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# Sirindhorn International Institute of Technology Thammasat University 

Final Exam Answers: Semester 1, 2011

Course Title: ITS323 Introduction to Data Communications
Instructor: Steven Gordon
Date/Time: Thursday 13 October 2011; 9:00-12:00

## Instructions:

- This examination paper has 21 pages (including this page).
- Conditions of Examination: Closed book; No dictionary; Non-programmable calculator is allowed
- Students are not allowed to be out of the exam room during examination. Going to the restroom may result in score deduction.
- Students are not allowed to have communication devices (e.g. mobile phone) in their possession.
- Write your name, student ID, section, and seat number clearly on the front page of the exam, and on any separate sheets (if they exist).
- Assume bits are ordered from left to right. For example, for the data 00001111, the first (1st) bit is 0 and the last (8th) bit is 1.
- Assume the speed of transmission is $3 \times 10^{8} \mathrm{~m} / \mathrm{s}$
- Reference material included at the end of the exam may be used.

Introduction to Data Communications, Semester 1, 2011
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## Question 1 [10 marks]

Consider datagram packet switching being used to send $n$ packets across a single path with $h$ hops, where $n \gg h$. Each packet has a length of $l$ bits. Each link has a data rate of $b$ bits per second. Each link has a propagation delay of $p$ seconds. Although there is processing delay and queuing delay at each node, they are both so small that you can assume they are both 0 .
(a) Write an equation that gives the total time, $T_{\text {datagram }}$, it takes from when the source host initiates the data transfer until the destination host has received all of the data. [6 marks]

Answer. $\quad T_{\text {datagram }}=n \times \frac{l}{b}+h \times p+(h-1) \times \frac{l}{b}$
(b) If all the conditions above are the same except virtual circuit packet switching was used instead of datagram packet switching, then would the total time, $T_{\text {virtualcircuit }}$, be greater than $T_{\text {datagram }}$, less than $T_{\text {datagram }}$, equal to $T_{\text {datagram }}$ or there is insufficient information to know. Explain your answer. [2 marks]

Answer. $T_{\text {virtualcircuit }}$ would be greater than $T_{\text {datagram }}$ because in virtual circuit packet switching the data is delivered in the same manner as datagram packet switching, but there is an additional time before the data transfer for the virtual circuit to be established.
(c) Explain an advantage of virtual circuit packet switching compared to datagram packet switching. [2 marks]

Answer. Virtual circuit packet switching ensures all packets in a data transfer will follow the same path. One advantage of this is that the packets will be delivered in order, hence no need for re-ordering at destination. Another advantage is that resources can be allocated along the path to improve the performance for the data transfer.

## Question 2 [7 marks]

(a) Draw a diagram to illustrate the TCP connection setup process between client A and server B. For each segment you must indicate appropriate flags and sequence/ack numbers. Assume A chooses an initial sequence number of 399, and B chooses 1034. [3 marks]


Figure 1: TCP Connection Establishment
(b) Assume that after the TCP connection setup between A and B above, that A sends 1500 bytes of data in a single segment to B . Then B sends 3000 bytes of data in a single segment to A using piggybacking. Finally A responds with an acknowledgement. Draw a new diagram showing the data transfer, indicating flags and numbers for each segment. (You don't have to draw the connection setup again) [4 marks]


Figure 2: TCP Data Transfer

## Question 3 [13 marks]

Consider the network in Figure 3.


Figure 3: An internet
The addresses on each interface, $I F$, are given in Table 1. All subnet masks are /24.
Table 1: Interface Addresses

| Interface | MAC Address | IP Address |
| :---: | :---: | :---: |
| $I F_{1}$ | $01: 23: 45: 67: 89: \mathrm{AB}$ | 10.10 .10 .1 |
| $I F_{2}$ | $\mathrm{AA}: \mathrm{BB}: \mathrm{CC}: \mathrm{DD}: \mathrm{EE}: \mathrm{FF}$ | 10.10 .10 .2 |
| $I F_{3}$ | $22: 44: 66: 88: \mathrm{AA}: \mathrm{CC}$ | 20.20 .20 .3 |
| $I F_{4}$ | $11: 33: 55: 77: 99: \mathrm{BB}$ | 20.20 .20 .4 |
| $I F_{5}$ | $\mathrm{FE}: \mathrm{DC}: \mathrm{BA}: 98: 76: 54$ | 30.30 .30 .5 |
| $I F_{6}$ | $00: 00: 11: 11: 22: 22$ | 30.30 .30 .6 |

Host A is running a web browser and a voice-over-IP (VoIP) application, while host B is running a web server, a web browser and a VoIP application. The user on host A has clicked on the URL http://www.example.com/courses/index.html. Assume DNS has already been used and that host A has already established a connection to host B (host A chose initial sequence number 1234, host B chose 89). Also, host A and B have already started a voice call using their VoIP applications. The VoIP application sends UDP packets containing PCM encoded voice.

Assume all HTTP requests are 100 Bytes in length, all HTTP responses are 1,000 Bytes, and all VoIP packets are 50 Bytes. The transport layer protocols can send the application layer data in a single segment. All requested URLs exist and are accessible on the server.

Assume you can intercept any packet on any link in the internet and see the entire contents of the packet. For the following packets and locations, complete the corresponding tables giving the value of the header fields.
(a) The packet containing the HTTP request sent by A, intercepted on link 1. [6 marks]

| Layer | Field | Value |
| :--- | :--- | :---: |
| Data Link | Source Address | $01: 23: 45: 67: 89: A B$ |
| Data Link | Destination Address | AA:BB:CC:DD:EE:FF |
| Network | Protocol | 6 |
| Network | Total Length | 140 |
| Transport | Destination Port | 80 |
| Transport | Sequence Number | 1235 |

(b) The packet containing the HTTP response sent by B, intercepted on link 3. [4 marks]

| Layer | Field | Value |
| :--- | :--- | :---: |
| Network | Source Address | 30.30 .30 .6 |
| Network | Destination Address | 10.10 .10 .1 |
| Network | Total Length | 1040 |
| Transport | Sequence Number | 90 |

(c) A packet containing VoIP data sent by A, intercepted on link 4. [3 marks]

| Layer | Field | Value |
| :--- | :--- | :---: |
| Network | Source Address | 10.10 .10 .1 |
| Network | Destination Address | 30.30 .30 .6 |
| Transport | Total Length | 58 |

## Question 4 [14 marks]

Consider a network with two links: $\mathrm{A}-\mathrm{B}-\mathrm{C}$.
First consider the link from A to B with the following characteristics:

- DATA frame consists of 100 bits of header plus 900 bits of data (total size 1,000 bits)
- ACK frame consists only of 100 bits of header
- Link data rate is $10 \mathrm{Mb} / \mathrm{s}$
- Link distance is 240 km

The Sliding Window flow control protocol is used on this link. The receiver (B) sends an ACK frame immediately after receiving a DATA frame (there is no processing delay). A 4-bit sequence number is used within the header of the DATA frame (and ACK frame). The maximum window size is 15 .
(a) What is the maximum number of DATA frames node A can send before having to wait for an ACK? [1 mark]

Answer. With a maximum window size of 15 then 15 frames may be sent before having to wait for an ACK.
(b) Assuming node A always has data ready to send, and it starts transmission of its 1st DATA frame at time 0s, at what time can node A start transmitting the 2nd DATA frame? [2 marks]

Answer. Immediately after the 1st DATA frame has been transmitted. The transmission time of a DATA frame is:
$T_{D A T A}=\frac{900+100}{10 \times 10^{6}}=100 \mu s$
Therefore, 2nd DATA frame can be sent at time 100us.
(c) At what time can node A start transmitting the 16th DATA frame? [4 marks]

Answer. As the maximum window size is 15 , the transmitter must wait until at least the 1st DATA frame is ACKed before sending the 16th DATA frame.
The ACK transmission time and Propagation delay are:
$T_{A C K}=\frac{100}{10 \times 10^{6}}=10 \mu \mathrm{~s}$
$P_{A B}=\frac{240 \times 10^{3}}{3 \times 10^{8}}=800 \mu s$
The time at which the ACK for the 1st DATA frame is received is: $T_{D A T A}+P_{A B}+$ $T_{A C K}+P_{B A}=1710 \mu s$. Note that the first 15 DATA frames are transmitted within 1500us. Therefore the transmitter must wait until time 1710us before sending the 16th frame.
(d) What is the maximum throughput that can be achieved across the link from A to B? [3 marks]

Answer. 15 DATA frames sent within 1710us, gives a throughput of:
Throughput $=\frac{15 \times 900}{1710 \times 10^{6}}=7.89 \mathrm{Mb} / \mathrm{s}$
Now consider the link from B to C with the following characteristics:

- DATA frame consists of 100 bits of header plus 900 bits of data (total size 1,000 bits)
- ACK frame consists only of 100 bits of header
- Link distance is 2.1 km

The Stop and Wait flow control protocol is used in this link.
(e) What is the minimum data rate necessary for Link B to C such that the throughput from A to C is the same as calculated in part (d)? [4 marks]

Answer. At B, 15 DATA frames arrive every 1710us. Therefore to achieve the same throughput from B to C, 15 DATA frames must be delivered to C every 1710 us. Considering Stop and Wait is used from B to C:
$P_{B C}=\frac{2100}{3 \times 10^{8}}=7 \mu \mathrm{~s}$
Time for sending 15 DATA frames:
$15 \times\left(\frac{1000}{\text { rate }}+P_{B C}+\frac{100}{\text { rate }}+P_{B C}\right)=1710 \times 10^{-6}$
Re-arranging gives:
rate $=\frac{1100}{\frac{1710}{15}-2.7}=11 \mathrm{Mb} / \mathrm{s}$
The data rate necessary is $11 \mathrm{Mb} / \mathrm{s}$.

## Question 5 [15 marks]

Assuming classless IP addressing is used, answer the following questions by writing your answers in the table on the next page. Unless otherwise stated, give all IP addresses in dotted decimal notation.
(a) For a host with IP address 207.89.34.56/8: [3 marks]
i. What is the network address?
ii. What is the directed broadcast address?
(b) For a host with IP address 73.193.16.4 and subnet mask 255.224.0.0: [4.5 marks]
i. What is the network address?
ii. What is the directed broadcast address?
iii. What is the maximum number of IP devices that can attach to this subnet?
(c) A host does not yet have an IP address configured, nor does it know its network address. [4.5 marks]
i. Give an IP address that the host can send to in order to send to itself.
ii. Give an IP address that the host can send to in order to deliver an IP datagram to all nodes on its subnet.
iii. For the case of part (ii), give the source address of the IP datagram.
(d) For a router with IP address 01101111010011101110000100000011 and host on a different subnet to the router with IP address 111.78.25.3: [3 marks]
i. What is the IP address of the router (in dotted decimal notation)?
ii. What is the subnet mask of the router (in slash or dotted decimal notation) that allows the most hosts to attach to the routers subnet?

| Question | Answer |
| :---: | :---: |
| (a) i. | $207.0 .0 .0 / 8$ |
| (a) ii. | $207.255 .255 .255 / 8$ |
| (b) i. | 73.192 .0 .0 |
| (b) ii. | 73.223 .255 .255 |
| (b) iii. | $2^{21}-2=2097150$ |
| (c) i. | 127.0 .0 .1 |
| (c) ii. | 255.255 .255 .255 |
| (c) iii. | 0.0 .0 .0 |
| (d) i. | 111.78 .225 .3 |
| (d) ii. | $/ 17$ or 255.255 .128 .0 |

## Question 6 [8 marks]

Consider LAN A with 5 computers (stations).
(a) Draw a diagram illustrating LAN A if using a bus topology. [2 marks]


Figure 4: Bus LAN
(b) Draw a diagram illustrating LAN A if using a ring topology. [2 marks]


Figure 5: Ring LAN

Consider another three LANs, LAN B, LAN C and LAN D, each with 5 computers. (That is, there are four LANs, and a total of 20 computers). Each of the four LANs use a star topology. The four LANs are then connected together by using a mesh topology amongst the switches.
(c) Draw a diagram illustrating the single larger LAN showing all 20 computers. [4 marks]


Figure 6: Larger LAN

## Question 7 [6 marks]

Consider the network shown in Figure 7. The delay (measured in milliseconds) and data rate (measured in $\mathrm{Mb} / \mathrm{s}$ ) for each link are shown in the format: (data rate, delay).


Figure 7: Network
Assuming least-cost routing is used to find a path from A to H , indicate the path and total cost if the cost metric was:
(a) Number of hops [2 marks]

Path: $A-B-G-H \quad$ Cost: 3
(b) Link delay [2 marks]

Path: $A-B-D-C-E-F-H \quad$ Cost: 27
(c) $\frac{100 \mathrm{Mb} / \mathrm{s}}{\text { Datarate }}[2 \mathrm{marks}]$

Path: $A-B-D-F-H$
Cost: 10

## Question 8 [7 marks]

(a) Draw the Internet protocol stack (or Internet protocol architecture), naming each of the layers. [3 marks]

| Application |
| :---: |
| Transport |
| Network |
| Data Link |
| Physical |

Figure 8: Internet stack
(b) Explain the key difference between an IP host and an IP router. [1 mark]

Answer. An IP router forwards IP datagrams, whereas an IP host does not. An $I P$ host is either a original source of a datagram or final destination of a datagram. If it receives an IP datagram a IP host will not send to another device.
(c) Explain a purpose of the TTL field in an IP header. [1 mark]

Answer. Time To Live is decrement by each router. It limits the number of hops the datagram will traverse. This is useful if there is a routing error the creates a loop. Without a TTL the datagram will be transmitted for ever; with a TTL it will eventually be dropped by a router.
(d) IP fragmentation/reassembly allows hosts and routers to perform fragmentation, but only hosts can reassemble. Explain the advantage of this approach compared to only allowing hosts to fragment and reassemble (i.e. not allowing routers to fragment or reassemble). [2 marks]

Answer. If routers cannot fragment then it means the source host must know the maximum packet sizes across the entire path. Hence there is extra time/complexity needed in discovering this. With IP the source host (and subsequent routers) only need to know the maximum packet size for the next hop.

## Question 9 [7 marks]

TCP includes two error control mechanisms: basic retransmit and fast retransmit.
(a) What ARQ scheme is basic retransmit most similar to? [1 mark]

Answer. Selective-reject (or selective-retransmission) ARQ, since a source is allowed to send a window size of frames (hence not stop-and-wait $A R Q$ ) and if a segment is lost only the lost segment needs to be retransmitted-other segments are buffered (hence not go-back-N ARQ).
(b) Explain how basic retransmit works in TCP. [2 marks]

Answer. When the source sends a segment it starts a timer. If an ACK is received before the timer expires than the timer is stopped, and the segment successfully delivered. If the timer expires before an ACK is received then the source retransmits the segment.
(c) Explain how fast retransmit works in TCP. [2 marks]

Answer. If the source receives four ACKs with the same acknowledgement number, then it assumes the segment that contains that acknowledgement number is lost and hence retransmits the segment.
(d) Explain an advantage of fast retransmit (compared to basic retransmit). [2 marks]

Answer. The source potentially retransmits a lost segment earlier than if waiting for a timeout, and hence spends less time waiting and leads to higher efficiency.

## Question 10 [10 marks]

Consider the network in Figure 9. Each router has multiple numbered interfaces (e.g. Router A has interface A1 and A2). There are four LANs shown, with each LAN containing multiple hosts. Router H connects, via interface H1, to another router which then may connect to many more routers and LANs (in other words, assume more LANs can be reached via router H ).


Figure 9: Routed Network
Assuming the destinations may be in any of the four LANs or in any LANs beyond router H , complete the following routing tables. You may use as many rows as necessary. Use * to indicate a wild card (any value). Use direct to indicate direct delivery to the destination. Use the LAN name to refer to a network (e.g. LAN1). Use the router name and interface to refer to a next router (e.g. A1). Assume rows in the routing table are processed in order (from top to bottom).

Answer. Variations of the following answer are possible (e.g. not relying on direct).


Router B

| Dest | Next |
| :---: | :---: |
| LAN1 | A2 |
| LAN2 | direct |
| $*$ | C1 |
|  |  |
|  |  |
|  |  |

Router F

| Dest | Next |
| :---: | :---: |
| LAN1 | E2 |
| LAN2 | E2 |
| LAN3 | E2 |
| LAN4 | G1 |
| $*$ | H2 |
|  |  |

LAN1 Host

| Dest | Next |
| :---: | :---: |
| LAN1 | direct |
| $*$ | A1 |
|  |  |
|  |  |
|  |  |
|  |  |

## Question 11 [3 marks]

(a) With respect to point-to-point transmission links, what is multiplexing? [1 mark]

Answer. Multiplexing is combining signals from multiple transmitting sources into one signal that is transmitted on a single link. At the end of the link the individual component signals are separated and sent to their intended destinations.
(b) Explain the difference between TDM and FDM. [2 marks]

Answer. Frequency Division Multiplexing allocates different frequencies for the input user signals. Time Division Multiplexing allocates different time slots for the input user signals.

## Reference Material

Selected well-known ports:

- FTP 20 and 21
- SSH 22
- Telnet 23
- SMTP 25
- DNS 53
- HTTP 80
- HTTPS 443


Figure 10: IP Datagram Format. Flags: Reserved, Don’t Fragment, More Fragments


Figure 11: TCP Segment Format. Flags: CWR, ECE, URG, ACK, PSH, RST, SYN, FIN

|  | 16 |  |
| :---: | :---: | :---: |
| $\stackrel{¢}{6}$ | Source Port | Destination Port |
| 盛 | Total Length | Checksum |
|  | Data |  |

Figure 12: UDP Datagram Format


Figure 13: Ethernet Frame Format

Selected Protocol numbers:

- 1 ICMP
- 6 TCP
- 17 UDP

Selected HTTP Status Codes:

- 200 Ok
- 304 Not Modified
- 401 Unauthorized
- 404 Not Found

