Question 1
Your roommates use Skype to talk to their families in Eastern Europe. They know you are taking CE 150/L this quarter and asked you to explain to them why, when they call home via Skype, the quality of the call varies during the call. (6 points)

(a) What do you tell them?

The Internet uses packet switching in order to share its resources amongst concurrent application flows. As such, flows are serviced on demand, i.e., network resources are not reserved ahead of time. Consequently, as network load changes over time, the call quality may fluctuate. This is especially problematic for real-time applications.

(b) They also observed that when they call home using the traditional telephone network, they have a completely different experience, i.e., the quality of the call is the same throughout the call. They also asked you to explain why that is. What do you tell them?

The traditional phone network is circuit-switched. This means a dedicated circuit is established and resources are reserved for every call. Consequently, the same quality will be experienced during a call independent of network load fluctuations.

Question 2
Network performance (12 points)

(a) Write a mathematical expression for network latency. Explain each term in your expression specifying which ones can vary with time (and why) and which ones don't vary with time (and why). (2 points)

Total delay = propagation + queuing + transmission + processing

- **propagation - based on speed of light (varies with distance)**
  It could be argued that if a path changes then the propagation time might change slightly, but it would probably be negligible
- **queuing - time in router's queue waiting to be serviced and forwarded; varies with congestion so does vary with time.**
- **transmission - time to transmit information out of a node; it is given by the size of the information (e.g., in bits) divided by the link bandwidth (e.g., bits per second). Unless links are replaced/upgraded, transmission time doesn't change with time.**

- **processing - time to process the first packet in the router's queue to decide where to forward packet. It does not vary with time.**

(b) In class, we discussed different ways loss can occur as data is transferred over the network. List and provide a brief explanation of the different types of data loss discussed in class. (2 points)

- **Losses due to packet drops: packets can be dropped by routers when their queues fill up due to network congestion.**
- **Losses due to transmission errors: transmission errors may cause information to get corrupted (e.g., bits may flip) on the way from source to destination.**

(c) Throughput is one of the metrics of network performance we covered in class. How is average throughput defined?

Amount of data transmitted / unit of time; it's usually expressed in terms of bits/sec (bps), bytes/sec (Bps), packets/sec, messages/sec.

(d) What is "bottleneck link"? Describe a scenario exemplifying a bottleneck link.

The bottleneck link is the link with the lowest bandwidth in a connection. Example: If your upload/download rate to the Internet is 10Mbps and all links in between yourself and the intended server are 1Gbps you will be "bottlenecked" by your 10Mbps link, meaning you will not be able to send or receive data faster than 10Mbps.

e) Suppose the UCSC campus network connects to the Internet through a 10Gbps link and the CE 150/L was allocated a dedicated 100 Mbps link to access the UCSC campus network. Alice is a student taking CE 150/L. What is the average throughput of Alice's connection to the Internet when she is the only CE 150/L student connecting to the Internet? Explain.

Alice's average throughput when connecting to the Internet is 100 Mbps, as Alice is bottlenecked by the 100Mbps link from the CE 150/L to the UCSC campus network.

(f) Same scenario as above but now Alice is one of 100 CE 150/L students connecting to the Internet? What is Alice's connection average throughput assuming each user gets a fair share of the networks capacity? Explain. (4 points)

1Mbps (100Mbps/100), as Alice must now share the bottlenecked link between 100 students.

Question 3

In class, we discussed three different locations for Web caches, namely on the client's host, in the Internet service provider's network, and in the content provider's network. Explain how each of these caches can improve performance in terms of user response time, Internet traffic load, and content provider server load. (9 points)

- **Client's host cache: lower user response time; less traffic going on the Internet, and less load on content provider's servers.**
- **ISP cache: lower user response time; less Internet-bound traffic; less load on content provider's servers.**
- **Content provider cache: less load on content provider's servers (assuming content provider cache can either serve the requested content or direct request to lighter loaded servers); possibly somewhat lower response time (request still goes on the Internet but may avoid highly loaded content provider servers).**

Question 4

A video broadcasting service uses a medium-earth orbit (MEO) satellite system whose orbit is at an altitude of 24,000km. Content providers beam content up to the satellite which then sends content to service subscribers. Feedback from subscribers is sent to content providers via a terrestrial back channel with latency of approximately 150ms. (12 points)

(a) What is the round-trip time between content providers and subscribers considering that the propagation speed is 2.4x10^8m/s? Show your work. (4 points)

Notice that, in this scenario, the network topology is a triangle with the satellite, content provider, and subscriber at its vertices.

```
          Sat
         ^   \
   A   /       \  B
```

```
          /              \
       /          C        v
    CP  <------------  SS
```

Propagation time CP -> Sat: (24,000,000m/240,000,000m/s) = 0.1s
Propagation time Sat -> SS: 0.1s
Propagation time SS -> CP = 0.15s

RTT: CP -> Sat -> SS -> CP = 0.1s + 0.1s + 0.15s = 0.35s

(b) For this scenario, what is the channel utilization for Stop-and-Wait with 10Kbit segment size if the channel's data rate is 10 Mbps (assuming no losses)? Show your work. (4 points)

Channel utilization = (size of data/channel data rate)/[(size of data/channel data rate)+RTT] = 10Kbit/10Mbps/(10Kbit/10Mbps+0.35) = (10*10^3/10*10^6)/[(10*10^3/10*10^6)+.35] = 10^-3/10^-3+.35 = .001/.001+.35 = 0.002849 = 0.2849%

(c) Suppose you were hired to increase network utilization by a factor of 10 without incurring any additional cost. You then propose to use "pipelining". Describe how your protocol will work and its additional complexity compared to Stop-and-Wait. (4 points)

Send 10 segments at once, i.e., use a window of 10 segments. But you need sequence numbers to track which ones were correctly received and ACKs that include the sequence number of which segments were correctly received. As ACKs are received, the window can "slide" by the number of segments acknowledged so additional segments can be transmitted.

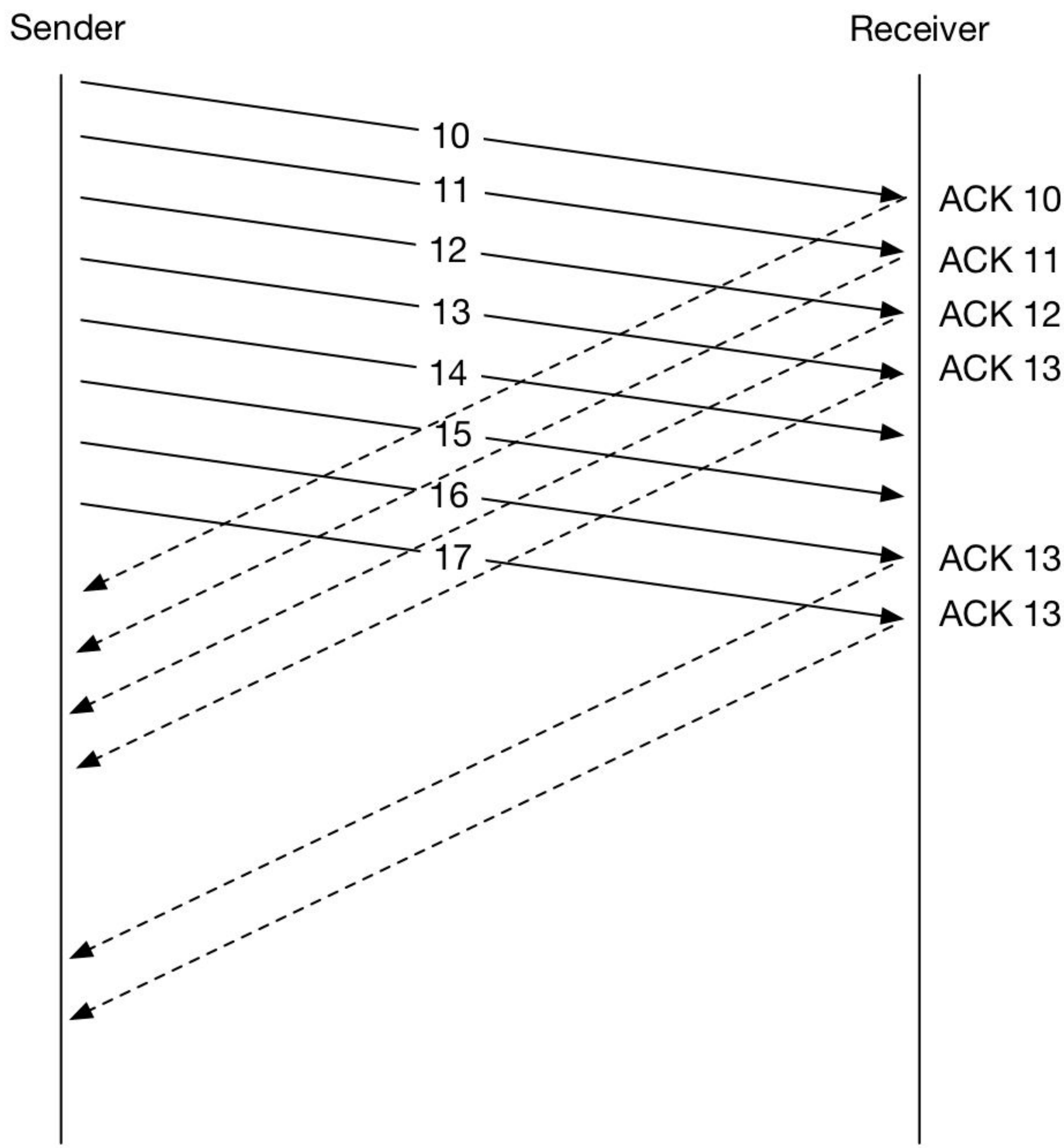
Question 5 ARQ protocols
(a) Suppose that a sender is using an ARQ protocol with a 8-segment window to transfer data reliably to a receiver. Does this protocol use pipelining? Explain. (2 points)

Yes, it has an 8-segment window size.

(b) The ARQ sender sends segments 0 to 4. Then the application generates additional data. Can the ARQ sender send additional segments without receiving acknowledgments for segments 0 to 4? If so, why and which segments? If not, why not? (3 points)

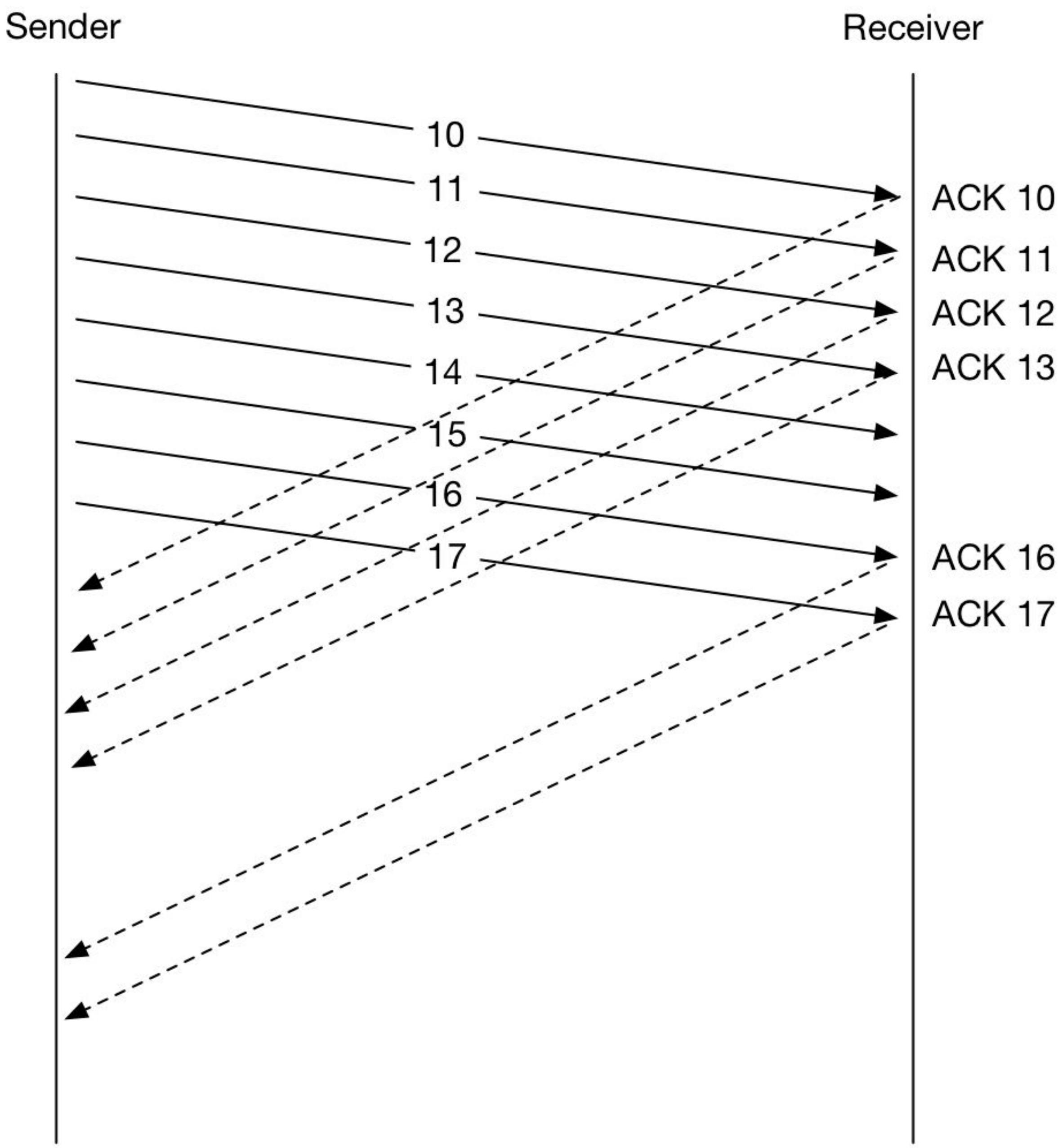
Yes it can send 3 more, 5 through 7, if 4 is inclusive (4 more, 4 through 7, if 4 is not inclusive) because the window size is 8.

(c) Later, the sender sends segments 10 to 17 (having received acknowledgments for all previously sent segments). The receiver receives segments 10, 11, 12, 13, 16, 17 and generates acknowledgments for the received segments. The sender then receives the acknowledgments generated by the receiver. Assume segments that were not received were lost. Draw the time diagram for this scenario if Go- Back-N ARQ is used including the acknowledgments generated by the receiver. Which segment(s) will be retransmitted by the sender, if any? Explain. (3 points)



14, 15, 16, and 17 will be retransmitted.

(d) Suppose the same situation as above but sender and receiver use Selective-Repeat ARQ. Draw the time diagram for this case. Which segment(s) will be retransmitted, if any? Explain. (3 points)



14 and 15 will be retransmitted. Selective repeat allows for out of order segments.

(e) Can cumulative ACKs be used in the scenario described in (c)? And in (d)? Explain. (4 points)

Cumulative ACKs can be used for Go-Back-N as an ACK N represents everything up to N has been received. Selective-Repeat cannot use cumulative ACKs as it won't allow for out of order segments to be ACKed.

(f) Describe the additional function(s) the Selective-Repeat ARQ receiver has to perform when compared to the Go-Back-N receiver. Explain. (3 points)

Selective-Repeat has extra complexity as it has to keep track of which packets were lost and which were received out of order, whereas Go-Back-N only needs to know what was the the highest sequence number acknowledged. The Selective-Repeat receiver also has to buffer out-of-order segments (since they will not be retransmitted by the sender) so that when lost segments are retransmitted, they can be put in the right order and deliver all in-order segments to the application.

Question 6 UDP
(a) If a network application is using UDP as its transport protocol, does it need to establish a connection to the receiver before sending data? Explain. (2 points)

No. UDP is connection-less (or transaction-oriented), since it provides a best-effort delivery service at the transport layer, i.e., it does not guarantee reliable delivery. As such, it does not need to maintain a connection (and connection state) between the communicating end points.

(b) UDP does not guarantee reliable data delivery. However, UDP may carry an optional checksum in its header. What kind of losses do checksums try to recover from? Explain. (2 points)

Checksums can recover from transmission errors (e.g., flipped bits) that can happen as data is transmitted over the network.

(c) How does UDP use its checksum? Describe what takes place both at the UDP sender and UDP receiver. (4 points)

The UDP sender uses a mathematical formula to calculate a checksum for the data that will be transmitted. The sender includes the calculated checksum on the header of the segment to be transmitted. The receiver applies the same formula to the received data and compares the result to the checksum sent by the sender. If the two checksums match, the receiver assumes no transmission errors occurred; otherwise, it discards the segment.

Question 7 TCP (points)
(a) What is TCP’s retransmission timeout? Why should it be set as a function of the round-trip time (RTT)? (3 points)

TCP’s retransmission timeout, or RTO, is the time that TCP waits before deciding that a segment was lost and retransmitting it. The RTO needs to be a function of the RTT to adapt to changing network conditions: the TCP sender should not retransmit without giving enough time for the receiver’s ACK to arrive, nor it should wait too long to recover from a lost segment.

(b) TCP tries to piggyback ACKs onto data segments. How does that work? Why is it used? Are timers needed? (4 points)

Since a TCP connection is full-duplex, each side of the connection tries to use data segments to also carry acknowledgments thereby reducing the amount of signalling overhead. As such, TCP piggybacks the ACK information (i.e., the sequence number being acknowledged) in the same packets as data it is transmitting. To signal to the other end that an ACK is piggybacked on a data segment, TCP sets the ACK bit on the data segment header. Timers are still needed so that an ACK can be generated when there is no data flowing in that direction.

Question 8
The NGO Internet4All has been created to help bridge the so-called “Digital Divide” and will help interconnect villages in remote parts of countries in Africa to the Internet. The project will interconnect the village’s community center, school, local bank, etc. using a 100Mbps local
area network (LAN). However, there will only be enough funds to get each village connected to the Internet through a 10Mbps link. Assume that the delay to retrieve an object from the Internet is on average 2.5s, while the delay to get an object of similar size residing in the LAN is 15ms. You were hired to propose a way to improve the average response time. So you decide to install a Web cache in the village’s network whose miss ratio is 40%. Assume that the time to transmit an object is negligible. (8 points)

(a) What is the average response time to get a page from the Web assuming that on average, Web pages consist of the main page plus 4 embedded objects? Assume non-persistent HTTP is used. Show your work.

We need one RTT for the connection plus one for the request. The RTT to establish the connection will be 2.5s, while the RTT for the request can be reduced by the cache and will be on average 0.015+.4*2.5 = 1.015s. We have one page and four embedded objects for a total of 5 objects, so the total time spent is 5*(2.5 + 1.015) = 17.575s

(b) Same as above using persistent HTTP.

Now we have only one RTT to establish the connection, plus 5 requests. So the total time spent is 2.5 + 5*1.015) = 7.575s

Question 9

Under the program described in Question 8 above, each remote village in Africa will have its own DNS domain using the format: VillageName.CountryCode. Each village will also have its own DNS local name server. (6 points)

(a) Suppose that a user logged to host1 in a village called Tucano in Kenya (tucano.ke) is issuing a request for an object from www.pbs.org. Show the sequence of steps to resolve the Web's server name before the request can be issued. Assume name resolution is done recursively.

Full recursive (half right - root does not do recursion):
host -> local dns server
local dns server -> root server
root server -> .org tld server
.org tld server -> authoritative nameserver for pbs.org
authoritative nameserver for pbs.org -> .org tld server
.org tld server -> root server
root server -> local dns server
local dns server -> host

Recursive at local:
host -> local dns server
local dns server -> root server
root server -> local dns server
local dns server -> .org tld server
.org tld server -> local dns server
local dns server -> authoritative nameserver for pbs.org

authoritative nameserver for pbs.org -> local dns server
local dns server -> host

(b) Subsequently, another user from the same village requests an object from www.fordfoundation.org. Assuming that information about the .org name server was cached by the local DNS name server, show the sequence of steps to resolve www.fordfoundation.org.

host -> local dns nameserver
local dns server -> .org tld server
.org tld server -> local dns server
local dns server -> authoritative nameserver for fordfoundation.org
authoritative nameserver for fordfoundation.org -> local dns server
local dns server -> host

Recursive:
Host -> local DNS NS
Local DNS NS -> .org TLD NS
.org TLD NS -> authoritative NS for fordfoundation.org
authoritative NS for fordfoundation.org -> .org TLD NS
.org TLD NS -> local DNS NS
Local DNS NS -> host