NETWORK PROTOCOLS

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TRANSPORT LAYER
AND
TRANSPORT LAYER PROTOCOLS

LECTURE 4 PART B

2204 - 2025

21 October

Our goal

Our goal In this lecture is to:

❑ **understand principles behind transport layer services:**

- multiplexing, demultiplexing
- **reliable data transfer**
- **flow control**
- **congestion control**

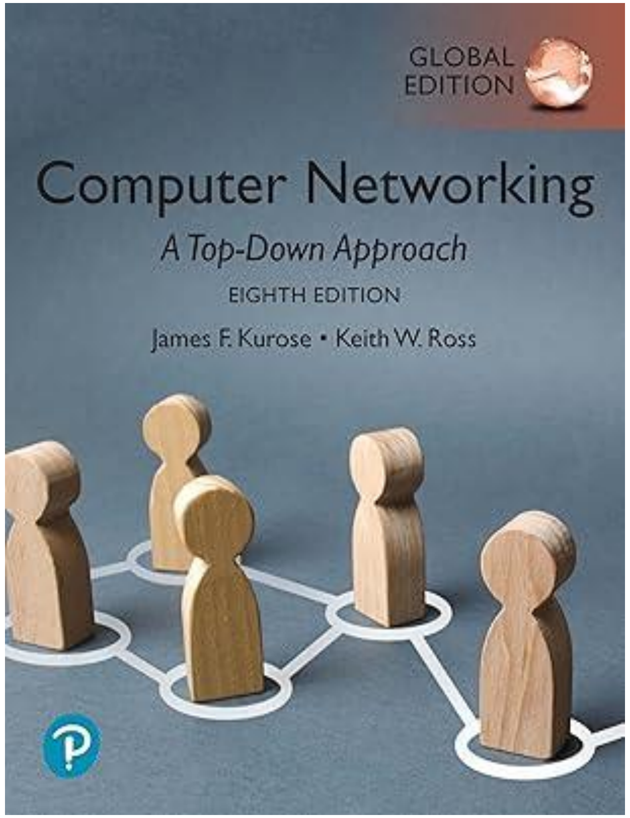
❑ **learn about Internet transport layer protocols:**

- **UDP: connectionless transport**
- **TCP: connection-oriented reliable transport**
- **TCP congestion control**

Transport layer: roadmap

In this lecture part B will talk about the following:

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
- Principles of congestion control
- TCP congestion control
- Evolution of transport-layer functionality



Transport layer: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
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- TCP congestion control
- Evolution of transport-layer functionality

UDP: User Datagram Protocol

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

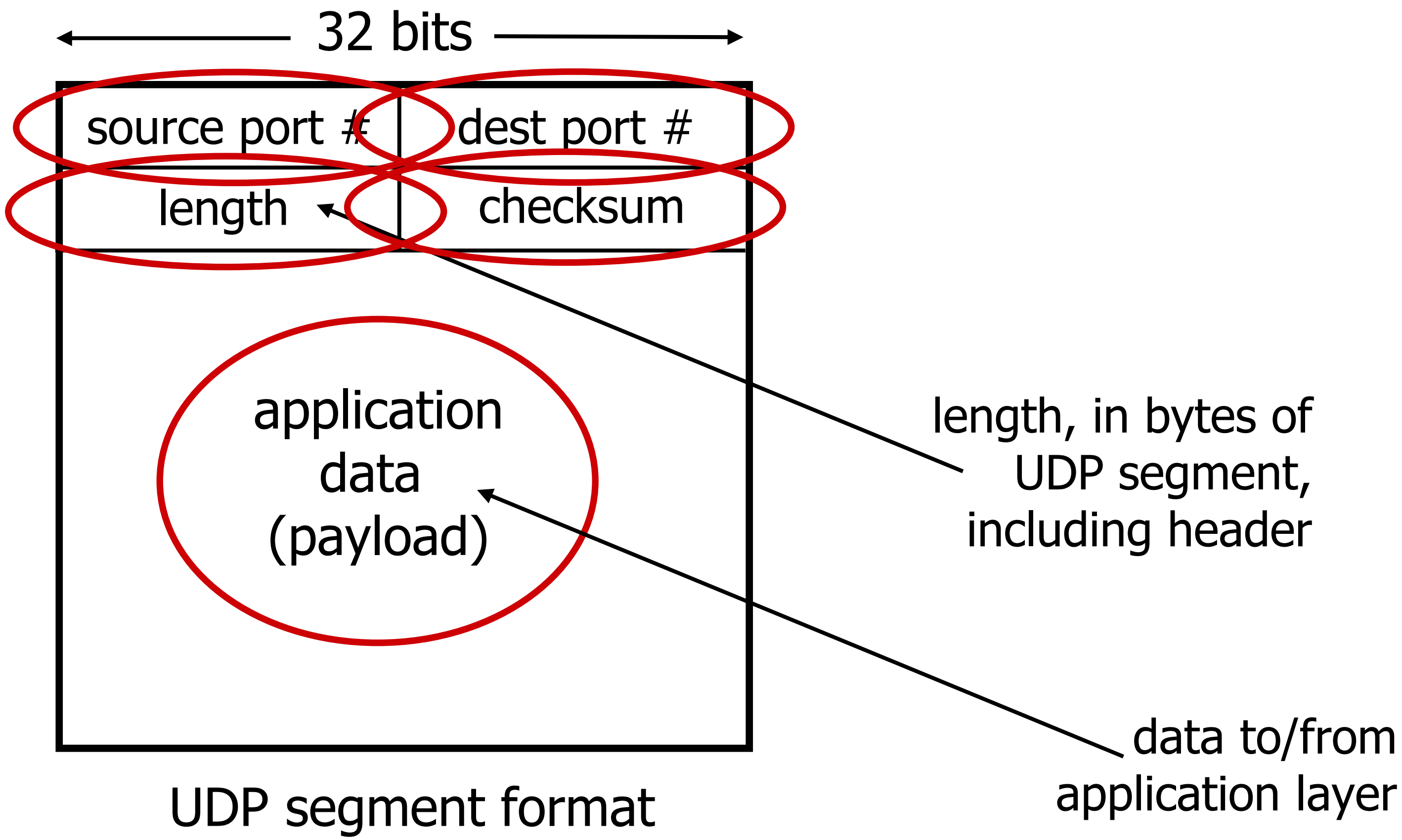
Why is there a UDP?

- no connection establishment (which can add **round-trip time RTT** delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control
  - UDP can blast away as fast as desired!
  - can function in the face of congestion

UDP: User Datagram Protocol

- UDP use:
  - streaming multimedia apps
  - DNS
  - SNMP (Simple Network Management Protocol)
- if reliable transfer needed over UDP:
  - add needed reliability at application layer
  - add congestion control at application layer

UDP segment header

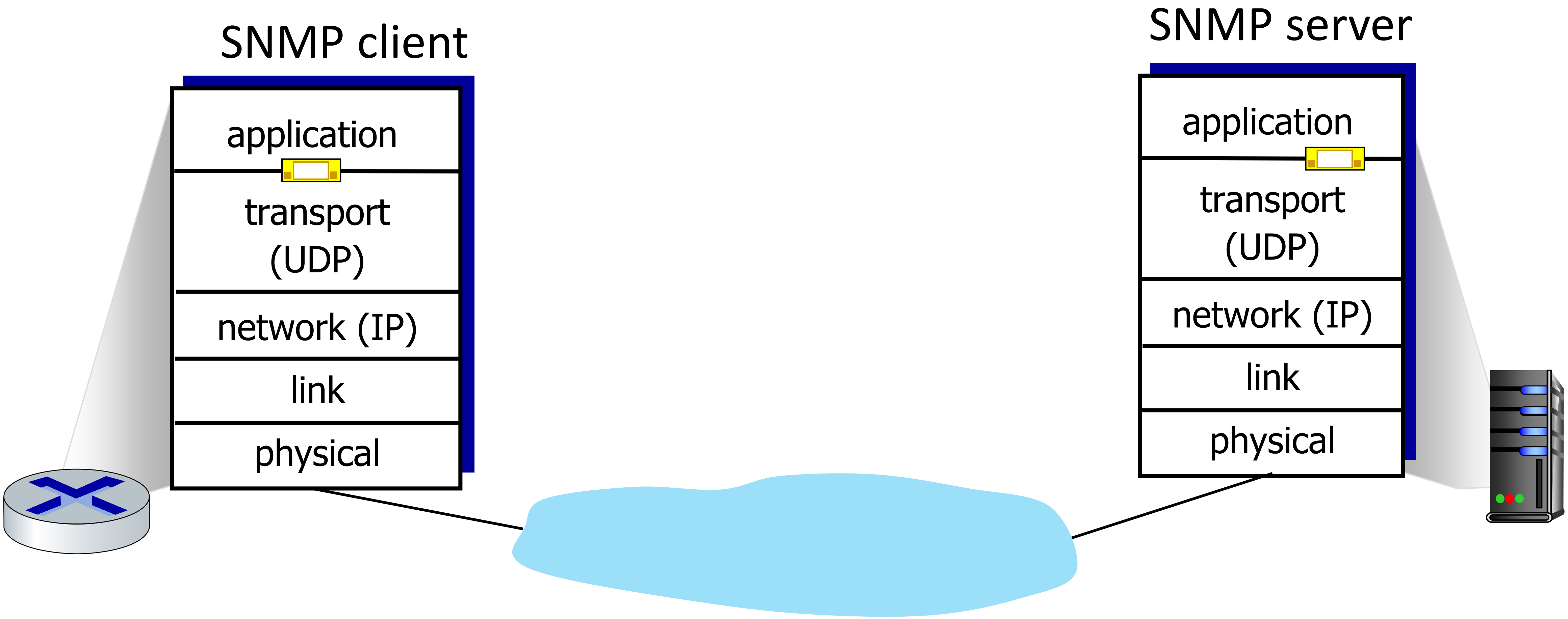


UDP segment format

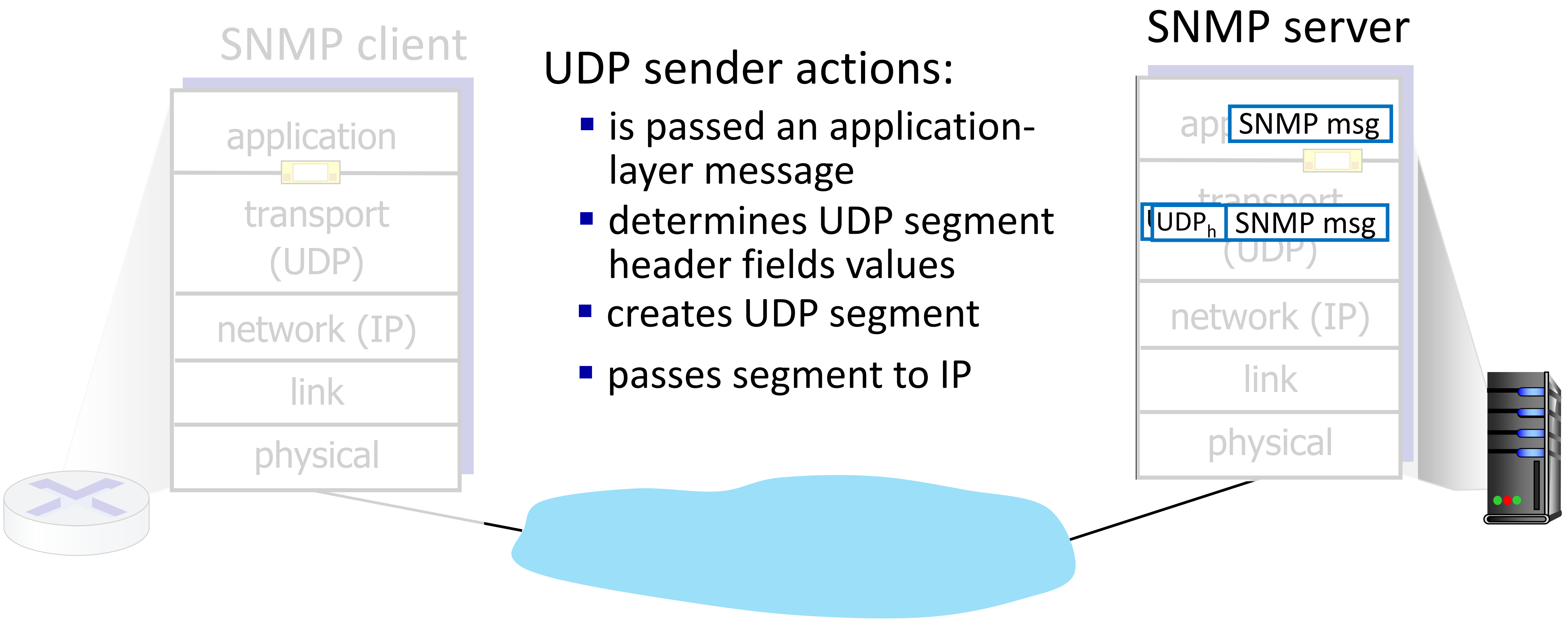
length, in bytes of UDP segment, including header

data to/from application layer

UDP: Transport Layer Actions



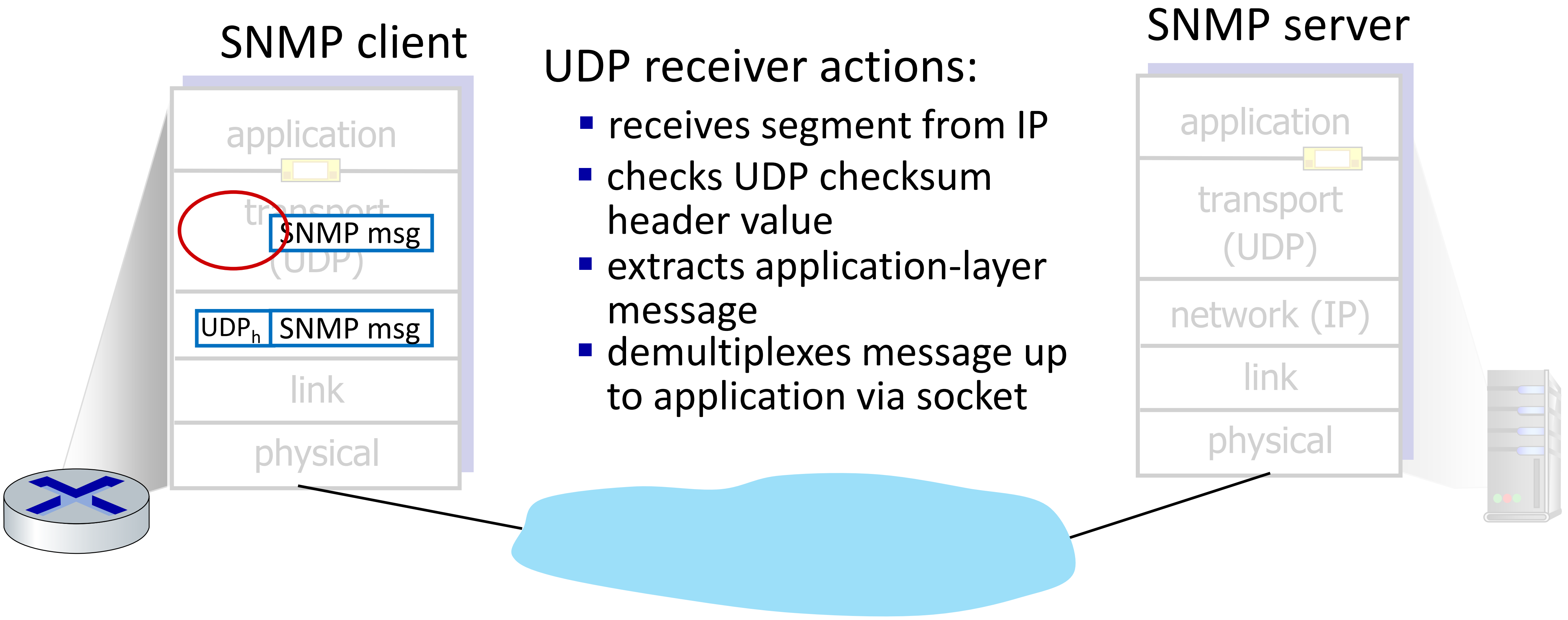
UDP: Transport Layer Actions



UDP sender actions:

- is passed an application-layer message
- determines UDP segment header fields values
- creates UDP segment
- passes segment to IP

UDP: Transport Layer Actions

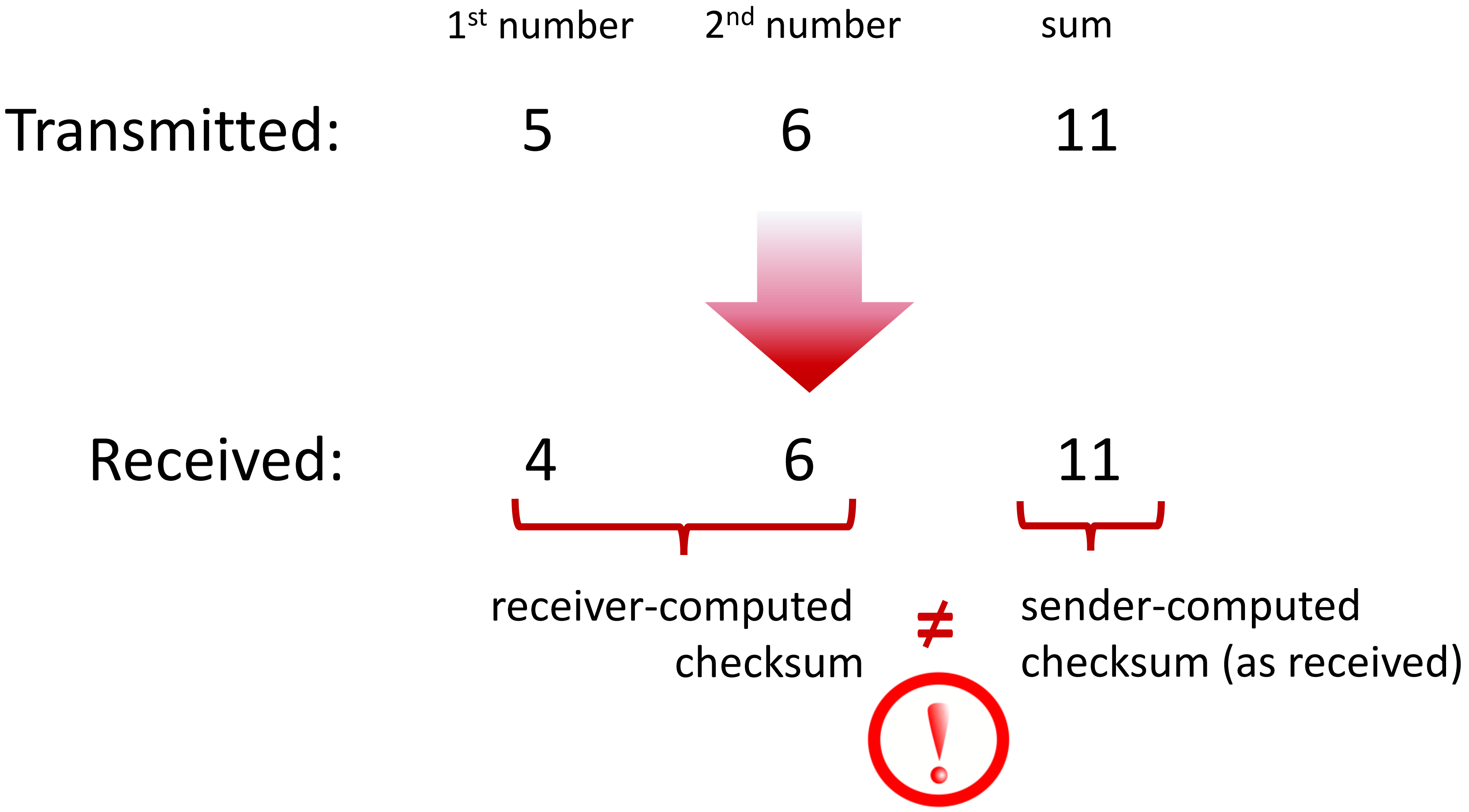


UDP receiver actions:

- receives segment from IP
- checks UDP checksum header value
- extracts application-layer message
- demultiplexes message up to application via socket

UDP checksum

Goal: detect errors (*i.e.*, flipped bits) in transmitted segment



	1st number	2nd number	sum
Transmitted:	5	6	11
Received:	4	6	11

receiver-computed checksum ≠ sender-computed checksum (as received)

Internet checksum

Goal: detect errors (*i.e.*, flipped bits) in transmitted segment

sender:

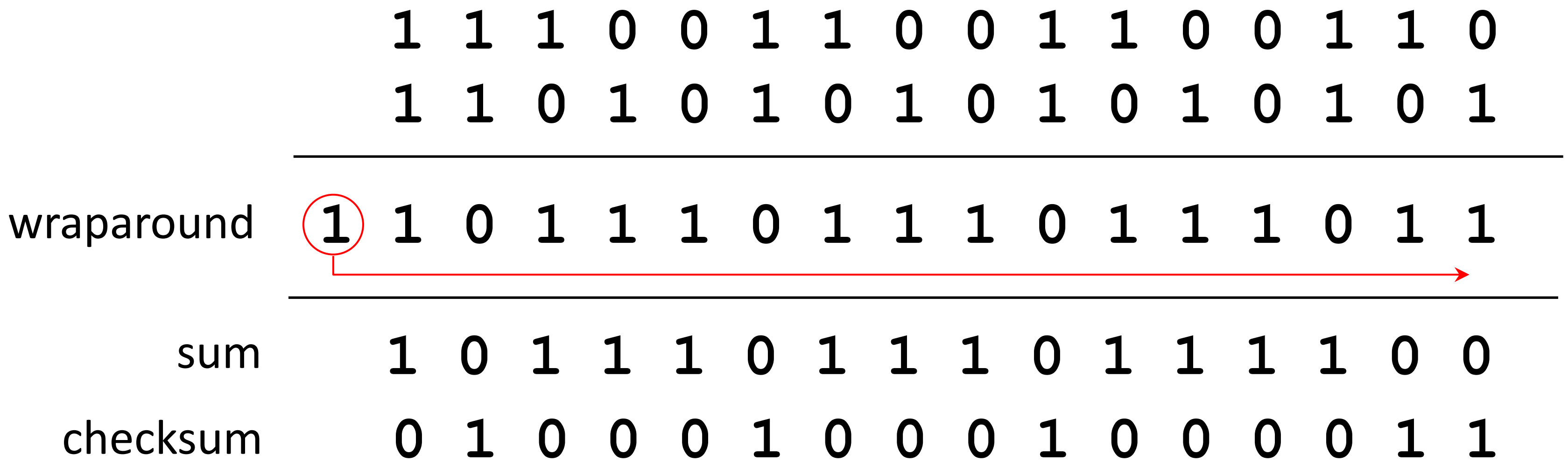
- treat contents of UDP segment (including UDP header fields and IP addresses) as sequence of 16-bit integers
- checksum: addition (one's complement sum) of segment content
- checksum value put into UDP checksum field

receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - not equal - error detected
  - equal - no error detected. *But maybe errors nonetheless?* More later ….

Internet checksum: an example

example: add two 16-bit integers



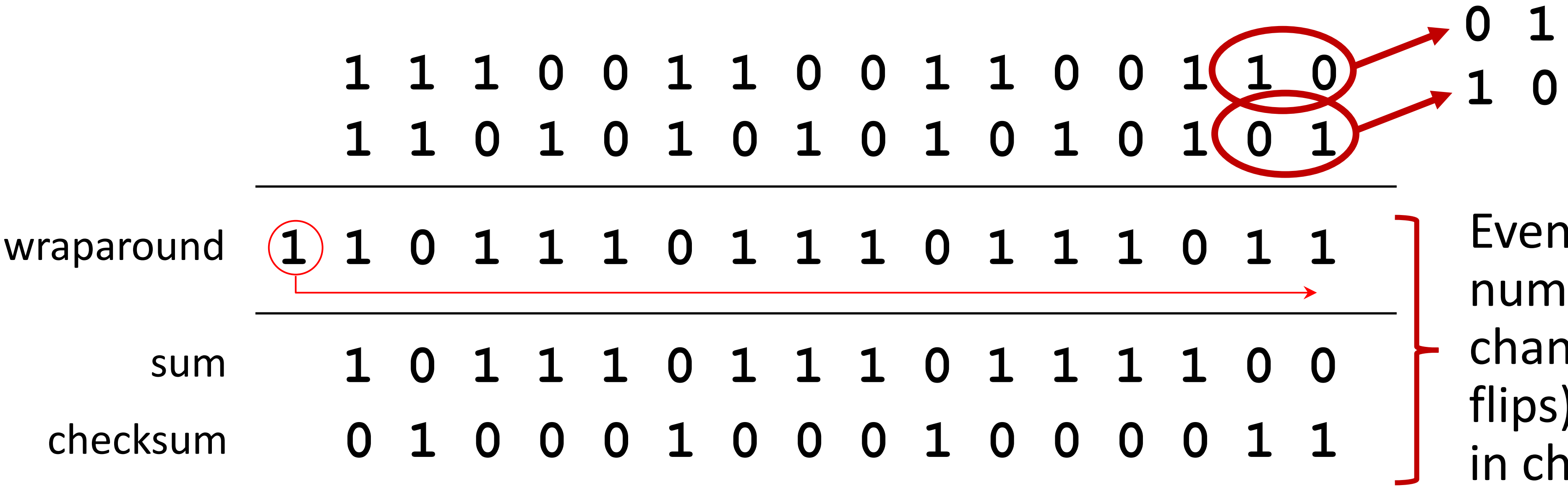
		1 1 1 0 0 1 1 0 0 1 1 0 0 1 1 0
		1 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1
wraparound	1	1 0 1 1 1 0 1 1 1 0 1 1 1 0 1 1
sum		1 0 1 1 1 0 1 1 1 0 1 1 1 1 0 0
checksum		0 1 0 0 0 1 0 0 0 1 0 0 0 0 1 1

Note: when adding numbers, a carryout from the most significant bit needs to be added to the result

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

Internet checksum: weak protection!

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

Even though numbers have changed (bit flips), *no* change in checksum!

Transport layer: roadmap

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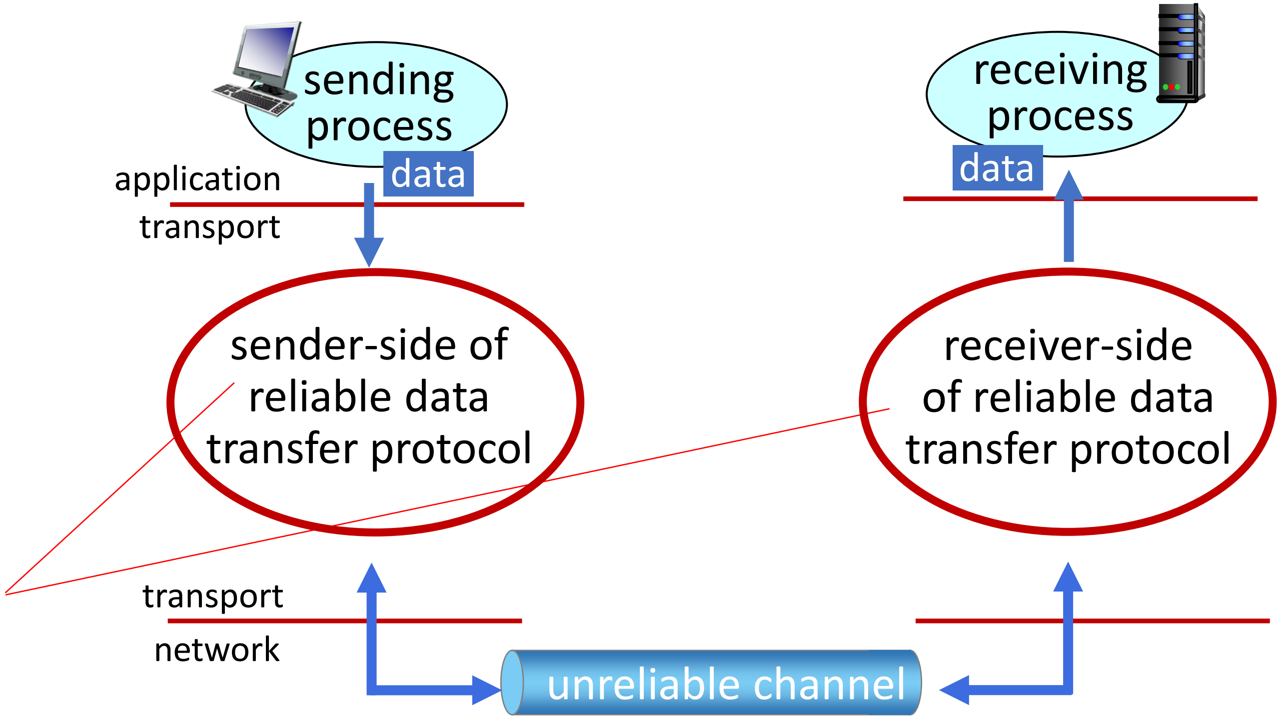
Principles of reliable data transfer

Complexity of reliable data transfer protocol will depend (strongly) on characteristics of unreliable channel (lose, corrupt, reorder data?)

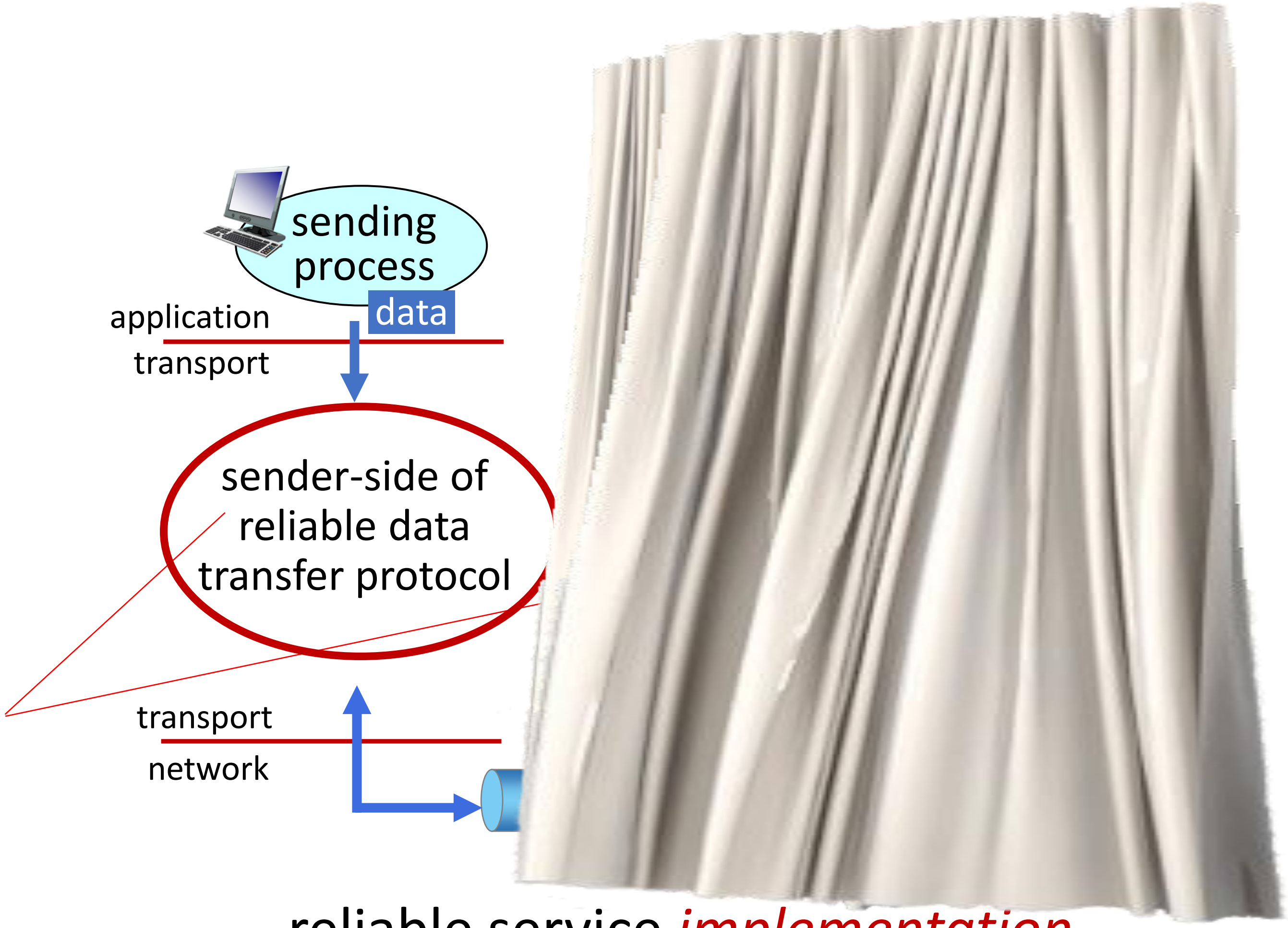


reliable service *implementation*

Principles of reliable data transfer

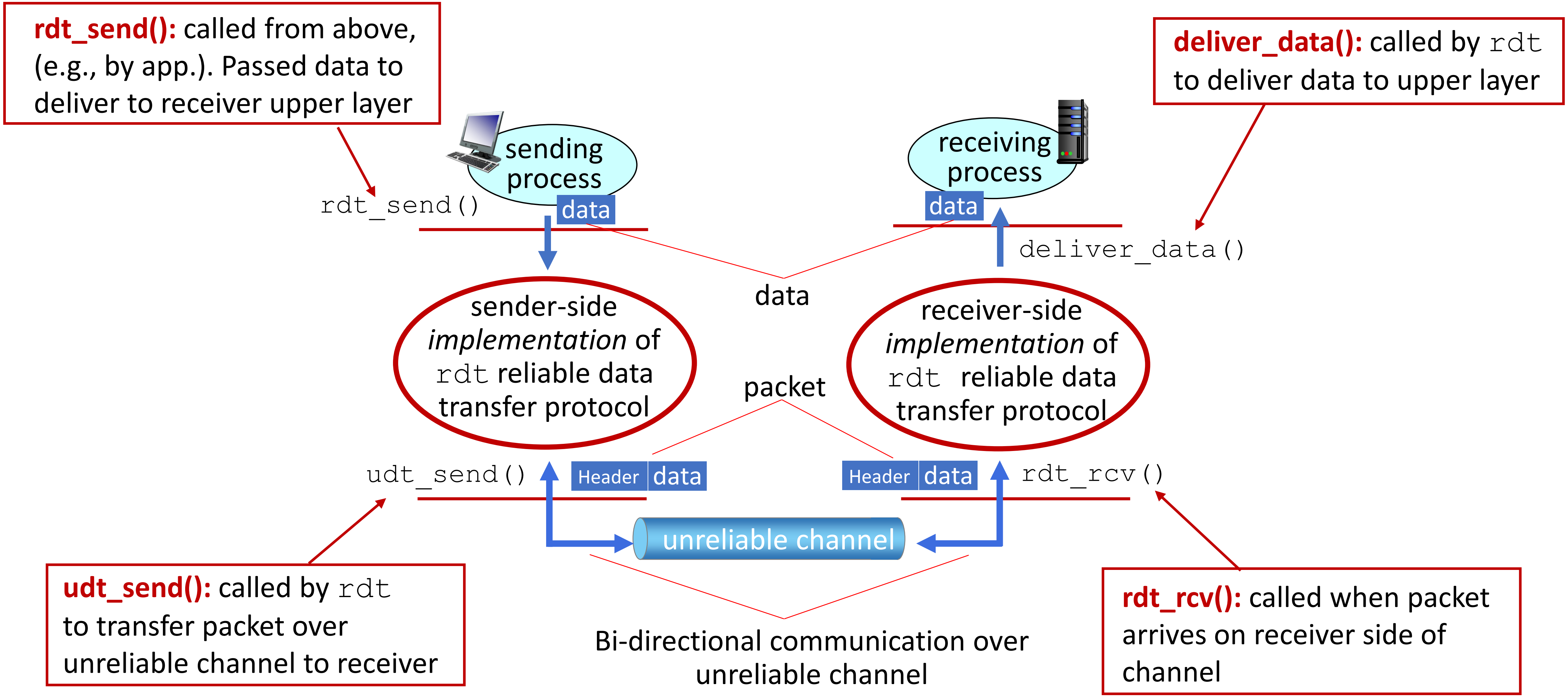
Sender, receiver do *not* know the “state” of each other, e.g., was a message received?

- unless communicated via a message



reliable service *implementation*

Reliable data transfer protocol (rdt): interfaces



rdt_send(): called from above, (e.g., by app.). Passed data to deliver to receiver upper layer

deliver_data(): called by `rdt` to deliver data to upper layer

sending process

receiving process

`rdt_send()`

data

data

`deliver_data()`

data

sender-side *implementation* of `rdt` reliable data transfer protocol

receiver-side *implementation* of `rdt` reliable data transfer protocol

packet

`udt_send()`

Header data

Header data

`rdt_rcv()`

unreliable channel

Bi-directional communication over unreliable channel

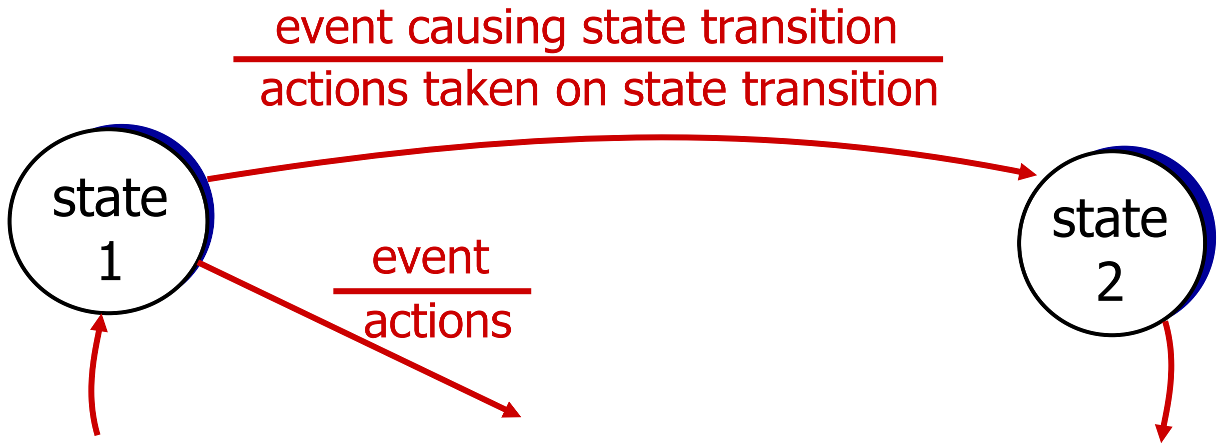
udt_send(): called by `rdt` to transfer packet over unreliable channel to receiver

rdt_rcv(): called when packet arrives on receiver side of channel

Reliable data transfer: getting started

We will:

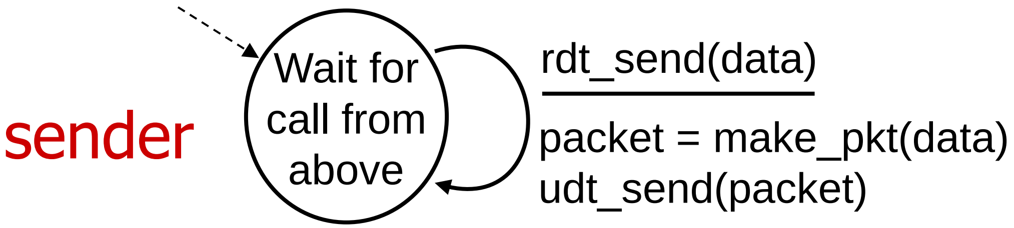
- incrementally develop sender, receiver sides of reliable data transfer protocol (`rdt`)
- consider only unidirectional data transfer
  - but control info will flow in both directions!
- use finite state machines (FSM) to specify sender, receiver

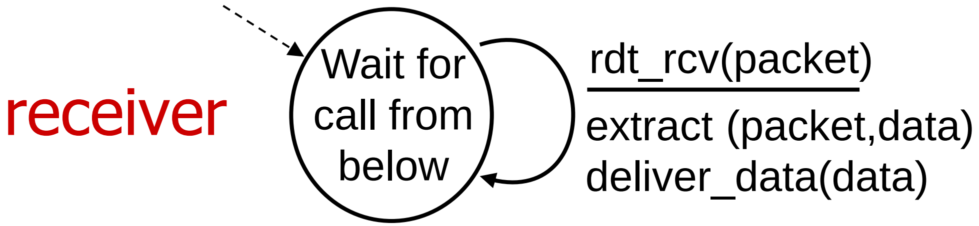


state: when in this "state" next state uniquely determined by next event

rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- *separate* FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver reads data from underlying channel





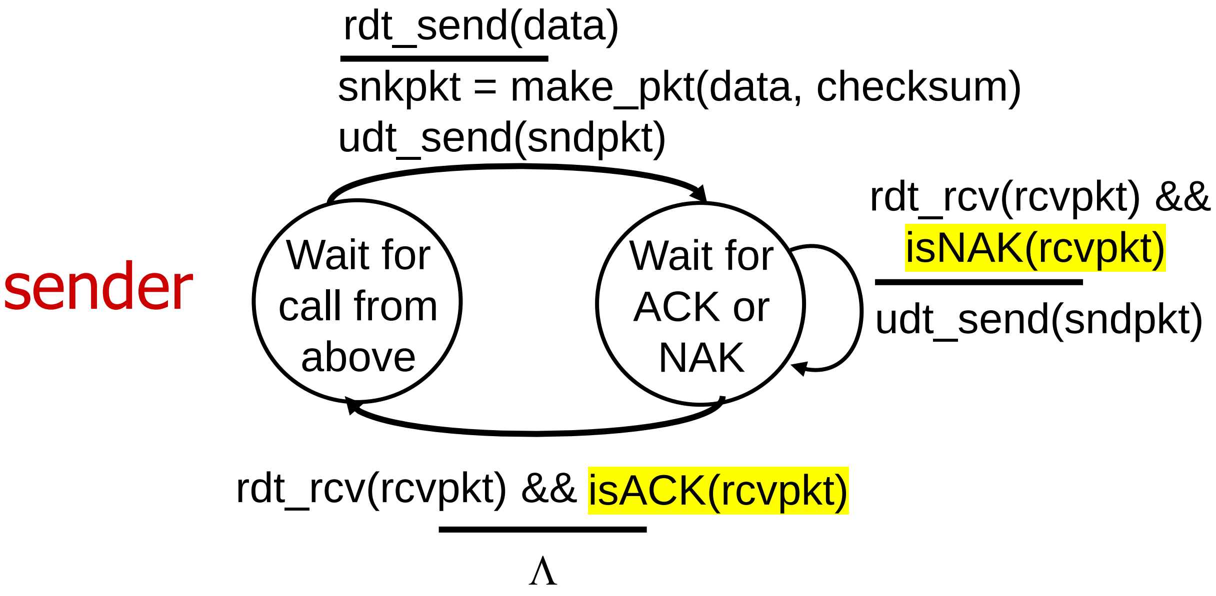
rdt2.0: channel with bit errors

- 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

stop and wait

sender sends one packet, then waits for receiver response

rdt2.0: FSM specification



sender

rdt_send(data)

snkpkt = make_pkt(data, checksum)
udt_send(sndpkt)

Wait for call from above

Wait for ACK or NAK

rdt_rcv(rcvpkt) &&
isNAK(rcvpkt)

udt_send(sndpkt)

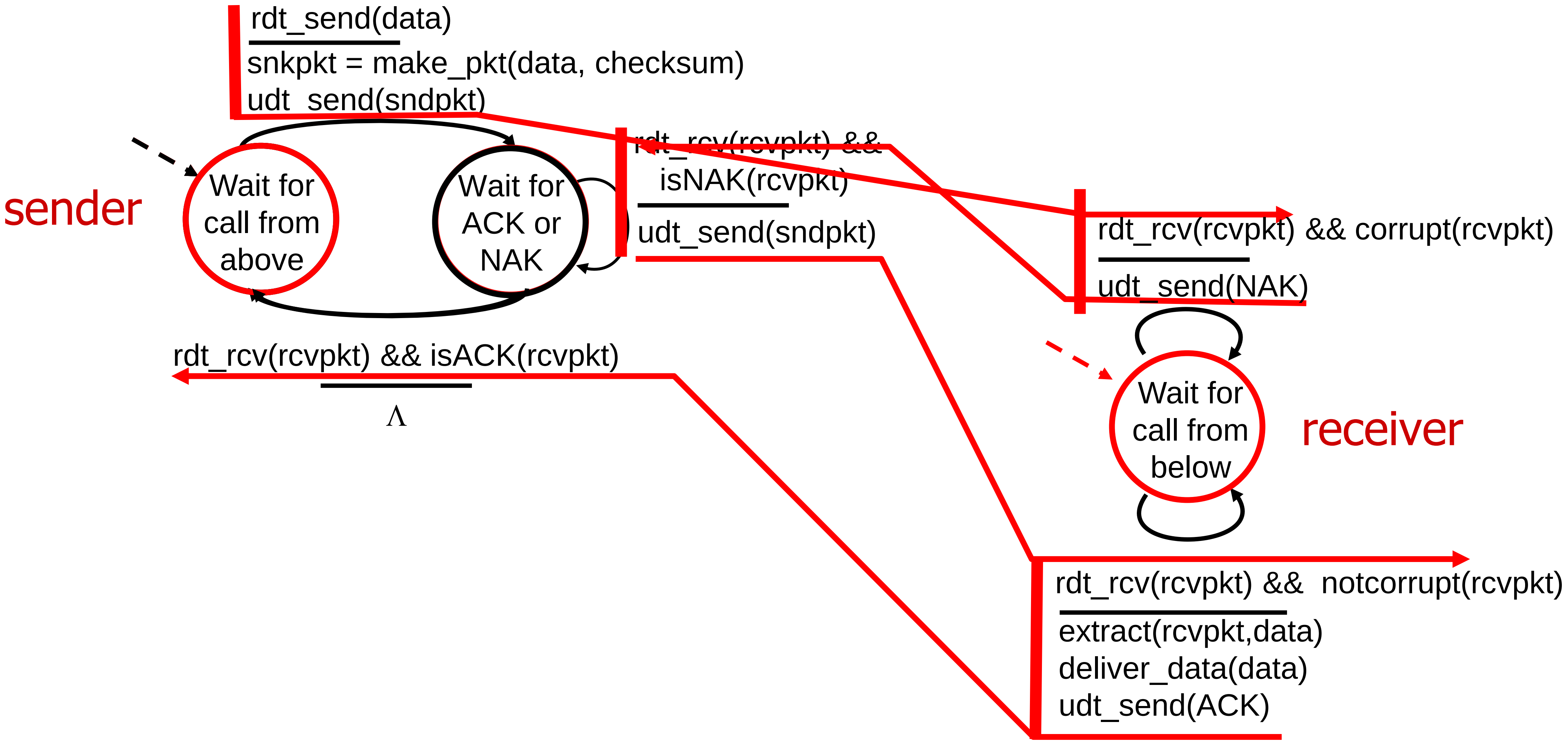
rdt_rcv(rcvpkt) && isACK(rcvpkt)

Λ

Note: "state" of receiver (did the receiver get my message correctly?) isn't known to sender unless somehow communicated from receiver to sender

- that's why we need a protocol!

rdt2.0: corrupted packet scenario



rdt_send(data)
snkpkt = make_pkt(data, checksum)
udt_send(sndpkt)

sender

Wait for call from above

Wait for ACK or NAK

rdt_rcv(rcvpkt) &&
isNAK(rcvpkt)
udt_send(sndpkt)

rdt_rcv(rcvpkt) && isACK(rcvpkt)
Λ

rdt_rcv(rcvpkt) && corrupt(rcvpkt)
udt_send(NAK)

Wait for call from below

receiver

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
extract(rcvpkt,data)
deliver_data(data)
udt_send(ACK)

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:

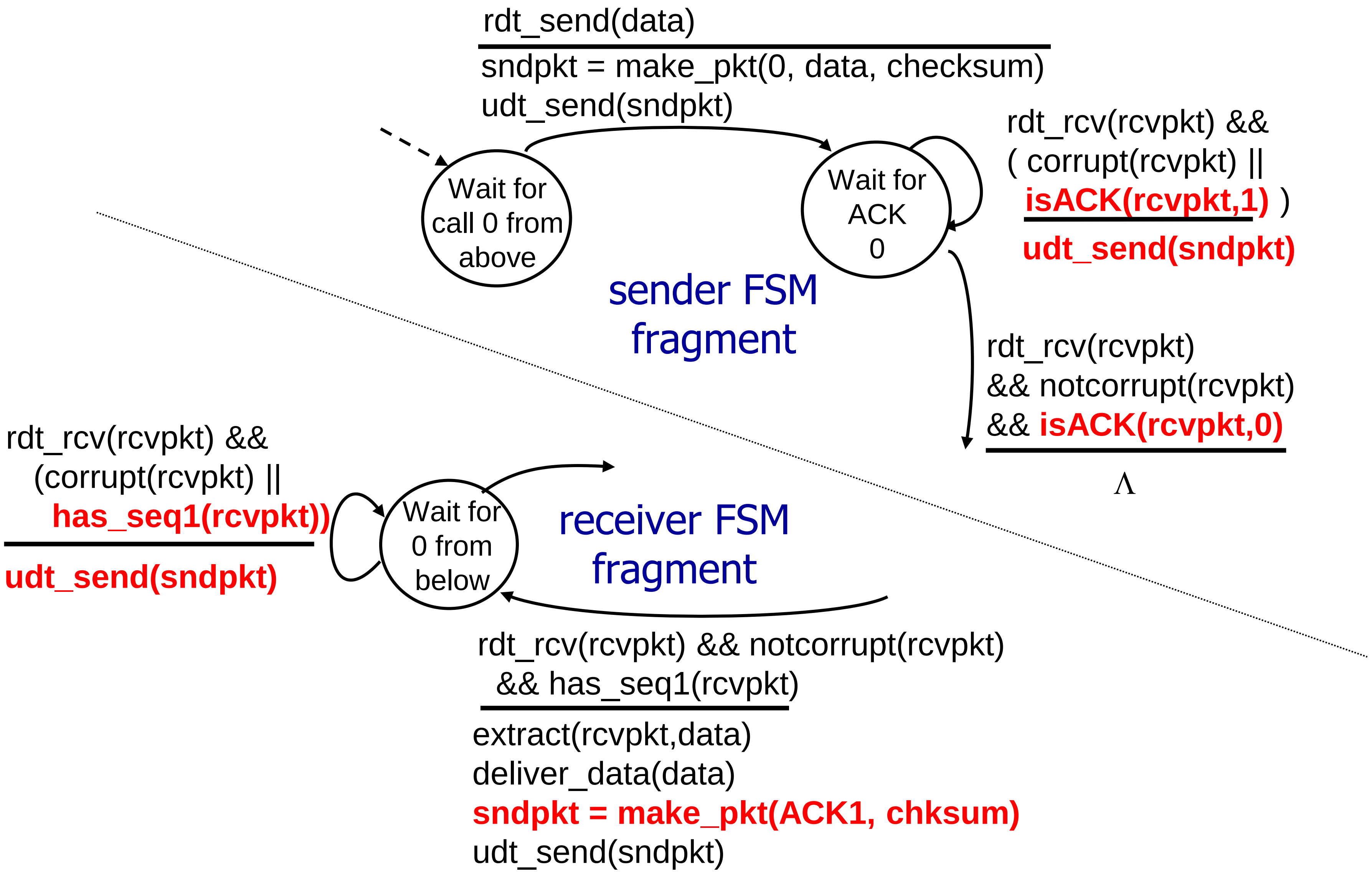
- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

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*

As we will see, TCP uses this approach to be NAK-free

rdt2.2: sender, receiver fragments



rdt_send(data)

sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,1))

udt_send(sndpkt)

sender FSM fragment

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& **isACK(rcvpkt,0)**

Λ

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
has_seq1(rcvpkt))

udt_send(sndpkt)

receiver FSM fragment

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq1(rcvpkt)

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK1, chksum)
udt_send(sndpkt)

rdt3.0: channels with errors *and* loss

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

- checksum, sequence #s, ACKs, retransmissions will be of help … but not quite enough

Q: How do *humans* handle lost sender-to-receiver words in conversation?

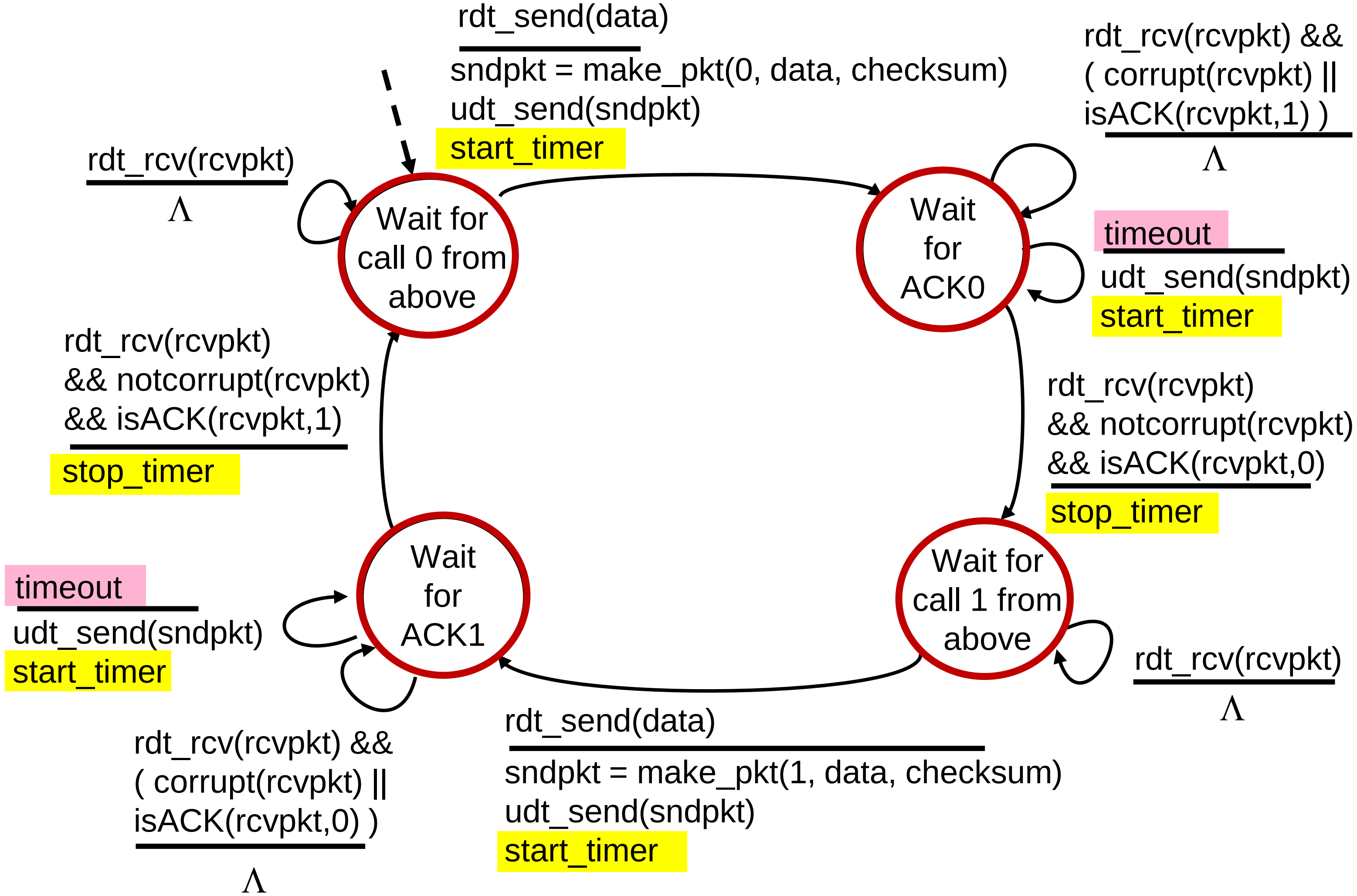
rdt3.0: channels with errors *and* loss

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 packet being ACKed
- use countdown timer to interrupt after "reasonable" amount of time

timeout

rdt3.0 sender



rdt_send(data)
sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)
Λ

Wait for call 0 from above

Wait for ACK0

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,1))
Λ

timeout
udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt,1)
stop_timer

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt,0)
stop_timer

Wait for ACK1

Wait for call 1 from above

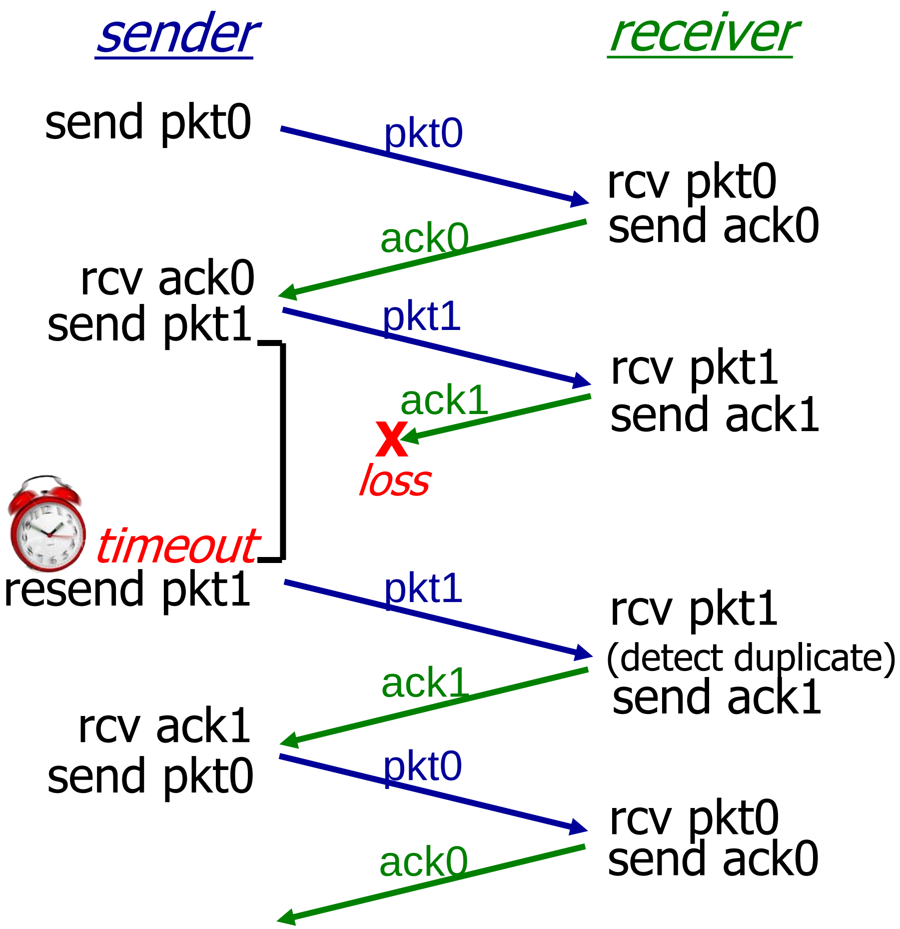
timeout
udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)
Λ

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,0))
Λ

rdt_send(data)
sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)
start_timer

rdt3.0 in action



sender

receiver

send pkt0

pkt0

rcv pkt0
send ack0

ack0

rcv ack0
send pkt1

pkt1

rcv pkt1
send ack1

ack1

X
loss

timeout
resend pkt1

pkt1

rcv pkt1
(detect duplicate)
send ack1

ack1

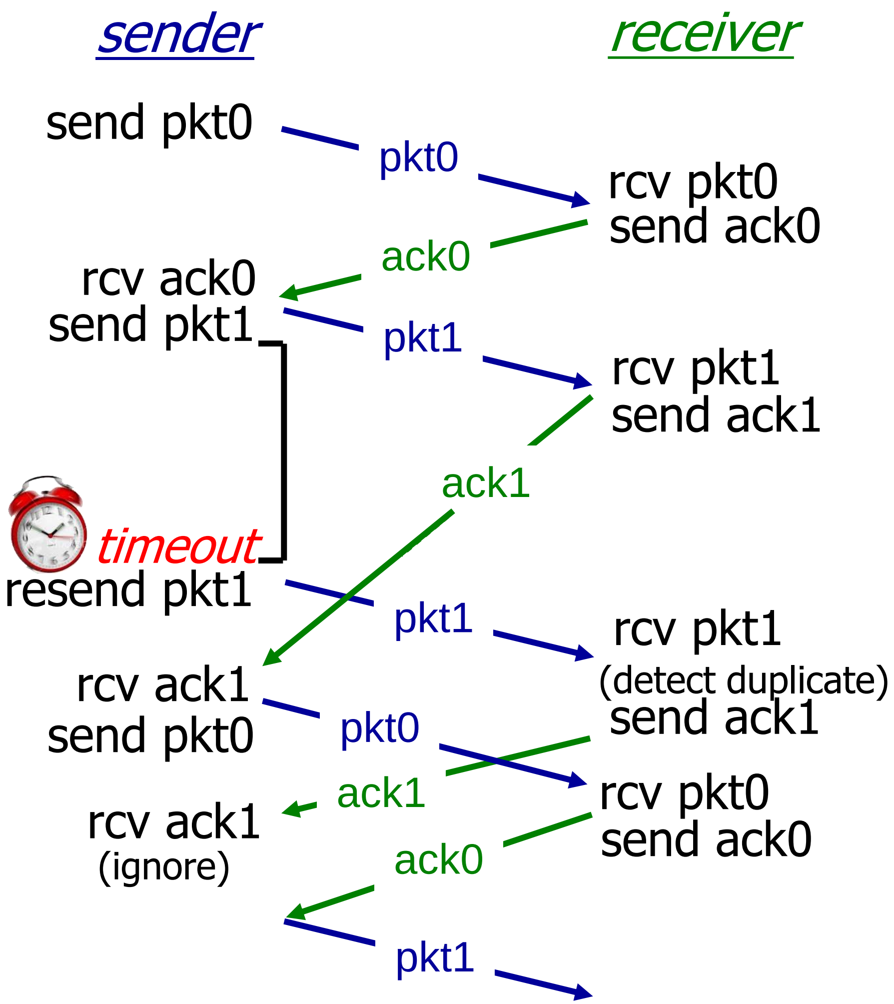
rcv ack1
send pkt0

pkt0

rcv pkt0
send ack0

ack0

(c) ACK loss



sender

receiver

send pkt0

pkt0

rcv pkt0
send ack0

ack0

rcv ack0
send pkt1

pkt1

rcv pkt1
send ack1

ack1

timeout
resend pkt1

pkt1

rcv pkt1
(detect duplicate)
send ack1

rcv ack1
send pkt0

pkt0

ack1

rcv pkt0
send ack0

rcv ack1
(ignore)

ack0

pkt1

(d) premature timeout/ delayed ACK

Performance of rdt3.0 (stop-and-wait)

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

- example: 1 Gbps link, 15 ms prop. delay, 8000 bit packet
  - time to transmit packet into channel:

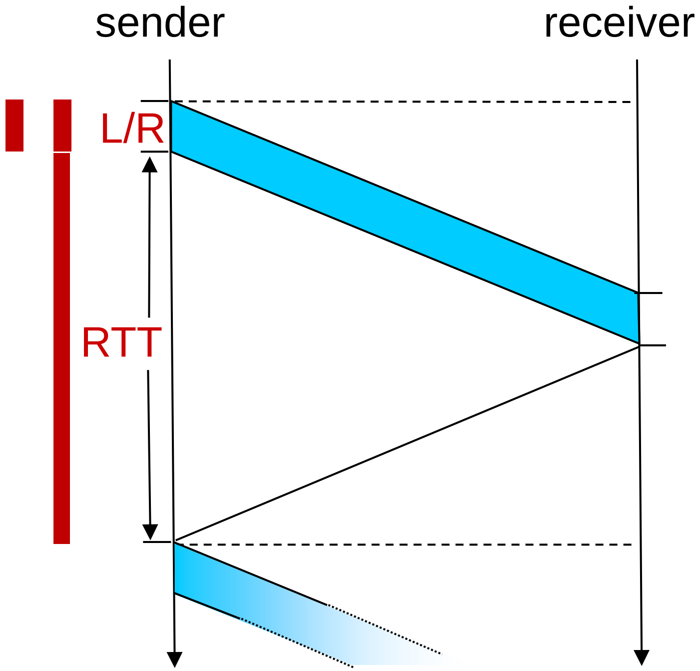
$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs}$$

rdt3.0: stop-and-wait operation

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

$$= \frac{.008}{30.008}$$

$$= 0.00027$$

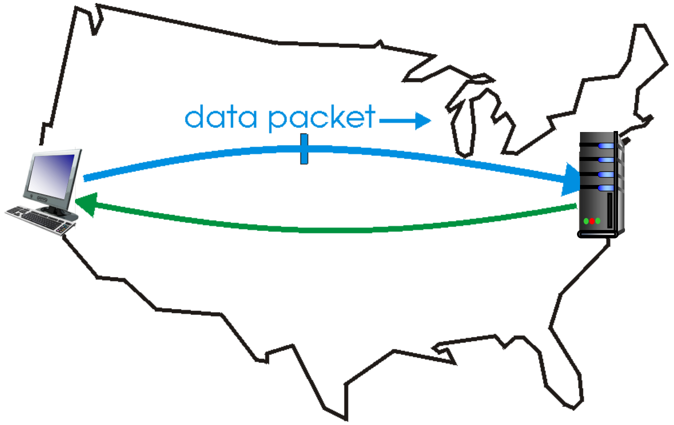


- rdt 3.0 protocol performance stinks!
- Protocol limits performance of underlying infrastructure (channel)

rdt3.0: pipelined protocols operation

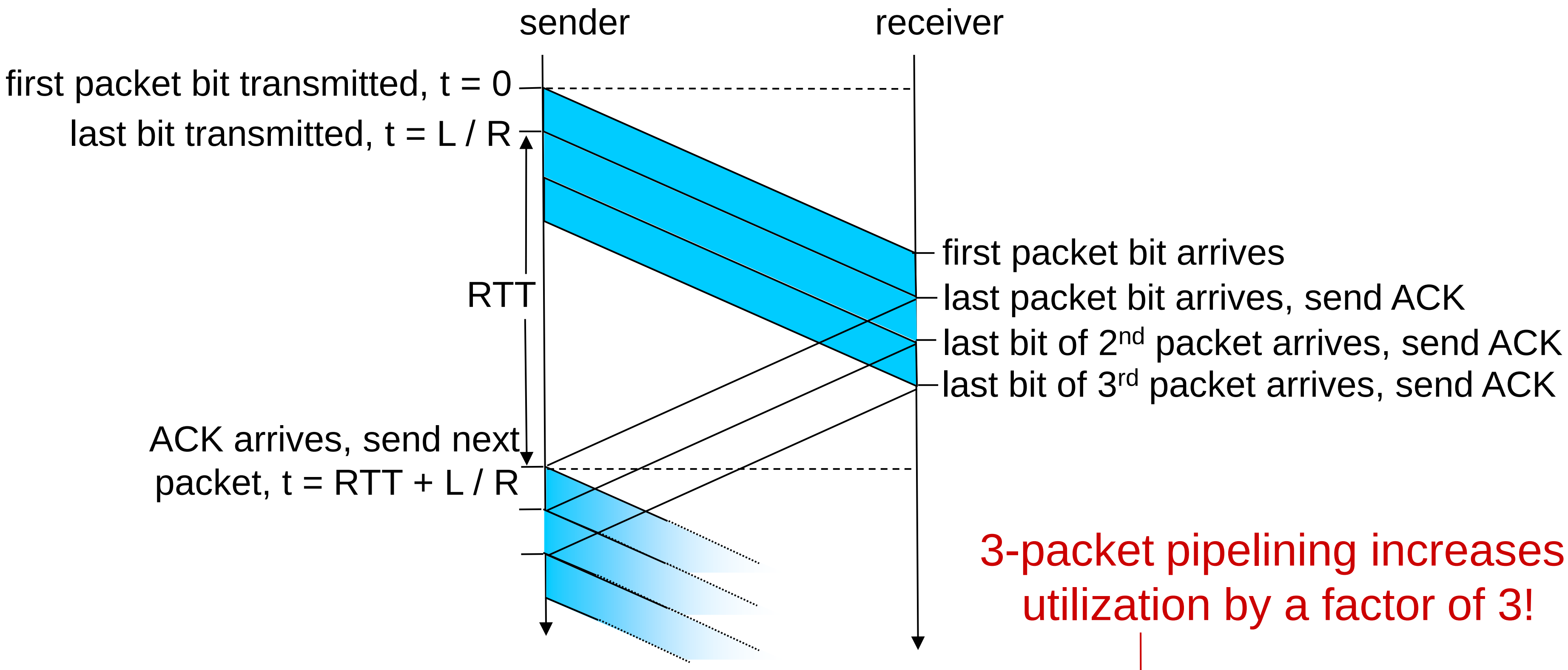
pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged packets

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



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

Pipelining: increased utilization

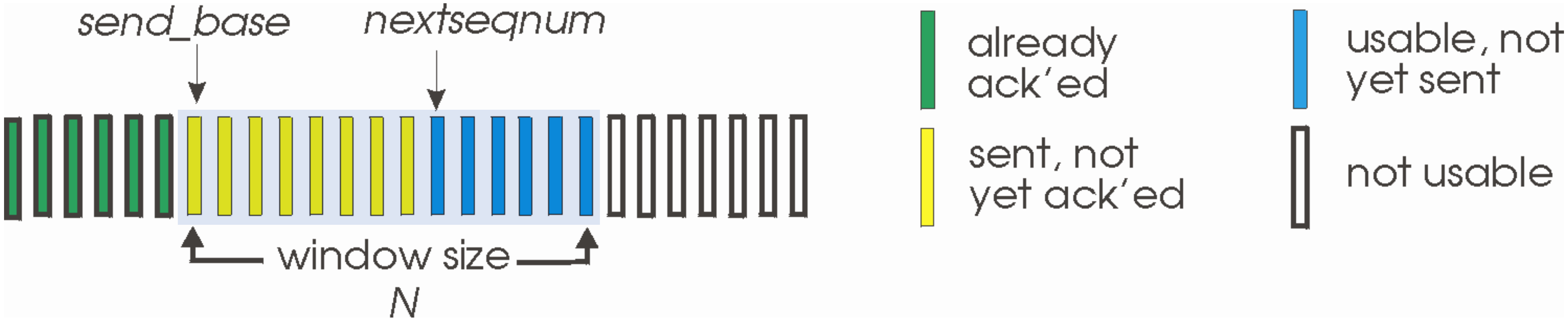


3-packet pipelining increases utilization by a factor of 3!

$$U_{\text{sender}} = \frac{3L / R}{RTT + L / R} = \frac{.0024}{30.008} = 0.00081$$

Go-Back-N: sender

- sender: “window” of up to N, consecutive transmitted but unACKed pkts
  - k-bit seq # in pkt header

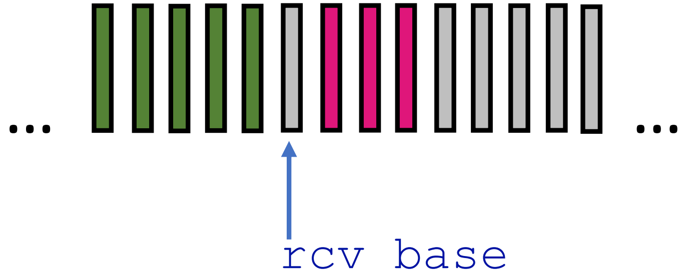


- *cumulative ACK:* ACK(*n*): ACKs all packets up to, including seq # *n*
  - on receiving ACK(*n*): move window forward to begin at *n+1*
- timer for oldest in-flight packet
- *timeout(n):* retransmit packet n and all higher seq # packets in window

Go-Back-N: receiver

- ACK-only: always send ACK for correctly-received packet so far, with highest *in-order* seq #
  - may generate duplicate ACKs
  - need only remember `rcv_base`
- on receipt of out-of-order packet:
  - can discard (don't buffer) or buffer: an implementation decision
  - re-ACK pkt with highest in-order seq #

Receiver view of sequence number space:

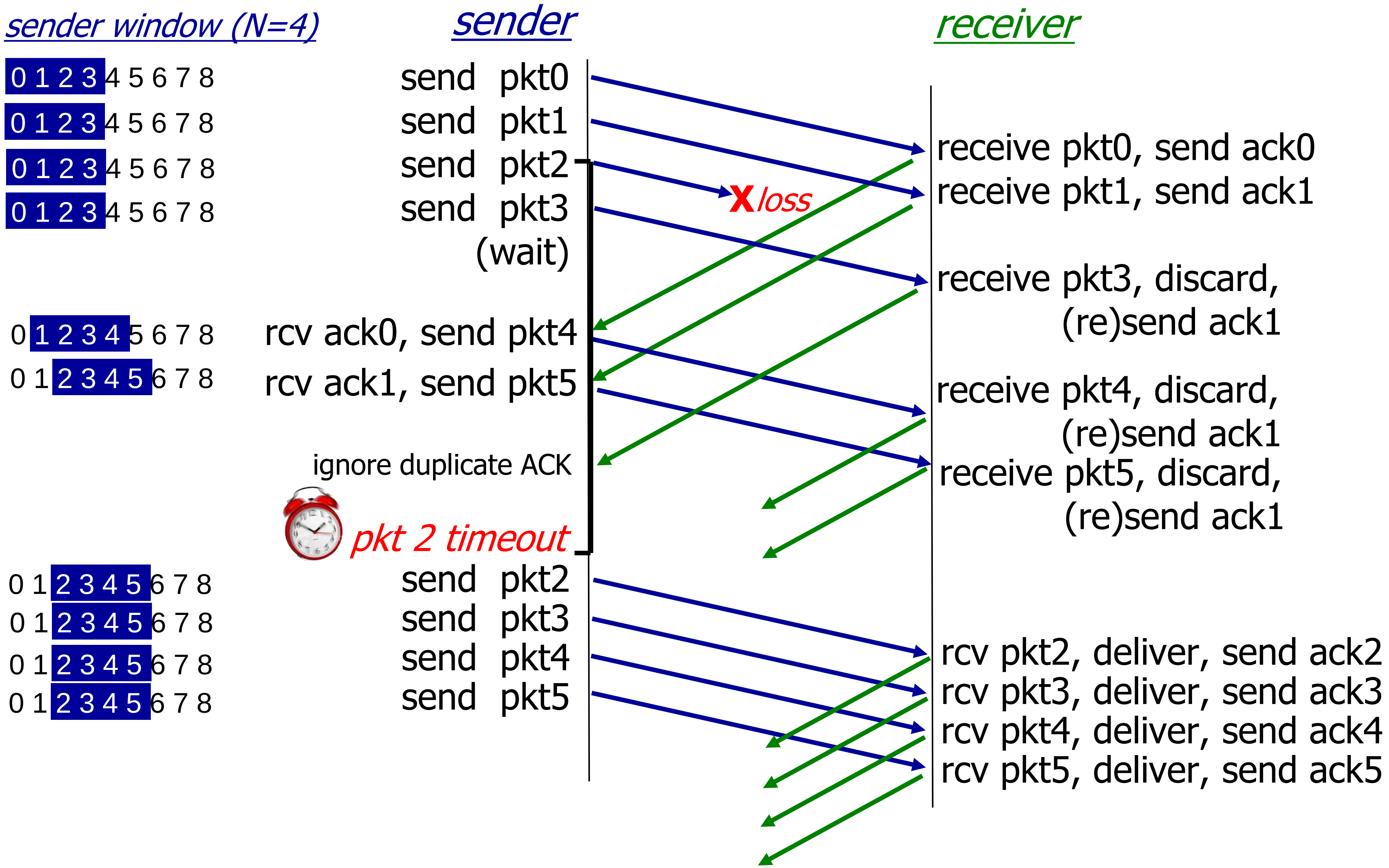


🟩 received and ACKed

🟥 Out-of-order: received but not ACKed

⬜ Not received

Go-Back-N in action



sender window (N=4)

sender

receiver

0 1 2 3 4 5 6 7 8 send pkt0

0 1 2 3 4 5 6 7 8 send pkt1

0 1 2 3 4 5 6 7 8 send pkt2

0 1 2 3 4 5 6 7 8 send pkt3

(wait)

X *loss*

receive pkt0, send ack0

receive pkt1, send ack1

receive pkt3, discard,
(re)send ack1

0 1 2 3 4 5 6 7 8 rcv ack0, send pkt4

0 1 2 3 4 5 6 7 8 rcv ack1, send pkt5

receive pkt4, discard,
(re)send ack1

receive pkt5, discard,
(re)send ack1

ignore duplicate ACK

pkt 2 timeout

0 1 2 3 4 5 6 7 8 send pkt2

0 1 2 3 4 5 6 7 8 send pkt3

0 1 2 3 4 5 6 7 8 send pkt4

0 1 2 3 4 5 6 7 8 send pkt5

rcv pkt2, deliver, send ack2

rcv pkt3, deliver, send ack3

rcv pkt4, deliver, send ack4

rcv pkt5, deliver, send ack5

Selective repeat: the approach

- *pipelining*: *multiple* packets in flight
- *receiver individually ACKs* all correctly received packets
  - buffers packets, as needed, for in-order delivery to upper layer
- sender:
  - maintains (conceptually) a timer for each unACKed pkt
    - timeout: retransmits single unACKed packet associated with timeout
  - maintains (conceptually) "window" over *N* consecutive seq #s
    - limits pipelined, "in flight" packets to be within this window

Selective repeat: sender and receiver

sender

data from above:

- if next available seq # in window, send packet

timeout(*n*):

- resend packet *n*, restart timer

ACK(*n*) in [sendbase,sendbase+N-1]:

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

receiver

packet *n* in [rcvbase, rcvbase+N-1]

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

packet *n* in [rcvbase-N,rcvbase-1]

- ACK(*n*)

otherwise:

- ignore

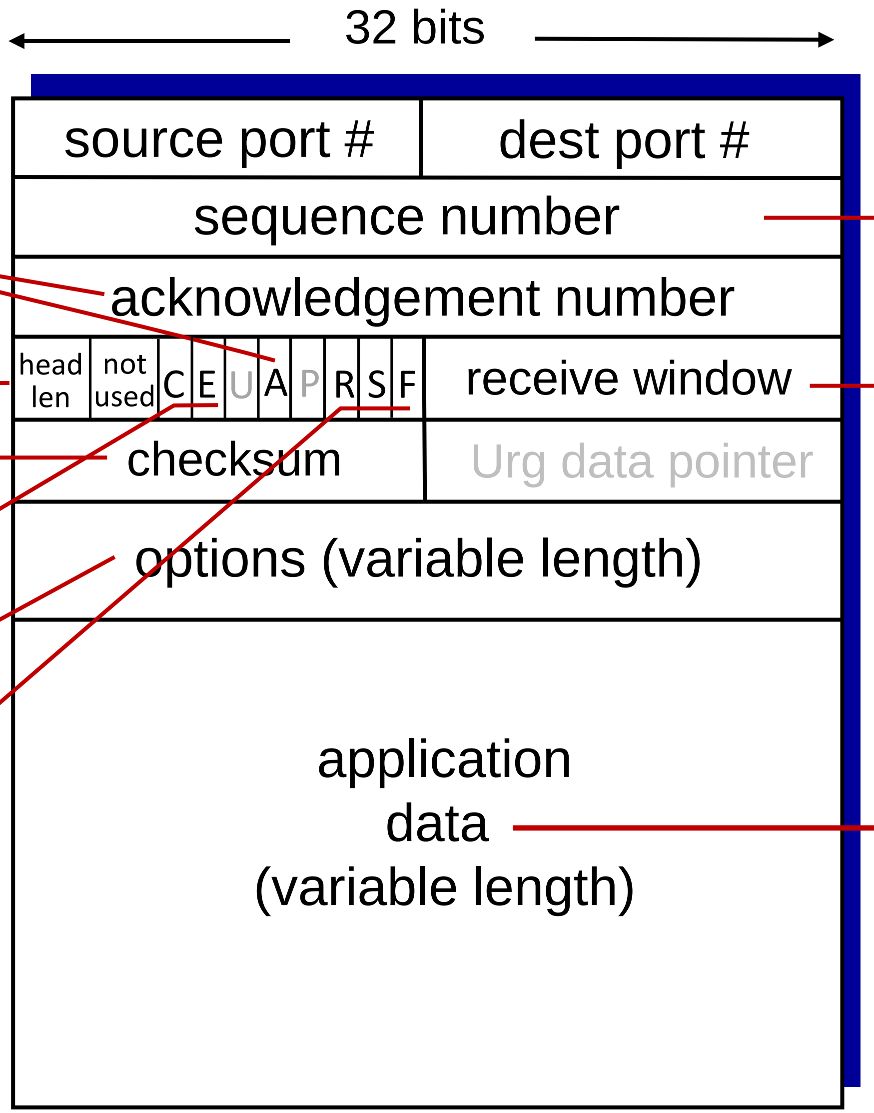
Transport layer: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- Principles of congestion control
- TCP congestion control

TCP: overview RFCs: 793,1122, 2018, 5681, 7323

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - no “message boundaries"
- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- cumulative ACKs
- pipelining:
  - TCP congestion and flow control set window size
- connection-oriented:
  - handshaking (exchange of control messages) initializes sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure



ACK: seq # of next expected byte; A bit: this is an ACK

length (of TCP header)

Internet **checksum**

C, E: congestion notification

TCP **options**

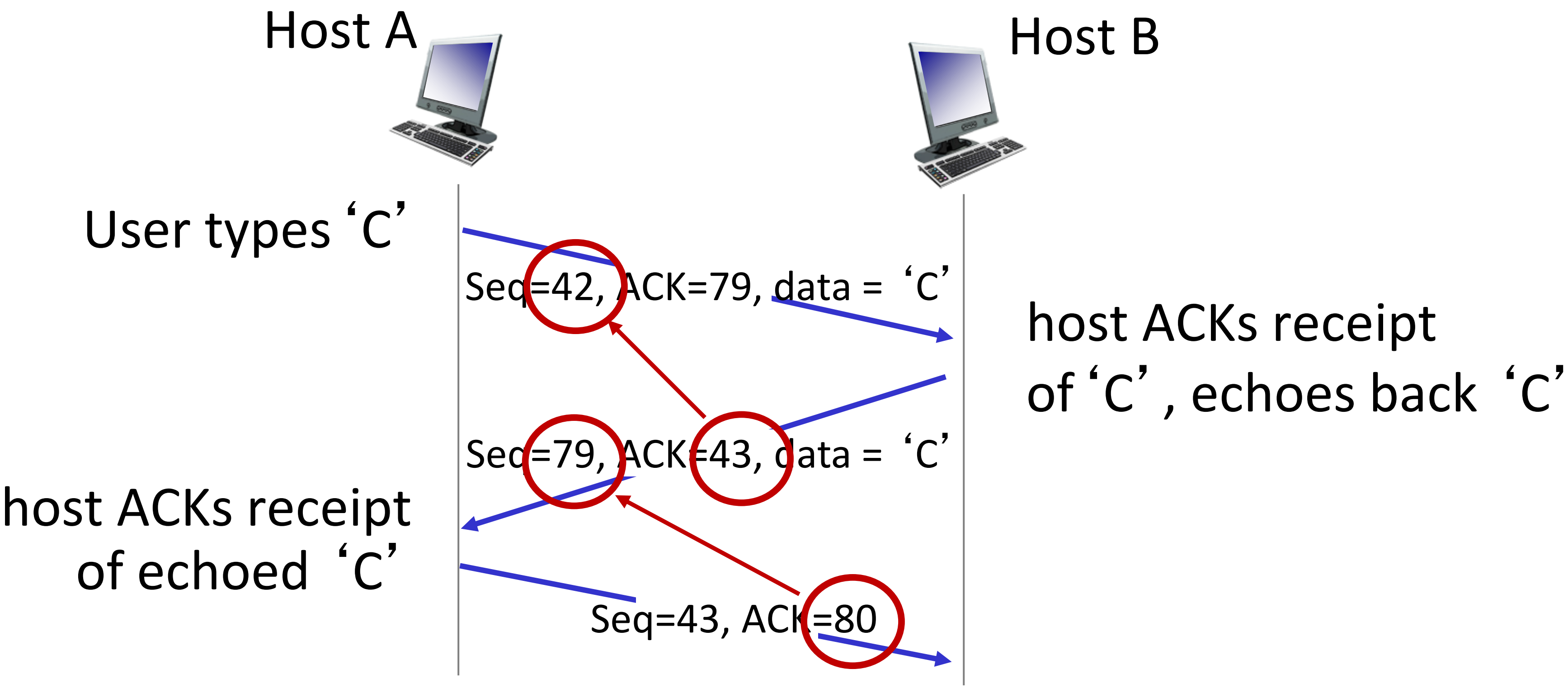
RST, SYN, FIN: connection management

segment seq #: counting bytes of data into bytestream (not segments!)

flow control: # bytes receiver willing to accept

data sent by application into TCP socket

TCP sequence numbers, ACKs



simple telnet scenario

TCP round trip time, timeout

Q: how to set TCP timeout value?

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

Q: how to estimate RTT?

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

TCP round trip time, timeout

- timeout interval: **`EstimatedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`**: want a larger safety margin

```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

estimated RTT

"safety margin"

- **`DevRTT`**: EWMA of **`SampleRTT`** deviation from **`EstimatedRTT`**:

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

(typically, β = 0.25)

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

TCP Sender (simplified)

event: data received from application

- create segment with seq #
- seq # is byte-stream number of first data byte in segment
- start timer if not already running
  - think of timer as for oldest unACKed segment
  - expiration interval: **`TimeOutInterval`**

event: timeout

- retransmit segment that caused timeout
- restart timer

event: ACK received

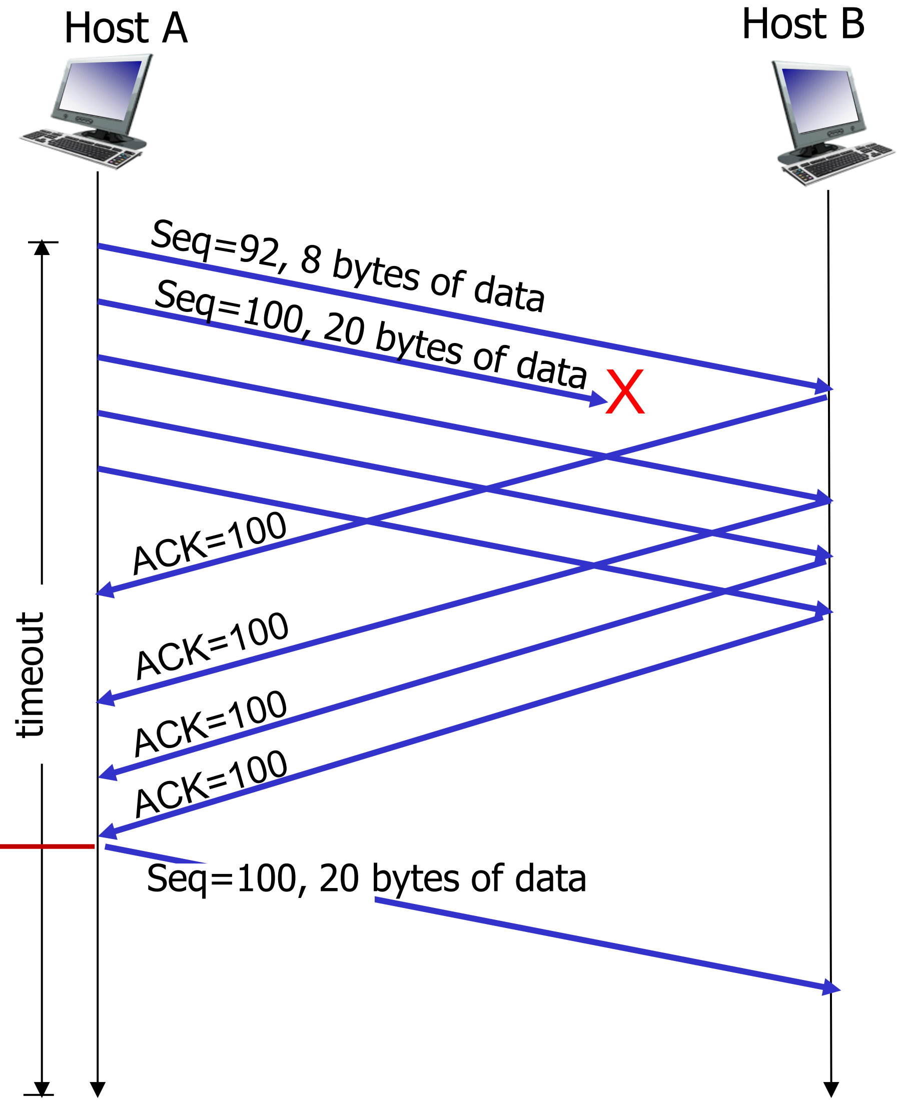
- if ACK acknowledges previously unACKed segments
  - update what is known to be ACKed
  - start timer if there are still unACKed segments

TCP fast retransmit

TCP fast retransmit

if sender receives 3 additional ACKs for same data ("triple duplicate ACKs"), resend unACKed segment with smallest seq #

- likely that unACKed segment lost, so don't wait for timeout



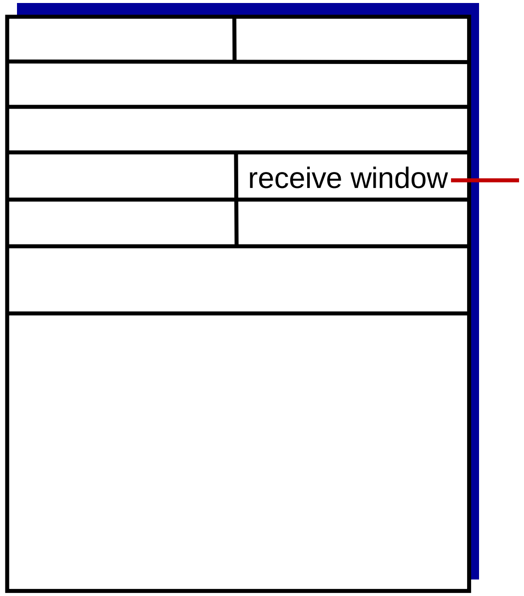
Receipt of three duplicate ACKs indicates 3 segments received after a missing segment – lost segment is likely. So retransmit!

Transport layer: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- Principles of congestion control
- TCP congestion control

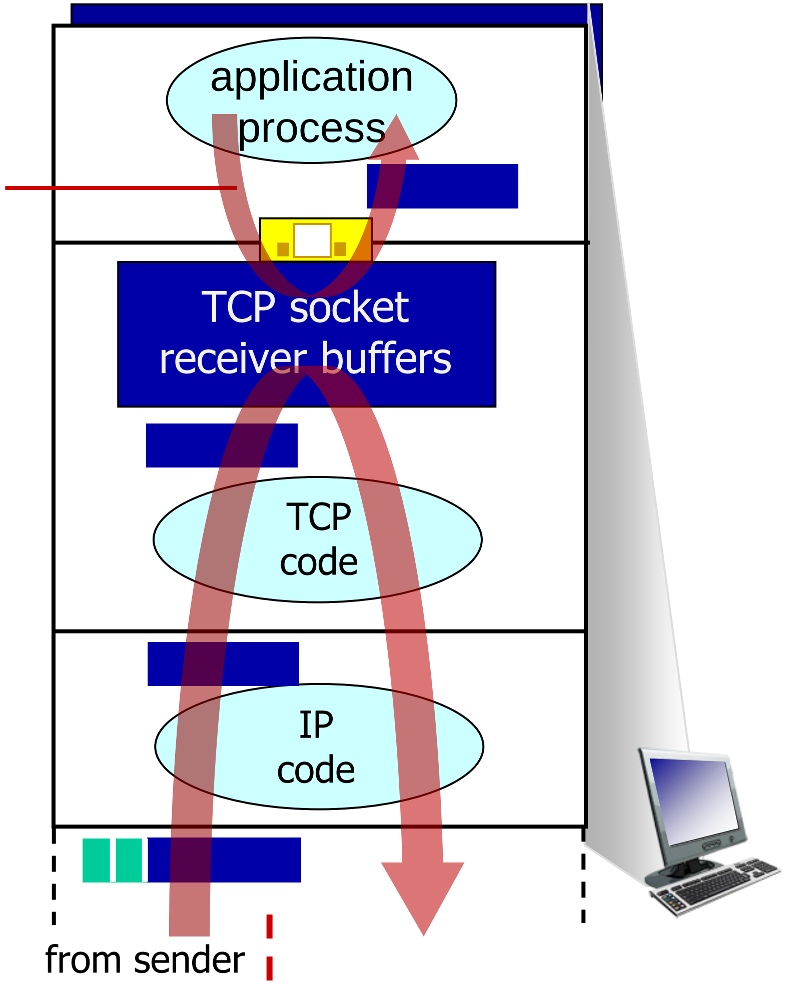
TCP flow control

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



flow control: # bytes receiver willing to accept

Application removing data from TCP socket buffers



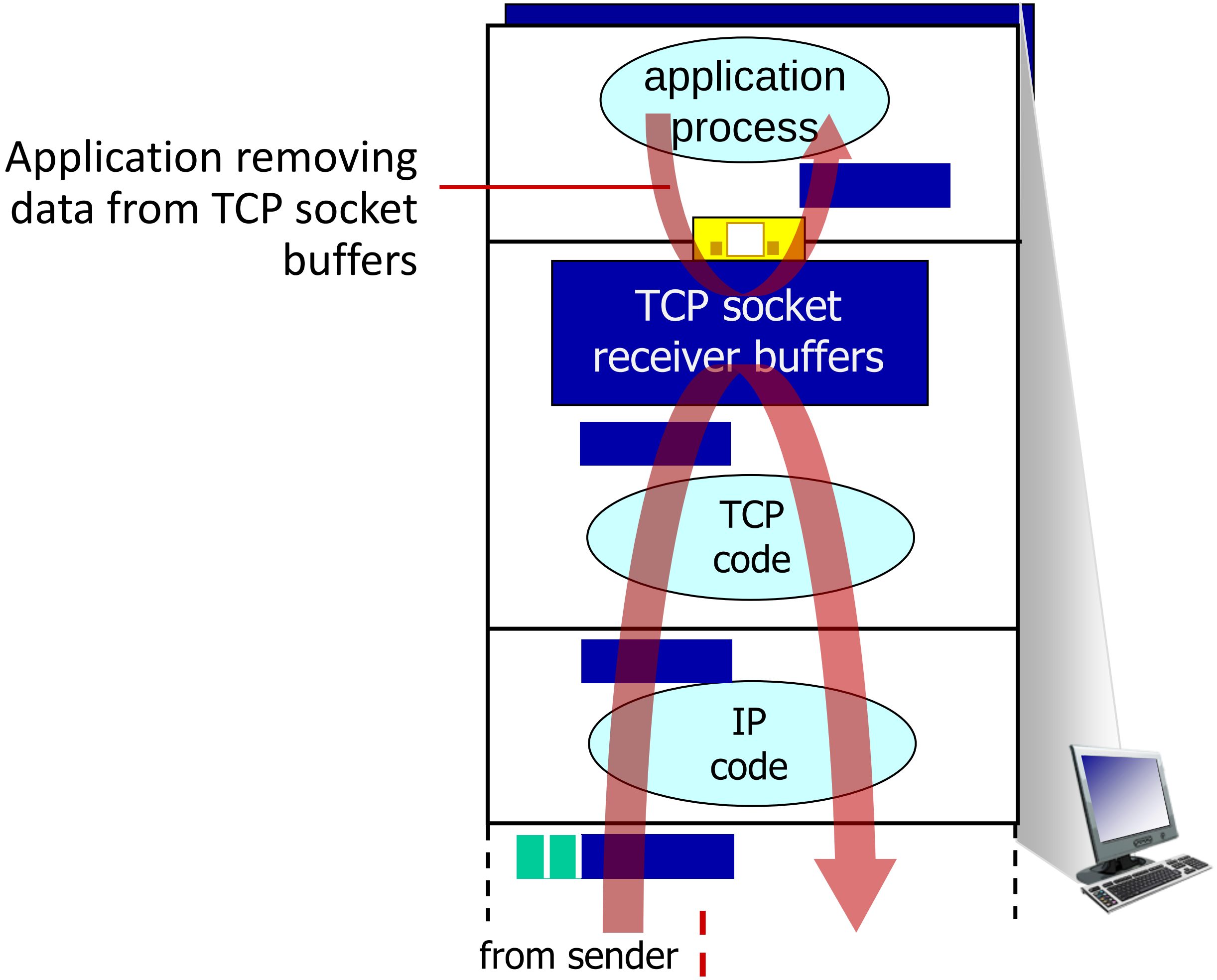
receiver protocol stack

TCP flow control

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

flow control

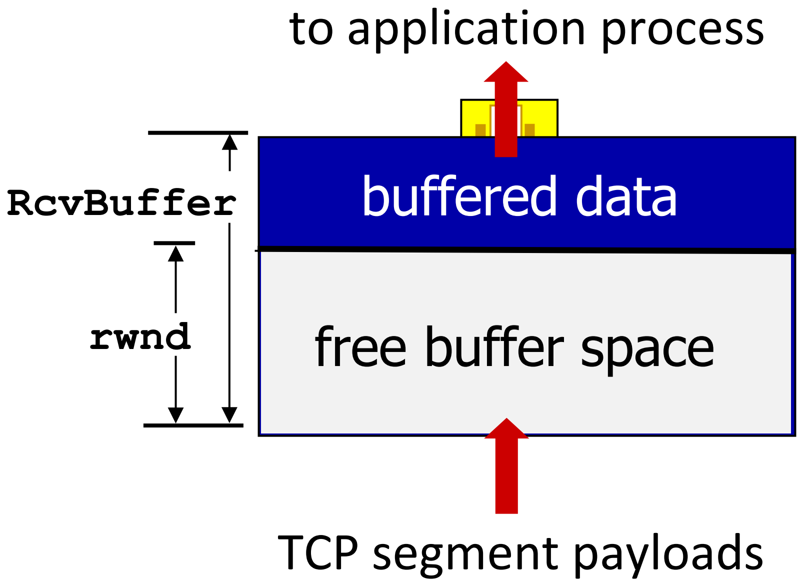
receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast



receiver protocol stack

TCP flow control

- TCP receiver "advertises" free buffer space in **`rwnd`** field in TCP header
  - **`RcvBuffer`** size set via socket options (typical default is 4096 bytes)
  - many operating systems auto-adjust **`RcvBuffer`**
- sender limits amount of unACKed ("in-flight") data to received **`rwnd`**
- guarantees receive buffer will not overflow



TCP receiver-side buffering

TCP flow control

- TCP receiver "advertises" free buffer space in **`rwnd`** field in TCP header
  - **`RcvBuffer`** size set via socket options (typical default is 4096 bytes)
  - many operating systems auto-adjust **`RcvBuffer`**
- sender limits amount of unACKed ("in-flight") data to received **`rwnd`**
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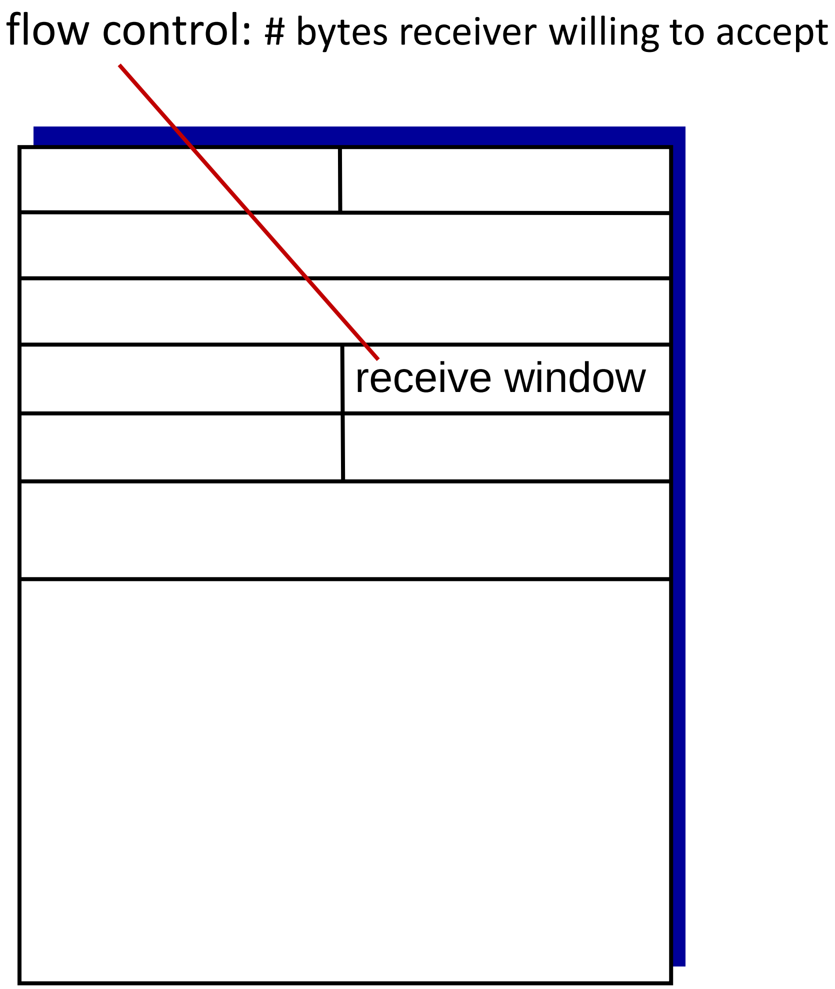
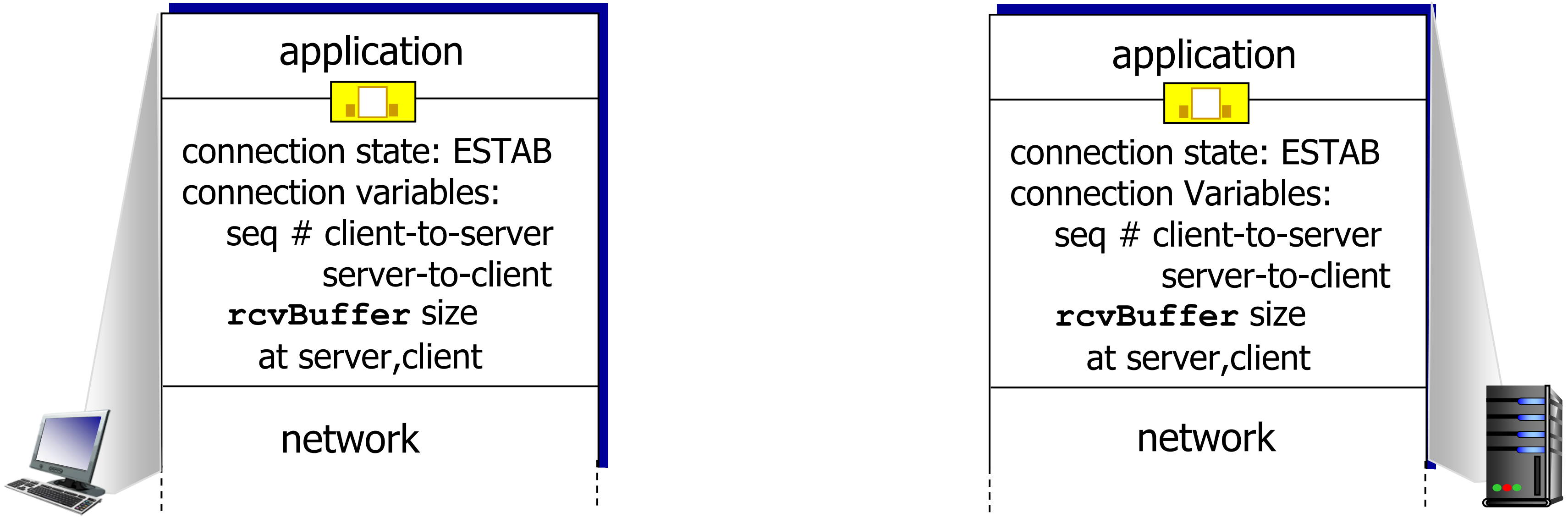


TCP segment format

TCP connection management

before exchanging data, sender/receiver "handshake":

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters (e.g., starting seq #s)



application

connection state: ESTAB
connection variables:
seq # client-to-server
server-to-client
`rcvBuffer` size
at server,client

network

application

connection state: ESTAB
connection Variables:
seq # client-to-server
server-to-client
`rcvBuffer` size
at server,client

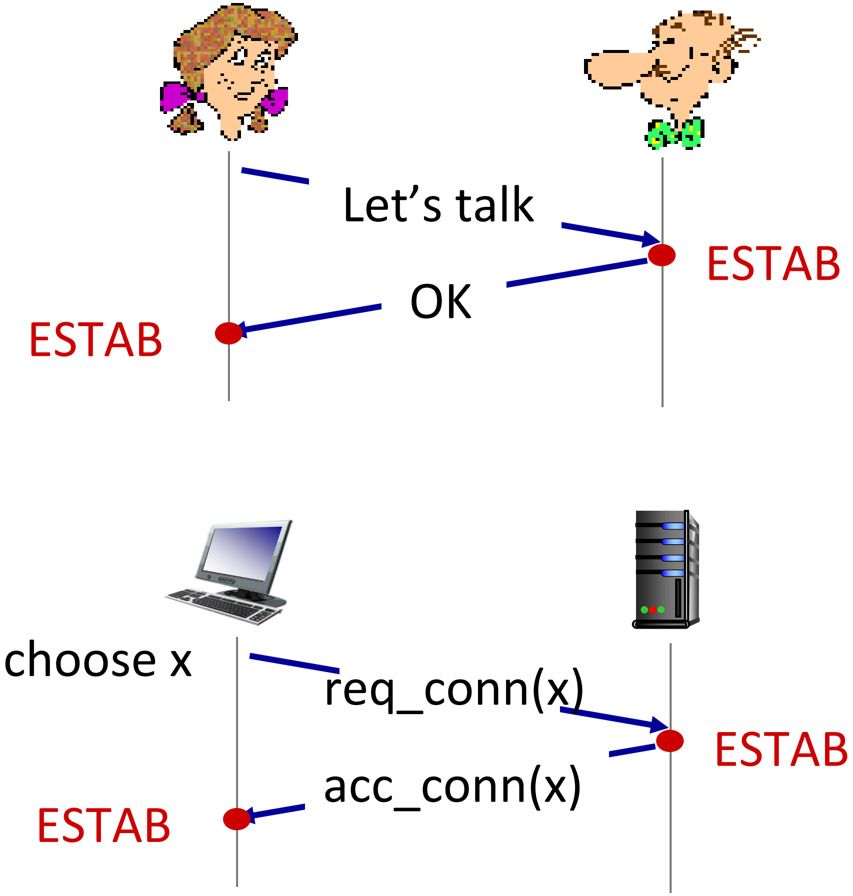
network

```
Socket clientSocket =
  newSocket("hostname","port number");
```

```
Socket connectionSocket =
  welcomeSocket.accept();
```

Agreeing to establish a connection

2-way handshake:



Q: will 2-way handshake always work in network?

- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can’t “see” other side

Closing a TCP connection

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

Transport layer: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
- **Principles of congestion control**
- TCP congestion control
- Evolution of transport-layer functionality

Principles of congestion control

Congestion:

- informally: "too many sources sending too much data too fast for *network* to handle"
- manifestations:
  - long delays (queueing in router buffers)
  - packet loss (buffer overflow at routers)
- different from flow control!
- a top-10 problem!

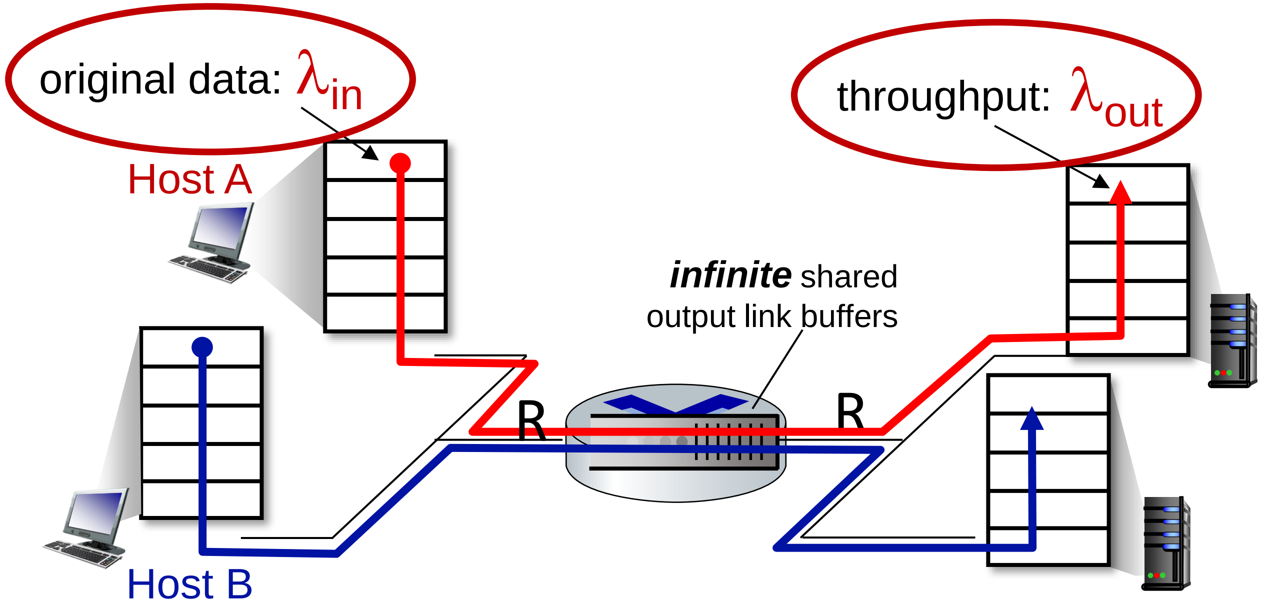
congestion control: too many senders, sending too fast

flow control: one sender too fast for one receiver

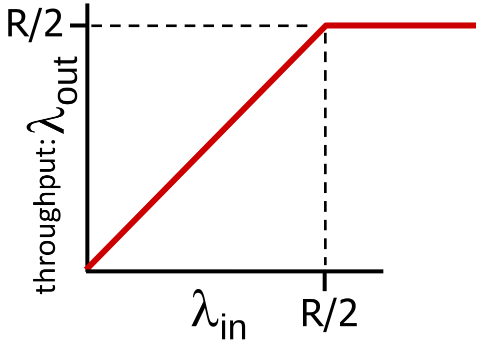
Causes/costs of congestion: scenario 1

Simplest scenario:

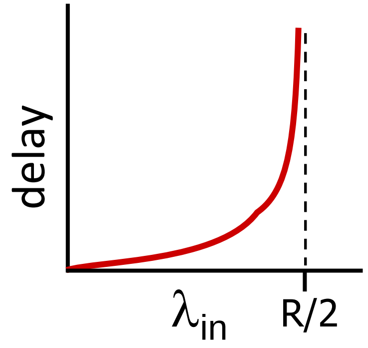
- one router, infinite buffers
- input, output link capacity: R
- two flows
- no retransmissions needed



Q: What happens as arrival rate λ_{in} approaches R/2?



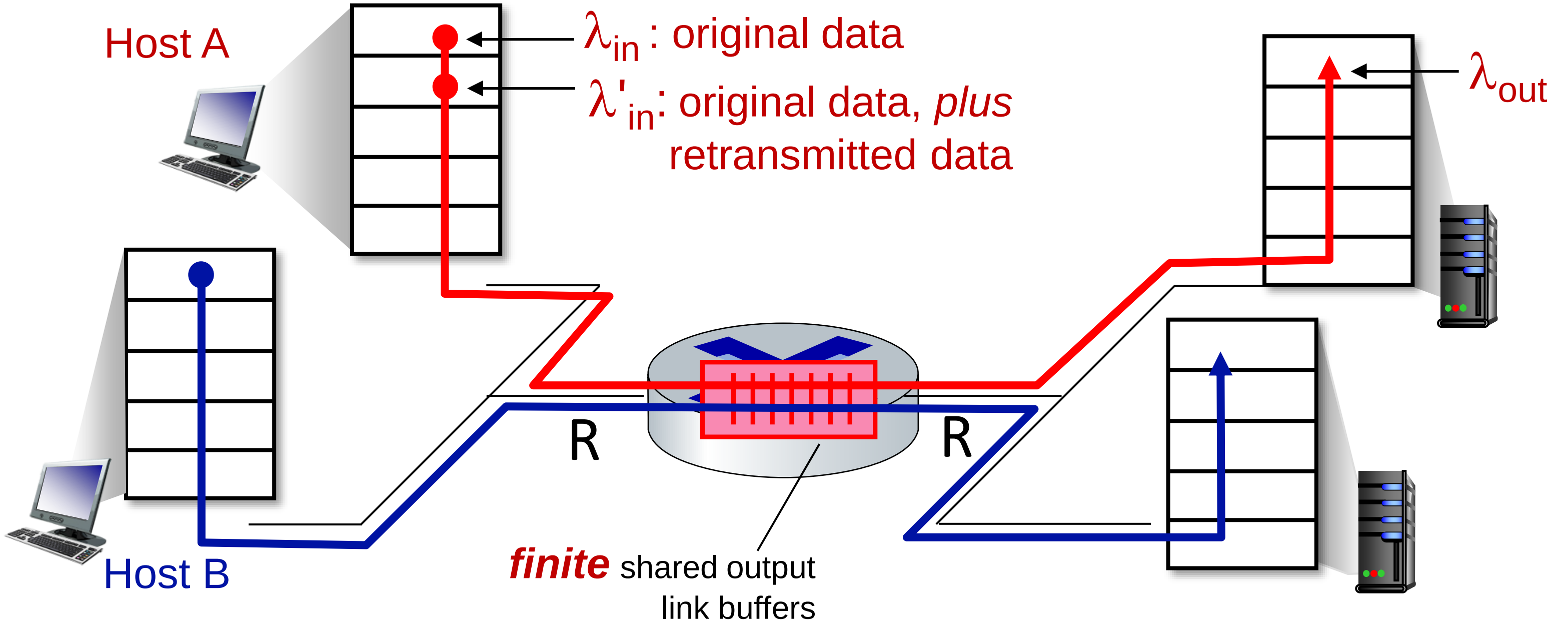
maximum per-connection throughput: R/2



large delays as arrival rate λινε approaches capacity

Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmits lost, timed-out packet
  - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
  - transport-layer input includes *retransmissions* : $\lambda'_{in} \geq \lambda_{in}$

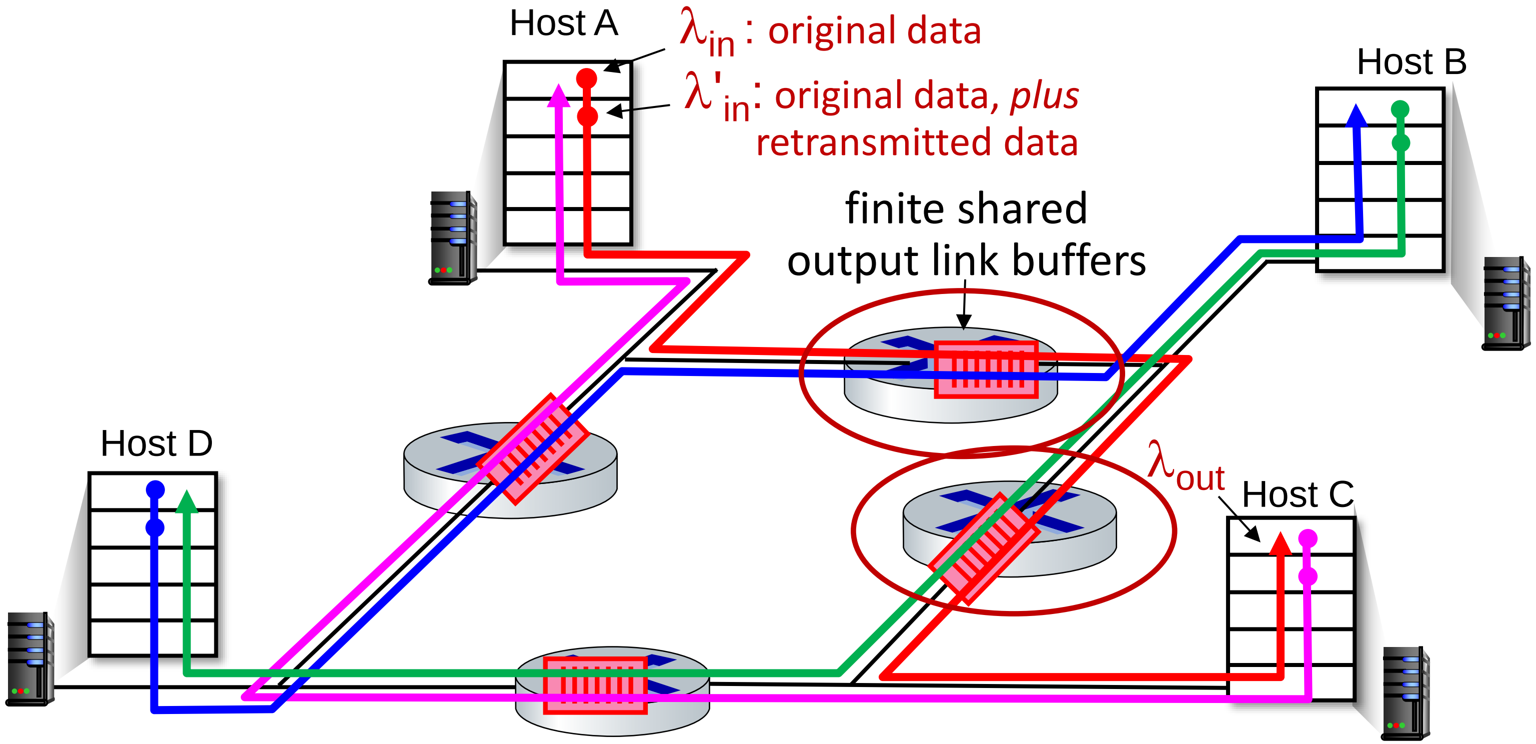


Causes/costs of congestion: scenario 3

- *four* senders
- *multi-hop* paths
- timeout/retransmit

Q: what happens as λ_{in} and λ_{in}' increase ?

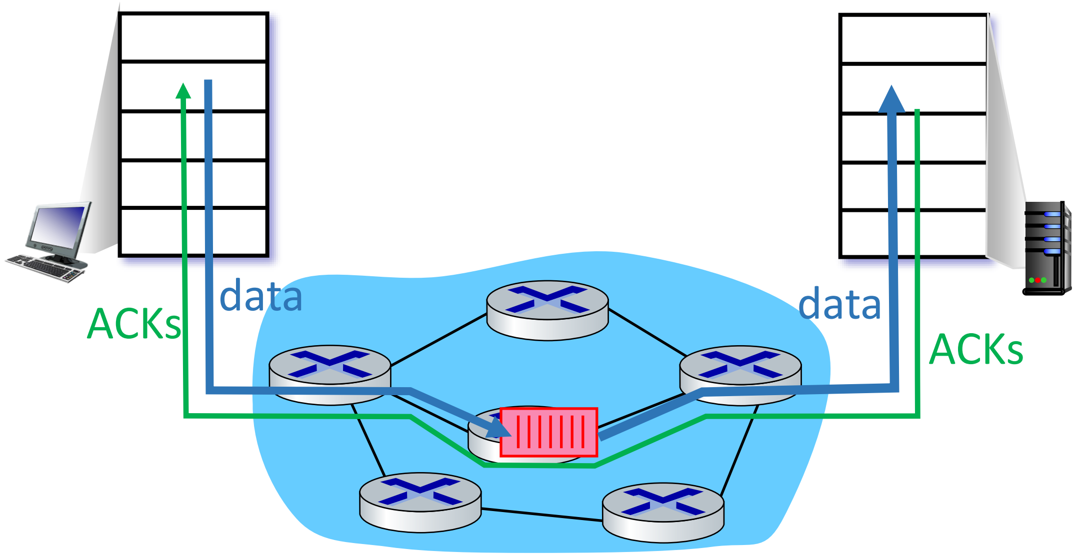
A: as red λ_{in}' increases, all arriving blue pkts at upper queue are dropped, blue throughput → 0



Approaches towards congestion control

End-end congestion control:

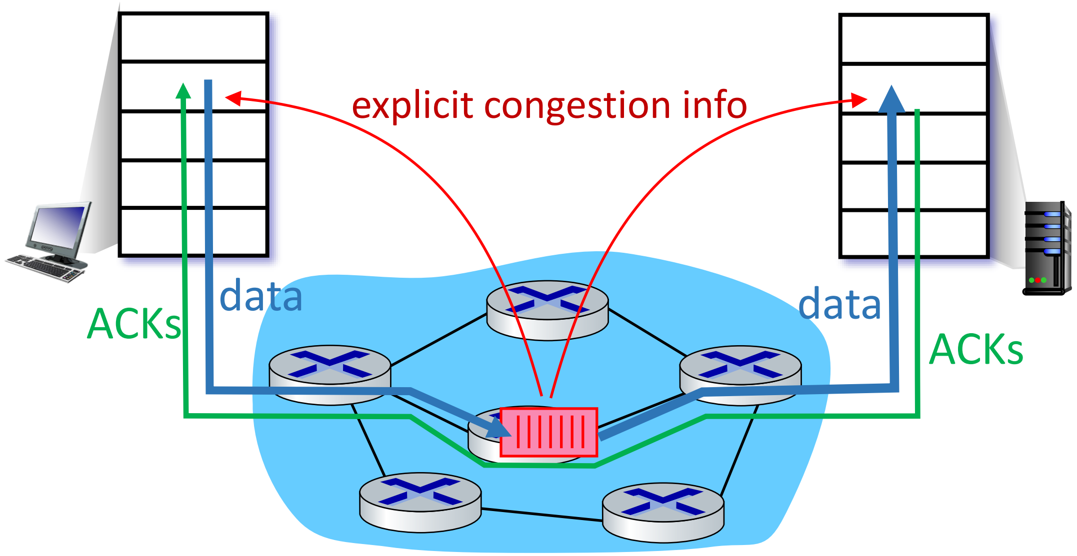
- no explicit feedback from network
- congestion *inferred* from observed loss, delay
- approach taken by TCP



Approaches towards congestion control

Network-assisted congestion control:

- routers provide *direct* feedback to sending/receiving hosts with flows passing through congested router
- may indicate congestion level or explicitly set sending rate
- TCP ECN, ATM, DECbit protocols



Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
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TCP congestion control: AIMD

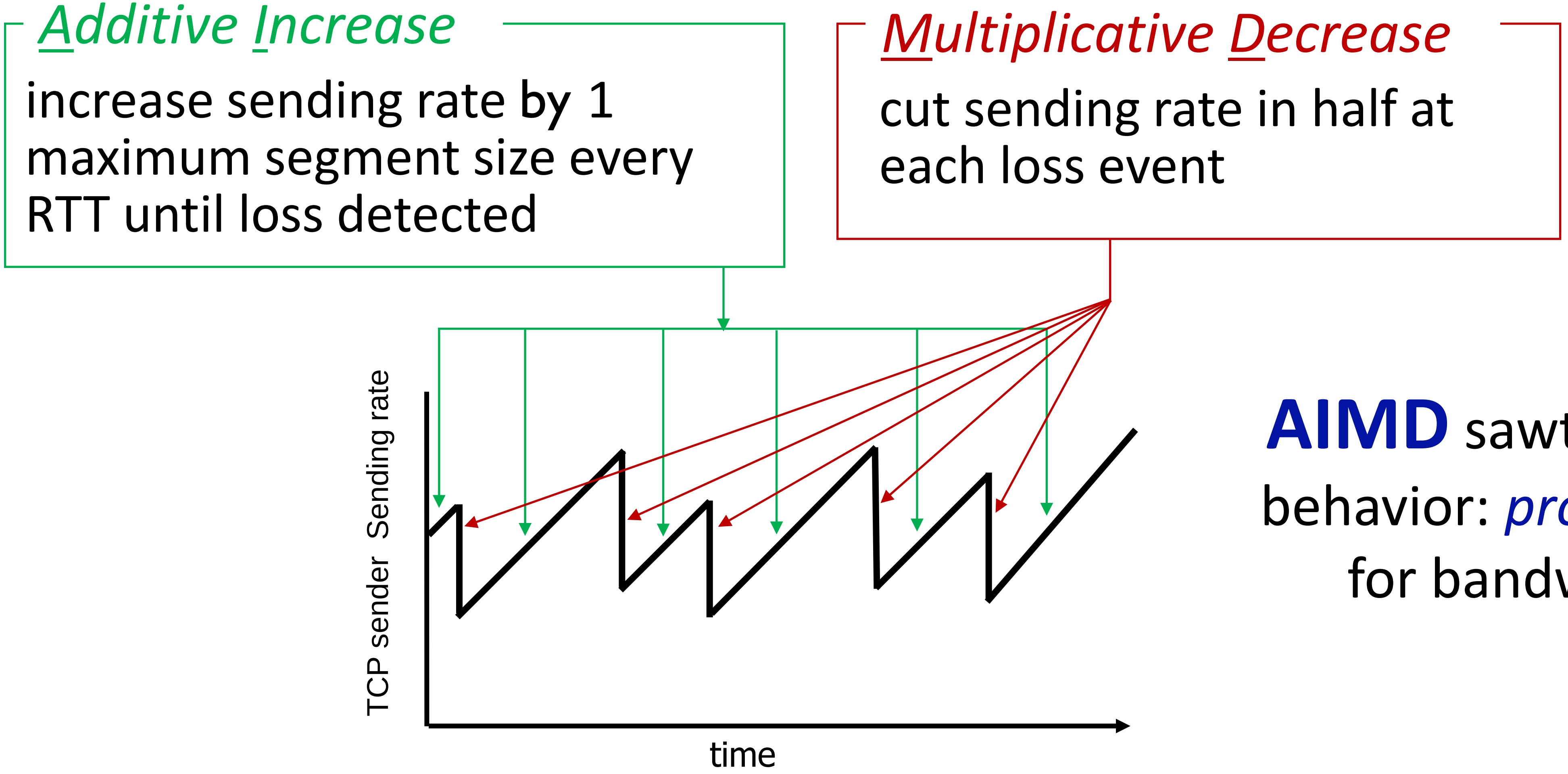
- *approach:* senders can increase sending rate until packet loss (congestion) occurs, then decrease sending rate on loss event

Additive Increase

increase sending rate by 1 maximum segment size every RTT until loss detected

Multiplicative Decrease

cut sending rate in half at each loss event



AIMD sawtooth behavior: *probing* for bandwidth

TCP AIMD: more

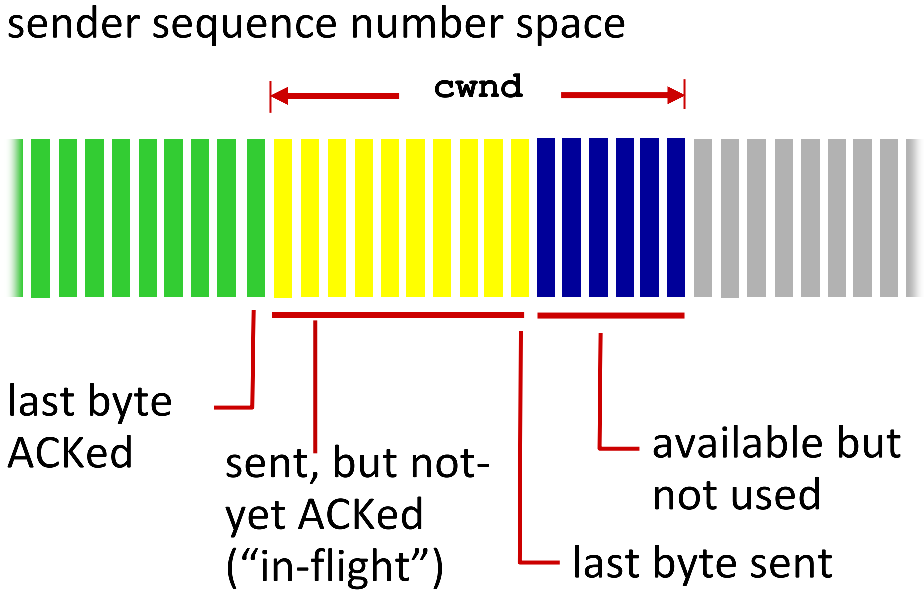
Multiplicative decrease detail: sending rate is

- Cut in half on loss detected by triple duplicate ACK (TCP Reno)
- Cut to 1 MSS (maximum segment size) when loss detected by timeout (TCP Tahoe)

Why AIMD?

- AIMD – a distributed, asynchronous algorithm – has been shown to:
  - optimize congested flow rates network wide!
  - have desirable stability properties

TCP congestion control: details



TCP sending behavior:

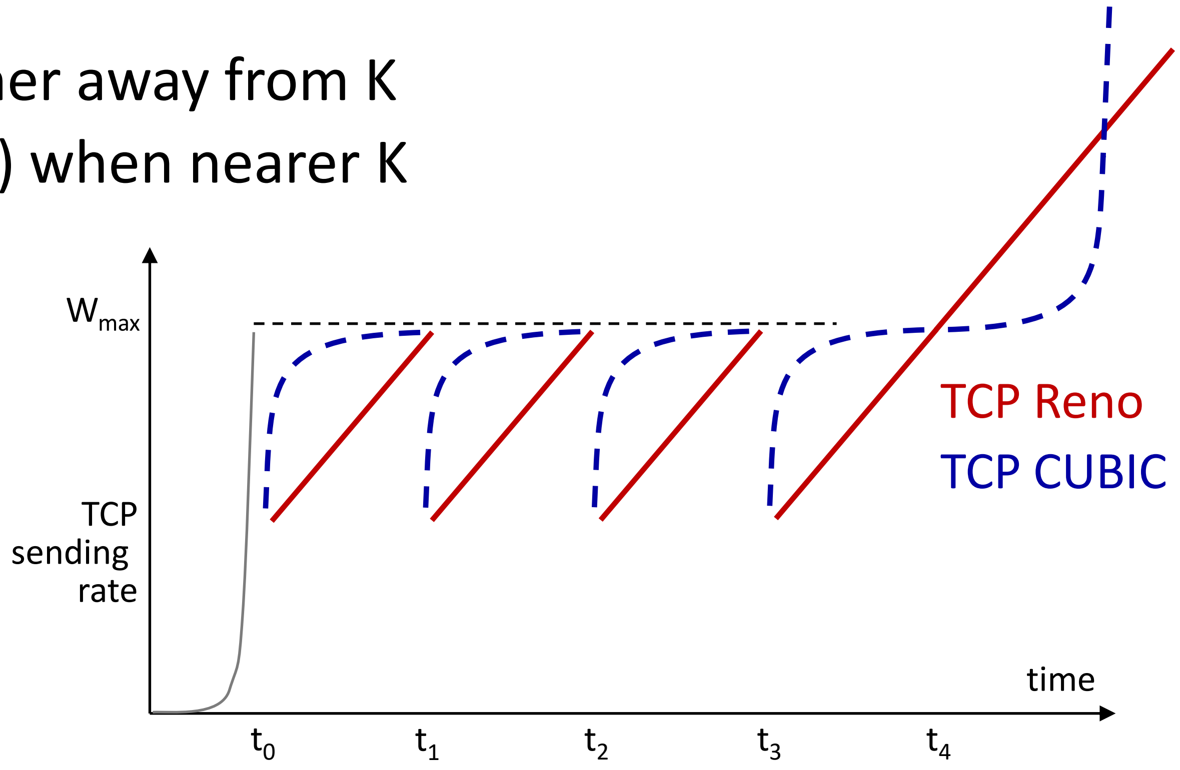
- *roughly:* send `cwnd` bytes, wait RTT for ACKS, then send more bytes

$$\text{TCP rate} \approx \frac{\texttt{cwnd}}{\text{RTT}} \text{ bytes/sec}$$

- TCP sender limits transmission: **`LastByteSent- LastByteAcked ≤ cwnd`**
- `cwnd` is dynamically adjusted in response to observed network congestion (implementing TCP congestion control)

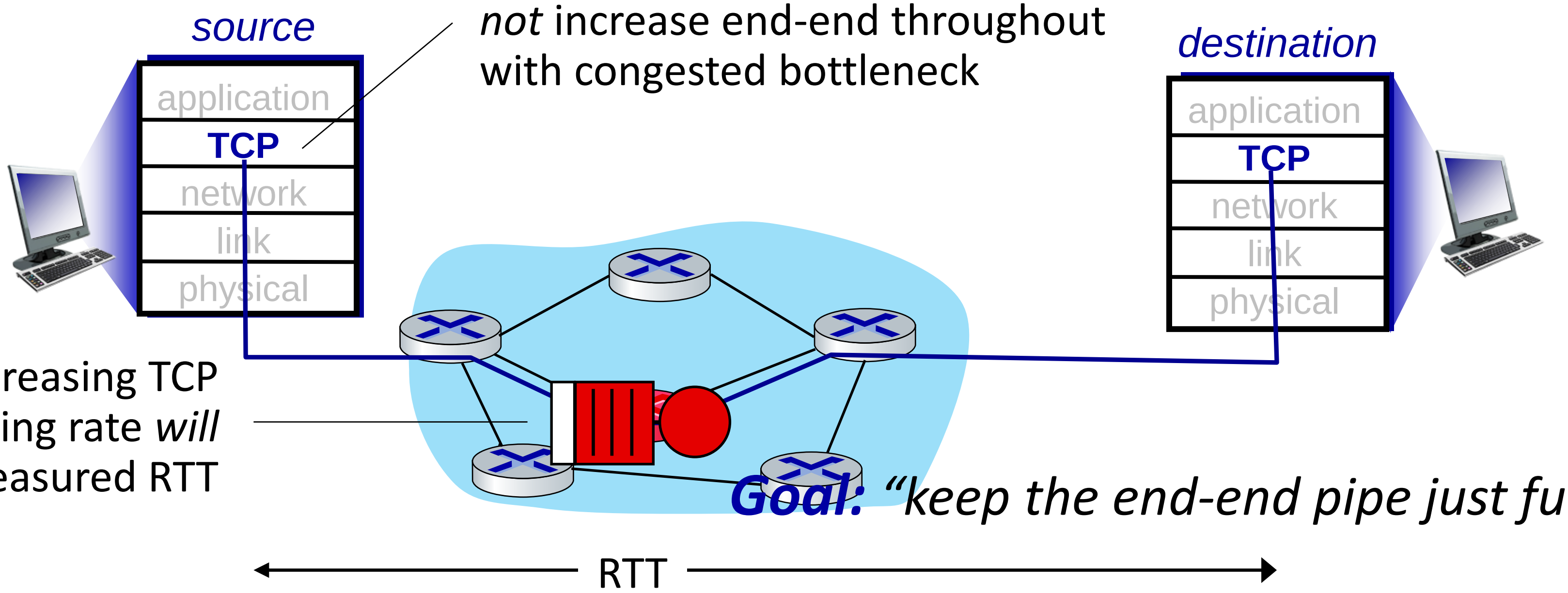
TCP CUBIC

- K: point in time when TCP window size will reach W_{max}
  - K itself is tunable
- increase W as a function of the *cube* of the distance between current time and K
  - larger increases when further away from K
  - smaller increases (cautious) when nearer K
- TCP CUBIC default in Linux, most popular TCP for popular Web servers



TCP and the congested “bottleneck link”

- TCP (classic, CUBIC) increase TCP’s sending rate until packet loss occurs at some router’s output: the *bottleneck link*
- understanding congestion: useful to focus on congested bottleneck link



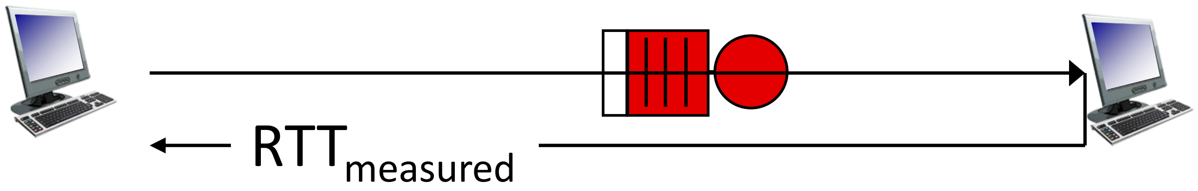
insight: increasing TCP sending rate will *not* increase end-end throughout with congested bottleneck

insight: increasing TCP sending rate *will* increase measured RTT

Goal: *“keep the end-end pipe just full, but not fuller”*

Delay-based TCP congestion control

Keeping sender-to-receiver pipe “just full enough, but no fuller”: keep bottleneck link busy transmitting, but avoid high delays/buffering



$$\text{measured throughput} = \frac{\text{\# bytes sent in last RTT interval}}{RTT_{measured}}$$

Delay-based approach:

- RTT_{min} - minimum observed RTT (uncongested path)
- uncongested throughput with congestion window `cwnd` is $cwnd/RTT_{min}$

```
if measured throughput “very close” to  uncongested throughput
    increase cwnd linearly               /* since path not congested */
else if measured throughput “far below” uncongested throughout
    decrease cwnd  linearly              /* since path is congested */
```

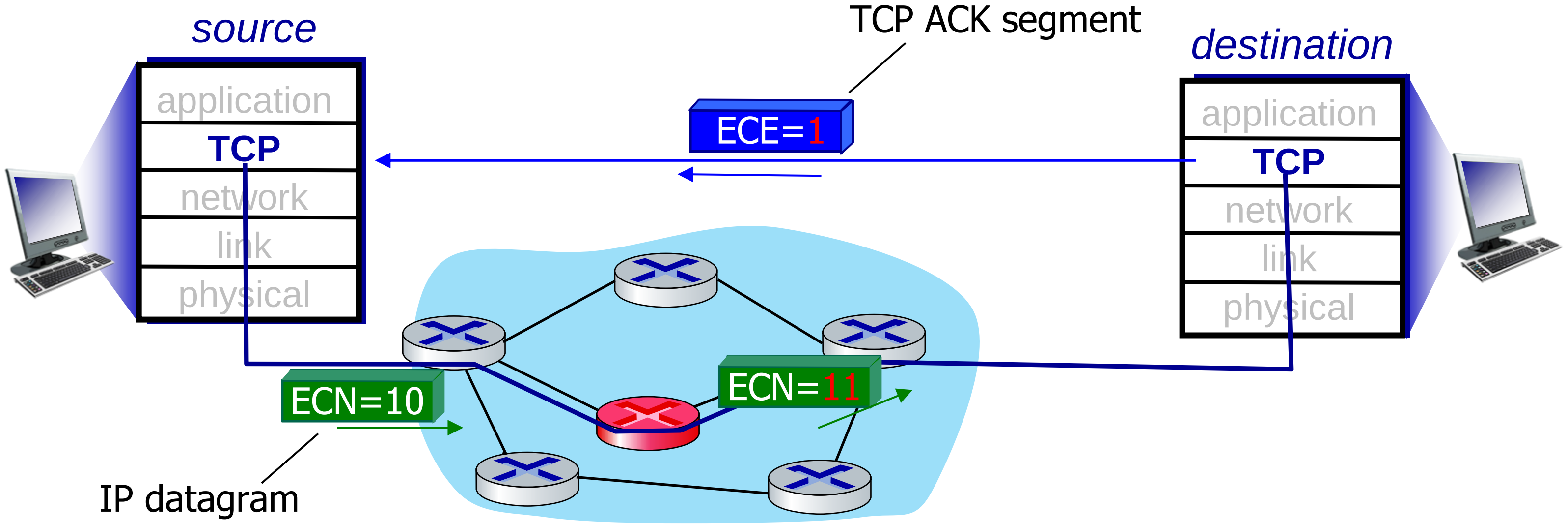
Delay-based TCP congestion control

- congestion control without inducing/forcing loss
- maximizing throughout ("keeping the just pipe full… ") while keeping delay low ("…but not fuller")
- a number of deployed TCPs take a delay-based approach
    - BBR deployed on Google's (internal) backbone network

Explicit congestion notification (ECN)

TCP deployments often implement *network-assisted* congestion control:

- two bits in IP header (ToS field) marked *by network router* to indicate congestion
  - *policy* to determine marking chosen by network operator
- congestion indication carried to destination
- destination sets ECE bit on ACK segment to notify sender of congestion
- involves both IP (IP header ECN bit marking) and TCP (TCP header C,E bit marking)



Fairness: must all network apps be "fair"?

Fairness and UDP

- multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- instead use UDP:
  - send audio/video at constant rate, tolerate packet loss
- there is no "Internet police" policing use of congestion control

Fairness, parallel TCP connections

- application can open *multiple* parallel connections between two hosts
- web browsers do this , e.g., link of rate R with 9 existing connections:
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets R/2

Transport layer: roadmap

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Evolving transport-layer functionality

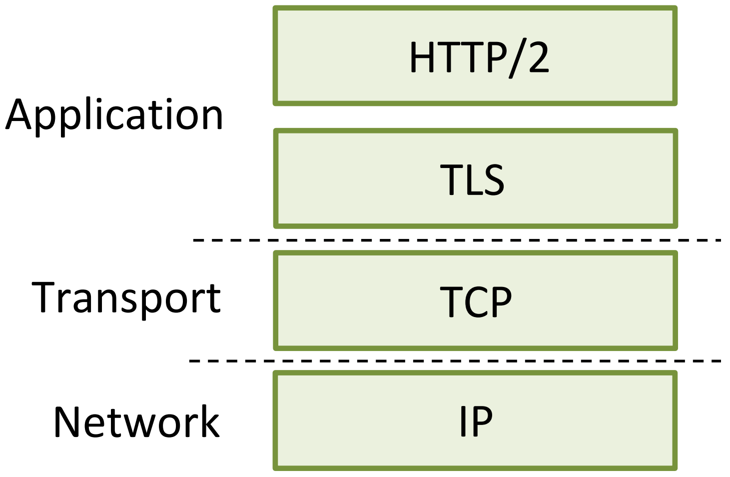
- TCP, UDP: principal transport protocols for 40 years
- different “flavors” of TCP developed, for specific scenarios:

Scenario	Challenges
Long, fat pipes (large data transfers)	Many packets “in flight”; loss shuts down pipeline
Wireless networks	Loss due to noisy wireless links, mobility; TCP treat this as congestion loss
Long-delay links	Extremely long RTTs
Data center networks	Latency sensitive
Background traffic flows	Low priority, “background” TCP flows

- moving transport–layer functions to application layer, on top of UDP
  - HTTP/3: QUIC

QUIC: Quick UDP Internet Connections

- application-layer protocol, on top of UDP
  - increase performance of HTTP
  - deployed on many Google servers, apps (Chrome, mobile YouTube app)



HTTP/2 over TCP

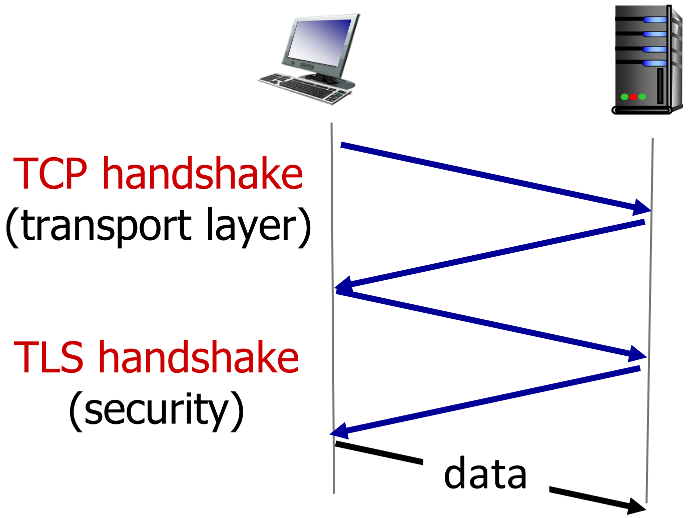
QUIC: Quick UDP Internet Connections

adopts approaches we've studied in this chapter for connection establishment, error control, congestion control

- **error and congestion control:** "Readers familiar with TCP's loss detection and congestion control will find algorithms here that parallel well-known TCP ones." [from QUIC specification]
- **connection establishment:** reliability, congestion control, authentication, encryption, state established in one RTT

- multiple application-level "streams" multiplexed over single QUIC connection
  - separate reliable data transfer, security
  - common congestion control

QUIC: Connection establishment



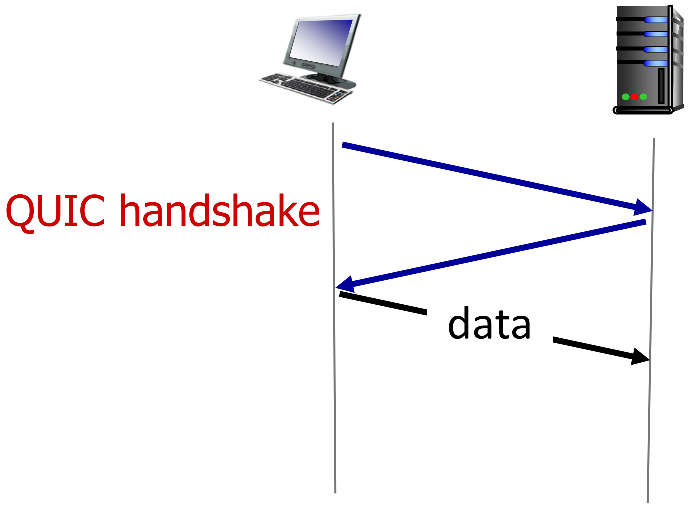
TCP handshake
(transport layer)

TLS handshake
(security)

data

TCP (reliability, congestion control state) + TLS (authentication, crypto state)

- 2 serial handshakes



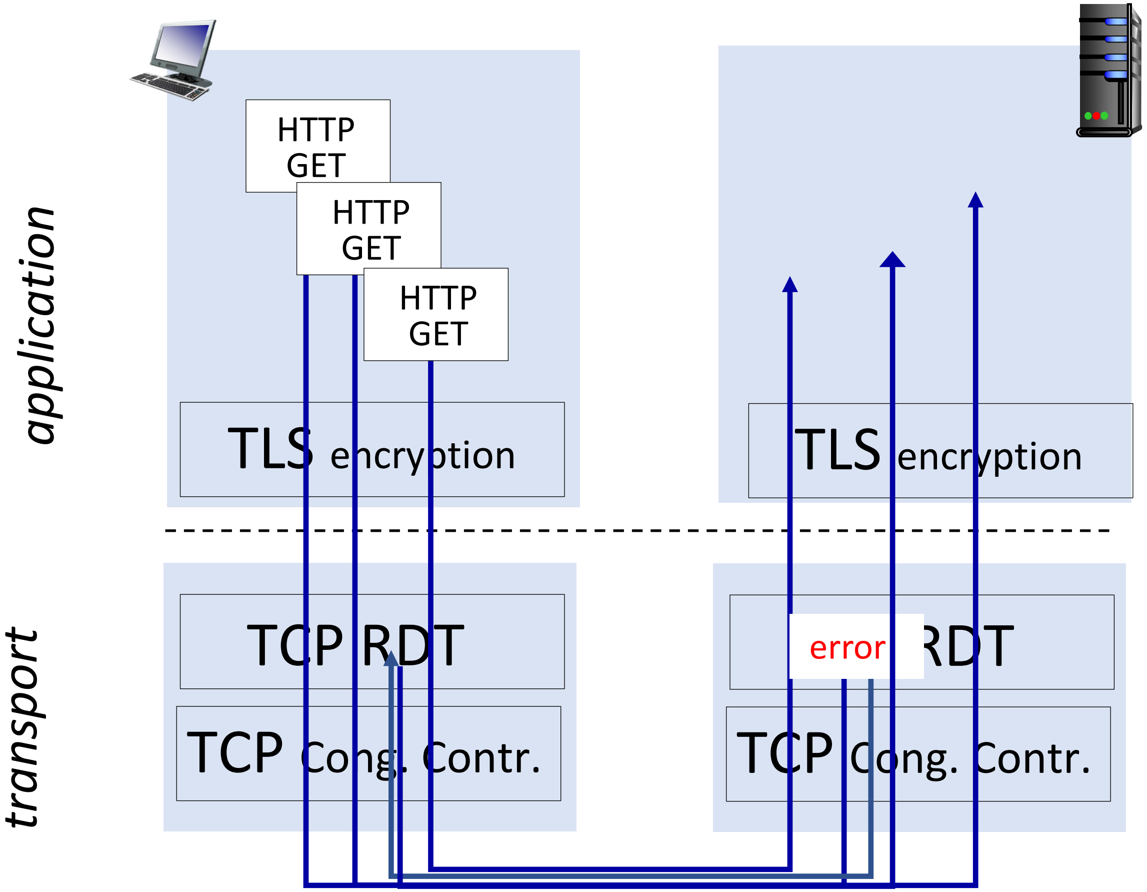
QUIC handshake

data

QUIC: reliability, congestion control, authentication, crypto state

- 1 handshake

QUIC: streams: parallelism, no HOL blocking



(a) HTTP 1.1

TCP over “long, fat pipes”

- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- requires W = 83,333 in-flight segments
- throughput in terms of segment loss probability, L [Mathis 1997]:

$$\text{TCP throughput} = \frac{1.22 \cdot \text{MSS}}{\text{RTT}\sqrt{L}}$$

➜ to achieve 10 Gbps throughput, need a loss rate of L = $2 \cdot 10^{-10}$ *– a very small loss rate!*

- versions of TCP for long, high-speed scenarios

Acknowledgment

- **These lecture slides are based on:**

1) **Chapter 3 (P 211-312)** from the book “Computer Networking: A Top-Down Approach, Eighth Edition, Global Edition” by (James F. Kurose and Keith W. Ross’s).

END OF LECTURE (4)

Keep connected with the classroom

lmzcbsf

THANK YOU FOR YOUR ATTENTION