Transport Layer: TCP and UDP

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Overview

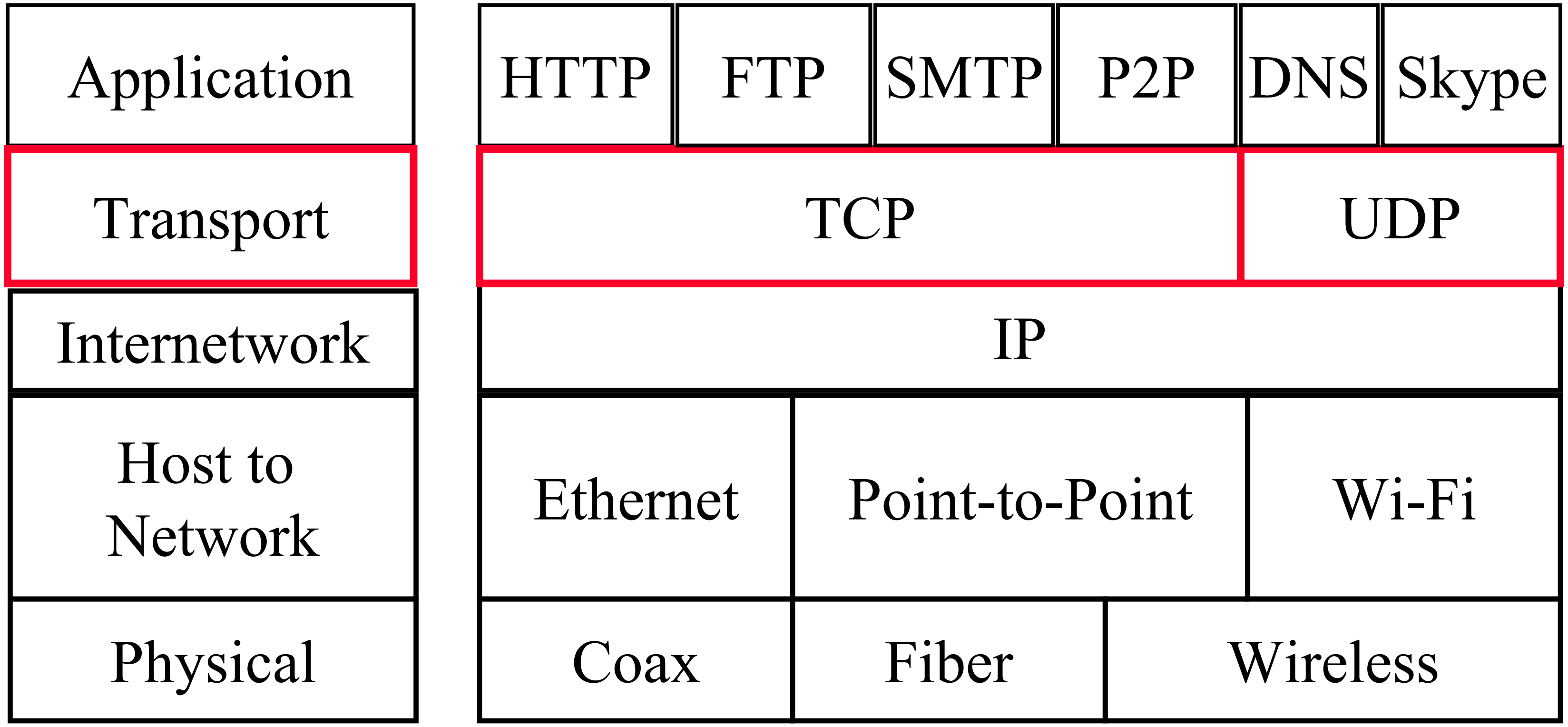
- Transport Layer Design Issues:
  - Multiplexing/Demultiplexing
  - Flow control
  - Error control
- UDP
- TCP
  - Header format, connection management, checksum
  - Slow Start Congestion Control
- **Note**: This class lecture is based on Chapter 3 of the textbook (Kurose and Ross) and the figures provided by the authors.

Transport Layer Design Issues

1. Transport Layer Functions
2. Multiplexing and Demultiplexing
3. Error Detection: Checksum
4. Flow Control
5. Efficiency Principle
6. Error Control: Retransmissions

Protocol Layers

- Top-Down approach



Application — HTTP, FTP, SMTP, P2P, DNS, Skype

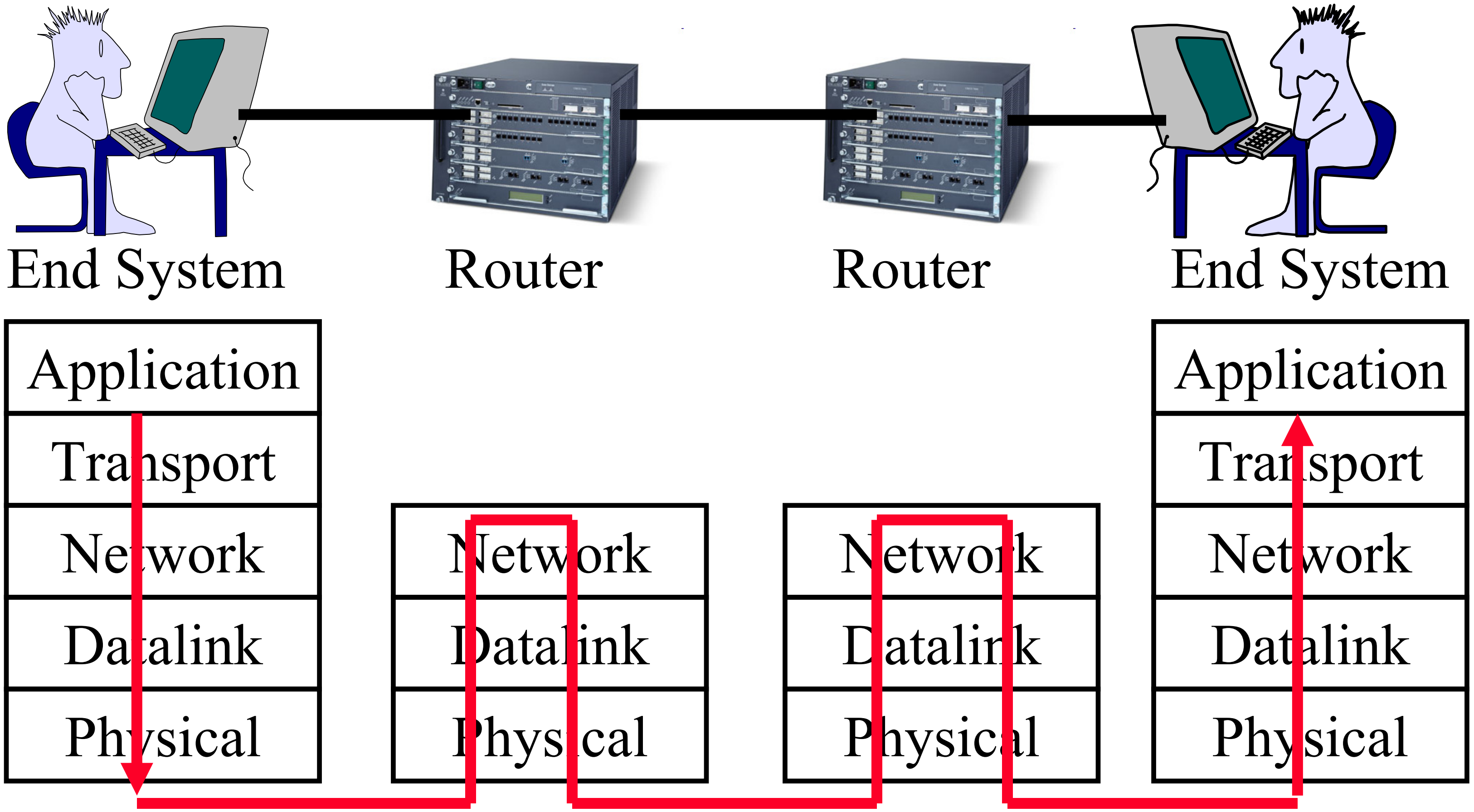
Transport — TCP, UDP

Internetwork — IP

Host to Network — Ethernet, Point-to-Point, Wi-Fi

Physical — Coax, Fiber, Wireless

Transport Layer



End System	Router	Router	End System
Application			Application
Transport			Transport
Network	Network	Network	Network
Datalink	Datalink	Datalink	Datalink
Physical	Physical	Physical	Physical

- Transport = End-to-End Services
  Services required at source and destination systems
  Not required on intermediate hops

Transport Layer Functions

1. **Multiplexing and demultiplexing**: among applications and processes at end systems
2. **Error detection**: Bit errors
3. **Loss detection**: Lost packets due to buffer overflow at intermediate systems (Sequence numbers and acks)
4. **Error/loss recovery**: Retransmissions
5. **Flow control**: Ensuring receiver has buffers
6. **Congestion Control**: Ensuring network has capacity

Not all transports provide all functions

Flow control makes sure that the receiver has the capacity and congestion control makes sure that the network has the capacity

Multiplexing and Demultiplexing

❑ Transport layer at the sender side receives data from different Applications, encapsulates every packet with a Transport Layer header and pass it on to the underlying Network Layer. This job of transport layer is known as **Multiplexing**.

❑ At the receiver's side, the transport gathers the data, examines it socket and passes the data to the correct Application. This is known as **De-Multiplexing**.

Ex: Multiplexing and Demultiplexing

- Suppose that there are two houses. One is in India and other is in America. In the house in India, lives a person James along with his 5 children. In the house in America, lives a person Steve along with his 4 children.

- Now all 5 children of James write a letter to every children of Steve on every Sun-day. Therefore total number of letters will be 20. Thus, all the children writes the letter, put them in envelopes and hand over it to James. Then James write source house address and the destination house address on the envelope and give it to the postal service of India.

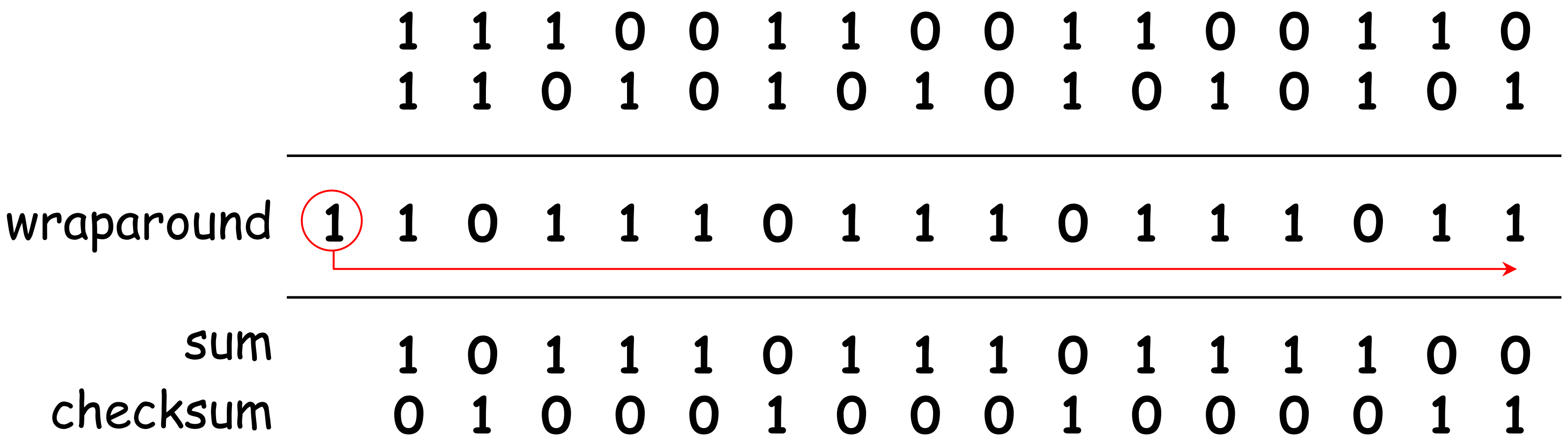
- The postal service of India puts some other addresses corresponding to the country and delivers it to the America postal Service. The American Postal sees the destination address on the envelopes and will deliver those 20 letters to the Steve House. Steve collects the letter from the postman and after considering the name of his respective children on the envelopes, he gives the letter to each of them.

In this example we have processes and the layers.
Processes = children
Application Layer messages = envelopes
Hosts = The two Houses
Transport Layer Protocol = James and Steve
Network Layer protocol = Postal Service

- When James collects all the letters from his children, he multiplexes all and encapsulates them with the respective children name on the letter and house address and give it to the Indian postal service. On the receiving side, Steve collects all the letters from postal service of America and de-multiplexes them to see, which letter is for which child and delivers it respectively.

Error Detection: Checksum

- ❑ **Cyclic Redundancy Check (CRC)**: Powerful but generally requires hardware
- ❑ **Checksum**: Weak but easily done in software
  - ❑ **Example**: *1's complement* of 1’s complement sum of 16-bit words with overflow wrapped around



		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

The 1s complement is obtained by converting all the 0s to 1s and converting all the 1s to 0s. Thus the 1s complement of the sum 1011101110111100 is 0100010001000011, which becomes the checksum.

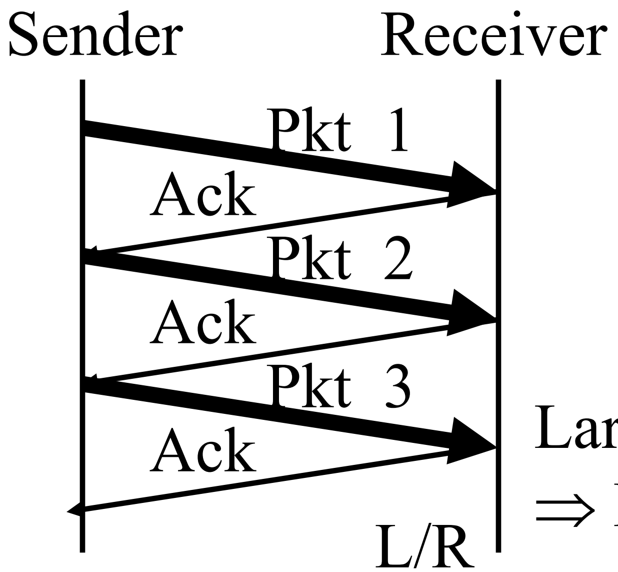
At the receiver, all three 16-bit words are added, including the checksum, and do wraparound if required. If no errors are introduced into the packet, then clearly the sum at the receiver will be 1111111111111111. If one of the bits is a 0, then we know that errors have been introduced into the packet.

Flow Control

Window is the number of packets that can be outstanding without ACK.

- Flow Control Goals:
  1. Sender does not flood the receiver,
  2. Maximize throughput

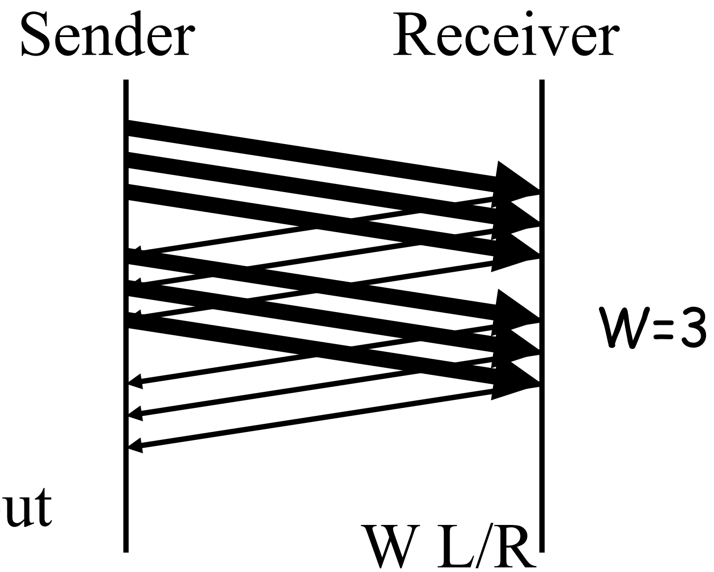
Stop and Wait Flow Control



Large RTT
⇒ Low Thruput

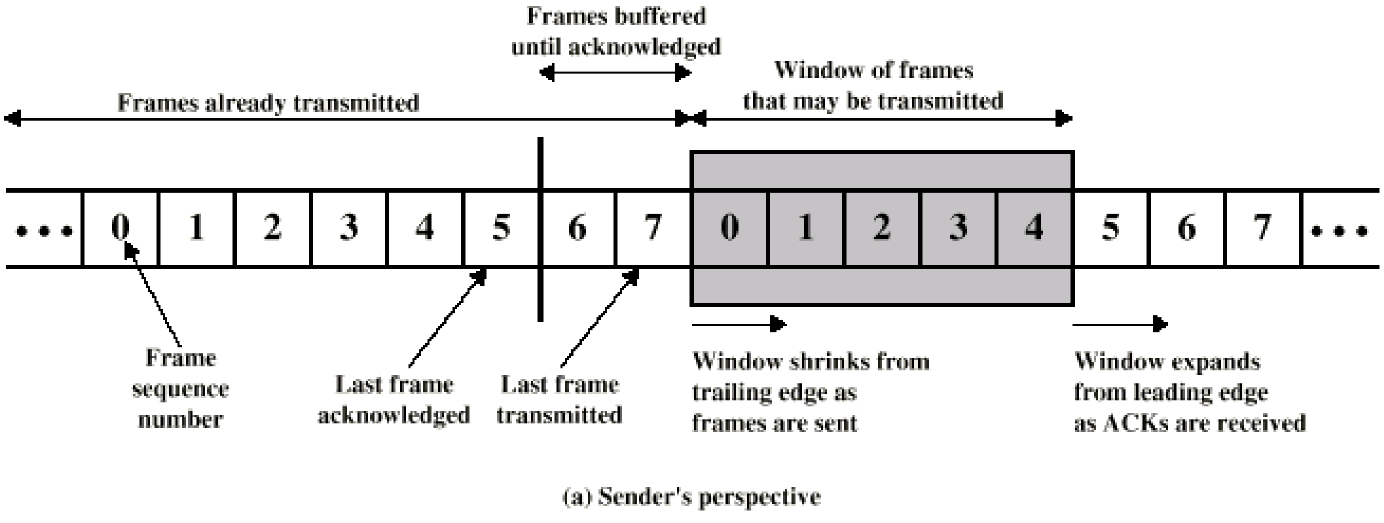
$$\text{Throughput} = \frac{L/R}{RTT+L/R}$$

Window Flow Control

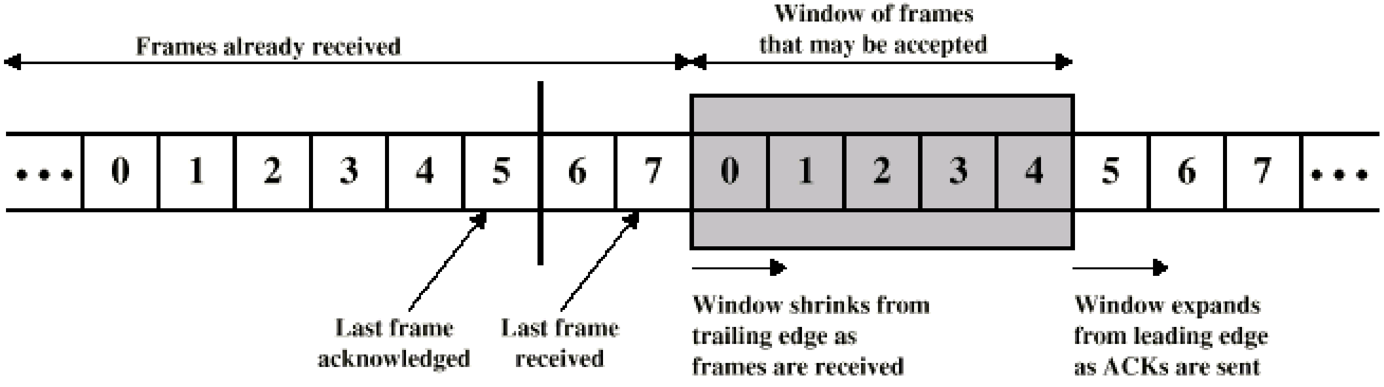


$$\text{Throughput} = \frac{W\ L/R}{RTT+L/R}$$

Sliding Window Diagram

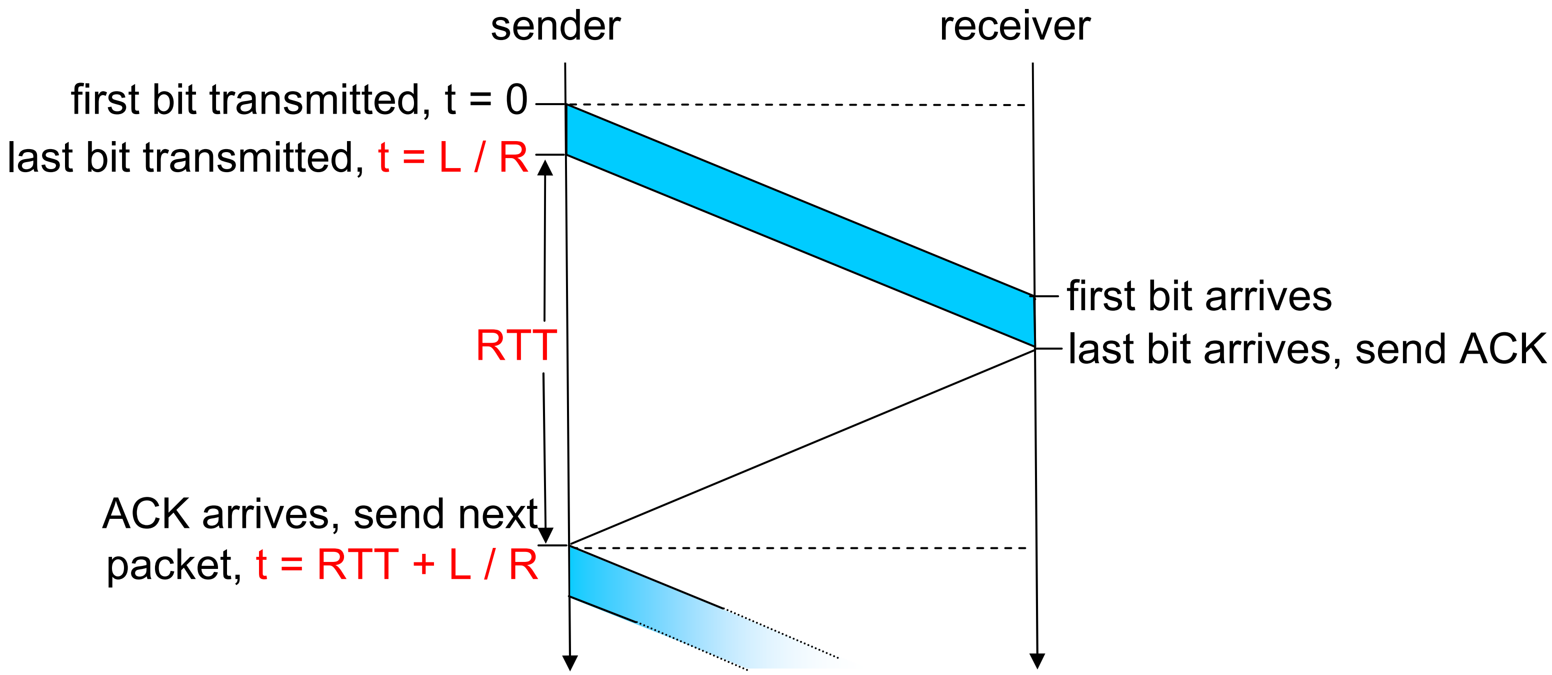


(a) Sender's perspective



(b) Receiver's perspective

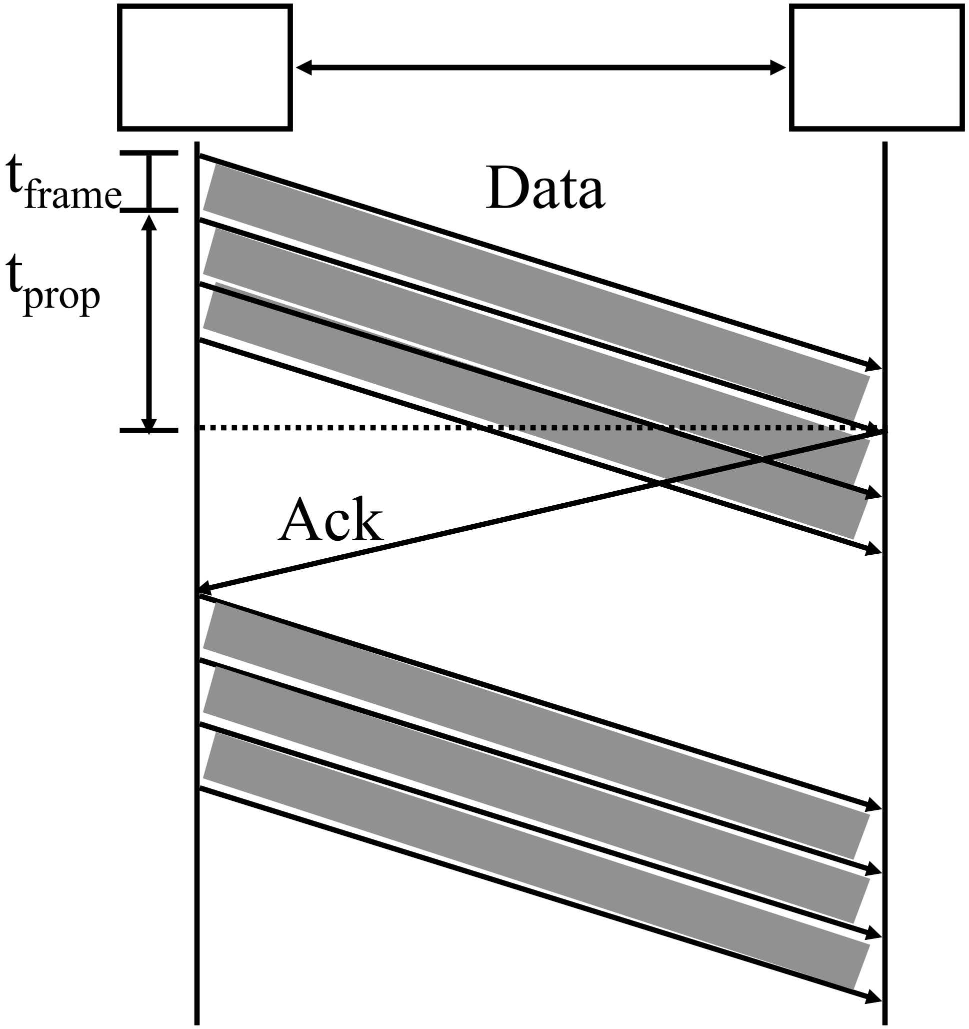
Stop and Wait Flow Control


sender
receiver
first bit transmitted, t = 0
last bit transmitted, t = L / R
RTT
first bit arrives
last bit arrives, send ACK
ACK arrives, send next packet, t = RTT + L / R

$$U = \frac{L/R}{RTT + L/R} = \frac{t_{frame}}{2t_{prop}+t_{frame}} = \frac{1}{2\alpha + 1}$$

Here, $\alpha = t_{prop}/t_{frame}$

Sliding Window Protocol Efficiency



$$U = \frac{W\ t_{frame}}{2t_{prop} + t_{frame}}$$

$$= \begin{cases} \dfrac{W}{2\alpha+1} \\ 1 \text{ if } W > 2\alpha+1 \end{cases}$$

Here, $\alpha = t_{prop}/t_{frame}$

W=1 ⇒ Stop and Wait

Utilization: Examples

Satellite Link: One-way Propagation Delay = 270 ms

RTT=540 ms

Frame Size L = 500 Bytes = 4 kb

Data rate R = 56 kbps ⇒ t_{frame} = L/R= 4/56 = 71 ms

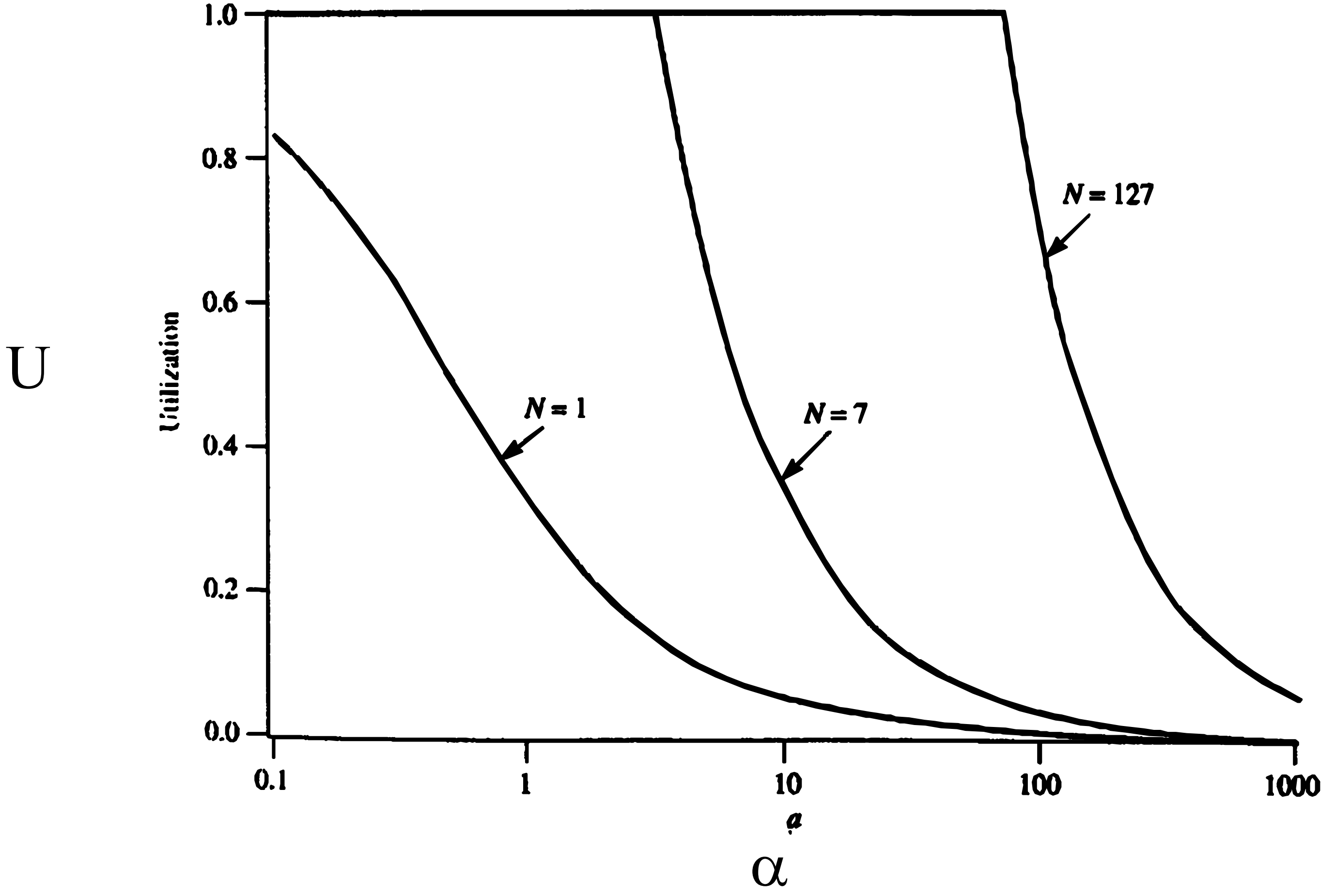
$\alpha = t_{prop}/t_{frame} = 270/71 = 3.8$

$U = 1/(2\alpha+1) = 0.12$

- Short Link: 1 km = 5 μs,
  Rate=10 Mbps,
  Frame=500 bytes ⇒ t_{frame}= 4k/10M= 400 μs
  $\alpha=t_{prop}/t_{frame}=5/400=0.012 \Rightarrow U=1/(2\alpha+1)=0.98$

Note: The textbook uses RTT in place of t_{prop} and L/R for t_{frame}

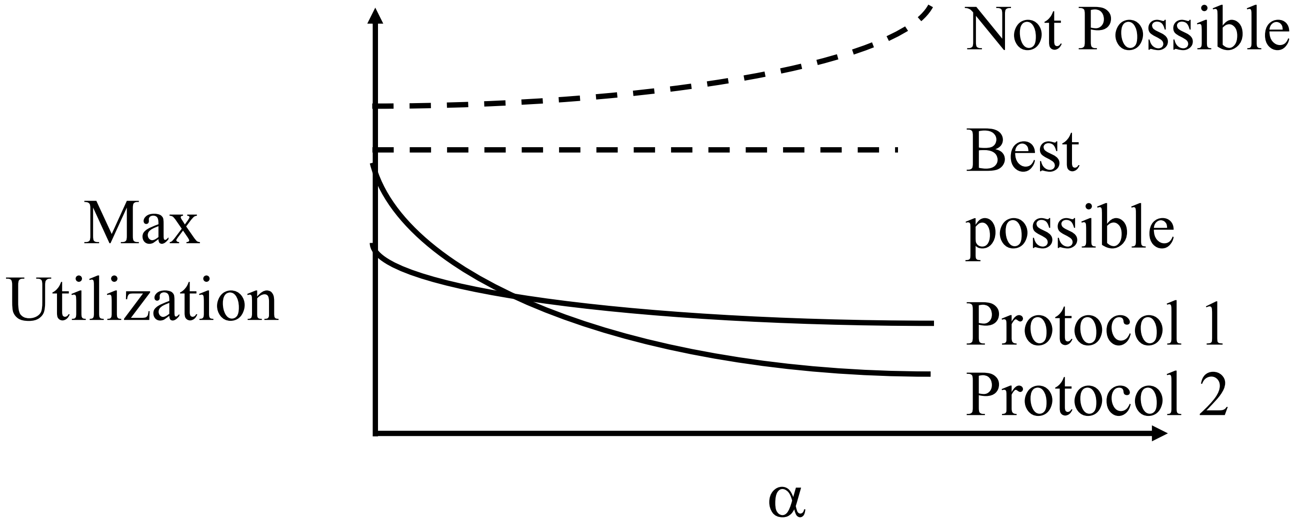
Effect of Window Size



- Larger window is better for larger α

Efficiency Principle

- For **all** protocols, the maximum utilization (efficiency) is a *non-increasing* function of α.



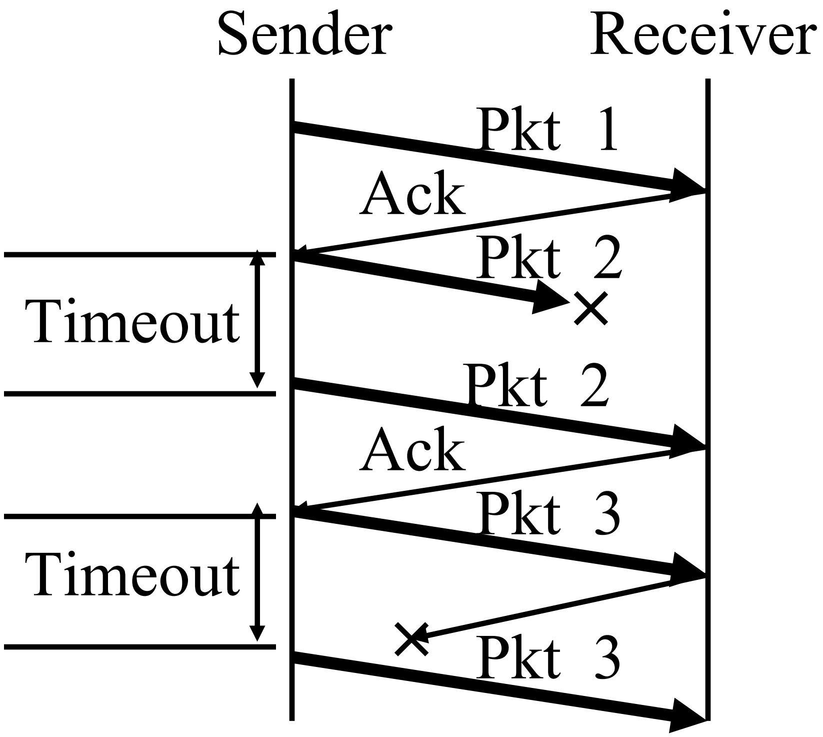
$$\alpha = \frac{t_{prop}}{t_{frame}} = \frac{\text{Distance/Speed of Signal}}{\text{Bits Transmitted /Bit rate}}$$

$$= \frac{\text{Distance} \times \text{Bit rate}}{\text{Bits Transmitted} \times \text{Speed of Signal}}$$

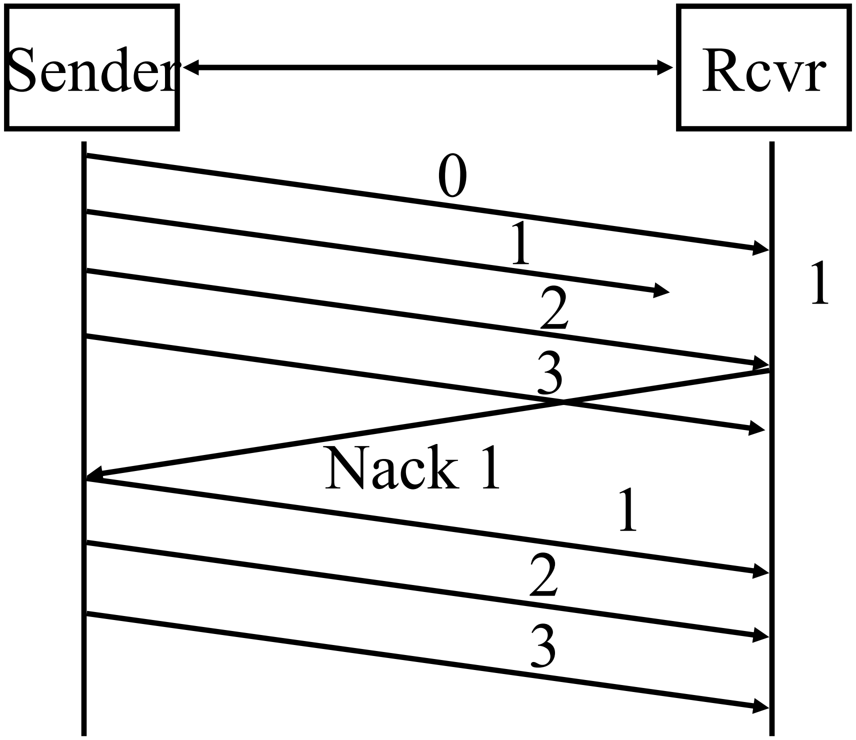
Error Control: Retransmissions

- Retransmit lost packets ⇒ **A**utomatic **R**epeat re**Q**uest (ARQ)

Stop and Wait ARQ

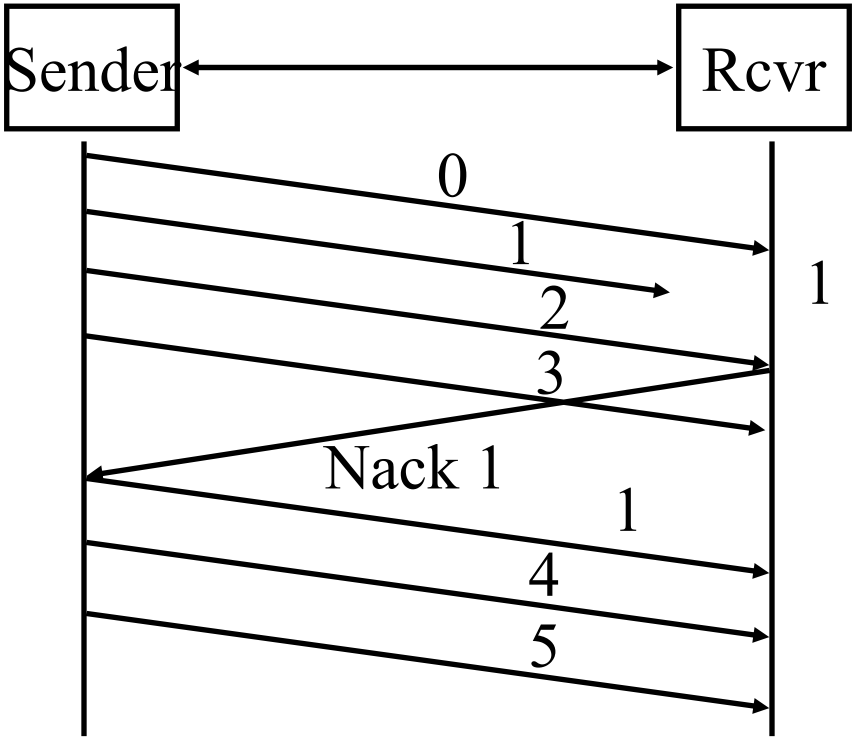


Go-Back-N ARQ



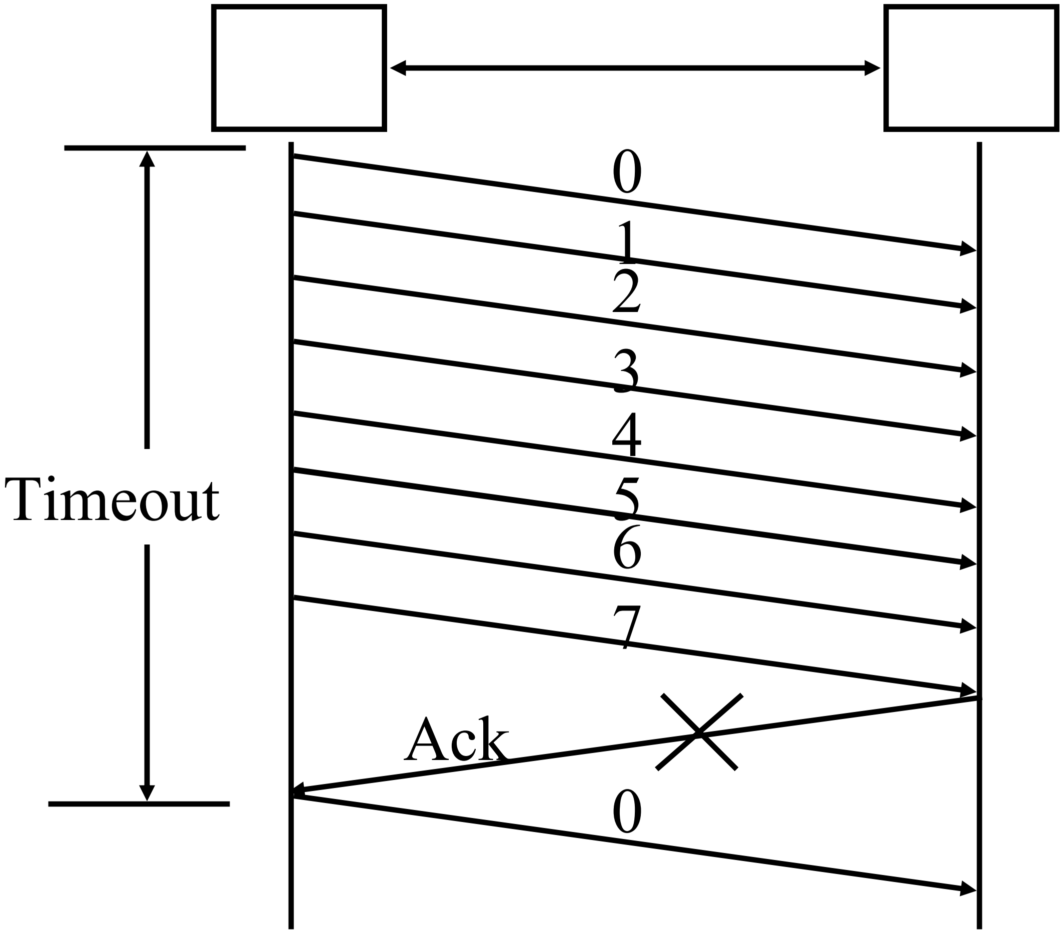
- Receiver does not cache out-of-order frames
- Sender has to ***go back*** and retransmit all frames after the lost frame

Selective Repeat ARQ



- Receiver caches out-of-order frames
- Sender retransmits only the lost frame
- Also known as selective *reject* ARQ

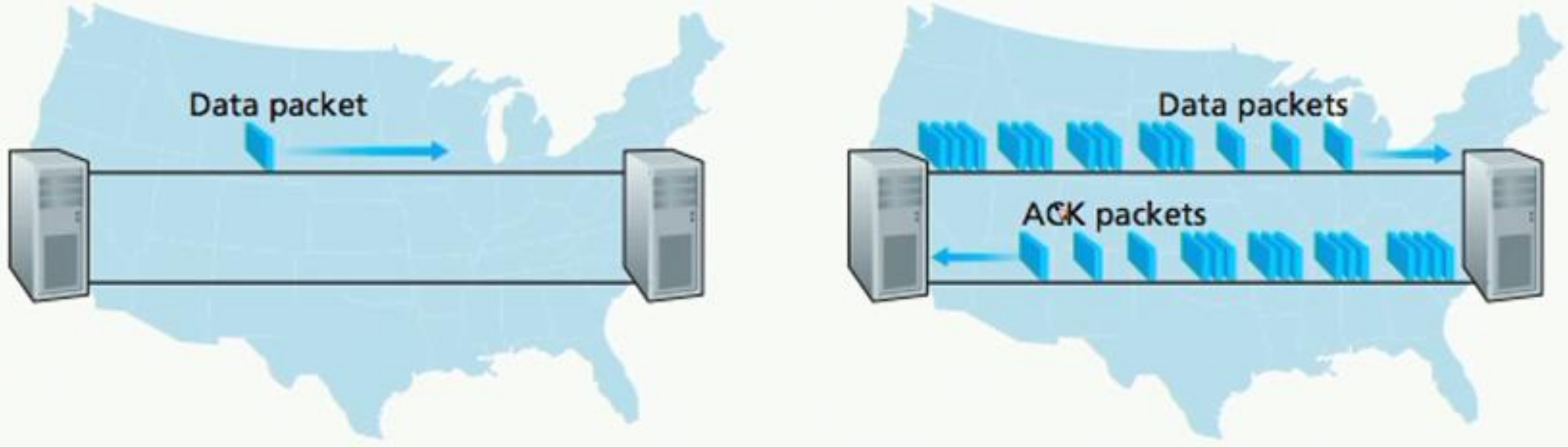
Selective Repeat: Window Size



Sequence number space ≥ 2 window size

Window size ≤ 2^{n-1}

Stop-and-wait vs Sliding windows



a. A stop-and-wait protocol in operation

b. A pipelined protocol in operation

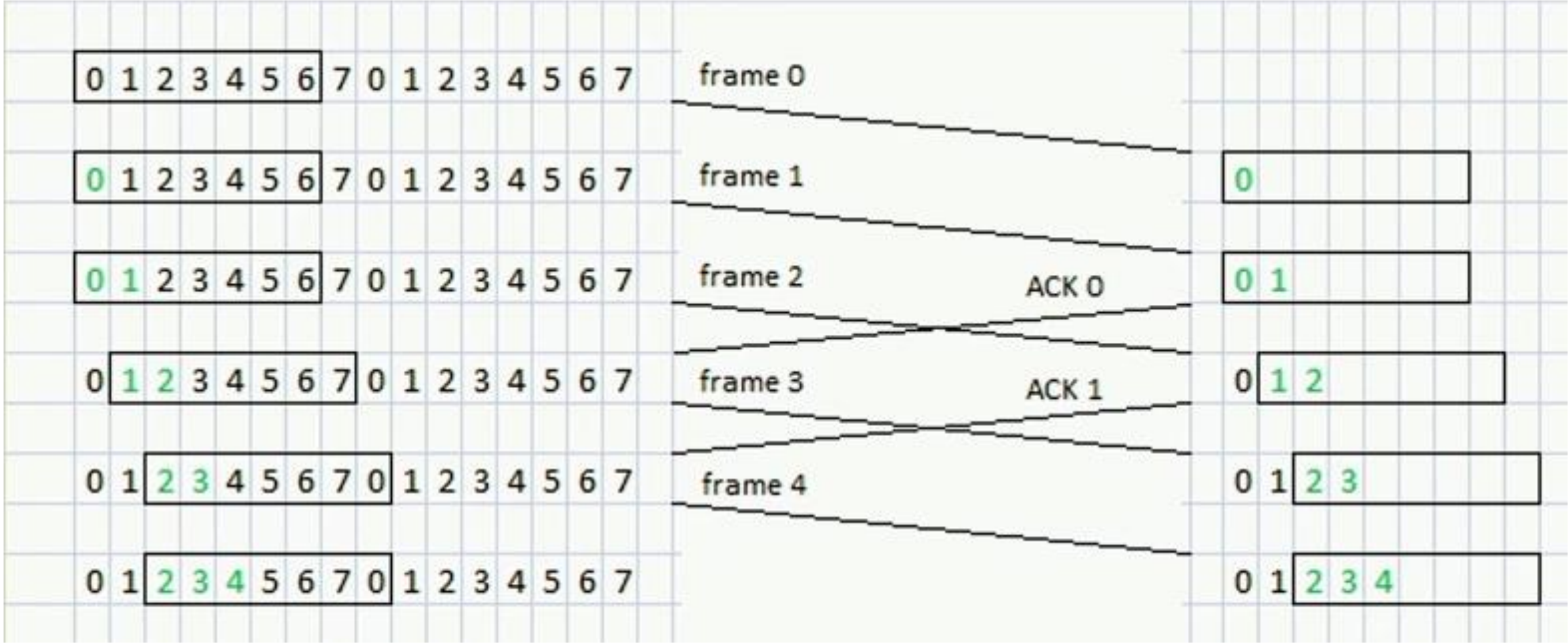
Sliding windows protocol

- Each outbound segment contains a sequence number – from 0 to some maximum (2^n-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 windows protocol: Sender and receiver windows



Sliding windows protocol in noisy channel

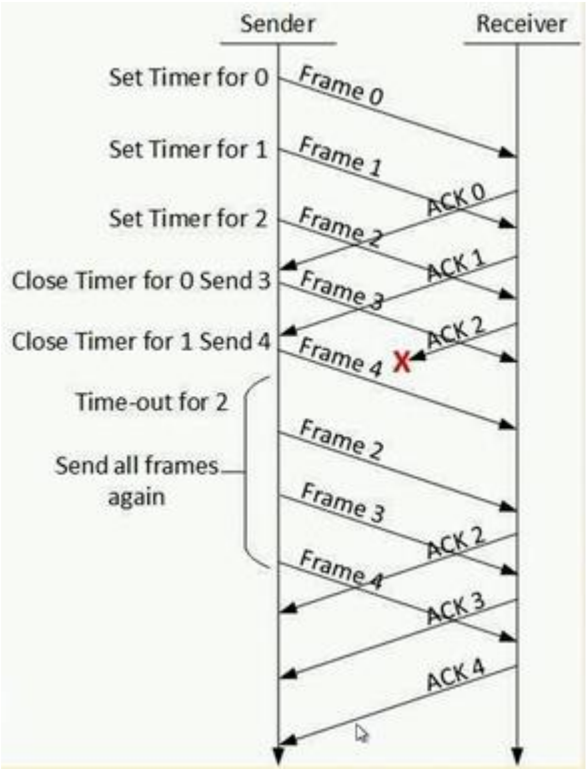
- **A timeout occurs if a segment (or the acknowledgment) gets lost**

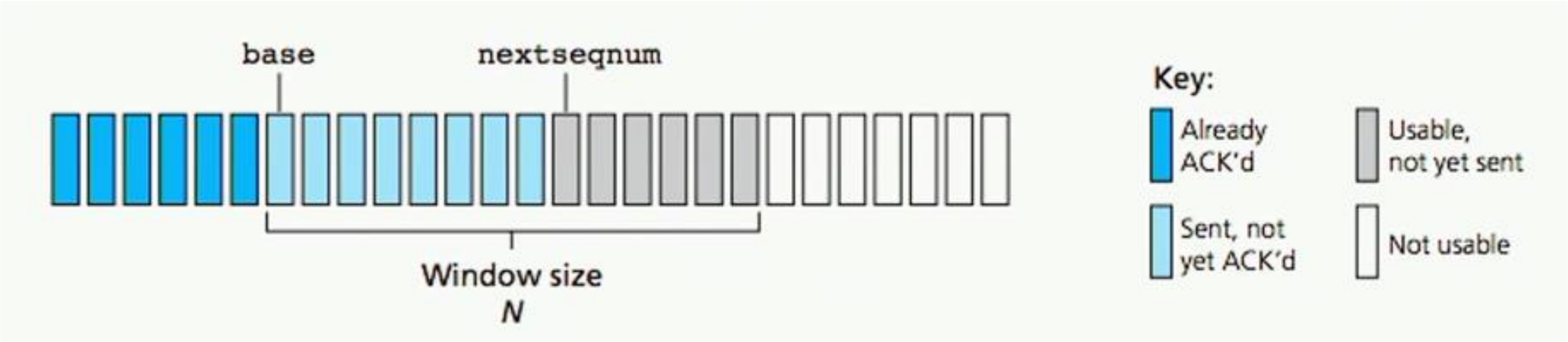
- **How does the flow and error control protocol handle a timeout?**

- **Go Back N ARQ:** If segment N is lost, all the segments from segment 0 (start of the sliding window) to 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: Sender window control





Go back N ARQ: A bound on window size

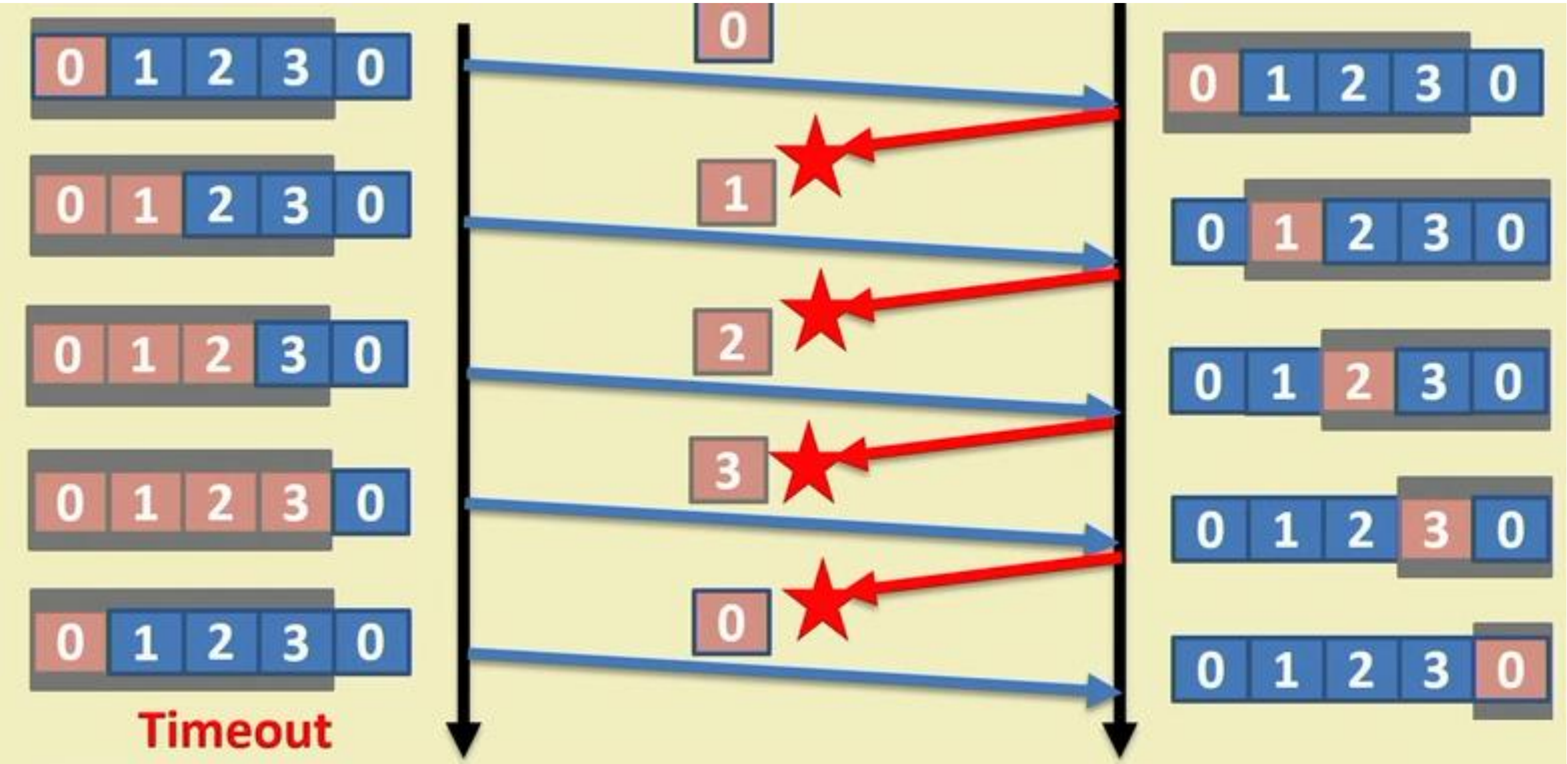
- **Outstanding Frames –** Frames that have been transmitted, but not yet acknowledged

- **Maximum Sequence Number (MAX_SEQ): MAX_SEQ+1** distinct sequence numbers are there
  - **0,1,…,MAX_SEQ**

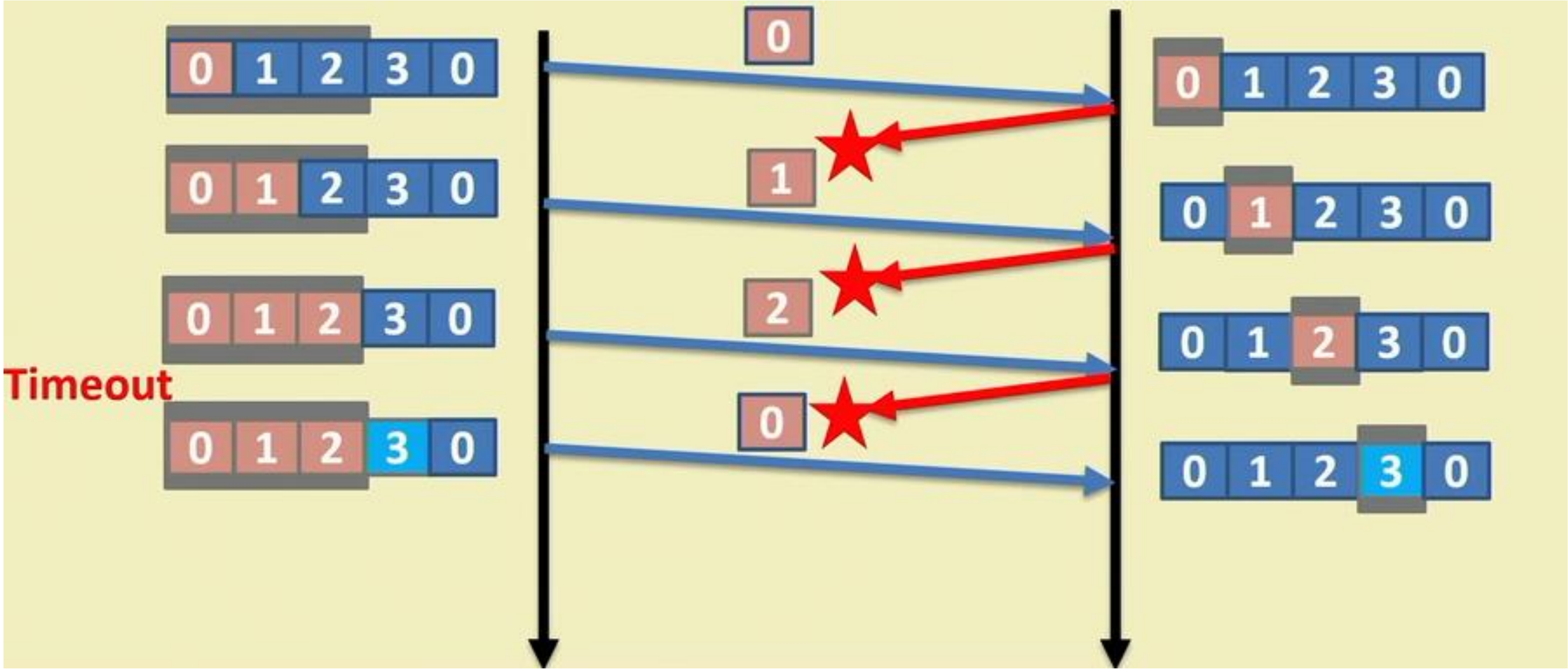
- **Maximum Number of Outstanding Frames (=Window Size): MAX_SEQ**

- **Example:** Sequence Numbers (0,1,2,…,7) – 3 bit sequence numbers, number of outstanding frames = 7 **(Not 8)**

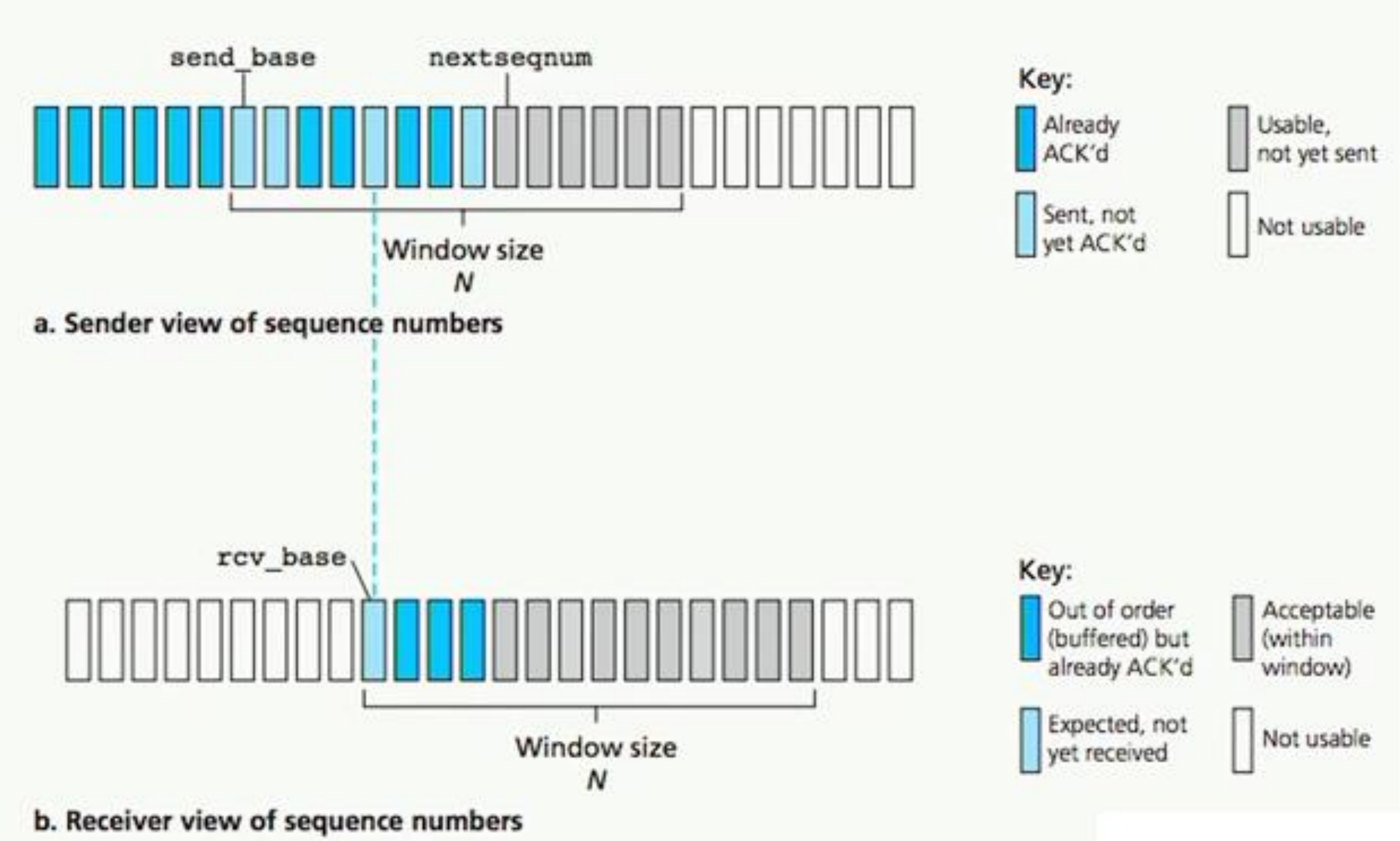
Go Back N ARQ: MAX_SEQ = 3, Window size= 4



Go Back N ARQ: MAX_SEQ = 3, Window size = 3



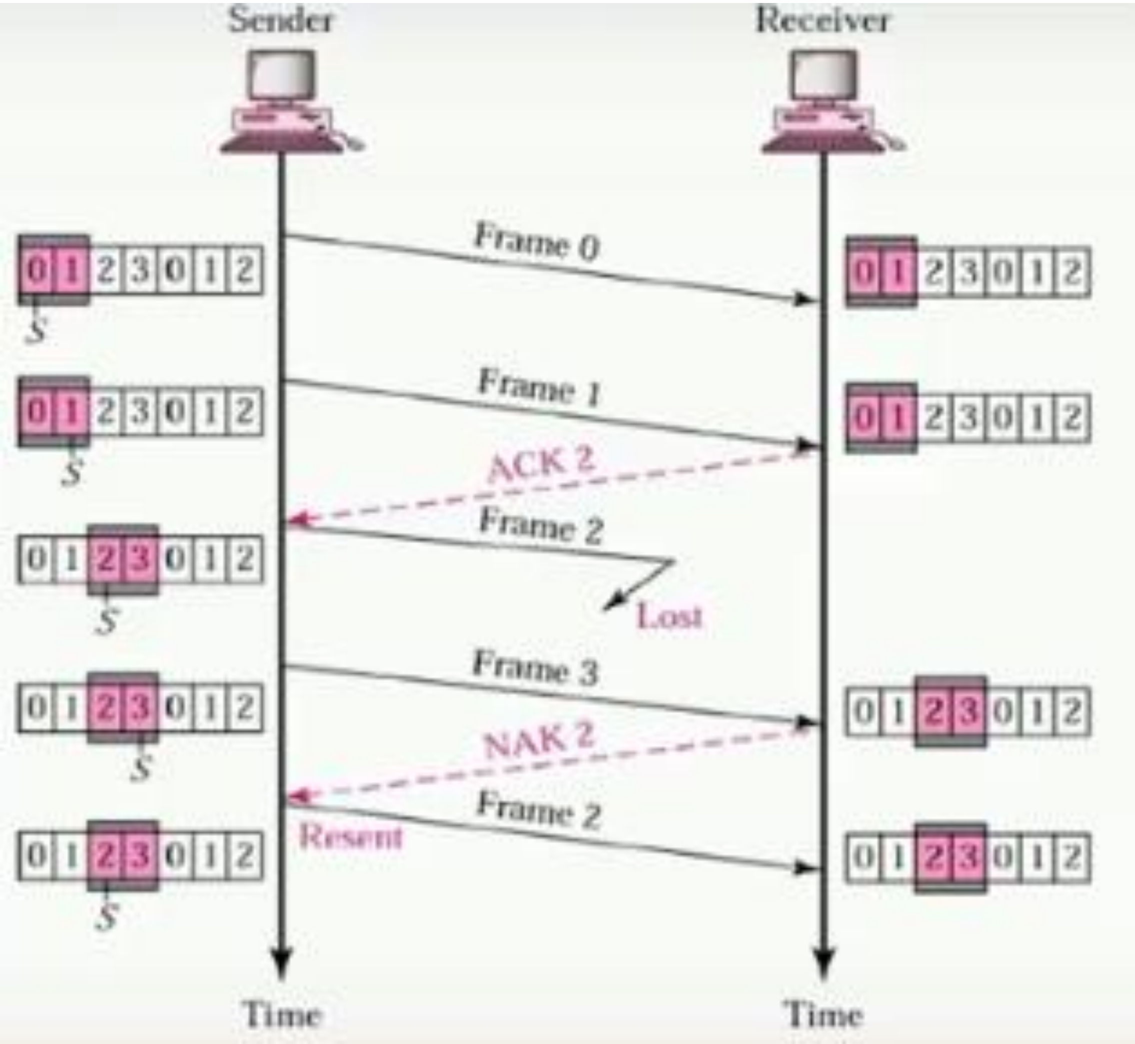
Selective Repeat (SR): Window control



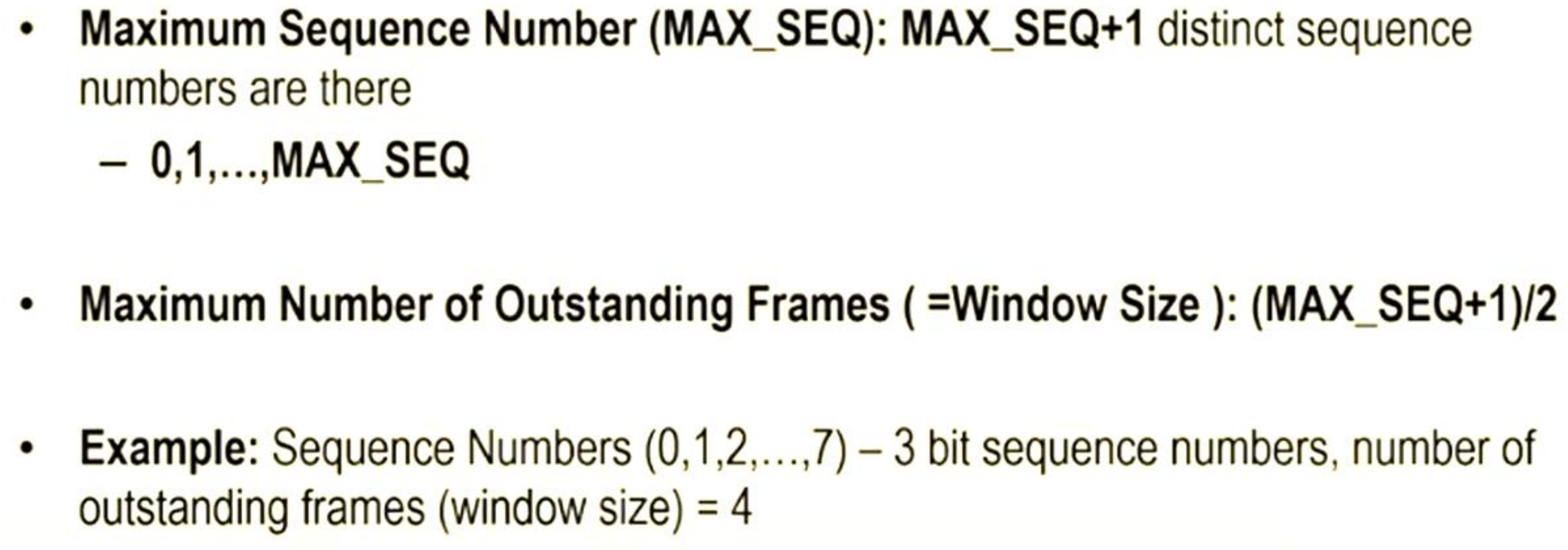
a. Sender view of sequence numbers

b. Receiver view of sequence numbers

Selective Repeat ARQ



Selective Repeat ARQ: A bound on window size

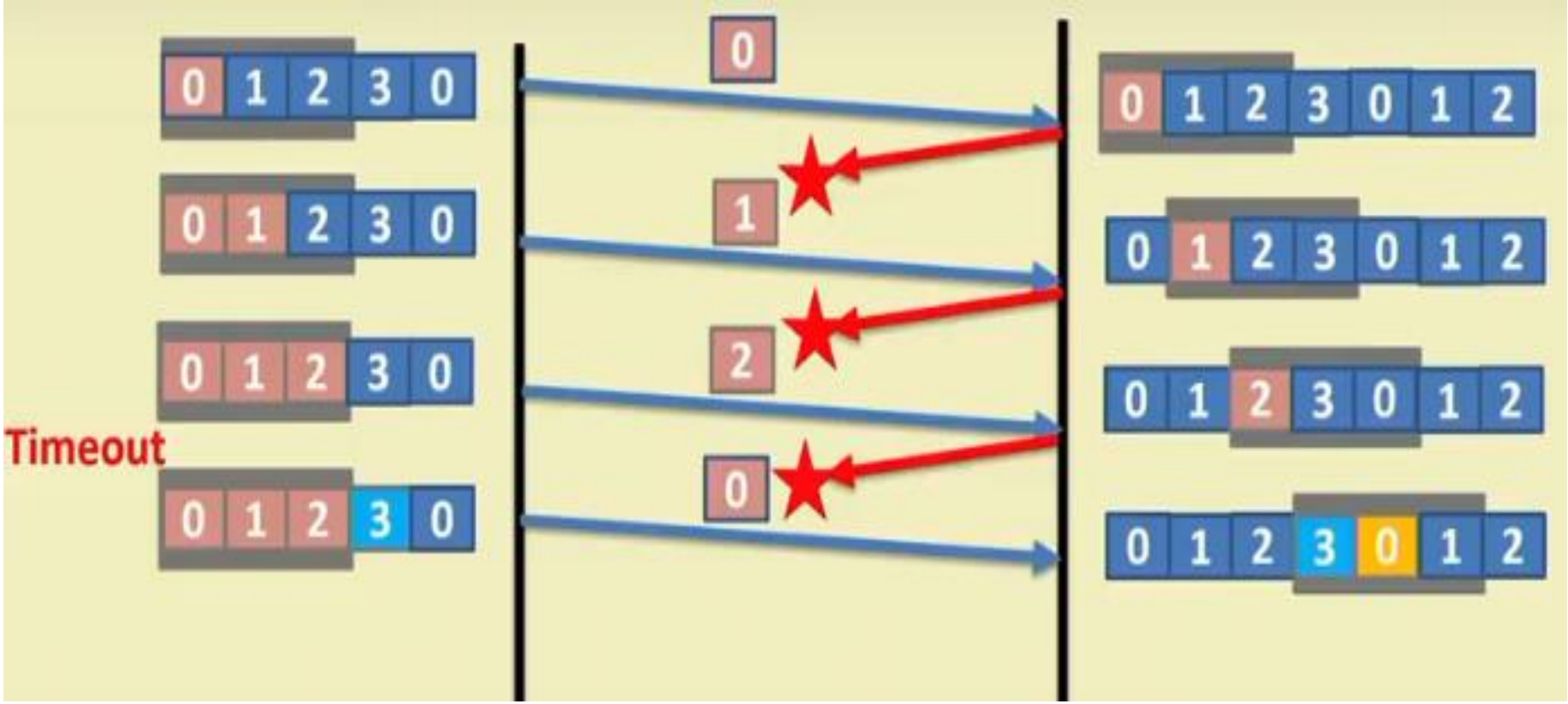


- **Maximum Sequence Number (MAX_SEQ): MAX_SEQ+1** distinct sequence numbers are there
  - **0,1,…,MAX_SEQ**

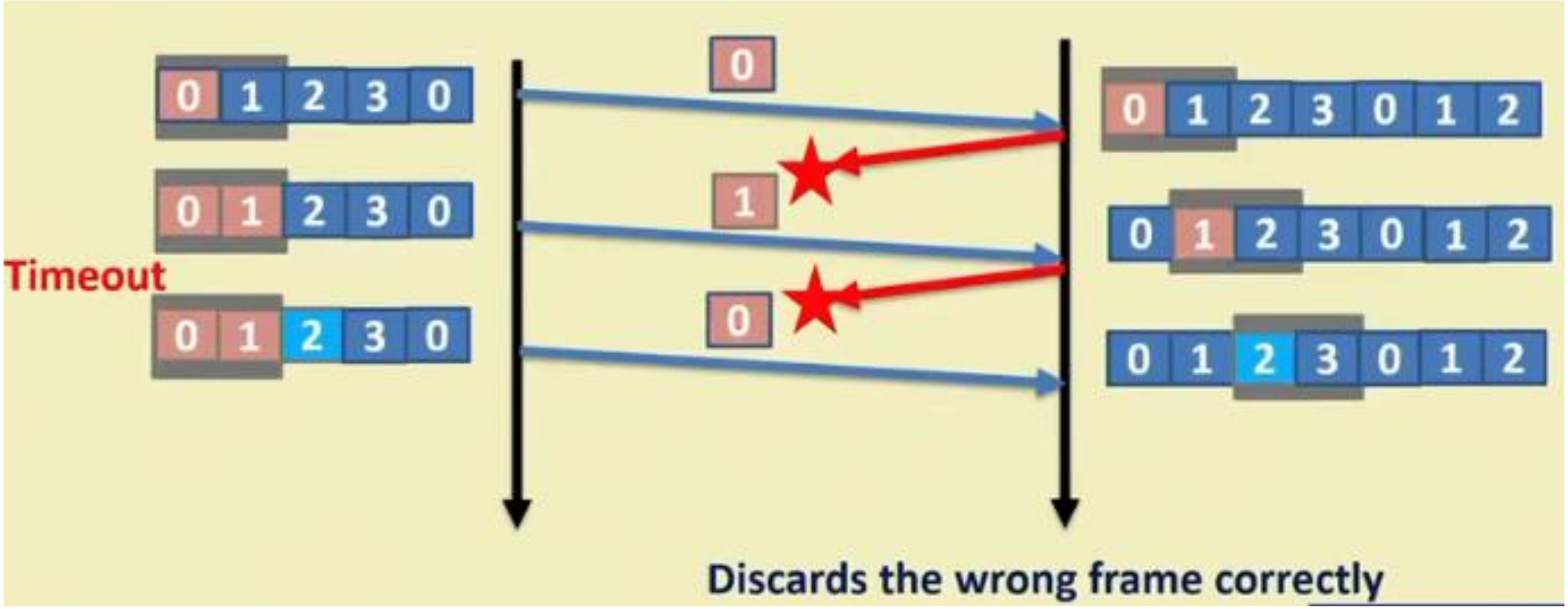
- **Maximum Number of Outstanding Frames (=Window Size): (MAX_SEQ+1)/2**

- **Example:** Sequence Numbers (0,1,2,…,7) – 3 bit sequence numbers, number of outstanding frames (window size) = 4

SR ARQ: MAX_SEQ = 3, Window size = 3



SR ARQ: MAX_SEQ = 3, Window size = 2



Qus: Consider a Selective Repeat Sliding window Protocol that use a frame size of 1 KB to send data on a 1.5 Mbps link with a one-wa c latency of 50 ms. To acheive a link utilization of 60%, the minimum no. of bit required to represent the sequence field is ___ ?

Ans: Frame Size $(L) = 1Kb = 1 \times 8 \times 10^3 = 8000$ bits

Transmission Rate $(R) = 1.5$ Mbps $= 1.5 \times 10^6$ b/s

one-way Propation time = 50 msec.

Required Channel Utilization = 60% = .6

$$U = \frac{W \cdot t_{frame}}{2t_{prop} + t_{frame}} \geq .6$$

$$t_{frame} = \frac{F}{R} = \frac{8000 \text{ bits}}{1.5 \times 10^6 \text{ bits}} = \frac{8000}{1500000} = 0.00533$$

$= 5.33$ ms

$$.6 \leq \frac{W \times 5.33}{5.33 + 100}$$

or $$.6 \leq \frac{5.33\,W}{105.33}$$

or $5.33\,W \geq 0.6 \times 105.33$

or $$W \geq \frac{63.198}{5.33} = 11.85$$

For Selective Repeat Sliding window Protocol, if n bits are used for sequence no, the window size $= 2^{n-1}$

$$\left(\frac{\text{Max seq no} + 1}{2}\right) \Rightarrow \frac{2^n - 1 + 1}{2} = 2^{n-1}$$

$2^{n-1} \geq 11.85$

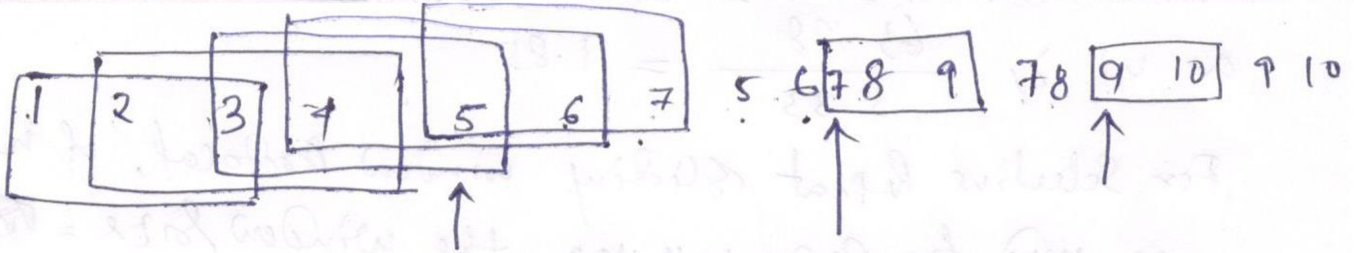
or, $\frac{2^n}{2} \geq 11.85$

or $2^n \geq 23.714$ or $n \log_2 2 \geq \log 23.714$

$\log_2 2^n \geq \log_2 23.714$ $n \geq 4.56$ $\boxed{n = 5}$

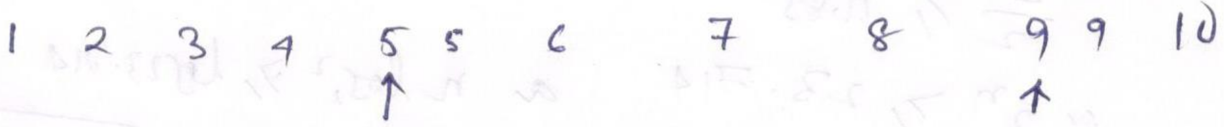
Ques:. Let the window size is 3. 10 Packets are going to sent in a errorieous channel and every 5th Packet is lost during transmission. Total how many transmission is required for Select-Repeat ARQ and Go-back-N-ARQ?

Go-back-N ARQ :-



total transmission = 18

Selective Reject ARQ!



total transmission = 12

Performance: Maximum Utilization

- **Stop and Wait Flow Control**: $U = 1/(1+2\alpha)$
- **Window Flow Control**:

$$U = \begin{cases} 1 & W \geq 2\alpha+1 \\ W/(2\alpha+1) & W < 2\alpha+1 \end{cases}$$

- **Stop and Wait ARQ**: $U = (1-P)/(1+2\alpha)$
- **Go-back-N ARQ**:

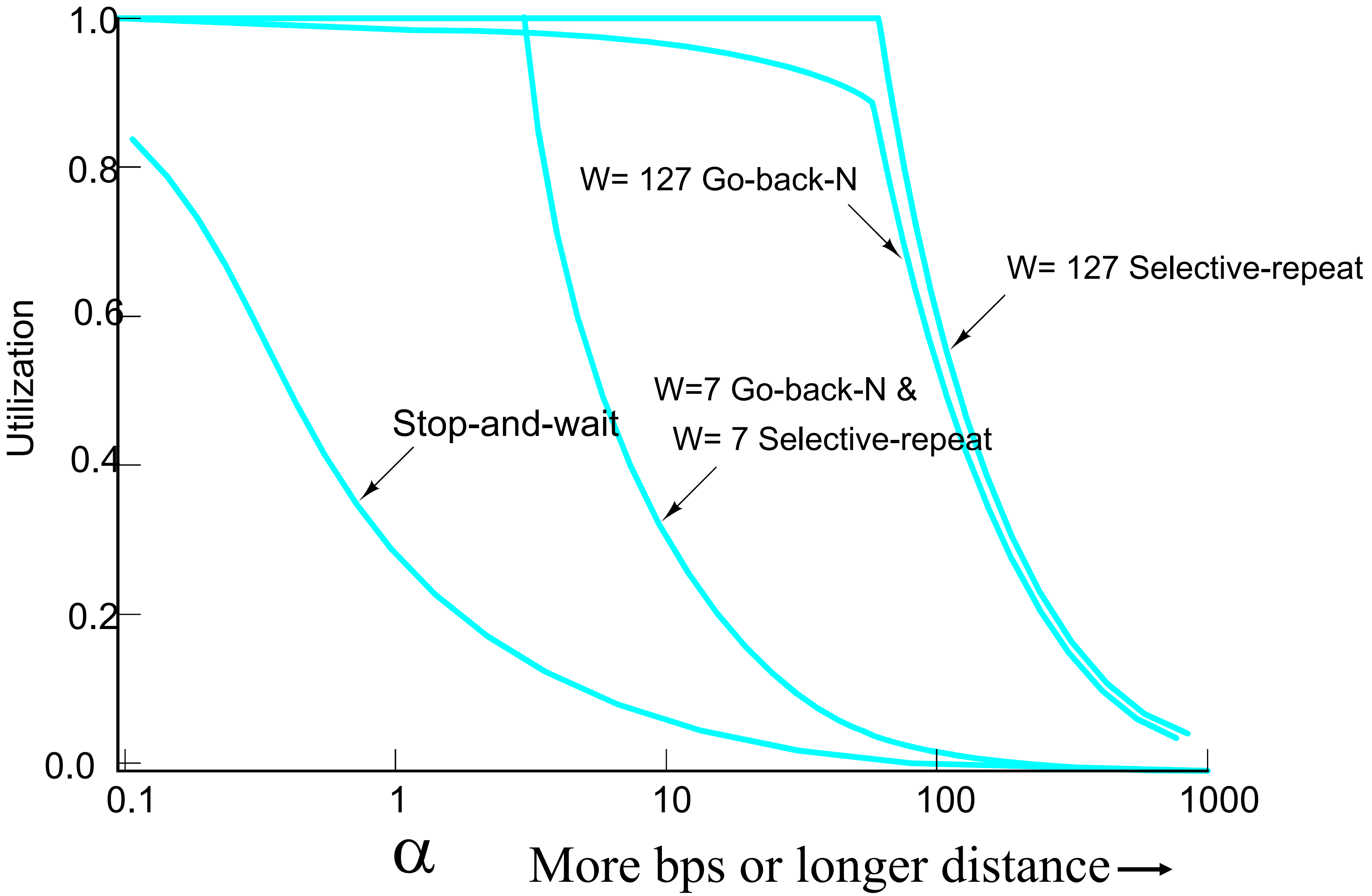
P = Probability of Loss

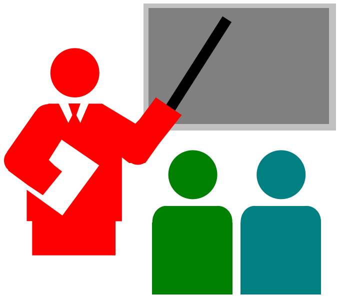
$$U = \begin{cases} (1-P)/(1+2\alpha P) & W \geq 2\alpha+1 \\ W(1-P)/[(2\alpha+1)(1-P+WP)] & W < 2\alpha+1 \end{cases}$$

- **Selective Repeat ARQ**:

$$U = \begin{cases} (1-P) & W \geq 2\alpha+1 \\ W(1-P)/(2\alpha+1) & W < 2\alpha+1 \end{cases}$$

Performance Comparison





Transport Layer Design Issues

1. Multiplexing/demultiplexing by a combination of source and destination IP addresses and port numbers.
2. Window flow control is better for long-distance or high-speed networks
3. Longer distance or higher speed
   ⇒ Larger α ⇒ Larger window is better
4. Stop and and wait flow control is ok for short distance or low-speed networks
5. Selective repeat is better stop and wait ARQ
   Only slightly better than go-back-N

Homework 3A

Problem 19 on page 302 of the textbook:

Consider the GBN protocol with a sender window size of 3 and a sequence number range of 1,024. Suppose that at time t, the next in-order packet that the receiver is expecting has a sequence number of k. Assume that the medium does not reorder messages. Answer the following questions:

A. What are the possible sets of sequence numbers insdie the sender's window at time t? Justify your answer.

B. What are all possible values of the ACK field in all possible messages currently propagating back to the sender at time t? Justify your answer.

Window Flow Control:

C. How big window (in number of packets) is required for the channel utilization to be greater than 60% on a cross-country link of 4000 km running at 20 Mbps using 1 kByte packets?

Efficiency Principle:

D. Ethernet V1 access protocol was designed to run at 10 Mbps over 2.5 Km using 1500 byte packets. This same protocol needs to be used at 100 Mbps at the same efficiency. What distance can it cover if the frame size is not changed?

UDP and TCP

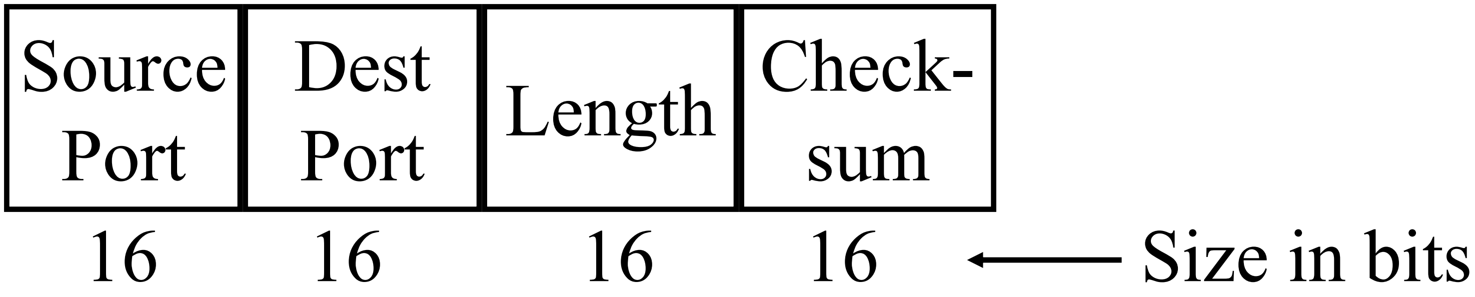
1. User Datagram Protocol (UDP)
2. TCP Header Format, Options, Checksum
3. TCP Connection Management
4. Round Trip Time Estimation
5. Principles of Congestion Control
6. Slow Start Congestion Control

Transports

TCP	UDP
Reliable data transfer	Unreliable Data Transfer
Packet Sequence # required	Sequence # optional
Every packet is acked	Not Acked
Lost packets are retransmitted	No Retransmission
May cause long delay	Quick and Lossy
Connection-oriented service	Connection-less Service
Good for Reliable and delay-insenstive applications	Good for loss-tolerant and delay sensitive applications
Applications: email, http, ftp, Remote terminal access	Telephony, Streaming Multimedia

User Datagram Protocol (UDP)

- Connectionless end-to-end service
- No flow control. No error recovery (no acks)
- Provides multiplexing via ports
- Error detection (Checksum) optional. Applies to pseudo-header (same as TCP) and UDP segment. If not used, it is set to zero.
- Used by network management, DNS, Streamed multimedia (Applications that are loss tolerant, delay sensitive, or have their own reliability mechanisms)



Source Port	Dest Port	Length	Check-sum	
16	16	16	16	← Size in bits

TCP Segment Format

Source Port	Dest Port	Seq No	Ack No	Data Offset	Resvd	U	A	P	R	S	F
16	16	32	32	4	6	1	1	1	1	1	1

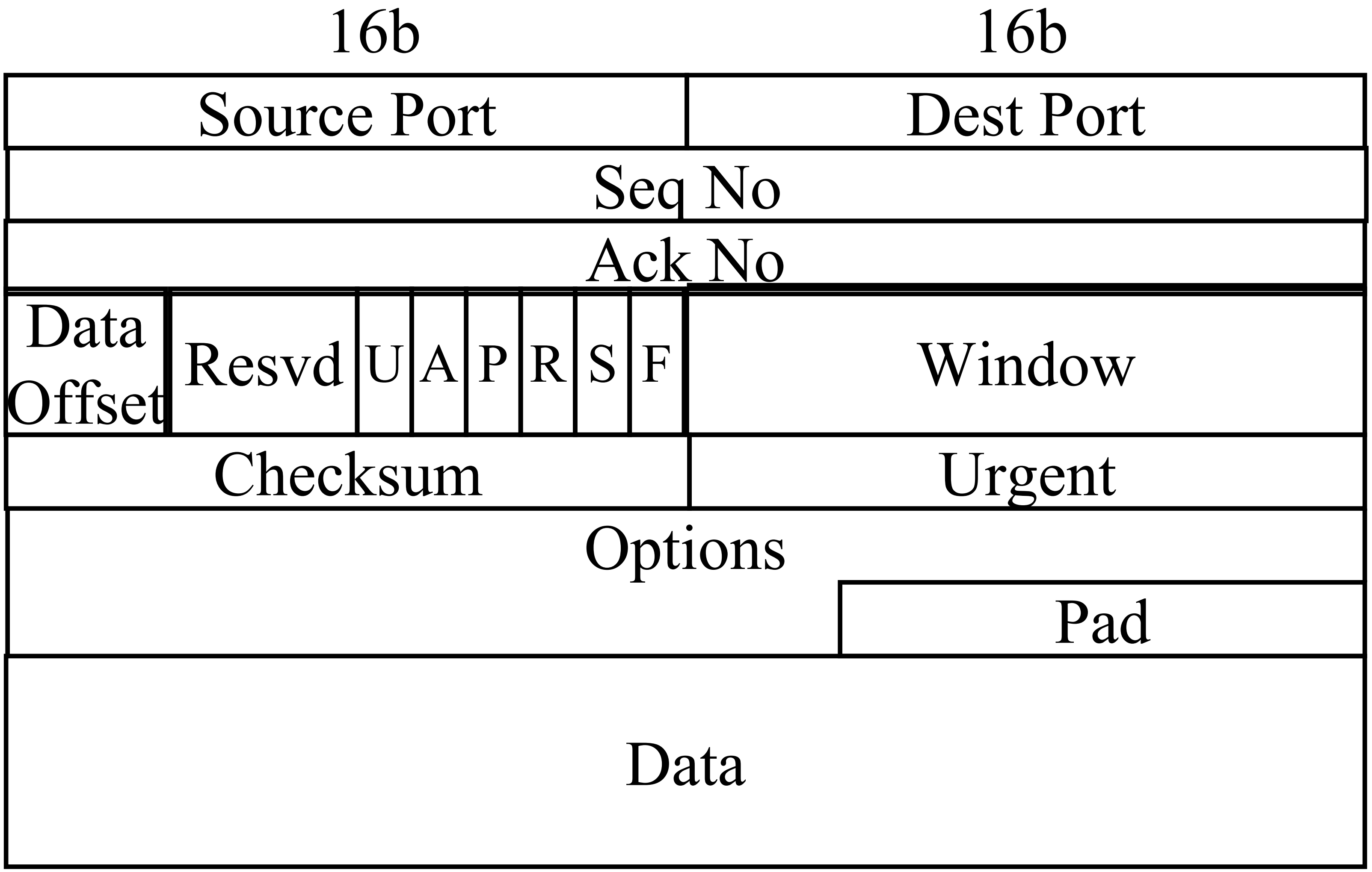
Window	Check-sum	Urgent	Options	Pad	Data
16	16	16	x	y	

← Size in bits

TCP

- Transmission Control Protocol
- Key Services:
  - **Send**: Please send when convenient
  - **Data stream push**: Please send it all now, if possible.
  - **Urgent data signaling**: Destination TCP! please give this urgent data to the user
    (Urgent data is delivered in sequence. Push at the source should be explicit if needed.)
  - Note: Push has no effect on delivery.
    Urgent requests quick delivery

TCP Segment Format (Cont)



16b 16b

Source Port | Dest Port

Seq No

Ack No

Data Offset | Resvd | U | A | P | R | S | F | Window

Checksum | Urgent

Options | Pad

Data

TCP Header Fields

- **Source Port** (16 bits): Identifies source user process
- **Destination Port** (16 bits)
  21 = FTP, 23 = Telnet, 53 = DNS, 80 = HTTP, ...
- **Sequence Number** (32 bits): Sequence number of the first byte in the segment. If SYN is present, this is the initial sequence number (ISN) and the first data byte is ISN+1.
- **Ack number** (32 bits): Next byte expected
- **Data offset** (4 bits): Number of 32-bit words in the header
- **Reserved** (6 bits)

TCP Header (Cont)

- **Control** (6 bits):Urgent pointer field significant,
  Ack field significant,
  Push function,
  Reset the connection,
  Synchronize the sequence numbers,
  No more data from sender

URG	ACK	PSH	RST	SYN	FIN

- **Window** (16 bits): Advertise the window size

TCP Header (Cont)

- **Checksum** (16 bits): covers the segment plus a pseudo header. Includes the following fields from IP header: source and dest adr, protocol, segment length. Protects from IP misdelivery.
- **Urgent pointer** (16 bits): Points to the byte following urgent data. Lets receiver know how much data it should deliver right away.
- **Options** (variable):
  Max segment size (does not include TCP header, default 536 bytes), Window scale factor, Selective Ack permitted, Timestamp, No-Op, End-of-options

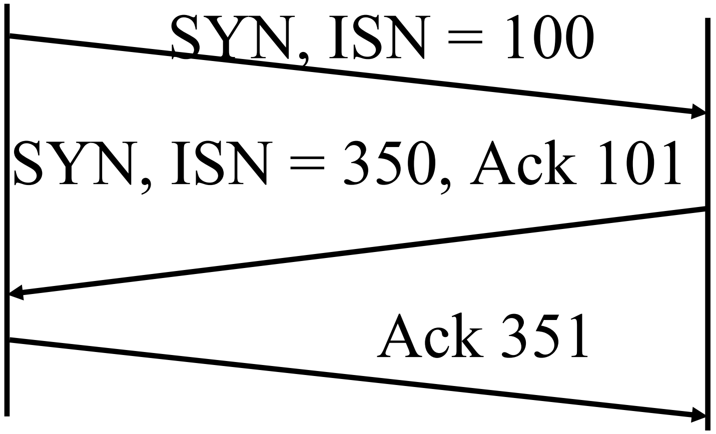
TCP Options

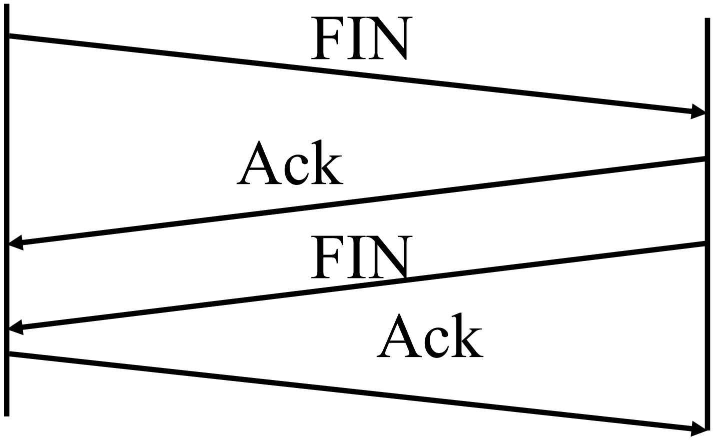
Kind	Length	Meaning
0	1	End of Valid options in header
1	1	No-op
2	4	Maximum Segment Size
3	3	Window Scale Factor
8	10	Timestamp

- **End of Options**: Stop looking for further option
- **No-op**: Ignore this byte. Used to align the next option on a 4-byte word boundary
- **Max Segment Size (MSS):** Does not include TCP header

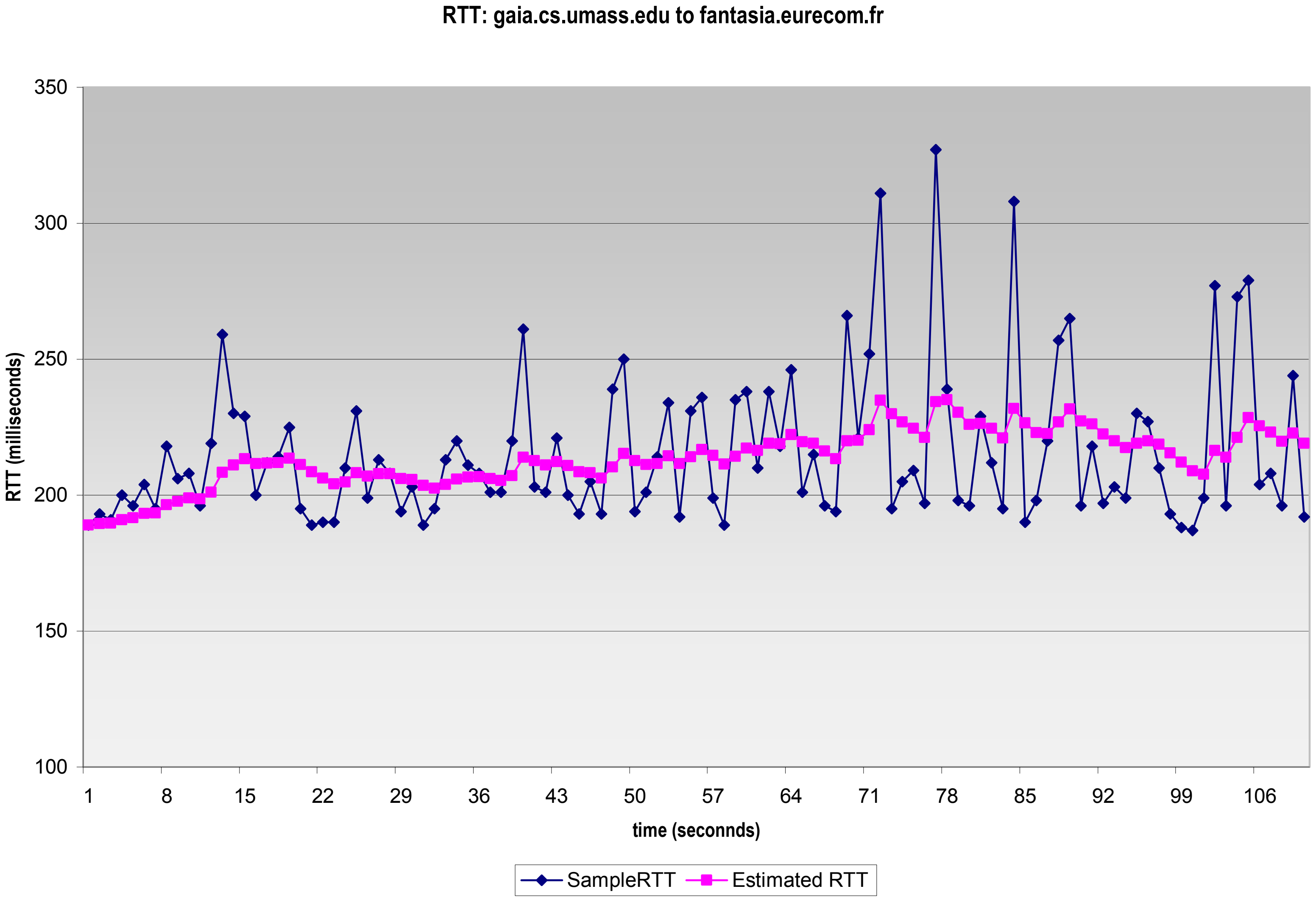
TCP Connection Management

- ❑ Connection Establishment
  - ❑ Three way handshake
  - ❑ SYN flag set
    ⇒ Request for connection
- ❑ Connection Termination
  - ❑ Close with FIN flag set
  - ❑ Abort



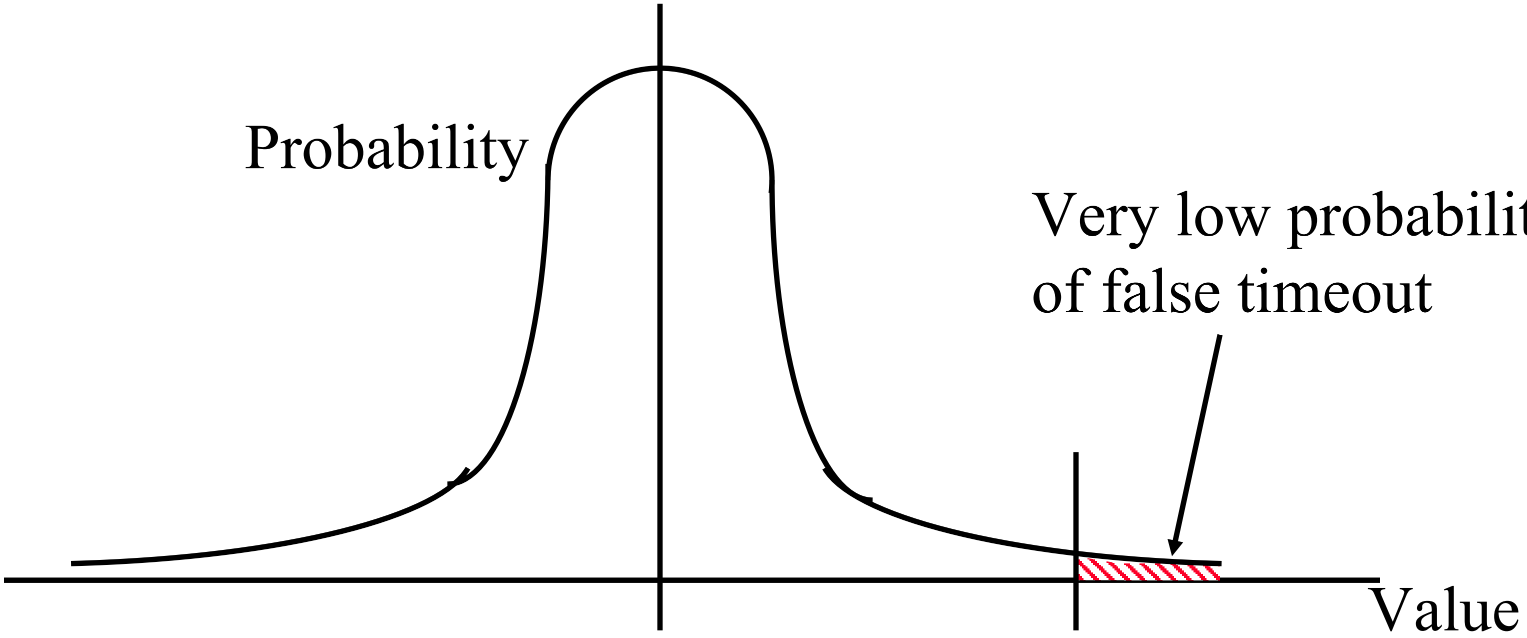


Example RTT estimation:



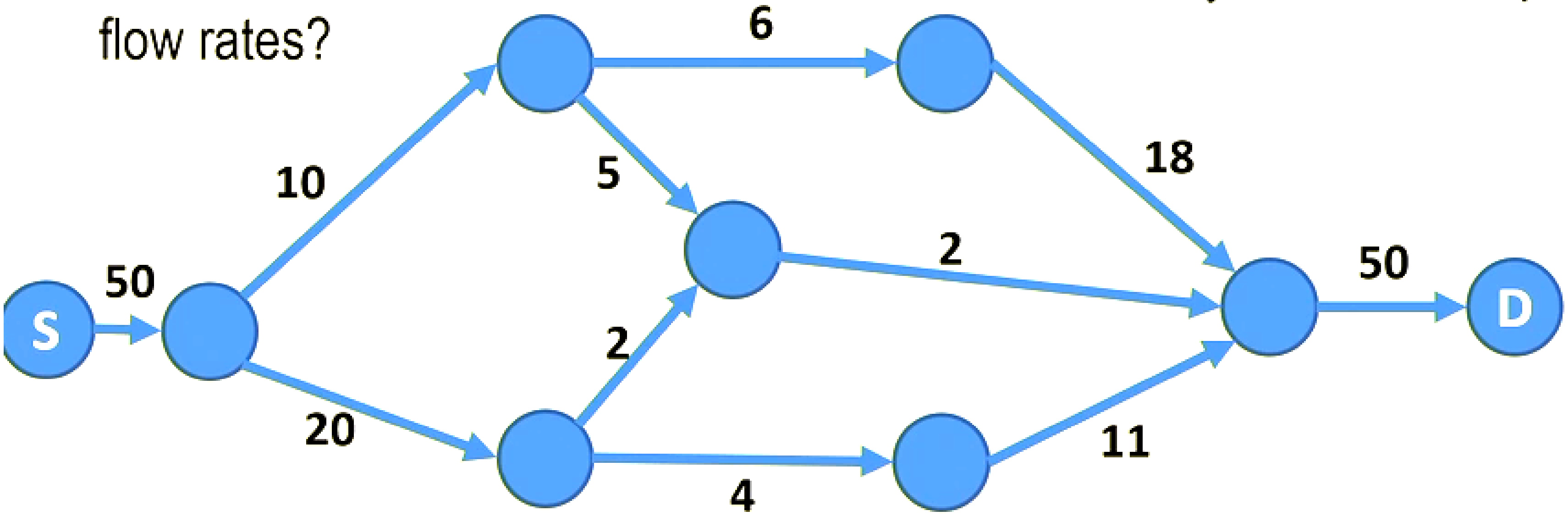
Round Trip Time Estimation

- Measured round trip time (SampleRTT) is very random.
- EstimatedRTT=(1- α)EstimatedRTT+α SampleRTT
- DevRTT = (1-β)DevRTT+ β |SampleRTT-EstmatedRTT|
- TimeoutInterval=EstimatedRTT+**4** DevRTT



Congestion control in a network

- Consider a centralized network scenario – how can you maintain optimal flow rates?



Congestion control in a network

- Flows enter and exit network dynamically – so applying an algorithm for congestion control is difficult

- **Congestion avoidance:** Regulate the sending rate based on what the network can support

  Sending Rate = minimum (network rate, Receiver rate)

Congestion control and fairness

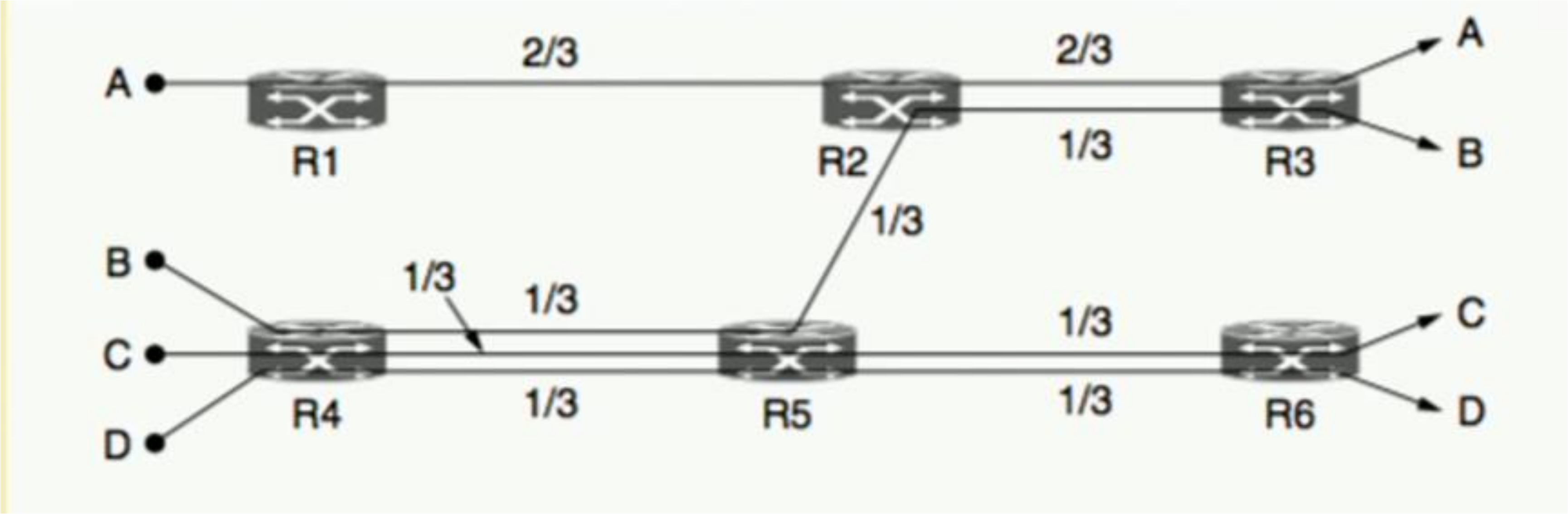
- Ensure that the rate of all the flows in the network is controlled in a **fair way**

- A bad congestion control algorithm may affect fairness - Some flows can get starved

- Hard fairness in a decentralized network is difficult to implement

- **Max-Min Fairness**: An allocation is max-min fair if the bandwidth given to one flow cannot be increased without decreasing the bandwidth given to another flow with an allocation.

Max-Min Fairness – An example



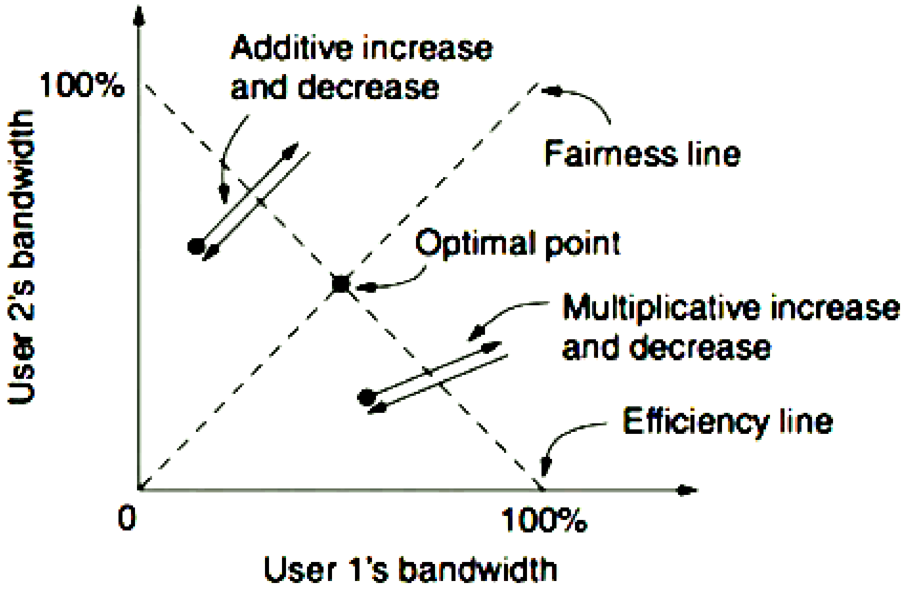
AIMD: Additive Increase Multiplicative Decrease

- **Additive Increase Multiplicative Decrease (AIMD)** – Chiu and Jain (1989)

- Let *w(t)* be the sending rate. *a (a > 0)* is the additive increase factor, and *b (0<b<1)* is the multiplicative decrease factor

$$w(t+1) = \begin{cases} w(t) + a & \text{if congestion is not detected} \\ w(t) \times b & \text{if congestion is detected} \end{cases}$$

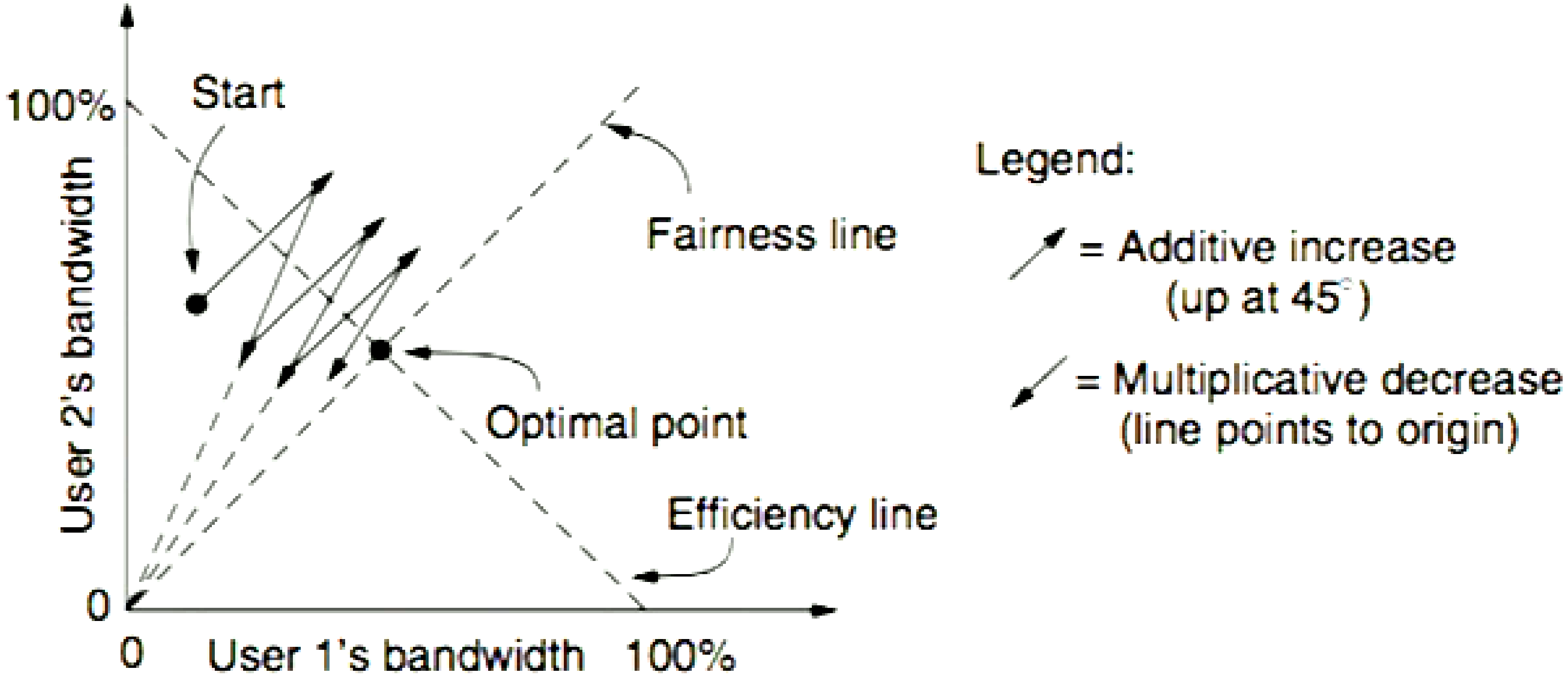
AIMD: Two flows example



- **AIAD** – Oscillate across the efficiency line

- **MIMD** – Oscillate across the efficiency line (different slope from AIAD)

AIMD: Two flows example



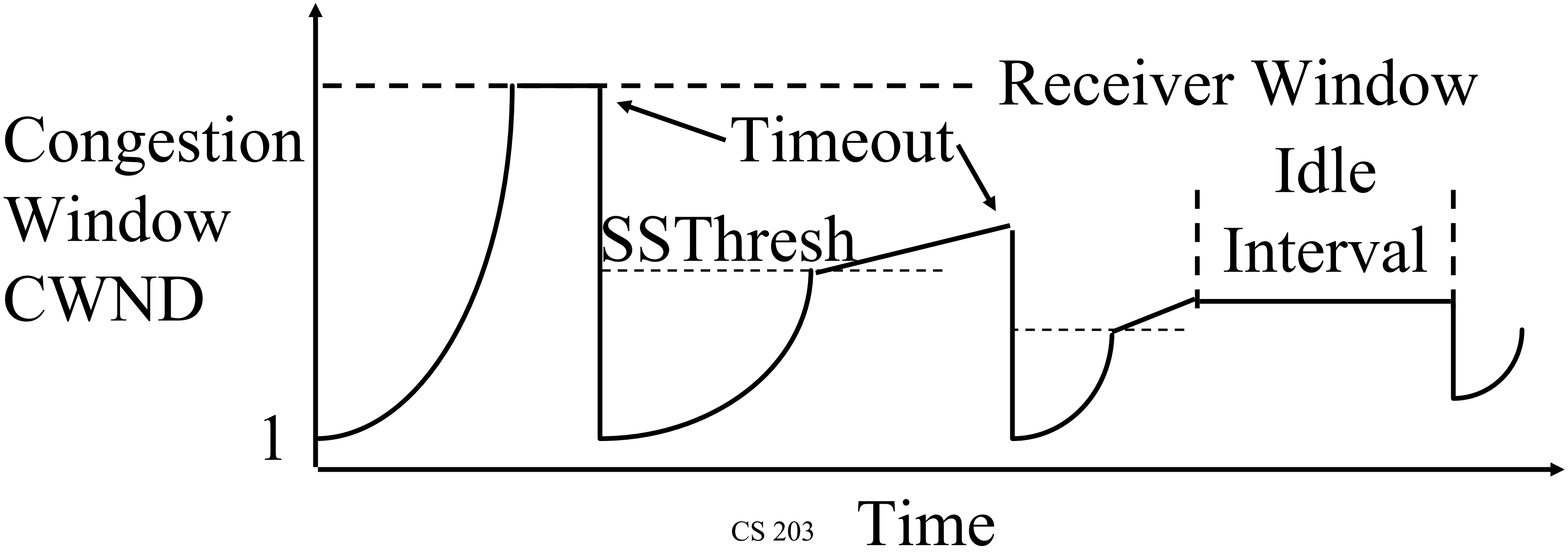
- The path converges towards the optimal point
- Used by TCP - Adjust the size of the sliding window

Slow Start Congestion Control

- Window = Flow Control Avoids receiver overrun
- Need congestion control to avoid network overrun
- The sender maintains two windows:
  Credits from the receiver
  Congestion window from the network
  Congestion window is always less than the receiver window
- Starts with a congestion window (CWND) of 1 segment (one max segment size)
  ⇒ Do not disturb existing connections too much.
- Increase CWND by 1 MSS every time an ack is received
- Assume CWND is in bytes

Slow Start (Cont)

- If segments lost, remember slow start threshold (SSThresh) to CWND/2
  Set CWND to 1 MSS
  Increment by 1MSS per ack until SSthresh
  Increment by 1 MSS*MSS/CWND per ack afterwards



Slow Start (Cont)

- At the beginning, SSThresh = Receiver window
- After a long idle period (exceeding one round-trip time), reset the congestion window to one.
- Exponential growth phase is also known as “Slow start” phase
- The linear growth phase is known as “congestion avoidance phase”

Fast Recovery

- Optional – implemented in TCP Reno (Earlier version was TCP Tahoe)
- Duplicate Ack indicates a lost/out-of-order segment
- On receiving 3 duplicate acks (4th ack for the same segment):
  - Enter Fast Recovery mode
    - Retransmit missing segment
    - Set SSTHRESH=CWND/2
    - Set CWND=SSTHRESH+3 MSS
    - Every subsequent duplicate ack: CWND=CWND+1MSS
  - When a new ack (not a duplicate ack) is received
    - Exit fast recovery
    - Set CWND=SSTHRESH

TCP Average Throughput

- Average Throughput = $\dfrac{1.22\ \text{MSS}}{\text{RTT}\ \sqrt{\text{P}}}$

- Here, P = Probability of Packet loss

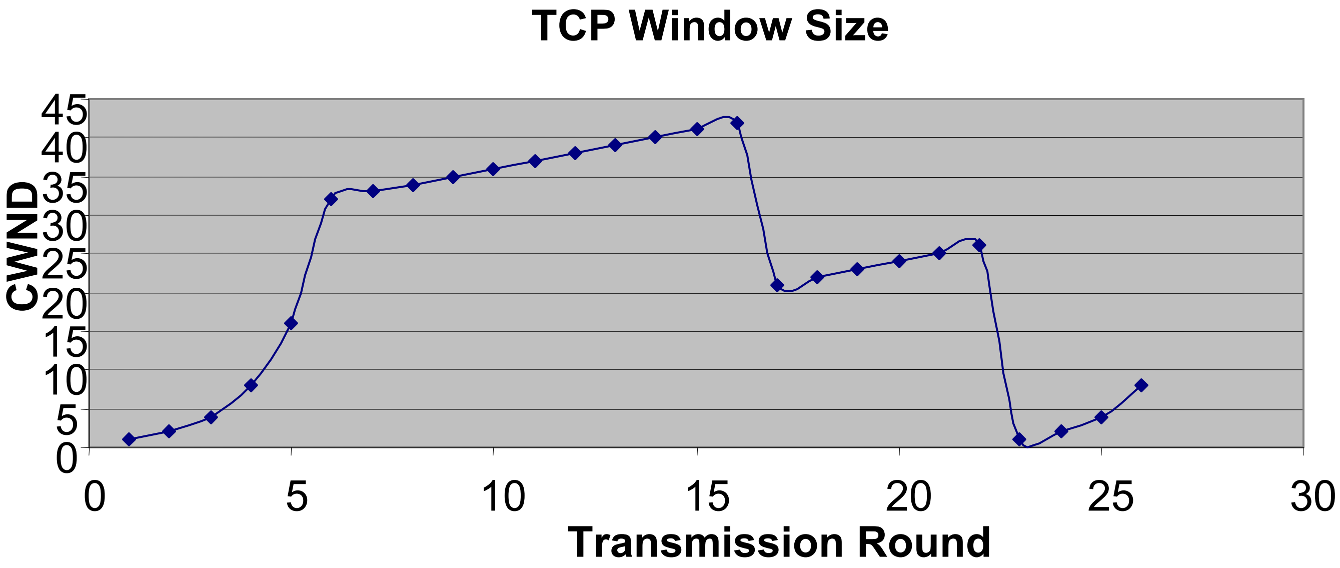
UDP and TCP: Summary

1. UDP provides flow multiplexing and optional checksum
2. Both UDP and TCP use port numbers for multiplexing
3. TCP provides reliable full-duplex connections.
4. TCP is stream based and has credit flow control
5. Slow-start congestion control works on timeout

Homework 3B

- Problem P37 on page 306 of the textbook:
- Consider Figure 3.58. Assuming TCP Reno is the protocol experiencing the behavior shown above, answer the following questions. In all cases, you should provide a short discussion justifying your answer.

Round	CWND
1	1
2	2
3	4
4	8
5	16
6	32
7	33
8	34
9	35
10	36
11	37
12	38
13	39
14	40
15	41
16	42
17	21
18	22
19	23
20	24
21	25
22	26
23	1
24	2
25	4
26	8



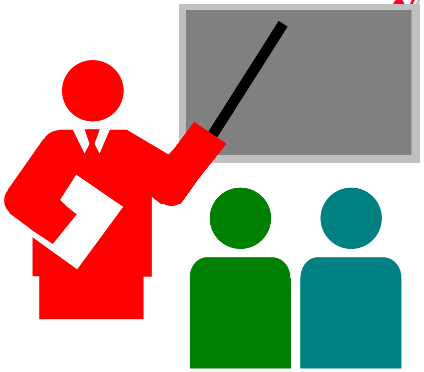
Homework 3B (Cont)

- A. Identify the interval of time when TCP slow start is operating.
- B. Identify the intervals of time when TCP congestion avoidance is operating.
- C. After the 16th transmission round, is segment loss detected by a triple duplicate ACK or by a timeout?
- D. After the 22nd transmission round, is segment loss detected by a triple duplicate ACK or by a timeout?
- E. What is the initial value of ssthresh at the first transmission round?
- F .What is the value of ssthresh at the 18th transmission round?
- G. What is the value of ssthresh at the 24th transmission round?

Homework 3B (Cont)

- H. During what transmission round is the 70th segment sent?
- I. Assuming a packet loss is detected after the 26th round by the receipt of a triple duplicate ACK, what will be the values of the congestion window size and of ssthresh?
- J. Suppose TCP Tahoe is used (instead of TCP Reno), and assume that triple duplicate ACKs are received at the 16th round. What are the ssthresh and the congestion window size at the 19th round?
- K. Again suppose TCP Tahoe is used, and there is a timeout event at the end of 22nd round. How many packets have been sent out from 17th round till 22nd round, inclusive?

Summary



1. Multiplexing/demultiplexing by a combination of source and destination IP addresses and port numbers.
2. Longer distance or higher speed
   ⇒ Larger α ⇒ Larger window is better
3. Window flow control is better for long-distance or high-speed networks
4. UDP is connectionless and simple.
   No flow/error control. Has error detection.
5. TCP provides full-duplex connections with flow/error/congestion control.