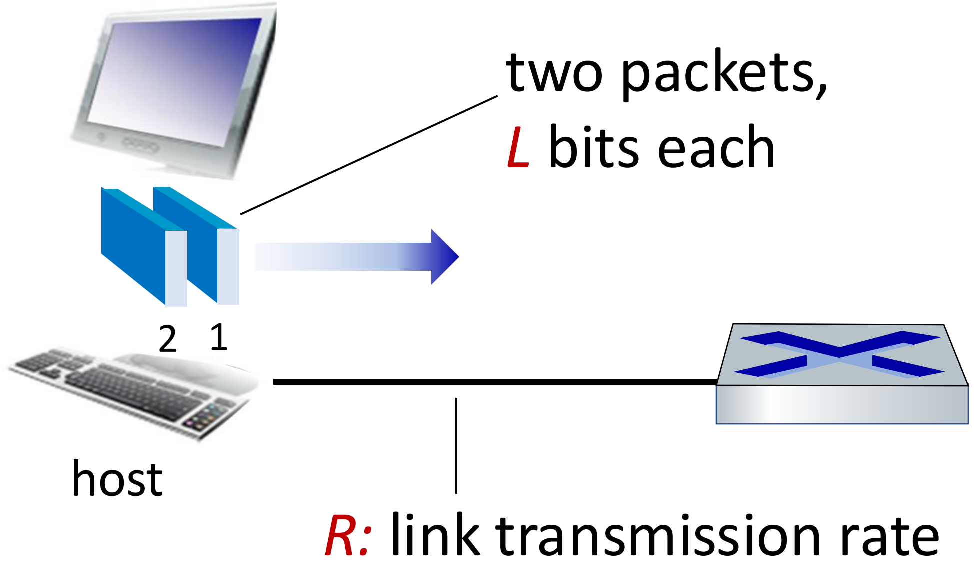
Review

Chapter 1

Host: sends *packets* of data

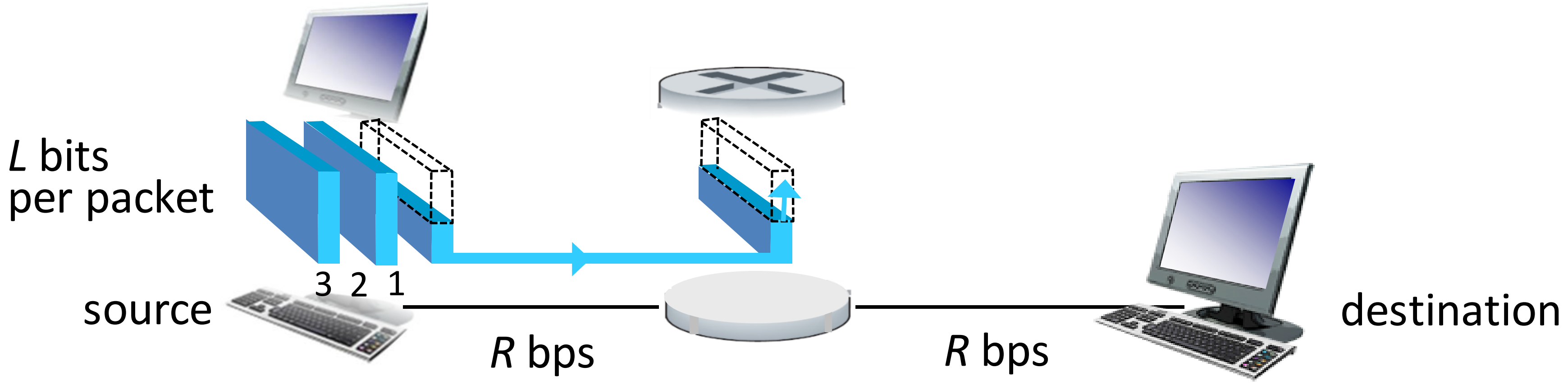
host sending function:

- 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*



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

Packet-switching: store-and-forward

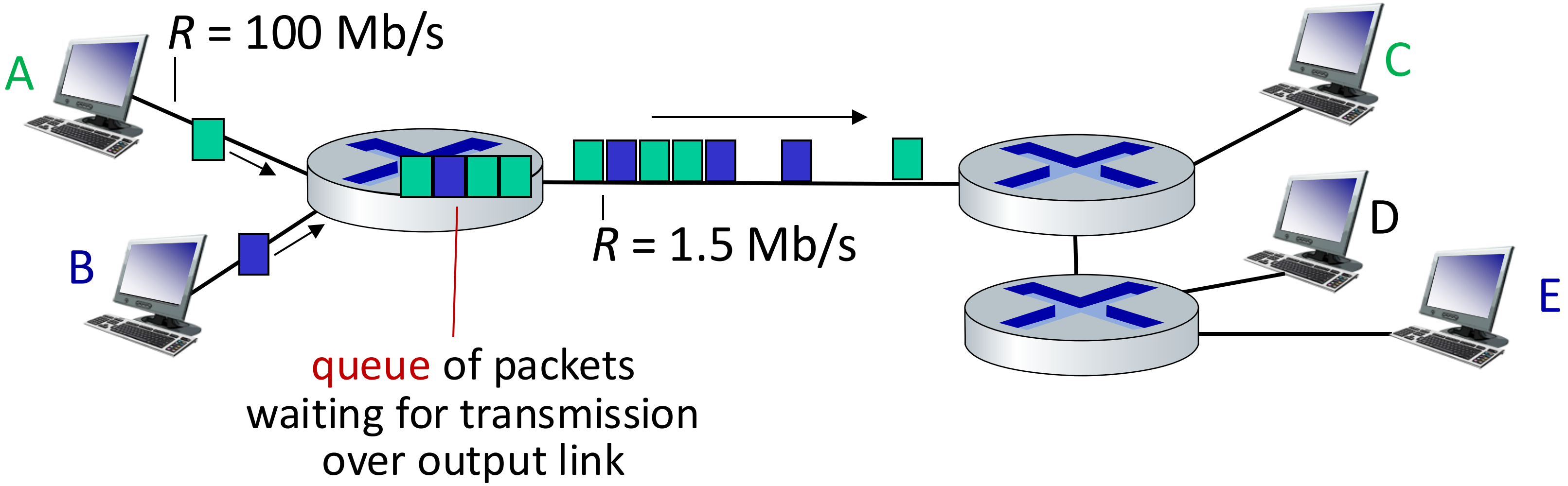


- packet transmission delay: takes L/R seconds to transmit (push out) L-bit packet into link at R bps
- *store and forward: entire* packet must arrive at router before it can be transmitted on next link

One-hop numerical example:

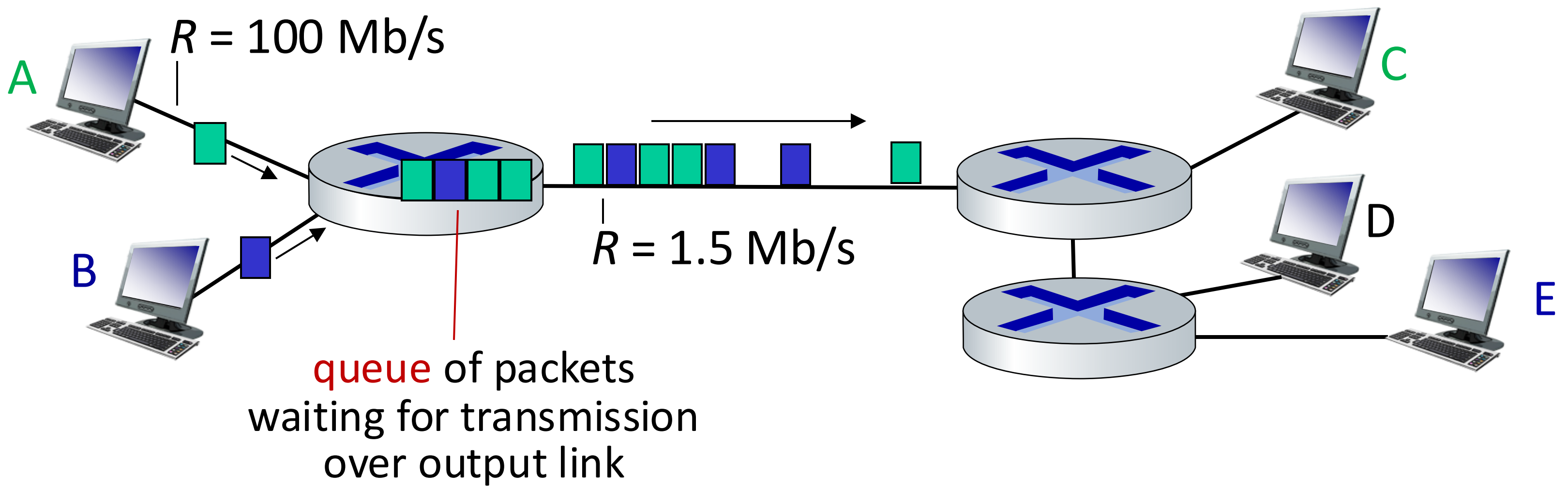
- L = 10 Kbits
- R = 100 Mbps
- one-hop transmission delay = 0.1 msec

Packet-switching: queueing



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

Packet-switching: queueing



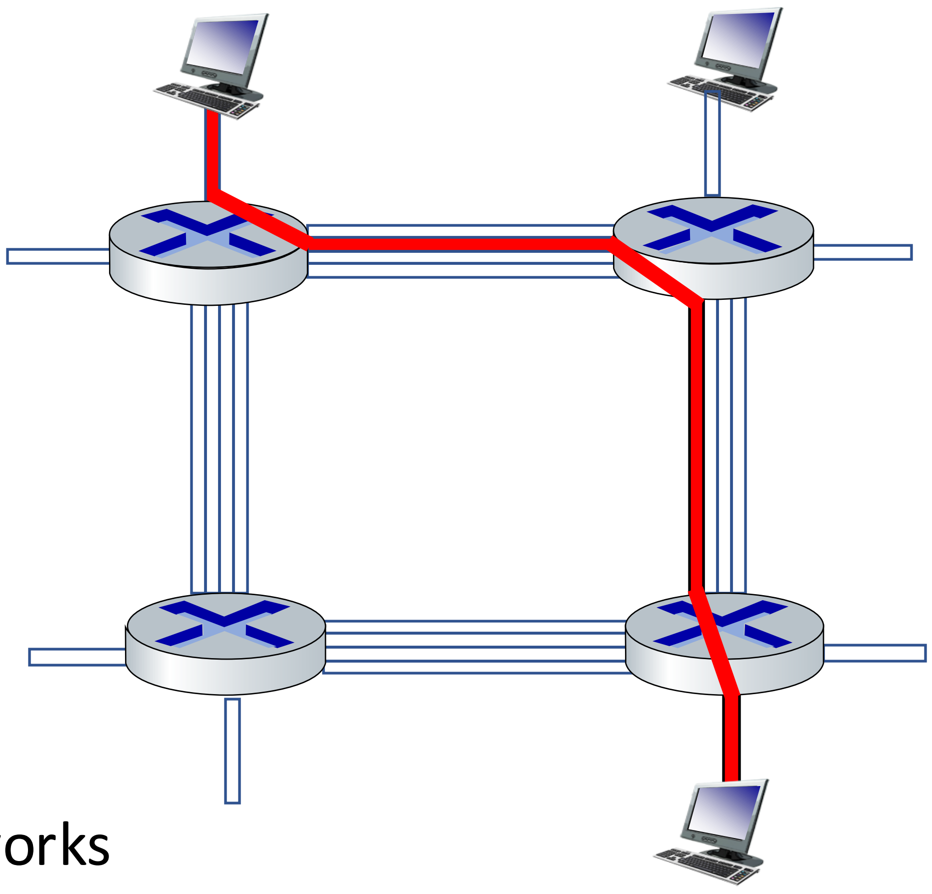
Packet queuing and loss: if arrival rate (in bps) to link exceeds transmission rate (bps) of link for some period of time:

- 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



Packet switching versus circuit switching

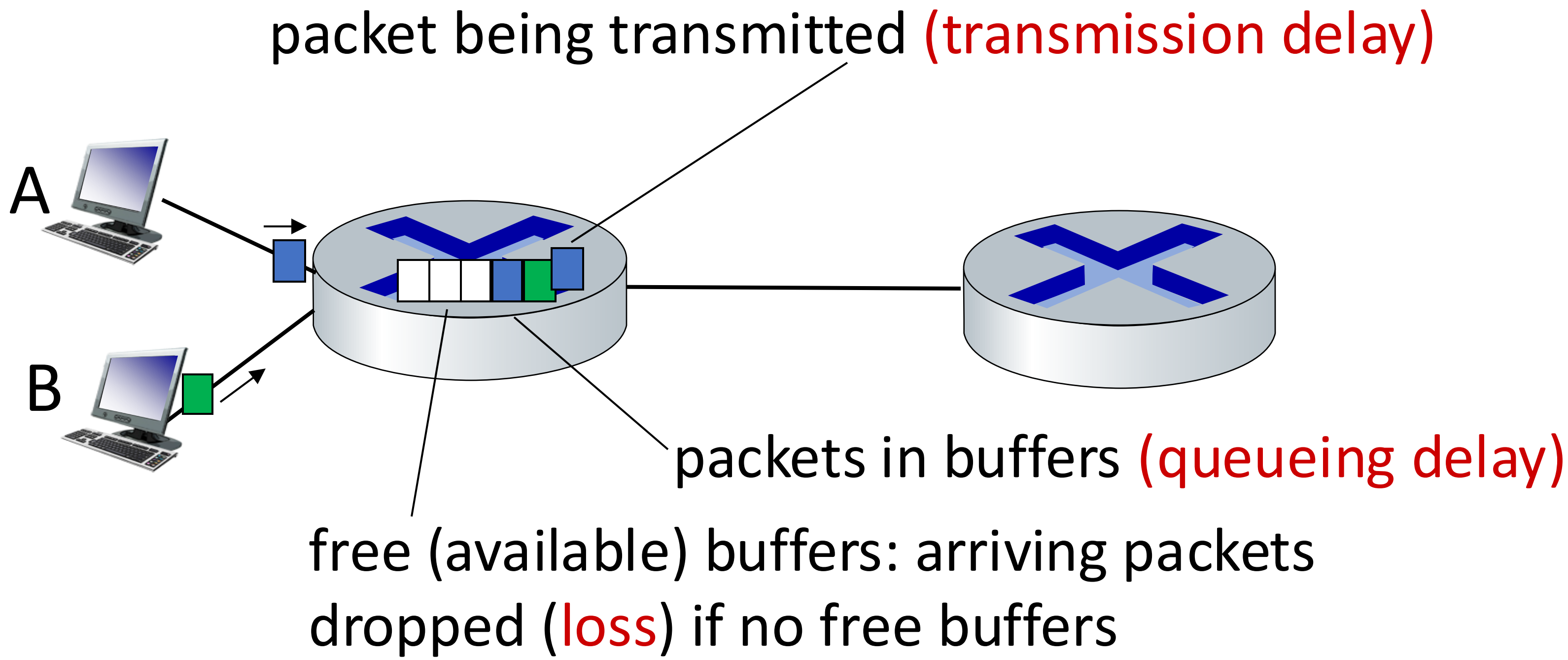
Is packet switching a "slam dunk winner"?

- 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.

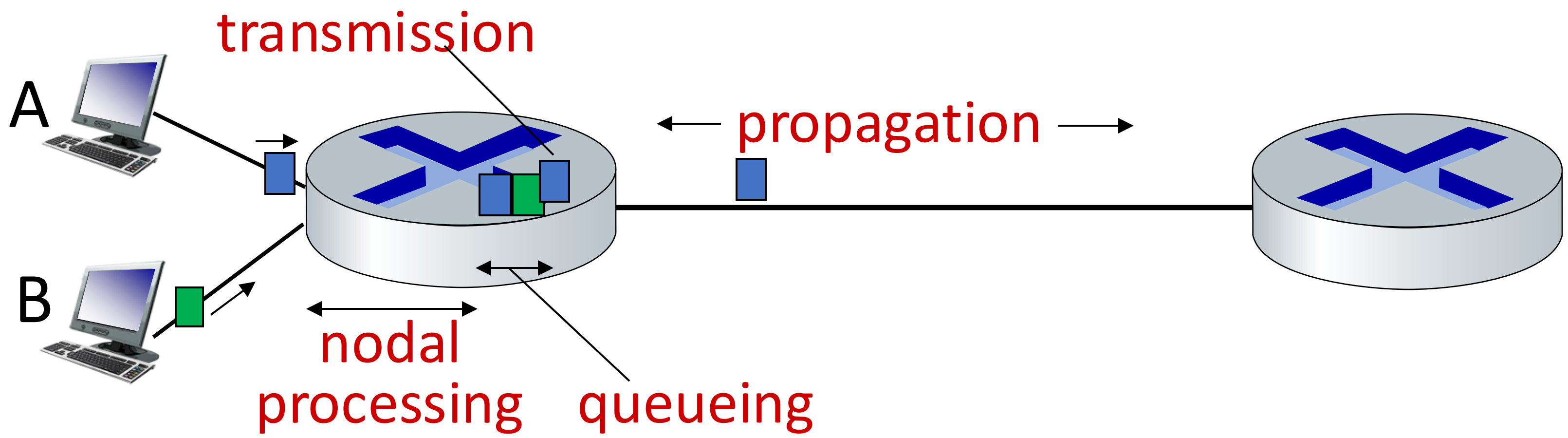
Q: human analogies of reserved resources (circuit switching) versus on-demand allocation (packet switching)?

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



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

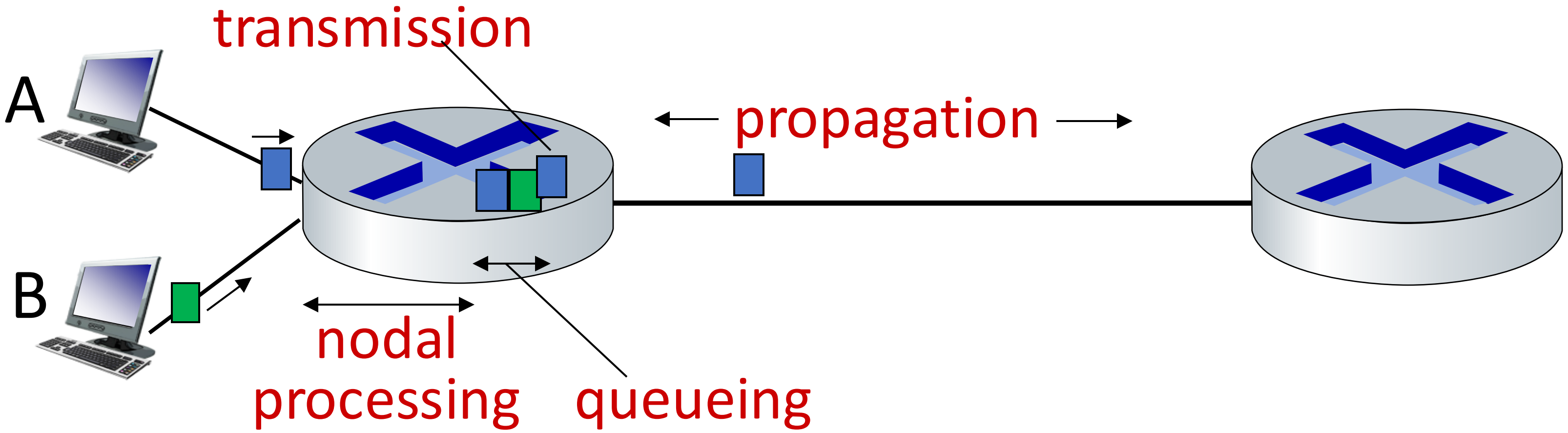
d_{proc}: nodal processing

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

d_{queue}: queueing delay

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

Packet delay: four sources



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

d_{trans}: transmission delay:

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

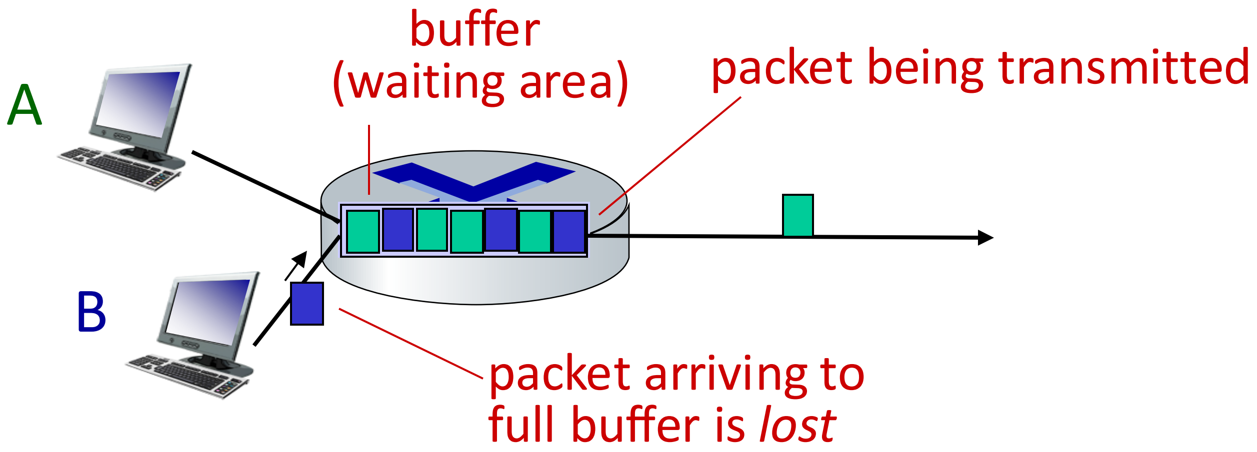
d_{prop}: propagation delay:

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

d_{trans} and d_{prop}
very different

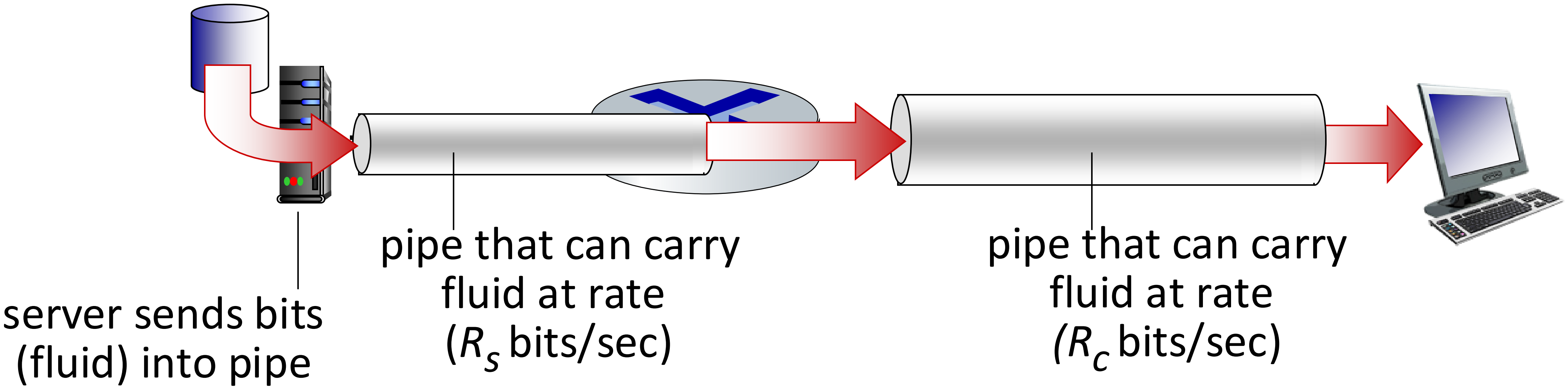
Packet loss

- 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



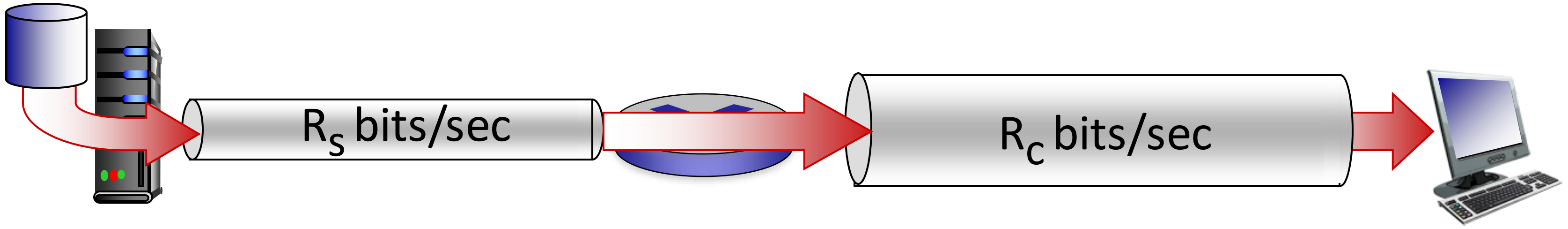
Throughput

- *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

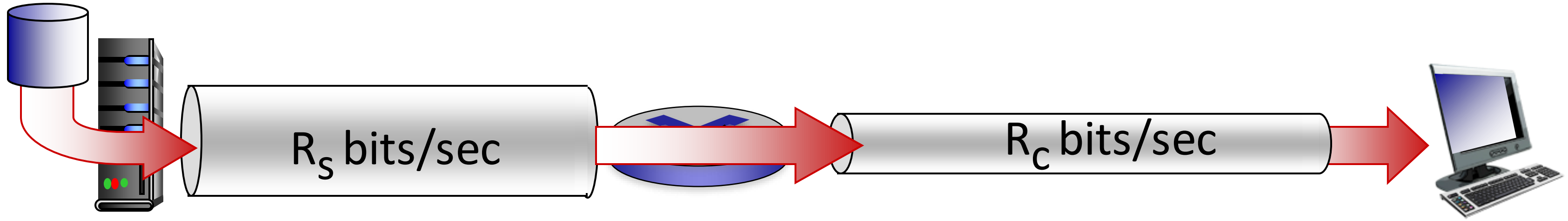


Throughput

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



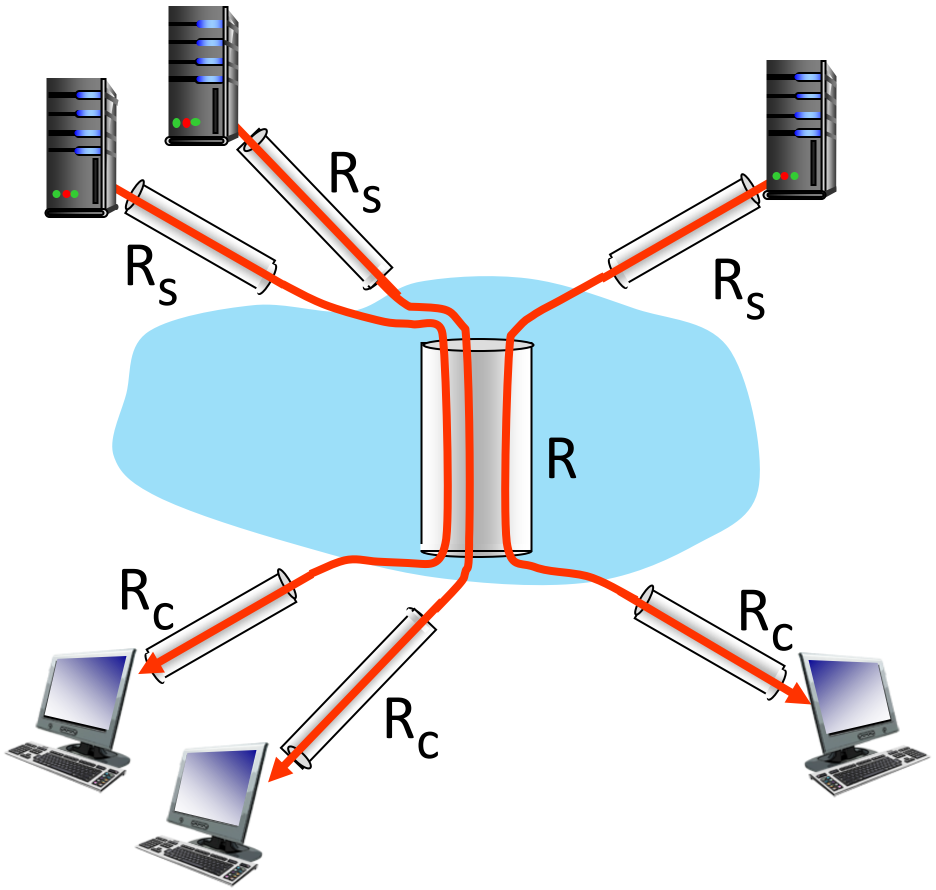
$R_s > R_c$ What is average end-end throughput?



bottleneck link

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

Throughput: network scenario

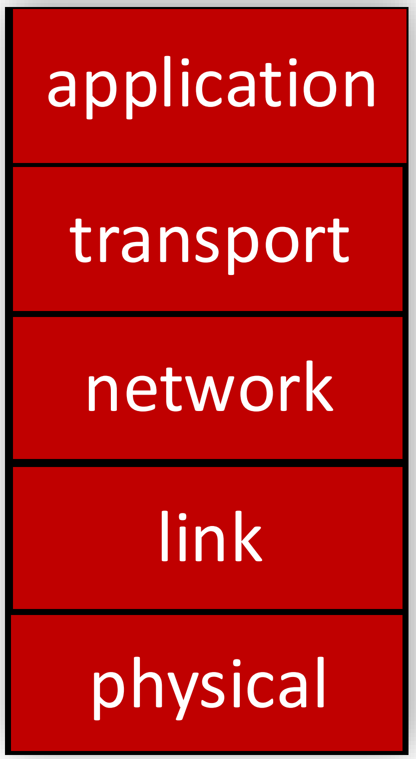


10 connections (fairly) share backbone bottleneck link *R* bits/sec

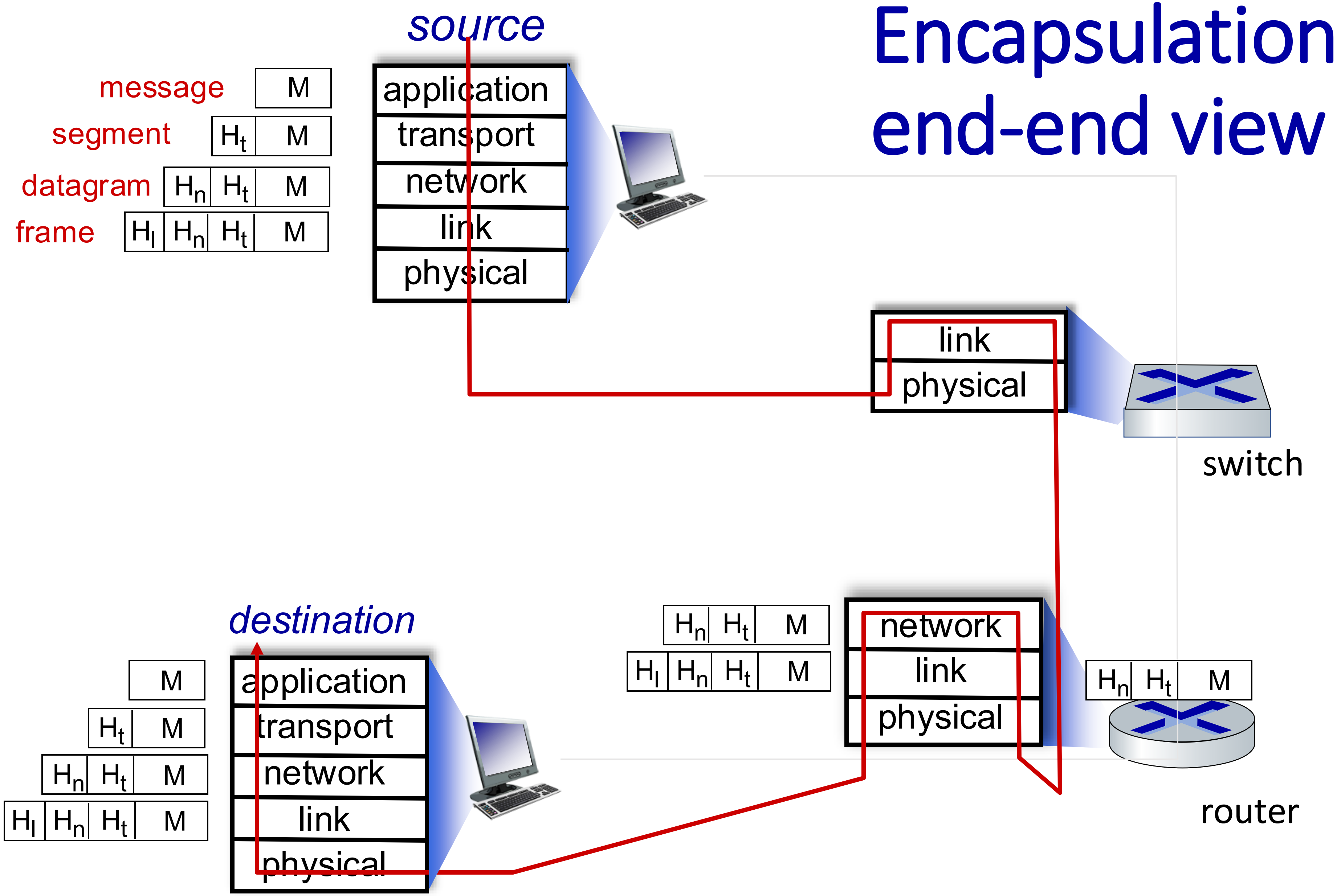
- per-connection end-end throughput: *min(R_c,R_s,R/10)*
- in practice: R_c or R_s is often bottleneck

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”



Encapsulation: an end-end view



Chapter 2
Application Layer

Client-server paradigm

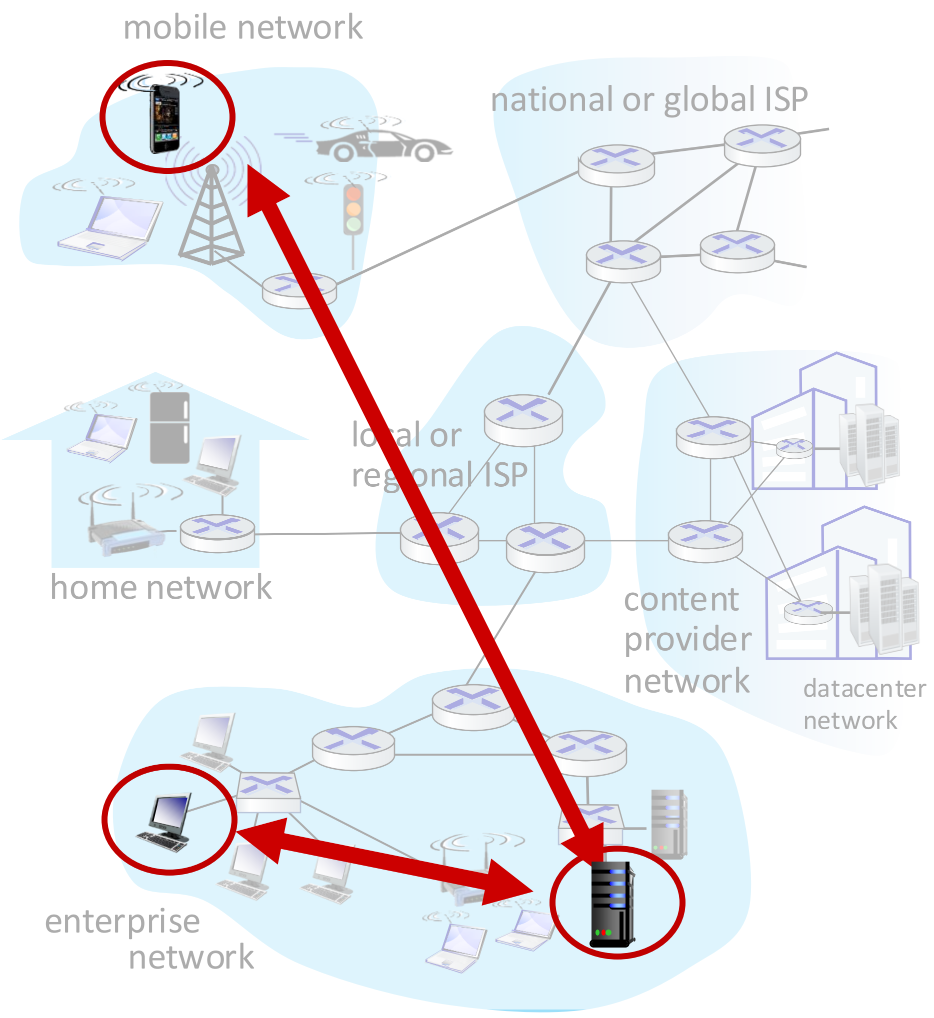
server:

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

clients:

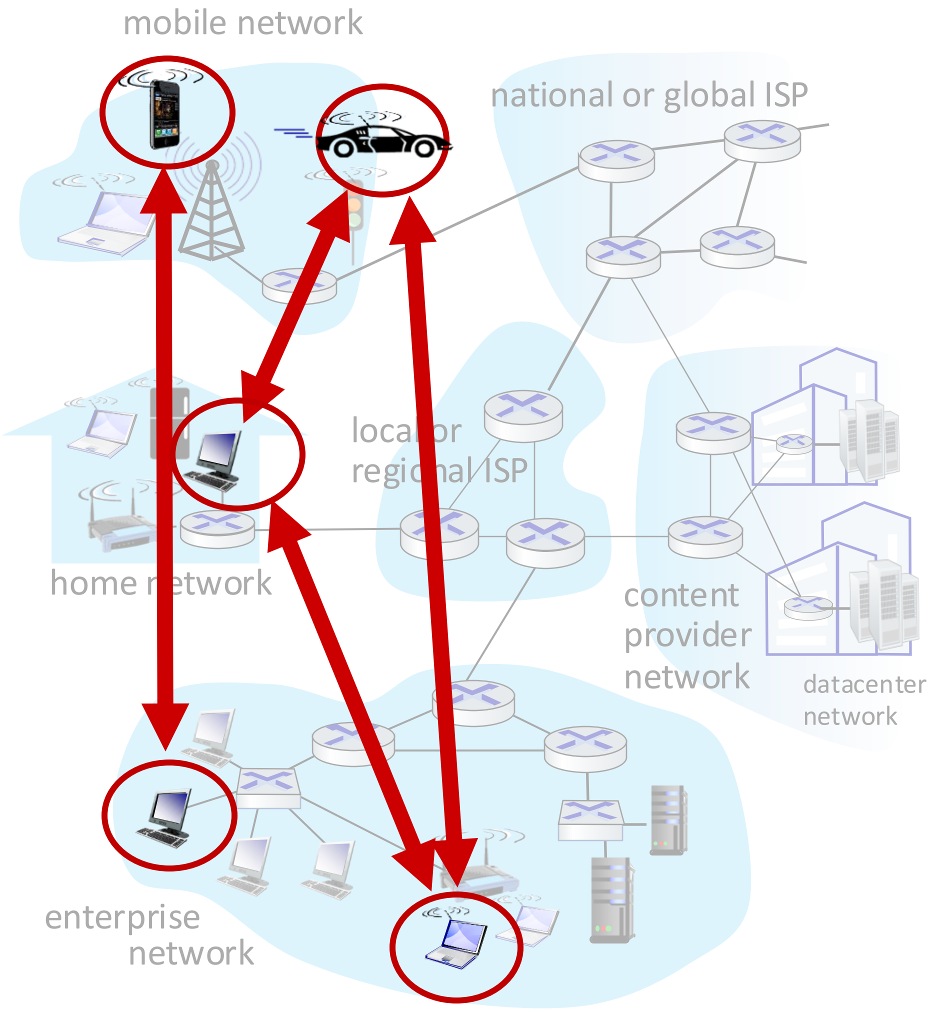
- contact, communicate with server
- may be intermittently connected
- may have dynamic IP addresses
- do *not* communicate directly with each other

- examples: HTTP, IMAP, FTP



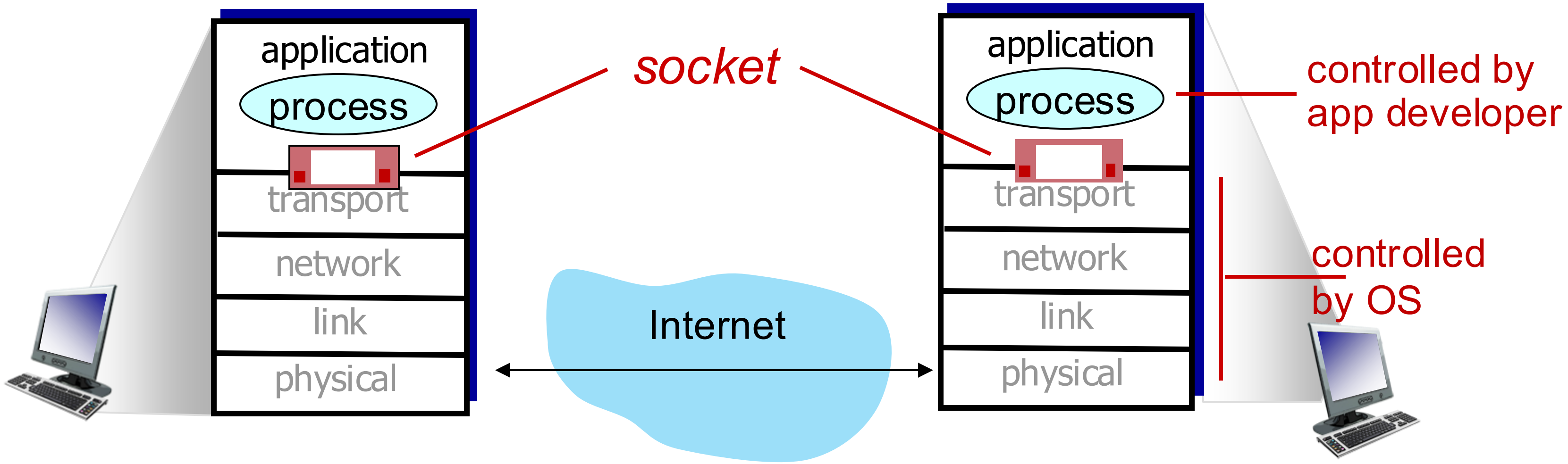
Peer-peer architecture

- *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]



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



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?

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

HTTP request message

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

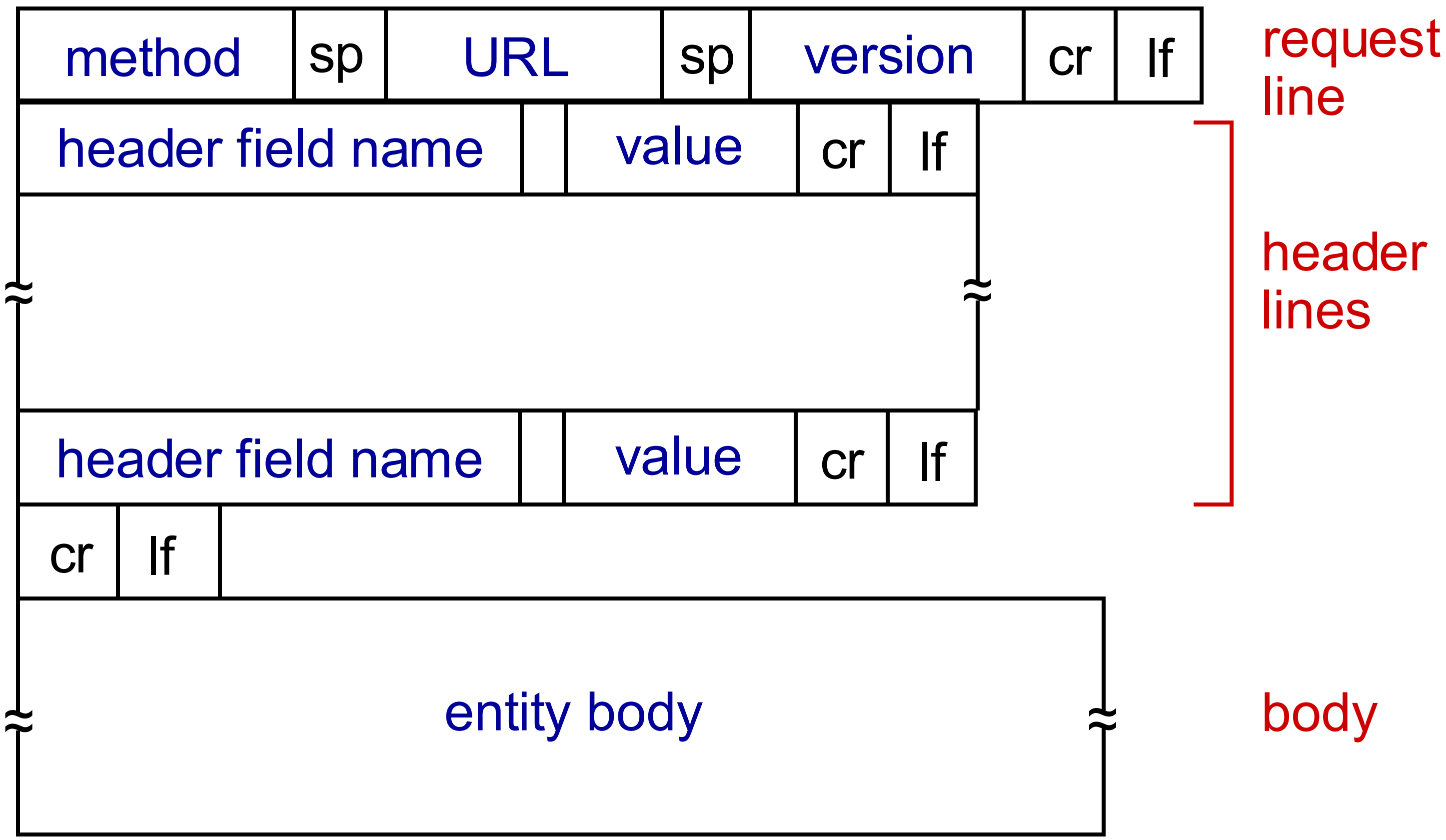
carriage return character

line-feed character

request line (GET, POST, HEAD commands)

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

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 '?'):

```
www.somesite.com/animalsearch?monkeys&banana
```

HEAD method:

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

PUT method:

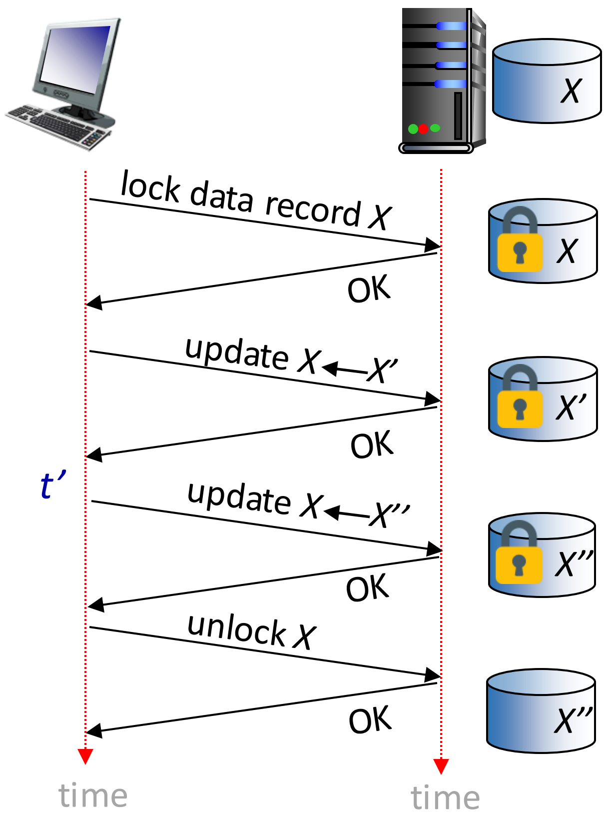
- uploads new file (object) to server
- completely replaces file that exists at specified URL with content in entity body of POST HTTP request message

Maintaining user/server state: cookies

Recall: HTTP GET/response interaction is *stateless*

- no notion of multi-step exchanges of HTTP messages to complete a Web "transaction"
  - no need for client/server to track "state" of multi-step exchange
  - all HTTP requests are independent of each other
  - no need for client/server to "recover" from a partially-completed-but-never-completely-completed transaction

a stateful protocol: client makes two changes to X, or none at all



Q: what happens if network connection or client crashes at *t′* ?

Maintaining user/server state: cookies

Web sites and client browser use *cookies* to maintain some state between transactions

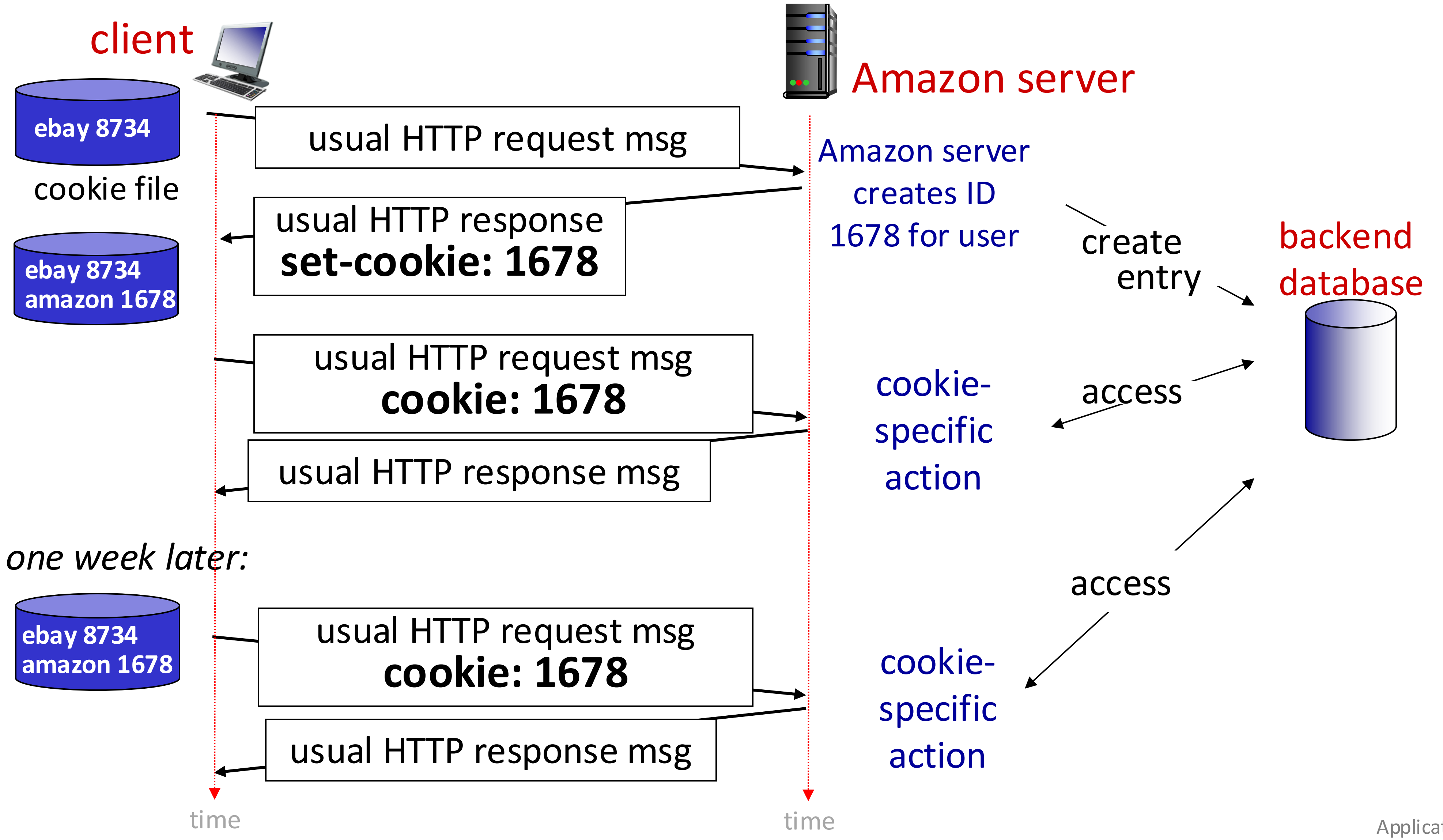
four components:

1) cookie header line of HTTP *response* message
2) cookie header line in next HTTP *request* message
3) cookie file kept on user's host, managed by user's browser
4) back-end database at Web site

Example:

- Susan uses browser on laptop, visits specific e-commerce site for first time
- when initial HTTP requests arrives at site, site creates:
  - unique ID (aka "cookie")
  - entry in backend database for ID
- subsequent HTTP requests from Susan to this site will contain cookie ID value, allowing site to "identify" Susan

Maintaining user/server state: cookies



HTTP cookies: comments

What cookies can be used for:

- authorization
- shopping carts
- recommendations
- user session state (Web e-mail)

Challenge: How to keep state?

- *at protocol endpoints:* maintain state at sender/receiver over multiple transactions
- *in messages:* cookies in HTTP messages carry state

aside

cookies and privacy:

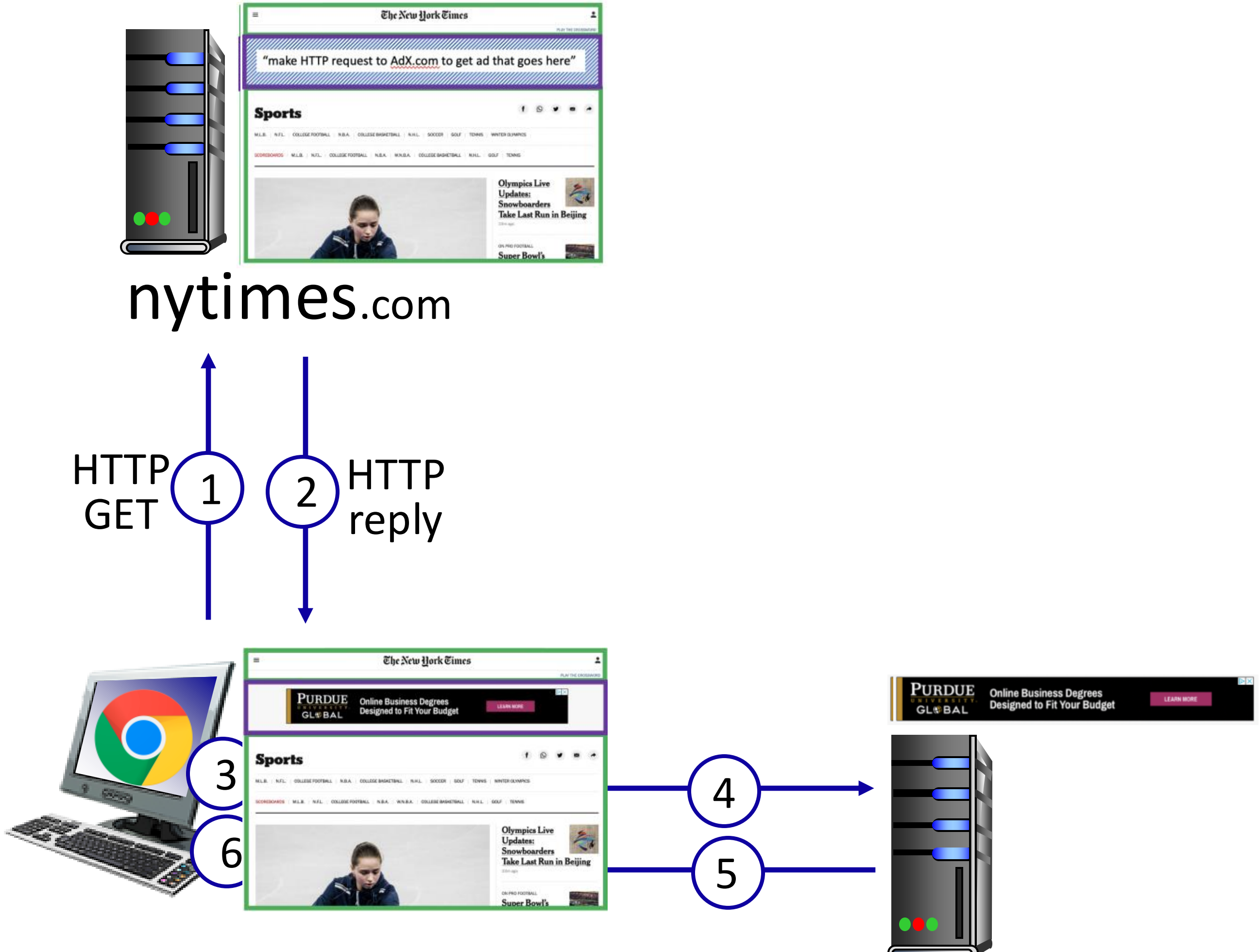
- cookies permit sites to *learn* a lot about you on their site.
- third party persistent cookies (tracking cookies) allow common identity (cookie value) to be tracked across multiple web sites

Example: displaying a NY Times web page

1 2 GET base html file from nytimes.com

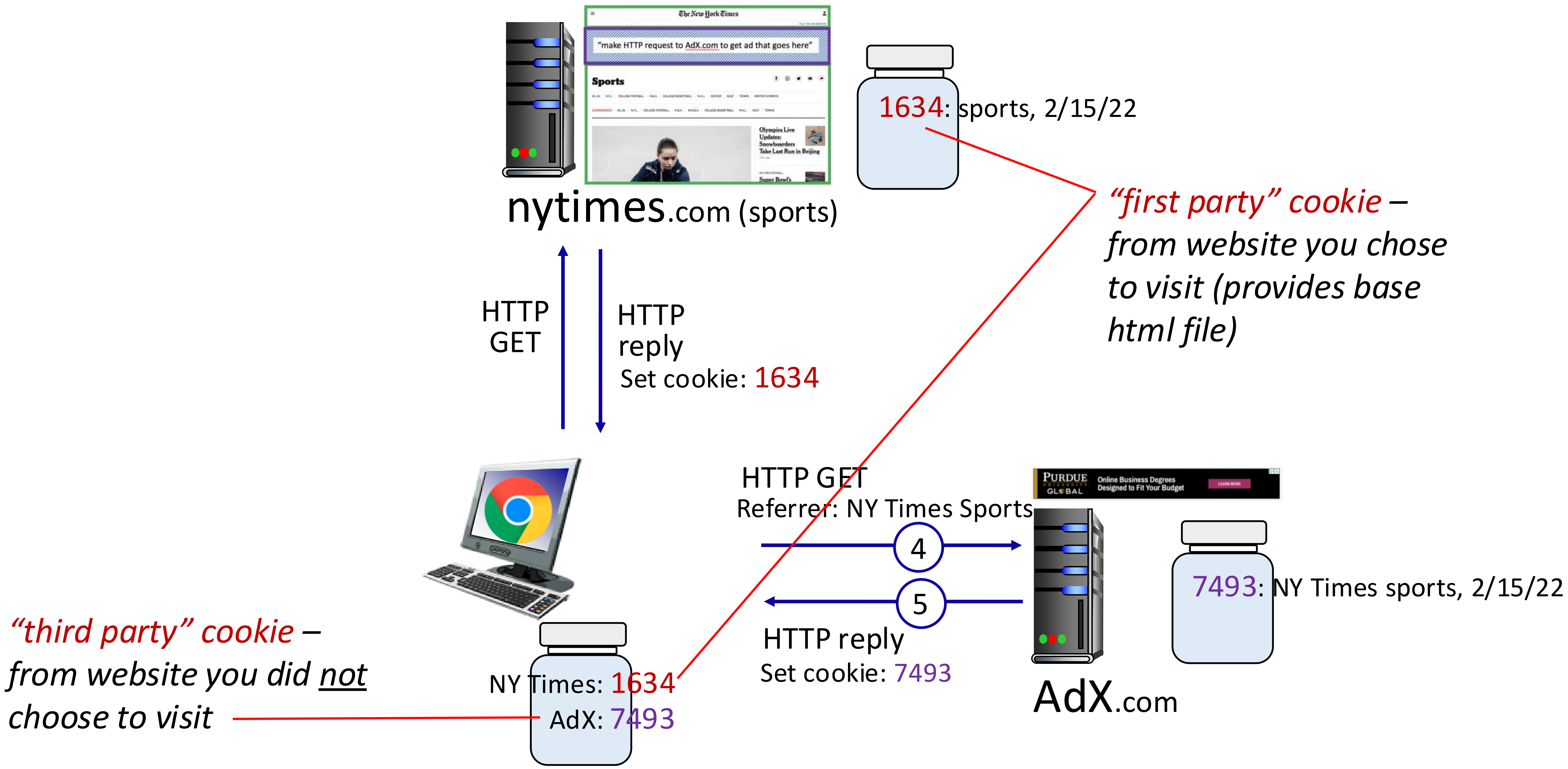
4 5 fetch ad from AdX.com

7 display composed page

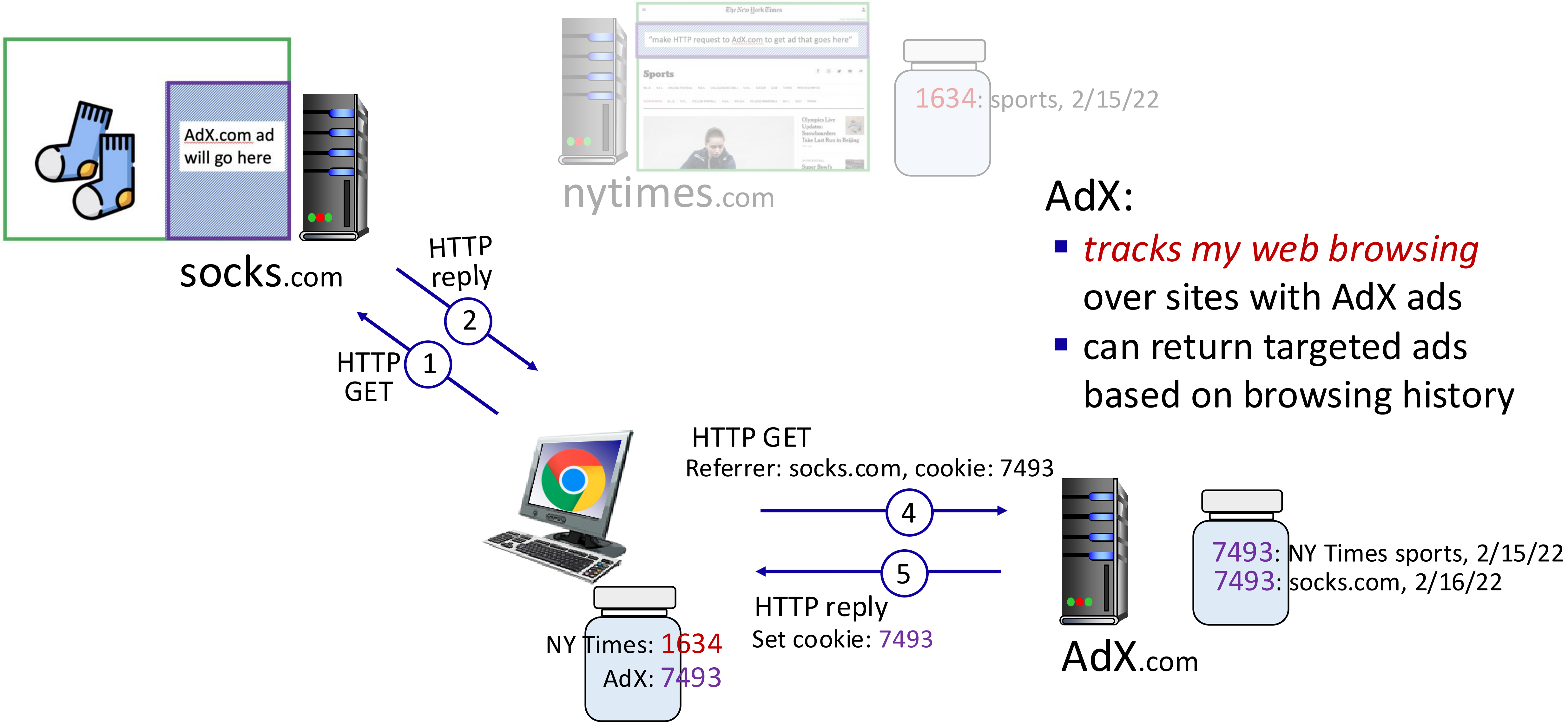


NY times page with embedded ad displayed

Cookies: tracking a user's browsing behavior



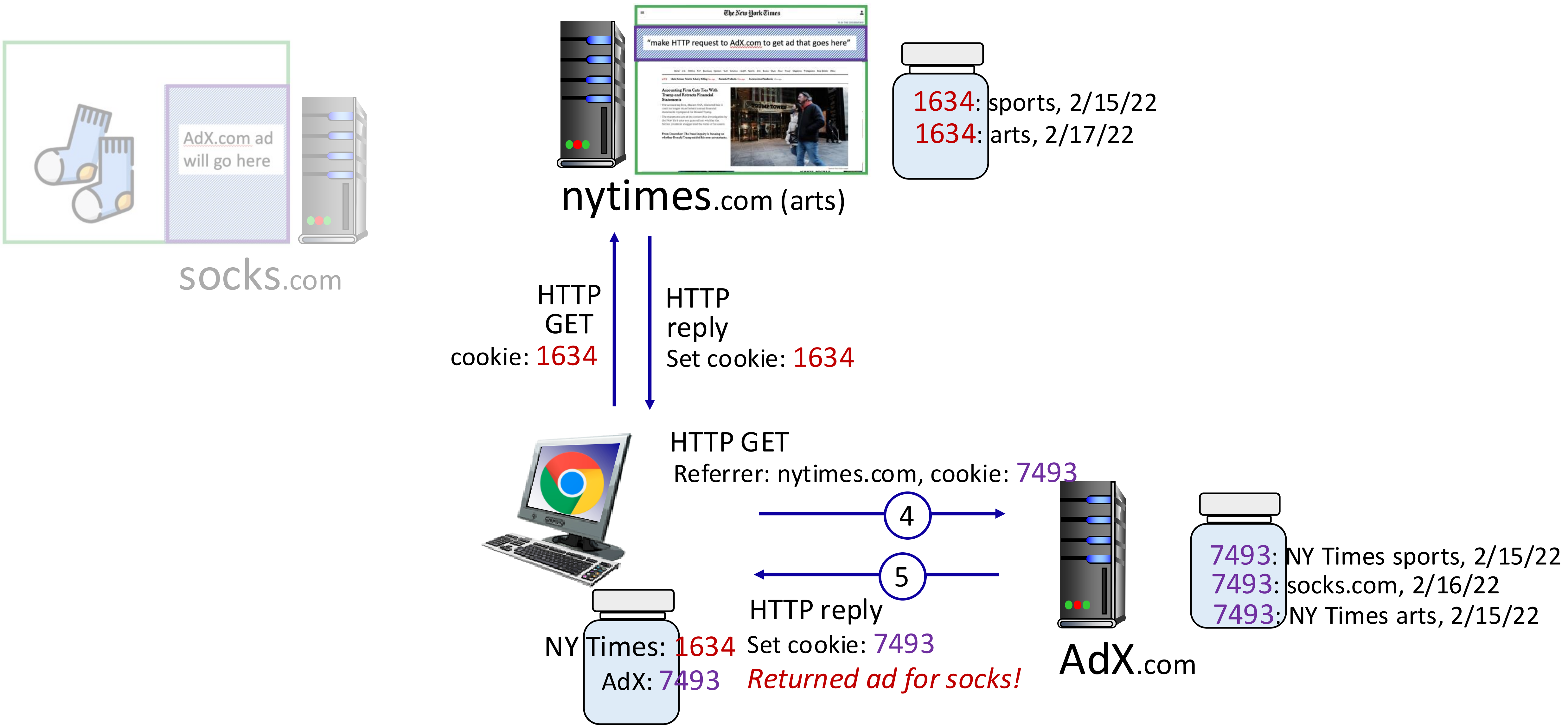
Cookies: tracking a user's browsing behavior



AdX:

- *tracks my web browsing* over sites with AdX ads
- can return targeted ads based on browsing history

Cookies: tracking a user's browsing behavior (one day later)



Cookies: tracking a user's browsing behavior

Cookies can be used to:

- track user behavior on a given website (first party cookies)
- track user behavior across multiple websites (third party cookies) without user ever choosing to visit tracker site (!)
- tracking may be *invisible* to user:
  - rather than displayed ad triggering HTTP GET to tracker, could be an invisible link

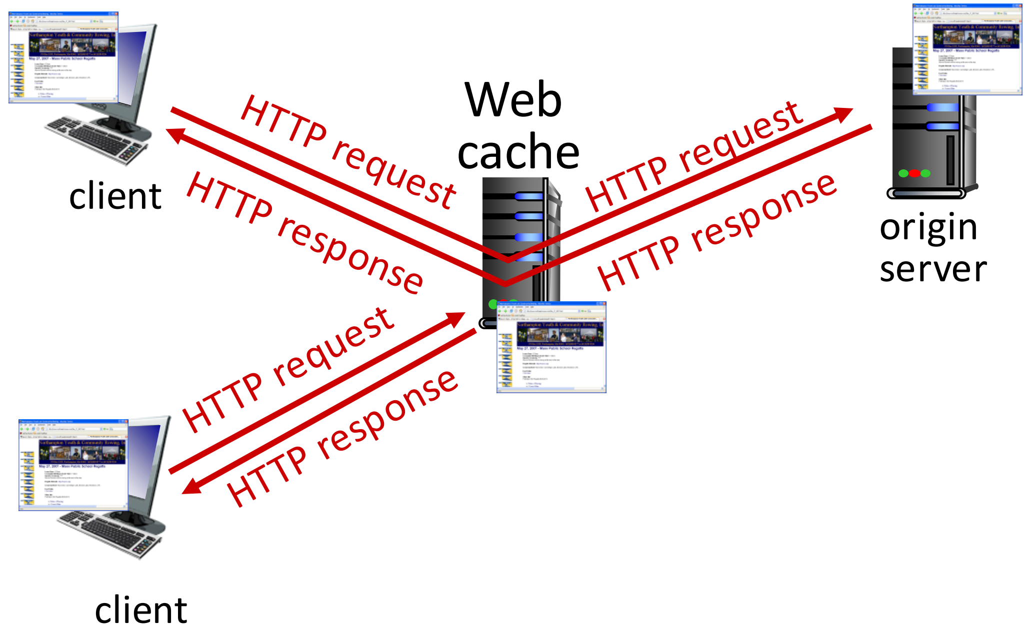
third party tracking via cookies:

- disabled by default in Firefox, Safari browsers
- to be disabled in Chrome browser in 2023

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

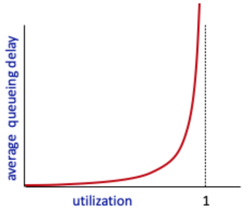
Scenario:

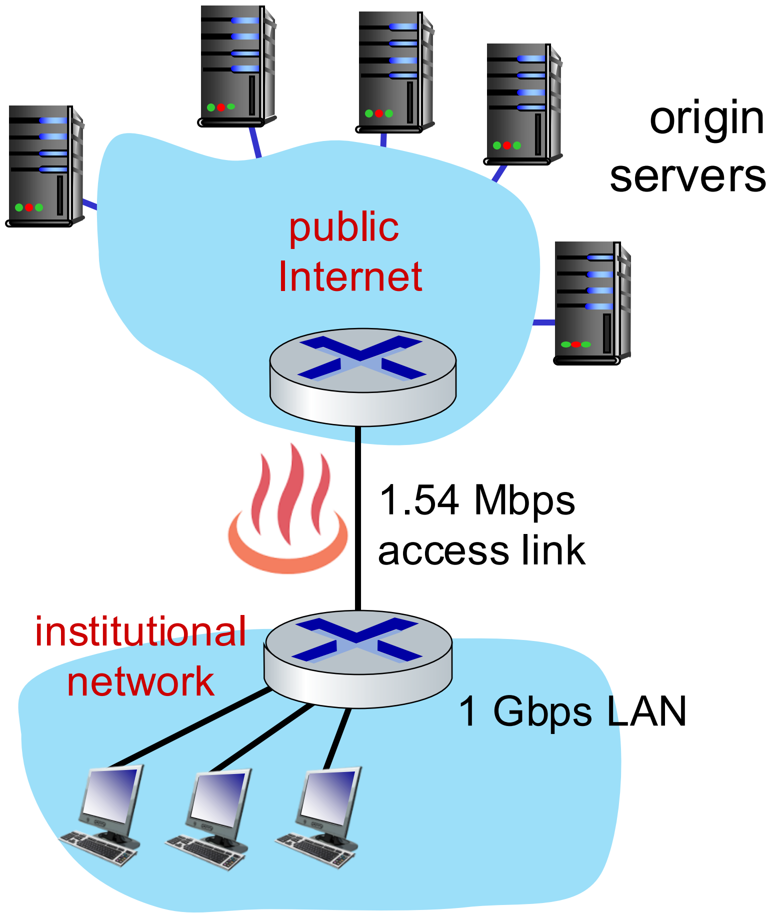
- access link rate: 1.54 Mbps
- RTT from institutional router to server: 2 sec
- web object size: 100K bits
- average request rate from browsers to origin servers: 15/sec
  - avg data rate to browsers: 1.50 Mbps

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!

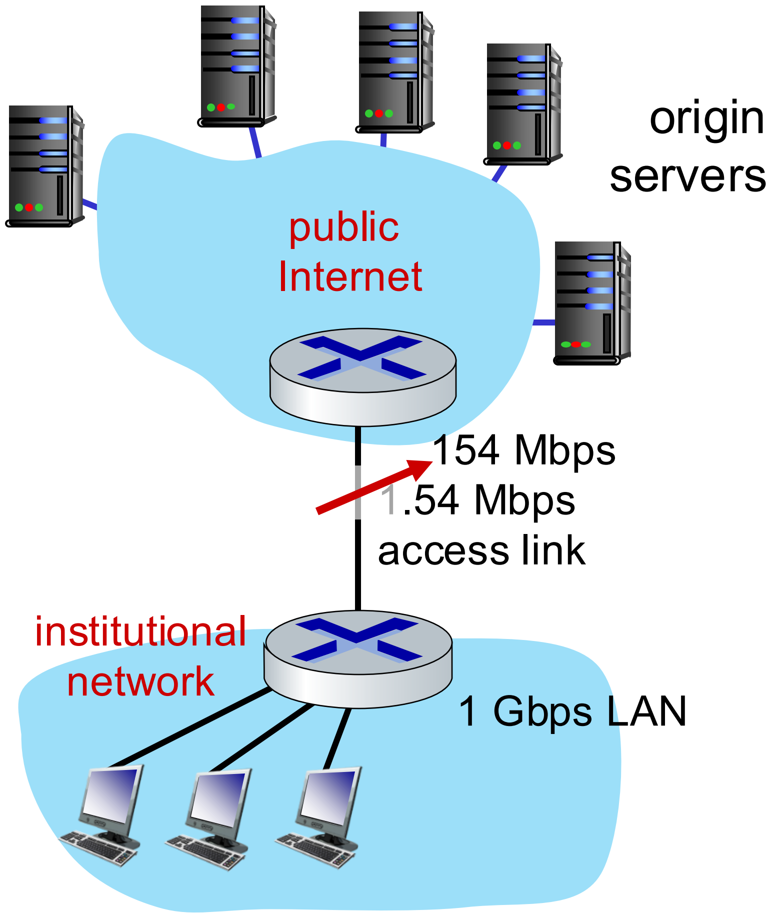




Option 1: buy a faster access link

Scenario:

- access link rate: ~~1.54~~ 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~~ msecs + usecs

Cost: faster access link (expensive!)

Option 2: install a web cache

Scenario:

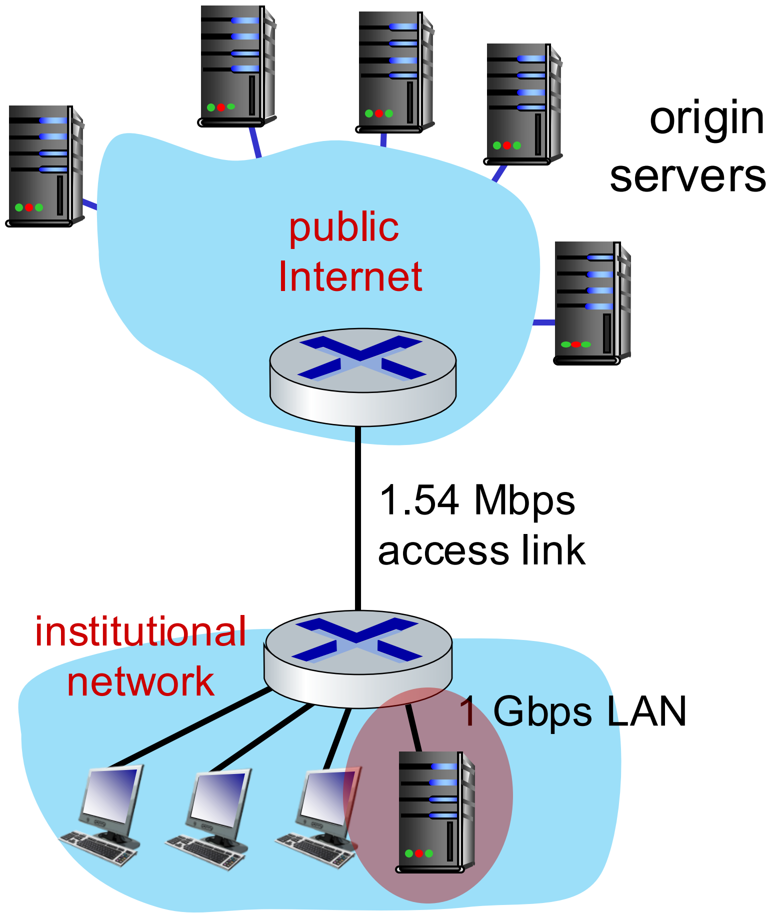
- access link rate: 1.54 Mbps
- RTT from institutional router to server: 2 sec
- web object size: 100K bits
- average request rate from browsers to origin servers: 15/sec
  - avg data rate to browsers: 1.50 Mbps

Cost: web cache (cheap!)

Performance:

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

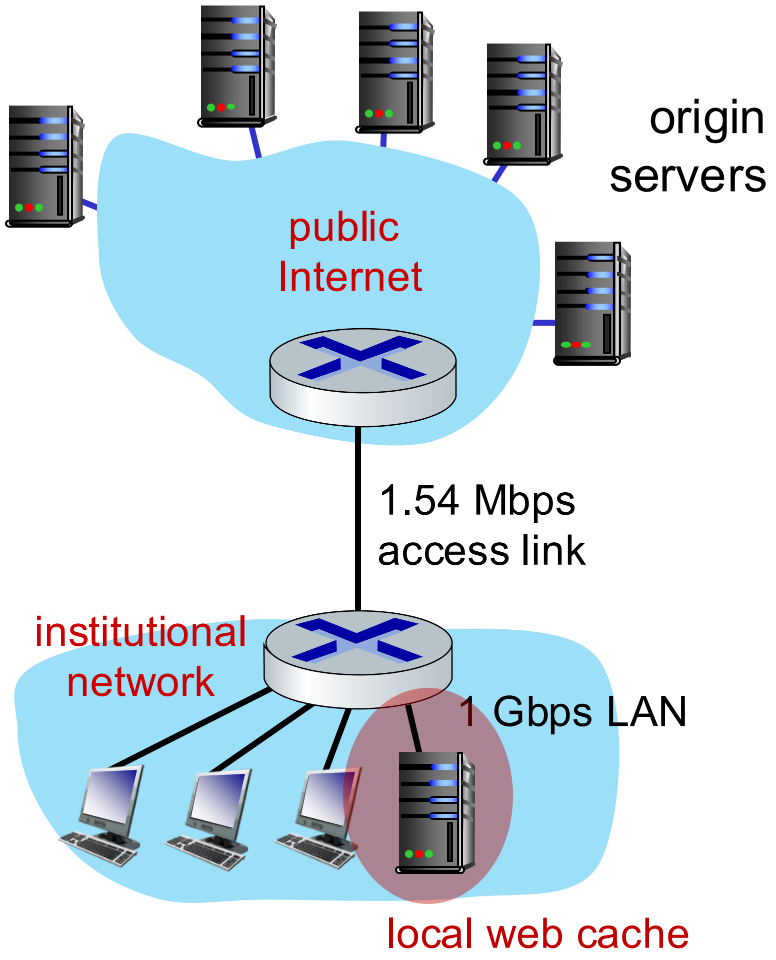
How to compute link utilization, delay?



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 secs) + 0.4 (~msecs) = ~ 1.2 secs



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

DNS: Domain Name System

people: many identifiers:
- name, passport #

Internet hosts, routers:
- IP address (32 bit) - used for addressing datagrams
- "name", e.g., cs.hitsz.edu.cn - 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!

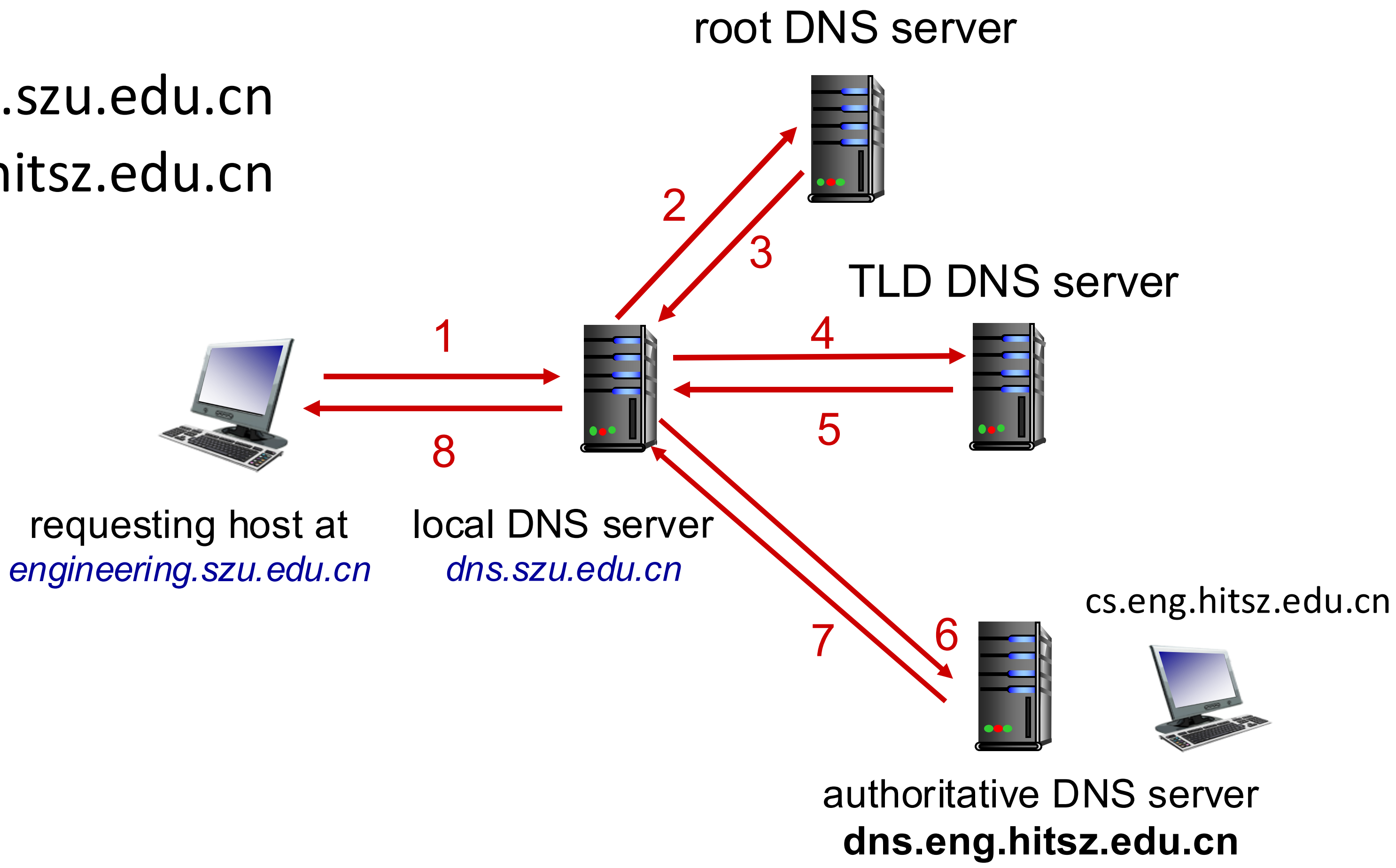
- Comcast DNS servers alone: 600B DNS queries/day
- Akamai DNS servers alone: 2.2T DNS queries/day

DNS name resolution: iterated query

Example: host at engineering.szu.edu.cn wants IP address for cs.eng.hitsz.edu.cn

Iterated query:

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

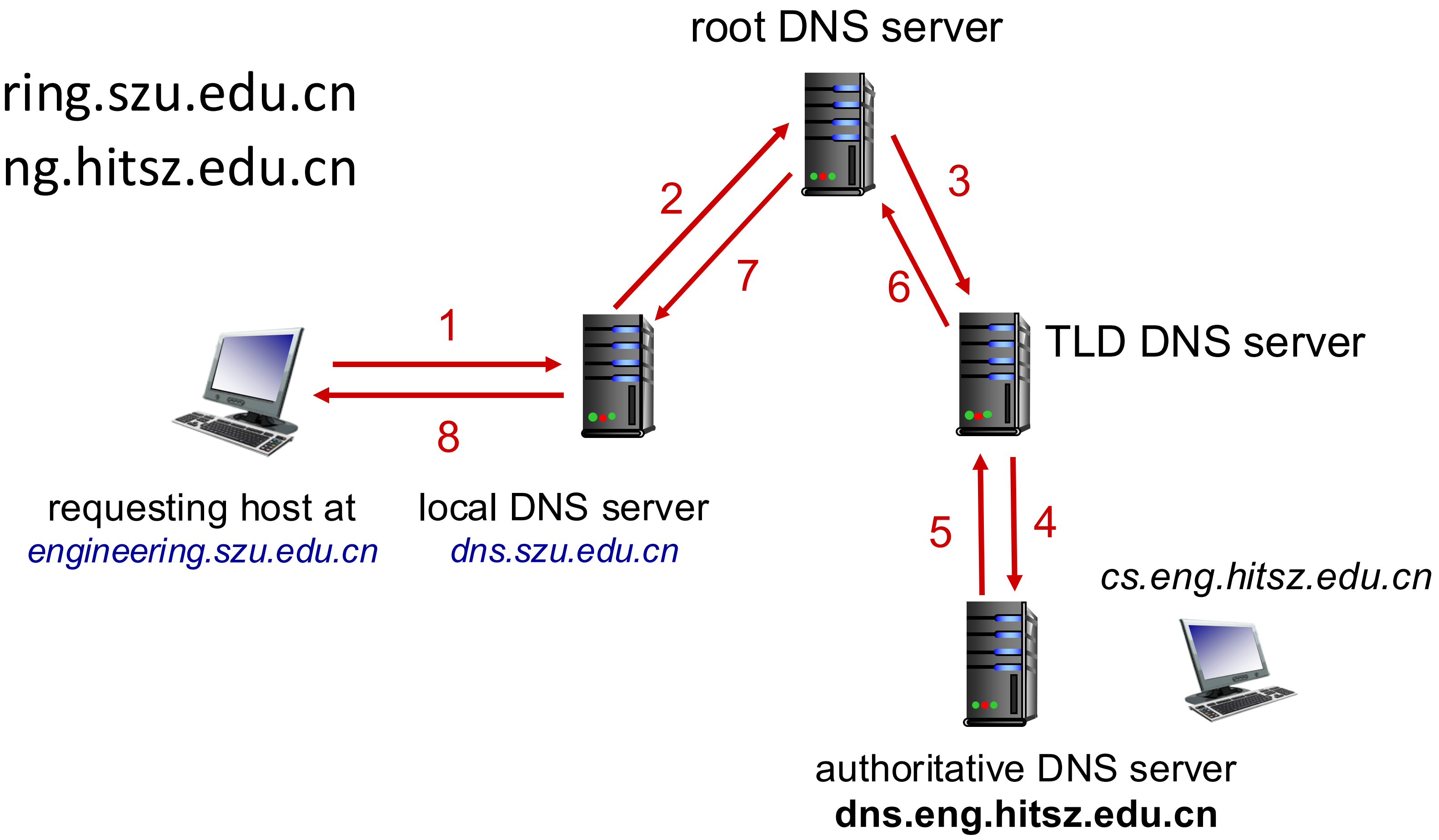


DNS name resolution: recursive query

Example: host at engineering.szu.edu.cn wants IP address for cs.eng.hitsz.edu.cn

Recursive query:

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



DNS records

DNS: distributed database storing resource records (RR)

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

type=A
- `name` is hostname
- `value` is IP address

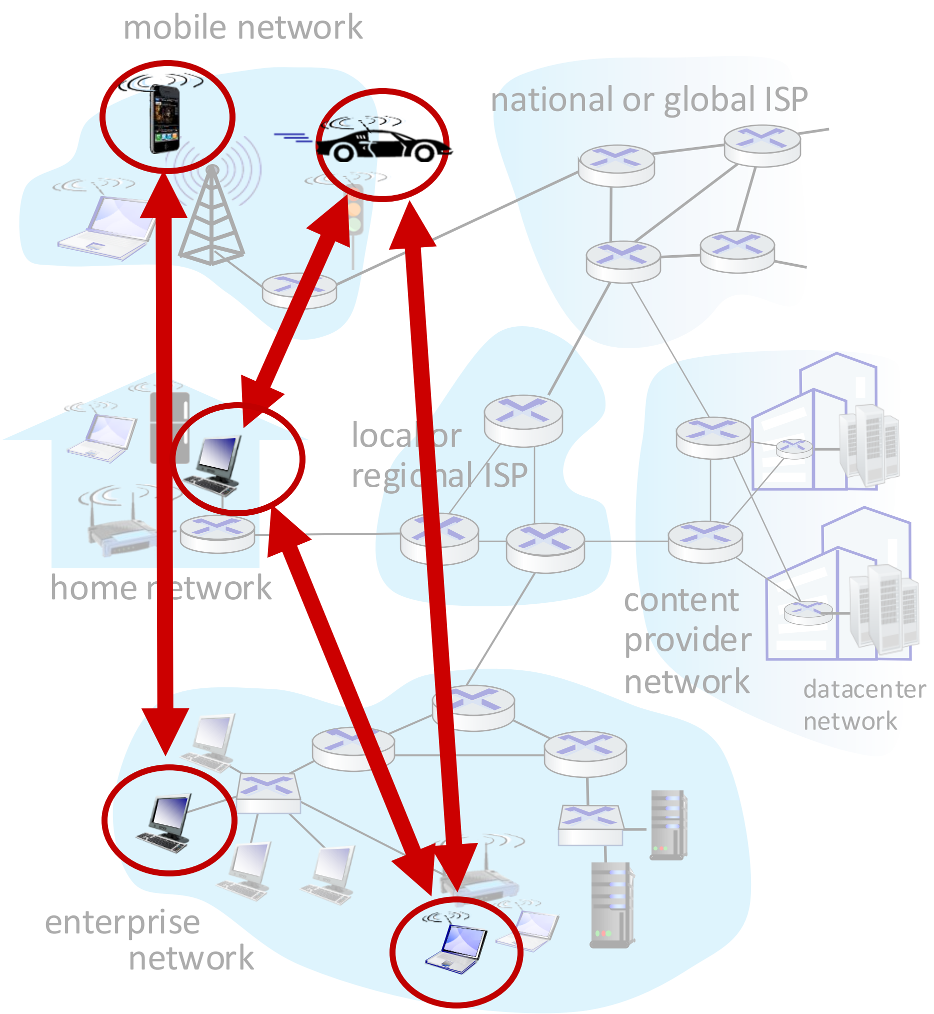
type=NS
- `name` is domain (e.g., foo.com)
- `value` is hostname of authoritative name server for this domain

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`

Peer-to-peer (P2P) architecture

- *no* always-on server
- arbitrary end systems directly communicate
- peers request service from other peers, provide service in return to other peers
  - *self scalability* – new peers bring new service capacity, and new service demands
- peers are intermittently connected and change IP addresses
  - complex management
- examples: P2P file sharing (BitTorrent), streaming (KanKan), VoIP (Skype)



Chapter 3
Transport Layer

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

Internet checksum: weak protection!

example: add two 16-bit integers

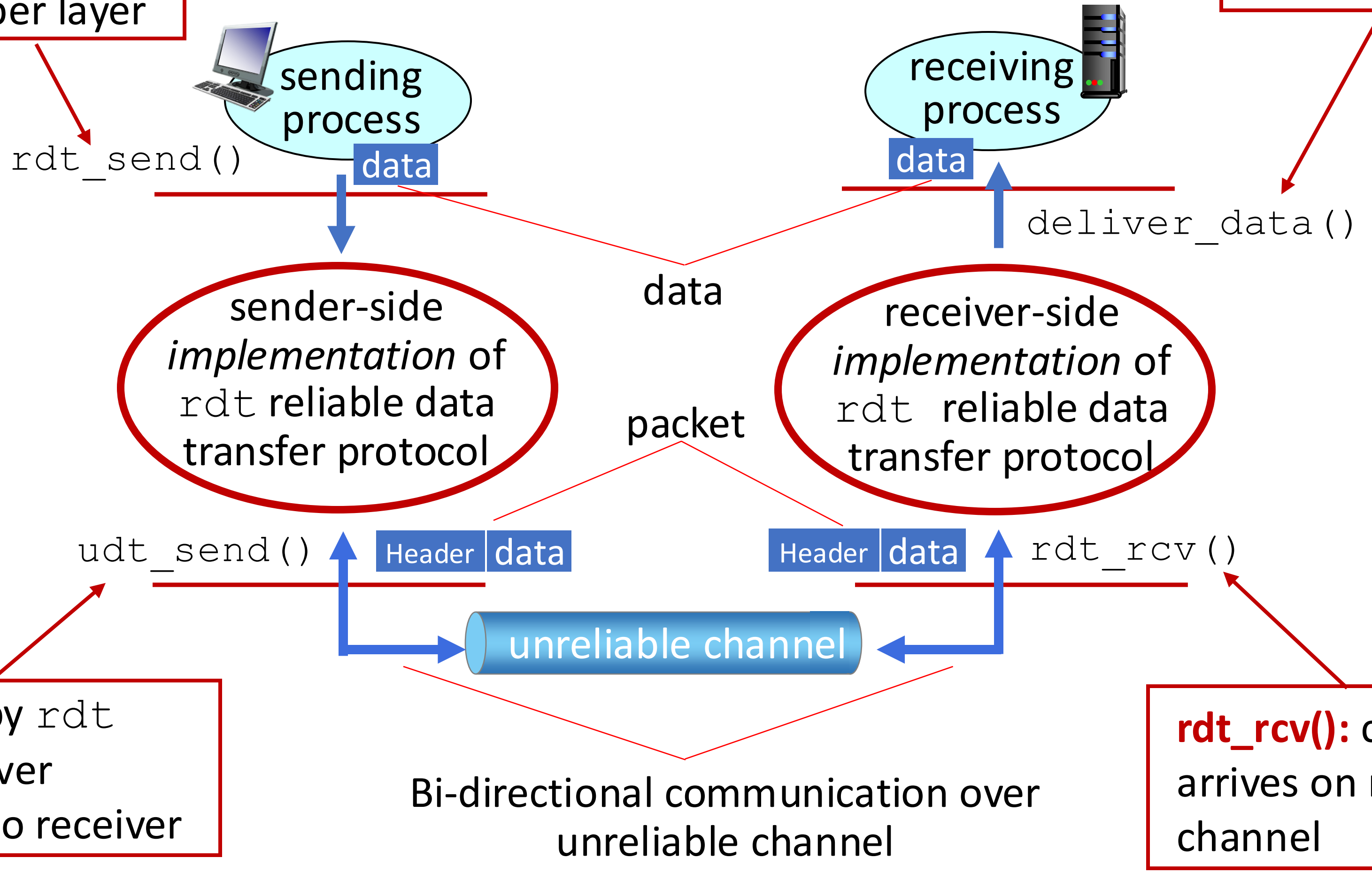
	1 1 1 0 0 1 1 0 0 1 1 0 0 1 1 0	0 1
	1 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1	1 0
wraparound	1 1 0 1 1 1 0 1 1 1 0 1 1 1 0 1 1	
sum	1 0 1 1 1 0 1 1 1 0 1 1 1 1 0 0	
checksum	0 1 0 0 0 1 0 0 0 1 0 0 0 0 1 1	

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

Reliable data transfer protocol (rdt): interfaces

rdt_send(): called from above, (e.g., by app.). Passed data to deliver to receiver upper layer

deliver_data(): called by `rdt` to deliver data to upper layer



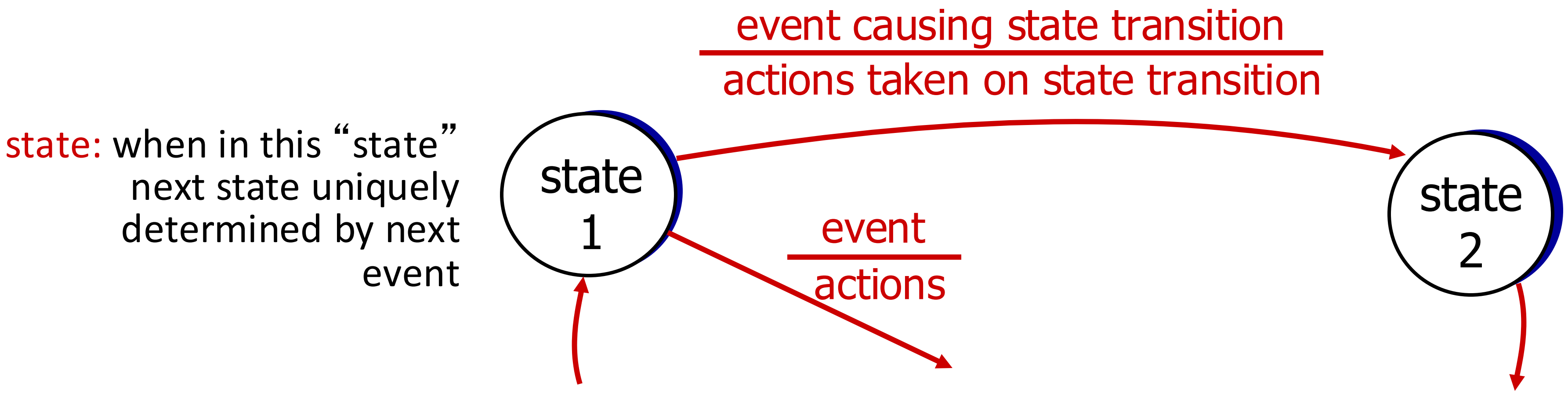
udt_send(): called by `rdt` to transfer packet over unreliable channel to receiver

rdt_rcv(): called when packet arrives on receiver side of channel

Reliable data transfer: getting started

We will:

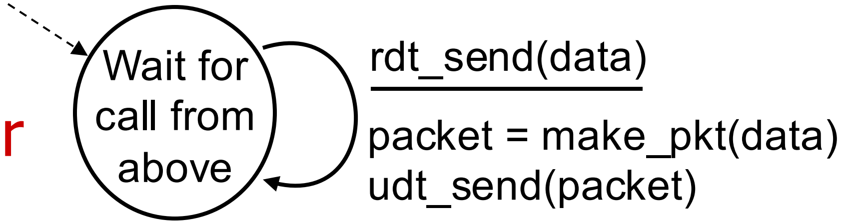
- incrementally develop sender, receiver sides of reliable data transfer protocol (`rdt`)
- consider only unidirectional data transfer
  - but control info will flow in both directions!
- use finite state machines (FSM) to specify sender, receiver



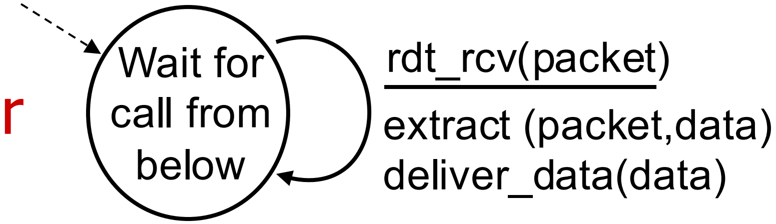
rdt1.0: reliable transfer over a reliable channel

- 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?

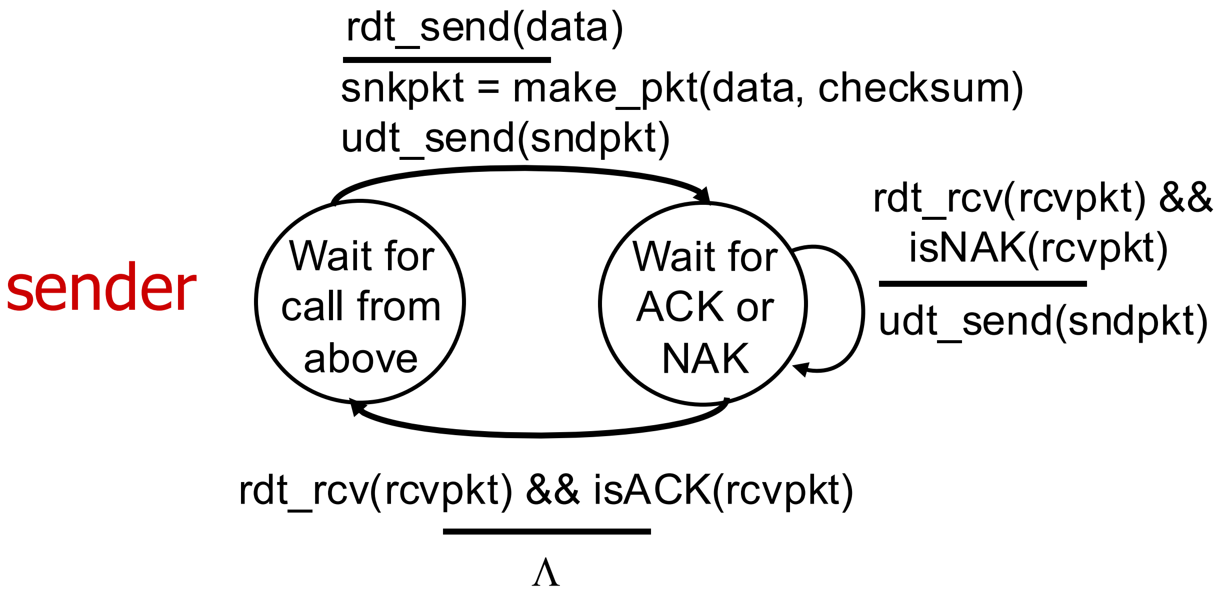
rdt2.0: channel with bit errors

- 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

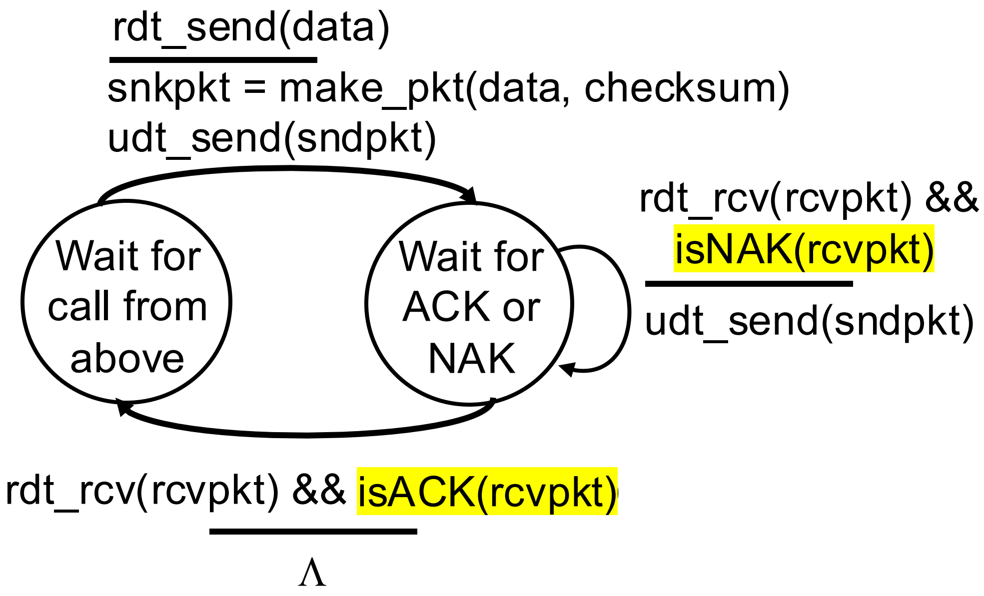
sender sends one packet, then waits for receiver response

rdt2.0: FSM specifications



rdt2.0: FSM specification

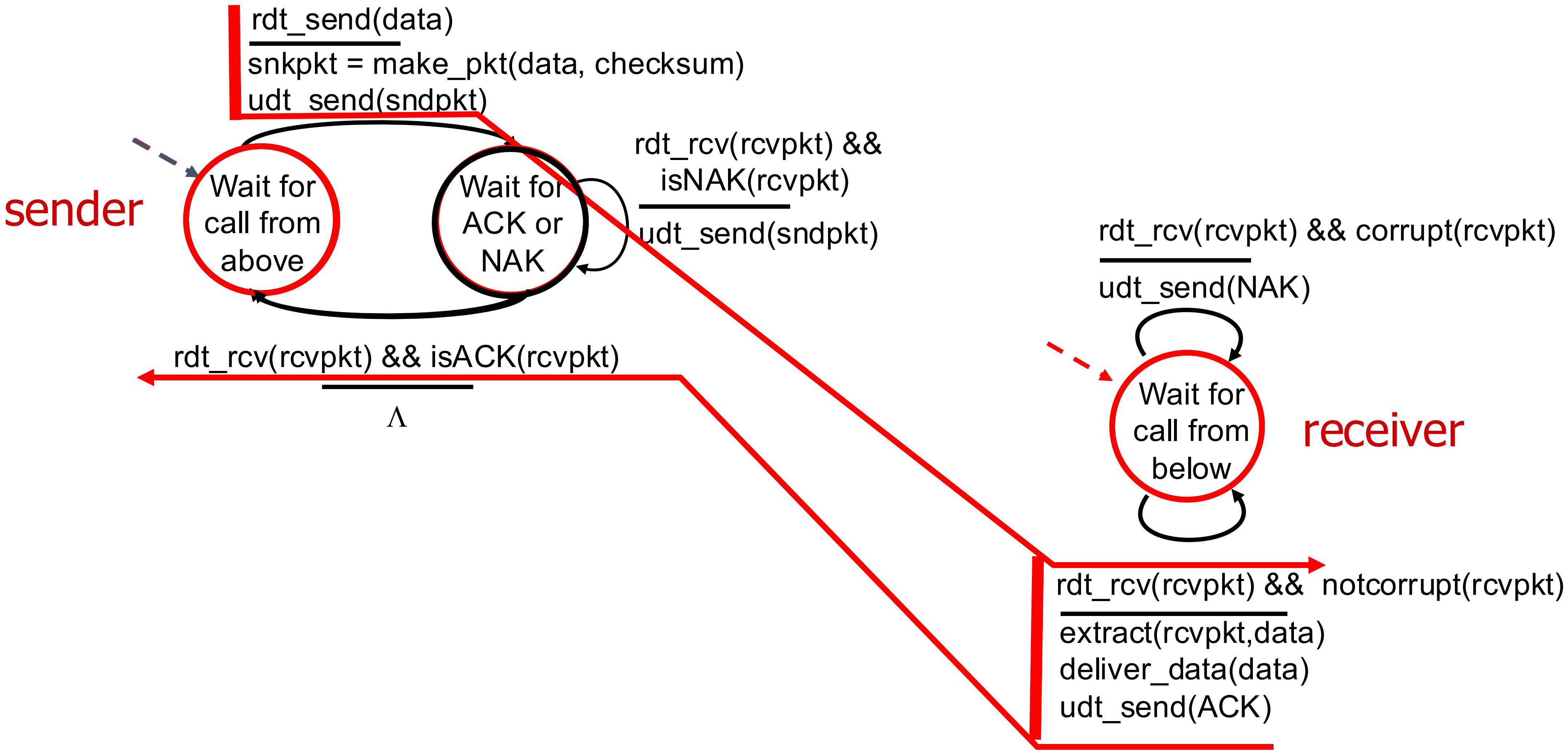
sender



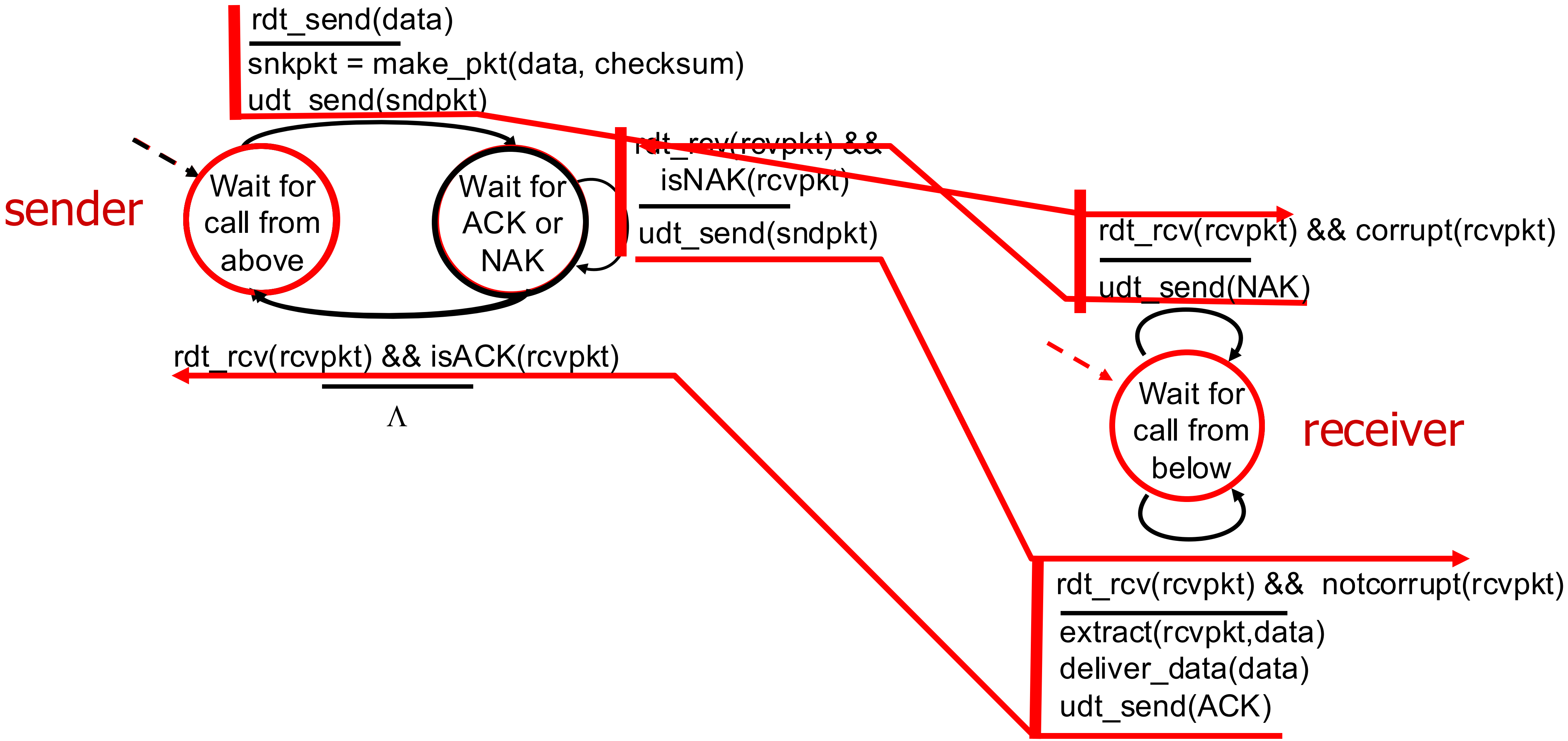
Note: “state” of receiver (did the receiver get my message correctly?) isn’t known to sender unless somehow communicated from receiver to sender

- 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

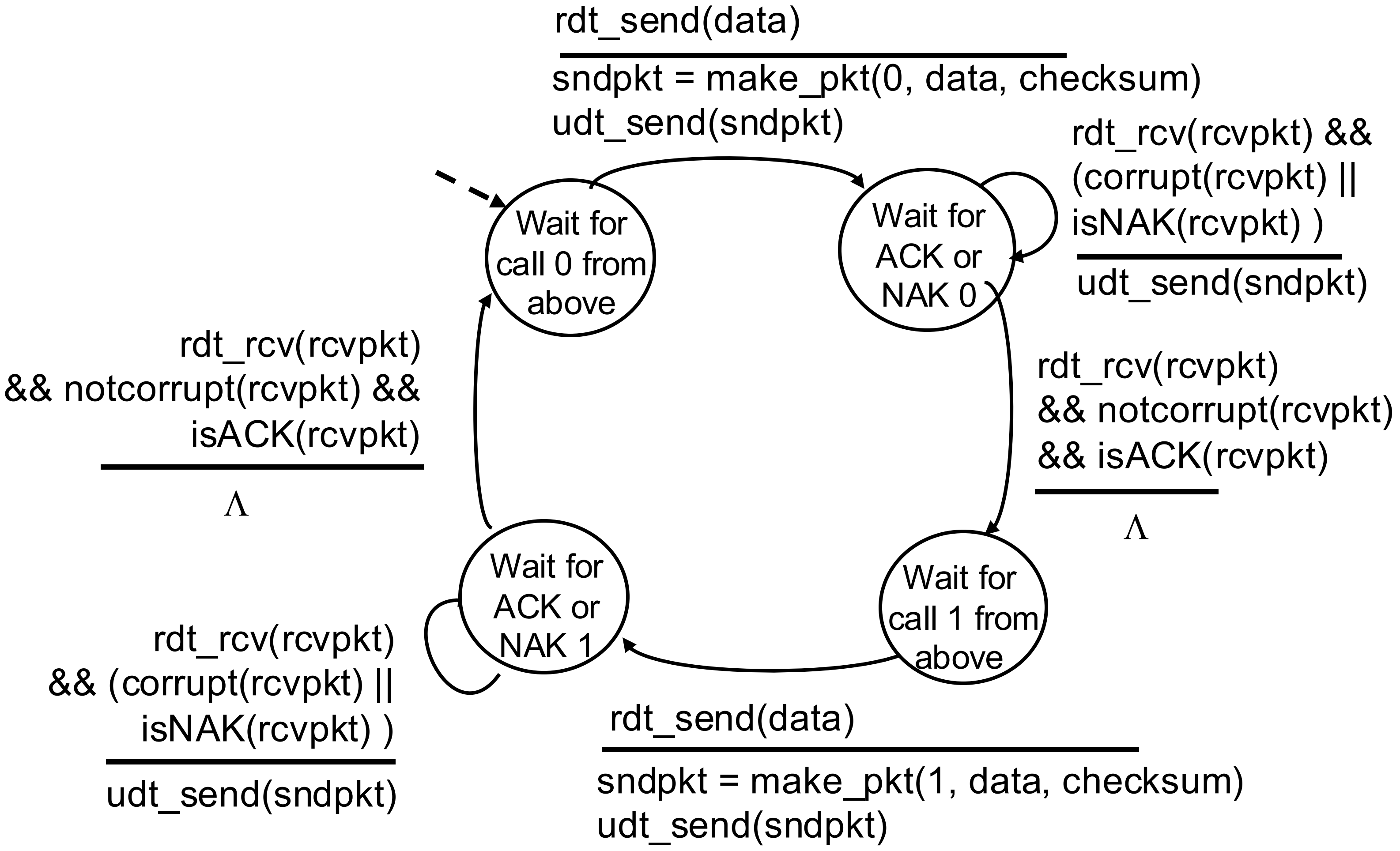
handling duplicates:

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

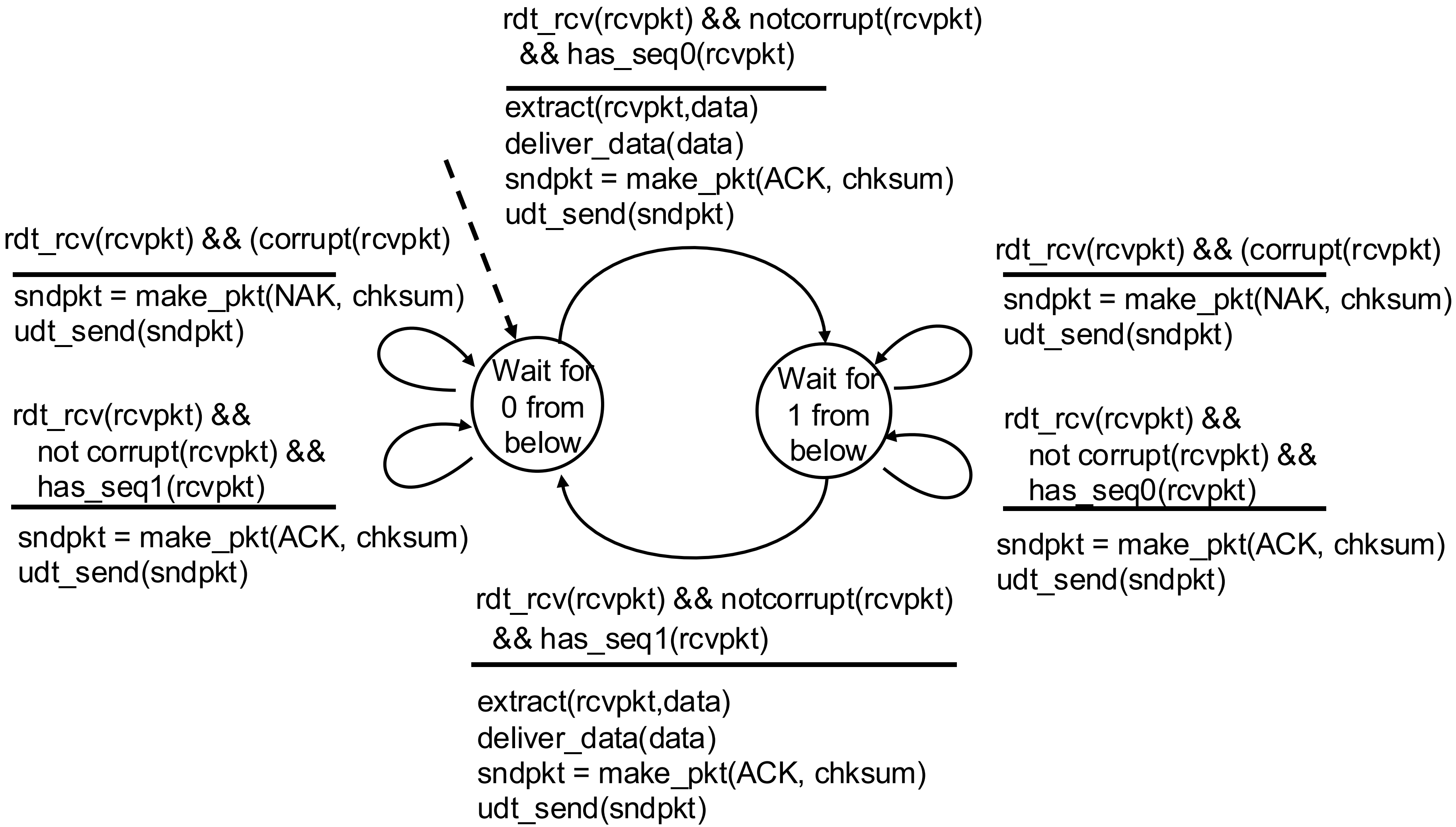
stop and wait

sender sends one packet, then waits for receiver response

rdt2.1: sender, handling garbled ACK/NAKs



rdt2.1: receiver, handling garbled ACK/NAKs



rdt2.1: discussion

sender:

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

receiver:

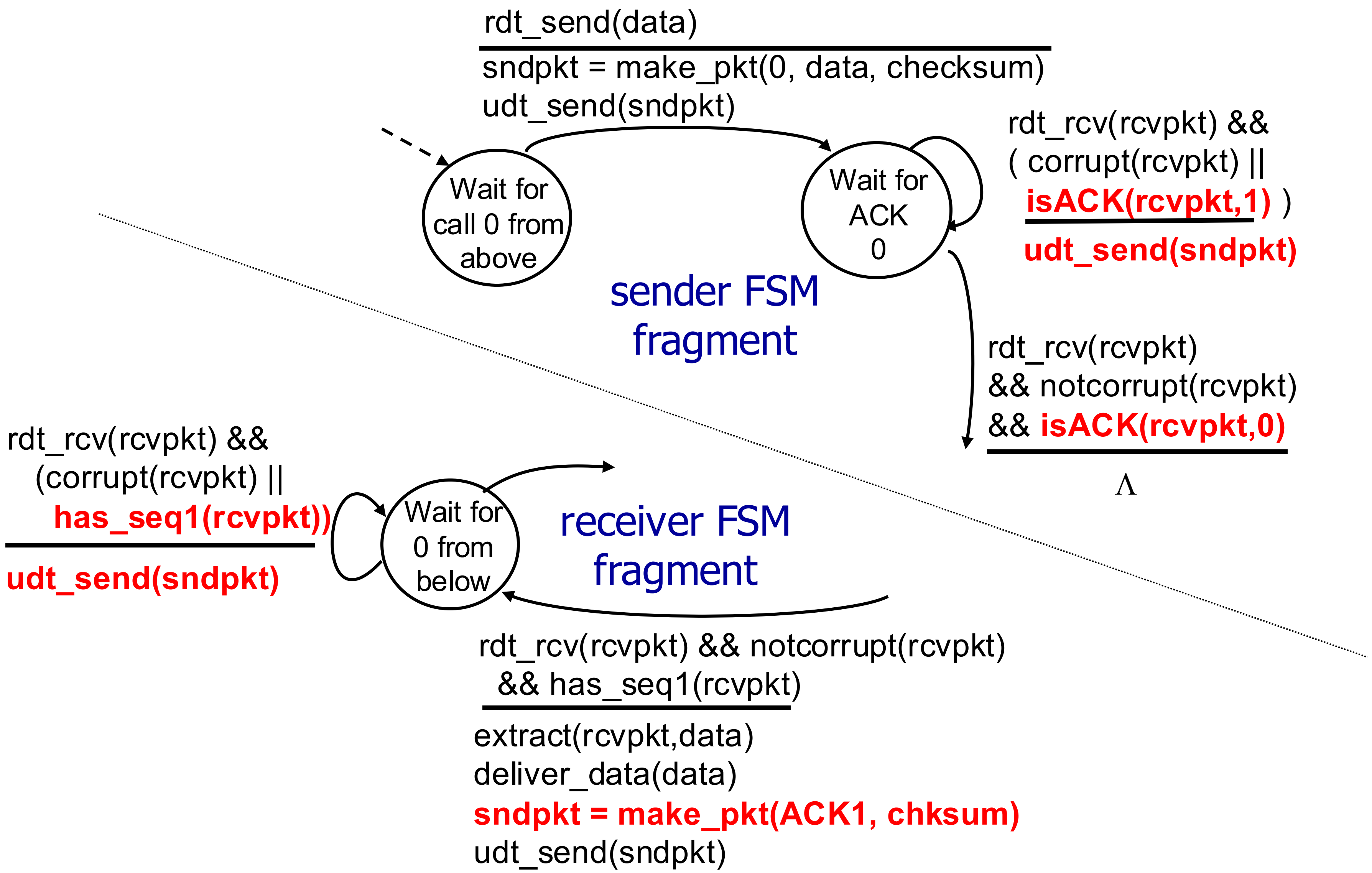
- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

rdt2.2: a NAK-free protocol

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

As we will see, TCP uses this approach to be NAK-free

rdt2.2: sender, receiver fragments



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

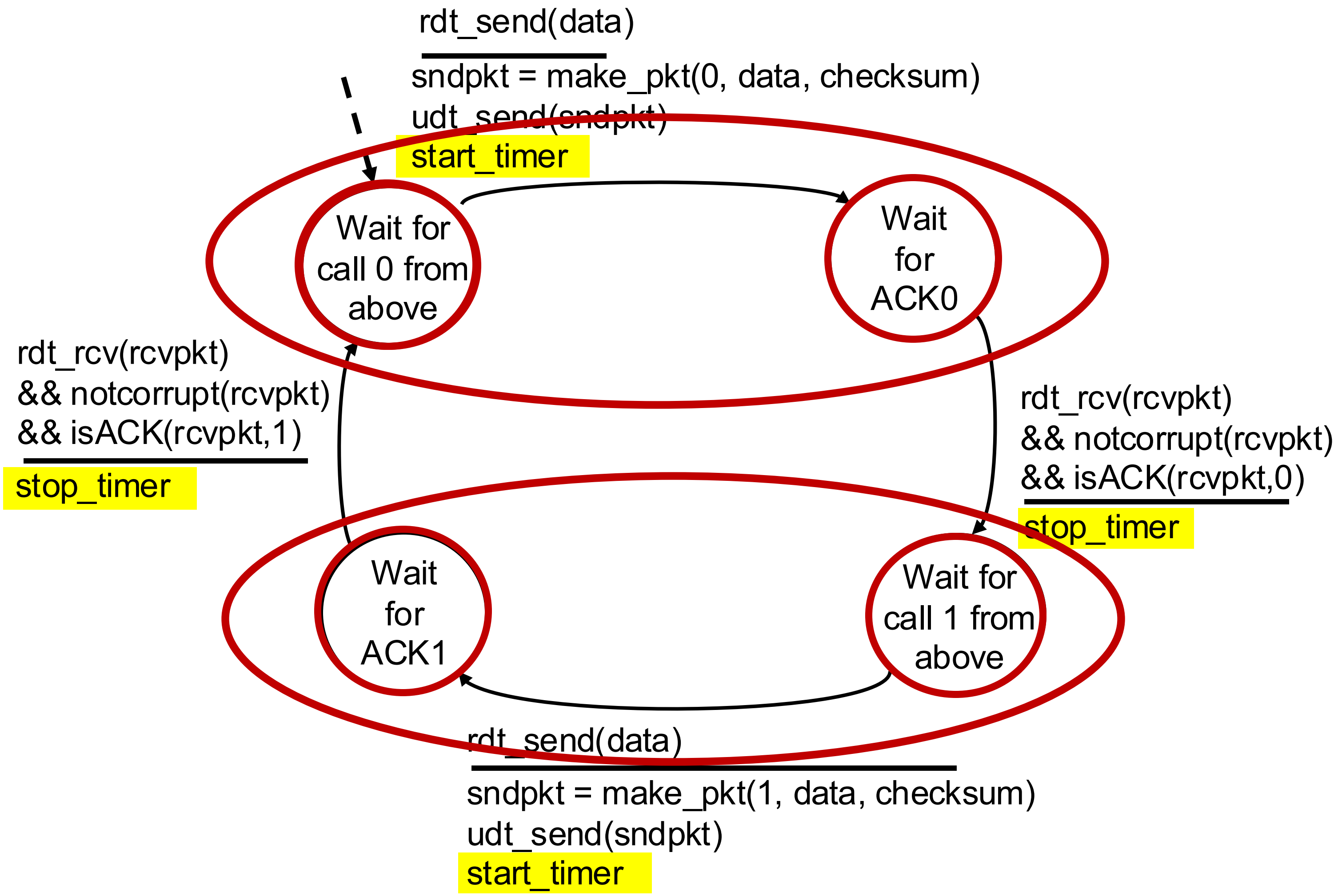
Q: How do *humans* handle lost sender-to-receiver words in conversation?

rdt3.0: channels with errors *and* loss

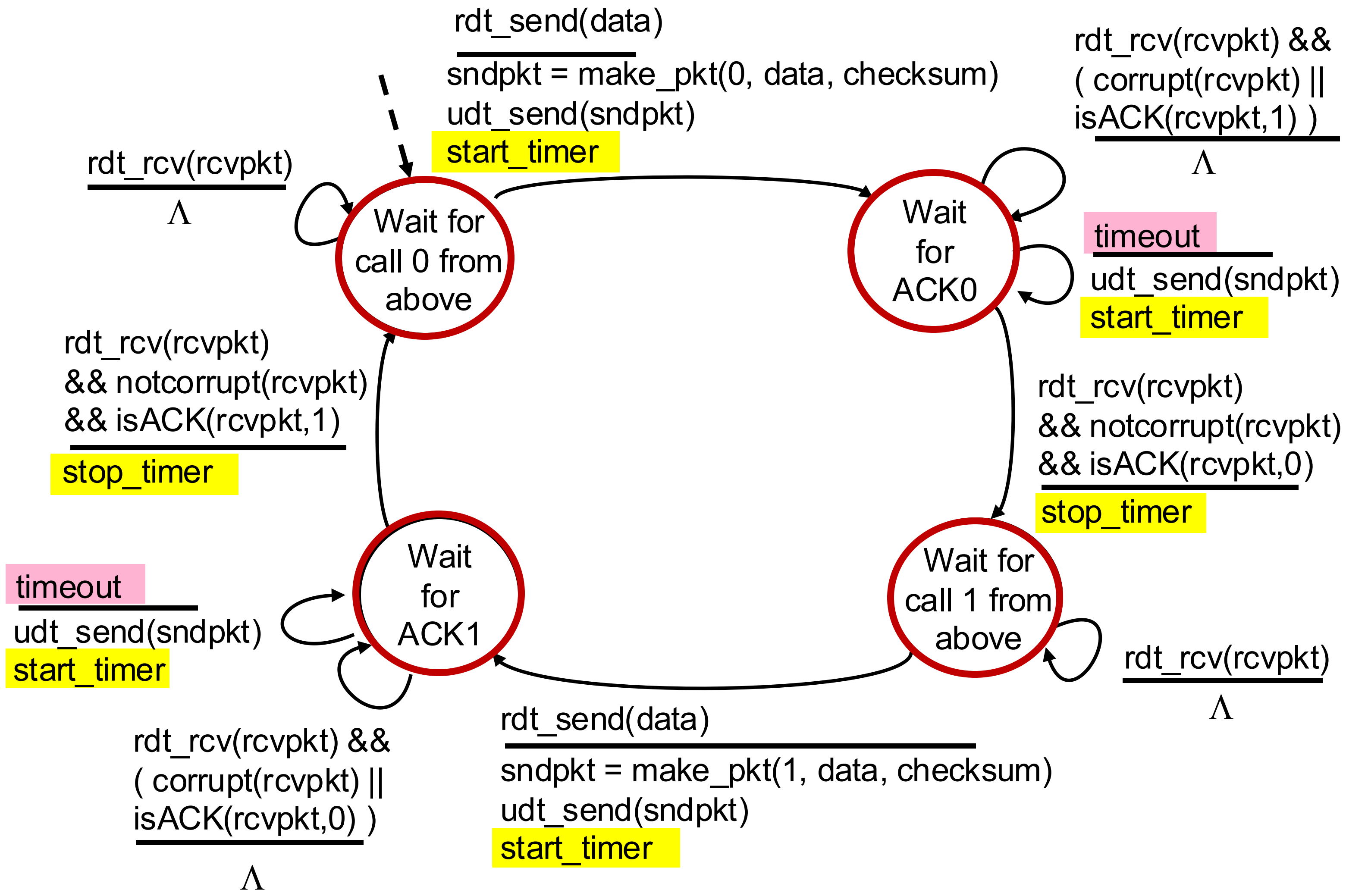
Approach: sender waits "reasonable" amount of time for ACK

- 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

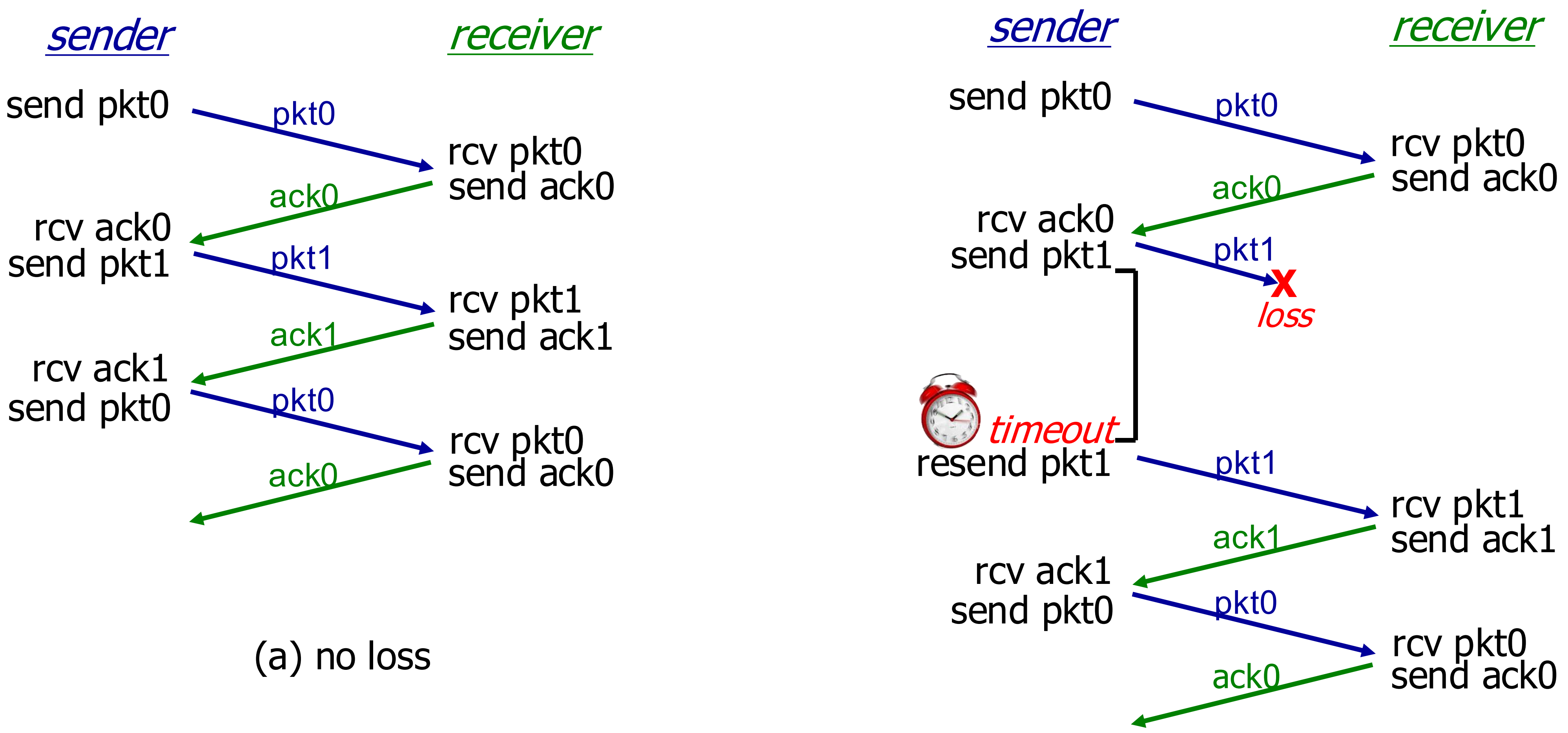
rdt3.0 sender



rdt3.0 sender



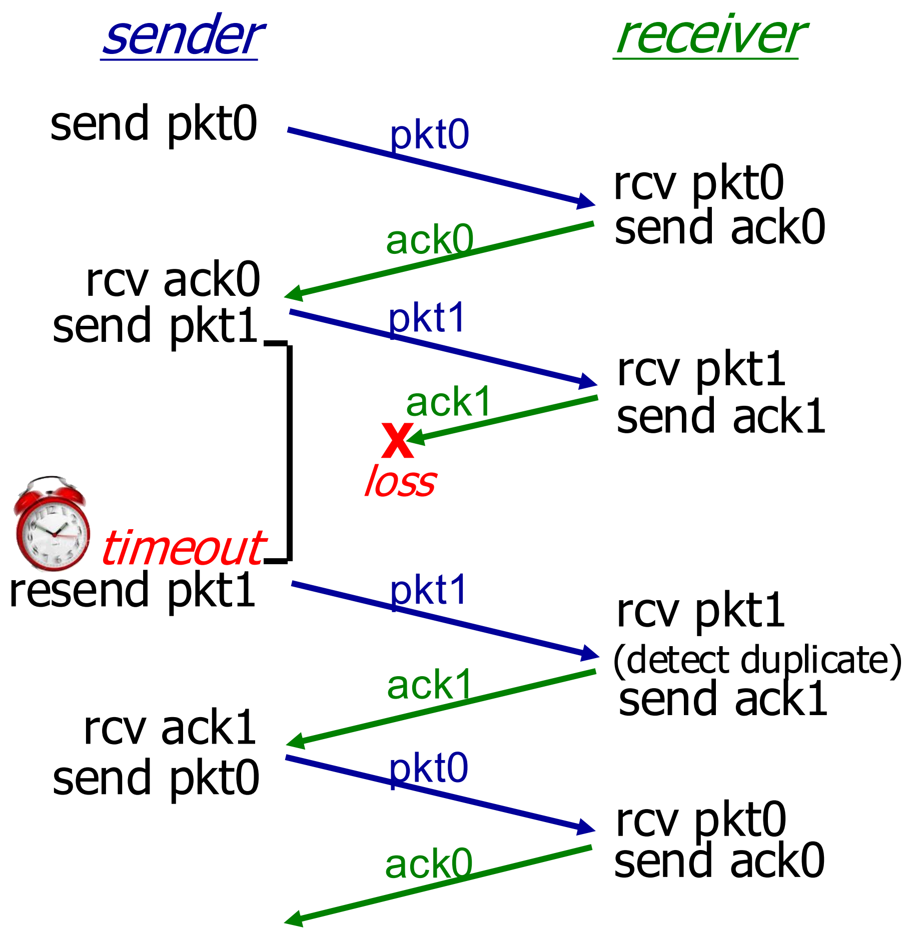
rdt3.0 in action



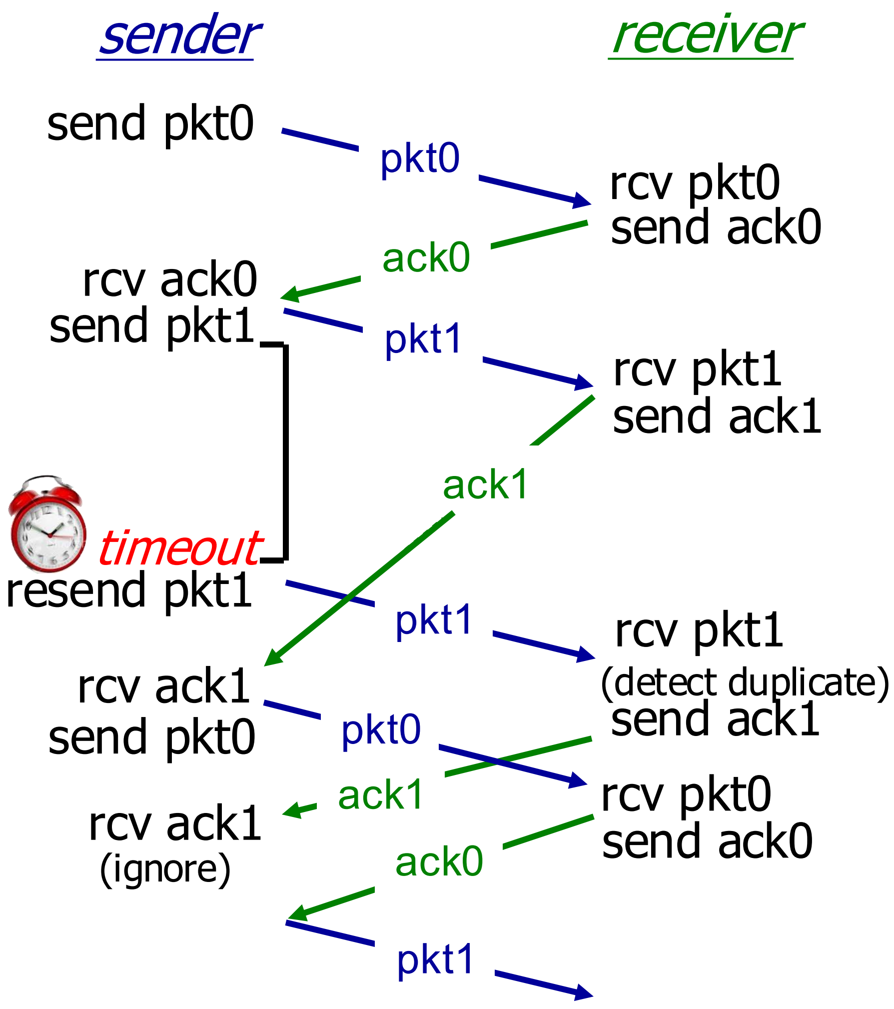
(a) no loss

(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

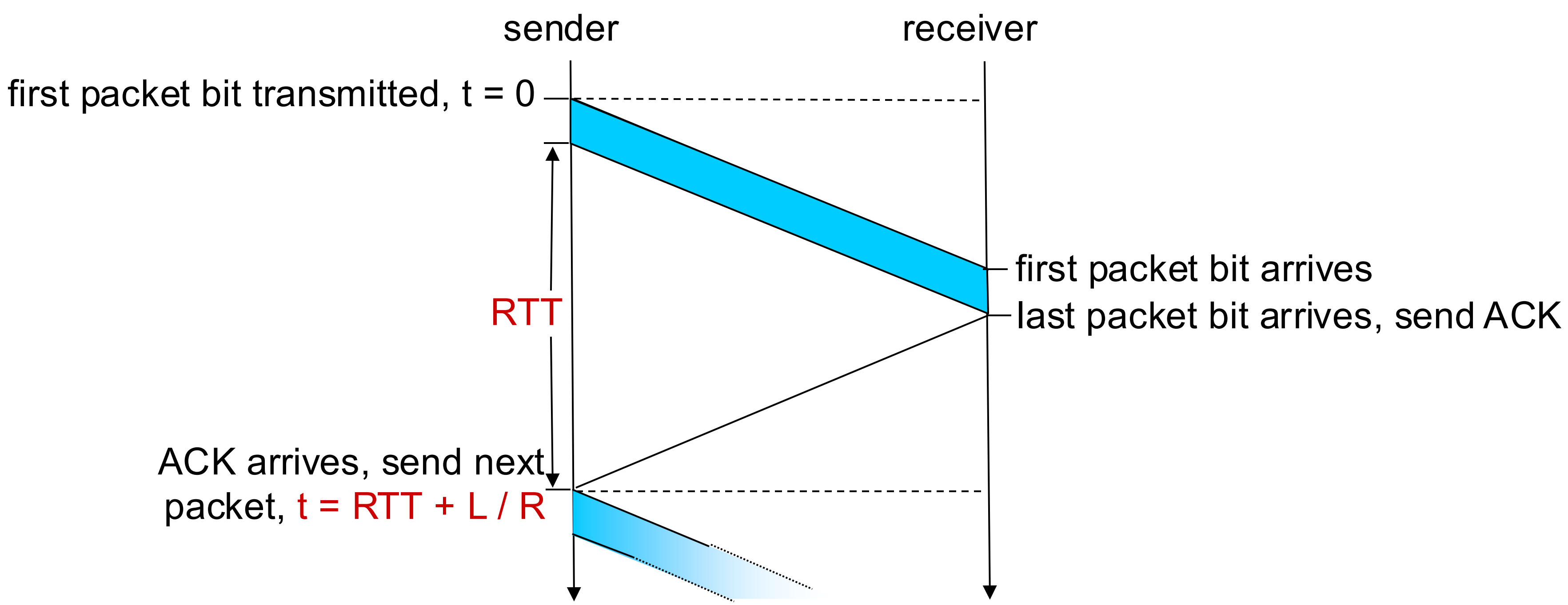
Performance of rdt3.0 (stop-and-wait)

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

- example: 1 Gbps link, 30 ms RTT, 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

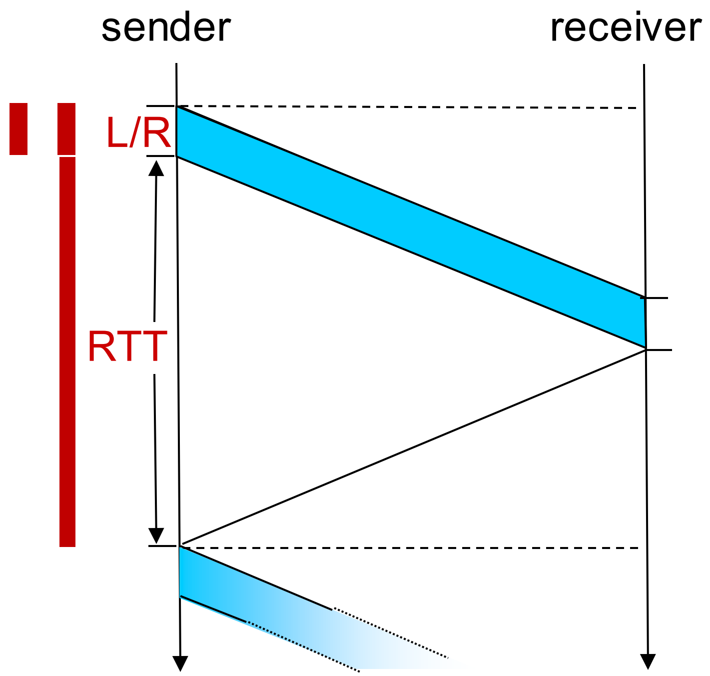


rdt3.0: stop-and-wait operation

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

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

$$= 0.00027$$

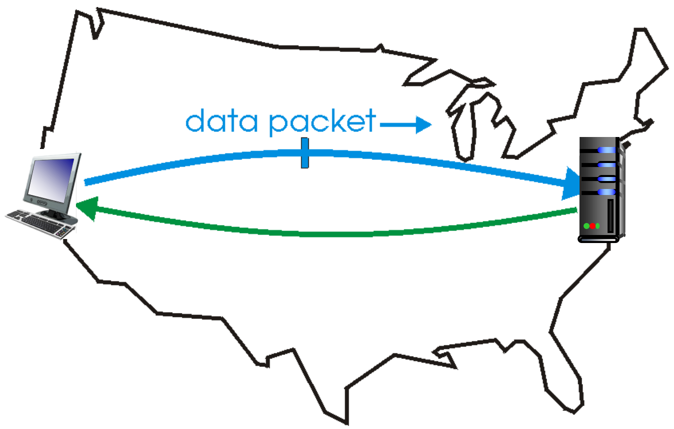


- rdt 3.0 protocol performance stinks!
- Protocol limits performance of underlying infrastructure (channel)

rdt3.0: pipelined protocols operation

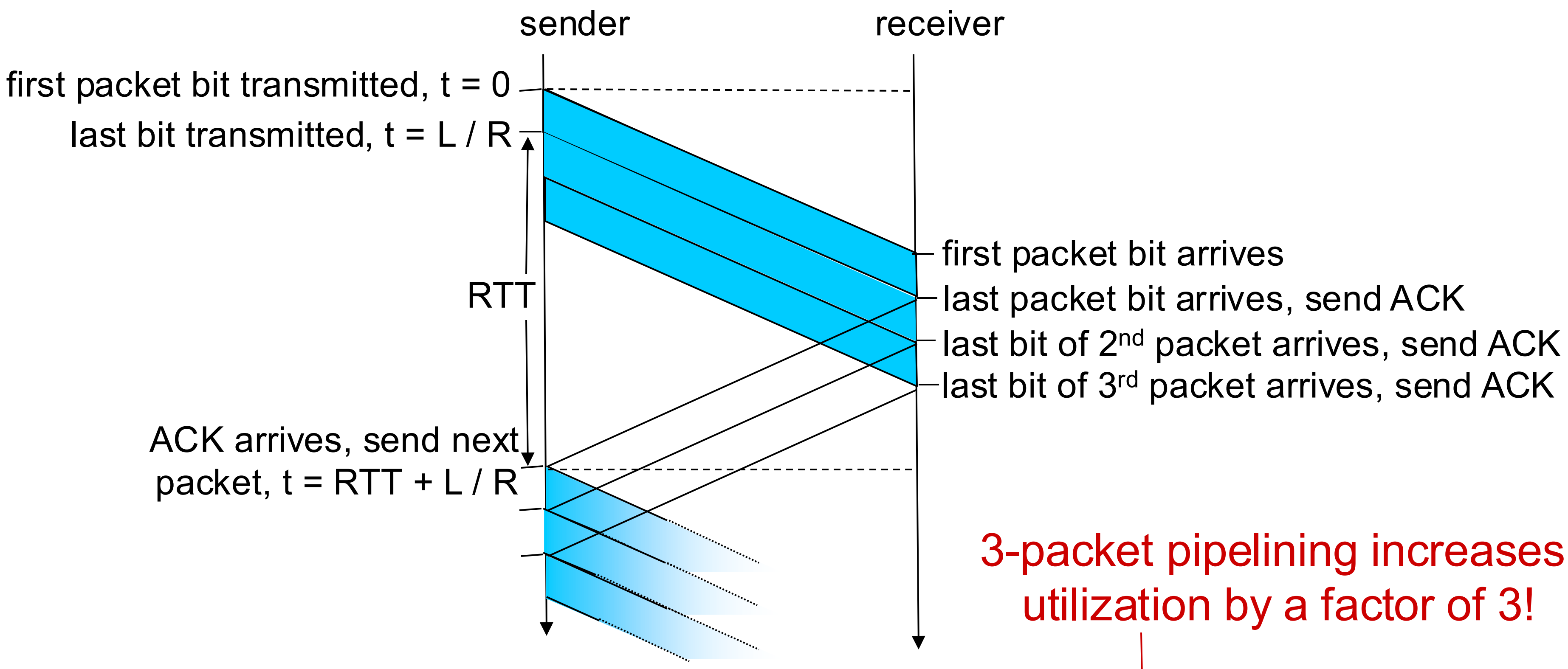
pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged packets

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



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

Pipelining: increased utilization

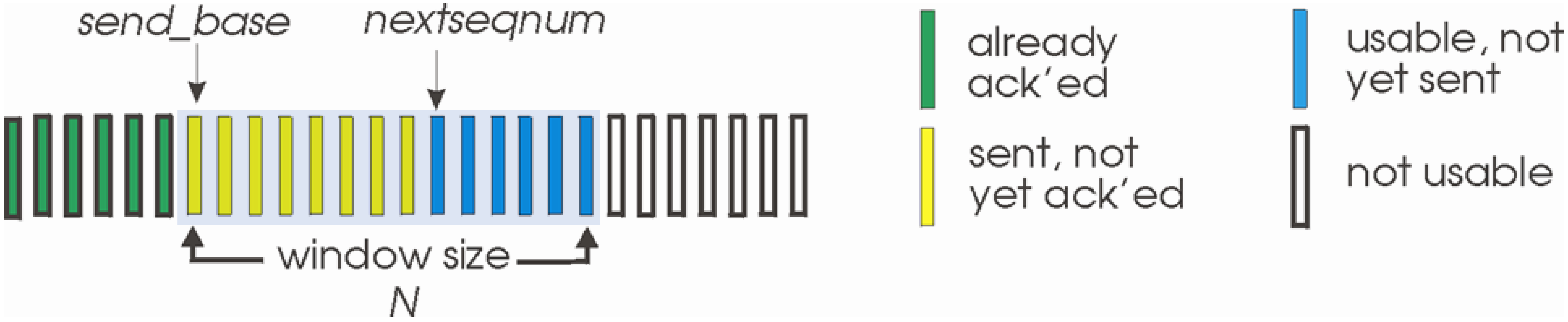


3-packet pipelining increases utilization by a factor of 3!

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

Go-Back-N: sender

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



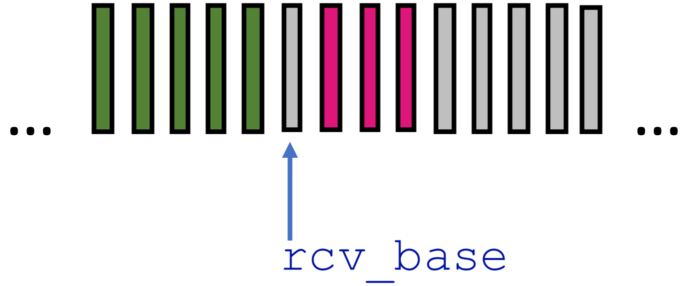
- *cumulative ACK:* ACK(*n*): ACKs all packets up to, including seq # *n*
  - on receiving ACK(*n*): move window forward to begin at *n+1*
- timer for oldest in-flight packet
- *timeout(n):* retransmit packet n and all higher seq # packets in window

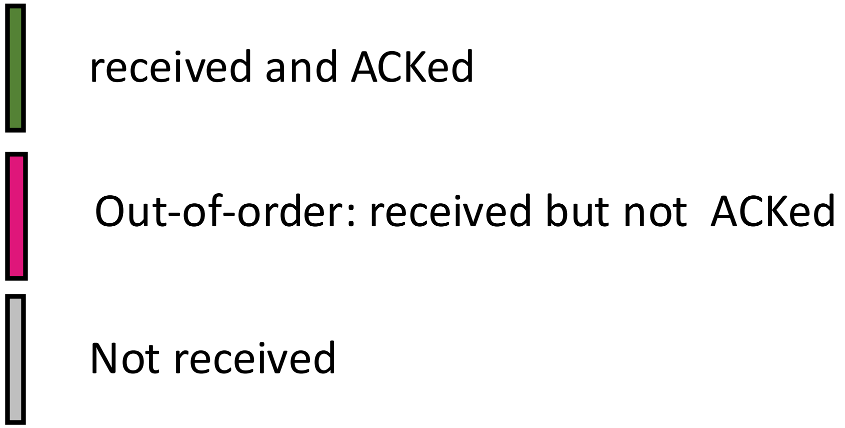
Go-Back-N: receiver

- ACK-only: always send ACK for correctly-received packet so far, with highest *in-order* seq #
  - may generate duplicate ACKs

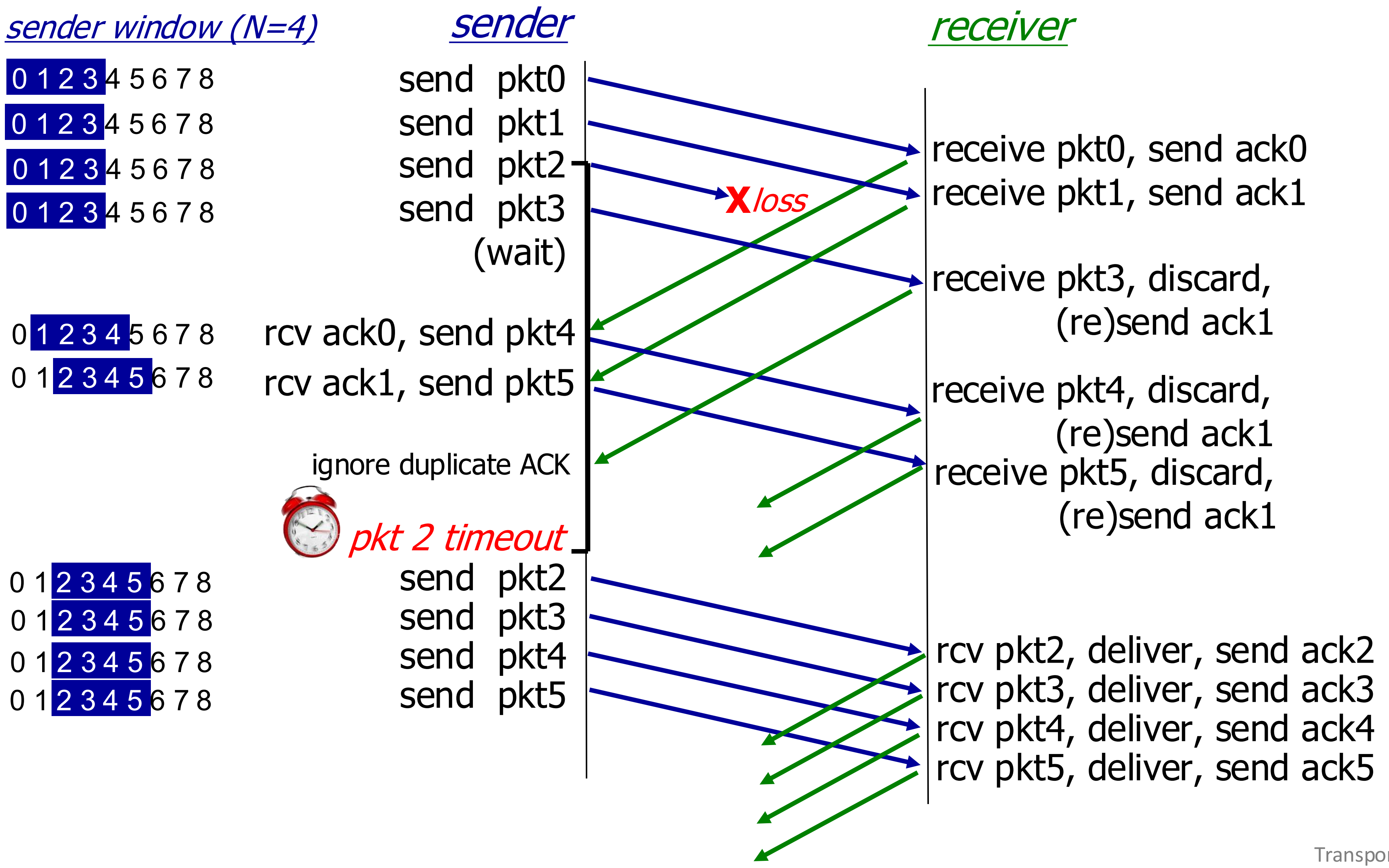
- on receipt of out-of-order packet:
  - can discard (don’t buffer) or buffer: an implementation decision
  - re-ACK pkt with highest in-order seq #

Receiver view of sequence number space:





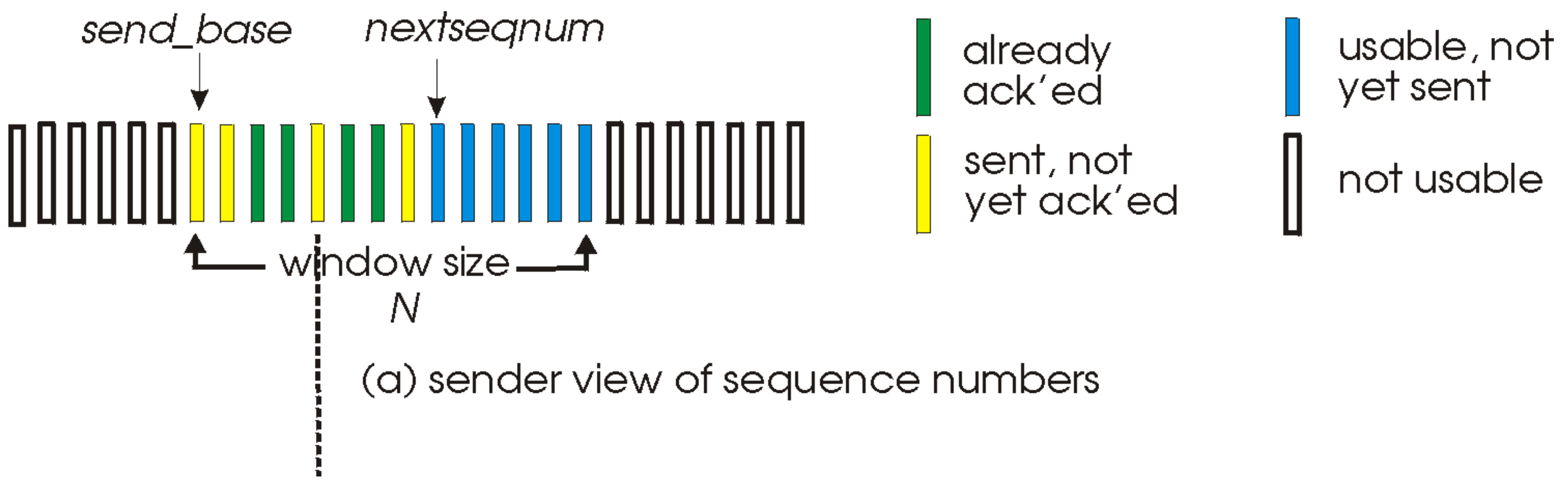
Go-Back-N in action



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



(a) sender view of sequence numbers

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

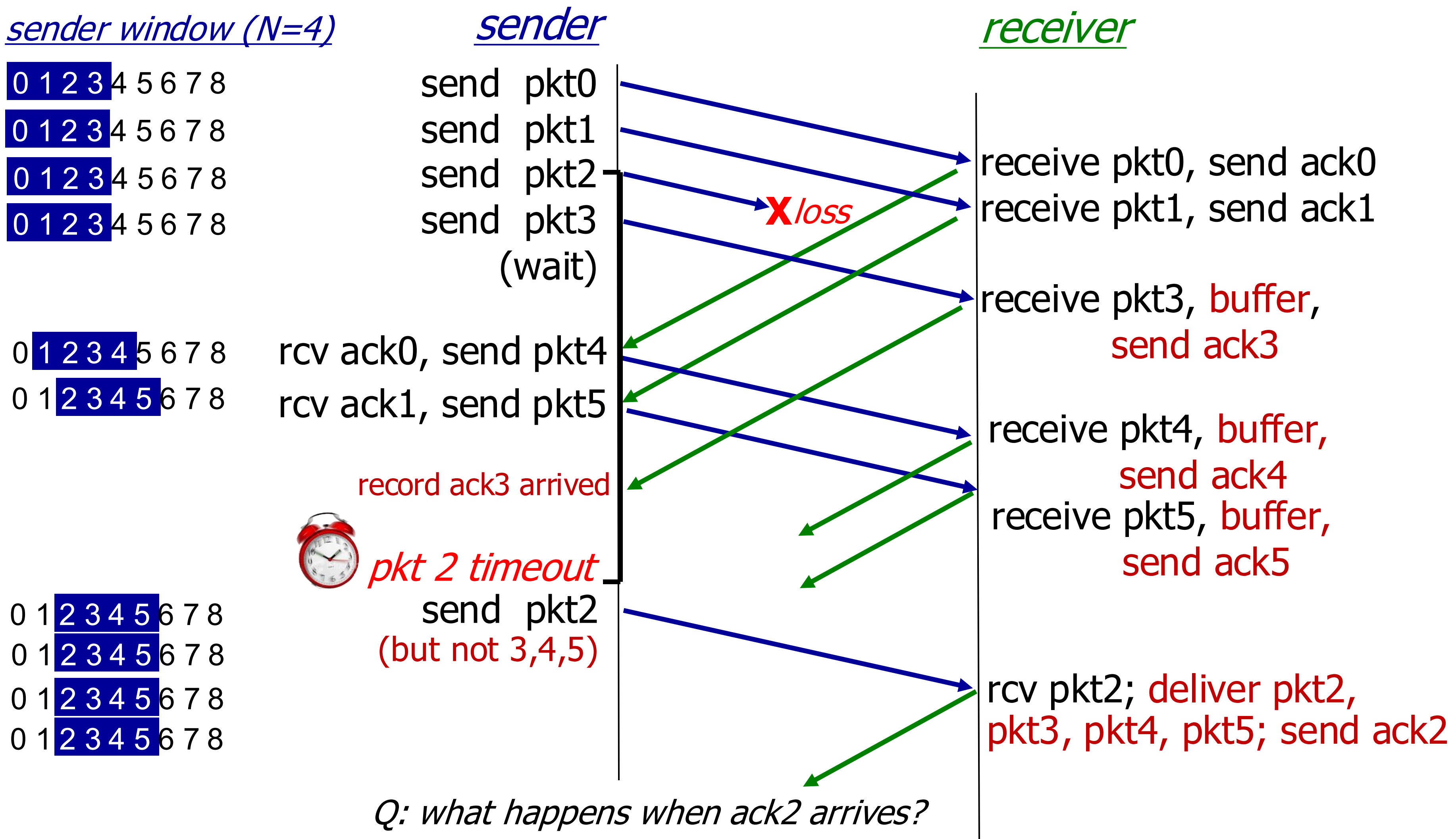
packet *n* in [rcvbase-N,rcvbase-1]

- ACK(*n*)

otherwise:

- ignore

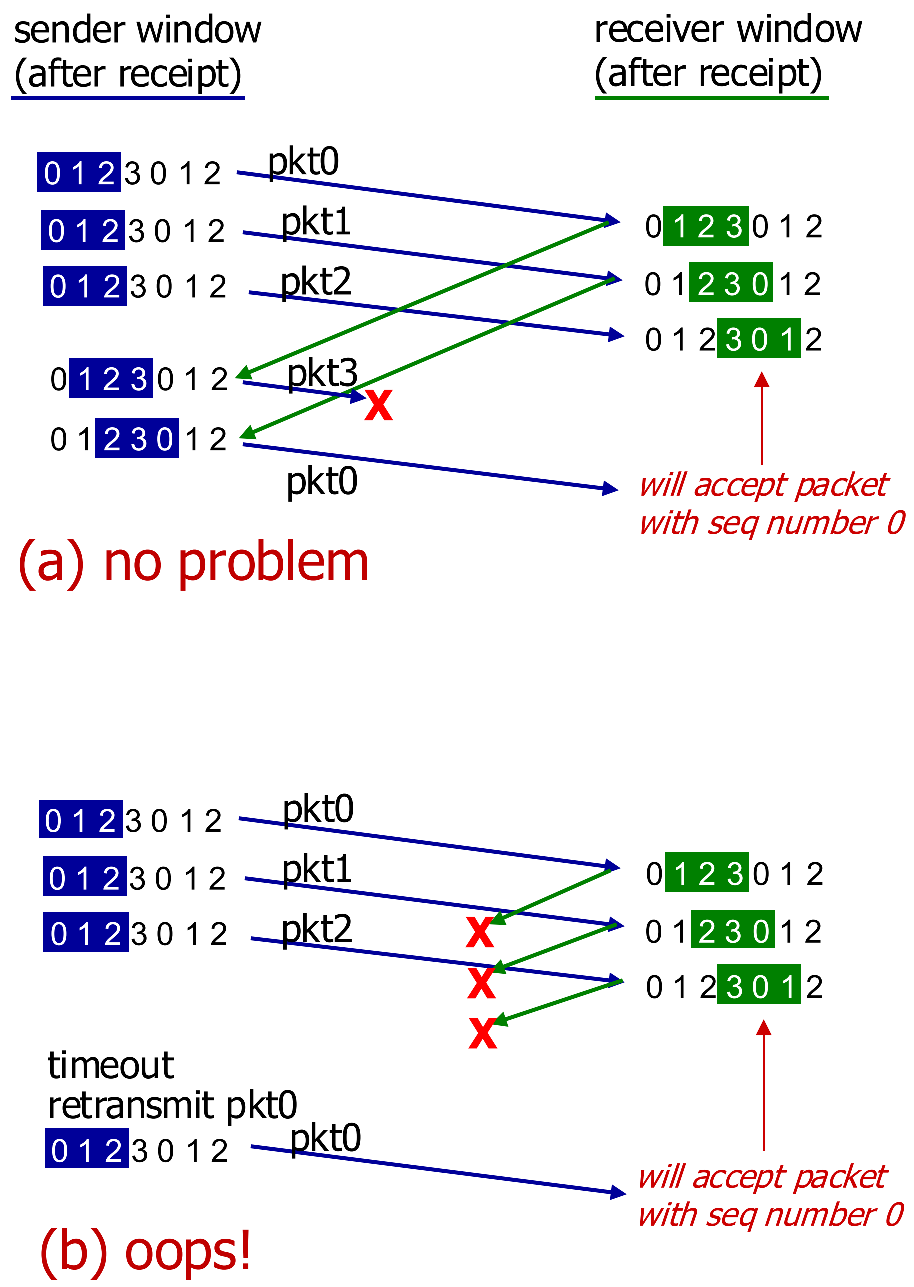
Selective Repeat in action



Selective repeat: a dilemma!

example:

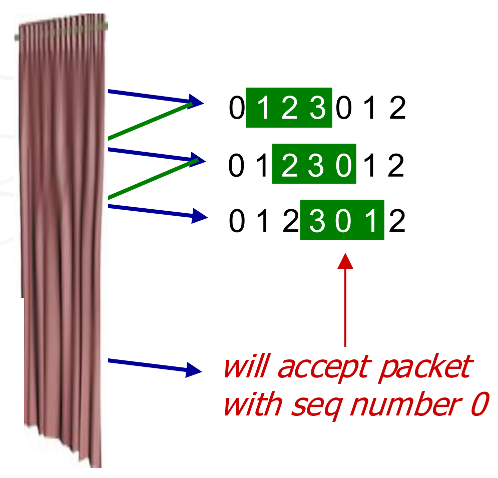
- seq #s: 0, 1, 2, 3 (base 4 counting)
- window size=3



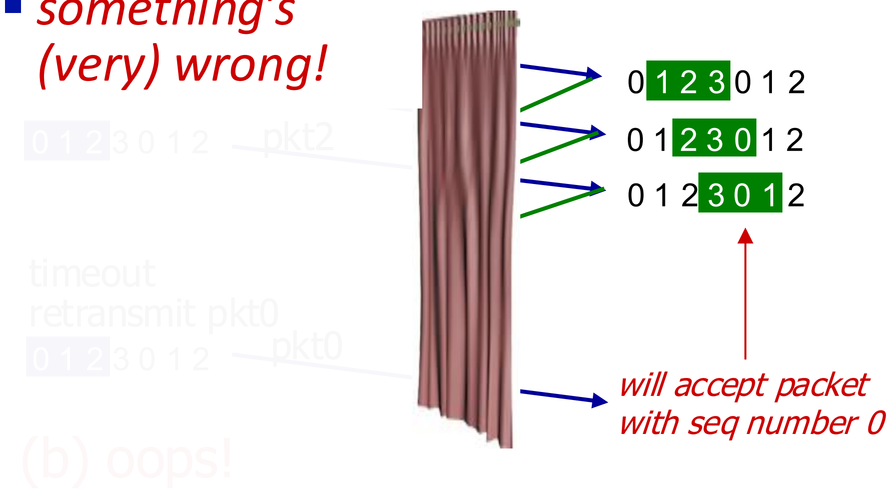
Selective repeat: a dilemma!

example:

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



- *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)?

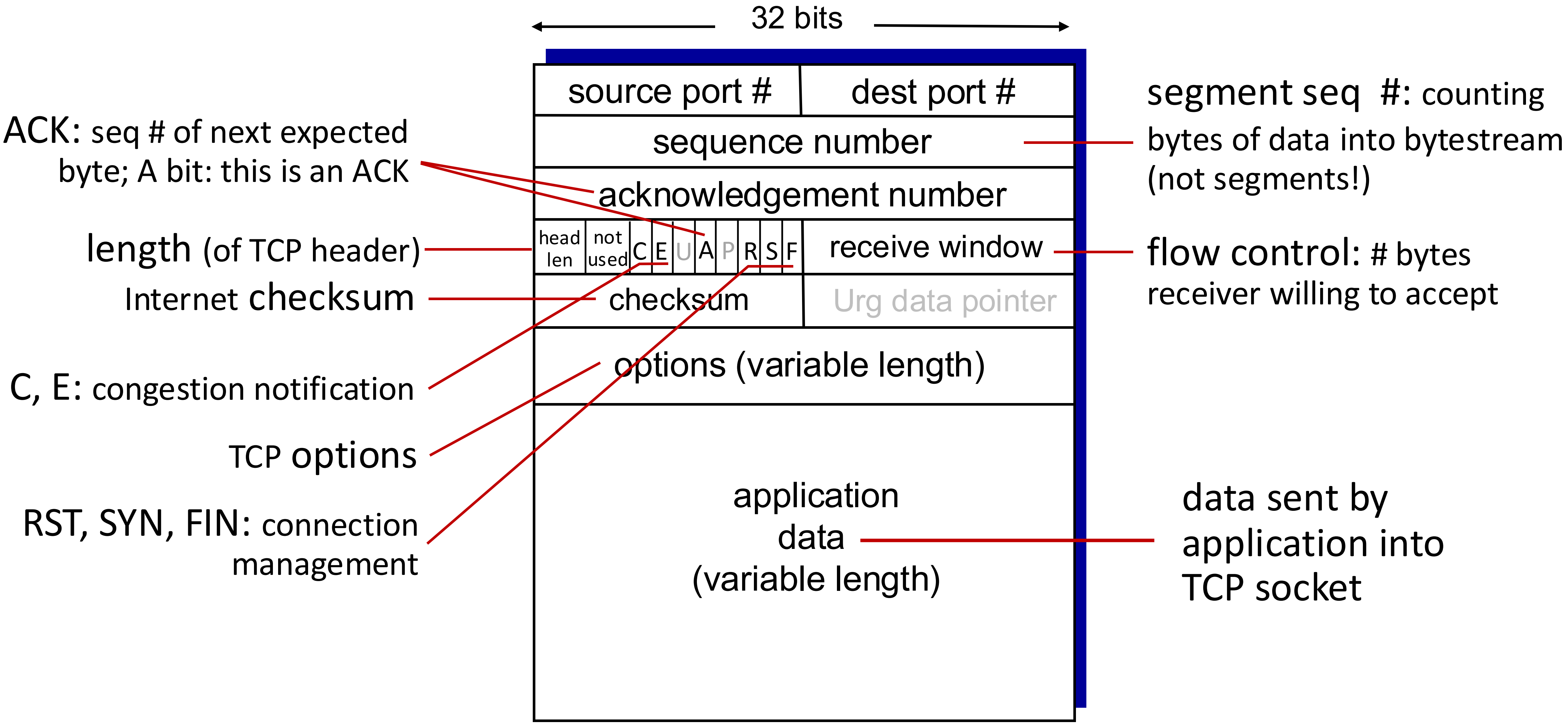
Chapter 3: roadmap

- 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:

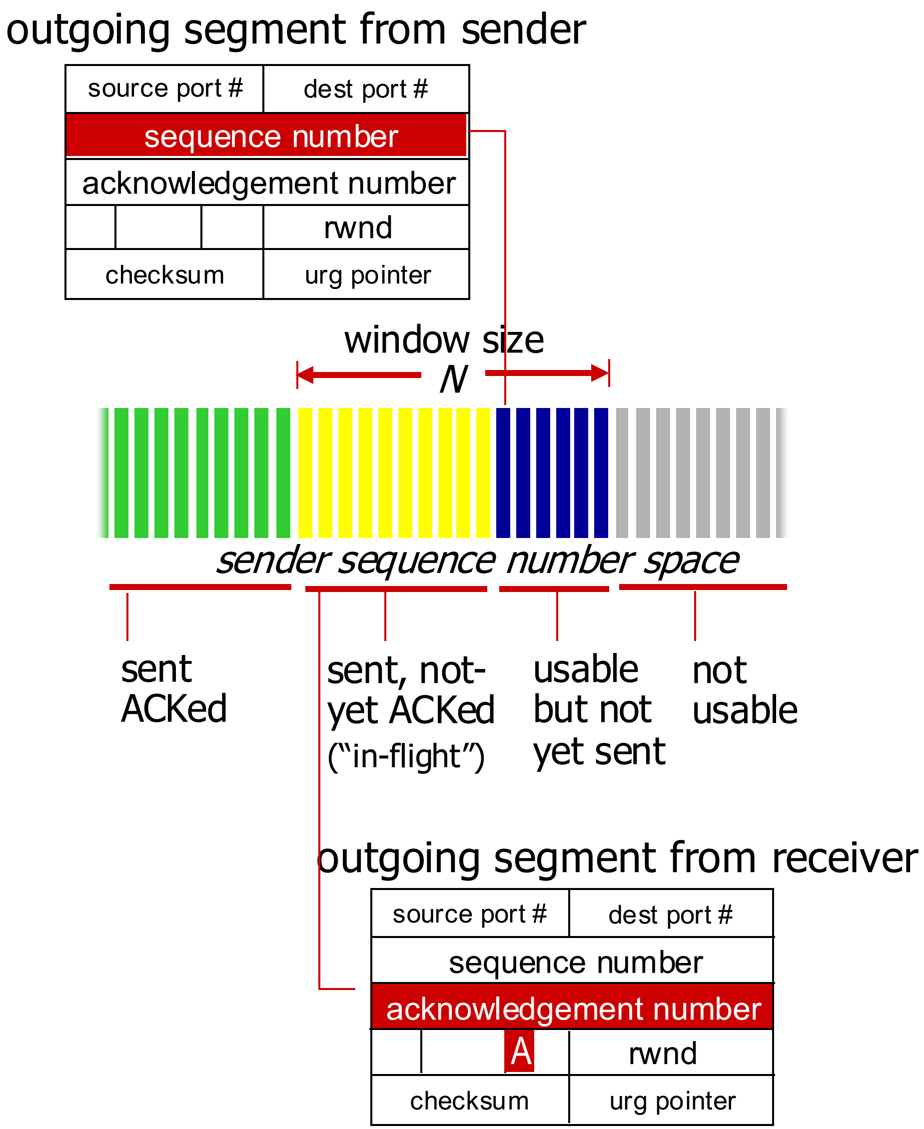
- byte stream "number" of first byte in segment's data

Acknowledgements:

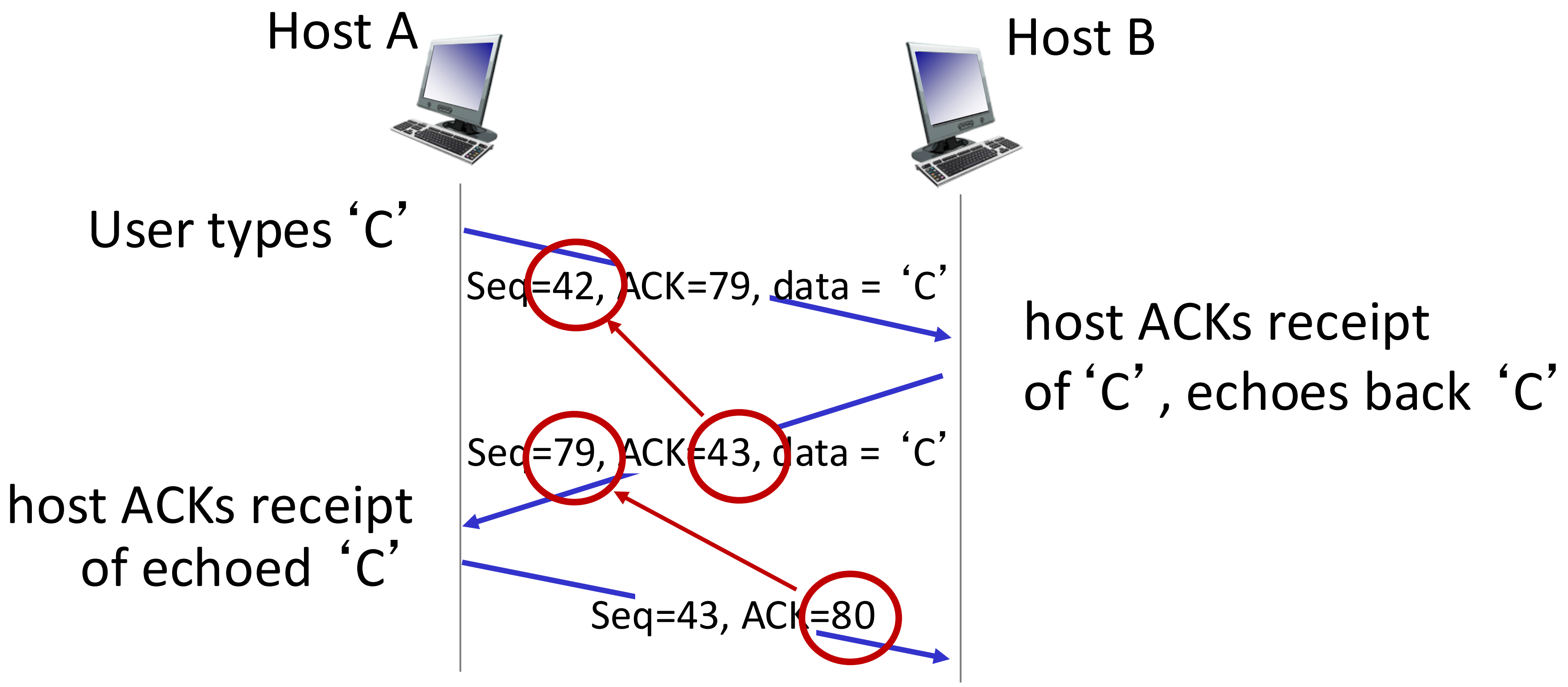
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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



TCP sequence numbers, ACKs



simple telnet scenario

TCP round trip time, timeout

Q: how to set TCP timeout value?

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

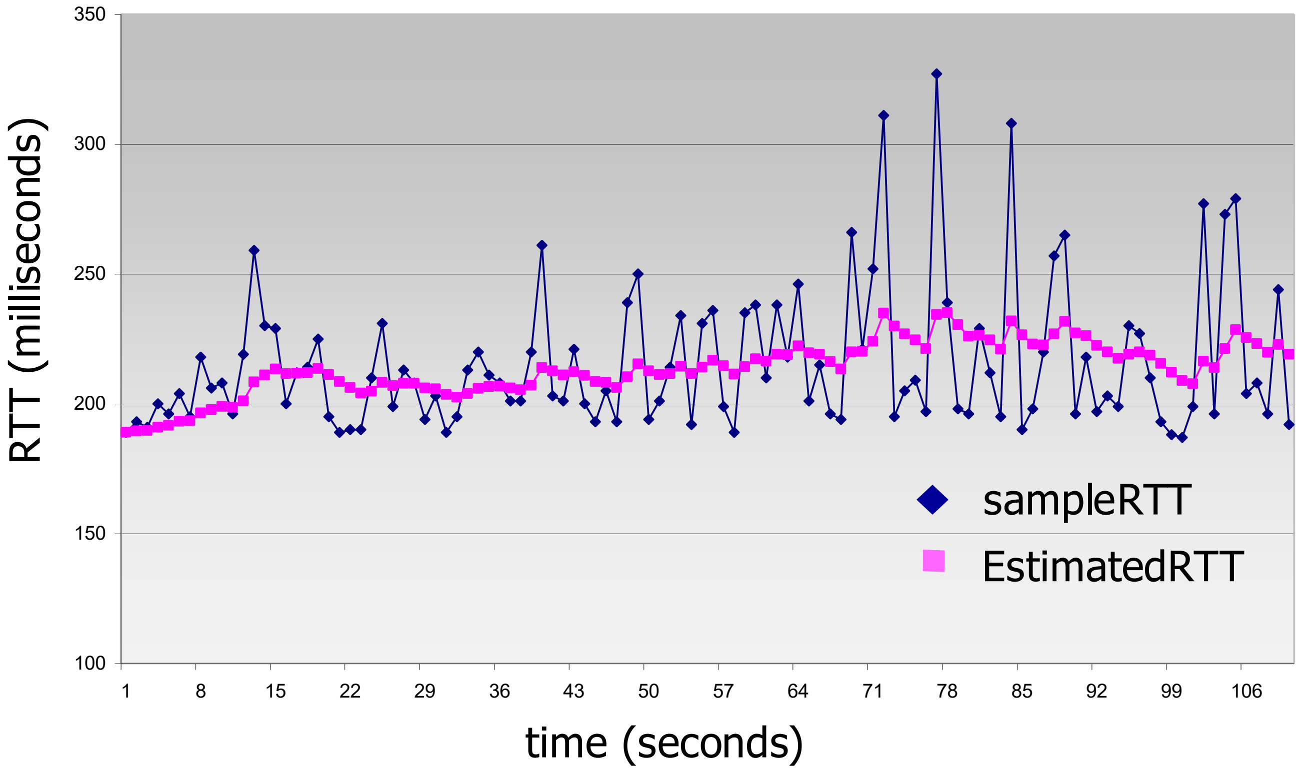
Q: how to estimate RTT?

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

TCP round trip time, timeout

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

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



TCP round trip time, timeout

- timeout interval: **`EstimatedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT:`** want a larger safety margin

```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

- **`DevRTT`**: EWMA of **`SampleRTT`** deviation from **`EstimatedRTT`**:

```
DevRTT = (1-β)*DevRTT + β*|SampleRTT-EstimatedRTT|
```

(typically, β = 0.25)

TCP Sender (simplified)

event: data received from application

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

event: timeout

- retransmit segment that caused timeout
- restart timer

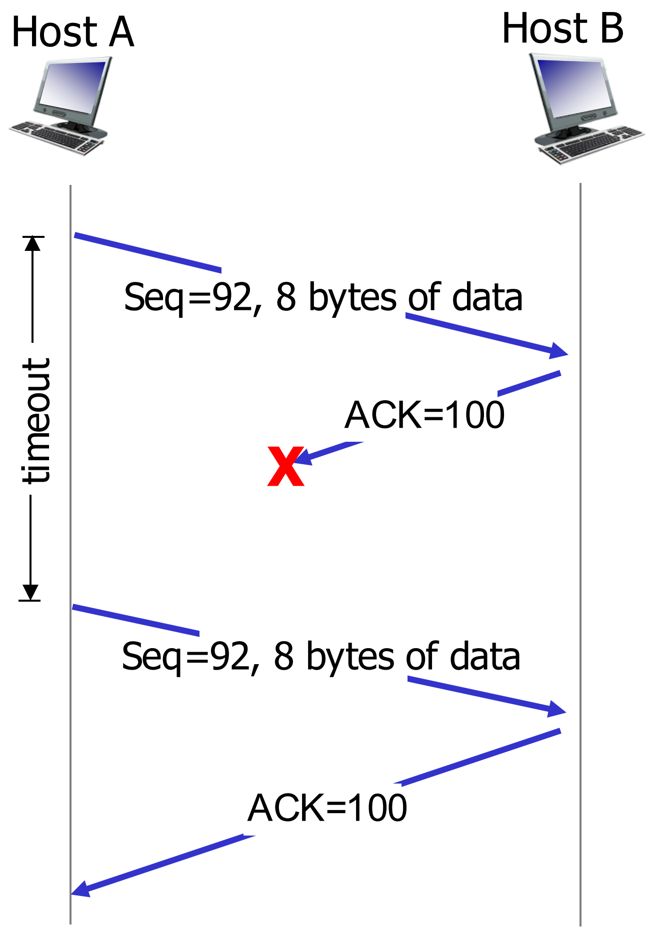
event: ACK received

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

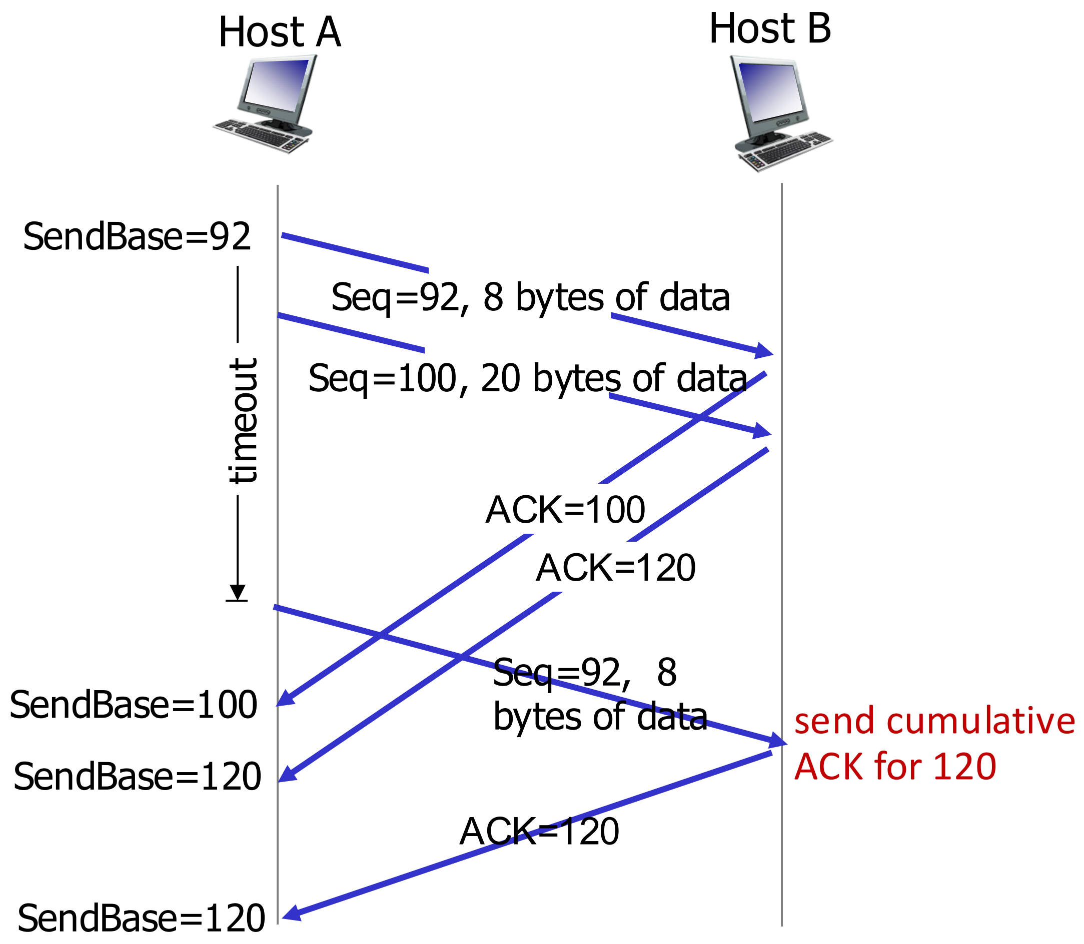
TCP Receiver: ACK generation [RFC 5681]

Event at receiver	*TCP receiver action*

TCP: retransmission scenarios

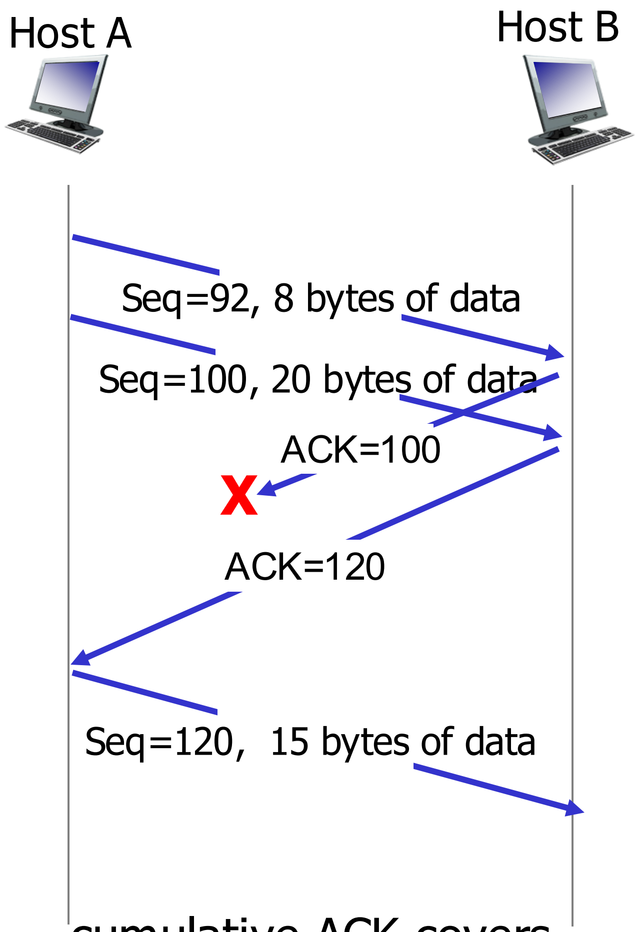


lost ACK scenario



premature timeout

TCP: retransmission scenarios



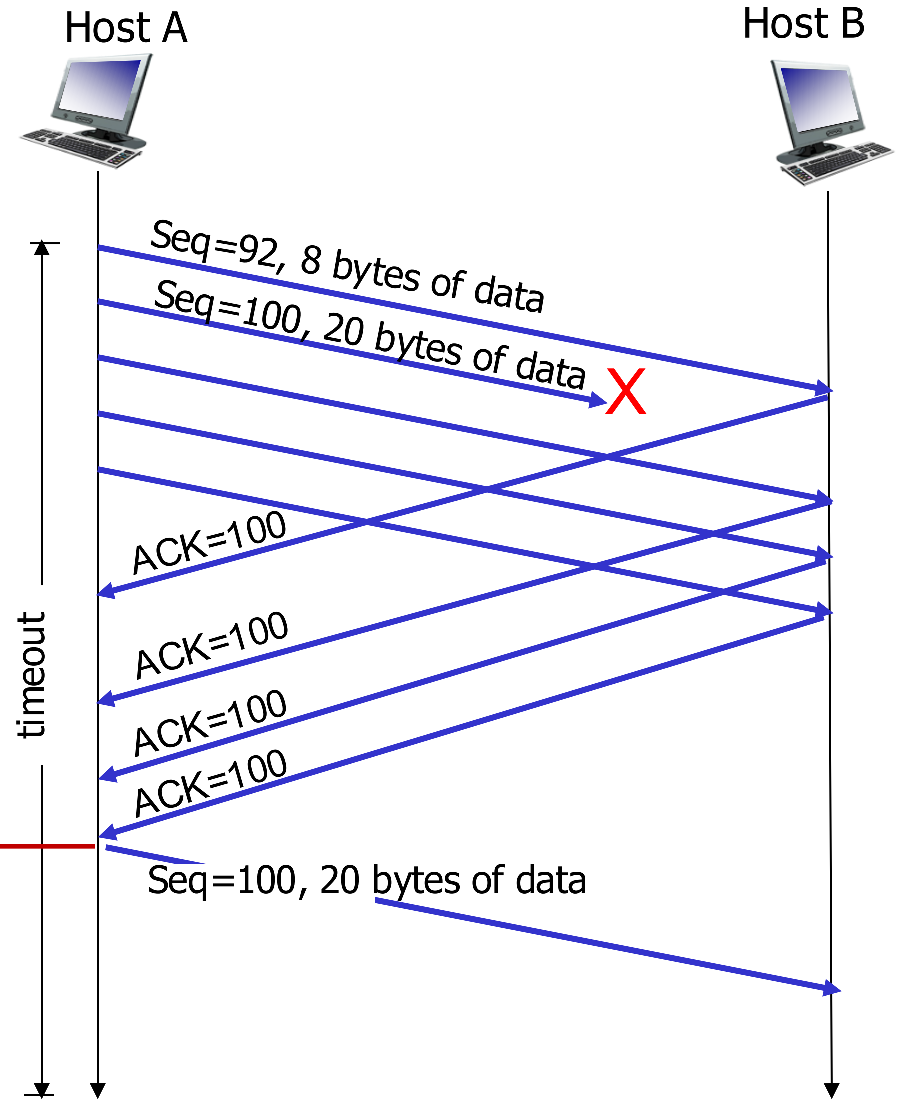
cumulative ACK covers
for earlier lost ACK

TCP fast retransmit

TCP fast retransmit

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

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



Receipt of three duplicate ACKs indicates 3 segments received after a missing segment – lost segment is likely. So retransmit!

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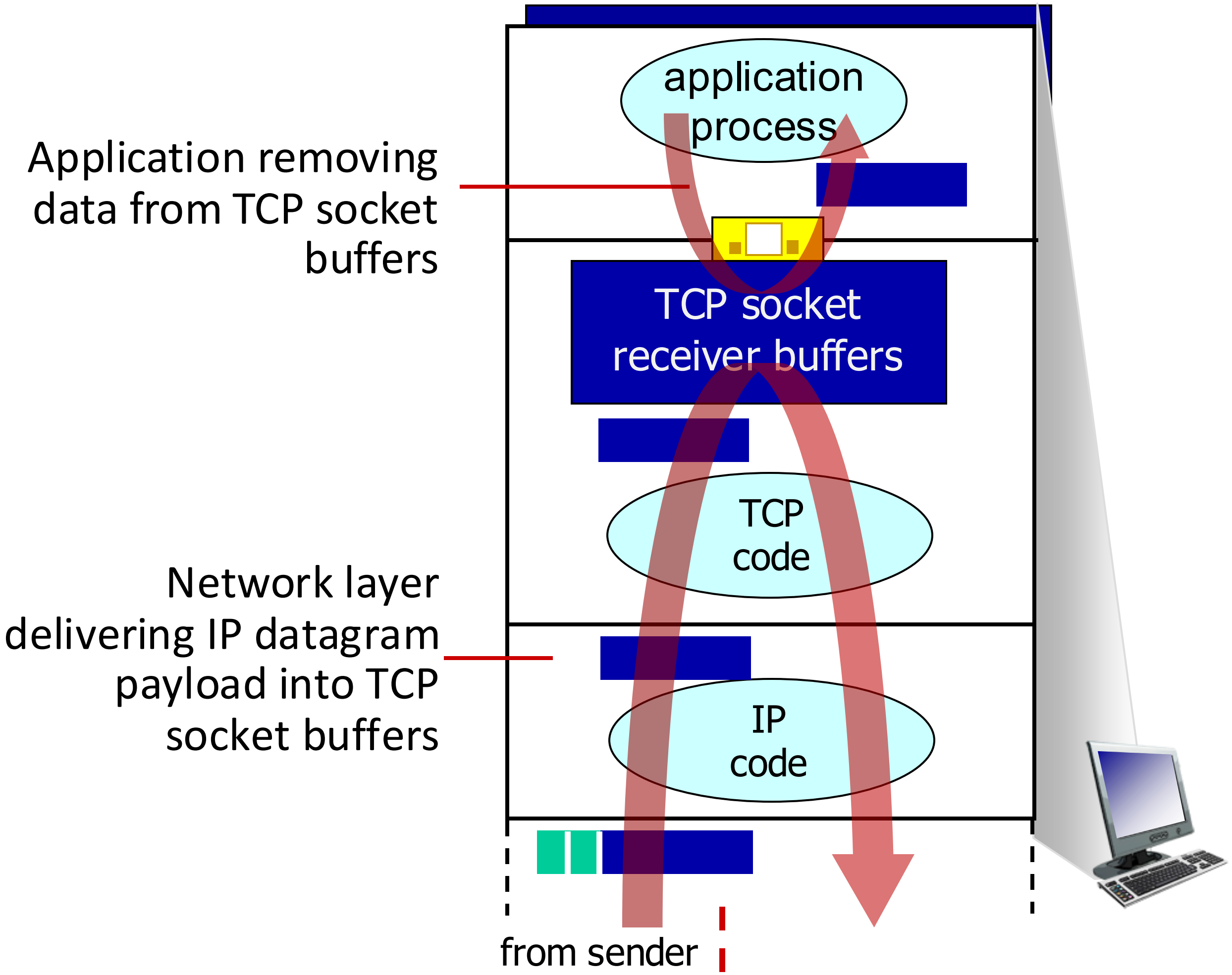
TCP flow control

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



receiver protocol stack

TCP flow control

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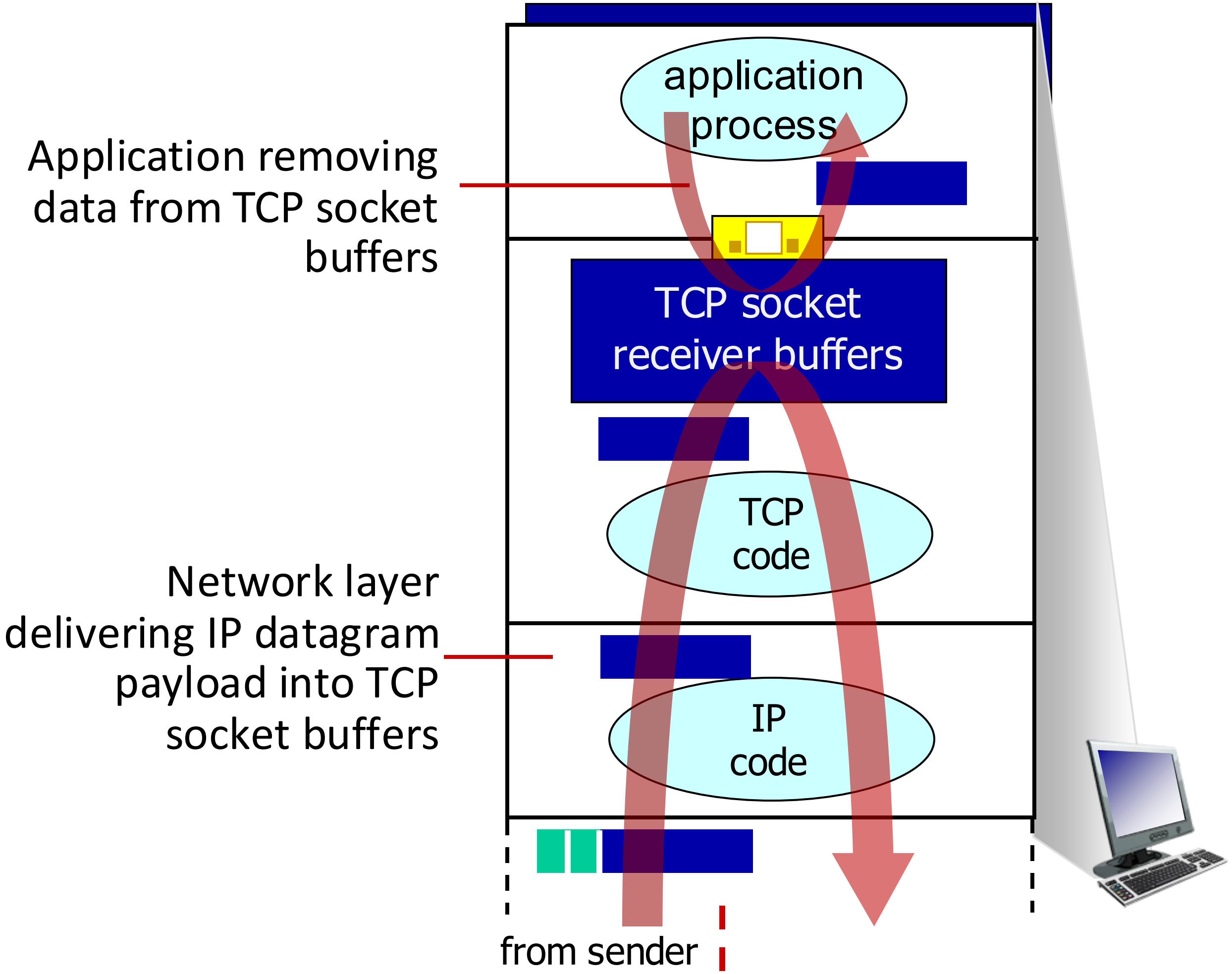

receiver protocol stack

TCP flow control

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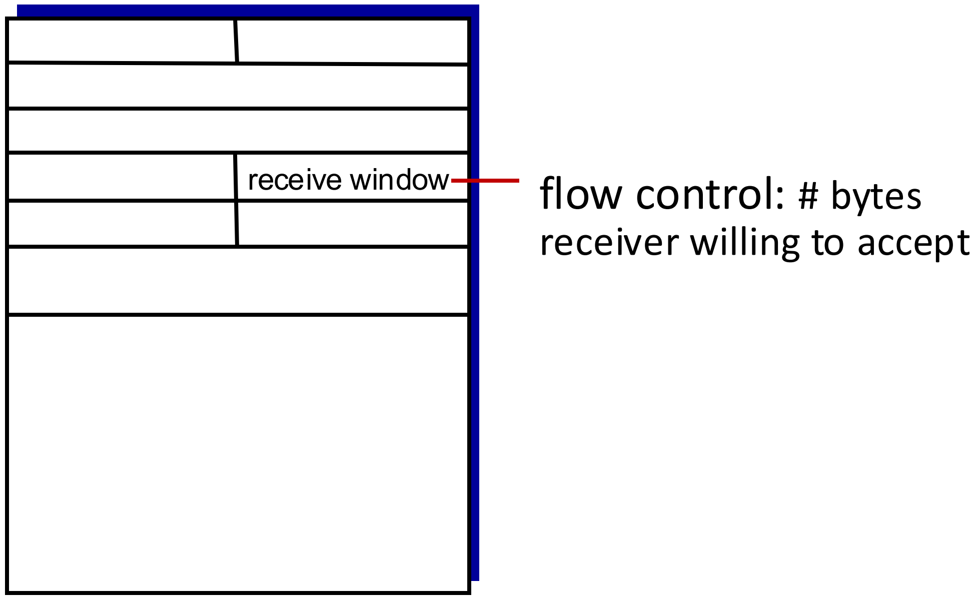
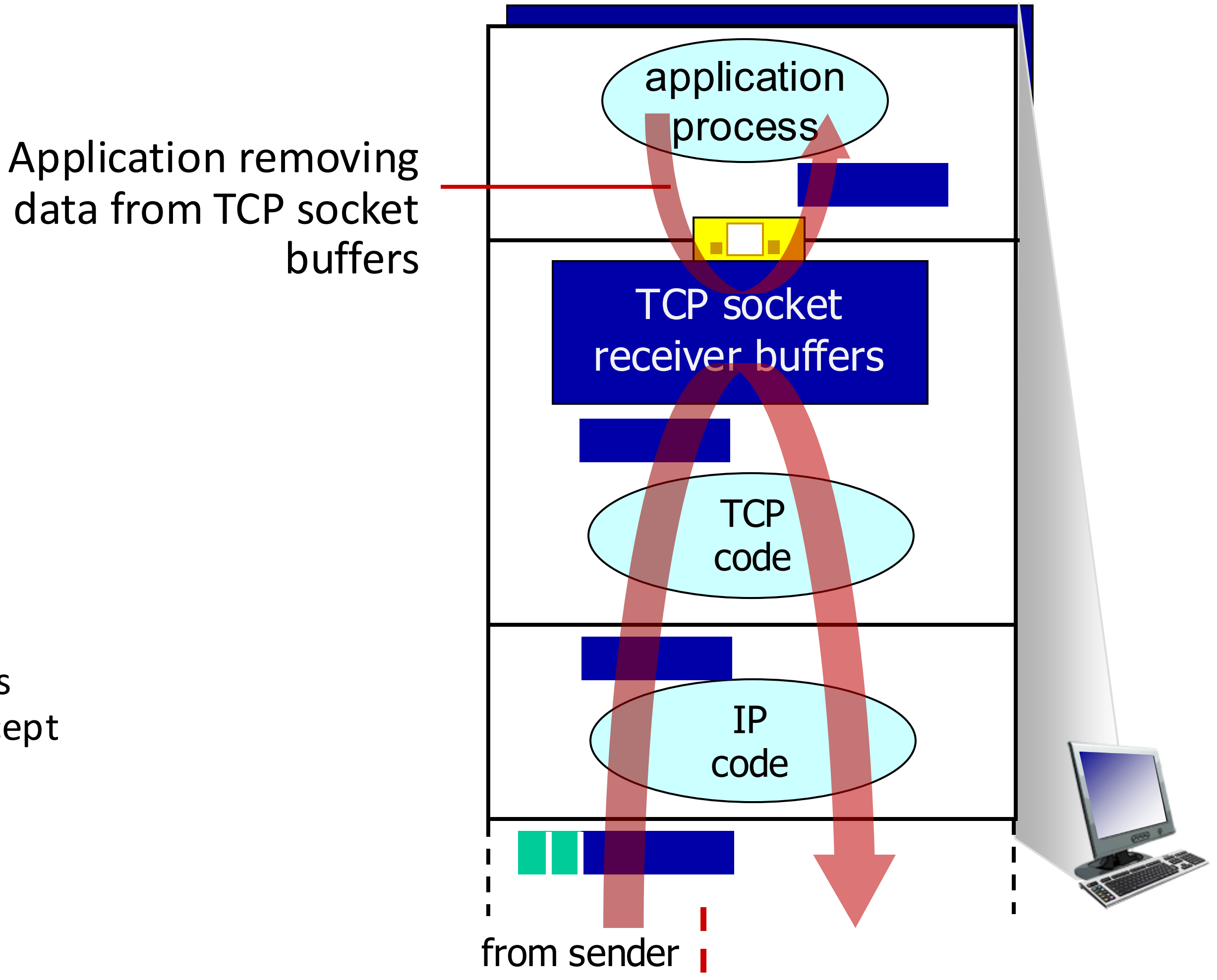




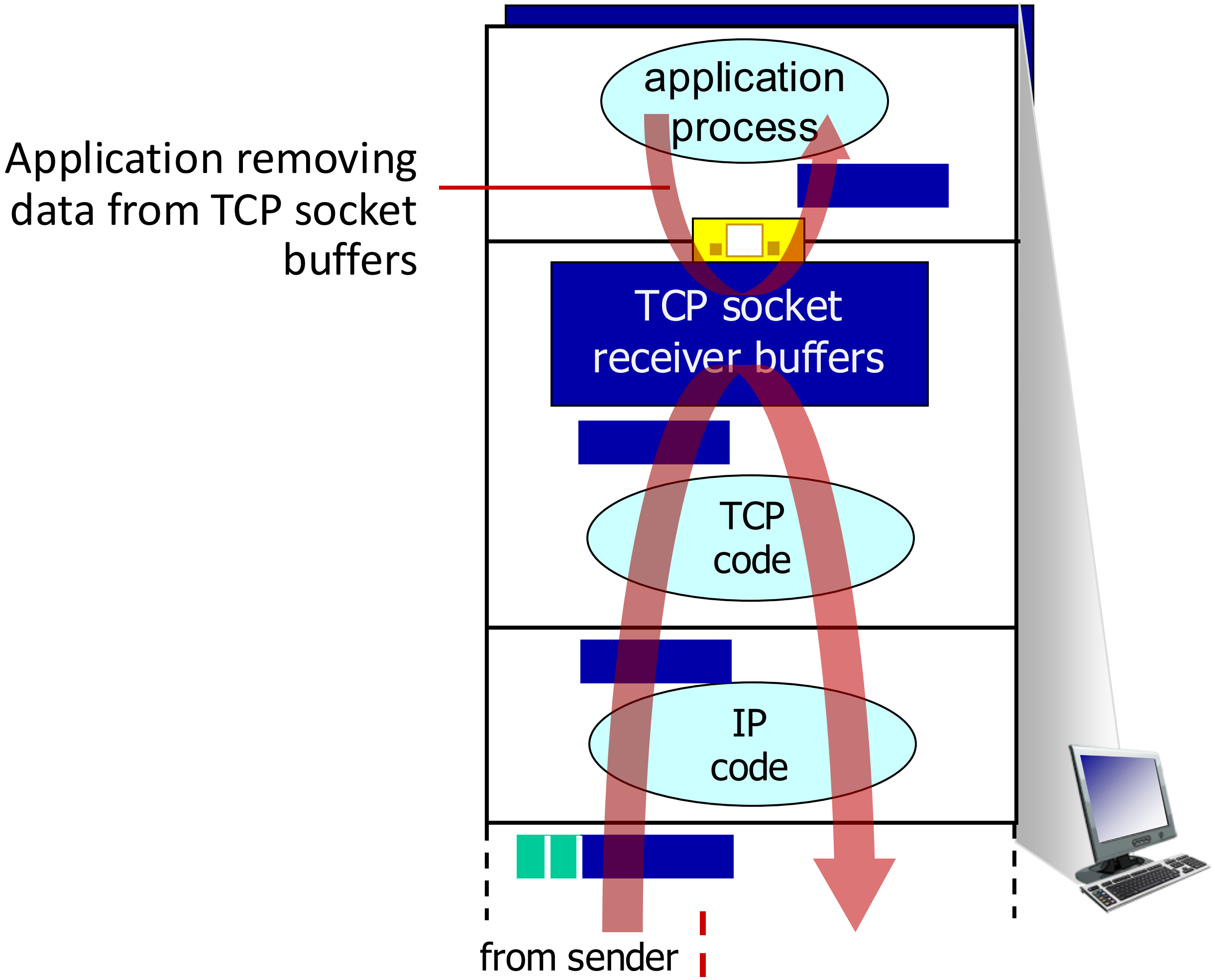
receiver protocol stack

TCP flow control

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

flow control

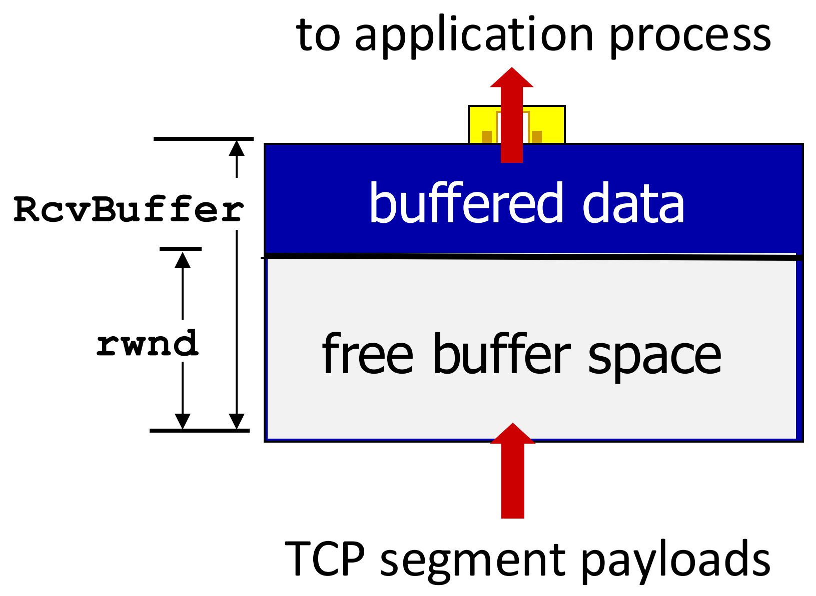
receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast



receiver protocol stack

TCP flow control

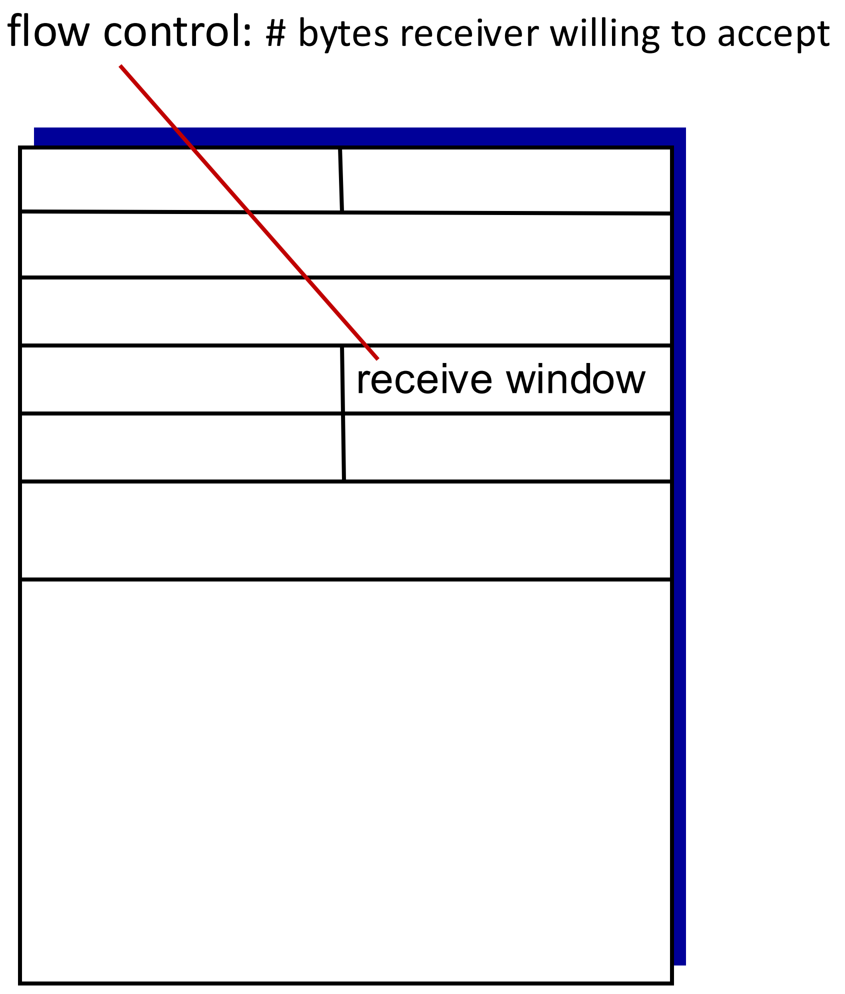
- TCP receiver "advertises" free buffer space in **rwnd** field in TCP header
  - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
  - many operating systems auto-adjust **RcvBuffer**
- sender limits amount of unACKed ("in-flight") data to received **rwnd**
- guarantees receive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

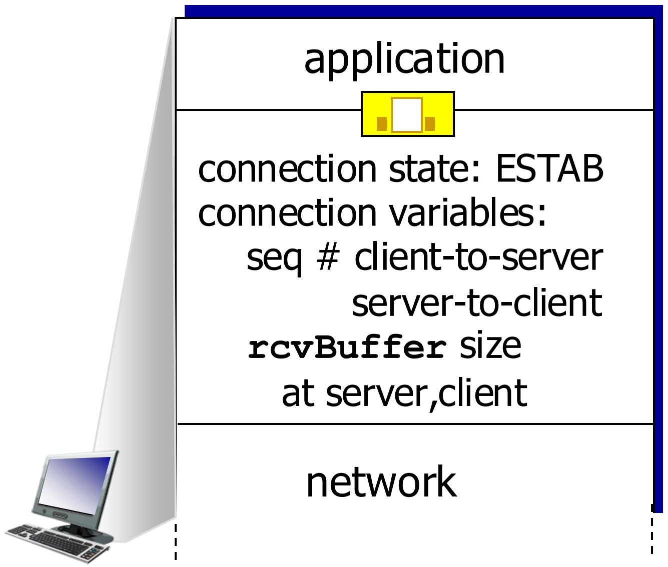


TCP segment format

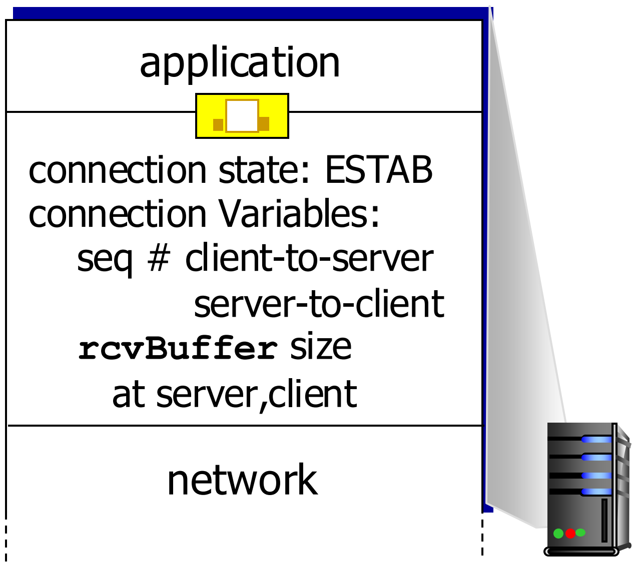
TCP connection management

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)



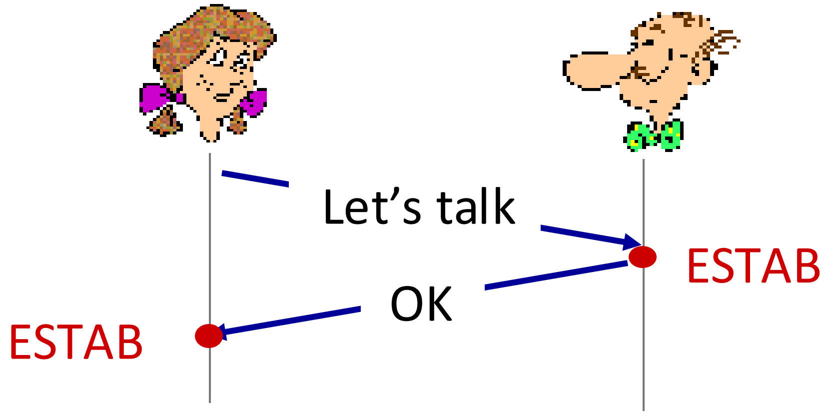

```
Socket clientSocket =
  newSocket("hostname","port number");
```

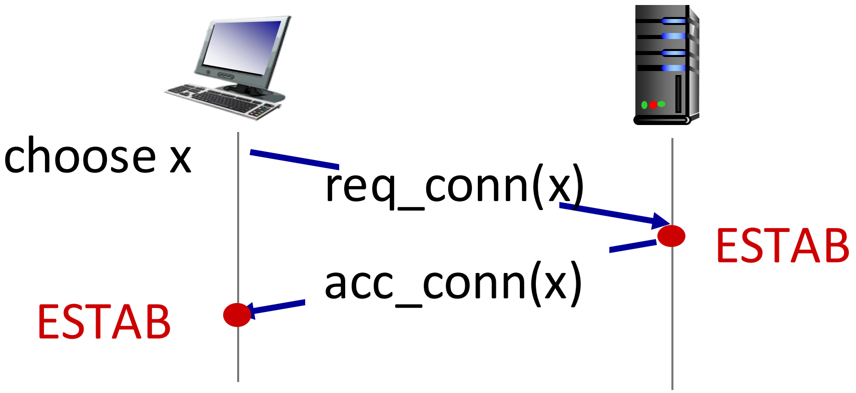



```
Socket connectionSocket =
  welcomeSocket.accept();
```

Agreeing to establish a connection

2-way handshake:

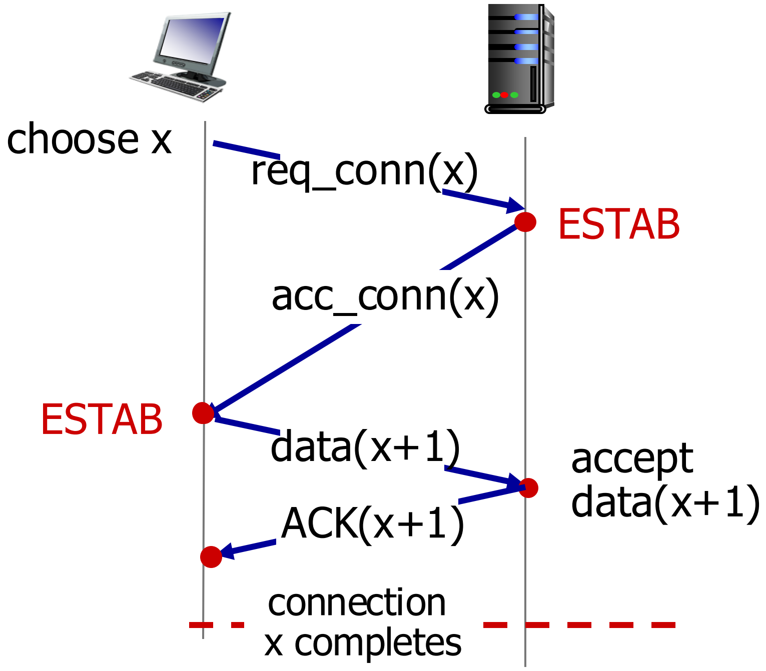




Q: will 2-way handshake always work in network?

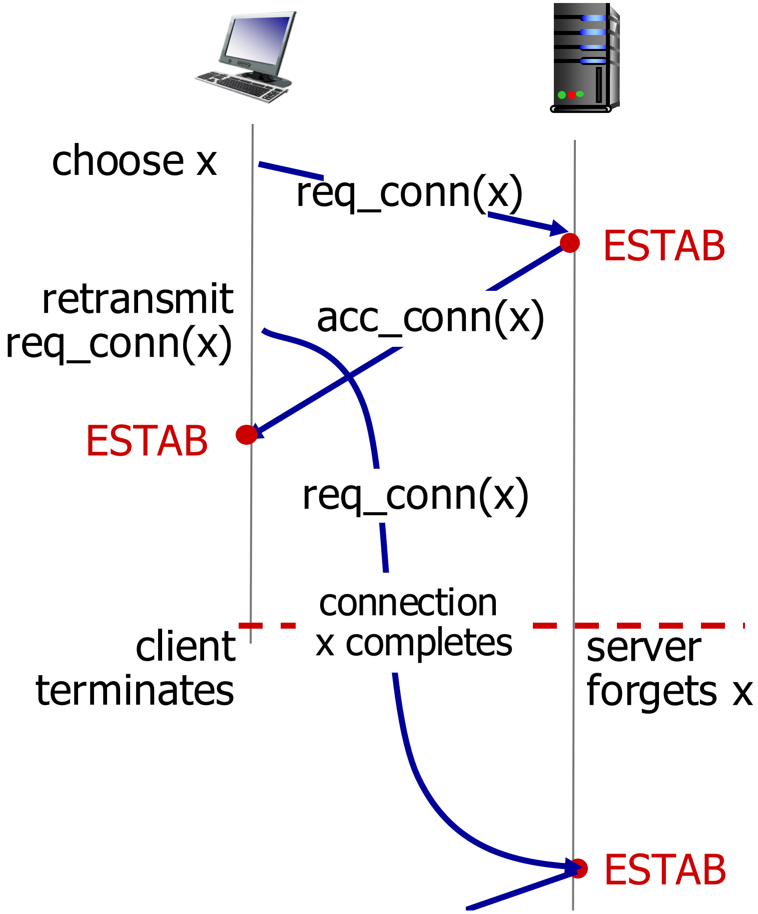
- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side

2-way handshake scenarios



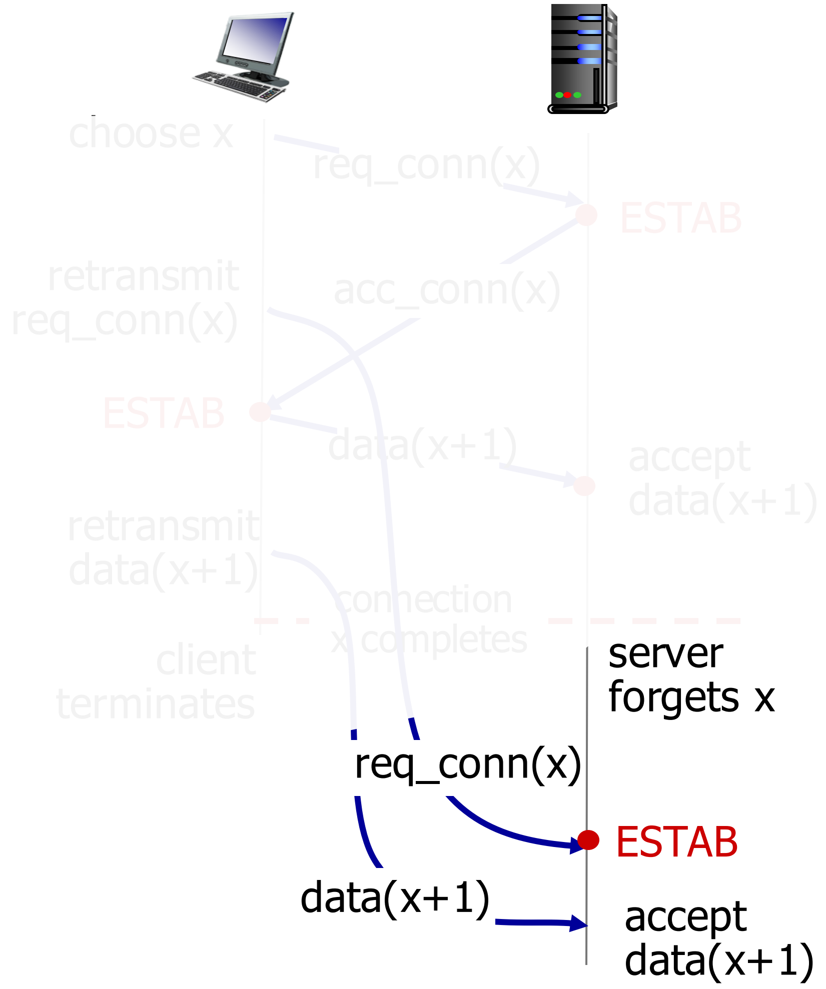
No problem!

2-way handshake scenarios



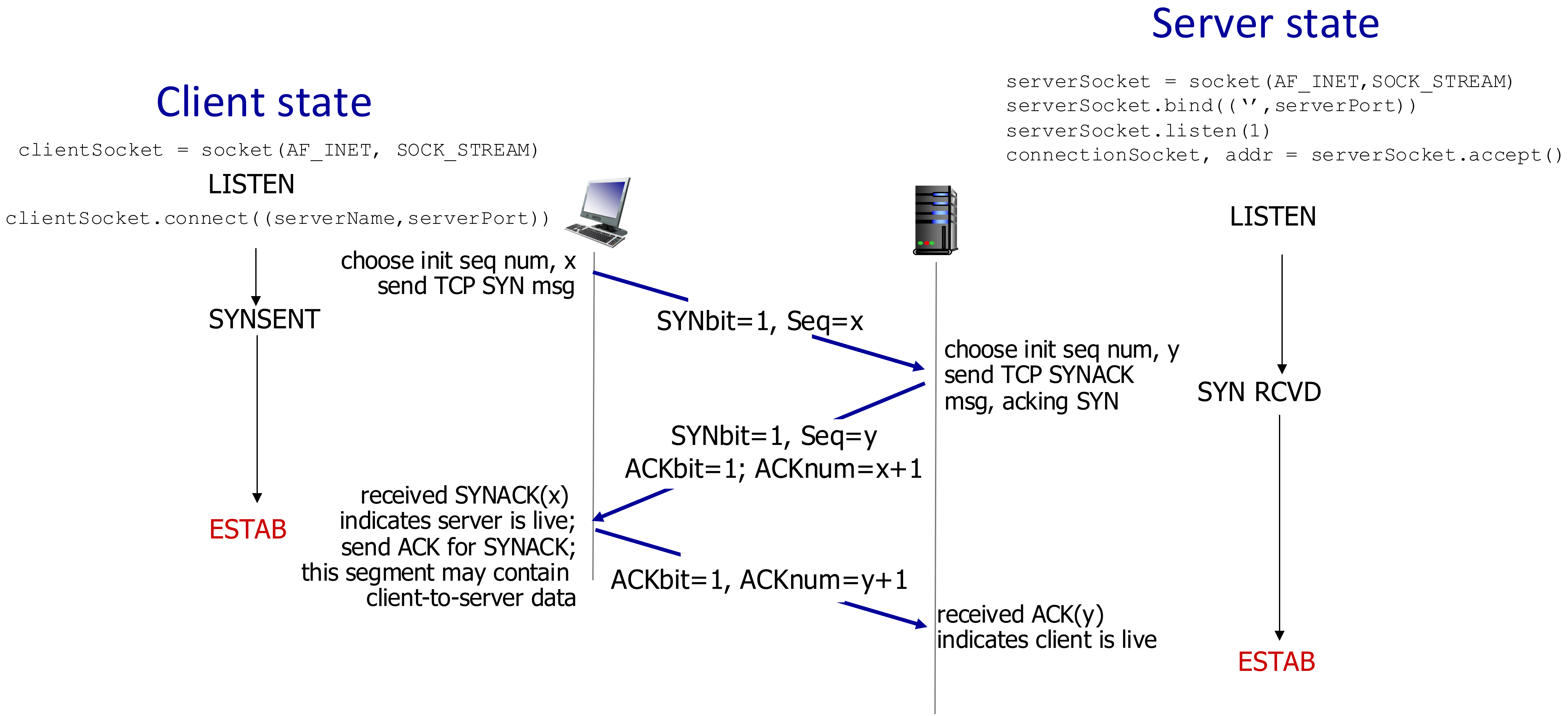
Problem: half open connection! (no client)

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 3: 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

TCP congestion control: AIMD

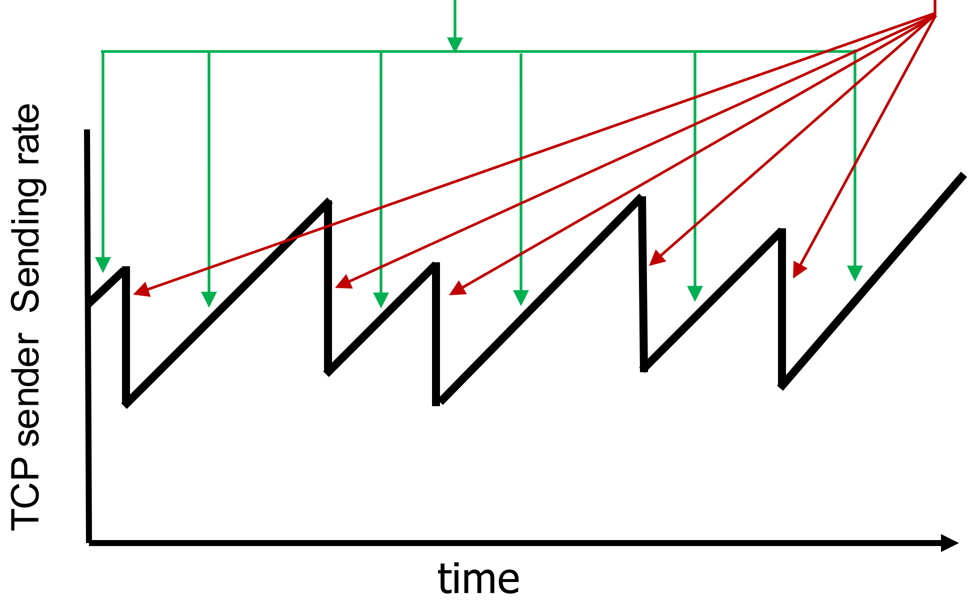
- *approach:* senders can increase sending rate until packet loss (congestion) occurs, then decrease sending rate on loss event

Additive Increase

increase sending rate by 1 maximum segment size every RTT until loss detected

Multiplicative Decrease

cut sending rate in half at each loss event



AIMD sawtooth behavior: *probing* for bandwidth

TCP AIMD: more

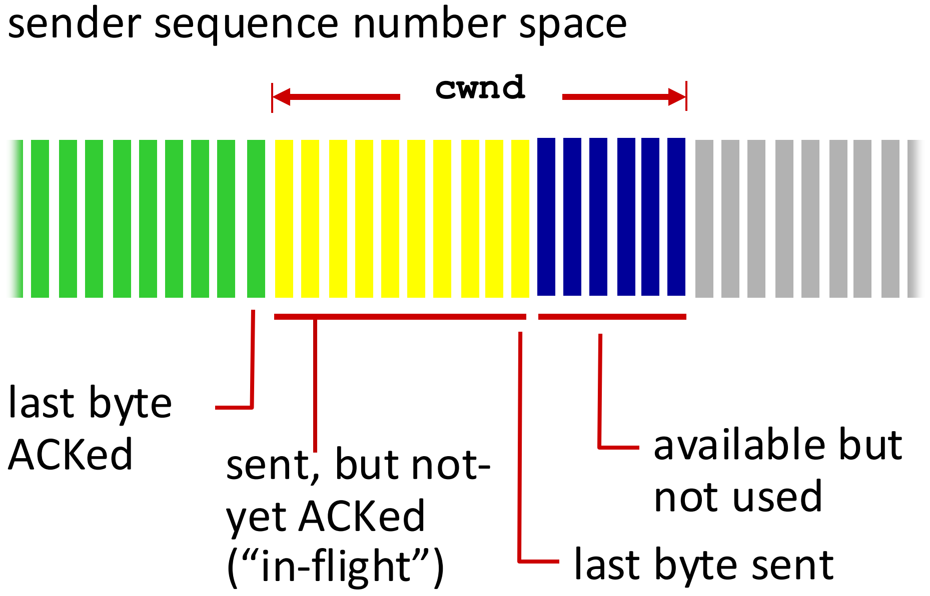
Multiplicative decrease detail: sending rate is

- Cut in half on loss detected by triple duplicate ACK (TCP Reno)
- Cut to 1 MSS (maximum segment size) when loss detected by timeout (TCP Tahoe)

Why AIMD?

- AIMD – a distributed, asynchronous algorithm – has been shown to:
  - optimize congested flow rates network wide!
  - have desirable stability properties

TCP congestion control: details



TCP sending behavior:

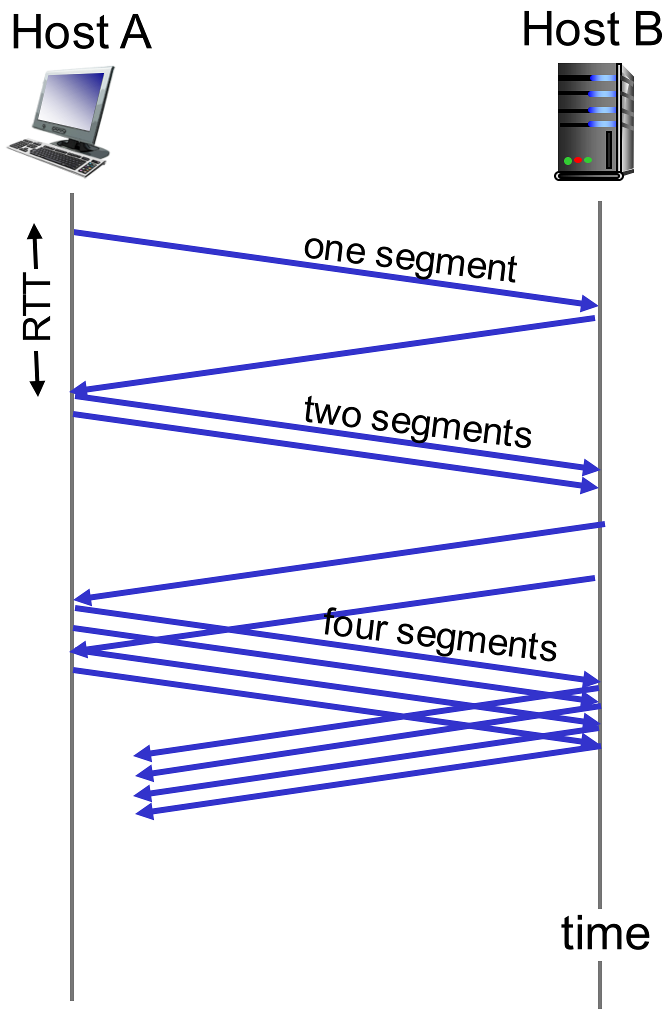
- *roughly:* send `cwnd` bytes, wait RTT for ACKS, then send more bytes

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

- TCP sender limits transmission: **`LastByteSent- LastByteAcked` ≤ `cwnd`**
- `cwnd` is dynamically adjusted in response to observed network congestion (implementing TCP congestion control)

TCP slow start

- when connection begins, increase rate exponentially until first loss event:
  - initially **cwnd** = 1 MSS
  - double **cwnd** every RTT
  - done by incrementing **cwnd** for every ACK received
- *summary:* initial rate is slow, but ramps up exponentially fast



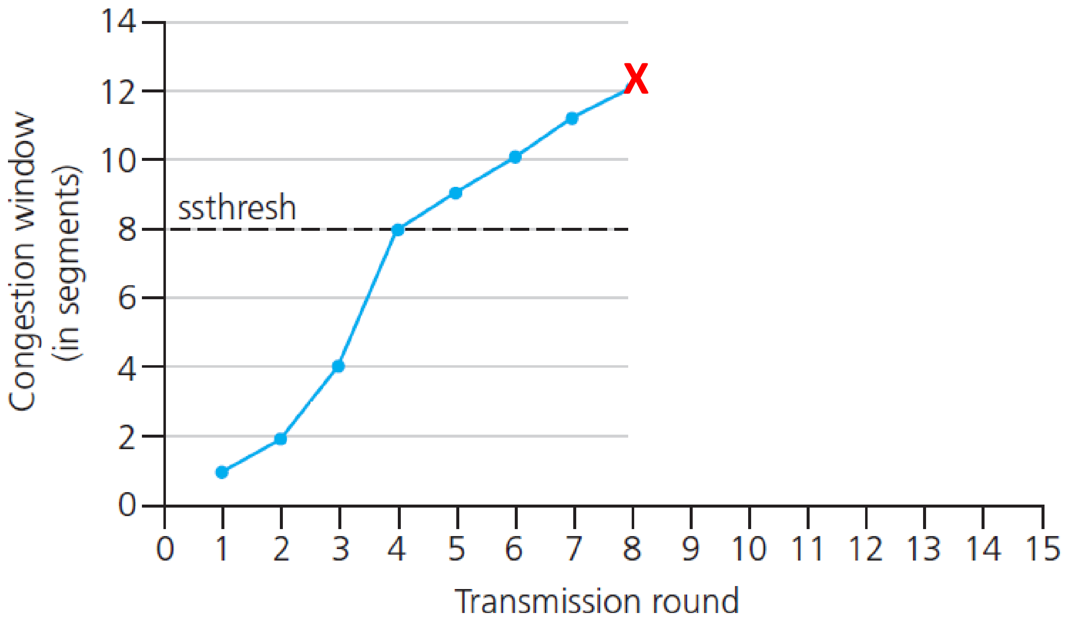
TCP: from slow start to congestion avoidance

Q: when should the exponential increase switch to linear?

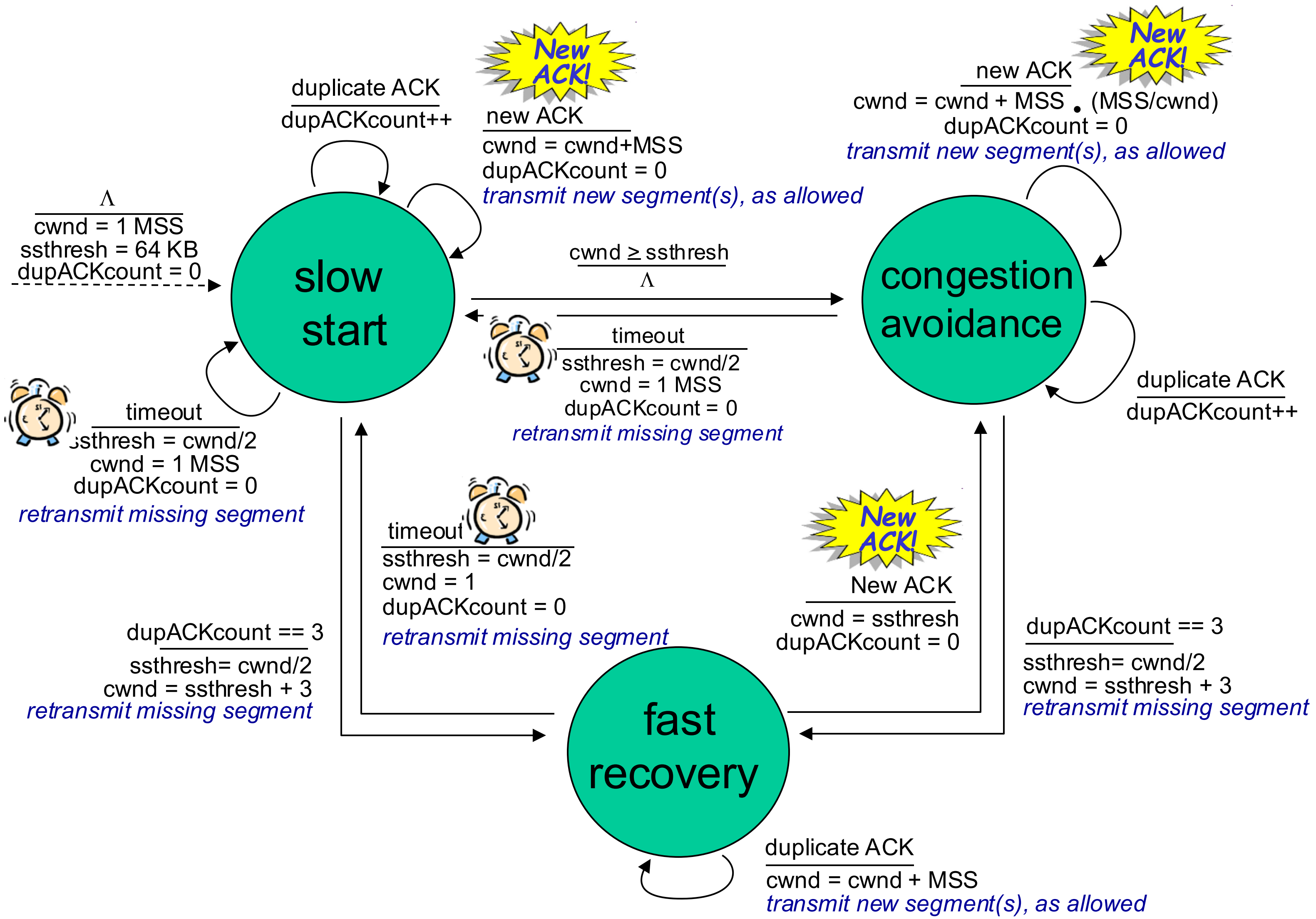
A: when **`cwnd`** gets to 1/2 of its value before timeout.

Implementation:

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



Summary: TCP congestion control

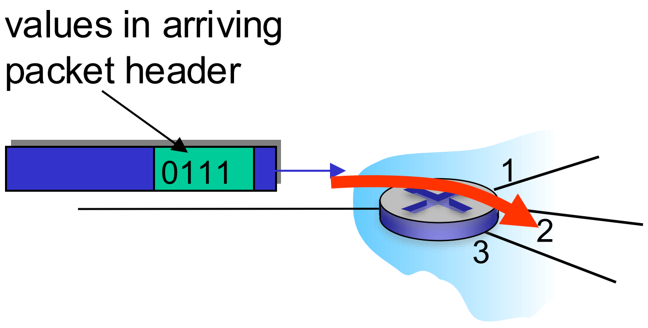


Chapter 4
Network Layer:
Data Plane

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

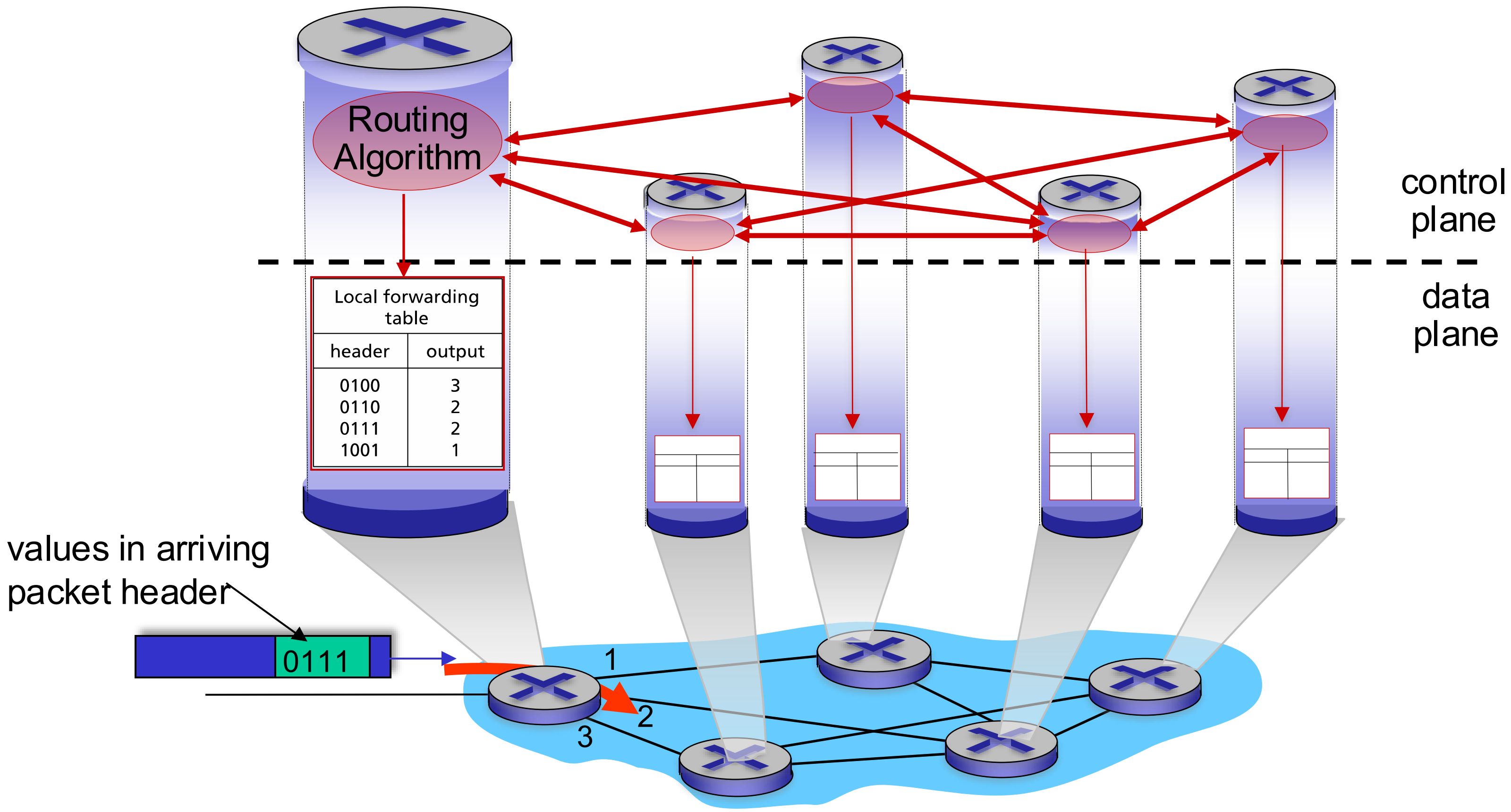


Control plane

- *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

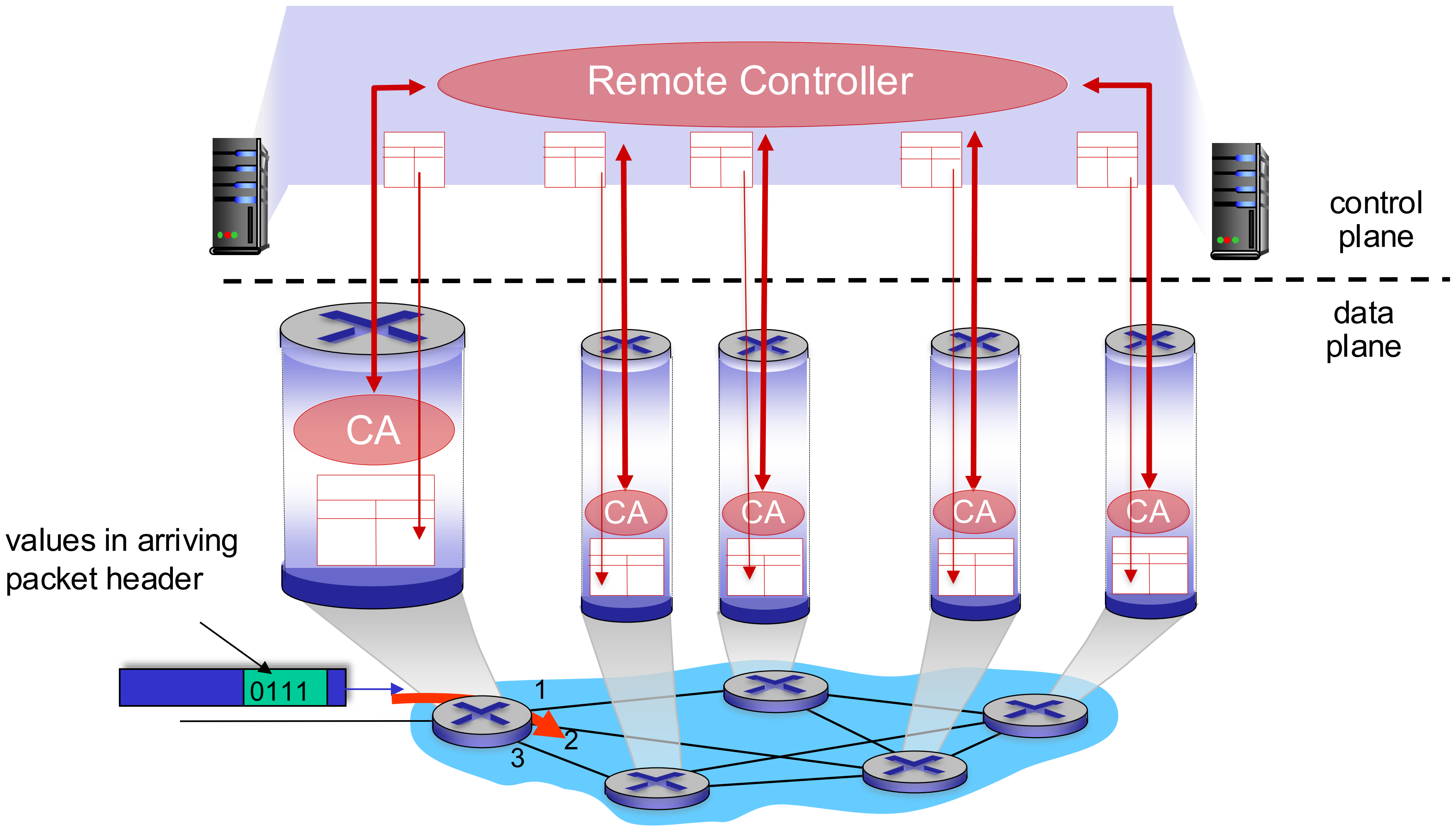
Per-router control plane

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



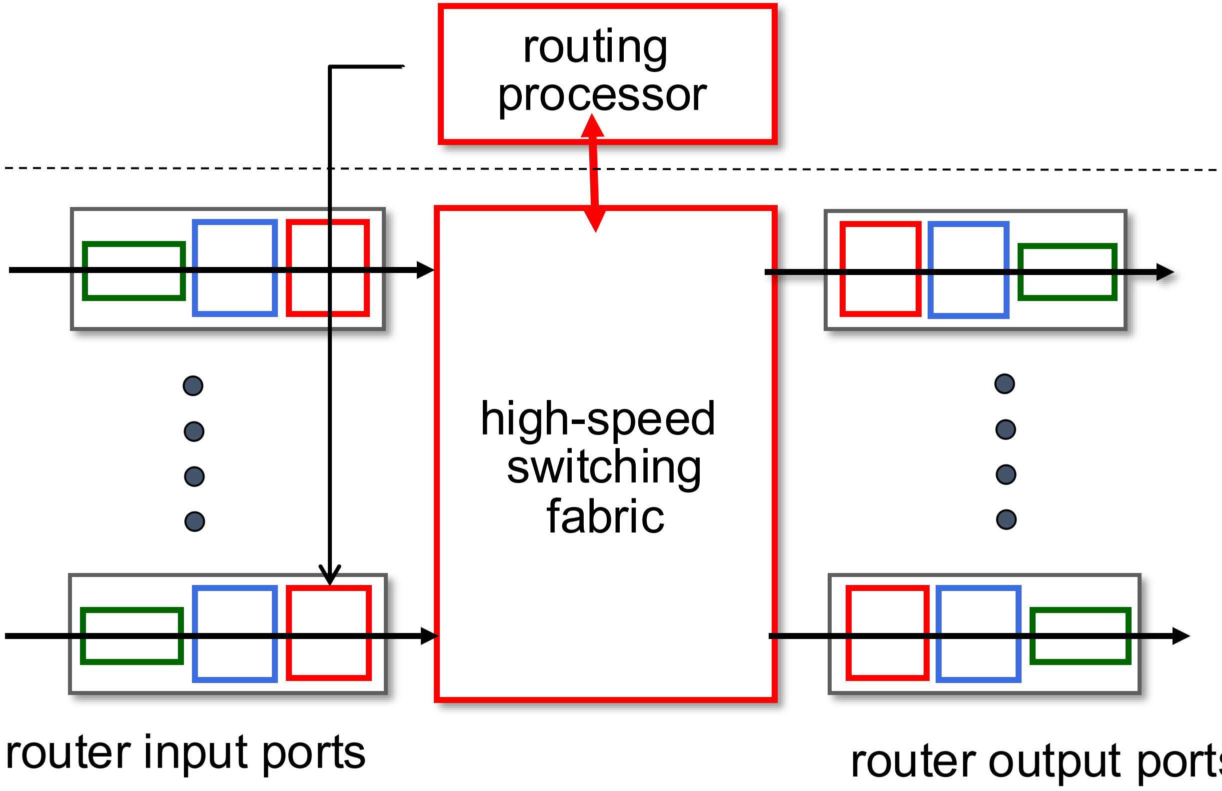
Software-Defined Networking (SDN) control plane

Remote controller computes, installs forwarding tables in routers



Router architecture overview

high-level view of generic router architecture:

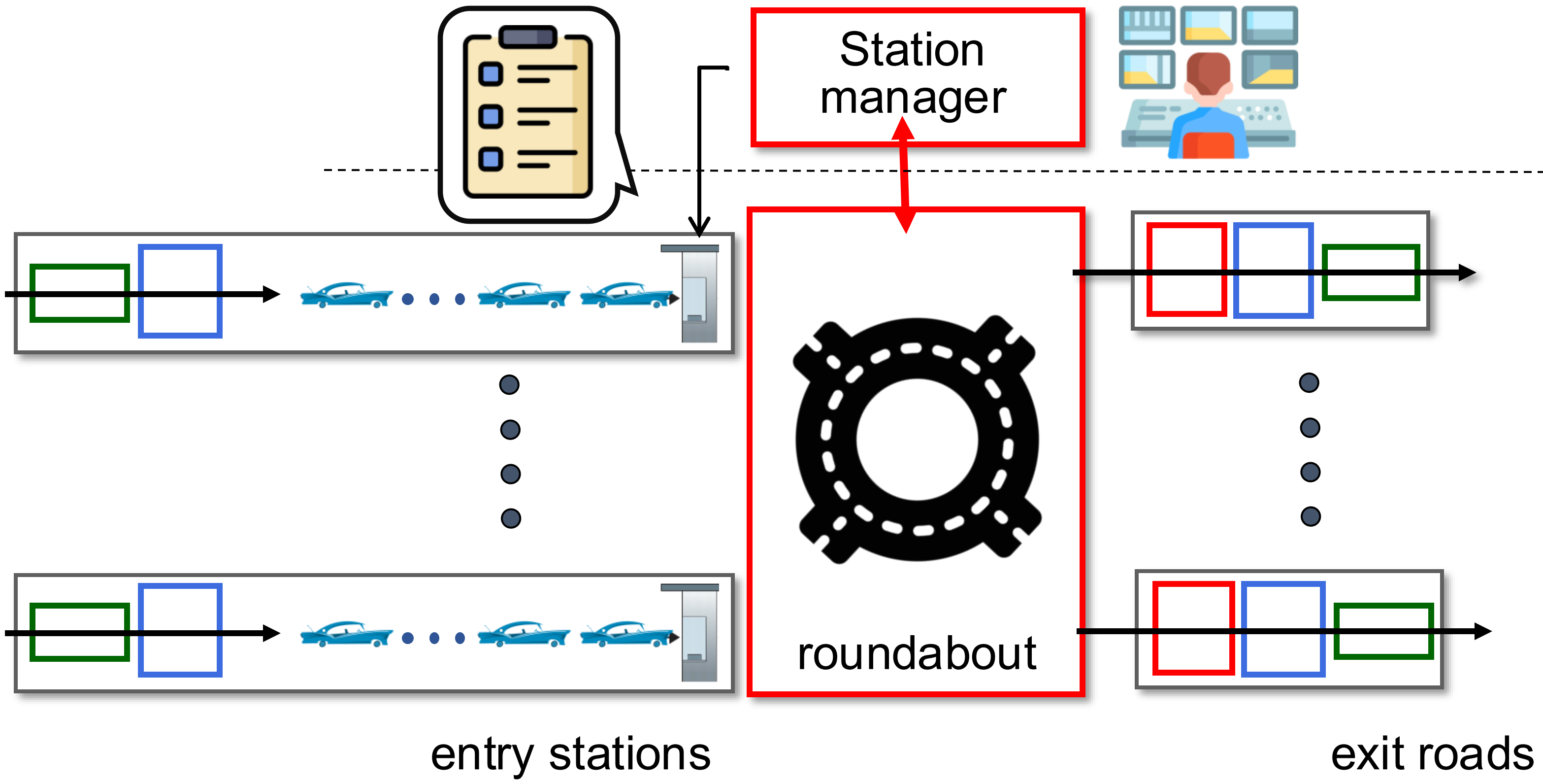


routing, management
control plane (software)
operates in millisecond
time frame

forwarding data plane
(hardware) operates
in nanosecond
timeframe

Router architecture overview

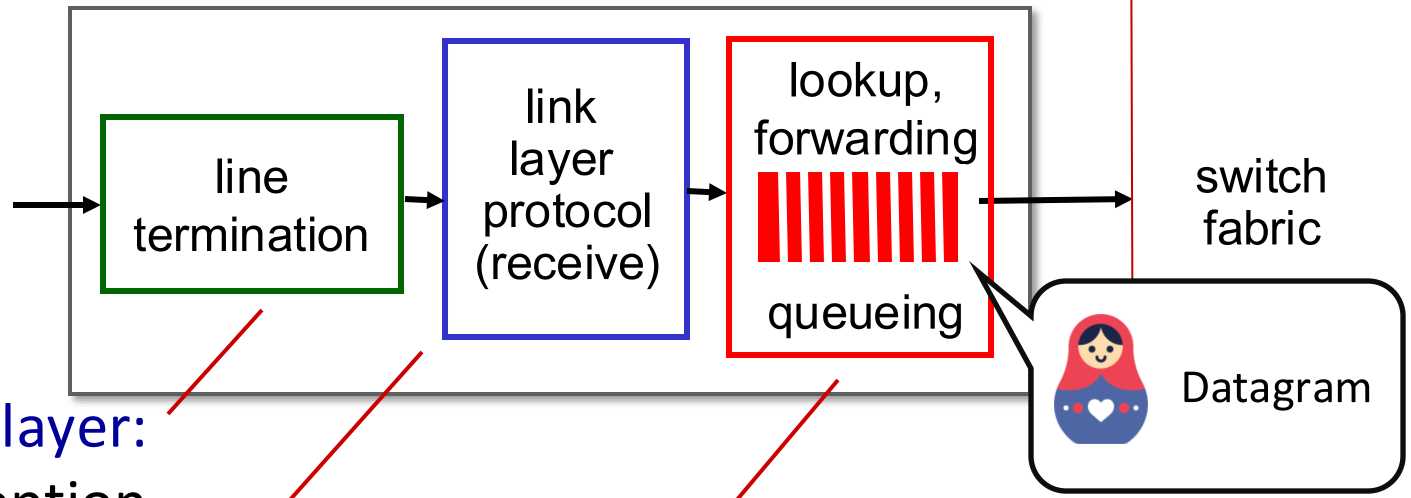
analogy view of generic router architecture:



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

forwarding data plane
(hardware) operates
in nanosecond
timeframe

Input port functions



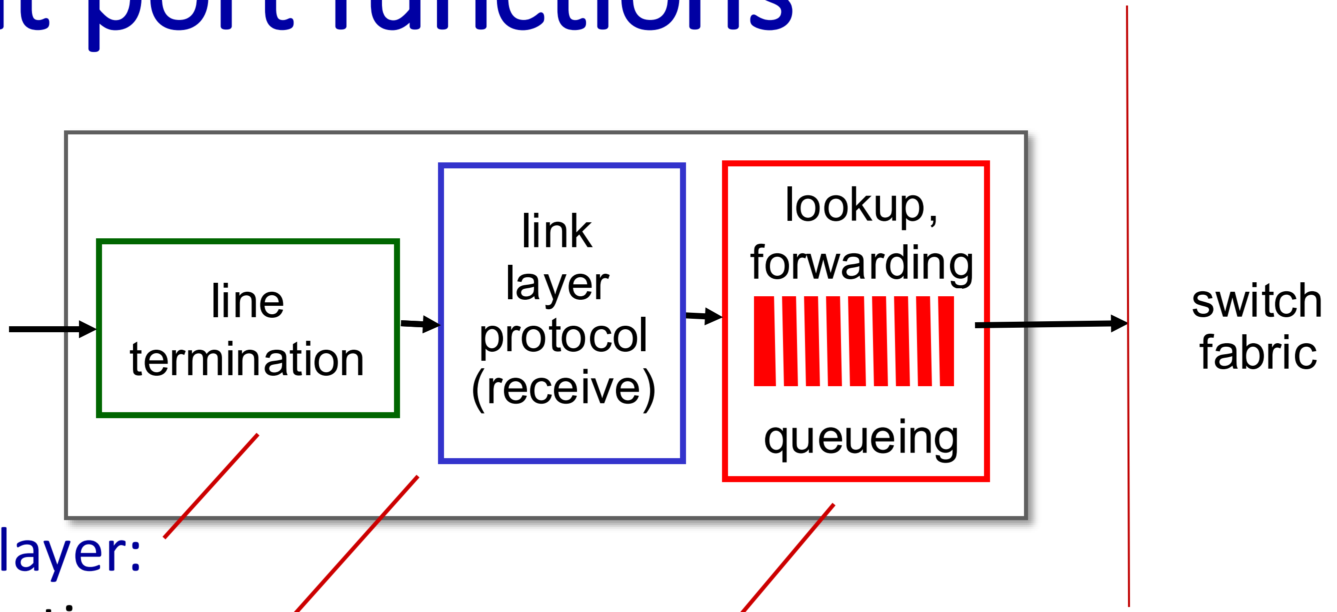
physical layer:
bit-level reception

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	0
11001000 00010111 00010000 00000100 through 11001000 00010111 00010000 00000111	3
11001000 00010111 00011000 11111111	
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 00010111 00011000 ********	1
11001000 00010111 00011*** ********	2
otherwise	3

match!

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	00010111	00011000	********	1
11001000	00010111	00011***	********	2
otherwise				3

match!

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	00010111	00011000	********	1
11001000	00010111	00011***	********	2
otherwise				3

match!

examples:

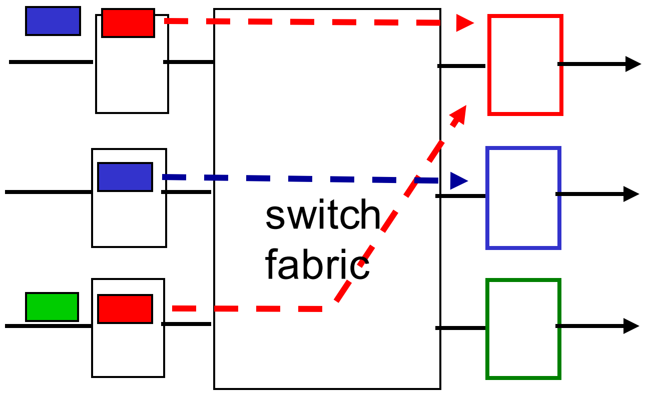
11001000	00010111	00010110	10100001	which interface?
11001000	00010111	00011000	10101010	which interface?

Longest prefix matching

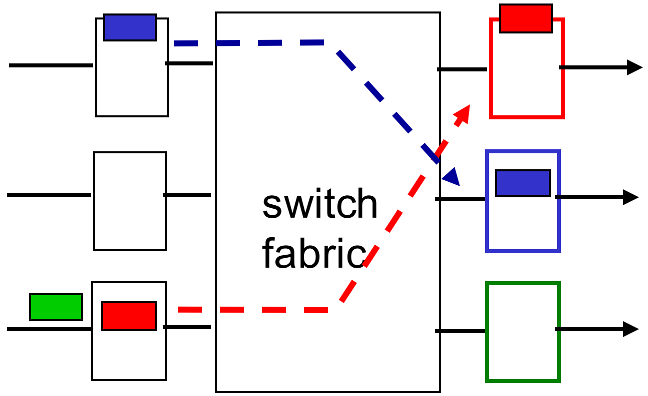
- we’ll see *why* longest prefix matching is used shortly, when we study addressing
- longest prefix matching: often performed using ternary content addressable memories (TCAMs)
  - *content addressable:* present address to TCAM: retrieve address in one clock cycle, regardless of table size
  - Cisco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

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

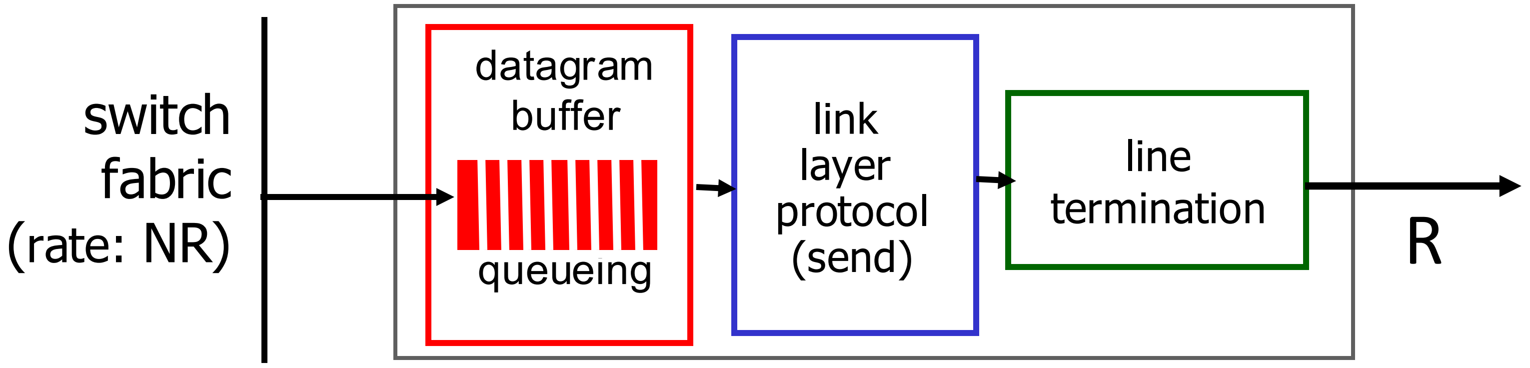


output port contention: only one red datagram can be transferred. lower red packet is *blocked*



one packet time later: green packet experiences HOL blocking

Output port queuing



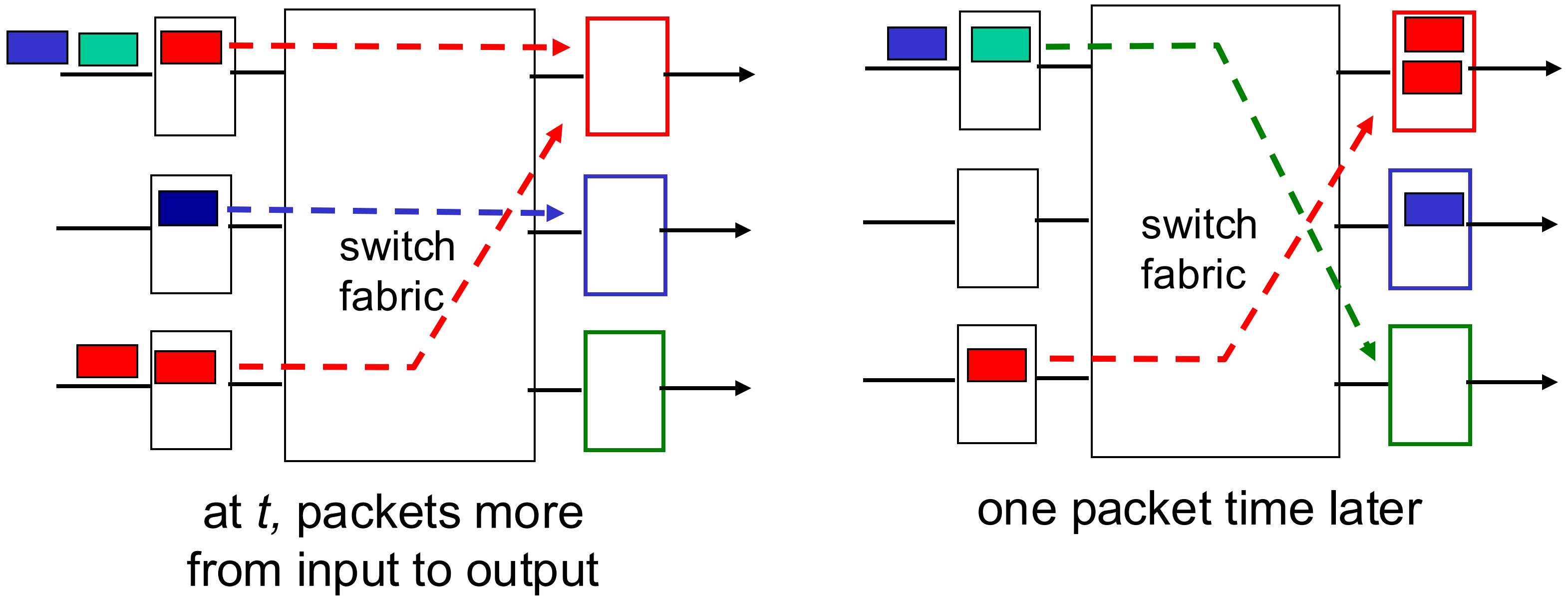
- *Buffering* required when datagrams arrive from fabric faster than link transmission rate. *Drop policy:* which datagrams to drop if no free buffers?

Datagrams can be lost due to congestion, lack of buffers

- *Scheduling discipline* chooses among queued datagrams for transmission

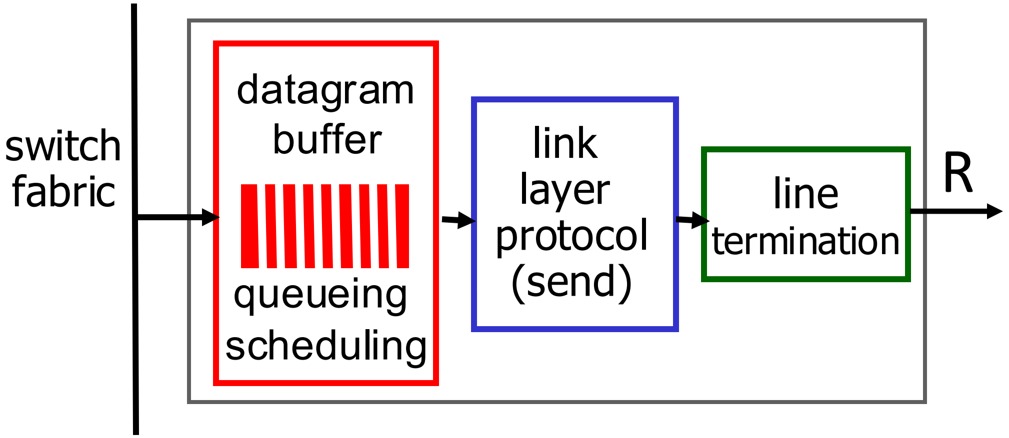
Priority scheduling – who gets best performance, network neutrality

Output port queuing

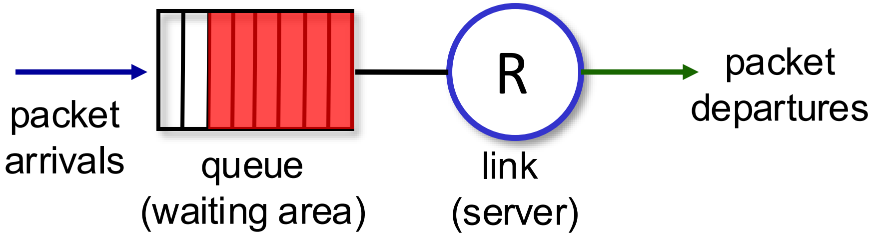


- buffering when arrival rate via switch exceeds output line speed
- *queueing (delay) and loss due to output port buffer overflow!*

Buffer Management



Abstraction: queue



buffer management:

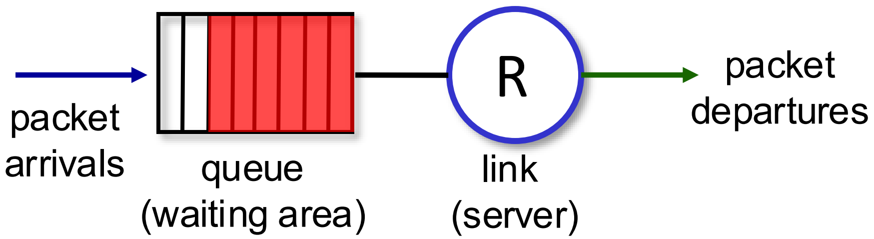
- drop: which packet to add, drop when buffers are full
  - tail drop: drop arriving packet
  - priority: drop/remove on priority basis
- marking: which packets to mark to signal congestion (ECN, RED)

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



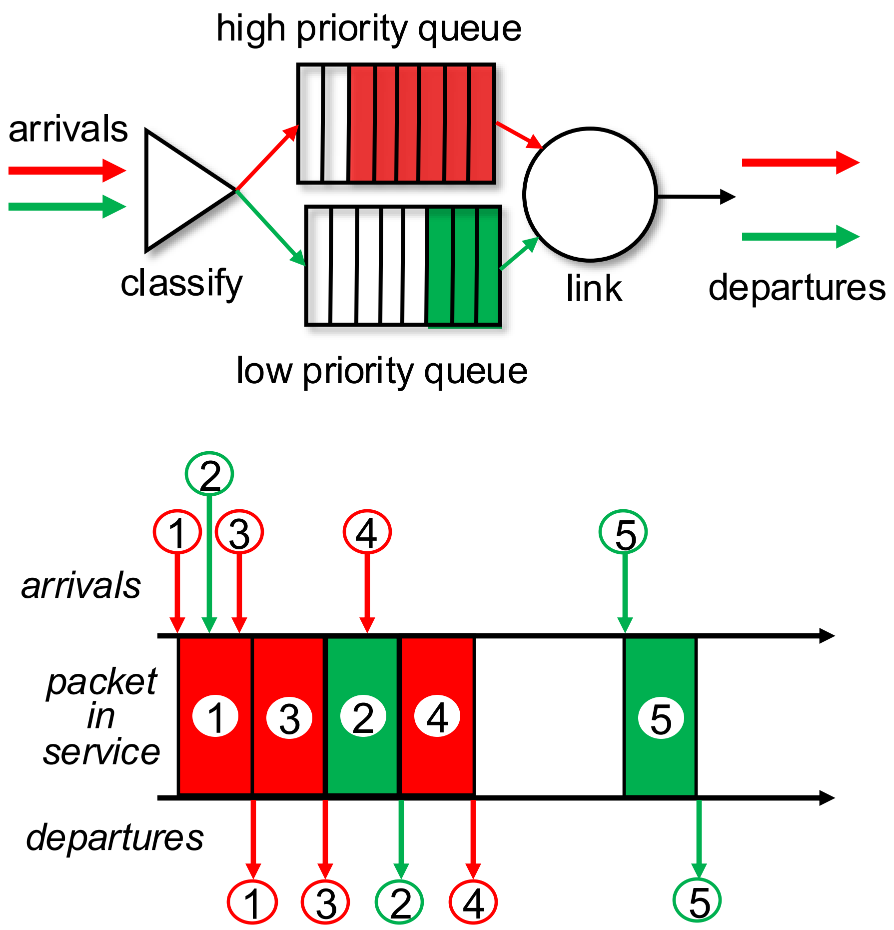
FCFS: packets transmitted in order of arrival to output port

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

Scheduling policies: priority

Priority scheduling:

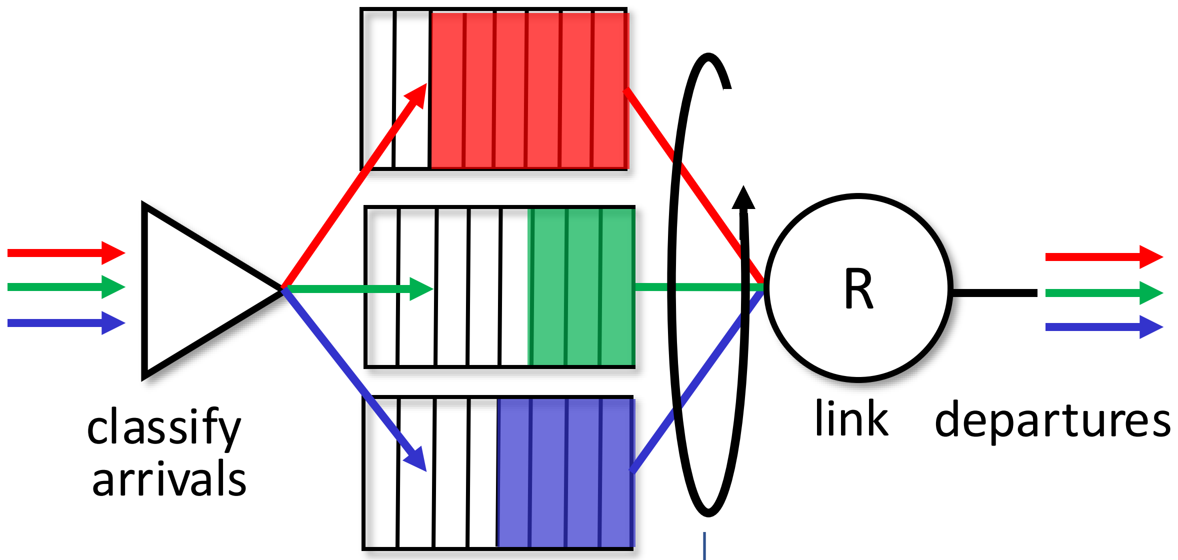
- arriving traffic classified, queued by class
  - any header fields can be used for classification
- send packet from highest priority queue that has buffered packets
  - FCFS within priority class



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



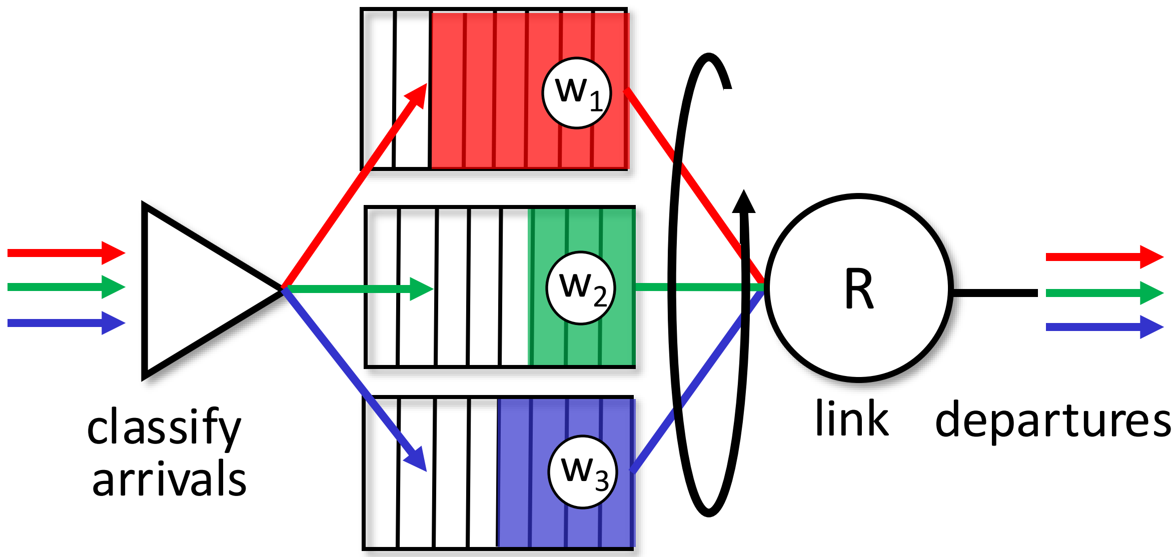
Scheduling policies: weighted fair queueing

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)



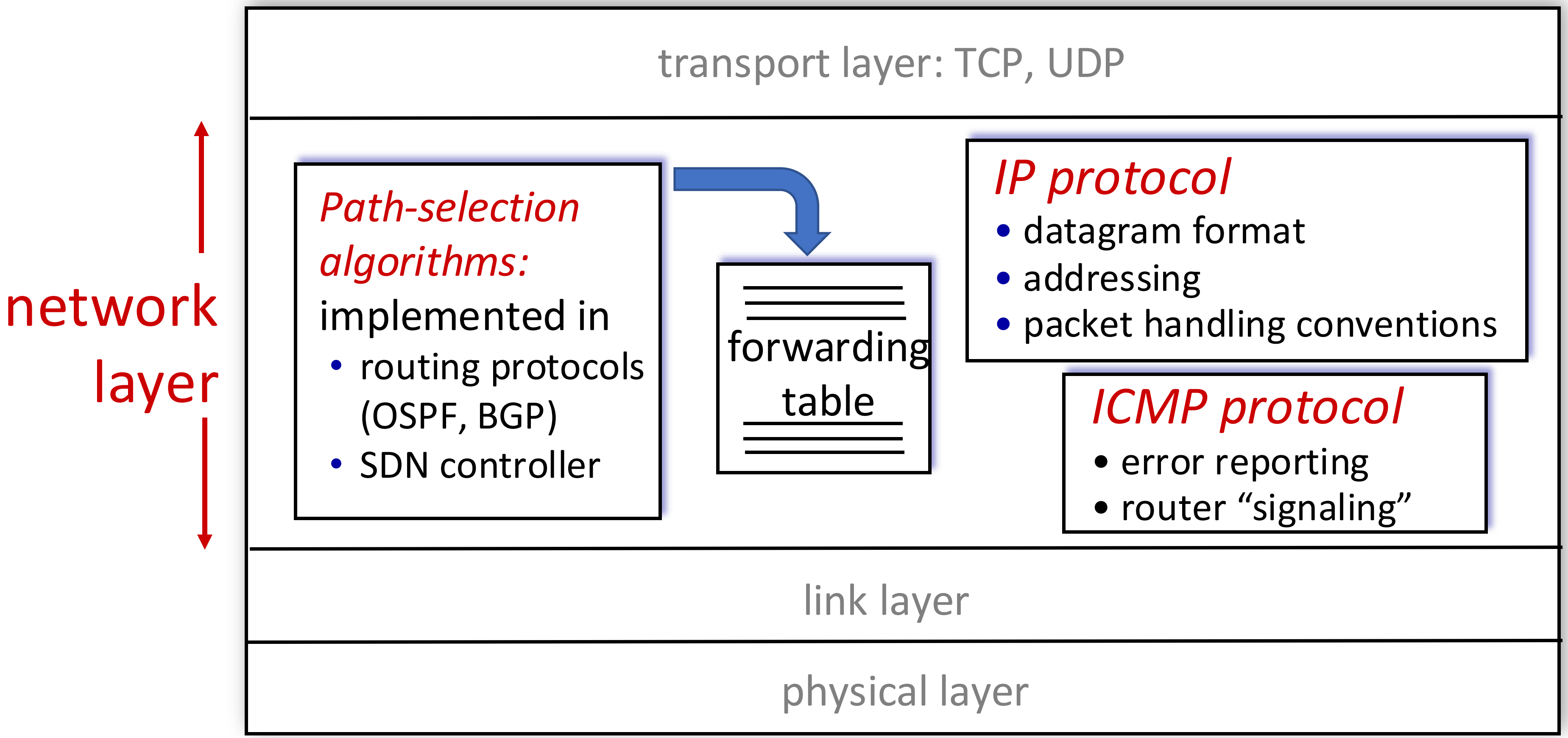
Network layer: "data plane" roadmap

- 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

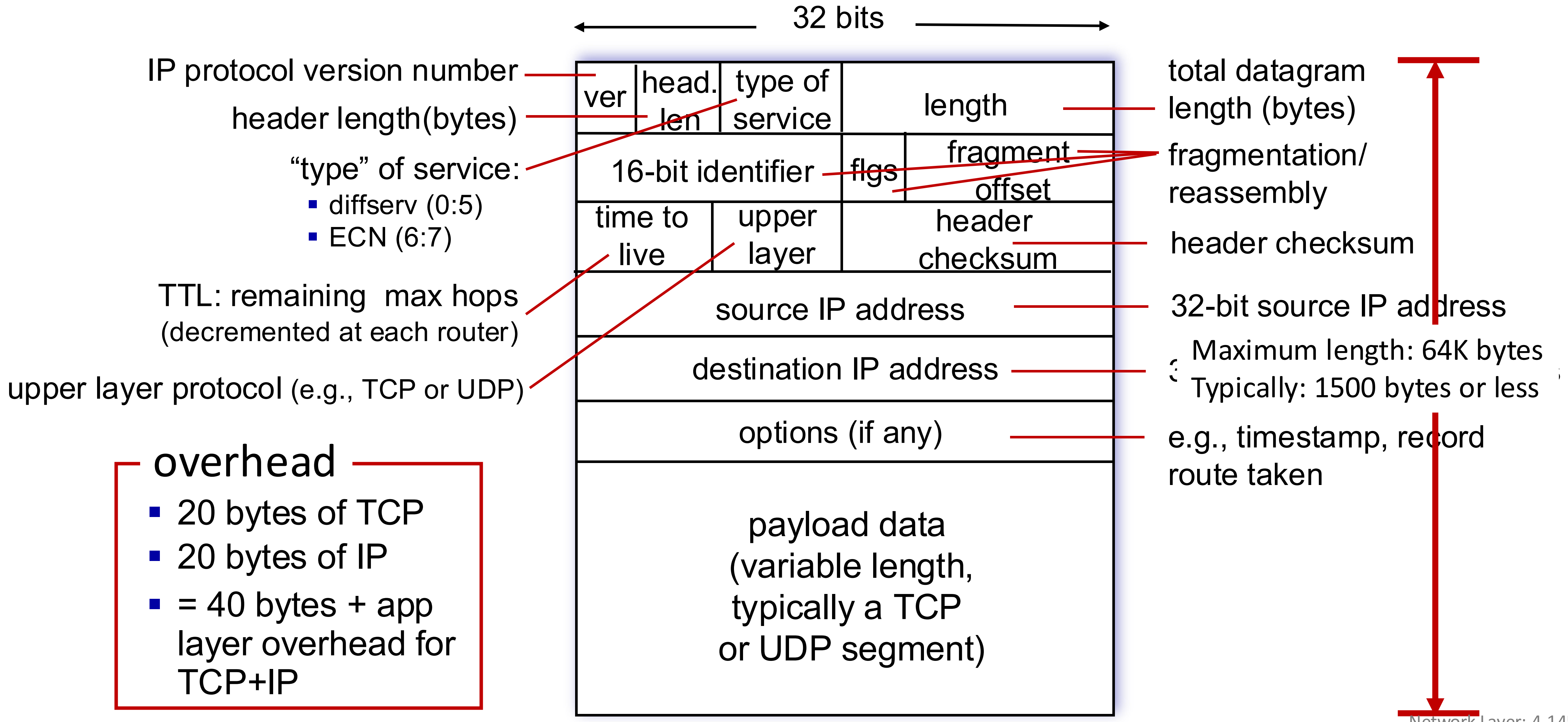
- Generalized Forwarding, SDN
  - match+action
  - OpenFlow: match+action in action
- Middleboxes

Network Layer: Internet

host, router network layer functions:



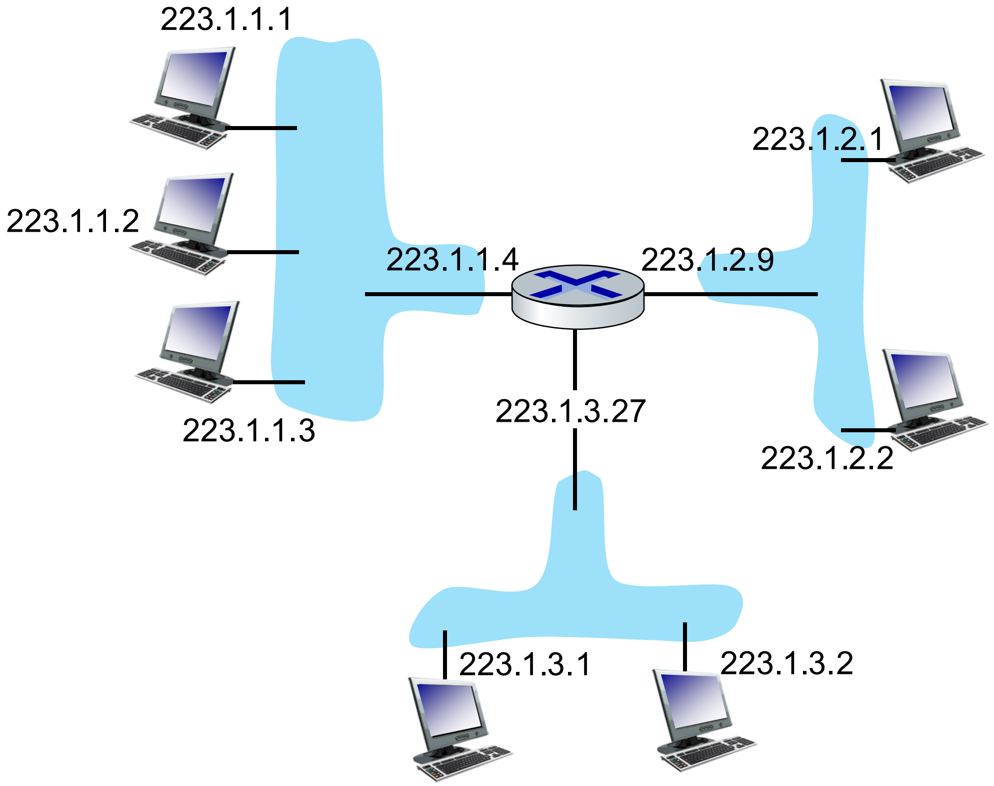
IP Datagram format



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:

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

IP addressing: introduction

- IP address: 32-bit identifier associated with each host or router *interface*
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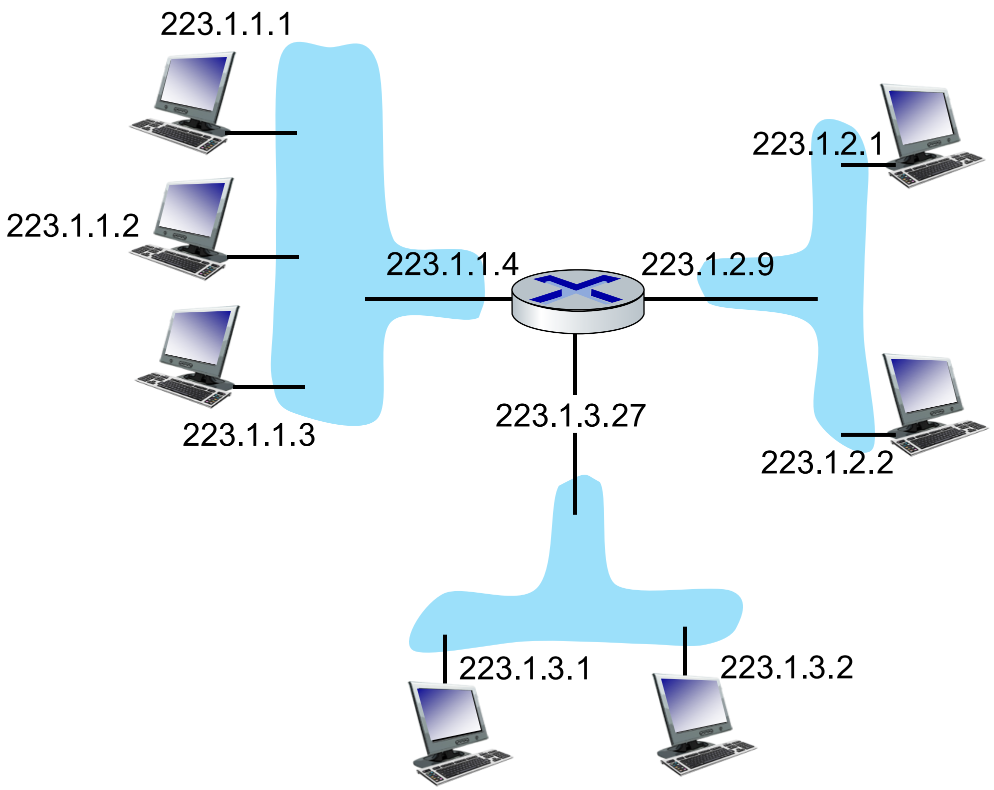


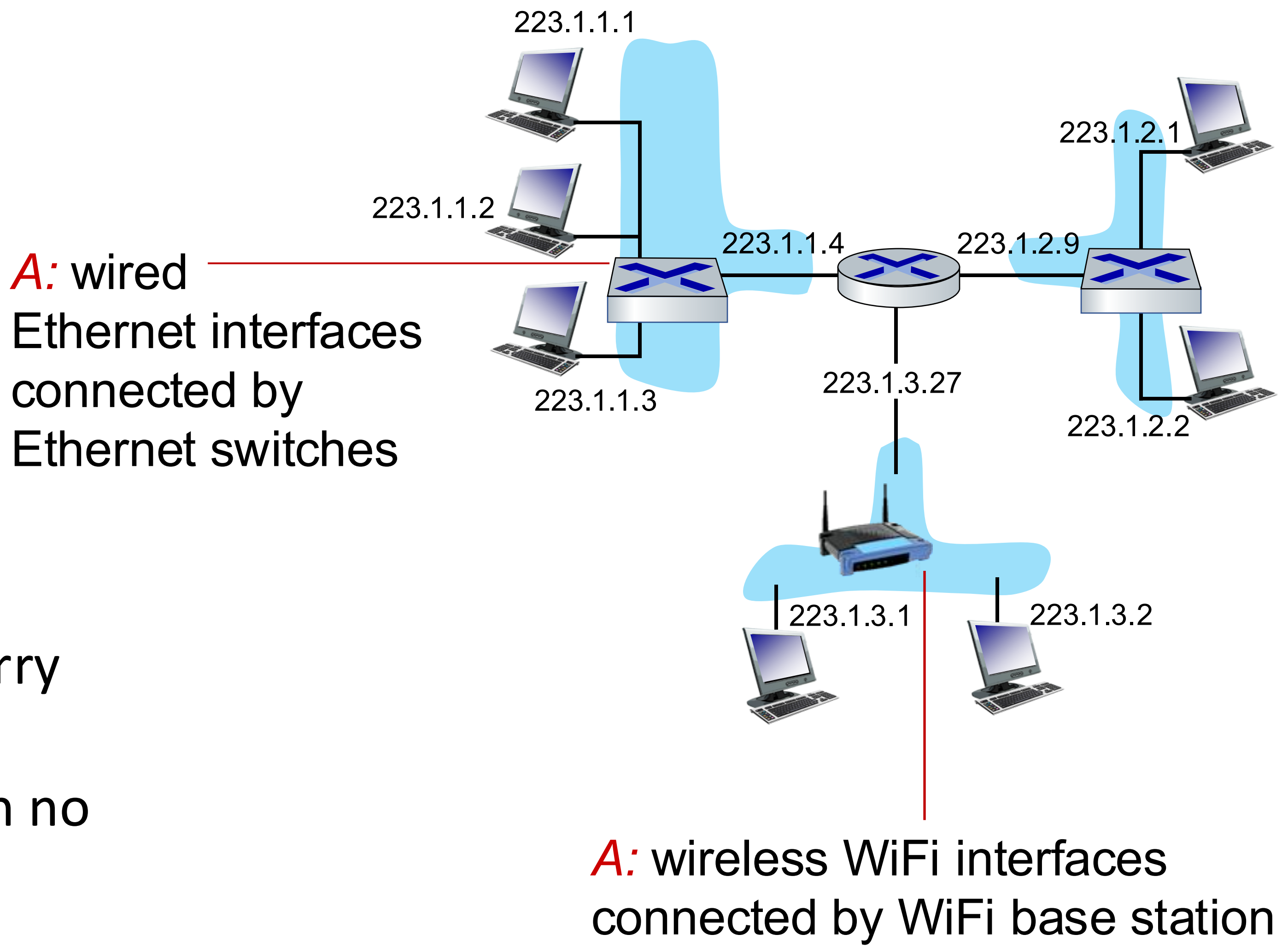
dotted-decimal IP address notation:

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

IP addressing: introduction

Q: how are interfaces actually connected?

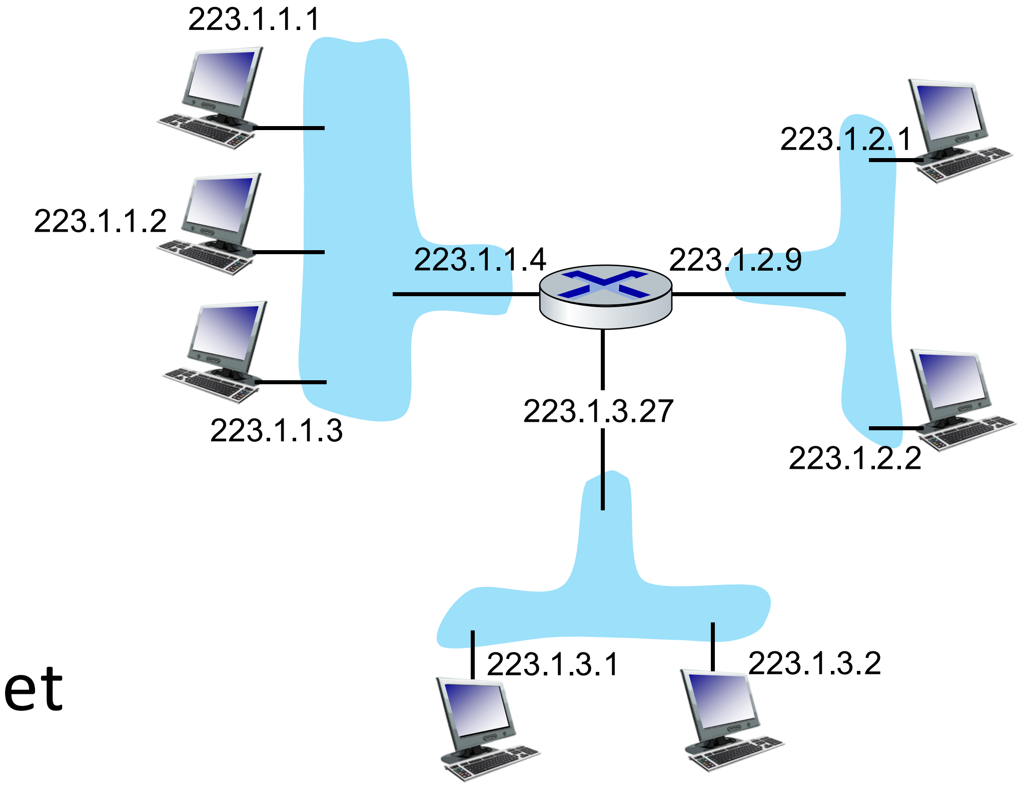
A: we'll learn about that in chapters 6, 7



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

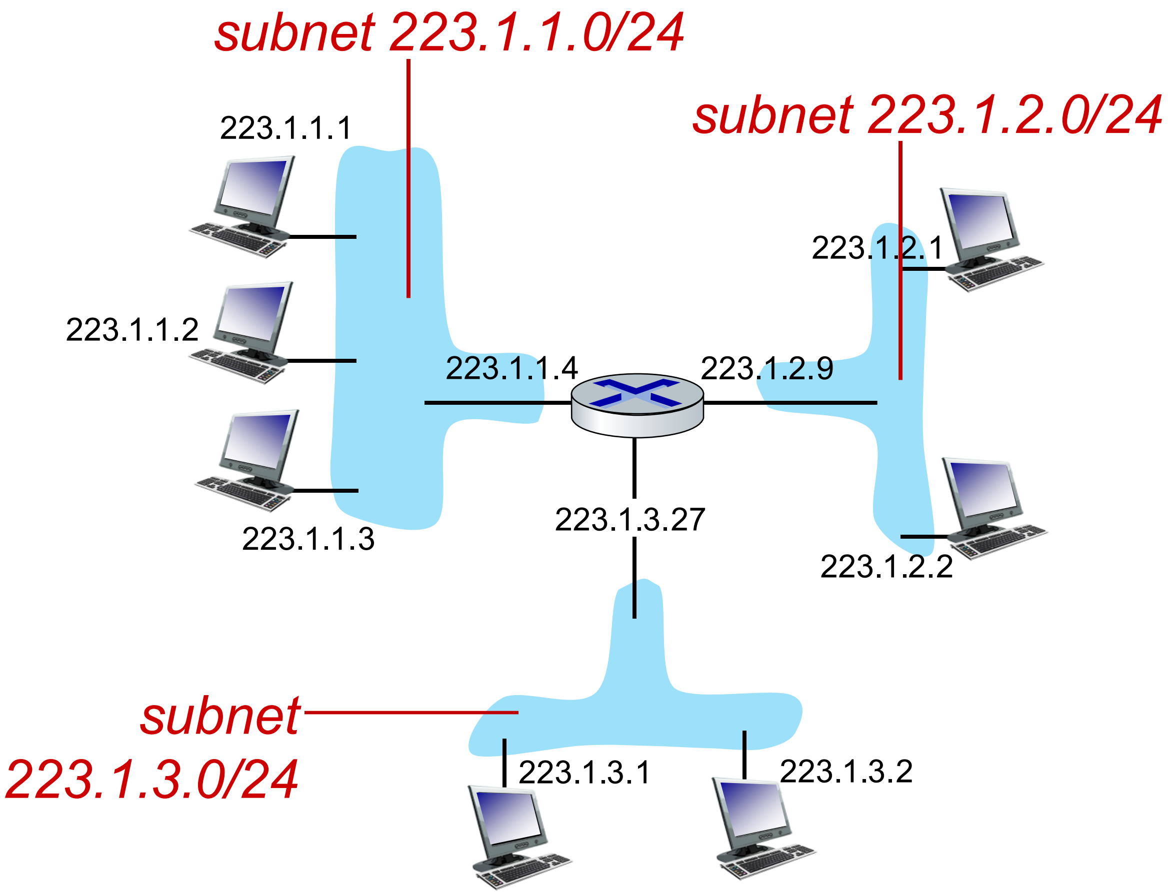


network consisting of 3 subnets

Subnets

Recipe for defining subnets:

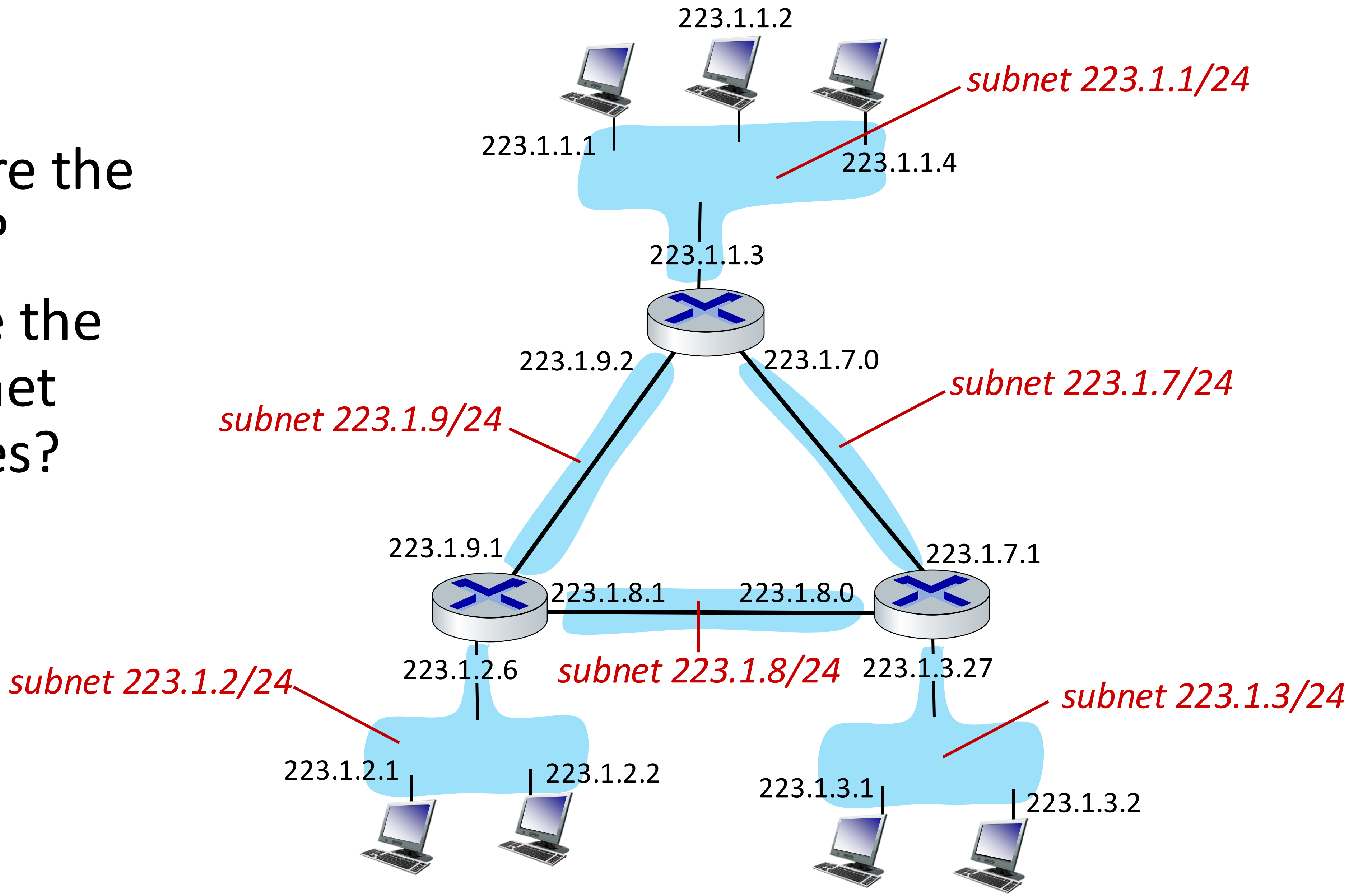
- detach each interface from its host or router, creating "islands" of isolated networks
- each isolated network is called a *subnet*



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

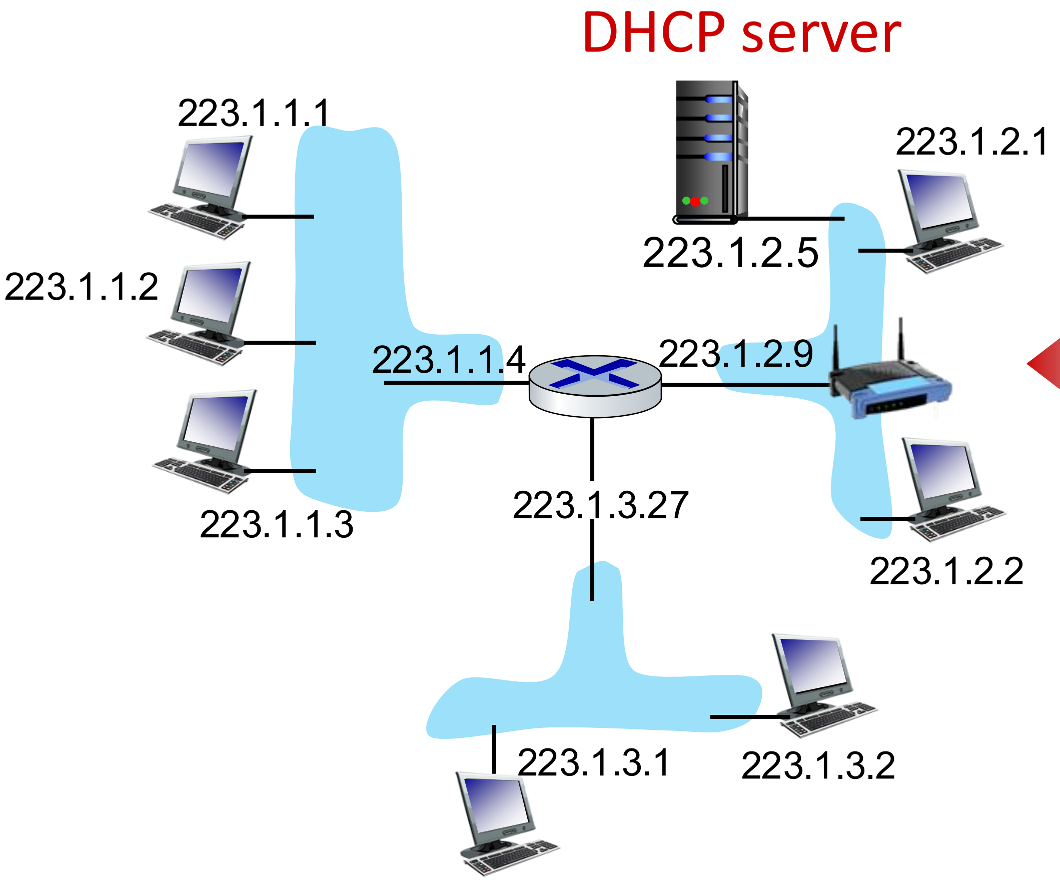
goal: host *dynamically* obtains IP address from network server when it "joins" network

- 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

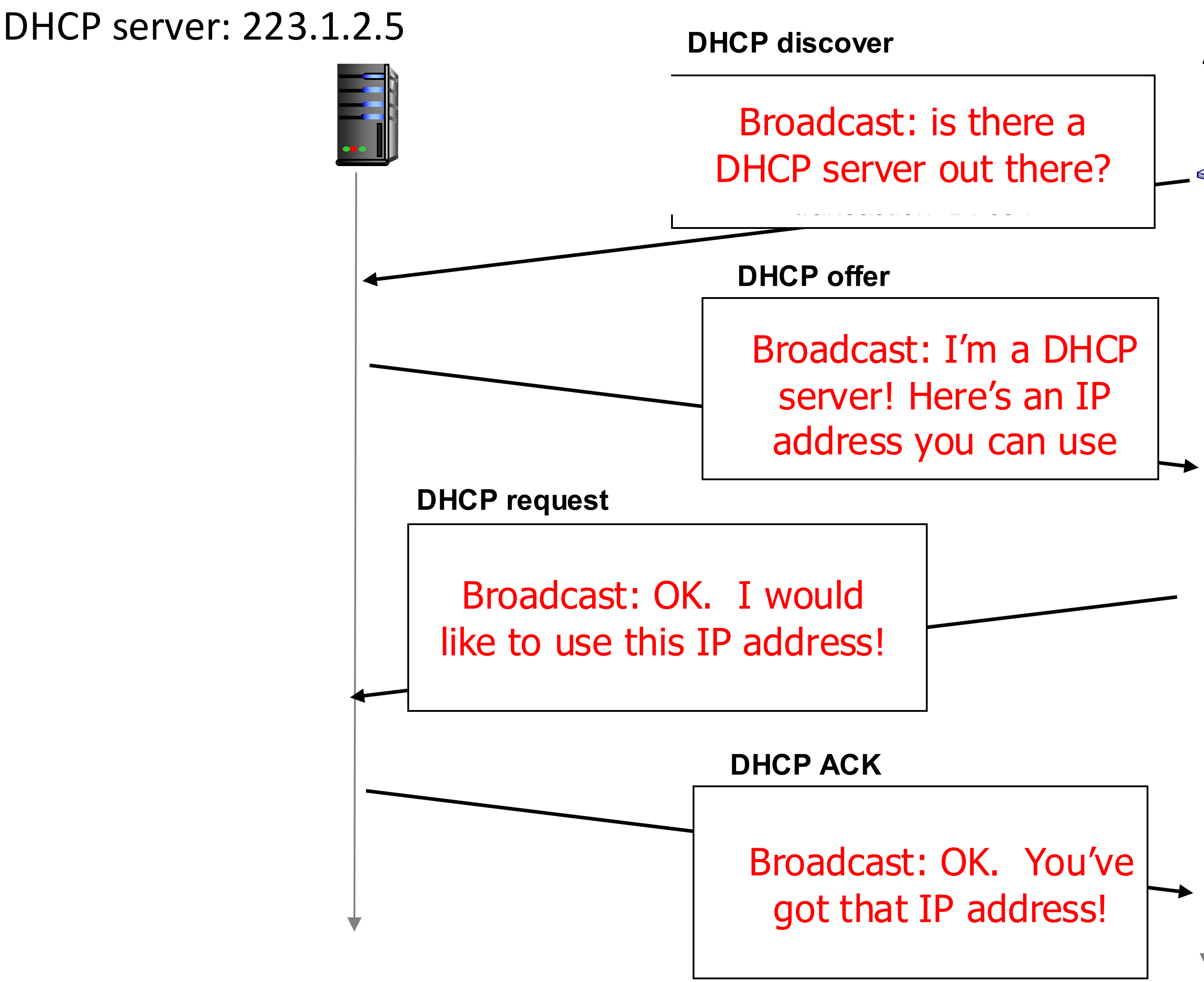
DHCP client-server scenario



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



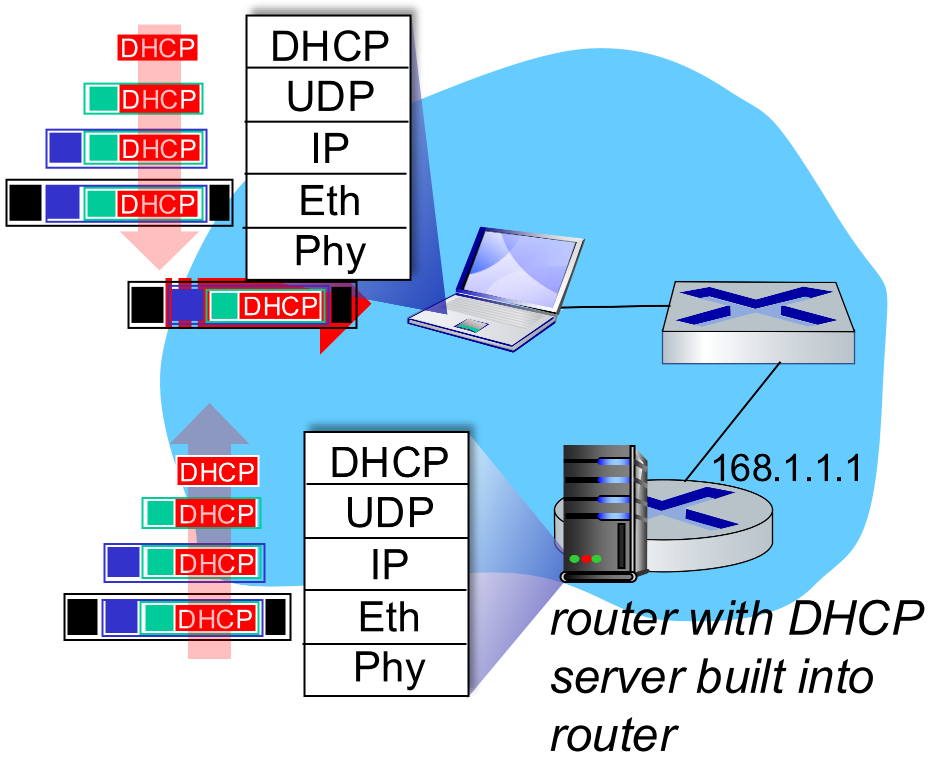
The two steps above can be skipped "if a client remembers and wishes to reuse a previously allocated network address" [RFC 2131]

DHCP: more than IP addresses

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

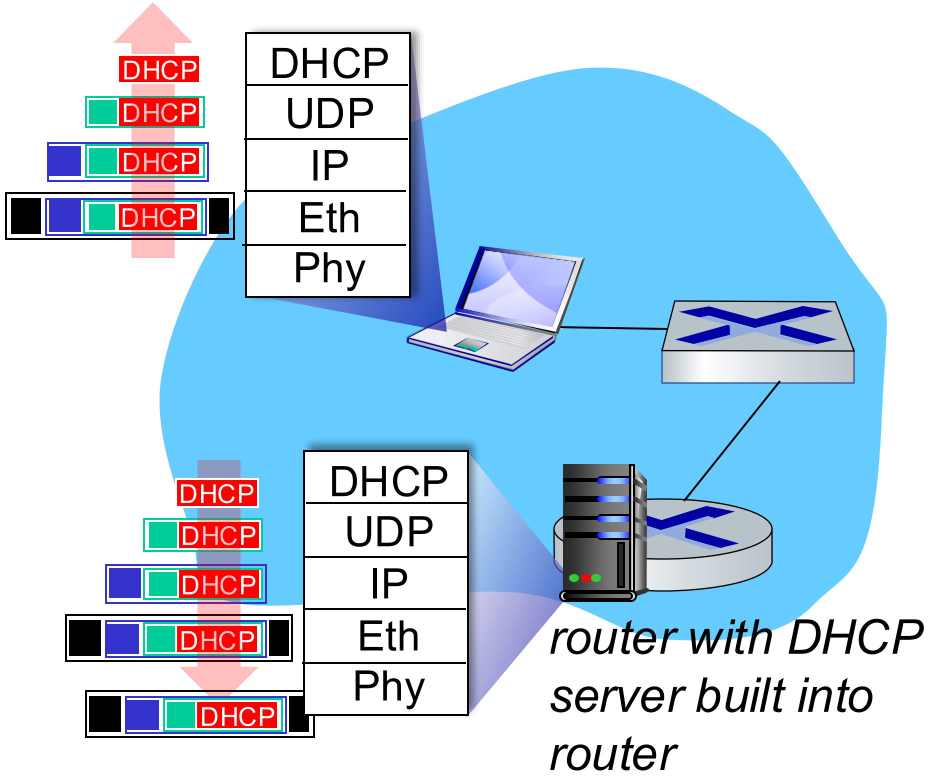
- address of first-hop router for client
- name and IP address of DNS sever
- network mask (indicating network versus host portion of address)

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

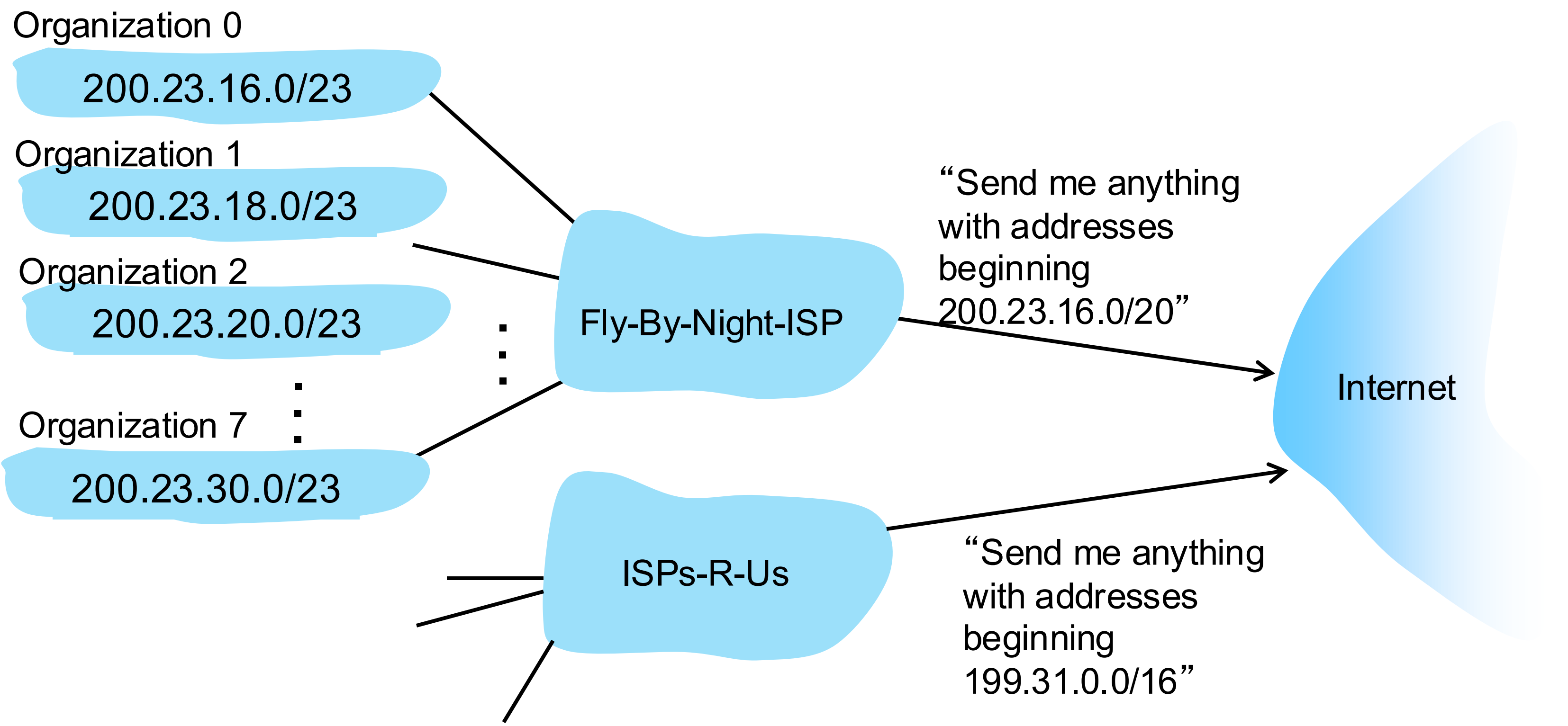
ISP's block	11001000 00010111 00010000 00000000	200.23.16.0/20

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

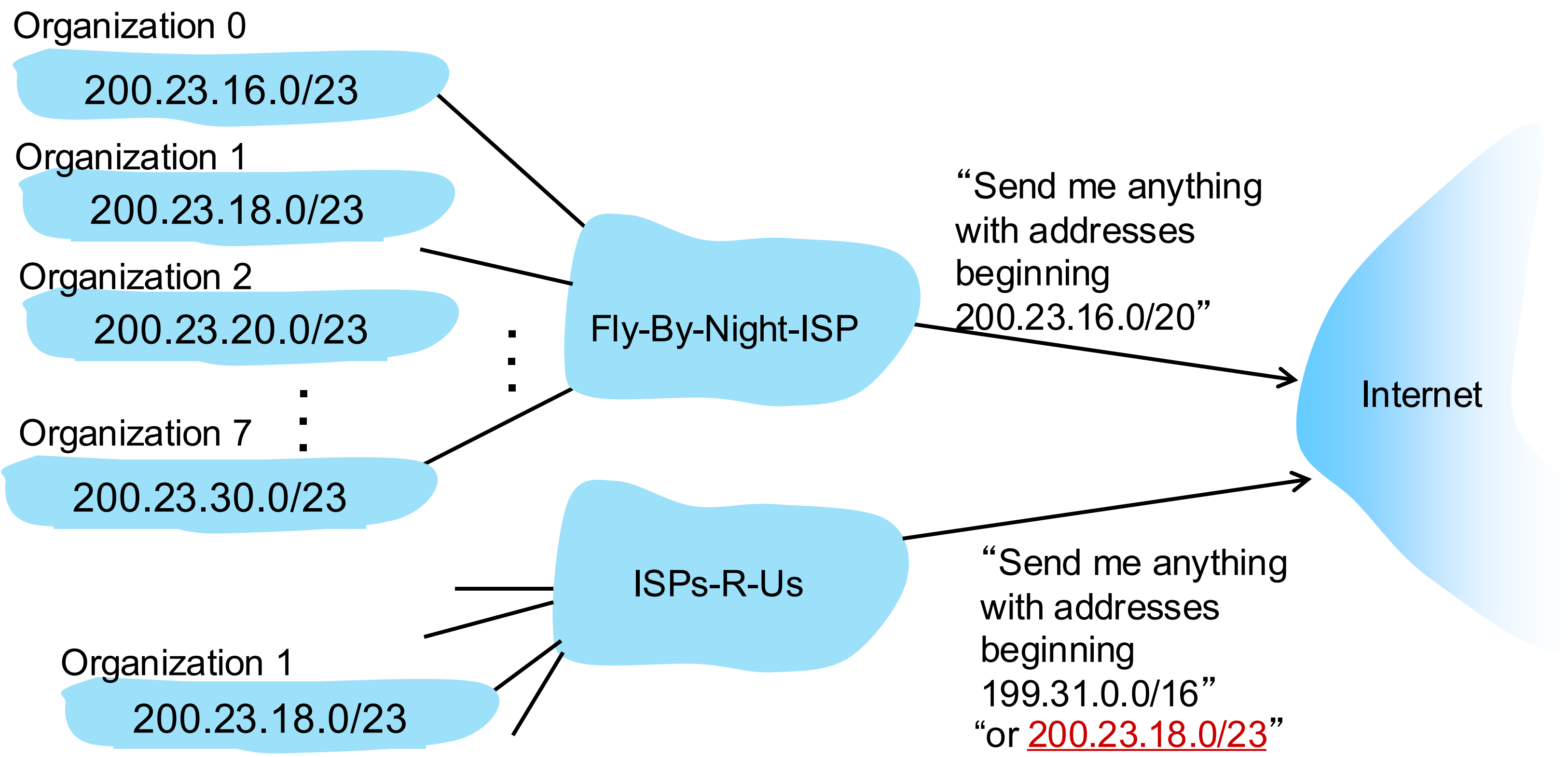
Hierarchical addressing: route aggregation

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



Hierarchical addressing: more specific routes

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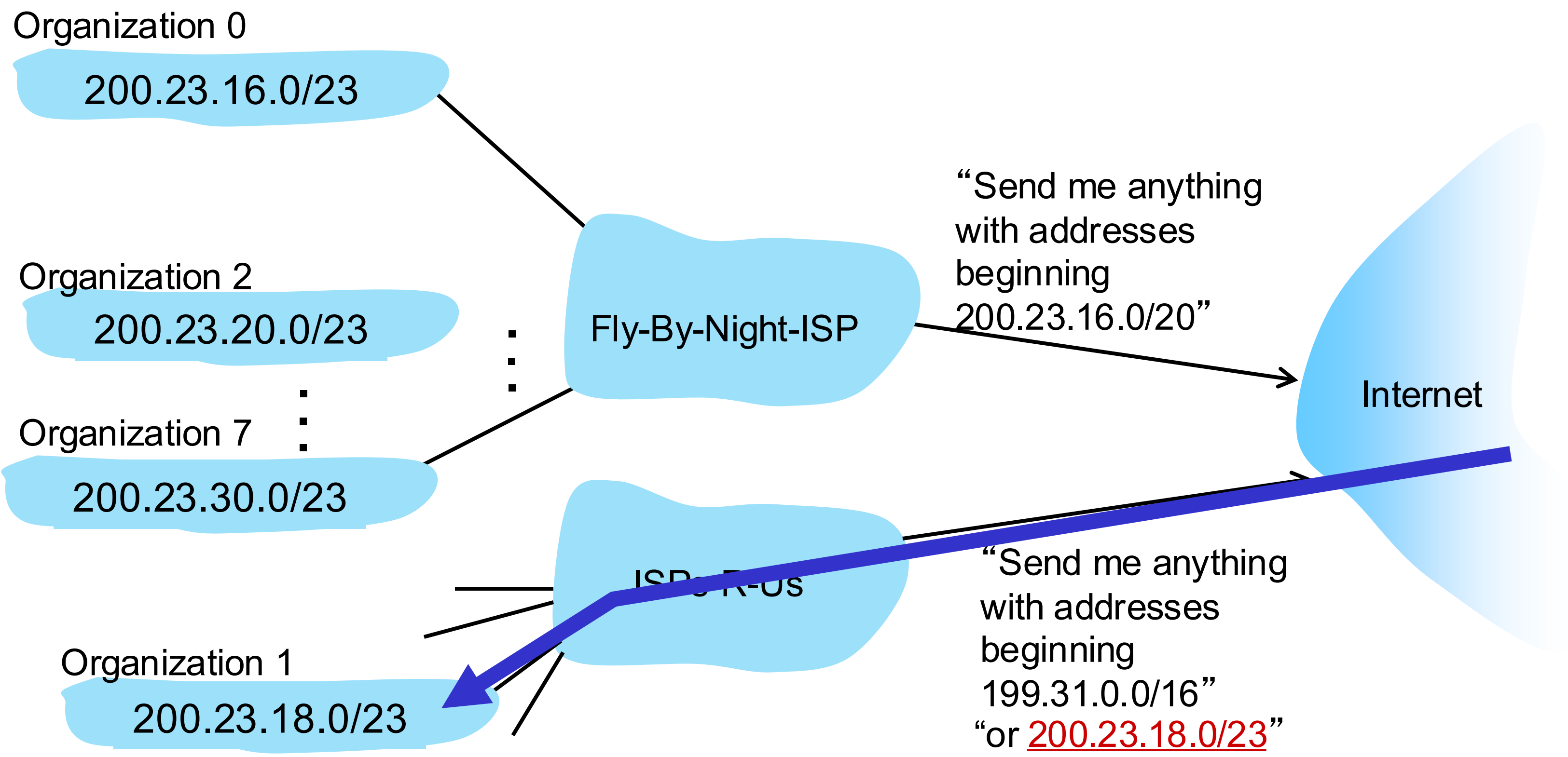


IP addressing: last words ...

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)

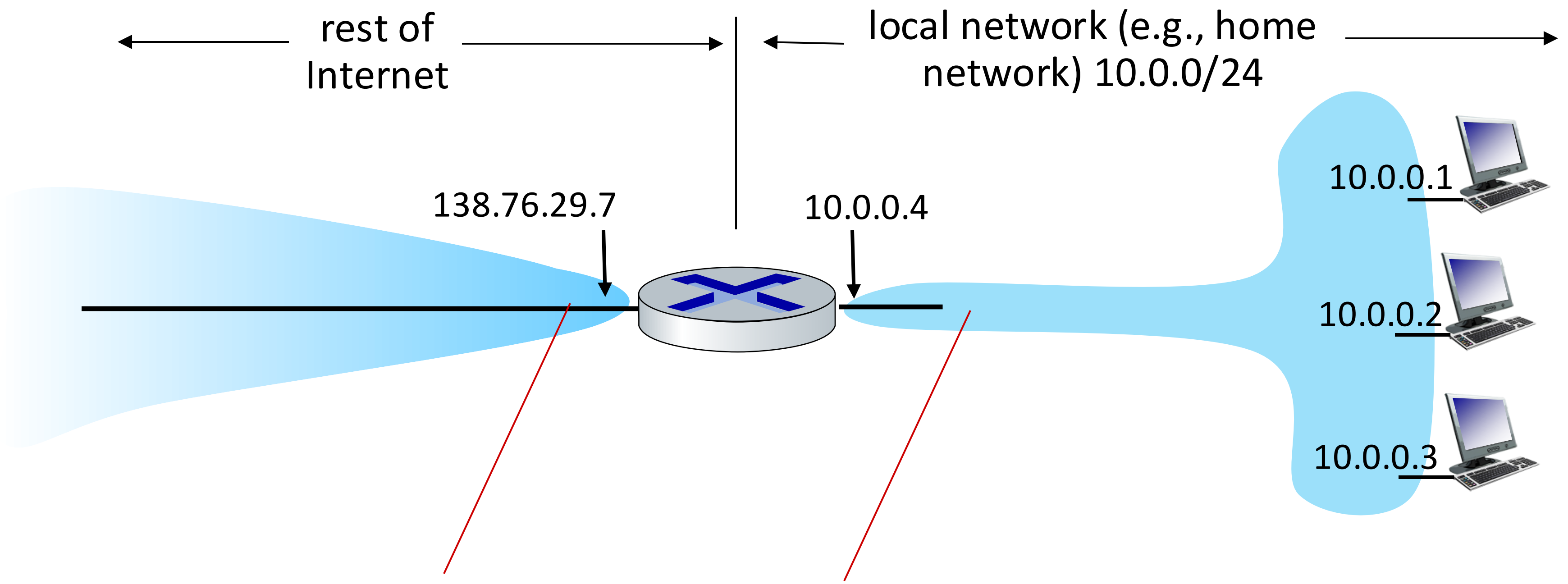
Network layer: “data plane” roadmap

- 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

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

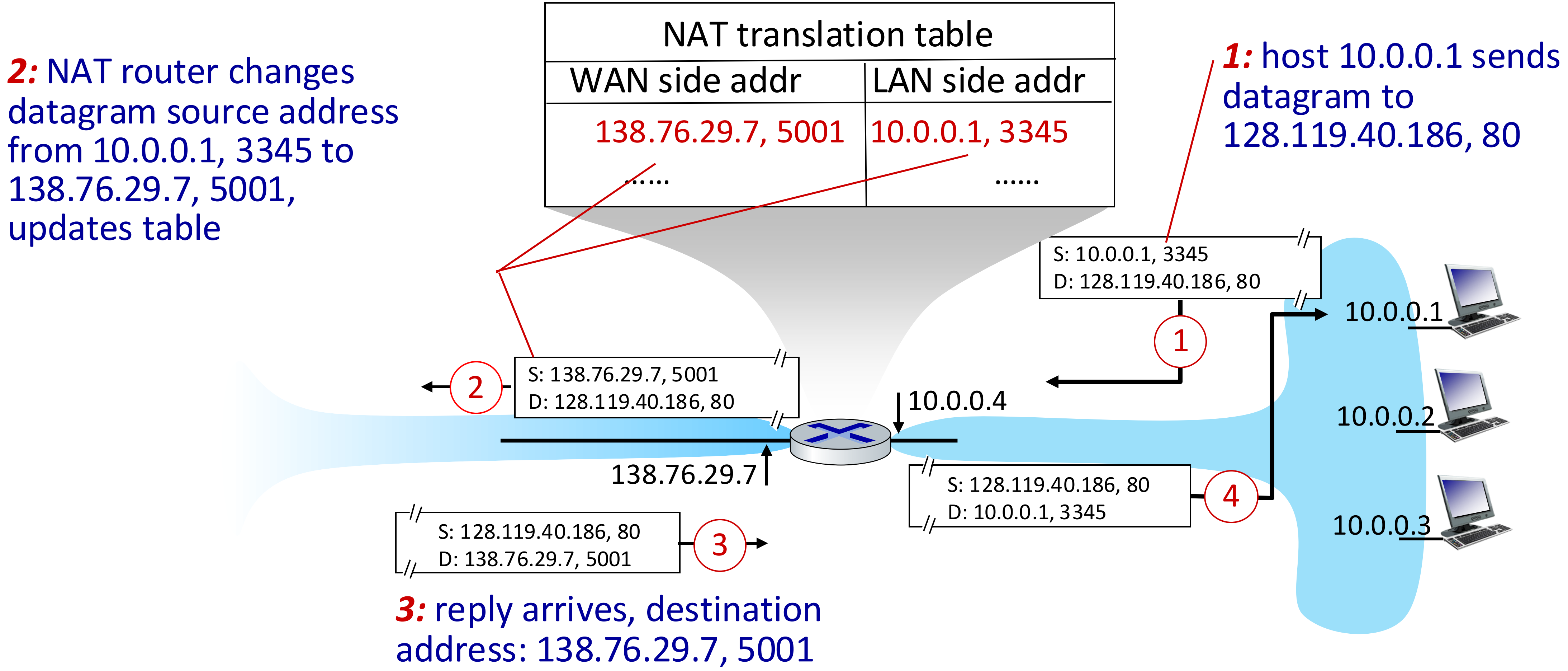
- all devices in local network have 32-bit addresses in a "private" IP address space (10/8, 172.16/12, 192.168/16 prefixes) that can only be used in local network
- advantages:
  - just one IP address needed from provider ISP for *all* devices
  - can change addresses of host in local network without 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

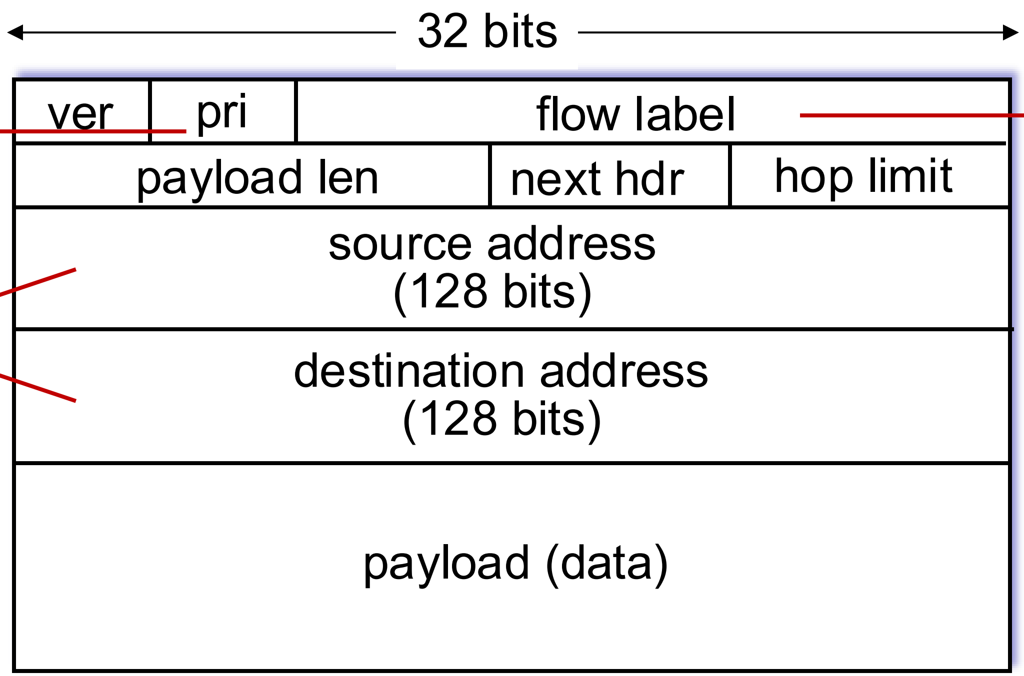
IPv6: motivation

- 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

priority: identify priority among datagrams in flow

128-bit IPv6 addresses



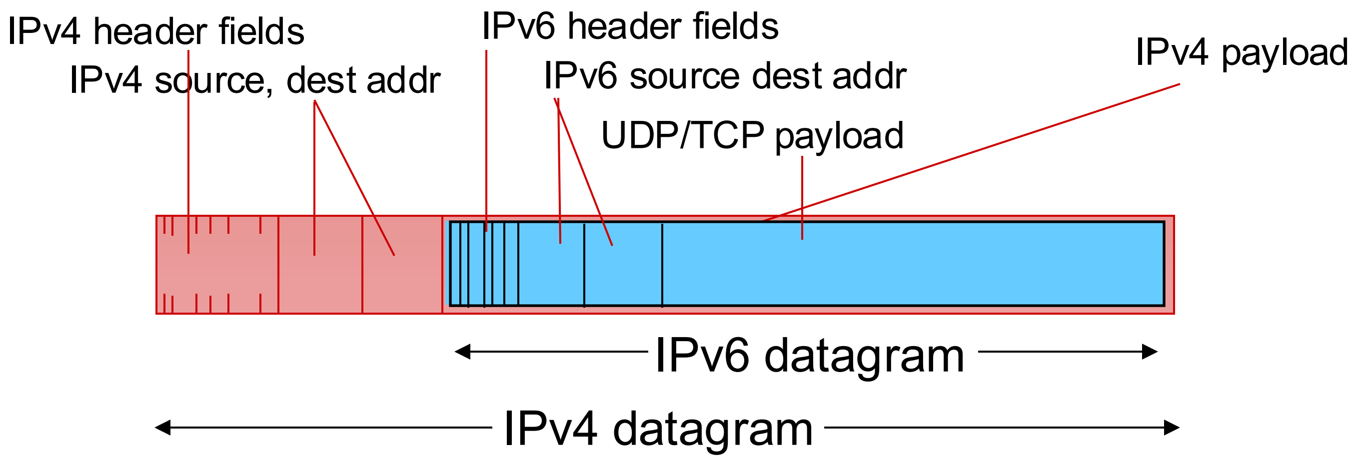
flow label: identify datagrams in same "flow." (concept of "flow" not well defined).

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)

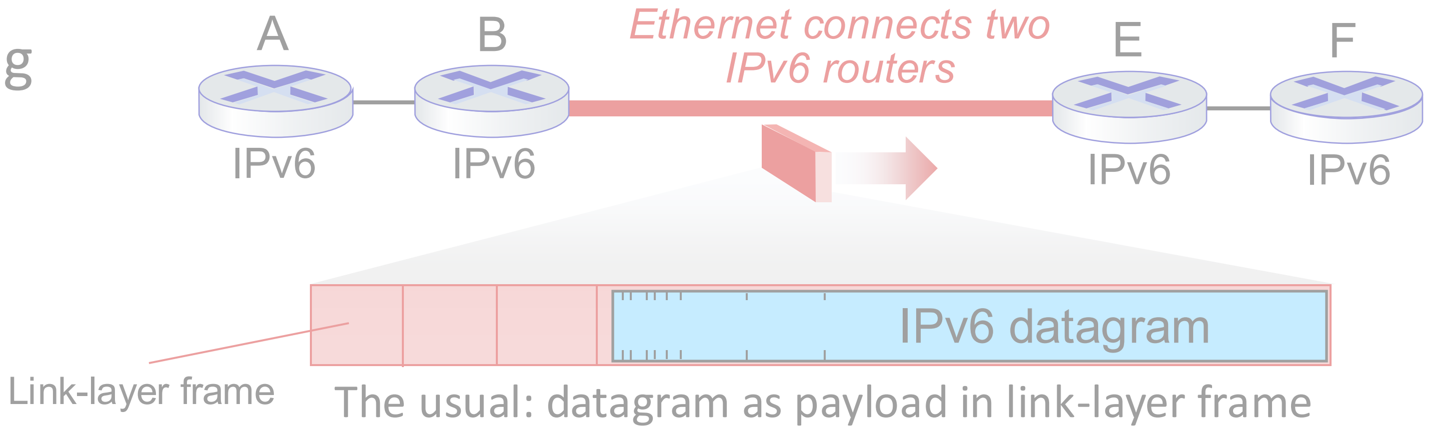
Transition from IPv4 to IPv6

- not all routers can be upgraded simultaneously
  - no "flag days"
  - how will network operate with mixed IPv4 and IPv6 routers?
- tunneling: IPv6 datagram carried as *payload* in IPv4 datagram among IPv4 routers ("packet within a packet")
  - tunneling used extensively in other contexts (4G/5G)

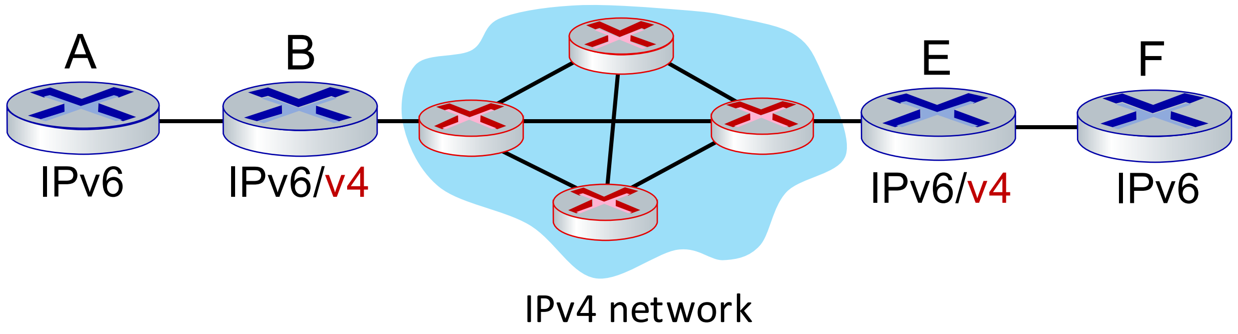


Tunneling and encapsulation

Ethernet connecting two IPv6 routers:

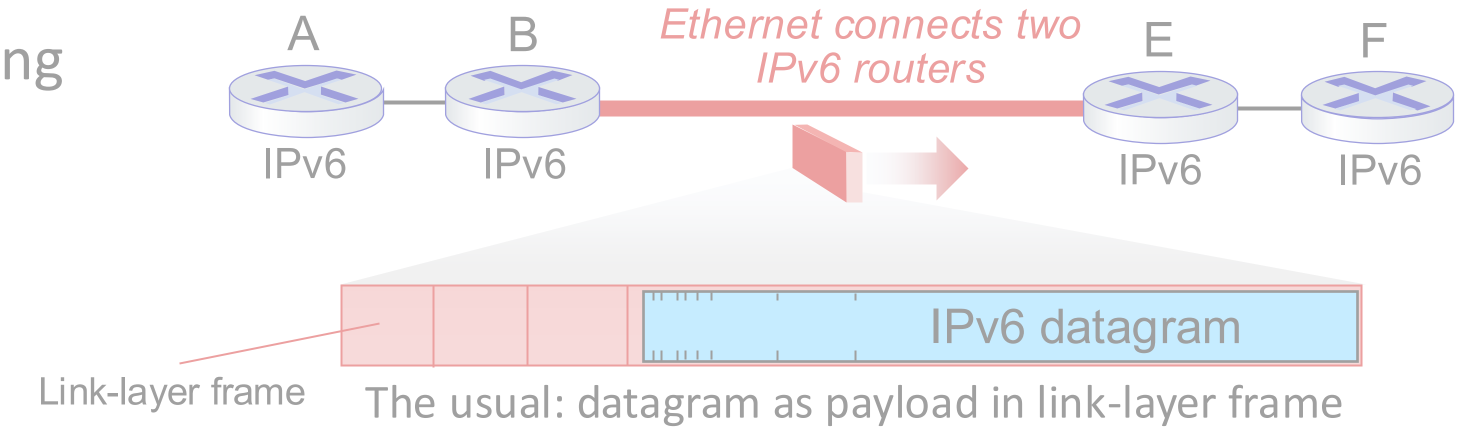


IPv4 network connecting two IPv6 routers



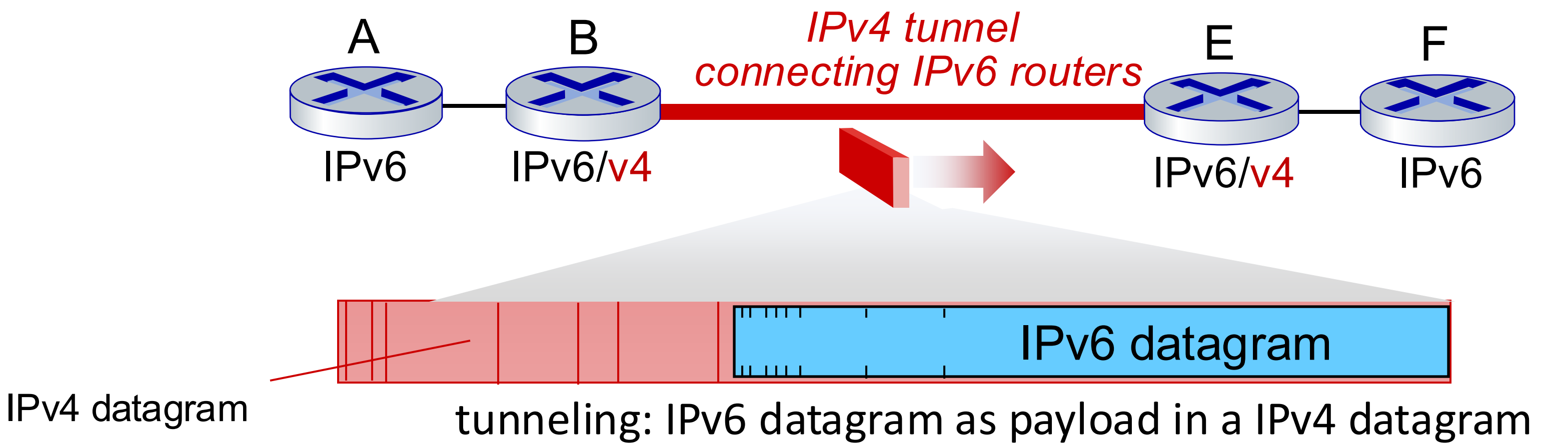
Tunneling and encapsulation

Ethernet connecting two IPv6 routers:



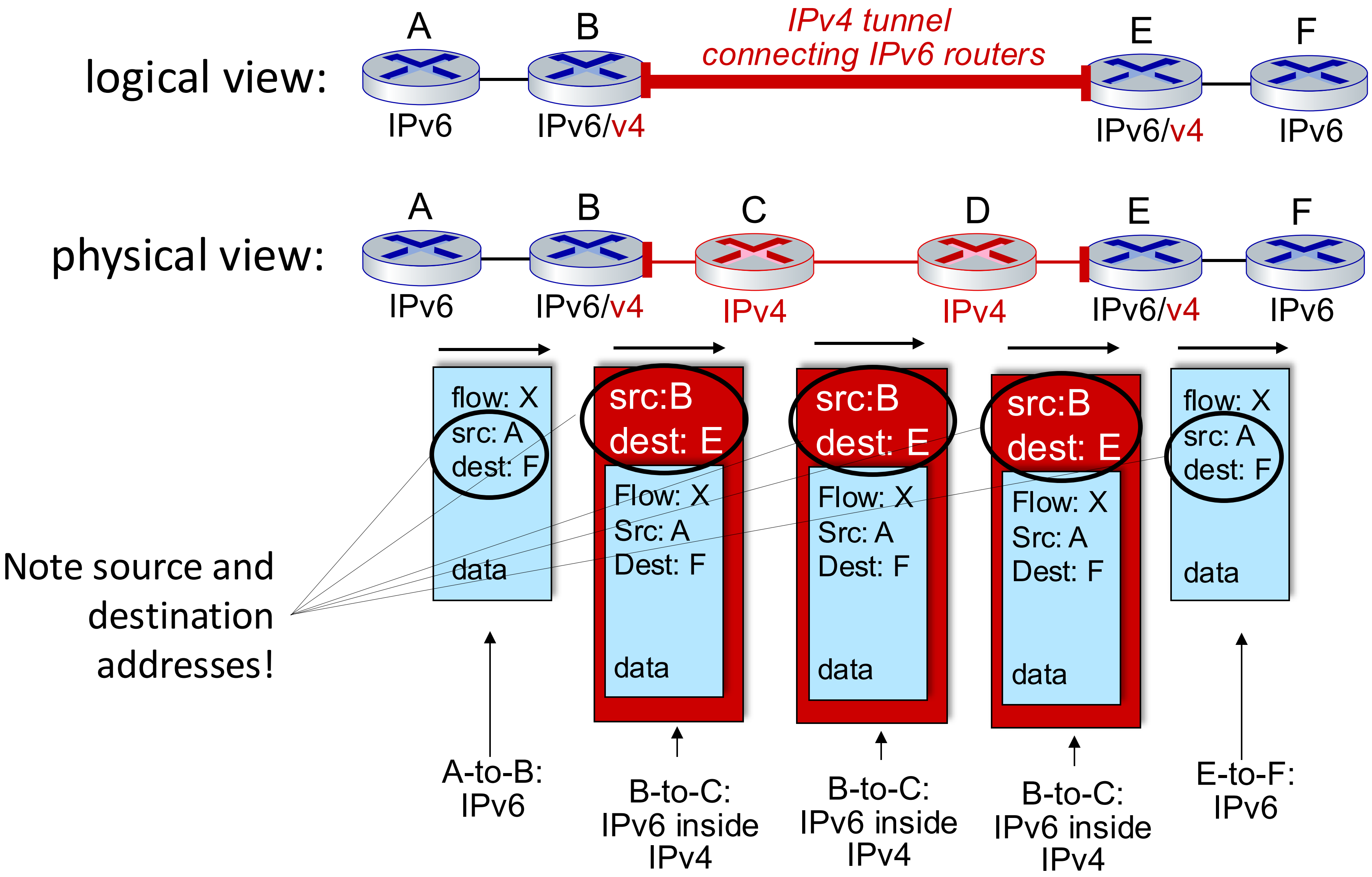
The usual: datagram as payload in link-layer frame

IPv4 tunnel connecting two IPv6 routers



tunneling: IPv6 datagram as payload in a IPv4 datagram

Tunneling


logical view:
A
B
IPv4 tunnel
connecting IPv6 routers
E
F
IPv6
IPv6/v4
IPv6/v4
IPv6
physical view:
A
B
C
D
E
F
IPv6
IPv6/v4
IPv4
IPv4
IPv6/v4
IPv6
flow: X
src: A
dest: F
data
src:B
dest: E
Flow: X
Src: A
Dest: F
data
src:B
dest: E
Flow: X
Src: A
Dest: F
data
src:B
dest: E
Flow: X
Src: A
Dest: F
data
flow: X
src: A
dest: F
data
Note source and destination addresses!
A-to-B: IPv6
B-to-C: IPv6 inside IPv4
B-to-C: IPv6 inside IPv4
B-to-C: IPv6 inside IPv4
E-to-F: IPv6

Chapter 5
Network Layer:
Control Plane

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

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*; = ∞ 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

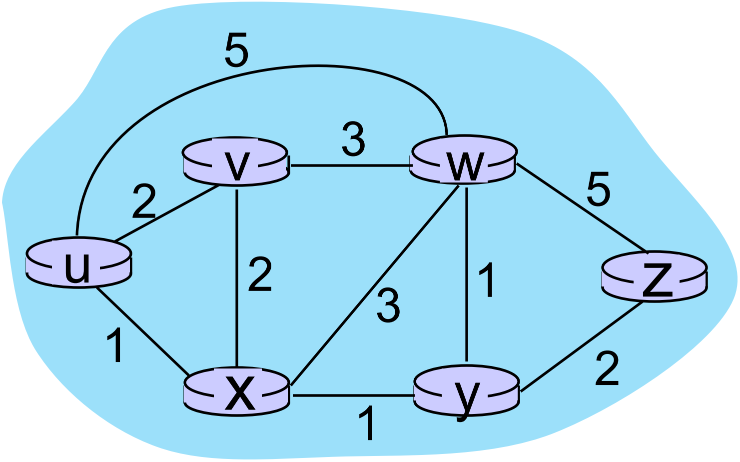
Dijkstra's link-state routing algorithm

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						

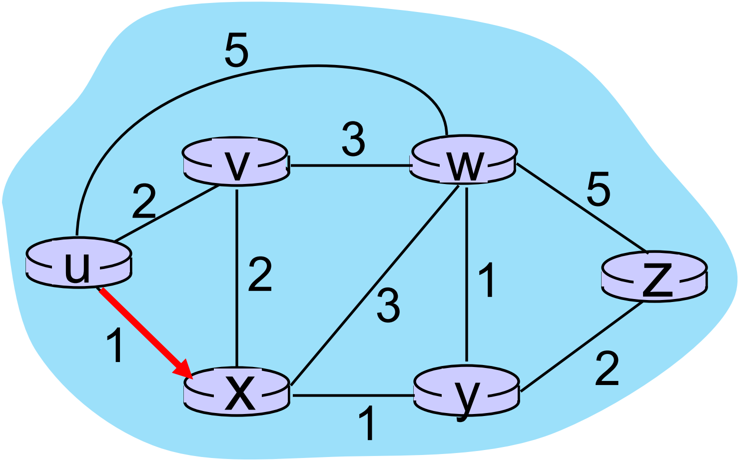


Initialization (step 0):

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						



8 *Loop*

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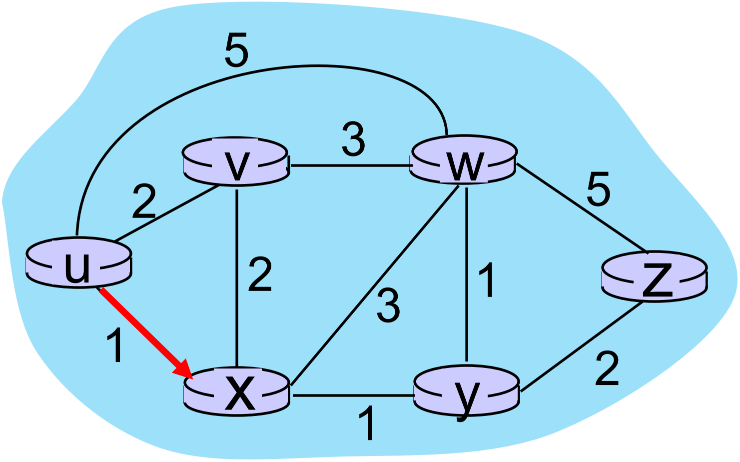
10 add *a* to *N'*

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						
3						
4						
5						



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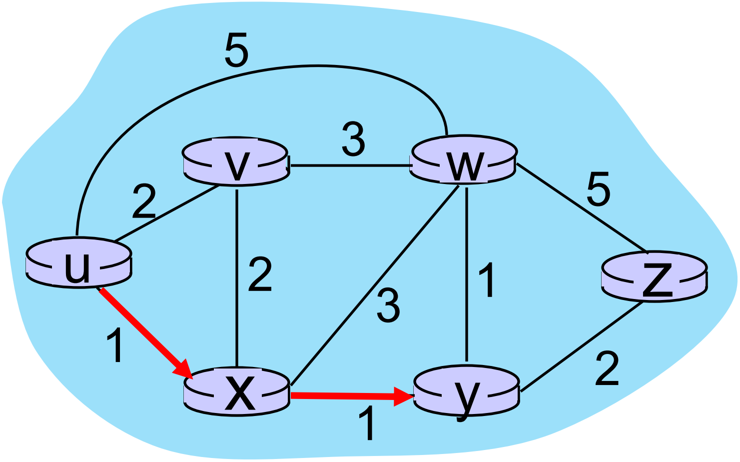
$D(b) = \min (D(b), D(a) + c_{a,b})$

$D(v) = \min (D(v), D(x) + c_{x,v}) = \min(2, 1+2) = 2$
$D(w) = \min (D(w), D(x) + c_{x,w}) = \min (5, 1+3) = 4$
$D(y) = \min (D(y), D(x) + c_{x,y}) = \min(inf,1+1) = 2$

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						



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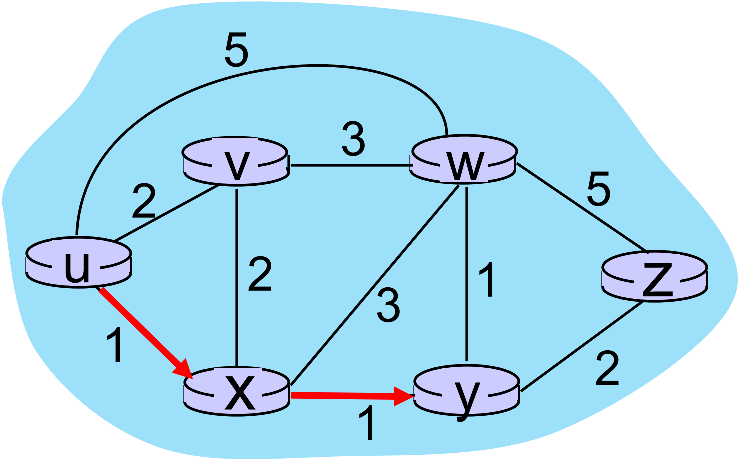
Dijkstra’s algorithm: an example

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						
4						
5						



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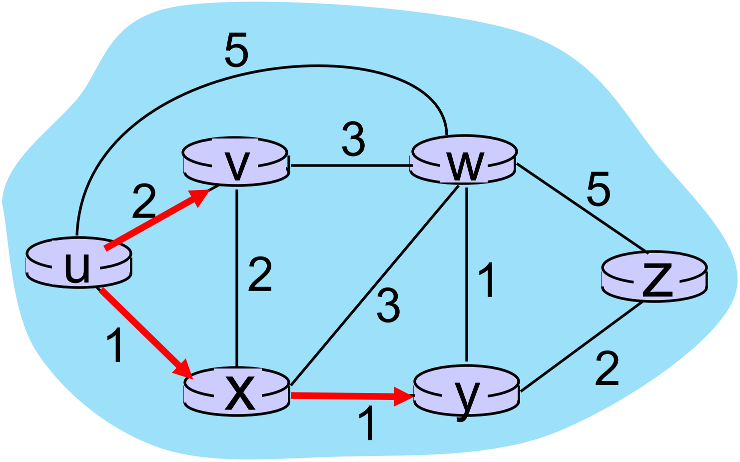
$$\mathbf{D(b) = min\ (\ D(b),\ D(a) + c_{a,b}\ )}$$

$D(w) = min\ (\ D(w),\ D(y) + c_{y,w}\ ) = min\ (4,\ 2{+}1) = 3$ 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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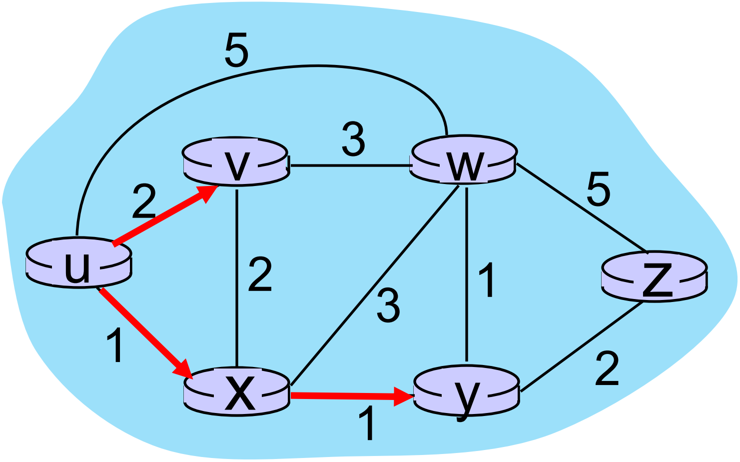
Dijkstra's algorithm: an example

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						
5						



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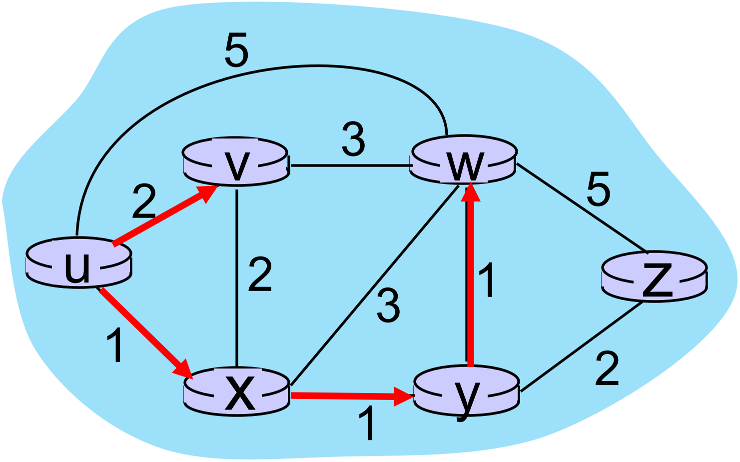
Dijkstra's algorithm: an example

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					
5						



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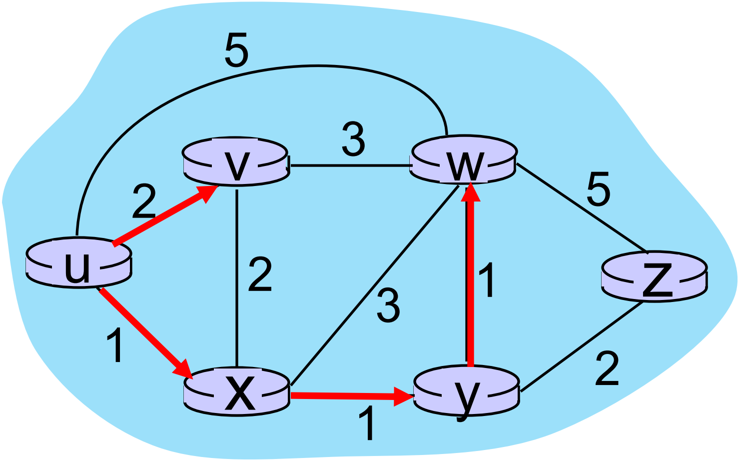
Dijkstra’s algorithm: an example

		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						



Loop

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add *a* to *N'*

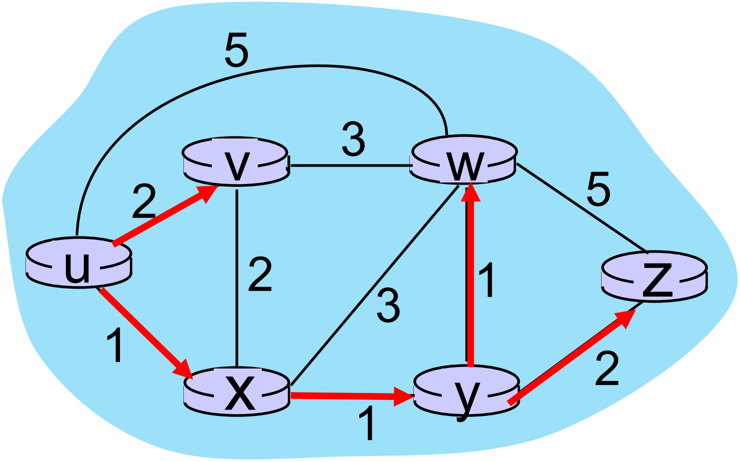
update *D(b)* for all *b* adjacent to *a* and not in *N'* :

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

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

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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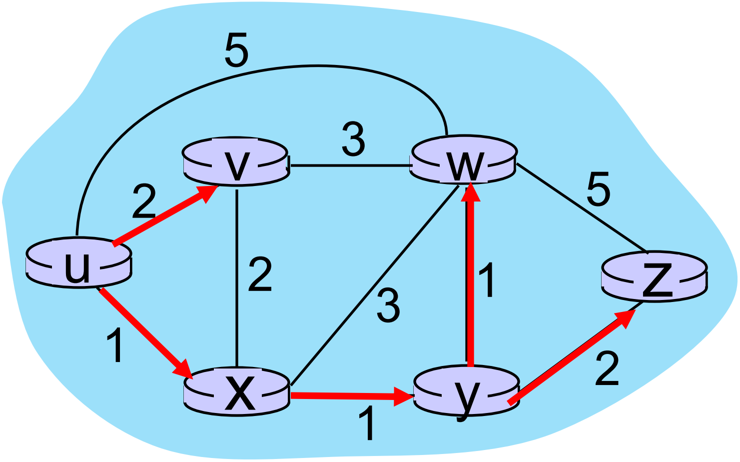
Dijkstra's algorithm: an example

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					



8 *Loop*

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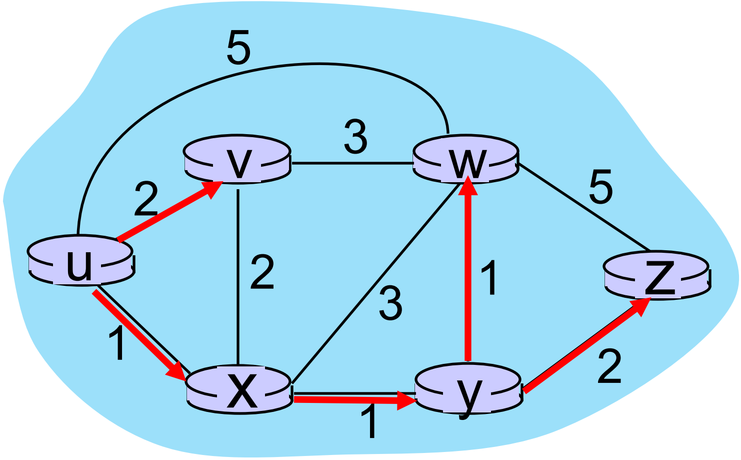
10 add *a* to *N'*

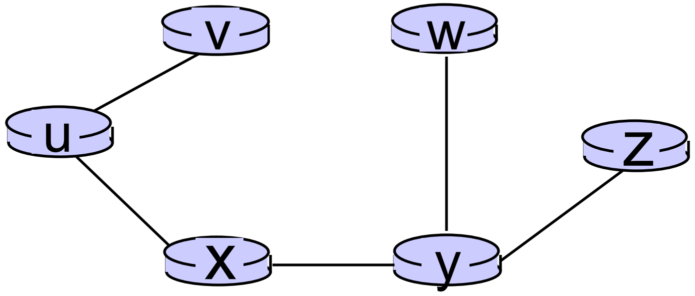
11 update *D(b)* for all *b* adjacent to *a* and not in *N'* :

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

Dijkstra's algorithm: an example



resulting least-cost-path tree from u:



resulting forwarding table in u:

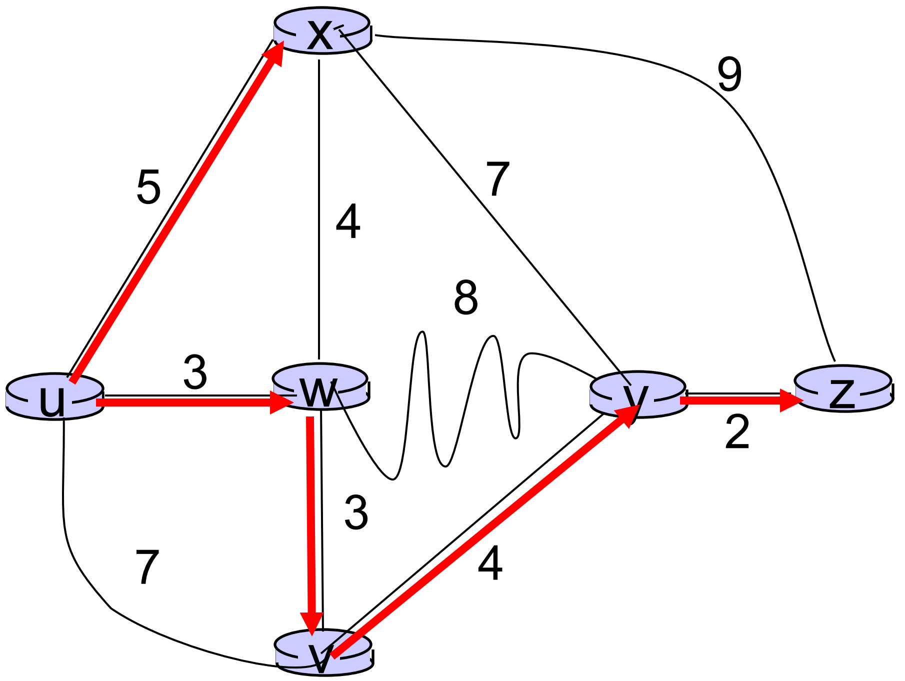
destination	outgoing link
v	(u,v)
x	(u,x)
y	(u,x)
w	(u,x)
x	(u,x)

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

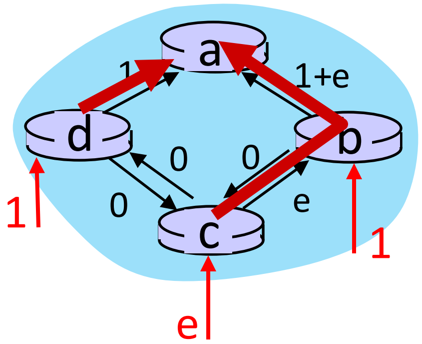
- each of *n* iteration: need to check all nodes, *w*, not in *N*
- $n(n+1)/2$ comparisons: $O(n^2)$ complexity
- more efficient implementations possible: $O(n \log n)$

message complexity:

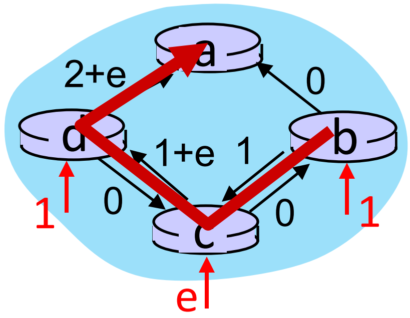
- each router must *broadcast* its link state information to other *n* routers
- efficient (and interesting!) broadcast algorithms: $O(n)$ link crossings to disseminate a broadcast message from one source
- each router's message crosses $O(n)$ links: overall message complexity: $O(n^2)$

Dijkstra's algorithm: oscillations possible

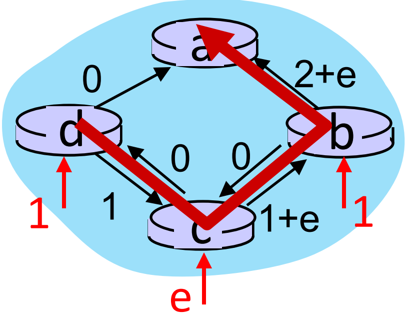
- when link costs depend on traffic volume, route oscillations possible
- sample scenario:
  - routing to destination a, traffic entering at d, c, e with rates 1, e (<1), 1
  - link costs are directional, and volume-dependent



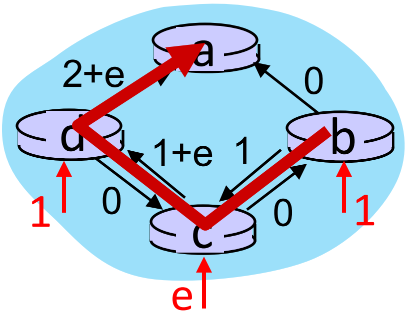
initially



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

- 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

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:

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

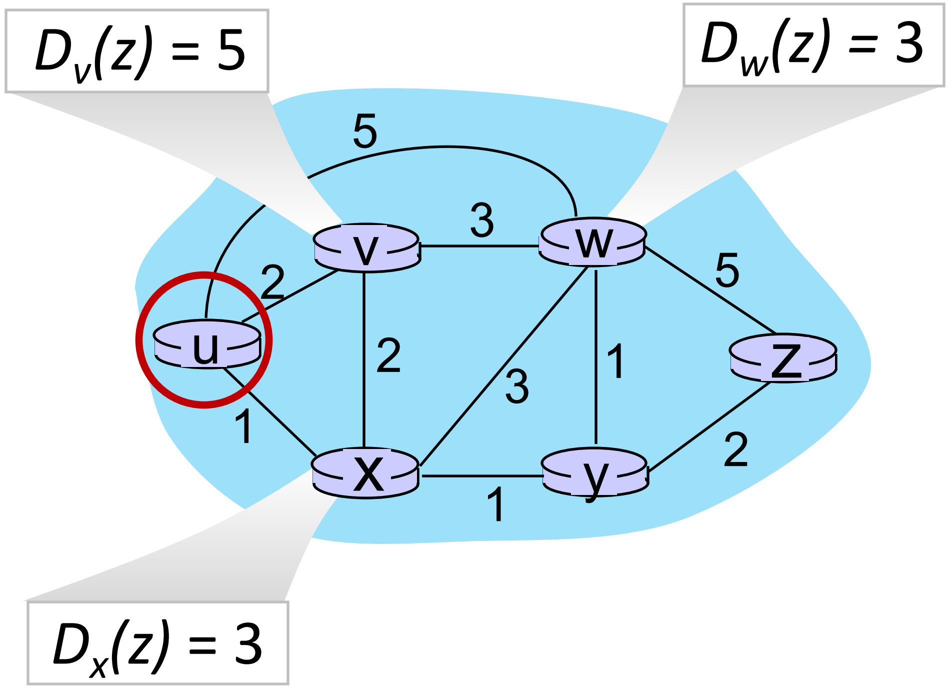
v's estimated least-cost-path cost to y

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:

$$D_u(z) = \min \{ c_{u,v} + D_v(z), \; c_{u,x} + D_x(z), \; c_{u,w} + D_w(z) \}$$

$$= \min \{2 + 5, \; 1 + 3, \; 5 + 3\} = 4$$

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:

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

- 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

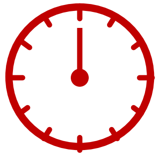
iterative, asynchronous: each local iteration caused by:

- 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!

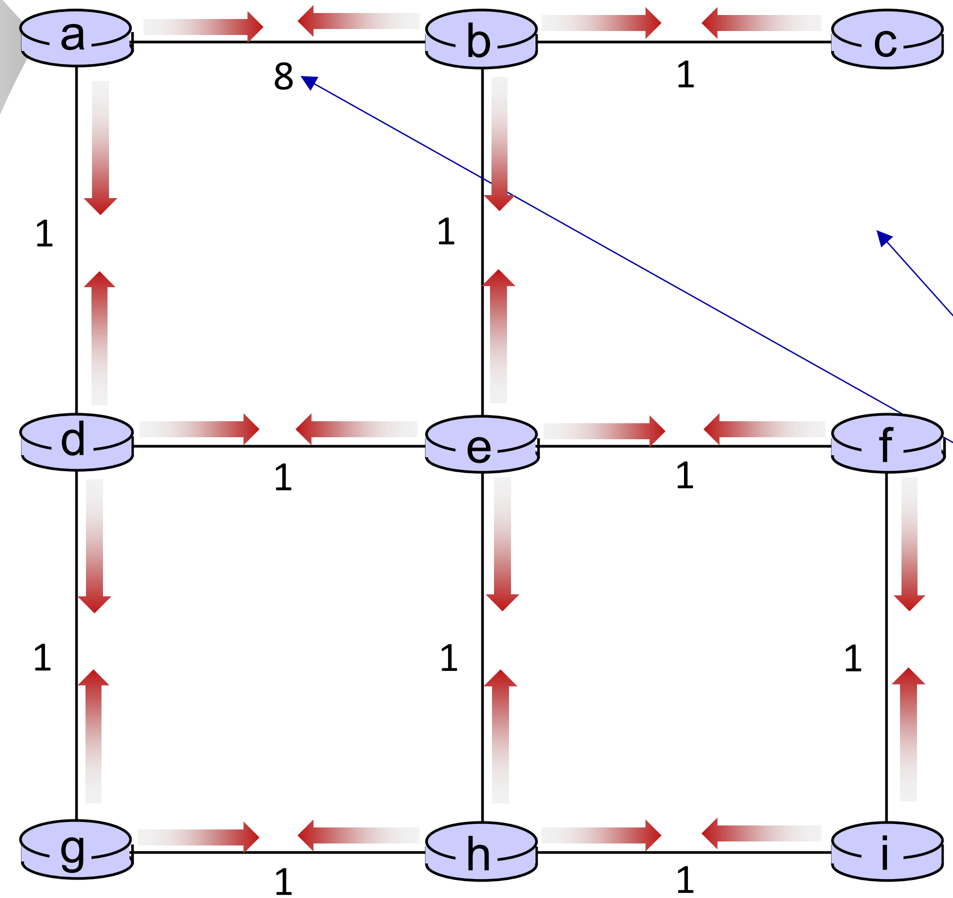
Distance vector: example



t=0

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

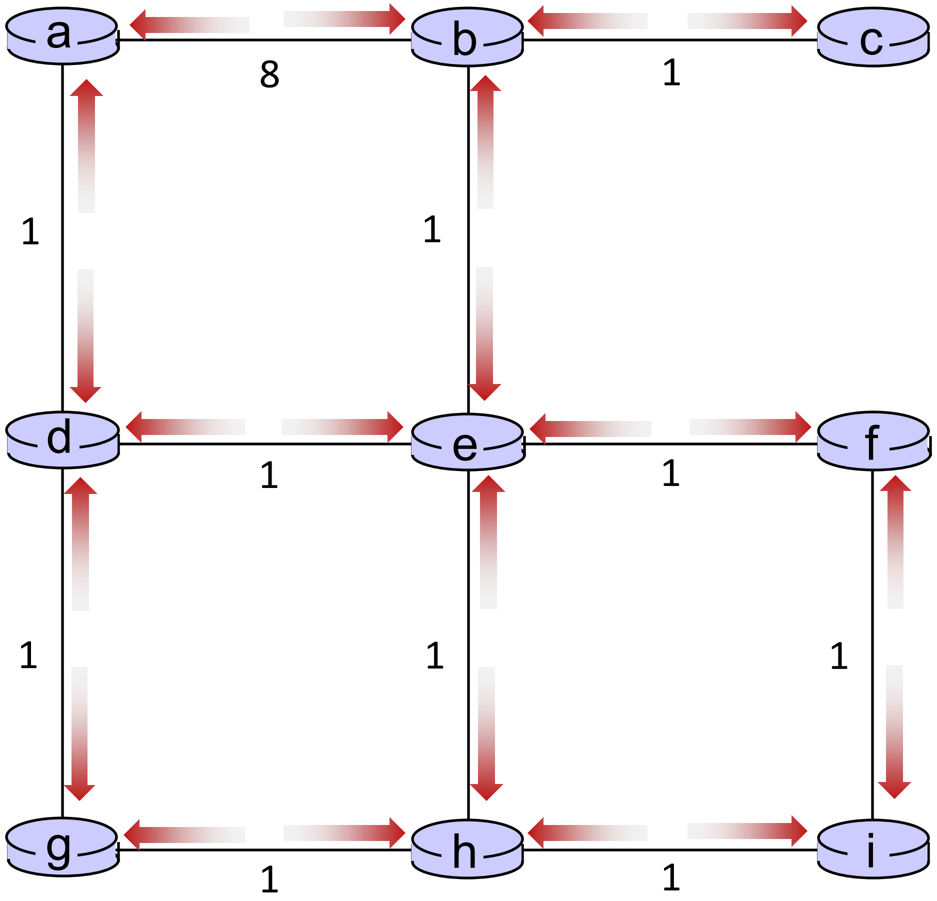
DV in a:
$D_a(a)=0$
$D_a(b) = 8$
$D_a(c) = \infty$
$D_a(d) = 1$
$D_a(e) = \infty$
$D_a(f) = \infty$
$D_a(g) = \infty$
$D_a(h) = \infty$
$D_a(i) = \infty$



A few asymmetries:
- missing link
- larger cost

Distance vector example: iteration

t=1



All nodes:

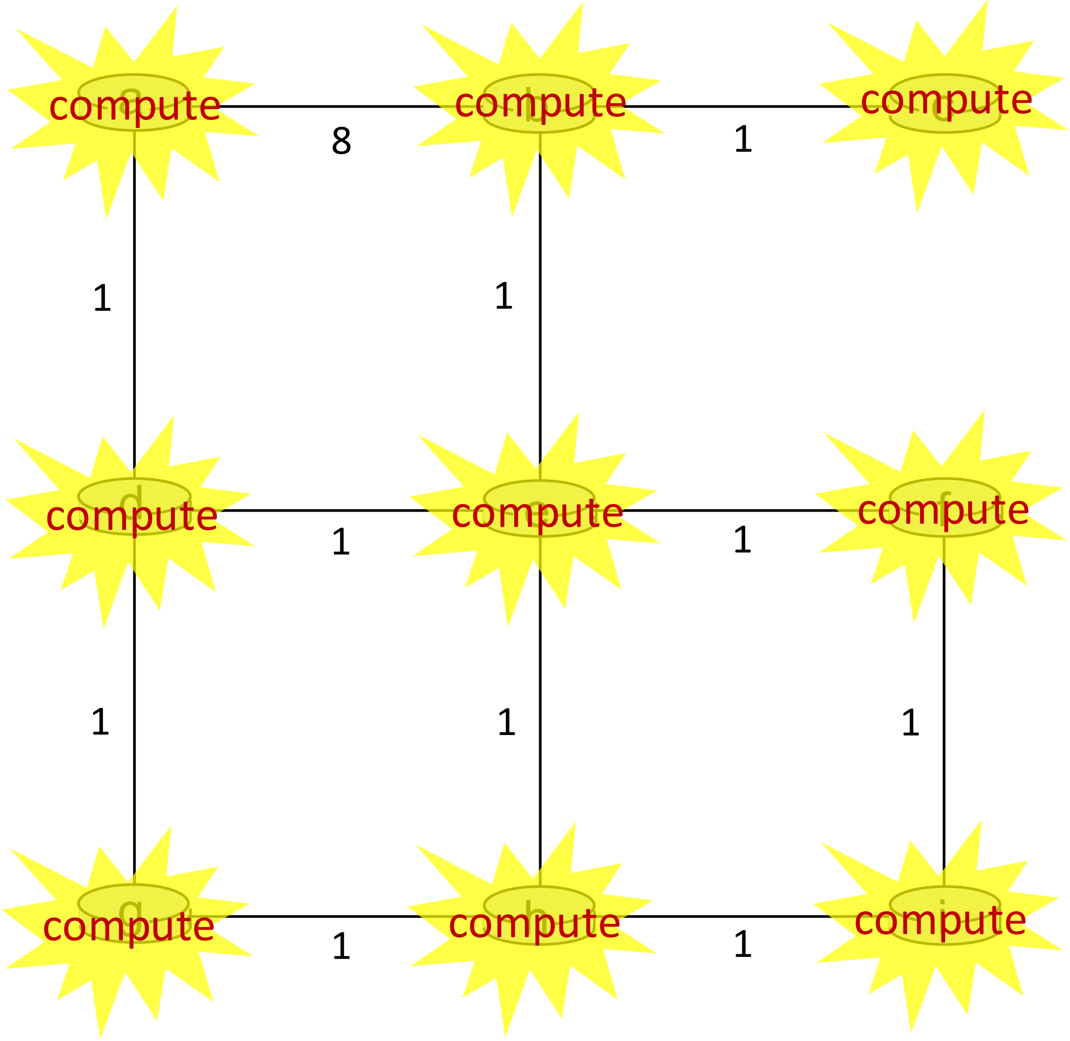
- receive distance vectors from neighbors
- compute their new local distance vector
- send their new local distance vector to neighbors

Distance vector example: iteration

t=1

All nodes:

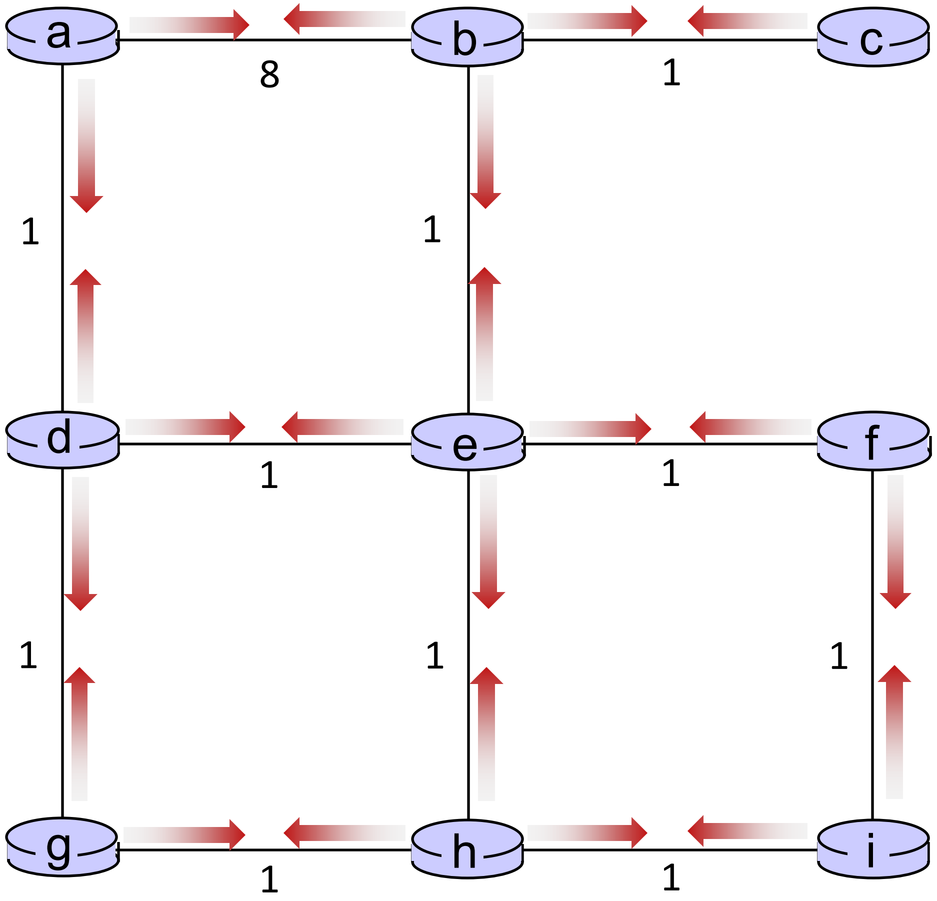
- receive distance vectors from neighbors
- compute their new local distance vector
- send their new local distance vector to neighbors



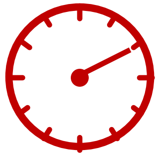
Distance vector example: iteration

t=1

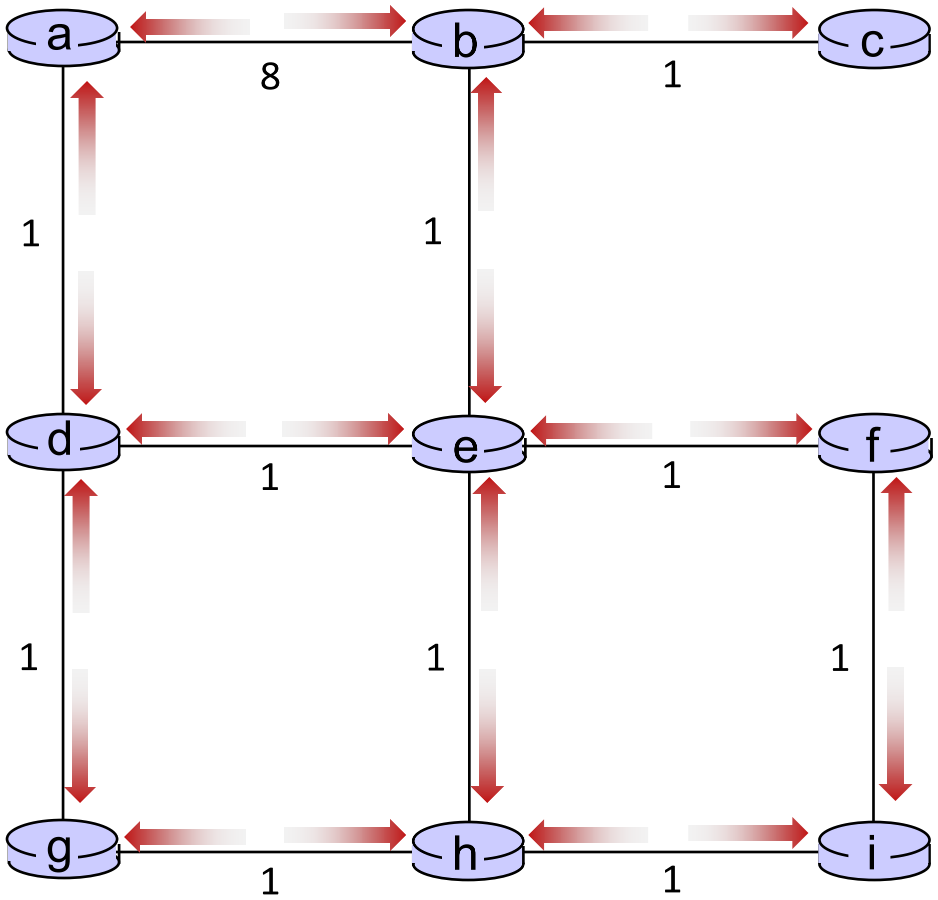
All nodes:
- receive distance vectors from neighbors
- compute their new local distance vector
- send their new local distance vector to neighbors



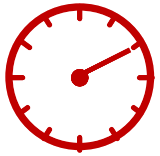
Distance vector example: iteration



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



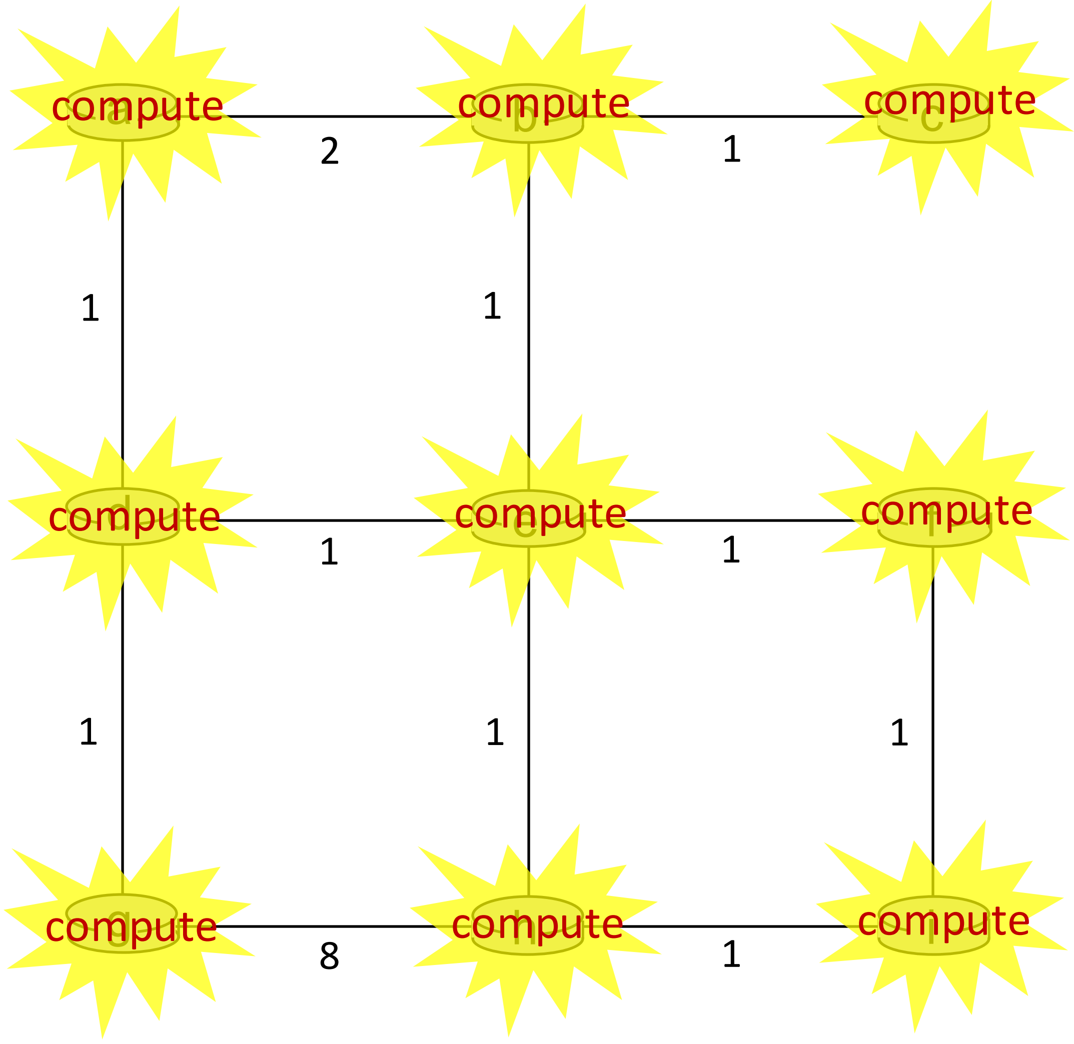
Distance vector example: iteration



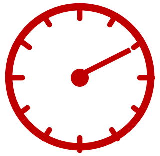
t=2

All nodes:

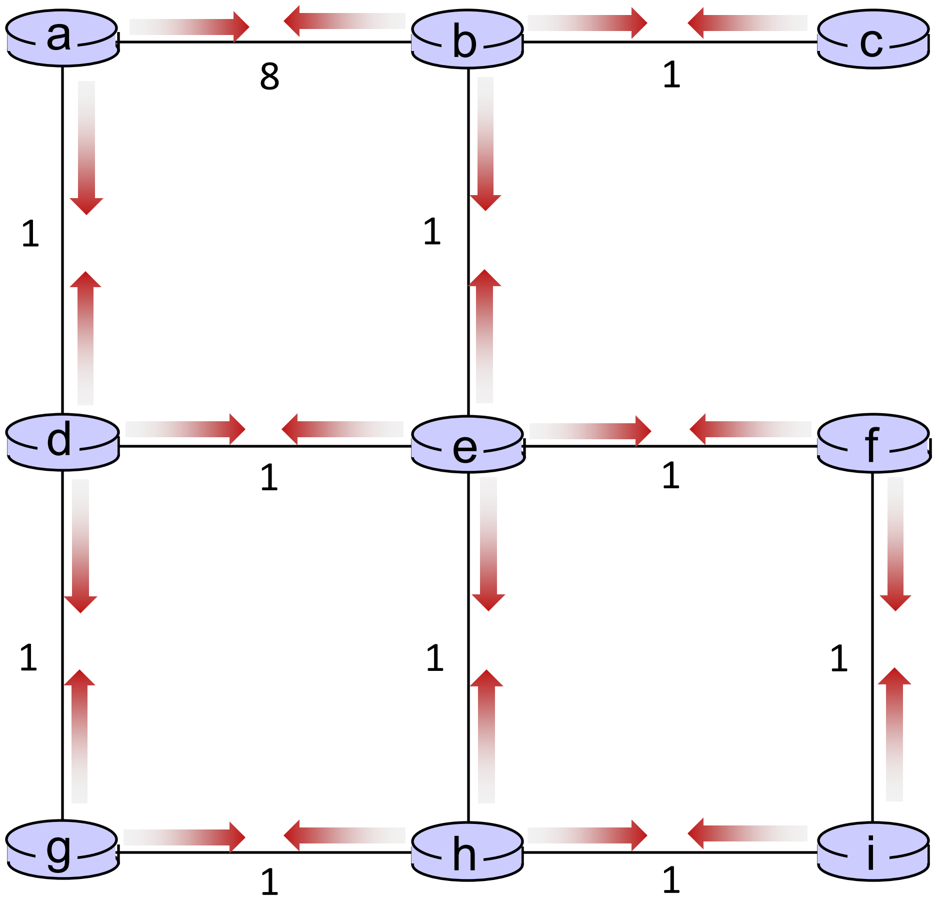
- receive distance vectors from neighbors
- compute their new local distance vector
- send their new local distance vector to neighbors



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

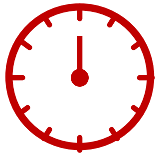


Distance vector example: iteration

…. 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:

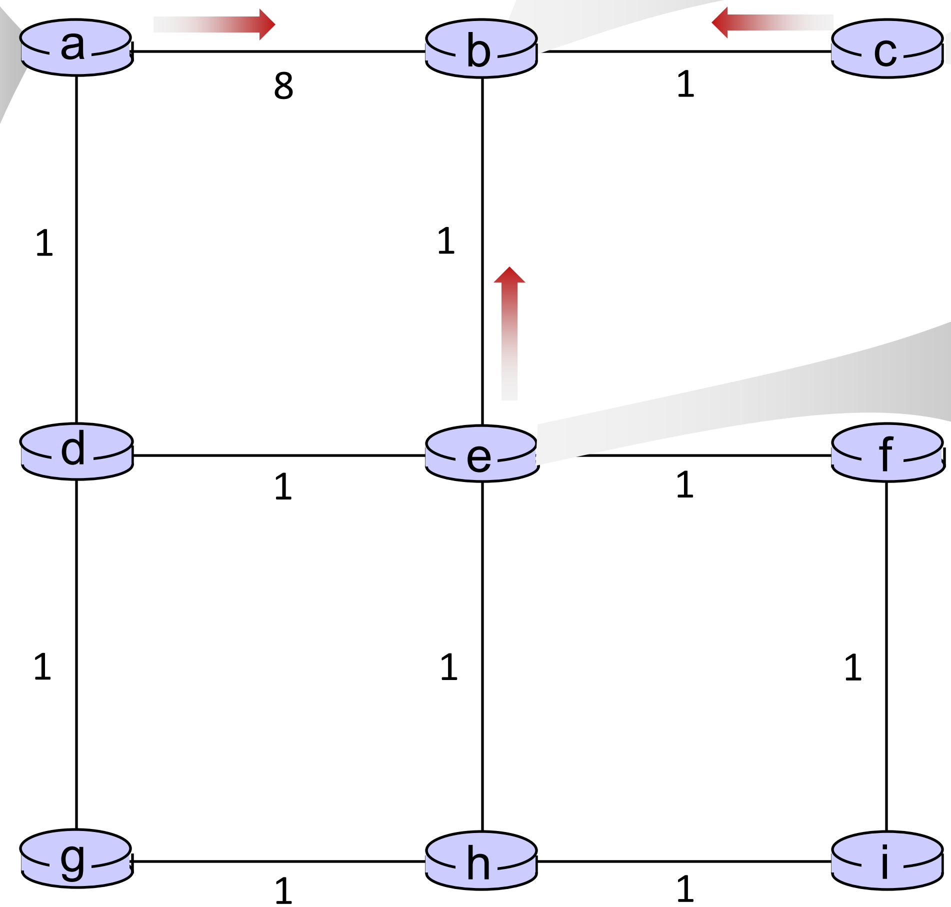
$D_b(a) = 8$	$D_b(f) = \infty$
$D_b(c) = 1$	$D_b(g) = \infty$
$D_b(d) = \infty$	$D_b(h) = \infty$
$D_b(e) = 1$	$D_b(i) = \infty$

DV in c:

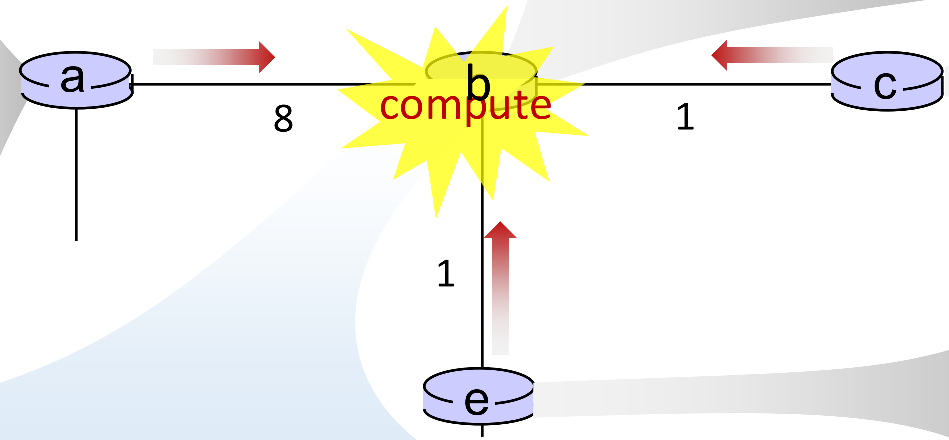
$D_c(a) = \infty$
$D_c(b) = 1$
$D_c(c) = 0$
$D_c(d) = \infty$
$D_c(e) = \infty$
$D_c(f) = \infty$
$D_c(g) = \infty$
$D_c(h) = \infty$
$D_c(i) = \infty$

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 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:

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

t=1

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

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

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

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

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

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

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

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

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

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:

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:

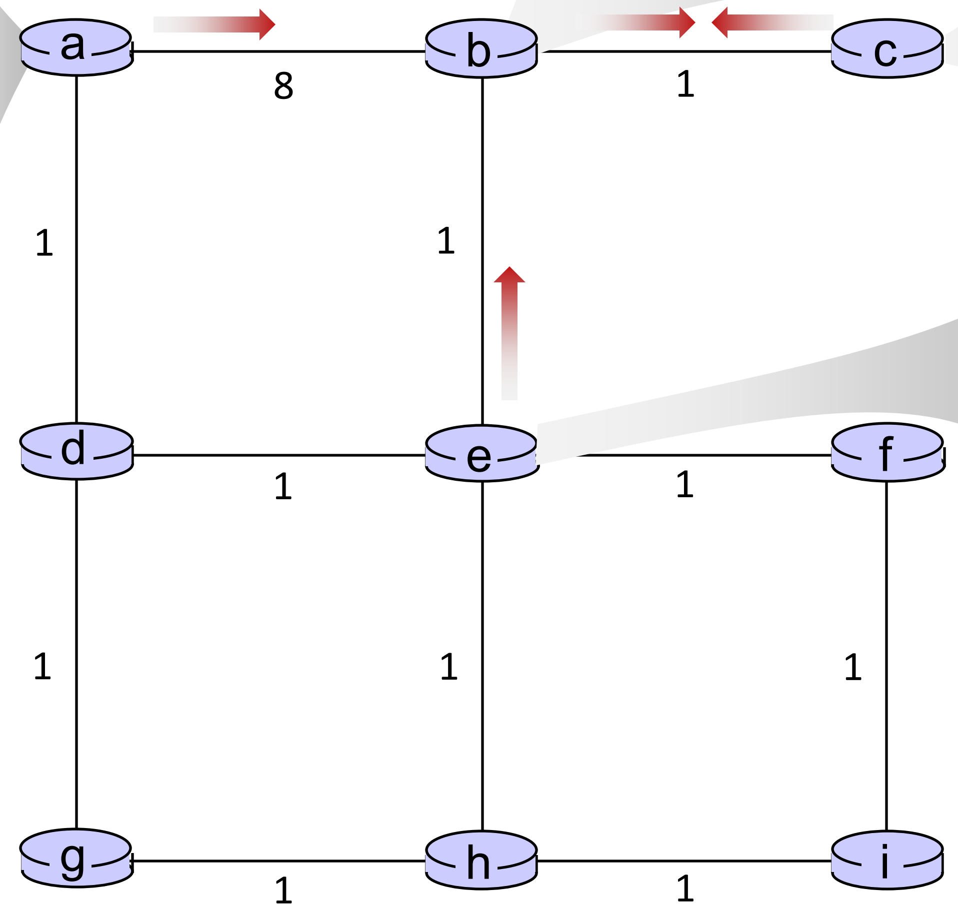
$D_b(a) = 8$	$D_b(f) = \infty$
$D_b(c) = 1$	$D_b(g) = \infty$
$D_b(d) = \infty$	$D_b(h) = \infty$
$D_b(e) = 1$	$D_b(i) = \infty$

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:

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



t=1

- c receives DVs from b

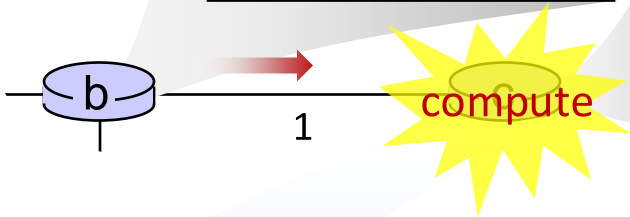
Distance vector example:

DV in b:

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

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:

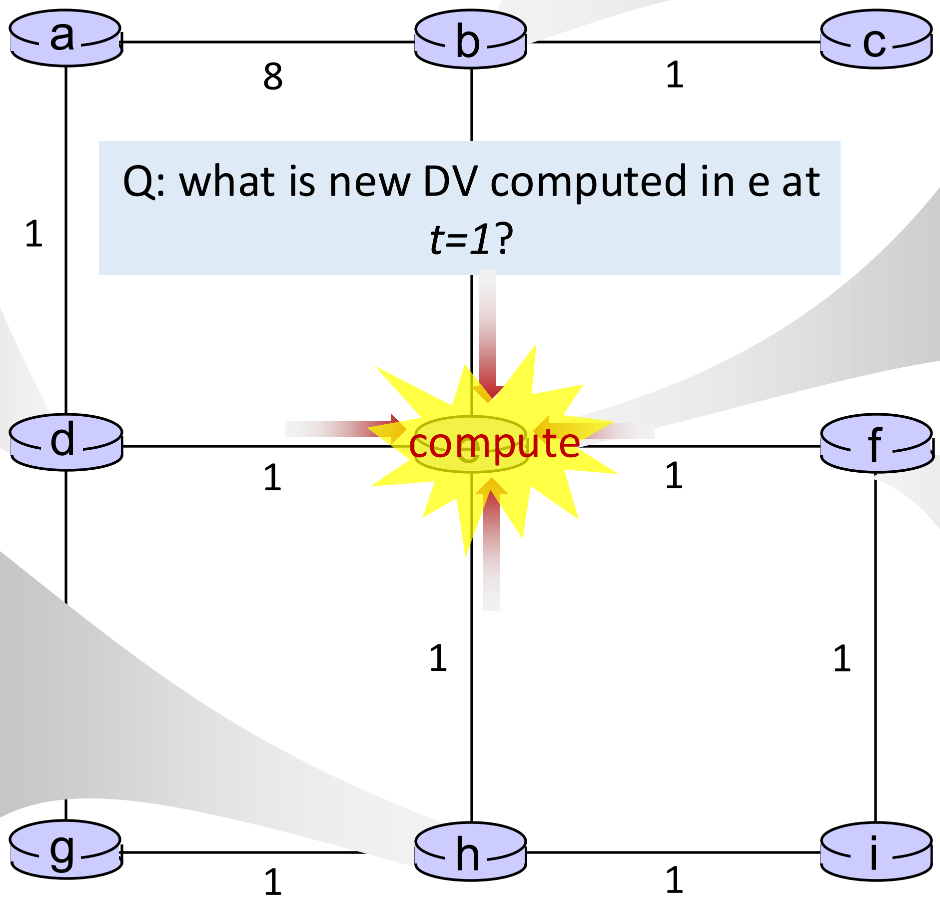
$D_c(a) = 1$
$D_c(b) = \infty$
$D_c(c) = \infty$
$D_c(d) = 0$
$D_c(e) = 1$
$D_c(f) = \infty$
$D_c(g) = 1$
$D_c(h) = \infty$
$D_c(i) = \infty$

DV in 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:

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



Q: what is new DV computed in e at *t=1*?

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 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:

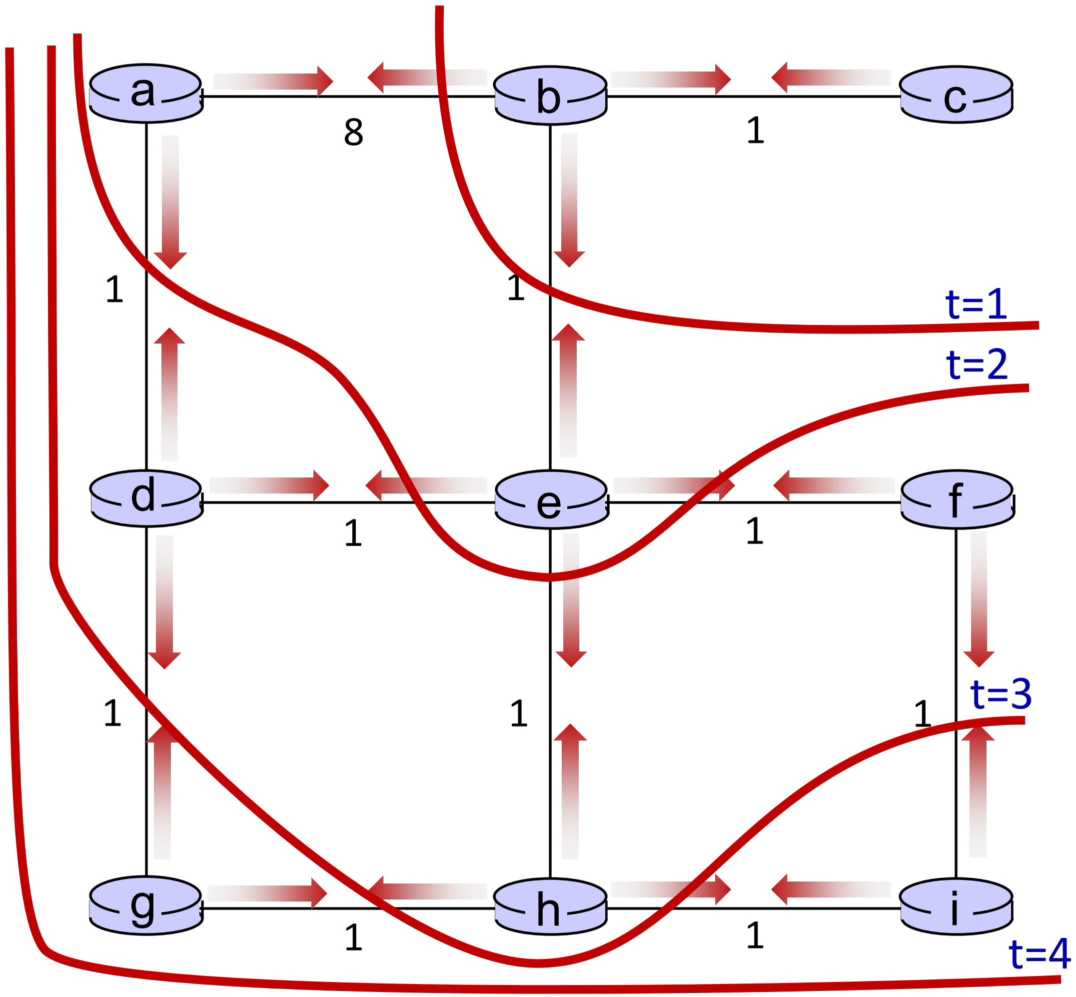
t=0 c's state at t=0 is at c only

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

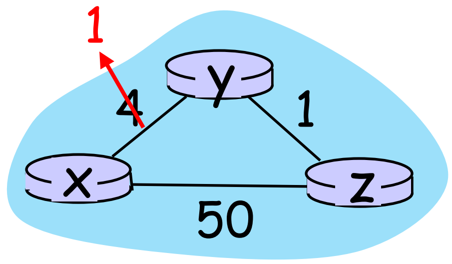
t=4 c's state at t=0 may influence distance vector computations up to **4** hops away, i.e., at g, i



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



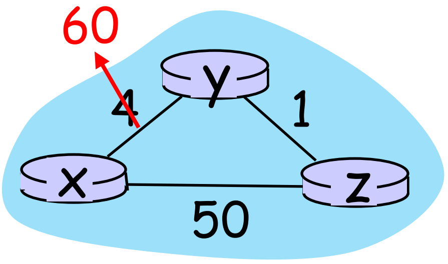
"good news travels fast"

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*.

  …

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

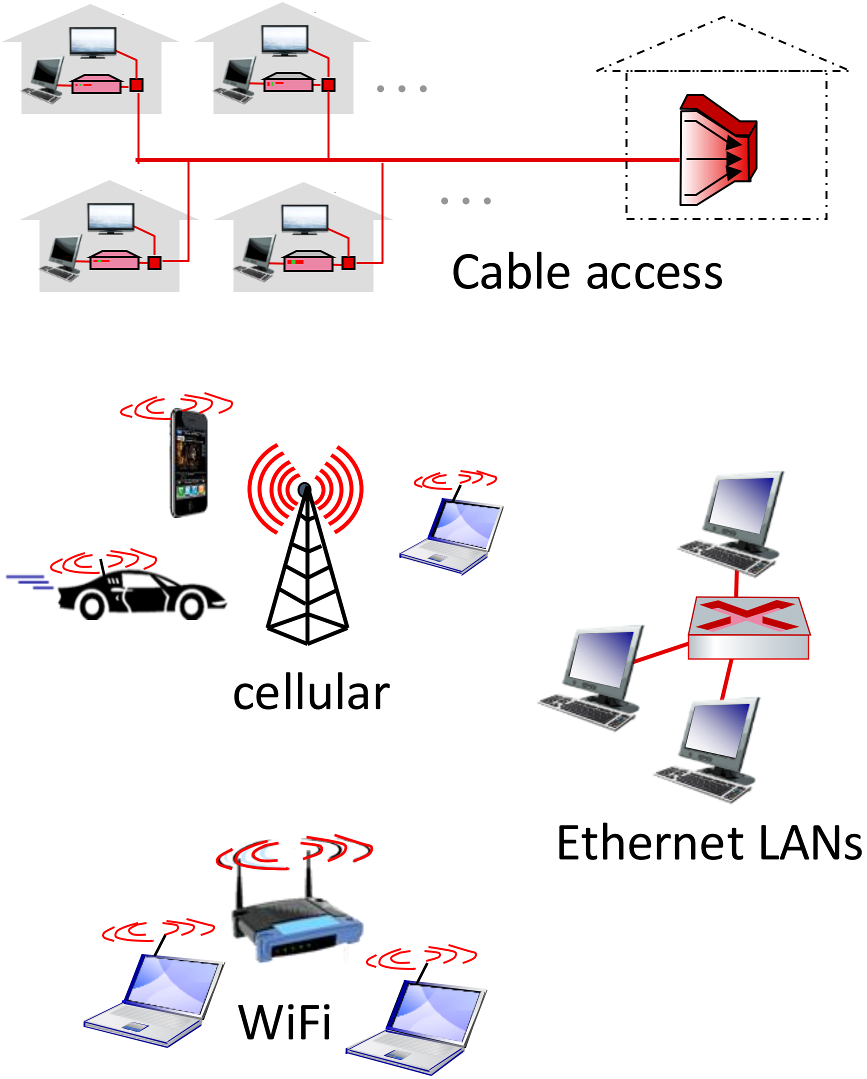
DV:

- 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

Chapter 6
The Link Layer and LANs

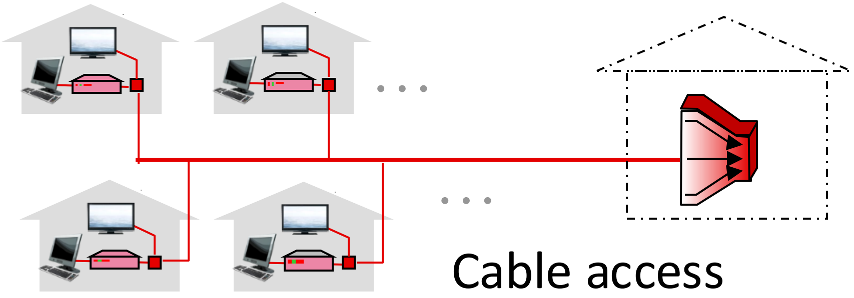
Link layer: services

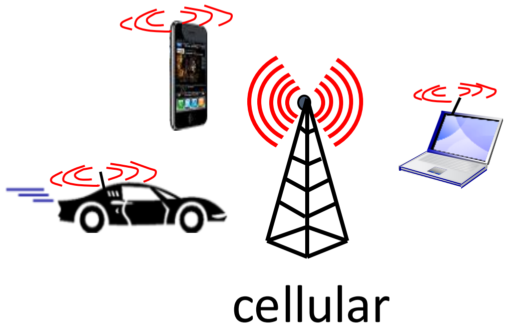
- **framing, link access:**
  - encapsulate datagram into frame, adding header, trailer
  - channel access if shared medium
  - "MAC" addresses in frame headers identify source, destination (different from IP address!)
- **reliable delivery between adjacent nodes**
  - we already know how to do this!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

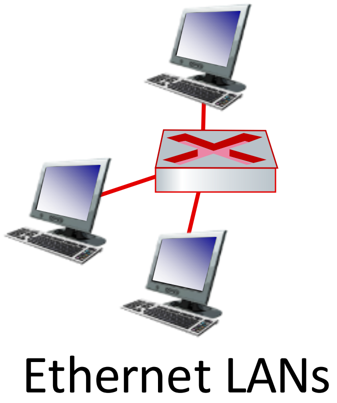


Link layer: services (more)

- **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







Link layer, LANs: roadmap

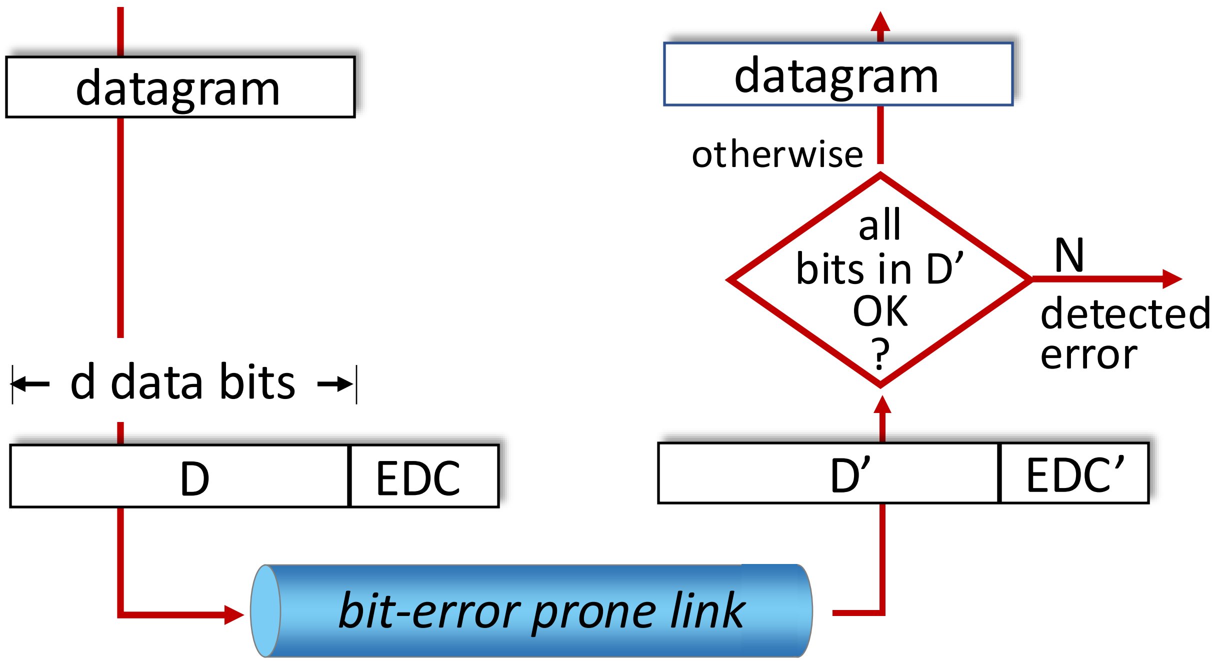
- introduction
- error detection, correction
- multiple access protocols
- LANs
  - addressing, ARP
  - Ethernet
  - switches
  - VLANs
- link virtualization: MPLS
- data center networking

- 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

Internet checksum (review, see section 3.3)

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

sender:

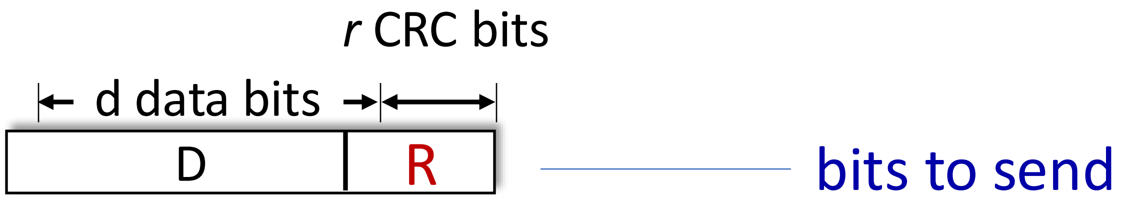
- treat contents of UDP segment (including UDP header fields and IP addresses) as sequence of 16-bit integers
- checksum: addition (one's complement sum) of segment content
- checksum value put into UDP checksum field

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)



$\langle D,R \rangle = D*2^r \text{ XOR } R$ — 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

G

1 0 1 0 1 1

1 0 0 1) 1 0 1 1 1 0 0 0 0

1 0 0 1

1 0 1

0 0 0

1 0 1 0

1 0 0 1

1 1 0

0 0 0

1 1 0 0

1 0 0 1

1 0 1 0

1 0 0 1

0 1 1

R

$D*2^r$ (here, r=3)

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

Multiple access links, protocols

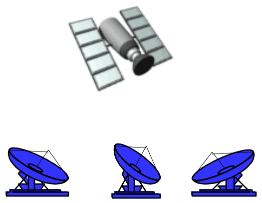
two types of "links":

- 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 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

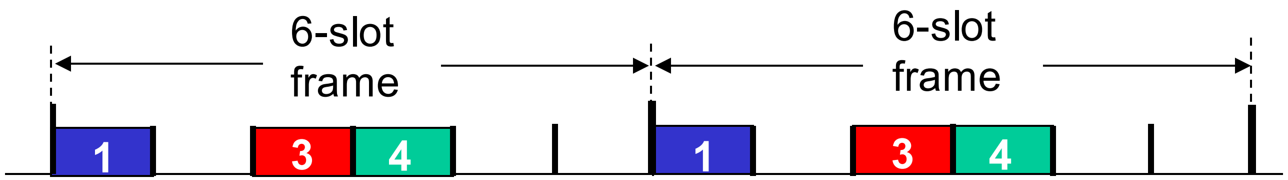
three broad classes:

- 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

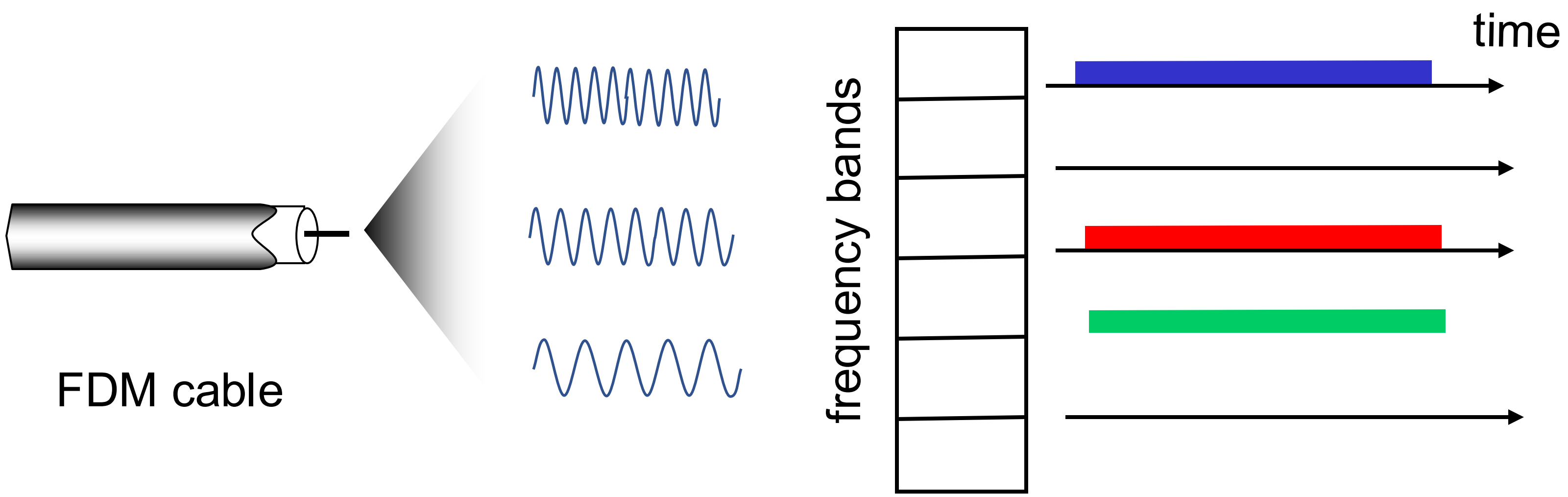
- access to channel in "rounds"
- each station gets fixed length slot (length = packet transmission time) in each round
- unused slots go idle
- example: 6-station LAN, 1,3,4 have packets to send, slots 2,5,6 idle



Channel partitioning MAC protocols: FDMA

FDMA: frequency division multiple access

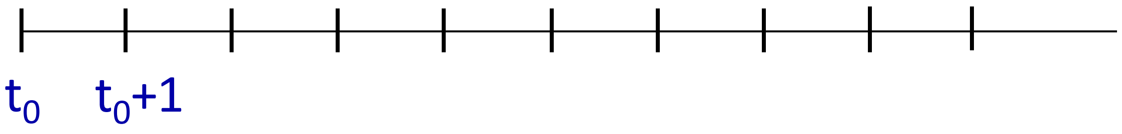
- channel spectrum divided into frequency bands
- each station assigned fixed frequency band
- unused transmission time in frequency bands go idle
- example: 6-station LAN, 1,3,4 have packet to send, frequency bands 2,5,6 idle



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



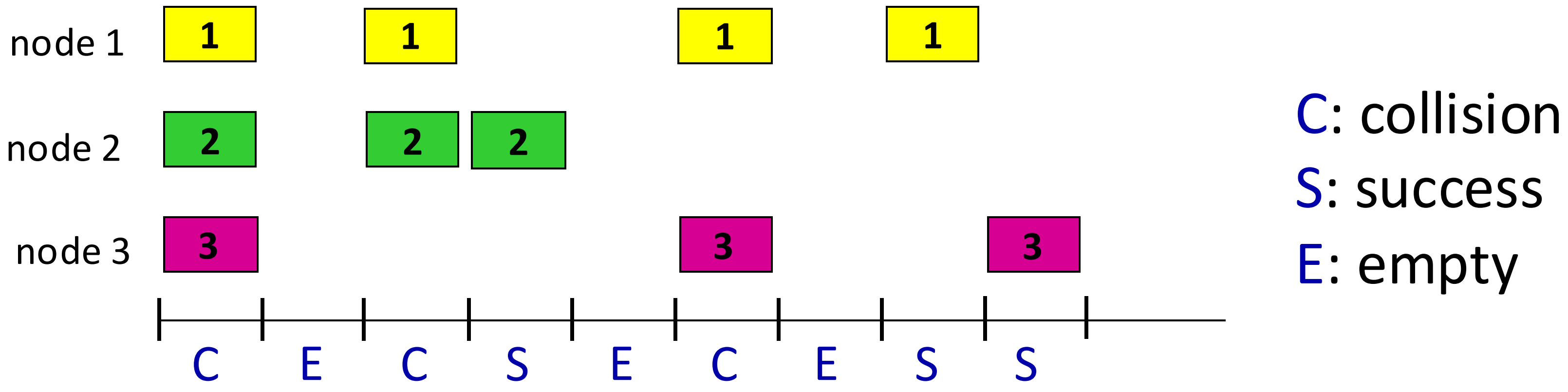
assumptions:

- 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

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:

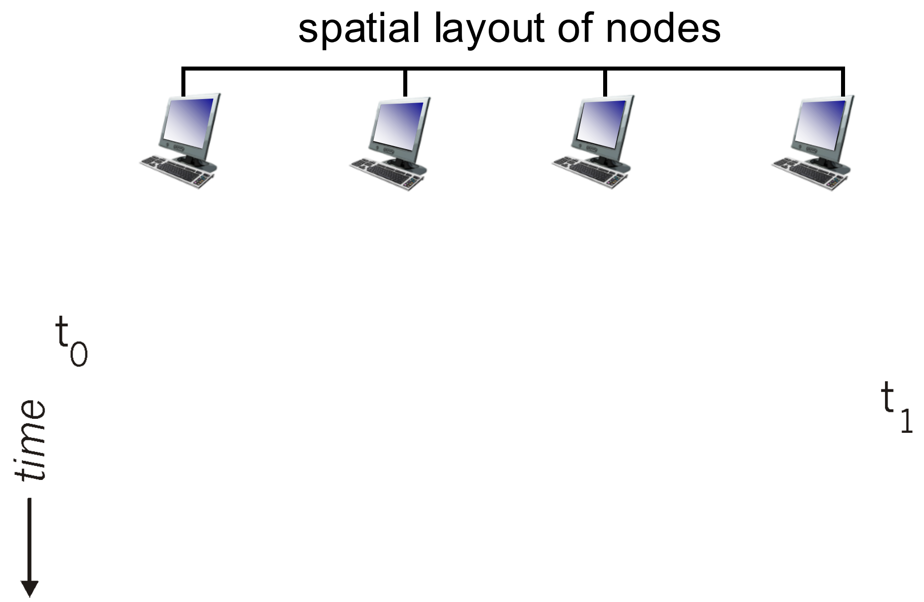
- if channel sensed idle: transmit entire frame
- if channel sensed busy: defer transmission

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

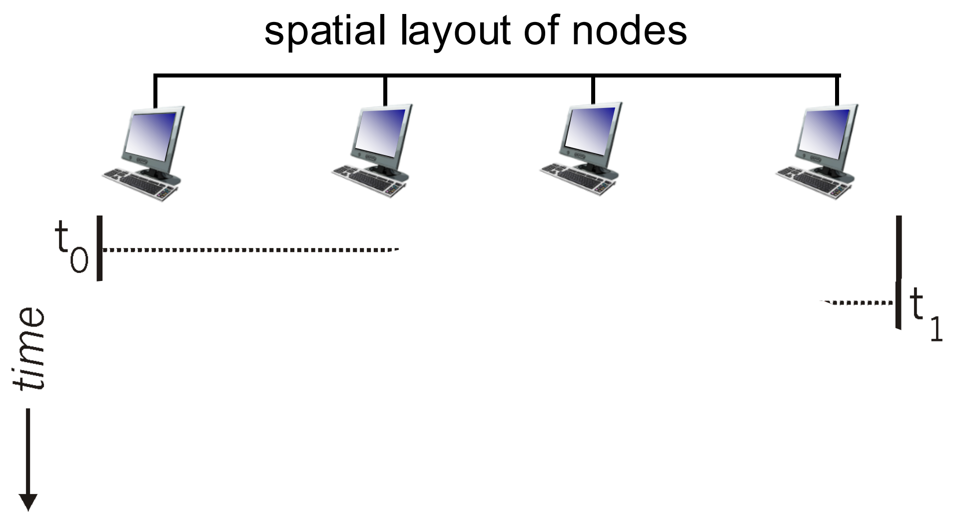
CSMA: collisions

- 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 mth 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

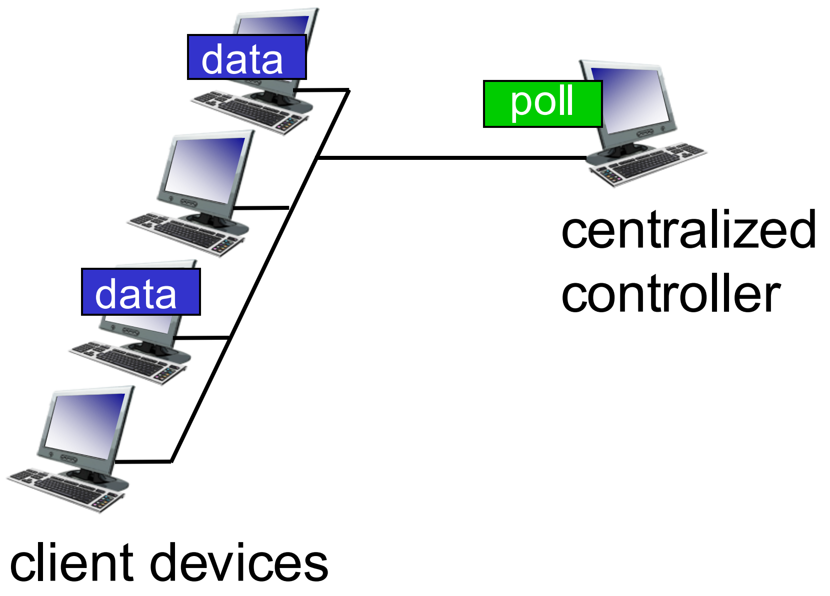
“taking turns” protocols

- look for best of both worlds!

“Taking turns” MAC protocols

polling:

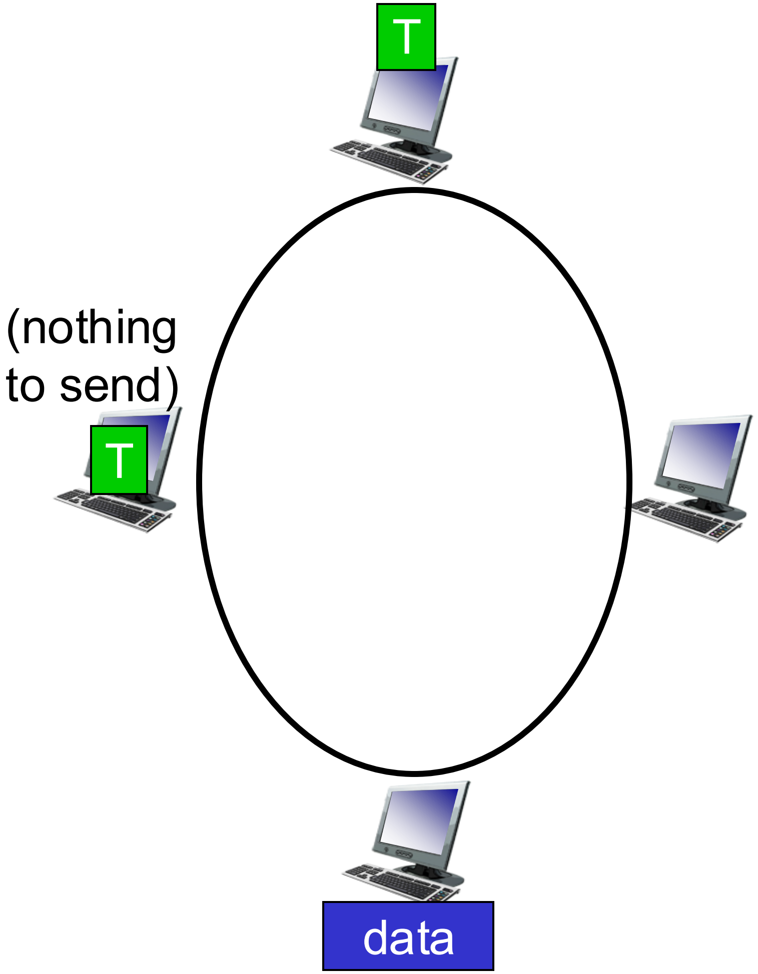
- centralized controller “invites” other nodes to transmit in turn
- typically used with “dumb” devices
- concerns:
  - polling overhead
  - latency
  - single point of failure (master)
- Bluetooth uses polling



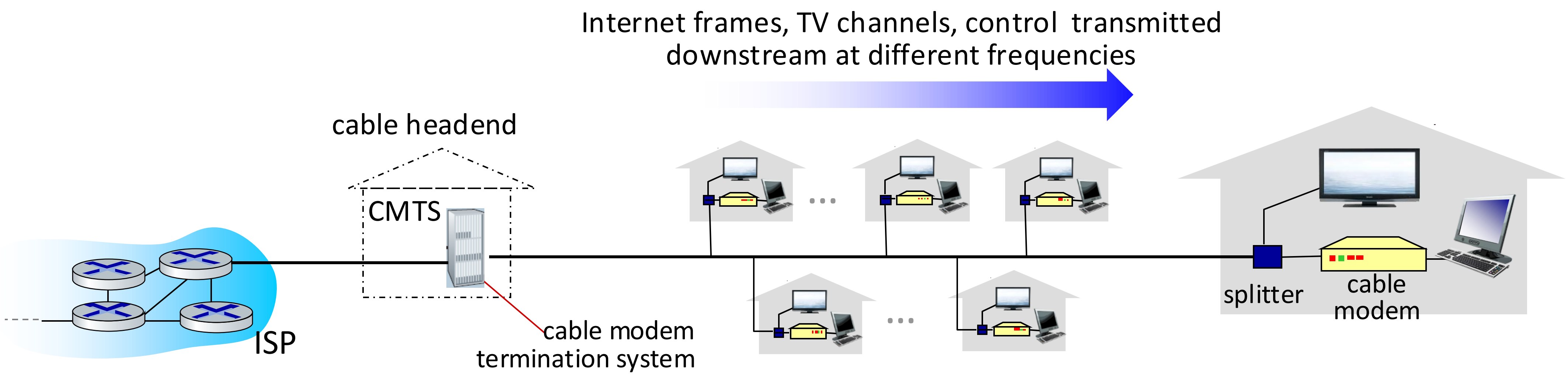
"Taking turns" MAC protocols

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)

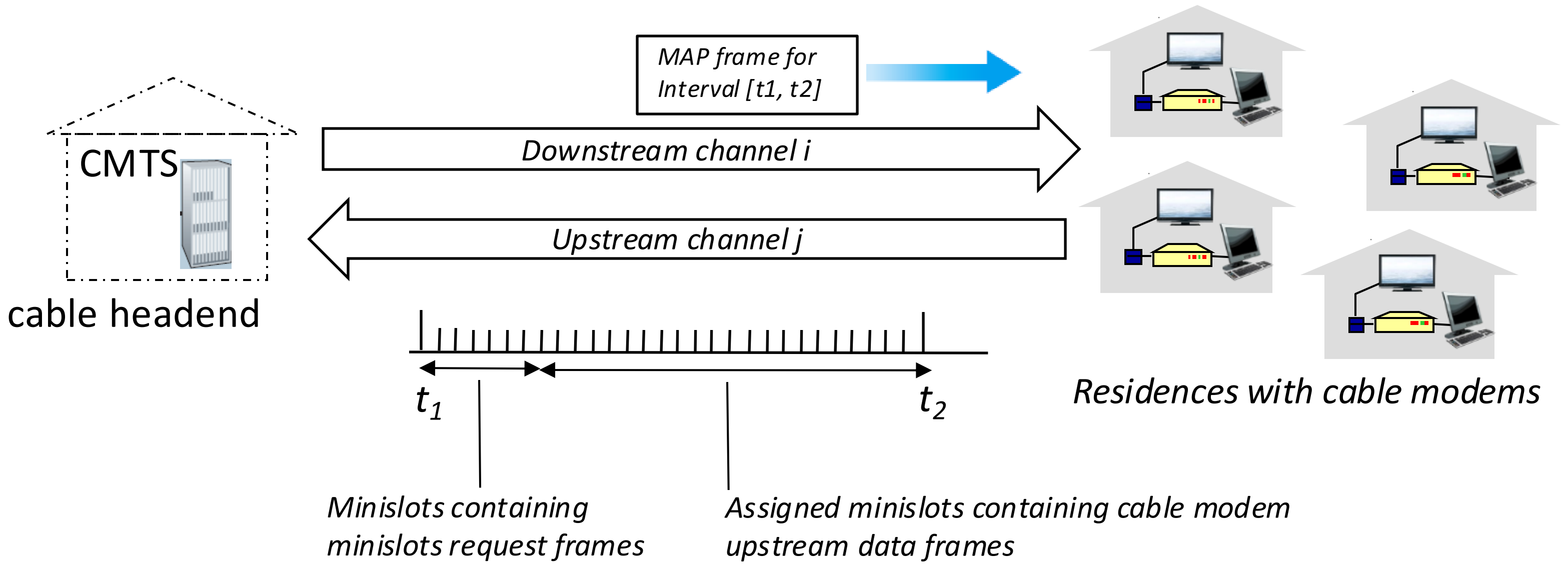


Cable access network: FDM, TDM *and* random access!



- 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
- data center networking

- a day in the life of a web request

MAC addresses

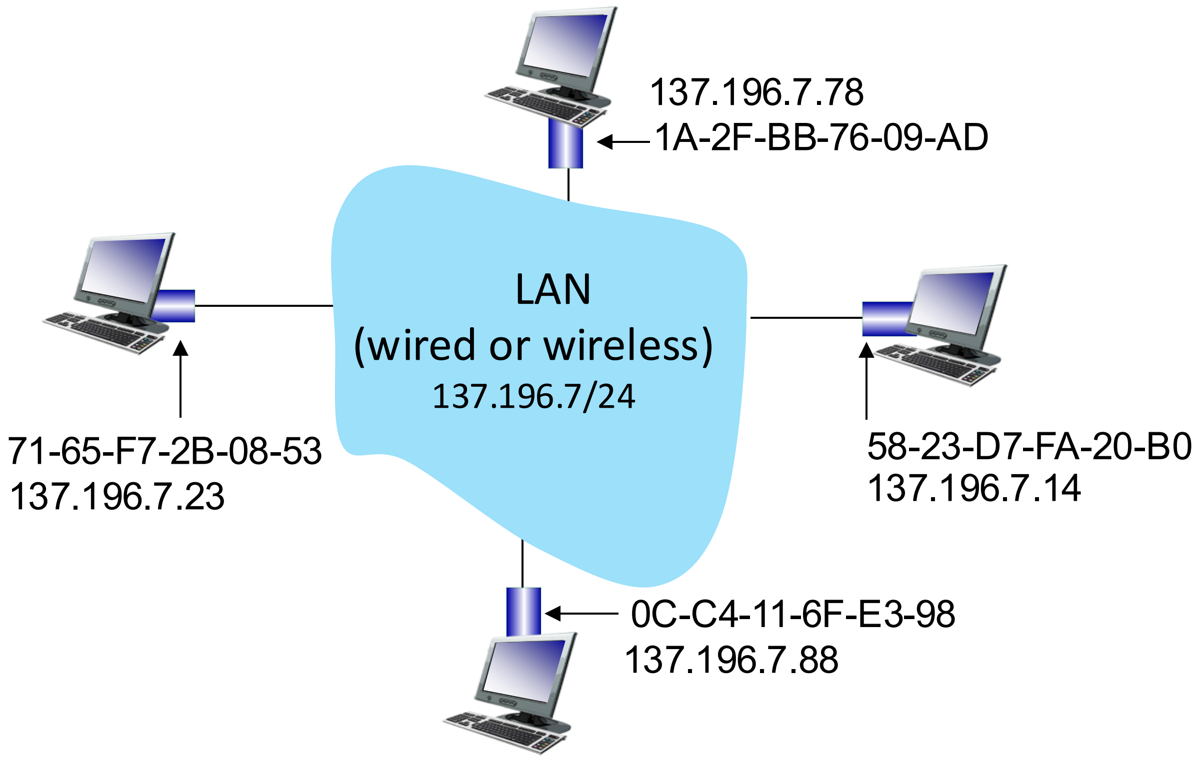
- 32-bit IP address:
  - *network-layer* address for interface
  - used for layer 3 (network layer) forwarding
  - e.g.: 128.119.40.136
- MAC (or LAN or physical or Ethernet) address:
  - function: used "locally" to get frame from one interface to another physically-connected interface (same subnet, 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)

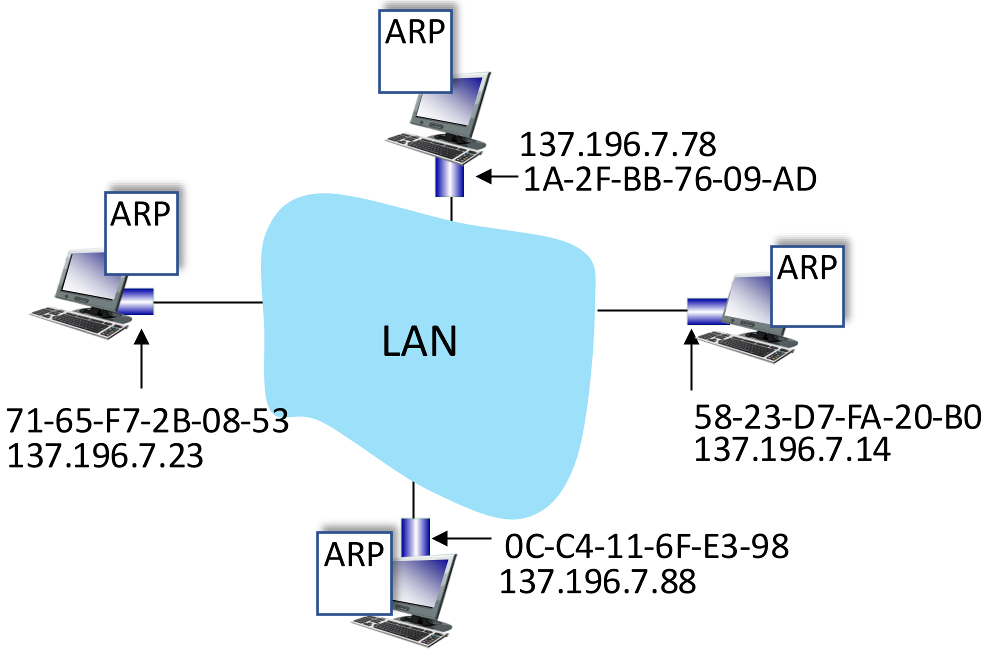


MAC addresses

- MAC address allocation administered by IEEE
- manufacturer buys portion of MAC address space (to assure uniqueness)
- 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

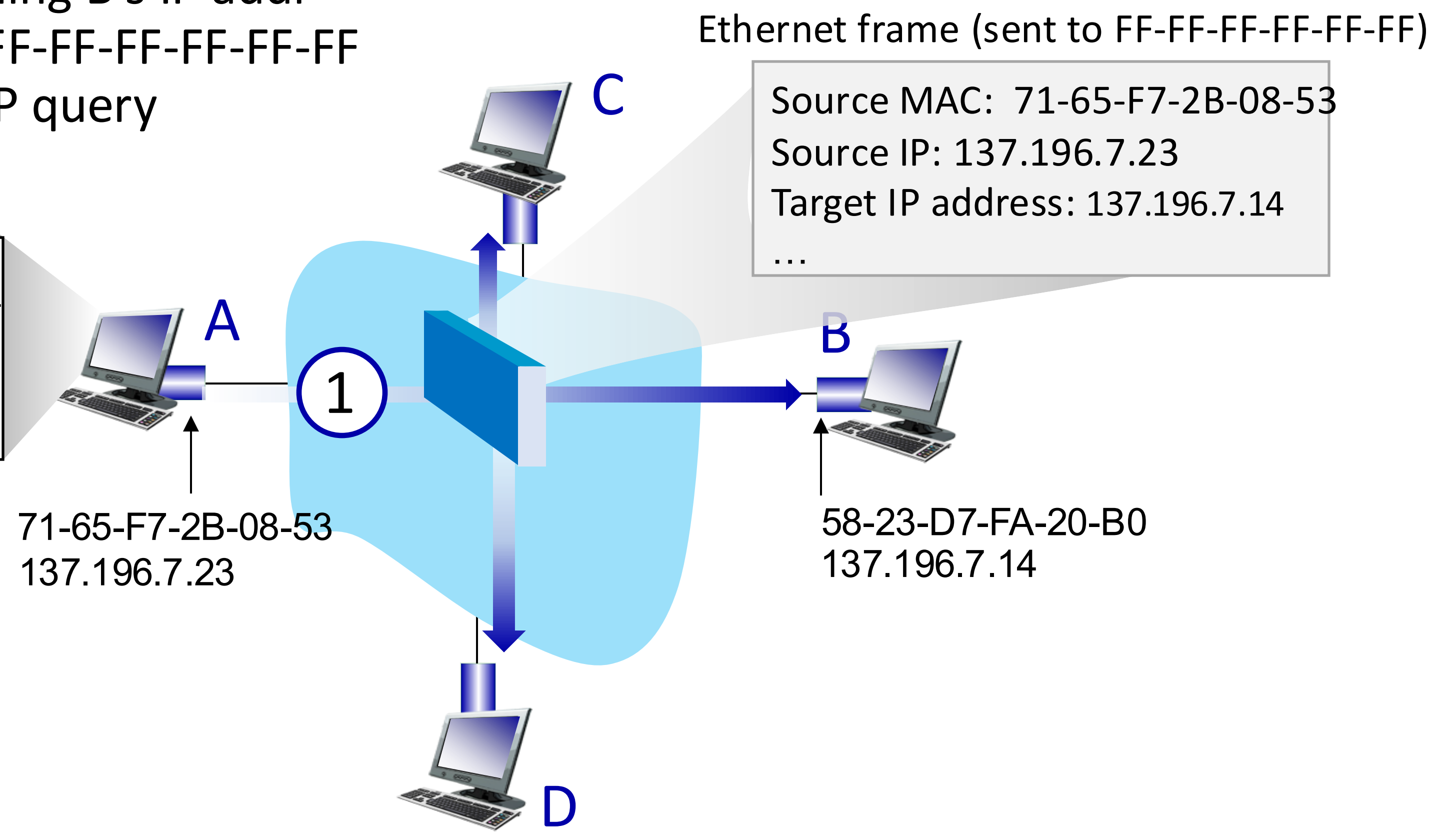
example: A wants to send datagram to B

- 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

example: A wants to send datagram to B

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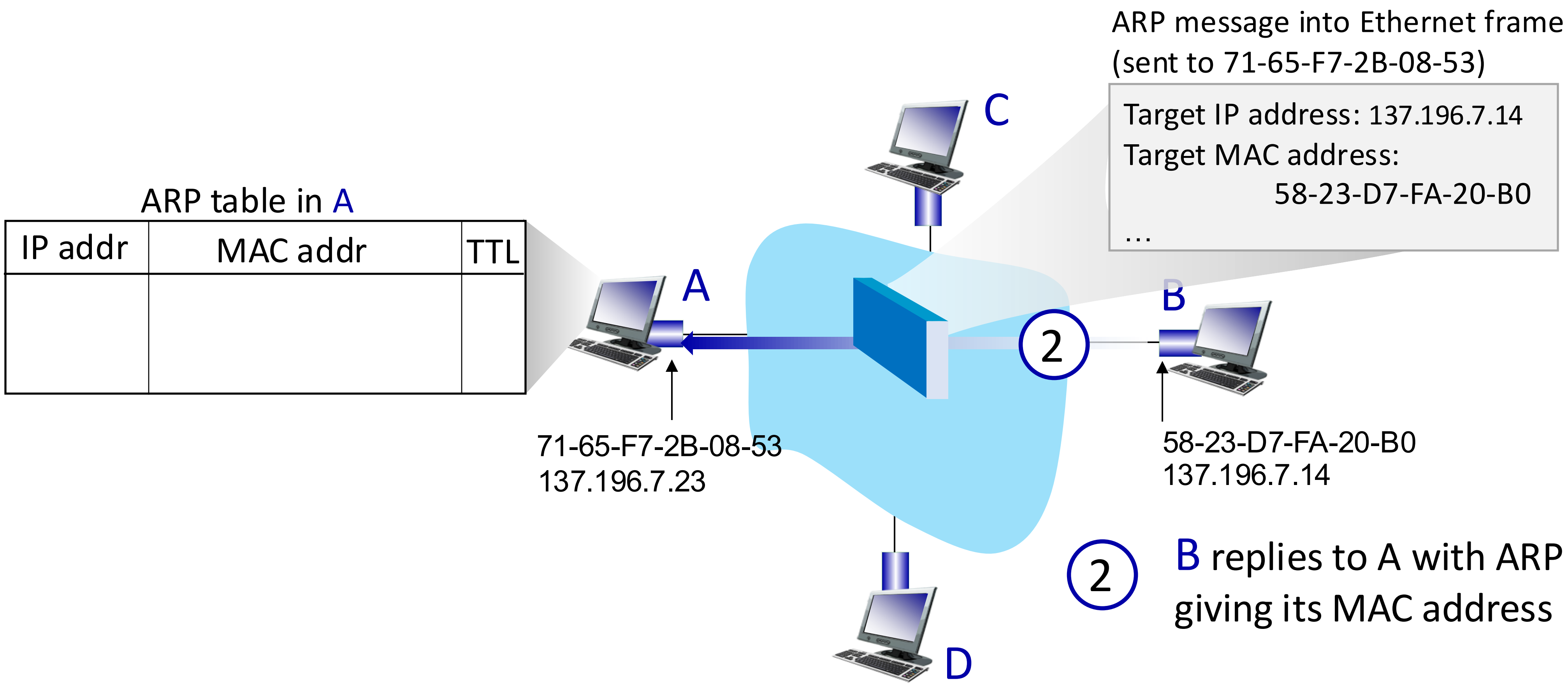


2 B replies to A with ARP response, giving its MAC address

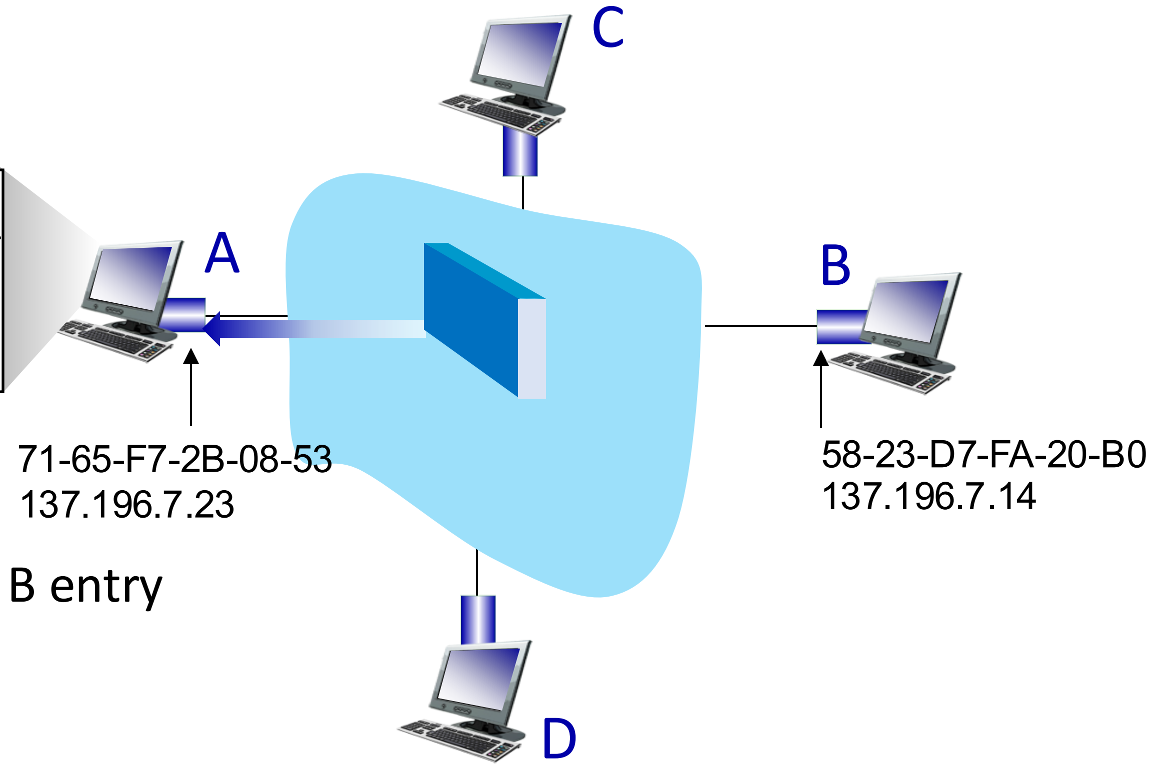
ARP protocol in action

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

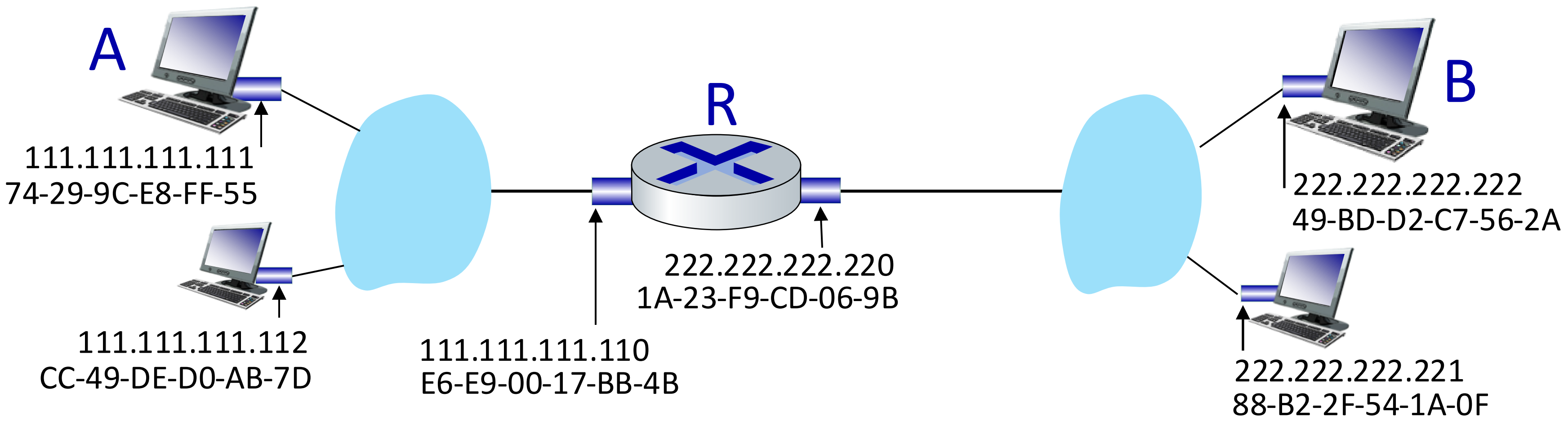


3 A receives B's reply, adds B entry into its local ARP table

Routing to another subnet: addressing

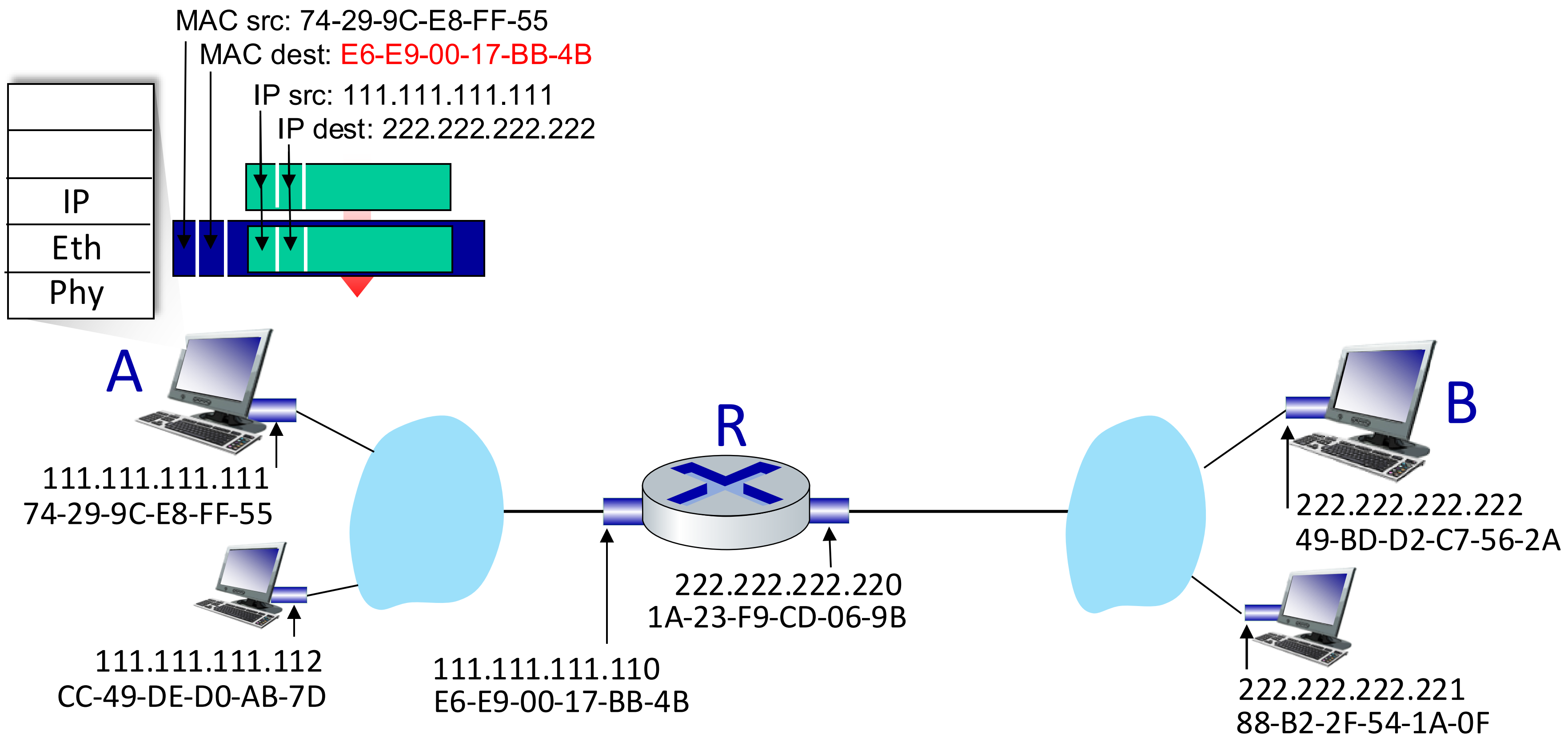
walkthrough: sending a datagram from *A* to *B* via *R*

- 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
  - A knows R's MAC address (how?)



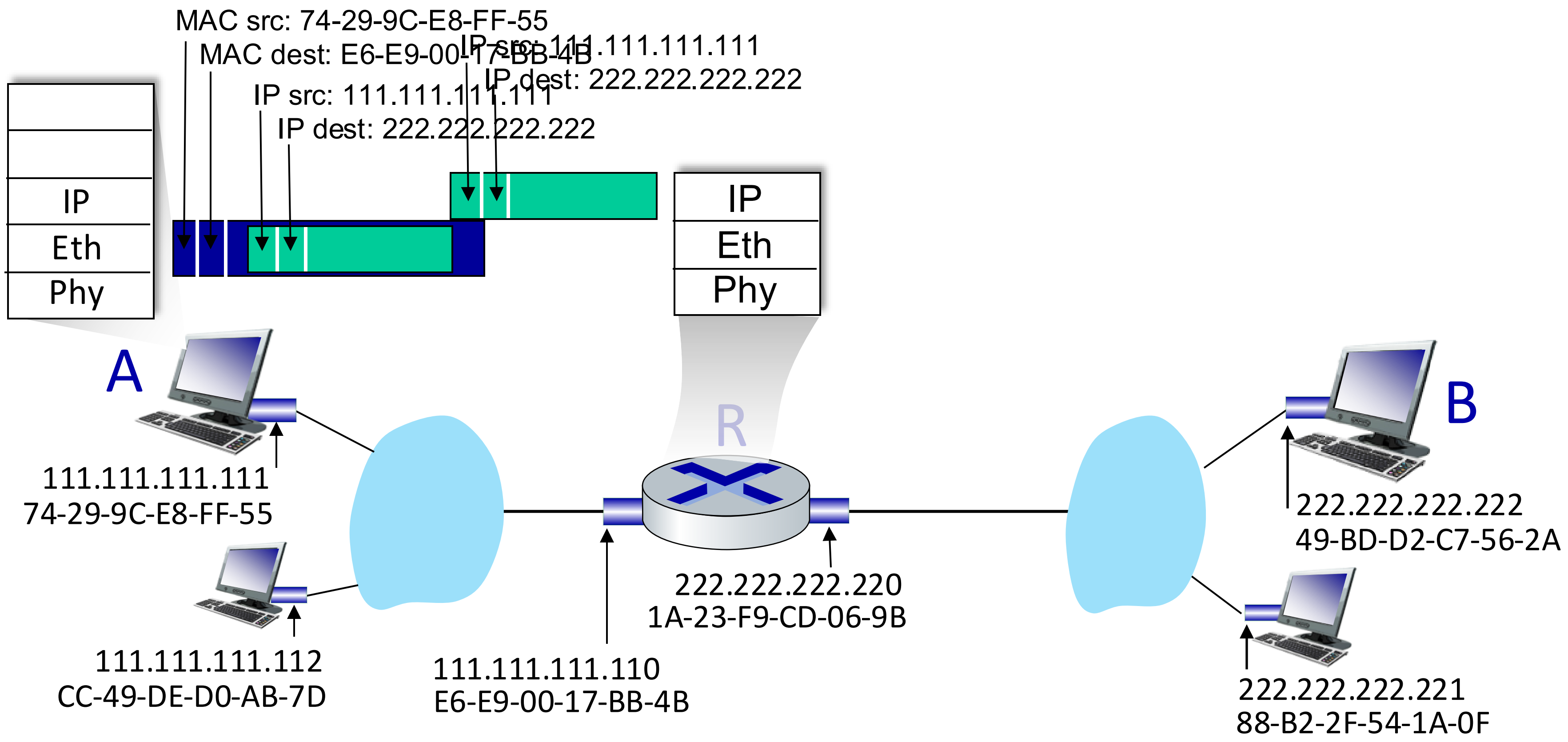
Routing to another subnet: addressing

- 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



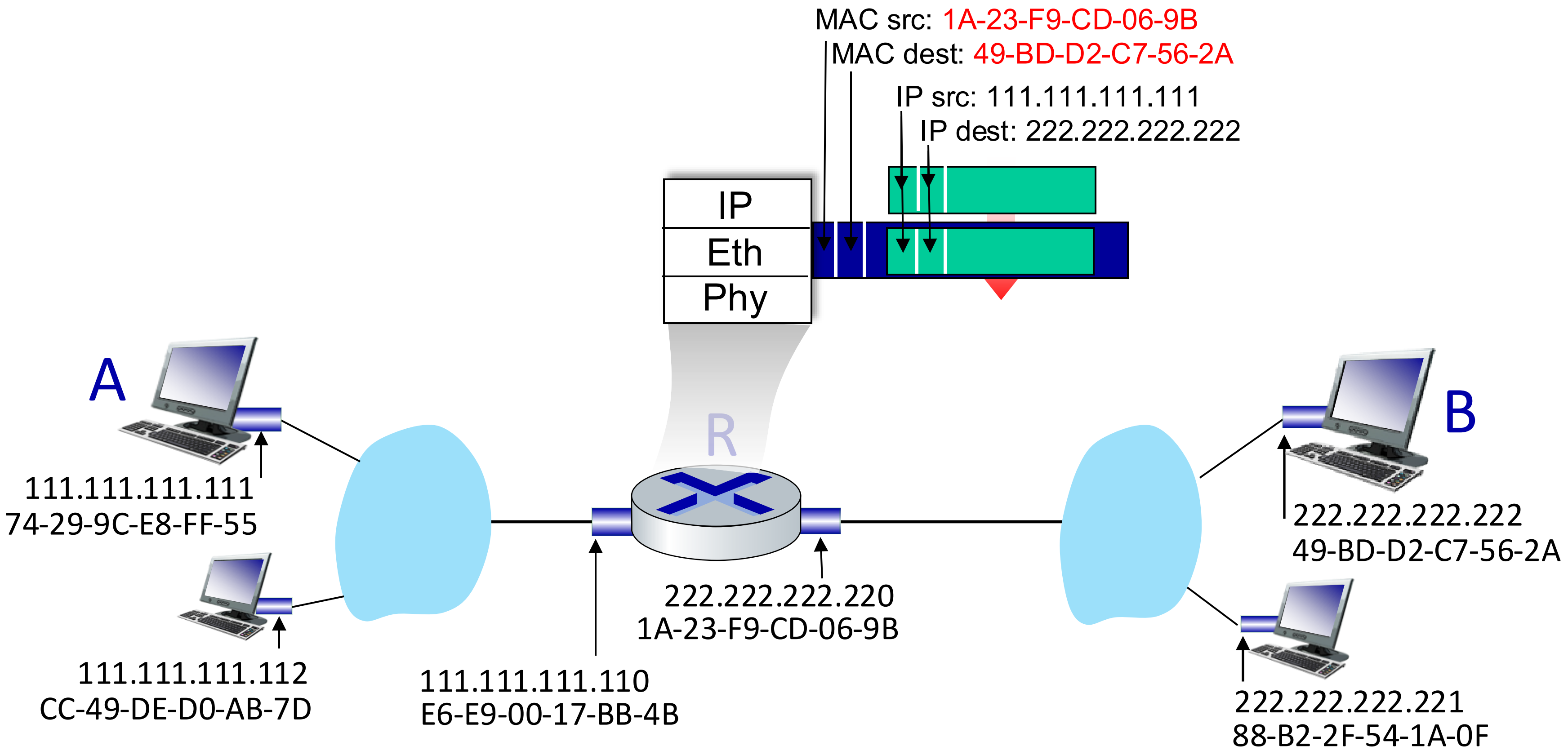
Routing to another subnet: addressing

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



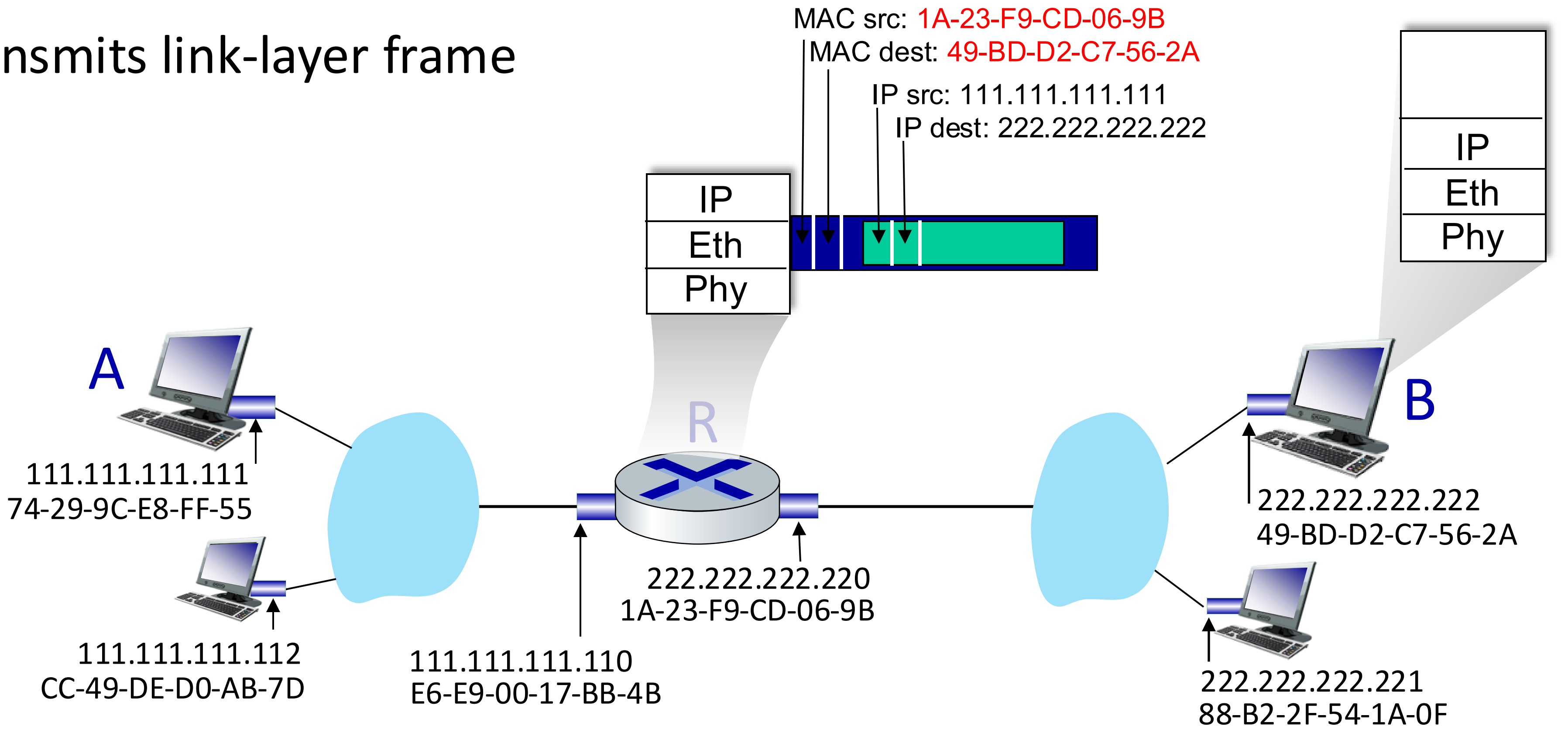
Routing to another subnet: addressing

- 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



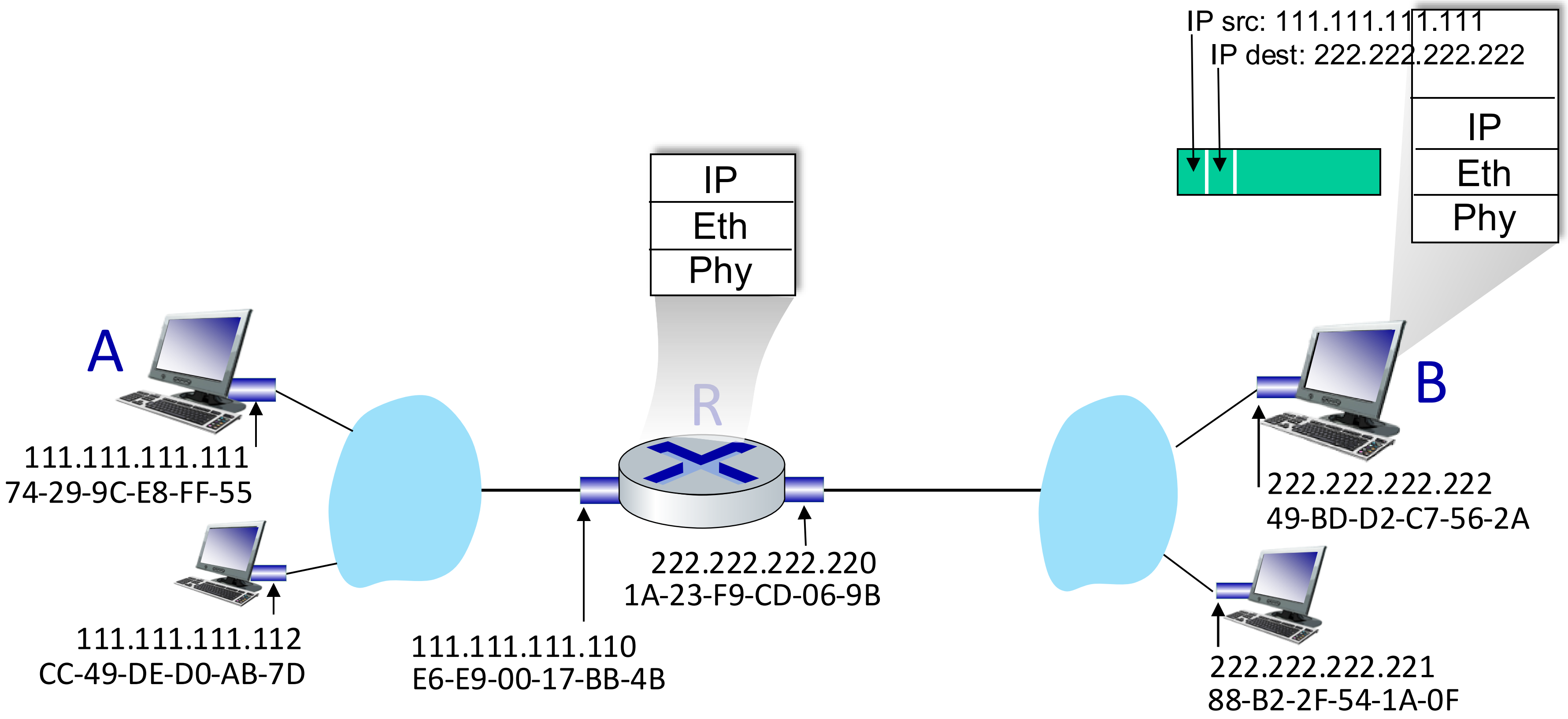
Routing to another subnet: addressing

- 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



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- link virtualization: MPLS
- data center networking

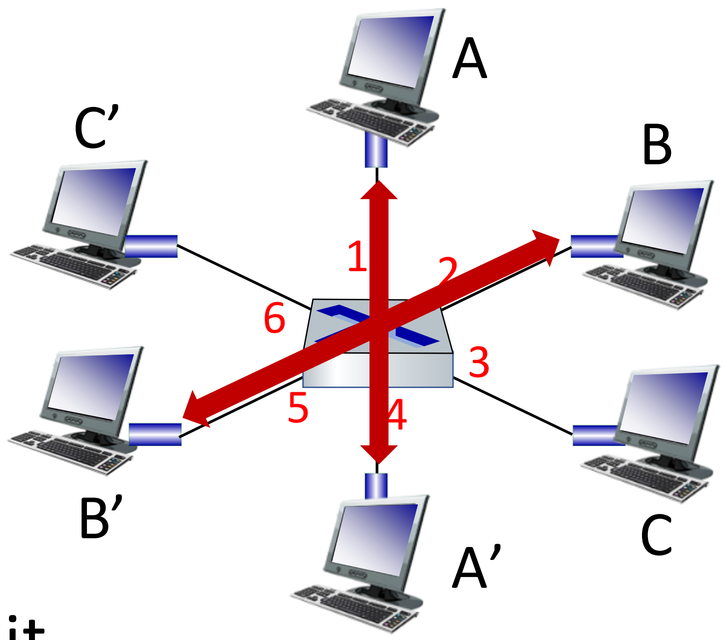
- a day in the life of a web request

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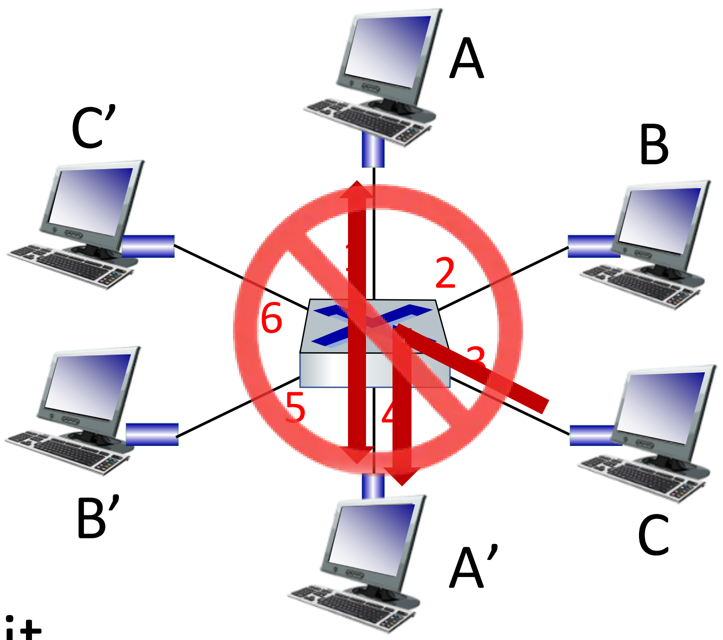
- switches buffer packets
- Ethernet protocol used on *each* incoming link, so:
  - no collisions; full duplex
  - each link is its own collision domain
- switching: A-to-A' and B-to-B' can transmit simultaneously, without collisions
  - but A-to-A' and C to A' can *not* happen simultaneously



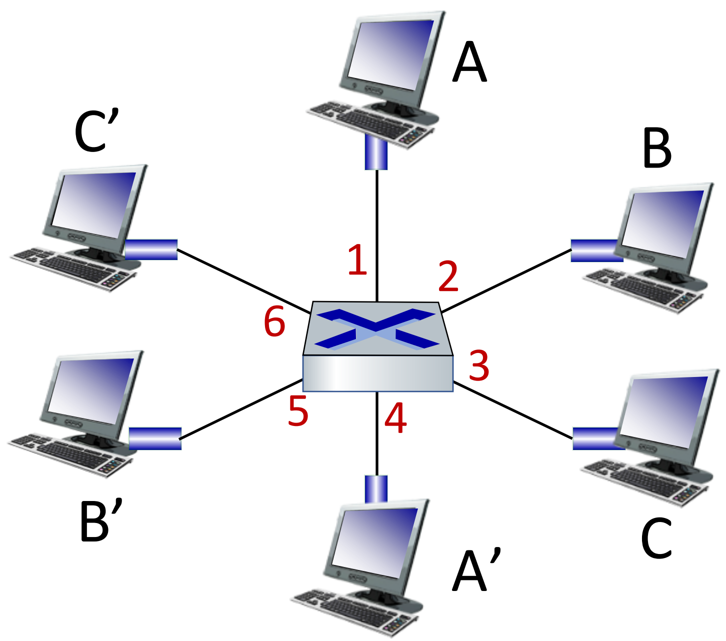
switch with six interfaces (1,2,3,4,5,6)

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!

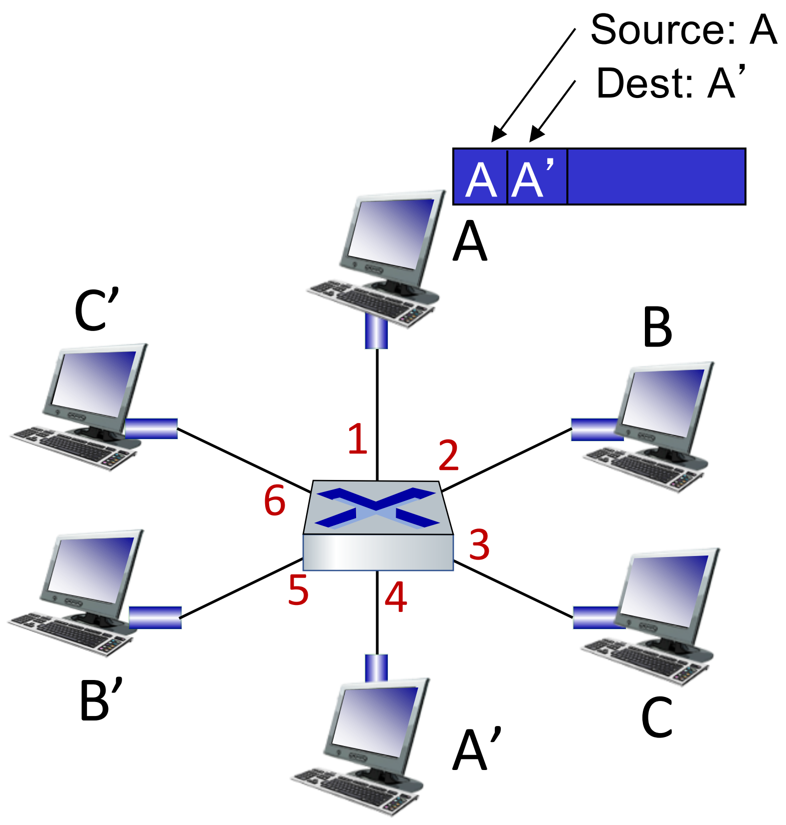


Q: how are entries created, maintained in switch table?

- 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)

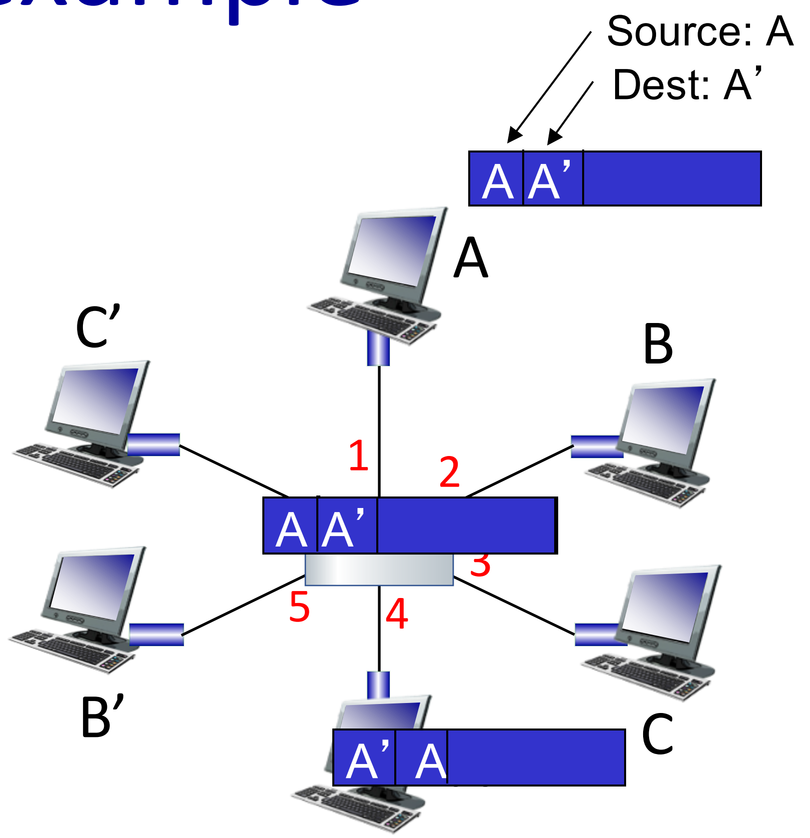
Switch: frame filtering/forwarding

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

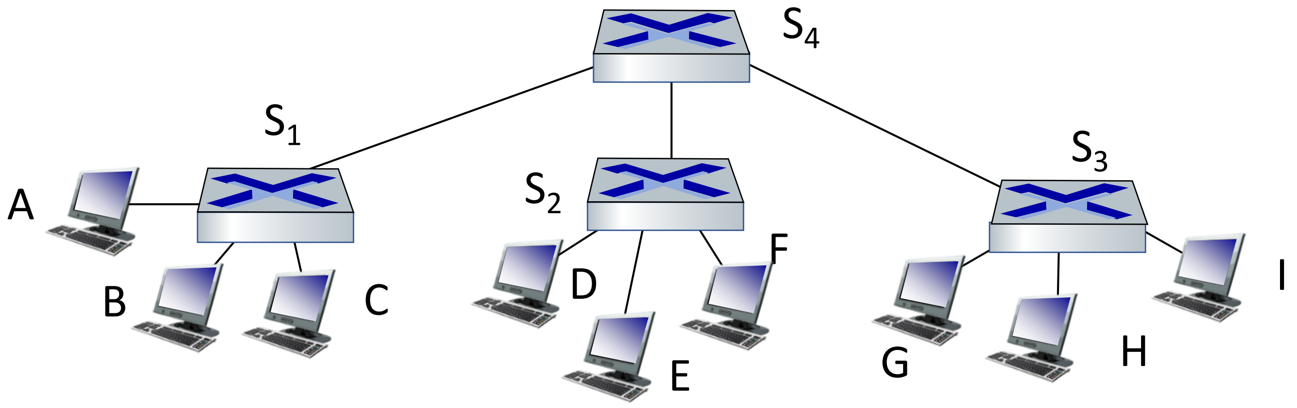


MAC addr	interface	TTL
A	*1*	*60*
A'	*4*	*60*

switch table (initially empty)

Interconnecting switches

self-learning switches can be connected together:

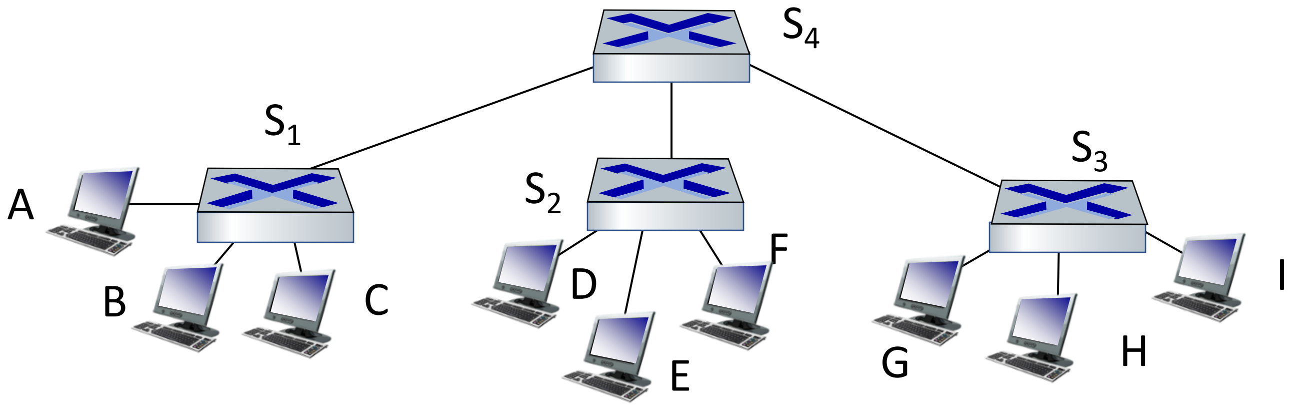


Q: sending from A to G - how does S_1 know to forward frame destined to G via S_4 and S_3?

- *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

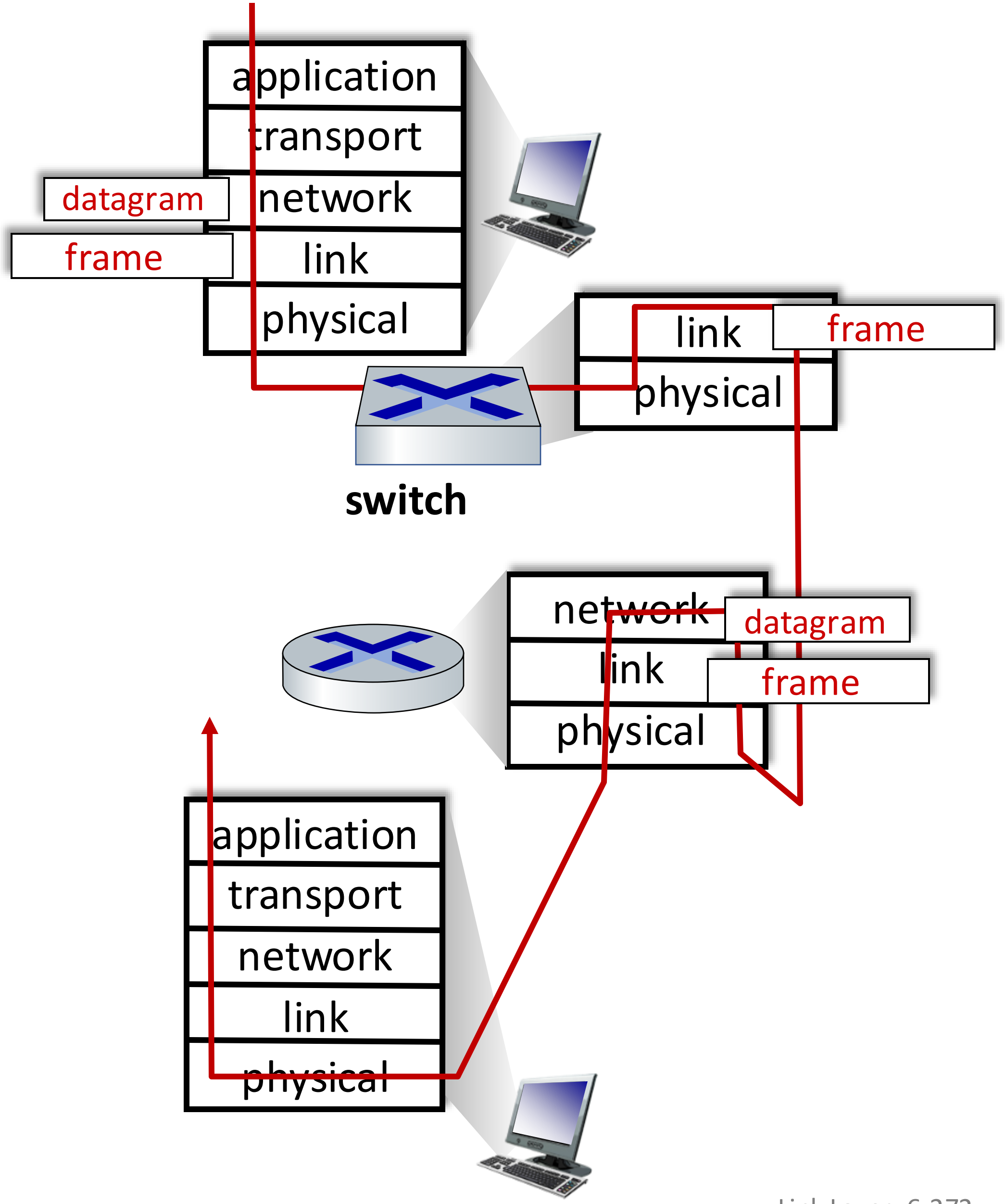
Switches vs. routers

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



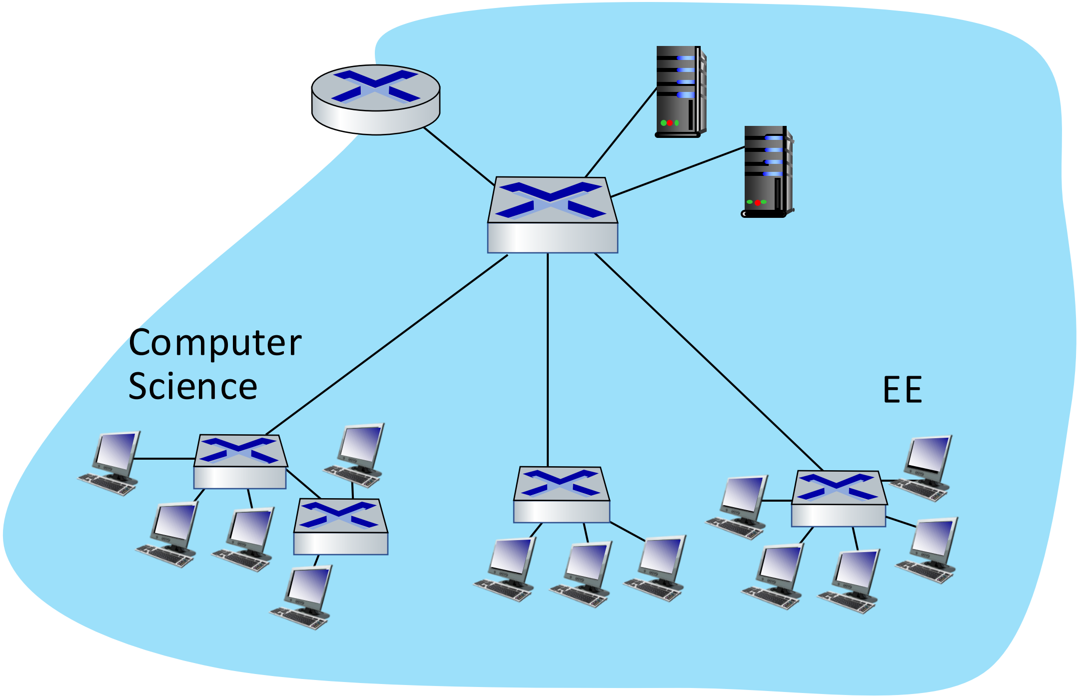
Link layer, LANs: roadmap

- 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

Virtual LANs (VLANs): motivation

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

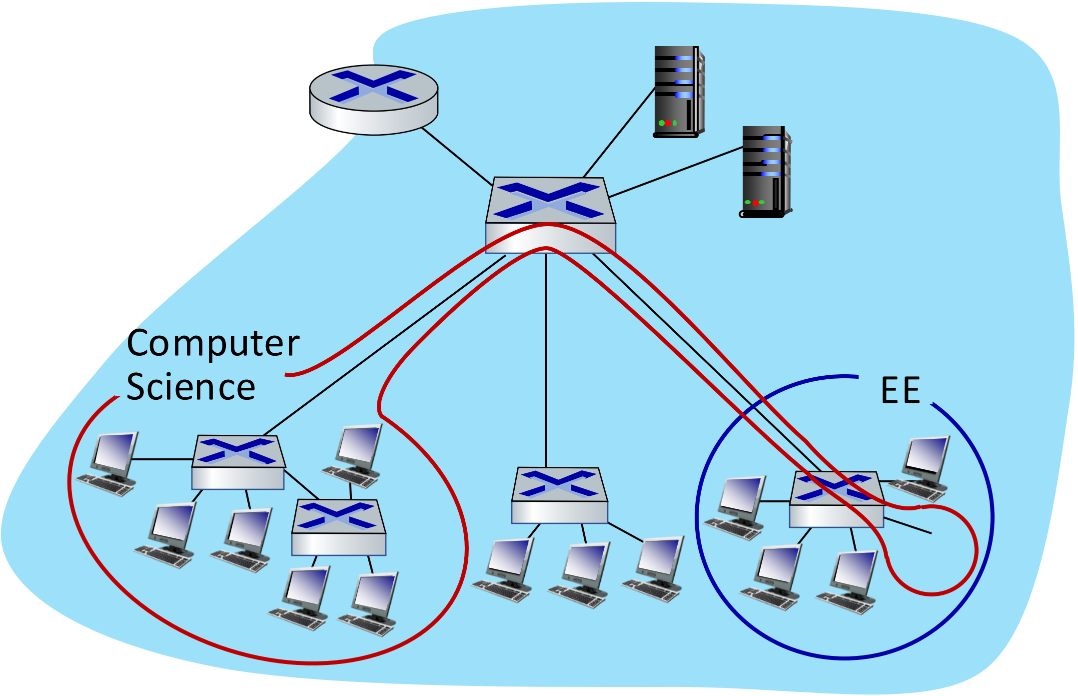


single broadcast domain:

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

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, efficiency issues

administrative issues:

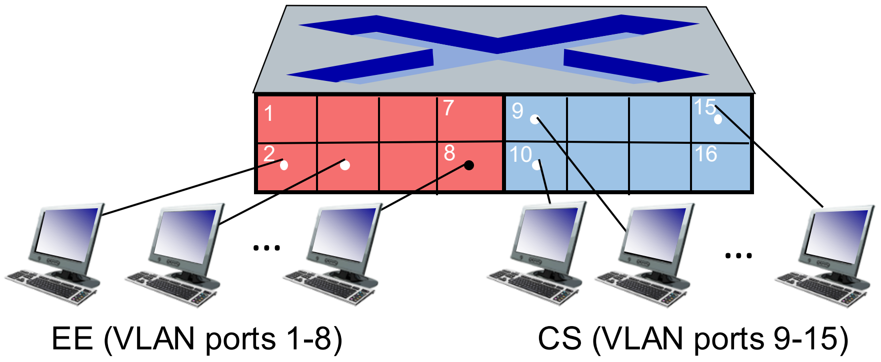
- CS user moves office to EE - *physically* attached to EE switch, but wants to remain *logically* attached to CS switch

Port-based VLANs

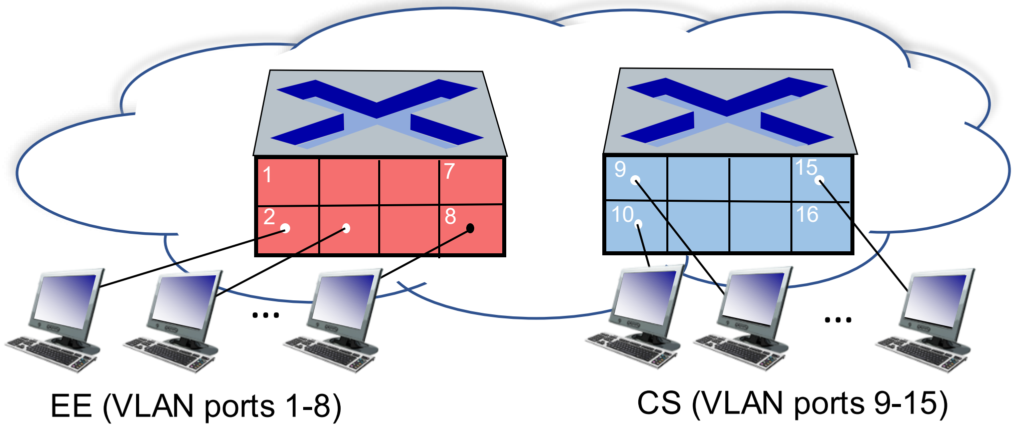
Virtual Local Area Network (VLAN)

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 ……

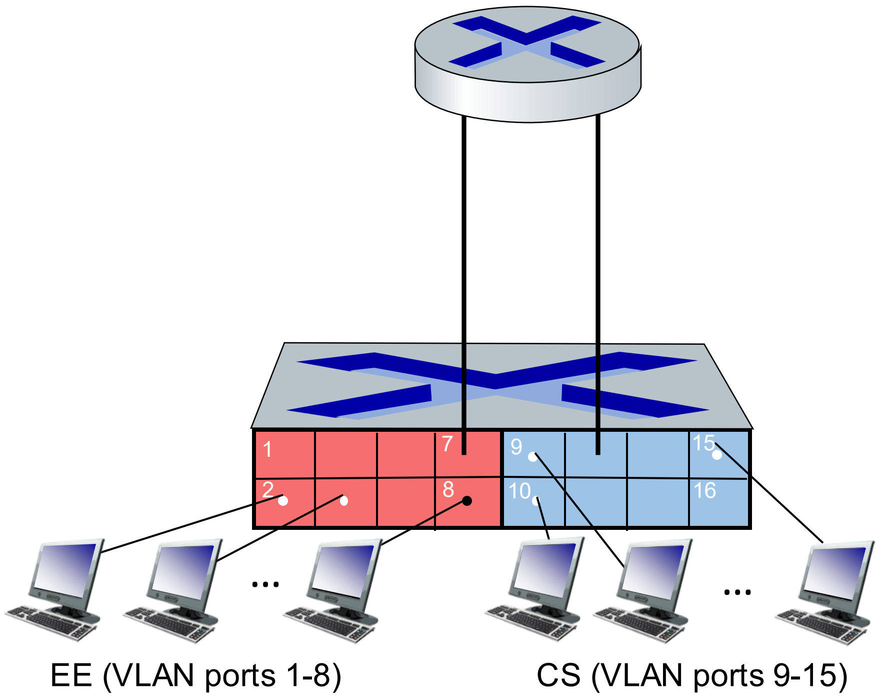


… operates as multiple virtual switches

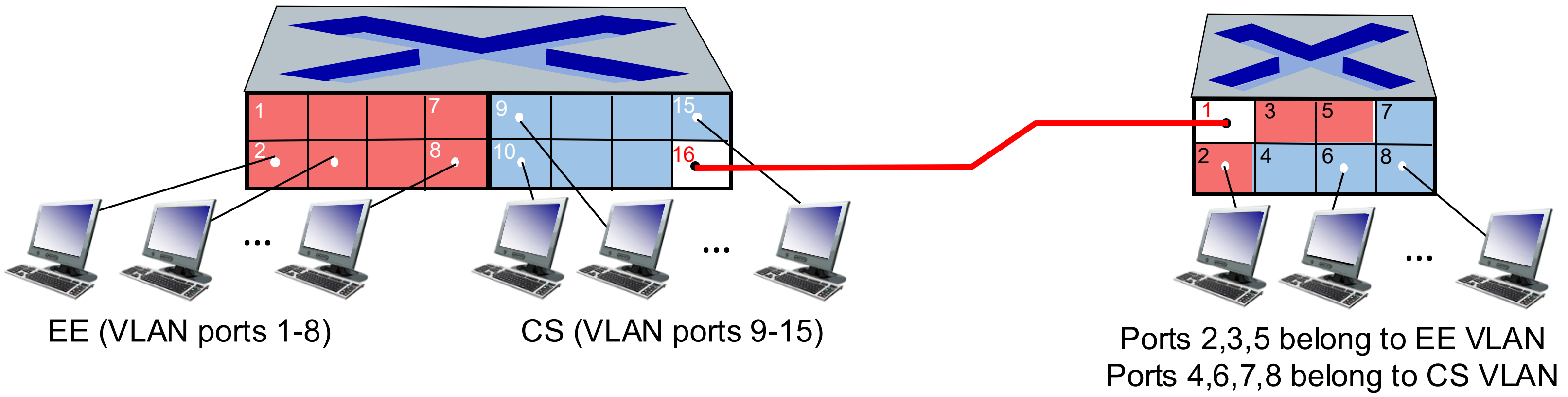


Port-based VLANs

- 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

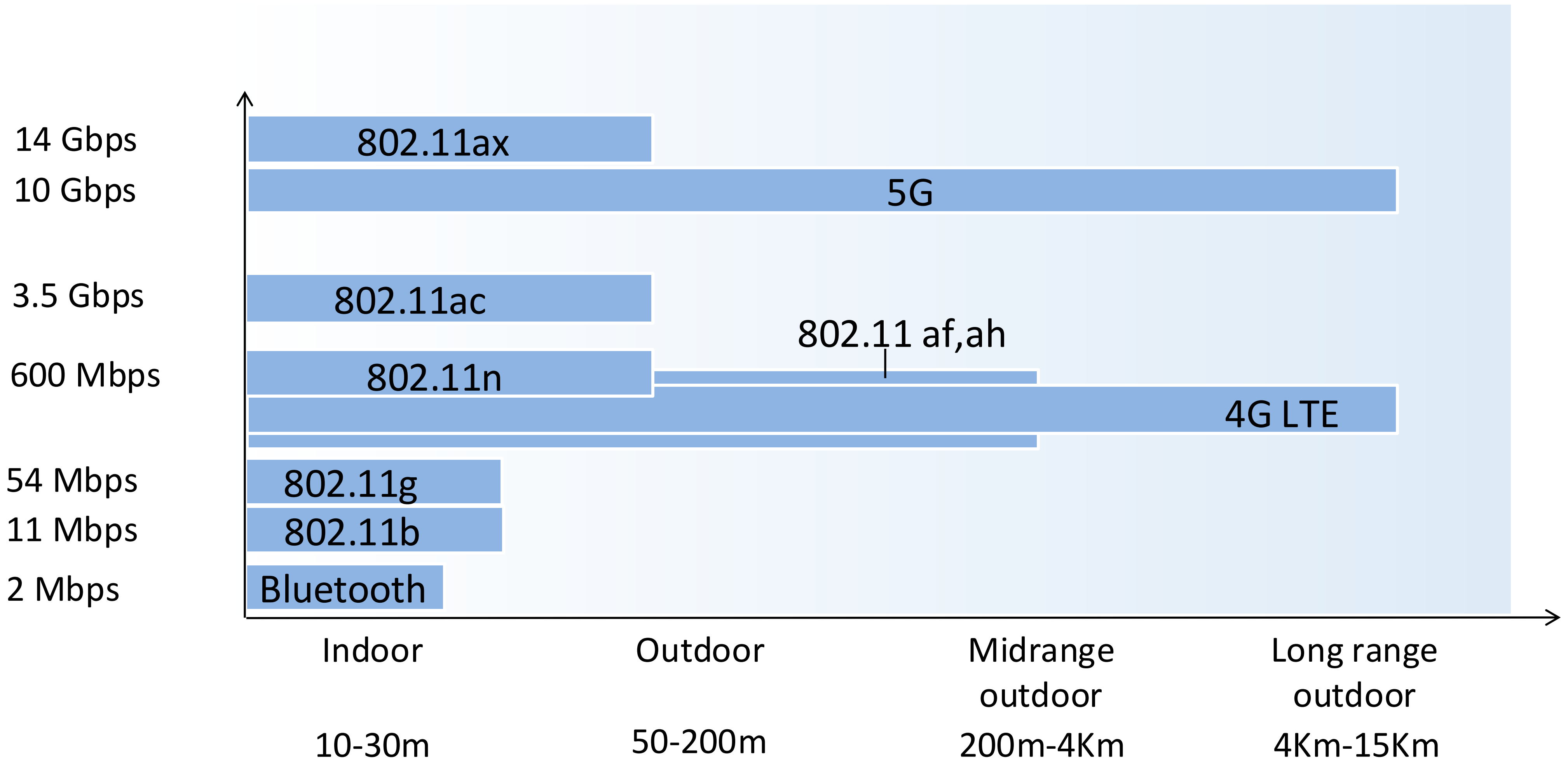


trunk port: carries frames between VLANS defined over multiple physical switches

- 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

Chapter 7
Wireless and Mobile Networks

Characteristics of selected wireless links



Chapter 7 outline

- Introduction

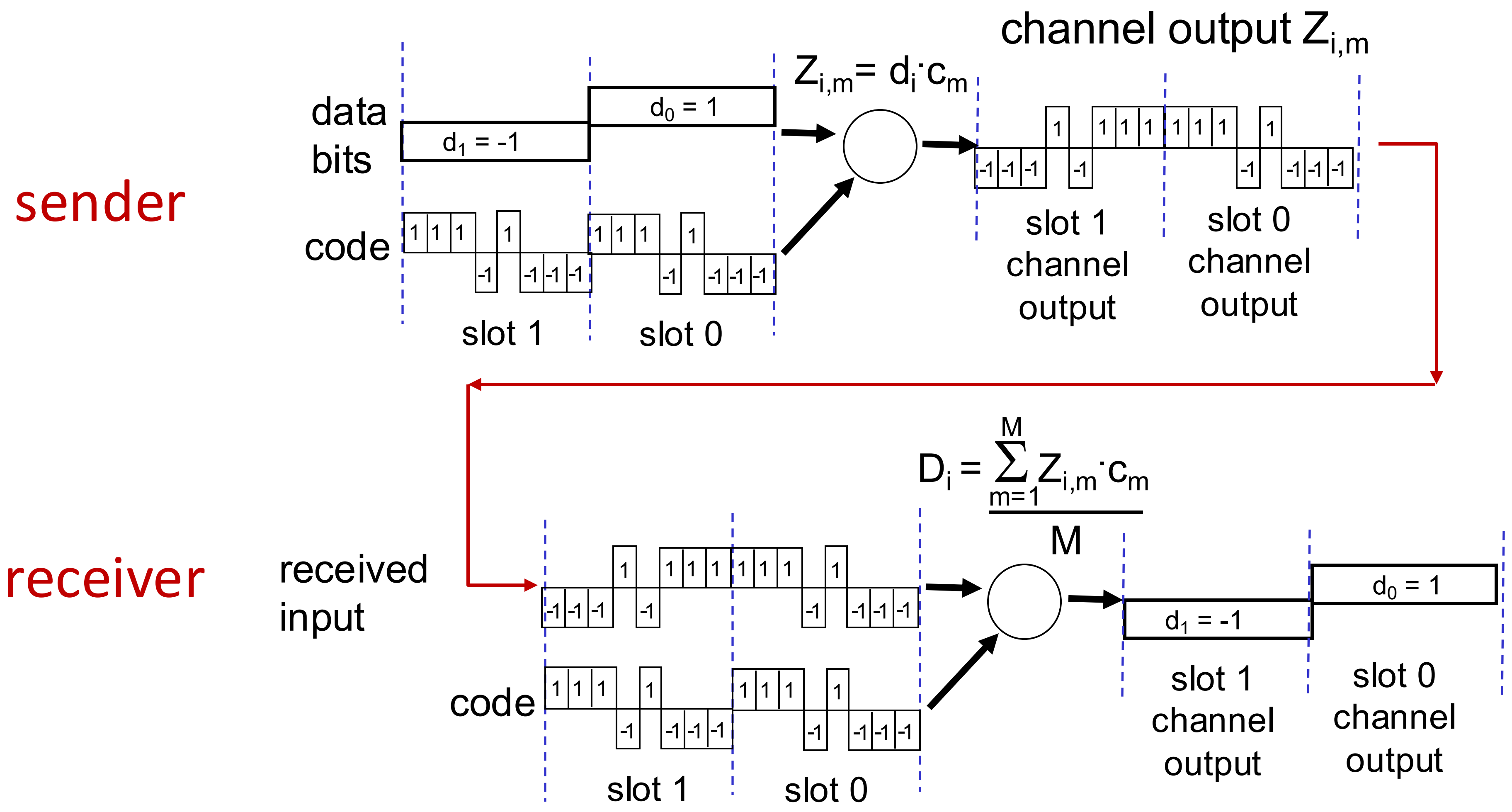
Wireless

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

Code Division Multiple Access (CDMA)

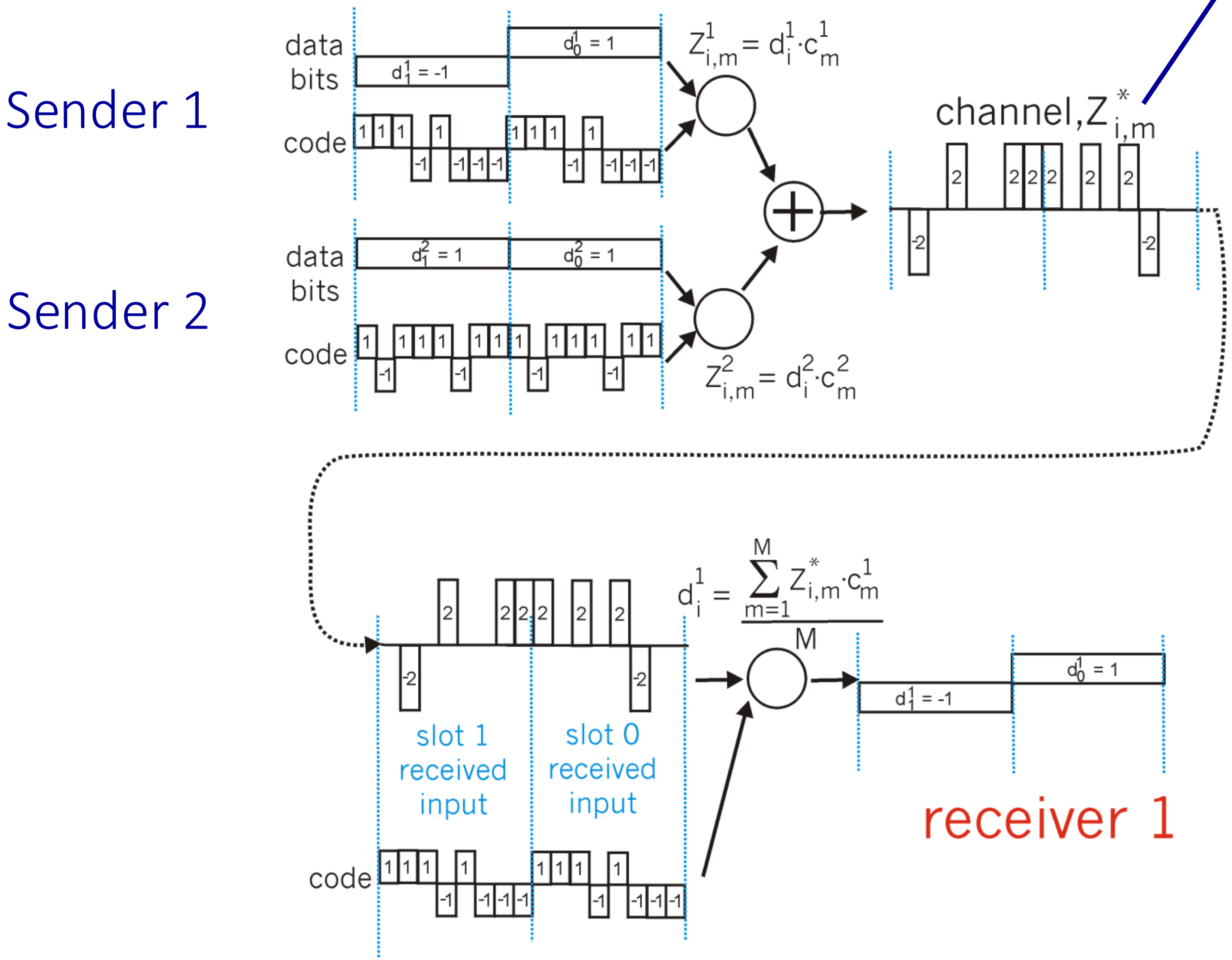
- unique "code" assigned to each user; i.e., code set partitioning
  - all users share same frequency, but each user has own "chipping" sequence (i.e., code) to encode data
  - allows multiple users to "coexist" and transmit simultaneously with minimal interference (if codes are "orthogonal")
- encoding: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

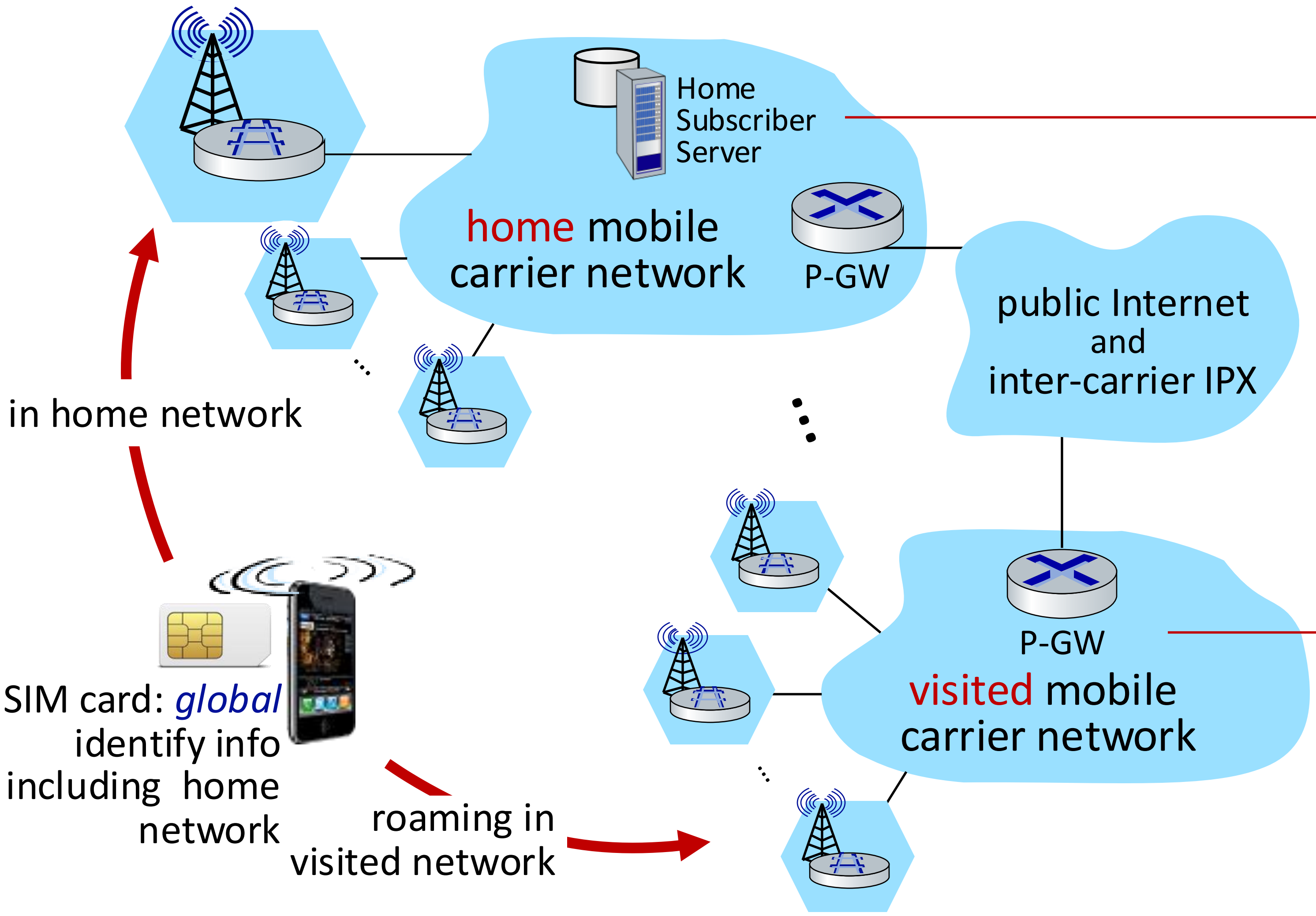


channel sums together transmissions by sender 1 and 2

using same code as sender 1, receiver recovers sender 1's original data from summed channel data!

... now *that's* useful!

Home network, visited network: 4G/5G



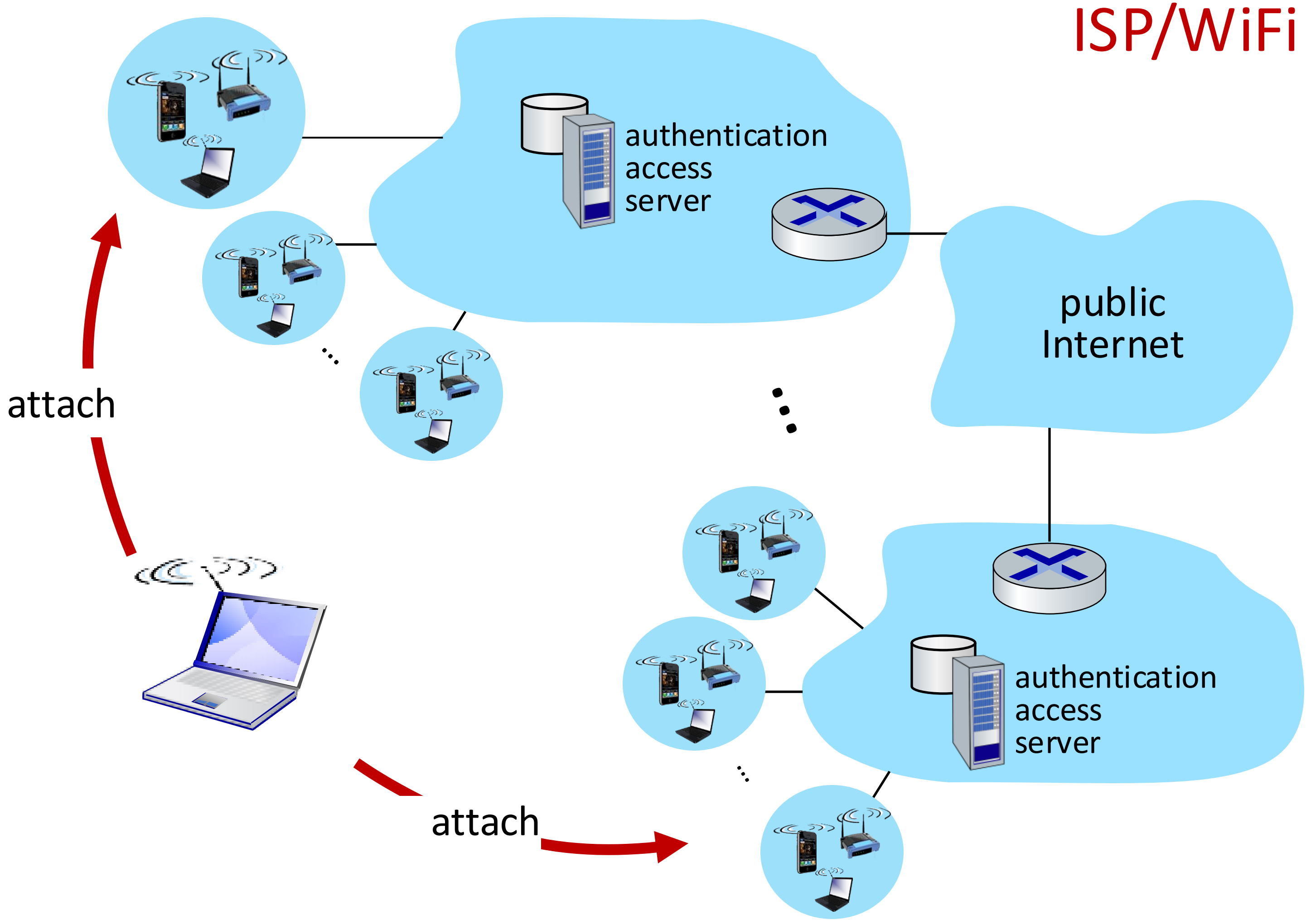
home network:

- (paid) service plan with cellular provider, e.g., Unicom, CMCC
- 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

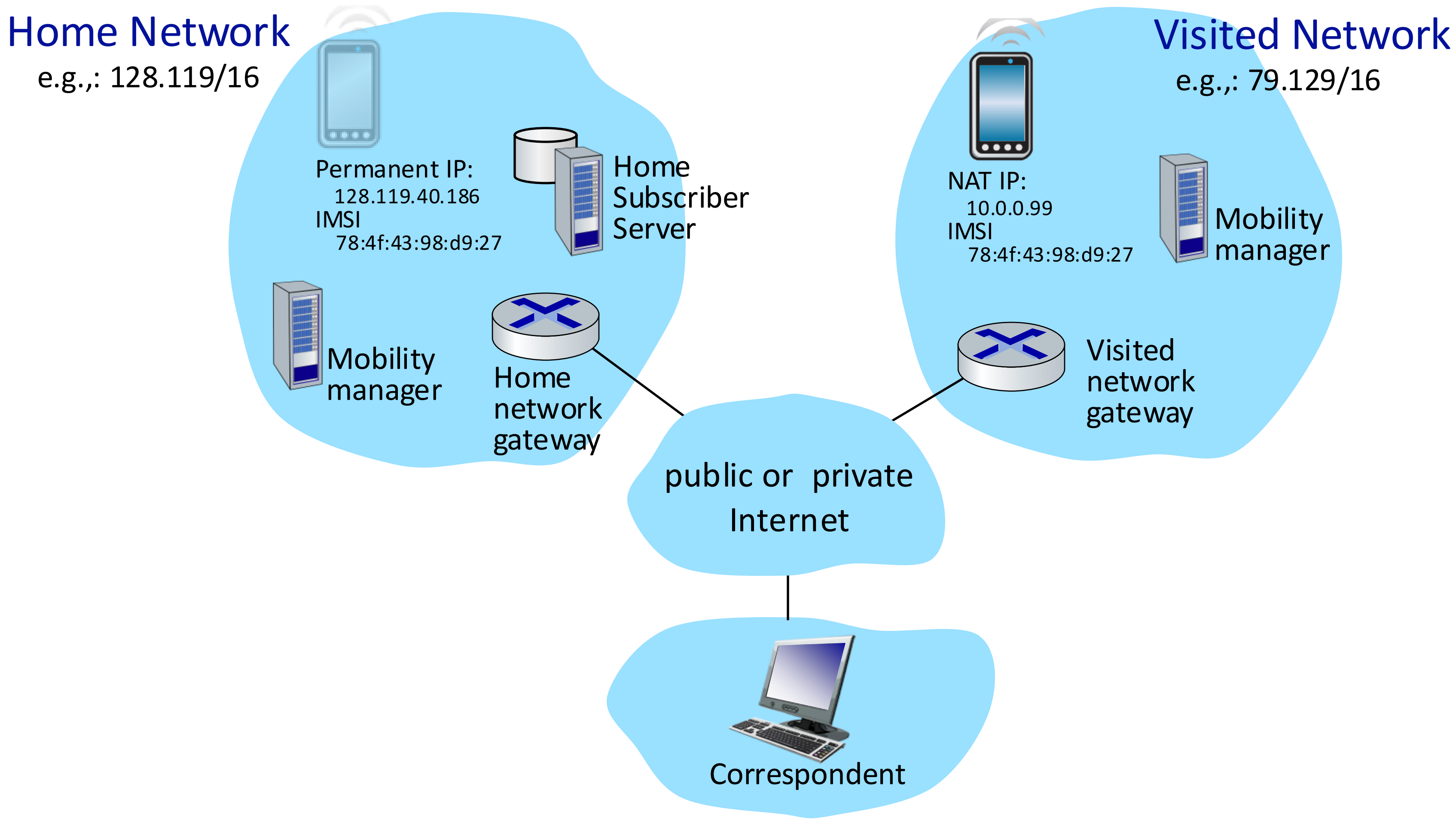
Home network, visited network: ISP/WiFi



ISP/WiFi: no notion of global "home"

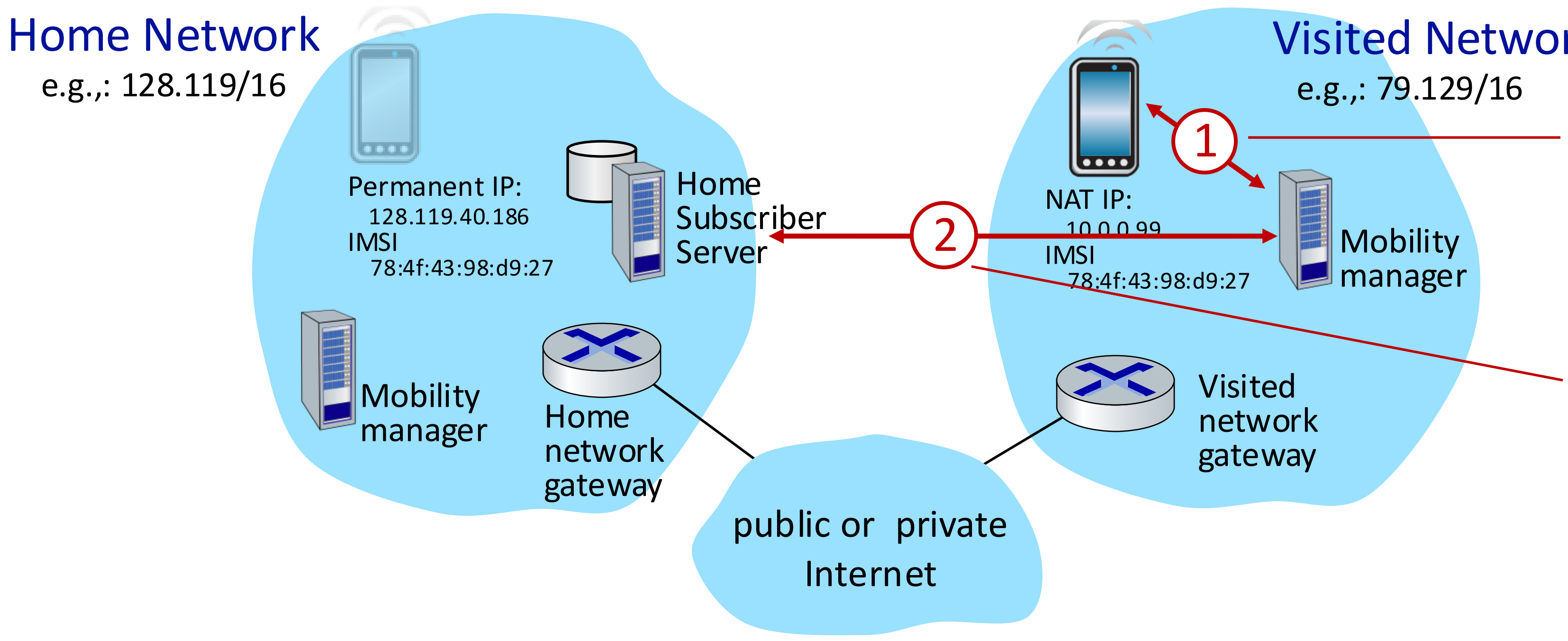
- credentials from ISP (e.g., username, password) stored on device or with user
- different networks: different credentials
  - some exceptions (e.g., eduroam)
  - architectures exist (mobile IP) for 4G-like mobility, but not used

Home network, visited network: generic



IMSI (International Mobile Subscriber Identity)

Registration: home needs to know where you are!



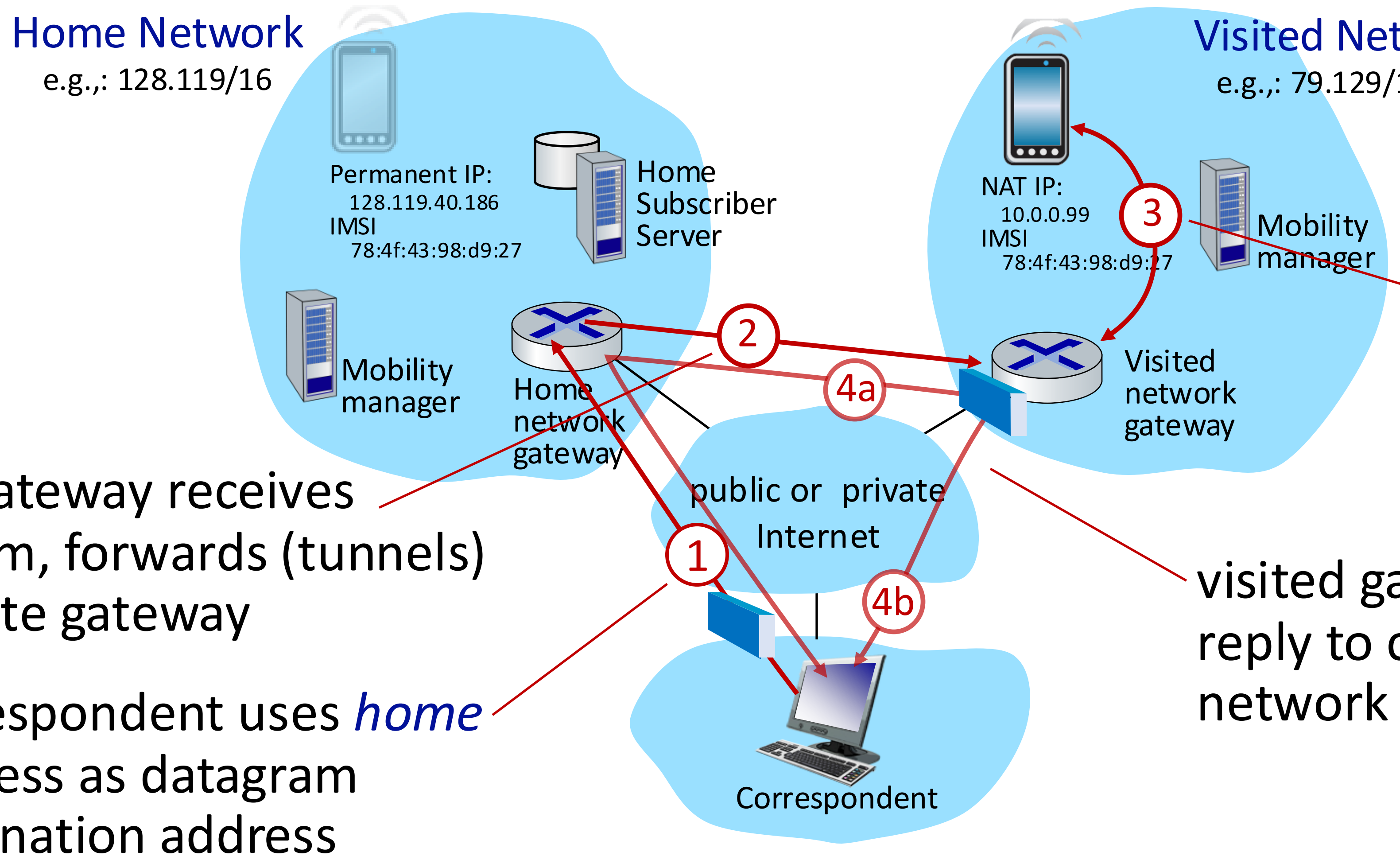
mobile *associates* with visited mobility manager

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



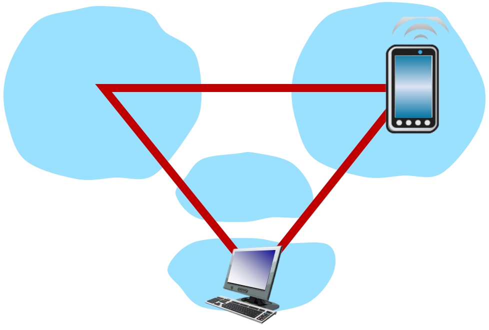
visited gateway router forwards to mobile

home gateway receives datagram, forwards (tunnels) to remote gateway

correspondent uses *home* address as datagram destination address

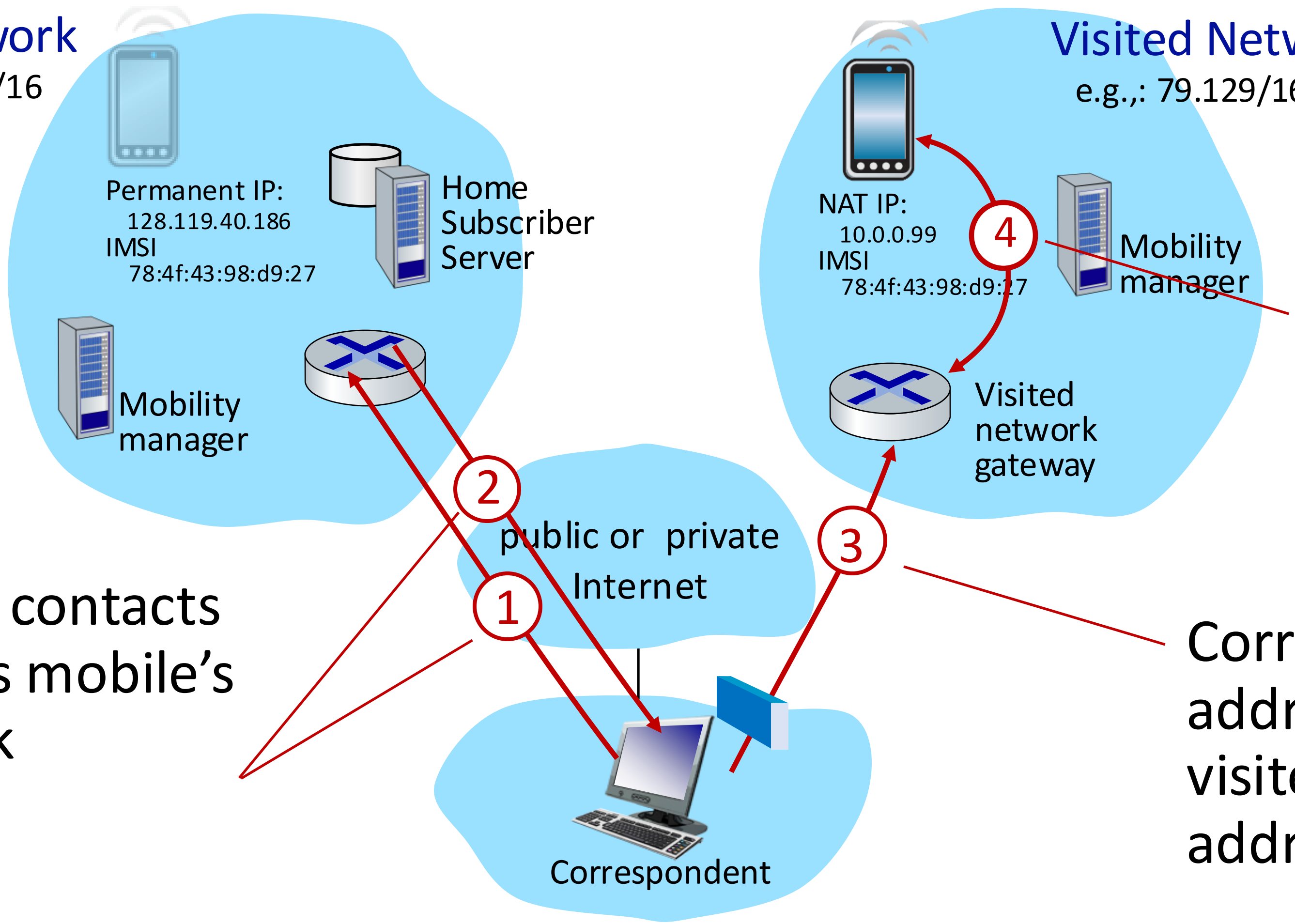
visited gateway router forwards reply to correspondent via home network (4a) or directly (4b)

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

Chapter 8
Security

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

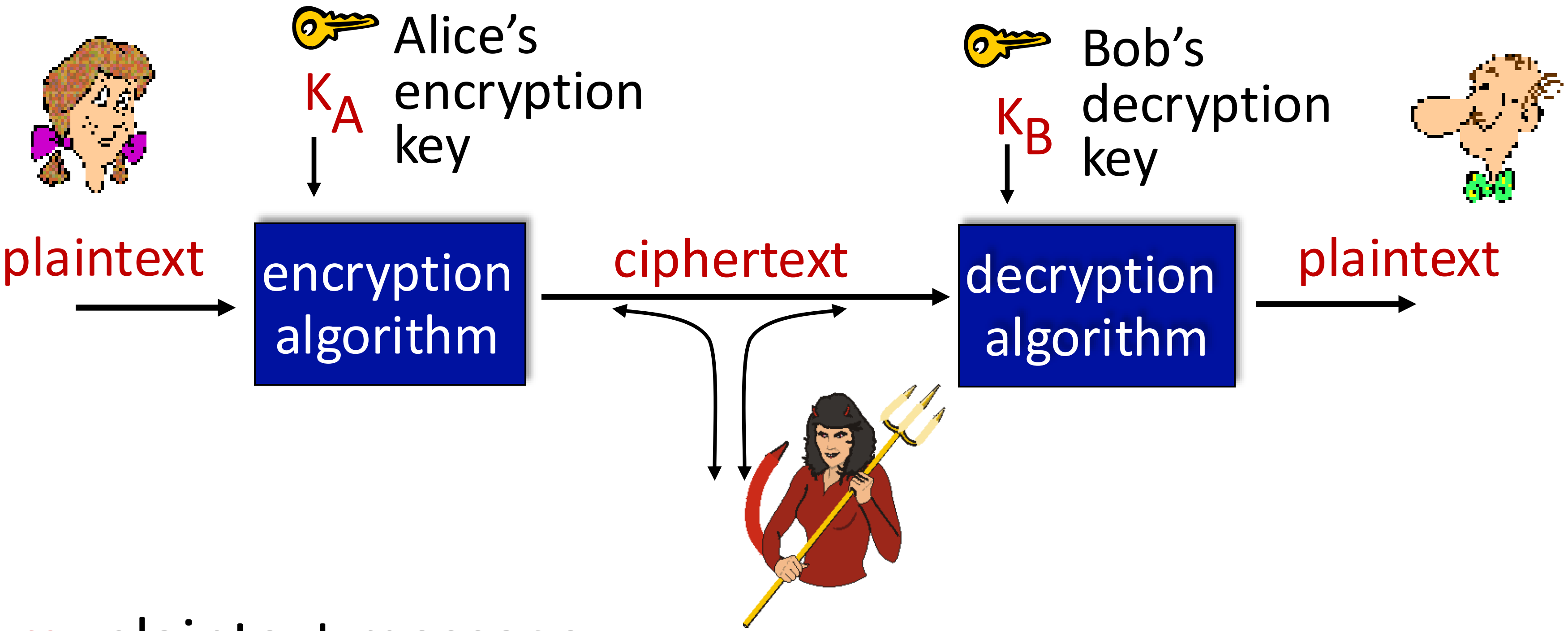
message integrity: sender, receiver want to ensure message not altered (in transit, or afterwards) without detection

access and availability: services must be accessible and available to users

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

The language of cryptography



m: plaintext message

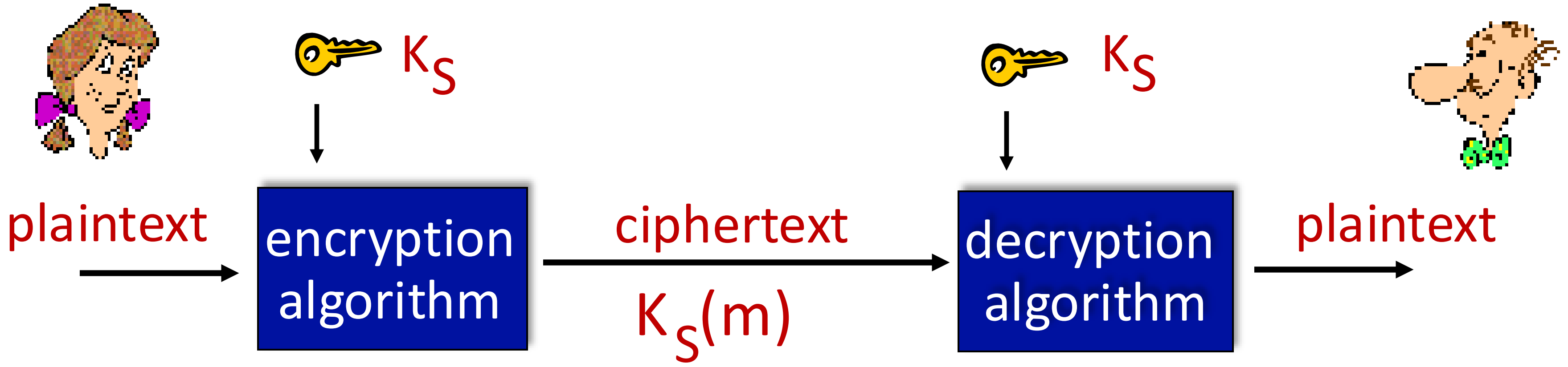
$K_A(m)$: ciphertext, encrypted with key K_A

$m = K_B(K_A(m))$

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

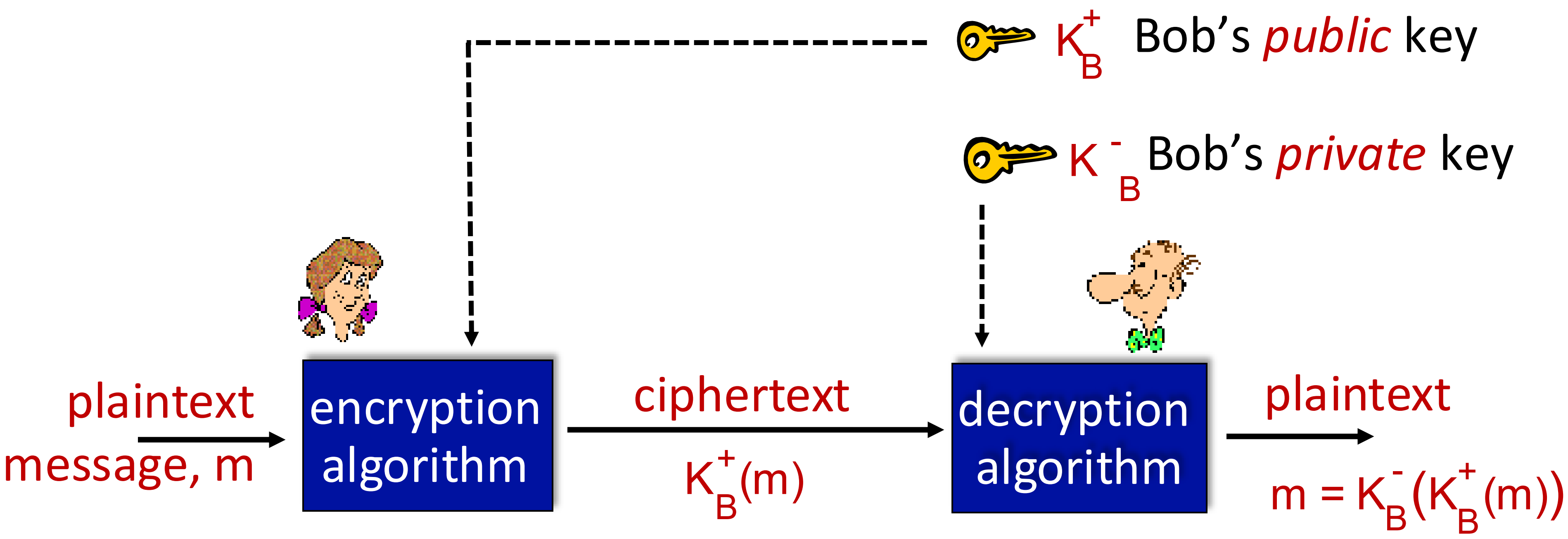
symmetric key crypto:

- 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!

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: $\underbrace{\text{bit pattern}}_{00001100}$ $\underbrace{m}_{12}$ $\underbrace{m^e}_{24832}$ $\underbrace{c = m^e \bmod n}_{17}$

decrypt: $\underbrace{c}_{17}$ $\underbrace{c^d}_{481968572106750915091411825223071697}$ $\underbrace{m = c^d \bmod n}_{12}$

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

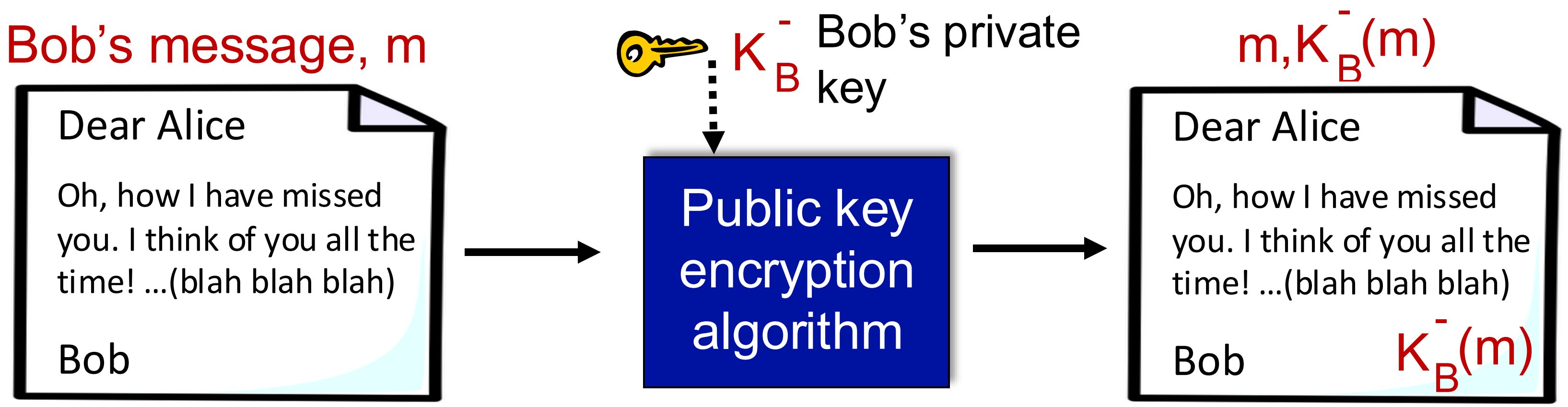
session key, K_S

- Bob and Alice use RSA to exchange a symmetric session key K_S
- once both have K_S, they use symmetric key cryptography

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:

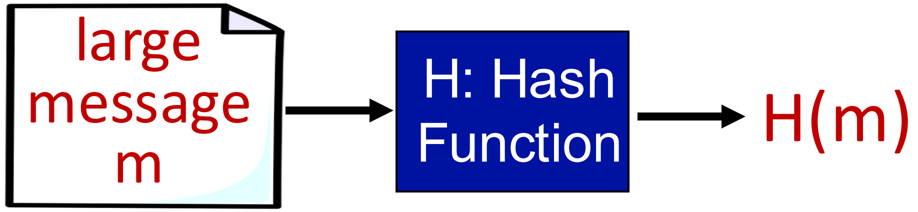
✓ Alice can take m, and signature $K_B^-(m)$ to court and prove that Bob signed m

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:

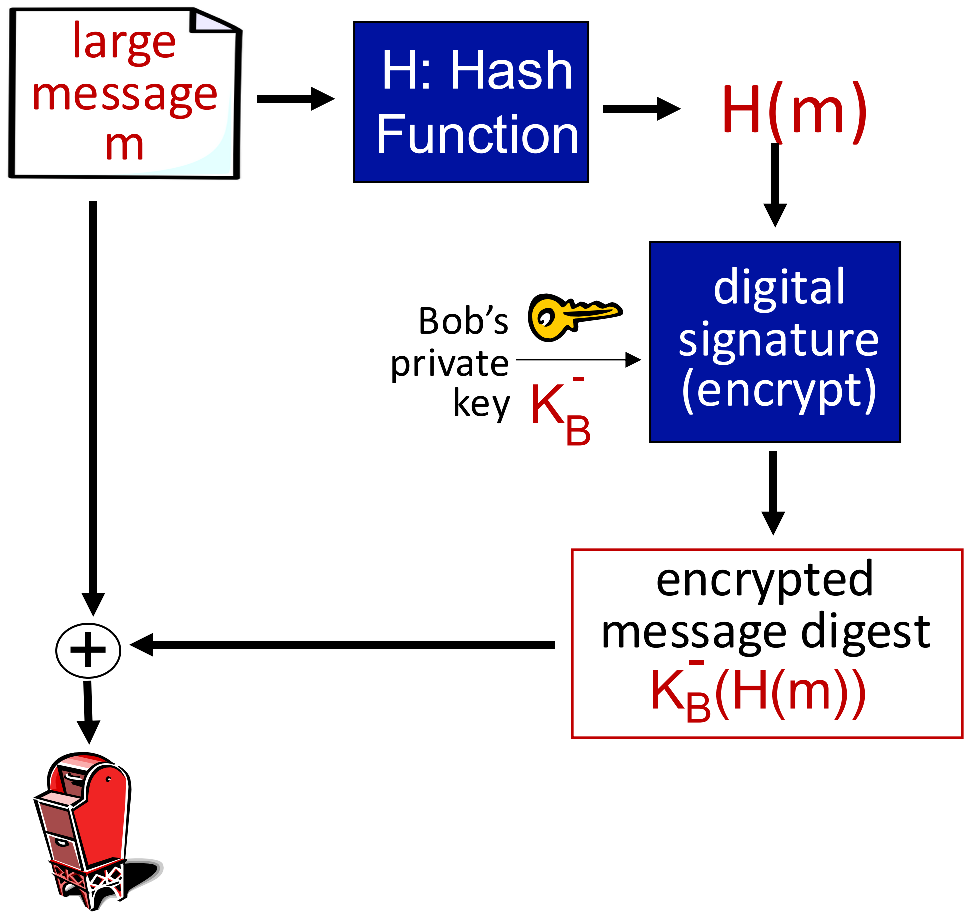
message	ASCII format
I O U 1	49 4F 55 31
0 0 . 9	30 30 2E 39
9 B O B	39 42 D2 42
	B2 C1 D2 AC

message	ASCII format
I O U 9	49 4F 55 39
0 0 . 1	30 30 2E 31
9 B O B	39 42 D2 42
	B2 C1 D2 AC

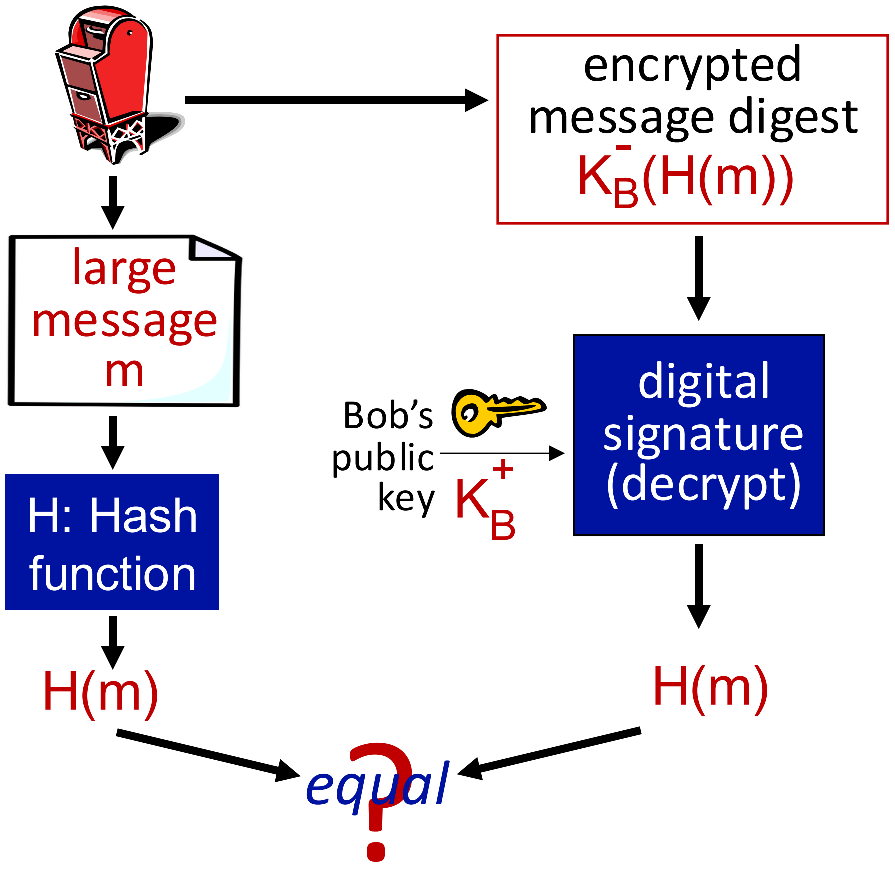
different messages but identical checksums!

Digital signature = signed message digest

Bob sends digitally signed message:



Alice verifies signature, integrity of digitally signed message:



Hash function algorithms

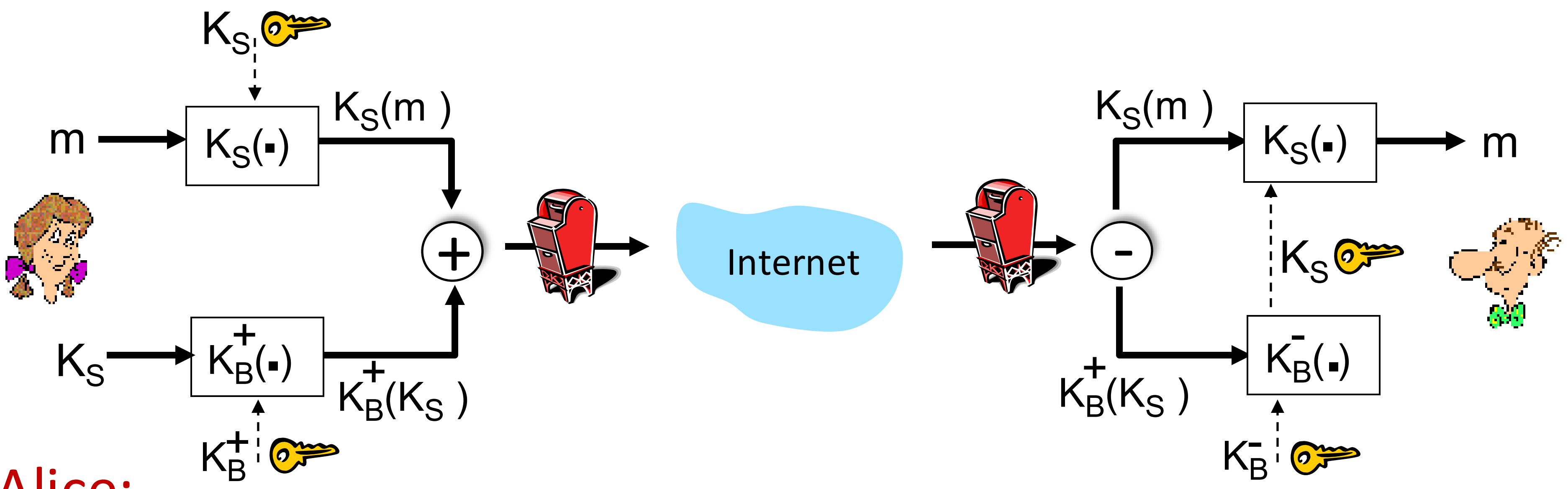
- MD5 hash function widely used (RFC 1321)
  - computes 128-bit message digest in 4-step process.
  - arbitrary 128-bit string x, appears difficult to construct msg m whose MD5 hash is equal to x
- SHA-1 is also used
  - US standard [NIST, FIPS PUB 180-1]
  - 160-bit message digest

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.

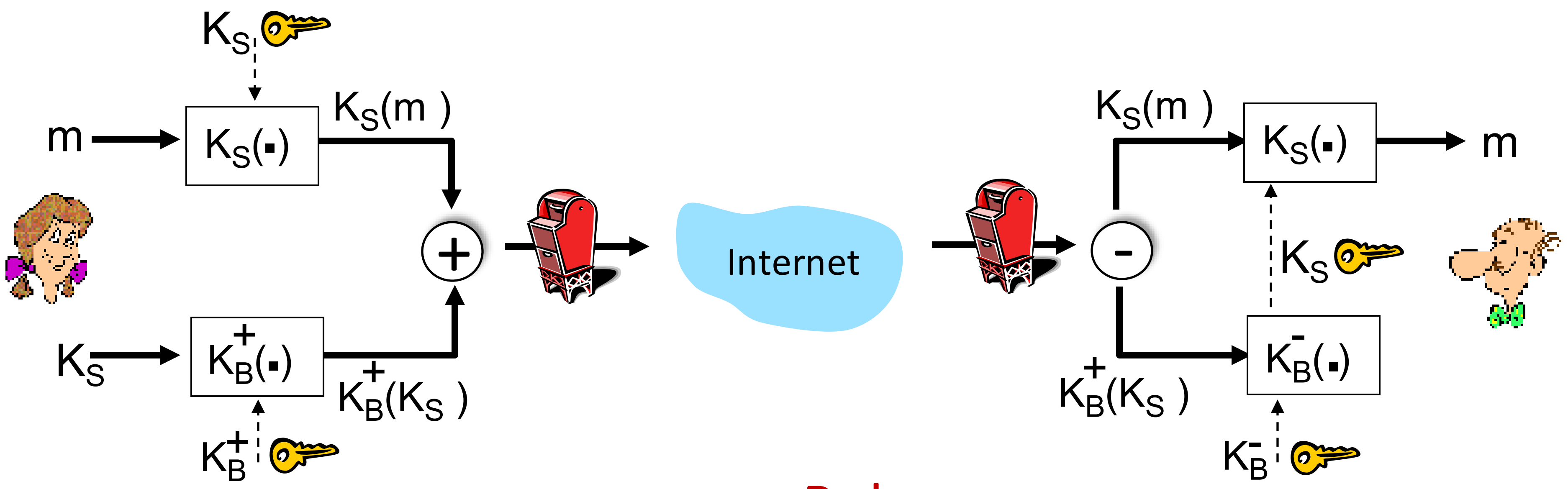


Alice:

- 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.

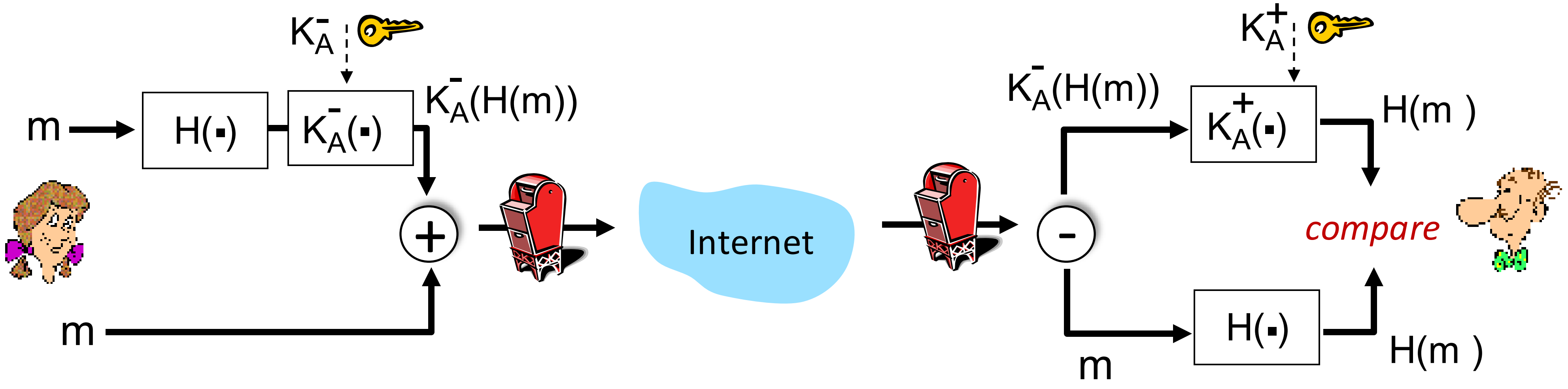


Bob:

- 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

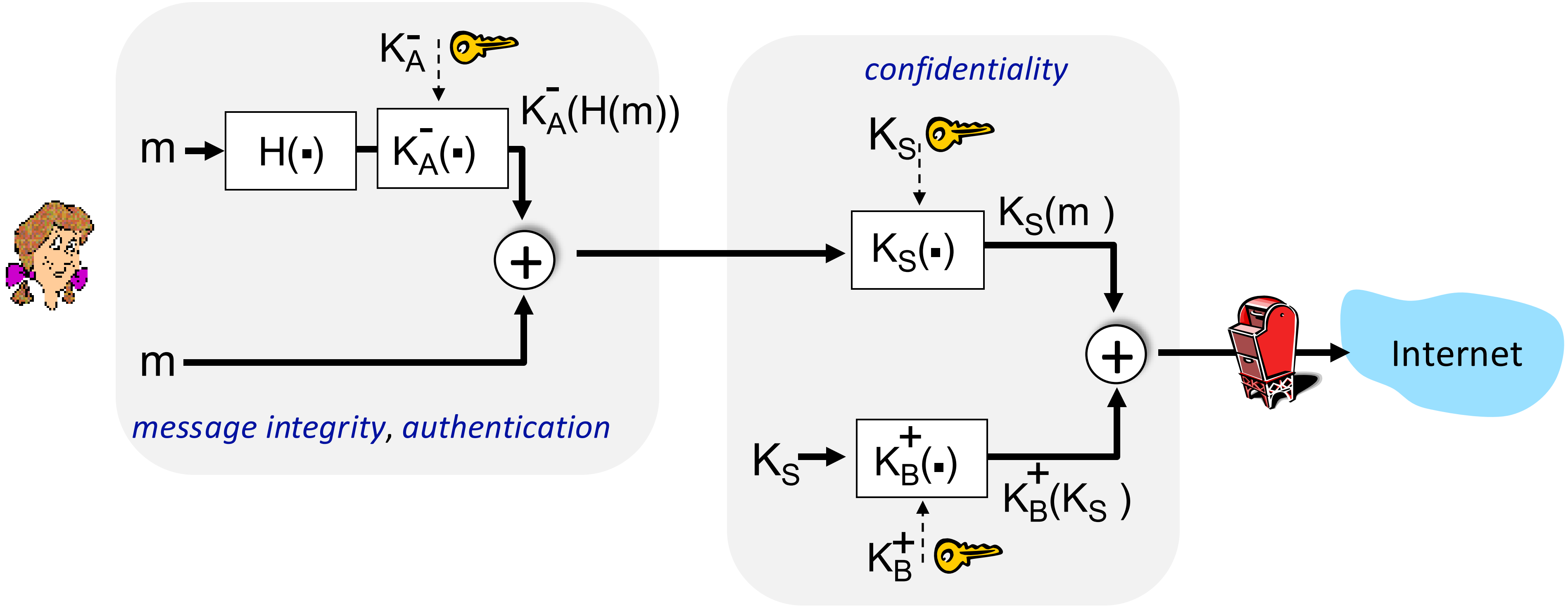
Alice wants to send m to Bob, with *message integrity*, *authentication*



- 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?