CS 755 – System and Network Architectures and Implementation

Module 3 - Transport

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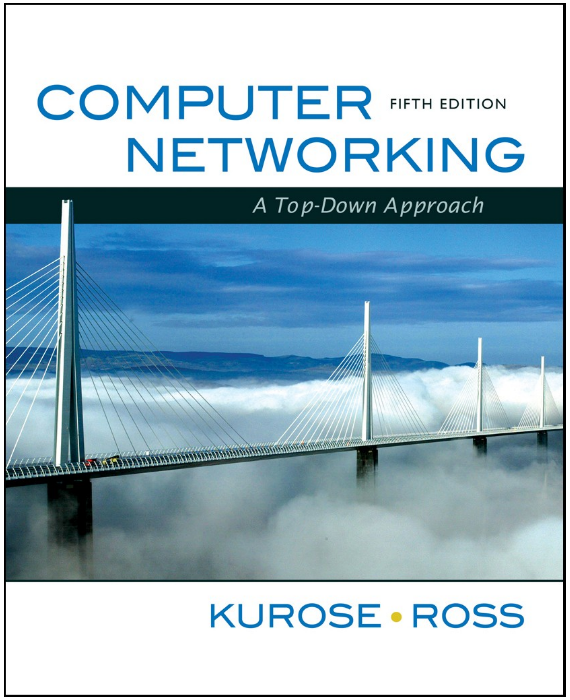

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Computer Networking: A Top Down Approach
5th edition.
Jim Kurose, Keith Ross
Addison-Wesley, April 2009.

Overview

- multiplexing, virtual channel
- reliable transmission
- flow and congestion control
- connection management and semantics

Networks – Review

- network graph
- goal: facilitate any-to-any communication
- main concerns
  - routing
  - addressing
  - scalability
- virtualization

Transport

- virtual channel
- end-to-end transmission performance
  - reliable transmission
  - rate control
- connection management

Hop by Hop vs. End to End?

- some services only hop by hop
  - delay control
  - throughput guarantees
- others also end to end
  - multiplexing
  - loss control – reliable transmission
  - rate control
- principle: if possible, use end to end

Multiplexing

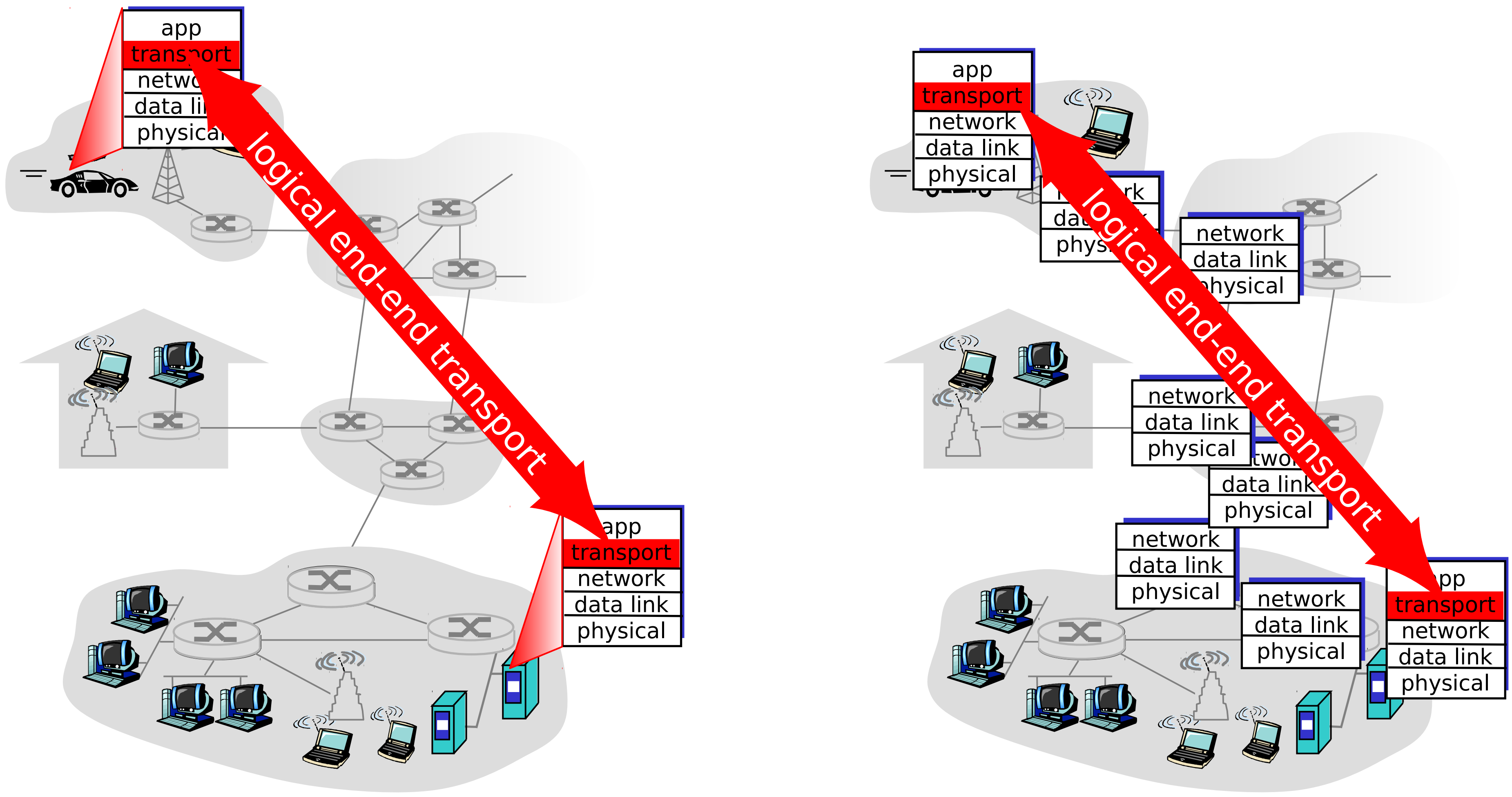
- multiple logical sessions over same channel
- here: IP connectivity provides “virtual channel”
- transport session also provides “virtual channel”
- multiplexing
  - encapsulation / stacking of multiplex label
- demultiplexing
  - forwarding according to multiplex label
  - decapsulation / remove multiplex label

Multiplexing

- Internet addressing convention:
  - IP address – network node address
  - transport *port* – transport session identifier

- Other Approaches
  - virtual circuit – integrated with network layer
  - hop-by-hop transport service
  - session identified locally by *virtual circuit identifier*

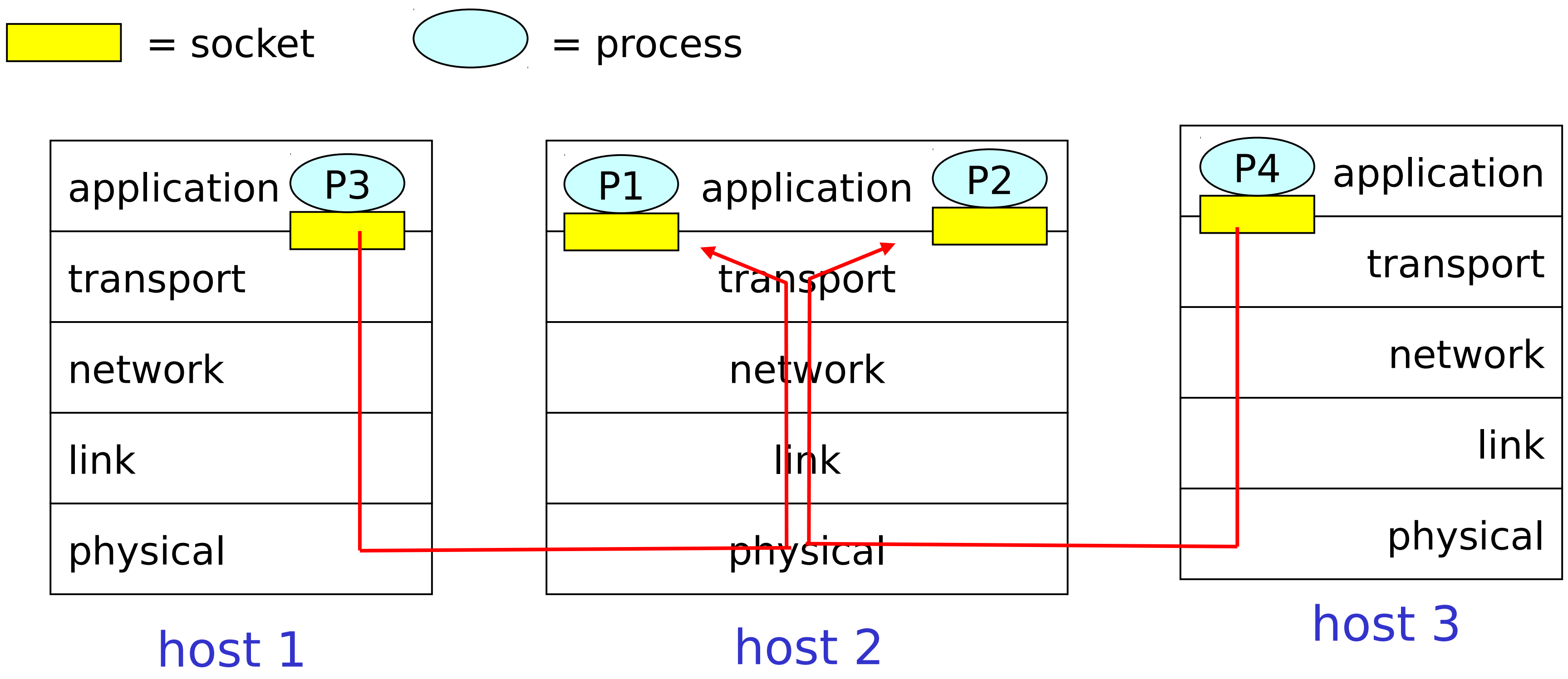
Layered Service



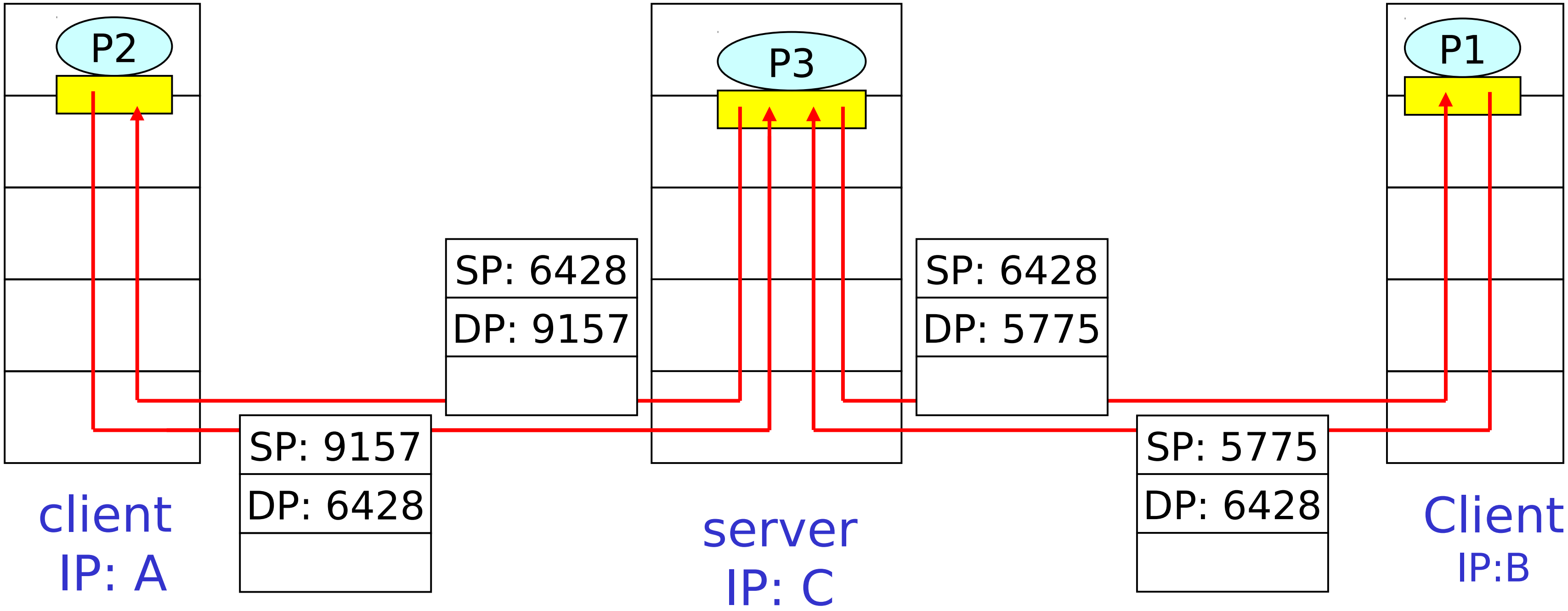
Operating System Integration

- OS implements transport protocol
  - handles asynchronous execution
  - provides send/receive queue at *socket*
    - socket: named communication endpoint
- OS *system calls*
  - create/remove sockets
  - establish names, connections
  - send/receive data

Multiplexing – Multiple Sockets

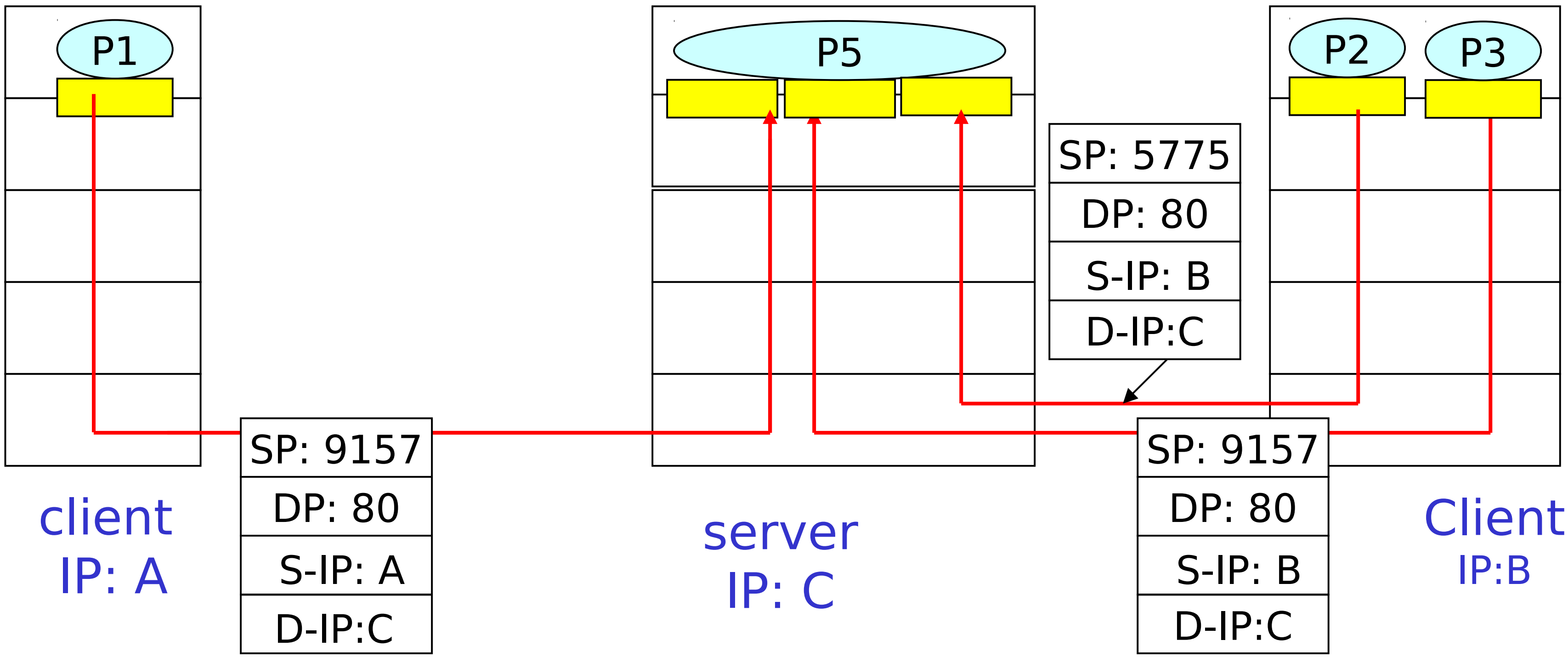


Multiplexing – Single Socket



SP provides “return address”

Multiplexing – Multiple Connections

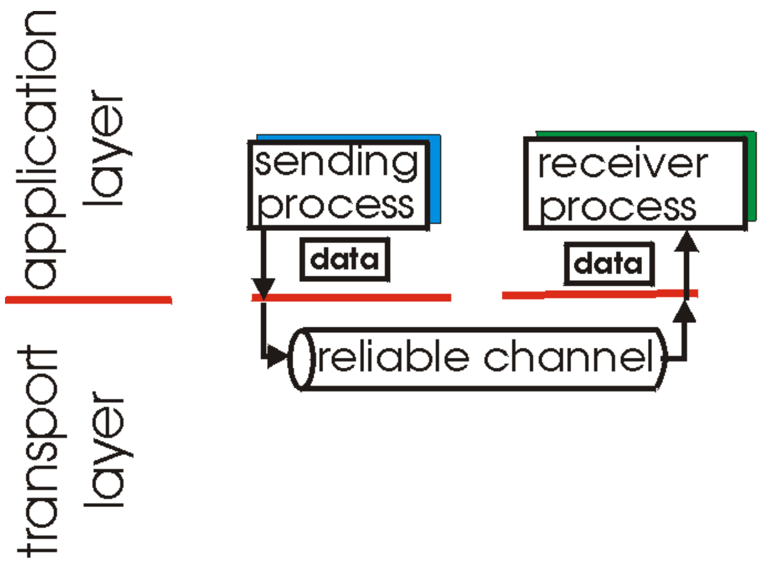


Reliable Transmission

- use acknowledgements to indicate receipt
  - sender knows data has been received
  - BUT: two-army problem
- look at functionality first, then performance

Principles of Reliable data transfer

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

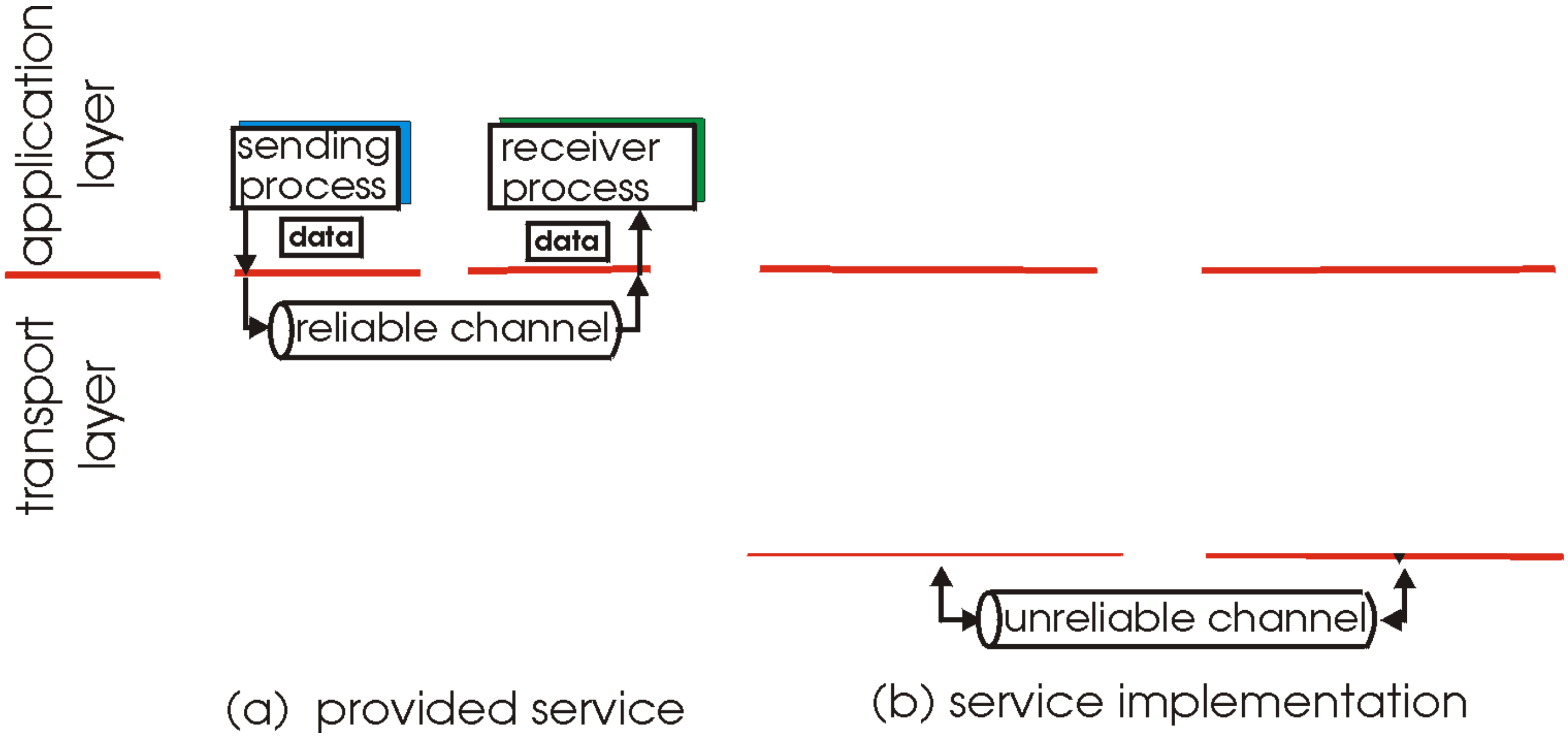


(a) provided service

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

Principles of Reliable data transfer

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



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Principles of Reliable data transfer

- important in app., transport, link layers
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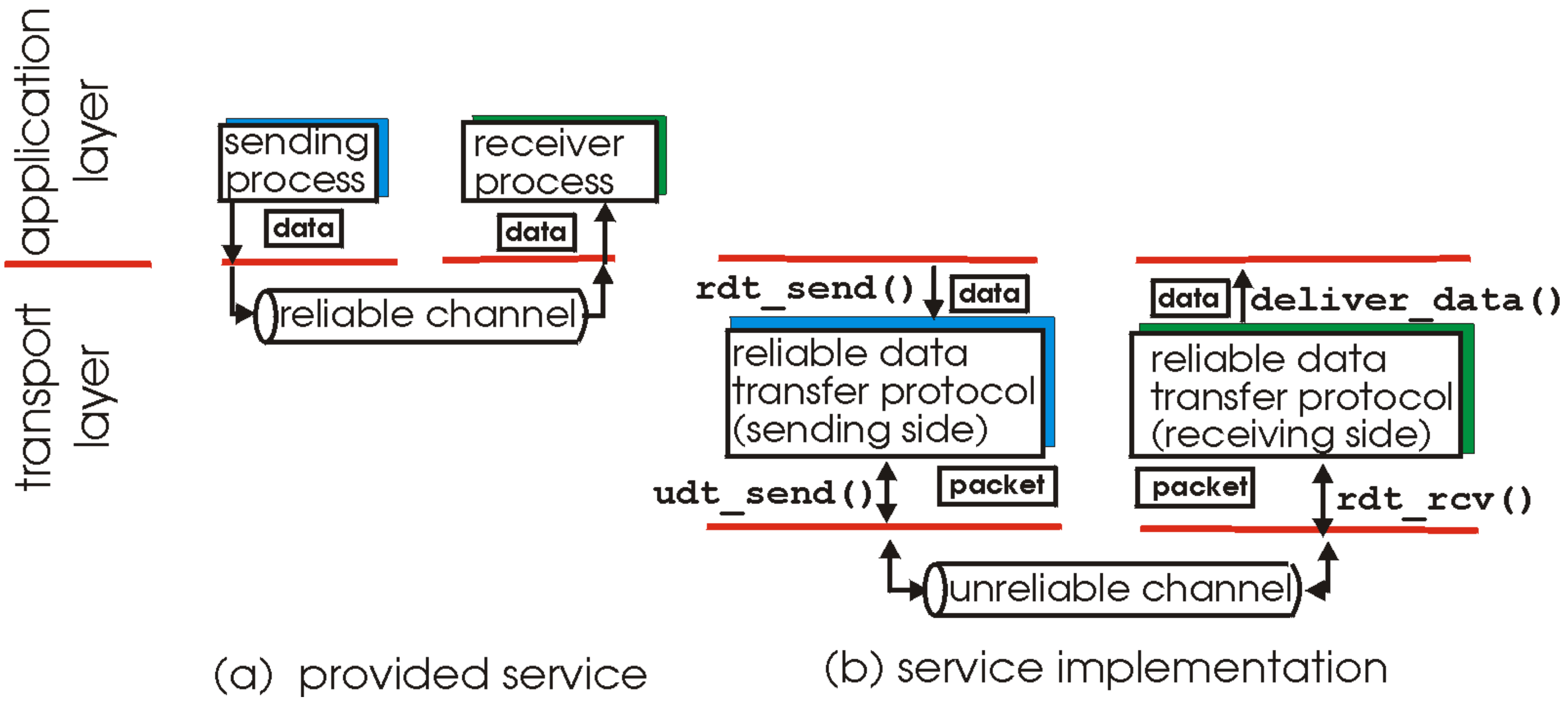
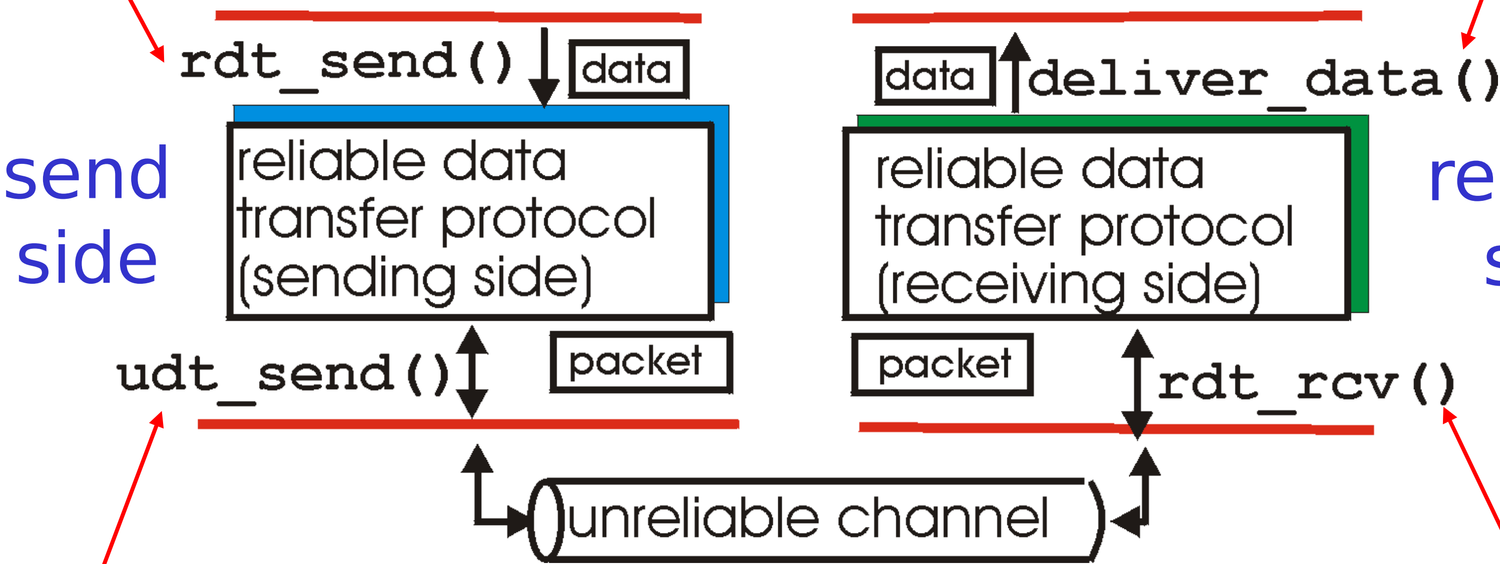


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

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 layer



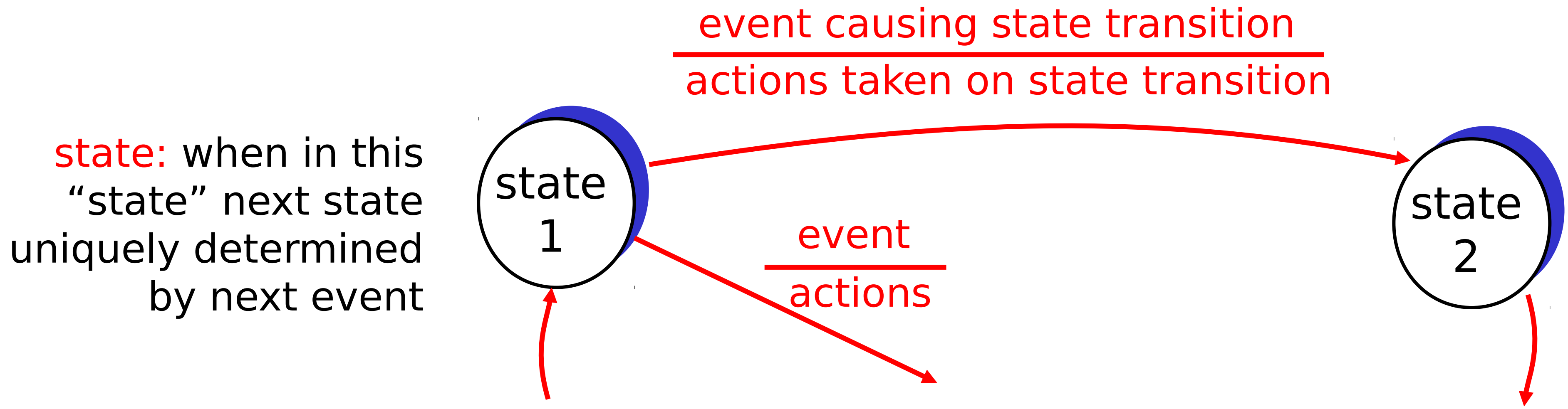
udt_send(): called by rdt, to transfer packet over unreliable channel to receiver

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

Reliable data transfer: getting started

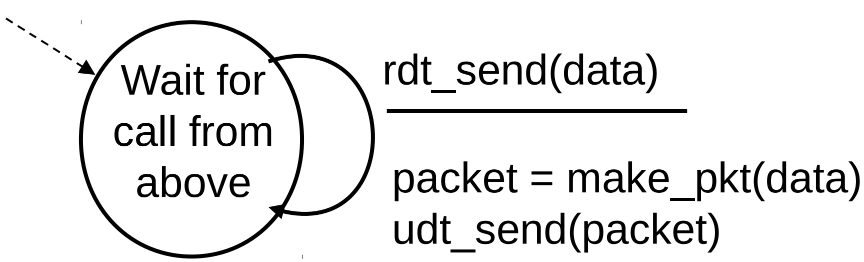
We'll:

- 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

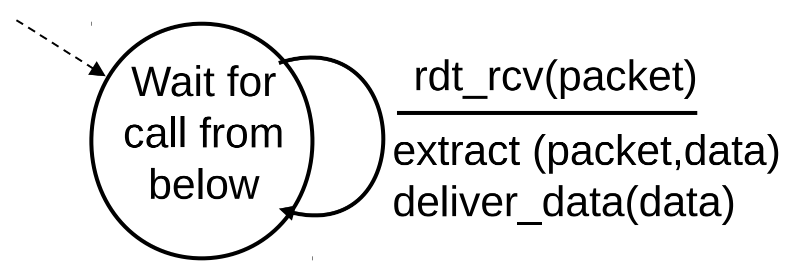


Rdt1.0: reliable transfer over a reliable channel

- 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



sender

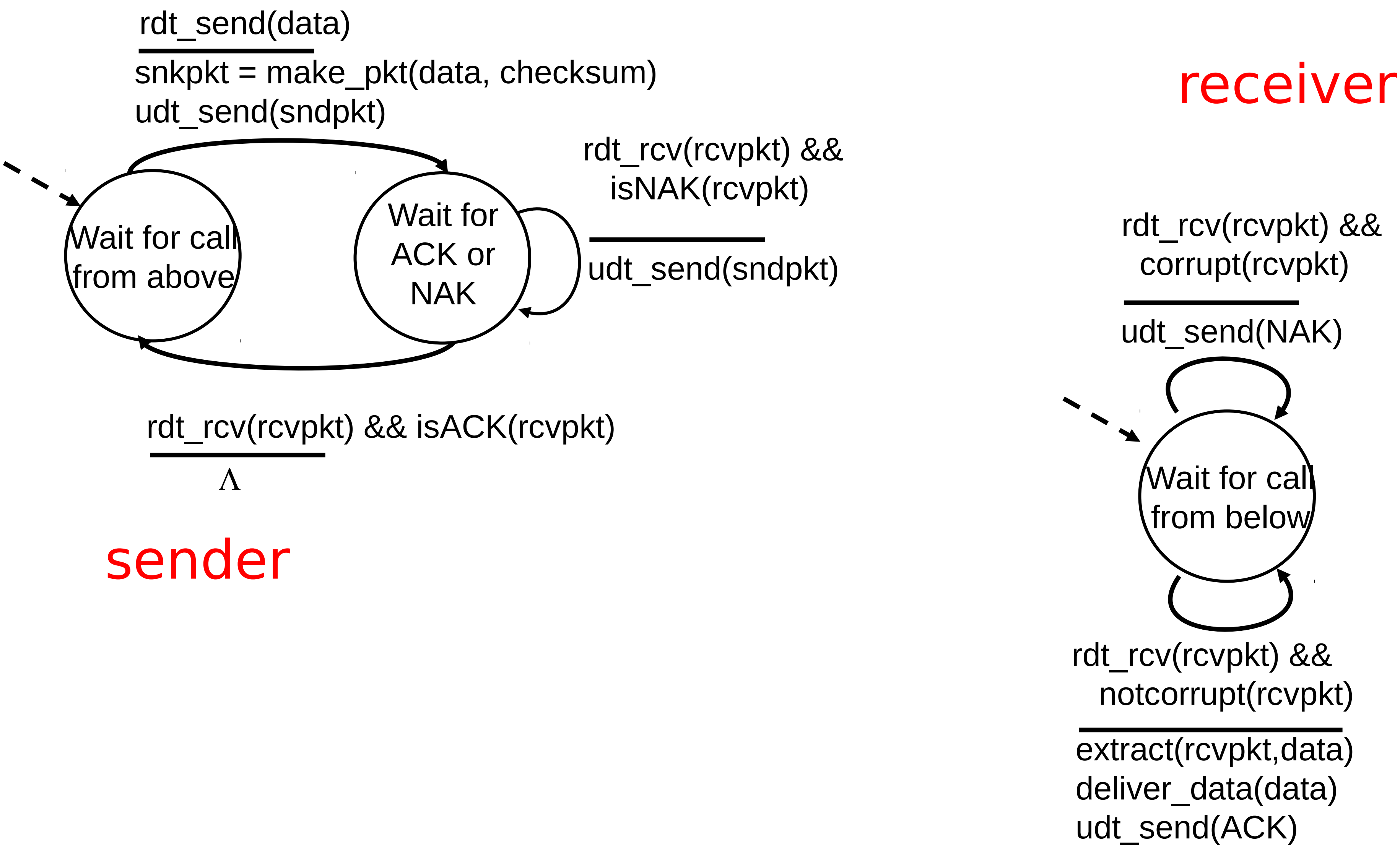


receiver

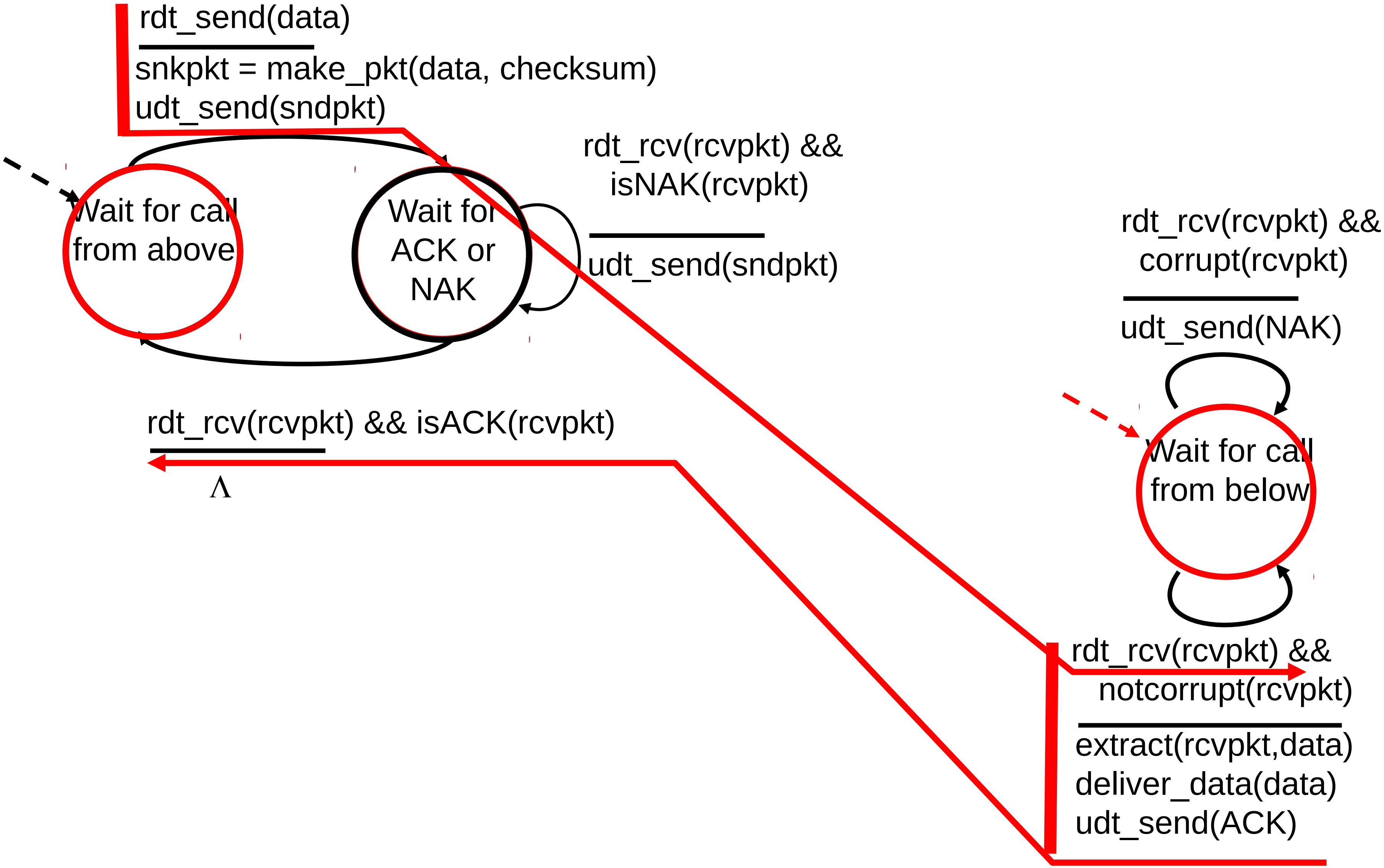
Rdt2.0: channel with bit errors

- 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

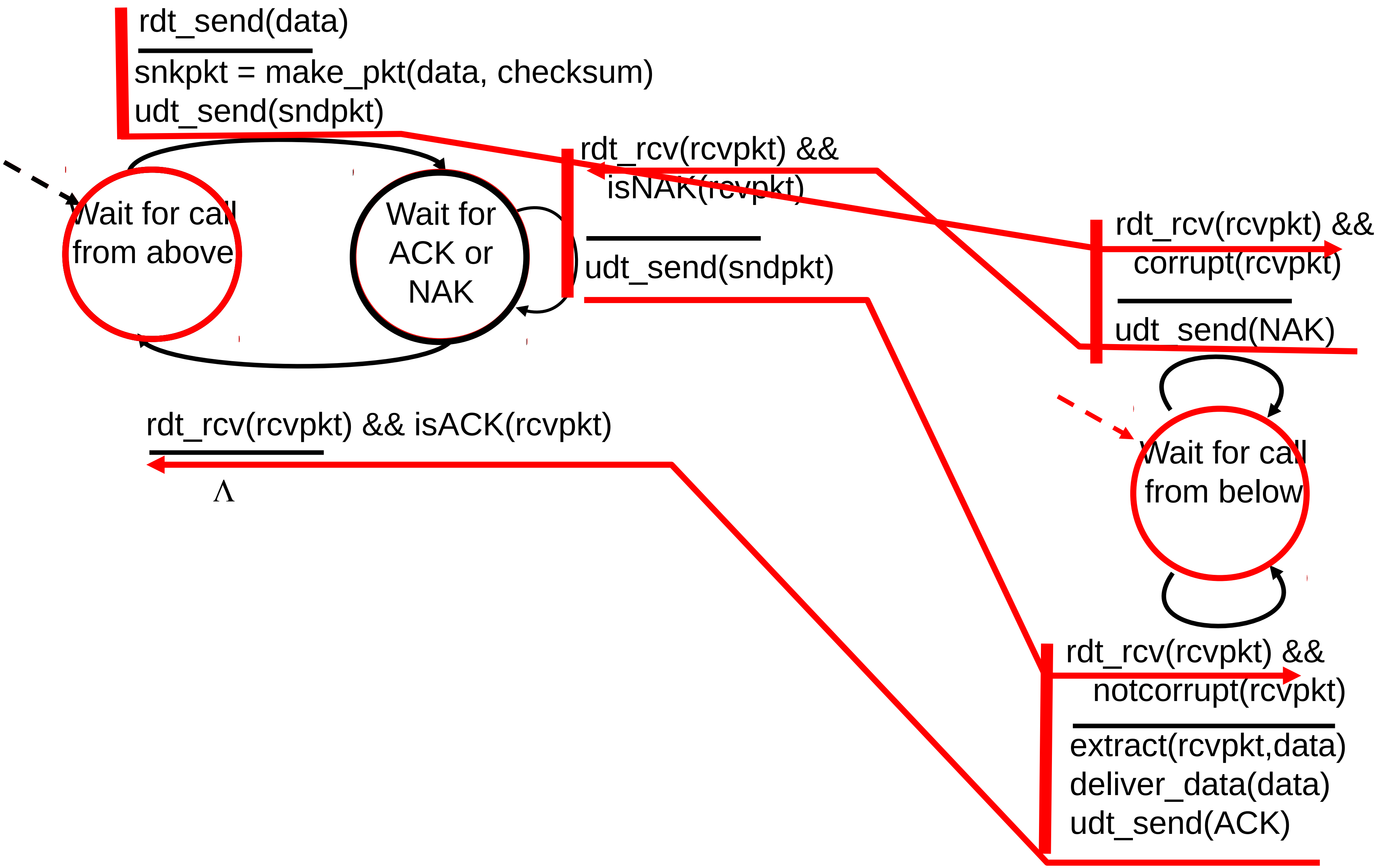
rdt2.0: FSM specification



rdt2.0: operation with no errors



rdt2.0: error scenario



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

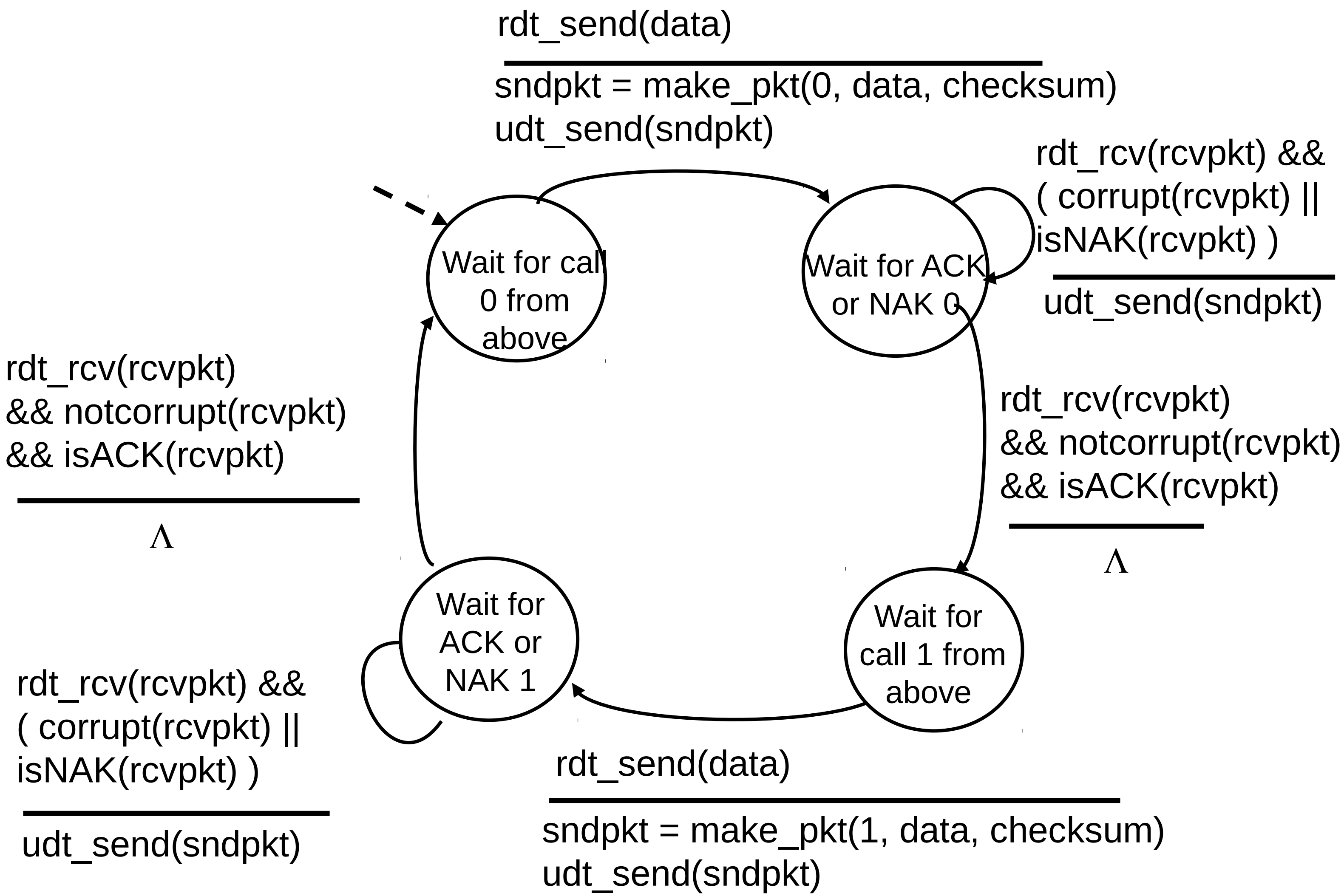
Handling duplicates:

- sender retransmits current pkt if ACK/NAK garbled
- sender adds *sequence number* to each pkt
- receiver discards (doesn't deliver up) duplicate pkt

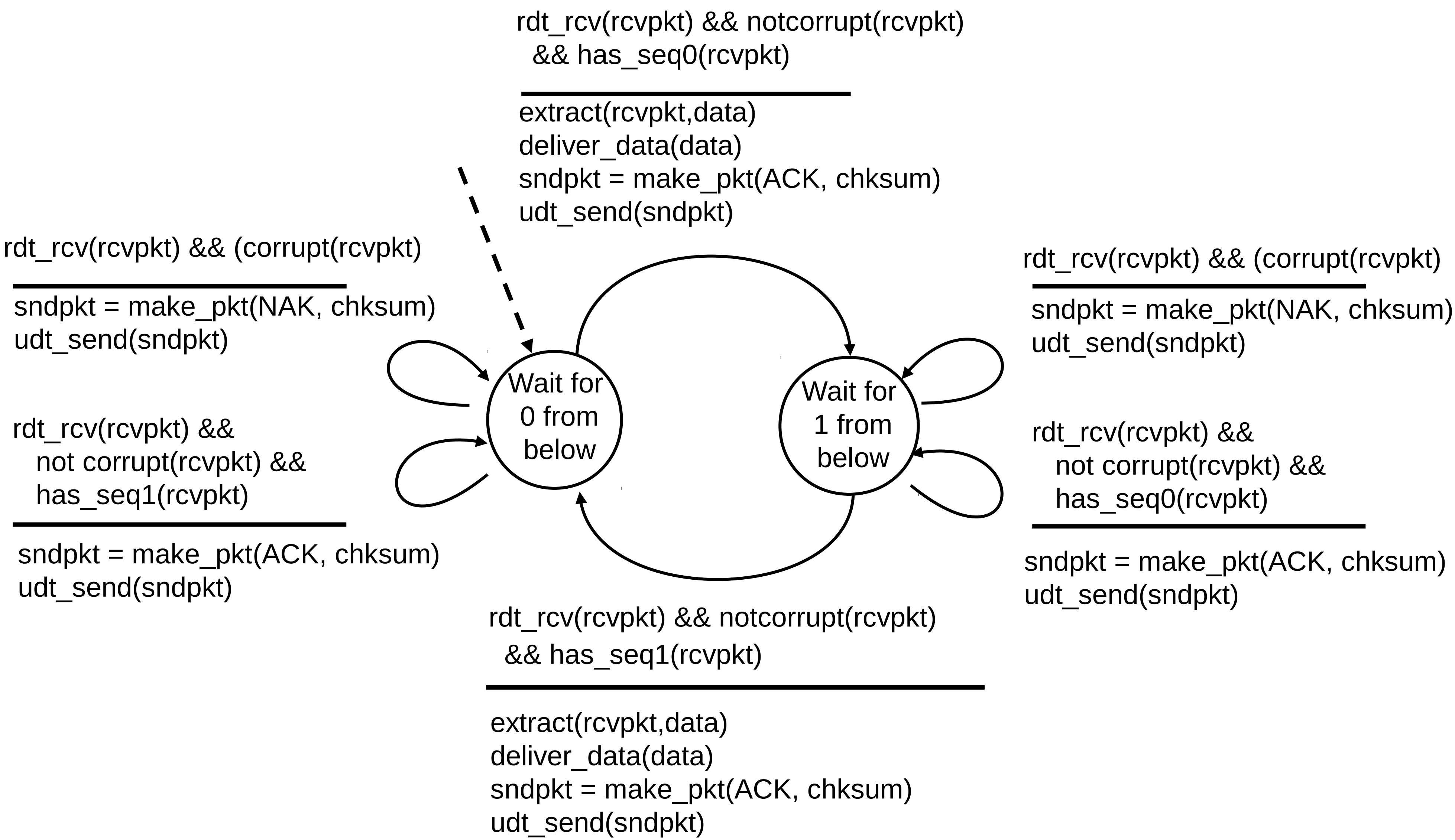
stop and wait

Sender sends one packet, then waits for receiver response

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



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 "current" pkt has 0 or 1 seq. #

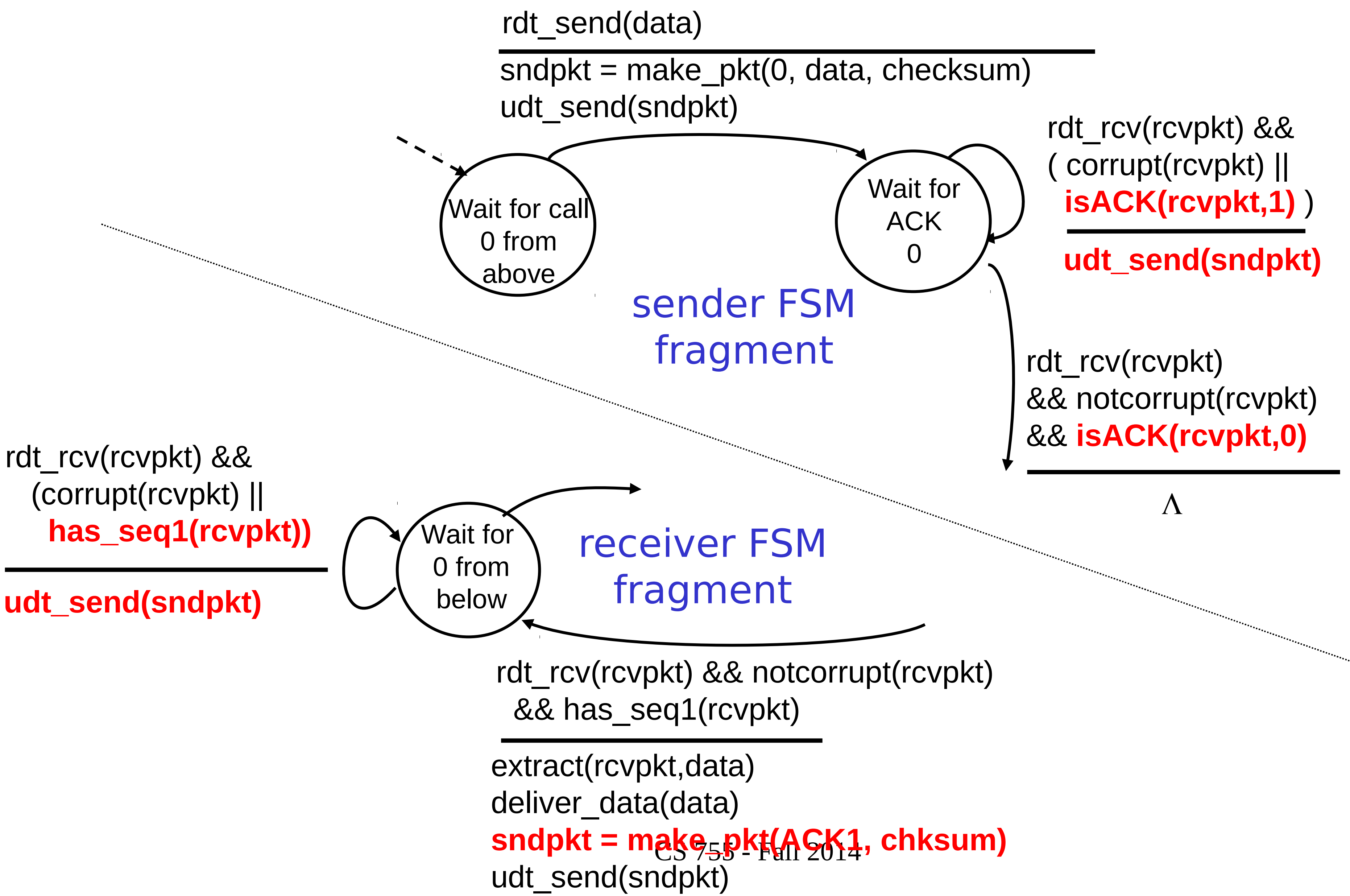
Receiver:

- 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*

rdt2.2: sender, receiver fragments



rdt3.0: channels with errors *and* loss

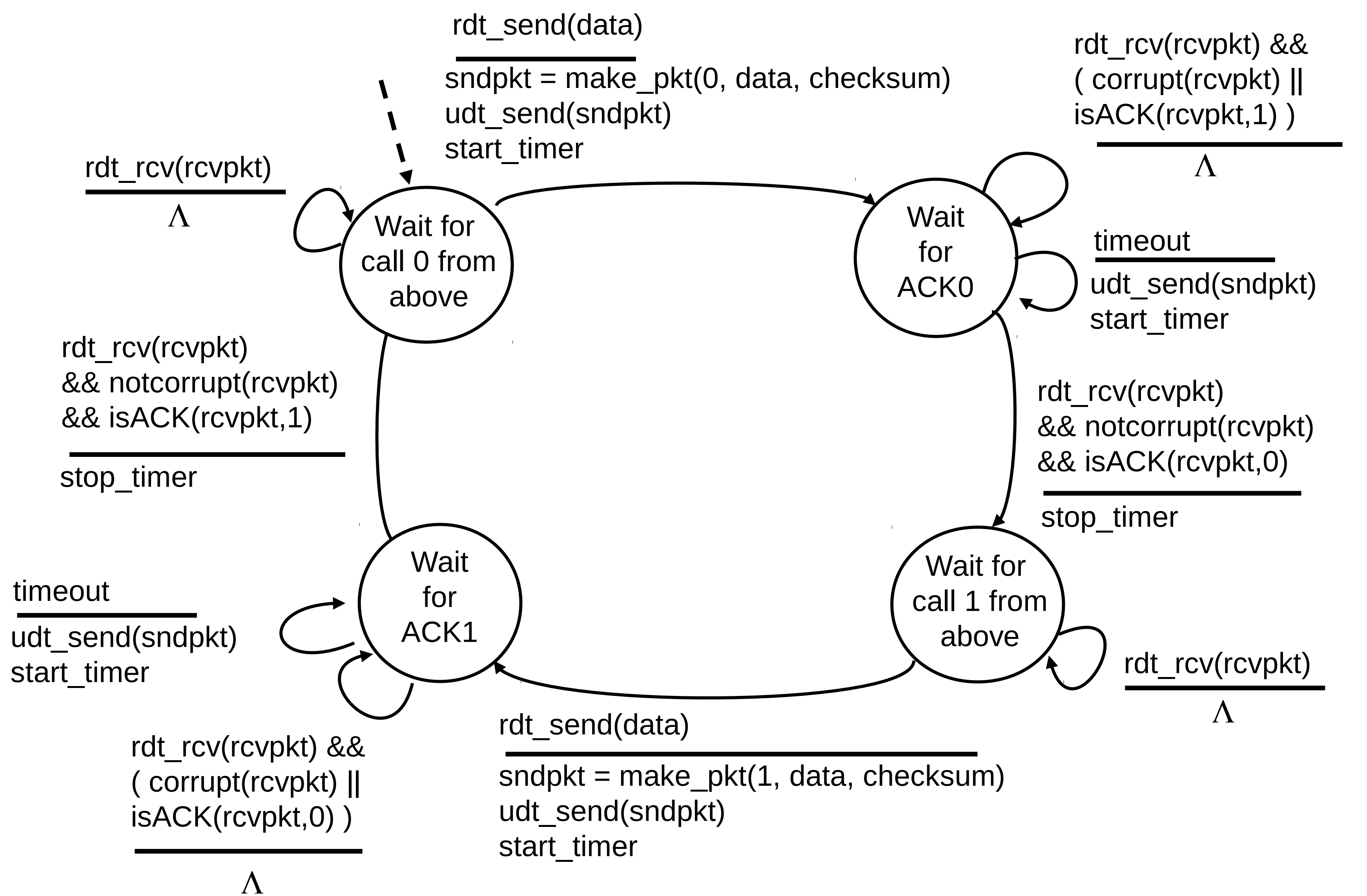
New assumption: underlying channel can also lose packets (data or ACKs)

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

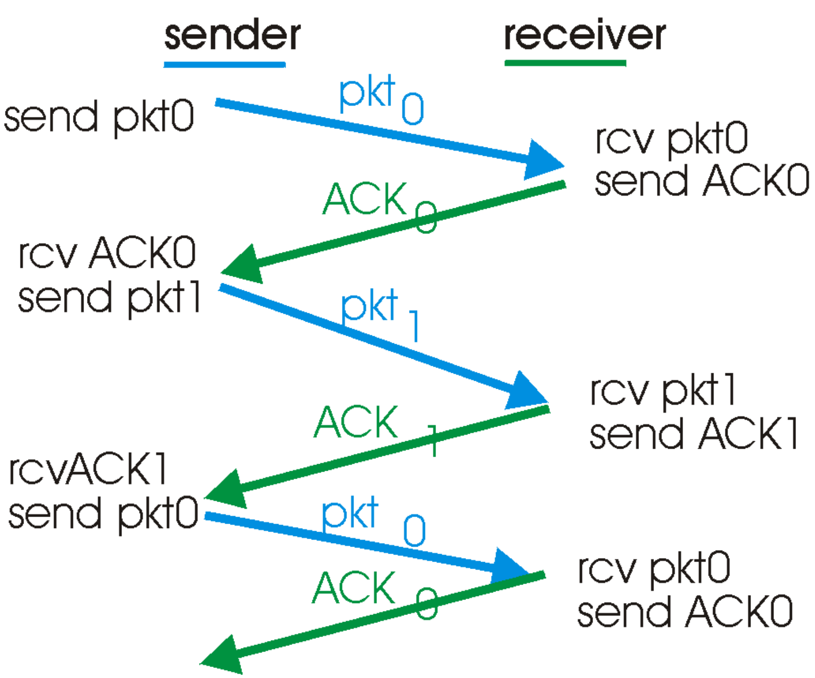
Approach: sender waits "reasonable" amount of time for ACK

- 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

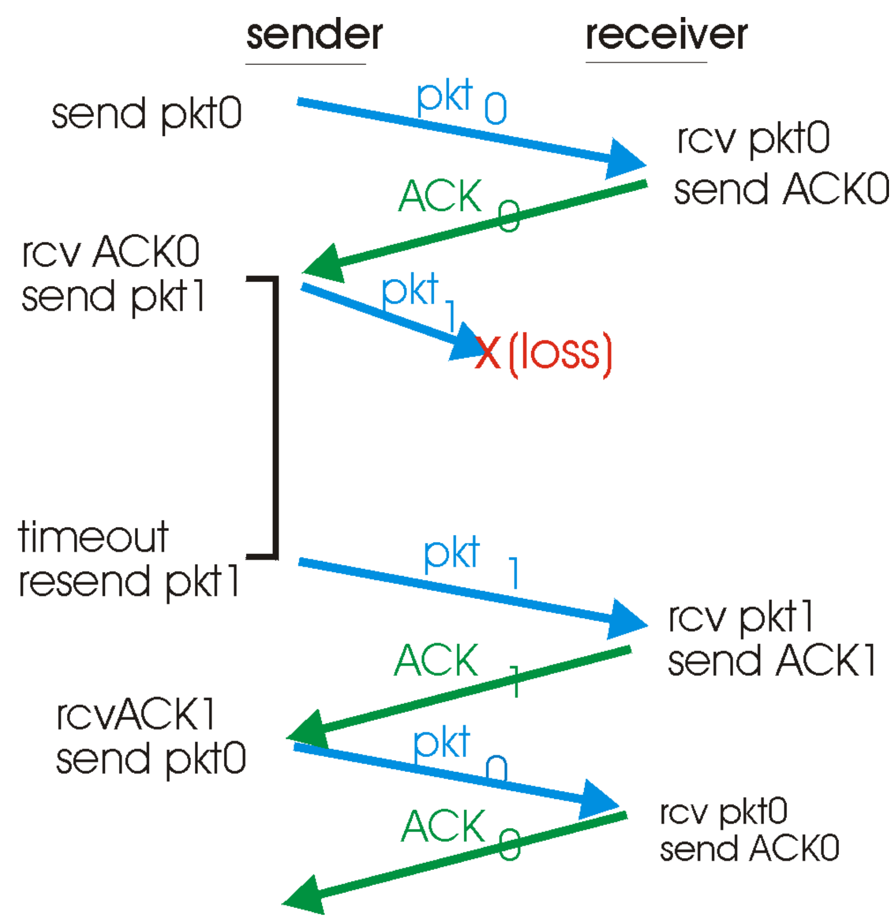
rdt3.0 sender



rdt3.0 in action

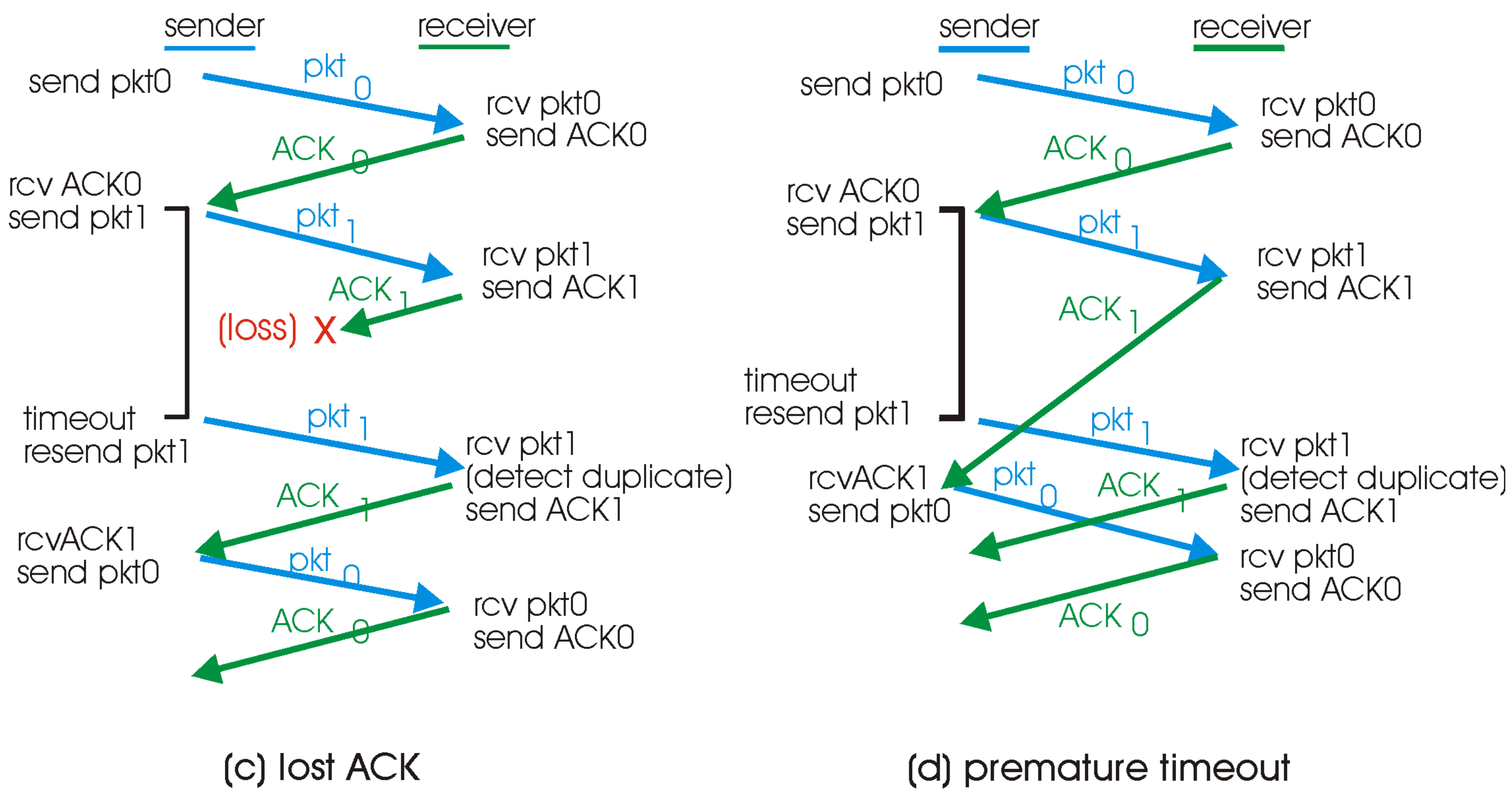


(a) operation with no loss



(b) lost packet

rdt3.0 in action



Performance of rdt3.0

- rdt3.0 works, but performance stinks
- example: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

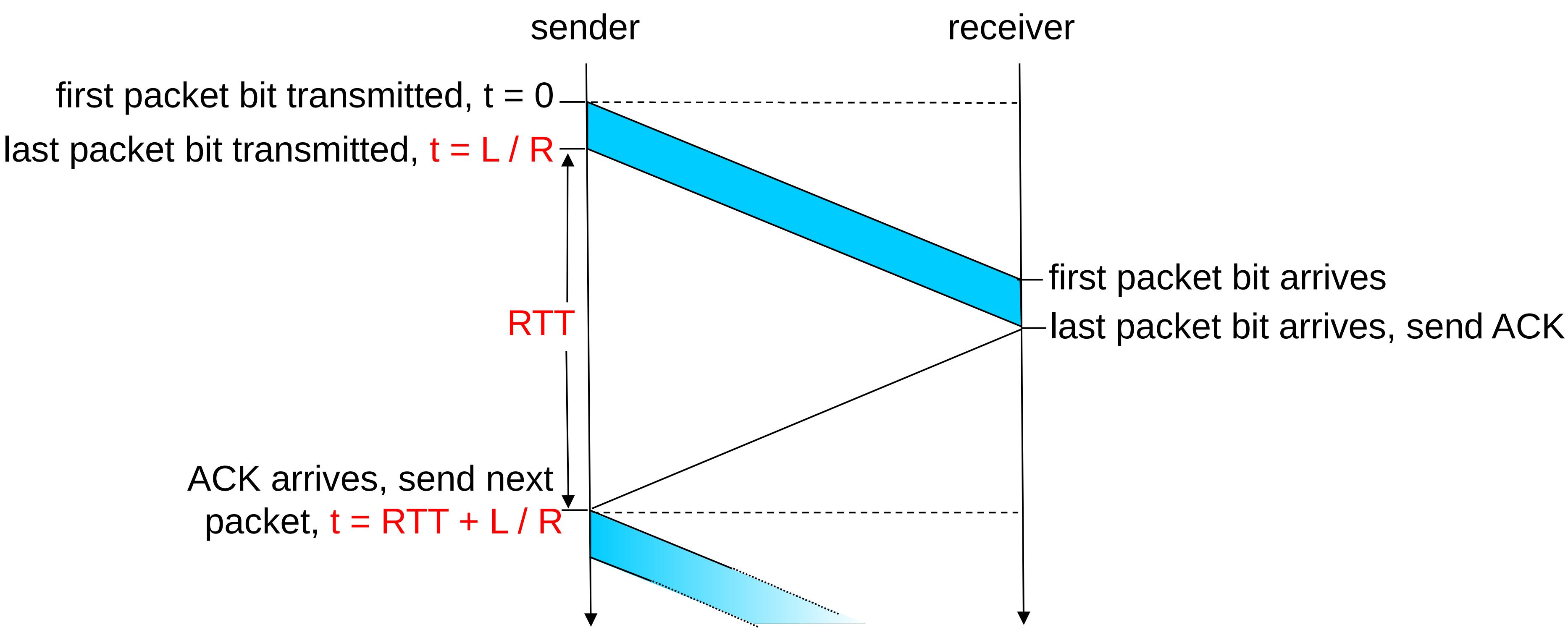
$$d_{trans} = \frac{L}{R} = \frac{8000\text{bits}}{10^9\text{bps}} = 8\,\text{microseconds}$$

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

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

- 1KB pkt every 30 msec -> 33kB/sec thruput over 1 Gbps link
- network protocol limits use of physical resources!

rdt3.0: stop-and-wait operation

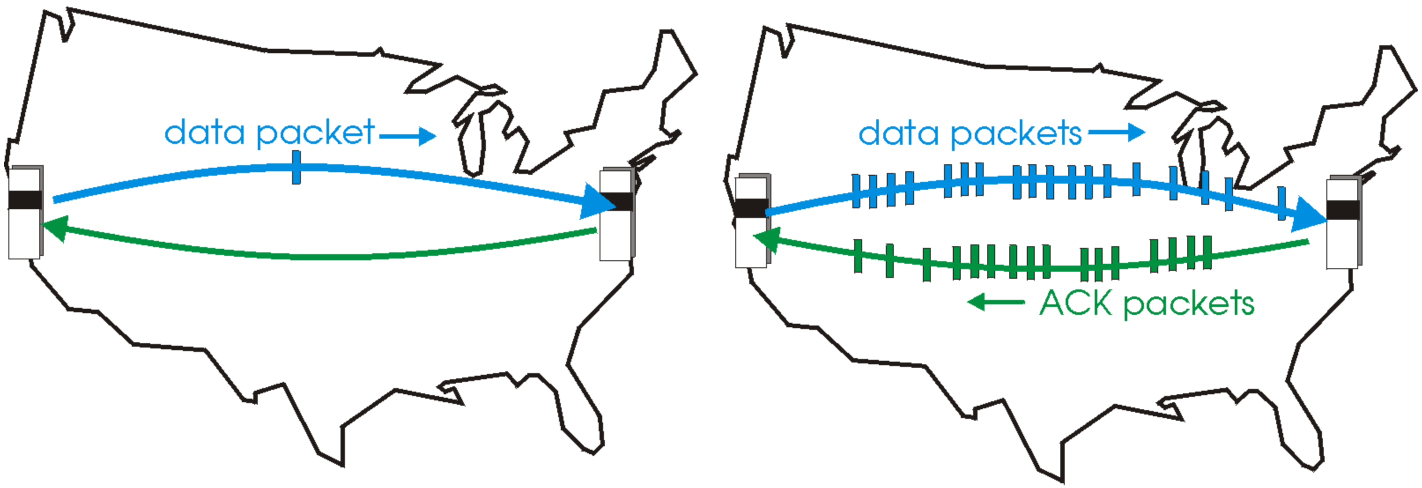


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

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

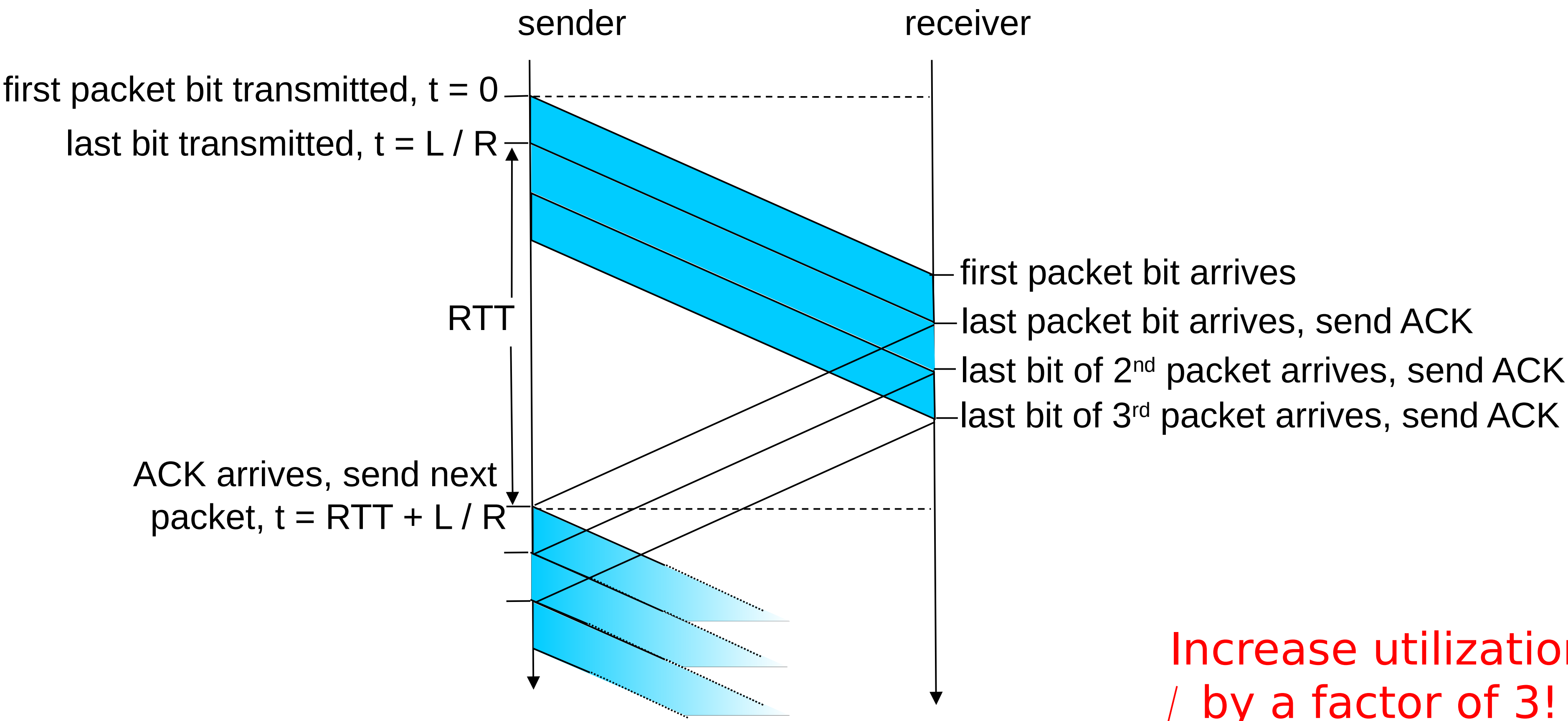


(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

- Two generic forms of pipelined protocols: *go-Back-N, selective repeat*

Pipelining: increased 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

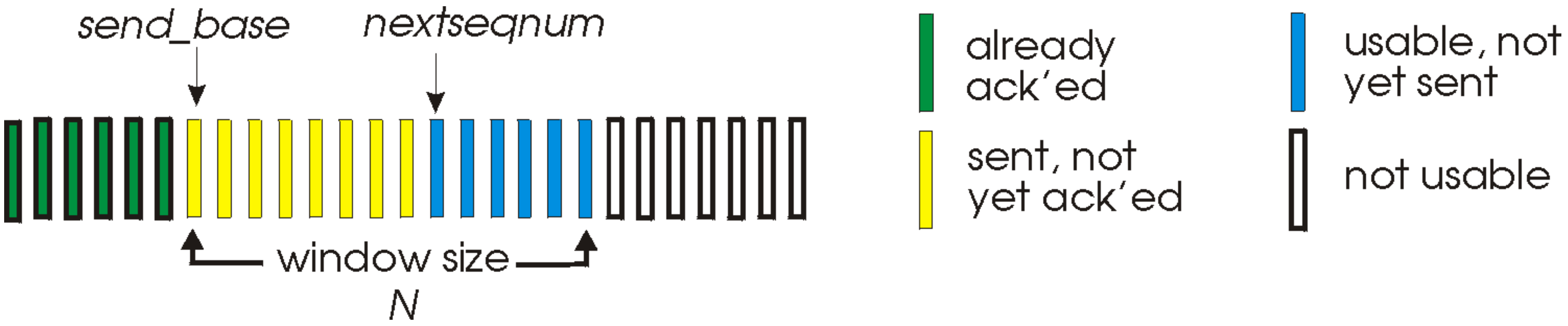
Selective Repeat: overview

- *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

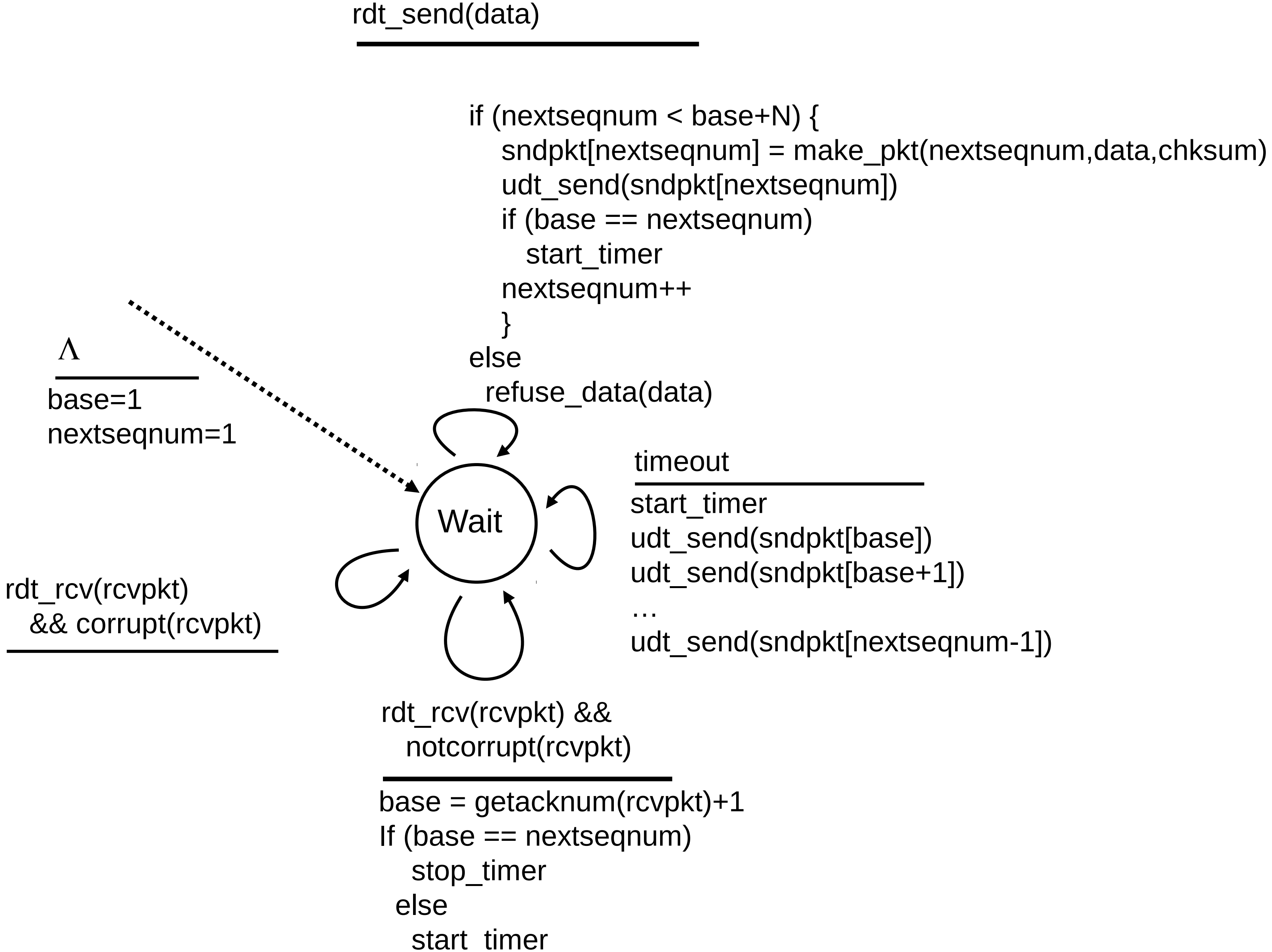
Sender:

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

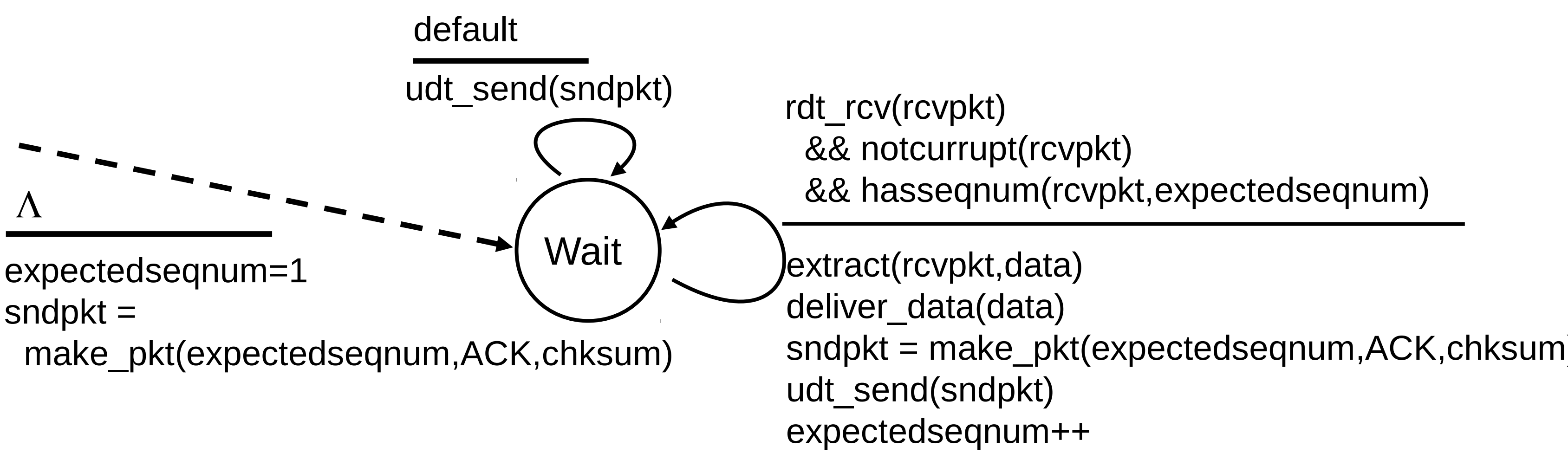


- ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"
  - may receive duplicate ACKs (see receiver)
- timer for each in-flight pkt
- *timeout(n):* retransmit pkt n and all higher seq # pkts in window

GBN: sender extended FSM



GBN: receiver extended FSM

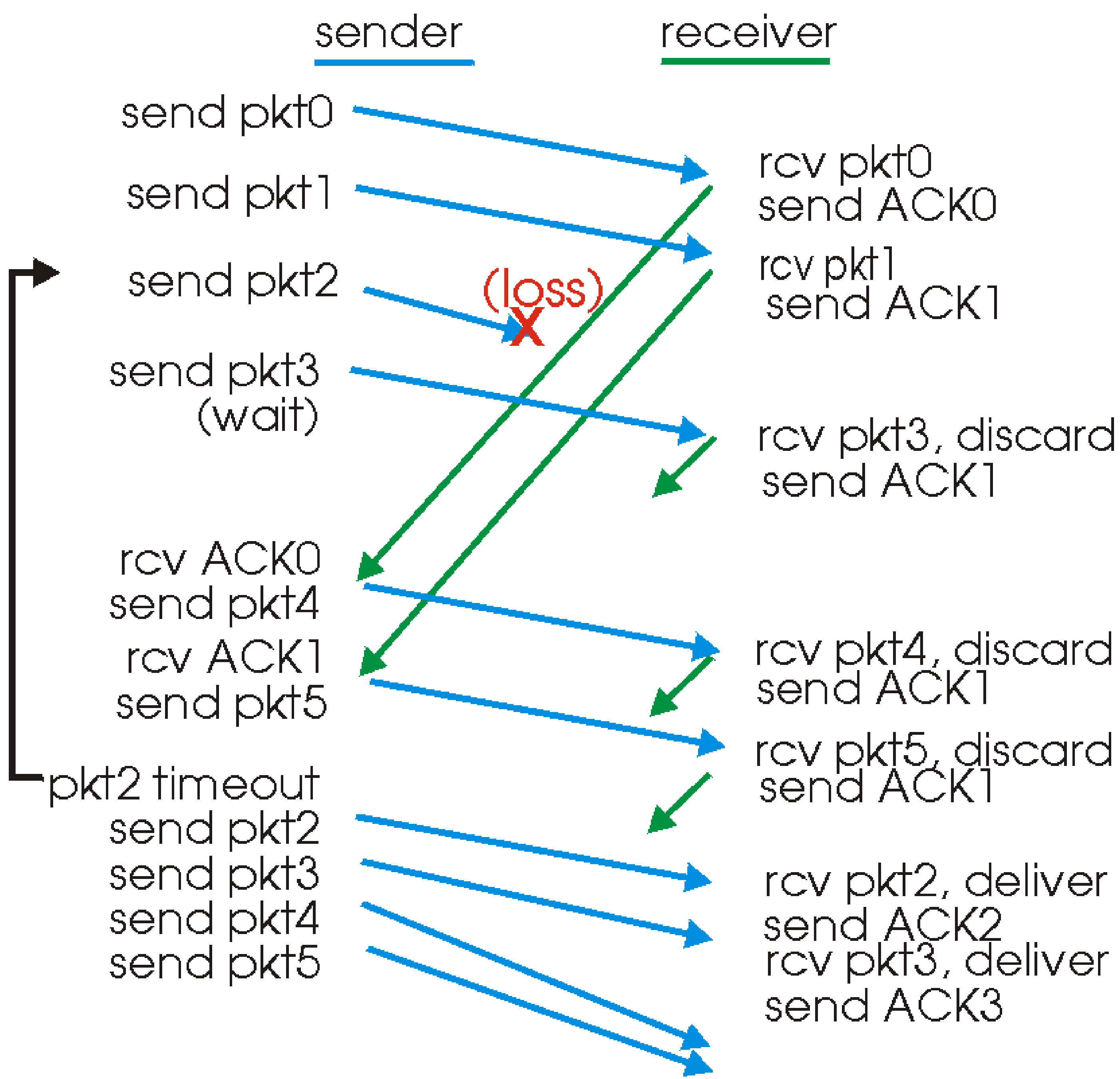


ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- 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 #

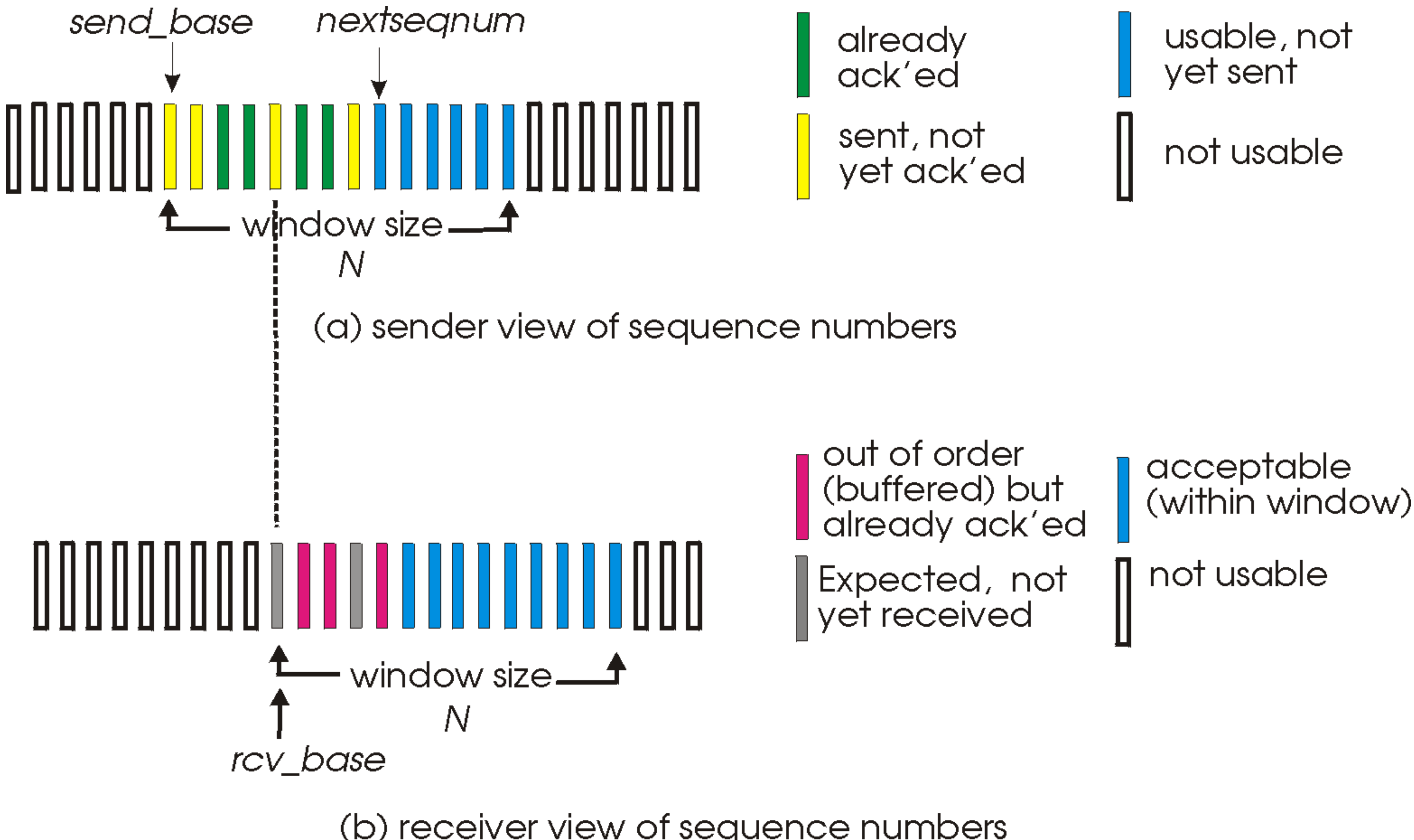
GBN in action



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
  - again limits seq #s of sent, unACKed pkts

Selective repeat: sender, receiver windows



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

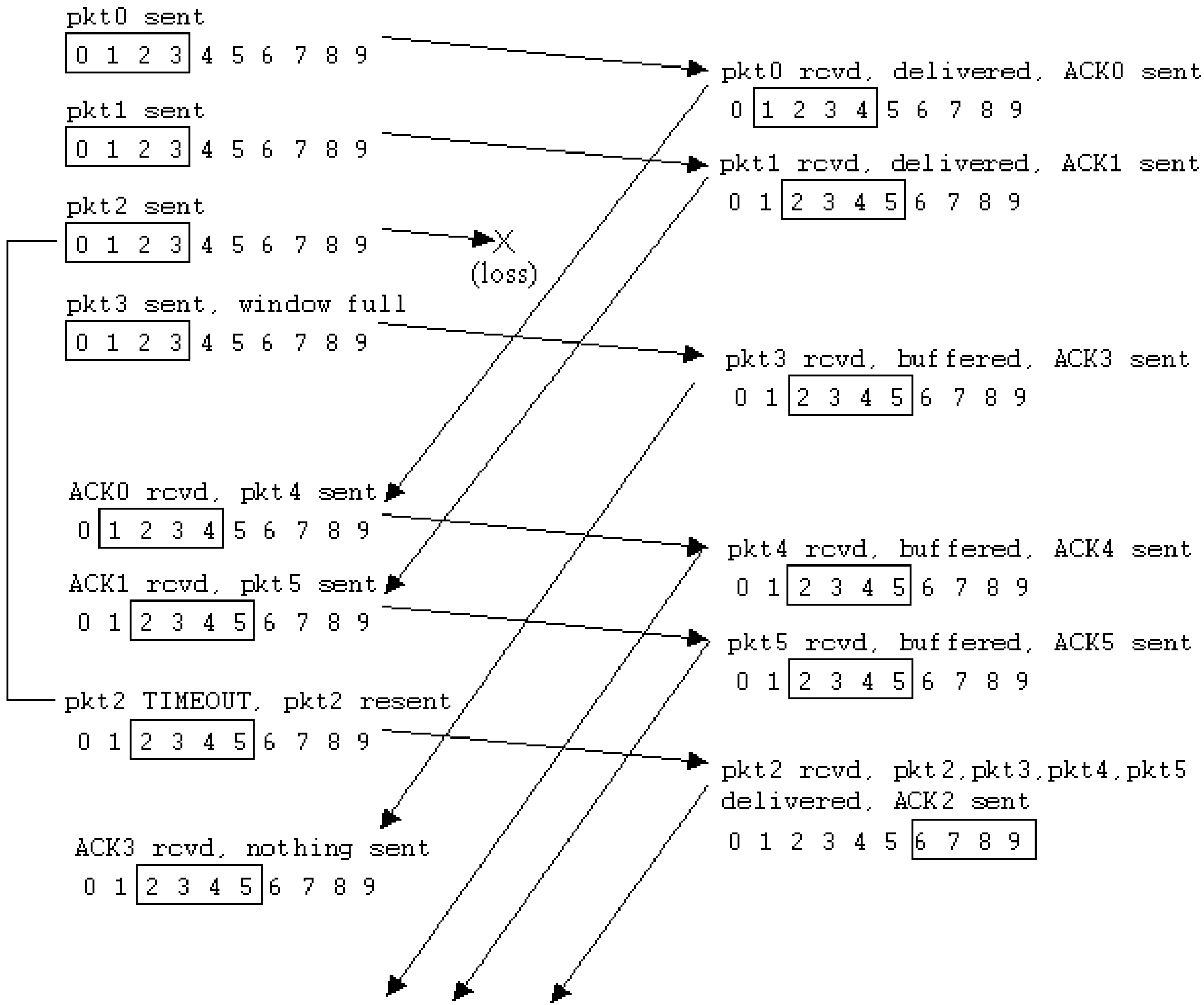
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

Selective repeat in action



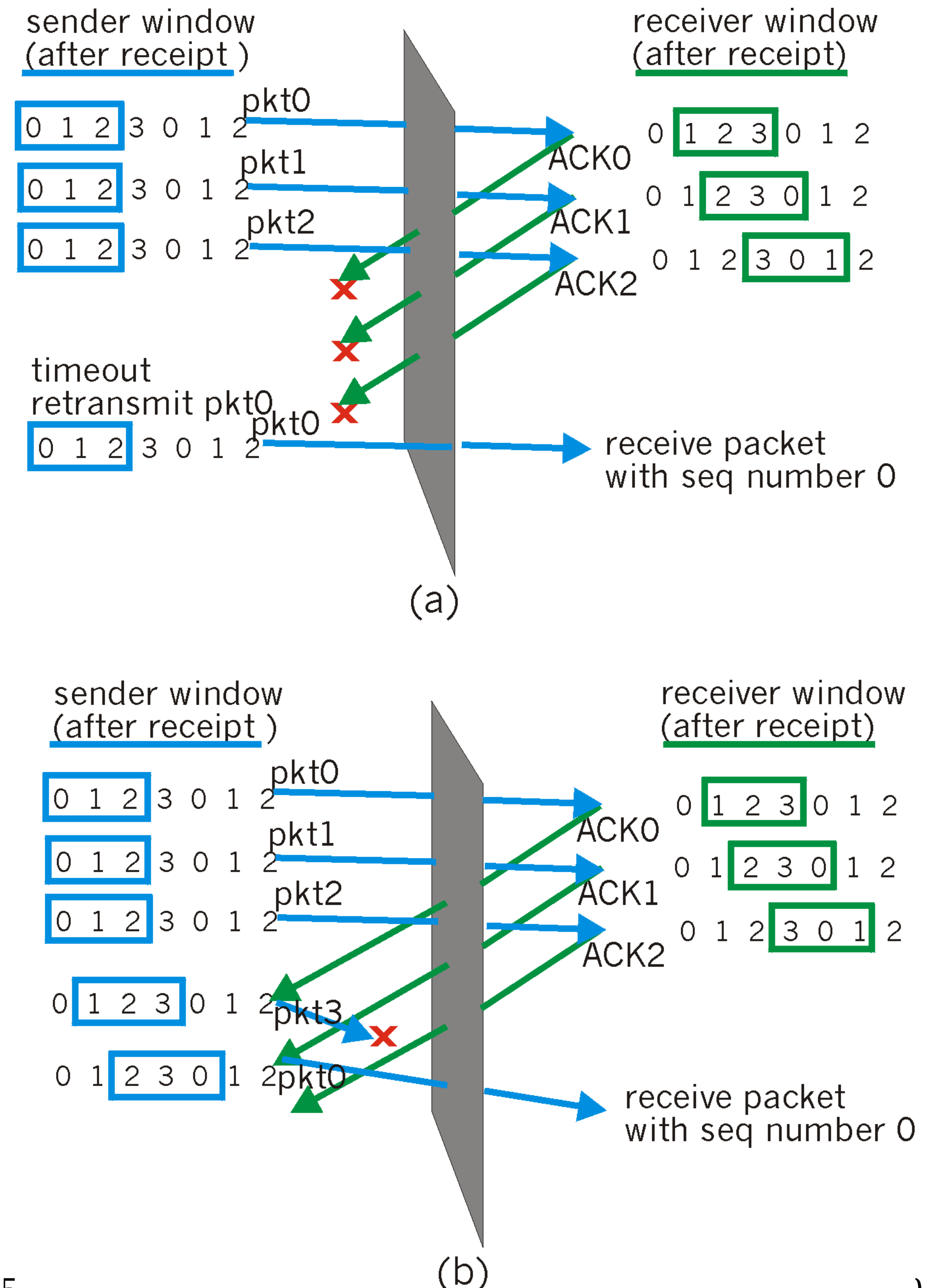
Selective repeat: dilemma

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 relationship between seq # size and window size?

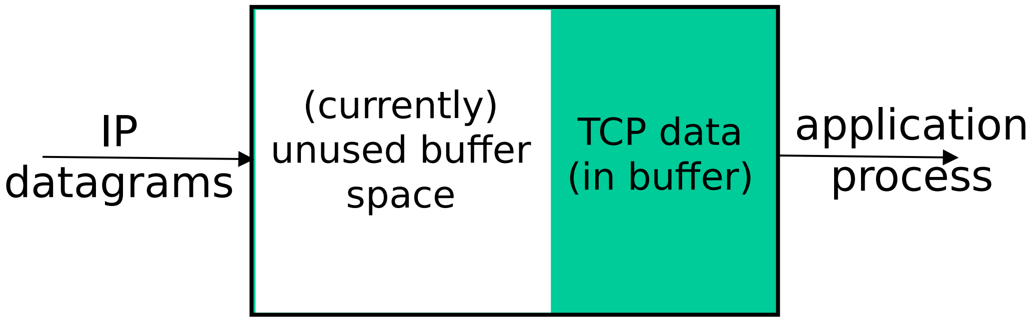


Flow Control: TCP

- receive side of TCP connection has a receive buffer:

flow control

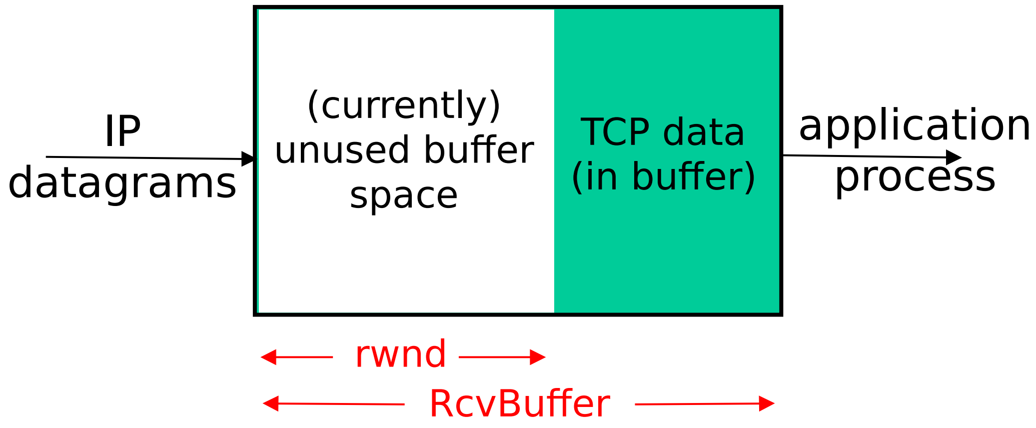
sender won't overflow receiver's buffer by transmitting too much, too fast



- *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

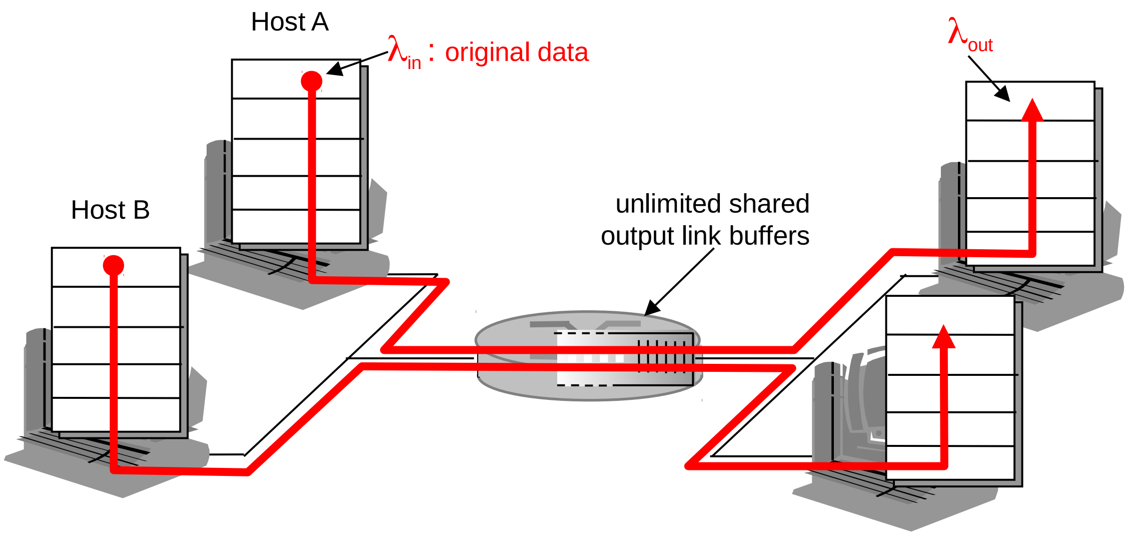
Congestion Control

- decoupled network and transport service: multiple senders might overwhelm routers => packet delay and loss
- certain situations: congestion collapse

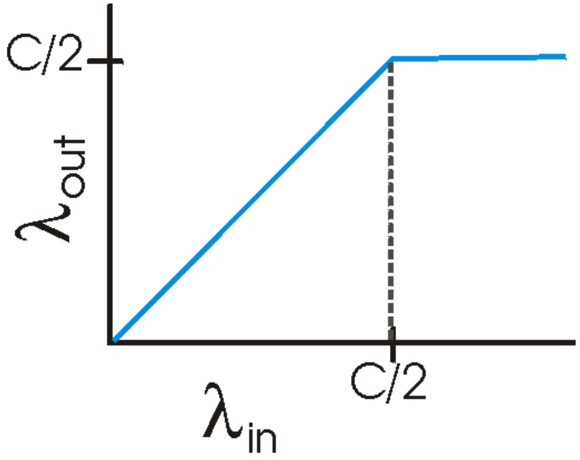
- another goal: “fair” sharing of network resources

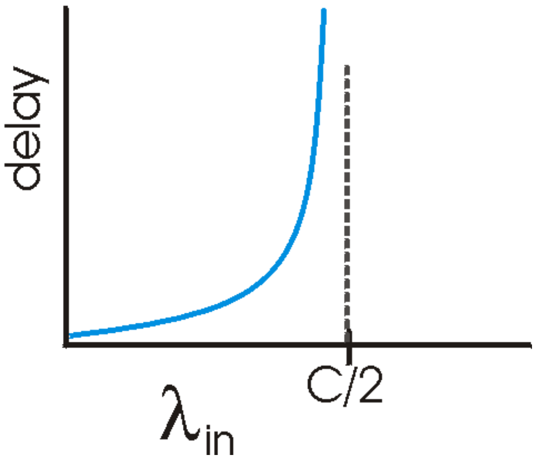
Overload without Reliability

- one router, infinite buffers
- no retransmission



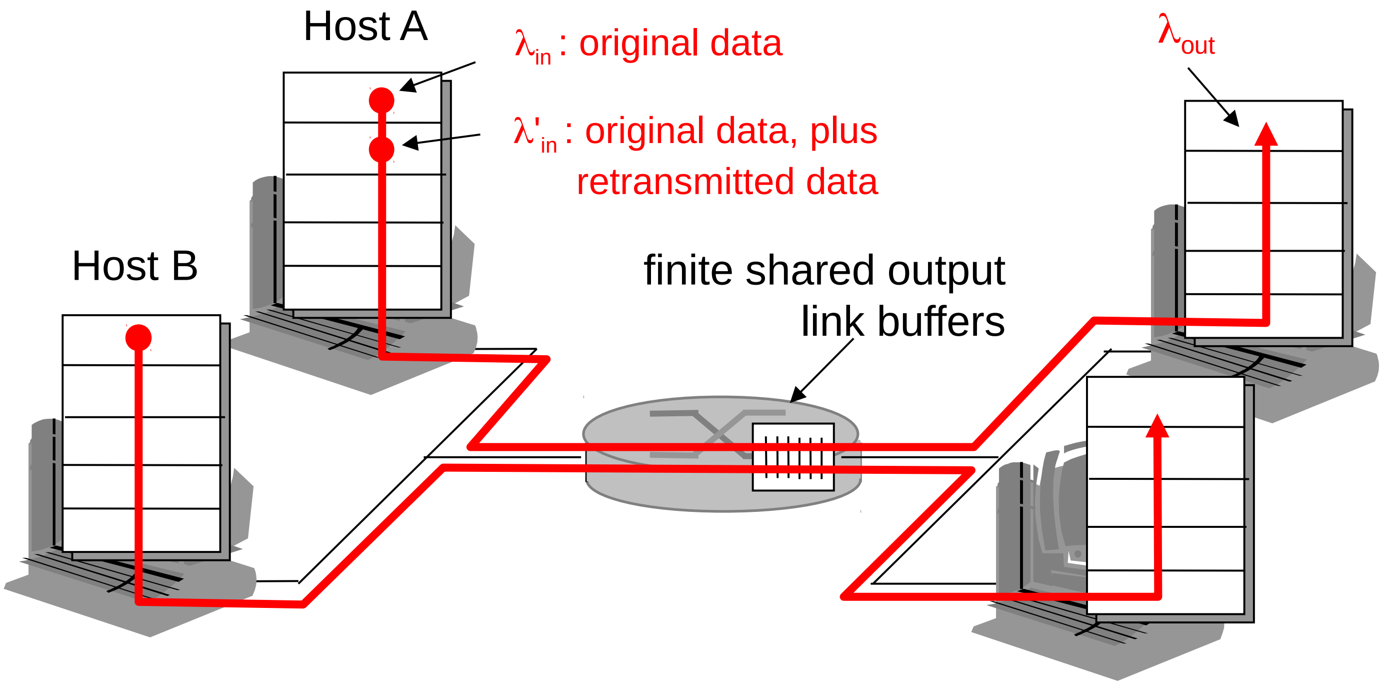
- large delays
- maximum throughput





Overload with Reliability

- one router, finite buffers
- retransmission of lost packets

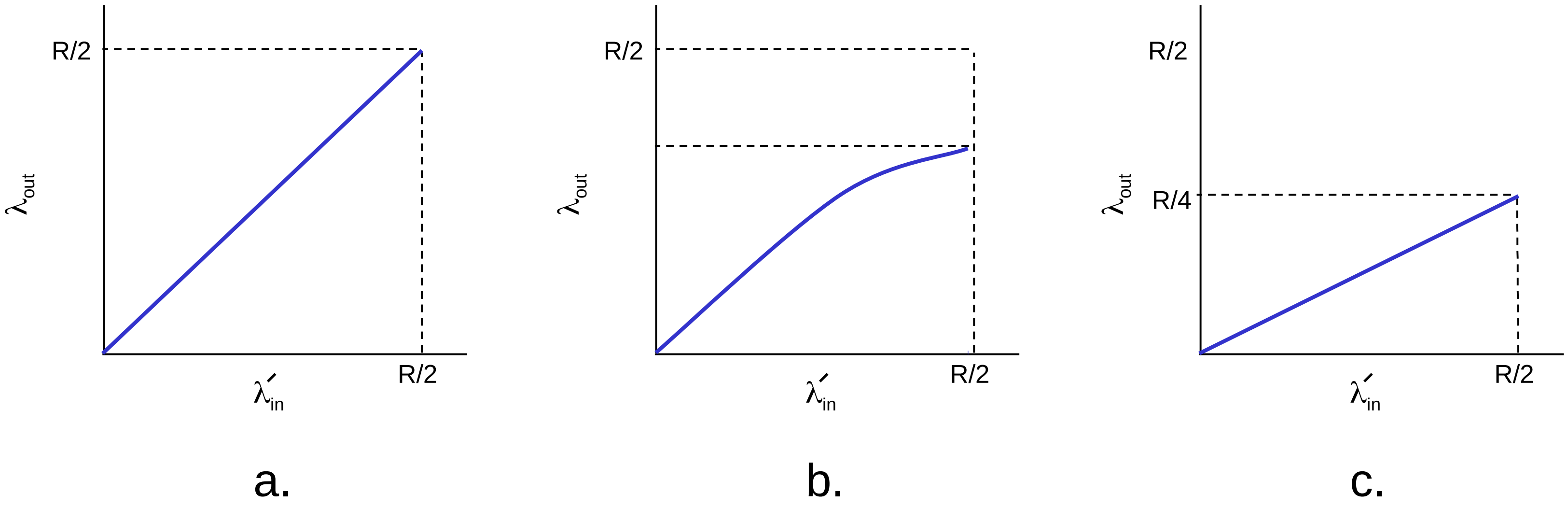


Overload with Reliability

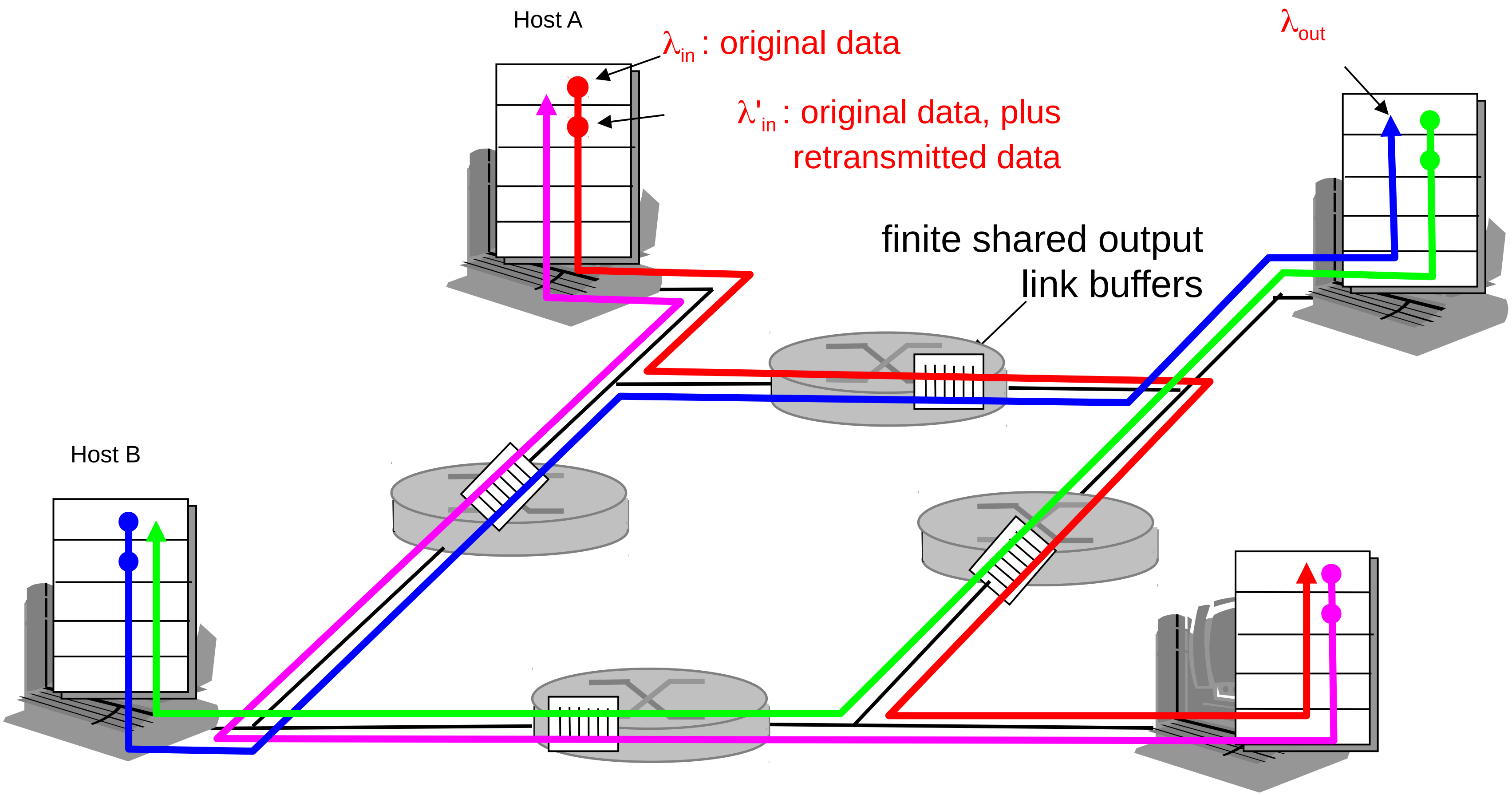
a) perfect send rate

b) finite buffer -> loss & retransmission

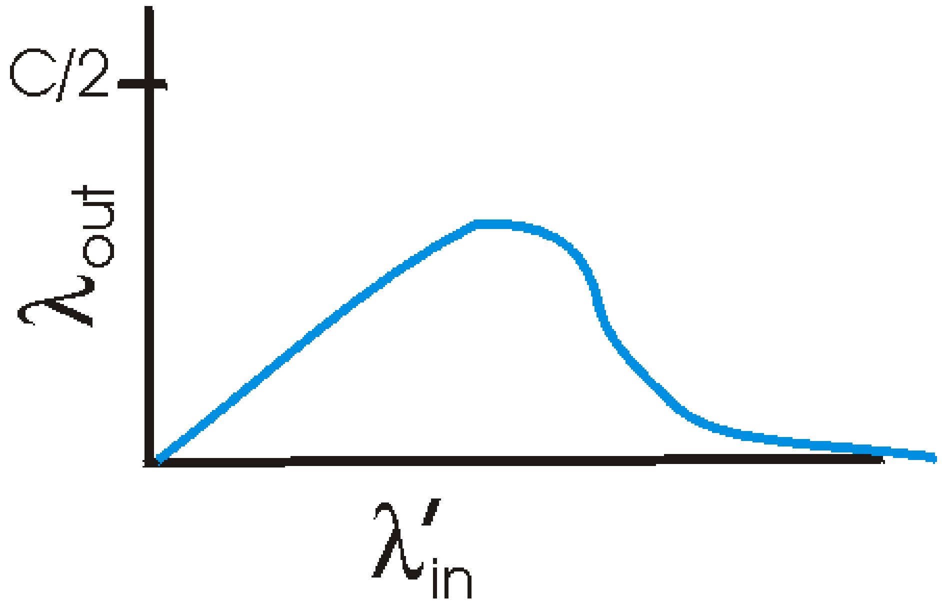
c) retransmission too eager (timeout to small)

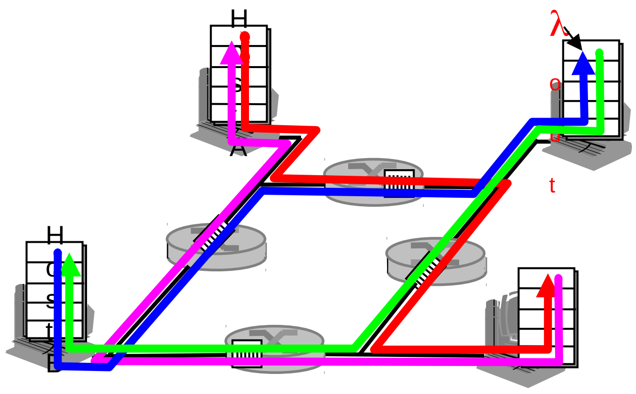


Circular Bottlenecks



Congestion Collapse





- packet drop after upstream bottleneck

  => upstream capacity wasted

Congestion Control

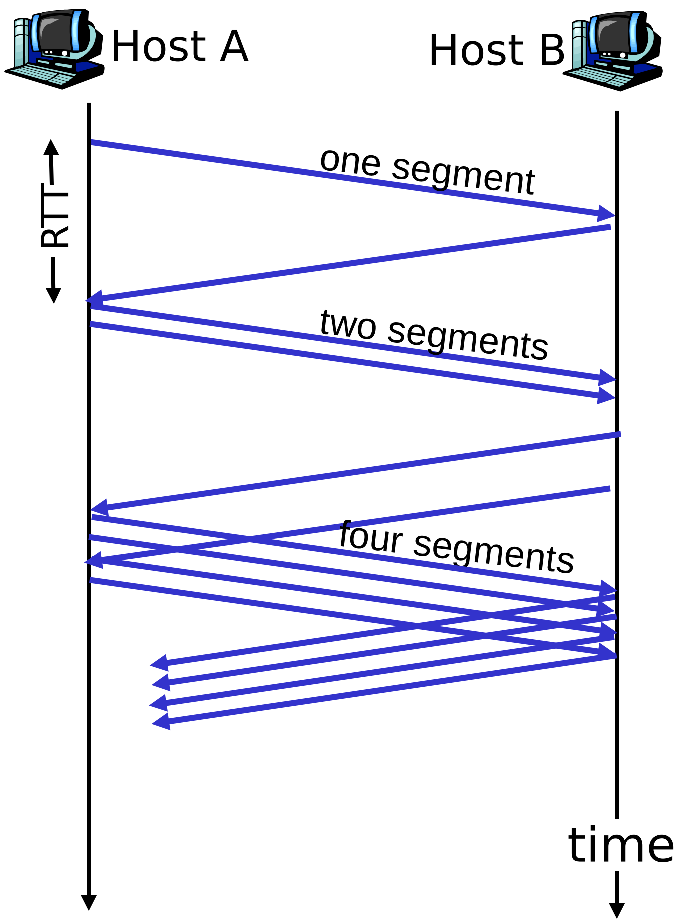
- senders must control rate to avoid permanent network overload
- input signals?
  - direct feedback from network? overhead!
  - indirect feedback through receiver? reaction time!
    - TCP congestion control
    - use drop or packet marking to indicate overload

TCP Congestion Control

- limit depth of pipeline – *congestion window*
- sending rate (roughly): CongWin / RTT
- adjust CongWin based on network feedback
  - sender infers packet loss
    - duplicate ACK -> assume light overload
    - timeout -> assume severe overload
- adaptation regimes/phases
  - *slow start*: start at very small rate, increase fast
  - *congestion avoidance*: hold rate, increase slow

TCP Slow Start

- start with small fixed CongWin
- increase exponentially until first loss
  - double CongWin every RTT, i.e.:
  - increment CongWin for every ACK
- start slow, but increase fast

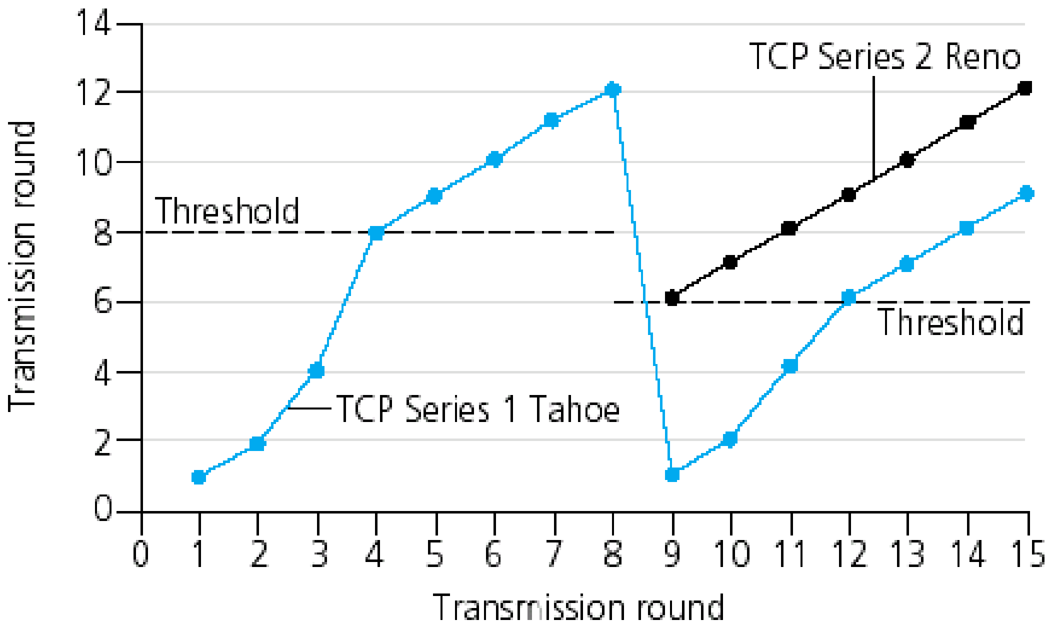


TCP Congestion Avoidance

- regular operation (no loss):
  increase CongWin by fixed amount per RTT
- receiver detects missing segment
  -> send duplicate ACK for previous one
- sender receives 3 duplicate ACKs
  -> reduce CongWin in half
- but after sender timeout:
  -> restart Slow Start procedure

TCP Rate Control

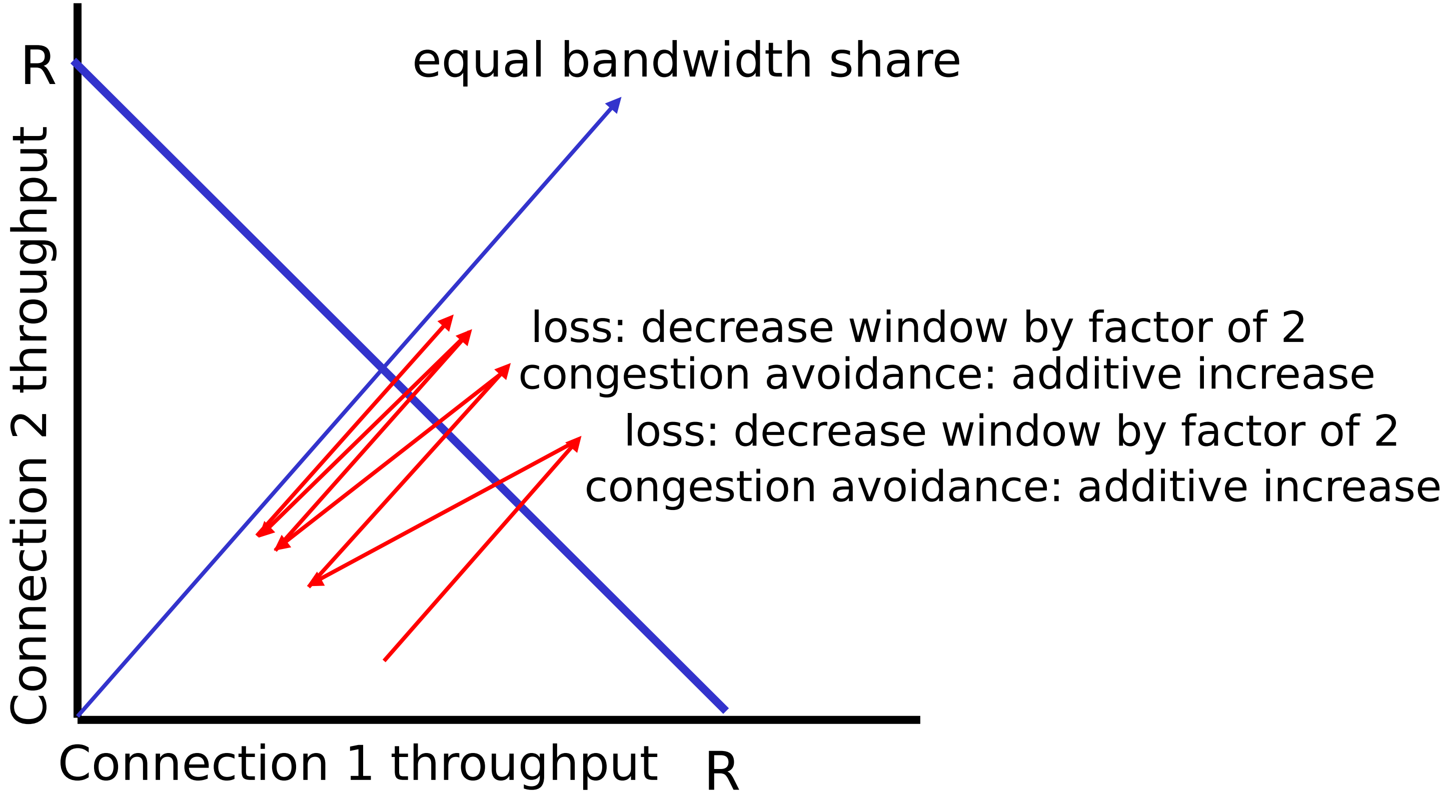
- Slow Start -> Congestion Avoidance
  - based on threshold (CongWin/2 of last timeout)



TCP Fairness

Two competing sessions:

- Additive increase gives slope of 1, as throughout increases
- multiplicative decrease decreases throughput proportionally



TCP Discussion

- hybrid of Go-Back-N and Selective Repeat
  - SACK: more precise acknowledgements (limited)
- reduction of CongWin -> pause sending
  - until ACKs catch up with outstanding data
- refinements
  - fast retransmit & fast recovery -> resume sending faster during dupack losses
  - keep sending at ACK-clocked pace

TCP Discussion

- TCP fairness relies on configuration values
  - initial window for slow start
  - additive increase during congestion avoidance

-> problem with highspeed / long-delay links

- more agile congestion control -> robustness?
- per-session fairness?

- lots of other approaches in the literature
  - very little real-world adoption

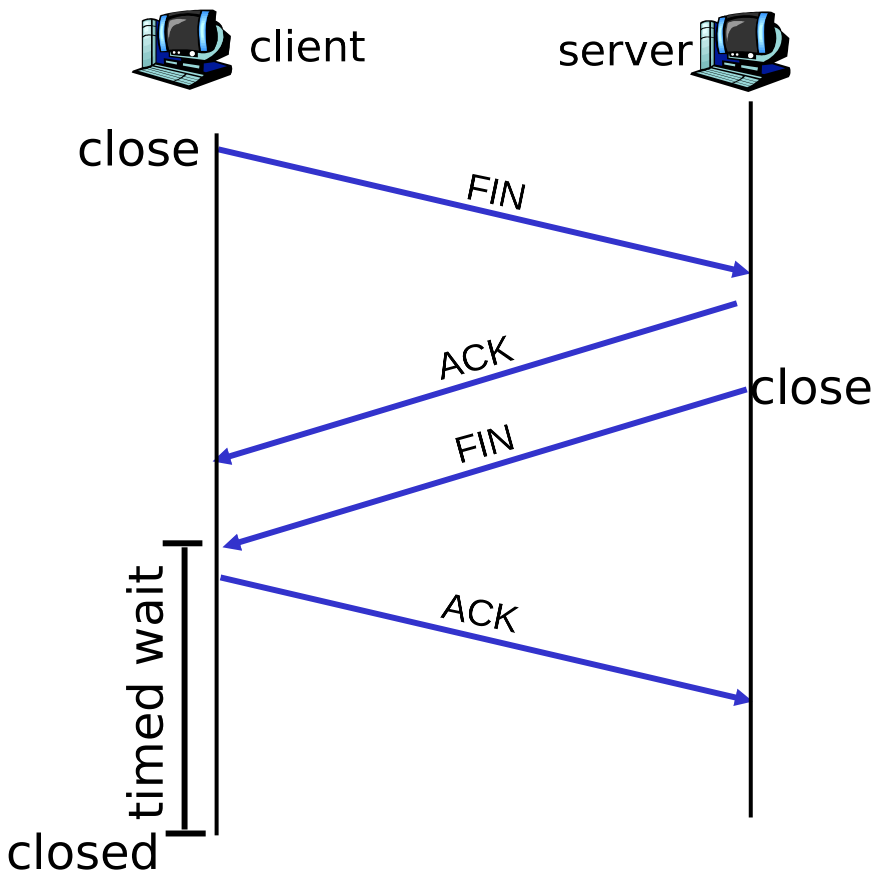
Connection Management: TCP

- Connection Establishment: 3-Way Handshake
- Step 1: initiator sends SYN to responder
  - sets up initial variables, e.g., sequence number
- Step 2: responder responds with SYNACK
  - sets up initial variables, e.g., sequence number
  - responder allocates internal buffer
- Step 3: initiator responds with ACK
  - might already send data along

Connection Management: TCP

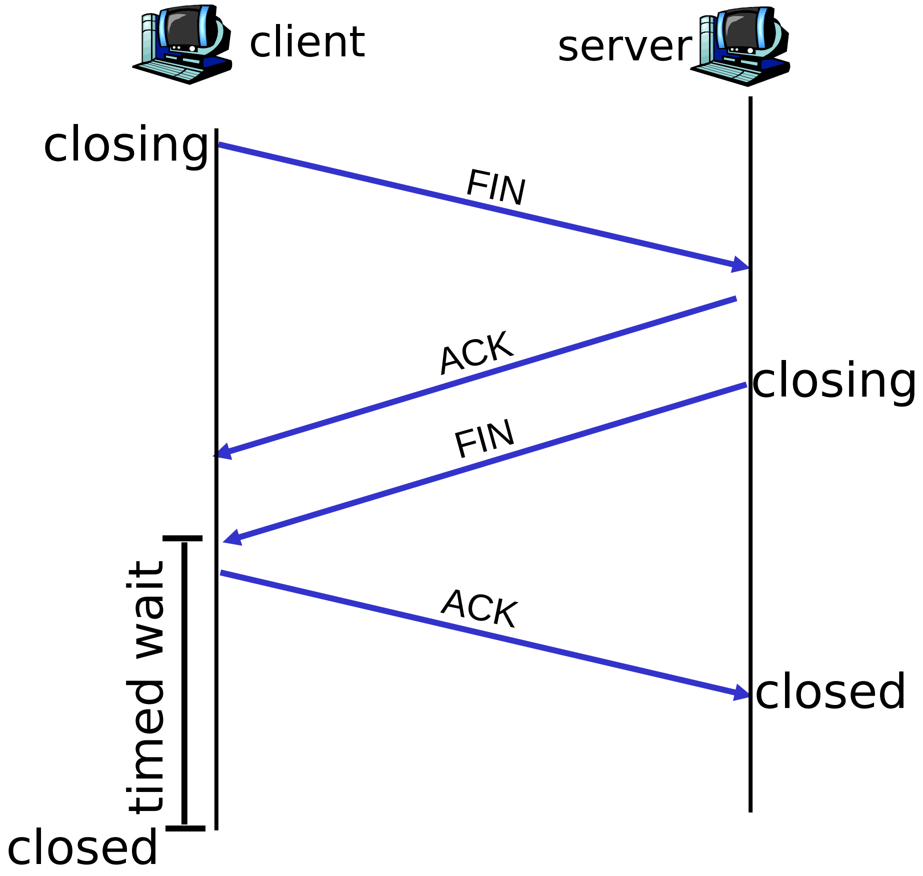
- Connection Teardown – Be Aware of Reliability!
- Step 1: client sends TCP FIN to server
- Step 2: server responds with ACK & sends FIN

- FIN -> no more data
- ACK -> all data received



Connection Management: TCP

- Step 3: client responds with ACK
  - enters “TIMED WAIT”
  - in case ACK is lost
- Step 4: server receives ACK
  - connection closed



Interface Semantics

- message interface
  - message boundaries preserved across transport
- byte-stream interface
  - message boundaries not preserved
  - simpler and more flexible for implementation