Communication Protocols and Internet Architectures
Harvard University
CSCI S-40

Homework Assignment #1 Solutions
Total points: 17

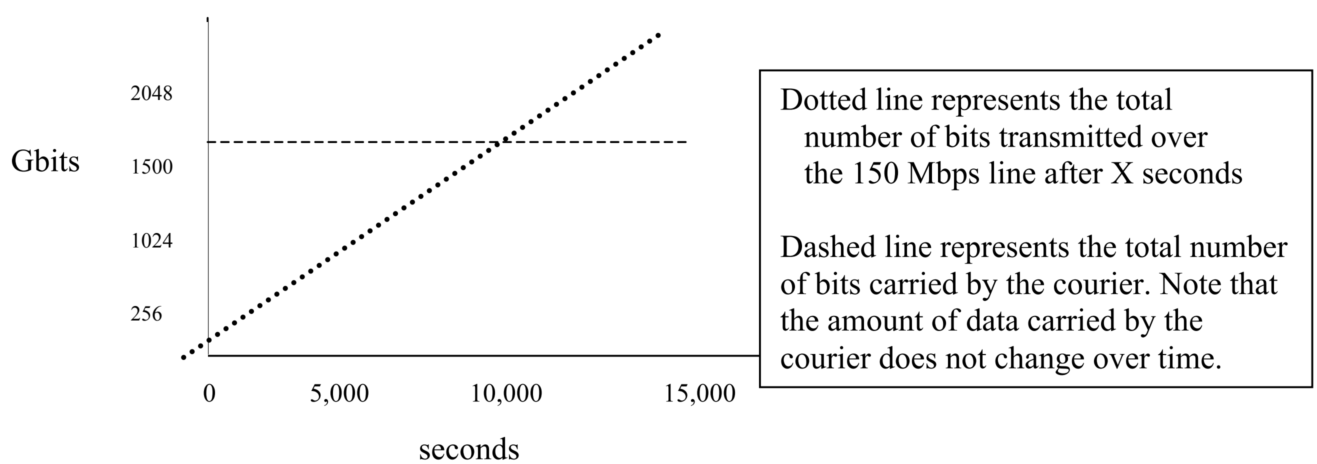
1) Imagine that a bicycle messenger is given three (3) USB memory sticks, each of which contains 64 gigabytes of data. Given that the courier can travel at 20 km per hour through traffic, for what range of distance does the courier have a higher data rate than a transmission line whose data rate (excluding overhead) is 150 Mbps? [2 points]

Answer:
The amount of data to be transmitted is 1536 gigabits. The calculation is as follows:

- 3 USBs contain 192 Gbytes
- 192 GBytes = 192 x 8 bits/byte, or = 1536 gigabits
  (This assumes that 1 GByte equals exactly 10**9 bytes; your answer will be different if you use 2 **30 bytes.)

Given the above, it takes about 10,240 seconds to transmit the same amount of data over a communications line that operates at 150 Mbps. This is computed as follows:

$$\frac{1{,}536{,}000 \text{ Mb}}{150 \text{ Mb/second}} = 10240\ 6{,}826 \text{ seconds (or about 2 hours and 51 minutes)}$$



In 10,240 seconds, the courier can go a distance of approximately 56.8 km. This is computed as follows:

20 km/hour = .00555 km/sec
.00555 km/second x 10,240 seconds = 56.8 km

Therefore for a distance of up to approximately 56.8 km, the courier carries (or transmits) the data faster. After that distance, the courier is still carrying (transmitting) the same amount of data, but the line has now transmitted more data.

2) Assume that you are viewing a web page on the course web site from a computer located at your home or your office…. See homework for complete question [2 points]

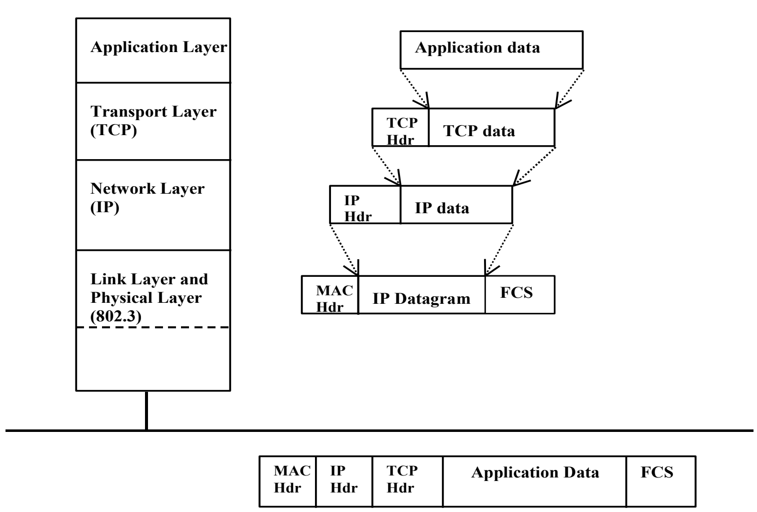
Answer:

Assume for this answer that you are viewing the web page from an office at MIT and that the computer in the office is connected to an ethernet local area network (LAN.) You view the class web page with a browser on this computer using the HTTP protocol, which is an application level protocol. This application layer protocol does not put its information directly onto the ethernet but rather it is passed to TCP, the transport layer protocol. TCP takes the application data and applies its own header information. IP, the network layer protocol then gets the information from TCP, places it in the payload portion of a packet, and adds it's header information. At the data link layer, the 802.3 protocol places the IP information in its payload, adds its own header information and passes it to the physical layer. This process is called encapsulation and it typically takes place at each layer of the protocol stack. (An important point to remember is that the payload of a lower layer is the header and payload of the layer above it) The encapsulation process will be reversed when the packet reaches its destination. Note that IP is an unreliable, connectionless protocol that provides best effort delivery so TCP is used to deliver the web page since TCP provides reliable service.

In the topology described above, it is determined that the destination IP address is not part of the local network so the frame is sent to the default router (a.k.a. gateway) for that network. This router will then look at the destination IP network address, compare that information to the information in its routing table, and determine where to route the packet. Since you are at an office at MIT, the router on your office network will forward the packet to one of the other routers on the MIT campus network, and from there to a router on a network connecting MIT and Harvard. This process will continue until the packet reaches a router that is connected to the Harvard network where the course web server is located.

When the packet reaches this final router in the path (which is where the web server is located), the router builds an ethernet frame (which contains the IP packet) and puts the frame on that LAN. The Ethernet frame will pass through ethernet switches on its way to the server. When it reaches the network interface card (NIC) of the server, the "encapsulation" process begins again, only in reverse order. This way the lower layer protocols know what protocols they need to pass the payload information up to. If any data is lost or corrupted along the way TCP will handle error detection and correction. Note that the process described here will occur for every exchange between the browser and the server, but that this description is just an overview of the steps that take place.

Attached is a simple diagram, which shows the encapsulation process.



3) Join three different IETF mailing lists.

Answer:

You will receive three points for having joined three IETF mailing lists.

4) Explain the use of the sliding window for both flow control and error control. Show how the same sequence numbers that are used for flow control are also used for error control. Illustrate your answer using a time sequence diagram. [3 points]

Answer: A sliding window provides the mechanism necessary for flow control by limiting the number of unacknowledged frames that can be sent before the transmitting station must wait for an acknowledgement. The sequence numbers in the frames are the mechanism used to identify the individual frames and both the transmitter and receiver keep track of them. The number of frames that can be sent before acknowledgement is known as the "window" and once this number has been sent, the window is "closed."

An acknowledgement "opens" up the window, thus allowing more frames to be sent. The sequence numbers in the additional frames are of course larger than the prior frames (until they wrap-around) and the range of sequence numbers is determined by the size of the field. A three-bit field (as used in the HDLC) provides eight sequence numbers and given this range, the window size would be set between one and seven.

The first diagram below shows a simple example of flow control with a window size of two. The window in the second diagram below has a window size of four. This second diagram shows how the sequence number in the received frame is used to identify that a frame was lost and to request retransmission. Note that an ACK of the prior frame that was correctly received would not cause the sender to resend the frame that was lost. A unique type of frame that indicates that there was an error is required. (This is the REJ in the protocol below.)

Diagram 1:

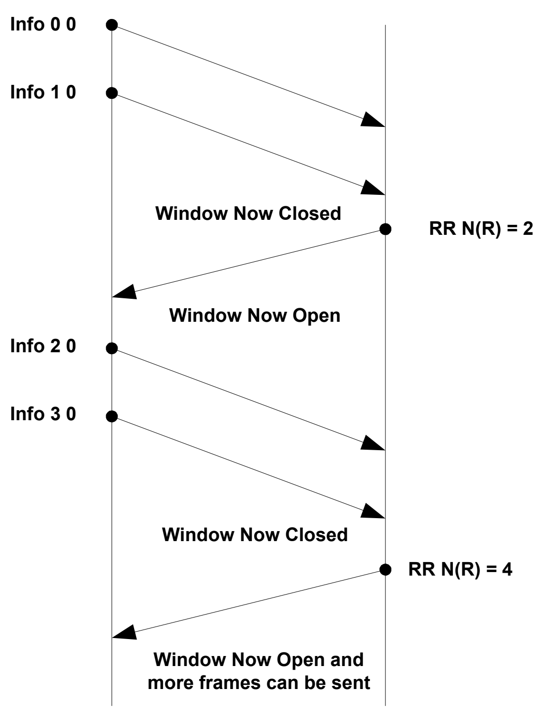
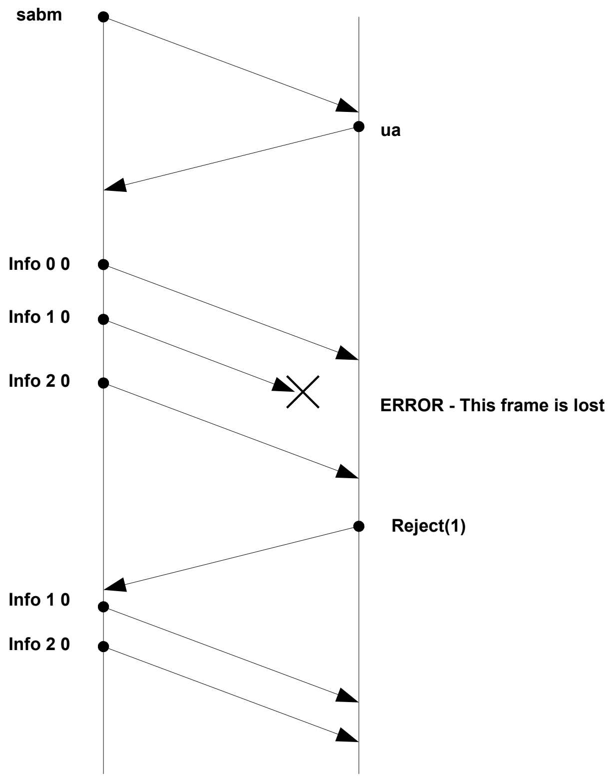


Diagram 2:



5a.) Research the network architecture and technology of cable modems and xDSL service for providing access to the Internet from your home. Describe in detail … see the homework for the complete question. [3 points for this part]
5b.) Research new technology …… Describe in detail … see the homework for the complete question. [1 points for this part]

Answer to Part A.
Assume that a computer is connected to the Internet via a cable modem. It communicates via an Ethernet cable to the Cable Modem, and then via coaxial cable or a combination of coax and fiber to the Cable Modem Termination System at the cable system's head end (in the cable company office).

Frequency Division Multiplexing (FDM) is used to separate channels for the upstream and downstream data transmission on a coax cable system. (Note that we are talking about a coax and not a fiber system.) On a number of cable systems, the upstream data channels are from 5-42 MHz and the downstream data is transmitted in the 550-750 MHz range. The channels are usually 6-8MHz wide. Other channels are of course used for cable TV. (Other types of modulation would be used on a fiber based system; your answer just needed to discuss one type of system.)

In analog cable systems, analog modulation is typically required for the transmission of digital signals over the long coaxial cables. [Optional details: On the downstream channels, QAM-64 is typically used (6 bits/baud or 6 bits/symbol) or, if the cable quality is very good, QAM-256 (8 bits/symbol minus overhead) is used. For upstream channels, there is typically too much noise, so a lower modulation rate is typically used.

On a coax based system, when the computer has data to send, a control packet is first sent by the cable modem to the head end requesting bandwidth, and the head end assigns the appropriate number of mini-slots for the amount of data to be sent, thereby using Time Division Multiplexing (TDM) for this transmission. The minislots do not have to be used in the downstream communication since the head end is the only source, and it can regulate its own communications, i.e., there is no contention for the channel.

In contrast to cable service, xDSL service uses the existing phone line (known as a local loop) between the house and the equipment located in the telephone company's Central Office (CO). However, as compared to a standard telephone line, additional equipment is added at the CO and filters on the telephone line are changed so that almost the entire capacity of the local loop is available for use. With these changes, the bandwidth available for data is approximately 1 MHz or higher. (Note that the bandwidth of the normal telephone channel for telephone server (called POTS) is not changed; it remains approximately 3,100 Hz.) Copper based xDSL service uses frequency division multiplexing (FDM) to divide the frequency spectrum between normal telephone service, and upstream and downstream data service and a number of different modulation schemes. Since the portions of the bandwidth available for upstream and downstream transmission are generally different, the speed of data transmission will be different, and hence the word Asymmetric prefixes the term DSL, i.e., ADSL.

The throughput of the xDSL system over copper wire depends upon which technology is used and the subscriber's distance from the CO, but it can easily be a few Mbps. (Again, this is bandwidth which is split between the up-stream and down-stream paths.) Since ADSL has dedicated bandwidth at the local loop, the transmission rates are more consistent than cable service; however modulation must be optimized on each channel depending on the line quality and noise that is present at that moment, so data rates can and do vary in xDSL. In addition, the service provider may lower the bandwidth of the service to increase the distance from its end office to the customer, or to manage the load on the routers located in the central office and other equipment in its network. (This answer focused on copper-based systems versus FIOS type systems.)

Answer to Part B:
You will receive one point for correctly describing a new technology in this area.

6a.) What is a virtual circuit? Given it is "virtual", how does it compare to a "real" circuit?
6b.) Describe, compare and contrast, connection-oriented and connectionless communication.
6c.) It is very common to hear folks say that connection-oriented communications is the same as circuit switching or circuit-switched communication. Does this statement make sense? How are the two concepts and technologies related to each other, and/or how are they different?
[3 points for entire question]

Answer to 6a : It is helpful and important to understand the historical context when answering this question. Many years ago (let's say 70 or so), a dedicated physical path would have been established between the two parties when a circuit-switched telephone call was placed. (Refer to the picture of the telephone operators in the lecture handout.) This physical circuit was available for the exclusive use of the two parties for the duration of the call; in other words, the circuit was not shared. With the introduction of multiplexing and then packet switching, physical circuits were no longer dedicated to a specific conversation between two parties but rather the circuits were shared between many different users. Each user (or set of users) was allocated just a portion of the available capacity of the physical circuit for a limited period of time, and the circuit was used by other users at other times. However, the users are not aware that the circuit is being shared. They believe that they have a dedicated circuit available to them, where in fact they have a shared "virtual circuit" available for their use. The primary reason to share circuits is that systems built in this way are much more efficient.

As we will learn later in the term, the performance of this virtual circuit, as defined by parameters such as available bandwidth, error rate, delay and jitter, is determined by the technology that is used to implement the shared path. Some technology such as fixed TDM provide a well defined set of performance parameters while the performance of packet switching can vary greatly.

Answer to 6b and 6c : The most common description of connection-oriented communication is that it provides for a reliable, sequenced transmission and delivery of data where a connection (or circuit) between the source and the destination is established before the transmission of the data starts. It is sometimes referred to as circuit-switched although this is overly simplistic and not accurate as described below. TCP is an example of reliable connection-oriented communication.

Connectionless communication is the unreliable transmission and delivery of data where a connection (or circuit) between the source and the destination is not established before the transmission of the data begins. It is sometimes called datagram service. There are no acknowledgments for the packets that are sent, nor is the sequence of the packets guaranteed. This type of service is used where higher protocols take care of the reliability (given that it is required by the user.) IP is the most common example of a connectionless protocol. Note that both of these types of communication (connection oriented and connectionless) provide for addressing and error detection.

There is a fair amount of confusion between the definitions of circuit switched communication and connection oriented communication; they are not the same thing. In this course, when we discuss circuit switched communication we will be talking about how the underlying physical network is implemented. As noted in part a, a dedicated circuit or specific network resources are put in place for use by a pair of end-points for the duration of their conversation. However, it is important to note that the use of these dedicated circuits or network resources does not mean that the communication is reliable: consider that a dial-up modem uses a circuit switched telephone line but it is not reliable. When we talk about connection-oriented communication in this class we mean that the communication between the two end points is reliable. One final point to understand regarding connection-oriented communication is that it does not specify whether one is using circuit switching or packet switching. Although packet switching is almost always used today, either type of switching could be used.