INTRODUCTION & REVIEW
INTERNET PROTOCOL AND SERVICES

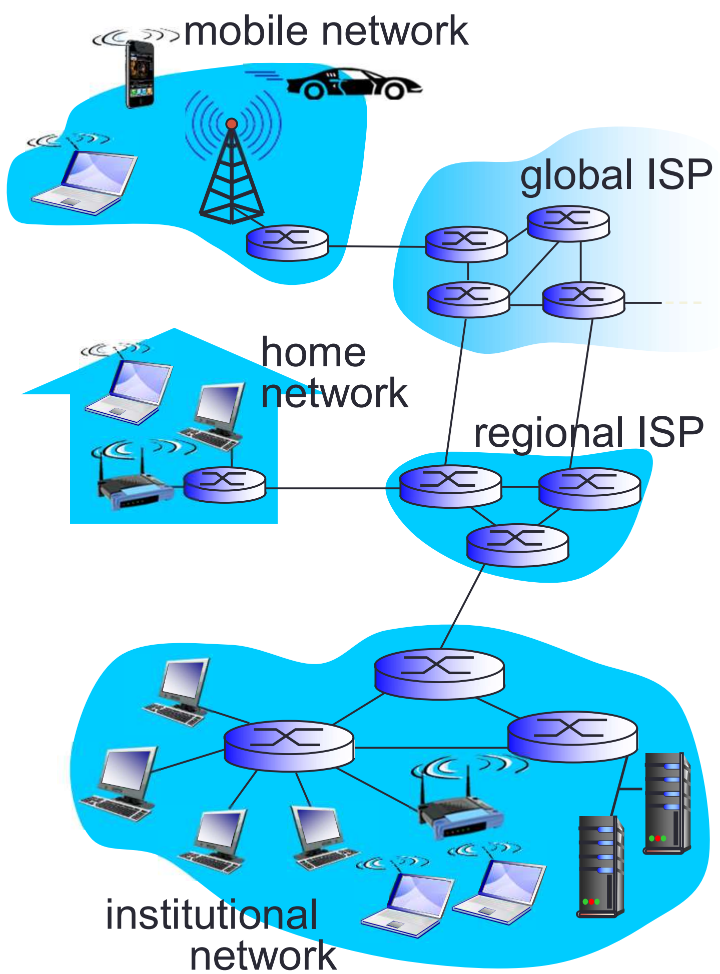
Some slides have been taken from: *Computer Networking: A Top Down Approach Featuring the Internet*, 3rd edition. Jim Kurose, Keith Ross. Addison-Wesley, July 2004.

Contents

- Internet protocol stack
- Application layer
- TCP & UDP
- Internet layer

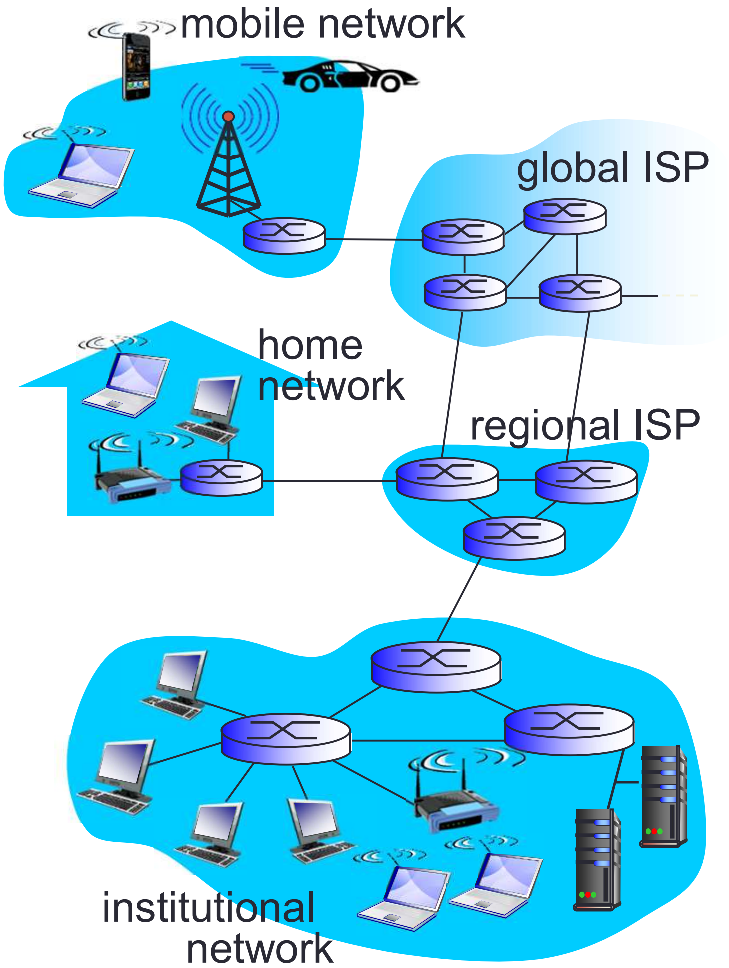
What's the Internet?

- *Internet:* "network of networks"
  - Interconnected ISPs
- *protocols* control sending, receiving of msgs
  - e.g., TCP, IP, HTTP, Skype, 802.11
- *Internet standards*
  - RFC: Request for comments
  - IETF: Internet Engineering Task Force



What’s the Internet?

- *Infrastructure that provides services to applications:*
  - Web, VoIP, email, games, e-commerce, social nets, …
- *provides programming interface to apps*
  - hooks that allow sending and receiving app programs to “connect” to Internet
  - provides service options, analogous to postal service



What's a protocol?

human protocols:

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

… specific msgs sent

… specific actions taken when msgs received, or other events

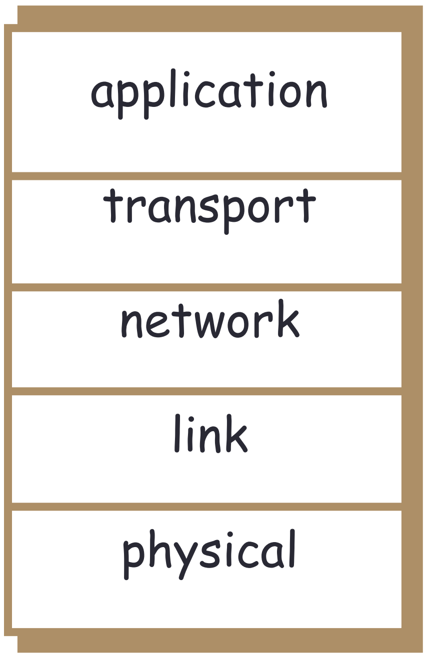
network protocols:

- machines rather than humans
- all communication activity in Internet governed by protocols

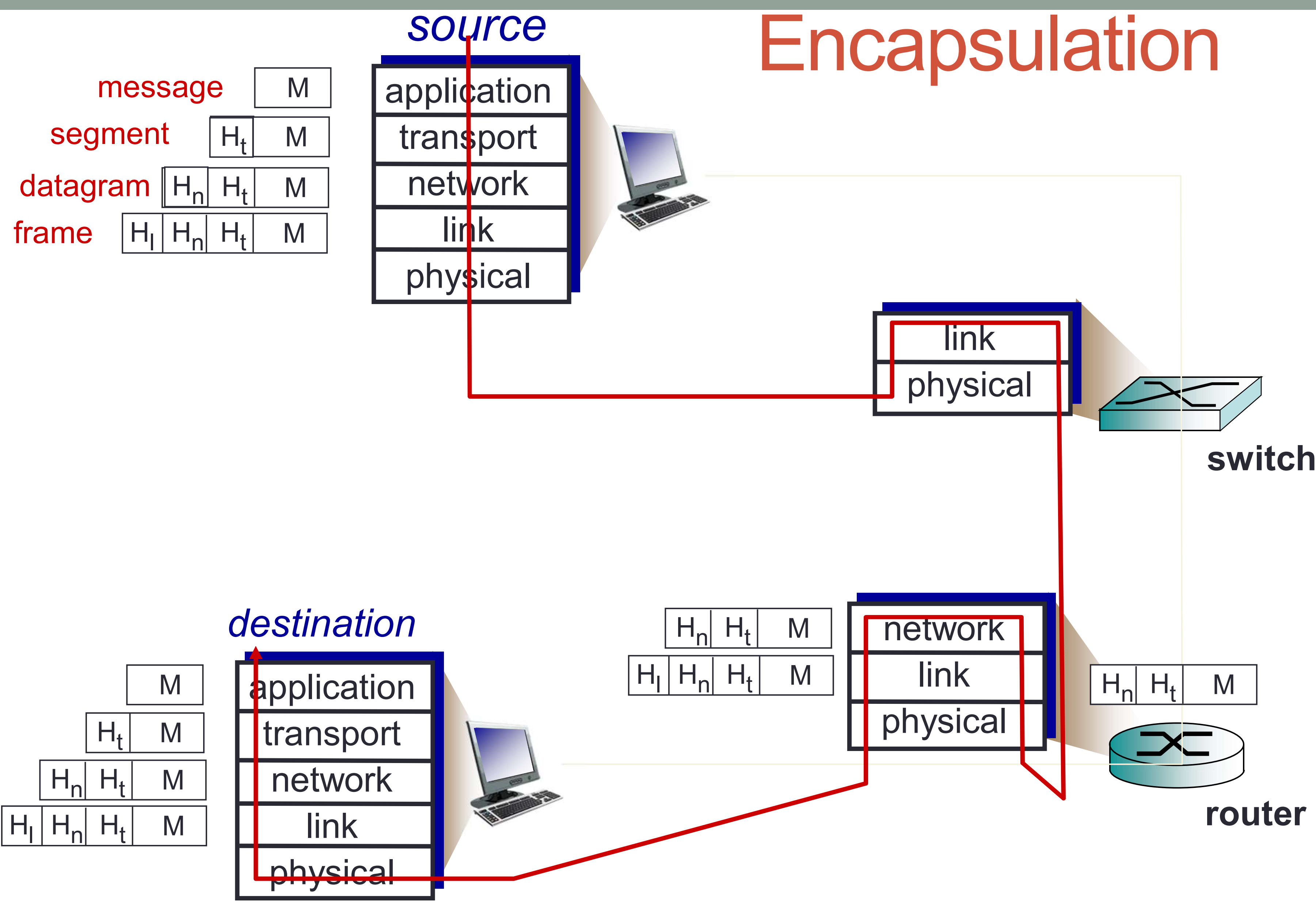
protocols define format, order of msgs sent and received among network entities, and actions taken on msg transmission, receipt

TCP/IP protocol stack

- application: supporting network applications
  - FTP, SMTP, STTP
- transport: host-host data transfer
  - TCP, UDP
- network: routing of datagrams from source to destination
  - IP, routing protocols
- link: data transfer between neighboring network elements
  - PPP, Ethernet
- physical: bits “on the wire”



Encapsulation



Application layer

- E-mail
- Web
- Instant messaging
- Remote login
- P2P file sharing
- Multi-user network games
- Streaming stored video clips
- Internet telephone
- Real-time video conference
- Massive parallel computing

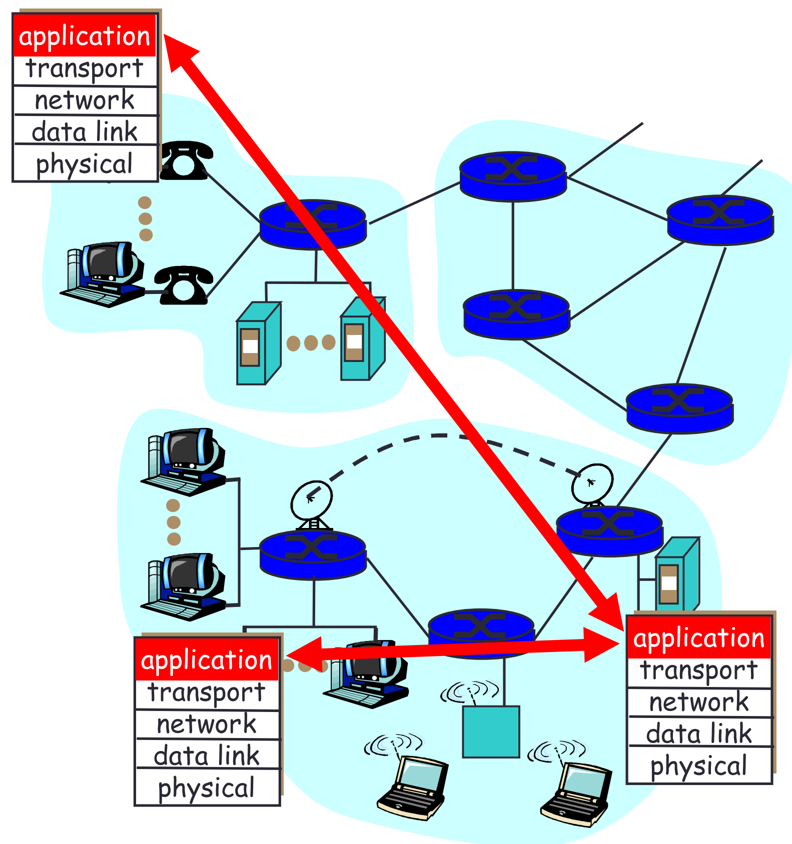
Creating a network app

Write programs that

- run on different end systems and
- communicate over a network.
- e.g., Web: Web server software communicates with browser software

No software written for devices in network core

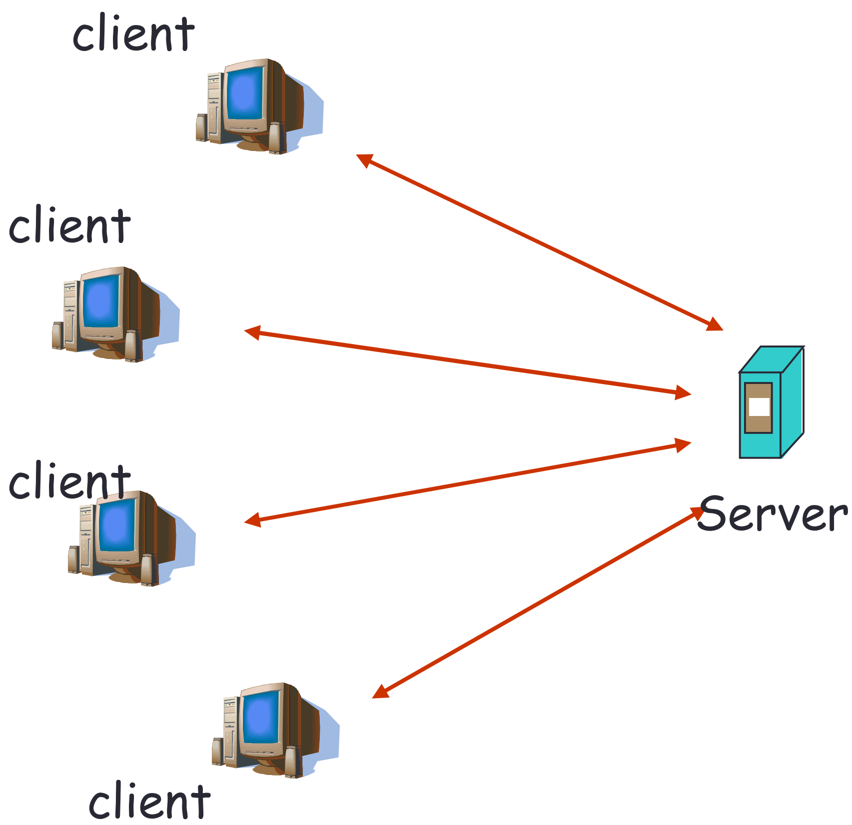
- Network core devices do not function at app layer
- This design allows for rapid app development



Application architectures

- Client-server
- Peer-to-peer (P2P)
- Hybrid of client-server and P2P

Client-server architecure



server:

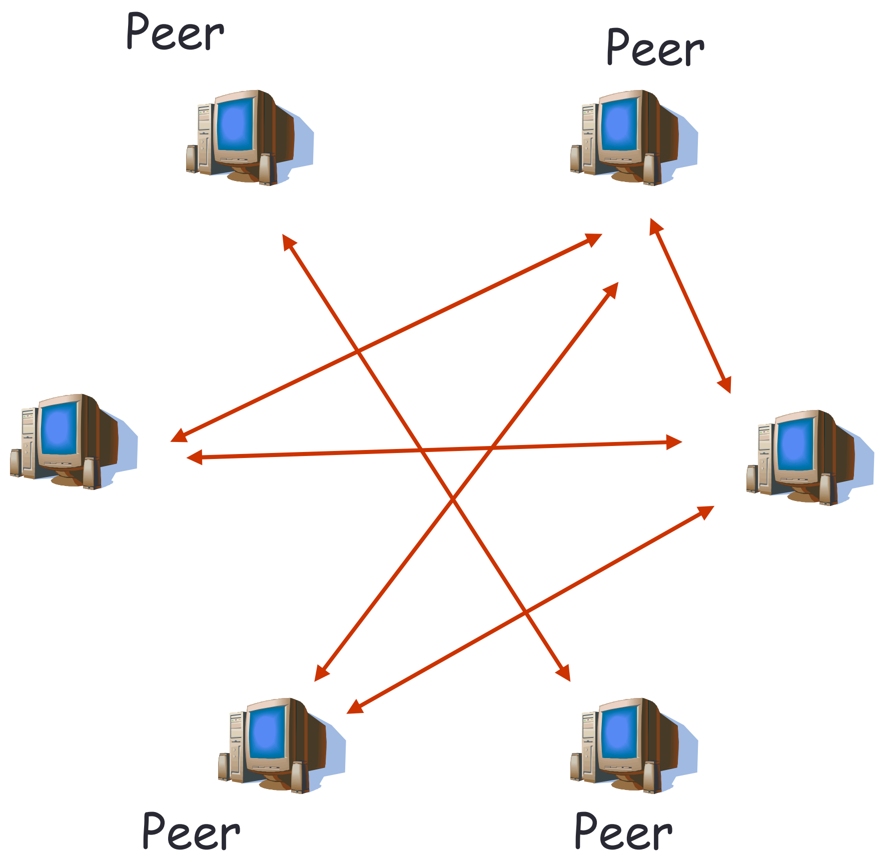
- always-on host
- permanent IP address
- server farms for scaling

clients:

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

Pure P2P architecture

- no always on server
- arbitrary end systems directly communicate
- peers are intermittently connected and change IP addresses
- example: Gnutella



Highly scalable

But difficult to manage

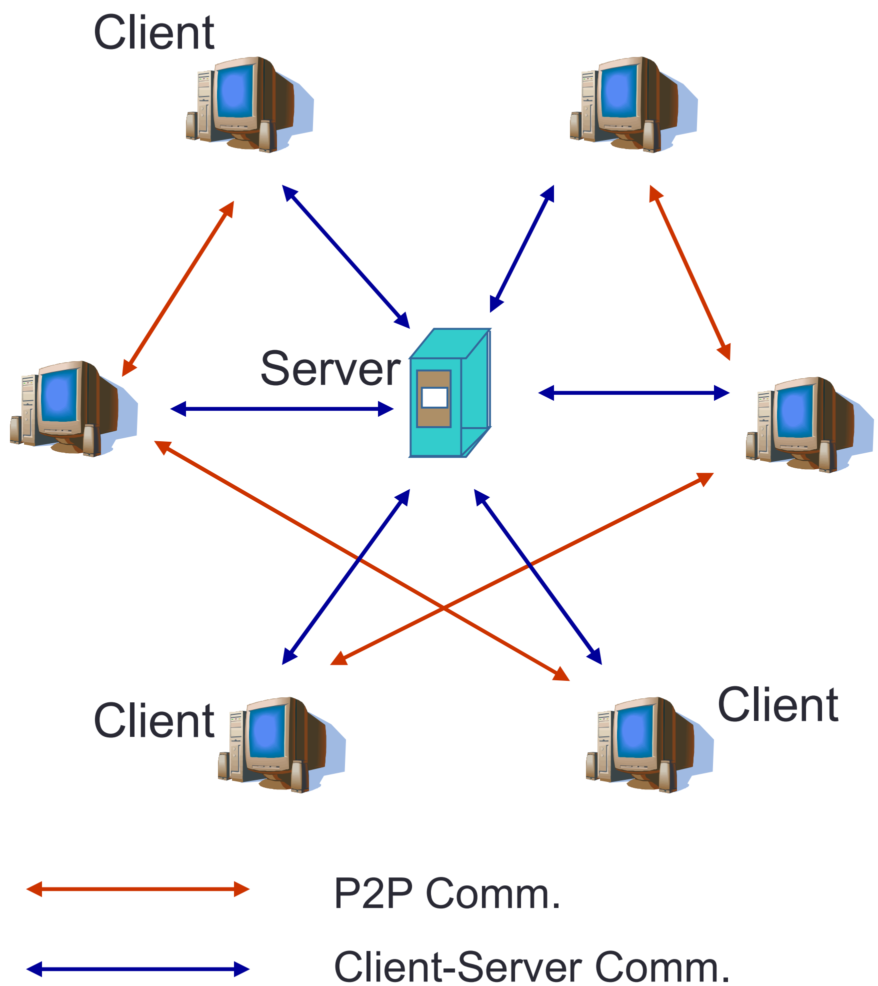
Hybrid of client-server and P2P

BitTorrent

- File transfer P2P
- File search centralized:
  - Peers register content at central server
  - Peers query same central server to locate content

Instant messaging

- Chatting between two users is P2P
- Presence detection/location centralized:
  - User registers its IP address with central server when it comes online
  - User contacts central server to find IP addresses of buddies



Processes communicating

Process: program running within a host.

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

Client process: process that initiates communication

Server process: process that waits to be contacted

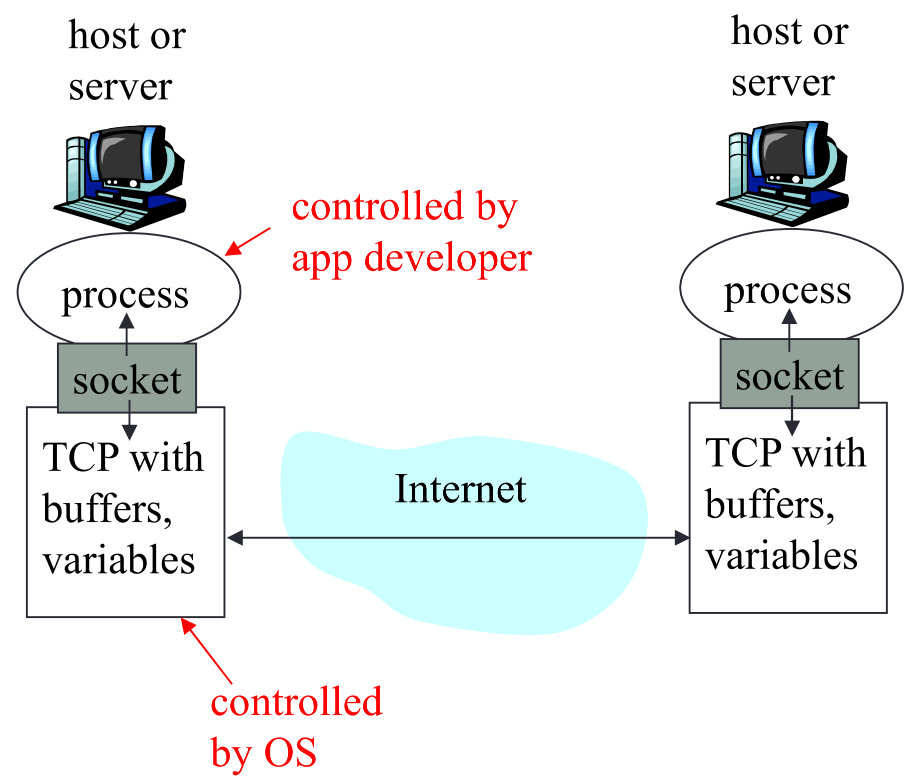
- Note: applications with P2P architectures have client processes & server processes

Sockets

- process sends/receives messages to/from its socket
- Defined by
  - Port number
  - IP Address
  - TCP/UDP

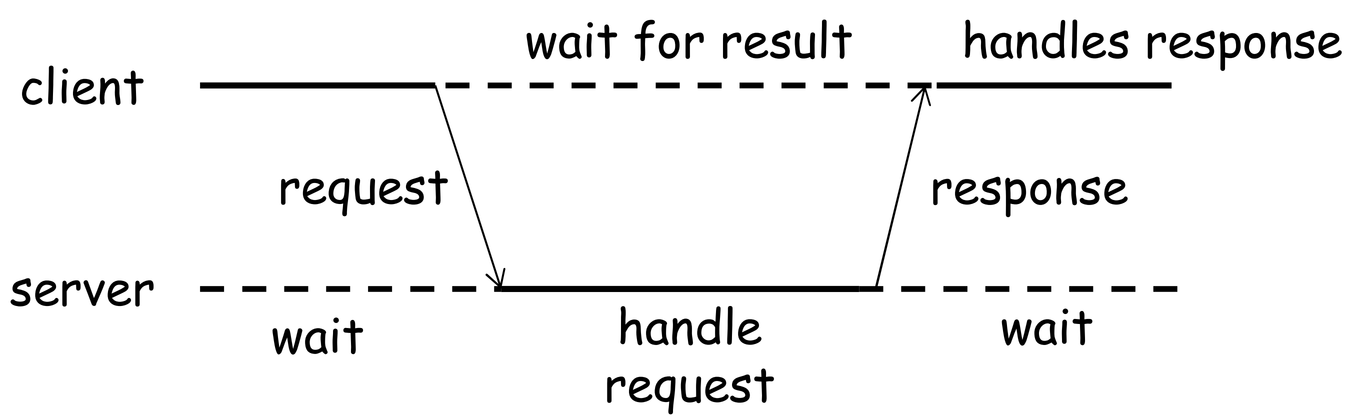
Socket address

- API: (1) choice of transport protocol; (2) ability to fix a few parameters



Processes communicating

- Client process: sends request
- Server process: replies response
- Typically: single server - multiple clients
- The server does not need to know anything about the client
- The client should always know something about the server
  - at least the socket address of the server



App-layer protocol defines

- Types of messages exchanged, eg, request & response messages
- Syntax of message types: what fields in messages & how fields are delineated
- Semantics of the fields, ie, meaning of information in fields
- Rules for when and how processes send & respond to messages

What transport service does an app need?

Data loss

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

Timing

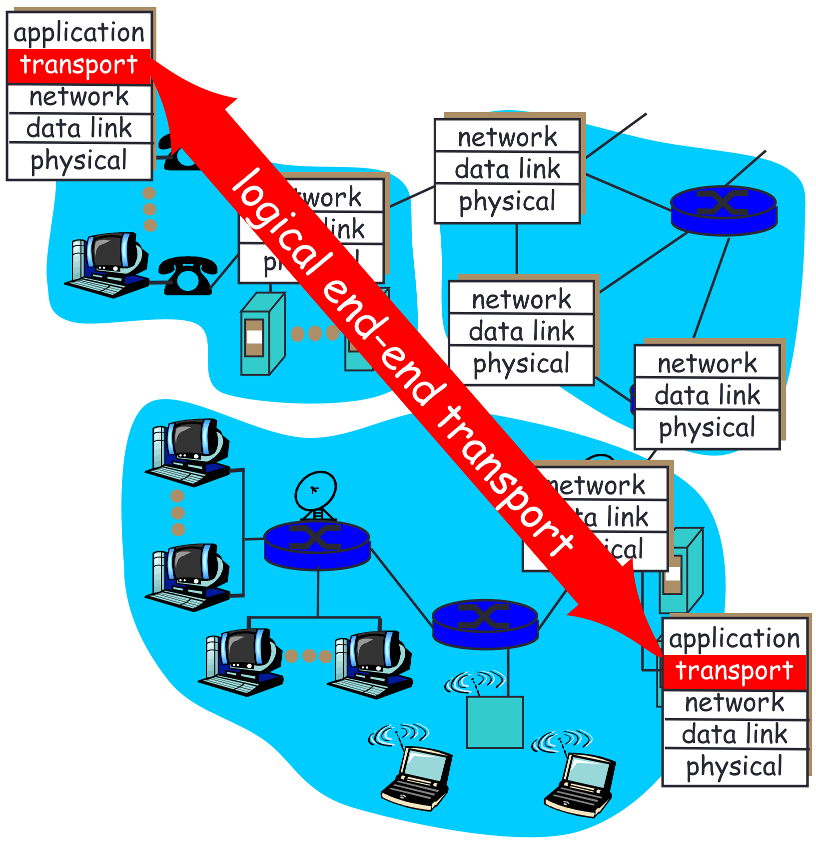
- some apps (e.g., Internet telephony, interactive games) require low delay to be "effective"

Bandwidth

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

Transport services and protocols

- provide *logical communication* between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into segments, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP



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
- *does not provide:* timing, minimum throughput guarantee, security
- *connection-oriented:* setup required between client and server processes

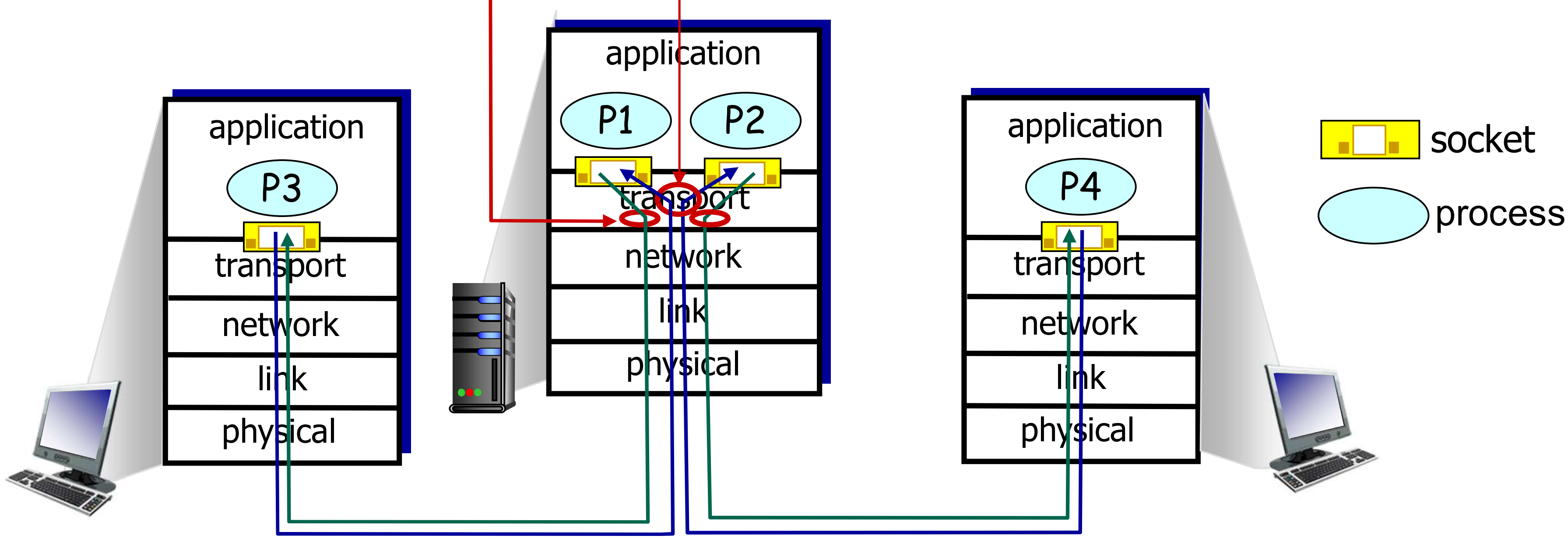
UDP service:

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

Multiplexing/demultiplexing

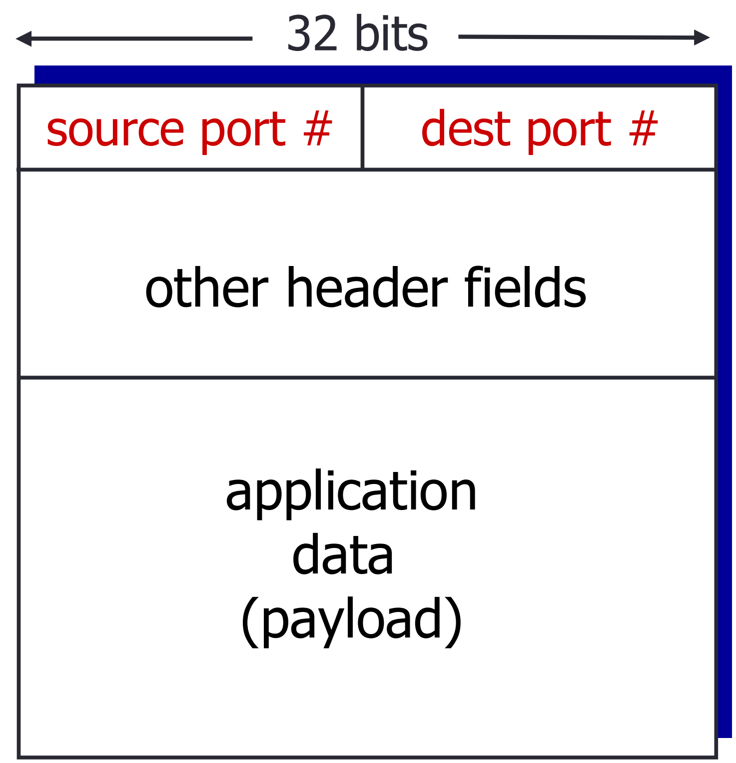
multiplexing at sender:
handle data from multiple sockets, add transport header (later used for demultiplexing)

demultiplexing at receiver:
use header info to deliver received segments to correct socket



How demultiplexing works

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



TCP/UDP segment format

UDP: User Datagram Protocol [RFC 768]

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

Why is there a UDP?

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small segment header
- no congestion control: UDP can blast away as fast as desired

UDP demultiplexing

- Create sockets with port numbers:

```
mySocket = socket(AF_INET,
  SOCK_DGRAM, 0)
```

- UDP socket identified by two-tuple:

(dest IP address, dest port number)

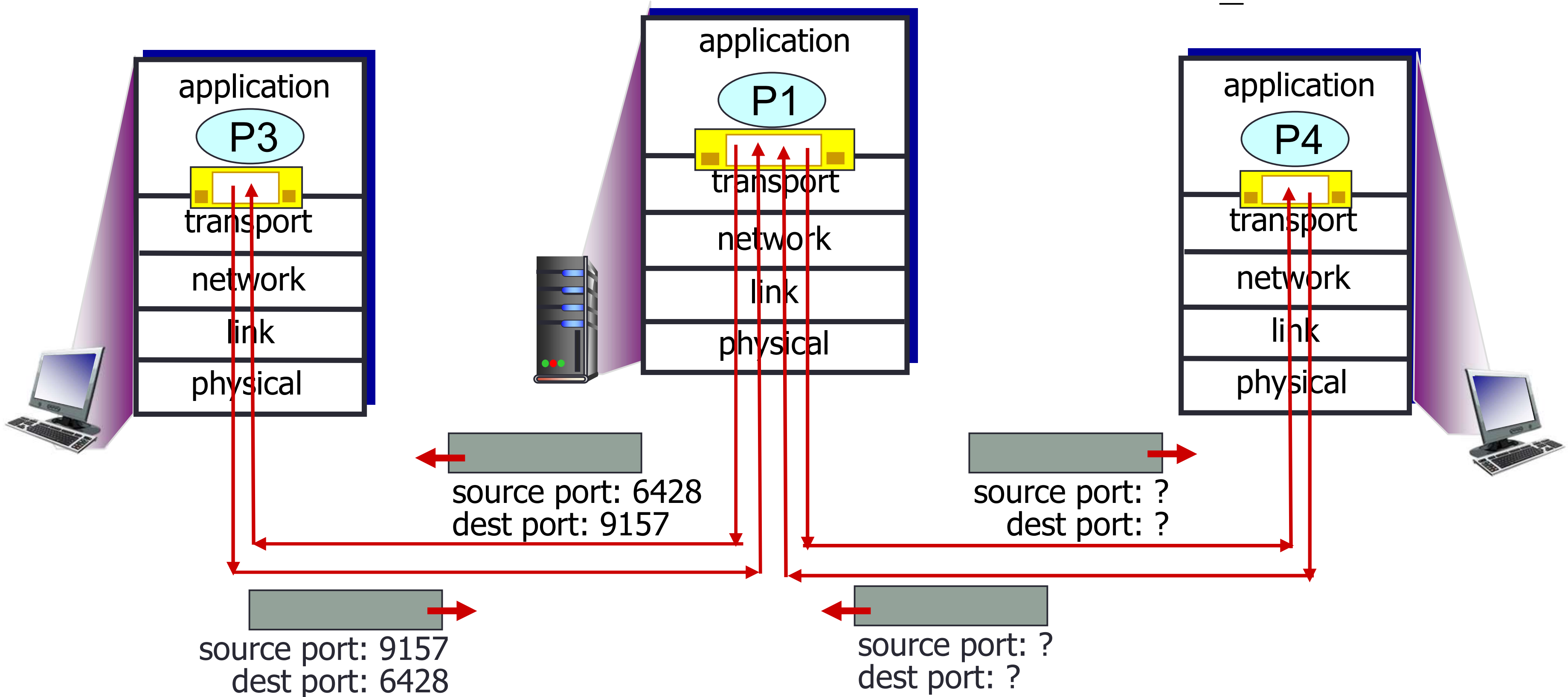
- When host receives UDP segment:
  - checks destination port number in segment
  - directs UDP segment to socket with that port number
- IP datagrams with different source IP addresses and/or source port numbers directed to same socket

UDP demux

```
mySocket =
 socket(AF_INET,
 SOCK_DGRAM, 0);
```

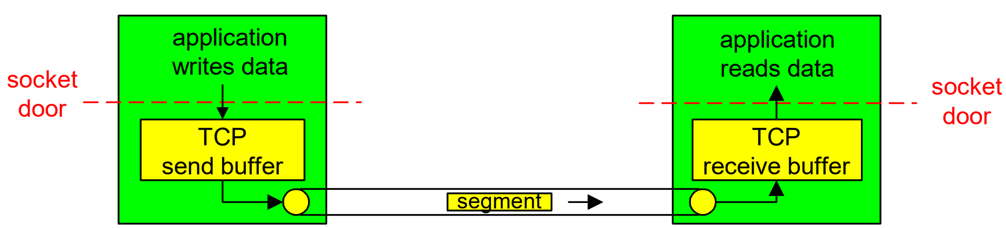
```
serverSocket =
  socket(AF_INET,
  SOCK_DGRAM, 0);
bind(…)
```

```
mySocket =
 socket(AF_INET,
 SOCK_DGRAM, 0);
```



TCP: Overview RFCs: 793, 1122, 1323, 2018, 2581

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - no "message boundaries"
- pipelined:
  - TCP congestion and flow control set window size
- *send & receive buffers*



- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control msgs) init's sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP Connection Management: Setup

Recall: TCP sender, receiver establish "connection" before exchanging data segments

- initialize TCP variables:
  - seq. #s
  - buffers, flow control info (e.g. **`RcvWindow`**)
- *client:* connection initiator
  - connect()
- *server:* contacted by client
  - accept()

Three way handshake:

Step 1: client host sends TCP SYN segment to server

- specifies initial seq #
- no data

Step 2: server host receives SYN, replies with SYNACK segment

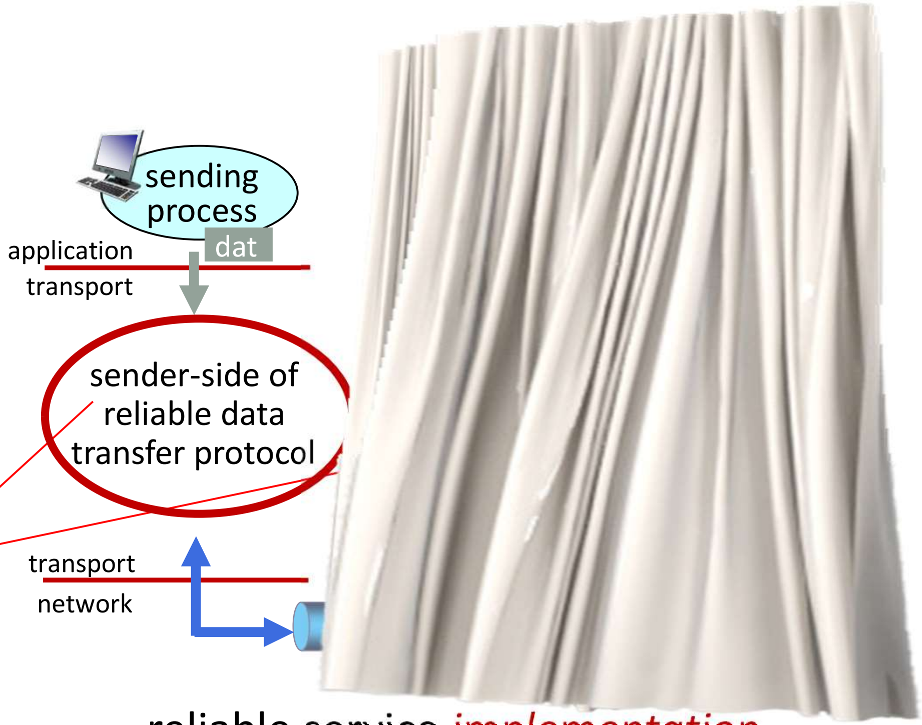
- server allocates buffers
- specifies server initial seq. #

Step 3: client receives SYNACK, replies with ACK segment, which may contain data

Principles of reliable data transfer

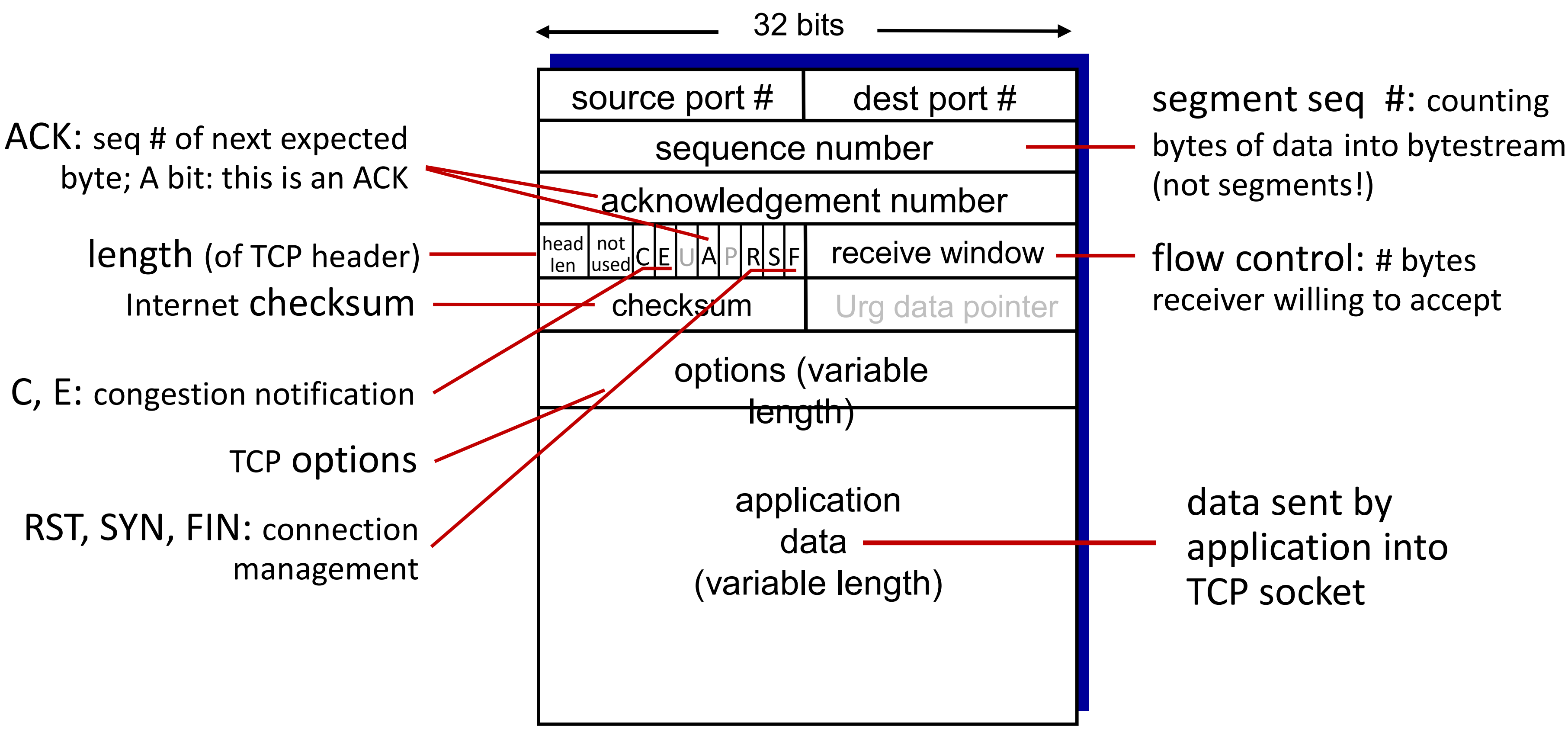
Sender, receiver do *not* know the “state” of each other, e.g., was a message received?

- unless communicated via a message



reliable service *implementation*

TCP segment structure



TCP sequence numbers, ACKs

Sequence numbers:

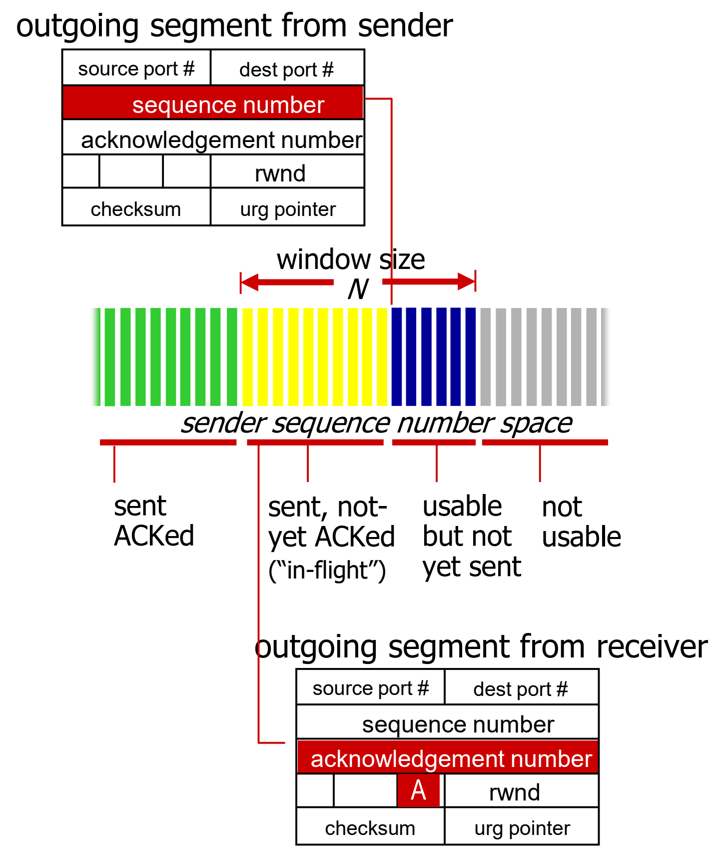
- byte stream "number" of first byte in segment's data

Acknowledgements:

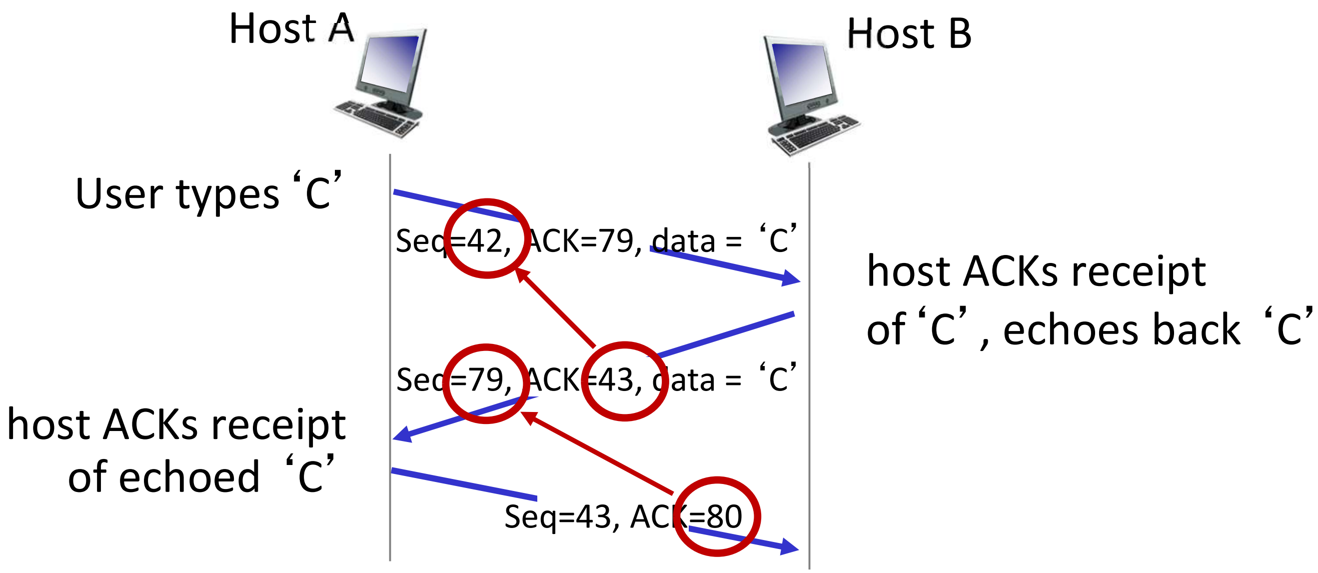
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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

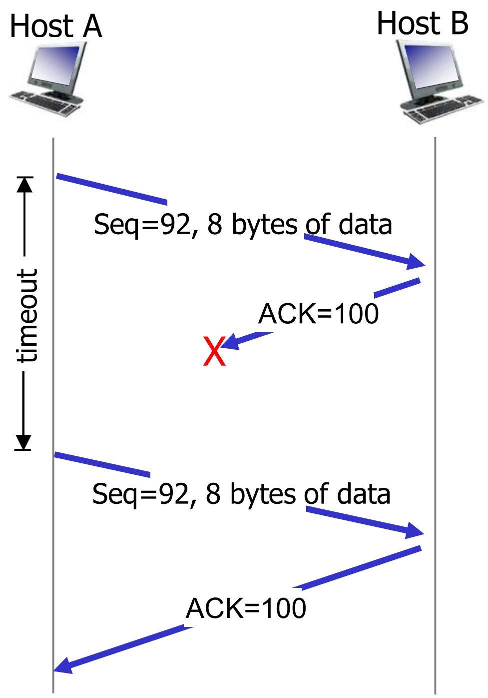


TCP sequence numbers, ACKs

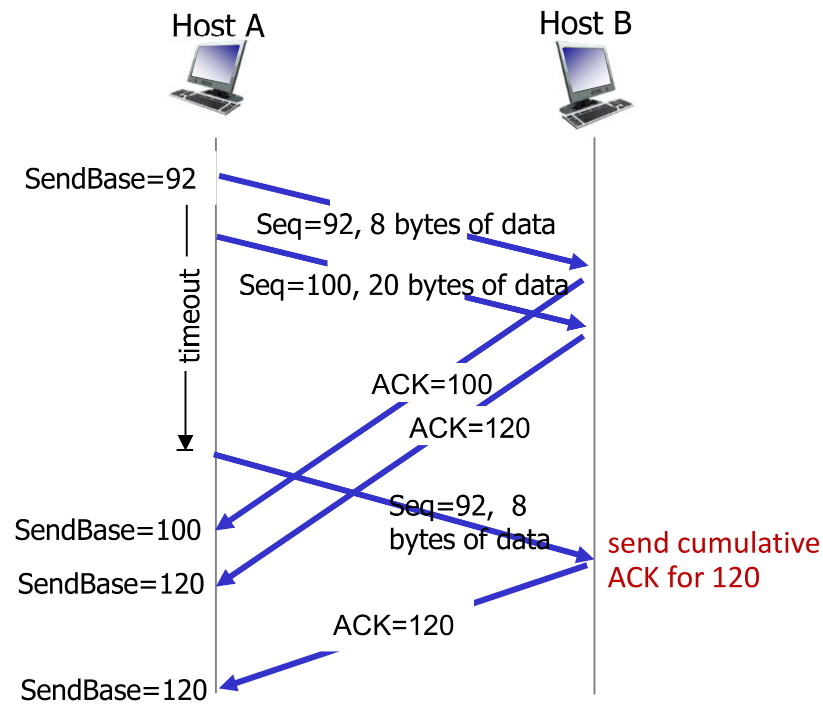


simple telnet scenario

TCP: retransmission scenarios

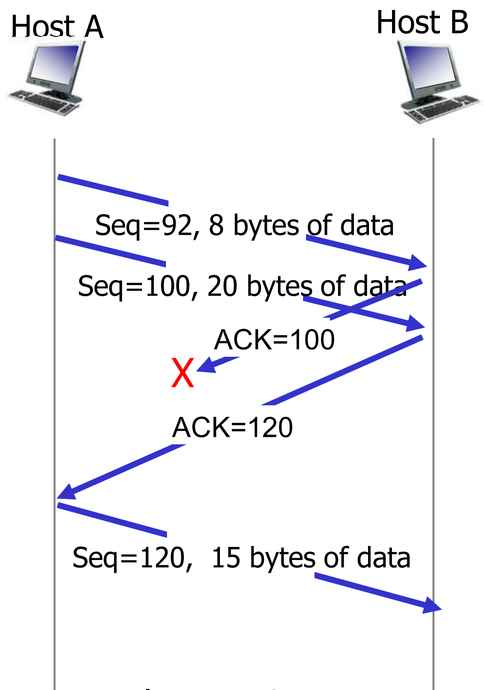


lost ACK scenario



premature timeout

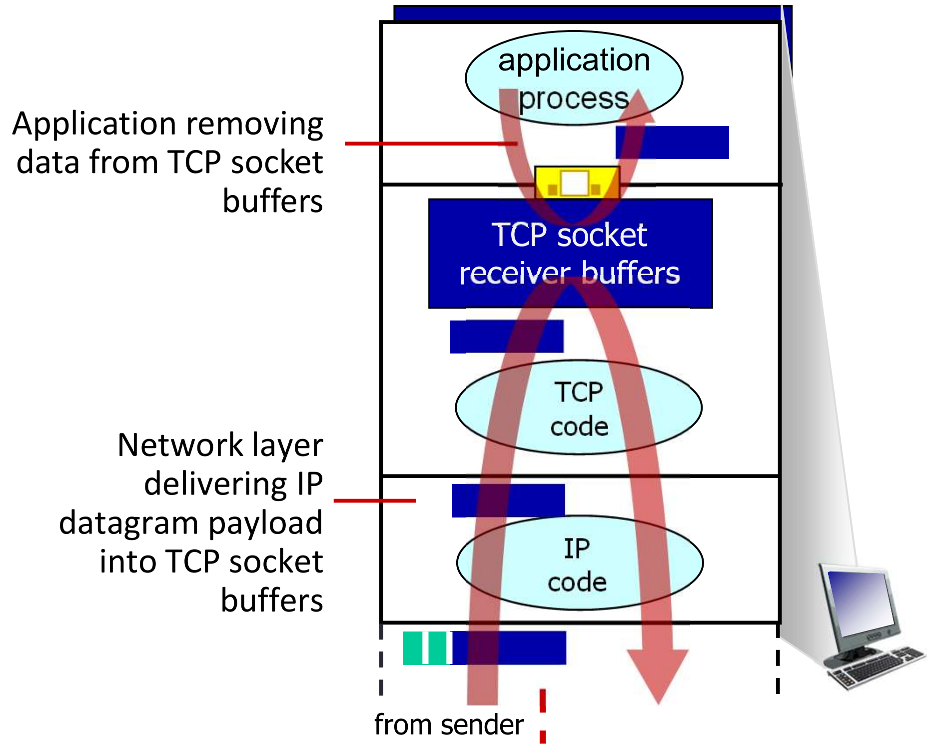
TCP: retransmission scenarios



cumulative ACK covers
for earlier lost ACK

TCP flow control

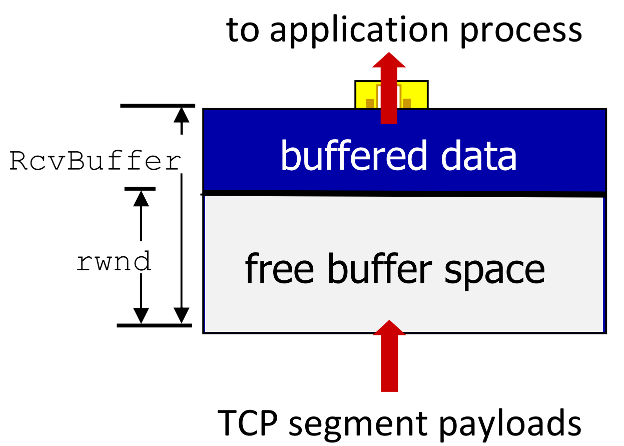
Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?



receiver protocol stack

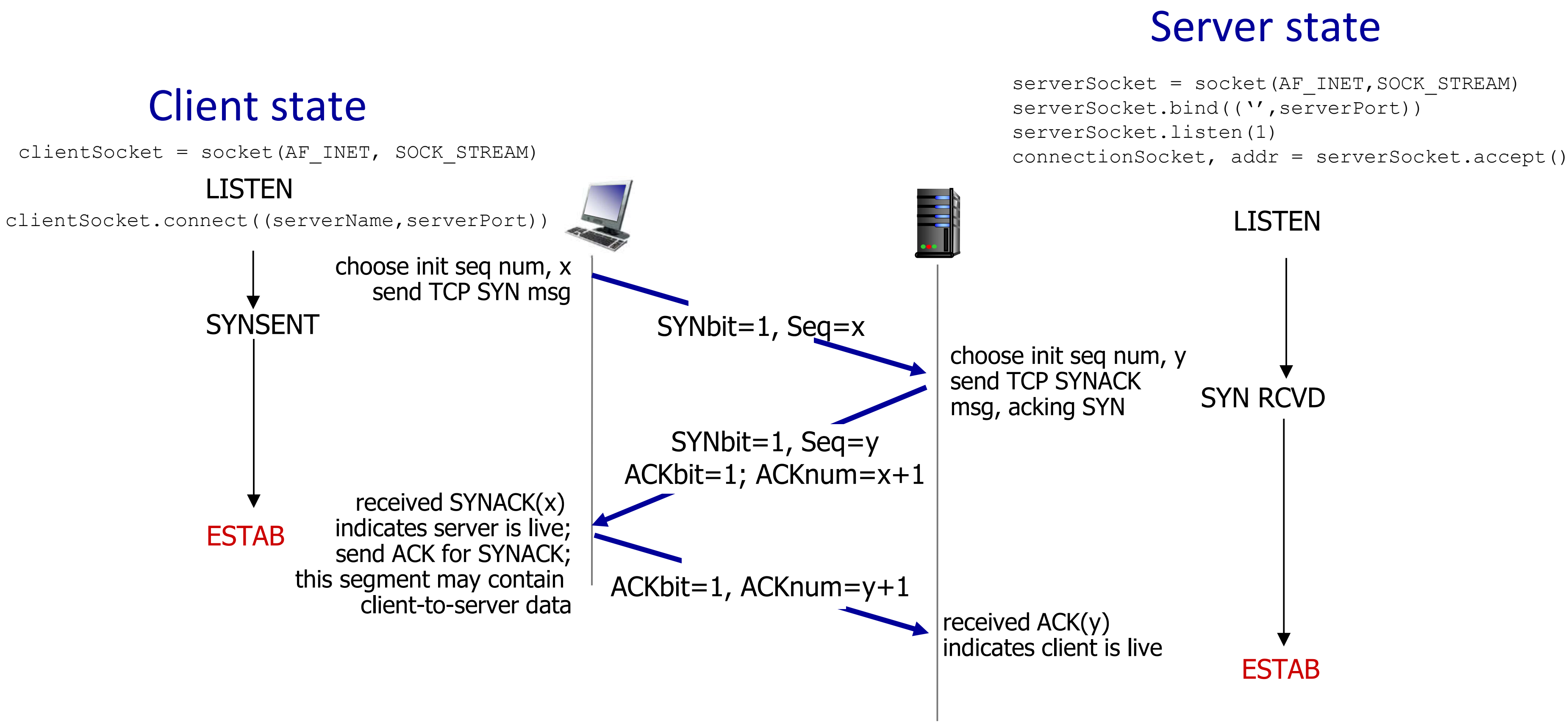
TCP flow control

- 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 autoadjust `RcvBuffer`
- sender limits amount of unACKed ("in-flight") data to received `rwnd`
- guarantees receive buffer will not overflow



TCP receiver-side buffering

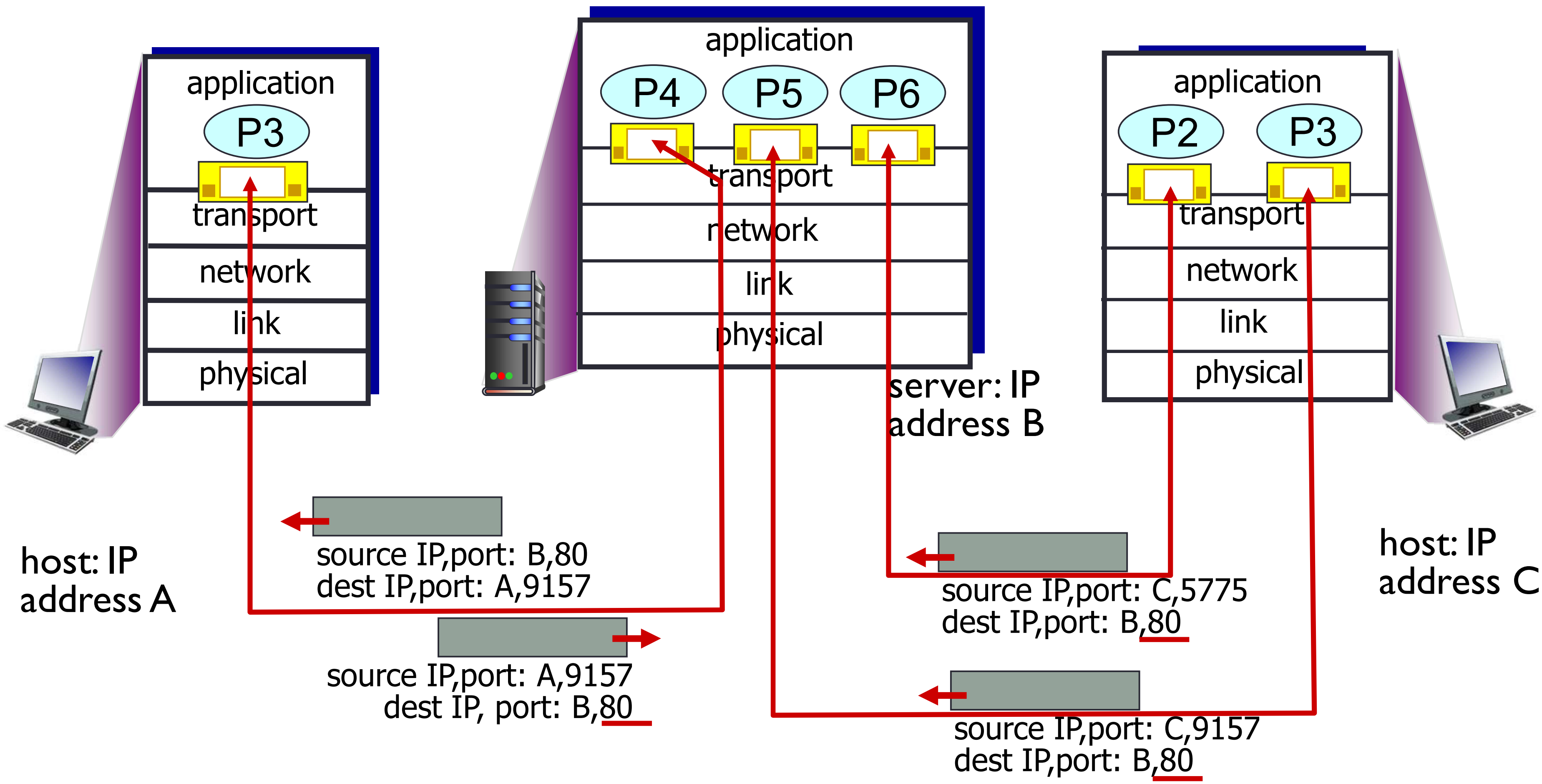
TCP 3-way handshake



Connection-oriented demux

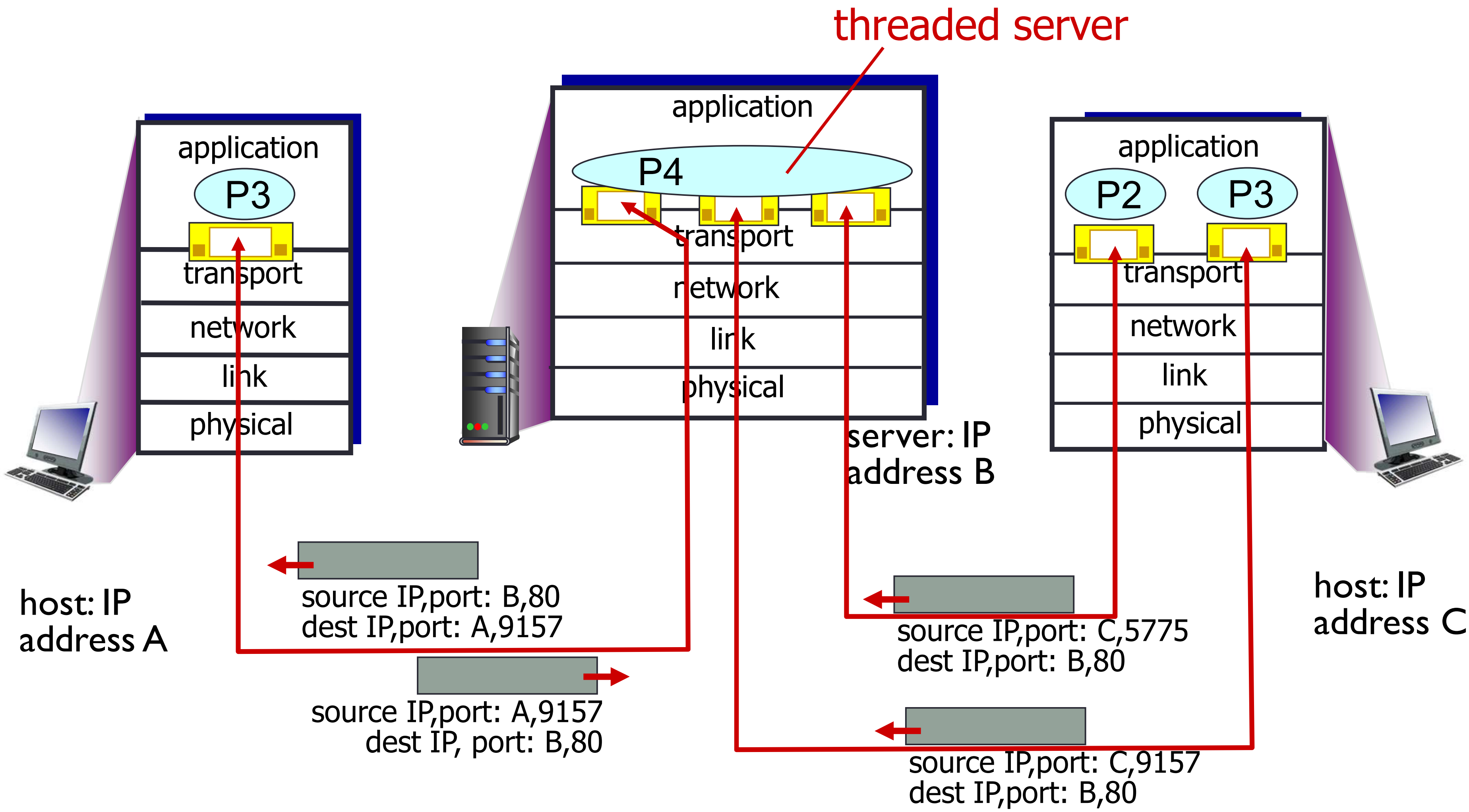
- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- recv host uses all four values to direct segment to appropriate socket
- Server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- Web servers have different sockets for each connecting client

Connection-oriented demux: example



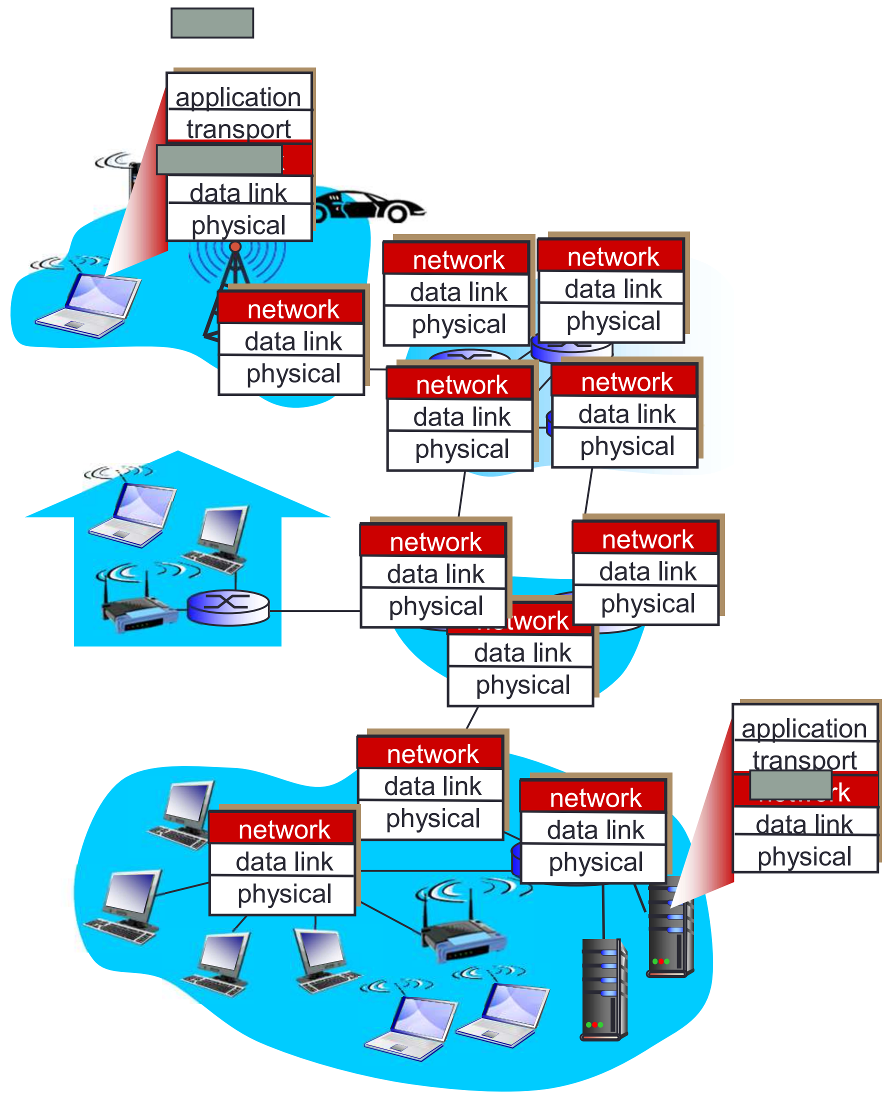
three segments, all destined to IP address: B,
dest port: 80 are demultiplexed to *different* sockets

Connection-oriented demux: example



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



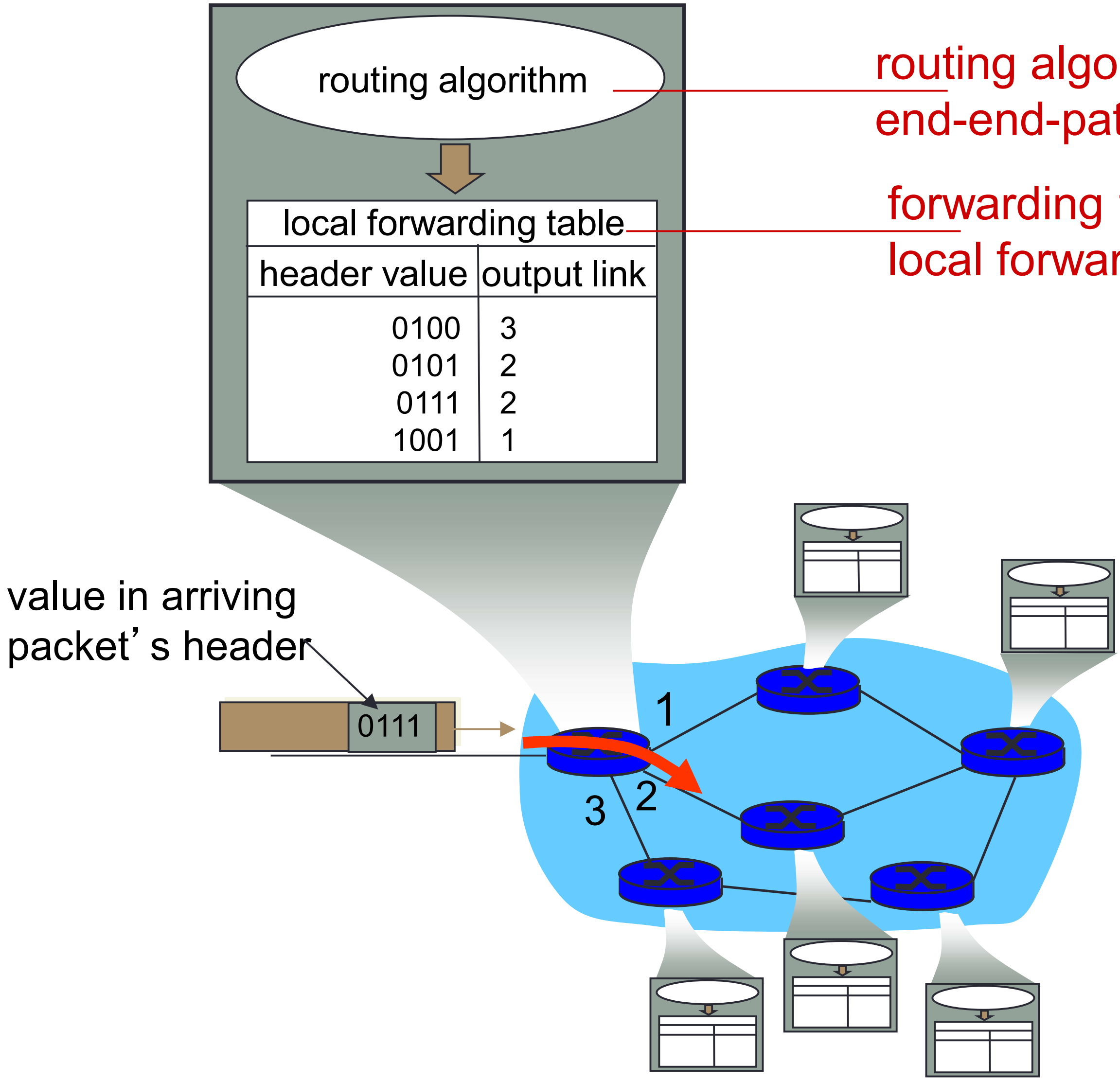
Two key network-layer functions

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

analogy:

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

Interplay between routing and forwarding

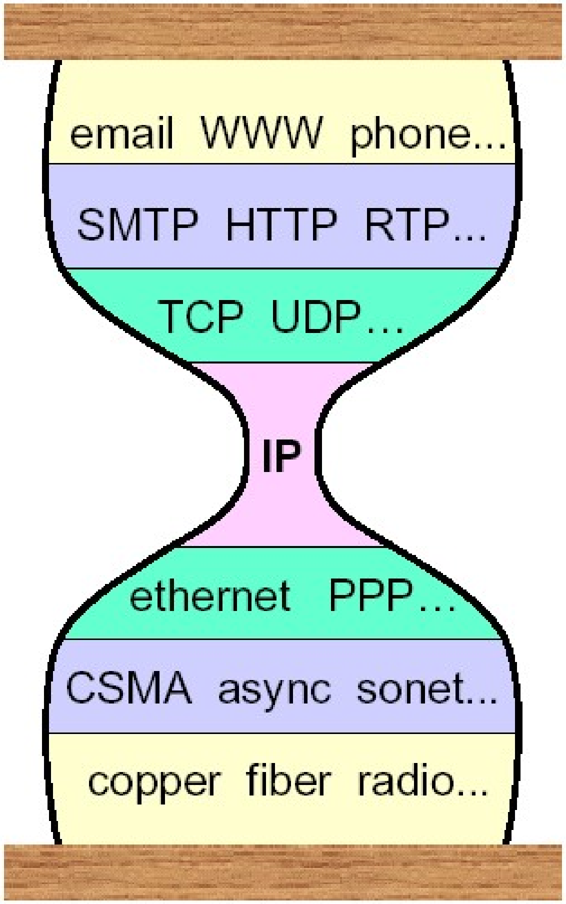


routing algorithm determines end-end-path through network

forwarding table determines local forwarding at this router

Why an internet layer?

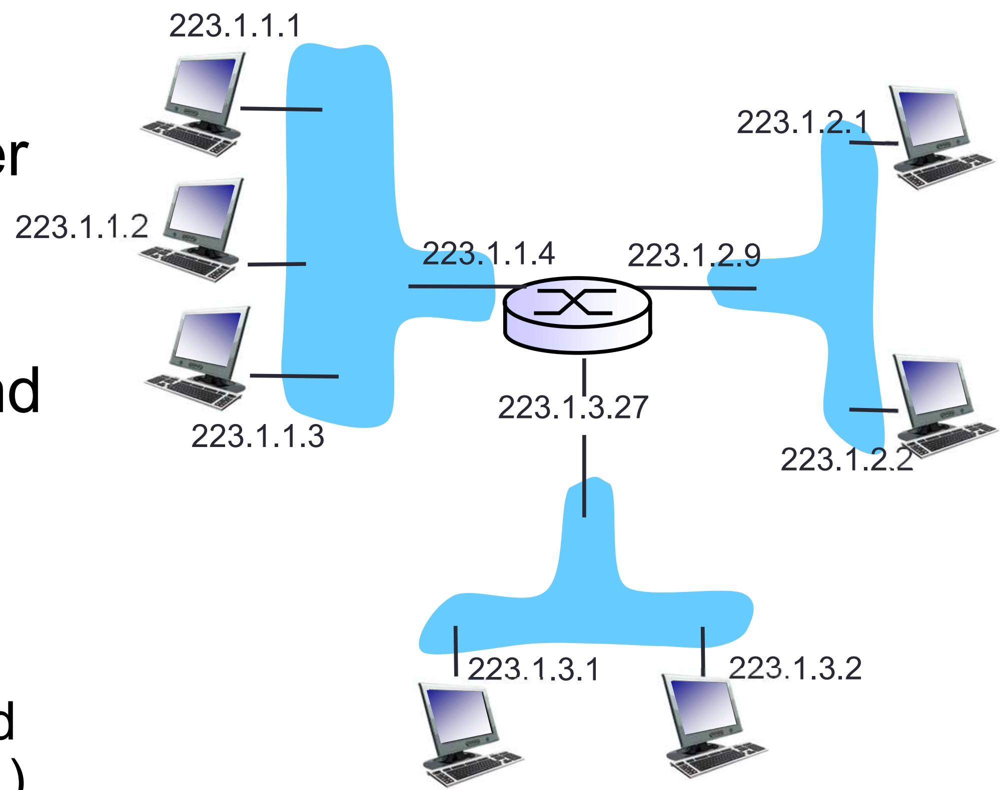
- Why not one big flat LAN?
  - Different LAN protocols
  - Flat address space not scalable
- IP provides:
  - Global addressing
  - Scaling to WANs
  - Virtualization of network isolates end-to-end protocols from network details/changes



"hourglass model"
(Steve Deering)

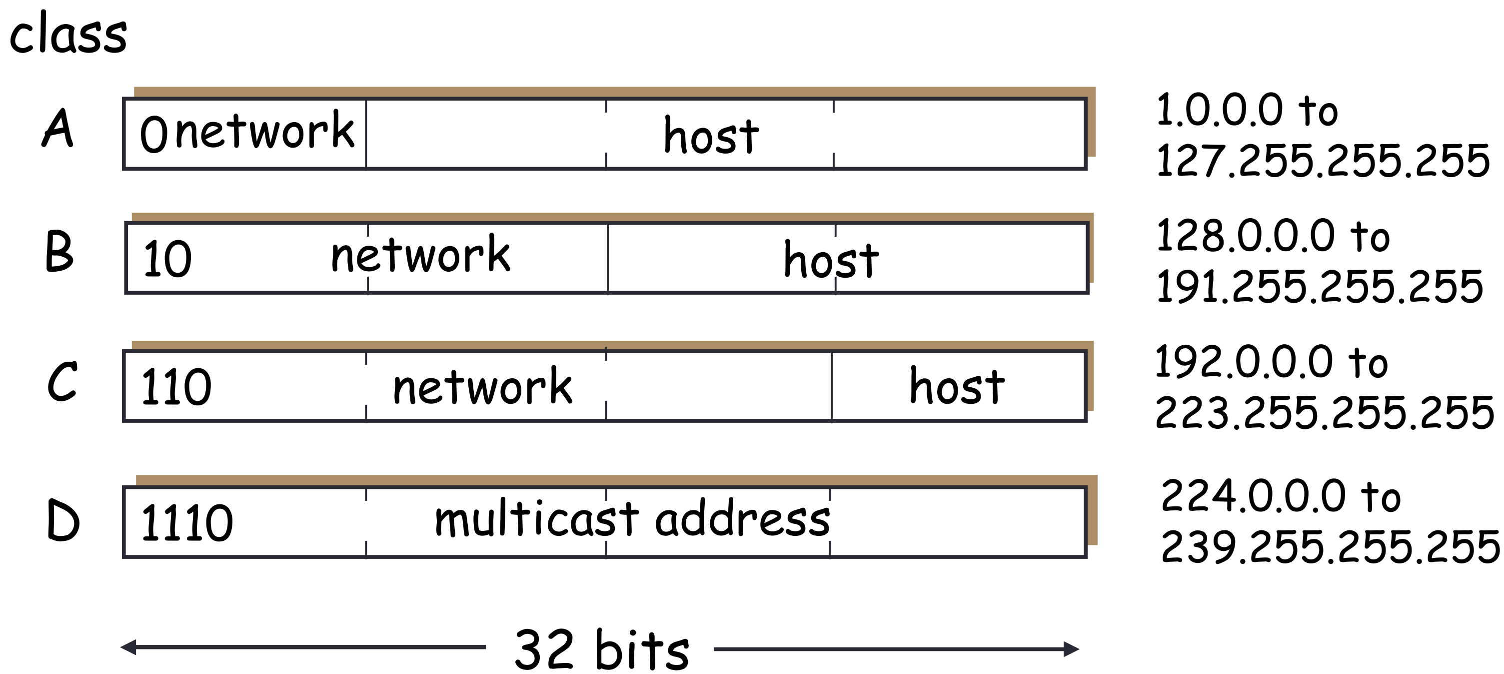
IP addressing: introduction

- *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*



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

IP addressing: “class-full”



- Classful addressing:
  - inefficient use of address space, address space exhaustion
  - e.g., class B net allocated enough addresses for 65K hosts, even if only 2K hosts in that network

IP addressing: “class-less”

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

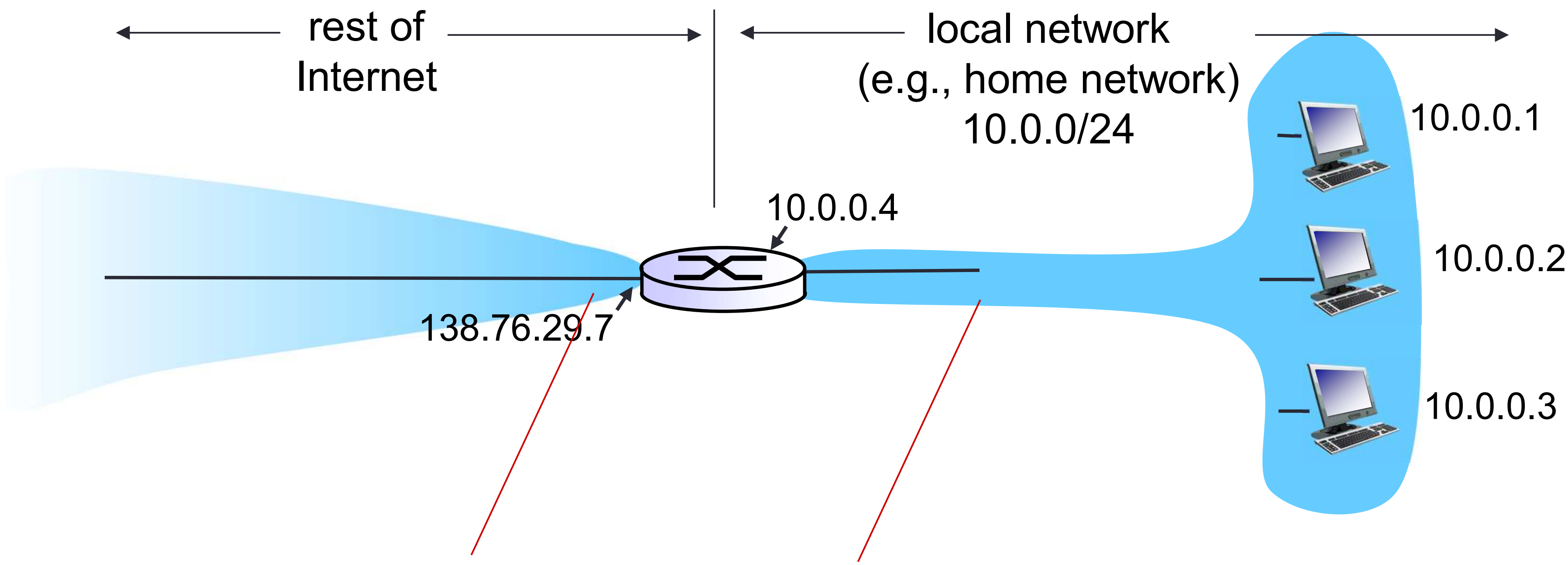
Address Allocation for Private Internets

- RFC1918

Private address	10.0.0.0/8 172.16.0.0/16 → 172.31.0.0/16 192.168.0.0/24 → 192.168.255.0 /24
Loopback address	127.0.0.0 /8
Multicast address	224.0.0.0 ～239.255.255.255

- Link local address: 169.254.0.0/16

NAT: network address translation



all datagrams *leaving* local network have *same* single source NAT IP address: 138.76.29.7,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

motivation: local network uses just one IP address as far as outside world is concerned:

- range of addresses not needed from ISP: just one IP address for all devices
- can change addresses of devices in local network without notifying outside world
- can change ISP without changing addresses of devices in local network
- devices inside local net not explicitly addressable, visible by outside world (a security plus)

NAT: network address translation

implementation: NAT router must:

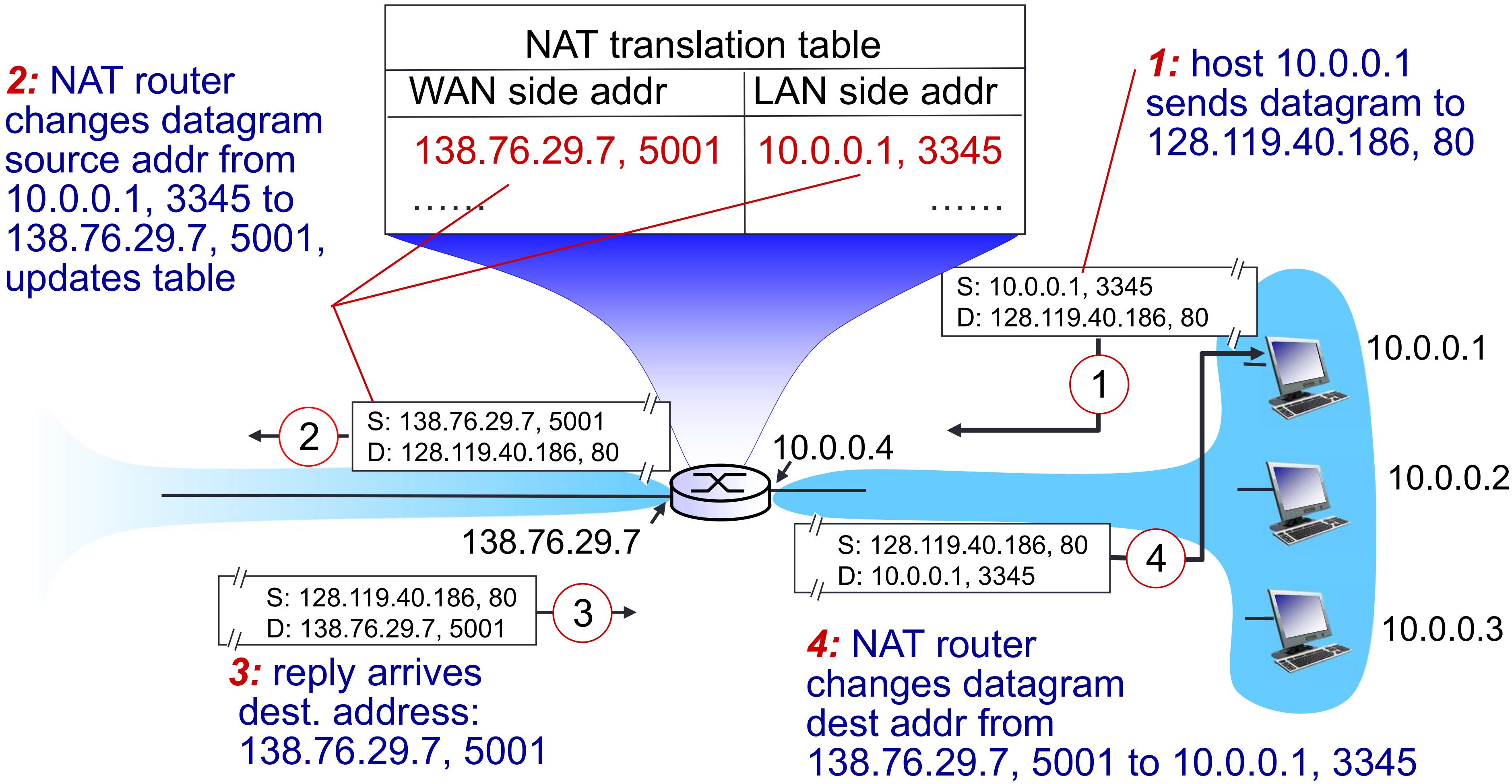
- *outgoing datagrams: replace* (source IP address, port #) of every outgoing datagram to (NAT IP address, new port #)

  ... remote clients/servers will respond using (NAT IP address, new port #) as destination addr

- *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 dest fields of every incoming datagram with corresponding (source IP address, port #) stored in NAT table

NAT: network address translation

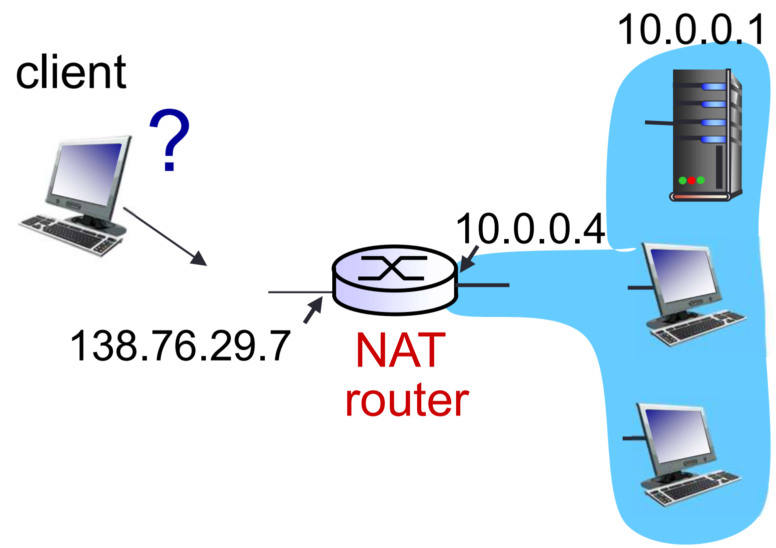


NAT: network address translation

- 16-bit port-number field:
  - 60,000 simultaneous connections with a single LAN-side address!
- NAT is controversial:
  - routers should only process up to layer 3
  - violates end-to-end argument
    - NAT possibility must be taken into account by app designers, e.g., P2P applications
  - address shortage should instead be solved by IPv6

NAT traversal problem

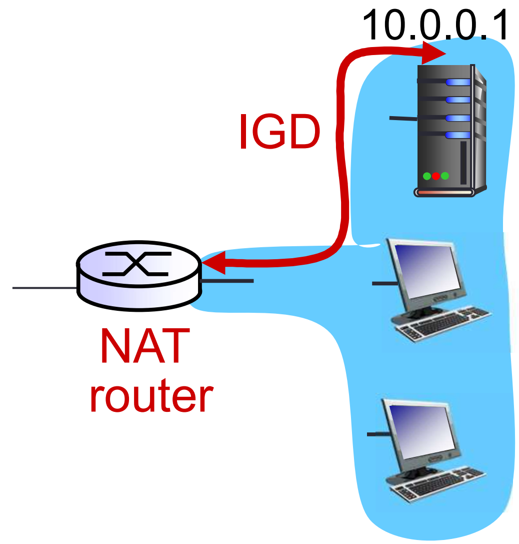
- client wants to connect to server with address 10.0.0.1
  - server address 10.0.0.1 local to LAN (client can't use it as destination addr)
  - only one externally visible NATed address: 138.76.29.7
- *solution1:* statically configure NAT to forward incoming connection requests at given port to server
  - e.g., (123.76.29.7, port 2500) always forwarded to 10.0.0.1 port 25000



NAT traversal problem

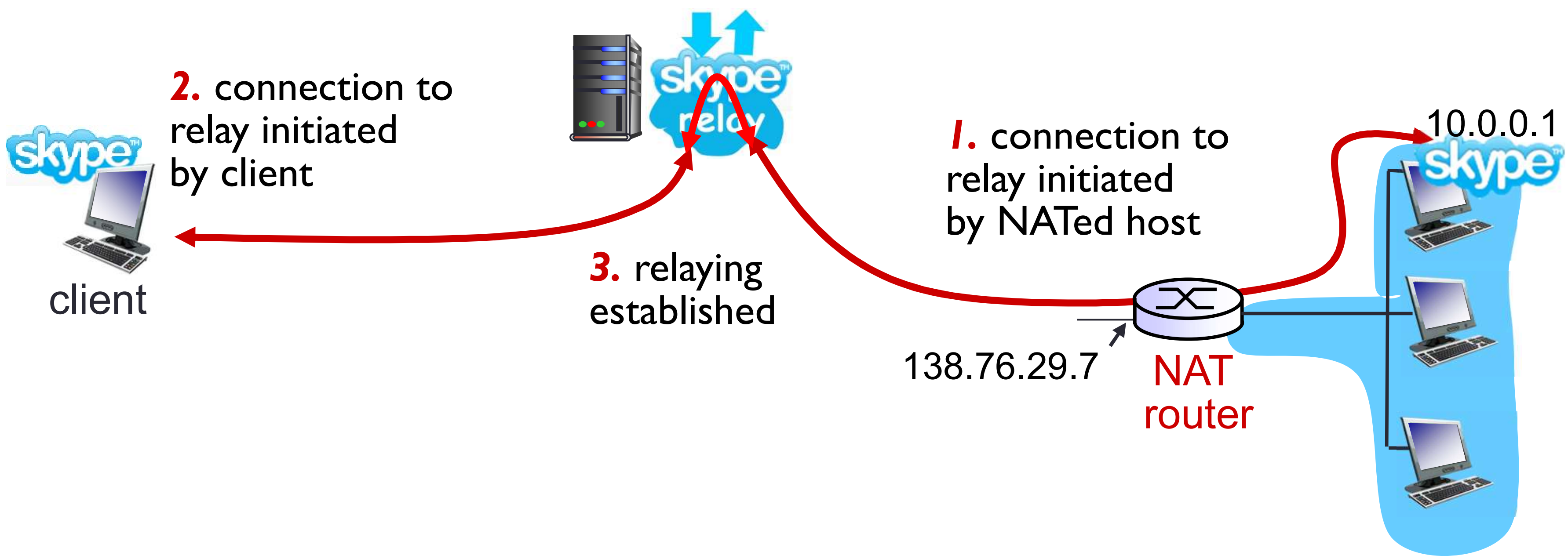
- *solution 2:* Universal Plug and Play (UPnP) Internet Gateway Device (IGD) Protocol. Allows NATed host to:
  - learn public IP address (138.76.29.7)
  - add/remove port mappings (with lease times)

i.e., automate static NAT port map configuration



NAT traversal problem

- *solution 3:* relaying (used in Skype)
  - NATed client establishes connection to relay
  - external client connects to relay
  - relay bridges packets between to connections



NAT traversal problem

- *solution 4:* NAT hole punching. Example: STUN protocol