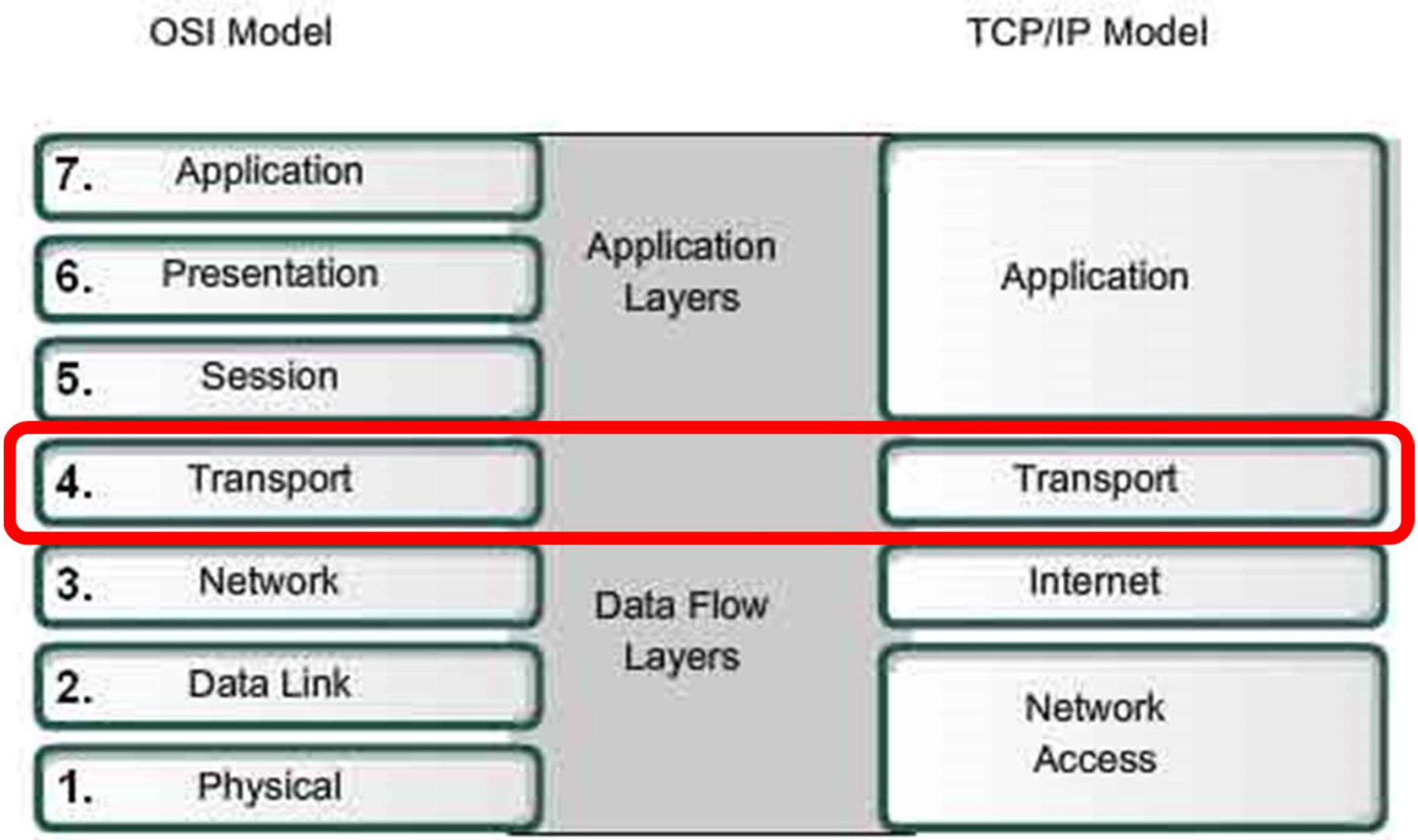
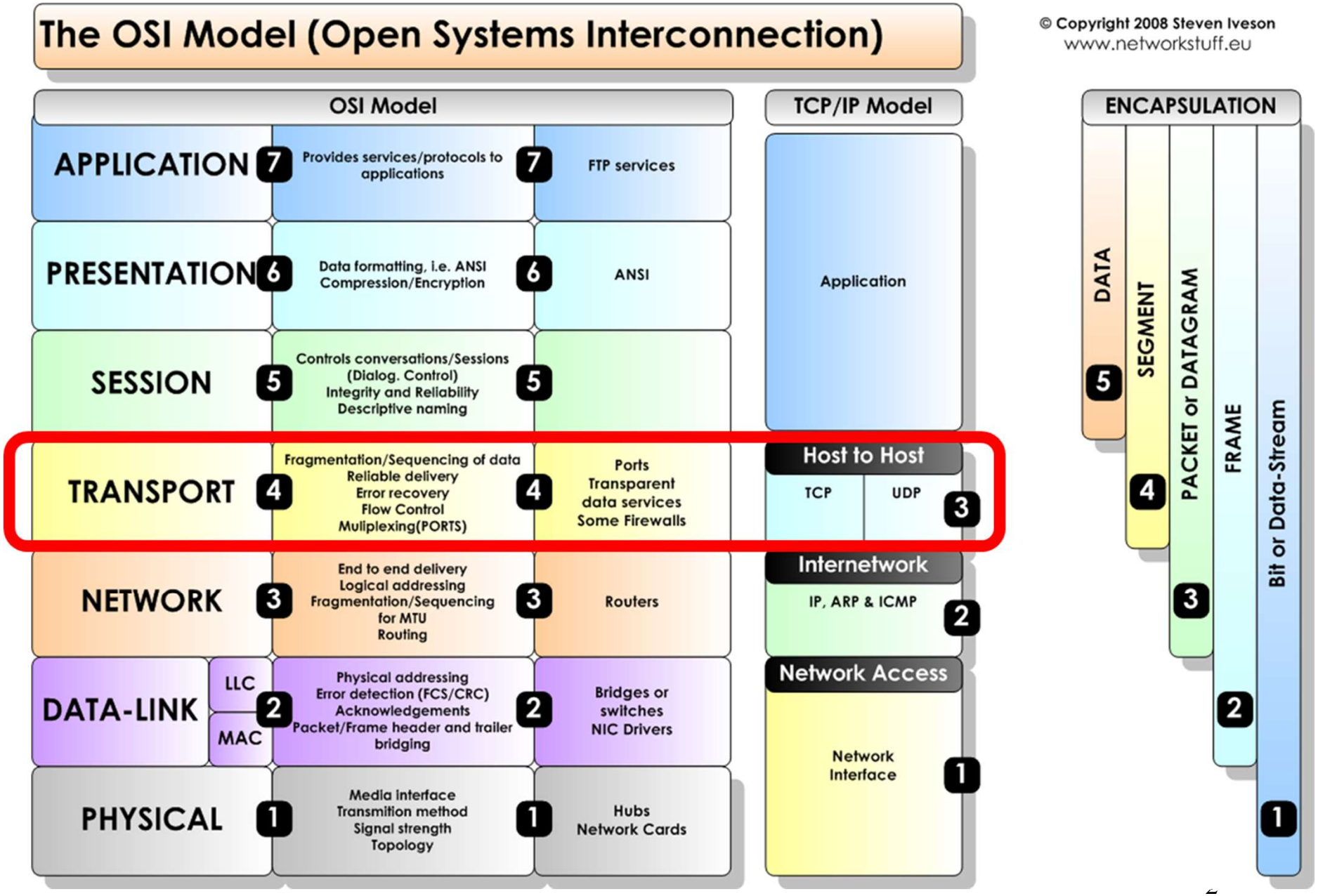

OSI Model
TCP/IP Model
7. Application
6. Presentation
5. Session
4. Transport
3. Network
2. Data Link
1. Physical
Application Layers
Data Flow Layers
Application
Transport
Internet
Network Access


The OSI Model (Open Systems Interconnection)
© Copyright 2008 Steven Iveson
www.networkstuff.eu
OSI Model
TCP/IP Model
ENCAPSULATION
APPLICATION 7
Provides services/protocols to applications 7
FTP services
PRESENTATION 6
Data formatting, i.e. ANSI Compression/Encryption 6
ANSI
SESSION 5
Controls conversations/Sessions (Dialog. Control) Integrity and Reliability Descriptive naming 5
Application
TRANSPORT 4
Fragmentation/Sequencing of data Reliable delivery Error recovery Flow Control Muliplexing(PORTS) 4
Ports Transparent data services Some Firewalls
Host to Host
TCP
UDP
3
NETWORK 3
End to end delivery Logical addressing Fragmentation/Sequencing for MTU Routing 3
Routers
Internetwork
IP, ARP & ICMP
2
DATA-LINK
LLC
MAC
2
Physical addressing Error detection (FCS/CRC) Acknowledgements Packet/Frame header and trailer bridging 2
Bridges or switches NIC Drivers
Network Access
Network Interface
1
PHYSICAL 1
Media interface Transmition method Signal strength Topology 1
Hubs Network Cards
DATA
5
SEGMENT
4
PACKET or DATAGRAM
3
FRAME
2
Bit or Data-Stream
1

Transport-level Protocols

Connection Oriented

Transmission Control Protocol (TCP)

Logical connection

Establishment

Maintenance, termination

Reliable services

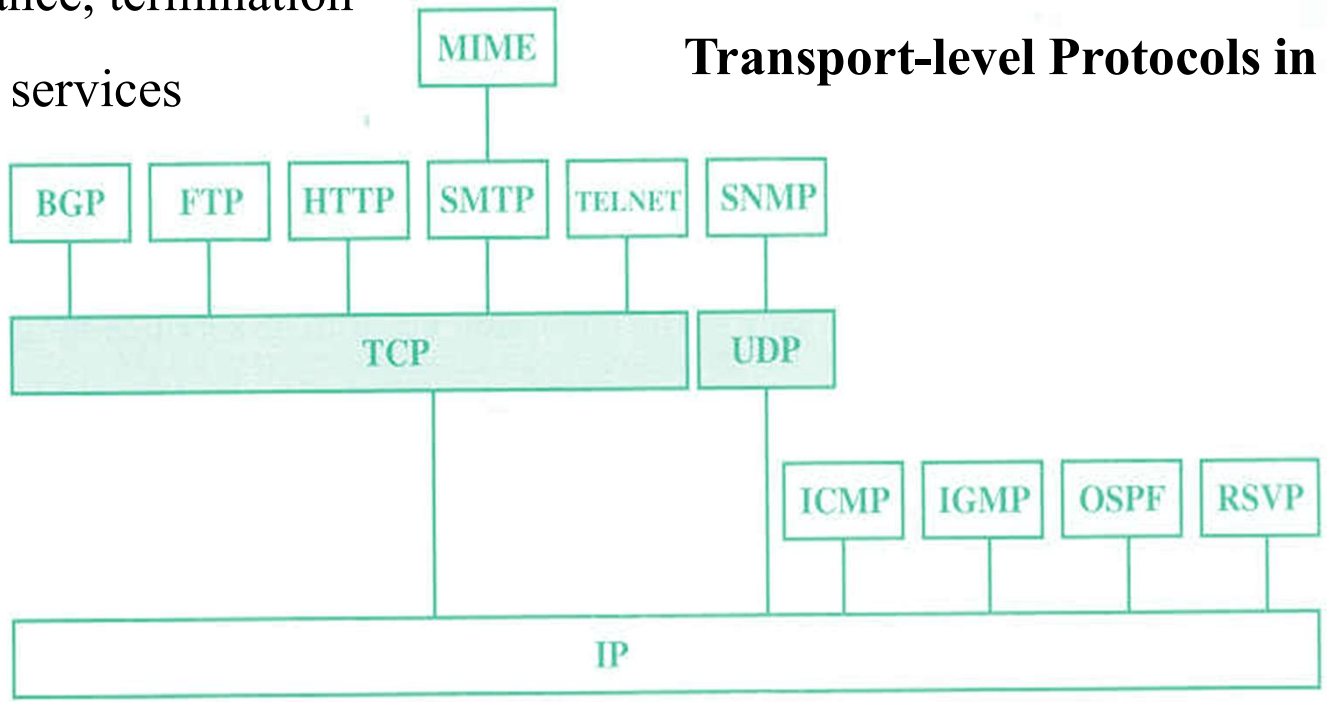
Connectionless

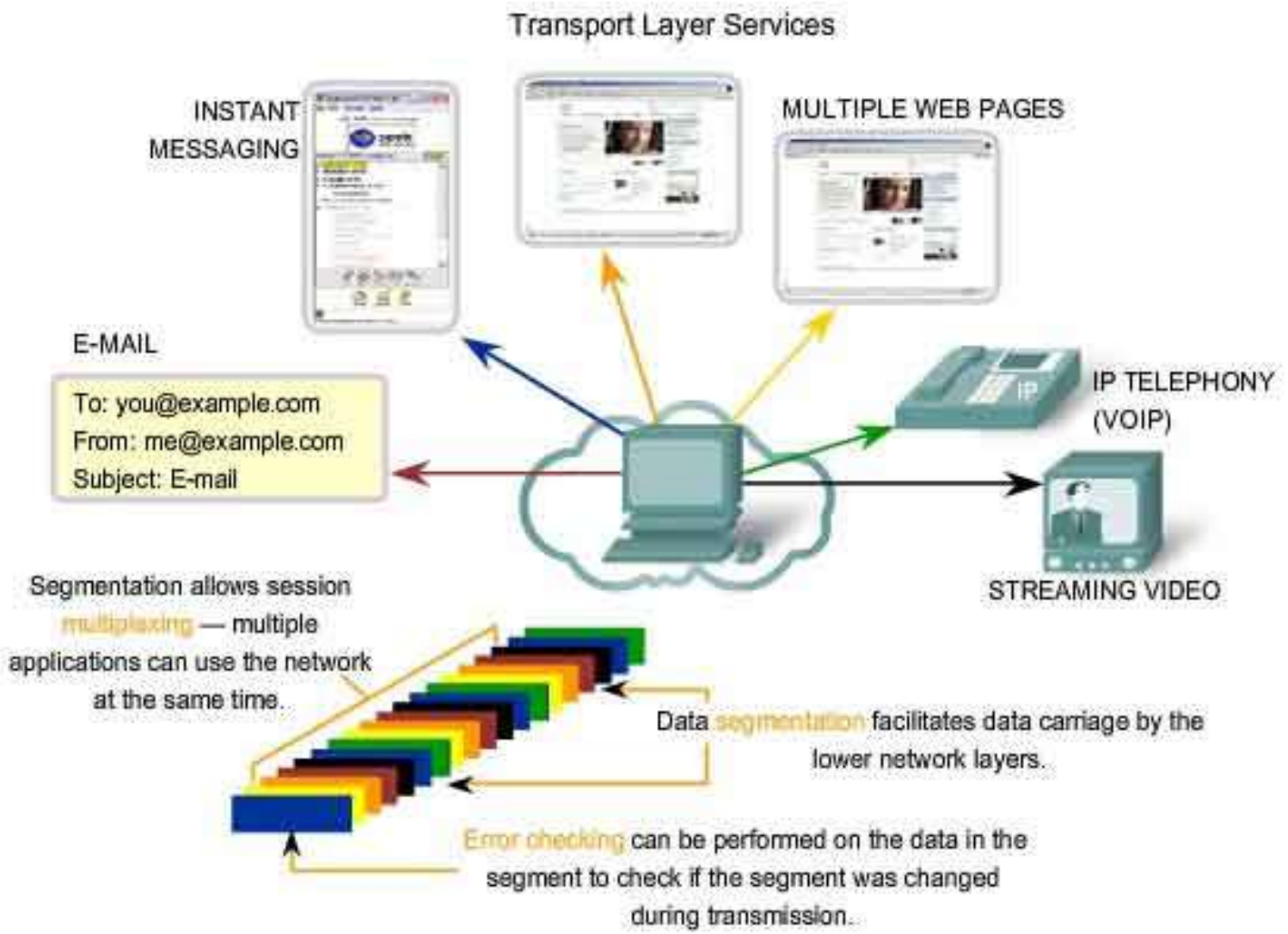
User Datagram Protocol (UDP)

Connectionless

'Best-effort' delivery

Transport-level Protocols in Context




Transport Layer Services
INSTANT MESSAGING
MULTIPLE WEB PAGES
E-MAIL
To: you@example.com
From: me@example.com
Subject: E-mail
IP TELEPHONY (VOIP)
STREAMING VIDEO
Segmentation allows session multiplexing — multiple applications can use the network at the same time.
Data segmentation facilitates data carriage by the lower network layers.
Error checking can be performed on the data in the segment to check if the segment was changed during transmission.

Transmission Control Protocol (TCP)

Connection oriented

Reliable byte stream over unreliable IP

IP may be transported over many different network technologies

RFC 793

Each end: a socket

Socket is a pair of: IP address + port number

Multiple connections may be active on a socket

Full duplex connections

Point to point links

No multicast or broadcast

Other protocols used for these

May buffer information to increase amount sent

PUSH flag requests that buffered data be sent

Communicating computers must agree on a port number

`Server' opens selected port and waits for incoming messages

`Client' selects local port and sends message to selected port

Services provided by many computers use reserved, *well-known* port numbers:

TFTP, DNS, Echo

Other services use *dynamically assigned* port numbers

Port	**Name**	**Description**
7	echo	Echo input back to sender
9	discard	Discard input
11	systat	System statistics
13	daytime	Time of day (ASCII)
17	quote	Quote of the day
19	chargen	Character generator
37	time	System time (seconds since 1970)
53	domain	DNS
69	tftp	Trivial File Transfer Protocol (TFTP)
123	ntp	Network Time Protocol (NTP)
161	snmp	Simple Network Management Protocol (SNMP)

Transmission Control Protocol (TCP)

TCP general features

- *Connection oriented*: Application requests connection to destination and then uses connection to deliver data to transfer data
- *Point-to-point*: A TCP connection has two endpoints
- *Reliability*: TCP guarantees data will be delivered without loss, duplication or transmission errors
- *Full duplex*: The endpoints of a TCP connection can exchange data in both directions simultaneously
- *Stream interface*: Application delivers data to TCP as a continuous *stream*, with no record boundaries; TCP makes no guarantees that data will be received in same blocks as transmitted
- *Reliable connection establishment*: *Three-way handshake* guarantees reliable, synchronized startup between endpoints
- *Graceful connection termination*: TCP guarantees delivery of all data after endpoint shutdown by application

TCP Services

Reliable communication between pairs of processes (applications)

Across variety of reliable and unreliable networks and internets

Two labeling facilities:

- Data stream push
  - TCP user can require transmission of all data up to push flag
  - Receiver will deliver in same manner
  - Avoids waiting for full buffers
- Urgent data signal
  - Indicates urgent data is upcoming in stream
  - User decides how to handle it

TCP uses IP for data delivery

•Endpoints are identified by ports (sockets)

-16-bit number

--System Ports range (0-1023): “well-known ports” which provide well-known services; assignment through IANA

https://www.iana.org/assignments/service-names-port-numbers/service-names-port-numbers.xhtml

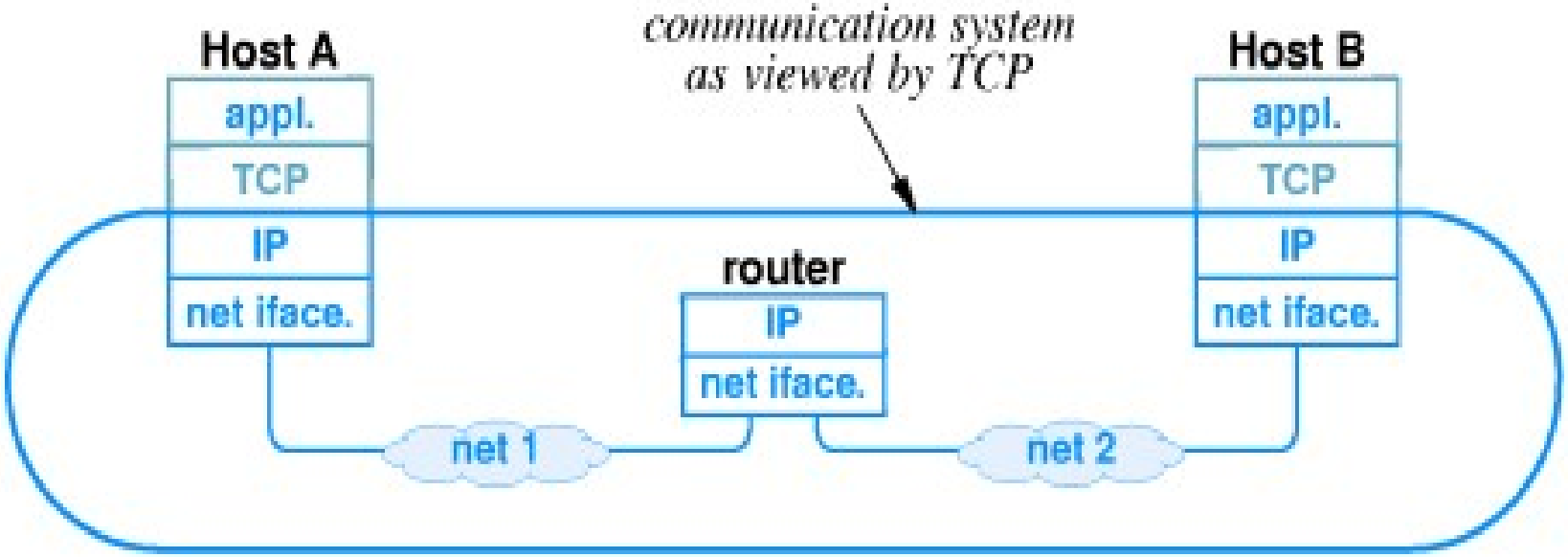
--User Ports range (1024-49151) are available for assignment through IANA

--Dynamic Ports range (49152-65535) have been specifically set aside for local and dynamic use and cannot be assigned through IANA.

•Allows multiple connections on each host

•Ports may be associated with an application or a process

•IP treats TCP like data and does not interpret any contents of the TCP message



TCP services at the boundary with the process (application level) are defined using the concepts of abstract service primitives (TCP ASPs) and service-access points (SAPs). The primitives are implemented using the segment header fields or passing some parameters at the IP level.

Items Passed to IP

TCP passes some parameters down to IP

- Precedence of segments
- Normal delay/low delay
- Normal throughput/high throughput
- Normal reliability/high reliability
- Security

Also, TCP Protocol:

Breaks application messages into *segments* (TCP data units)

Each segment has an (at least) 20 byte header

Segments are sized based on:

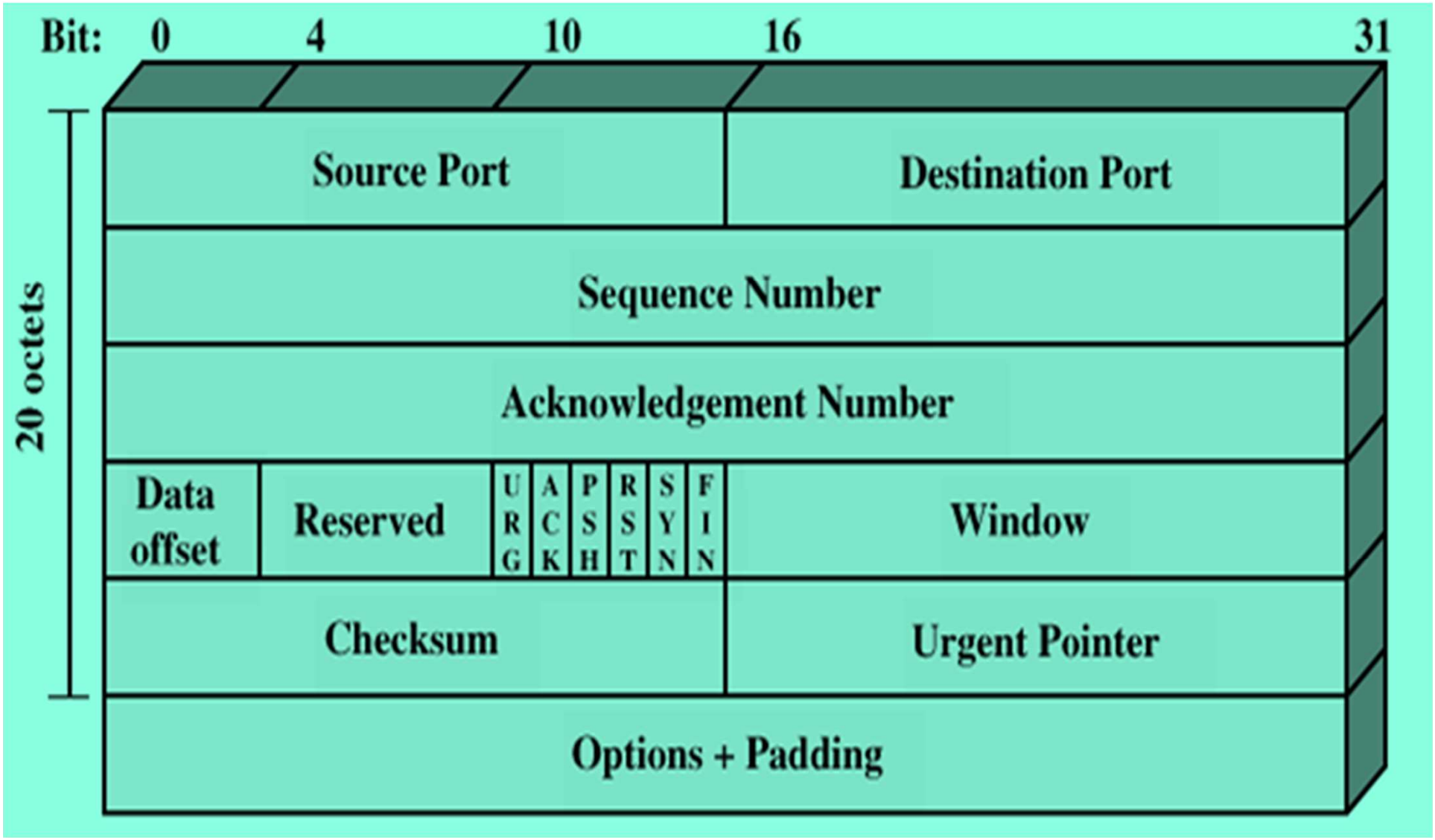
being less the 64koctets (the path Maximum Transmission Unit (MTU))

Segments may be *fragmented* during transmission if MTU smaller than packet

TCP Header Fields

Source port – source TCP user
Destination port – destination TCP user
Sequence number – sequence number of the first data octet in this segment
Acknowledgement number – piggybacking acknowledgement, contains sequence number of next data octet the destination TCP entity expects to receive
Data offset – number of words (32 bit) in this header (header flexible length)
Reserved – for future use & developments

TCP Header


Bit:
0
4
10
16
31
20 octets
Source Port
Destination Port
Sequence Number
Acknowledgement Number
Data offset
Reserved
URG
ACK
PSH
RST
SYN
FIN
Window
Checksum
Urgent Pointer
Options + Padding

Flags – 6 bits:

URG – urgent pointer field significant

ACK – acknowledgement field significant

PSH – Push function

RST – reset connection

SYN – synchronize the sequence numbers (used in connection establishment)

FIN – no more data from sender (used in connection termination)

Window – flow control credit allocation in octets (window size)

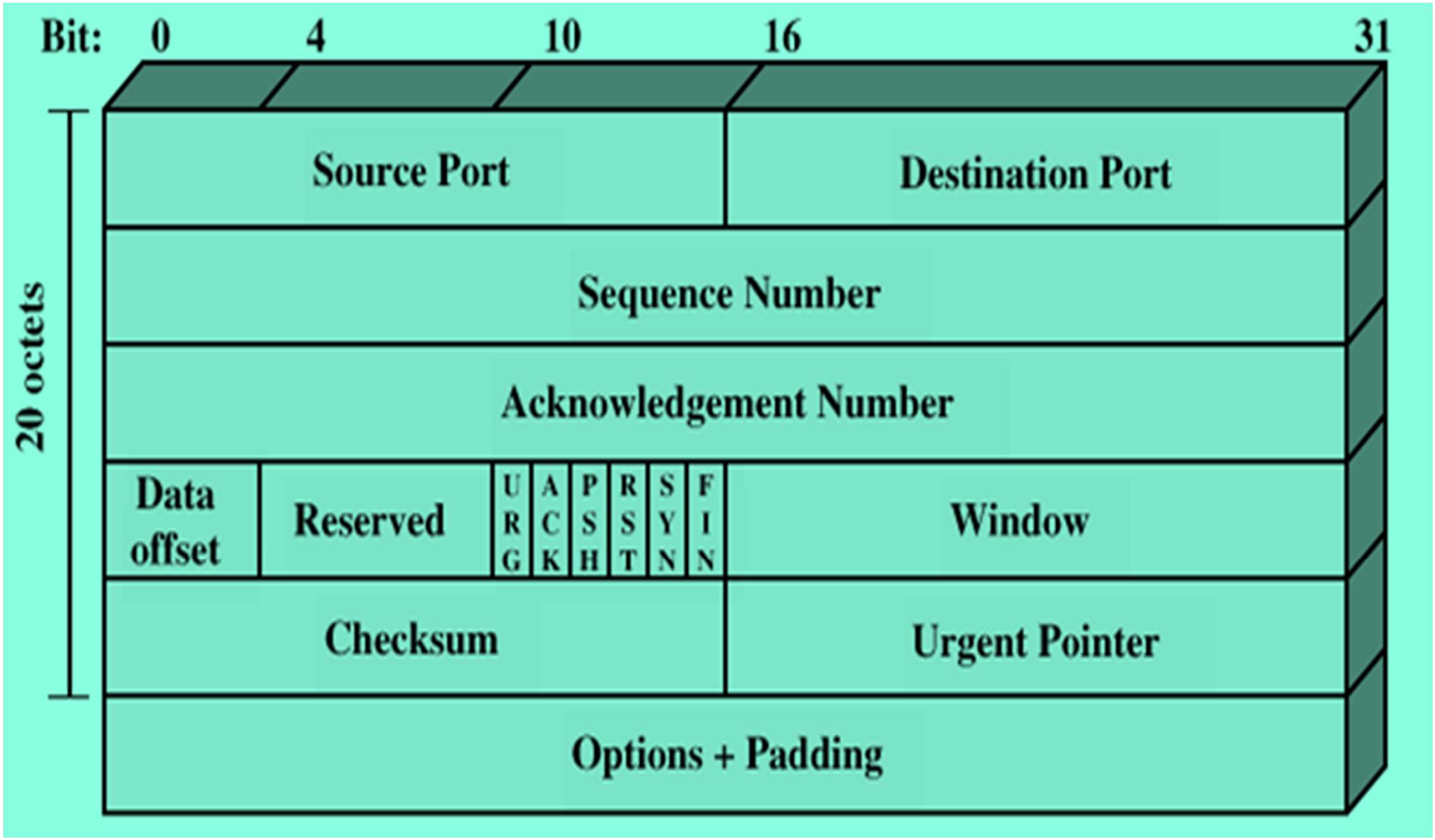
Checksum – ones complement of the sum modulo 2^{16}-1 of all 16-bit words in the segment (plus eventually pseudo-header)

Urgent Pointer – pointer to the last octet in a sequence of urgent data

Options – variable field

Padding – to meet segment length of multiple of 32 bit

TCP Header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RST
SYN
FIN
Window
Checksum
Urgent Pointer
Options + Padding

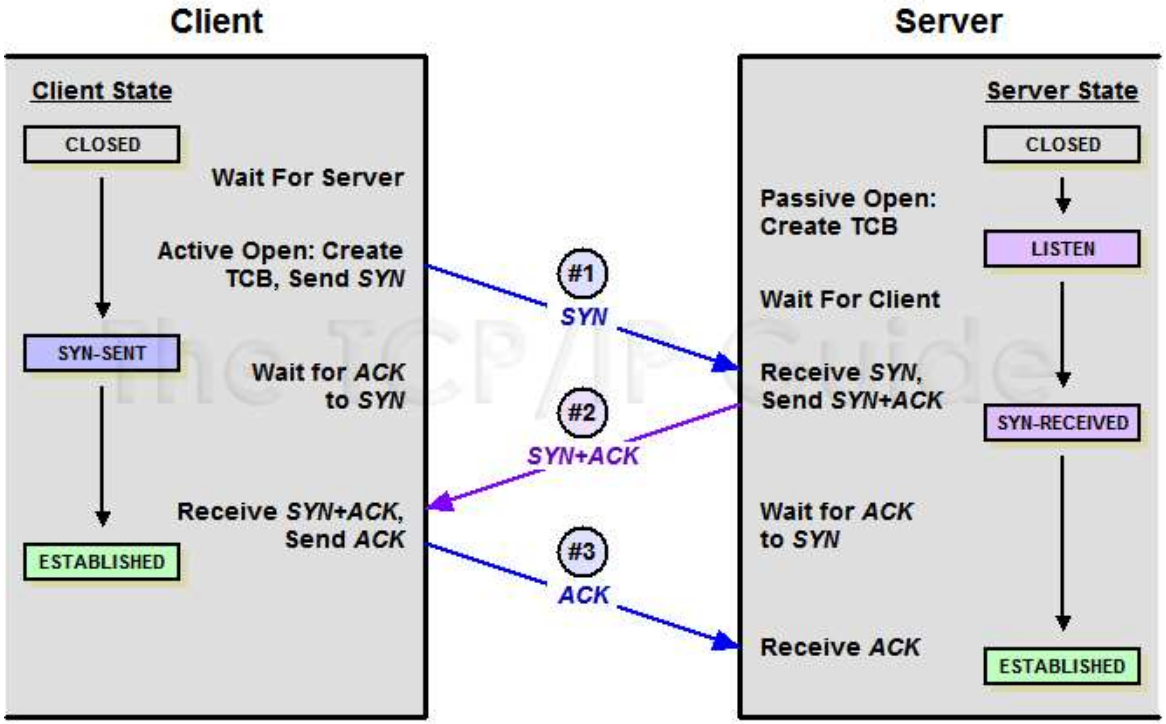
TCP Mechanisms

Connection establishment

Three way handshake used for connection establishment (exchange of SYNs)

Each TCP connection is established between pair of ports

One port can connect to multiple destinations (may support multiple connections)



TCP Mechanisms

Connection establishment

- Three way handshake used for connection establishment (exchange of SYNs)
- Each TCP connection is established between pair of ports
- One port can connect to multiple destinations (may support multiple connections)

Data transfer

- Logically, data is considered a stream of octets
- Octets numbered modulo 2^{32}
- Data is transferred over a TCP connection in segments
- Flow control by credit allocation of number of octets
- Data buffered at transmitter and receiver
- Use of *PUSH* flag for forcing transmission of so far accumulated date (end-of-block function)
- User may specify *urgent* data transmission

Connection termination

- Graceful close (normal exchange of FIN info)
  - TCP users issues CLOSE primitive
  - Transport entity sets FIN flag on last segment sent
- Abrupt termination by ABORT primitive
  - Entity abandons all attempts to send or receive data
  - RST segment transmitted (connection Reset)

Implementation Policy Options (allows for possible TCP implementations)

- Send policy
- Deliver policy
- Accept policy
- Retransmit policy
- Acknowledge policy

Send policy

If no *Push* flag or CLOSE indication, a TCP entity transmits at its own convenience

- Used Data buffered at transmit buffer
- TCP entity may construct segment per data batch as provided by user
- May wait for certain amount of data

Policy depends on performance considerations (header overhead/response speed)

Delivery Policy

In absence of *Push*, receiving TCP entity may deliver data to the user at own convenience

- May deliver as each in order segment received
- May buffer data from more than one segment

Policy depends on how promptly user needs data, or how much processing involved (each delivery = application software interrupts)

Acknowledgement policy

- Immediate ACK
- Cumulative ACK

Accept policy

Segments may arrive out of order; options:

- In order
  - Only accept segments in order
  - Discard out of order segments
- In windows
  - Accept all segments within receiver window

Retransmit policy

TCP maintains queue of segments transmitted but not acknowledged

Policy depends mainly on receiver acceptance policy (in order or in window)

TCP will retransmit if not ACKed in given time; retransmission options:

- First only segment from the queue (necessary one timer for entire queue)
- Batch – all segments from queue (one timer for entire queue)
- Individual – one timer for each segment in queue; retransmits the individual segment

TCP Congestion Control

Basic idea in congestion avoidance: don't insert a new packet until an old one leaves

Other Basic idea: timeouts associated with retransmission are due to congestion state

Approaches based on re-transmission timer & window size management:

Timers for re-transmission: time-out management

TCP Congestion control estimate round trip delay, by observing delay pattern in recent segments, and then sets the timer to a value a little bit greater

Estimations based on:

Simple average

Average Round-Trip Time (ARTT) for a segment *i*, is the simple average of delays for the precedent *k* transmitted segments.

Exponential Average

Gives a better prediction of the next RTT value

Binary exponential Backoff algorithm (see CSMA/CD) may be used for obtaining values for the retransmission time-out values (RTOs)

Exponential RTO Backoff

Since timeout value is probably due to congestion (dropped packet or long round trip), maintaining same RTO is not a good idea

RTO increased each time a segment is re-transmitted

RTO = q*RTO

Commonly q=2

Known as Binary exponential backoff

Window management

TCP Window size may affect transmission parameters; need for managing size:

Used Techniques

Slow Start

If start using window size as provided by past connection, may cause excessive flow, internet conditions may differ

Solution: sender starting with a smaller window size

Two windows: allowed window and congestion window, sized in segments, no octets

Allwd_wndw = min (cngst_wndw, gained_credit)

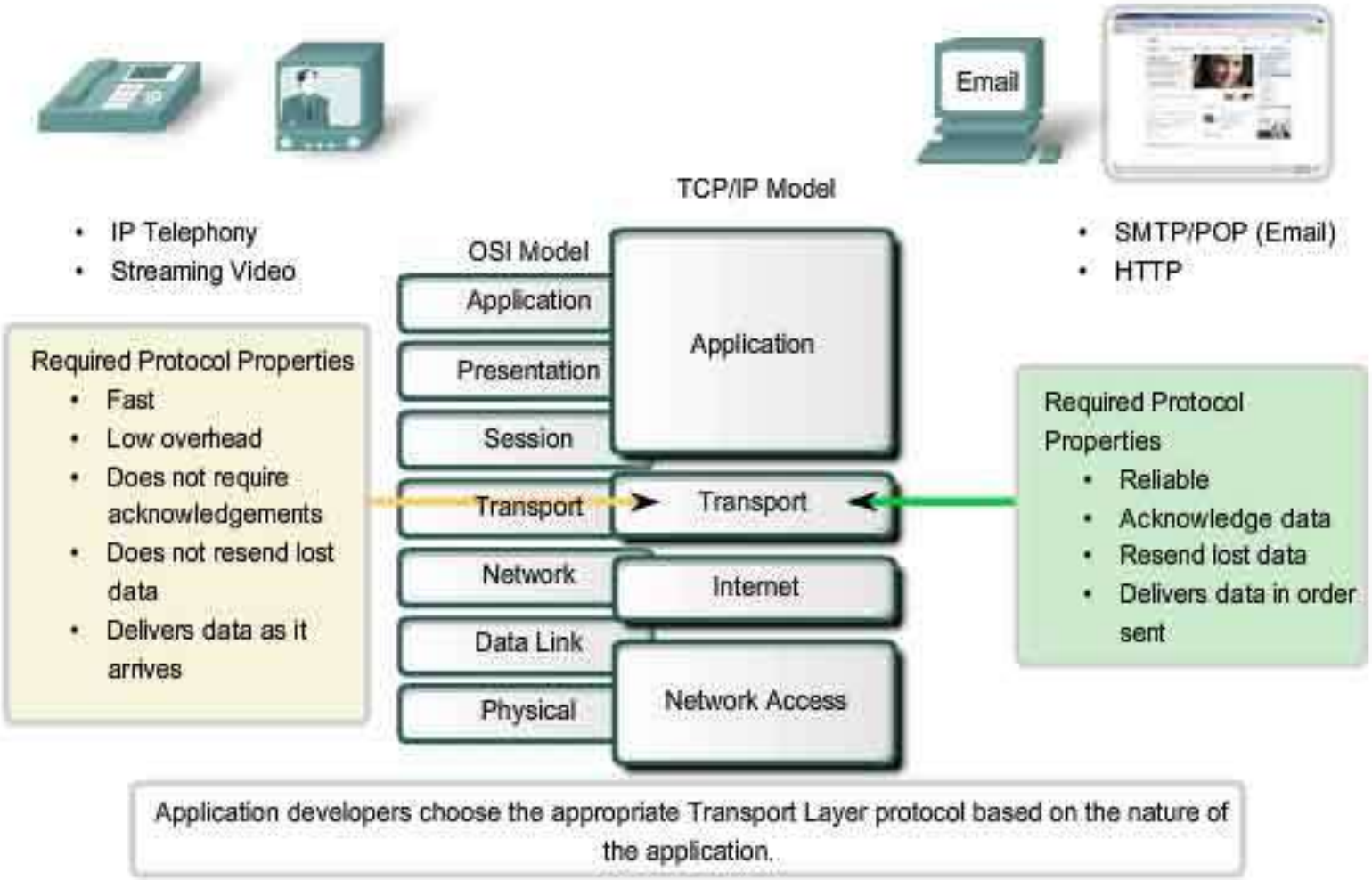
cngst_wndw starts with value 1, then increments every acknowledged segment

Dynamic window sizing on congestion

A possible scenario: When first segment lost, means collision sign, so reset value of cngst_wndw to 1 and begin ‘slow start’

There are more others ….

Transport Layer Protocols



User Datagram Protocol (UDP)

Connectionless

Less overhead

Used mostly for

real-time applications (voice, video, telemetry), with no need for retransmissions

non-critical functions: inward data collections (monitoring …), outward data collections (broadcasted announcements …)

Specified by RFC 768

Unreliable

Delivery and duplication control not guaranteed

UDP delivers independent messages, called *datagrams* between applications or processes on host computers

`Best effort' delivery - datagrams may be lost, delivered out of order, etc.

Checksum (optionally) guarantees integrity of data

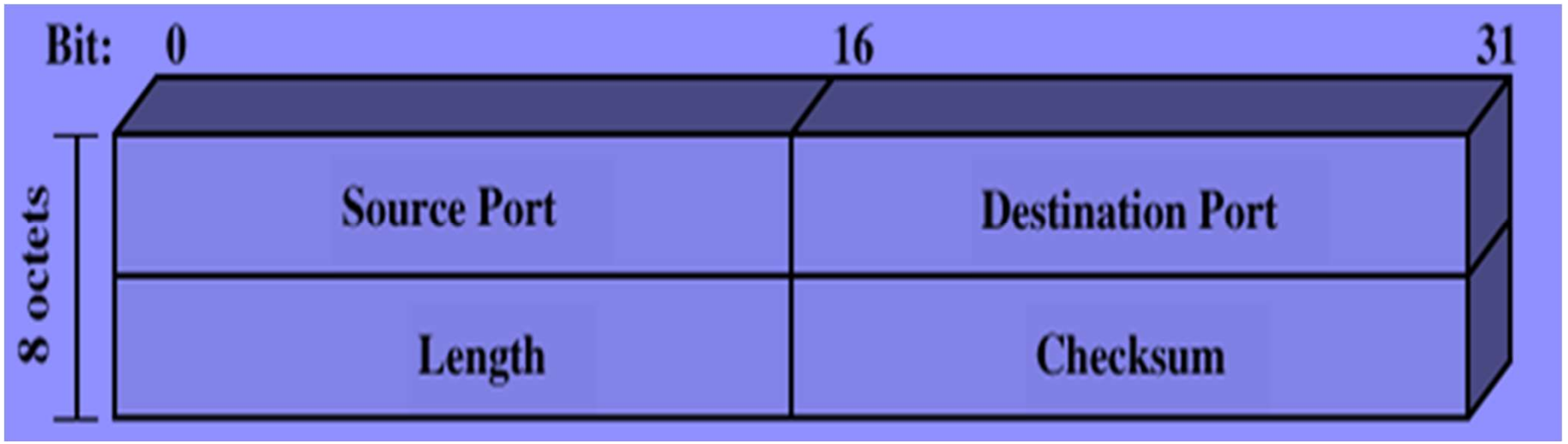
For generality, endpoints of UDP are called *protocol ports* or *ports*

Each UDP data transmission identifies the internet address and port number of the destination and the source of the message (port, socket … as TCP)

UDP header is very simple:

- Port numbers
- Message length
- Checksum (optional, if yes, use same 1s complement checksum as IP)

UDP Header



Introduction to Internetworking

Introductory terms

Communications Network

Facility that provides data transfer services

An internet

Collection of communications networks interconnected by bridges and/or routers

The Internet - note upper case I

The *global collection* of thousands of individual machines and networks

Intranet

Corporate internet operating within the organization

Uses Internet (TCP/IP and http) technology to deliver documents and resources

End System (ES)

Device attached to one of the networks of an internet

Supports end-user applications or services

Intermediate System (IS)

Device used to connect two networks

Permits communication between end systems attached to different networks

Bridge

IS used to connect two LANs using similar LAN protocols

Address filter passing on packets to the required network only

OSI layer 2 (Data Link)

Router

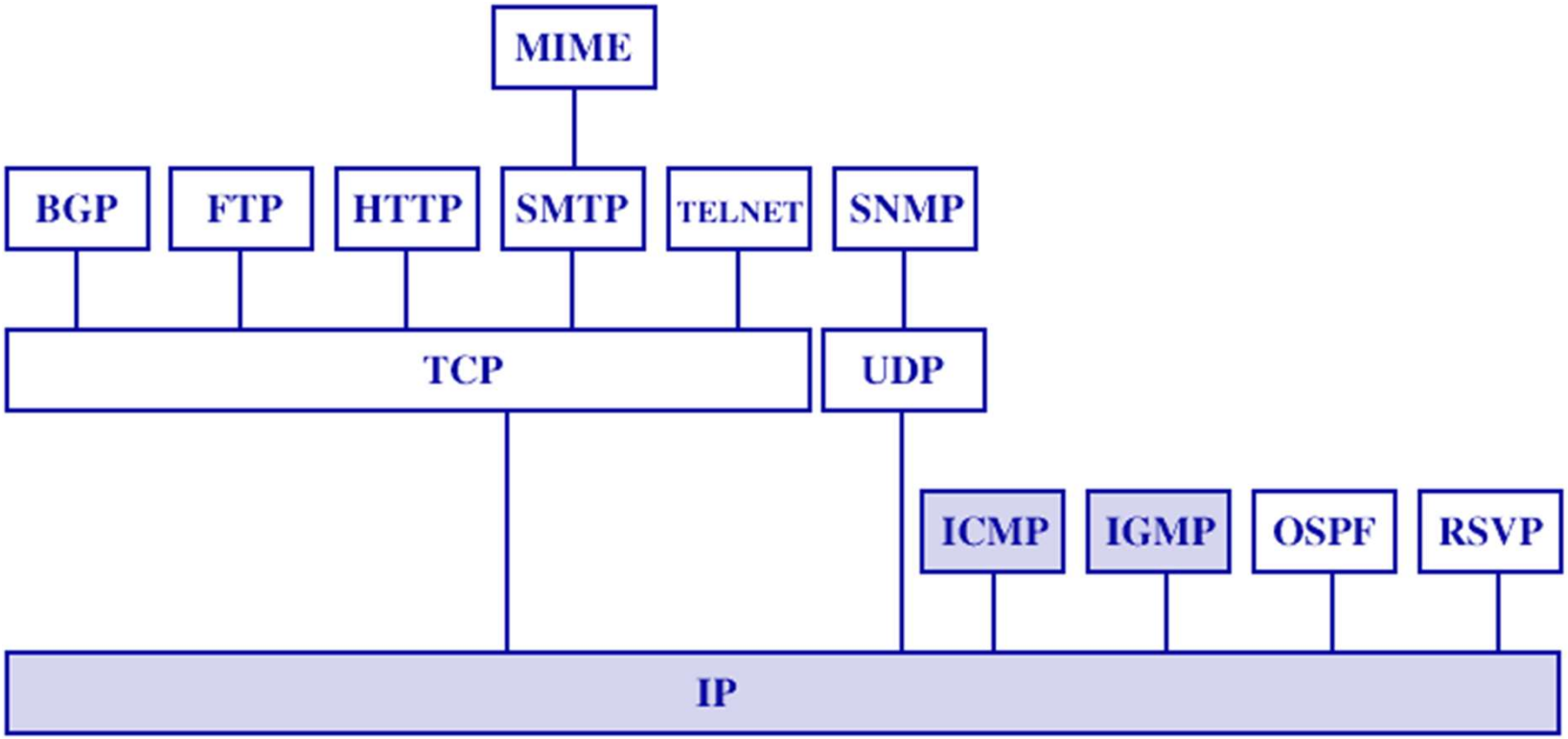
Connects two (possibly dissimilar) networks

Uses internet protocol present in each router and end system

OSI Layer 3 (Network)

Internetworking Protocols

TCP/IP stack and suite of internetworking protocols



Requirements of Internetworking

Link between networks

Minimum physical and link layer

Routing and delivery of data between processes on different networks

Accounting services and status info

Independent of network architectures

Network Architecture Specific Features

Addressing

Packet size

Access mechanism

Timeouts

Error recovery

Status reporting

Routing

User access control

Architectural Approaches

Connection oriented

Connectionless

Connection Oriented

Assume that each network is connection oriented

IS connect two or more networks

- IS appear as DTE to each network
- Logical connection set up between DTEs
  - Concatenation of logical connections across networks
- Individual network virtual circuits joined by IS

May require enhancement of local network services

- 802, FDDI are datagram services

Connection Oriented IS Functions

Relaying

Routing

e.g. X.75 used to interconnect X.25 packet switched networks

Connection oriented not often used

Connectionless Operation

Corresponds to datagram mechanism in packet switched networks

Each NPDU treated separately

Network layer protocol common to all DTEs and routers

- Known generically as the internet protocol

Internet Protocol

- One such internet protocol developed for ARPANET
- RFC 791
- Lower layer protocol needed to access particular network

Connectionless Internetworking

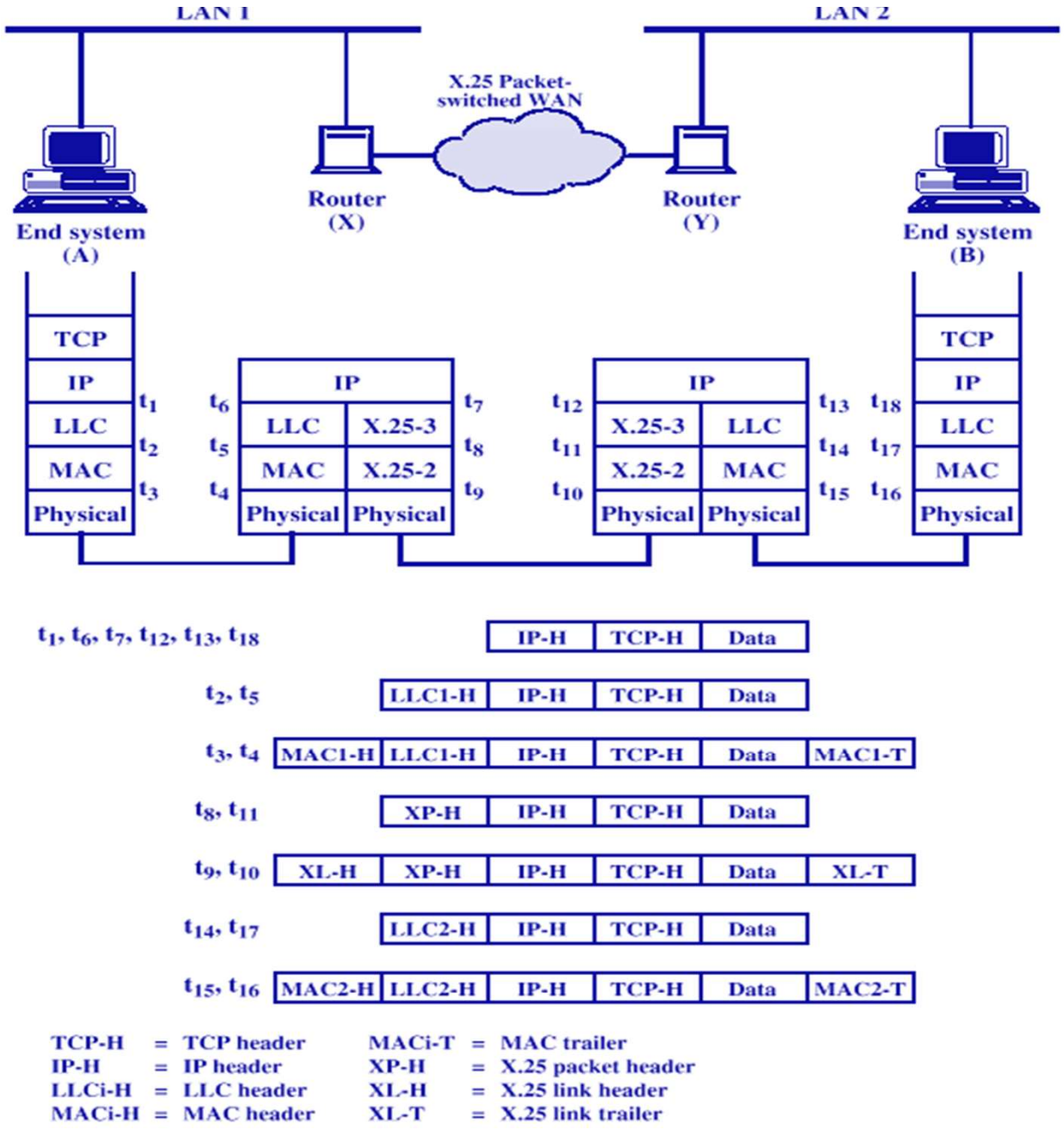
Advantages

- Flexibility
- Robust
- No unnecessary overhead

Main drawback: Unreliable

- Not guaranteed delivery
- Not guaranteed order of delivery
  - Packets can take different routes
- Reliability is responsibility of next layer up (e.g. TCP)

Example of an IP protocol operations, acting over a X.25 packet switched WAN network



Design Issues

- Routing
- Datagram lifetime
- Fragmentation and re-assembly
- Error control
- Flow control

Routing

End systems and routers maintain routing tables

- Indicate next router to which datagram should be sent
- Static
  - May contain alternative routes
- Dynamic
  - Flexible response to congestion and errors

Source routing

- Source specifies route as sequential list of routers to be followed
- Security
- Priority

Route recording

Datagram Lifetime

Datagrams could loop indefinitely

- Consumes resources
- Transport protocol may need upper bound on datagram life

Datagram marked with lifetime

- Time To Live field in IP
- Once lifetime expires, datagram discarded (not forwarded)
- Hop count
  - Decrement time to live on passing through a each router
- Time count
  - Need to know how long since last router

Fragmentation and Re-assembly

Different packet sizes

When to re-assemble

- At destination
  - Results in packets getting smaller as data traverses internet
- Intermediate re-assembly
  - Need large buffers at routers
  - Buffers may fill with fragments
  - All fragments must go through same router
    - Inhibits dynamic routing

IP re-assembles at destination only

Uses fields in header

Data Unit Identifier (ID)

- Identifies end system originated datagram
  - Source and destination address
  - Protocol layer generating data (e.g. TCP)
  - Identification supplied by that layer

Data length

- Length of user data in octets

Offset

- Position of fragment of user data in original datagram
- In multiples of 64 bits (8 octets)

More flag

- Indicates that this is not the last fragment

Dealing with Failure

Re-assembly may fail if some fragments get lost

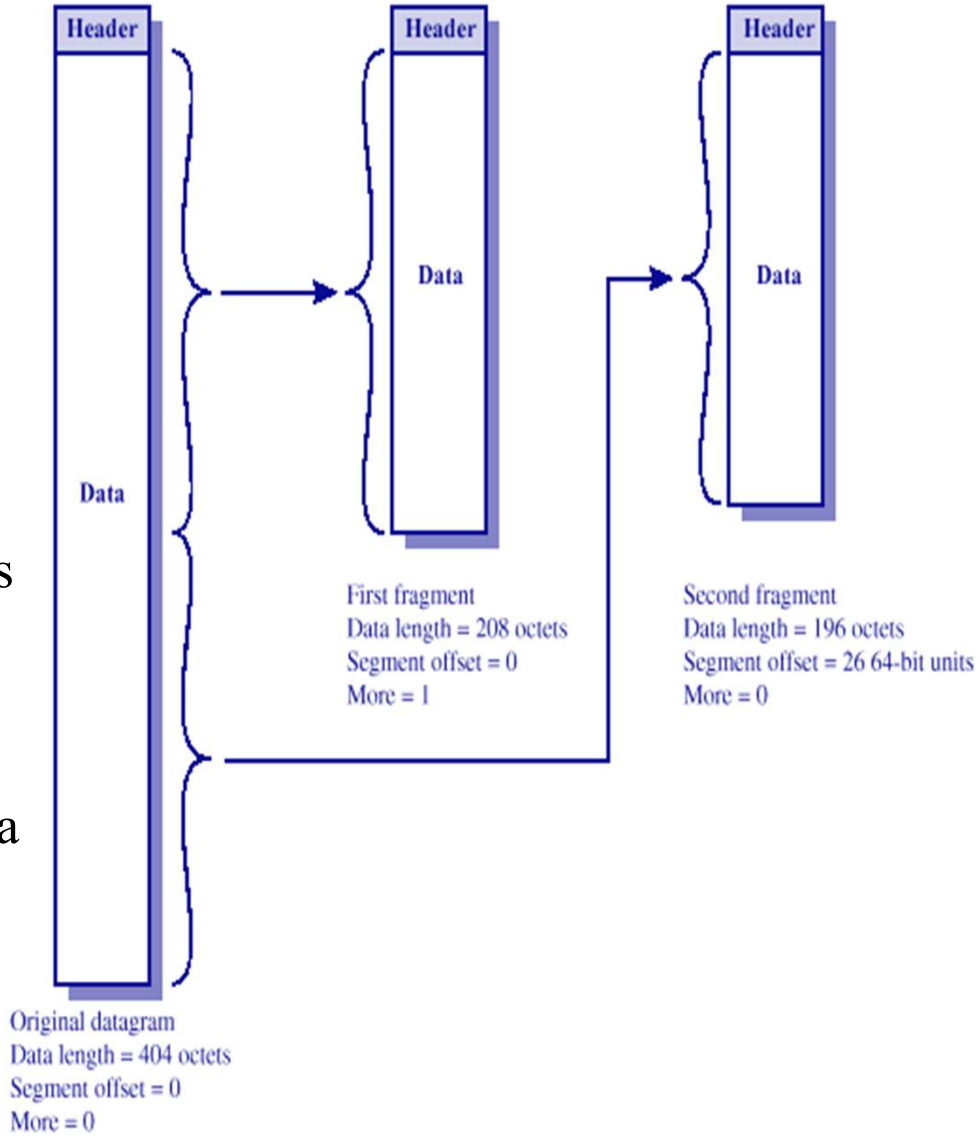
Need to detect failure

Re-assembly time out

- Assigned to first fragment to arrive
- If timeout expires before all fragments arrive, discard partial data

Use packet lifetime (time to live in IP)

- If time to live runs out, kill partial data



Error Control

Not guaranteed delivery

Router should attempt to inform source if packet discarded

e.g. for time to live expiring

Source may modify transmission strategy

May inform high layer protocol

Datagram identification needed

(Look up ICMP)

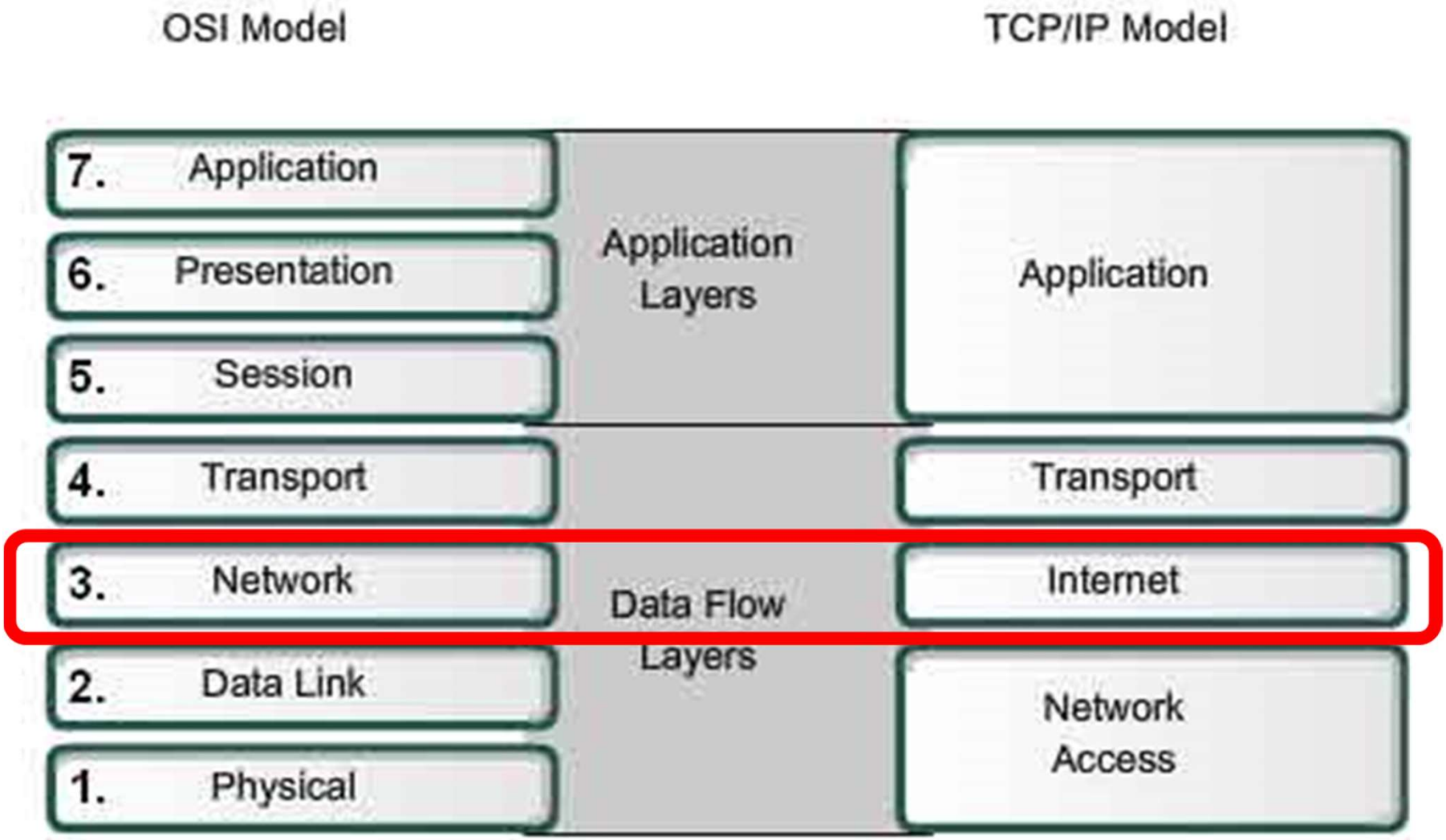
Flow Control

Allows routers and/or stations to limit rate of incoming data

Limited in connectionless systems

Send flow control packets

Requesting reduced flow, e.g. ICMP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The OSI Model (Open Systems Interconnection)