Computer Networks and Applications

COMP 3331/COMP 9331

Week 4

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

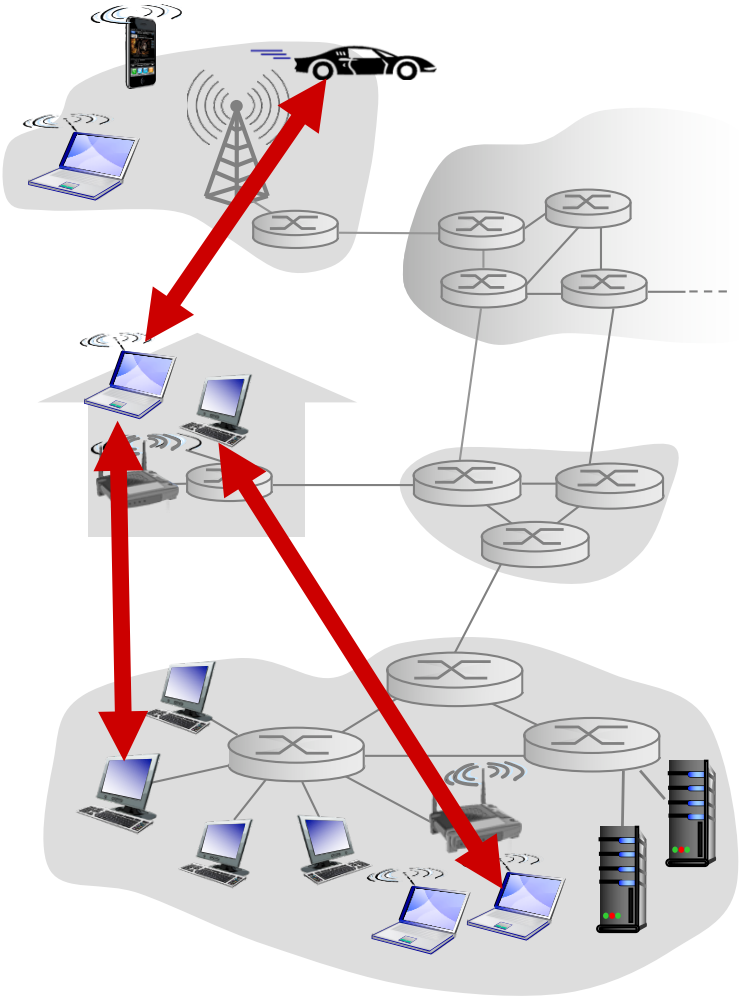
2.5 P2P applications

2.6 video streaming and content distribution networks (CDNs)

2.7 socket programming with UDP and TCP

Pure P2P architecture

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



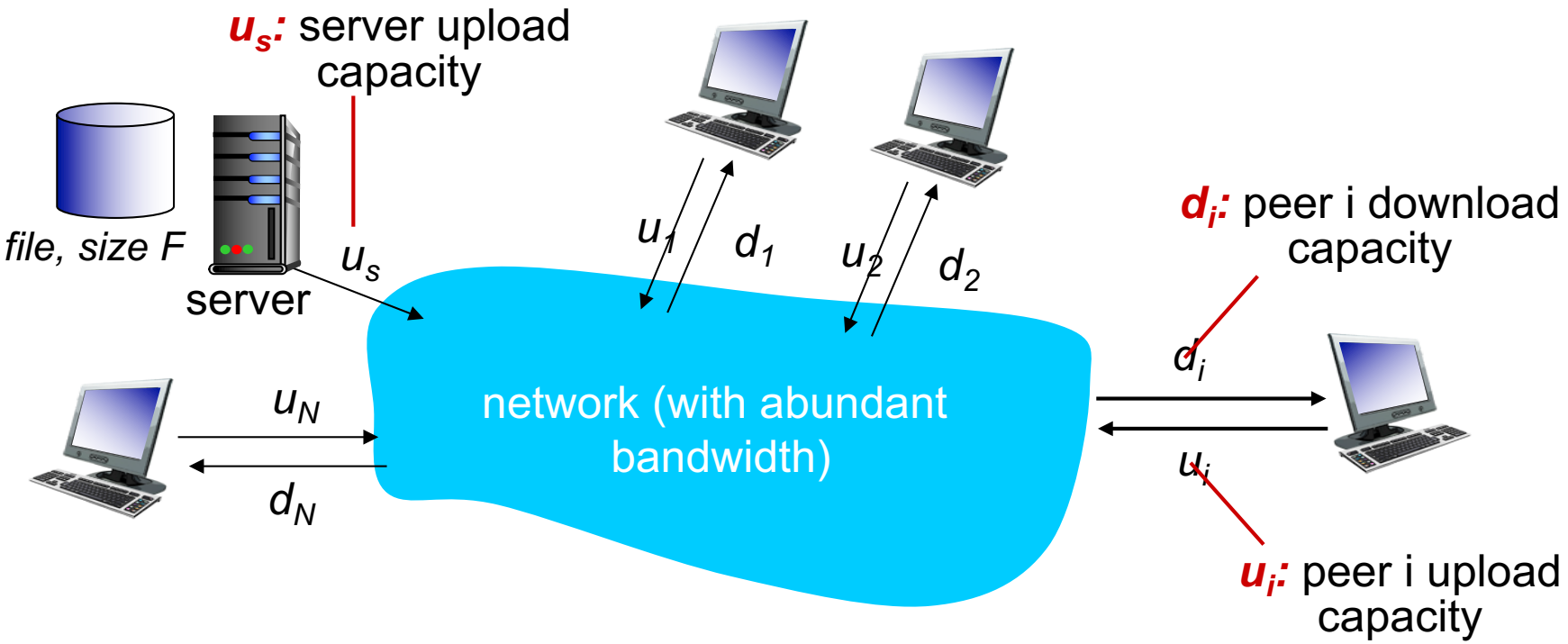
examples:

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

File distribution: client-server vs P2P

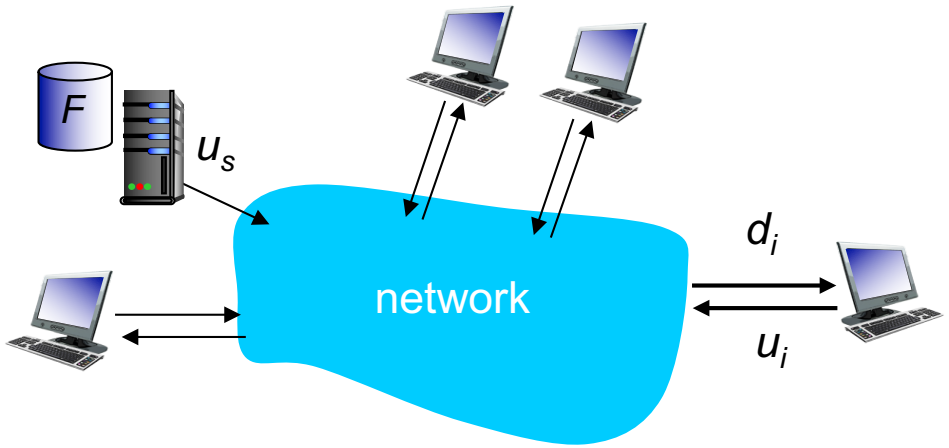
Question: how much time to distribute file (size *F*) from one server to *N* peers?

- 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}



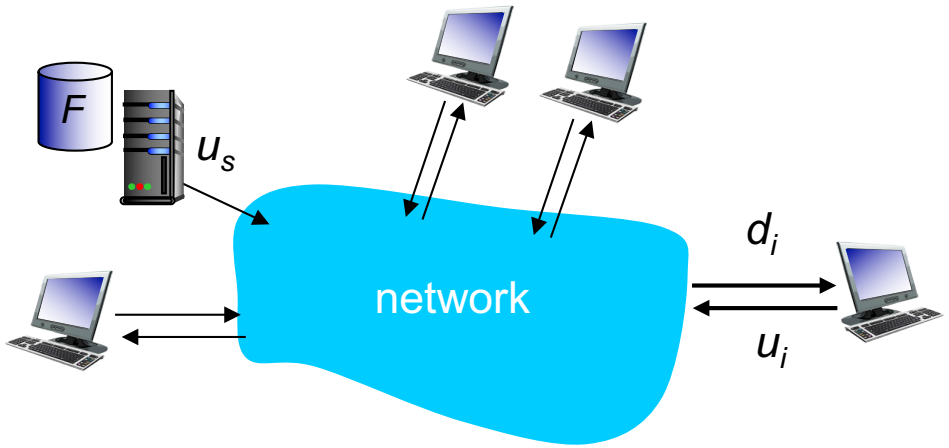
time to distribute F to N clients using client-server approach

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

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

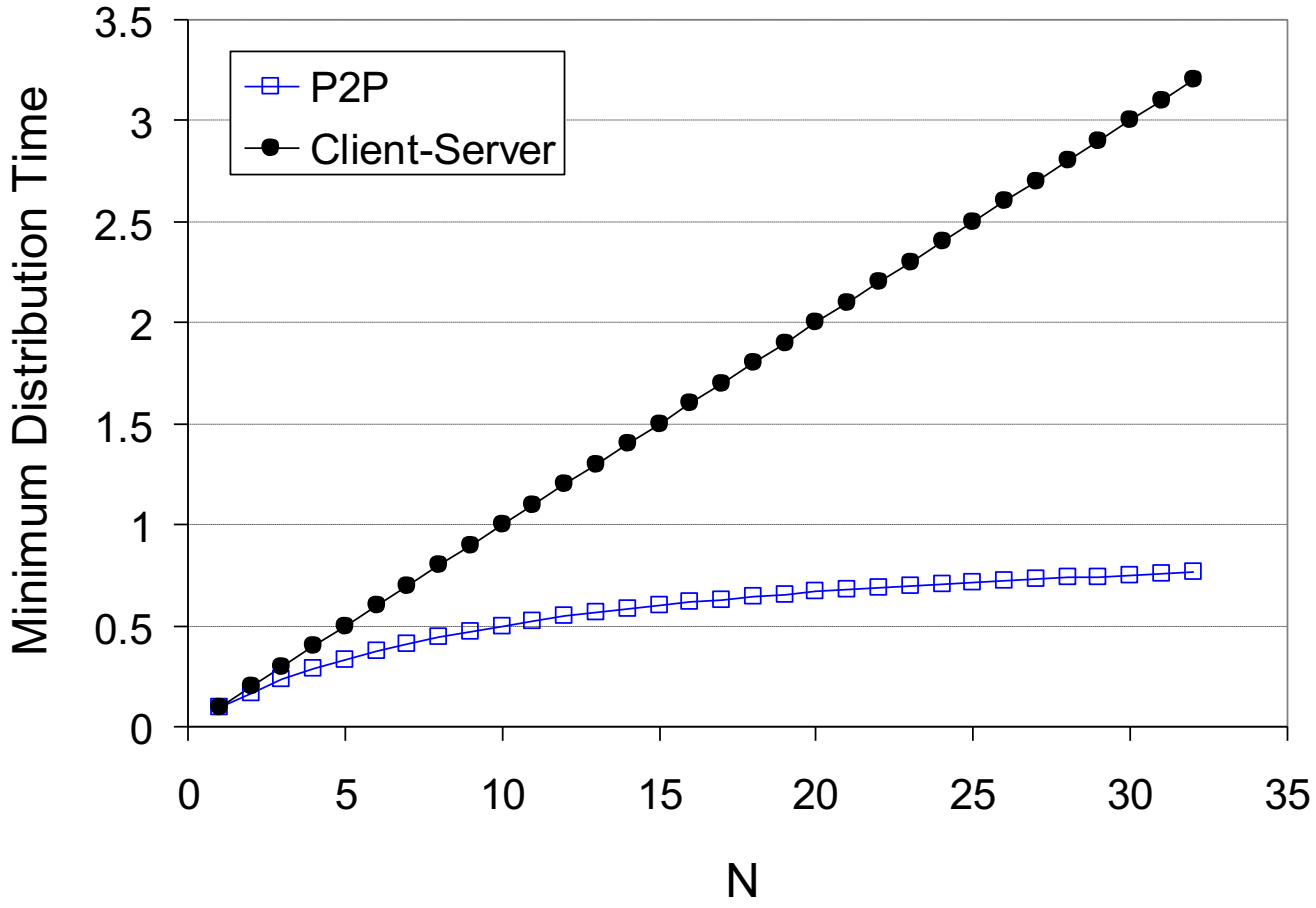
$$D_{P2P} \geq max\{F/u_s, F/d_{min}, NF/(u_s + \Sigma u_i)\}$$

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, $u_s = 10u$, $d_{min} \geq u_s$

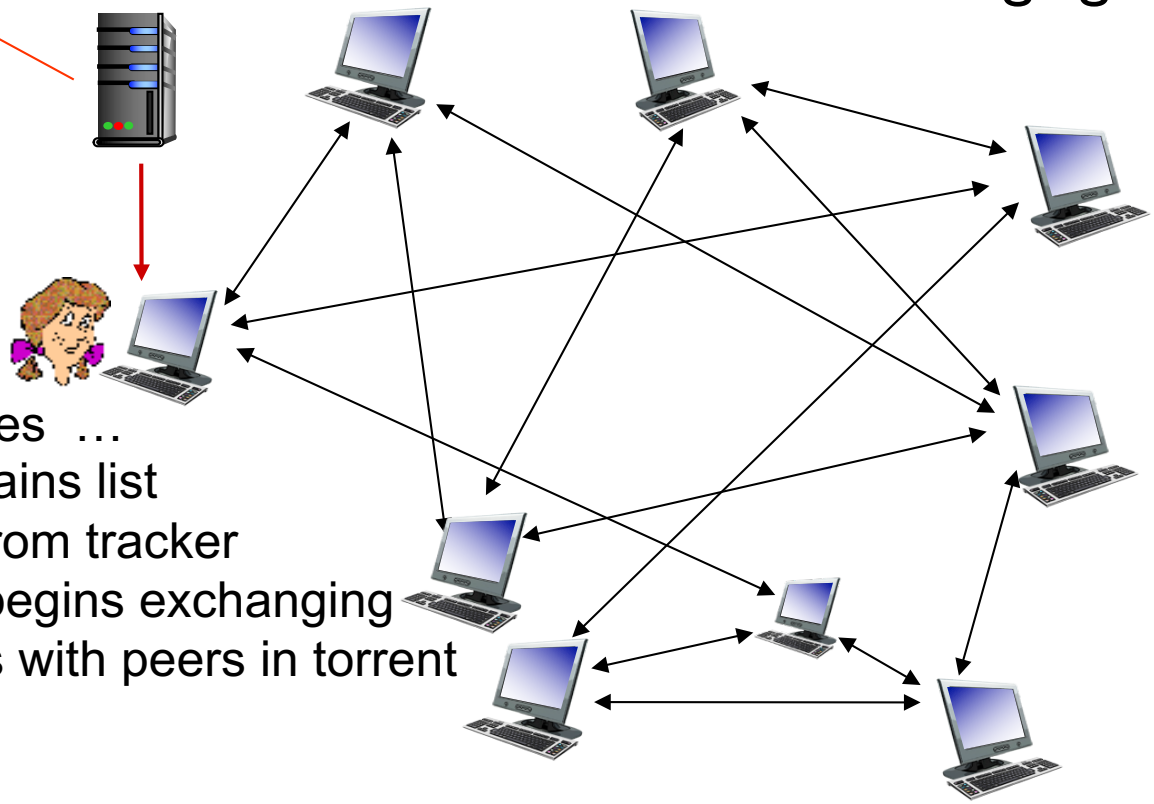


P2P file distribution: BitTorrent

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

tracker: tracks peers participating in torrent

torrent: group of peers exchanging chunks of a file



Alice arrives …
… obtains list of peers from tracker
… and begins exchanging file chunks with peers in torrent

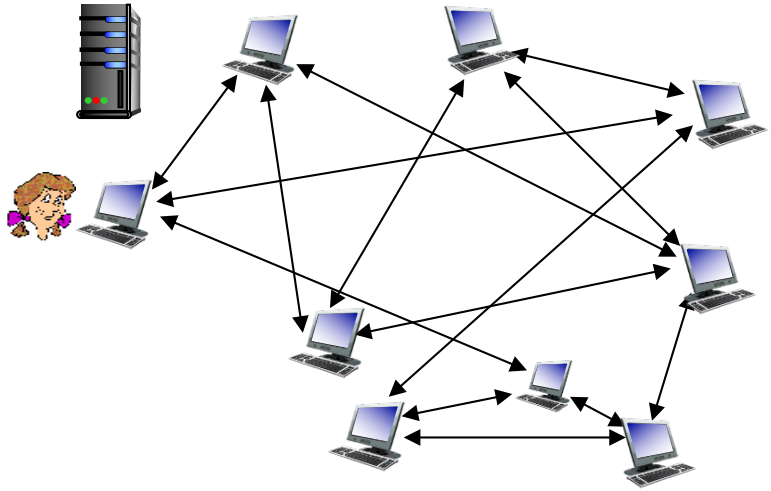
.torrent files

- Contains address of trackers for the file
  - Where can I find other peers?
- Contain a list of file chunks and their cryptographic hashes
  - This ensures that chunks are not modified

Title	**End-systems that have it**
House of Cards Season 4	124.45.66.128, 88.45.34.214
Walking Dead Season 6	111.111.111.111, 432.125.43.43
Game of Thrones Season 5	42.12.43.32
Better Call Saul Season 2	123.134.2.123

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

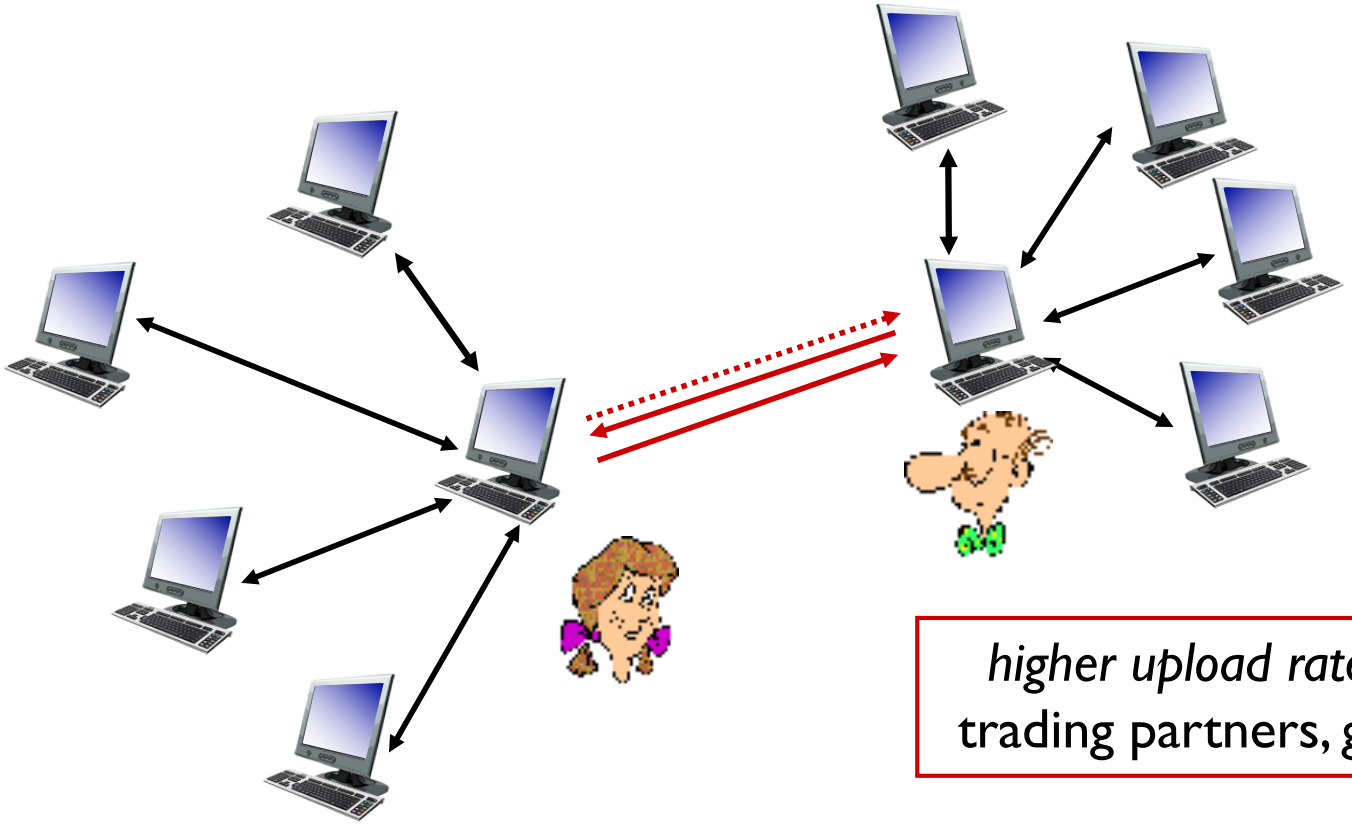
- Alice sends chunks to those four peers currently sending her chunks *at highest rate*
  - other peers are choked by Alice (do not receive chunks from her)
  - re-evaluate top 4 every10 secs
- every 30 secs: randomly select another peer, starts sending chunks
  - "optimistically unchoke" this peer
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

Original Research Paper on BitTorrent linked to OpenLearning

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

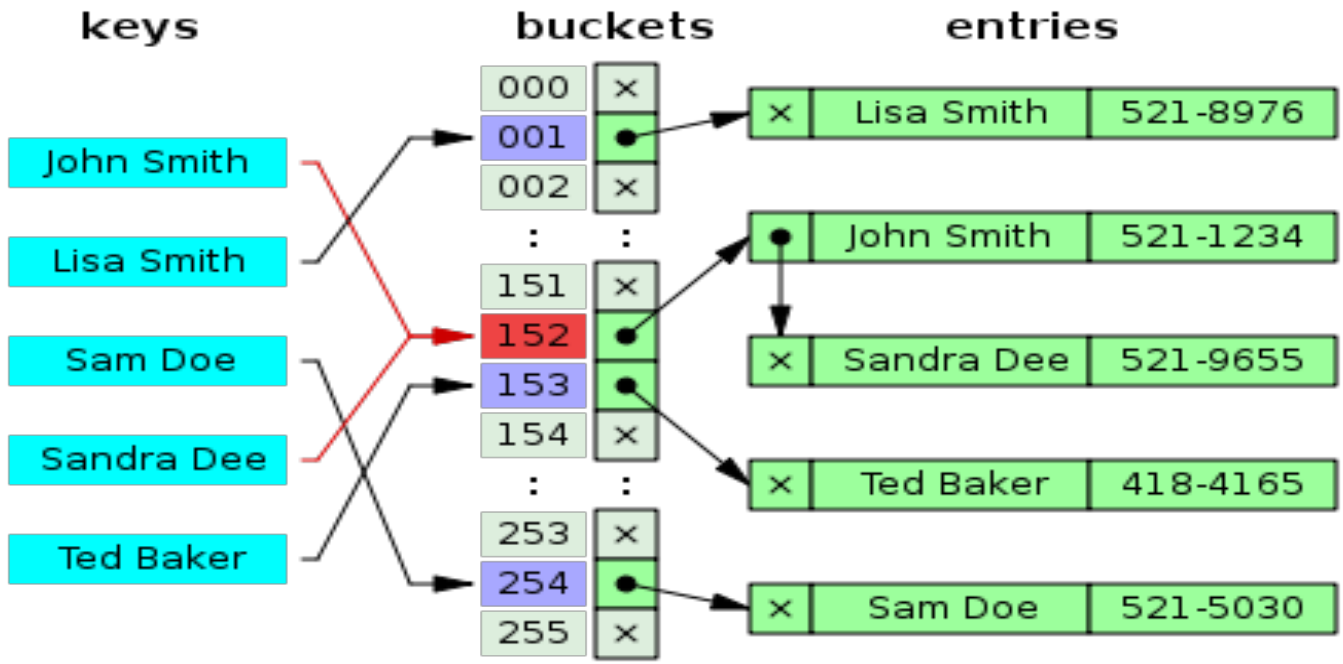


Figure src: http://en.wikipedia.org/wiki/Hash_table

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

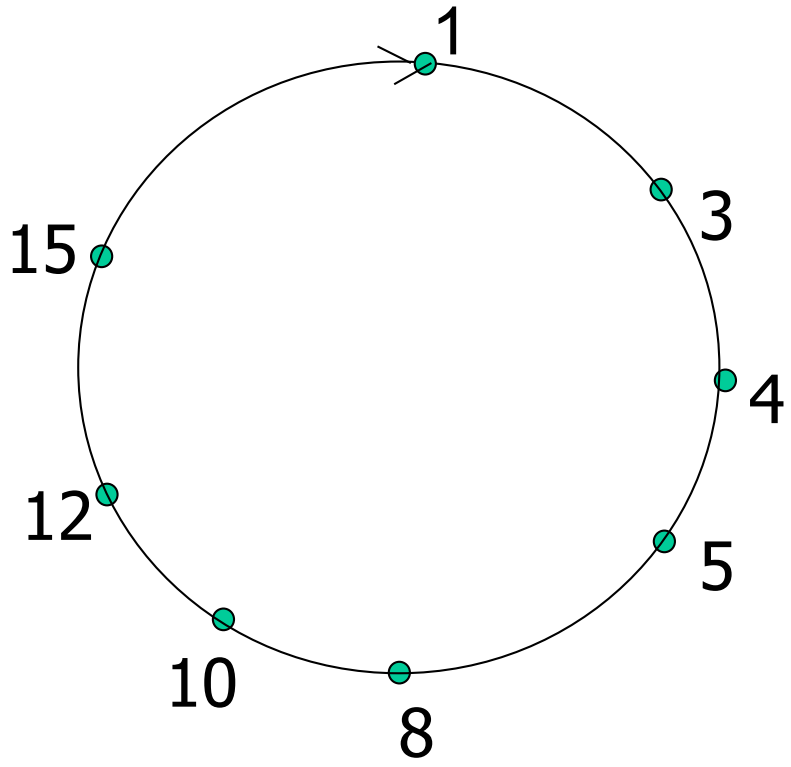
DHT identifiers: Consistent Hashing

- 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

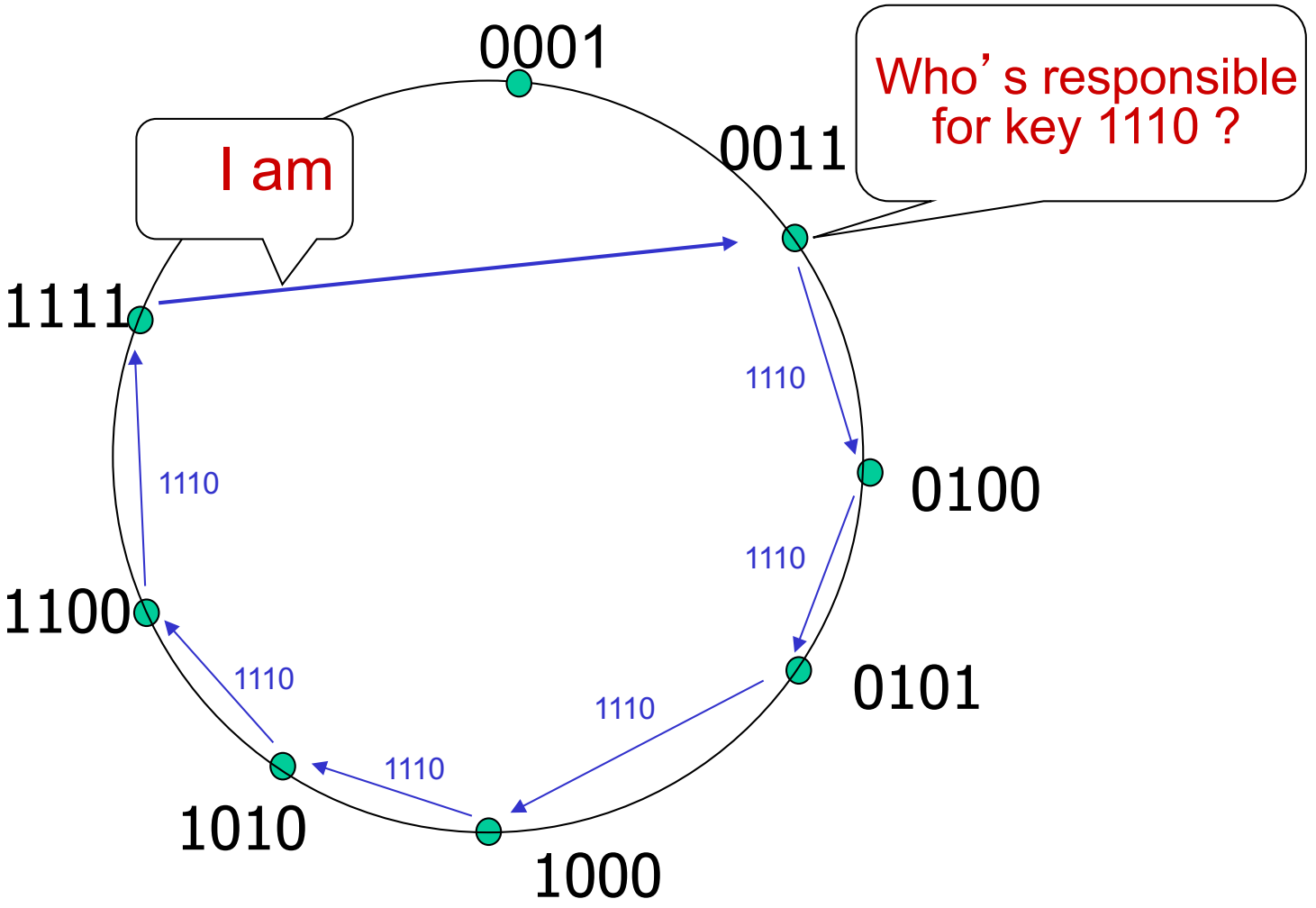
- rule: assign key to the peer that has the *closest* ID.
- common convention: closest is the *immediate successor* of the key.
- e.g., *n*=4; peers: 1,3,4,5,8,10,12,14;
  - key = 13, then successor peer = 14
  - key = 15, then successor peer = 1

Circular DHT (1)



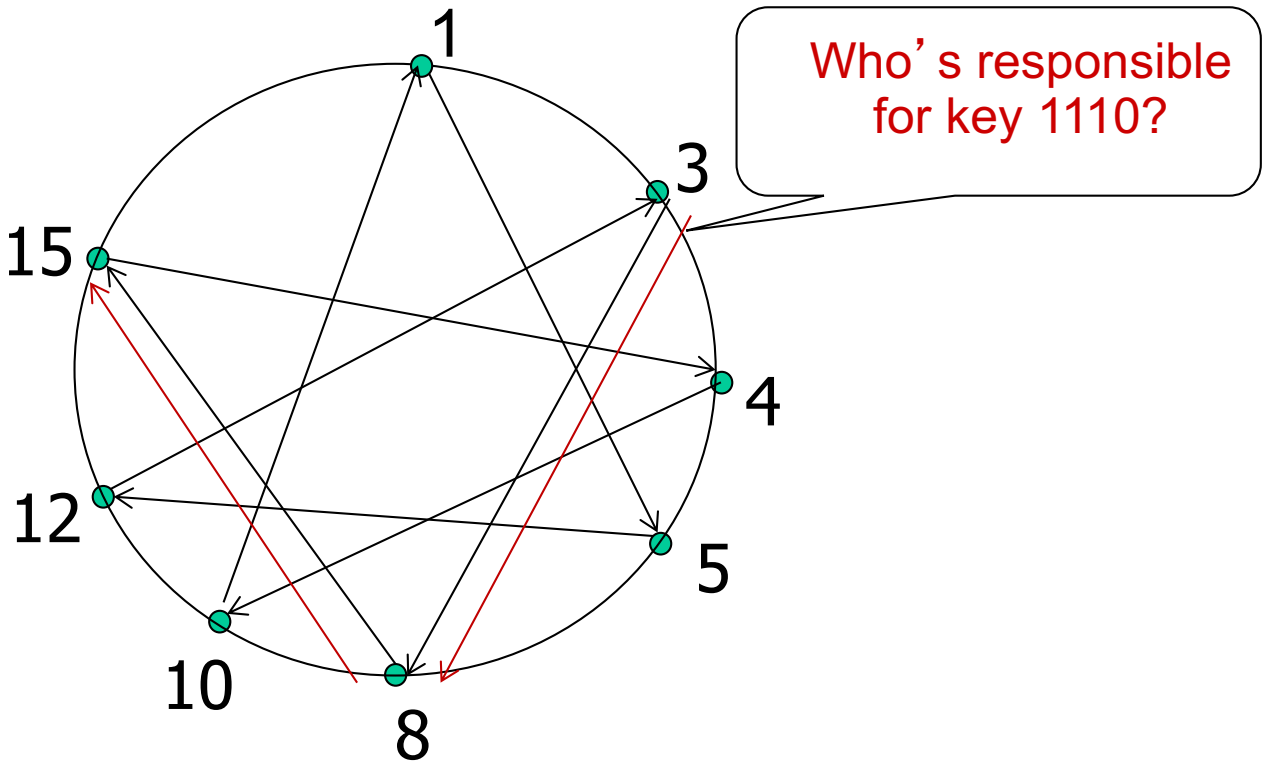
- each peer *only* aware of immediate successor and predecessor.
- "overlay network"

Circular DHT (2)



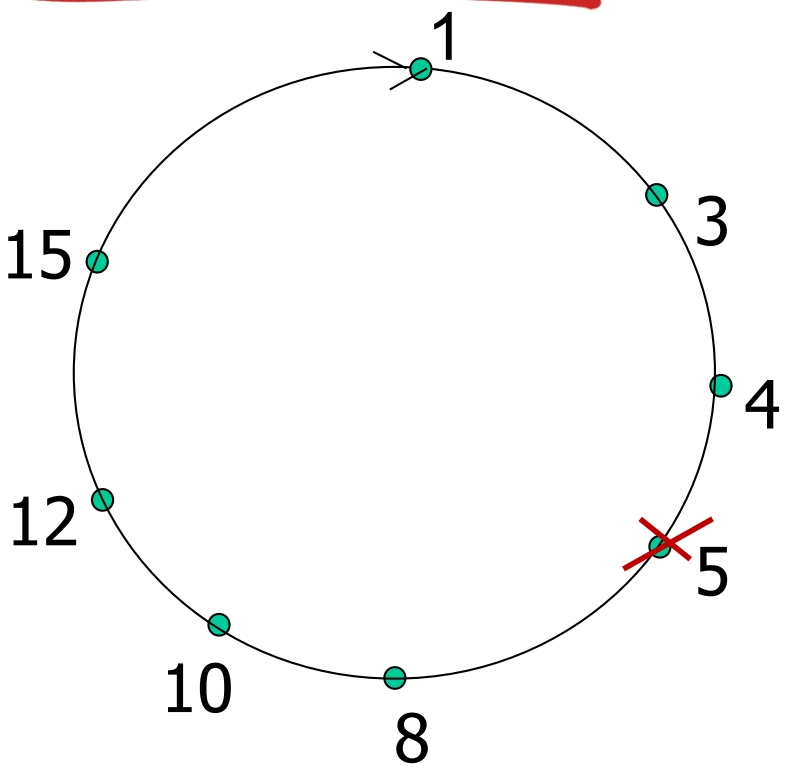
Define closest as closest successor

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 Papers (on the webpage):
Chord: A Scalable Peer-to-Peer Lookup Service for Internet Applications
Dynamo: Amazon's Highly Available Key-value Store

Complete all DHT quiz questions on OpenLearning

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

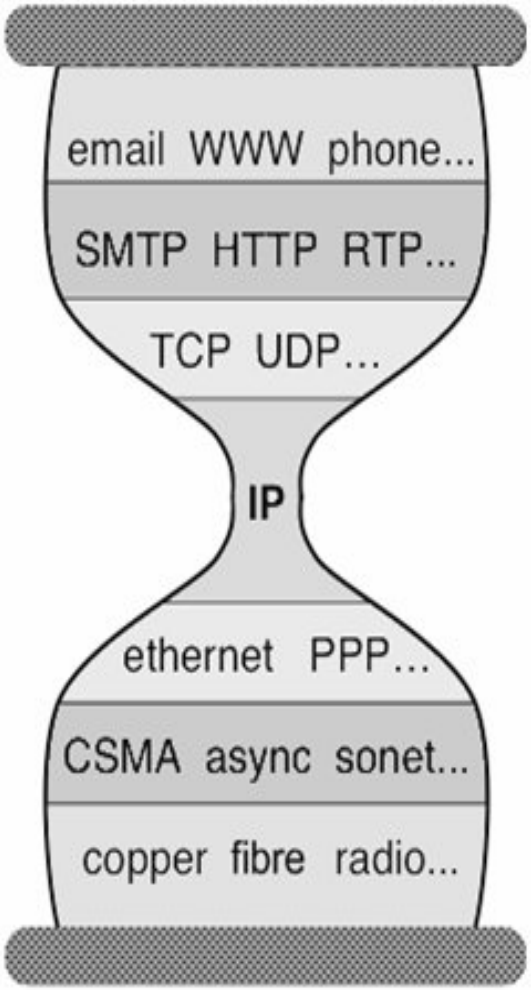
- segment structure
- reliable data transfer
- flow control
- connection management

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 !!

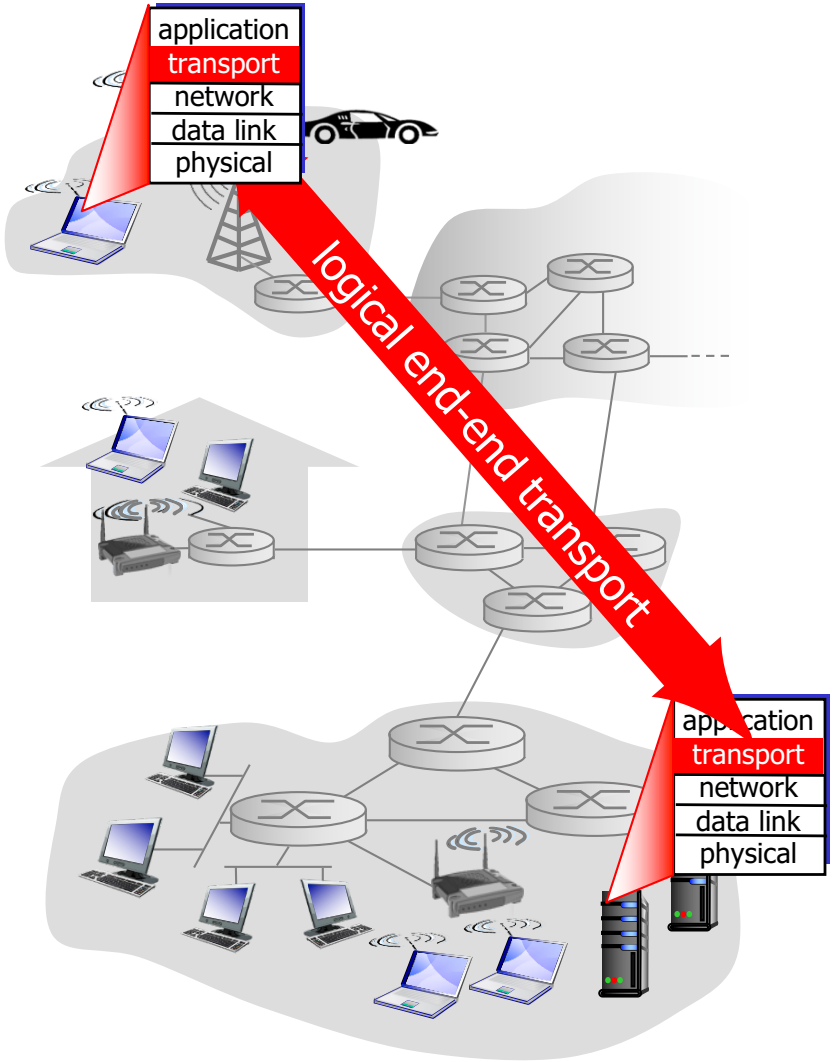


Network layer (context)

- 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!!)

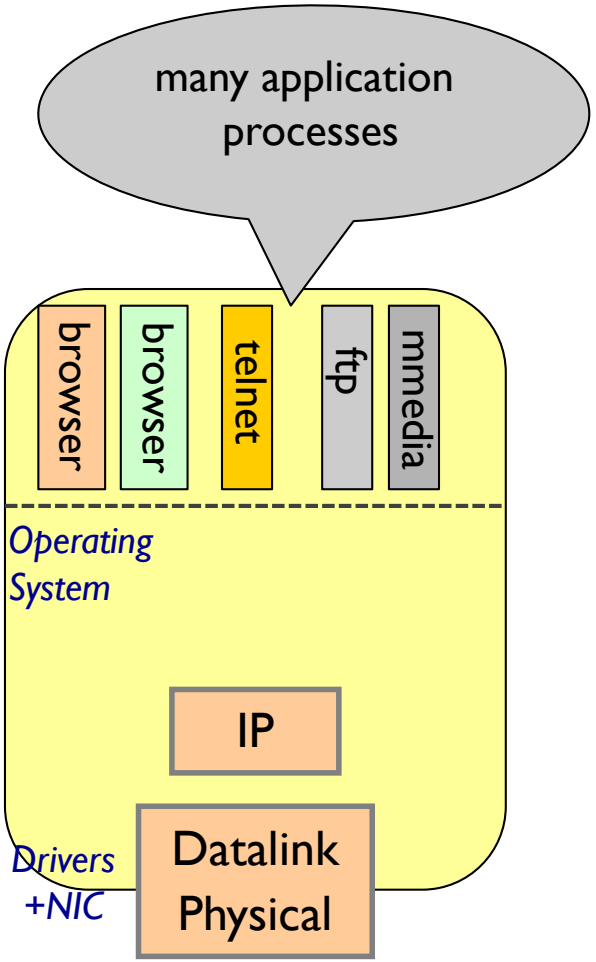
Transport services and protocols

- 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

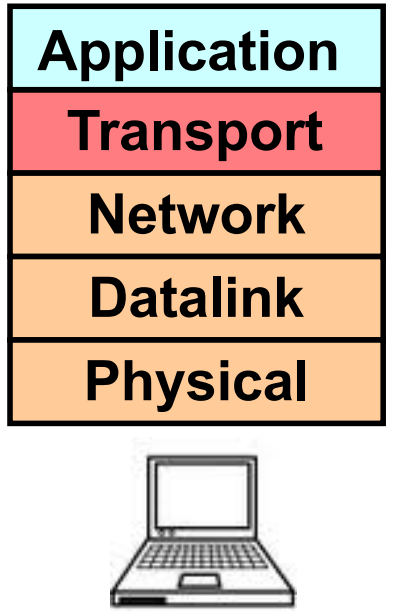


Network layer provides logical communication between hosts

Why a transport layer?

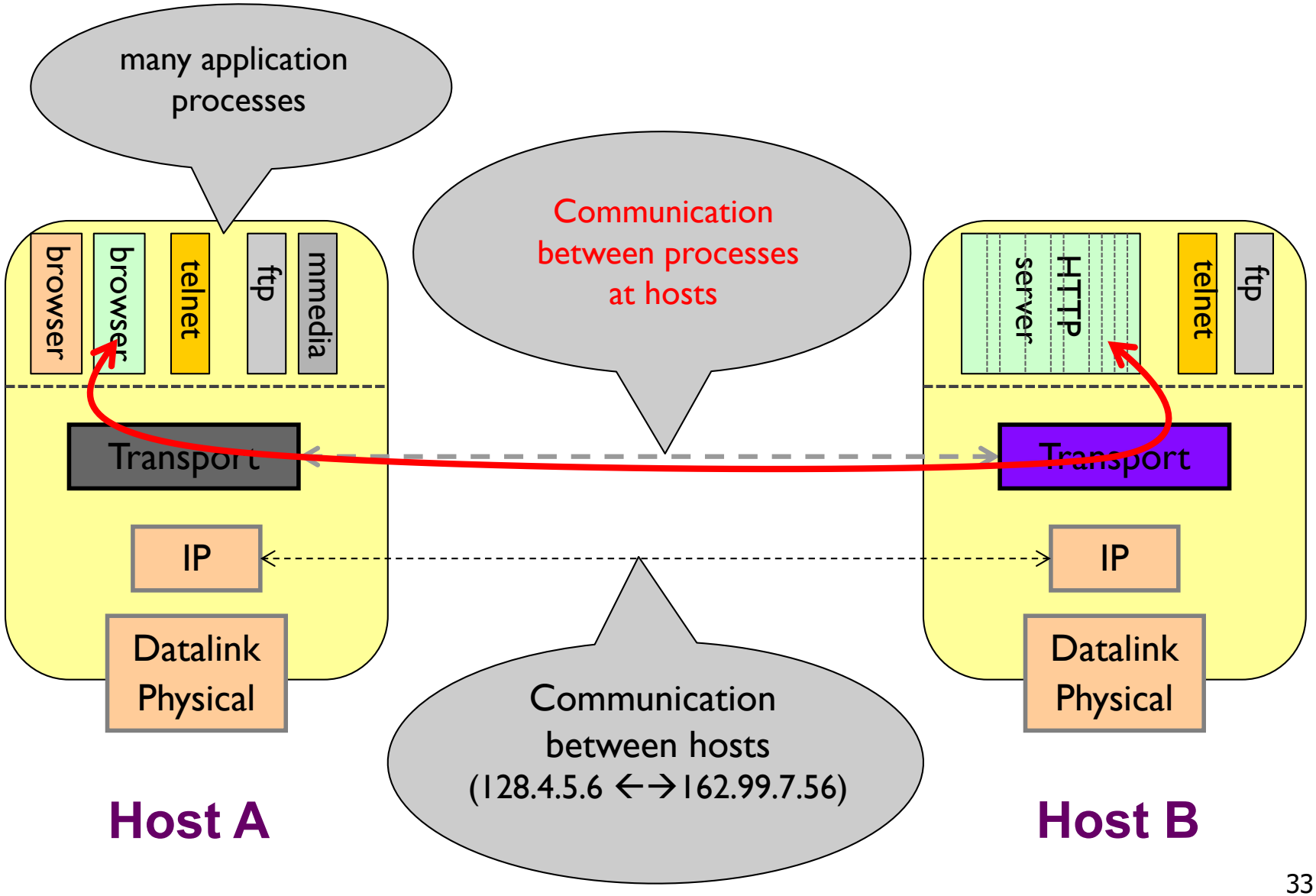


Host A



Host B

Why a transport layer?



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

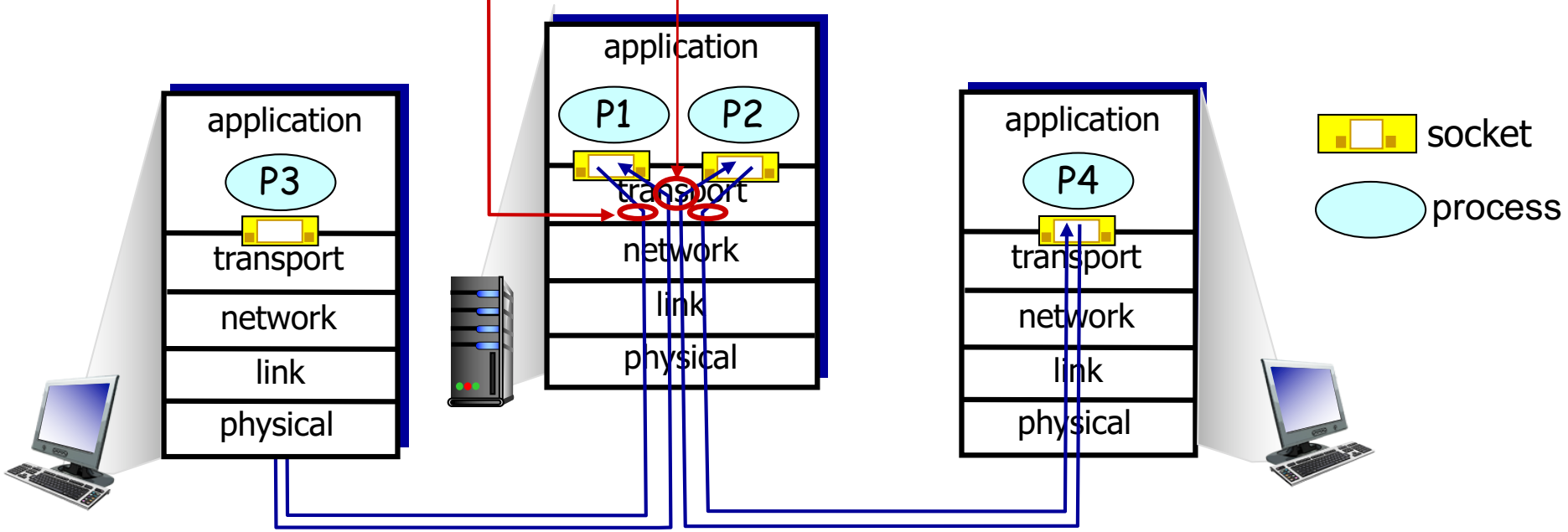
3.6 principles of congestion control

3.7 TCP congestion control

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 #

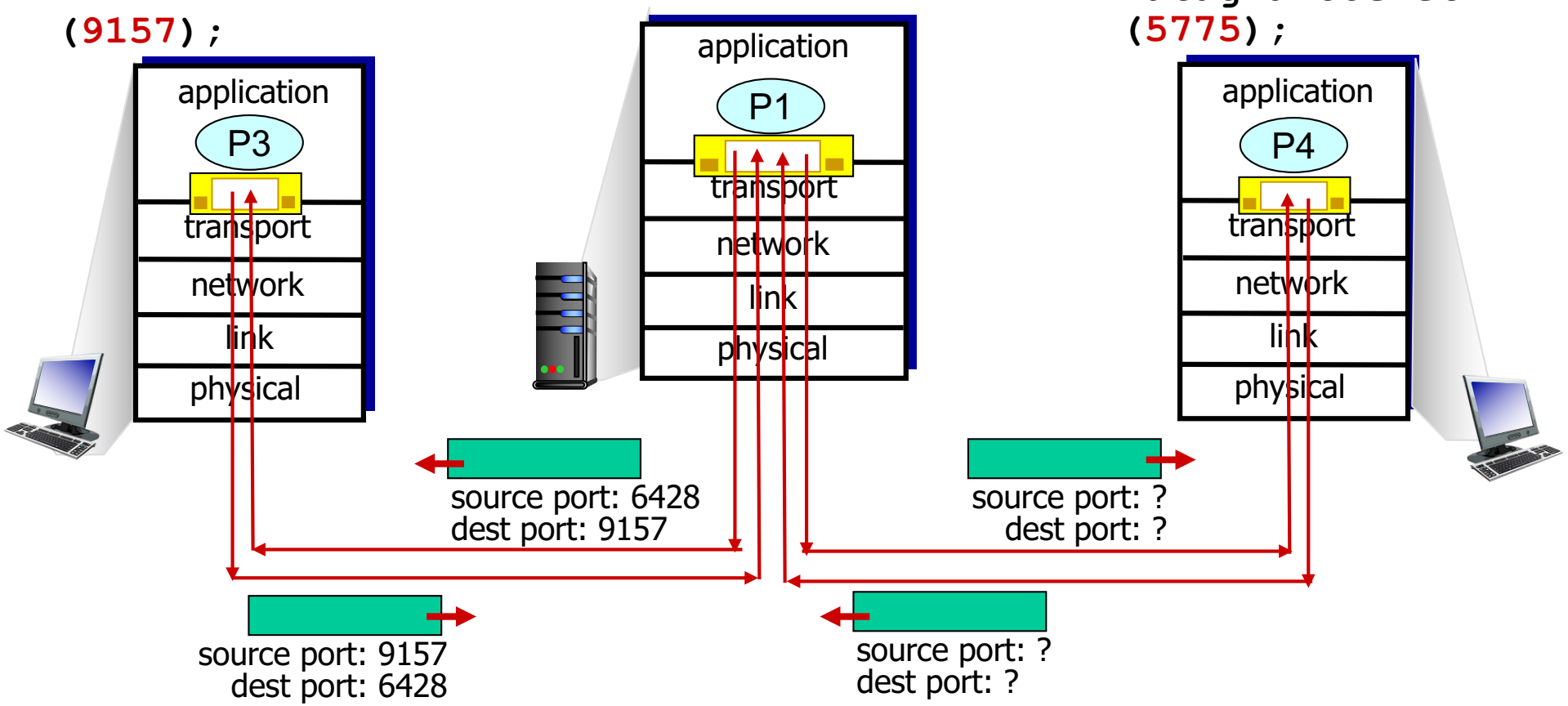
IP datagrams with *same dest. port #,* but different source IP addresses and/or source port numbers will be directed to *same socket* at dest

Connectionless demux: example

```
DatagramSocket
 mySocket2 = new
 DatagramSocket
 (9157);
```

```
DatagramSocket
 serverSocket = new
 DatagramSocket
 (6428);
```

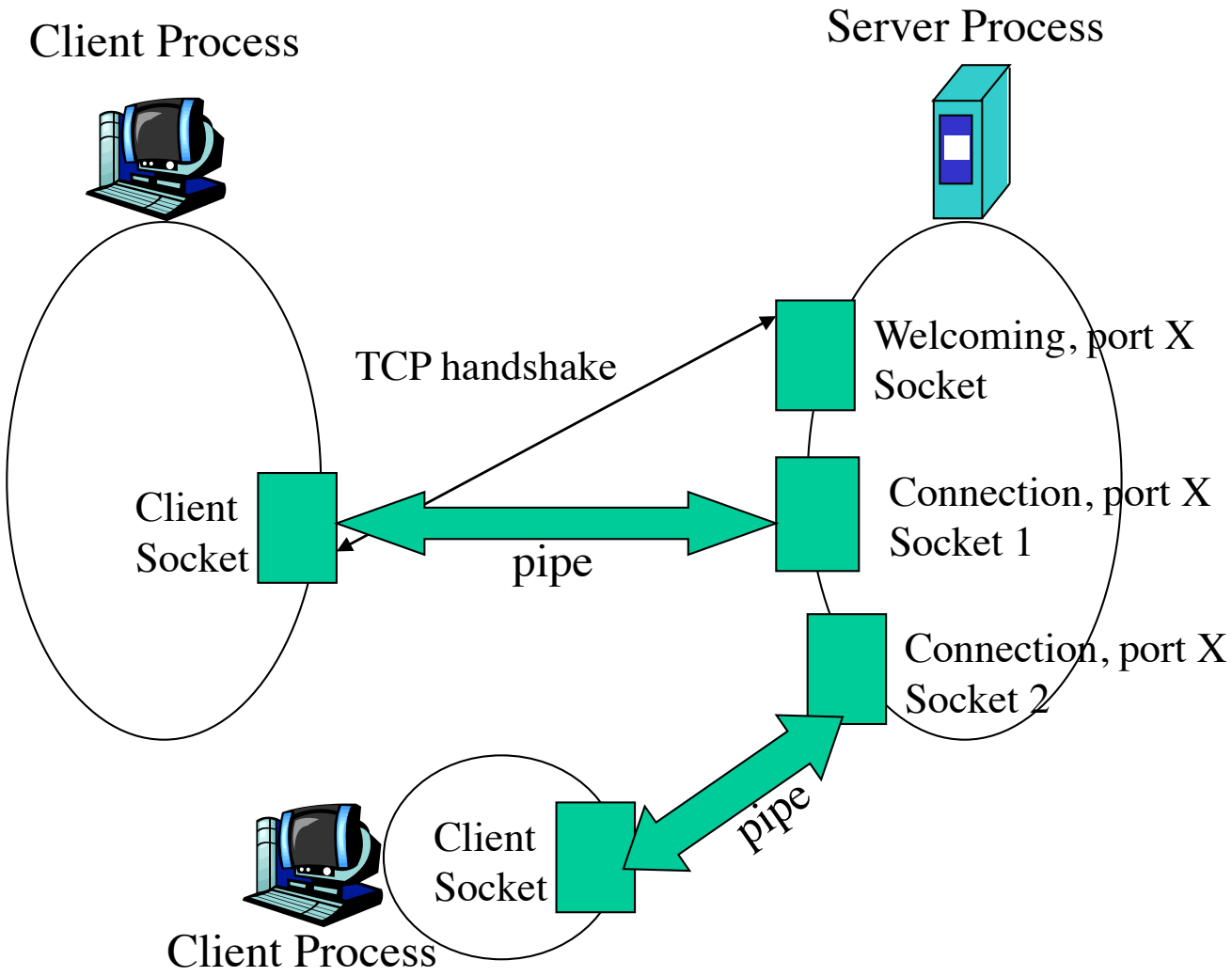
```
DatagramSocket
 mySocket1 = new
 DatagramSocket
 (5775);
```



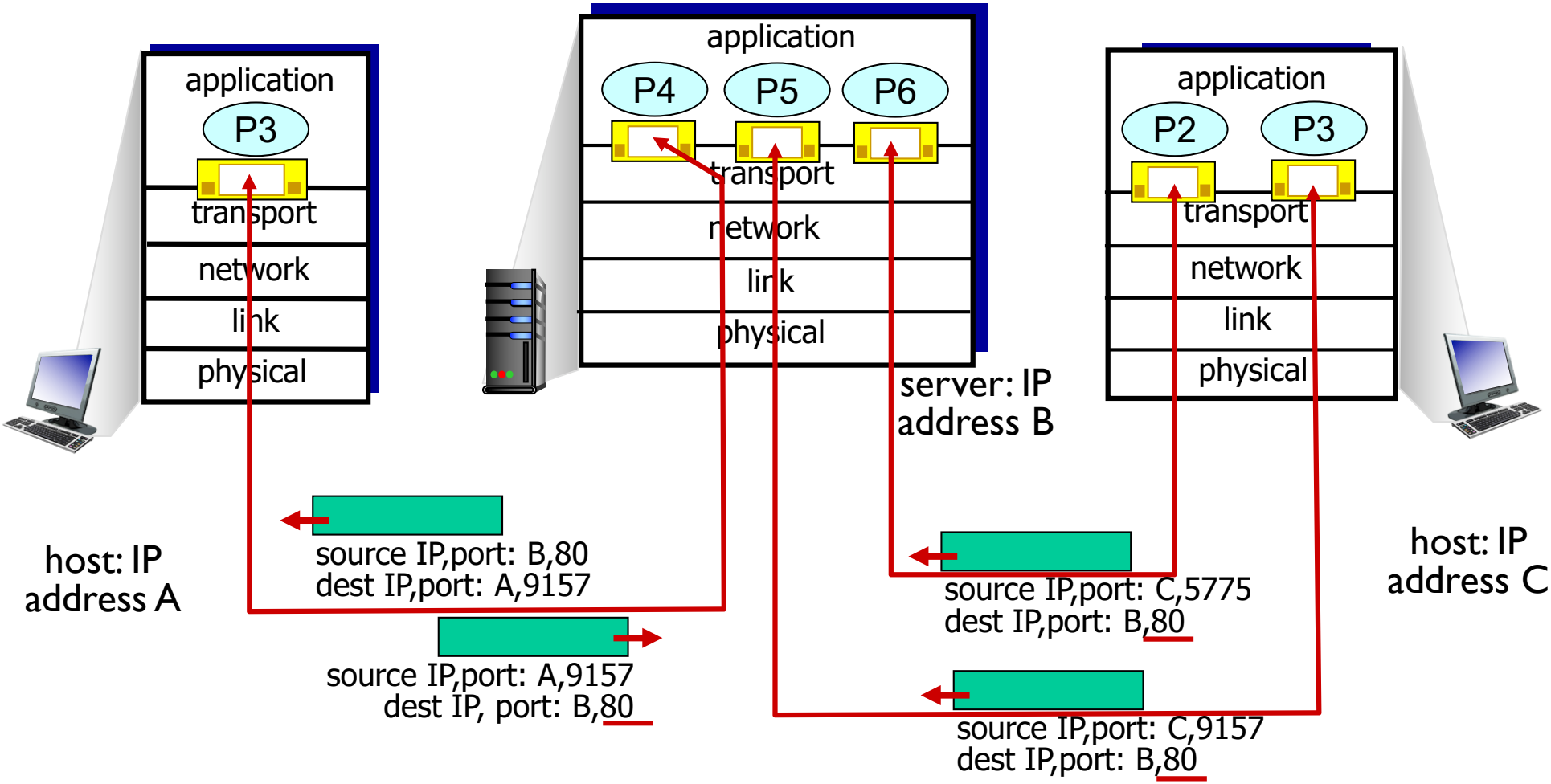
Connection-oriented demux

- 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



Connection-oriented demux: example



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

Quiz: Sockets

- Suppose that a Web Server runs in Host C on port 80. Suppose this server uses persistent connections, and is currently receiving requests from two different Hosts, A and B.
  - Are all of the requests being sent through the same socket at host C ?
  - If they are being passed through different sockets, do both of the sockets have port 80?

Complete all other questions on OpenLearning

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

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

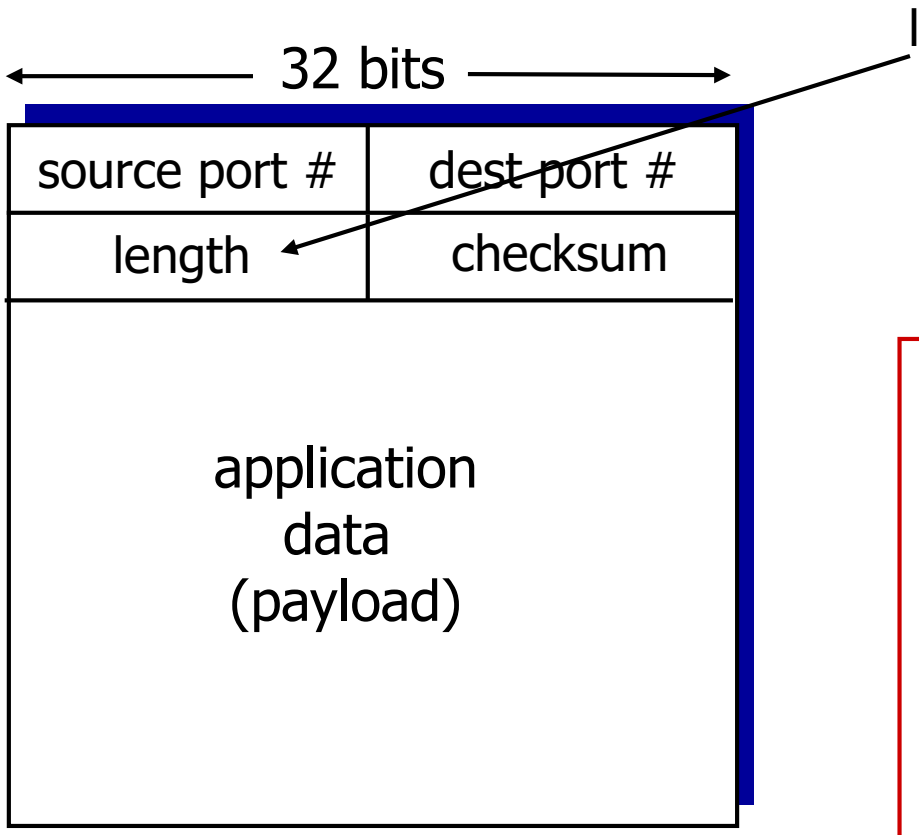
3.7 TCP congestion control

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

length, in bytes of UDP segment, including header

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:

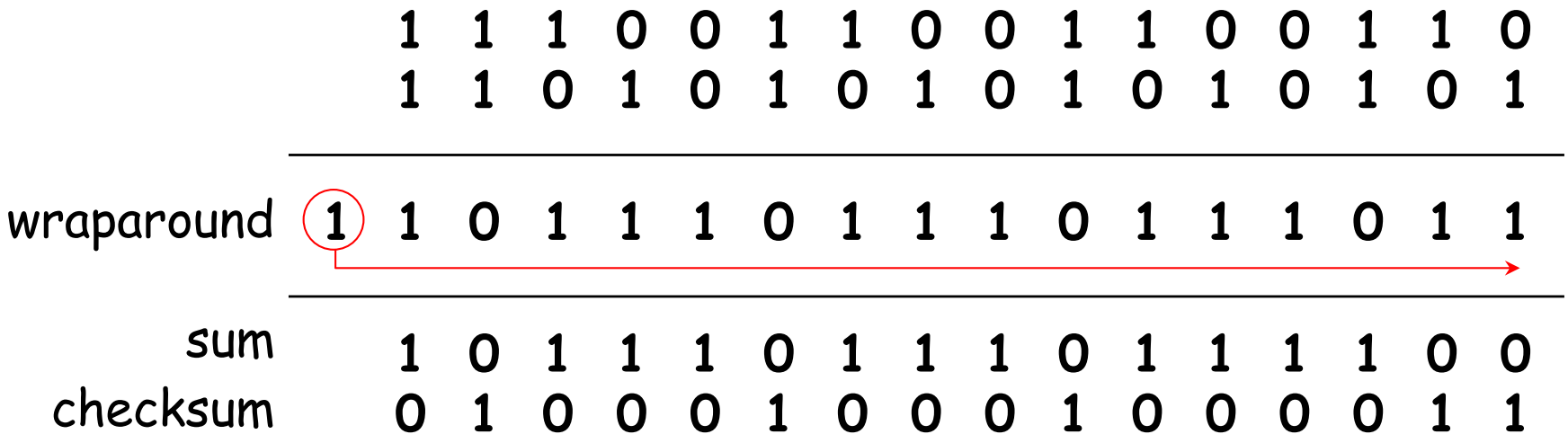
- treat segment contents, including header fields, as sequence of 16-bit integers
- checksum: addition (one's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

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



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)

There are a number of self-check questions on the UDP page on OpenLearning. Please complete them.

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

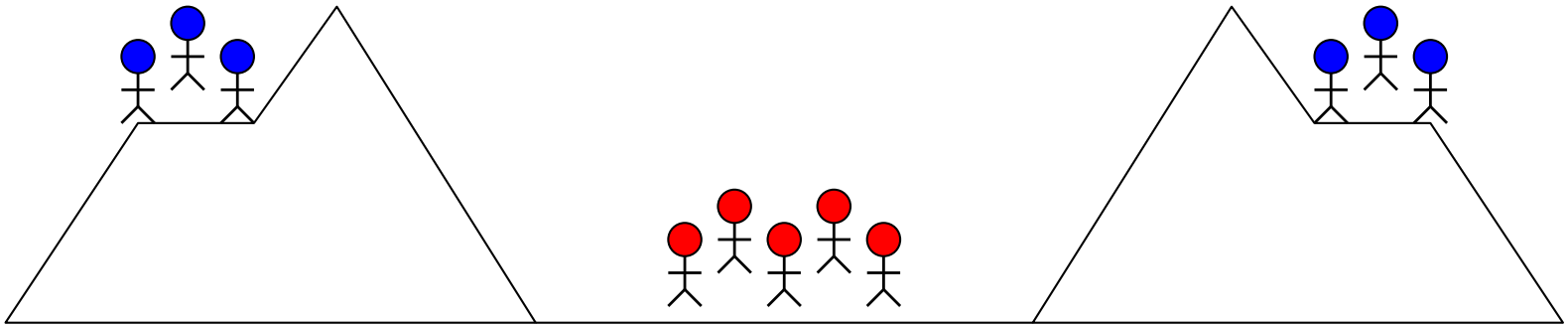
3.6 principles of congestion control

3.7 TCP congestion control

Reliable Transport

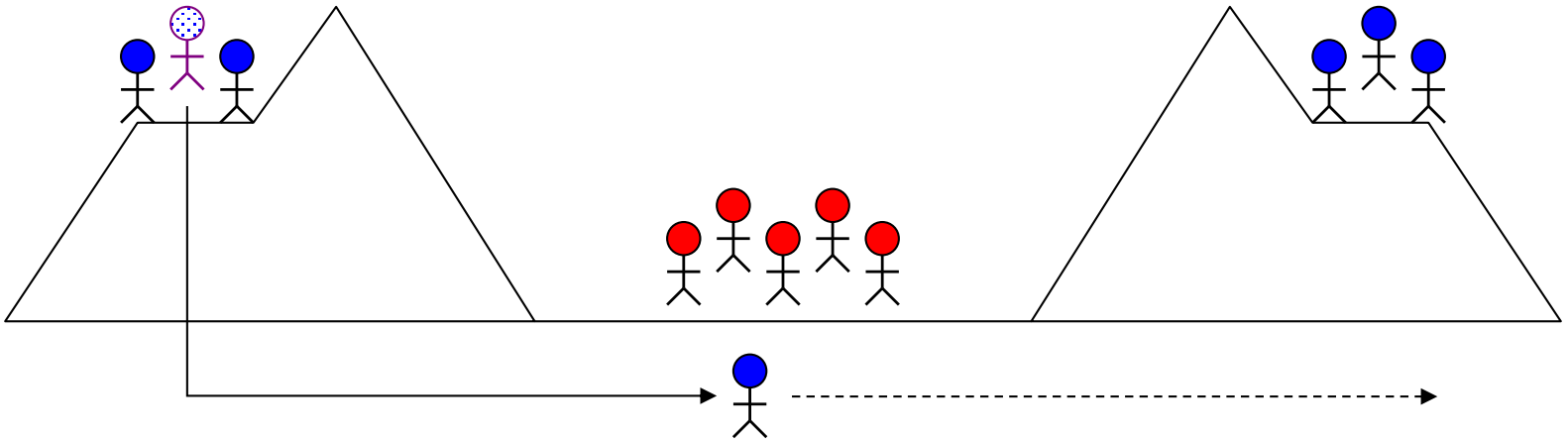
- In a perfect world, reliable transport is easy
- All the bad things best-effort can do
  - a packet is corrupted (bit errors)
  - a packet is lost
  - a packet is delayed (*why?*)
  - packets are reordered (*why?*)
  - a packet is duplicated (*why?*)

The Two Generals Problem



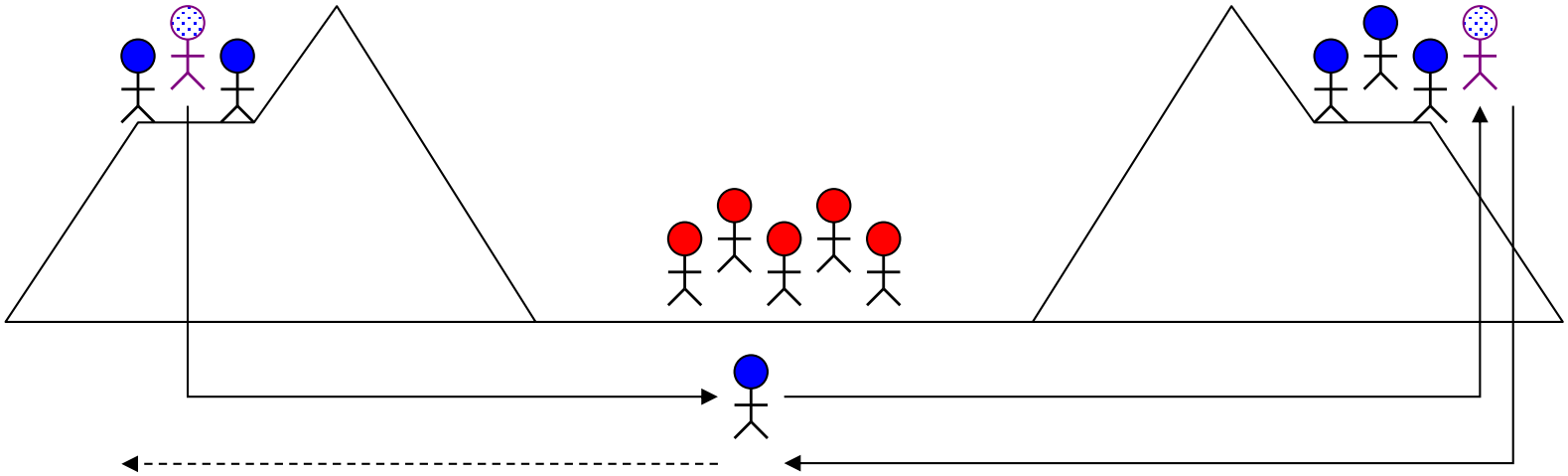
- Two army divisions (blue) surround enemy (red)
  - Each division led by a general
  - Both must agree when to simultaneously attack
  - If either side attacks alone, defeat
- Generals can only communicate via messengers
  - Messengers may get captured (unreliable channel)

The Two Generals Problem



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

The Two Generals Problem

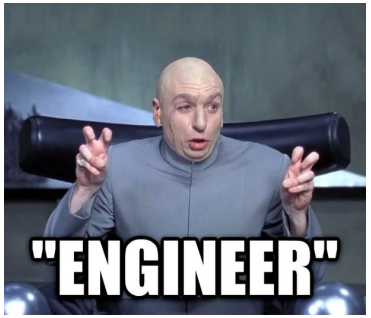


- How to be sure messenger made it?
  - Send acknowledgement: "We received message"

Quiz: Reliability

- In the "two generals problem", can the two armies reliably coordinate their attack?
  - A: Yes (explain how)
  - B: No (explain why not)

Engineering



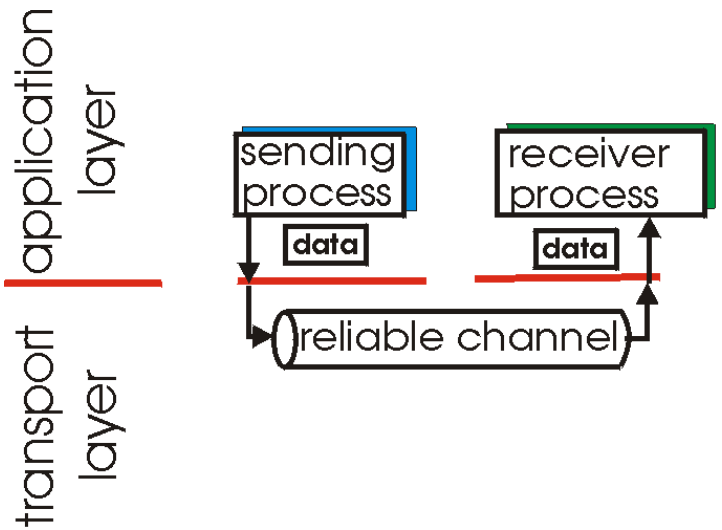
- Concerns
  - Message corruption
  - Message duplication
  - Message loss
  - Message reordering
  - Performance
- Our toolbox
  - Checksums
  - Timeouts
  - Acks and Nacks
  - Sequence numbering
  - Pipelining

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!

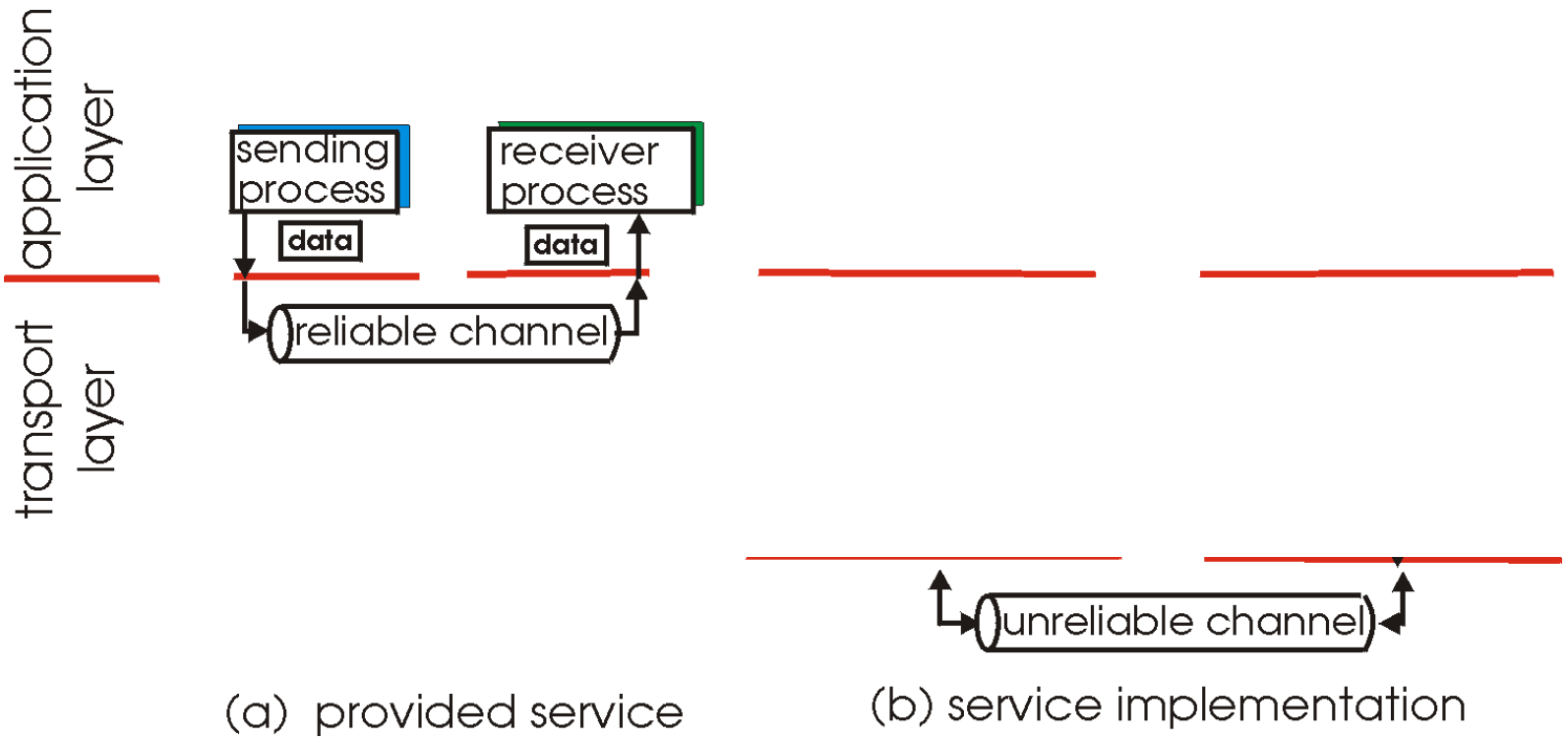


(a) provided service

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

Principles of reliable data transfer

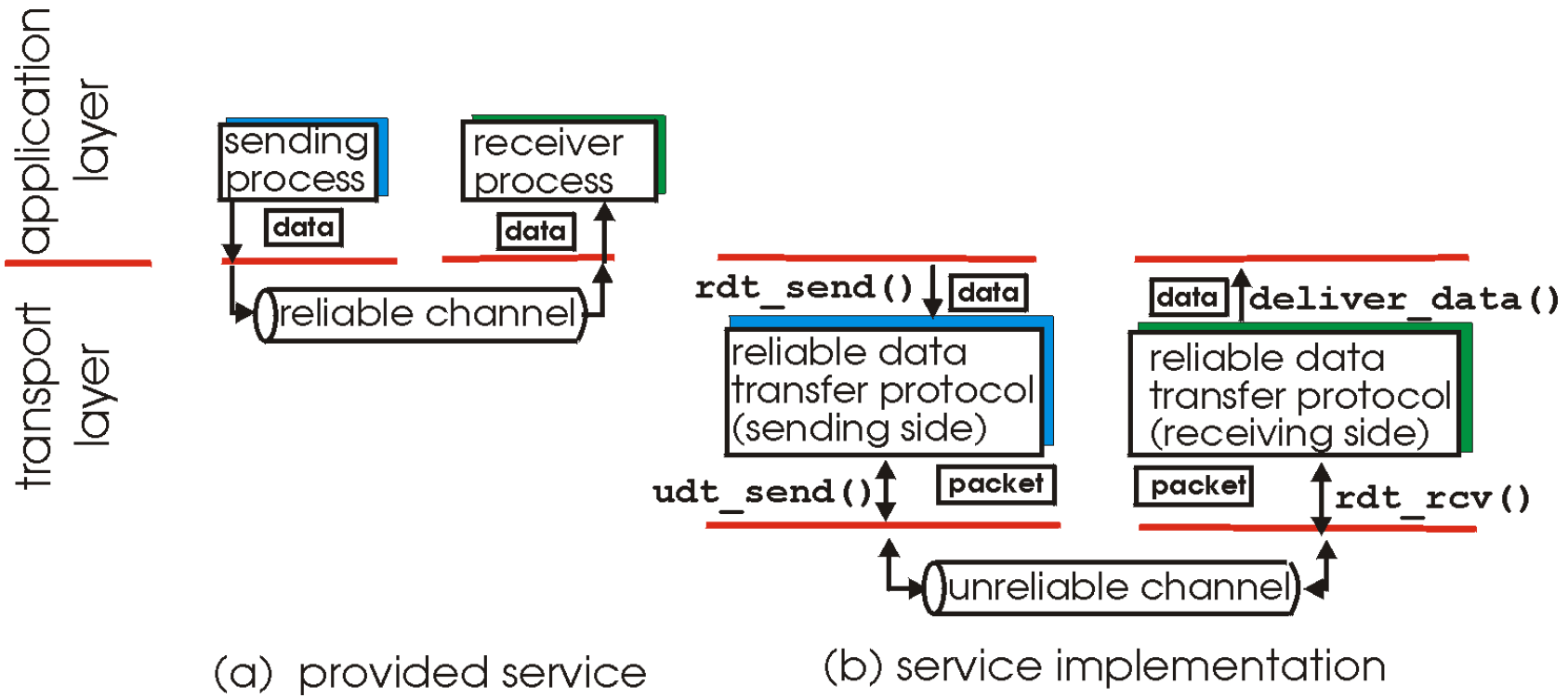
- important in application, transport, link layers
  - top-10 list of important networking topics!



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

Principles of reliable data transfer

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



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

NOTE

- The OpenLearning pages outline state machines for the different versions of protocols that we will discuss in these slides. Please read through them. We will use examples to explain the core ideas in the lectures rather than state machines.

There are a number of self-check questions and problems on RDT protocols on OpenLearning. Please complete them. This is a rather important topic.

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

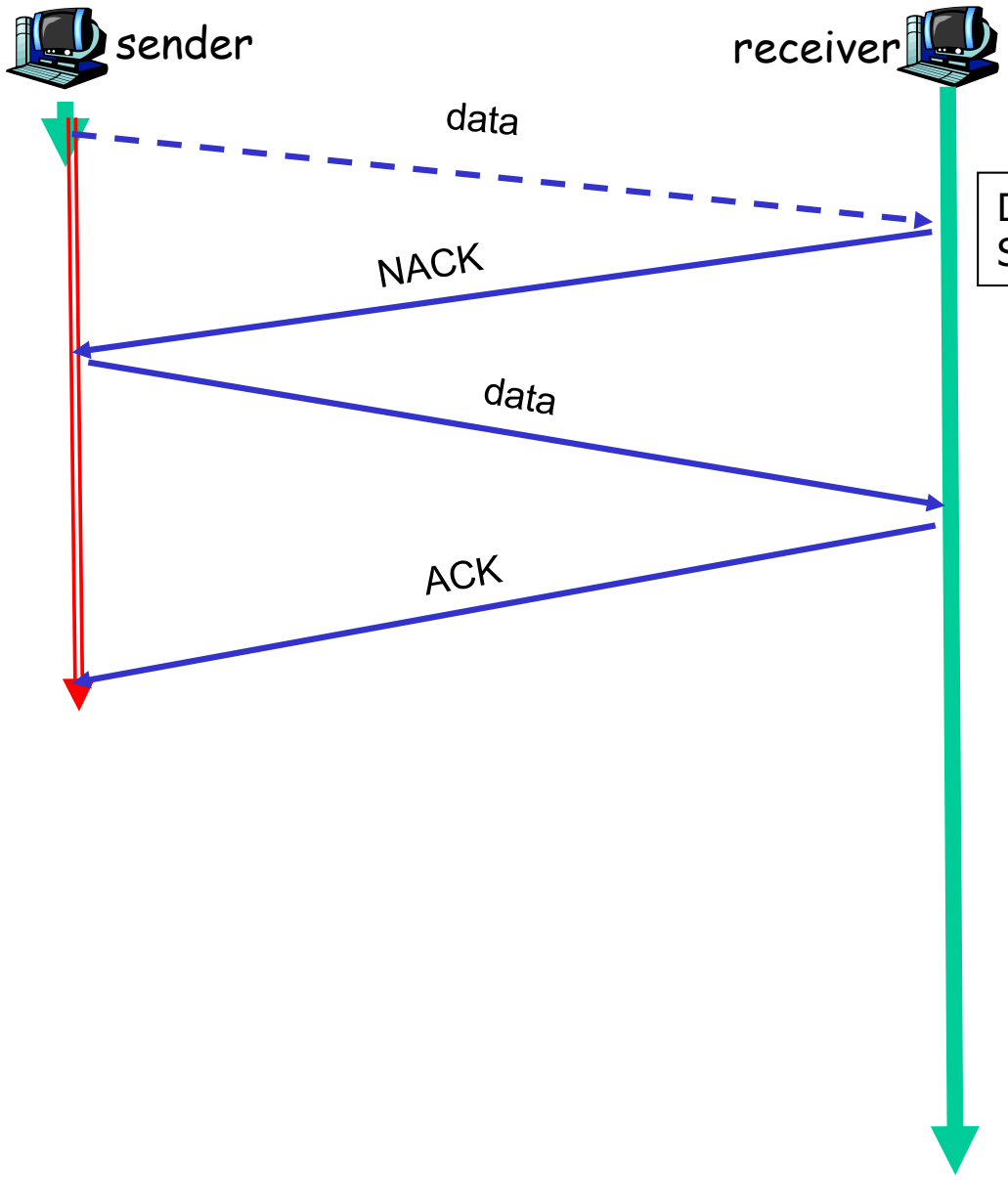
- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors:

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:

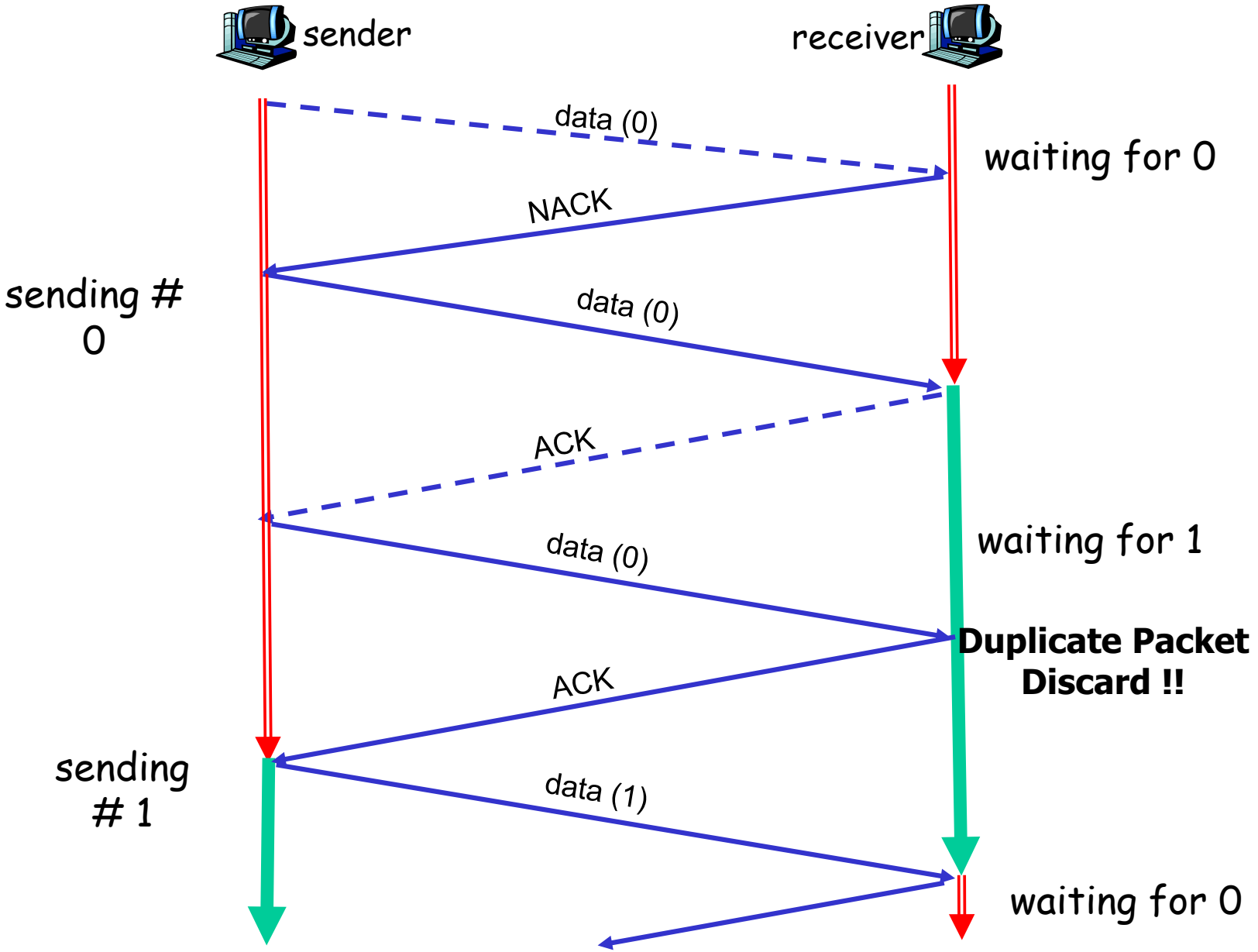
- seq # added to pkt
- two seq. #'s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "expected" pkt should have seq # of 0 or 1

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

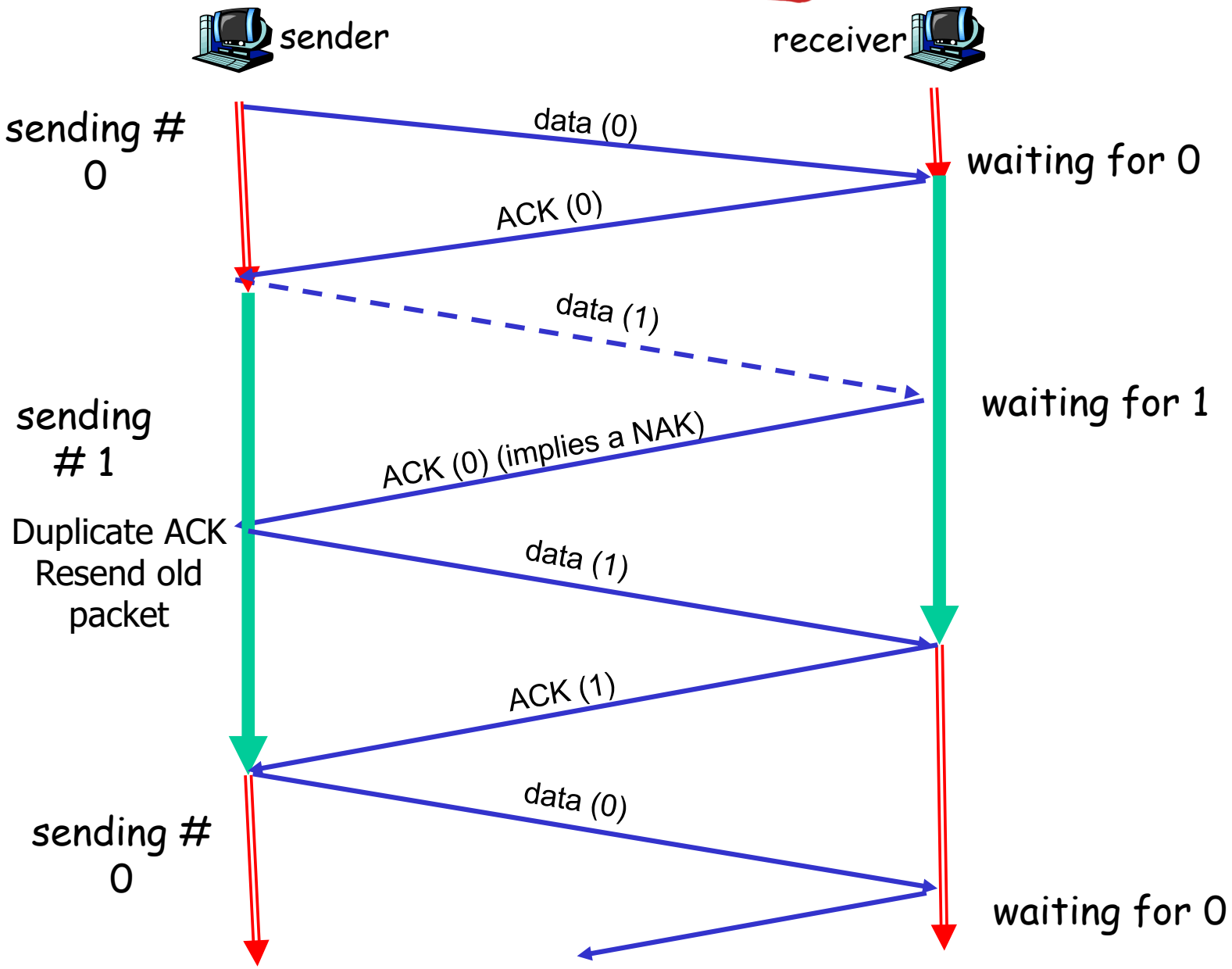


rdt2.2: a NAK-free protocol

- 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:

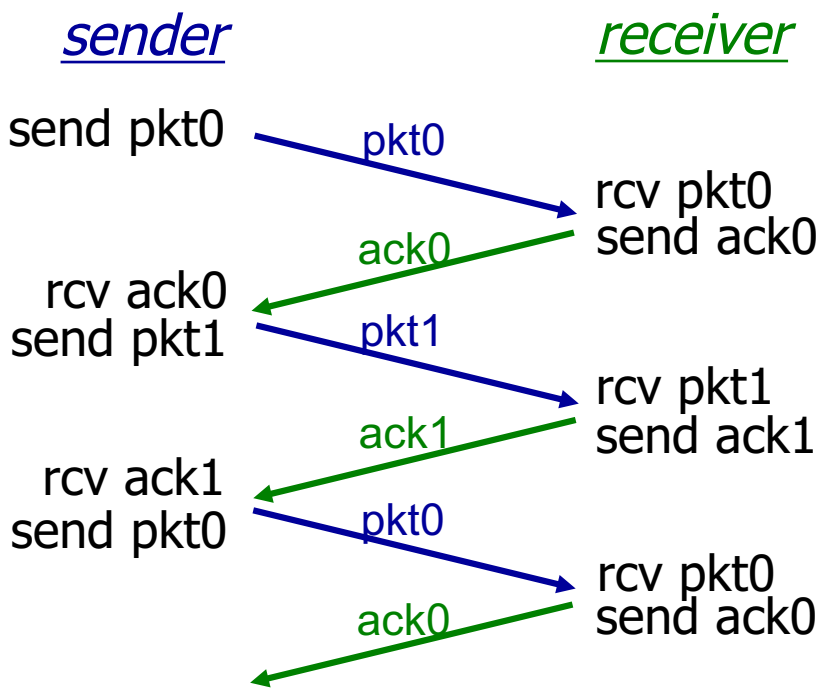
underlying channel can also lose packets (data, ACKs)

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

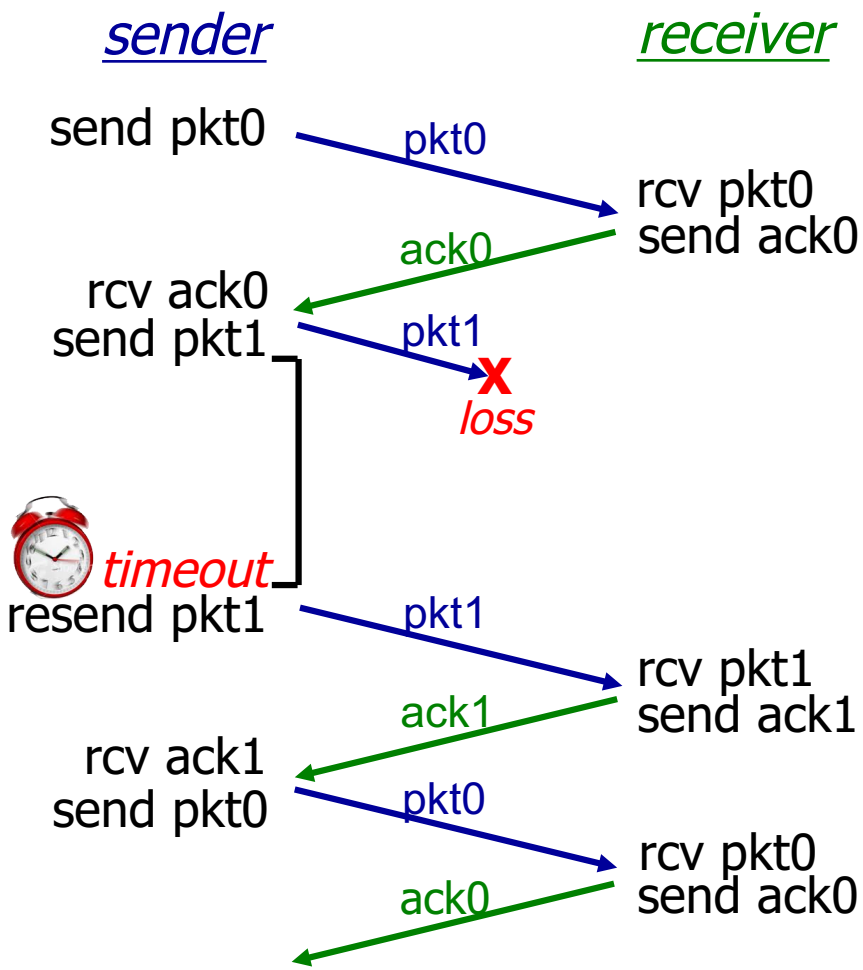
approach: sender waits "reasonable" amount of time for ACK

- 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

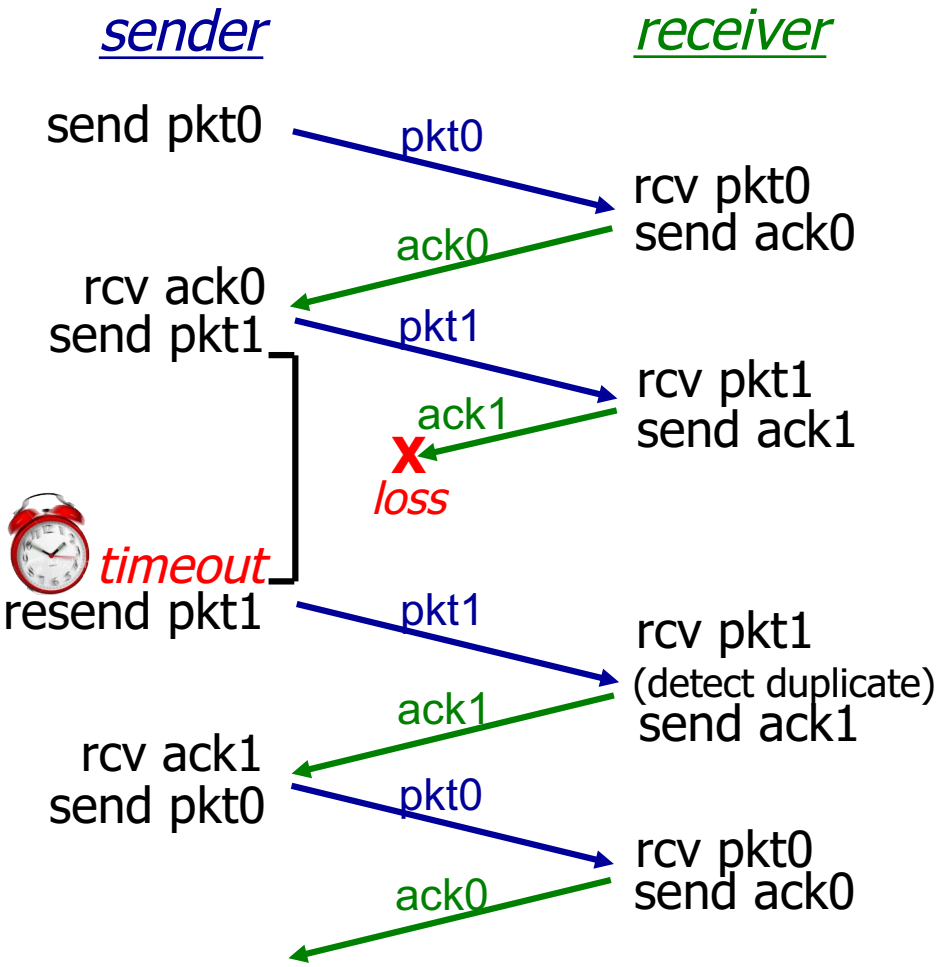


(a) no loss

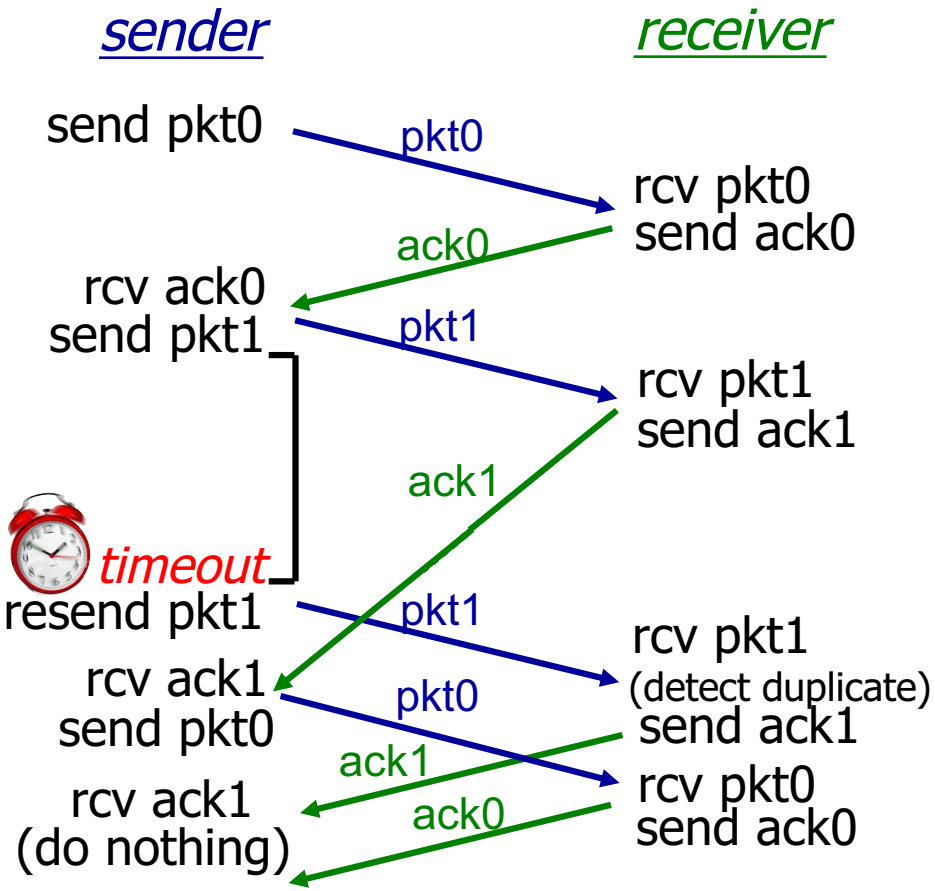


(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

Quiz: RDT 3.0

Suppose that the RTT between the sender and receiver is constant and known to the sender. Would a timer still be necessary in RDT 3.0 assuming that packets can be lost?

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