

# Major Exam, COL 334/672

November 21, 2016, Duration: 2 hours, Maximum Marks: 72

1. (12 marks) Explain the following terms in the context of TCP Reno's congestion control protocol. For each point below, explain how the congestion window is modified, and also the intuition about why the window is changed in this way. If ss\_thresh is also modified, then state how it is changed and give the reasons for why it is modified in that way.
  - (a) (3 marks) Slow start phase
  - (b) (3 marks) Additive Increase in congestion avoidance phase
  - (c) (3 marks) Multiplicative decrease on receipt of 3 DUP ACKs
  - (d) (3 marks) Multiplicative decrease on timeout.

2. (8 marks) Consider Figure 1. Assume that all routers are BGP speakers. All routers and hosts in AS1 and AS2 have IP addresses matching with the prefixes shown. AS-1 hears BGP advertisements about prefix 132.14.0.0/16 from r8, r9, and r15. Similarly AS-2 hears BGP advertisements about prefix 124.17.0.0/16 from r1, r2, and r16. The nodes C and D are hosts connected to routers in their own ASes.

- (a) (2 marks) How can AS-2 recommend to AS-1, using BGP attributes, the use of r1 - r8, over r2 - r9 for traffic destined for prefix 132.14.0.0/16 coming from AS-1?
- (b) (2 marks) Despite the recommendation from AS-2 to use r1 - r8, how can AS-1 ensure, using appropriate BGP attributes, that packets sent from C to D use the link r6 - r15?
- (c) (4 marks) Suppose the attributes for BGP announcements regarding prefix 124.17/16 (in AS-2) are such that r10 adopts hot-potato routing for packets with destination IP address matching 124.17/16. Assume that the IGP cost (metric for intra-domain routing protocol) is 1 for every link depicted, except for links r8 - r10 and r9 - r11 which have costs 5 and 3 respectively. What path will traffic from host D to host C take? Why does it take this path?

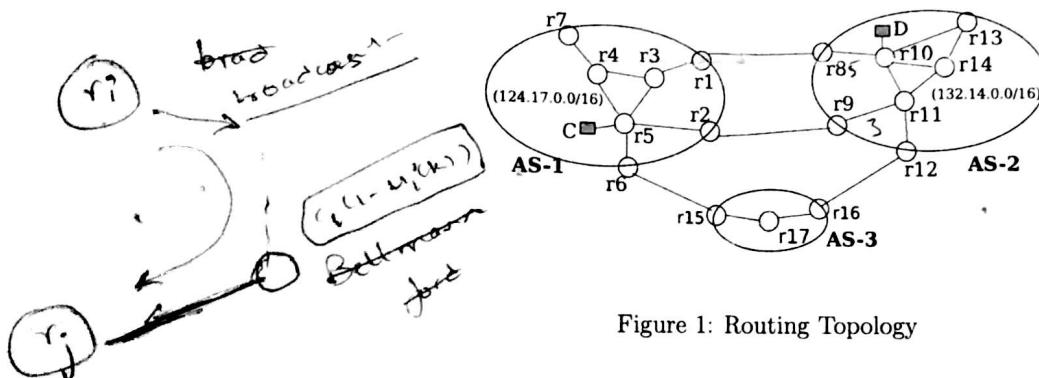


Figure 1: Routing Topology

3. (8 marks) Explain the following about the Domain Name System (DNS).
  - (a) (2 marks) How many logical DNS root servers are there in the Internet and why was this number chosen?
  - (b) (2 marks) In DNS resource records what does Type field equal to A, NS, CNAME, and MX mean?
  - (c) (4 marks) Suppose a host at Stanford does a DNS lookup for sri.cse.iitd.ernet.in and its local DNS server at Stanford has IP addresses only of DNS root servers and nothing more. Assume that each DNS server in the DNS hierarchy only has resource records for DNS servers at the next lower level, and that the IP address of sri.cse.iitd.ernet.in is stored only in the DNS server of cse.iitd.ernet.in. Explain in your own words, with suitable diagrams if necessary, how the local Stanford DNS server performs an iterative search to resolve the IP address of sri.cse.iitd.ernet.in.

**Note:** The following questions do not have unique solutions. Be creative and state any assumptions/approximations you make. More elegant solutions will get more marks.

4. (14 marks) Consider a single autonomous system consisting of several IP routers  $R_1, R_2, \dots, R_n$ . These routers are connected to each other via point-to-point wired links and form an arbitrary network topology. The links are bidirectional and have arbitrary link bandwidths. Any router,  $R_i$ , may generate a "flow" of UDP data (set of packets) destined for some other router  $R_j$ . Such a flow can start and terminate at arbitrary time instants. The data travels across the network from source ( $R_i$ ) to destination ( $R_j$ ) using an intra-domain routing protocol which you must design.

We define available bandwidth of a path as follows. Suppose a path consists of  $m$  links with link speeds  $c_l$  ( $l = 1, 2, \dots, m$ ). For each link  $l$ , in time interval  $[k\tau, (k+1)\tau]$  where  $k = \dots, -1, 0, 1, \dots$  and  $\tau$  is a time unit set by the network administrator, we compute the utilization (fraction of link speed used to transmit data) and denote it by  $u_l(k)$ . For example, if  $\tau = 10$  seconds, and for link  $l$ , between time 30 and 40 seconds, only 30% of the link speed is used for transmitting data then  $u_l(3) = 0.3$ . Now the available bandwidth of the path in interval  $[k\tau, (k+1)\tau]$  is defined as,

$$\min_l c_l (1 - u_l(k))$$

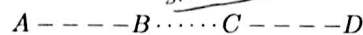
Design an intra-domain routing protocol which ensures that any flow travels across a path which has the maximum available bandwidth among all paths between source and destination. Note that this path may change over time as available bandwidth itself changes. You may use any of the ideas discussed in class.

Your solution must clearly state:

- (2 marks) What information about utilization is computed by each router in the network over time.
- (4 marks) How this information is shared (or proactively obtained) by other nodes in the network. You can create your own protocol (running on top of IP) and message types for this. State what each message in your protocol will contain.
- (8 marks) How the best path is identified and used for routing each flow.

5. (14 marks) The TCP protocols we considered in class assume that packet drops occur always due to network congestion, that is, due to queues filling up in the network etc. This assumption may not hold in wireless networks where significant packet loss can occur due to reasons other than congestion, such as bad wireless channel properties.

Consider the network topology below. Links  $A - B$  and  $C - D$  are wired with negligible bit error rates. Link  $B - C$  is a wireless link with a high bit error rate.



We assume that because of this high bit error rate, a packet transmitted from  $B$  arrives "corrupted" at  $C$ , with non-negligible probability,  $p$ , and is then discarded by the link layer protocol at  $C$ . Assume that consecutive packet transmissions on this link face corruption "independent" of each other, and that the link layer protocol does not retransmit corrupted packets (i.e. there is no ARQ). Further, all nodes employ a drop-tail FIFO queuing mechanism.

Suppose  $A$  transmits an infinitely large data set to  $D$  using TCP-Vegas. Assume no other data traffic exists on the network.

- (2 marks) How will TCP's throughput change as  $p$  increases (a qualitative answer will do)? Explain why.
- (12 marks) Make modifications to TCP-Vegas to reduce the negative effect on throughput of packet losses caused due to bit-error on the wireless link. Your new protocol must not require any explicit modification at (or information from) intermediate routers. It must intelligently try to distinguish if a packet loss is due to FIFO queue overflow (droptail) or if the loss was due to bit-error on the wireless link, and modify its window appropriately.

6. (16 marks) A peer-to-peer file-sharing network has the following properties.

- Each file is stored (replicated) in a maximum of 5 peer nodes.
- Unlike Napster, this application uses no centralized server that stores the location(s) of different files.
- Any peer (person wanting a file) may download a chunk (i.e. a portion) of a file using TCP-Reno from another peer who has this file. He can specify the starting and ending byte of the chunk he wants. For example, he can state that he wants a particular file from byte 1200 to byte 1999. We assume byte 0 represents the first byte of a file.
- Every client wishes to download a file from his peers as fast as possible. Simultaneous downloads from multiple peers is allowed.
- The peer may not know the exact bit-rate in the near future for data transfer from another peer to itself but can predict this from recent downloads from that peer.
- Nodes may join and leave the P2P network.

Design such a P2P network and describe briefly the following aspects of your design. State any assumptions you make clearly.

- (2 marks) How does a node join the P2P network?
- (2 marks) How is a given file stored in multiple locations?
- (2 marks) How does a client wanting a file determine which peer(s) possess it?
- (2 marks) How is the P2P network robust to node failure or node leaving the network?
- (8 marks) How does a client minimize the total time to obtain a particular file?