上 海 交 通 大 学 试 卷（A卷答案）

（ 2019– 2020第一学期 ）

班级号：_____ **学号:** _____ **姓名:** _____

课程名称: Computer Networking **成绩:** _____

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1. (8 x 5 points) First judge the following statements true or false and then justify your answer.

(a) The Internet architecture puts much of its complexity at the core routers instead of the end systems.

False. The Internet architecture puts much of its complexity at the end systems instead of the core routers.

(b) Applications using UDP service are always unreliable.

False. Applications using UDP can implement RDT itself, e.g. DNS.

(c) When N users connect to the same www server, there will be N different sockets created at N different port numbers at the www server.

False. www server will create N different sockets for N users but with same port number.

(d) TCP will crowd out UDP when congestion occurs.

False. TCP has congestion control but UPD hasn't. So UDP will crowd out TCP when congestion occurs.

我承诺，我将严格遵守考试纪律。

承诺人：_____

题号	1	2	3	4	5	6	7
得分							
批阅人(流水阅卷教师签名处)							

(e) A protocol is always implemented in software.

False. Most link layer protocol is implemented in software and hardware.

(f) DVR has count-to-infinity problem because it uses the amount of carried traffic as the link cost, i.e. load sensitive.

False. DVR has count-to-infinity problem because it uses local information to update the routing table.

(g) OSPF uses LSR and Hierarchical Routing to find the least-cost path, whereas BGP uses DVR, together with local policies, to find a good path instead of best path, with explicit AS path to avoid count-to-infinity problem.

True. OSPF is intra-AS routing, so it's performance dominant, whereas BGP is inter-AS routing, it's policy dominant.

(h) Within block of IP addresses 202.120.30.0 ~ 202.120.43.255, 202.120.30.0 ~ 202.120.33.0 could be assigned to a company in need of 1000 IP addresses.

False. With 202.120.30.0 ~ 202.120.33.0 assigned, it couldn't use CIDR to aggregate routing table. 202.120.32.0 ~ 202.120.35.0 prefix is recommended.

2. (10 points) Please describe TCP/IP model and explain the hourglass philosophy of the TCP/IP model. And explain the single technology in the definition of “Collection of autonomous computers interconnected by a single technology”

Solutions:

(4 Points) Most network software are organized as a stack of layers or levels, each one built upon the one below it. TCP/IP model has 4 layers: Application layer includes HTTP, FTP, SMTP protocols, Application layer includes TCP and UDP protocols, Network layer includes IP, ICMP and Routing protocols, and Link layer includes different kinds of networks includes Ethernet, WiFi, IBM Toke Ring etc.
(4 points) The “Hourglass” philosophy of Internet is: IP bridges different applications over different networks. If everybody just supports IP, can use many different applications over many different networks.
(2 points) The single technology in the definition of “Collection of autonomous computers interconnected by a single technology” means IP protocol.

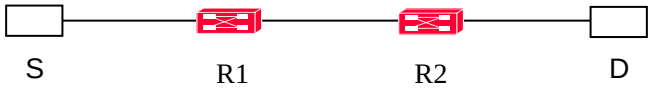
3. (10 points) Please explain the Reliable Data Transfer and Media Access Control methods for Ethernet and WiFi respectively.

Solutions:

(2 points) Ethernet RDT: Ethernet is unreliable and connectionless. It doesn’t have flow control, it uses CRC to detect bit errors but don’t support ACK and retransmission.
(3 points) Ethernet MAC: CSMA/CD. When frame ready, sense the channel: If it’s idle for 96 bit times, transmit, do collision detection; If it’s busy, wait until idle. When collision detected, abandon and send a 48-bit jam signal, then use binary exponential backoff to retransmit.

(2 points) WiFi RDT: reliable, using stop and wait to do flow control, and using ARQ to do error control.
(3 points) WiFi MAC: DCF + PCF. All 802.11 implementations must support DCF but PCF is optional. For DCF, it uses CSMA/CA. CSMA/CA uses exchange of short messages RTS/CTS to do collision avoidance. When a potential sender sends RTS to potential receiver to request to communicate, the receiver will response a CTS to confirm the approval. At the same time, all the stations within receiver’s range will receive CTS and will keep silence during the transmission, thus solve the hidden terminal problem.

4. (10 points) Consider a transport of a file from source S to destination D with packet switching (as depicted in the following figure). The file has a length 9000 bytes and is segmented into messages each with 1200 bytes data, which will be sent using TCP/IP. The Ethernet links of S-R1 and R2-D have the same transmission rate of 1Mbps; the PPP link of R1-R2 has the rate of 10Mbps. All the links have the same propagation delay of 2 ms. Each router has processing delay of 1ms, and the network traffic is very light, i.e., the queuing delay can be ignored. How long does it take the file to be delivered from S to D? (note: TCP and IP headers are of 20 bytes each, Ethernet header+trailer is of 18 bytes, and PPP header+trailer is of 8 bytes)



Solutions:

- （2 points） Number of packets: np = 9000/1200 = 8 packets
- (4 points) The first frame length of Ethernet links: le = 1200 + protocol headers (20 + 20 + 18) = 1258 bytes; The first frame length of PPP links: lp = 1200 + protocol headers (20 + 20 + 8) = 1248 bytes; The end-to-end delay of the first packet: df = transmit delay (2* 1258*8/1000 ms + 1248*8/ 10000 ms) + propagation delay (2*3 ms) + processing delay (1*2 ms) = 20.128 + 0.998 + 6 + 2 = 29 ms
- (2 points) The transmit delay of the remaining packets dt=(9000 – 1200 + 7 protocol headers (7* (20+20+18))) * 8/1000ms = 66 ms
- (2 points) The total delay: D = The end-to-end delay of the first packet df + The transmit delay of the remaining packets dt = 29 + 66 = 95 ms

5. （10 points）What's TCP fairness? Please explain with congestion control and AIMD (Additive Increase Multiplicative Decrease) game.

Solutions:

(4 points) TCP fairness: TCP provides per-connection, not per-application, fairness. If K TCP sessions share same bottleneck link of capacity R, each would have average rate of R/K.

(6 points) AIMD Game: during congestion control , congestion window will be additively increased until packet loss(AI) , and congestion window will be decreased by factor of 2 when loss detected(MD). In this way, we can achieve good bandwidth allocation to be fair and efficient.

6. (20 points) RDT can be implemented at any layer.

a) Please explain the technologies of Reliable Data Transfer introduced in this course

b) Please give examples of RDT implementation at application layer, transport layer and data link layer respectively and give reasons why they choose to implement RDT at these layers.

c) How to choose error control methods of Error Detection and Retransmission or Foreword Error Correction? Please explain and give examples of different scenarios.

Solutions:

a) (6 points) RDT technologies:

(1) Error control technologies: Error Detection and Retransmission, Forward Error Detection

(2) Packet Loss: Timer and Retransmission

(3) Disorder: Sequence No.

(4) Flow Control: Stop and Wait, Sliding Window

b) (2 points) Application layer: DNS. DNS uses UDP service which is unreliable but efficient, whereas UDP need to be reliable, so it implements RDT at application layer.

(2 points) Transport layer: TCP. TCP provide reliable service to upper layer applications.

(2 points) Data Link layer: HDLC, WiFi. They provide RDT at data link layer to deal with error in shortest time.

c)

(4 points) Forward Error Correction: using checking code to detect and correct bit errors. It’s suitable when errors are expected or when no time for retransmission.

(4 points) Error Detection and Retransmission: using checking code to detect bit errors and do retransmission when error detected. It’s more efficient when errors are not expected or burst errors when they do occur

(1 point) Which is better depends on the pattern of errors. e.g. 1000 bit messages with a bit error rate(BER) of 1 in 10000,

- in random: overhead FEC ~10, EDR ~100
- in burst of 100: overhead FEC >100, EDR ~30