CS/ECE 438: Computer Networks **Spring 2026**

Homework 4

Handed Out: April 11, 2026 *Due: 11:59pm, May 1, 2026*

TA: Nishant Sheikh

- Homework assignments must be submitted online through Gradescope. No handwritten solutions will be accepted. LaTeX formatting is recommended, but not enforced.
- Each question is worth the same number of points (e.g., 10 points). Show your work for full credit. You will not receive points for a correct answer with no work shown. Partial credit may be awarded for correct reasoning even if the final answer is incorrect.
- Please come to office hours if you have questions about the homework.
- While we encourage discussion within and outside of the class, cheating and copying is strictly prohibited. Copied solutions will result in the entire assignment being discarded from grading at the very least and a report filed in the FAIR system.

Updated April 15, 2026: see 2a, 3, 5, 7

1 Sequence Number Space

Consider the Go-Back-N protocol, with $SWS = 10$, and sequence numbers mod 1024. Suppose that at time t, the next in-order packet that the receiver is expecting has a sequence number of k. Assume that the medium may drop packets, but does not reorder messages.

a. Describe the sets of sequence numbers that could be inside the sender's window at time t. Justify your answer.

b. Let's say that at time t, there is a message propagating back to the sender. What are all possible values of the ACK field in that message? Justify your answer.

c. With the Go-Back-N protocol, is it possible for the sender to receive an ACK for a packet that falls outside of its current window? Justify your answer with an example.

2 Sliding Window Protocols - Part 1

This question concerns tradeoffs given different sliding window parameters.

a. For each of the following situations, draw two timeline diagrams — one with cumulative ACKs, one with Selective Repeat ACKs — for the sliding window algorithm. Use a fixed $SWS = 6$ frames, and $RWS = 5$ frames. Use a timeout interval of $2 \times RTT$. Also, assume that $BDP > SWS$.

   Label your timeline diagrams with sequence & ACK numbers, and show when packets are buffered or time out.

   i. Frame 1 is lost.

   ii. Frames 1-5 are lost.

b. Consider the sliding window algorithm with $SWS = RWS = 6$ and sequence numbers mod 10. This means that the nth packet, P_n, contains n mod 10 in its sequence number field. Show an example in which the algorithm becomes confused. No packets may arrive out of order.

3 Sliding Window Protocols - Part 2

You want to transfer a file from Champaign to Chicago. For this problem, assume:

- The file size is 100 000 B.
- The file will be transferred in data packets. Data packets are 1300 B long, with 100 B of header and 1200 B of payload. For all data packets, the payload is padded up to 1200 B.
- The size of acknowledgement packets, including header, is 100 B.
- Each individual packet is acknowledged.
- Hosts can send and receive at the same time. Communication is bidirectional, and the bandwidth is 300 Mbps in each direction.
- The propagation time between Champaign and Chicago is 10 ms in both directions.
- Computation time for packet processing is negligible. Once a packet has been received in full, the receiver can send an acknowledgement immediately.
- Likewise, the sender can send packets back-to-back (limited by its window size).
- There is no packet loss, corruption or reordering.

Tip: for this question, we suggest making timeline diagrams to help you visualize events and timings. We will not grade these diagrams.

Based on these parameters, answer the following questions:

a. How long will it take to transfer the file using Stop-and-Wait?

b. What is the corresponding goodput?

c. If $SWS = RWS = 8$, how long will it take to transfer the file using a sliding window protocol? What is the goodput?

d. If $SWS = RWS = 2 \times BDP$, how long will it take to transfer the file using a sliding window protocol? What is the goodput?

e. If $SWS = RWS = 4 \times BDP$, how long will it take to transfer the file using a sliding window protocol? What is the goodput? Explain why.

4 TCP RTT Estimation

One difficulty with the original TCP SRTT estimator is the choice of an initial value. In the absence of any special knowledge of network conditions, the typical approach is to pick an arbitrary value, such as 3 seconds, and hope this will converge quickly to an accurate value. If this estimate is too small, TCP will perform unnecessary retransmissions. If it is too large, TCP will wait a long time before retransmitting if the first segment is lost. Also, the convergence might be slow.

a. Choose $\alpha = 0.5$ and SRTT(0) $= 1.5\,$s, and assume all measured RTT values $= 1\,$s with no packet loss. What is SRTT(10)? Recall:

$$SRTT(k+1) = \alpha\, SRTT(k) + (1-\alpha)\, RTT(k+1)$$

Describe your solution approach, and provide the numerical result to 4 decimal places.

b. Using the same values as in part a, what happens if we use $\alpha = 0.3$ or $\alpha = 0.8$? Provide a numerical result for SRTT(10) in both cases, then describe the effect of a larger or smaller α on the RTT estimation feature.

c. What is the retransmission ambiguity problem addressed by the Karn-Partridge algorithm? How does the algorithm avoid the ambiguity?

5 TCP Slow Start

Although slow start with congestion avoidance is an effective technique for coping with congestion, it can result in long recovery times in high-speed networks.

a. Assume a RTT delay of 150 ms, a 1 Gbps link, and a segment size of ~~2 kB~~ 2048 B. Determine the window size needed to keep the pipe full, and the time it will take to reach that window size after a timeout using the Jacobson Algorithm, assuming that the sender was transmitting at that full window before the timeout.

b. Repeat for a segment size of ~~16 kB~~ 16 384 B.

6 HTTP

Suppose a webpage has nothing but 30 large images, each 4 MB in size. The size of the webpage itself is negligible. A client wants to access the webpage and load the images in their browser. Between the client and server, the RTT is 60 ms, and the transmission rate is 50 Mbps. How long will it take to load the webpage in each of the following cases?

Be sure to account for the time needed to establish the connection and retrieve the webpage itself. You do not need to account for the delay from closing the connection.

a. Using Non-Persistent HTTP?

b. Using Persistent HTTP?

c. Using Pipelined Persistent HTTP?

7 DNS

The task requires using the `dig` command to provide answers. To ensure accurate results, it is recommended to perform these steps from a computer located on a campus network. The user can refer to the `dig` documentation to understand how to utilize it.

a. Starting from one of the root servers `{a-m}.root-servers.net`, perform an iterative lookup for the host `images.nasawestprime.com`. For example, you could initiate the search using the following command:

   ```
   dig @a.root-servers.net images.nasawestprime.com
   ```

   Please provide a list of the following information for each name server you visit during the lookup process:

   i. Can you specify the domain name of the name server being visited?

   ii. Can you provide the IP address of the name server that is currently being used?

   iii. How long did the query take?

   iv. How long will this result be cached for?

   v. What is the round-trip time (RTT) to the server? *Hint: try `ping`!*

   Some nameservers may not respond to your pings. If that is the case, you can write "Did not respond to pings."

b. Perform a recursive query of the name `images.nasawestprime.com` using the resolver at `resolver.illinois.edu`. Was this query faster or slower than the sum of the iterative steps? Why do you think that is?

c. Using `dig -x`, perform an iterative reverse-mapping query for the address of `images.nasawestprime.com` you found in the previous steps. Provide the same information as in part a of this question.

d. Looking back on what you've done for the previous parts of this question, can you explain why the DNS protocol typically uses UDP over TCP?