

**Birla Institute of Technology & Science, Pilani**  
**Department of Computer Science and Information Systems**  
**First Semester 2017-2018**  
**Advanced Computer Networks (CS G525)**  
**MID SEMESTER TEST (REGULAR) CLOSE BOOK**

**Duration: 1.5 Hrs**

**Date: 10/10/2017 (Pilani & Hyd)**

**MM: 20**

**Note: Answer sub parts of a question (if any) at one place in sequence.**

Q.1 Answer the following questions:

a) According to the end-to-end principle of system design, local implementations (i.e. lower layer implementation) of functions may enhance performance above that achievable using end-to-end implementations alone. Describe the performance benefits of localized implementation of “error control” and “ensuring delay requirements” functions with suitable examples.

b) Name Data Networking (NDN) architecture provides support to observe data plane performance whereas IP architecture does not. Justify this argument.

c) How is a TCP connection is differ from a Virtual Circuit (VC) connection? **[2+1+1=4M]**

Q.2 a) Consider the following scheduling schemes for packet level Fair Queuing (FQ) algorithm:

i) Send the packet which has the largest finish time.

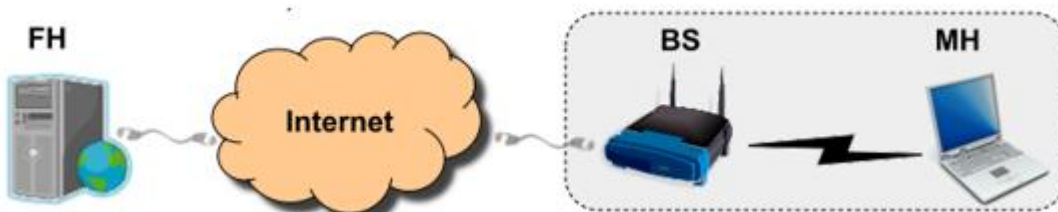
ii) Send the packet which has the smallest start time.

iii) Send the packet which has smallest finish time.

Scrutinize these scheduling schemes for achieving fairness and delay guarantees.

b) Suppose a buffer of total length 100 packets is organized as 4 queues that are serviced by FQ algorithm. Assume the buffer is full, the service rate is 5 packets/sec, and that no new packets arrive to the buffer. What is the maximum queuing delay for any packet? How does this change if we use Weighted-FQ and the queues have weights 1, 2, 3, 4 respectively? **[3+1=4M]**

Q.3 Consider a TCP connection between Mobile Host (MH) and a Fixed Host (FH) as shown in the Fig. 1. The MH is sender and FH is receiver. Suggest a suitable solution to shield TCP sender (i.e. MH) from the packet losses on wireless link. Write any limitations of your solution (if any). **[3M]**

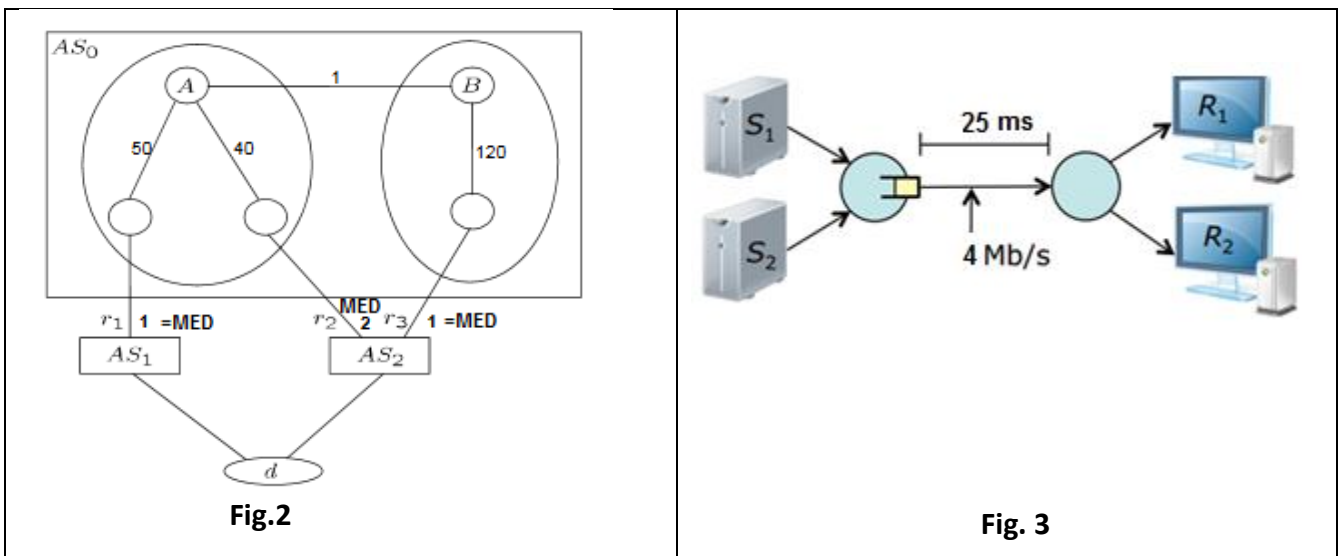


**Fig. 1**

Q.4 a) How does a MPTCP flow differ from a single TCP flow that is splitted into multiple subflows and by using ECMP routing these subflows are transmitted over different paths? Write your observations from the traffic distribution and link utilization point of view.

b) Let's assume a single path TCP user with 100 Mb/s WiFi access link, who then adds a 10 Mb/s 4G access link. What throughput should this user now gets with MPTCP connection. Assume semi coupled congestion control algorithm is used here. You can also assume there is no other competing traffic flow exists. [1.5 + 1.5=3M]

Q.5 Consider the AS topology with three ASes (i.e.  $AS_0$ ,  $AS_1$  and  $AS_2$ ) as shown in Fig. 2. The  $AS_0$  is internally divided into two sibling ASes having BGP routers A and B. There are two routes (i.e.  $r_1$  and  $r_2$ ) are available from A to reach prefix d. Similarly, route  $r_3$  is available from B to reach prefix d. The MED value announced by  $AS_1$  for route  $r_1$  is 1. Similarly,  $AS_2$  announced MED values 2 and 1 for routes  $r_2$  and  $r_3$  respectively. The nos. 40, 50, 1 and 120 represent IGP cost of respective links in the  $AS_0$ . Routers A and B import and export routes as per BGP policies. Will there be route oscillation happens for prefix d? Explain. [3M]



Q.6 Two TCP senders  $S_1$  and  $S_2$  and the corresponding receivers  $R_1$  and  $R_2$  are shown in Fig. 3. Both senders use TCP Reno. Assume that the one-way propagation delay for both connections is 25 ms and the link joining the two routers has a bandwidth of 4 Mbit/s. Let  $cwnd_1$  and  $cwnd_2$  be the values of the senders' congestion windows. Maximum Segment Size (MSS) is 1KB for both the connections.

- What is the smallest amount of Bytes that keeps the link busy all the time?
- Assume that the link buffer overflows whenever  $cwnd_1 + cwnd_2 \geq 36\text{KB}$  and that time,  $cwnd_1 = 12\text{KB}$  and  $cwnd_2 = 24\text{KB}$ . What are the values of  $cwnd_1$  and  $cwnd_2$  one RTT later? Assume that all packet losses are detected by a triple duplicate ack.
- How many RTTs pass before  $cwnd_1 + cwnd_2 = 36\text{KB}$  again? What are the values of  $cwnd_1$  and  $cwnd_2$  at this point? [1+1+1 = 3M]

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