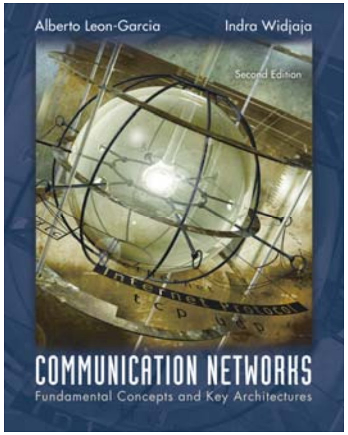
# Chapter 5 Peer-to-Peer Protocols and Data Link Layer




PART I: Peer-to-Peer Protocols

Peer-to-Peer Protocols and Service Models

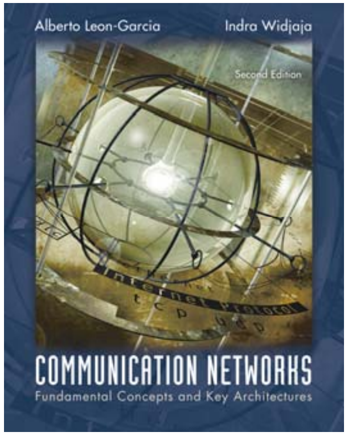
ARQ Protocols and Reliable Data Transfer

Flow Control

Timing Recovery

TCP Reliable Stream Service & Flow Control

# Chapter 5
# Peer-to-Peer Protocols and Data Link Layer




PART II: Data Link Controls

Framing

Point-to-Point Protocol

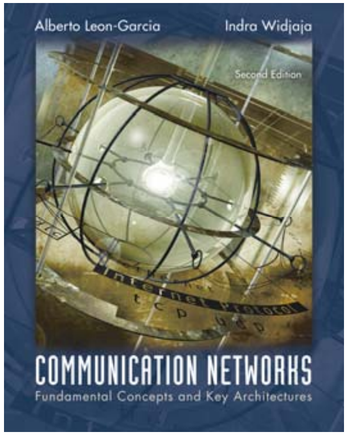
High-Level Data Link Control

Link Sharing Using Statistical Multiplexing

# Chapter Overview

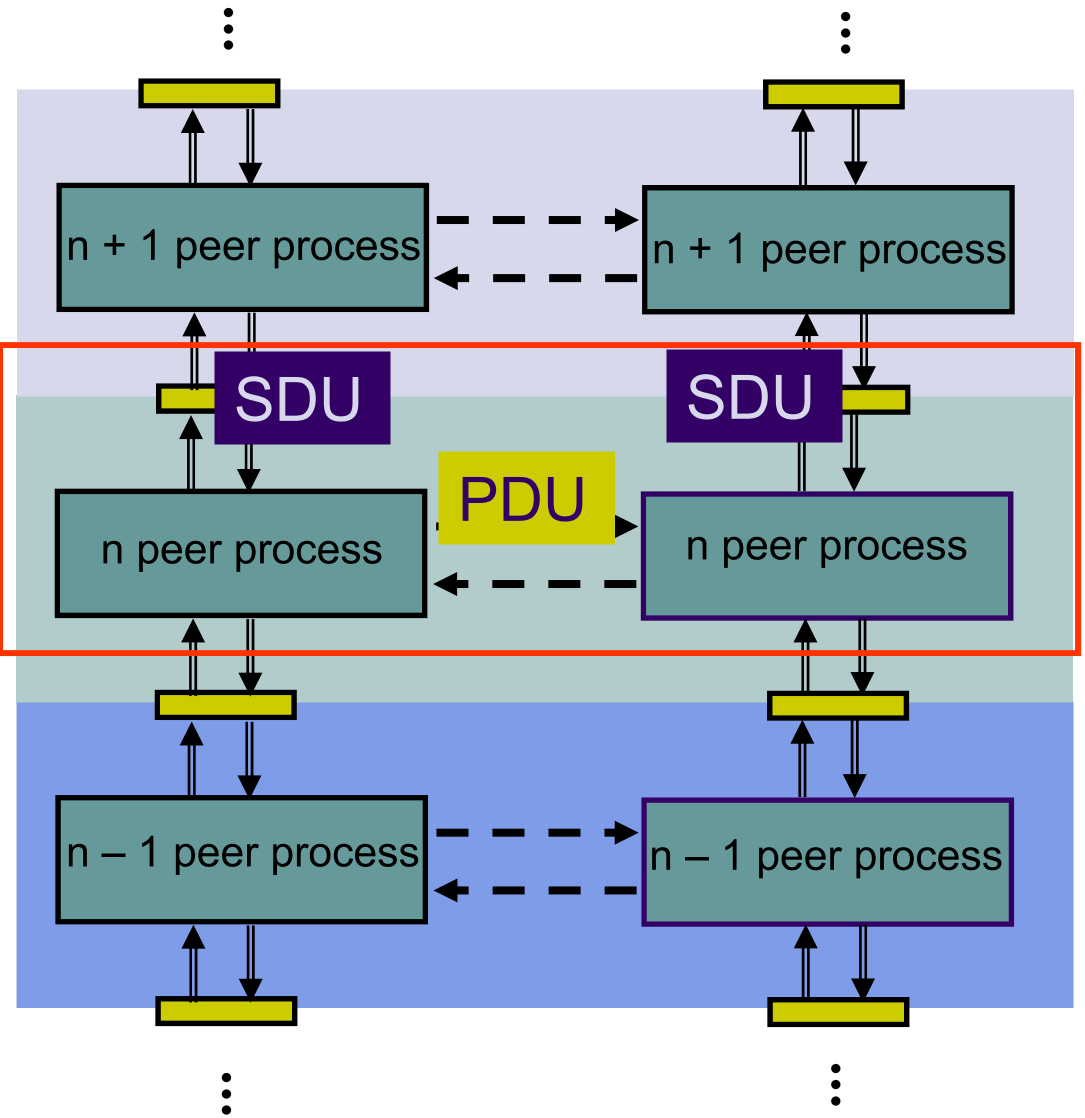
- Peer-to-Peer protocols:  many protocols involve the interaction between two peers
  - Service Models are discussed & examples given
  - Detailed discussion of ARQ provides example of development of peer-to-peer protocols
  - Flow control, TCP reliable stream, and timing recovery
- Data Link Layer
  - Framing
  - PPP & HDLC protocols
  - Statistical multiplexing for link sharing

# Chapter 5
# Peer-to-Peer Protocols and Data Link Layer




## *Peer-to-Peer Protocols and Service Models*

# Peer-to-Peer Protocols



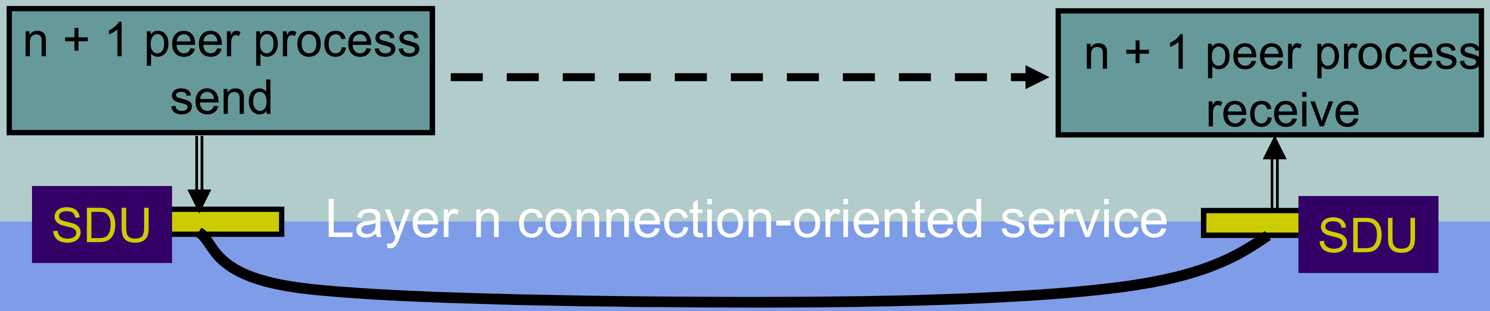

- *Peer-to-Peer processes* execute layer-n protocol to provide service to layer-(n+1)
- Layer-(n+1) peer calls layer-n and passes Service Data Units (SDUs) for transfer
- Layer-n peers exchange Protocol Data Units (PDUs) to effect transfer
- Layer-n delivers SDUs to destination layer-(n+1) peer

# Service Models

- The *service model* specifies the information transfer service layer-n provides to layer-(n+1)
- The most important distinction is whether the service is:
  - Connection-oriented
  - Connectionless
- Service model possible features:
  - Arbitrary message size or structure
  - Sequencing and Reliability
  - Timing, Pacing, and Flow control
  - Multiplexing
  - Privacy, integrity, and authentication

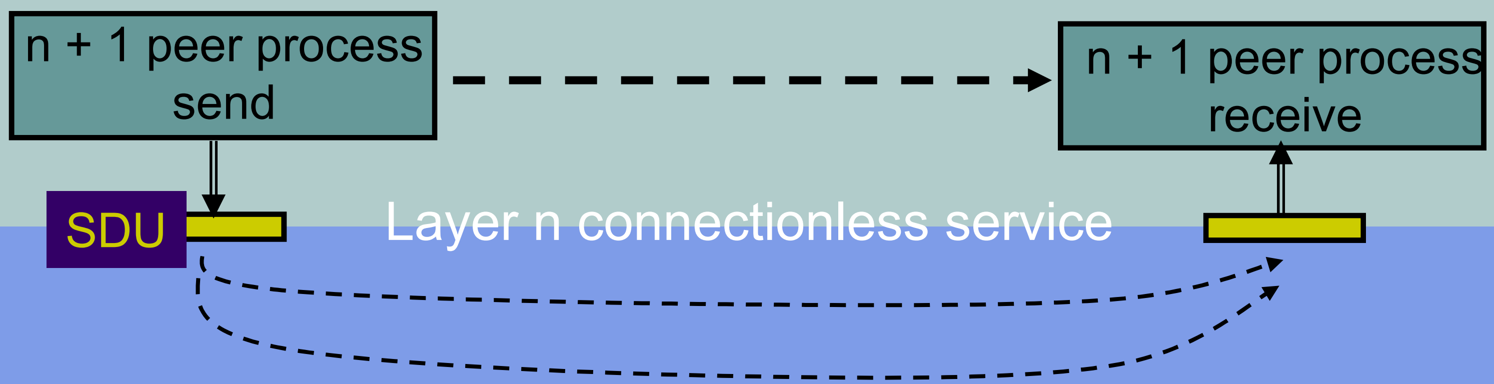
# Connection-Oriented Transfer Service

- Connection Establishment
  - Connection must be established between layer-(n+1) peers
  - Layer-n protocol must: Set initial parameters, e.g. sequence numbers; and Allocate resources, e.g. buffers
- Message transfer phase
  - Exchange of SDUs
- Disconnect phase
- Example: TCP, PPP



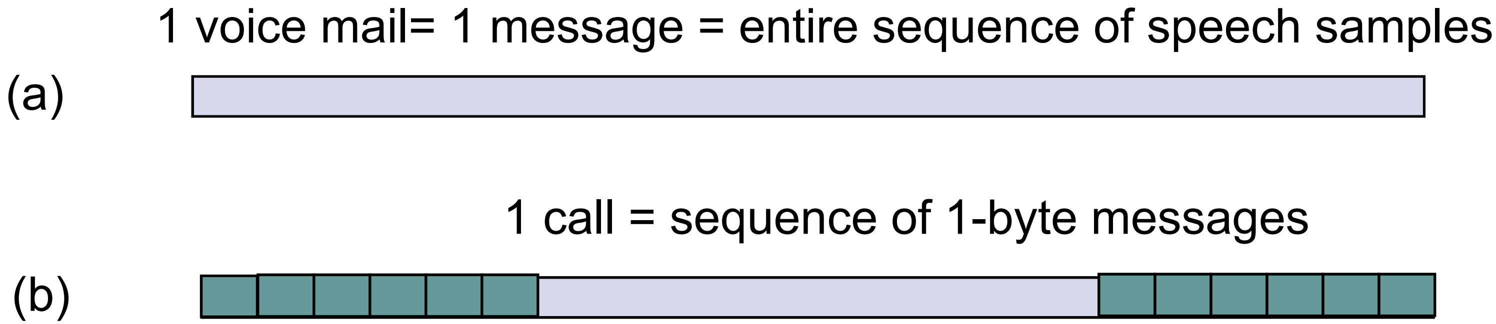
# Connectionless Transfer Service

- No Connection setup, simply send SDU
- Each message send independently
- Must provide all address information per message
- Simple & quick
- Example:  UDP, IP



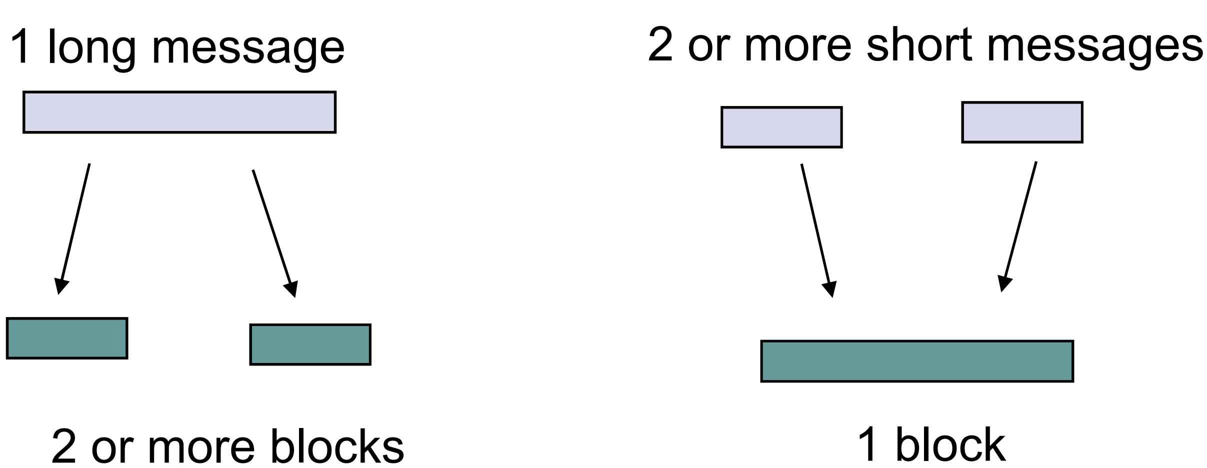
# Message Size and Structure

- What message size and structure will a service model accept?
  - Different services impose restrictions on size & structure of data it will transfer
  - Single bit?  Block of bytes?  Byte stream?
  - Ex: Transfer of voice mail = 1 long message
  - Ex: Transfer of voice call = byte stream



# Segmentation & Blocking

- To accommodate arbitrary message size, a layer may have to deal with messages that are too long or too short for its protocol
- *Segmentation & Reassembly*: a layer breaks long messages into smaller blocks and reassembles these at the destination
- *Blocking & Unblocking*: a layer combines small messages into bigger blocks prior to transfer



# Reliability & Sequencing

- *Reliability*: Are messages or information stream delivered error-free and without loss or duplication?
- *Sequencing*: Are messages or information stream delivered in order?
- *ARQ protocols* combine error detection, retransmission, and sequence numbering to provide reliability & sequencing
- Examples: TCP and HDLC

# Pacing and Flow Control

- Messages can be lost if receiving system does not have sufficient buffering to store arriving messages
- If destination layer-(n+1) does not retrieve its information fast enough, destination layer-n buffers may overflow
- *Pacing & Flow Control* provide backpressure mechanisms that control transfer according to availability of buffers at the destination
- Examples:  TCP and HDLC

# Timing

- Applications involving voice and video generate units of information that are related temporally
- Destination application must reconstruct temporal relation in voice/video units
- Network transfer introduces delay & jitter
- *Timing Recovery* protocols use *timestamps* & *sequence numbering* to control the delay & jitter in delivered information
- Examples:  RTP & associated protocols in Voice over IP

# Multiplexing

- *Multiplexing* enables multiple layer-(n+1) users to share a layer-n service
- A multiplexing tag is required to identify specific users at the destination
- Examples: UDP, IP

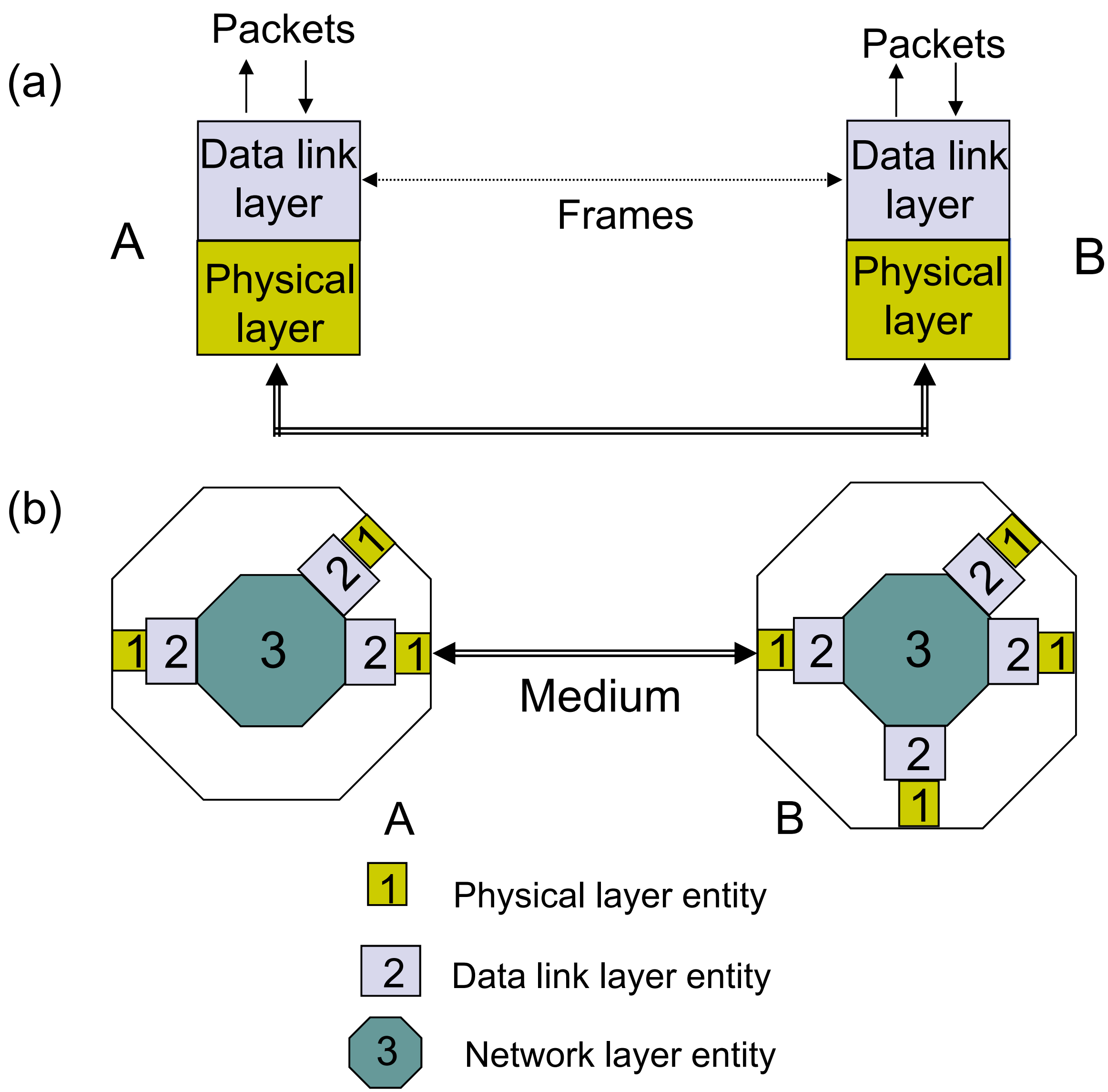
# Privacy, Integrity, & Authentication

- *Privacy*: ensuring that information transferred cannot be read by others
- *Integrity*: ensuring that information is not altered during transfer
- *Authentication*: verifying that sender and/or receiver are who they claim to be
- *Security protocols* provide these services and are discussed in Chapter 11
- Examples: IPSec, SSL

# End-to-End vs. Hop-by-Hop

- A service feature can be provided by implementing a protocol
  - end-to-end across the network
  - across every hop in the network
- Example:
  - Perform error control at every hop in the network or only between the source and destination?
  - Perform flow control between every hop in the network or only between source & destination?
- We next consider the tradeoffs between the two approaches

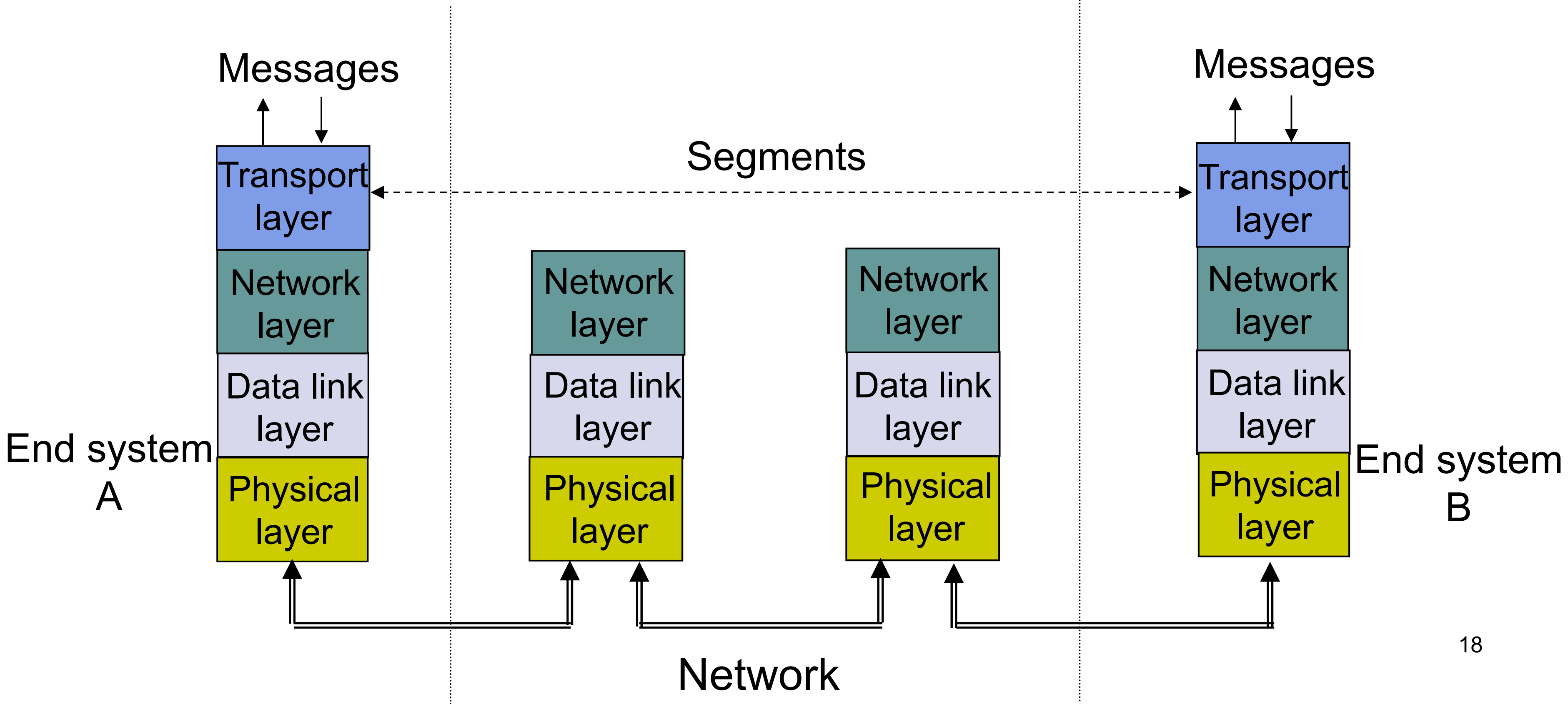
# Error control in Data Link Layer



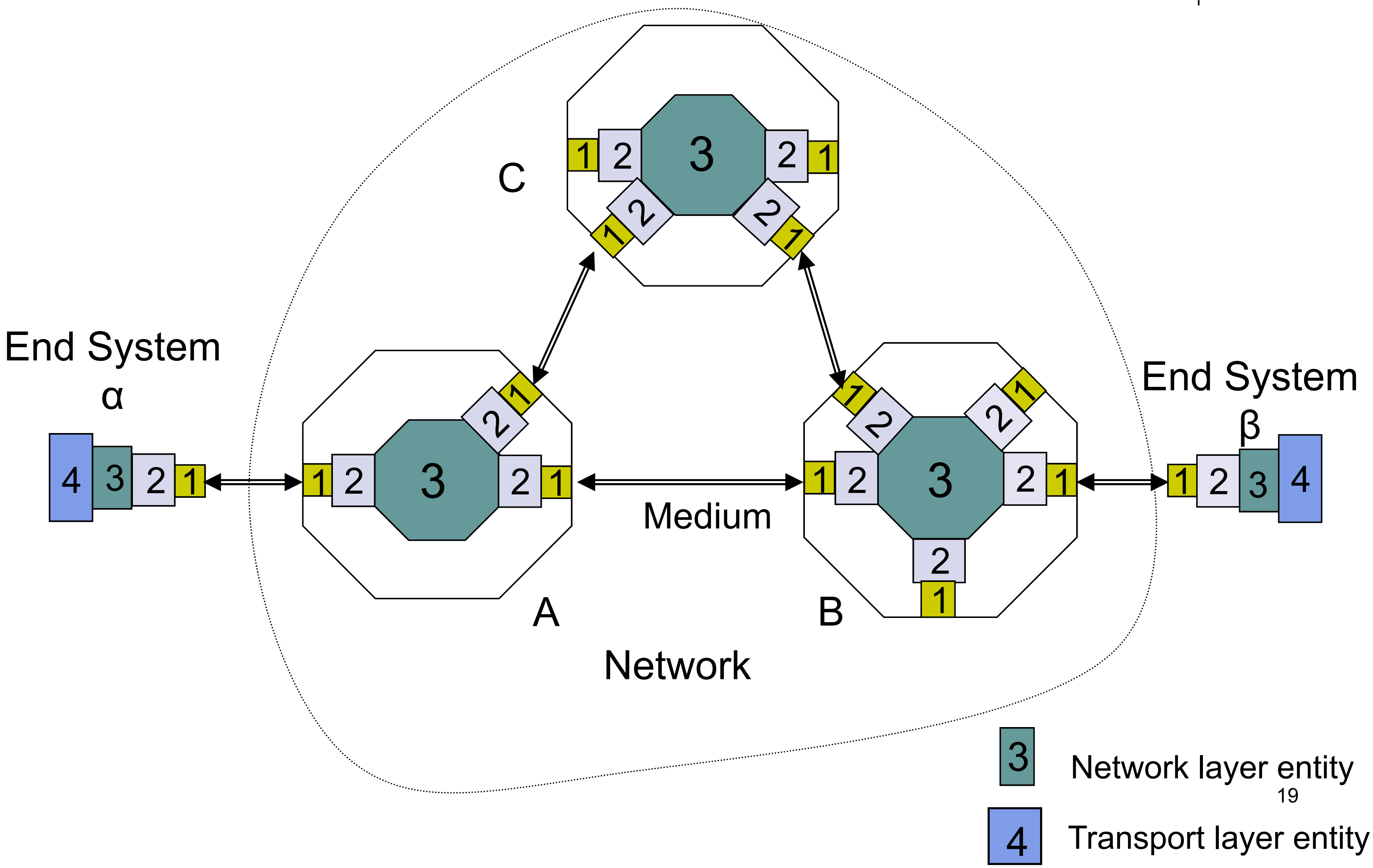

- Data Link operates over wire-like, directly-connected systems
- Frames can be corrupted or lost, but arrive in order
- Data link performs error-checking & retransmission
- Ensures error-free packet transfer between two systems

# Error Control in Transport Layer

- Transport layer protocol (e.g. TCP) sends segments across network and performs end-to-end error checking & retransmission
- Underlying network is assumed to be unreliable

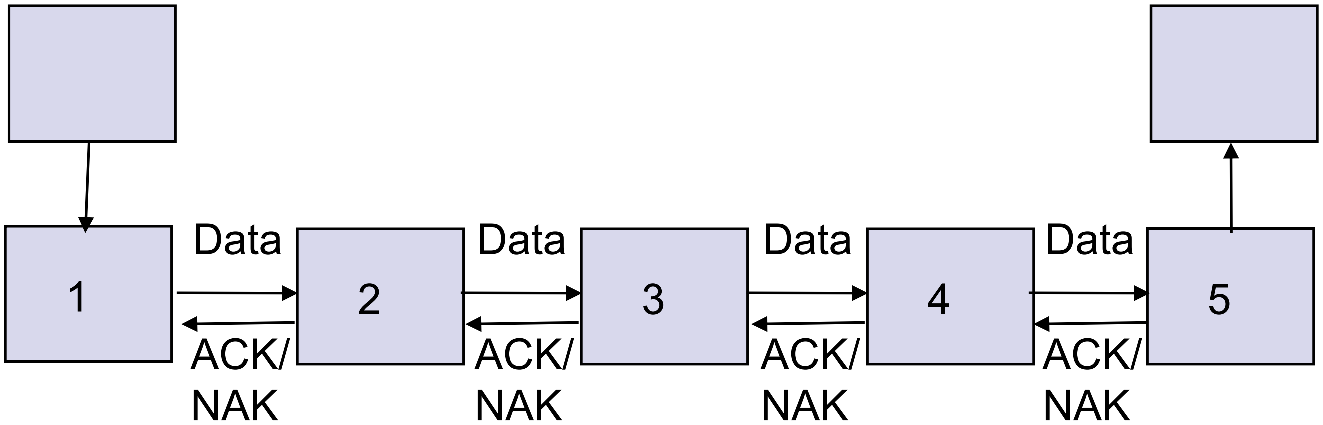


- Segments can experience long delays, can be lost, or arrive out-of-order because packets can follow different paths across network
- End-to-end error control protocol more difficult



# End-to-End Approach Preferred

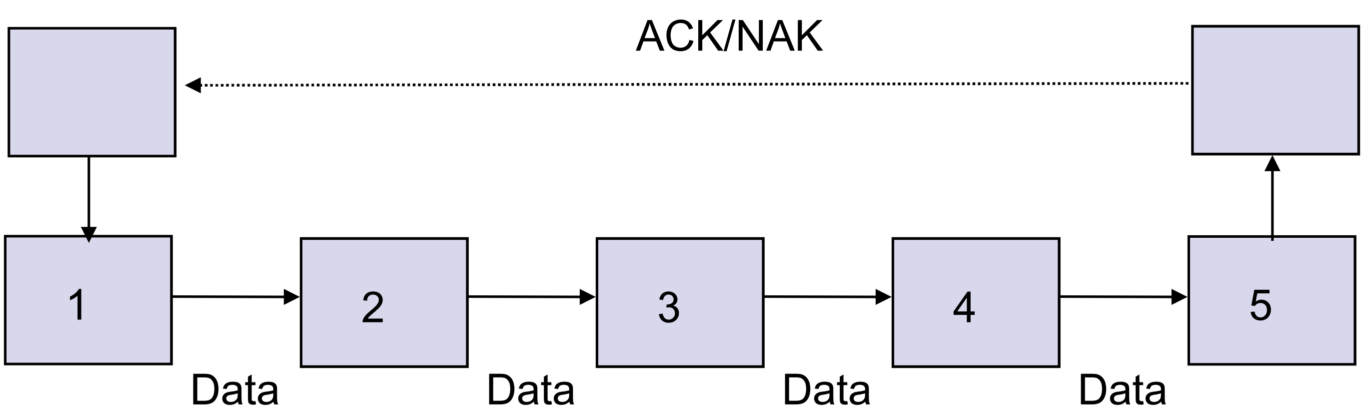
Hop-by-hop




Hop-by-hop cannot ensure E2E correctness

Faster recovery

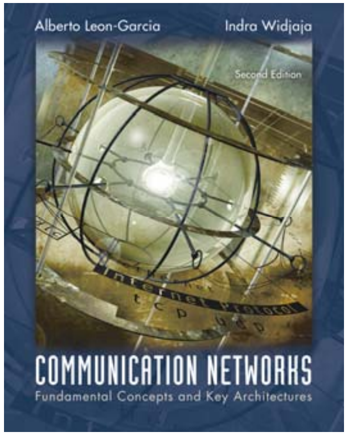
End-to-end




Simple inside the network

More scalable if complexity at the edge

# Chapter 5
# Peer-to-Peer Protocols and Data Link Layer



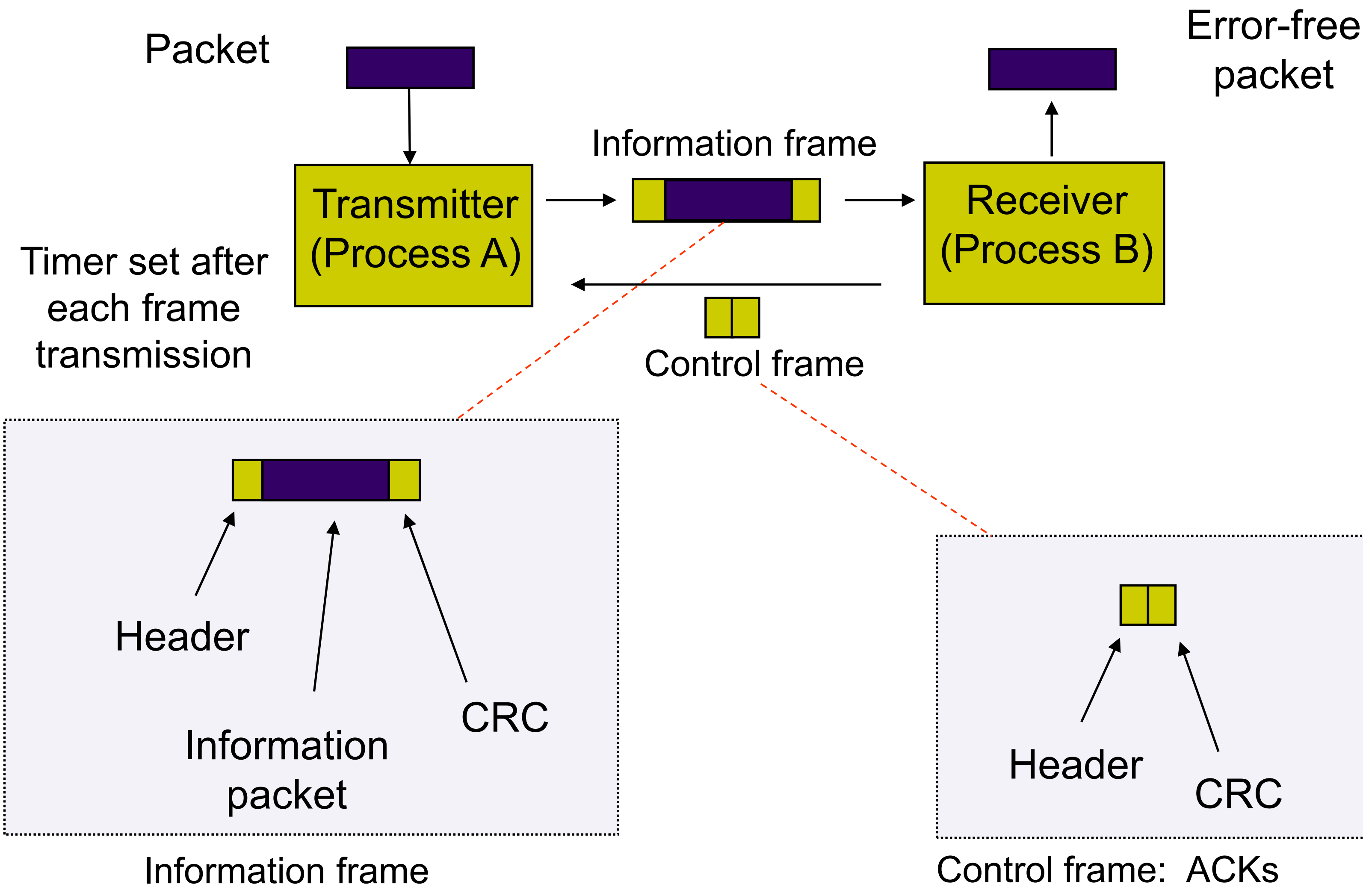

## *ARQ Protocols and Reliable Data Transfer*

# Automatic Repeat Request (ARQ)

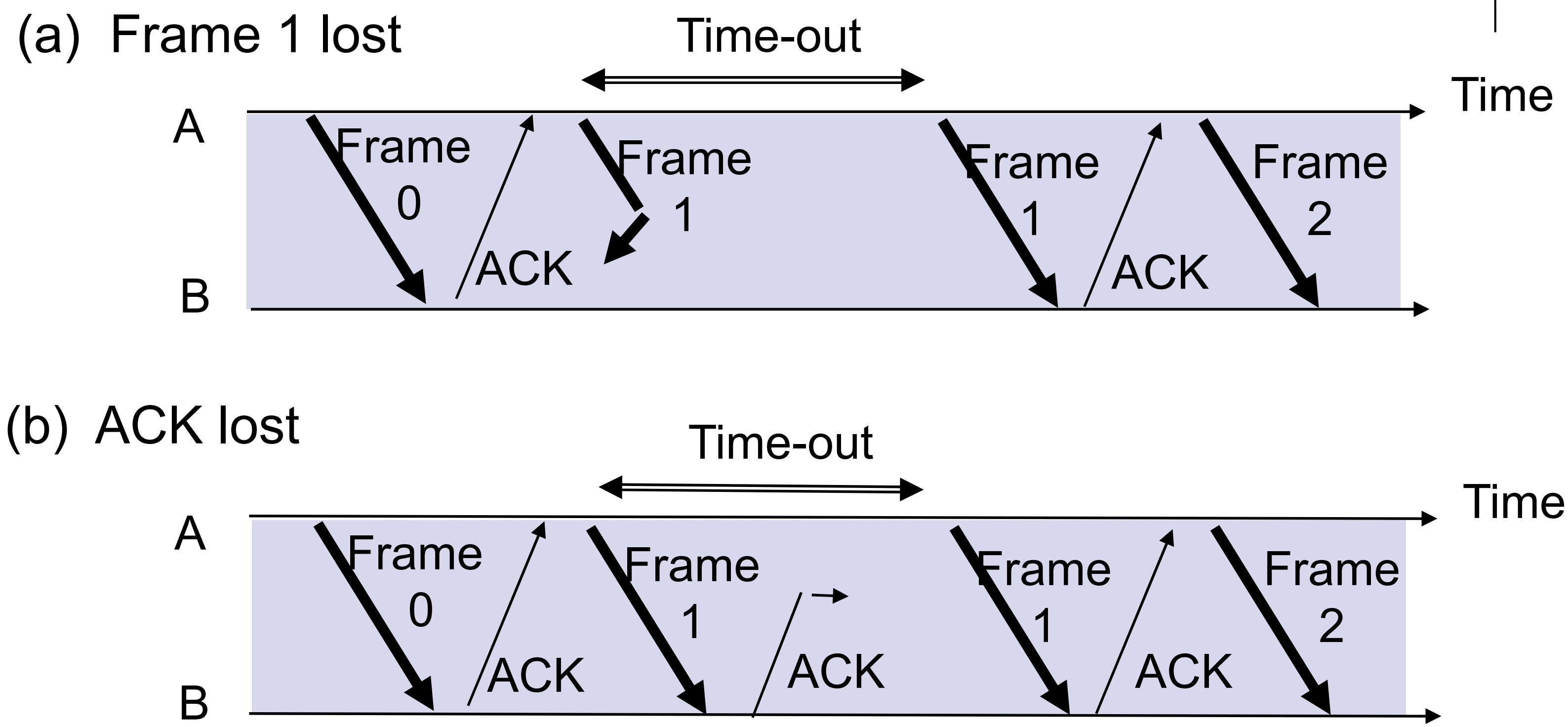
- *Purpose*: to ensure a sequence of information packets is delivered in order and without errors or duplications despite transmission errors & losses
- We will look at:
  - Stop-and-Wait ARQ
  - Go-Back N ARQ
  - Selective Repeat ARQ
- Basic elements of ARQ:
  - *Error-detecting code* with high error coverage
  - *ACKs* (positive acknowledgments
  - *NAKs* (negative acknowlegments)
  - *Timeout mechanism*

# Stop-and-Wait ARQ

Transmit a frame, wait for ACK



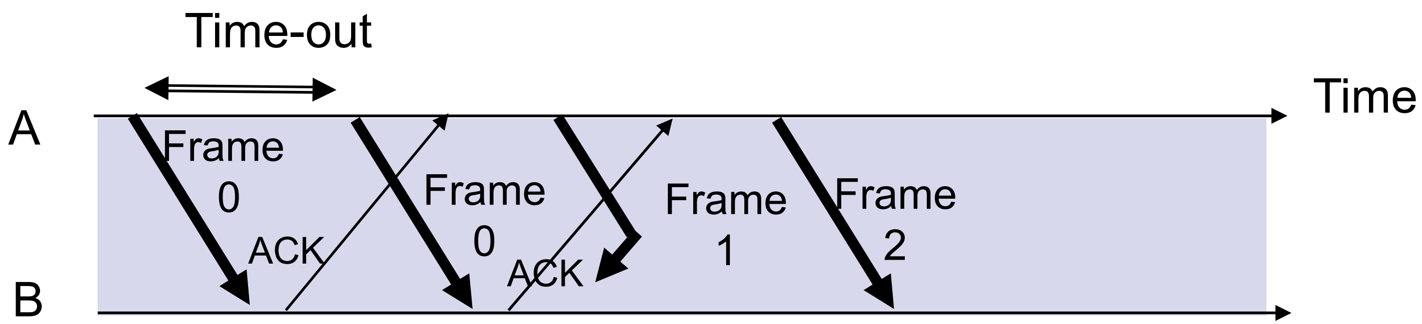
# Need for Sequence Numbers




- In cases (a) & (b) the transmitting station A acts the same way
- But in case (b) the receiving station B accepts frame 1 twice
- Question: How is the receiver to know the second frame is also frame 1?
- Answer: ***Add frame sequence number in header***
- $S_{last}$ is sequence number of most recent transmitted frame

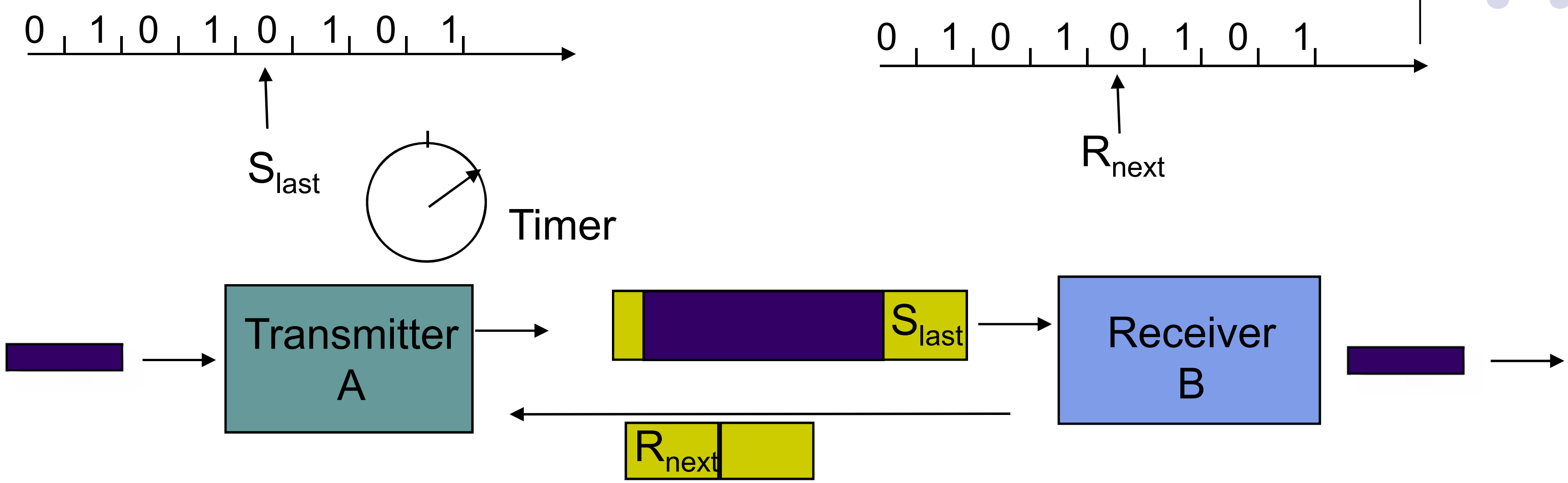
# Sequence Numbers

(c) Premature Time-out

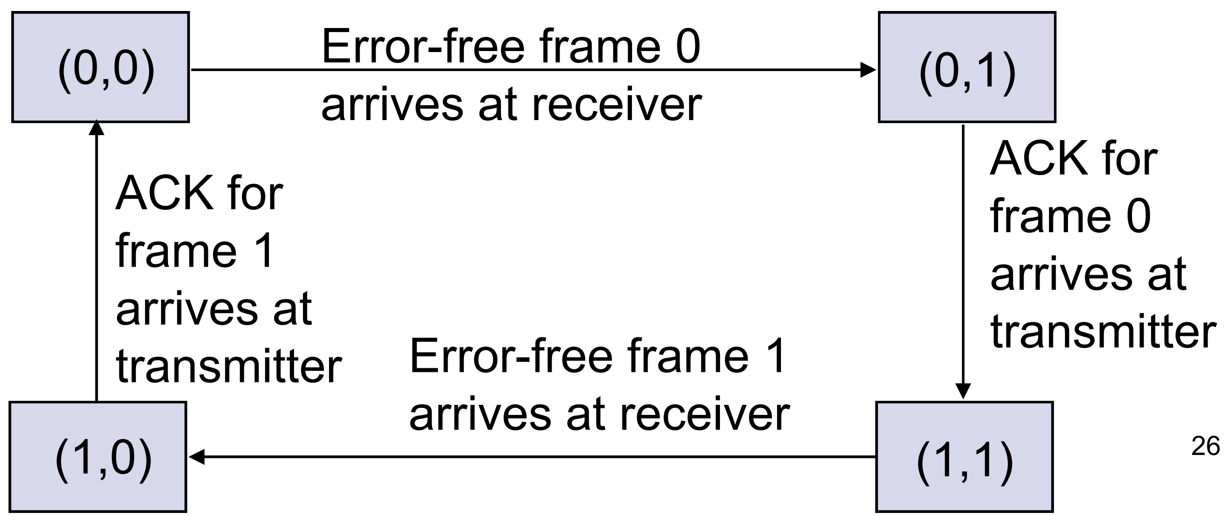



- The transmitting station A misinterprets duplicate ACKs
- Incorrectly assumes second ACK acknowledges Frame 1
- Question: How is the receiver to know second ACK is for frame 0?
- Answer: ***Add frame sequence number in ACK header***
- $R_{next}$ is sequence number of next frame expected by the receiver
- Implicitly acknowledges receipt of all prior frames

# 1-Bit Sequence Numbering Suffices




Global State:
($S_{last}$, $R_{next}$)



# Stop-and-Wait ARQ

**Transmitter**

Ready state

- Await request from higher layer for packet transfer
- When request arrives, transmit frame with updated $S_{last}$ and CRC
- Go to Wait State

Wait state

- Wait for ACK or timer to expire; block requests from higher layer
- If timeout expires
  - retransmit frame and reset timer
- If ACK received:
  - If sequence number is incorrect or if errors detected: ignore ACK
  - If sequence number is correct ($R_{next} = S_{last} + 1$): accept frame, go to Ready state

**Receiver**

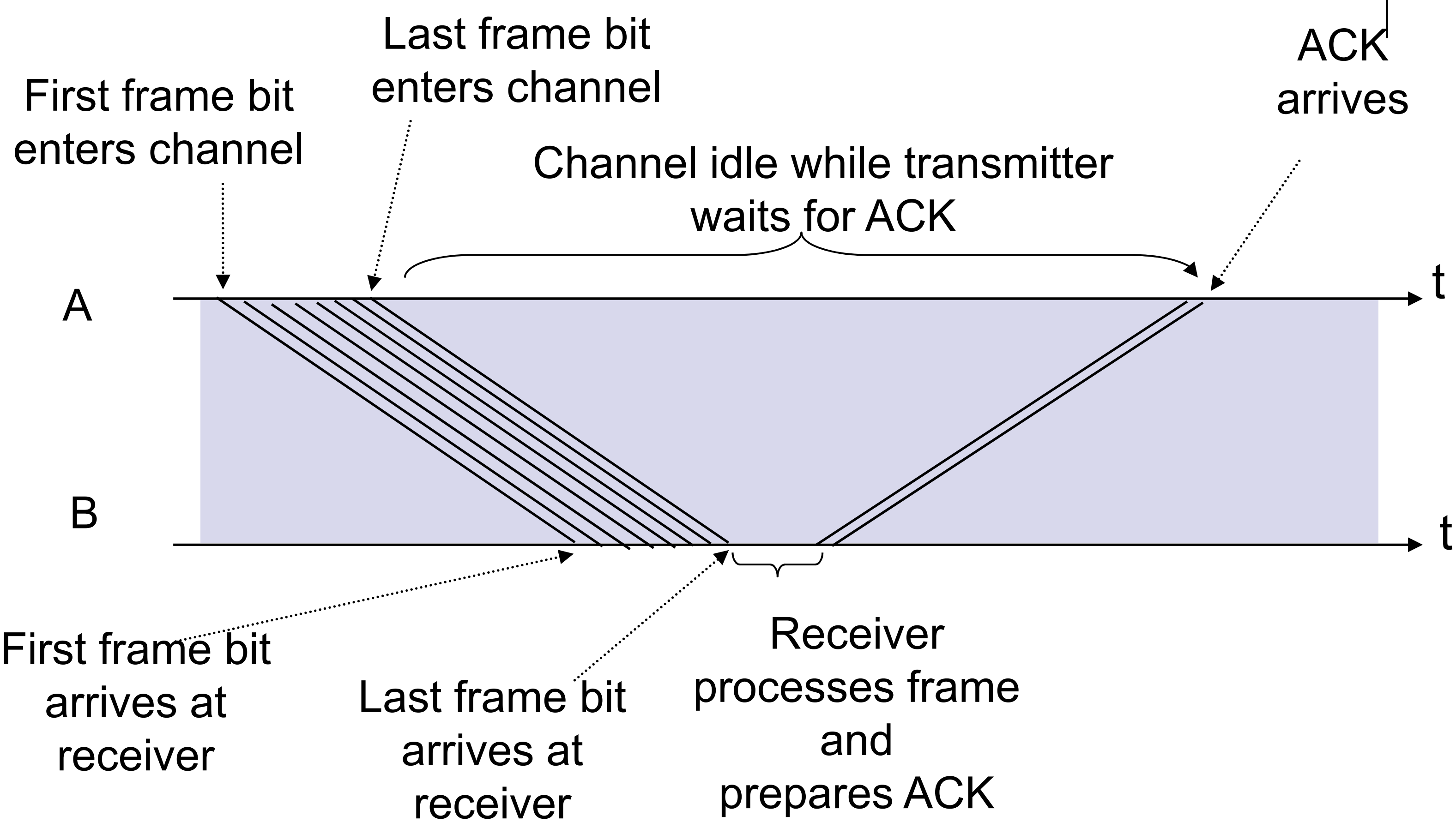
Always in Ready State

- Wait for arrival of new frame
- When frame arrives, check for errors
- If no errors detected and sequence number is correct ($S_{last}$=$R_{next}$), then
  - accept frame,
  - update $R_{next}$,
  - send ACK frame with $R_{next}$,
  - deliver packet to higher layer
- If no errors detected and wrong sequence number
  - discard frame
  - send ACK frame with $R_{next}$
- If errors detected
  - discard frame

# Applications of Stop-and-Wait ARQ

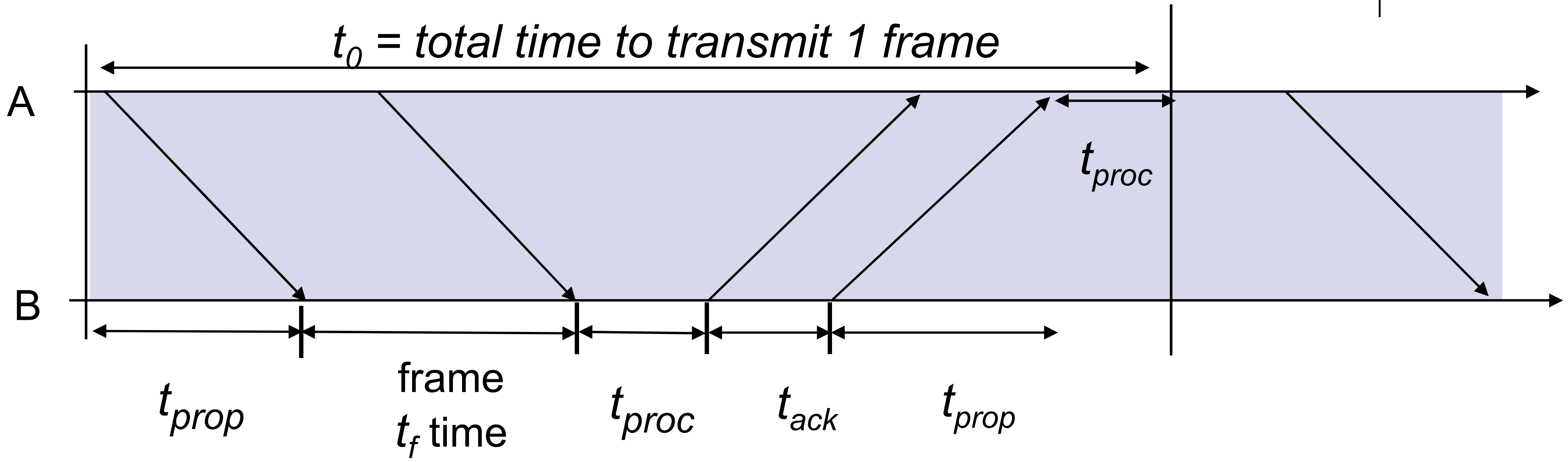
- IBM *Binary Synchronous Communications protocol* (Bisync): character-oriented data link control
- *Xmodem*: modem file transfer protocol
- *Trivial File Transfer Protocol* (RFC 1350): simple protocol for file transfer over UDP

# Stop-and-Wait Efficiency




- 10000 bit frame @ 1 Mbps takes 10 ms to transmit
- If wait for ACK = 1 ms, then efficiency = 10/11= 91%
- If wait for ACK = 20 ms, then efficiency =10/30 = 33%

# Stop-and-Wait Model




$$t_0 = 2t_{prop} + 2t_{proc} + t_f + t_{ack}$$

$$= 2t_{prop} + 2t_{proc} + \frac{n_f}{R} + \frac{n_a}{R}$$

bits/info frame

bits/ACK frame

channel transmission rate

# S&W Efficiency on Error-free channel

***Effective transmission rate:***

bits for header & CRC

$$R_{eff}^{0} = \frac{\text{number of information bits delivered to destination}}{\text{total time required to deliver the information bits}} = \frac{n_f - n_o}{t_0},$$

***Transmission efficiency:***

$$\eta_0 = \frac{R_{eff}}{R} = \frac{\frac{n_f - n_o}{t_0}}{R} = \frac{1 - \frac{n_o}{n_f}}{1 + \frac{n_a}{n_f} + \frac{2(t_{prop} + t_{proc})R}{n_f}}.$$

Effect of frame overhead

Effect of ACK frame

Effect of ***Delay-Bandwidth Product***

# Example: Impact of Delay-Bandwidth Product

$n_f$=1250 bytes = 10000 bits, $n_a$=$n_o$=25 bytes = 200 bits

| | Delay × Bandwidth Product<br>Efficiency | | | |
|---|---|---|---|---|
| Reaction time<br>Distance | 1 ms<br>200 km | 10 ms<br>2000 km | 100 ms<br>20000 km | 1 sec<br>200000 km |
| 1 Mbps | $10^3$<br>88% | $10^4$<br>49% | $10^5$<br>9% | $10^6$<br>1% |
| 1 Gbps | $10^6$<br>1% | $10^7$<br>0.1% | $10^8$<br>0.01% | $10^9$<br>0.001% |

*Stop-and-Wait does not work well for very high speeds or long propagation delays*

# S&W Efficiency in Channel with Errors

- Let $1 - P_f$ = probability frame arrives w/o errors
- Avg. # of transmissions to first correct arrival is then $1/(1-P_f)$
- "If 1-in-10 get through without error, then avg. 10 tries to success"
- Avg. Total Time per frame is then $t_0/(1 - P_f)$

$$\eta_{SW} = \frac{R_{eff}}{R} = \frac{\dfrac{n_f - n_o}{t_0 / (1-P_f)}}{R} = \frac{1 - \dfrac{n_o}{n_f}}{1 + \dfrac{n_a}{n_f} + \dfrac{2(t_{prop} + t_{proc})R}{n_f}}(1-P_f)$$

Effect of frame loss

# Example: Impact of Bit Error Rate

$n_f$=1250 bytes = 10000 bits, $n_a$=$n_o$=25 bytes = 200 bits

Find efficiency for random bit errors with p=0, $10^{-6}$, $10^{-5}$, $10^{-4}$

$$1 - P_f = (1-p)^{n_f} \approx e^{-n_f p} \text{ for large } n_f \text{ and small } p$$

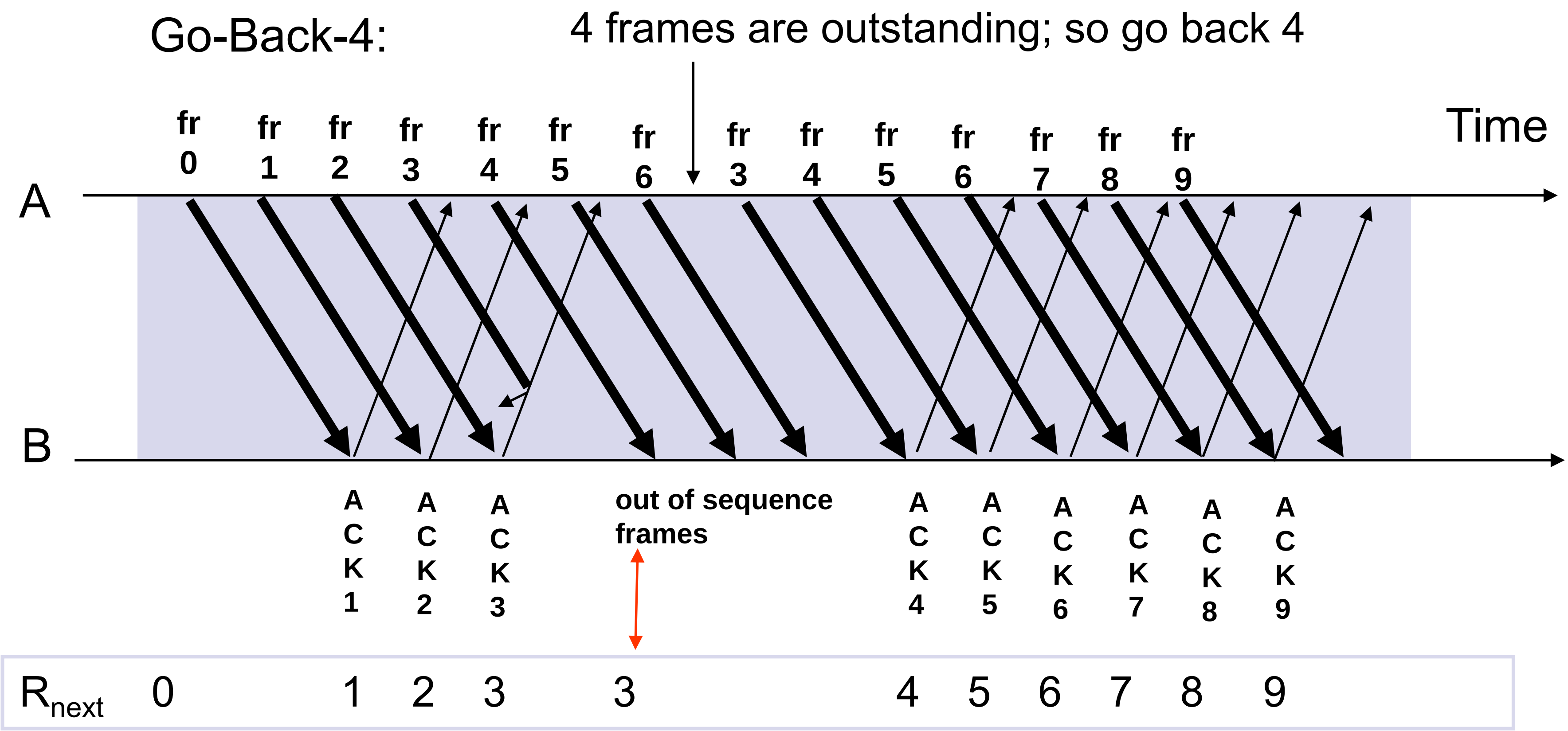
| | Delay × Bandwidth Product Efficiency | | | |
|---|---|---|---|---|
| Bit error p | 0 | $10^{-6}$ | $10^{-5}$ | $10^{-4}$ |
| 1 Mbps at 1 ms | 1<br>88% | 0.99<br>86.6% | 0.905<br>79.2% | 0.368<br>32.2% |

*Bit errors impact performance as* $n_f$ x p *approaches 1*

# Go-Back-N

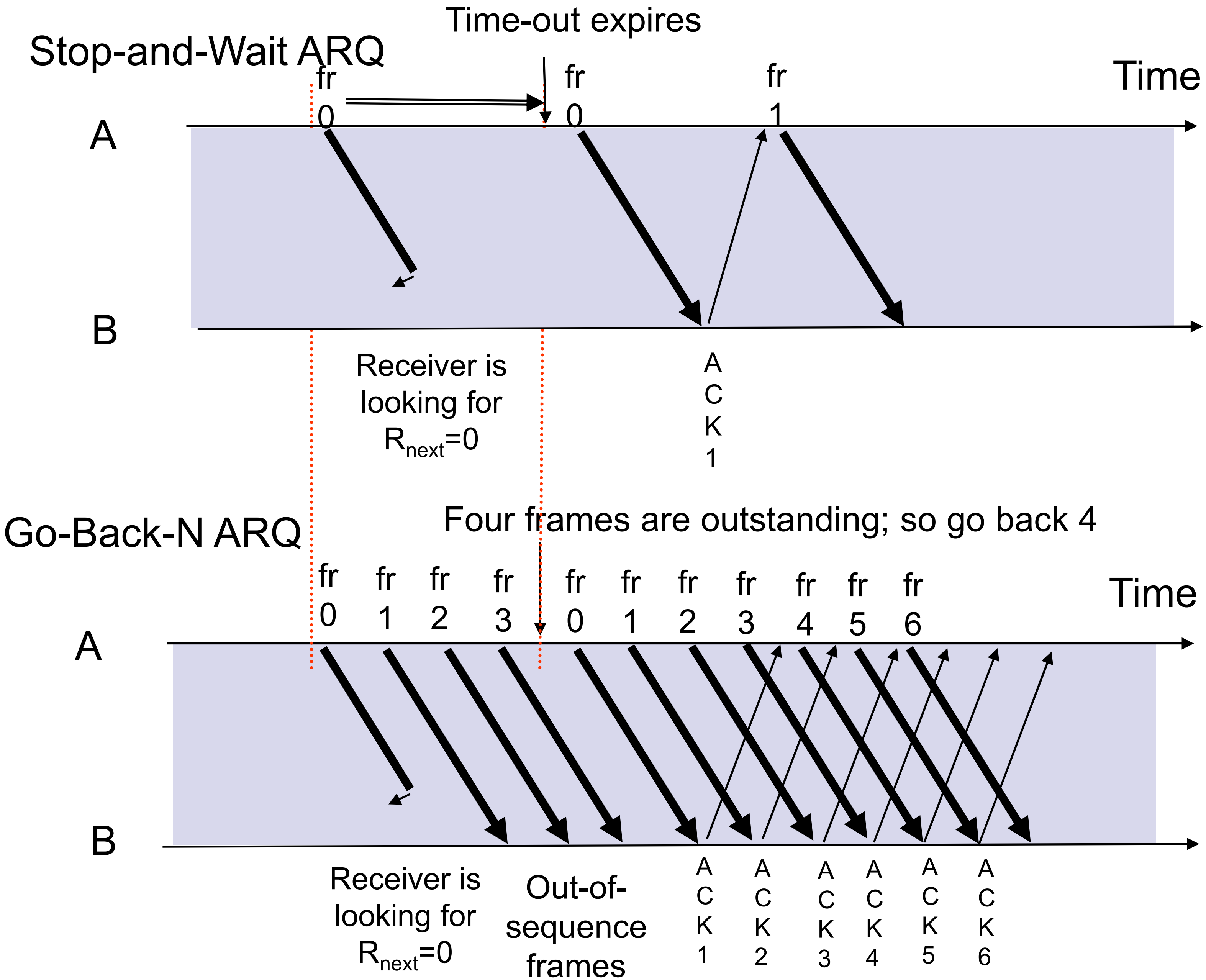
- Improve Stop-and-Wait by not waiting!
- Keep channel busy by continuing to send frames
- Allow a window of up to $W_s$ outstanding frames
- Use *m*-bit sequence numbering
- If ACK for oldest frame arrives before window is exhausted, we can continue transmitting
- If window is exhausted, pull back and retransmit all outstanding frames
- Alternative: Use timeout

# Go-Back-N ARQ




- Frame transmission are *pipelined* to keep the channel busy
- Frame with errors and subsequent out-of-sequence frames are ignored
- Transmitter is forced to go back when window of 4 is exhausted

# *Window size long enough to cover round trip time*

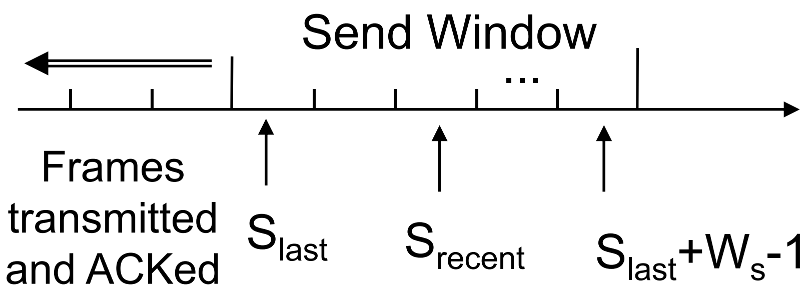


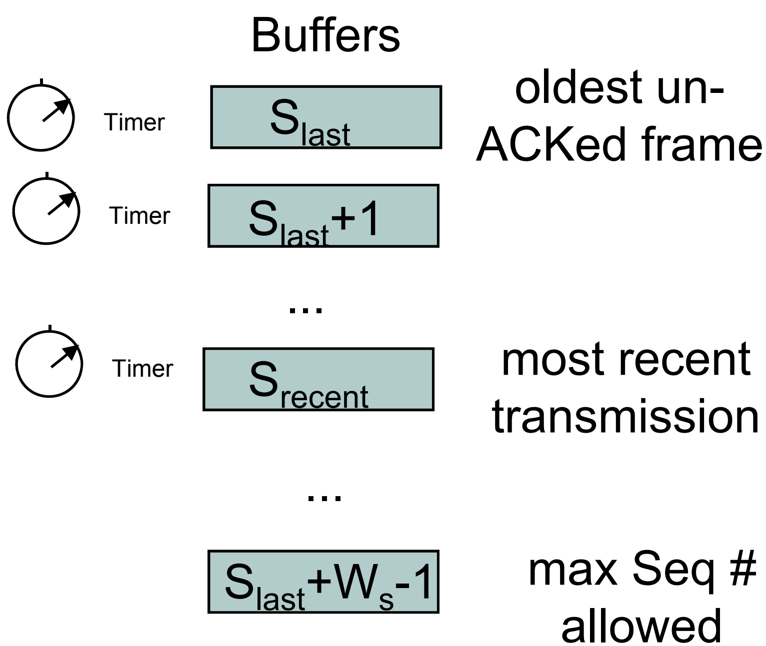
# Go-Back-N with Timeout

- Problem with Go-Back-N as presented:
  - If frame is lost and source does not have frame to send, then window will not be exhausted and recovery will not commence
- Use a timeout with each frame
  - When timeout expires, resend all outstanding frames

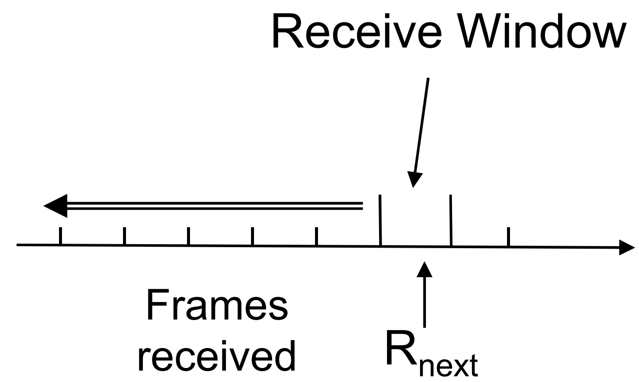
# Go-Back-N Transmitter & Receiver

## Transmitter




## Receiver

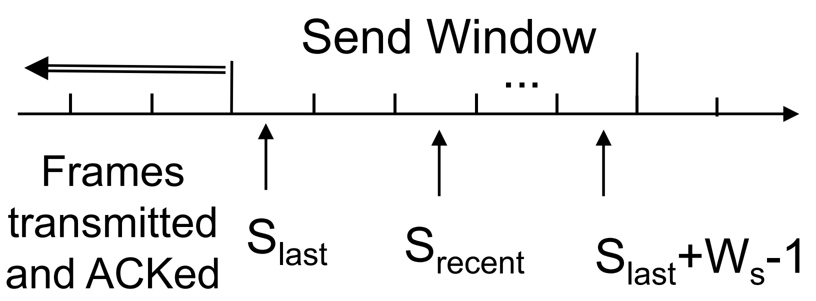



Receiver will only accept a frame that is error-free and that has sequence number $R_{next}$

When such frame arrives $R_{next}$ is incremented by one, so the *receive window slides forward* by one

# Sliding Window Operation

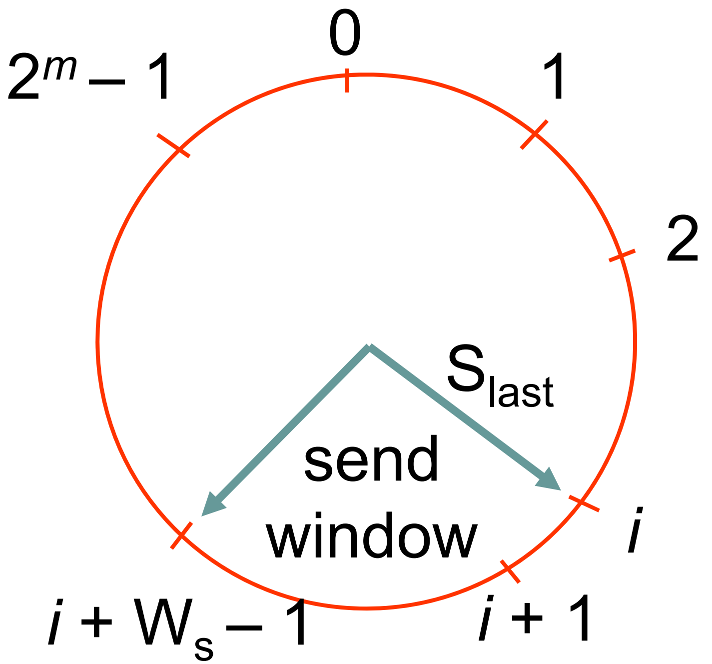
Transmitter



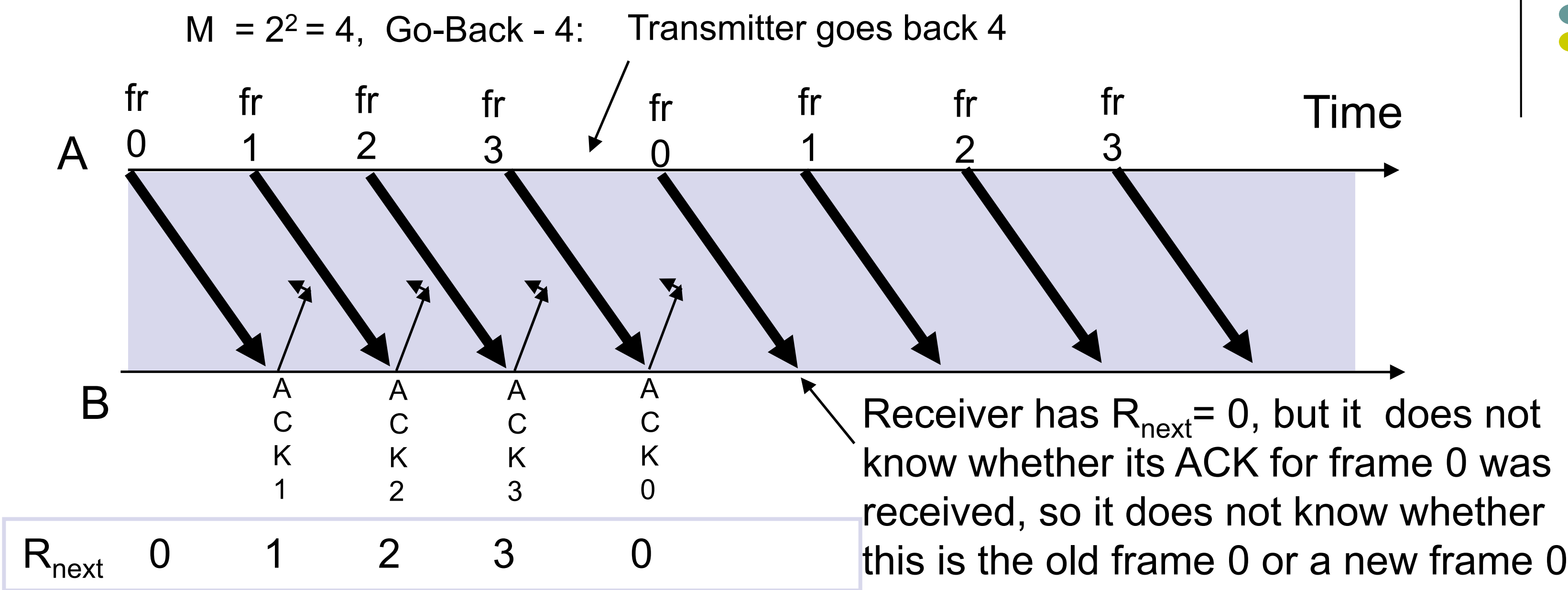

Transmitter waits for error-free ACK frame with sequence number $S_{last}$

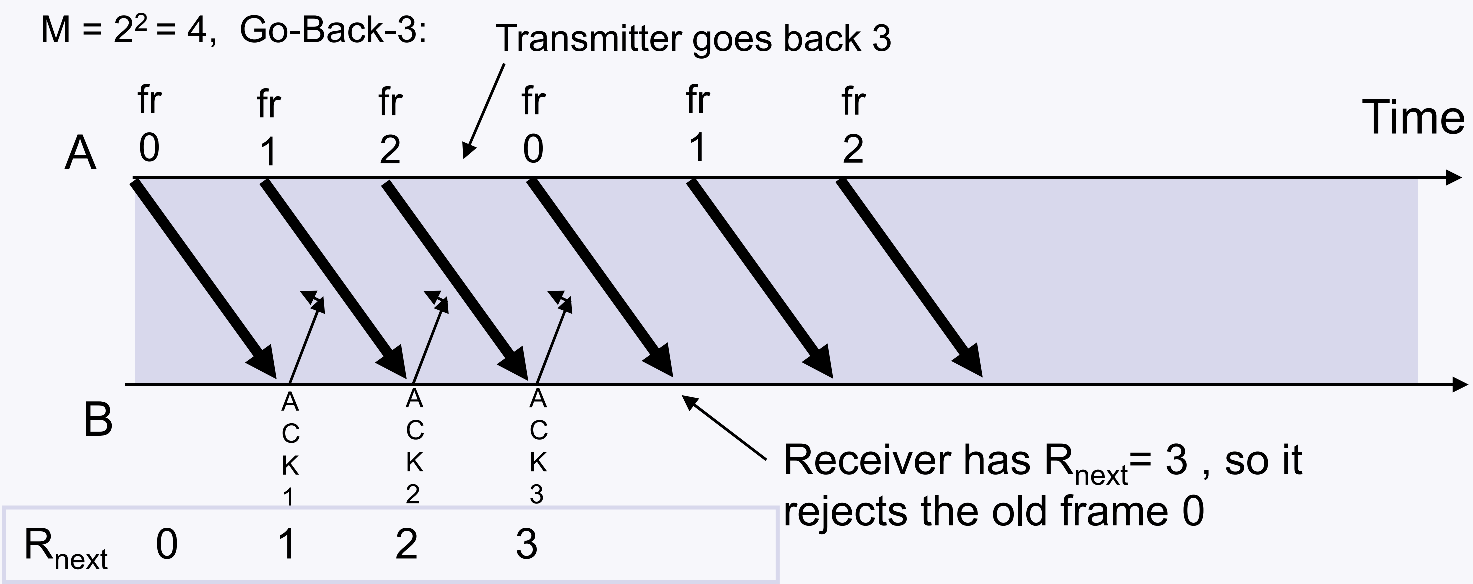
When such ACK frame arrives, $S_{last}$ is incremented by one, and the *send window slides forward* by one

*m*-bit Sequence Numbering

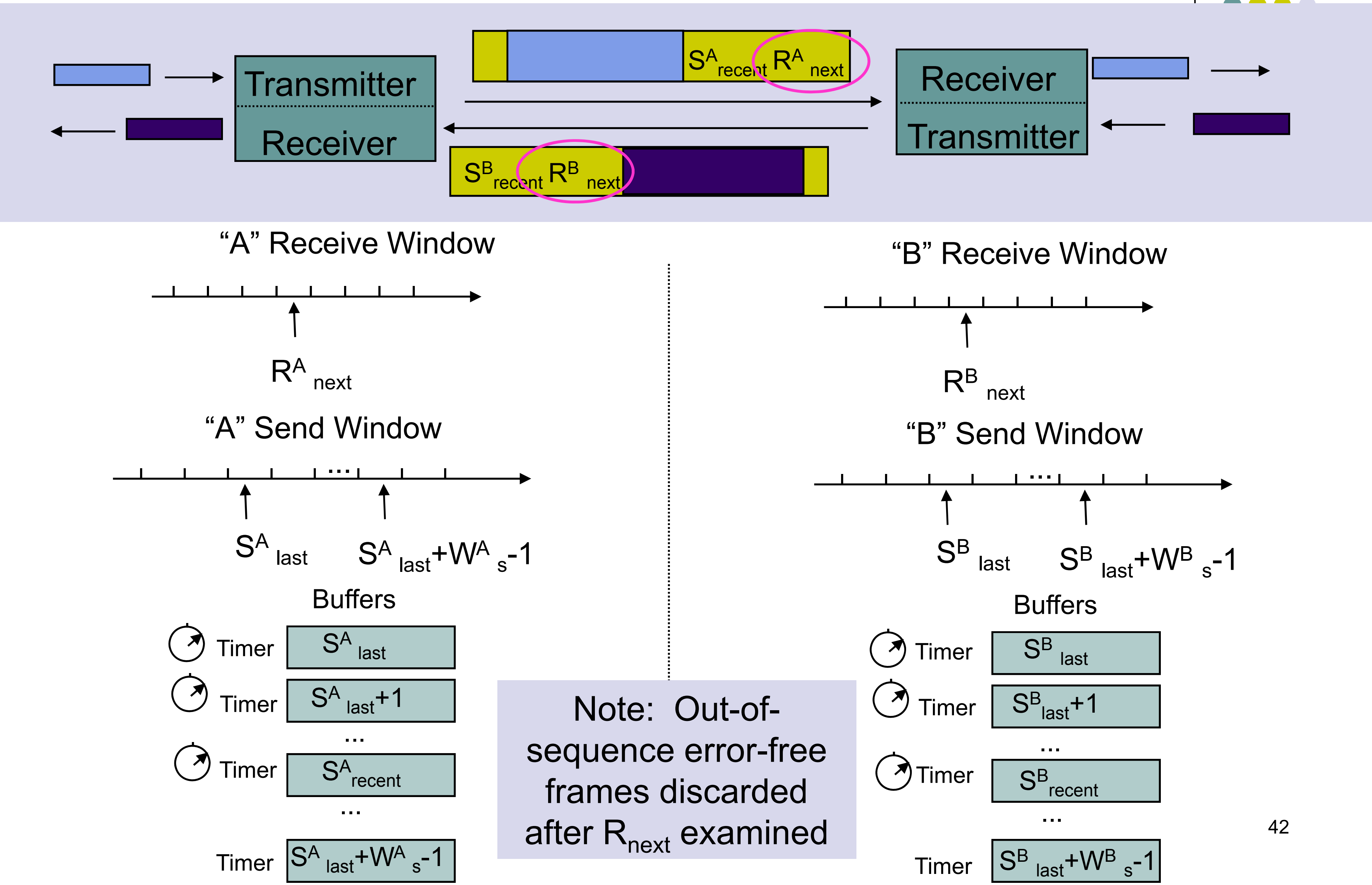


# *Maximum Allowable Window Size is $W_s = 2^m - 1$*



# ACK Piggybacking in Bidirectional GBN

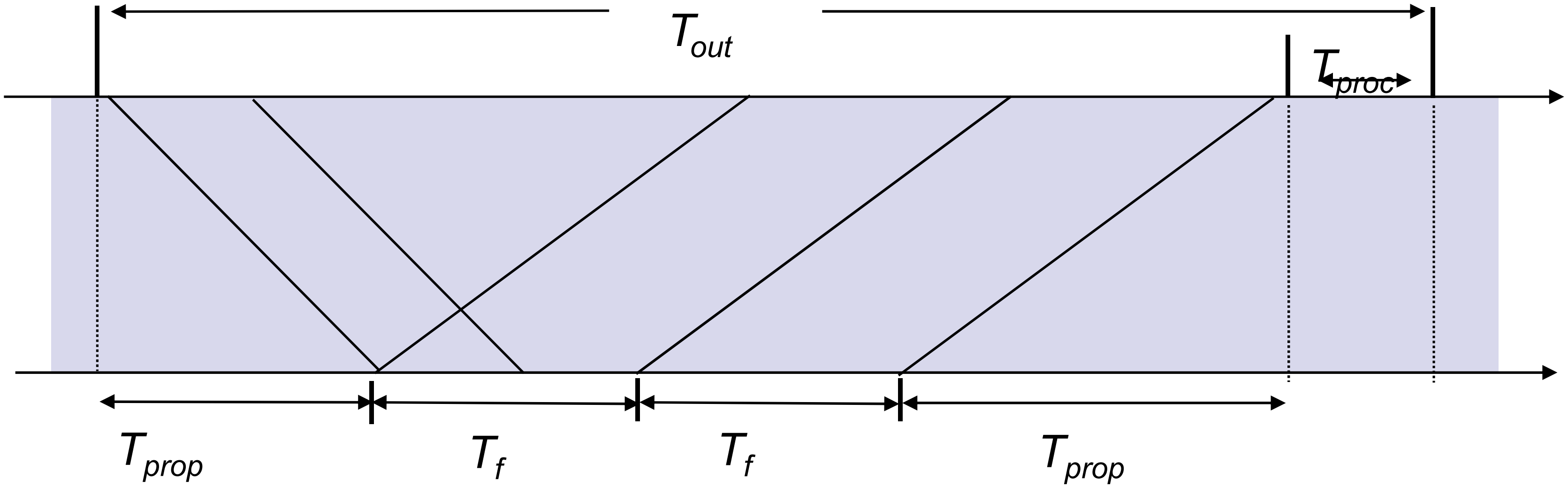



Note: Out-of-sequence error-free frames discarded after $R_{next}$ examined

# Applications of Go-Back-N ARQ

- *HDLC* (High-Level Data Link Control): bit-oriented data link control
- *V.42 modem*: error control over telephone modem links

# Required Timeout & Window Size




- Timeout value should allow for:
  - Two propagation times + 1 processing time: $2\ T_{prop} + T_{proc}$
  - A frame that begins transmission right before our frame arrives $T_f$
  - Next frame carries the ACK, $T_f$
- $W_s$ should be large enough to keep channel busy for $T_{out}$

# Required Window Size for Delay-Bandwidth Product

| Frame = 1250 bytes =10,000 bits, *R* = 1 Mbps | | |
|---|---|---|
| **$2(t_{prop} + t_{proc})$** | **2 x Delay x BW** | **Window** |
| 1 ms | 1000 bits | 1 |
| 10 ms | 10,000 bits | 2 |
| 100 ms | 100,000 bits | 11 |
| 1 second | 1,000,000 bits | 101 |

# Efficiency of Go-Back-N

- GBN is completely efficient, if $W_s$ large enough to keep channel busy, and if channel is error-free
- Assume $P_f$ frame loss probability, then time to deliver a frame is:
  - $t_f$ if first frame transmission succeeds $(1 – P_f)$
  - $T_f + W_s t_f/(1-P_f)$ if the first transmission does not succeed $P_f$

$$t_{GBN} = t_f(1-P_f) + P_f\{t_f + \frac{W_s t_f}{1-P_f}\} = t_f + P_f \frac{W_s t_f}{1-P_f} \quad \text{and}$$

$$\eta_{GBN} = \frac{\frac{n_f - n_o}{t_{GBN}}}{R} = \frac{1-\frac{n_o}{n_f}}{1+(W_s-1)P_f}(1-P_f)$$

Delay-bandwidth product determines $W_s$

# Example: Impact Bit Error Rate on GBN

$n_f$=1250 bytes = 10000 bits, $n_a$=$n_o$=25 bytes = 200 bits

Compare S&W with GBN efficiency for random bit errors with $p$ = 0, $10^{-6}$, $10^{-5}$, $10^{-4}$ and $R$ = 1 Mbps & 100 ms

1 Mbps x 100 ms = 100000 bits = 10 frames → Use $W_s$ = 11

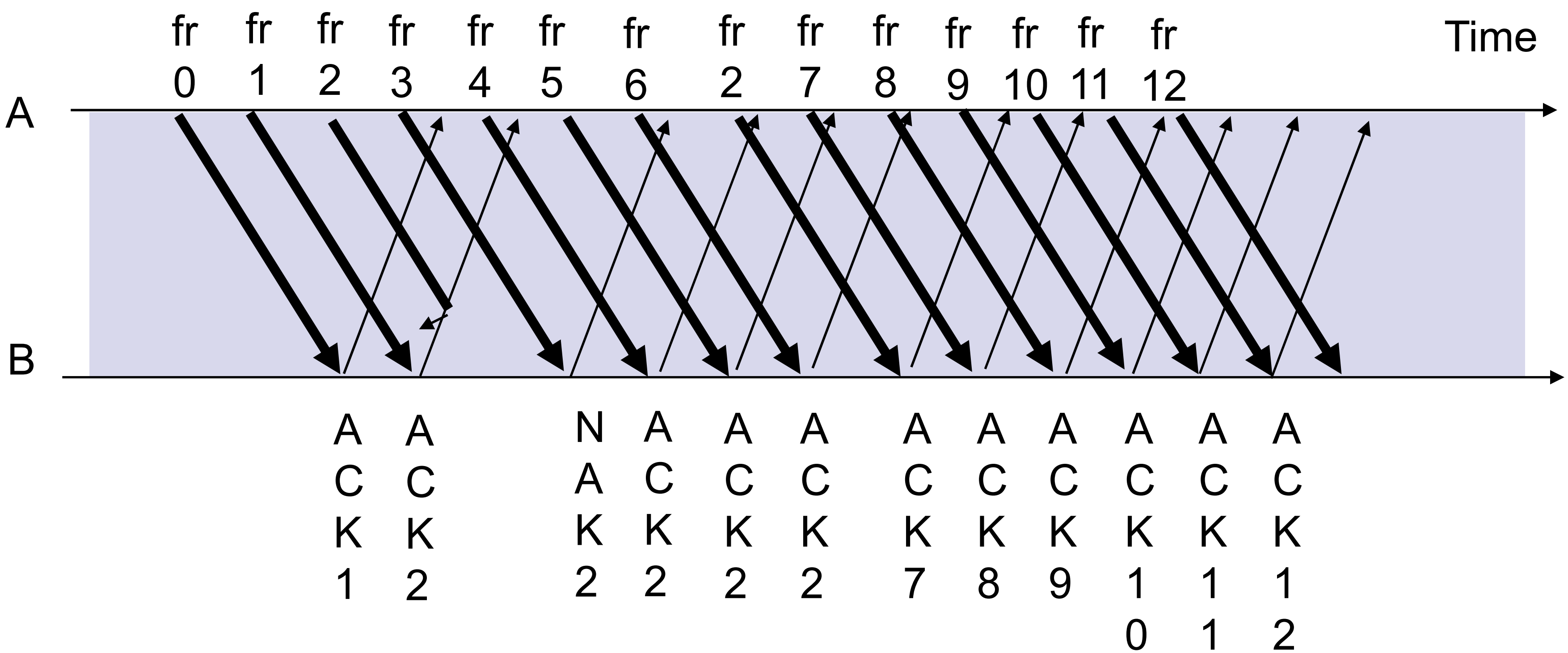
| Efficiency | 0 | $10^{-6}$ | $10^{-5}$ | $10^{-4}$ |
|---|---|---|---|---|
| S&W | 8.9% | 8.8% | 8.0% | 3.3% |
| GBN | 98% | 88.2% | 45.4% | 4.9% |

- *Go-Back-N significant improvement over Stop-and-Wait for large delay-bandwidth product*
- *Go-Back-N becomes inefficient as error rate increases*

# Selective Repeat ARQ

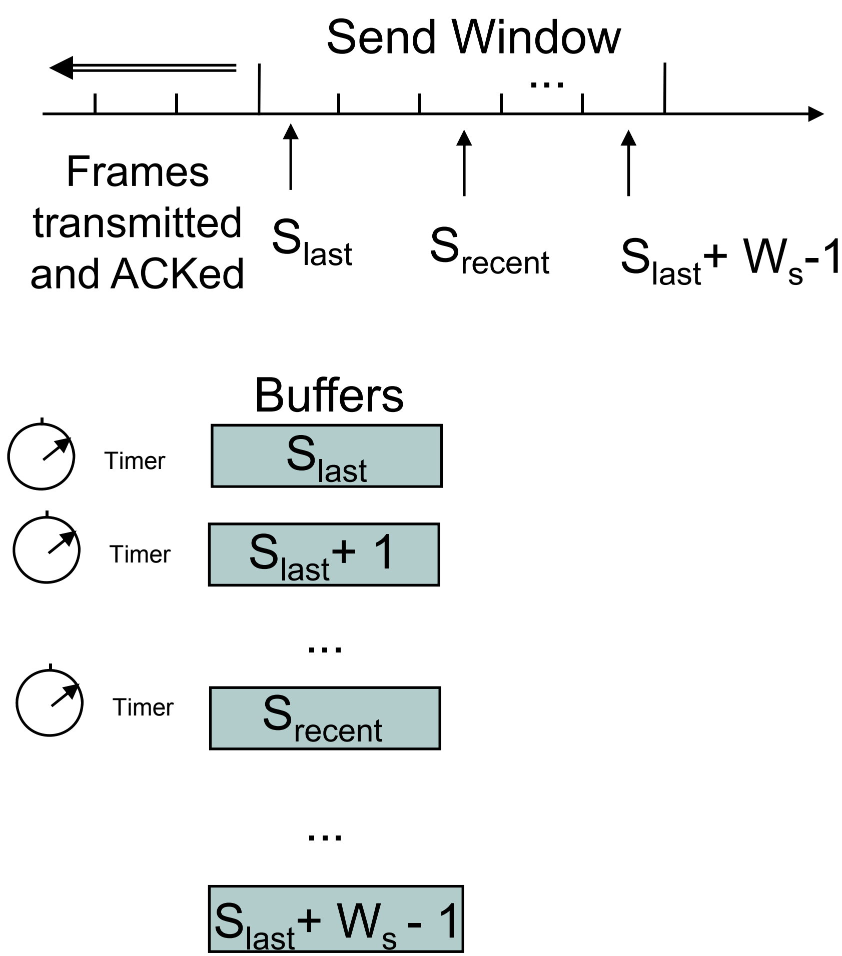
- Go-Back-N ARQ inefficient because *multiple* frames are resent when errors or losses occur
- Selective Repeat retransmits *only an individual frame*
  - Timeout causes individual corresponding frame to be resent
  - NAK causes retransmission of oldest un-acked frame
- Receiver maintains a *receive window* of sequence numbers that can be accepted
  - Error-free, but out-of-sequence frames with sequence numbers within the receive window are buffered
  - Arrival of frame with $R_{next}$ causes window to slide forward by 1 or more

# Selective Repeat ARQ

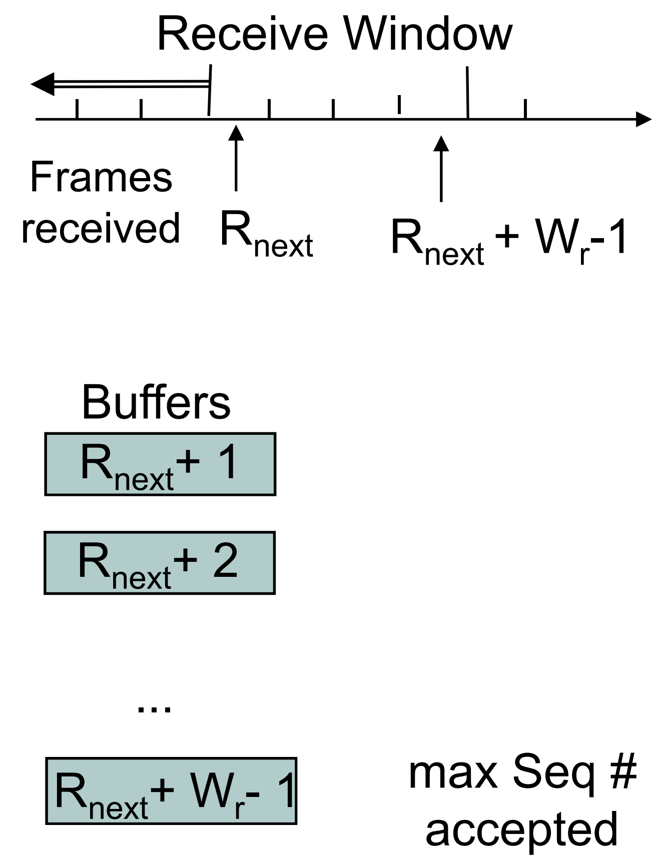


# Selective Repeat ARQ

## Transmitter

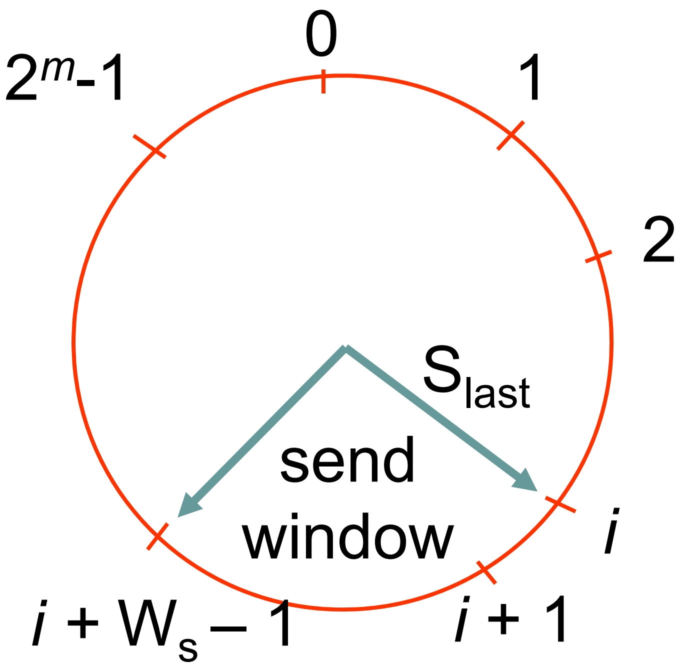



## Receiver



# Send & Receive Windows

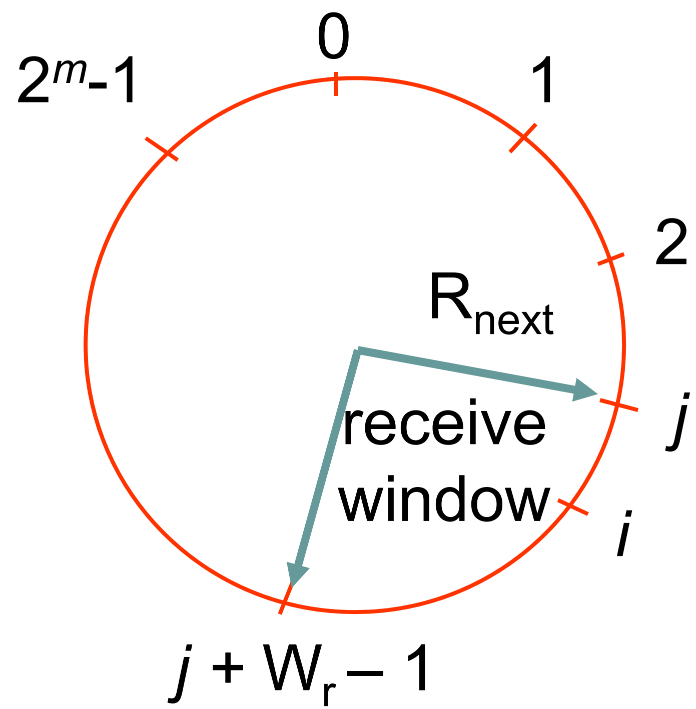
Transmitter




Moves k forward when ACK arrives with $R_{next} = S_{last} + k$

$k = 1, \ldots, W_s-1$

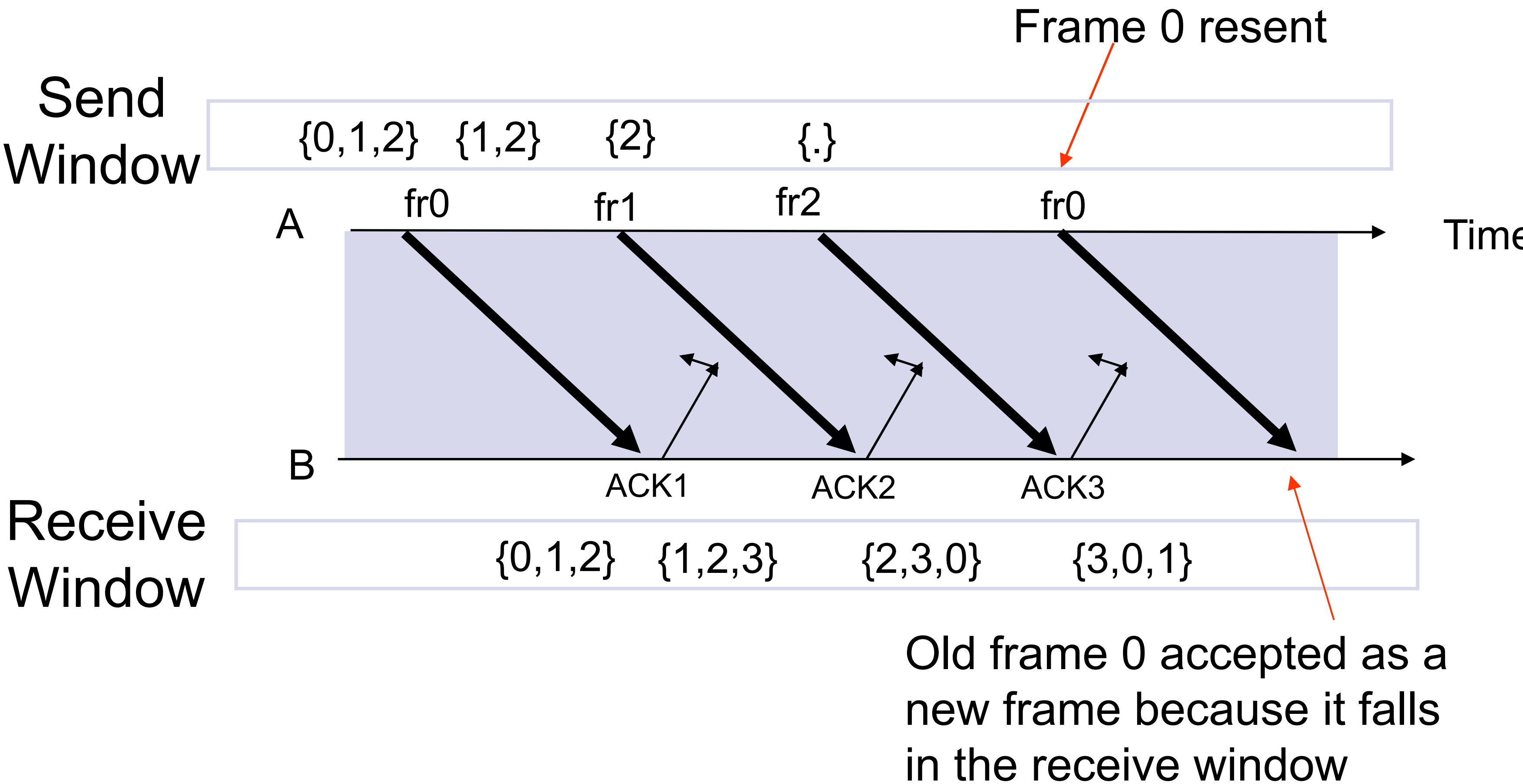
Receiver




Moves forward by 1 or more when frame arrives with Seq. # = $R_{next}$

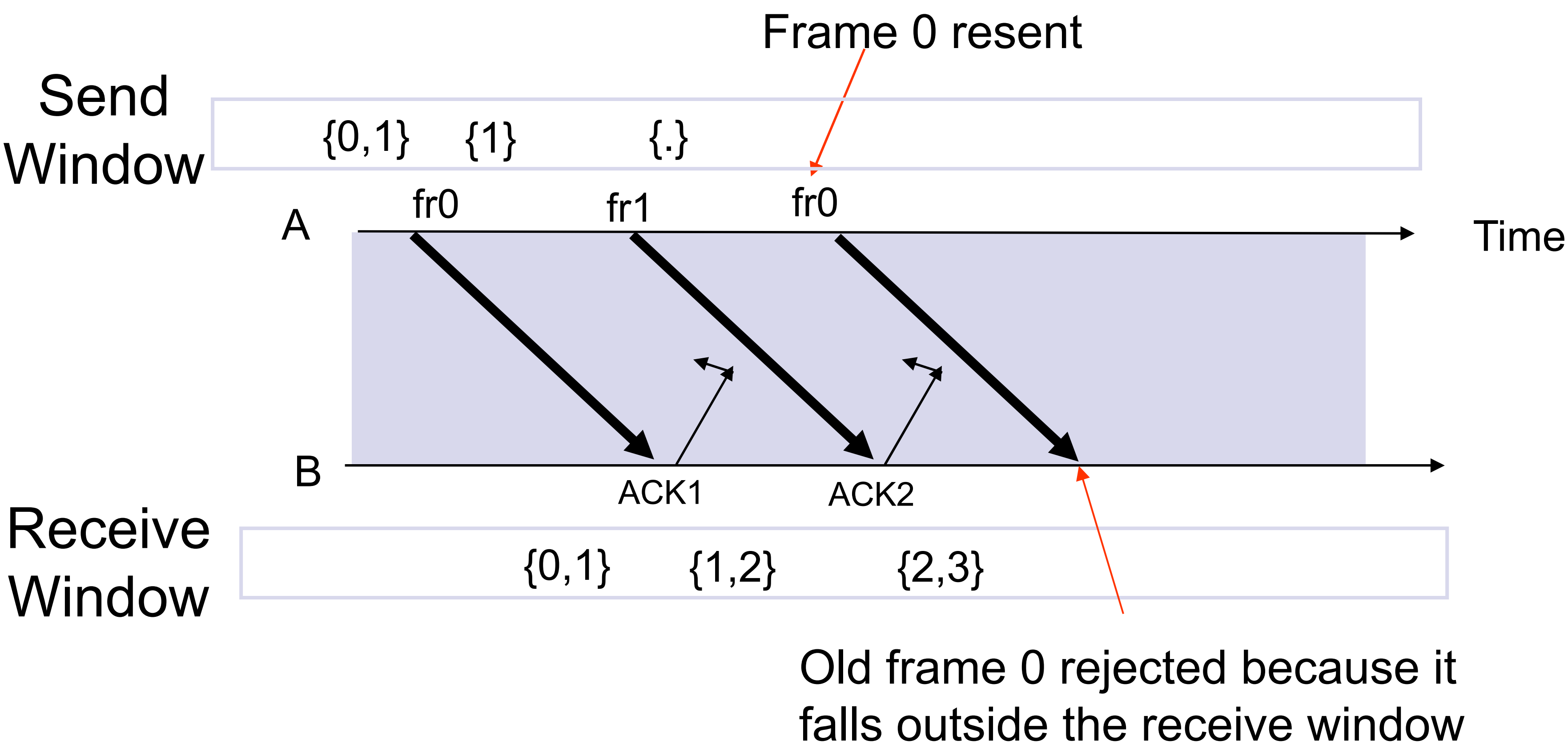
# What size $W_s$ and $W_r$ allowed?

- Example: $M=2^2=4$, $W_s=3$, $W_r=3$



# $W_s + W_r = 2^m$ is maximum allowed

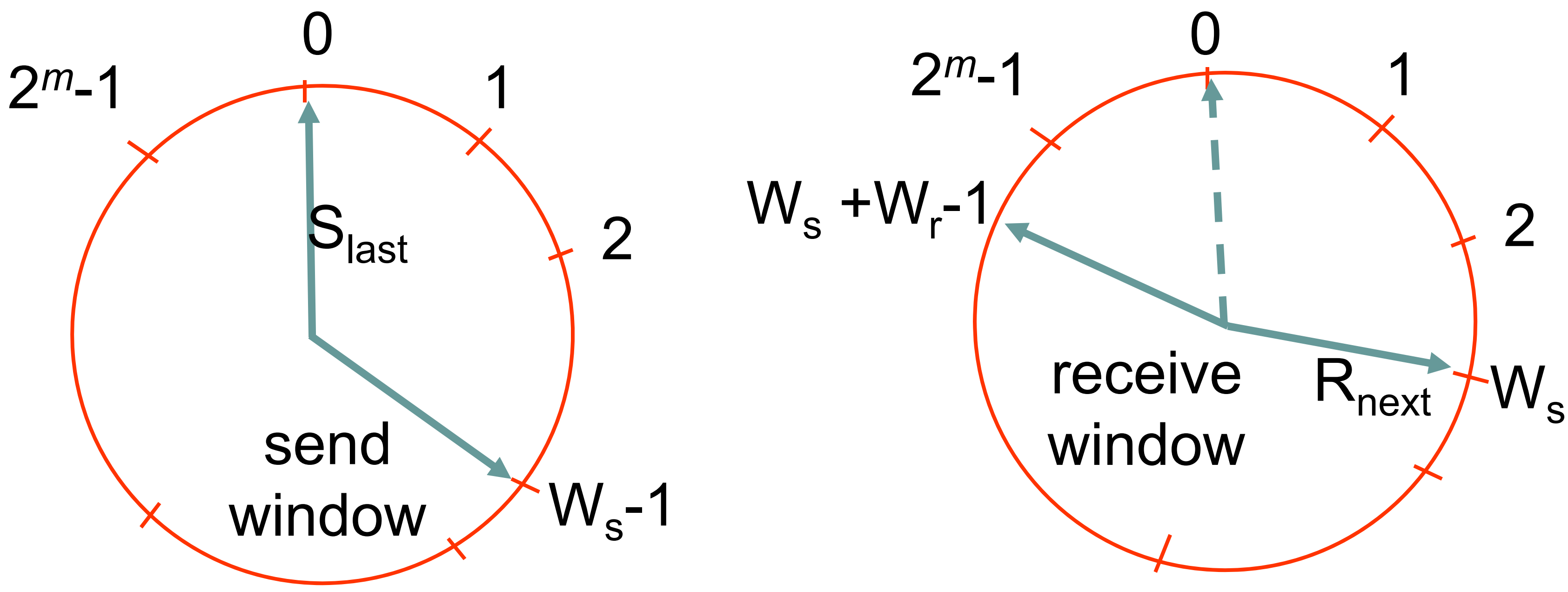
- Example: $M=2^2=4$, $W_s=2$, $W_r=2$



# Why $W_s + W_r = 2^m$ works

- Transmitter sends frames 0 to Ws-1; send window empty
- All arrive at receiver
- All ACKs lost
- Transmitter resends frame 0

- Receiver window starts at $\{0, \ldots, W_r\}$
- Window slides forward to $\{W_s, \ldots, W_s+W_r-1\}$
- Receiver rejects frame 0 because it is outside receive window



# Applications of Selective Repeat ARQ

- *TCP* (Transmission Control Protocol): transport layer protocol uses variation of selective repeat to provide reliable stream service
- *Service Specific Connection Oriented Protocol*: error control for signaling messages in ATM networks

# Efficiency of Selective Repeat

- Assume $P_f$ frame loss probability, then number of transmissions required to deliver a frame is:
  - $t_f / (1-P_f)$

$$\eta_{SR} = \frac{\dfrac{n_f - n_o}{t_f/(1-P_f)}}{R} = (1 - \frac{n_o}{n_f})(1-P_f)$$

# Example: Impact Bit Error Rate on Selective Repeat

$n_f$=1250 bytes = 10000 bits, $n_a$=$n_o$=25 bytes = 200 bits

Compare S&W, GBN & SR efficiency for random bit errors with $p$=0, $10^{-6}$, $10^{-5}$, $10^{-4}$ and $R$= 1 Mbps & 100 ms

| **Efficiency** | 0 | $10^{-6}$ | $10^{-5}$ | $10^{-4}$ |
|---|---|---|---|---|
| S&W | 8.9% | 8.8% | 8.0% | 3.3% |
| GBN | 98% | 88.2% | 45.4% | 4.9% |
| SR | 98% | 97% | 89% | 36% |

- *Selective Repeat outperforms GBN and S&W, but efficiency drops as error rate increases*

# Comparison of ARQ Efficiencies

Assume $n_a$ and $n_o$ are negligible relative to $n_f$, and $L = 2(t_{prop}+t_{proc})R/n_f=(W_s-1)$, then

Selective-Repeat:

$$\eta_{SR} = (1-P_f)(1-\frac{n_o}{n_f}) \approx (1-P_f)$$

Go-Back-N:

*For $P_f \approx 0$, SR & GBN same*

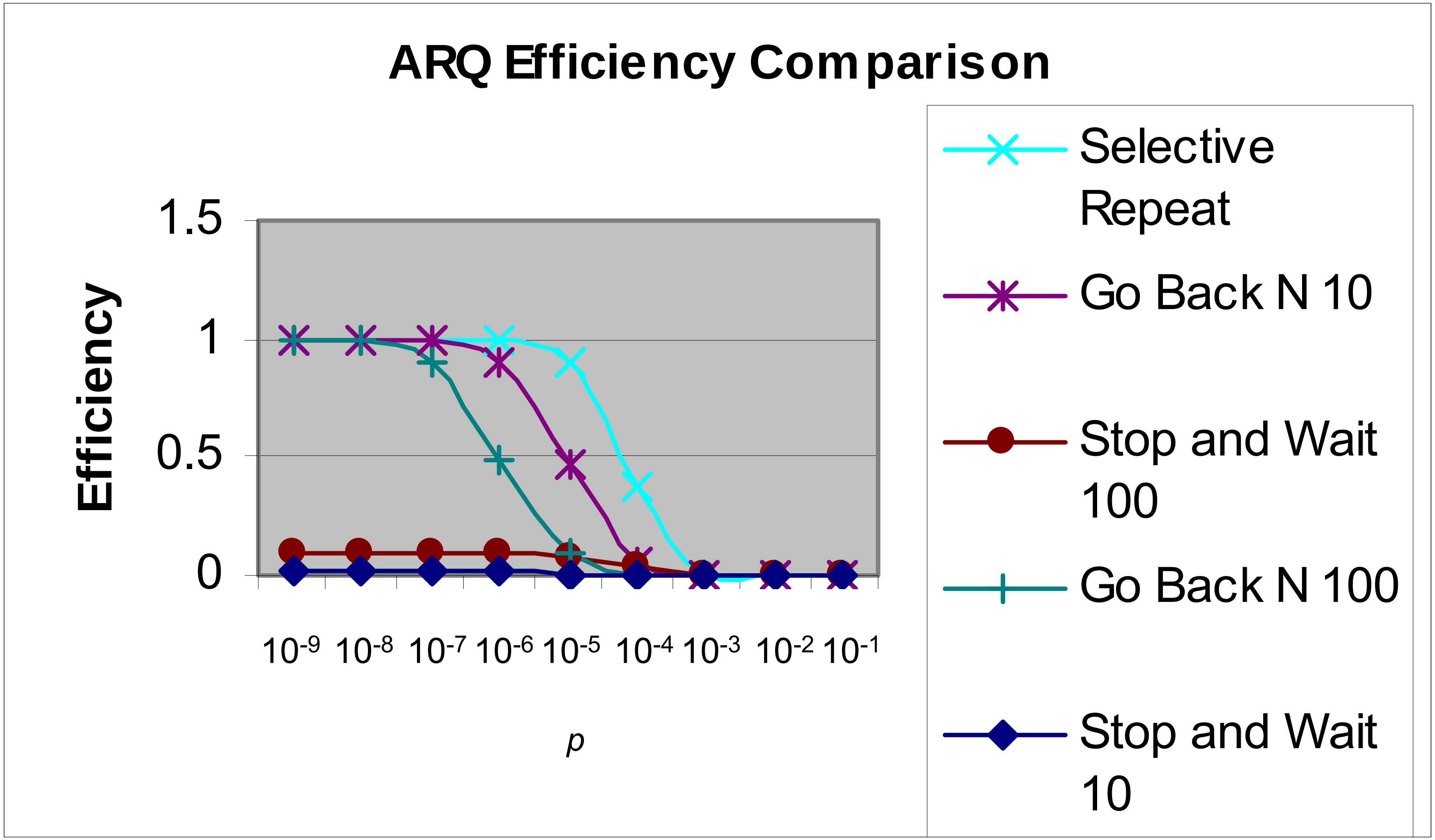
$$\eta_{GBN} = \frac{1-P_f}{1+(W_S-1)P_f} = \frac{1-P_f}{1+LP_f}$$

*For $P_f \to 1$, GBN & SW same*

Stop-and-Wait:

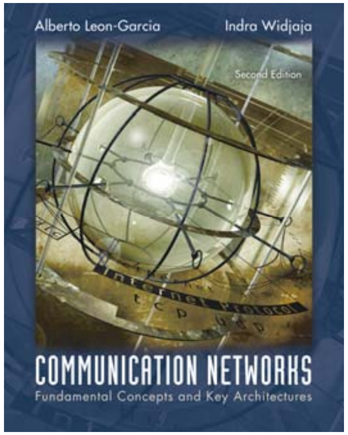
$$\eta_{SW} = \frac{(1-P_f)}{1+\frac{n_a}{n_f}+\frac{2(t_{prop}+t_{proc})R}{n_f}} \approx \frac{1-P_f}{1+L}$$

# ARQ Efficiencies



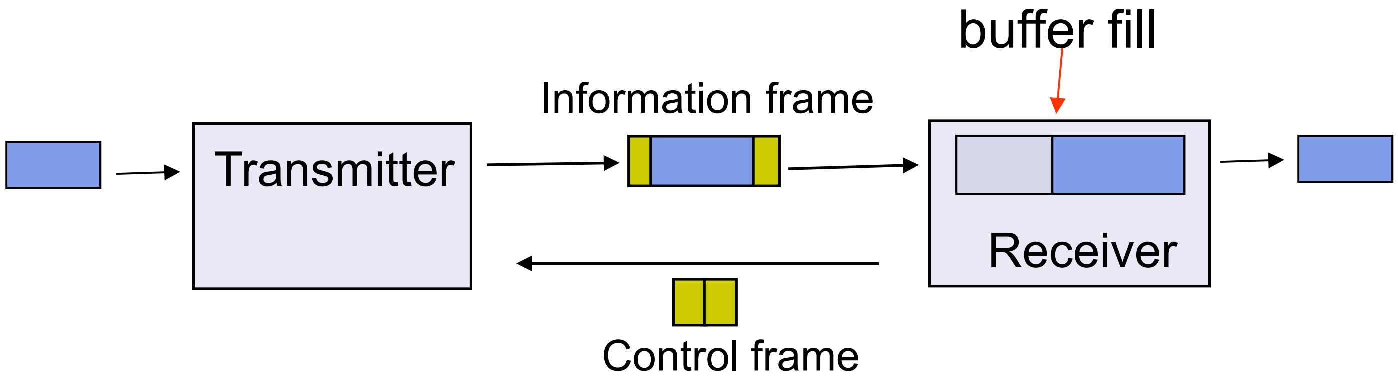

Delay-Bandwidth product = 10, 100

# Chapter 5
# Peer-to-Peer Protocols and Data Link Layer



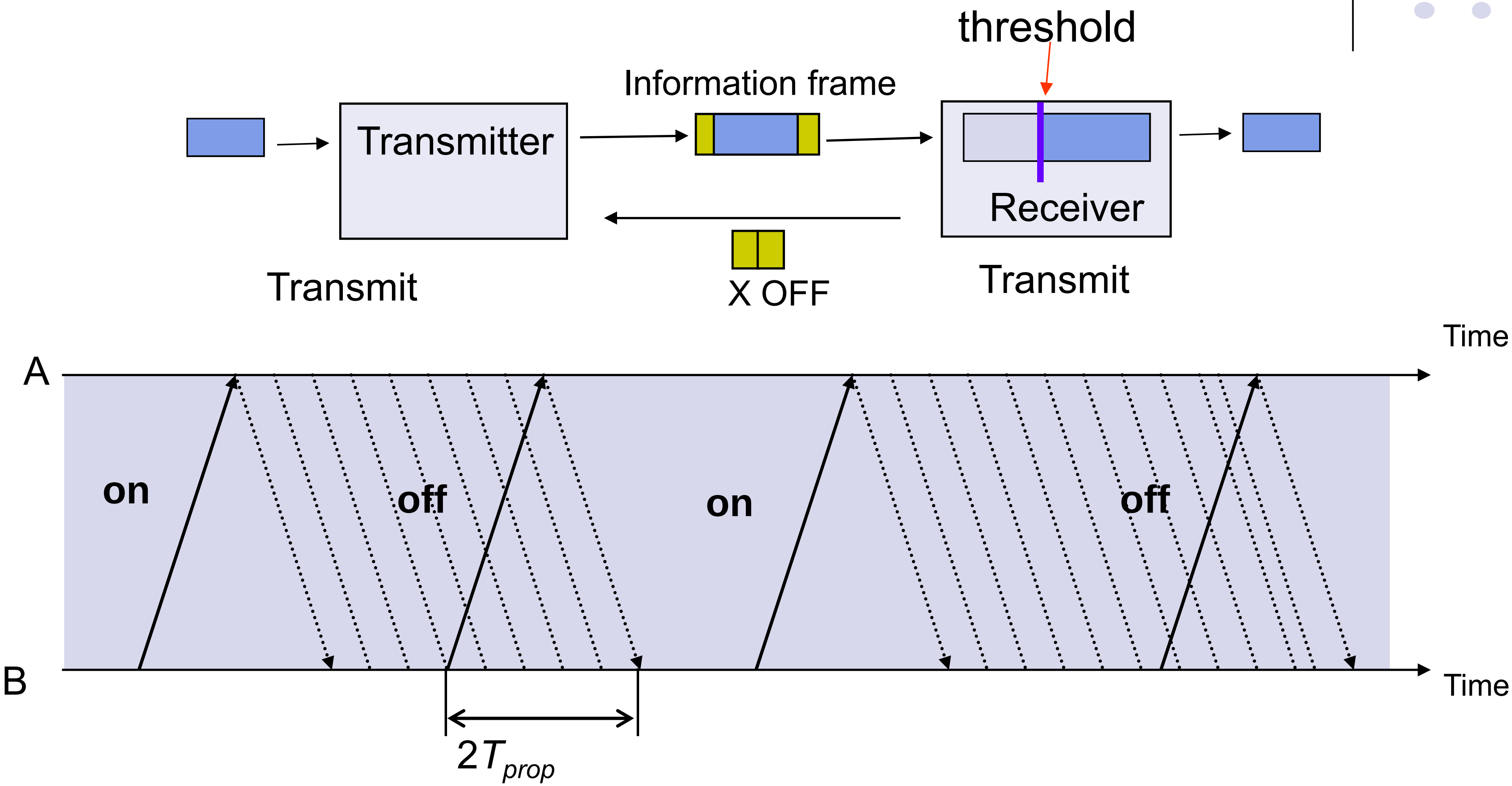

## *Flow Control*

# Flow Control



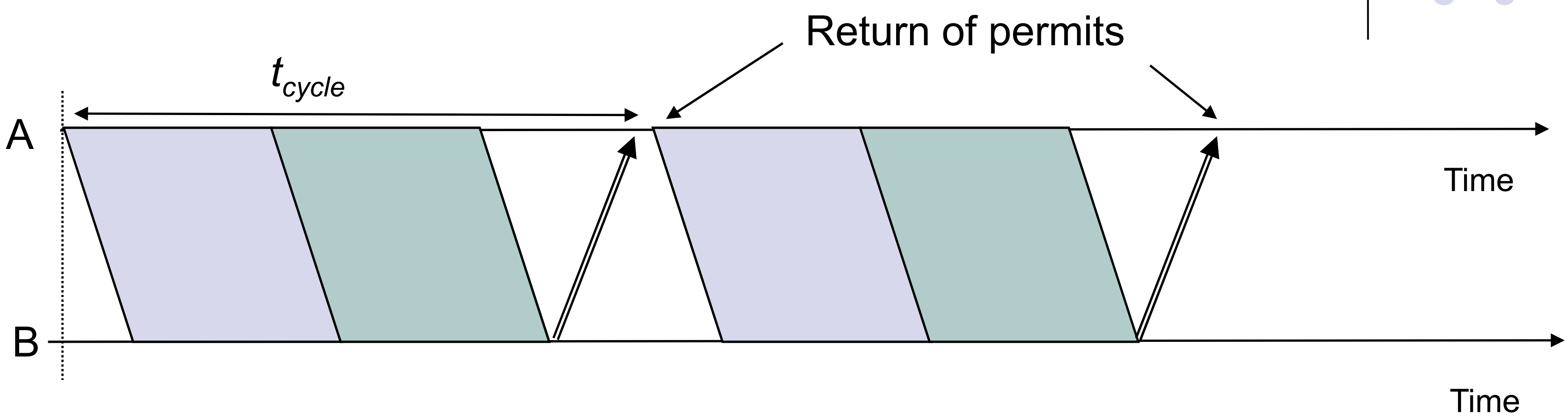

- Receiver has limited buffering to store arriving frames
- Several situations cause buffer overflow
  - Mismatch between sending rate & rate at which user can retrieve data
  - Surges in frame arrivals
- *Flow control* prevents buffer overflow by regulating rate at which source is allowed to send information

# X ON / X OFF




Threshold must activate OFF signal while 2 $T_{prop}$ $R$ bits still remain in buffer

# Window Flow Control




- Sliding Window ARQ method with $W_s$ equal to buffer available
  - Transmitter can never send more than $W_s$ frames
- ACKs that slide window forward can be viewed as permits to transmit more
- Can also pace ACKs as shown above
  - Return permits (ACKs) at end of cycle regulates transmission rate
- Problems using sliding window for both error & flow control
  - Choice of window size
  - Interplay between transmission rate & retransmissions
  - TCP separates error & flow control

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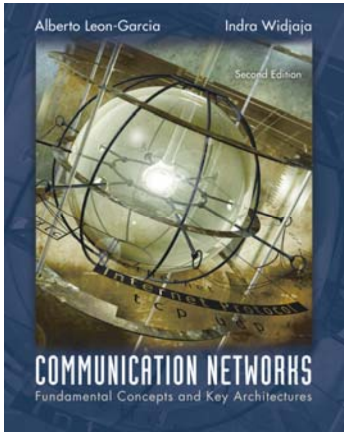
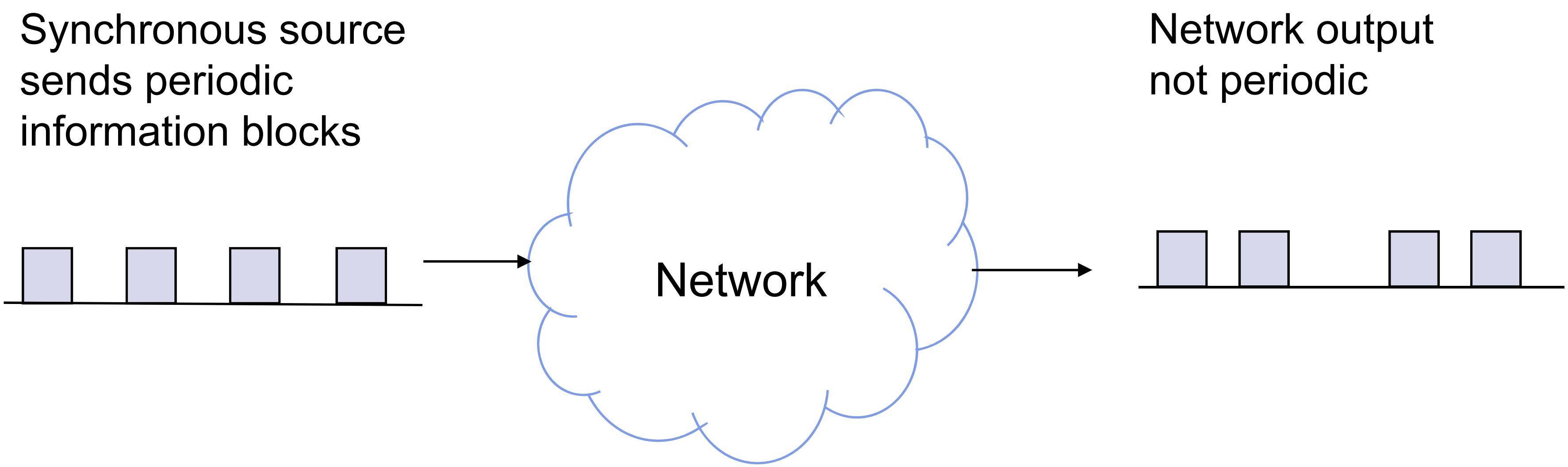


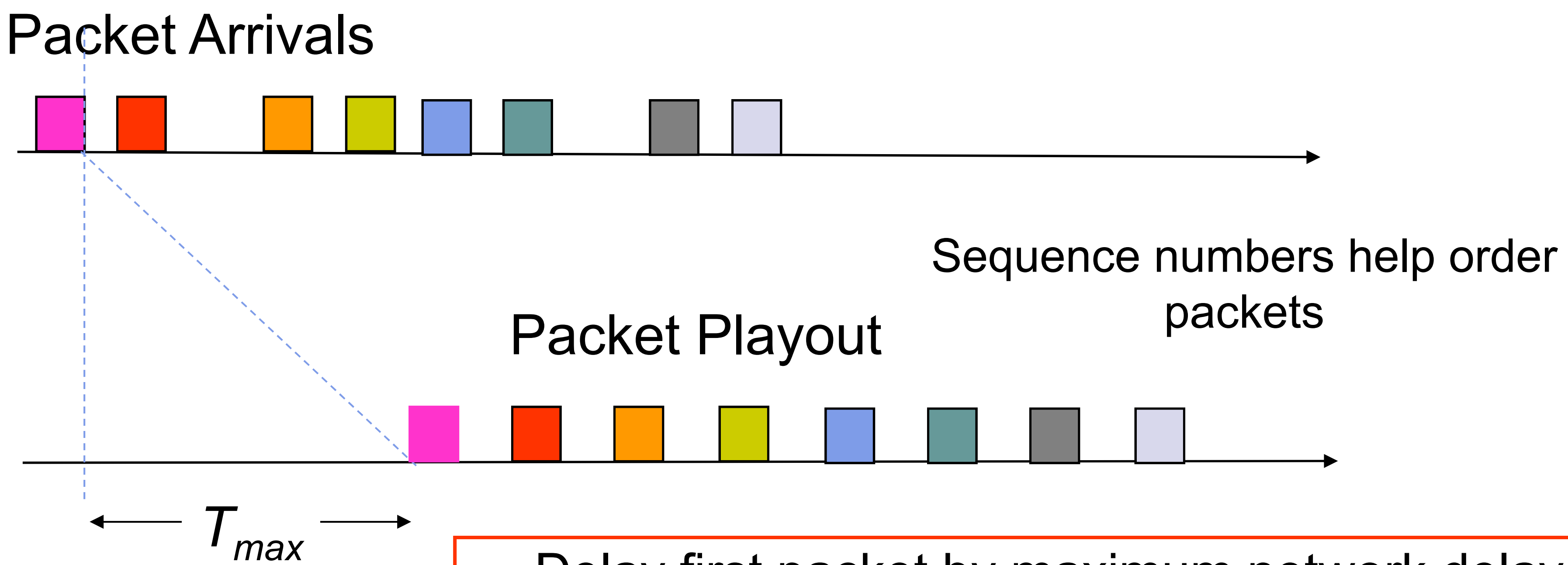

## *Timing Recovery*

# Timing Recovery for Synchronous Services



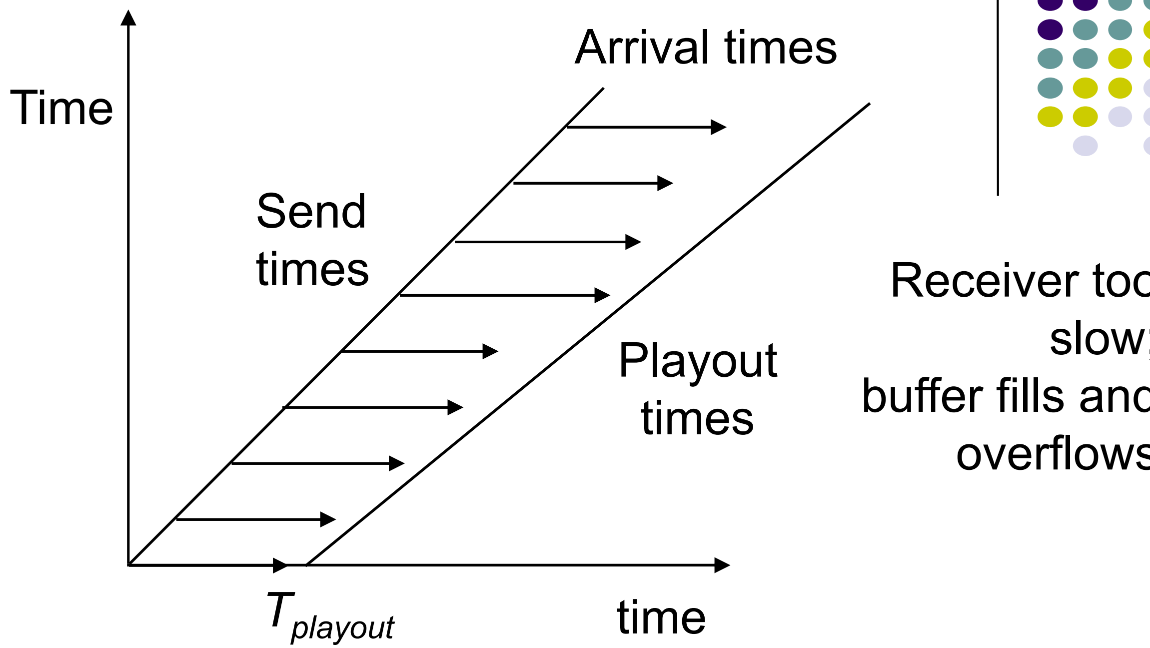

- Applications that involve voice, audio, or video can generate a synchronous information stream
- Information carried by equally-spaced fixed-length packets
- Network multiplexing & switching introduces random delays
  - Packets experience variable transfer delay
  - Jitter (variation in interpacket arrival times) also introduced
- Timing recovery re-establishes the synchronous nature of the stream

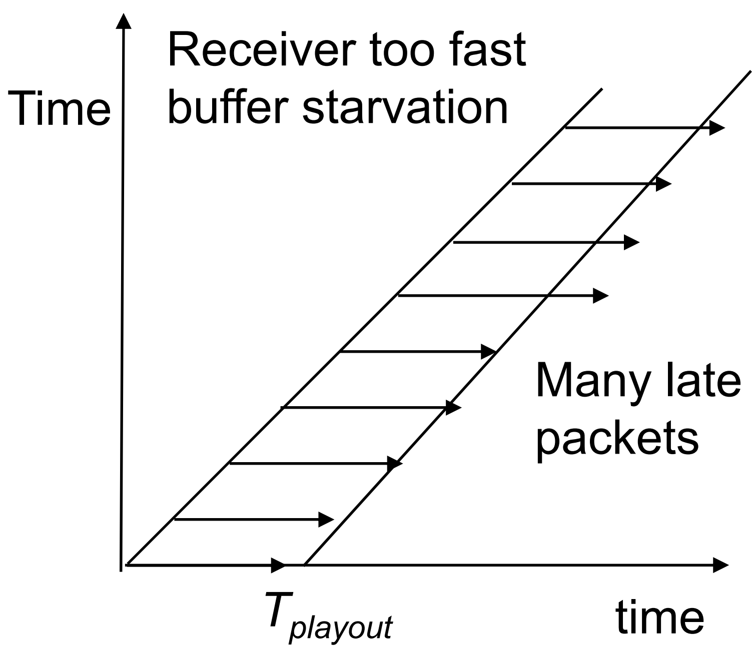
# Introduce Playout Buffer

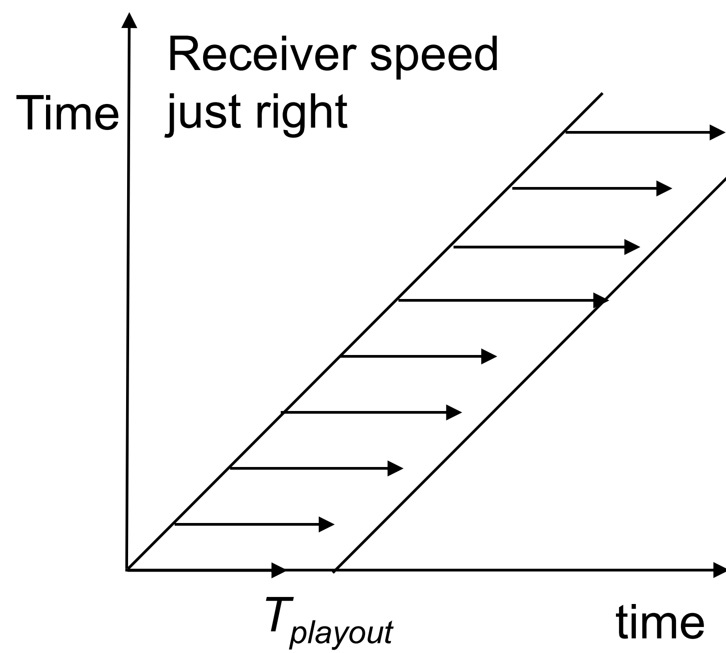




- Delay first packet by maximum network delay
- All other packets arrive with less delay
- Playout packet uniformly thereafter

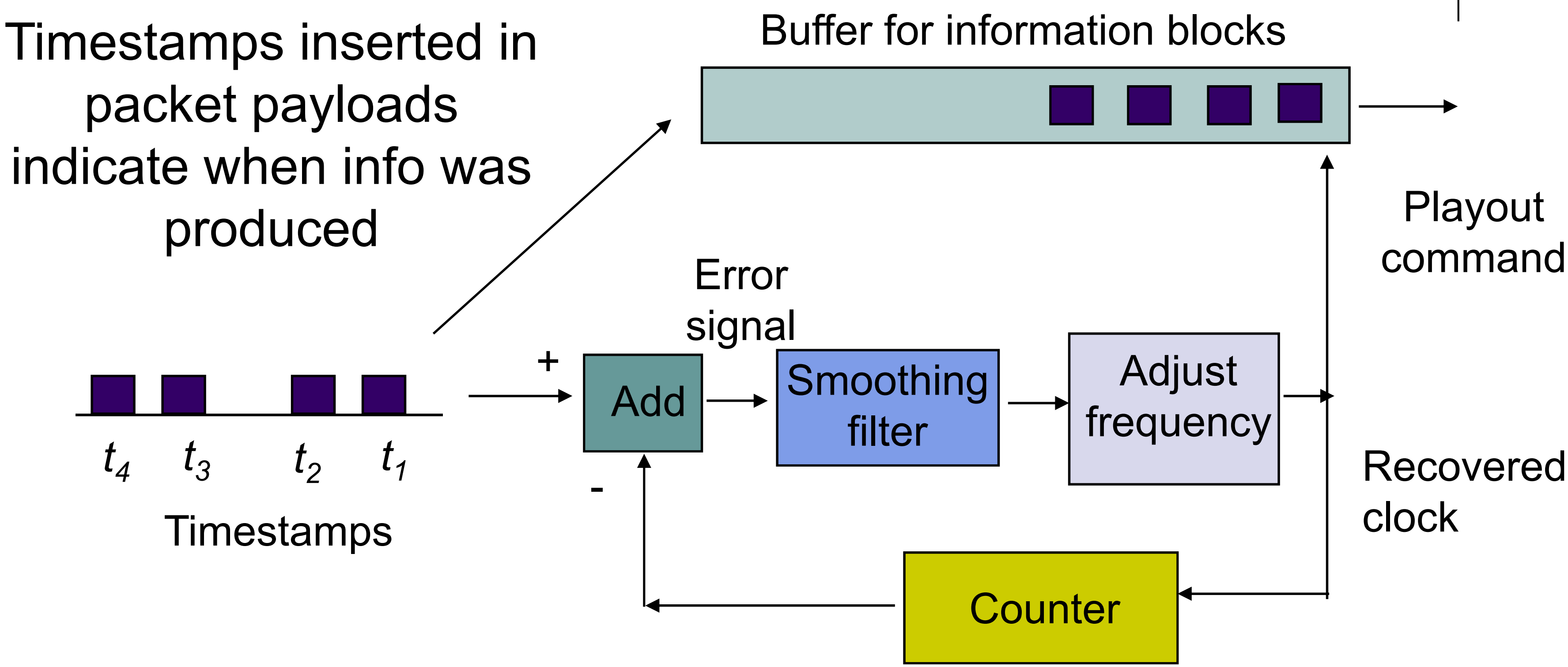
*Playout clock must be synchronized to transmitter clock*



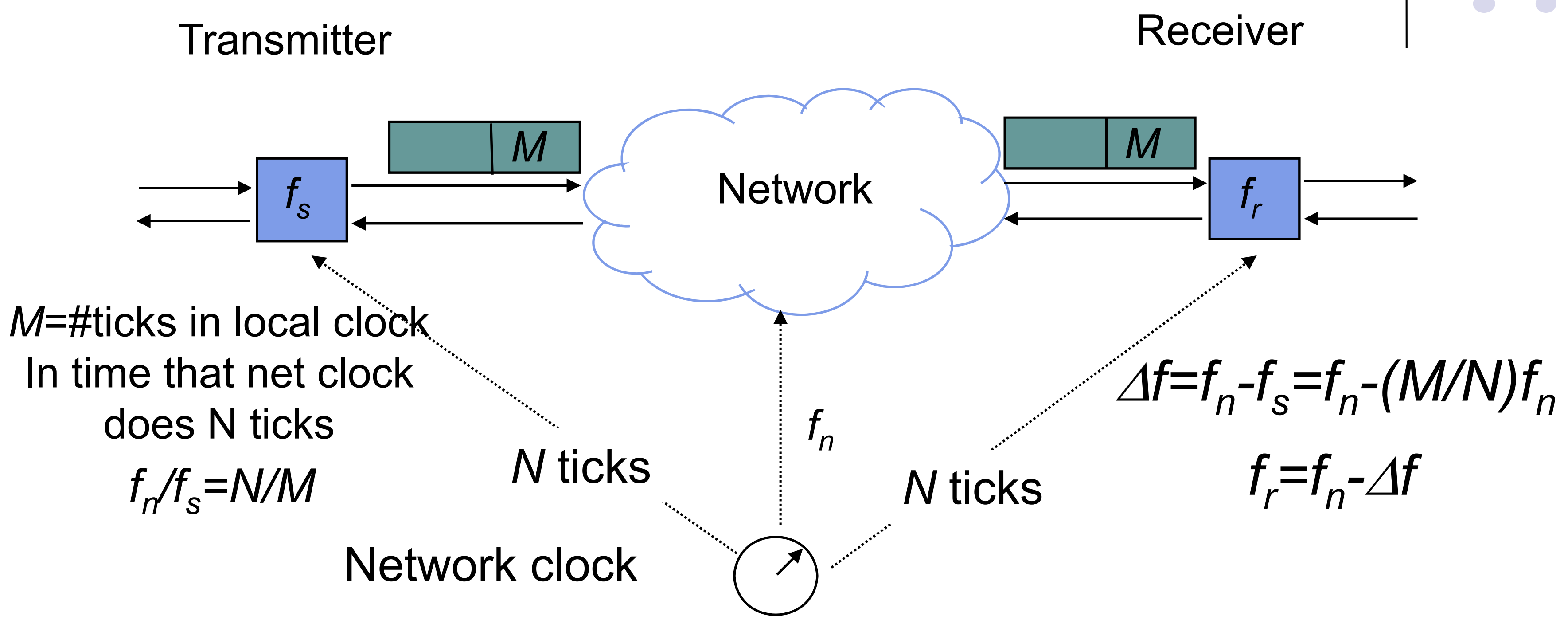



# Clock Recovery




- Counter attempts to replicate transmitter clock
- Frequency of counter is adjusted according to arriving timestamps
- Jitter introduced by network causes fluctuations in buffer & in local clock

# Synchronization to a Common Clock




- Clock recovery simple if a common clock is available to transmitter & receiver
  - E.g. SONET network clock; Global Positioning System (GPS)
- Transmitter sends $\Delta f$ of its frequency & network frequency
- Receiver adjusts network frequency by $\Delta f$
- Packet delay jitter can be removed completely

# Example:  Real-Time Protocol

- RTP (RFC 1889) designed to support real-time applications such as voice, audio, video
- RTP provides means to carry:
  - Type of information source
  - Sequence numbers
  - Timestamps
- Actual timing recovery must be done by higher layer protocol
  - MPEG2 for video, MP3 for audio

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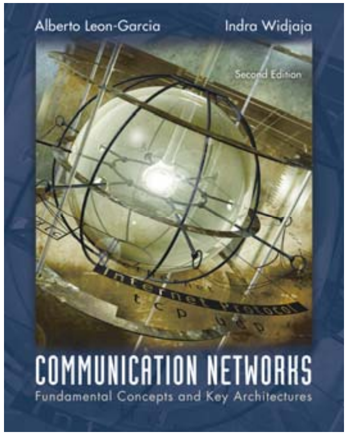
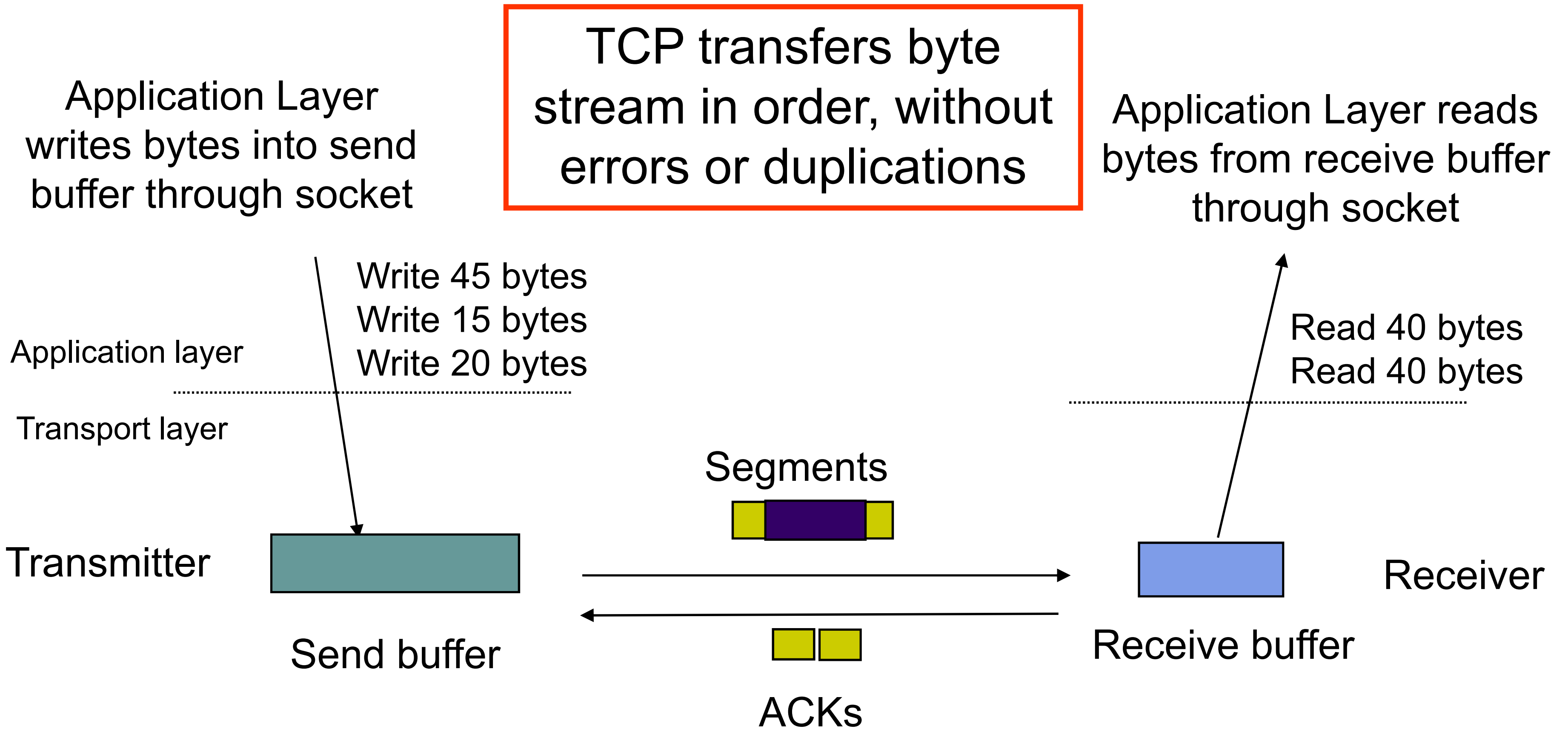



## *TCP Reliable Stream Service & Flow Control*

# TCP Reliable Stream Service

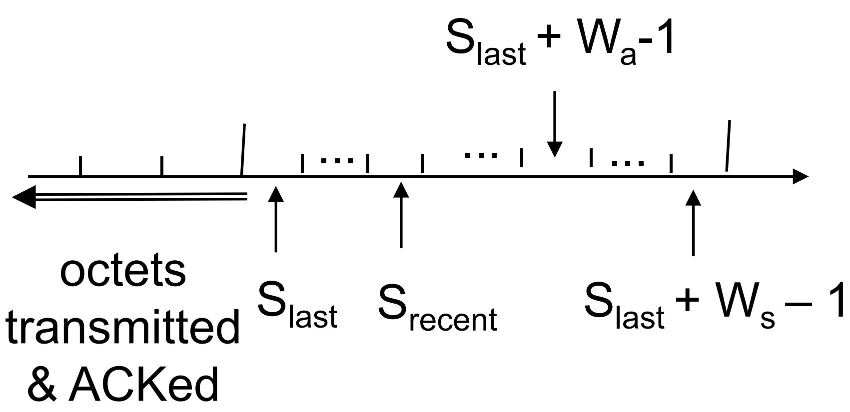


# TCP ARQ Method

- TCP uses *Selective Repeat ARQ*
  - Transfers byte stream without preserving boundaries
- Operates over best effort service of IP
  - Packets can arrive with errors or be lost
  - Packets can arrive out-of-order
  - Packets can arrive after very long delays
  - Duplicate segments must be detected & discarded
  - Must protect against segments from previous connections
- Sequence Numbers
  - Seq. # is number of first byte in segment payload
  - Very long Seq. #s (32 bits) to deal with long delays
  - Initial sequence numbers negotiated during connection setup (to deal with very old duplicates)
  - Accept segments within a receive window

## Transmitter

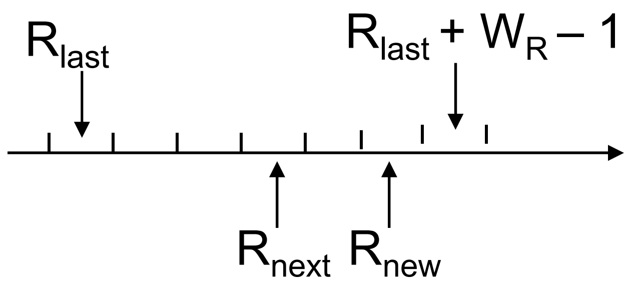
Send Window




$S_{last}$ oldest unacknowledged byte
$S_{recent}$ highest-numbered transmitted byte
$S_{last}+W_a-1$ highest-numbered byte that can be transmitted
$S_{last}+W_s-1$ highest-numbered byte that can be accepted from the application

## Receiver

Receive Window

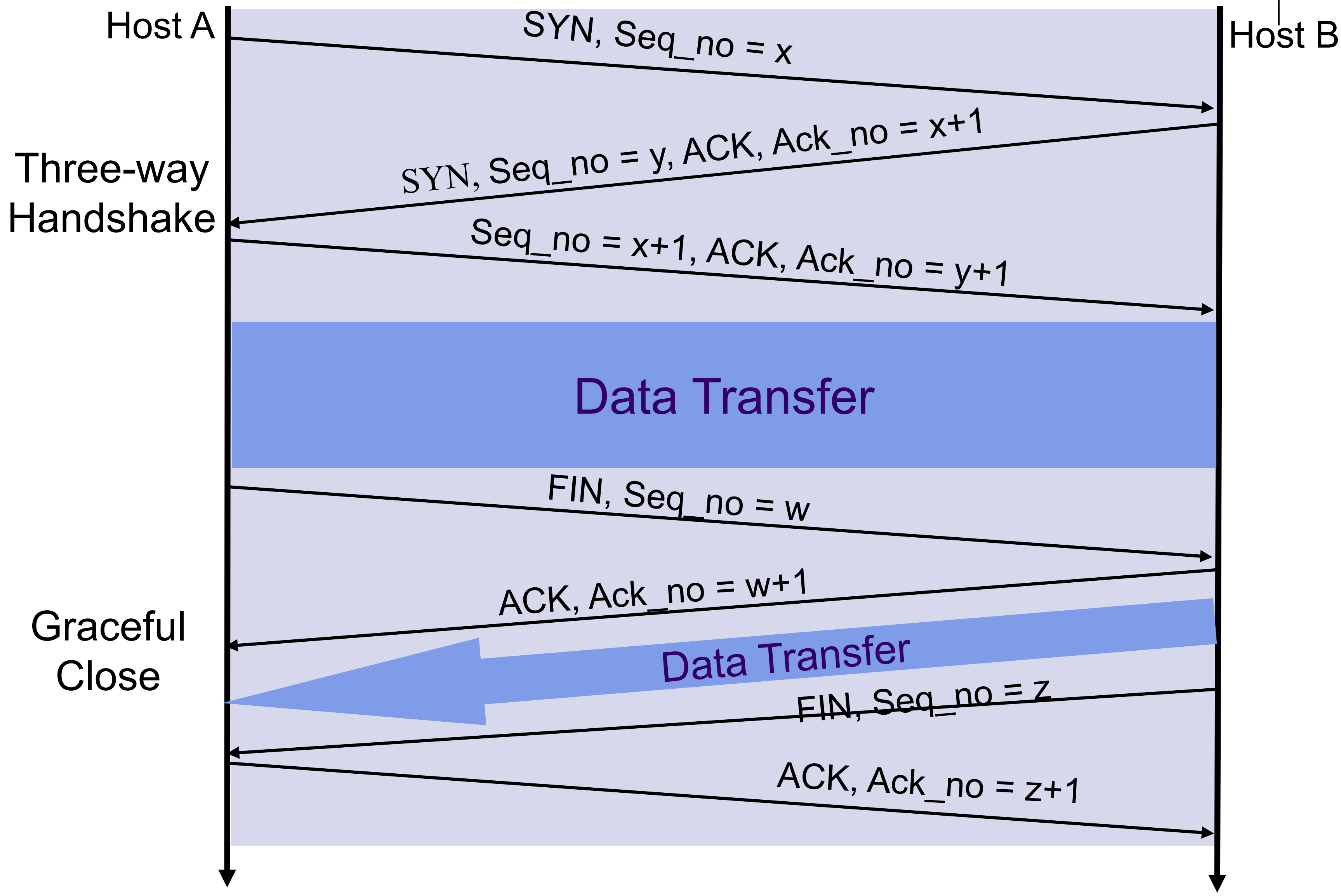



$R_{last}$ highest-numbered byte not yet read by the application
$R_{next}$ next expected byte
$R_{new}$ highest numbered byte received correctly
$R_{last}+W_R-1$ highest-numbered byte that can be accommodated in receive buffer

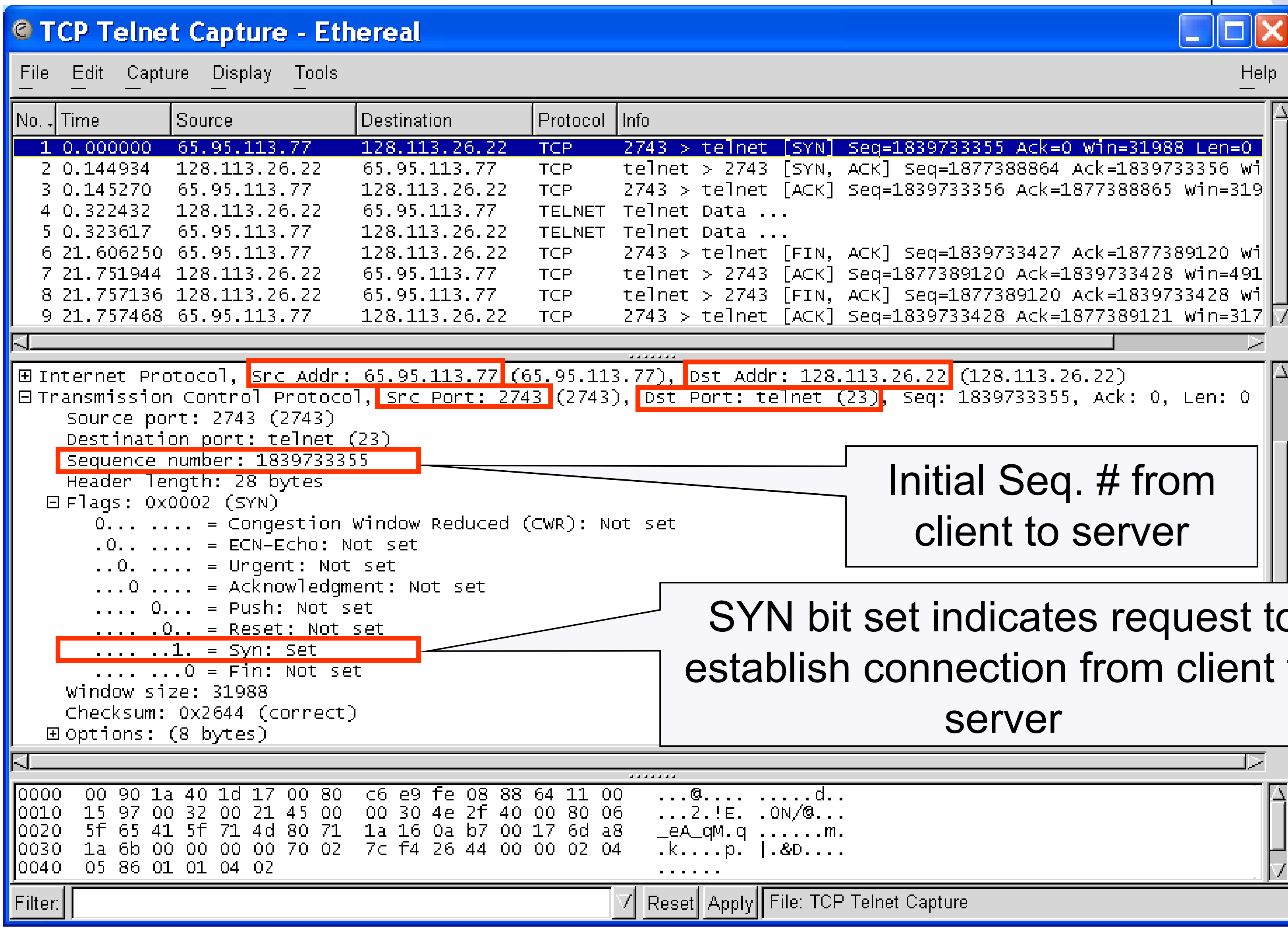
# TCP Connections

- TCP Connection
  - One connection each way
  - Identified uniquely by Send IP Address, Send TCP Port #, Receive IP Address, Receive TCP Port #
- Connection Setup with Three-Way Handshake
  - Three-way exchange to negotiate initial Seq. #'s for connections in each direction
- Data Transfer
  - Exchange segments carrying data
- Graceful Close
  - Close each direction separately

# Three Phases of TCP Connection



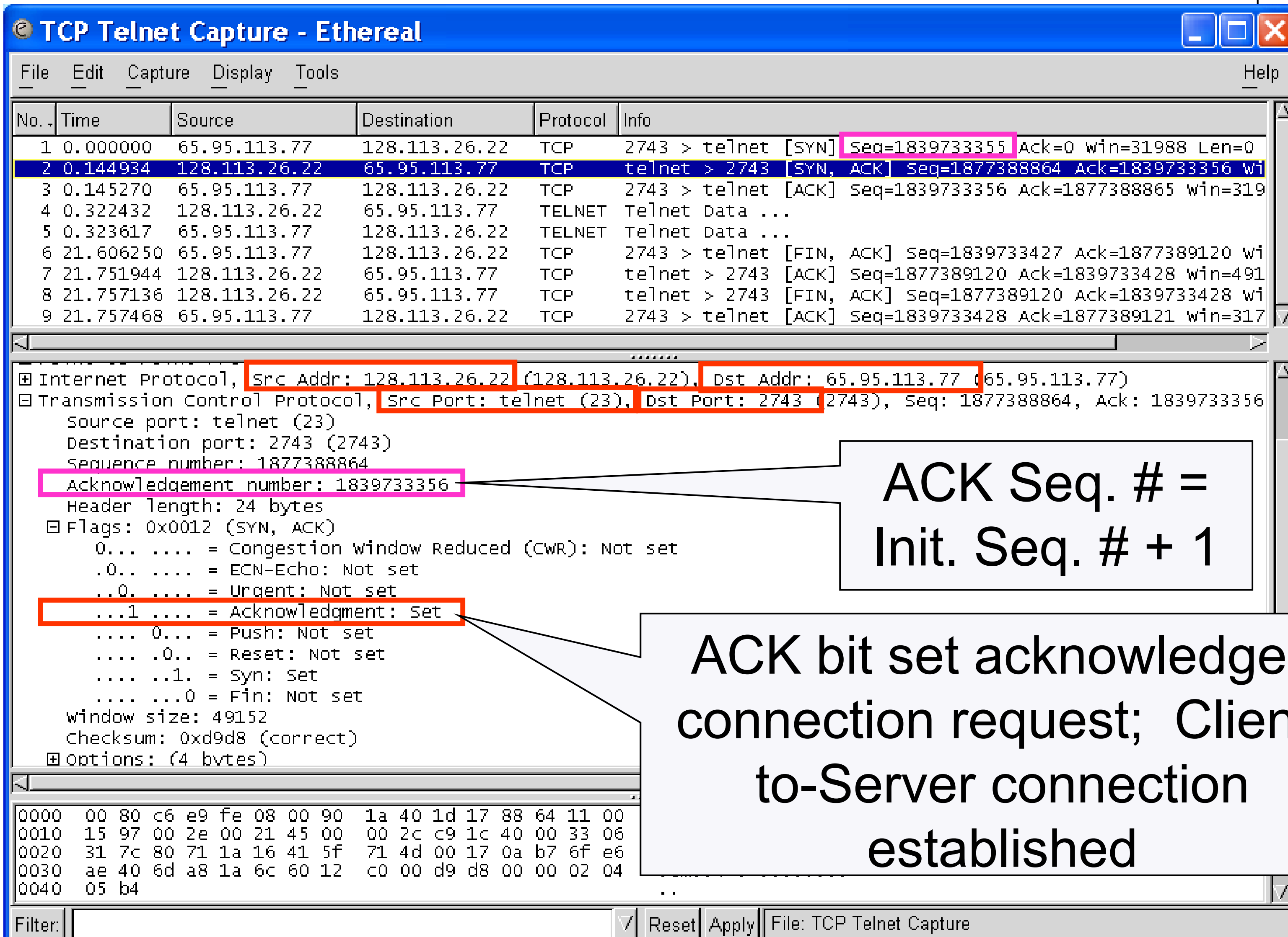
# 1st Handshake: Client-Server Connection Request




Initial Seq. # from client to server

SYN bit set indicates request to establish connection from client to server

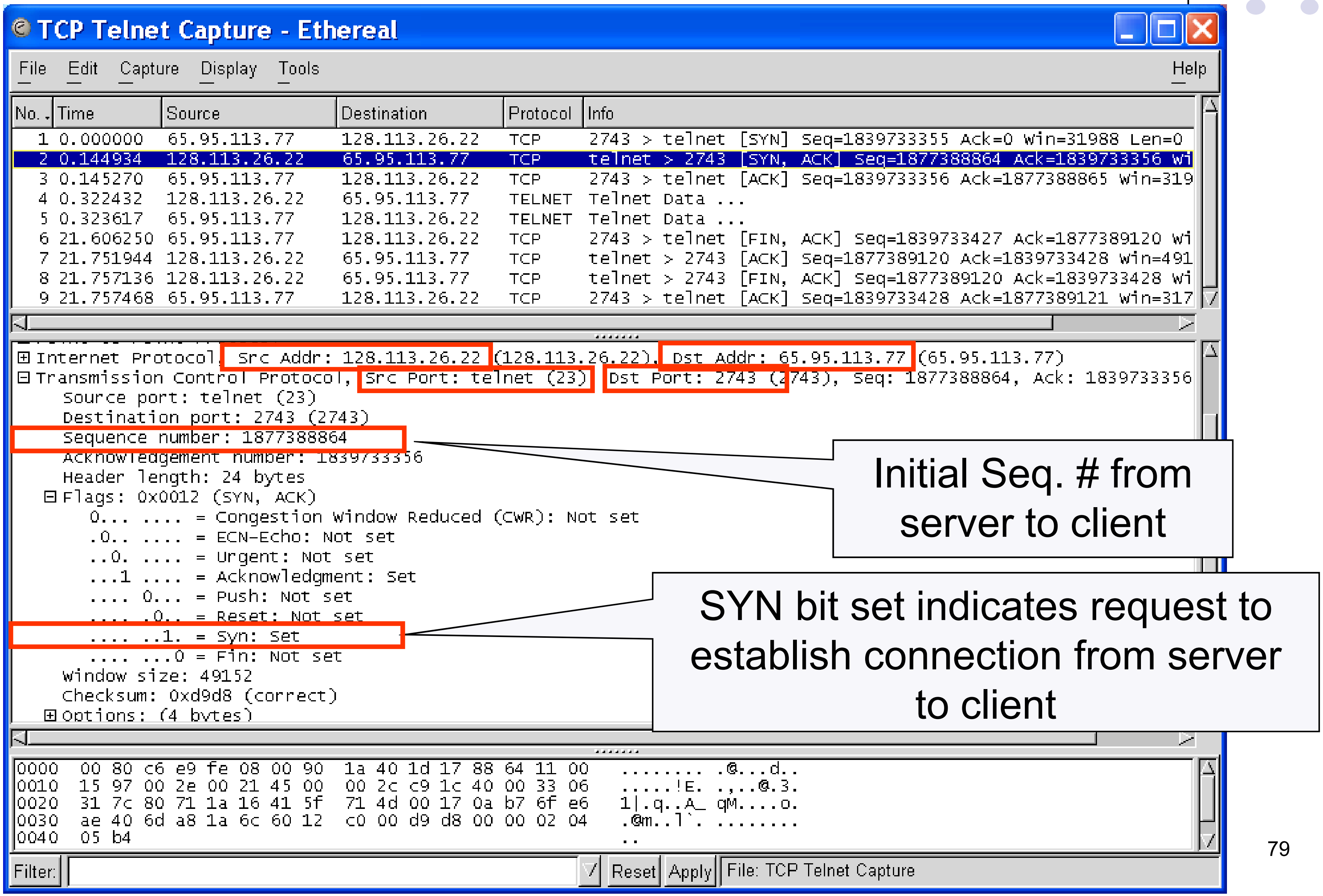
# 2nd Handshake:  ACK from Server



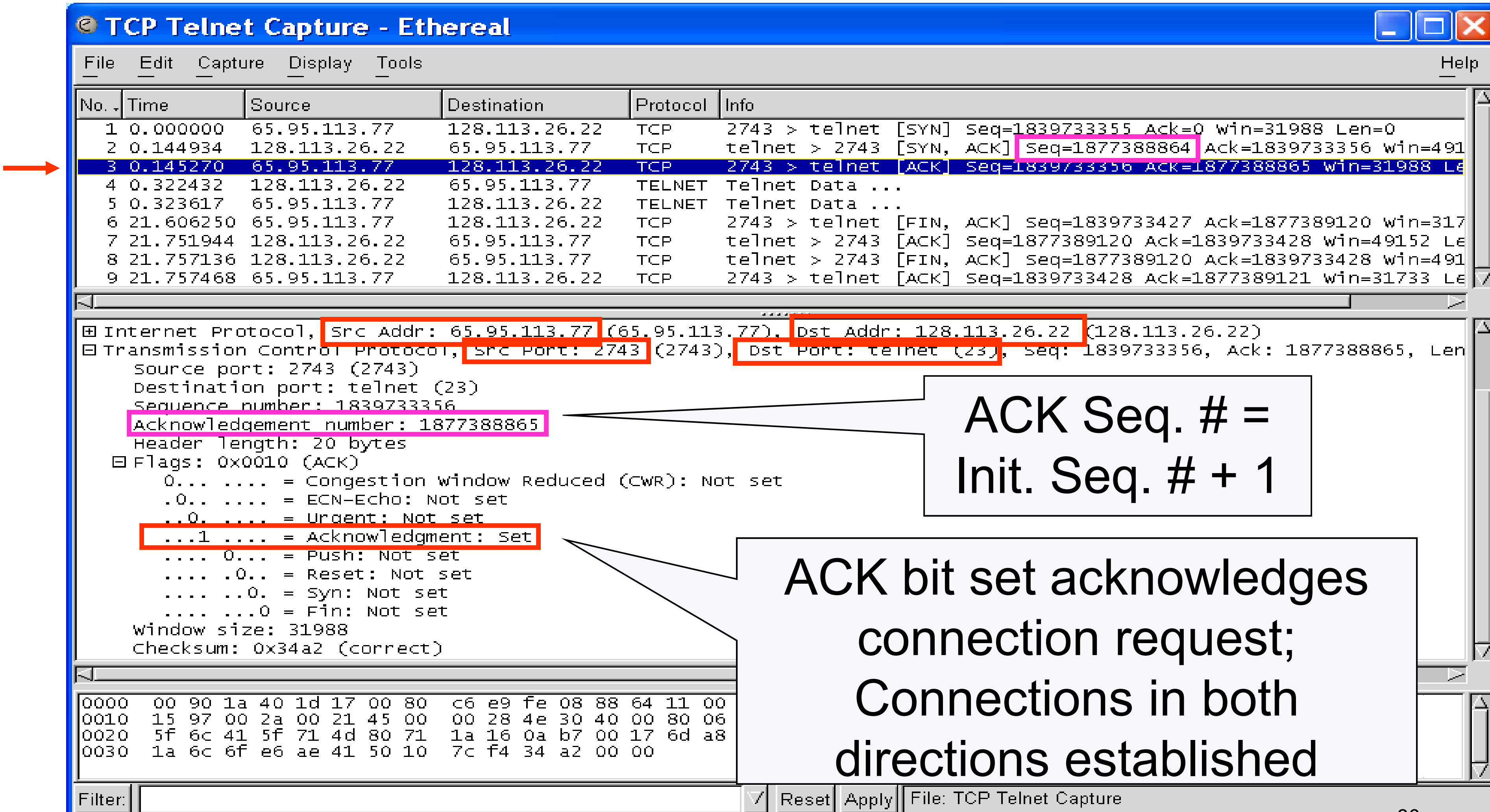

ACK Seq. # = Init. Seq. # + 1

ACK bit set acknowledges connection request;  Client-to-Server connection established

# 2nd Handshake: Server-Client Connection Request



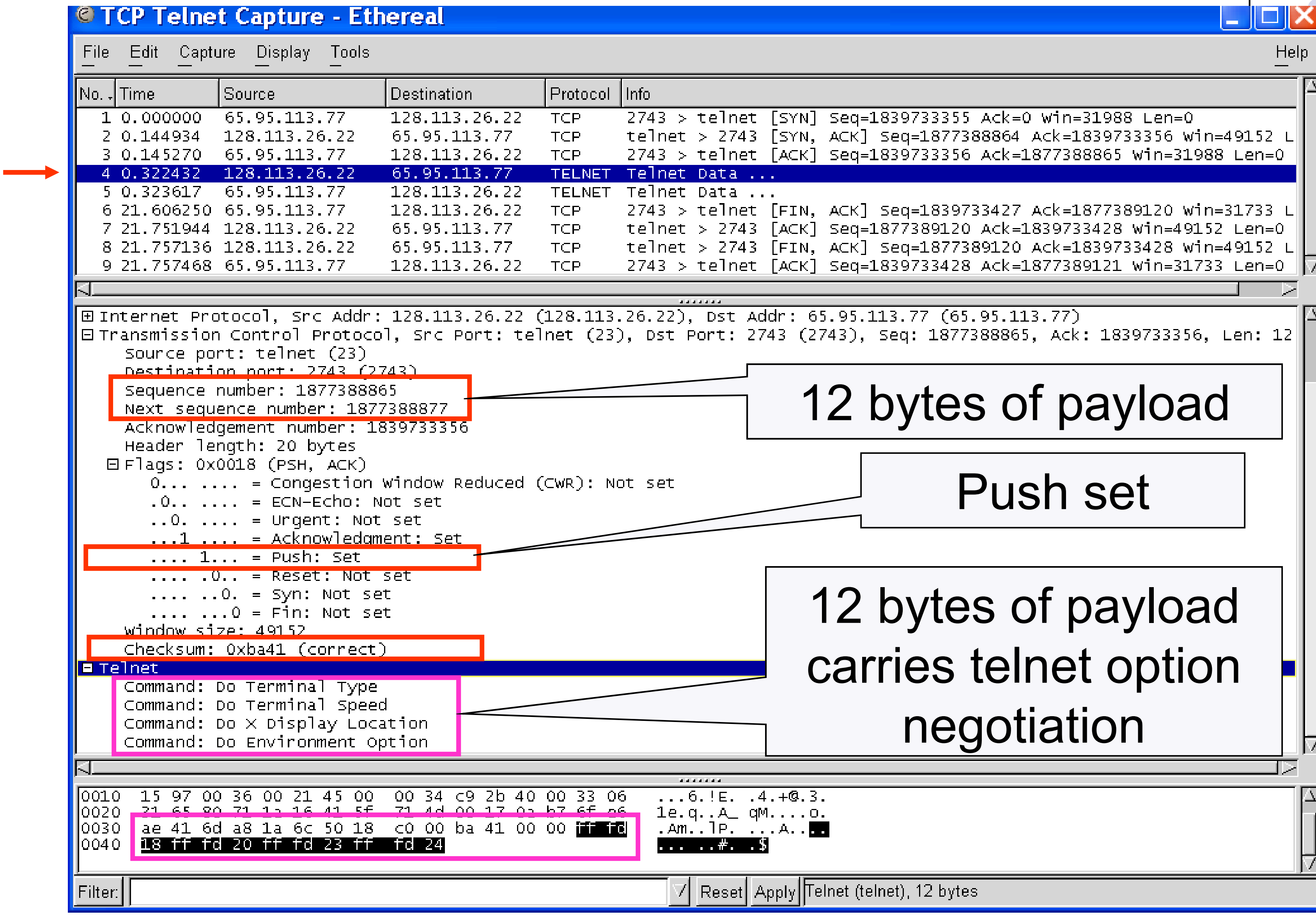
# 3rd Handshake: ACK from Client



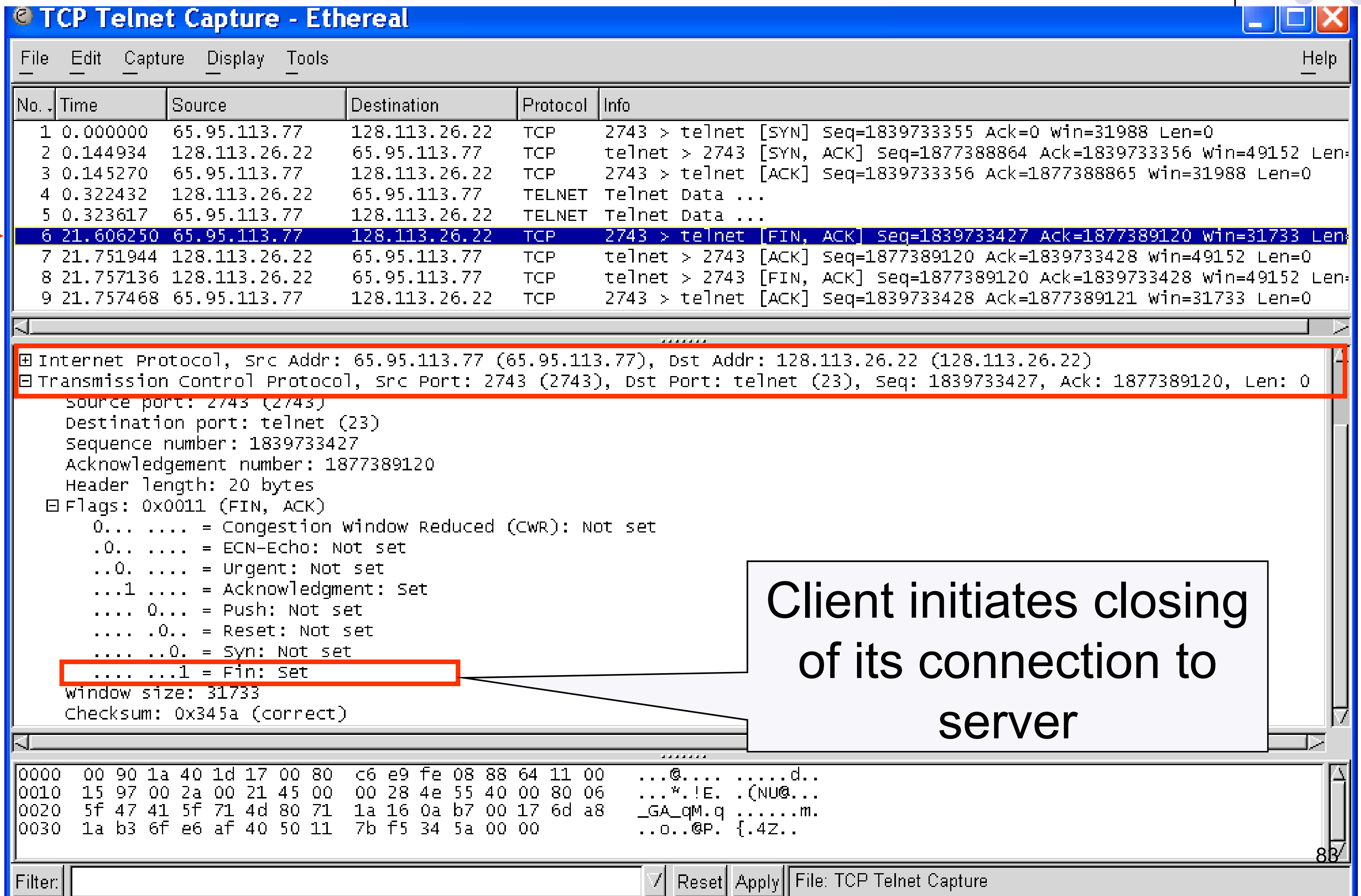
# TCP Data Exchange

- Application Layers write bytes into buffers
- TCP sender forms segments
  - When bytes exceed threshold or timer expires
  - Upon PUSH command from applications
  - Consecutive bytes from buffer inserted in payload
  - Sequence # & ACK # inserted in header
  - Checksum calculated and included in header
- TCP receiver
  - Performs selective repeat ARQ functions
  - Writes error-free, in-sequence bytes to receive buffer

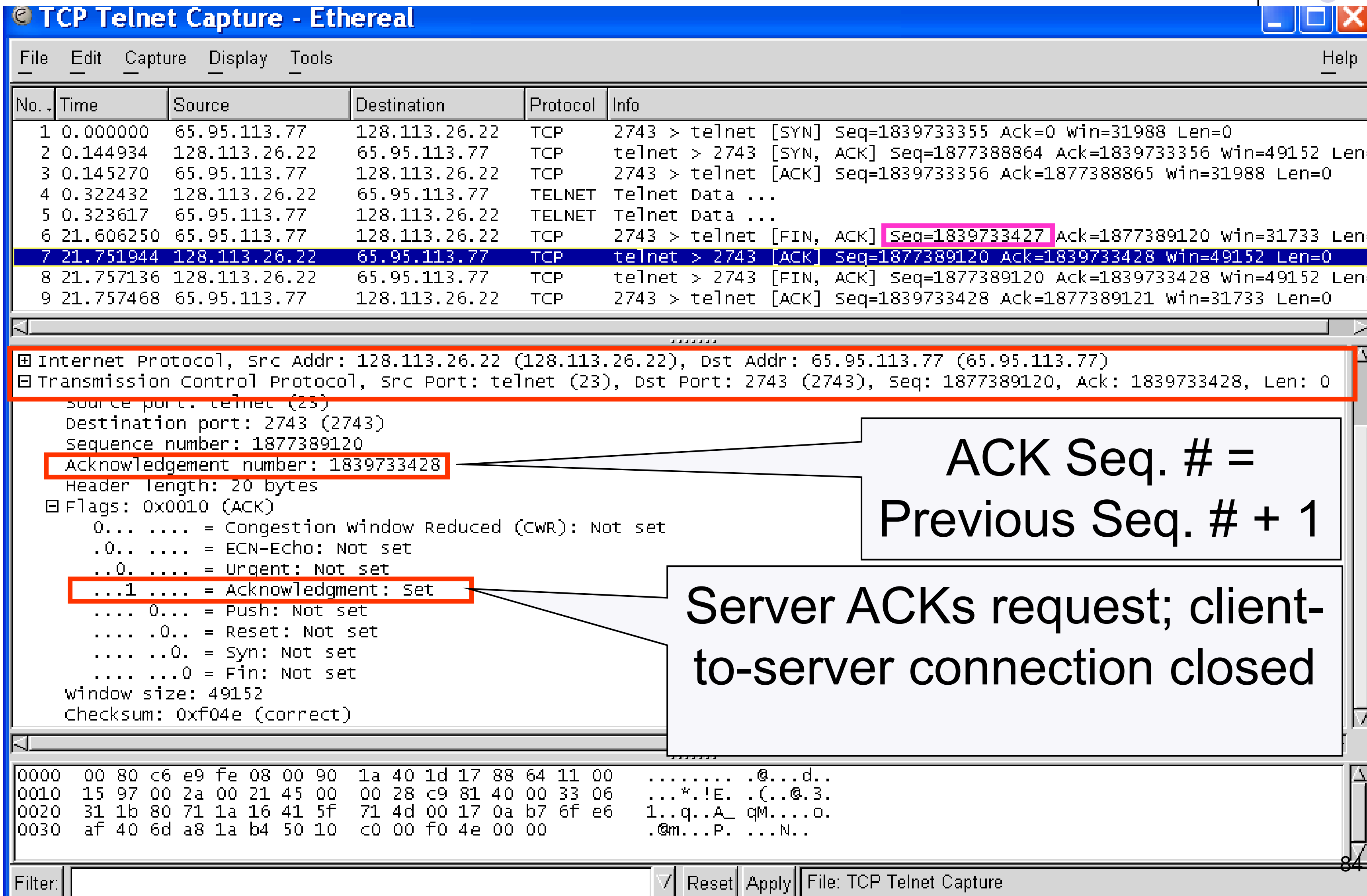
# Data Transfer: Server-to-Client Segment



# Graceful Close: Client-to-Server Connection



# Graceful Close: Client-to-Server Connection




ACK Seq. # = Previous Seq. # + 1

Server ACKs request; client-to-server connection closed

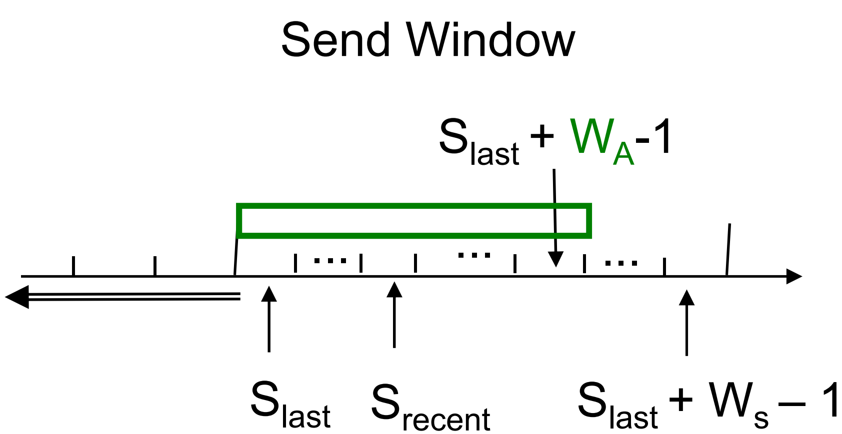
# Flow Control

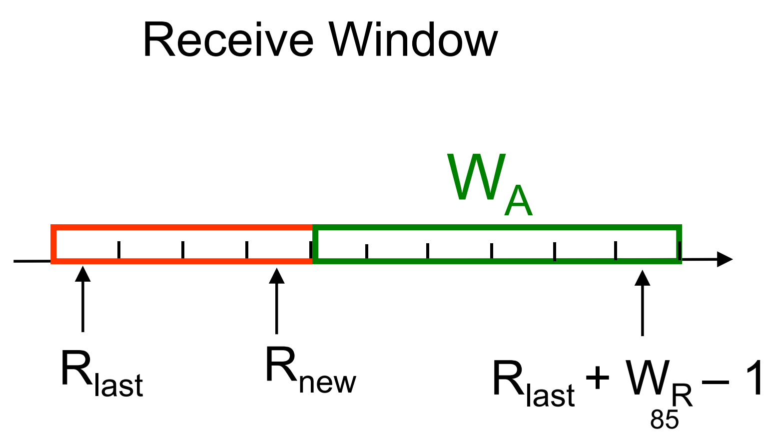
- TCP receiver controls rate at which sender transmits to prevent buffer overflow
- TCP receiver advertises a window size specifying number of bytes that can be accommodated by receiver

$$W_A = W_R - (R_{new} - R_{last})$$

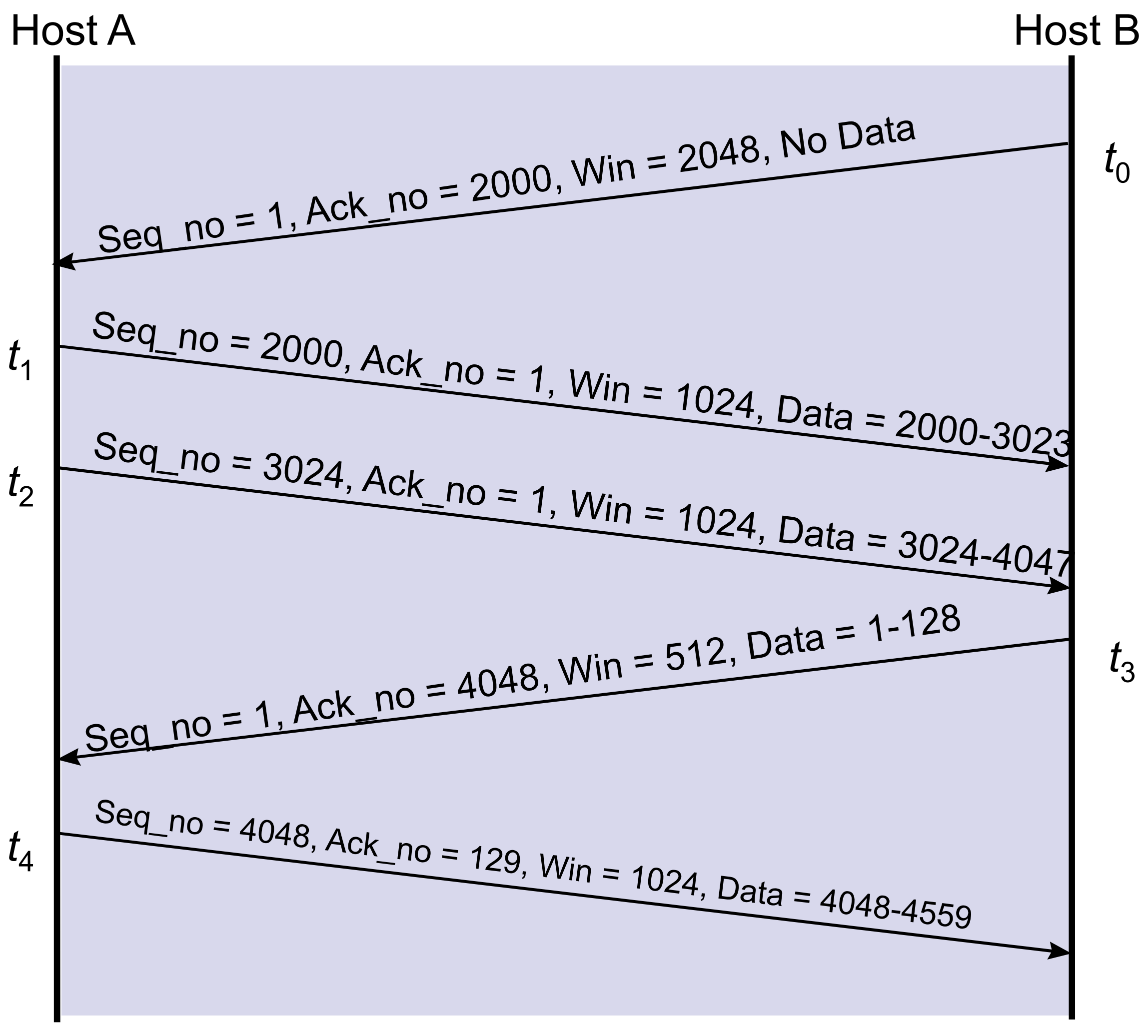
- TCP sender obliged to keep # outstanding bytes below $W_A$

$$(S_{recent} - S_{last}) \leq W_A$$



# TCP window flow control



# TCP Retransmission Timeout

- TCP retransmits a segment after timeout period
  - Timeout too short: excessive number of retransmissions
  - Timeout too long: recovery too slow
  - Timeout depends on RTT: time from when segment is sent to when ACK is received
- Round trip time (RTT) in Internet is highly variable
  - Routes vary and can change in mid-connection
  - Traffic fluctuates
- TCP uses adaptive estimation of RTT
  - Measure RTT each time ACK received: $\tau_n$

$$t_{RTT}(\text{new}) = \alpha\, t_{RTT}(\text{old}) + (1 - \alpha)\, \tau_n$$

  - $\alpha = 7/8$ typical

# RTT Variability

- Estimate variance $\sigma^2$ of RTT variation
- Estimate for timeout:

$$t_{out} = t_{RTT} + k\ \sigma_{RTT}$$

- If RTT highly variable, timeout increase accordingly
- If RTT nearly constant, timeout close to RTT estimate

- Approximate estimation of deviation

$$d_{RTT}(new) = \beta\ d_{RTT}(old) + (1-\beta)\ |\ \tau_n - t_{RTT}\ |$$

$$t_{out} = t_{RTT} + 4\ d_{RTT}$$