## *Example 1*

Imagine a TCP connection is transferring a file of 6000 bytes. The first byte is numbered 10010. What are the sequence numbers for each segment if data are sent in five segments with the first four segments carrying 1000 bytes and the last segment carrying 2000 bytes?

## *Solution*

The following shows the sequence number for each segment:

Segment 1 ==> sequence number: 10,010 (range: 10,010 to 11,009)

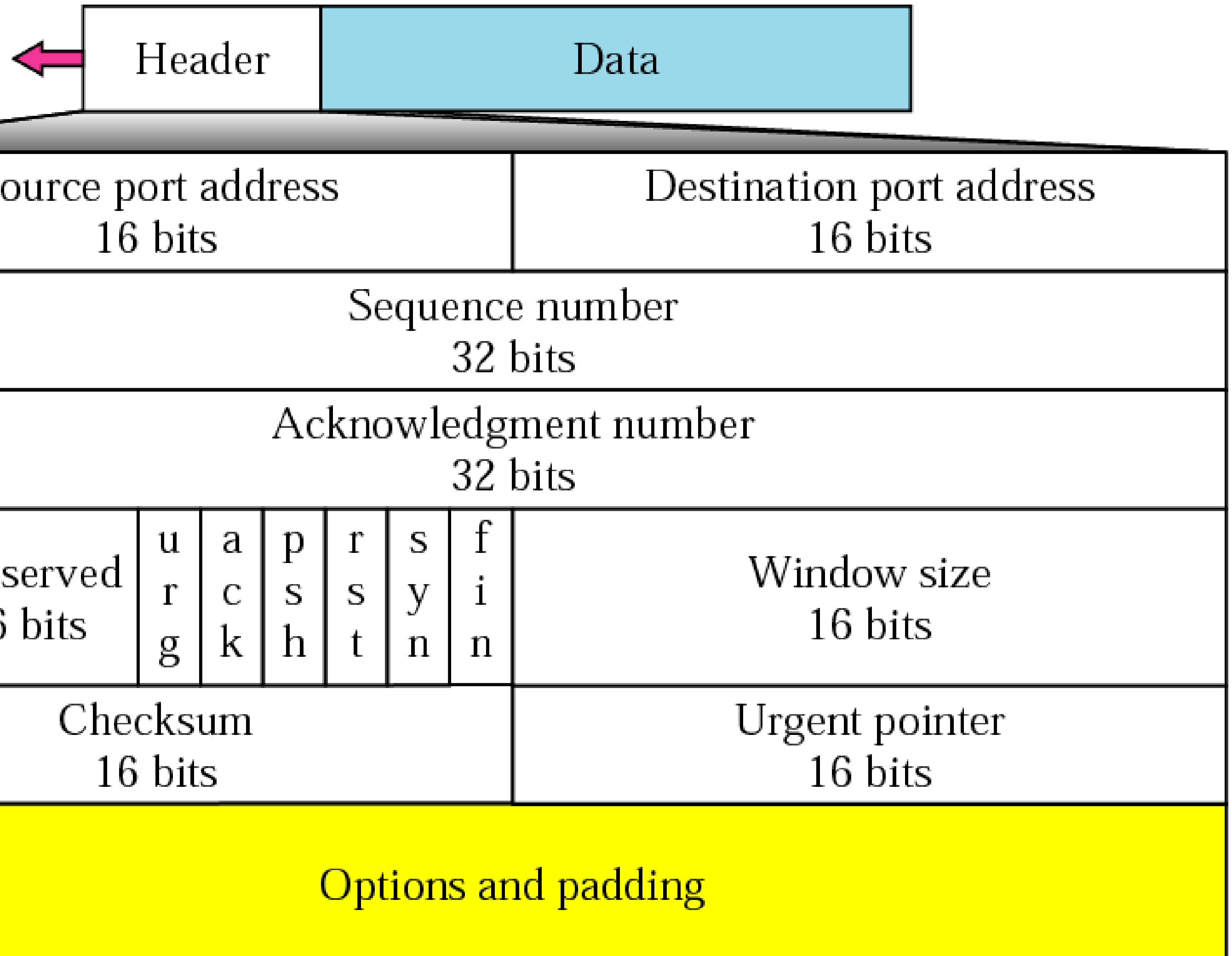
Segment 2 ==> sequence number: 11,010 (range: 11,010 to 12,009)

Segment 3 ==> sequence number: 12,010 (range: 12,010 to 13,009)

Segment 4 ==> sequence number: 13,010 (range: 13,010 to 14,009)

Segment 5 ==> sequence number: 14,010 (range: 14,010 to 16,009)

# TCP segment format



The segment consists of a 20- to 60-byte header, followed by data from the application program.

The header is **20 bytes** if there are no options and up to **60 bytes** if it contains options.

**Reserved:**
This is a 6-bit field reserved for future use.

**Control:**
This field defines 6 different control bits or flags.

One or more of these bits can be set at a time.

These bits enable **flow control, connection establishment and termination, connection abortion**, and the mode of data transfer in TCP.

# Control field

URG: Urgent pointer is valid
ACK: Acknowledgment is valid
PSH: Request for push
RST: Reset the connection
SYN: Synchronize sequence numbers
FIN: Terminate the connection

| | | | | | |
|---|---|---|---|---|---|
| URG | ACK | PSH | RST | SYN | FIN |

## *Description of flags in the control field*

| Flag | Description |
|---|---|
| **URG** | **The value of the urgent pointer field is valid.** |
| **ACK** | **The value of the acknowledgment field is valid.** |
| **PSH** | **Push the data.** |
| **RST** | **The connection must be reset.** |
| **SYN** | **Synchronize sequence numbers during connection.** |
| **FIN** | **Terminate the connection.** |

**Window size:**

This field defines the size of the window, in bytes, that the other party must maintain.

The length of this field is 16 bits, which means that the maximum size of the window is **65,535** bytes.

This value is normally referred to as the receiving window (rwnd) and is determined by the receiver.

The sender must obey the dictation of the receiver in this case.

## Checksum:

This 16-bit field contains the checksum. The calculation of the checksum for TCP follows the same procedure as the one described for UDP.

However, the inclusion of the checksum in the UDP datagram is optional, whereas the inclusion of the checksum for TCP is mandatory.

The same pseudoheader, serving the same purpose, is added to the segment. For the TCP pseudoheader, the value for the protocol field is 6.

**Urgent pointer:**

This l6-bit field, which is valid only if the urgent flag is set, is used when the segment contains urgent data.

**Options:**

There can be up to 40 bytes of optional information in the TCP header.

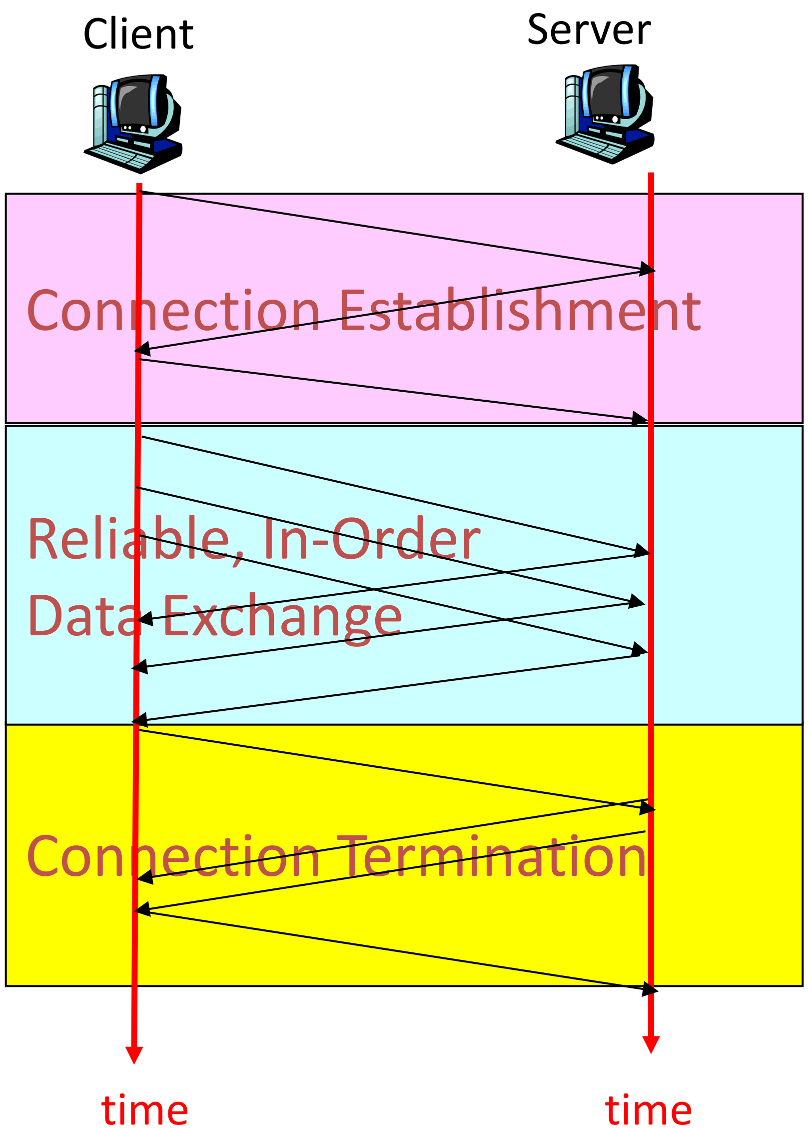
# A TCP Connection

The point is that a TCP connection is **virtual, not physical.** TCP operates at a higher level. TCP uses the services of IP to deliver individual segments to the receiver, but it controls the connection itself.

If a segment is lost or corrupted, it is retransmitted. Unlike TCP, IP is unaware of this retransmission.

If a segment arrives out of order, TCP holds it until the missing segments arrive; IP is unaware of this reordering.

# Typical TCP Transaction




- A TCP Transaction consists of 3 Phases
  1. Connection Establishment
     - Handshaking between client and server
  2. Reliable, In-Order Data Exchange
     - Recover any lost data through retransmissions and ACKs
  3. Connection Termination
     - Closing the connection

In TCP, connection-oriented transmission requires three phases:

**1] connection establishment.**

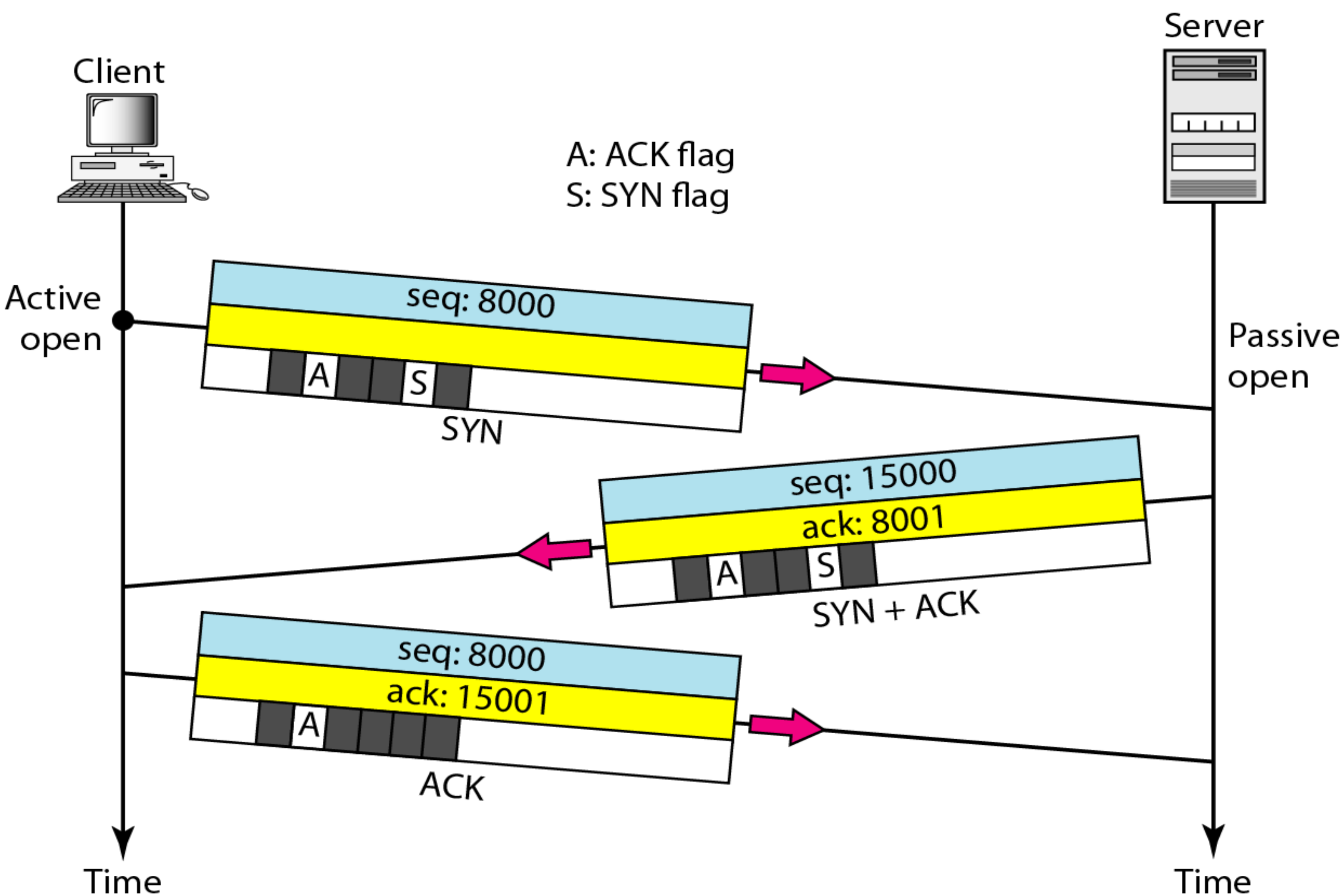
**2] data transfer and**

**3] connection termination.**

# 1] connection establishment

- Three-Way Handshaking
- Simultaneous Open

# Connection establishment using three-way handshaking



The server program tells its TCP that it is ready to accept a connection. This is called a request for a ***passive open***.
Although the server TCP is ready to accept any connection from any machine in the world, it cannot make the connection itself.

The client program issues a request for an ***active open***. A client that wishes to connect to an open server tells its TCP that it needs to be connected to that particular server.

**1]** The client sends the first segment, **a SYN segment**, in which only the SYN flag is set. This segment is for synchronization of sequence numbers. It consumes **one sequence number**.

When the data transfer starts, the sequence number is incremented by **1**.

We can say that the SYN segment carries no real data, but we can think of it as containing 1 imaginary byte.

**2]** The server sends the second segment, a **SYN +ACK segment**, with 2 flag bits set: **SYN and ACK**.

This segment has a dual purpose. It is a SYN segment for communication in the other direction and serves as the acknowledgment for the SYN segment. It consumes one sequence number.

**3]** The client sends the third segment. This is just an **ACK segment**. It acknowledges the receipt of the second segment with the ACK flag and acknowledgment number field.

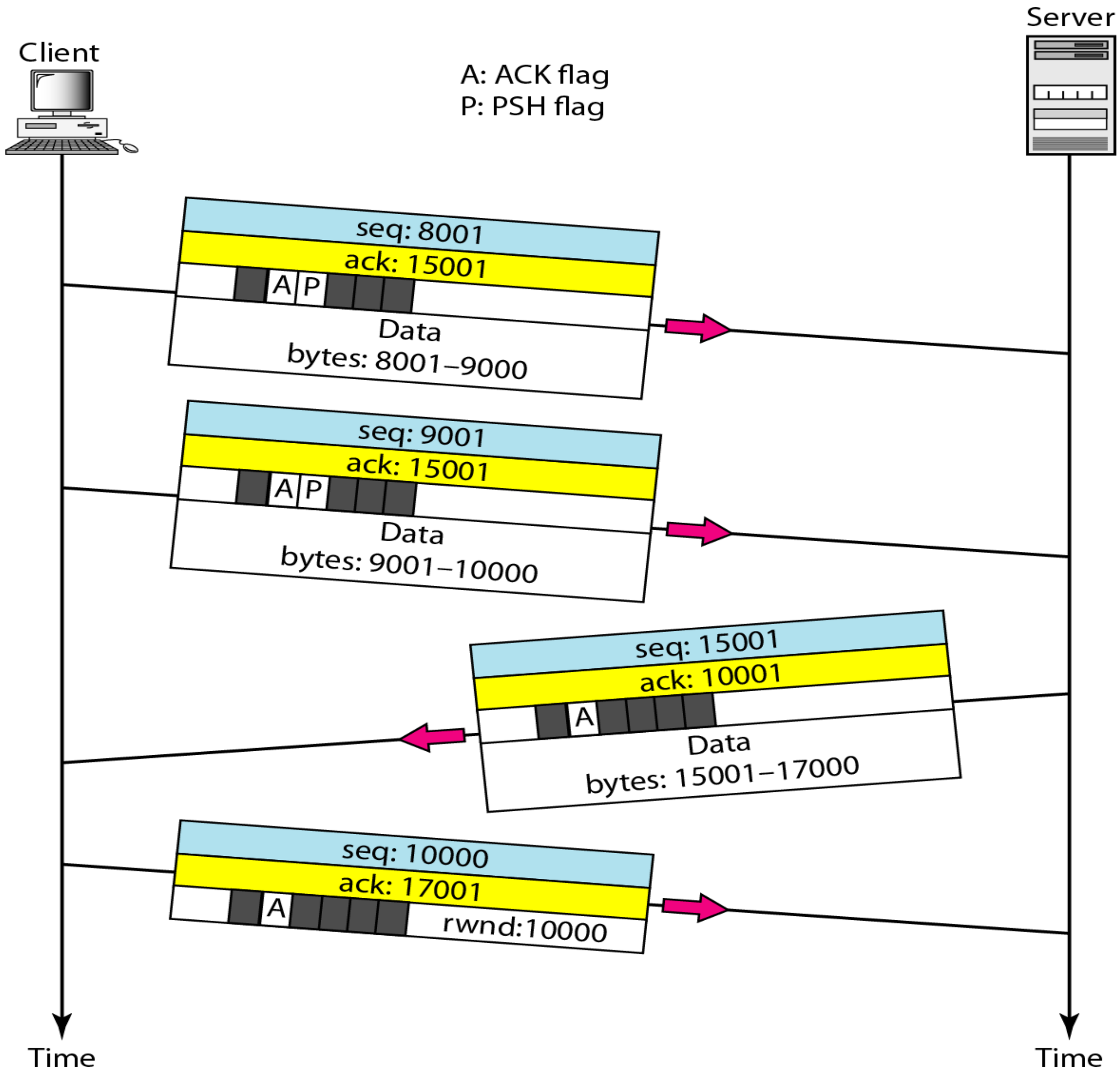
Note that the sequence number in this segment is the same as the one in the SYN segment; the ACK segment does not consume any sequence numbers.

# Connection establishment using Simultaneous Open

A rare situation, called a simultaneous open, may occur when both processes issue an **active open**.

In this case, both TCPs transmit a **SYN + ACK** segment to each other, and one single connection is established between them.

# 2] Data transfer



# Data transfer

After connection is established, bidirectional data transfer can take place. The client and server can both send data and acknowledgments.

It is enough to know that data traveling in the same direction as an acknowledgment are carried on the same segment. The acknowledgment is **piggybacked** with the data.

The data segments sent by the client have the **PSH (push)** flag set. so that the server TCP knows to deliver data to the server process as soon as they are received.

The segment from the server, on the other hand, **does not set the push flag**.

Most TCP implementations have the option to set or not set this flag.

# 3] Connection termination

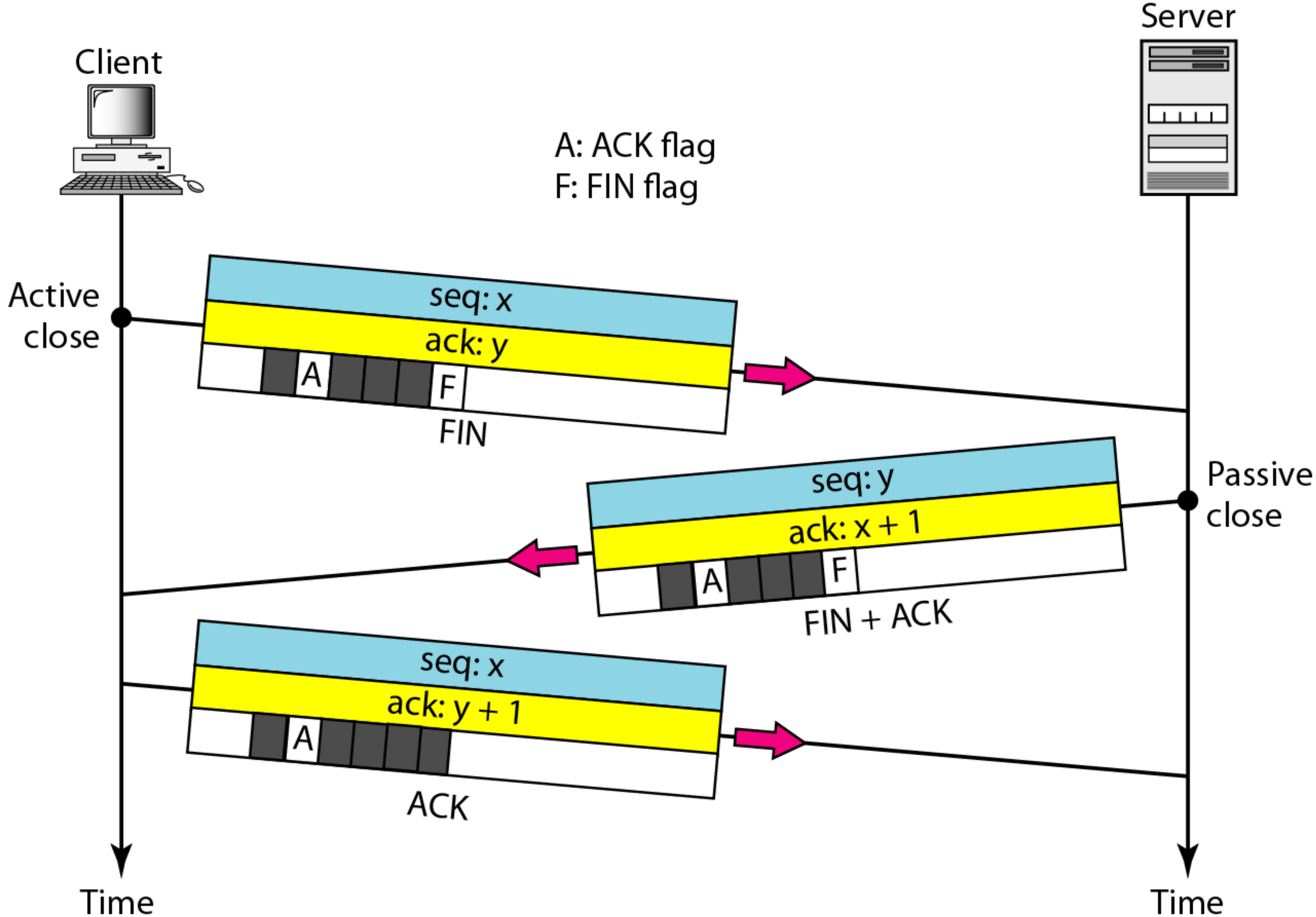
Any of the two parties involved in exchanging data (client or server) can close the connection, although it is usually initiated by **the client**.

Two options for connection termination:

**A] Three-way handshaking and**

**B] Four-way handshaking with a half-close option.**

# Connection termination using three-way handshaking



**1]** In a normal situation, the client TCP, after receiving a close command from the client process, sends the first segment, a **FIN segment** in which the FIN flag is set.

Note that a FIN segment can include the last chunk of data sent by the client, or it can be just a control segment .

If it is only a control segment, it consumes only one sequence number.

**2]** The server TCP, after receiving the FIN segment, informs its process of the situation and sends the second segment, **a FIN +ACK segment**, to confirm the receipt of the FIN segment from the client and at the same time to announce the closing of the connection in the other direction.

This segment can also contain the last chunk of data from the server. If it does not carry data, it consumes only one sequence number.

**3]** The client TCP sends the last segment, an ACK segment, to confirm the receipt of the FIN segment from the TCP server.

This segment contains the acknowledgment number, which is 1 plus the sequence number received in the FIN segment from the server.

This segment cannot carry data and consumes no sequence numbers.

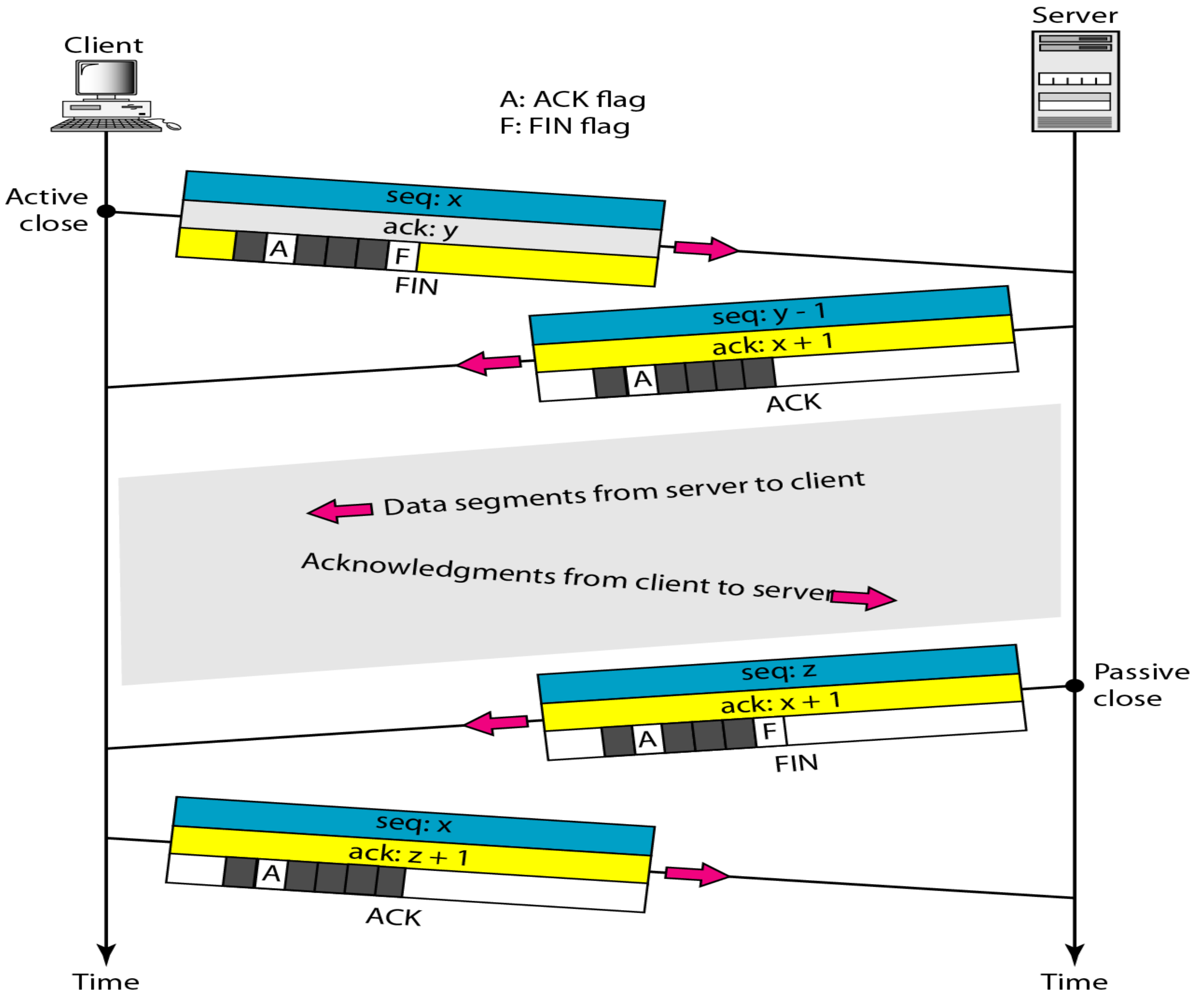
# B] Four-way handshaking with a half-close option.

Half-Close In TCP, one end can stop sending data while still receiving data. This is called **a half-close.**

Although either end can issue a half-close, it is normally initiated by the client.

It can occur when the server needs all the data before processing can begin.

# Half-close


Client
Server
A: ACK flag
F: FIN flag
Active close
seq: x
ack: y
A
F
FIN
seq: y - 1
ack: x + 1
A
ACK
Data segments from server to client
Acknowledgments from client to server
Passive close
seq: z
ack: x + 1
A
F
FIN
seq: x
ack: z + 1
A
ACK
Time
Time

A good example is **sorting**.

When the client sends data to the server to be sorted, the server needs to receive all the data before sorting can start. This means the client, after sending all the data, can close the connection in the outbound direction.

However, the inbound direction must remain open to receive the sorted data. The server, after receiving the data, still needs time for sorting; its outbound direction must remain open.

The client half-closes the connection by sending a FIN segment. The server accepts the half-close by sending the ACK segment. The data transfer from the client to the server stops. The server, however, can still send data. When the server has sent all the processed data, it sends a FIN segment, which is acknowledged by an ACK from the client.

After half-closing of the connection, data can travel from the server to the client and acknowledgments can travel from the client to the server.

The client cannot send any more data to the server. Note the sequence numbers we have used. The second segment (ACK) consumes no sequence number.

Although the client has received sequence number *y - 1 and is expecting y, the server sequence number is still y - 1. When the connection* finally closes, the sequence number of the last ACK segment is still *x, because no* sequence numbers are consumed during data transfer in that direction.

# TCP Flow Control

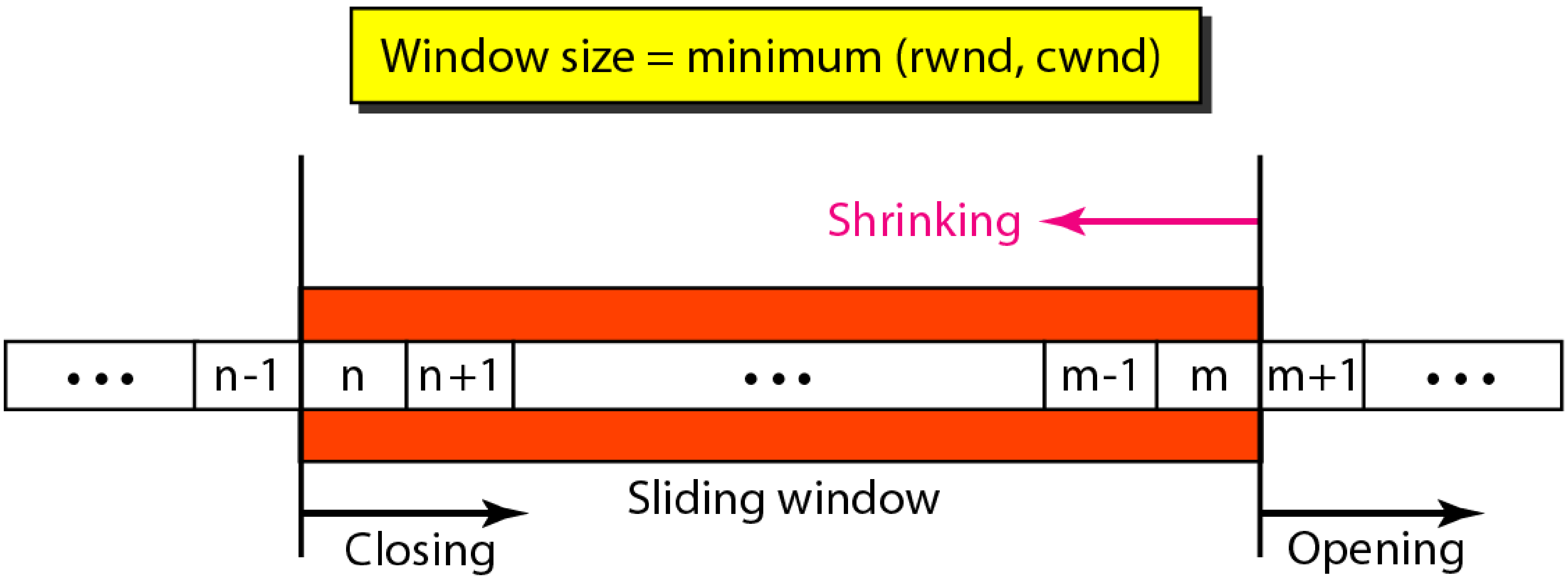
TCP uses a sliding window, to handle flow control.

The sliding window protocol used by TCP, however, is something between the ***Go-Back-N*** **and Selective Repeat sliding window**.

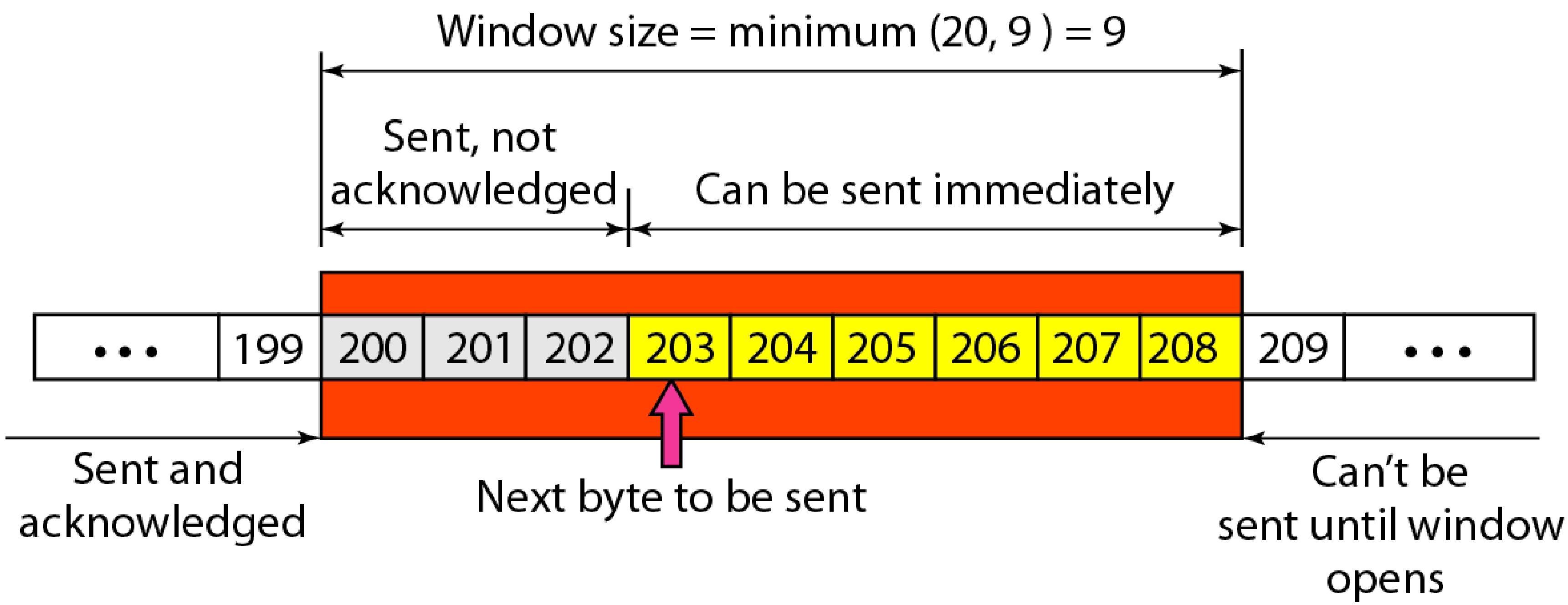
The sliding window protocol in TCP looks like the Go-Back-N protocol because it does not use NAKs;

it looks like Selective Repeat because the receiver holds the out-of-order segments until the missing ones arrive.

# Sliding window



# Example


Window size = minimum (20, 9 ) = 9
Sent, not acknowledged
Can be sent immediately
• • •
199
200
201
202
203
204
205
206
207
208
209
• • •
Sent and acknowledged
Next byte to be sent
Can't be sent until window opens

**Opening a window** means moving the right wall to the right. This allows more new bytes in the buffer that are eligible for sending.

**Closing the window** means moving the left wall to the right. This means that some bytes have been acknowledged and the sender need not worry about them anymore.

**Shrinking the window** means moving the right wall to the left. This is strongly discouraged and not allowed in some implementations.
because it means revoking the eligibility of some bytes for sending.

This is a problem if the sender has already sent these bytes. Note that the left wall cannot move to the left because this would revoke some of the previously sent acknowledgments.

The size of the window at one end is determined by the lesser of two values: ***receiver window (rwnd) or congestion window (cwnd).***

The receiver window is the value advertised by the opposite end in a segment containing acknowledgment. It is the number of bytes the other end can accept before its buffer overflows and data are discarded.

The congestion window is a value determined by the network to avoid congestion.

# RTP

Real-time Transport Protocol (RTP) is the protocol designed to handle real-time traffic on the Internet.

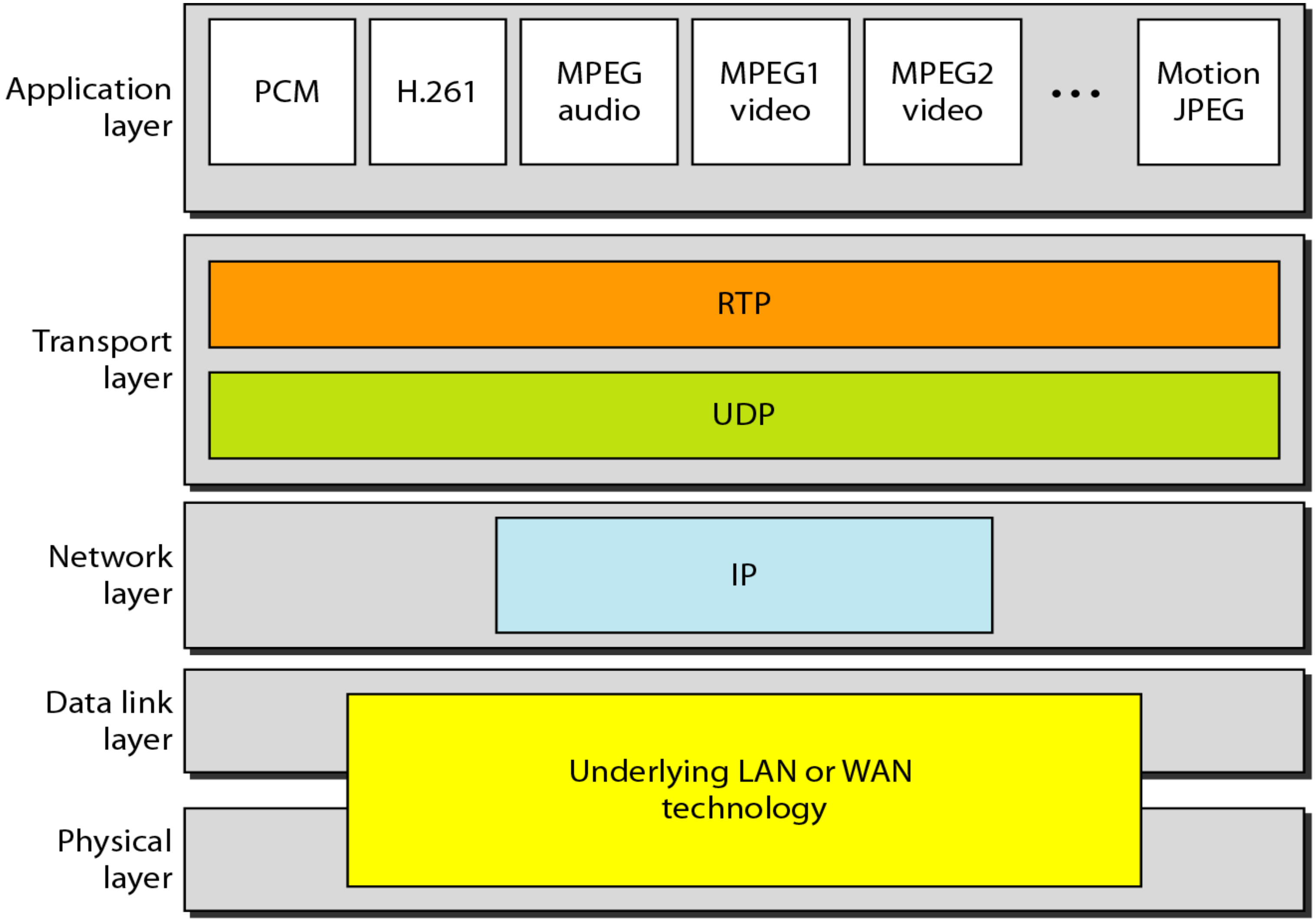
RTP does not have a delivery mechanism; it must be used with UDP.

RTP stands between UDP and the application program. The main contributions of RTP are **time-stamping, sequencing, and mixing facilities**.

The Real-time Transport Protocol (RTP) is a network protocol for delivering audio and video over IP networks.

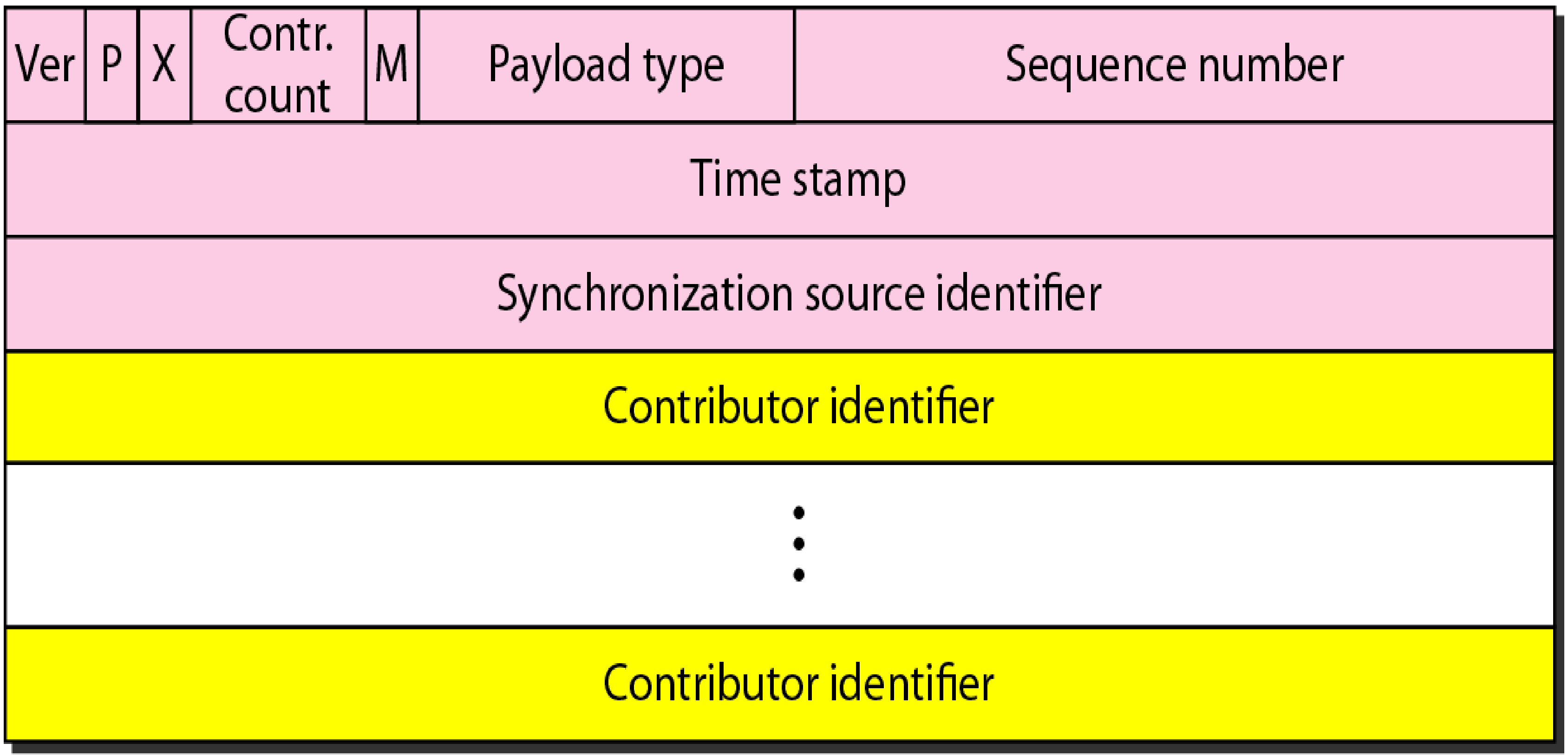
RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications, television services .

# RTP



Application layer
PCM
H.261
MPEG audio
MPEG1 video
MPEG2 video
• • •
Motion JPEG
Transport layer
RTP
UDP
Network layer
IP
Data link layer
Physical layer
Underlying LAN or WAN technology

# RTP packet header format



**2] P:-**

This 1-bit field, if set to 1, indicates the presence of padding at the end of the packet.

In this case, the value of the last byte in the padding defines the length of the padding.

There is no padding if the value of the P field is O.

**3] X: -** This 1-bit field, if set to 1, indicates an extra extension header between the basic header and the data.

There is no extra extension header if the value of this field is O.

**4] Contributor count :-** This 4-bit field indicates the number of contributors. Note that
we can have a maximum of 16 contributors because a 4-bit field only allows a number between 0 and 15.

**5] M: -** This 1-bit field is a marker used by the application to indicate, for example, the end of its data.

**6] Payload type :-** This 7-bit field indicates the type of the payload.

# Payload types

| *Type* | *Application* | *Type* | *Application* | *Type* | *Application* |
|---|---|---|---|---|---|
| 0 | PCMμ Audio | 7 | LPC audio | 15 | G728 audio |
| 1 | 1016 | 8 | PCMA audio | 26 | Motion JPEG |
| 2 | G721 audio | 9 | G722 audio | 31 | H.261 |
| 3 | GSM audio | 10–11 | L16 audio | 32 | MPEG1 video |
| 5–6 | DV14 audio | 14 | MPEG audio | 33 | MPEG2 video |

**7] Sequence number :-** This field is 16 bits in length. It is used to number the RTP packets. The sequence number of the first packet is chosen randomly; it is incremented by 1 for each subsequent packet. The sequence number is used by the receiver to detect lost or out-of-order packets.

**8] Timestamp :-** This is a 32-bit field that indicates the time relationship between packets. The timestamp for the first packet is a random number.

**9] Synchronization source identifier:-**

If there is only one source, this 32-bit field defines the source. However, if there are several sources, the mixer is the synchronization source and the other sources are contributors.

The value of the source identifier is a random number chosen by the source. The protocol provides a strategy in case of conflict (two sources start with the same sequence number).

**10] Contributor identifier:** -

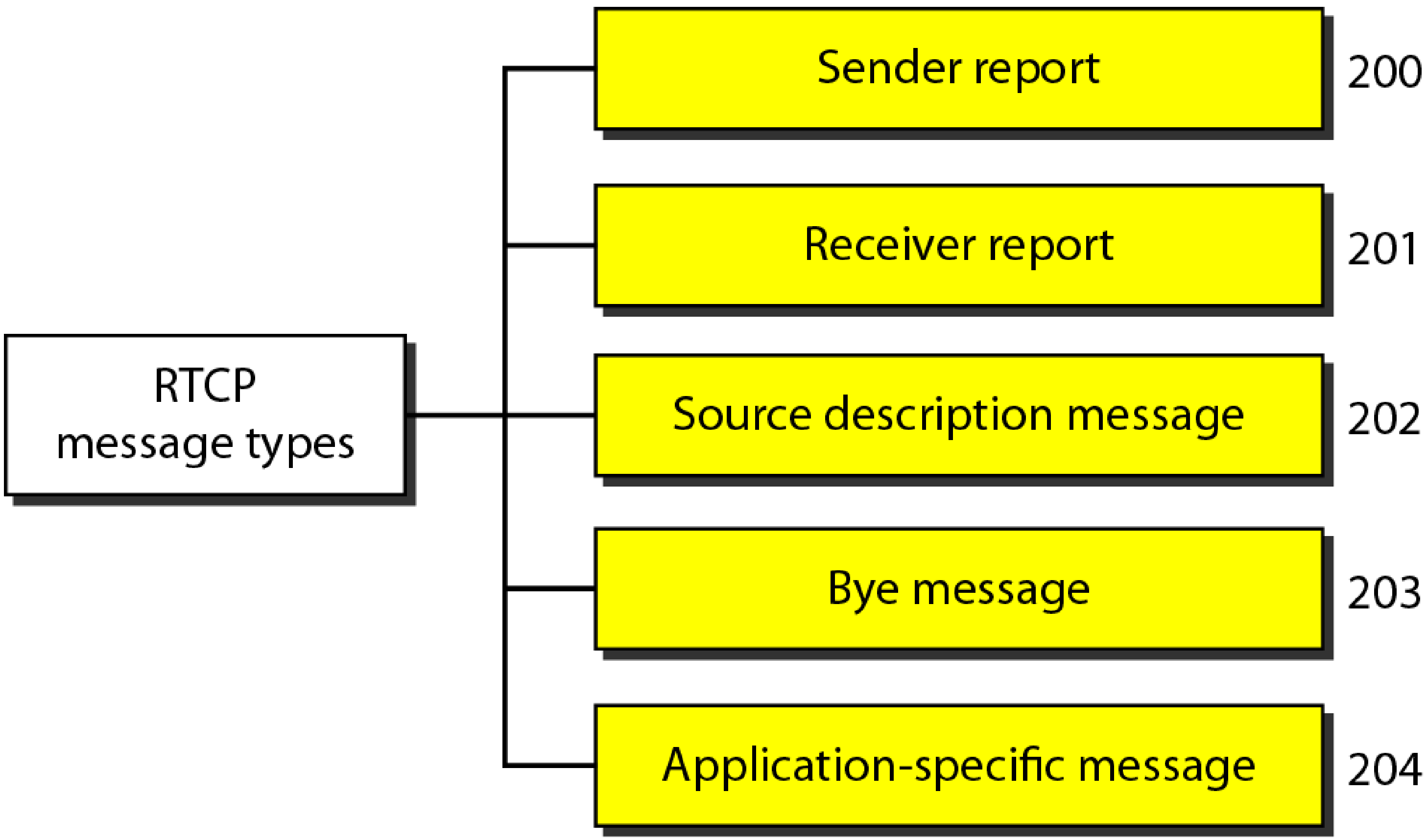
Each of these 32-bit identifiers defines a source. When there is more than one source in a session, the mixer is the synchronization source and the remaining sources are the contributors.

# RTCP

RTP allows only one type of message, one that carries data from the source to the destination.

In many cases, there is a need for other messages in a session. These messages control the flow and quality of data and allow the recipient to send feedback to the source or sources. **Real-time Transport Control Protocol (RTCP)** is a protocol designed for this purpose.

# RTCP message types



# SCTP

Stream Control Transmission Protocol (SCTP) is a new reliable, message-oriented transport layer protocol.

SCTP, however, is mostly designed for Internet applications that have recently been introduced. **For eg: IUA(ISDN over IP).**

These new applications need a more sophisticated service than TCP can provide.

SCTP is a protocol for transmitting multiple streams of data at the same time between two end points that have established a connection in a network.

SCTP is a message-oriented, reliable protocol that combines the best features of UDP and TCP.

It can detects lost data, duplicate data, and out-of-order data.

It also has congestion control and flow control mechanisms.

Sometimes referred to as "**next generation TCP**"

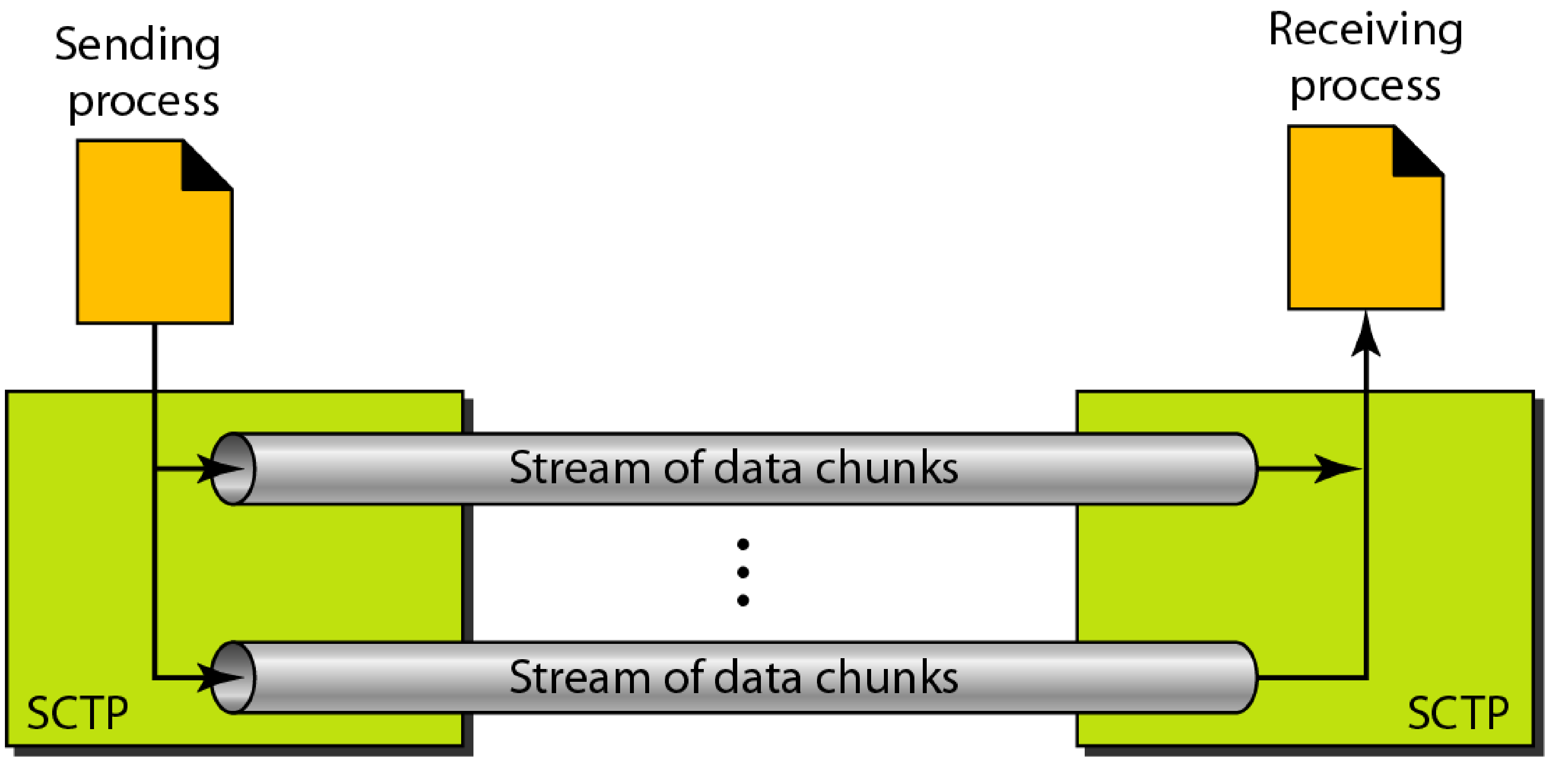
# SCTP Services

1] Process-to-Process Communication

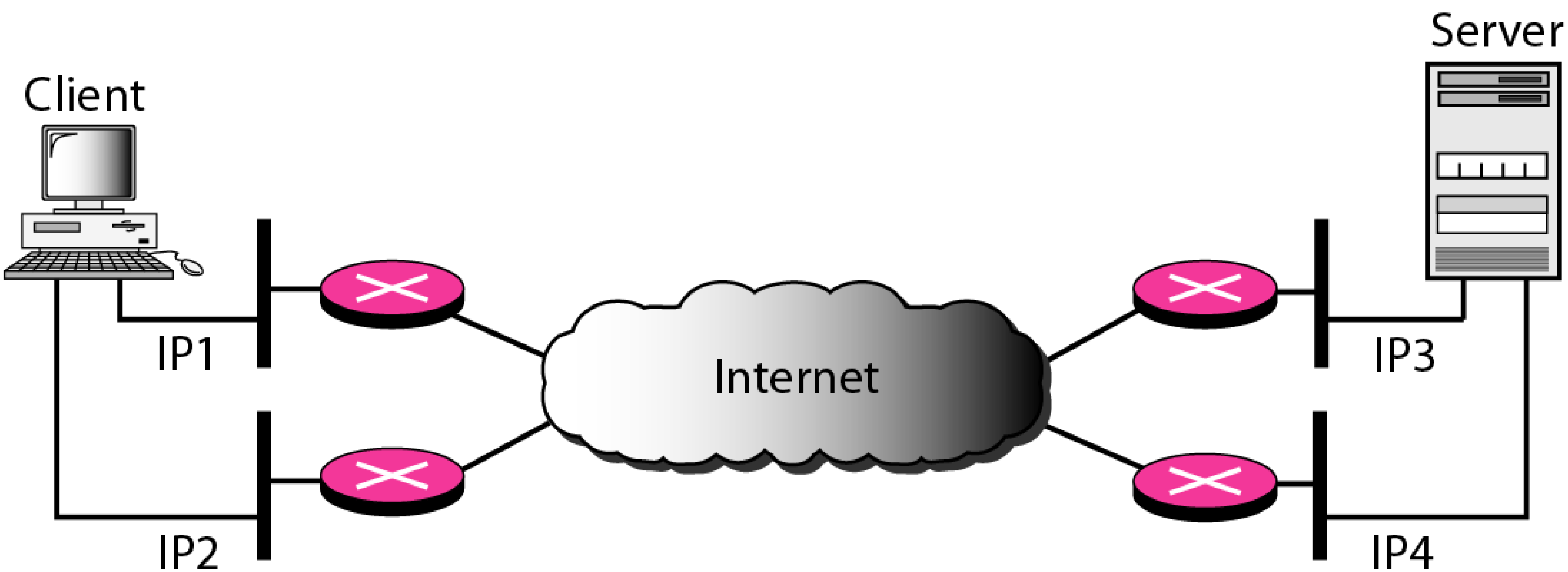
# Some SCTP applications

| *Protocol* | *Port Number* | *Description* |
|---|---|---|
| IUA | 9990 | ISDN over IP |
| M2UA | 2904 | SS7 telephony signaling |
| M3UA | 2905 | SS7 telephony signaling |
| H.248 | 2945 | Media gateway control |
| H.323 | 1718, 1719, 1720, 11720 | IP telephony |
| SIP | 5060 | IP telephony |

# 2] Multiple Streams



# 3] Multihoming concept



A TCP connection involves one source and one destination IP address. This means that even if the sender or receiver is a multihomed host (connected to more than one physical address with multiple IP addresses), only one of these IP addresses per end can be utilized during the connection.

In this fault-tolerant approach, when one path fails, another interface can be used for data delivery without interruption. This fault-tolerant feature is very helpful when we are sending and receiving a real-time payload such as Internet telephony.

**Internet telephony:** A category of hardware and software that enables people to use the Internet as the transmission medium for telephone calls.

There are many Internet telephony applications available. Some, like **CoolTalk and NetMeeting.**

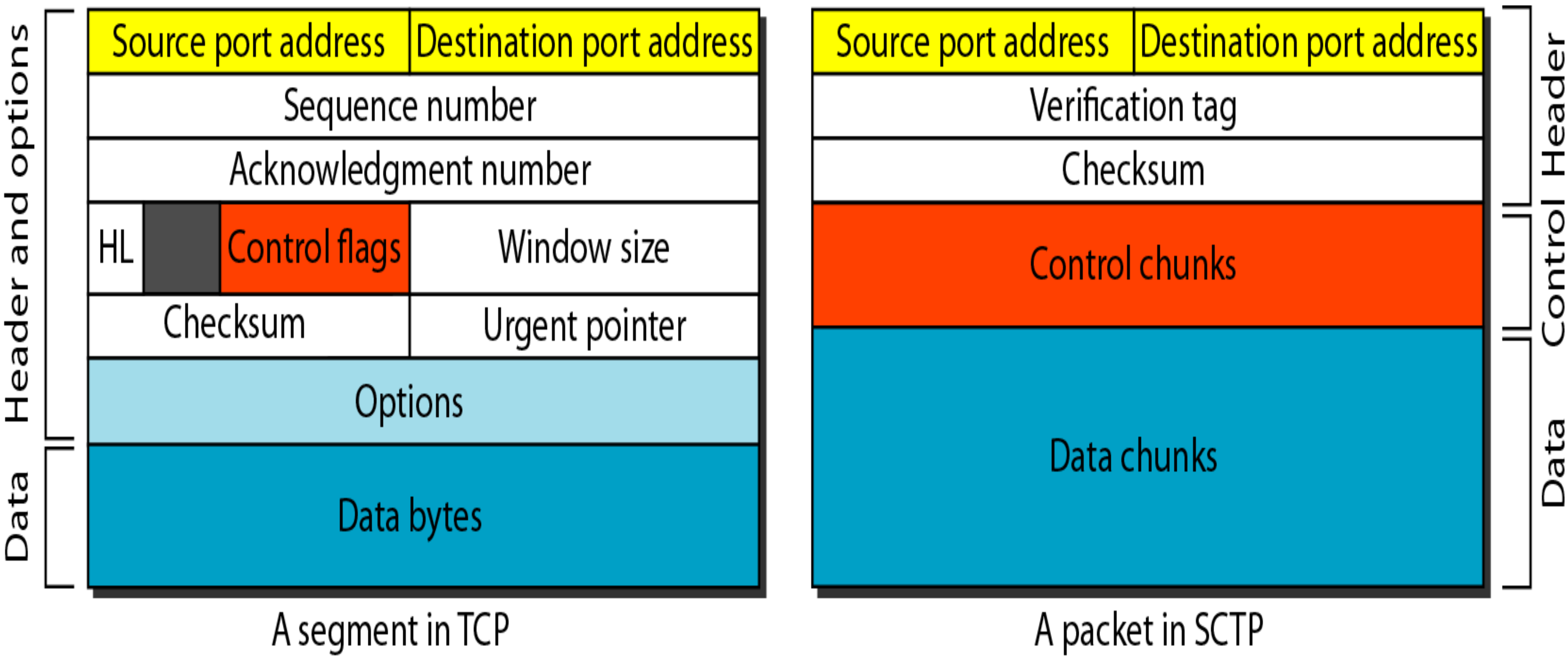
Others are stand-alone products. Internet telephony products are sometimes called **IP telephony, Voice over the Internet (VOI) or Voice over IP (VOIP)products.**

4] Full-Duplex Communication

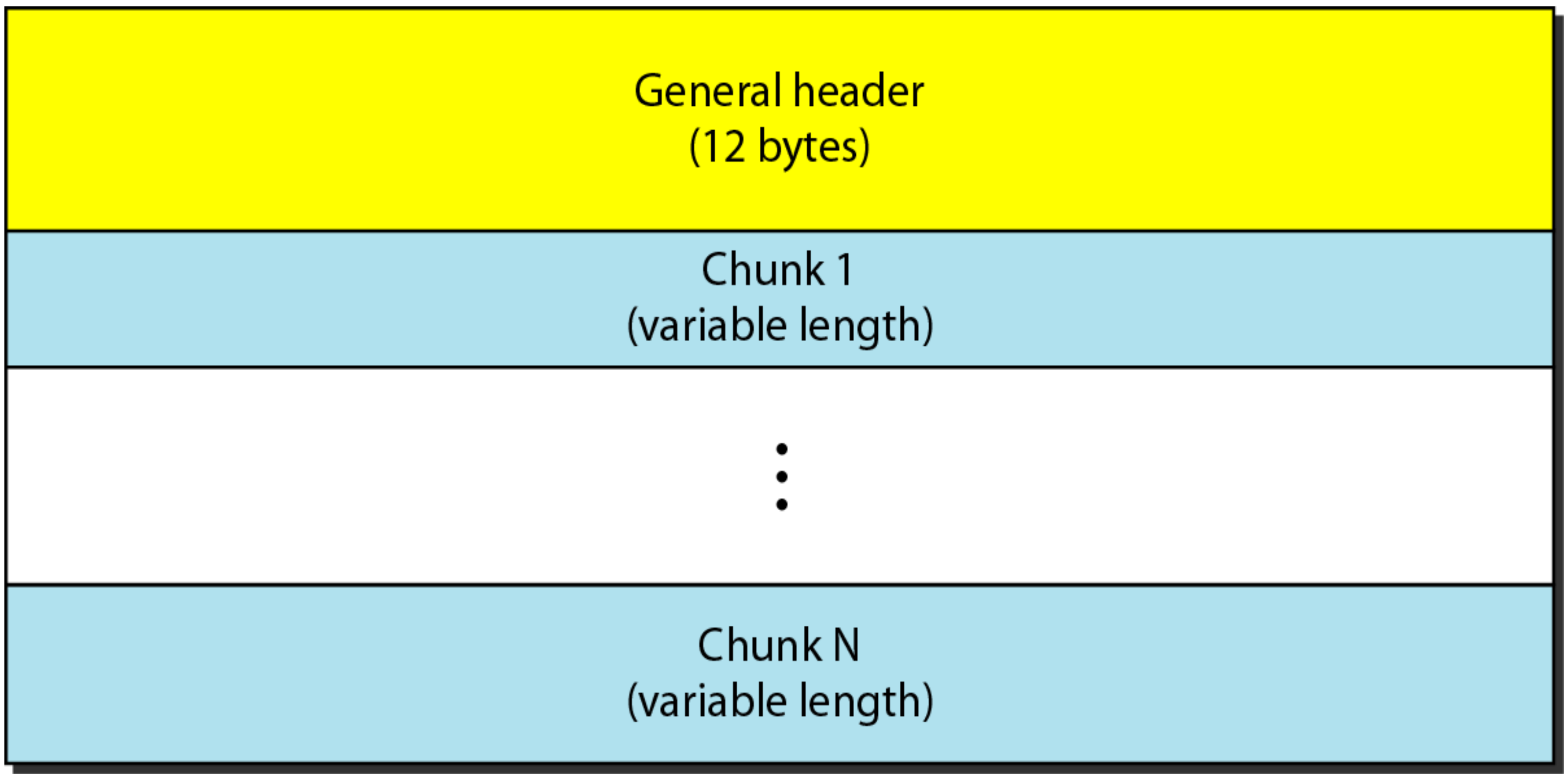
5] Connection-Oriented Service

6] Reliable Service.

# Comparison between a TCP segment and an SCTP packet



# SCTP packet format



# General header

| | |
|---|---|
| Source port address<br>16 bits | Destination port address<br>16 bits |
| Verification tag<br>32 bits | |
| Checksum<br>32 bits | |

An SCTP packet has a mandatory general header and a set of blocks called chunks.

There are two types of chunks: **control chunks and data chunks.**

A control chunk controls and maintains the association;

A data chunk carries user data.

In a packet, the control chunks come before the data chunks.

# Chunks

| *Type* | *Chunk* | *Description* |
|---|---|---|
| **0** | **DATA** | User data |
| **1** | **INIT** | Sets up an association |
| **2** | **INIT ACK** | Acknowledges INIT chunk |
| **3** | **SACK** | Selective acknowledgment |
| **4** | **HEARTBEAT** | Probes the peer for liveliness |
| **5** | **HEARTBEAT ACK** | Acknowledges HEARTBEAT chunk |
| **6** | **ABORT** | Aborts an association |
| **7** | **SHUTDOWN** | Terminates an association |
| **8** | **SHUTDOWN ACK** | Acknowledges SHUTDOWN chunk |
| **9** | **ERROR** | Reports errors without shutting down |
| **10** | **COOKIE ECHO** | Third packet in association establishment |
| **11** | **COOKIE ACK** | Acknowledges COOKIE ECHO chunk |
| **14** | **SHUTDOWN COMPLETE** | Third packet in association termination |
| **192** | **FORWARD TSN** | For adjusting cumulative TSN |

**Verification tag :**
This is a number that matches a packet to an association. This prevents a packet from a previous association from being mistaken as a packet in this association.

It serves as an identifier for the association; it is repeated in every packet during the association. There is a separate verification used for each direction in the association.

**Checksum: -**

This 32-bit field contains a CRC-32 checksum. Note that the size of the checksum is increased from 16 (in UDP, TCP, and IP) to 32 bits to allow the use of the CRC-32 checksum.

# SCTP Features

## 1] Transmission Sequence Number (TSN): -

Data transfer in SCTP is controlled by numbering the data chunks. SCTP uses a transmission sequence number (TSN) to number the data chunks. In other words, the TSN in SCTP plays the analogous role to the sequence number in TCP. TSNs are 32 bits long and randomly initialized between 0 and $2^{32}$ - 1. Each data chunk must carry the corresponding TSN in its header.

## 2] Stream Identifier: -

Each stream in SCTP needs to be identified by using a stream identifier (SI).

To distinguish between different streams, SCTP uses an SI.

The SI is a 16-bit number starting from O.

## 3] Stream Sequence Number :-

To distinguish between different data chunks Belonging to the same stream, SCTP uses **SSNs**.

When a data chunk arrives at the destination SCTP, it is delivered to the appropriate stream and in the proper order.

SCTP defines each data chunk in each stream with a stream sequence number (SSN).

## 4] Packets :-

In TCP, a segment carries data and control information. Data are carried as a collection of bytes; control information is defined by six control flags in the header.

The design of SCTP is totally different: **data are carried as data chunks, control information is carried as control chunks**.

Several control chunks and data chunks can be packed together in a **packet**. A packet in SCTP plays the same role as a segment in TCP

## 5] Flow Control

## 6] Error Control

## 7] Congestion Control

# Congestion Control in TCP

# Congestion Window

The sender's window size is determined not only by the receiver but also by congestion in the network.

The sender has two pieces of information: the receiver-advertised window size and the congestion window size.

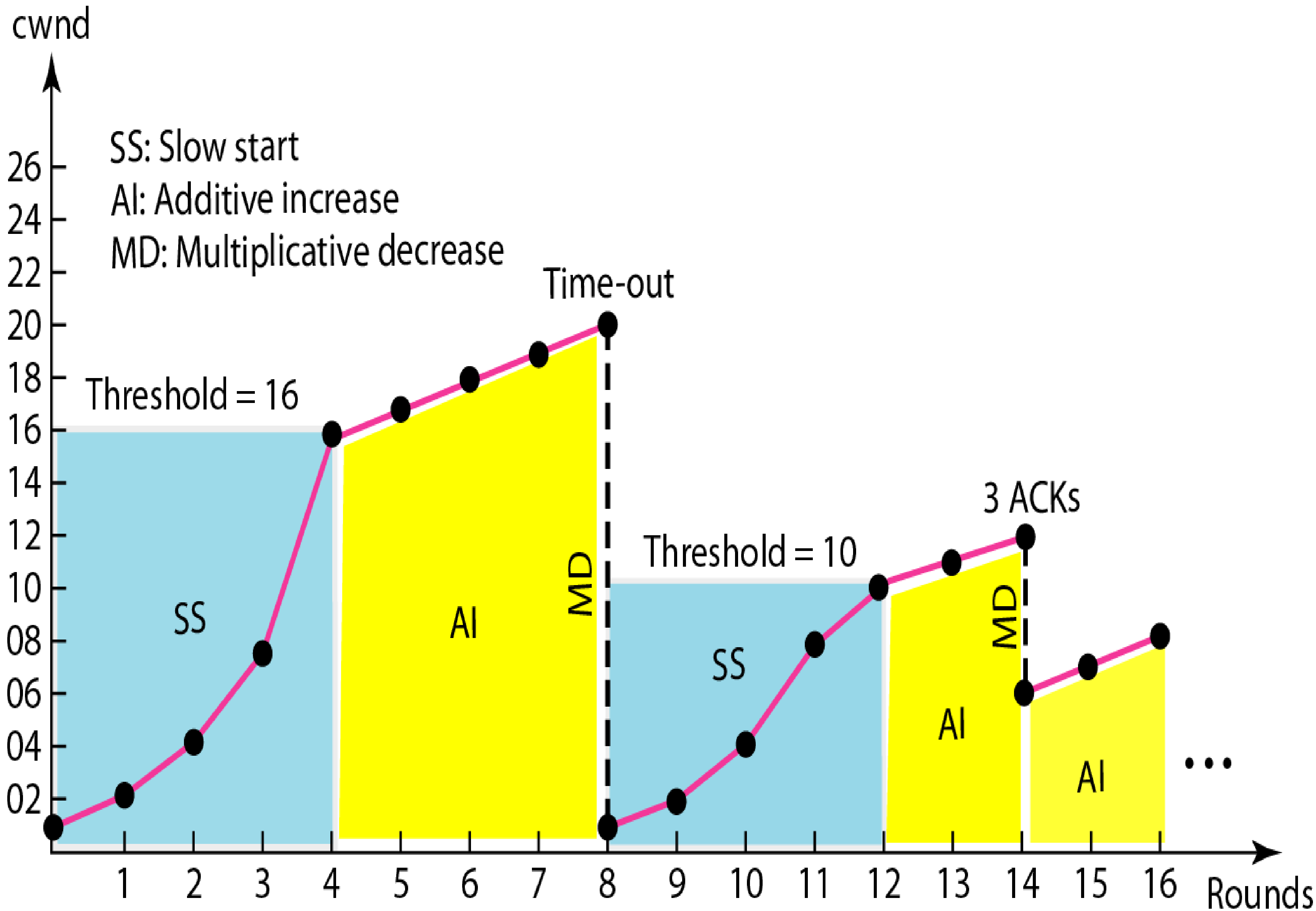
The actual size of the window is the minimum of these two.

**Actual window size = minimum (rwnd, cwnd)**

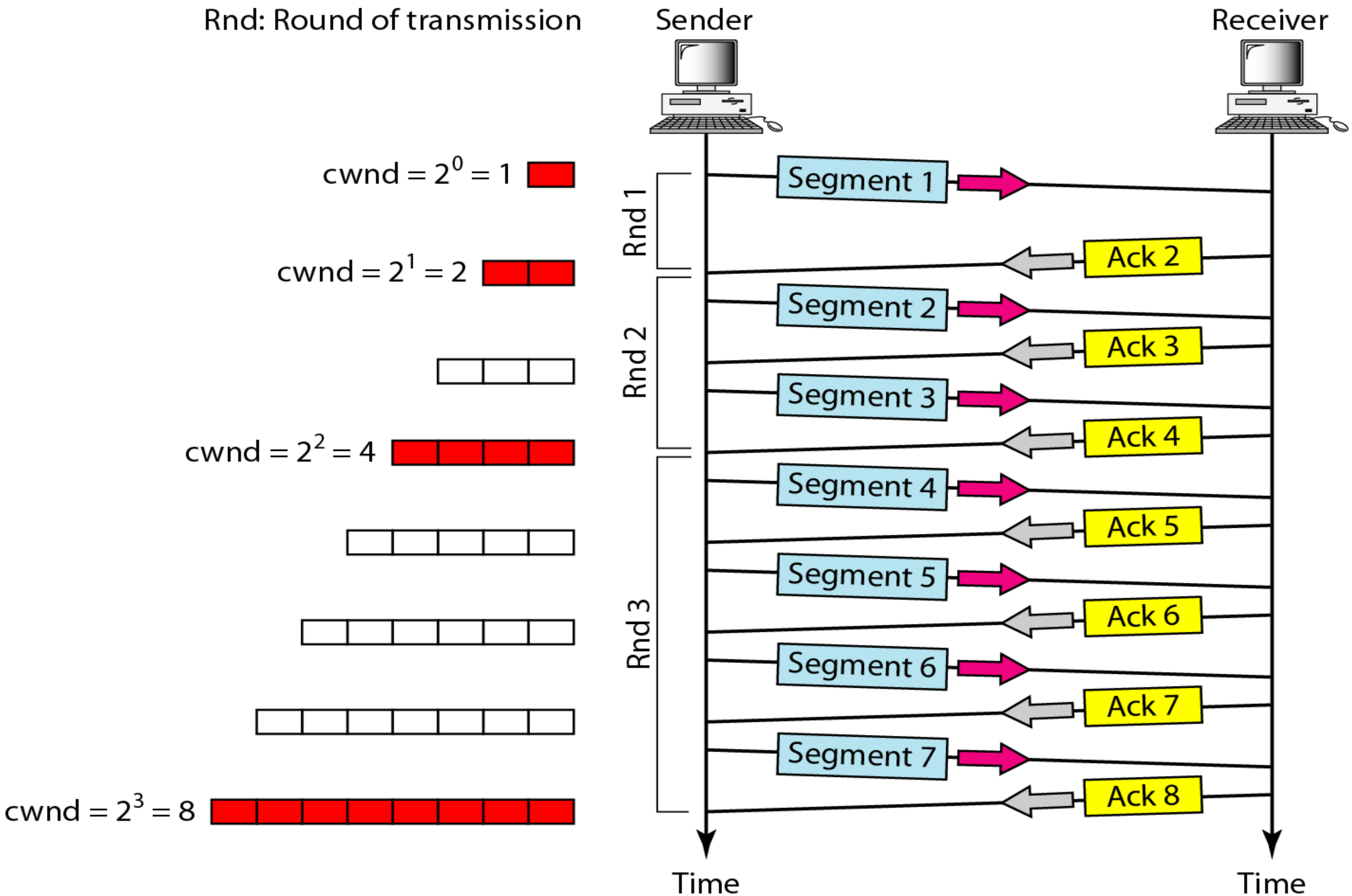
**Congestion Policy :-**

TCP's general policy for handling congestion is based on three phases: **slow start, congestion avoidance, and congestion detection**.

In the slow-start phase, the sender starts with a very slow rate of transmission, but increases the rate rapidly to reach a threshold. When the threshold is reached, the data rate is reduced to avoid congestion. Finally if congestion is detected, the sender goes back to the slow-start or congestion avoidance phase based on how the congestion is detected.


cwnd
SS: Slow start
AI: Additive increase
MD: Multiplicative decrease
Time-out
Threshold = 16
Threshold = 10
3 ACKs
SS
AI
MD
SS
AI
MD
AI
26
24
22
20
18
16
14
12
10
08
06
04
02
1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
Rounds

# Slow start, exponential increase

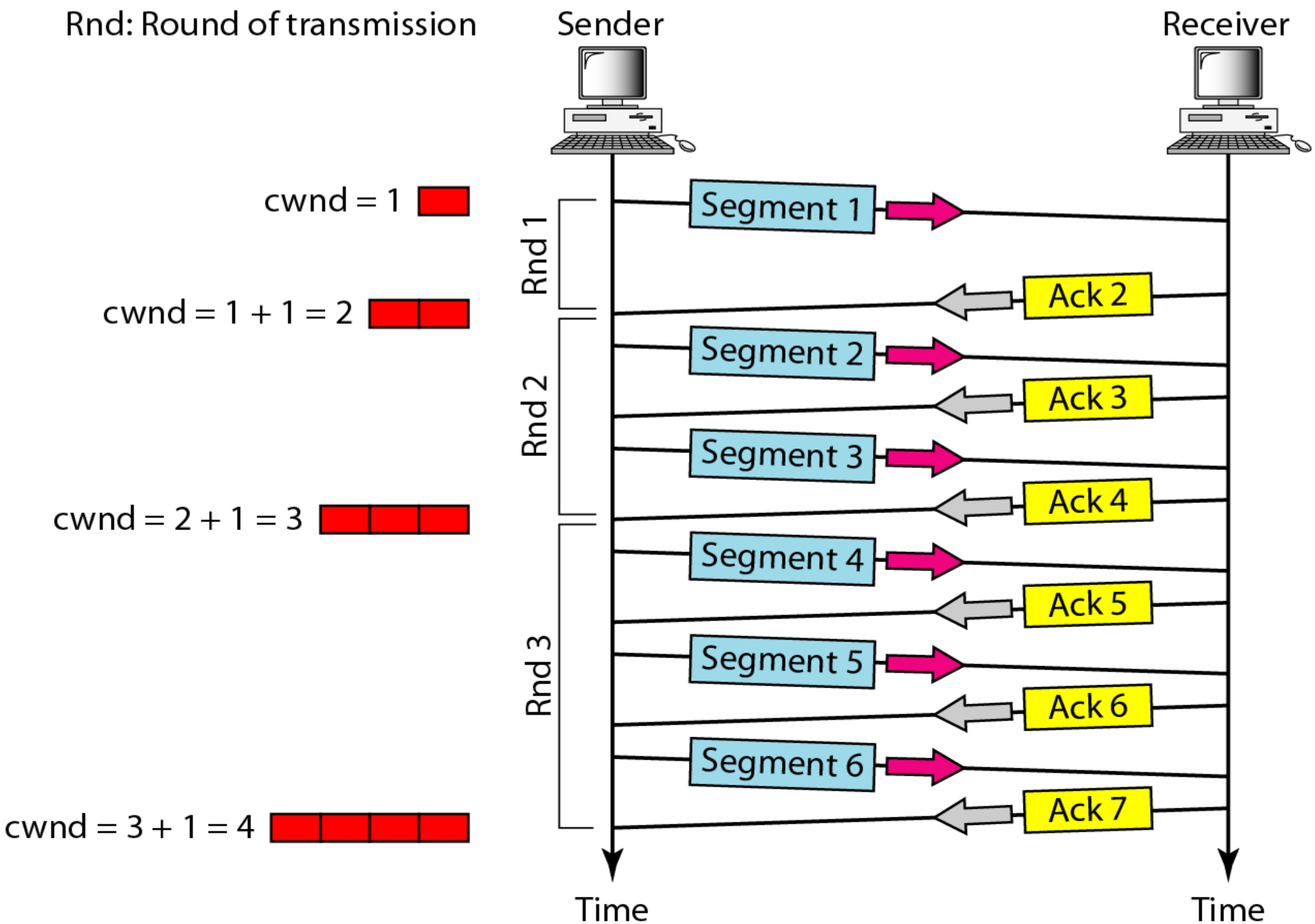


In the slow-start algorithm, the size of the congestion window increases exponentially until it reaches a threshold.

If we look at the size of cwnd in terms of rounds (acknowledgment of the whole window of segments).

Start → cwnd= 1.
After round 1 → cwnd=$2^1$ = 2.
After round 2 → cwnd=$2^2$ = 4.
After round 3 → cwnd=$2^3$ =8.

# Congestion avoidance, additive increase



In the congestion avoidance algorithm, the size of the congestion window increases additively until congestion is detected.

When the size of the congestion window reaches the slow-start threshold, the slow-start phase stops and the additive phase begins. In this algorithm, each time the whole window of segments is acknowledged (one round), the size of the congestion window is increased by 1.

In this case, after the sender has received acknowledgments for a complete window size of segments, the size of the window is increased by one segment.

Start $\rightarrow$ cwnd= 1.
After round 1 $\rightarrow$ cwnd= 1+1 = 2.
After round 2 $\rightarrow$ cwnd= 2+1 = 3.
After round 3 $\rightarrow$ cwnd= 3+1 = 4.

An implementation reacts to **congestion detection** in one of the following ways:

- If detection is by time-out, a new slow start phase starts.
- If detection is by three ACKs, a new congestion avoidance phase starts.

If three ACKs are received, there is a weaker possibility of congestion; a segment may have been dropped, but some segments after that may have arrived safely since three ACKs are received. **This is called fast transmission and fast recovery.**

# Congestion example