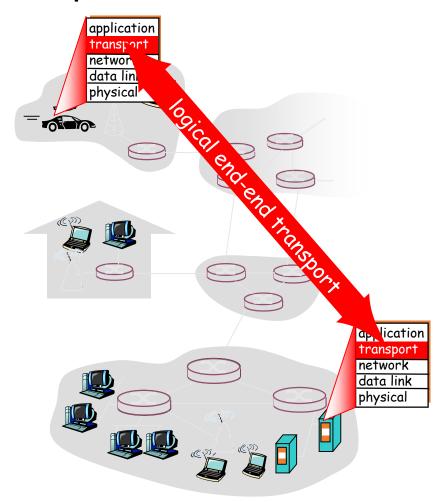
Transport Layer

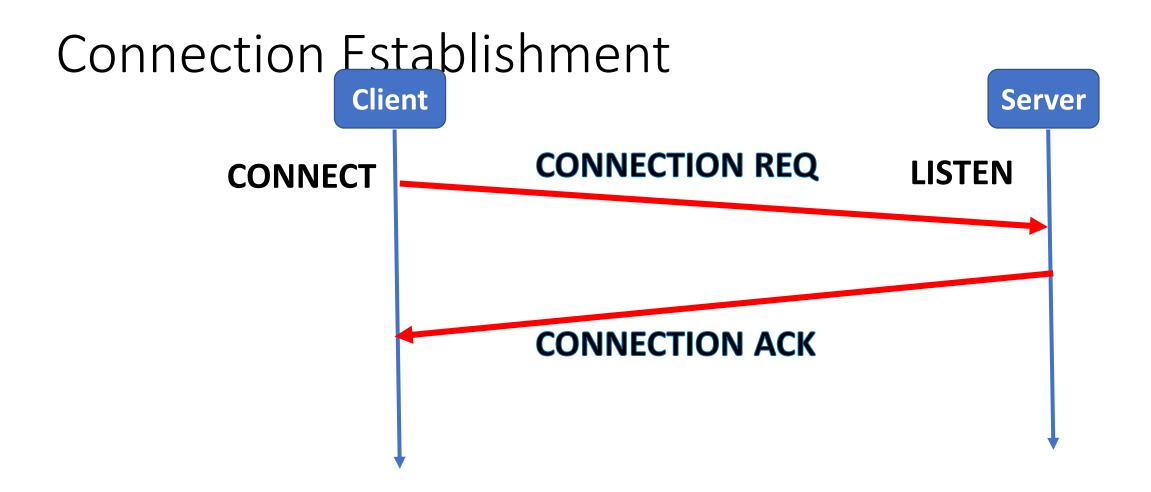
Transport services and protocols

- provide *logical communication* between app processes running on
 different hosts
- transport protocols run in end systems
 - send side: breaks app messages into segments, passes to network layer
 - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
 - Internet: TCP and UDP

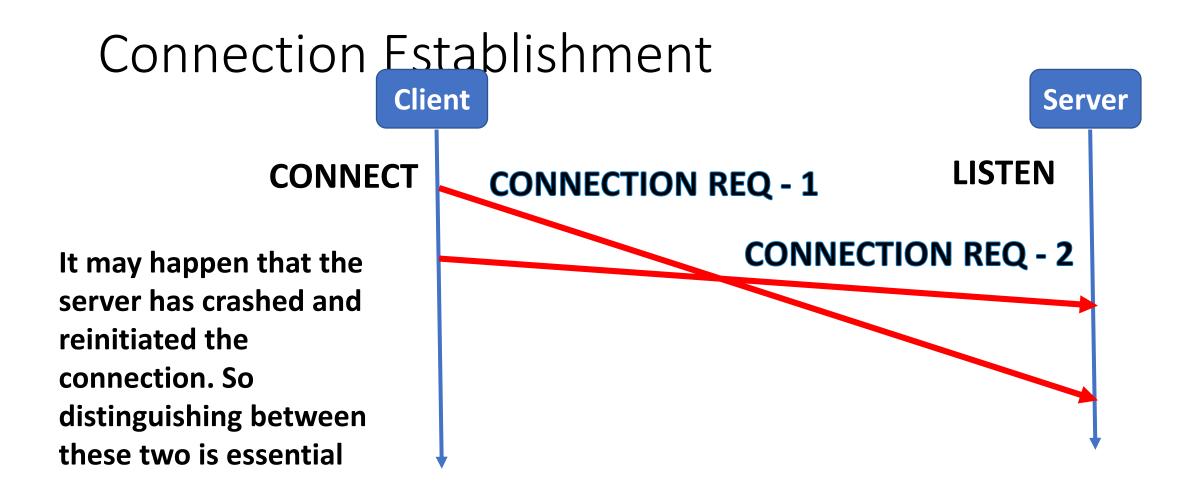


Transport Layer Services

- Connection establishment
- Error and flow control
- Congestion control



 This is a simple primitive for connection establishment – but does this work good?



• How will the server differentiate whether CONNECTION REQ-1 is a new connection request or a duplicate of the CONNECTION REQ-2?

Connection Establishment

 Consider a scenario when the network can lose, delay, corrupt and duplicate packets (the underline network layer uses unreliable data delivery)

- Consider retransmission for ensuring reliability every packet uses different paths to reach the destination
- Packets may be delayed and got struck in the network congestion, after the timeout, the sender assumes that the packets have been dropped, and retransmits the packets

Connection Establishment

• **Delayed duplicates** create a huge confusion in the packet switching network.

 A major challenge in packet switching network is to develop correct or at least acceptable protocols for handling delayed duplicates

How to handle delayed duplicates

- Solution 1: Give each connection a unique identifier chosen by the initiating party and put in each segment
 - Can you see any problem in this approach?

How to handle delayed duplicates

- Solution 1: Give each connection a unique identifier chosen by the initiating party and put in each segment
 - Can you see any problem in this approach?
 - Problem 1: Need to store the identifier (or sequence numbers) in a table at both sender and receiver side.

How to handle delayed duplicates

- Solution 1: Give each connection a unique identifier chosen by the initiating party and put in each segment
 - Can you see any problem in this approach?
 - Problem 1: Need to store the identifier (or sequence numbers) in a table at both sender and receiver side.

 Solution 2: Devise a mechanism to kill off aged packets that are still hobbling about (Restrict the packet lifetime)

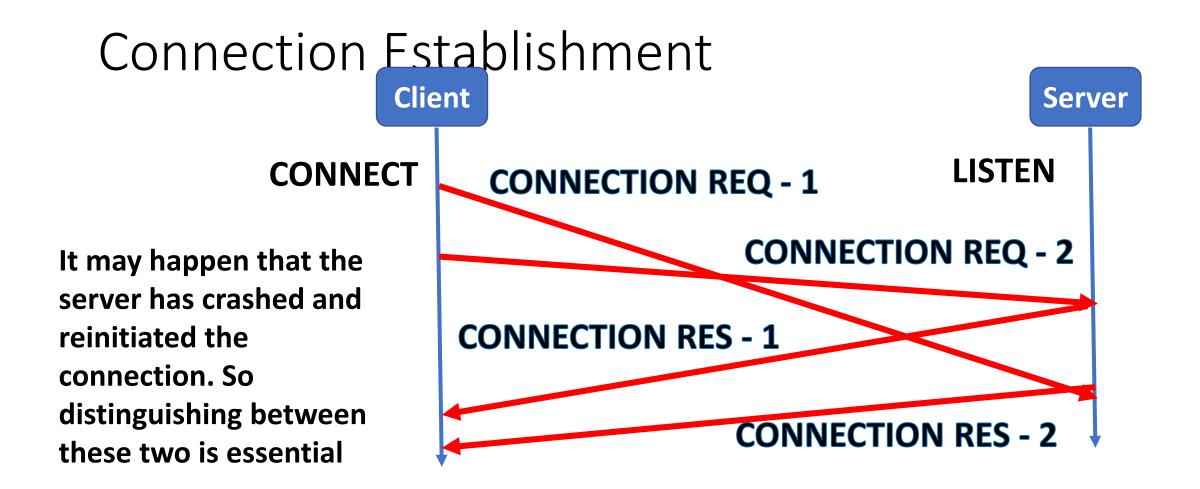
 Solution 2: Devise a mechanism to kill off aged packets that are still hobbling about (Restrict the packet lifetime)

- Two ways to restrict packet lifetime
 - Timestamping each packet define the lifetime of a packet in the network, need time synchronization across each router.

 Solution 2: Devise a mechanism to kill off aged packets that are still hobbling about (Restrict the packet lifetime)

- Two ways to restrict packet lifetime
 - Putting a hop count in each packet initialize to a maximum value and decrement each time the packet traverses a single hop (most feasible implementation)
 - Timestamping each packet define the lifetime of a packet in the network, need time synchronization across each router.

- Solution 2: Devise a mechanism to kill off aged packets that are still hobbling about (Restrict the packet lifetime)
- Two ways to restrict packet lifetime
 - Putting a hop count in each packet initialize to a maximum value and decrement each time the packet traverses a single hop (most feasible implementation)
 - Timestamping each packet define the lifetime of a packet in the network, need time synchronization across each router.
- Design Challenge: We need to guarantee not only that a packet is dead, but also that all acknowledgements to it are also dead



 How will the server differentiate whether CONNECTION RES-1 is a new connection response or a response of the CONNECTION RES-2?

 Design Challenge: We need to guarantee not only that a packet is dead, but also that all acknowledgements to it are also dead

 May be, we can use Solution 1 and 2 together for meeting the challenge

Let us define a maximum packet lifetime T – If we wait a time T secs after a
packet has been sent, we can be sure that all traces of it (packet and its
acknowledgement) are now gone

• Let us define a maximum packet lifetime T

 Rather than a physical clock (clock synchronization in the Internet is difficult to achieve), let us use a virtual clock – sequence number will represent a virtual clock

- Let us define a maximum packet lifetime T
- sequence number generated based on the clock ticks
- Label segments with sequence numbers that will not be reused within T secs.

 At most one packet with a given sequence number may be outstanding at any given time

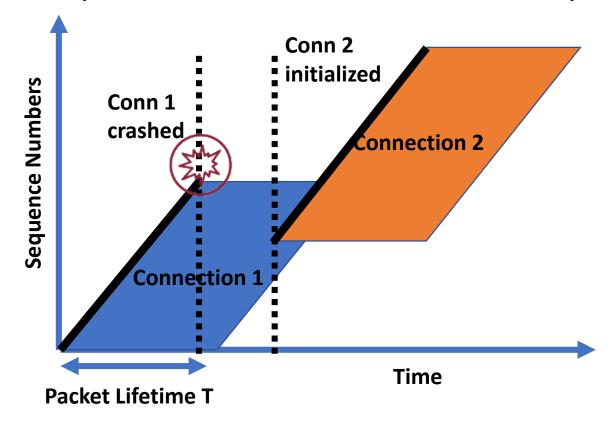
 At most one packet with a given sequence number may be outstanding at any given time

• If a receiver receives two segments having the same sequence number within a duration T, then one packet must be the duplicate. The receiver then discards the duplicate packets.

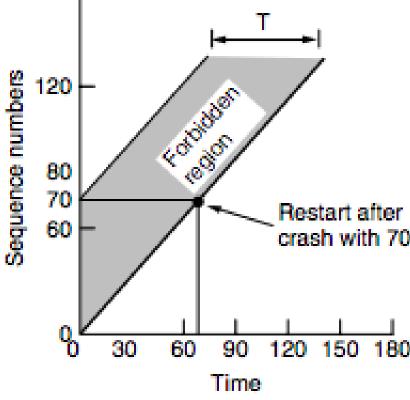
What happens in case of crash of sender/receiver??

Where to start from, what sequence number to use??

Why Initial Sequence Number is Important

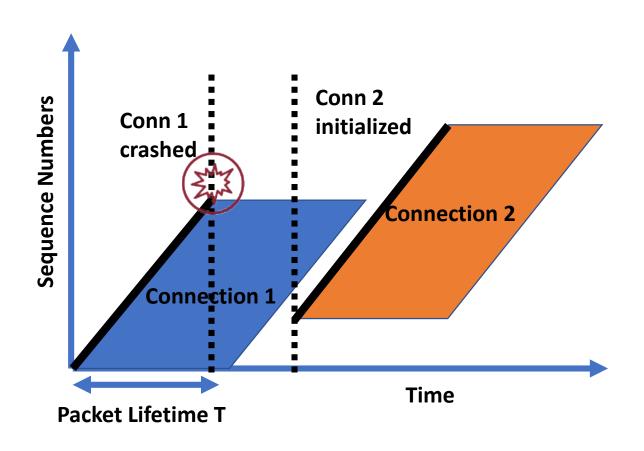


 A Delayed duplicate packet of connection 1 can create a confusion for connection 2

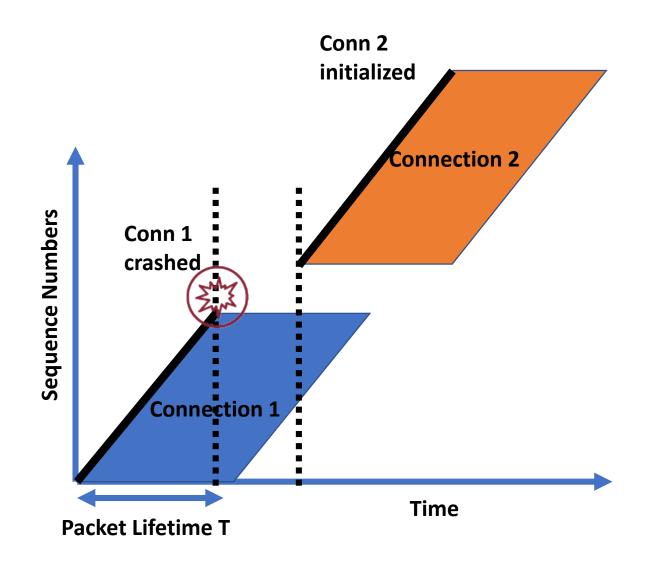


Adjust the initial sequence numbers properly - A host does not restart
with a sequence number in the forbidden region, based on the sequence
number it used before crash and the time duration T.

What We Ideally Want? Or ...



What We Ideally Want? Either ...

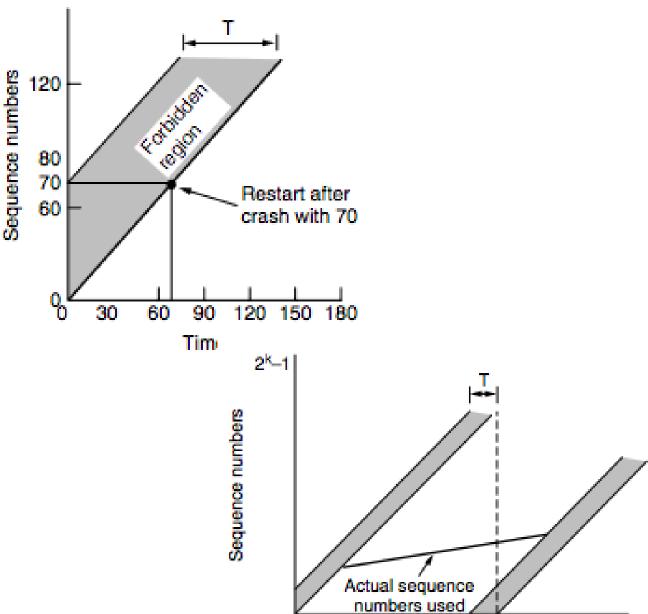


One problem while choosing a sequence number

 The period T and the rate of packets per second determine the size of the sequence number How do We Ensure that Packet Sequence Numbers are Out of the Forbidden Region

- Two possible source of problems
 - A host sends too much data too fast on a newly opened connection

 The data rate is too slow that the sequence number for a previous connection enters the forbidden region for the next connection



Adjusting the Sending Rate based on Sequence Numbers

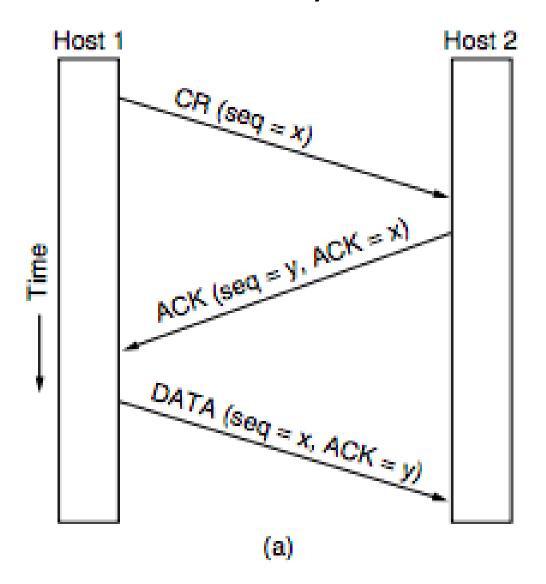
- The maximum data rate on any connection is one segment per clock tick
 - Clock ticks (inter-packet transmission duration) is adjusted based on the sequences acknowledged – ensure that no two packets are there in the network with same sequence number
 - We call this mechanism as self-clocking (used in TCP)
 - Ensures that the sequence numbers do not warp around too quickly (RFC 1323)

Adjusting the Sending Rate based on Sequence Numbers

- The maximum data rate on any connection is one segment per clock tick
 - Clock ticks (inter-packet transmission duration) is adjusted based on the sequences acknowledged – ensure that no two packets are there in the network with same sequence number
 - We call this mechanism as self-clocking (used in TCP)
 - Ensures that the sequence numbers do not warp around too quickly (RFC 1323)

- We do not remember sequence number at the receiver: Use a three way handshake to ensure that the connection request is not a repetition of an old connection request
 - The individual peers validate their own sequence number by looking at the acknowledgement (ACK)
 - Positive synchronization among the sender and the receiver

Three Way Handshake

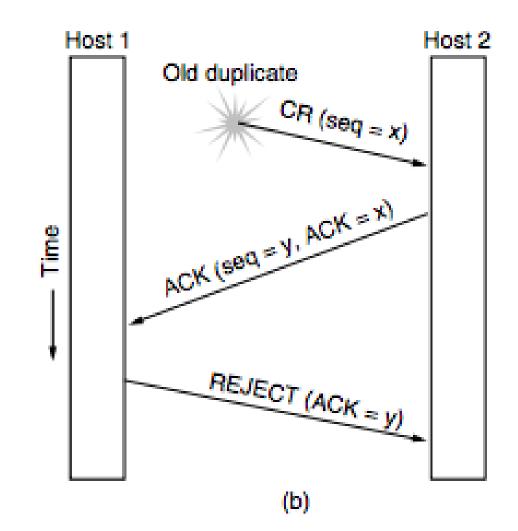


 By looking at the ACK, Host 1 ensures that Sequence number x does not belong to the forbidden region of any previously established connection

By looking at the ACK in DATA, Host 2
 ensures that sequence number y does not
 belong to the forbidden region of any
 previously established connection

Three Way Handshake – CONNECTION REQUEST is a Delayed Duplicate

Source: Computer Networks (5th Edition) by Tanenbaum, Wetherell



Three Way Handshake – CONNECTION REQUEST and ACKNOWLEDGEMENT both are Delayed

Duplicates

Host 1 Host 2 Old duplicate ACK (SEQ = Y, ACK = X) Old duplicate REJECT (ACK ≅ y) (c)

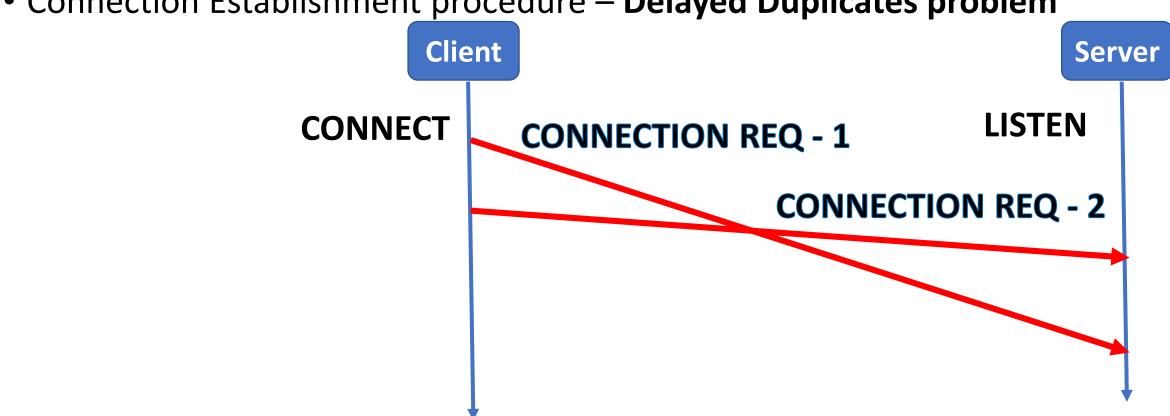
Source: Computer Networks (5th Edition) by Tanenbaum, Wetherell

Sequence Number Adjustment

- Two important requirements (*Tomlinson 1975, Selecting Sequence Numbers*)
 - Sequence numbers must be chosen such that a particular sequence number never refers to more than one byte (for byte sequence numbers) at any one time (how to choose the initial sequence number)
 - The valid range of sequence numbers must be positively synchronized between the sender and the receiver, whenever a connection is used (three way handshaking followed by the flow control mechanism – once connection is established, only send the data with expected sequence numbers)

Recap

Connection Establishment procedure – Delayed Duplicates problem



Recap

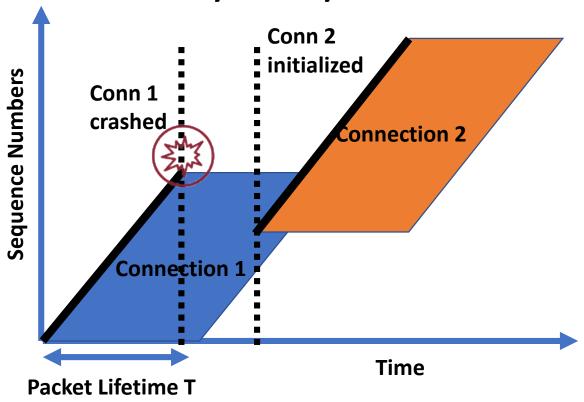
Connection Establishment – Handling Delayed Duplicates

- Solution:
 - Let us define a maximum packet lifetime T
 - Label segments with sequence numbers that will not be reused within T secs.

 At most one packet with a given sequence number may be outstanding at any given time

Recap

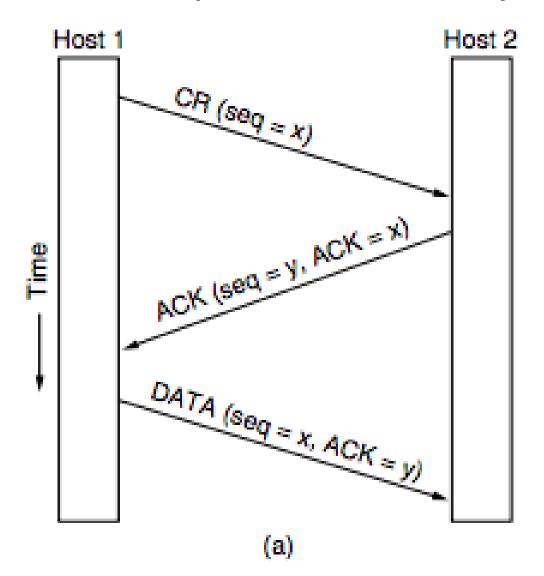
- Sequence Number Adjustment
 - Choosing an Initial sequence number is important as it does not clash with the sequence numbers currently used by another connection



Recap

- Sequence Number Adjustment
 - **3-way handshake:** The valid range of sequence numbers must be positively synchronized between the sender and the receiver, whenever a connection is used.
 - Receiver need to inform sender correctly which connection it is acknowledging, so that the sender can accordingly distinguish an old connection request from the current/new one

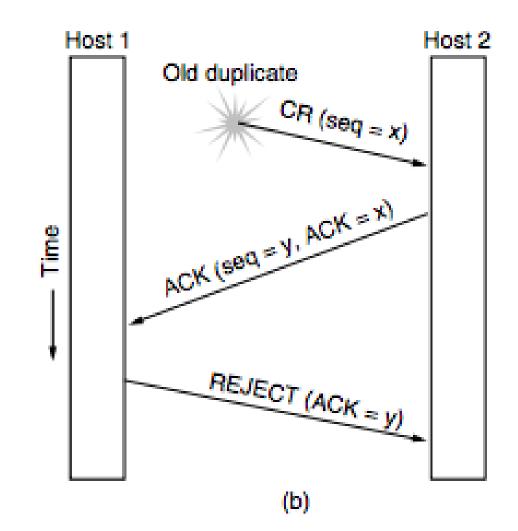
Recap - Three Way Handshake



 By looking at the ACK, Host 1 ensures that Sequence number x does not belong to the forbidden region of any previously established connection

By looking at the ACK in DATA, Host 2
 ensures that sequence number y does not
 belong to the forbidden region of any
 previously established connection

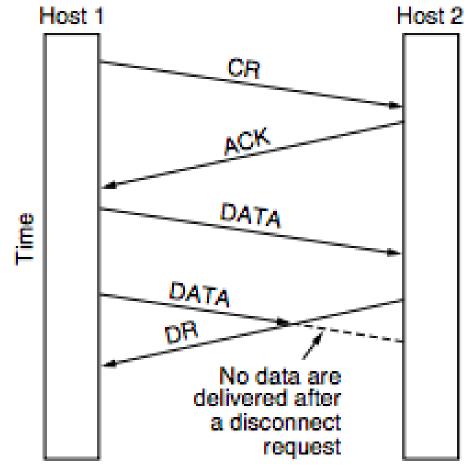
Three Way Handshake – CONNECTION REQUEST is a Delayed Duplicate



Connection Release – Asymmetric Release

• When one party hangs up, the connection is broken

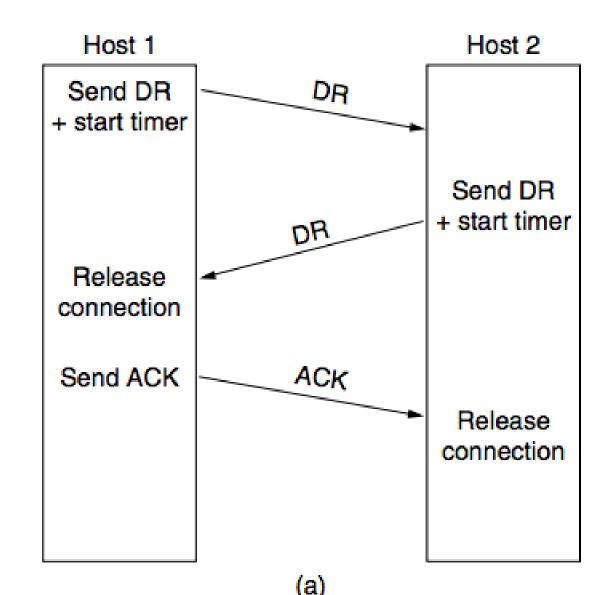
This may results in data loss



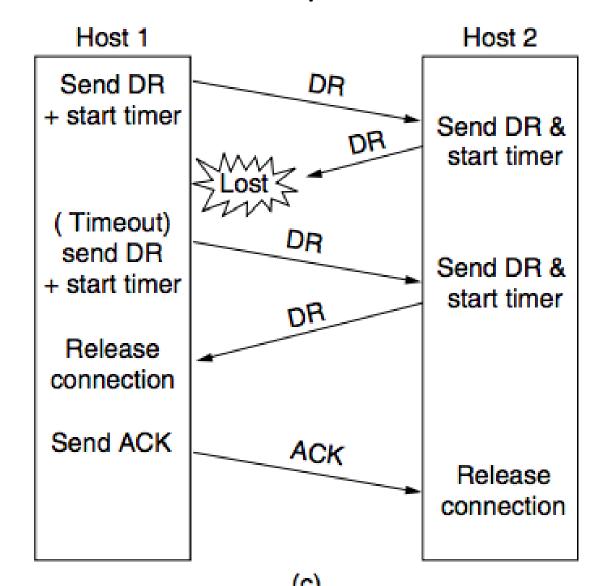
Connection Release – Symmetric Release

- Treats the connection as two separate unidirectional connections and requires each one to be released separately
- Does the job when each process has a fixed amount of data to send and clearly knows when it has sent it.
- What can be a protocol for this?
 - Host 1: "I am done"
 - Host 2: "I am done too"
- Does this protocol work good always?

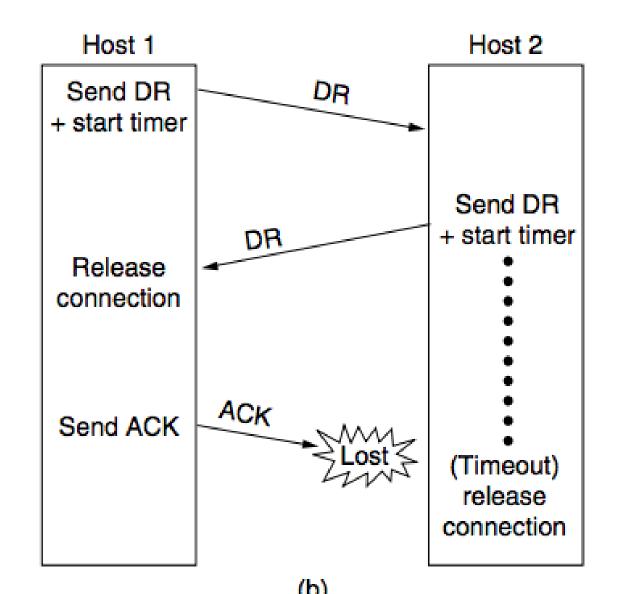
Connection Release



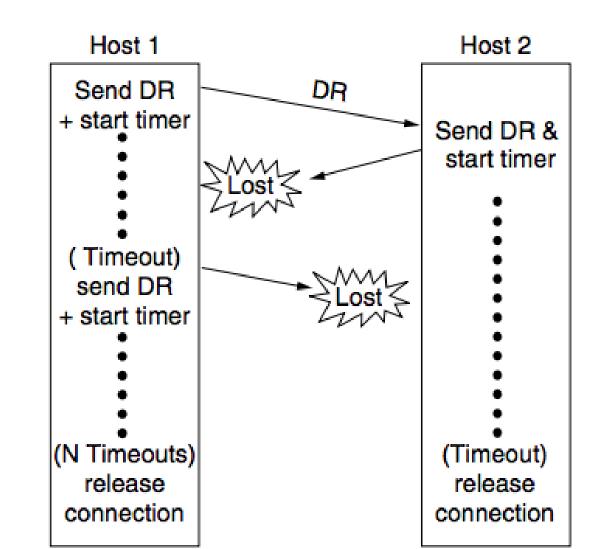
Connection Release – Response Lost



Connection Release – Final ACK Lost



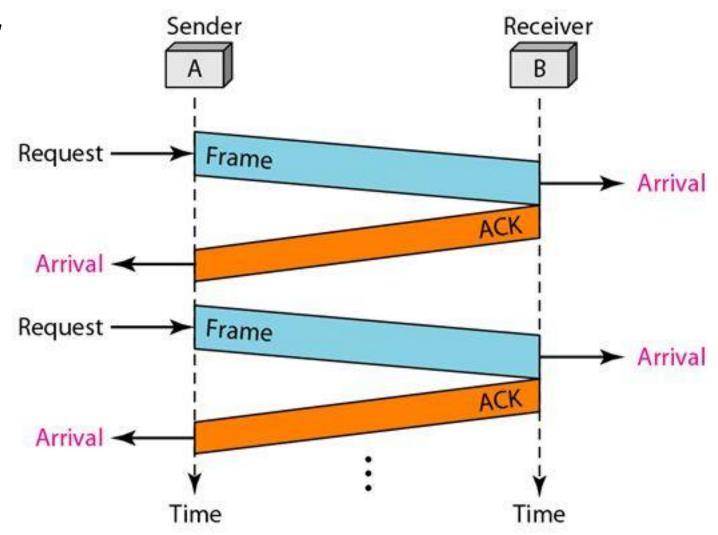
Connection Release – Response Lost and Subsequent DRs Lost



Flow Control Algorithms

Stop and Wait Flow

sender sends one frame and then waits for an acknowledgement before proceeding to send next frame.



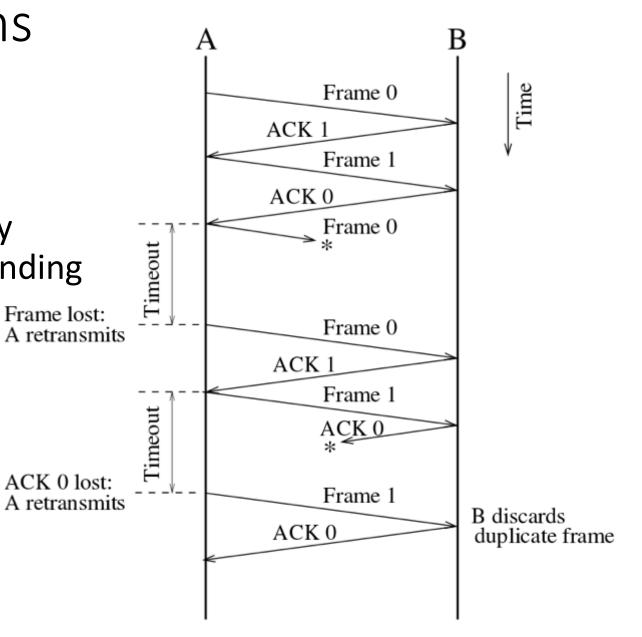
Flow Control Algorithms

Stop and Wait (Noisy Channel):

Use sequence numbers to individually identify each frame and the corresponding acknowledgement

Frame lost:

Automatic Repeat Request (ARQ)



Flow Control Algorithms

- Three things to consider when designing a reliable flow control algorithm
 - Sequence Numbers for distinguishing older packets and new ones (both data as well as ACKs).
 - Receiver Feedback using ACKs and NAKs
 - Retransmission after a fixed amount of Time, in case of lost data or ACKs

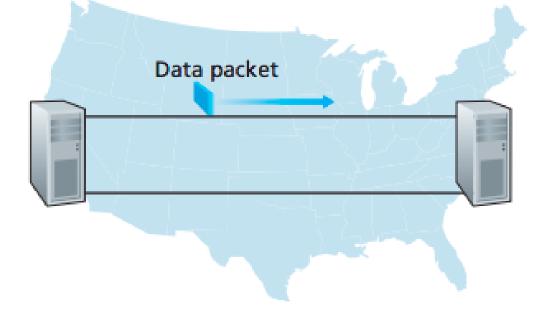
Stop-and-Wait ARQ:

- When a frame is sent, the sender starts the timeout counter.
- If acknowledgement of frame comes in time, the sender transmits the next frame in queue.
- If acknowledgement does not come in time, the sender assumes that either the frame or its acknowledgement is lost in transit. Sender retransmits the frame and starts the timeout counter.
- If a negative acknowledgement is received, the sender retransmits the frame.

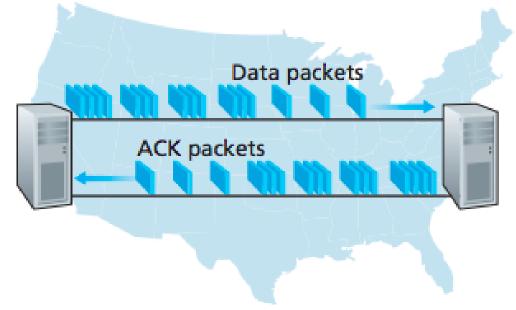
Problem with Stop and Wait

- Every packet needs to wait for the acknowledgement of the previous packet.
- For bidirectional connections use two instances of the stop and wait protocol at both directions – further waste of resources
- A possible solution: Piggyback data and acknowledgement from both the directions
- Reduce resource waste based on sliding window protocols (a pipelined protocol)

Stop and Wait versus Sliding Window (Pipelined)







b. A pipelined protocol in operation

Source: Computer Networks, Kurose, Ross

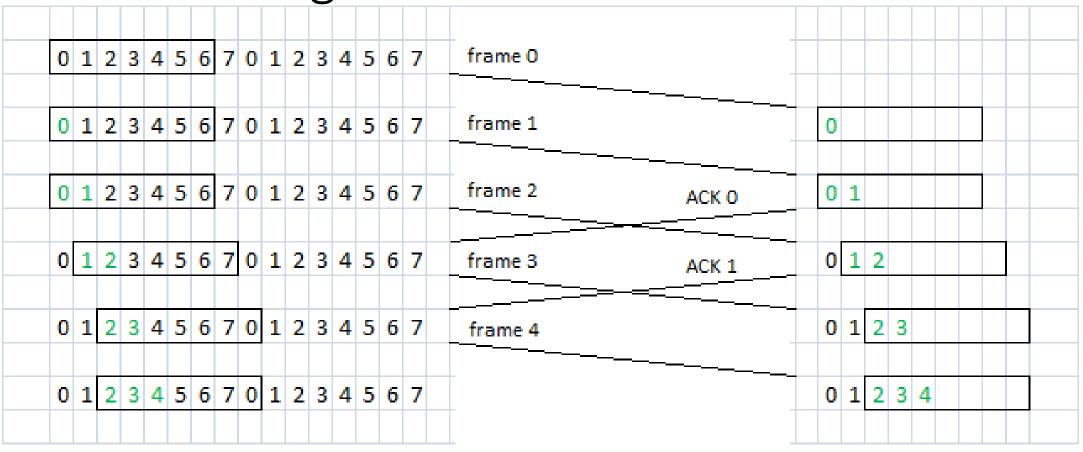
Sliding Window Protocols

• Each outbound segment contains a sequence number – from 0 to some maximum (2ⁿ-1 for a n bit sequence number)

 The sender maintains a set of sequence numbers corresponding to frames it is permitted to send (sending window)

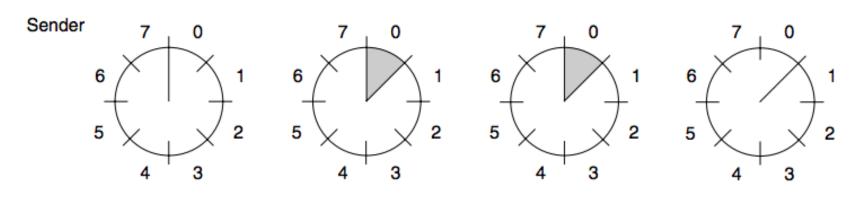
• The receiver maintains a set of frames it is permitted to accept (receiving window)

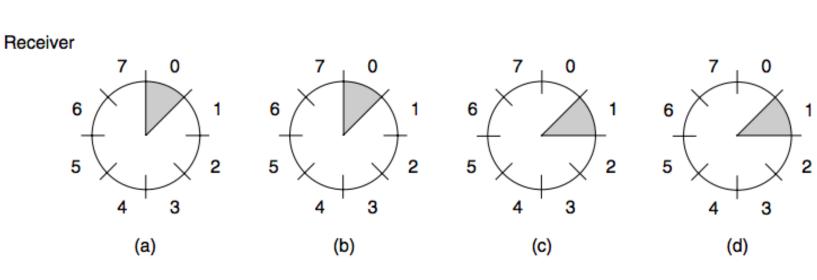
Sliding Window Protocols – Sending Window and Receiving Window



Sliding window Protocol
Source: http://ironbark.xtelco.com.au/subjects/DC/lectures/13/

Sliding Window for a 3 bit Sequence Number





Source: Computer Networks (5th Edition) by Tanenbaum, Wetherell

Figure 3-15. A sliding window of size 1, with a 3-bit sequence number. (a) Initially. (b) After the first frame has been sent. (c) After the first frame has been received. (d) After the first acknowledgement has been received.

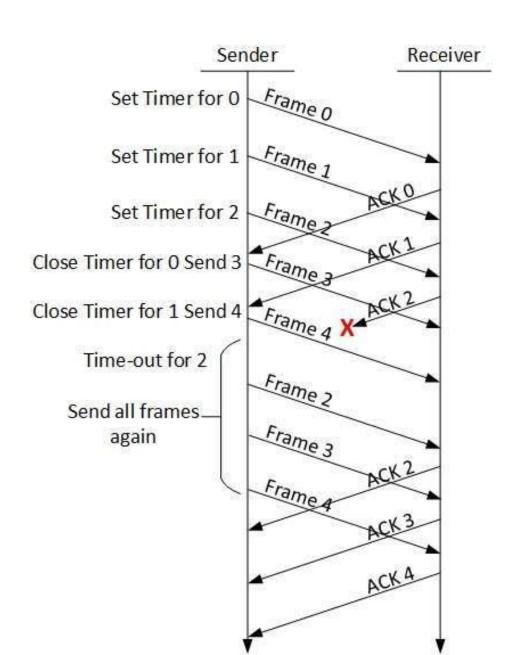
Sliding Window Protocols in Noisy Channels

- A timeout occurs if a segment (or the acknowledgment) gets lost
- How does the flow and error control protocol handle a timeout?
- Go Back N ARQ: If segment N is lost, all the segments from segment N are retransmitted
- Selective Repeat (SR) ARQ: Only the lost packets are selectively retransmitted
 - Negative Acknowledgement (NAK) or Selective Acknowledgements (SACK): Informs
 the sender about which packets need to be retransmitted (not received by the
 receiver)

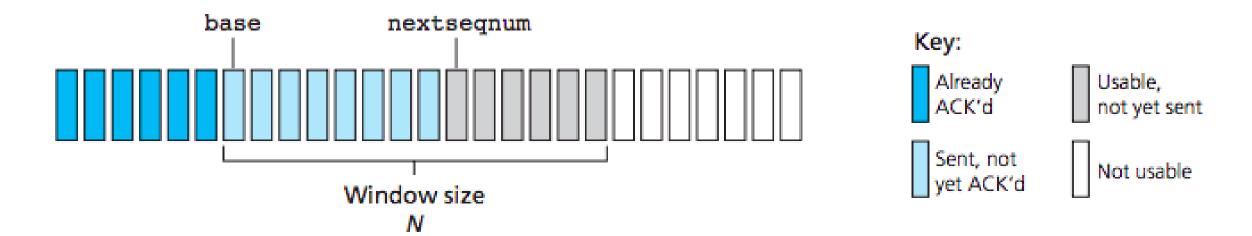
Go Back N ARQ

Source

https://www.tutorialspoint.com/data_com munication_computer_network/data_link control_and_protocols.htm



Go Back N ARQ – Sender Window Control

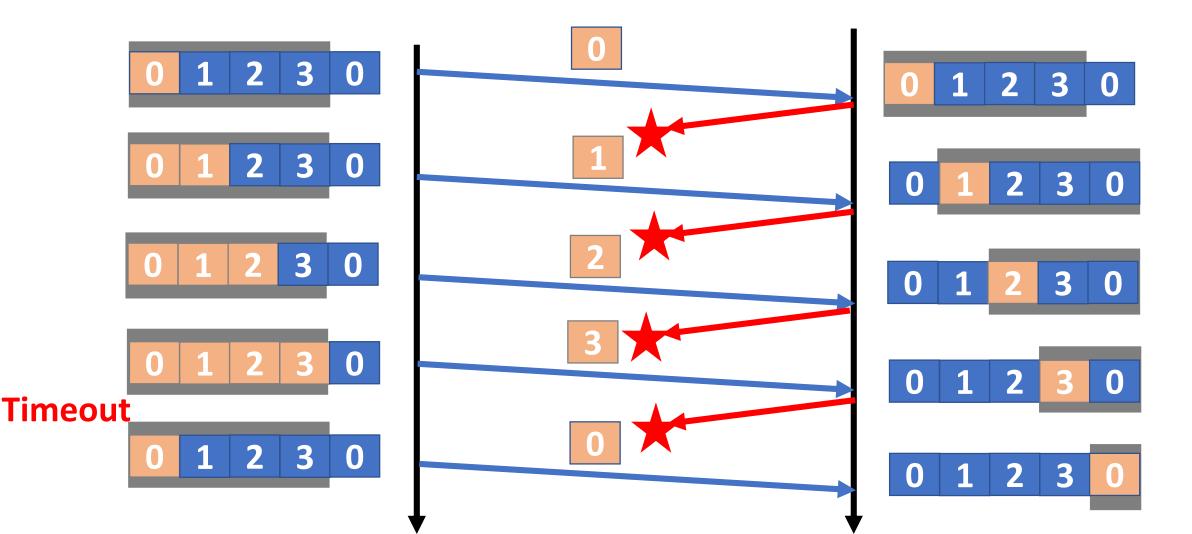


Source: Computer Networks, Kurose, Ross

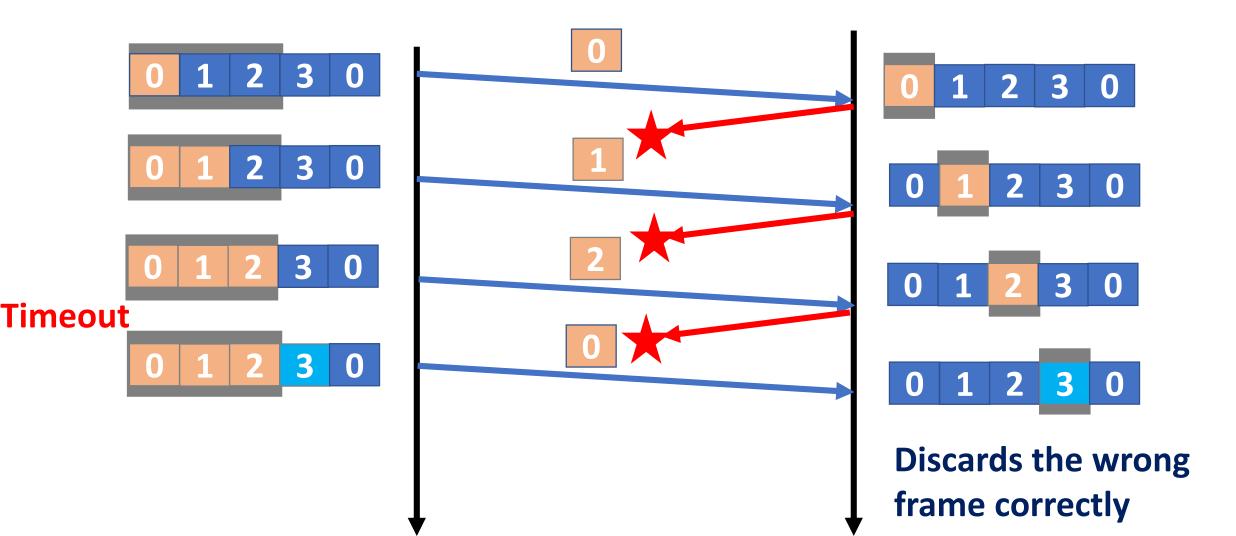
Go Back N ARQ – A Bound on Window Size

- Outstanding Frames Frames that have been transmitted, but not yet acknowledged
- Maximum Sequence Number (MAX_SEQ): MAX_SEQ+1 distinct sequence numbers are there
 - 0,1,...,MAX_SEQ
- Maximum Number of Outstanding Frames (=Window Size): MAX_SEQ
- Example: Sequence Numbers (0,1,2,...,7) 3 bit sequence numbers, number of outstanding frames = 7 (Not 8)

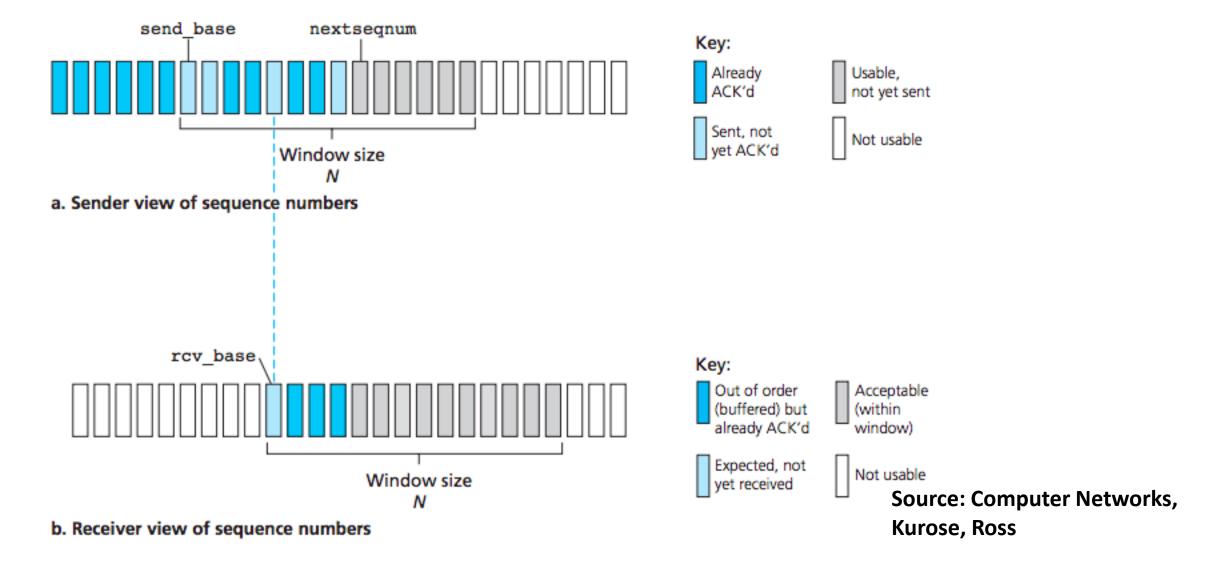
Go Back N ARQ — A Bound on Window Size • Let MAX_SEQ = 3, Window Size = 4



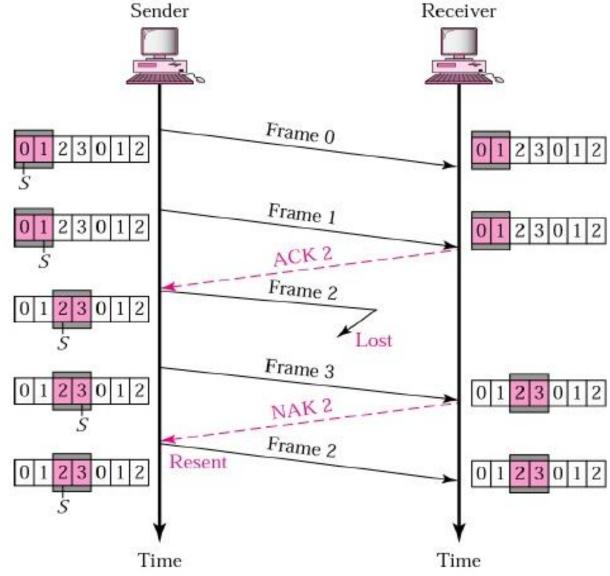
Go Back N ARQ — A Bound on Window Size • Let MAX_SEQ = 3, Window Size = 3



Selective Repeat (SR) – Window Control



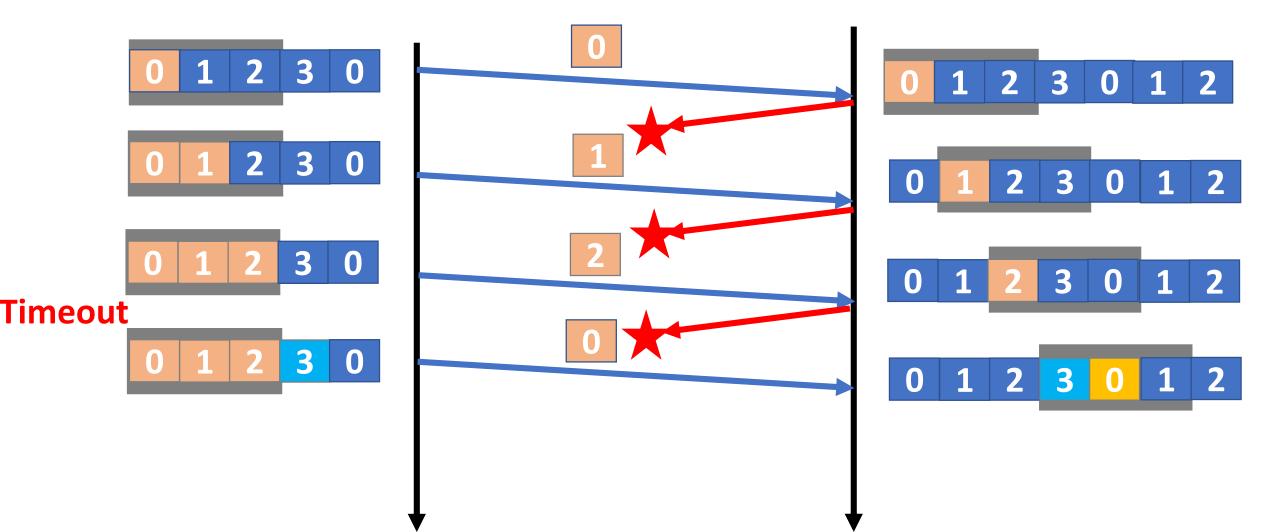
Selective Repeat ARQ



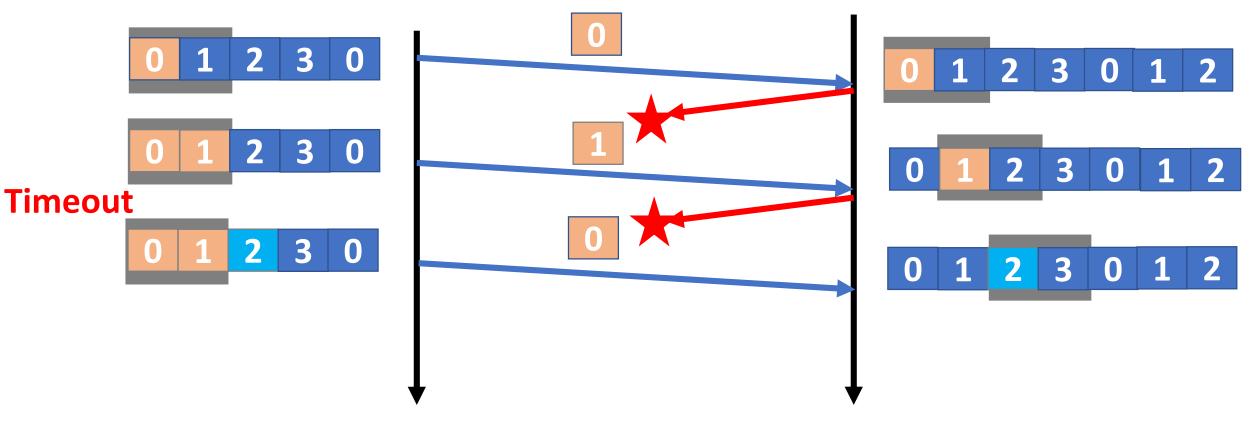
Selective Repeat – A Bound on Window Size

- Maximum Sequence Number (MAX_SEQ): MAX_SEQ+1 distinct sequence numbers are there
 - 0,1,...,MAX_SEQ
- Maximum Number of Outstanding Frames (=Window Size): (MAX_SEQ+1)/2
- Example: Sequence Numbers (0,1,2,...,7) 3 bit sequence numbers, number of outstanding frames (window size) = 4

Selective Repeat — A Bound on Window Size • Let MAX_SEQ = 3, Window Size = 3 [(MAX_SEQ+1)/2+1]



Selective Repeat — A Bound on Window Size • Let MAX_SEQ = 3, Window Size = 2



Discards the wrong frame correctly

Bandwidth Delay Product

- Bandwidth Delay Product (BDP) = Link Bandwidth x Link Delay an important metric for flow control
- Consider Bandwidth = 50 Kbps, one way transit time (delay) = 250 msec
 - BDP 12.5 Kbit
 - Assume 1000 bit segment size; BDP = 12.5 segments
- Consider the event of a segment transmission and the corresponding ACK reception – this takes a round trip time (RTT) – twice the one way latency.
- Maximum number of segments that can be outstanding during this duration = $12.5 \times 2 = 25$ segments

Bandwidth Delay Product – Implication on Window Size

- Maximum number of segments that can be outstanding within this duration = 25 + 1 (as the ACK is sent only when the first segment is received) = 26
 - This gives the maximum link utilization the link will always be busy in transmitting data segments
- Let BD denotes the number of frames equivalent to the BDP, w is the maximum window size
- So, w = 2BD + 1 gives the maximum link utilization this is an important concept to decide the window size for a window based flow control mechanism

Implication of BDP on Protocol Design Choice

- Consider the link bandwidth = 1Mbps, Delay = 1ms
- Consider a network, where segment size is 1 KB (~1000 bytes)
- Which protocol is better for flow control?
 - (a) stop and wait,
 - (b) Go back N,
 - (c) Selective Repeat
- BDP = 1 Mbps x 1ms = 1 Kb (~1000 bits)
- The segment size is eight times larger than the BDP -> the link can not hold an entire segment completely
- Sliding window protocols do not improve performance
- Stop and Wait is better less complexity