This exam is closed book. However, you may refer to a sheet of 8.5"x11" paper of your own design.

* Keep your answer concise.

* Show your reasoning clearly. If your reasoning is correct, but your final answer is wrong, you will receive most of the credit. If you just show the answer without reasoning, and your answer is wrong, you may receive no points at all.
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1. [37 points] **Short Questions**

Socket API

a) [4 points] What is the intention of setting the backlog parameter of a TCP listening (server) socket?

b) [5 points] Someone claims that UDP offers more control (over TCP) to applications. Can you list two scenarios where UDP offers more control to applications?

c) [5 points] Consider the following code segment of a UDP client:

```java
byte[] receiveData = new byte[1024];

while(true) {
    DatagramPacket receivePacket =
        new DatagramPacket(receiveData, receiveData.length);
    serverSocket.receive(receivePacket);
    String sentence = new String(receivePacket.getData());
}
```

Which line indicates that the code may not be interoperable across platforms?

*Default char set coding. If say 1024 size, -2*
[6 points] Consider three window-based reliability protocols: SR (Selective Repeat); GBN (Go-Back-N); and TCP. Assume that there are only packet losses and corruptions. Please mark true/false for the following statements.

__ SR requires more bits to represent sequence number than GBN for the same window size.

__ SR can be as high as W times more efficient than GBN, where W is the window size, and efficiency is measured by the average number of times that a packet is transmitted.

__ SR requires keeping track of more timeouts than GBN and TCP.

Connection management:

d) [5 points] Someone claims that TCP Three-Way-Handshake (TWH) introduces extra delays and more parameters. In particular, for one-way data transfer, a simplified handshake on the right for current the Internet can be enough. What do you think?
e) [6 points] What is the purpose of the TIME_WAIT state? Is it possible to eliminate it?

![Diagram showing TCP handshake]

f) [6 points] What does it mean when we say that statistical multiplexing can bring in performance advantage? Can you give a concrete setting to demonstrate the quantitative gain?
2. [28 points] DHT and P2P Design

a) [5 points] Distributed Hash Tables are a new networking primitive. An example DHT design we have seen is Chord. In Chord, each node with id maintains a successor table for item ids of \( id + 2^i \), \( i=0, \ldots \). For a query of an item with \( qid \), the next stop is the largest \( id + 2^i \) entry that does not exceed \( qid \). Consider the Chord network shown in the figure above. Please fill in their succ entries.

b) [4 points] Assume that two keys are inserted into the DHT: key 4 and key 7. On which nodes should the two items be stored?

c) [5 points] Assume that a query for \( qid=7 \) is issued at node 1. Please label on the figure how the query is forwarded.
d) [5 points] Someone claims that the Chord design follows the Kleinberg small-world design formula, where the connection probability of two nodes is proportional to \((1/{\text{distance}})^{\text{dimension}}\). In what sense can we say that Chord indeed follows this formula?

e) [5 points] DHT can be used to identify a set of nodes sharing the same content. Suppose that a lookup for key 7 identifies a set of BitTorrent clients with upload capacities \(x_1, x_2, \ldots, x_n\). The clients are downloading files from a server (seed) with upload capacity \(x_0\). Assume that upload is the bottleneck. What is the throughput of each client if it is the traditional client-server architecture (e.g., clients do not exchange data)? What is a good upper bound on the throughput of using BitTorrent?

f) [4 points] There can be multiple issues limiting the preceding upper bound. Can you list two factors that may make achieving the theoretical upper bound difficult?
3. [35 points] Congestion Control and Optimization
Consider the following plot of TCP/Reno window size as a function of time.

![Plot of TCP/Reno window size over time](image)

a) [4 points] Identify the intervals of time when TCP slow start is operating (plus or minus 1 is fine; similar below).

b) [4 points] Identify the intervals of time when TCP congestion avoidance is operating.

c) [4 points] Are there timeouts happened in the Figure?

d) [4 points] If the congestion control protocol is TCP/Vegas, how may the figure look like?

e) [4 points] Can you give two potential reasons for the lacking of TCP/Vegas deployment in the current Internet?
f) [5 points] One concern about TCP/Reno is the coupling between loss rate and throughput. Since many US cross-country backbones have upgraded to 10 Gbps, it is important that the backbone fibers are protected to have low loss rate. Assume segment size of 1000 bytes. What is (approximately) the maximum fiber loss rate for a single TCP/Reno flow to fully utilize a 10 Gbps backbone link with one-way propagation delay of 25 ms?


g) [5 points] Consider a simple network shown below. Four flows running TCP/Reno share two links each with 1 unit of bandwidth. How much bandwidth share will each flow acquire?

\[
\begin{align*}
\text{if misses 2p, } -4
\end{align*}
\]

h) [5 points] What does it mean when we say that TCP/Reno has the utility function below?

\[
U_f(x_f) = -\frac{2}{RTT^2 x_f}
\]