

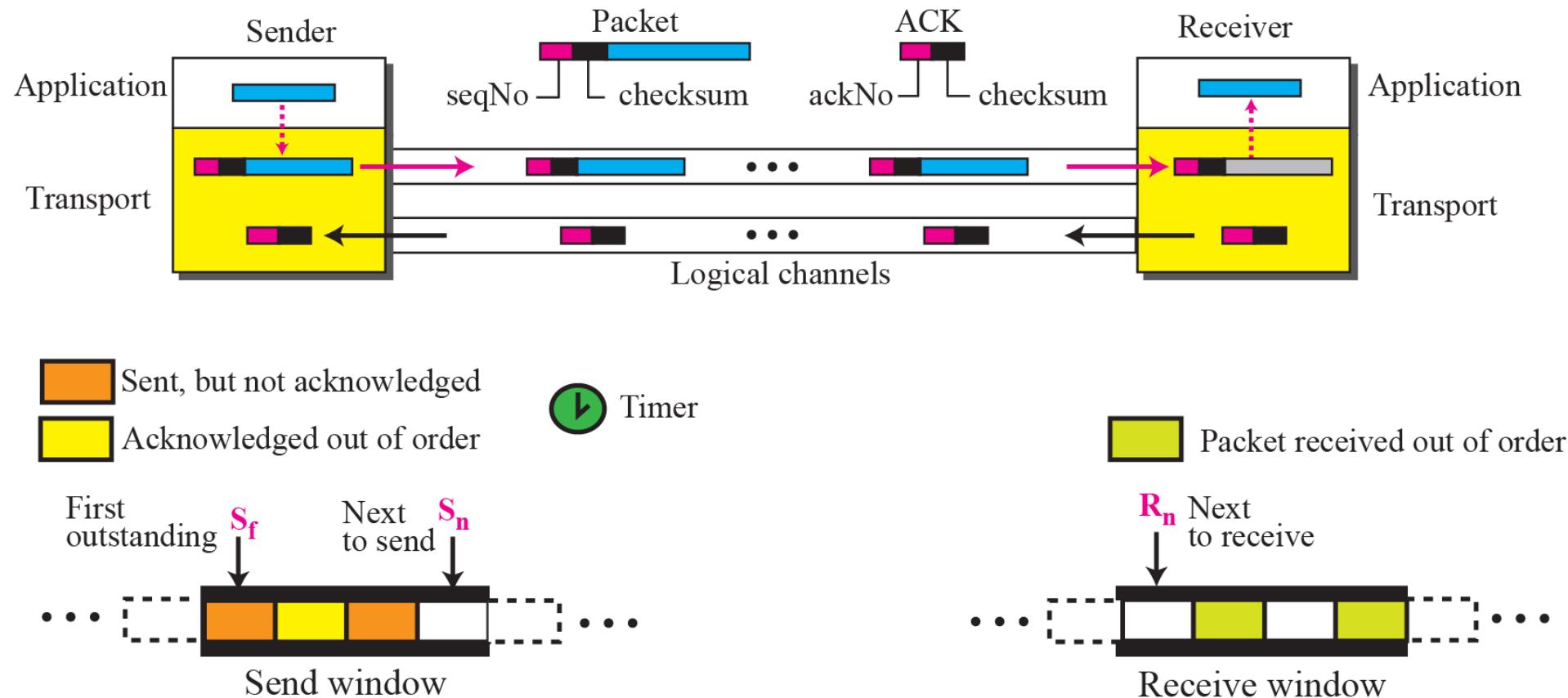
# Transport Layer

Anand Baswade

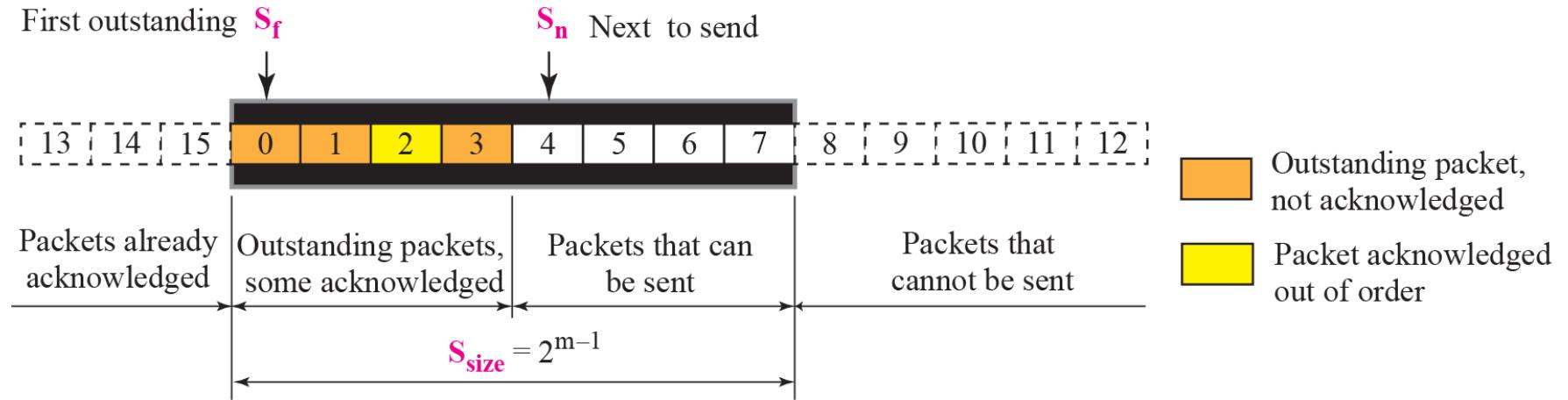
[anand@iitbhilai.ac.in](mailto:anand@iitbhilai.ac.in)



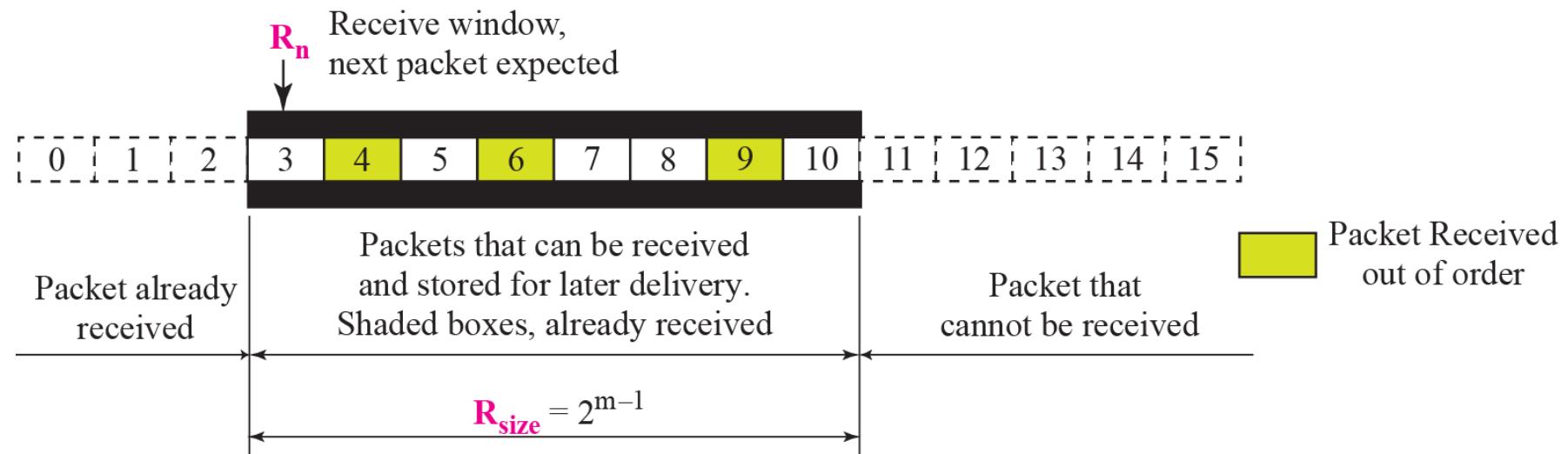
# Outline of Selective-Repeat



# *Send window for Selective-Repeat protocol*



# Receive window for Selective-Repeat protocol



# Example

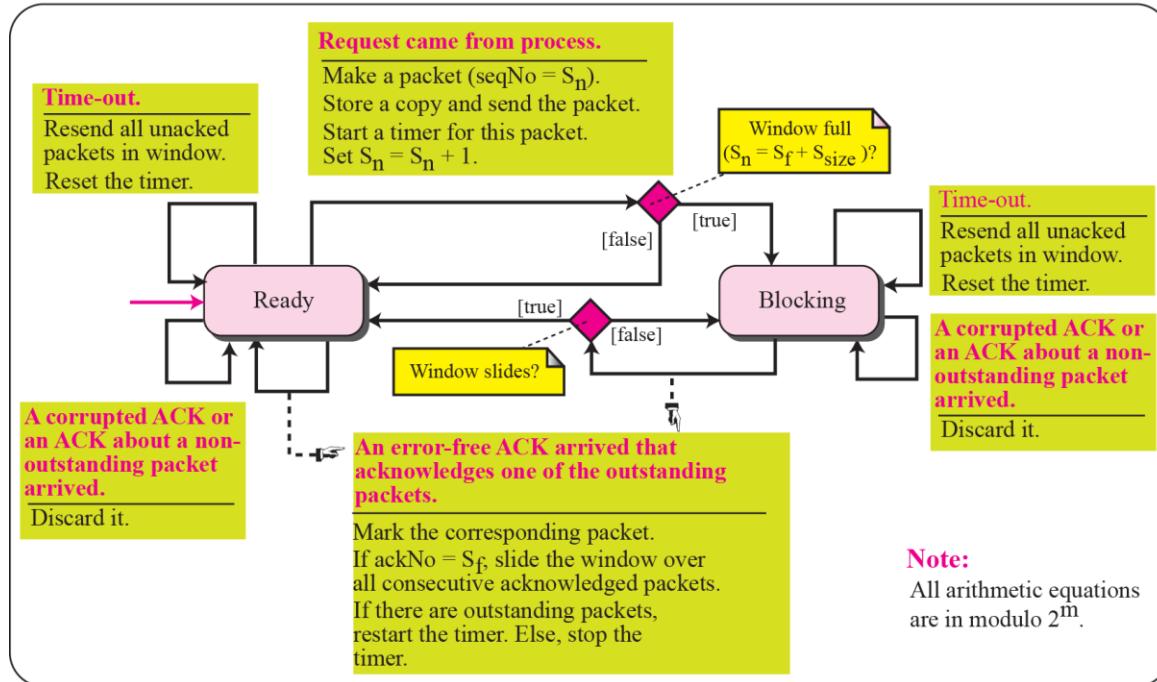
Assume a sender sends 6 packets: packets 0, 1, 2, 3, 4, and 5. The sender receives an ACK with ackNo = 3. What is the interpretation if the system is using GBN or SR?

## *Solution*

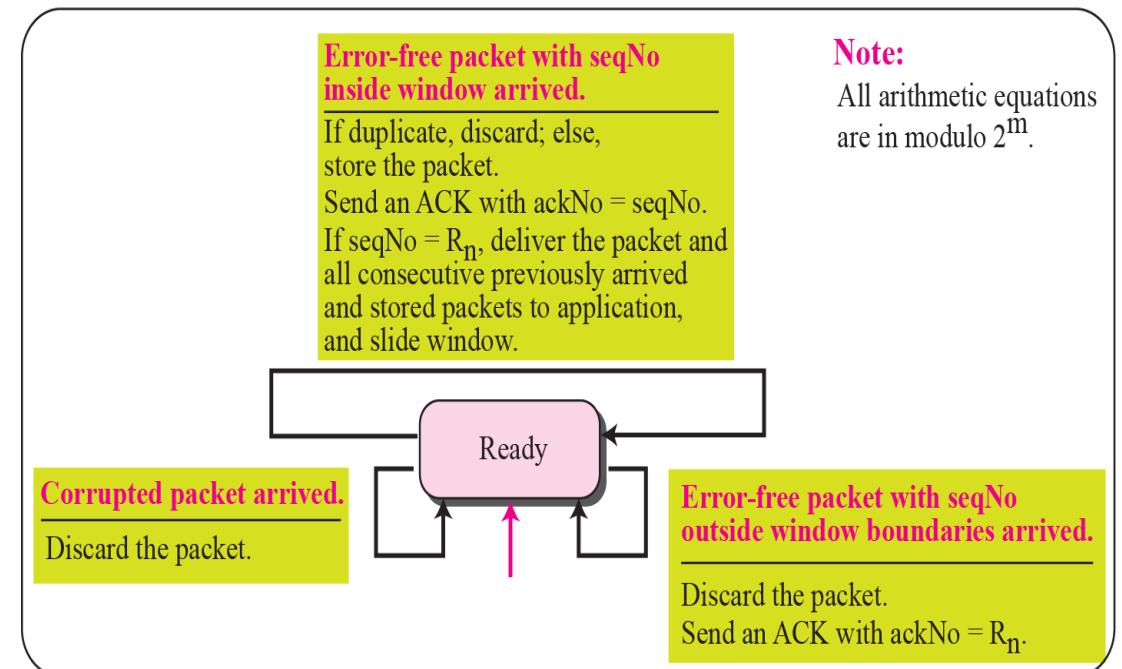
If the system is using GBN, it means that packets 0, 1, and 2 have been received uncorrupted and the receiver is expecting packet 3. If the system is using SR, it means that packet 3 has been received uncorrupted; the ACK does not say anything about other packets.

# FSMs for SR protocol

Sender

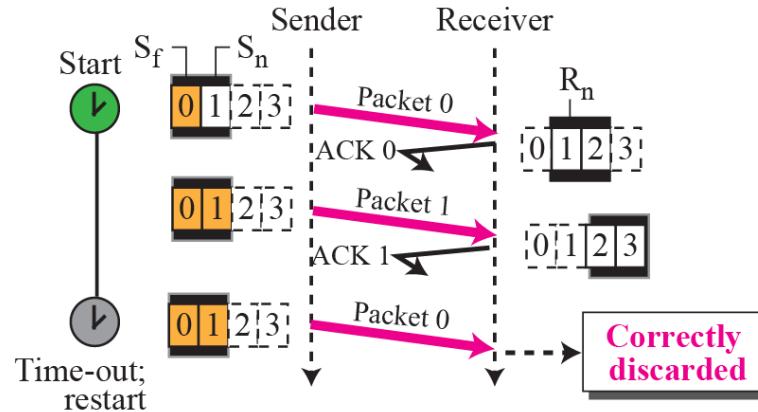


Receiver

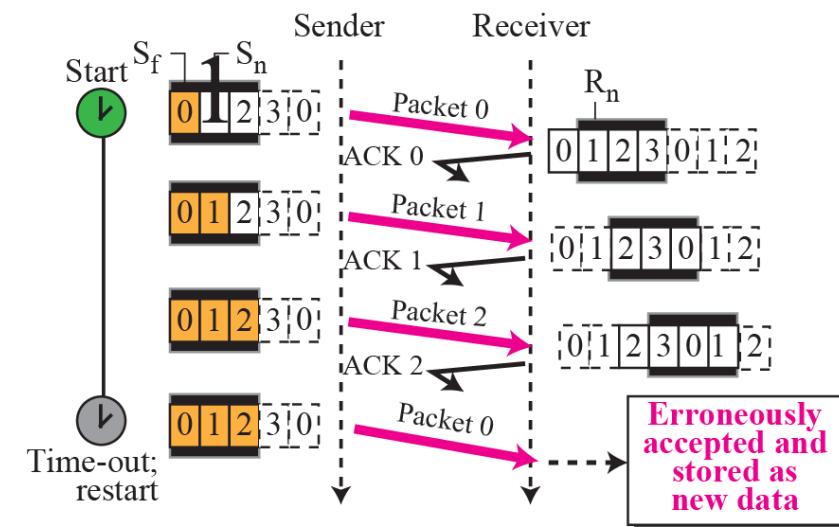


TCP/IP Protocol Suite

# Selective-Repeat window size

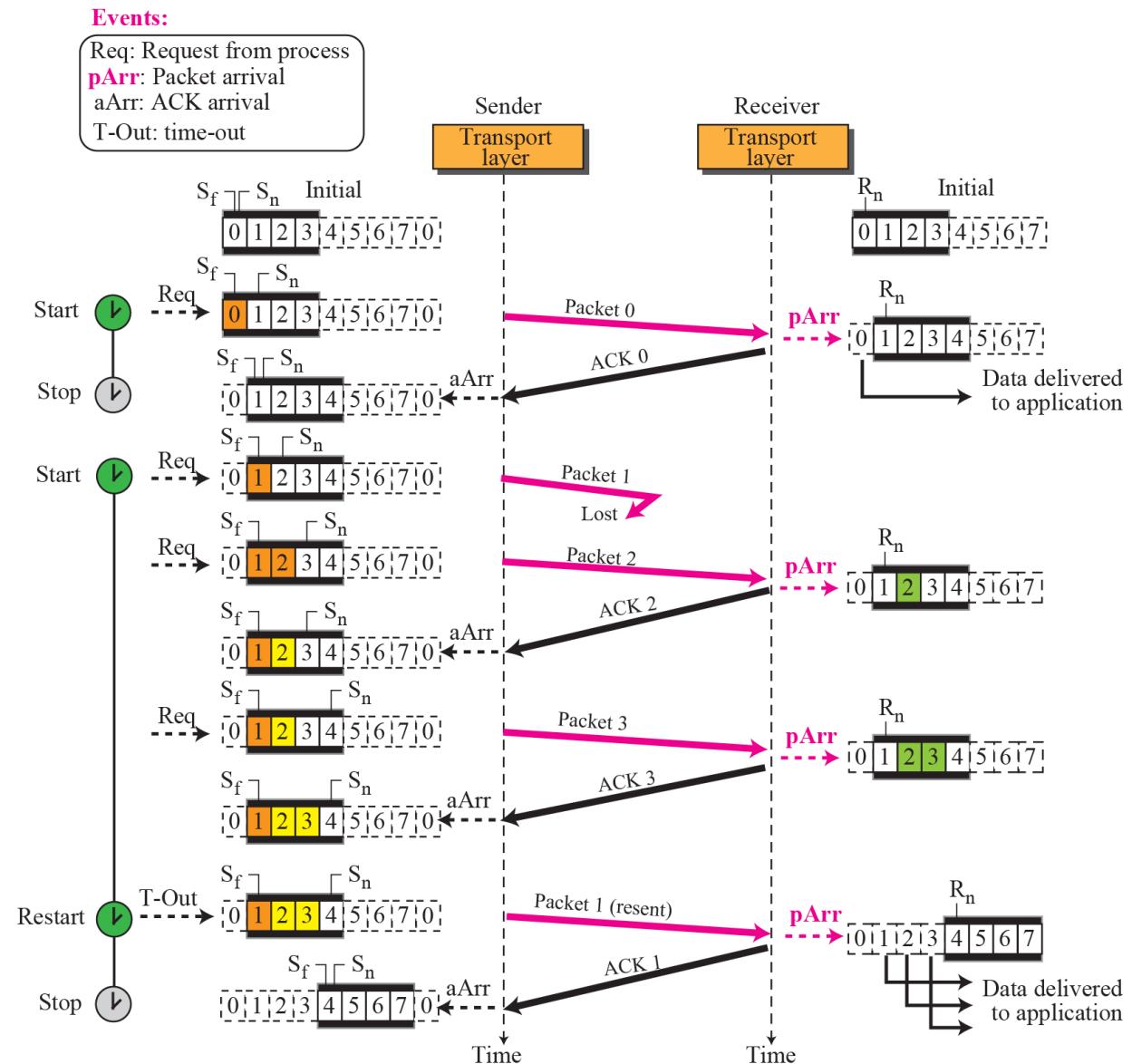


a. Send and receive windows  
of size =  $2^m - 1$



b. Send and receive windows  
of size  $> 2^m - 1$

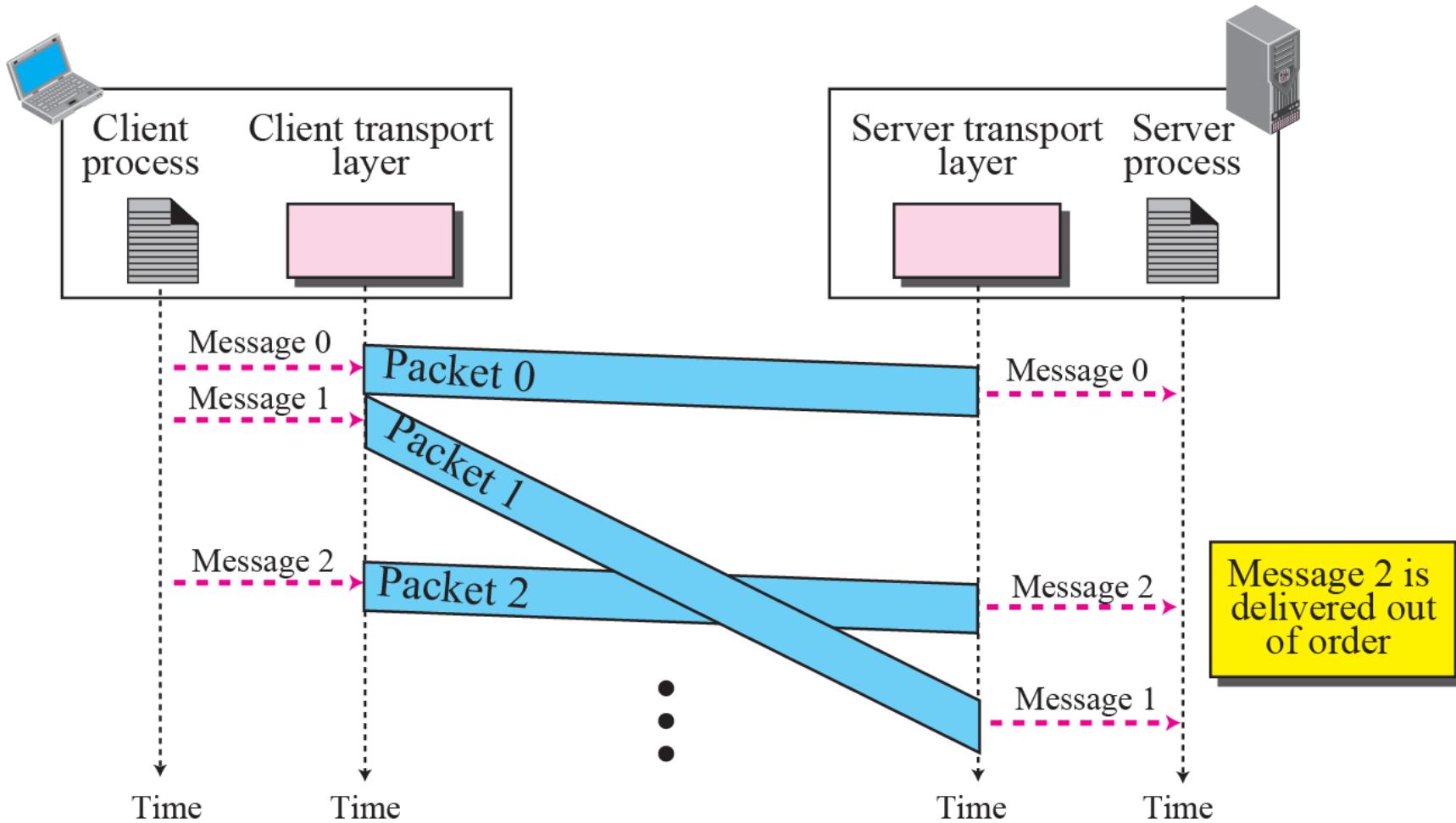
# Example



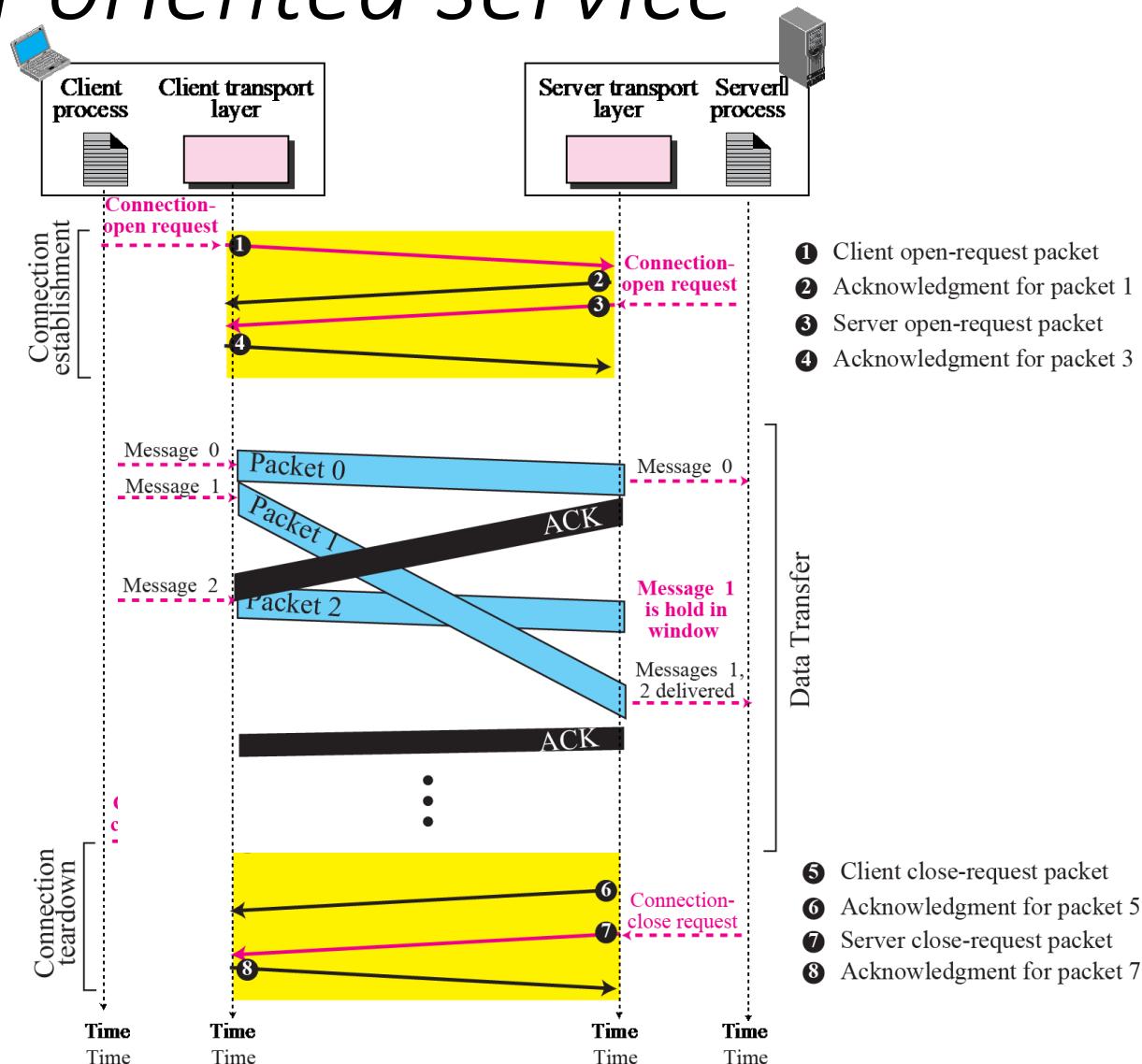
# *Selective-Repeat*

This is the most efficient among the ARQ schemes, but the sender must be more complex so that it can send out-of-order frames. The receiver also must have storage space to store the post-NAK frames and processing power to reinsert frames in proper sequence.

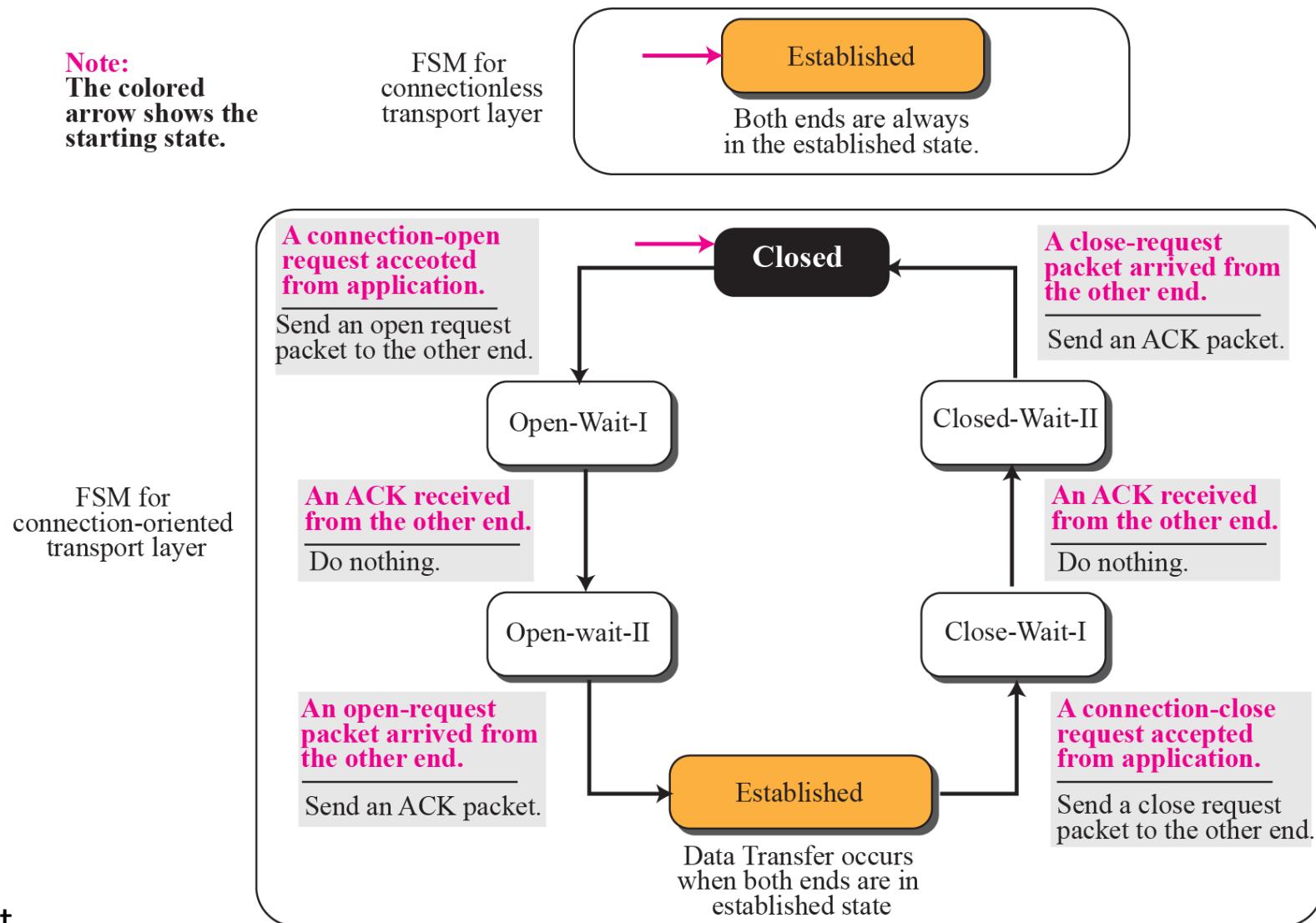
# Connectionless Service



# Connection-oriented service



# Connectionless and connection-oriented services as FSMs



# Transport Layer Protocols

- We can create a transport-layer protocol by combining a set of services described in the previous sections.
- The TCP/IP protocol uses a transport layer protocol that is either a modification or a combination of some of these protocols.

# Chapter 3: roadmap

- Transport-layer services
- **Connectionless transport: UDP**
- Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- Principles of congestion control
- TCP congestion control



# UDP: User Datagram Protocol

- Simple and quick Internet transport protocol
- “best effort” service, UDP segments may be:
  - lost
  - delivered out-of-order to app
- *connectionless*:
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

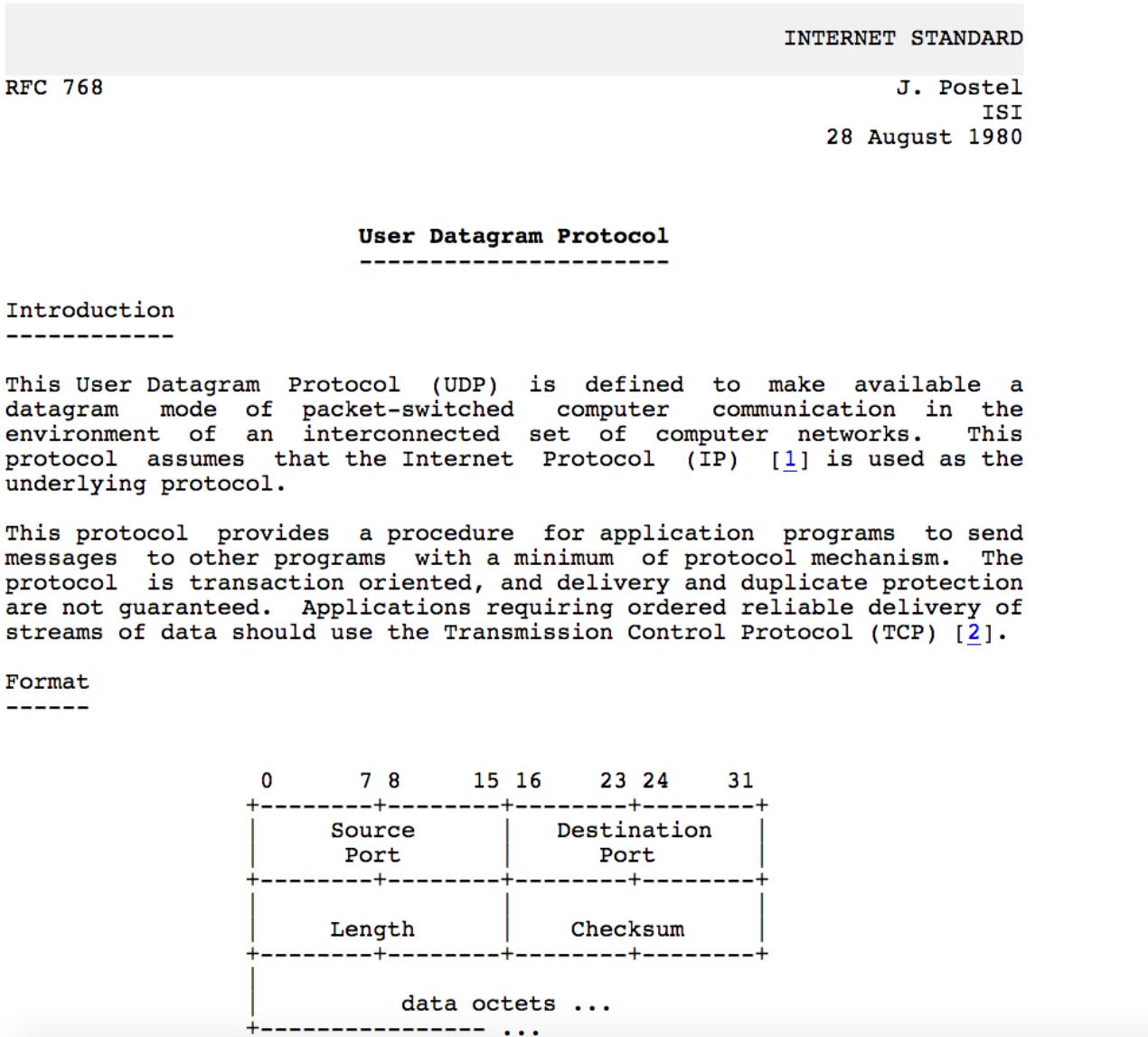
## Why is there a UDP?

- no connection establishment (which can add RTT delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control
  - UDP can blast away as fast as desired!
  - can function in the face of congestion

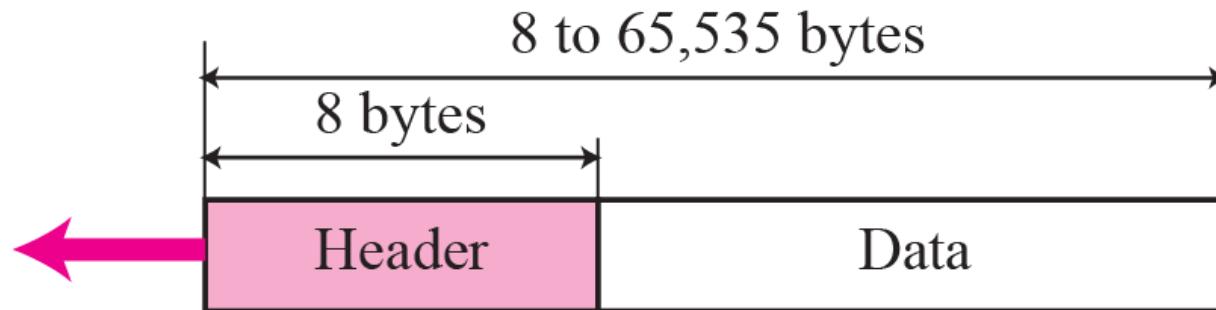
# UDP: User Datagram Protocol

- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
  - SNMP
  - HTTP/3
- if reliable transfer needed over UDP (e.g., HTTP/3):
  - add needed reliability at application layer
  - add congestion control at application layer

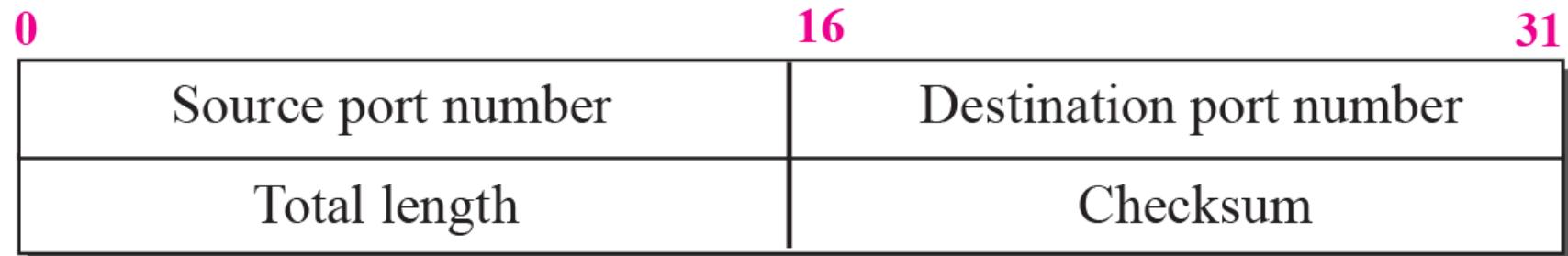
# UDP: User Datagram Protocol [RFC 768]



# UDP Header

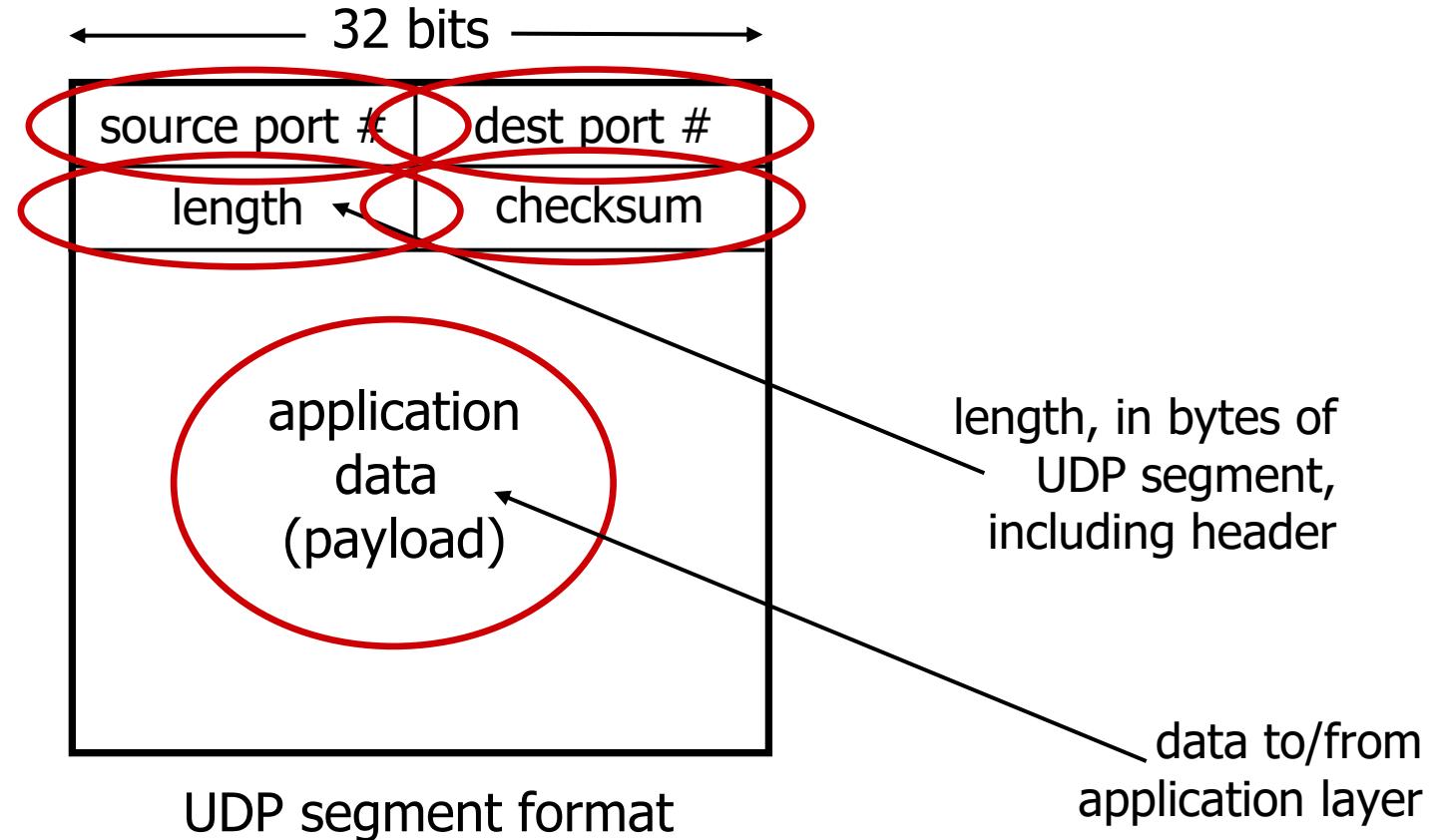


a. UDP user datagram



b. Header format

# UDP segment header cont..



# Questions

The following is a dump of a UDP header in hexadecimal format.

**CB84000D001C001C**

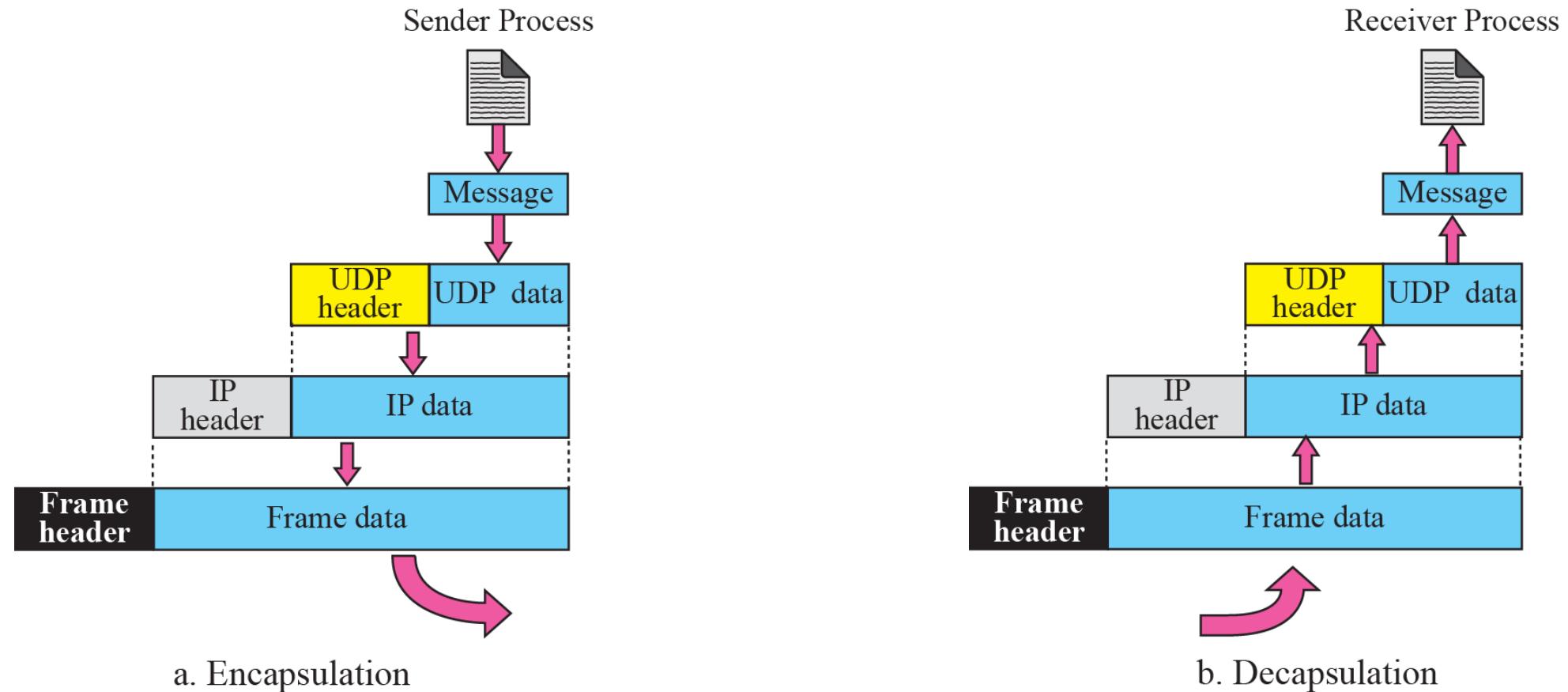
- a. What is the source port number?
- b. What is the destination port number?
- c. What is the total length of the user datagram?
- d. What is the length of the data?
- e. Is the packet directed from a client to a server or vice versa?
- f. What is the client process?

# Answers

## *Solution*

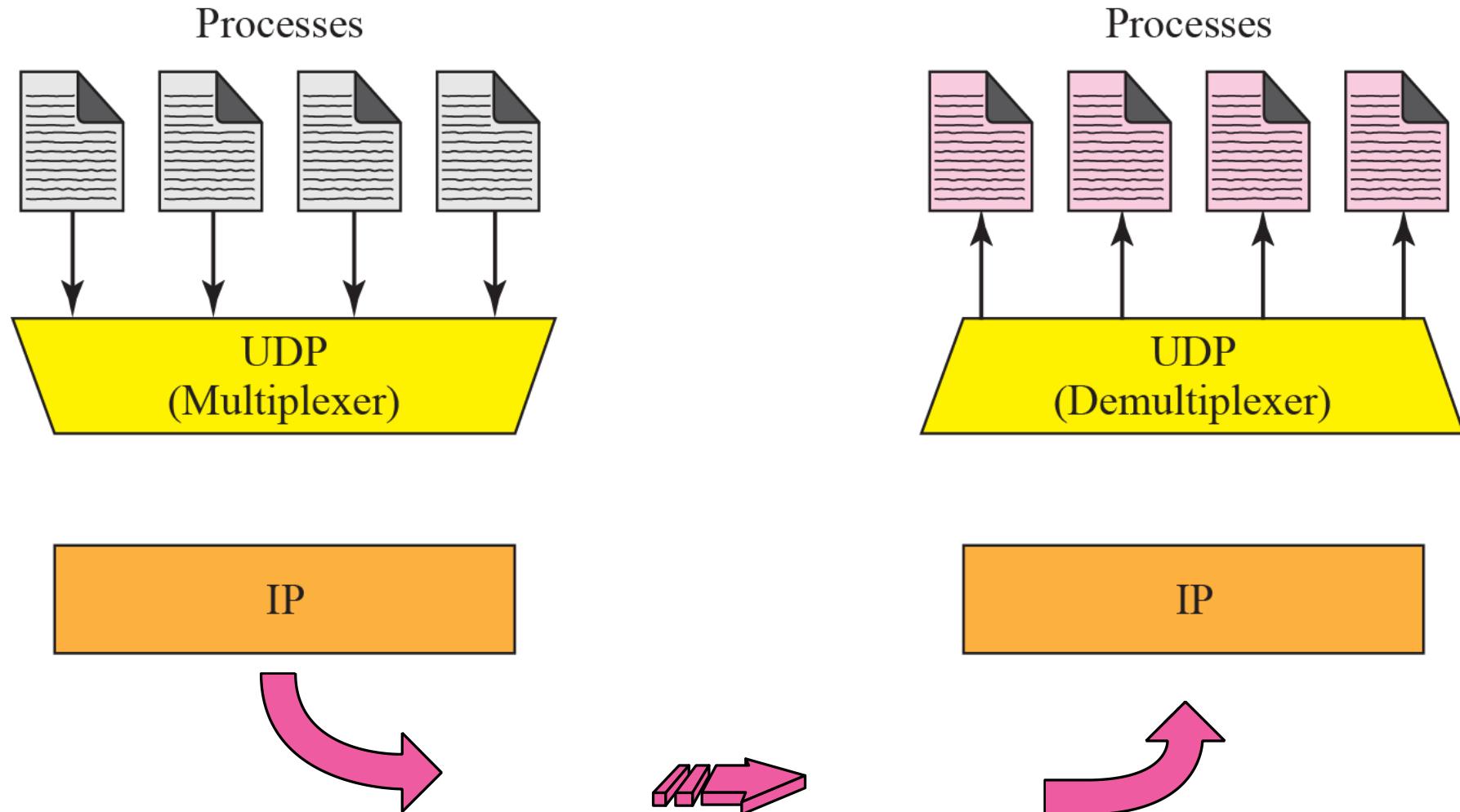
- a. The source port number is the first four hexadecimal digits  $(CB84)_{16}$  or 52100.
- b. The destination port number is the second four hexadecimal digits  $(000D)_{16}$  or 13.
- c. The third four hexadecimal digits  $(001C)_{16}$  define the length of the whole UDP packet as 28 bytes.
- d. The length of the data is the length of the whole packet minus the length of the header, or  $28 - 8 = 20$  bytes.
- e. Since the destination port number is 13 (well-known port), the packet is from the client to the server.
- f. The client process is the Daytime (see Table 14.1).

# Encapsulation and decapsulation



Source: TCP/IP Protocol Suite

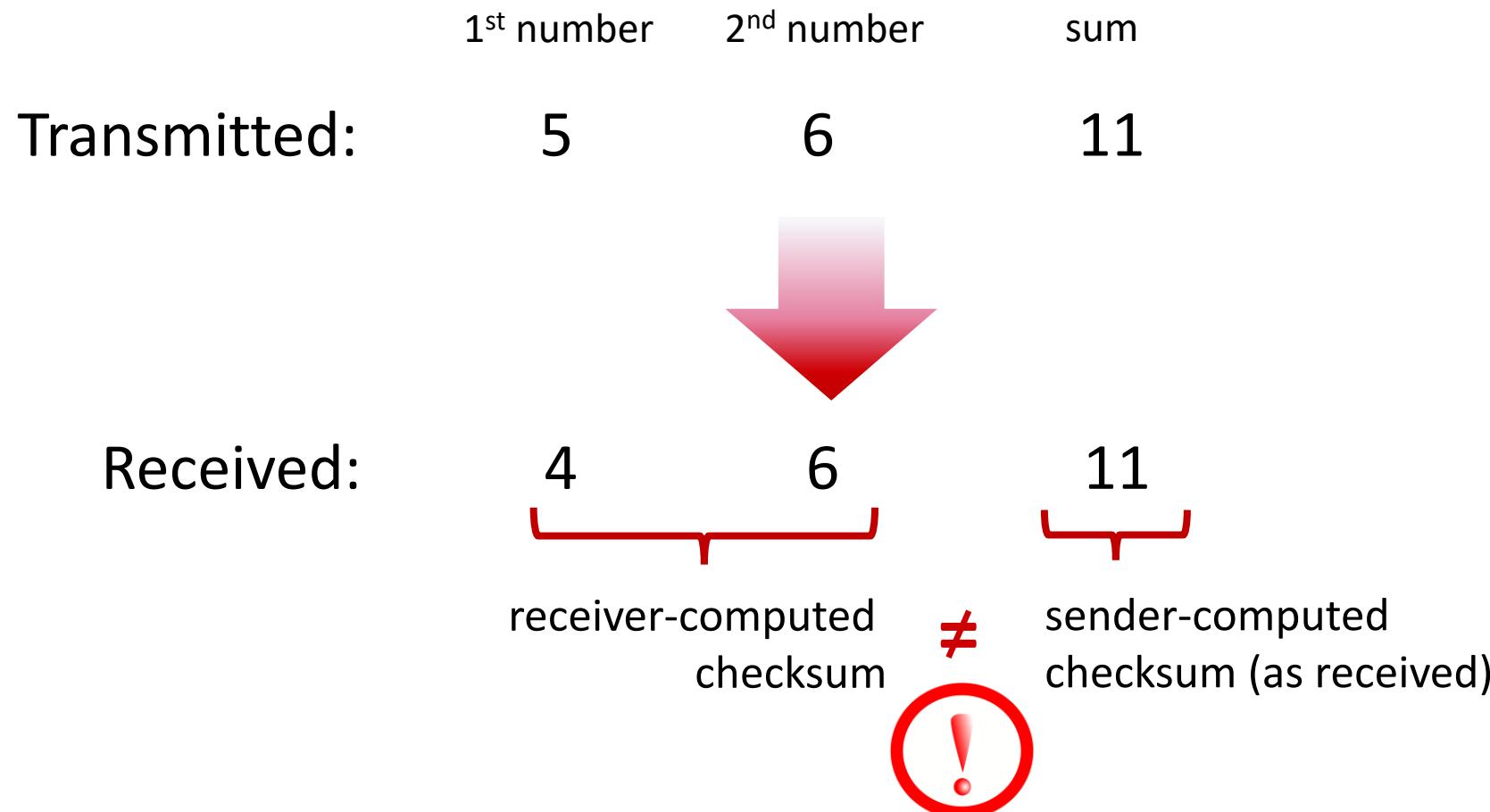
# *Multiplexing and demultiplexing*



SOURCE: TCP/IP Protocol Suite

# UDP checksum

**Goal:** detect errors (*i.e.*, flipped bits) in transmitted segment



# UDP checksum

**Goal:** detect errors (*i.e.*, flipped bits) in transmitted segment

## sender:

- treat contents of UDP segment (including UDP header fields and IP addresses) as sequence of 16-bit integers
- **checksum:** addition (one's complement sum) of segment content
- checksum value put into UDP checksum field

## receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - Not equal - error detected
  - Equal - no error detected. *But maybe errors nonetheless? More later ...*

# Internet checksum: an example

example: add two 16-bit integers

	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
<hr/>																
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1
<hr/>																
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

Note: when adding numbers, a carryout from the most significant bit needs to be added to the result

\* Check out the online interactive exercises for more examples: [http://gaia.cs.umass.edu/kurose\\_ross/interactive/](http://gaia.cs.umass.edu/kurose_ross/interactive/)

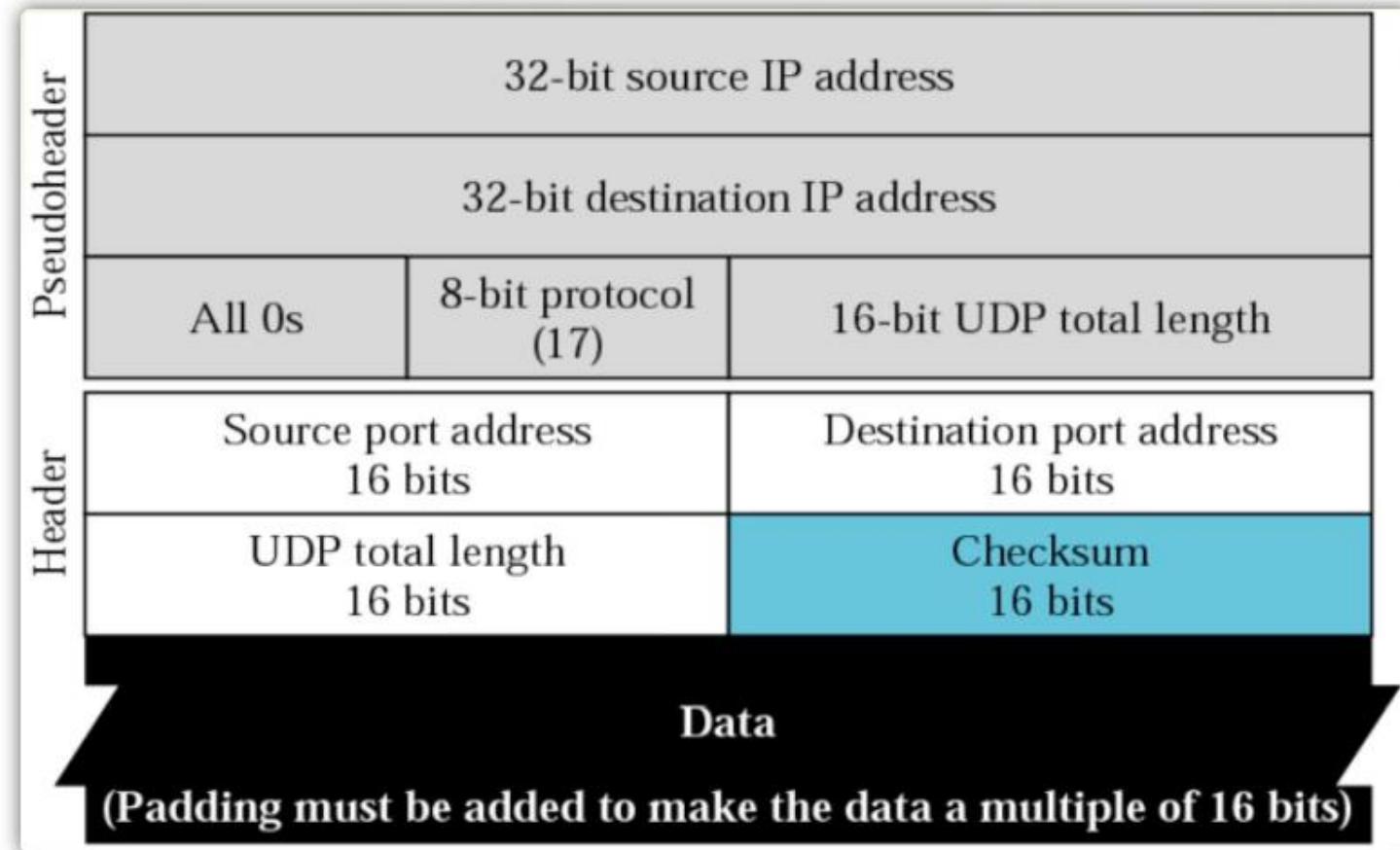
# Internet checksum: weak protection!

example: add two 16-bit integers

	1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	1	0	1
	1	1	0	1	0	1	0	1	0	1	0	1	0	1	1	0	1	0	1
<hr/>																			
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1	0	1
sum	1	0	1	1	1	0	1	1	1	0	1	1	1	1	1	0	0	0	0
checksum	0	1	0	0	0	1	0	0	0	1	0	0	0	0	0	1	1	0	0

Even though numbers have changed (bit flips), **no** change in checksum!

# UDP Checksum Calculations



Cont..

153.18.8.105			10011001 00010010 → 153.18
171.2.14.10			00001000 01101001 → 8.105
All 0s	17	15	10101011 00000010 → 171.2
1087			00001110 00001010 → 14.10
15	All 0s		00000000 00010001 → 0 and 17
T	E	S	00000000 00001111 → 15
I	N	G	00000100 00111111 → 1087
			00000000 00001101 → 13
			00000000 00001111 → 15
			00000000 00000000 → 0 (checksum)
			01010100 01000101 → T and E
			01010011 01010100 → S and T
			01001001 01001110 → I and N
			01000111 00000000 → G and 0 (padding)
			<hr/>
			<b>10010110 11101011</b> → Sum
			<b>01101001 00010100</b> → Checksum

At receiver, add everything including checksum and complement if solution is zero then packet is correctly received.

Source: TCP/IP Protocol Suite, Forouzan

# Questions

What value is sent for the checksum in one of the following hypothetical situations?

- a. The sender decides not to include the checksum.
- b. The sender decides to include the checksum, but the value of the sum is all 1s.
- c. The sender decides to include the checksum, but the value of the sum is all 0s.

# Answers

## *Solution*

- a. The value sent for the checksum field is all 0s to show that the checksum is not calculated.
- b. When the sender complements the sum, the result is all 0s; the sender complements the result again before sending. The value sent for the checksum is all 1s. The second complement operation is needed to avoid confusion with the case in part a.
- c. This situation never happens because it implies that the value of every term included in the calculation of the sum is all 0s, which is impossible; some fields in the pseudoheader have nonzero values.

# Point to Note

- UDP is an example of the connectionless simple protocol we discussed in as a part of Transport layer services with the exception of an optional checksum added to packets for error detection.

# Summary: UDP

- Simple protocol:
  - segments may be lost, delivered out of order
  - best effort service: “send and hope for the best”
- UDP has its plusses:
  - no setup/handshaking needed (no RTT incurred)
  - can function when network service is compromised
  - helps with reliability (checksum) → **Optional**
- build additional functionality on top of UDP in application layer (e.g., HTTP/3)