Computer Networks and Applications

COMP 3331/COMP 9331

Week 4

Transport Layer Part 1

Reading Guide:
Chapter 3, Sections 3.1 – 3.5

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

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

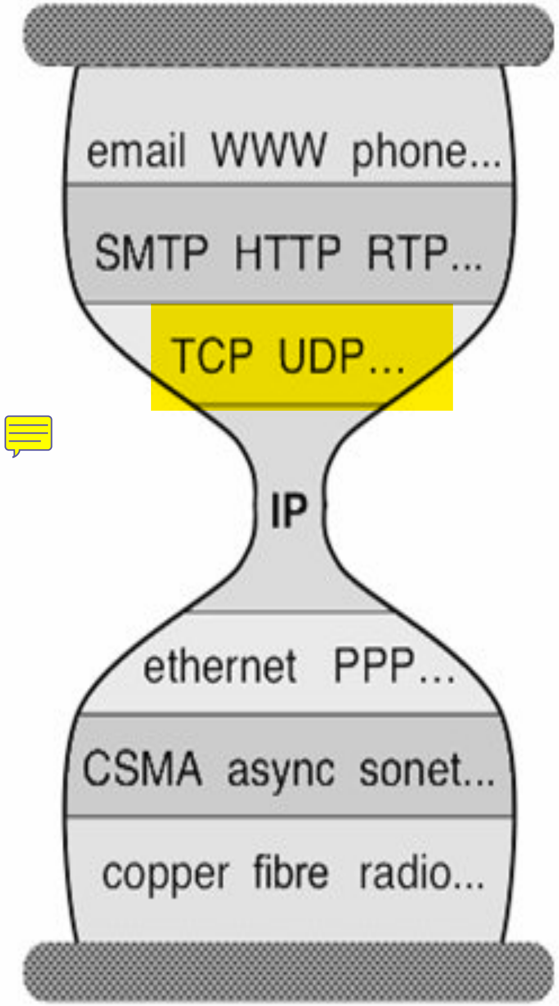
3.5 connection-oriented transport: TCP
- segment structure
- reliable data transfer
- flow control
- connection management

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)

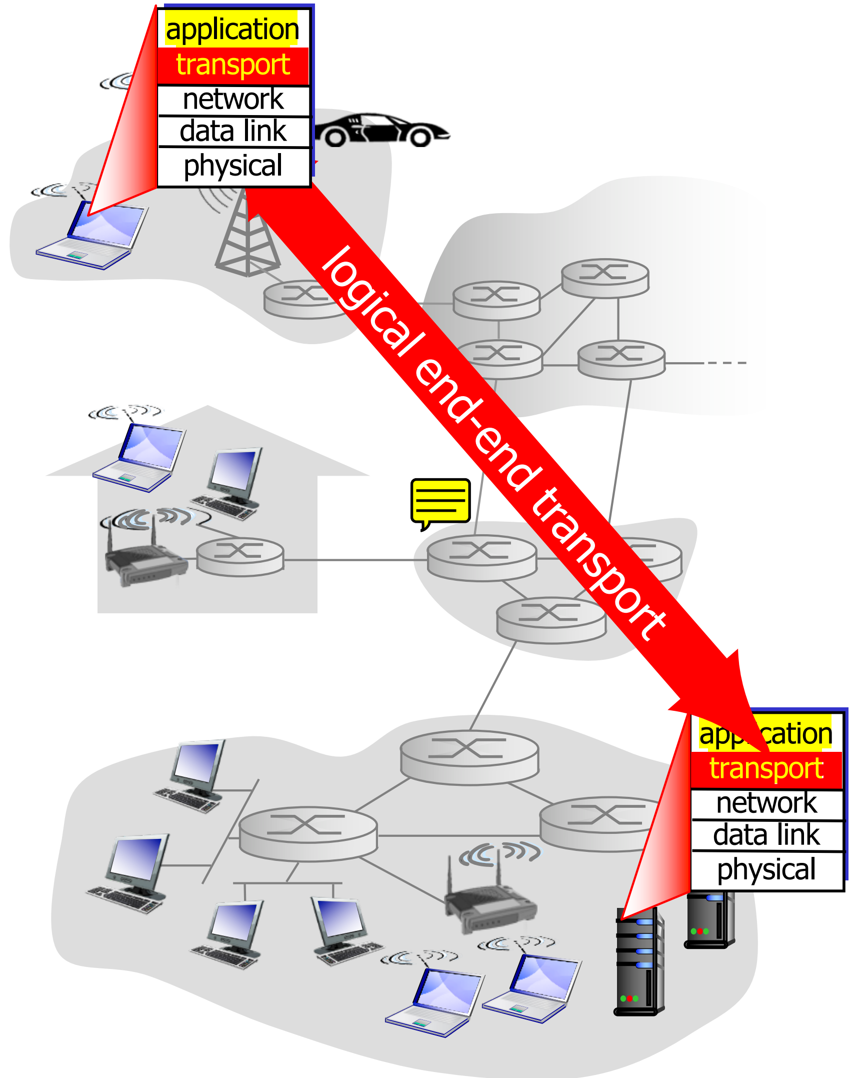
- What it does: finds paths through network
  - Routing from one end host to another

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

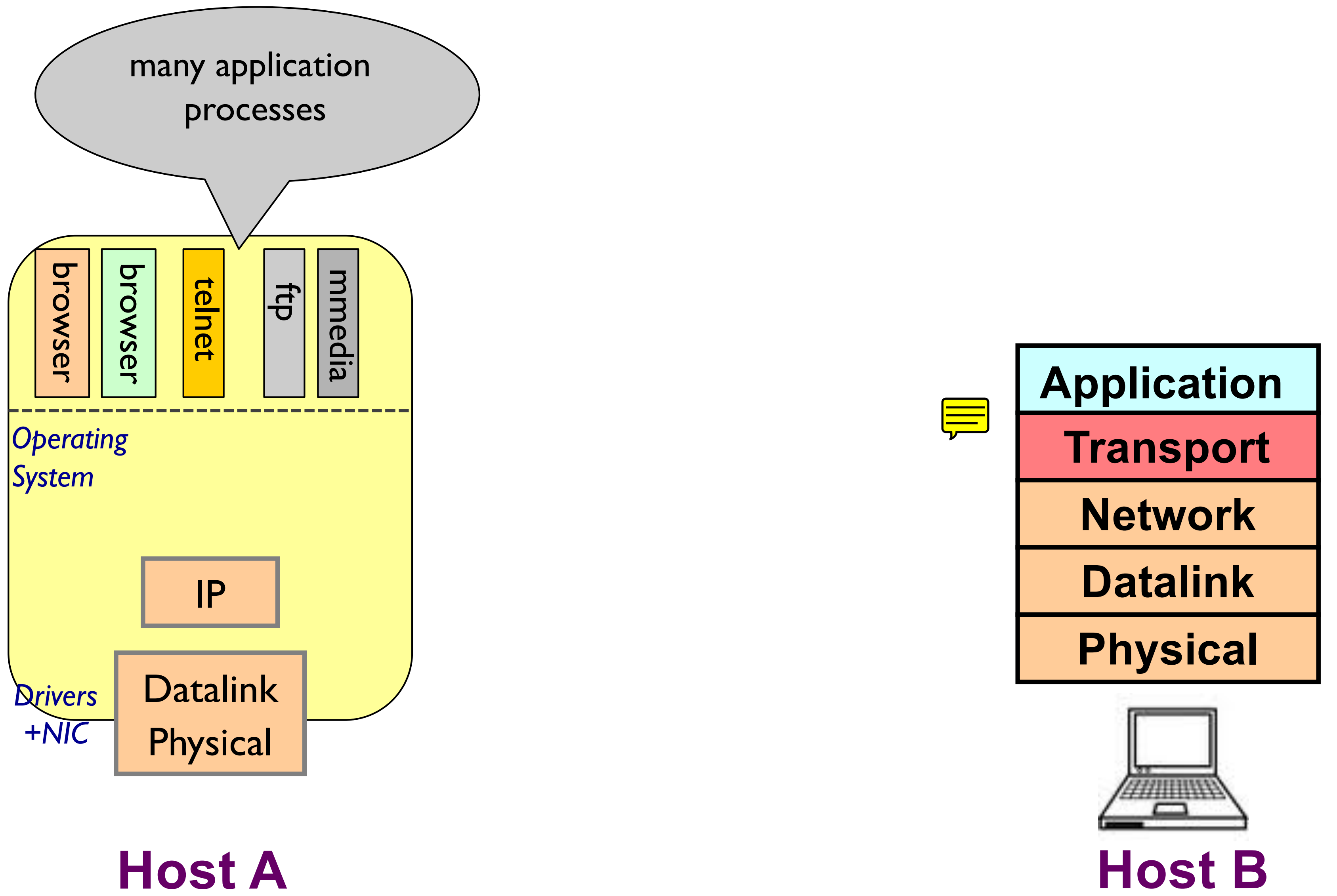
- For now, think of the network layer as giving us an “API” with one function: *sendtohost(data, host)*
  - Promise: the data will go to that (usually!!)

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



Why a transport layer?



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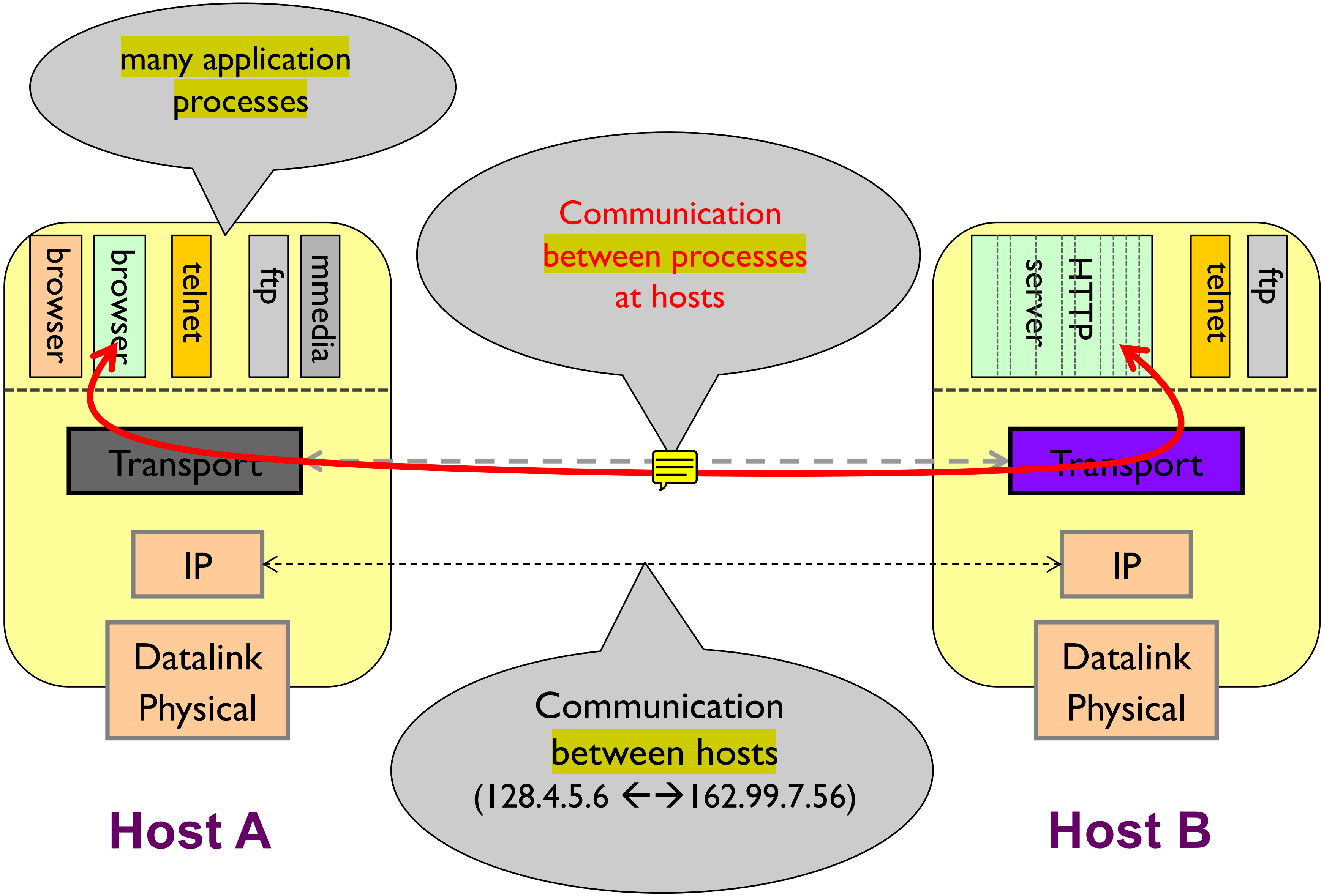

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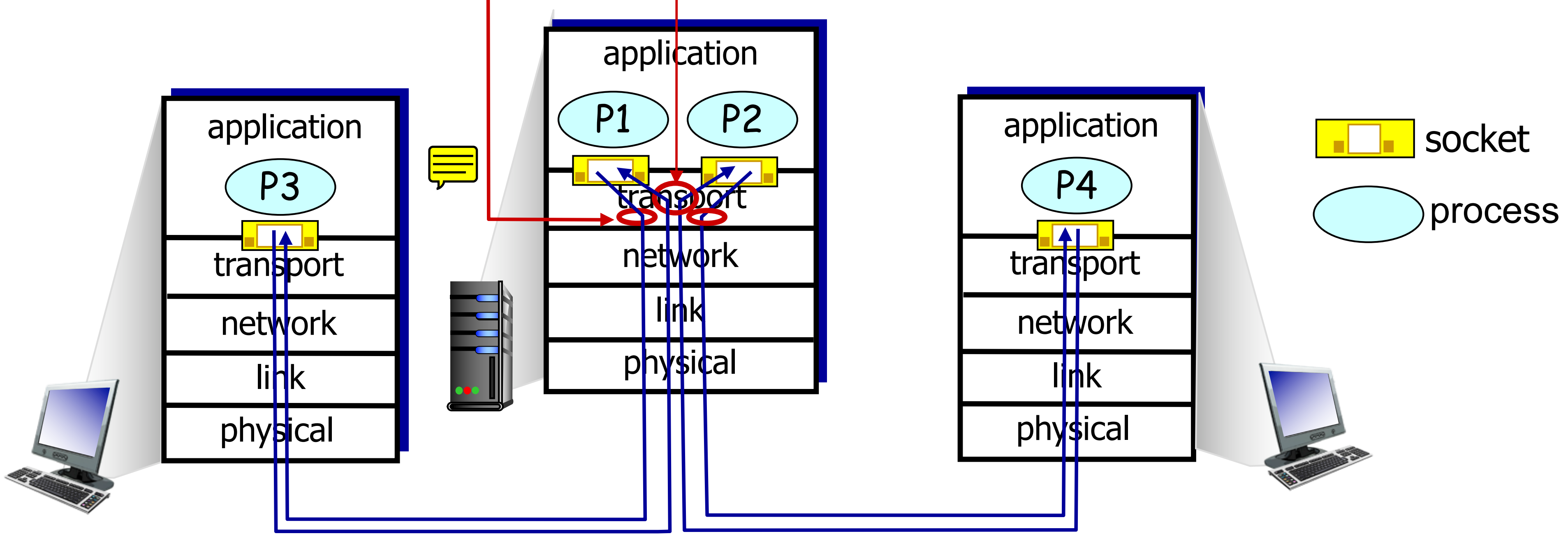
Multiplexing/demultiplexing

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 #:

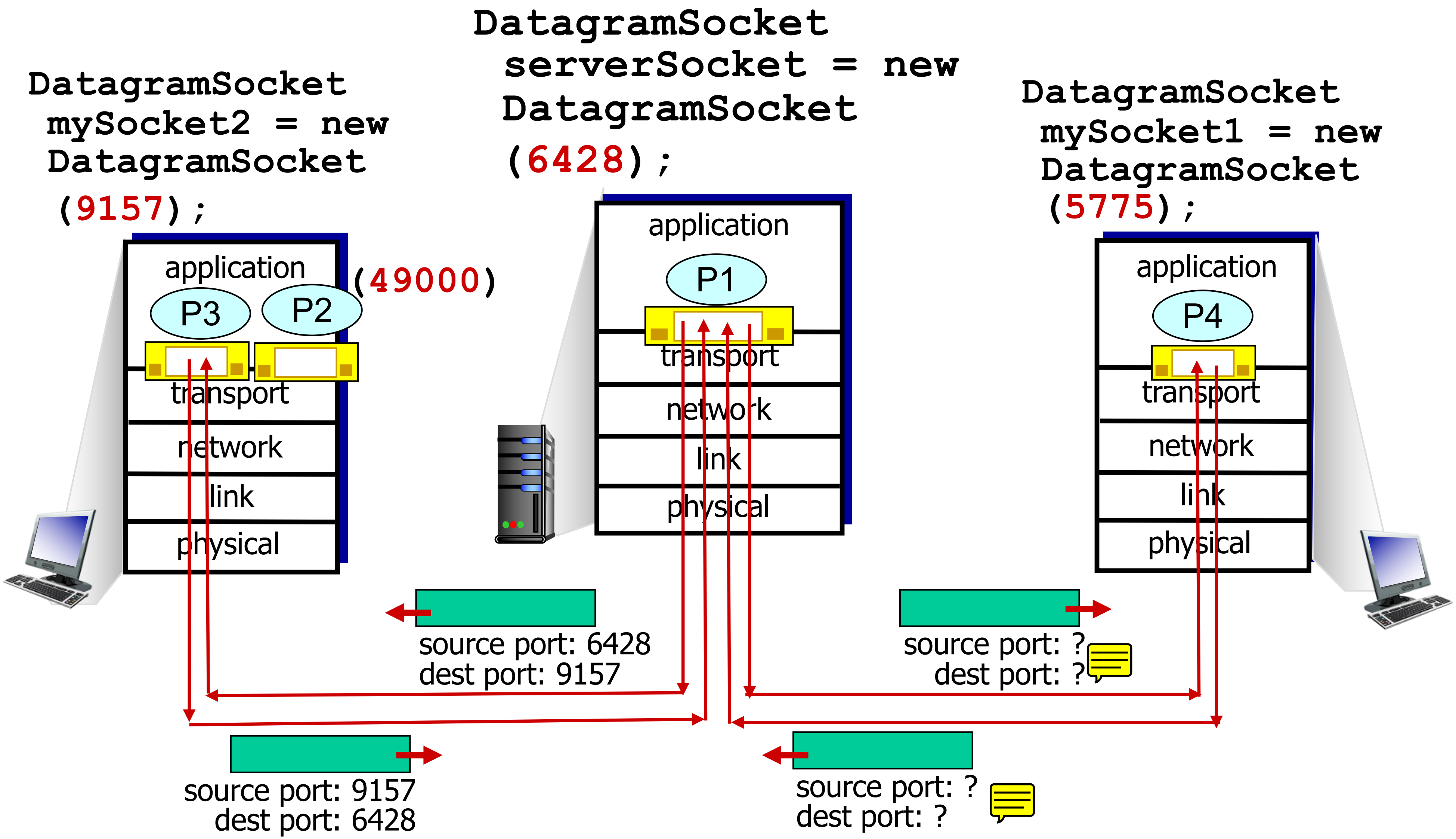
```
DatagramSocket mySocket1
= new DatagramSocket(12534);
```

- *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

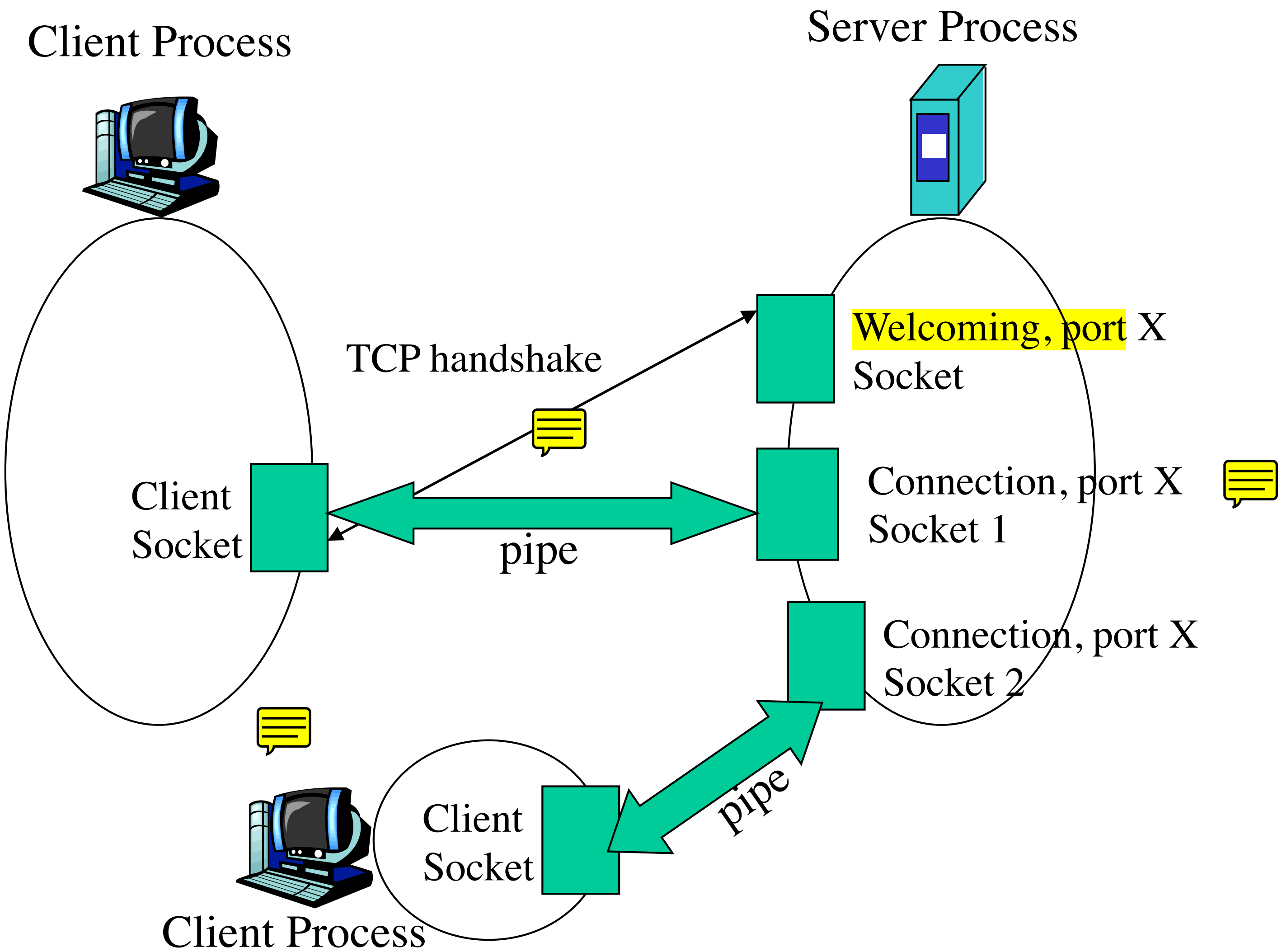
Connectionless demux: example



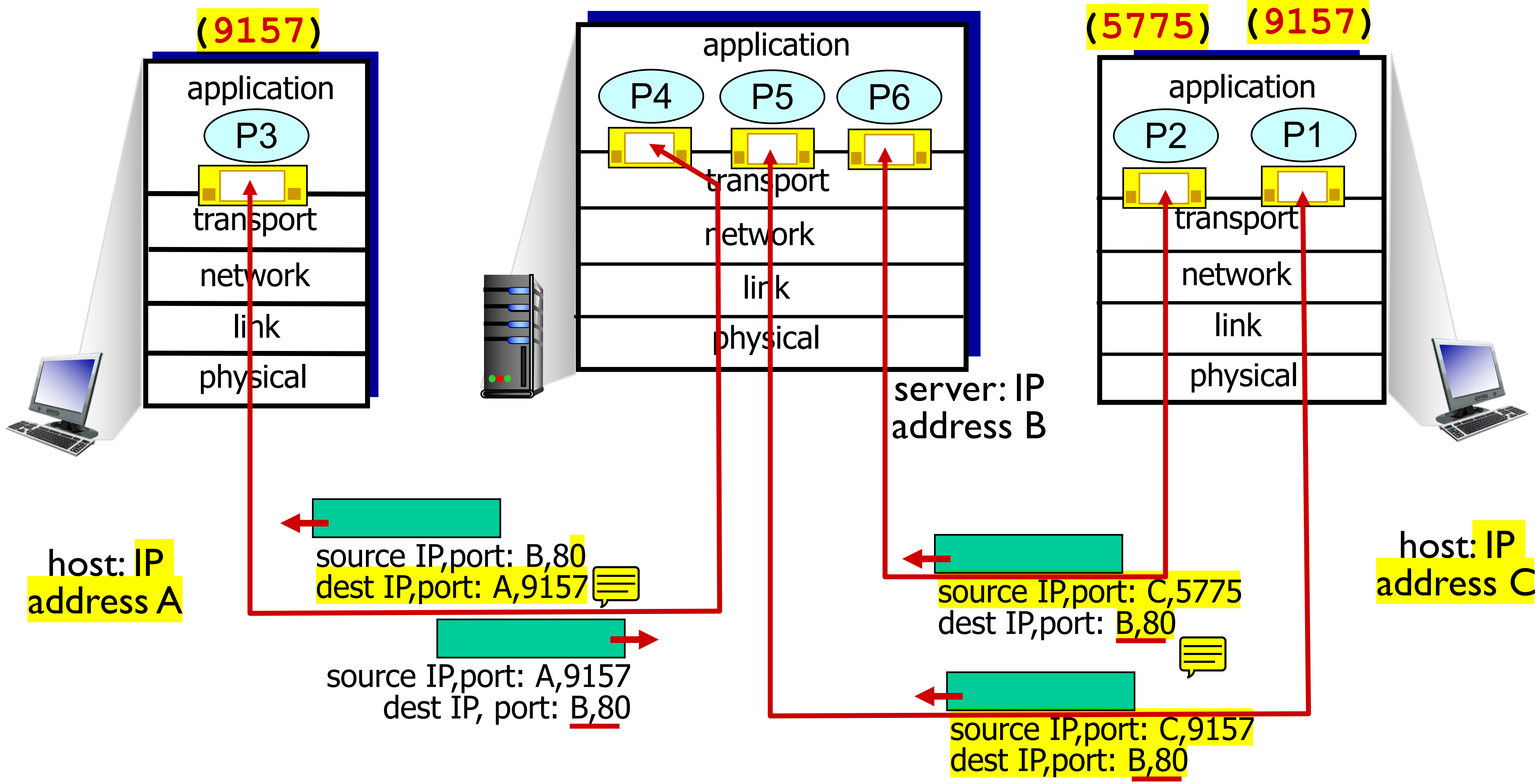
Connection-oriented demux

- 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



Connection-oriented demux: example



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

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 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

Quiz: UDP Sockets

- Suppose we use UDP instead of TCP for designing a web server where all requests and responses fit in a single UDP segment. Suppose 100 clients are simultaneously communicating with this web server. How many sockets are respectively at the server and each client?

a) 1, 1
b) 2, 1
c) 200, 2
d) 100, 1
e) 101, 1

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Quiz: TCP Sockets

- Suppose 100 clients are simultaneously communicating with a traditional HTTP/TCP web server. How many sockets are respectively at the server and each client?

  a) 1, 1
  b) 2, 1
  c) 200, 2
  d) 100, 1
  e) 101, 1

Quiz: TCP Sockets

- Suppose 100 clients are simultaneously communicating with a traditional HTTP/TCP web server. Do all of the sockets at the server have the same server-side port number?

a) Yes

b) No

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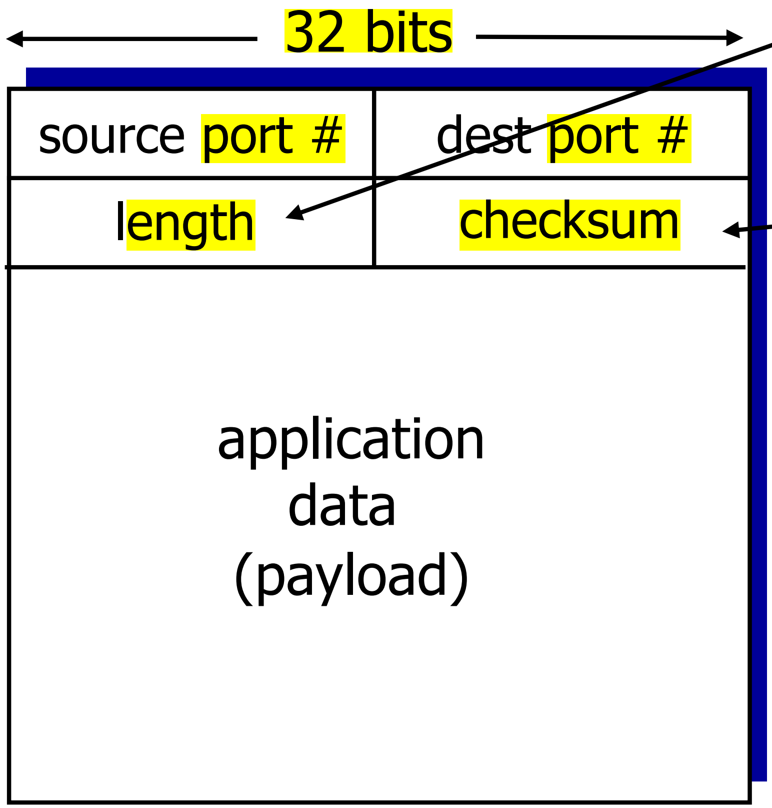
UDP: User Datagram Protocol [RFC 768]

- "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

length, in bytes of UDP segment, including header

2 bytes Optional Checksum

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: Checksum

- *Checksum is the16-bit* one's complement of the one's complement sum of a pseudo header of information from the IP header, the UDP header, and the data, padded with zero octets at the end (if necessary) to make a multiple of two octets.
- Checksum header, data and pre-pended IP pseudo-header
- But the header contains the checksum itself?
- What's IP pseudo-header?

bits	0 – 7	8 – 15	16 – 23	24 – 31
0	Source address			
32	Destination address			
64	Zeros	Protocol	UDP length	
96	Source Port		Destination Port	
128	Length		Checksum	
160+	Data			

IP pseduo-header

UDP header

UDP payload

Internet checksum: example

10011001 00010010	→	153.18
00001000 01101001	→	8.105
10101011 00000010	→	171.2
00001110 00001010	→	14.10
00000000 00010001	→	0 and 17
00000000 00001111	→	15
00000100 00111111	→	1087
00000000 00001101	→	13
00000000 00001111	→	15
00000000 00000000	→	0 (checksum)
01010100 01000101	→	T and E
01010011 01010100	→	S and T
01001001 01001110	→	I and N
01000111 00000000	→	G and 0 (padding)
10010110 11101011	→	Sum
01101001 00010100	→	Checksum

UDP Applications

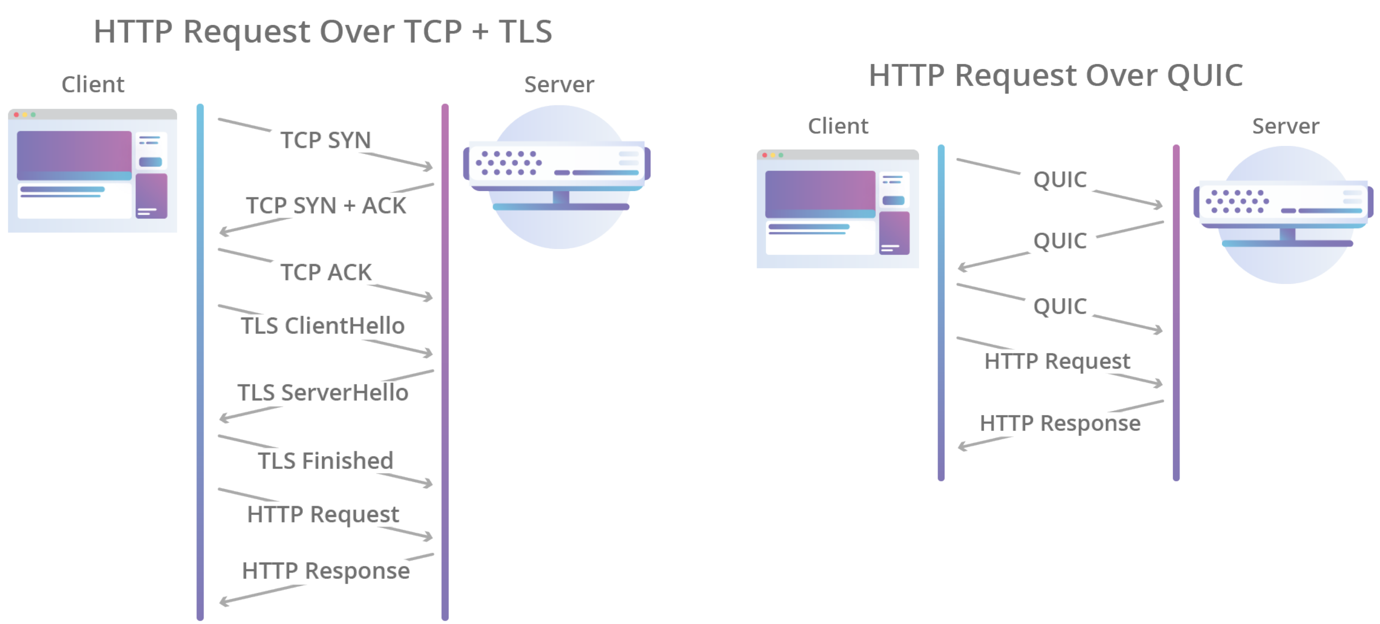
- Latency sensitive/time critical
  - Quick request/response (DNS, DHCP)
  - Network management (SNMP)
  - Routing updates (RIP)
  - Voice/video chat
  - Gaming (especially FPS)

- Error correction unnecessary (periodic messages)

THIS SLIDE IS NOT ASSESSABLE!!!

QUIC: Quick UDP Internet Connections

- Core idea: HTTP/2 over UDP
  - Faster connection establishment
  - Overcomes HoL blocking due to lost packets
  - Improved congestion control
  - Forward error correction
  - Connection migration



https://www.chromium.org/quic

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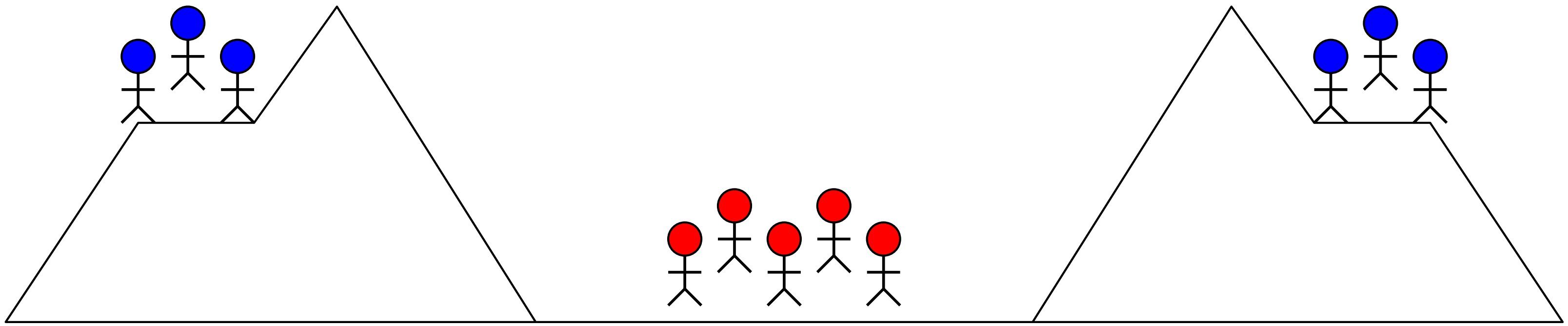
Reliable Transport

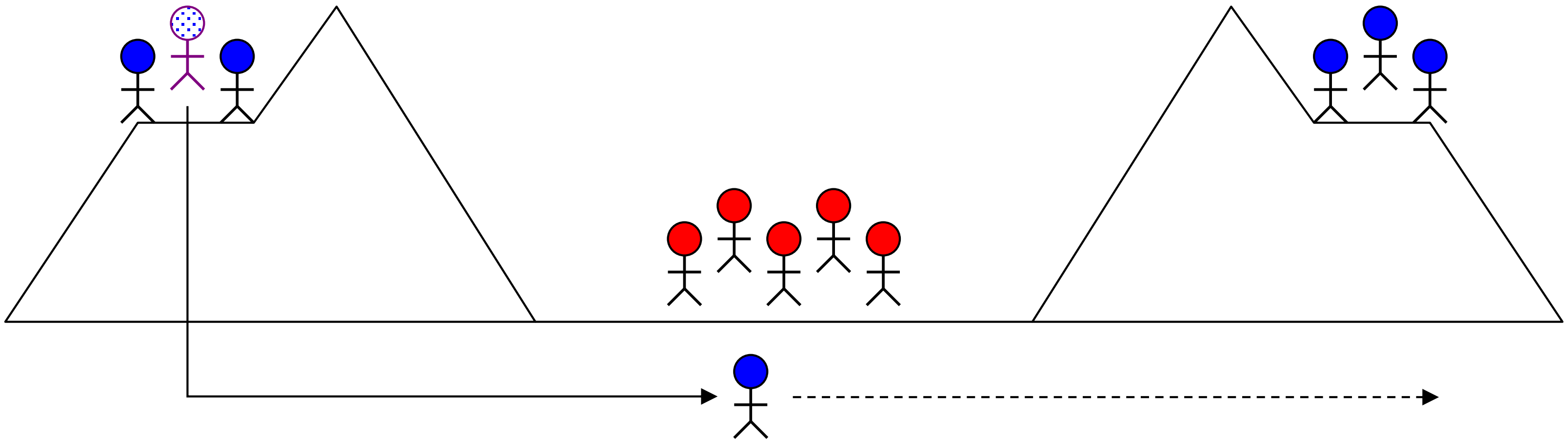
- In a perfect world, reliable transport is easy
- All the bad things best-effort can do
  - a packet is corrupted (bit errors) checksum usually prevents this
  - a packet is lost (*why?*) gets dropped because router queue is full
  - a packet is delayed (*why?*) because of buffering, queueing
  - packets are reordered (*why?*) because 2 packets can take different path, first may be longer
  - a packet is duplicated (*why?*) thinks packet is lost, not actually lost, resends the packet, then both arrive

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

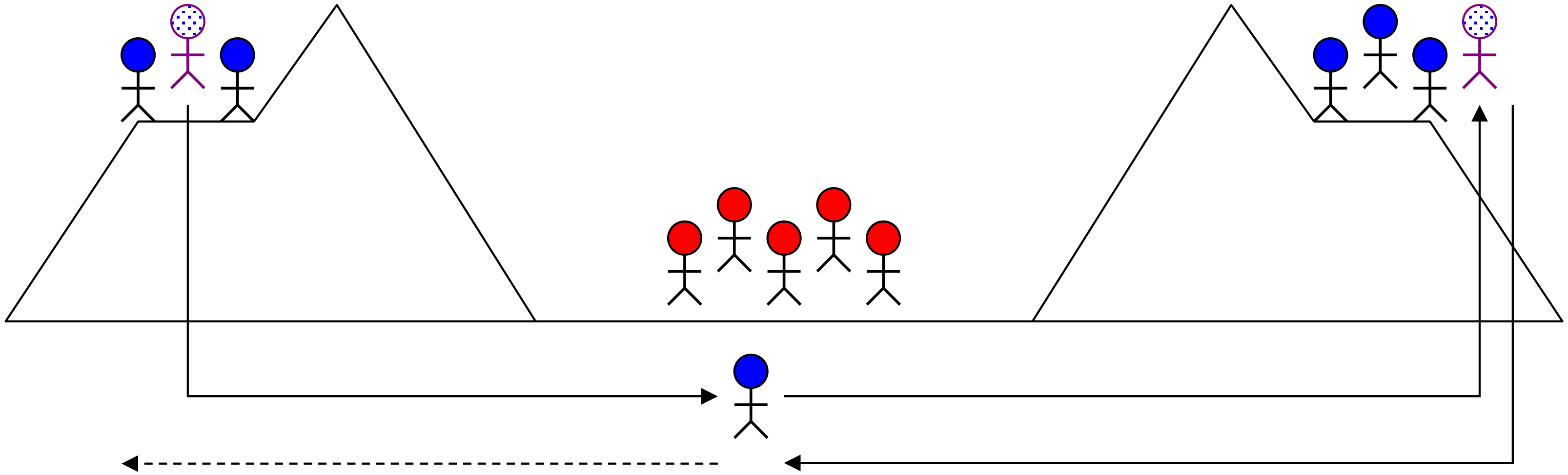


1. either he makes it and you both attack at dawn (win)
2. or he doesn't make it and only you attack at dawn and die

so you NEED TO KNOW WHETHER HE MADE IT

- How to coordinate?
  - Send messenger: "Attack at dawn"
  - What if messenger doesn't make it?

The Two Generals Problem



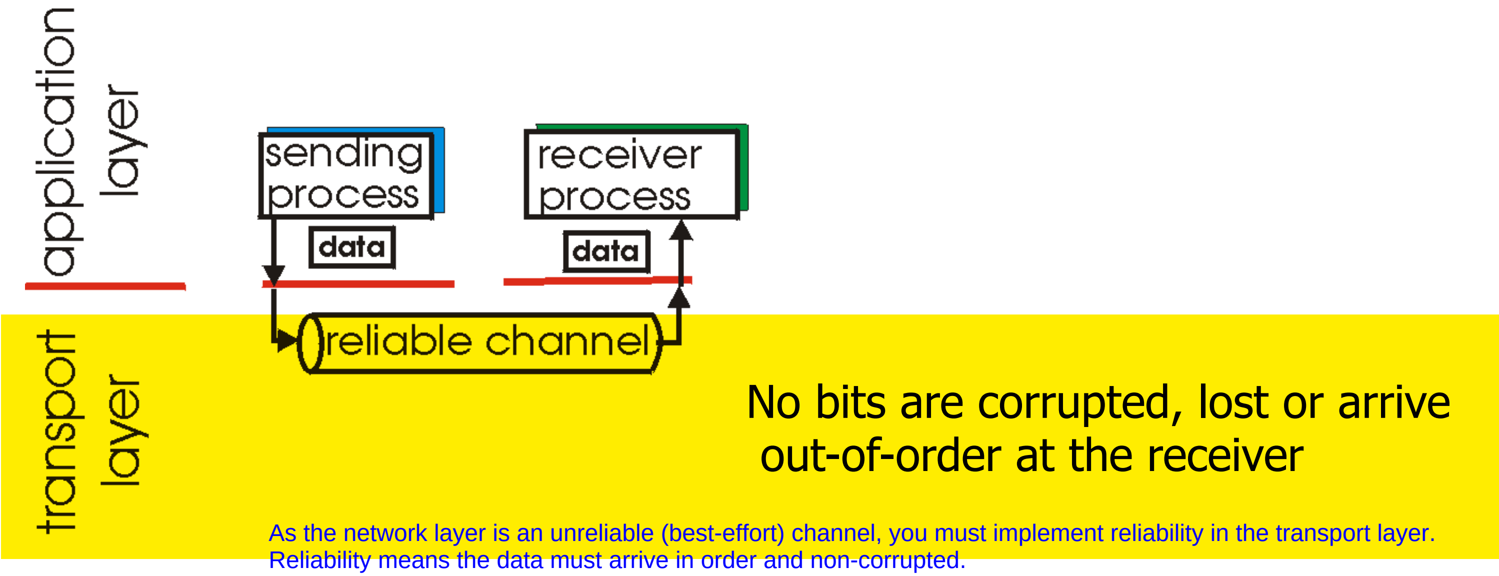
now assuming your guy made it, you need to know about it;
1. they send a guy back and he makes it (you both attack at dawn) (win)
2. they send a guy back but he doesn't make it, they attack and die (lose)

so now they are in the same situation as you were above, i.e. they need to know that their guy made it to you (and we just tried to prove this <- here and showed it doesn't work

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

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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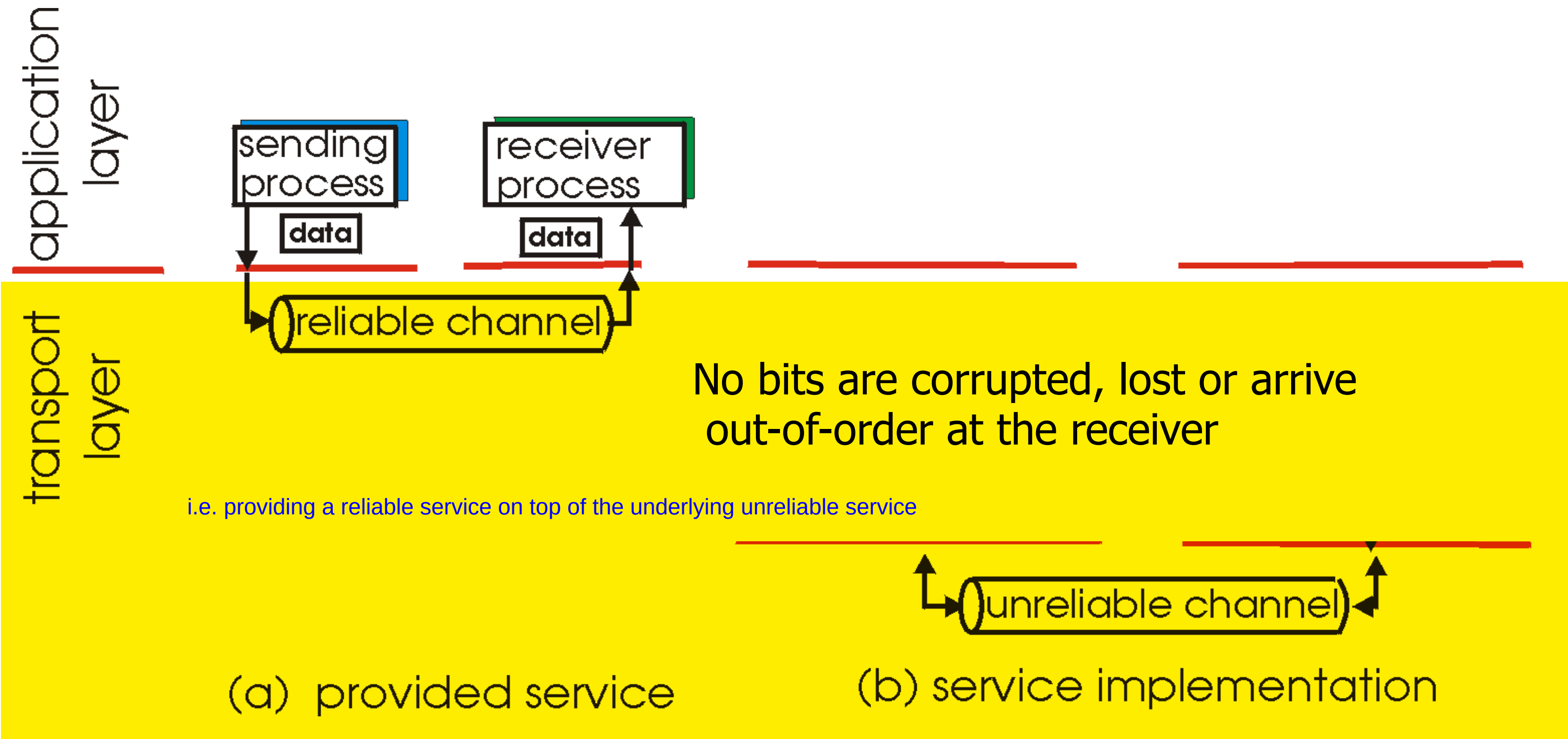

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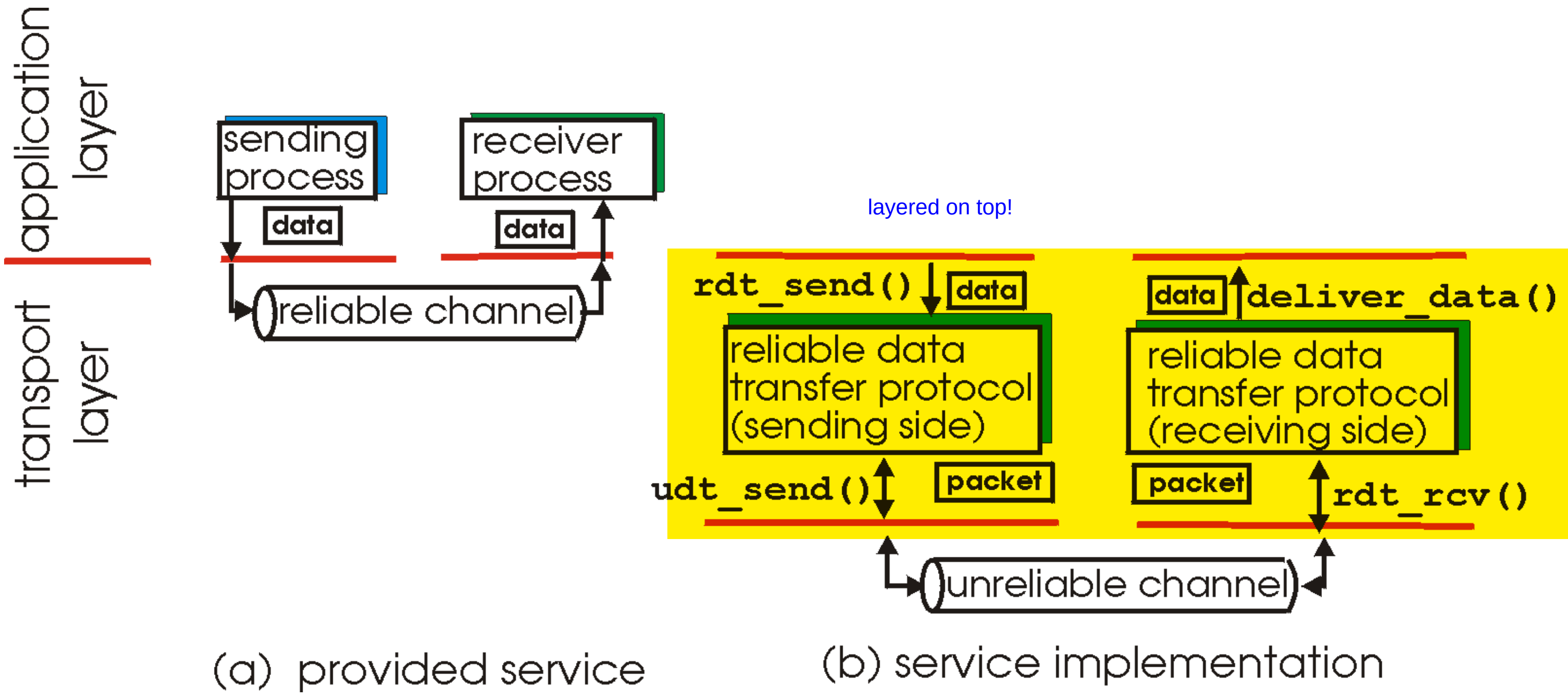


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable data transfer: getting started

We'll:

- Incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- Consider only unidirectional data transfer
  - but control info will flow on both directions!

i.e. need ACKS coming from client to sender

- Channel will not re-order packets

if it is reliable, and you are doing stop and wait, then there is no need to re-order packets!

stop and wait

sender sends one packet, then waits for receiver response

rdt1.0: reliable transfer over a reliable channel

- Underlying channel perfectly reliable

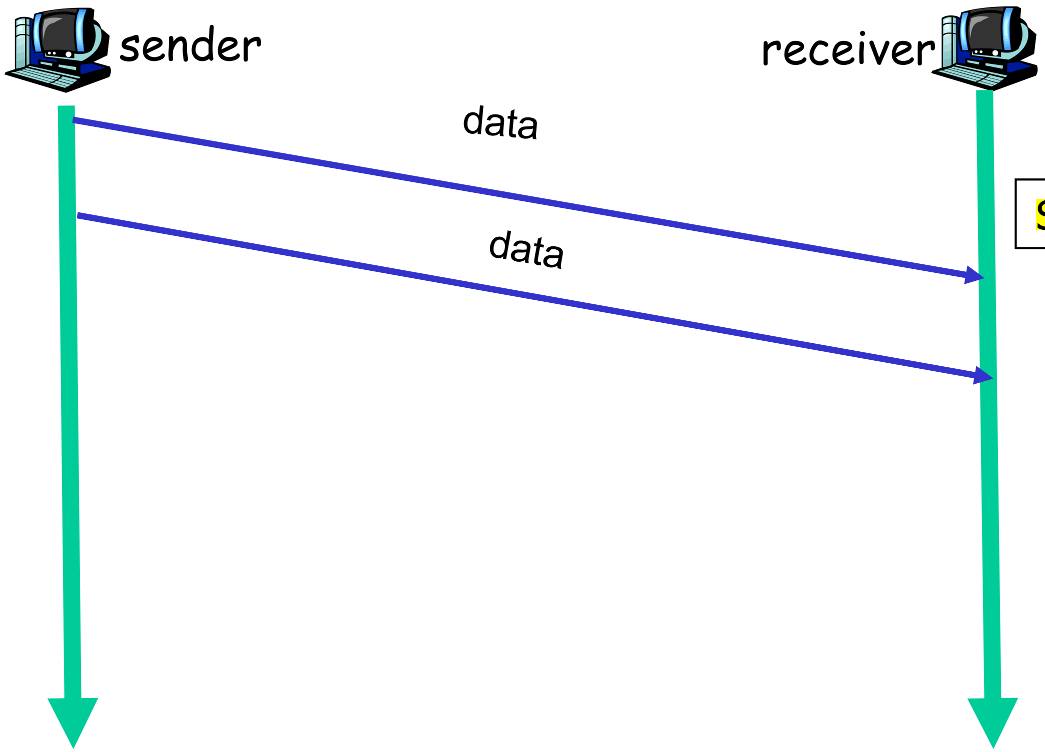
i.e IF network layer was 100% reliable

  - no bit errors
  - no loss of packets

and stop and wait so no possibility of packet re-ordering

- Transport layer does nothing !

Global Picture of rdt1.0



Solid line: error-free transmission

rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - checksum to detect bit errors

still guaranteed to get the packet there though, even if erroneous data

- *the* question: how to recover from errors:

so we've already talked about how to detect (most) errors, but how to recover??

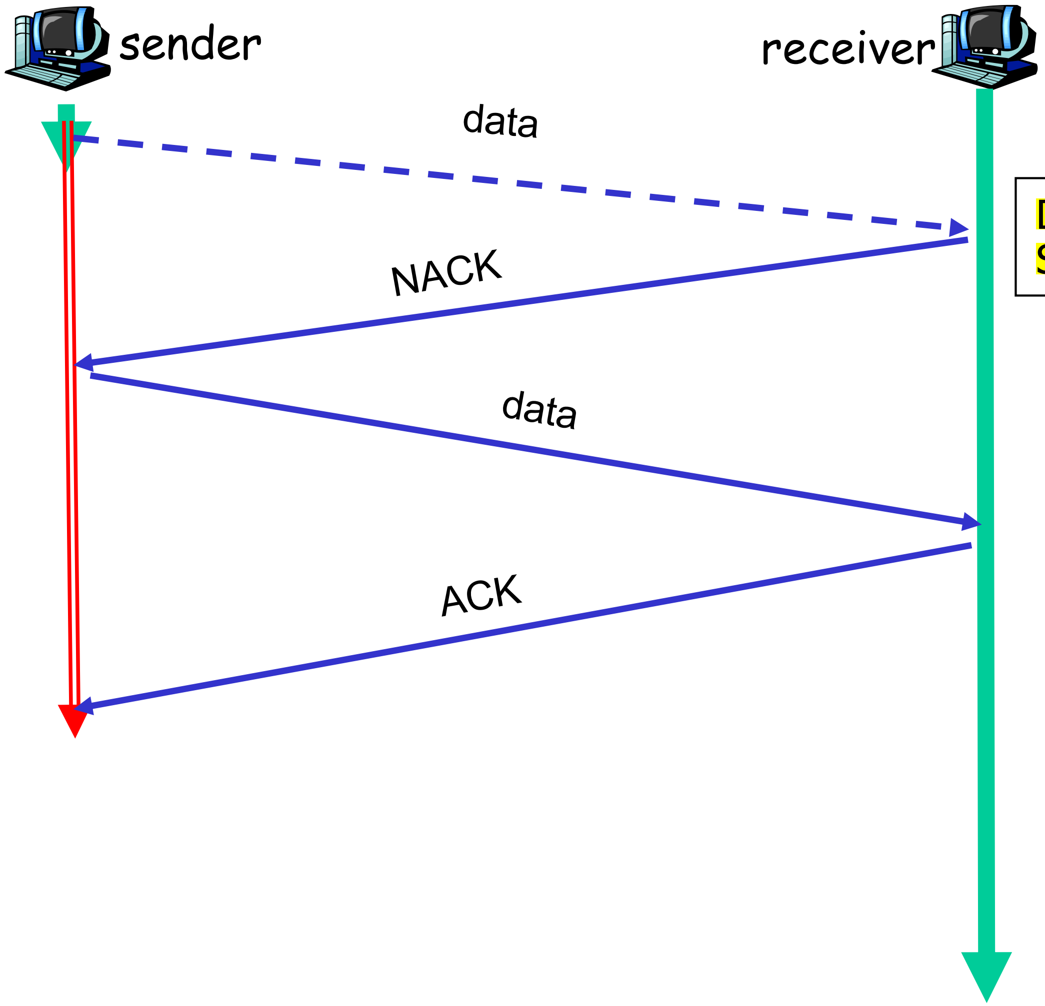
How do humans recover from "errors" during conversation?

we repeat, saying what we were supposed to say

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 via checksum
  - feedback: control msgs (ACK,NAK) from receiver to sender to recover from an error (i.e. receiver tests checksum and it is wrong and so sends back NAK to re-get)
  - retransmission sender will re-send when it gets a NAK back

Global Picture of rdt2.0



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

rdt2.0 has a fatal flaw!

we know it gets there because that is guaranteed in this example, but it could be erroneous.
Sender doesn't know what it is "ahh is this an ACK or a NAK or something else?"

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:

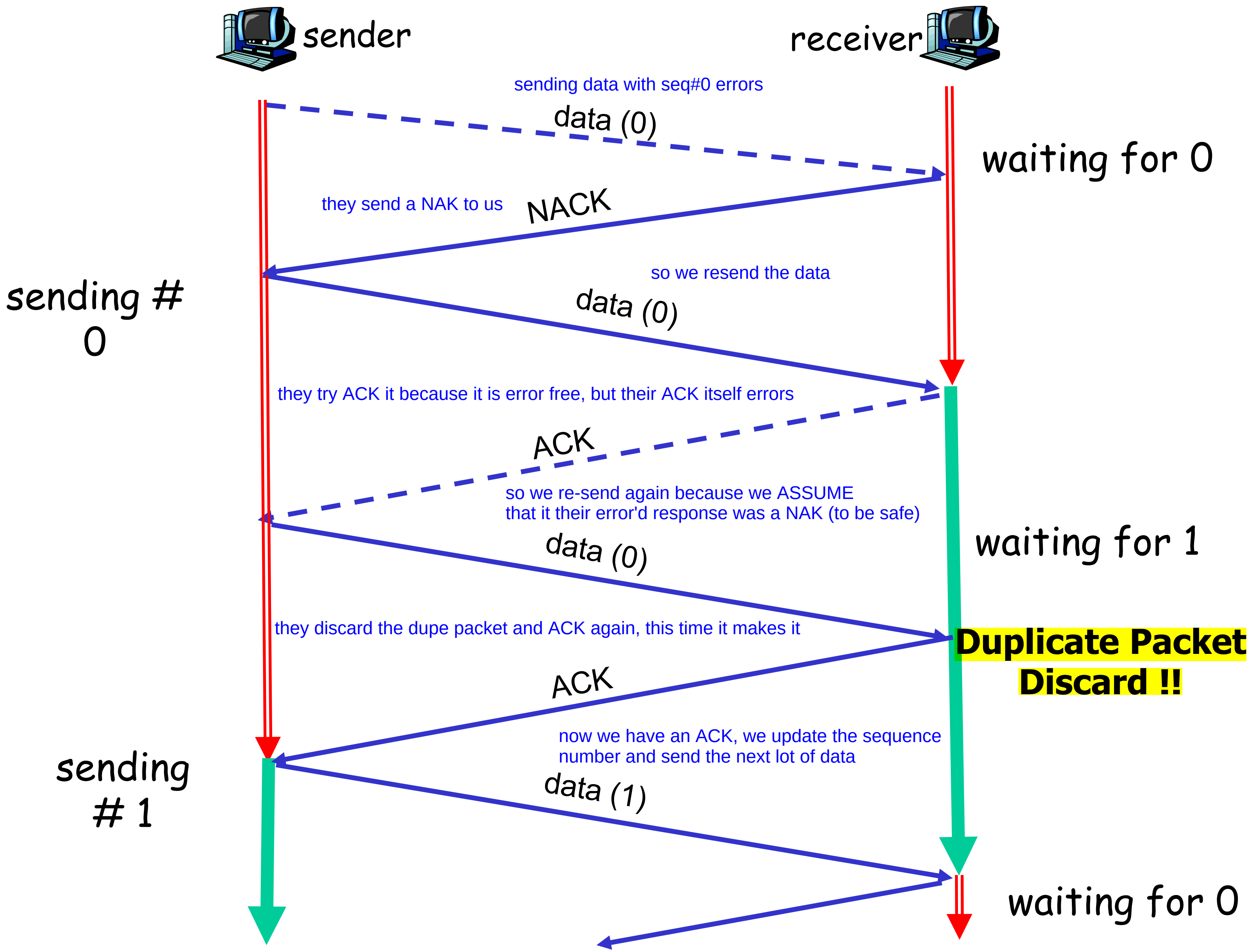
- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

it can corrupt just like these other packets can
this is the reason we may be getting sent duplicates in the first place!

New Measures: Sequence Numbers, Checksum for ACK/NACK, Duplicate detection

Another Look at rdt2.1

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

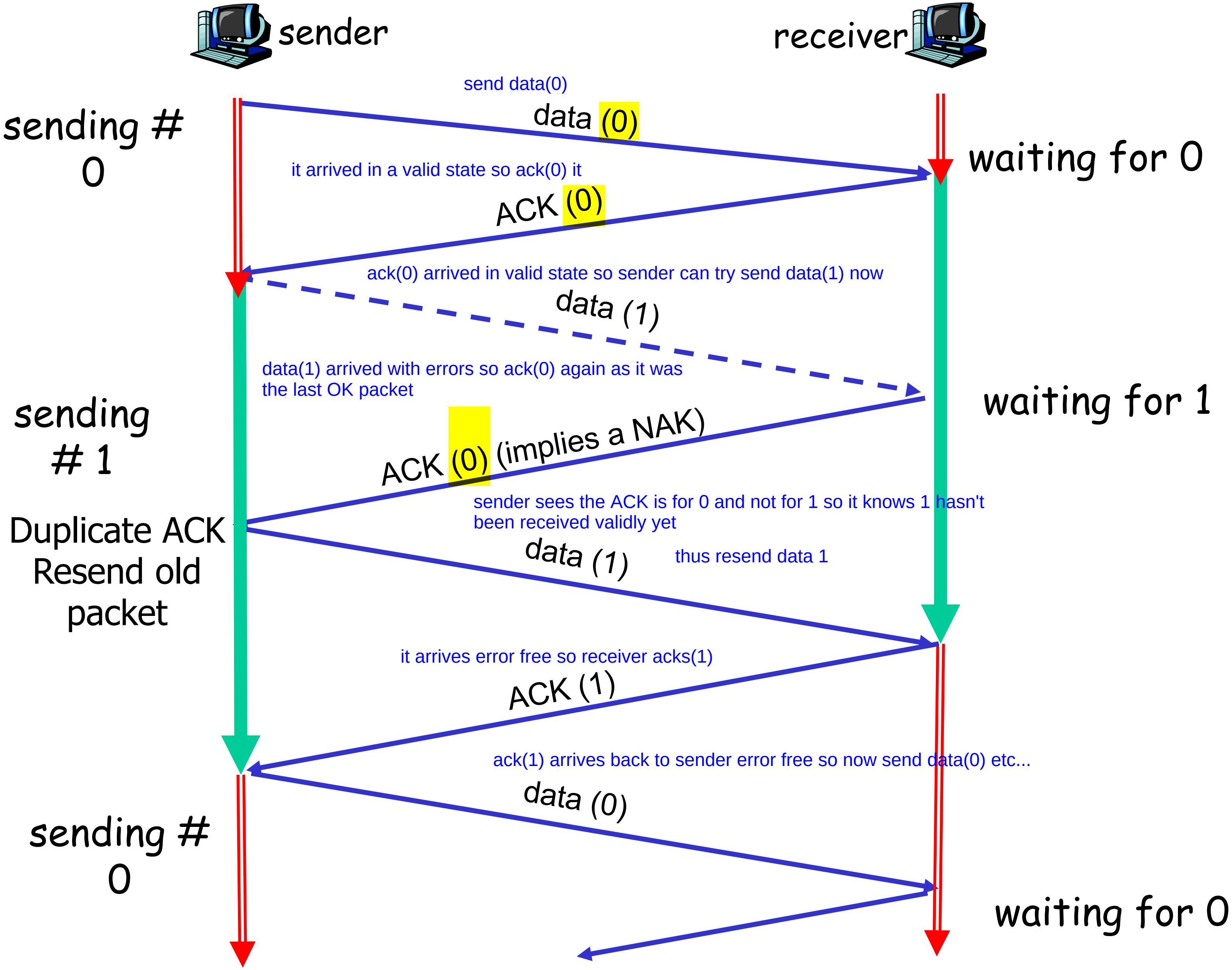


rdt2.2: a NAK-free protocol

- same functionality as rdt2.1, using ACKs only
- instead of NAK, receiver sends ACK for last pkt received OK   ACK(last okay packet seq#)
  - 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

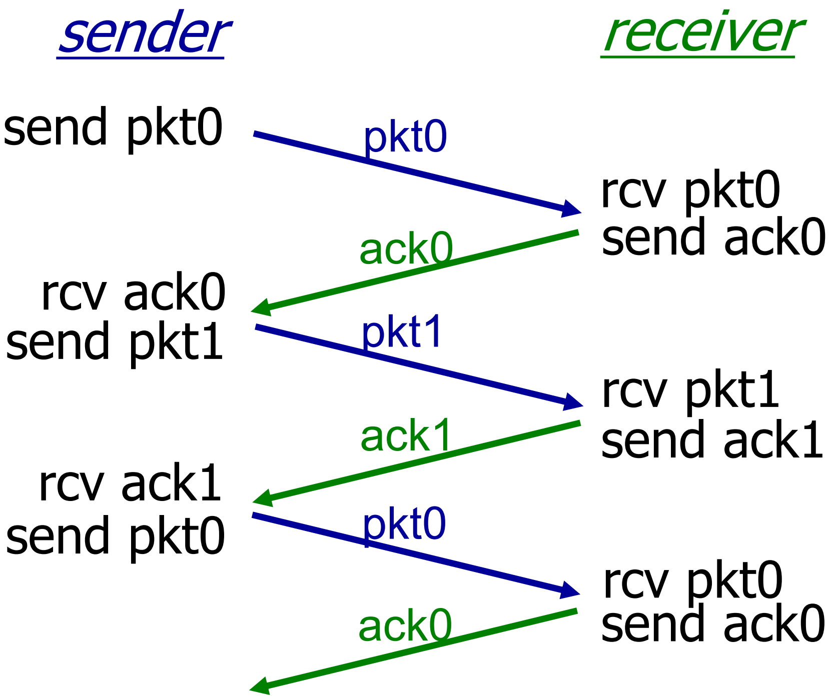
new assumption: underlying channel can also loose packets (data, ACKs)

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

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

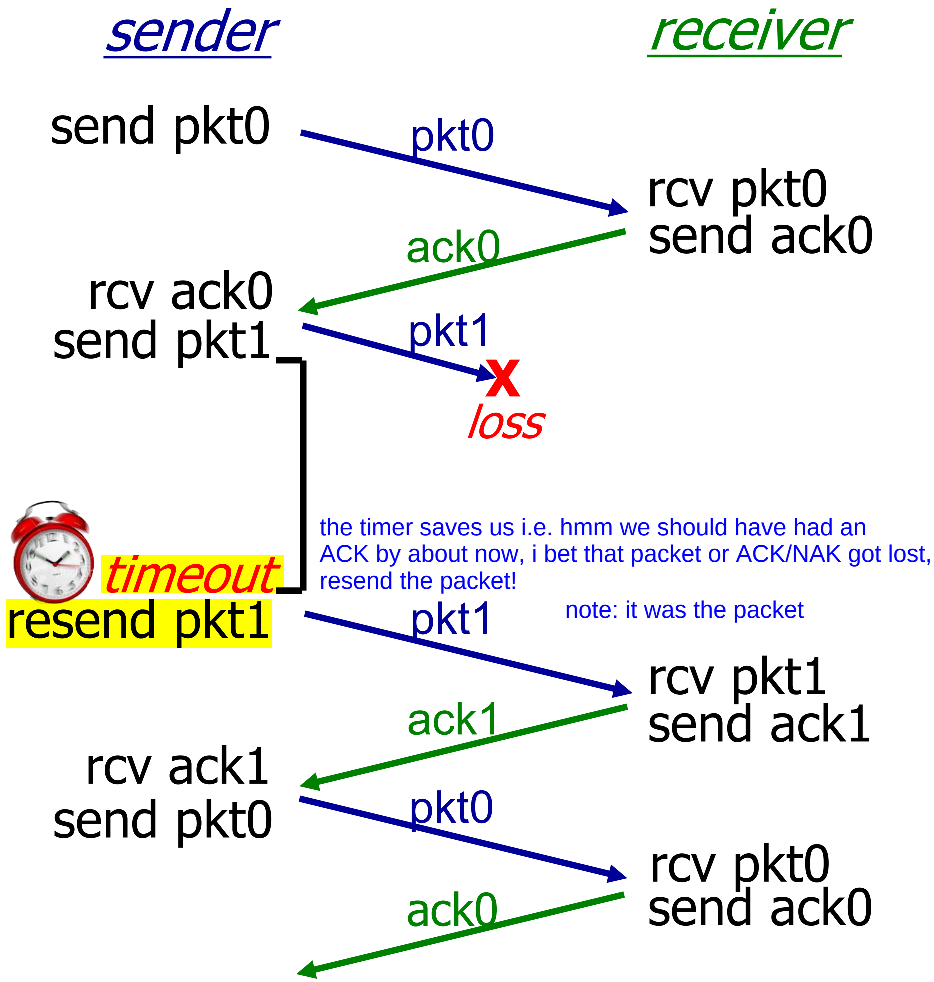
- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed this is already known
- requires countdown timer

rdt3.0 in action



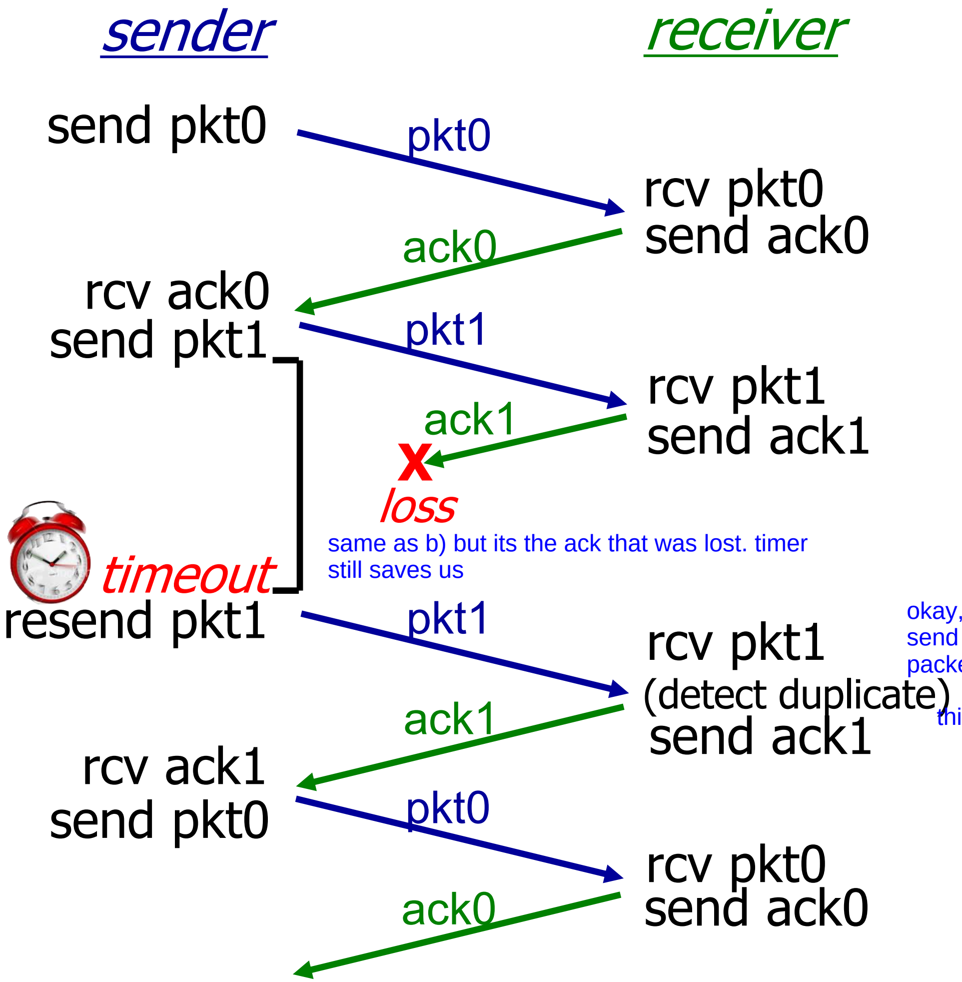
(a) no loss

having no loss is equivalent to the old system where loss wasn't possible, so of course this is fine!

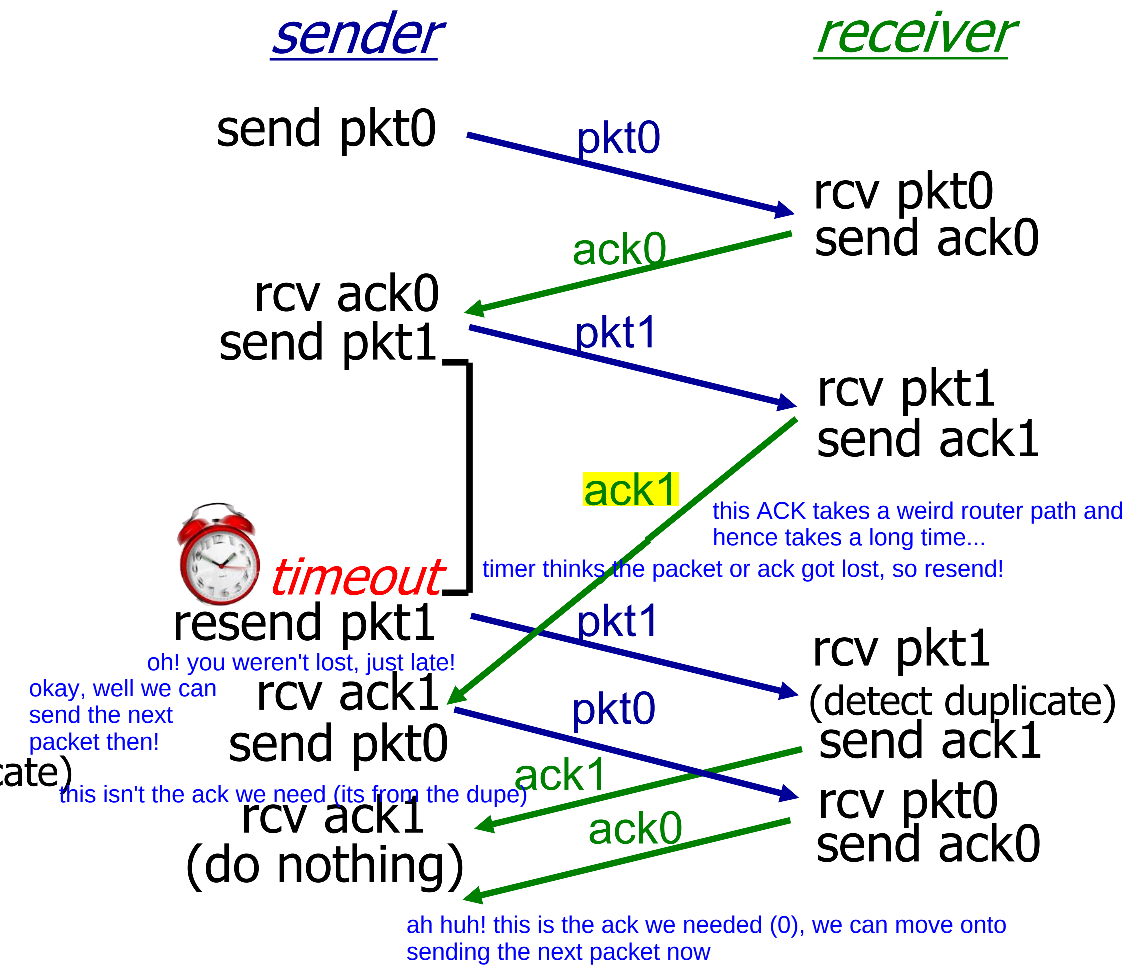


(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

was timer configured too early
OR did packet take a shoddy path?

Quiz: Reliable Data Transfer

- Which of the following are needed for reliable data transfer with only packet corruption (and no loss or reordering)? Use only as much as is strictly needed.

a) Checksums
b) Checksums, ACKs, NACKs
c) Checksums, ACKs
d) Checksums, ACKs, sequence numbers
e) Checksums, ACKs, NACKs, sequence numbers

ack+nack is not enough, how do you know if it's a duplicate packet that the sender is sending you because your last ack didn't make it back!

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Quiz: Reliable Data Transfer

- If packets (and ACKs and NACKs) could be lost which of the following is true of RDT 2.1 (or 2.2)?

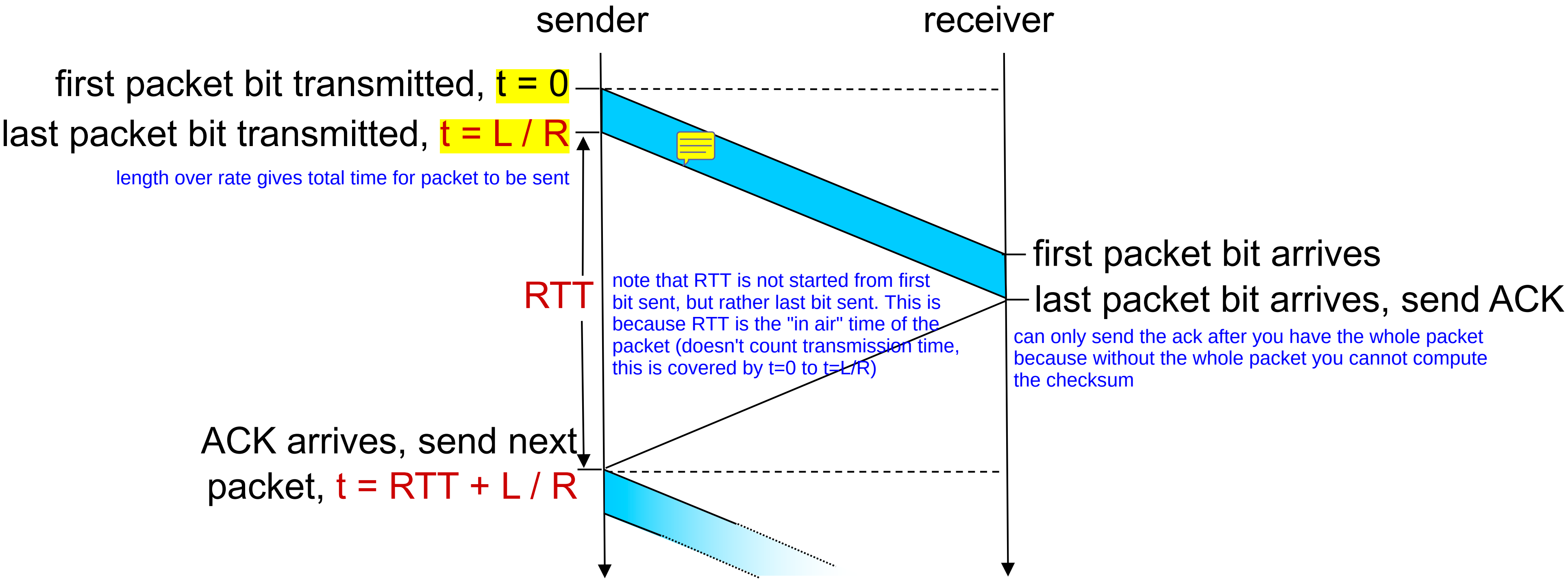
  a) Reliable in-order delivery is still achieved
  b) The protocol will get stuck
  c) The protocol will continue making progress but may skip delivering some messages

Quiz: Reliable Data Transfer

- Which of the following are needed for reliable data transfer to handle packet corruption and loss? Use only as much as is strictly needed.

  a) Checksums, timeouts
  b) Checksums, ACKs, sequence numbers
  c) Checksums, ACKs, timeouts
  d) Checksums, ACKs, timeouts, sequence numbers
  e) Checksums, ACKs, NACKs, timeouts, sequence numbers

rdt3.0: stop-and-wait operation



$$U_{sender} = \frac{L / R}{RTT + L / R}$$

time it takes to push packet out

time it takes from packet sent to be able to start sending next packet

hence the utility of the sender is ^ because for up to t = RTT + L/R, all the sender was able to do it push one packet out but then it just waiting doing nothing for the ACK to come back. Not very utilised!

Performance of rdt3.0

- rdt3.0 is correct, but performance stinks
- e.g.: 1 Gbps link, 8000 bit packet and 30msec RTT:

= 0.000008 seconds

$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs}$$

packet len

transmission rate

10^9 = 1,000,000,000 = 1Gb

note that the rate e.g. bps (bits per second), gives the time factor, so the answer is in seconds!

- U_{sender}: *utilization* – fraction of time sender busy sending

$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

- RTT=30 msec, 1KB pkt every 30.008 msec: 33kB/sec thruput over 1 Gbps link

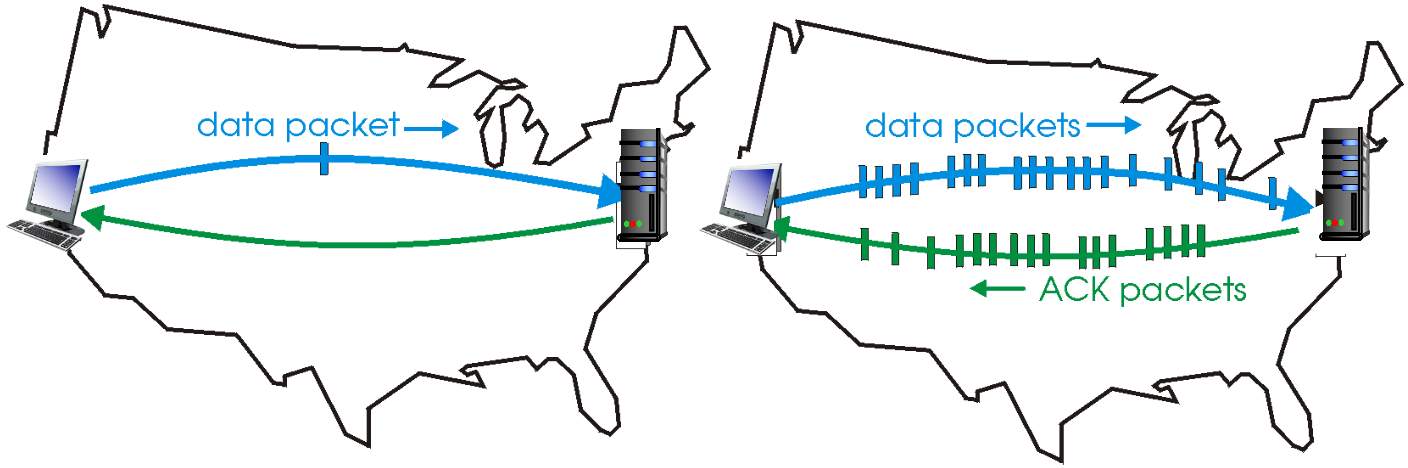
- Network protocol limits use of physical resources!

which is obvs bad, this would like be buying a supercomputer but then writing an O(n^3) function to run on it when there is an O(n) one possible!!!

Pipelined protocols

pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged pkts

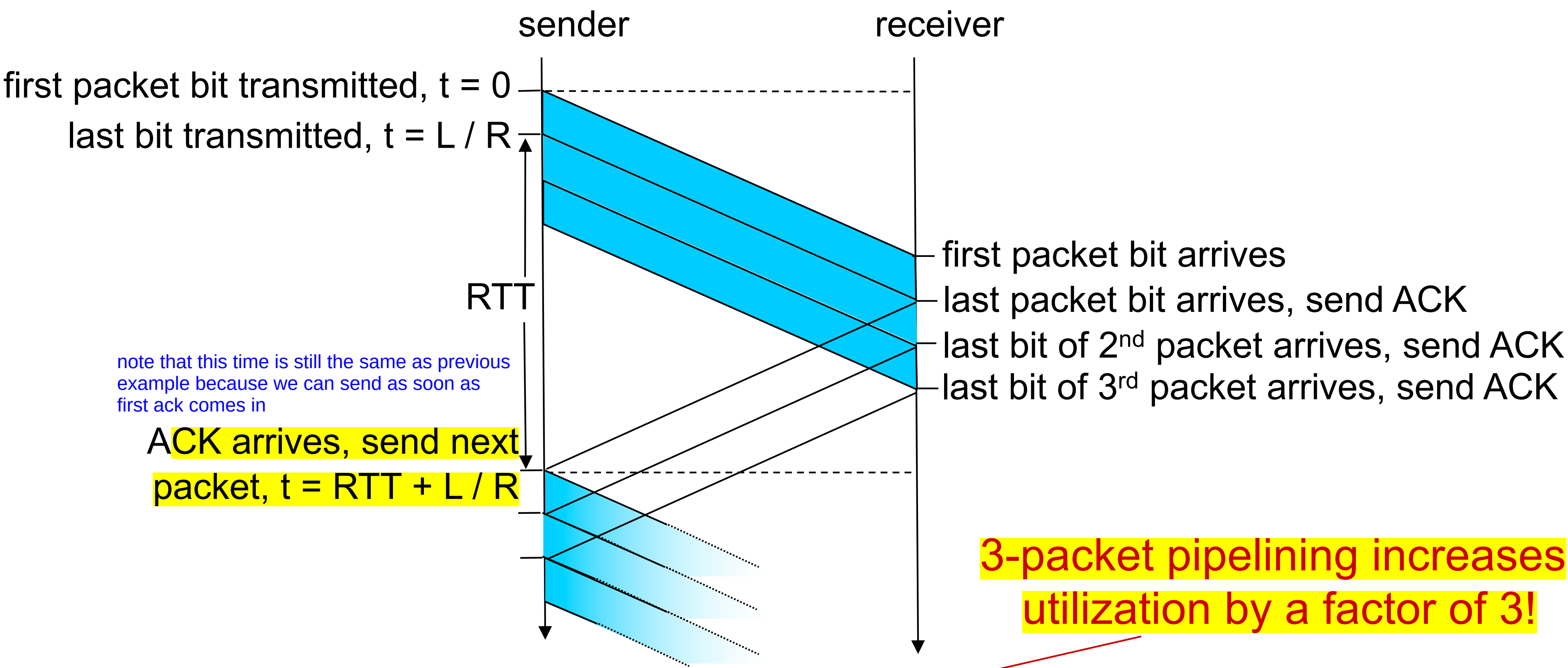
- range of sequence numbers must be increased
- buffering at sender and/or receiver



(a) a stop-and-wait protocol in operation (b) a pipelined protocol in operation

- two generic forms of pipelined (sliding window) protocols: *go-Back-N*, *selective repeat*

Pipelining: increased utilization



3-packet pipelining increases utilization by a factor of 3!

$$U_{sender} = \frac{3L / R}{RTT + L / R}$$

it's not 3L on the bottom because you can start sending again as soon as the first packet is acked

Pipelined protocols: overview

Go-Back-N:

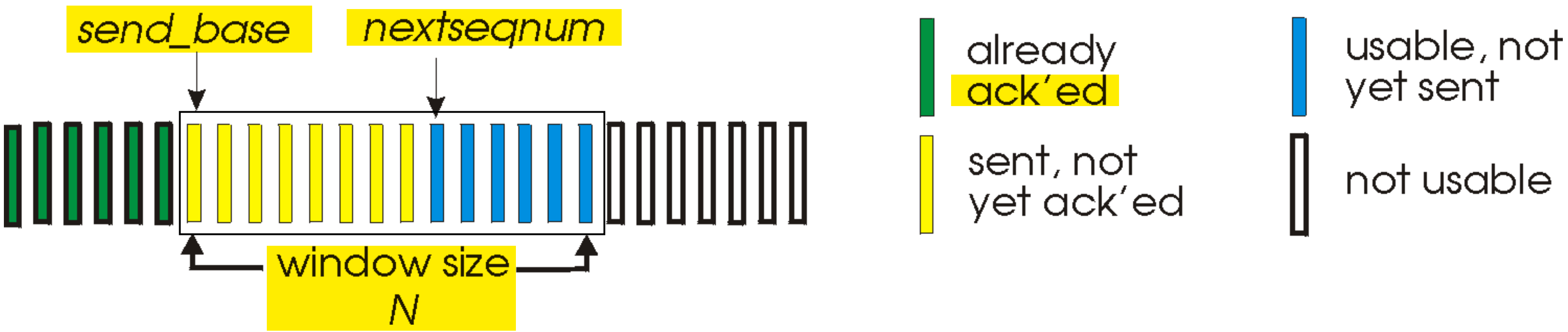
- Sender can have up to N unacked packets in pipeline
- Sender has single timer for oldest unacked packet, when timer expires, retransmit *all* unacked packets
- There is no buffer available at Receiver, out of order packets are discarded
- Receiver only sends *cumulative ack*, doesn't ack new packet if there's a gap

Selective Repeat:

- Sender can have up to N unacked packets in pipeline
- Sender maintains timer for each unacked packet, when timer expires, retransmit only that unacked packet
- Receiver has buffer, can accept out of order packets
- Receiver sends *individual ack* for each packet

Go-Back-N: sender

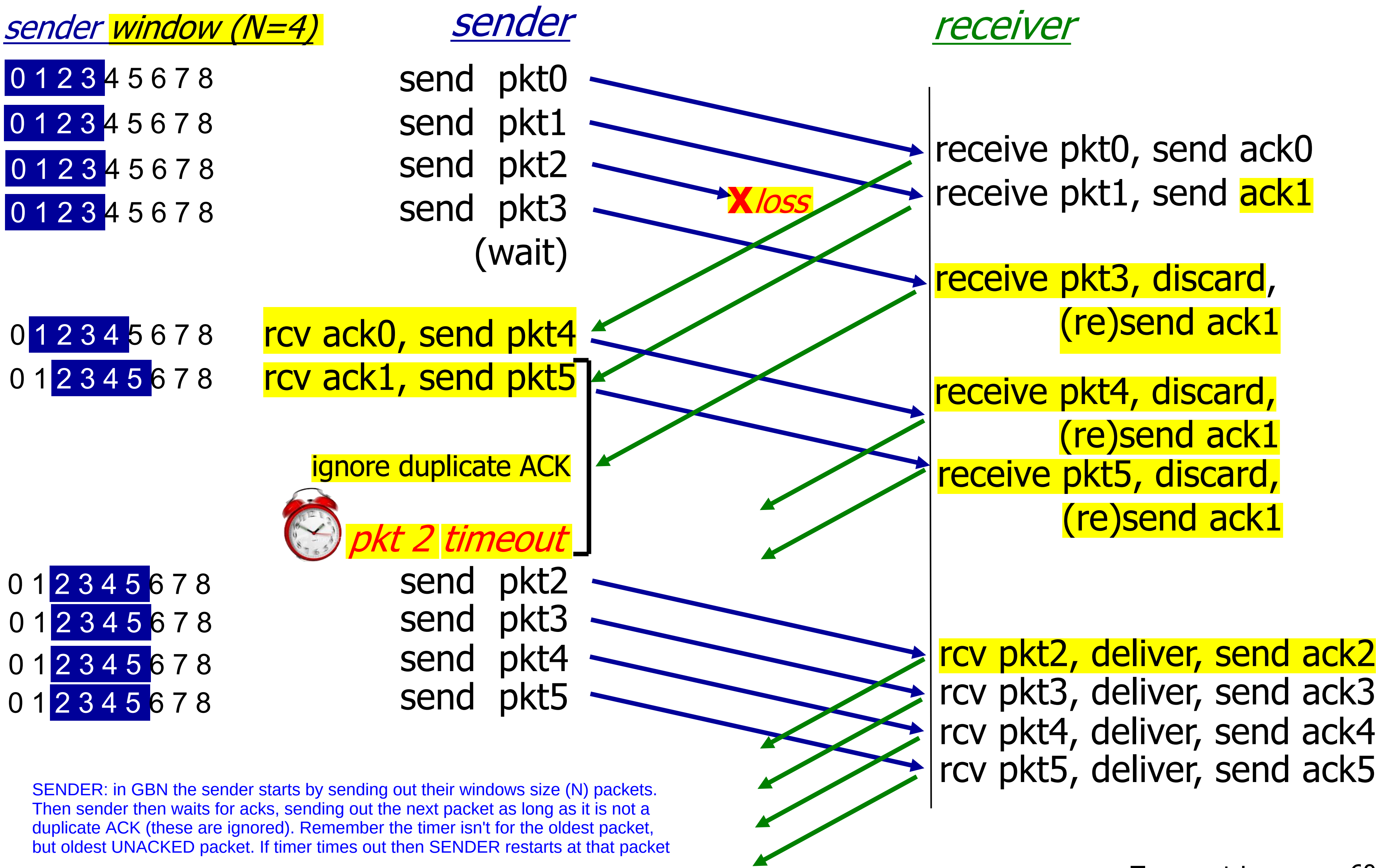
- k-bit seq # in pkt header
- "window" of up to N, consecutive unack'ed pkts allowed



- ACK(n): ACKs all pkts up to, including seq # n - *"cumulative ACK"*
  - may receive duplicate ACKs (see receiver)
- timer for oldest in-flight pkt
- *timeout(n):* retransmit packet n and all higher seq # pkts in window

Applets: http://media.pearsoncmg.com/aw/aw_kurose_network_2/applets/go-back-n/go-back-n.html
http://www.ccs-labs.org/teaching/rn/animations/gbn_sr/

GBN in action



SENDER: in GBN the sender starts by sending out their windows size (N) packets. Then sender then waits for acks, sending out the next packet as long as it is not a duplicate ACK (these are ignored). Remember the timer isn't for the oldest packet, but oldest UNACKED packet. If timer times out then SENDER restarts at that packet

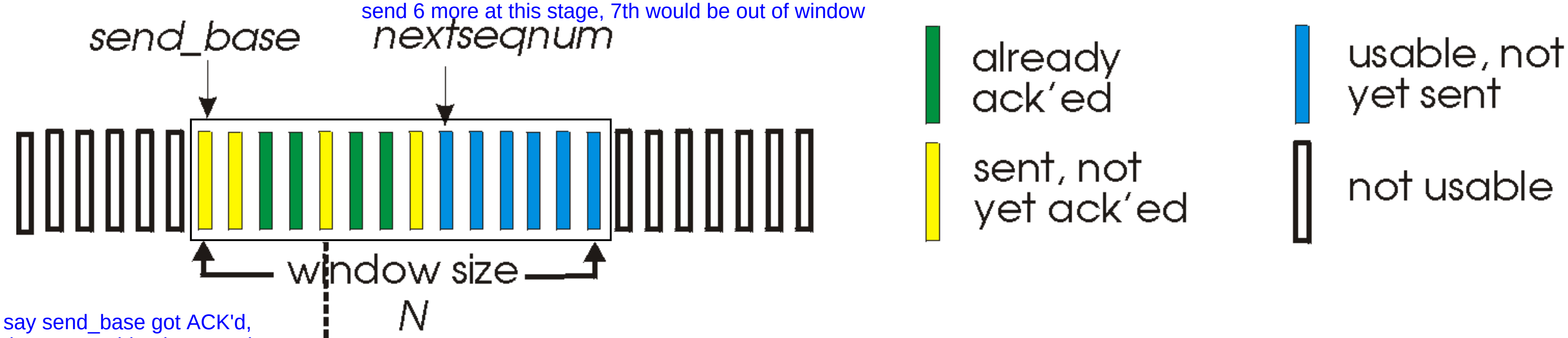
Selective repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
  - *N* consecutive seq #'s
  - limits seq #s of sent, unACKed pkts

Applet: http://media.pearsoncmg.com/aw/aw_kurose_network_3/applets/SelectRepeat/SR.html

Selective repeat: sender, receiver windows

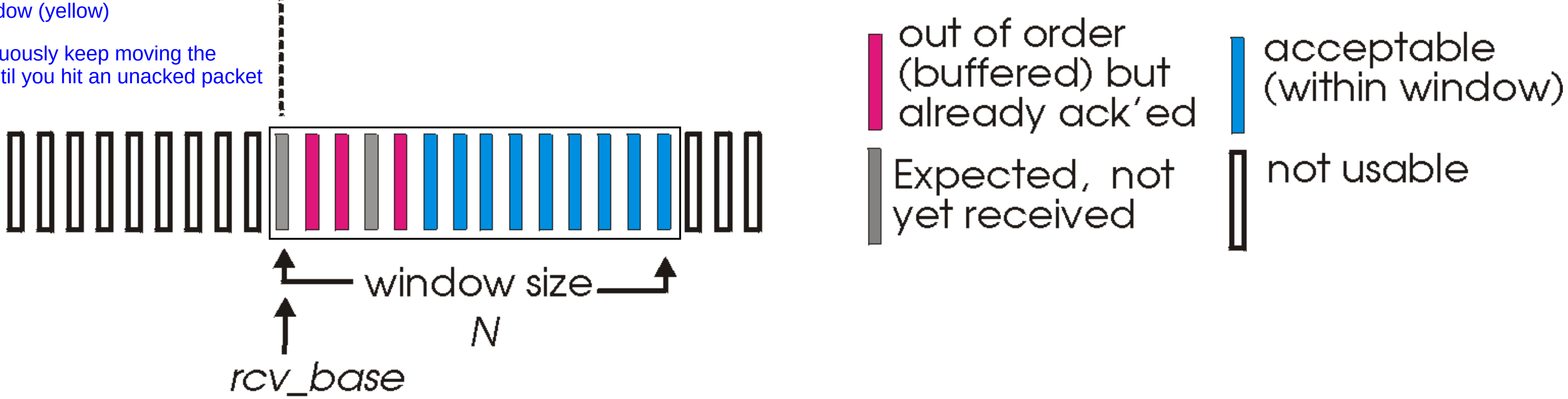
we are able to send another packet because the nextseq# falls within the window. Assuming the window didn't progress, we'd only be able to send 6 more at this stage, 7th would be out of window



(a) sender view of sequence numbers

say send_base got ACK'd, then we could only move the window up by 1, but then if that one got ACK'd we could move the window up 3! this is because the window can only move past ACK'd packets (green), and must keep UNACK'd packets in the window (yellow)

i.e. you continuously keep moving the send_base until you hit an unacked packet



(b) receiver view of sequence numbers

Selective repeat

sender

data from above:

- if next available seq # in window, send pkt

timeout(n):

- resend pkt n, restart timer

this is trivial

i.e. the current window

ACK(n) in [sendbase,sendbase+N]:

- mark pkt n as received
- if n smallest unACKed pkt, advance window base to next unACKed seq #

receiver

current window

pkt n in [rcvbase, rcvbase+N-1]

- send ACK(n)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

prev window

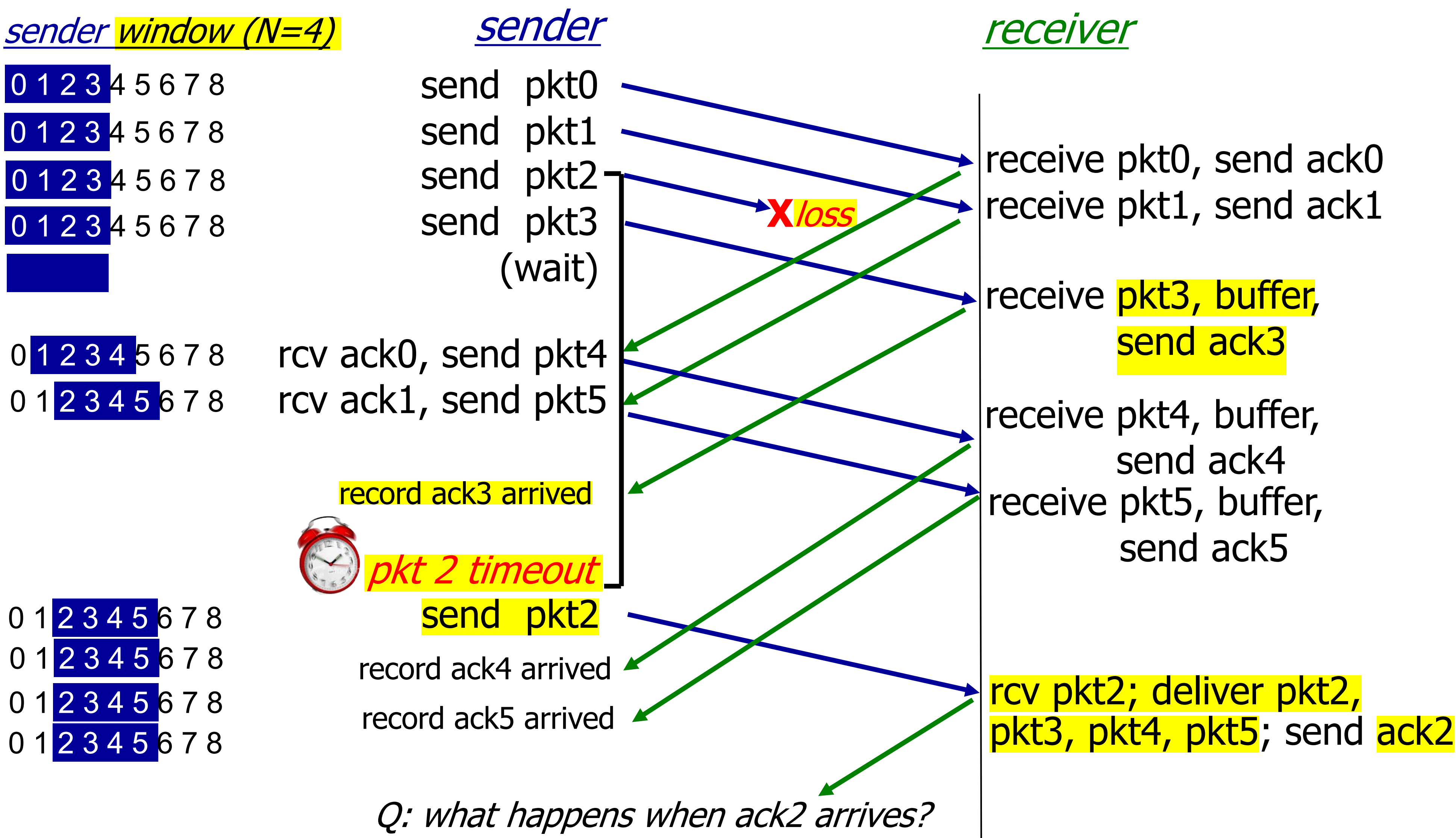
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

Selective repeat in action



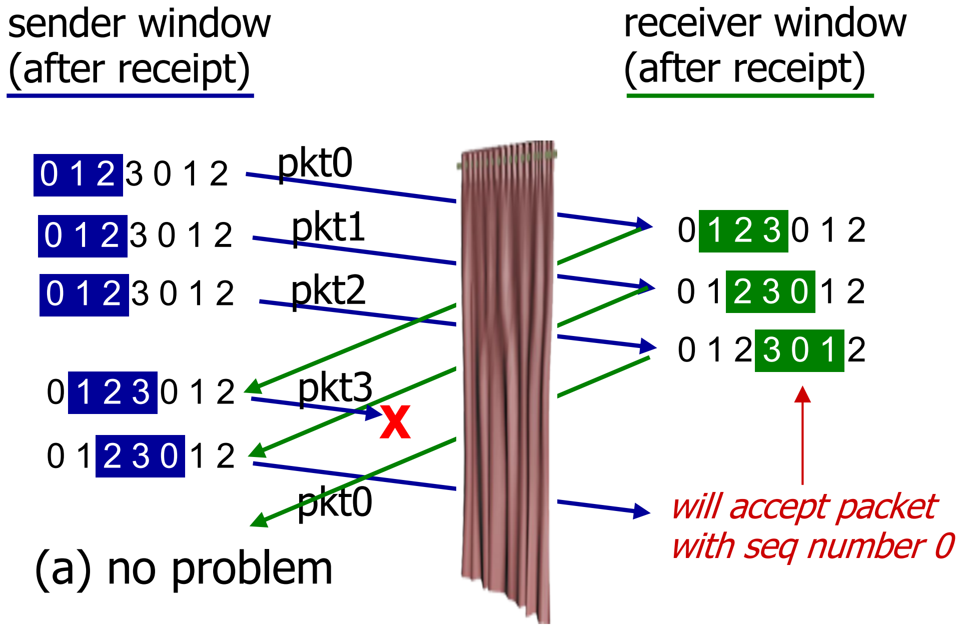
Selective repeat: dilemma

example:

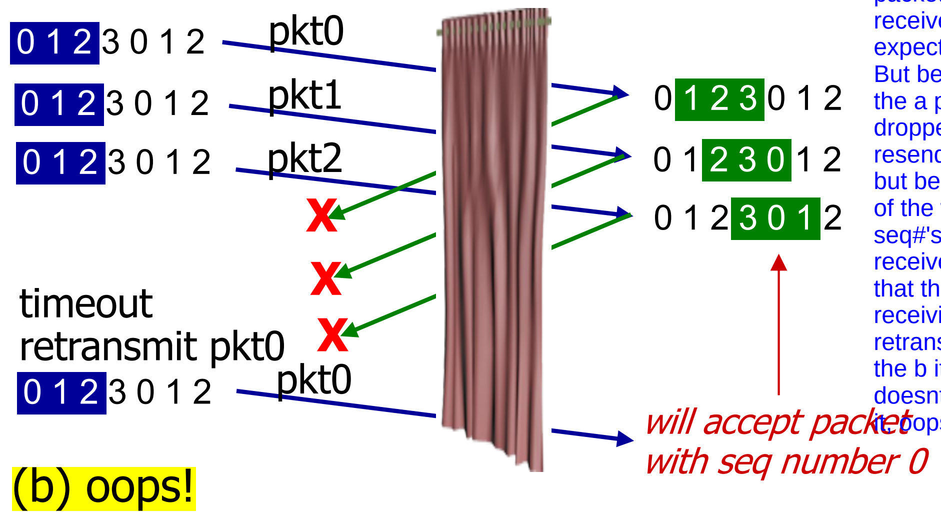
- seq #'s: 0, 1, 2, 3 i.e. 2bit seq#
- window size=3
- receiver sees no difference in two scenarios!
- duplicate data accepted as new in (b)

Q: what relationship between seq # size and window size to avoid problem in (b)?

A: Sender window size <= ½ of Sequence number space



receiver can't see sender side.
receiver behavior identical in both cases!
something's (very) wrong!



say we have seq#0a - seq#2a and seq #0b - #2b (a representing first batch, b representing seconds batch). All 'a' packets make it to the receiver, so now it is expecting the b packets. But because the acks for the a packets are dropped, the sender is resending the a packets, but because of the size of the window, the seq#'s overlap, and the receiver doesn't know that the seq#0 it is receiving is actually a retransmit of a and not the b it wants, but it doesnt know so it takes it. oops!

take the worst case, even if the window is half the seq# size (say window=2 and seq#'s=4), then if acks for 1 and 2 drop, the window can only move up to 3 and 4, and so does not overrun and use seq#' s 1 and 2 again before those are seen to be acked. (they will timeout and resend again). i..e this stop sequence numbers from being re-used before they're done with.

Recap: components of a solution

- Checksums (for error detection)
- Timers (for loss detection)
- Acknowledgments
  - cumulative
  - selective
- Sequence numbers (duplicates, windows) duplicates and pipelining
- Sliding Windows (for efficiency) pipelining

- Reliability protocols use the above to decide when and what to retransmit or acknowledge i.,e. TCP

Quiz: GBN, SR

- Which of the following is not true?

  a) GBN uses cumulative ACKs, SR uses individual ACKs
  b) Both GBN and SR use timeouts to address packet loss
  c) GBN maintains a separate timer for each outstanding packet
  d) SR maintains a separate timer for each outstanding packet
  e) Neither GBN nor SR use NACKs

Quiz: GBN, SR

- Suppose a receiver that has received all packets up to and including sequence number 24 and next receives packet 27 and 28. In response, what are the sequence numbers in the ACK(s) sent out by the GBN and SR receiver, respectively?

a) [27, 28], [28, 28]
b) [24, 24], [27, 28]   GBN re-acks last in-order ack in the event of getting an out of order packet. SR acks what it gets
c) [27, 28], [27, 28]
d) [25, 25], [25, 25]
e) [nothing], [27, 28]

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

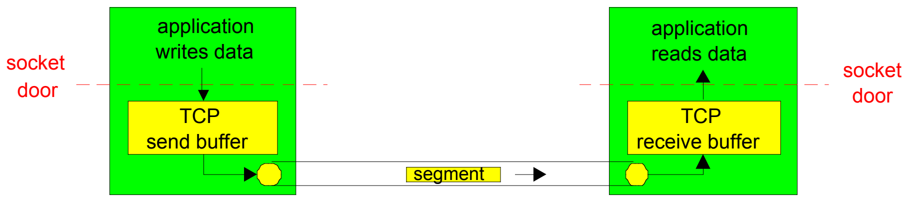
Practical Reliability Questions

- How do the sender and receiver keep track of outstanding pipelined segments?
- How many segments should be pipelined?
- How do we choose sequence numbers?
- What does connection establishment and teardown look like?
- How should we choose timeout values?

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

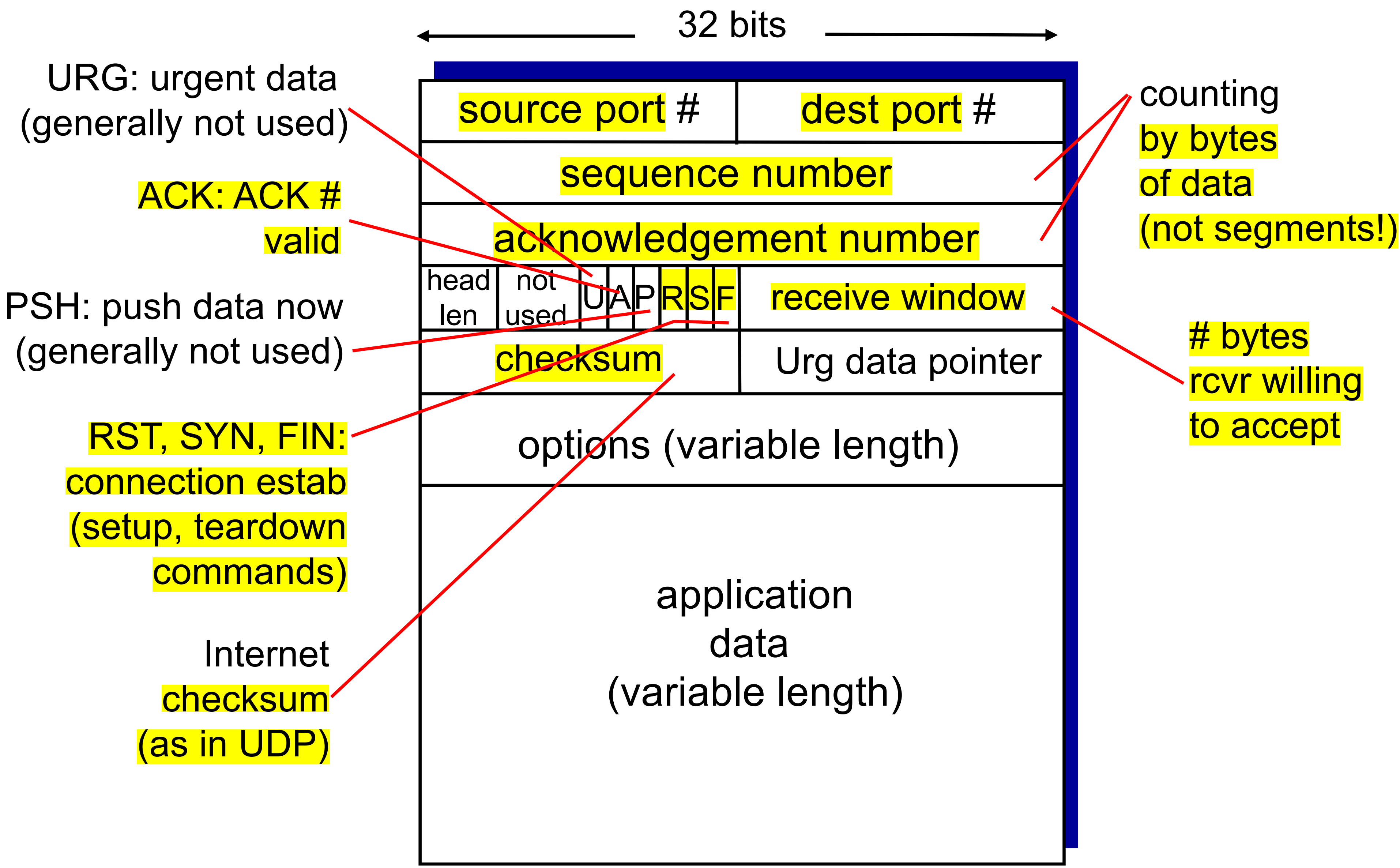
- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte stream:*
  - no "message boundaries"
- pipelined:

  dynamic window based on traffic
  - TCP congestion and flow control set window size
- send and receive buffers

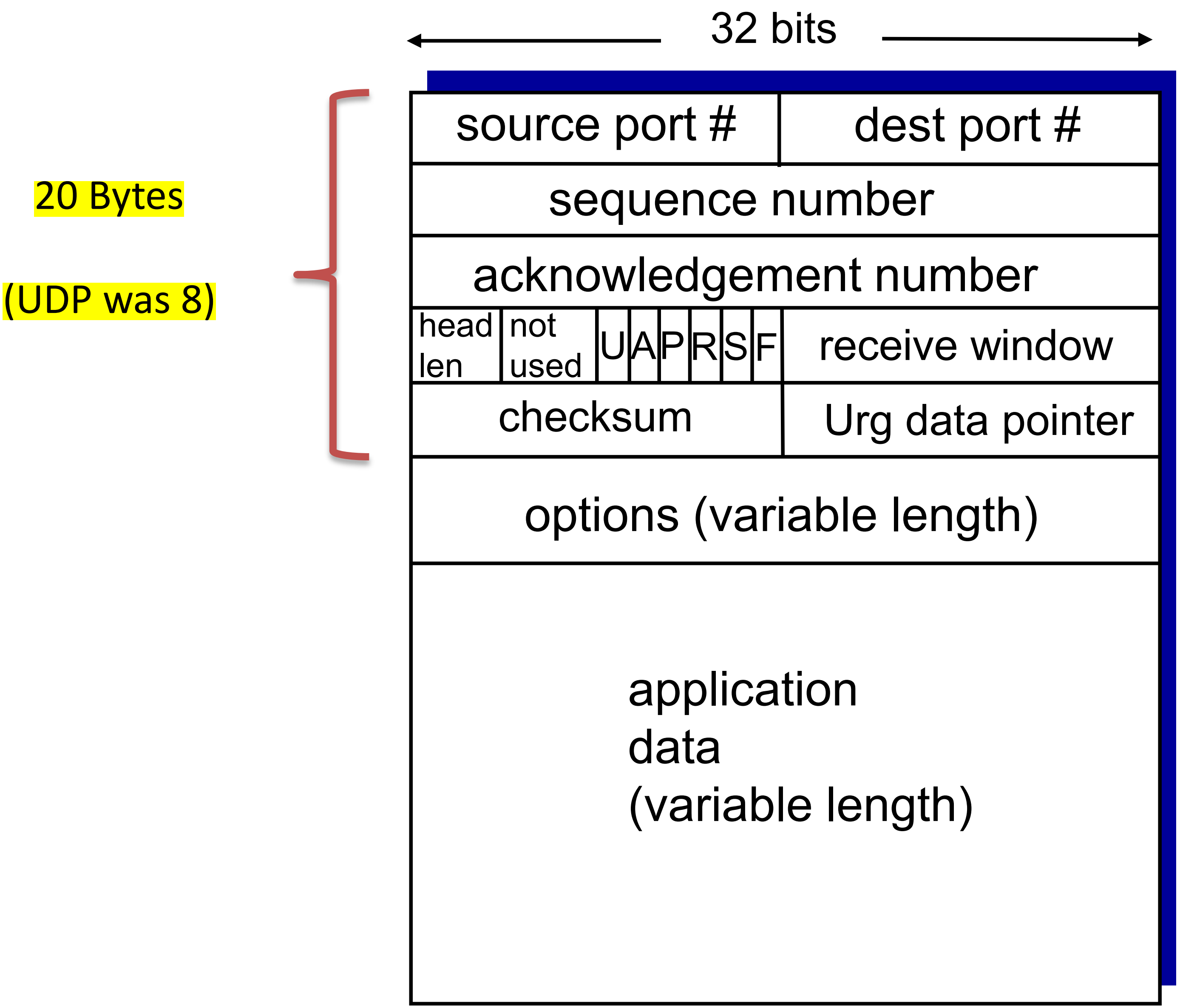


- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control msgs) inits sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure



TCP segment structure



Transport Layer Outline

3.1 transport-layer services

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Recall: Components of a solution for reliable transport

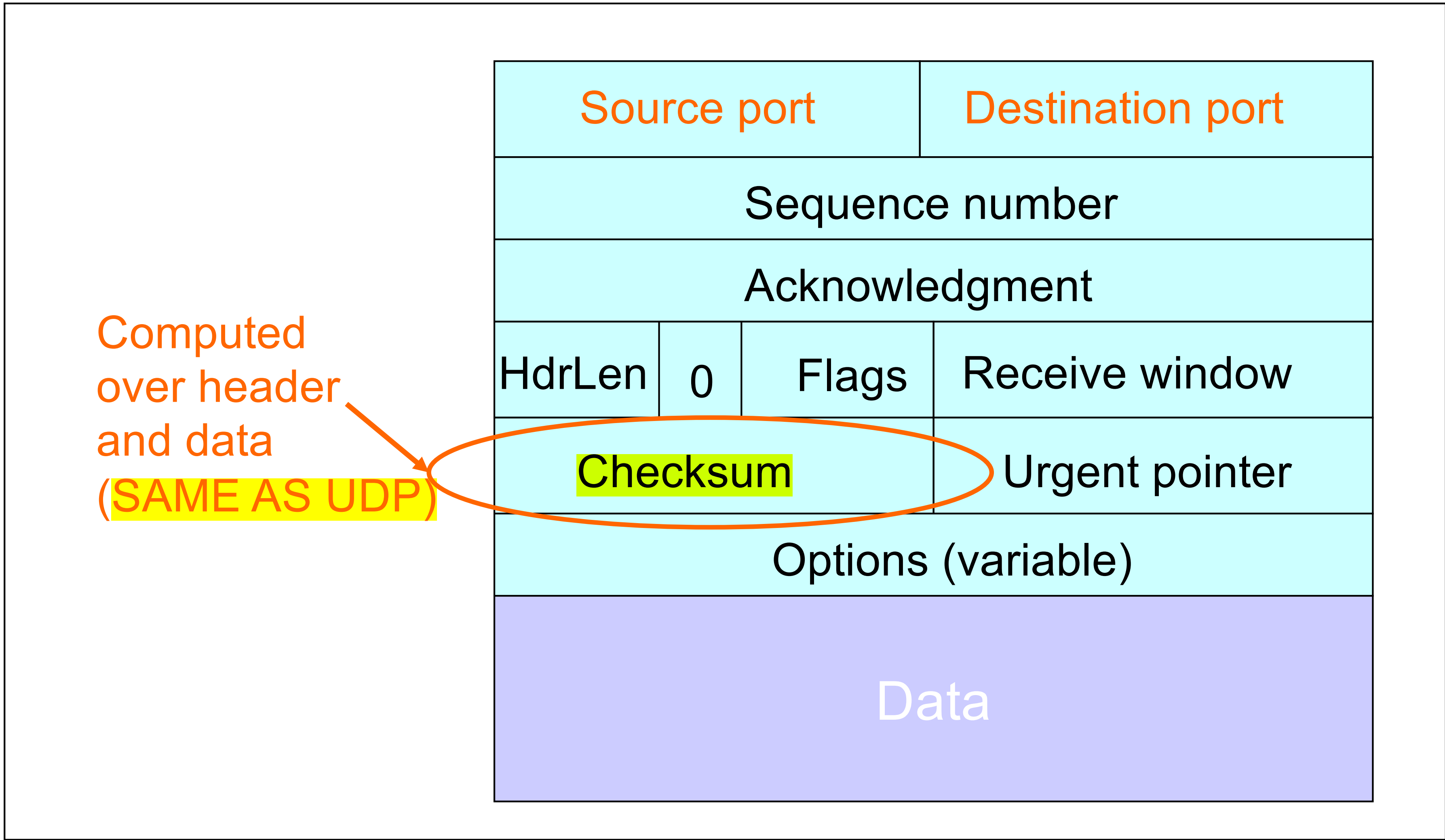
- Checksums (for error detection)
- Timers (for loss detection)
- Acknowledgments
  - cumulative
  - selective
- Sequence numbers (duplicates, windows)
- Sliding Windows (for efficiency)
  - Go-Back-N (GBN)
  - Selective Repeat (SR)

What does TCP do?

Many of our previous ideas, but some key differences

- Checksum

TCP Header


Source port
Destination port
Sequence number
Acknowledgment
HdrLen
0
Flags
Receive window
Checksum
Urgent pointer
Options (variable)
Data
Computed over header and data (SAME AS UDP)

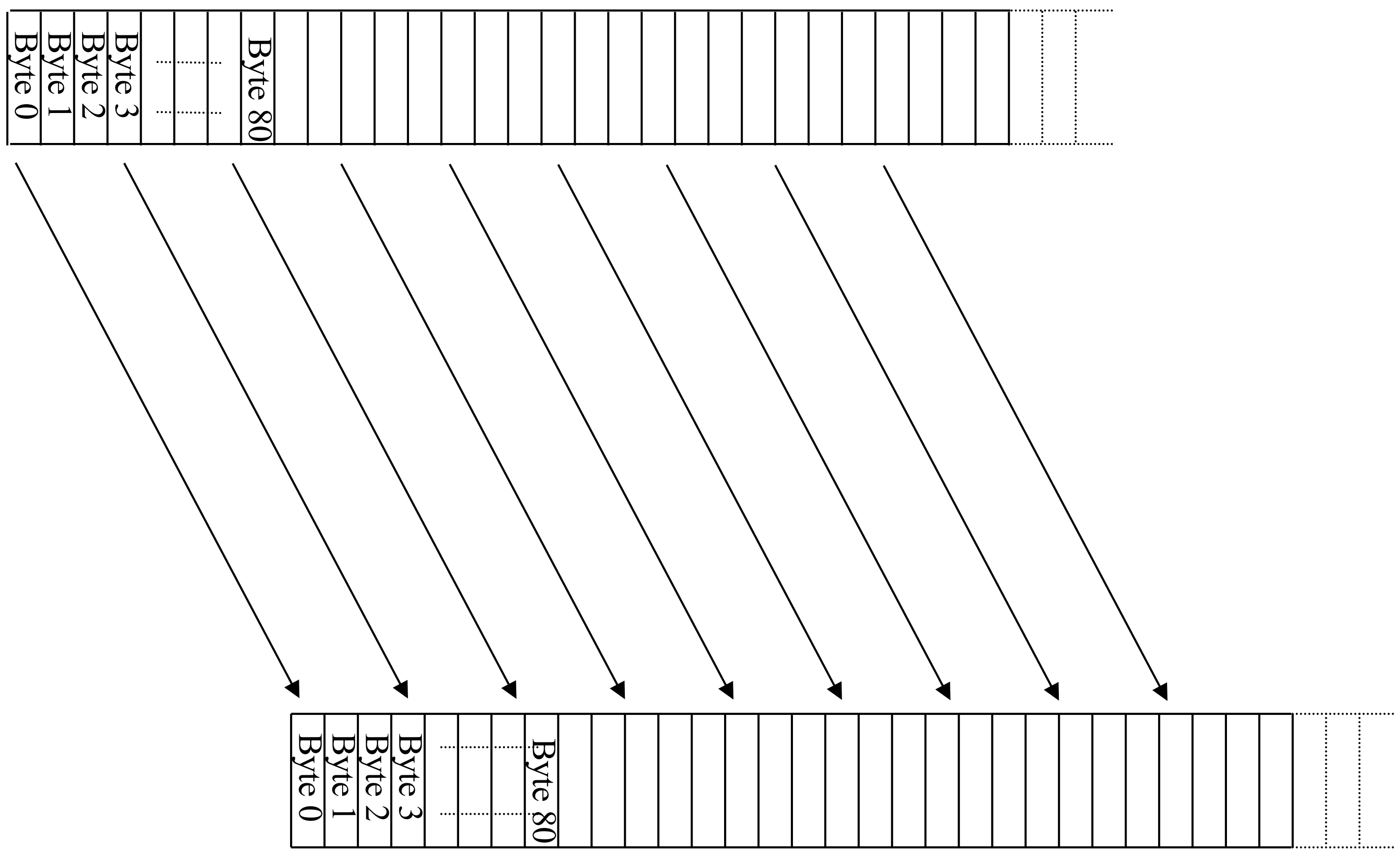
What does TCP do?

Many of our previous ideas, but some key differences

- Checksum
- **Sequence numbers are byte offsets**

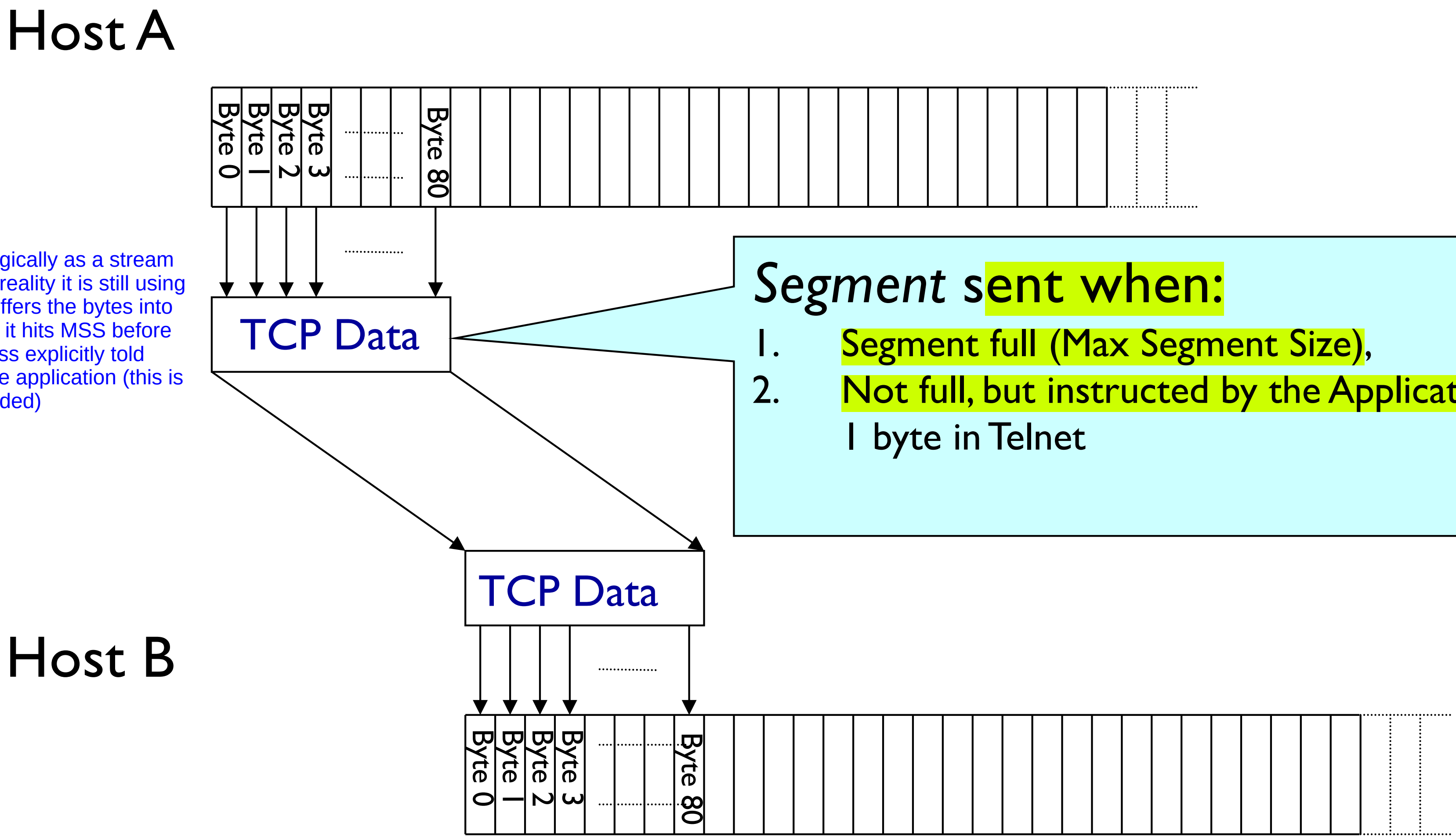
TCP "Stream of Bytes" Service ..

Application @ Host A



Application @ Host B

.. Provided Using TCP "Segments"

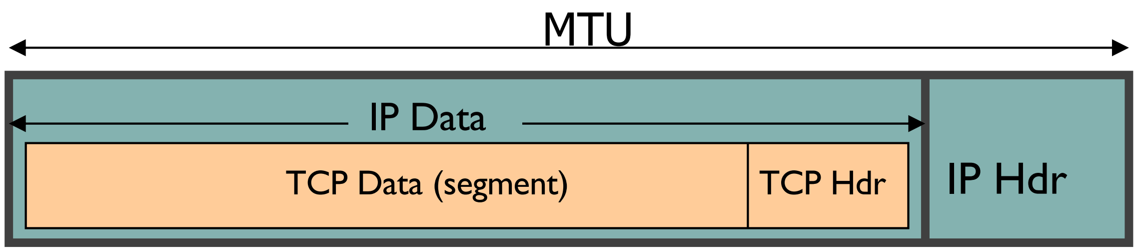


so it appears logically as a stream of bytes, but in reality it is still using segments. It buffers the bytes into a segment until it hits MSS before sending it, unless explicitly told otherwise by the application (this is sometimes needed)

Segment sent when:

1. Segment full (Max Segment Size),
2. Not full, but instructed by the Application e.g., 1 byte in Telnet

TCP Maximum Segment Size



- IP packet
  - No bigger than Maximum Transmission Unit (MTU)
  - E.g., up to 1500 bytes with Ethernet

so TCP funnels down to the network layer which eventually funnels down to IP, so whatever IP is bottle-necked by, so is network and TCP. So we are capped at 1500 bytes (for ethernet)

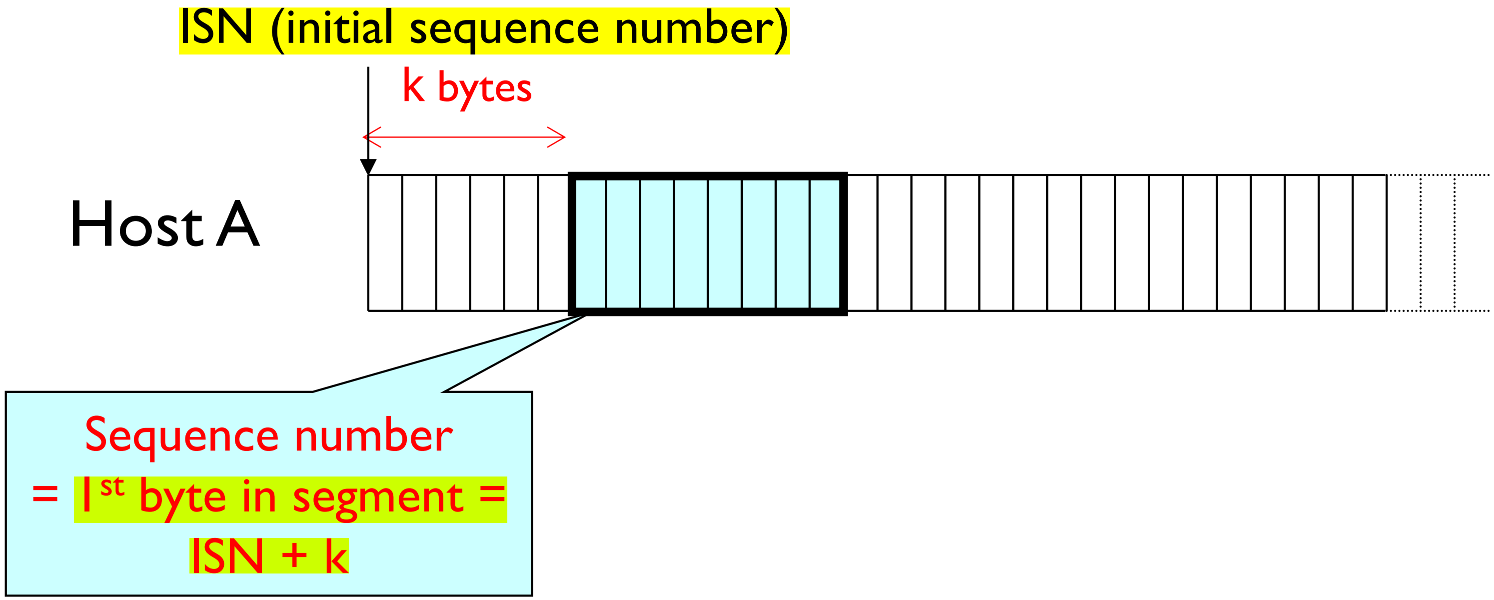
- TCP packet
  - IP packet with a TCP header and data inside
  - TCP header ≥ 20 bytes long

but the IP packet isn't all data, it has the IP header (min 20 bytes), and the TCP header which is also min 20 bytes.

- TCP **segment**
  - No more than Maximum Segment Size (MSS) bytes
  - E.g., up to 1460 consecutive bytes from the stream
  - MSS = MTU – 20 (min IP header) – 20 (min TCP header)

so this means MSS = MTU - len(IP header) - len(TCP header), i.e. in this case, 1500 - 20 - 20 = 1460

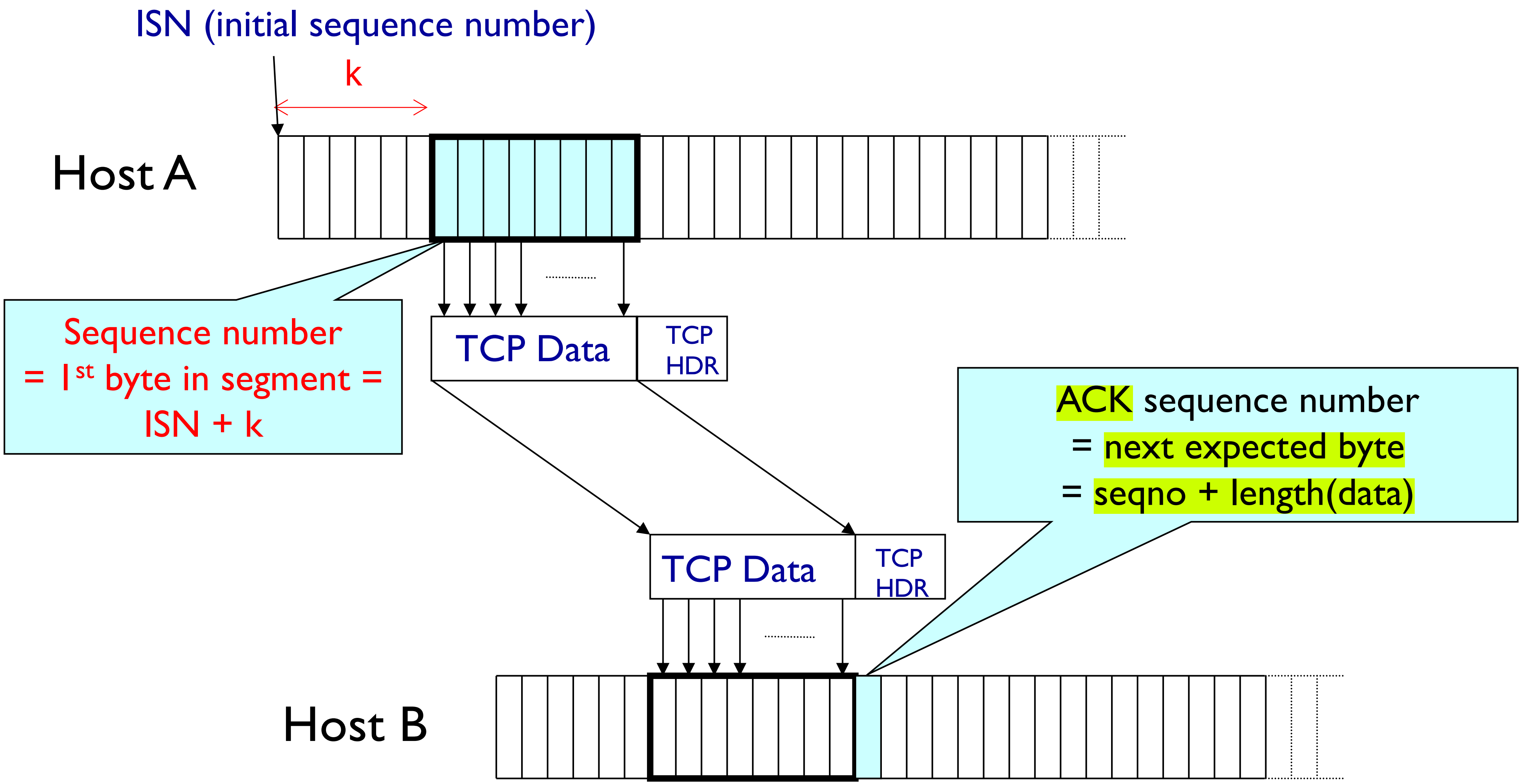
Sequence Numbers



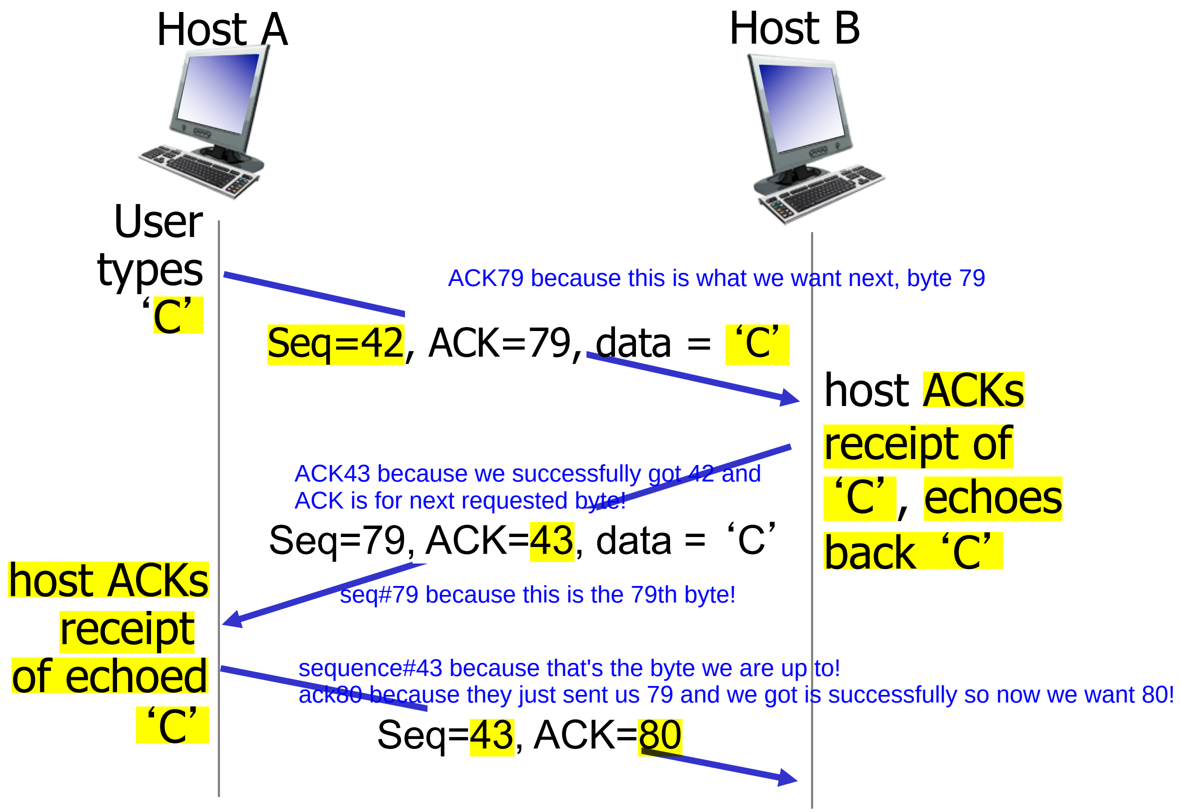
Sequence numbers:

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

Sequence & Ack Numbers



TCP seq. numbers, ACKs



simple telnet scenario

Transport Part 1: Summary

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

- **Next Week:**
  - TCP
    - TCP Flow Control
    - TCP Connection Management