2. a) Single hop, infrastructure-based
   b) Single hop, infrastructure-less
   c) Multi-hop, infrastructure-based
   d) Multi-hop, infrastructure-less

3. Path loss is due to the attenuation of the electromagnetic signal when it travels through matter. Multipath propagation results in blurring of the received signal at the receiver and occurs when portions of the electromagnetic wave reflect off objects and ground, taking paths of different lengths between a sender and receiver. Interference from other sources occurs when the other source is also transmitting in the same frequency range as the wireless network.

5. APs transmit beacon frames. An AP's beacon frames will be transmitted over one of the 11 channels. The beacon frames permit nearby wireless stations to discover and identify the AP.

6. False

7. APs transmit beacon frames. An AP's beacon frames will be transmitted over one of the 11 channels. The beacon frames permit nearby wireless stations to discover and identify the AP.

10. No, there wouldn't be any advantage. Suppose there are two stations that want to transmit at the same time, and they both use RTS/CTS. If the RTS frame is as long as a DATA frames, the channel would be wasted for as long as it would have been wasted for two colliding DATA frames. Thus, the RTS/CTS exchange is only useful when the RTS/CTS frames are significantly smaller than the DATA frames.

11. Initially the switch has an entry in its forwarding table which associates the wireless station with the earlier AP. When the wireless station associates with the new AP, the new AP creates a frame with the wireless station's MAC address and broadcasts the frame. The frame is received by the switch. This forces the switch to update its forwarding table, so that frames destined to the wireless station are sent via the new AP.

18. No. A node can remain connected to the same access point throughout its connection to the Internet (hence, not be mobile). A mobile node is the one that changes its point of attachment into the network over time. Since the user is always accessing the Internet through the same access point, she is not mobile.

21. The home network in GSM maintains a database called the home location register (HLR), which contains the permanent cell phone number and subscriber profile information about each of its subscribers. The HLR also contains information about the current locations of these subscribers. The visited network maintains a database known as the visitor location register (VLR) that contains an entry for each mobile user that is currently in the portion of the network served by the VLR. VLR entries thus come and go as mobile users enter and leave the network.
The edge router in home network in mobile IP is similar to the HLR in GSM and the edge router in foreign network is similar to the VLR in GSM.

Problem 3

$$d_2^1 = \frac{1\times1+(-1)\times(-1)+1\times1+1\times1+1\times1+(-1)\times(-1)+1\times1+1\times1}{8} = 1$$
$$d_2^2 = \frac{1\times1+(-1)\times(-1)+1\times1+1\times1+1\times1+(-1)\times(-1)+1\times1+1\times1}{8} = 1$$

Problem 5

a) The two APs will typically have different SSIDs and MAC addresses. A wireless station arriving to the café will associate with one of the SSIDs (that is, one of the APs). After association, there is a virtual link between the new station and the AP. Label the APs AP1 and AP2. Suppose the new station associates with AP1. When the new station sends a frame, it will be addressed to AP1. Although AP2 will also receive the frame, it will not process the frame because the frame is not addressed to it. Thus, the two ISPs can work in parallel over the same channel. However, the two ISPs will be sharing the same wireless bandwidth. If wireless stations in different ISPs transmit at the same time, there will be a collision. For 802.11b, the maximum aggregate transmission rate for the two ISPs is 11 Mbps.

b) Now if two wireless stations in different ISPs (and hence different channels) transmit at the same time, there will not be a collision. Thus, the maximum aggregate transmission rate for the two ISPs is 22 Mbps for 802.11b.

Problem 6

Suppose that wireless station H1 has 1000 long frames to transmit. (H1 may be an AP that is forwarding an MP3 to some other wireless station.) Suppose initially H1 is the only station that wants to transmit, but that while half-way through transmitting its first frame, H2 wants to transmit a frame. For simplicity, also suppose every station can hear every other station's signal (that is, no hidden terminals). Before transmitting, H2 will sense that the channel is busy, and therefore choose a random backoff value.

Now suppose that after sending its first frame, H1 returns to step 1; that is, it waits a short period of times (DIFS) and then starts to transmit the second frame. H1's second frame will then be transmitted while H2 is stuck in backoff, waiting for an idle channel. Thus, H1 should get to transmit all of its 1000 frames before H2 has a chance to access the channel. On the other hand, if H1 goes to step 2 after transmitting a frame, then it too chooses a random backoff value, thereby giving a fair chance to H2. Thus, fairness was the rationale behind this design choice.

Problem 7

A frame without data is 32 bytes long. Assuming a transmission rate of 11 Mbps, the time to transmit a control frame (such as an RTS frame, a CTS frame, or an ACK frame) is (256 bits)/(11 Mbps) = 23 usec. The time required to transmit the data frame is (8256 bits)/(11 Mbps) = 751

DIFS + RTS + SIFS + CTS + SIFS + FRAME + SIFS + ACK

= DIFS + 3SIFS + (3*23 + 751) usec = DIFS + 3SIFS + 820 usec

Problem 8

a) 1 message/ 2 slots
b) 2 messages/slot
c) 1 message/slot

d) i) 1 message/slot
   ii) 2 messages/slot
   iii) 2 messages/slot

e) i) 1 message/4 slots
   ii) slot 1: Message A→ B, message D→ C
      slot 2: Ack B→ A
      slot 3: Ack C→ D

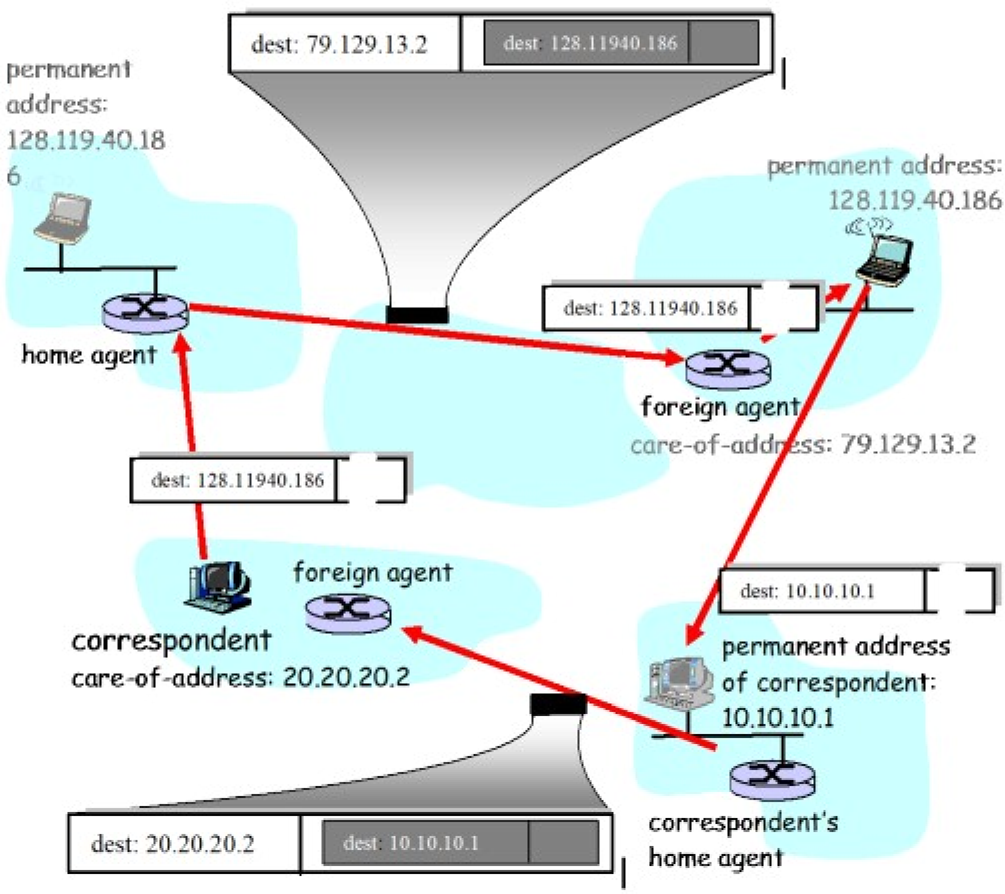
      = 2 messages/ 3 slots

iii)
  slot 1: Message C→ D
  slot 2: Ack D→C, message A→ B
  slot 3: Ack B→ A

Repeat

      = 2 messages/3 slots

Problem 12



If the correspondent is mobile, then any datagrams destined to the correspondent would have to pass through the **correspondent's home agent**. The **foreign agent** in the network being visited would also need to be involved, since it is this foreign agent that notifies the correspondent's home agent of the location of the correspondent. Datagrams received by the correspondent's home agent would need to be encapsulated/tunneled between the correspondent's home agent and foreign agent, (as in the case of the encapsulated diagram at the top of Figure 6.23.

Problem 13

Because datagrams must be first forward to the home agent, and from there to the mobile, the delays will generally be longer than via direct routing. Note that it *is* possible, however, that the direct delay from the correspondent to the mobile (i.e., if the datagram is not routed through the home agent) could actually be smaller than the sum of the delay from the correspondent to the home agent and from there to the mobile. It would depend on the delays on these various path segments. Note that indirect routing also adds a home agent processing (e.g., encapsulation) delay.

Problem 15

Two mobiles could certainly have the same care-of-address in the same visited network. Indeed, if the care-of-address is the address of the foreign agent, then this address would be the same. Once the foreign agent decapsulates the tunneled datagram and determines the address of the mobile, then separate addresses would need to be used to send the datagrams separately to their different destinations (mobiles) within the visited network.

Chapter 9

3. Quantizing a sample into 1024 levels means 10 bits per sample. The resulting rate of the PCM digital audio signal is 160 Kbps.

4. Streaming stored audio/video: In this class of applications, the underlying medium is prerecorded video, such as a movie, a television show, or a prerecorded sporting event. These prerecorded videos are played on servers, and users send requests to the servers to view the videos on demand. Many internet companies today provide streaming video, including YouTube, Netflix, and Hulu.
Conversational Voice- and Video-over-IP: Real-time conversational voice over the Internet is often referred to as Internet telephony, since, from the user's perspective, it is similar to the traditional circuit-switched telephone service. It is also commonly called Voice-over-IP (VOIP).Conversational video is similar except that it includes the video of the participants as well as their voices. Conversational voice and video are widely used in the Internet today, with the Internet companies like Skype and Google Talk boasting hundreds of millions of daily users.
Streaming Live Audio and Video: These applications allow users to receive a live radio or television transmission over the Internet. Today, thousands of radio and television stations around the world are broadcasting content over the internet.

6. The three significant drawbacks of UDP Streaming are:
1. Due to unpredictable and varying amount of available bandwidth between server and client, constant-rate UDP streaming can fail to provide continuous play out.
2. It requires a media control server, such as an RTSP server, to process client-to-server interactivity requests and to track client state for each ongoing client session.
3. Many firewalls are configured to block UDP traffic, preventing users behind these firewalls from receiving UDP video.

7. No. On the client side, the client application reads bytes from the TCP receive buffer and places the bytes in the client application buffer.

8. The initial buffering delay is tp = Q/x = 4 seconds.

9. End-to-end delay is the time it takes a packet to travel across the network from source to destination. Delay jitter is the fluctuation of end-to-end delay from packet to the next packet.

12. RTP streams in different sessions: different multicast addresses; RTP streams in the same session: SSRC field; RTP packets are distinguished from RTCP packets by using distinct port numbers.

13. The role of a SIP registrar is to keep track of the users and their corresponding IP addresses which they are currently using. Each SIP registrar keeps track of the users that belong to its domain. It also forwards INVITE messages (for users in its domain) to the IP address which the user is currently using. In this regard, its role is similar to

that of an authoritative name server in DNS.

Problem 6

a) $160+h$ bytes are sent every 20 msec. Thus the transmission rate is
$\frac{(160+h)\cdot 8}{20} Kbps = (64+.4h) Kbps$

b)

IP header: 20 bytes
UDP header: 8 bytes
RTP header: 12 bytes

c) h=40 bytes (a 25% increase in the transmission rate!)

Problem 7

a) Denote $d^{(n)}$ for the estimate after the nth sample.

$$d^{(1)} = r_4 - t_4$$

$$d^{(2)} = u(r_3 - t_3) + (1-u)(r_4 - t_4)$$

$$d^{(3)} = u(r_2 - t_2) + (1-u)\left[u(r_3 - t_3) + (1-u)(r_4 - t_4)\right]$$

$$= u(r_2 - t_2) + (1-u)u(r_3 - t_3) + (1-u)^2(r_4 - t_4)$$

$$d^{(4)} = u(r_1 - t_1) + (1-u)d^{(3)}$$

$$= u(r_1 - t_1) + (1-u)u(r_2 - t_2) + (1-u)^2 u(r_3 - t_3) + (1-u)^3(r_4 - t_4)$$

b)

$$d^{(n)} = u\sum_{j=1}^{n-1}(1-u)^j(r_j - t_j) + (1-u)^n(r_n - t_n)$$

c)

$$d^{(\infty)} = \frac{u}{1-u}\sum_{j=1}^{\infty}(1-u)^j(r_j - t_j)$$

$$= \frac{1}{9}\sum_{j=1}^{\infty}.9^j(r_j - t_j)$$

The weight given to past samples decays exponentially.

Problem 9

a) $r1 - t1 + r2 - t2 + \ldots + rn\text{-}1\text{-}tn\text{-}1 = (n\text{-}1)dn\text{-}1$

Substituting this into the expression for dn gives

$$d_n = \frac{n-1}{n} d_{n-1} + \frac{r_n - t_n}{n}$$

b) The delay estimate in part (a) is an average of the delays. It gives equal weight to recent delays and to "old" delays. The delay estimate in Section 6.3 gives more weight to recent delays; delays in the distant past have relatively little impact on the estimate.

Problem 11

a) The delay of packet 2 is 7 slots. The delay of packet 3 is 9 slots. The delay of packet 4 is 8 slots. The delay of packet 5 is 7 slots. The delay of packet 6 is 9 slots. The delay of packet 7 is 8 slots. The delay of packet 8 is > 8 slots.

b) Packets 3, 4, 6, 7, and 8 will not be received in time for their playout if playout begins at t-8.

c) Packets 3 and 6 will not be received in time for their playout if playout begins at t=9.

d) No packets will arrive after their playout time if playout time begins at t=10.

Problem 12

The answers to parts a and b are in the table below:

Packet Number	$r_i - t_i$	d_i	v_i
1	7	7	0
2	8	7.10	0.09
3	8	7.19	0.162
4	7	7.17	0.163
5	9	7.35	0.311
6	9	7.52	0.428
7	8	7.57	0.429
8	8	7.61	0.425

Problem 13

a) Both schemes require 25% more bandwidth. The first scheme has a playback delay of 5 packets. The second scheme has a delay of 2 packets.

b) The first scheme will be able to reconstruct the original high-quality audio encoding. The second scheme will use the low quality audio encoding for the lost packets and will therefore have lower overall quality.

c) For the first scheme, many of the original packets will be lost and audio quality will be very poor. For the second scheme, every audio chunk will be available at the receiver, although only the low quality version will be available for every other chunk. Audio quality will be acceptable.

Problem 15

a) As discussed in Chapter 2, UDP sockets are identified by the two-tuple consisting of destination IP address and destination port number. So the two packets will indeed pass through the same socket.

b) Yes, Alice only needs one socket. Bob and Claire will choose different SSRC's, so Alice will be able distinguish between the two streams. Another question we could have asked is: How does Alice's software know which stream (i.e. SSRC) belongs to Bob and which stream belongs to Alice? Indeed, Alice's software may want to display the sender's name when the sender is talking. Alice's software gets the SSRC to name mapping from the RTCP source description reports.

Problem 16

a) True

b) True

c) No, RTP streams can be sent to/from any port number. See the SIP example in Section 6.4.3

d) No, typically they are assigned different SSRC values.

e) True

f) False, she is indicating that she wishes to receive GSM audio

g) False, she is indicating that she wishes to receive audio on port 48753

h) True, 5060 for both source and destination port numbers

i) True

j) False, this is a requirement of H.323 and not SIP.