

# Transport Layer



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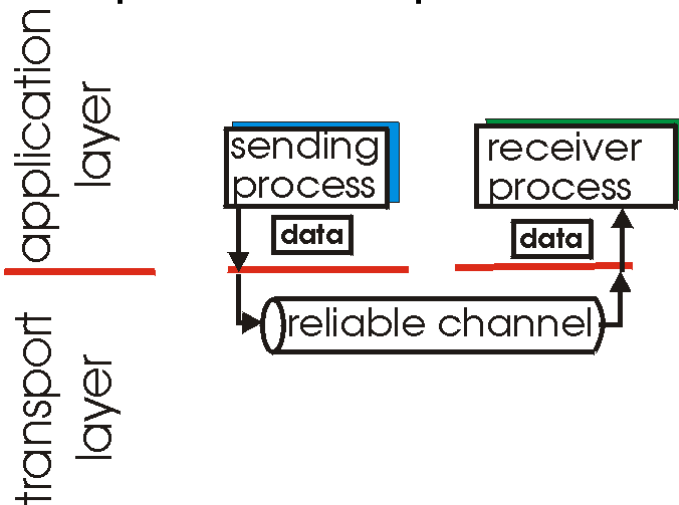
Instructor: C. Pu (Ph.D., Assistant Professor)

Lecture 09

*puc@marshall.edu*

# Principles of Reliable Data Transfer

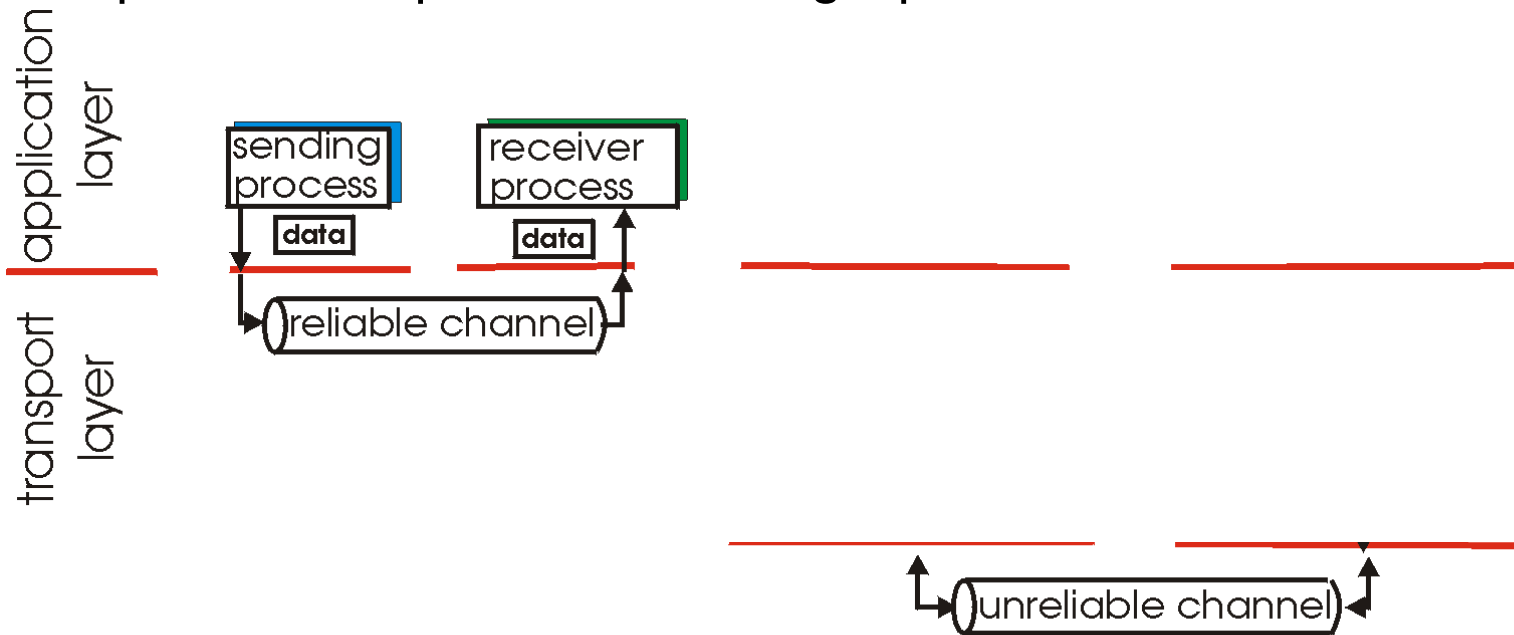
- important in app., transport, link layers
- top-10 list of important networking topics!



(a) provided service

# Principles of Reliable Data Transfer (cont.)

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



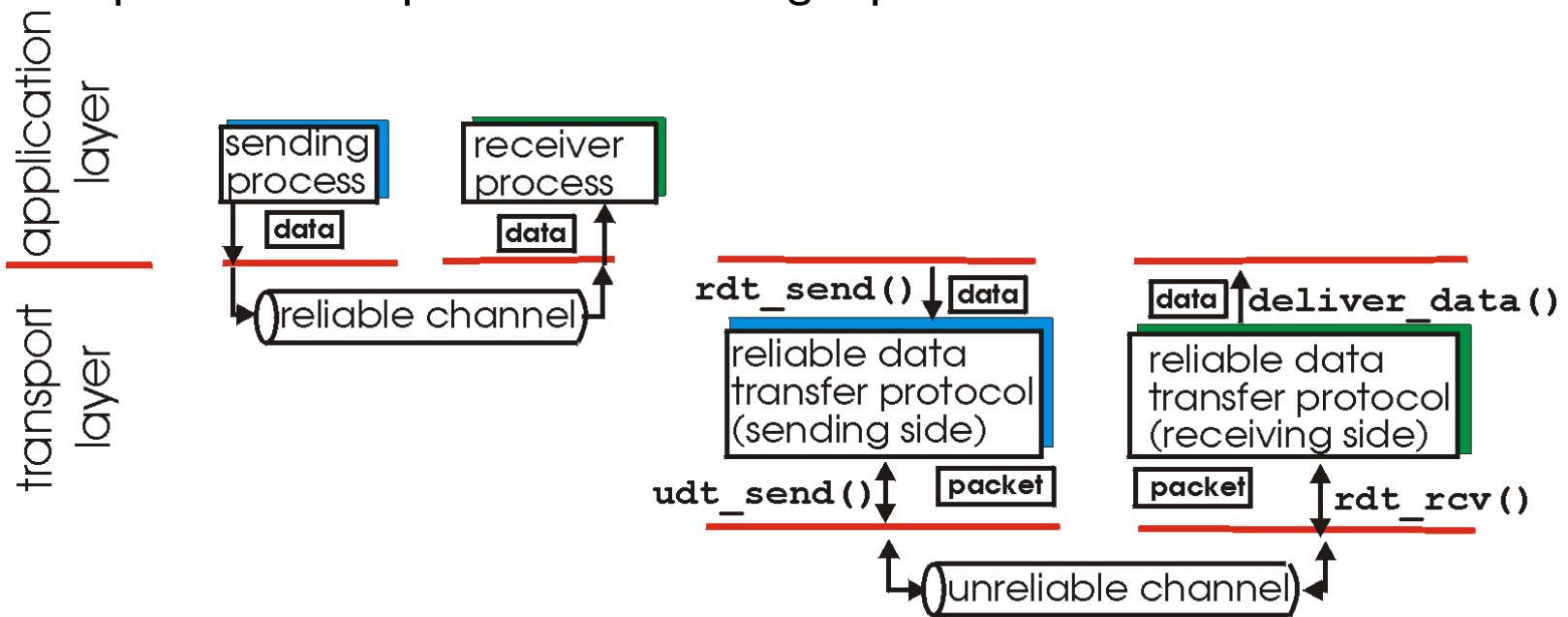
(a) provided service

(b) service implementation

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

# Principles of Reliable Data Transfer (cont.)

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



(a) provided service

(b) service implementation

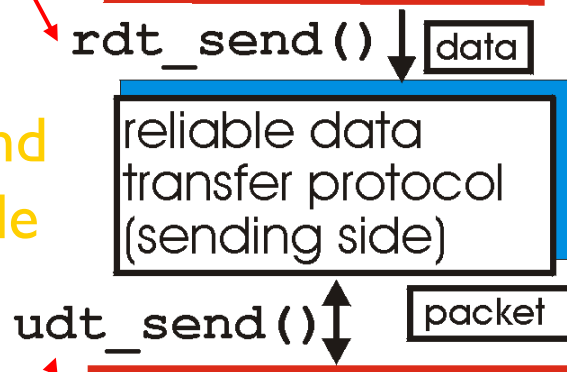
- characteristics of **unreliable channel** will determine complexity of reliable data transfer (rdt) protocol

# Reliable Data Transfer: Getting Started

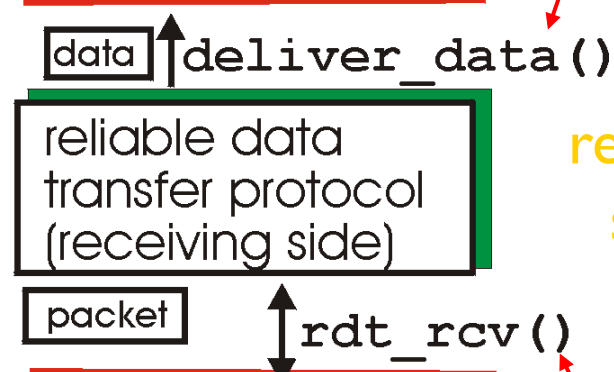
**rdt\_send():** called from **above**, (e.g., by app.). Pass data to be delivered to receiver upper layer

**deliver\_data():** called by **rdt** to deliver data to upper layer

send  
side



receive  
side



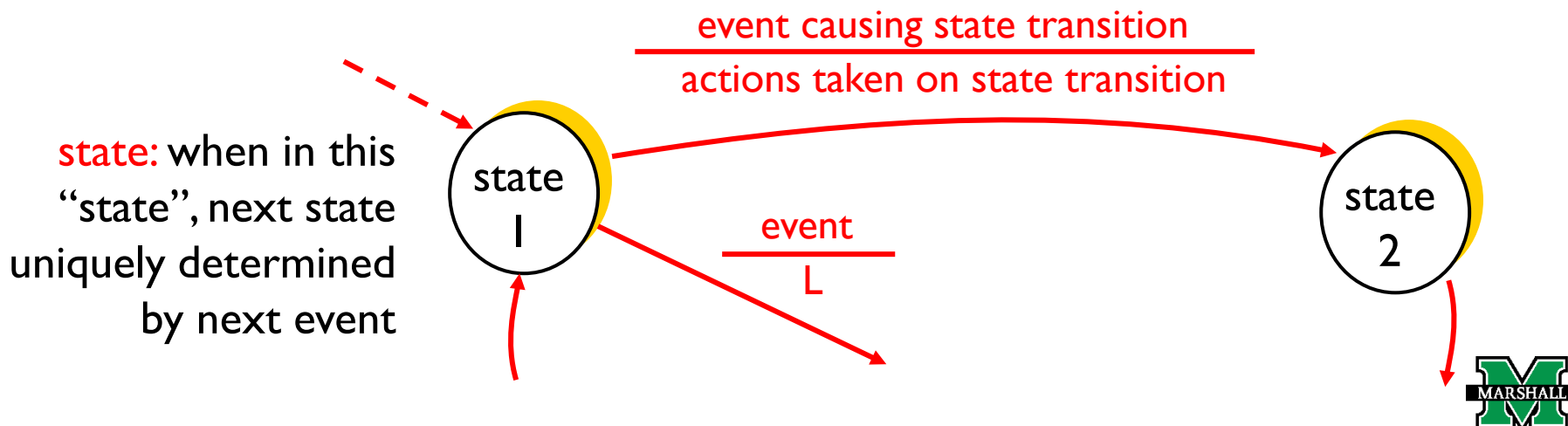
**udt\_send():** called by **rdt**, to transfer packet over unreliable channel to receiver

**rdt\_rcv():** called **when packet arrives** on receiver side of channel



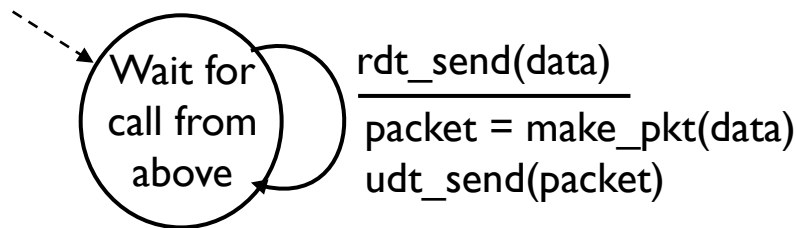
# Reliable Data Transfer: Getting Started (cont.)

- We'll:
  - incrementally develop sender and 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 the behaviors of sender and receiver

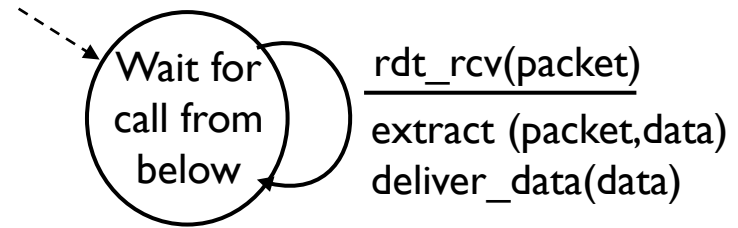


# rdt1.0: Reliable Transfer over a Reliable Channel

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



sender



receiver

with a perfectly reliable channel, there is no need for the receiver side to provide any feedback to the sender since nothing can go wrong!



## rdt2.0: Channel with Bit Errors

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- consider a more realistic model
  - underlying channel may flip bits in packet
    - bit errors occur in the physical components of a network as a packet is transmitted, propagates, or is buffered
    - checksum to detect bit errors
- how people might deal with such a situation?
  - conversation over a phone
  - the message taker might say “OK” after each sentence has been heard, understood, and recorded
  - if the message taker hears garbled sentence, the message sender is asked to repeat



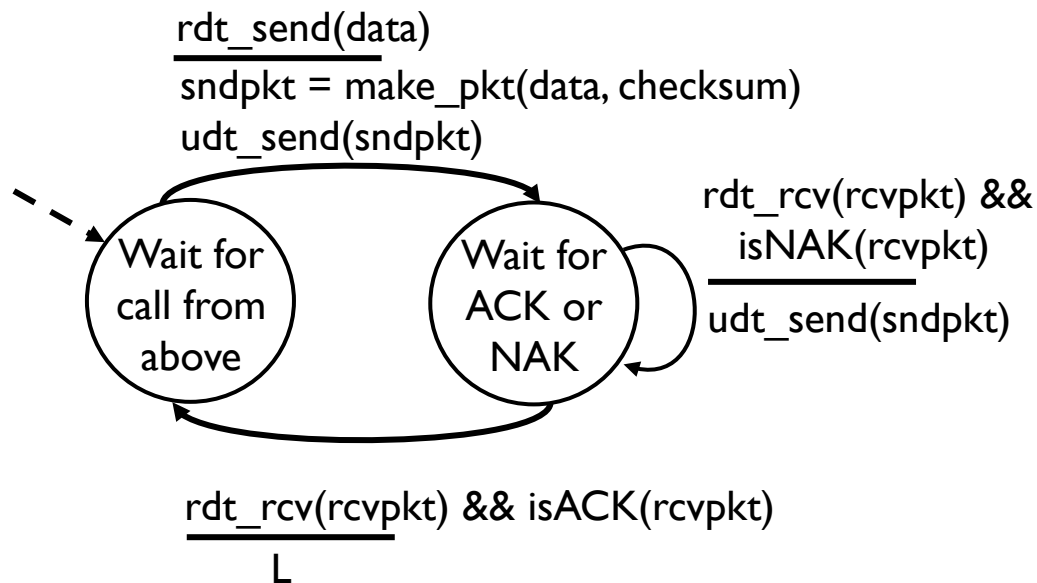


# rdt2.0: Channel with Bit Errors

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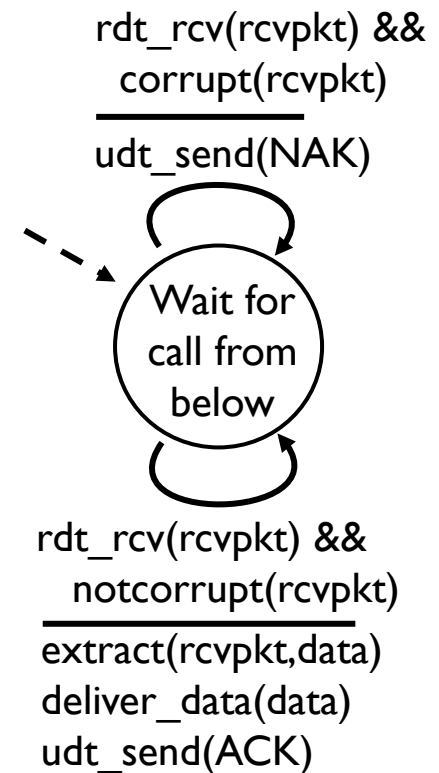
- the question: how to recover from errors?
  - *positive acknowledgements (ACKs)*:
    - receiver explicitly tells sender that pkt received OK with *ACKs*
  - *negative acknowledgements (NAKs)*:
    - receiver explicitly tells sender that pkt had errors with *NAK*
    - sender retransmits pkt on receipt of *NAK*
- new mechanisms in **rdt2.0** (beyond **rdt1.0**):
  - error detection
  - *receiver feedback*: control msgs (*ACKs* and *NAKs*) from receiver to sender
    - positive acknowledgement (*ACKs*)
    - negative acknowledgement (*NAKs*)

# rdt2.0: FSM specification

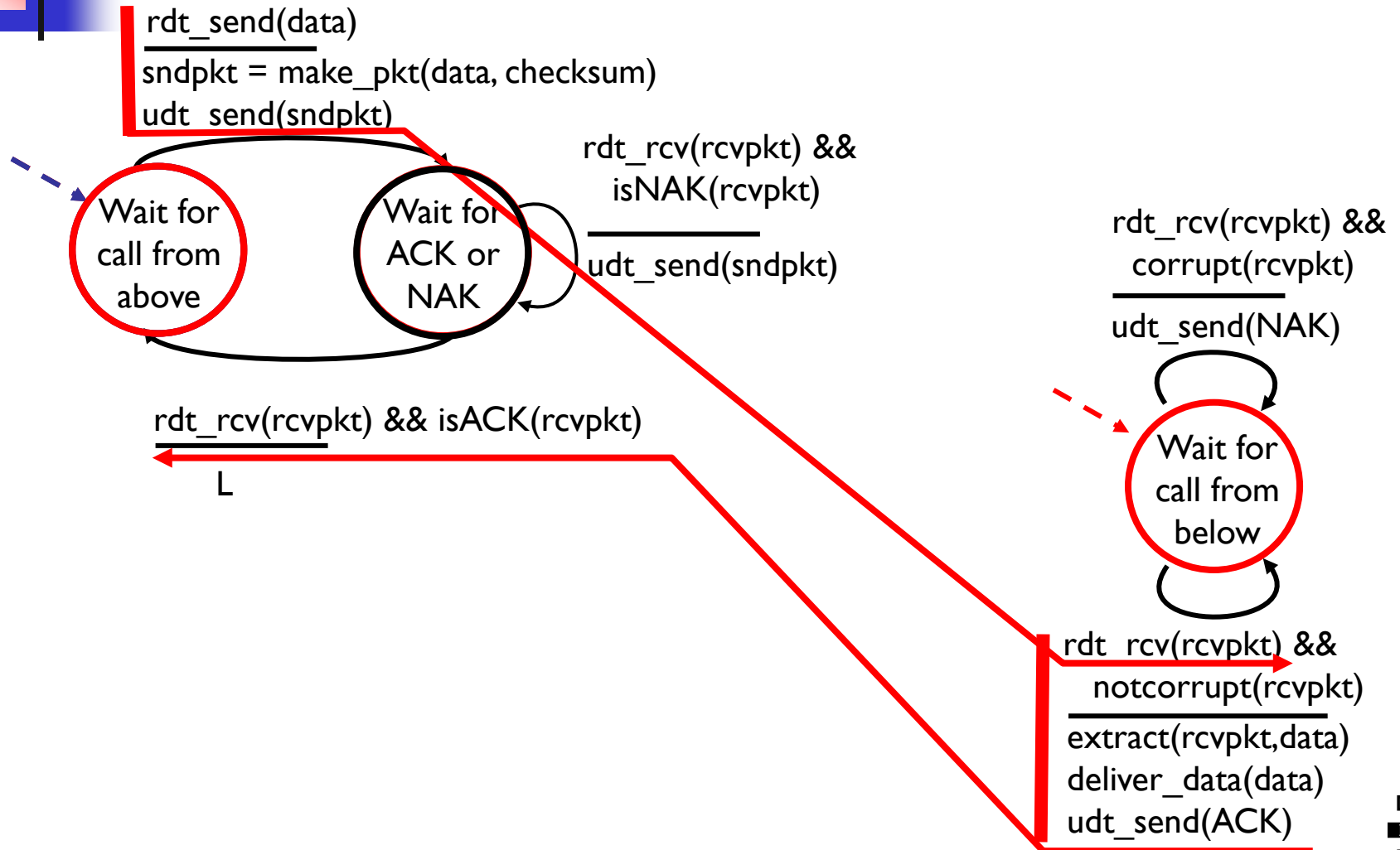


sender

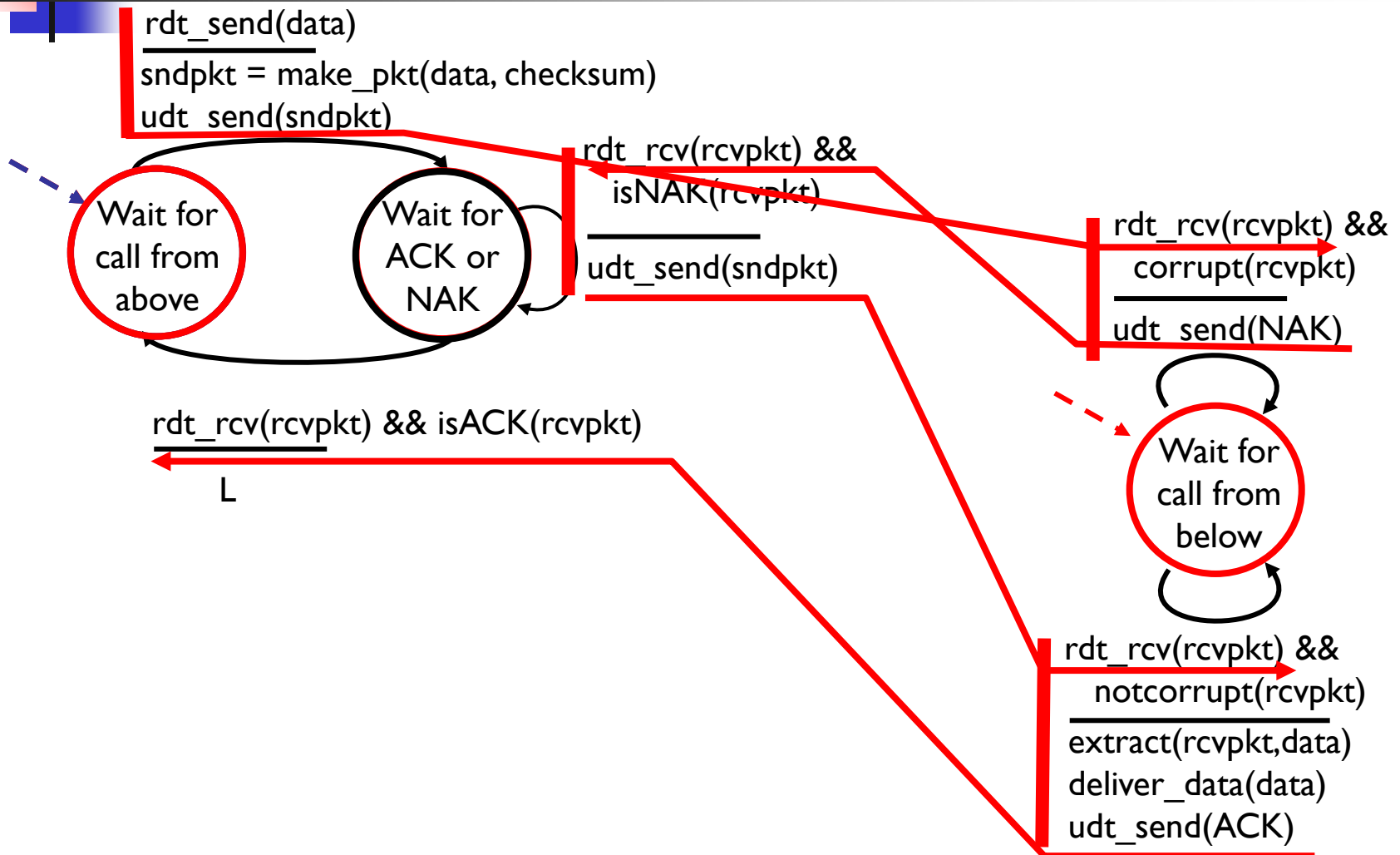
receiver



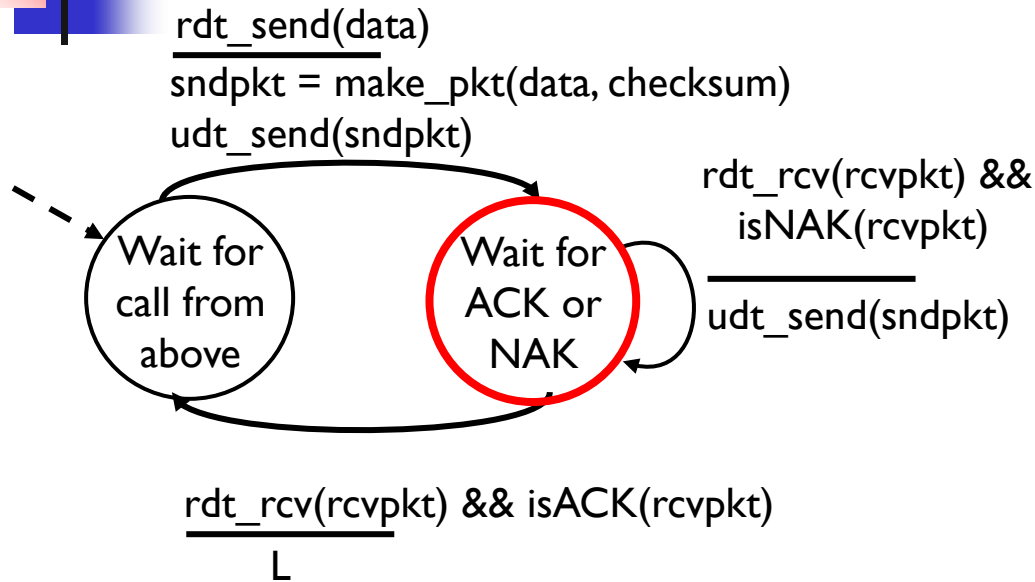
# rdt2.0: Operation with No Errors



## rdt2.0: Error Scenario



# rdt2.0: FSM specification



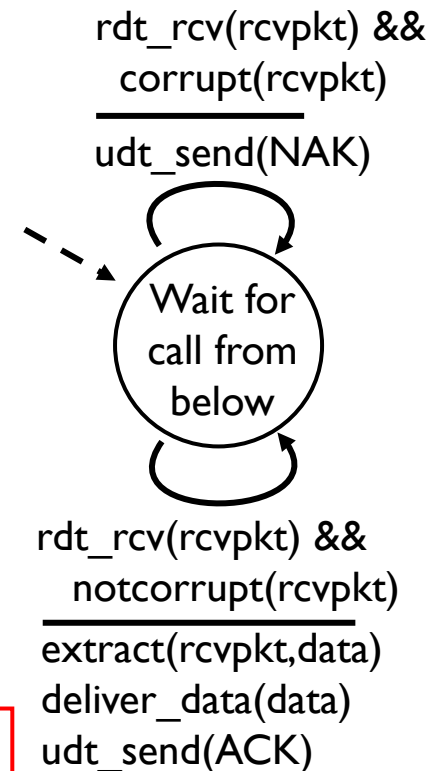
sender

## stop and wait

*sender sends one packet, then waits for receiver response.*

sender will not send a new data until it is sure that the receiver has correctly received the current packet.

receiver





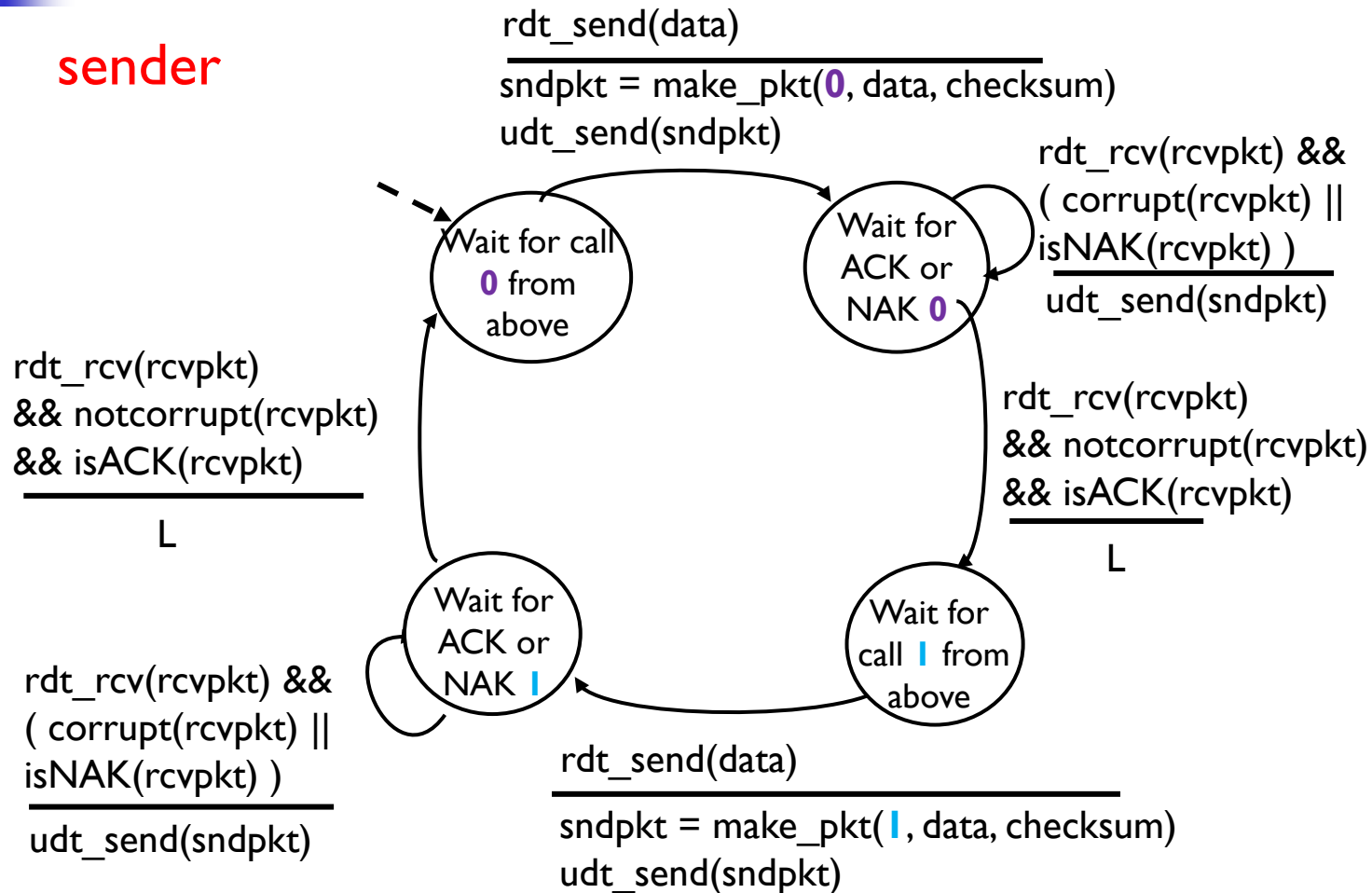
# rdt2.0 has a Fatal Flaw!

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- What happens if ACK/NAK corrupted?
  - add checksum to ACK/NAK to detect errors
  - more difficult question: how to *recover* from errors in ACK/NAK
  - sender doesn't know what happened at receiver!
    - sender does not know whether or not the receiver has correctly received the last piece of transmitted data
  - can't just retransmit: possible duplicate
- **Handling duplicates:**
  - sender retransmits current pkt if ACK/NAK garbled
  - sender adds *sequence number*, *1-bit*, to each pkt
  - receiver discards (doesn't deliver up) duplicate pkt

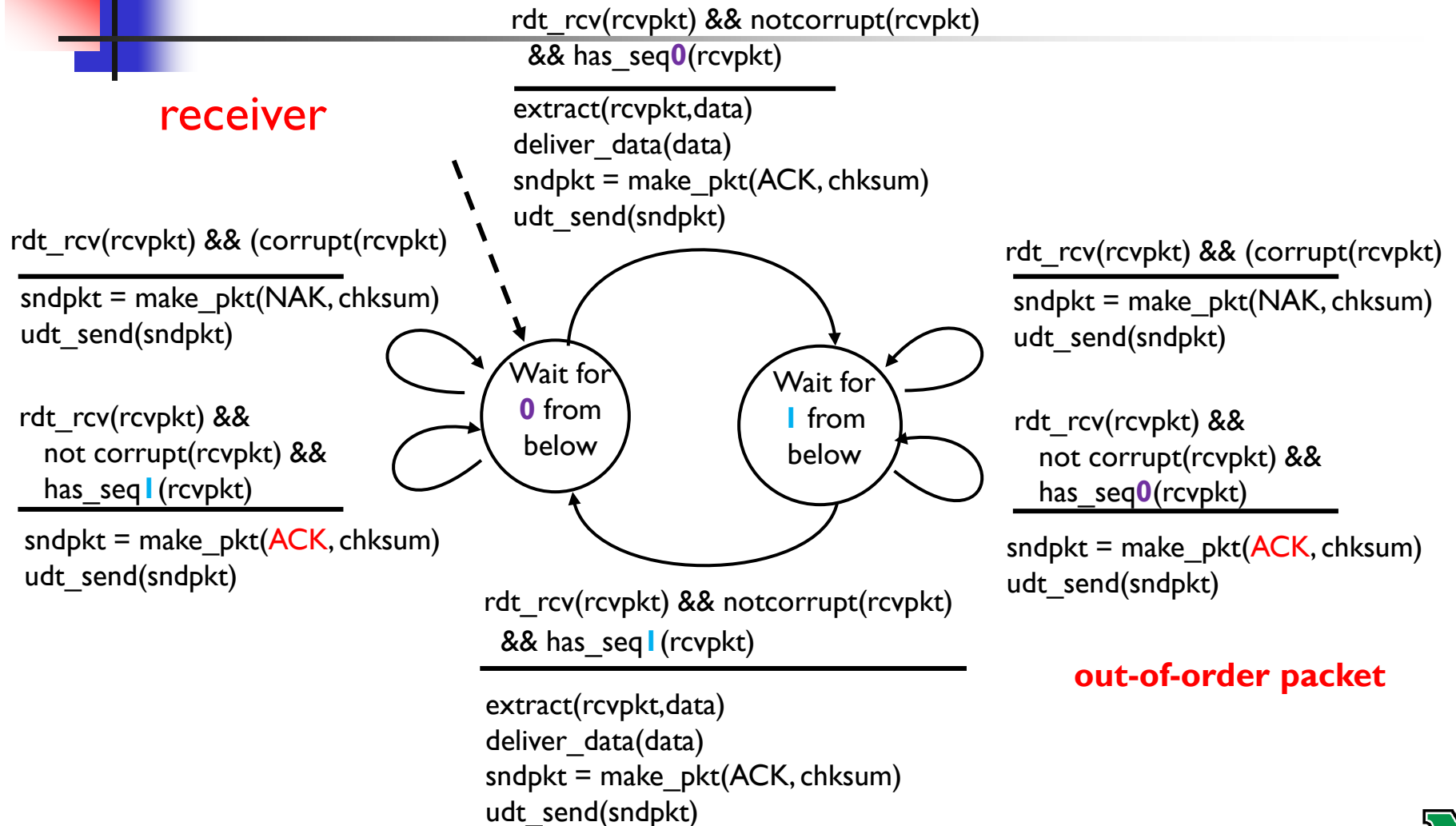
# rdt2.1: Sender, handles garbled ACK/NAKs

sender



# rdt2.1: Receiver, handles garbled ACK/NAKs

receiver







# rdt2.1: Discussion

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- 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



# rdt3.0: Channels with Errors and Loss

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- **new assumption:** underlying channel can also *lose packets* (data or ACKs)
  - two additional concerns:
    - how to detect packet loss?
    - what to do when packet loss occurs?
      - checksum, seq. #, ACKs, retransmissions will be of help
- **approach:** sender *waits* “reasonable” amount of *time* for ACK
  - detecting & recovering from lost packets from the sender side
  - how long must the sender wait?
    - at least as long as a *round trip time (RTT)* between sender and receiver



# rdt3.0: Channels with Errors and Loss (cont.)

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- **approach:** sender waits “reasonable” amount of time for ACK (cont.)
  - 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** (the sender):
    - (1) start the timer each time when a packet is sent
    - (2) respond to a timer interrupt
    - (3) stop the timer



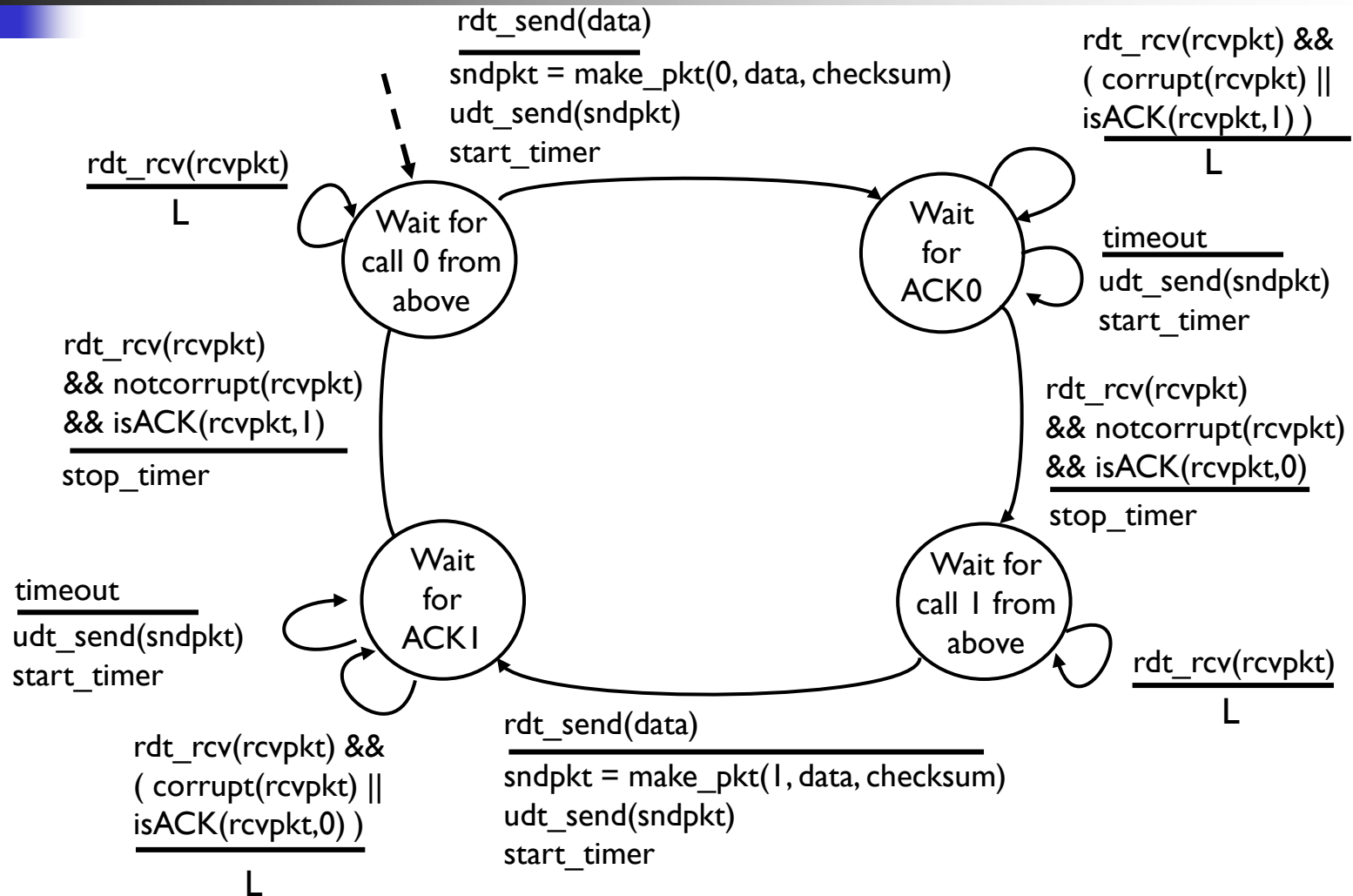
# rdt3.0: Channels with Errors and Loss (cont.)

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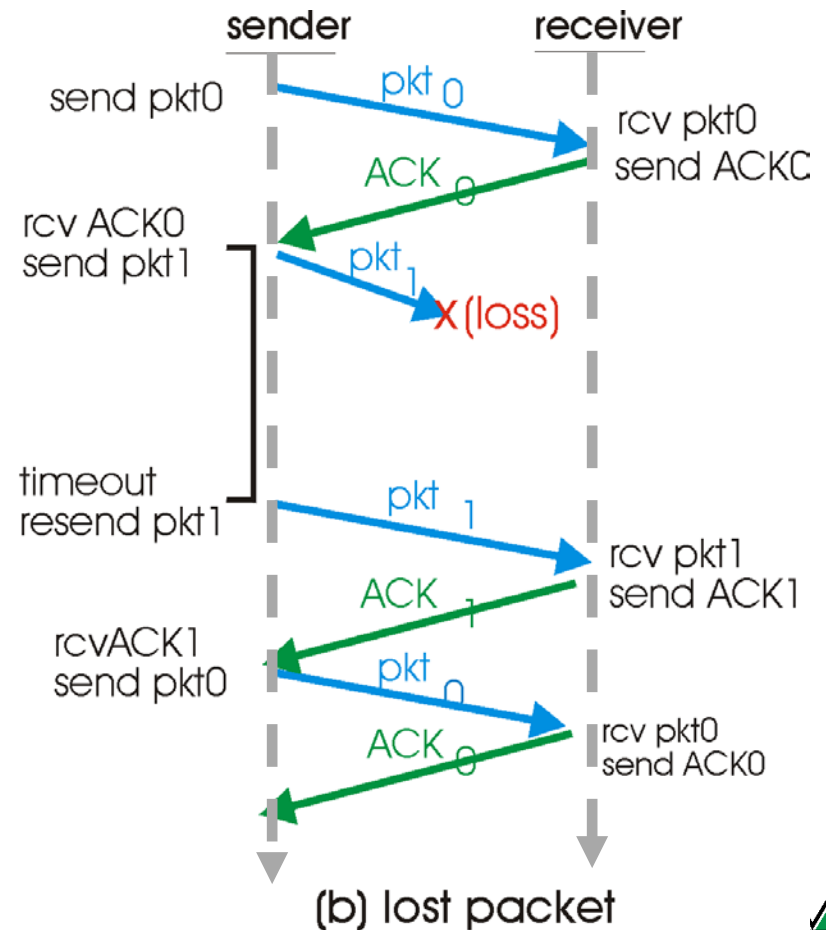
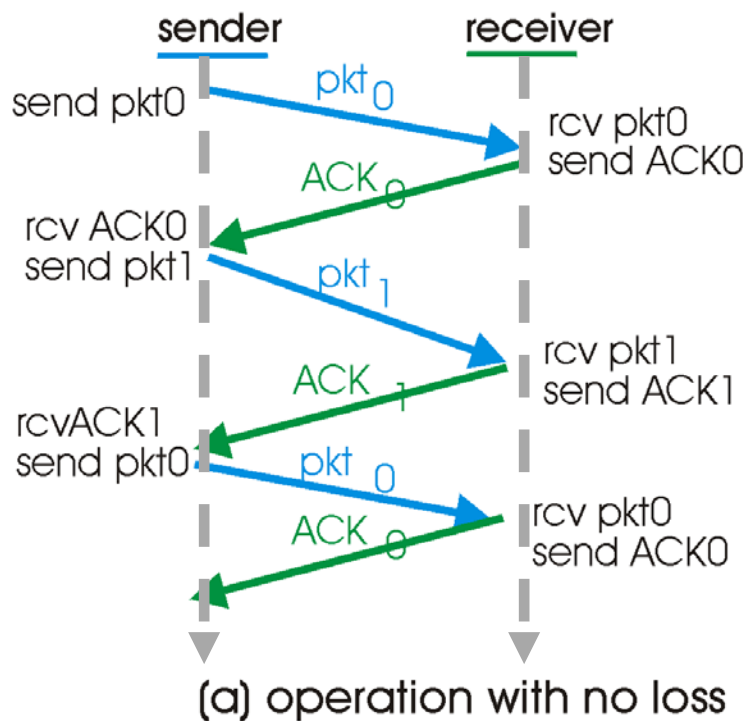
- from the sender's point of view, the sender does NOT know whether
  - a data packet was lost
  - an ACK was lost
  - if the packet or ACK was simply overly delayed
  - → in all case, the sender just retransmits!!

# rdt3.0: Sender

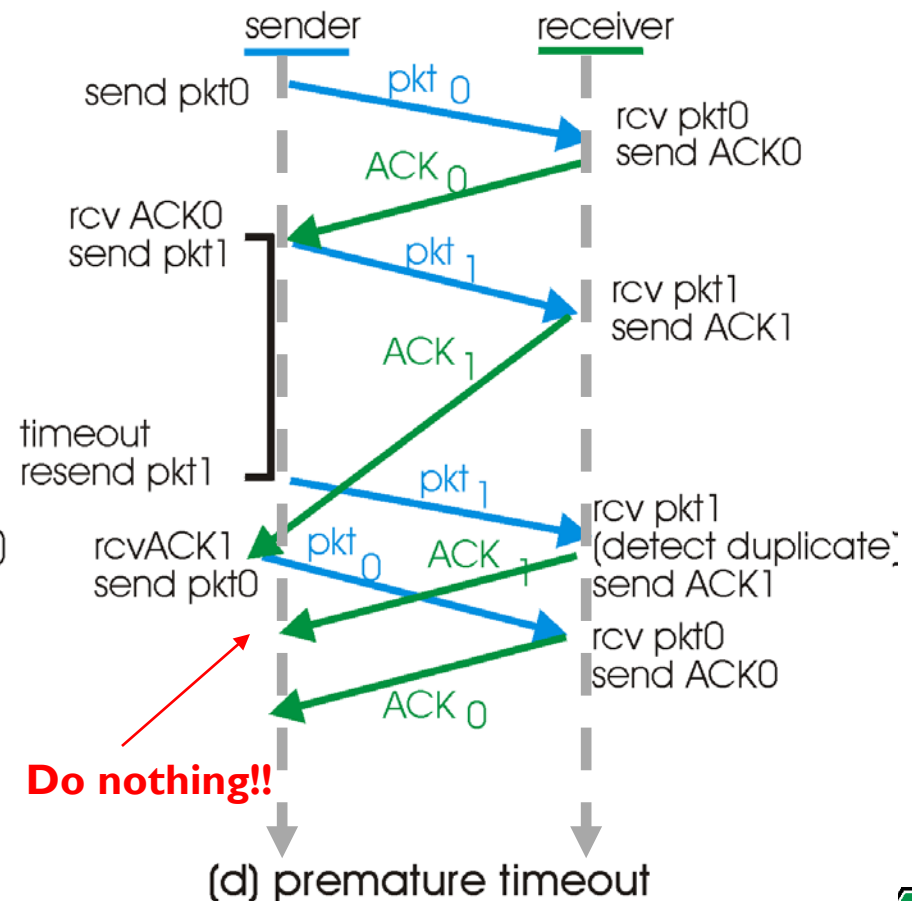
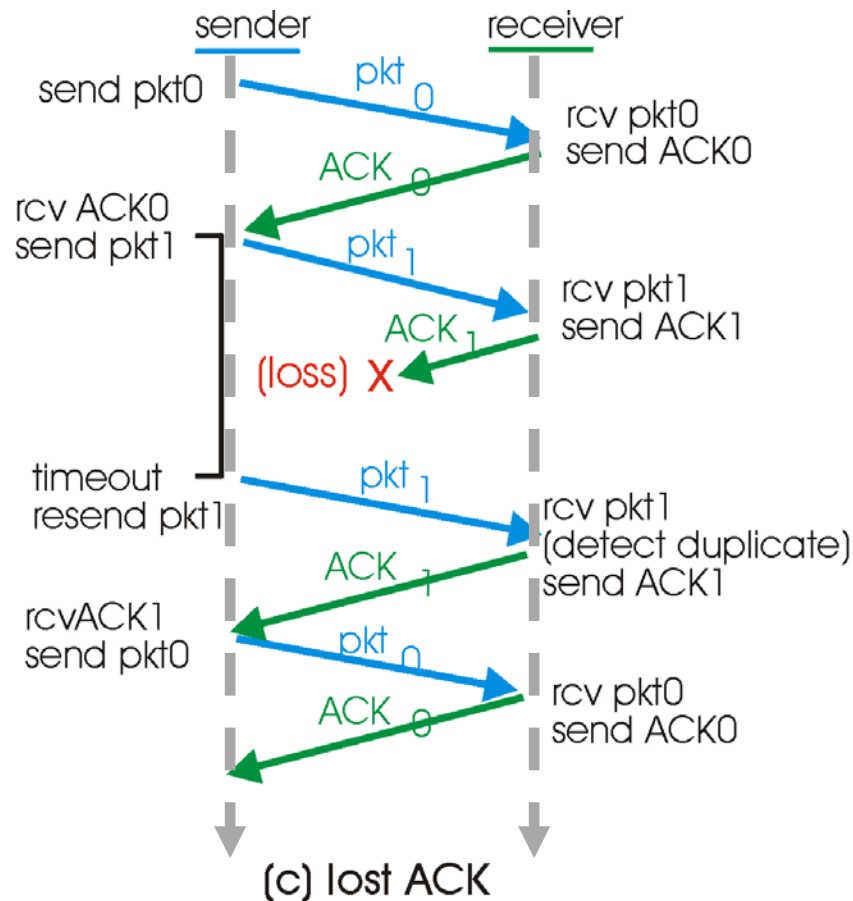
What about a receiver side?



# rdt3.0 in Action



## rdt3.0 in Action (cont.)





# Performance of rdt3.0

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- rdt3.0 works, but **performance stinks**
  - For example: 1 Gbps link, 15 ms end-to-end delay, 1KB packet:

$$T_{\text{transmit}} = \frac{L \text{ (packet length in bits)}}{R \text{ (transmission rate, bps)}} = \frac{8000 \text{ bit/pkt}}{10^{**}9 \text{ bit/sec}} = 8 \text{ microsec}$$

- RTT: the speed-of-light propagation delay, approximately 30 msec
  - the time  $t$  when the last bit of the packet emerging at the receiver

$$t = \text{RTT}/2 + L/R = 15.008 \text{ msec}$$

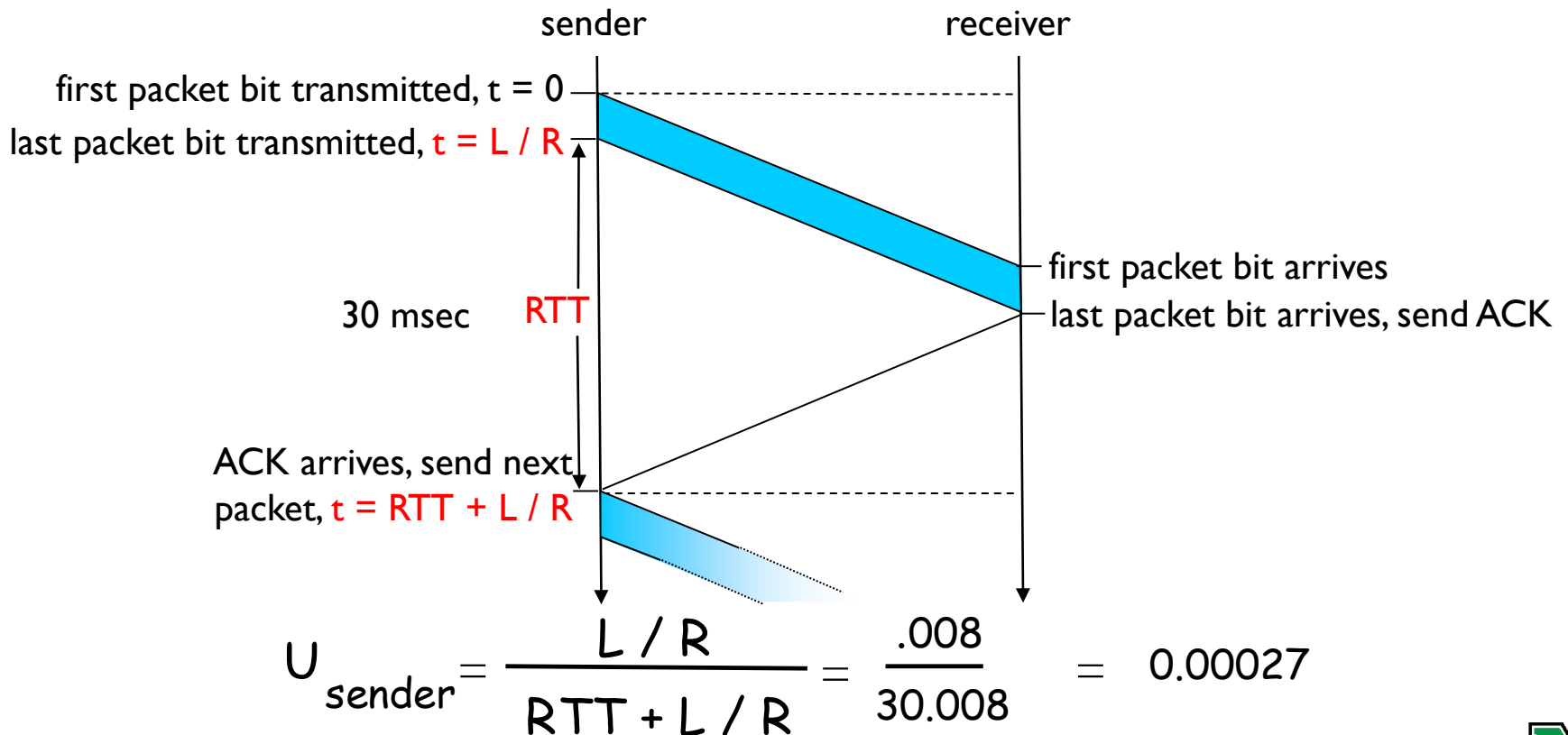
- When the ACK will come back?

$$t = \text{RTT} + L/R = 30.008 \text{ msec}$$



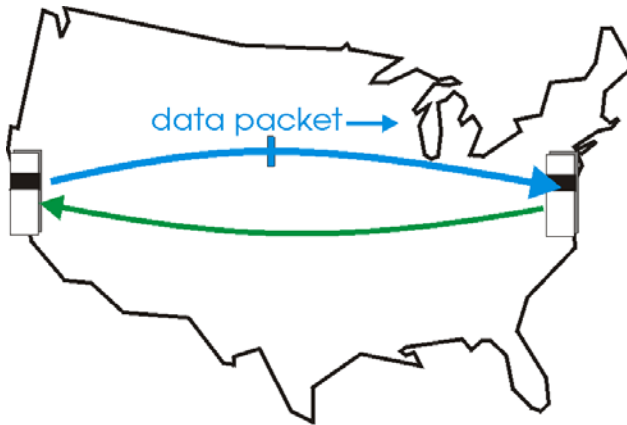
# rdt3.0: stop-and-wait Operation

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

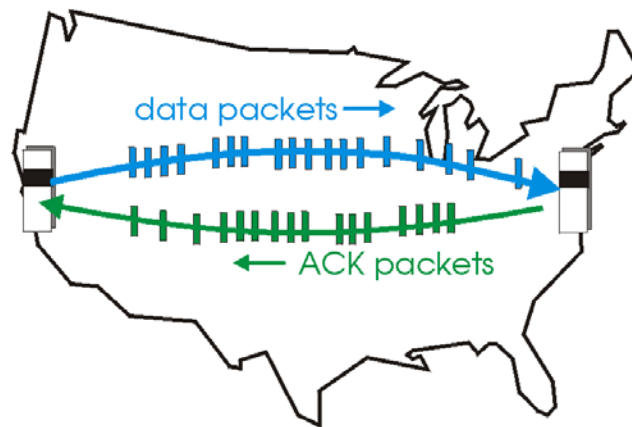


# Pipelined Protocols

- **pipelining**: sender allows multiple, “in-flight”, yet-to-be-acknowledged pkts. **sending multiple pkts without waiting for acknowledgments**
  - range of sequence numbers must be increased
  - buffering at sender and/or receiver
  - two generic forms of pipelined protocols: *Go-Back-N*, *selective repeat*

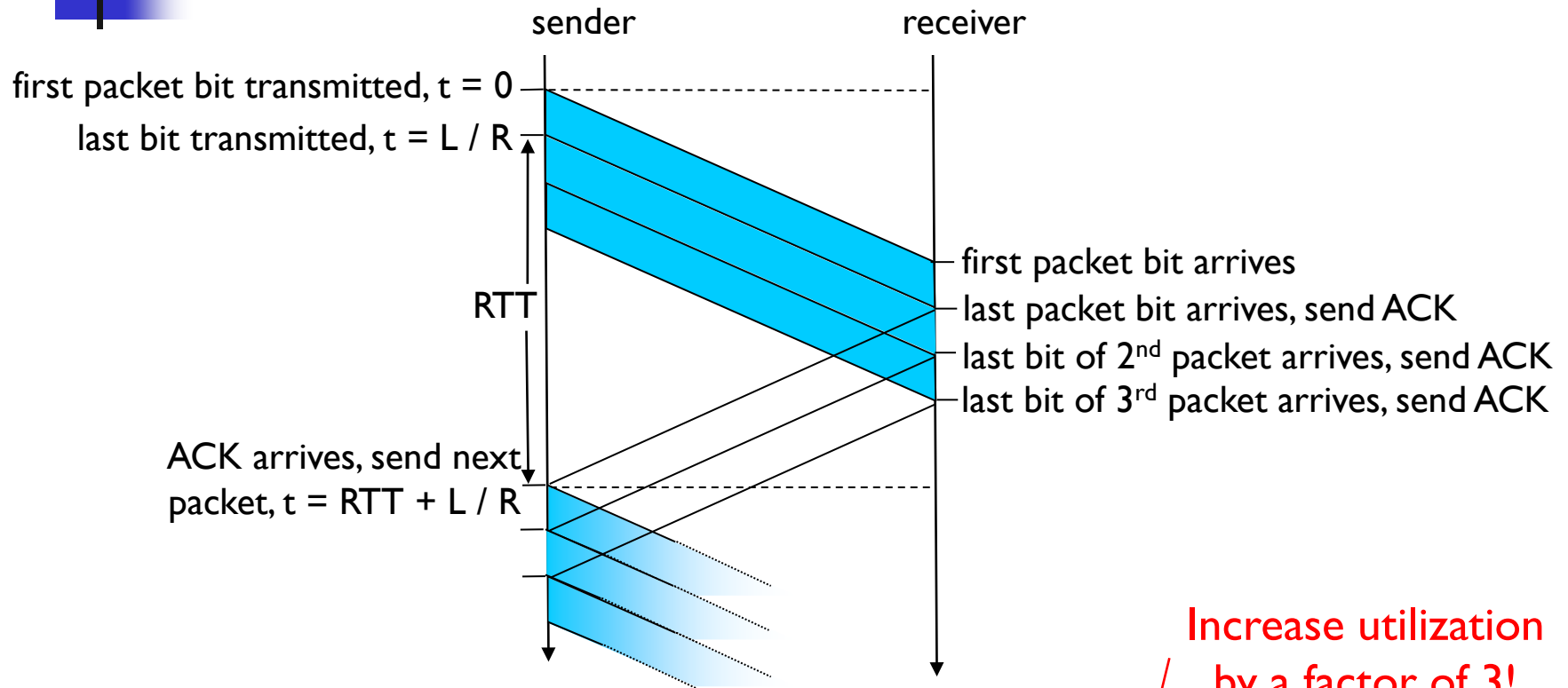


(a) a stop-and-wait protocol in operation



(b) a pipelined protocol in operation

# Pipelining: Increased Utilization



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