Transport Layer

WPI

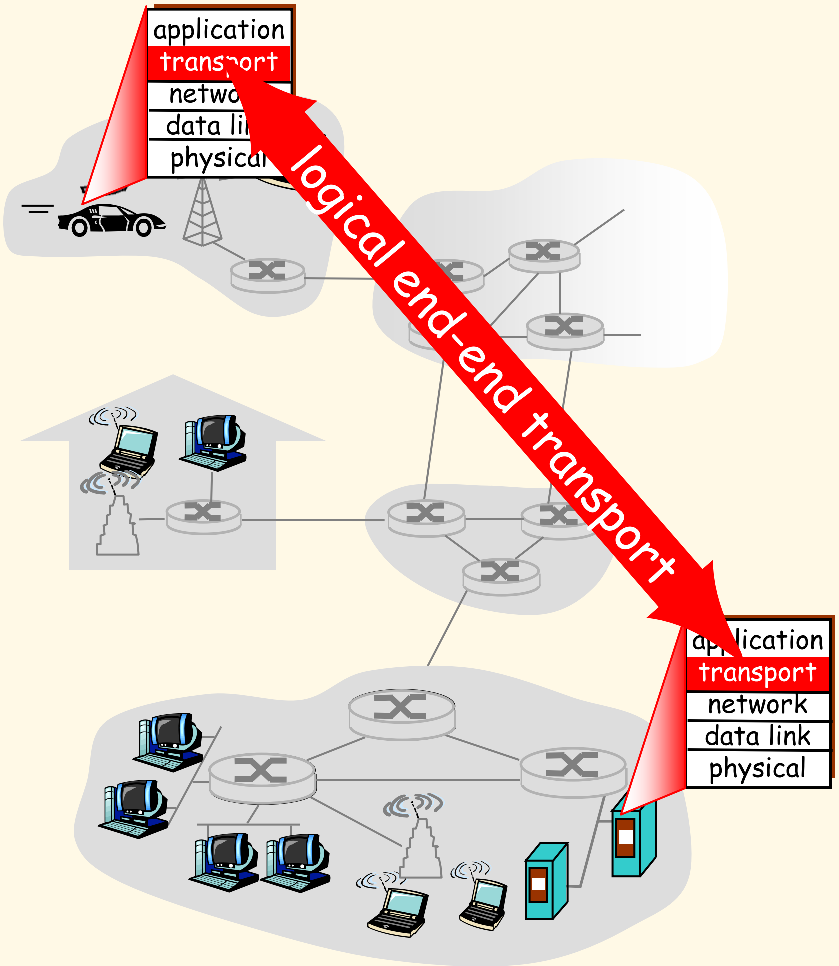
Computer Networks
Term A15

Kurose's Chapter 3 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

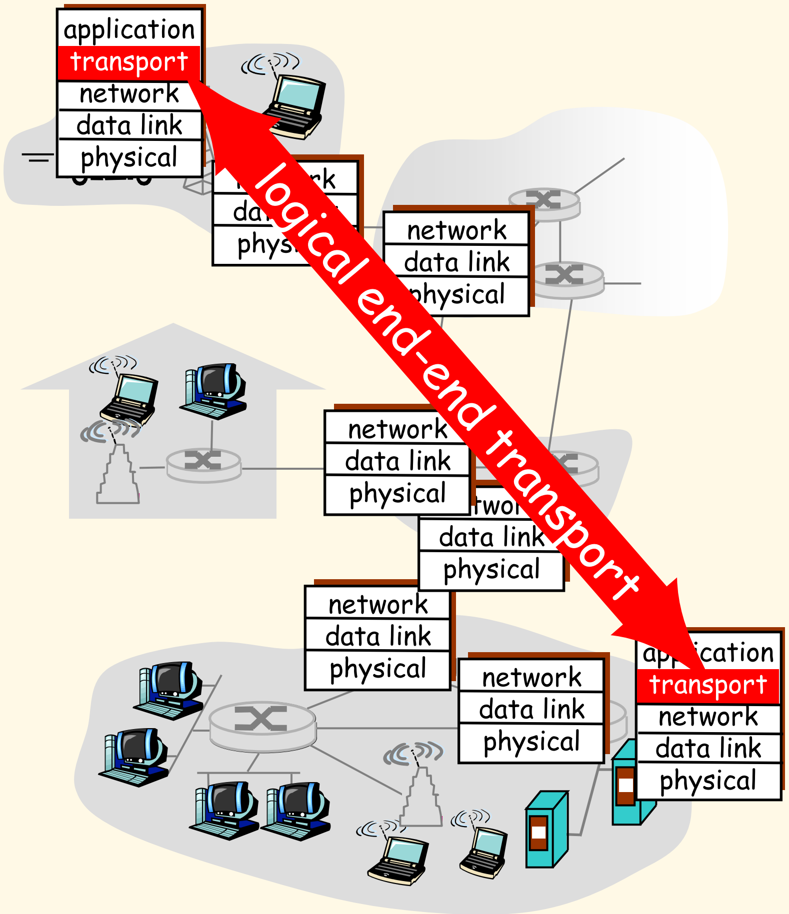
Transport Services and Protocols

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



Internet Transport Layer Protocols

- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



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- 3.2 Multiplexing and demultiplexing [Brief Look]
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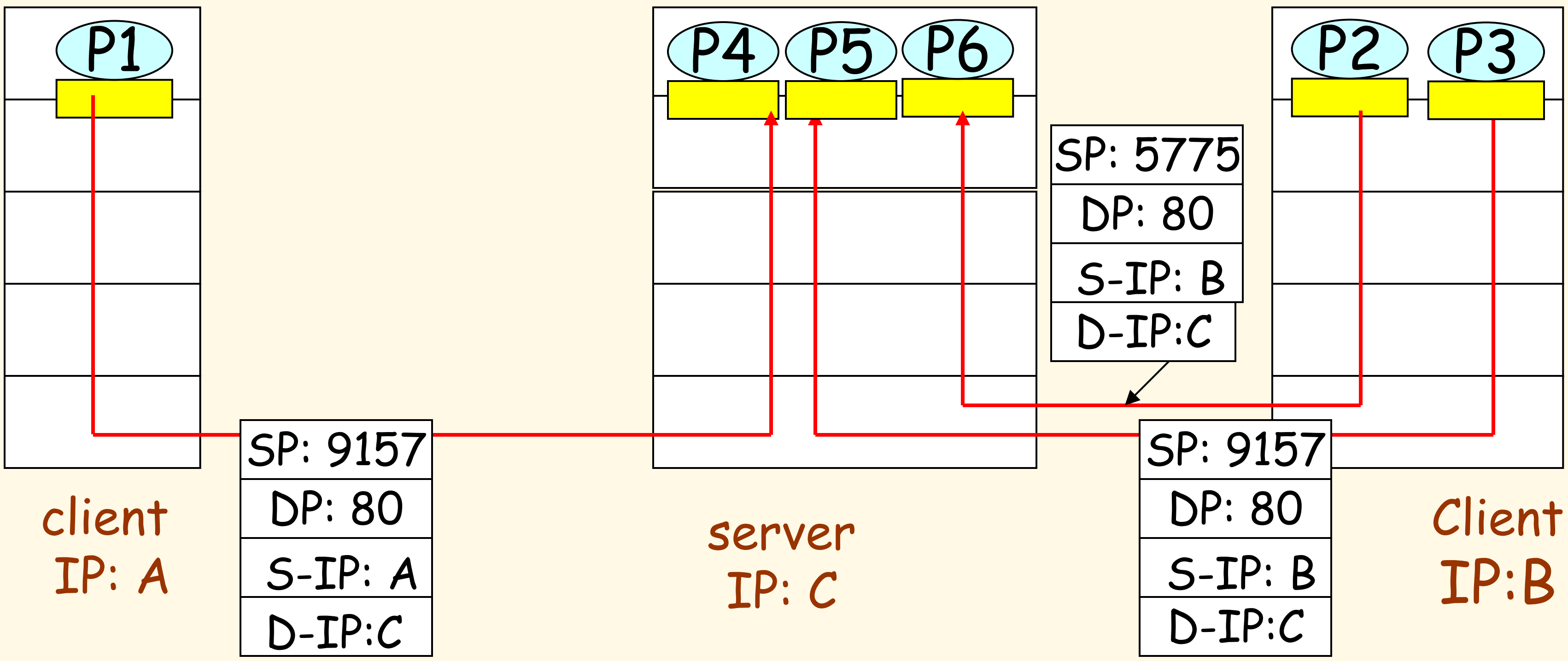
Connection-Oriented Demux

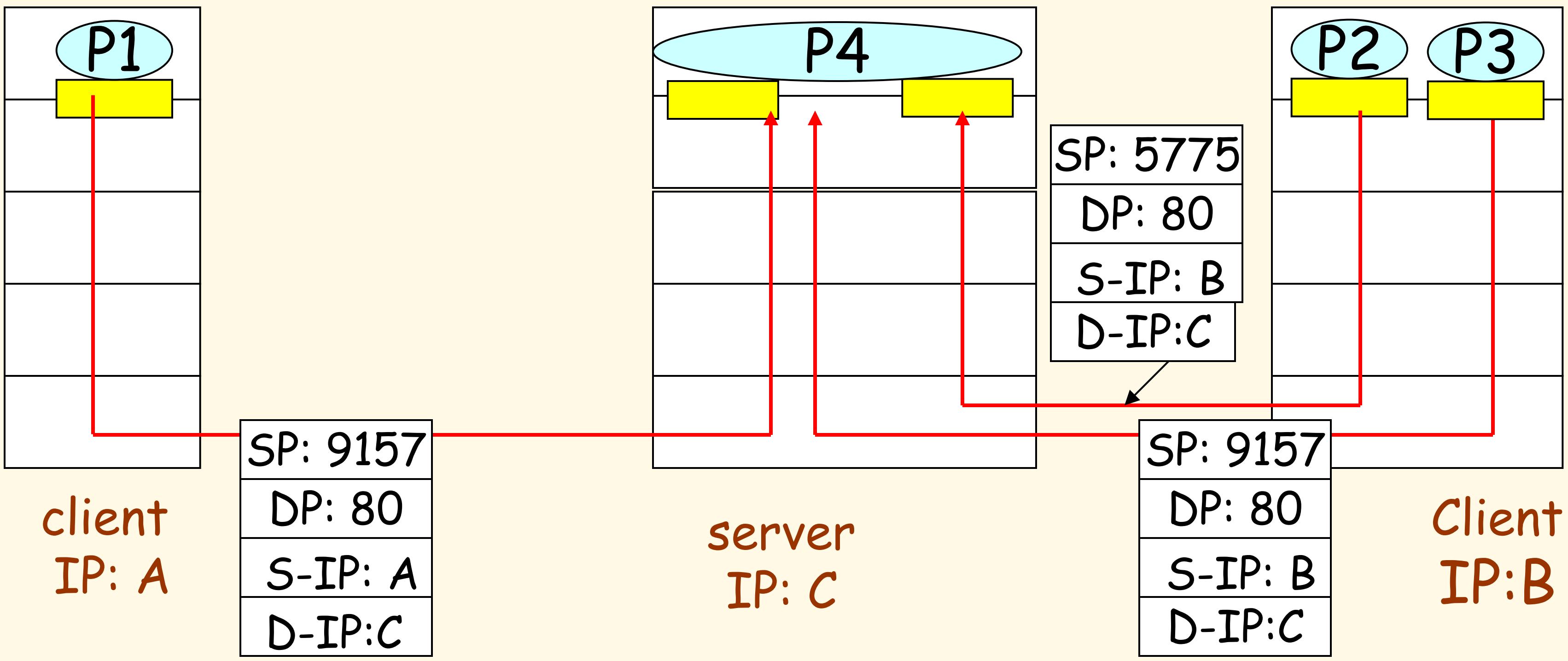
- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- receiving host uses all four values to direct segment to appropriate socket.
- Server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple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

Connection-Oriented Demux



Connection-Oriented Demux

Threaded Web Server



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UDP: User Datagram Protocol [RFC 768]

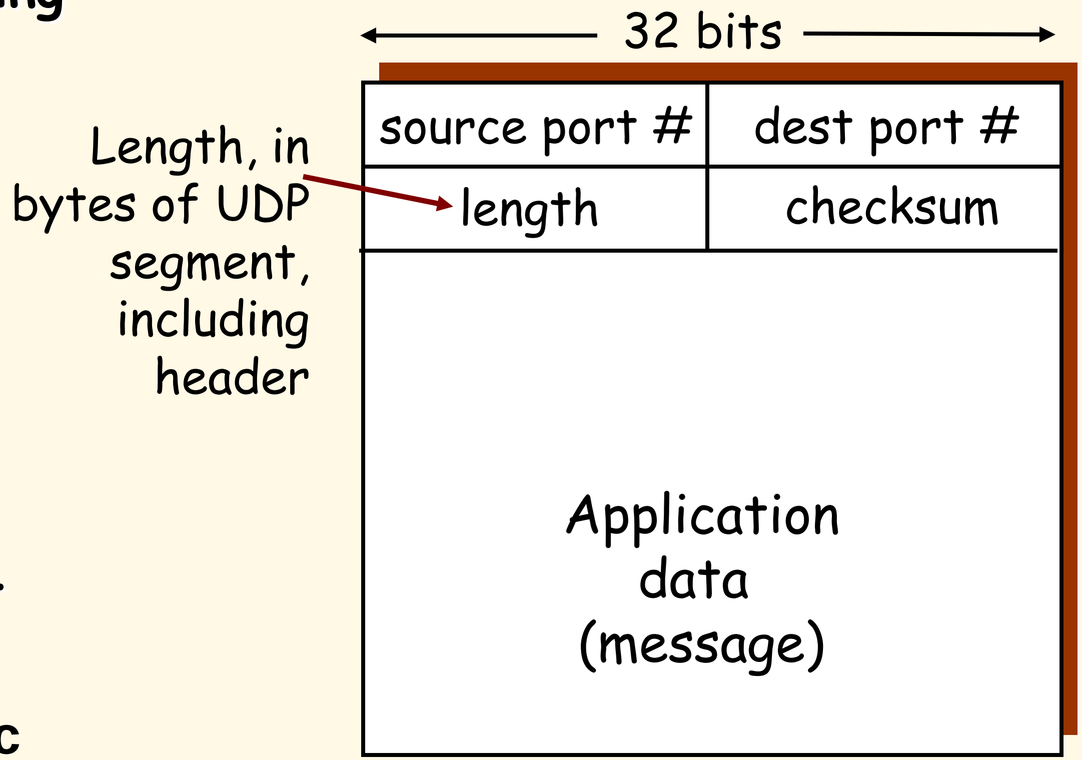
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out of order to app
- connectionless:
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others.

Why is there a UDP?

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

UDP Details

- often used for streaming multimedia apps
  - loss tolerant
  - rate sensitive
- other UDP uses
  - DNS
  - SNMP
- reliable transfer over UDP: add reliability at application layer
  - application-specific error recovery!



UDP segment format

UDP Checksum

Goal: detect "errors" (e.g., flipped bits) in transmitted segment

Sender:

- treat segment contents as sequence of 16-bit integers
- checksum: addition (1's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

Receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonetheless?* More later ….

Internet Checksum Example

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

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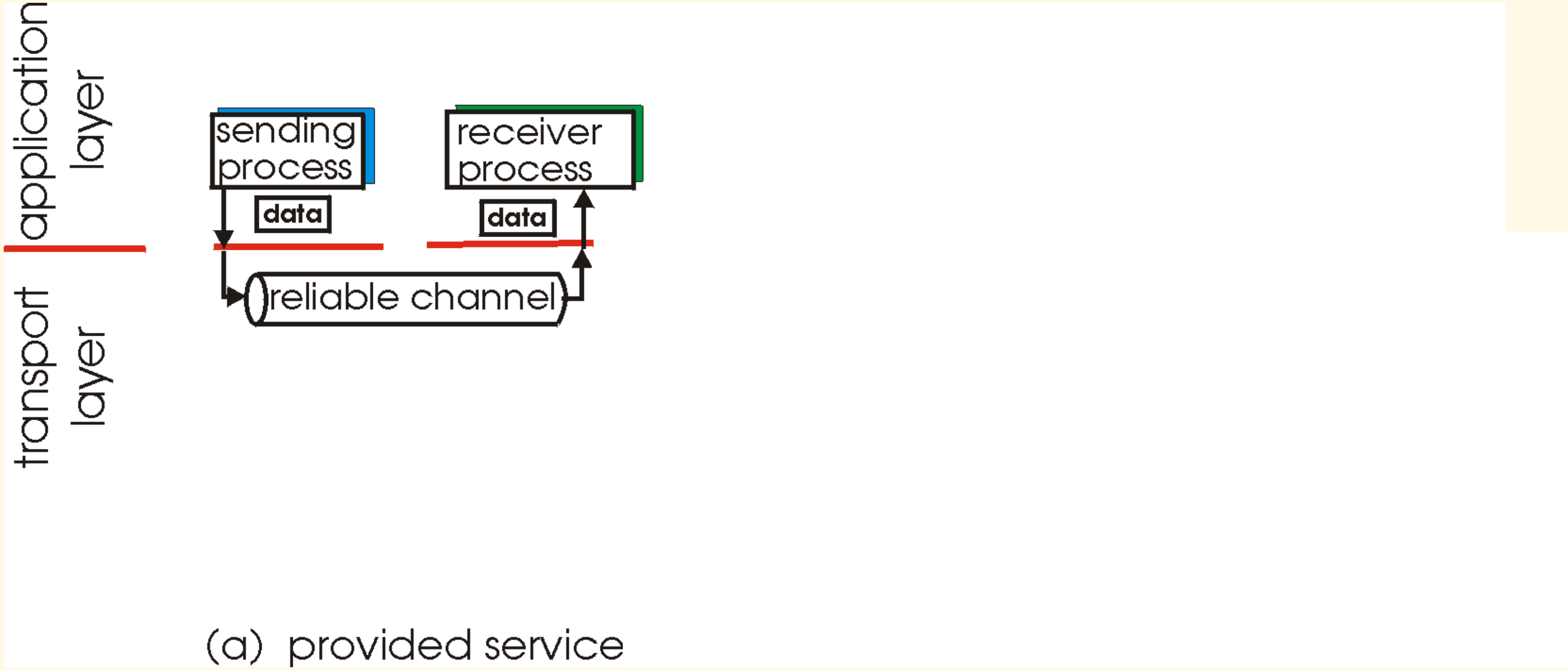
We will use Tanenbaum's Data Link Layer Treatment to study this in place of K&R's Transport Layer Discussion.

Principles of Reliable Data Transfer

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



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

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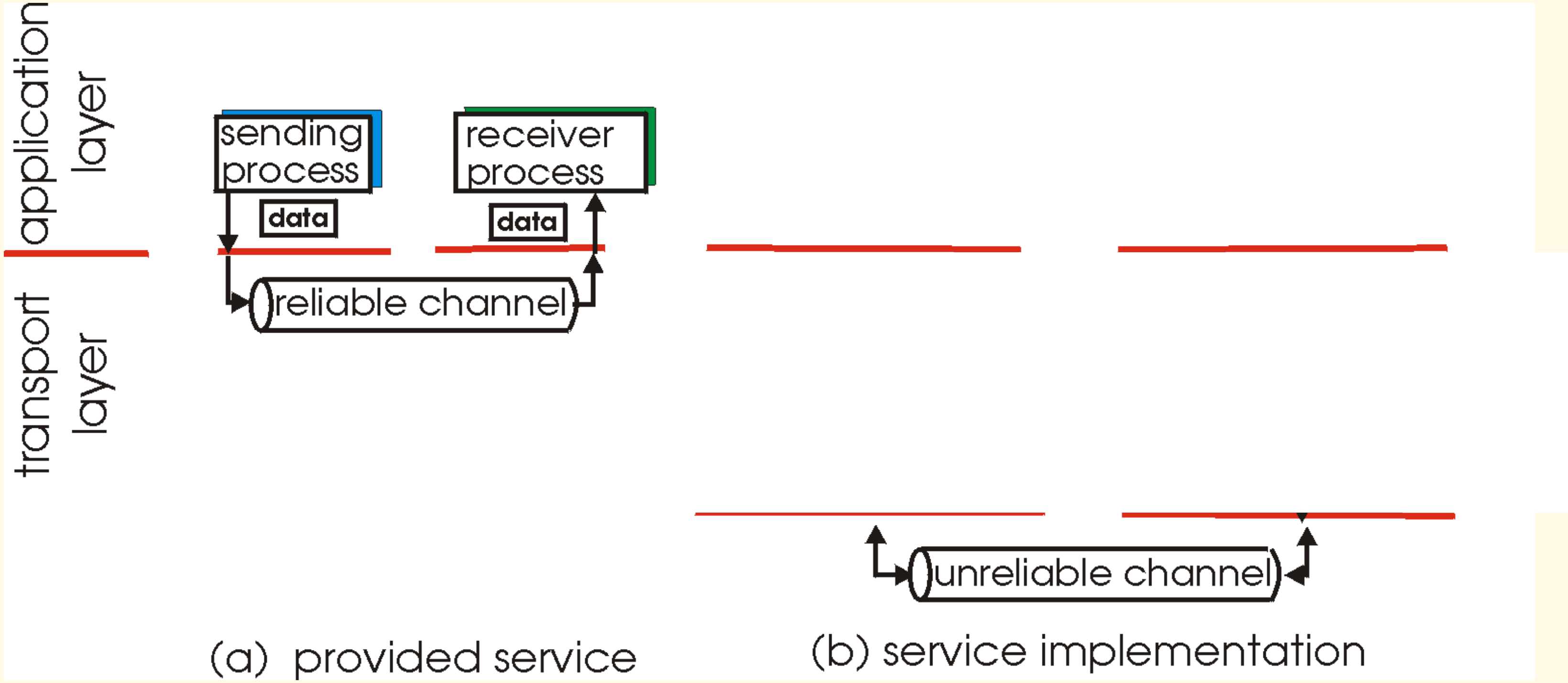

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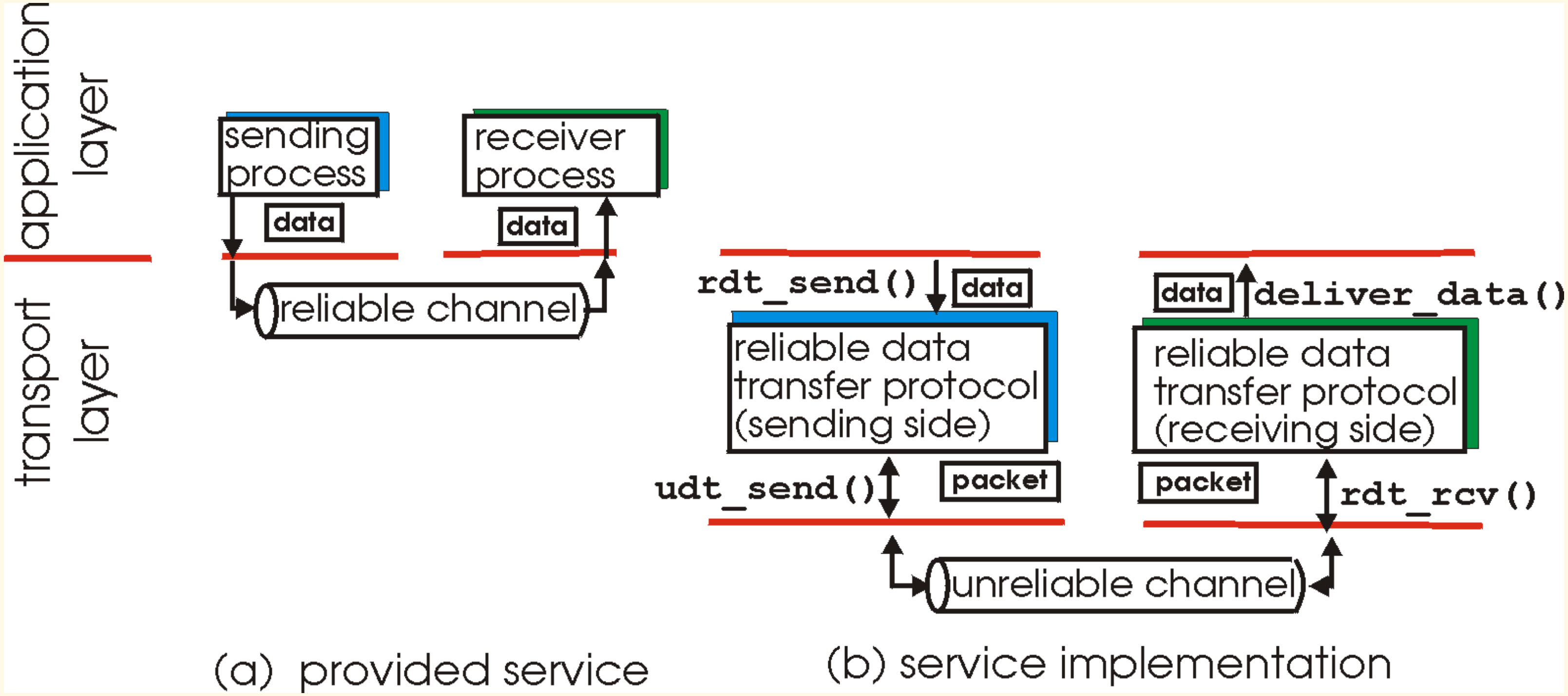
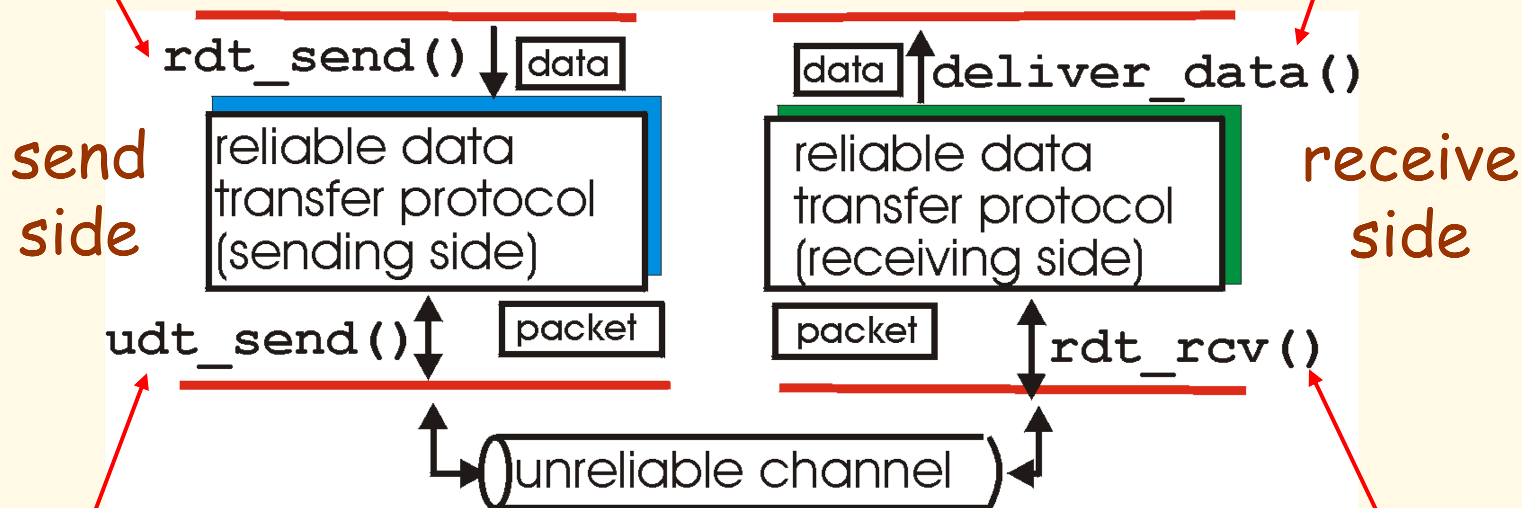


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt).

Reliable Data Transfer: Getting Started

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



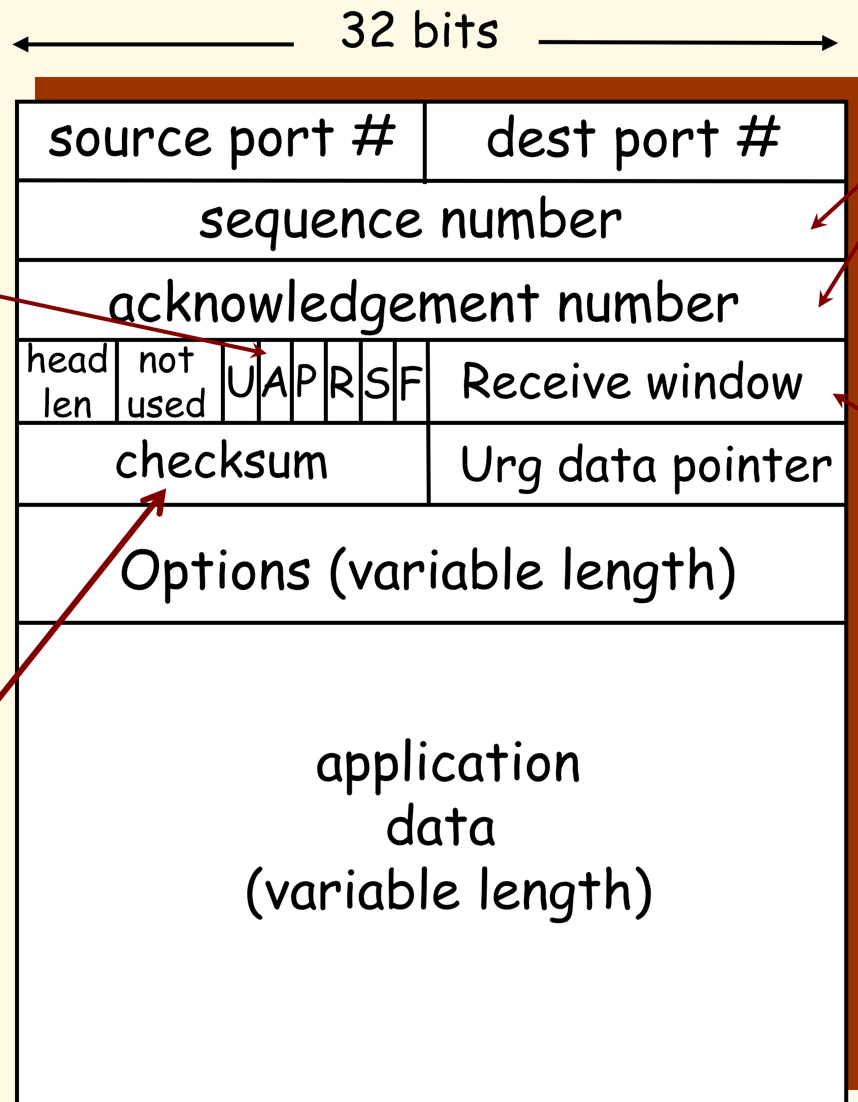
udt_send(): called by rdt, to transfer packet over unreliable channel to receiver

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

TCP Segment Structure

URG: urgent data (generally not used)

ACK: ACK # valid



counting by bytes of data (not segments!)

bytes rcvr willing to accept

Internet checksum (as in UDP) **{needed now for error detection}**

WARNING

Explanation of

Reliable Data Transport

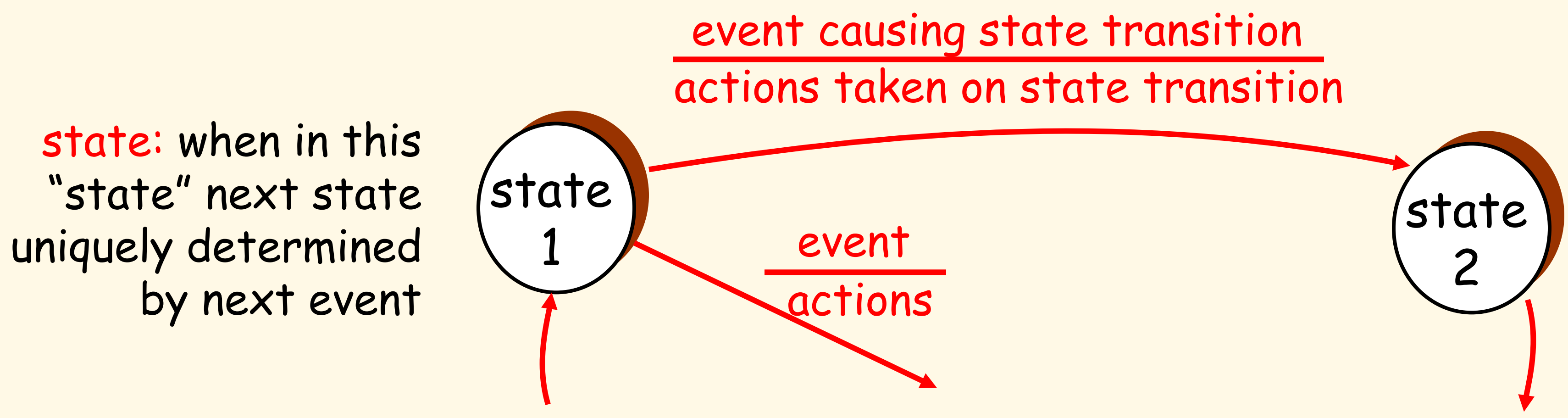
will now be explained using the

Data Link Layer

Reliable data transfer: getting started

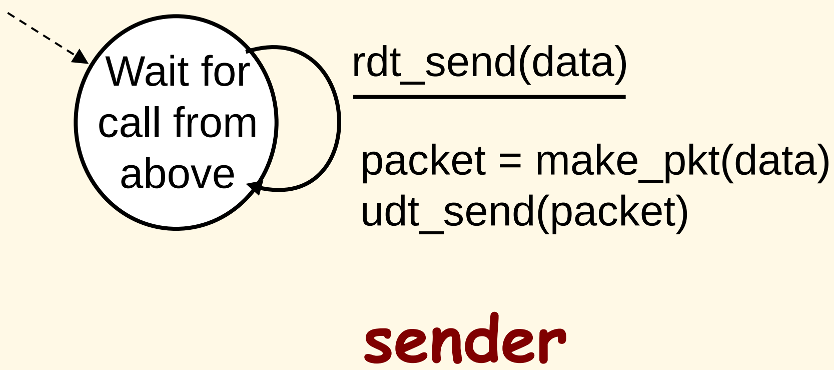
We'll:

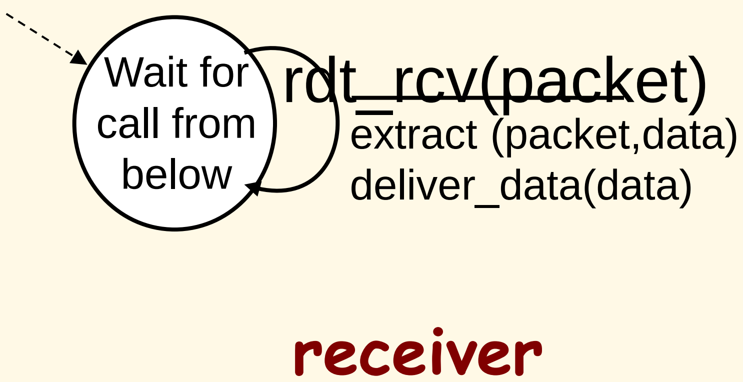
- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver



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 read 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.
- new mechanisms in `rdt2.0` (beyond `rdt1.0`):
  - error detection
  - receiver feedback: control msgs (ACK,NAK) rcvr->sender

rdt3.0: Channels with Errors *and* Loss

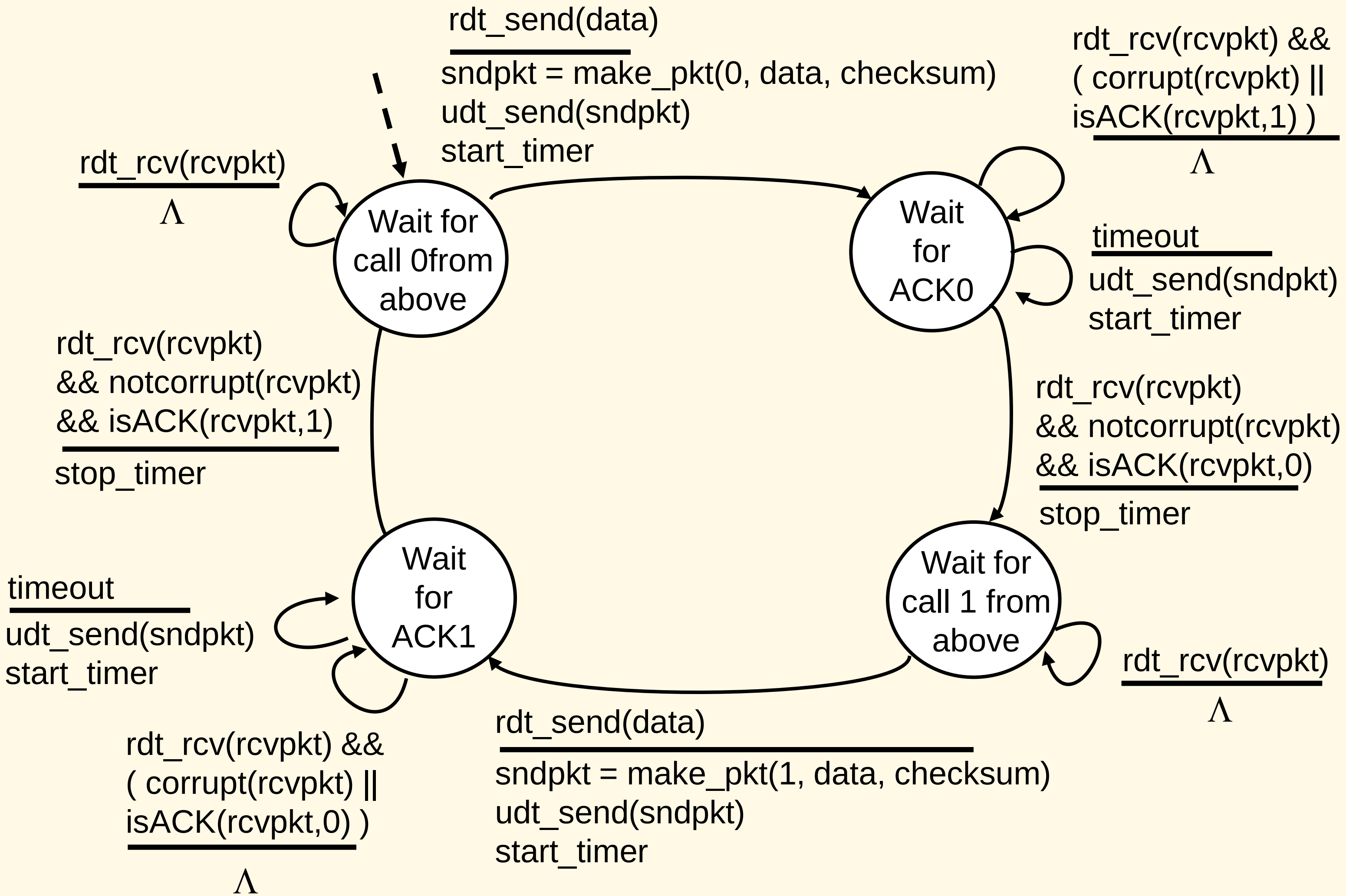
New assumption: underlying channel can also lose packets (data or ACKs)

- 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 use of seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

rdt3.0 Sender



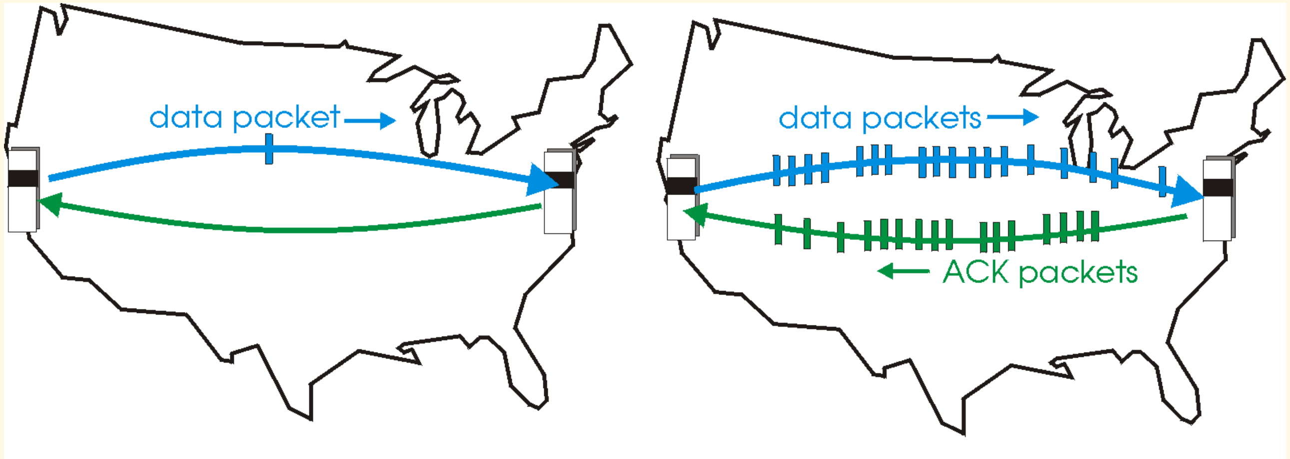
Pipelining and Sliding Windows

- Lecture returns back to this point after Data Link Layer.
- Diagrams from textbook!!

Pipelined Protocols

Pipelining:: sender allows multiple, "in-flight", yet-to-be-acknowledged packets.

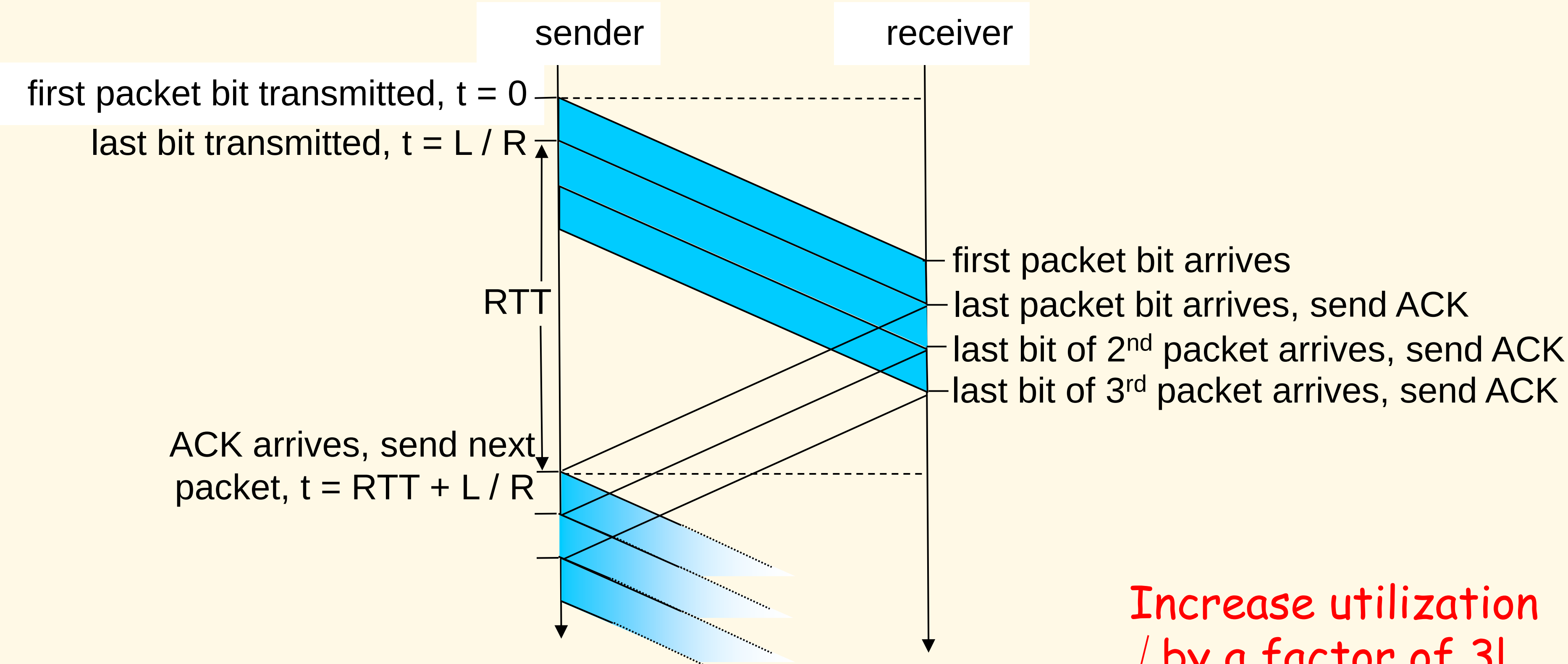
- range of sequence numbers must be increased
- buffering at sender and/or receiver



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

- Two generic forms of pipelined protocols: **Go-Back-N and Selective Repeat**

Pipelining increases Utilization



Increase utilization by a factor of 3!

$$U_{sender} = \frac{3 * L / R}{RTT + L / R} = \frac{.024}{30.008} = 0.0008$$

Pipelining Protocols

Go-back-N: overview

- *sender:* up to N unACKed pkts in pipeline
- *receiver:* only sends cumulative ACKs
  - doesn't ACK pkt if there's a gap
- *sender:* has timer for oldest unACKed pkt
  - if timer expires: retransmit all unACKed packets

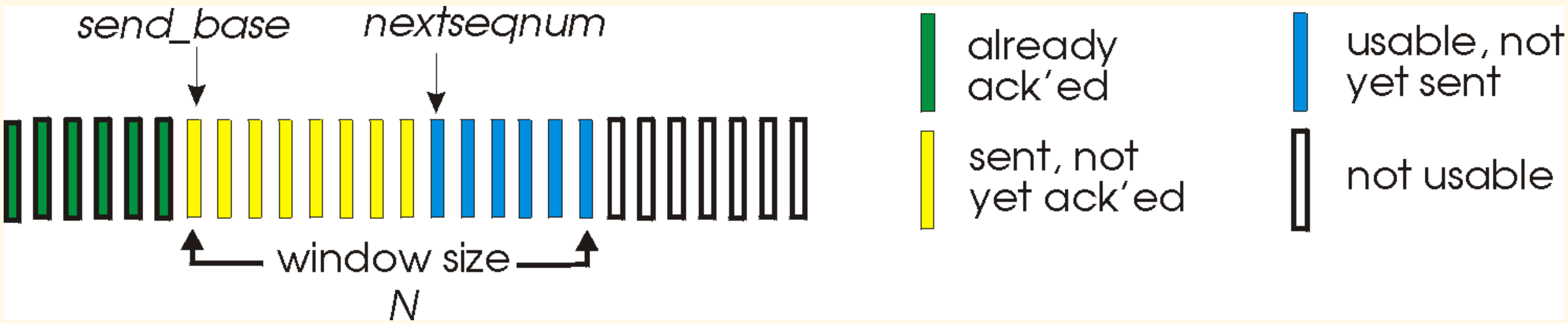
Selective Repeat: overview

- *sender:* up to N unACKed packets in pipeline
- *receiver:* ACKs individual pkts
- *sender:* maintains timer for each unACKed pkt
  - if timer expires: retransmit only unACKed packet.

Go-Back-N

Sender:

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



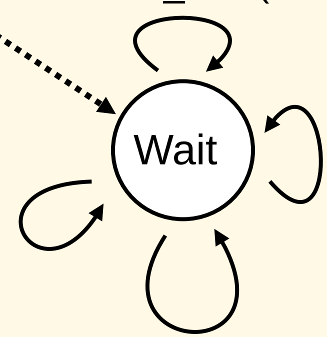
- ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"
  - may receive duplicate ACKs (see receiver)
- timer for each in-flight pkt
- *timeout(n):* retransmit pkt n and all higher seq # pkts in window.

GBN: Sender Extended FSM

```
rdt_send(data)
_________________________________
if (nextseqnum < base+N) {
    sndpkt[nextseqnum] = make_pkt(nextseqnum,data,chksum)
    udt_send(sndpkt[nextseqnum])
    if (base == nextseqnum)
       start_timer
    nextseqnum++
    }
else
  refuse_data(data)
```

Λ

```
_____________
base=1
nextseqnum=1
```

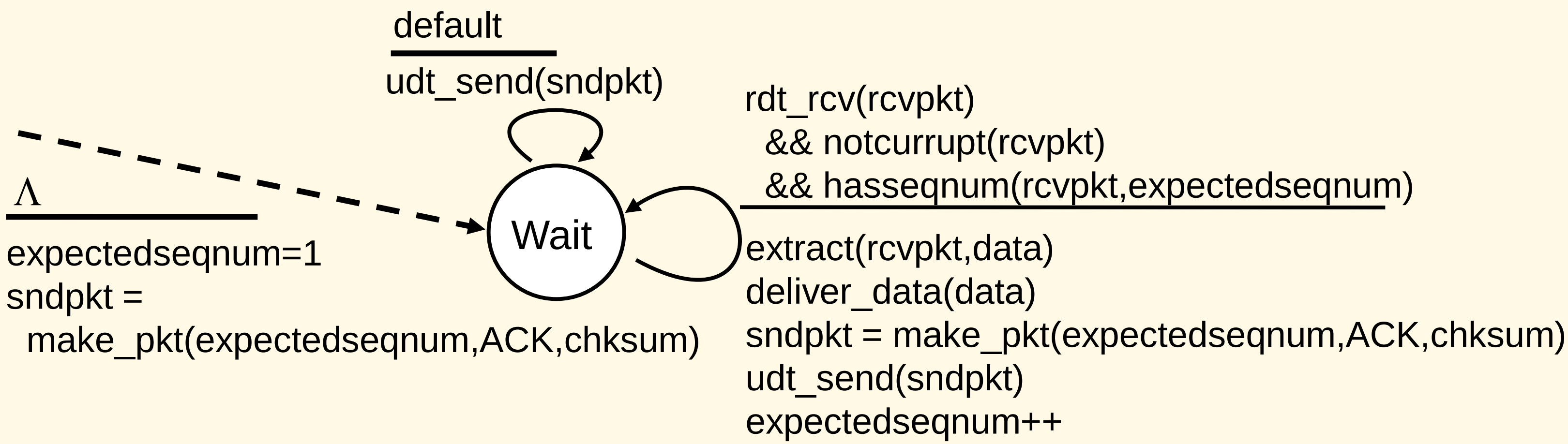



```
timeout
_________________________________
start_timer
udt_send(sndpkt[base])
udt_send(sndpkt[base+1])
…
udt_send(sndpkt[nextseqnum-1])
```

```
rdt_rcv(rcvpkt)
   && corrupt(rcvpkt)
_________________________________
```

```
rdt_rcv(rcvpkt) &&
   notcorrupt(rcvpkt)
_________________________________
base = getacknum(rcvpkt)+1
If (base == nextseqnum)
    stop_timer
  else
    start_timer
```

GBN: Receiver Extended FSM



ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

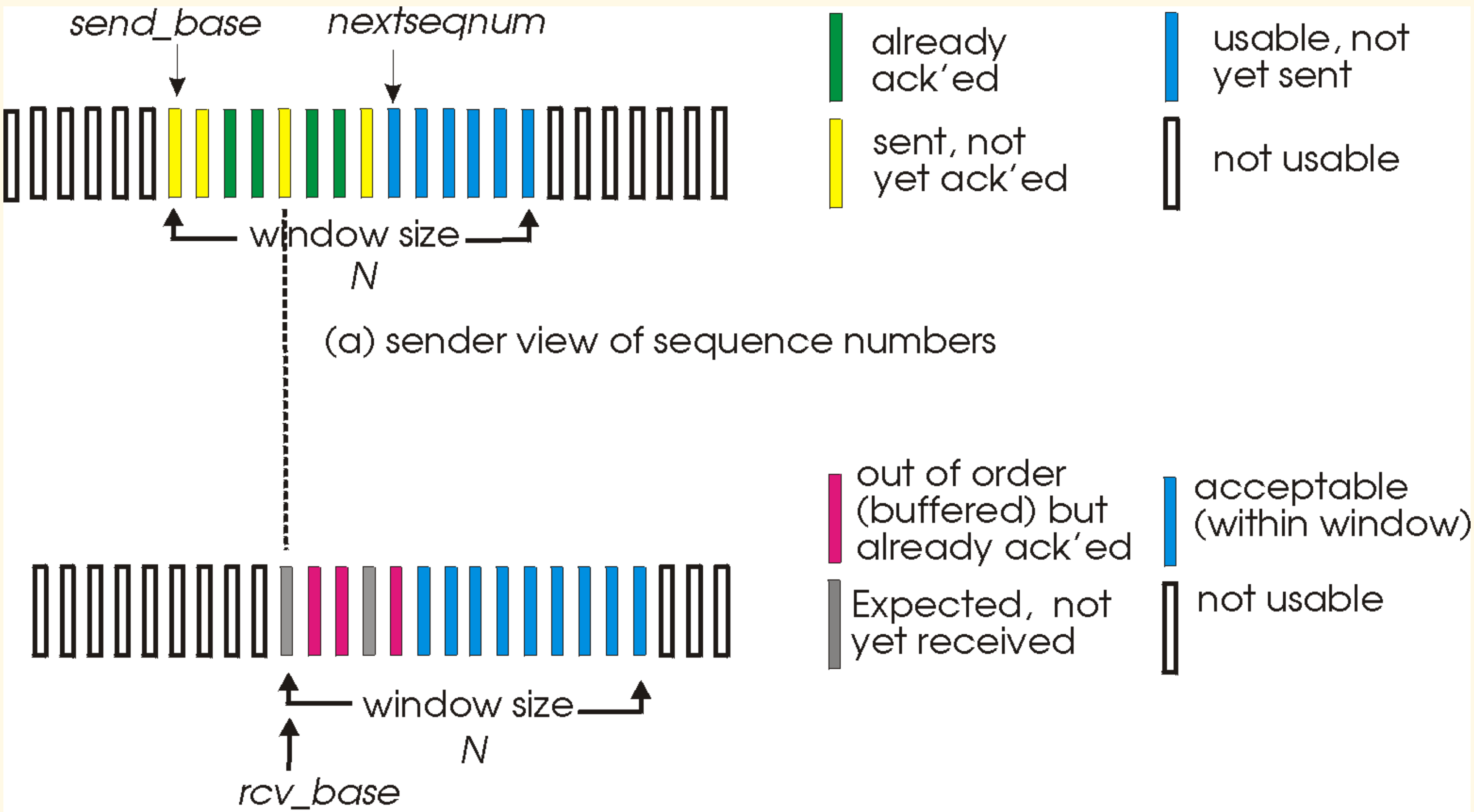
- may generate duplicate ACKs
- need only remember `expectedseqnum`

- **out-of-order pkt:**
  - discard (don't buffer) -> **no receiver buffering!**
  - Re-ACK pkt with highest in-order seq #

Selective Repeat

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

Selective Repeat
Sender, Receiver Windows



(a) sender view of sequence numbers

(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

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

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

receiver

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

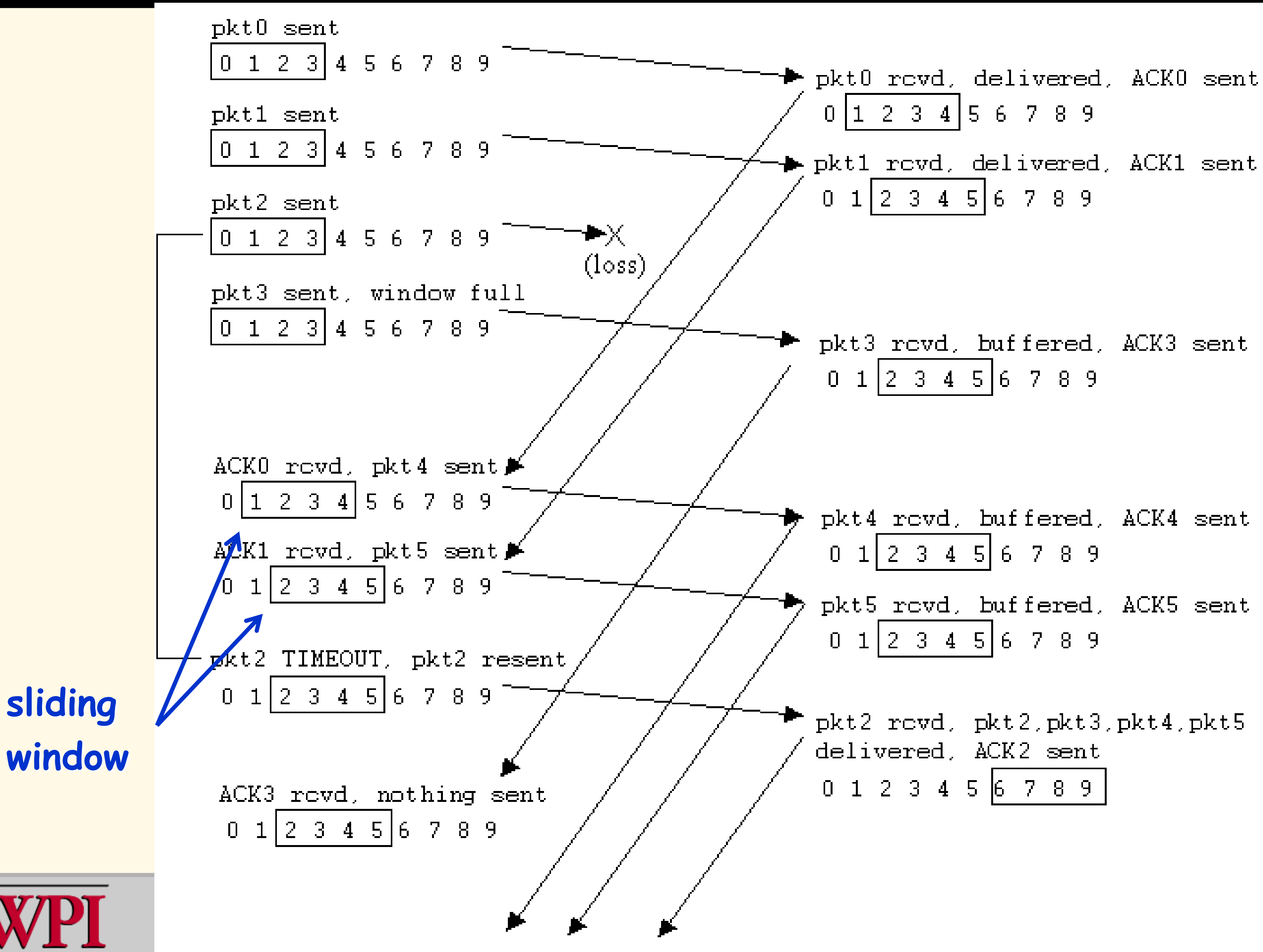
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

Selective Repeat in Action



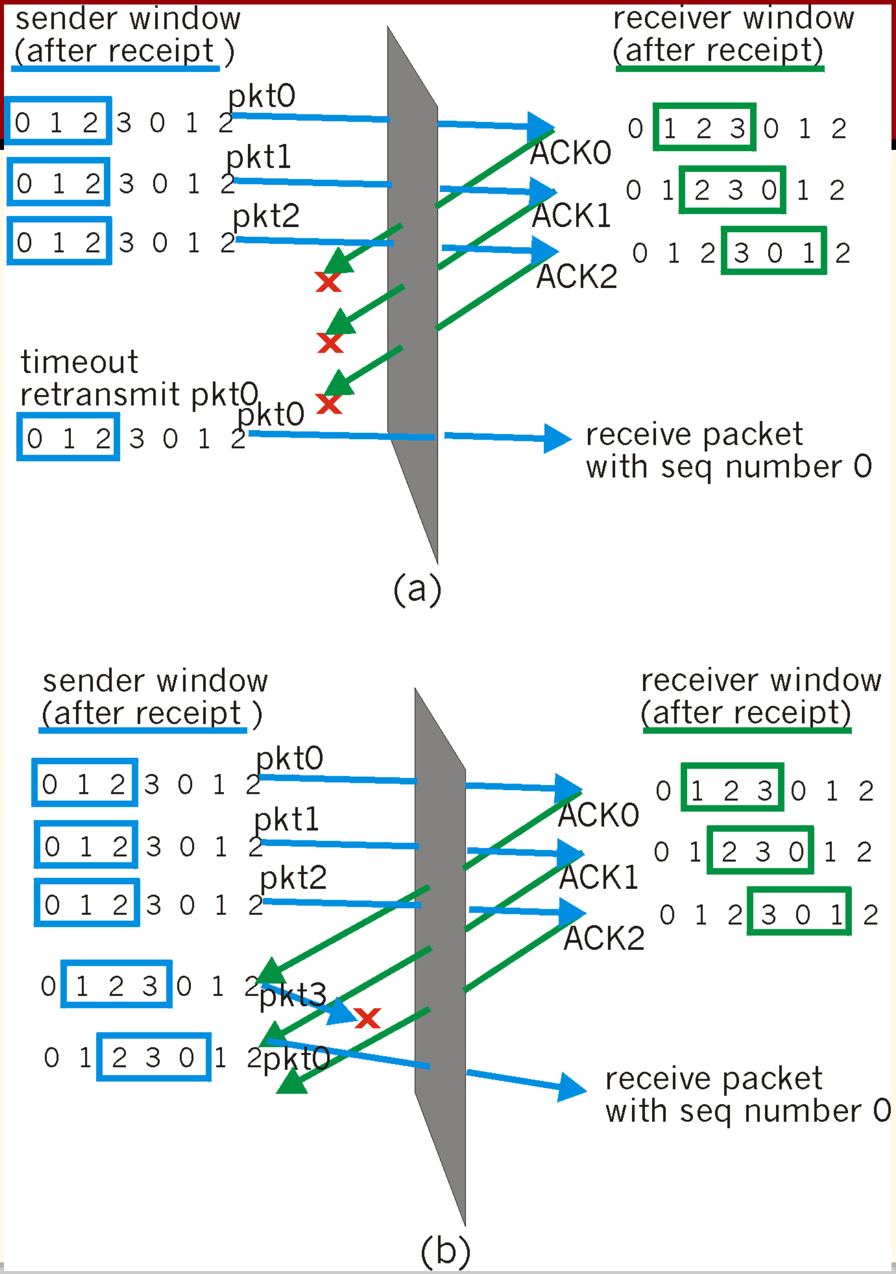
Selective Repeat Dilemma

Example:

- seq #'s: 0, 1, 2, 3
- window size=3

- receiver sees no difference in two scenarios!
- incorrectly passes duplicate data as new in (a)

Q: What is the required relationship between seq # size and window size?



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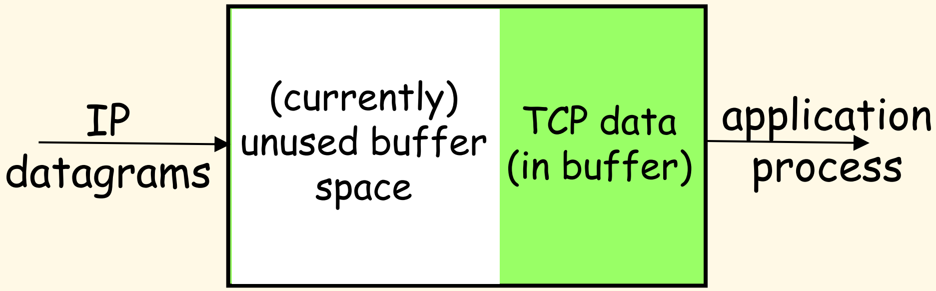
TCP Flow Control

- **receive side of TCP connection has a receive buffer:**

flow control

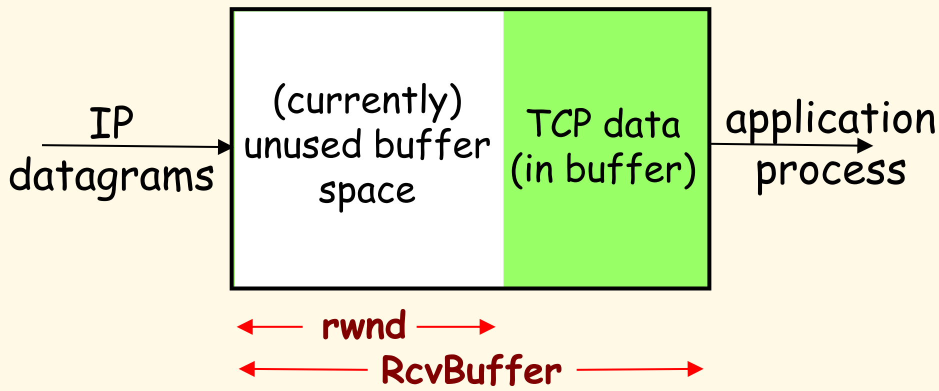
sender won't overflow receiver's buffer by transmitting too much, too fast



- **speed-matching service: matching send rate to receiving application's drain rate.**

- app process may be slow at reading from buffer.

TCP Flow Control: how it works



(suppose TCP receiver discards out-of-order segments)

- unused buffer space:

= `rwnd`

= `RcvBuffer-[LastByteRcvd - LastByteRead]`

- receiver: advertises unused buffer space by including `rwnd` value in segment header
- sender: limits # of unACKed bytes to `rwnd`
  - guarantees receiver's buffer doesn't overflow.
- `rwnd` known as the receiver's advertised window.

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