CSE3213 Computer Network I

Service Model, Error Control, Flow Control, and Link Sharing

(Ch. 5.1 – 5.3.1 and 5.7.1)

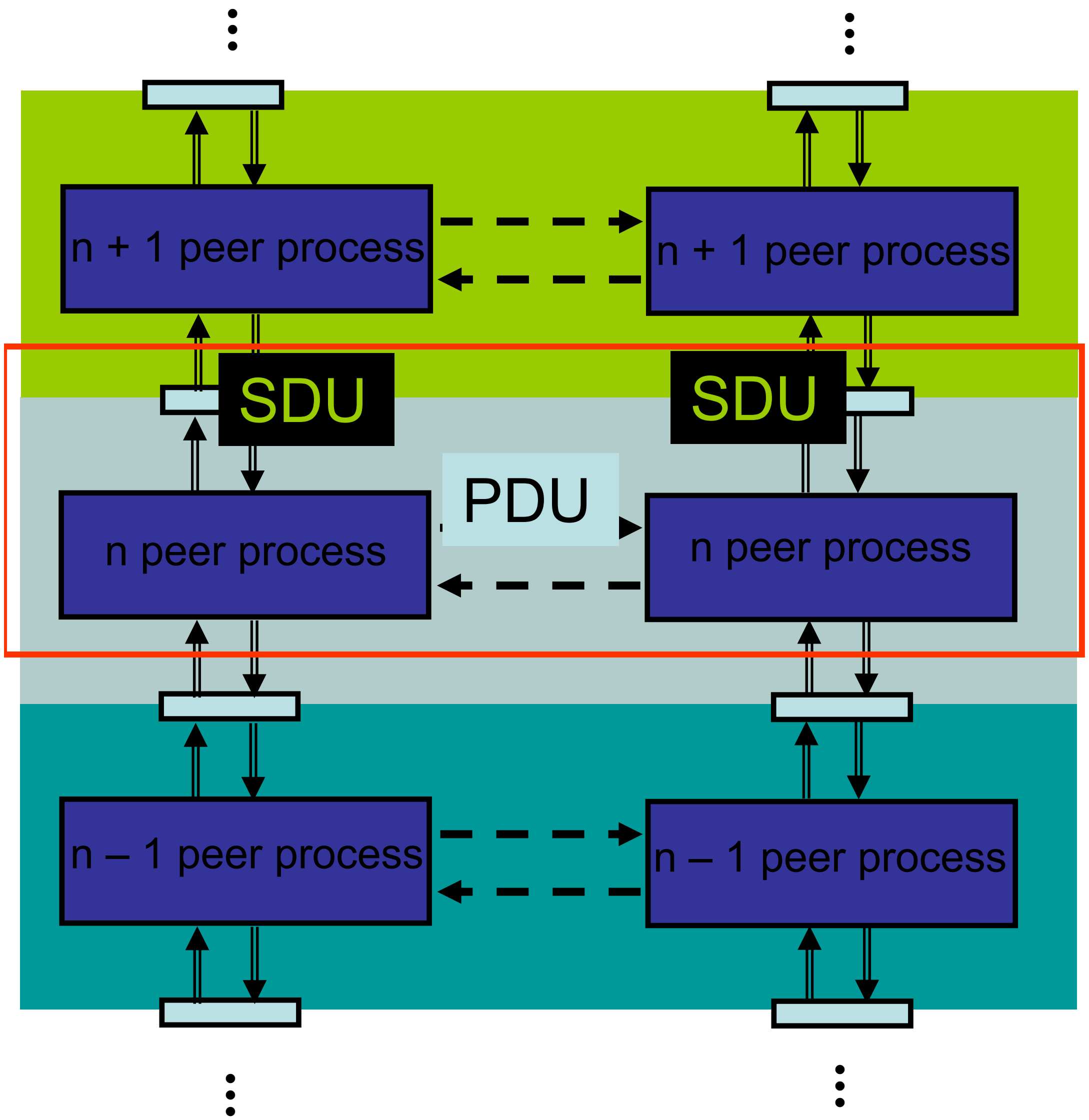
Course page:

http://www.cse.yorku.ca/course/3213

Slides modified from Alberto Leon-Garcia and Indra Widjaja

Peer-to-Peer Protocols and Service Models

Peer-to-Peer Protocols



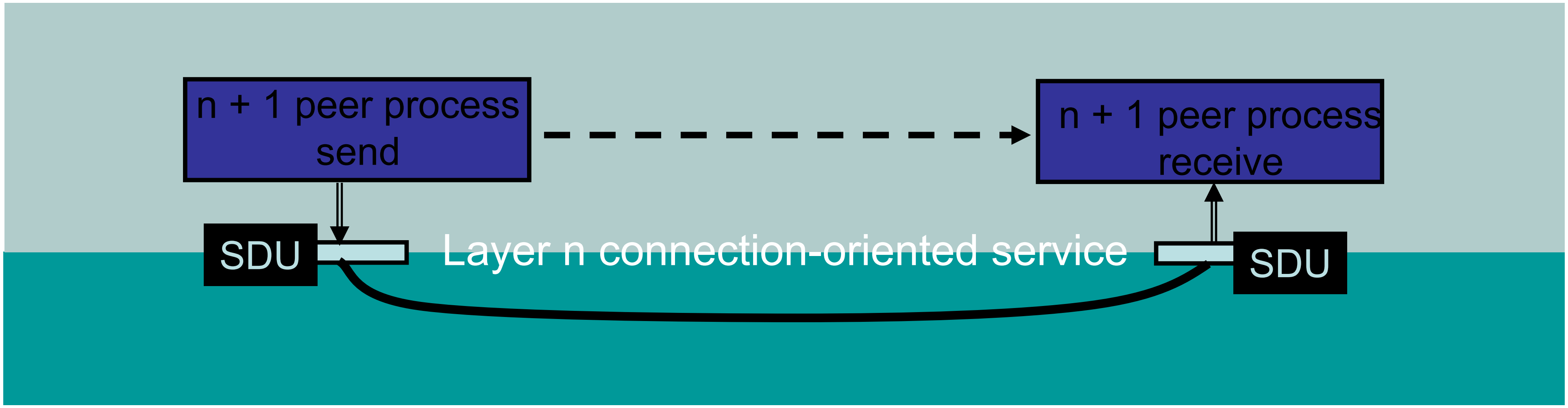
- *Peer-to-Peer processes* execute layer-n protocol to provide service to layer-(n+1)
- Layer-(n+1) peer calls layer-n and passes Service Data Units (SDUs) for transfer
- Layer-n peers exchange Protocol Data Units (PDUs) to effect transfer
- Layer-n delivers SDUs to destination layer-(n+1) peer

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

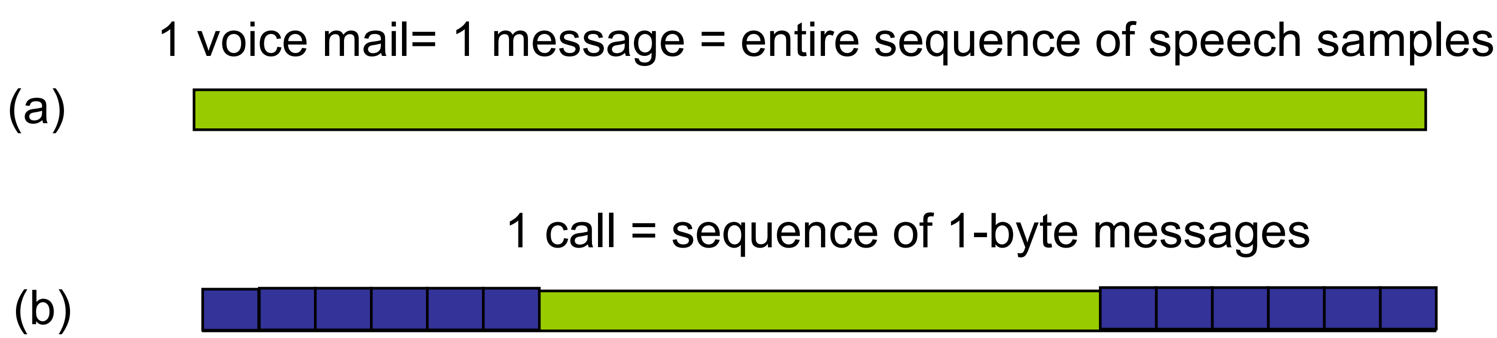


Connectionless Transfer Service

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

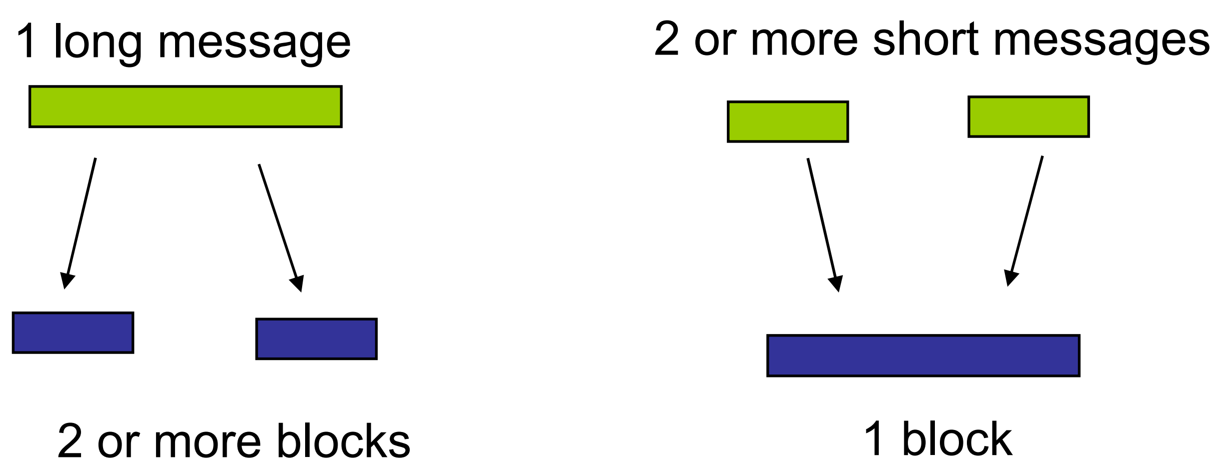
Message Size and Structure

- 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

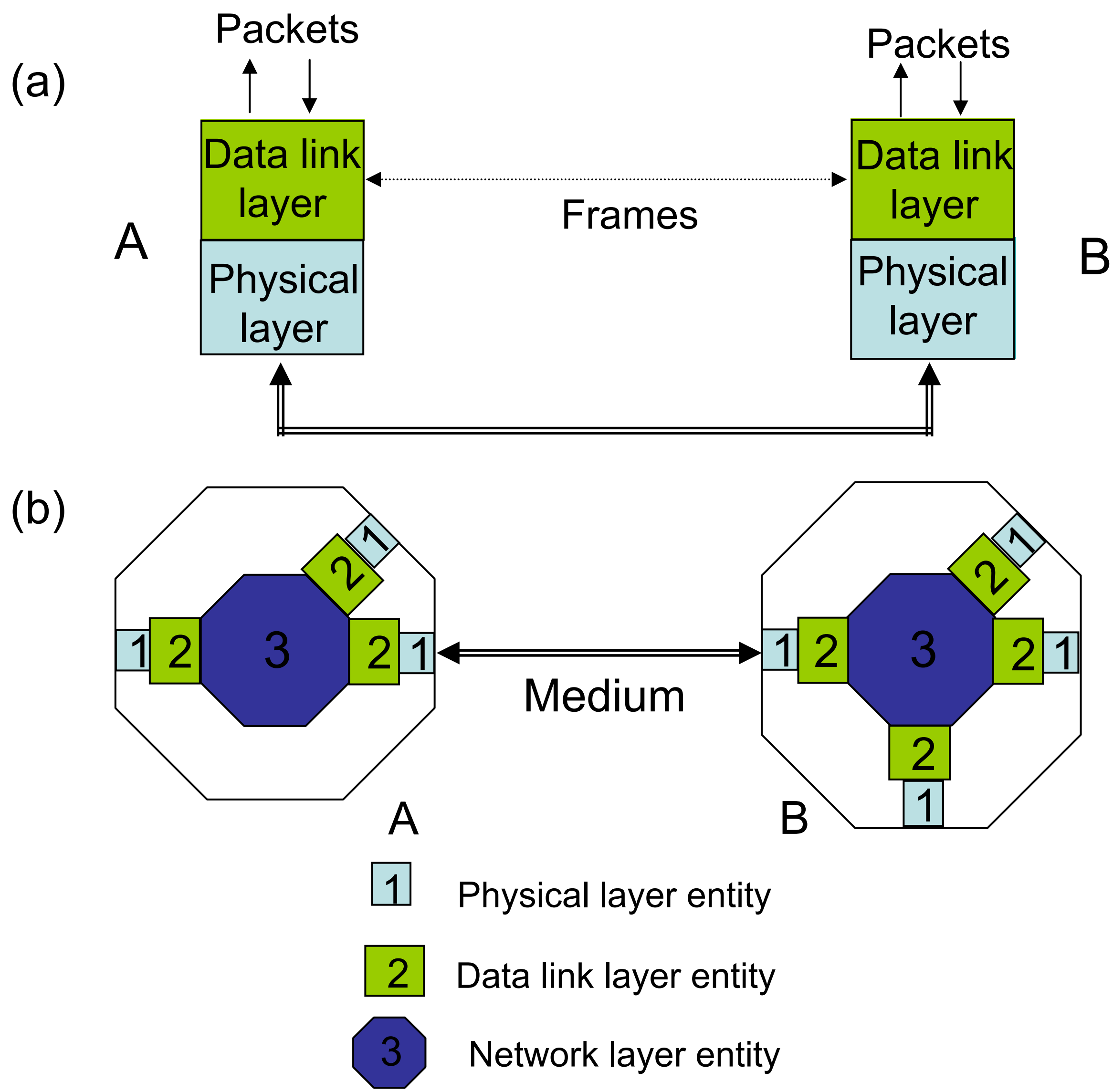
Privacy, Integrity, & Authentication

- *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 a single 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

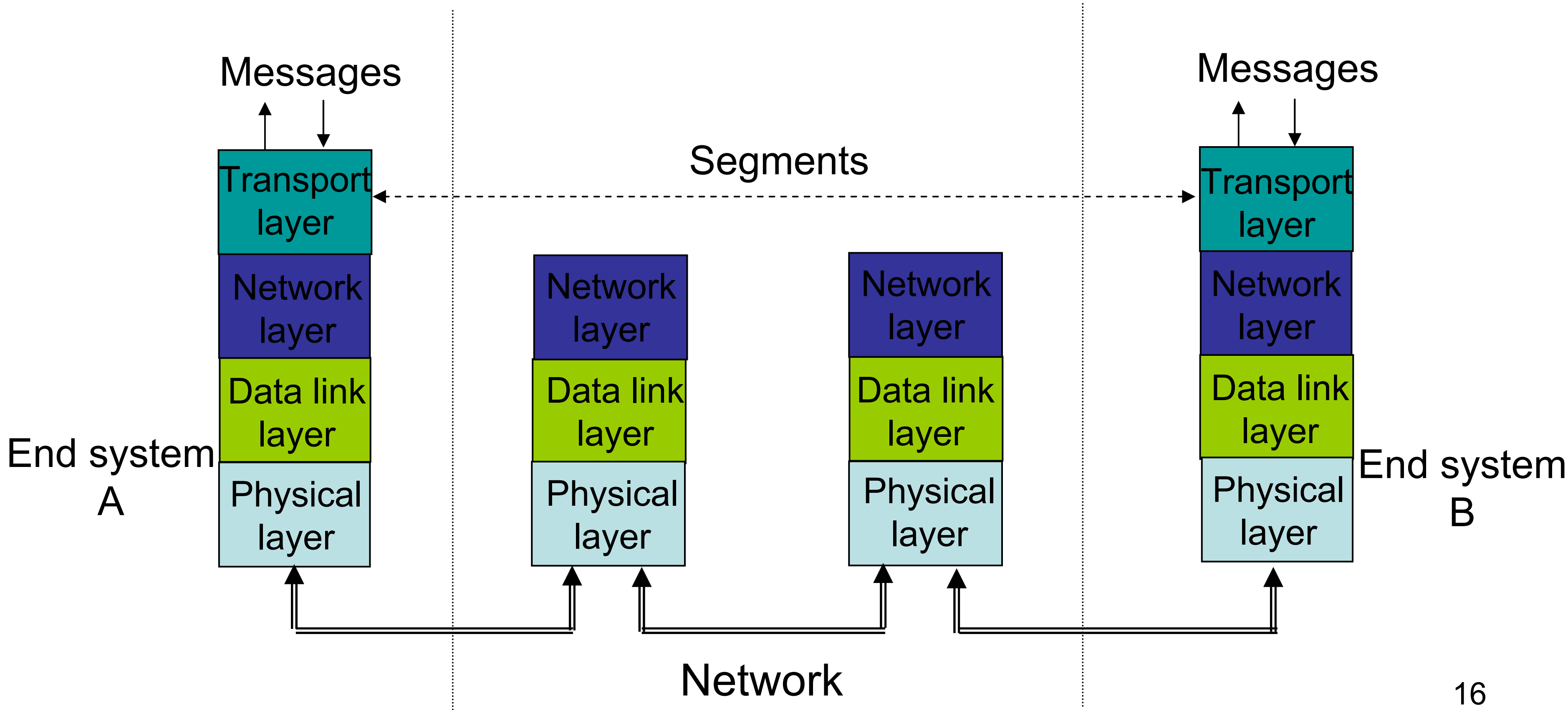
Error control in Data Link Layer



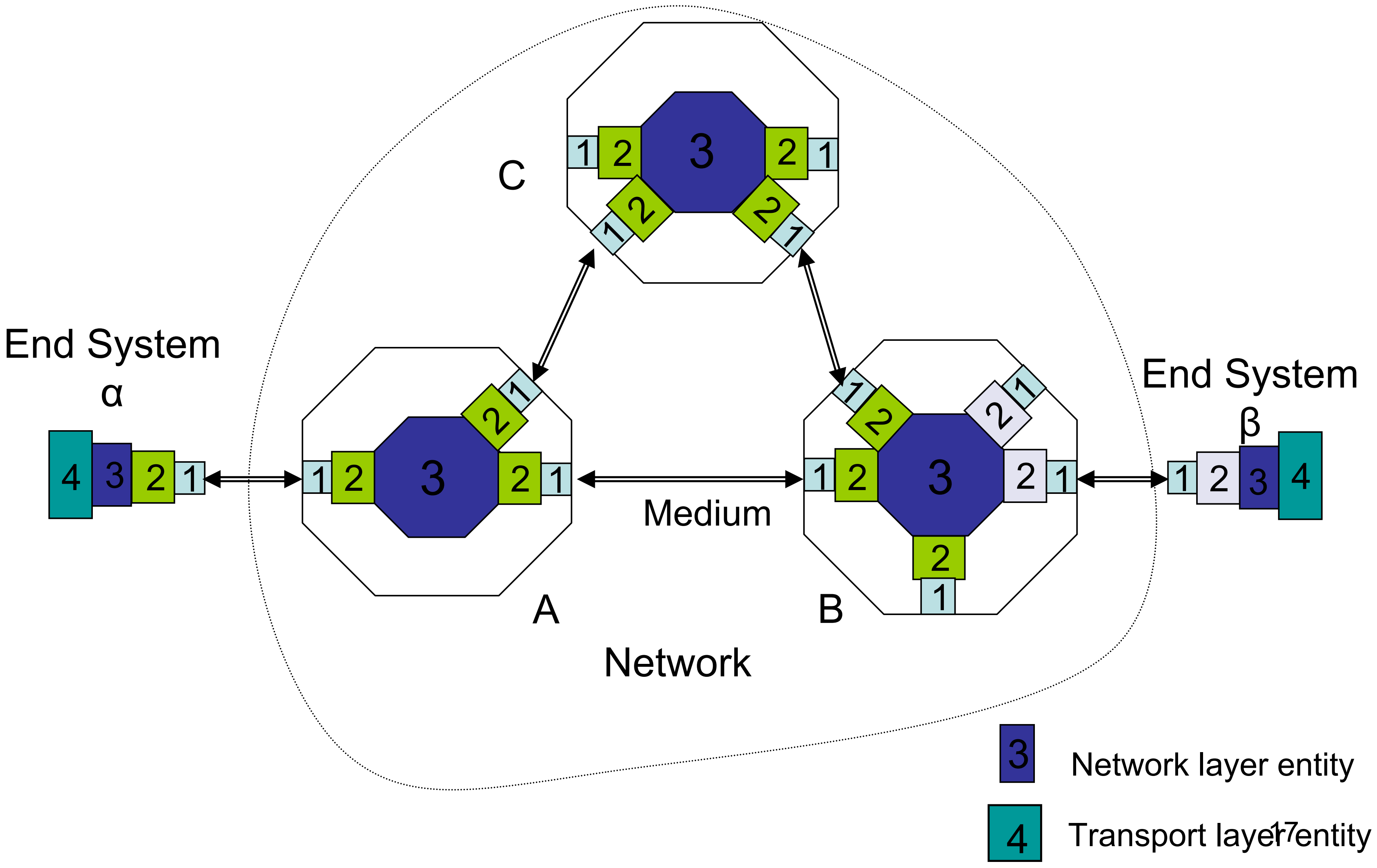
- Data Link operates over wire-like, directly-connected systems
- Frames can be corrupted or lost, but arrive in order
- Data link performs error-checking & retransmission
- Ensures error-free packet transfer between two systems

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

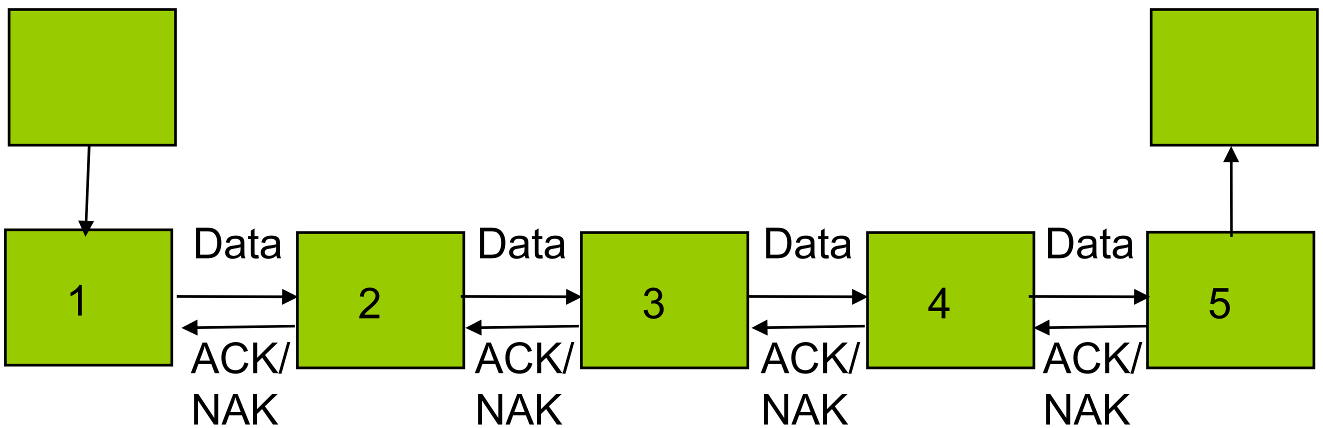


- Segments can experience long delays, can be lost, or arrive out-of-order because packets can follow different paths across network
- End-to-end error control protocol more difficult



End-to-End Approach Preferred

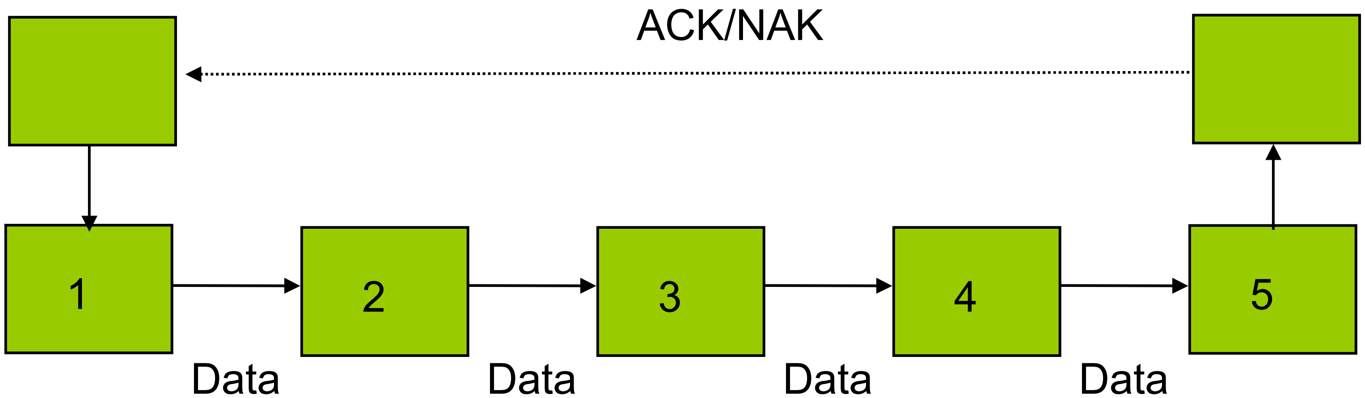
Hop-by-hop



Hop-by-hop cannot ensure E2E correctness

Faster recovery

End-to-end



Simple inside the network

More scalable if complexity at the edge

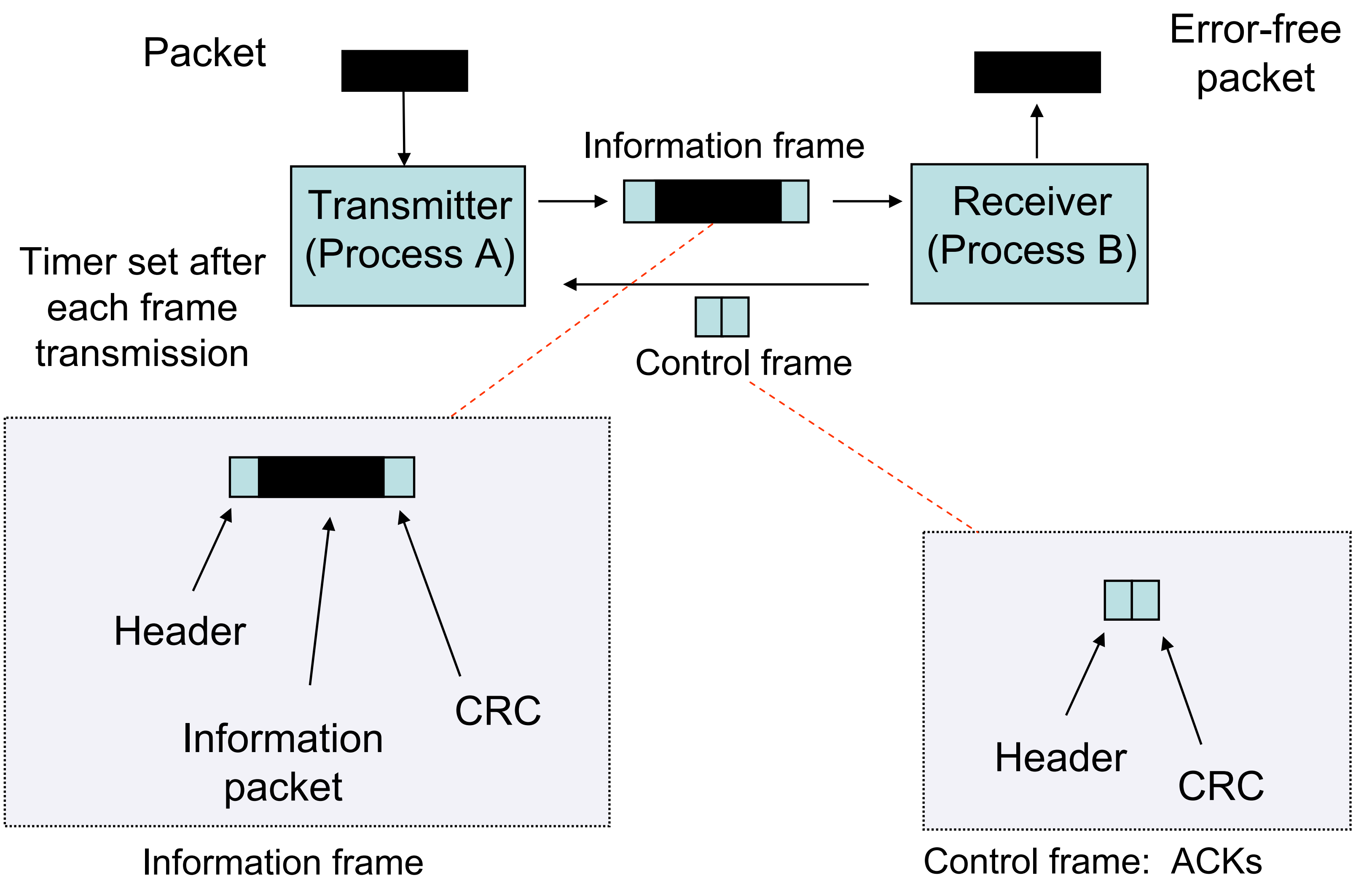
ARQ Protocols and Reliable Data Transfer

Automatic Repeat Request (ARQ)

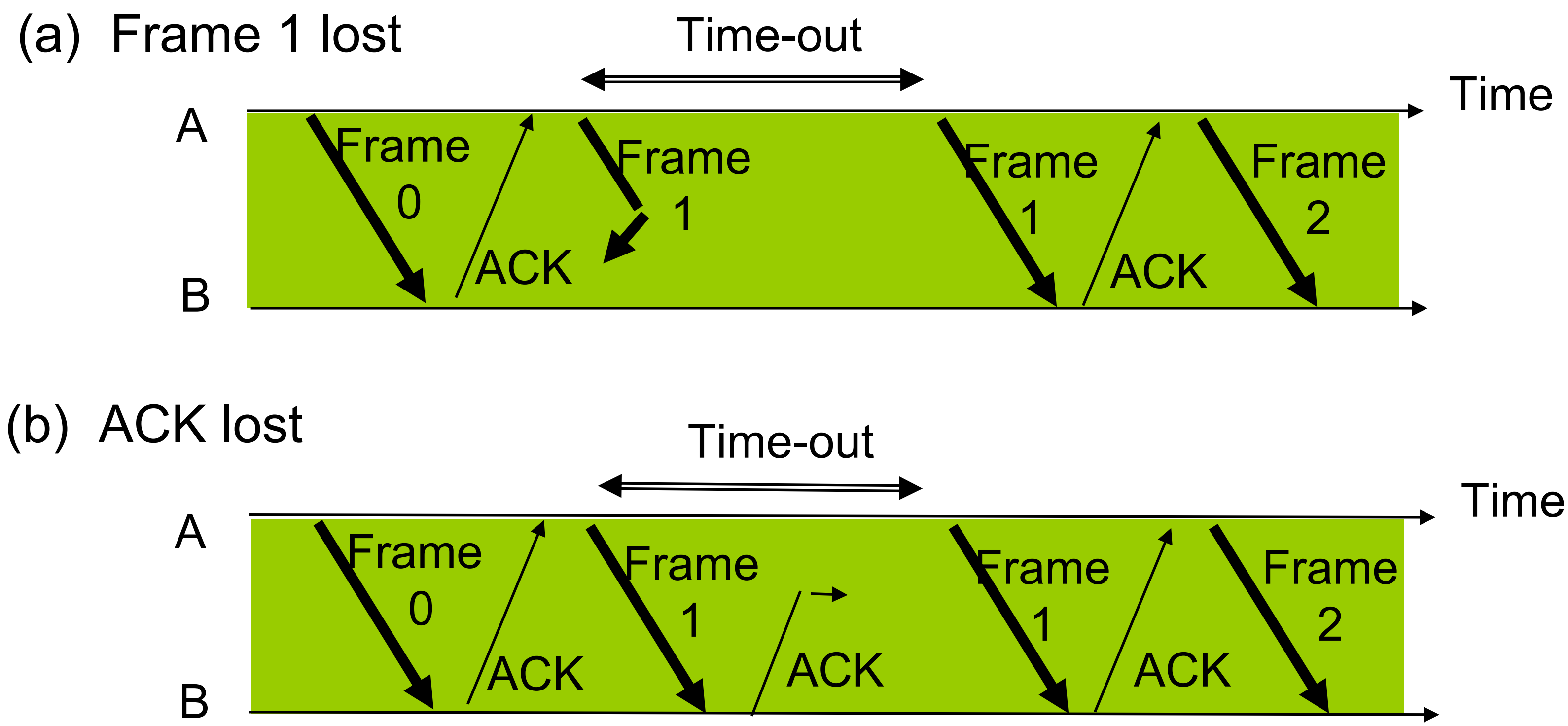
- *Purpose*: to ensure a sequence of information packets is delivered in order and without errors or duplications despite transmission errors & losses
- We will look at:
  - Stop-and-Wait ARQ
  - Go-Back N ARQ
  - Selective Repeat ARQ
- Basic elements of ARQ:
  - *Error-detecting code* with high error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

Stop-and-Wait ARQ

Transmit a frame, wait for ACK



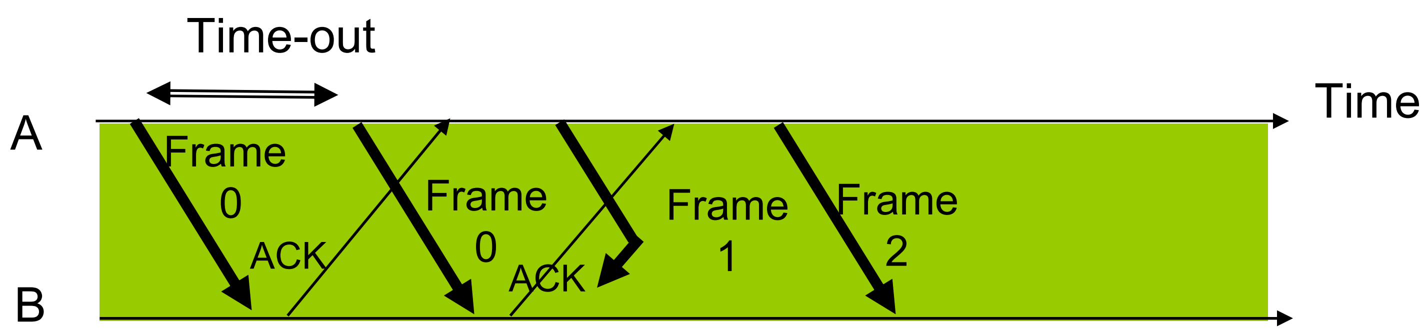
Need for Sequence Numbers



- 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

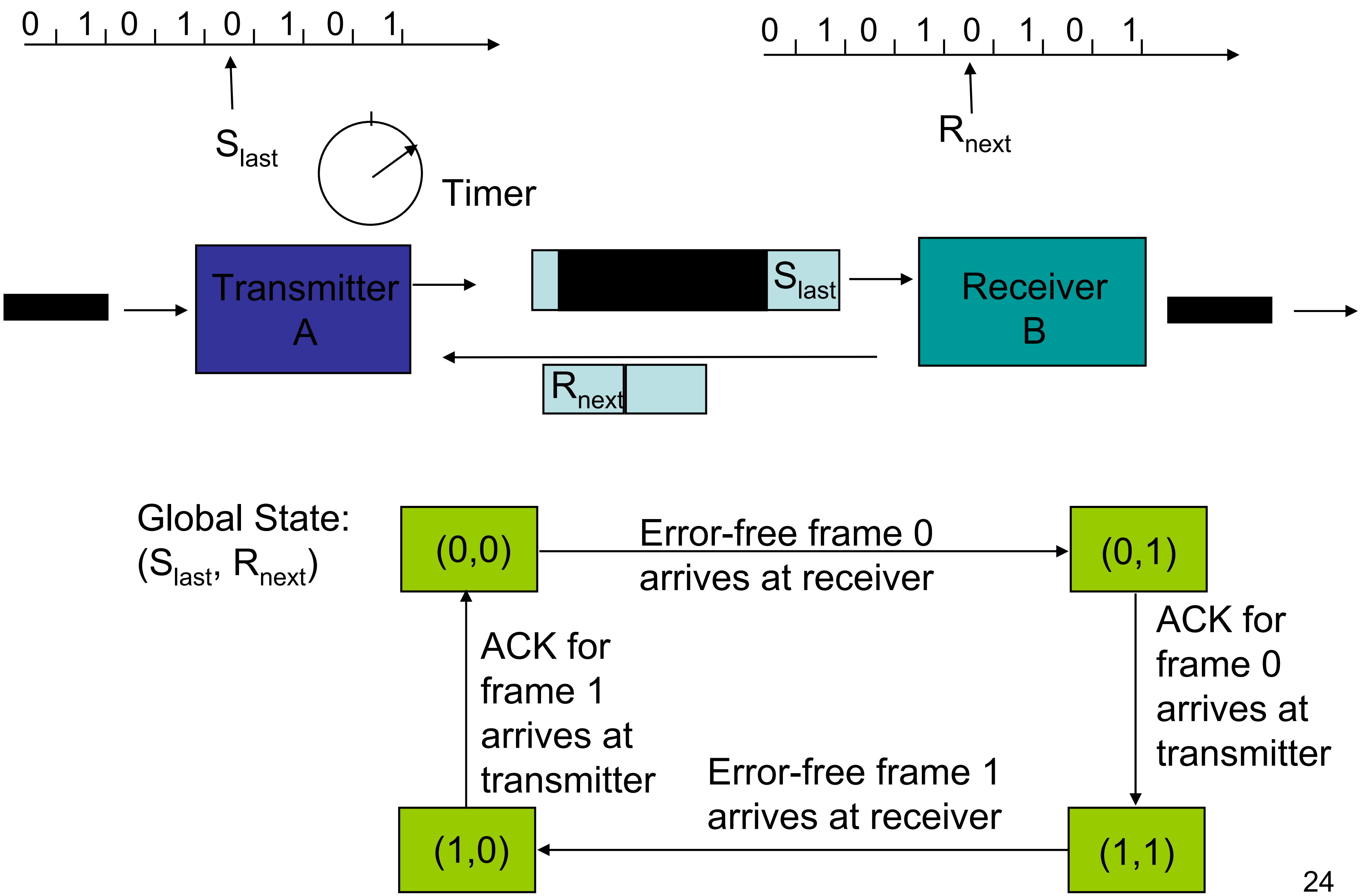
Sequence Numbers

(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



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

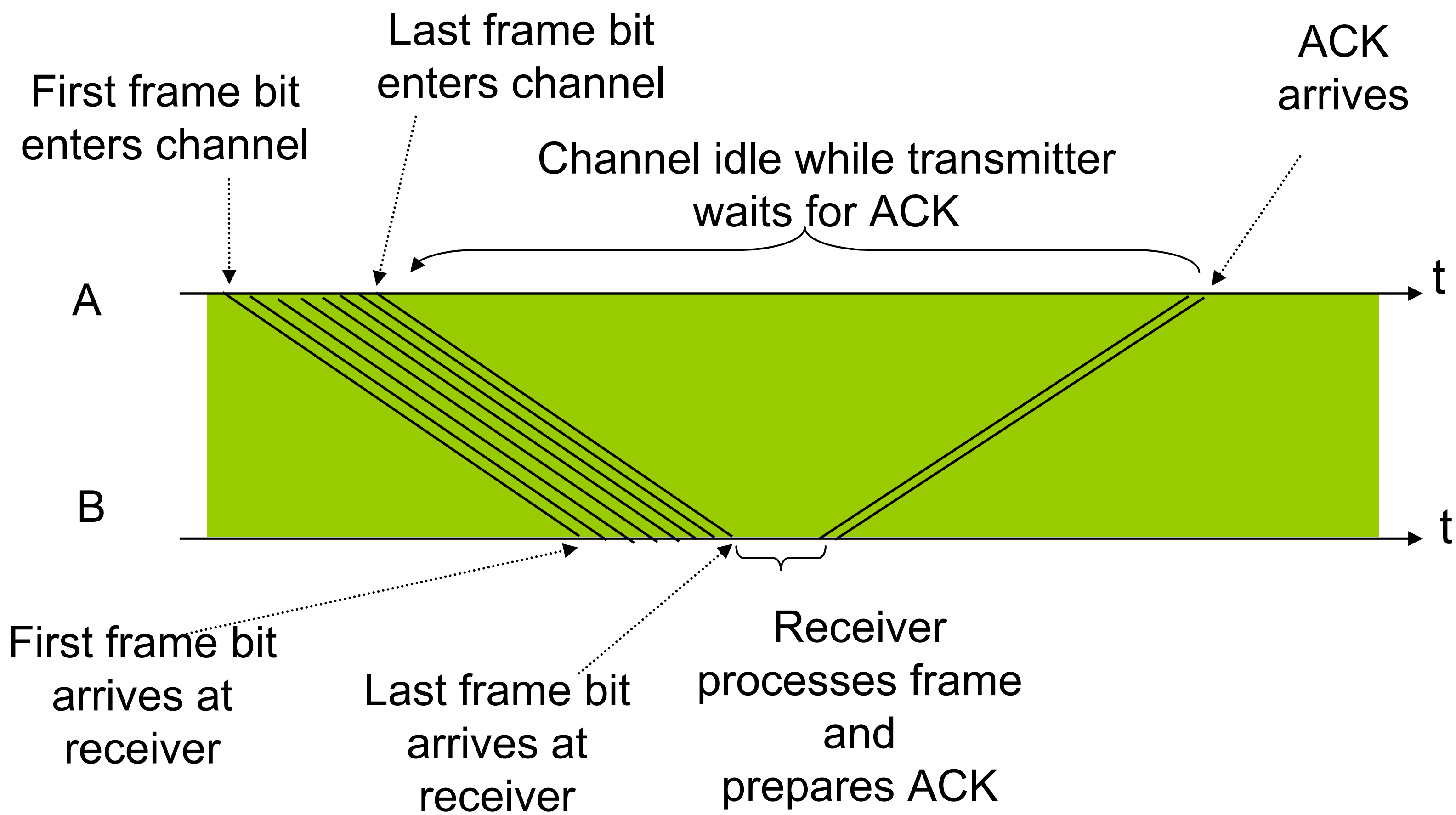
- Wait for ACK or timer to expire; block requests from higher layer
- If timeout expires
  - retransmit frame and reset timer
- If ACK received:
  - If sequence number is incorrect or if errors detected: ignore ACK
  - If sequence number is correct ($R_{next} = S_{last} +1$): accept frame, go to Ready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

Receiver

Always in Ready State

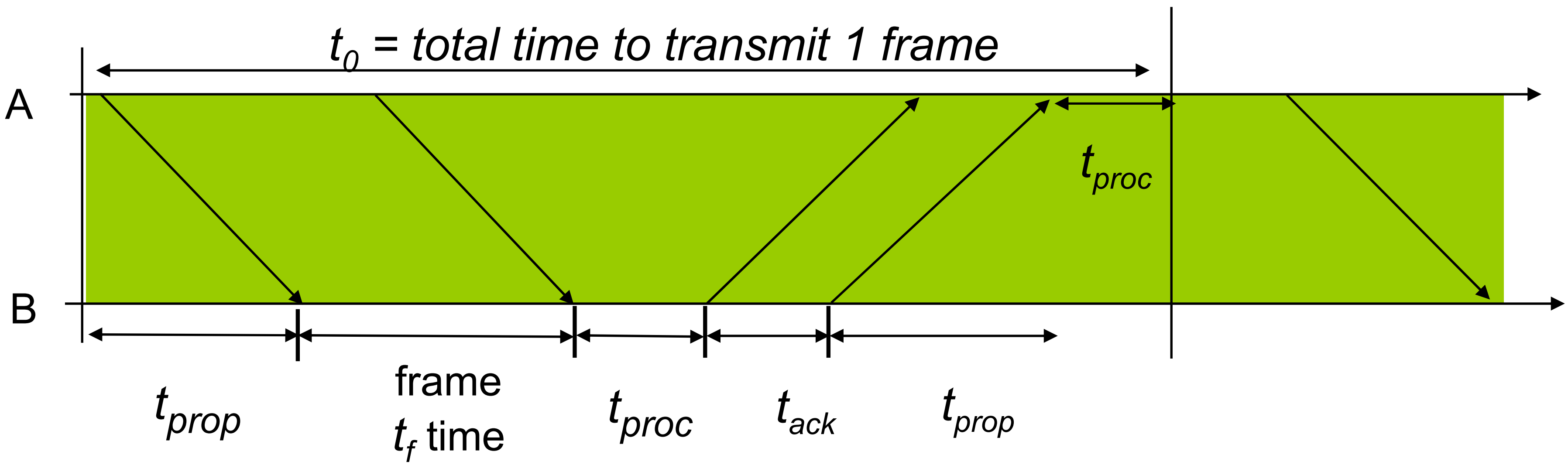
- Wait for arrival of new frame
- When frame arrives, check for errors
- If no errors detected and sequence number is correct (S_{last}=R_{next}), then
  - accept frame,
  - update R_{next},
  - send ACK frame with R_{next},
  - deliver packet to higher layer
- If no errors detected and wrong sequence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

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:

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

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

2xDelayxBW Efficiency	1 ms 200 km	10 ms 2000 km	100 ms 20000 km	1 sec 200000 km
1 Mbps	10^3 88%	10^4 49%	10^5 9%	10^6 1%
1 Gbps	10^6 1%	10^7 0.1%	10^8 0.01%	10^9 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 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}

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

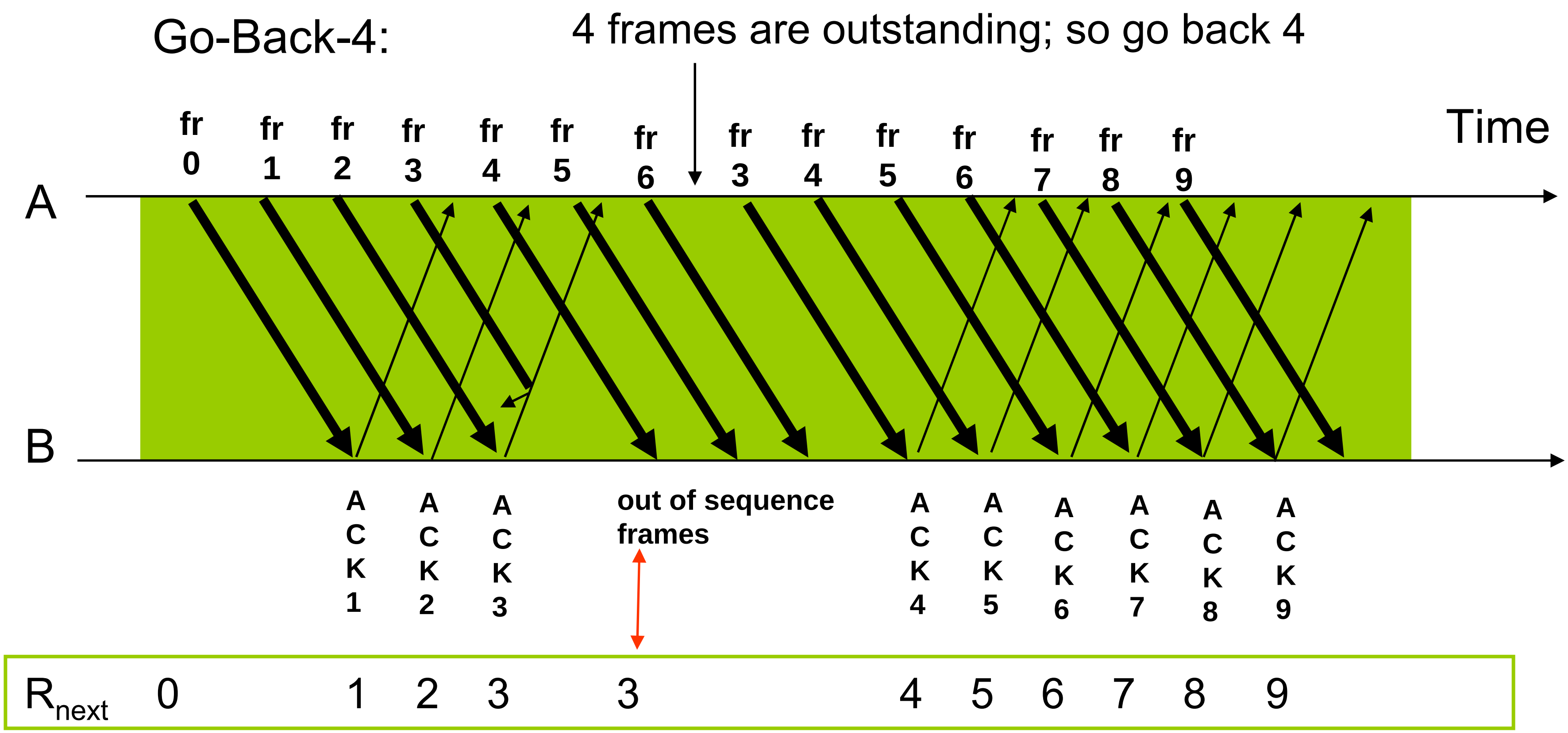
$1 - P_f$ Efficiency	0	10^{-6}	10^{-5}	10^{-4}
1 Mbps & 1 ms	1 88%	0.99 86.6%	0.905 79.2%	0.368 32.2%

Bit errors impact performance as $n_f p$ *approach 1*

Go-Back-N

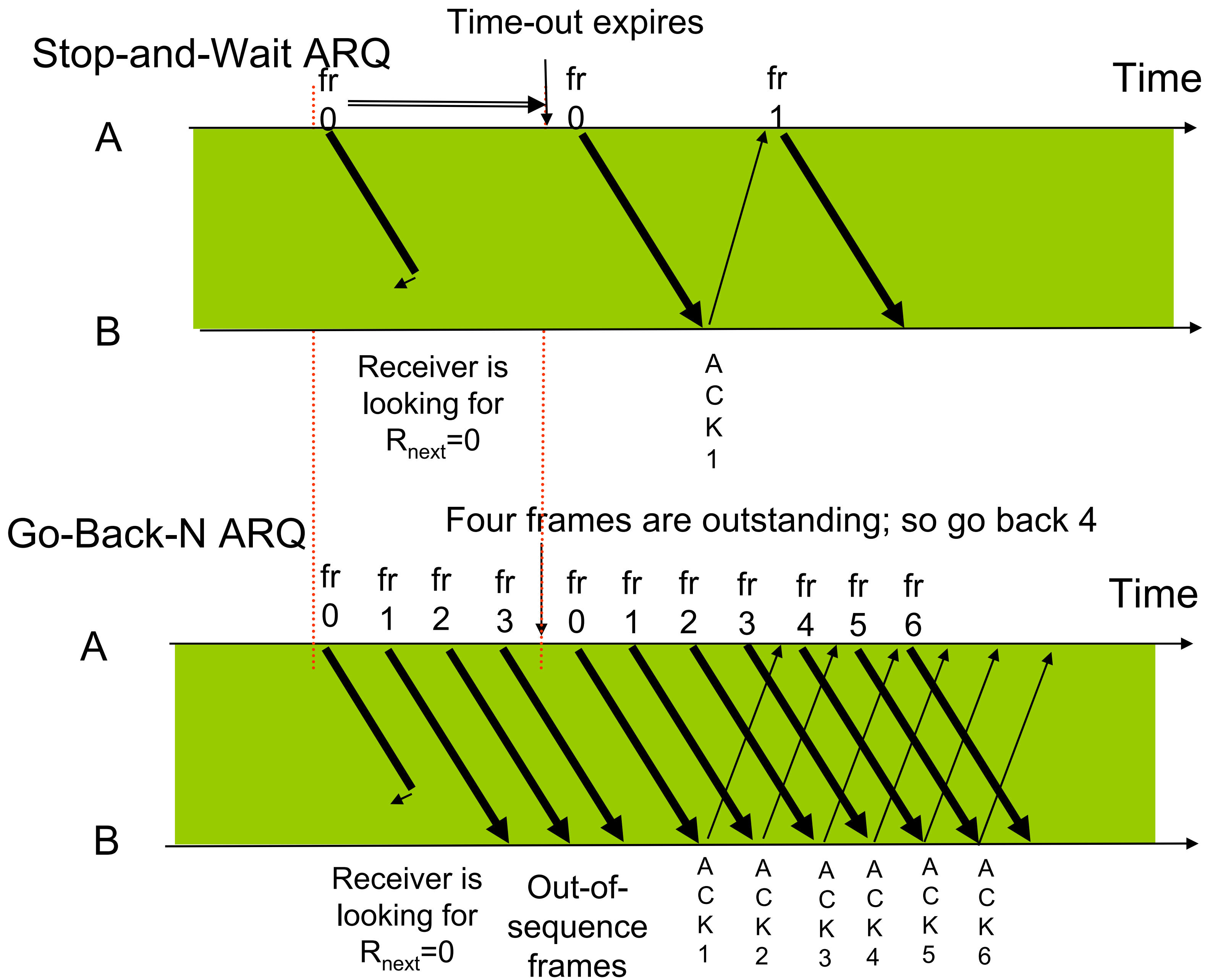
- Improve Stop-and-Wait by not waiting!
- Keep channel busy by continuing to send frames
- Allow a window of up to W_s outstanding frames
- Use *m*-bit sequence numbering
- If ACK for oldest frame arrives before window is exhausted, we can continue transmitting
- If window is exhausted, pull back and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

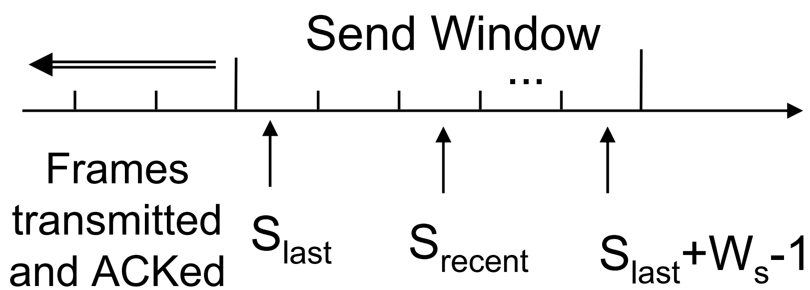


Go-Back-N with Timeout

- 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

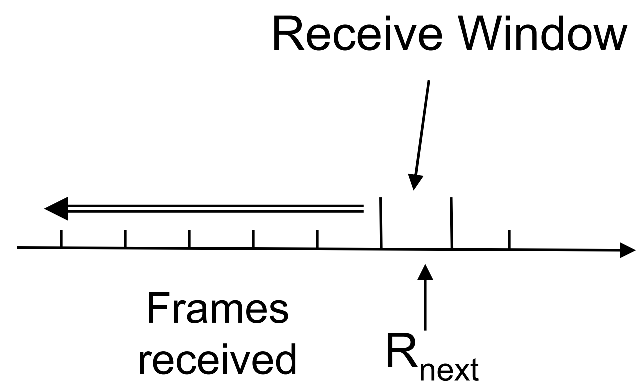
Go-Back-N Transmitter & Receiver

Transmitter



Buffers

Receiver

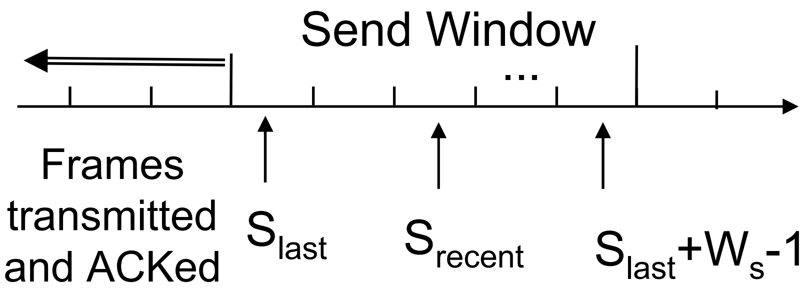


Receiver will only accept a frame that is error-free and that has sequence number R_{next}

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

Sliding Window Operation

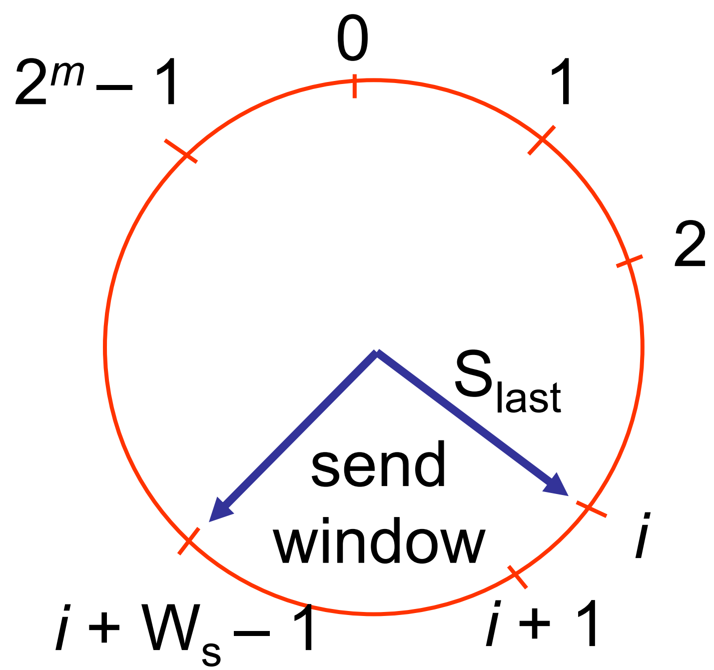
Transmitter



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

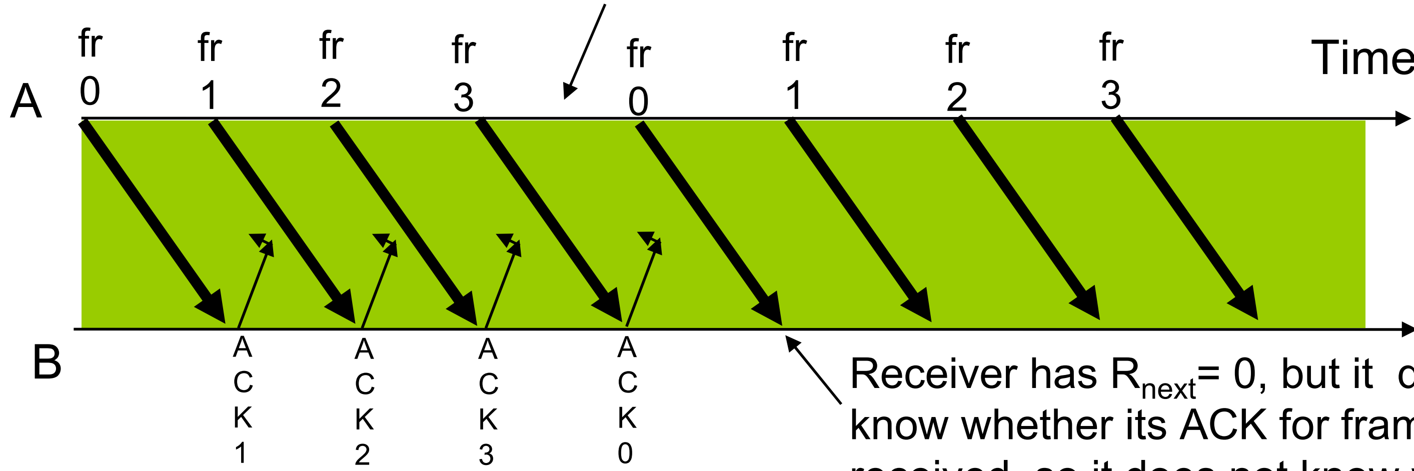
When such ACK frame arrives, S_{last} is incremented by one, and the *send window slides forward* by one

m-bit Sequence Numbering



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

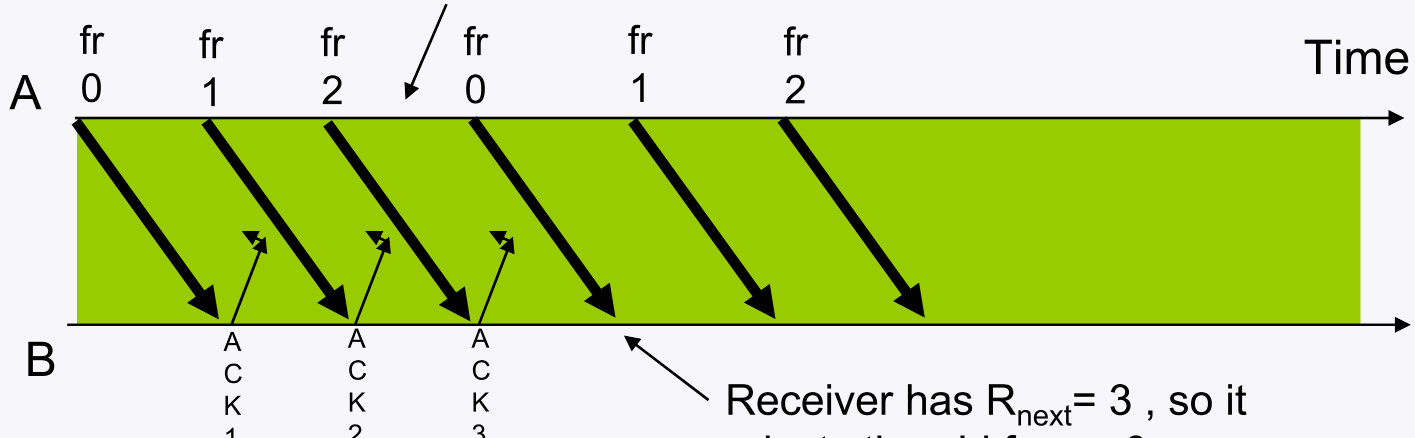
$M = 2^2 = 4$, Go-Back - 4: Transmitter goes back 4



R_{next}	0	1	2	3	0

Receiver has R_{next}= 0, but it does not know whether its ACK for frame 0 was received, so it does not know whether this is the old frame 0 or a new frame 0

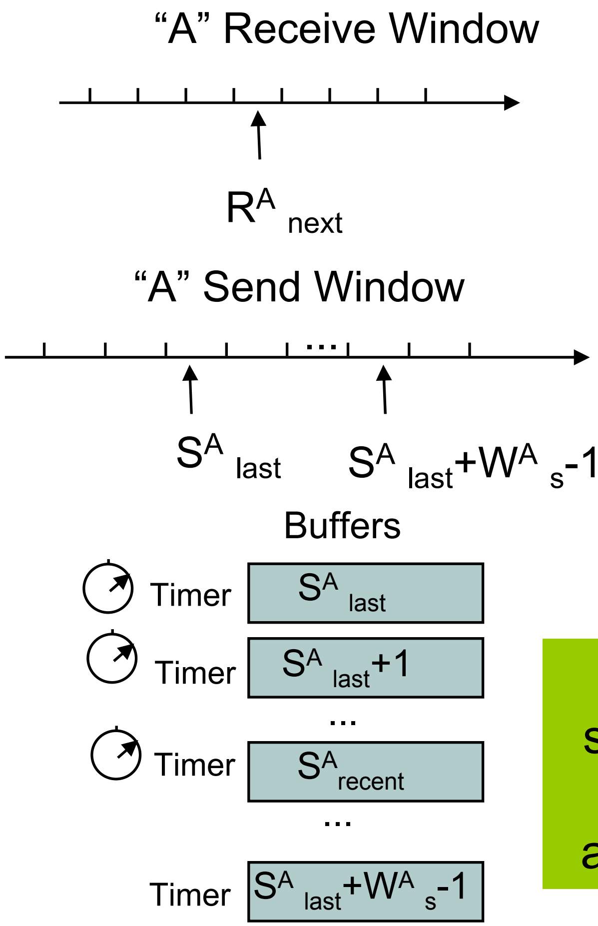
$M = 2^2 = 4$, Go-Back-3: Transmitter goes back 3



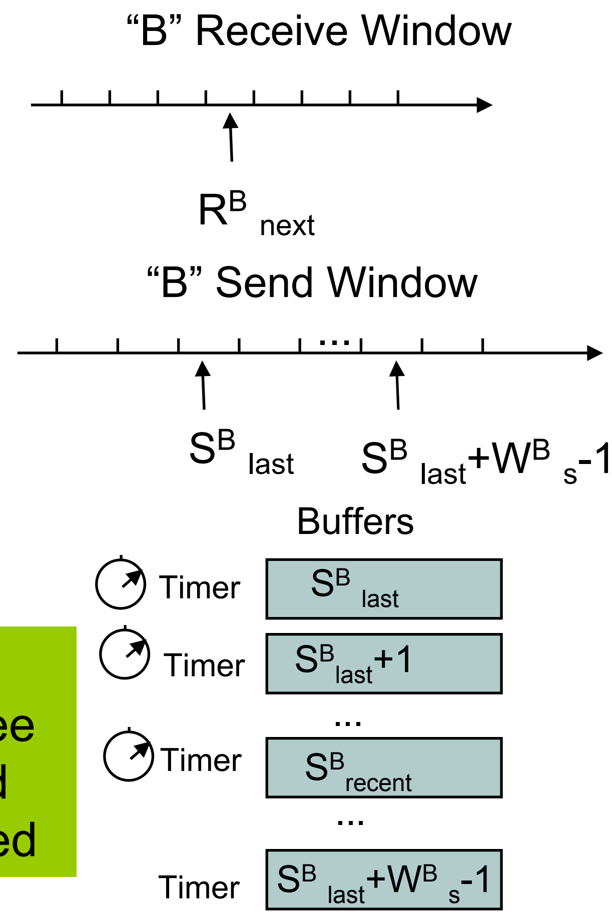
R_{next}	0	1	2	3

Receiver has R_{next}= 3 , so it rejects the old frame 0

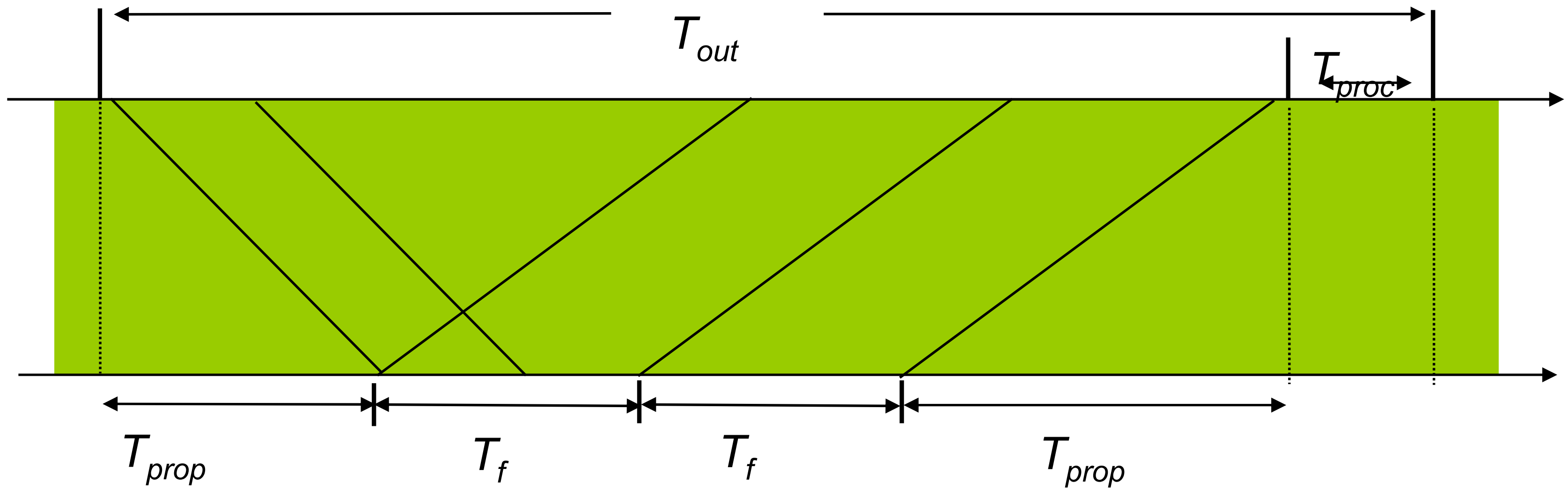
ACK Piggybacking in Bidirectional GBN



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



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}

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

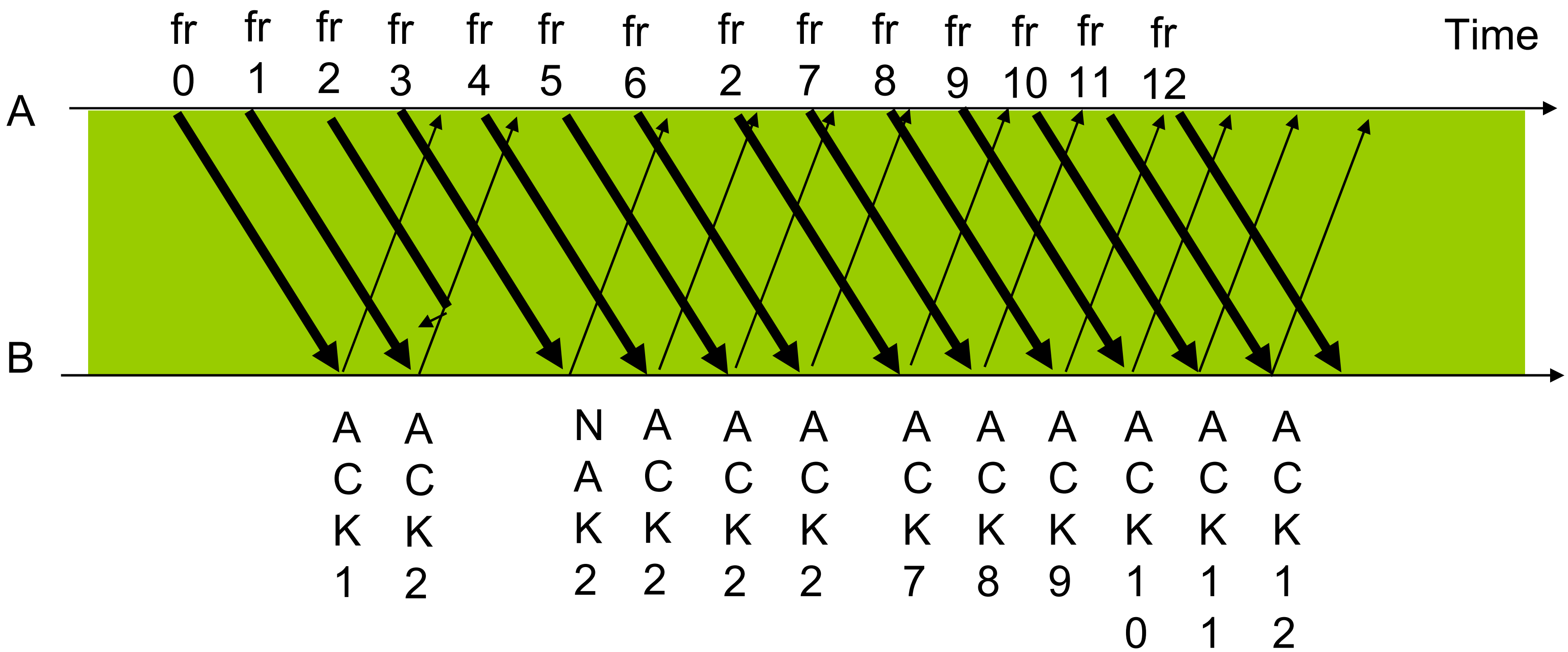
Efficiency	0	10^{-6}	10^{-5}	10^{-4}
S&W	8.9%	8.8%	8.0%	3.3%
GBN	98%	88.2%	45.4%	4.9%

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

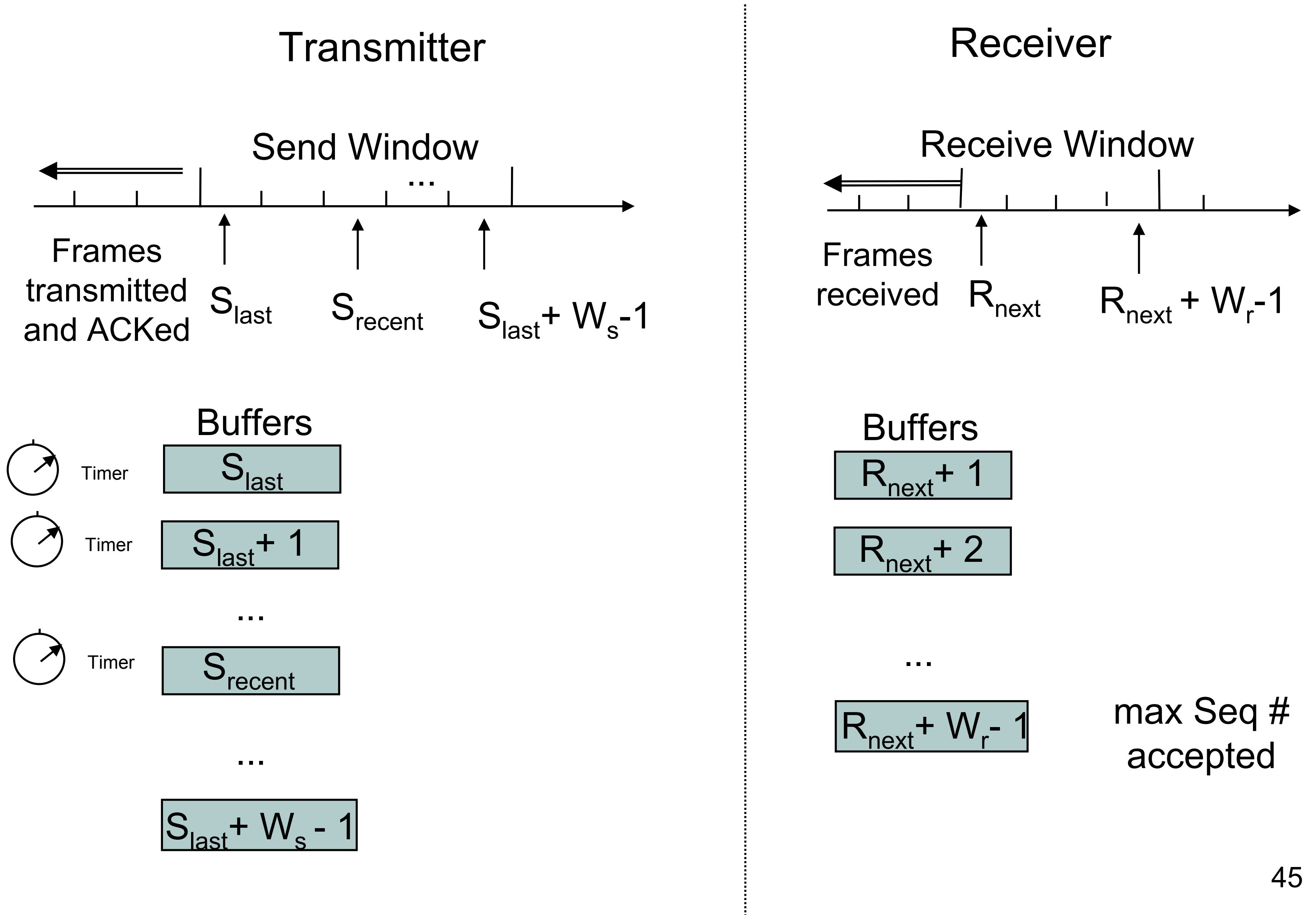
Selective Repeat ARQ

- 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

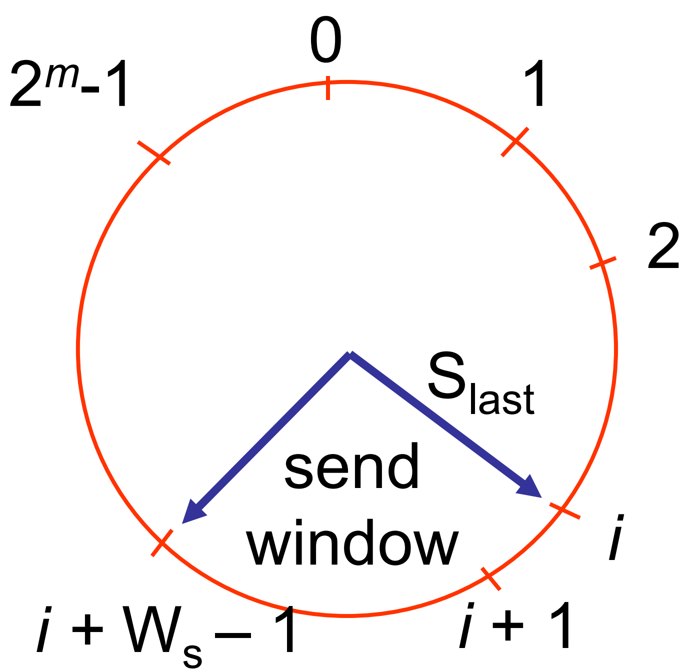


Selective Repeat ARQ



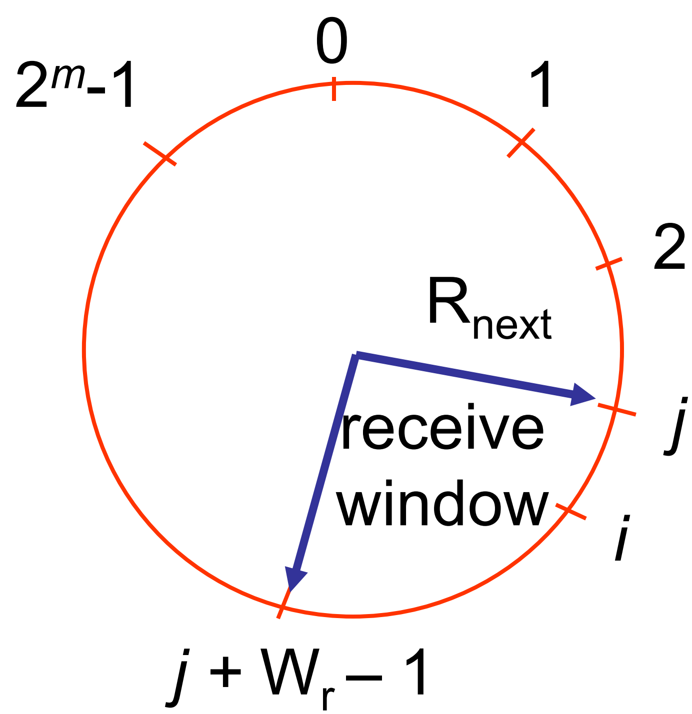
Send & Receive Windows

Transmitter



Moves k forward when ACK arrives with $R_{next} = S_{last} + k$
$k = 1, \ldots, W_s$-1

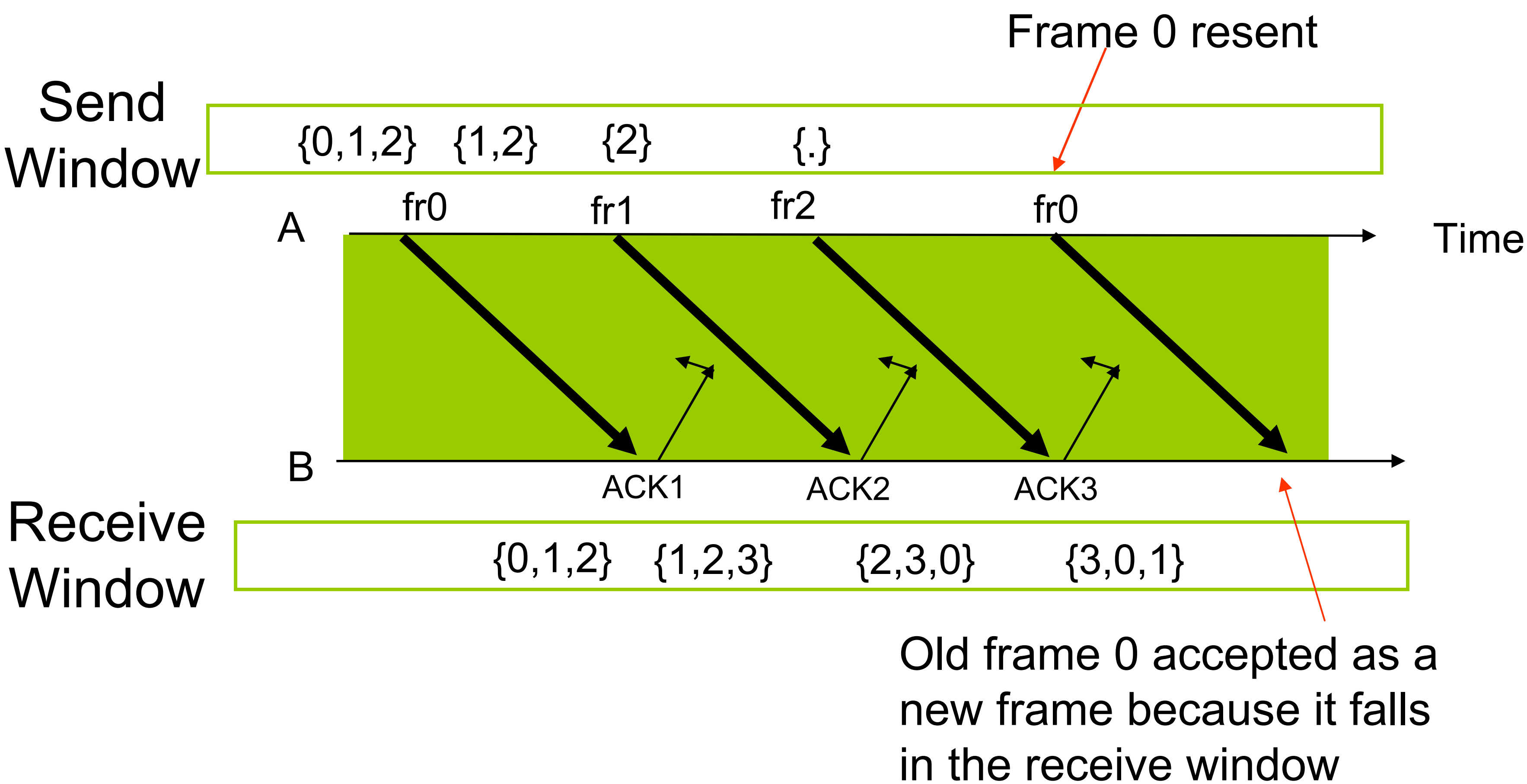
Receiver



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

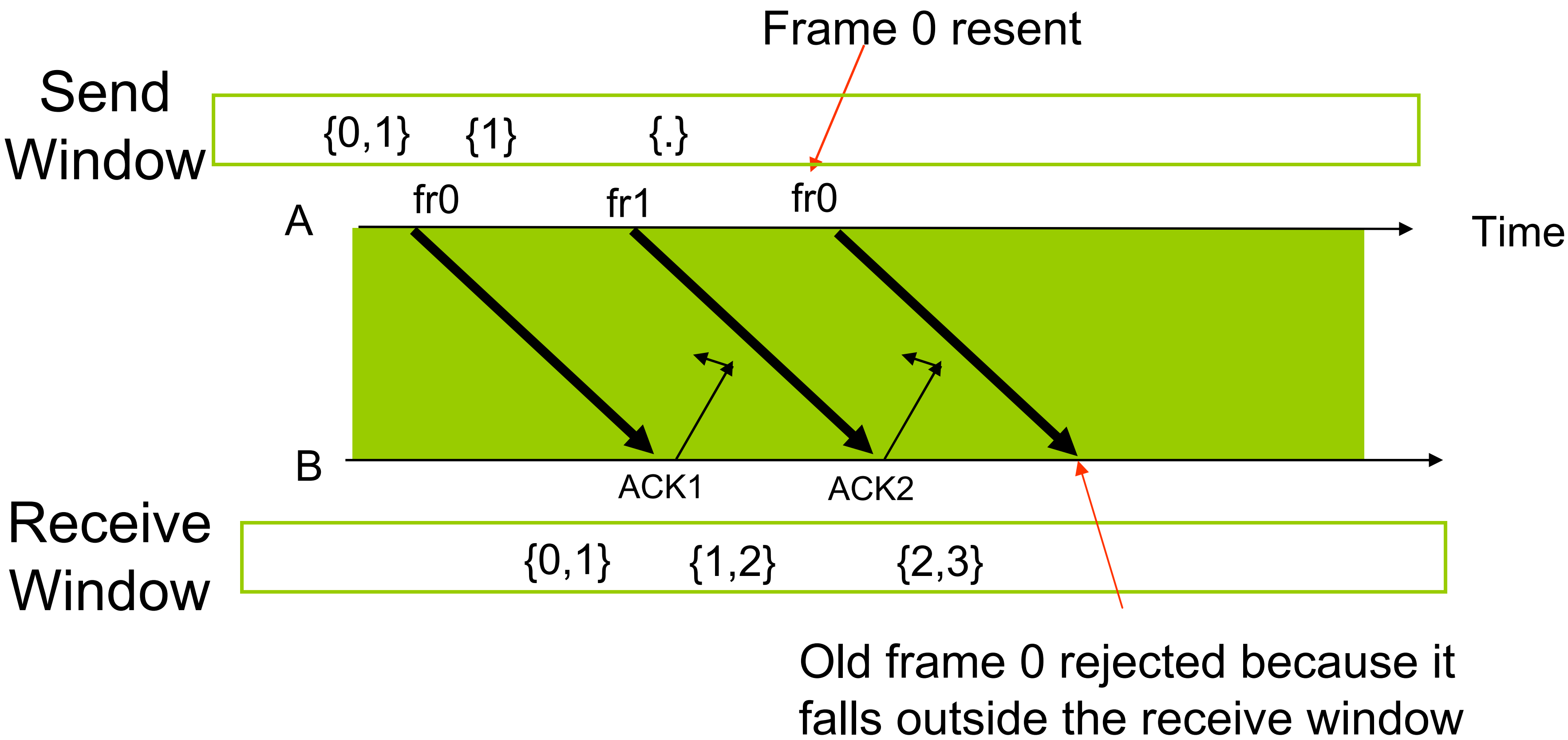
What size W_s and W_r allowed?

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



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

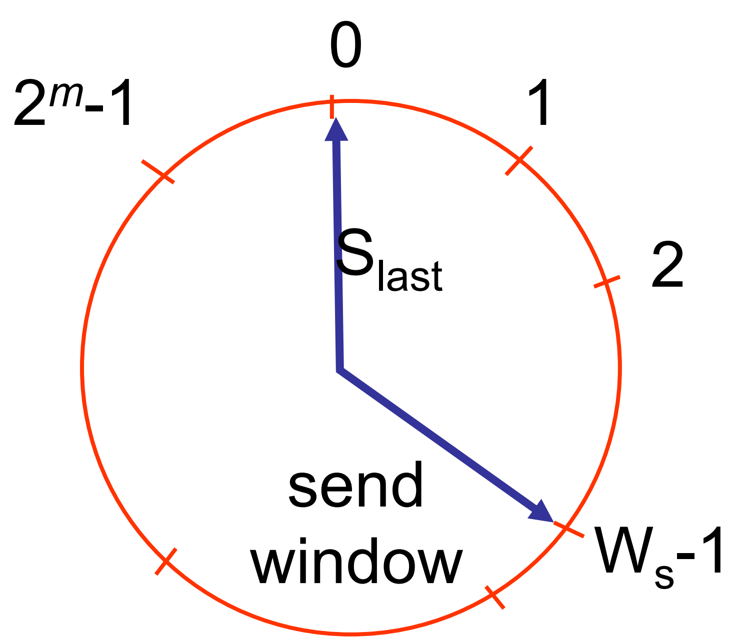
- Example: $M=2^2=4$, $W_s=2$, $W_r=2$

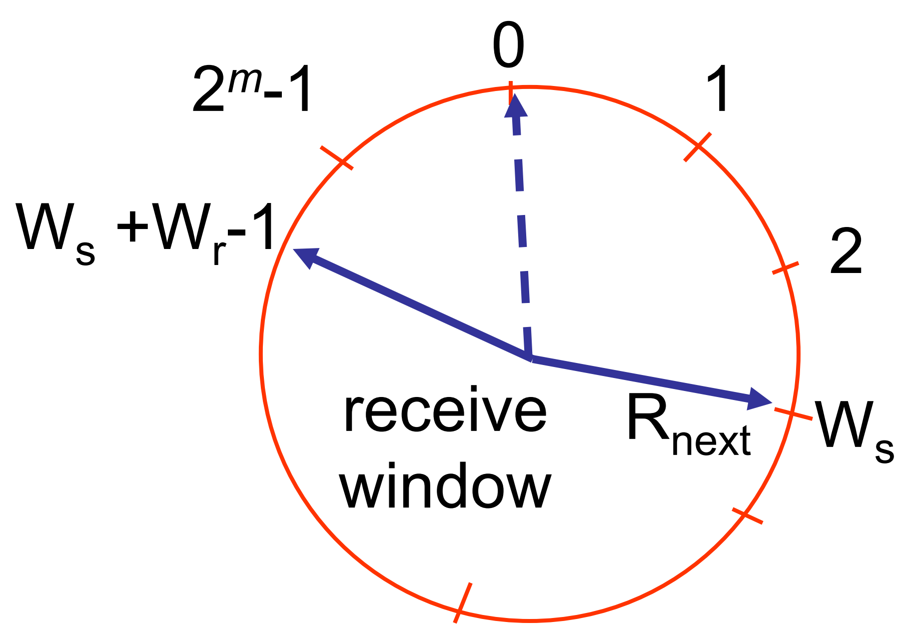


Why $W_s + W_r = 2^m$ works

- 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, …, W_r}
- Window slides forward to {W_s,…,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

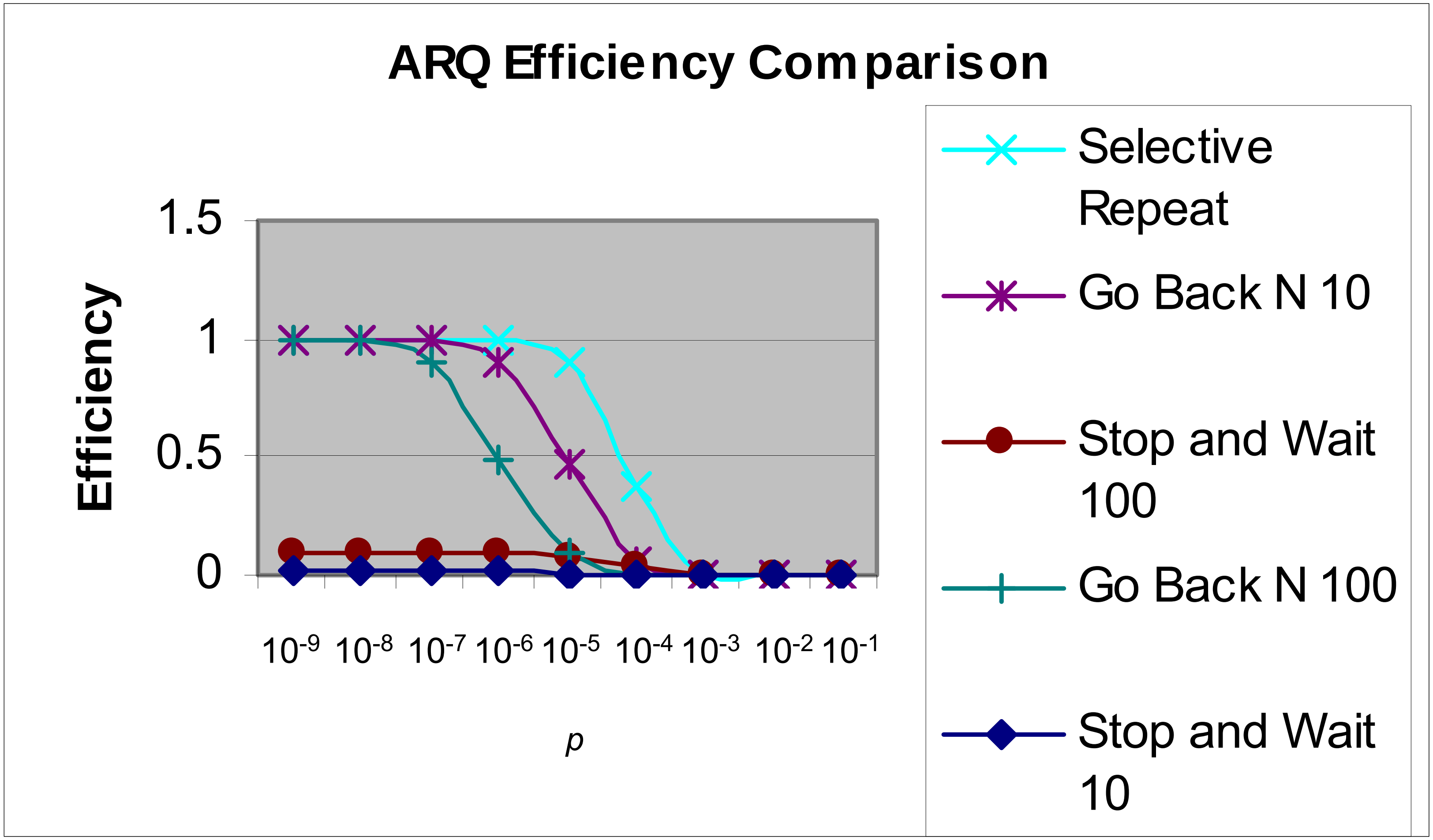
$$\eta_{GBN} = \frac{1-P_f}{1+(W_S-1)P_f} = \frac{1-P_f}{1+LP_f}$$

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

Stop-and-Wait:

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

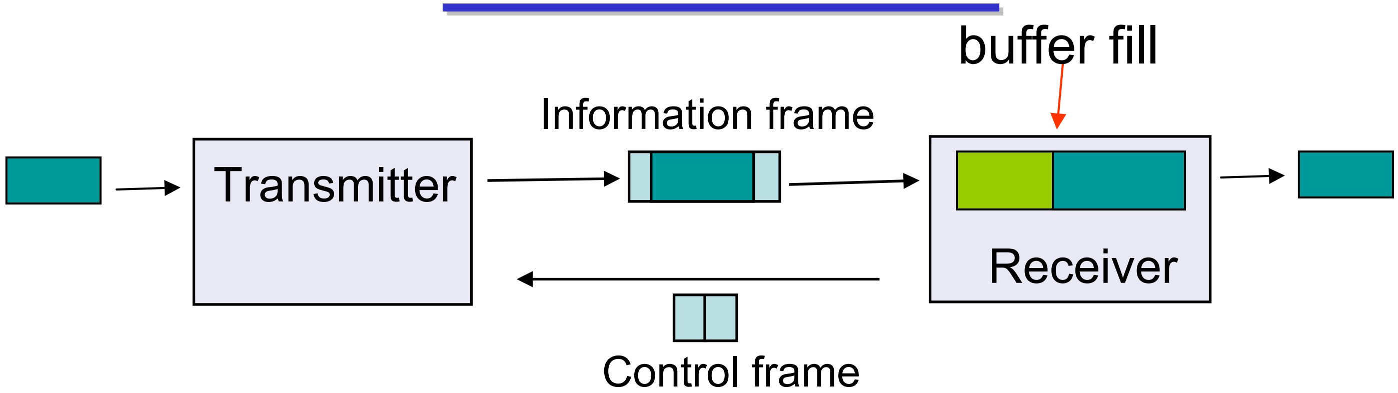
ARQ Efficiencies



Delay-Bandwidth product = 10, 100

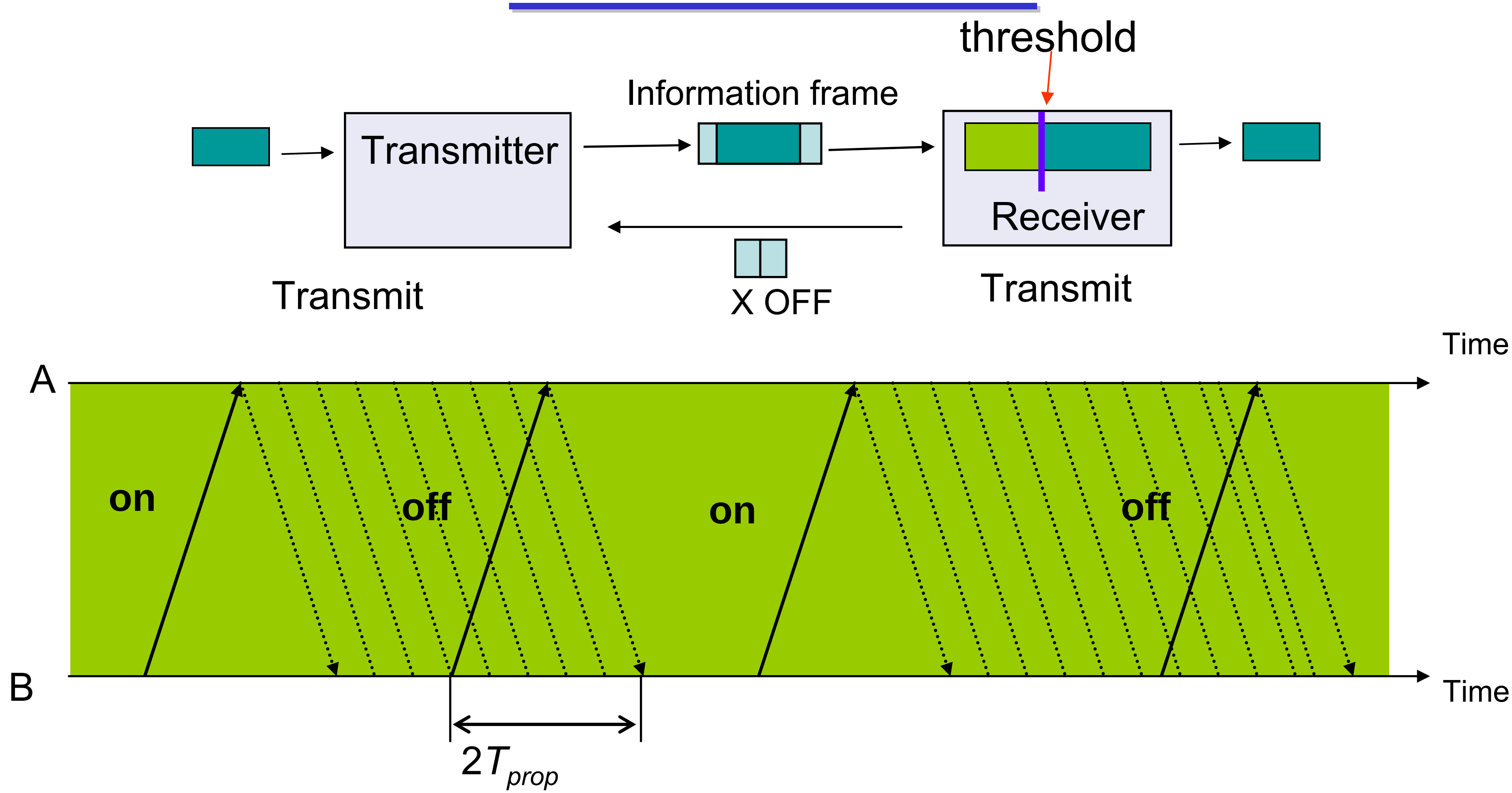
Flow Control

Flow Control



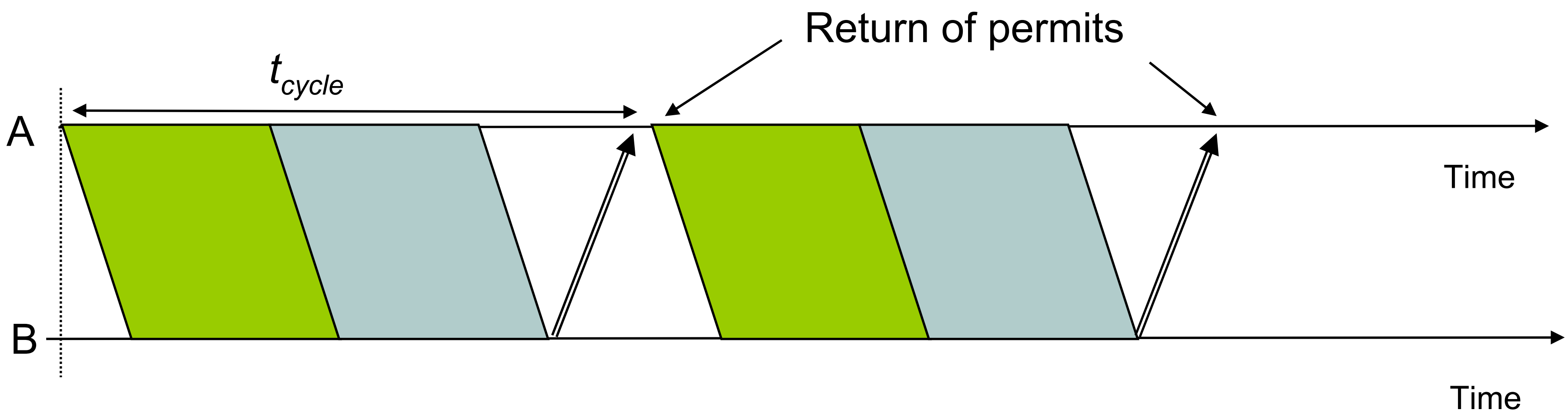
- Receiver has limited buffering to store arriving frames
- Several situations cause buffer overflow
  - Mismatch between sending rate & rate at which user can retrieve data
  - Surges in frame arrivals
- *Flow control* prevents buffer overflow by regulating rate at which source is allowed to send information

X ON / X OFF



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

Window Flow Control

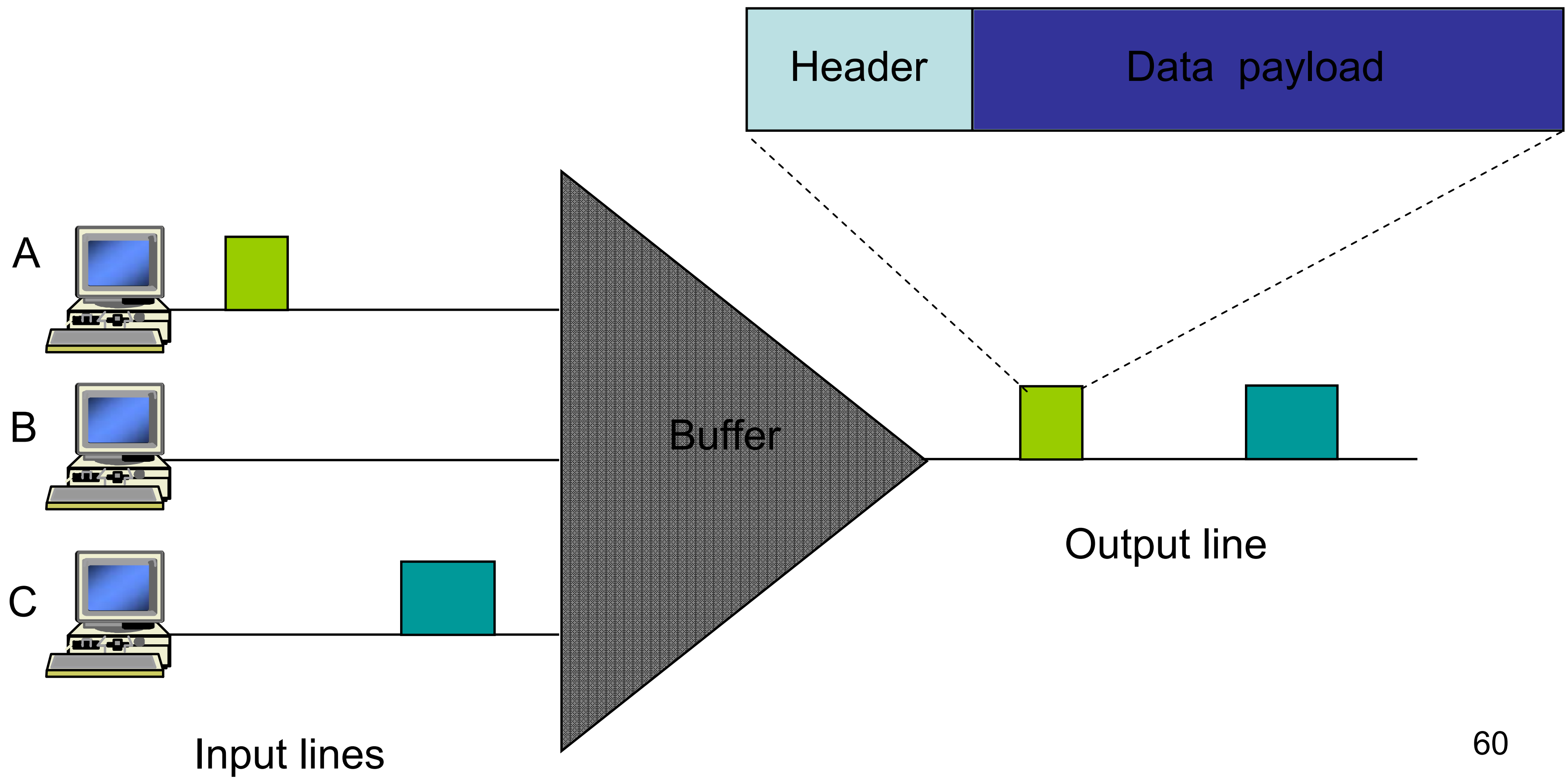


- Sliding Window ARQ method with W_s equal to buffer available
  - Transmitter can never send more than W_s frames
- ACKs that slide window forward can be viewed as permits to transmit more
- Can also pace ACKs as shown above
  - Return permits (ACKs) at end of cycle regulates transmission rate
- Problems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

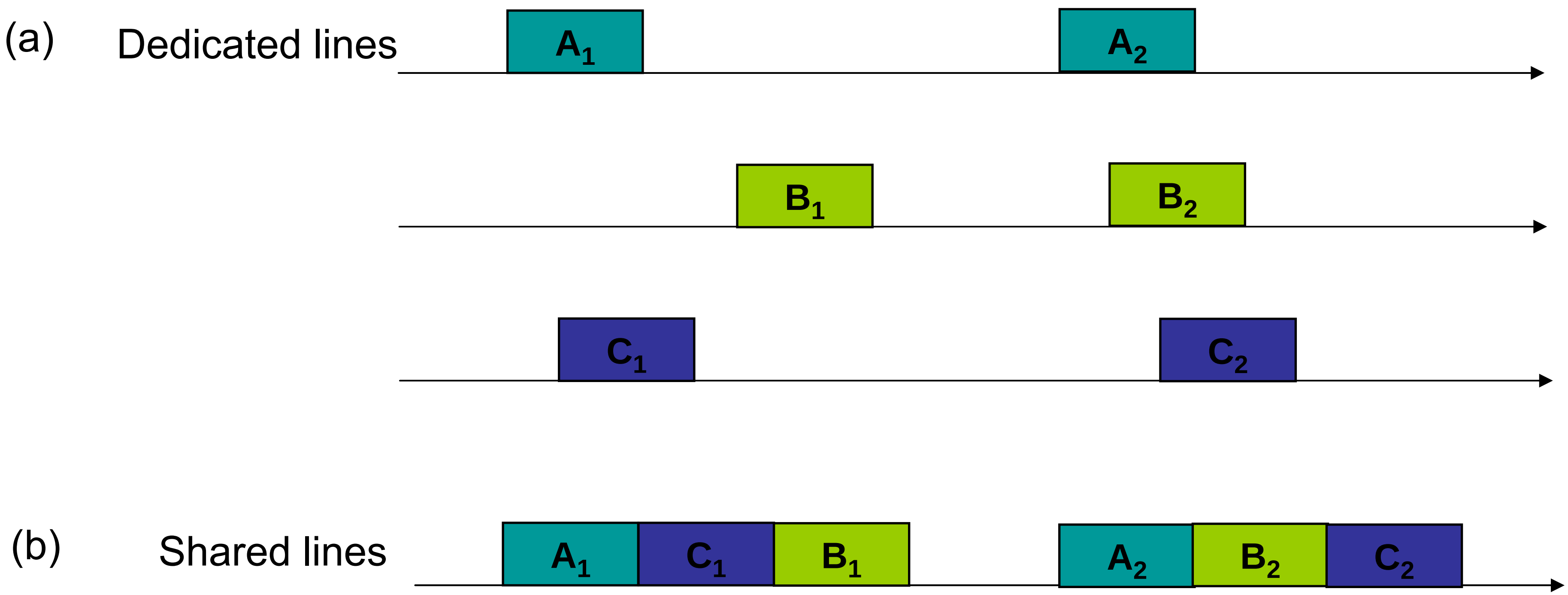
Link Sharing Using Statistical Multiplexing

Statistical Multiplexing

- Multiplexing concentrates bursty traffic onto a shared line
- Greater efficiency and lower cost

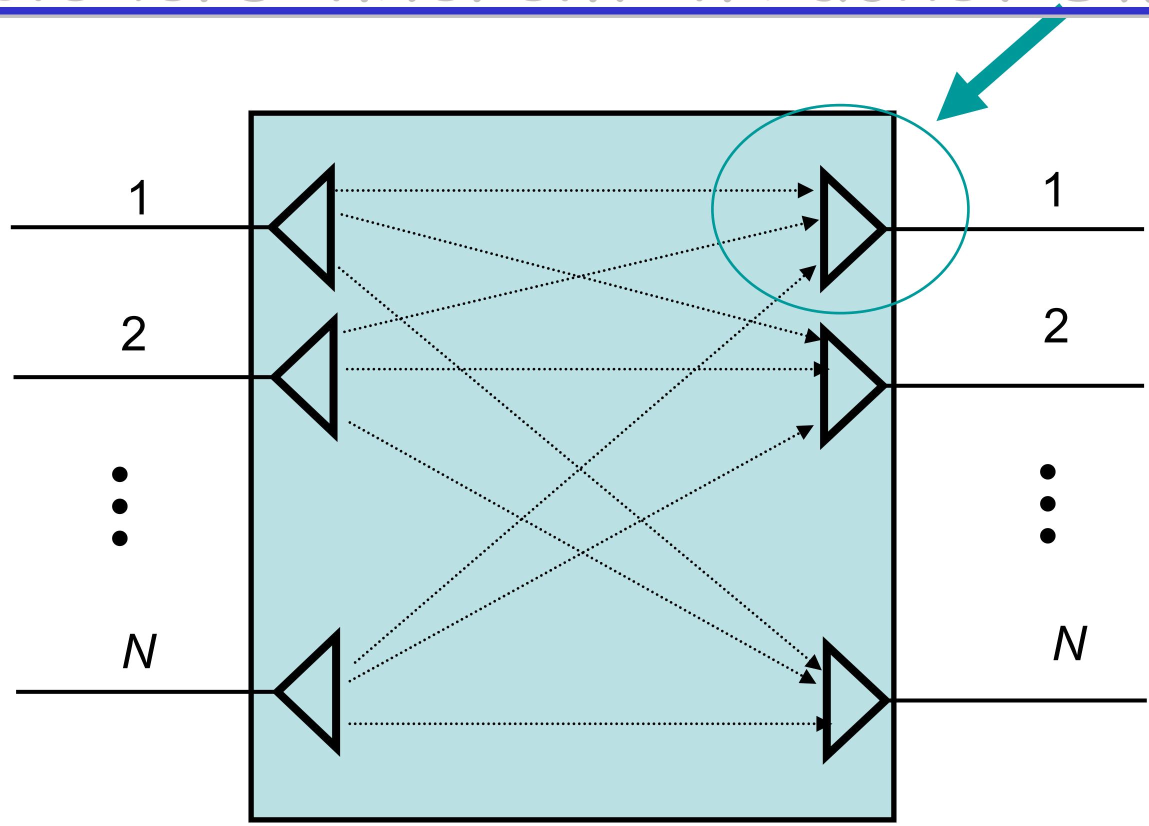


Tradeoff Delay for Efficiency



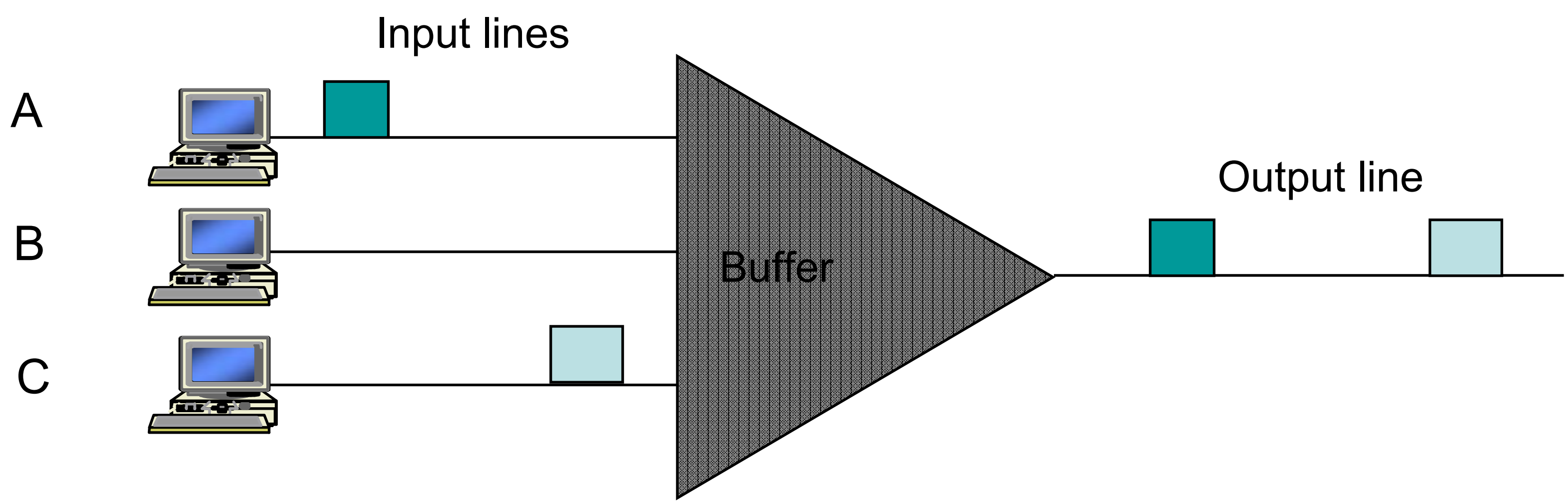
- Dedicated lines involve not waiting for other users, but lines are used inefficiently when user traffic is bursty
- Shared lines concentrate packets into shared line; packets buffered (delayed) when line is not immediately available

Multiplexers inherent in Packet Switches



- Packets/frames forwarded to buffer prior to transmission from switch
- Multiplexing occurs in these buffers

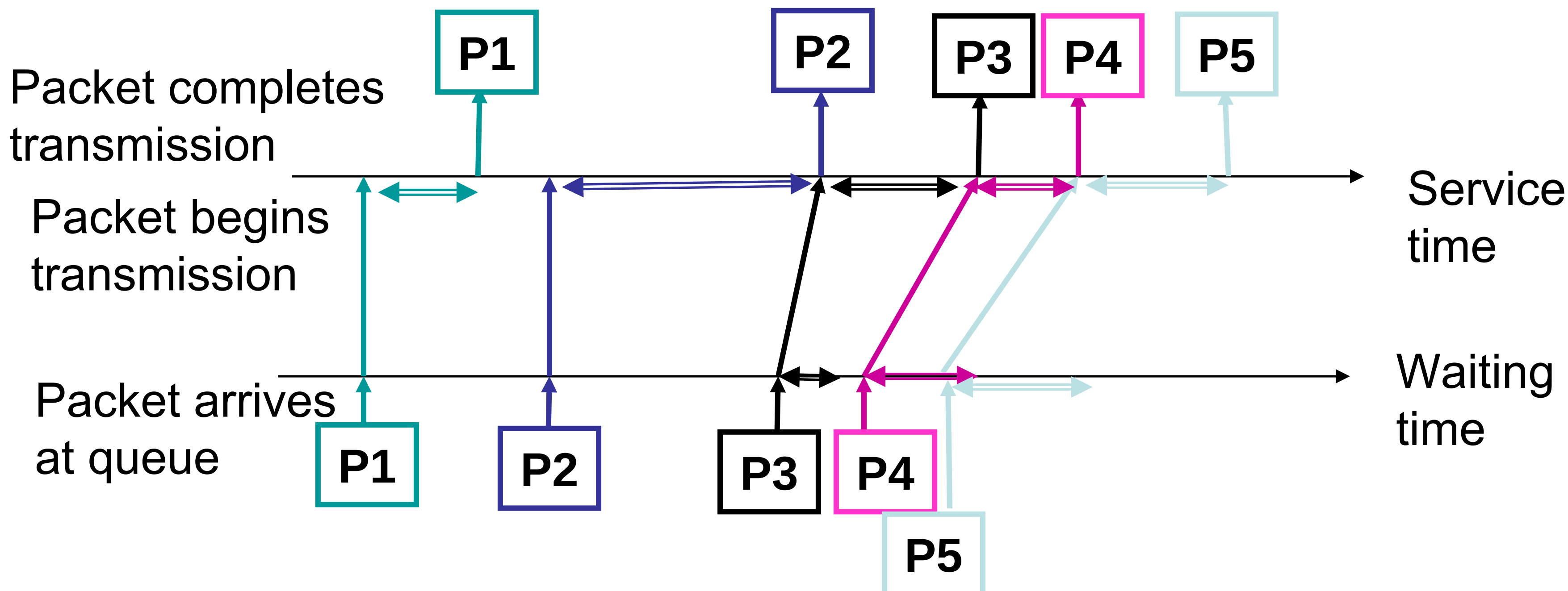
Multiplexer Modeling



- Arrivals: What is the packet interarrival pattern?
- Service Time: How long are the packets?
- Service Discipline: What is order of transmission?
- Buffer Discipline: If buffer is full, which packet is dropped?

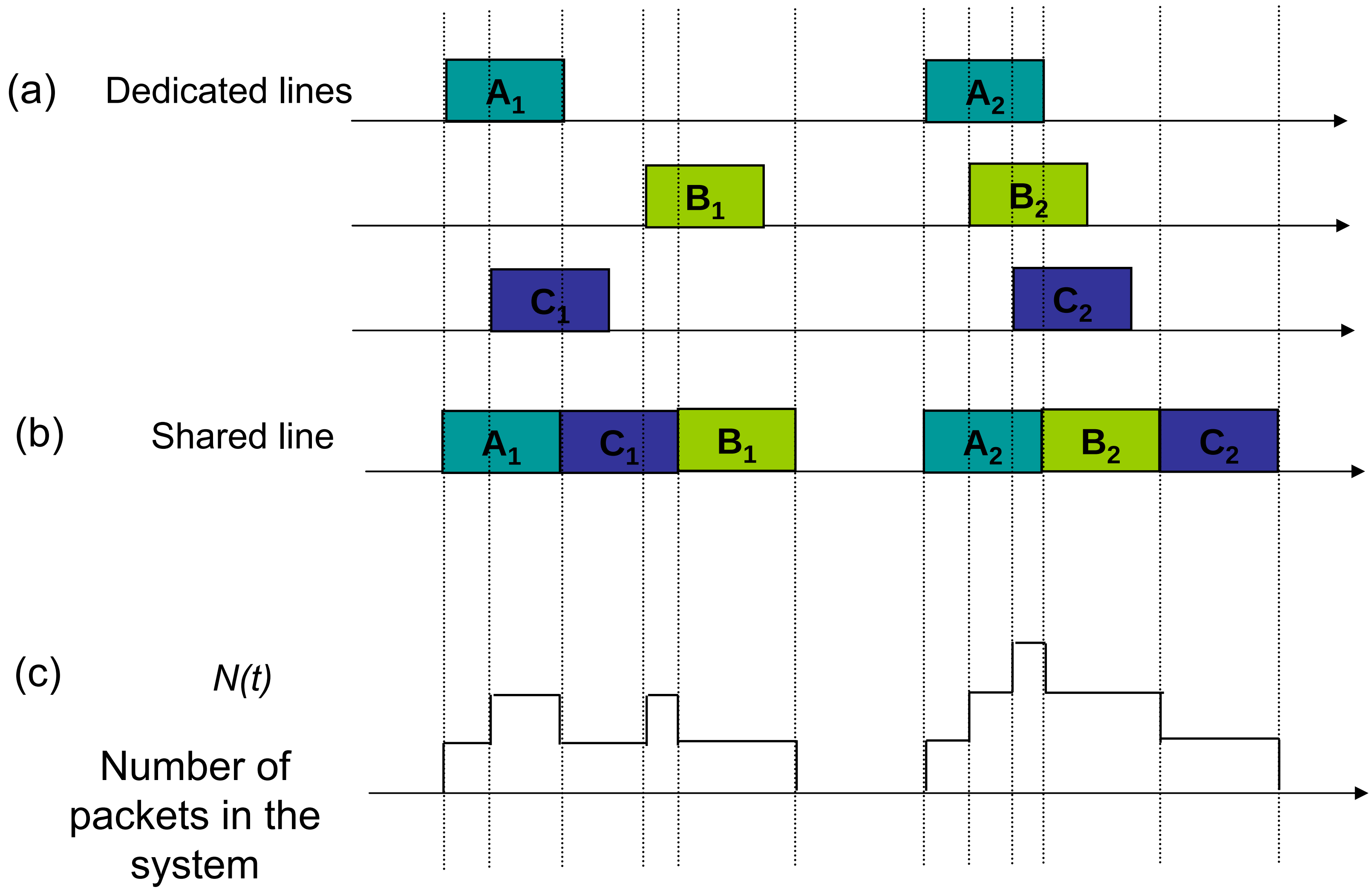
- Performance Measures:
- *Delay Distribution; Packet Loss Probability; Line Utilization*

Delay = Waiting + Service Times



- Packets arrive and wait for service
- Waiting Time: from arrival instant to beginning of service
- Service Time: time to transmit packet
- Delay: total time in system = waiting time + service time

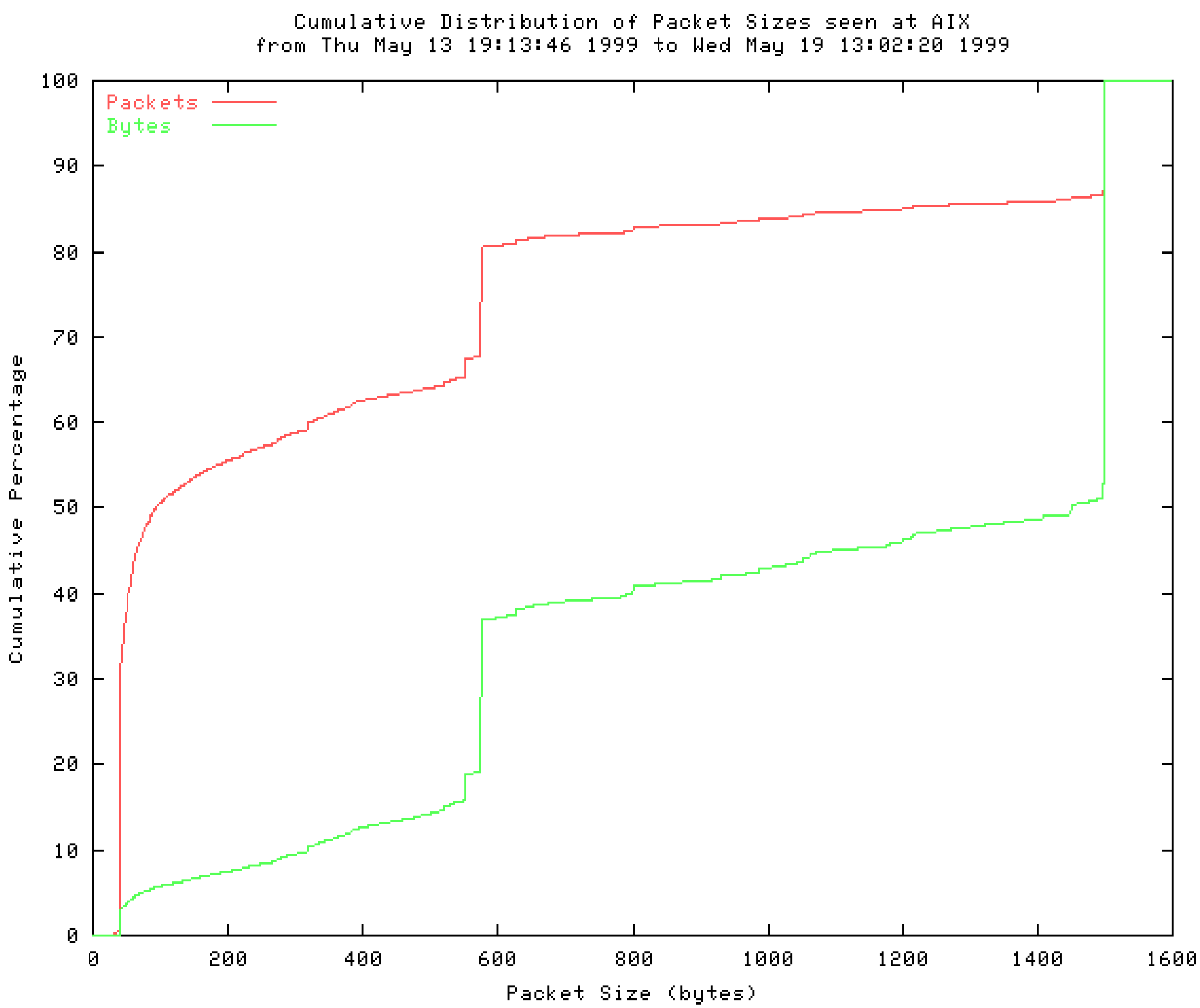
Fluctuations in Packets in the System



Packet Lengths & Service Times

- R bits per second transmission rate
- L = # bits in a packet
- $X = L/R$ = time to transmit ("service") a packet
- Packet lengths are usually variable
  - Distribution of lengths → Dist. of service times
  - Common models:
    - Constant packet length (all the same)
    - Exponential distribution
    - Internet Measured Distributions fairly constant
      - See next chart

Measure Internet Packet Distribution



- Dominated by TCP traffic (85%)
- ~40% packets are minimum-sized 40 byte packets for TCP ACKs
- ~15% packets are maximum-sized Ethernet 1500 frames
- ~15% packets are 552 & 576 byte packets for TCP implementations that do not use path MTU discovery
- Mean=413 bytes
- Stand Dev=509 bytes
- Source: caida.org