COSC264
Introduction to Computer Networks and the Internet

Reliable data transfer: ARQ protocols

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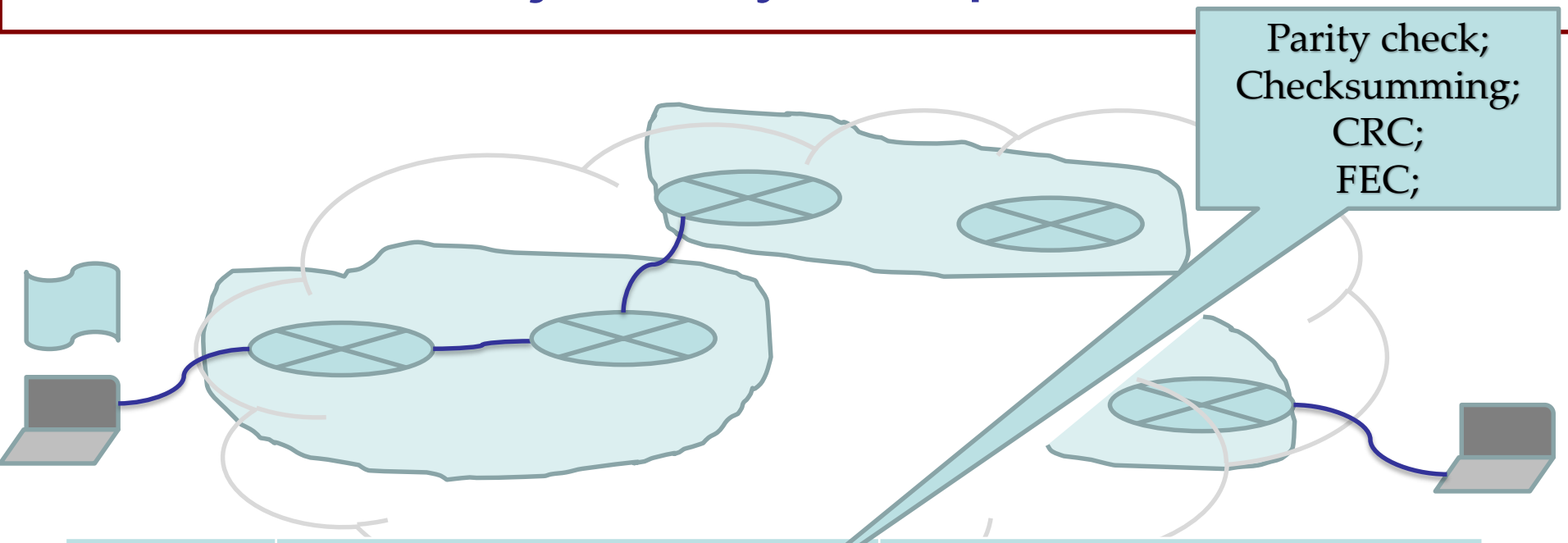
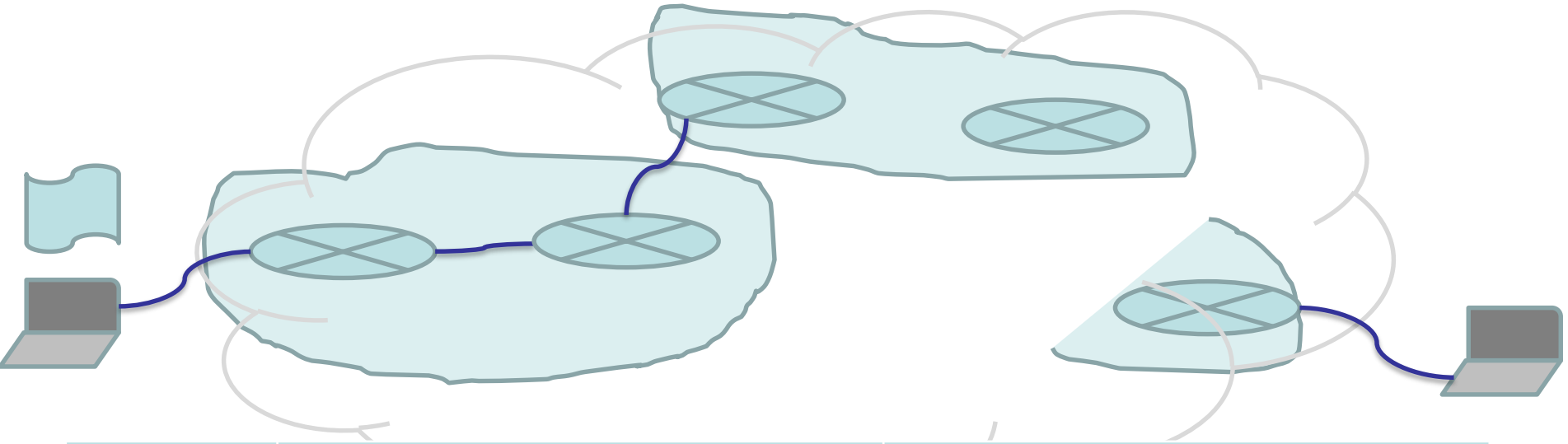
The journey of a packet



Problems	Causes	Solutions
Bit error	e.g., signal attenuation/noise	Error detection and correction
Buffer overflow	e.g., Speed-mismatch; Too much traffic;	Flow control and congestion control
Lost packet	e.g., buffer overflow at host/router	Acknowledgement and retransmission (ARQ)
Out of order	e.g. an early packet gets lost and retransmitted; a later one arrives first.	Acknowledgement and retransmission (ARQ)

The journey of a packet



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Outline

- ARQ (Automatic Repeat reQuest) Protocols
  - Alternating Bit Protocol
  - Go-Back-N
  - Selective Repeat (Selective Reject)
- Summary

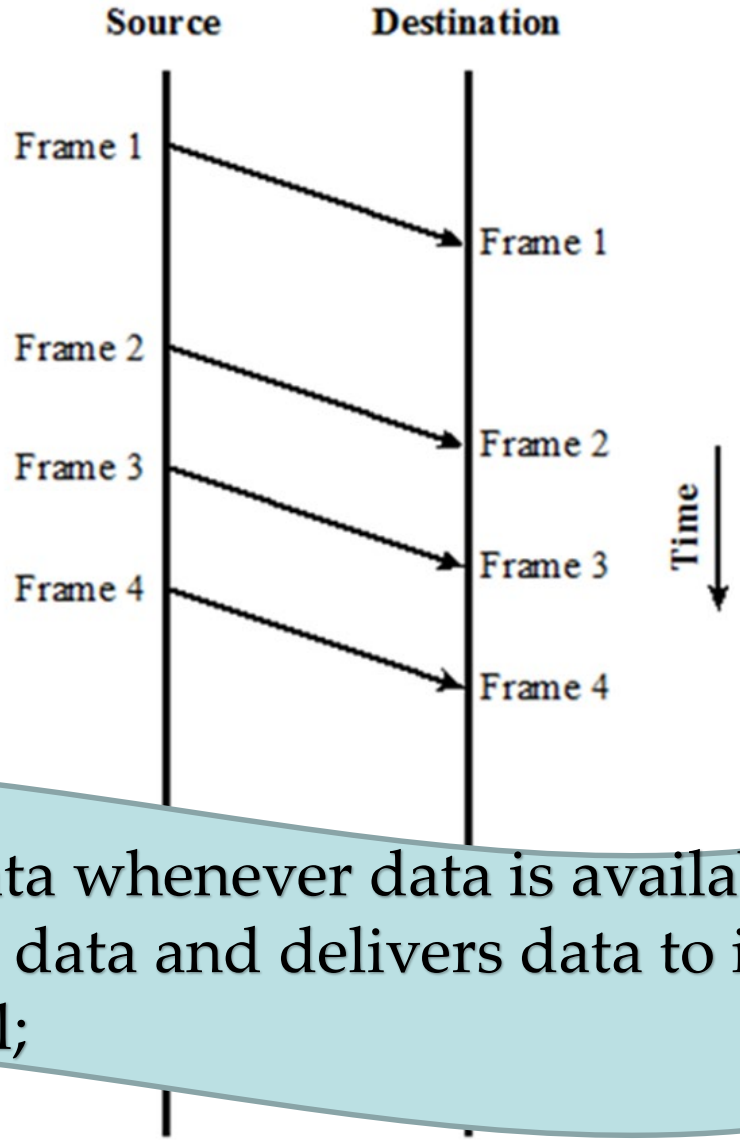
In a computer network setting, reliable data transfer protocols based on retransmission are known as ARQ (Automatic Repeat reQuest) protocols. [KR3]

Outline

- Reliable transfer protocols
  - rdt1.0: for a reliable channel
  - rdt2.0: for a channel with bit errors
  - rdt2.1: sender, handles garbled ACK/NAKs
  - rdt2.2: a NAK-free protocol
  - rdt3.0: channels with errors *and* loss
  - Pipelined protocols
    - Go-back-N
    - Selective repeat (Selective reject)

rdt 1.0: reliable data transfer over a perfectly reliable channel

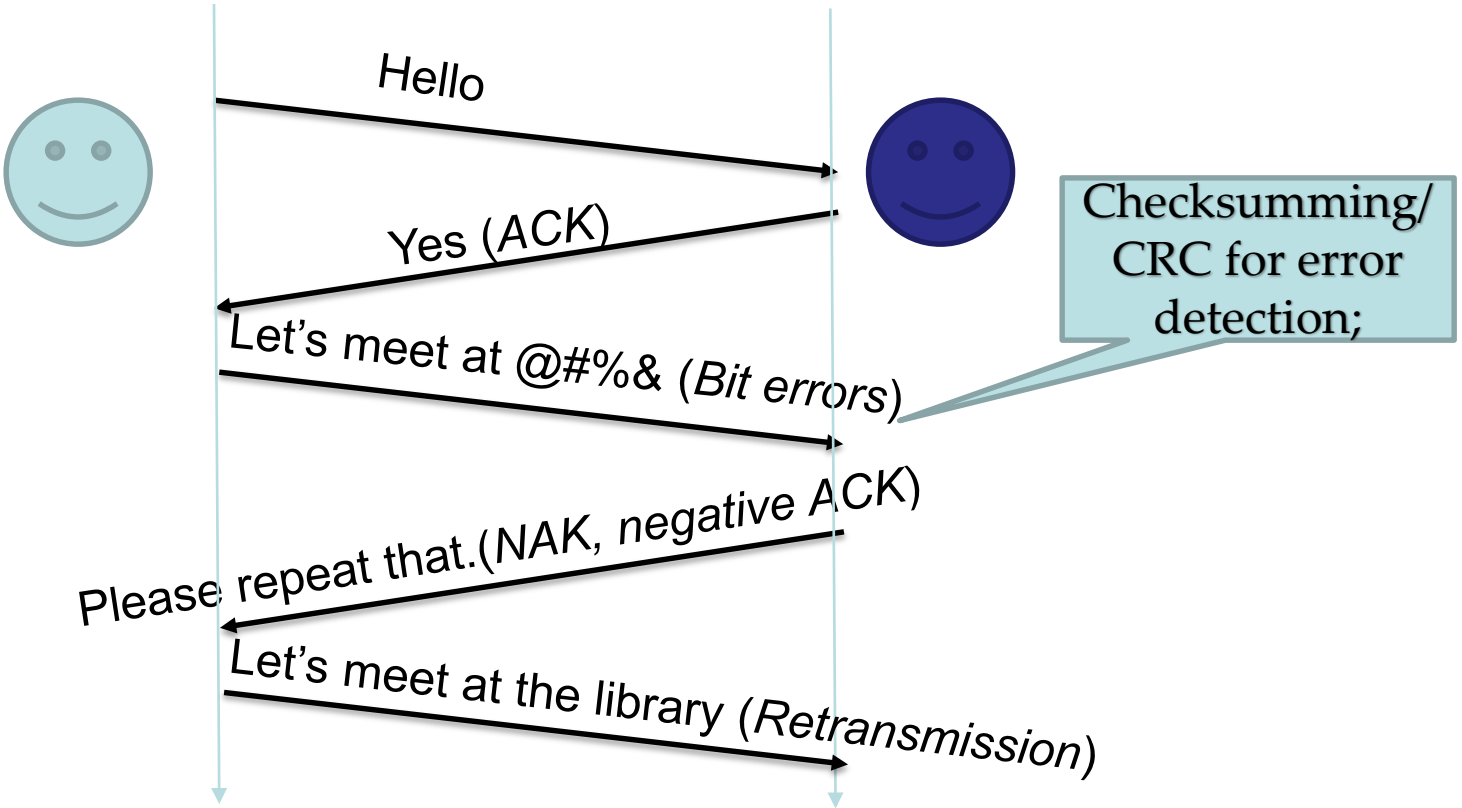
- A perfectly reliable channel;
  - no bit errors
  - no buffer overflow
  - no out-of-order
  - no packet loss



The sender sends data whenever data is available; the receiver receives data and delivers data to its upper-layer protocol;

rdt 2.0: reliable data transfer over a *lossless* channel with bit errors

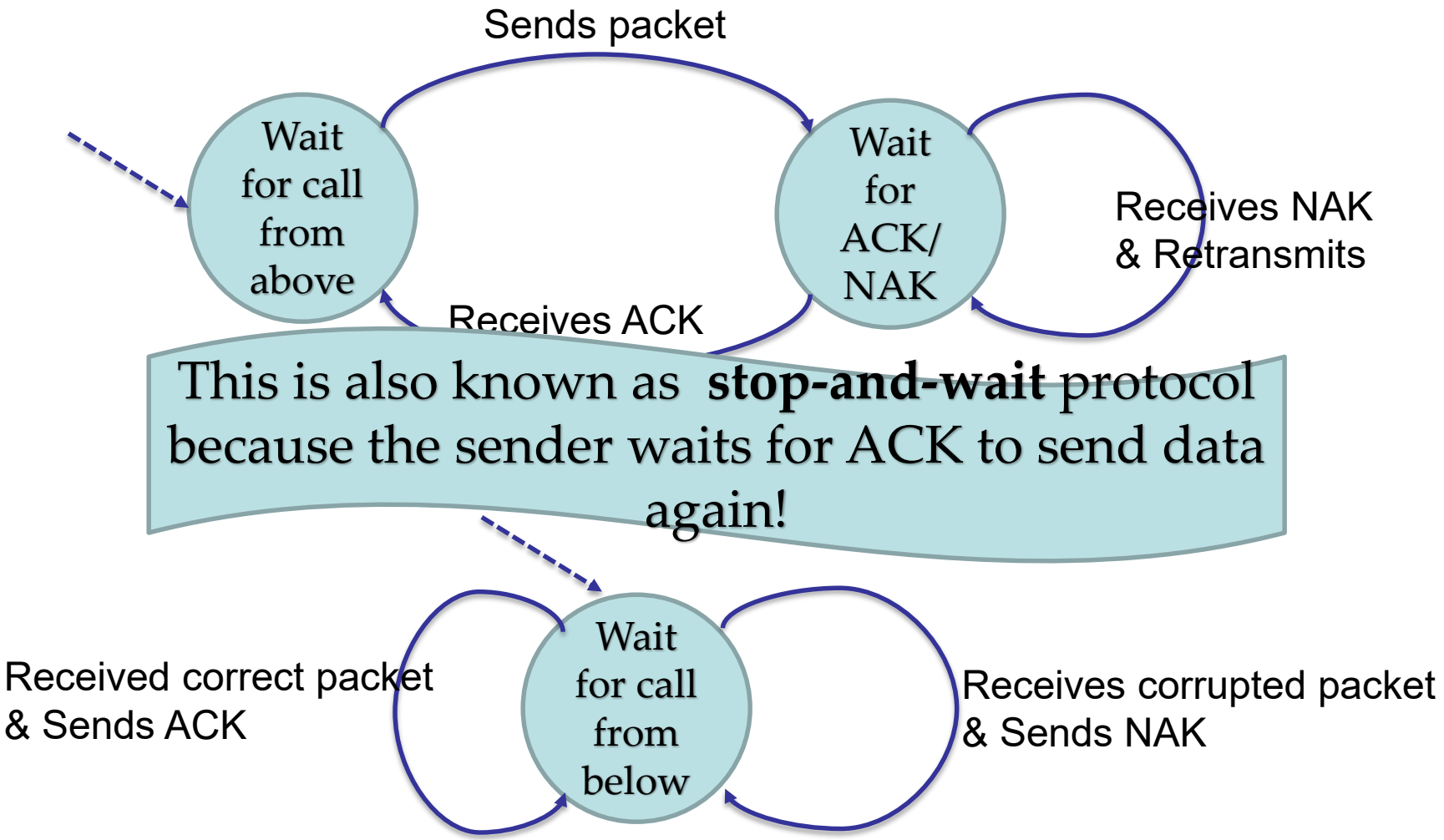
- A channel with only bit errors;



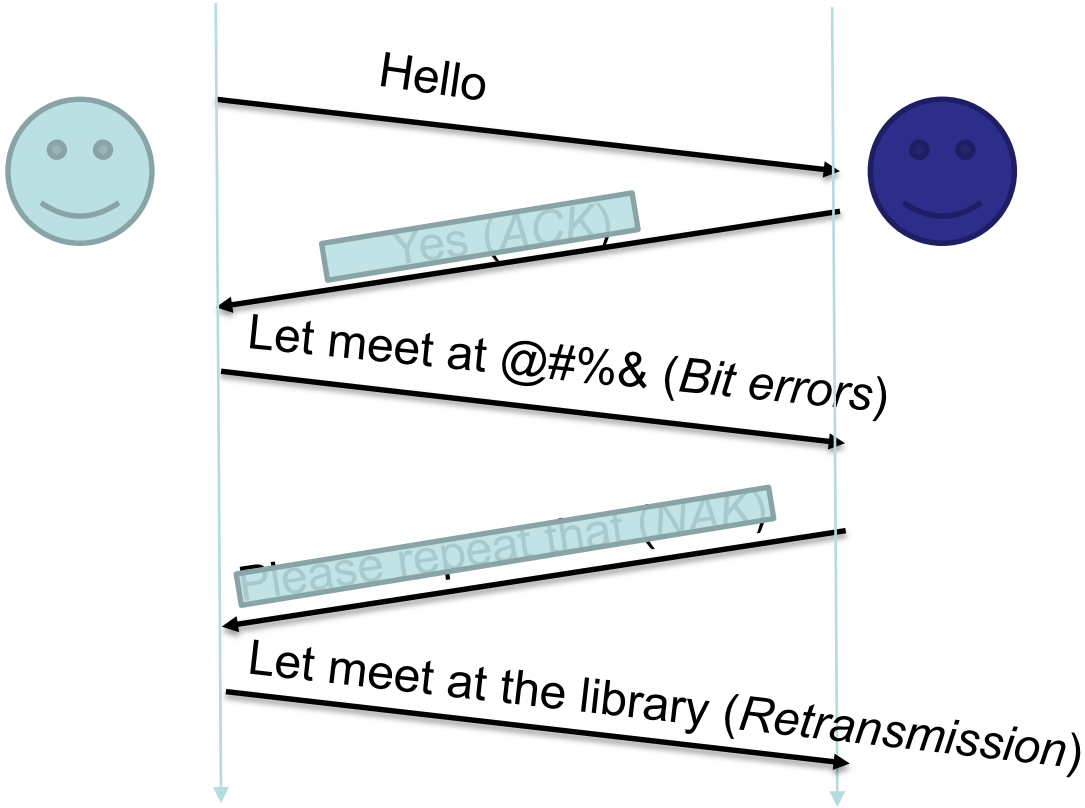
rdt 2.0 protocol

- Sender
  - When packet is ready, send it and wait for ACK/NAK;
  - If NAK is received, retransmits the last packet and wait for ACK/NAK;
  - If ACK is received, wait for next packet to be ready;
- Receiver
  - If packet is received without error, sends ACK;
  - If packet is received with error, sends NAK;

rdt 2.0: FSM (Finite-state machine)

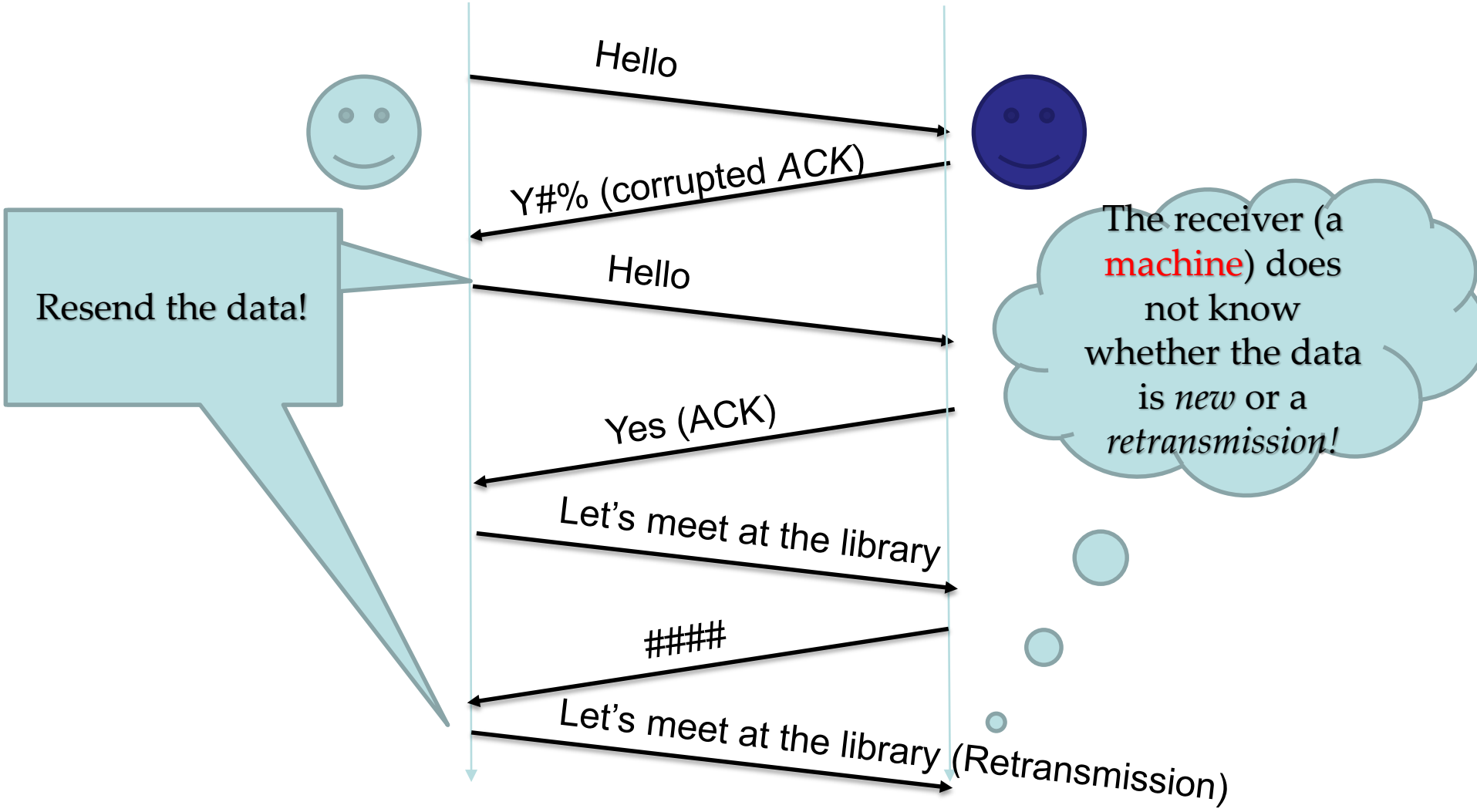


rdt 2.0: A flaw-(corrupted ACK/NAK)



Again we can detect bit errors in ACK/NAK!
But we only know something is corrupted but do not know it is an ACK or NAK exactly;

rdt 2.0: A flaw-corrupted ACK/NAK



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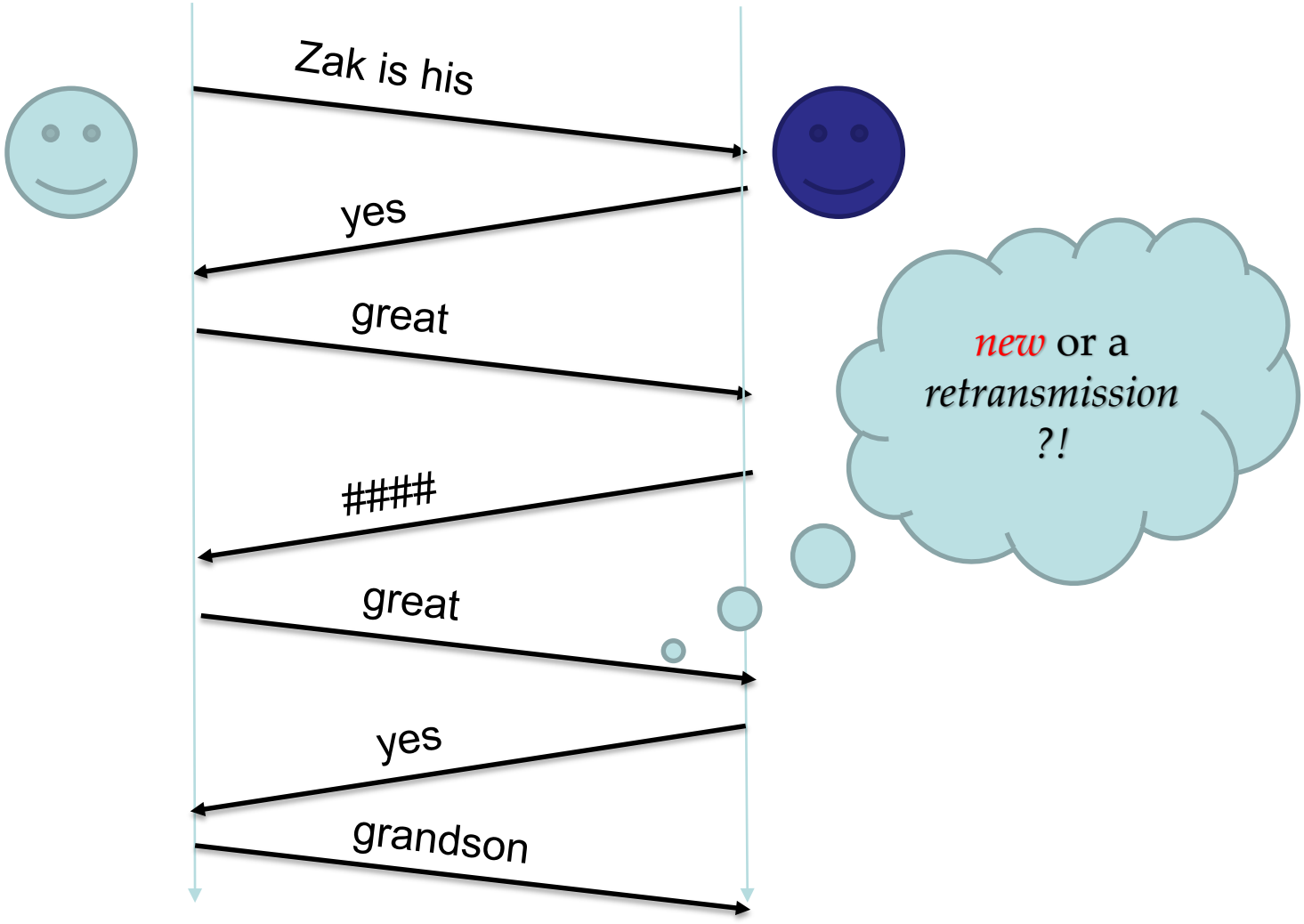

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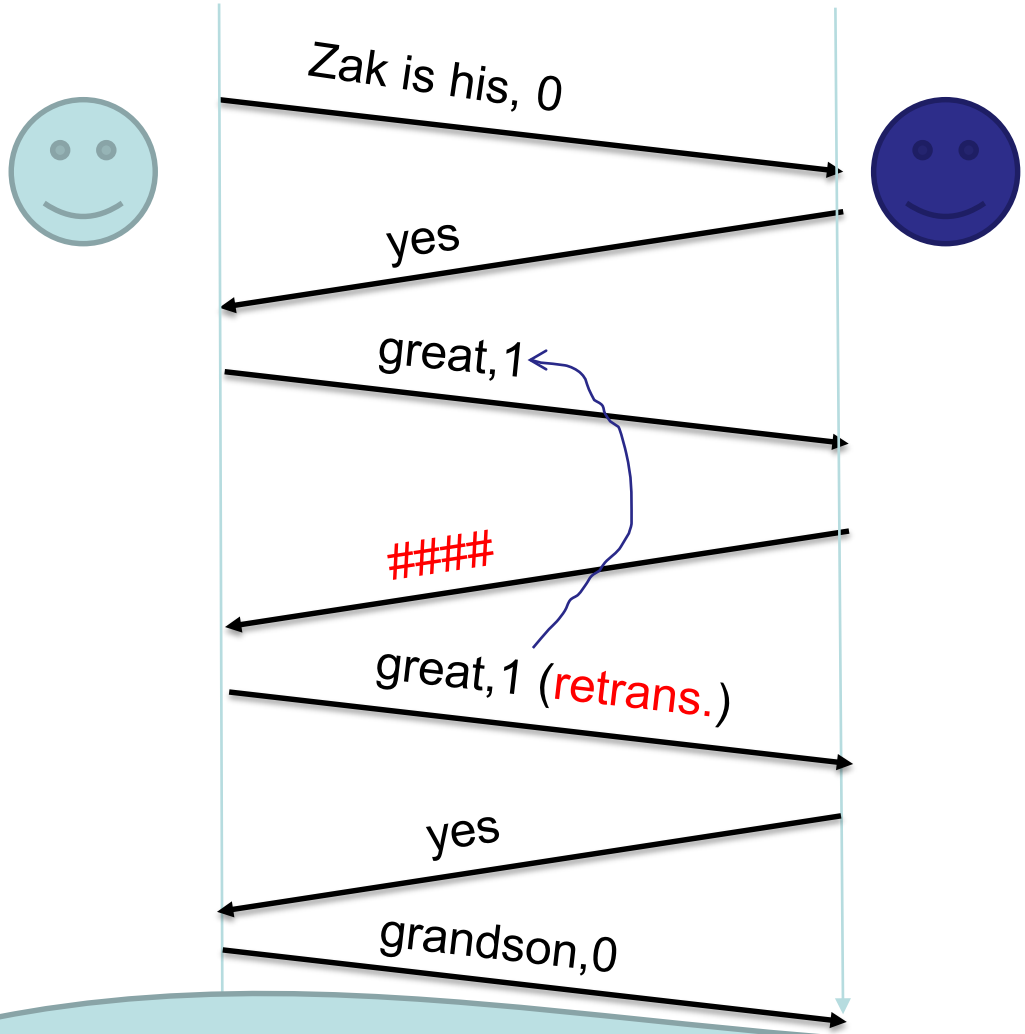
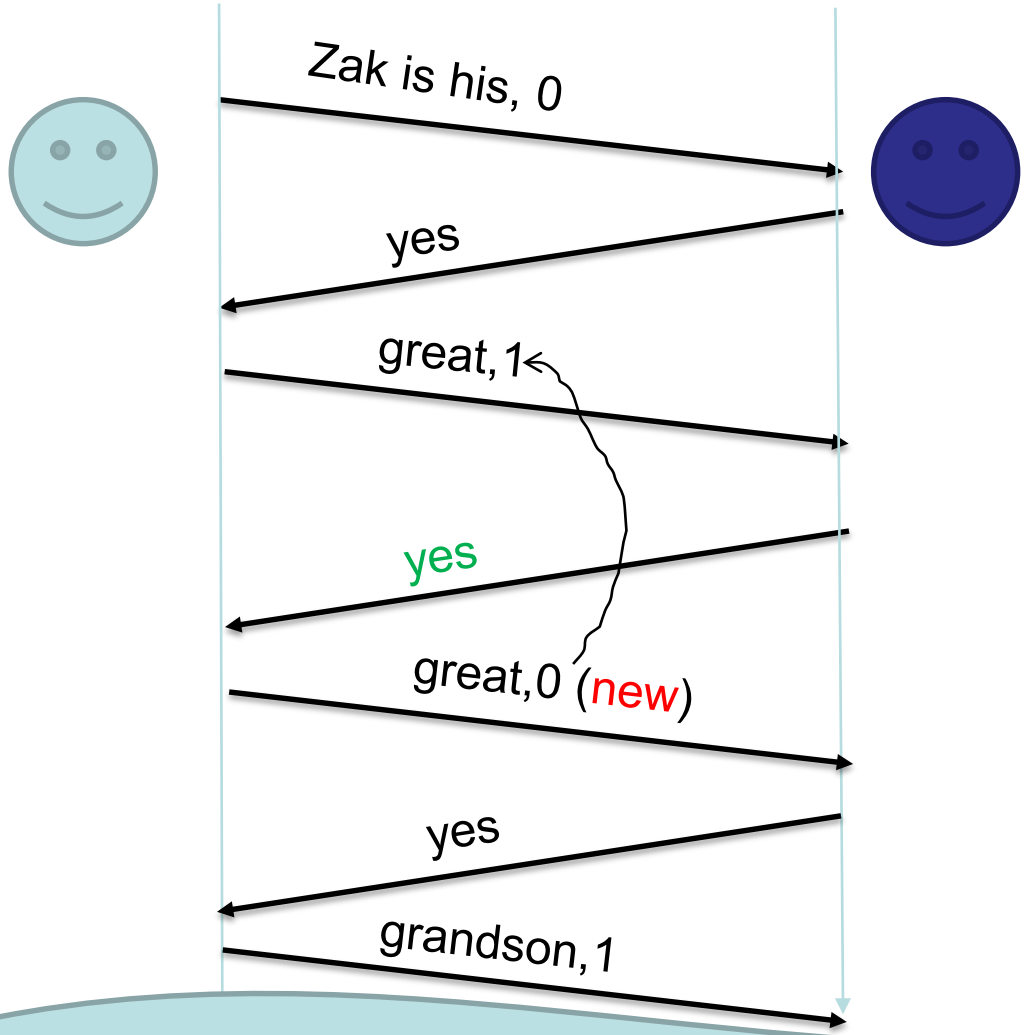


Zak is his great grandson; the receiver will not be confused!

rdt 2.0: A flaw-corrupted ACK/NACK

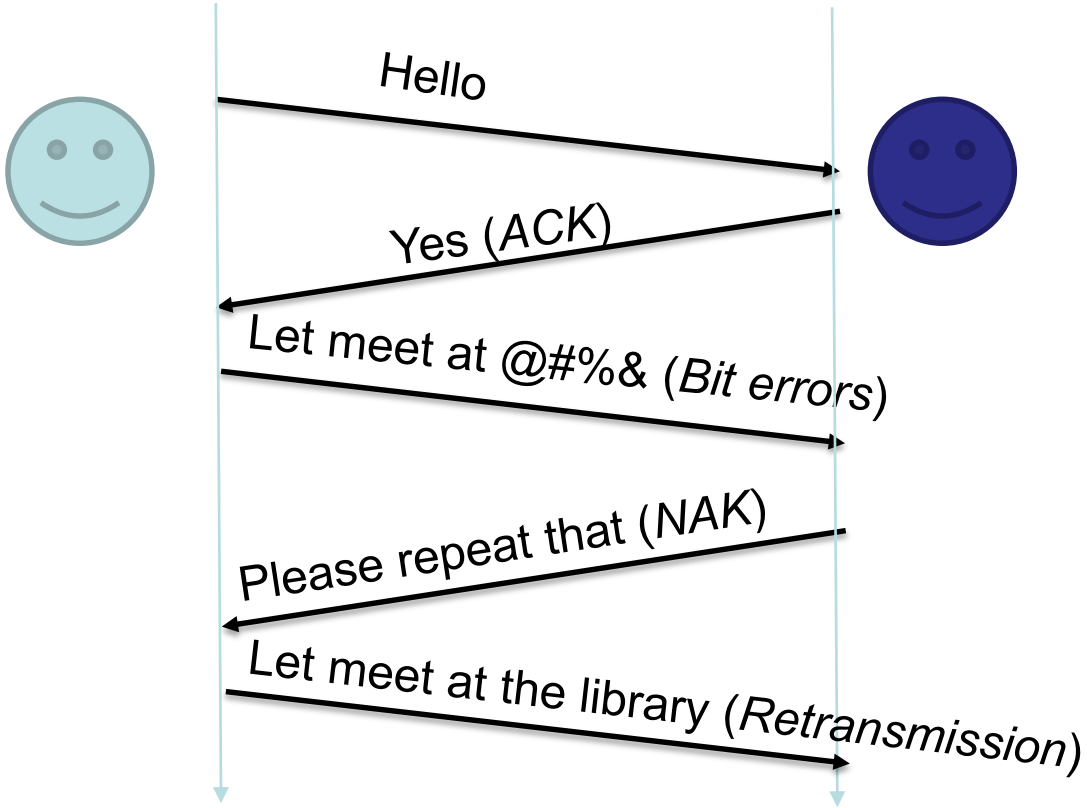


rdt 2.1: A flaw: corrupted ACK/NACK

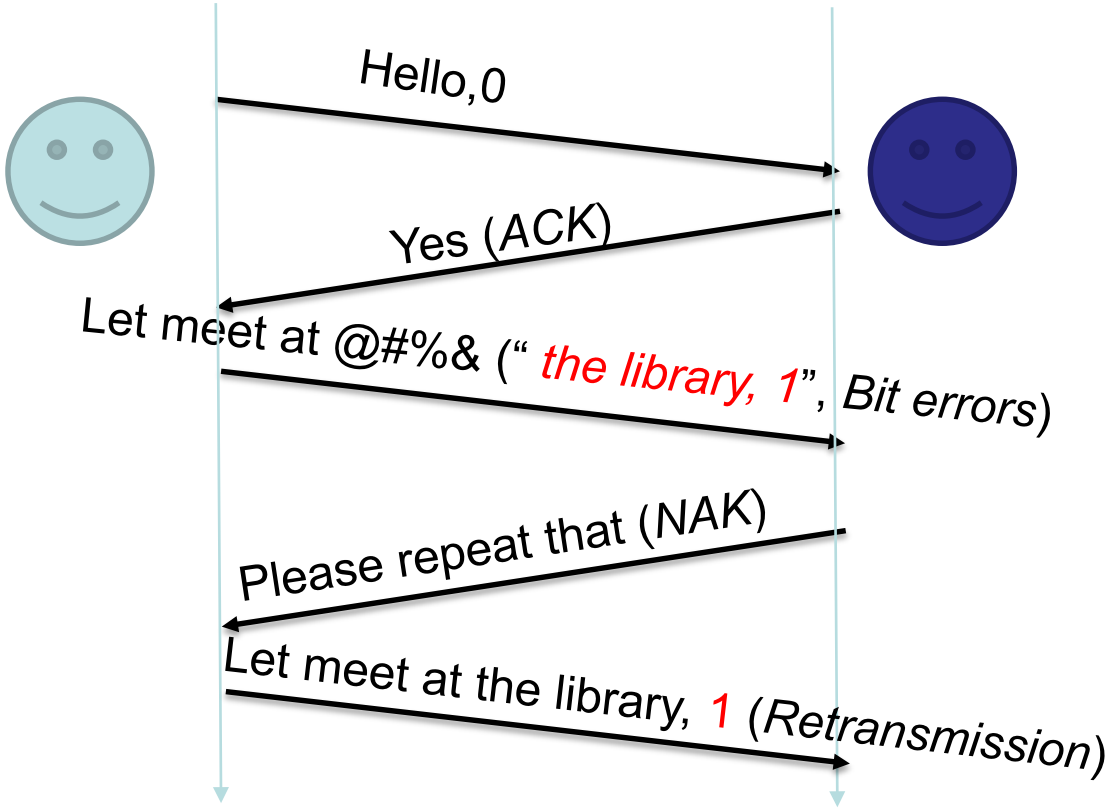
- Data packet with a new field (1 bit sequence number);
  - retransmission (same as previous)
  - new(different from previous)

Two cases: corrupted data (ACK/NAK); corrupted acknowledgements (retrans. + 1 bit seq #).
Assumption: no packet loss;

rdt 2.2: a NAK-free protocol



rdt 2.2: a NAK-free protocol



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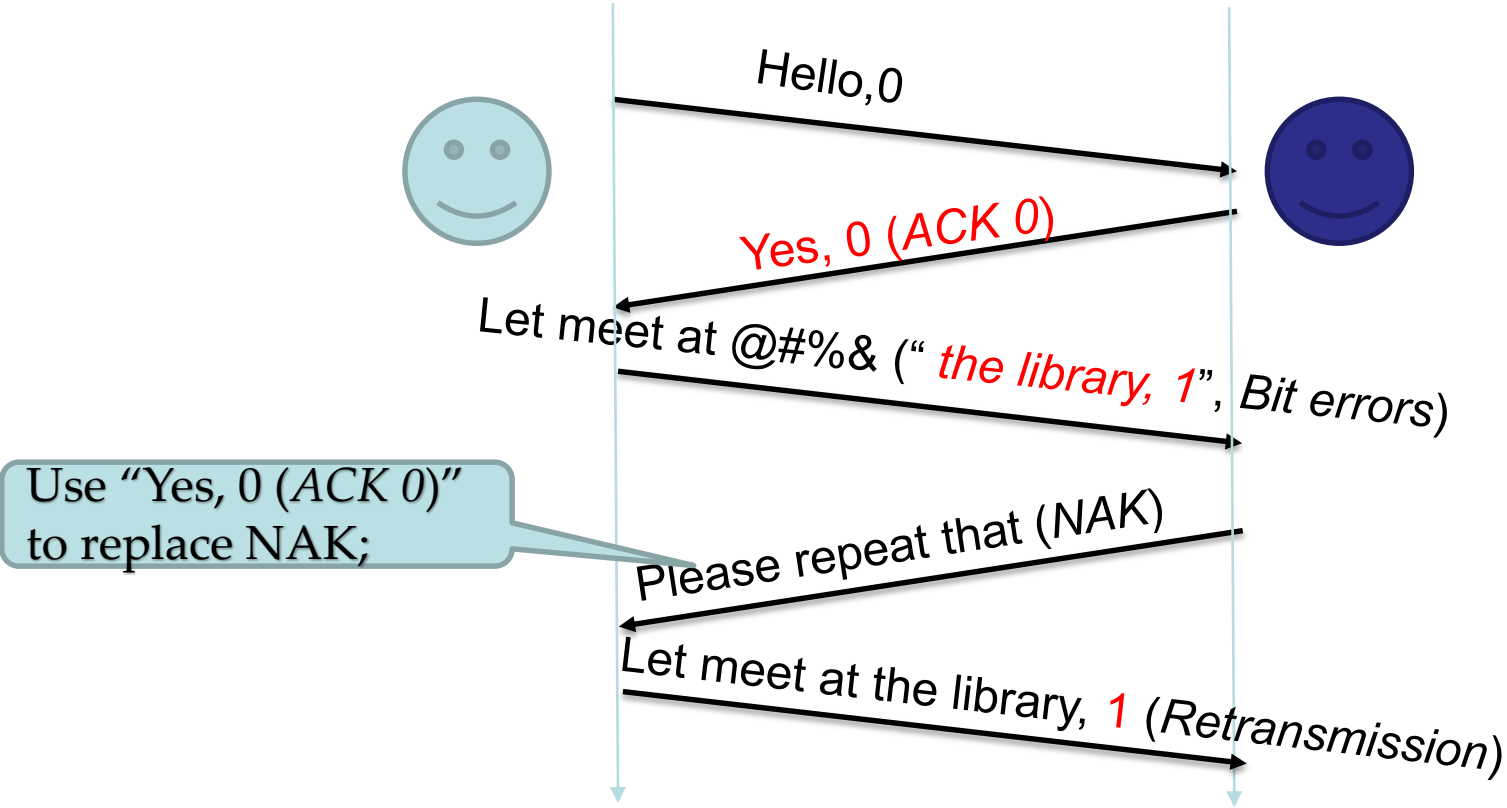
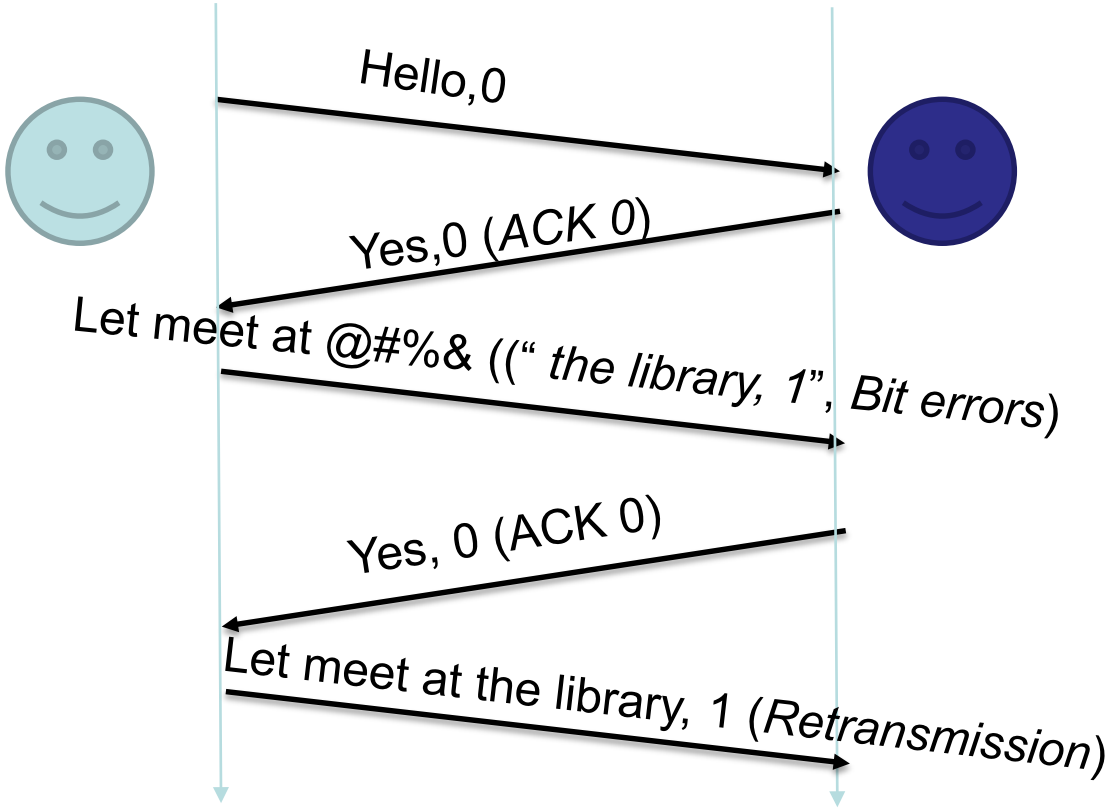


rdt 2.2: A NAK-free protocol



We can use duplicate ACKs to replace NAK.

Two cases: corrupted data (ACK/NAK); corrupted acknowledgements (retrans. + 1 bit seq#).
NAK-free: use duplicate ACKs;
Assumption: no packet loss;

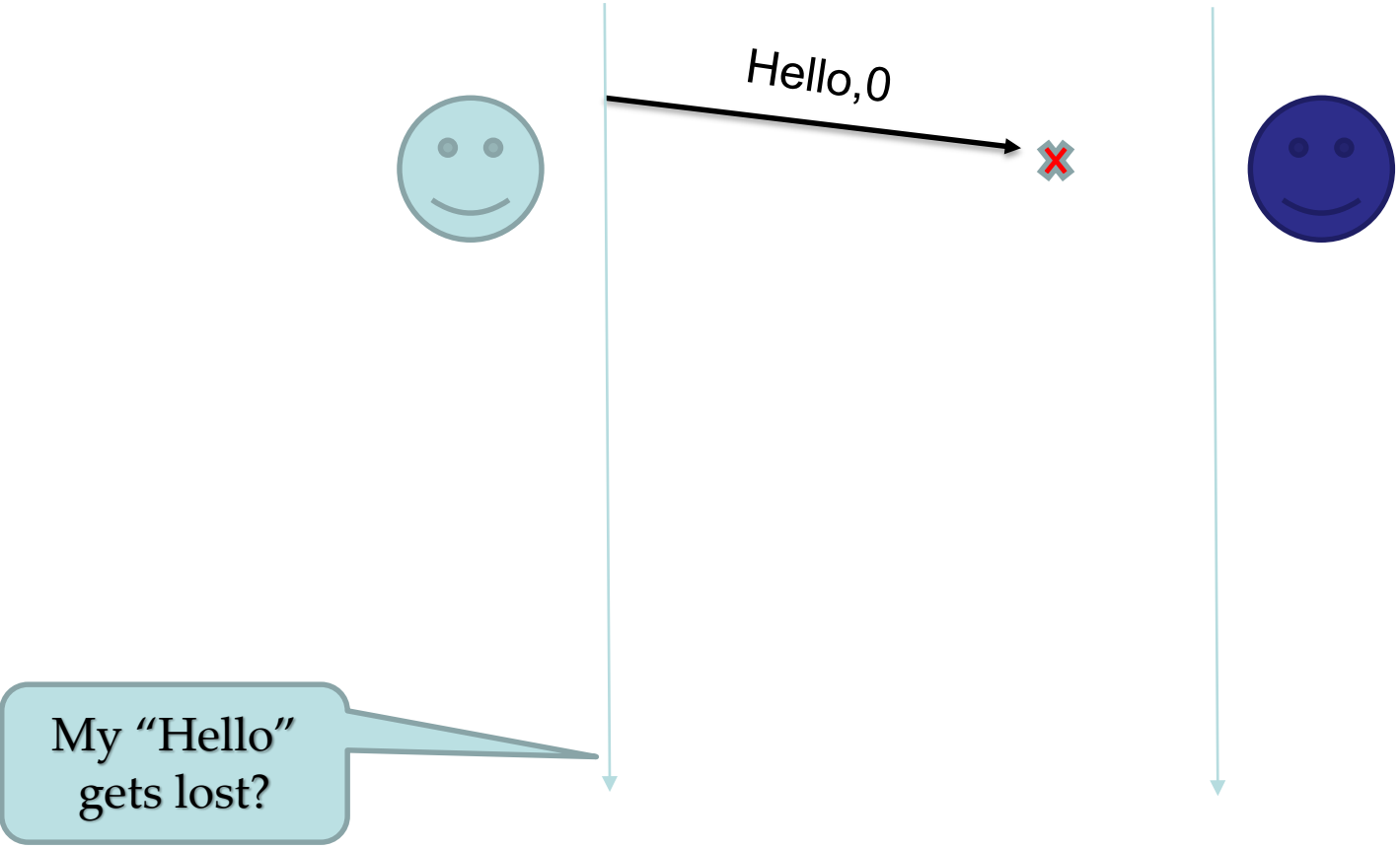
rdt 3.0: Reliable data transfer over a *lossy* channel with bit errors

- Packets can get lost.
  - How to detect packet loss?
  - What to do when packet loss occurs?

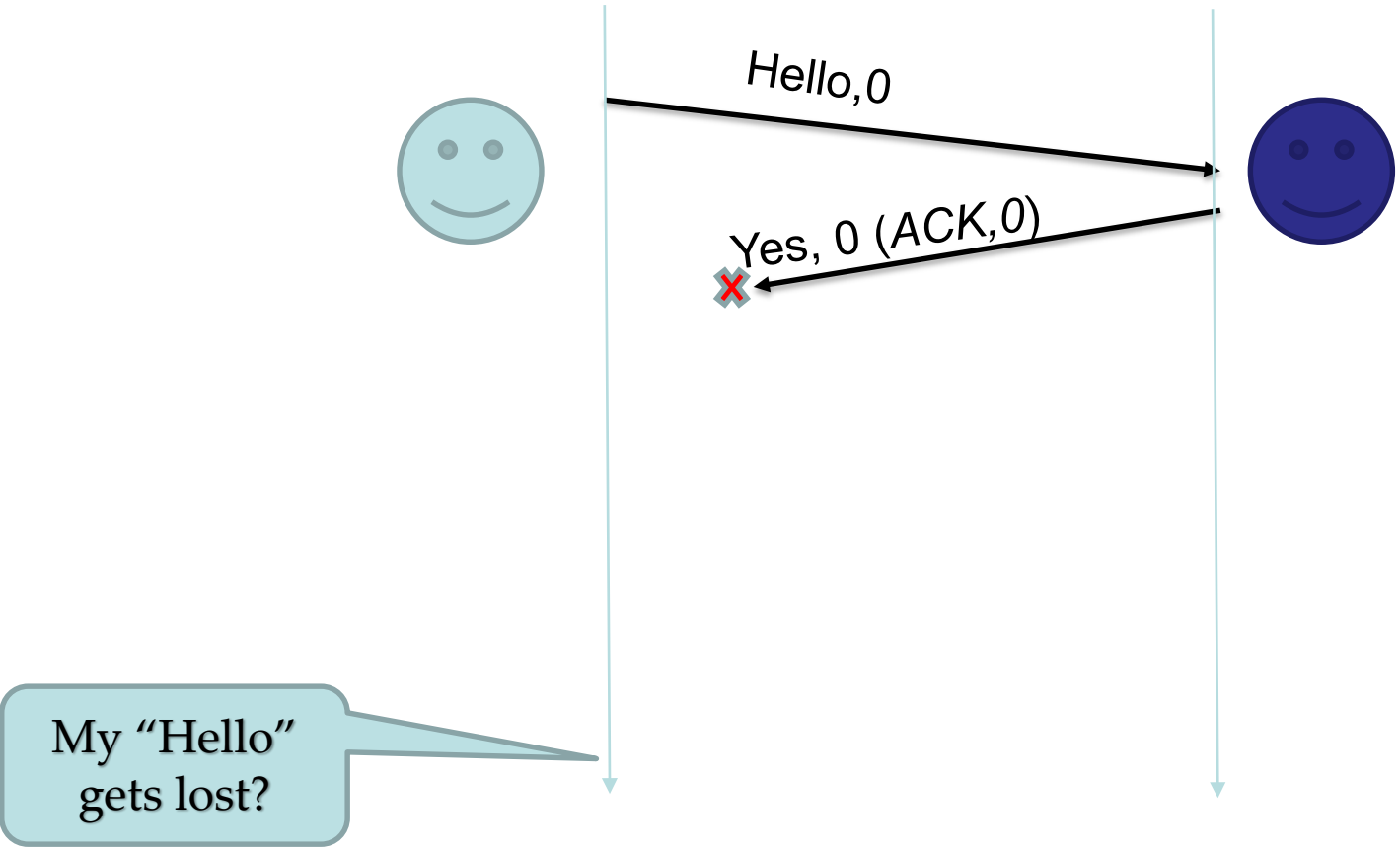
What to do when packet loss occurs

- Retransmission (A panacea!)
  - Data packet loss
  - ACK loss
  - Data/ACK delayed
  - (bit errors)

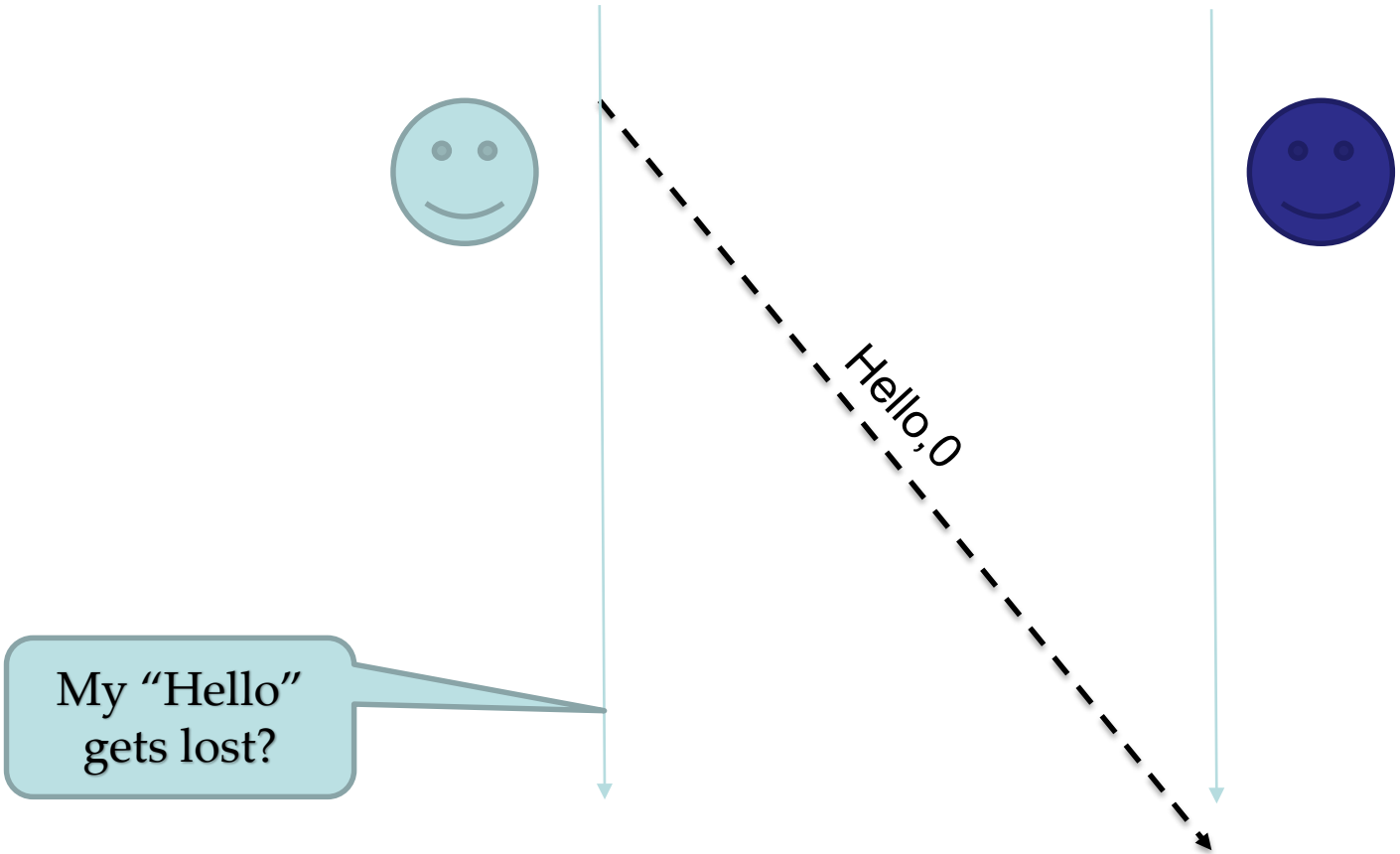
Data loss



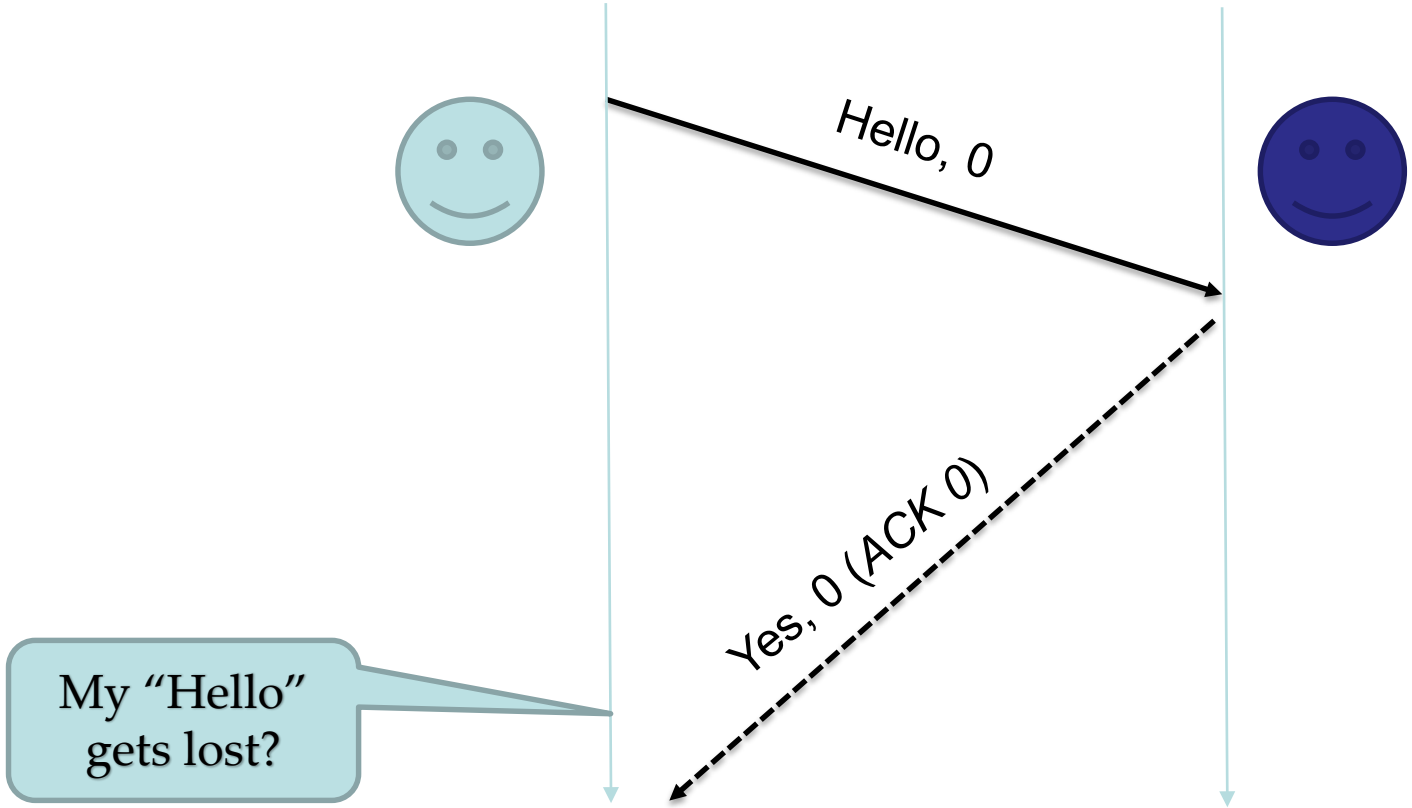
ACK loss



Data Delayed



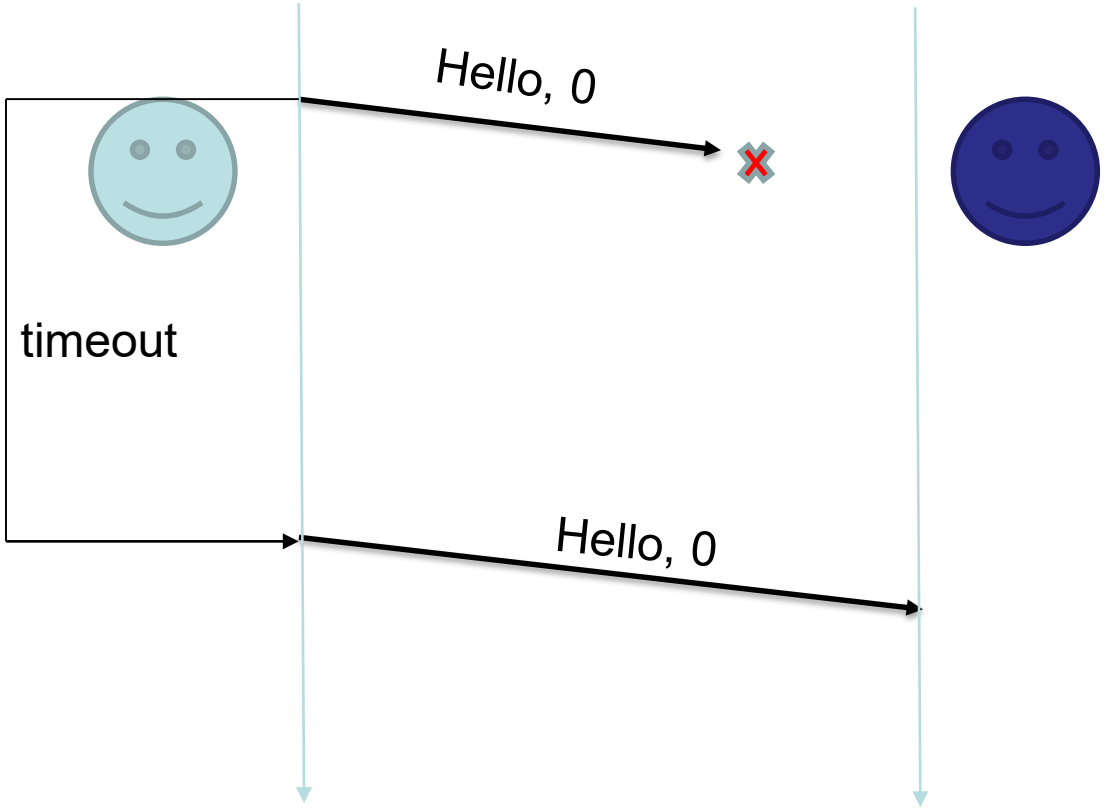
ACK Delayed



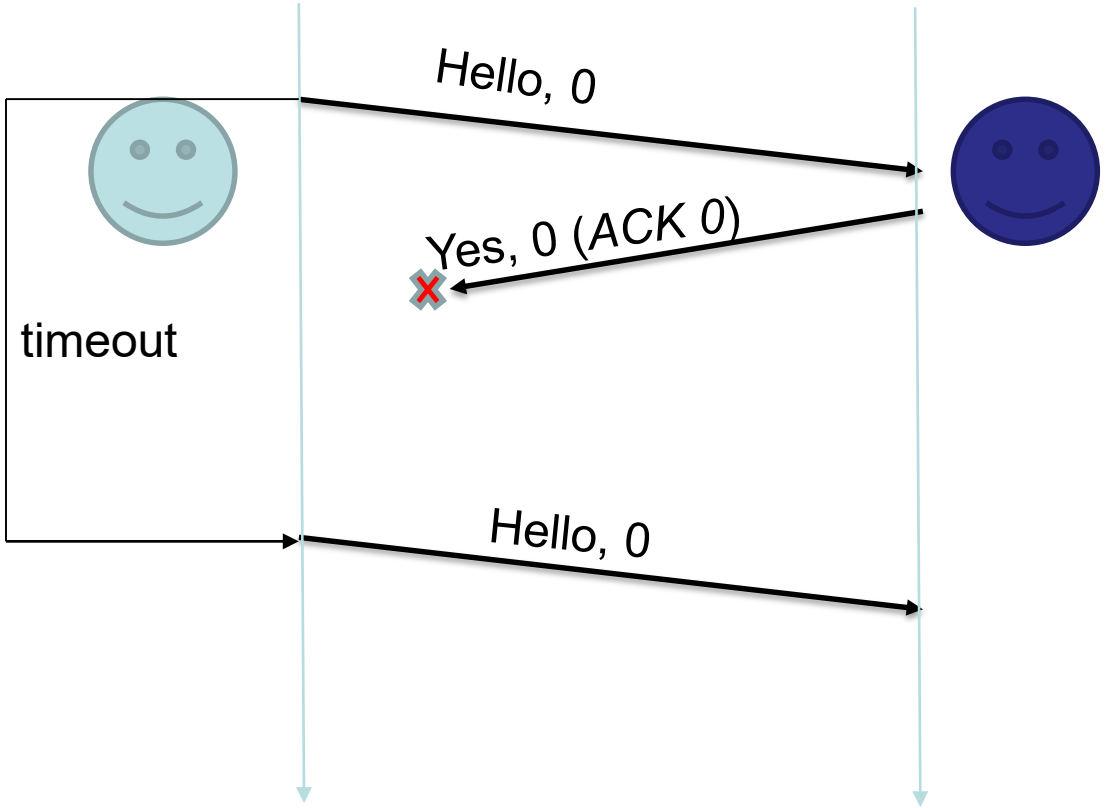
How to detect packet loss

- Wait, wait and wait!
- How long should the sender wait?
  - Too short → duplicate data packets
  - Too long → decreased performance
  - The solution: to judiciously choose a time value

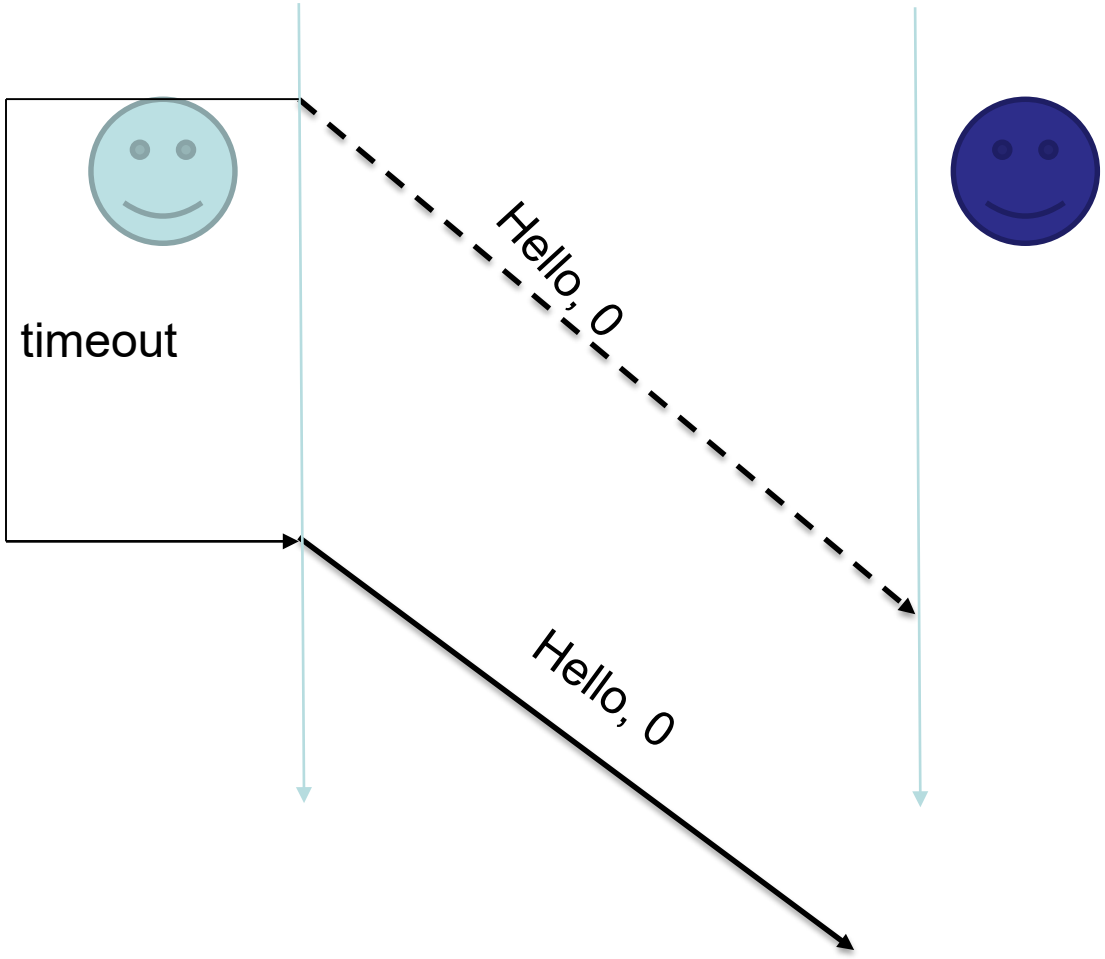
Data loss



ACK loss



Delayed



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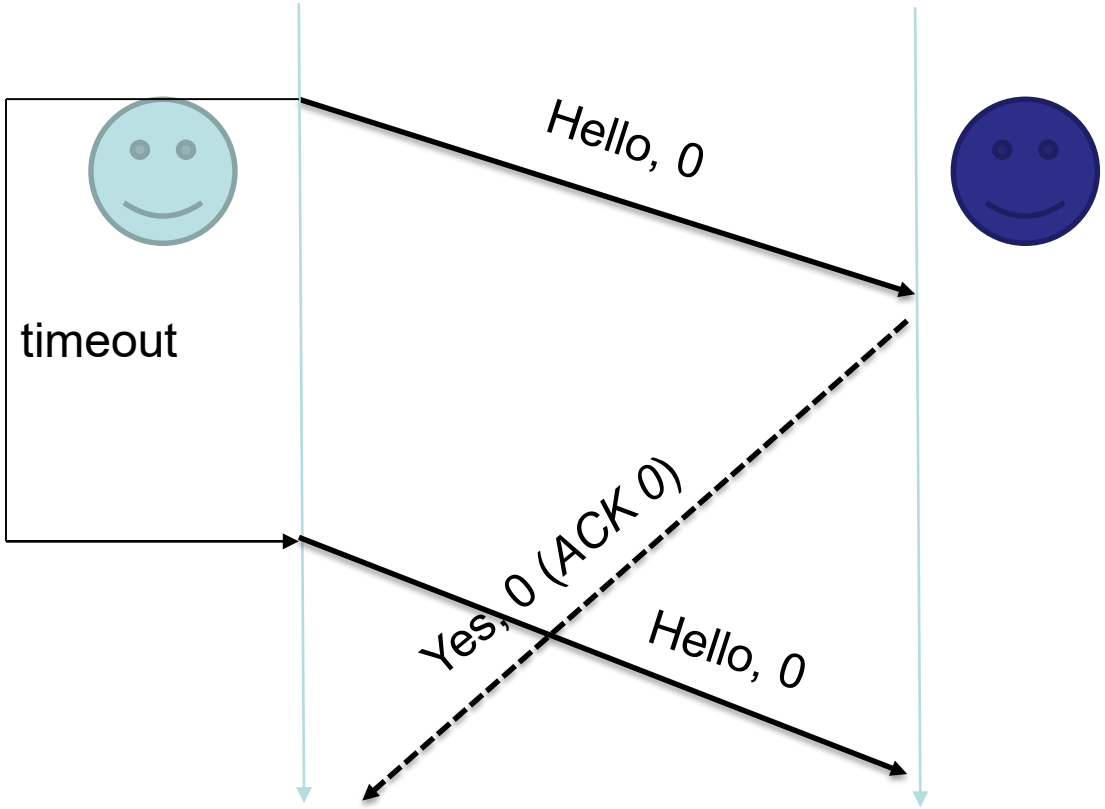
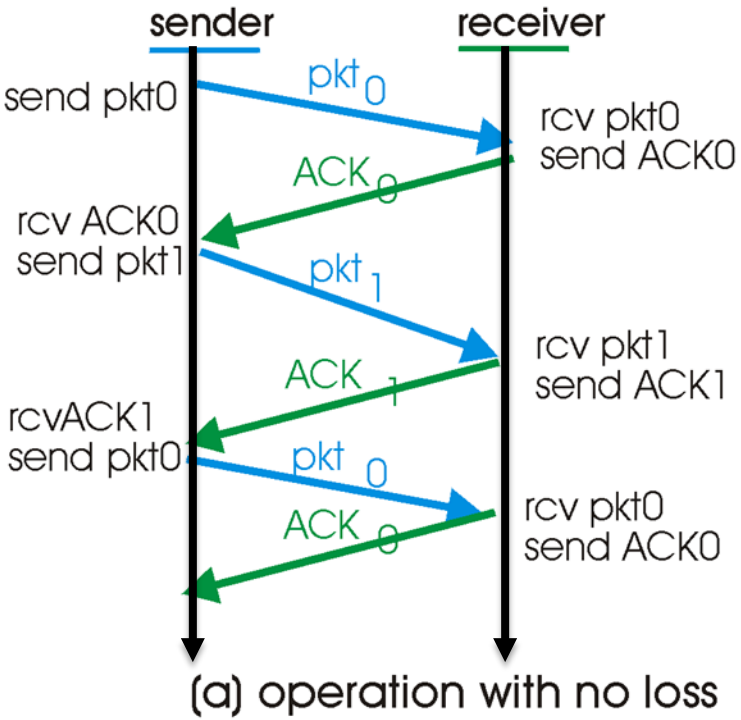
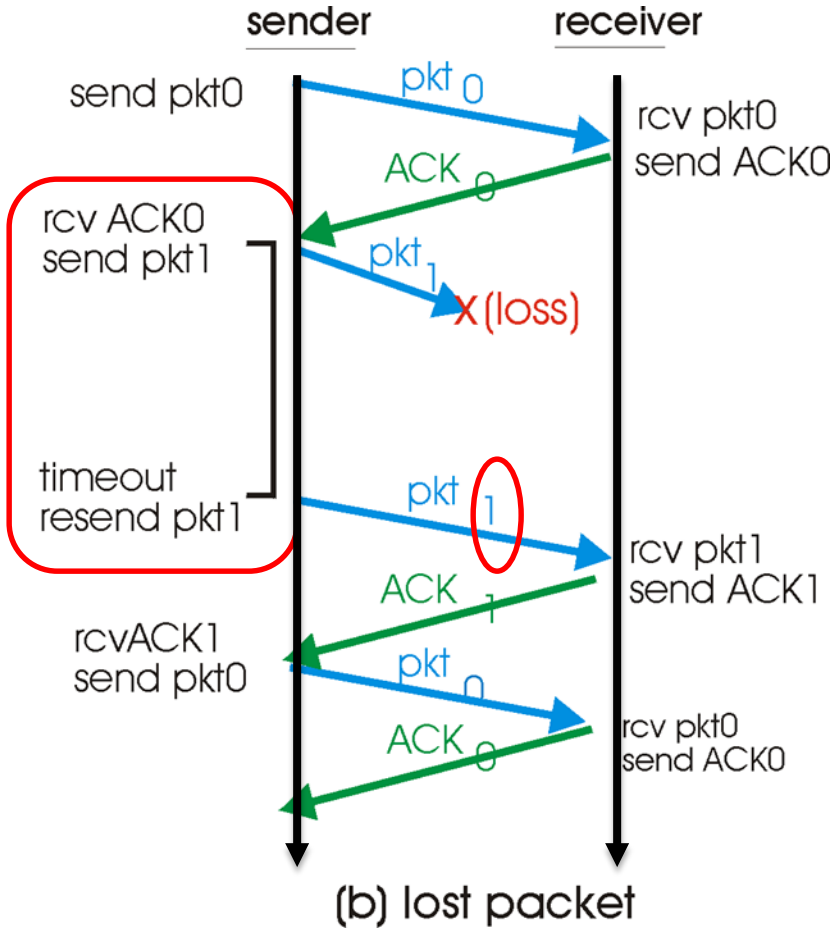


Use timeout to detect packet loss (*choose a time value wisely*);
Use retransmission to recover from packet loss (*duplicate data packets – use seq#*);

rdt 3.0: An illustration



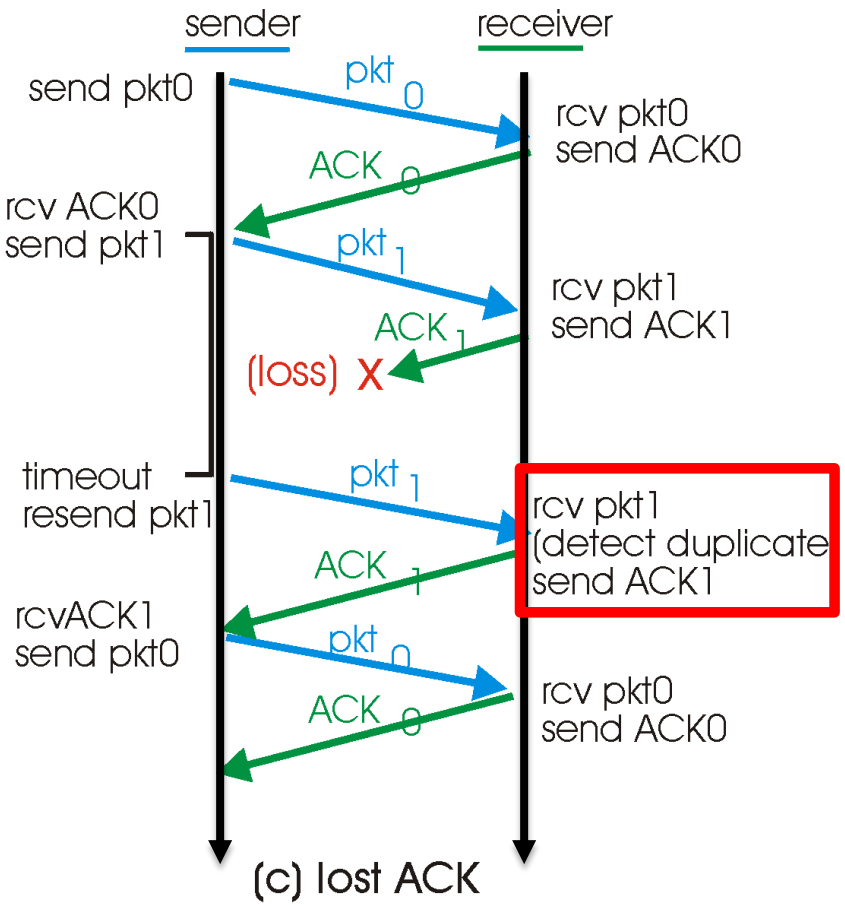
(a) operation with no loss



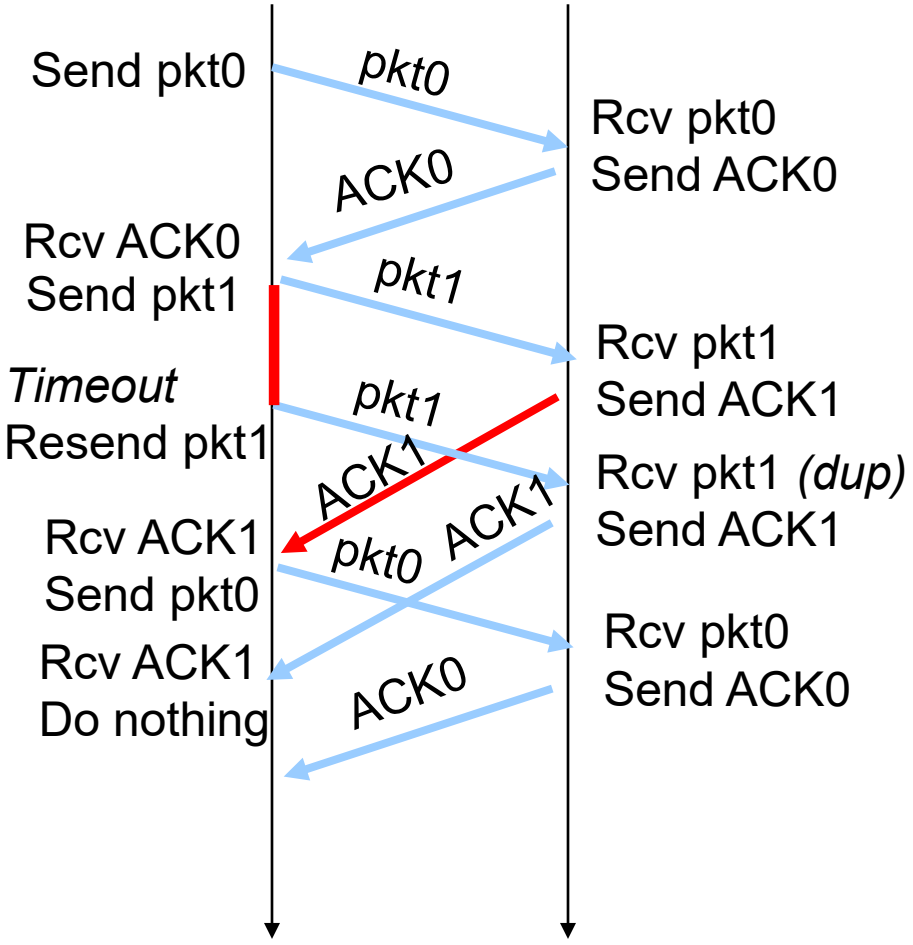
(b) lost packet

Also known as alternating-bit protocol

rdt 3.0: An illustration



(c) lost ACK

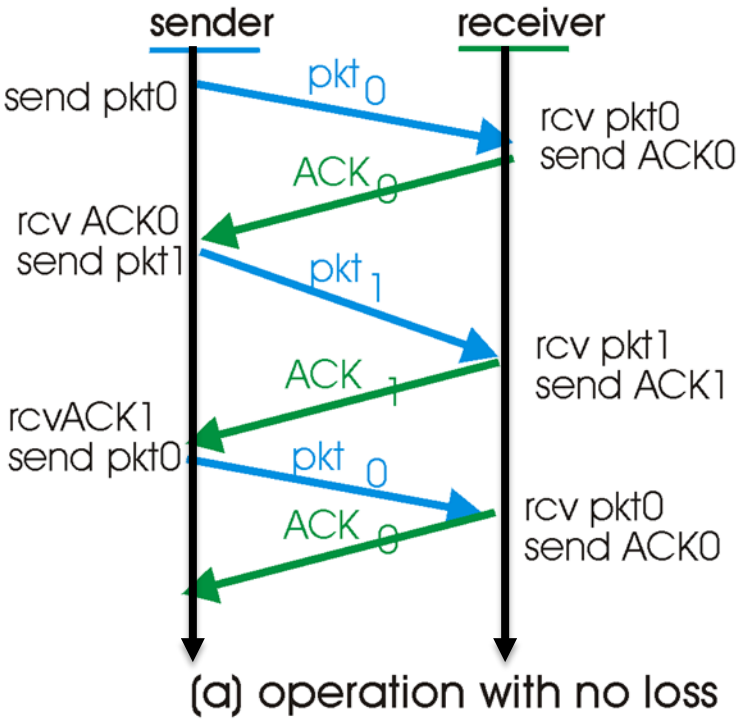


(d) Premature timeout

Mechanisms we have learned

- Checksum (to detect bit error)
- ACK/NAK (to notify the sender)
- Sequence number (duplicate data packets)
- Duplicate ACKs (NAK-free)
- Timer (data packet loss)
- Retransmission (a panacea)

rdt 3.0: An illustration - review



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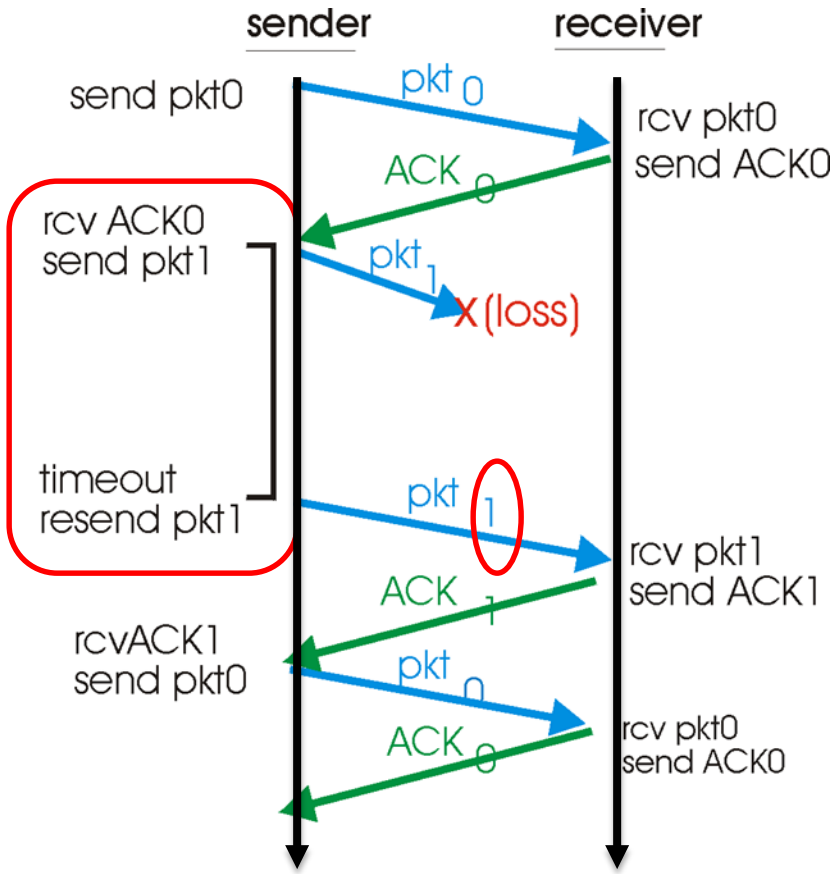

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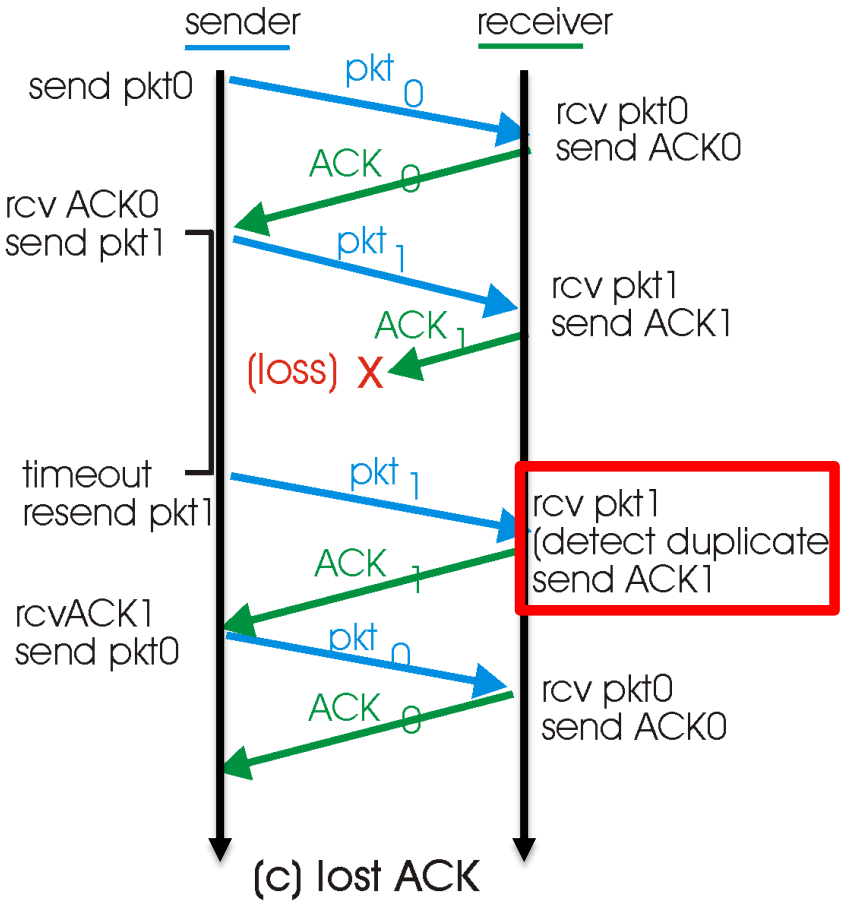
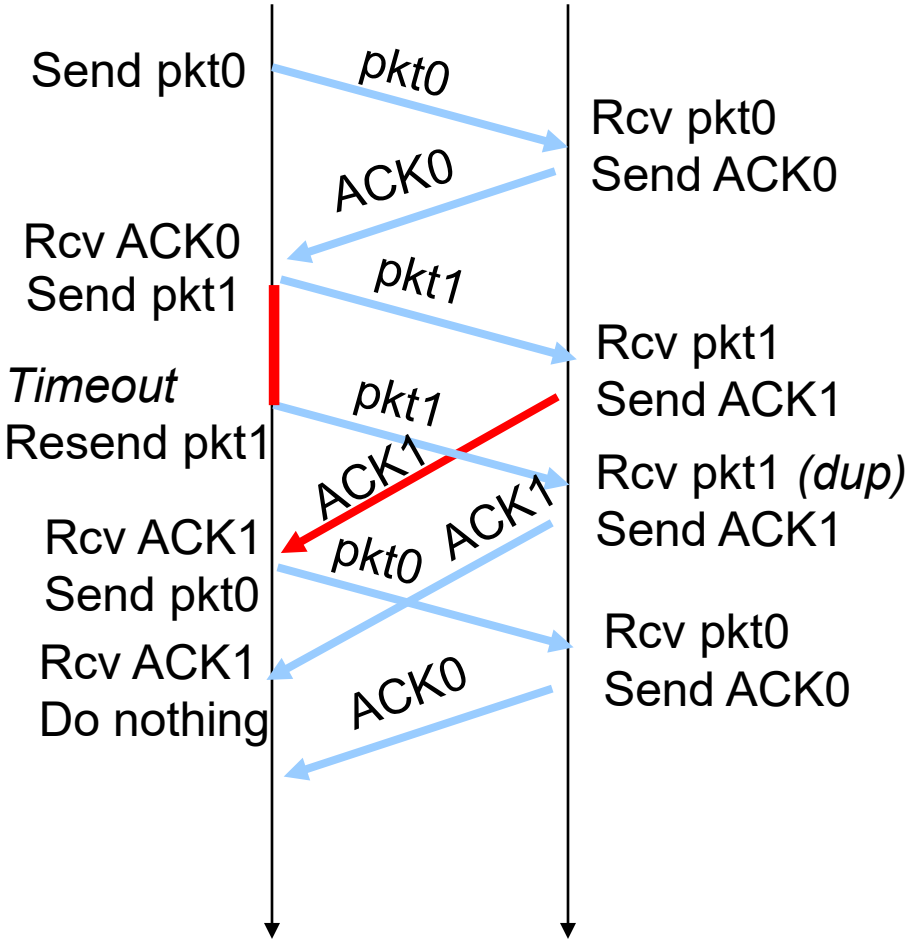


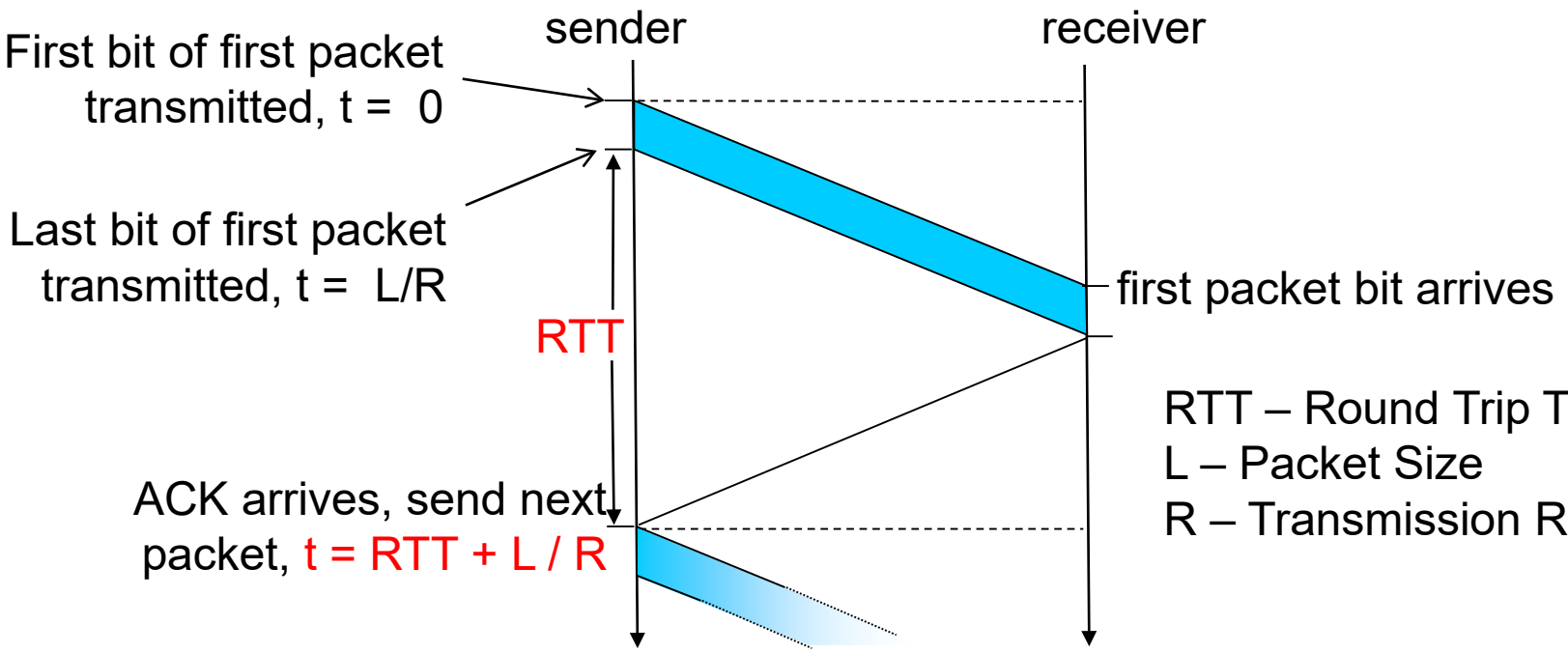
(c) lost ACK



(d) Premature timeout

The performance problem of rdt 3.0

- It is essentially a stop-and-wait protocol.



The Sender Utilisation: $U_{sender} = \frac{L / R}{RTT + L / R}$

The performance problem of rdt 3.0

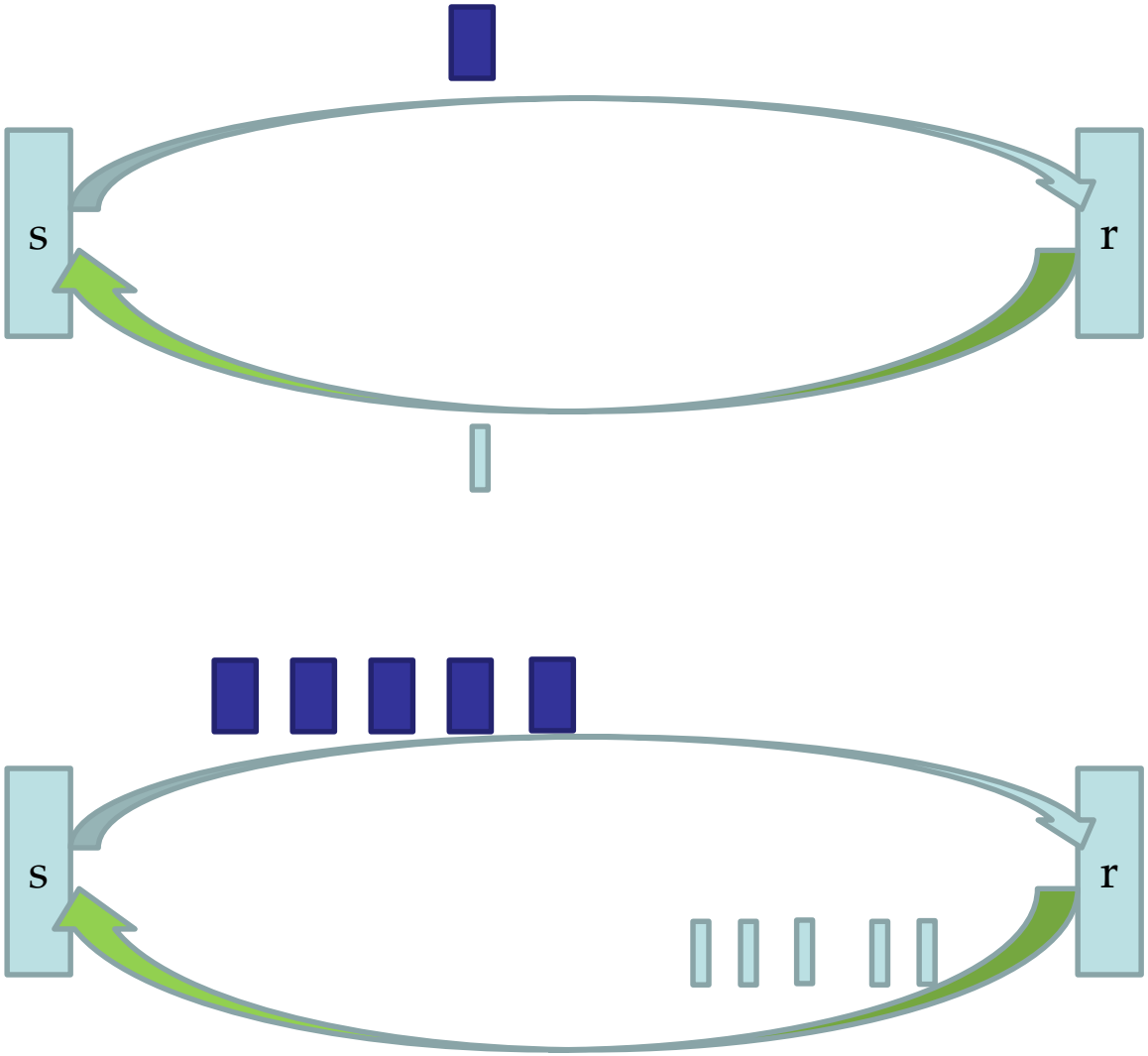
- An example: 1 Gbps link, 15 ms end-to-end delay (RTT, 30ms) , 1KB packet:

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

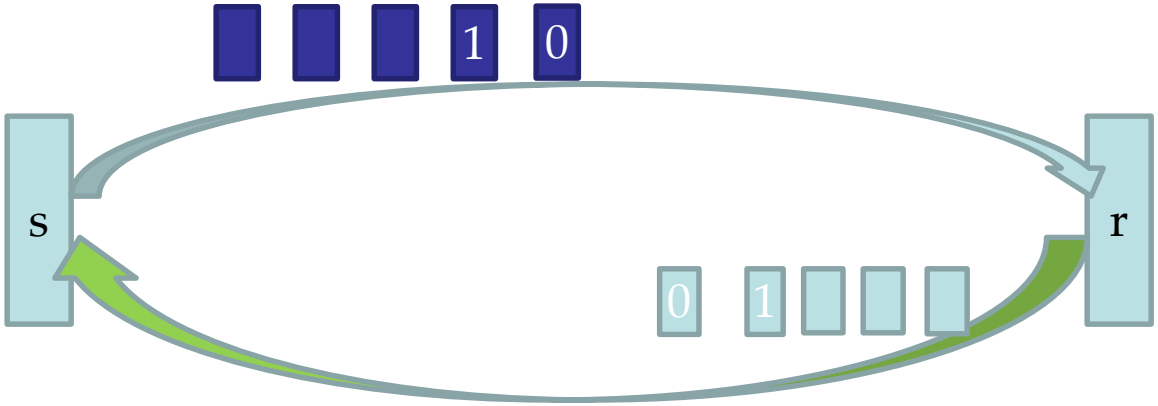
$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

The sender is waiting in most time!
Very low utilisation! Not mention we have neglected lower layer processing times and queuing delays.

Pipelining



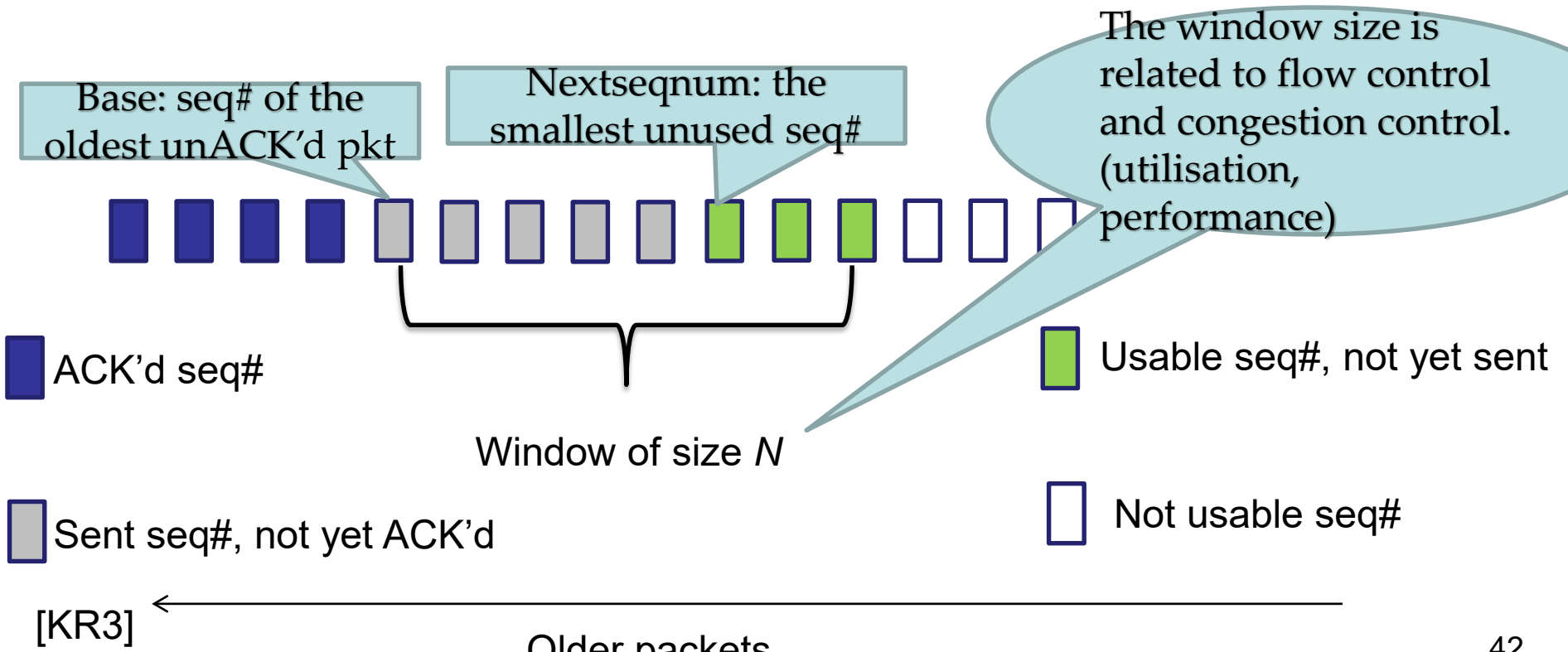
Pipelining



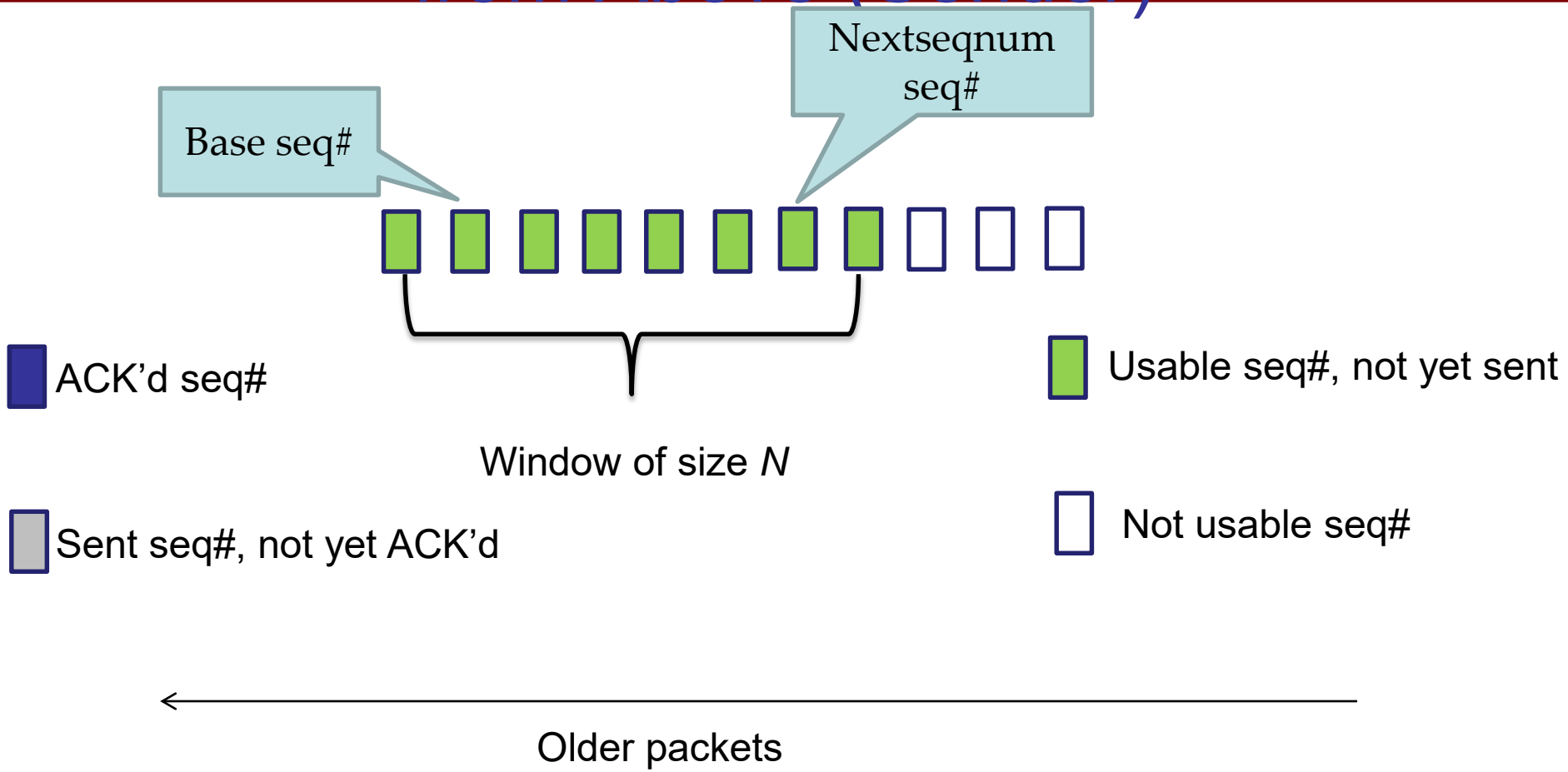
- Using Seq# [0,1] is insufficient;
  - In practice, seq # field occupies k bits, ranging from 0 to 2^k -1. (TCP uses 32 bits for seq#.)
- Sender and receiver may have to buffer more than one packet;
- Then we have two basic approaches based on pipelining.

Go-Back-N protocol

- The sender is allowed to send multiple packets without waiting for an acknowledgement, but is constrained to have *no more than N unacknowledged packets* in the pipeline. (Sender cannot send too fast.)

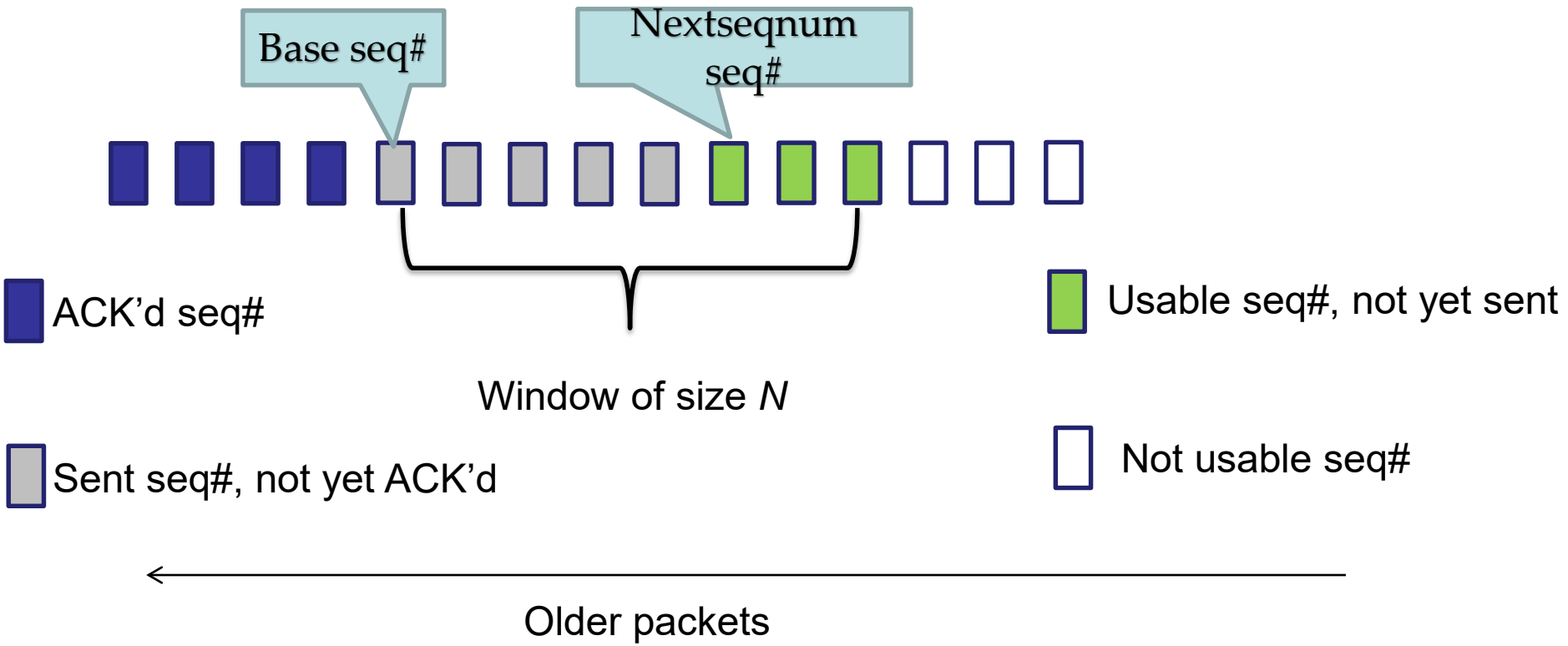


Go-Back-N – Initialisation & Invocation from Above (Sender)



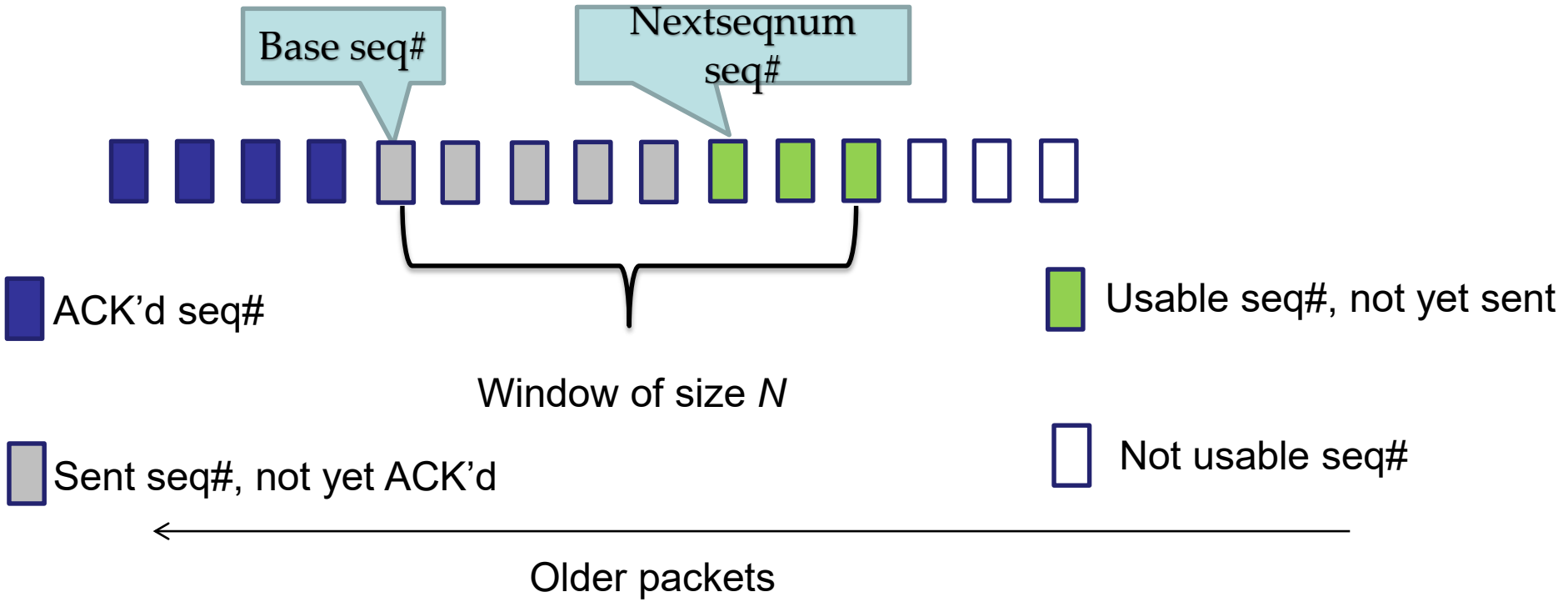
[KR3]

Go-Back-N – Receipt of an ACK (Sender)



[KR3]

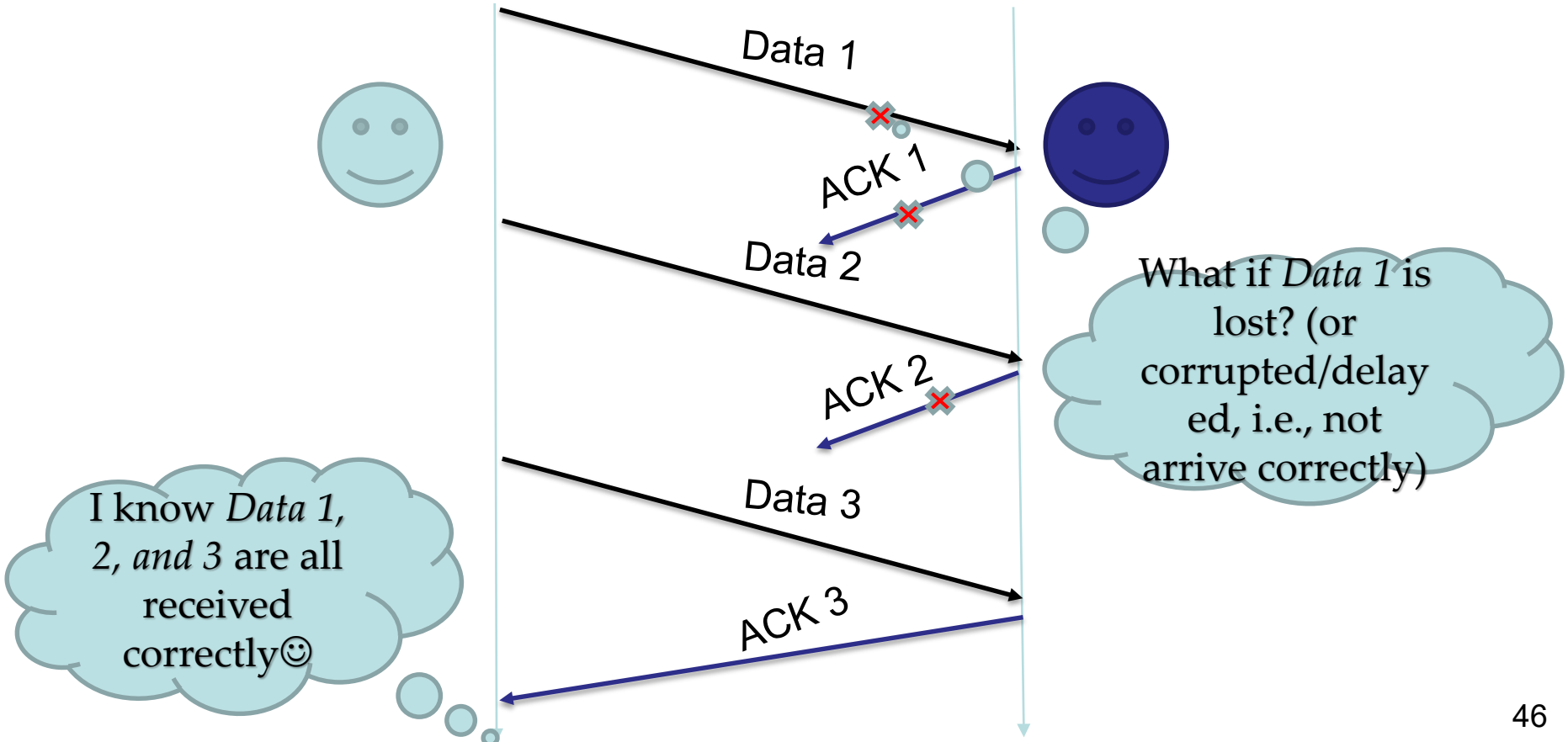
Go-Back-N- Cumulative Acknowledgment (Sender)



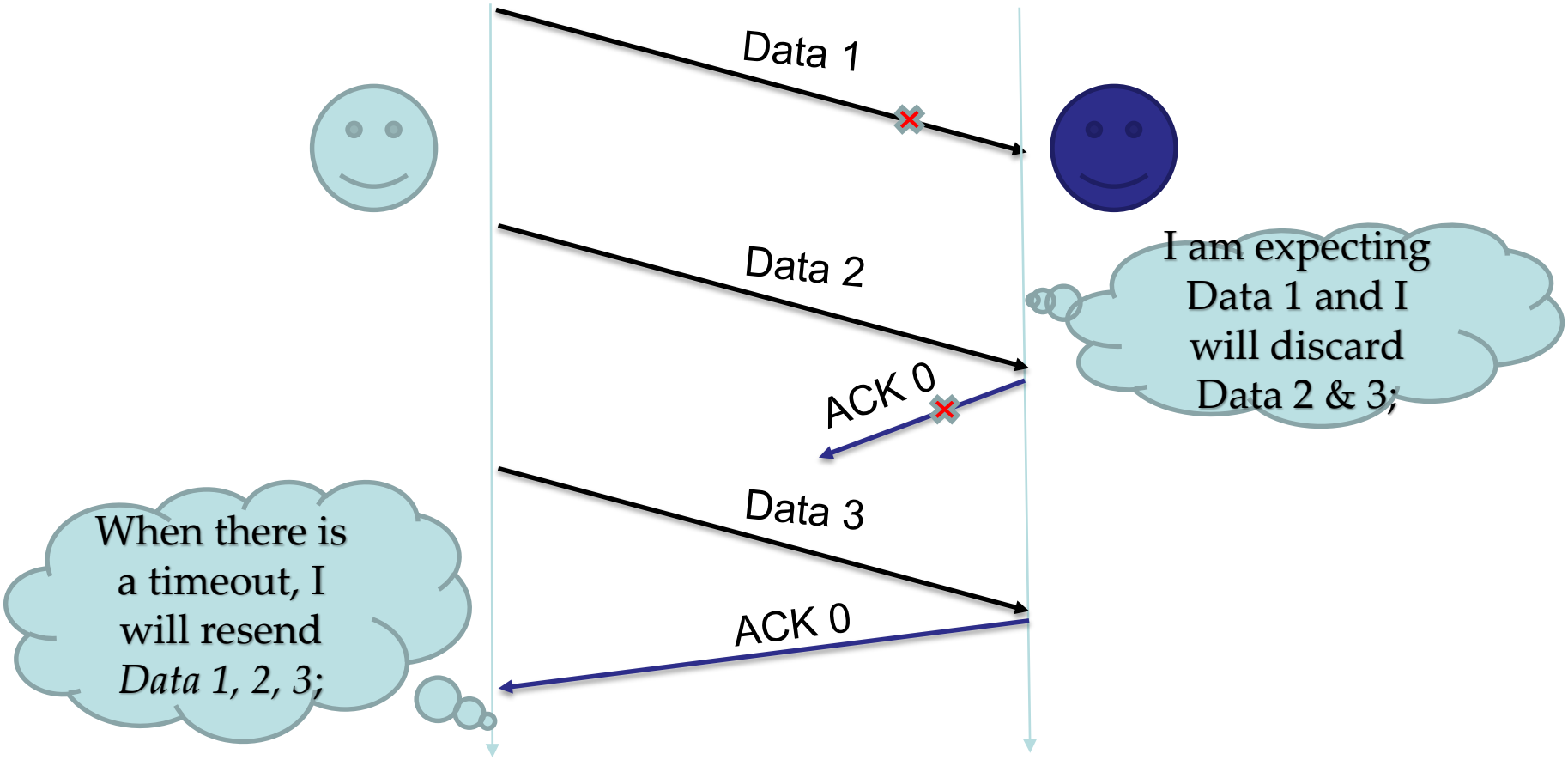
[KR3]

Go-Back-N: Cumulative ACK (Sender)

- An ACK with seq # *n* means that all packets with a seq # up to (and including) *n* have been correctly received at the receiver.



Go-Back-N: Cumulative ACK (Sender)



Receiver: My rule is always expecting the next in-order pkt; if unexpected pkt comes, I always discard them and resend an ACK for the latest correctly received in-order pkt.

Go-Back-N: Cumulative ACK (Sender)

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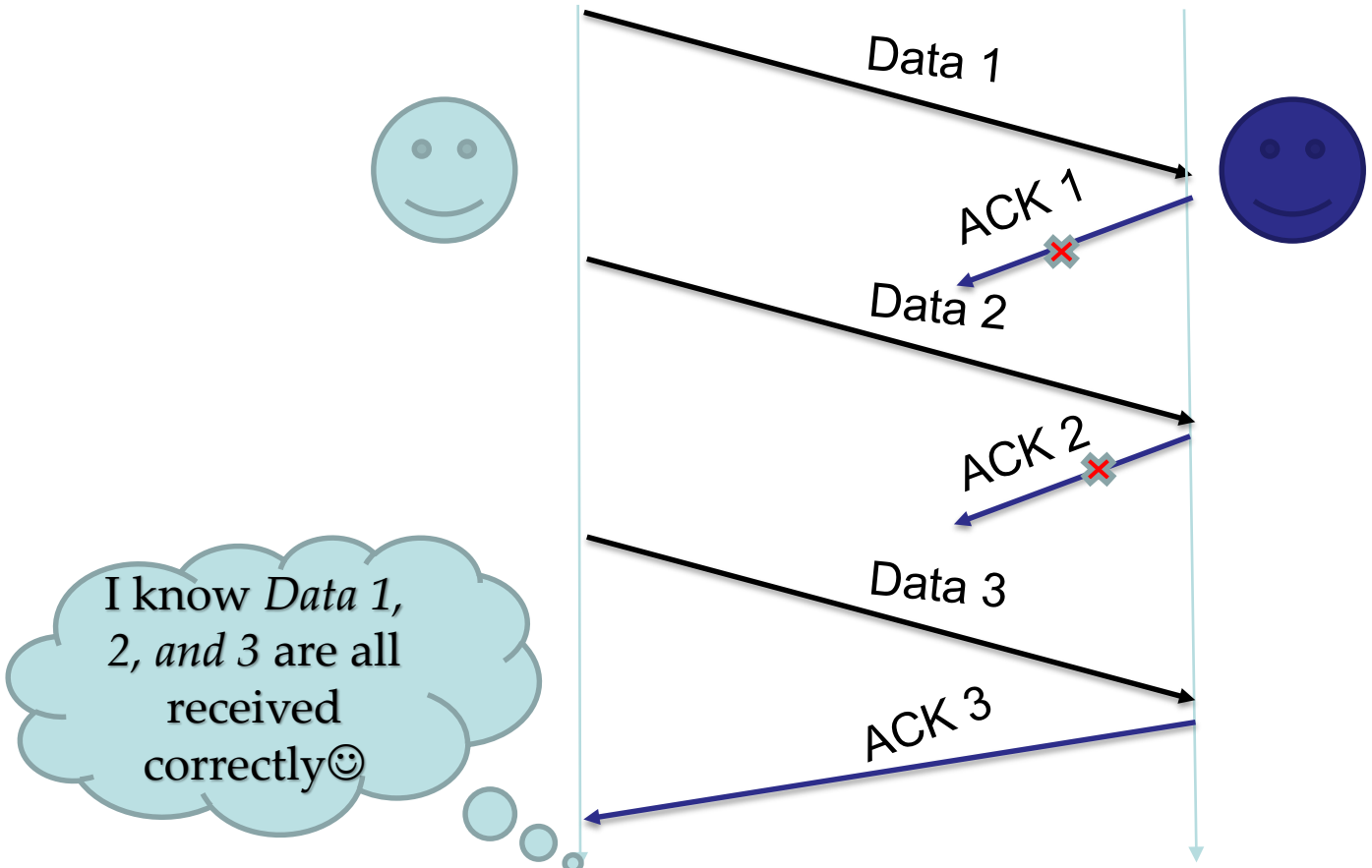
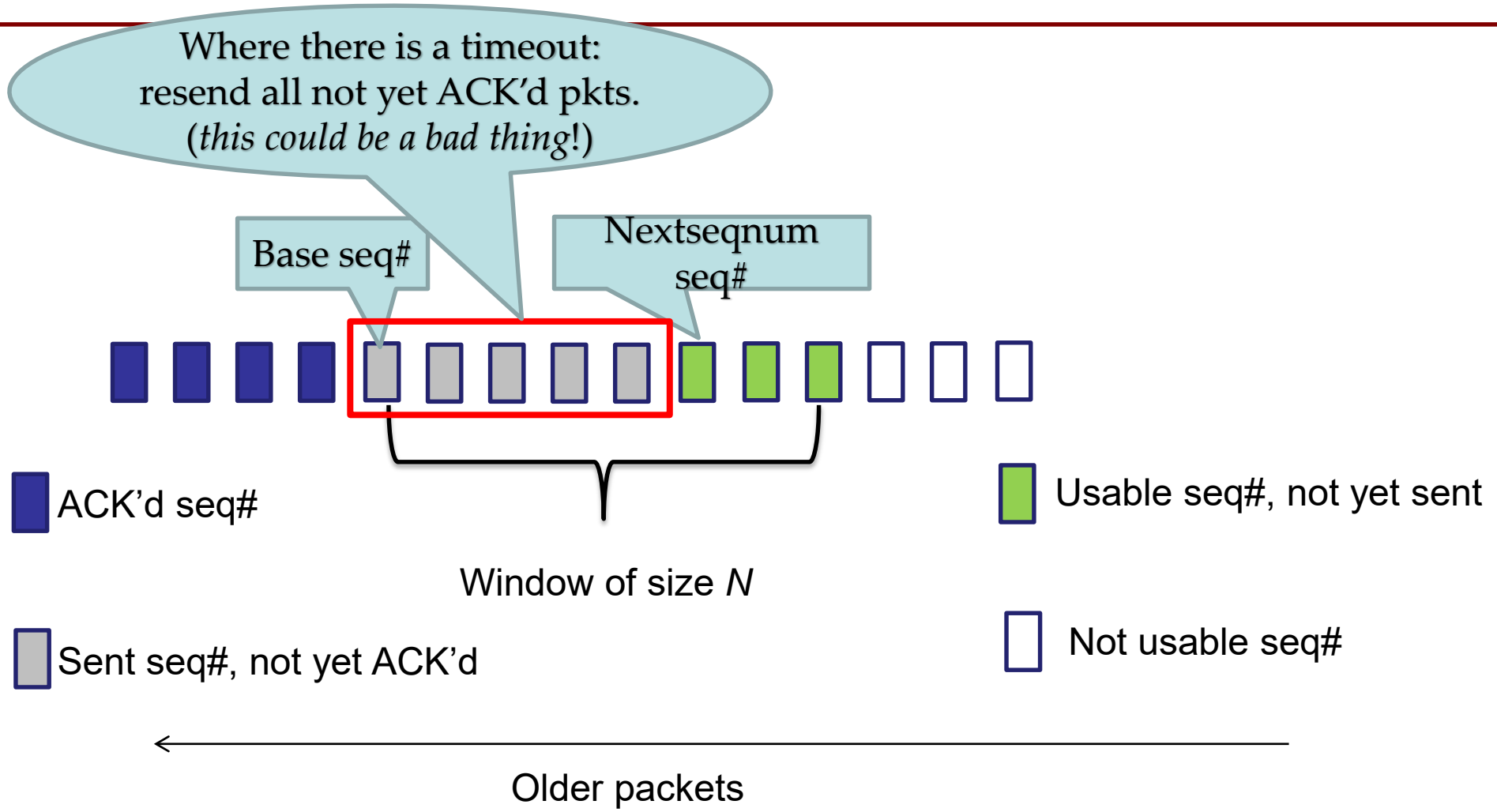


Go-Back-N -timeout



[KR3]

The Receiver Side

Initially, expecting packet 0

When packet 0 arrives uncorrupted,
Deliver data to above;
Send ACK 0;
Now expecting packet 1;

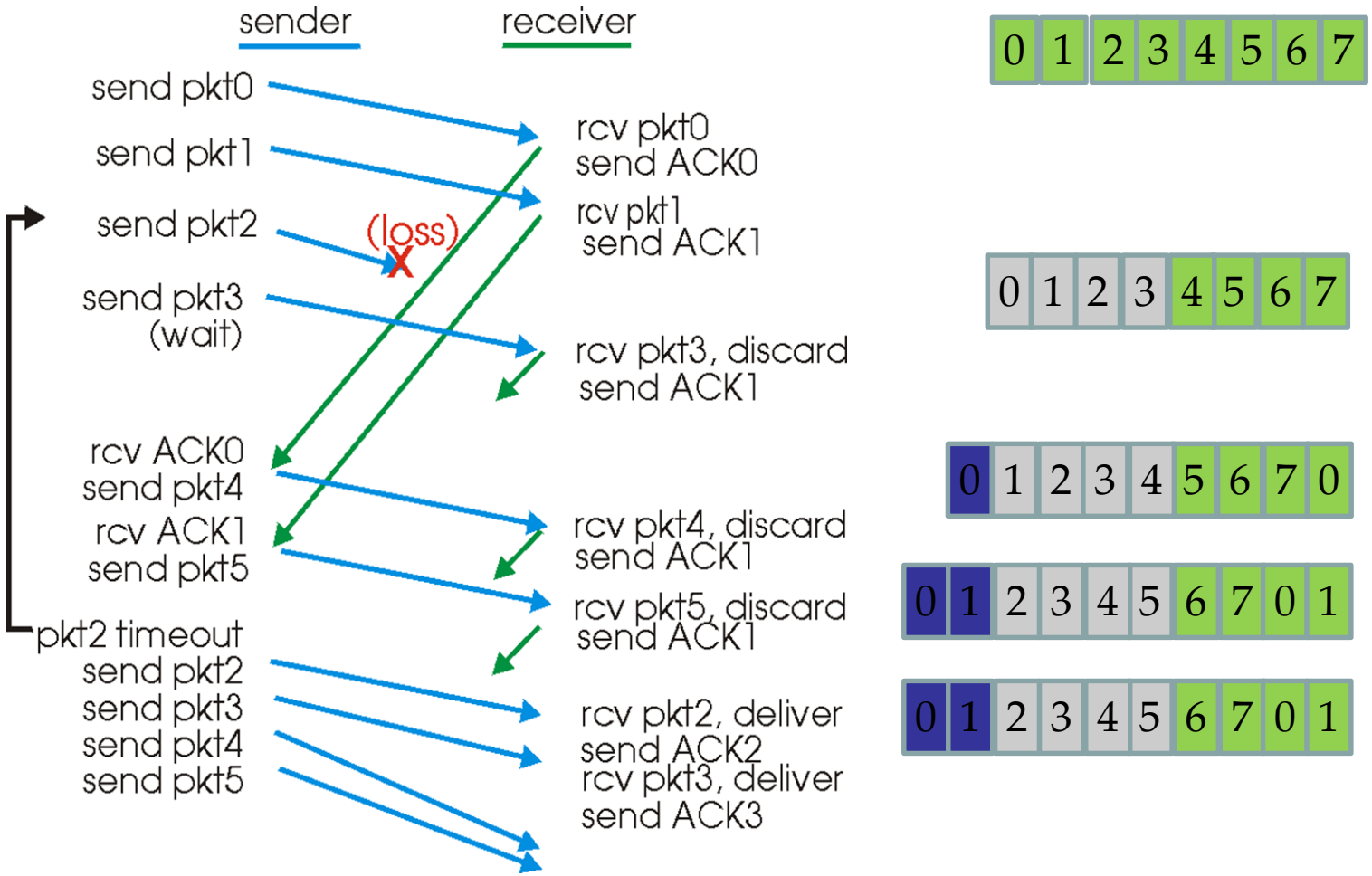
Then packet 2 (*out-of-order*) arrives uncorrupted,
Discard it;
Still send ACK 0;
Still expecting packet 1;

The receiver could have buffered packet 2 for later delivery; but if pkt 1 is lost, pkts 1 & 2 will be retransmitted later. For simplicity!

Then packet 1 arrives uncorrupted,
Deliver data to above;
Send ACK 1;
Now expecting packet 2;

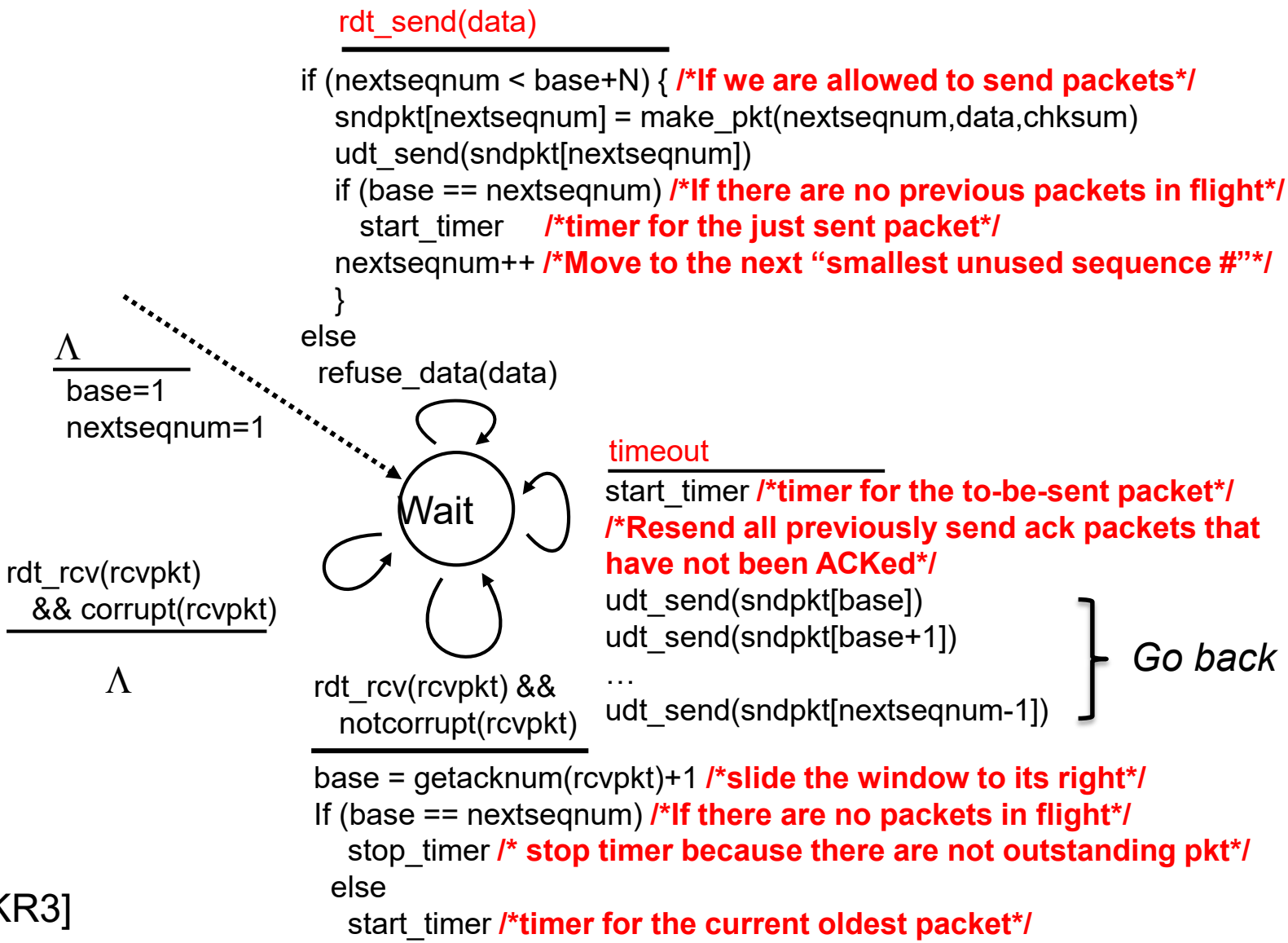
The receiver only needs to keep track of *expectedseqnum* (a variable).

Go-Back-N: an example



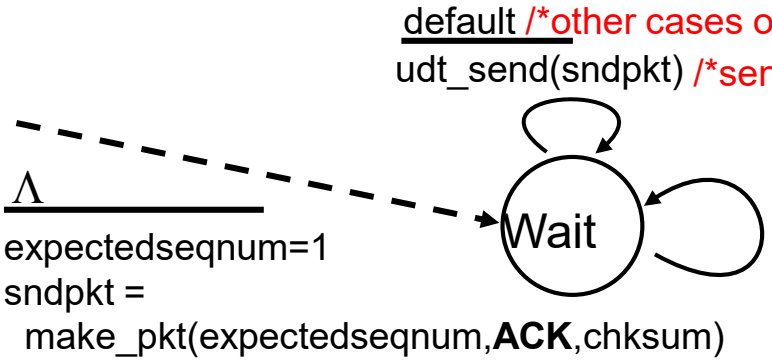
[AK05, KR3]

Go-Back-N: sender extended FSM

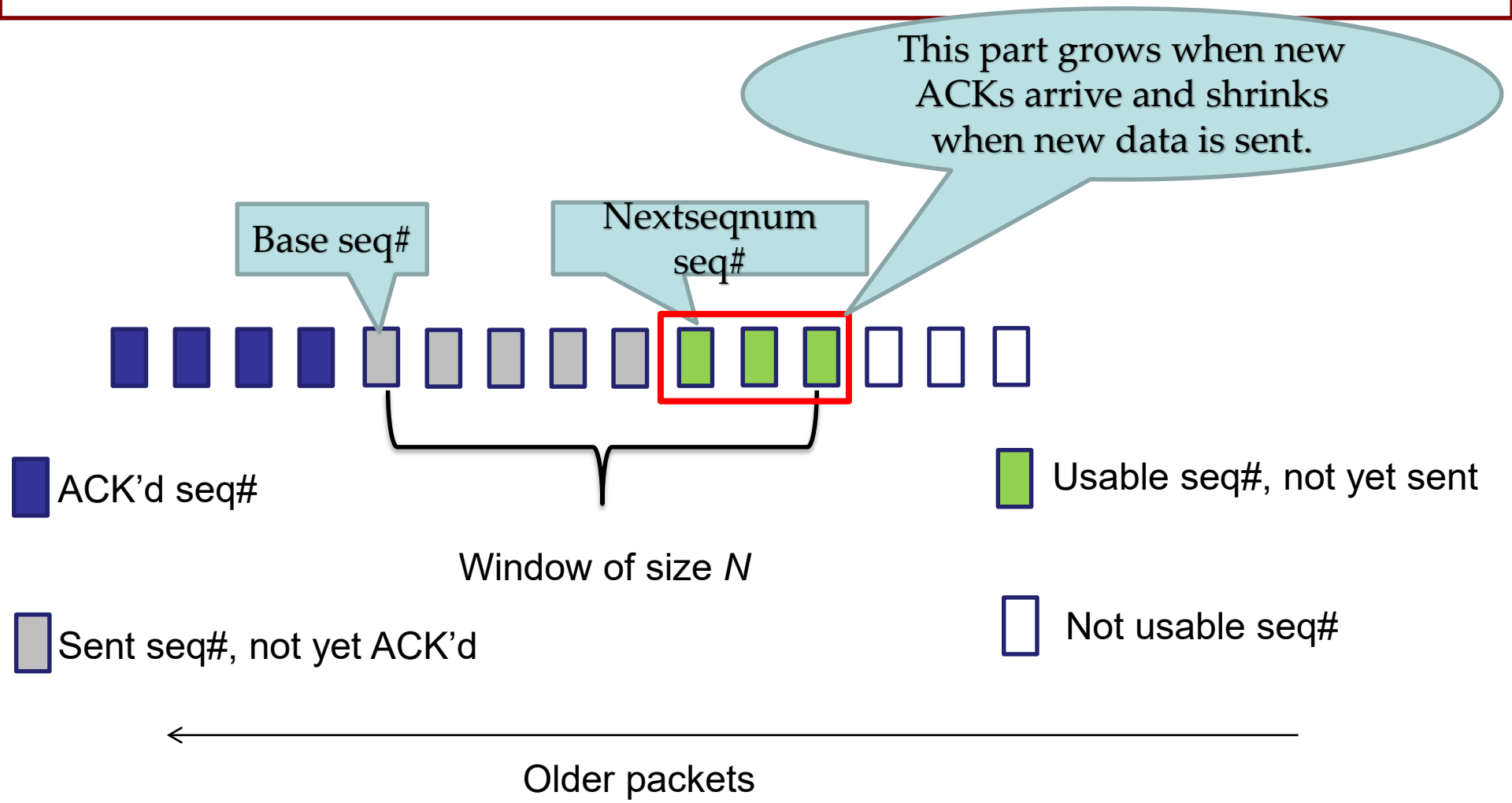


[AK05, KR3]

Go-Back-N: Receiver extended FSM



[AK05, KR3]

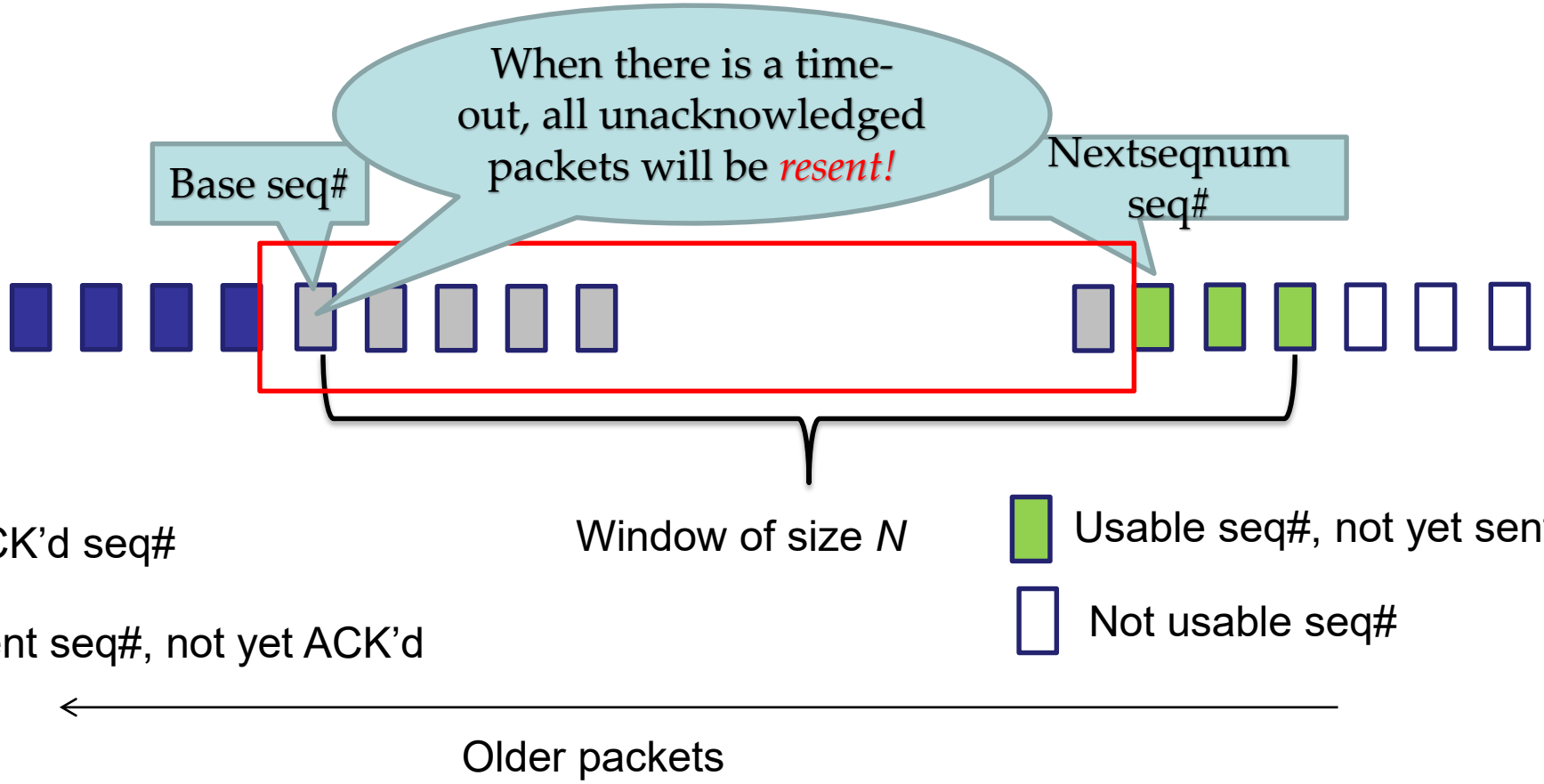

This part grows when new ACKs arrive and shrinks when new data is sent.
Base seq#
Nextseqnum seq#
ACK'd seq#
Usable seq#, not yet sent
Window of size N
Sent seq#, not yet ACK'd
Not usable seq#
Older packets

Tricks so far

- Checksum (bit errors)
- Seq # (new or duplicate)
- Duplicate ACK (NAK-free)
- Timer (to detect packet loss)
- Retransmission (to recover from transmission errors)
- Cumulative ACK (ACK all packets with no bigger seq#, in Go-Back-N)
- Window, pipelining (to improve sender utilisation, flow control, congestion control)

Go-Back-N is great!... But it has its own performance problem☹

Go-Back-N: Many packets in pipeline

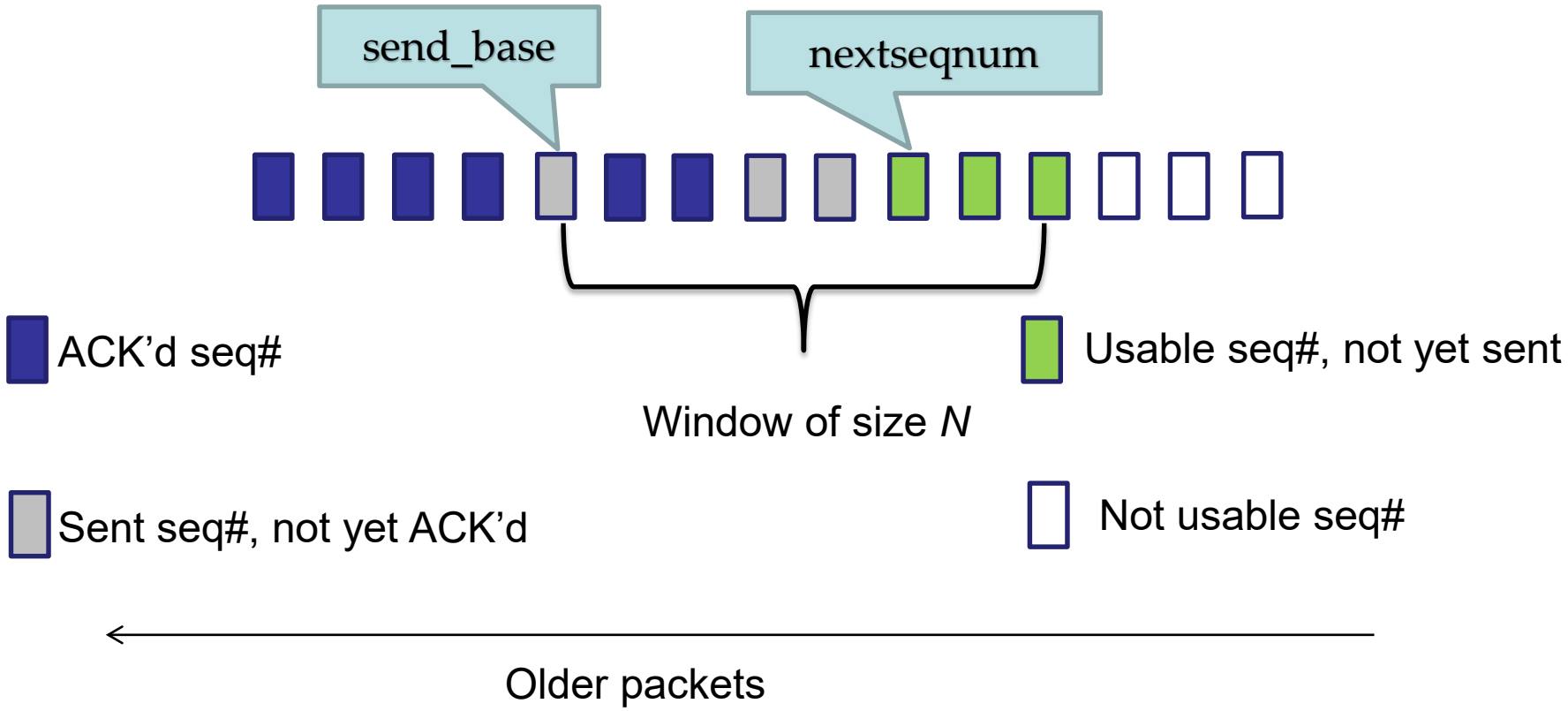


Selective Repeat/Reject (SR)

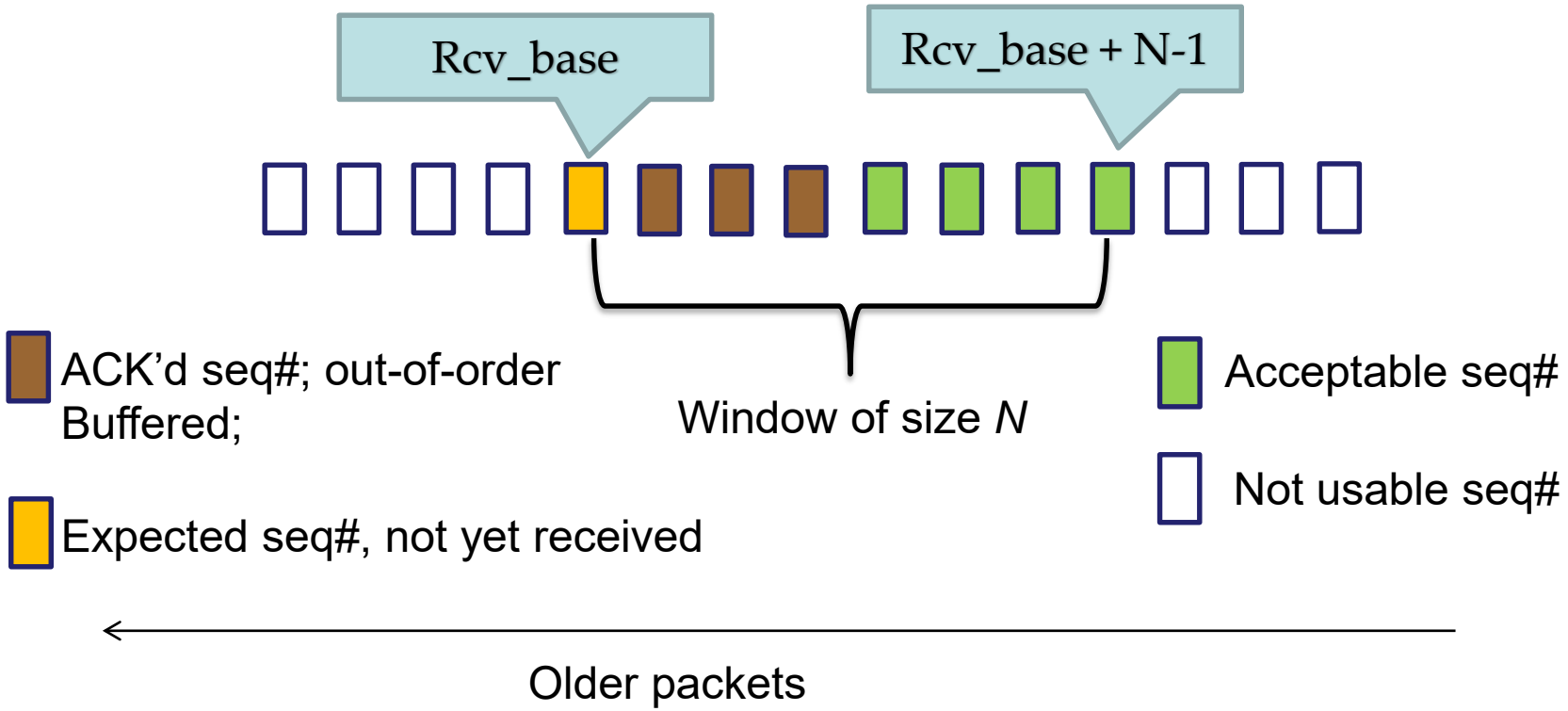
- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender has timer for each unACKed pkt
- sender window
  - N consecutive seq #'s
  - again limits seq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

[AK05]

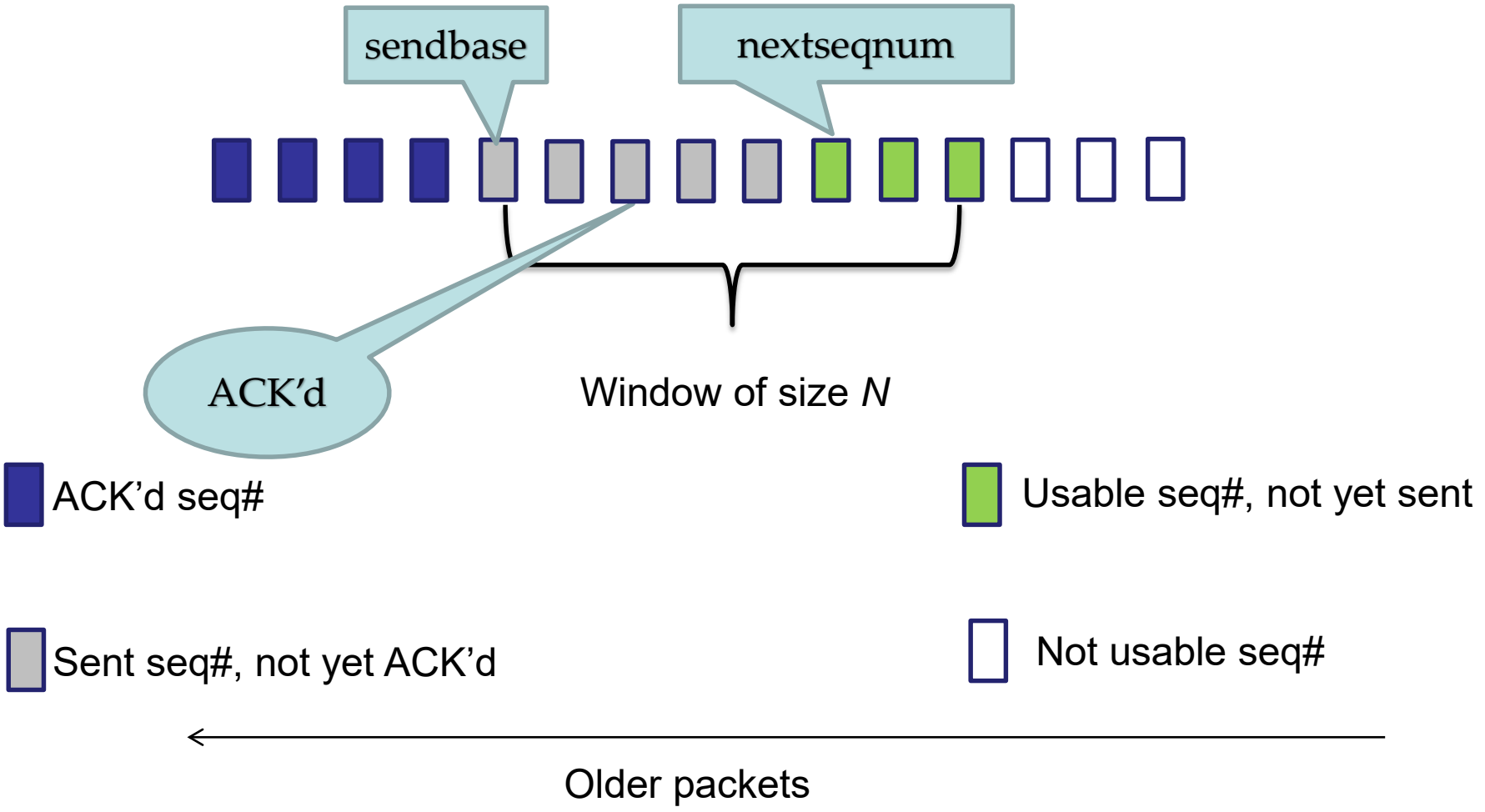
SR: Sender window



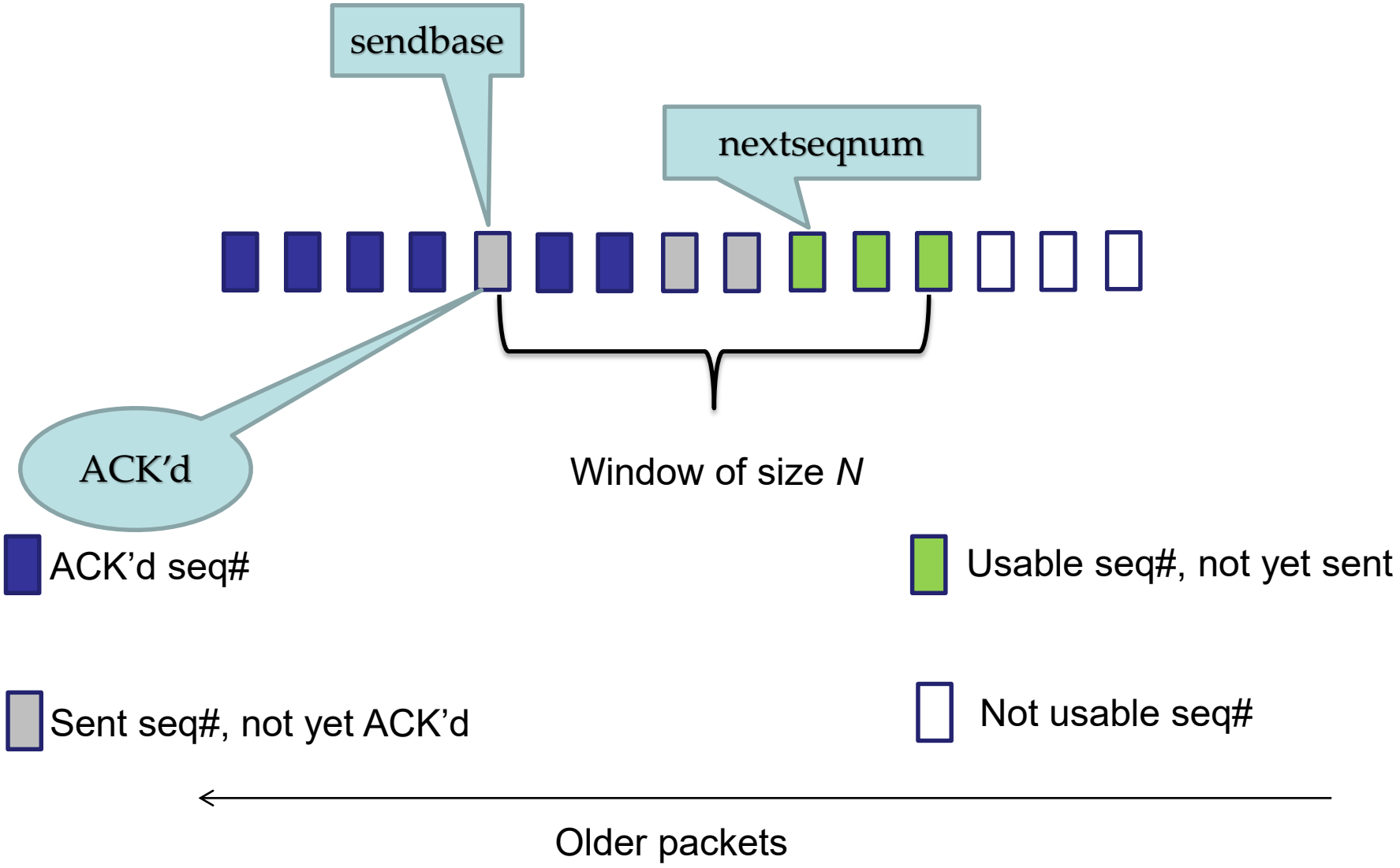
SR: Receiver window



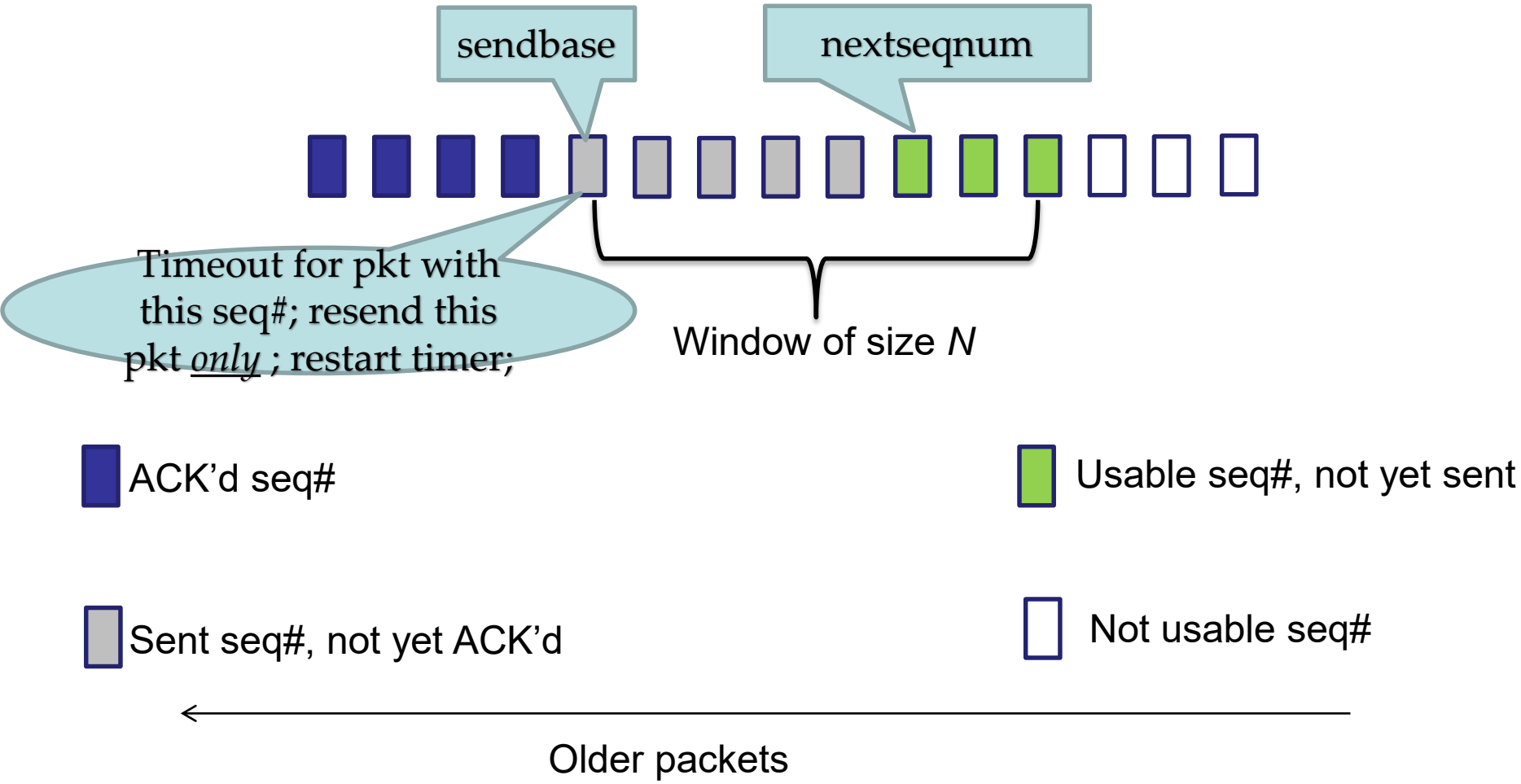
SR: Sender window – receipt of an ACK(1)



SR: Sender window – receipt of an ACK(2)



SR: Sender window – timeout



SR: sender side

sender

data from above :

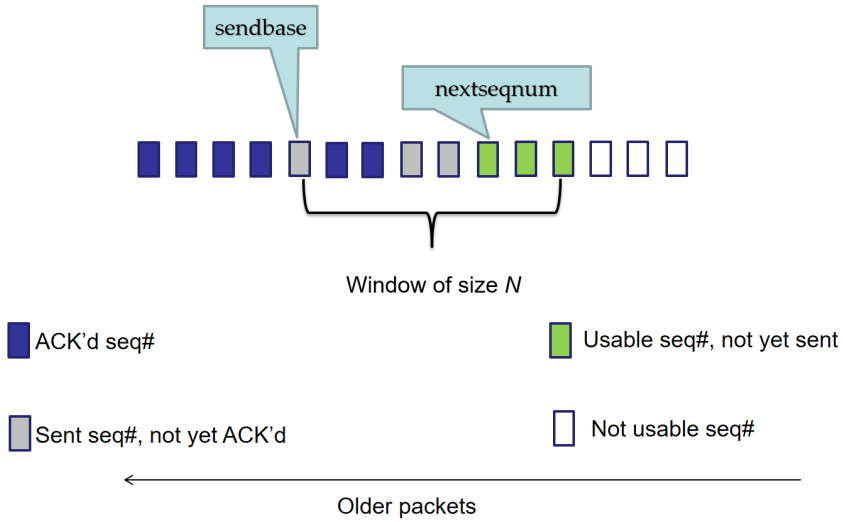
- if next available seq # in window, send pkt

timeout(n):

- resend pkt n, restart timer

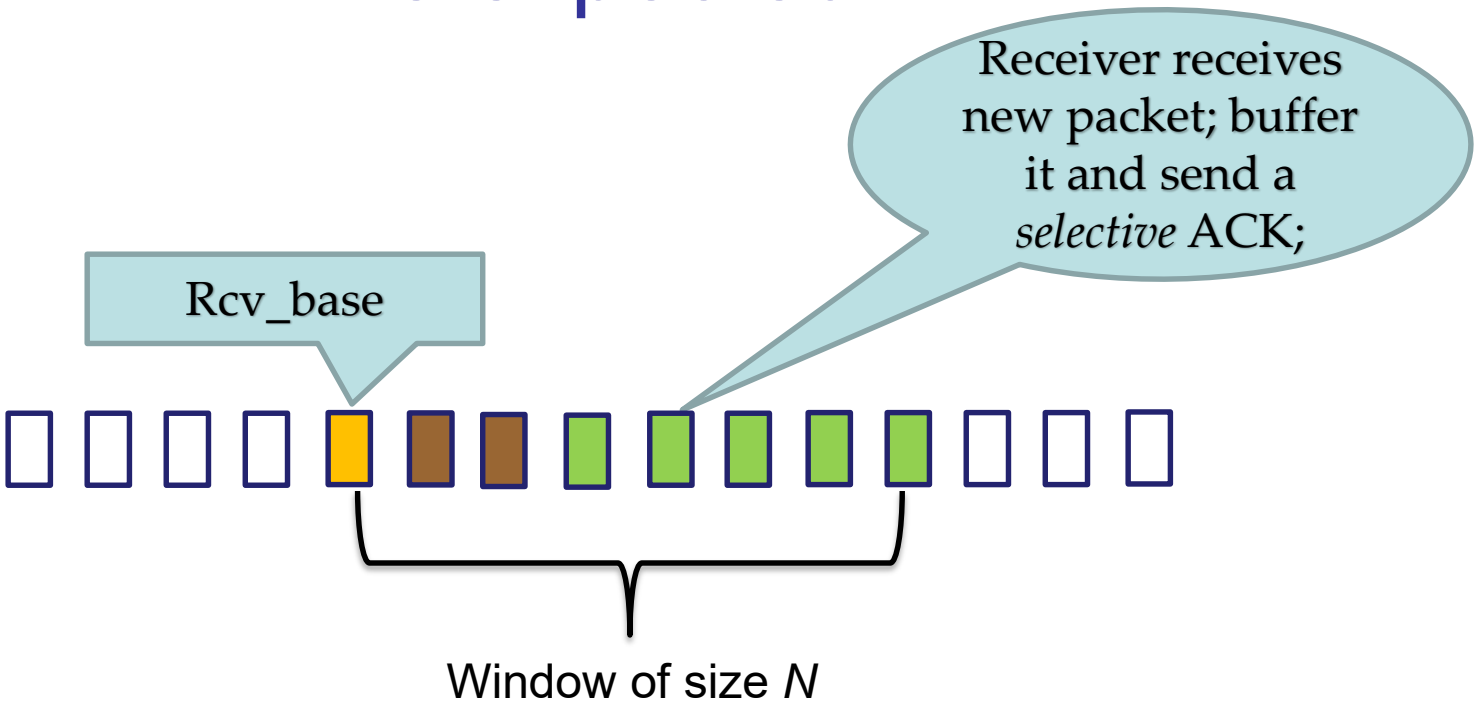
ACK(n) in [sendbase,sendbase+N-1]:

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



The only case when sender window moves.

SR: Receiver window (1) – new pkt not the expected



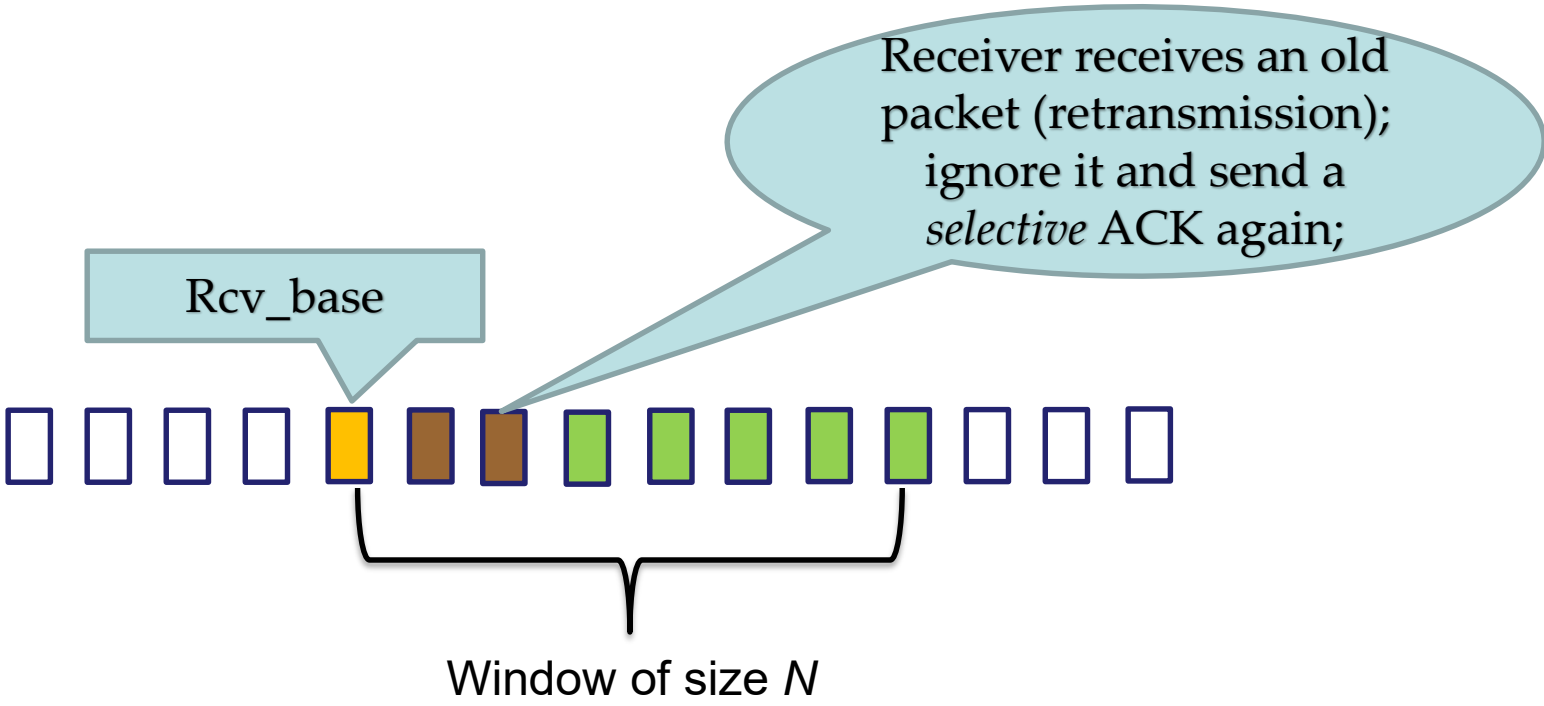
ACK'd seq#; out-of-order Buffered;

expected seq#, not yet received

acceptable seq#

Not usable seq#

SR: Receiver window (2) – old pkt



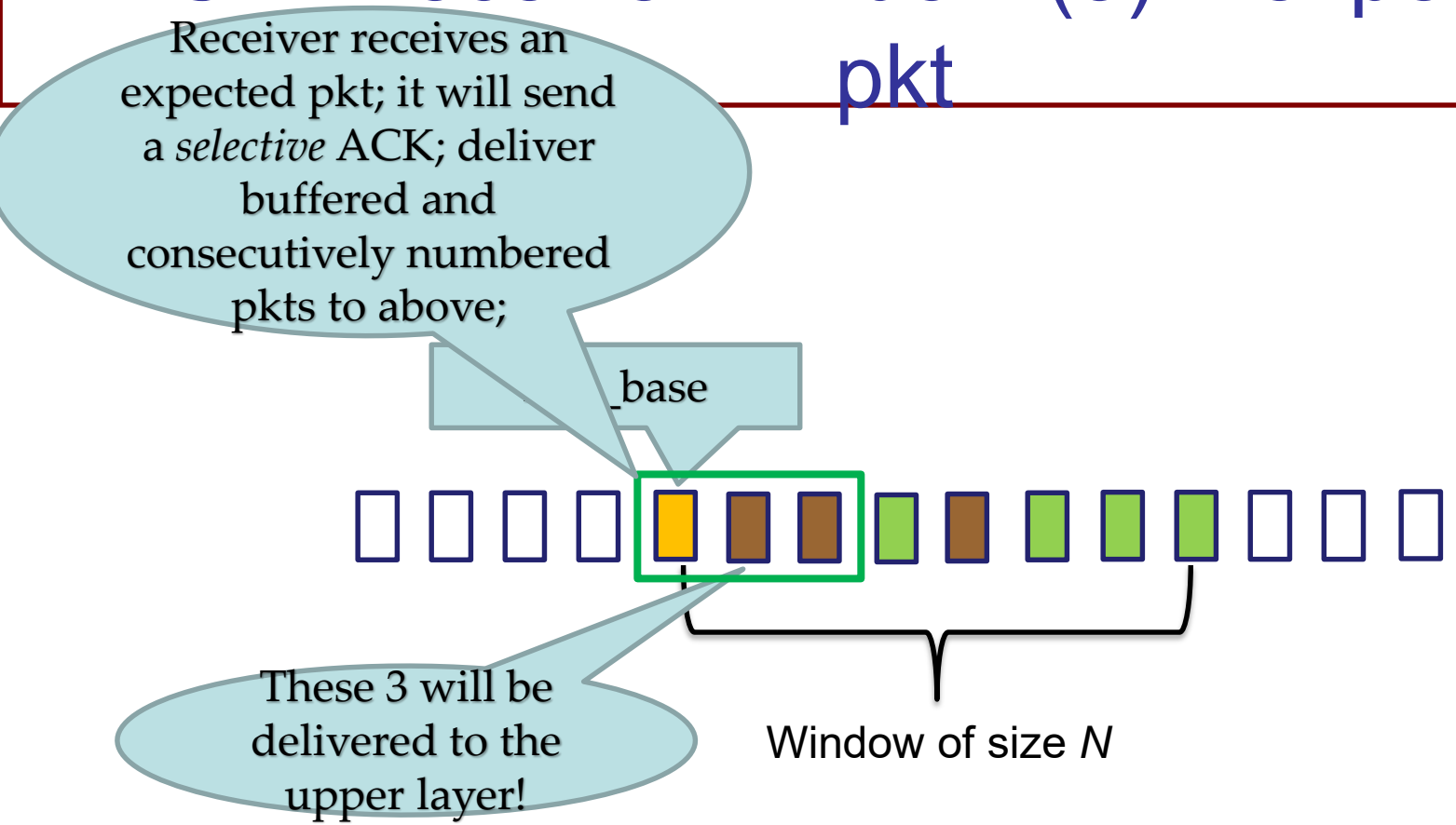
ACK'd seq#; out-of-order Buffered;

expected seq#, not yet received

acceptable seq#

Not usable seq#

SR: Receiver window (3) – expected pkt



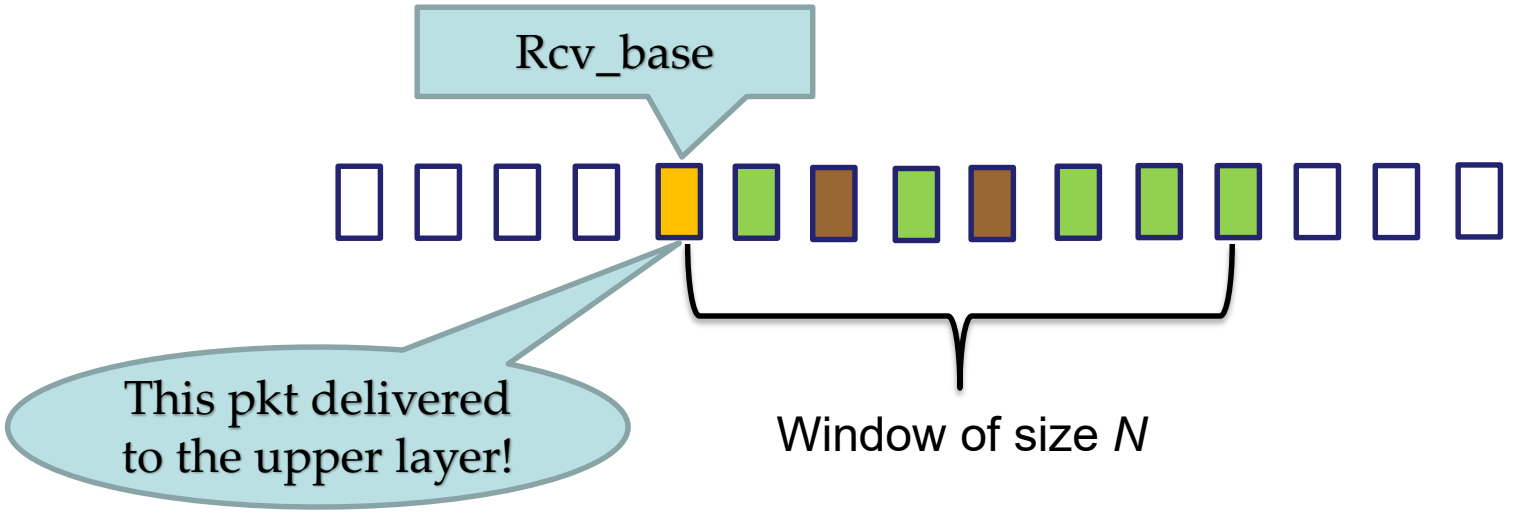
ACK'd seq#; out-of-order Buffered;

expected seq#, not yet received

acceptable seq#

Not usable seq#

SR: Receiver window (3) – expected pkt



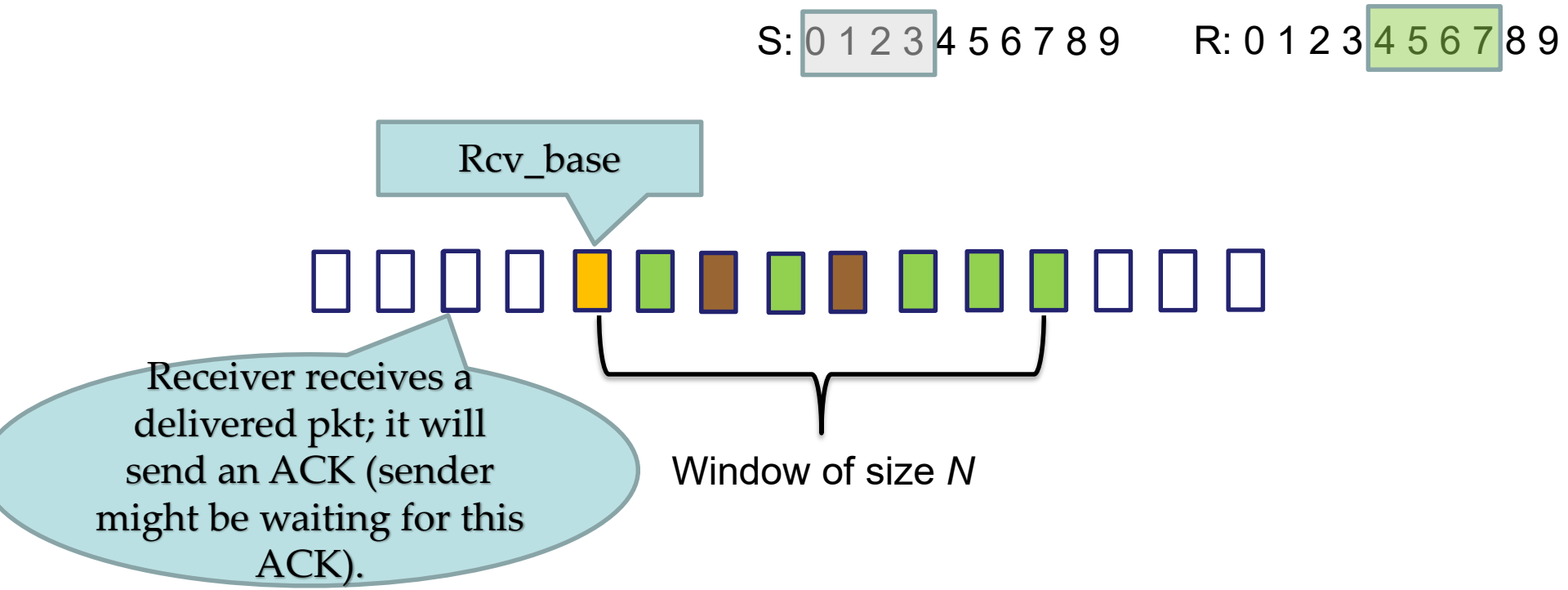
ACK'd seq#; out-of-order Buffered;

expected seq#, not yet received

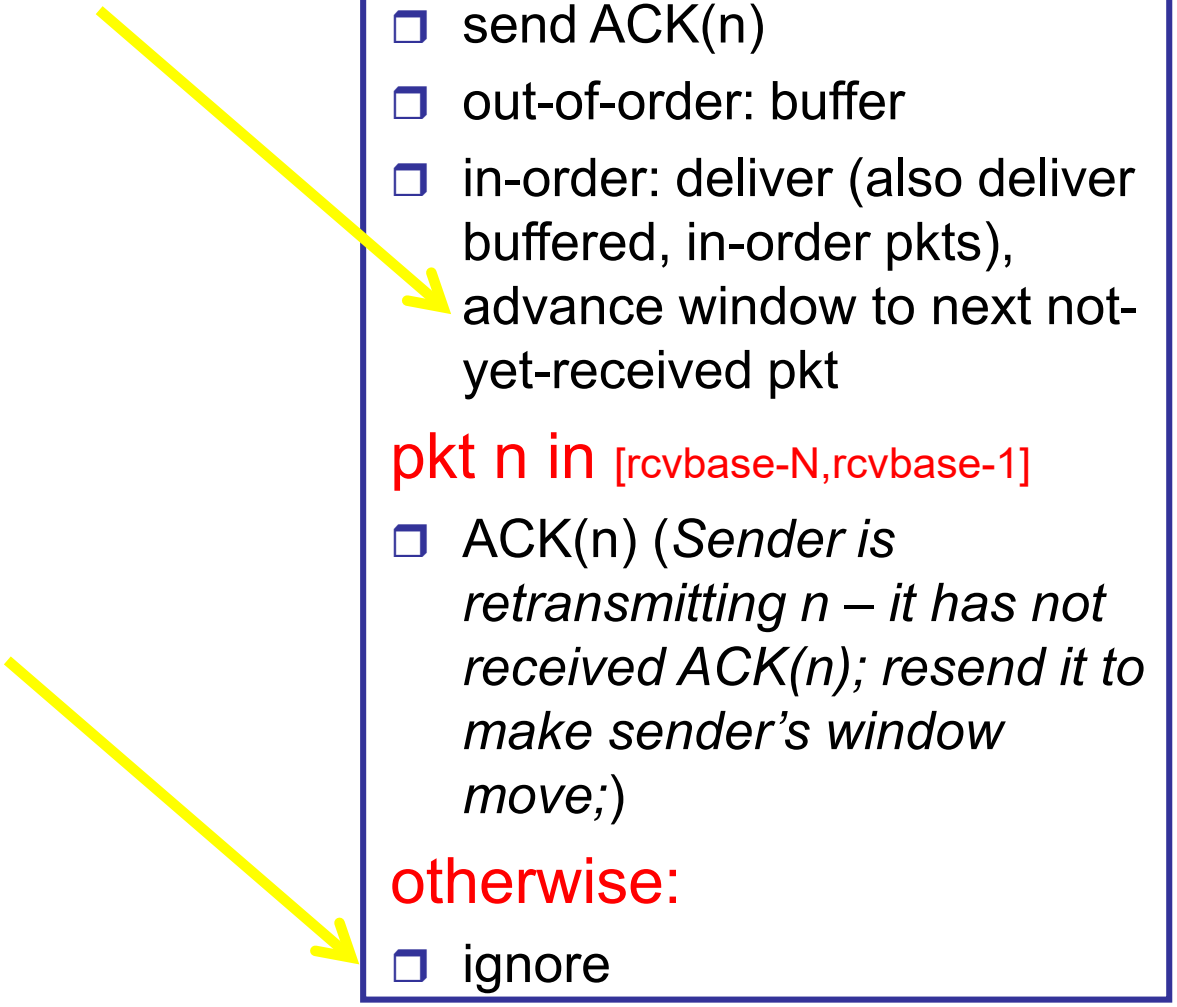
acceptable seq#

Not usable seq#

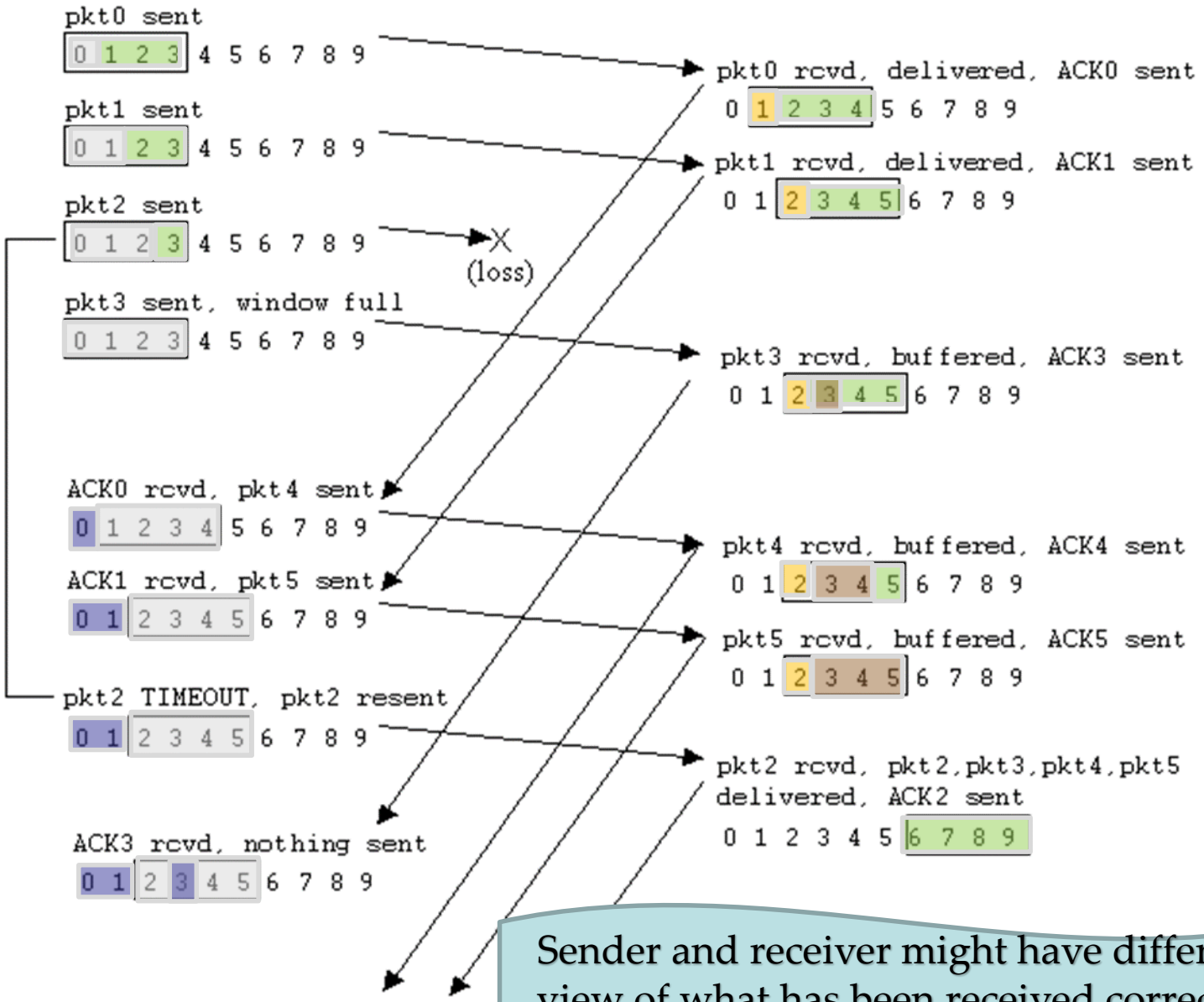
SR: Receiver window (3) – a delivered pkt - seq# is in [rcv_base-N, rcvbase-1]



SR: sender and receiver protocols



SR: an example (after-receipt window state)



Sender and receiver might have different view of what has been received correctly.

SR: receiver dilemma with too large window

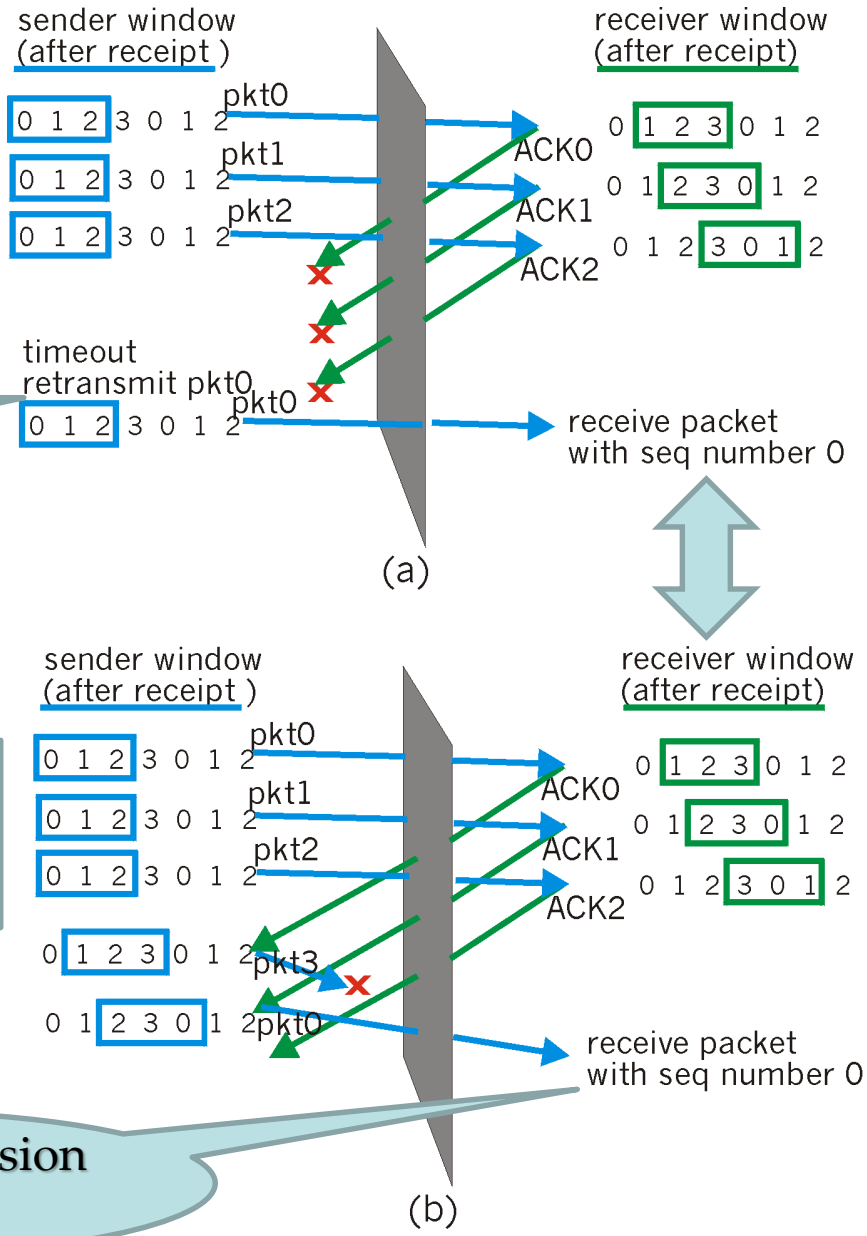
Example:

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

A retransmission with seq# 0

- receiver sees no difference in two scenarios!

Q: what relationship between seq # space and window size?



A new transmission with seq# 0

[AK05, KR3]

Window size vs seq# space

Q: what relationship between seq # space and window size?

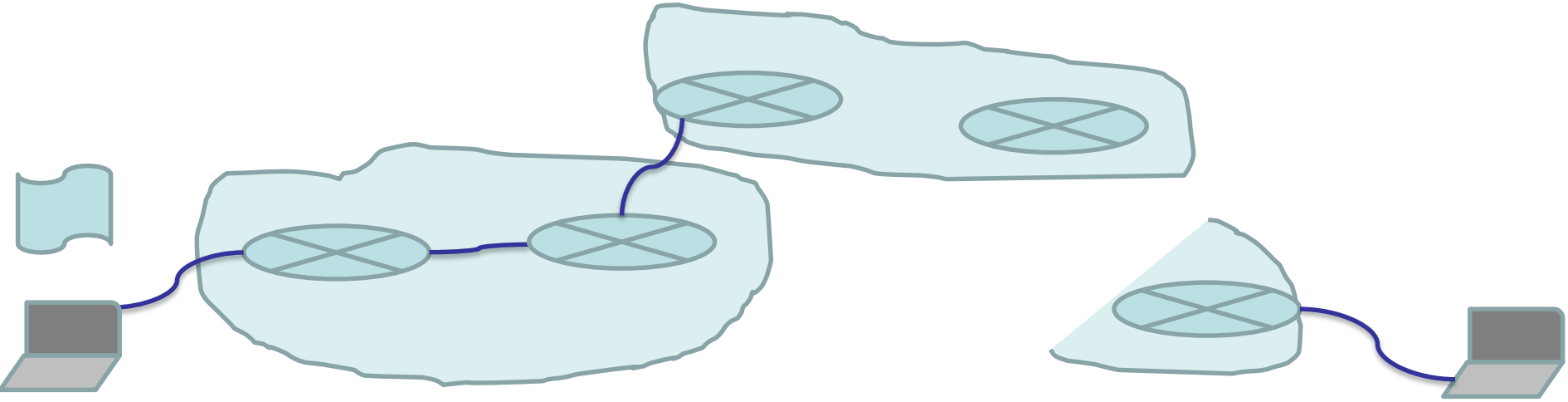
A: window size $\leq \frac{1}{2}$ (size of *seq# space*).

e.g., we use an 8-bit field for seq#; the window size should be no greater than 2^8/2 = 2^7 = 128;

On the other hand, if we use a window of size 4, we should at least use a 3-bit seq# field.

A hidden assumption

- We assume a packet will NOT be reordered within the channel between the sender and the receiver;
  - Reasonable for a single link;
  - Reordering could happen when the sender and receiver is connected by a multi-hop path;



- Old copies of a packet or an ACK with seq# *x* can appear. Neither sender's window or receiver's window contains *x*;
  - Give a limited "live" time to every packet in the network; *up to 3 mins in TCP ext. for high-speed network [RFC 1323].*
  - Refer to the "ignore" case in SR Receiver side protocol;

The tricks so far

Mechanism	Use, Comments
Checksum	To detect bit errors
Timer	To timeout/retransmit a packet (lost/premature packet)
Sequence number	To detect a lost packet (gap in seq#) To detect a duplicate packet (duplicate seq#)
Acknowledgement	To notify successful reception (individual/cumulative)
Negative ACK	To notify unsuccessful reception
Window, pipelining	To improve sender utilisation (utilisation vs performance)

Summary

- Reliable transfer protocols
  - rdt1.0: for a reliable channel
  - rdt2.0: for a channel with bit errors
  - rdt2.1: sender, handles garbled ACK/NAKs
  - rdt2.2: a NAK-free protocol
  - rdt3.0: channels with errors *and* loss
  - Pipelined protocols
    - Go-back-N
    - Selective repeat

References

- [KR3] James F. Kurose, Keith W. Ross, *Computer networking: a top-down approach featuring the Internet*, 3rd edition.
- [PD5] Larry L. Peterson, Bruce S. Davie, *Computer networks: a systems approach*, 5th edition
- [TW5] Andrew S. Tanenbaum, David J. Wetherall, *Computer network*, 5th edition
- [LHBi]Y-D. Lin, R-H. Hwang, F. Baker, *Computer network: an open source approach*, International edition

Acknowledgements

- All slides are developed based on slides from the following two sources:
  - Dr DongSeong Kim's slides for COSC264, University of Canterbury;
  - Prof Aleksandar Kuzmanovic's lecture notes for CS340,Northwestern University, https://users.cs.northwestern.edu/~akuzma/classes/CS340-w05/lecture_notes.htm