

**CSCE 463/612**

**Networks and Distributed Processing**

**Spring 2024**

**Transport Layer III**

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# Chapter 3: Roadmap

3.1 Transport-layer services

3.2 Multiplexing and demultiplexing

3.3 Connectionless transport: UDP

**3.4 Principles of reliable data transfer**

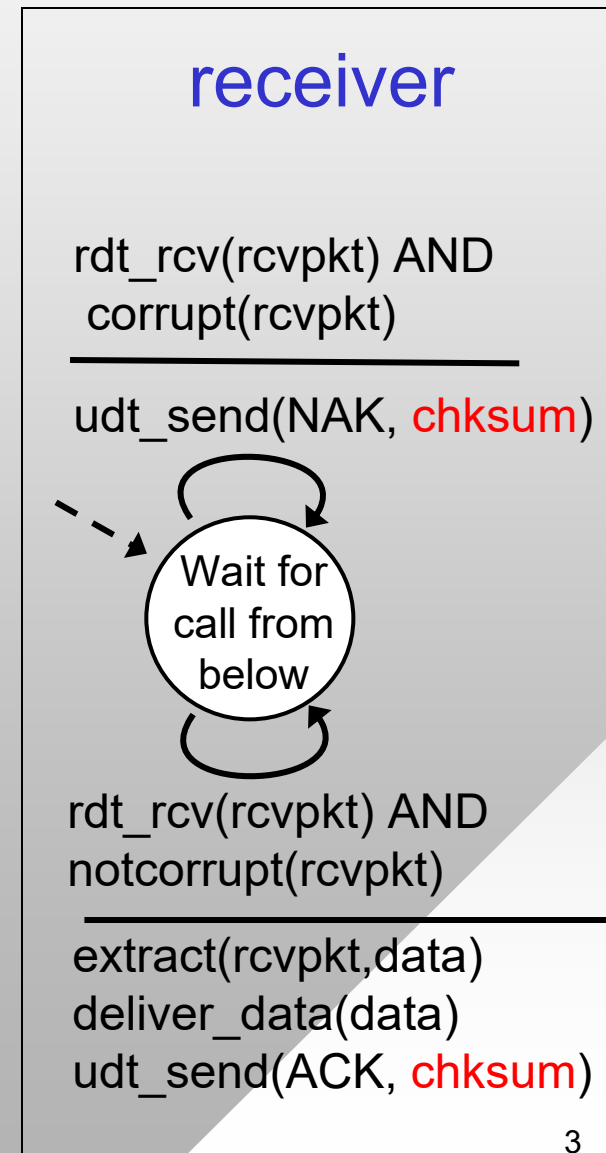
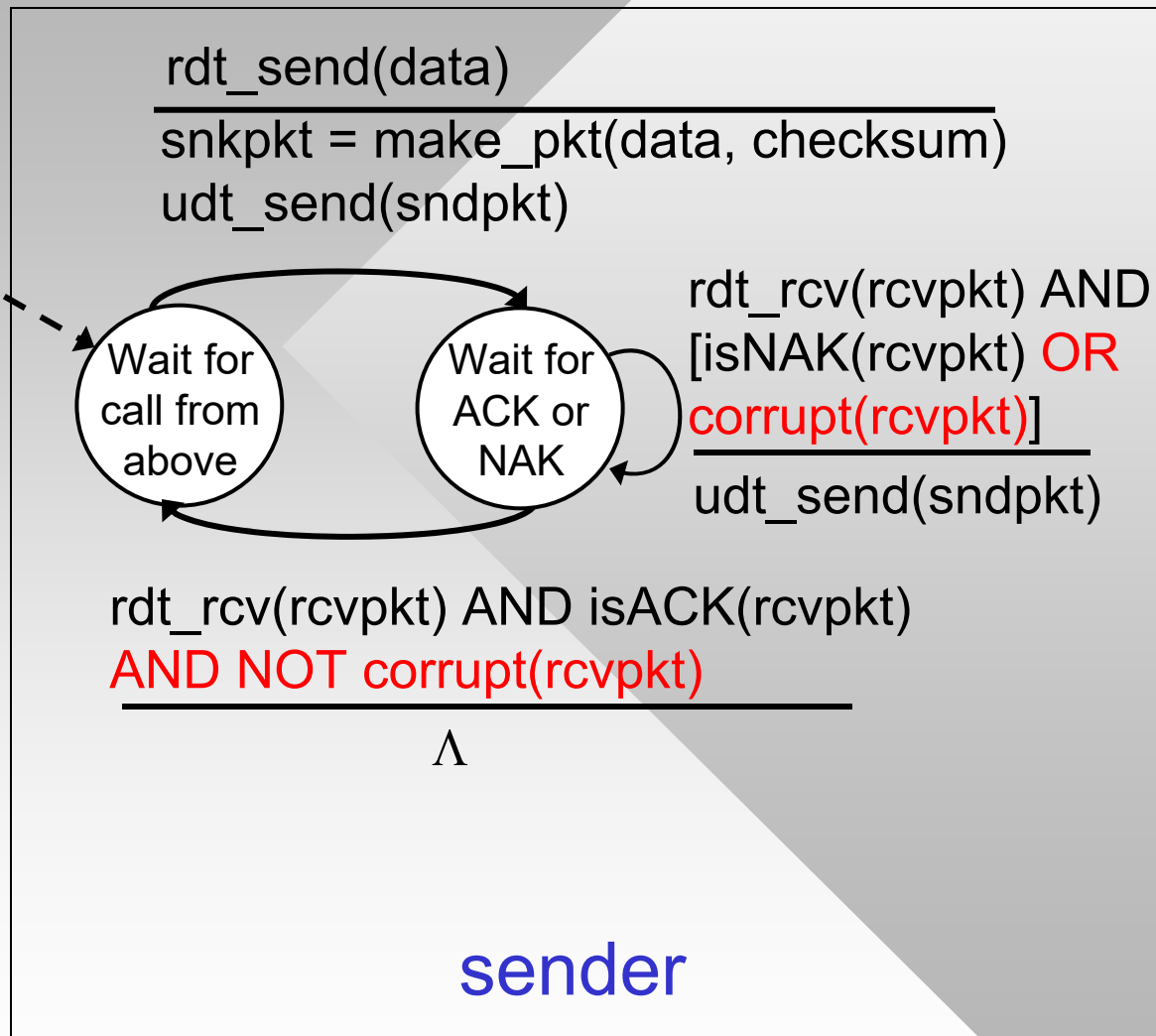
3.5 Connection-oriented transport: TCP

- Segment structure
- Reliable data transfer
- Flow control
- Connection management

3.6 Principles of congestion control

3.7 TCP congestion control

# Rdt2.0a: Handles Corrupted Feedback



Any problems?

# Rdt2.0 and Rdt2.0a Have Fatal Flaws

- Rdt 2.0 does not work when ACK/NAK is corrupted
  - Sender doesn't know what happened at receiver!
- Rdt 2.0a delivers duplicate packets to application

## Proper algorithm:

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

**Stop-and-Wait protocol:** sender sends one packet, then waits for receiver's response

# Rdt2.1: Sender, Handles Garbled ACK/NAKs

rdt\_send(data)

sndpkt = make\_pkt(0, data, checksum)

udt\_send(sndpkt)

rdt\_rcv(rcvpkt) AND  
[corrupt(rcvpkt) OR  
isNAK(rcvpkt)]

udt\_send(sndpkt)

rdt\_rcv(rcvpkt) AND  
NOT corrupt(rcvpkt)  
AND isACK(rcvpkt)

$\Lambda$

rdt\_rcv(rcvpkt) AND  
NOT corrupt(rcvpkt) AND  
isACK(rcvpkt)

$\Lambda$

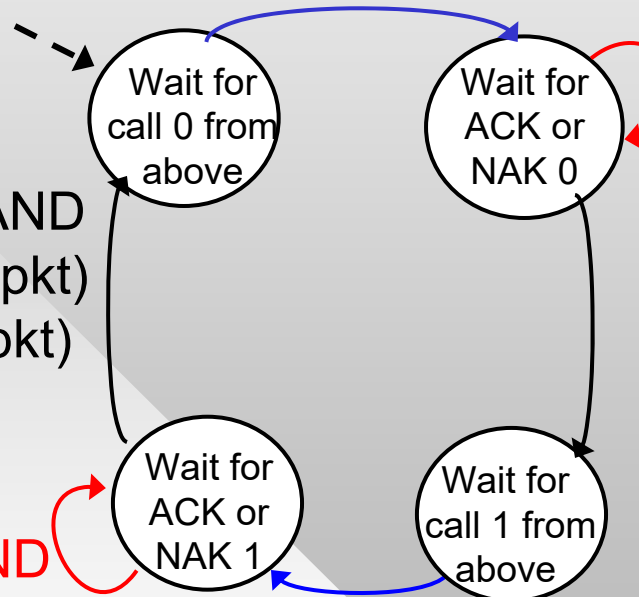
rdt\_rcv(rcvpkt) AND  
[corrupt(rcvpkt) OR  
isNAK(rcvpkt)]

udt\_send(sndpkt)

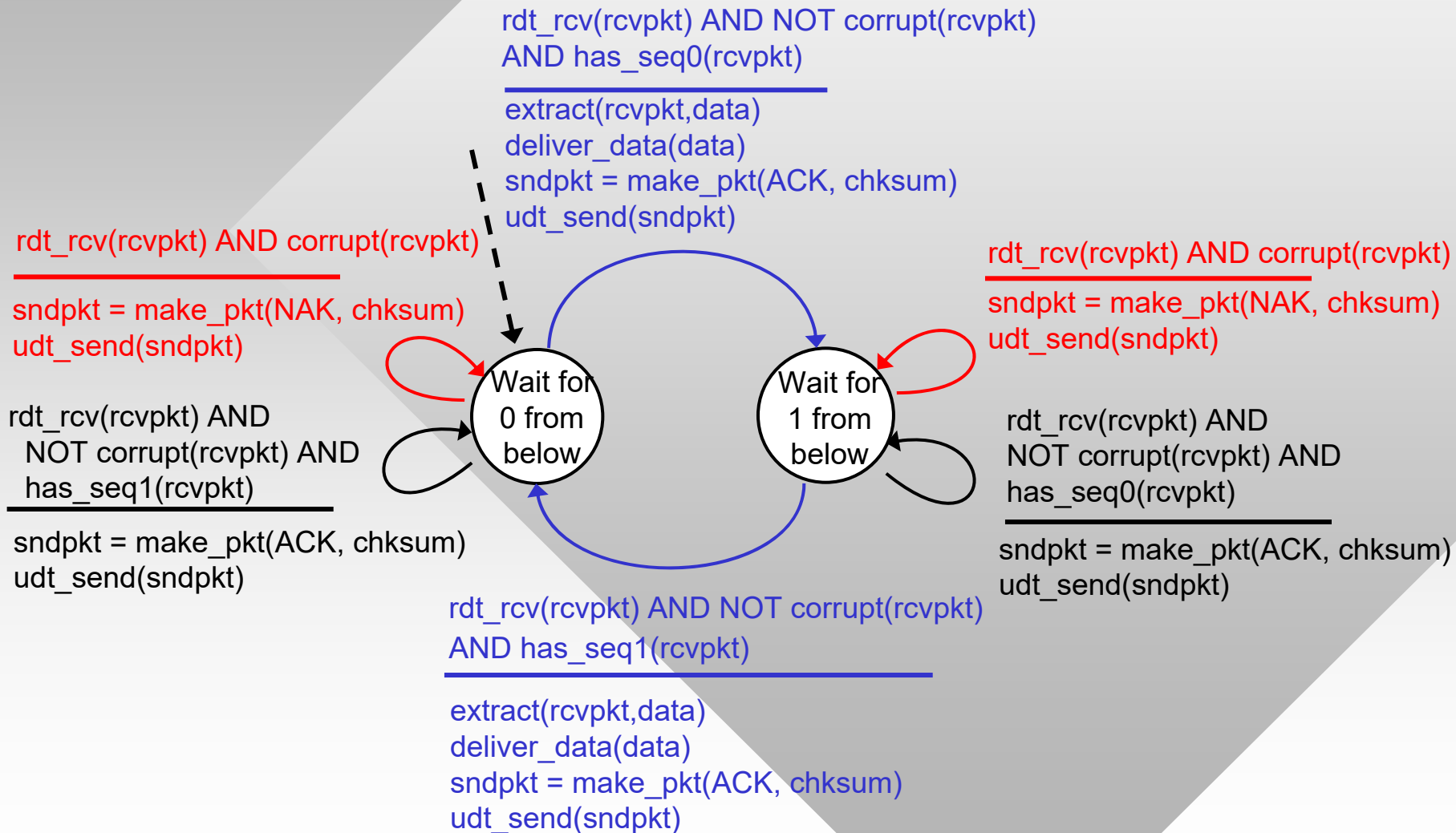
rdt\_send(data)

sndpkt = make\_pkt(1, data, checksum)

udt\_send(sndpkt)



# 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
  - Protocol must remember whether current pkt has 0 or 1 sequence number

## Receiver:

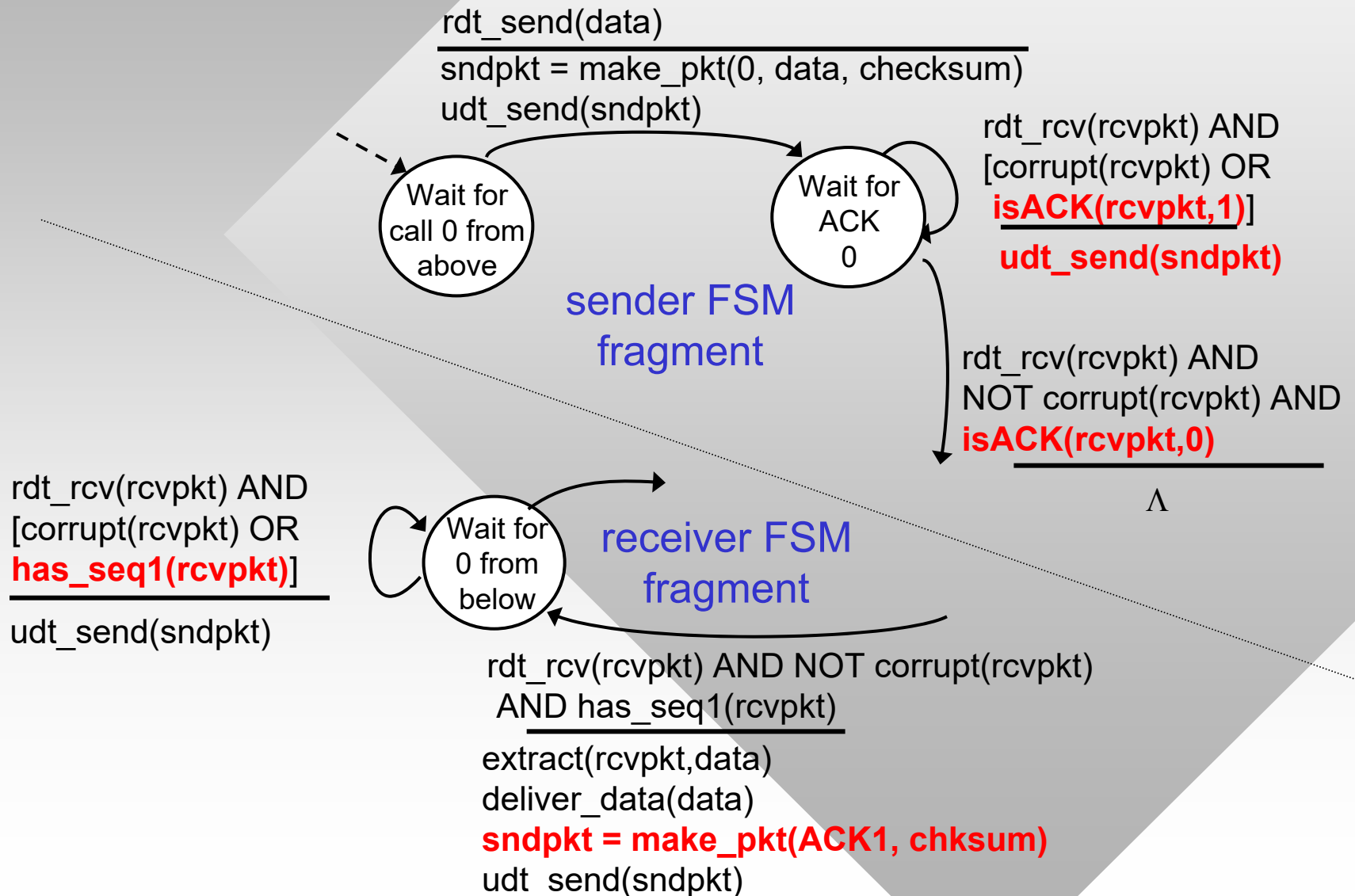
- Must check if received packet is duplicate
  - State indicates whether 0 or 1 is the expected packet seq #
- Note: receiver *cannot* know if its last ACK/NAK was received correctly at sender

## Rdt2.2: NAK-Free Protocol

- Same functionality as rdt2.1, using ACKs only
  - Most protocols are easier to generalize without NAKs
- Instead of NAKs, receiver sends an ACK for **last packet received correctly**
  - Receiver must *explicitly* include seq # of pkt being ACKed
- Duplicate ACK at sender results in same action as NAK: *retransmit current pkt*



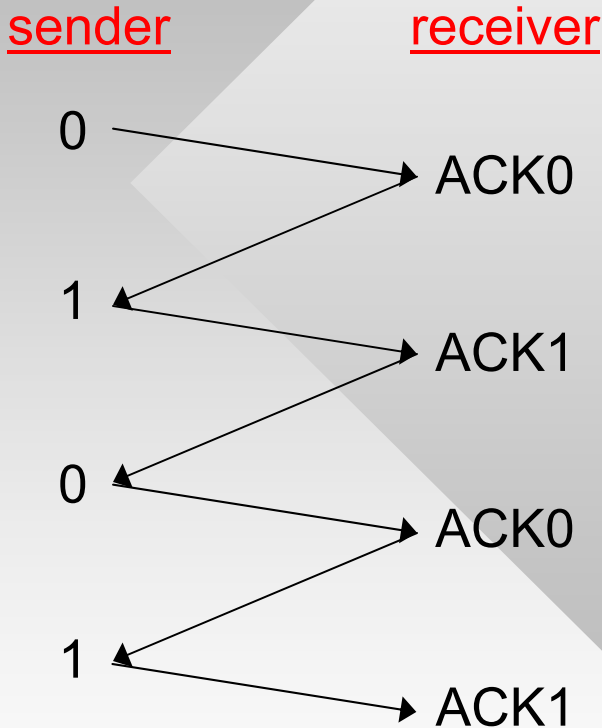
# Rdt2.2: Sender, Receiver Fragments



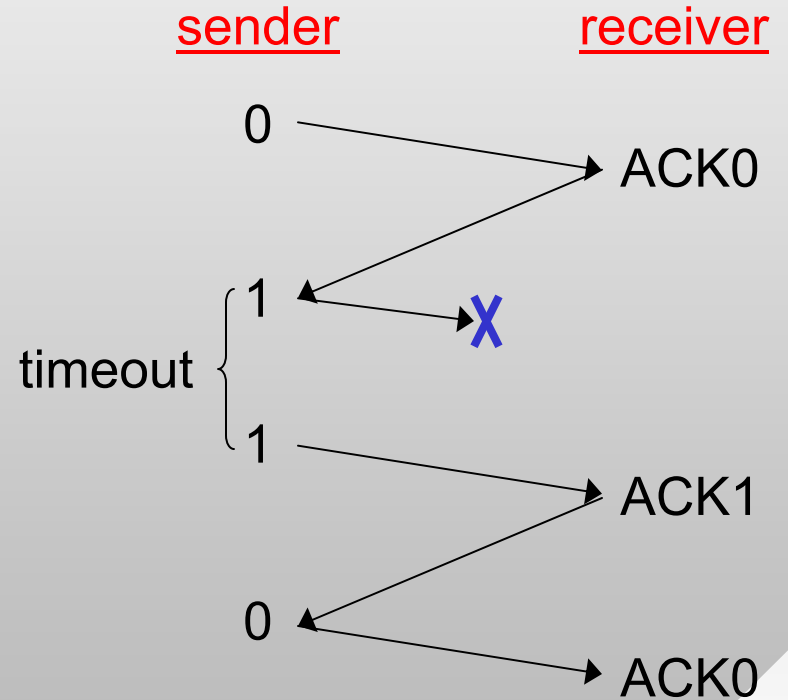
# Rdt3.0: Channels With Errors *and* Loss

- New assumption: underlying channel can also lose data packets or ACKs
  - Still no reordering
- Checksum, sequence numbers, ACKs, retransmissions will be of help, but **not enough**
  - Why not?
- Approach: sender waits a “reasonable” amount of time for ACK
  - Retransmits if no ACK received in this time
  - Sender requires a timer
- Redundant retransmission is now possible
  - ACK is lost or arrives after the timer expires
  - The use of seq. #'s in data pkts and ACKs already handles this

# Rdt3.0 in Action (No Corruption)

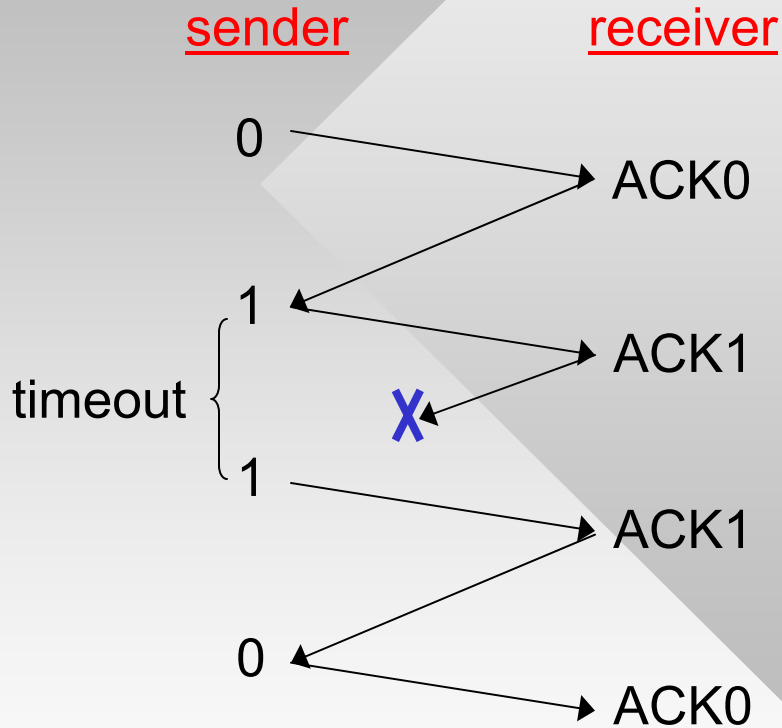


no loss

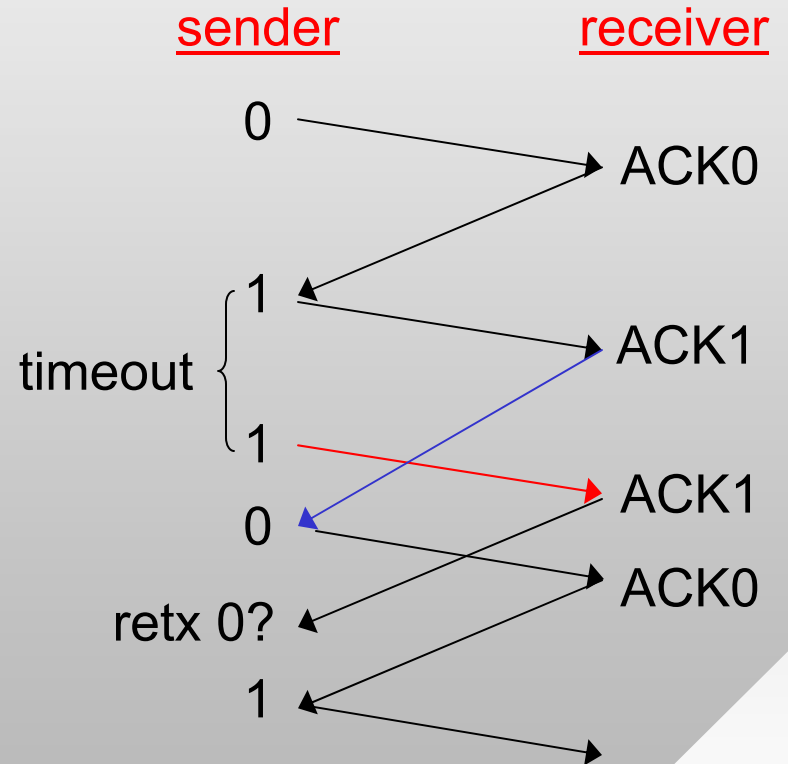


forward loss

# Rdt3.0 in Action (No Corruption)



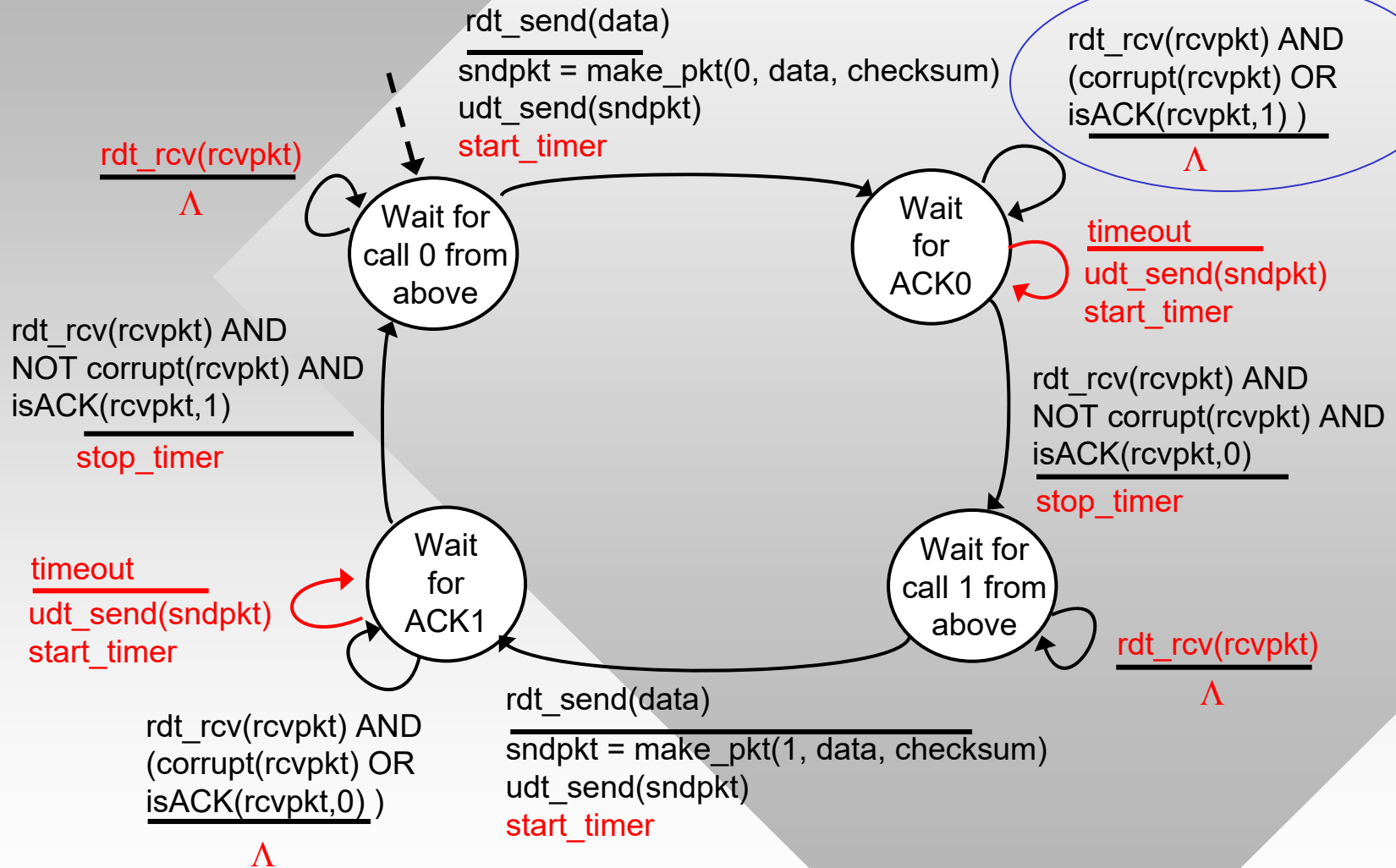
reverse loss



premature timeout

# Rdt3.0 Sender

Must not retx: ACK1 may be from a premature timeout on pkt1



# Performance of Rdt3.0

Notation: KB = Kilobyte;  
Kbps = Kilobits/sec  
Gbps = Gigabits/sec

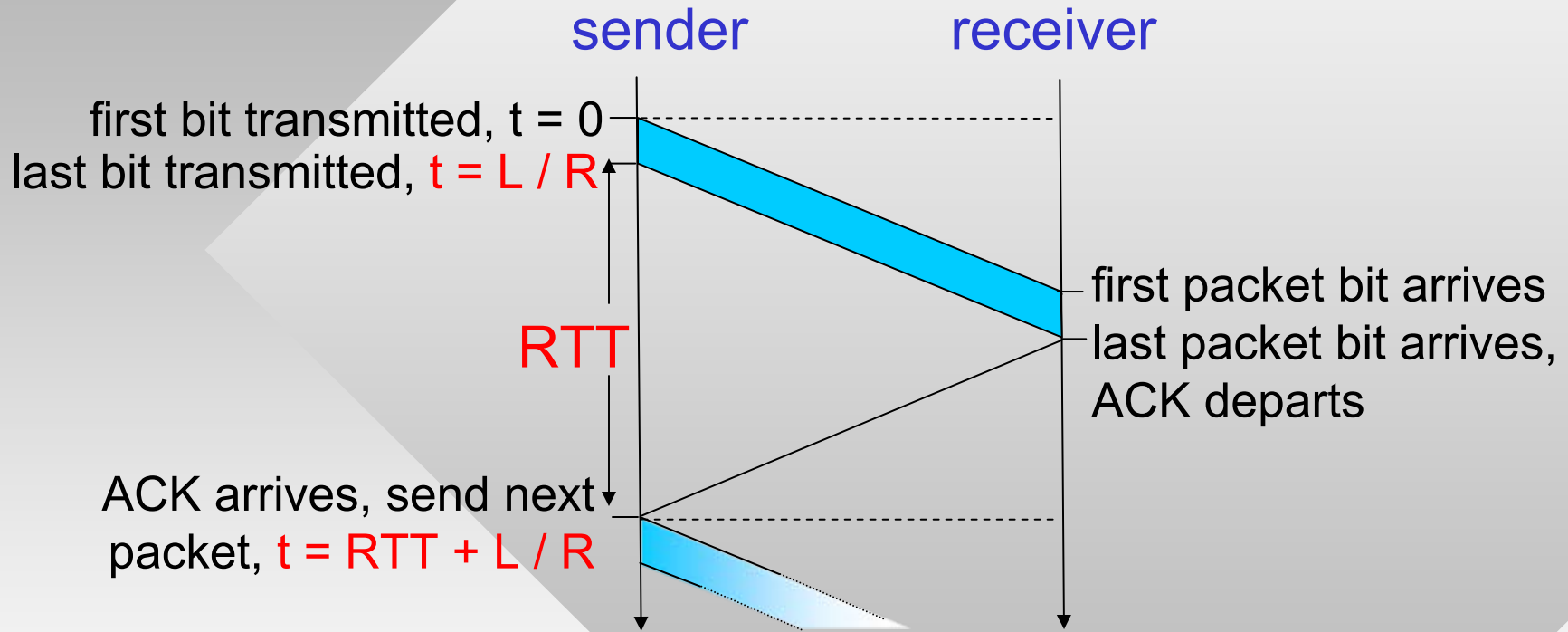
- Rdt 3.0 works, but performance is low
- Example: 1 Gbps link, 15 ms end-to-end propagation delay, 1 KB packets, no loss or corruption:

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

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

- Server spends 0.008 ms being busy and 30 ms being idle, thus its link utilization is **only 0.027%**
- 1-KB pkt every 30 ms → 264 Kbps throughput
- *Network protocol limits use of physical resources!*

# Rdt3.0: Stop-and-Wait Operation



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

## Performance of Rdt3.0

- Next assume that 10% of data packets are corrupted/lost (no loss in retransmissions or ACKs) and the timeout is 1 second
  - 90% of packets take  $(RTT + L/R) \approx 30$  ms to complete, while 10% require  $[\text{timeout} + RTT + 2L/R] \approx 1.03$  sec
  - Average per-packet delay  $0.9 \cdot 0.03 + 0.1 \cdot 1.03$  sec = 130 ms
  - Average rate 7.7 pkts/s or 61.5 Kbps
- Rdt3.0 similar to HTTP 1.0 or non-pipelined HTTP 1.1
- Next time we'll improve this using [pipelining](#), which allows multiple unack'ed packets at any time