Computer Networks and Applications

COMP 3331/COMP 9331

Week 5

P2P + Transport Layer Part 1

Reading Guide: Chapter 2, 2.5 + Chapter 3, Sections 3.1 – 3.4

2. Application Layer: outline

2.1 principles of network applications
- app architectures
- app requirements

2.2 Web and HTTP

2.3 electronic mail
- SMTP, POP3, IMAP

2.4 DNS

2.5 P2P applications

2.6 video streaming and content distribution networks (CDNs)

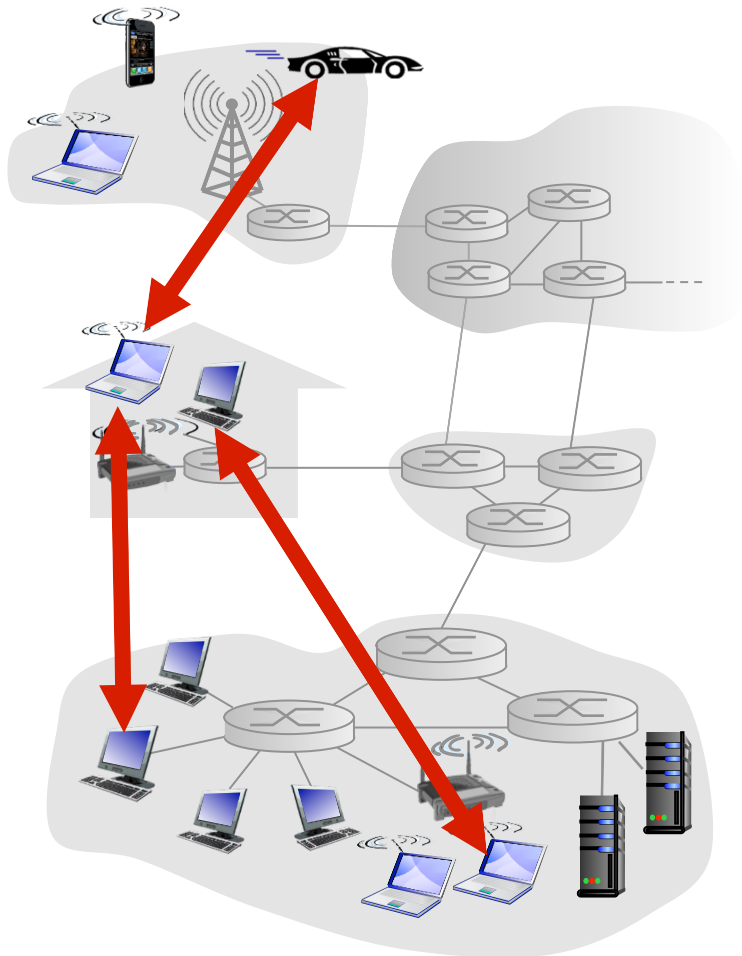
2.7 socket programming with UDP and TCP

P2P architecture

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

examples:

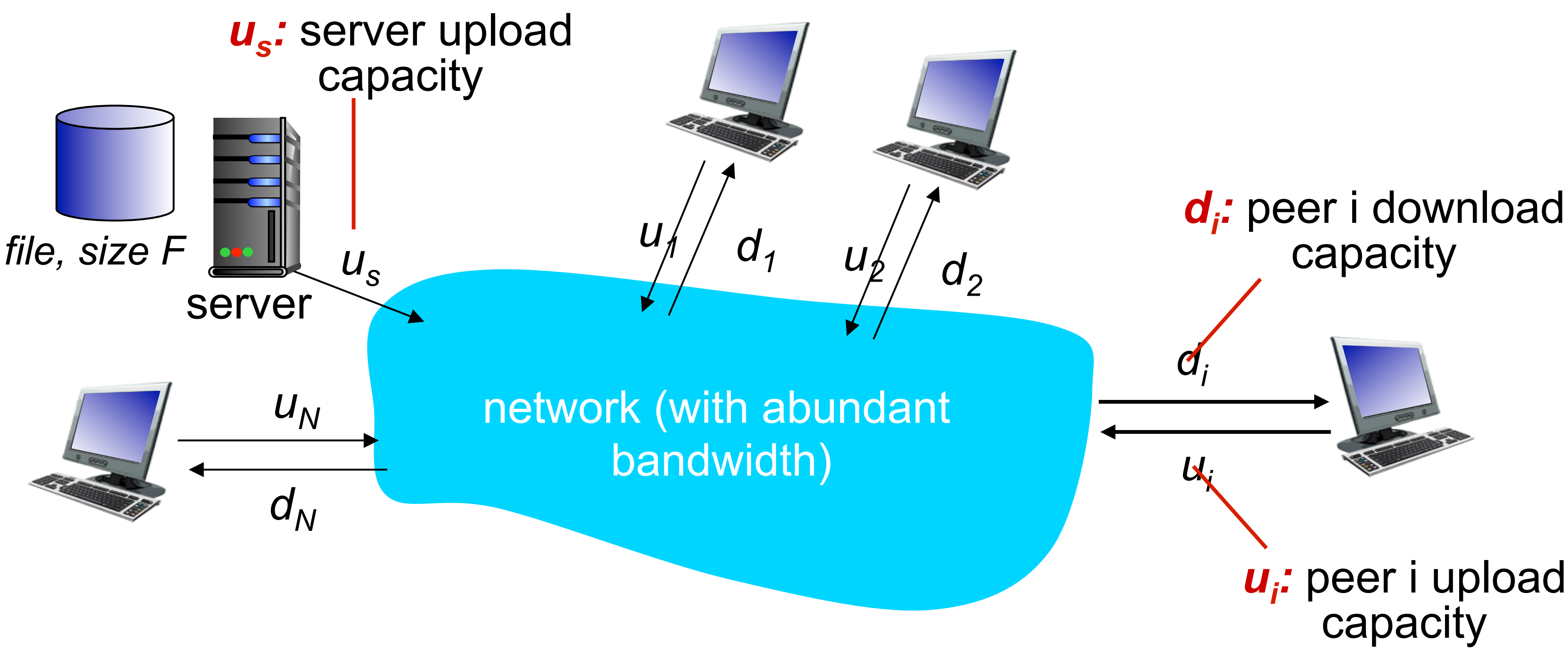
- file distribution (BitTorrent)
- Streaming (KanKan)
- VoIP (Skype)



File distribution: client-server vs P2P

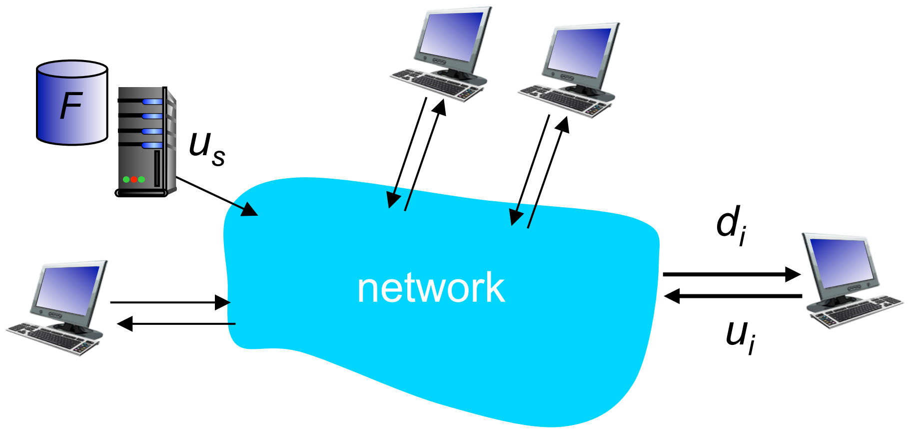
Question: how much time to distribute file (size *F*) from one server to *N* peers?

- peer upload/download capacity is limited resource



File distribution time: client-server

- *server transmission:* must sequentially send (upload) N file copies:
  - time to send one copy: F/u_s
  - time to send N copies: NF/u_s



- *client:* each client must download file copy
  - d_{min} = min client download rate
  - min client download time: F/d_{min}

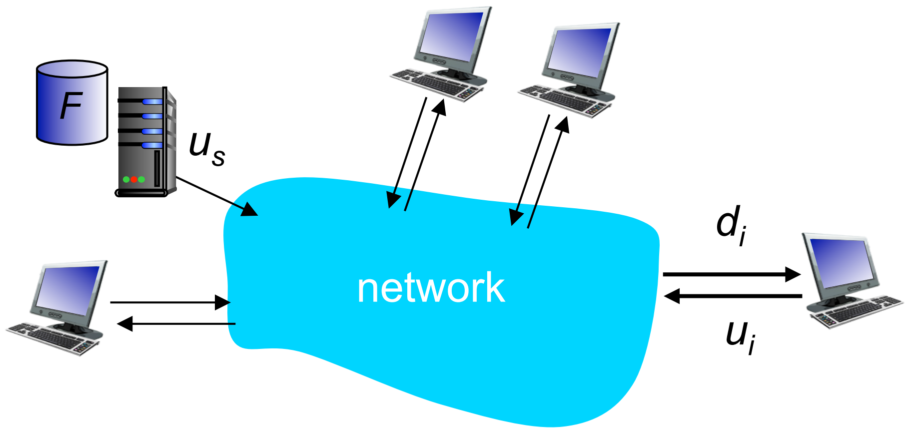
time to distribute F to N clients using client-server approach

$$D_{c\text{-}s} \geq max\{NF/u_{s,},F/d_{min}\}$$

increases linearly in N

File distribution time: P2P

- *server transmission:* must upload at least one copy
  - time to send one copy: F/u_s
- *client:* each client must download file copy
  - min client download time: F/d_{min}



- *clients:* as aggregate must download *NF* bits
  - max upload rate (limiting max download rate) is $u_s + \Sigma u_i$

time to distribute F to N clients using P2P approach

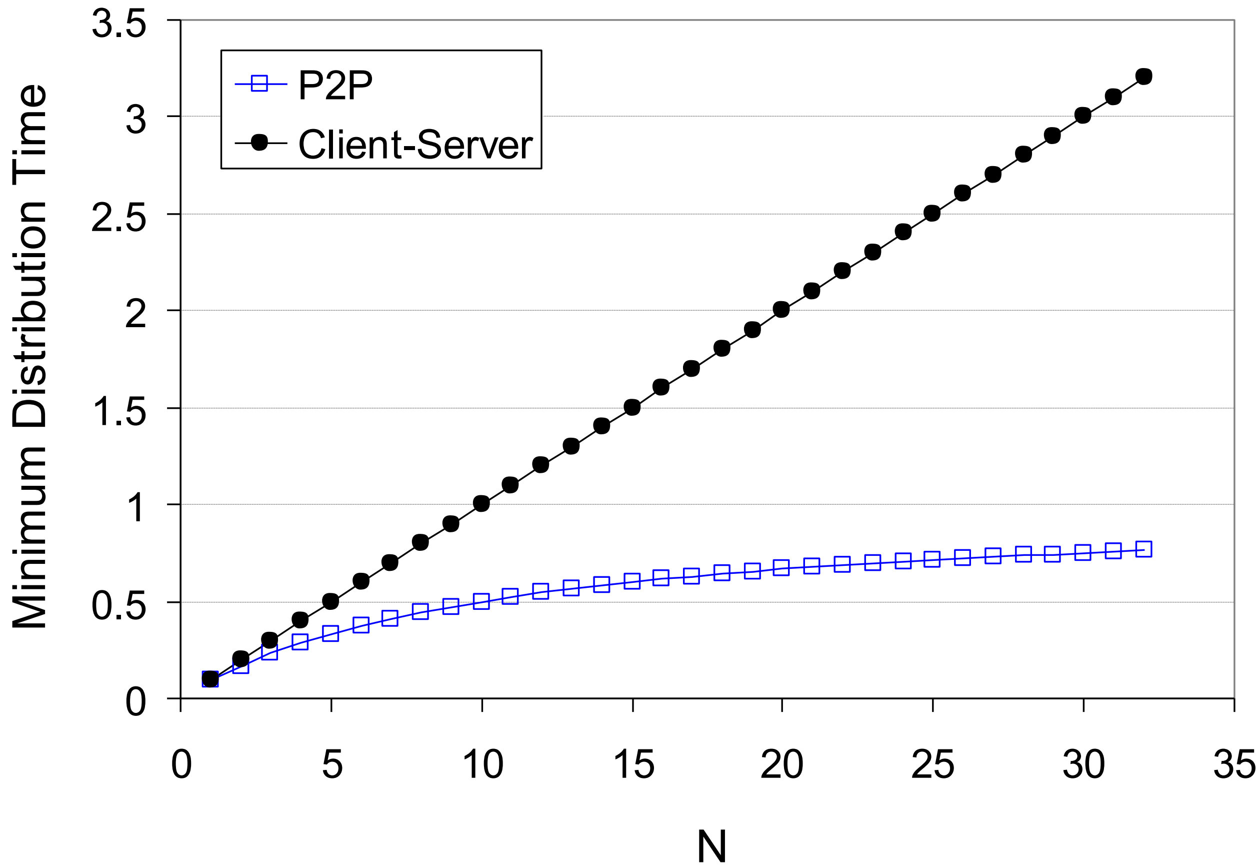
$$D_{P2P} \geq max\{F/u_s, F/d_{min}, NF/(u_s + \Sigma u_i)\}$$

increases linearly in *N* …

… but so does this, as each peer brings service capacity

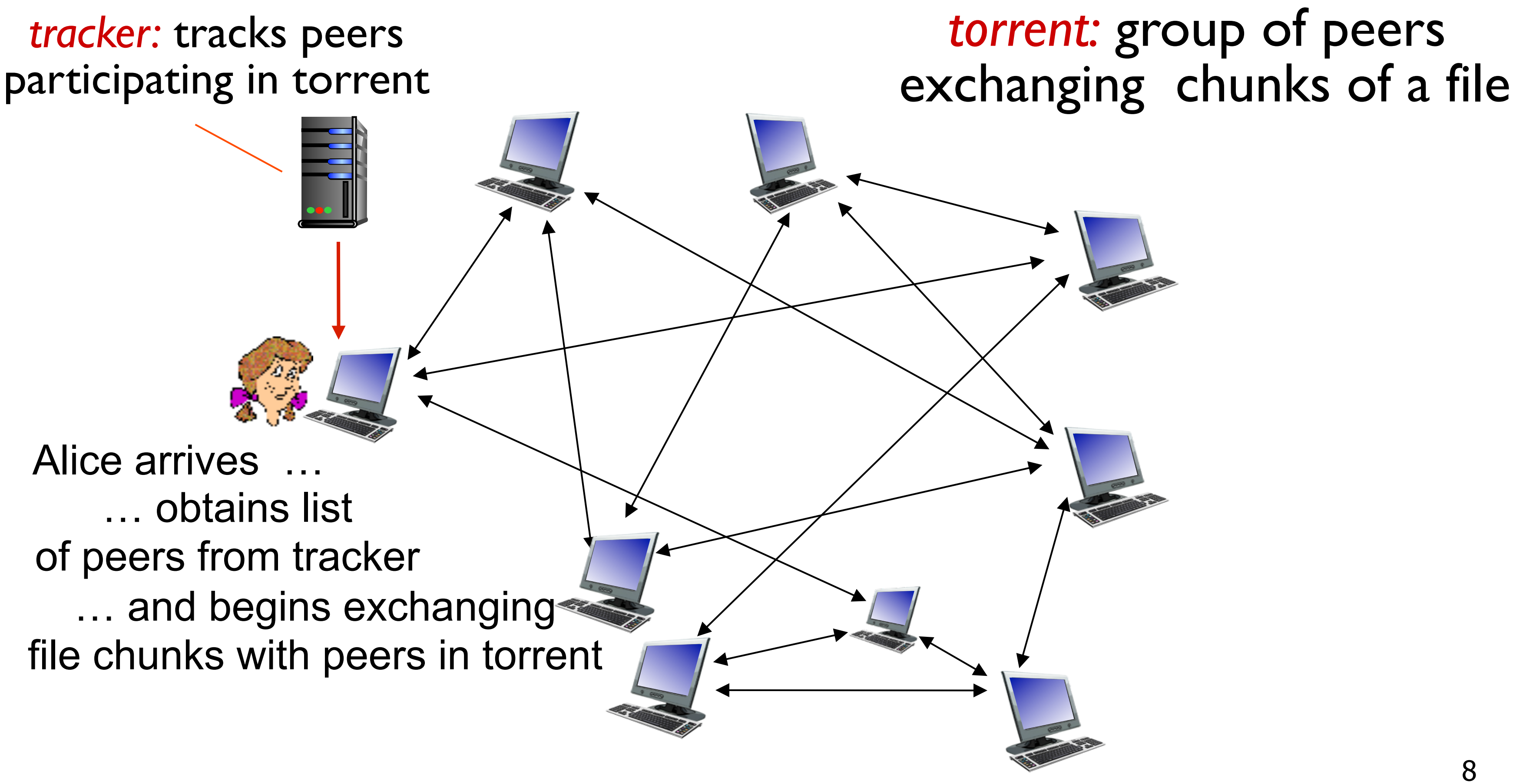
Client-server vs. P2P: example

client upload rate = u, F/u = 1 hour, $u_s = 10u$, $d_{min} \geq u_s$

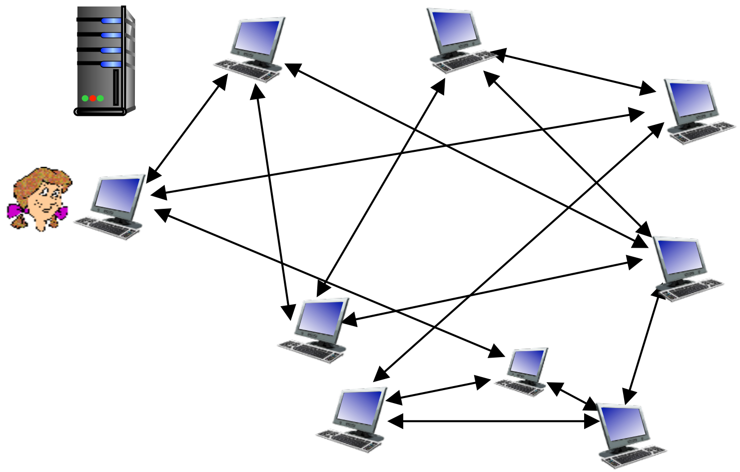


P2P file distribution: BitTorrent

- file divided into 256Kb chunks
- peers in torrent send/receive file chunks



P2P file distribution: BitTorrent



- peer joining torrent:
  - has no chunks, but will accumulate them over time from other peers
  - registers with tracker to get list of peers, connects to subset of peers ("neighbours")

- while downloading, peer uploads chunks to other peers
- peer may change peers with whom it exchanges chunks
  - *churn:* peers may come and go
- once peer has entire file, it may (selfishly) leave or (altruistically) remain in torrent

BitTorrent: requesting, sending file chunks

requesting chunks:

- at any given time, different peers have different subsets of file chunks
- periodically, Alice asks each peer for list of chunks that they have
- Alice requests missing chunks from peers, rarest first
- **Q:** Why rarest first?

sending chunks: tit-for-tat

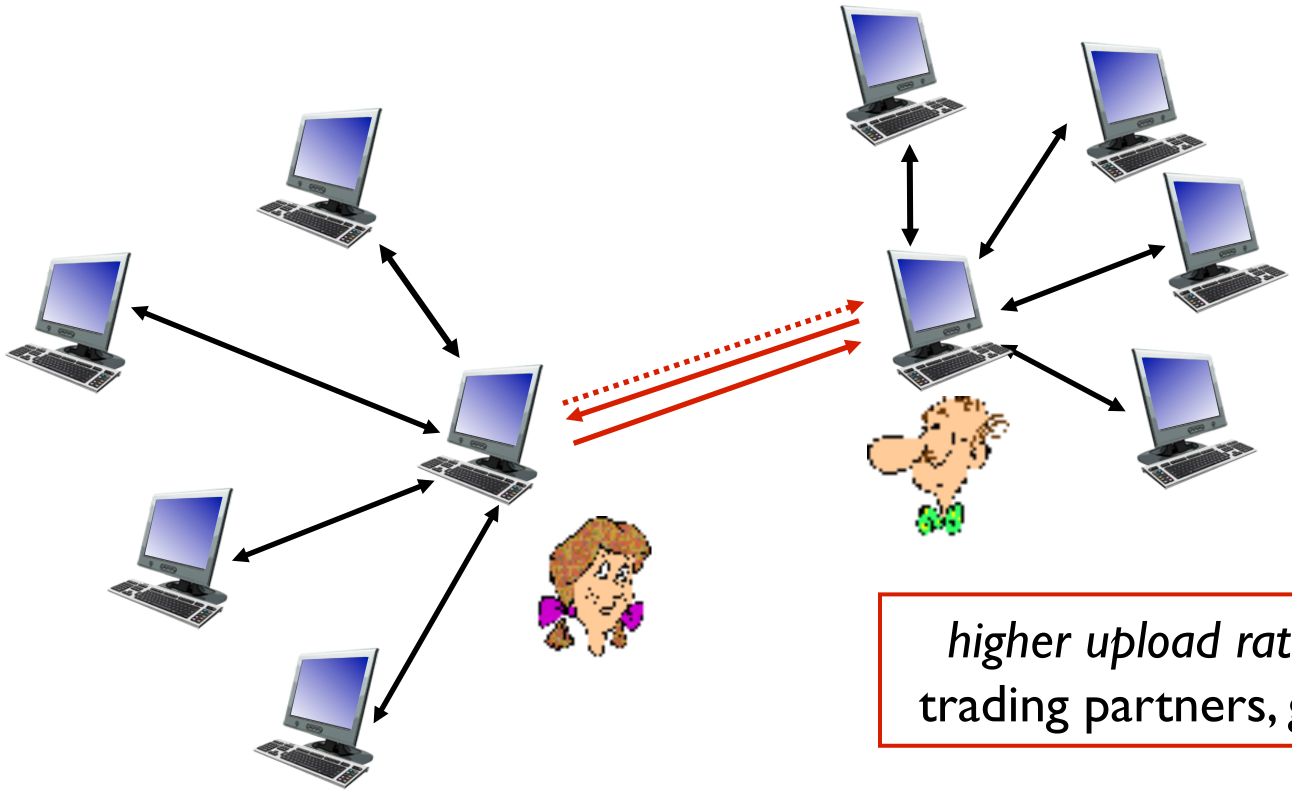
- Alice sends chunks to those four peers currently sending her chunks *at highest rate*
  - other peers are choked by Alice (do not receive chunks from her)
  - re-evaluate top 4 every10 secs
- every 30 secs: randomly select another peer, starts sending chunks
  - "optimistically unchoke" this peer
  - newly chosen peer may join top 4

BitTorrent: tit-for-tat

(1) Alice "optimistically unchokes" Bob

(2) Alice becomes one of Bob's top-four providers; Bob reciprocates

(3) Bob becomes one of Alice's top-four providers



higher upload rate: find better trading partners, get file faster !

Quiz: Free-riding

- Suppose Todd joins a BitTorrent torrent, but he does not want to upload any data to any other peers. Todd claims that he can receive a complete copy of the file that is shared by the swarm. Is Todd's claim possible? Why or Why not?

Getting rid of the server/tracker

- Distribute the tracker information using a Distributed Hash Table (DHT)

- A DHT is a lookup structure
  - Maps keys to an arbitrary value
  - Works a lot like, well …. hash table

Hash table - review

- (key,value) pairs
- Centralised hash table – all (key,value) pairs on 1 node
- Distributed hash tables – each node has a “section” of (key,value) pairs

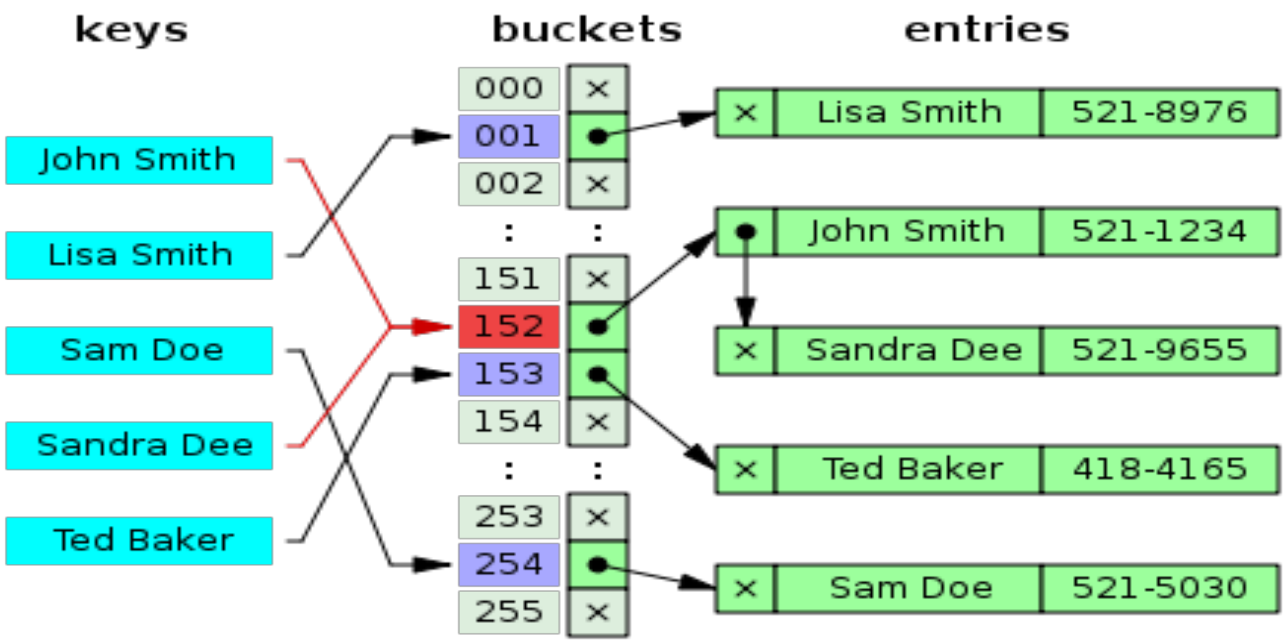


Figure src: http://en.wikipedia.org/wiki/Hash_table

Distributed Hash Table (DHT)

- DHT: a *distributed P2P database*
- database has (key, value) pairs; examples:
  - key: TFN number; value: human name
  - key: file name; value: BT tracker peer(s)
- Distribute the (key, value) pairs over the (millions of peers)
- a peer queries DHT with key
  - DHT returns values that match the key
- peers can also insert (key, value) pairs

Challenges

- How do we assign (key, value) pairs to nodes?
- How do we find them again quickly?
- What happens if nodes join/leave?

Q: how to assign keys to peers?

- basic idea:
  - convert each key to an integer
  - Assign integer to each peer
  - put (key,value) pair in the peer that is closest to the key

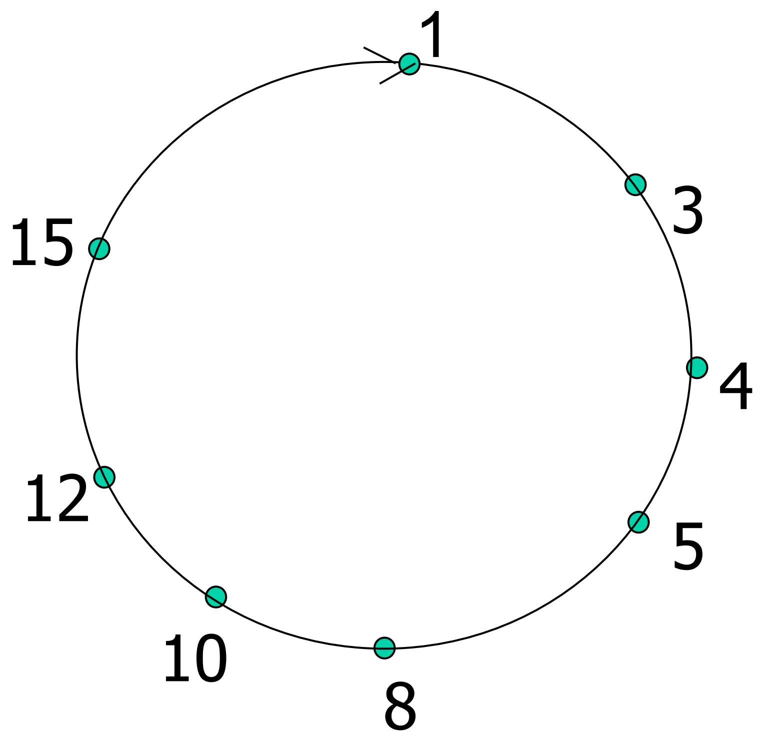
DHT identifiers: Consistent Hashing

- assign integer identifier to each peer in range $[0,2^n-1]$ for some *n*-bit hash function
  - E.g., node ID is hash of its IP address
- require each key to be an integer in same range
- to get integer key, hash original key
  - e.g., key = hash("House of Cards Season 4")
  - this is why its is referred to as a *distributed "hash" table*

Assign keys to peers

- rule: assign key to the peer that has the *closest* ID.
- common convention: closest is the *immediate successor* of the key.
- e.g., *n*=4; peers: 1,3,4,5,8,10,12,14;
  - key = 13, then successor peer = 14
  - key = 15, then successor peer = 1

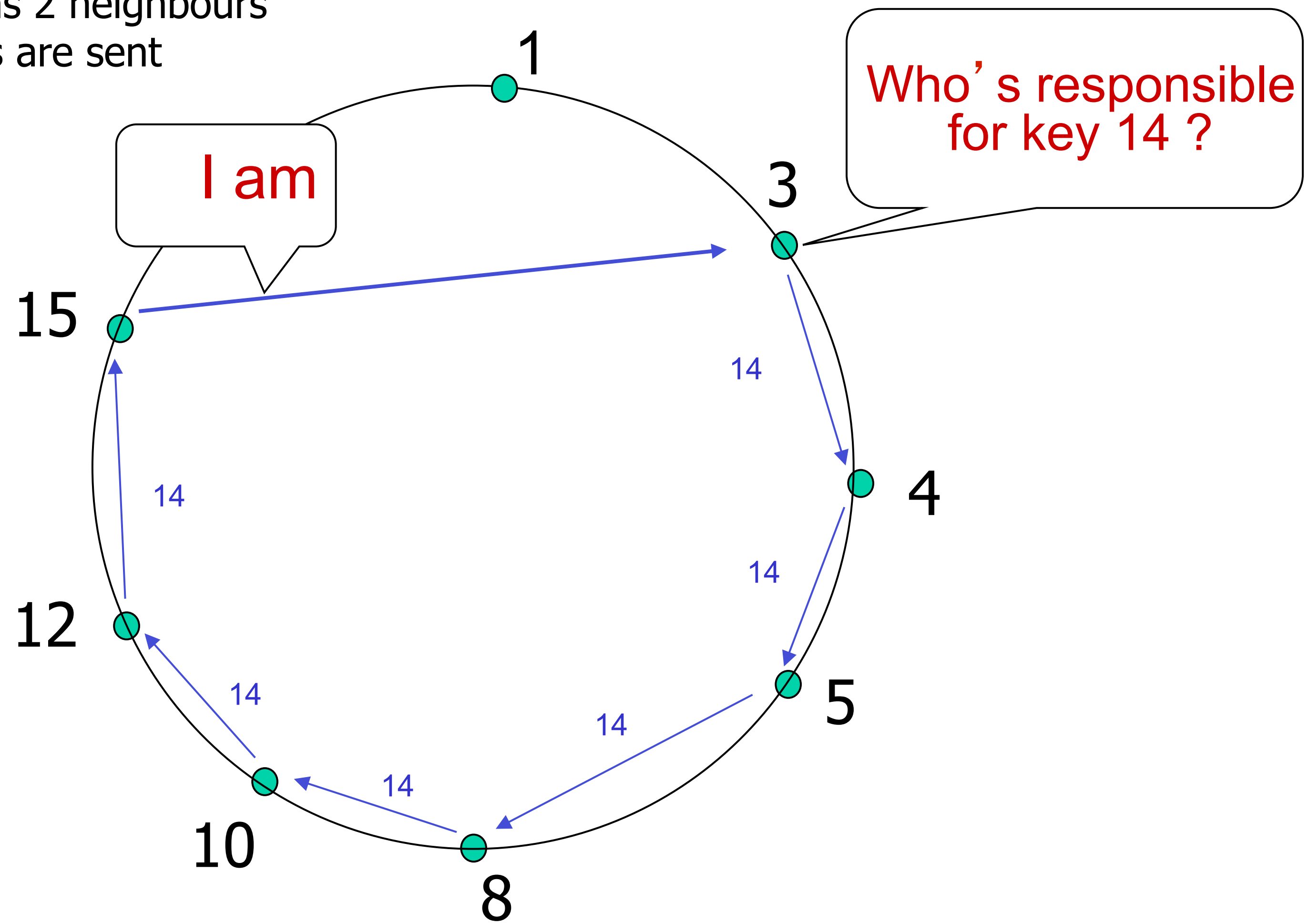
Circular DHT (1)



- each peer *only* aware of immediate successor and predecessor.
- "overlay network"

Circular DHT (2)

- Each per maintains 2 neighbours
- 6 query messages are sent

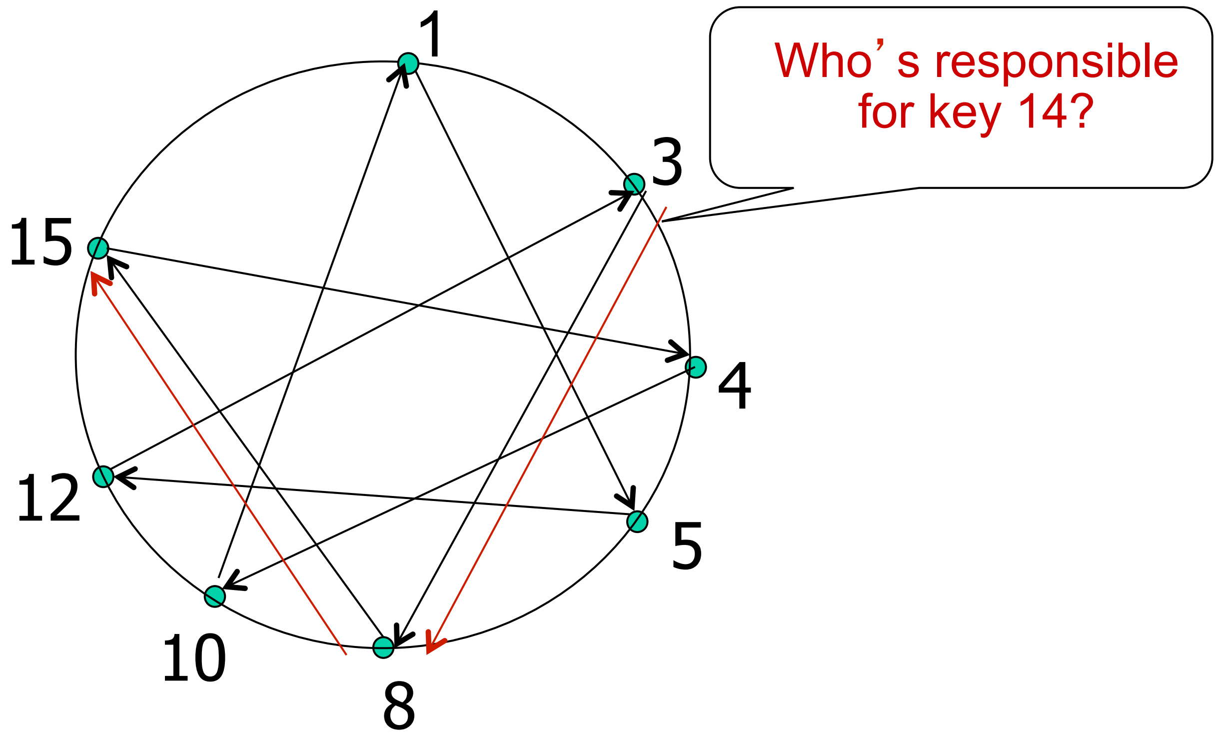


Define <u>closest</u> as closest successor

Tradeoff: #of neighbours vs. #of messages

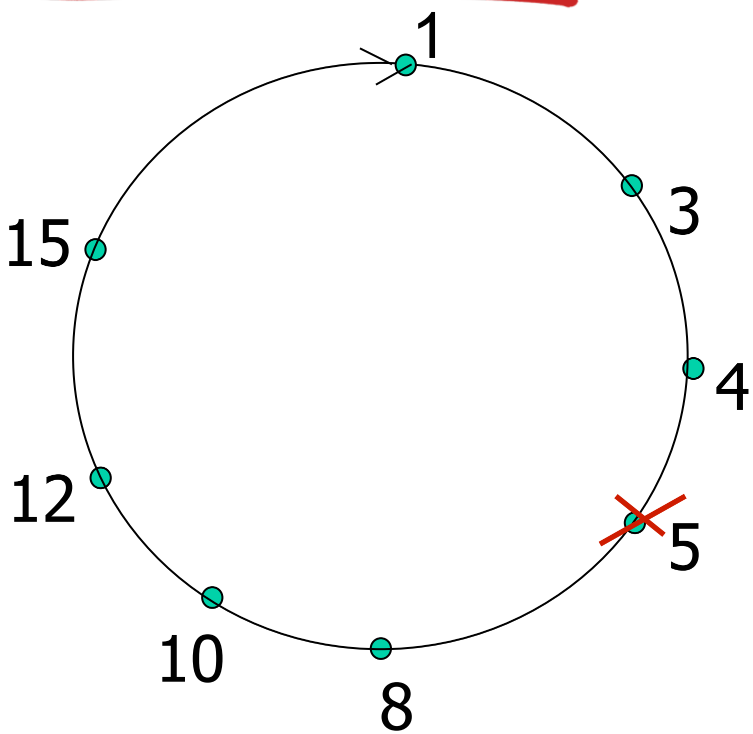
- The smaller the number of neighbours to maintain, the larger the number of messages have to be sent;
- Previous example: 2 neighbours, 6 messages
- Worst case: N messages
- Average: N/2 messages

Circular DHT with shortcuts



- each peer keeps track of IP addresses of predecessor, successor, short cuts
- reduced from 6 to 2 messages.
- possible to design shortcuts so *O(log N)* neighbours, *O(log N)* messages in query

Peer churn



handling peer churn:

❖peers may come and go (churn)
❖each peer knows address of its two successors
❖each peer periodically pings its two successors to check aliveness
❖if immediate successor leaves, choose next successor as new immediate successor

example: peer 5 abruptly leaves

❖peer 4 detects peer 5 departure; makes 8 its immediate successor; asks 8 who its immediate successor is; makes 8's immediate successor its second successor.

More DHT info

- How do nodes join?
- How does cryptographic hashing work?
- How much state does each node store?

Research Articles (available from Moodle page):
Chord: A Scalable Peer-to-Peer Lookup Service for Internet Applications
Dynamo: Amazon's Highly Available Key-value Store

DHT: Applications

- File sharing and backup [CFS, Ivy, OceanStore, PAST, Pastiche …]
- Web cache and replica [Squirrel, Croquet Media Player]
- Censor-resistant stores [Eternity]
- DB query and indexing [PIER, Place Lab, VPN Index]
- Event notification [Scribe]
- Naming systems [ChordDNS, Twine, INS, HIP]
- Distributed BitTorrent tracker [Kademlia, Vuze]
- Communication primitives [I3, …]
- Host mobility [DTN Tetherless Architecture]

Transport Layer

our goals:

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

Transport Layer 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

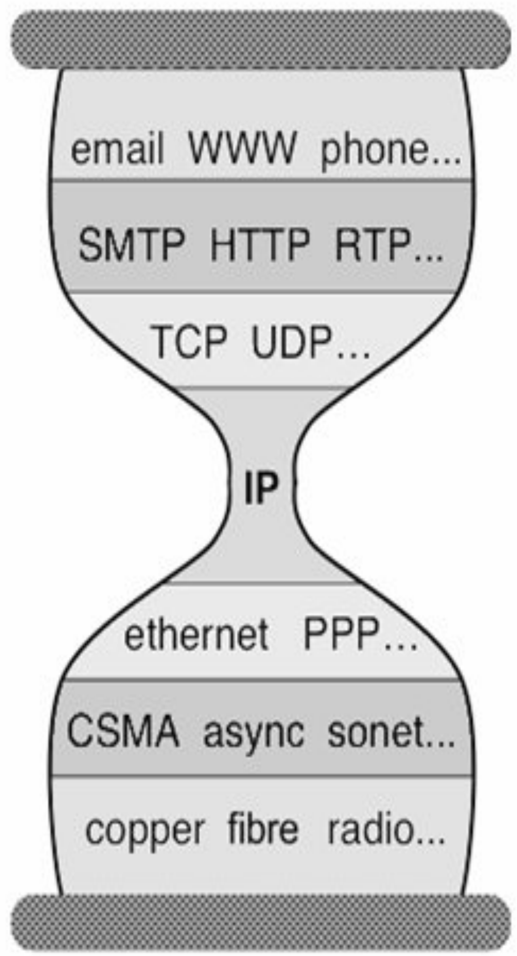
3.6 principles of congestion control

3.7 TCP congestion control

Transport layer

- Moving "down" a layer

- Current perspective:
  - Application is the boss….
  - Usually executing within the OS Kernel
  - The network layer is ours to command !!



Network layer (context)

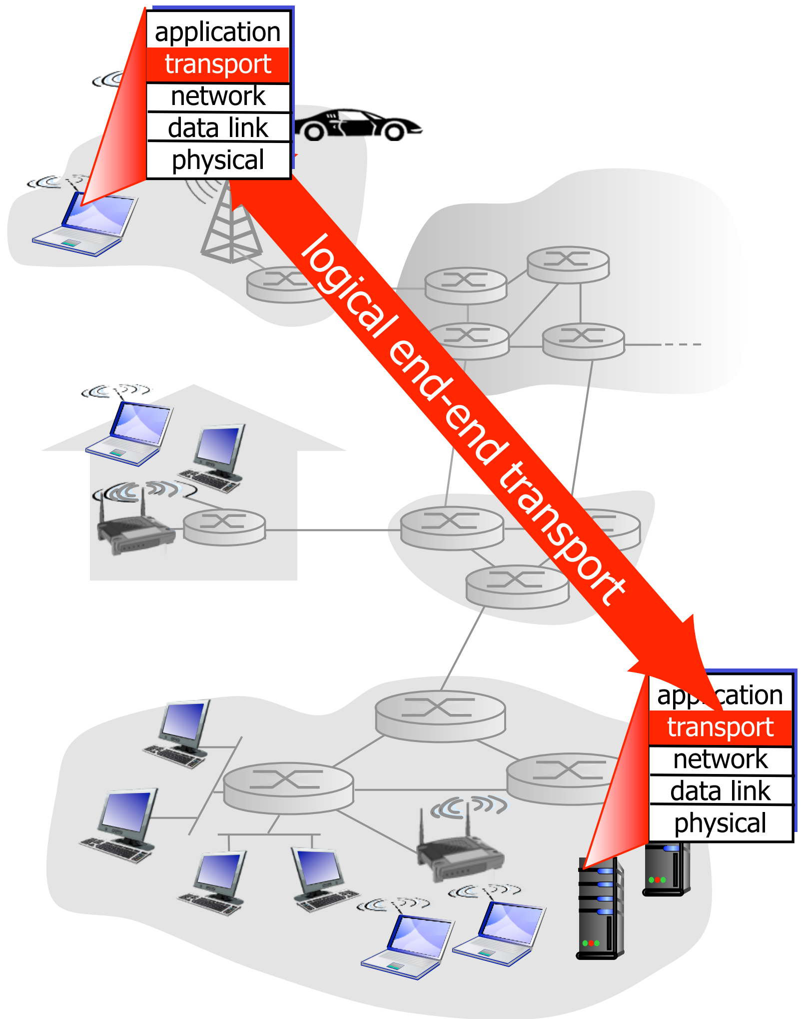
- What it does: finds paths through network
  - Routing from one end host to another

- What it doesn't:
  - Reliable transfer: "best effort delivery"
  - Guarantee paths
  - Arbitrate transfer rates

- For now, think of the network layer as giving us an "API" with one function: *sendtohost(data, host)*
  - Promise: the data will go to that (usually!!)

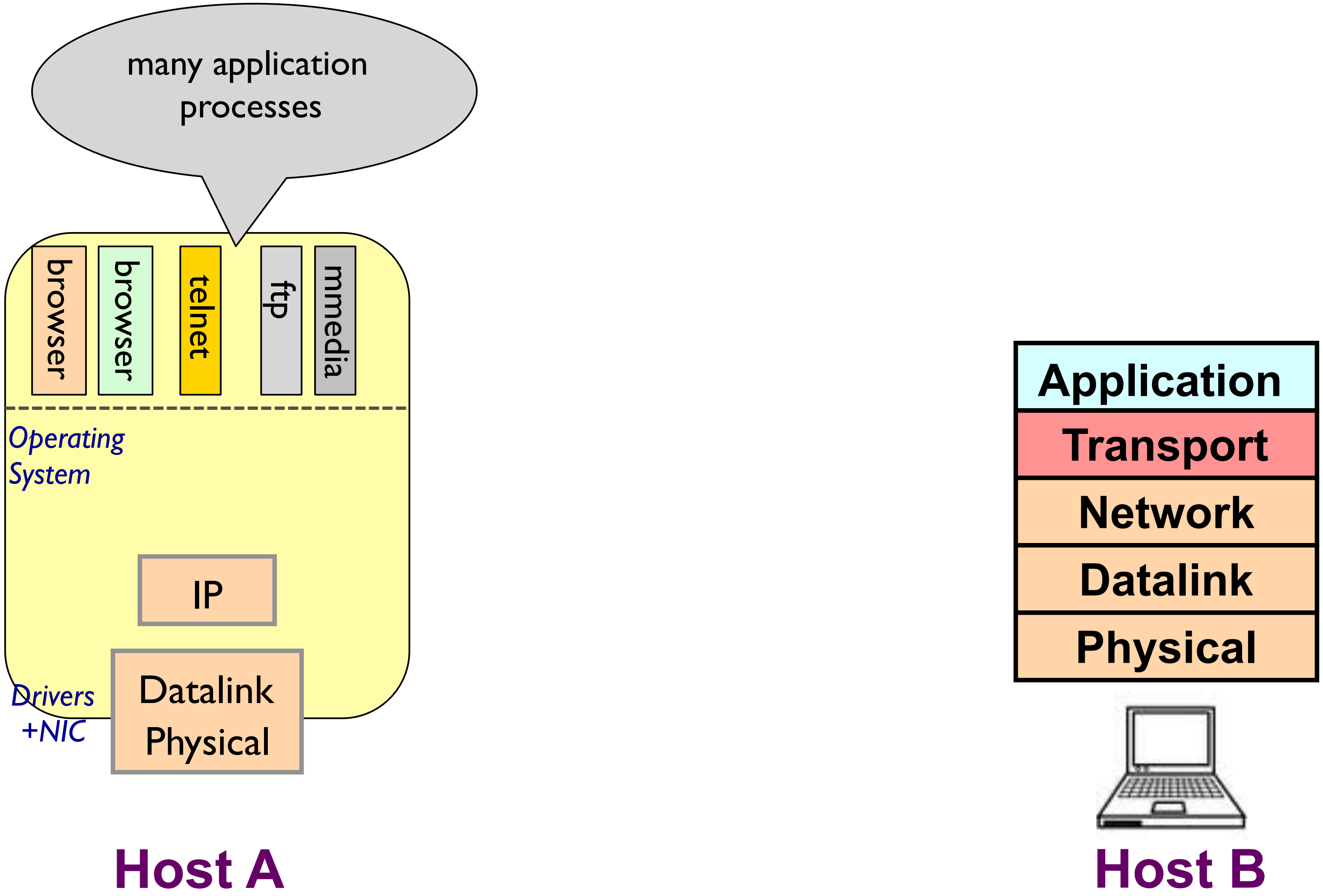
Transport services and protocols

- 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
  - Exports services to application that network layer does not provide

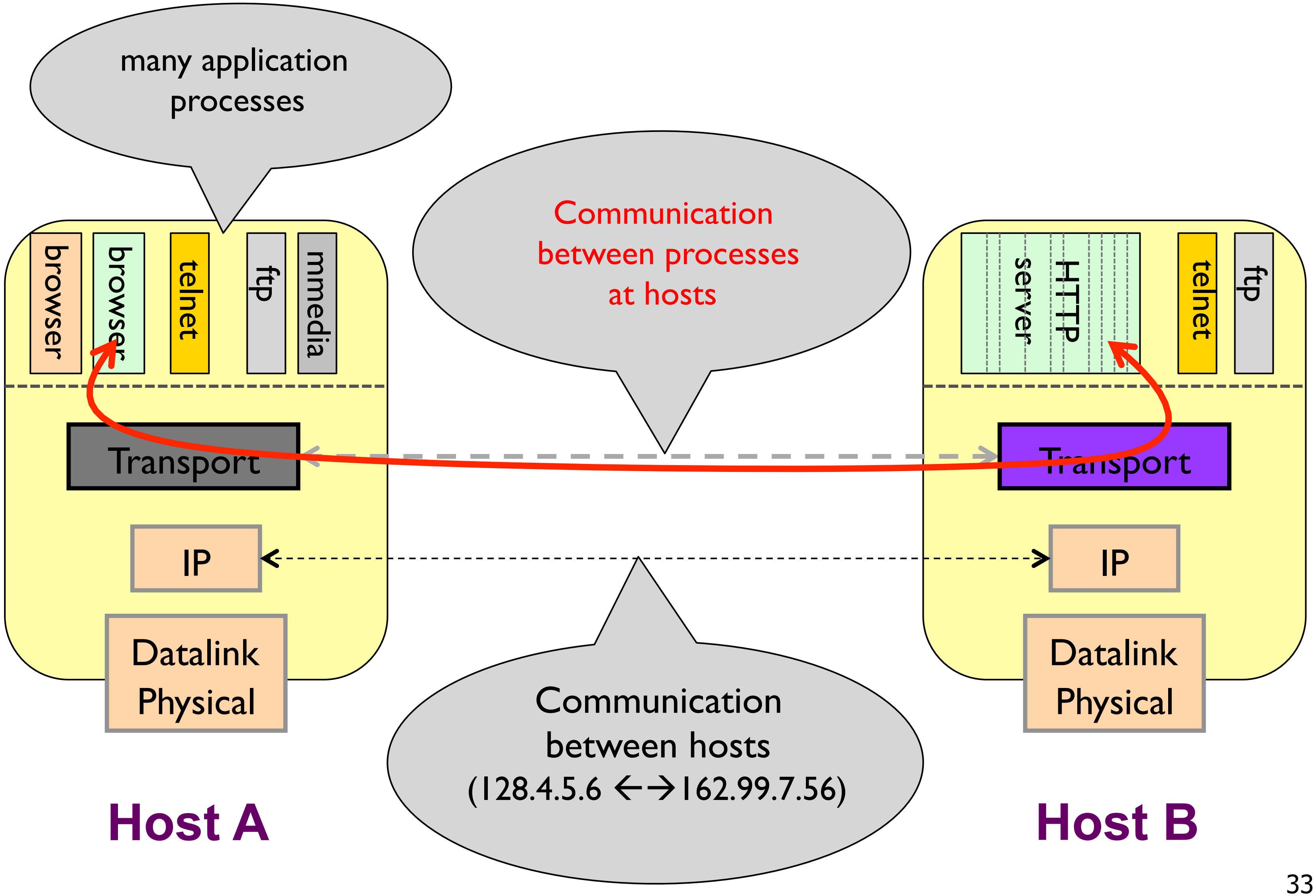


Network layer provides logical communication between hosts

Why a transport layer?



Why a transport layer?



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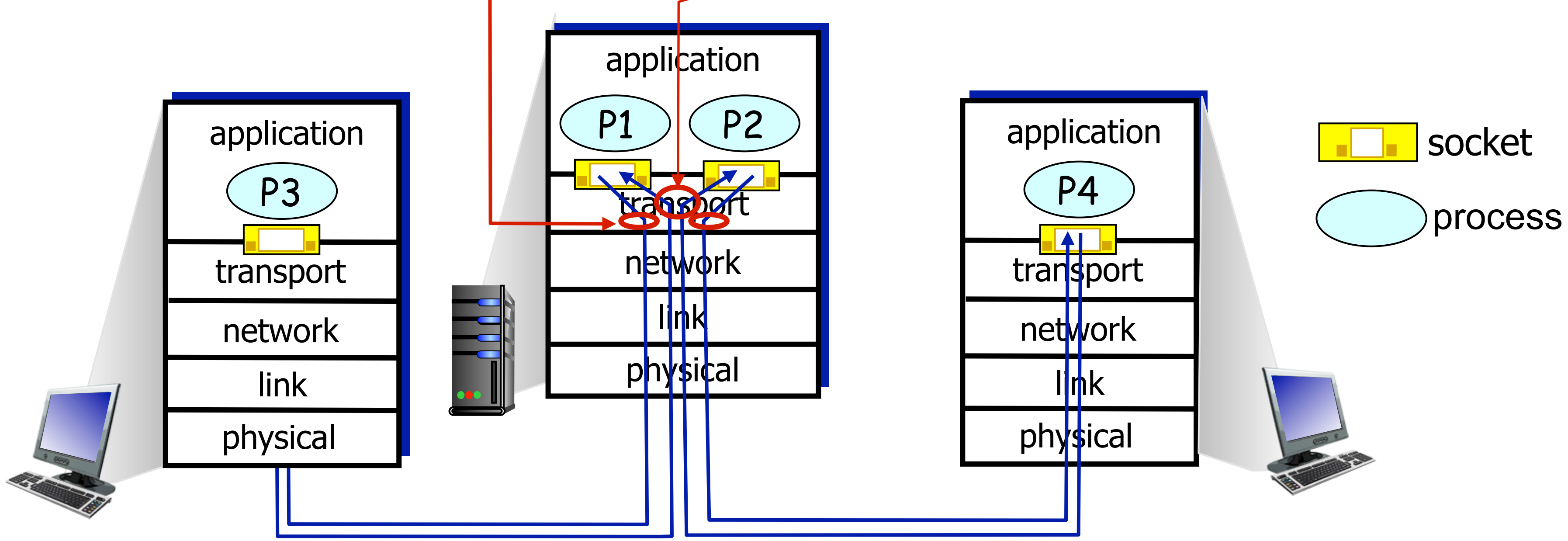
Multiplexing/demultiplexing

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



Note: The network is a shared resource. It does not care about your applications, sockets, etc.

Connectionless demultiplexing

- *recall:* created socket has host-local port #:

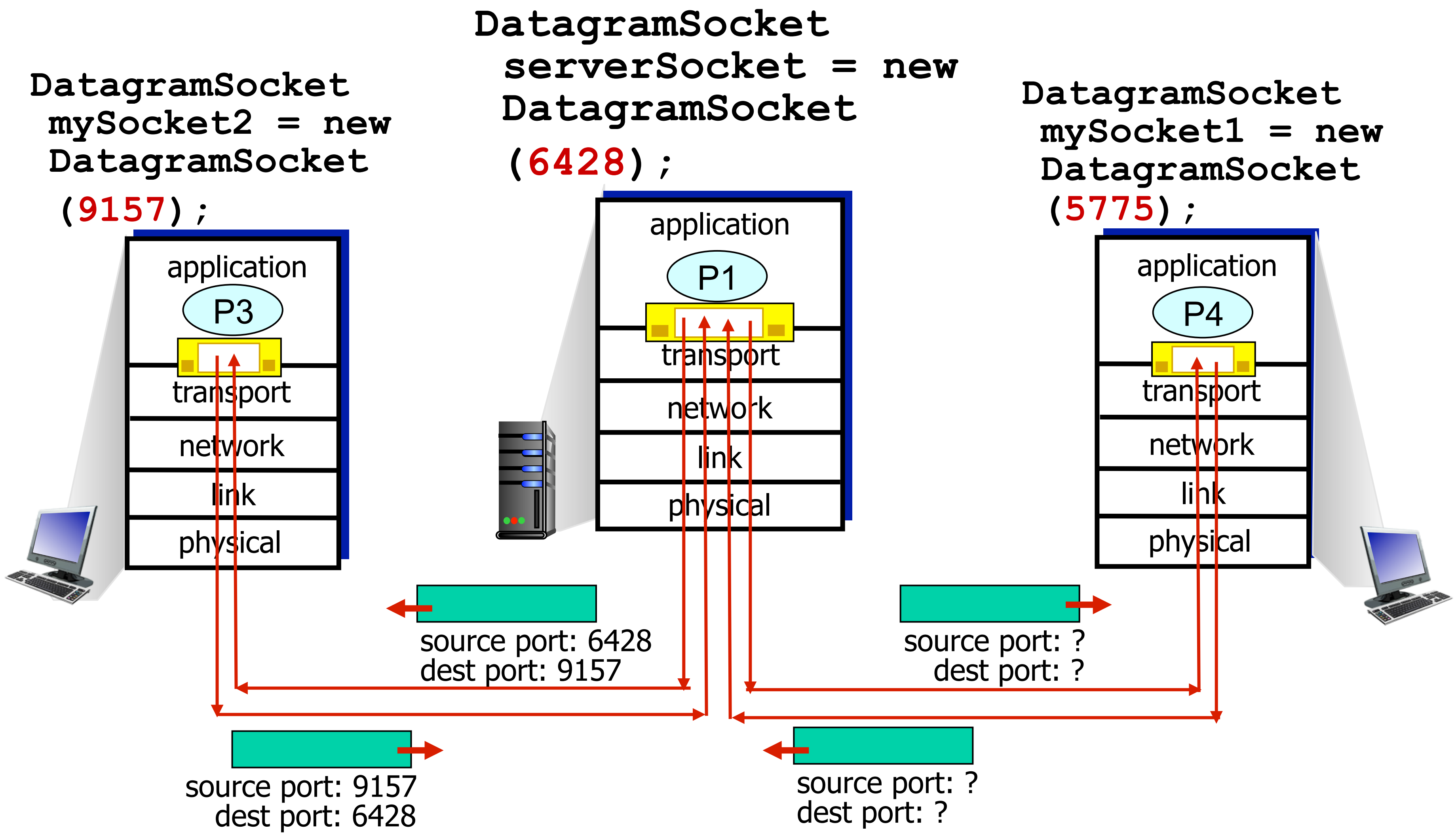
```
DatagramSocket mySocket1
= new DatagramSocket(12534);
```

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

- when host receives UDP segment:
  - checks destination port # in segment
  - directs UDP segment to socket with that port #

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

Connectionless demux: example

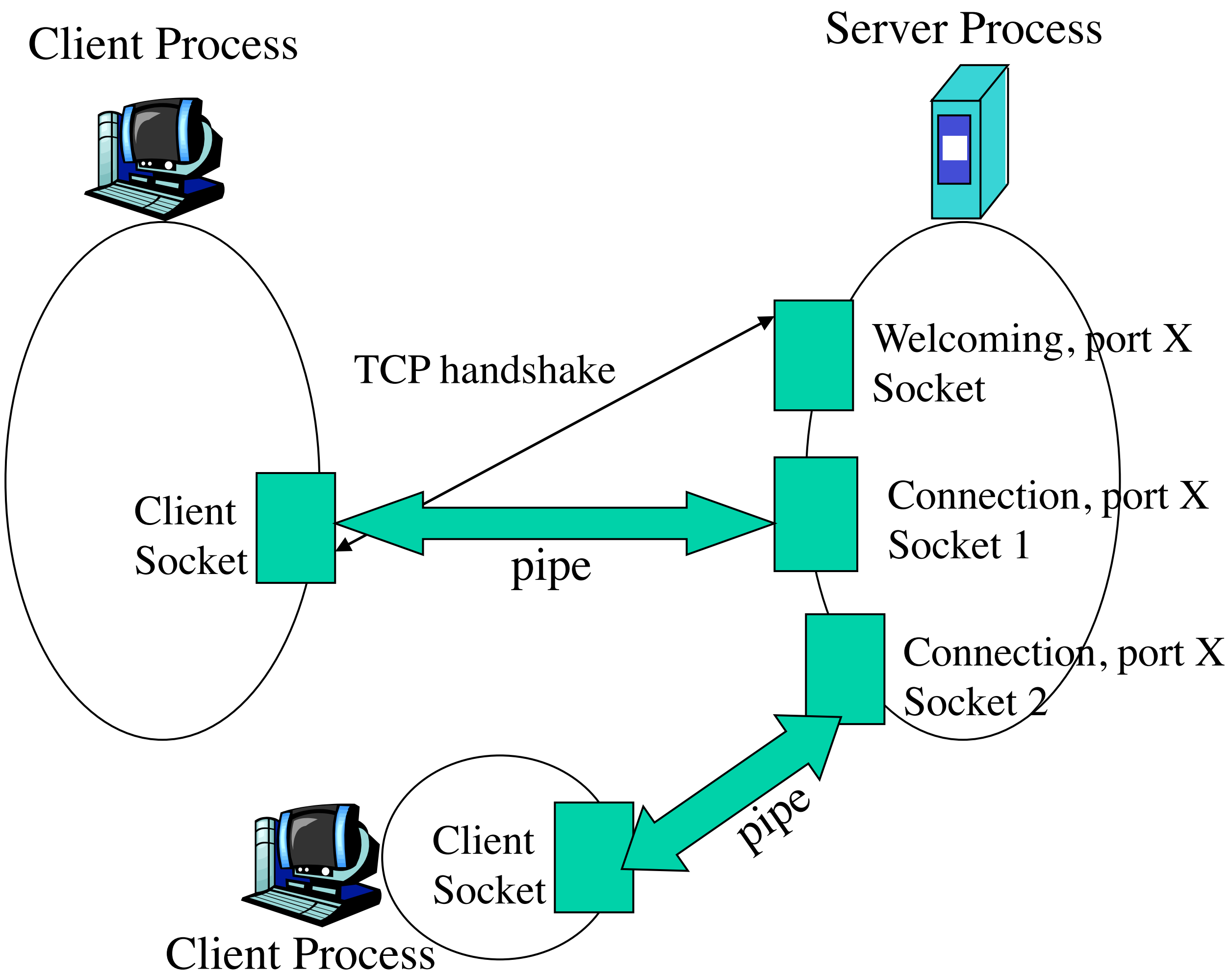


Connection-oriented demux

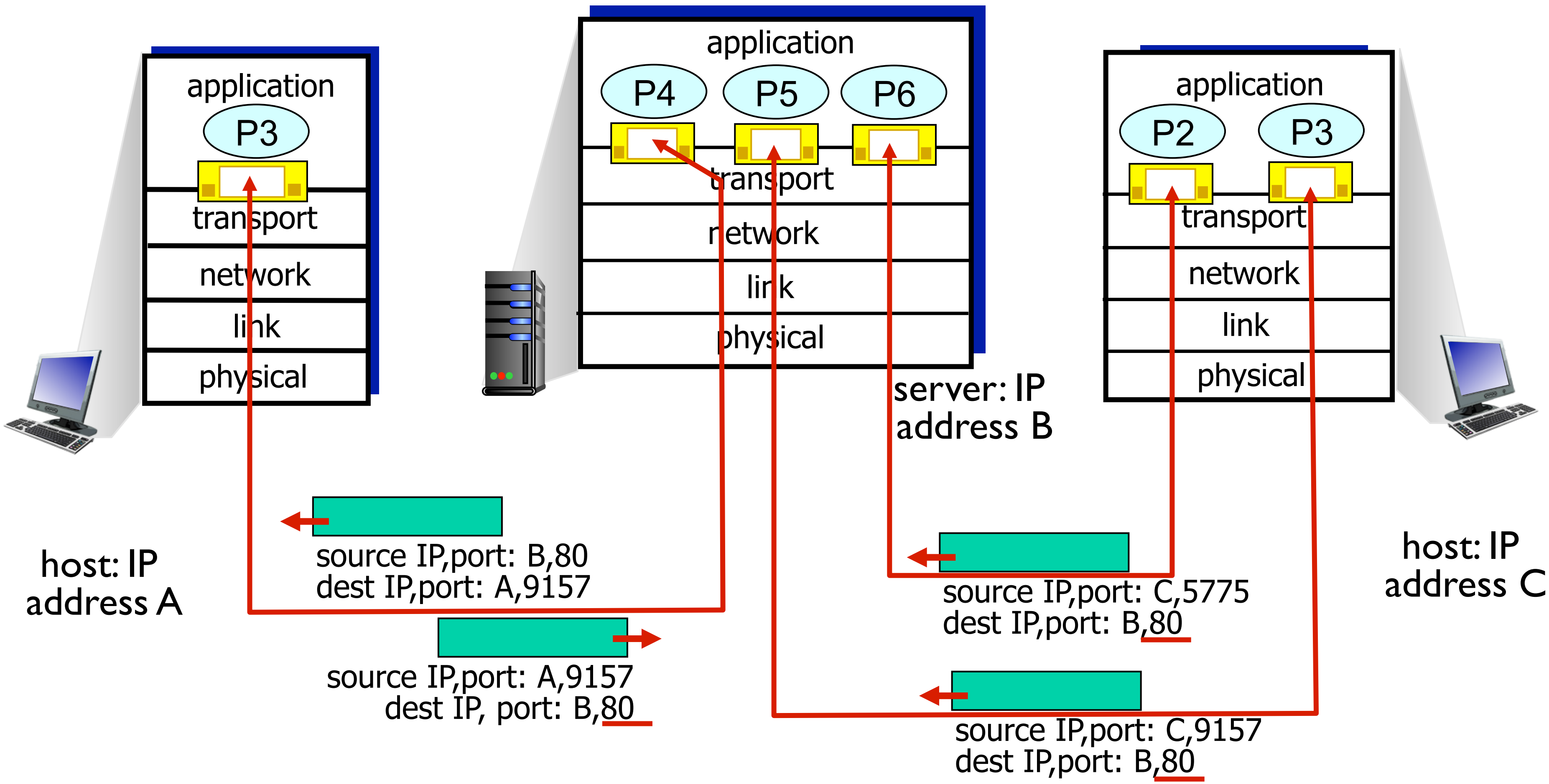
- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- demux: receiver uses all four values to direct segment to appropriate socket
- server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

Revisiting TCP Sockets

Server process creates (spawns) two more sockets for two different TCP connections



Connection-oriented demux: example



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

Take-away message

- UDP is identified by 2-tuple
  - Server maintains a single socket for all incoming UDP packets
- TCP is identified by 4-tuple
  - Server creates new sockets for each TCP connection

May I scan your ports?

http://netsecurity.about.com/cs/hackertools/a/aa121303.htm

- Servers wait at open ports for client requests
- Hackers often perform *port scans* to determine open, closed and unreachable ports on candidate victims
- Several ports are well-known
  - <1024 are reserved for well-known apps
  - Other apps also use known ports
    - MS SQL server uses port 1434 (udp)
    - Sun Network File System (NFS) 2049 (tcp/udp)
- Hackers can use exploit known flaws with these known apps
  - Example: Slammer worm exploited buffer overflow flaw in the SQL server
- How do you scan ports?
  - Nmap, Superscan, etc

http://www.auditmypc.com/

https://www.grc.com/shieldsup

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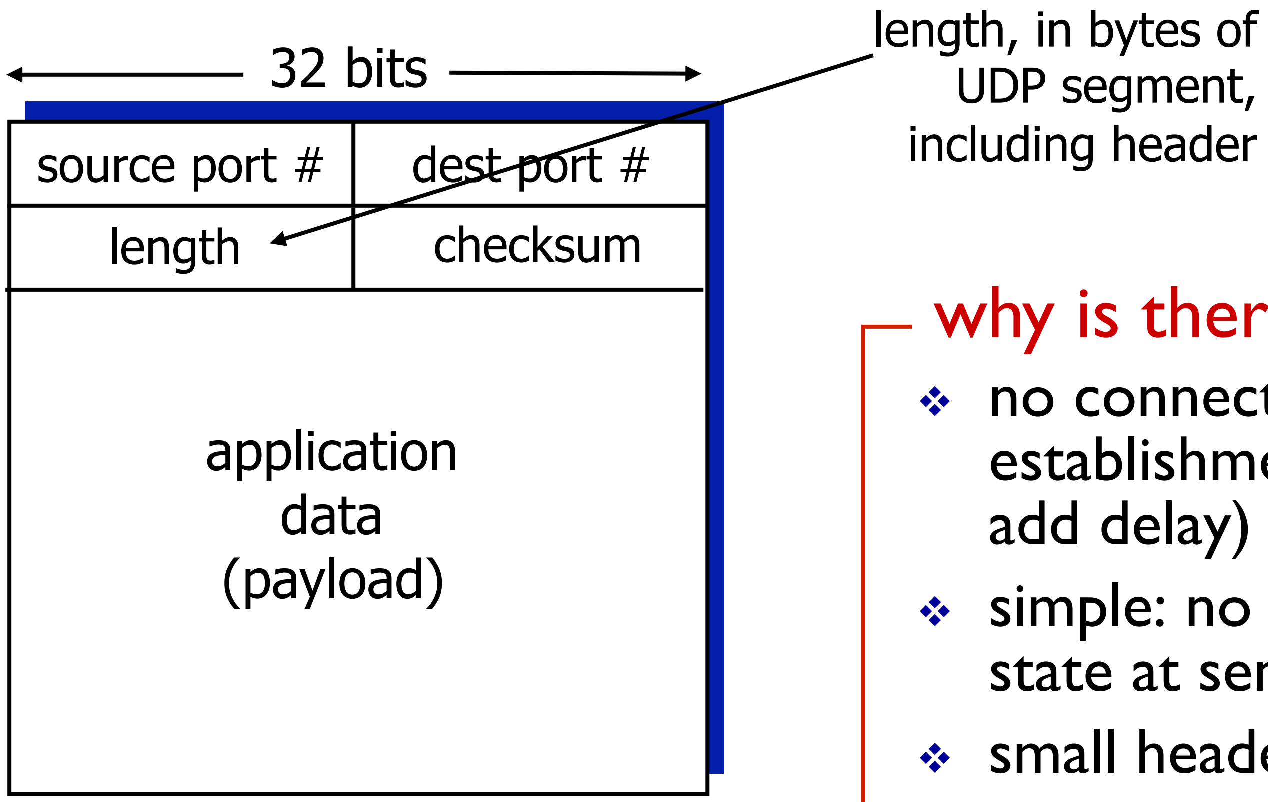
UDP: User Datagram Protocol [RFC 768]

- "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

UDP: segment header



UDP segment format

why is there a UDP?

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

UDP checksum

- *Goal:* detect "errors" (e.g., flipped bits) in transmitted segment
  - Router memory errors
  - Driver bugs
  - Electromagnetic interference

sender:

- treat segment contents, including header fields, as sequence of 16-bit integers
- checksum: addition (one's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

receiver:

- Add all the received together as 16-bit integers
- Add that to the checksum
- If the result is not 1111 1111 1111 1111, there are errors !

Internet checksum: 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

UDP Applications

- Latency sensitive/time critical
  - Quick request/response (DNS, DHCP)
  - Network management (SNMP)
  - Routing updates (RIP)
  - Voice/video chat
  - Gaming (especially FPS)
- Error correction unnecessary (periodic messages)

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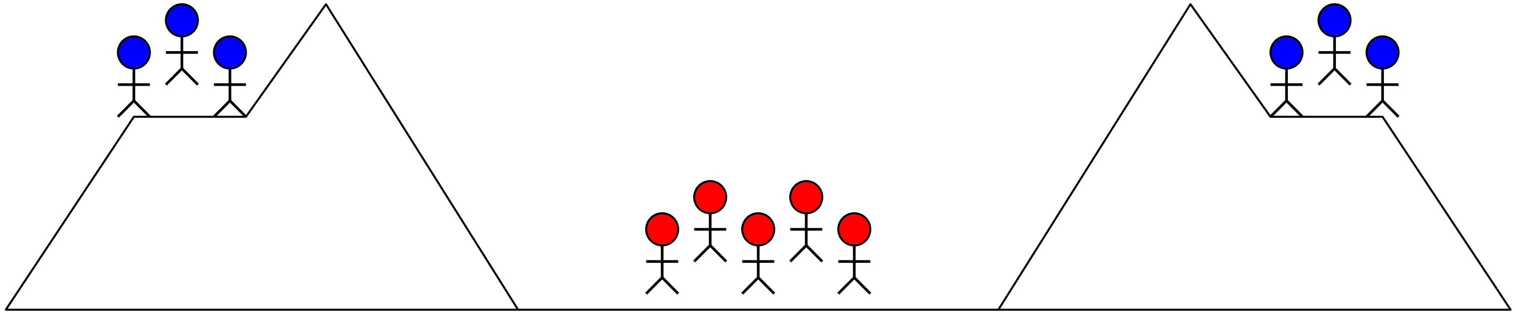
Reliable Transport

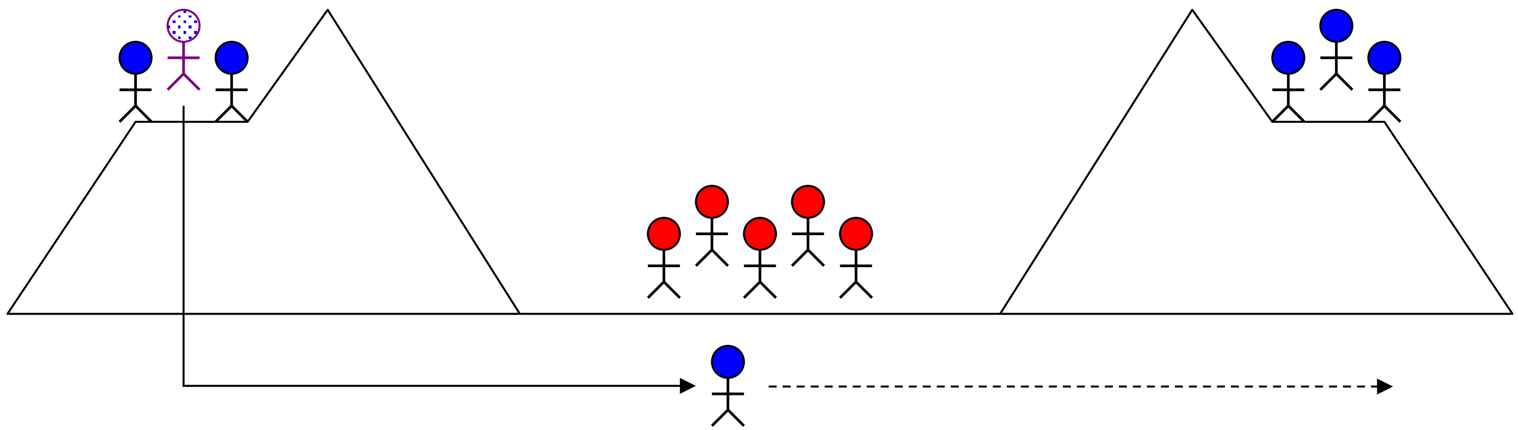
- In a perfect world, reliable transport is easy
- All the bad things best-effort can do
  - a packet is corrupted (bit errors)
  - a packet is lost
  - a packet is delayed (*why?*)
  - packets are reordered (*why?*)
  - a packet is duplicated (*why?*)

The Two Generals Problem



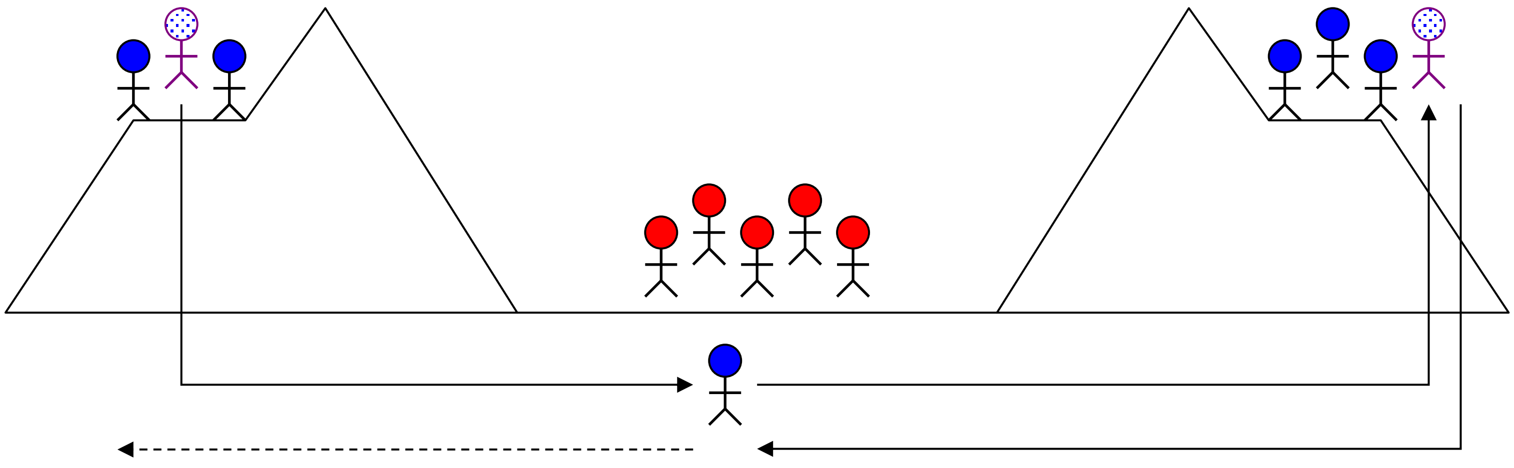
- Two army divisions (blue) surround enemy (red)
  - Each division led by a general
  - Both must agree when to simultaneously attack
  - If either side attacks alone, defeat
- Generals can only communicate via messengers
  - Messengers may get captured (unreliable channel)

The Two Generals Problem



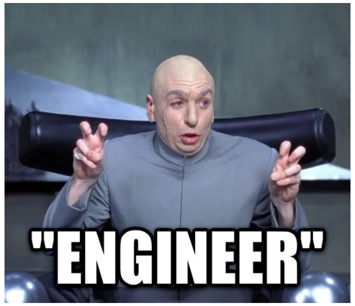
- How to coordinate?
  - Send messenger: “Attack at dawn”
  - What if messenger doesn’t make it?

The Two Generals Problem



- How to be sure messenger made it?
  - Send acknowledgement: “We received message”

Engineering



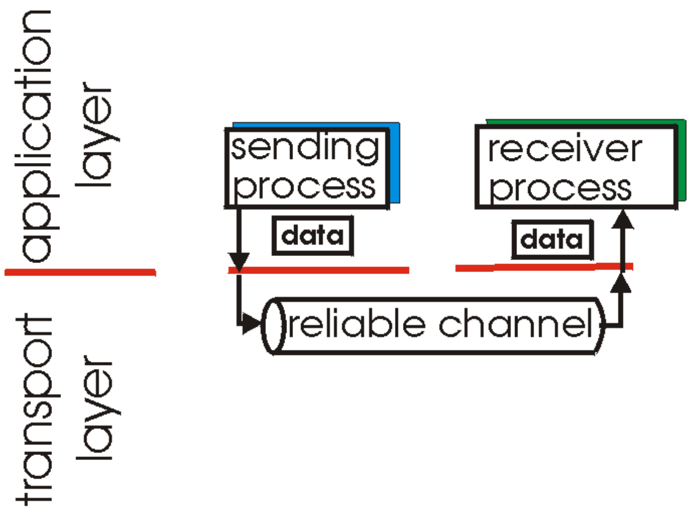
- Concerns
  - Message corruption
  - Message duplication
  - Message loss
  - Message reordering
  - Performance
- Our toolbox
  - Checksums
  - Timeouts
  - Acks and Nacks
  - Sequence numbering
  - Pipelining

We will use these to build Automatic Repeat Request (ARQ) protocols

- Stop-and-wait
- Pipelining
  - Go-back-N
  - Selective Repeat

Principles of reliable data transfer

- important in application, transport, link layers
  - top-10 list of important networking topics!



(a) provided service

- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Principles of reliable data transfer

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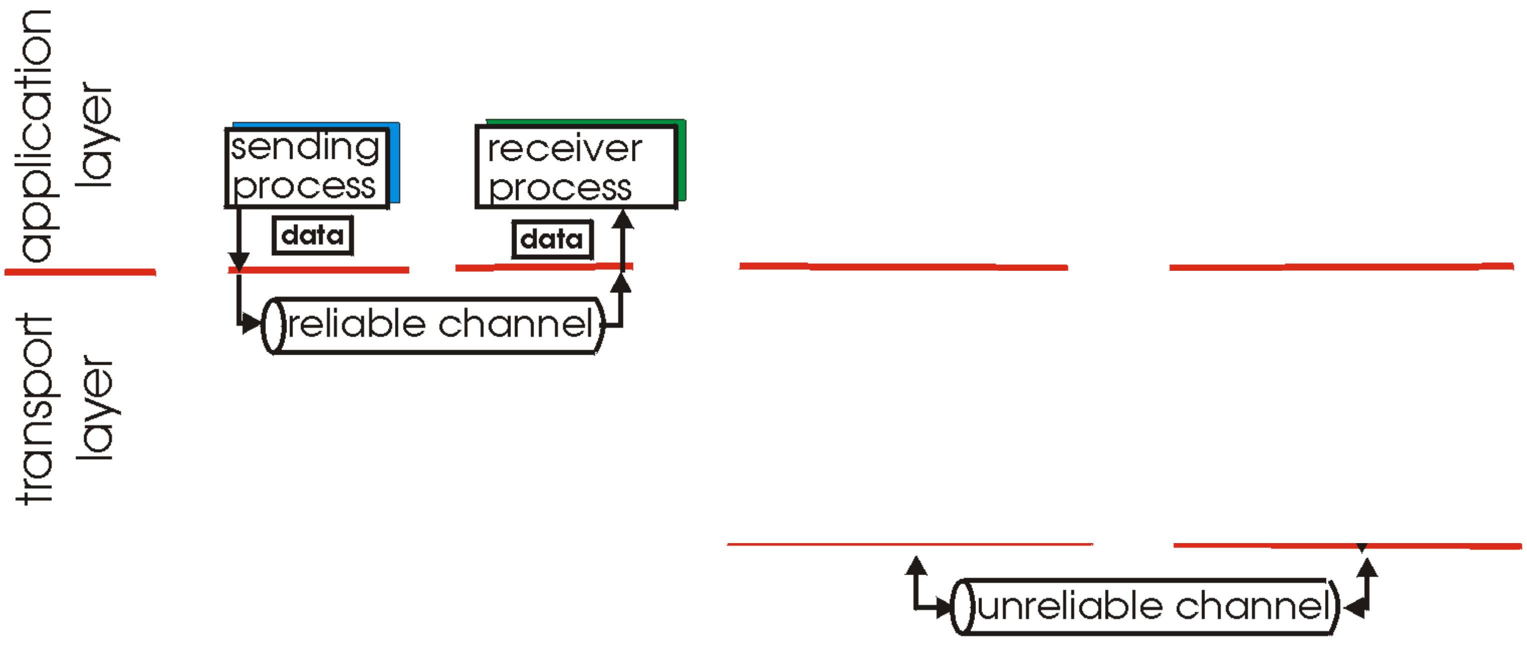

(a) provided service (b) service implementation

- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Principles of reliable data transfer

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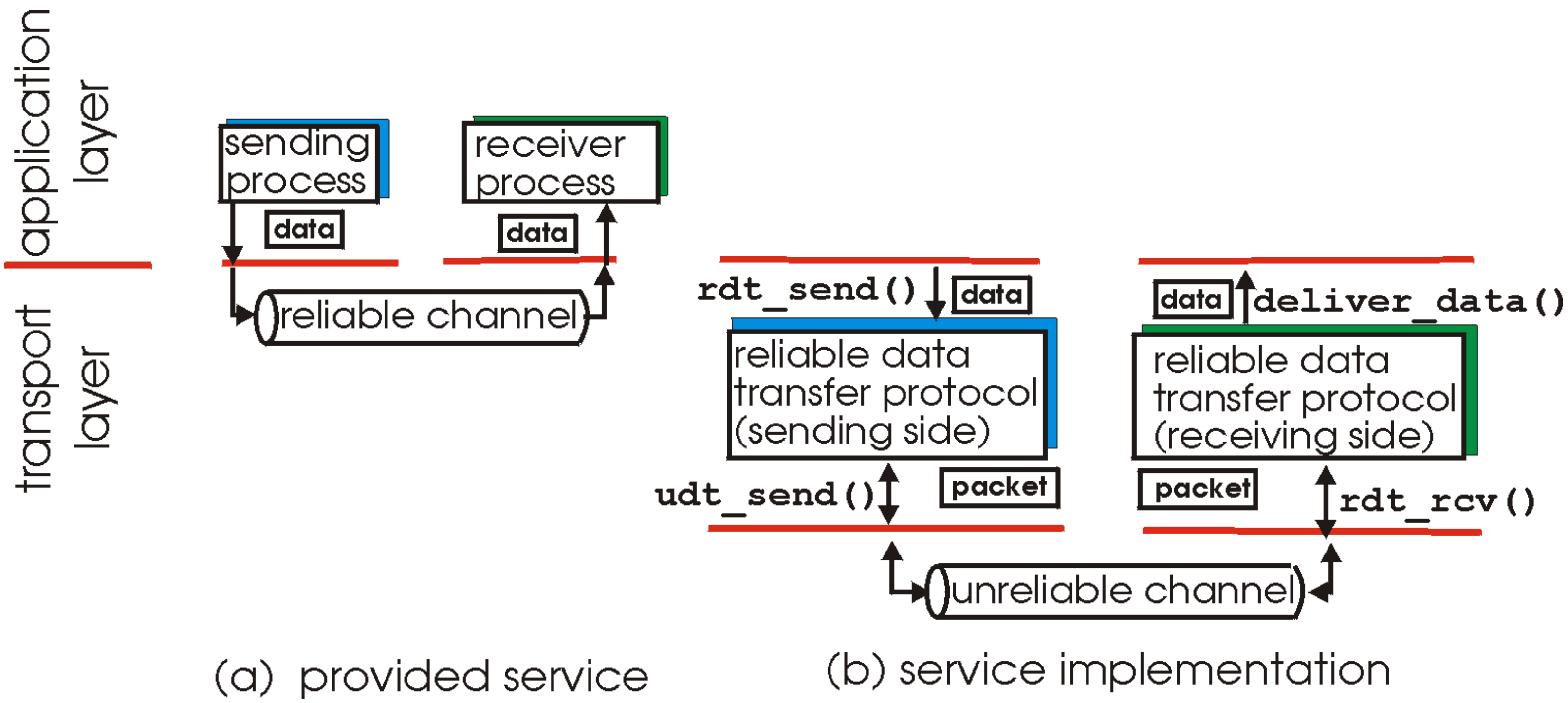


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- Transport layer does not have to do anything !!

rdt2.0: channel with bit errors

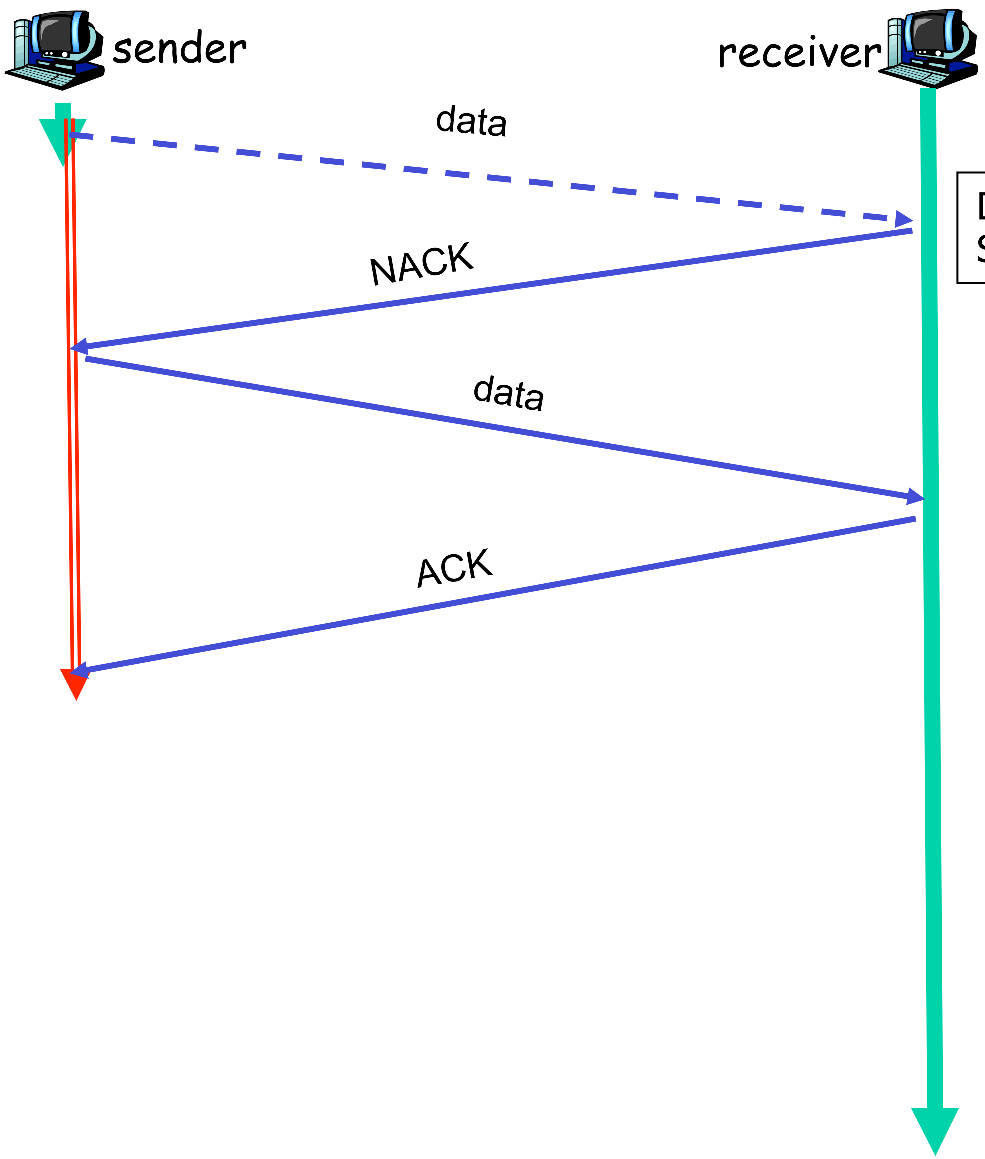
- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors:

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
- new mechanisms in **`rdt2.0`** (beyond **`rdt1.0`**):
  - error detection
  - feedback: control msgs (ACK,NAK) from receiver to sender

Global Picture of rdt2.0



Dotted line: erroneous transmission
Solid line: error-free transmission

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: discussion

sender:

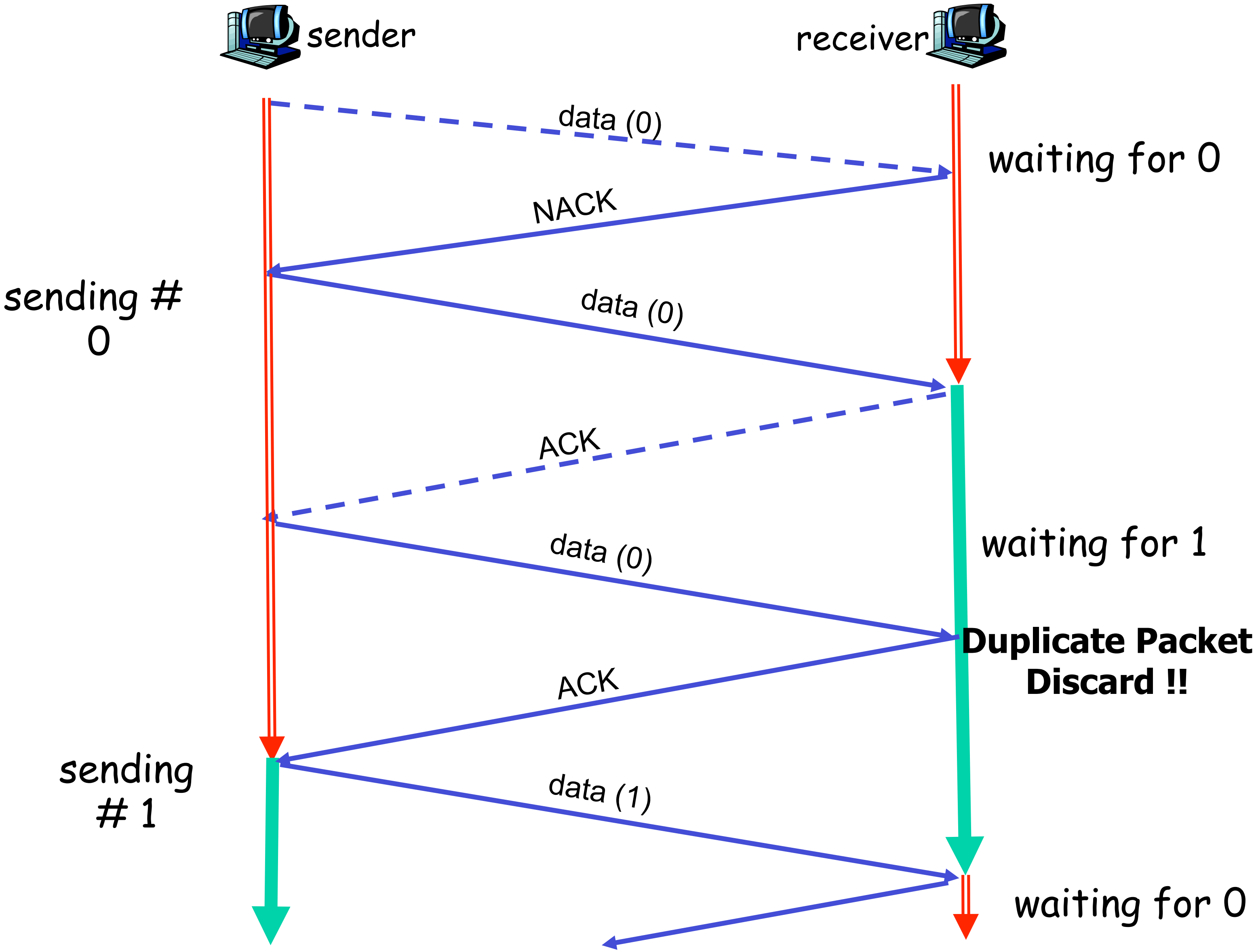
- seq # added to pkt
- two seq. #'s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "expected" pkt should have seq # of 0 or 1

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

Another Look at rdt2.1

Dotted line: erroneous transmission
Solid line: error-free transmission

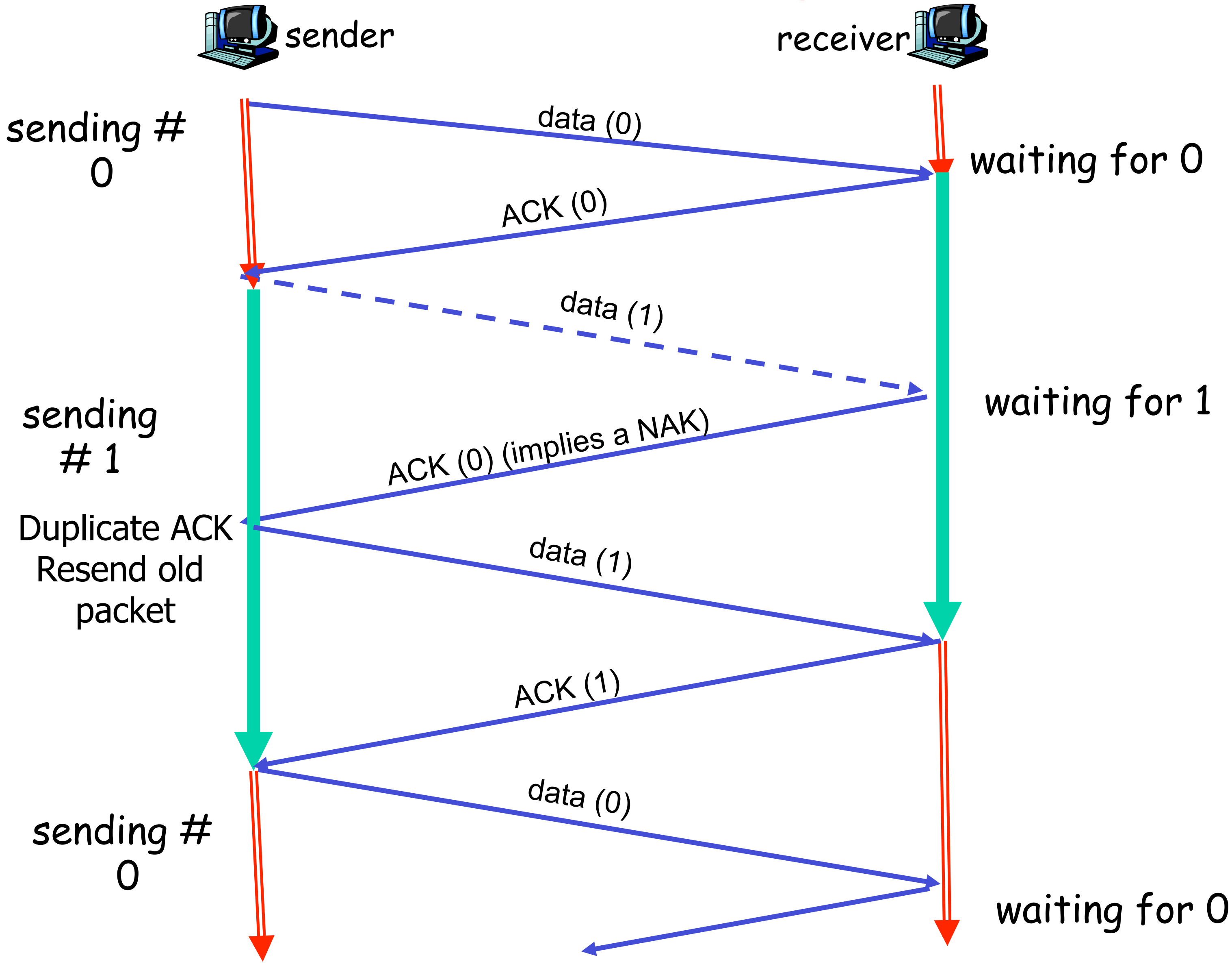


rdt2.2: a NAK-free protocol

- 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*

rdt2.2: Example

Dotted line: erroneous transmission
Solid line: error-free transmission



rdt3.0: channels with errors *and* loss

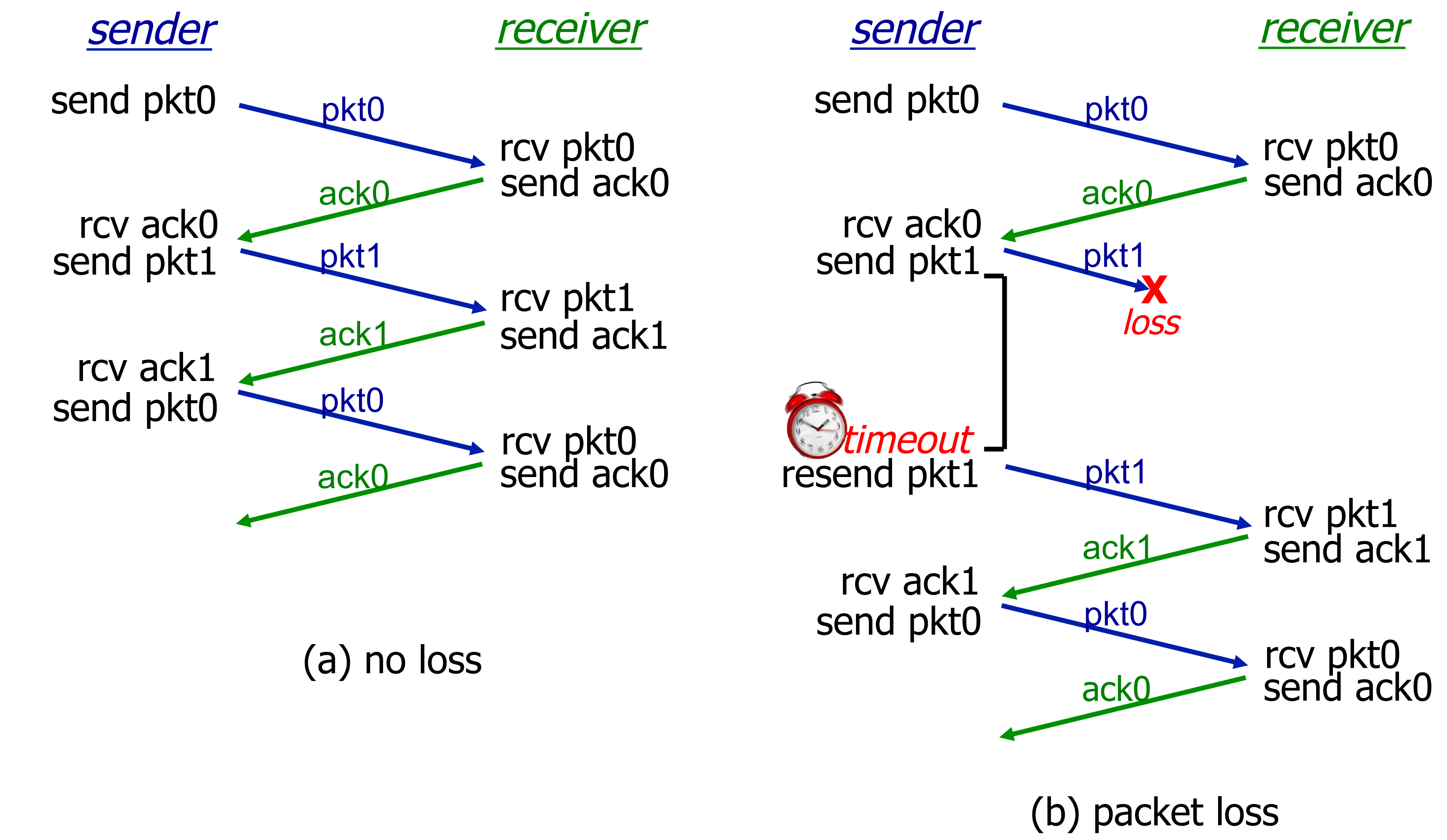
new assumption: underlying channel can also lose packets (data, ACKs)

- checksum, seq. #, ACKs, retransmissions will be of help … but not enough

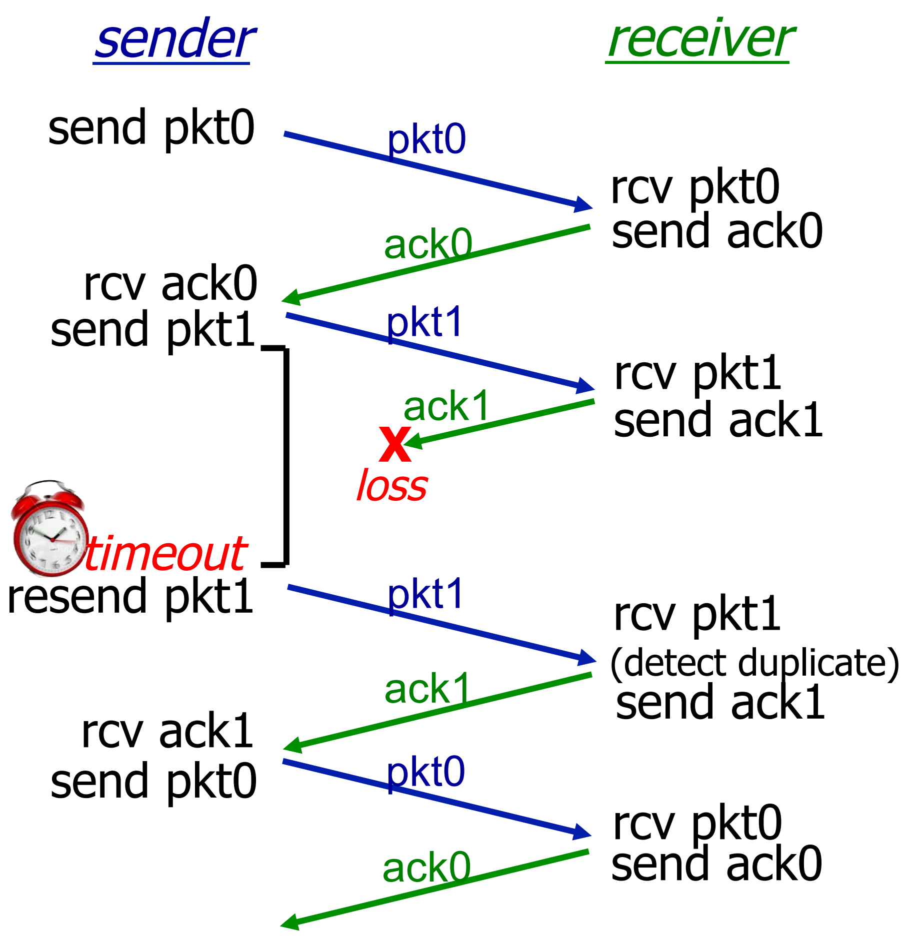
approach: sender waits "reasonable" amount of time for ACK

- 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 pkt being ACKed
- requires countdown timer

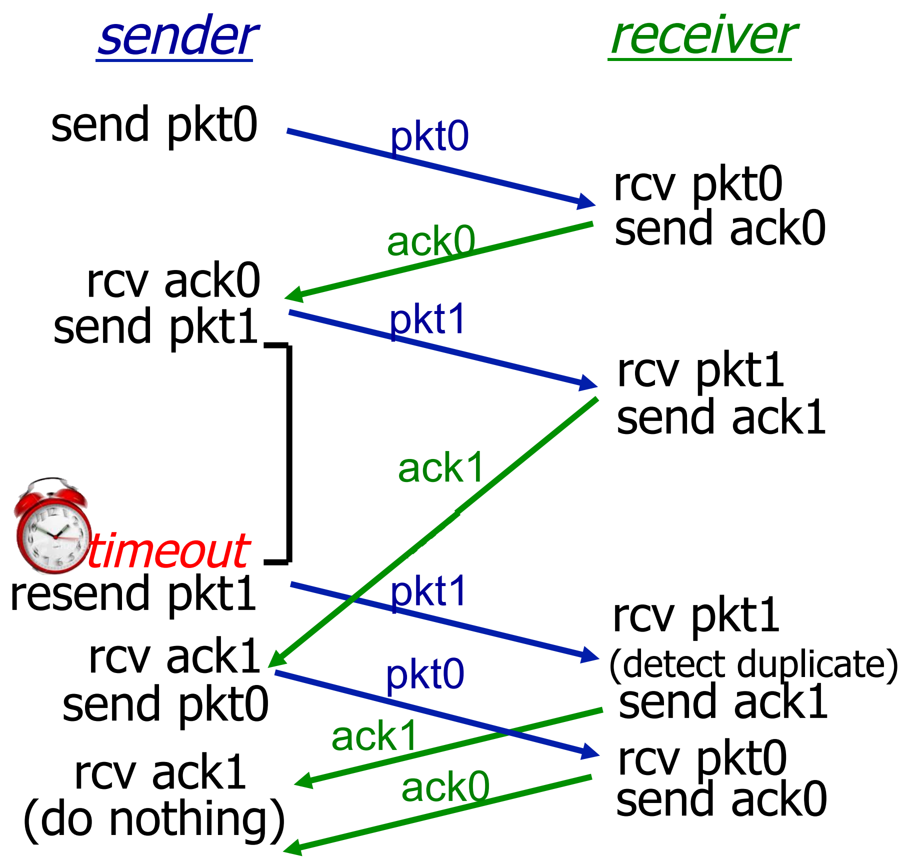
rdt3.0 in action



rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

Transport Part 1: Summary

- principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
- UDP

- **Next Week:**
  - Pipelined Protocols for reliable data transfer
  - TCP
    - TCP Flow Control
    - TCF Connection Management
    - TCP Congestion Control