College of Information Studies
University of Maryland Hornbake Library Building College Park, MD 20742-4345

Routers

Session 12

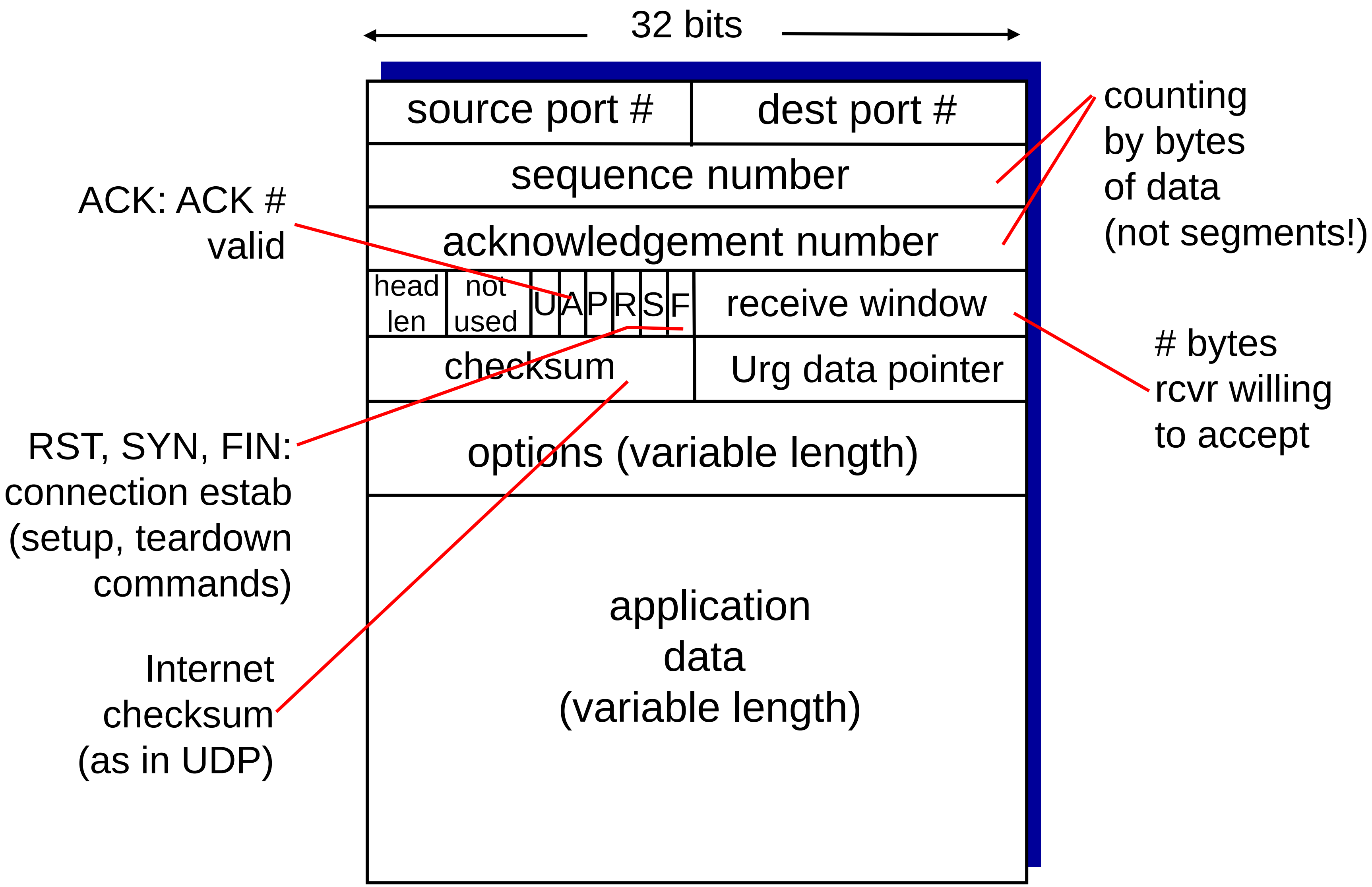
INST 346

Technologies, Infrastructure and Architecture

Goals for Today

- Finish up TCP
  - Flow control, timeout selection, close connection
- Network layer overview
- Structure of a router
- Getahead: IPv4 addresses

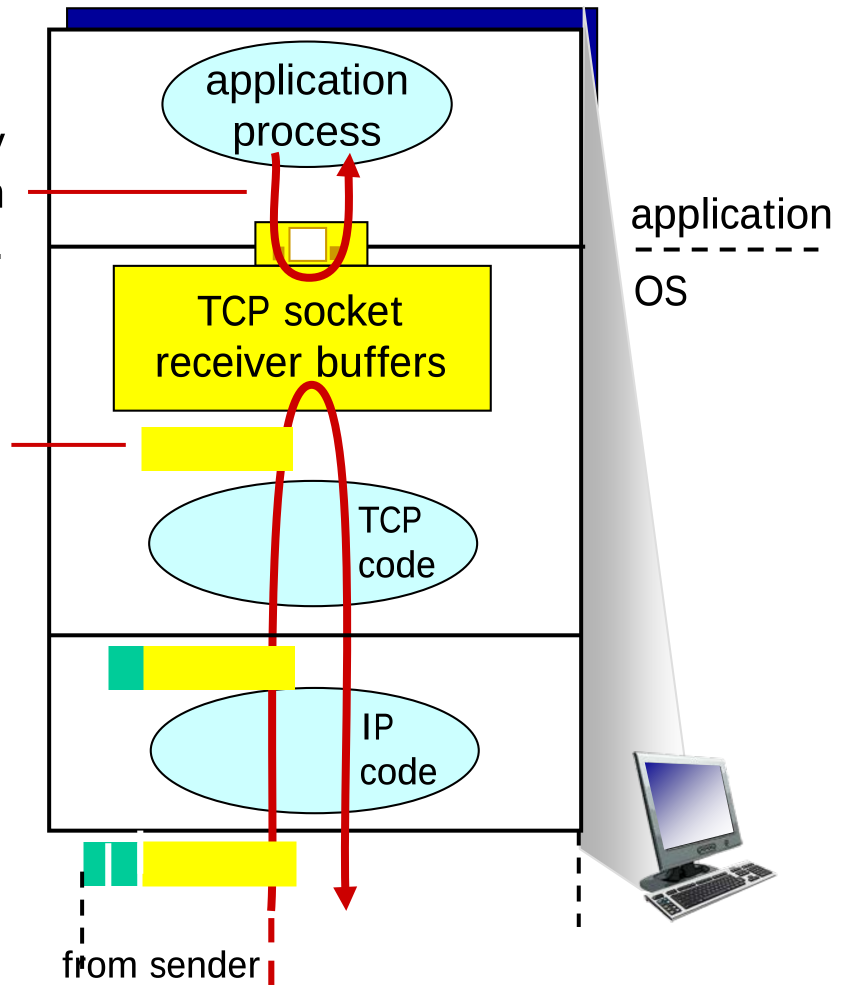
TCP segment structure



TCP flow control

application may remove data from TCP socket buffers ....

... slower than TCP receiver is delivering (sender is sending)



receiver protocol stack

flow control

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

TCP round trip time, timeout

Q: how to set TCP timeout value?

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

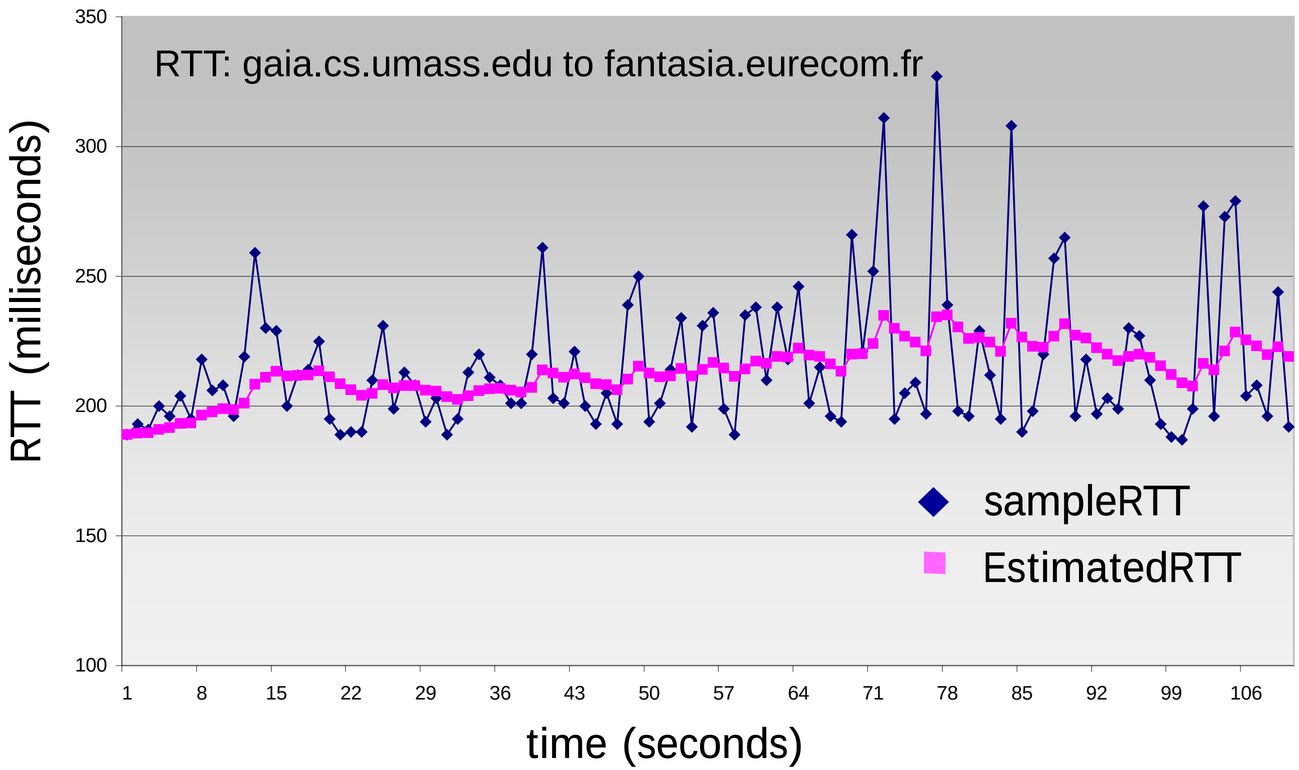
Q: how to estimate RTT?

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

TCP round trip time, timeout

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

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



TCP round trip time, timeout

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

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

(typically, β = 0.25)

`TimeoutInterval = EstimatedRTT + 4*DevRTT`

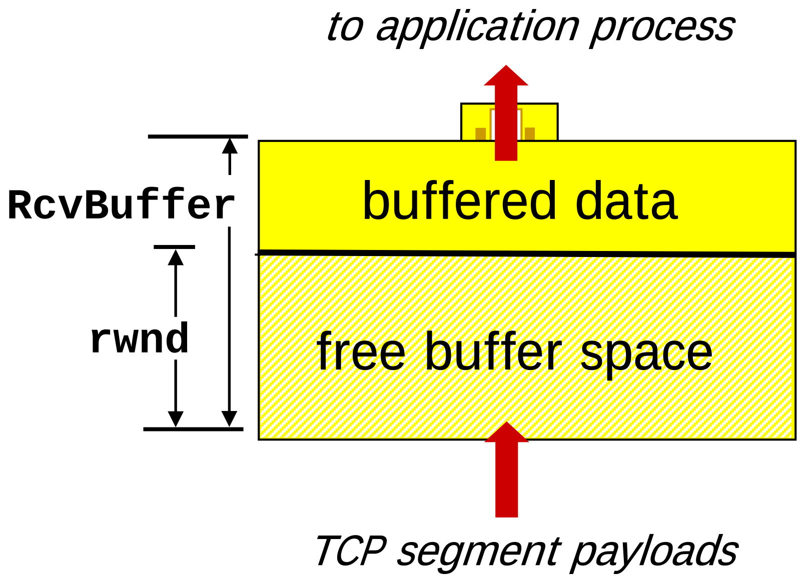
estimated RTT

"safety margin"

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

TCP flow control

- receiver "advertises" free buffer space by including **rwnd** value in TCP header of receiver-to-sender segments
  - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **RcvBuffer**
- sender limits amount of unacked ("in-flight") data to receiver's **rwnd** value
- guarantees receive buffer will not overflow

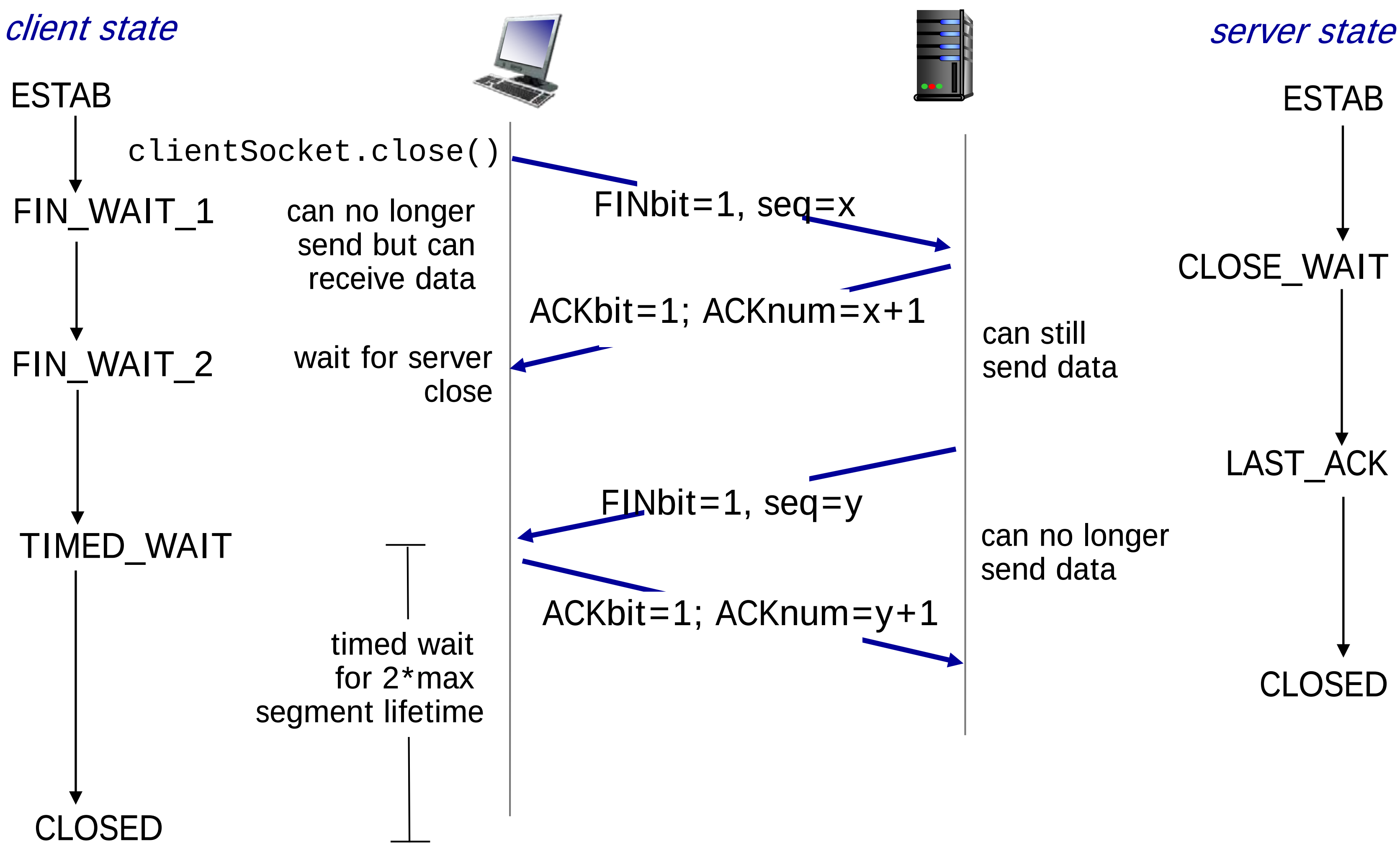


receiver-side buffering

TCP: closing a connection

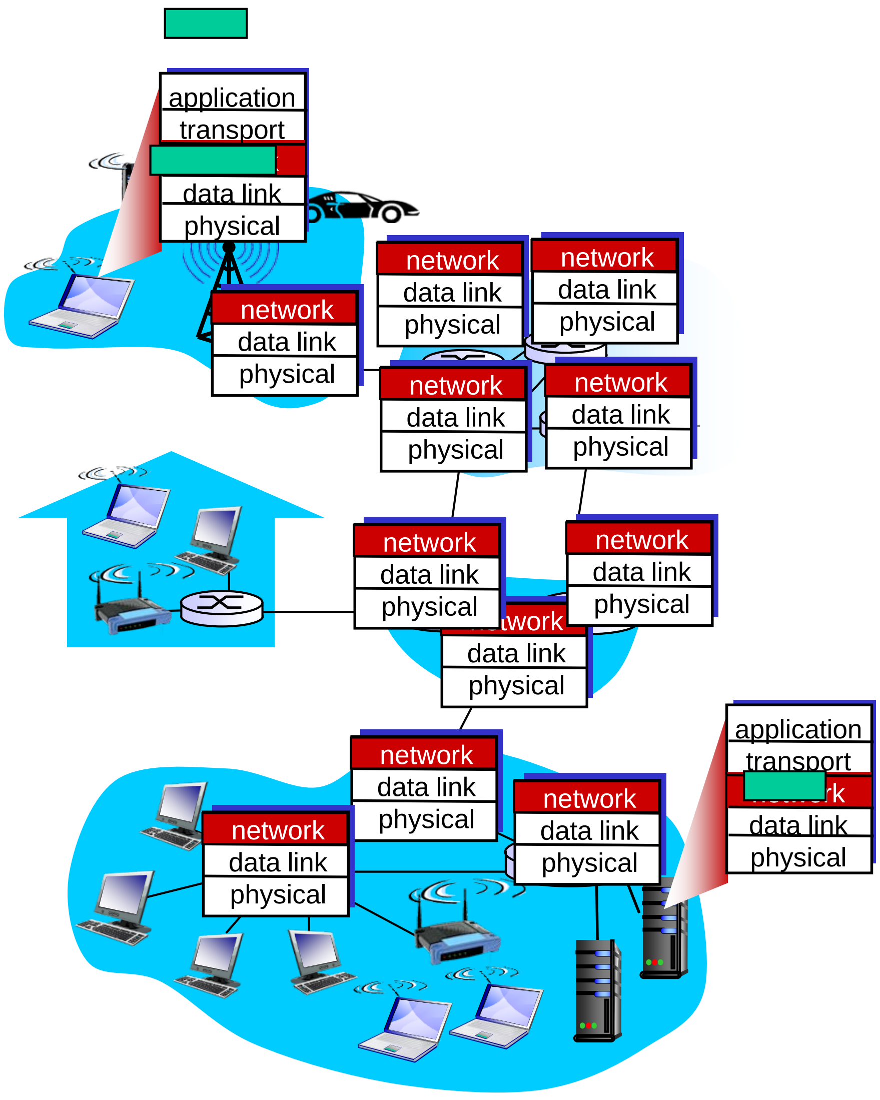
- client, server each close their side of connection
  - send TCP segment with FIN bit = 1
- respond to received FIN with ACK
  - on receiving FIN, ACK can be combined with own FIN
- simultaneous FIN exchanges can be handled

TCP: closing a connection



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-layer functions

network-layer functions:

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

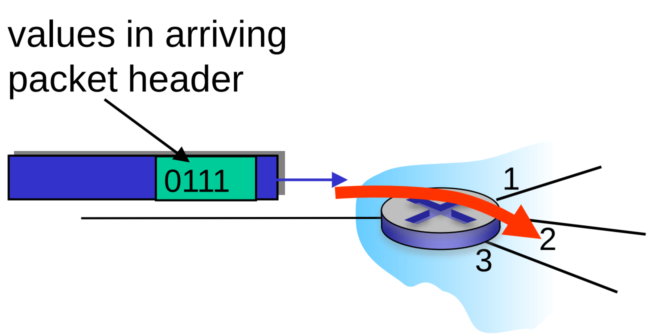
analogy: taking a trip

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

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
- forwarding function

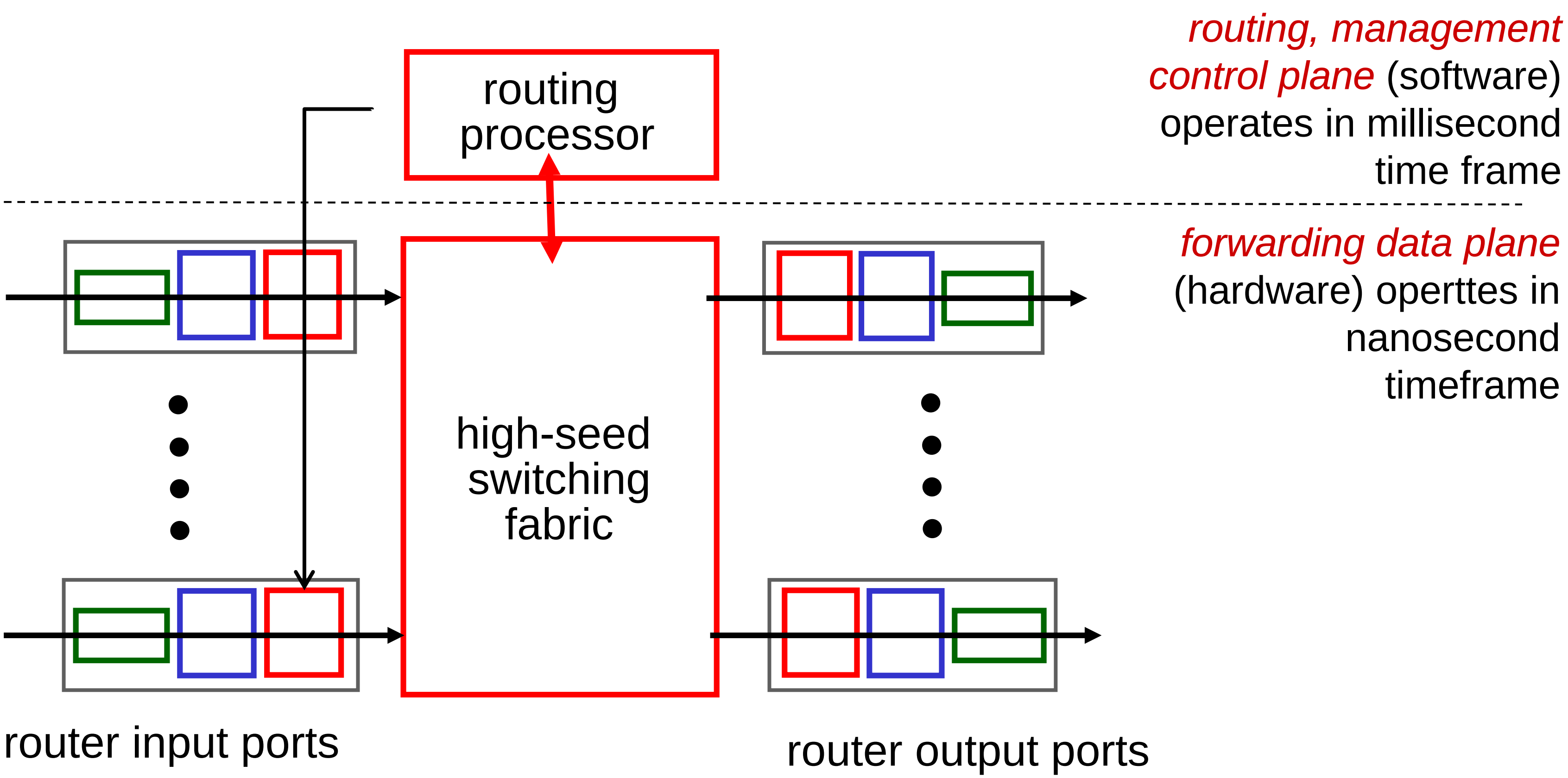


Control plane

- 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

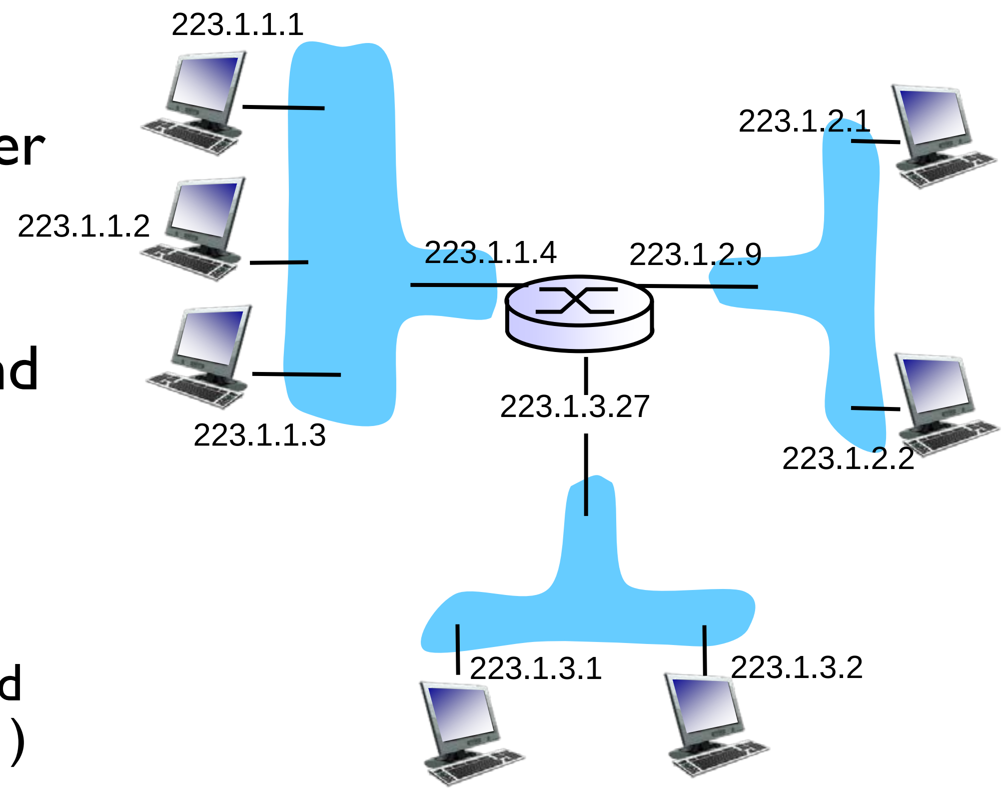
Router architecture overview

- high-level view of generic router architecture:



IP addressing: introduction

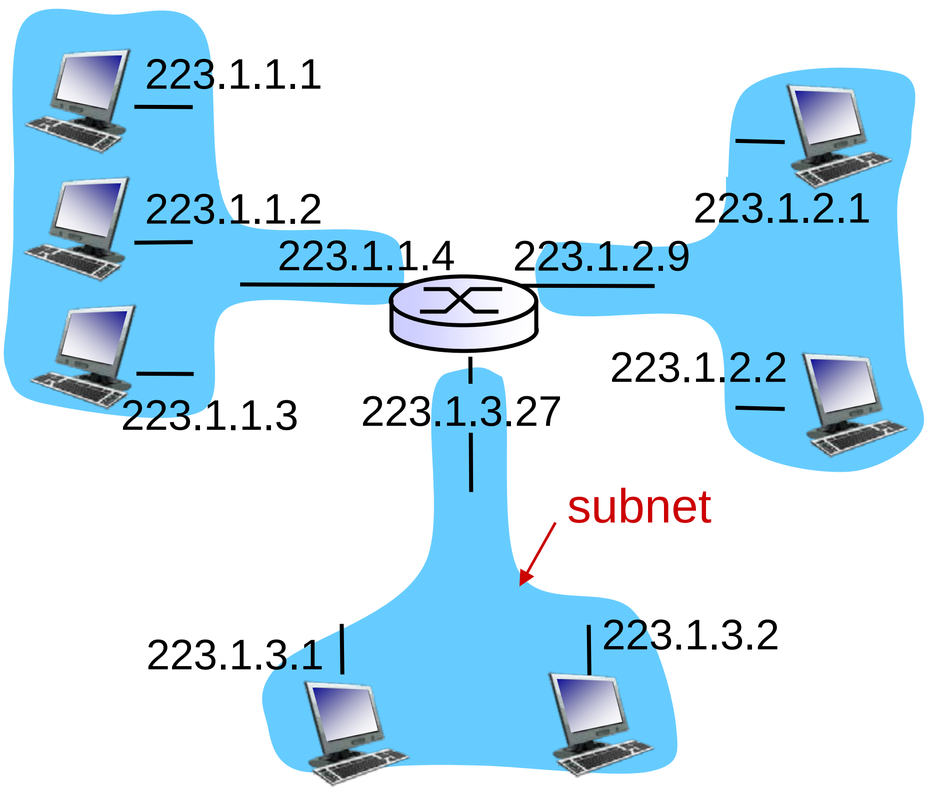
- *IP address:* 32-bit identifier for host, router *interface*
- *interface:* connection between host/router and physical link
  - router' s typically have multiple interfaces
  - host typically has one or two interfaces (e.g., wired Ethernet, wireless 802.11)
- *IP addresses associated with each interface*



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

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

IP addressing: CIDR

CIDR: Classless InterDomain Routing

- 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?

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 0001000	0 00000000	200.23.16.0/23
Organization 1	11001000 00010111 0001001	0 00000000	200.23.18.0/23
Organization 2	11001000 00010111 0001010	0 00000000	200.23.20.0/23
...	.....	....	....
Organization 7	11001000 00010111 0001111	0 00000000	200.23.30.0/23

IP addressing: the last word...

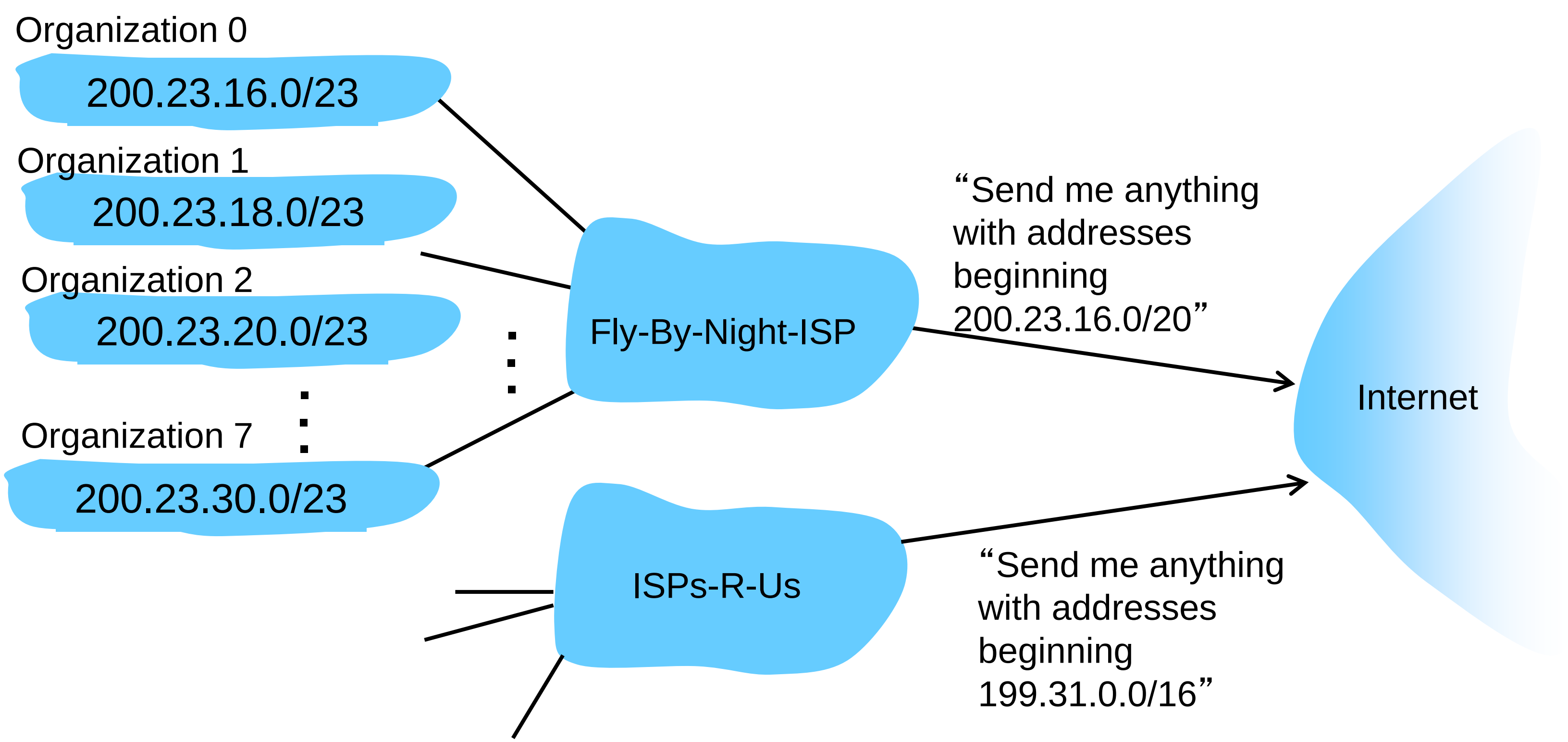
Q: how does an ISP get block of addresses?

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

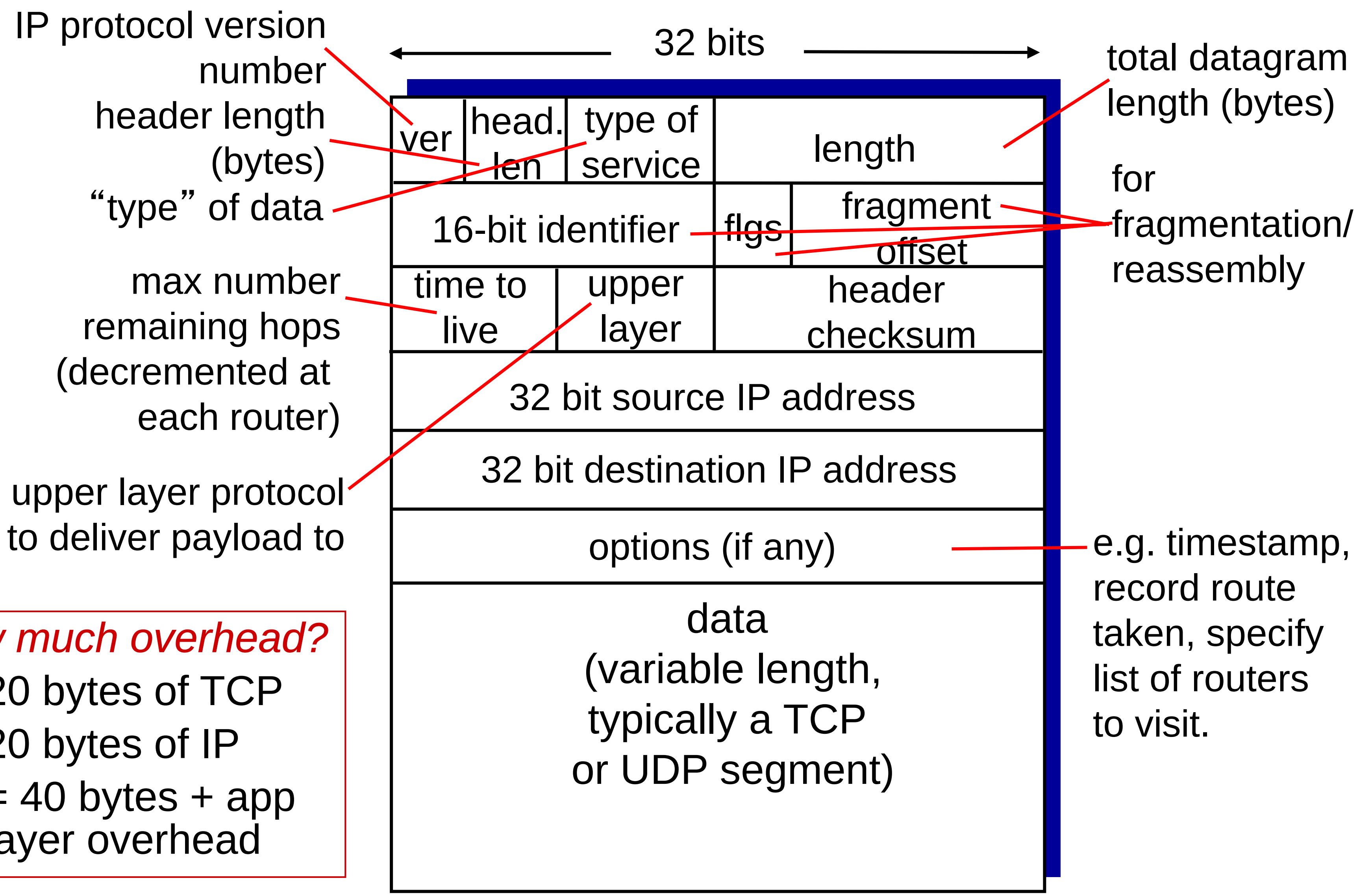
- allocates addresses
- manages DNS
- assigns domain names, resolves disputes

Hierarchical addressing: route aggregation

hierarchical addressing allows efficient advertisement of routing information:



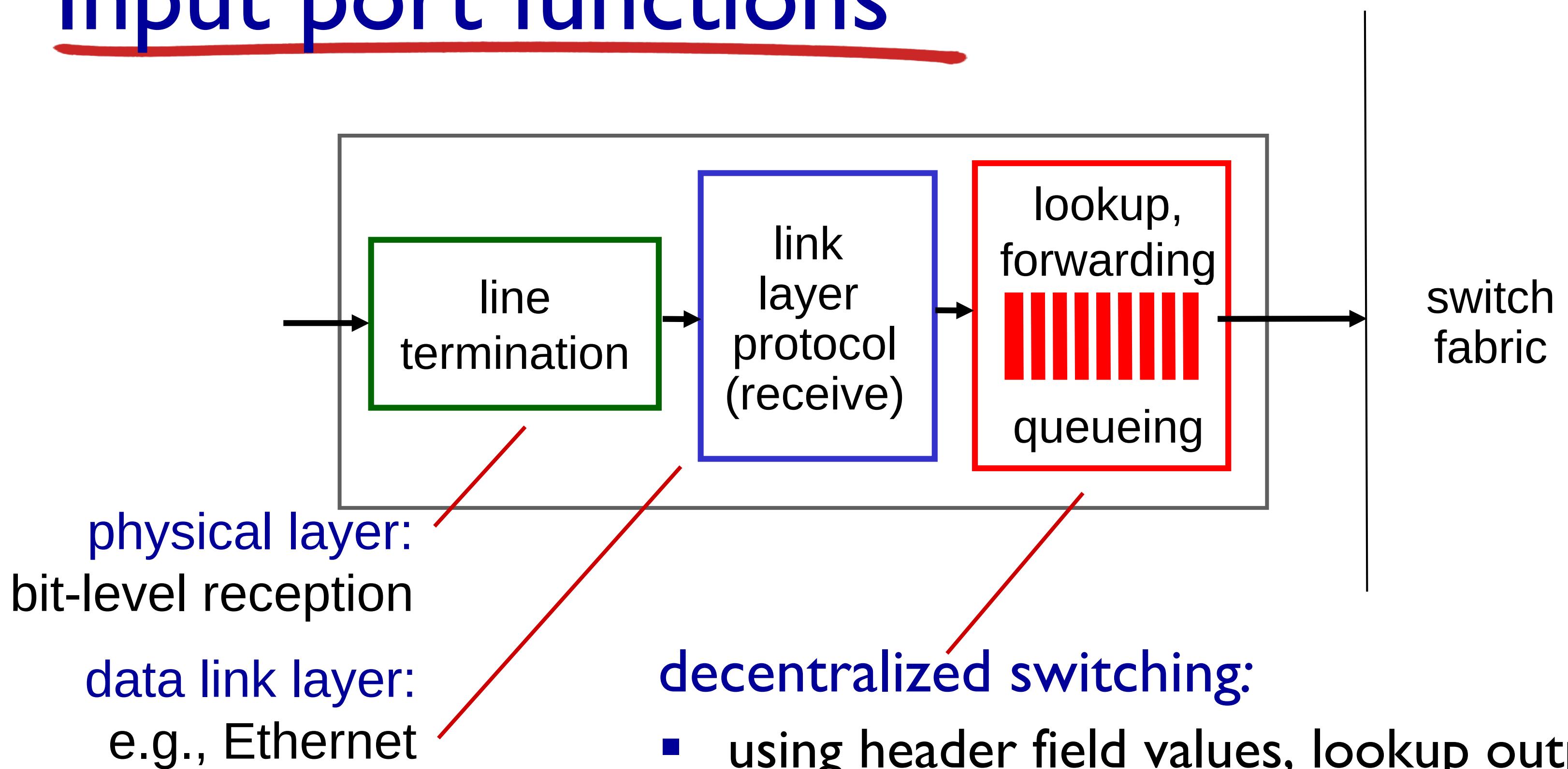
IP datagram format



how much overhead?

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

Input port functions



decentralized switching:

- using header field values, lookup output port using forwarding table in input port memory
- 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?

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?

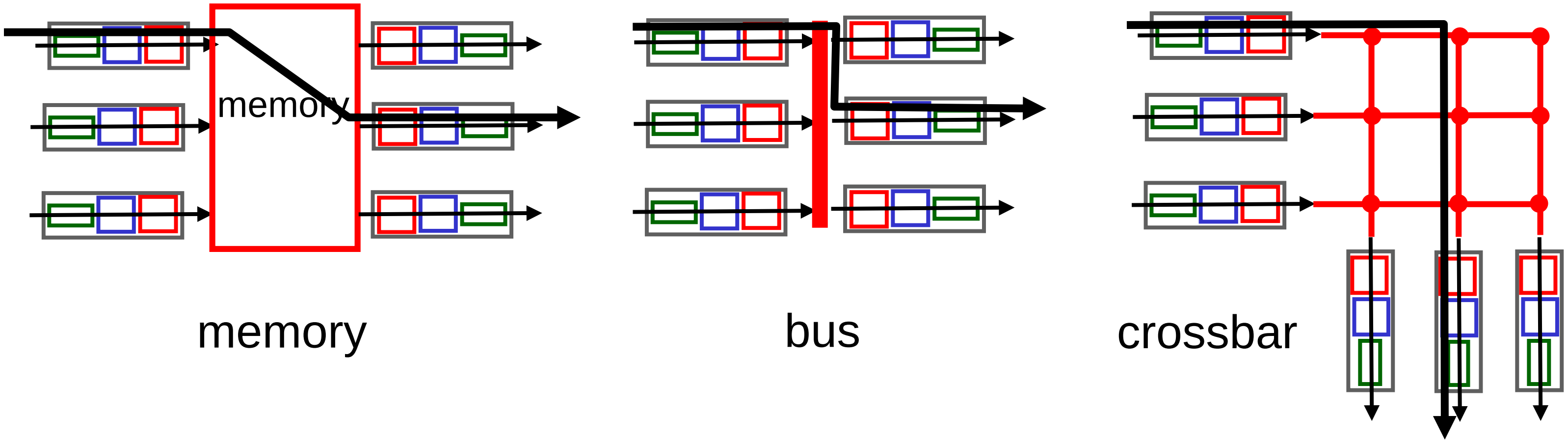
DA: 11001000 00010111 00011000 10101010 which interface?

Longest prefix matching

- 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: can up ~1M routing table entries in TCAM

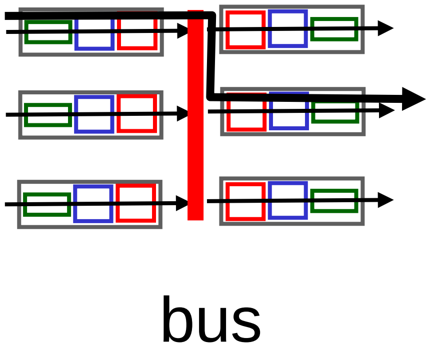
Switching fabrics

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



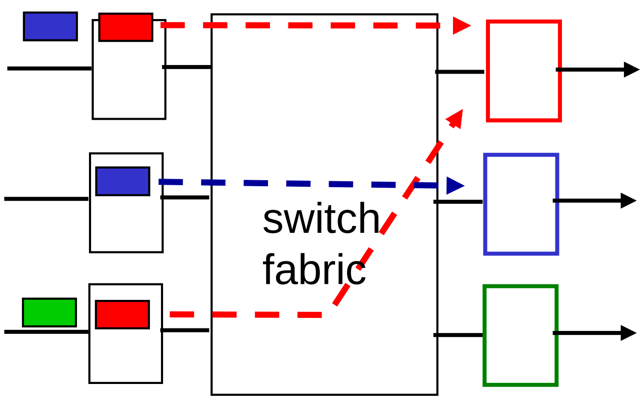
Switching via a bus

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

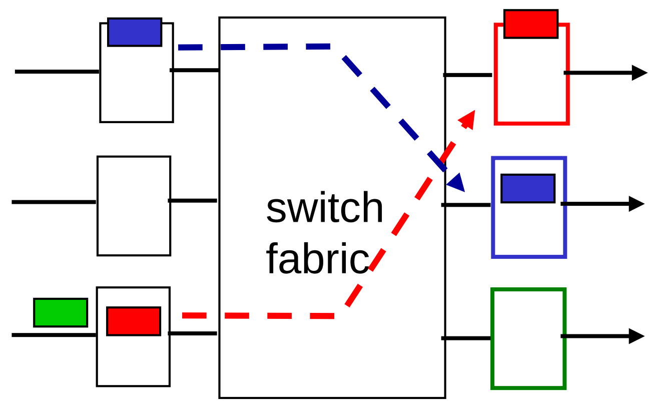


Input port queuing

- 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



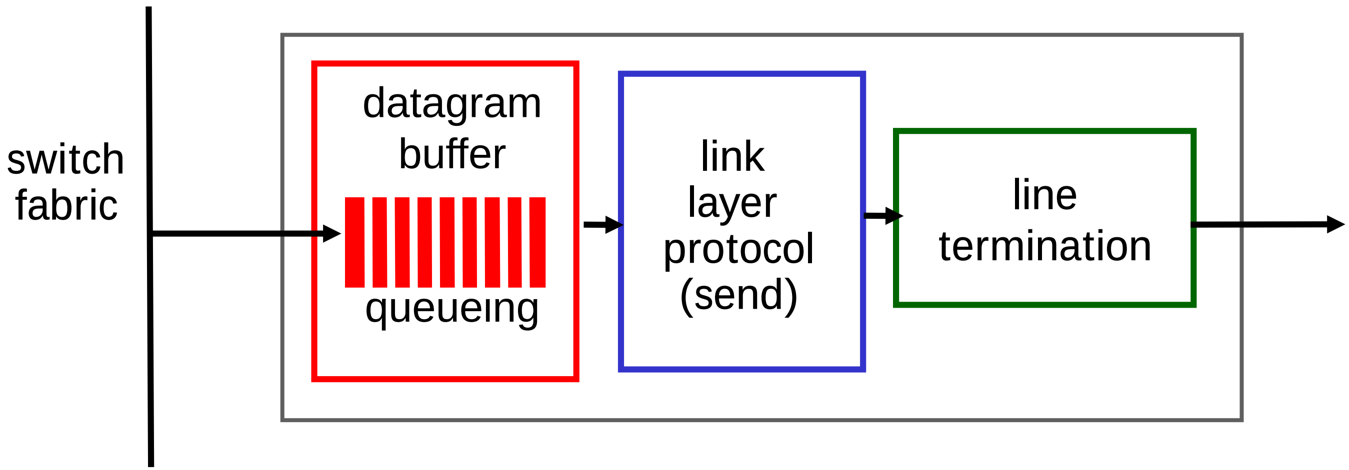
output port contention:
only one red datagram can be transferred.
lower red packet is blocked



one packet time later:
green packet experiences HOL blocking

Output ports

This slide in HUGELY important!



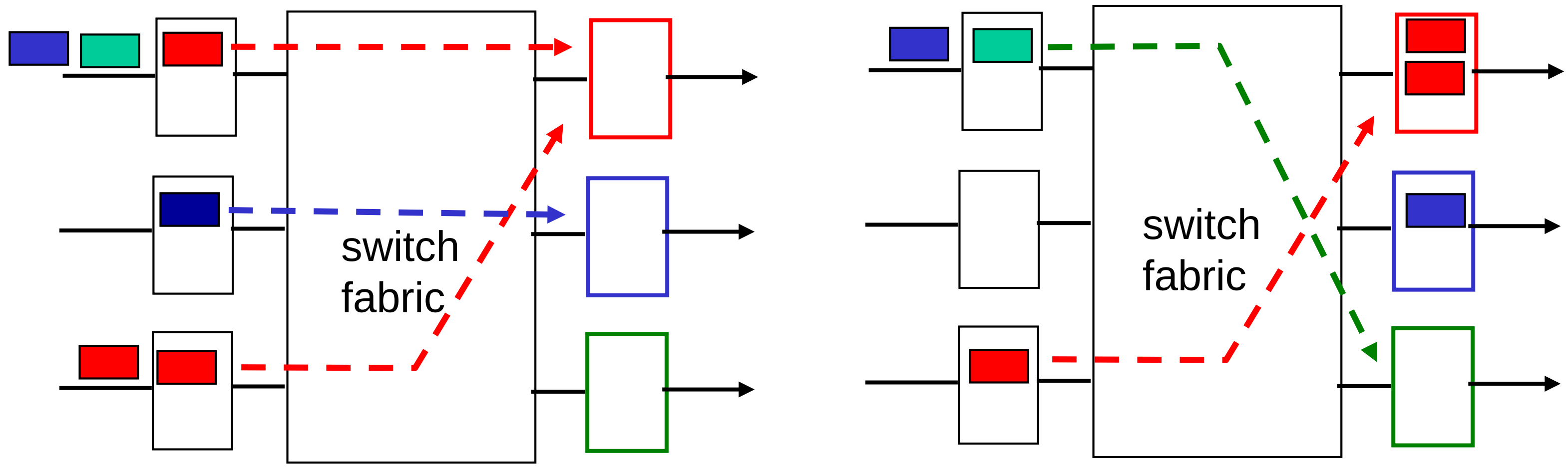
- *buffering* required
  from fabric faster
  rate

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

- *scheduling*
  datagrams

Priority scheduling – who gets best performance, network neutrality

Output port queueing



at *t*, packets more from input to output

one packet time later

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

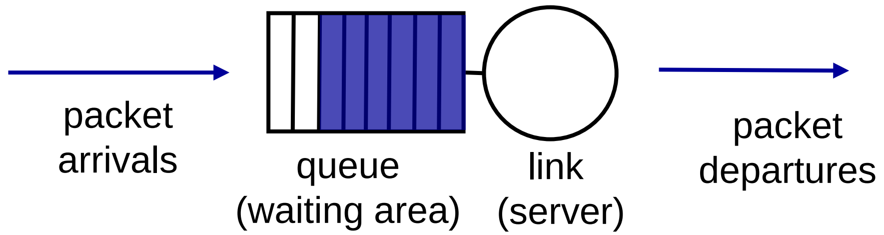
How much buffering?

- 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

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

Scheduling policies

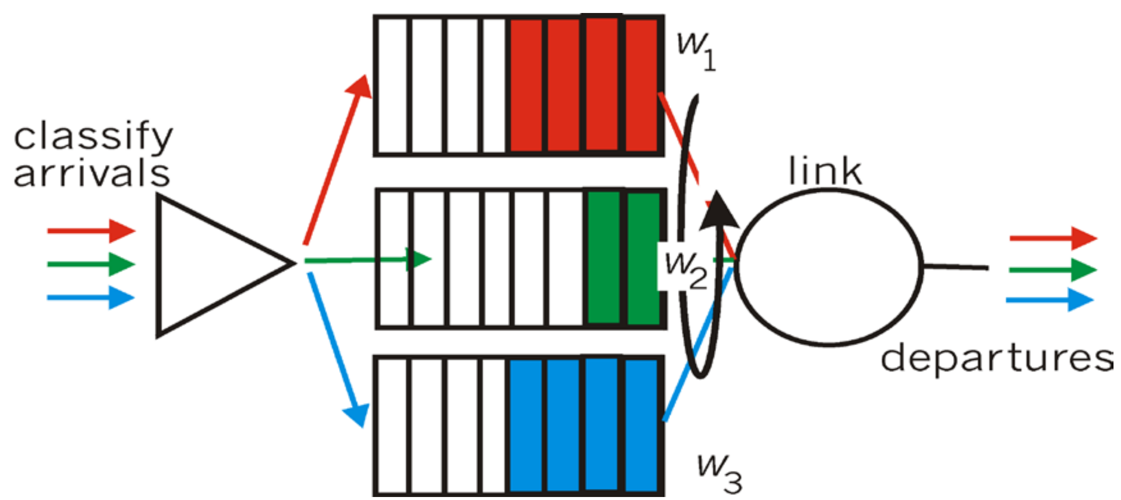
- *scheduling:* choose next packet to send on link
- *FIFO (first in first out) scheduling:* send in order of arrival to queue
  - real-world example?
  - *discard policy:* if packet arrives to full queue: who to discard?
    - *tail drop:* drop arriving packet
    - *priority:* drop/remove on priority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



Scheduling policies

Weighted Fair Queuing (WFQ):

- generalized Round Robin
- each class gets weighted amount of service in each cycle



Before You Go

On a sheet of paper, answer the following (ungraded) question (no names, please):

What one or two possible improvements to the way the class is being taught would make the most difference?