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CS3640

Transport Layer (2): Reliable Data Transfer

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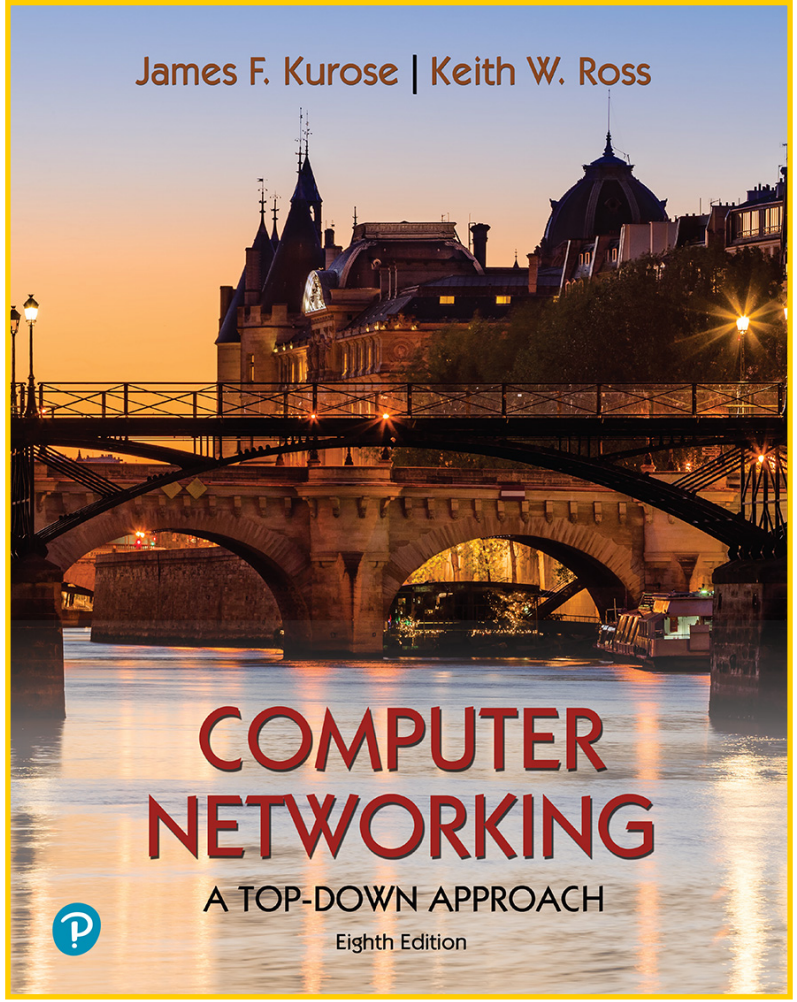
Lecture goals

how to achieve reliable data transport over an unreliable network such as the Internet?

- *Challenges and techniques*
- *Stop-and-Go protocols*
- *Performance analysis*



Chapter 3.4

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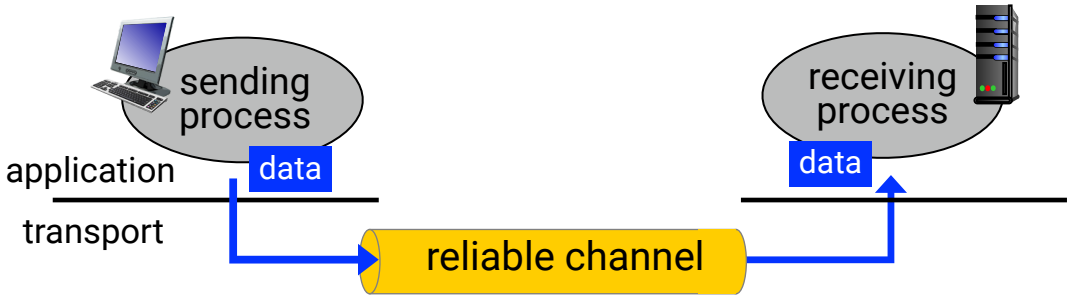
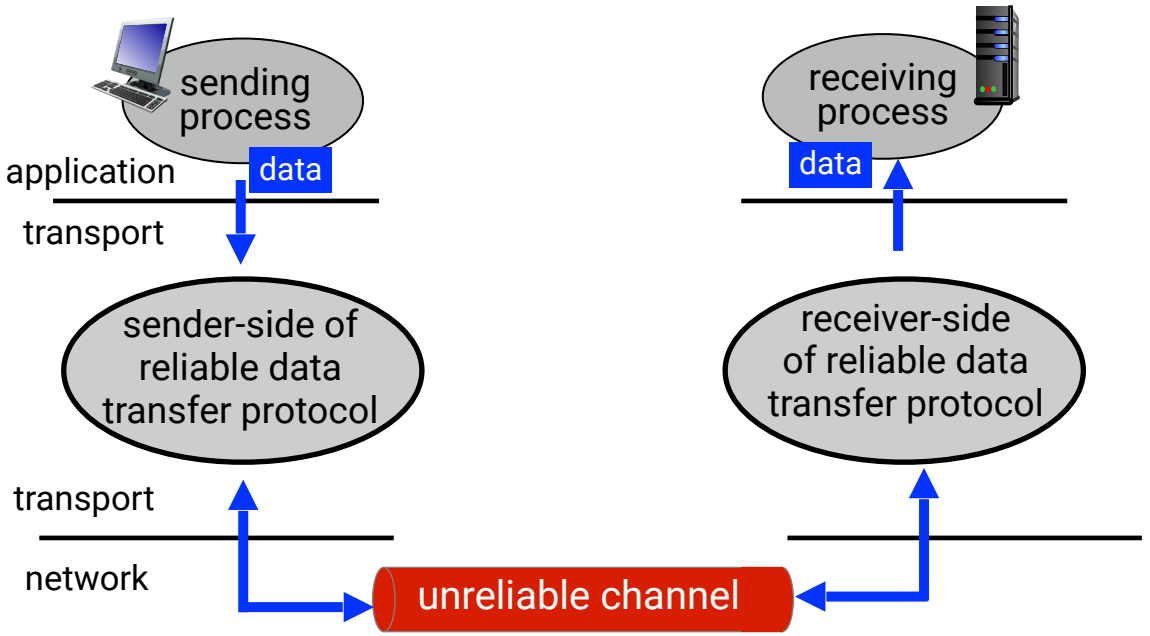
Principles of reliable data transfer

reliable service abstraction



reliable service implementation



Key Challenges

Sender, receiver do not know the “state” of each other, or what happens in the channel (unless communicated via a message)

Complexity of reliable data transfer protocol. This depends on the characteristics of unreliable channel (lose, corrupt, reorder data?)

Designing Reliable Data Transfer (RDT) Protocols

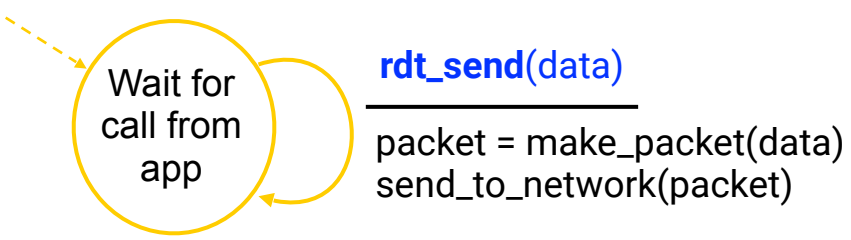
- first, we define characteristics of the underlying unreliable channel
- then, incrementally develop the sender and receiver sides of RDT
- using finite state machines (FSM) to specify the sender and receiver specifications

1. Channel with perfect reliability
2. Channel with bit errors
3. Channel with bit errors & packet loss

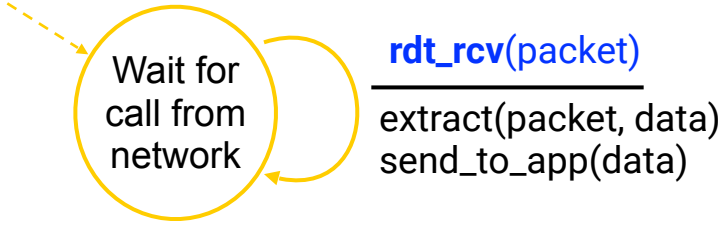
RDT-1: reliable transport over a perfectly reliable channel

- underlying network channel provides all the required functionality
- So, RDT-1 does not have to handle bit errors, loss of packets, or duplicates
- This renders the sender and receiver FSMs to be simple

RDT-1 sender



RDT-1 receiver



RDT-2: reliable transport over a channel with bit errors

how do we detect bit errors?

- employ checksums

how do we recover from errors?

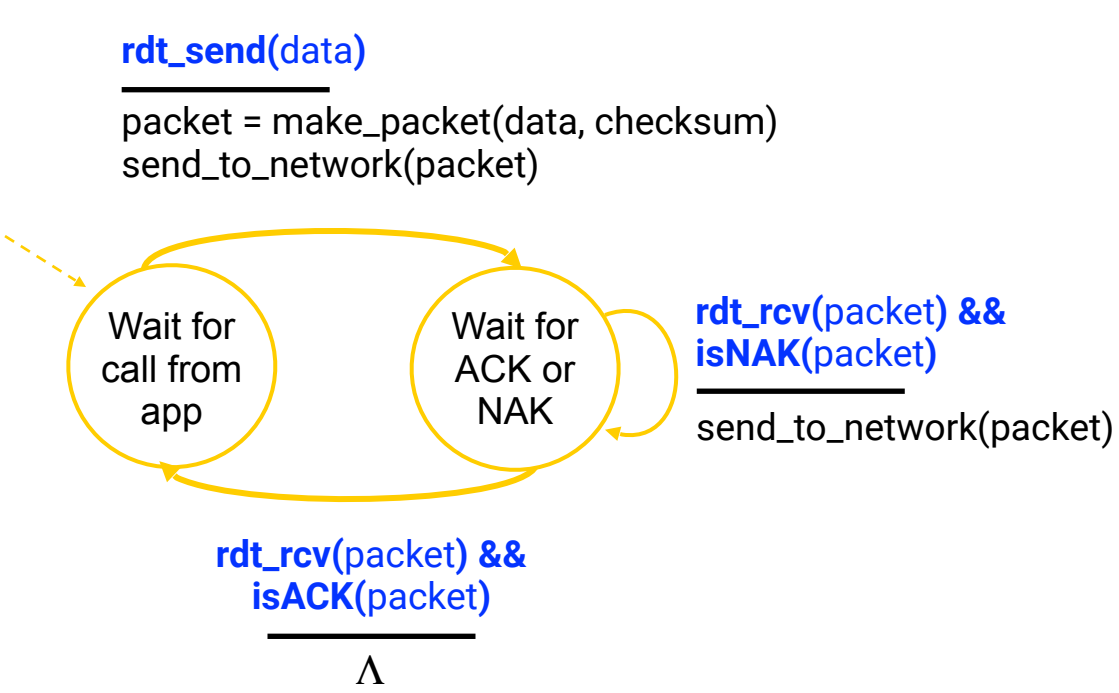
- acknowledgements (**ACKs**): receiver explicitly tells sender that packet was received OK
- negative acknowledgements (**NAKs**): receiver explicitly tells sender that packet had errors
- retransmit the packet on receipt of NAK

stop and wait protocol

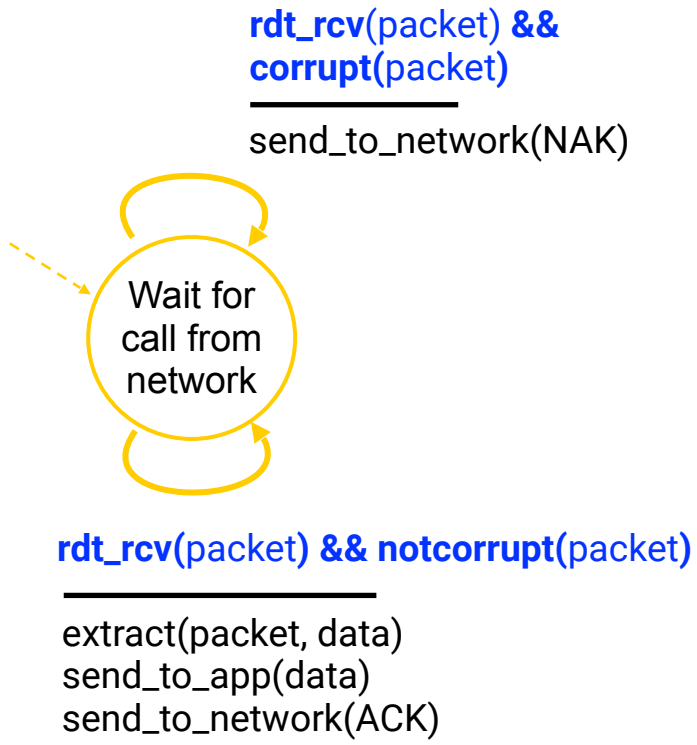
sender sends one packet, then waits for receiver response

RDT-2: FSM Specifications

RDT-2 sender



RDT-2 receiver



RDT-2 has a serious flaw!

What happens if ACK/NAK corrupted?

- sender doesn't know what happened at receiver!
- sender can't just retransmit as it may result in duplicate packets

Propose **RDT-2.1** with two changes:

1. ***retransmit on corrupted ACK/NAK***
2. ***use sequence# to detect duplicates***

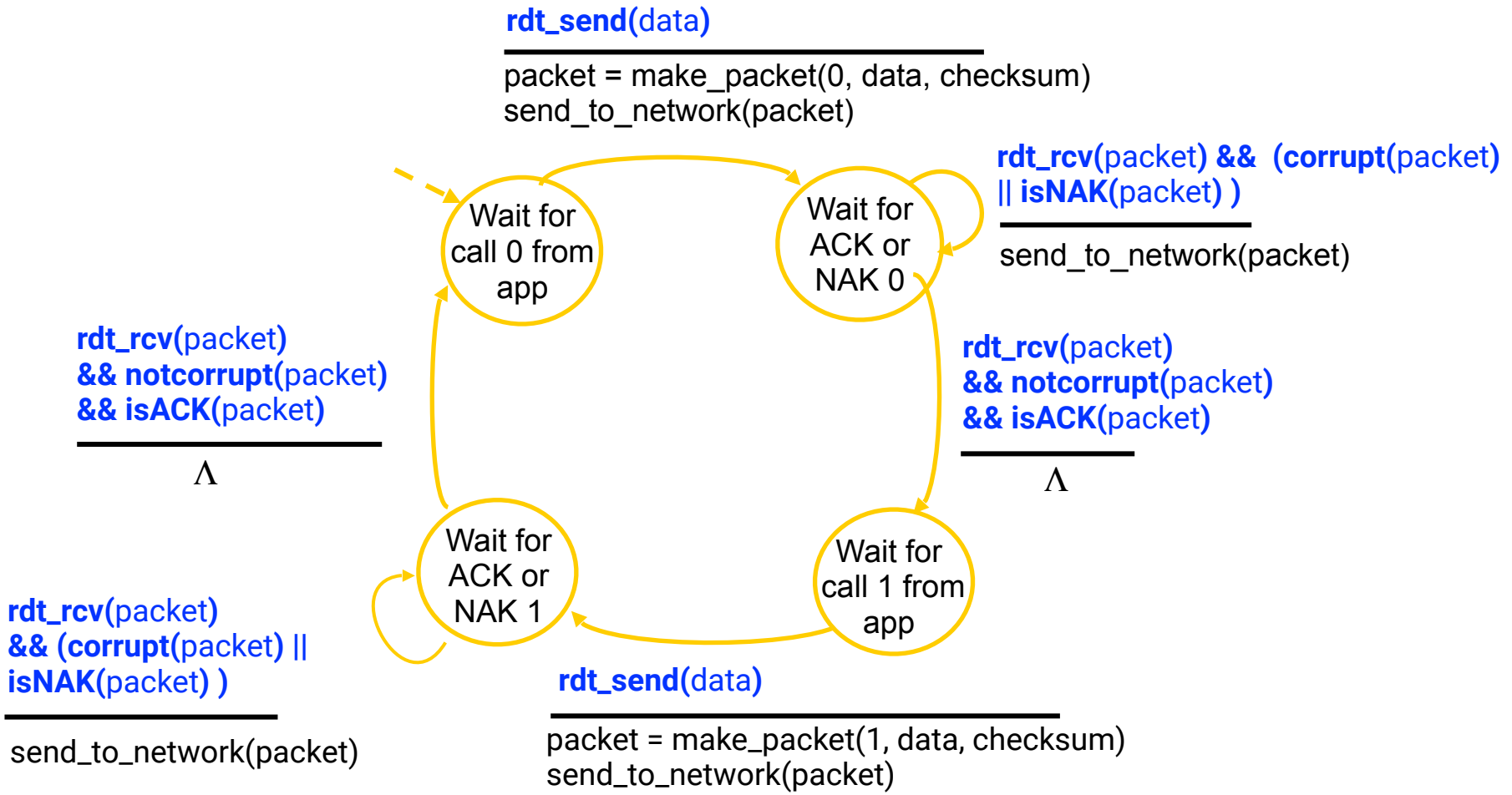
Changes to the Sender

- must check if received ACK/NAK corrupted
- add a sequence number to packets; however, only two seq# (0,1) will suffice. Why?
- results in twice as many states as before

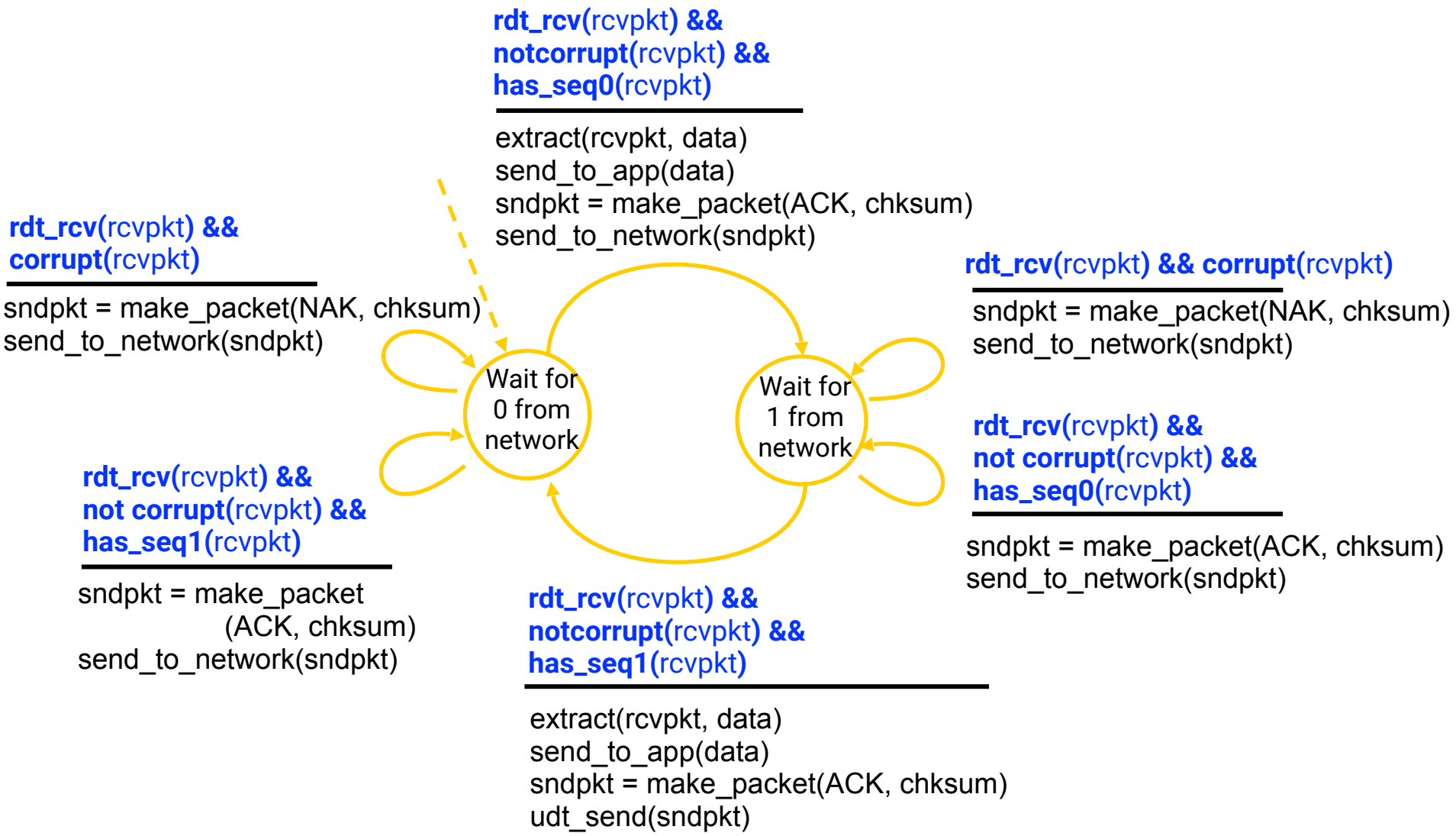
Changes to the Receiver

- must check if received packet is a duplicate (using sequence numbers)
- receiver won't know if its last ACK/NAK received OK at sender

RDT-2.1: sender, handling garbled ACK/NAKs



RDT-2.1: receiver, handling garbled ACK/NAKs



RDT-3: reliable transport over a channel with errors and loss

Could sender and receiver recover from packet losses with existing mechanisms?

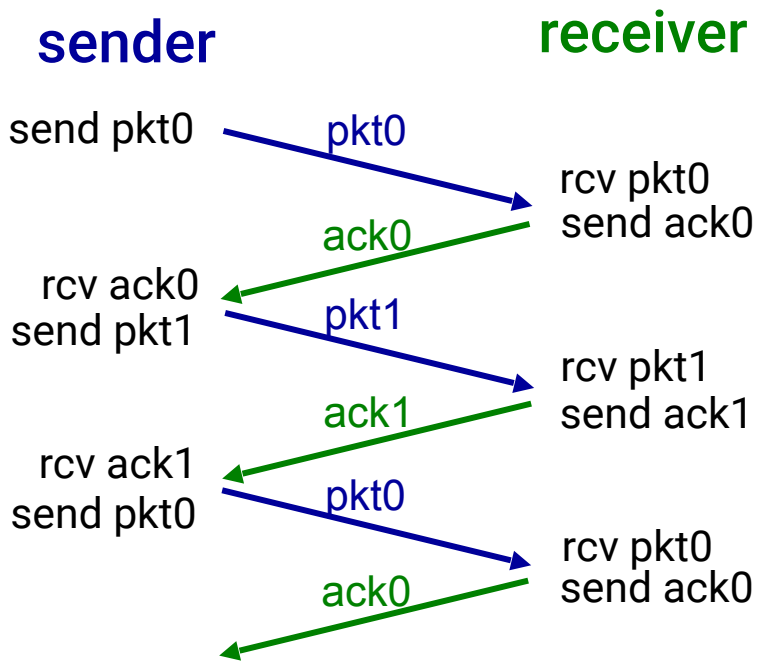
- sequence#, ACK, and retransmissions will help but not quite enough

Approach

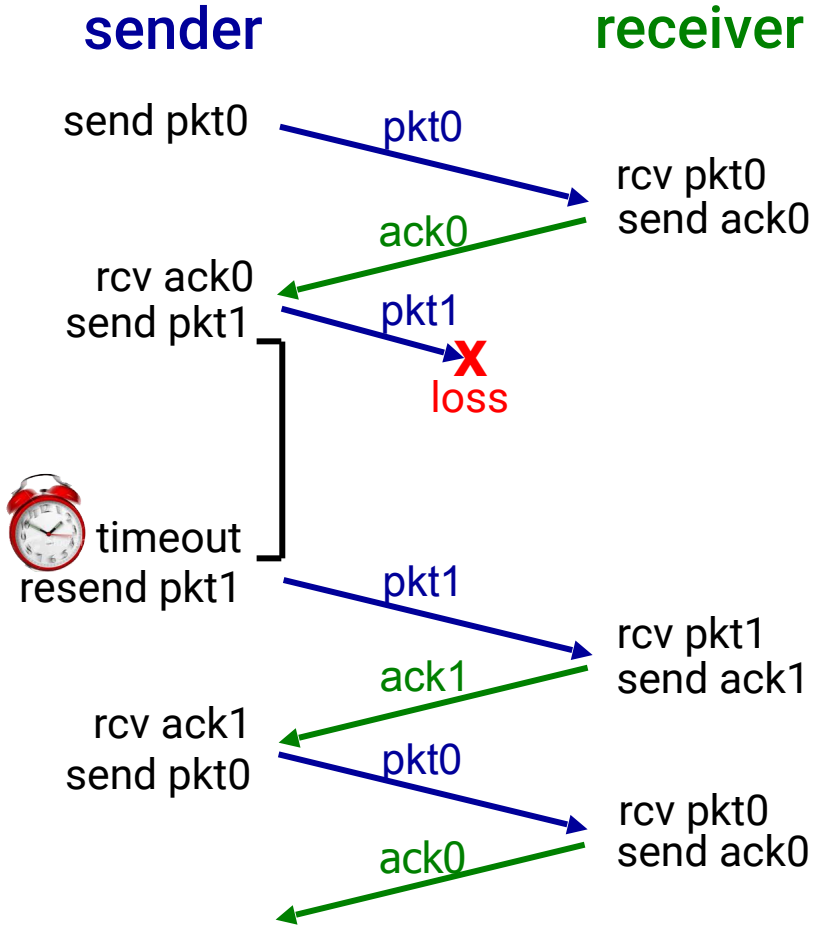
- Modeled after the human behavior… the notion of ***timeout and retry***
- sender waits a reasonable amount of time for ACK; if no ACK is received in that time, retransmits the packet again
- receiver already has the mechanism to check for duplicates (via sequence#); however, since multiple ACK can be on the wire now, they must carry the sequence# of the packet being ACKed

RDT-3 in action

Case-1: No loss

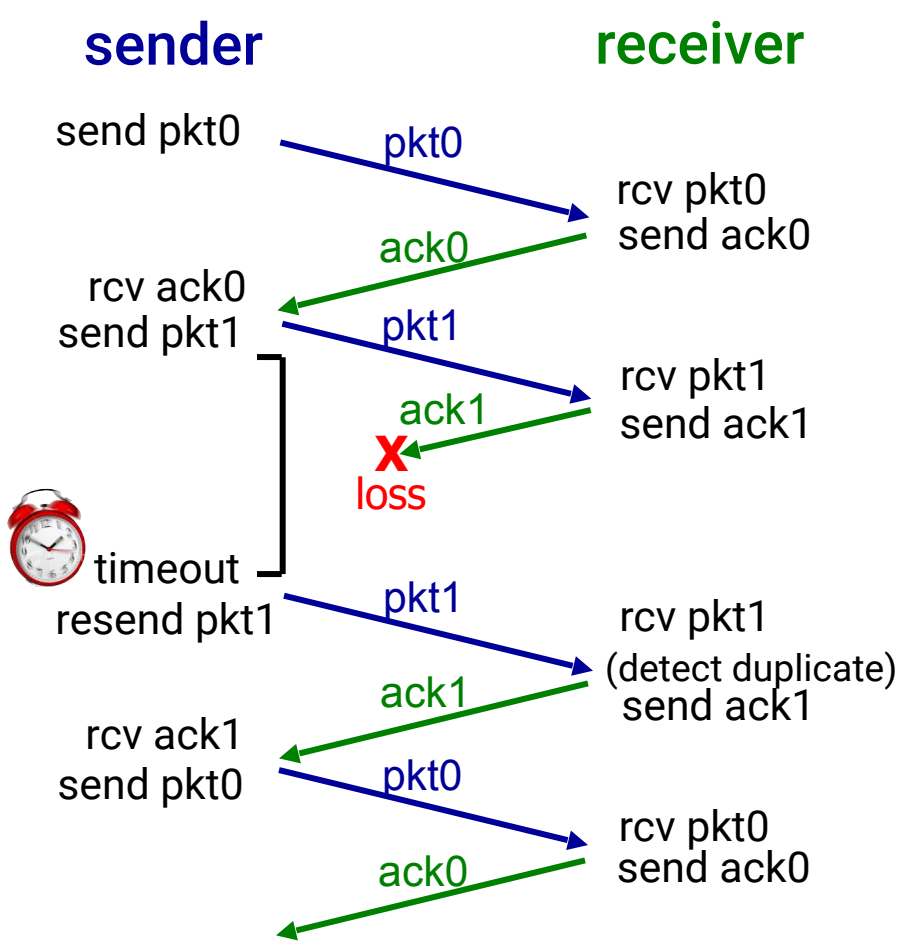


Case-2: Packet loss

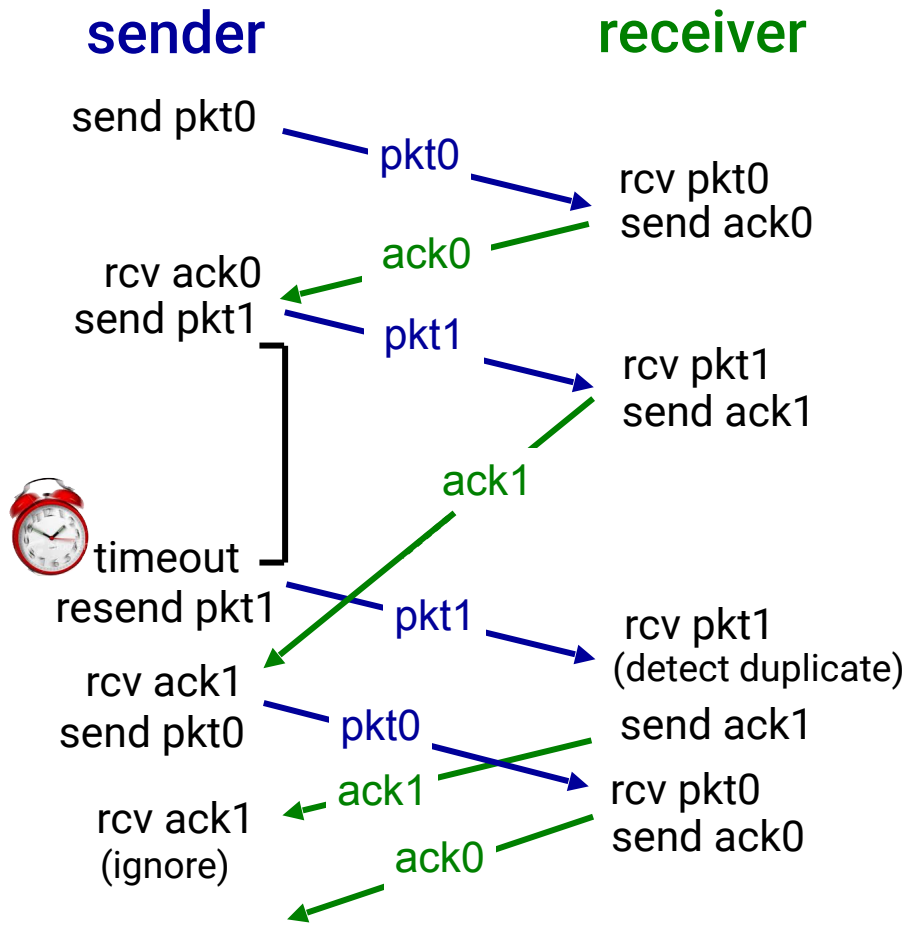


RDT-3 in action

Case-3: ACK loss

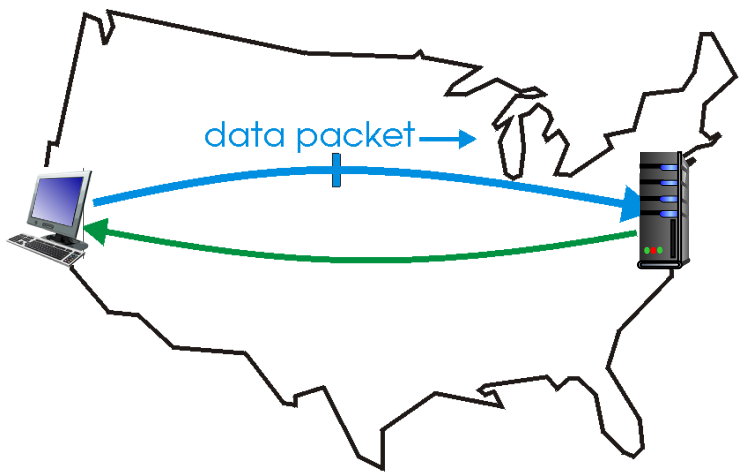


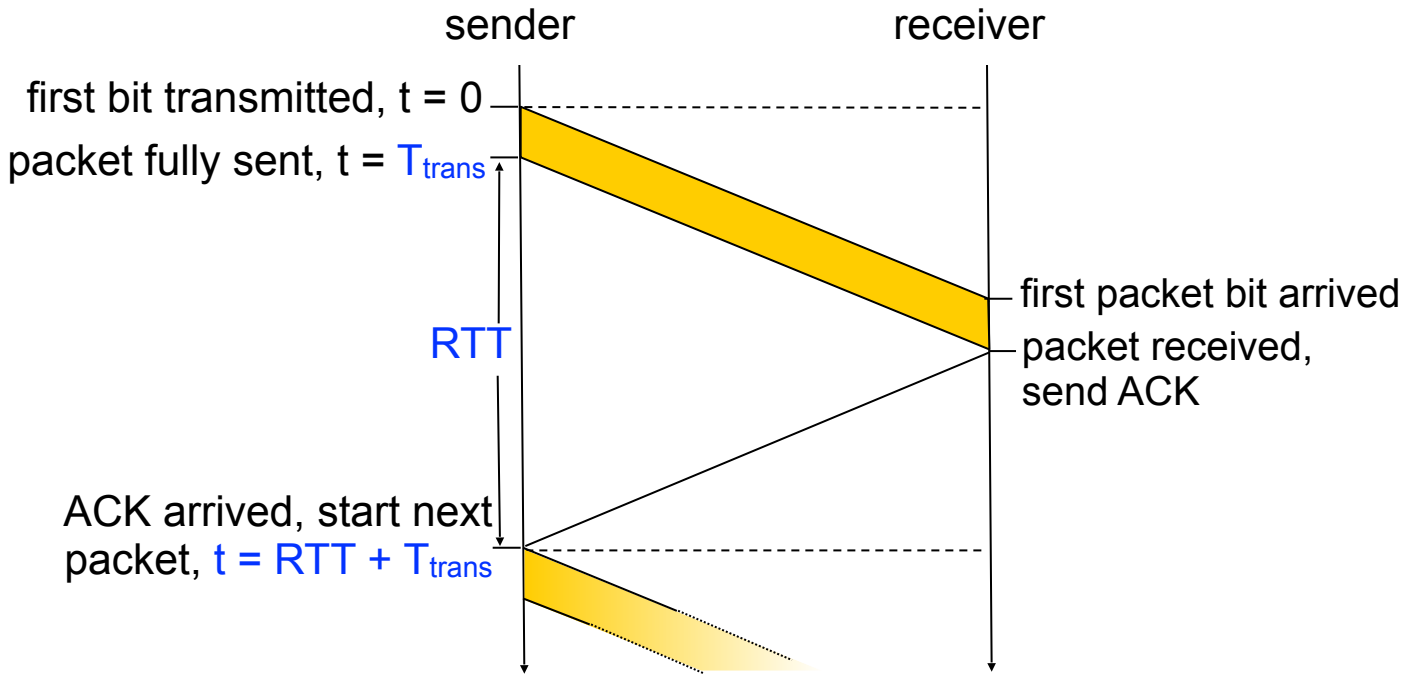
Case-4: Delayed ACK



Performance of **Stop-and-Wait**

Visualizing stop-and-wait in operation





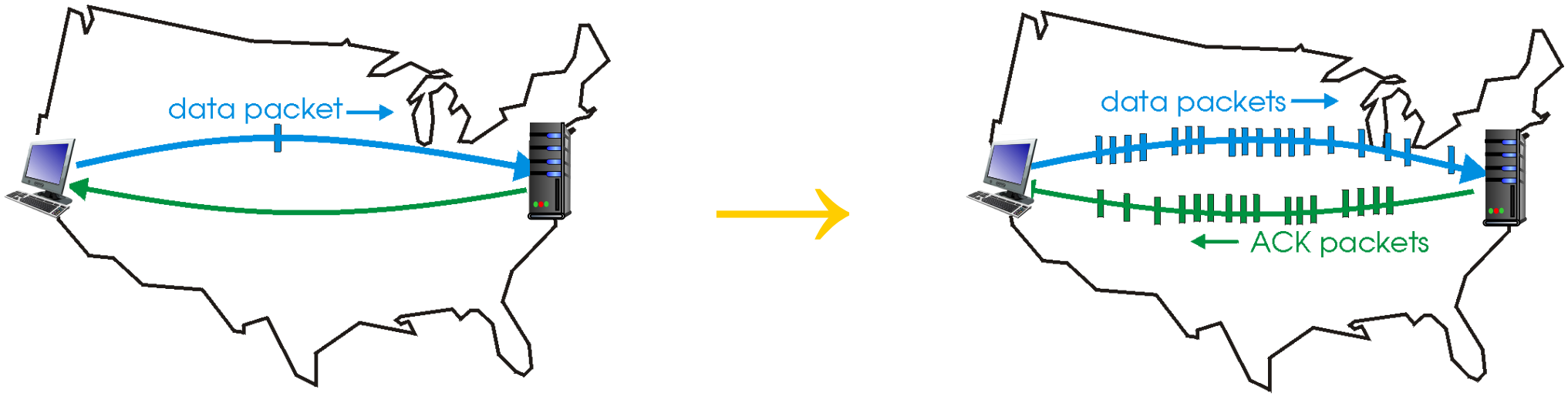
An example

- NY-SF propagation, D_{prop} = 15 ms
- Packet length, L = 8000 bits
- Transmission rate, R = 1 Gbps
- Transmission time, T_{trans} = L/R = 8 μs

Channel Utilization: *fraction of the time bits are sent into the channel*

- Utilization = T_{trans} / (RTT + T_{trans})
- Utilization = 8 μs / 30.008 ms = 0.00027 = **0.027%**

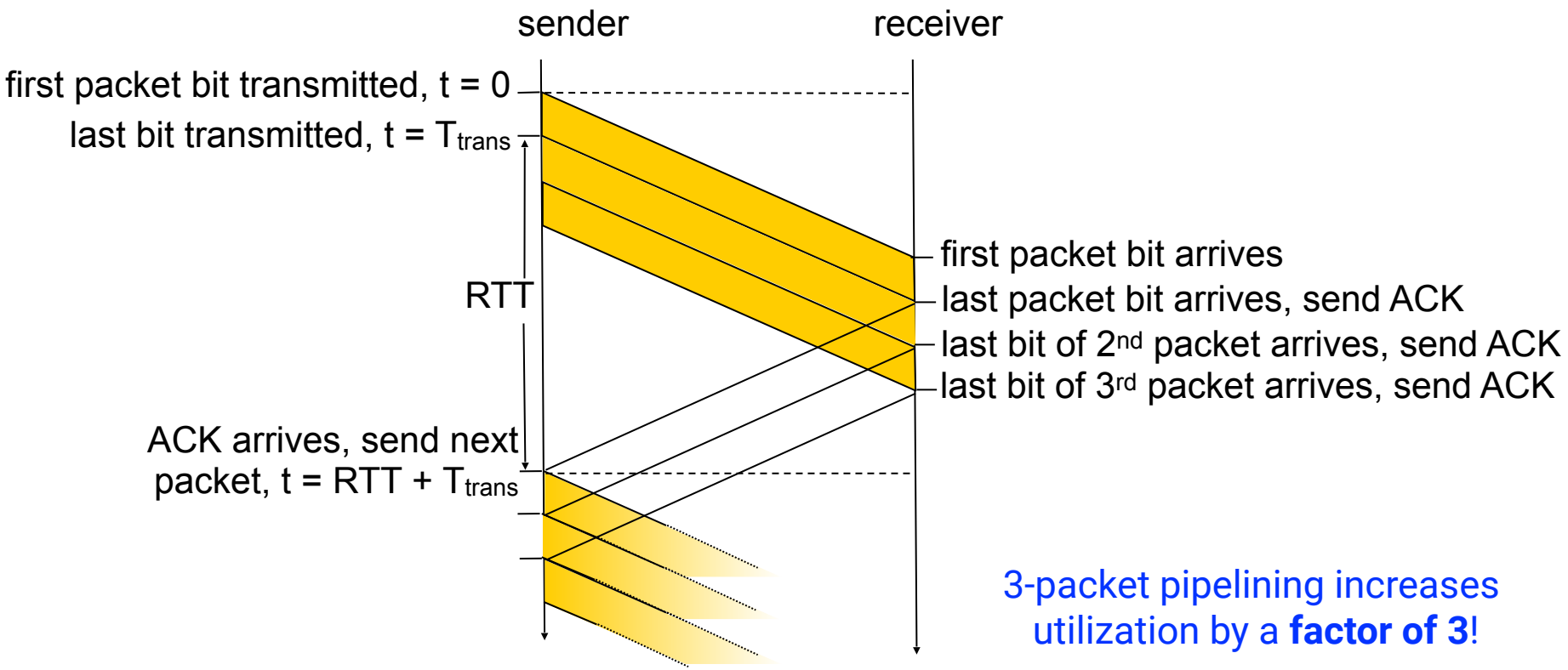
Increased Utilization w/ Pipelining



How does pipelining affect RDT-3 protocol?

- range of sequence# must be increased since there may be multiple, in-transit, unACKed packets
- sender and receiver sides need to buffer more than one packet

Increased Utilization w/ Pipelining



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

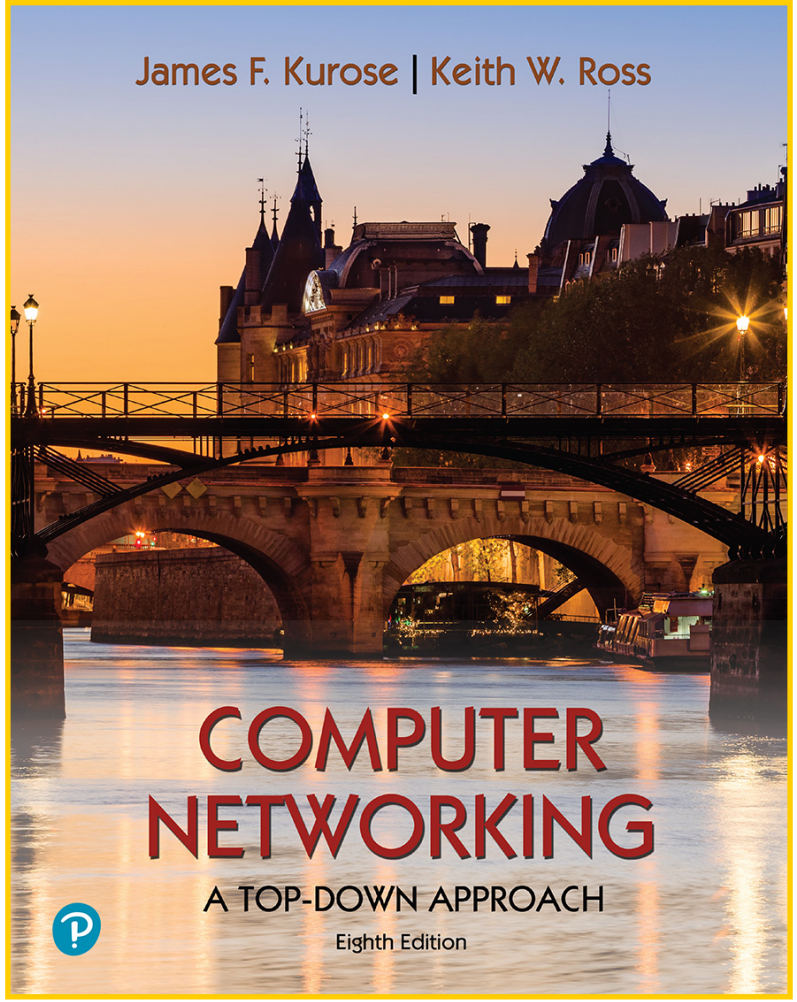
Channel Utilization: *fraction of the time bits are sent into the channel*

- Utilization = **3** * T_{trans} / (RTT + T_{trans})
- Utilization = **24** µs / 30.008 ms = 0.00081 = **0.081%**

Next lecture

continued discussion of reliable data transfer, followed by congestion control

- *Two pipelined RDT protocols*
- *Congestion control*



Chapter 3.4, 3.6

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Spot Quiz (ICON)