CS 305: Computer Networks

Fall 2022

Lecture 10： Network Layer – The Data Plane

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Chapter 3 outline

3.1 transport-layer services
3.2 multiplexing and demultiplexing
3.3 connectionless transport: UDP
3.4 principles of reliable data transfer
3.5 connection-oriented transport: TCP
- segment structure
- reliable data transfer
- flow control
- connection management

3.6 principles of congestion control
3.7 TCP congestion control

TCP congestion control: additive increase multiplicative decrease

Questions for achieving congestion control:

Q1: How does a TCP sender limit the rate at which it sends traffic into its connection?

Congestion window: **cwnd**

LastByteSent – LastByteAcked ≤ min{cwnd, rwnd}

Q2: How does a TCP sender perceive that there is congestion on the path between itself and the destination?

Timeout; three duplicate ACKs

Q3: What algorithm should the sender use to change its send rate as a function of perceived end-to-end congestion?

TCP congestion control: additive increase multiplicative decrease

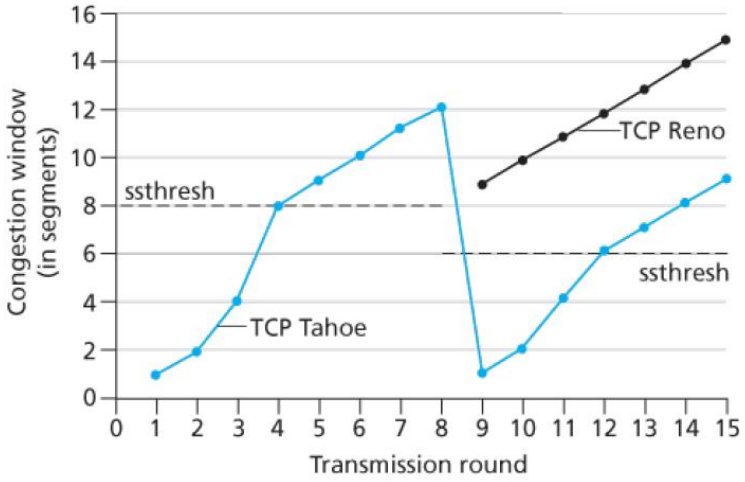
Q3: What algorithm should the sender use to change its send rate as a function of perceived end-to-end congestion?

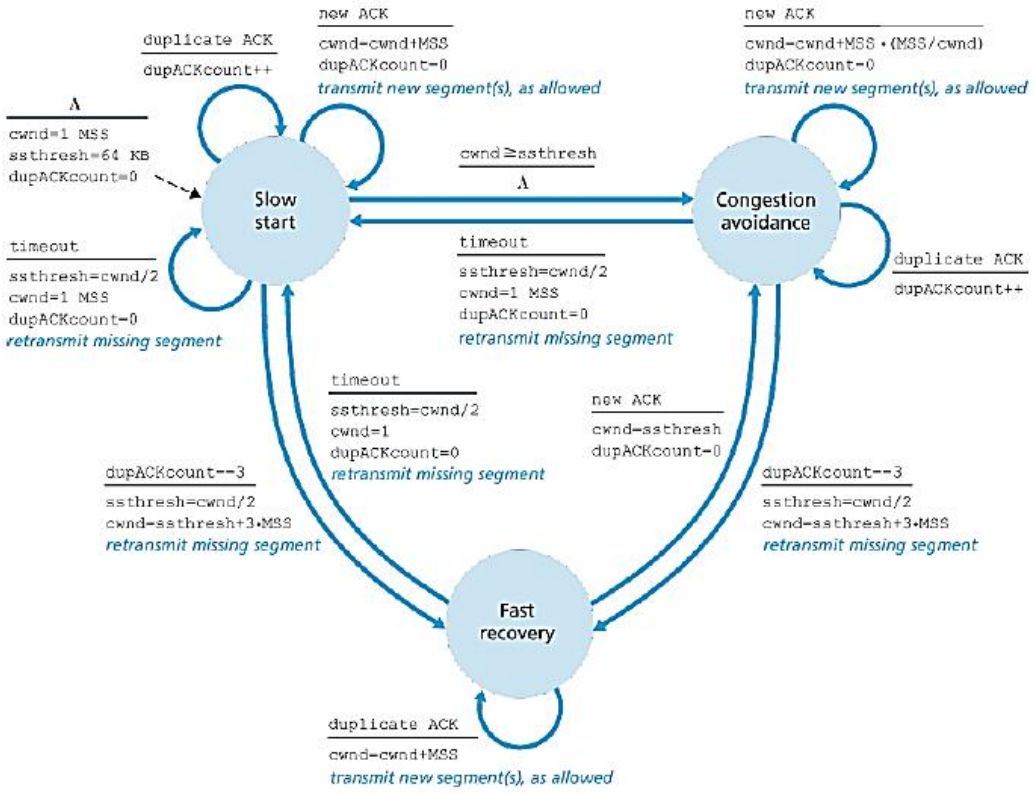
- A lost segment → congestion → decrease rate
- An acknowledged segment → the network is fine → increase rate
- Bandwidth **probing**: network condition may change

TCP Congestion Control: details

The congestion control algorithm has three major components:

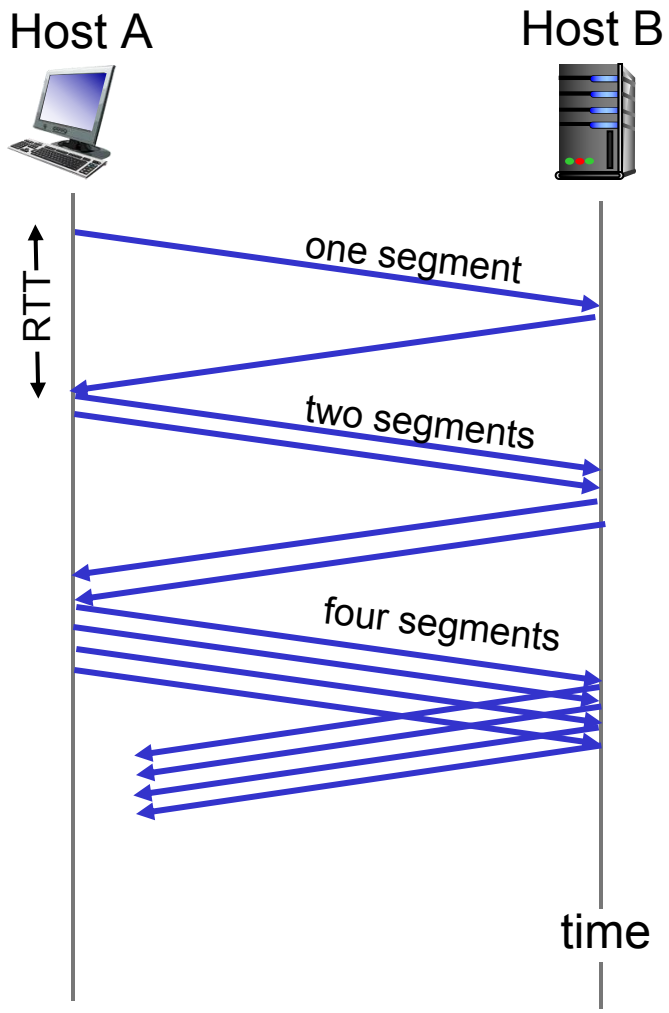
- Slow start: exponentially increase
- Congestion avoidance: linearly increase
- Fast recovery:



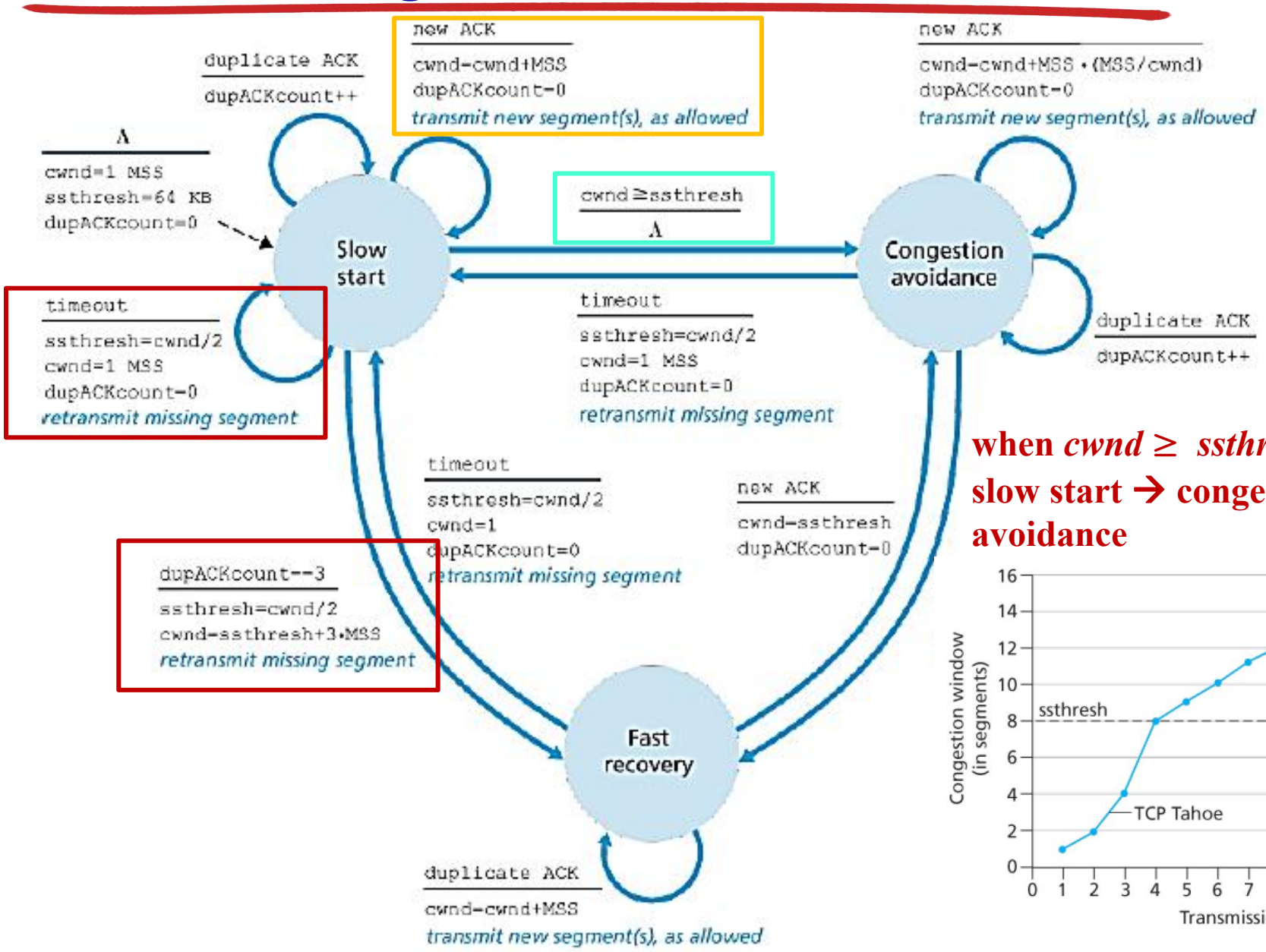


TCP Slow Start

- when connection begins or timeout occurs, increase rate exponentially until packet lost:
  - 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 Congestion Control: FSM



when *cwnd* ≥ *ssthresh:* slow start → congestion avoidance

TCP: detecting, reacting to loss

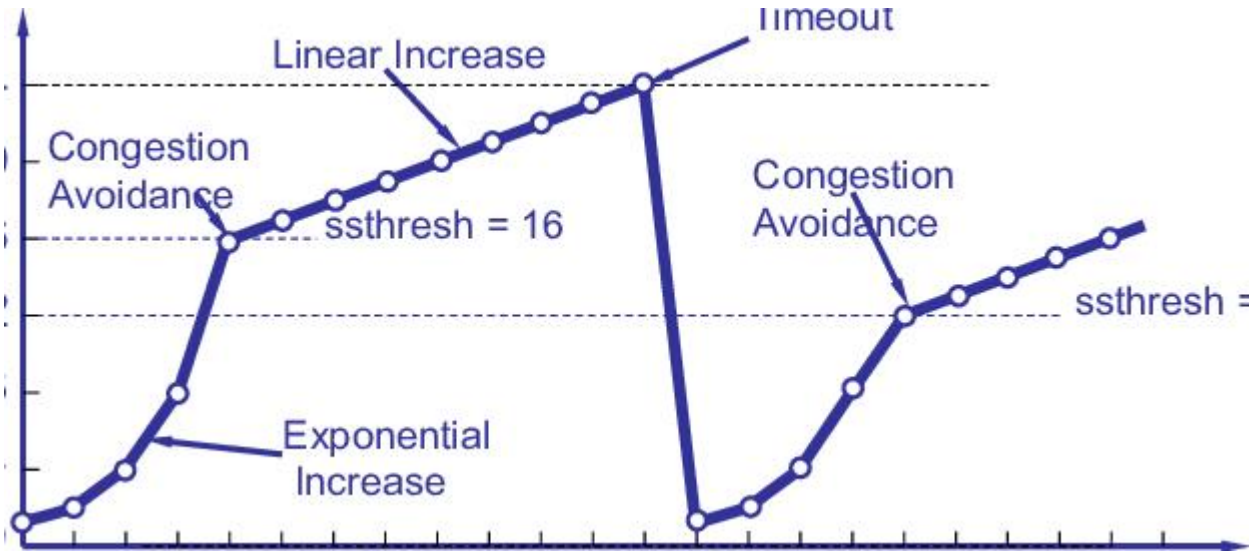
- loss indicated by timeout:
  - **cwnd** set to 1 MSS; **ssthresh** = **cwnd**/2
  - window then grows exponentially (as in slow start) to threshold, then grows linearly
- loss indicated by 3 duplicate ACKs:
  - TCP Tahoe always sets **cwnd** to 1 (timeout or 3 duplicate acks) → Slow start
    - **cwnd** set to 1 MSS; **ssthresh** = **cwnd**/2
  - TCP RENO
    - dup ACKs indicate network capable of delivering some segments → Fast Recovery
    - ssthresh = cwnd / 2; cwnd = ssthresh + 3MSS

TCP: switching 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

Thus, when timeout occurs, ssthresh = cwnd/2

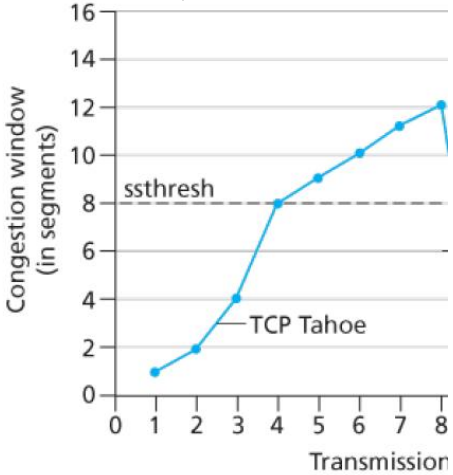


TCP Congestion Avoidance

Trigger: *cwnd* ≥ ssthresh

Increases *cwnd* linearly: by one MSS every RTT

- Increase *cwnd* by (MSS/*cwnd*)MSS bytes whenever a new acknowledgment arrives.
- E.g., if MSS is 1,460 bytes and *cwnd* is 14,600 bytes, then 10 segments are being sent within an RTT.
  - Each arriving ACK (assuming one ACK per segment) increases the congestion window size by 1/10 MSS,



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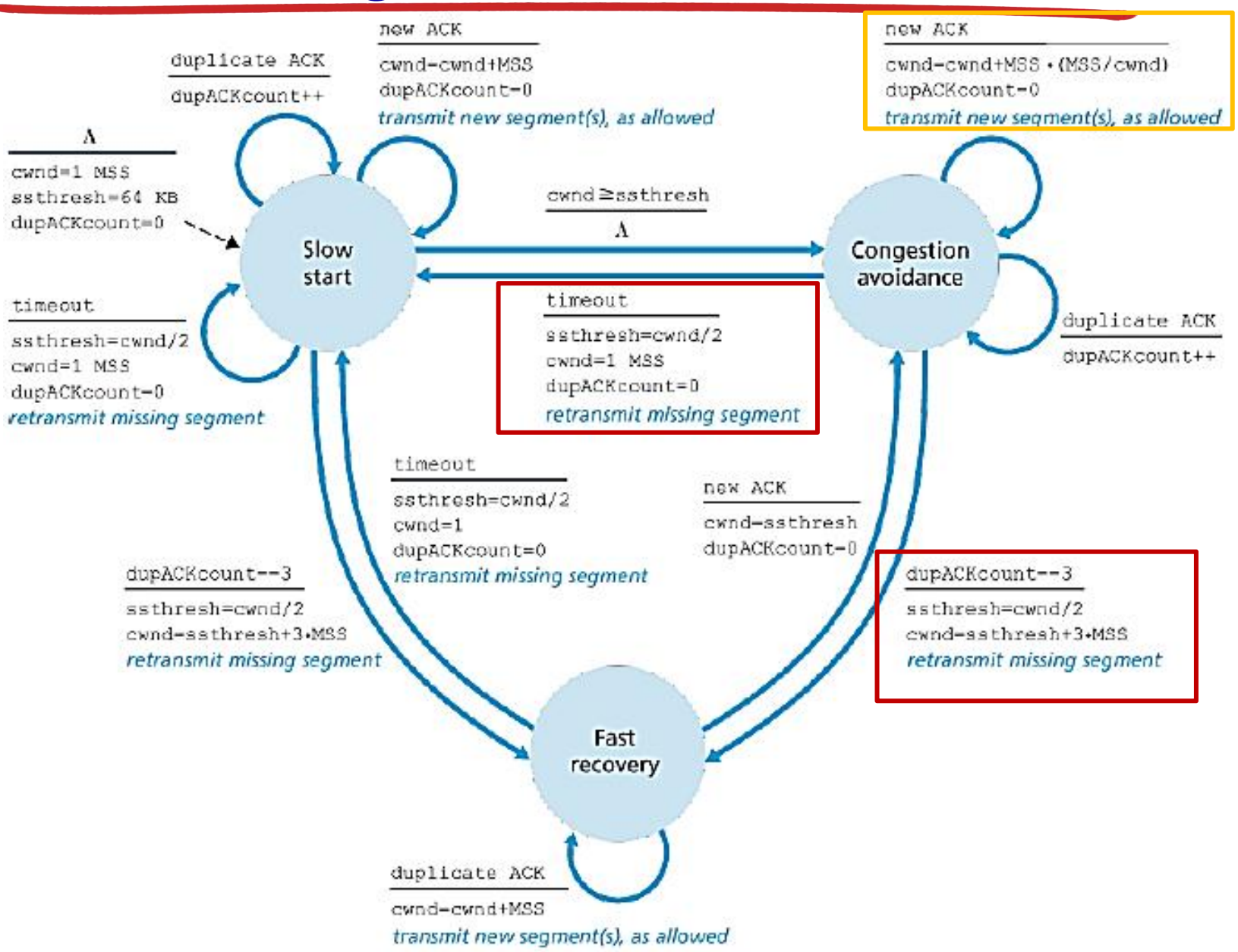
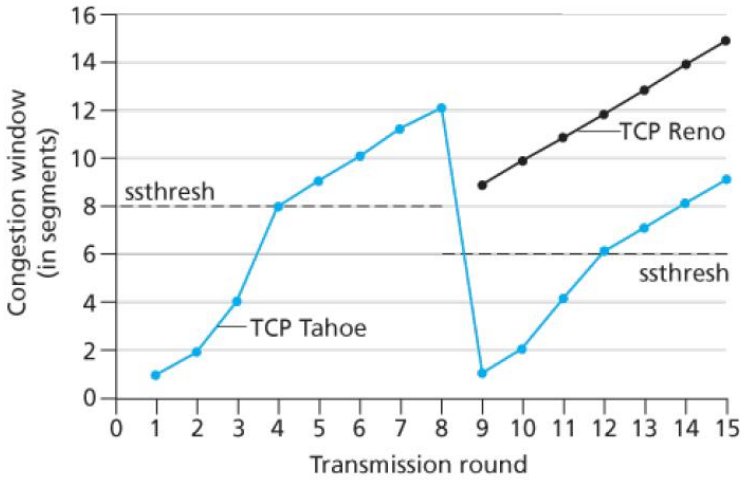


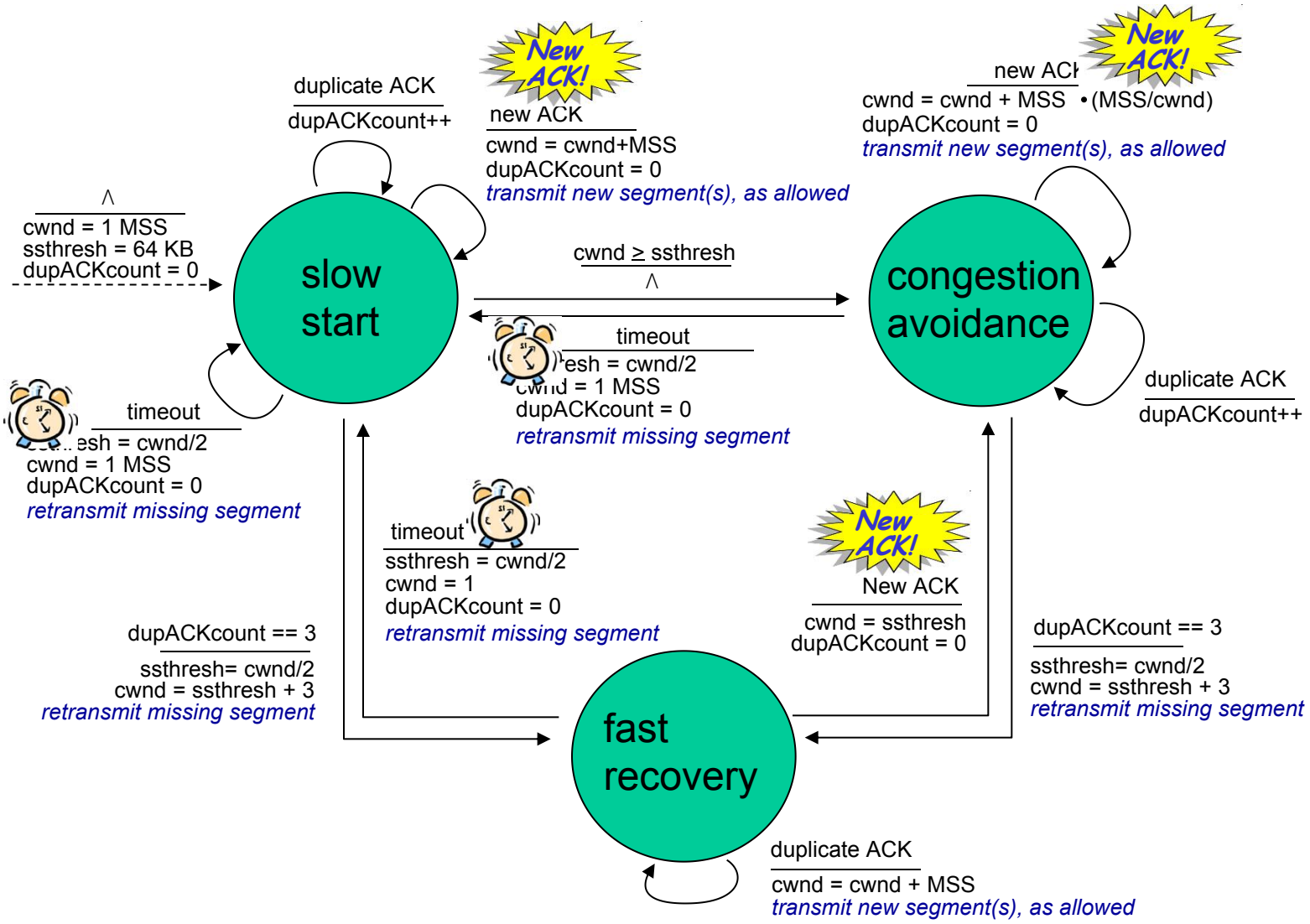
TCP Fast Recovery

- Trigger （RENO）： triple duplicate ACKs
- ***ssthresh = cwnd / 2; cwnd = ssthresh + 3MSS***

- The value of *cwnd* is increased by 1 MSS for every duplicate ACK received for the missing segment that caused TCP to enter the fast-recovery state
- when an ACK arrives for the missing segment, enter congestion avoidance



Summary: TCP Congestion Control



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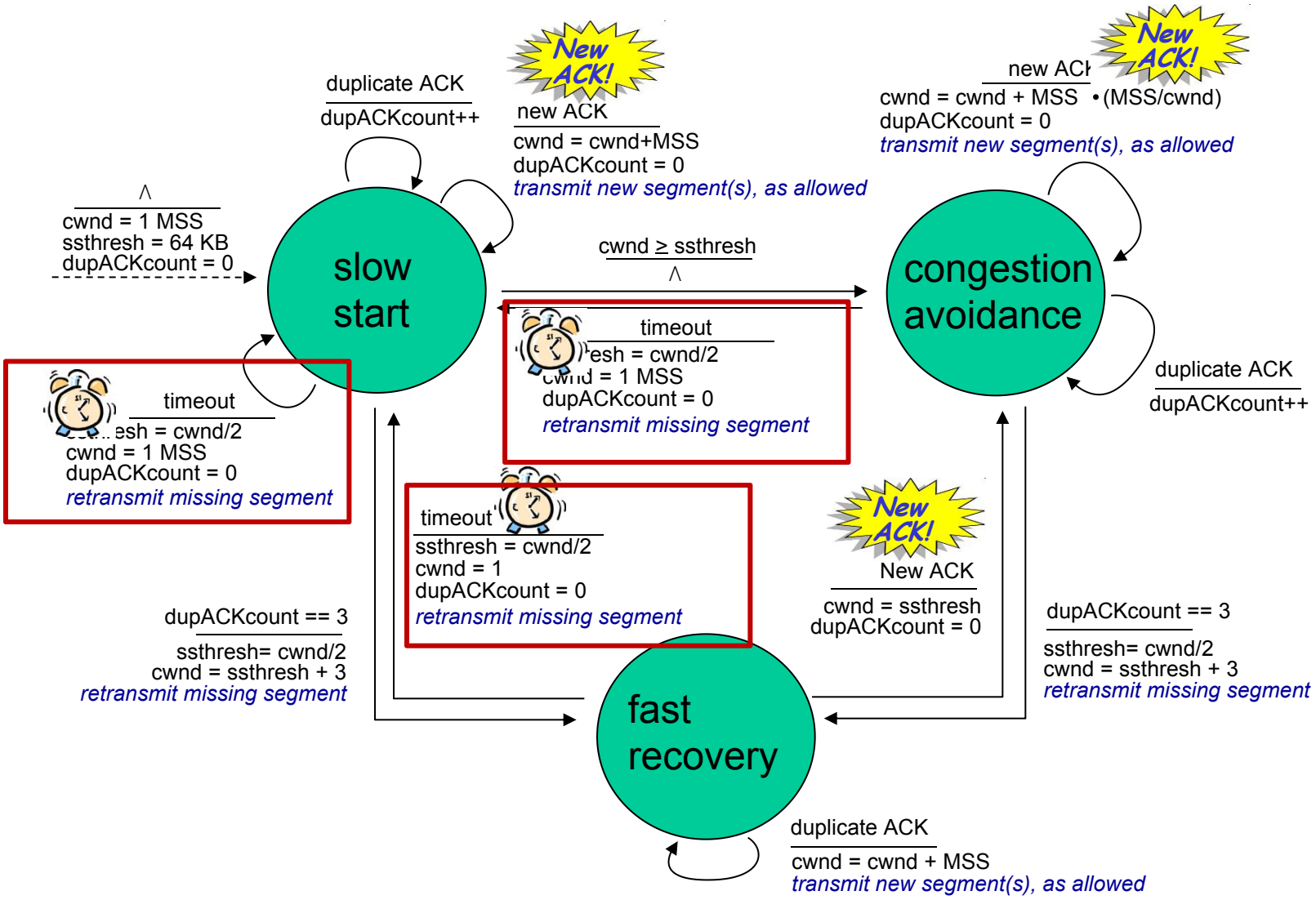

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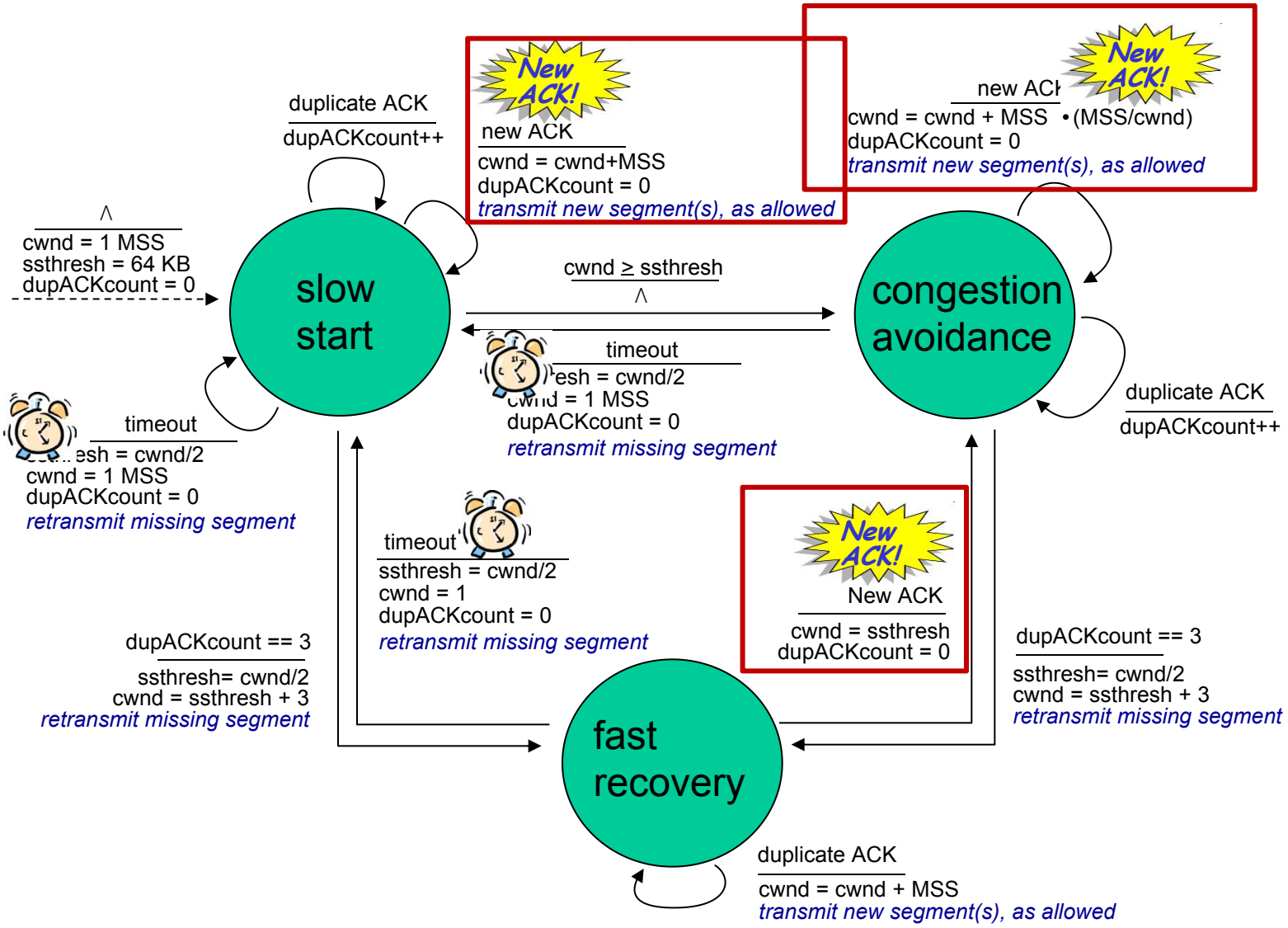

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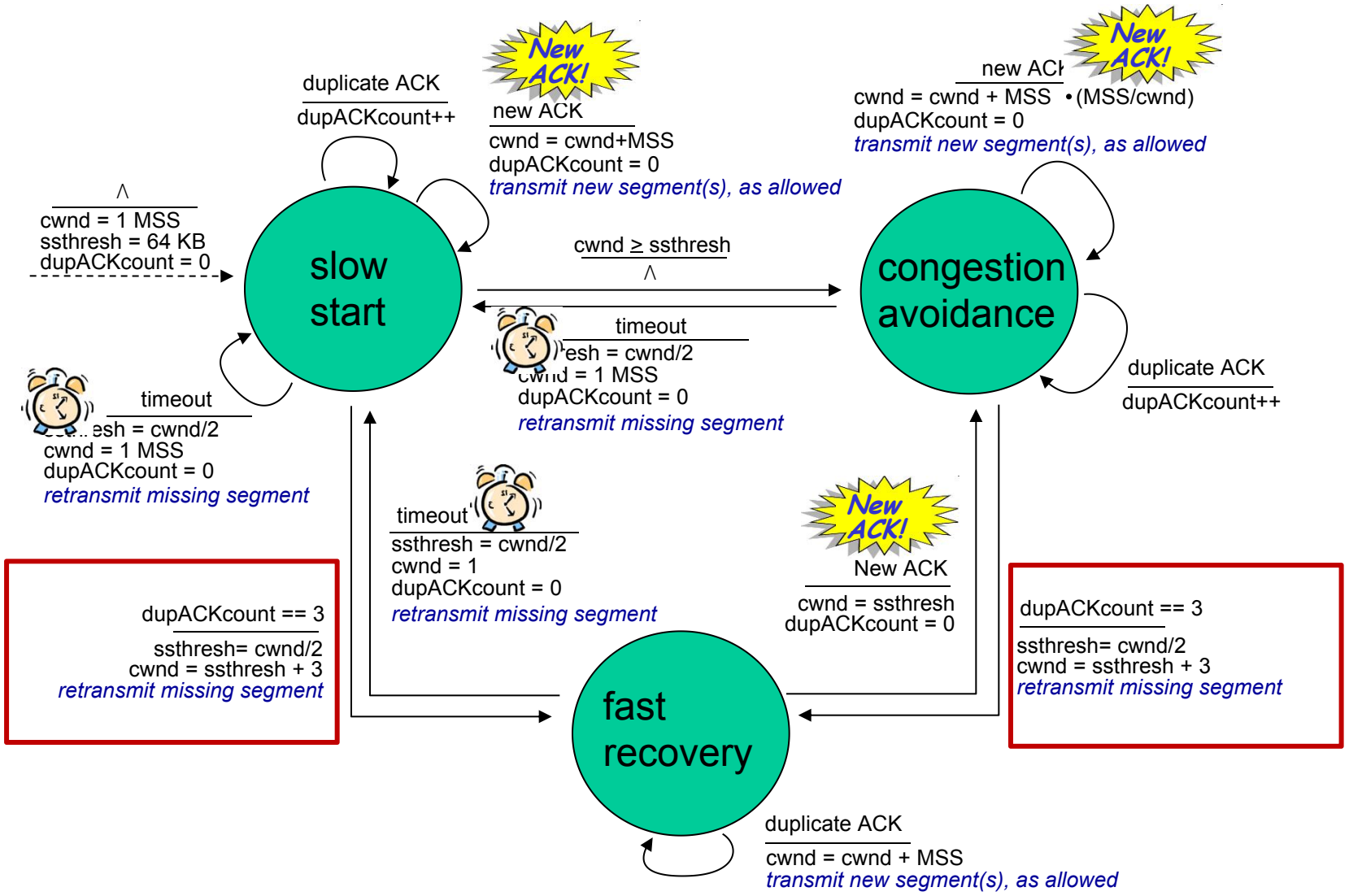
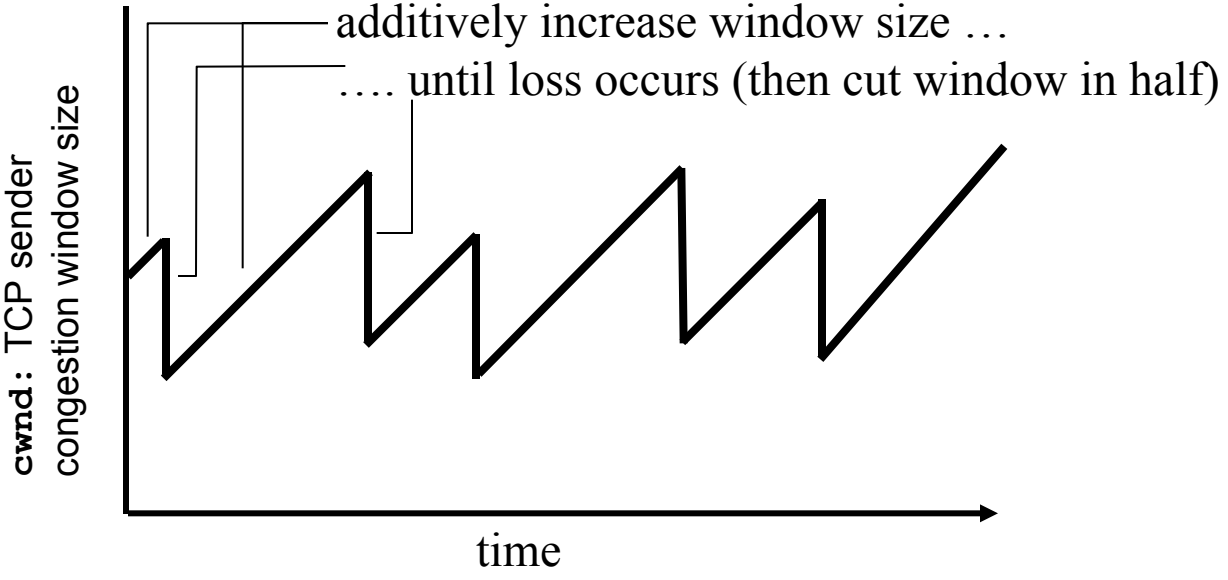


TCP congestion control: additive increase multiplicative decrease

Approach: sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs

- *additive increase:* increase **cwnd** by 1 MSS every RTT until loss detected
- *multiplicative decrease*: cut **cwnd** in half after loss

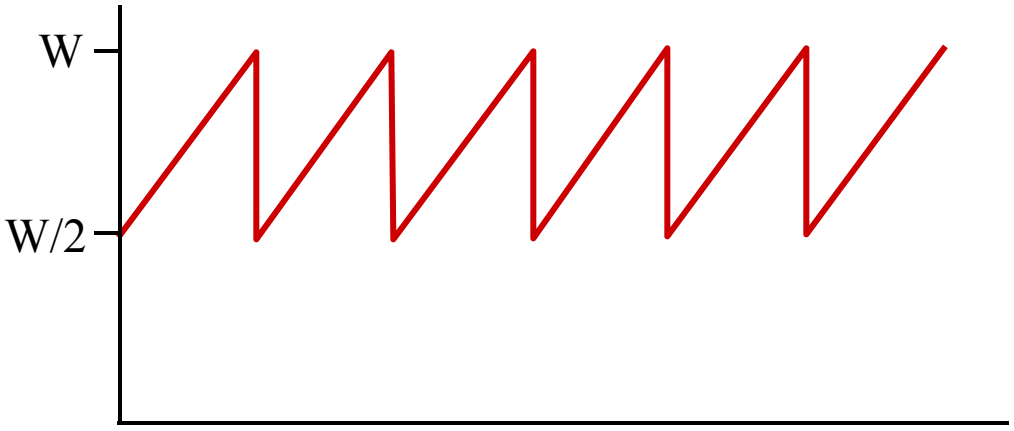
AIMD: probing for bandwidth



TCP throughput

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

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



TCP Futures: TCP over "long, fat pipes"

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

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

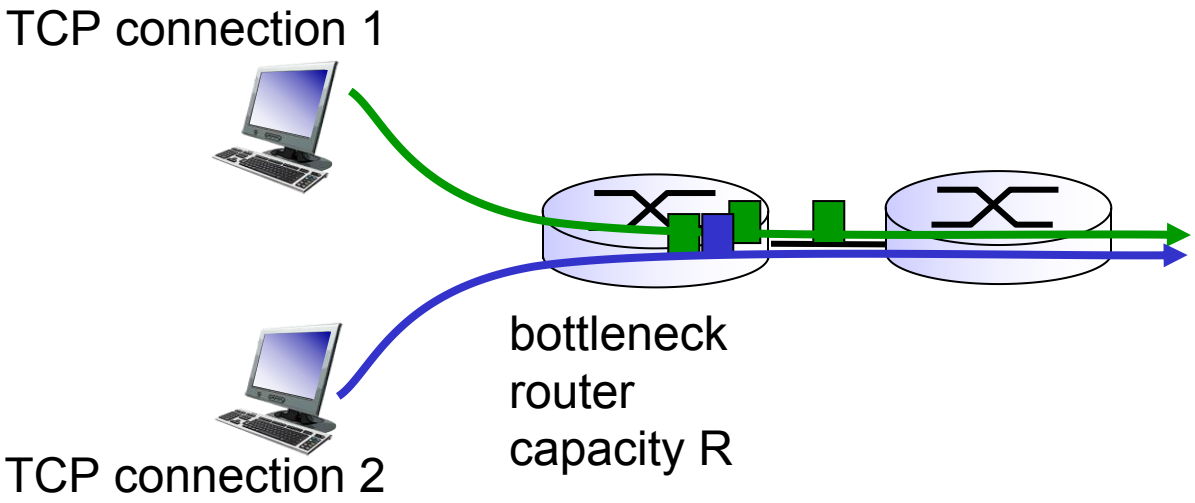
$$\text{TCP throughput} = \frac{1.22 \cdot MSS}{RTT\sqrt{L}}$$

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

- new versions of TCP for high-speed

TCP Fairness

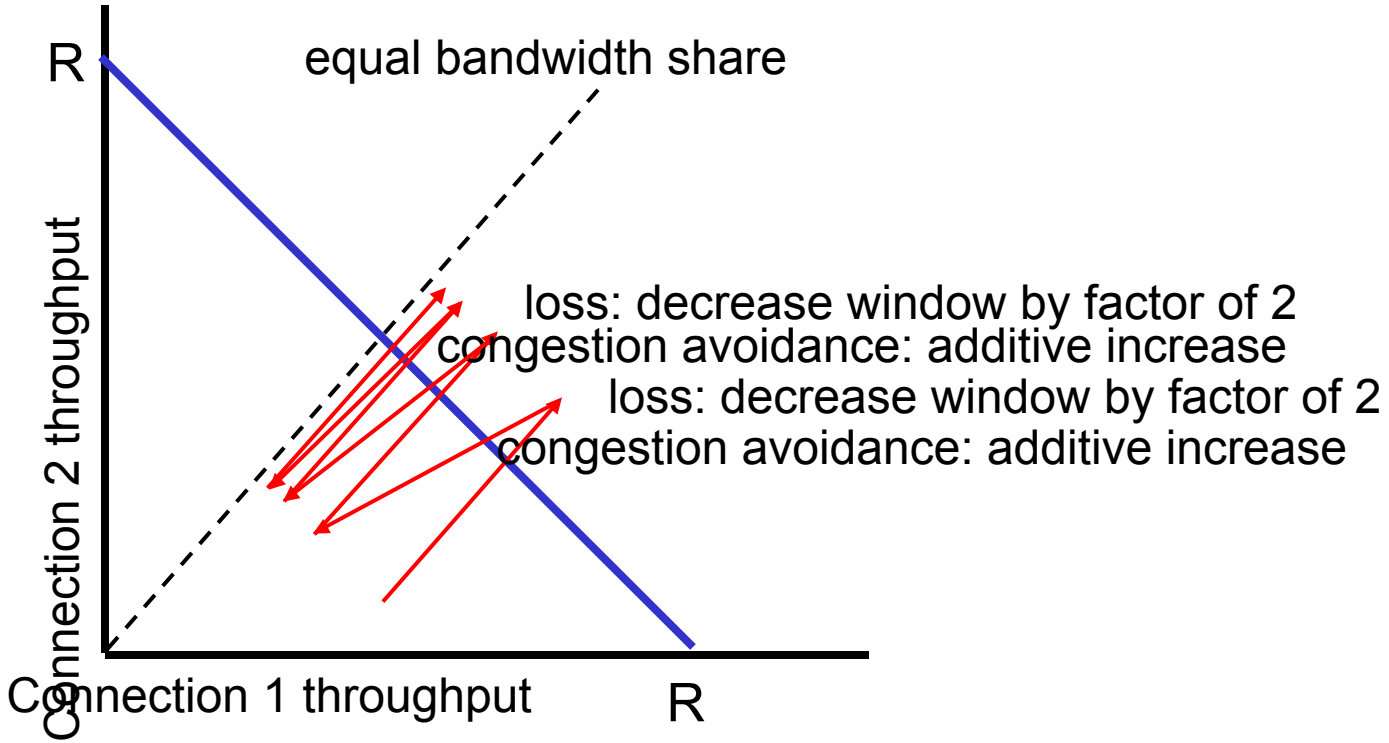
Fairness goal: if *K* TCP sessions share same bottleneck link of bandwidth *R*, each should have average rate of *R/K*



Why is TCP fair?

two competing sessions:

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



Fairness (more)

Fairness and UDP

- multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- use UDP:
  - send audio/video at constant rate, tolerate packet loss
- UDP sources to crowd out TCP traffic

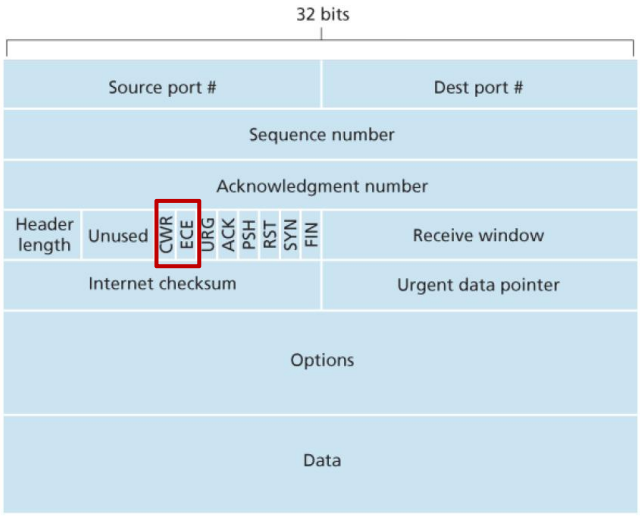
Fairness, parallel TCP connections

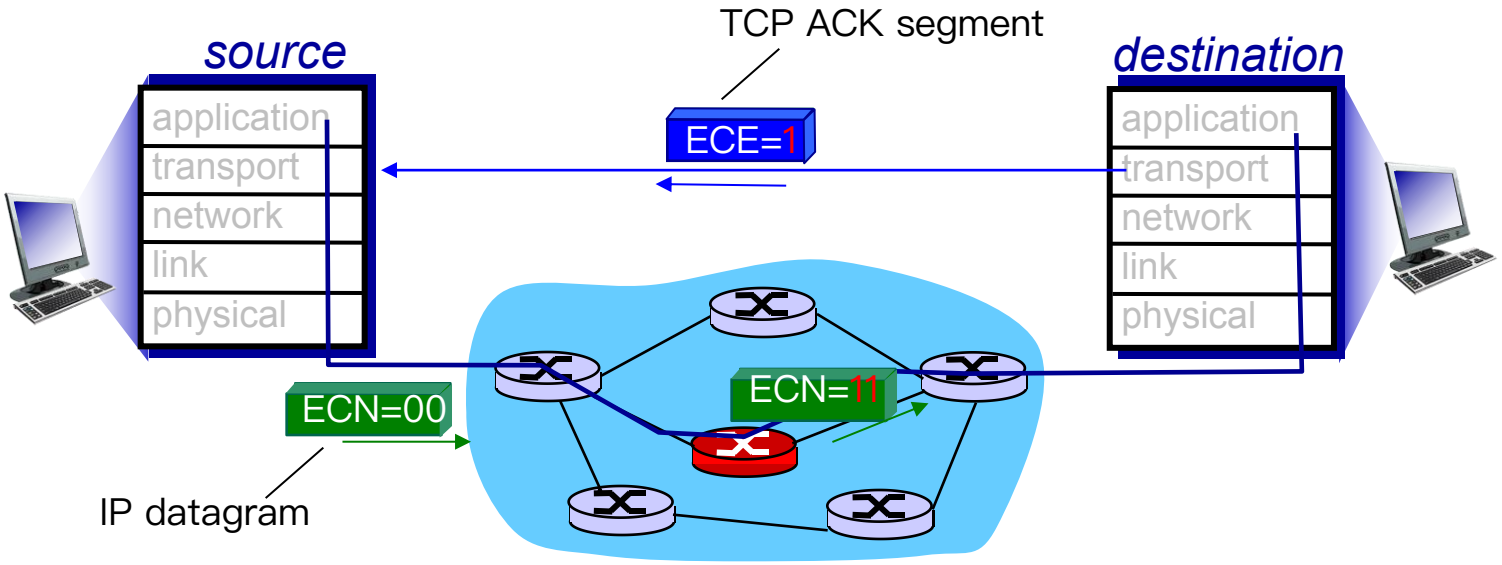
- application can open multiple parallel connections between two hosts
- web browsers do this
- e.g., link of rate R with 9 existing connections:
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets more than R/2

Explicit Congestion Notification (ECN)

network-assisted congestion control:

- two bits in IP header (ToS field) marked *by network router* to indicate congestion
- congestion indication carried to receiving host
- receiver sets ECE bit on receiver-to-sender ACK segment to notify sender of congestion





Chapter 3: summary

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

next:

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

Chapter 4: outline

4.1 Overview of Network layer
- data plane
- control plane

4.2 What's inside a router

4.3 IP: Internet Protocol
- datagram format
- fragmentation
- IPv4 addressing
- network address translation
- IPv6

4.4 Generalized Forward and SDN
- match
- action
- OpenFlow examples of match-plus-action in action

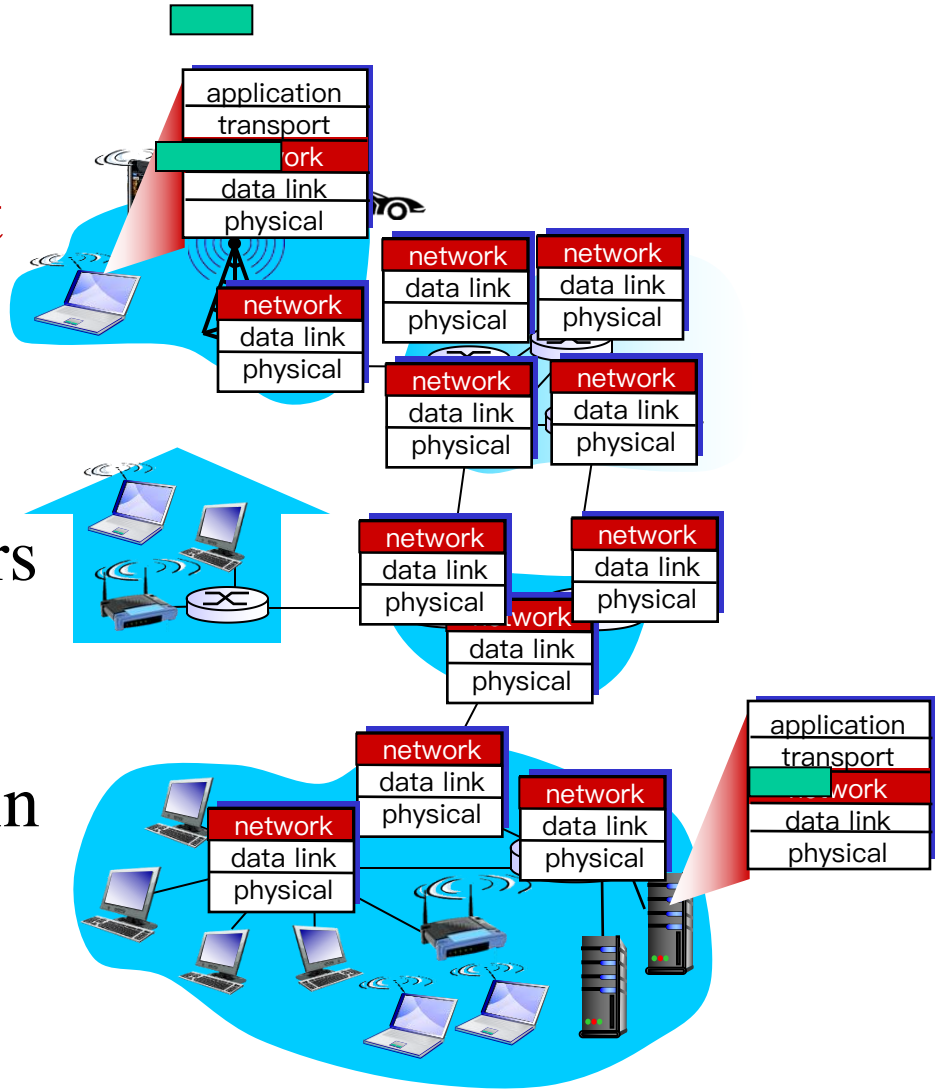
Chapter 4: network layer

Chapter goals:

- Understand principles behind network layer services, focusing on data plane:
  - network layer service models
  - forwarding versus routing
  - how a router works
  - generalized forwarding
- Instantiation, implementation in the Internet

Network layer

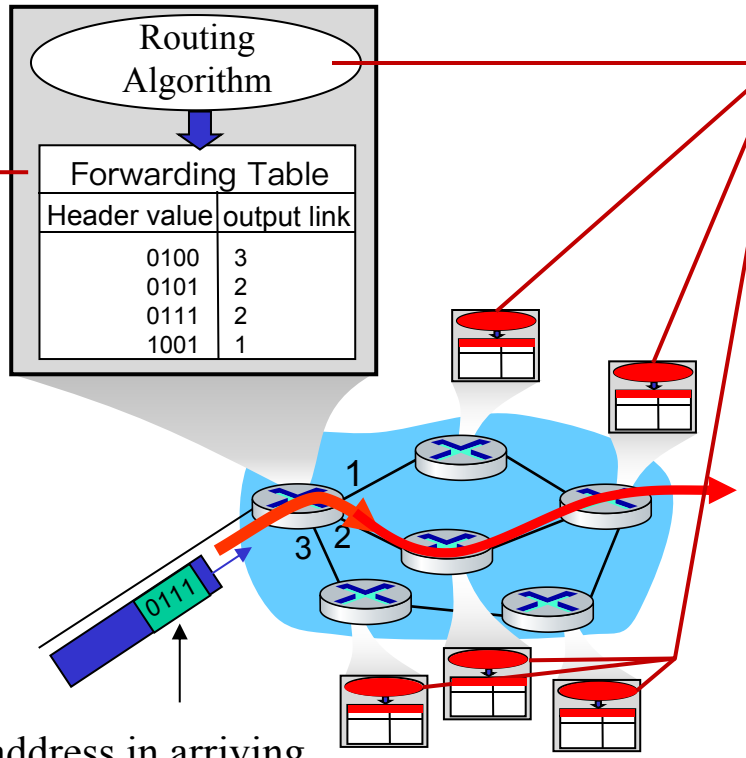
- transport segment from sending to receiving host
- on sending side encapsulates segments into datagrams
- on receiving side, delivers segments to transport layer
- network layer protocols in *every* host, router
- router examines header fields in all IP datagrams passing through it



Two key network-core functions

Forwarding:

local action: move arriving packets from router's input link to appropriate router output link



destination address in arriving packet's header

Routing:

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

Two key network-layer functions

Network-layer functions:

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

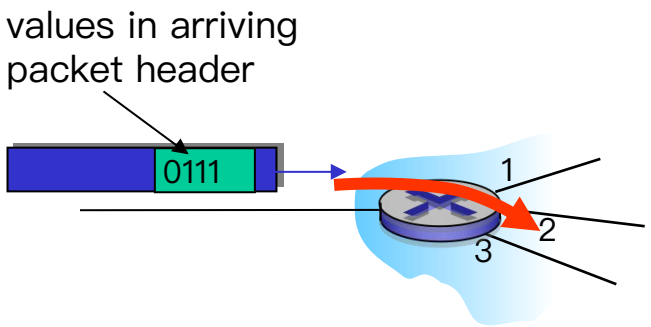
Analogy: taking a trip

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

Network layer: data plane, control plane

Data plane

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



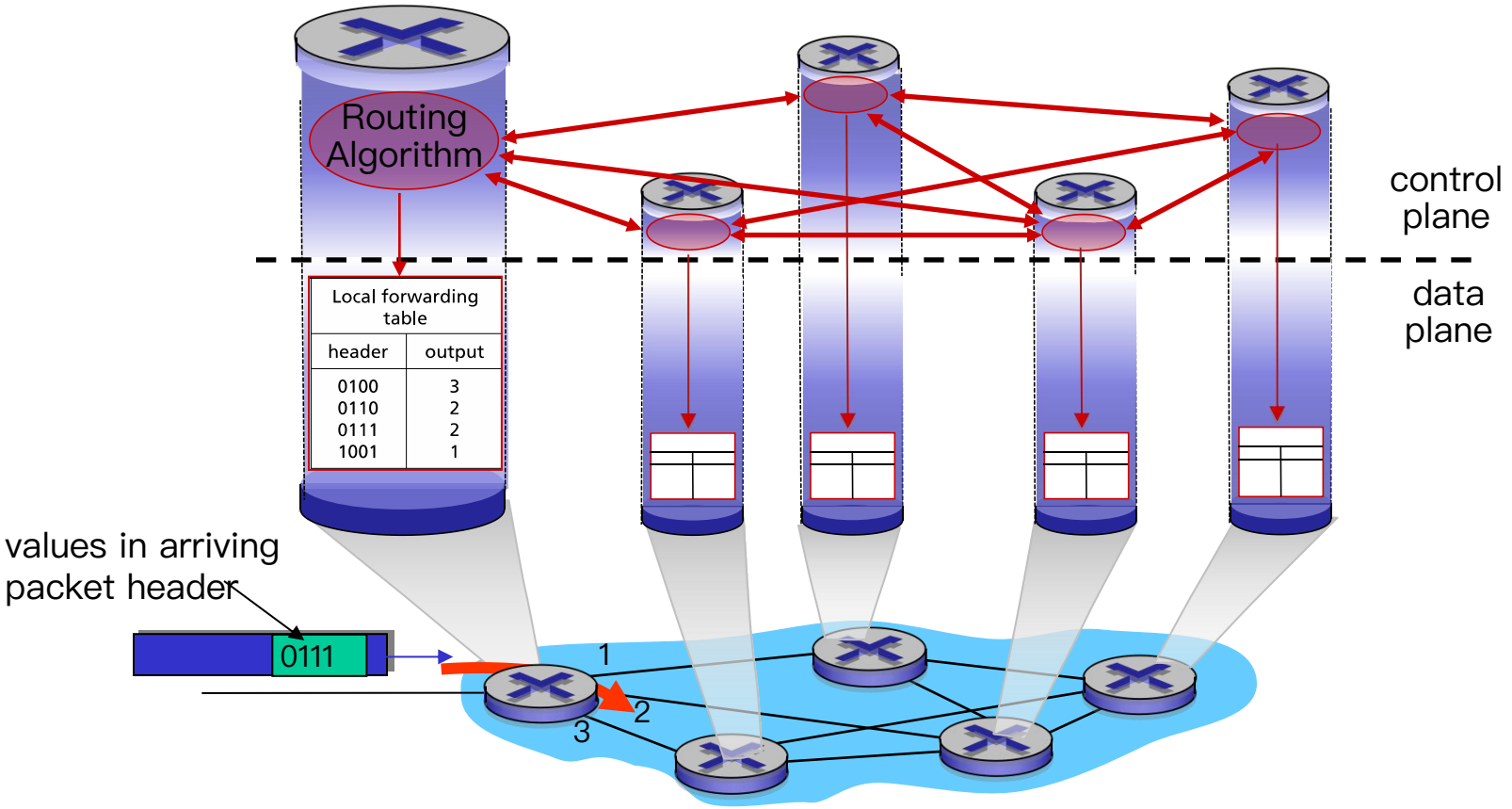
Control plane

- network-wide logic, software
- 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

Control plane: Traditional Approach

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

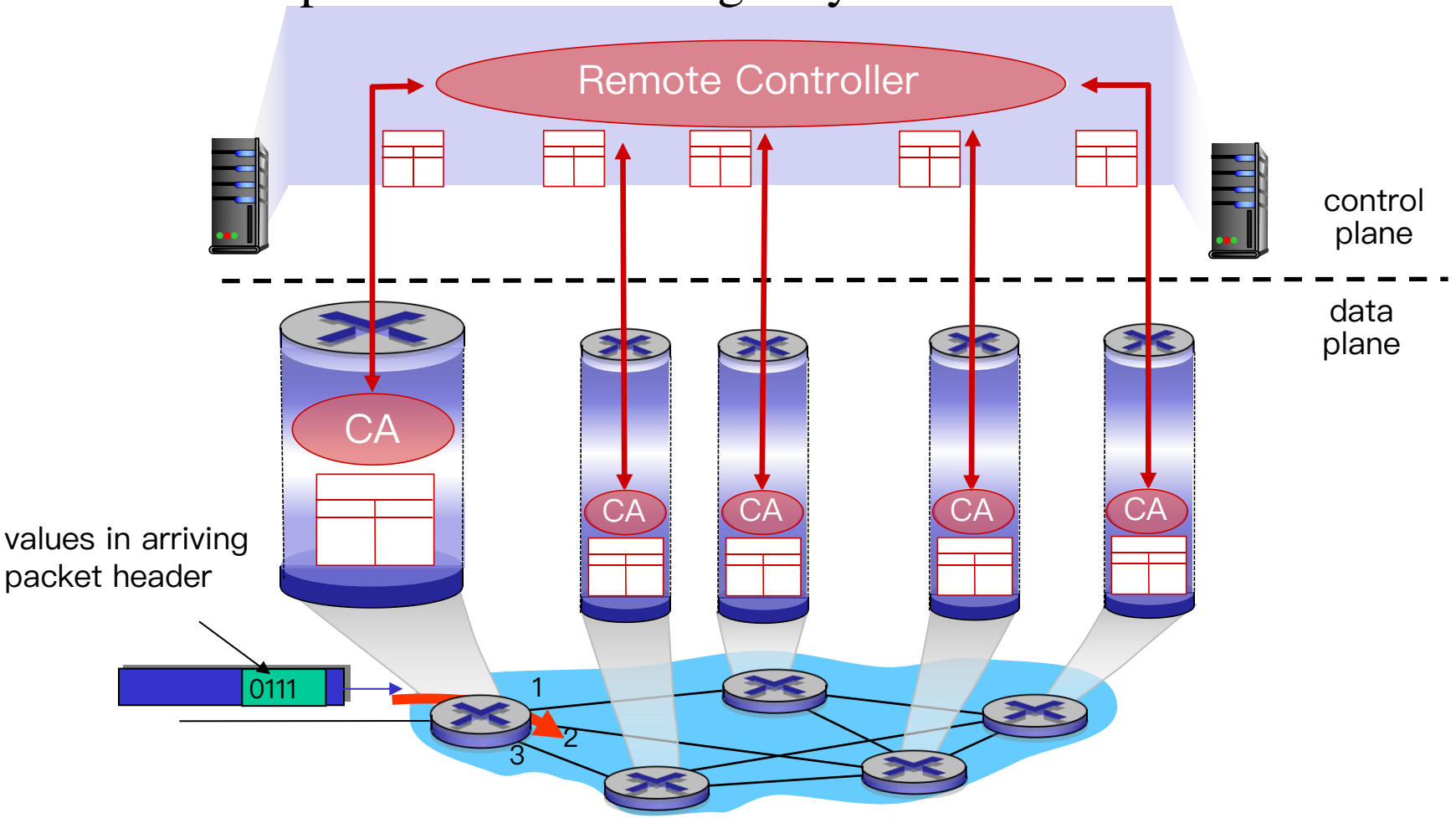
- Forwarding and routing functions are contained within a router



Control Plane: The SDN Approach

Logically centralized control plane: A distinct (typically remote) controller interacts with local control agents (CAs)

- Router performs forwarding only



Network service model

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

example services for individual datagrams:

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

example services for a flow of datagrams:

- in-order datagram delivery
- guaranteed minimum bandwidth to flow

other example services: security

Internet service model provide "best effort" service, no guarantee on bandwidth, loss, order or timing.

Chapter 4: outline

4.1 Overview of Network layer
- data plane
- control plane

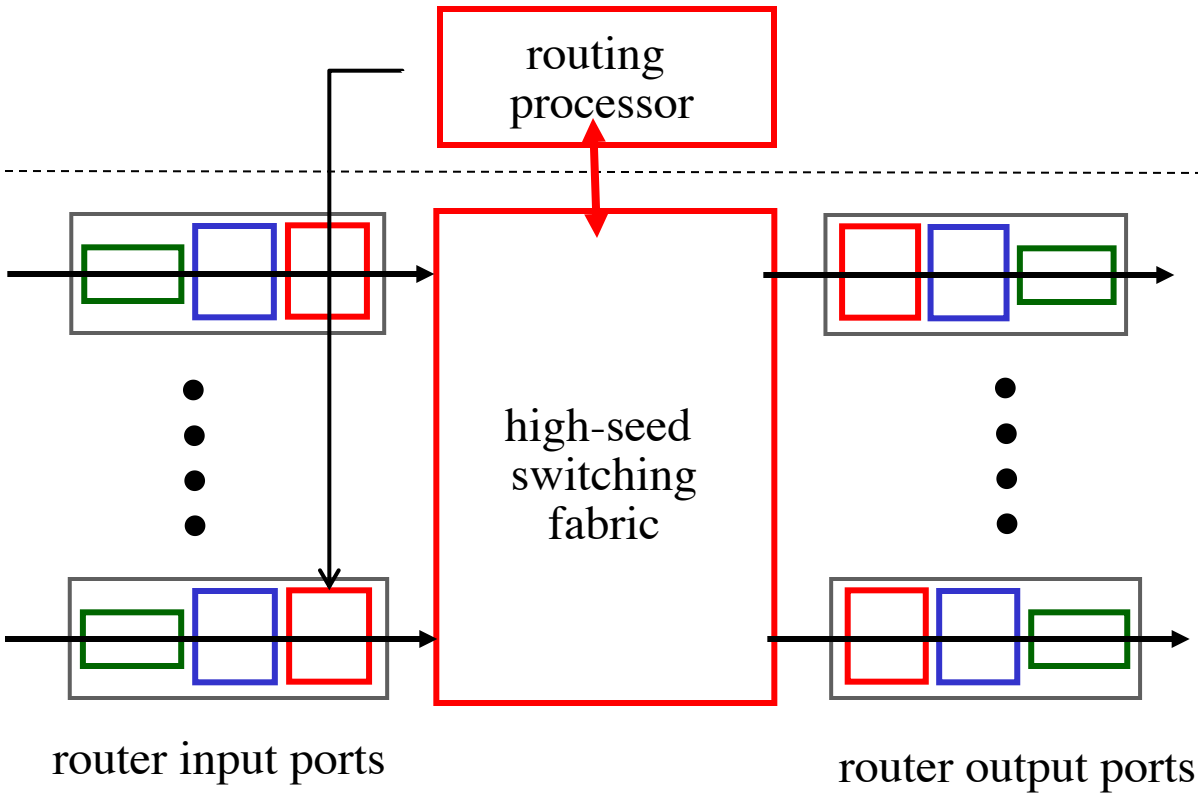
4.2 What's inside a router

4.3 IP: Internet Protocol
- datagram format
- fragmentation
- IPv4 addressing
- network address translation
- IPv6

4.4 Generalized Forward and SDN
- match
- action
- OpenFlow examples of match-plus-action in action

Router architecture overview

High-level view of generic router architecture:

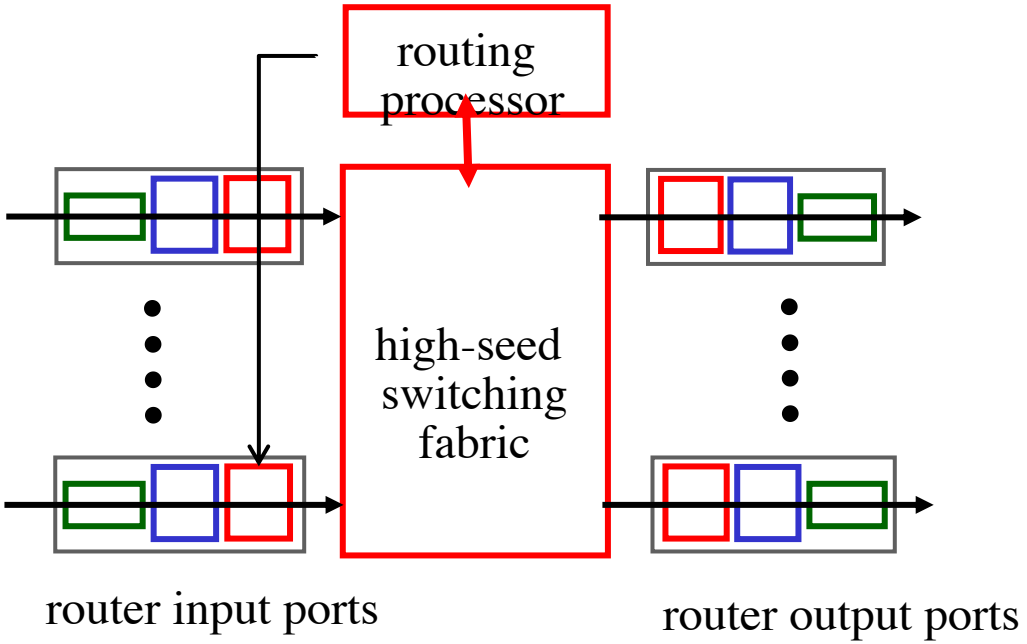


routing, management control plane (software) operates in millisecond time frame

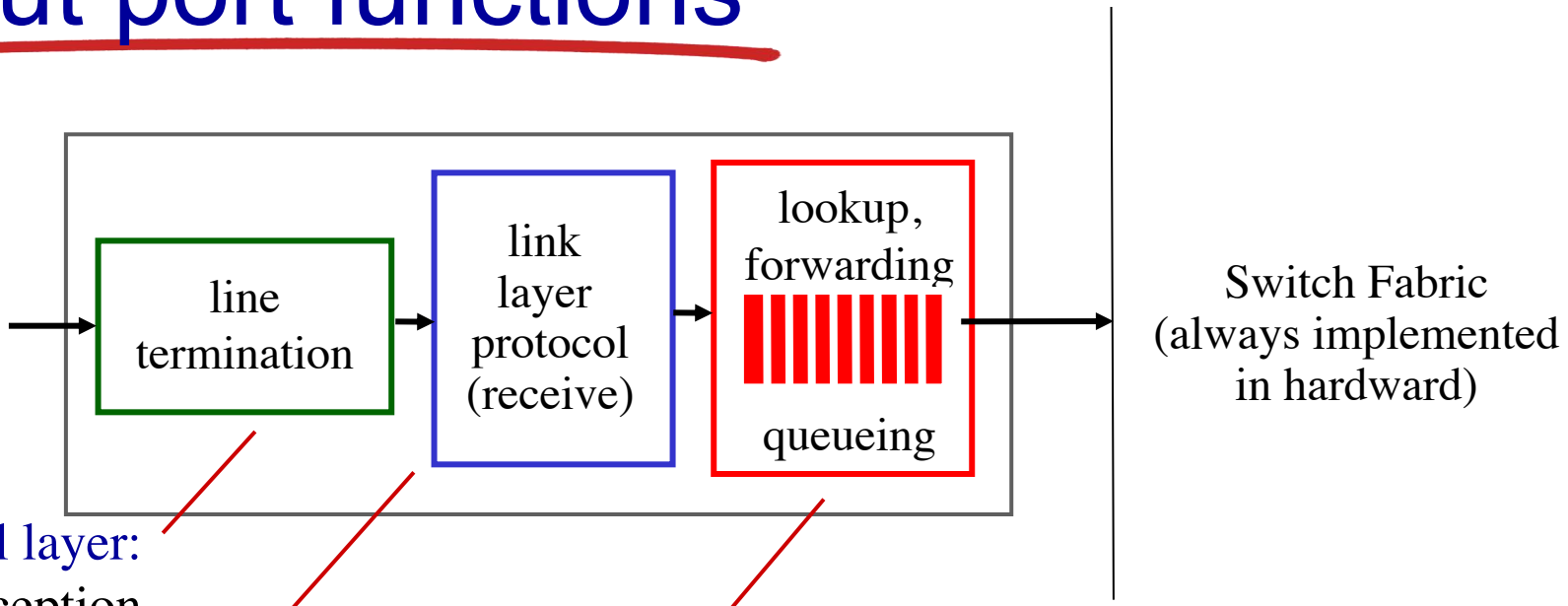
forwarding data plane (hardware) operates in nanosecond timeframe

Router Overview

- Input port
- Switch fabrics
- Output port
- Queuing
  - Input port queue
  - Output port queue
  - Scheduling



Input port functions



physical layer:
bit-level reception

link layer:
e.g., Ethernet
see chapter 5

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'
- queuing: if datagrams arrive faster than forwarding rate into switch fabric

Destination-based forwarding

forwarding table

Destination Address Range	Link Interface
11001000 00010111 00010000 00000000 through 11001000 00010111 00010111 11111111	0
11001000 00010111 00011000 00000000 through 11001000 00010111 00011000 11111111	1
11001000 00010111 00011001 00000000 through 11001000 00010111 00011111 11111111	2
otherwise	3

Q: but what happens if ranges don't divide up so nicely (i.e., overlap between entities)?

Longest prefix matching

longest prefix matching

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

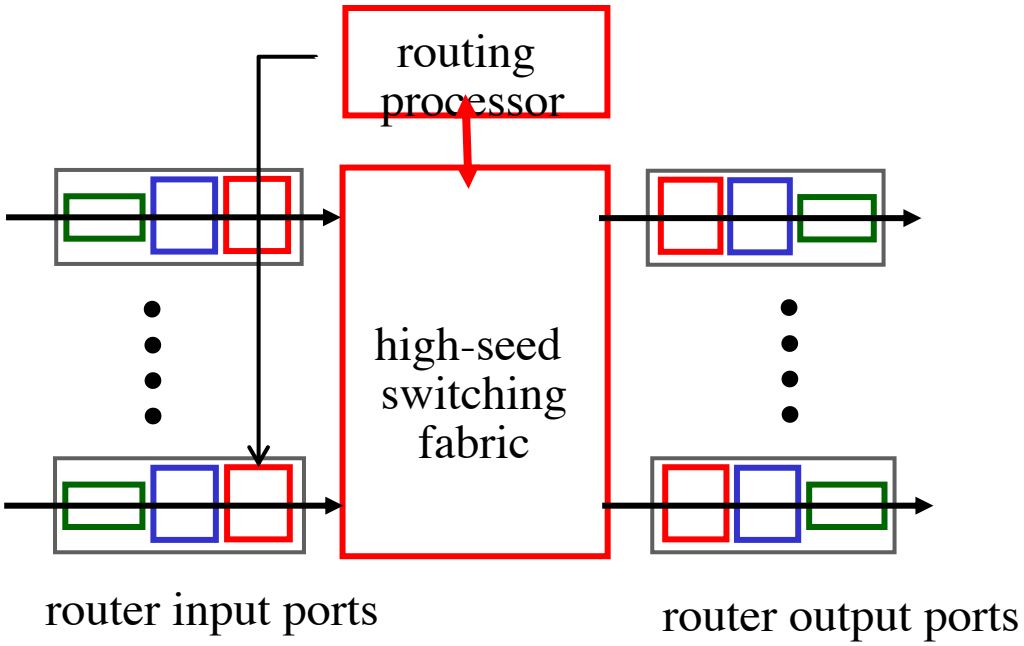
examples:

DA: 11001000 00010111 00010110 10100001 which interface?

DA: 11001000 00010111 00011000 10101010 which interface?

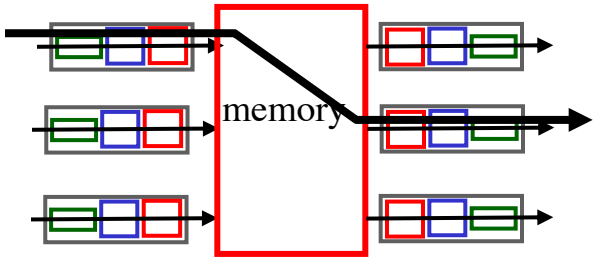
Router Overview

- Input port
- Switching fabrics
- Output port
- Queuing
  - Input port queue
  - Output port queue
  - Scheduling



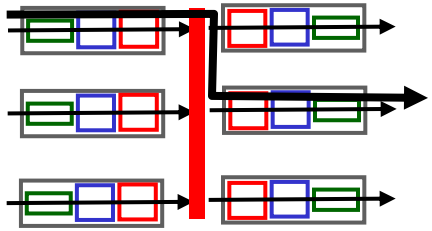
Switching fabrics

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



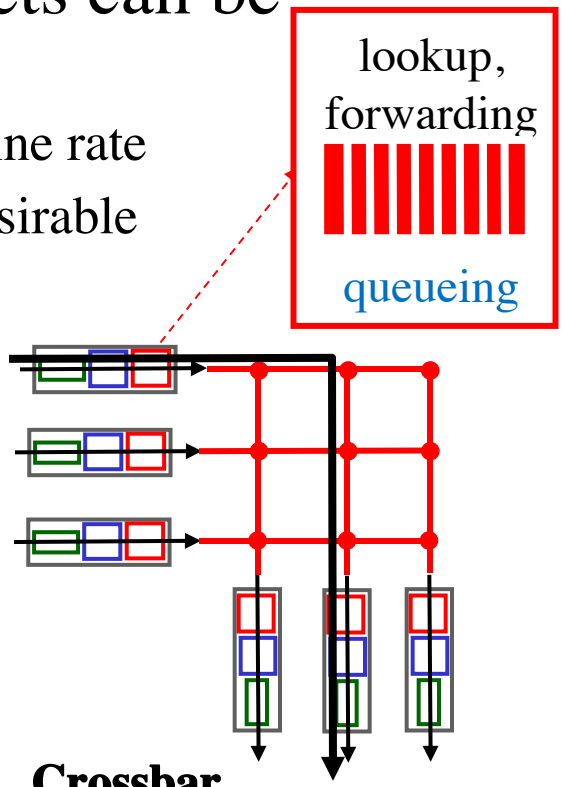
Switch via memory

- Interrupt; write and read
- Two packets cannot be forwarded at the same time



Switch via bus

- Broadcast; label
- One packet can cross at a time



Crossbar

- Multiple packets in parallel
- Non-blocking

Router Overview

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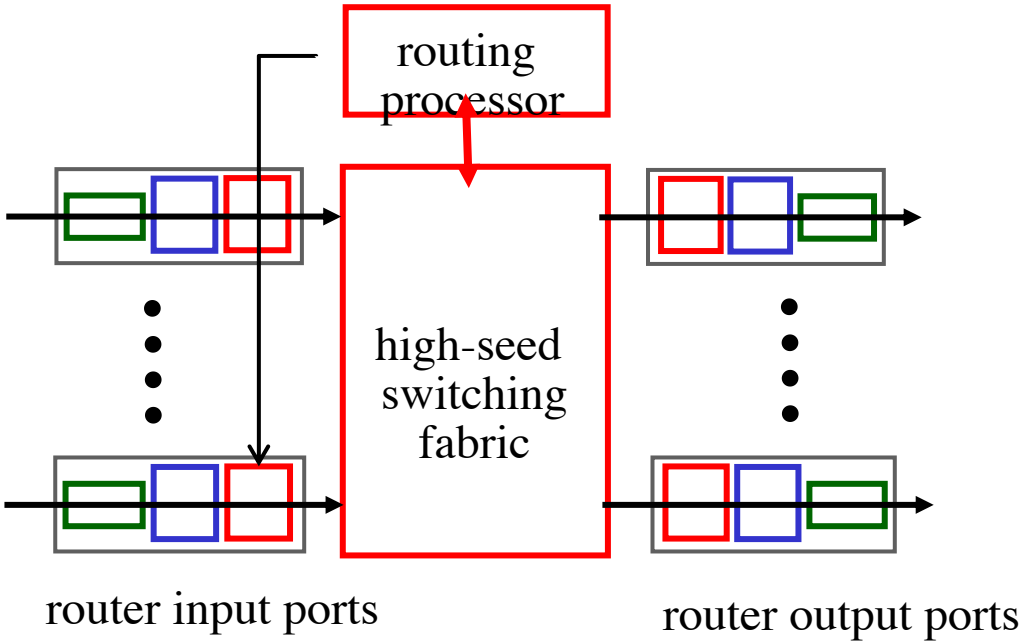
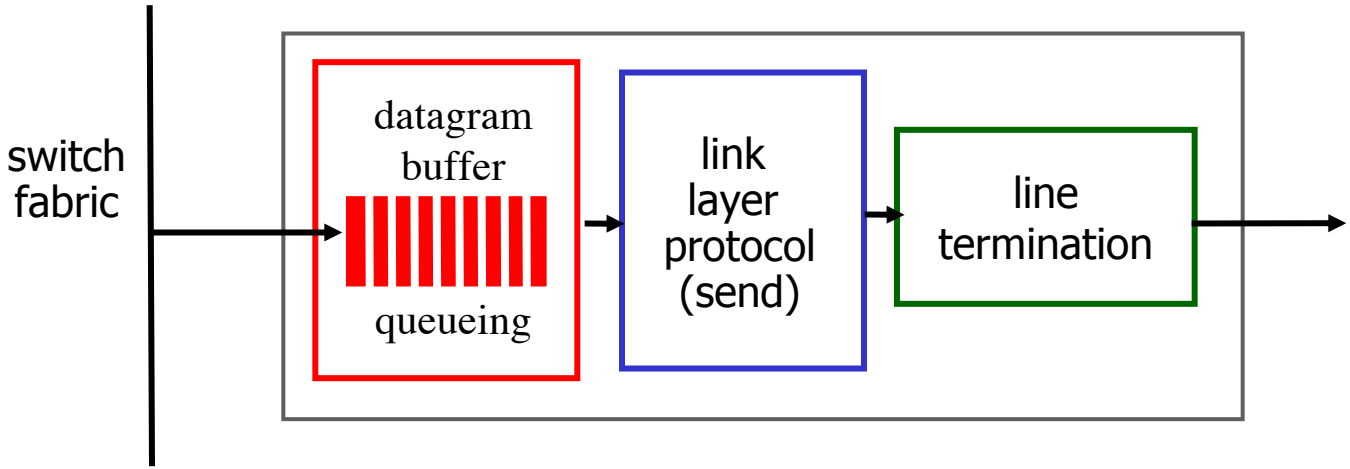


Output ports



- *buffering* required when datagrams arrive from fabric faster than the transmission rate

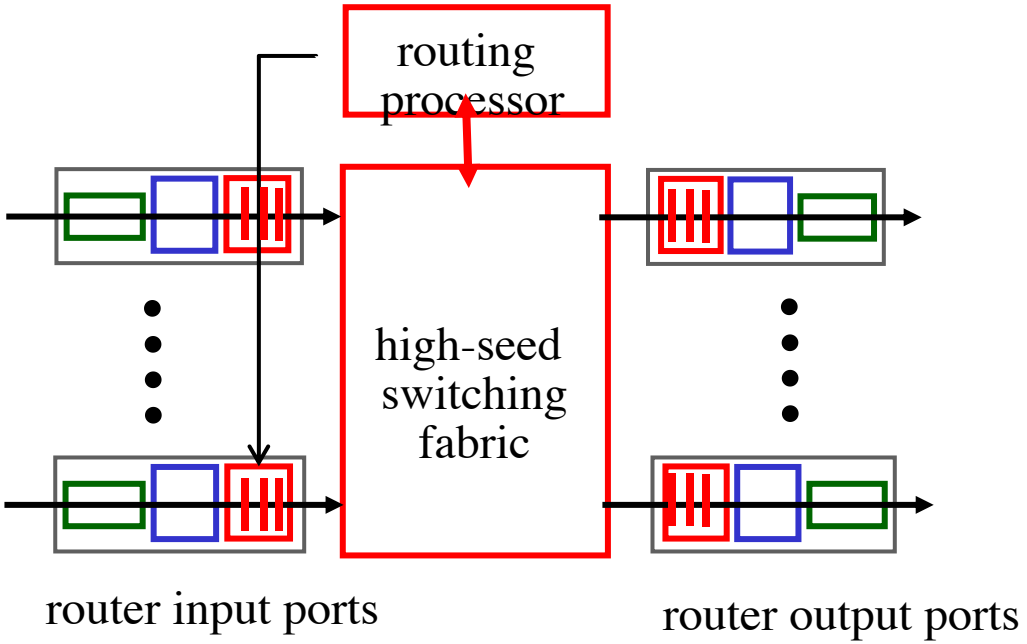
Datagram (packets) can be lost due to congestion, lack of buffers

- *scheduling discipline* chooses among queued datagrams for transmission

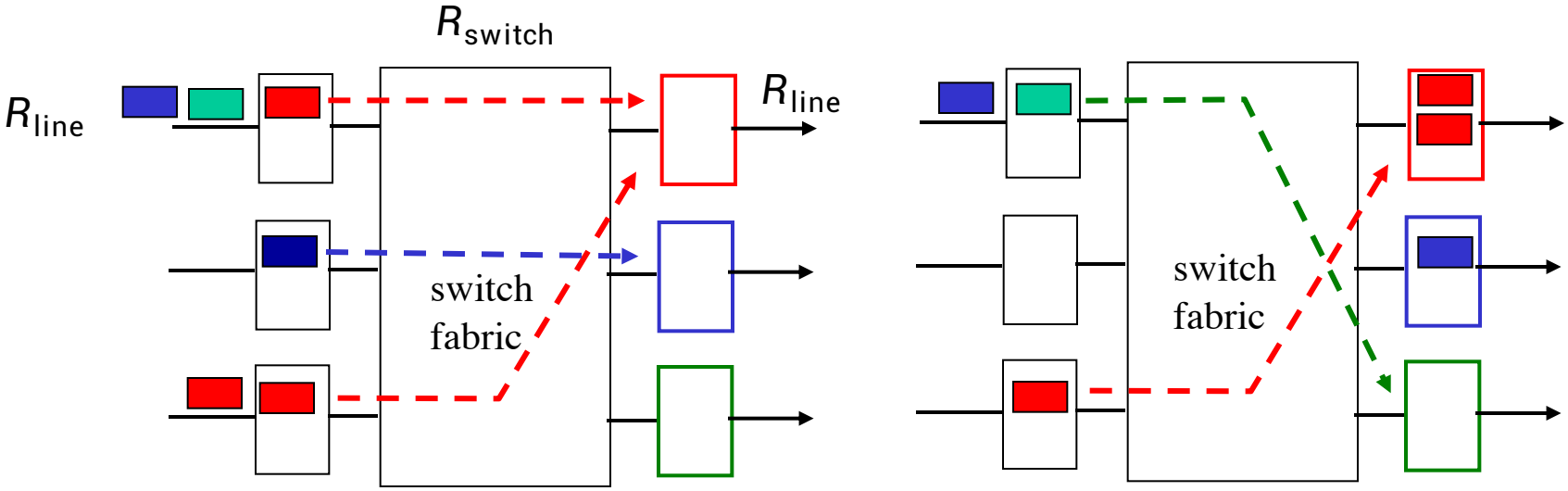
Priority scheduling – who gets best performance, network neutrality

Router Overview

- Input port
- Switch fabrics
- Output port
- Queuing
  - Input port queue
  - Output port queue
  - Queue scheduling



Input port queueing



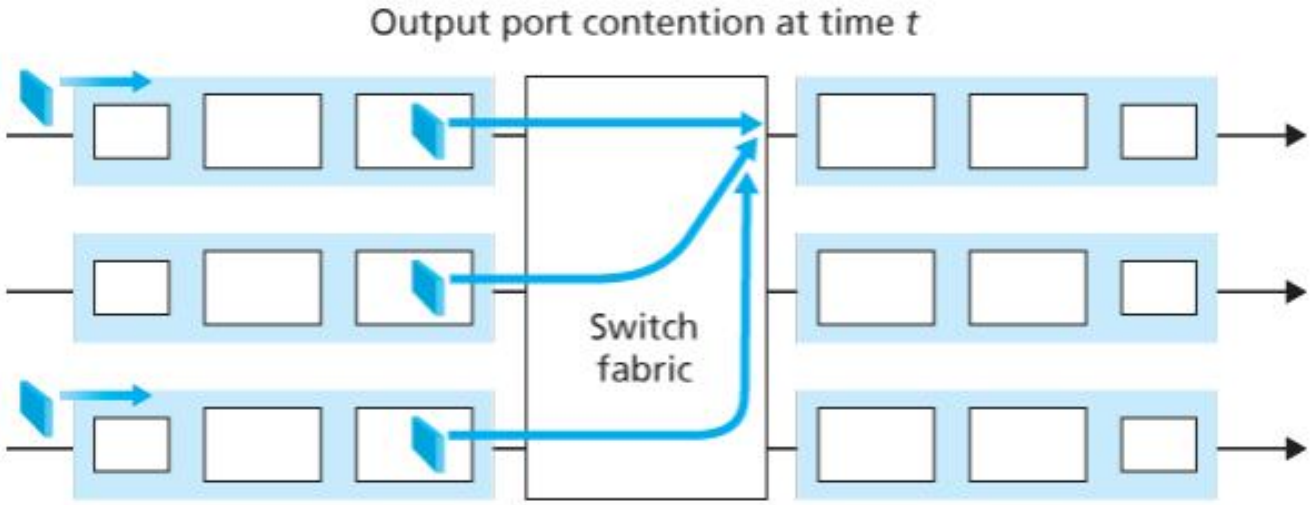
at *t*, packets more from input to output

one packet time later

Switch fabric is not fast enough (e.g., suppose $R_{switch} = R_{line}$)

- Packet queuing occur at input port
- Crossbar, and multiple packets must be transferred to the same port

Output port queueing



- If R_{switch} is N times faster than R_{line}, then negligible queuing at the input ports.
- buffering when arrival rate via switch exceeds output line speed
- *queueing (delay) and loss due to output port buffer overflow!*

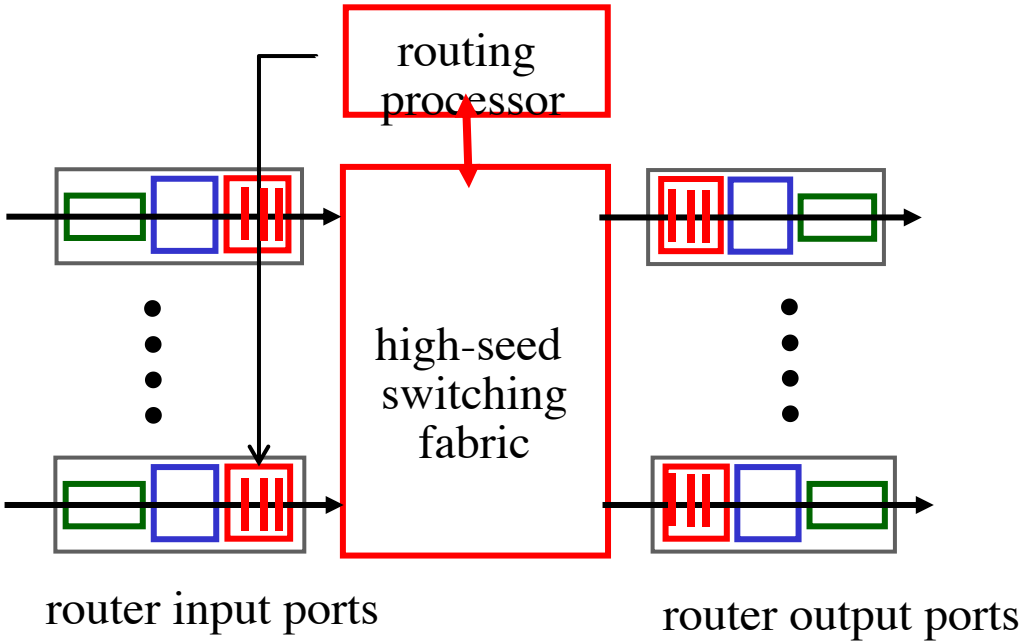
How much buffering?

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

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

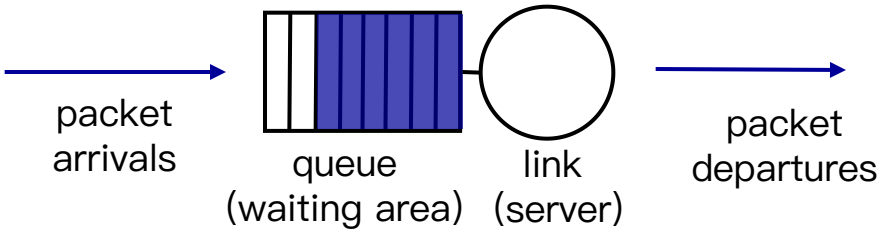
Scheduling mechanisms

- *scheduling:* choose next packet to send on link



Scheduling policies: FIFO

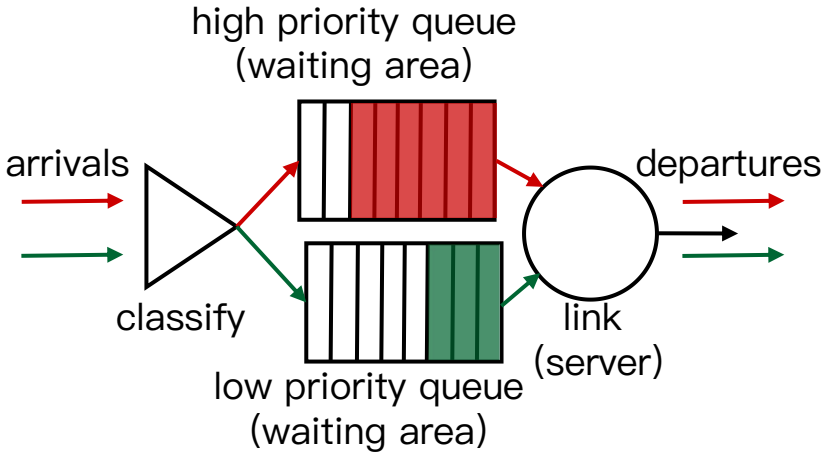
- *FIFO (first in first out) queuing:* send in order of arrival to queue
- *discard policy:* if packet arrives to full queue: who to discard?
  - *tail drop:* drop arriving packet
  - *priority:* drop/remove on priority basis
  - *random:* drop/remove randomly

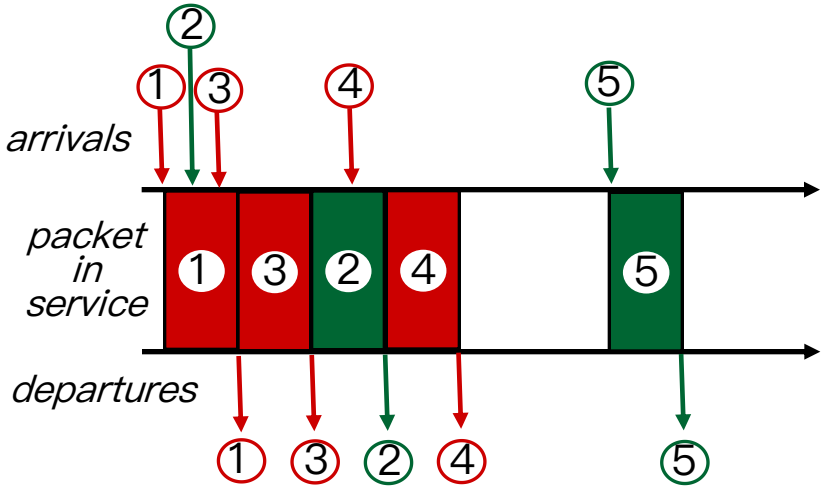


Scheduling policies: priority

priority queuing: send highest priority queued packet

- multiple *classes*, with different priorities
  - class may depend on marking or other header info, e.g. IP source/dest, port numbers, etc.
- Transmit a packet from the highest priority class
- Non-preemptive priority queuing

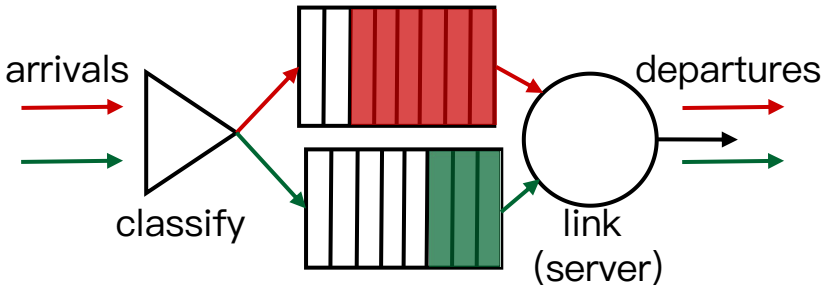


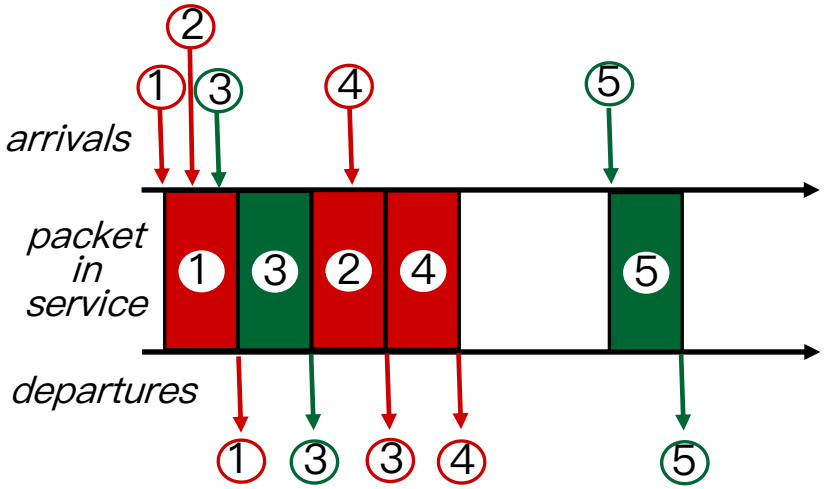


Scheduling policies: still more

Round Robin (RR) scheduling:

- multiple classes
- cyclically scan class queues, sending one complete packet from each class (if available)

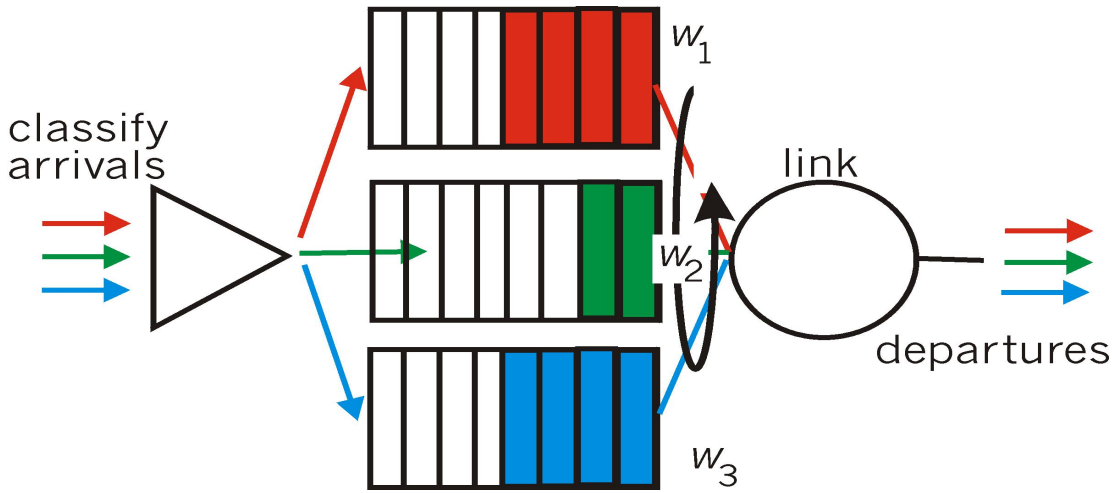




Scheduling policies

Weighted Fair Queuing (WFQ):

- generalized Round Robin
- each class gets weighted amount of service in each cycle
  - $w_i / \sum_{j \in \mathcal{Q}^{busy}} w_j$ of the bandwidth (throughput)
  - $\mathcal{Q}^{busy}$: all classes that have queued packets
  - Worst case: all queues have packets; $w_i / \sum_{j \in \mathcal{Q}} w_j$



Chapter 4: outline

4.1 Overview of Network layer
- data plane
- control plane

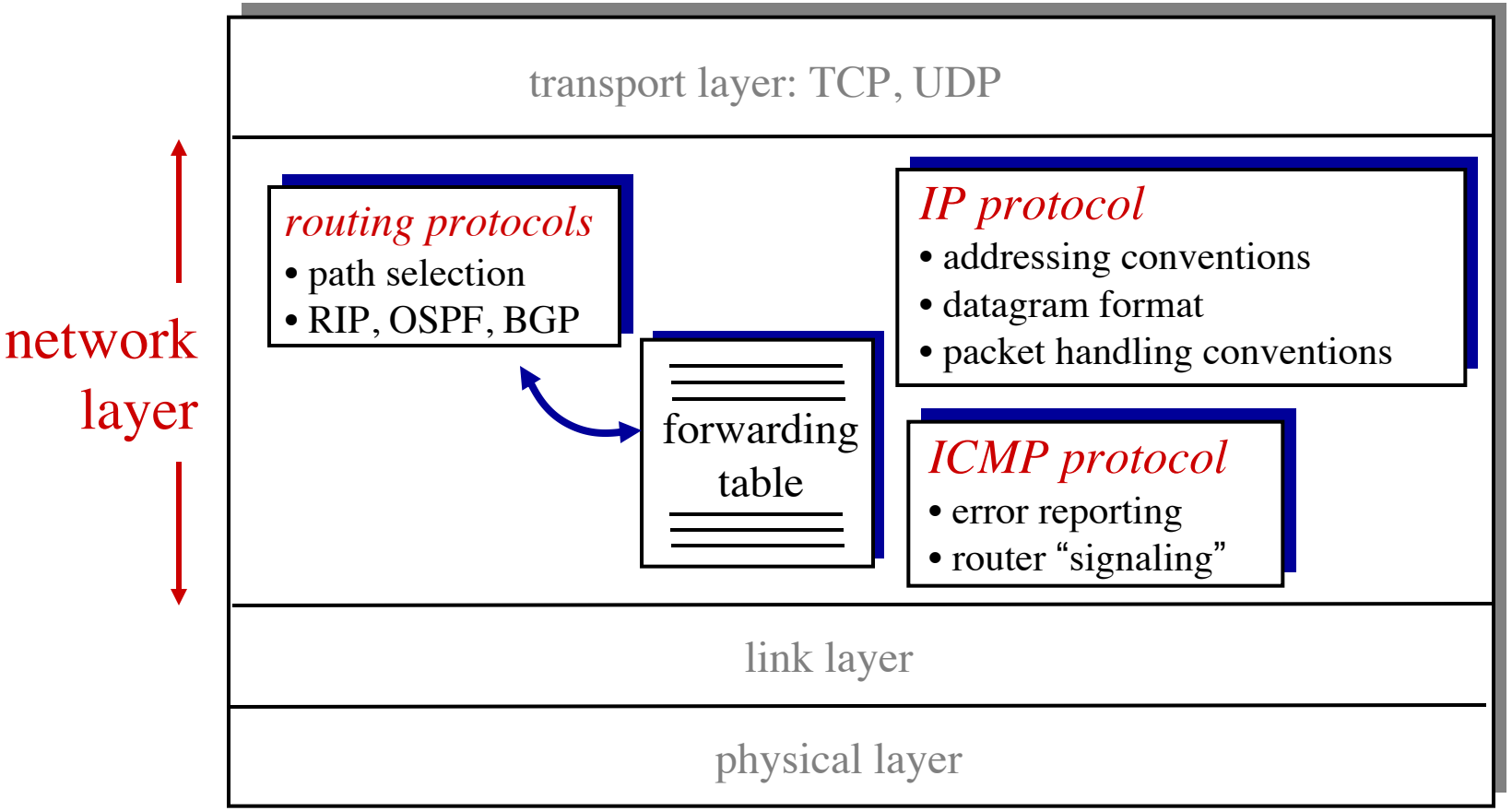
4.2 What's inside a router

4.3 IP: Internet Protocol
- datagram format
- fragmentation
- IPv4 addressing
- network address translation
- IPv6

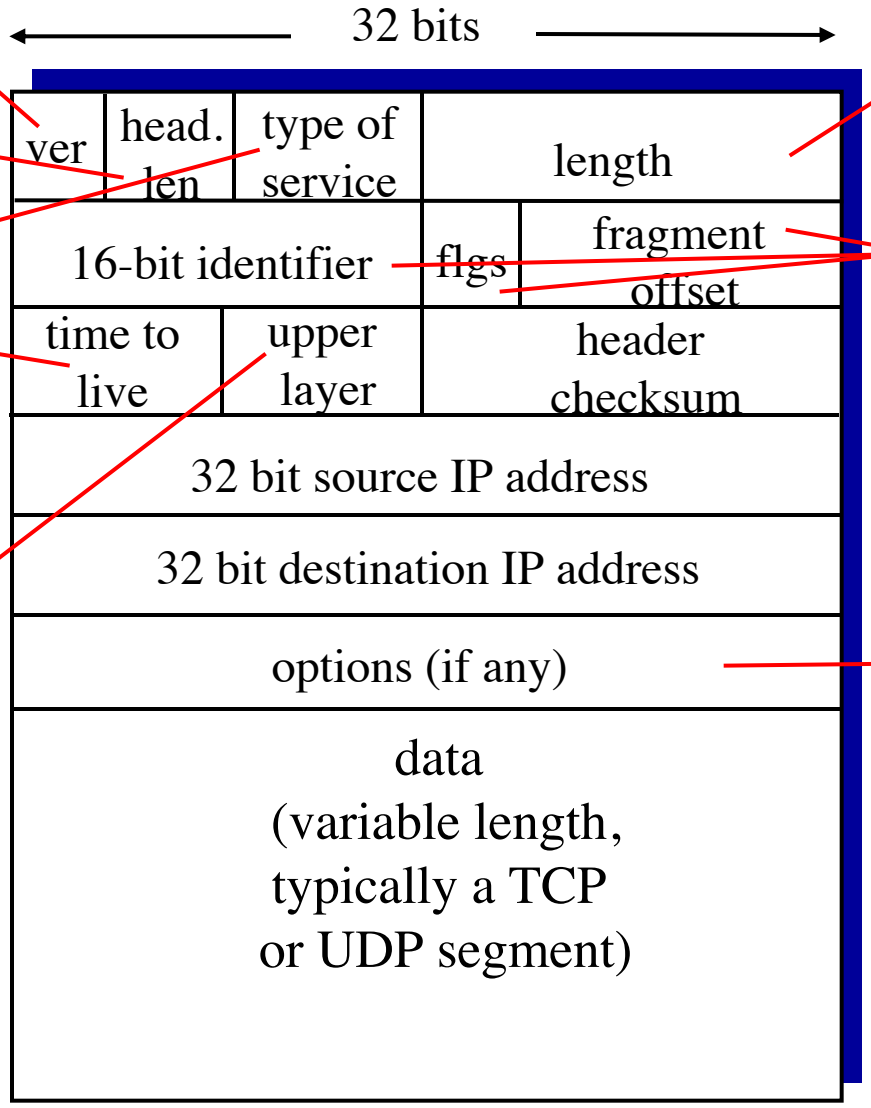
4.4 Generalized Forward and SDN
- match
- action
- OpenFlow examples of match-plus-action in action

The Internet network layer

host, router network layer functions:



IP datagram format



IP protocol version number

header length (bytes)

"type" of data

max number remaining hops (decremented at each router)

upper layer protocol to deliver payload to

total datagram length (bytes)

for fragmentation/ reassembly

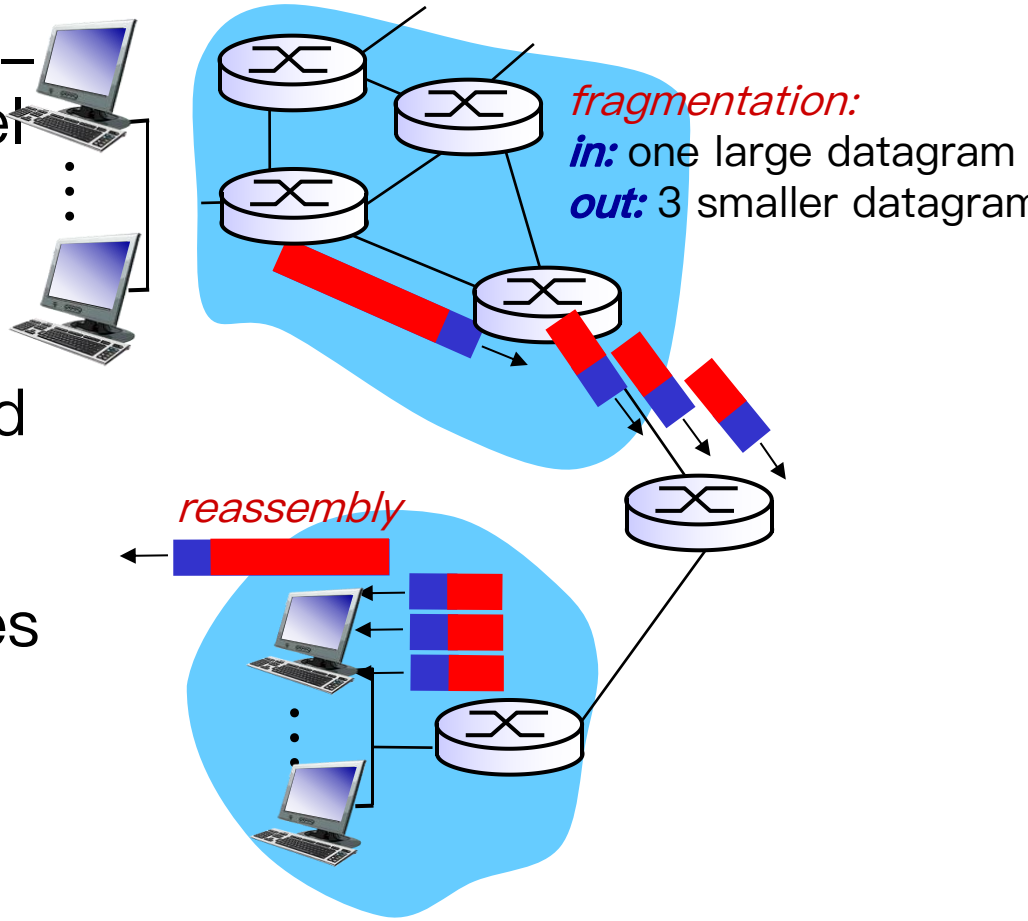
e.g. timestamp, record route taken, specify list of routers to visit.

how much overhead?

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

IP fragmentation, reassembly

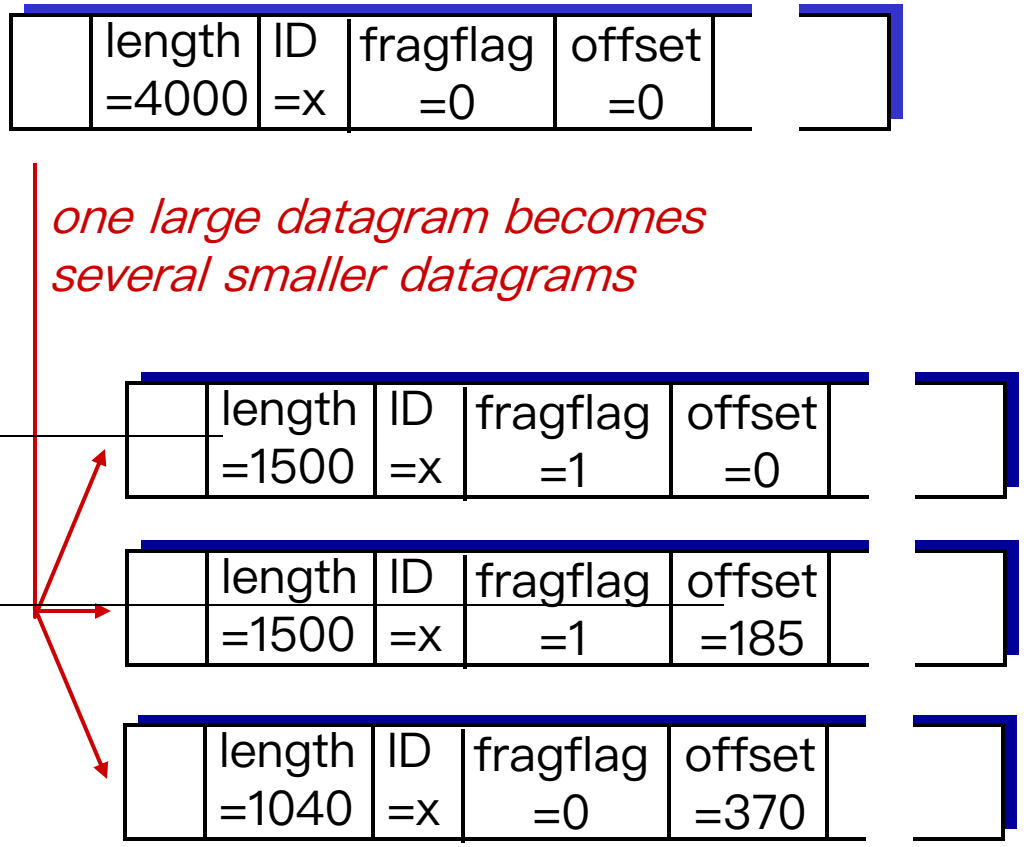
- network links have max transmission unit (MTU) – largest possible link–level frame
  - different link types, different MTUs
- large IP datagram divided ("fragmented") within net
  - one datagram becomes several datagrams
  - "reassembled" only at final destination
  - IP header bits used to identify, order related fragments



IP fragmentation, reassembly

example:

- 4000 byte datagram
- MTU = 1500 bytes



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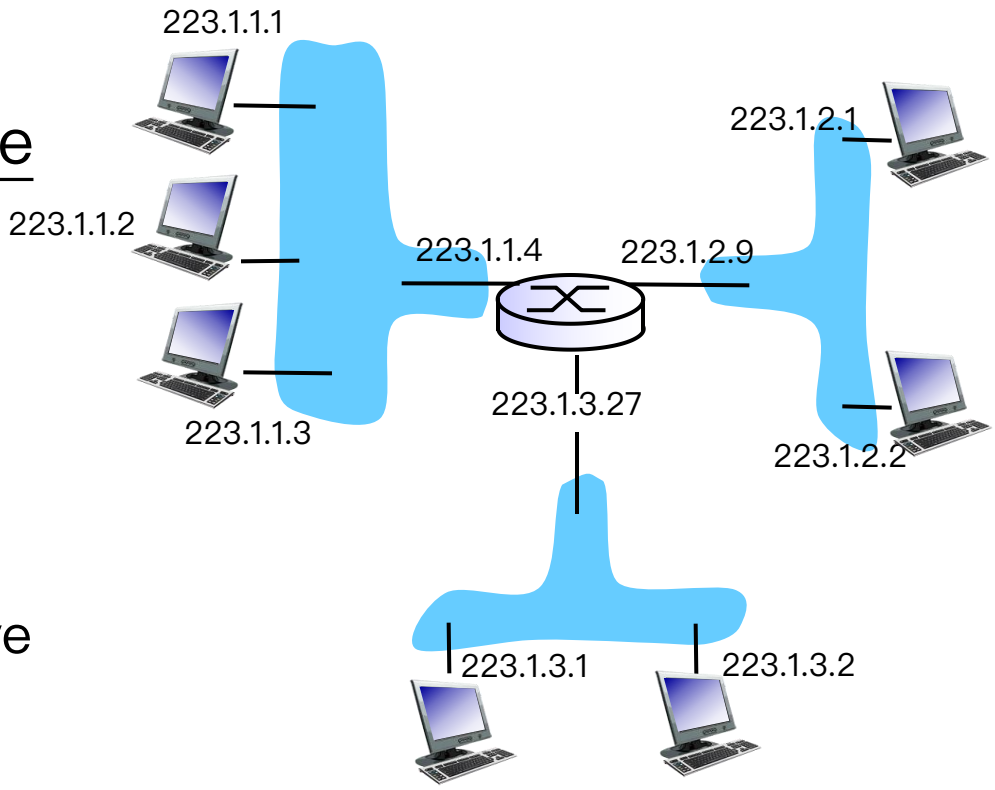
Overview

- IP addressing
- Subnet
- How to assign/obtain IP address?

IP addressing: introduction

- *IP address:* 32-bit identifier for interface of hosts and routers
- *interface:* (network interface card) 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)
- *IP addresses associated with each interface*



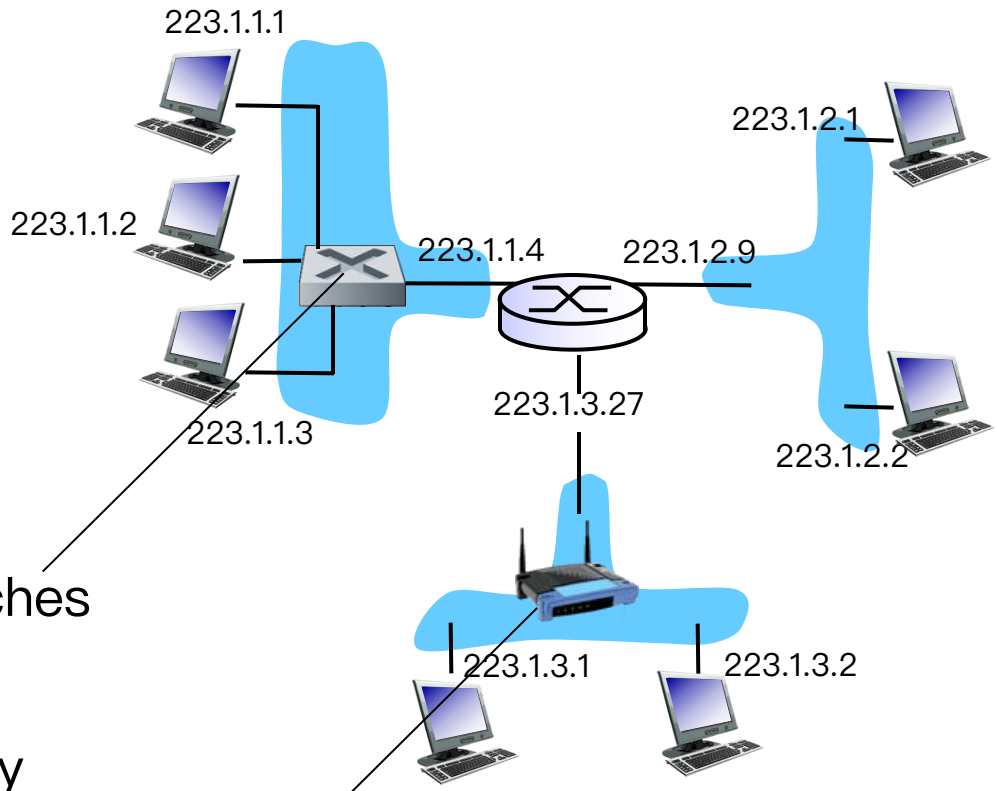
223.1.1.1 = 11011111 00000001 00000001 000000

223 1 1 1

IP addressing: introduction

Q: how are interfaces actually connected?

A: we'll learn about that in chapter 5, 6.



A: wired Ethernet interfaces connected by Ethernet switches

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

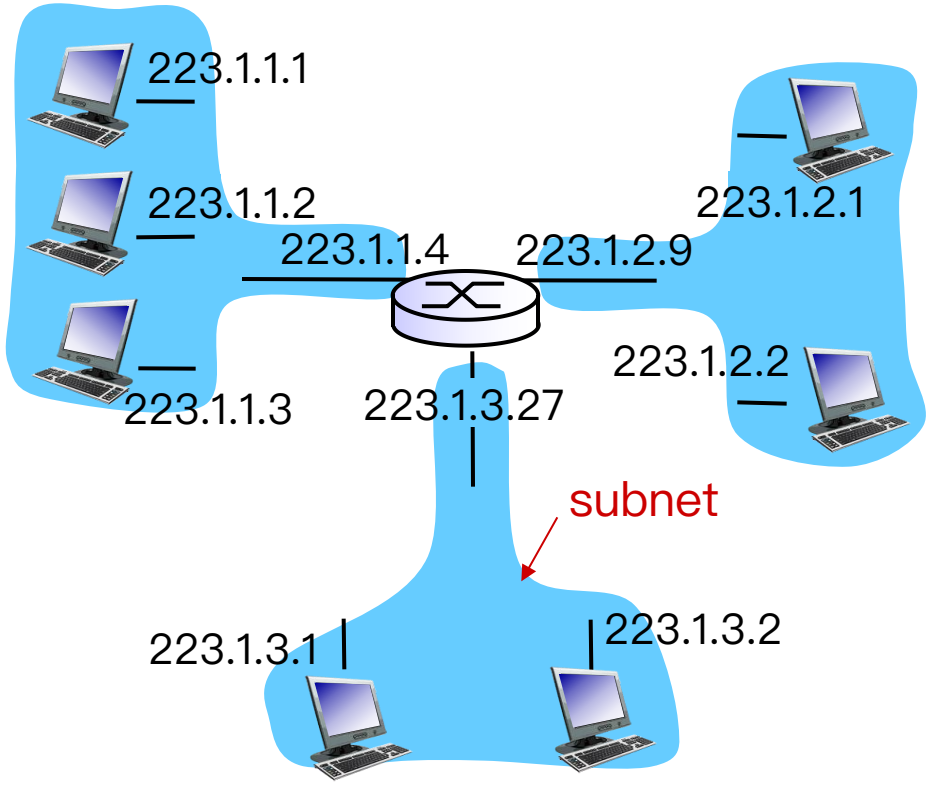
A: wireless WiFi interfaces connected by WiFi base station

Overview

- IP addressing
- Subnet
- How to assign/obtain IP address?

Subnets

- IP address:
  - subnet part – high order bits
  - host part – low order bits
- *what's a subnet ?*
  - device interfaces with same subnet part of IP address
  - can physically reach each other *without intervening router*

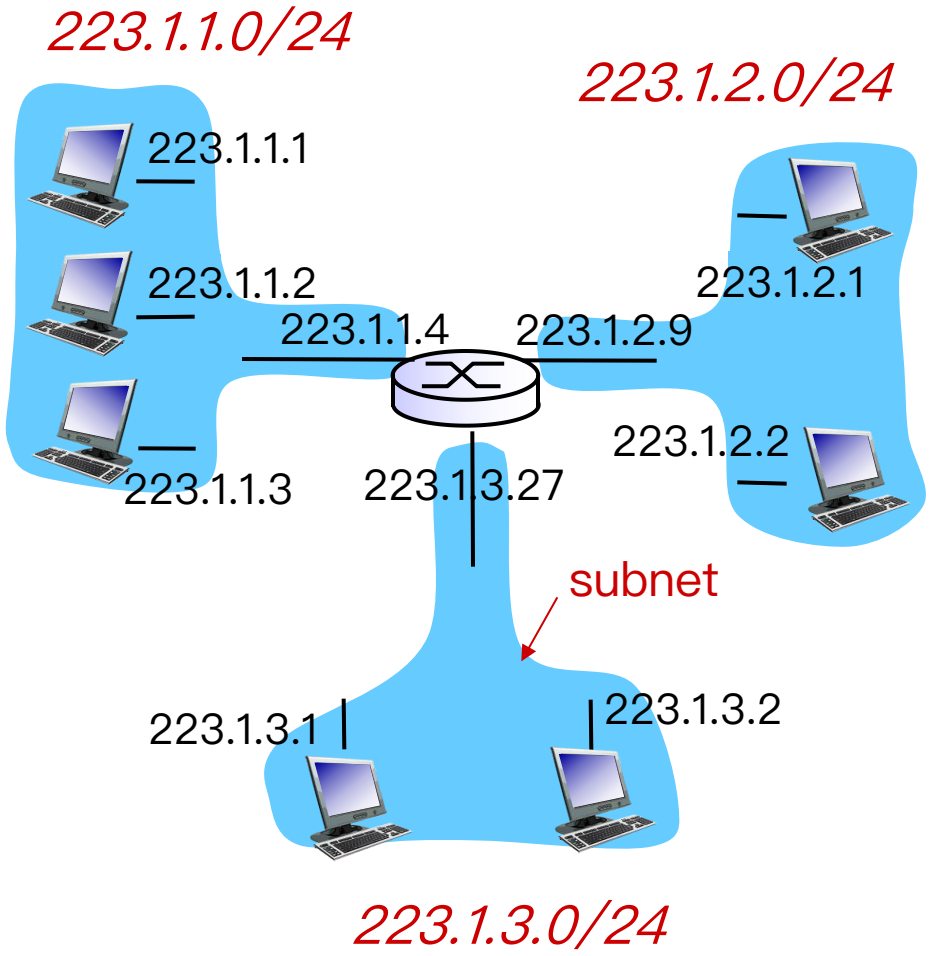


network consisting of 3 subnets

Subnets

recipe

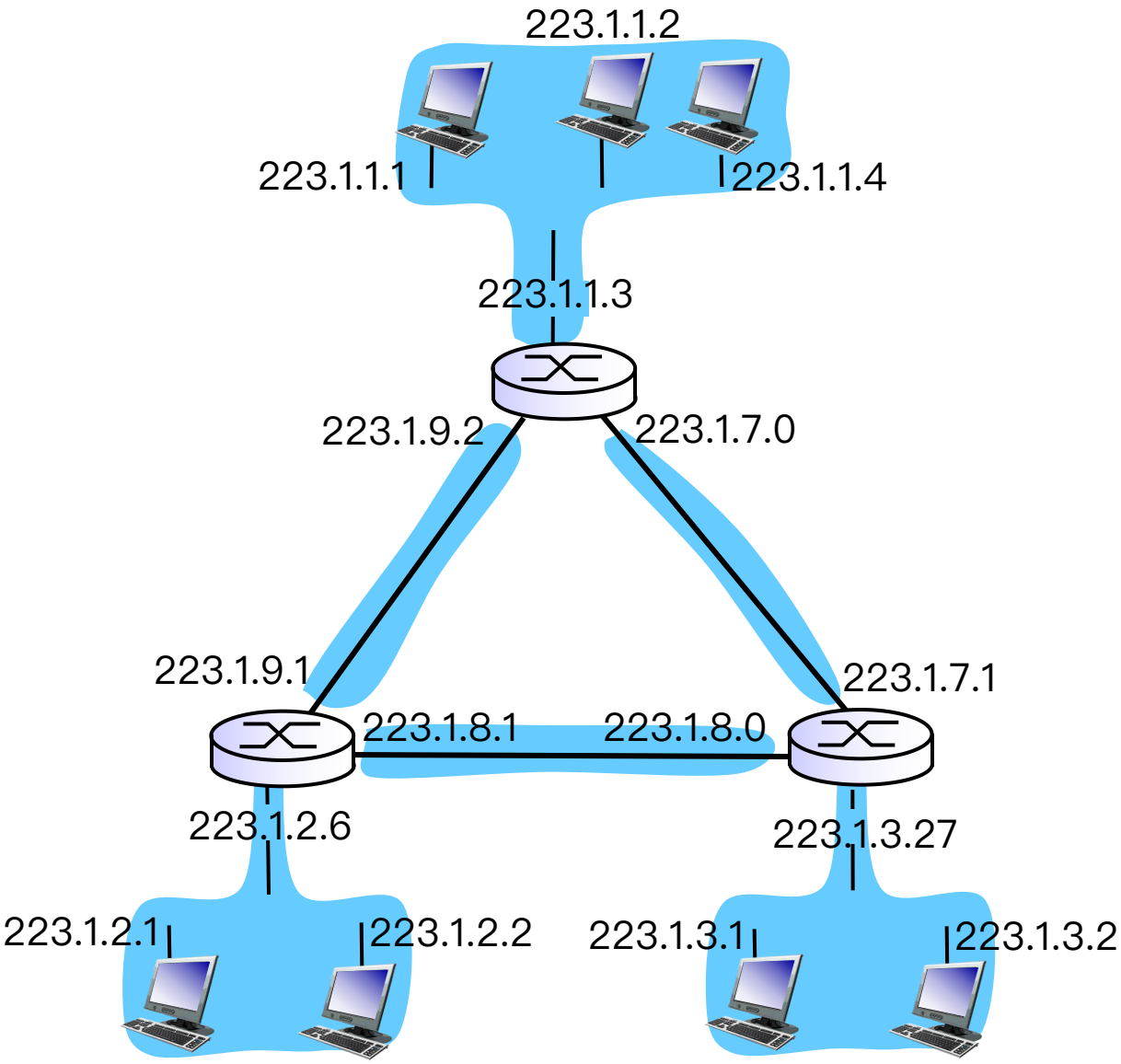
- to determine the subnets, detach each interface from its host or router, creating islands of isolated networks
- each isolated network is called a *subnet*



subnet mask: /24
255.255.255.0

Subnets

how many?



Overview

- IP addressing
- Subnet
- How to assign/obtain IP address?

IP addressing: CIDR

CIDR: Classless InterDomain Routing

- A method to assign blocks of IP address
- subnet portion of address of arbitrary length
- address format: a.b.c.d/x, where x is # bits in subnet portion of address

11001000 00010111 00010000 00000000

200.23.16.0/23

Subnet mask: 255.255.254.0

How does an ISP get block of addresses?

Q: how does an ISP get block of addresses?

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

- allocates addresses
- manages DNS
- assigns domain names, resolves disputes

How does a subnet get block of address

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

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

ISP's block	11001000	00010111	00010000	00000000	200.23.16.0/20
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

How does a subnet get block of address

Suppose all of the interfaces in each of these three subnets are required to have the prefix 166.4.20.128/25.

- Subnet 1 is required to support at least 62 interfaces
- Subnet 2 is required to support at least 30 interfaces
- Subnet 3 is required to support at least 28 interfaces

Provide three network addresses (of the form a.b.c.d/x) that satisfy these constraints.

How does a subnet get block of address

Block of addresses: 166.4.20.128/25. Subnet 1: at least 62 interfaces; Subnet 2: at least 30 interfaces; Subnet 3: at least 28 interfaces

This block of IP addresses can be written as

10100110 00000100 00010100 10000000

Subnet 1: since $2^6 = 64 > 62$, we can assign the following block

10100110 00000100 00010100 10000000

which can be represented as 166.4.20.128/26.

Subnet 2: since $2^5 = 32 > 30$, we can assign the following block

10100110 00000100 00010100 11000000

which can be represented as 166.4.20.192/27.

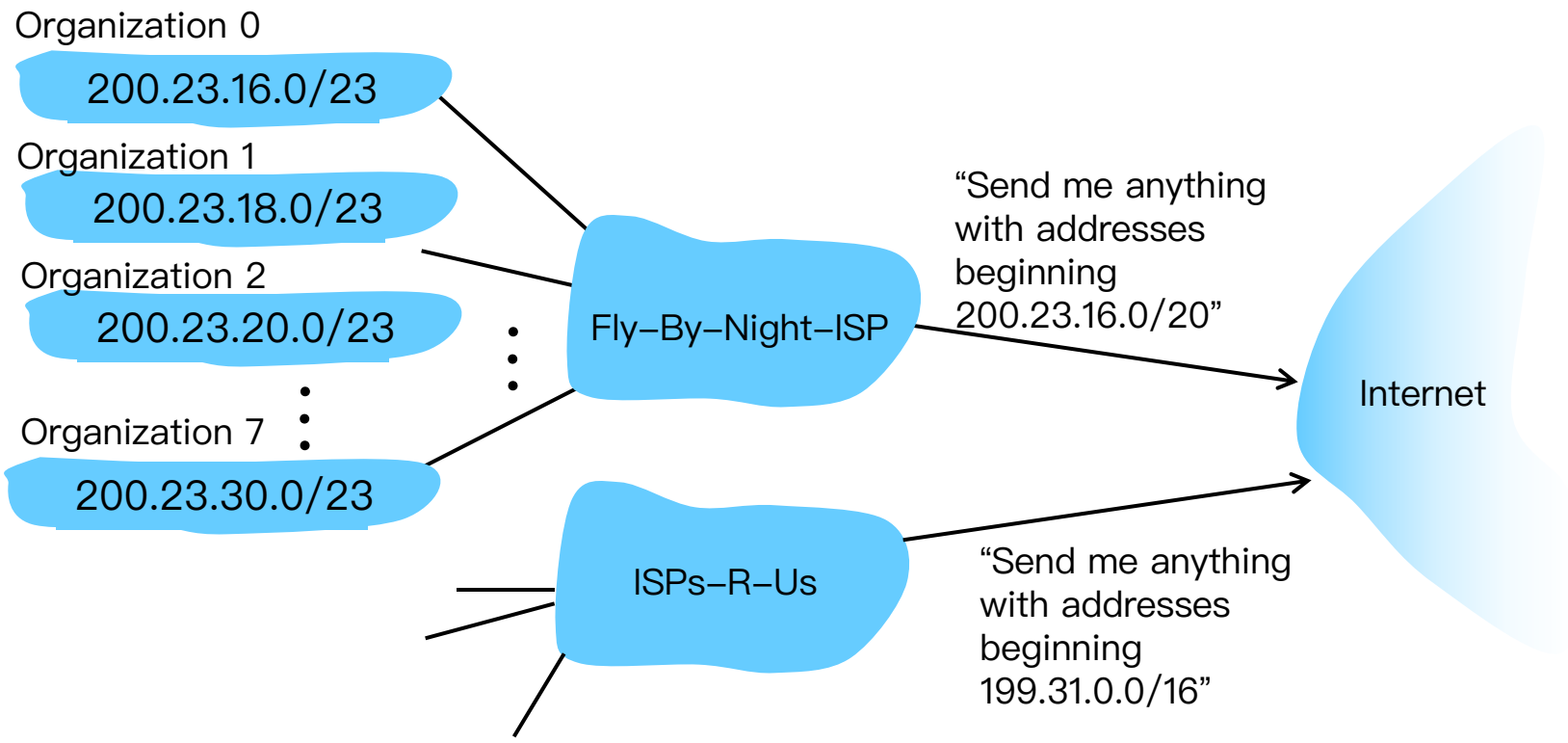
Subnet 3: since $2^5 = 32 > 38$, we can assign the following block

10100110 00000100 00010100 11100000

which can be represented as 166.4.20.224/27.

Hierarchical addressing: route aggregation

hierarchical addressing allows efficient advertisement of rou
information:

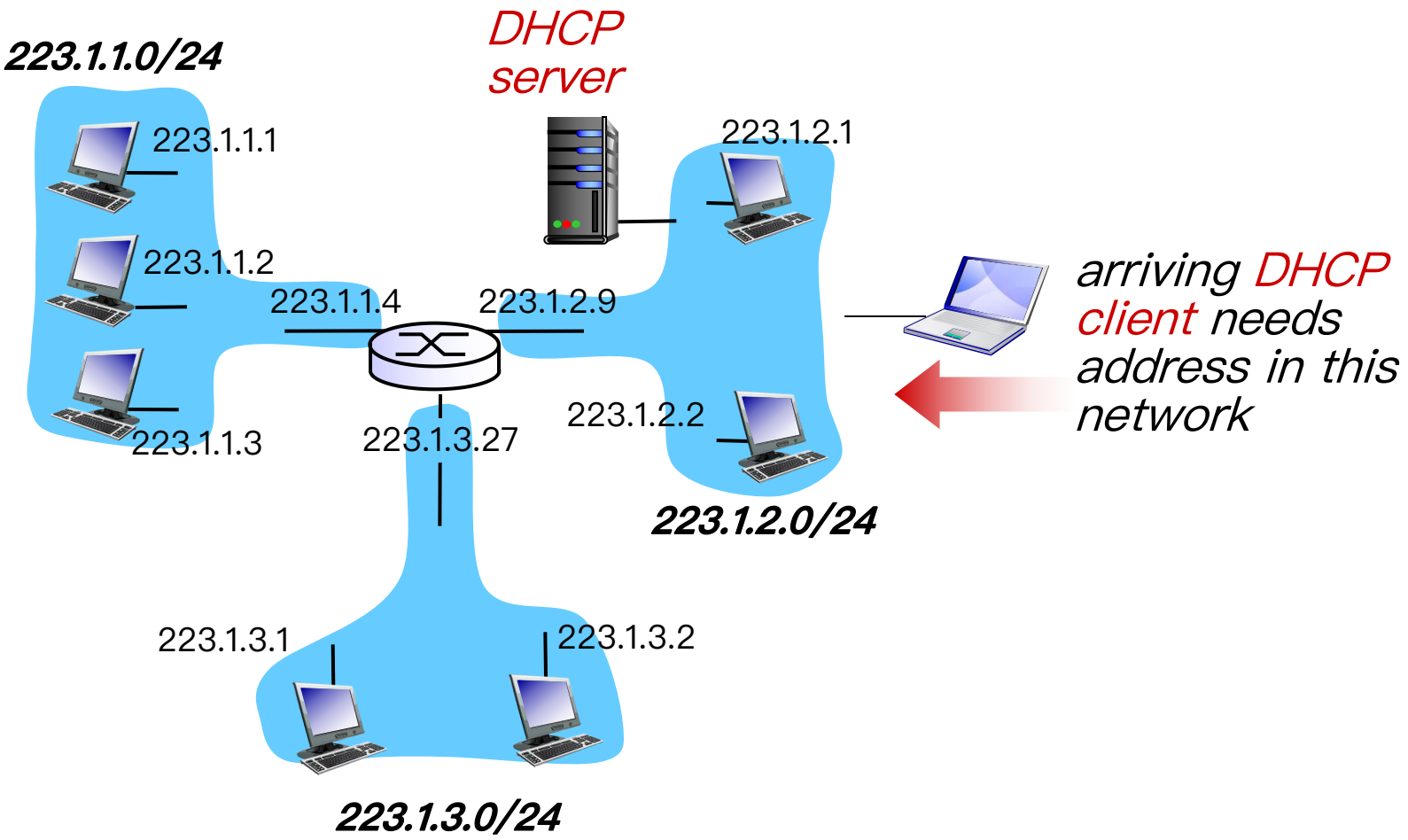


How does a host get an IP?

Q: How does a *host* get IP address?

- hard–coded by system admin in a file
  - Windows: control–panel–>network–>configuration–>tcp/ip–>properties
  - UNIX: /etc/rc.config
- DHCP: Dynamic Host Configuration Protocol: dynamically get address from as server
  - "plug–and–play"

DHCP client–server scenario



DHCP: Dynamic Host Configuration Protocol

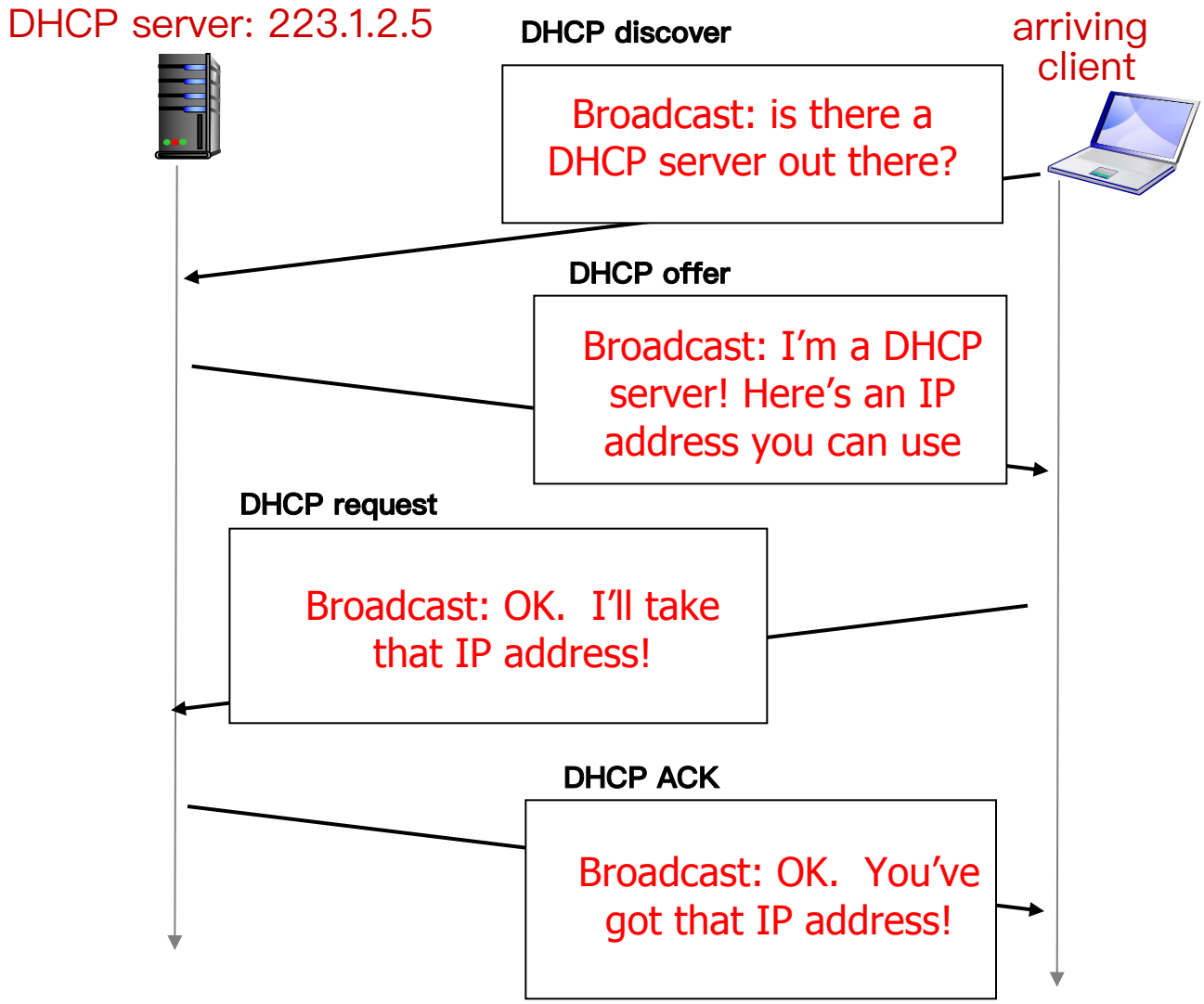
goal: allow host to *dynamically* obtain its IP address from network server when it joins network

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

DHCP overview:

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

DHCP client–server scenario



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)