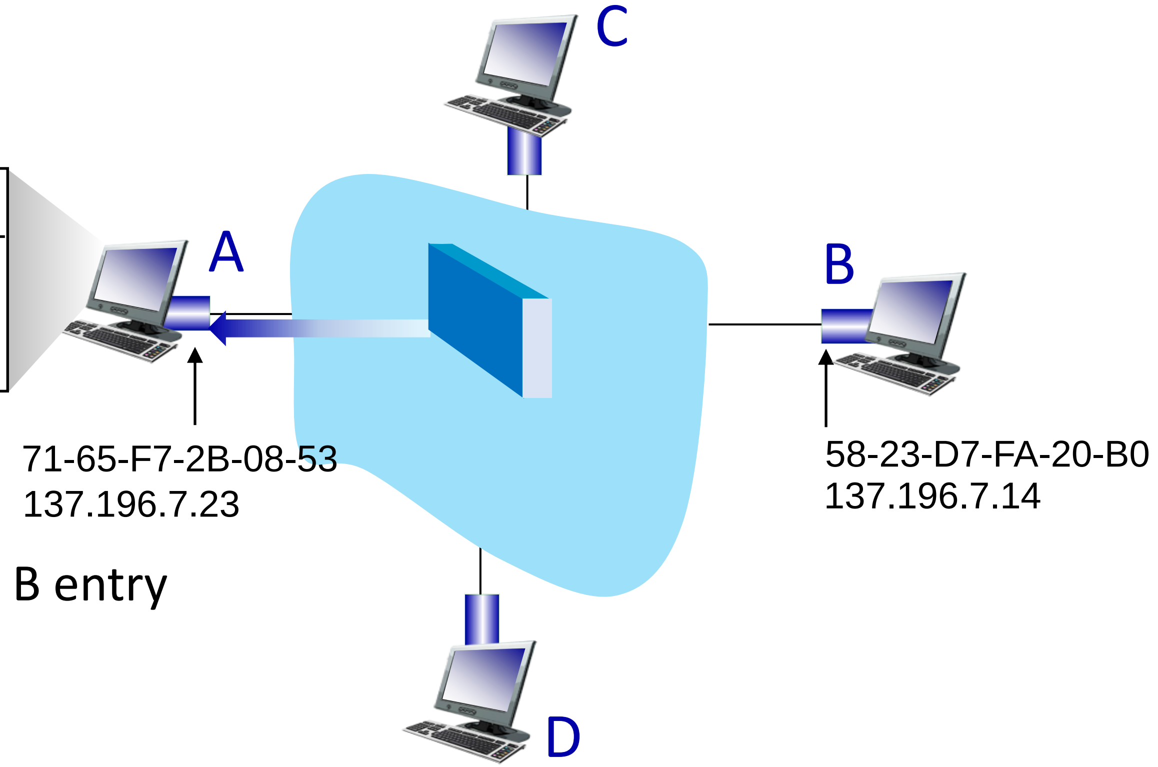
ARP protocol in action

example: A wants to send datagram to B

- B's MAC address not in A's ARP table, so A uses ARP to find B's MAC address

ARP table in A

IP addr	MAC addr	TTL
137.196.7.14	58-23-D7-FA-20-B0	500

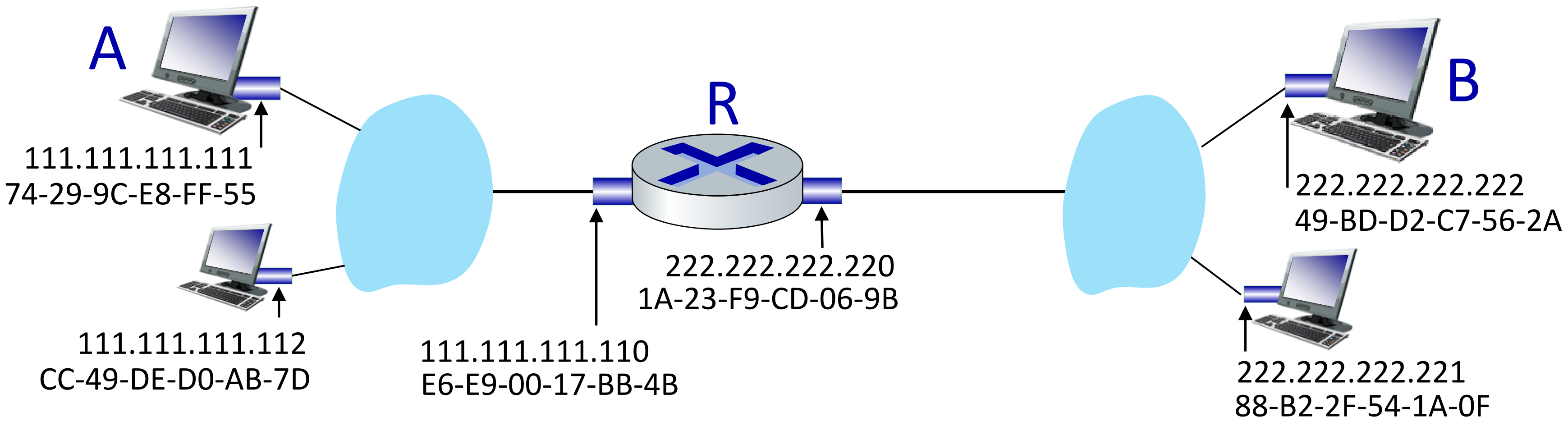


3 A receives B's reply, adds B entry into its local ARP table

Routing to another subnet: addressing

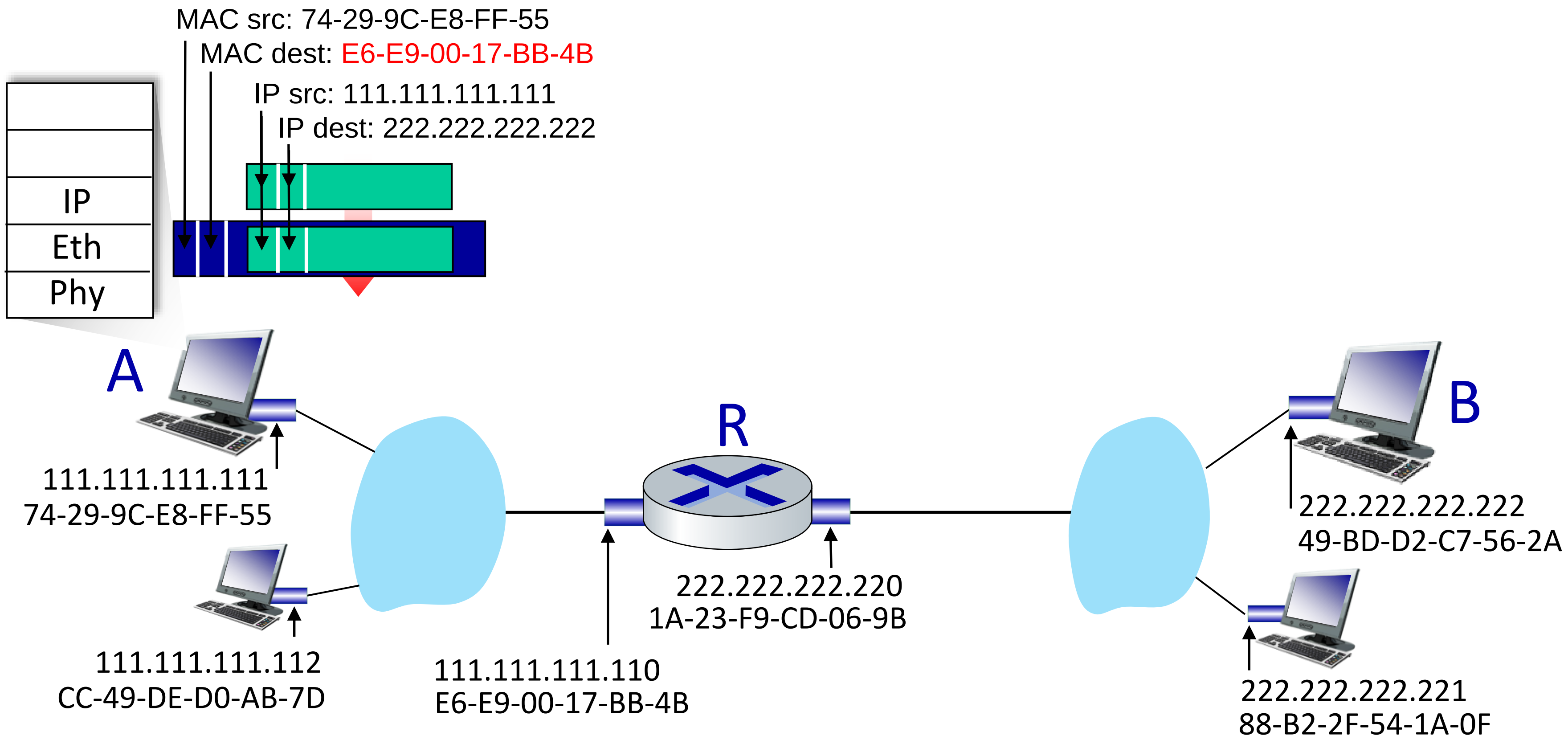
walkthrough: sending a datagram from *A* to *B* via *R*

- focus on addressing – at IP (datagram) and MAC layer (frame) levels
- assume that:
  - A knows B's IP address
  - A knows IP address of first hop router, R (how?)
  - A knows R's MAC address (how?)



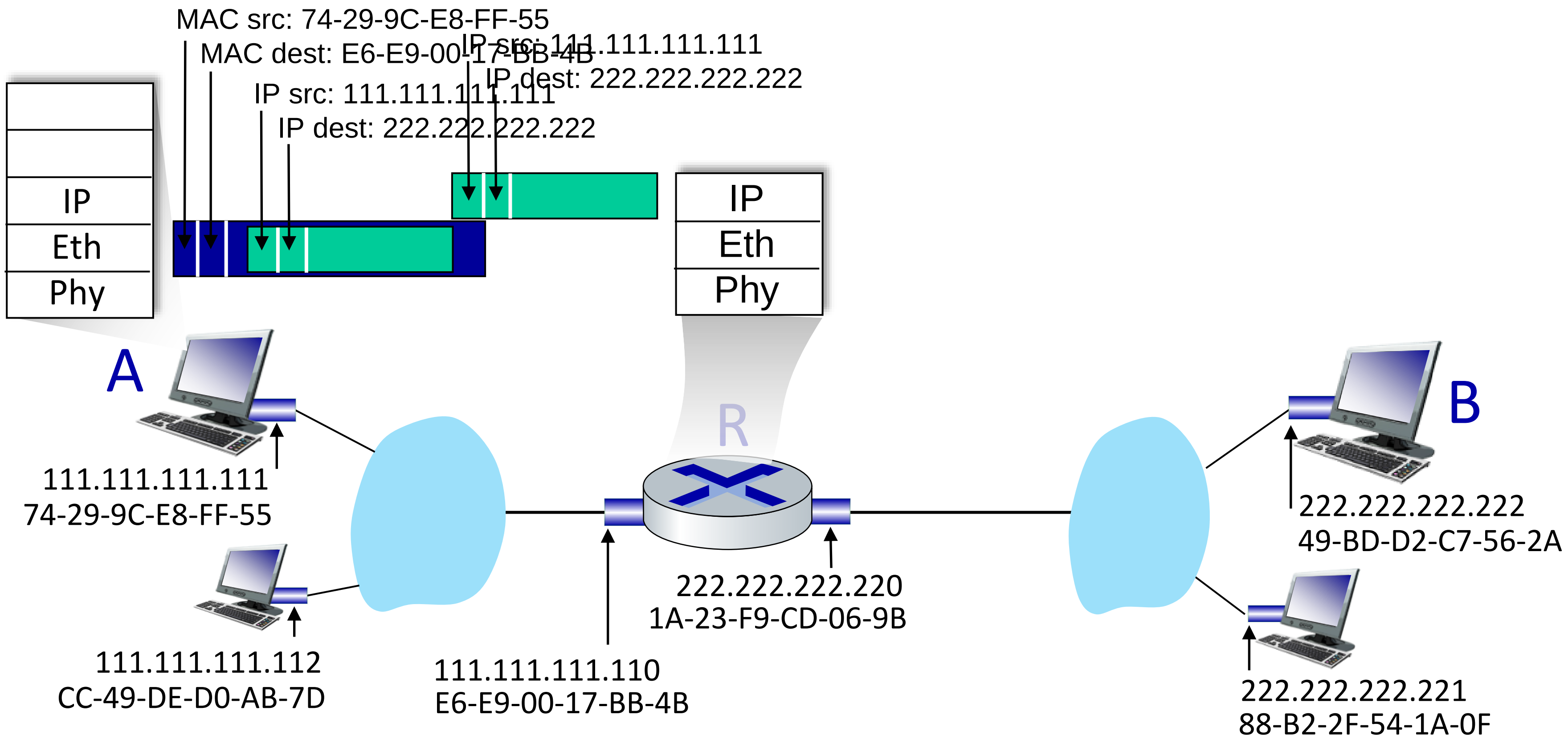
Routing to another subnet: addressing

- A creates IP datagram with IP source A, destination B
- A creates link-layer frame containing A-to-B IP datagram
  - R's MAC address is frame's destination



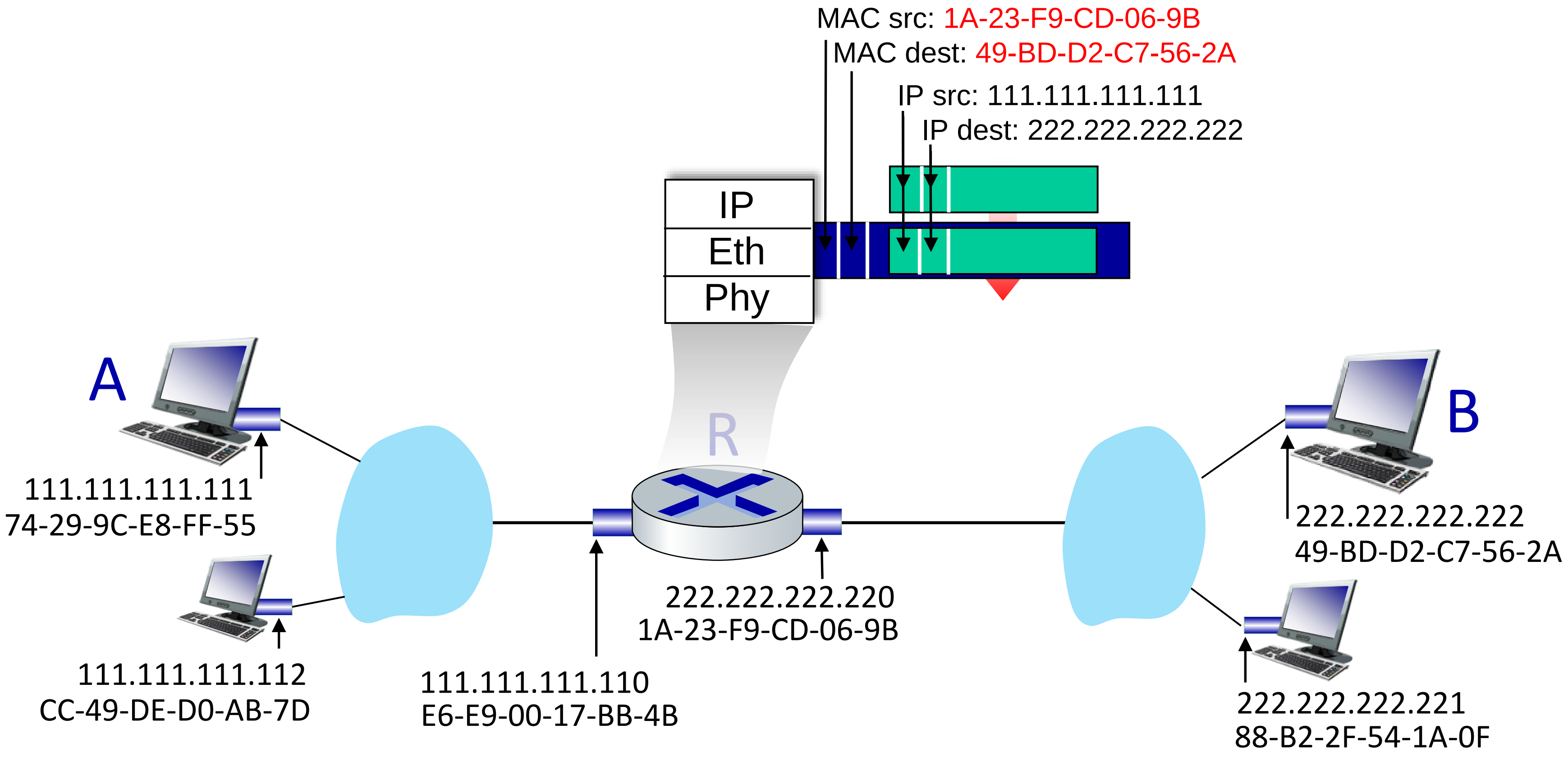
Routing to another subnet: addressing

- frame sent from A to R
- frame received at R, datagram removed, passed up to IP



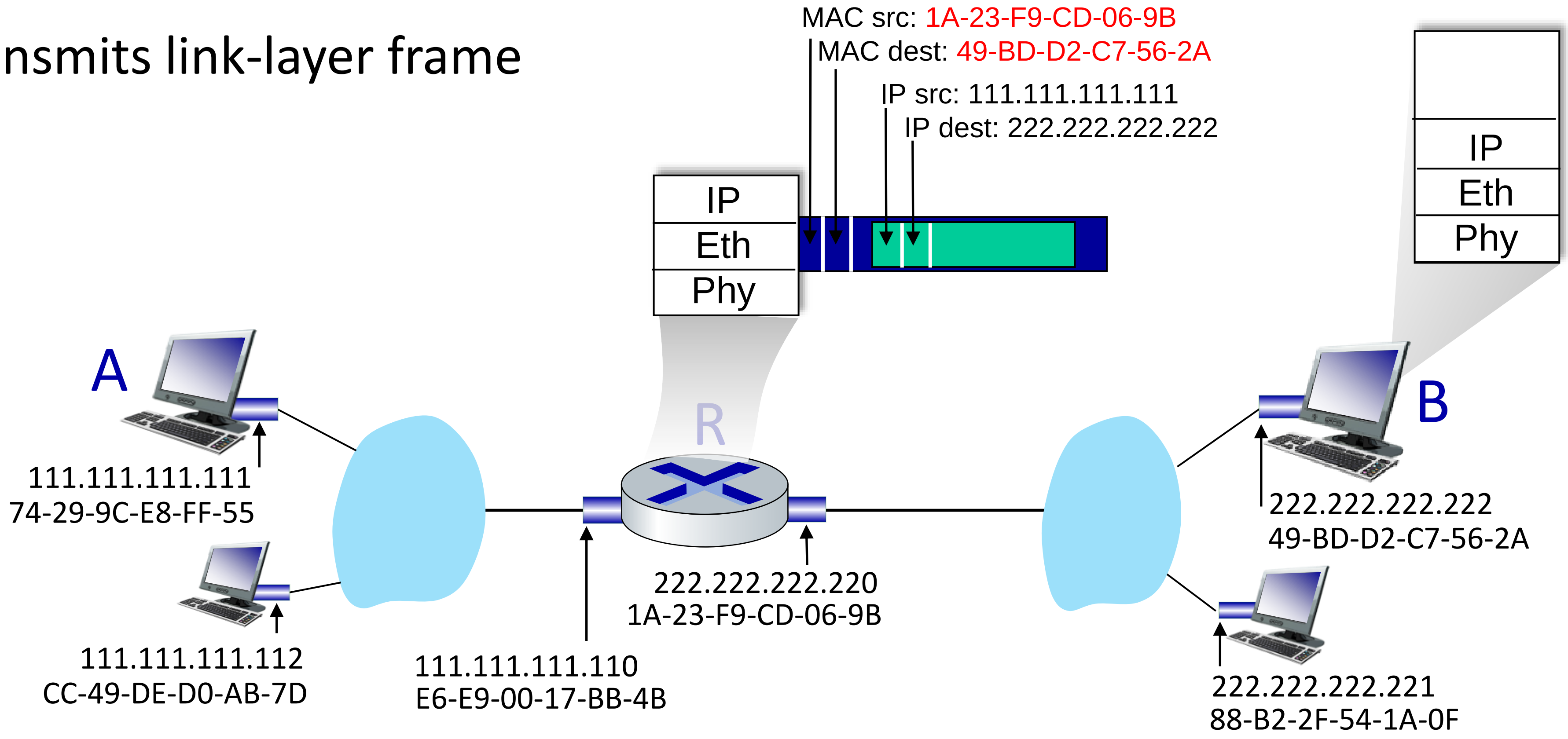
Routing to another subnet: addressing

- R determines outgoing interface, passes datagram with IP source A, destination B to link layer
- R creates link-layer frame containing A-to-B IP datagram. Frame destination address: B's MAC address



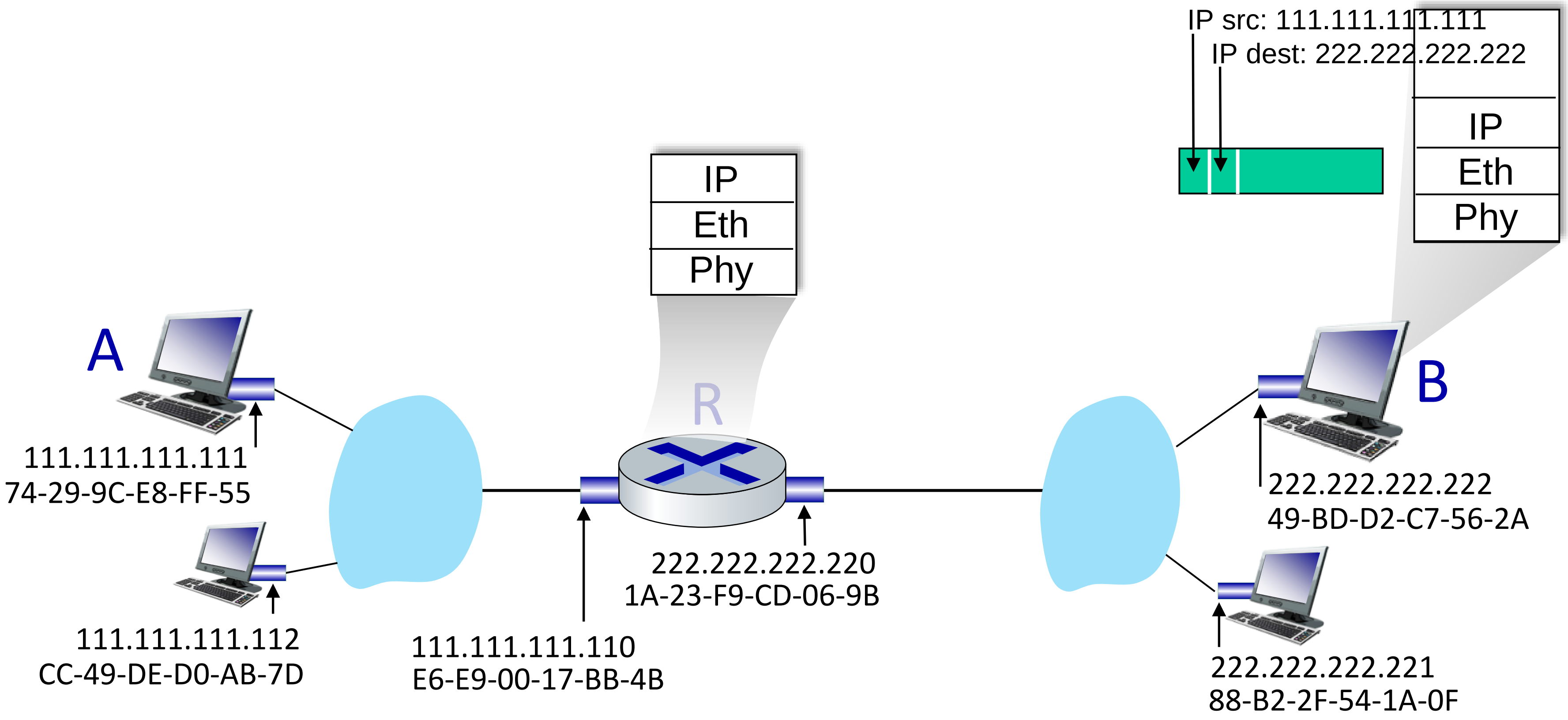
Routing to another subnet: addressing

- R determines outgoing interface, passes datagram with IP source A, destination B to link layer
- R creates link-layer frame containing A-to-B IP datagram. Frame destination address: B's MAC address
- transmits link-layer frame



Routing to another subnet: addressing

- B receives frame, extracts IP datagram destination B
- B passes datagram up protocol stack to IP



Link layer, LANs: roadmap

- introduction
- error detection, correction
- multiple access protocols
- LANs
  - addressing, ARP
  - Ethernet
  - switches
  - VLANs
- link virtualization: MPLS
- data center networking
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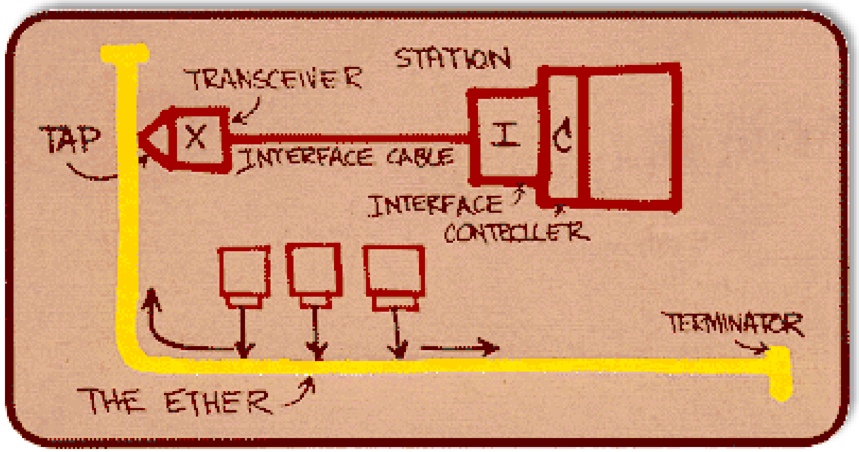
Ethernet

"dominant" wired LAN technology:

- first widely used LAN technology
- simpler, cheap
- kept up with speed race: 10 Mbps – 400 Gbps
- single chip, multiple speeds (e.g., Broadcom BCM5761)

Bob Metcalfe: Ethernet co-inventor, 2022 ACM Turing Award recipient

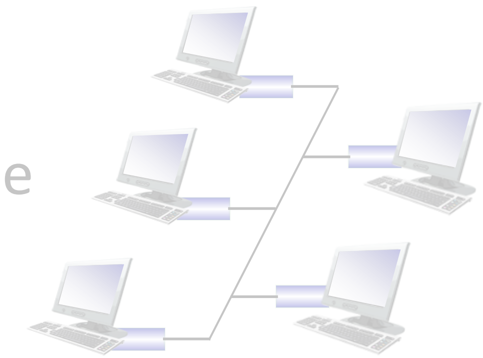
Metcalfe's Ethernet sketch

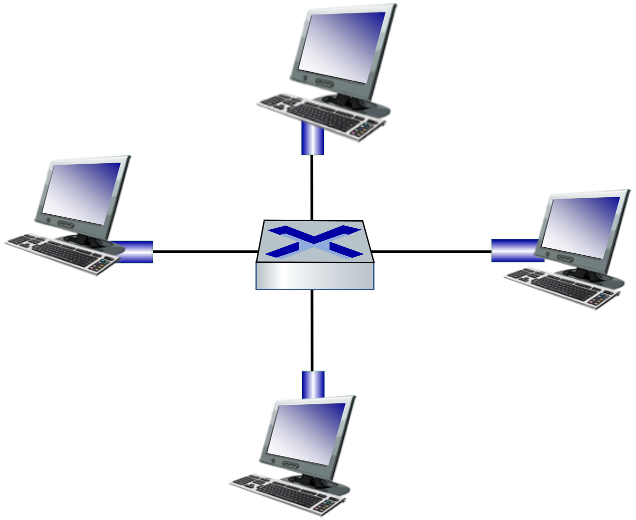


Ethernet: physical topology

- bus: popular through mid 90s
  - all nodes in same collision domain (can collide with each other)
- switched: prevails today
  - active link-layer 2 *switch* in center
  - each "spoke" runs a (separate) Ethernet protocol (nodes do not collide with each other)

bus: coaxial cable





switched

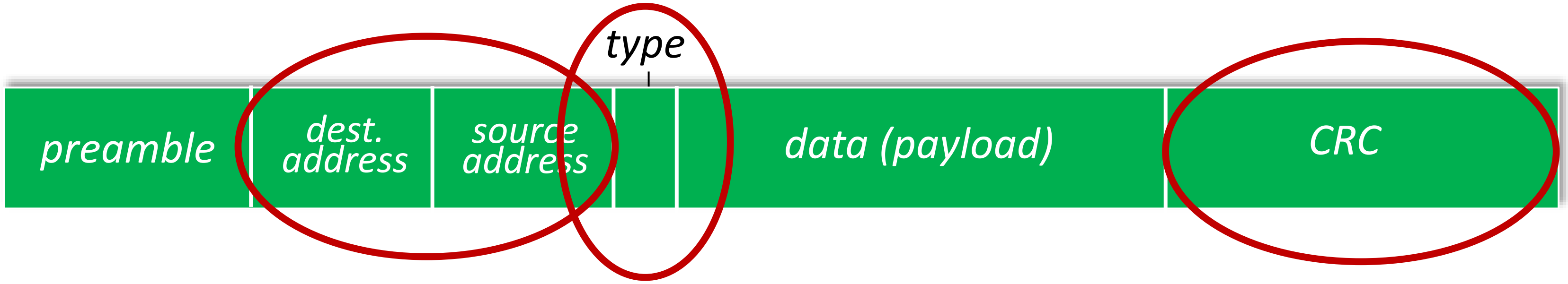
Ethernet frame structure

sending interface encapsulates IP datagram (or other network layer protocol packet) in Ethernet frame

preamble:

- used to synchronize receiver, sender clock rates
- 7 bytes of 10101010 followed by one byte of 10101011

Ethernet frame structure (more)



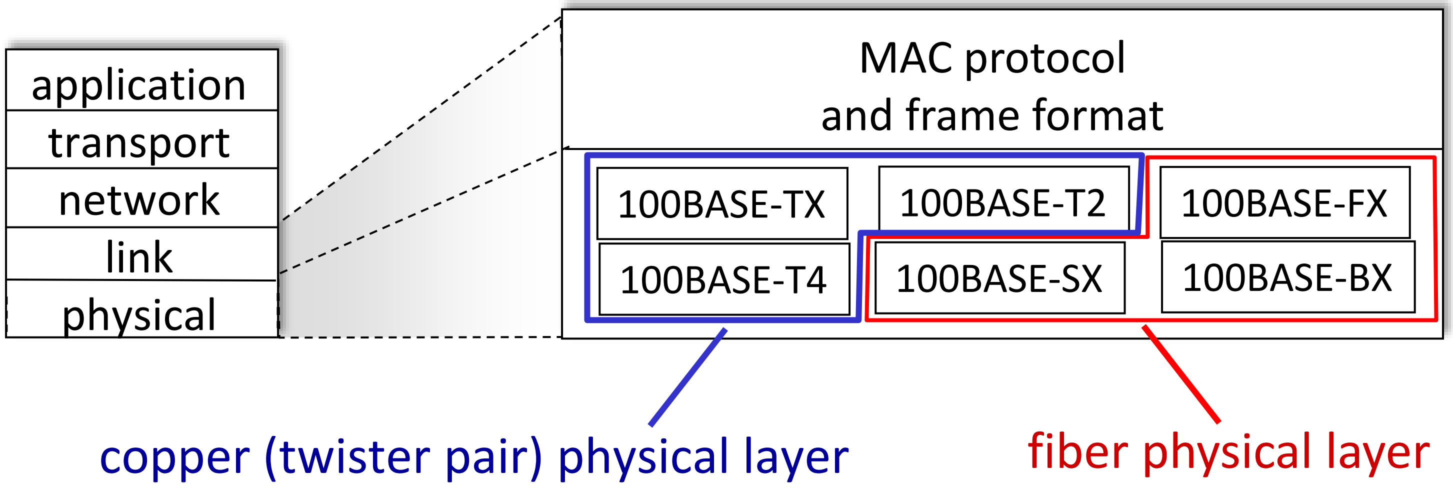
- addresses: 6 byte source, destination MAC addresses
  - if adapter receives frame with matching destination address, or with broadcast address (e.g., ARP packet), it passes data in frame to network layer protocol
  - otherwise, adapter discards frame
- type: indicates higher layer protocol
  - mostly IP but others possible, e.g., Novell IPX, AppleTalk
  - used to demultiplex up at receiver
- CRC: cyclic redundancy check at receiver
  - error detected: frame is dropped

Ethernet: unreliable, connectionless

- connectionless: no handshaking between sending and receiving NICs
- unreliable: receiving NIC doesn't send ACKs or NAKs to sending NIC
  - data in dropped frames recovered only if initial sender uses higher layer rdt (e.g., TCP), otherwise dropped data lost
- Ethernet's MAC protocol: unslotted CSMA/CD with binary backoff

802.3 Ethernet standards: link & physical layers

- *many* different Ethernet standards
  - common MAC protocol and frame format
  - different speeds: 2 Mbps, ... 100 Mbps, 1Gbps, 10 Gbps, 40 Gbps, 80 Gbps
    - different physical layer media: fiber, cable



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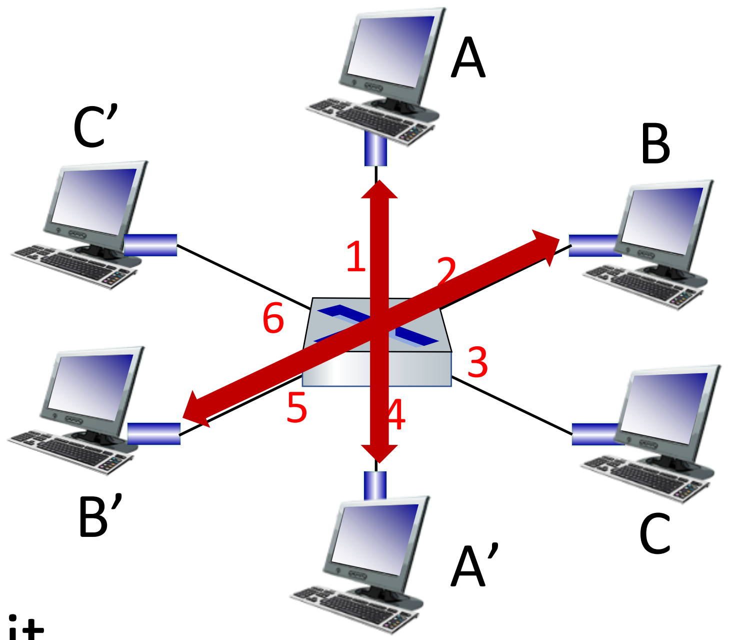
Ethernet switch

- Switch is a link-layer device: takes an *active* role
  - store, forward Ethernet (or other type of) frames
  - examine incoming frame's MAC address, *selectively* forward frame to one-or-more outgoing links when frame is to be forwarded on segment, uses CSMA/CD to access segment
- transparent: hosts *unaware* of presence of switches
- plug-and-play, self-learning
  - switches do not need to be configured

Switch: multiple simultaneous transmissions

- hosts have dedicated, direct connection to switch
- switches buffer packets
- Ethernet protocol used on *each* incoming link, so:
  - no collisions; full duplex
  - each link is its own collision domain
- switching: A-to-A' and B-to-B' can transmit simultaneously, without collisions



switch with six interfaces (1,2,3,4,5,6)

Switch: multiple simultaneous transmissions

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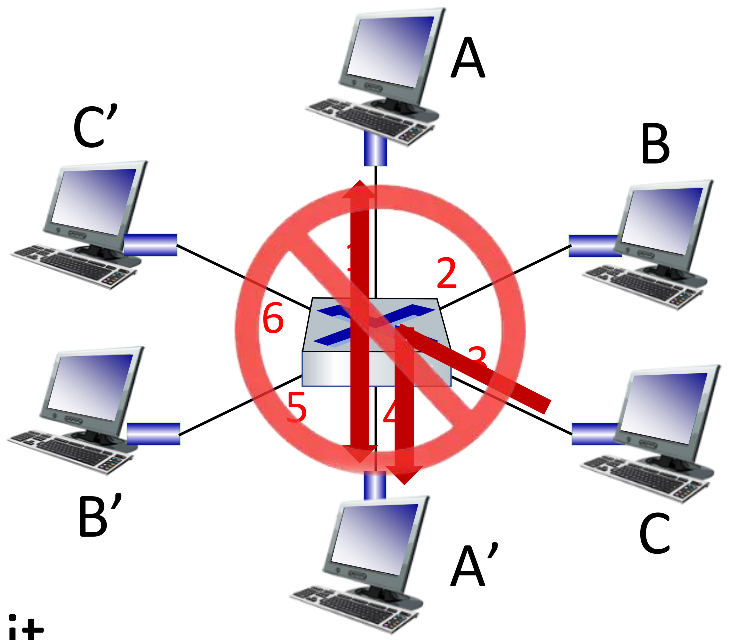
- Ethernet protocol used on *each* incoming link, so:
  - no collisions; full duplex
  - each link is its own collision domain
- switching: A-to-A' and B-to-B' can transmit simultaneously, without collisions
  - but A-to-A' and C to A' can *not* happen simultaneously



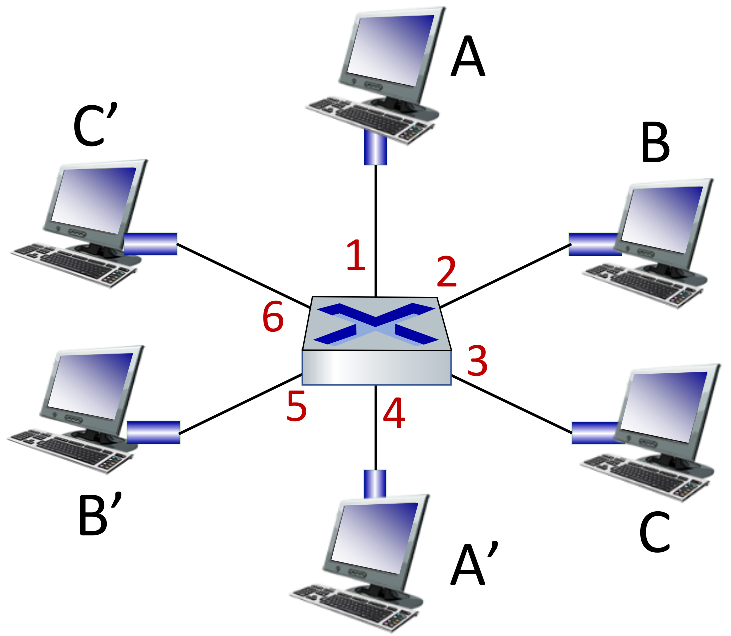
switch with six interfaces (1,2,3,4,5,6)

Switch forwarding table

Q: how does switch know A' reachable via interface 4, B' reachable via interface 5?

A: each switch has a switch table, each entry:

- (MAC address of host, interface to reach host, time stamp)
- looks like a routing table!

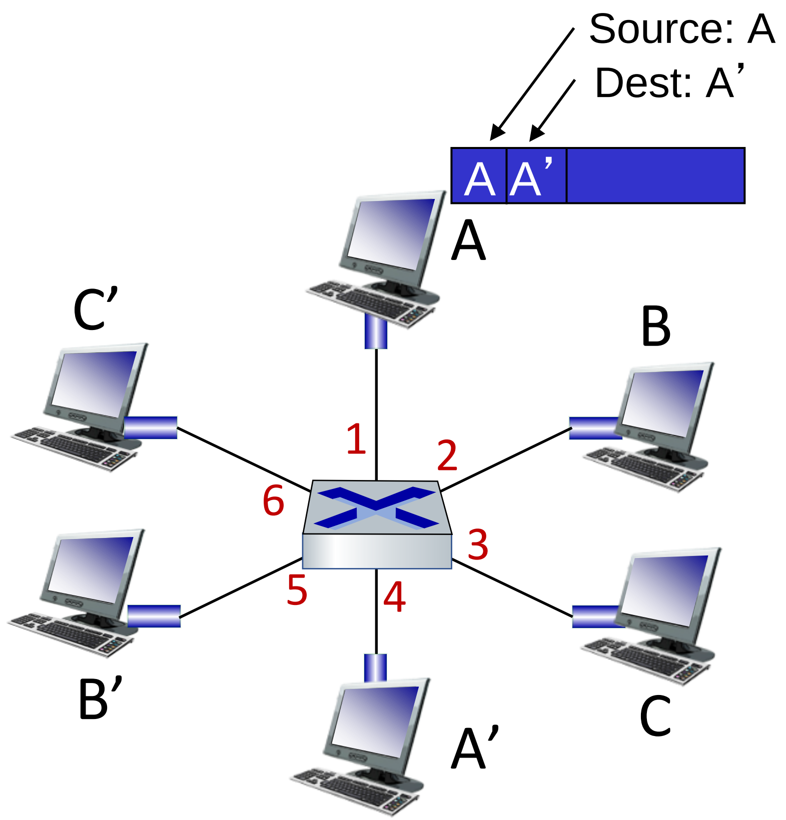


Q: how are entries created, maintained in switch table?

- something like a routing protocol?

Switch: self-learning

- switch *learns* which hosts can be reached through which interfaces
  - when frame received, switch "learns" location of sender: incoming LAN segment
  - records sender/location pair in switch table



MAC addr	interface	TTL
A	*1*	*60*

Switch table (initially empty)

Switch: frame filtering/forwarding

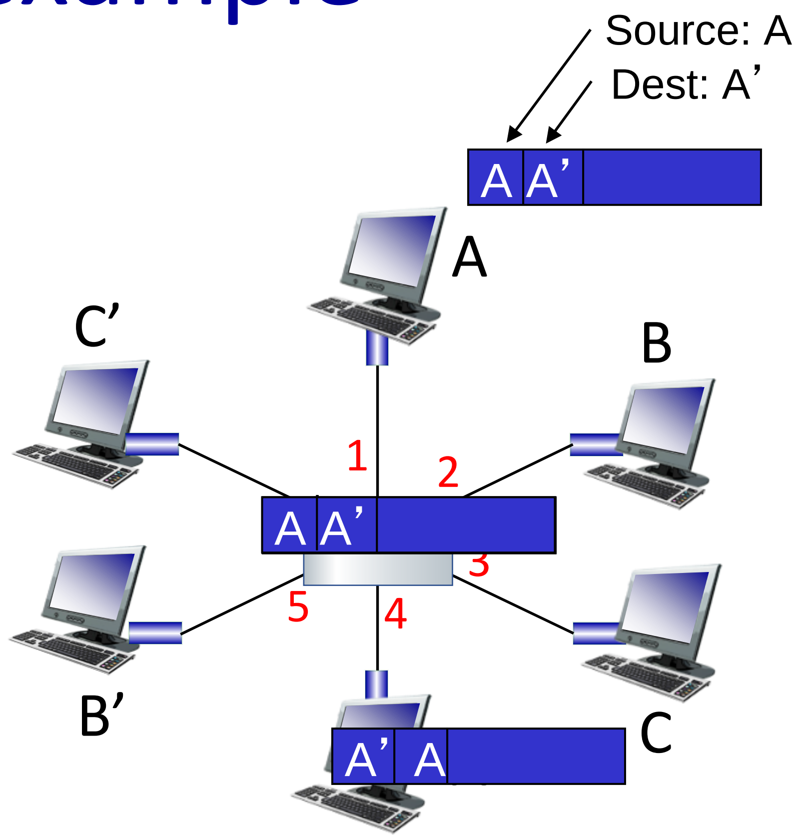
when frame received at switch:

1. record incoming link, MAC address of sending host
2. index switch table using MAC destination address
3. if entry found for destination

```
then {
if destination on segment from which frame arrived
    then drop frame
     else forward frame on interface indicated by entry
  }
 else flood  /* forward on all interfaces except arriving interface */
```

Self-learning, forwarding: example

- frame destination, A', location unknown: flood
- destination A location known: selectively send on just one link

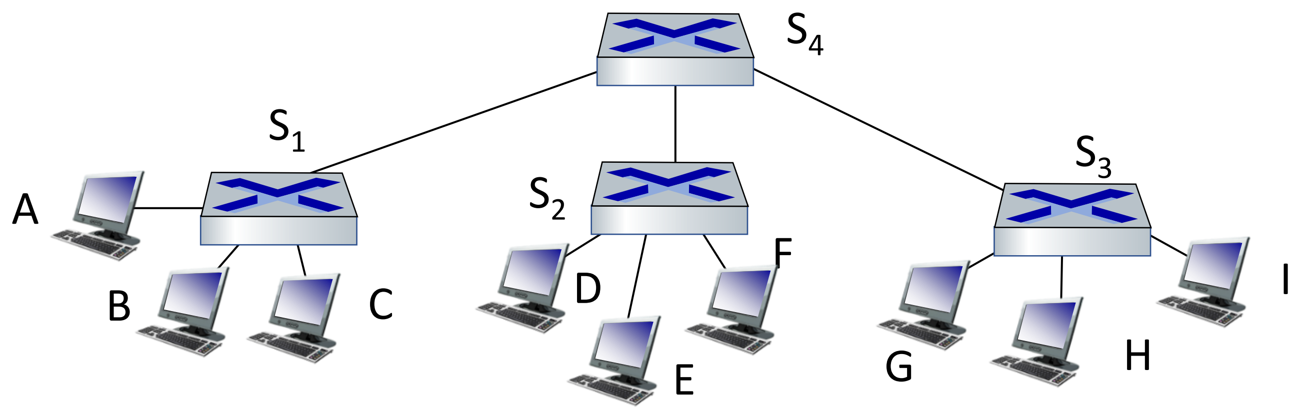


MAC addr	interface	TTL
A	*1*	*60*
A'	*4*	*60*

switch table (initially empty)

Interconnecting switches

self-learning switches can be connected together:

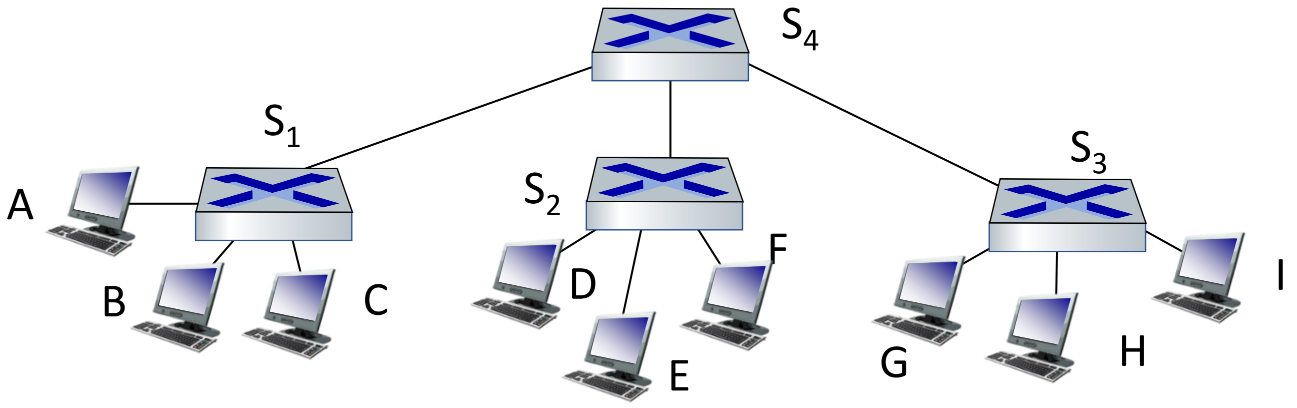


Q: sending from A to G - how does S_1 know to forward frame destined to G via S_4 and S_3?

- *A:* self learning! (works exactly the same as in single-switch case!)

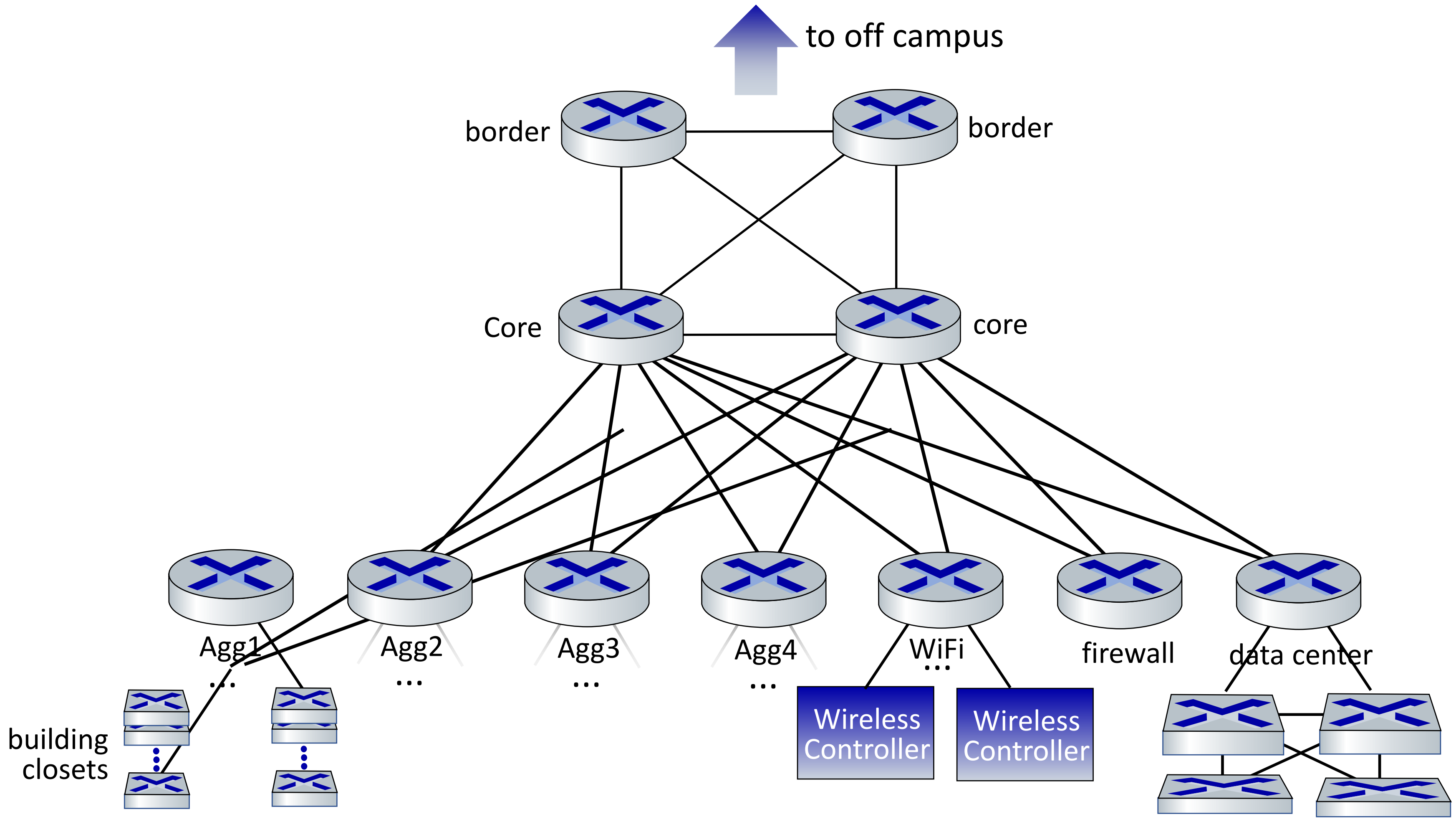
Self-learning multi-switch example

Suppose C sends frame to I, I responds to C



Q: show switch tables and packet forwarding in S_1, S_2, S_3, S_4

UMass Campus Network - Detail

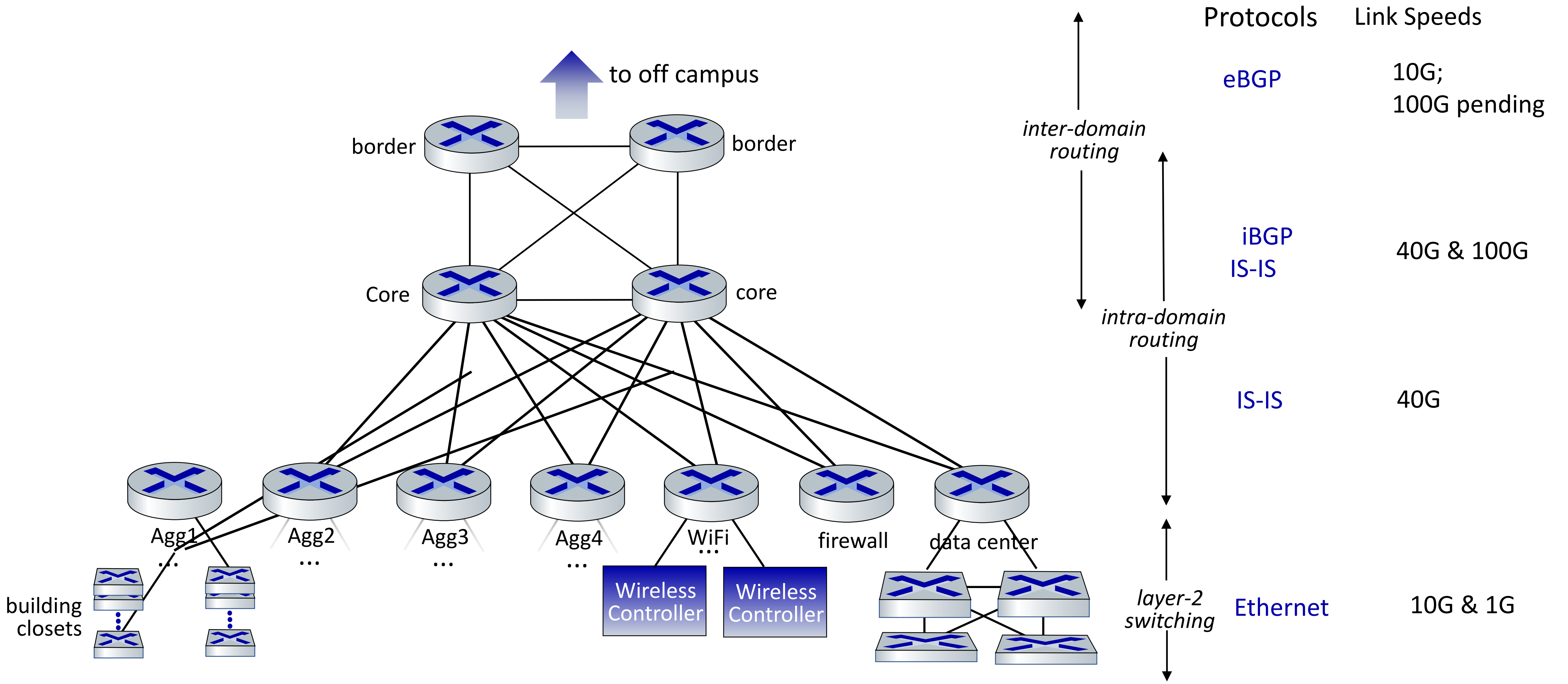


UMass network:

- 4 firewalls
- 10 routers
- 2000+ network switches
- 6000 wireless access points
- 30000 active wired network jacks
- 55000 active end-user wireless devices

... all built, operated, maintained by ~15 people

UMass Campus Network - Detail



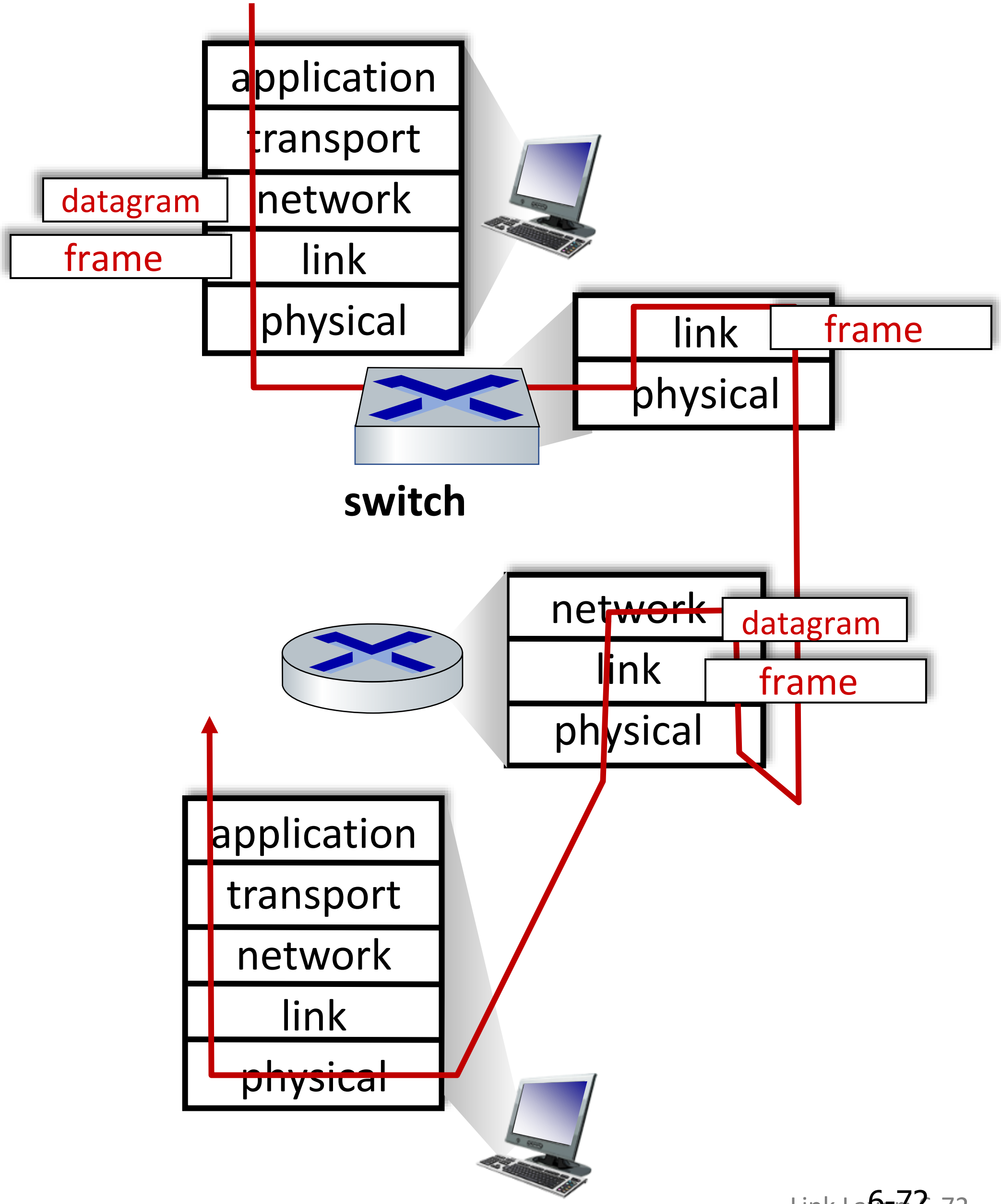
Switches vs. routers

both are store-and-forward:

- *routers*: network-layer devices (examine network-layer headers)
- *switches:* link-layer devices (examine link-layer headers)

both have forwarding tables:

- *routers:* compute tables using routing algorithms, IP addresses
- *switches:* learn forwarding table using flooding, learning, MAC addresses



Link layer, LANs: roadmap

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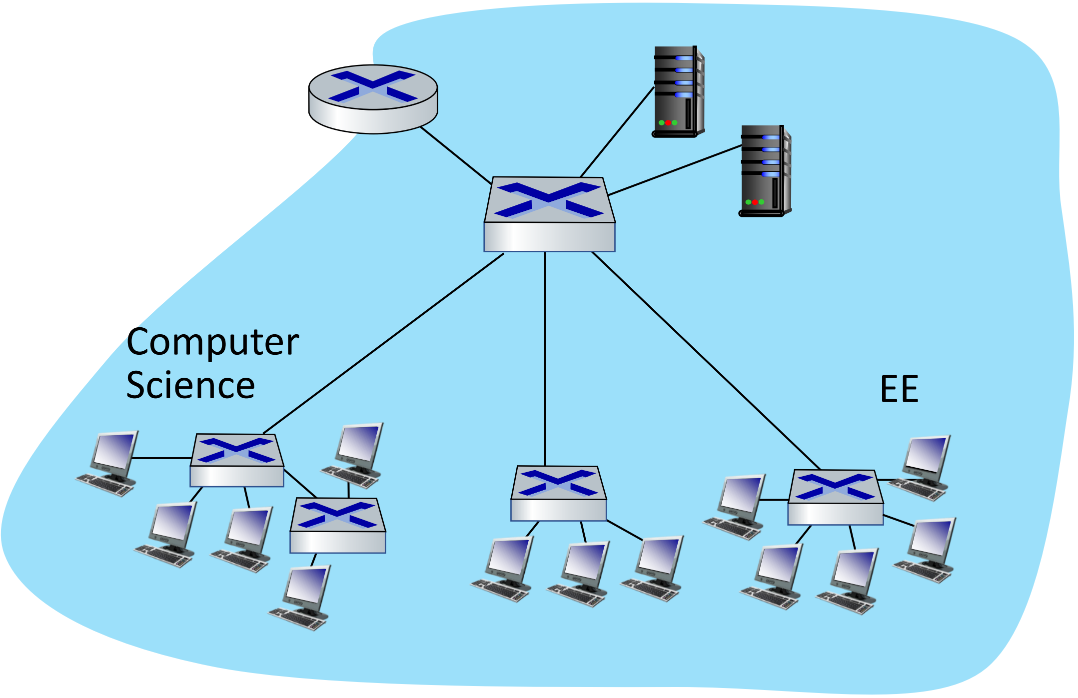
Virtual LANs (VLANs): motivation

Q: what happens as LAN sizes scale, users change point of attachment?



single broadcast domain:

- *scaling:* all layer-2 broadcast traffic (ARP, DHCP, unknown MAC) must cross entire LAN
- efficiency, security, privacy issues

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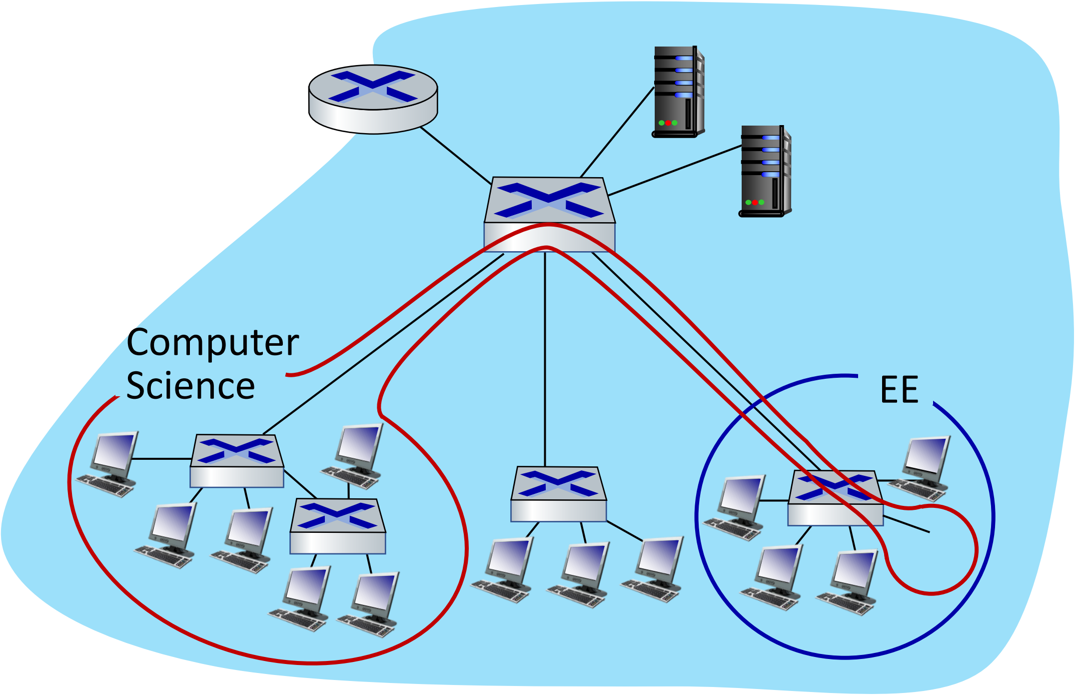


single broadcast domain:

- *scaling:* all layer-2 broadcast traffic (ARP, DHCP, unknown MAC) must cross entire LAN
- efficiency, security, privacy, efficiency issues

administrative issues:

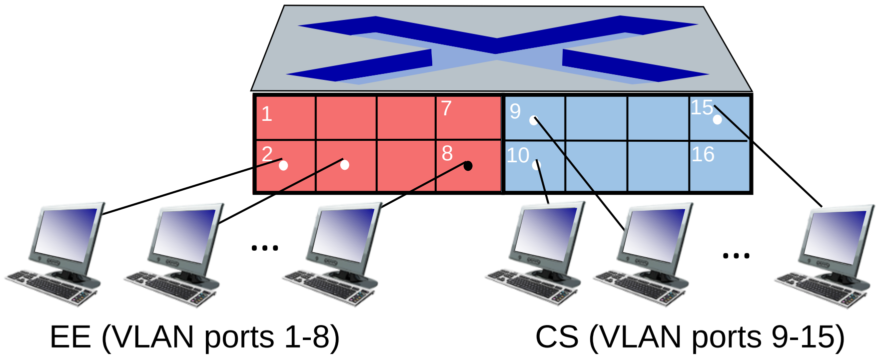
- CS user moves office to EE - *physically* attached to EE switch, but wants to remain *logically* attached to CS switch

Port-based VLANs

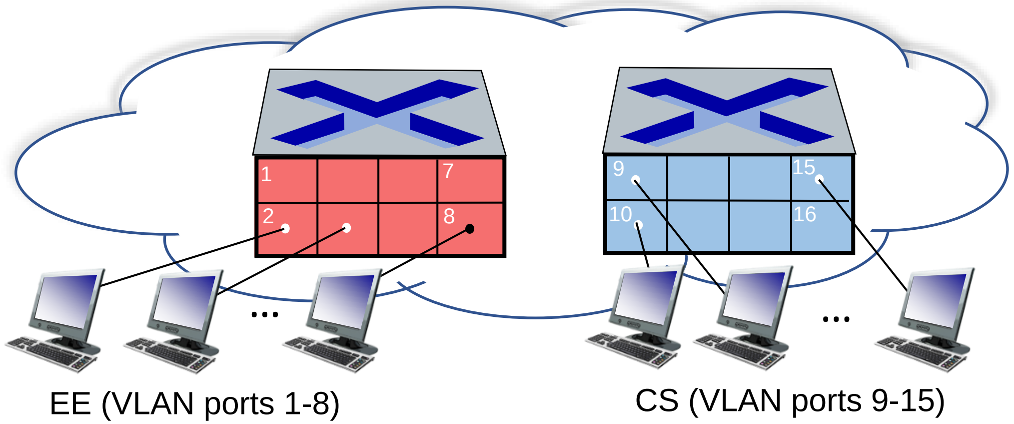
Virtual Local Area Network (VLAN)

switch(es) supporting VLAN capabilities can be configured to define multiple *virtual* LANS over single physical LAN infrastructure.

port-based VLAN: switch ports grouped (by switch management software) so that *single* physical switch ……

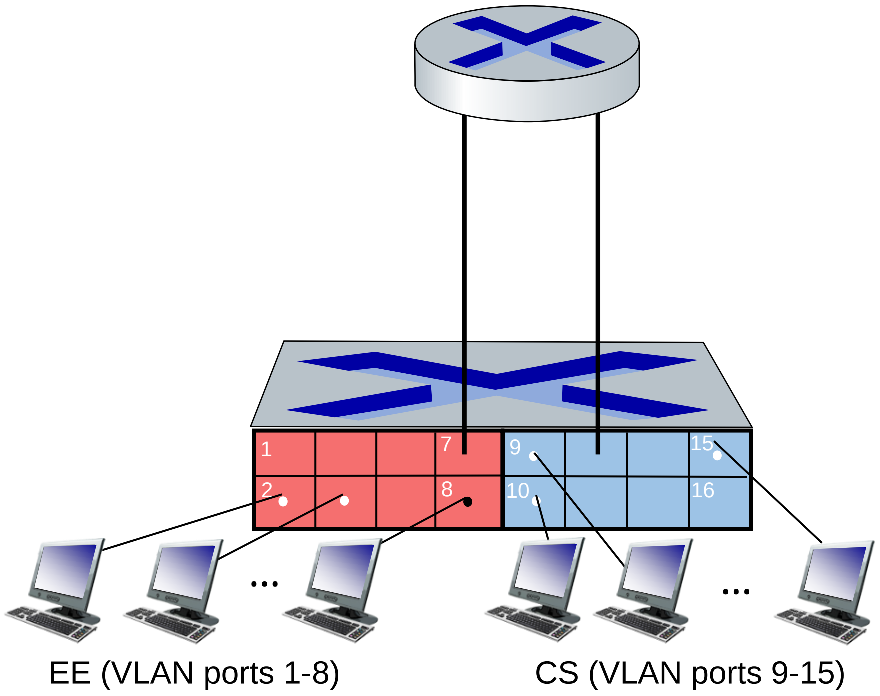


… operates as **multiple** virtual switches

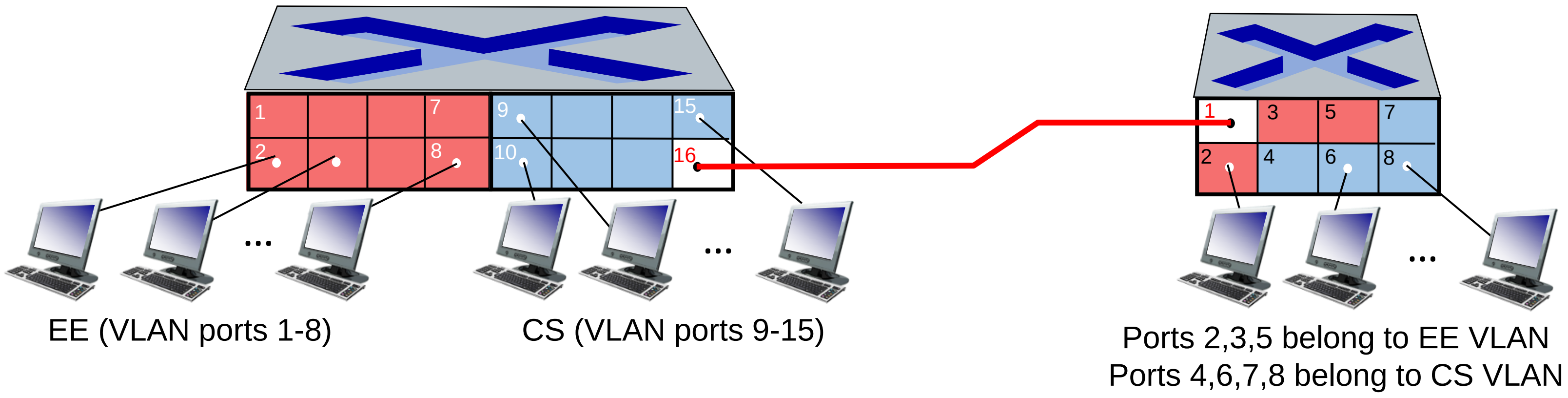


Port-based VLANs

- traffic isolation: frames to/from ports 1-8 can *only* reach ports 1-8
  - can also define VLAN based on MAC addresses of endpoints, rather than switch port
- dynamic membership: ports can be dynamically assigned among VLANs
- forwarding between VLANS: done via routing (just as with separate switches)
  - in practice vendors sell combined switches plus routers



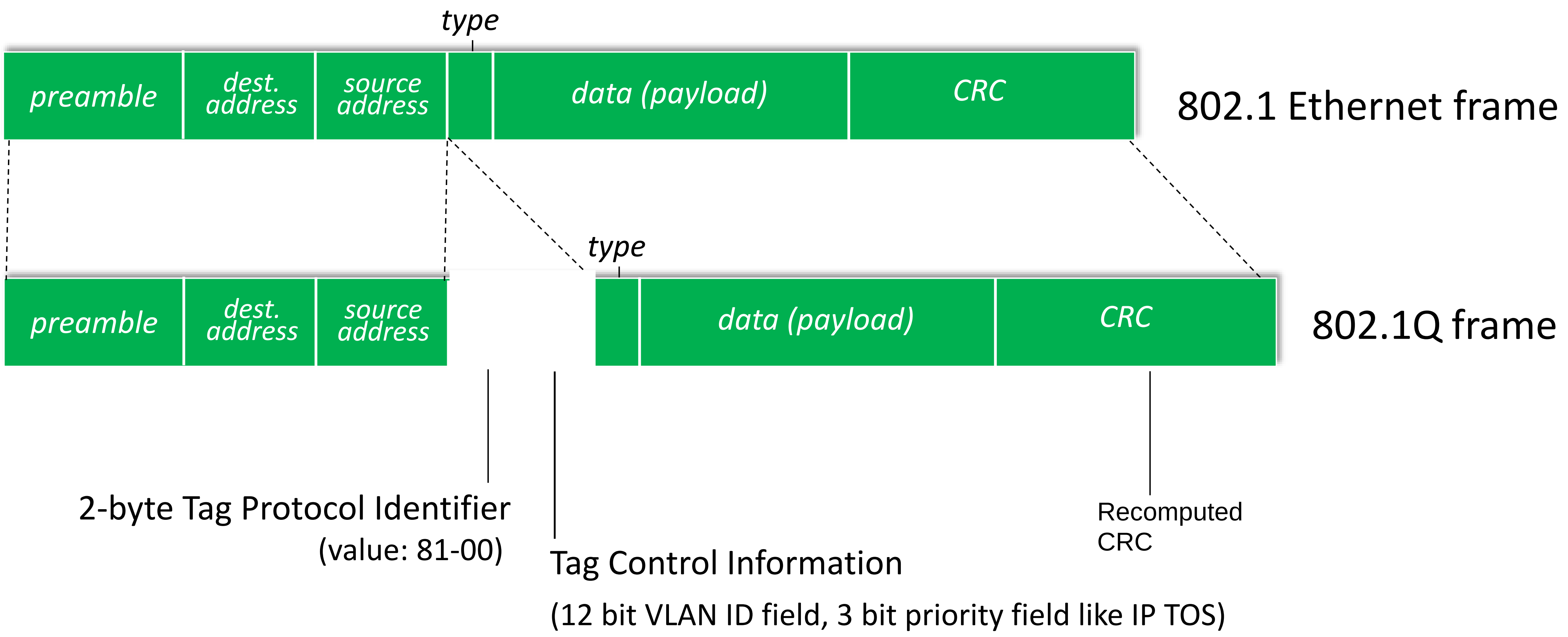
VLANS spanning multiple switches



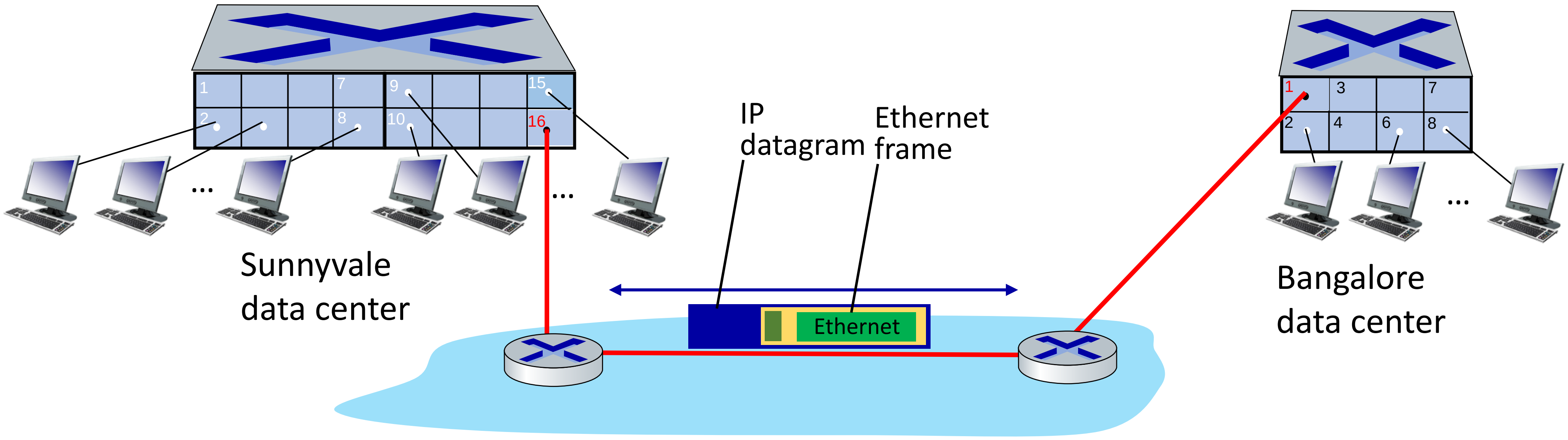
trunk port: carries frames between VLANS defined over multiple physical switches

- frames forwarded within VLAN between switches can't be vanilla 802.1 frames (must carry VLAN ID info)
- 802.1q protocol adds/removed additional header fields for frames forwarded between trunk ports

802.1Q VLAN frame format



EVPN: Ethernet VPNs (aka VXLANs)



Layer-2 Ethernet switches *logically* connected to each other (e.g., using IP as an *underlay*)

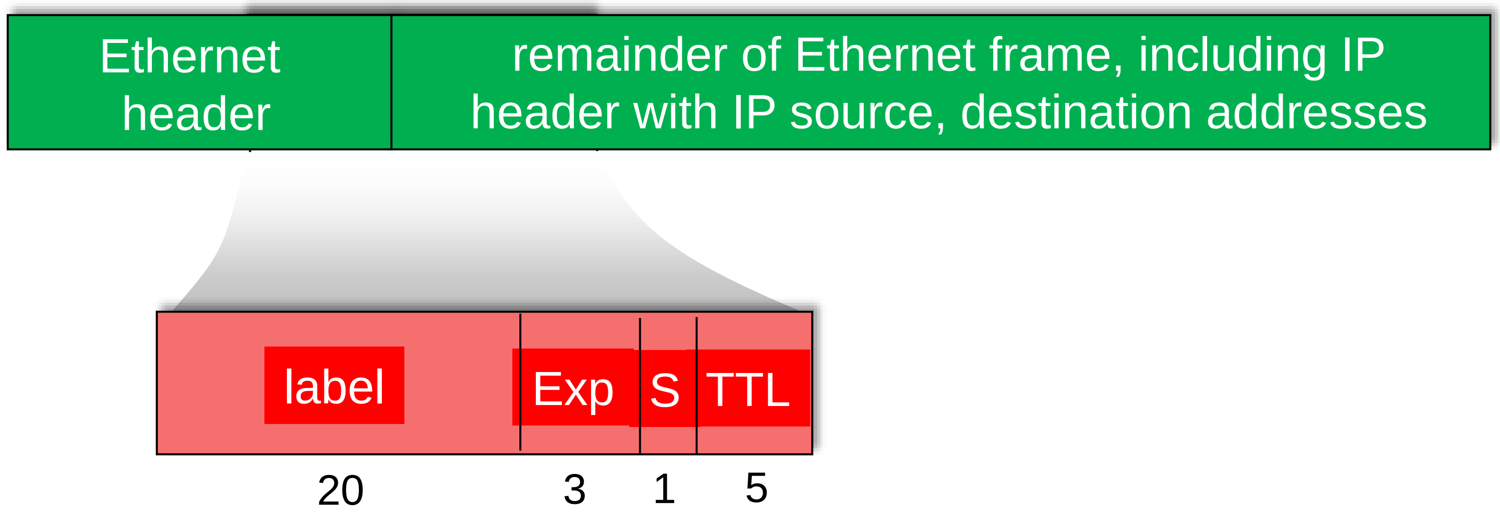
- Ethernet frames carried *within* IP datagrams between sites
- "*tunneling* scheme to *overlay Layer 2 networks on top of Layer 3 networks* ... runs over the existing networking infrastructure and provides a means to "stretch" a Layer 2 network." [RFC 7348]

Link layer, LANs: roadmap

- introduction
- error detection, correction
- multiple access protocols
- LANs
  - addressing, ARP
  - Ethernet
  - switches
  - VLANs
- link virtualization: MPLS
- data center networking
- a day in the life of a web request

Multiprotocol label switching (MPLS)

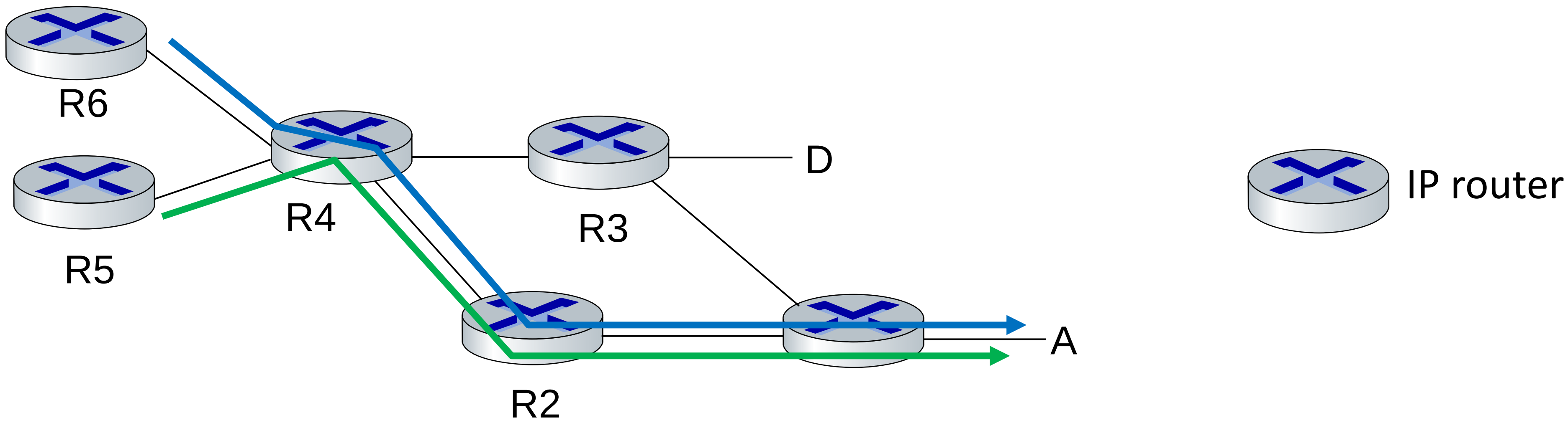
- goal: high-speed IP forwarding among network of MPLS-capable routers, using fixed length label (instead of shortest prefix matching)
  - faster lookup using fixed length identifier
  - borrowing ideas from Virtual Circuit (VC) approach
  - but IP datagram still keeps IP address!



MPLS capable routers

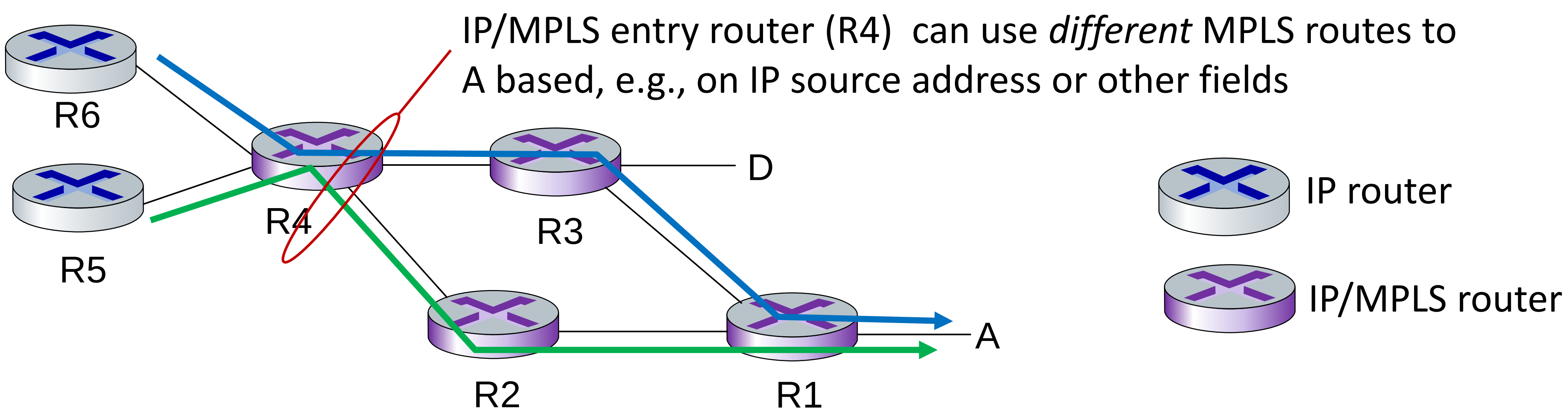
- a.k.a. label-switched router
- forward packets to outgoing interface based only on label value (*don't inspect IP address*)
  - MPLS forwarding table distinct from IP forwarding tables
- *flexibility:* MPLS forwarding decisions can *differ* from those of IP
  - use destination *and* source addresses to route flows to same destination differently (traffic engineering)
  - re-route flows quickly if link fails: pre-computed backup paths

MPLS versus IP paths



- IP routing: path to destination determined by destination address alone

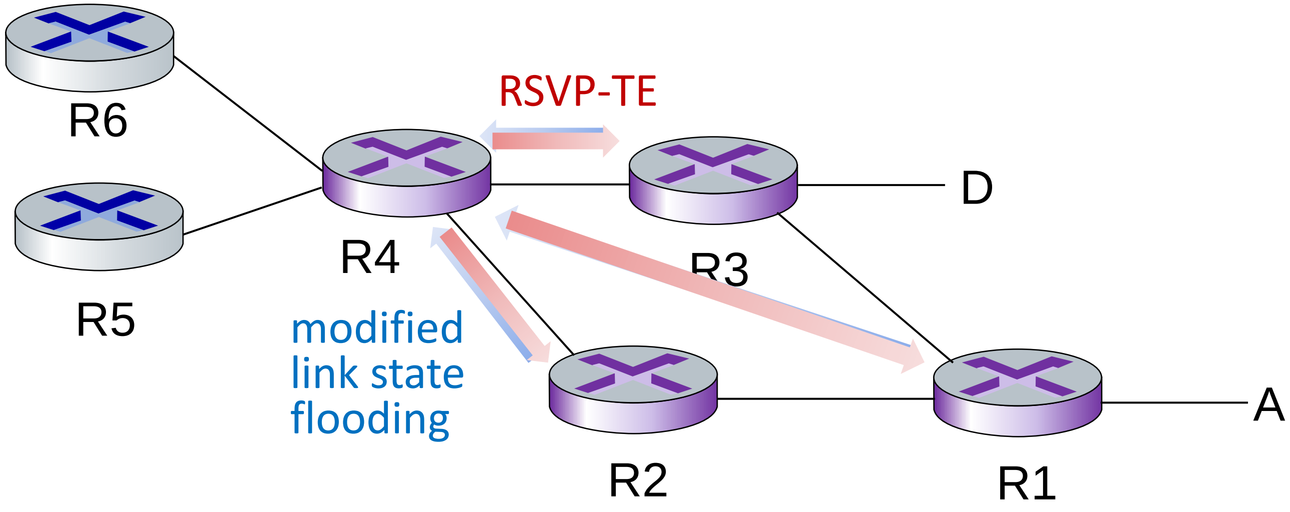
MPLS versus IP paths



- IP routing: path to destination determined by destination address alone
- MPLS routing: path to destination can be based on source *and* destination address
  - flavor of generalized forwarding (MPLS 10 years earlier)
  - *fast reroute:* precompute backup routes in case of link failure

MPLS signaling

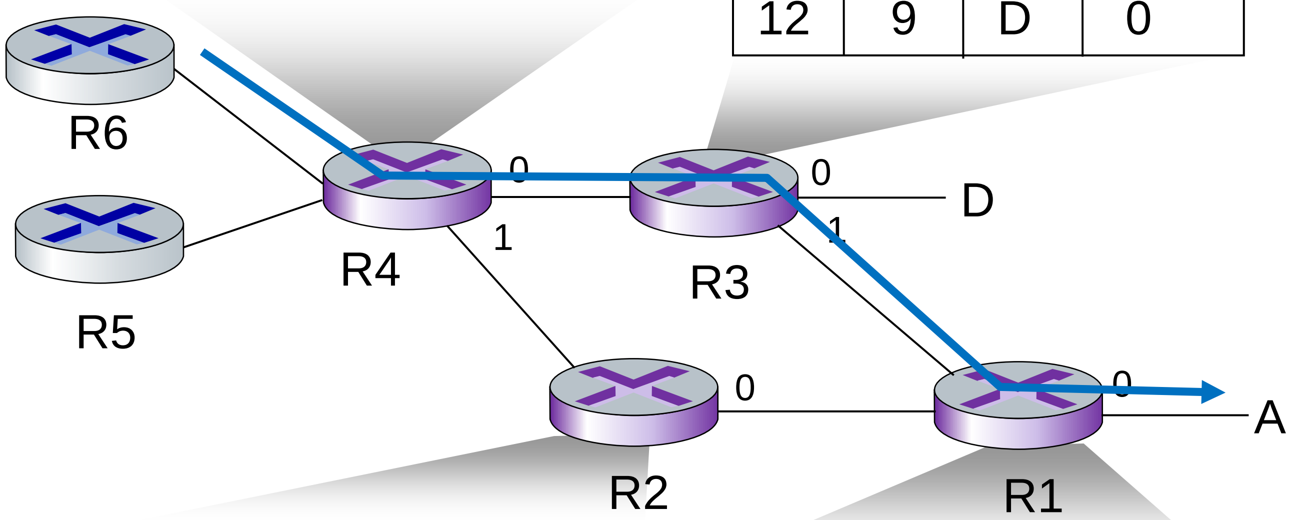
- modify OSPF, IS-IS link-state flooding protocols to carry info used by MPLS routing:
  - e.g., link bandwidth, amount of "reserved" link bandwidth
- entry MPLS router uses RSVP-TE signaling protocol to set up MPLS forwarding at downstream routers



MPLS forwarding tables

in label	out label	dest	out interface
	10	A	0
	12	D	0
	8	A	1

in label	out label	dest	out interface
10	6	A	1
12	9	D	0



in label	out label	dest	out interface
8	6	A	0

in label	out label	dest	out interface
6	-	A	0

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Datacenter networks

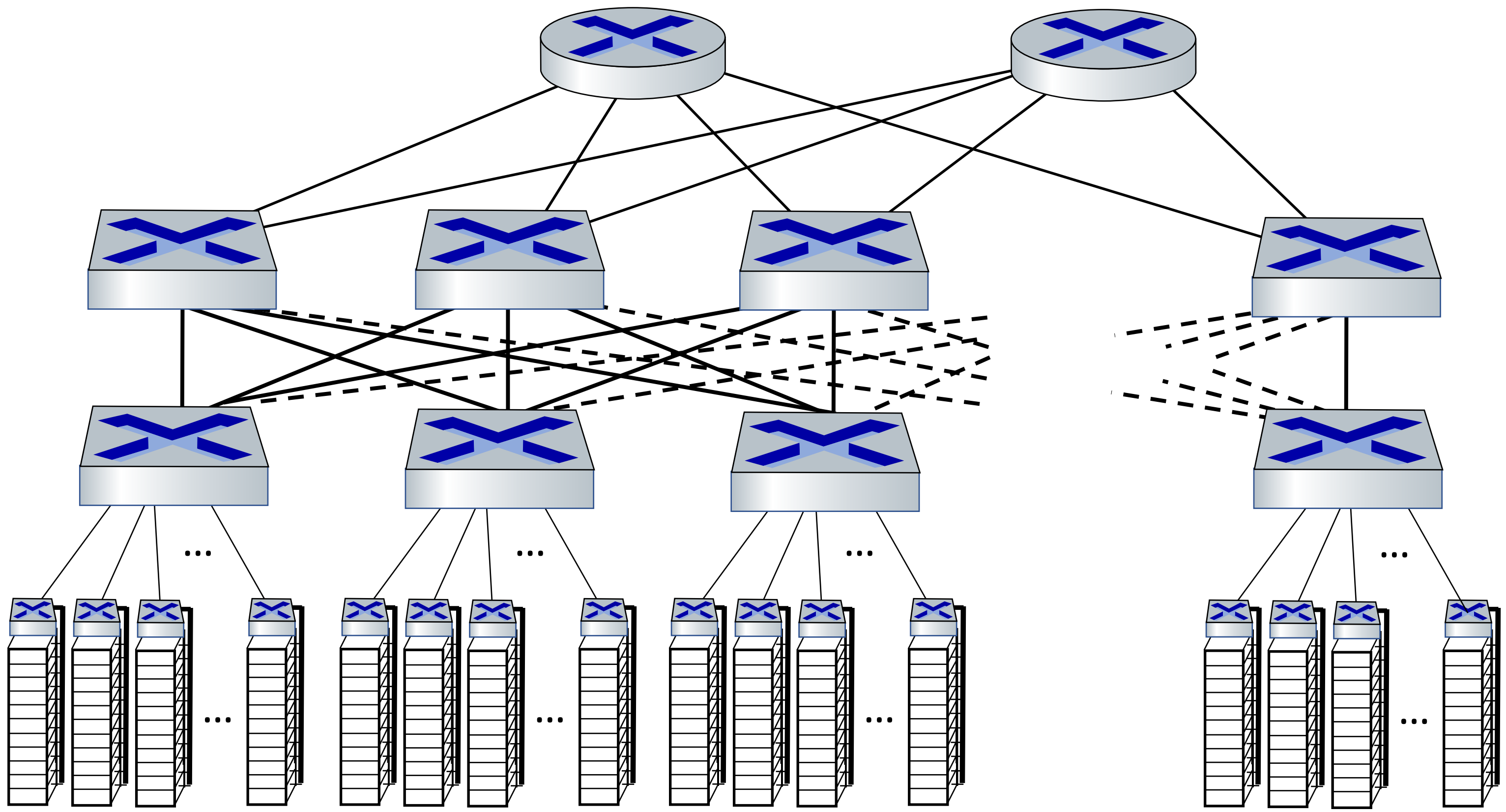
10's to 100's of thousands of hosts, often closely coupled, in close proximity:

- e-business (e.g. Amazon)
- content-servers (e.g., YouTube, Akamai, Apple, Microsoft)
- search engines, data mining (e.g., Google)

challenges:

- multiple applications, each serving massive numbers of clients
- reliability
- managing/balancing load, avoiding processing, networking, data bottlenecks

Datacenter networks: network elements



Border routers
- connections outside datacenter

Tier-1 switches
- connecting to ~16 T-2s below

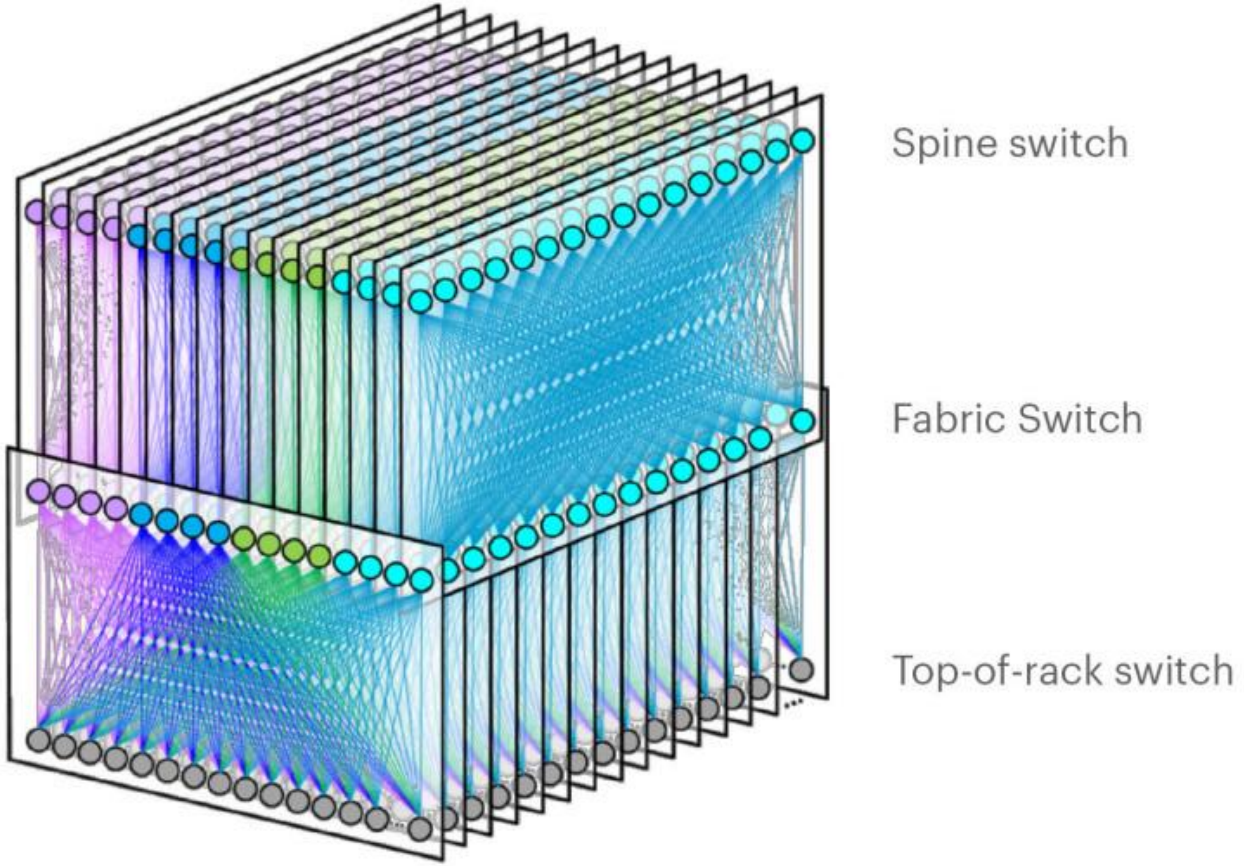
Tier-2 switches
- connecting to ~16 TORs below

Top of Rack (TOR) switch
- one per rack
- 100G-400G Ethernet to blades

Server racks
- 20- 40 server blades: hosts

Datacenter networks: network elements

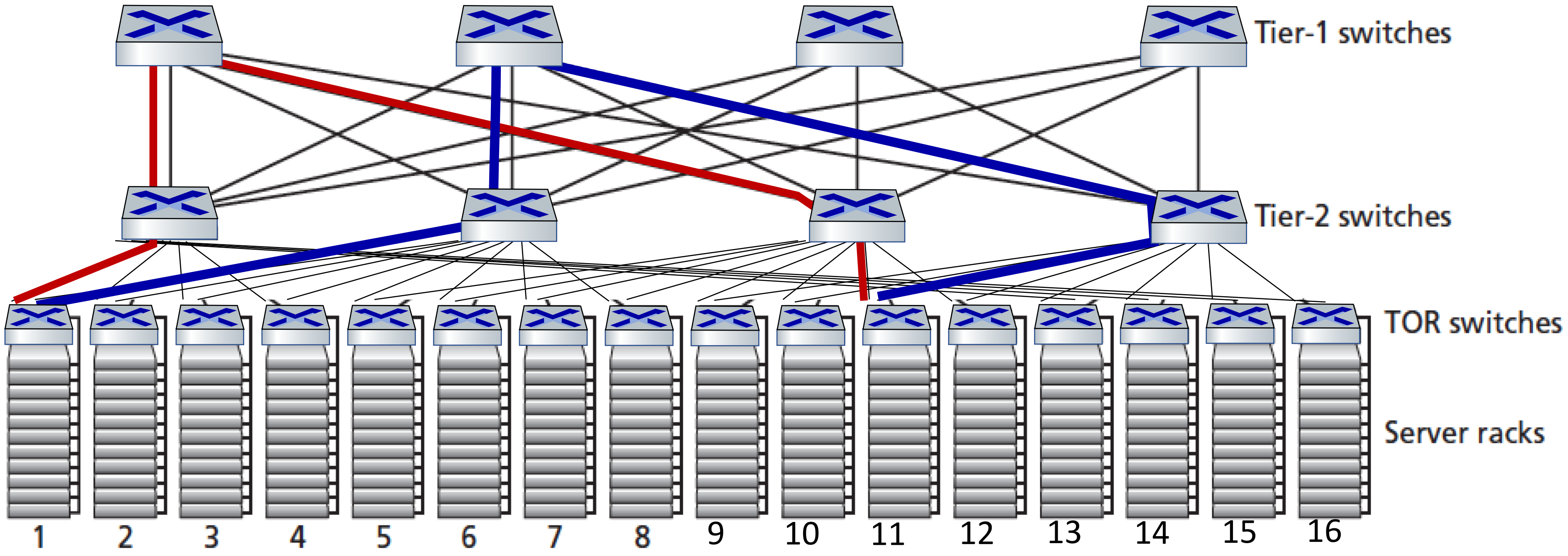
Facebook F16 data center network topology:



https://engineering.fb.com/data-center-engineering/f16-minipack/ (posted 3/2019)

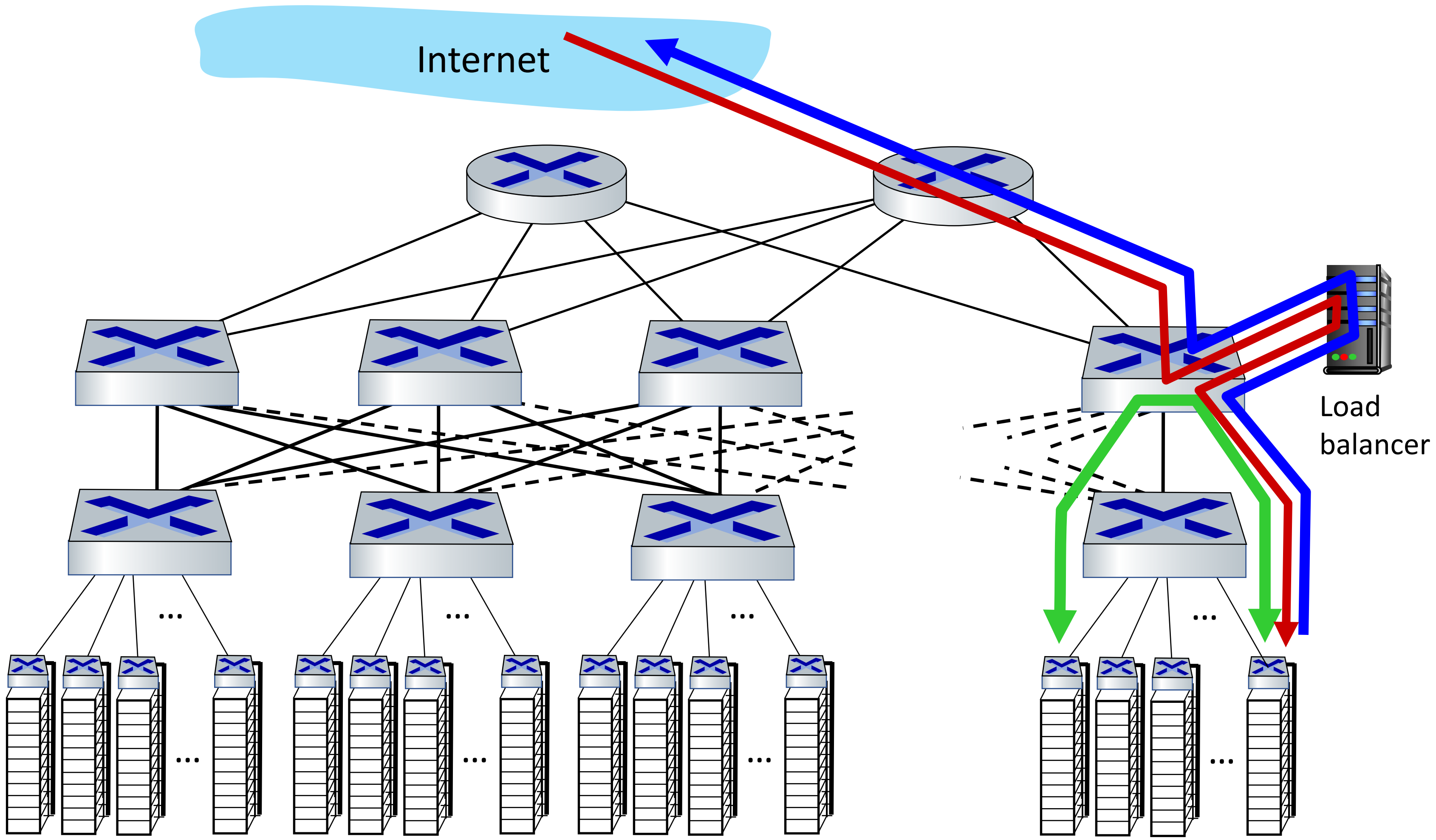
Datacenter networks: multipath

- rich interconnection among switches, racks:
  - increased throughput between racks (multiple routing paths possible)
  - increased reliability via redundancy



two disjoint paths highlighted between racks 1 and 11

Datacenter networks: application-layer routing



load balancer: application-layer routing

- receives external client requests
- directs workload within data center
- returns results to external client (hiding data center internals from client)

Datacenter networks: protocol innovations

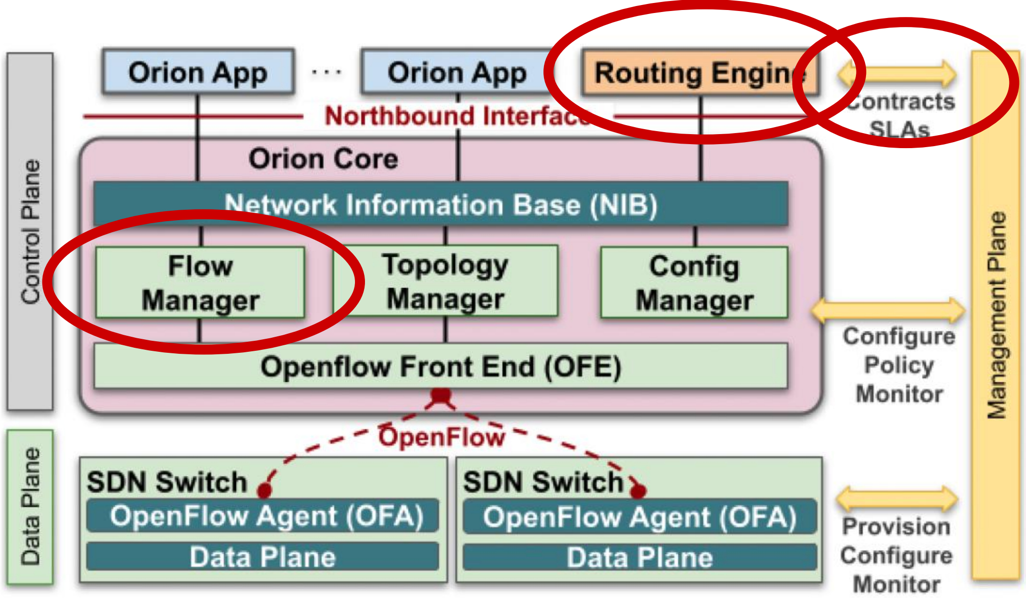
- link layer:
  - RoCE: remote DMA (RDMA) over Converged Ethernet
- transport layer:
  - ECN (explicit congestion notification) used in transport-layer congestion control (DCTCP, DCQCN)
  - experimentation with hop-by-hop (backpressure) congestion control
- routing, management:
  - SDN widely used within/among organizations' datacenters
  - place related services, data as close as possible (e.g., in same rack or nearby rack) to minimize tier-2, tier-1 communication

Google Networking: Infrastructure and Selected Challenges (Slides: https://networkingchannel.eu/google-networking-infrastructure-and-selected-challenges/

ORION: Google's new SDN control plane for internal datacenter (Jupiter) + wide area (B4) network

- routing (intradomain, iBGP), traffic engineering: implemented in *applications* on top of ORION core
- edge-edge flow-based controls (e.g., CoFlow scheduling) to meet contract SLAs
- management: pub-sub distributed microservices in Orion core, OpenFlow for switch signaling/monitoring

Orion SDN architecture and core apps



Note:

- no routing protocols, congestion control (partially) also managed by SDN rather than by protocol
- are protocols dying?

Link layer, LANs: roadmap

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  - Ethernet
  - switches
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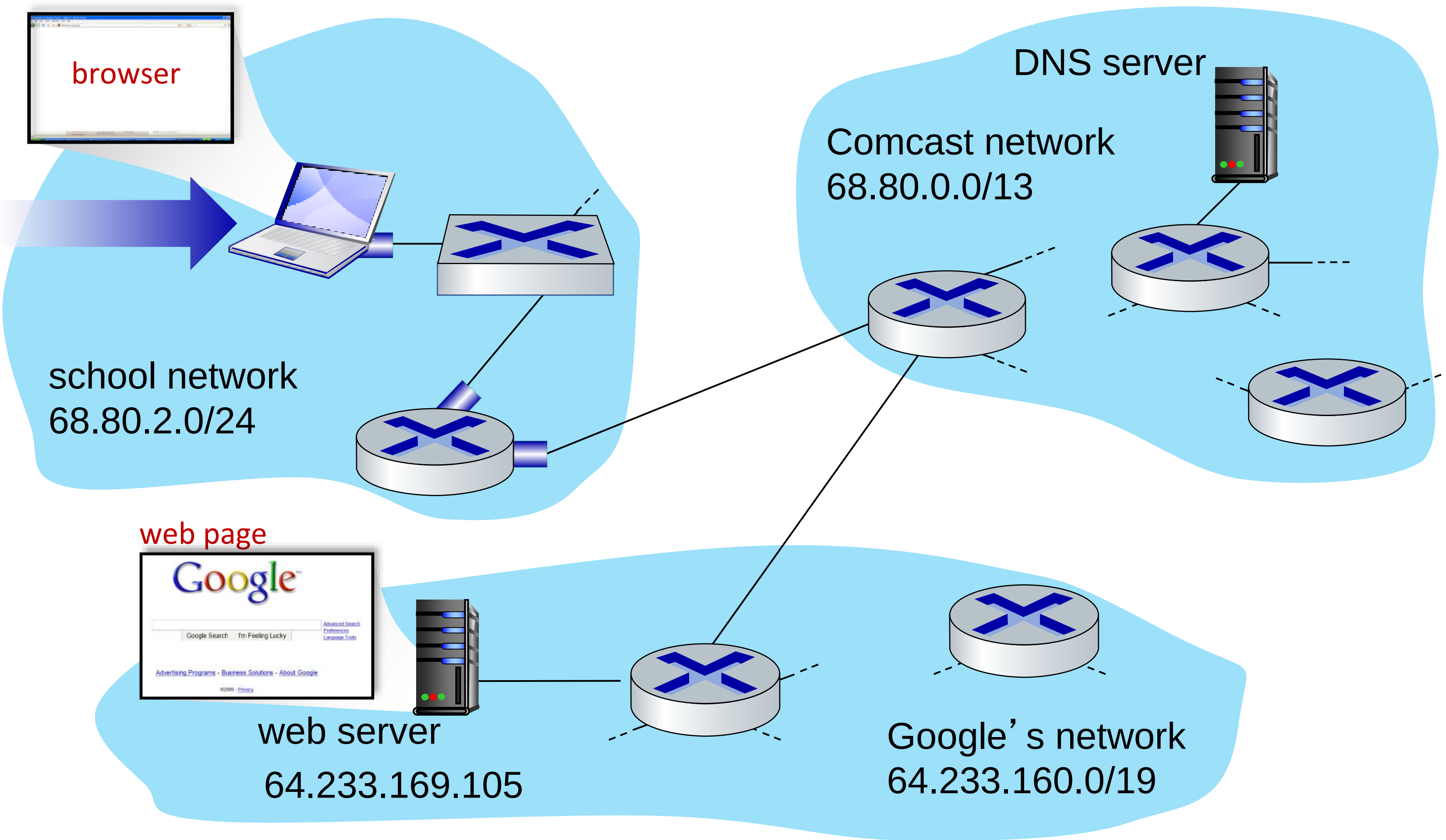
Synthesis: a day in the life of a web request

- our journey down the protocol stack is now complete!
  - application, transport, network, link
- putting-it-all-together: synthesis!
  - *goal:* identify, review, understand protocols (at all layers) involved in seemingly simple scenario: requesting www page
  - *scenario:* student attaches laptop to campus network, requests/receives www.google.com

A day in the life: scenario

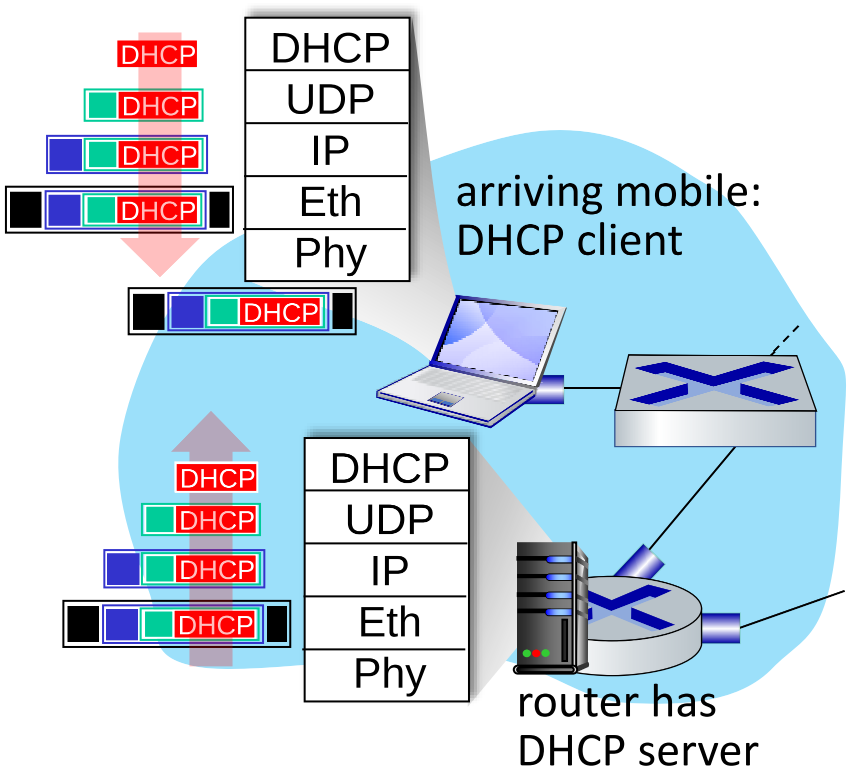


scenario:

- arriving mobile client attaches to network …
- requests web page: www.google.com

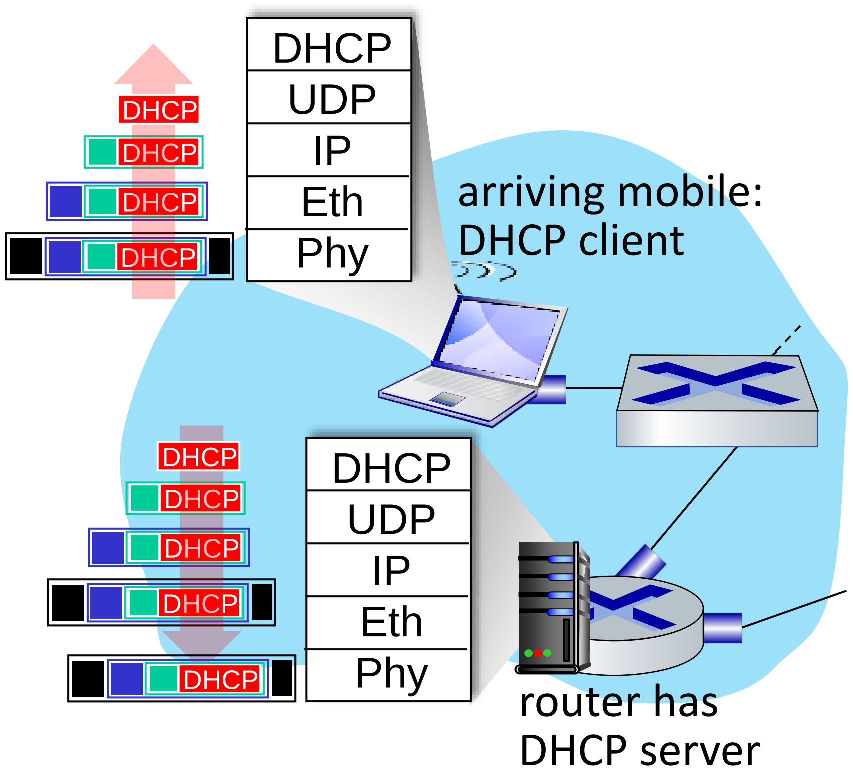
Sounds simple!

A day in the life: connecting to the Internet



- connecting laptop needs to get its own IP address, addr of first-hop router, addr of DNS server: use DHCP
- DHCP request encapsulated in UDP, encapsulated in IP, encapsulated in 802.3 Ethernet
- Ethernet frame broadcast (dest: FFFFFFFFFFFF) on LAN, received at router running DHCP server
- Ethernet de-muxed to IP de-muxed, UDP de-muxed to DHCP

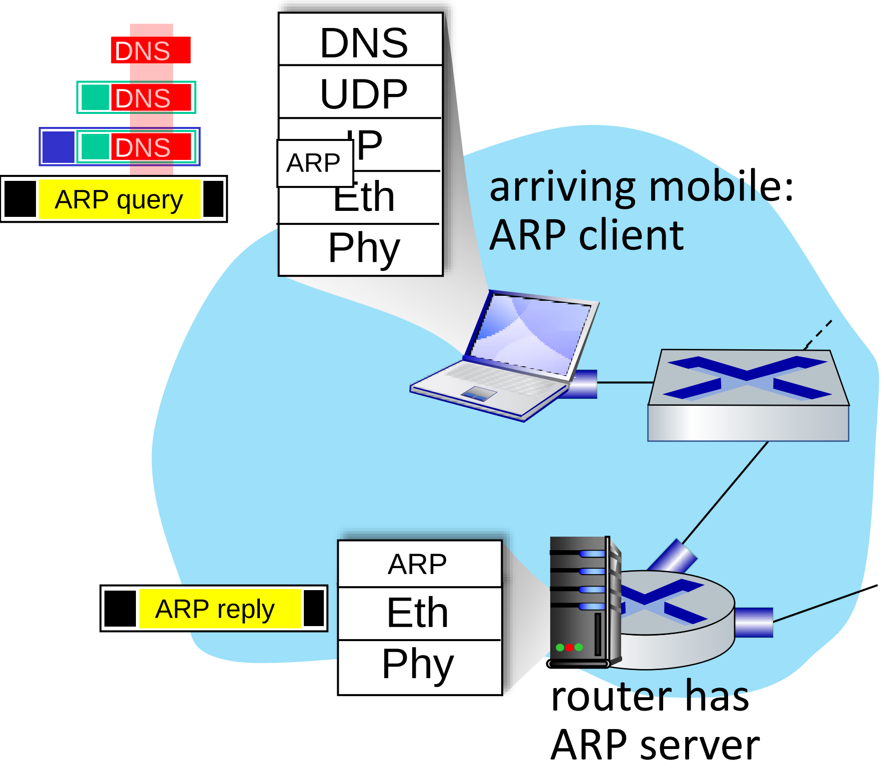
A day in the life: connecting to the Internet



- DHCP server formulates DHCP ACK containing client's IP address, IP address of first-hop router for client, name & IP address of DNS server
- encapsulation at DHCP server, frame forwarded (switch learning) through LAN, demultiplexing at client
- DHCP client receives DHCP ACK reply

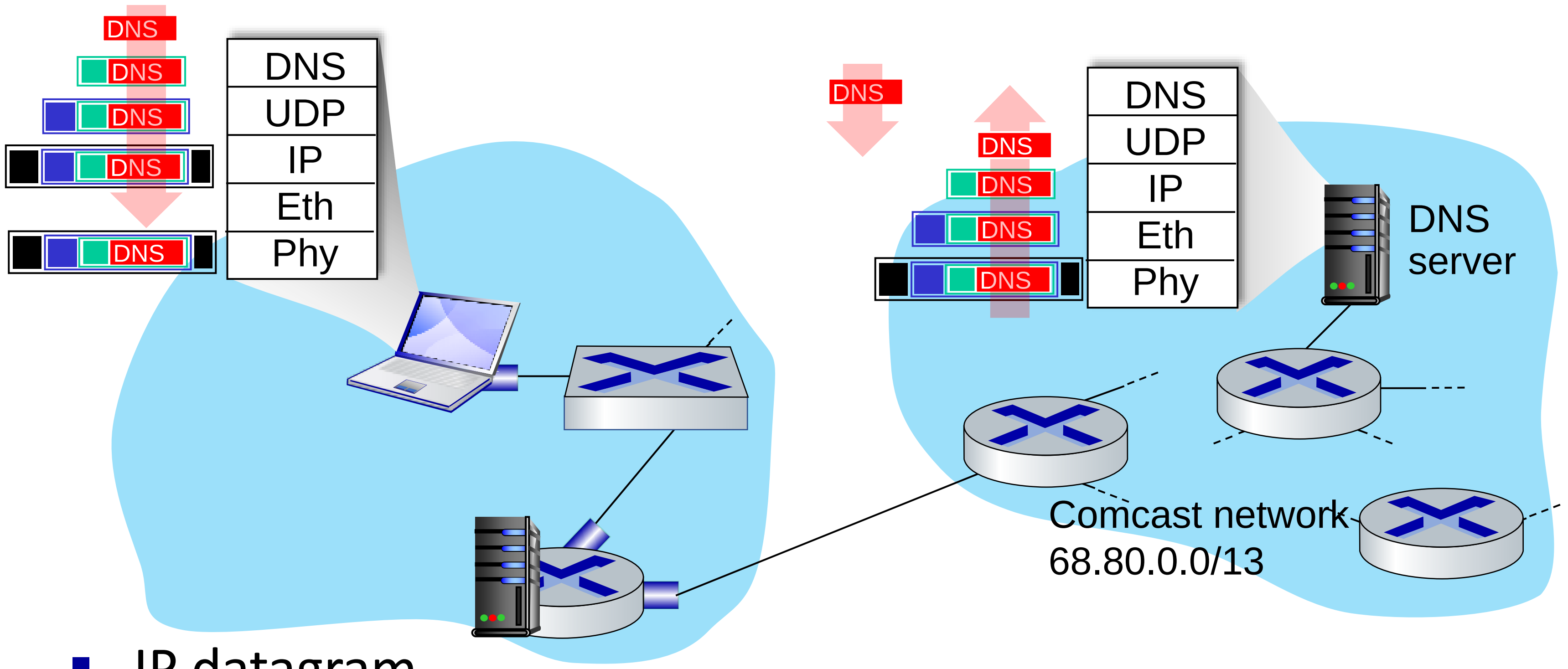
Client now has IP address, knows name & addr of DNS server, IP address of its first-hop router

A day in the life… ARP (before DNS, before HTTP)



- before sending HTTP request, need IP address of www.google.com: DNS
- DNS query created, encapsulated in UDP, encapsulated in IP, encapsulated in Eth. To send frame to router, need MAC address of router interface: ARP
- ARP query broadcast, received by router, which replies with ARP reply giving MAC address of router interface
- client now knows MAC address of first hop router, so can now send frame containing DNS query

A day in the life… using DNS

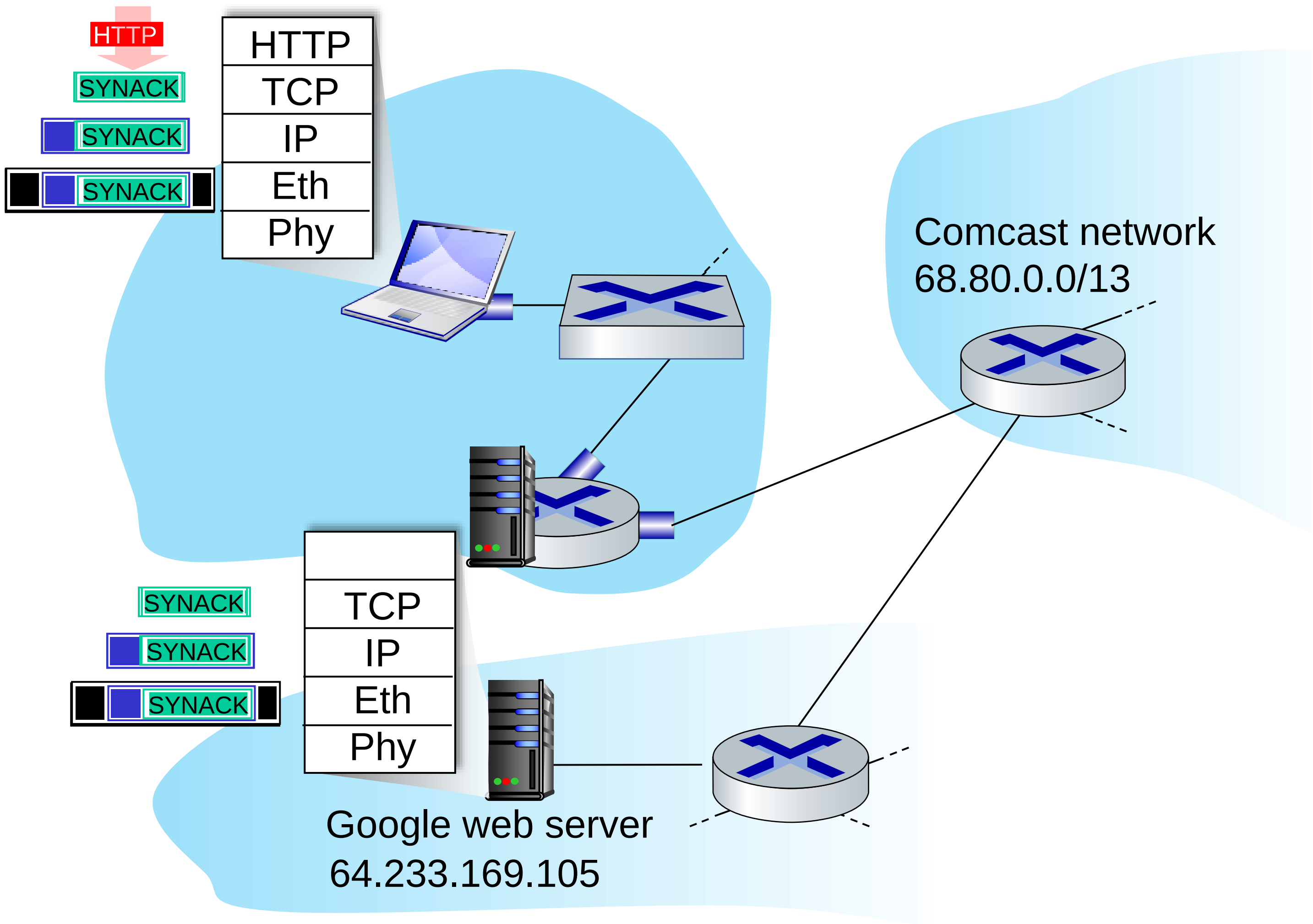


- de-muxed to DNS
- DNS replies to client with IP address of www.google.com

- IP datagram containing DNS query forwarded via LAN switch from client to 1st hop router

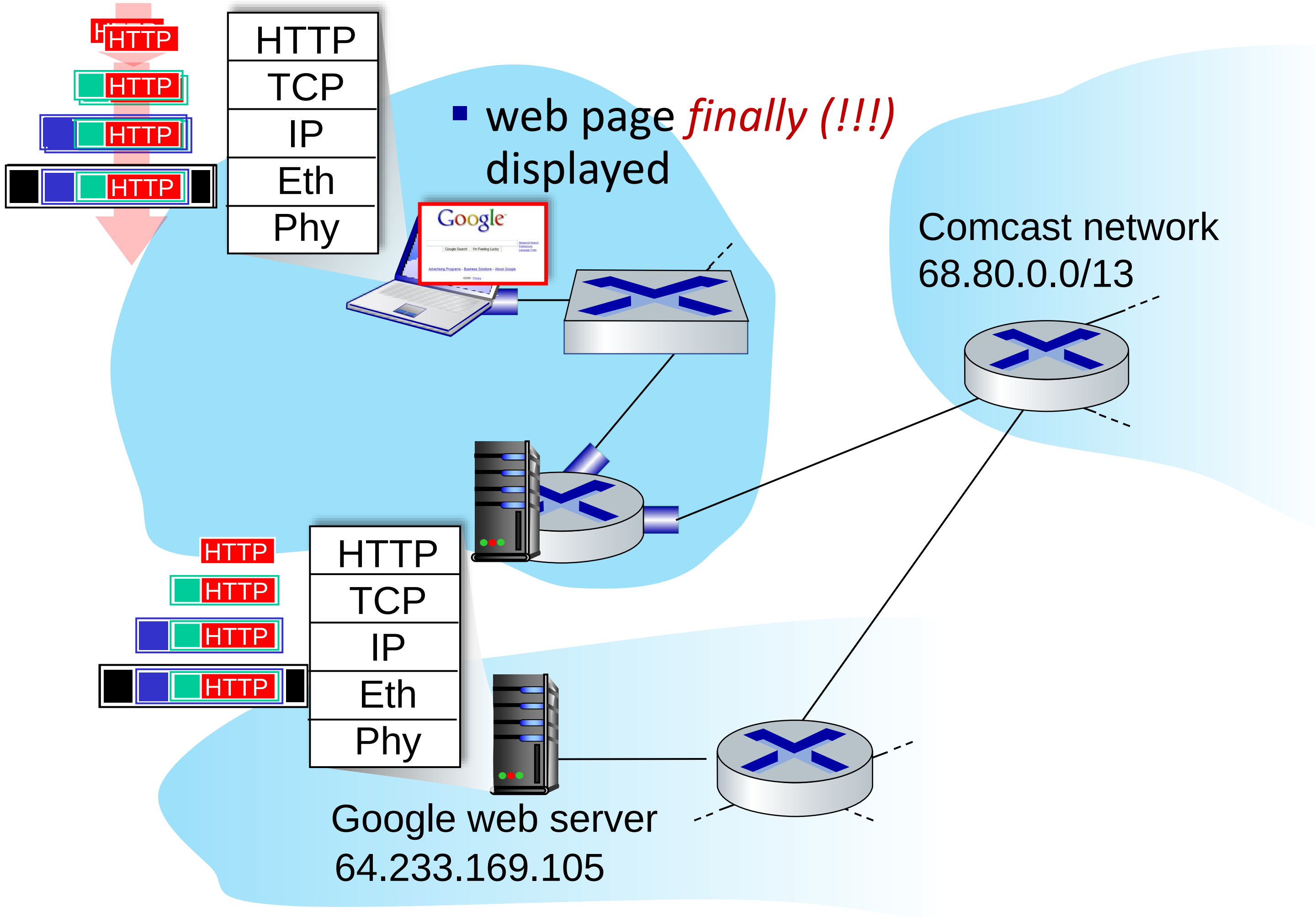
- IP datagram forwarded from campus network into Comcast network, routed (tables created by RIP, OSPF, IS-IS and/or BGP routing protocols) to DNS server

A day in the life...TCP connection carrying HTTP



- to send HTTP request, client first opens TCP socket to web server
- TCP SYN segment (step 1 in TCP 3-way handshake) inter-domain routed to web server
- web server responds with TCP SYNACK (step 2 in TCP 3-way handshake)
- TCP connection established!

A day in the life… HTTP request/reply



- HTTP request sent into TCP socket
- IP datagram containing HTTP request routed to www.google.com
- web server responds with HTTP reply (containing web page)
- IP datagram containing HTTP reply routed back to client

Chapter 7
Wireless and Mobile Networks

Wireless and Mobile Networks: context

- more wireless (mobile) phone subscribers than fixed (wired) phone subscribers (10-to-1 in 2019)!
- more mobile-broadband-connected devices than fixed-broadband-connected devices devices (5-1 in 2019)!
  - 4G/5G cellular networks now embracing Internet protocol stack, including SDN
- two important (but different) challenges
  - wireless: communication over wireless link
  - mobility: handling the mobile user who changes point of attachment to network

Chapter 7 outline

- Introduction

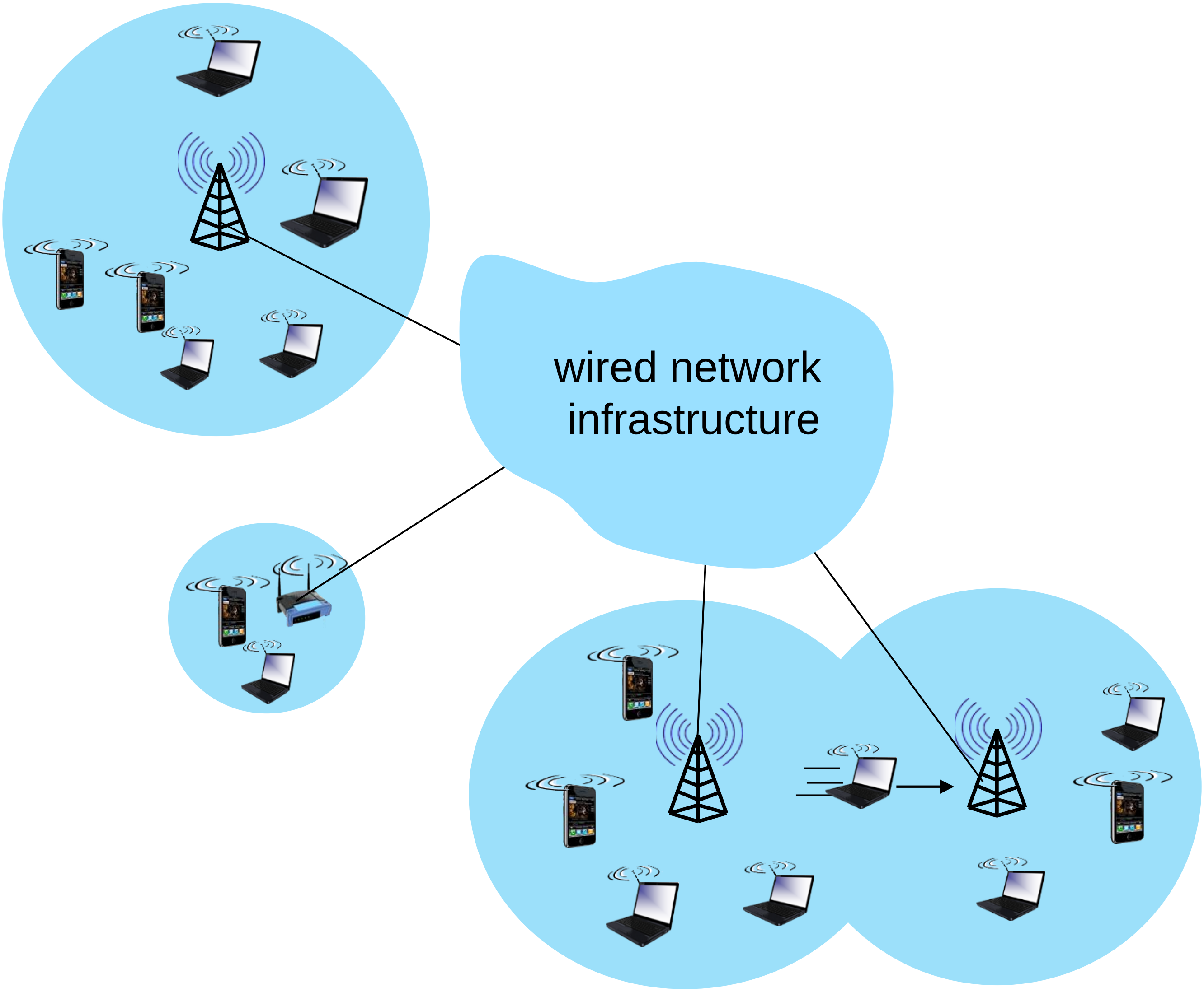
Wireless

- Wireless Links and network characteristics
- WiFi: 802.11 wireless LANs
- Cellular networks: 4G and 5G

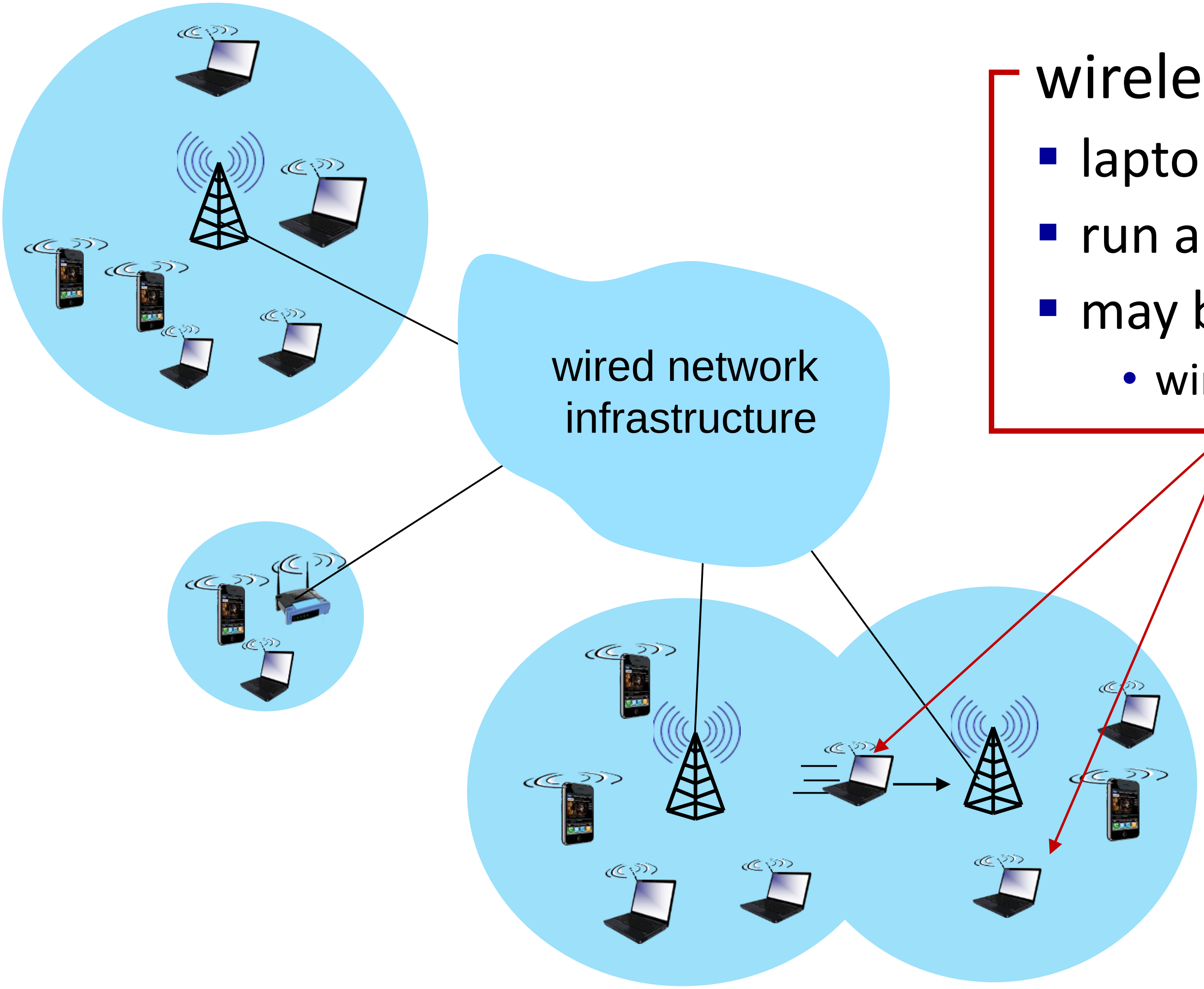
Mobility

- Mobility management: principles
- Mobility management: practice
  - 4G/5G networks
  - Mobile IP
- Mobility: impact on higher-layer protocols

Elements of a wireless network



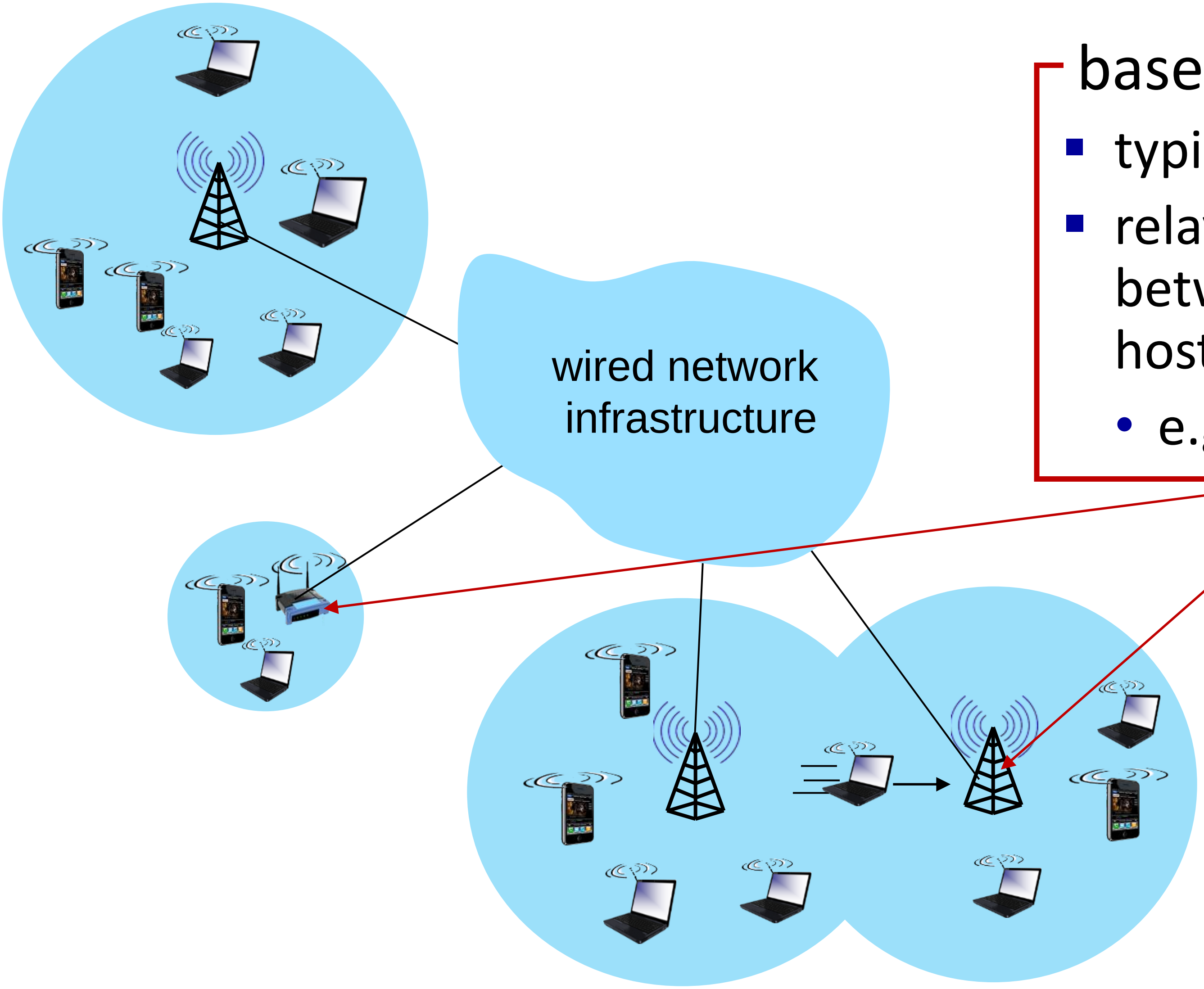
Elements of a wireless network



wireless hosts

- laptop, smartphone, IoT
- run applications
- may be stationary (non-mobile) or mobile
  - wireless does *not* always mean mobility!

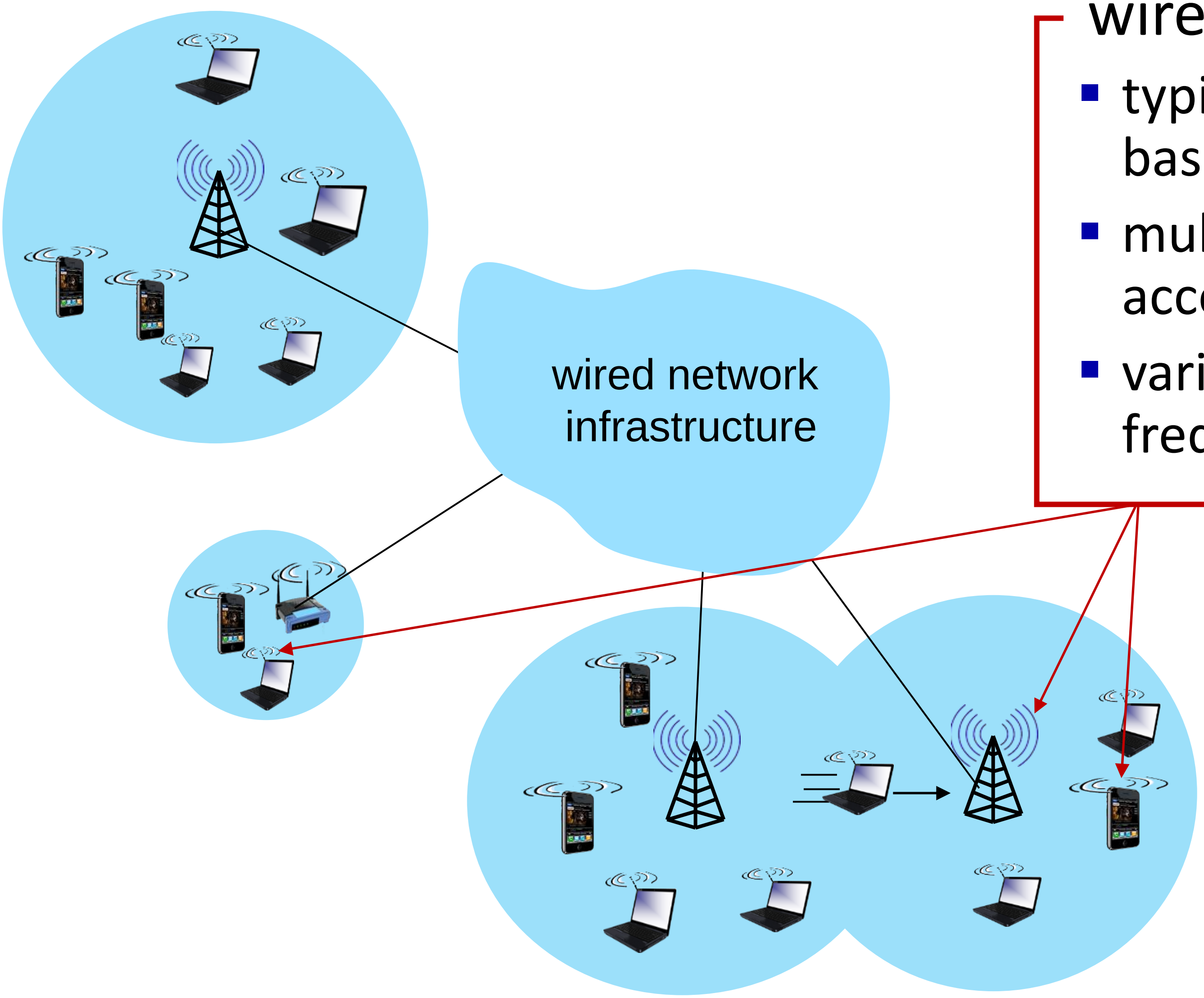
Elements of a wireless network



base station 

- typically connected to wired network
- relay - responsible for sending packets between wired network and wireless host(s) in its “area”
  - e.g., cell towers, 802.11 access points

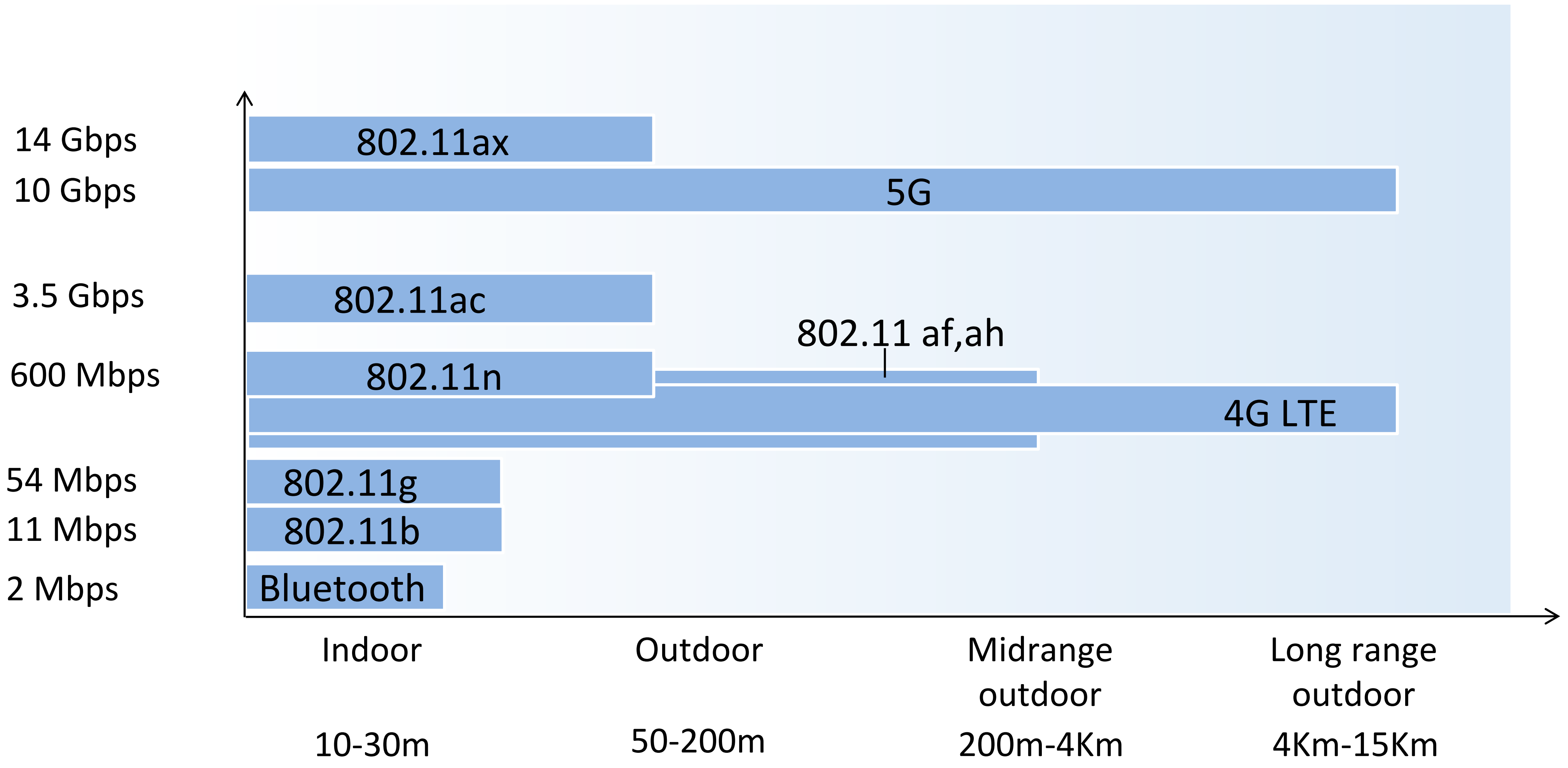
Elements of a wireless network



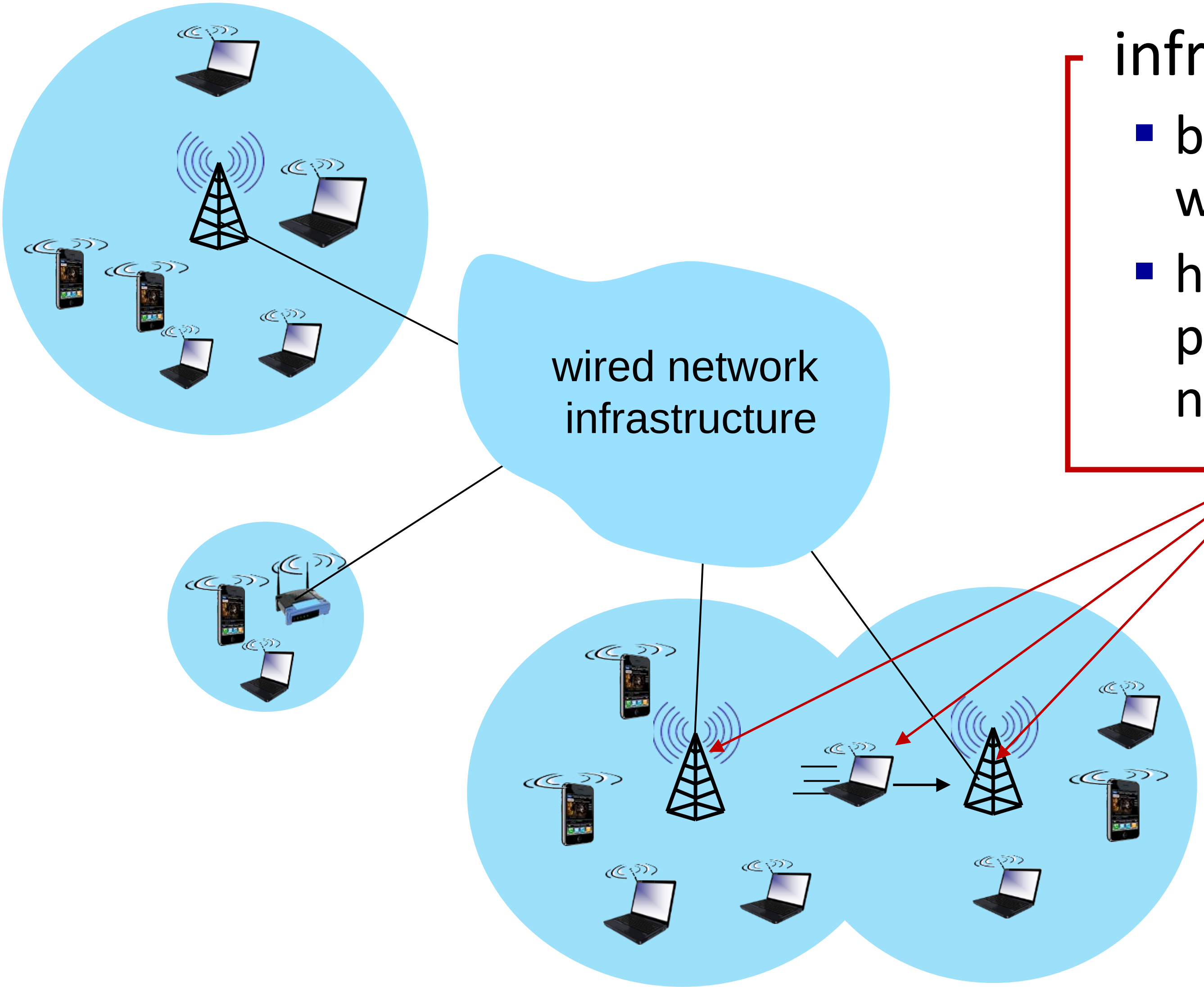
wireless link

- typically used to connect mobile(s) to base station, also used as backbone link
- multiple access protocol coordinates link access
- various transmission rates and distances, frequency bands

Characteristics of selected wireless links



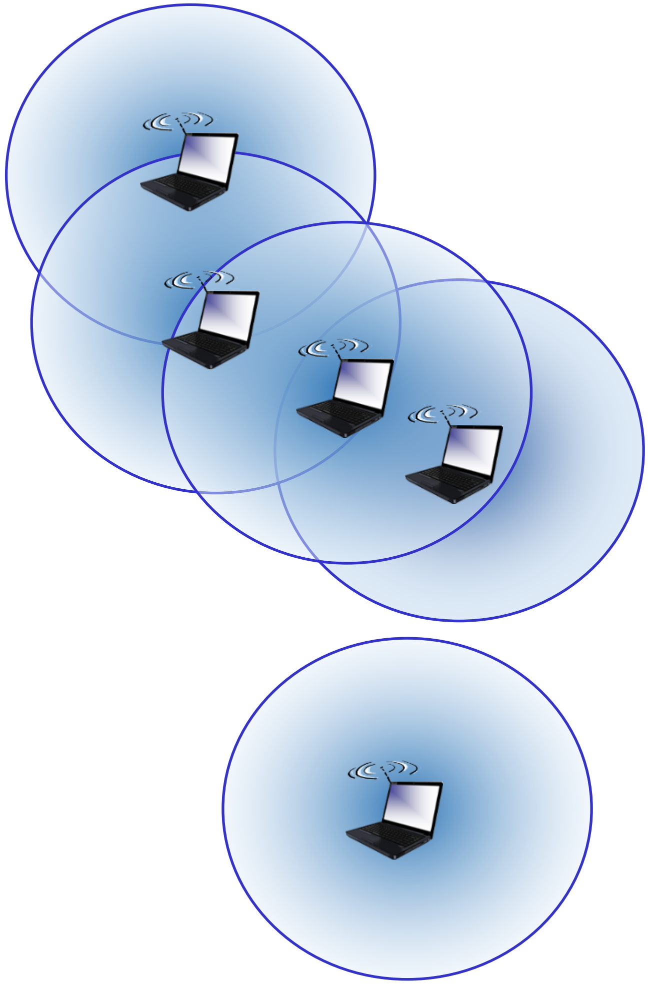
Elements of a wireless network



infrastructure mode

- base station connects mobiles into wired network
- handoff: mobile changes base station providing connection into wired network

Elements of a wireless network



ad hoc mode

- no base stations
- nodes can only transmit to other nodes within link coverage
- nodes organize themselves into a network: route among themselves

Wireless network taxonomy

	single hop	multiple hops
infrastructure (e.g., APs)	host connects to base station (WiFi, cellular) which connects to larger Internet	host may have to relay through several wireless nodes to connect to larger Internet: *mesh net*
no infrastructure	no base station, no connection to larger Internet (Bluetooth, ad hoc nets)	no base station, no connection to larger Internet. May have to relay to reach other a given wireless node MANET, VANET

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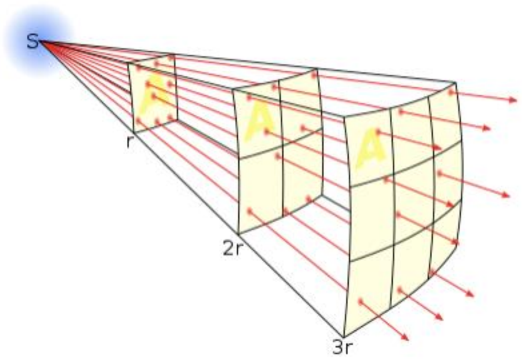
Wireless link characteristics: fading (attenuation)

Wireless radio signal attenuates (loses power) as it propagates (free space "path loss")

Free space path loss ~ $(fd)^2$

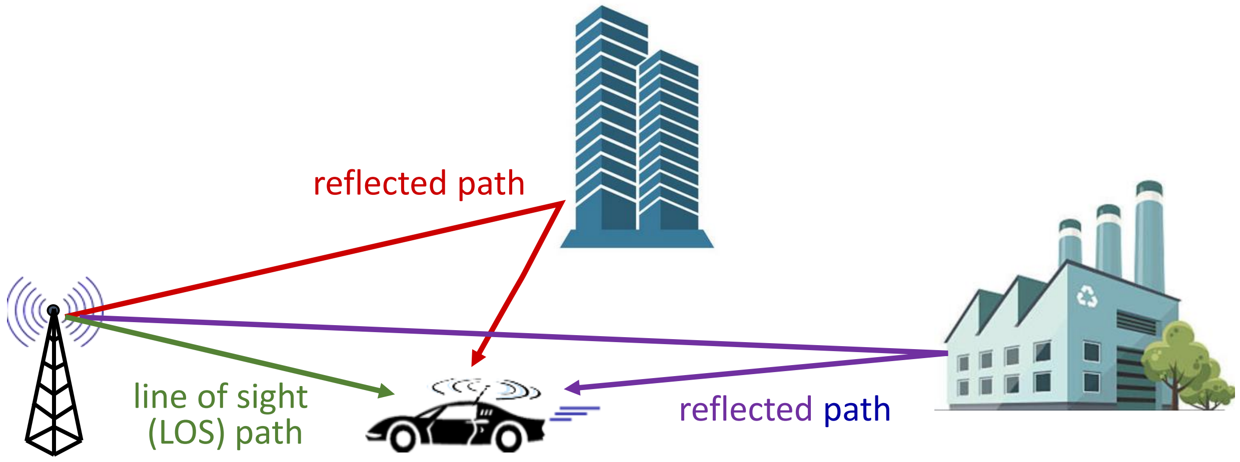
f: frequency
d: distance



higher frequency or longer distance → larger free space path loss

Wireless link characteristics: multipath

multipath propagation: radio signal reflects off objects ground, built environment, arriving at destination at slightly different times



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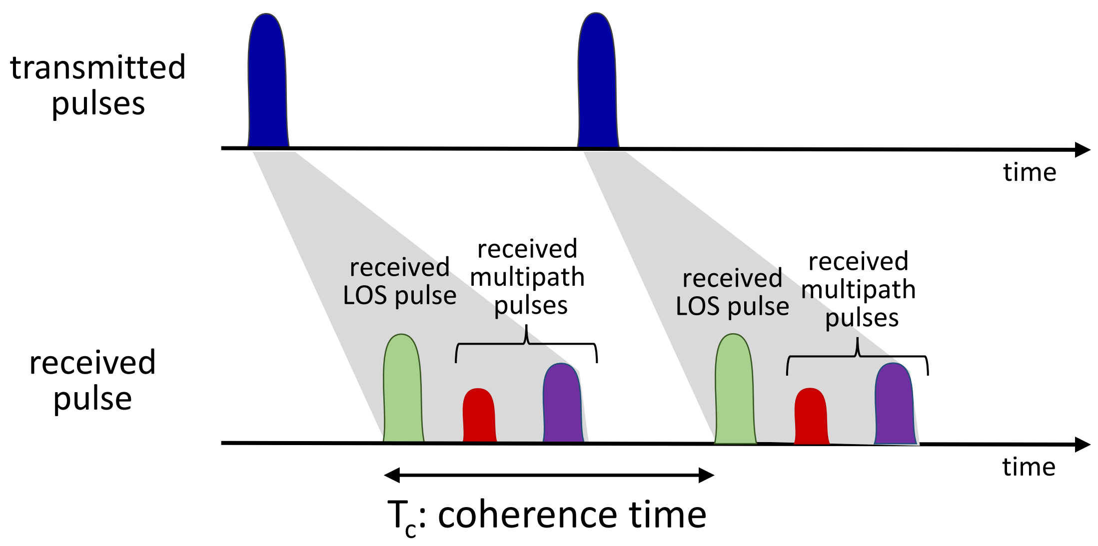
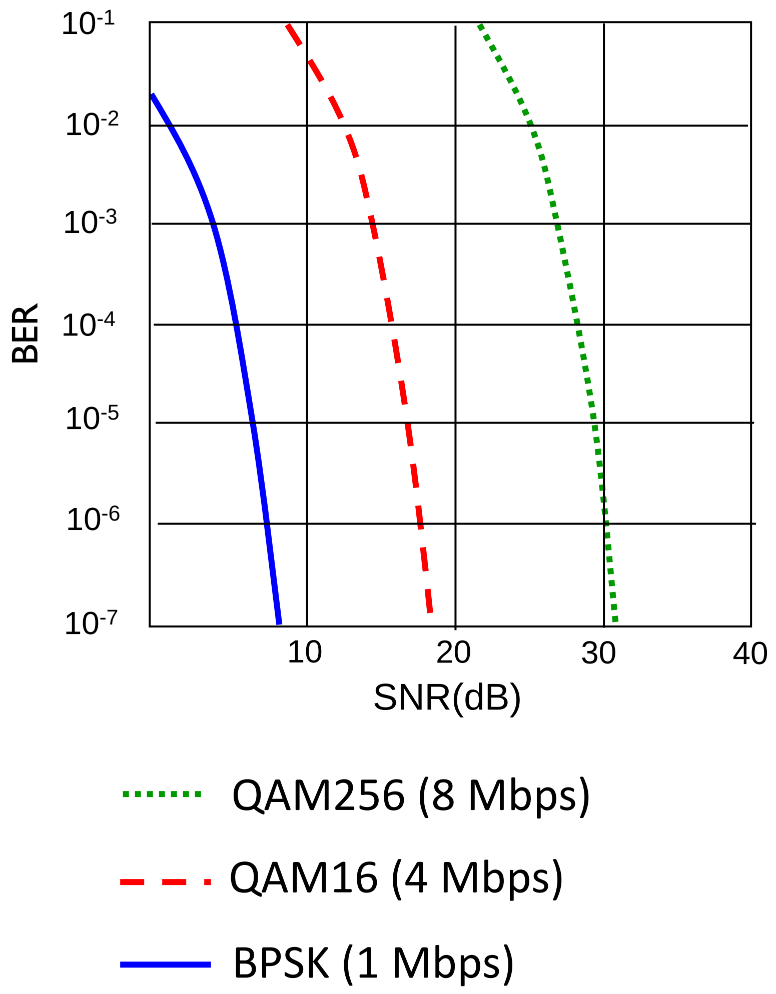


Coherence time:

- amount of time bit is present in channel to be received
- influences maximum possible transmission rate, since coherence times can not overlap
- inversely proportional to
  - frequency
  - receiver velocity

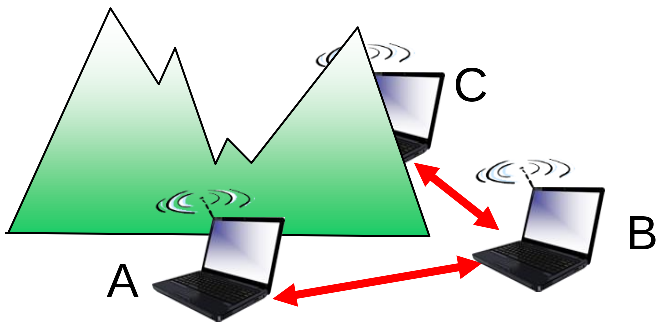
Wireless link characteristics: noise

- interference from other sources on wireless network frequencies: motors, appliances
- SNR: signal-to-noise ratio
  - larger SNR – easier to extract signal from noise (a "good thing")
- SNR versus BER tradeoff
  - *given physical layer:* increase power -> increase SNR->decrease BER
  - SNR may change with mobility: dynamically adapt physical layer (modulation technique, rate)



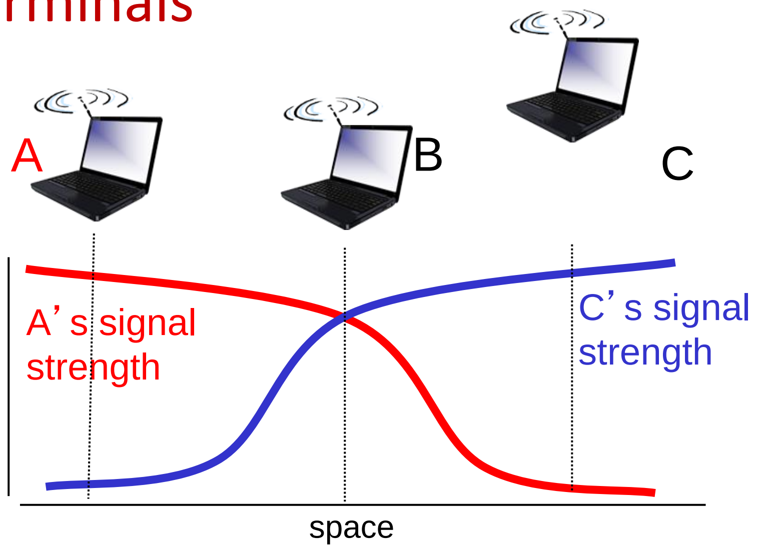
Wireless link characteristics: hidden terminals

Hidden terminal problem



- B, A hear each other
- B, C hear each other
- A, C can not hear each other means A, C unaware of their interference at B

Attenuation also causes "hidden terminals"



- B, A hear each other
- B, C hear each other
- A, C can not hear each other interfering at B

Chapter 7 outline

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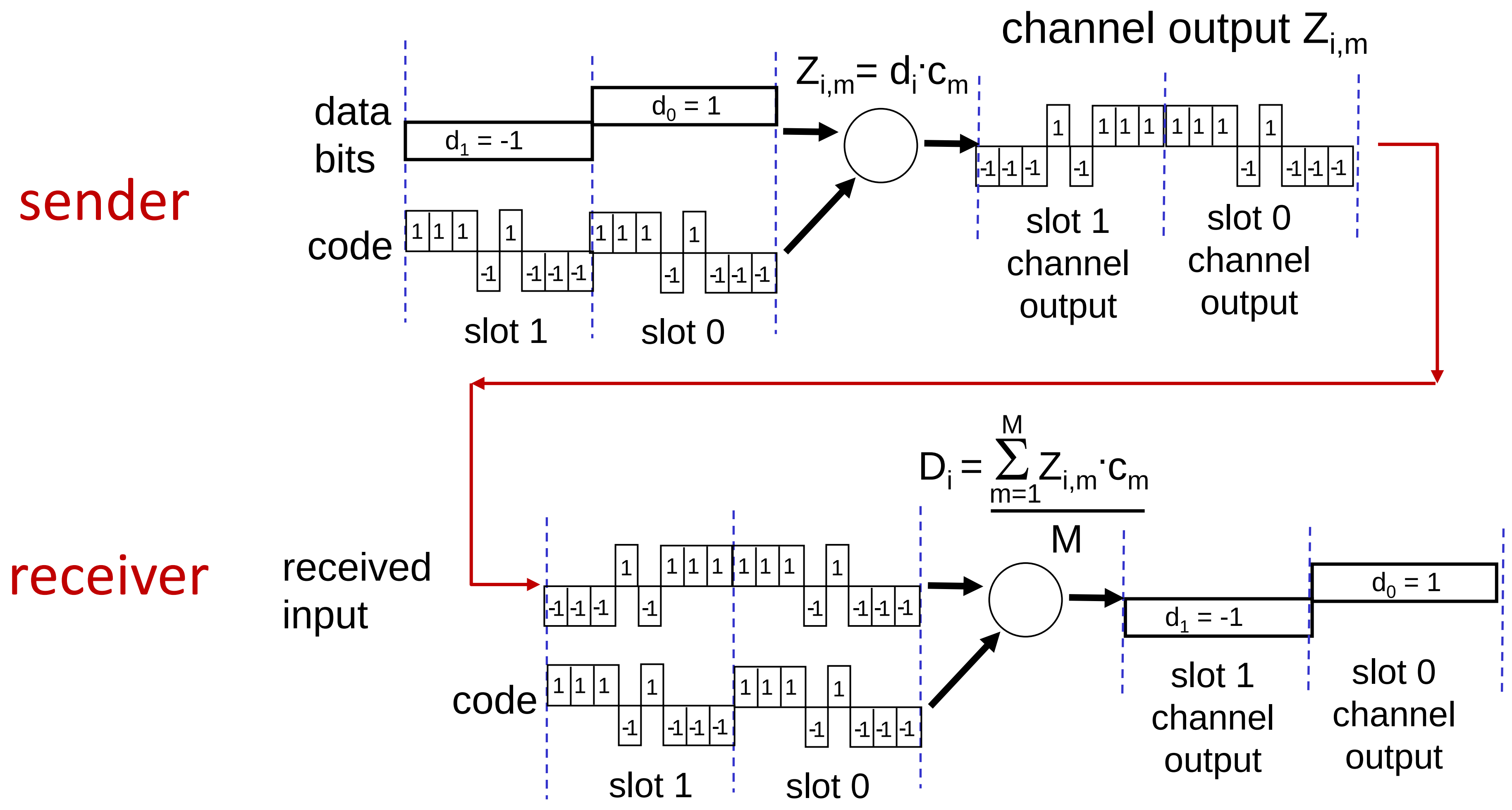
Wireless

- Wireless links and network characteristics
- CDMA: code division multiple access
- WiFi: 802.11 wireless LANs
- Bluetooth

Code Division Multiple Access (CDMA)

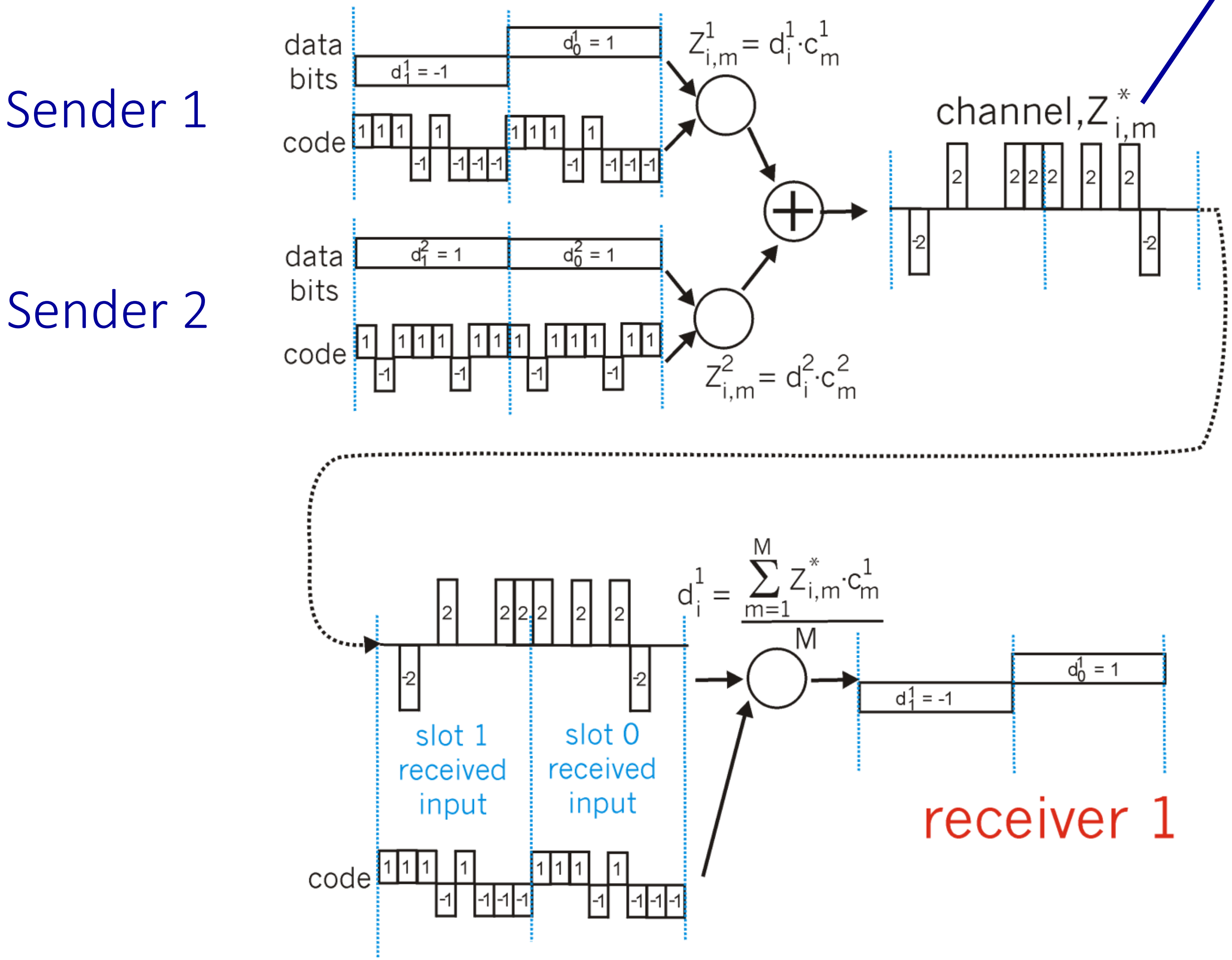
- unique "code" assigned to each user; i.e., code set partitioning
  - all users share same frequency, but each user has own "chipping" sequence (i.e., code) to encode data
  - allows multiple users to "coexist" and transmit simultaneously with minimal interference (if codes are "orthogonal")
- encoding: inner product: (original data) X (chipping sequence)
- decoding: summed inner-product: (encoded data) X (chipping sequence)

CDMA encode/decode



... but this isn't really useful yet!

CDMA: two-sender interference



channel sums together transmissions by sender 1 and 2

using same code as sender 1, receiver recovers sender 1's original data from summed channel data!

… now *that's* useful!

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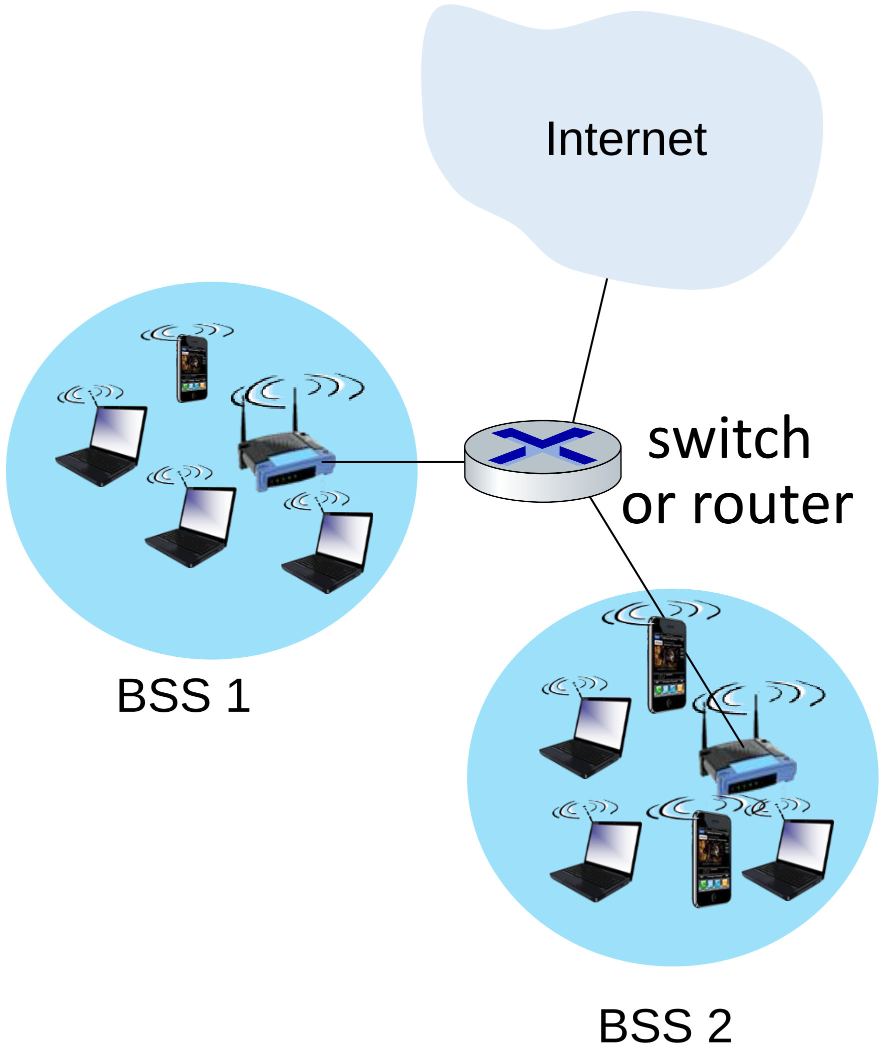
IEEE 802.11 Wireless LAN

IEEE 802.11 standard	Year	Max data rate	Range	Frequency
802.11b	1999	11 Mbps	30 m	2.4 Ghz
802.11g	2003	54 Mbps	30m	2.4 Ghz
802.11n (WiFi 4)	2009	600	70m	2.4, 5 Ghz
802.11ac (WiFi 5)	2013	3.47Gpbs	70m	5 Ghz
802.11ax (WiFi 6)	2020 (exp.)	14 Gbps	70m	2.4, 5 Ghz
802.11af	2014	35 – 560 Mbps	1 Km	unused TV bands (54-790 MHz)
802.11ah	2017	347Mbps	1 Km	900 Mhz

- all use CSMA/CA for multiple access, and have base-station and ad-hoc network versions

802.11 LAN architecture

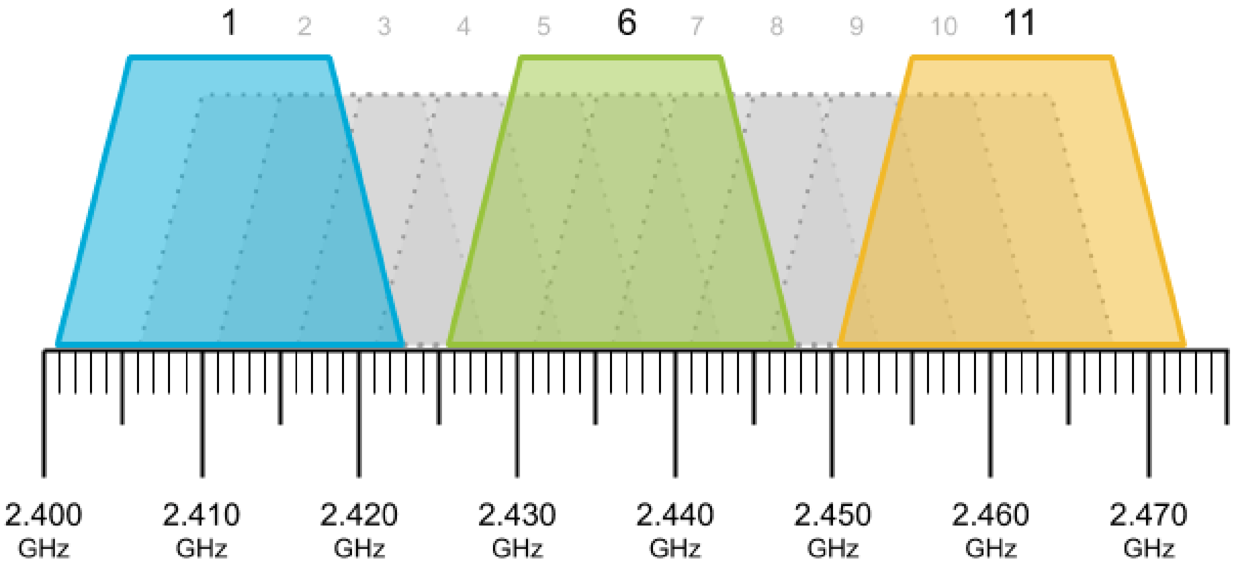


- wireless host communicates with base station
  - base station = access point (AP)
- Basic Service Set (BSS) (aka “cell”) in infrastructure mode contains:
  - wireless hosts
  - access point (AP): base station
  - ad hoc mode: hosts only

802.11: Channels

- spectrum divided into channels at different frequencies
  - AP admin chooses frequency for AP
  - interference possible: channel can be same as that chosen by neighboring AP!

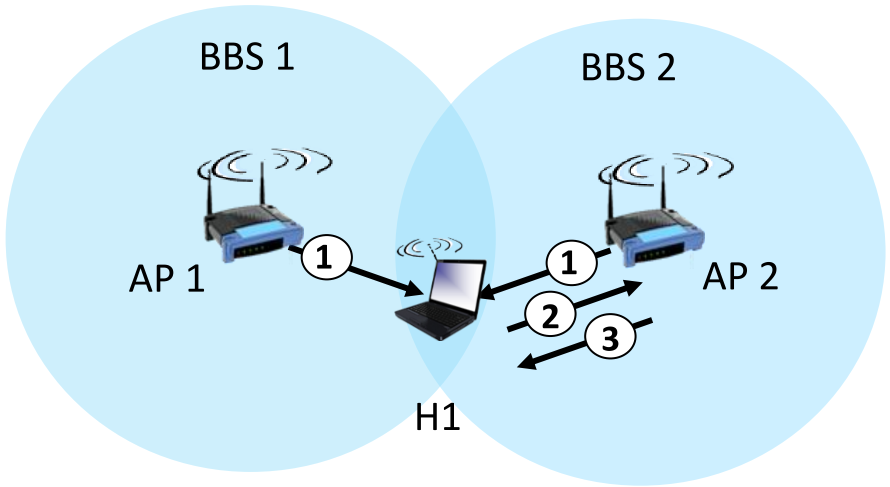
Example: 2.4 GHz

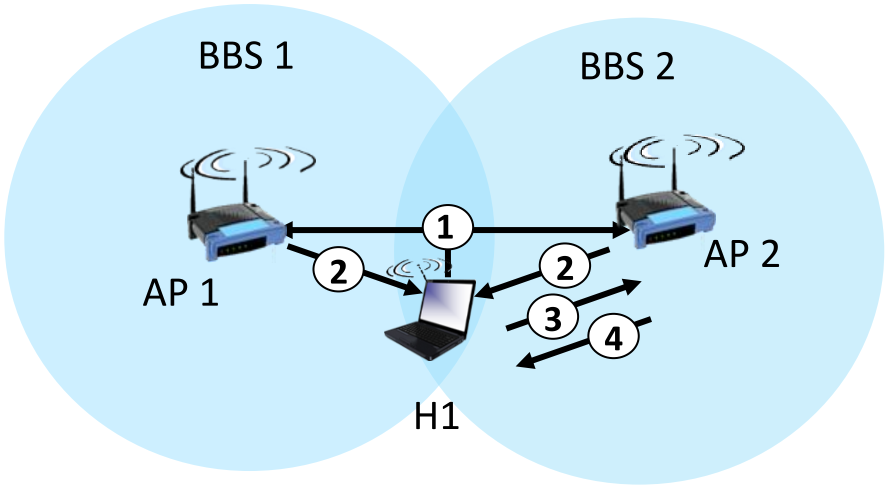


802.11: Association

- arriving host: must associate with an AP
  - scans channels, listening for *beacon frames* containing AP's name (SSID) and MAC address
  - selects AP to associate with
  - then may perform authentication [Chapter 8]
  - then typically run DHCP to get IP address in AP's subnet

802.11: passive/active scanning





passive scanning:

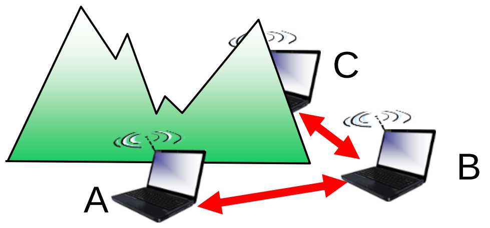
(1) beacon frames sent from APs
(2) association Request frame sent: H1 to selected AP
(3) association Response frame sent from selected AP to H1

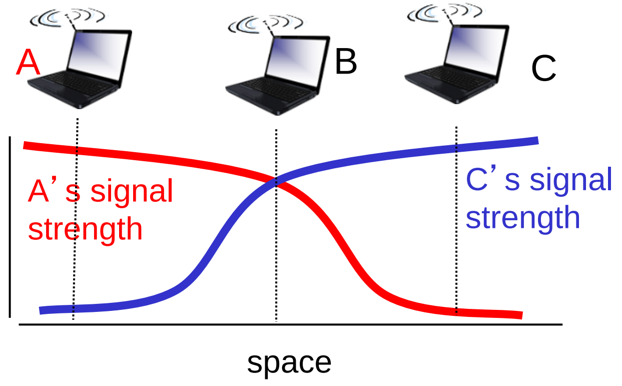
active scanning:

(1) Probe Request frame broadcast from H1
(2) Probe Response frames sent from APs
(3) Association Request frame sent: H1 to selected AP
(4) Association Response frame sent from selected AP to H1

IEEE 802.11: multiple access

- avoid collisions: 2^+ nodes transmitting at same time
- 802.11: CSMA - sense before transmitting
  - don't collide with detected ongoing transmission by another node
- 802.11: *no* collision detection!
  - difficult to sense collisions: high transmitting signal, weak received signal due to fading
  - can't sense all collisions in any case: hidden terminal, fading
  - goal: *avoid collisions:* CSMA/<u>C</u>ollision<u>A</u>voidance



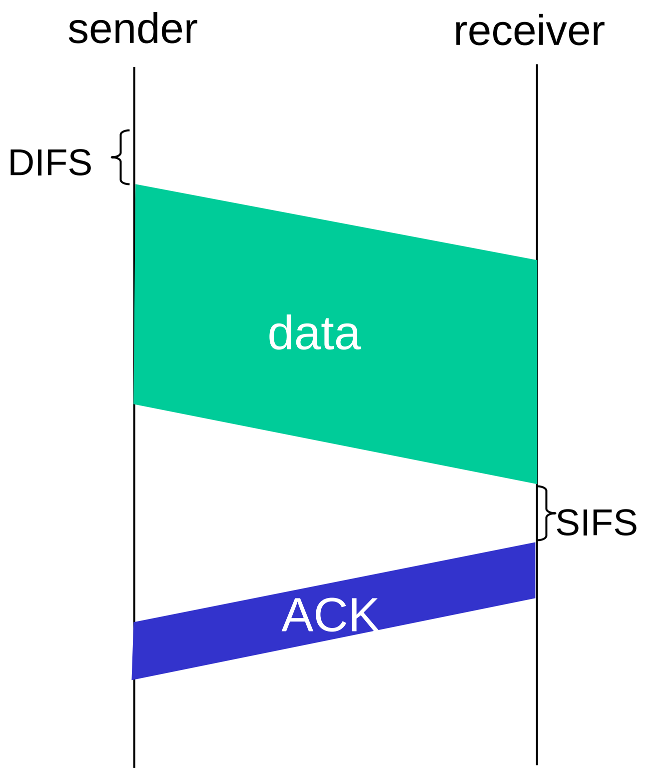


IEEE 802.11 MAC Protocol: CSMA/CA

802.11 sender

1 if sense channel idle for **DIFS** then
    transmit entire frame (no CD)

2 if sense channel busy then
    start random backoff time
    timer counts down while channel idle
    transmit when timer expires
    if no ACK, increase random backoff interval, repeat 2



802.11 receiver

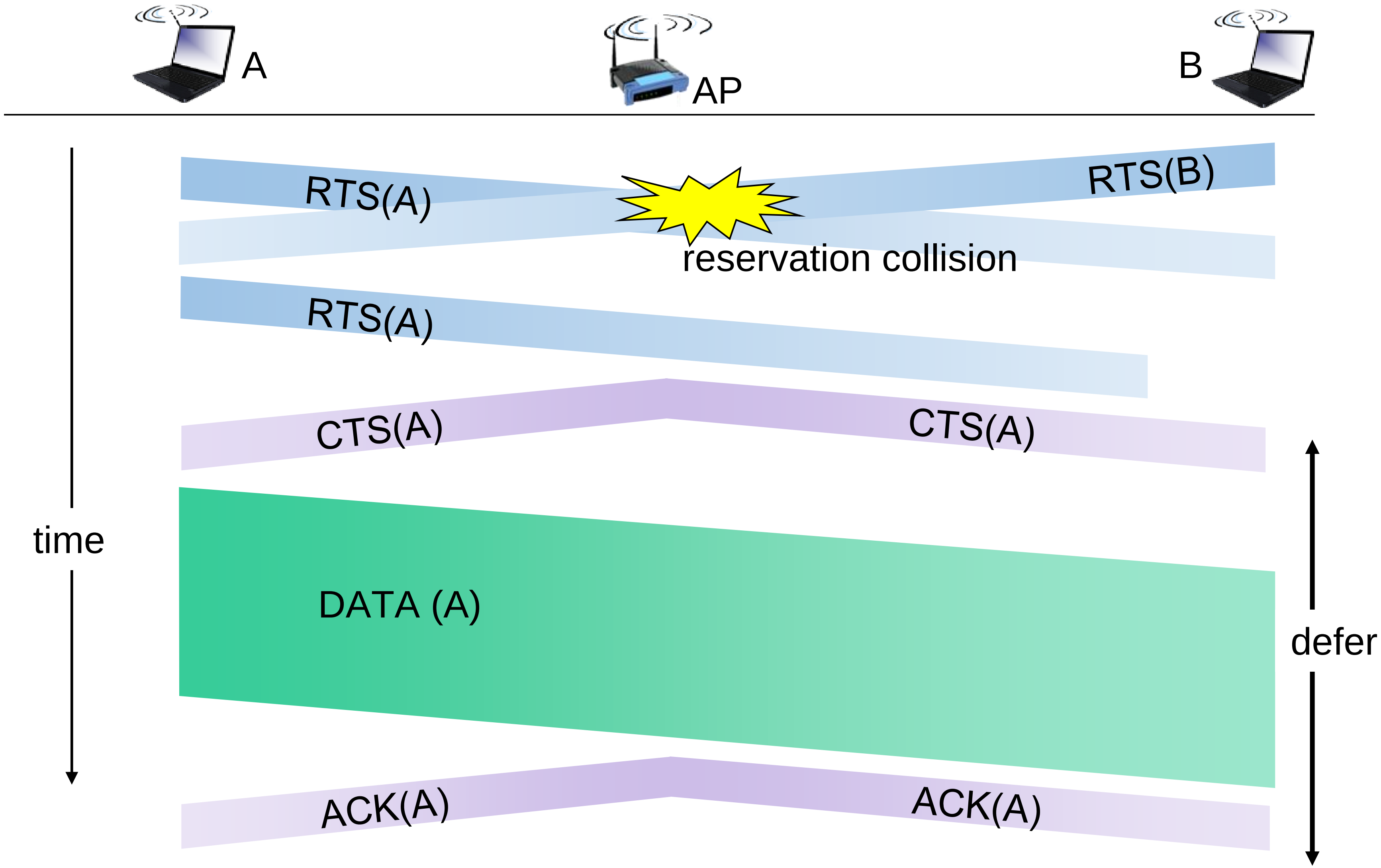
if frame received OK
  return ACK after **SIFS** (ACK needed due to hidden terminal problem)

Avoiding collisions (more)

idea: sender "reserves" channel use for data frames using small reservation packets

- sender first transmits *small* request-to-send (RTS) packet to BS using CSMA
  - RTSs may still collide with each other (but they're short)
- BS broadcasts clear-to-send CTS in response to RTS
- CTS heard by all nodes
  - sender transmits data frame
  - other stations defer transmissions

Collision Avoidance: RTS-CTS exchange



802.11 frame: addressing

2	2	6	6	6	2	6	0 - 2312	4
frame control	duration	address 1	address 2	address 3	seq control	address 4	payload	CRC

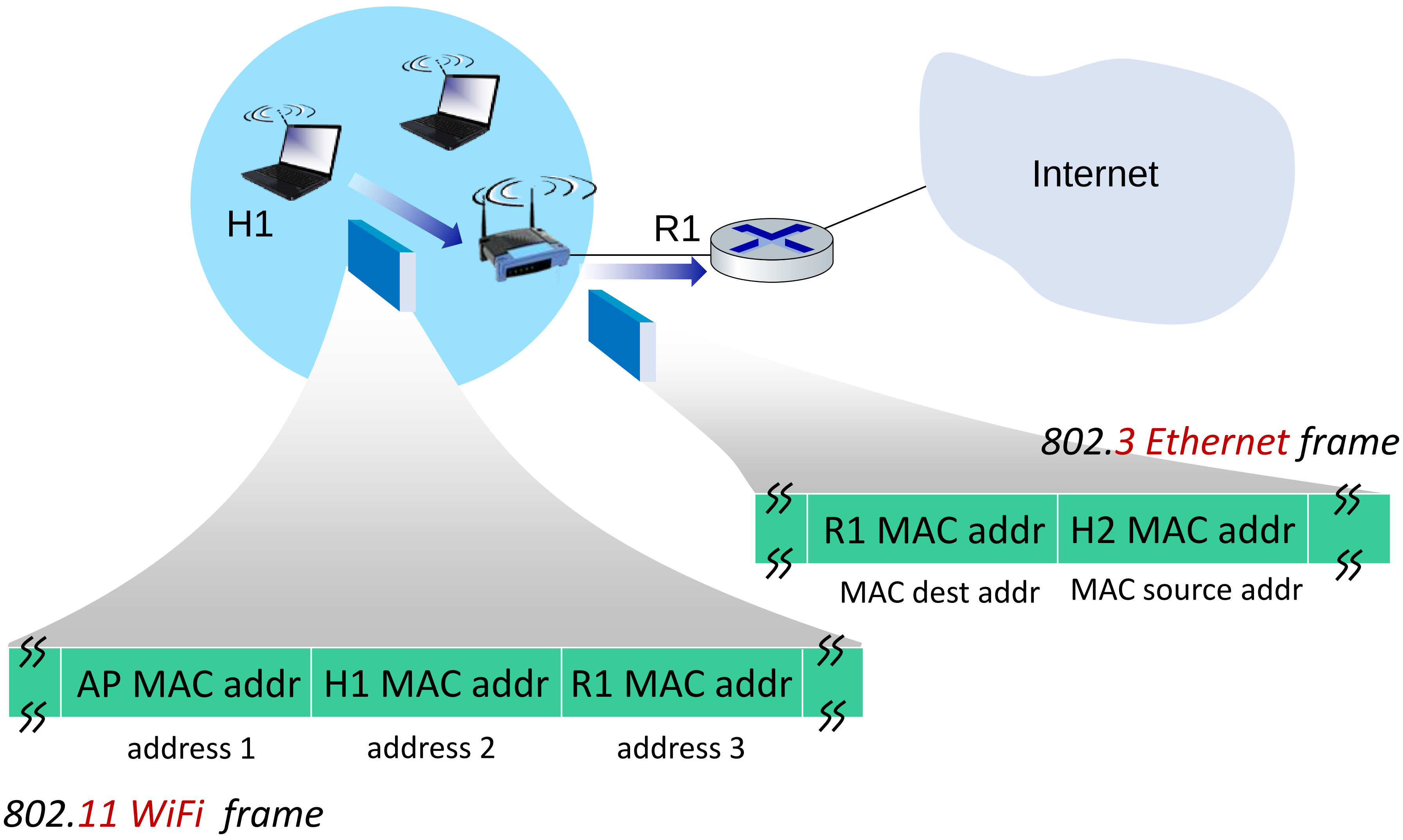
Address 1: MAC address of wireless host or AP to receive this frame

Address 2: MAC address of wireless host or AP transmitting this frame

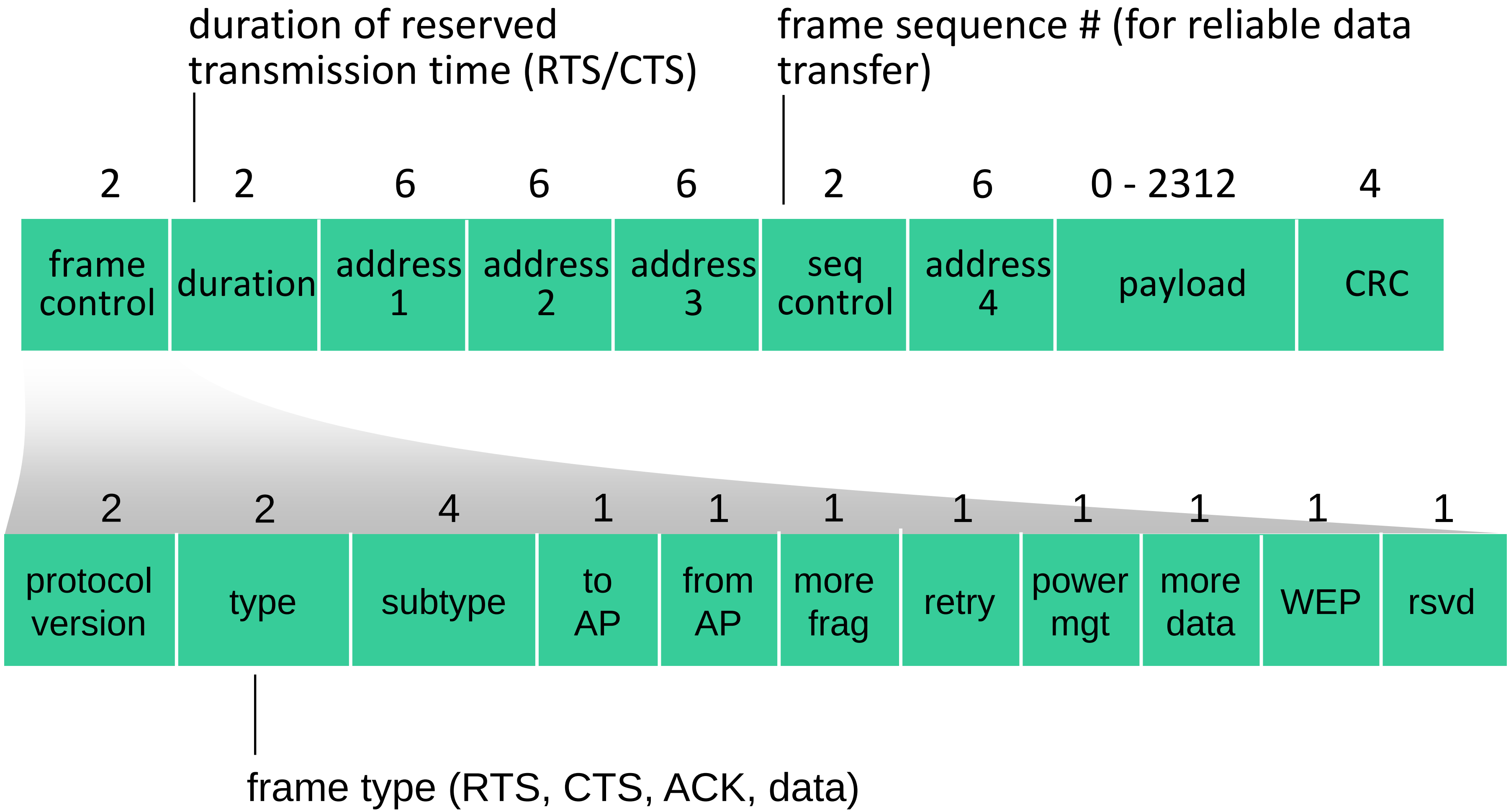
Address 3: MAC address of router interface to which AP is attached

Address 4: used only in ad hoc mode

802.11 frame: addressing

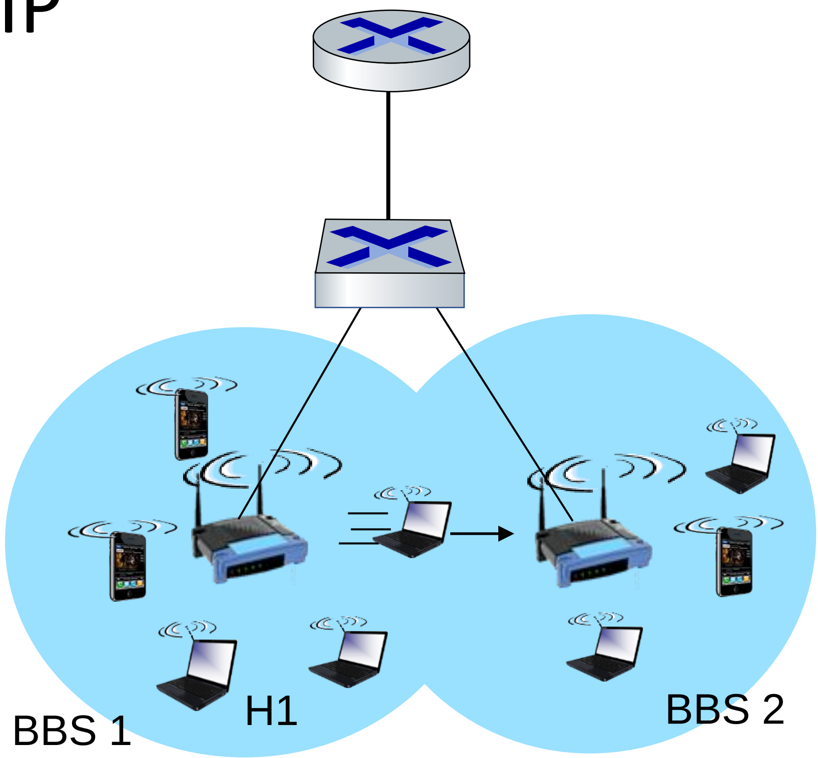


802.11 frame: addressing



802.11: mobility within same subnet

- H1 remains in same IP subnet: IP address can remain same
- switch: which AP is associated with H1?
  - self-learning (Ch. 6): switch will see frame from H1 and "remember" which switch port can be used to reach H1



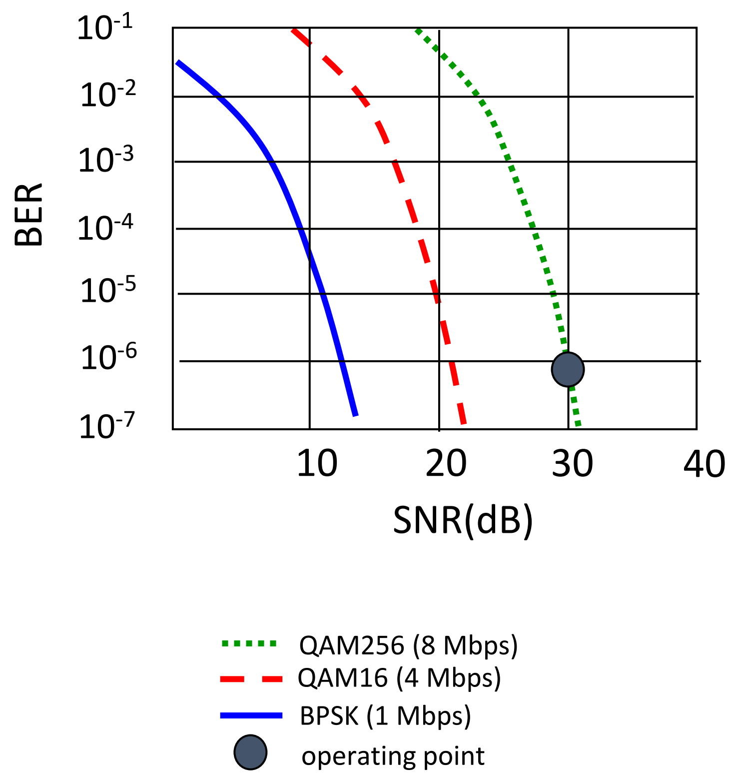
802.11: advanced capabilities

Rate adaptation

- base station, mobile dynamically change transmission rate (physical layer modulation technique) as mobile moves, SNR varies

1. SNR decreases, BER increase as node moves away from base station

2. When BER becomes too high, switch to lower transmission rate but with lower BER



802.11: advanced capabilities

power management

- node-to-AP: "I am going to sleep until next beacon frame"
  - AP knows not to transmit frames to this node
  - node wakes up before next beacon frame
- beacon frame: contains list of mobiles with AP-to-mobile frames waiting to be sent
  - node will stay awake if AP-to-mobile frames to be sent; otherwise sleep again until next beacon frame

Chapter 7 outline

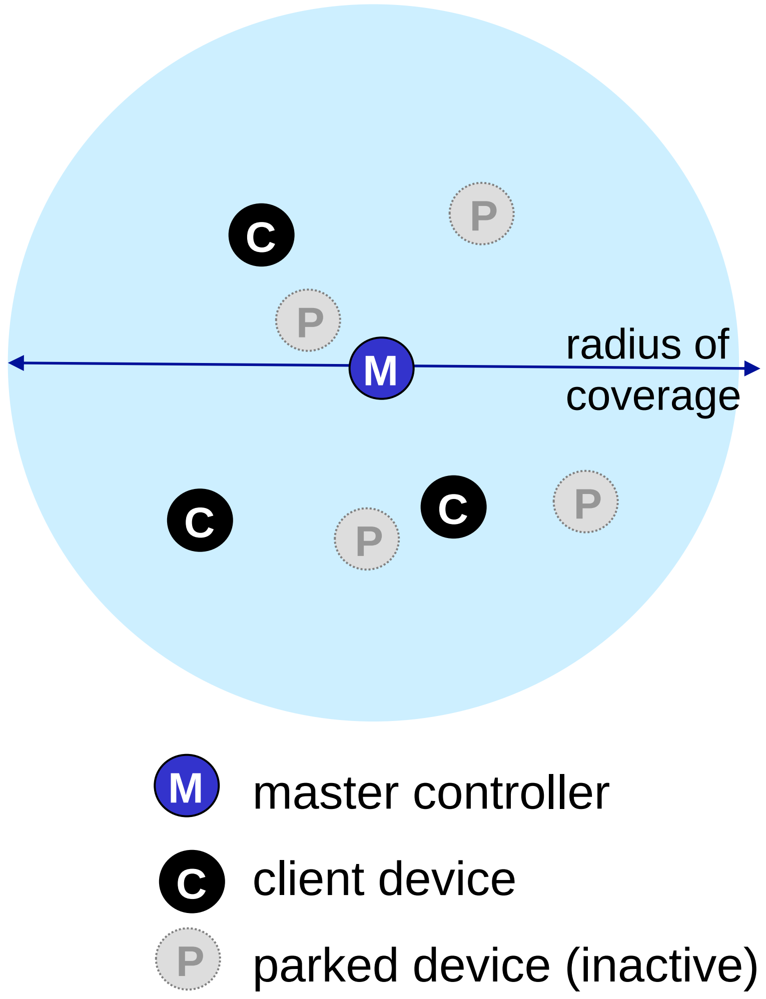
- Introduction

Wireless

- Wireless links and network characteristics
- CDMA: code division multiple access
- WiFi: 802.11 wireless LANs
- Bluetooth

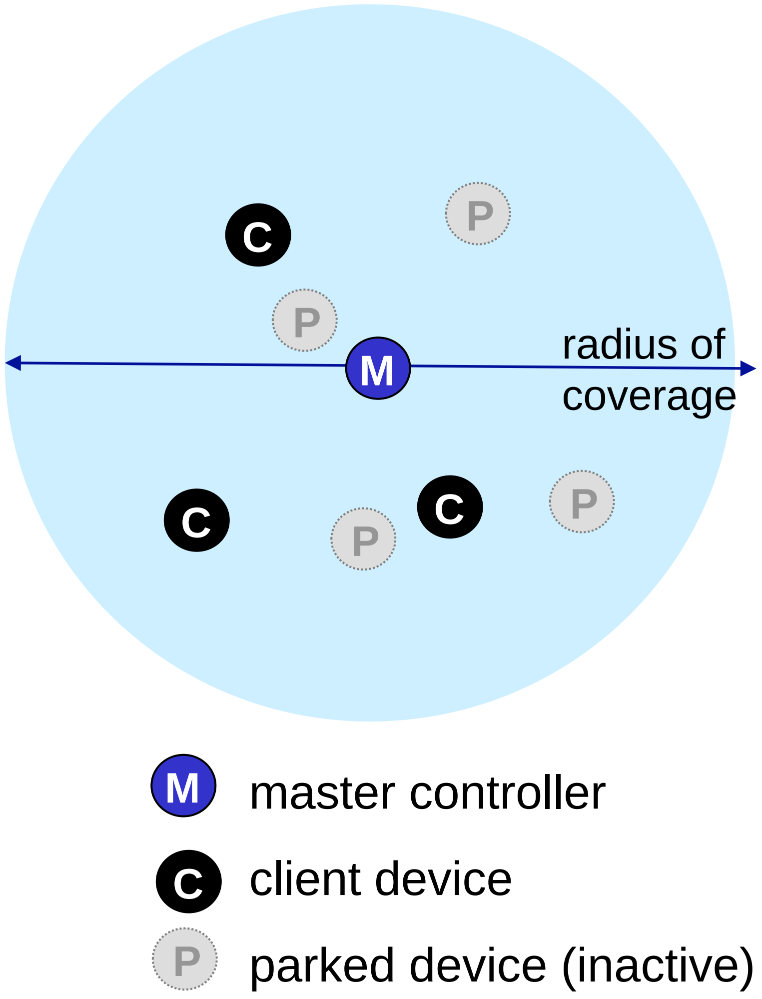
Personal area networks: Bluetooth

- less than 10 m diameter
- replacement for cables (mouse, keyboard, headphones)
- ad hoc: no infrastructure
- 2.4-2.5 GHz ISM radio band, up to 3 Mbps
- master controller / client devices:
  - master polls clients, grants requests for client transmissions

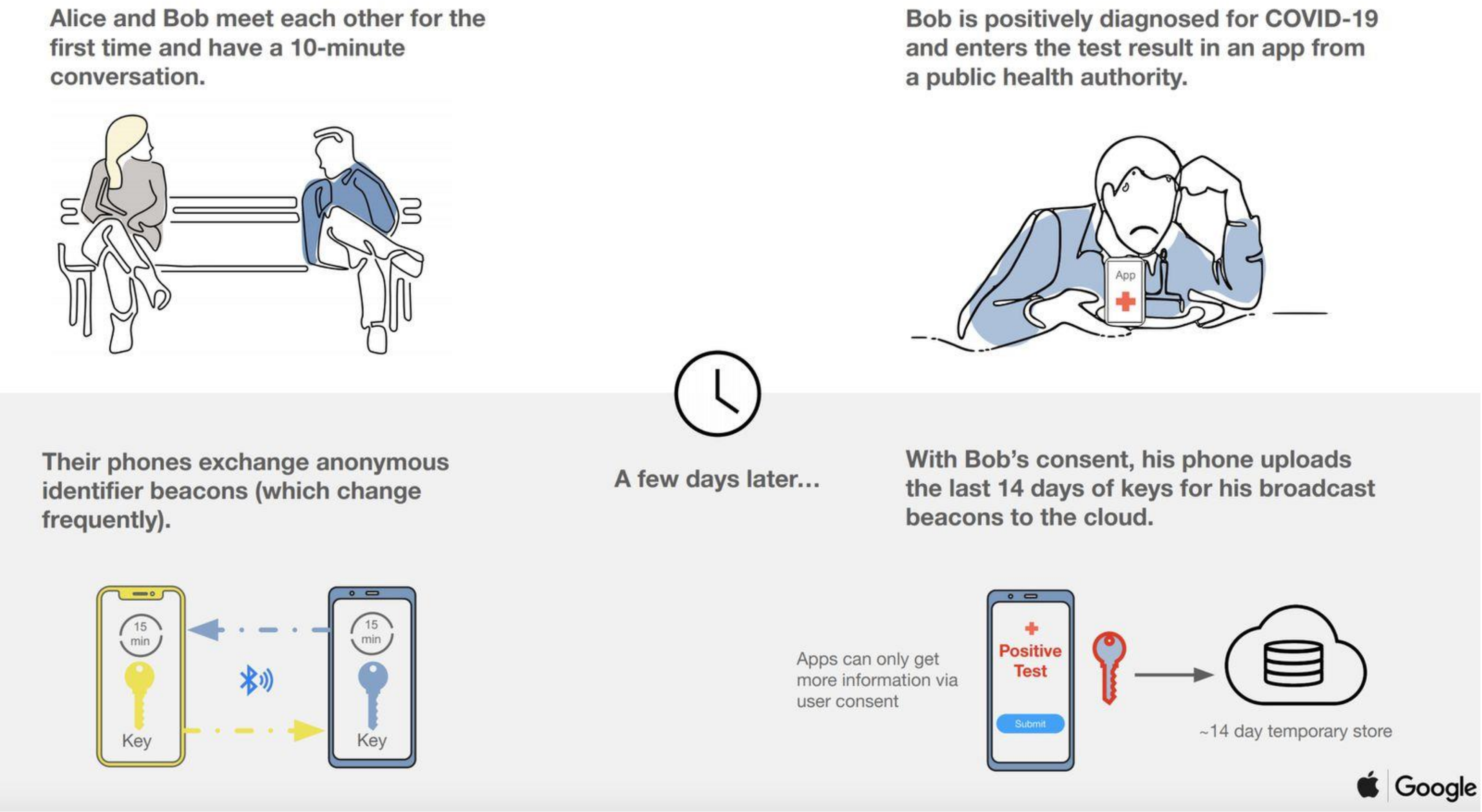


Personal area networks: Bluetooth

- TDM, 625 μsec sec. slot
- FDM: sender uses 79 frequency channels in known, pseudo-random order slot-to-slot (spread spectrum)
  - other devices/equipment not in piconet only interfere in some slots
- parked mode: clients can "go to sleep" (park) and later wakeup (to preserve battery)
- bootstrapping: nodes self-assemble (plug and play) into piconet



Pandemic + Bluetooth



Source: https://www.macrumors.com/guide/exposure-notification/

Chapter 7 outline

- Introduction

Wireless

- Wireless links and network characteristics
- WiFi: 802.11 wireless LANs
- Cellular networks: 4G and 5G

Mobility

- Mobility management: principles
- Mobility management: practice
  - 4G/5G networks
  - Mobile IP
- Mobility: impact on higher-layer protocols

4G/5G cellular networks

- *the* solution for wide-area mobile Internet
- widespread deployment/use:
  - more mobile-broadband-connected devices than fixed-broadband-connected devices devices (5-1 in 2019)!
  - 4G availability: 97% of time in Korea (90% in US)
- transmission rates up to 100’s Mbps
- technical standards: 3rd Generation Partnership Project (3GPP)
  - wwww.3gpp.org
  - 4G: Long-Term Evolution (LTE)standard

4G/5G cellular networks

similarities to wired Internet

- edge/core distinction, but both belong to same carrier
- global cellular network: a network of networks
- widespread use of protocols we've studied: HTTP, DNS, TCP, UDP, IP, NAT, separation of data/control planes, SDN, Ethernet, tunneling
- interconnected to wired Internet

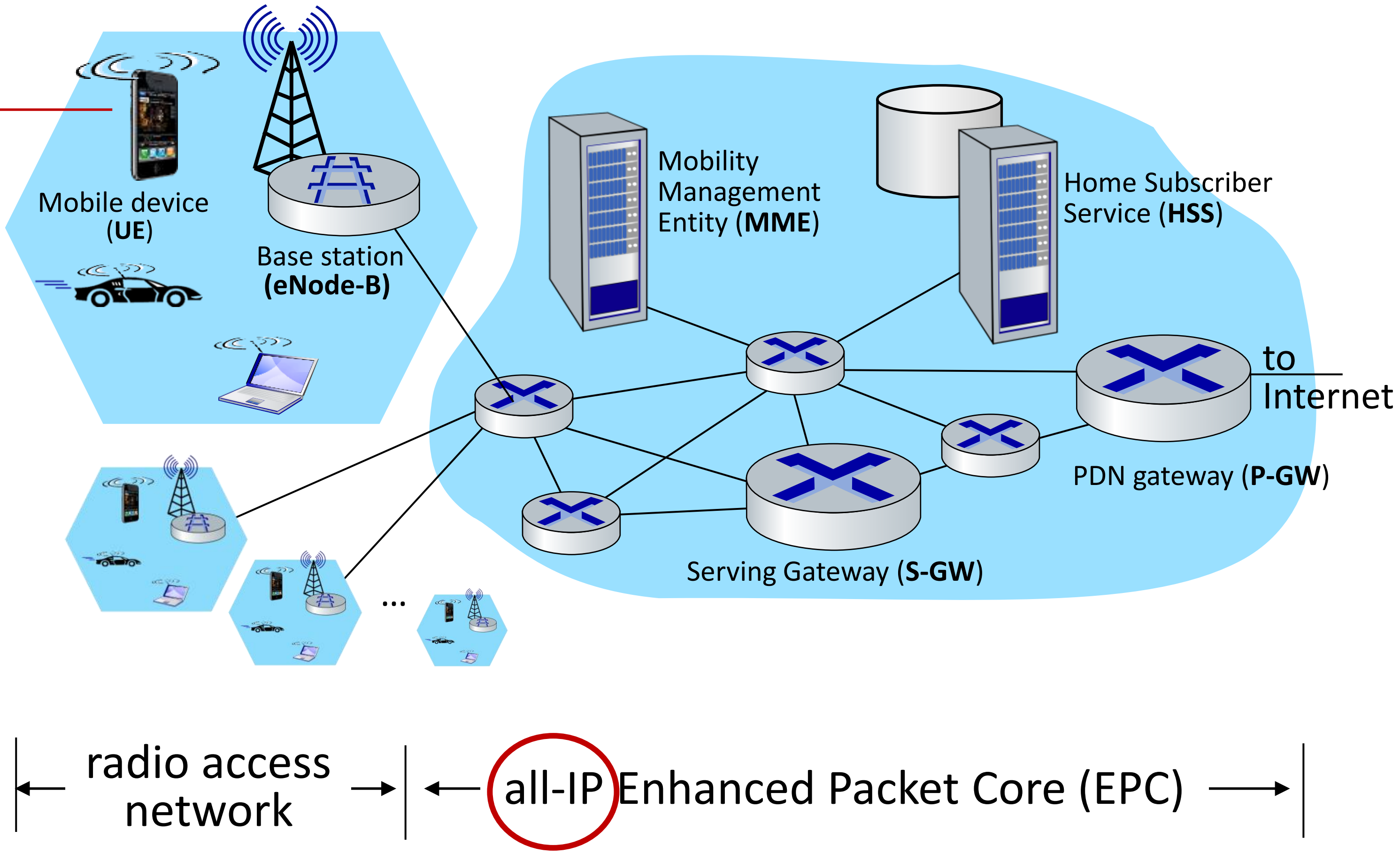
differences from wired Internet

- different wireless link layer
- mobility as a 1st class service
- user "identity" (via SIM card)
- business model: users subscribe to a cellular provider
  - strong notion of "home network" versus roaming on visited nets
  - global access, with authentication infrastructure, and inter-carrier settlements

Elements of 4G LTE architecture

Mobile device:

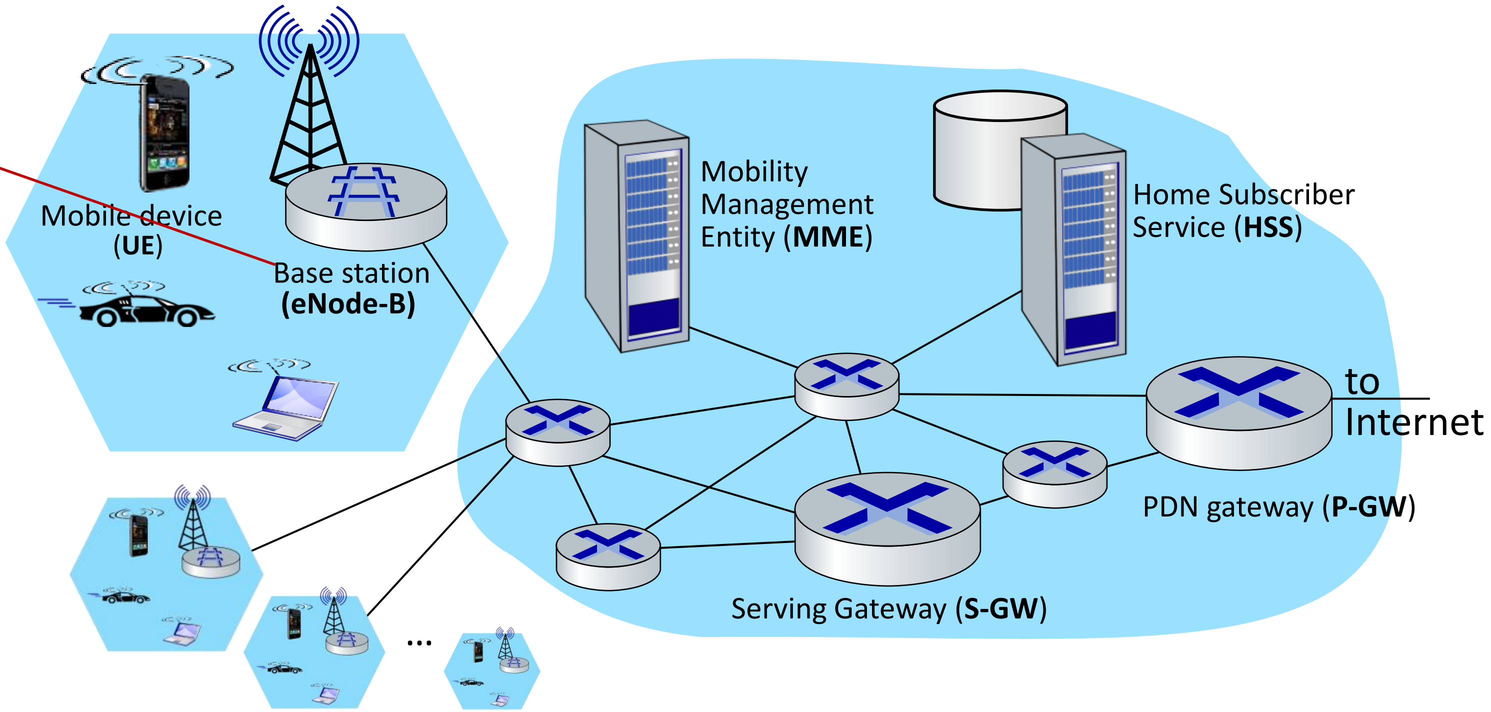
- smartphone, tablet, laptop, IoT, ... with 4G LTE radio
- 64-bit International Mobile Subscriber Identity (IMSI), stored on SIM (Subscriber Identity Module) card
- LTE jargon: User Equipment (UE)



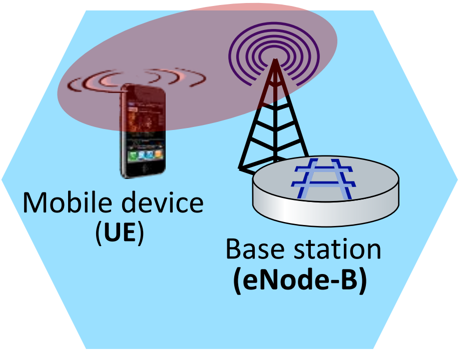
Elements of 4G LTE architecture

Base station:

- at "edge" of carrier's network
- manages wireless radio resources, mobile devices in its coverage area ("cell")
- coordinates device authentication with other elements
- similar to WiFi AP but:
  - active role in user mobility
  - coordinates with nearly base stations to optimize radio use
- LTE jargon: eNode-B

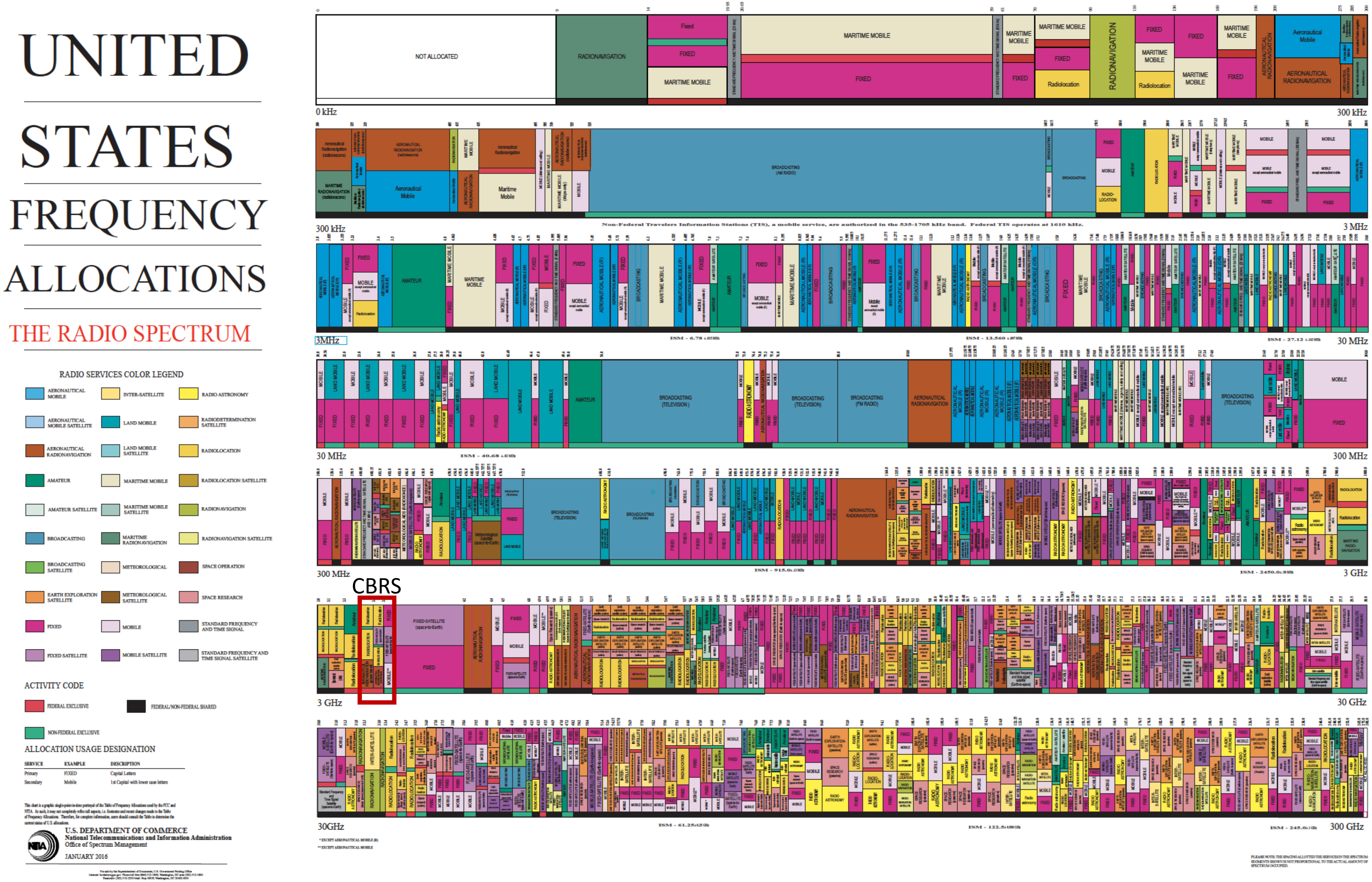


Radio Access Network: 4G radio



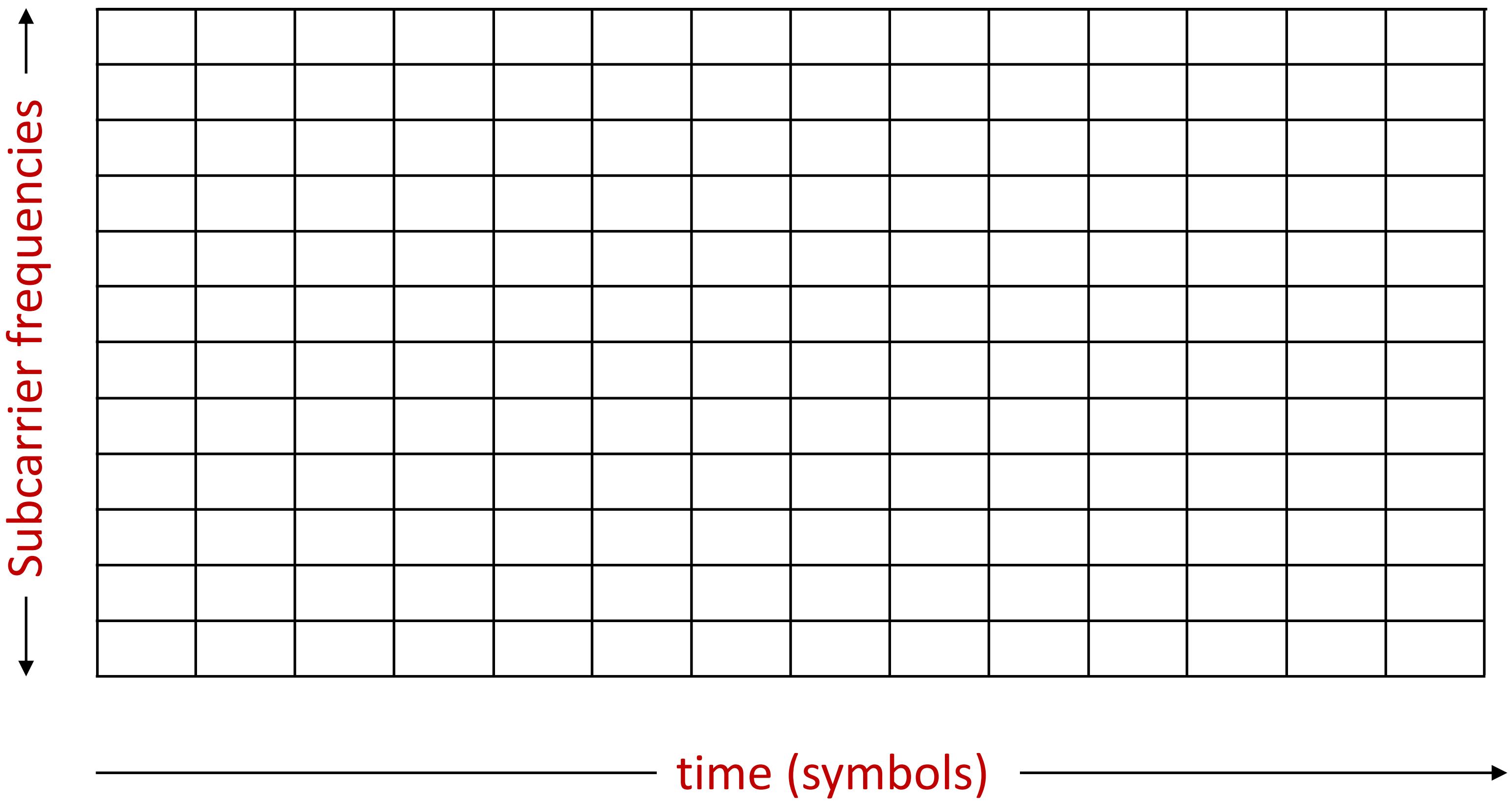
- connects device (UE) to a base station (eNode-B)
  - multiple devices connected to each base station
- many different possible frequencies bands, multiple channels in each band
  - popular bands: 600, 700, 850, 1500, 1700, 1900, 2100, 2600, 3500 MHz
  - separate upstream and downstream channels
- sharing 4G radio channel among users:
  - OFDM: Orthogonal Frequency Division Multiplexing
  - combination of FDM, TDM
- 100's Mbps possible per user/device

Spectrum



UNITED
STATES
FREQUENCY
ALLOCATIONS
THE RADIO SPECTRUM
RADIO SERVICES COLOR LEGEND
AERONAUTICAL MOBILE
AERONAUTICAL MOBILE SATELLITE
AERONAUTICAL RADIONAVIGATION
AMATEUR
AMATEUR SATELLITE
BROADCASTING
BROADCASTING SATELLITE
EARTH EXPLORATION SATELLITE
FIXED
FIXED SATELLITE
INTER-SATELLITE
LAND MOBILE
LAND MOBILE SATELLITE
MARITIME MOBILE
MARITIME MOBILE SATELLITE
MARITIME RADIONAVIGATION
METEOROLOGICAL
METEOROLOGICAL SATELLITE
MOBILE
MOBILE SATELLITE
RADIO ASTRONOMY
RADIODETERMINATION SATELLITE
RADIOLOCATION
RADIOLOCATION SATELLITE
RADIONAVIGATION
RADIONAVIGATION SATELLITE
SPACE OPERATION
SPACE RESEARCH
STANDARD FREQUENCY AND TIME SIGNAL
STANDARD FREQUENCY AND TIME SIGNAL SATELLITE
ACTIVITY CODE
FEDERAL EXCLUSIVE
FEDERAL/NON-FEDERAL SHARED
NON-FEDERAL EXCLUSIVE
ALLOCATION USAGE DESIGNATION
SERVICE
EXAMPLE
DESCRIPTION
Primary
FIXED
Capital Letters
Secondary
Mobile
1st Capital with lower case letters
U.S. DEPARTMENT OF COMMERCE
National Telecommunications and Information Administration
Office of Spectrum Management
JANUARY 2016
0 kHz
300 kHz
NOT ALLOCATED
RADIONAVIGATION
MARITIME MOBILE
FIXED
3 MHz
BROADCASTING (AM RADIO)
3MHz
30 MHz
AMATEUR
BROADCASTING (TELEVISION)
BROADCASTING (FM RADIO)
AERONAUTICAL RADIONAVIGATION
300 MHz
3 GHz
CBRS
FIXED-SATELLITE (space-to-Earth)
30 GHz
30GHz
300 GHz

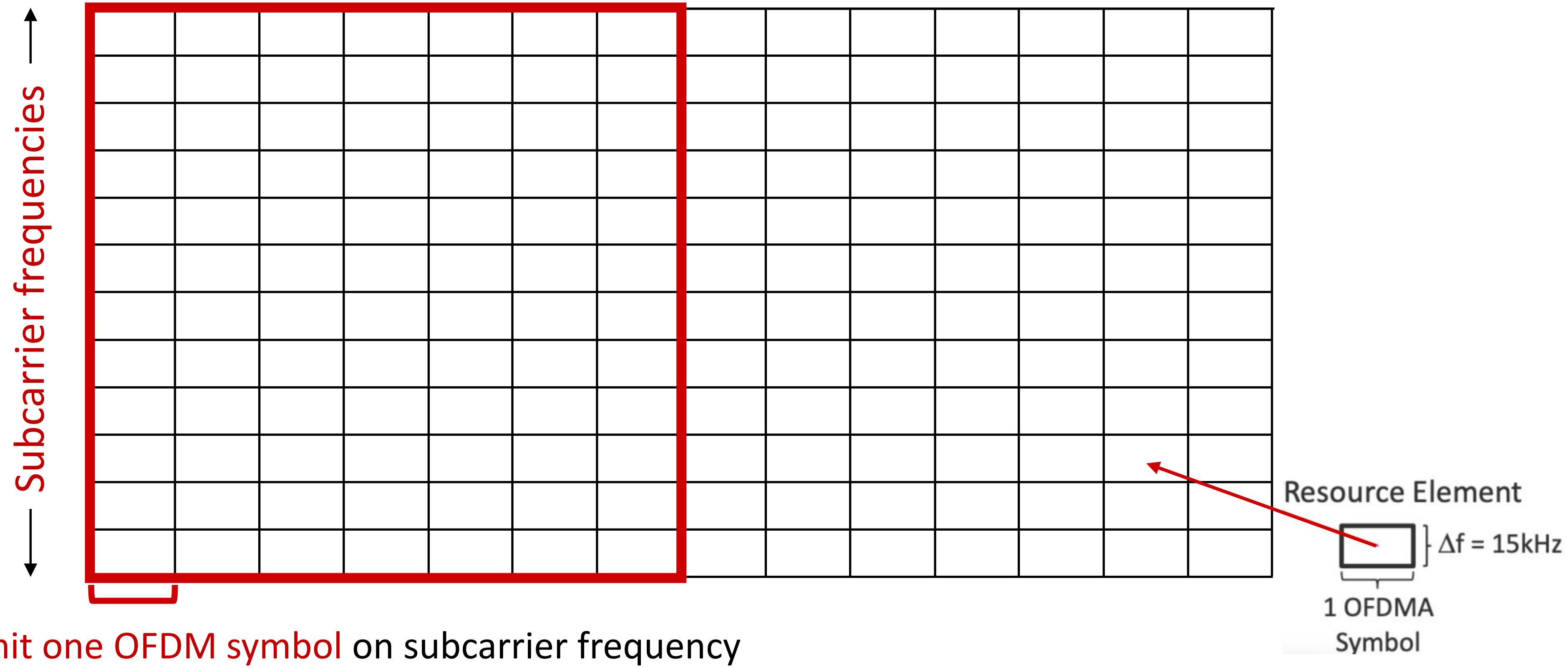
OFDMA: <u>time</u> division (LTE)



OFDMA: time division (LTE)

Physical Resource Block (PRB): blocks of 7x12=84 resource elements

- unit of transmission scheduling

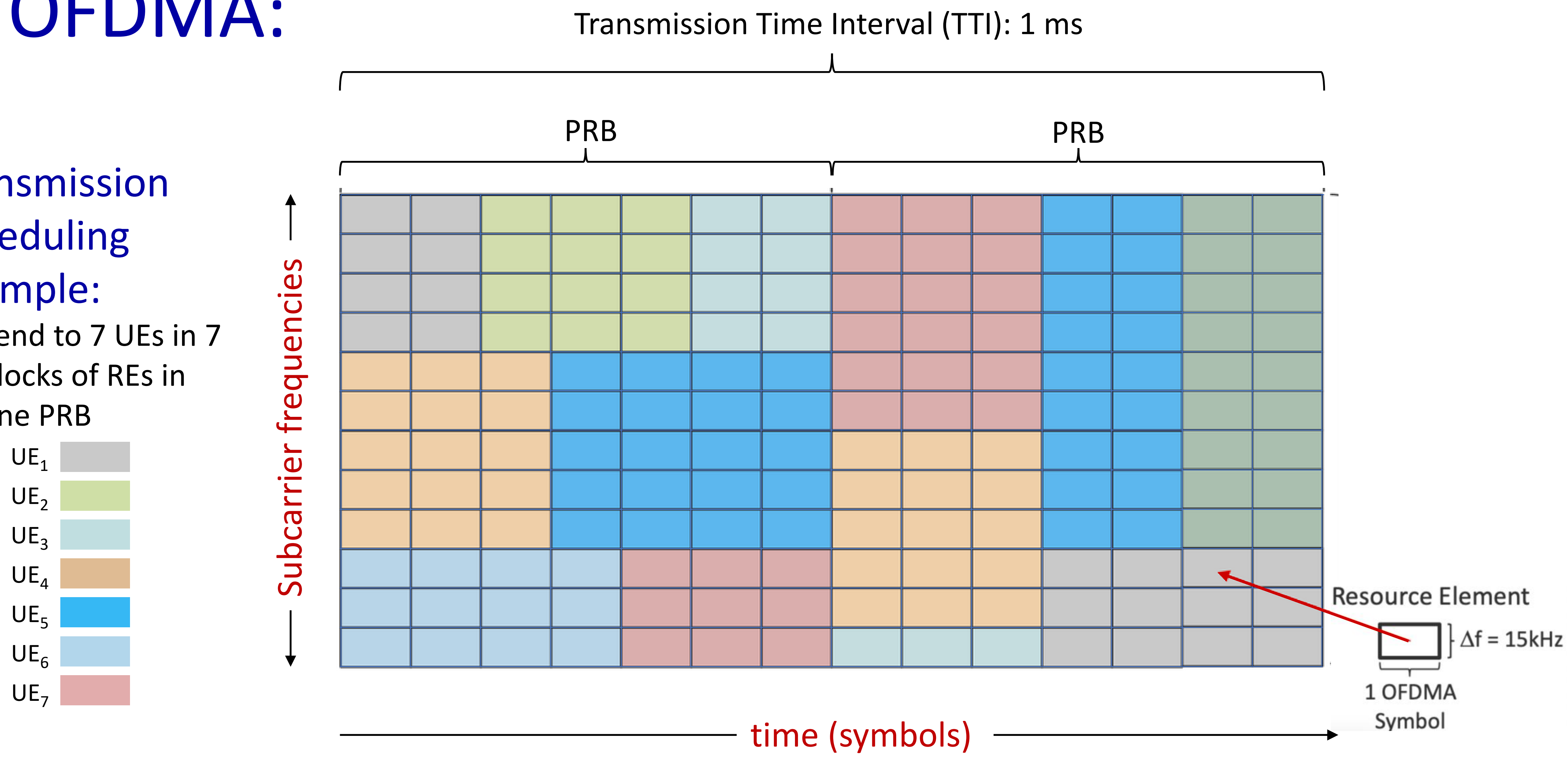


time to transmit one OFDM symbol on subcarrier frequency

OFDMA:

Transmission scheduling example:

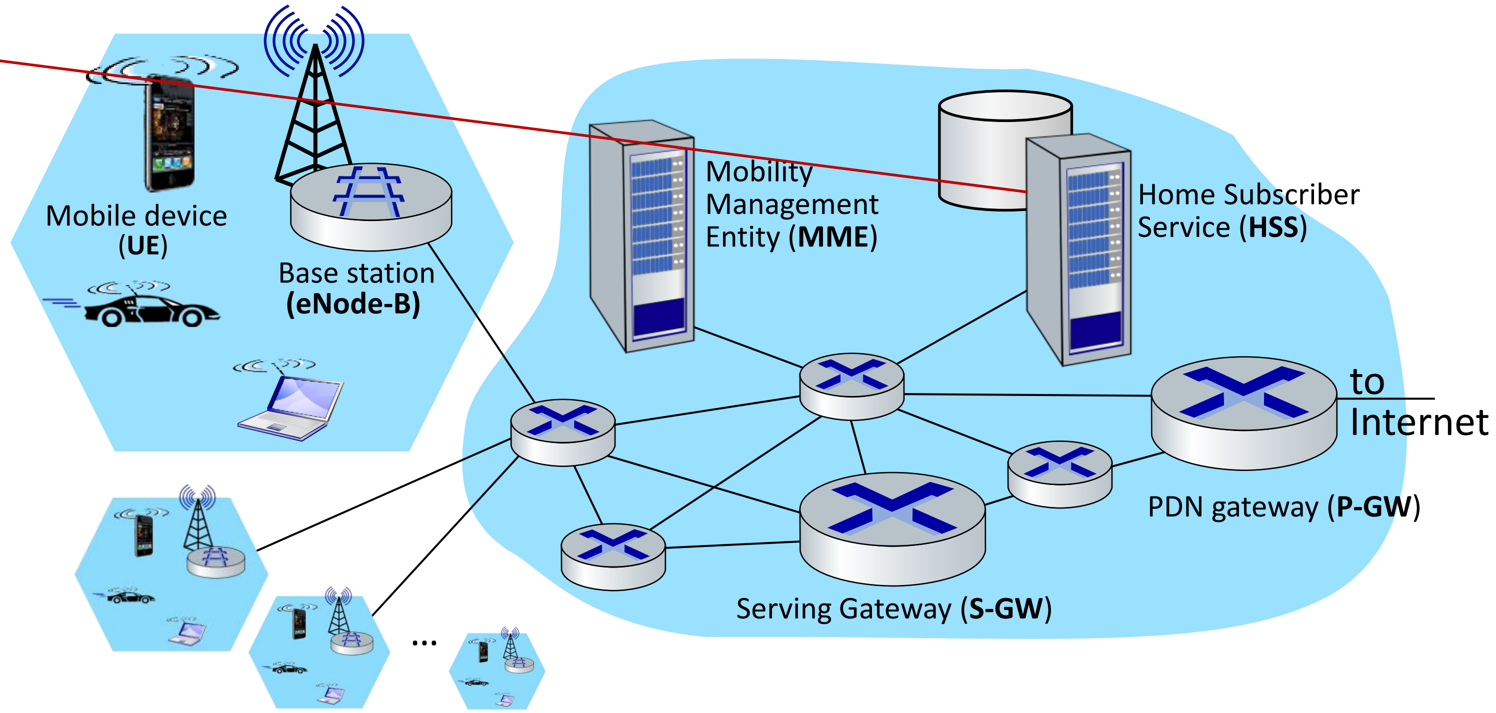
- Send to 7 UEs in 7 blocks of REs in one PRB



Elements of 4G LTE architecture

Home Subscriber Service

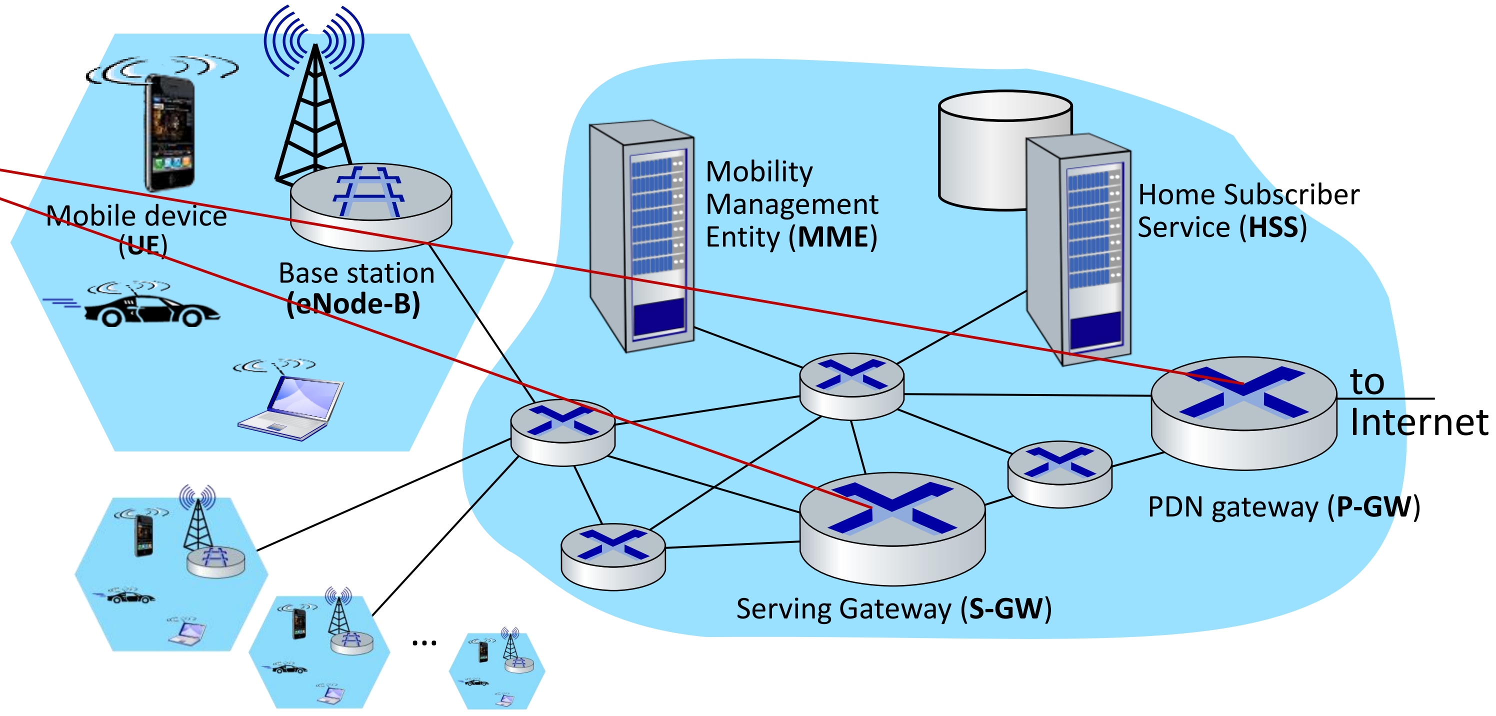
- stores info about mobile devices for which the HSS's network is their "home network"
- works with MME in device authentication



Elements of 4G LTE architecture

Serving Gateway (S-GW), PDN Gateway (P-GW)

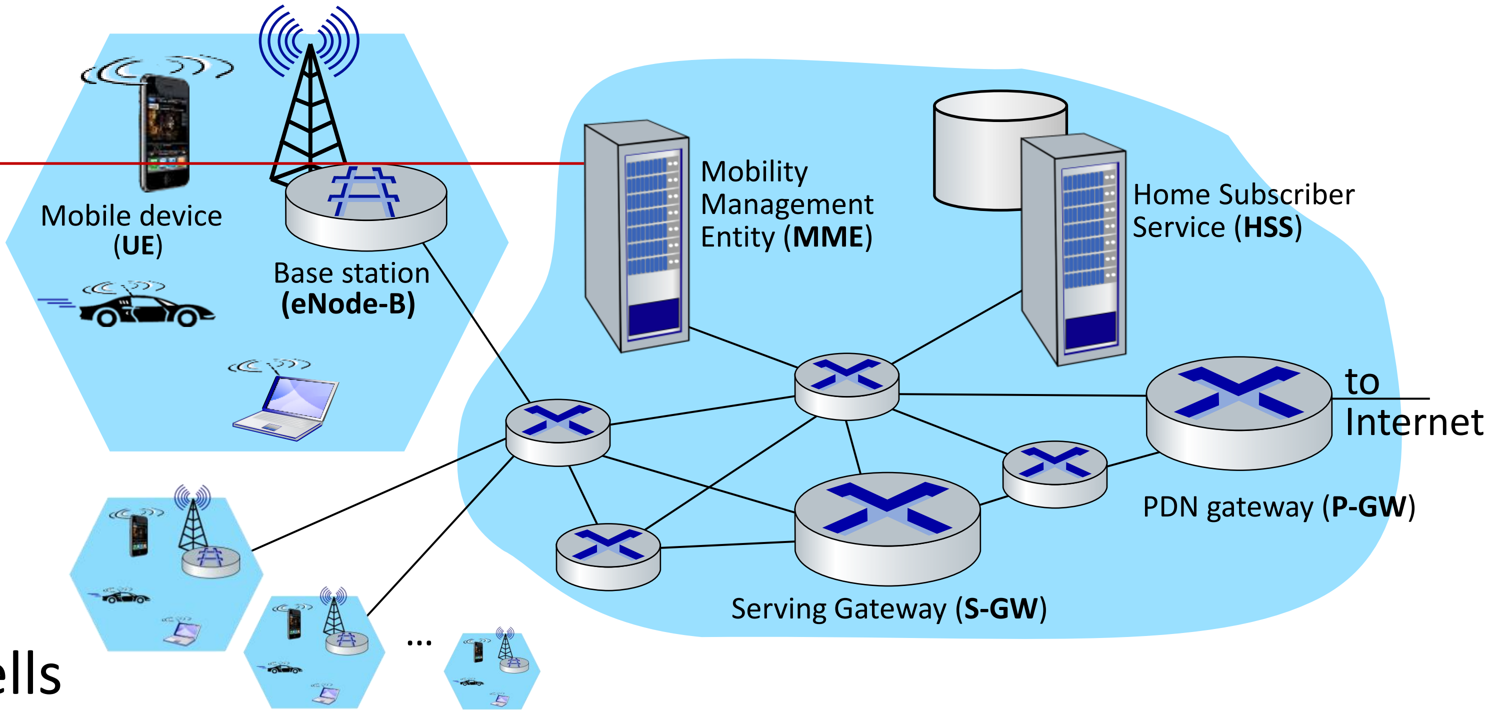
- lie on data path from mobile to/from Internet
- P-GW
  - gateway to mobile cellular network
  - Looks like nay other internet gateway router
  - provides NAT services
- other routers:
  - extensive use of tunneling



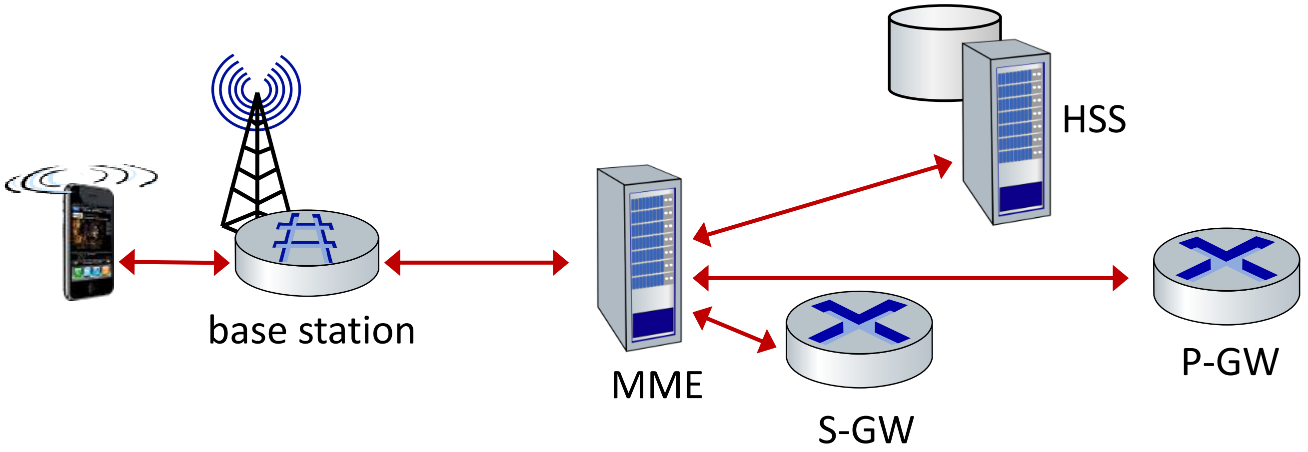
Elements of 4G LTE architecture

Mobility Management Entity

- device authentication (device-to-network, network-to-device) coordinated with mobile home network HSS
- mobile device management:
  - device handover between cells
  - tracking/paging device location
- path (tunneling) setup from mobile device to P-GW

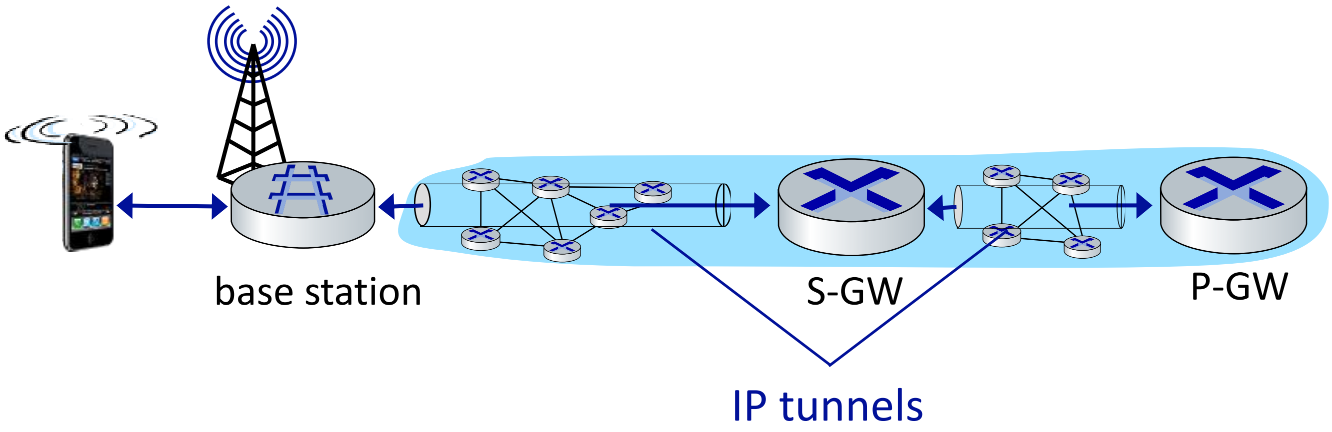


LTE: data plane control plane separation



control plane

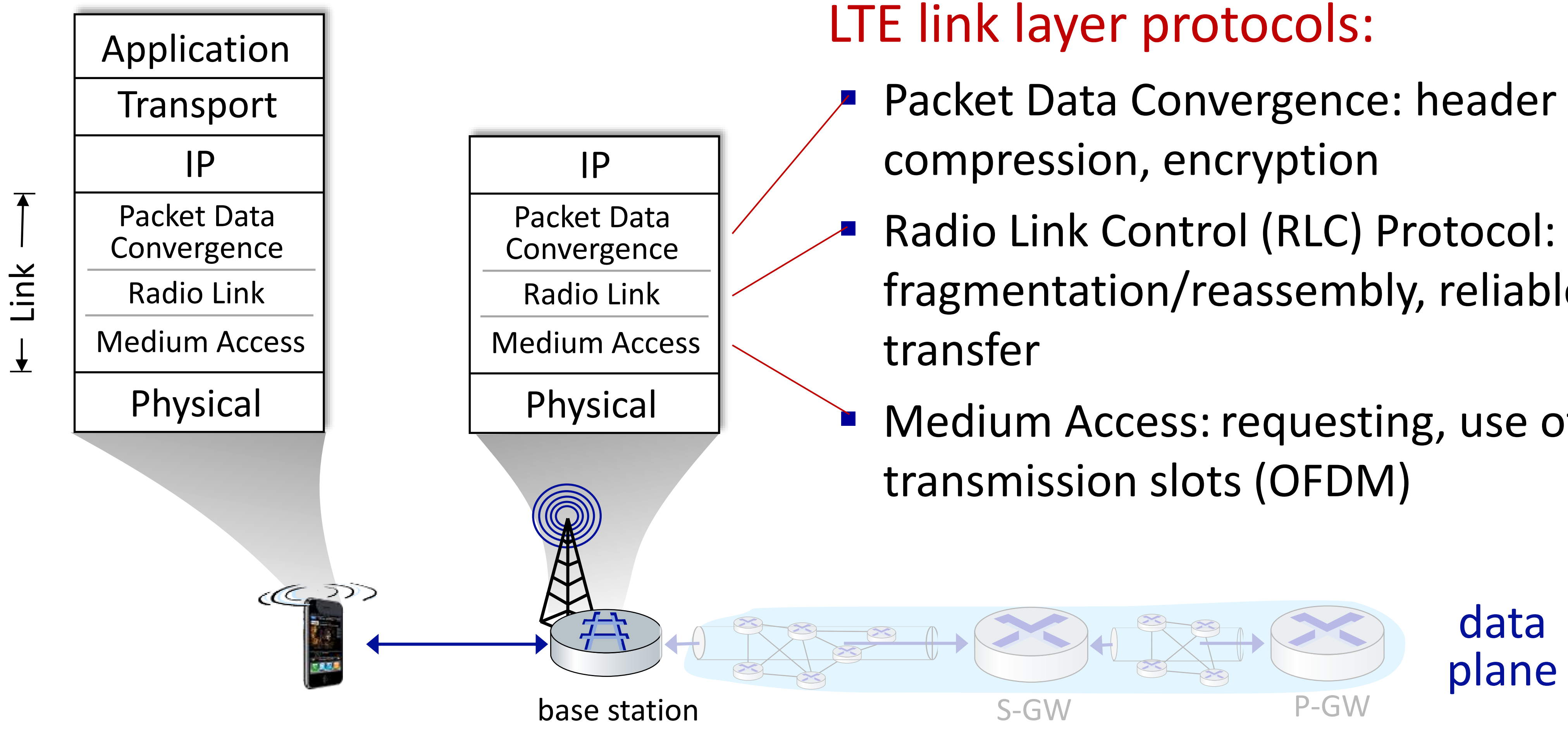
- new protocols for mobility management , security, authentication (later)



data plane

- new protocols at link, physical layers
- extensive use of tunneling to facilitate mobility

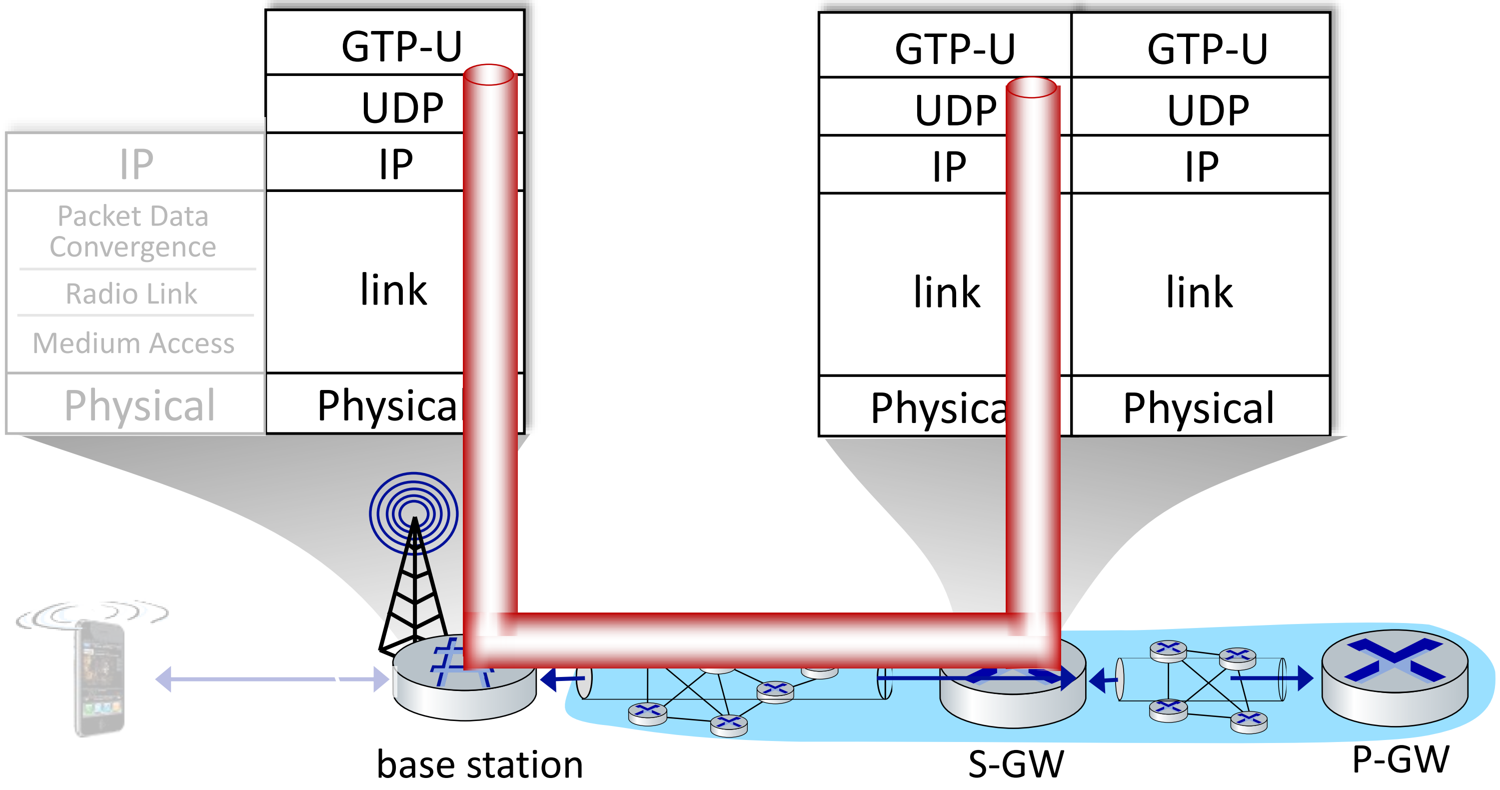
LTE data plane protocol stack: first hop



LTE link layer protocols:

- Packet Data Convergence: header compression, encryption
- Radio Link Control (RLC) Protocol: fragmentation/reassembly, reliable data transfer
- Medium Access: requesting, use of radio transmission slots (OFDM)

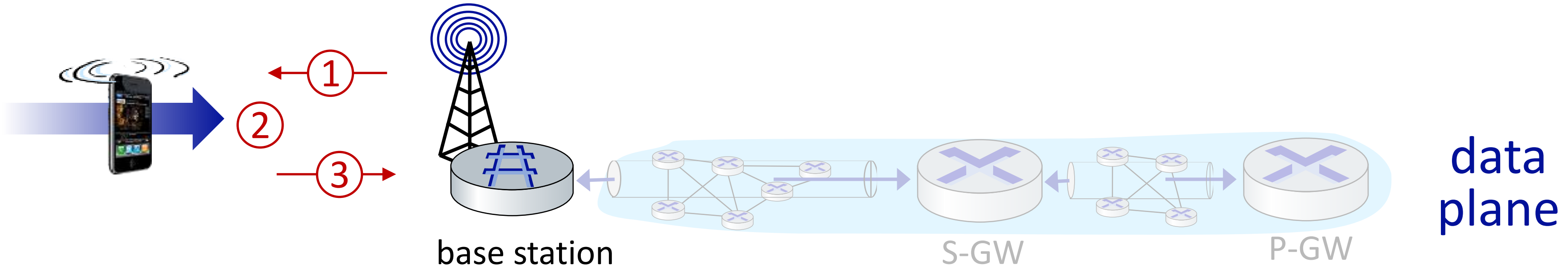
LTE data plane protocol stack: packet core



tunneling:

- mobile datagram encapsulated using GPRS Tunneling Protocol (GTP), sent inside UDP datagram to S-GW
- S-GW re-tunnels datagrams to P-GW
- supporting mobility: only tunneling endpoints change when mobile user moves

LTE data plane: associating with a BS



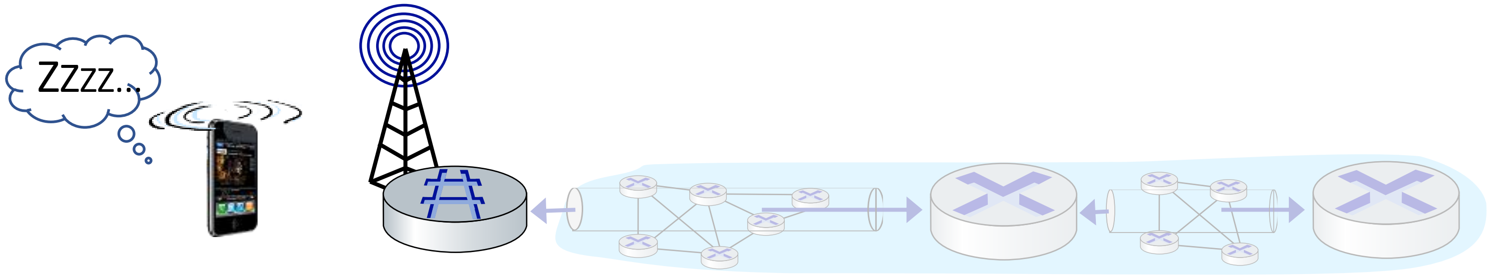
1 BS broadcasts primary synch signal every 5 ms on all frequencies
   - BSs from multiple carriers may be broadcasting synch signals

2 mobile finds a primary synch signal, then locates 2nd synch signal on this freq.
   - mobile then finds info broadcast by BS: channel bandwidth, configurations; BS's cellular carrier info
   - mobile may get info from multiple base stations, multiple cellular networks

3 mobile selects which BS to associate with (*e.g.,* preference for home carrier)

4 more steps still needed to authenticate, establish state, set up data plane

LTE mobiles: sleep modes

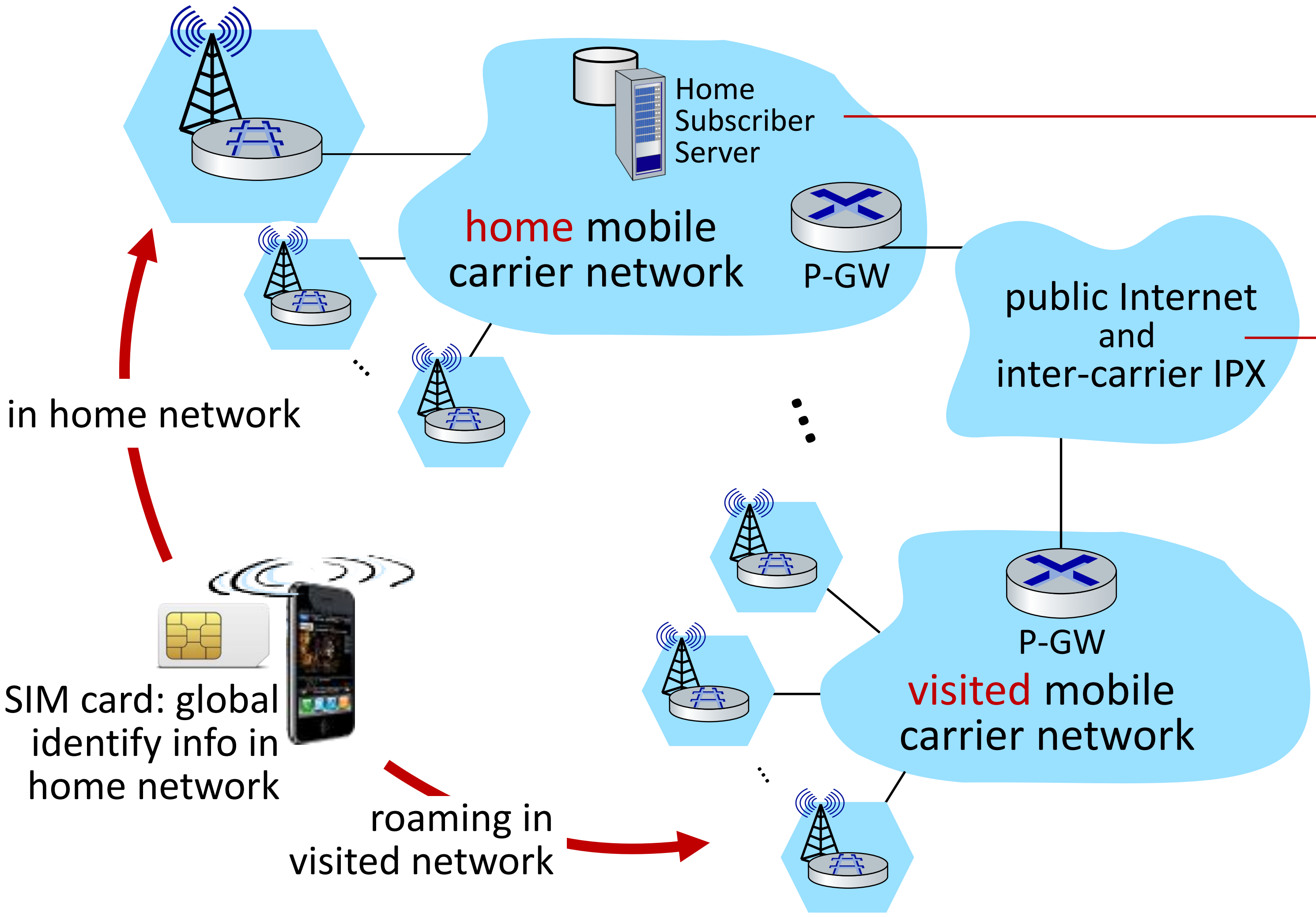


data plane

as in WiFi, Bluetooth: LTE mobile may put radio to “sleep” to conserve battery:

- light sleep: after 100’s msec of inactivity
  - wake up periodically (100’s msec) to check for downstream transmissions
- deep sleep: after 5-10 secs of inactivity
  - mobile may change cells while deep sleeping – need to re-establish association

Global cellular network: a network of IP networks



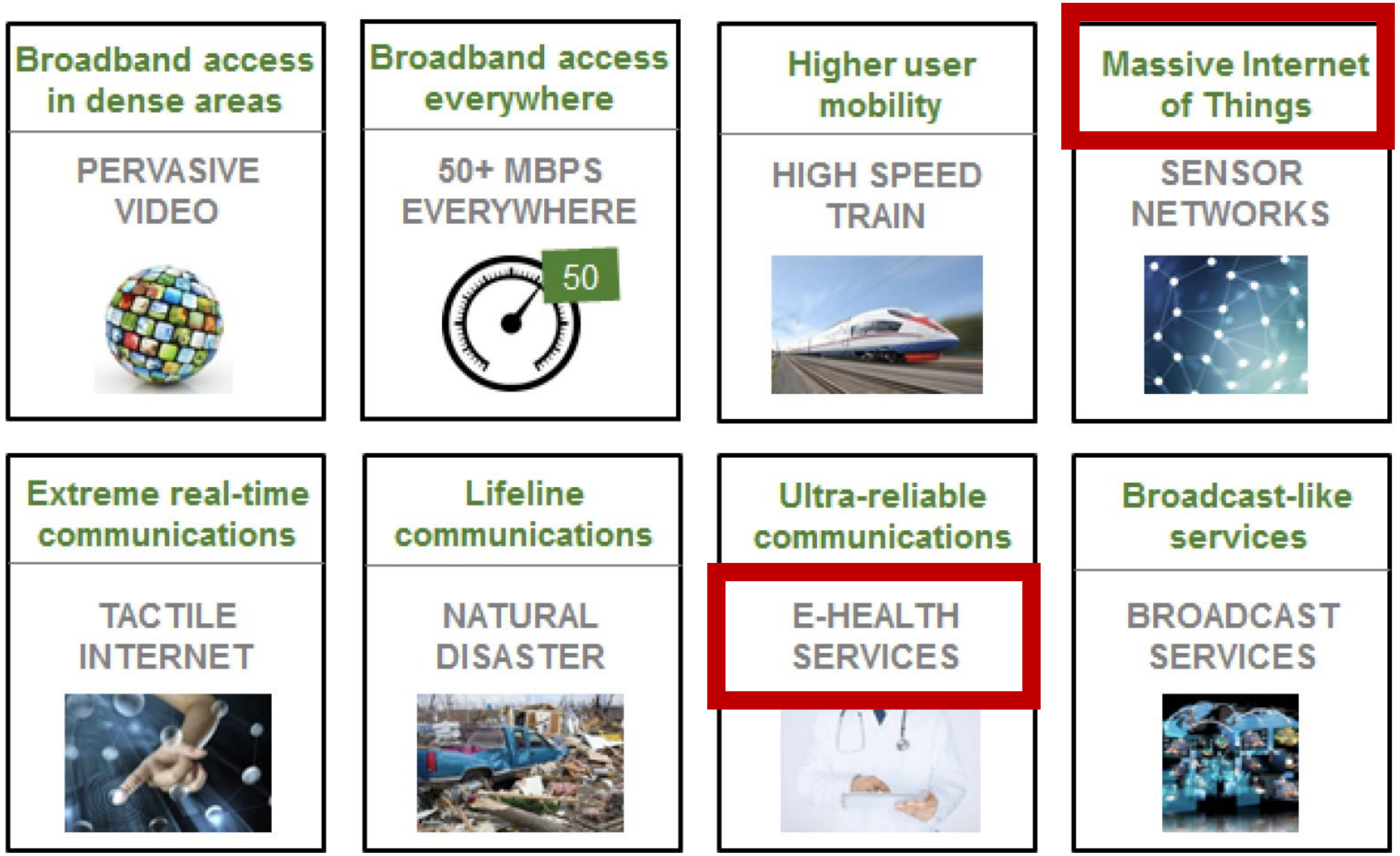
home network HSS:

- identify & services info, while in home network and roaming

all IP:

- carriers interconnect with each other, and public internet at exchange points
- legacy 2G, 3G: not all IP, handled otherwise

On to 5G: motivation



From Next Generation Mobile Networks (NGMS) alliance: 2020 white paper

Hype/wishes need to be separated from reality or likely nearer-term reality

On to 5G: motivation

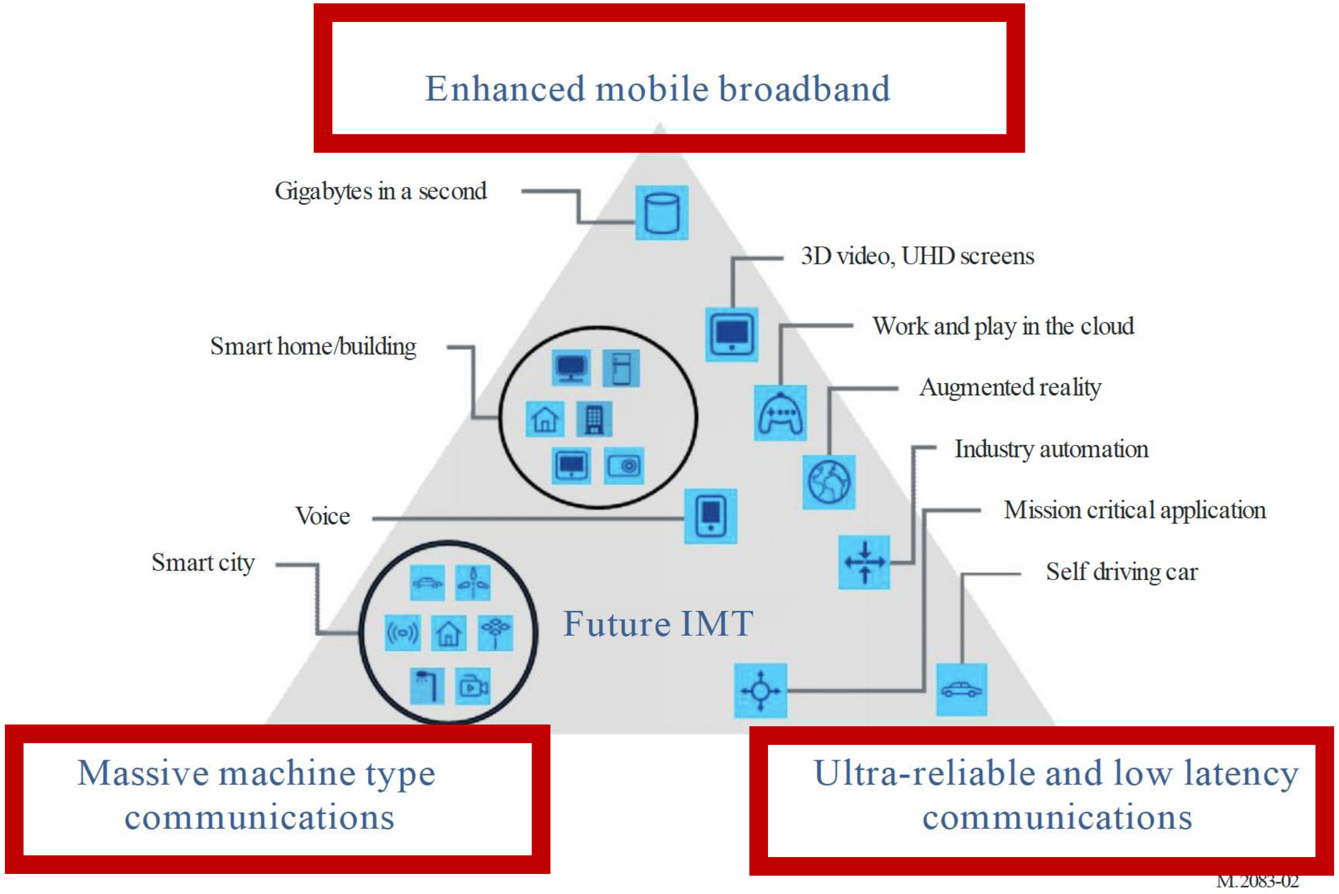
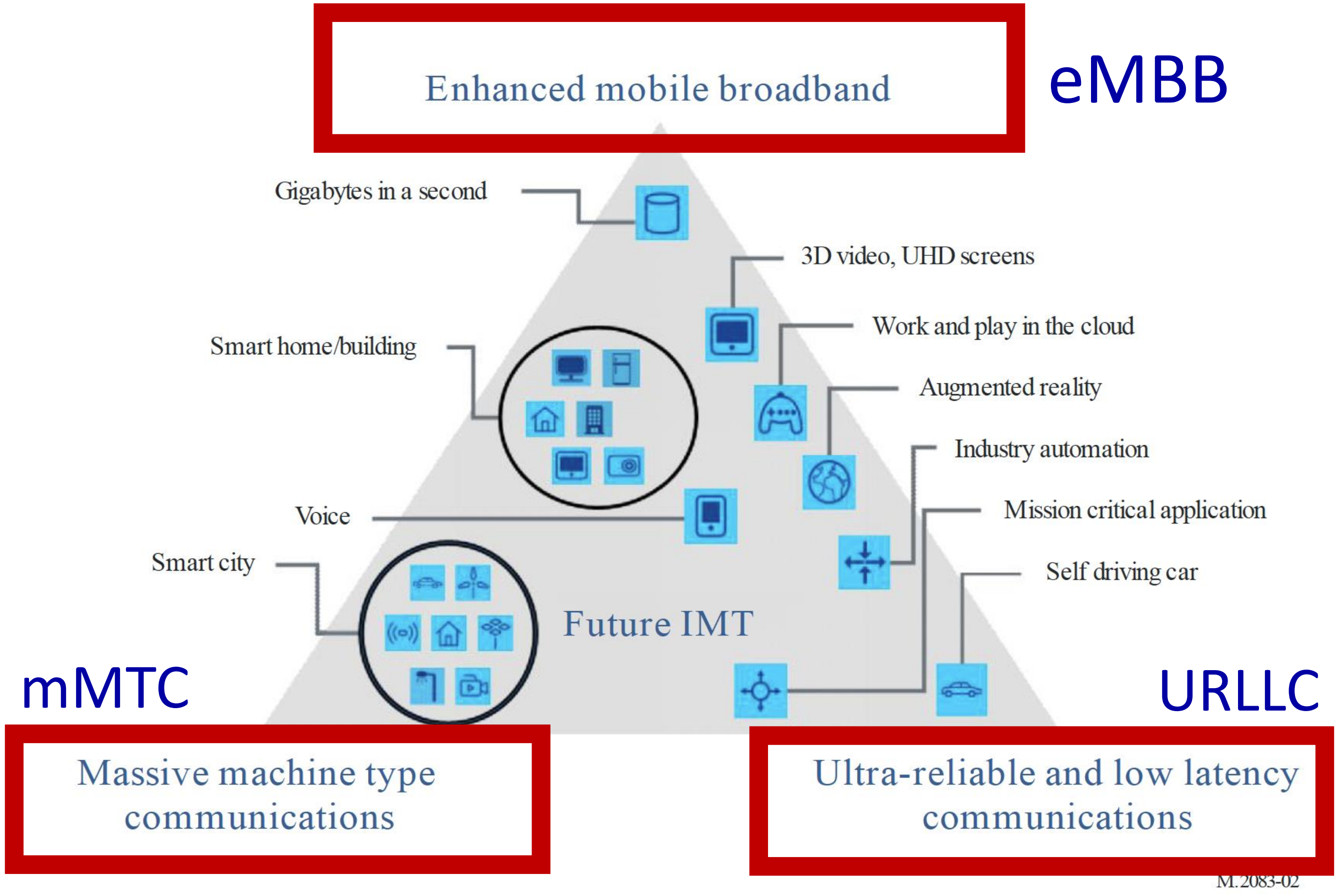


Figure: from Recommendation ITU-R M.2083-0 (2015)

"initial standards and launches have mostly focused on enhanced Mobile Broadband, 5G is expected to increasingly enable new business models and countless new use cases, in particular those of massive Machine Type Communications and Ultra-reliable and Low Latency Communications."

On to 5G: motivation



Industry verticals:

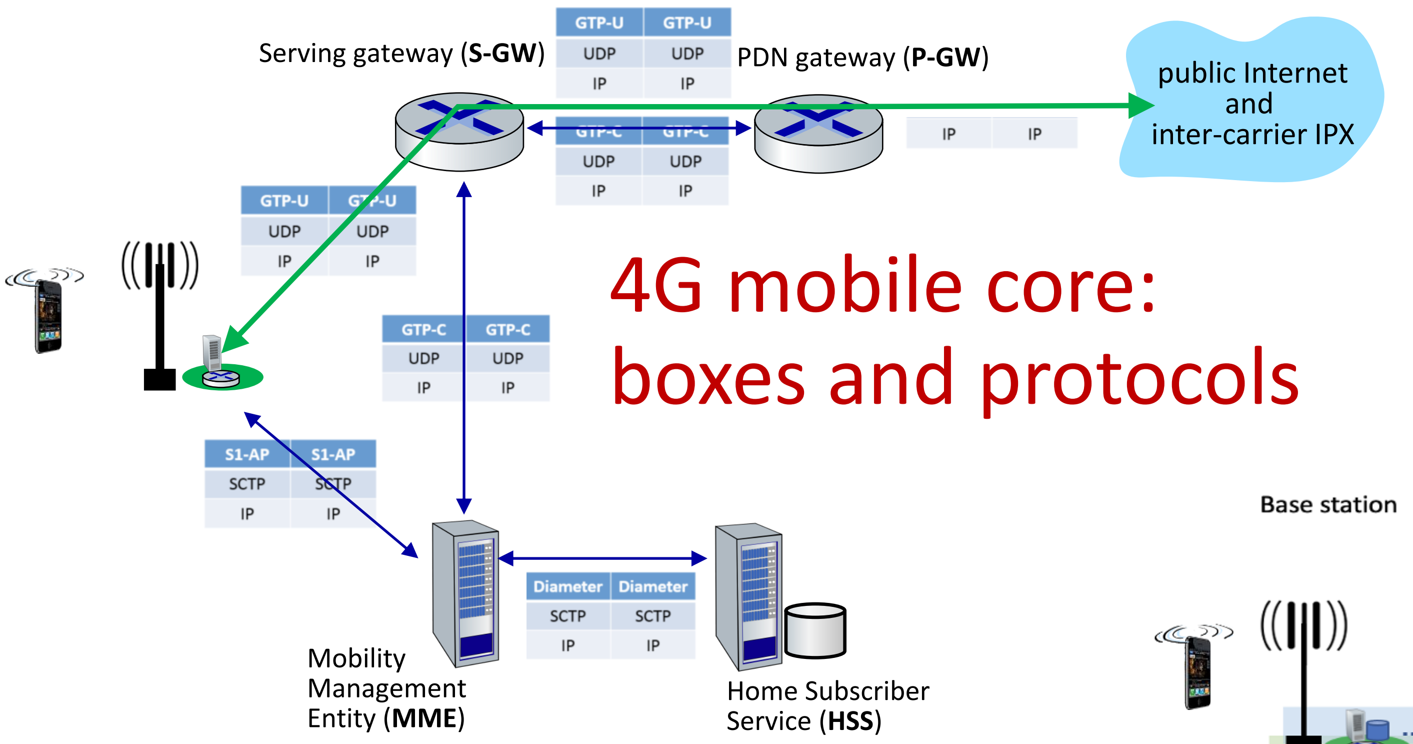
- Manufacturing
- Constructions
- Transport
- Health
- Smart communities
- Education
- Tourism
- Agriculture
- Finance

K. Schwab, "The Fourth Industrial Revolution," World Economic Forum.

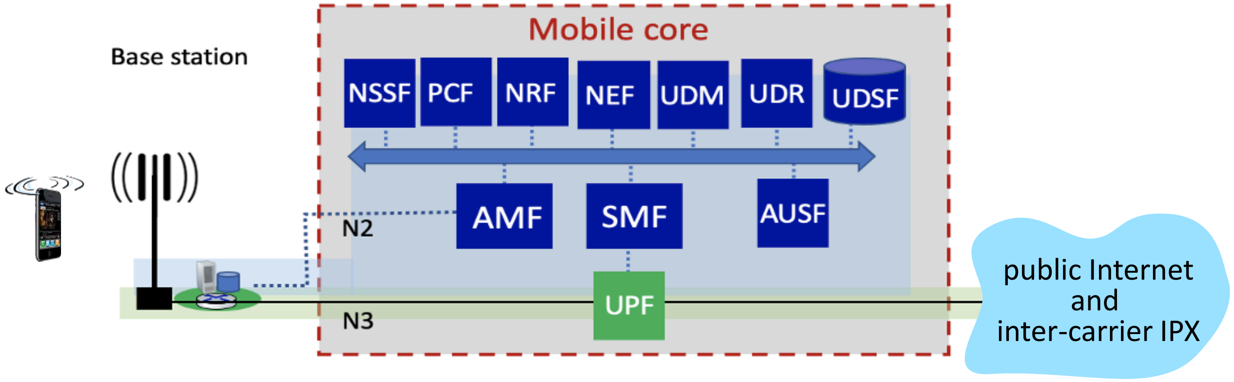
On to 5G: Radio

- goal: 10x increase in peak bitrate, 10x decrease in latency, 100x increase in traffic capacity over 4G
- 5G NR (new radio):
  - two frequency bands: FR1 (450 MHz–6 GHz) and FR2 (24 GHz–52 GHz): millimeter wave frequencies
  - not backwards-compatible with 4G
  - MIMO: multiple directional antennae
- millimeter wave frequencies: much higher data rates, but over shorter distances
  - pico-cells: cells diameters: 10-100 m
  - massive, dense deployment of new base stations required

On to 5G: SDN-like architecture

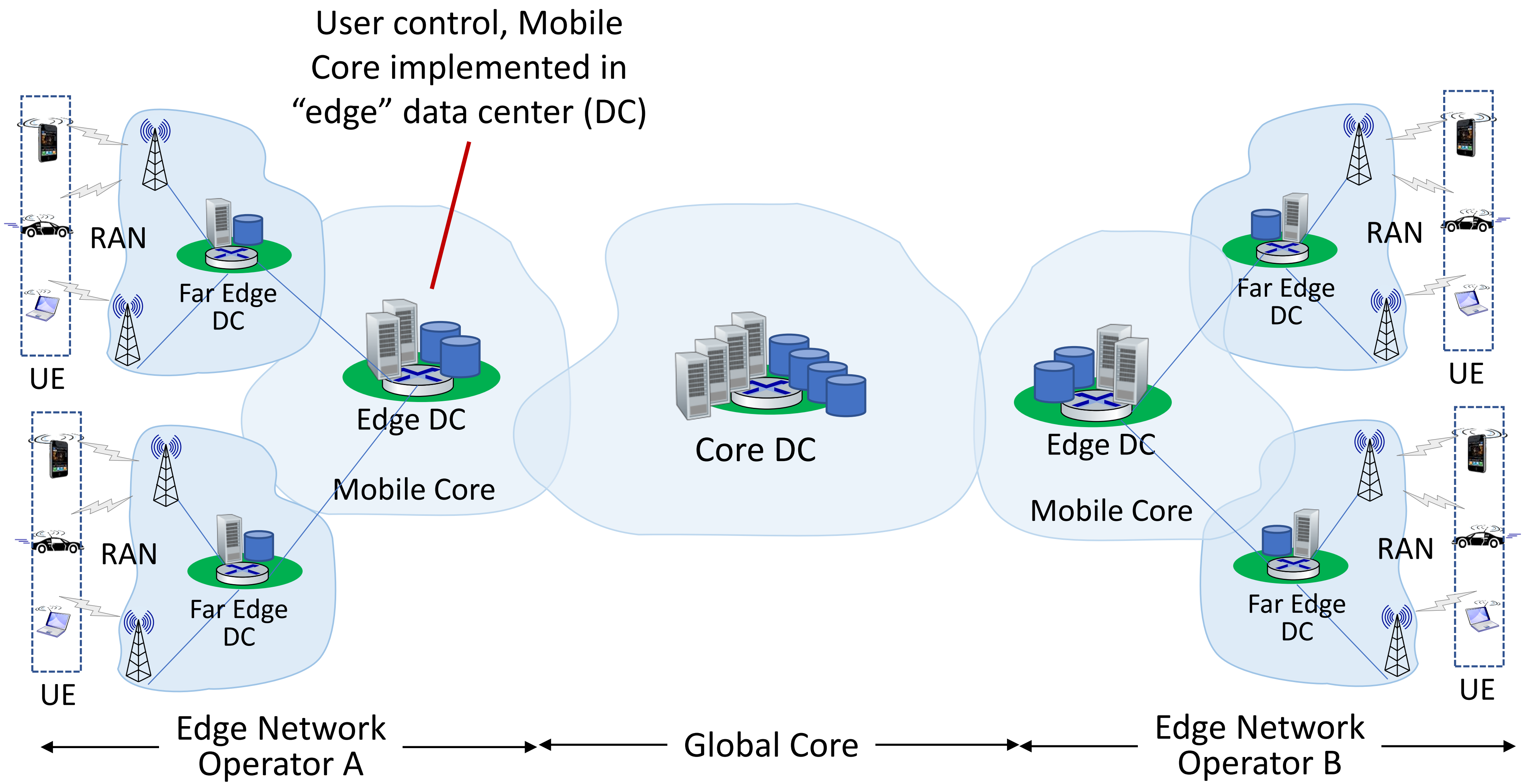


4G mobile core: boxes and protocols

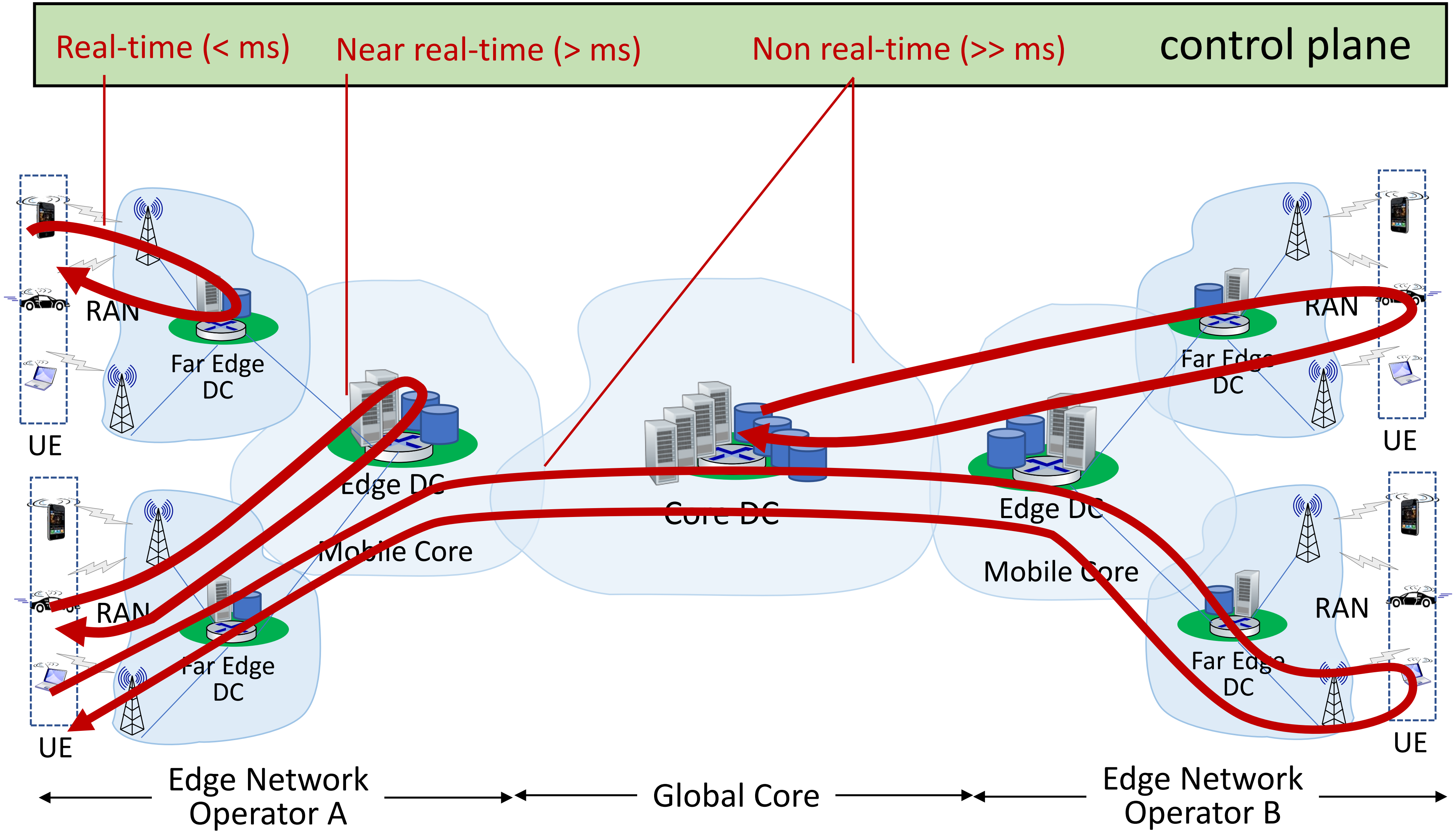


5G: microservice-like architecture

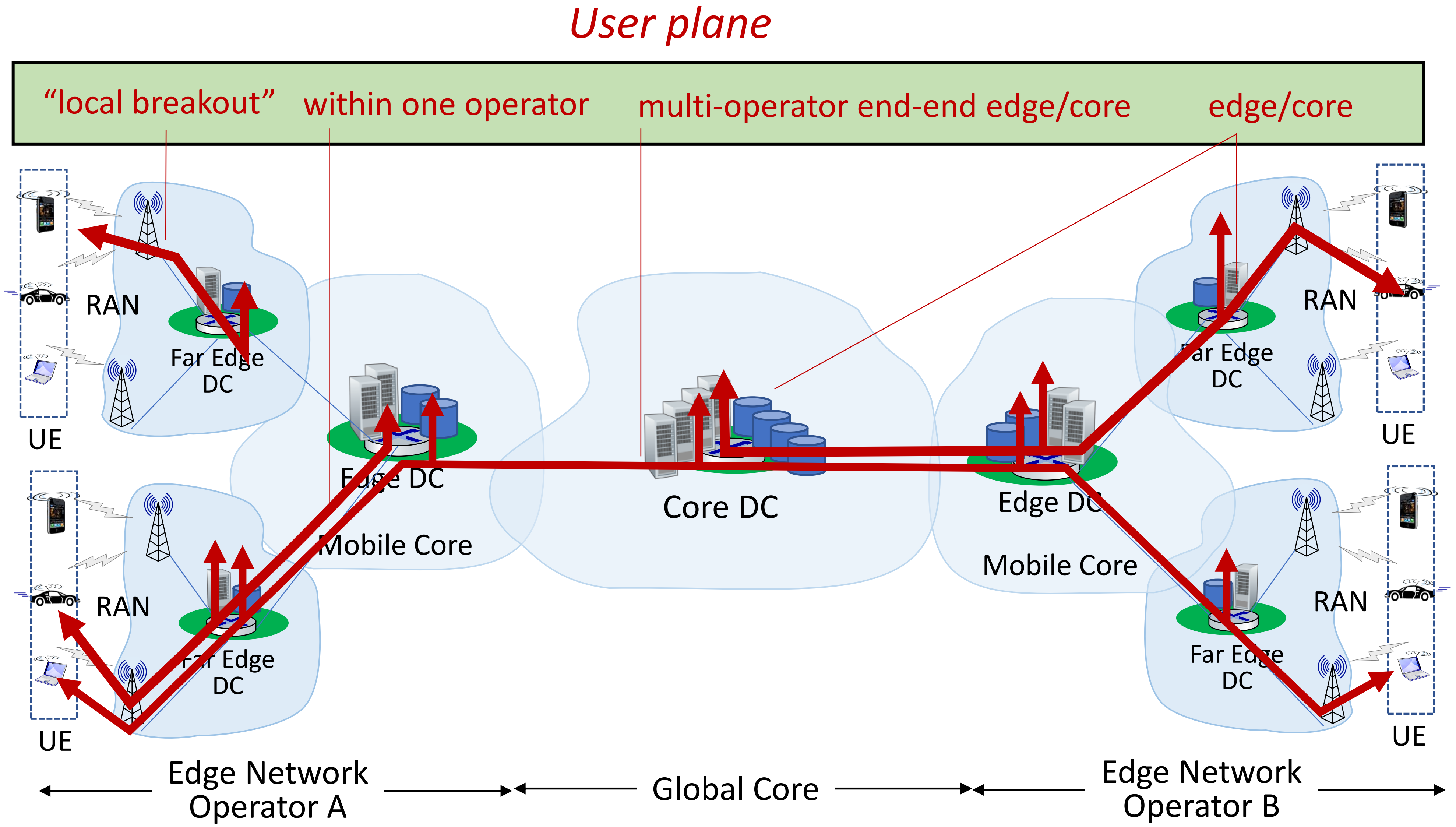
Functional elements: communication, computation, data



Control plane: resource control



User plane: resources, as used by users (application)



On beyond 5G?

- "6G" not obviously next: "NextG" and "Beyond 5G" heard more often than "6G"
- 5G on an evolutionary path (like the Internet)
  - agility: cloud technologies (SDN) mean new features can be introduced rapidly, deployed continuously
  - customization: change can be introduced bottom-up (e.g., by enterprises and edge cloud partners with Private 5G)
    - No need to wait for standardization
    - No need to reach agreement (among all incumbent stakeholders)

Chapter 7 outline

- Introduction

Wireless

- Wireless links and network characteristics
- WiFi: 802.11 wireless LANs
- Cellular networks: 4G and 5G

Mobility

- Mobility management: principles
- Mobility management: practice
  - 4G/5G networks
  - Mobile IP
- Mobility: impact on higher-layer protocols

What is mobility?

- spectrum of mobility, from the network perspective:

no mobility

high mobility

device moves between networks, but powers down while moving

device moves within same AP in one provider network

device moves among APs in one provider network

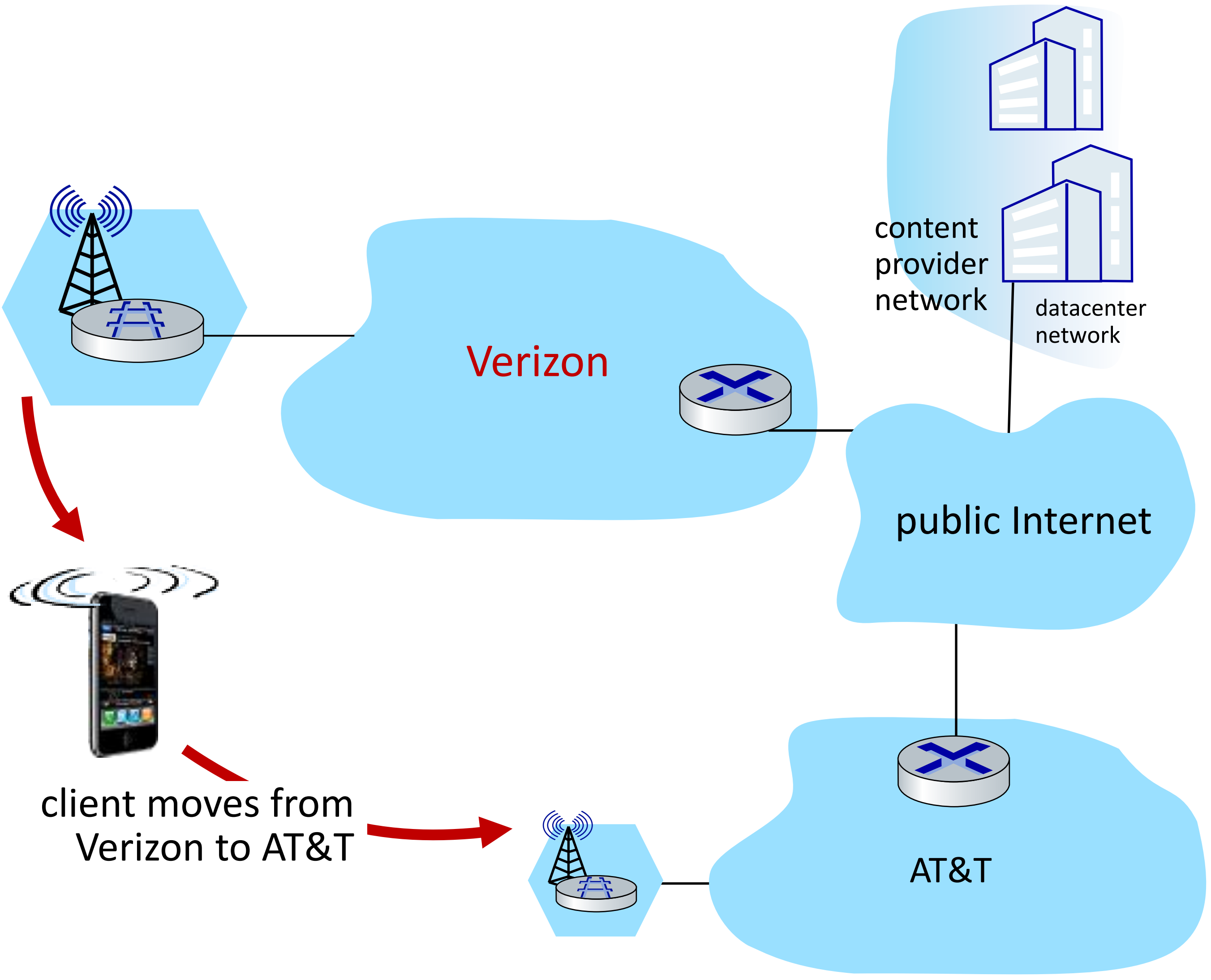
device moves among multiple provider networks, while maintaining ongoing connections

We're interested in these!

Mobility challenge:

If a device moves from one network another:

- How will the "network" know to forward packets to the *new* network?



Mobility approaches

- let network (routers) handle it:
  - routers advertise well-known name, address (e.g., permanent 32-bit IP address), or number (e.g., cell #) of visiting mobile node via usual routing table exchange
  - Internet routing could do this already *with no* changes! Routing tables indicate where each mobile located via longest prefix match!

Mobility approaches

- let network (routers) handle it:
  - routers advertise well-kn… address (e.g., permanent 32-bit IP address), or numb…) of visiting mobile node via usual routing table exch…
  - Internet routing could do t… …dy *with no* changes! Routing tables indicate where each mobile located via longest prefix match!

not scalable to billions of mobiles

- let end-systems handle it: functionality at the "edge"
  - *indirect routing:* communication from correspondent to mobile goes through home network, then forwarded to remote mobile
  - *direct routing:* correspondent gets foreign address of mobile, send directly to mobile

Contacting a mobile friend:

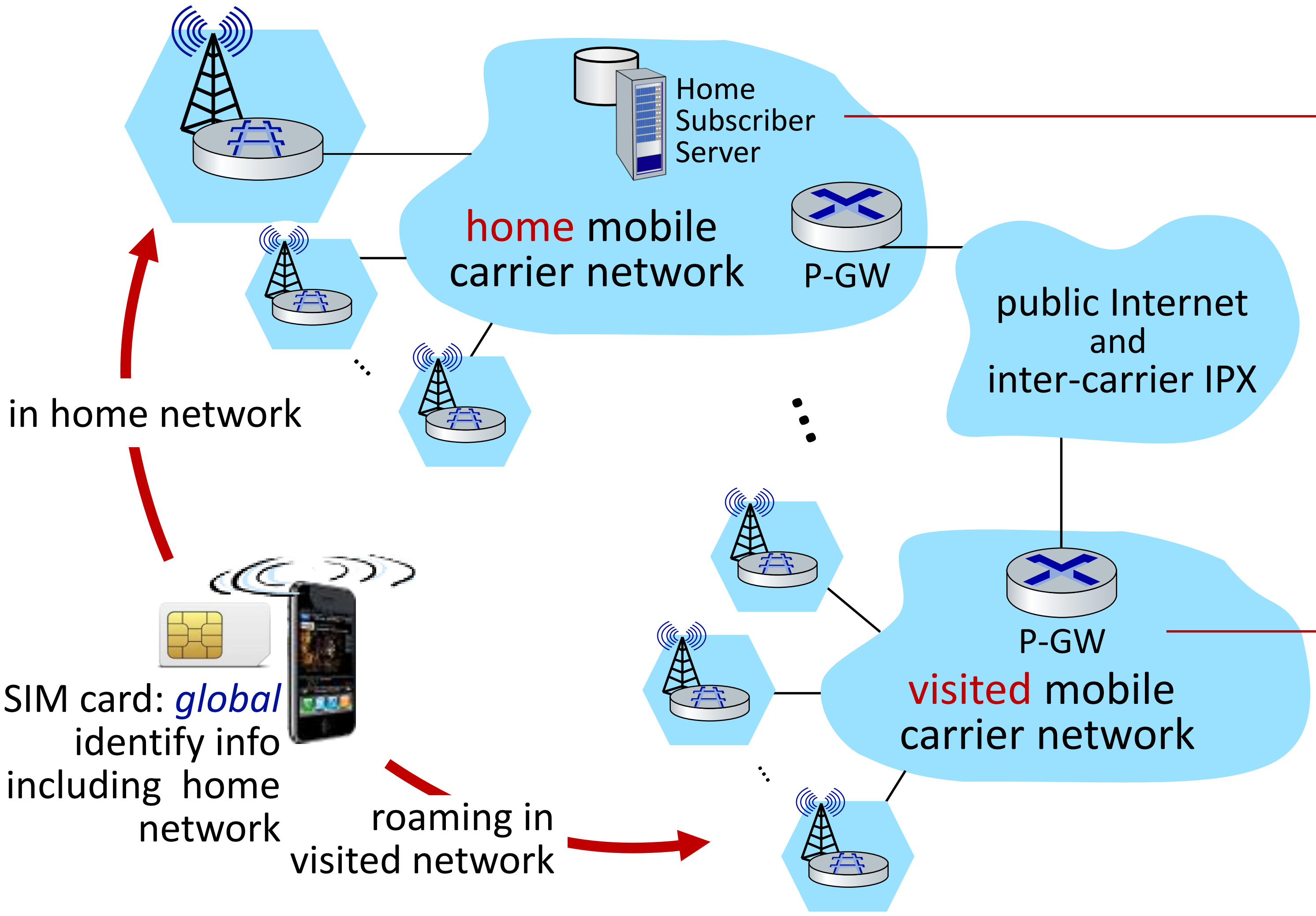
Consider friend frequently changing locations, how do you find him/her?

- search all phone books?
- expect her to let you know where he/she is?
- call his/her parents?
- Facebook!

The importance of having a "home":

- a definitive source of information about you
- a place where people can find out where you are

Home network, visited network: 4G/5G



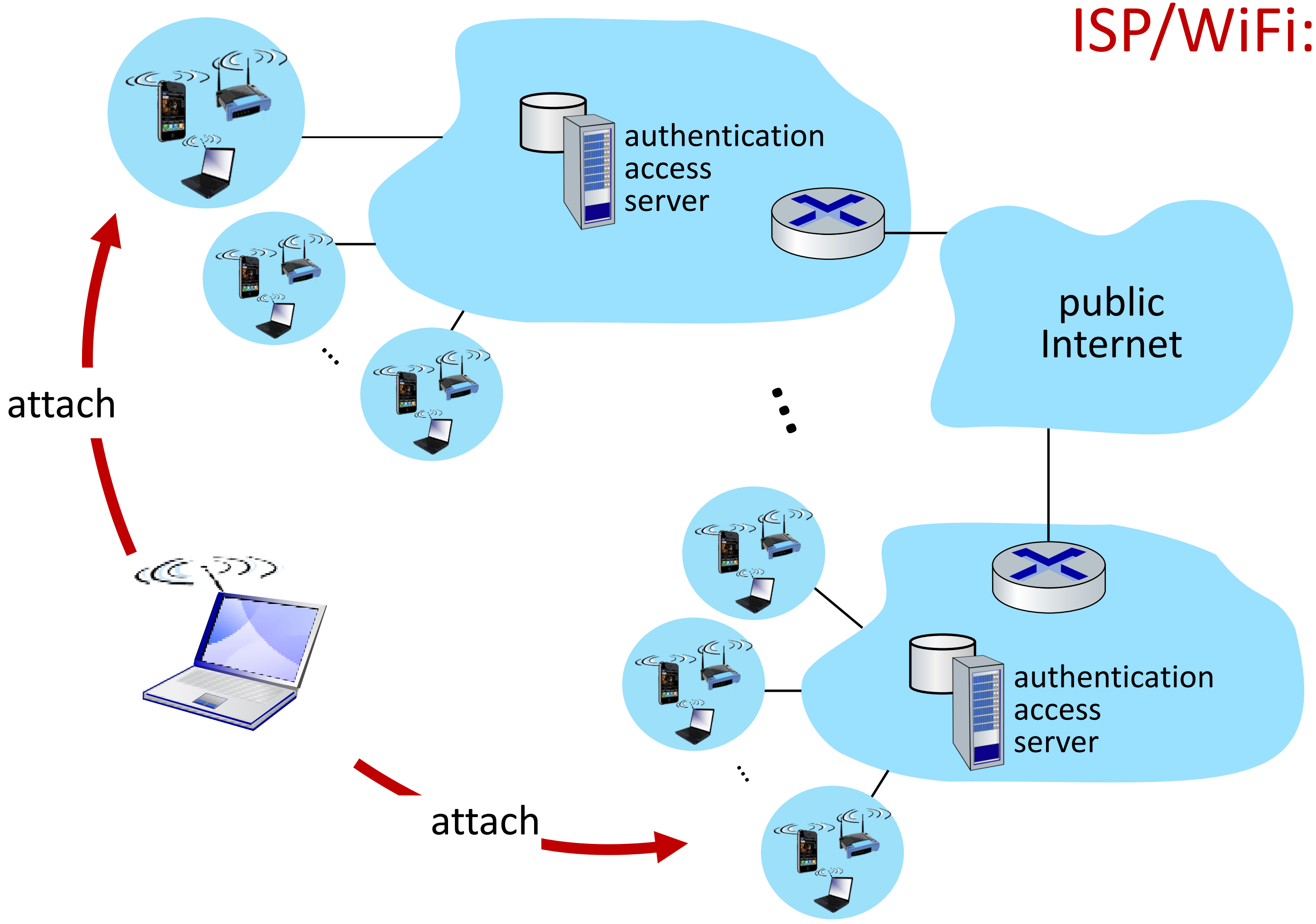
home network:

- (paid) service plan with cellular provider, e.g., Verizon, Orange
- home network HSS stores identify & services info

visited network:

- any network other than your home network
- service agreement with other networks: to provide access to visiting mobile

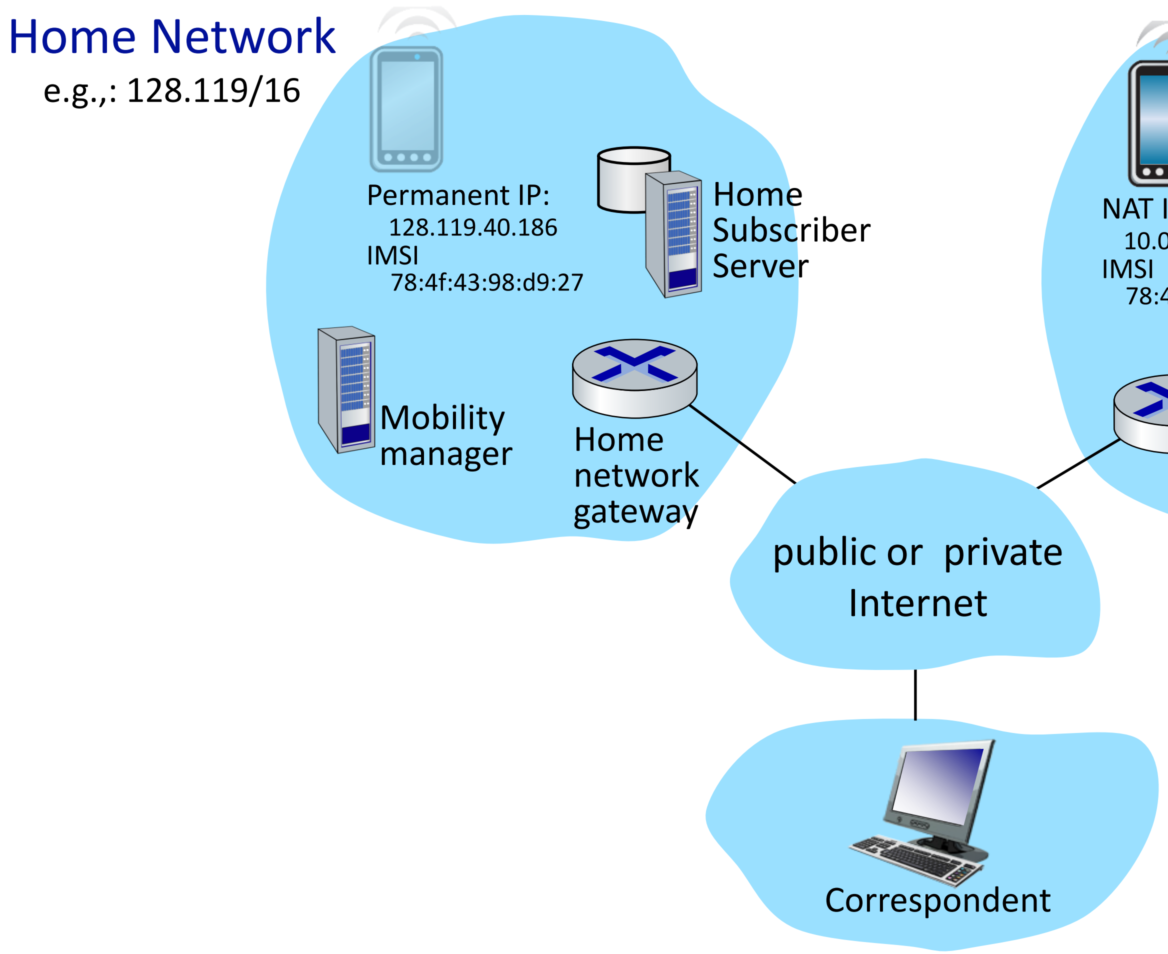
Home network, visited network: ISP/WiFi



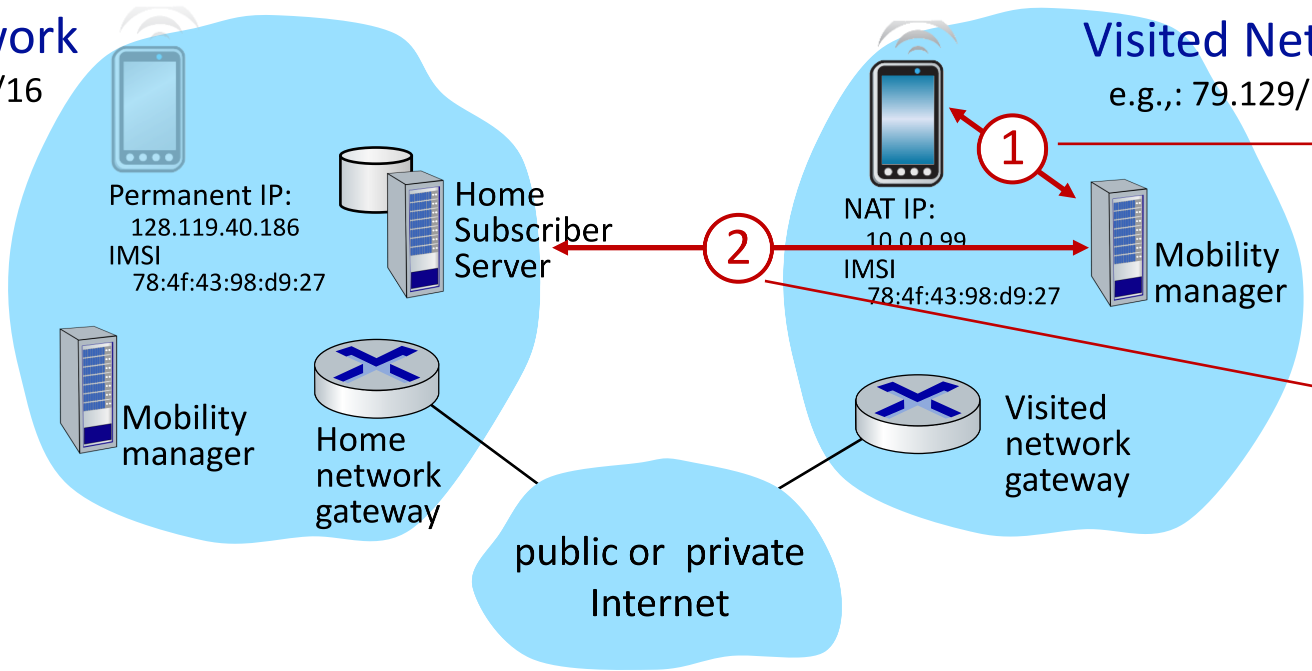
ISP/WiFi: no notion of global "home"

- credentials from ISP (e.g., username, password) stored on device or with user
- ISPs may have national, international presence
- different networks: different credentials
  - some exceptions (e.g., eduroam)
  - architectures exist (mobile IP) for 4G-like mobility, but not used

Home network, visited network: generic



Registration: home needs to know where you are!



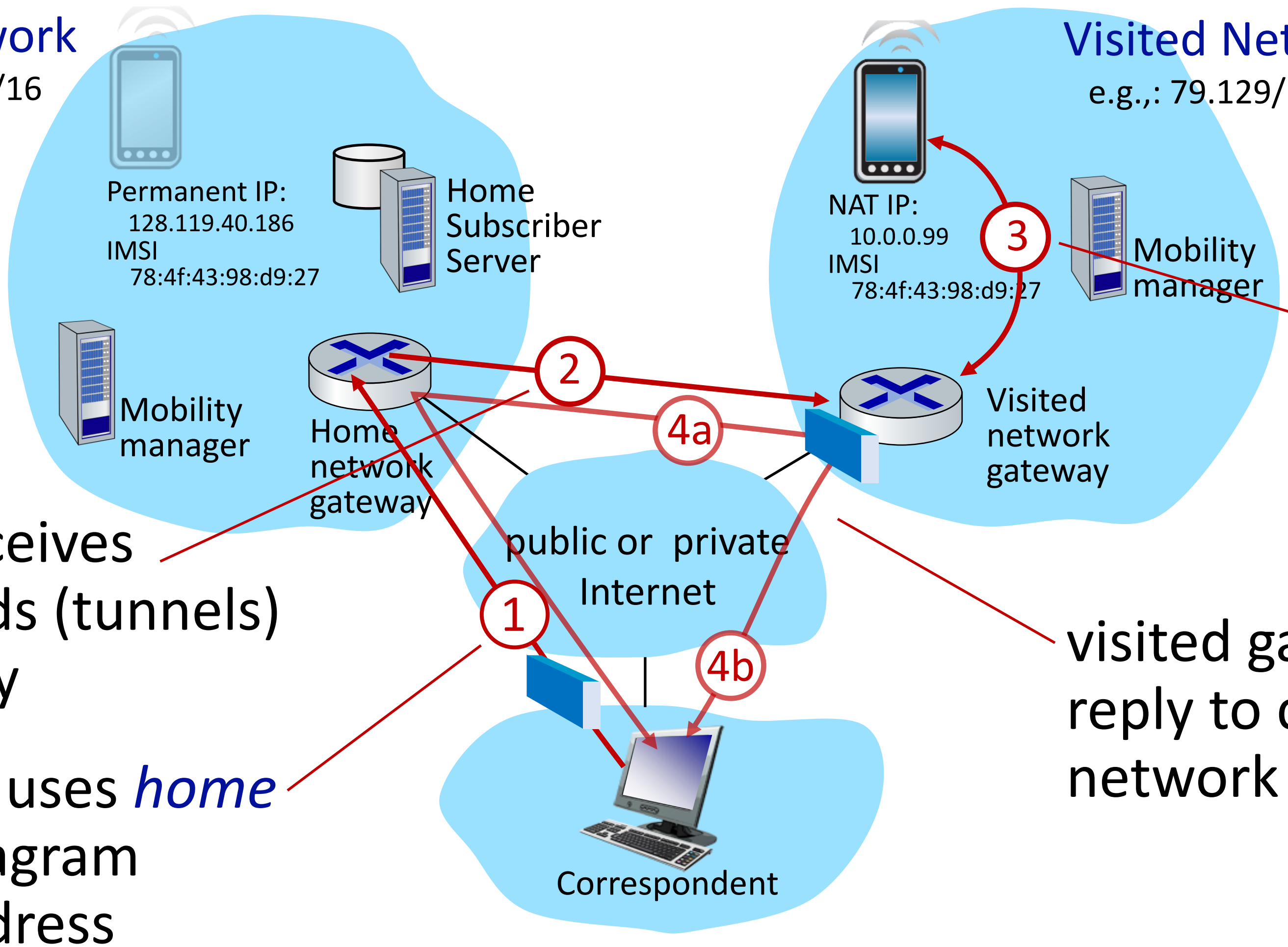
mobile *associates* with visited mobility manager

visited mobility manager *registers* mobile's location with home HSS

end result:

- visited mobility manager knows about mobile
- home HSS knows location of mobile

Mobility with indirect routing



visited gateway router forwards to mobile

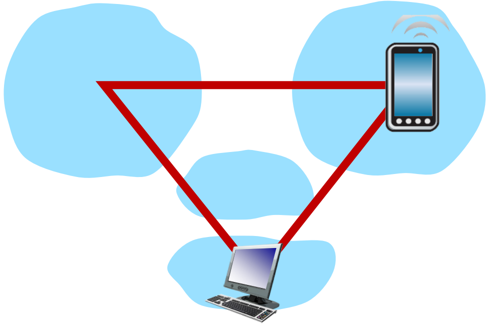
home gateway receives datagram, forwards (tunnels) to remote gateway

visited gateway router forwards reply to correspondent via home network (4a) or directly (4b)

correspondent uses *home* address as datagram destination address

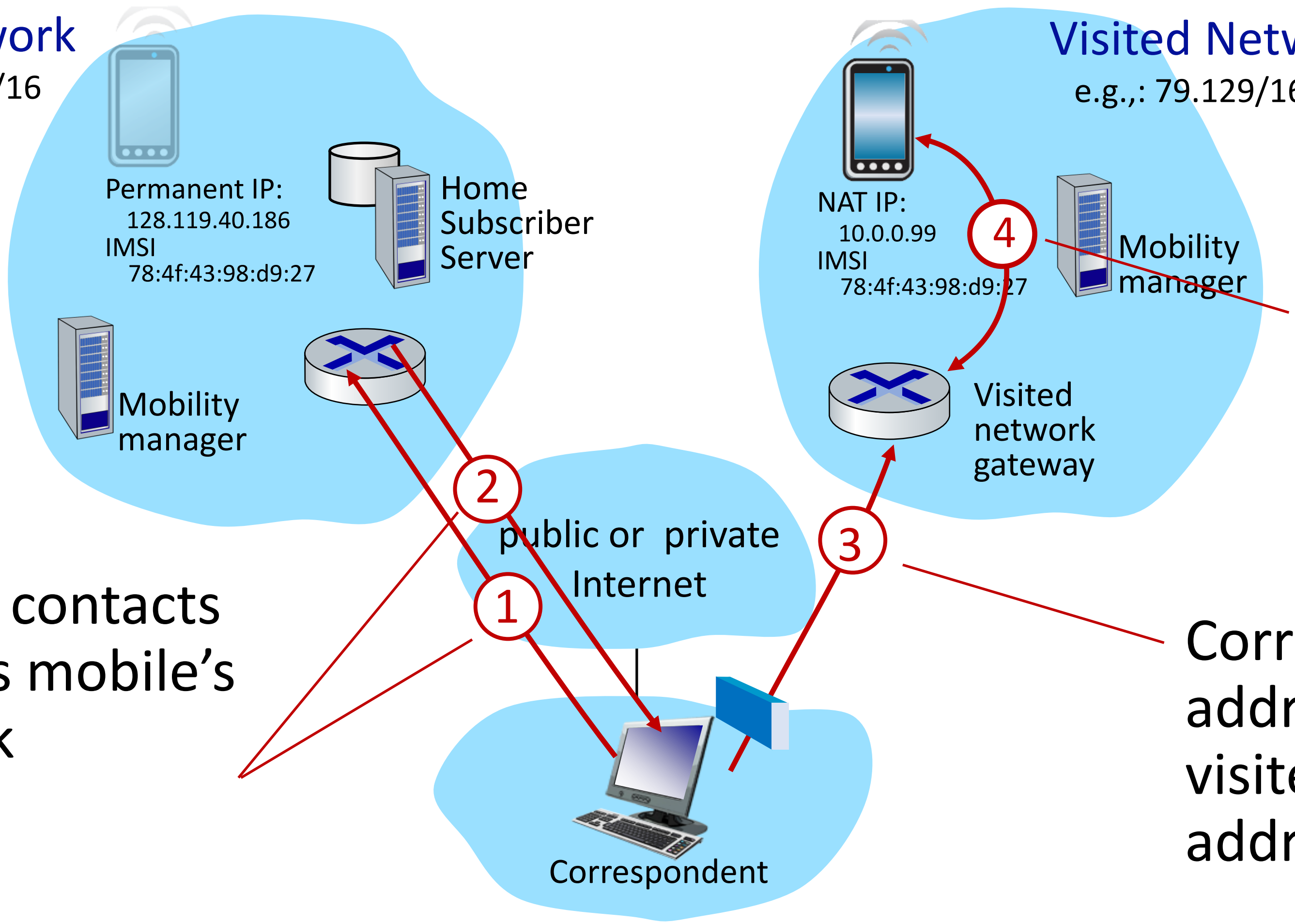
Mobility with indirect routing: comments

- triangle routing:
  - inefficient when correspondent and mobile are in same network



- mobile moves among visited networks: transparent to correspondent!
  - registers in new visited network
  - new visited network registers with home HSS
  - datagrams continue to be forwarded from home network to mobile in new network
  - *on-going (e.g., TCP) connections between correspondent and mobile can be maintained!*

Mobility with direct routing



correspondent contacts *home* HSS, gets mobile's visited network

Correspondent addresses datagram to visited network address

visited gateway router forwards to mobile

Mobility with direct routing: comments

- overcomes triangle routing inefficiencies
- *non-transparent to correspondent:* correspondent must get care-of-address from home agent
- what if mobile changes visited network?
    - can be handled, but with additional complexity

Chapter 7 outline

- Introduction

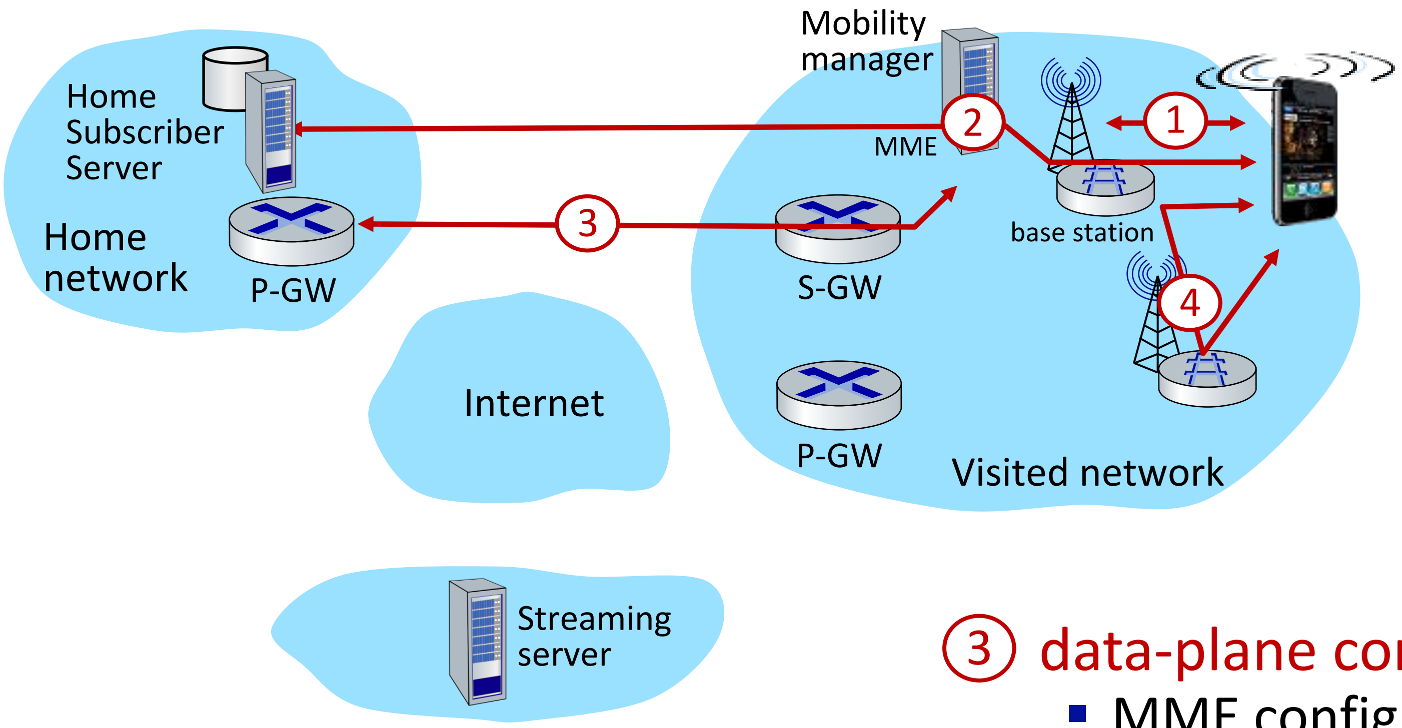
Wireless

- Wireless links and network characteristics
- WiFi: 802.11 wireless LANs
- Cellular networks: 4G and 5G

Mobility

- Mobility management: principles
- Mobility management: practice
  - 4G/5G networks
  - Mobile IP
- Mobility: impact on higher-layer protocols

Mobility in 4G networks: major mobility tasks



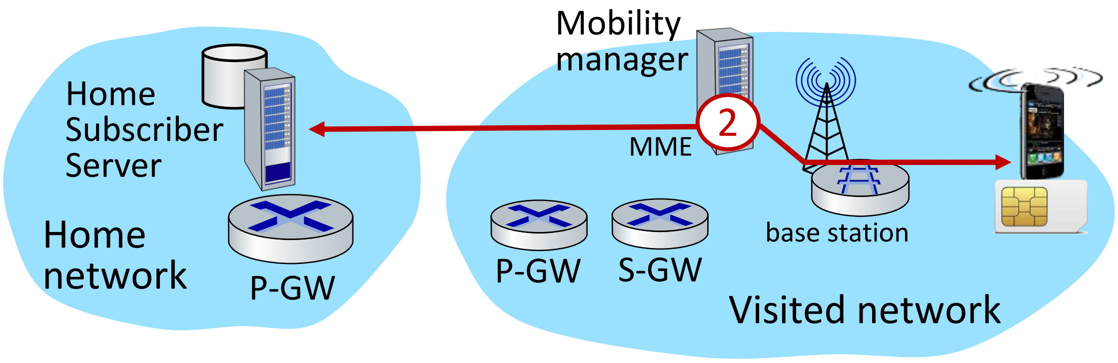
1. base station association:
   - covered earlier
   - mobile provides IMSI – identifying itself, home network

2. control-plane configuration:
   - MME, home HSS establish control-plane state - mobile is in visited network

3. data-plane configuration:
   - MME configures forwarding tunnels for mobile
   - visited, home network establish tunnels from home P-GW to mobile

4. mobile handover:
   - mobile device changes its point of attachment to visited network

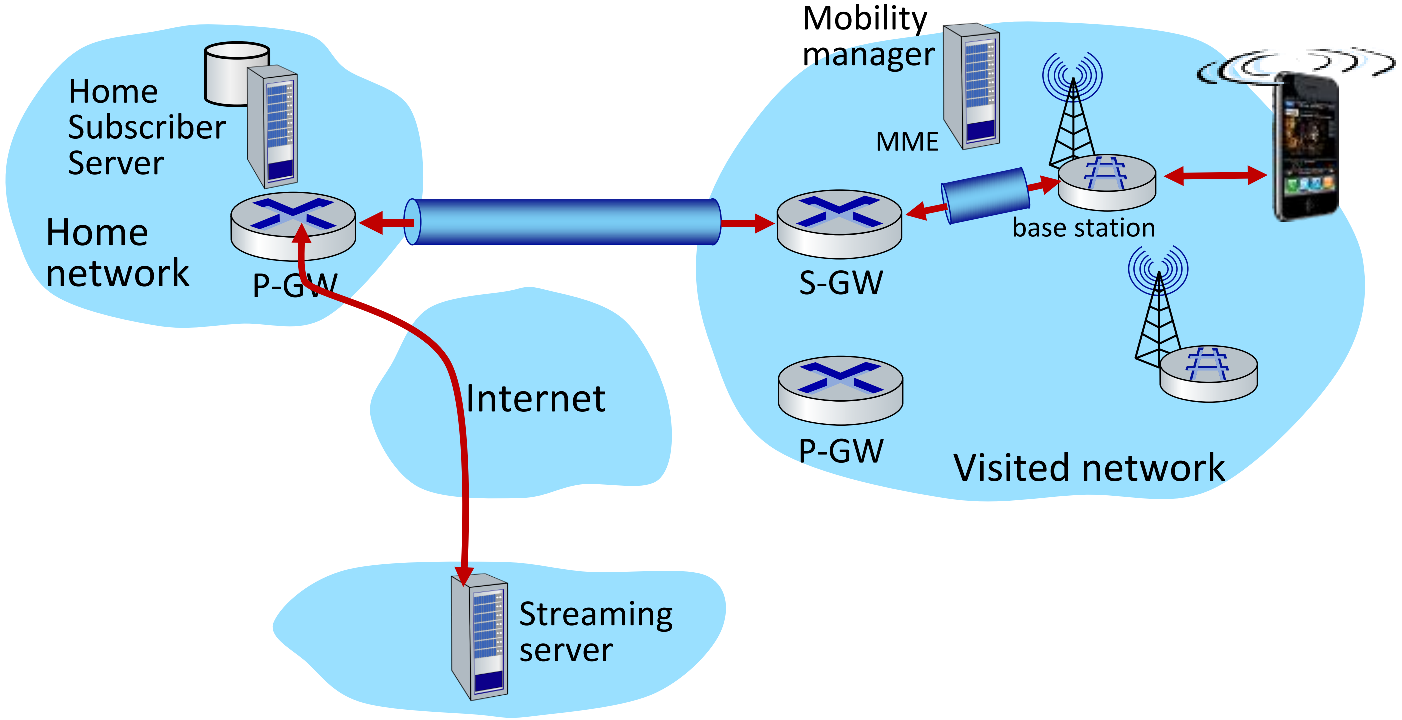
Configuring LTE control-plane elements



- Mobile communicates with local MME via BS control-plane channel
- MME uses mobile's IMSI info to contact mobile's home HSS
  - retrieve authentication, encryption, network service information
  - home HHS knows mobile now resident in visited network
- BS, mobile select parameters for BS-mobile data-plane radio channel

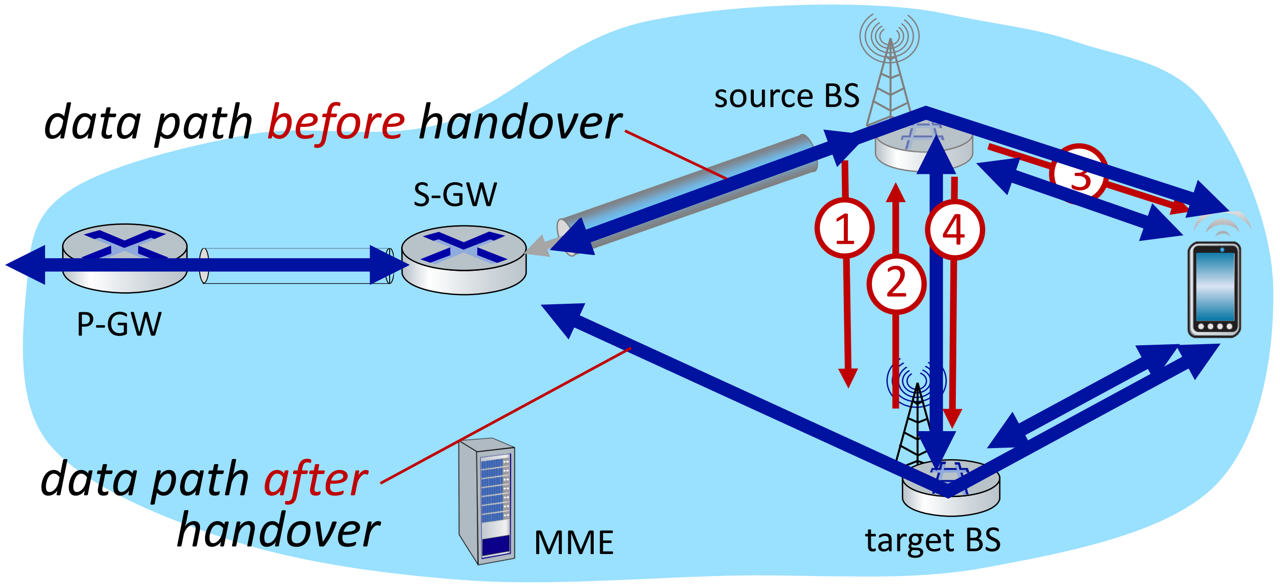
Configuring data-plane tunnels for mobile

- S-GW to BS tunnel: when mobile changes base stations, simply change endpoint IP address of tunnel
- S-GW to home P-GW tunnel: implementation of indirect routing



- tunneling via GTP (GPRS tunneling protocol): mobile's datagram to streaming server encapsulated using GTP inside UDP, inside datagram

Handover between BSs in same cellular network



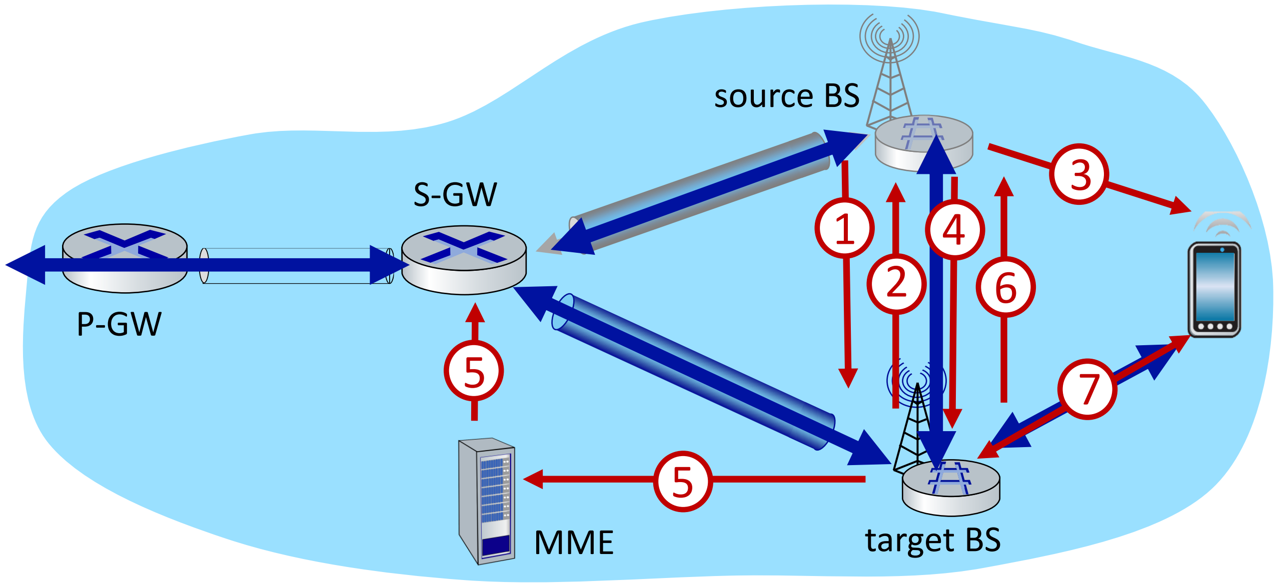
1. current (source) BS selects target BS, sends *Handover Request message* to target BS

2. target BS pre-allocates radio time slots, responds with HR ACK with info for mobile

3. source BS informs mobile of new BS
   - mobile can now send via new BS - handover *looks* complete to mobile

4. source BS stops sending datagrams to mobile, instead forwards to new BS (who forwards to mobile over radio channel)

Handover between BSs in same cellular network



5 target BS informs MME that it is new BS for mobile

- MME instructs S-GW to change tunnel endpoint to be (new) target BS

6 target BS ACKs back to source BS: handover complete, source BS can release resources

7 mobile's datagrams now flow through new tunnel from target BS to S-GW

Mobile IP

- mobile IP architecture standardized ~20 years ago [RFC 5944]
  - long before ubiquitous smartphones, 4G support for Internet protocols
  - did not see wide deployment/use
  - perhaps WiFi for Internet, and 2G/3G phones for voice were "good enough" at the time
- mobile IP architecture:
  - indirect routing to node (via home network) using tunnels
  - mobile IP home agent: combined roles of 4G HSS and home P-GW
  - mobile IP foreign agent: combined roles of 4G MME and S-GW
  - protocols for agent discovery in visited network, registration of visited location in home network via ICMP extensions

Wireless, mobility: impact on higher layer protocols

- logically, impact *should* be minimal …
  - best effort service model remains unchanged
  - TCP and UDP can (and do) run over wireless, mobile
- … but performance-wise:
  - packet loss/delay due to bit-errors (discarded packets, delays for link-layer retransmissions), and handover loss
  - TCP interprets loss as congestion, will decrease congestion window un-necessarily
  - delay impairments for real-time traffic
  - bandwidth a scare resource for wireless links

Chapter 8
Security

Security: overview

Chapter goals:

- understand principles of network security:
  - cryptography and its *many* uses beyond “confidentiality”
  - authentication
  - message integrity
- security in practice:
  - firewalls and intrusion detection systems
  - security in application, transport, network, link layers

Chapter 8 outline

- What is network security?
- Principles of cryptography
- Message integrity, authentication
- Securing e-mail
- Securing TCP connections: TLS
- Network layer security: IPsec
- Security in wireless and mobile networks
- Operational security: firewalls and IDS

What is network security?

confidentiality: only sender, intended receiver should "understand" message contents

- sender encrypts message
- receiver decrypts message

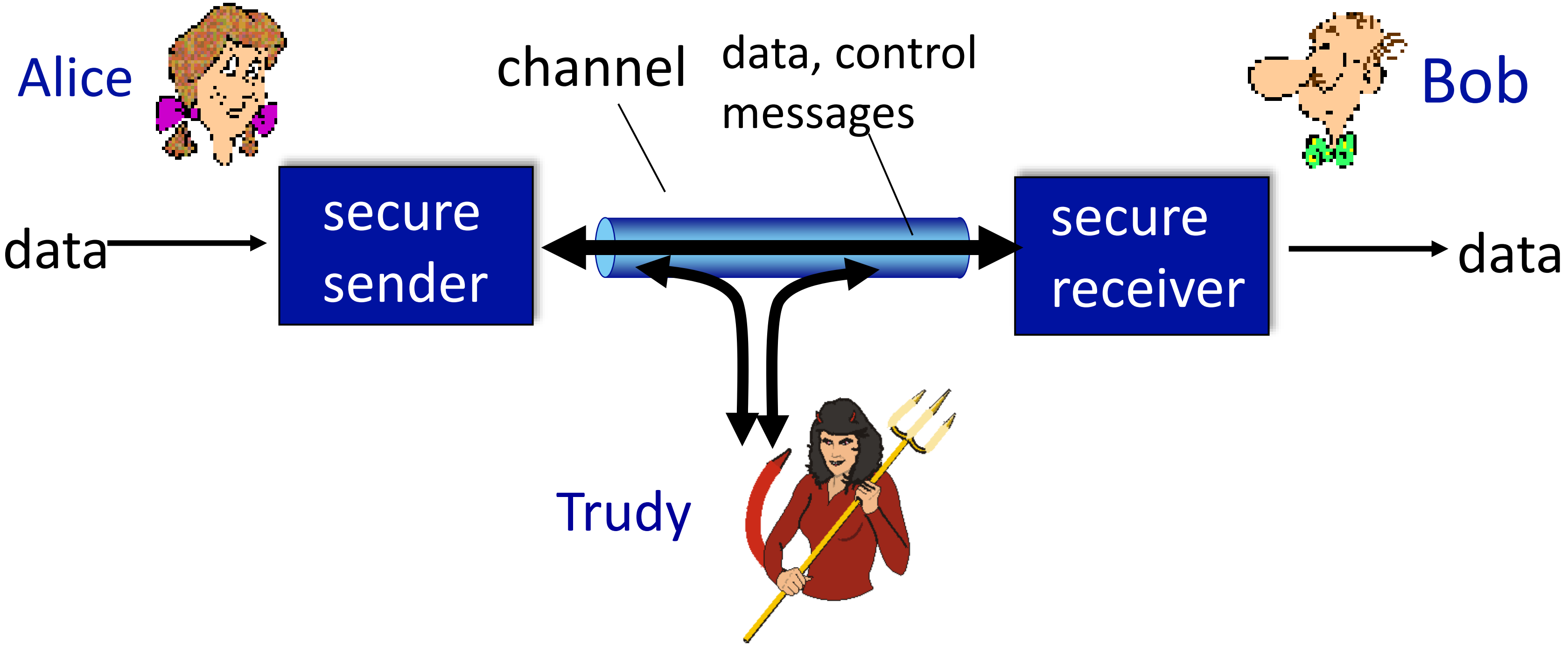
authentication: sender, receiver want to confirm identity of each other

message integrity: sender, receiver want to ensure message not altered (in transit, or afterwards) without detection

access and availability: services must be accessible and available to users

Friends and enemies: Alice, Bob, Trudy

- well-known in network security world
- Bob, Alice (lovers!) want to communicate "securely"
- Trudy (intruder) may intercept, delete, add messages



Friends and enemies: Alice, Bob, Trudy

Who might Bob and Alice be?

- … well, *real-life* Bobs and Alices!
- Web browser/server for electronic transactions (e.g., on-line purchases)
- on-line banking client/server
- DNS servers
- BGP routers exchanging routing table updates
- other examples?

There are bad guys (and girls) out there!

Q: What can a "bad guy" do?

A: A lot! (recall section 1.6)

- eavesdrop: intercept messages
- actively insert messages into connection
- impersonation: can fake (spoof) source address in packet (or any field in packet)
- hijacking: "take over" ongoing connection by removing sender or receiver, inserting himself in place
- denial of service: prevent service from being used by others (e.g., by overloading resources)

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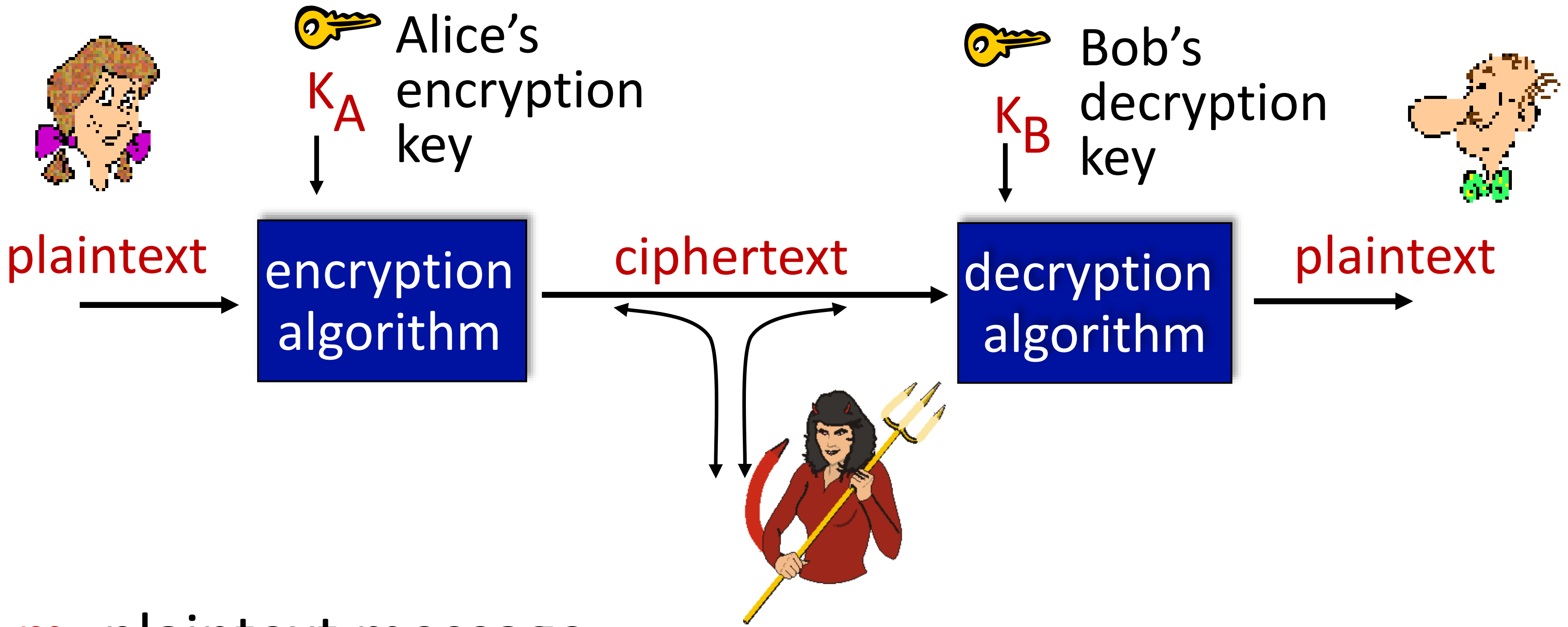
The language of cryptography



m: plaintext message

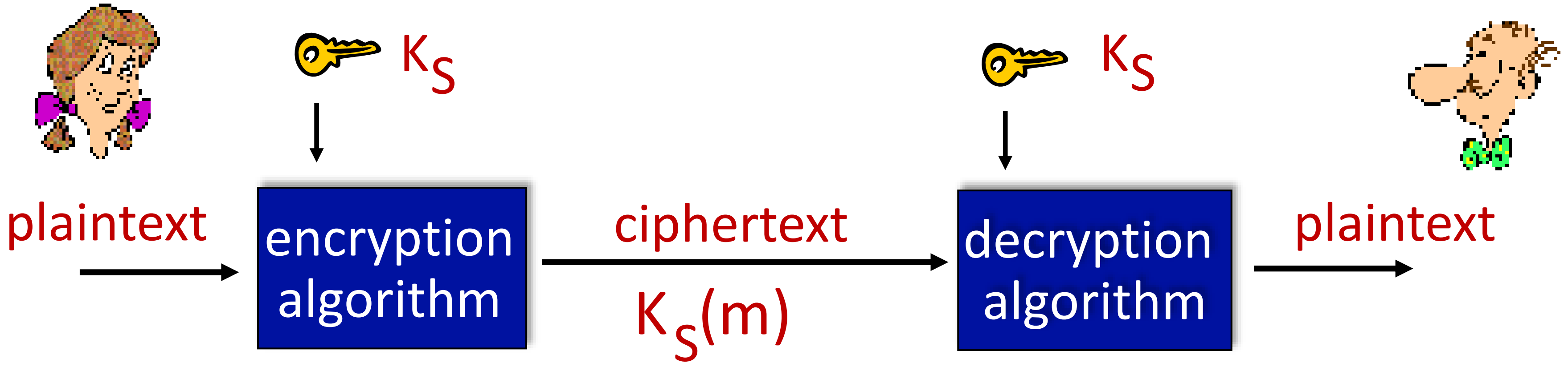
$K_A(m)$: ciphertext, encrypted with key K_A

$m = K_B(K_A(m))$

Breaking an encryption scheme

- cipher-text only attack: Trudy has ciphertext she can analyze
- two approaches:
  - brute force: search through all keys
  - statistical analysis
- known-plaintext attack: Trudy has plaintext corresponding to ciphertext
  - *e.g.*, in monoalphabetic cipher, Trudy determines pairings for a,l,i,c,e,b,o,
- chosen-plaintext attack: Trudy can get ciphertext for chosen plaintext

Symmetric key cryptography



symmetric key crypto: Bob and Alice share same (symmetric) key: K

- *e.g.,* key is knowing substitution pattern in mono alphabetic substitution cipher

Q: how do Bob and Alice agree on key value?

Simple encryption scheme

substitution cipher: substituting one thing for another

- monoalphabetic cipher: substitute one letter for another

```
 plaintext:   abcdefghijklmnopqrstuvwxyz

ciphertext:   mnbvcxzasdfghjklpoiuytrewq
```

e.g.:
```
 Plaintext: bob. i love you. alice
ciphertext: nkn. s gktc wky. mgsbc
```

Encryption key: mapping from set of 26 letters to set of 26 letters

A more sophisticated encryption approach

- n substitution ciphers, $M_1, M_2, \ldots, M_n$
- cycling pattern:
  - e.g., n=4: M_1,M_3,M_4,M_3,M_2; M_1,M_3,M_4,M_3,M_2; ..
- for each new plaintext symbol, use subsequent substitution pattern in cyclic pattern
  - dog: d from M_1, o from M_3, g from M_4

Encryption key: n substitution ciphers, and cyclic pattern

- key need not be just n-bit pattern

Symmetric key crypto: DES

DES: Data Encryption Standard

- US encryption standard [NIST 1993]
- 56-bit symmetric key, 64-bit plaintext input
- block cipher with cipher block chaining
- how secure is DES?
  - DES Challenge: 56-bit-key-encrypted phrase decrypted (brute force) in less than a day
  - no known good analytic attack
- making DES more secure:
  - 3DES: encrypt 3 times with 3 different keys

AES: Advanced Encryption Standard

- symmetric-key NIST standard, replaced DES (Nov 2001)
- processes data in 128 bit blocks
- 128, 192, or 256 bit keys
- brute force decryption (try each key) taking 1 sec on DES, takes 149 trillion years for AES

Public Key Cryptography

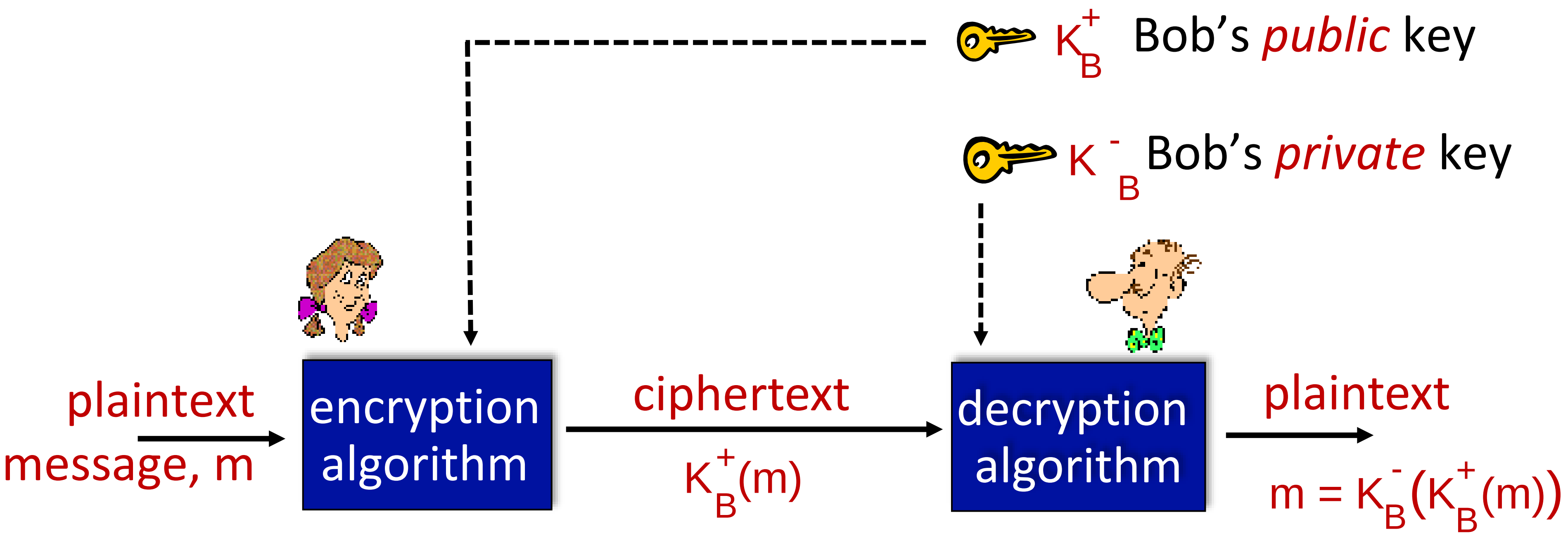
symmetric key crypto:

- requires sender, receiver know shared secret key
- Q: how to agree on key in first place (particularly if never "met")?

public key crypto

- *radically* different approach [Diffie-Hellman76, RSA78]
- sender, receiver do *not* share secret key
- *public* encryption key known to *all*
- *private* decryption key known only to receiver

Public Key Cryptography



Wow - public key cryptography revolutionized 2000-year-old (previously only symmetric key) cryptography!

- similar ideas emerged at roughly same time, independently in US and UK (classified)

Public key encryption algorithms

requirements:

1. need $K_B^+(\cdot)$ and $K_B^-(\cdot)$ such that

$$K_B^-(K_B^+(m)) = m$$

2. given public key K_B^+, it should be impossible to compute private key K_B^-

RSA: Rivest, Shamir, Adelson algorithm

Prerequisite: modular arithmetic

- x mod n = remainder of x when divide by n
- facts:

  [(a mod n) + (b mod n)] mod n = (a+b) mod n
  [(a mod n) - (b mod n)] mod n = (a-b) mod n
  [(a mod n) * (b mod n)] mod n = (a*b) mod n
- thus

  $(a \bmod n)^d \bmod n = a^d \bmod n$
- example: x=14, n=10, d=2:

  $(x \bmod n)^d \bmod n = 4^2 \bmod 10 = 6$

  $x^d = 14^2 = 196 \quad x^d \bmod 10 = 6$

RSA: getting ready

- message: just a bit pattern
- bit pattern can be uniquely represented by an integer number
- thus, encrypting a message is equivalent to encrypting a number

example:

- m= 10010001. This message is uniquely represented by the decimal number 145.
- to encrypt m, we encrypt the corresponding number, which gives a new number (the ciphertext).

RSA: Creating public/private key pair

1. choose two large prime numbers *p, q.* (e.g., 1024 bits each)

2. compute $n = pq$, $z = (p-1)(q-1)$

3. choose e (with $e<n$) that has no common factors with z (e, z are "relatively prime").

4. choose d such that $ed-1$ is exactly divisible by z. (in other words: $ed \bmod z = 1$).

5. *public* key is $\underbrace{(n,e)}_{K_B^+}$. *private* key is $\underbrace{(n,d)}_{K_B^-}$.

RSA: encryption, decryption

0. given (n,e) and (n,d) as computed above

1. to encrypt message m ($<n$), compute

   $c = m^e \bmod n$

2. to decrypt received bit pattern, c, compute

   $m = c^d \bmod n$

magic happens! $m = (\underbrace{m^e \bmod n}_{c})^d \bmod n$

RSA example:

Bob chooses *p=5, q=7*. Then *n=35, z=24*.

e=5 (so *e, z* relatively prime).

d=29 (so *ed-1* exactly divisible by z).

encrypting 8-bit messages.

encrypt:	bit pattern	m	m^e	$c = m^e \bmod n$
	0000l000	12	24832	17
decrypt:	c	c^d		$m = c^d \bmod n$
	17	481968572106750915091411825223071697		12

Why does RSA work?

- must show that $c^d \bmod n = m$, where $c = m^e \bmod n$
- fact: for any x and y: $x^y \bmod n = x^{(y \bmod z)} \bmod n$
  - where $n= pq$ and $z = (p-1)(q-1)$
- thus,

$$
\begin{aligned}
c^d \bmod n &= (m^e \bmod n)^d \bmod n \\
&= m^{ed} \bmod n \\
&= m^{(ed \bmod z)} \bmod n \\
&= m^1 \bmod n \\
&= m
\end{aligned}
$$

RSA: another important property

The following property will be *very* useful later:

$$\underbrace{K_B^-(K_B^+(m))}_{\text{use public key first, followed by private key}} = m = \underbrace{K_B^+(K_B^-(m))}_{\text{use private key first, followed by public key}}$$

result is the same!

Why $K_B^-(K_B^+(m)) = m = K_B^+(K_B^-(m))$?

follows directly from modular arithmetic:

$$
\begin{aligned}
(m^e \bmod n)^d \bmod n &= m^{ed} \bmod n \\
&= m^{de} \bmod n \\
&= (m^d \bmod n)^e \bmod n
\end{aligned}
$$

Why is RSA secure?

- suppose you know Bob's public key (n,e). How hard is it to determine d?
- essentially need to find factors of n without knowing the two factors p and q
  - fact: factoring a big number is hard

RSA in practice: session keys

- exponentiation in RSA is computationally intensive
- DES is at least 100 times faster than RSA
- use public key crypto to establish secure connection, then establish second key – symmetric session key – for encrypting data

session key, K_S

- Bob and Alice use RSA to exchange a symmetric session key K_S
- once both have K_S, they use symmetric key cryptography

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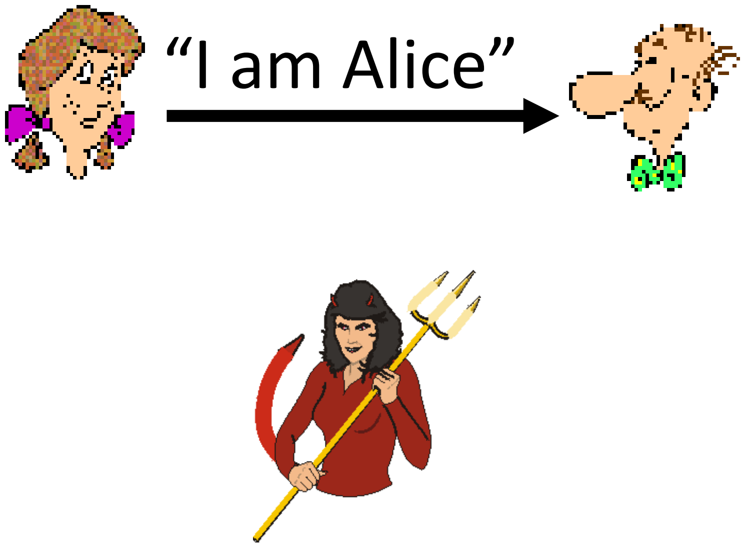
Authentication

Goal: Bob wants Alice to "prove" her identity to him

Protocol ap1.0: Alice says "I am Alice"



failure scenario??

Authentication

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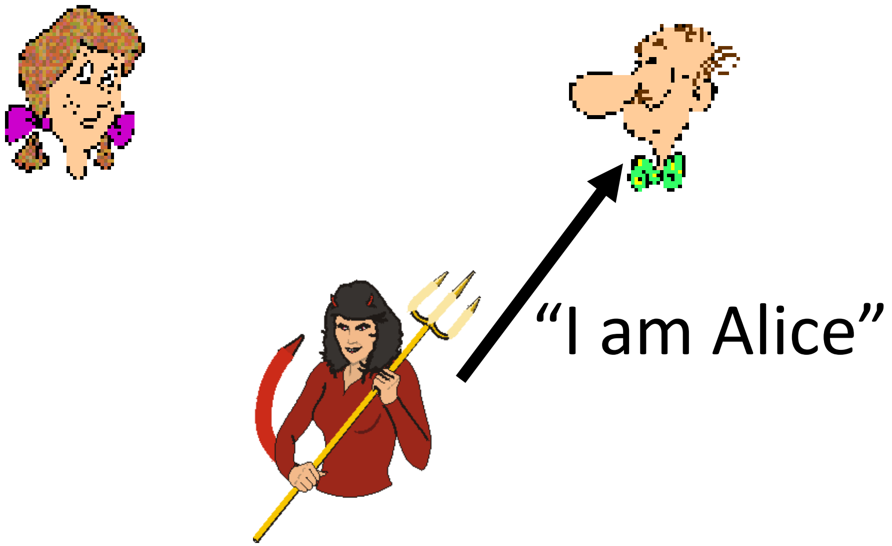


in a network, Bob can not "see" Alice, so Trudy simply declares herself to be Alice

Authentication: another try

Goal: Bob wants Alice to "prove" her identity to him

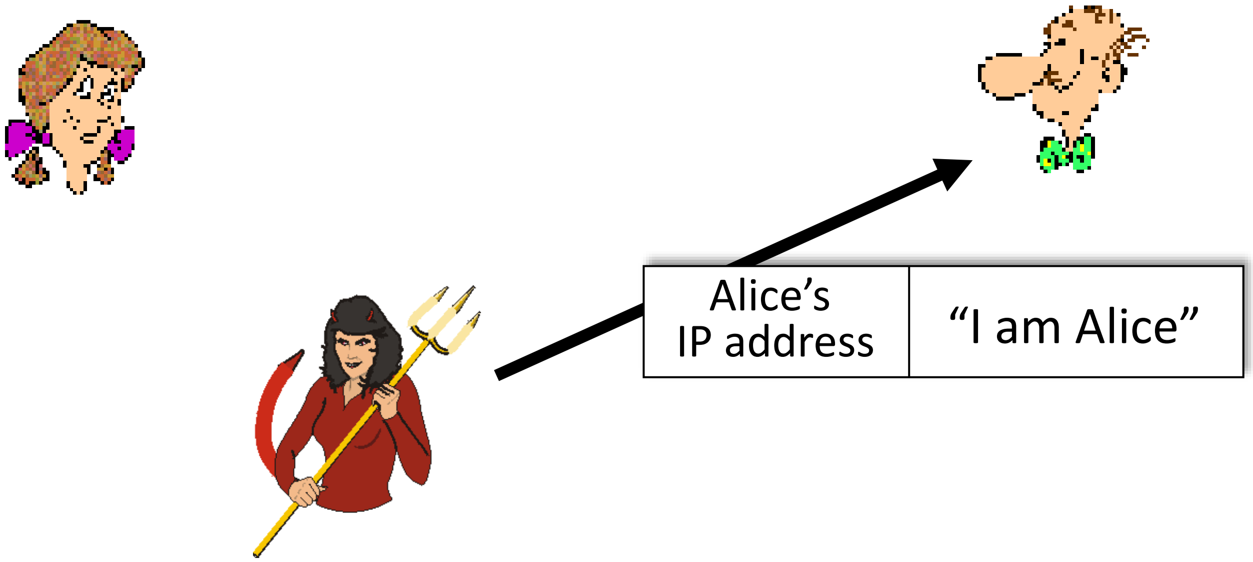
Protocol ap2.0: Alice says "I am Alice" in an IP packet containing her source IP address

failure scenario??

Authentication: another try

Goal: Bob wants Alice to “prove” her identity to him

Protocol ap2.0: Alice says “I am Alice” in an IP packet containing her source IP address

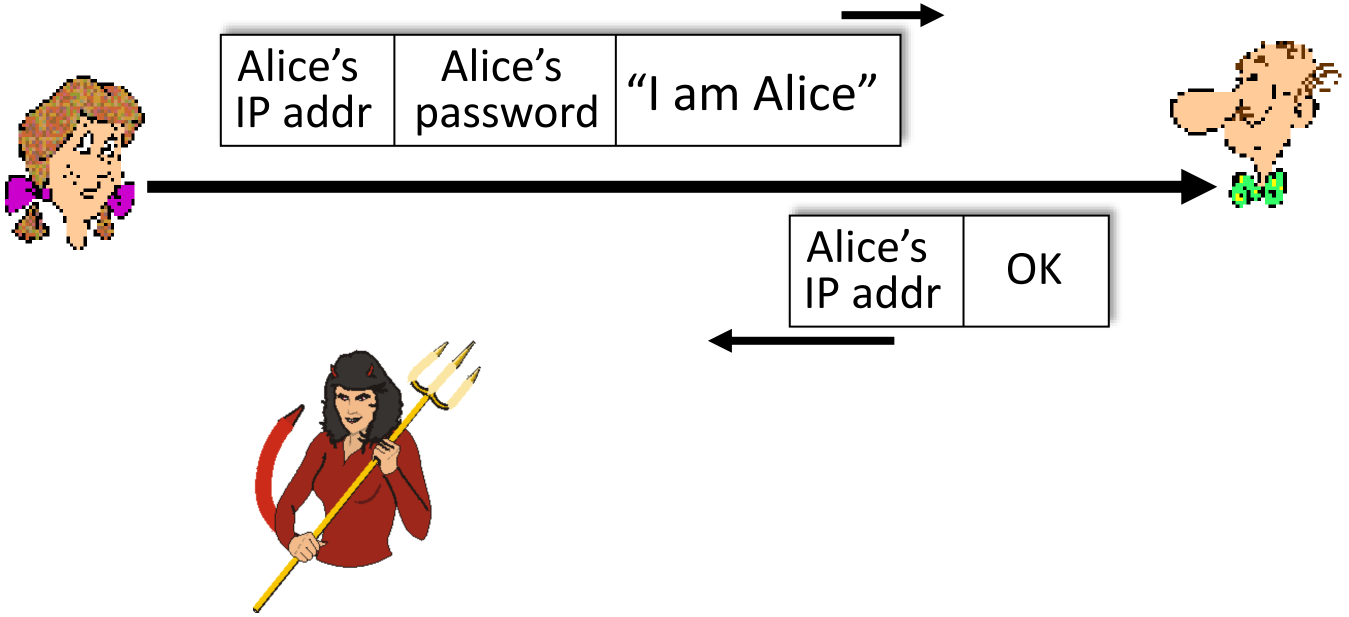


Trudy can create a packet “spoofing” Alice’s address

Authentication: a third try

Goal: Bob wants Alice to “prove” her identity to him

Protocol ap3.0: Alice says “I am Alice” and sends her secret password to “prove” it.



failure scenario??

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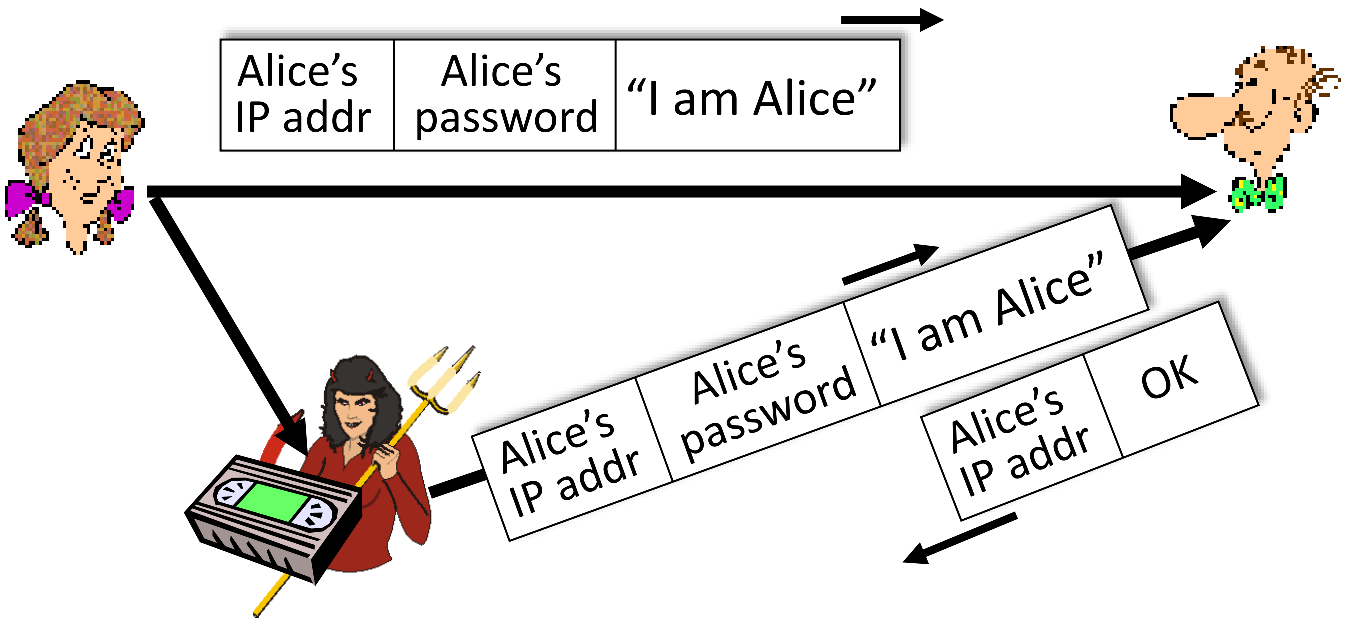
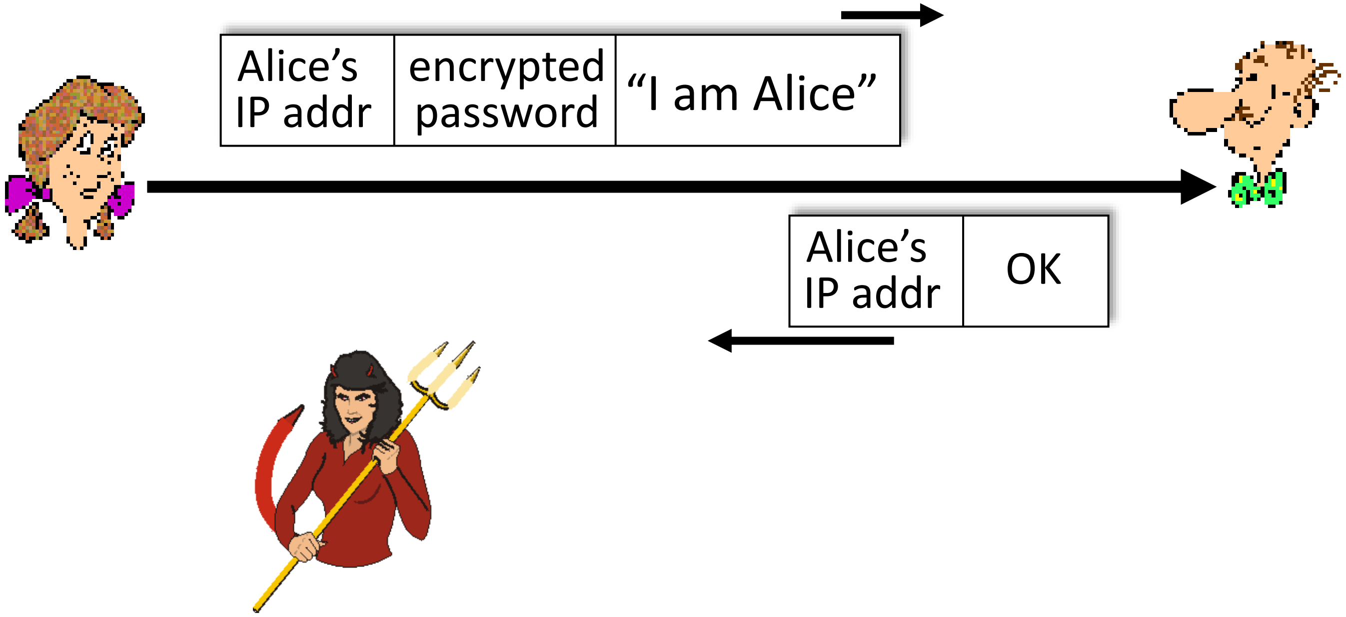


playback attack: Trudy records Alice's packet and later plays it back to Bob

Authentication: a modified third try

Goal: Bob wants Alice to “prove” her identity to him

Protocol ap3.0: Alice says “I am Alice” and sends her encrypted secret password to “prove” it.



failure scenario??

Authentication: a modified third try

Goal: Bob wants Alice to "prove" her identity to him

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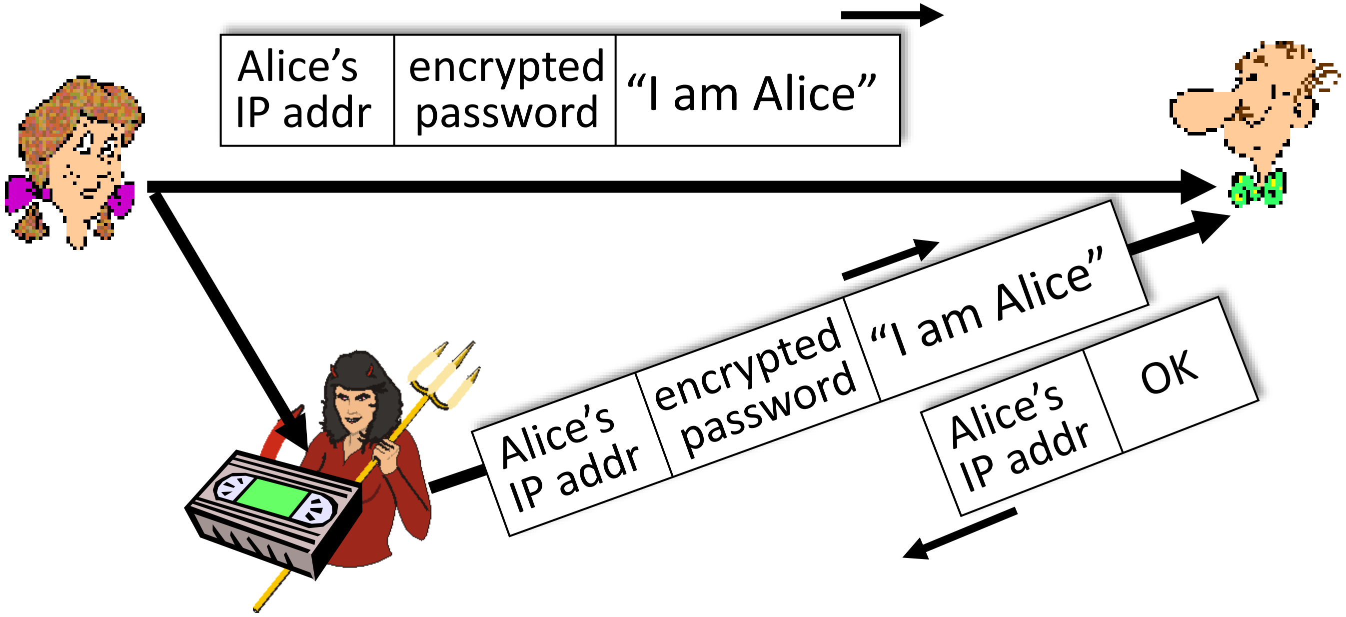


playback attack still works: Trudy records Alice's packet and later plays it back to Bob

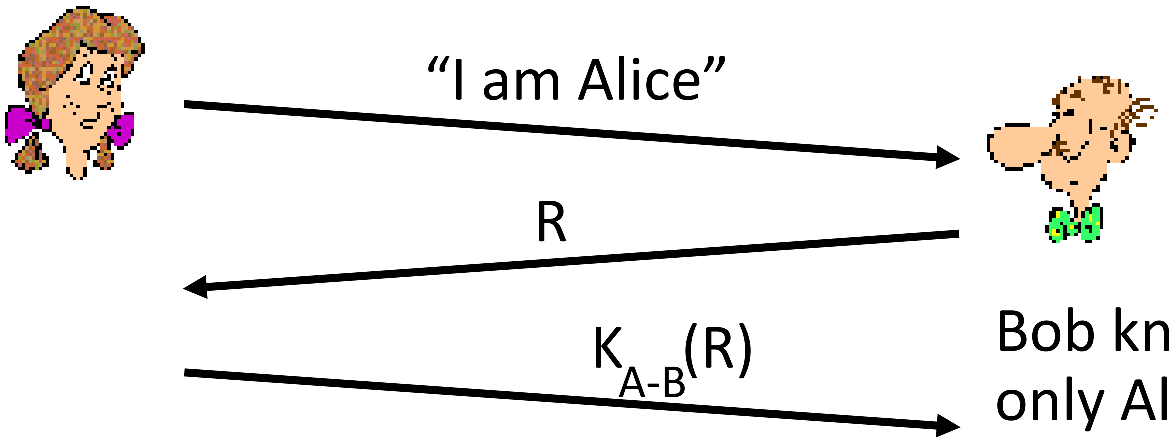
Authentication: a fourth try

Goal: avoid playback attack

nonce: number (R) used only once-in-a-lifetime

protocol ap4.0: to prove Alice "live", Bob sends Alice nonce, R

- Alice must return R, encrypted with shared secret key



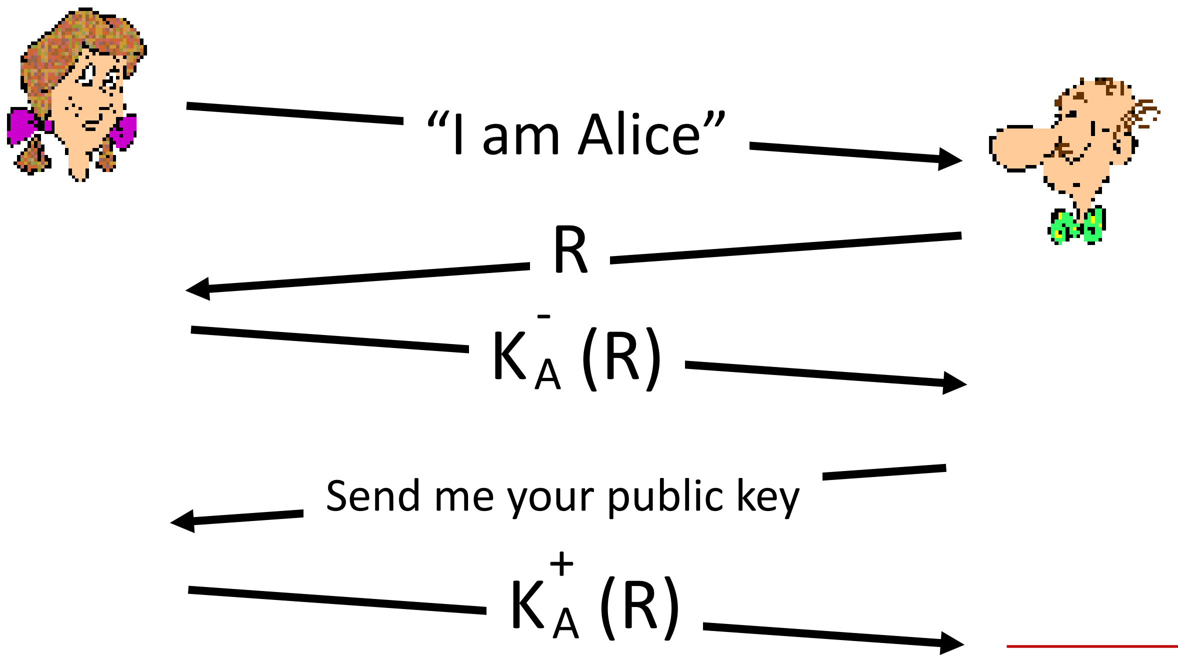
Bob know Alice is live, and only Alice knows key to encrypt nonce, so it must be Alice!

Failures, drawbacks?

Authentication: ap5.0

ap4.0 requires shared symmetric key - can we authenticate using public key techniques?

ap5.0: use nonce, public key cryptography



Bob computes

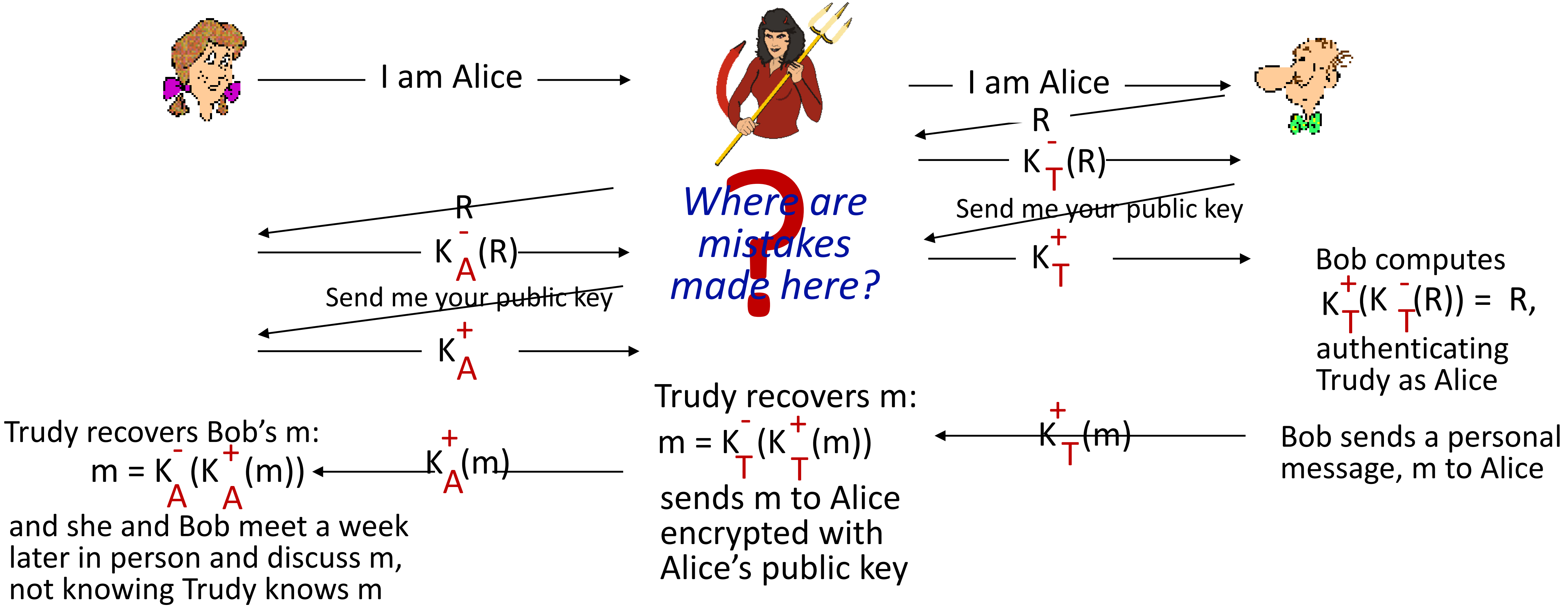
$K_A^+(K_A^-(R)) = R$

and knows only Alice could have the private key, that encrypted R such that

$K_A^+(K_A^-(R)) = R$

Authentication: ap5.0 – there's still a flaw!

man (or woman) in the middle attack: Trudy poses as Alice (to Bob) and as Bob (to Alice)



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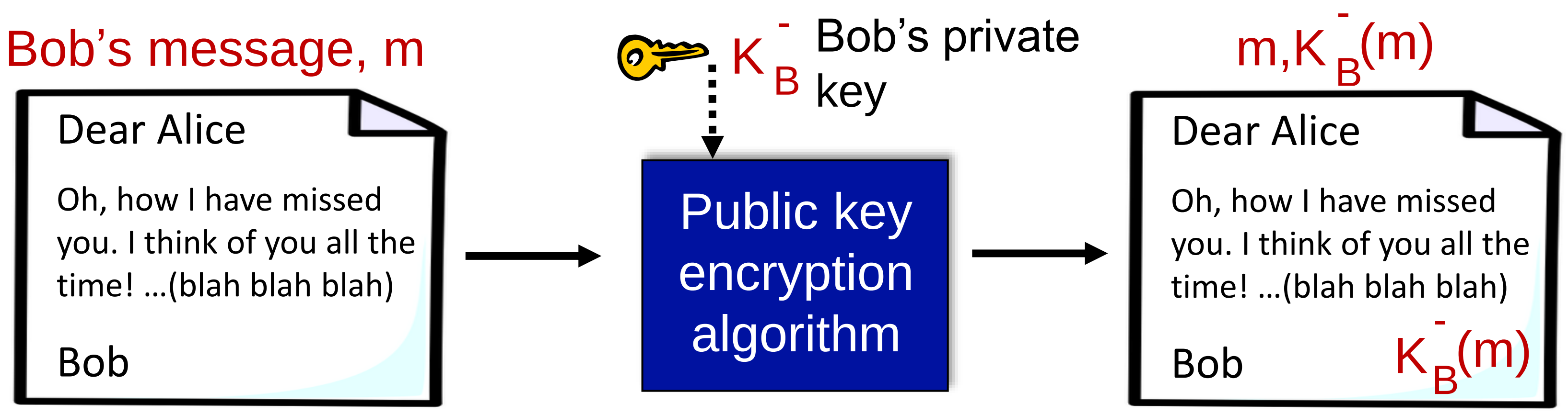
Digital signatures

cryptographic technique analogous to hand-written signatures:

- sender (Bob) digitally signs document: he is document owner/creator.
- *verifiable, nonforgeable:* recipient (Alice) can prove to someone that Bob, and no one else (including Alice), must have signed document
- simple digital signature for message m:
  - Bob signs m by encrypting with his private key K_B, creating "signed" message, $K_B^-(m)$



Digital signatures

- suppose Alice receives msg m, with signature: m, $K_B^-(m)$
- Alice verifies m signed by Bob by applying Bob's public key K_B^+ to $K_B^-(m)$ then checks $K_B^+(K_B^-(m)) = m$.
- If $K_B^+(K_B^-(m)) = m$, whoever signed m must have used Bob's private key

Alice thus verifies that:

- Bob signed m
- no one else signed m
- Bob signed m and not m'

non-repudiation:

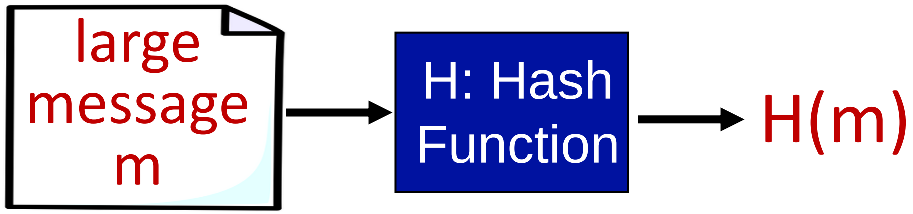
✓ Alice can take m, and signature $K_B^-(m)$ to court and prove that Bob signed m

Message digests

computationally expensive to public-key-encrypt long messages

goal: fixed-length, easy- to-compute digital "fingerprint"

- apply hash function H to *m*, get fixed size message digest, *H(m)*



Hash function properties:

- many-to-1
- produces fixed-size msg digest (fingerprint)
- given message digest *x*, computationally infeasible to find *m* such that *x = H(m)*

Internet checksum: poor crypto hash function

Internet checksum has some properties of hash function:

- produces fixed length digest (16-bit sum) of message
- is many-to-one

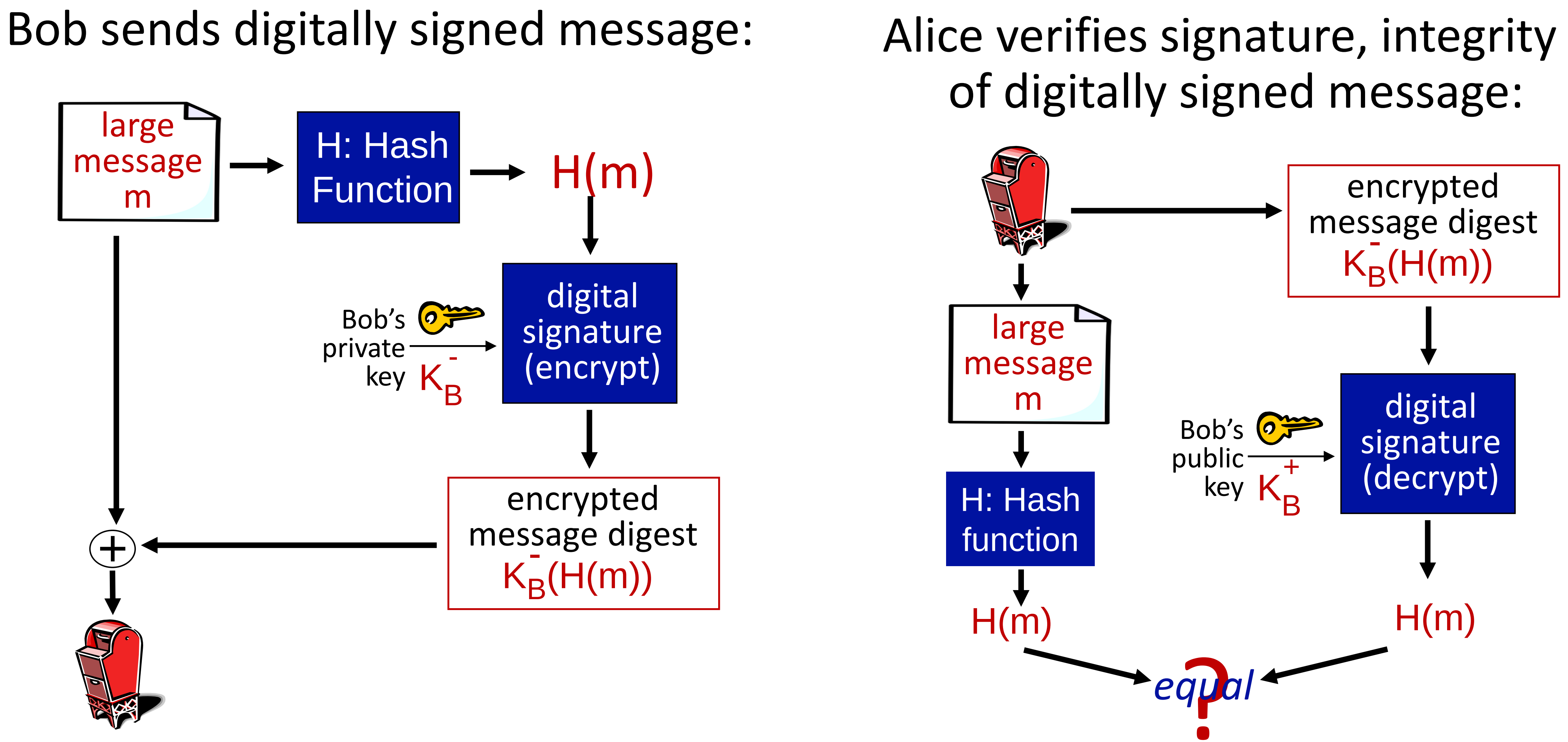
but given message with given hash value, it is easy to find another message with same hash value:

message	ASCII format
I O U 1	49 4F 55 31
0 0 . 9	30 30 2E 39
9 B O B	39 42 D2 42
	B2 C1 D2 AC

message	ASCII format
I O U <u>9</u>	49 4F 55 <u>39</u>
0 0 . <u>1</u>	30 30 2E <u>31</u>
9 B O B	39 42 D2 42
	B2 C1 D2 AC

different messages but identical checksums!

Digital signature = signed message digest

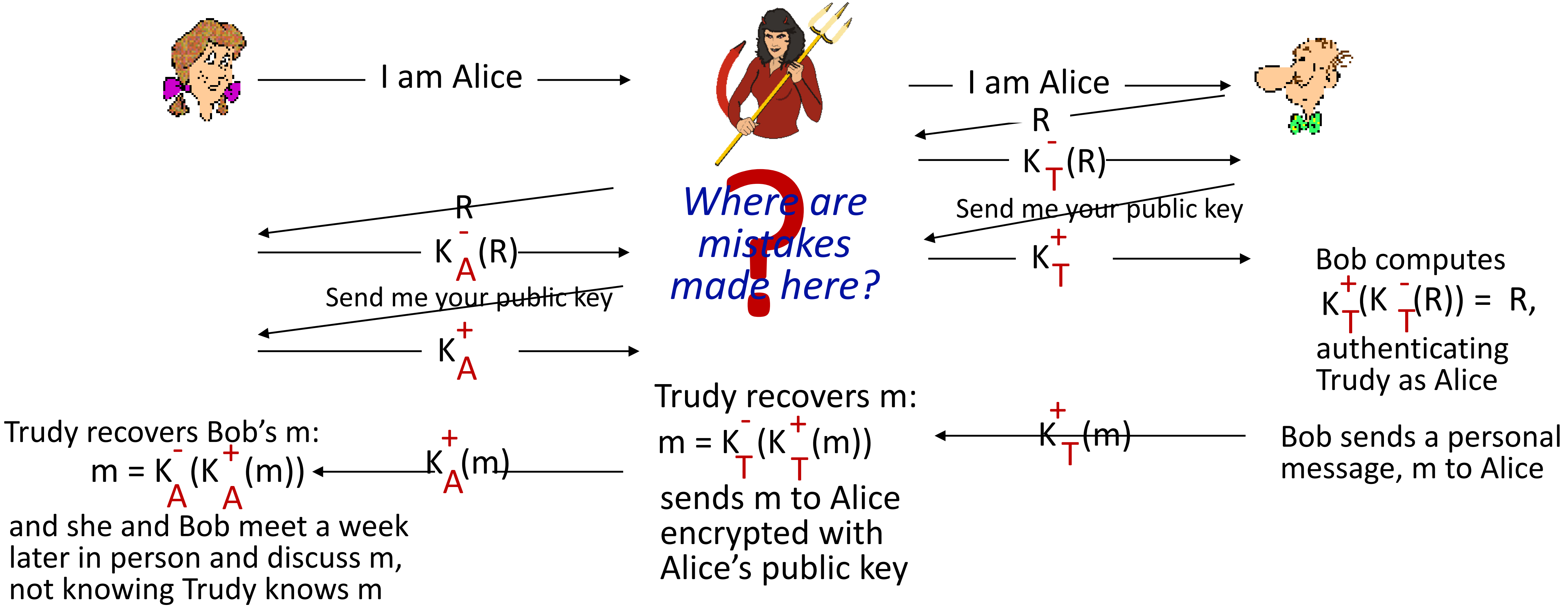


Hash function algorithms

- MD5 hash function widely used (RFC 1321)
  - computes 128-bit message digest in 4-step process.
  - arbitrary 128-bit string x, appears difficult to construct msg m whose MD5 hash is equal to x
- SHA-1 is also used
  - US standard [NIST, FIPS PUB 180-1]
  - 160-bit message digest

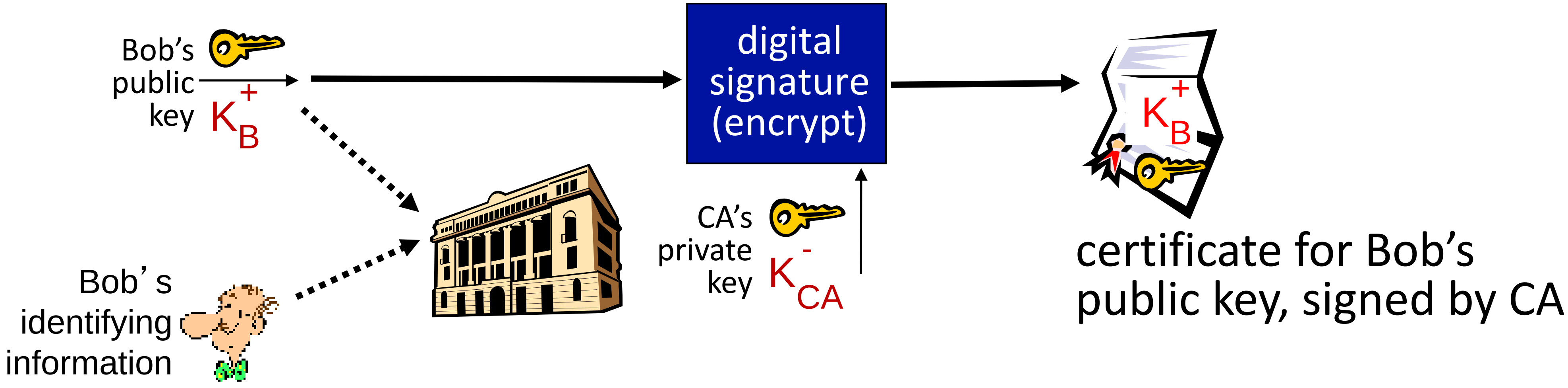
Authentication: ap5.0 – let's fix it!!

Recall the problem: Trudy poses as Alice (to Bob) and as Bob (to Alice)



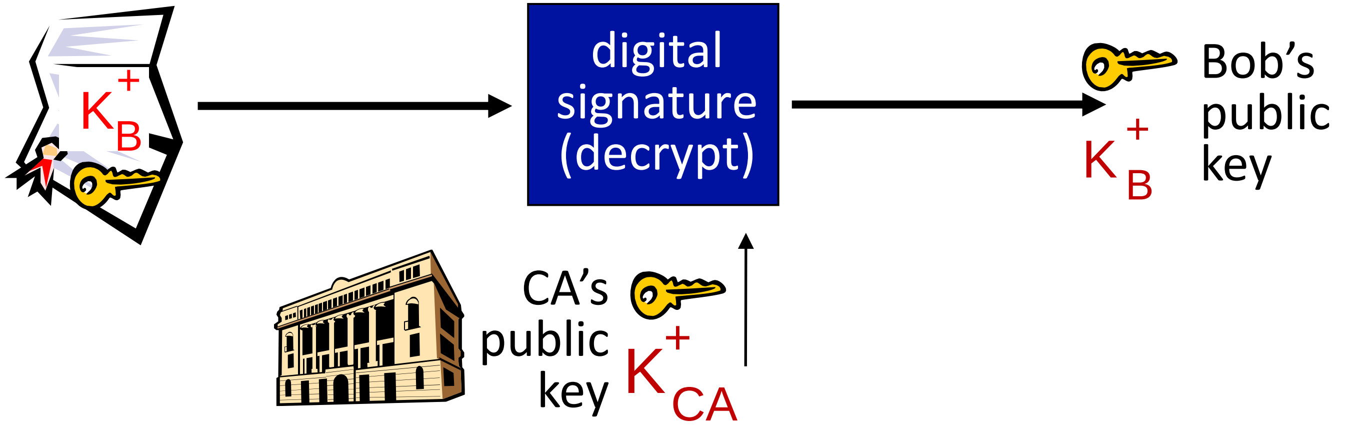
Public key Certification Authorities (CA)

- certification authority (CA): binds public key to particular entity, E
- entity (person, website, router) registers its public key with CE provides "proof of identity" to CA
  - CA creates certificate binding identity E to E's public key
  - certificate containing E's public key digitally signed by CA: CA says "this is E's public key"



Public key Certification Authorities (CA)

- when Alice wants Bob's public key:
  - gets Bob's certificate (Bob or elsewhere)
  - apply CA's public key to Bob's certificate, get Bob's public key

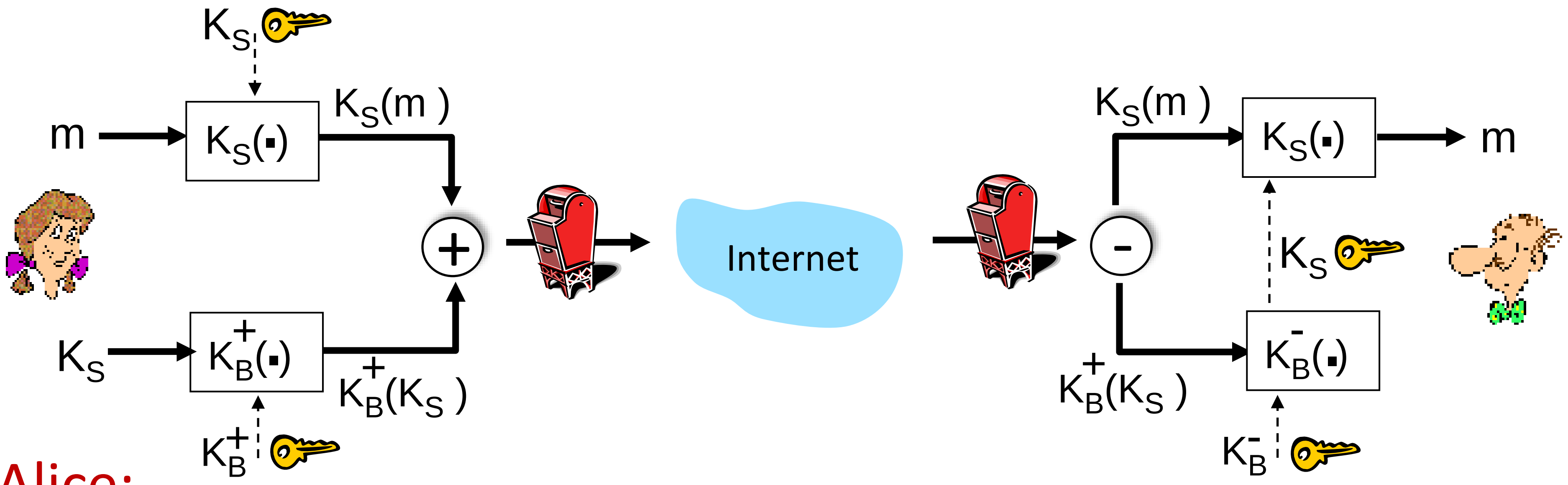


Chapter 8 outline

- What is network security?
- Principles of cryptography
- Authentication, message integrity
- Securing e-mail
- Securing TCP connections: TLS
- Network layer security: IPsec
- Security in wireless and mobile networks
- Operational security: firewalls and IDS

Secure e-mail: confidentiality

Alice wants to send *confidential* e-mail, m, to Bob.

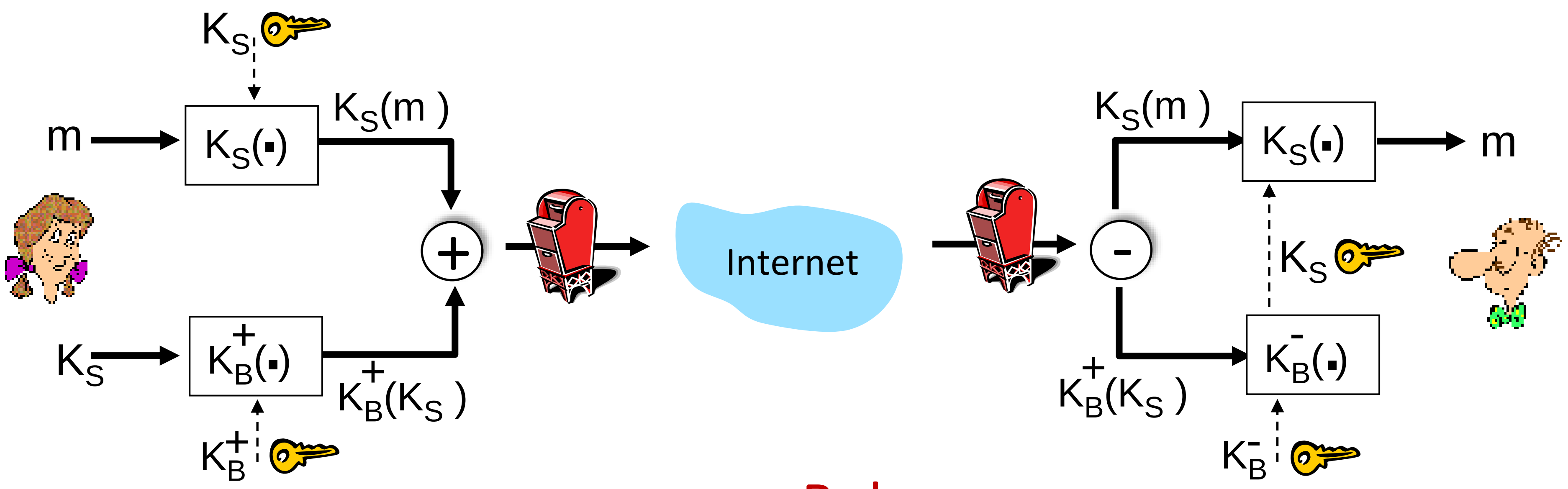


Alice:

- generates random *symmetric* private key, K_S
- encrypts message with K_S (for efficiency)
- also encrypts K_S with Bob's public key
- sends both $K_S(m)$ and $K^+_B(K_S)$ to Bob

Secure e-mail: confidentiality (more)

Alice wants to send *confidential* e-mail, m, to Bob.

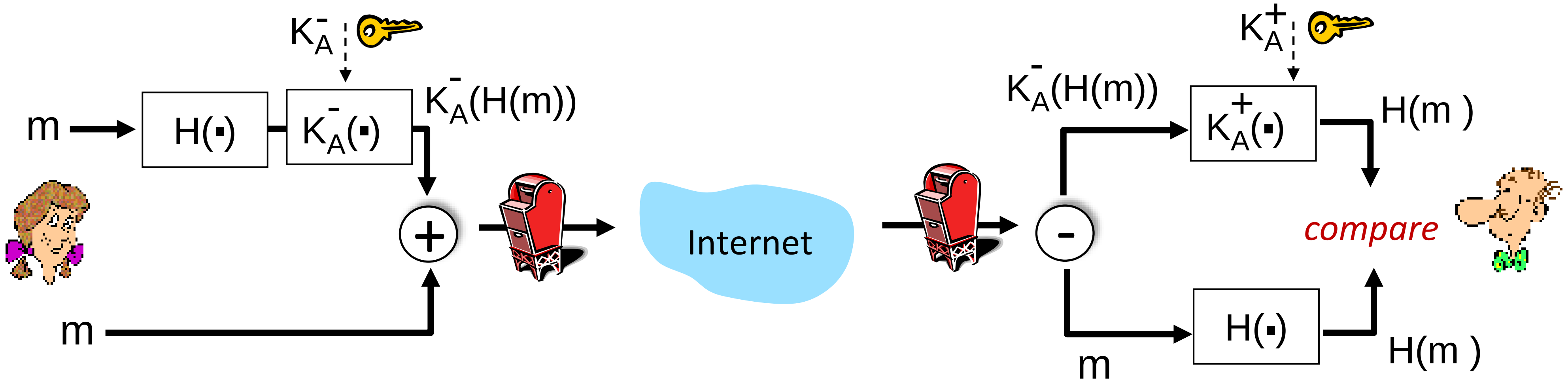


Bob:

- uses his private key to decrypt and recover K_S
- uses K_S to decrypt $K_S(m)$ to recover m

Secure e-mail: integrity, authentication

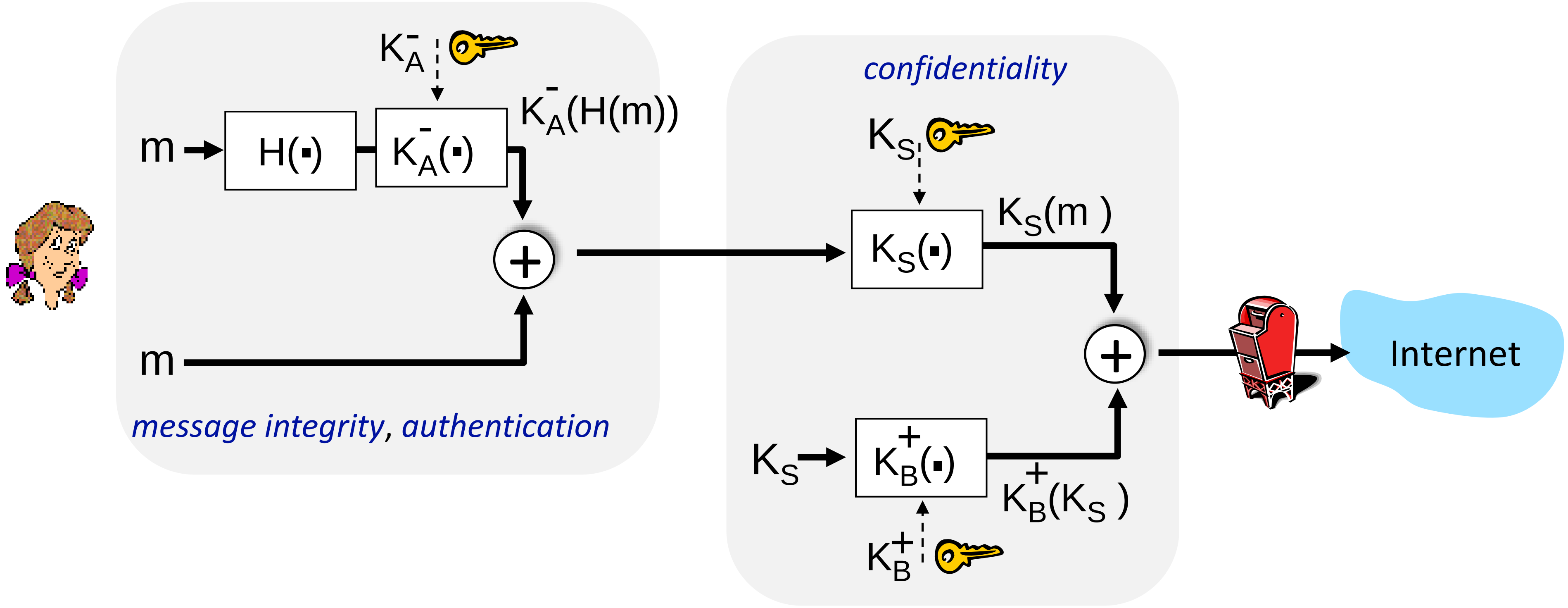
Alice wants to send m to Bob, with *message integrity*, *authentication*



- Alice digitally signs hash of her message with her private key, providing integrity and authentication
- sends both message (in the clear) and digital signature

Secure e-mail: integrity, authentication

Alice sends m to Bob, with *confidentiality, message integrity, authentication*



Alice uses three keys: her private key, Bob's public key, new symmetric key

What are Bob's complementary actions?

Chapter 8 outline

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Transport-layer security (TLS)

- widely deployed security protocol above the transport layer
  - supported by almost all browsers, web servers: https (port 443)
- provides:
  - confidentiality: via *symmetric encryption*
  - integrity: via *cryptographic hashing*
  - authentication: via *public key cryptography*

  all techniques we have studied!
- history:
  - early research, implementation: secure network programming, secure sockets
  - secure socket layer (SSL) deprecated [2015]
  - TLS 1.3: RFC 8846 [2018]

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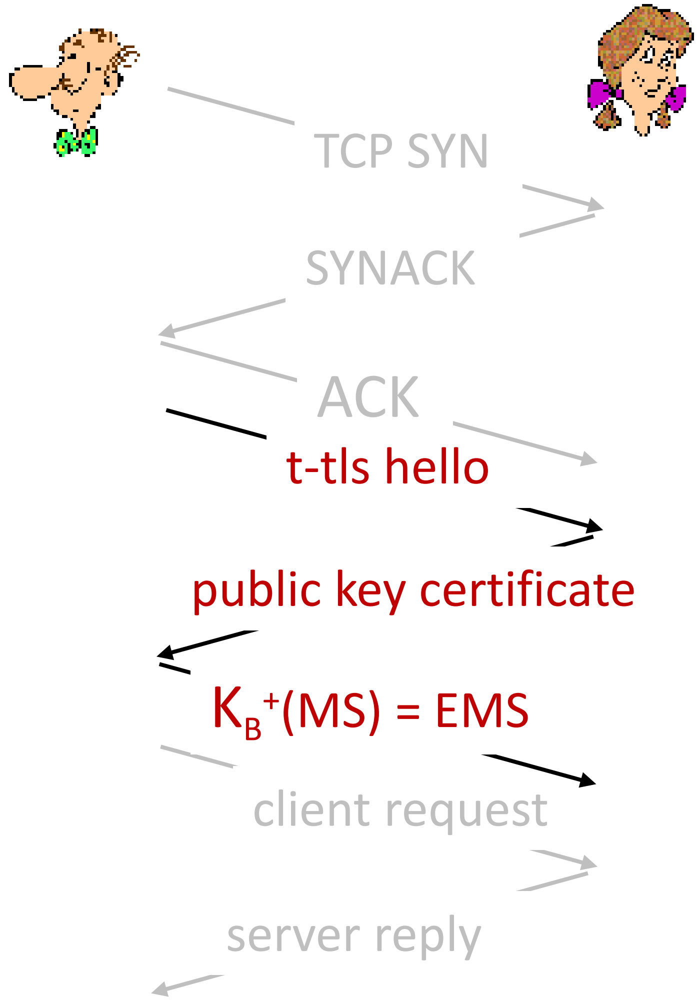
Transport-layer security: what's needed?

- let's *build* a toy TLS protocol, *t-tls,* to see what's needed!
- we've seen the "pieces" already:
  - handshake: Alice, Bob use their certificates, private keys to authenticate each other, exchange or create shared secret
  - key derivation: Alice, Bob use shared secret to derive set of keys
  - data transfer: stream data transfer: data as a series of records
    - not just one-time transactions
  - connection closure: special messages to securely close connection

t-tls: initial handshake



t-tls handshake phase:

- Bob establishes TCP connection with Alice
- Bob verifies that Alice is really Alice
- Bob sends Alice a master secret key (MS), used to generate all other keys for TLS session
- potential issues:
  - 3 RTT before client can start receiving data (including TCP handshake)

t-tls: cryptographic keys

- considered bad to use same key for more than one cryptographic function
  - different keys for message authentication code (MAC) and encryption
- four keys:
  - K_c : encryption key for data sent from client to server
  - M_c : MAC key for data sent from client to server
  - K_s : encryption key for data sent from server to client
  - M_s : MAC key for data sent from server to client
- keys derived from key derivation function (KDF)
  - takes master secret and (possibly) some additional random data to create new keys

t-tls: encrypting data

- recall: TCP provides data *byte stream* abstraction
- Q: can we encrypt data in-stream as written into TCP socket?
  - *A:* where would MAC go? If at end, no message integrity until all data received and connection closed!
  - *solution:* break stream in series of "records"
    - each client-to-server record carries a MAC, created using M_c
    - receiver can act on each record as it arrives
- t-tls record encrypted using symmetric key, K_c, passed to TCP:

$K_c($ | *length* | *data* | *MAC* | $)$

t-tls: encrypting data (more)

- possible attacks on data stream?
  - *re-ordering:* man-in middle intercepts TCP segments and reorders (manipulating sequence #s in unencrypted TCP header)
  - *replay*
- solutions:
  - use TLS sequence numbers (data, TLS-seq-# incorporated into MAC)
  - use nonce

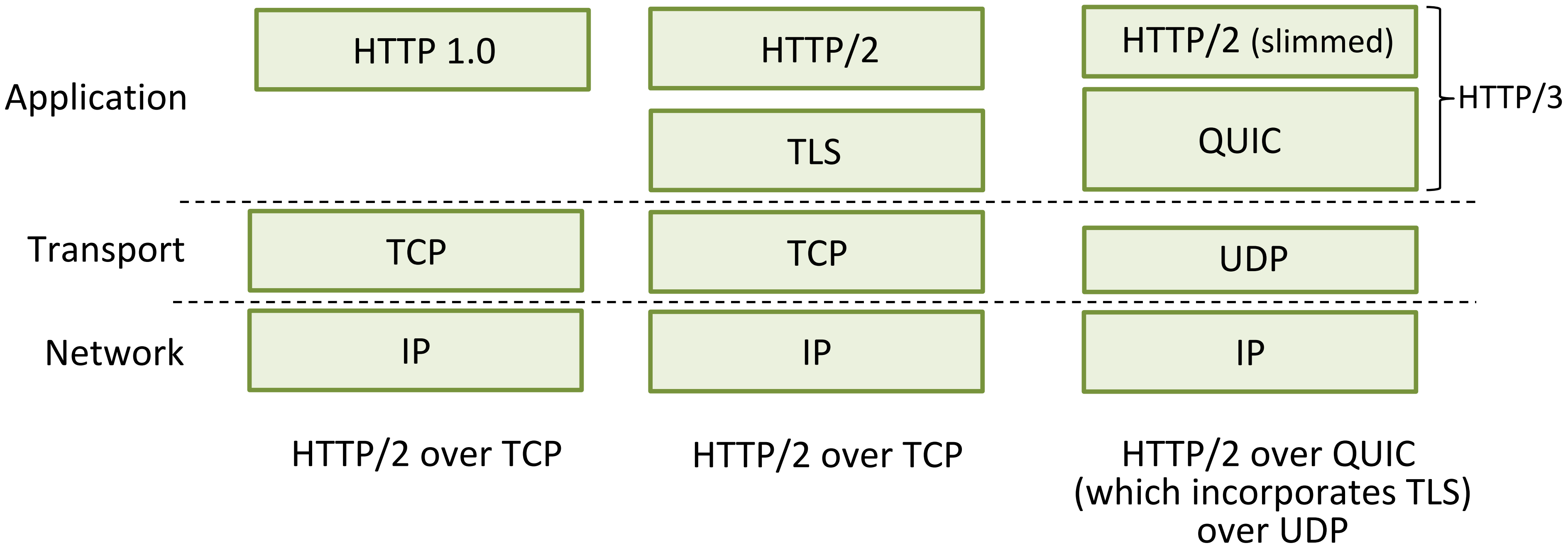
t-tls: connection close

- truncation attack:
  - attacker forges TCP connection close segment
  - one or both sides thinks there is less data than there actually is
- solution: record types, with one type for closure
  - type 0 for data; type 1 for close
- MAC now computed using data, type, sequence #

$K_c($ | *length* | *type* | *data* | *MAC* | $)$

Transport-layer security (TLS)

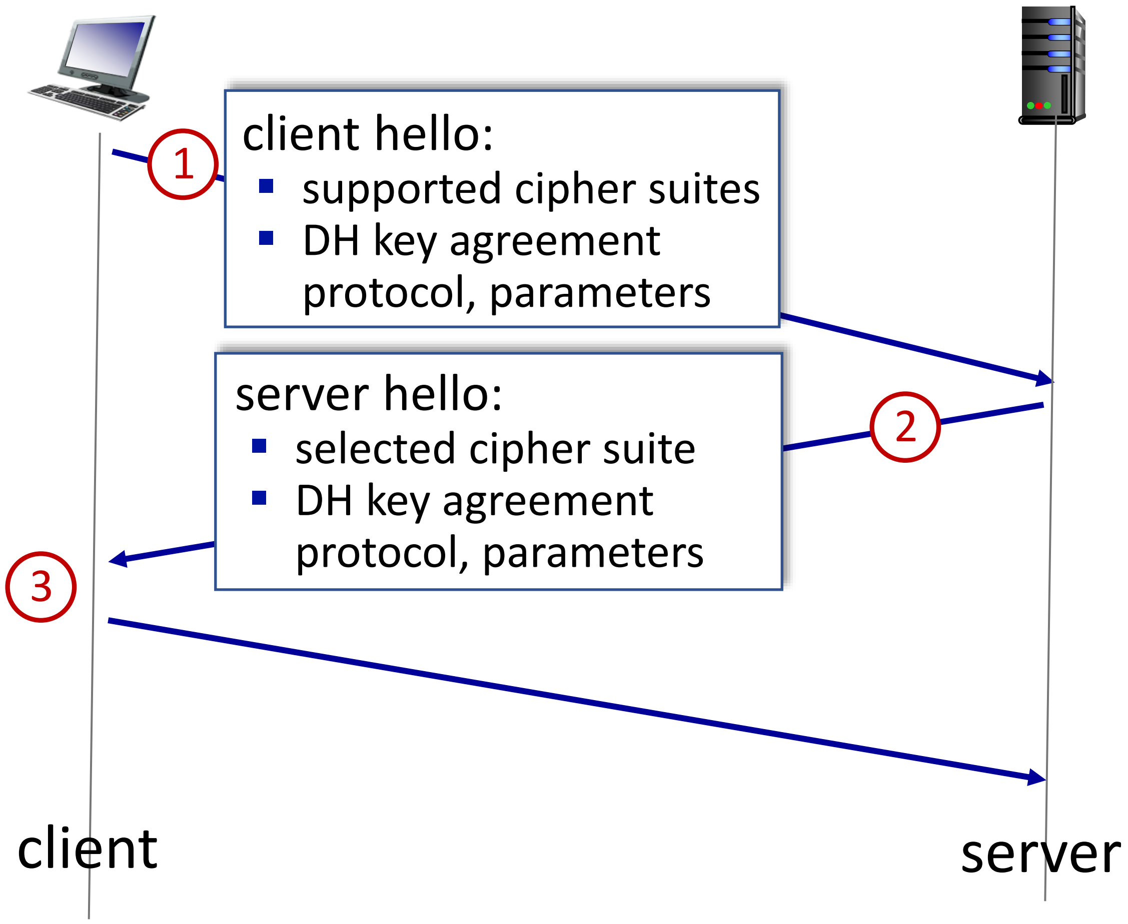
- TLS provides an API that *any* application can use
- an HTTP view of TLS:



TLS: 1.3 cipher suite

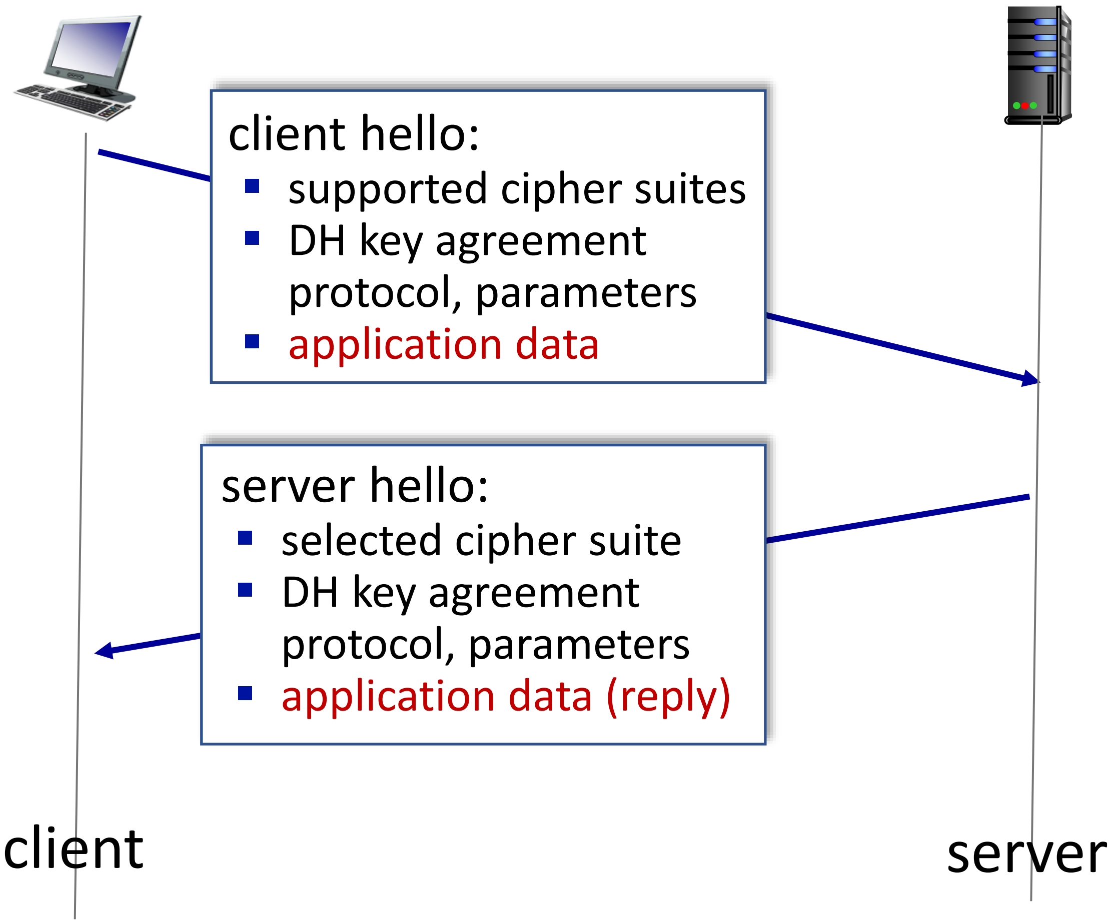
- "cipher suite": algorithms that can be used for key generation, encryption, MAC, digital signature
- TLS: 1.3 (2018): more limited cipher suite choice than TLS 1.2 (2008)
  - only 5 choices, rather than 37 choices
  - *requires* Diffie-Hellman (DH) for key exchange, rather than DH or RSA
  - combined encryption and authentication algorithm ("authenticated encryption") for data rather than serial encryption, authentication
    - 4 based on AES
  - HMAC uses SHA (256 or 284) cryptographic hash function

TLS 1.3 handshake: 1 RTT



1. client TLS hello msg:
   - *guesses* key agreement protocol, parameters
   - indicates cipher suites it supports
2. server TLS hello msg chooses
   - key agreement protocol, parameters
   - cipher suite
   - server-signed certificate
3. client:
   - checks server certificate
   - generates key
   - can now make application request (e.g.., HTTPS GET)

TLS 1.3 handshake: 0 RTT



- initial hello message contains encrypted application data!
  - "resuming" earlier connection between client and server
  - application data encrypted using "resumption master secret" from earlier connection
- vulnerable to replay attacks!
  - maybe OK for get HTTP GET or client requests not modifying server state

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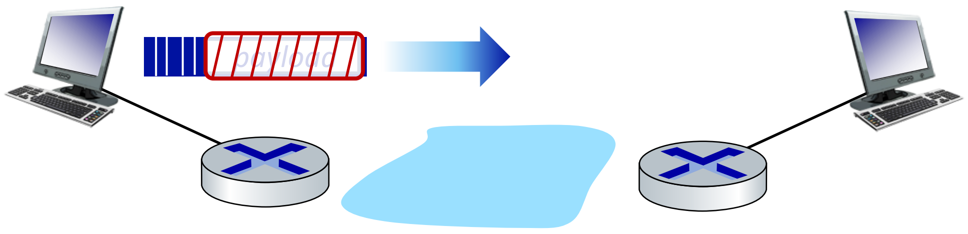
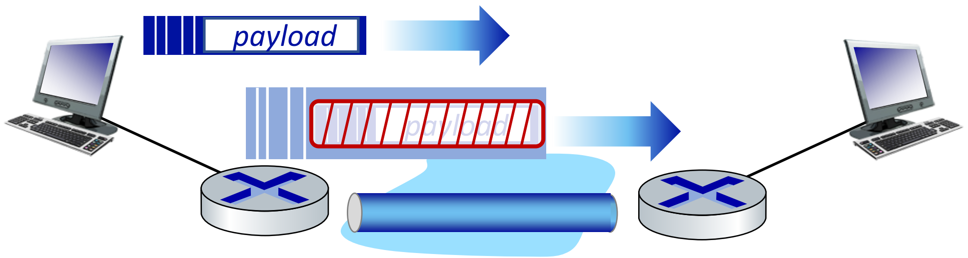
IP Sec

- provides datagram-level encryption, authentication, integrity
  - for both user traffic and control traffic (e.g., BGP, DNS messages)
- two "modes":



transport mode:

- *only* datagram *payload* is encrypted, authenticated



tunnel mode:

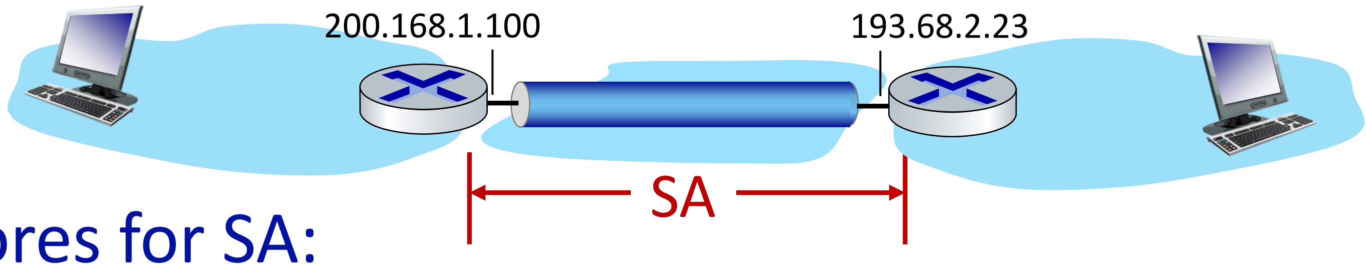
- entire datagram is encrypted, authenticated
- encrypted datagram encapsulated in new datagram with new IP header, tunneled to destination

Two IPsec protocols

- Authentication Header (AH) protocol [RFC 4302]
  - provides source authentication & data integrity but *not* confidentiality
- Encapsulation Security Protocol (ESP) [RFC 4303]
  - provides source authentication, data integrity, *and confidentiality*
  - more widely used than AH

Security associations (SAs)

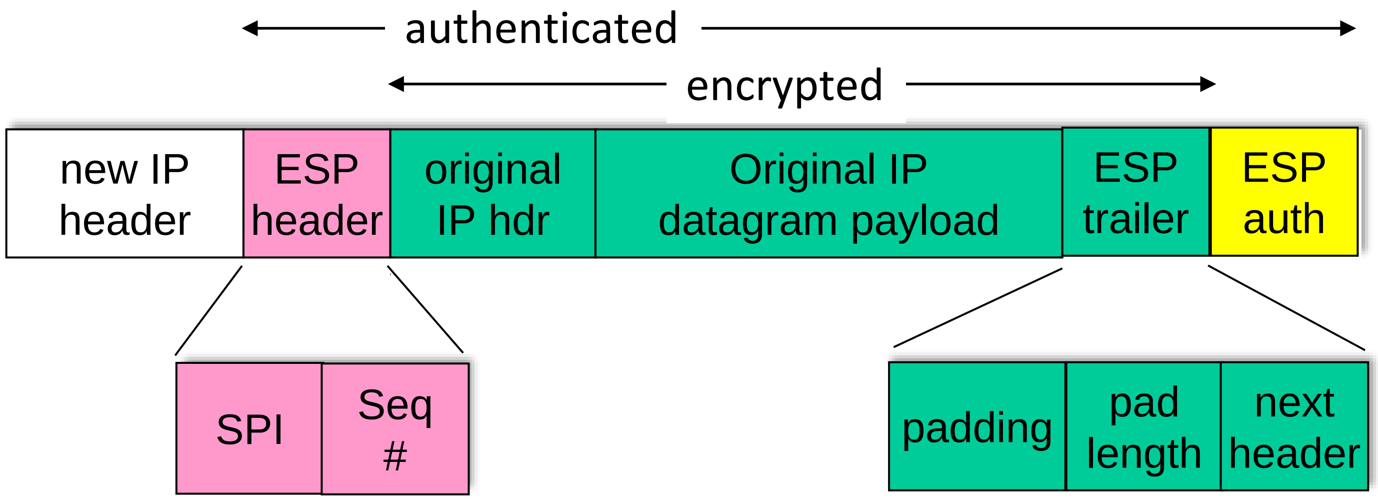
- before sending data, security association (SA) established from sending to receiving entity (directional)
- ending, receiving entitles maintain *state information* about SA
  - recall: TCP endpoints also maintain state info
  - IP is connectionless; IPsec is connection-oriented!



R1 stores for SA:

- 32-bit identifier: *Security Parameter Index (SPI)*
- origin SA interface (200.168.1.100)
- destination SA interface (193.68.2.23)
- type of encryption used
- encryption key
- type of integrity check used
- authentication key

IPsec datagram



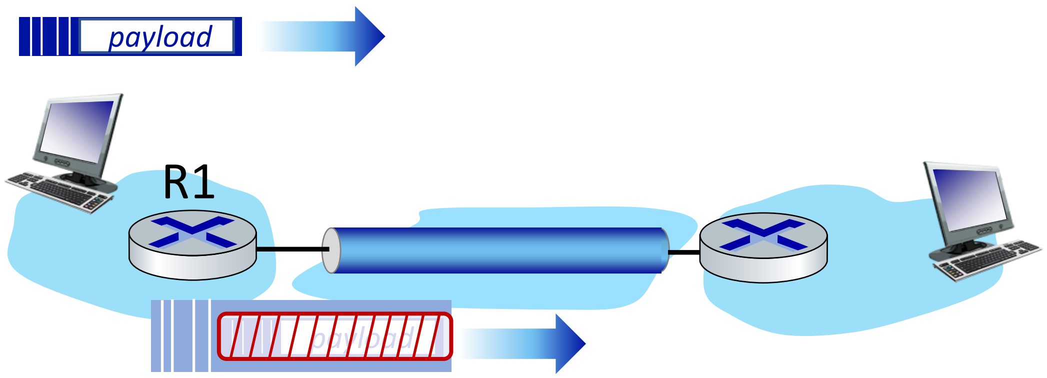
tunnel mode ESP

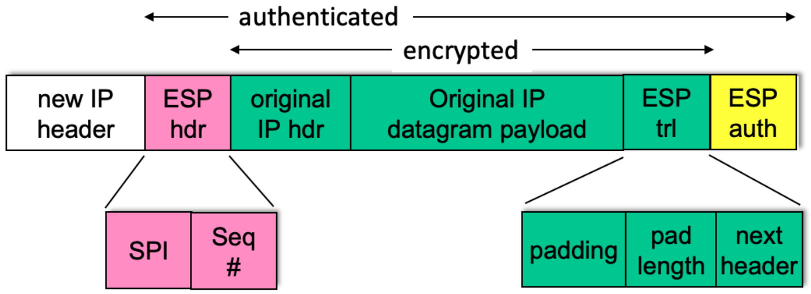
- ESP trailer: padding for block ciphers
- ESP header:
  - SPI, so receiving entity knows what to do
  - sequence number, to thwart replay attacks
- MAC in ESP auth field created with shared secret key

ESP tunnel mode: actions

at R1:

- appends ESP trailer to original datagram (which includes original header fields!)
- encrypts result using algorithm & key specified by SA
- appends ESP header to front of this encrypted quantity
- creates authentication MAC using algorithm and key specified in SA
- appends MAC forming *payload*
- creates new IP header, new IP header fields, addresses to tunnel endpoint





IPsec sequence numbers

- for new SA, sender initializes seq. # to 0
- each time datagram is sent on SA:
  - sender increments seq # counter
  - places value in seq # field
- goal:
  - prevent attacker from sniffing and replaying a packet
  - receipt of duplicate, authenticated IP packets may disrupt service
- method:
  - destination checks for duplicates
  - doesn't keep track of *all* received packets; instead uses a window

IPsec security databases

Security Policy Database (SPD)

- policy: for given datagram, sender needs to know if it should use IP sec
- policy stored in security policy database (SPD)
- needs to know which SA to use
  - may use: source and destination IP address; protocol number

SAD: "how" to do it

Security Assoc. Database (SAD)

- endpoint holds SA state in security association database (SAD)
- when sending IPsec datagram, R1 accesses SAD to determine how to process datagram
- when IPsec datagram arrives to R2, R2 examines SPI in IPsec datagram, indexes SAD with SPI, processing
- datagram accordingly.

SPD: "what" to do

Summary: IPsec services

Trudy sits somewhere between R1, R2. she doesn’t know the keys

- will Trudy be able to see original contents of datagram? How about source, dest IP address, transport protocol, application port?
- flip bits without detection?
- masquerade as R1 using R1’s IP address?
- replay a datagram?

IKE: Internet Key Exchange

- *previous examples:* manual establishment of IPsec SAs in IPsec endpoints:

  Example SA:

  SPI: 12345
  Source IP: 200.168.1.100
  Dest IP: 193.68.2.23
  Protocol: ESP
  Encryption algorithm: 3DES-cbc
  HMAC algorithm: MD5
  Encryption key: 0x7aeaca…
  HMAC key:0xc0291f…

- manual keying is impractical for VPN with 100s of endpoints
- instead use IPsec IKE (Internet Key Exchange)

IKE: PSK and PKI

- authentication (prove who you are) with either
  - pre-shared secret (PSK) or
  - with PKI (pubic/private keys and certificates).
- PSK: both sides start with secret
  - run IKE to authenticate each other and to generate IPsec SAs (one in each direction), including encryption, authentication keys
- PKI: both sides start with public/private key pair, certificate
  - run IKE to authenticate each other, obtain IPsec SAs (one in each direction).
  - similar with handshake in SSL.

IKE phases

- IKE has two phases
  - *phase 1:* establish bi-directional IKE SA
    - note: IKE SA different from IPsec SA
    - aka ISAKMP security association
  - *phase 2:* ISAKMP is used to securely negotiate IPsec pair of SAs
- phase 1 has two modes: aggressive mode and main mode
  - aggressive mode uses fewer messages
  - main mode provides identity protection and is more flexible

IPsec summary

- IKE message exchange for algorithms, secret keys, SPI numbers
- either AH or ESP protocol (or both)
  - AH provides integrity, source authentication
  - ESP protocol (with AH) additionally provides encryption
- IPsec peers can be two end systems, two routers/firewalls, or a router/firewall and an end system

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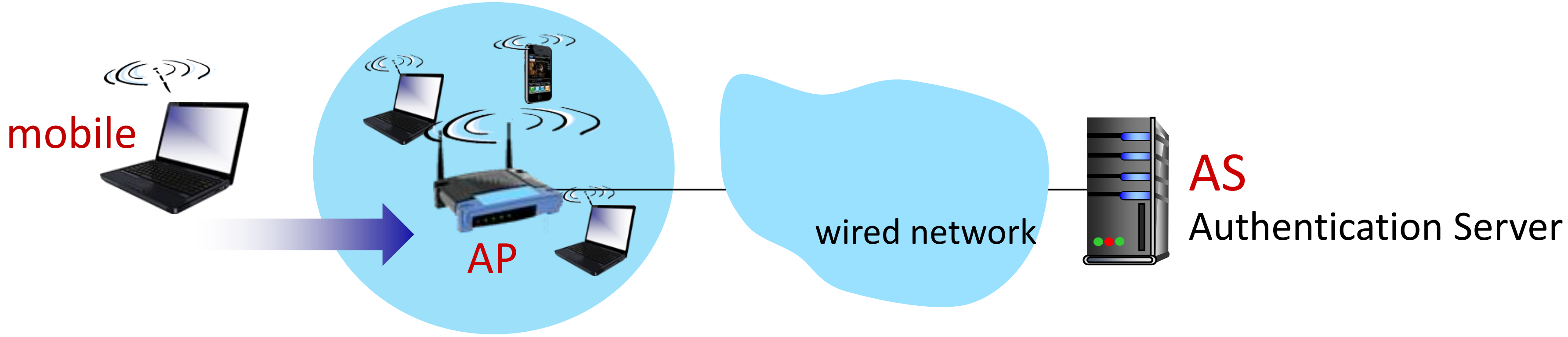
  - 4G/5G
- Operational security: firewalls and IDS

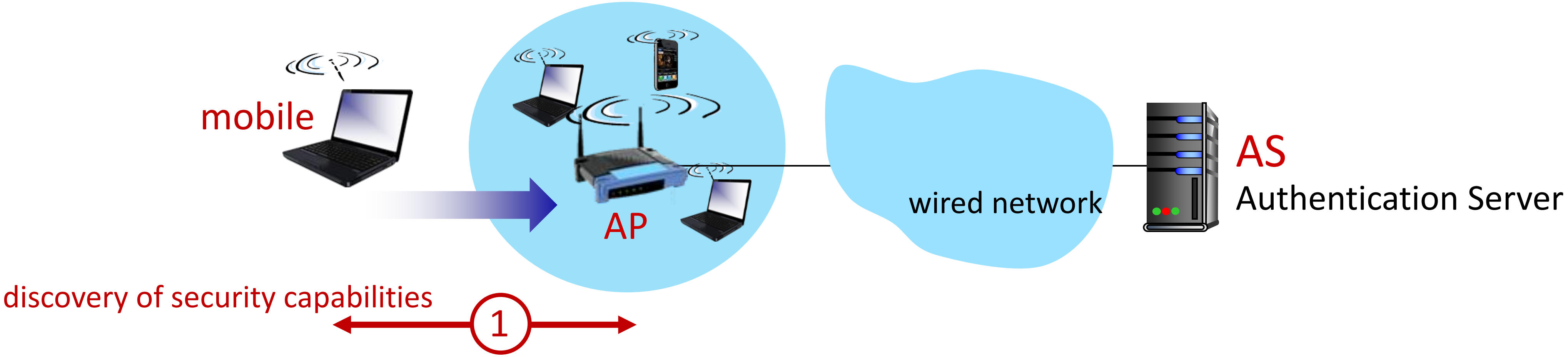
802.11: authentication, encryption



Arriving mobile must:

- associate with access point: (establish) communication over wireless link
- authenticate to network

802.11: authentication, encryption

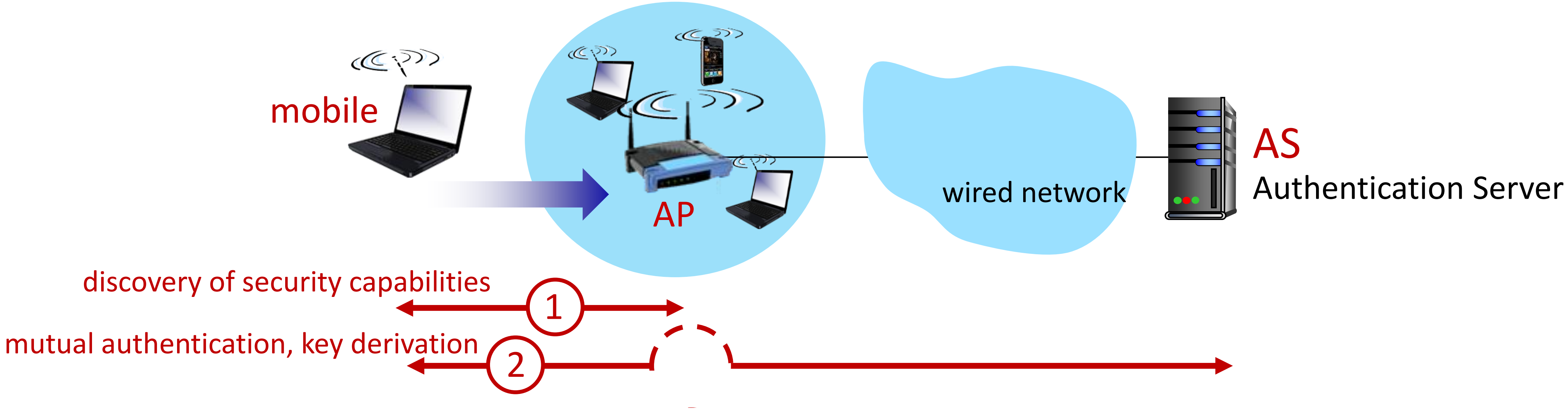


1 discovery of security capabilities:

- AP advertises its presence, forms of authentication and encryption provided
- device requests specific forms authentication, encryption desired

although device, AP already exchanging messages, device not yet authenticated, does not have encryption keys

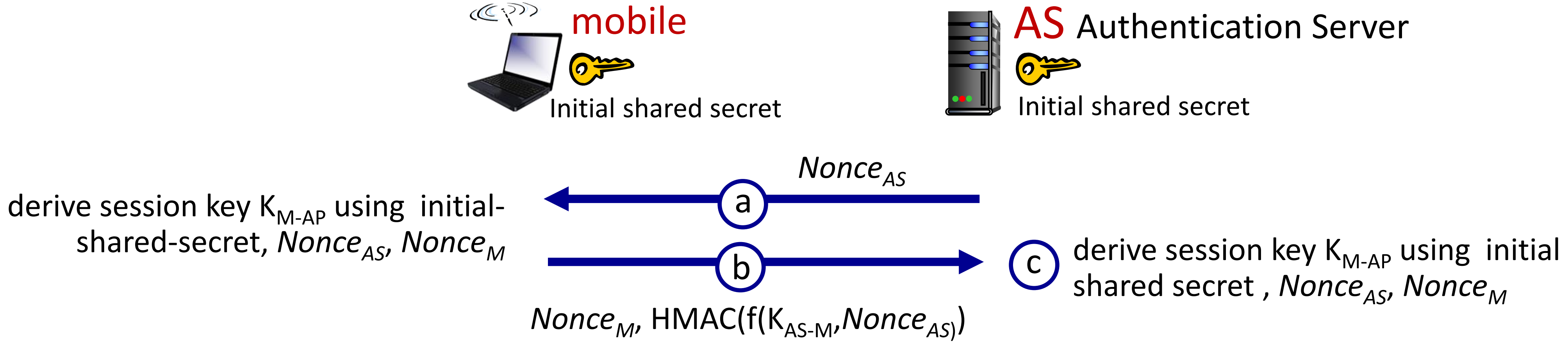
802.11: authentication, encryption



2 mutual authentication and shared symmetric key derivation:

- AS, mobile already have shared common secret (e.g., password)
- AS, mobile use shared secret, nonces (prevent relay attacks), cryptographic hashing (ensure message integrity) to authenticating each other
- AS, mobile derive symmetric session key

802.11: WPA3 handshake

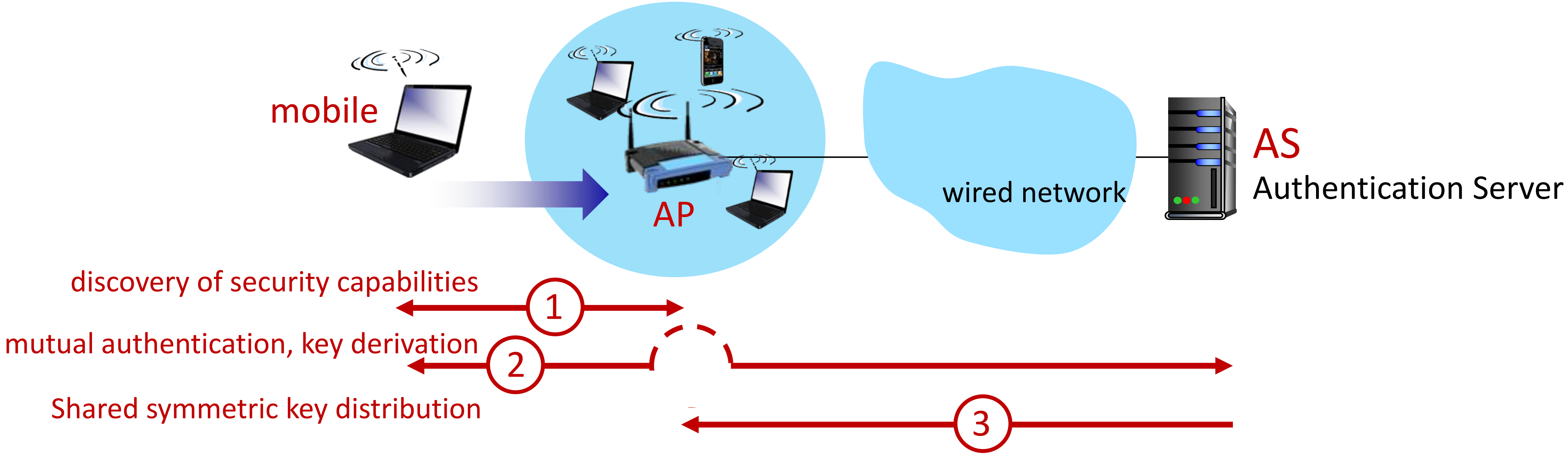


a AS generates $Nonce_{AS}$, sends to mobile

b mobile receives $Nonce_{AS}$
- generates $Nonce_M$
- generates symmetric shared session key $K_{M\text{-}AP}$ using $Nonce_{AS}$, $Nonce_M$, and initial shared secret
- sends $Nonce_M$, and HMAC-signed value using Nonce_{AS} and initial shared secret

c AS derives symmetric shared session key $K_{M\text{-}AP}$

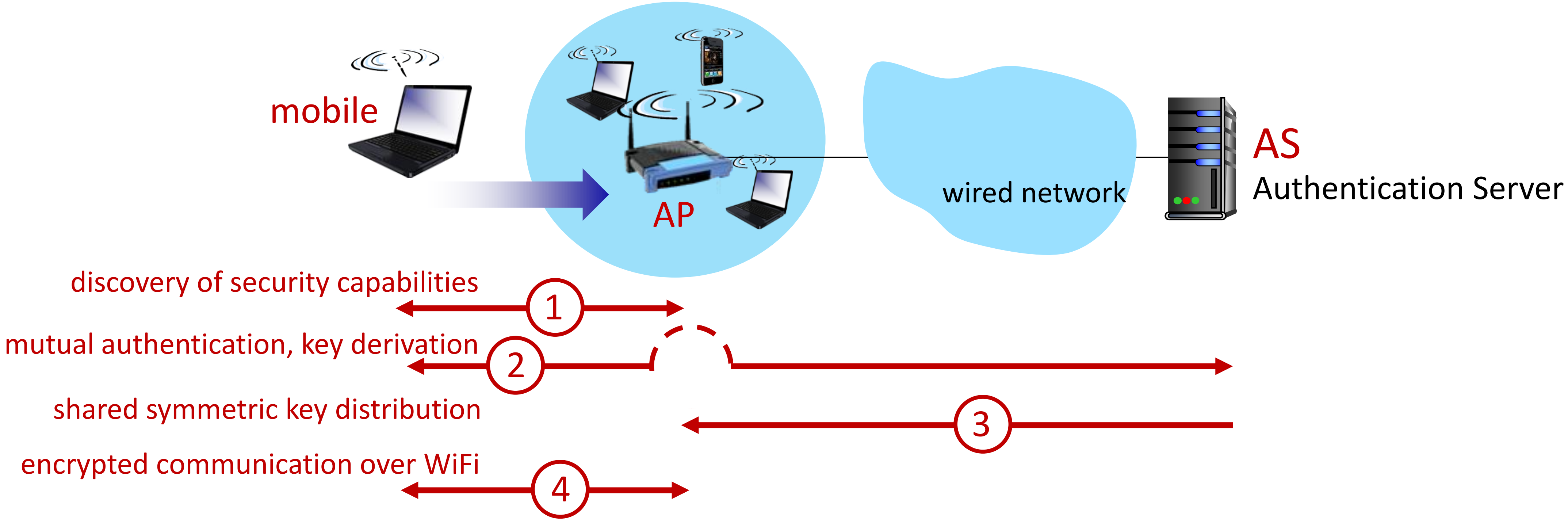
802.11: authentication, encryption



3 shared symmetric session key distribution (e.g., for AES encryption)

- same key derived at mobile, AS
- AS informs AP of the shared symmetric session

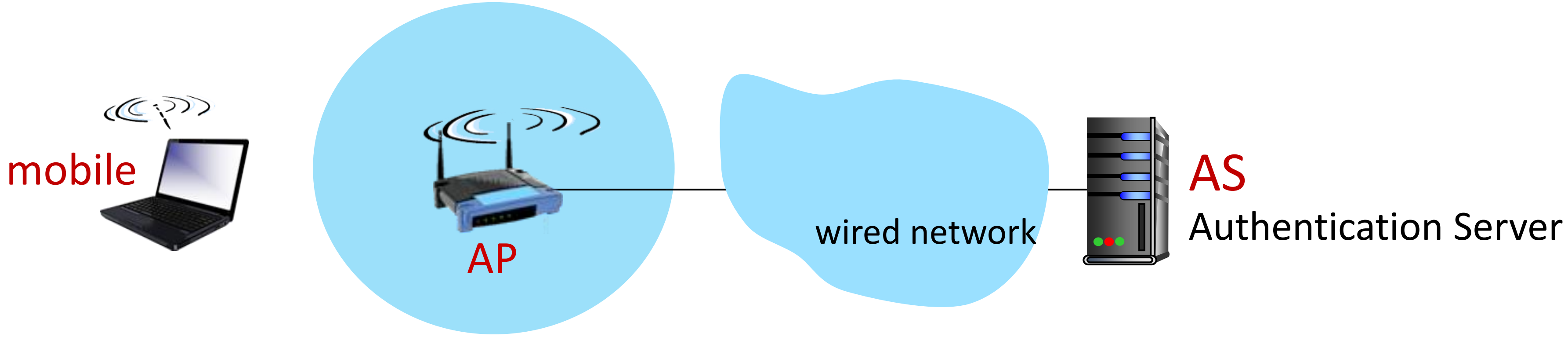
802.11: authentication, encryption



4 encrypted communication between mobile and remote host via AP

- same key derived at mobile, AS
- AS informs AP of the shared symmetric session

802.11: authentication, encryption



EAP TLS	
EAP	
EAP over LAN (EAPoL)	RADIUS
IEEE 802.11	UDP/IP

- Extensible Authentication Protocol (EAP) [RFC 3748] defines end-to-end request/response protocol between mobile device, AS

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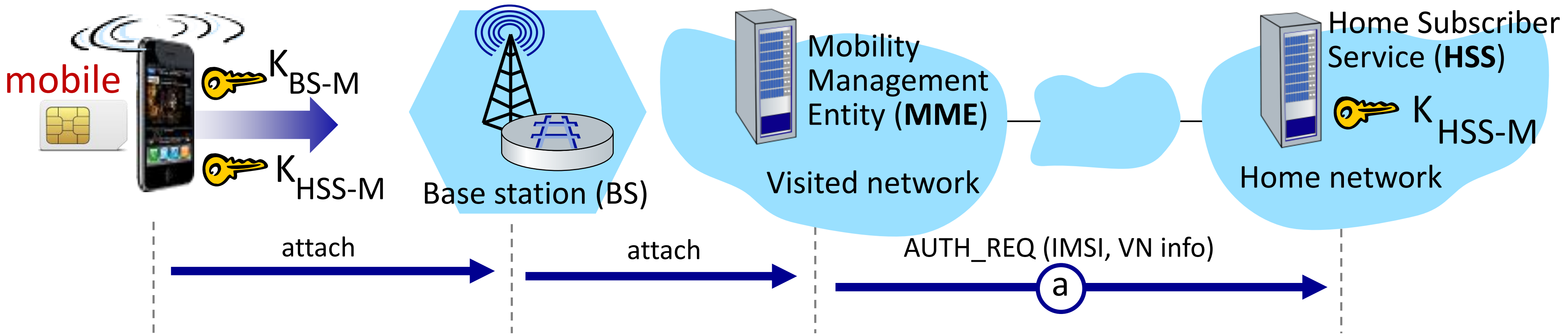
Authentication, encryption in 4G LTE

- arriving mobile must:
  - associate with BS: (establish) communication over 4G wireless link
  - authenticate itself to network, and authenticate network
- notable differences from WiFi
  - mobile's SIMcard provides global identity, contains shared keys
  - services in visited network depend on (paid) service subscription in home network

Authentication, encryption in 4G LTE

- mobile, BS use derived session key $K_{BS\text{-}M}$ to encrypt communications over 4G link
- MME in visited network + HHS in home network, together play role of WiFi AS
  - ultimate authenticator is HSS
  - trust and business relationship between visited and home networks

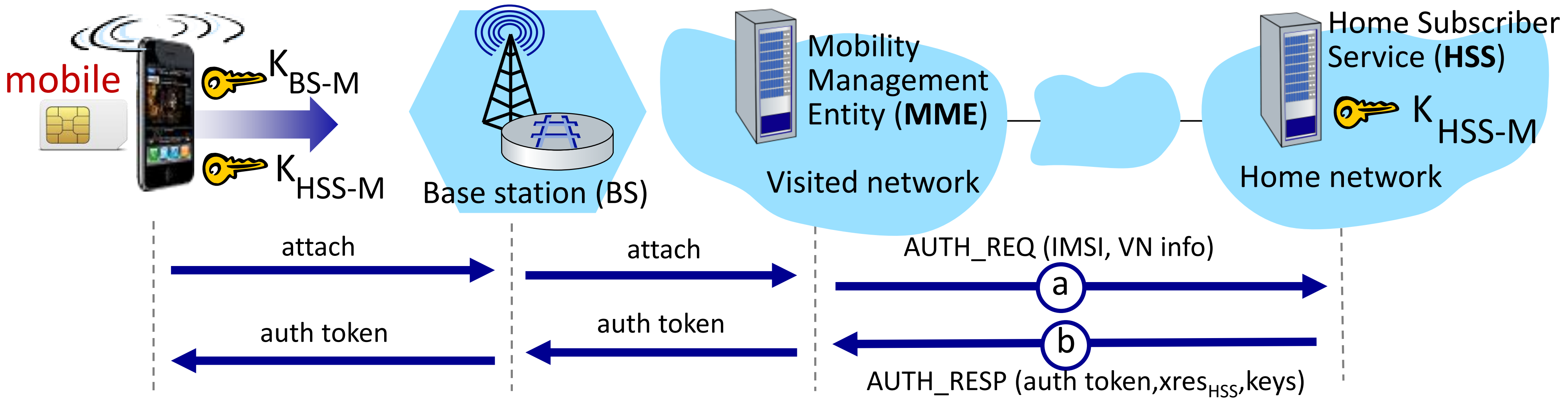
Authentication, encryption in 4G LTE



ⓐ authentication request to home network HSS

- mobile sends attach message (containing its IMSI, visited network info) relayed from BS to visited MME to home HHS
- IMSI identifies mobile's home network

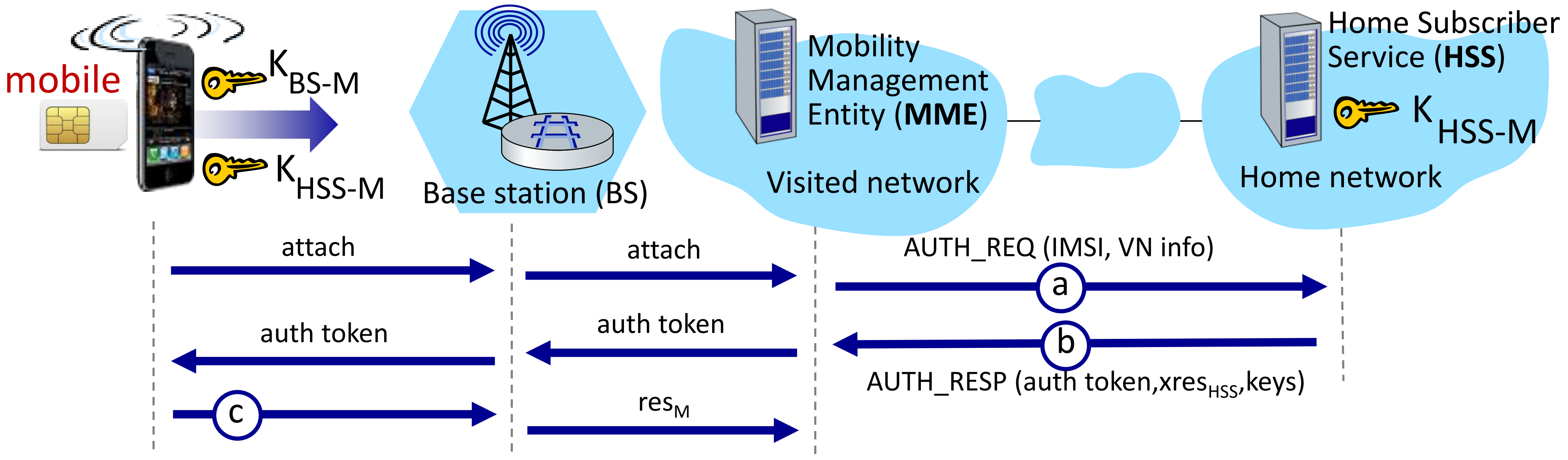
Authentication, encryption in 4G LTE



ⓑ HSS use shared-in-advance secret key, K_{HSS-M}, to derive authentication token, *auth_token*, and expected authentication response token, *$xres_{HSS}$*

- *auth_token* contains info encrypted by HSS using K_{HSS-M} , allowing mobile to know that whoever computed *auth_token* knows shared-in-advance secret
- mobile has authenticated network
- visited HSS keeps *$xres_{HSS}$* for later use

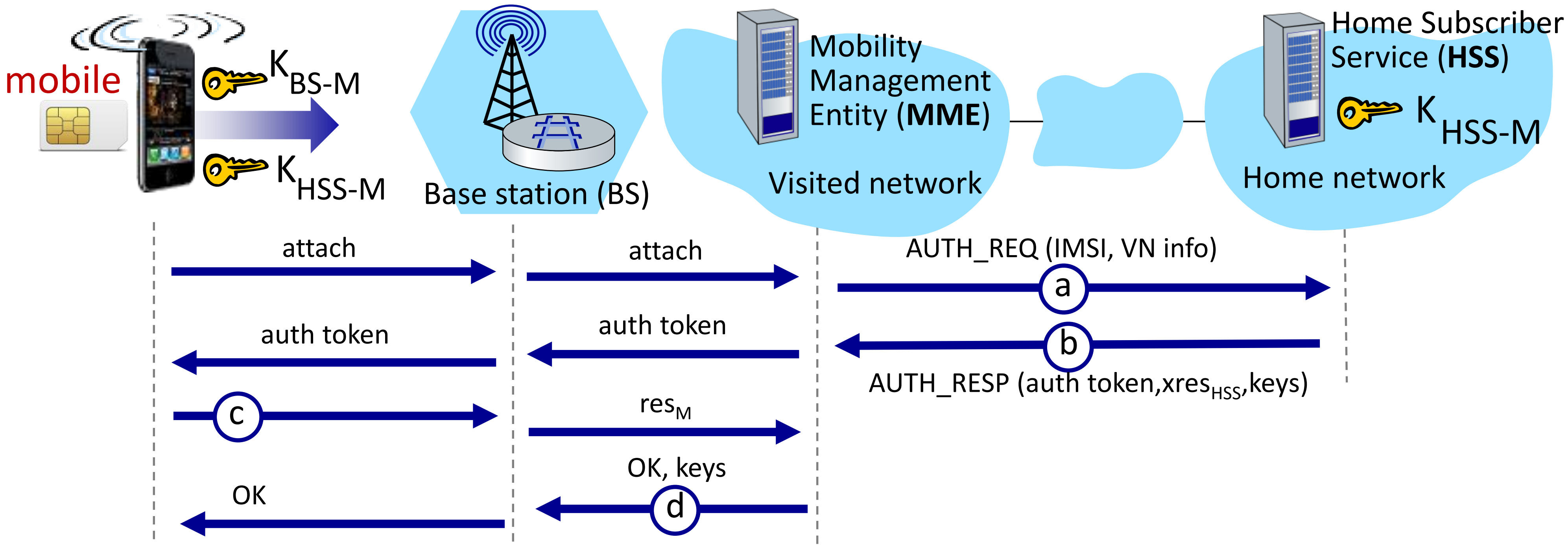
Authentication, encryption in 4G LTE



c authentication response from mobile:

- mobile computes res_M using its secret key to make same cryptographic calculation that HSS made to compute $xres_{HSS}$ and sends res_M to MME

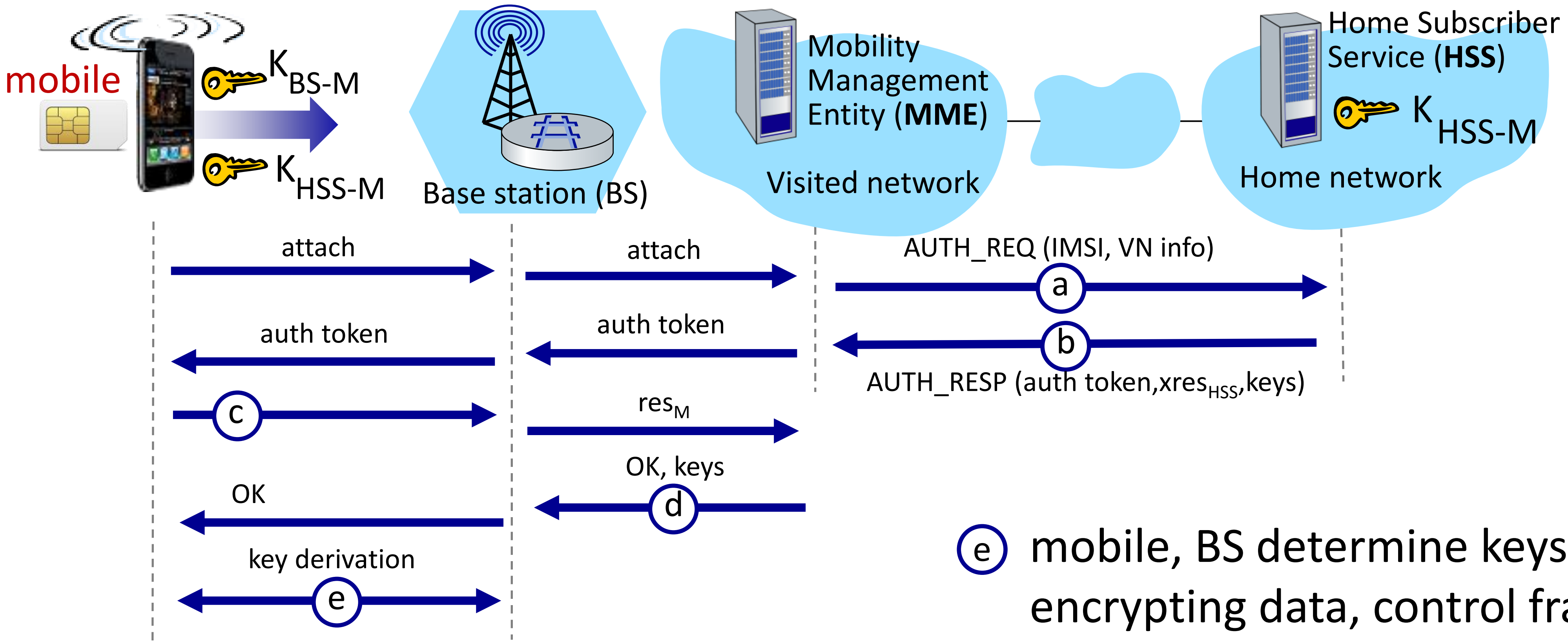
Authentication, encryption in 4G LTE



d mobile is authenticated by network:

- MMS compares mobile-computed value of res_M with the HSS-computed value of $xres_{HSS}$. If they match, mobile is authenticated ! (why?)
- MMS informs BS that mobile is authenticated, generates keys for BS

Authentication, encryption in 4G LTE



e mobile, BS determine keys for encrypting data, control frames over 4G wireless channel

- AES can be used

Authentication, encryption: from 4G to 5G

- 4G: MME in visited network makes authentication decision
- 5G: home network provides authentication decision
  - visited MME plays "middleman" role but can still reject

- 4G: uses shared-in-advance keys
- 5G: keys not shared in advance for IoT

- 4G: device IMSI transmitted in cleartext to BS
- 5G: public key crypto used to encrypt IMSI

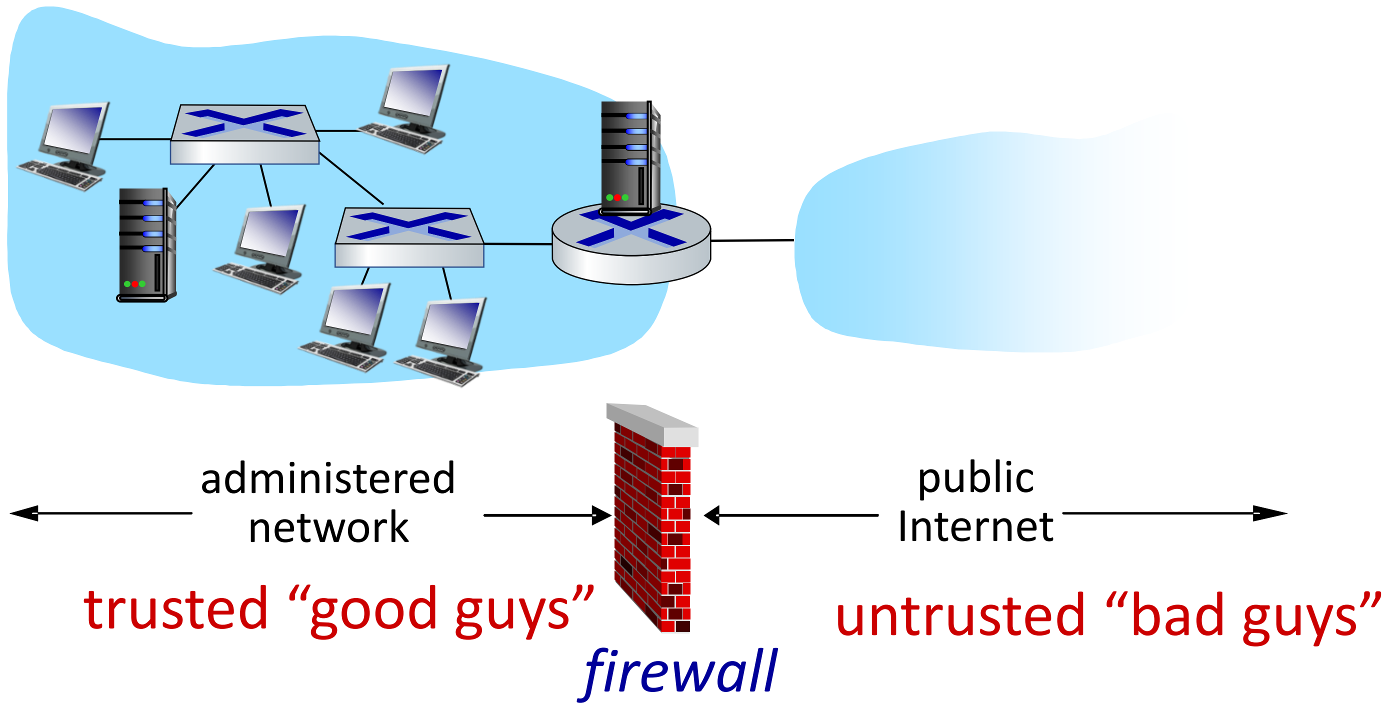
Chapter 8 outline

- What is network security?
- Principles of cryptography
- Authentication, message integrity
- Securing e-mail
- Securing TCP connections: TLS
- Network layer security: IPsec
- Security in wireless and mobile networks
- Operational security: firewalls and IDS

Firewalls

firewall

isolates organization's internal network from larger Internet, allowing some packets to pass, blocking others



Firewalls: why

prevent denial of service attacks:

- SYN flooding: attacker establishes many bogus TCP connections, no resources left for "real" connections

prevent illegal modification/access of internal data

- e.g., attacker replaces CIA's homepage with something else

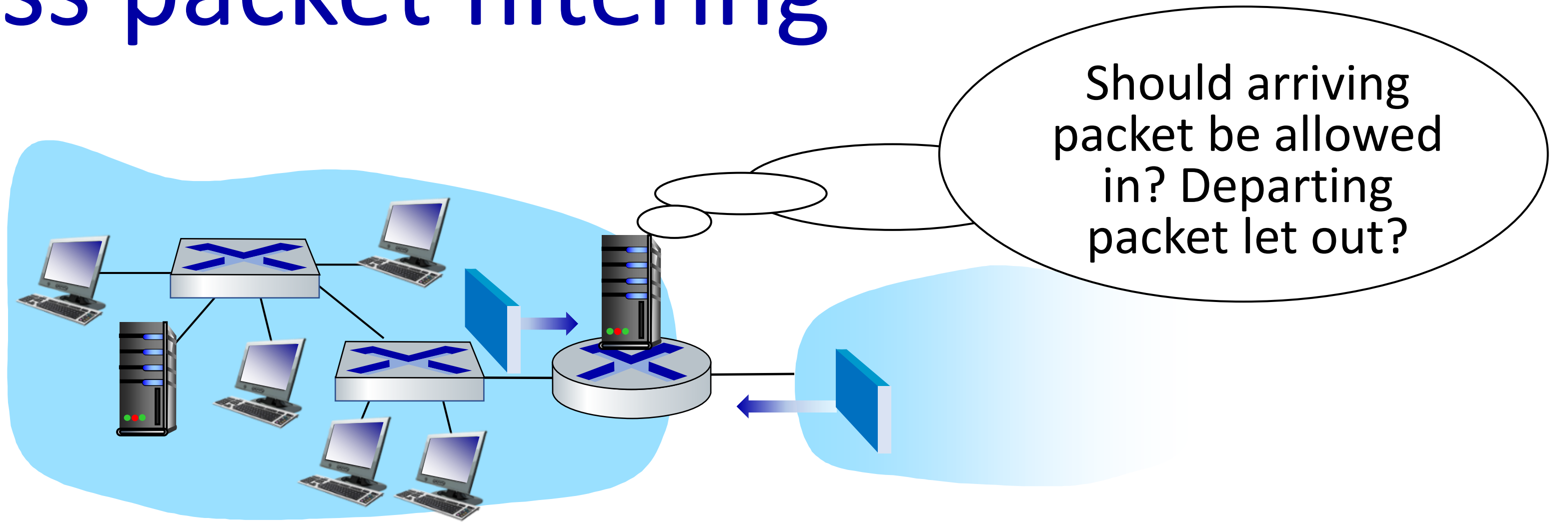
allow only authorized access to inside network

- set of authenticated users/hosts

three types of firewalls:

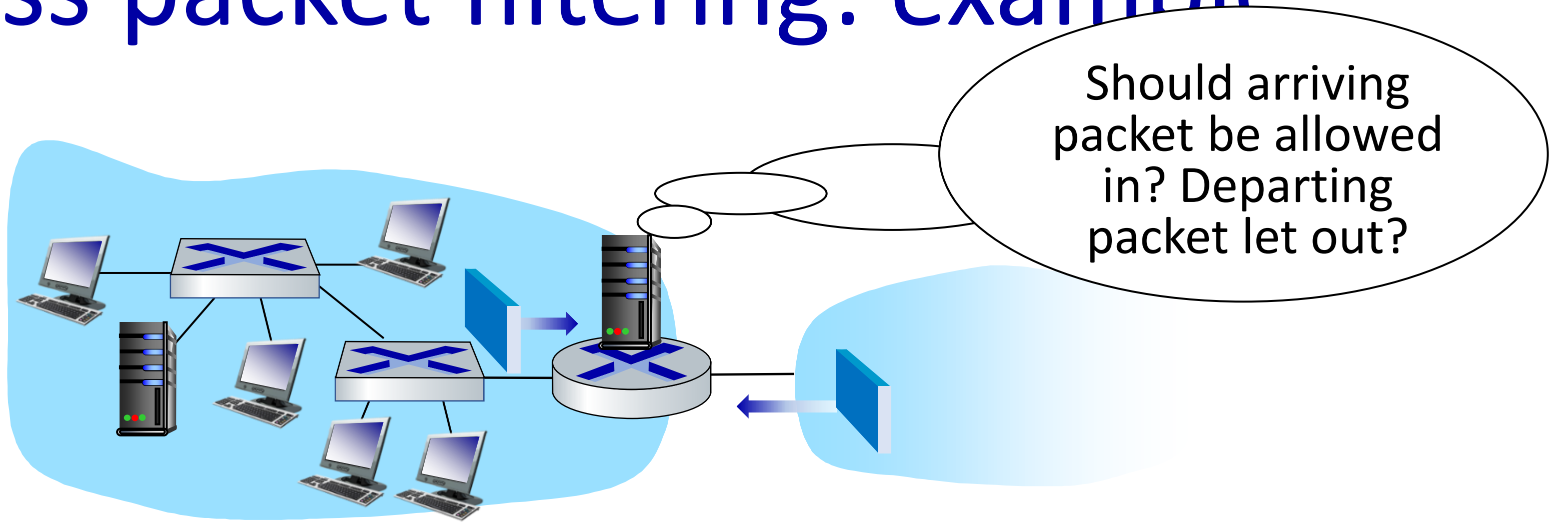
- stateless packet filters
- stateful packet filters
- application gateways

Stateless packet filtering



- internal network connected to Internet via router firewall
- filters packet-by-packet, decision to forward/drop packet based on:
  - source IP address, destination IP address
  - TCP/UDP source, destination port numbers
  - ICMP message type
  - TCP SYN, ACK bits

Stateless packet filtering: example



- example 1: block incoming and outgoing datagrams with IP protocol field = 17 and with either source or dest port = 23
  - result: all incoming, outgoing UDP flows and telnet connections are blocked
- example 2: block inbound TCP segments with ACK=0
  - result: prevents external clients from making TCP connections with internal clients, but allows internal clients to connect to outside

Stateless packet filtering: more examples

Policy	Firewall Setting
no outside Web access	drop all outgoing packets to any IP address, port 80
no incoming TCP connections, except those for institution's public Web server only.	drop all incoming TCP SYN packets to any IP except 130.207.244.203, port 80
prevent Web-radios from eating up the available bandwidth.	drop all incoming UDP packets - except DNS and router broadcasts.
prevent your network from being used for a smurf DoS attack.	drop all ICMP packets going to a "broadcast" address (e.g. 130.207.255.255)
prevent your network from being tracerouted	drop all outgoing ICMP TTL expired traffic

Access Control Lists

ACL: table of rules, applied top to bottom to incoming packets: (action, condition) pairs: looks like OpenFlow forwarding (Ch. 4)!

action	source address	dest address	protocol	source port	dest port	flag bit
allow	222.22/16	outside of 222.22/16	TCP	> 1023	80	any
allow	outside of 222.22/16	222.22/16	TCP	80	> 1023	ACK
allow	222.22/16	outside of 222.22/16	UDP	> 1023	53	---
allow	outside of 222.22/16	222.22/16	UDP	53	> 1023	----
deny	all	all	all	all	all	all

Stateful packet filtering

- *stateless packet filter:* heavy handed tool
  - admits packets that "make no sense," e.g., dest port = 80, ACK bit set, even though no TCP connection established:

action	source address	dest address	protocol	source port	dest port	flag bit
allow	outside of 222.22/16	222.22/16	TCP	80	> 1023	ACK

- *stateful packet filter:* track status of every TCP connection
  - track connection setup (SYN), teardown (FIN): determine whether incoming, outgoing packets "makes sense"
  - timeout inactive connections at firewall: no longer admit packets

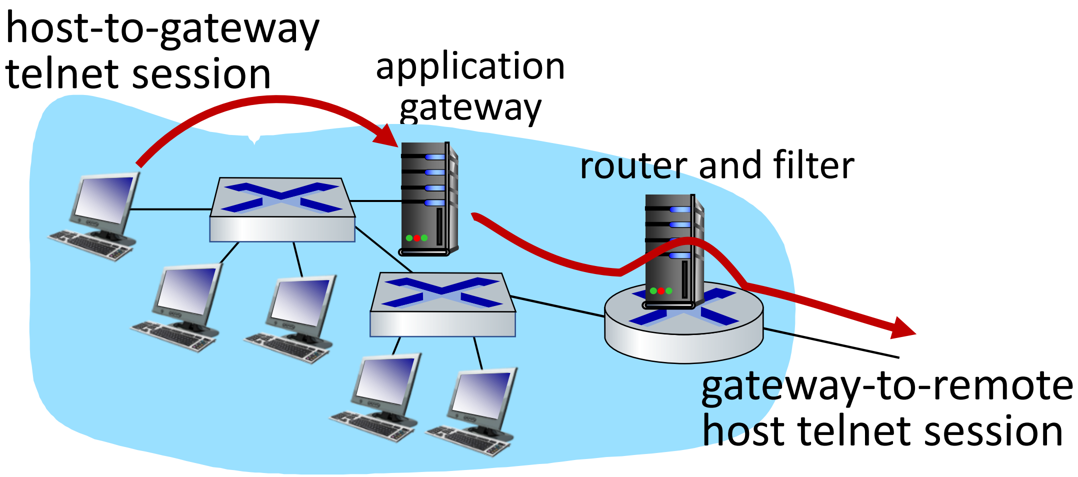
Stateful packet filtering

ACL augmented to indicate need to check connection state table before admitting packet

action	source address	dest address	proto	source port	dest port	flag bit	check connection
allow	222.22/16	outside of 222.22/16	TCP	> 1023	80	any	
allow	outside of 222.22/16	222.22/16	TCP	80	> 1023	ACK	X
allow	222.22/16	outside of 222.22/16	UDP	> 1023	53	---	
allow	outside of 222.22/16	222.22/16	UDP	53	> 1023	----	X
deny	all	all	all	all	all	all	

Application gateways

- filter packets on application data as well as on IP/TCP/UDP fields.
- *example:* allow select internal users to telnet outside



1. require all telnet users to telnet through gateway.
2. for authorized users, gateway sets up telnet connection to dest host
   - gateway relays data between 2 connections
3. router filter blocks all telnet connections not originating from gateway

Limitations of firewalls, gateways

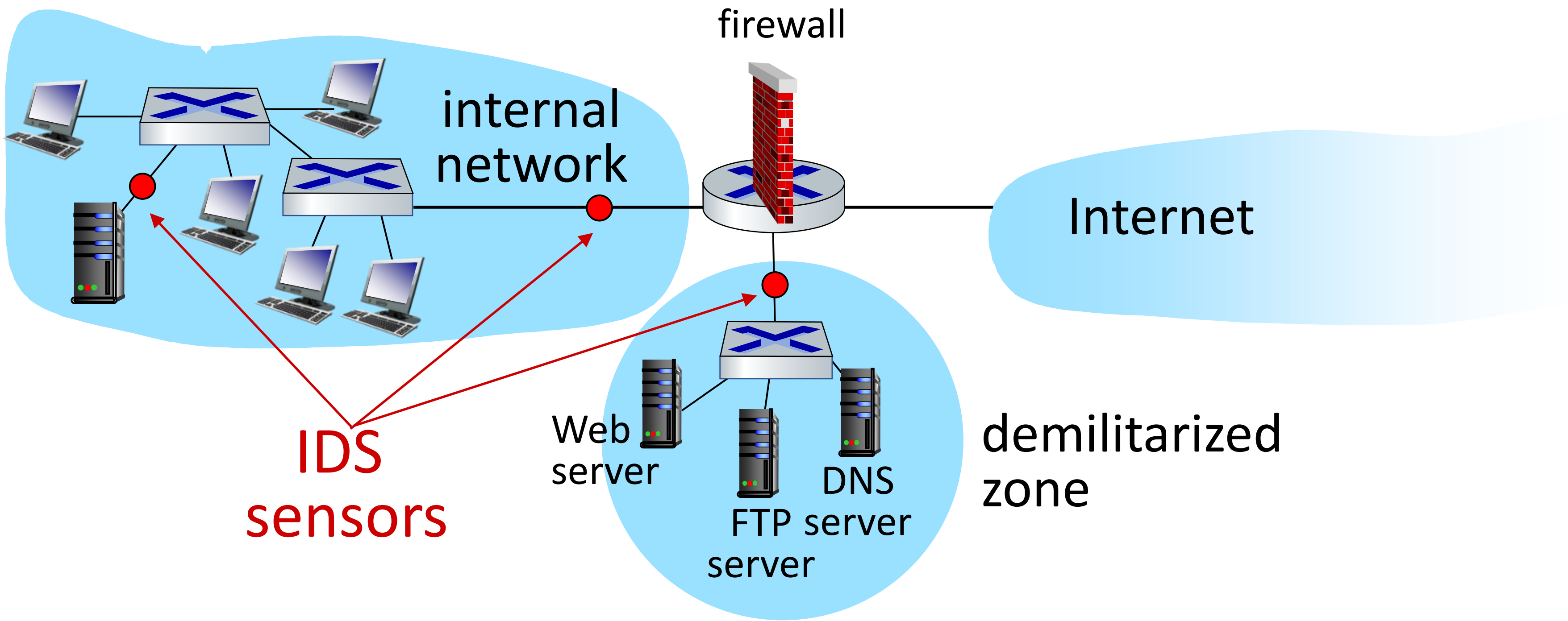
- IP spoofing: router can't know if data "really" comes from claimed source
- if multiple apps need special treatment, each has own app. gateway
- client software must know how to contact gateway
  - e.g., must set IP address of proxy in Web browser
- filters often use all or nothing policy for UDP
- *tradeoff:* degree of communication with outside world, level of security
- many highly protected sites still suffer from attacks

Intrusion detection systems

- packet filtering:
  - operates on TCP/IP headers only
  - no correlation check among sessions
- IDS: intrusion detection system
  - deep packet inspection: look at packet contents (e.g., check character strings in packet against database of known virus, attack strings)
  - examine correlation among multiple packets
    - port scanning
    - network mapping
    - DoS attack

Intrusion detection systems

multiple IDSs: different types of checking at different locations



Network Security (summary)

basic techniques……

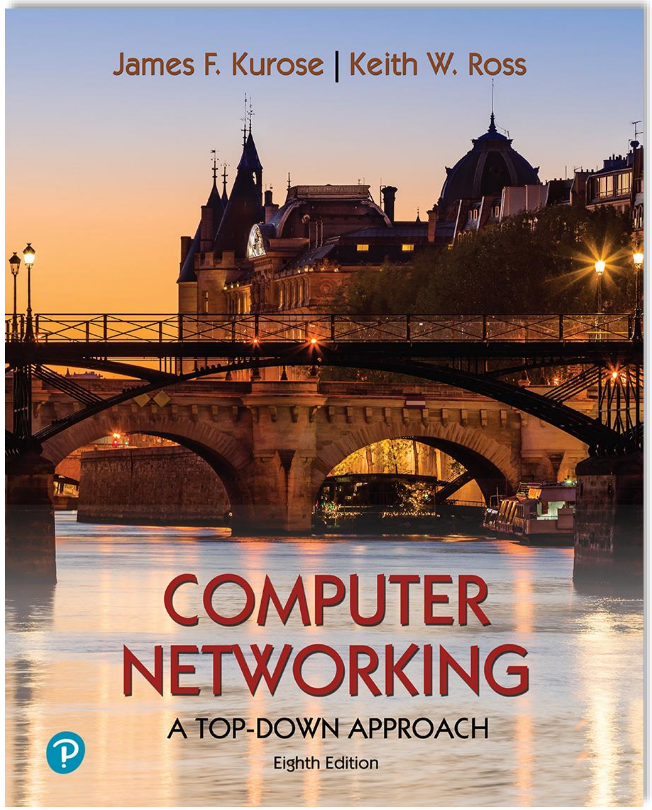
- cryptography (symmetric and public key)
- message integrity
- end-point authentication

…. used in many different security scenarios

- secure email
- secure transport (TLS)
- IP sec
- 802.11, 4G/5G

operational security: firewalls and IDS

References



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