UCLA-19S-CS118 Midterm

Spencer Benjamin Pugh

TOTAL POINTS

89 / 100

QUESTION 1

Problem 1 18 pts

1.1 1.1 D **2 / 2**

✓ **- 0 pts** Correct

1.2 1.2 C **2 / 2**

✓ **- 0 pts** Correct

1.3 1.3 B **2 / 2**

✓ **- 0 pts** Correct

1.4 1.4 B **2 / 2**

✓ **- 0 pts** Correct

1.5 1.5 D **2 / 2**

✓ **- 0 pts** Correct

1.6 1.6 C **2 / 2**

✓ **- 0 pts** Correct

1.7 1.7 BC **2 / 2**

✓ **- 0 pts** Correct

1.8 1.8 A **2 / 2**

✓ **- 0 pts** Correct

1.9 1.9 C **2 / 2**

✓ **- 0 pts** Correct

QUESTION 2

Problem 2 18 pts

2.1 3 / 3

✓ **- 0 pts** Correct

2.2 3 / 3

✓ **- 0 pts** Correct

2.3 3 / 3

✓ **- 0 pts** Correct

2.4 3 / 3

✓ **- 0 pts** Correct

2.5 3 / 3

✓ **- 0 pts** Correct

2.6 2 / 3

✓ **- 1 pts** Got 17. But wrong bits

QUESTION 3

3 Problem 3 8 / 8

✓ **+ 2 pts** One Item Correct

✓ **+ 2 pts** One Item Correct

✓ **+ 2 pts** One Item Correct

✓ **+ 2 pts** One Item Correct

QUESTION 4

Problem 4 12 pts

4.1 4 / 4

✓ **- 0 pts** Correct

4.2 3 / 4

✓ **- 1 pts** Click here to replace this description.

4.3 2 / 4

✓ **- 2 pts** Partially Correct

QUESTION 5

5 Problem 5 6 / 12

✓ - 1 pts ACK flag missing/wrong
✓ - 1 pts FIN flag missing/wrong
✓ - 2 pts Hdr len wrong
✓ - 2 pts UPRS flags missing/wrong

QUESTION 6

Problem 6 12 pts

6.1 3 / 3

✓ - 0 pts Correct

6.2 2 / 3

✓ - 1 pts No drop the packet

6.3 3 / 3

✓ - 0 pts Correct

6.4 3 / 3

✓ - 0 pts No restarting the timer

QUESTION 7

Problem 7.1 8 pts

7.1 2 / 2

✓ - 0 pts Correct

7.2 3 / 3

✓ - 0 pts Correct

7.3 3 / 3

✓ - 0 pts Correct

QUESTION 8

Problem 7.2 12 pts

8.1 3 / 3

✓ - 0 pts Correct

8.2 3 / 3

✓ - 0 pts Correct

8.3 3 / 3

✓ - 0 pts Correct

8.4 3 / 3

✓ - 0 pts Correct

8.5 Figure 0 / 0

✓ - 0 pts Correct

CS118 Midterm Exam, Spring 2019

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Student ID: 605006452

Notes:

1. This is a closed-book, closed-notes examination. But you can use the one-page cheat sheet.
2. You are not allowed to use your electronic devices.
3. Be **brief** and **concise** in your answers. Answer only within the space provided. If you need additional work sheets, use them but do NOT submit these sheets with the midterm examination.
4. Use the back pages for scratch paper. You should cross out your scratch work when you submit your exam paper. Any answers or work shown on the back of any page will NOT be considered or graded.
5. If you wish to be considered for partial credit, show all your work.
6. Make sure that you have 10 pages (including this page and one-page Appendix) before you begin.

PROBLEM	MAX SCORE	YOUR SCORE
1	18	
2	18	
3	8	
4	12	
5	12	
6	12	
7	20	
TOTAL	100	

DO NOT TURN TO THE NEXT PAGE UNLESS YOU GET PERMISSION !!

Problem 1: Multiple choices (2 points each). Select <u>all</u> the correct answers from the four choices. Note that there can be *multiple* correct answers.

For your reference, the following are the full names for the used Acronyms: HTTP: hypertext transfer protocol; DNS: Domain name system; SMTP: Simple mail transfer protocol; POP3: Post office protocol version 3; IMAP: Internet Mail access protocol; TCP: transmission control protocol; UDP: User datagram protocol.

1. What would be the correct feature(s) of a file-distribution application using the peer-to-peer (P2P) model rather than the client-server model?
   - Your answer D (A) All peers have to remain always on; (B) A peer cannot leave even after its file downloading completes; (C) P2P model scales worse than the client-server model; (D) A peer can act as a client by issuing requests while serving other peers.
2. Which protocol is not used when Bob uses his smartphone to browse the ESPN Web site?
   - Your answer C (A) TCP; (B) HTTP; (C) IMAP; (D) DNS.
3. Which field(s) in the UDP header are used in connectionless demultiplexing?
   - Your answer B (A) Source port number; (B) Destination port number; (C) Length; (D) Sequence number.
4. What is true regarding Dynamic Adaptive Streaming over HTTP (DASH) for video streaming?
   - Your answer B (A) The video is encoded into several different versions, which have different qualities but have the same bit rate; (B) It allows clients with different Internet access rates to stream in video; (C) The client selects different chunks one at a time with HTTP PUT request messages; (D) It does not allow a client to dynamically adjust its streaming rate once the streaming session starts.
5. Which is a benefit of packet switching but not circuit switching?
   - Your answer D (A) congestion may occur inside the network; (B) reservation is always needed before data delivery; (C) providing delay guaranteed services; (D) statistical multiplexing
6. When can the first TCP data segment starts its transmission during the TCP connection?
   - Your answer C (A) together with the first SYN message; (B) before the second SYN+ACK message is received; (C) together with the third ACK message; (D) After the third ACK message has arrived at the receiver.
7. Which of the following statement about DNS is true?
   - Your answer B, C (A) A local DNS server never queries the root DNS server; (B) DNS uses caching to improve performance; (C) Some of DNS queries can be iterative and others recursive, in the sequence of queries to translate a hostname; (D) Only authoritative DNS servers can respond to queries.
8. Which layers in the protocol stack are implemented at the end host?
   - Your answer A (A) application layer, transport layer, network layer, link layer, physical layer; (B) application layer, transport layer; (C) network layer, link layer, physical layer; (D) application layer only.

9. Which mechanism of HTTP is used to allow a cache to verify that its objects are up to date?

   - Your answer C (A) Cookies; (B) stateless HTTP server; (C) conditional GET; (D) HTTP with persistent connections.

Problem 2 (3 points each): Answer the following questions. Be brief and concise.

1. In an ongoing TCP connection, the TCP sender receives a new Acknowledgment segment, which has its 'Receive Window' field set as 11000 Bytes. Before receiving this Acknowledgment, the TCP sender has not perceived any segment loss with its *cwnd* being 10000 Bytes and its *ssthresh* as 8000 Bytes. What is the window size updated by TCP in its reliable transfer right after receiving this Acknowledgment? Assume the maximum segment size (MSS) is 1000 Bytes. Show your steps.

   - don't need to use flow control
   - congestion avoidance → increase window size by $\frac{MSS}{cwnd} = \frac{1000B}{10000B} = \frac{1}{10}$ (·MSS) ($\frac{1}{10}$ of a segment)

   window size is set to **10100 Bytes**

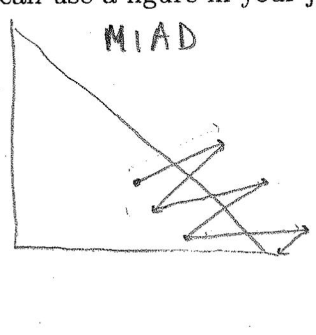
2. Consider the queuing delay in a router buffer (preceding an outbound link). Suppose all packets are L bits, the transmission rate is R bps, and that all N packets simultaneously arrive at the buffer at time t=0. Find the queuing delay of the packet that is transmitted last.

   first packet waits 0
   second packet waits $\frac{L}{R}$
   third packet waits $2\frac{L}{R}$
   ⋮
   last packet (N) waits $\frac{(N-1)L}{R}$

3. A new video streaming startup *v-start* decides to use the third-party CDN provider *limelight* to scale and reduce streaming latency. Explain how to leverage DNS to intercept and redirect the user clicks on `http://video.v-start.com/` to the CDN servers deployed by *limelight*.

   v-start uses the request from client to determine which CDN server the client should connect to. Then instead of directing client to sub-domain of v-start, the response DNS message from the v-start server contains the address of that CDN server.

4. Consider two TCP connections sharing a single link, with identical round-trip-times and segment sizes. The additive-increase, multiplicative-decrease (AIMD) mode is known to ensure fair share for both connections eventually. Now a programmer decides to use multiplicative-increase, additive-decrease (MIAD) mode in his implementation. Explain whether this change will result in starvation of a connection (i.e., if starting from an arbitrary window size, one connection will eventually be starved to the lowest speed while the other will get the highest speed) or not. You can use a figure in your justification.



   Yes, MIAD will lead to starvation. The larger connection will grow faster than the small one, until the small connection is starved (a multiplicative increase from 0 is still 0).

5. Consider that amazon.com uses a session cookie to track the shopping cart on Susan's browser. Can ebay.com reuse the same cookie? Can amazon.com reuse the same cookie on Bob's browser?

ebay.com can not use the same cookie for Susan
amazon.com can not use the cookie for Bob

The cookie is valid only for Susan and amazon.com together.

6. For the Go-Back-N protocol with the sender window size of 16, what is the minimum number of bits needed for the sequence number field?

$N+1 = 16+1 = 17$

Problem 3 (8 points): Joe is writing programs with a client and a server using stream sockets. The following is the SERVER code that Joe wrote. Can you help Joe to find four errors (there can be more) in his code? You can mark your answers in his code, and label the errors in the code. You can use the Appendix for references.

```
#include <server.h>  /* assume all headers are included correctly */
#define PORT 8080
int main(int argc, char const *argv[])
{
    int server_fd, new_fd; /* listen on server_fd, new connection on new_fd */
    struct sockaddr_in address;
    int addrlen = sizeof(address);
    char buffer[1024] = {0};
    char *hello = "Hello from server";

    if ((server_fd = socket(AF_INET, SOCK_STREAM, 0)) > 0) {
        perror("socket failed");
        exit(EXIT_FAILURE);
    }
    address.sin_family = PF_INET;
    address.sin_addr.s_addr = INADDR_ANY;
    address.sin_port = ntohs(PORT);

    if (bind(server_fd, (struct sockaddr *)&address, addrlen) < 0) {
        perror("bind failed");
        exit(EXIT_FAILURE);
    }
    if (accept(server_fd, 3) < 0) {
        exit(EXIT_FAILURE);
    }
    if ((new_fd = listen(server_fd, (struct sockaddr *)&address, (socklen_t*)&addrlen)) < 0) {
        exit(EXIT_FAILURE);
    }
    int valread = read(server_fd, buffer, 1024);
    printf("%s\n", buffer);
    sendto(new_fd, hello, strlen(hello), 0);
    printf("Hello message sent\n");
}
```

- At `> 0`: should be == / > is not an error
- At `ntohs`: should be htons
- Before `accept`: need to listen() before accept()
- At `new_fd =`: need to store result of accept here, not listen
- At `read(server_fd`: should be new_fd

Problem 4 (12 points): Delay in Application Layer Protocols One end host A visits `store.google.com` and `www.amazon.com` sequentially to compare the price of a smart watch.

- (4 points) (a) Assume Host A's local DNS server's cache is empty initially. Therefore, Host A needs to get the IP address of `store.google.com` via DNS query. Also assume that:
  - Only iterative DNS queries are used.
  - The RTT between Host A and the local DNS server is 20 ms.
  - The RTT between the local DNS server to any authoritative DNS server is 100 ms.
  - Ignore any DNS server's processing time.
  - TTL value for any record is 1 hour.
  - Any domain under `google.com` is hosted by `ns.google.com`; any domain under `amazon.com` is hosted by `ns.amazon.com`.

  How many milliseconds would have elapsed when Host A gets the IP of `store.google.com`?

20 ms to get top-level domain from local (.com)
100ms to get authoritative from top-level (google.com)
100ms to get ns.google.com from google.com
100ms to get store.google.com from ns.google.com

(20+100+100+100) ms = **320ms**

- (4 points) (b) Host A next visits `www.amazon.com` to compare the price of the same product. Assume `www.amazon.com` points to a 1000-Byte HTML file which references 4 images. Each referenced image size is 500 Bytes. The *one-way* propagation delay between Host A and Amazon's Web server is 100 ms and the transmission rate is 1 Mbps. When calculating the transmission delay, assume only HTTP response payload counts for that. If Host A uses HTTP 1.0, how many milliseconds would have elapsed when Host A receives all referenced images of `www.amazon.com` since it inputs the URL `www.amazon.com` in a browser's address bar?

404 × 4 = 1616; 1616 + 408 = 2024

DNS
.com is in cache
100 ms for amazon.com from .com
100ms for ns.amazon.com from amazon.com
100ms for www.amazon.com from ns.amazon.com
= 300ms for DNS

500B = 4000b
each image takes $\frac{4000b}{1Mbps}$ = 4ms to transmit
1000B = 8000b
index.html takes 8ms to transmit
to get index.html (assuming no trans. delay for set up connection messages) takes 100ms + 100ms + 100ms + 8ms + 100ms = 408 ms
each image takes 400 ms + 4 ms

total for HTTP = 408ms + 4·404ms = 2024ms

total delay = 300ms for DNS + 2024ms for HTTP = **2324 ms**

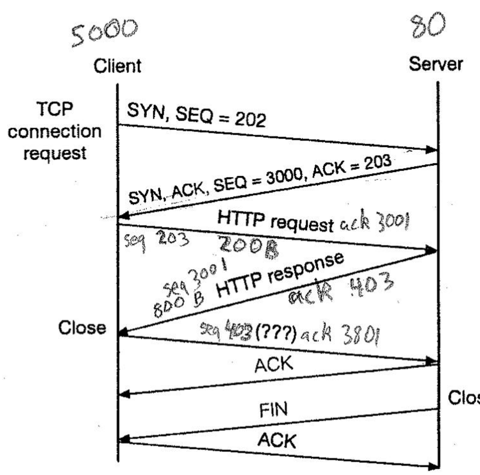
- (4 points) (c) Repeat (b), assuming Host A uses HTTP 1.0 with maximum 4 parallel connections. All other assumptions are the same.

Assuming (c) happens <u>instead of</u> (b) rather than <u>after</u> (b) (otherwise DNS would change)
DNS = 300ms still
HTTP = 200ms to open + 408ms for index + 200ms to open + 404ms for images in ||
= 1212ms

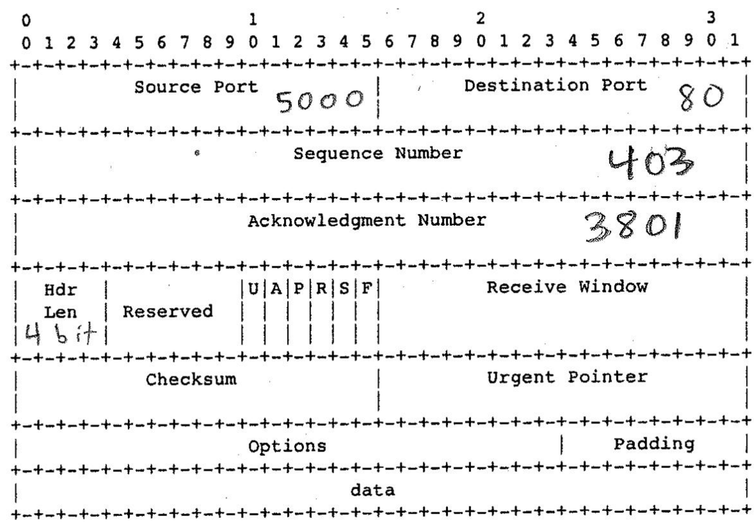
total = 300ms + 1212ms = **1512ms**

Problem 5 (12 points): TCP Protocol You are asked to fill in as many fields as possible in the TCP header (given in the right subfigure) for the fifth TCP message (marked with (???)) shown in the left figure. The left figure shows the sequence of messages being exchanged between the client and the Web server, starting from TCP connection establishment, HTTP request/response, to TCP connection close.

Write your answers *in decimal format* directly in the right figure. Assume that non-persistent HTTP is used. The HTTP request is 200 Bytes, while the HTTP response from the server is 800 Bytes. Client and Server's IP addresses are 111.111.111.111 and 222.222.222.222, respectively, and the client uses port 5000 for the TCP connection (hint: the TCP port number used on the Web server is 80).



(a) TCP Message Exchanges



(b) TCP Header

Problem 6 (12 points): Reliable Transfer Protocol In the following scenario, discuss how each reliable transfer protocol reacts:

- (3 points) Will premature timeout (i.e., the retransmission timer has a value smaller than RTT) affect the correctness of reliable transfer protocol? Briefly justify your answer.

  No. There will be lots of retransmission but correctness is maintained, since the sender will eventually receive ACKs.

- (3 points) In the Stop-and-Wait protocol, how does the receiver reacts if a duplicate data packet is received?

  The receiver sends a cumulative ACK of its last byte received (this will be a duplicate ACK).

- (3 points) Assume cumulative acknowledgment is used and each data segment size is 100B. In the Selective Repeat protocol, how does the sender react when receiving an ACK segment (with acknowledgment number 900B) right after receiving a duplicate ACK segment (with acknowledgment number 800B)?

The sender window is advanced to start at the packet with sequence number 900B. This is because cumulative ACKs guarantee that all packets before the sender window have been received. Packets that have entered the sender window are then transmitted sequentially.

- (3 points) In the Go-back-N protocol, if timeout is triggered at the sender before the sender receives the third duplicate ACK, how does the sender react? Justify your answer.

The sender, upon timeout, retransmits all packets in the sender window. This is always the case for GBN.

Problem 7 (20 points): TCP Congestion Control You will work on two scenarios on TCP congestion control. Note that Part 1 and Part 2 are considering different TCP connections; they are not correlated.

1. (8 points) Consider a scenario that a timeout event has been observed at the TCP sender. When the timeout occurs, the congestion window size *cwnd* at the sender is 9 segments. In this scenario, we assume that the receiver's advertised window size is always larger than 10 segments.

   - (2 points) How does TCP congestion control update its *cwnd* upon timeout?

     cwnd is set to 1 segment.

   - (3 points) How does TCP congestion control update its *ssthresh* (i.e., slow start threshold value) upon timeout? Show your steps.

     old value of cwnd = 9 segments $\left(ssthresh = \frac{cwnd}{2}\right)$

     $\frac{9 \text{ segments}}{2} = 4.5$ segments $\xrightarrow{\text{round down}}$ 4 segments

     ssthresh is set to 4 segments.

   - (3 points) Can the TCP sender transmit a new segment in addition to retransmitting the segment that experiences timeout? Briefly justify your answer.

     Not until the sender receives the ACK for the retransmitted segment because a timeout resets cwnd to 1 segment.

2. (12 points) Consider another new TCP connection. Assume that all algorithms are implemented in TCP congestion control: slow start, congestions avoidance, fast retransmit and fast recovery, and retransmission upon timeout. Right after fast retransmit/fast recovery phase, if *ssthresh* = *cwnd*, use the slow start algorithm. <u>You must draw a diagram to show the intermediate steps on Page 9 to receive full credit.</u>

- TCP uses reliable transfer. The used ACK for each segment is based on cumulative ACK and the acknowledgment number indicates the next expected segment. The receiver acknowledges every segment, and the sender always has data available for transmission.
- Initially *ssthresh* at the sender is set to 8, and *cwnd* as 1. Assume *cwnd* and *ssthresh* are counted in segments, and the transmission time for each segment is negligible (equivalently, you can assume that each segment size is conceptually one unit). Retransmission timeout (RTO) is initially set to 500ms and remains unchanged during the connection lifetime. The RTT is 100ms for all transmissions.
- The connection starts from the initial sequence number of 1 at t=0. Segments with sequence number 4 and 5 arrived at the receiver out of order (i.e., segment 5 arrives before segment 4). Segment with sequence number 7 is lost once. No other misbehavior is observed.

(a) (3 points) The receiver sends an ACK upon receiving segment with sequence number 5. What algorithm should the sender use when receiving this ACK? What is the updated value for *cwnd* and *ssthresh* upon this?

no action. cwnd remains 4, ssthresh remains 8
(ACK 4, first duplicate)

(b) (3 points) The receiver sends an ACK upon receiving segment with sequence number 4. What algorithm should the sender use when receiving this ACK? What is the updated value for *cwnd* and *ssthresh* upon this?

ACK 6, slow start. new ack so cwnd += 1 → cwnd = 5
ssthresh remains 8.

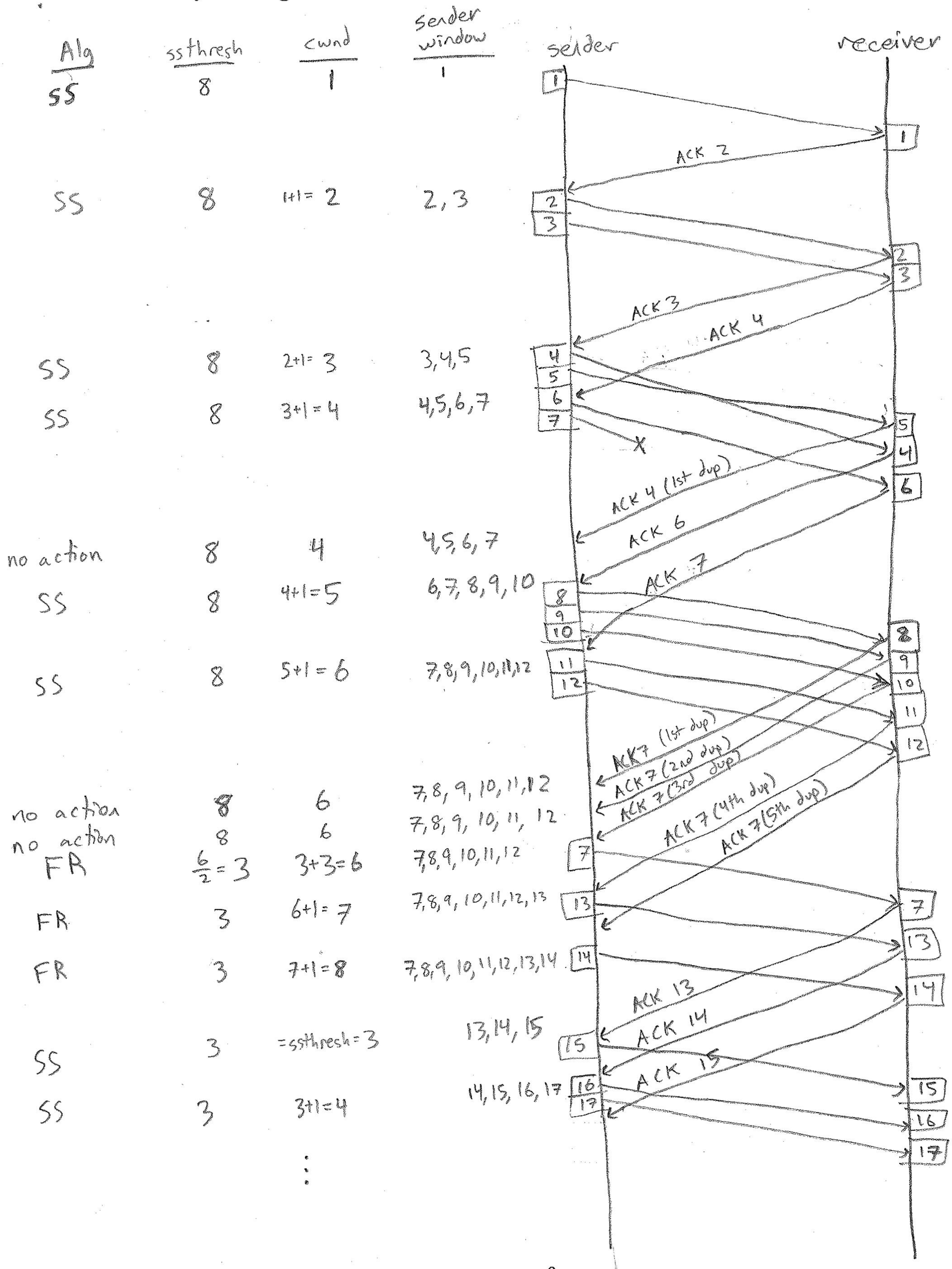
(c) (3 points) The receiver sends an ACK upon receiving the segment with sequence number 10. What is the sequence of actions the sender takes upon receiving this ACK?

(ACK 7, 3rd duplicate)
- switch to Fast Retransmit/Fast Recovery Algorithm
- set ssthresh to $\frac{cwnd}{2} = \frac{6}{2} = 3$
- set cwnd to ssthresh + 3 = 3 + 3 = 6
- retransmit packet with sequence number 7

(d) (3 points) What is the congestion window size *cwnd* at the sender when the sender transmits segment with sequence number 16?

4 (see diagram)

Draw your diagram for Problem 7.2 here.



Appendix. Socket Programming Function Calls.

- *struct in_addr { in_addr_t s_addr; /* 32-bit IP addr */}*
- *struct sockaddr_in {*
  short sin_family; / e.g., AF_INET */*
  ushort sin_port; / TCP/UDP port */*
  struct in_addr; / IP address */ }*
- *struct hostent* gethostbyaddr (const char* addr, size_t len, int family)*
  struct hostent gethostbyname (const char* hostname)*;
  char inet_ntoa (struct in_addr inaddr)*;
  int gethostname (char name, size_t namelen)*;
- *int socket (int family, int type, int protocol)*;
  [*family*: AF_INET (IPv4), AF_INET6 (IPv6), AF_UNIX (Unix socket); *type*: SOCK_STREAM (TCP), SOCK_DGRAM (UDP); *protocol*: 0 (typically)]
- *int bind (int sockfd, struct socketaddr* myaddr, int addrlen)*;
  [*sockfd*: socket file descriptor; *myaddr*: includes IP address and port number; *addrlen*: length of address structure == sizeof(struct sockaddr_in)]
  returns 0 on success, and sets *errno* on failure.
- *int sendto(int sockfd, char* buf, size_t nbytes, int flags, struct sockaddr* destaddr, int addrlen)*;
  [*sockfd*: socket file descriptor; *buf*: data buffer; *nbytes*: number of bytes to try to read; *flags*: typically use 0; *destaddr*: IP addr and port of destination socket; *addrlen*: length of address structure == sizeof(struct sockaddr_in)]
  returns number of bytes written or -1. Also sets *errno* on failure.
- *int listen (int sockfd, int backlog)*;
  [*sockfd*: socket file descriptor; *backlog*: bound on length of accepted connection queue]
  returns 0 on success, -1 and sets *errno* on failure.
- *int recvfrom (int sockfd, char* buf, size_t nbytes, int flags, struct sockaddr* srcaddr, int* addrlen)*;
  [*sockfd*: socket file descriptor; *buf*: data buffer; *nbytes*: number of bytes to try to read; *flags*: typically use 0; *destaddr*: IP addr and port of destination socket; *addrlen*: length of structure == sizeof(struct sockaddr_in)]
  returns number of bytes read or -1, also sets *errno* on failure.
- *int connect(int sockfd, struct sockaddr* servaddr, int addrlen)*;
  [*sockfd*: socket file descriptor; *servaddr*: IP addr and port of the server; *addrlen*: length of structure == sizeof(struct sockaddr_in)]
  returns 0 on success, -1 and sets *errno* on failure.
- *int close (int sockfd)*;
  returns 0 on success, -1 and sets *errno* on failure.
- *int accept (int sockfd, struct sockaddr* cliaddr, int* addrlen)*;
  [*sockfd*: socket file descriptor; *cliaddr*: IP addr and port of the client; *addrlen*: length of structure == sizeof(struct sockaddr_in)]
  returns file descriptor or -1 sets *errno* on failure
- *int shutdown (int sockfd, int howto)*;
  returns 0 on success, -1 and sets *errno* on failure.
- *int write(int sockfd, char* buf, size_t nbytes)*;
  [*sockfd*: socket file descriptor; *buf*: data buffer; *nbytes*: number of bytes to try to write]
  returns number of bytes written or -1.
- *int read(int sockfd, char* buf, size_t nbytes)*;
  [*sockfd*: socket file descriptor; *buf*: data buffer; *nbytes*: number of bytes to try to read]
  returns number of bytes read or -1.
- *int select(int maxfdp1, fd_set *readfds, fd_set *writefds, fd_set *exceptfds, struct timeval *tvptr)*;
- Convert multi-byte integer types from host byte order to network byte order:
  - *uint32_t htonl(uint32_t hostlong)*: host to network short
  - *uint16_t htons(uint16_t hostshort)*: host to network long
  - *uint32_t ntohl(uint32_t netlong)*: network to host short
  - *uint16_t ntohs(uint16_t netshort)*: network to host long