

1) Is fragmentation needed in concatenated virtual-circuit internet or only in datagram systems? Explain

ans) It is needed in both, even in a concatenated virtual-circuit networks along the path might accept 1024-byte packets and others might only accept 48 byte packets fragmentation is still needed.

2) Suppose that host A is connected to a router R1, R1 is connected to another router R2 and R2 is connected to host B

a) Suppose that a TCP message that contains 900 bytes of data & 20 bytes of TCP header is passed to IP code at host A for delivery to B

line A-R1

length=940

ID=x

DF=0

offset=0

b) show the total length, identification, DF, MF and fragmentation offset fields of the IP header in each packet transmitted over the three links.

link R1-R2

length=500

ID=x

DF=0

MF=1

offset=60

length=460

ID=x

DF=0
MF=0
offset=60

- c) Assume that
- link A-R1 can support a max frame size of 1024 bytes including 14 byte frame header
 - link R1-R2 can support a max frame size of 512 bytes including an 8 byte frame header
 - link R2-B support a max frame size of 512 bytes including an 12byte frame header

link R2-R

length=500
ID=x
DF=0
MF=1
offset=0

length=460
ID=x
DF=0
MF=0
offset=60

3) A computer on a 6MBPS network is regulated by a token bucket. The token bucket is filled at rate of 1MBPS. It is initially filled to capacity with 8MBPS. How long can the computer transmit at the full 6MBPS.

ANS) Now tokens are added at the rate of r bytes/sec which is 1MBPS in the given question

Capacity of the token bucket(b) = 8MBPS

Max possible transmission rate(m) = 6MBPS

Max burst time = $b/(m-1)$

$$= 8/(6-1)$$

$$= 8/5$$

$$= 1.6 \text{ sec}$$

In the above formula, 1 to the power of r is subtracted from M to calculate the max burst time. The reason for this subtraction new token are added at the rate of r while transmission happens at max transmission rate m

4) Find the subnet network address for the following

s.no	IP a/s	MASK
a)	141.181.14.16	255.255.224.0
b)	200.34.22.156	255.255.255.240
c)	125.35.12.57	255.255.0.0

ANS) a) subnetwork a/s

IP a/s	141.181.14.16	0000 1110
MASK	<u>255.255.224.0</u>	<u>1110 0000</u>
	141.181.0.0	0000 0000

b)

IP a/s	200.34.22.156	1001 1100
MASK	<u>255.255.255.240</u>	<u>1111 0000</u>
	200.34.22. 144	1001 0000

c)

IP a/s	125.35.12.57
MASK	<u>255.255.0.0</u>
	125.35.0.0

5) Give examples of applications where it is good to use UDP and TCP respectively

ANS) UDP: Anything where you don't care too much if you get all data always

- Tunneling/VPN (lost packets are ok - the tunneled protocol takes care of it)
- Media streaming (lost frames are ok)
- Games that don't care if you get *every* update
- Local broadcast mechanisms (same application running on different machines "discovering" each other)

TCP: Almost anything where you have to get all transmitted data

- Web
- SSH, FTP, telnet
- SMTP, sending mail
- IMAP/POP, receiving mail

6) The following is a dump of a UDP header in hexadecimal form : 06 32 00 0D 00 1C E2 17. What is the

- a) source port number
- b) Destination port number
- c) Total length of the UDP
- d) Length of the data

ANS) The UDP header has four parts, each of two bytes. That means we get the

following interpretation of the header.

(a) Source port number = $0632_{16} = 1586$

(b) Destination port number = $000D_{16} = 13$

(c) Total length = $001C_{16} = 28$ bytes

(d) Since the header is 8 bytes the data length is $28 - 8 = 20$ bytes.

7) If the TCP round - trip time, RTT, is currently 30 msec and the following acknowledgements come in after 26, 32 and 24 msec, respectively, what is the new RTT estimate using the Jacobson algorithm? Use $\alpha=0.1$.

ANS) Estimated RTT1

$$= 0.9 * 30 + 0.1 * 26 = 29.6$$

Estimated RTT2

$$= 0.9 * 29.6 + 0.1 * 32 = 29.84$$

Estimated RTT3

$$= 0.9 * 29.84 + 0.1 * 24 = 29.256$$

9) If packets on a certain network can carry a maximum of only 500 bytes in the data portion. An application using TCP/IP on a node on this network generates a TCP segment with 1000 bytes in the data portion. How many IP packets are transmitted to carry this TCP segment, and what are their sizes?

ANS) The TCP segment is of size 1,020 bytes. Therefore, we will need 3 IP packets of sizes 520, 520 and 40 bytes.

10) Write notes on the following:

- a) Anycast Routing
- b) Routing in adhoc network
- c) RTP Header
- d) Time Wait timer

ANS) a) **Anycast** is a network addressing and routing methodology in which datagrams from a side sender are routed to the topologically nearest node in a group of potential receivers, though it may be sent to several nodes, all identified by the same destination address

Applications

With the growth of the Internet, network services increasingly have high-availability requirements. As a result, operation of anycast services has grown in popularity among network operators.

Domain Name System

Nearly all Internet root name servers are as clusters of hosts using anycast addressing. 12 of the 13 root servers A-M exist in multiple locations, with 11 on multiple continents. (Root server H exists in two U.S. locations. Root server B exists in a single location in the Los Angeles Area.) The 12 servers with multiple locations use anycast address announcements to provide a decentralized service. This has accelerated the deployment of physical (rather than logical) root servers outside the US. RFC 3258 documents the use of anycast addressing to provide authoritative DNS services. Many commercial DNS providers have switched to an IP anycast

environment to increase query performance, redundancy, and to implement load balancing.

IPv6 transition

In IPV4 to IPV6 transitioning anycast addressing may be deployed to provide IPv6 compatibility to IPv4 hosts. This method, 6 to 8, uses a default gateway with the IP address 192.88.99.1, as described in RFC 3068. This allows multiple providers to implement 6 to 4 gateways without hosts having to know each individual provider's gateway addresses.

Content delivery networks

Content delivery networks may use anycast for actual HTTP connections to their distribution centers, or for DNS. Because most HTTP connections to such networks request static content such as images and style sheets, they are generally short-lived and stateless across subsequent TCP sessions. The general stability of routes and statelessness of connections makes anycast suitable for this application, even though it uses TCP.

b)

• Characteristics

- Dynamic topology
- Links are low bandwidth, variable capacity, sometimes unidirectional
- Limited battery power and other resources in the nodes
- More route alternatives (every node is a router)

• Typical applications

- Military environments (soldiers, tanks, planes)
- Emergency and rescue operations

- Meeting rooms
- Personal area networking, e.g. Bluetooth
- Wireless home networking
- Special applications (industrial control, taxis, boats)

- **Advantages**

- Only required routes are maintained

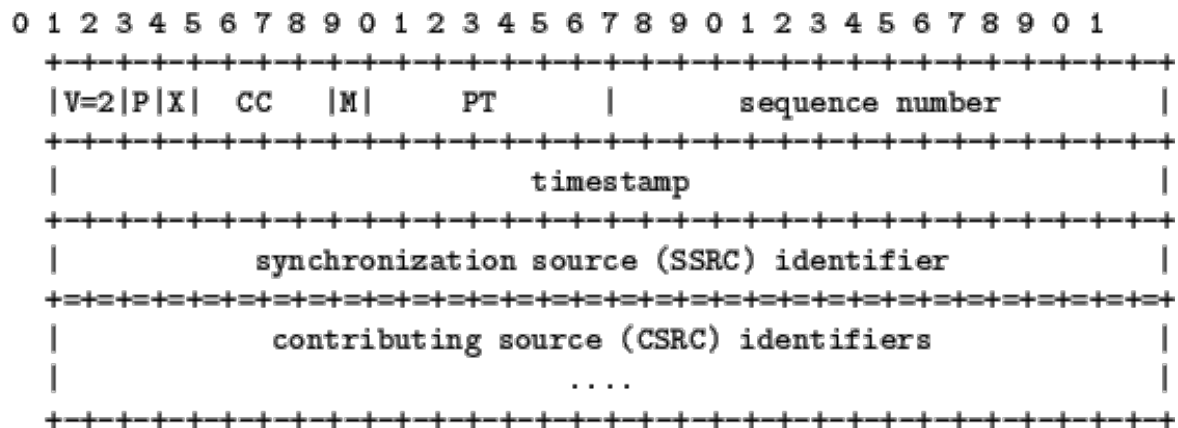
- **Disadvantages**

- Delay before the first packet can be sent
- Route discovery usually involves flooding

Routing protocols in adhoc n/w

- Many routing protocols have been proposed
 - Both proactive and reactive
 - Some protocols adapted from wired networks, some invented for mobile ad hoc networks
- No single protocol works
 - Attempts to combine different solutions, e.g. adaptive and combinations of proactive and reactive protocols
- Standardization in IETF
 - MANET (Mobile Ad hoc Network) working group
- Currently considered routing protocols: DSR, AODV, OLSR, TBRPF
 - MobileIP

c)



The first twelve octets are present in every RTP packet, while the list of CSRC identifiers is present only when inserted by a mixer.

version (V)

: 2 bits

This field identifies the version of RTP. The version defined by this specification is two (2).

padding (P)

: 1 bit

If the padding bit is set, the packet contains one or more additional padding octets at the end which are not part of the payload.

extension (X)

: 1 bit

If the extension bit is set, the fixed header is followed by exactly one header extension, with a format defined in Section 5.2.1.

CSRC count (CC)

: 4 bits

The CSRC count contains the number of CSRC identifiers that follow the fixed header.

marker (M)

: 1 bit

The interpretation of the marker is defined by a profile. It is intended to allow significant events such as frame boundaries to be marked in the packet stream.

payload type (PT)

: 7 bits

This field identifies the format of the RTP payload and determines its interpretation by the application.

sequence number

: 16 bits

The sequence number increments by one for each RTP data packet sent, and may be used by the receiver to detect packet loss and to restore packet sequence

timestamp

: 32 bits

SSRC

: 32 bits

The SSRC field identifies the synchronization source.

CSRC list

: 0 to 15 items, 32 bits each

The CSRC list identifies the contributing sources for the payload contained in this packet. The number of identifiers is given by the CC field. If there are more than 15 contributing sources, only 15 may be identified. CSRC identifiers are inserted by mixers, using the SSRC identifiers of contributing sources.

d)

The time wait(2msl) timer is used during connection termination

- The max segment life time(MSL) is amount of time any segment can exist in a n/w before being discarded

- The implementation needs to choose or value for MSL common values are 30 sec or 1 min or even 2 min
- The 2MSL timer used when TCP performs an active close and sends the final ACK in the ACK is cost. This requires that the RTO timer at the other end timer out and new FIN & ACK segments are resent.