6.263/16.37: Lectures 3 & 4

The Data Link Layer: ARQ Protocols

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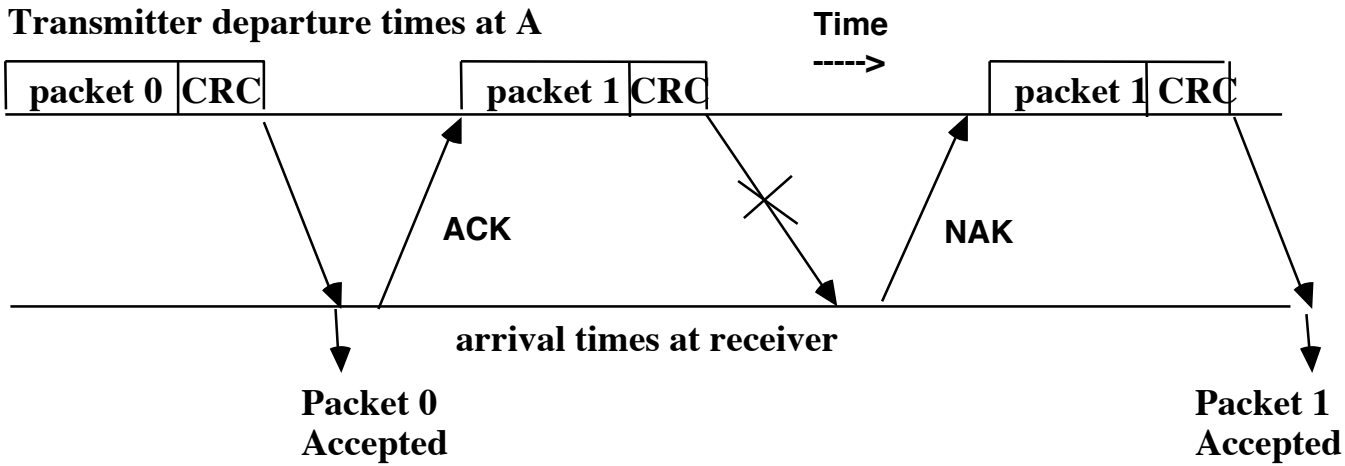
Automatic Repeat ReQuest (ARQ)

- When the receiver detects errors in a packet, how does it let the transmitter know to re-send the corresponding packet?

- Systems which automatically request the retransmission of missing packets or packets with errors are called ARQ systems.

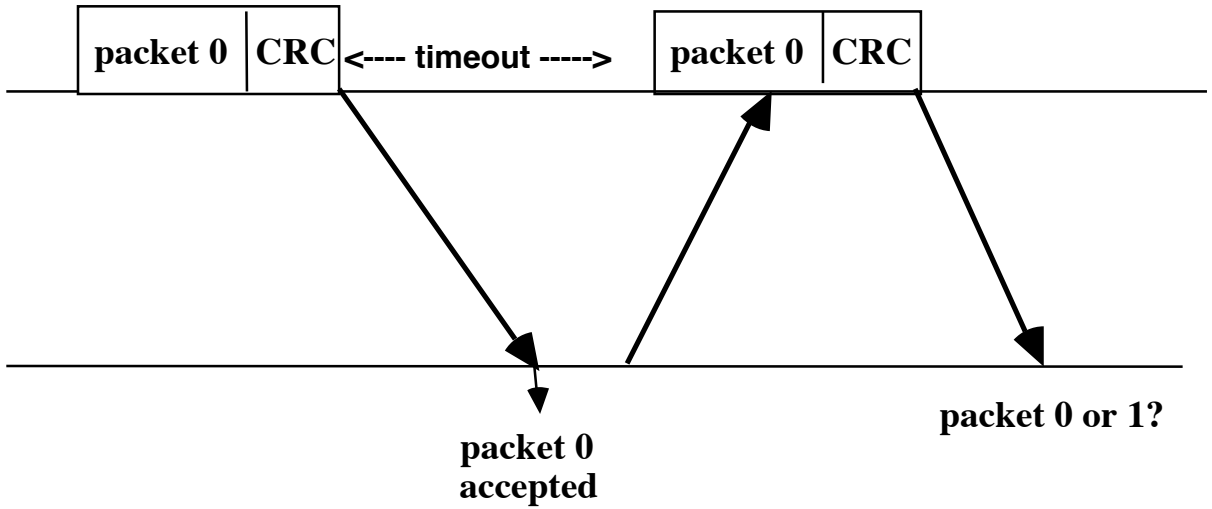
- Three common schemes
  - Stop & Wait
  - Go Back N
  - Selective Repeat

Pure Stop and Wait Protocol



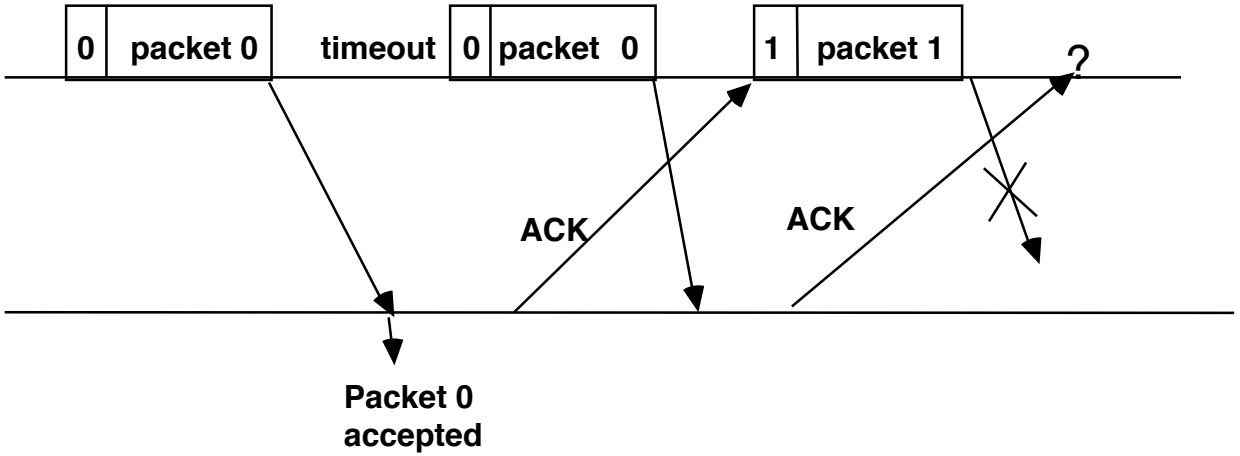
- Problem: Lost Packets
  - Sender will wait forever for an acknowledgement
- Packet may be lost due to framing errors
- Solution: Use time-out (TO)
  - Sender retransmits the packet after a timeout

The Use Of Timeouts For Lost Packets Requires Sequence Numbers



- Problem: Unless packets are numbered the receiver cannot tell which packet it received
- Solution: Use packet numbers (sequence numbers)

Request Numbers Are Required On ACKs To Distinguish Packet ACKed



- REQUEST NUMBERS:
  - Instead of sending "ack" or "nak", the receiver sends the number of the packet currently awaited.
  - Sequence numbers and request numbers can be sent modulo 2.

    This works correctly assuming that

    1) Frames travel in order (FCFS) on links
    2) The CRC never fails to detect errors
    3) The system is correctly initialized.

Stop and Wait Protocol
Algorithm at sender (node A)

(with initial condition SN=0)

1) Accept packet from higher layer when available;
assign number SN to it

2) Transmit packet SN in frame with sequence # SN

3) Wait for an error free frame from B

i. if received and it contains RN>SN in the request # field, set SN to RN and go to 1

ii. if not received within given time, go to 2

Stop and Wait
Algorithm at receiver (node B)

(with initial condition RN=0)

1) Whenever an error-free frame is received from A with a sequence # equal to RN, release received packet to higher layer and increment RN.

2) At arbitrary times, but within bounded delay after receiving any error free frame from A, transmit a frame to A containing RN in the request # field.

Correctness of stop & wait with integer SN, RN

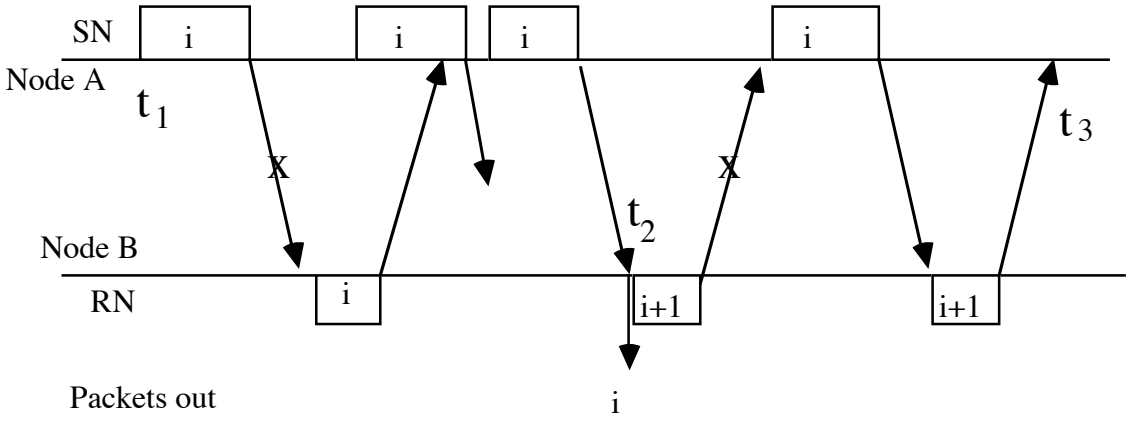
- Assume, for A to (from) B transmission, that
  - All errors are detected as errors
  - Initially no frames are on link, SN=0, RN=0
  - Frames may be arbitrarily delayed or lost
  - Each frame is correctly received with at least some probability $q>0$.
- Split proof of correctness into two parts:
  - SAFETY: show that no packet is ever released out of order or more than once
  - LIVENESS: show that every packet is eventually released

Safety

- No frames on link initially, packet 0 is first packet accepted at A, it is the only packet assigned SN=0, and must be the packet released by B if B ever releases a packet

- Subsequently (using induction) if B has released packets up to and including n-1, then RN is updated to n when n-1 is released, and only n can be released next

LIVENESS



t_1 = time at which A first starts to transmit packet i

t_2 = time at which B correctly receives & releases i, and increases RN to i+1

t_3 = time at which SN is increased to i+1

Will prove that $t_1 < t_2 < t_3 < \infty$. $\Rightarrow$ Liveness

Liveness Argument

- Let SN(t), RN(t) be values of SN and RN at time t

  From the algorithm,

  (1) SN(t) and RN(t) are increasing in t and $SN(t) \leq RN(t)$ for all t
  (2) From safety (since i has not been sent before t_1) $RN(t_1) \leq i$ and $SN(t_1) = i$

- From (1) and (2), $RN(t_1) = SN(t_1) = i$
- RN is incremented at t_2 and SN at t_3, so $t_2 < t_3$
- A transmits i repeatedly up to t_3, and thus to t_2 when it is correctly received. Since $q>0$, t_2 is finite
- B transmits RN=i+1 repeatedly until correctly received at t_3, and $q > 0$ implies that t_3 is finite.

Correctness of Stop & Wait with binary (finite) SN, RN

- Assume that frames travel on link in order

  Note that with integer SN, RN, either

  SN=RN (from t_1 to t_2) or (3)

  SN=RN-1 (from t_2 to t_3) (4)

  Since frames travel in order, the sequence numbers arriving at B and the request numbers arriving at A are increasing, so a single bit can resolve the ambiguity between (3) and (4)

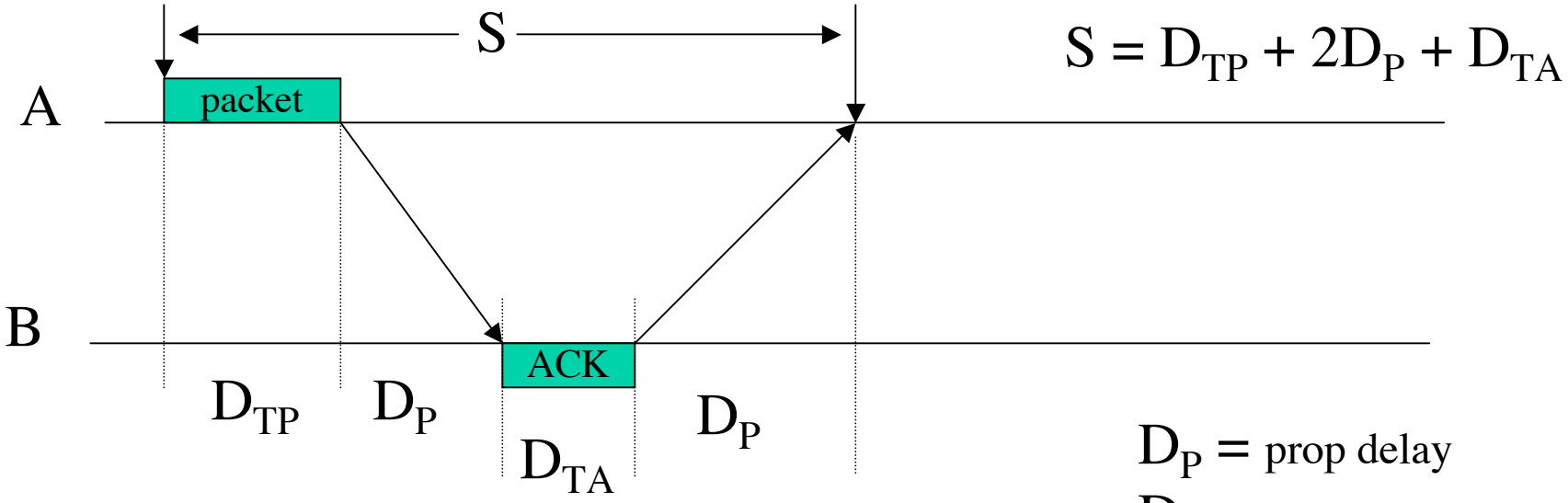
  - RN = 0 and SN = 1 or RN =1 and SN = 0
    => received packet is an old packet
  - RN = 0 and SN = 0 or RN = 1 and SN = 1
    => received packet is new

Efficiency of stop and wait

Let S = time between the transmission of a packet and reception of its ACK

D_{TP} = transmission time of the packet

Efficiency (no errors) = D_{TP}/S



$S = D_{TP} + 2D_P + D_{TA}$

D_P = prop delay

D_{TA} = ACK trans. Time

DTP = packet trans. time

$E = D_{TP}/(D_{TP} + 2D_P + D_{TA})$

Stop and wait in the presence of errors

Let P = the probability of an error in the transmission of a packet or in its acknowledgment

$S = D_{TP} + 2D_P + D_{TA}$

TO = the timeout interval

X = the amount of time that it takes to transmit a packet and receive its ACK. This time accounts for retransmissions due to errors

$E[X] = S + TO*P/(1-P)$, Efficiency = $D_{TP}/E[X]$

Where,

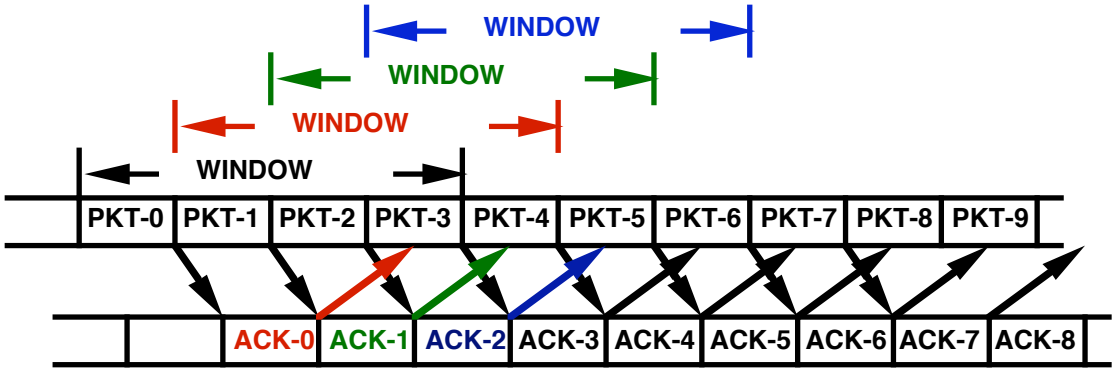
TO = D_{TP} in a full duplex system

TO = S in a half duplex system

Go Back N ARQ (Sliding Window)

- Stop and Wait is inefficient when propagation delay is larger than the packet transmission time
  - Can only send one packet per round-trip time
- Go Back N allows the transmission of new packets before earlier ones are acknowledged

- Go back N uses a window mechanism where the sender can send packets that are within a "window" (range) of packets
  - The window advances as acknowledgements for earlier packets are received

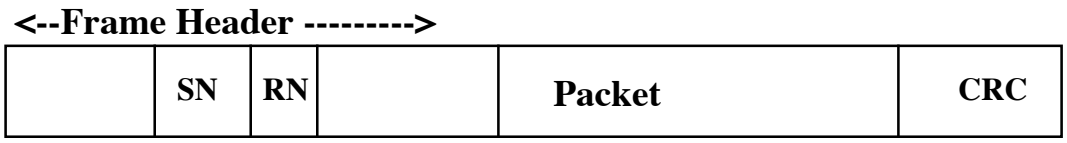


Features of Go Back N

- Window size = N
  - Sender cannot send packet i+N until it has received the ACK for packet i
- Receiver operates just like in Stop and Wait
  - Receive packets in order
  - Receiver cannot accept packet out of sequence
  - Send RN = i + 1 => ACK for all packets up to and including i

- Use of piggybacking
  - When traffic is bi-directional RN's are piggybacked on packets going in the other direction

    Each packet contains a SN field indicating that packet's sequence number and a RN field acknowledging packets in the other direction



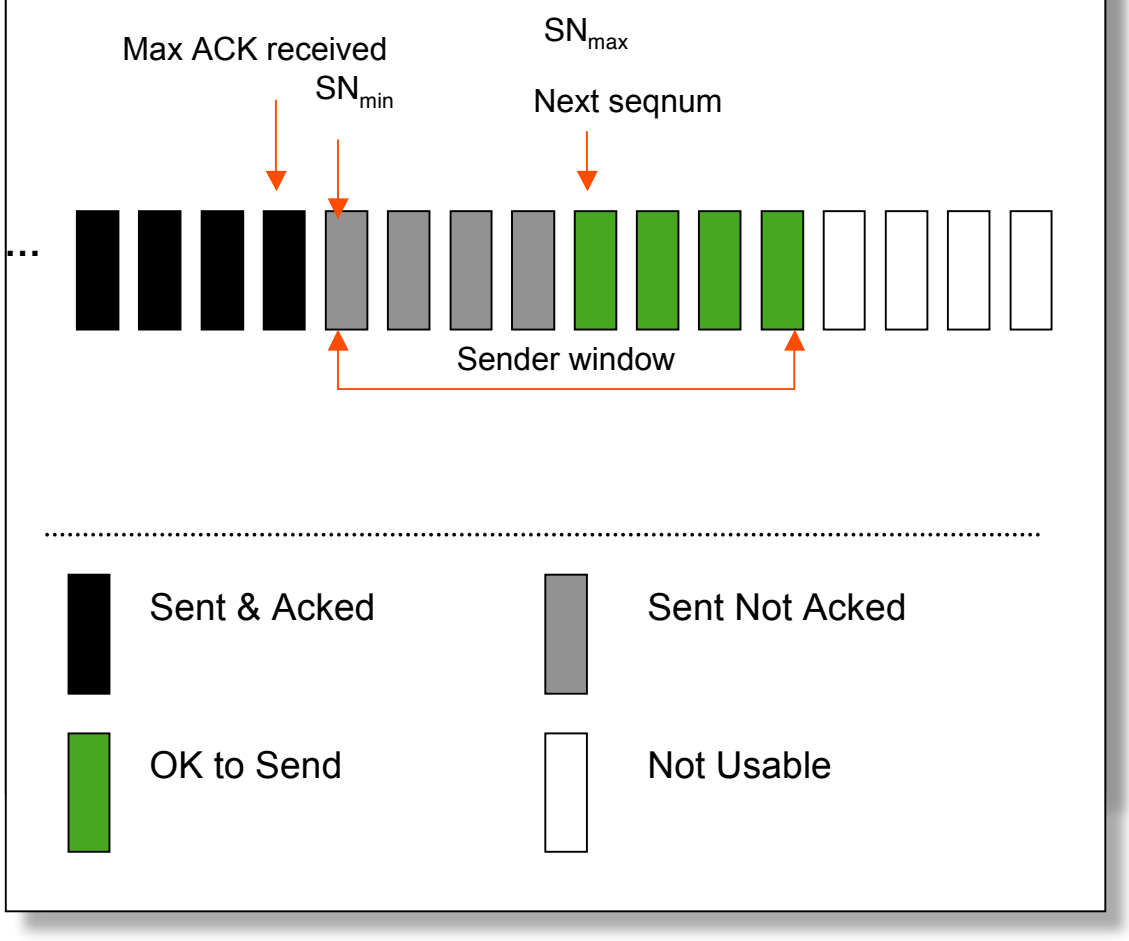
Go Back N ARQ

- The transmitter has a "window" of N packets that can be sent without acknowledgements
- This window ranges from the last value of RN obtained from the receiver (denoted SN_{min}) to SN_{min}+N-1
- When the transmitter reaches the end of its window, or times out, it goes back and retransmits packet SN_{min}

Let SN_{min} be the smallest number packet not yet ACKed

Let SN_{max} be the number of the next packet to be accepted from the higher layer (I.e., the next new packet to be transmitted)

Sender


Max ACK received
SN_{min}
SN_{max}
Next seqnum
Sender window
Sent & Acked
Sent Not Acked
OK to Send
Not Usable

Go Back N: Sender Rules

- $SN_{min} = 0$; $SN_{max} = 0$
- Repeat
  - If $SN_{max} < SN_{min} + N$ (entire window not yet sent)
    Send packet SN_{max} ;
    $SN_{max} = SN_{max} + 1$;
  - If packet arrives from receiver with $RN > SN_{min}$
    $SN_{min} = RN$;
  - If $SN_{min} < SN_{max}$ (there are still some unacknowledged packets) and sender cannot send any new packets
    Choose some packet between SN_{min} and SN_{max} and re-send it
- The last rule says that when you cannot send any new packets you should re-send an old (not yet ACKed) packet
  - There may be two reasons for not being able to send a new packet
    Nothing new from higher layer
    Window expired ($SN_{max} = SN_{min} + N$)
  - No set rule on which packet to re-send
    Least recently sent

Go Back N: Receiver Rules

- RN = 0;
- Repeat
  - When a good packet arrives, if SN = RN

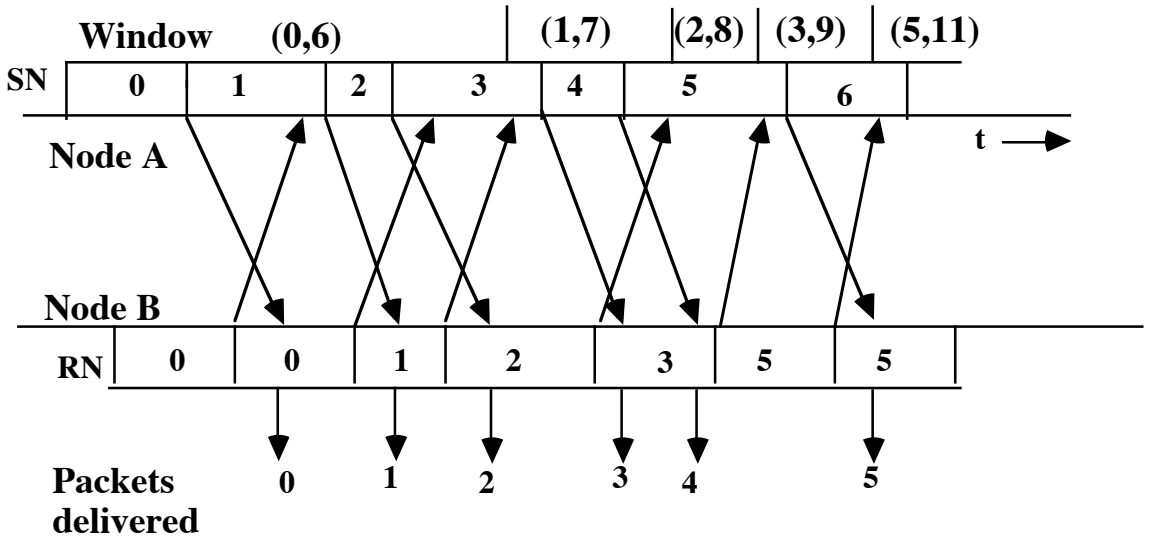
    Accept packet

    Increment RN = RN +1

- At regular intervals send an ACK packet with RN
  - Most DLCs send an ACK whenever they receive a packet from the other direction

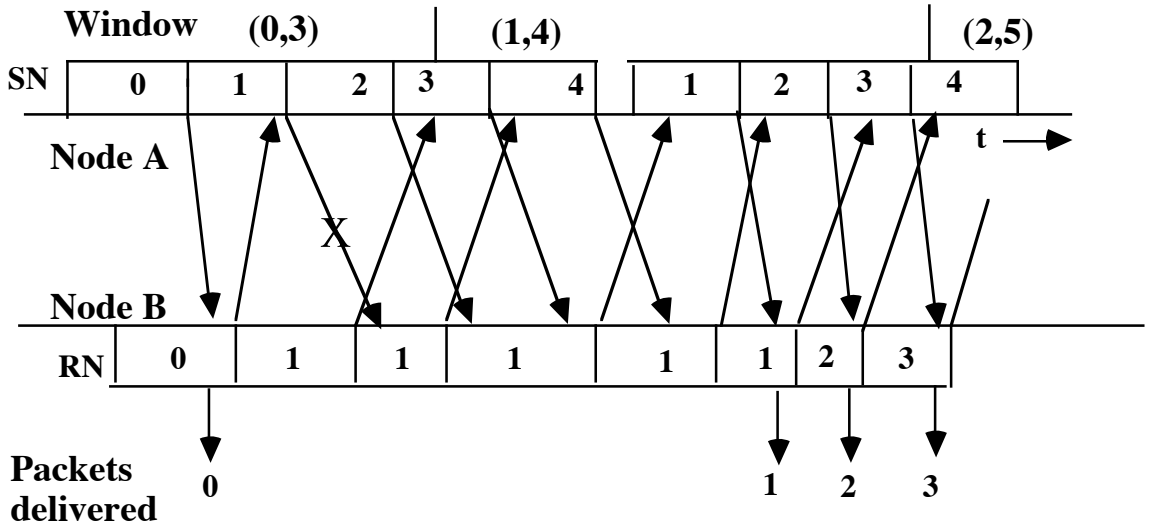
    Delayed ACK for piggybacking
- Receiver reject all packets with SN not equal RN
  - However, those packets may still contain useful RN numbers (see homework assignment)

Example of Go Back 7 ARQ



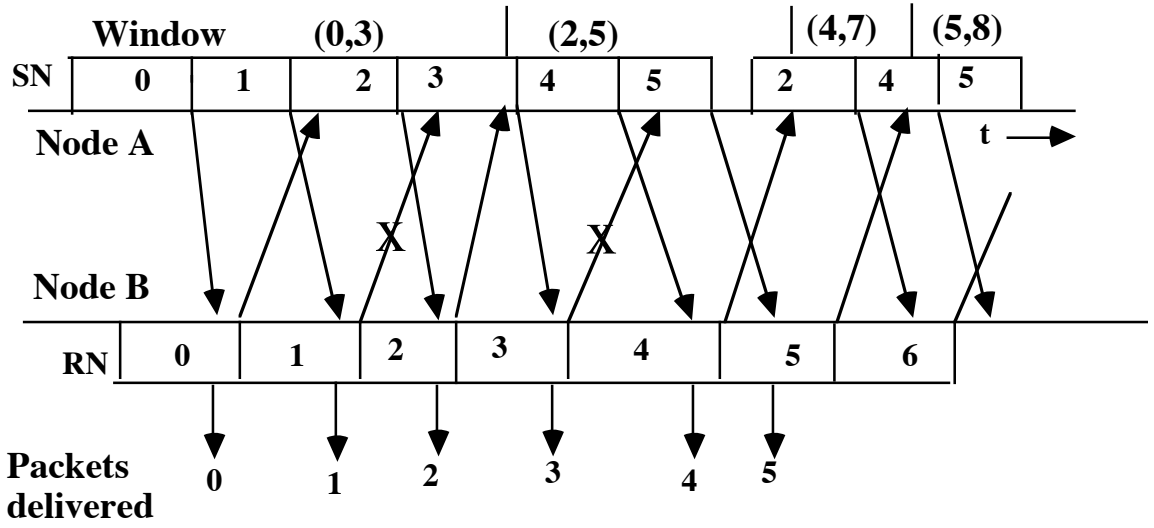
- Note that packet RN-1 must be accepted at B before a frame containing request RN can start transmission at B

RETRANSMISSION BECAUSE OF ERRORS FOR GO BACK 4 ARQ



- Note that the timeout value here is take to be the time to send a full window of packets
- Note that entire window has to be retransmitted after an error

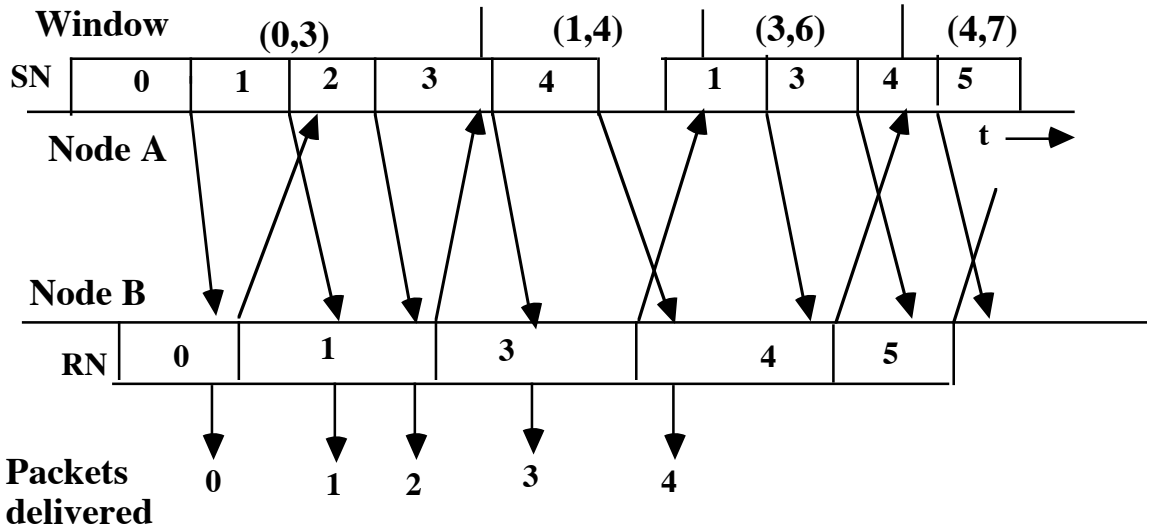
RETRANSMISSION DUE TO FEEDBACK ERRORS FOR GO BACK 4 ARQ



- When an error occurs in the reverse direction the ACK may still arrive in time. This is the case here where the packet from B to A with RN=2 arrives in time to prevent retransmission of packet 0
- Packet 2 is retransmitted because RN = 4 did not arrive in time, however it did arrive in time to prevent retransmission of packet 3
  - Was retransmission of packet 4 and 5 really necessary?

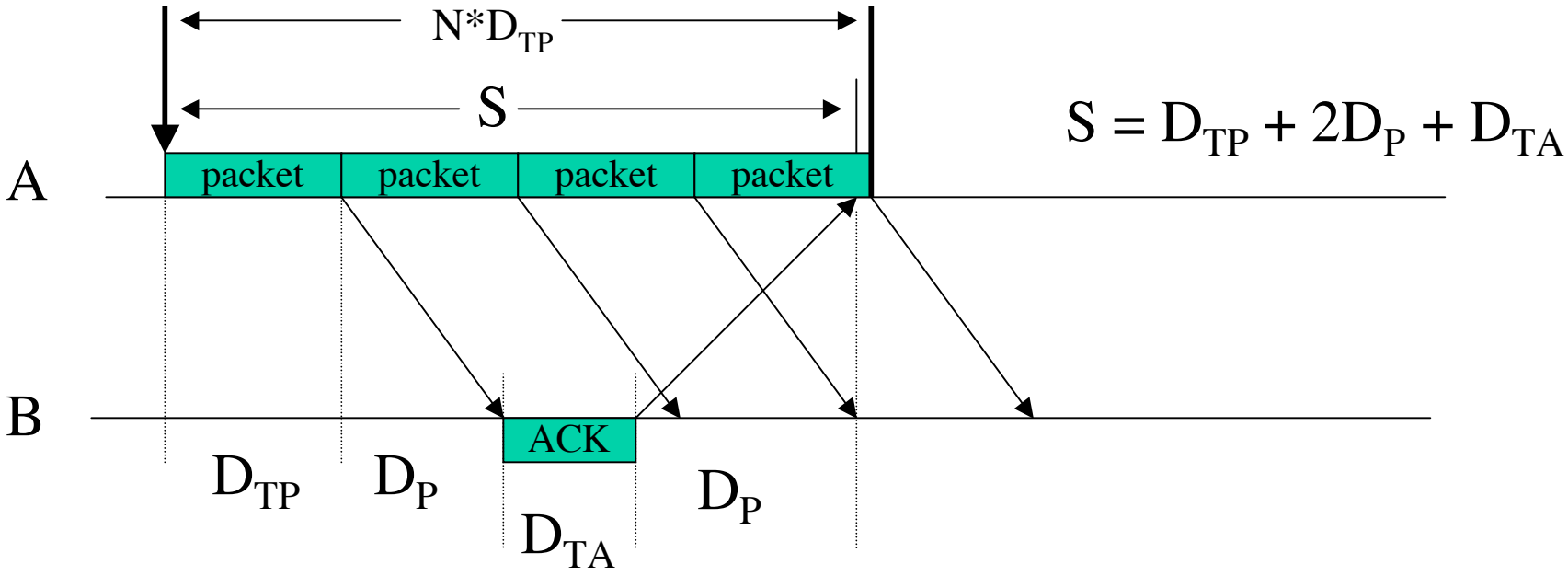
    Strictly no because the window allows transmission of packets 6 and 7 before further retransmissions. However, this is implementation dependent

EFFECT OF LONG FRAMES



- Long frames in feedback direction slow down the ACKs
  - This causes a transmitter with short frames to wait or go back
- Notice again that the retransmission of packets 3 and 4 was not strictly required because the sender could have sent new packets within the window
  - Again, this is implementation dependent

Efficiency of Go Back N



- We want to choose N large enough to allow continuous transmission while waiting for an ACK for the first packet of the window,

$$N > S/ D_{TP}$$

- Without errors the efficiency of Go Back N is,

$$E = \min\{1, N*D_{TP}/S\}$$

Efficiency of Go Back N with transmission errors
Approximate analysis

Assume: $N = \left\lceil \frac{S}{D_{TP}} \right\rceil$ $\qquad\qquad$ $TO = N*D_{TP}$

- When an error occurs the entire window of N packets must be retransmitted

Let X = the number of packets sent per successful transmission

$$E[X] = 1*(1-P) + (X+N)*P$$

$$= 1 + N*P/(1-P)$$

$$\text{Efficiency} = 1/E[X]$$

Go Back N Requirements

- Go Back N is guaranteed to work correctly, independent of the detailed choice of which packets to repeat, if

  1) System is correctly initialized
  2) No failures in detecting errors
  3) Packets travel in FCFS order
  4) Positive probability of correct reception
  5) Transmitter occasionally resends Sn_{min} (e.g., upon timeout)
  6) Receiver occasionally sends RN

Notes on Go Back N

- Requires no buffering of packets at the receiver
- Sender must buffer up to N packets while waiting for their ACK
- Sender must re-send entire window in the event of an error
- Packets can be numbered modulo M where M > N
  - Because at most N packets can be sent simultaneously
- Receiver can only accept packets in order
  - Receiver must deliver packets in order to higher layer
  - Cannot accept packet i+1 before packet i
  - This removes the need for buffering
  - This introduces the need to re-send the entire window upon error

- The major problem with Go Back N is this need to re-send the entire window when an error occurs. This is due to the fact that the receiver can only accept packets in order

Selective Repeat Protocol (SRP)

- Selective Repeat attempts to retransmit only those packets that are actually lost (due to errors)
  - Receiver must be able to accept packets out of order
  - Since receiver must release packets to higher layer in order, the receiver must be able to buffer some packets

- Retransmission requests
  - Implicit

    The receiver acknowledges every good packet, packets that are not ACKed before a time-out are assumed lost or in error

    Notice that this approach must be used to be sure that every packet is eventually received
  - Explicit

    An explicit NAK (selective reject) can request retransmission of just one packet

    This approach can expedite the retransmission but is not strictly needed
  - One or both approaches are used in practice

SRP Rules

- Window protocol just like GO Back N
  - Window size W
- Packets are numbered Mod M where M >= 2W
- Sender can transmit new packets as long as their number is with W of all un-ACKed packets
- Sender retransmit un-ACKed packets after a timeout
  - Or upon a NAK if NAK is employed
- Receiver ACKs all correct packets
- Receiver stores correct packets until they can be delivered in order to the higher layer

Need for buffering

- Sender must buffer all packets until they are ACKed
  - Up to W un-ACKed packet are possible
- Receiver must buffer packets until they can be delivered in order
  - I.e., until all lower numbered packets have been received
  - Needed for orderly delivery of packets to the higher layer
  - Up to W packets may have to be buffered (in the event that the first packet of a window is lost)
- Implication of buffer size = W
  - Number of un-ACKed packets at sender ≤ W

    Buffer limit at sender
  - Number of un-ACKed packets at sender cannot differ by more than W

    Buffer limit at the receiver (need to deliver packets in order)
  - Packets must be numbered modulo M ≥ 2W (using $\log_2(M)$ bits)

EFFICIENCY

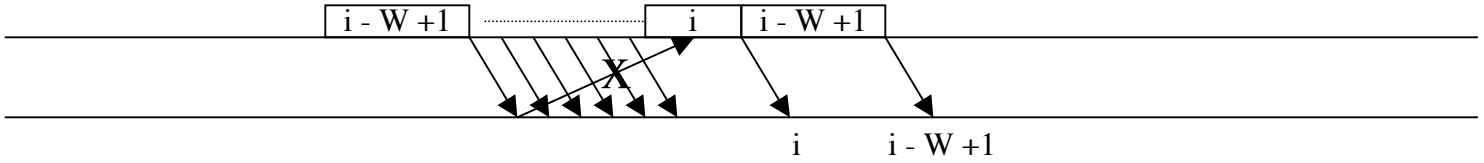
- For ideal SRP, only packets containing errors will be retransmitted
  - Ideal is not realistic because sometimes packets may have to be retransmitted because their window expired. However, if the window size is set to be much larger than the timeout value then this is unlikely
- With ideal SRP, efficiency = 1 - P
  - P = probability of a packet error
- Notice the difference with Go Back N where

$$\text{efficiency (Go Back N)} = 1/(1 + N*P/(1-P))$$

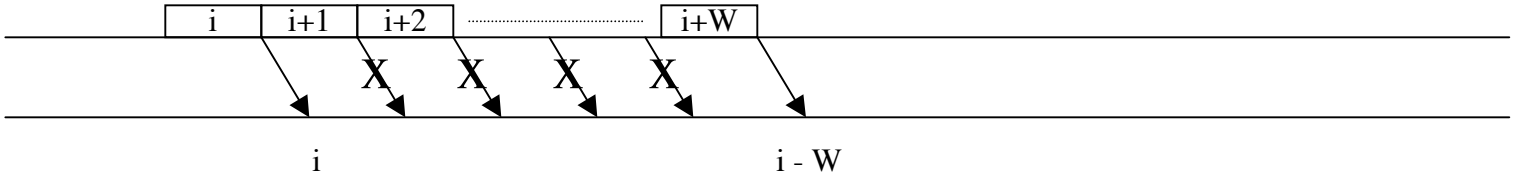
- When the window size is small performance is about the same, however with a large window SRP is much better
  - As transmission rates increase we need larger windows and hence the increased use of SRP

Why are packets numbered Modulo 2W?

- Lets consider the range of packets that may follow packet i at the receiver



Packet i may be followed by the first packet of the window (i -W+1) if it requires retransmission



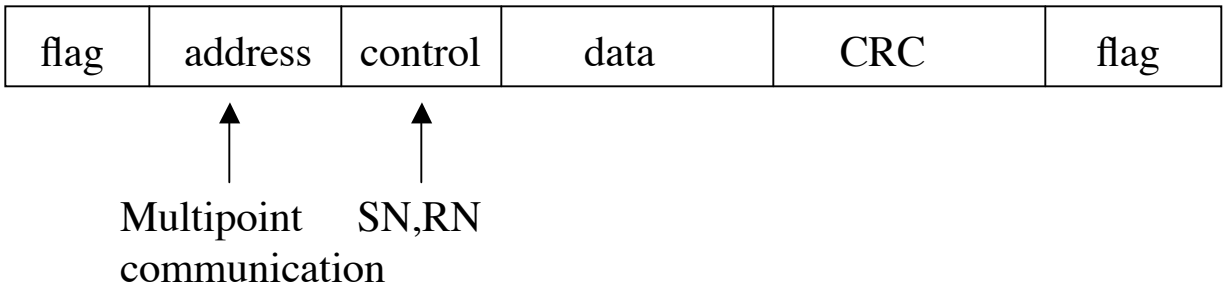
Packet i may be followed by the last packet of the window (i+W) if all Of the ACKs between i and i +W are lost

- Receiver must differentiate between packets i -W+1 ... i +W
  - These 2W packets can be differentiated using Mod 2W numbering

STANDARD DLC's

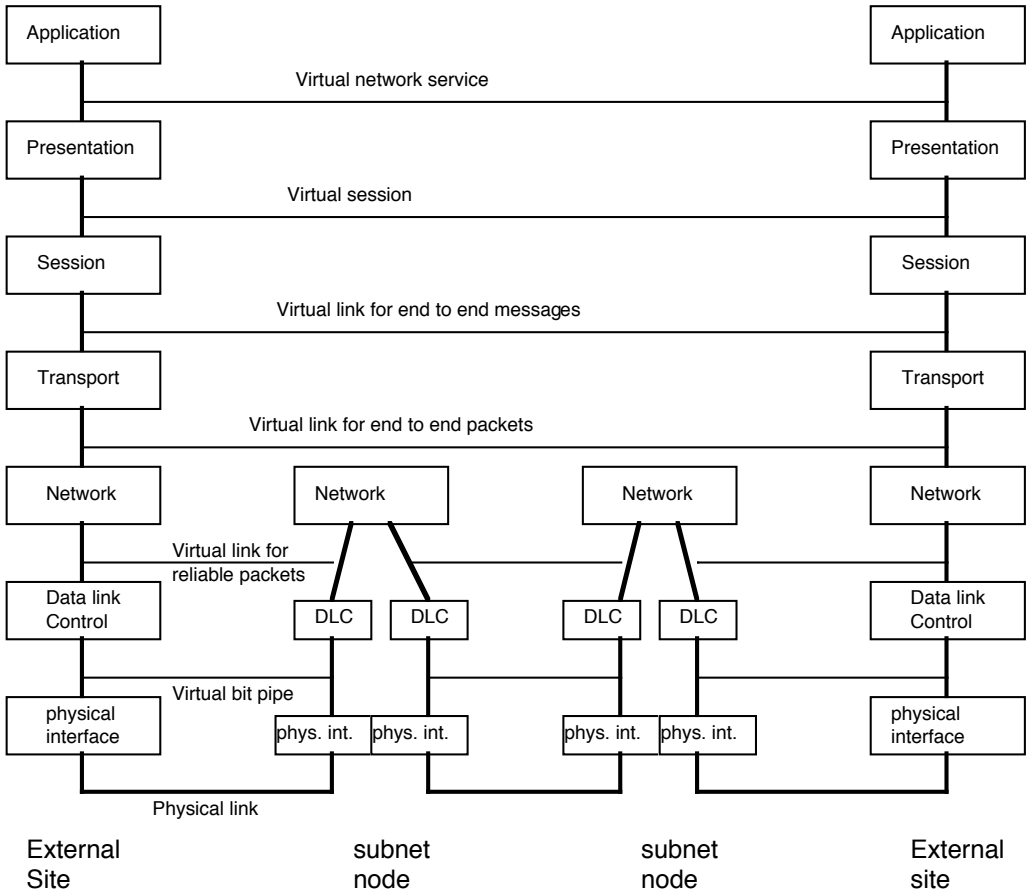
- HDLC, LAPB (X.25), and SDLC are almost the same
  - HDLC/ SDLC developed by IBM for IBM SNA networks
  - LAPB developed for X.25 networks
- They all use bit oriented framing with flag = 01111110
- They all use a 16-bit CRC for error detection
- They all use Go Back N ARQ with N = 7 or 127 (optional)

SDLC packet



- Older protocols (used for modems, e.g., xmodem) used stop and wait and simple checksums

Reliability in the Protocol Stack



FEC/ARQ

ARQ

ARQ

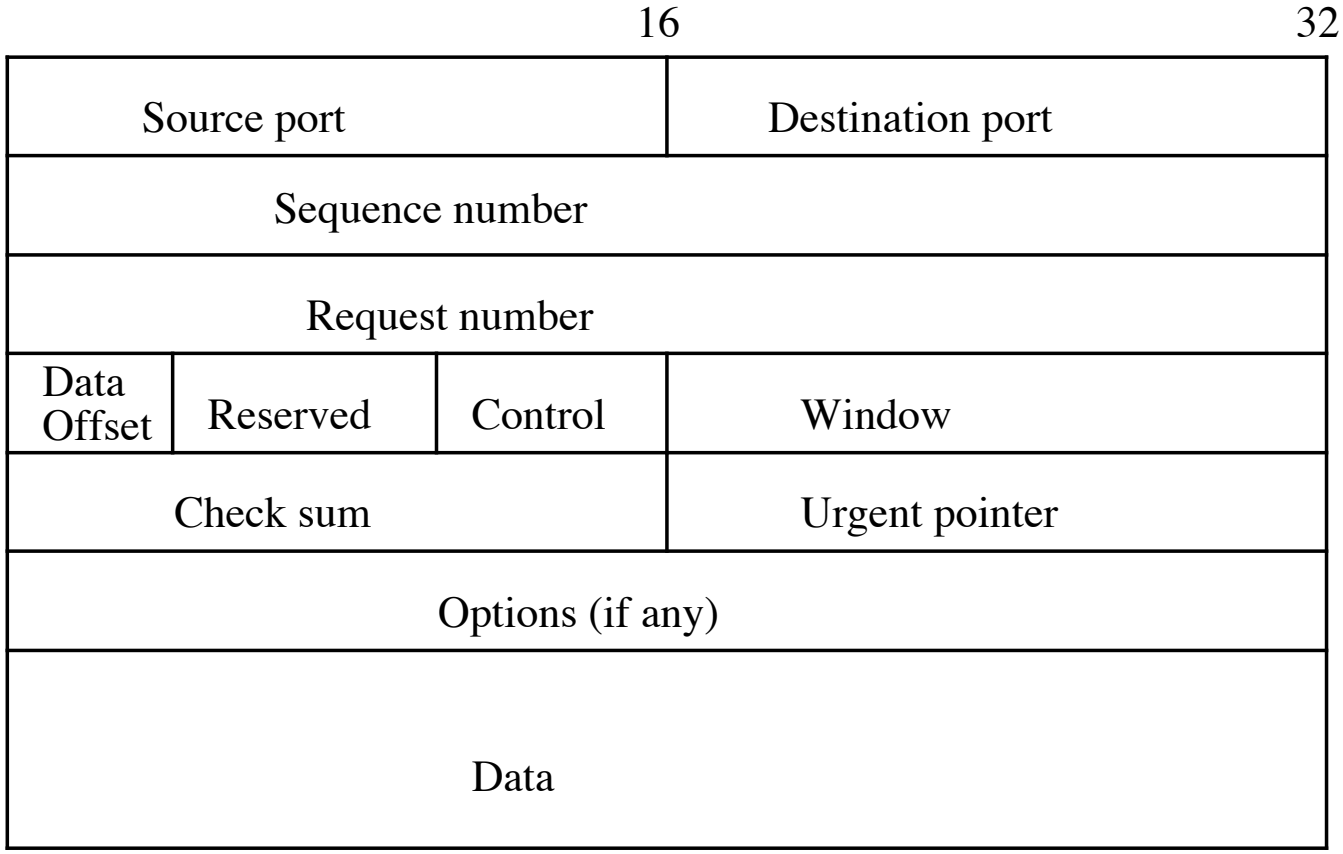
FEC coding

Transmission Control Protocol (TCP)

- Transport layer protocol
  - Reliable transmission of messages
- Connection oriented
  - Stream traffic
  - Must re-sequence out of order IP packets
- Reliable
  - ARQ mechanism
  - Notice that packets have a sequence number and an ack number
  - Notice that packet header has a window size (for Go Back N)

- Flow/congestion control mechanism
  - Limits the size of the window in response to congestion

TCP header fields



TCP error recovery

- Error recovery is done at multiple layers
  - Link, transport, application
- Transport layer error recovery is needed because
  - Packet losses can occur at network layer
    E.g., buffer overflow
  - Some link layers may not be reliable
- SN and RN are used for error recovery in a similar way to Go Back N at the link layer
  - Large SN needed for re-sequencing out of order packets
  - Notice difference from Link Layer ARQ
- TCP uses a timeout mechanism for packet retransmission
  - Timeout calculation
  - Fast retransmission

Retransmissions in TCP: A variation of Go-Back-N

- Sliding window with cumulative acks
  - Receiver can only return a single "ack" sequence number to the sender.
  - Acknowledges all bytes with a lower sequence number
  - Starting point for retransmission
  - Duplicate acks sent when out-of-order packet received
- Sender only retransmits a single packet at a time
  - Optimistic assumption: only one that it knows is lost
  - Network is congested ⇒ shouldn't overload it
- Error control is based on byte sequences, not packets.
  - Retransmitted packet can be different from the original lost packet (e.g., due to fragmentation)

TCP timeout calculation

- Based on round trip time measurement (RTT)
  - Weighted average

$$RTT\_AVE = a*(RTT\_measured) + (1-a)*RTT\_AVE \quad ; \quad a \sim 0.1$$

- Timeout is a multiple of RTT_AVE (usually two)
  - Short Timeout would lead to too many retransmissions
  - Long Timeout would lead to large delays and inefficiency

- In order to make Timeout be more tolerant of delay variations it has been proposed (Jacobson) to set the timeout value based on the standard deviation of RTT

$$Timeout = RTT\_AVE + 4*RTT\_SD$$

- In many TCP implementations the minimum value of Timeout is 500 ms due to the clock granularity

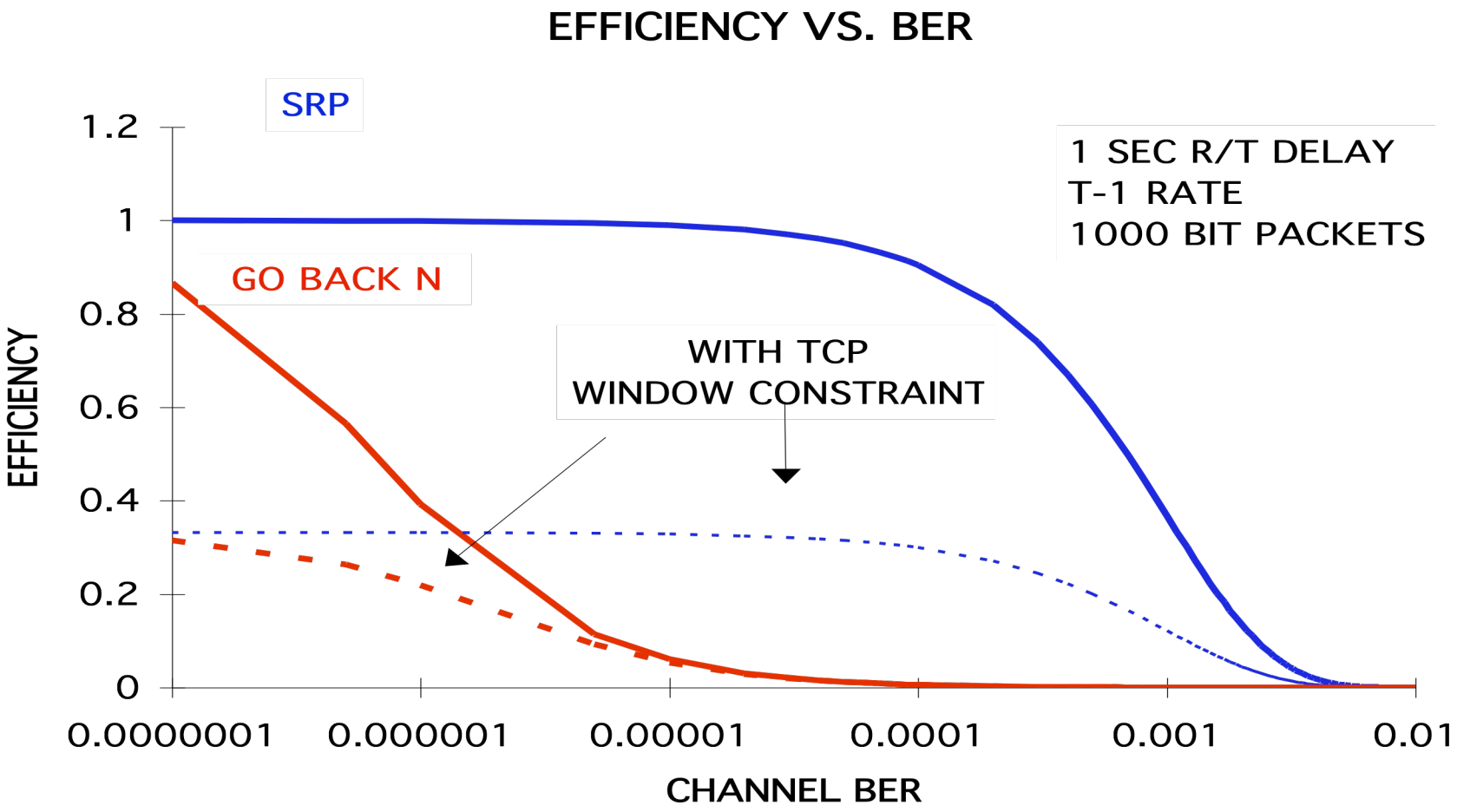
Fast Retransmit

- When TCP receives a packet with a SN that is greater than the expected SN, it sends an ACK packet with a request number of the expected packet SN
  - This could be due to out-of-order delivery or packet loss
- If a packet is lost then duplicate RNs will be sent by TCP until the packet it correctly received
  - But the packet will not be retransmitted until a Timeout occurs
  - This leads to added delay and inefficiency
- Fast retransmit assumes that if 3 duplicate RNs are received by the sending module that the packet was lost
  - After 3 duplicate RNs are received the packet is retransmitted
  - After retransmission, continue to send new data
- Fast retransmit allows TCP retransmission to behave more like Selective repeat ARQ

SACK: A variation on SRP ARQ

- Option for selective ACKs (SACK) also widely deployed
- Selective acknowledgement (SACK) essentially adds a bitmask of packets received
  - Implemented as a TCP option (extended TCP header)
  - Encoded as a set of received byte ranges (max of 4 ranges/often max of 3)
- When to retransmit?
  - Packets may experience different delays
  - Still need to deal with reordering
  - Wait for out of order by 3pkts

TCP Error Control



- Original TCP designed for low BER, low delay links
- New implementations (RFC 1323) allow for larger windows and selective retransmissions