WIRELESS COMMUNICATION NETWORKS AND SYSTEMS

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SOLUTIONS MANUAL

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Chapter 2 Transmission Fundamentals

ANSWERS TO QUESTIONS

2.1 A continuous or analog signal is one in which the signal intensity varies in a smooth fashion over time while a discrete or digital signal is one in which the signal intensity maintains one of a finite number of constant levels for some period of time and then changes to another constant level.

2.2 Amplitude, frequency, and phase are three important characteristics of a periodic signal.

2.3 2π radians.

2.4 The relationship is $\lambda f = v$, where λ is the wavelength, f is the frequency, and v is the speed at which the signal is traveling.

2.5 The spectrum of a signal consists of the frequencies it contains; the bandwidth of a signal is the width of the spectrum.

2.6 Attenuation is the gradual weakening of a signal over distance.

2.7 The rate at which data can be transmitted over a given communication path, or channel, under given conditions, is referred to as the channel capacity.

2.8 Bandwidth, noise, and error rate affect channel capacity.

2.9 With guided media, the electromagnetic waves are guided along an enclosed physical path, whereas unguided media provide a means for transmitting electromagnetic waves through space, air, or water, but do not guide them.

2.10 Point-to-point microwave transmission has a high data rate and less attenuation than twisted pair or coaxial cable. It is affected by rainfall, however, especially above 10 GHz. It is also requires line of sight and is subject to interference from other microwave transmission, which can be intense in some places.

2.11 Direct broadcast transmission is a technique in which satellite video signals are transmitted directly to the home for continuous operation.

2.12 A satellite must use different uplink and downlink frequencies for continuous operation in order to avoid interference.

2.13 Broadcast is omnidirectional, does not require dish shaped antennas, and the antennas do not have to be rigidly mounted in precise alignment.

2.14 Multiplexing is cost-effective because the higher the data rate, the more cost-effective the transmission facility.

2.15 Interference is avoided under frequency division multiplexing by the use of guard bands, which are unused portions of the frequency spectrum between subchannels.

2.16 A synchronous time division multiplexer interleaves bits from each signal and takes turns transmitting bits from each of the signals in a round-robin fashion.

ANSWERS TO PROBLEMS

2.1 Period = 1/1000 = 0.001 s = 1 ms.

2.2 **a.** $\sin(2\pi ft - \pi) + \sin(2\pi ft + \pi) = 2\sin(2\pi ft + \pi)$ or $2\sin(2\pi ft - \pi)$ or $-2\sin(2\pi ft)$
b. $\sin(2\pi ft) + \sin(2\pi ft - \pi) = 0$.

2.3

N	C		D		E		F		G		A		B		C
F	264		297		330		352		396		440		495		528
D		33		33		22		44		44		55		33	
W	1.25		1.11		1		0.93		0.83		0.75		0.67		0.63

N = note; F = frequency (Hz); D = frequency difference; W = wavelength (m)

2.4 $2\sin(4\pi t + \pi)$; $A = 2$, $f = 2$, $\phi = \pi$

2.5 $(1 + 0.1\cos 5t)\cos 100t = \cos 100t + 0.1\cos 5t\cos 100t$. From the trigonometric identity $\cos a\cos b = (1/2)(\cos(a + b) + \cos(a - b))$, this equation can be rewritten as the linear combination of three sinusoids:
$\cos 100t + 0.05\cos 105t + 0.05\cos 95t$

2.6 We have $\cos^2 x = \cos x \cos x = (1/2)(\cos(2x) + \cos(0)) = (1/2)(\cos(2x) + 1)$. Then: $f(t) = (10\cos t)^2 = 100\cos^2 t = 50 + 50\cos(2t)$. The period of $\cos(2t)$ is π and therefore the period of $f(t)$ is π.

2.7 If $f_1(t)$ is periodic with period X, then $f_1(t) = f_1(t + X) = f_1(t + nX)$ where n is an integer and X is the smallest value such that $f_1(t) = f_1(t + X)$. Similarly, $f_2(t) = f_2(t + Y) = f_2(t + mY)$. We have $f(t) = f_1(t) + f_2(t)$. If $f(t)$ is periodic with period Z, then $f(t) = f(t + Z)$. Therefore $f_1(t) + f_2(t) = f_1(t + Z) + f_2(t + Z)$. This last equation is satisfied if $f_1(t) = f_1(t$

+ Z) and $f_2(t) = f_2(t + Z)$. This leads to the condition Z = nX = mY for some integers n and m. We can rewrite this last as (n/m) = (Y/X). We can therefore conclude that if the ratio (Y/X) is a rational number, then f(t) is periodic.

2.8 The signal would be a low-amplitude, rapidly changing waveform.

2.9 Using Shannon's equation: $C = B \log_2 (1 + SNR)$
We have W = 300 Hz $(SNR)_{dB} = 3$
Therefore, $SNR = 10^{0.3}$
$C = 300 \log_2 (1 + 10^{0.3}) = 300 \log_2 (2.995) = 474$ bps

2.10 Using Nyquist's equation: $C = 2B \log_2 M$
We have C = 9600 bps
a. $\log_2 M = 4$, because a signal element encodes a 4-bit word
Therefore, $C = 9600 = 2B \times 4$, and
B = 1200 Hz
b. $9600 = 2B \times 8$, and B = 600 Hz

2.11 Nyquist analyzed the theoretical capacity of a noiseless channel; therefore, in that case, the signaling rate is limited solely by channel bandwidth. Shannon addressed the question of what signaling rate can be achieved over a channel with a given bandwidth, a given signal power, and in the presence of noise.

2.12 **a.** Using Shannon's formula: $C = 3000 \log_2 (1+400000) = 56$ Kbps
b. Due to the fact there is a distortion level (as well as other potentially detrimental impacts to the rated capacity, the actual maximum will be somewhat degraded from the theoretical maximum. A discussion of these relevant impacts should be included and a qualitative value discussed.

2.13 $C = B \log_2 (1 + SNR)$
$20 \times 10^6 = 3 \times 10^6 \times \log_2(1 + SNR)$
$\log2(1 + SNR) = 6.67$
$1 + SNR = 102$
$SNR = 101$

2.14 From Equation 2.1, we have $L_{dB} = 20 \log (4\pi d/\lambda) = 20 \log (4\pi df/v)$, where $\lambda f = v$ (see Question 2.4). If we double either d or f, we add a term 20 log(2), which is approximately 6 dB.

2.15

Decibels	1	2	3	4	5	6	7	8	9	10
Losses	0.8	0.63	0.5	0.4	0.32	0.25	0.2	0.16	0.125	0.1
Gains	1.25	1.6	2	2.5	3.2	4.0	5.0	6.3	8.0	10

2.16 For a voltage ratio, we have

$$N_{dB} = 30 = 20 \log(V_2/V_1)$$
$$V_2/V_1 = 10^{30/20} = 10^{1.5} = 31.6$$

2.17 Power (dBW) = 10 log (Power/1W) = 10 log20 = 13 dBW

Chapter 3 Communication Networks

ANSWERS TO QUESTIONS

3.1 Wide area networks (WANs) are used to connect stations over very large areas that may even be worldwide while local area networks (LANs) connect stations within a single building or cluster of buildings. Ordinarily, the network assets supporting a LAN belong to the organization using the LAN. For WANs, network assets of service providers are often used. LANs also generally support higher data rates than WANs.

3.2 It is advantageous to have more than one possible path through a network for each pair of stations to enhance reliability in case a particular path fails.

3.3 Telephone communications.

3.4 Static routing involves the use of a predefined route between any two end points, with possible backup routes to handle overflow. In alternate routing, multiple routes are defined between two end points and the choice can depend on time of day and traffic conditions.

3.5 This is a connection to another user set up by prior arrangement, and not requiring a call establishment protocol. It is equivalent to a leased line.

3.6 In the **datagram** approach, each packet is treated independently, with no reference to packets that have gone before. In the **virtual circuit** approach, a preplanned route is established before any packets are sent. Once the route is established, all the packets between a pair of communicating parties follow this same route through the network.

3.7 It is not efficient to use a circuit switched network for data since much of the time a typical terminal-to-host data communication line will be idle. Secondly, the connections provide for transactions at a constant data rate, which limits the utility of the network in interconnecting a variety of host computers and terminals.

3.8 If the video is having errors, there may be a high packet loss rate. If the video is pausing frequently for more buffering, the average data rate may be too low.

ANSWERS TO PROBLEMS

3.1 a. Circuit Switching

Total= $C_1 + C_2 + C_3$ where

C_1 = Call Setup Time

C_2 = Message Delivery Time

C_3 = Call Teardown Time

C_1 = S = 0.2

C_2 = Propagation Delay + Transmission Time

= $N \times D + L/B$

= $4 \times 0.001 + 3000/9600 = 0.3165$

C_3 = T = 0.02

Total= 0.2 + 0.3165 + 0.02 = 0.5365 sec

Datagram Packet Switching

There are P – H = 1080 – 80 = 1000 data bits per packet. A message of 3000 bits requires three packets (3000 bits/1000 bits/packet = 3 packets).

Total= $D_1 + D_2 + D_3 + D_4$ where

D_1 = Time to Transmit and Deliver all packets through first hop

D_2 = Time to Deliver last packet across second hop

D_3 = Time to Deliver last packet across third hop

D_4 = Time to Deliver last packet across fourth hop

D_1 = $3 \times t + p$ where

t = transmission time for one packet

p = propagation delay for one hop

D_1 = $3 \times (P/B) + D$

= $3 \times (1080/9600) + 0.001$

= 0.3385

D_2 = $D_3 = D_4 = t + p$

= $(P/B) + D$

= $(1080/9600) + 0.001 = 0.1135$

T = 0.3385 + 0.1135 + 0.1135 + 0.1135

= 0.6790 sec

Virtual Circuit Packet Switching

T = $V_1 + V_2 + V_3$ where

V_1 = Call Setup Time

V_2 = Datagram Packet Switching Time

V_3 = Call Teardown Time

T = S + 0.6790 +T = 0.2 + 0.6790 + 0.02 = 0.8990 sec

b. Circuit Switching vs. Diagram Packet Switching

T_c = End-to-End Delay, Circuit Switching

T_c = $S + N \times D + L/B + T$

T_d = End-to-End Delay, Datagram Packet Switching

N_p = Number of packets = $\left\lceil \frac{L}{P - H} \right\rceil$

T_d = $D_1 + (N - 1)D_2$

D_1 = Time to transmit and send all other packets from first hop

D_2 = Time to deliver last packet through N hops

D_1 = $(N_p - 1)(P/B)$

D_2 = $(P/B + D) \times N$

T_d = $(N_p - 1 + N)(P/B) + N \times D$

T_c = T_d

$S + L/B = (N_p + N - 1)(P/B)$

Circuit Switching vs. Virtual Circuit Packet Switching

T_V = End-to-End Delay, Virtual Circuit Packet Switching

T_V = $S + T_d + T$

$T_C = T_V$

$L/B = (N_p + N - 1)(P/B)$

Datagram vs. Virtual Circuit Packet Switching

$T_d = T_V$

T_d = $S + T_d + T$

Can never be true with S>0 or T>0

3.2 From Problem 3.1, we have

$$T_d = (N_p + N - 1)(P/B) + N \times D$$

For maximum efficiency, we assume that $N_p = L/(P - H)$ is an integer. Also, it is assumed that D = 0. Thus

$$T_d = (L/(P - H) + N - 1)(P/B)$$

To minimize as a function of P, take the derivative:

$$0 = dT_d/(dP)$$
$$0 = (1/B)(L/(P - H) + N - 1) - (P/B)L/(P - H)^2$$
$$0 = L(P - H) + (N - 1)(P - H)^2 - LP$$
$$0 = -LH + (N - 1)(P - H)^2$$

$(P - H)^2 = LH/(N - 1)$

$P = H + \sqrt{\dfrac{LH}{N - 1}}$

3.3 From above, we have

$T_d = (N_p + N - 1)(P/B) + N \times D$

$T_c = S + N \times D + L/B + T$

$T_d < T_c$

$S + N \times D + L/B + T < (N_p + N - 1)(P/B) + N \times D$

$S + L/B + T < (N_p + N - 1)(P/B)$

$0.6825 < 0.1125*(2+N)$

$N > 4.067$

Since N is an integer and is greater than 4.067, $N \geq 5$.

3.4 Total times for datagram switching and virtual circuit switching now become

$T_{Dat} = T_D$ = processing time for datagram switching

T_{VC} = processing time for datagram switching

$T_d = (N_p + N - 1)(P/B + T_{Dat}) + N \times D$

$T_V = S + (N_p + N - 1)(P/B + T_{VC}) + N \times D + T$

$T_V < T_d$

$S + (N_p + N - 1)(P/B + T_{VC}) + N \times D + T < (N_p + N - 1)(P/B + T_{Dat}) + N \times D$

$S + (N_p + N - 1)(P/B + T_{VC}) + T < (N_p + N - 1)(P/B + T_{Dat})$

$0.2 + (3 + 4 - 1)(1080/9600 + 0.0035) + 0.02 < (3 + 4 - 1)(1080/9600 + T_{Dat})$

$0.916 < 6(0.1125 + T_{Dat})$

$T_{Dat} > 0.04017$ s

3.5 For what range numbers of packets would the end-to-end delay to transfer all of the packets be less for virtual circuit switching than the datagram approach?

Say X is the number of packets.

For virtual circuit switching, end-to-end delay is

$T_V = S + (X + N - 1)(P/B + T_{VC}) + N \times D + T$

For datagram switching it is

$T_d = (X + N - 1)(P/B + T_{Dat}) + N \times D$

For virtual circuit switching to take less time

$S + (X + N - 1)(P/B + T_{VC}) + N \times D + T < (X + N - 1)(P/B + T_{Dat}) + N \times D$

$S + (X + N - 1)(P/B + T_{VC}) + T < (X + N - 1)(P/B + T_{Dat})$

$0.2 + (1080/9600 + 0.003)(X+3) + 0.02 < (1080/9600 + 0.0145)(X + 3)$

$0.22 < (0.0115)(X + 3)$

$X > 16.13$

Virtual circuit switching is better for 17 or more packets

For what range of sizes of the total message length does this correspond?

Note the following reordering and simplification of terms from above.

$S + (X + N - 1)(P/B + T_{VC}) + N \times D + T < (X + N - 1)(P/B + T_{Dat}) + N \times D$

$S + (X + N - 1)(P/B) + (X + N - 1)(T_{VC}) + N \times D + T$

$< (X + N - 1)(P/B) + (X + N - 1)(T_{Dat}) + N \times D$

$S + (X + N - 1)(T_{VC}) + T < (X + N - 1)(T_{Dat})$

$S + T < (X + N - 1)(T_{Dat} - T_{VC})$

$0.22 < (0.0115)(X + 3)$

$X > 16.13$

Transmission times (P/B) are the same for both approaches, so the terms cancel.

If we have a smaller packet size P_{last} for the 17th and last packet. The inequality becomes:

$S + (X - 1)(P/B) + (N)(P_{last}/B) + (X + N - 1)(T_{VC}) + N \times D + T$

$< (X - 1)(P/B) + (N)(P_{last}/B) + (X + N - 1)(T_{Dat}) + N \times D$

Again the P/B and P_{last}/B terms cancel and the dependence is only on the number of packets as above.

$S + (X + N - 1)(T_{VC}) + T < (X + N - 1)(T_{Dat})$

$X > 16.13$

Any message length which requires 17 packets makes the inequality favor virtual circuit switching. Hence, a message length of $16 \times (P - H)+1 =$ 16001 bits creates 17 packets.

Answer: Message length $L \geq 16001$ bits

3.6 Each telephone makes 0.5 calls/hour at 6 minutes each. Thus a telephone occupies a circuit for 3 minutes per hour. Twenty telephones can share a circuit (although this 100% utilization implies long queuing delays). Since 10% of the calls are long distance, it takes 200 telephones to occupy a long distance (4 kHz) channel full time. The interoffice trunk has $10^6/(4 \times 10^3) = 250$ channels. With 200 telephones per channel, an end office can support $200 \times 250 = 50{,}000$ telephones.

3.7 The argument ignores the overhead of the initial circuit setup and the circuit teardown.

3.8 Yes. A large noise burst could create an undetected error in the packet. If such an error occurs and alters a destination address field or virtual circuit identifier field, the packet would be misdelivered.

3.9 The number of hops is one less than the number of nodes visited.

- **a.** The fixed number of hops is 2.
- **b.** The furthest distance from a station is halfway around the loop. On average, a station will send data half this distance. For an N-node network, the average number of hops is (N/4) – 1.
- **c.** 1.

Chapter 4 Protocols and the TCP/IP Suite

ANSWERS TO QUESTIONS

4.1 The network access layer is concerned with the exchange of data between a computer and the network to which it is attached.

4.2 The transport layer is concerned with data reliability and correct sequencing.

4.3 A protocol is the set of rules or conventions governing the way in which two entities cooperate to exchange data.

4.4 A PDU is the combination of data from the next higher communications layer and control information.

4.5 The software structure that implements the communications function. Typically, the protocol architecture consists of a layered set of protocols, with one or more protocols at each layer.

4.6 Transmission Control Protocol/Internet Protocol (TCP/IP) are two protocols originally designed to provide low level support for internetworking. The term is also used generically to refer to a more comprehensive collection of protocols developed by the U.S. Department of Defense and the Internet community.

4.7 Layering decomposes the overall communications problem into a number of more manageable subproblems.

4.8 A router is a device that operates at the Network layer of the OSI model to connect dissimilar networks.

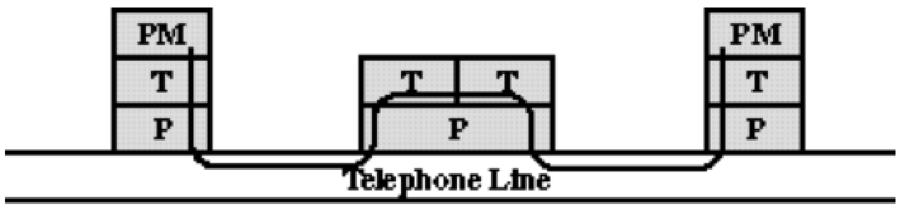
ANSWERS TO PROBLEMS

4.1 The guest effectively places the order with the cook. The host communicates this order to the clerk, who places the order with the cook. The phone system provides the physical means for the order to be transported from host to clerk. The cook gives the pizza to the clerk with the order form (acting as a "header" to the pizza). The clerk boxes the pizza with the delivery address, and the delivery van encloses all of the orders to be delivered. The road provides the physical path for delivery.

4.2 **a.**

The PMs speak as if they are speaking directly to each other. For example, when the French PM speaks, he addresses his remarks directly to the Chinese PM. However, the message is actually passed through two translators via the phone system. The French PM's translator translates his remarks into English and telephones these to the Chinese PM's translator, who translates these remarks into Chinese.

b.



An intermediate node serves to translate the message before passing it on.

4.3 Perhaps the major disadvantage is the processing and data overhead. There is processing overhead because as many as seven modules (OSI model) are invoked to move data from the application through the communications software. There is data overhead because of the appending of multiple headers to the data. Another possible disadvantage is that there must be at least one protocol standard per layer. With so many layers, it takes a long time to develop and promulgate the standards.

4.4 No. There is no way to be assured that the last message gets through, except by acknowledging it. Thus, either the acknowledgment process continues forever, or one army has to send the last message and then act with uncertainty.

4.5 A case could be made either way. **First**, look at the functions performed at the network layer to deal with the communications network (hiding the details from the upper layers). The network layer is responsible for routing data through the network, but with a broadcast network, routing is not needed. Other functions, such as sequencing, flow control, error control between end systems, can be accomplished at layer 2, because the link layer will be a protocol directly between the two end systems, with no intervening switches. So it would seem that a network layer is not needed. **Second**, consider the network layer from the point of view of the upper layer using it. The upper layer sees itself attached to an access point into a network supporting communication with multiple devices. The layer for assuring that data sent across a network is delivered to one of a number of other end systems is the network layer. This argues for inclusion of a network layer.

In fact, the OSI layer 2 is split into two sublayers. The lower sublayer is concerned with medium access control (MAC), assuring that only one end system at a time transmits; the MAC sublayer is also responsible for addressing other end systems across the LAN. The upper sublayer is called Logical Link Control (LLC). LLC performs traditional link control functions. With the MAC/LLC combination, no network layer is needed (but an internet layer may be needed).

4.6 The internet protocol can be defined as a separate layer. The functions performed by IP are clearly distinct from those performed at a network layer and those performed at a transport layer, so this would make good sense. This results in 8 layers.

The session and transport layer both are involved in providing an end-to-end service to the OSI user, and could easily be combined. This has been done in TCP/IP, which provides a direct application interface to TCP. This results in 6 layers

4.7 **a.** No. This would violate the principle of separation of layers. To layer (N – 1), the N-level PDU is simply data. The (N – 1) entity does not know about the internal format of the N-level PDU. It breaks that PDU into fragments and reassembles them in the proper order.

b. Each N-level PDU must retain its own header, for the same reason given in (a).

4.8 Data plus transport header plus internet header equals 1820 bits. This data is delivered in a sequence of packets, each of which contains 24 bits of network header and up to 776 bits of higher-layer headers and/or data. Three network packets are needed. Total bits delivered = $1820 + 3 \times 24 = 1892$ bits.

4.9 UDP provides the source and destination port addresses and a checksum that covers the data field. These functions would not normally be performed by protocols above the transport layer. Applications can't provide port addresses (they aren't really addresses). Port-ids have to be unambiguous between the application and the layer (or OS). If they were provided by the application then the same port-id could be assigned by different applications and layer (or OS) couldn't distinguish them. UDP therefore is a necessity. Thus UDP provides a useful, though limited, service.

4.10 In the case of IP and UDP, these are unreliable protocols that do not guarantee delivery, so they do not notify the source. TCP does guarantee delivery. However, the technique that is used is a timeout. If the source does not receive an acknowledgment to data within a given period of time, the source retransmits.

4.11 UDP has a fixed-sized header. The header in TCP is of variable length.

Chapter 5 Overview of Wireless Communications

ANSWERS TO QUESTIONS

5.1 An **isotropic antenna** is a point in space that radiates power in all directions equally.

5.2 A radiation pattern is a graphical representation of the radiation properties of an antenna as a function of space coordinates.

5.3 The term *fading* refers to the time variation of received signal power caused by changes in the transmission medium or path(s).

5.4 **Diffraction** occurs at the edge of an impenetrable body that is large compared to the wavelength of the radio wave. The edge in effect become a source and waves radiate in different directions from the edge, allowing a beam to bend around an obstacle. If the size of an obstacle is on the order of the wavelength of the signal or less, **scattering** occurs. An incoming signal is scattered into several weaker outgoing signals in unpredictable directions.

5.5 **Fast fading** refers to changes in signal strength between a transmitter and receiver as the distance between the two changes by a small distance of about one-half a wavelength. **Slow fading** refers to changes in signal strength between a transmitter and receiver as the distance between the two changes by a larger distance, well in excess of a wavelength.

5.6 **Flat fading**, or nonselective fading, is that type of fading in which all frequency components of the received signal fluctuate in the same proportions simultaneously. **Selective fading** affects unequally the different spectral components of a radio signal.

5.7 Cost, capacity utilization, and security and privacy are three major advantages enjoyed by digital transmission over analog transmission.

5.8 With amplitude-shift keying, binary values are represented by two different amplitudes of carrier frequencies. This approach is susceptible to sudden gain changes and is rather inefficient.

5.9 QAM takes advantage of the fact that it is possible to send two different signals simultaneously on the same carrier frequency, by using two copies of the carrier frequency, one shifted by 90˚ with respect to the other. For QAM, each carrier is ASK modulated.

5.10 A parity bit appended to an array of binary digits to make the sum of all the binary digits, including the parity bit, always odd (odd parity) or always even (even parity).

5.11 The CRC is an error detecting code in which the code is the remainder resulting from dividing the bits to be checked by a predetermined binary number.

5.12 The CRC has more bits and therefore provides more redundancy. That is, it provides more information that can be used to detect errors.

5.13 Detection of errors and retransmission of frames that are received in error.

5.14 Go-back-N ARQ is a form of error control in which a destination station sends a negative acknowledgment (NAK) when it receives an error. The source station receiving the NAK will retransmit the frame in error plus all succeeding frames transmitted in the interim.

5.15

- Orthogonal Frequency Division Multiplexing (OFDM) is a scheme that divides a broadband signal into multiple, orthogonal, low bit rate parallel transmissions.
- Orthogonal Frequency Division Multiple Access (OFDMA) is an OFDM MAC mechanism where multiple users share a channel by using different OFDM subcarriers.
- Single-carrier FDMA (SC-FDMA) is a variant of OFDM that performs extra DFT operations at the transmitter and receiver. This benefits the mobile user to provide better battery life, efficiency, and lower cost. It does not support multiple access; at any given point in time, all of the subcarriers must be dedicated to one user.

5.16 The bandwidth is wider after the signal has been encoded using spread spectrum.

5.17 **(1)** We can gain immunity from various kinds of noise and multipath distortion. **(2)** It can also be used for hiding and encrypting signals. Only a recipient who knows the spreading code can recover the encoded information. **(3)** Several users can independently use the same higher bandwidth with very little interference, using code division multiple access (CDMA).

5.18 With frequency hopping spread spectrum (FHSS), the signal is broadcast over a seemingly random series of radio frequencies, hopping from frequency to frequency at fixed intervals. A receiver, hopping between frequencies in synchronization with the transmitter, picks up the message.

5.19 With direct sequence spread spectrum (DSSS), each bit in the original signal is represented by multiple bits in the transmitted signal, using a spreading code.

5.20 CDMA allows multiple users to transmit over the same wireless channel using spread spectrum. Each user uses a different spreading code. The receiver picks out one signal by matching the spreading code.

ANSWERS TO PROBLEMS

5.1

Distance (km)	Radio (dB)	Wire (dB)
1	–6	–3
2	–12	–6
4	–18	–12
8	–24	–24
16	–30	–48

5.2 We have $\lambda f = c$; in this case $\lambda \times 30 = 3 \times 10^8$ m/sec, which yields a wavelength of 10,000 km. Half of that is 5,000 km which is comparable to the east-to-west dimension of the continental U.S. While an antenna this size is impractical, the U.S. Defense Department has considered using large parts of Wisconsin and Michigan to make an antenna many kilometers in diameter.

5.3 **a.** Using $\lambda f = c$, we have $\lambda = (3 \times 10^8 \text{ m/sec})/(300 \text{ Hz}) = 1{,}000$ km, so that $\lambda/2 = 500$ km.

b. The carrier frequency corresponding to $\lambda/2 = 1$ m is given by:
$f = c/\lambda = (3 \times 10^8 \text{ m/sec})/(2 \text{ m}) = 150$ MHz.

5.4 $\lambda = 2 \times 2.5 \times 10^{-3} \text{ m} = 5 \times 10^{-3} \text{ m}$
$f = c/\lambda = (3 \times 10^8 \text{ m/sec})/(5 \times 10^{-3} \text{ m}) = 6 \times 10^{10} \text{ Hz} = 60 \text{ GHz}$

5.5 $L_{dB} = 20 \log(f_{\mathrm{MHz}}) + 120 + 20 \log (d_{\mathrm{km}}) + 60 - 147.56$
$= 20 \log(f_{\mathrm{MHz}}) + 20 \log (d_{\mathrm{km}}) + 32.44$

5.6 **a.** From Appendix 2A, Power_{dBW} = 10 log (Power_{W}) = 10 log (50) = 17 dBW
Power_{dBm} = 10 log (Power_{mW}) = 10 log (50,000) = 47 dBm

b. Using Equation (5.2),
$L_{dB} = 20 \log(900 \times 10^6) + 20 \log (100) - 147.56 = 120 + 59.08 + 40 - 147.56 = 71.52$
Therefore, received power in dBm = 47 – 71.52 = –24.52 dBm

c $L_{dB} = 120 + 59.08 + 80 - 147.56 = 111.52$; $P_{r,dBm} = 47 - 111.52 = -64.52$ dBm

d The antenna gain results in an increase of 3 dB, so that $P_{r,dBm} = -61.52$ dBm

Source: [RAPP02]

5.7 From Equation 2.2, the ratio of transmitted power to received power is
$P_t/P_r = (4\pi d/\lambda)^2$
If we double the frequency, we halve λ, or if we double the distance, we double d, so the new ratio for either of these events is:
$P_t/P_{r2} = (8\pi d/\lambda)^2$
Therefore:
$10 \log (P_r/P_{r2}) = 10 \log (2^2) = 6$ dB

5.8 Any arithmetic scheme will work if applied in exactly the same way to the forward and reverse process. The modulo 2 scheme is easy to implement in circuitry. It also yields a remainder one bit smaller than binary arithmetic.

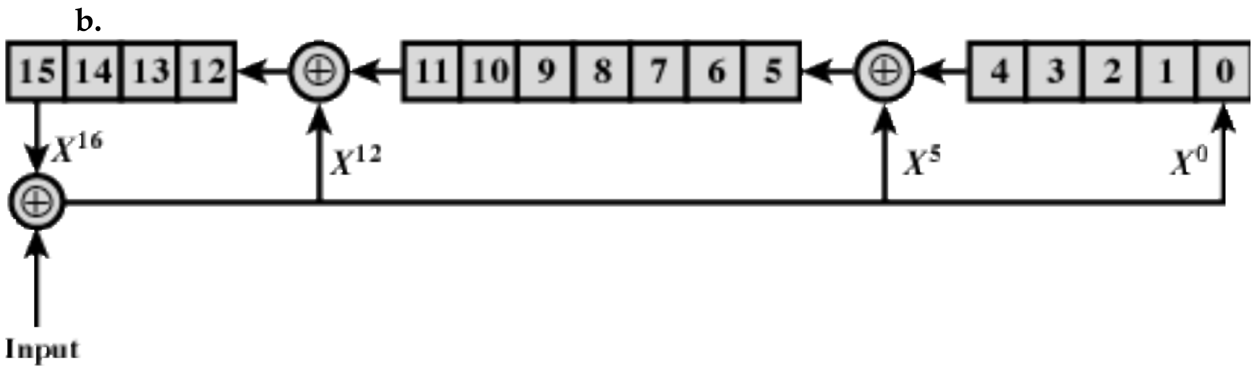
5.9 **a.** We have:
Pr [single bit in error] = 10^{-3}
Pr [single bit not in error] = $1 - 10^{-3} = 0.999$
Pr [8 bits not in error] = $(1 - 10^{-3})^8 = (0.999)^8 = 0.992$
Pr [at least one error in frame] = $1 - (1 - 10^{-3})^8 = 0.008$

b. Pr [at least one error in frame] = $1 - (1 - 10^{-3})^{10} = 1 - (0.999)^{10} = 0.01$

5.10 **a.**

```
                                  1000100010011000
10001000000100001/10000000000000000000000000000000
                  10001000000100001
                      10000000100001000
                      10001000000100001
                          10010001100010000
                          10001000000100001
                             11001100110001000
                             10001000000100001
                              10001001101010010
                              10001000000100001
                    remainder =   0001101110011000
```

b.



Shift	Shift Register																Input
	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	
1	0	0	0	1	0	0	0	0	0	0	1	0	0	0	0	1	1
2	0	0	1	0	0	0	0	0	0	1	0	0	0	0	1	0	0
3	0	1	0	0	0	0	0	0	1	0	0	0	0	1	0	0	0
4	1	0	0	0	0	0	0	1	0	0	0	0	1	0	0	0	0
5	0	0	0	1	0	0	1	0	0	0	1	1	0	0	0	1	0
6	0	0	1	0	0	1	0	0	0	1	1	0	0	0	1	0	0
7	0	1	0	0	1	0	0	0	1	1	0	0	0	1	0	0	0
8	1	0	0	1	0	0	0	1	1	0	0	0	1	0	0	0	0
9	0	0	1	1	0	0	1	1	0	0	1	1	0	0	0	1	0
10	0	1	1	0	0	1	1	0	0	1	1	0	0	0	1	0	0
11	1	1	0	0	1	1	0	0	1	1	0	0	0	1	0	0	0
12	1	0	0	0	1	0	0	1	1	0	1	0	1	0	0	1	0
13	0	0	0	0	0	0	1	1	0	1	1	1	0	0	1	1	0
14	0	0	0	0	0	1	1	0	1	1	1	0	0	1	1	0	0
15	0	0	0	0	1	1	0	1	1	1	0	0	1	1	0	0	0
16	0	0	0	1	1	0	1	1	1	0	0	1	1	0	0	0	0

CRC

5.11

```
                  10110110
        ______________
110011 / 11100011000000
         110011
           101111
           110011
            111000
            110011
              101100
              110011
               111110
               110011
      CRC =      11010
```

5.12

a.

	00000	10101	01010
00000	0	2	2
10101	3	0	5
01010	2	5	0

b.

	000000	010101	101010	110110
000000	0	3	3	4
010101	3	0	6	6
101010	3	6	0	3
110110	4	6	3	0

5.13 Suppose that the minimum distance between codewords is at least $2t + 1$. For a codeword **w** to be decoded as another codeword **w'**, the received sequence must be at least as close to **w'** as to **w**. For this to happen, at least $t + 1$ bits of **w** must be in error. Therefore all errors involving t or fewer digits are correctable.

5.14 **a.** Because only one frame can be sent at a time, and transmission must stop until an acknowledgment is received, there is little effect in increasing the size of the message if the frame size remains the same. All that this would affect is connect and disconnect time.

b. Increasing the number of frames would decrease frame size (number of bits/frame). This would lower line efficiency, because the propagation time is unchanged but more acknowledgments would be needed.

c. For a given message size, increasing the frame size decreases the number of frames. This is the reverse of (b).

5.15 $1/66.67 \times 10^{-6} = 15000$ symbols/sec = 15000 bits/sec per subcarrier
$(18 \times 10^6$ bits/sec)/(15000 bits/sec) = 1200 subcarriers

5.16 For 15000 bits/sec per subcarrier
$B_s = 15000$ Hz
180 kHz/15 kHz = 12 subcarriers per resource block

5.17 **a.** MFSK
b. $L = 2$
c. $M = 2^L = 4$
d. $k = 3$
e. slow FHSS
f. $2^k = 8$
g.

Time	0	1	2	3	4	5	6	7	8	9	10	11
Input data	0	1	1	1	1	1	1	0	0	0	1	0
Frequency	f_1		f_3		f_3		f_2		f_0		f_2	

Time	12	13	14	15	16	17	18	19
Input data	0	1	1	1	1	0	1	0
Frequency	f_1		f_3		f_2		f_2	

Chapter 6 The Wireless Channel

ANSWERS TO QUESTIONS

6.1 The two functions of an antenna are: (1) For transmission of a signal, radio-frequency electrical energy from the transmitter is converted into electromagnetic energy by the antenna and radiated into the surrounding environment (atmosphere, space, water); (2) for reception of a signal, electromagnetic energy impinging on the antenna is converted into radio-frequency electrical energy and fed into the receiver.

6.2 An **isotropic antenna** is a point in space that radiates power in all directions equally.

6.3 A sending directional antenna pattern has a main lobe directs most of the energy. It also has sidelobes that carry some unintended energy to the sides and back of the antenna array. The pattern also includes nulls between the main lobe and sidelobes in which very little energy is sent. A receiving antenna pattern includes the same features, except energy is received in a directional manner instead of sent.

6.4 A radiation pattern is a graphical representation of the radiation properties of an antenna as a function of space coordinates.

6.5 A parabolic antenna creates, in theory, a parallel beam without dispersion. In practice, there will be some beam spread. Nevertheless, it produces a highly focused, directional beam.

6.6 Effective area and wavelength.

6.7 Theoretical and measurement-based models have shown that beyond a certain distance the average received signal power decreases by a factor of $1/d^n$. This factor n is called the path loss exponent.

6.8 Free space loss.

6.9 **Thermal noise** is due to thermal agitation of electrons. **Intermodulation noise** produces signals at a frequency that is the sum or difference of the two original frequencies or multiples of those frequencies. **Crosstalk** is the unwanted coupling between signal paths. **Impulse noise** is noncontinuous, consisting of irregular pulses or noise spikes of short duration and of relatively high amplitude.

6.10 Refraction is the bending of a radio beam caused by changes in the speed of propagation at a point of change in the medium.

6.11 Multipath fading causes the signal to vary with location due to the combination of delayed multipath signal arrivals.

6.12 **Diffraction** occurs at the edge of an impenetrable body that is large compared to the wavelength of the radio wave. The edge in effect become a source and waves radiate in different directions from the edge, allowing a beam to bend around an obstacle. If the size of an obstacle is on the order of the wavelength of the signal or less, **scattering** occurs. An incoming signal is scattered into several weaker outgoing signals in unpredictable directions.

6.13 **Fast fading** refers to changes in signal strength between a transmitter and receiver as the distance between the two changes by a small distance of about one-half a wavelength. **Slow fading** refers to changes in signal strength between a transmitter and receiver as the distance between the two changes by a larger distance, well in excess of a wavelength.

6.14 **Flat fading**, or nonselective fading, is that type of fading in which all frequency components of the received signal fluctuate in the same proportions simultaneously. **Selective fading** affects unequally the different spectral components of a radio signal.

6.15 **Space diversity** involves the physical transmission path and typical refers to the use of multiple transmitting or receiving antennas. With **frequency diversity**, the signal is spread out over a larger frequency bandwidth or carried on multiple frequency carriers. **Time diversity** techniques aim to spread the data out over time so that a noise burst affects fewer bits.

ANSWERS TO PROBLEMS

6.1

Distance (km)	Radio (dB)	Wire (dB)
1	–6	–3
2	–12	–6
4	–18	–12
8	–24	–24
16	–30	–48

6.2 The length of a half-wave dipole is one-half the wavelength of the signal that can be transmitted most efficiently. Therefore, the optimum wavelength in this case is λ = 20 m. The optimum free space frequency is $f = c/\lambda = (3 \times 10^8)/20 = 15$ MHz.

6.3 **a.** Using $\lambda f = c$, we have $\lambda = (3 \times 10^8 \text{ m/sec})/(300 \text{ Hz}) = 1{,}000$ km, so that $\lambda/2 = 500$ km.

b. The carrier frequency corresponding to $\lambda/2 = 1$ m is given by:
$f = c/\lambda = (3 \times 10^8 \text{ m/sec})/(2 \text{ m}) = 150$ MHz.

6.4 **a.** First, take the derivative of both sides of the equation $y^2 = 2px$:

$$\frac{dy}{dx}y^2 = \frac{dy}{dx}(2px);\ \ 2y\frac{dy}{dx} = 2p;\ \ \frac{dy}{dx} = \frac{p}{y}$$

Therefore $\tan \beta = (p/y_1)$.

b. The slope of PF is $(y_1 - 0)/(x_1 - (p/2))$. Therefore:

$$\tan a = \frac{\dfrac{y_1}{x_1 - \dfrac{p}{2}} - \dfrac{p}{y_1}}{1 + \dfrac{y_1}{x_1 - \dfrac{p}{2}}\dfrac{p}{y_1}} = \frac{y_1^2 - px_1 + \dfrac{1}{2}p^2}{x_1 y_1 - \dfrac{1}{2}py_1 + py_1}$$

Because $y_1^2 = 2px_1$, this simplifies to $\tan \alpha = (p/y_1)$.

6.5 $L_{dB} = 20 \log(f_{\text{MHz}}) + 120 + 20 \log(d_{\text{km}}) + 60 - 147.56$
$= 20 \log(f_{\text{MHz}}) + 20 \log(d_{\text{km}}) + 32.44$

6.6 **a.** From Appendix 2A, $\text{Power}_{\text{dBW}} = 10 \log(\text{Power}_{\text{W}}) = 10 \log(50) = 17$ dBW
$\text{Power}_{\text{dBm}} = 10 \log(\text{Power}_{\text{mW}}) = 10 \log(50{,}000) = 47$ dBm

b. Using Equation (5.2),
$L_{dB} = 20 \log(900 \times 10^6) + 20 \log(100) - 147.56 = 120 + 59.08 + 40 - 147.56 = 71.52$
Therefore, received power in dBm = 47 – 71.52 = –24.52 dBm

c $L_{dB} = 120 + 59.08 + 80 - 147.56 = 111.52$; $P_{r,\text{dBm}} = 47 - 111.52 = -64.52$ dBm

d The antenna gain results in an increase of 3 dB, so that $P_{r,\text{dBm}} = -61.52$ dBm

Source: [RAPP02]

6.7 a. From Table 6.5: 2.7 to 3.5

b. Using Equation (6.6),
$L_{dB} = 20 \log(900 \times 10^6) + 10n \log(100) - 147.56 = 120 + 59.08 + 62 - 147.56 = 93.52$
Therefore, received power in dBm = 47 – 93.52 = –46.52 dBm

c. $L_{dB} = 120 + 59.08 + 124 - 147.56 = 155.52$; $P_{r,\text{dBm}} = 47 - 155.52 = -108.52$ dBm

d. The antenna gain results in an increase of 3 dB, so that $P_{r,\text{dBm}} = -105.52$ dBm

6.8 Using dBm calculations, the transmit power is 10 log(2W/.001) = 33 dBm

$L_{dB} \geq 33 - (-105) = 138$ dB

$L_{dB} = 20 \log(1.8 \times 10^9) + 10n \log(5200) - 147.56 \geq 138$

$n \geq (138 - 185.11 + 147.56) / 37.16$

$n \geq 2.703$

6.9 a. From Table 5.2, $G = 7A/\lambda^2 = 7Af^2/c^2 = (7\times\pi\times(0.6)^2\times(2\times10^9)^2]/(3\times10^8)^2 = 351.85$

$G_{dB} = 25.46$ dB

b. 0.1 W x 351.85 = 35.185 W

c. Use $L_{dB} = 20 \log(4\pi) + 20 \log(d) + 20 \log(f) - 20 \log(c) - 10 \log(G_r) - 10 \log(G_t)$

$L_{dB} = 21.98 + 87.6 + 186.02 - 169.54 - 25.46 - 25.46 = 75.14$ dB

The transmitter power, in dBm is 10 log (100) = 20.

The available received signal power is 20 – 75.14 = –55.14 dBm

6.10 From Equation 2.2, the ratio of transmitted power to received power is

$P_t/P_r = (4\pi d/\lambda)^2$

If we double the frequency, we halve λ, or if we double the distance, we double d, so the new ratio for either of these events is:

$P_t/P_{r2} = (8\pi d/\lambda)^2$

Therefore:

$10 \log(P_r/P_{r2}) = 10 \log(2^2) = 6$ dB

6.11 $L_{dB}(\text{suburban}) = L_{dB}(\text{urban small/medium city}) - 2[\log(f_c/28)]^2 - 5.4$

$L_{dB}(\text{urban small/medium city}) = 69.55 + 26.16(\log f_c) - 13.82 \log h_t - A(h_r) + (44.9 - 6.55 \log h_t) \log d$

$A(h_r) = (1.1 \log f_c - 0.7) h_r - (1.56 \log f_c - 0.8)$

$= (1.1 \log 900 - 0.7) \times 1.5 - (1.56 \log 900 - 0.8) = 0.0159$

$L_{dB}(\text{suburban}) = 69.55 + 26.16(\log f_c) - 13.82 \log h_t - A(h_r)$

$+ (44.9 - 6.55 \log h_t) \log d - 2[\log(f_c/28)]^2 - 5.4 = 134.0$ dB

6.12 This problem is not well formed and there is an error in the text for the Okumura model on p. 176, line 4. "+ 18.33 (log f_c)…" should replace "– 18.733 (log f_c)…"

Transmit power in dBW = $10 \log(150 \times 10^3) = 51.76$ dBW

Received power in dBW = $10 \log(1 \times 10^{-13}) = -130$ dBW

$L_{dB} = 51.76 - (-130) = 181.76$ dBW

$L_{dB}(\text{open}) = L_{dB}(\text{urban small/medium city}) - 4.78(\log f_c)^2 - 18.733(\log f_c) - 40.98$

$L_{dB}(\text{urban small/medium city}) = 69.55 + 26.16(\log f_c) - 13.82 \log h_t - A(h_r) + (44.9 - 6.55 \log h_t) \log d$

$A(h_r) = (1.1 \log f_c - 0.7) h_r - (1.56 \log f_c - 0.8)$

$= (1.1 \log 76 - 0.7) \times 1.5 - (1.56 \log 76 - 0.8) = -0.0807$

L_{dB}(open) = 69.55 + 26.16 (log f_c) – 13.82 log h_t – (–0.0807)

+ (44.9 – 6.55 log h_t) log d – 4.78 (log f_c) ^{2} – 18.733 (log f_c) – 40.98

L_{dB}(open) = 69.55 + 26.16 (log f_c) – (–0.0807)

+ 44.9 log d– 4.78 (log f_c) ^{2} – 18.733 (log f_c) – 40.98 + log h_t (– 13.82– 6.55 log d)

181.76 = 111.16 – 26.29 (log h_t)

10 ^ ((111.16 – 181.76) / 26.29) = h_t = 0.00206 m, which is an absurd result. Even a 0.002 meter tall antenna could achieve a 181.76 dB path loss., according to this calculation.

However, the text has the following error:
Page 176, line 4: Should be "+ 18.33 (log f_c)…" instead of "– 18.733 (log f_c)…"
This would create the following result:

L_{dB}(open) = 69.55 + 26.16 (log f_c) – (–0.0807)

+ 44.9 log d– 4.78 (log f_c) ^{2} + 18.33 (log f_c) – 40.98 + log h_t (– 13.82– 6.55 log d)

181.76 = 180.87 – 45.94 (log h_t)

10 ^ ((180.87 – 181.76) / 26.29) = h_t = 0.925 m, which is still not a realistic result.

If the received power requirement were to be 10^{-8} W

L_{dB} = 51.76 – (–80) = 131.76 dBW

10 ^ ((180.87 – 131.76) / 26.29) = h_t = 73.79 m

6.13 N = –228.6 dBW + 10 log T + 10 log B

We have T = 273.15 + 50 = 323.15 K, and B = 10,000

N = –228.6 dBW + 25.09 +40 = –163.51 dBW

Converting to watts, $N_W = 10^{N/10} = 4 \times 10^{-17}$ W

6.14 **a.** Output waveform:

$\sin(2\pi f_1 t) + 1/3 \sin(2\pi(3f_1)t) + 1/5 \sin(2\pi(5f_1)t) + 1/7 \sin(2\pi(7f_1)t)$

where $f_1 = 1/T = 1$ kHz

Output power = 1/2 (1 + 1/9 + 1/25 + 1/49) = 0.586 watt

b. Output noise power = 8 kHz × 0.1 μWatt/Hz = 0.8 mWatt

SNR = 0.586/0.0008 = 732.5 $(SNR)_{dB}$ = 28.65

6.15 **a.** Check slow fading, using a factor of 10 for much, much greater.

$T_b = 1/r_s = 1/50 \times 10^3 = 20$ μs

$T_c \gg T_b$?

$T_c > 10\ T_b$?

Test condition: 10 ms > 200 μs?

This is true, so slow fading.

b. Assume $B_S \approx r_S$ = 50 kHz. To check for slow fading, test the following,

$B_C \gg B_S$?

$B_C > 10B_S$?

Test condition: 600 kHz > 500 kHz?

This is true, so flat fading.

6.16 Assume $B_S \approx r_S$. To check for slow fading, test the following,

$B_C \gg B_S$?

$B_C > 10B_S$?

To have flat fading, $B_S < B_C/10$

B_S < 10 kHz

r_S < 10 kbps

6.17 For k signals, $P_b \approx 0.15^{k} (0.1) \approx 10^{-4}$

$k = \log (10^{-4}/0.1)/ \log (0.15) = 3.64$

More than 3.64 branches are needed, so $k \geq 4$.

6.18 Let *RI* = refractive index, α = angle of incidence, β = angle of refraction

$(\sin \alpha)/\sin \beta) = RI_{air}/RI_{water} = 1.0003/(4/3) = 0.75$

$\sin \beta = 0.5/0.75 = 0.66$; $\beta = 41.8°$

Chapter 7 Signal Encoding Techniques

ANSWERS TO QUESTIONS

7.1 In differential encoding, the signal is decoded by comparing the polarity of adjacent signal elements rather than determining the absolute value of a signal element.

7.2 A modem converts digital information into an analog signal, and conversely.

7.3 Cost, capacity utilization, and security and privacy are three major advantages enjoyed by digital transmission over analog transmission.

7.4 With amplitude-shift keying, binary values are represented by two different amplitudes of carrier frequencies. This approach is susceptible to sudden gain changes and is rather inefficient.

7.5 Non return-to-zero-level (NRZ-L) is a data encoding scheme in which a negative voltage is used to represent binary one and a positive voltage is used to represent binary zero. A disadvantage of NRZ transmission is that it is difficult to determine where one bit ends and the next bit begins.

7.6 The difference is that offset QPSK introduces a delay of one bit time in the Q stream

7.7 QAM takes advantage of the fact that it is possible to send two different signals simultaneously on the same carrier frequency, by using two copies of the carrier frequency, one shifted by 90° with respect to the other. For QAM, each carrier is ASK modulated.

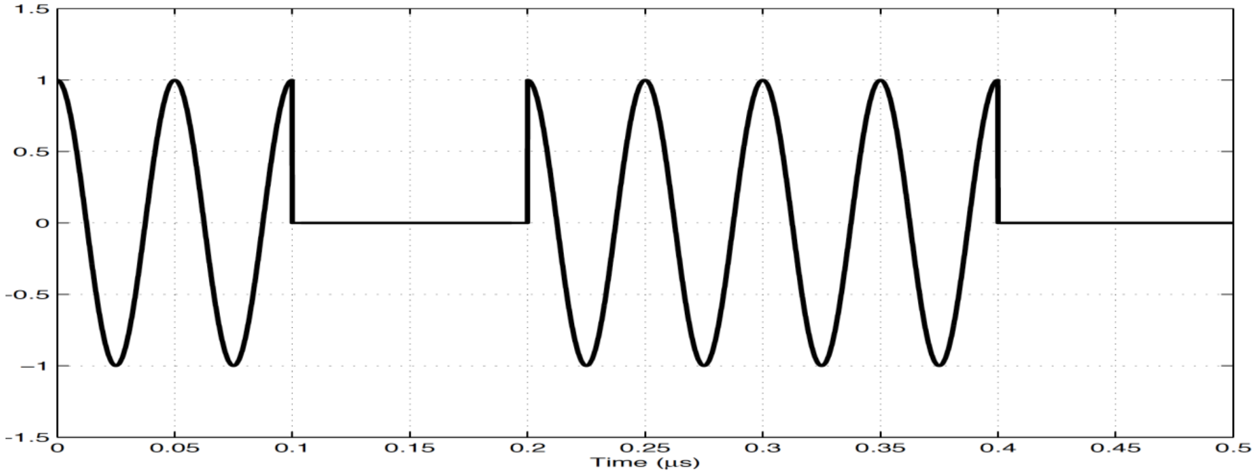
7.8 The sampling rate must be higher than twice the highest signal frequency.

7.9 Frequency modulation (FM) and phase modulation (PM) are special cases of angle modulation. For PM, the phase is proportional to the modulating signal. For FM, the derivative of the phase is proportional to the modulating signal.

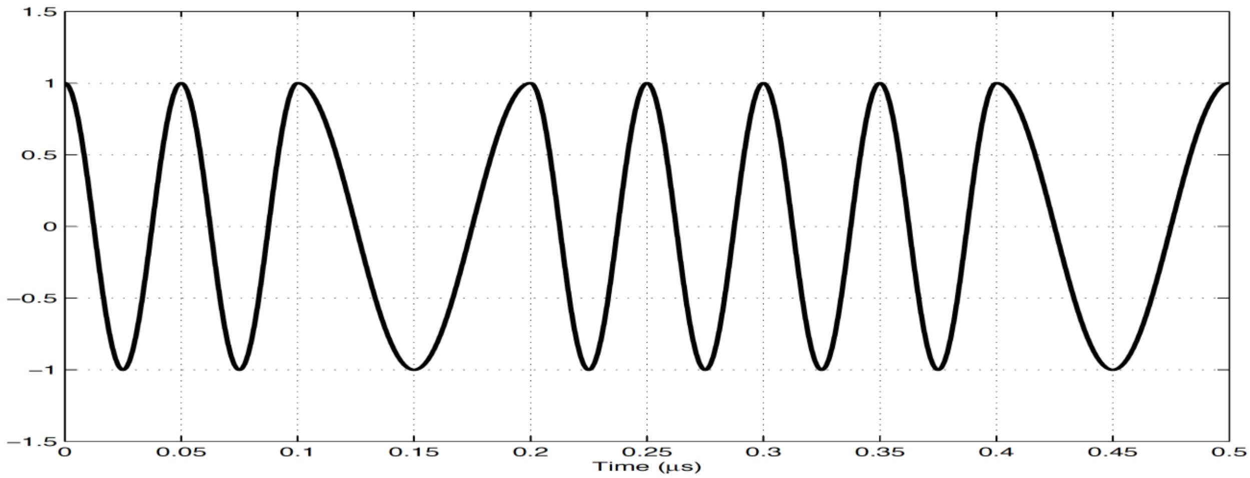
7.10 A waveform encoder takes the voice waveform and samples the values, maybe providing compression by only encoding differences between samples or between predictions. A vocoder detects the characteristics of the human voice and only transmits the associated parameters. The vocoder typically provides the lower bit rate.

ANSWERS TO PROBLEMS

7.1 All plots use the cosine function.



7.2

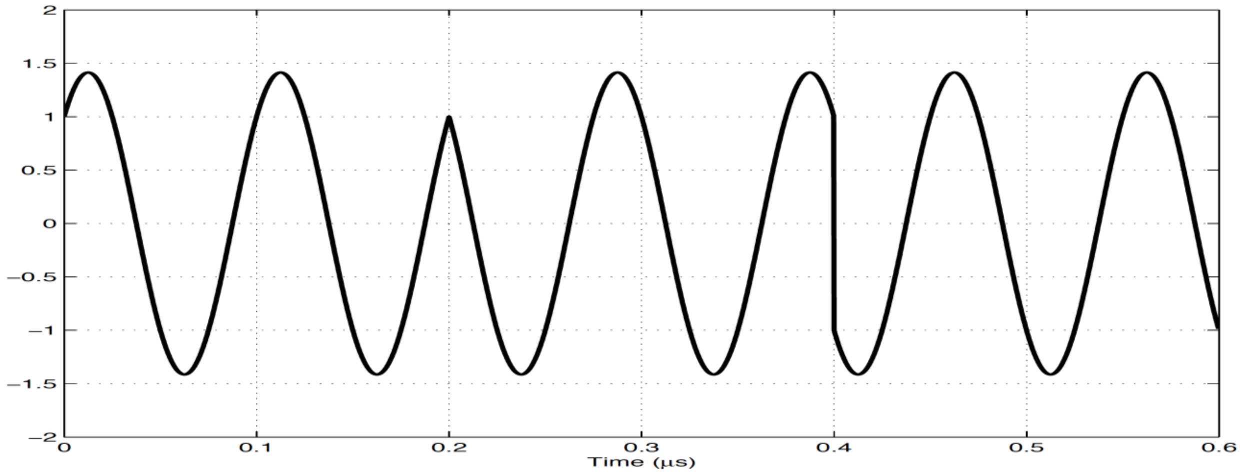


7.3

10 → amplitude $\sqrt{2}$ at $-\frac{\pi}{4} = -45°$

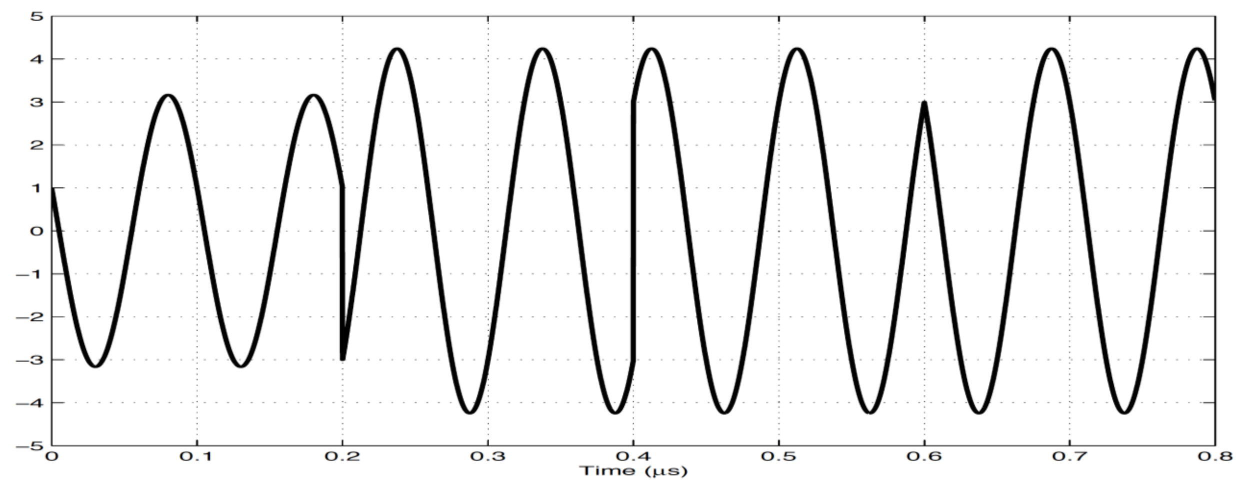
11 → amplitude $\sqrt{2}$ at $+\frac{\pi}{4} = +45°$

01 → amplitude $\sqrt{2}$ at $+\frac{3\pi}{4} = +135°$



7.4

1011 → amplitude $\sqrt{3^2+1^2}$ at – $\tan^{-1}\frac{3}{1} = \ 1.249\ rad = 71.56°$
0000 → amplitude $\sqrt{3^2+3^2}$ at $+\frac{5\pi}{4} = \ +225°$
1100 → amplitude $\sqrt{3^2+3^2}$ at $-\frac{\pi}{4} = \ -45°$
1111 → amplitude $\sqrt{3^2+3^2}$ at $\frac{\pi}{4} = \ +45°$



7.5 $s(t) = d_1(t)\cos w_ct + d_2(t)\sin w_ct$

Use the following identities: $\cos 2\alpha = 2\cos^2\alpha - 1$; $\sin 2\alpha = 2\sin\alpha \cos\alpha$

$$\begin{aligned} s(t)\cos w_ct &= d_1(t)\cos^2 w_ct + d_2(t)\sin w_ct \cos w_ct \\ &= (1/2)d_1(t) + (1/2)d_1(t)\cos 2w_ct + (1/2)d_2(t)\sin 2w_ct \end{aligned}$$

Use the following identities: $\cos 2\alpha = 1 - 2\sin^2\alpha$; $\sin 2\alpha = 2\sin\alpha \cos\alpha$

$$s(t)\sin w_ct = d_1(t)\cos w_ct \sin w_ct + d_2(t)\sin^2 w_ct$$

$$= (1/2)d_1(t)\sin 2w_c t + (1/2)d_2(t) - (1/2)d_2(t)\cos 2w_c t$$

All terms at $2w_c$ are filtered out by the low-pass filter, yielding:

$y_1(t) = (1/2)d_1(t)$; $y_2(t) = (1/2)d_2(t)$

7.6 T_s = signal element period; T_b = bit period; A = amplitude = 0.005

a. $T_s = T_b = 10^{-5}$ sec

$$P = \frac{1}{T_s}\int_0^{T_s} s^2(t) = \frac{A^2}{2}$$

$$E_b = P \times T_b = P \times T_s = \frac{A^2}{2} \times T_s; \quad N_0 = 2.5 \times 10^{-8} \times T_s$$

$$\frac{E_b}{N_0} = \frac{\left(A^2/2\right) \times T_s}{2.5 \times 10^{-8} \times T_s} = 500; \quad \left(E_b/N_0\right)_{\text{dB}} = 10\log 500 = 27\text{ dB}$$

b.

$$T_b = \frac{T_s}{2}; \; E_b = P \times \frac{T_s}{2}; \; N_0 = 2.5 \times 10^{-8} \times T_s$$

$$\left(E_b/N_0\right) = 250; \qquad \left(E_b/N_0\right)_{\text{dB}} = 10\log 250 = 24\text{ dB}$$

7.7 Each signal element conveys two bits. First consider NRZ-L. It should be clear that in this case, D = R/2. For the remaining codes, one must first determine the average number of pulses per bit. For example, for Biphase-M, there is an average of 1.5 pulses per bit. We have pulse rate of P, which yields a data rate of

R = P/1.5

D = P/2 = (1.5 x R)/2 = 0.75 × R

7.8 BPSK requires approximately 8.4 dB E_b/N_0.

DPSK requires approximately 9.3 dB E_b/N_0.

ASK/BFSK require approximately 12.4 dB E_b/N_0.

9.3 – 8.4 = 0.9 dB higher E_b/N_0 is required for DPSK than BPSK.

12.4 – 8.4 = 4.0 dB higher E_b/N_0 is required for ASK/BFSK then BPSK.

7.9 For rectangular pulses, $E_b = A^2T_b$

For BPSK, $E_{b,BPSK} = (0.01)^2T_b = 0.0001T_b$

For DPSK versus BPSK 0.9 dB = 10 log $((E_{b,DPSK}/N_0)/(E_{b,BPSK}/N_0))$

$10^{0.9/10} = 1.23 = E_{b,DPSK}/E_{b,BPSK} = (A_{DPSK}{}^2T_b)/(A_{BPSK}{}^2T_b) = (A_{DPSK}{}^2)/(0.0001)$

$A_{DPSK} = 0.0111$

Using the same approach, $A_{DPSK} = 0.0158$

7.10 $E_b/N_0 = (S/N)(B/R)$

$S/N = (R/B)(E_b/N_0) = 1 \times (E_b/N_0)$

$(S/N)_{dB} = (E_b/N_0)_{dB}$

For **FSK** and **ASK**, from Figure 4.10, $(E_b/N_0)_{dB} = 13.5$ dB

$(S/N)_{dB} = 13.5$ dB

For **PSK**, from Figure 4.10, $(E_b/N_0)_{dB} = 10.5$

$(S/N)_{dB} = 10.5$ dB

For **QPSK**, the effective bandwidth is halved, so that

$(R/B) = 2$

$(R/B)_{dB} = 3$

$(S/N)_{dB} = 3 + 10.5 = 13.5$ dB

7.11 As was mentioned in the text, analog signals in the voice band that represent digital data have more high frequency components than analog voice signals. These higher components cause the signal to change more rapidly over time. Hence, DM will suffer from a high level of slope overload noise. PCM, on the other hand, does not estimate changes in signals, but rather the absolute value of the signal, and is less affected than DM.

7.12 From the text, $(SNR)_{db} = 6.02\,n + 1.76$, where n is the number of bits used for quantization. In this case, $(SNR)_{db} = 60.2 + 1.76 = 61.96$ dB.

7.13 **a.** $(SNR)_{db} = 6.02\,n + 1.76 = 30$ dB

$n = (30 - 1.76)/6.02 = 4.69$

Rounded off, n = 5 bits

This yields $2^5 = 32$ quantization levels

b. R = 7000 samples/s × 5 bits/sample = 35 Kbps

7.14 The maximum slope that can be generated by a DM system is

$\delta/T_s = \delta f_s$

where T_s = period of sampling; f_s = frequency of sampling

Consider that the maximum frequency component of the signal is

$w(t) = A \sin 2\pi f_a t$

The slope of this component is

$dw(t)/dt = A 2\pi f_a \cos 2\pi f_a t$

and the maximum slope is A2 π f_a. To avoid slope overload, we require that

$$\delta f_s > A2\pi f_a$$

or $$d > \frac{2pf_aA}{f_s}$$

Source: [COUC01]

7.15 **a.** A total of 2^8 quantization levels are possible, so the normalized step size is 2^{-8} = 0.003906.

b. The actual step size, in volts, is:

$0.003906 \times 10V = 0.03906V$

c. The maximum normalized quantized voltage is $1 - 2^{-8} = 0.9961$. Thus the actual maximum quantized voltage is:

$0.9961 \times 10V = 9.961V$

d. The normalized step size is 2^{-8}. The maximum error that can occur is one-half the step size. Therefore, the normalized resolution is:

$\pm\ 1/2 \times 2^{-8} = 0.001953$

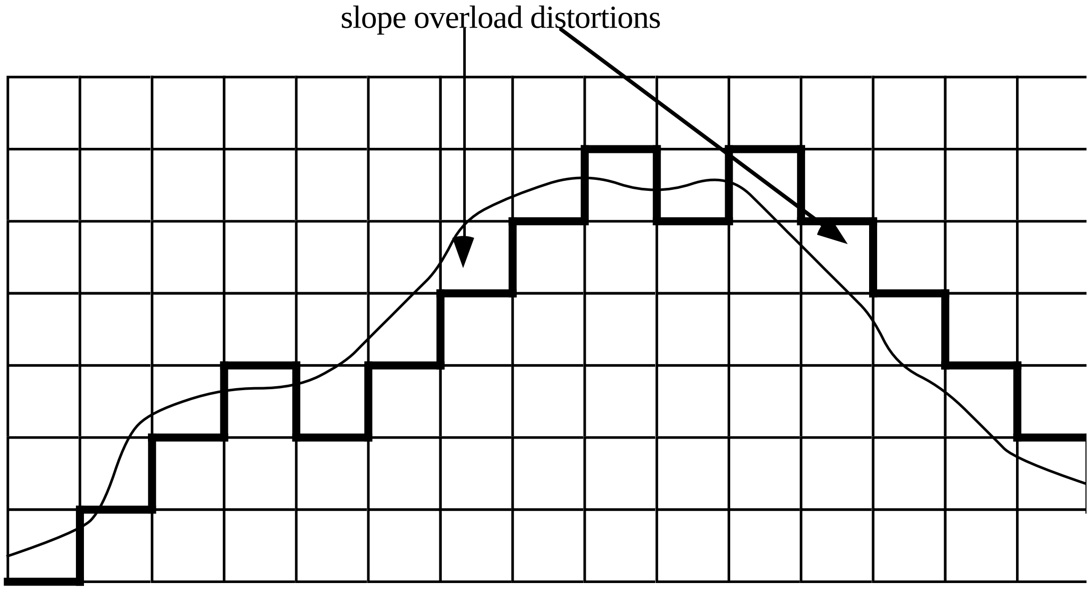
e. The actual resolution is

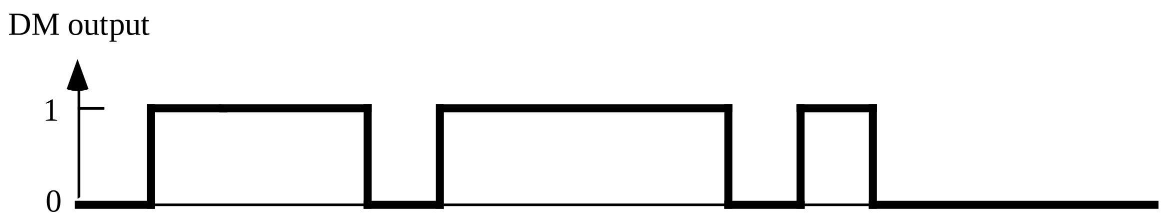
$\pm\ 0.001953 \times 10V = \pm\ 0.01953V$

f. The percentage resolution is

$\pm\ 0.001953 \times 100\% = \pm\ 0.1953\ \%$

7.16





7.17 $s(t) = A_c \cos[2\pi f_c t + \phi(t)] = 10 \cos[(10^8)\pi t + 5 \sin 2\pi(10^3)t]$

Therefore, $\phi(t) = 5 \sin 2\pi(10^3)t$, and the maximum phase deviation is 5 radians. For frequency deviation, recognize that the change in frequency is determined by the derivative of the phase:

$\phi'(t) = 5\,(2\pi)\,(10^3) \cos 2\pi(10^3)t$

which yields a frequency deviation of $\Delta f = (1/2\pi)[\,5\,(2\pi)\,(10^3)] = 5$ kHz

7.18 **a.** $s(t) = A_c \cos[2\pi f_c t + n_p m(t)] = 10 \cos[2\Pi(10^6)t + 0.1 \sin(10^3)\pi t]$

$A_c = 10$; $f_c = 10^6$

$10\, m(t) = 0.1 \sin(10^3)\pi t$, so $m(t) = 0.01 \sin(10^3)\pi t$

b. $s(t) = A_c \cos[2\pi f_c t + \phi(t)] = 10 \cos[2\Pi(10^6)t + 0.1 \sin(10^3)\pi t]$

$A_c = 10$; $f_c = 10^6$

$\phi(t) = 0.1 \sin(10^3)\pi t$, so $\phi'(t) = 100\pi \cos(10^3)\pi t = n_f m(t) = 10\, m(t)$

Therefore $m(t) = 10\pi \cos(10^3)\pi t$

7.19 **a.** For AM, $s(t) = [1 + m(t)] \cos(2\pi f_c t)$

$s_1(t) = [1 + m_1(t)] \cos(2\pi f_c t)$; $s_2(t) = [1 + m_2(t)] \cos(2\pi f_c t)$

For the combined signal $m_c(t) = m_1(t) + m_2(t)$,

$s_c(t) = [1 + m_1(t) + m_2(t)] \cos(2\pi f_c t) = s_1(t) + s_2(t) - 1$, which is a linear combination of $s_1(t)$ and $s_2(t)$.

b. For PM, $s(t) = A \cos(2\pi f_c t + n_p m(t))$

$s_1(t) = A \cos(2\pi f_c t + n_p m_1(t))$; $s_2(t) = A \cos(2\pi f_c t + n_p m_2(t))$

For the combined signal $m_c(t) = m_1(t) + m_b(t)$,

$s_c(t) = A \cos(2\pi f_c t + n_p[m_1(t) + m_2(t)])$, which is not a linear combination of $s_1(t)$ and $s_2(t)$.

Chapter 8 Orthogonal Frequency Division Multiplexing

ANSWERS TO QUESTIONS

8.1

- Orthogonal Frequency Division Multiplexing (OFDM) is a scheme that divides a broadband signal into multiple, orthogonal, low bit rate parallel transmissions.
- Orthogonal Frequency Division Multiple Access (OFDMA) is an OFDM MAC mechanism where multiple users share a channel by using different OFDM subcarriers.
- Single-carrier FDMA (SC-FDMA) is a variant of OFDM that performs extra DFT operations at the transmitter and receiver. This benefits the mobile user to provide better battery life, efficiency, and lower cost. It does not support multiple access; at any given point in time, all of the subcarriers must be dedicated to one user.

8.2 Subcarriers must be orthogonal. If the bit time of a subcarrier is T, then the frequency spacing should be 1/T.

8.3 OFDM overcomes multipath delay spread and efficiently uses the wireless spectrum.

8.4 The main technical problems with OFDM are the fact that the peak-to-average power ratio (PAPR) is higher than single-carrier signals and intercarrier interference (ICI) caused by subcarriers that are not in frequency synchronization.

8.5 OFDM is a scheme for transmitting a signal on numerous subcarriers. It is a physical layer modulation technique. OFDMA is a MAC layer mechanism for sharing those subcarriers among multiple users.

8.6 Instead of requiring N oscillators for N subcarriers, the IFFT approach requires one oscillator for N subcarriers. This reduction from N oscillators to one oscillator is true for both the transmitter and receiver.

8.7 Orthogonality allows signals to overlap each other in frequency if they are spaced properly. This is in contrast with traditional FDM that requires signals to have minimal overlap. For example in Figure 8.2, instead of requiring a spacing of $6f_b$, subcarriers can be spaced f_b apart, creating a sixfold increase in capacity.

8.8 The cyclic prefix addresses the problem of intersymbol interference caused by multipath delay spread. The format of the cyclic prefix also enables circular convolution.

8.9 Its longer bit duration creates special capabilities to overcome multipath fading by reducing the susceptibility to intersymbol interference. The symbol times can be substantially longer than the delay spread of the channel, so equalizers may not be needed. Also, frequency selective fading only adversely affects some of the subcarriers.

8.10

- Adjacent subcarriers – All subcarriers are assigned in a contiguous block of frequencies with relatively equivalent SINR for all of them. This allows the system to search for the best allocation of blocks of frequencies.
- Regularly spaced subcarriers – Subcarriers are distributed like a comb function. Frequency diversity provides a sufficient number of good carriers. This approach is simpler because it does not require substantial channel estimation.
- Randomly spaced subcarriers – This has similar benefits to regularly spaced subcarriers, but with reduced adjacent-cell interference.

8.11 To reduce the PAPR problem, which is especially helpful for mobile users with regards to battery life, power efficiency, and lower cost.

ANSWERS TO PROBLEMS

8.1 $1/66.67 \times 10^{-6}$ = 15000 symbols/sec = 15000 bits/sec per subcarrier
$(18 \times 10^6$ bits/sec)/(15000 bits/sec) = 1200 subcarriers

8.2 For 15000 bits/sec per subcarrier
$B_s = 15000$ Hz
180 kHz/15 kHz = 12 subcarriers per resource block

8.3 To be flat fading,
$B_C \gg B_S$
$B_C > 10B_S$
$80 \times 10^3 > 10B_S$
$B_S < 8 \times 10^3$
$B_S \approx r_b$
$\mathbf{16} \times 10^6 \,\mathbf{/}\, 8 \times 10^3 = 2000$ subcarriers

8.4 Maximum excess delay << 4.7 μs
Maximum excess delay < 0.47 μs
$(0.47 \times 10^{-6}) \times (3 \times 10^{8}\ \text{m/s}) = 141\ \text{m}$
This is distance traveled more than 1000 m, so the total distance is 1141 m.

8.5 $\int_0^{T_b} \cos(2\pi f_1 t)\cos(2\pi f_2 t)\, dt = 0$

$$\frac{1}{2}\int_0^{T_b} \cos(2\pi(f_1 + f_2)t) + \cos(2\pi(f_1 - f_2)t)\, dt = 0$$

$$\frac{1}{2}\left(\frac{\sin(2\pi(f_1 + f_2)T_b)}{2\pi(f_1 + f_2)} + \frac{\sin(2\pi(f_1 - f_2)T_b)}{2\pi(f_1 - f_2)}\right) = 0$$

$$2\pi(f_1 + f_2)T_b = \pi k, k\ integer$$

$$f_1 + f_2 = \frac{k}{2T_b}$$

$$2\pi(f_1 - f_2)T_b = 2\pi n, n\ integer$$

$$f_1 - f_2 = \frac{n}{2T_b}$$

8.6 $f_1 - f_2 = \frac{1}{2T_b}$

Chapter 9 Spread Spectrum

ANSWERS TO QUESTIONS

9.1 The bandwidth is wider after the signal has been encoded using spread spectrum.

9.2 **(1)** We can gain immunity from various kinds of noise and multipath distortion. **(2)** It can also be used for hiding and encrypting signals. Only a recipient who knows the spreading code can recover the encoded information. **(3)** Several users can independently use the same higher bandwidth with very little interference, using code division multiple access (CDMA).

9.3 With frequency hopping spread spectrum (FHSS), the signal is broadcast over a seemingly random series of radio frequencies, hopping from frequency to frequency at fixed intervals. A receiver, hopping between frequencies in synchronization with the transmitter, picks up the message.

9.4 Slow FHSS = multiple signal elements per hop; fast FHSS = multiple hops per signal element.

9.5 With direct sequence spread spectrum (DSSS), each bit in the original signal is represented by multiple bits in the transmitted signal, using a spreading code.

9.6 For an *N*-bit spreading code, the bit rate after spreading (usually called the chip rate) is *N* times the original bit rate.

9.7 CDMA allows multiple users to transmit over the same wireless channel using spread spectrum. Each user uses a different spreading code. The receiver picks out one signal by matching the spreading code.

ANSWERS TO PROBLEMS

9.1 a. We have $C = B \log_2 (1 + \text{SNR})$. For SNR = 0.1, B = 0.41 MHz; For SNR = 0.01, B = 3.9 MHz; for SNR = 0.001, B = 38.84 MHz. Thus, to achieve the desired SNR, the signal must be spread so that 56 KHz is carried in very large bandwidths.

b. For 1 bps/Hz, the equation $C = B \log_2 (1 + \text{SNR})$ becomes $\log_2 (1 + \text{SNR}) = 1$. Solving for SNR, we have SNR = 1. Thus a far higher SNR is required without spread spectrum.

9.2 The total number of tones, or individual channels is:

W_s/f_d = (400 MHz)/(100 Hz) = 4×10^6.

The minimum number of PN bits = $\lceil \log_2 (4 \times 10^6) \rceil = 22$

where $\lceil x \rceil$ indicates the smallest integer value not less than x. Source: [SKLA01]

9.3 $W_s = 1000\, f_d$; $W_d = 4\, f_d$; Using Equation 9.3 , $G_p = W_s/W_d = 250 = 24$ dB

9.4 **a.** MFSK

b. $L = 2$

c. $M = 2^L = 4$

d. $k = 3$

e. slow FHSS

f. $2^k = 8$

g.

Time	0	1	2	3	4	5	6	7	8	9	10	11
Input data	0	1	1	1	1	1	1	0	0	0	1	0
Frequency	f_1		f_3		f_3		f_2		f_0		f_2	

Time	12	13	14	15	16	17	18	19
Input data	0	1	1	1	1	0	1	0
Frequency	f_1		f_3		f_2		f_2	

9.5 **a.** MFSK

b. $L = 2$

c. $M = 2^L = 4$

d. $k = 3$

e. fast FHSS

f. $2^k = 8$

g. Same as for Problem 9.4

9.6 **a.** This is from the example 7.1.

f_1 = 75 kHz 000 f_2 =125 kHz 001 f_3 = 175 kHz 010 f_4 = 225 kHz 011

f_5 = 275 kHz 100 f_6 = 325 kHz 101 f_7 = 375 kHz 110 f_8 = 425 kHz 111

b. We need three more sets of 8 frequencies. The second set can start at 475 kHz, with 8 frequencies separated by 50 kHz each. The third set can start at 875 kHz, and the fourth set at 1275 kHz.

Chapter 10 Coding and Error Control

ANSWERS TO QUESTIONS

10.1 A parity bit appended to an array of binary digits to make the sum of all the binary digits, including the parity bit, always odd (odd parity) or always even (even parity).

10.2 The CRC is an error detecting code in which the code is the remainder resulting from dividing the bits to be checked by a predetermined binary number.

10.3 The CRC has more bits and therefore provides more redundancy. That is, it provides more information that can be used to detect errors.

10.4 Modulo 2 arithmetic, polynomials, and digital logic.

10.5 It is possible. You could design a code in which all codewords are at least a distance of 3 from all other codewords, allowing all single-bit errors to be corrected. Suppose that some but not all codewords in this code are at least a distance of 5 from all other codewords. Then for those particular codewords, but not the others, a double-bit error could be corrected.

10.6 An (n, k) block code encodes k data bits into n-bit codewords.

10.7 An (n, k, K) code processes input data k bits at a time and produces an output of n bits for each incoming k bits. The current output of n bits is a function of the last $K \times k$ input bits.

10.8 A trellis is a diagram that shows the state transitions over time in a convolutional code.

10.9 Detection of errors and retransmission of frames that are received in error.

10.10 Go-back-N ARQ is a form of error control in which a destination station sends a negative acknowledgment (NAK) when it receives an error. The source station receiving the NAK will retransmit the frame in error plus all succeeding frames transmitted in the interim.

10.11 Soft decision decoding information from previous transmissions can be combined with soft decision information from more recent transmissions to provide assurance levels of the decoding of certain bits.

ANSWERS TO PROBLEMS

10.1 Any arithmetic scheme will work if applied in exactly the same way to the forward and reverse process. The modulo 2 scheme is easy to implement in circuitry. It also yields a remainder one bit smaller than binary arithmetic.

10.2 **a.** We have:

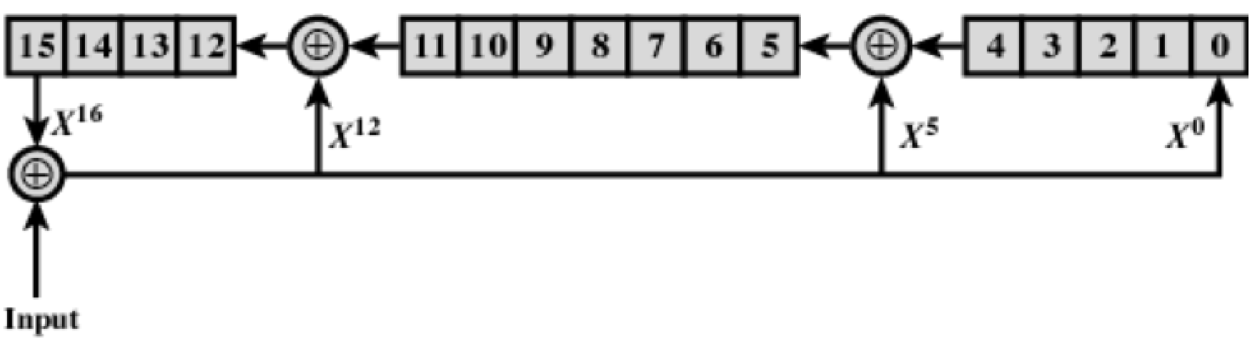
Pr [single bit in error] = 10^{-3}
Pr [single bit not in error] = $1 - 10^{-3} = 0.999$
Pr [8 bits not in error] = $(1 - 10^{-3})^8 = (0.999)^8 = 0.992$
Pr [at least one error in frame] = $1 - (1 - 10^{-3})^8 = 0.008$

b. Pr [at least one error in frame] = $1 - (1 - 10^{-3})^{10} = 1 - (0.999)^{10} = 0.01$

10.3 **a.**

```
                                   1000100010011000
10001000000100001/100000000000000000000000000000000
                  10001000000100001
                      10000001000010000
                      10001000000100001
                          10010001100010000
                          10001000000100001
                             11001100110001000
                             10001000000100001
                              10001001101010010
                              10001000000100001
remainder =                        0001101110011000
```

b.



Shift	Shift Register																Input
	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	
1	0	0	0	1	0	0	0	0	0	0	1	0	0	0	0	1	1
2	0	0	1	0	0	0	0	0	0	1	0	0	0	0	1	0	0
3	0	1	0	0	0	0	0	0	1	0	0	0	0	1	0	0	0
4	1	0	0	0	0	0	0	1	0	0	0	0	1	0	0	0	0
5	0	0	0	1	0	0	1	0	0	0	1	1	0	0	0	1	0

6	0	0	1	0	0	1	0	0	0	1	1	0	0	0	1	0	0
7	0	1	0	0	1	0	0	0	1	1	0	0	0	1	0	0	0
8	1	0	0	1	0	0	0	1	1	0	0	0	1	0	0	0	0
9	0	0	1	1	0	0	1	1	0	0	1	1	0	0	0	1	0
10	0	1	1	0	0	1	1	0	0	1	1	0	0	0	1	0	0
11	1	1	0	0	1	1	0	0	1	1	0	0	0	1	0	0	0
12	1	0	0	0	1	0	0	1	1	0	1	0	1	0	0	1	0
13	0	0	0	0	0	0	1	1	0	1	1	1	0	0	1	1	0
14	0	0	0	0	0	1	1	0	1	1	1	0	0	1	1	0	0
15	0	0	0	0	1	1	0	1	1	1	0	0	1	1	0	0	0
16	0	0	0	1	1	0	1	1	1	0	0	1	1	0	0	0	0

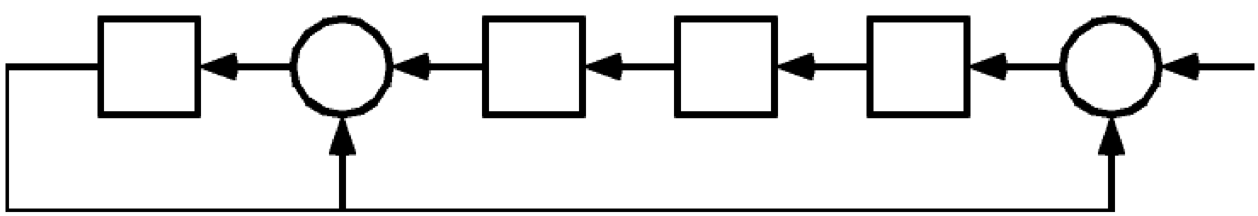
CRC

10.4 At the conclusion of the data transfer, just before the CRC pattern arrives, the shift register should contain the identical CRC result. Now, the bits of the incoming CRC are applied at point C_4 (Figure 8.3). Each 1 bit will merge with a 1 bit (exclusive-or) to produce a 0; each 0 bit will merge with a 0 bit to produce a zero.

10.5

```
                 10110110
       ___________________
110011/1110001100000
       110011
         101111
         110011
          111000
          110011
            101100
            110011
             111110
             110011
   CRC =      11010
```

10.6 a.



b. Data = 1 0 0 1 1 0 1 1 1 0 0

$M(X) = 1 + X^3 + X^4 + X^6 + X^7 + X^8$

$X^4M(X) = X^{12} + X^{11} + X^{10} + X^8 + X^7 + X^4$

$$\frac{X^4 M(X)}{P(X)} = X^{12} + X^{11} + X^{10} + X^8 + X^7 + \frac{X^2}{P(X)}$$

$R(X) = X^2$
$T(X) = X^4M(X) + R(X) = X^{12} + X^{11} + X^{10} + X^8 + X^7 + X^4 + X^2$
Code = 0 0 1 0 1 0 0 1 1 0 1 1 1 0 0

c. Code = 0 0 1 0 1 0 0 0 1 0 1 1 1 0 0

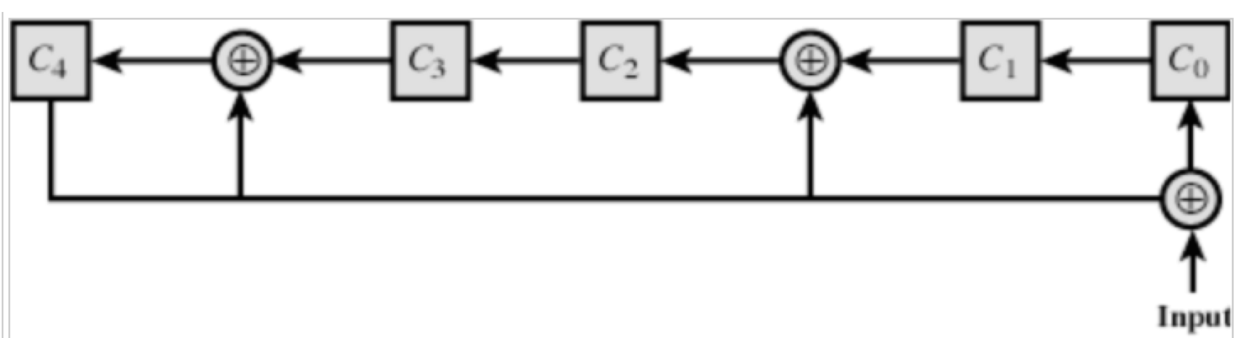
$\frac{T(X)}{P(X)}$ yields a nonzero remainder

10.7 a. The multiplication of M(X) by X^{16} corresponds to shifting M(X) 16 places and thus providing the space for a 16-bit FCS. The addition of $X^kL(X)$ to $X^{16}M(X)$ inverts the first 16 bits of G(X) (one's complements). The addition of L(X) to R(X) inverts all of the bits of R(X).

b. The HDLC standard provides the following explanation. The addition of $X^KL(X)$ corresponds to a value of all ones. This addition protects against the obliteration of leading flags, which may be non-detectable if the initial remainder is zero. The addition of L(X) to R(X) ensures that the received, error-free message will result in a unique, non-zero remainder at the receiver. The non-zero remainder protects against the potential non-detectability of the obliteration of trailing flags.

c. The implementation is the same as that shown in Solution 3b, with the following strategy. At both transmitter and receiver, the initial content of the register is preset to all ones. The final remainder, if there are no errors, will be 0001 1101 0000 1111.

10.8 a. For simplicity, we do not show the switches.



b.

C_4	C_3	C_2	C_1	C_0	$C_4 \oplus C_3$	$C_4 \oplus C_1$	$C_4 \oplus I$	Input
0	0	0	0	0	0	0	1	1
0	0	0	0	1	0	0	0	0
0	0	0	1	0	0	1	1	1

0	0	1	0	1	0	0	0	0
0	1	0	1	0	1	1	0	0
1	0	1	0	0	1	1	1	0
1	1	1	0	1	0	1	0	1
0	1	1	1	0	1	1	1	1
1	1	1	0	1	0	1	1	0
0	1	1	1	1	1	1	1	1
1	1	1	1	1	0	0	1	–
0	1	0	1	1	1	1	0	–
1	0	1	1	0	1	0	1	–
1	1	0	0	1	0	1	1	–
0	0	1	1	1	0	1	0	–
0	1	1	1	0				

c. The partial results from the long division show up in the shift register, as indicated by the shaded portions of the preceding table. Compare to long division example in Section 8.1.

d. Five additional steps are required to produce the result.

10.9 **a.**

	00000	10101	01010
00000	0	2	2
10101	3	0	5
01010	2	5	0

b.

	000000	010101	101010	110110
000000	0	3	3	4
010101	3	0	6	6
101010	3	6	0	3
110110	4	6	3	0

10.10 a. $p(\mathbf{v} \mid \mathbf{w}) = \beta^{d(w,v)}(1 - \beta)^{(n - d(w,v))}$

b. If we write $d_i = d(\mathbf{w}_i, \mathbf{v})$, then $\dfrac{p(v|w_1)}{p(v|w_2)} = \dfrac{b^{d_1}(1 - b)^{n-d_1}}{b^{d_2}(1 - b)^{n-d_2}} = \left(\dfrac{1 - b}{b}\right)^{d_2 - d_1}$

c. If $0 < \beta < 0.5$, then $(1 - \beta)/\beta > 1$. Therefore, by the equation of part b, $p(\mathbf{v} \mid \mathbf{w}_1)/p(\mathbf{v} \mid \mathbf{w}_2) > 1$ if an only if $d_1 < d_2$.

10.11 Suppose that the minimum distance between codewords is at least $2t + 1$. For a codeword **w** to be decoded as another codeword **w'**, the received sequence must be at least as close to **w'** as to **w**. For this to happen, at least $t + 1$ bits of **w** must be in error. Therefore all errors involving t or fewer digits are correctable.

10.12

$$\begin{array}{llllllllll}
C1 & = & D1 \oplus & D2 \oplus & & D4 \oplus & D5 \oplus & & D7 & \\
C2 & = & D1 \oplus & & D3 \oplus & D4 \oplus & & D6 \oplus & D7 & \\
C4 & = & & D2 \oplus & D3 \oplus & D4 \oplus & & & & D8 \\
C8 & = & & & & & D5 \oplus & D6 \oplus & D7 \oplus & D8
\end{array}$$

10.13 The transmitted block and check bit calculation are shown in Table 8.2a and b. Now suppose that the only error is in C8. Then the received block results in the following table:

Position	12	11	10	9	8	7	6	5	4	3	2	1
Bits	D8	D7	D6	D5	C8	D4	D3	D2	C4	D1	C2	C1
Block	0	0	1	1	1	1	0	0	1	1	1	1
Codes			1010	1001		0111				0011		

The check bit calculation after reception:

Position	**Code**
Hamming	1111
10	1010
9	1001
7	0111
3	0011
XOR = syndrome	1000

The nonzero result detects and error and indicates that the error is in bit position 8, which is check bit C8.

10.14 Data bits with value 1 are in bit positions 12, 11, 5, 4, 2, and 1:

Position	12	11	10	9	8	7	6	5	4	3	2	1
Bit	D8	D7	D6	D5	C8	D4	D3	D2	C4	D1	C2	C1
Block	1	1	0	0		0	0	1		0		
Codes	1100	1011						0101				

Check bit calculation:

Position	**Code**
12	1100
11	1011
5	0101
XOR = C8 C4 C2 C1	0010

10.15 The Hamming Word initially calculated was:

bit number:	12	11	10	9	8	7	6	5	4	3	2	1
	0	0	1	1	0	1	0	0	1	1	1	1

Doing an exclusive-OR of 0111 and 1101 yields 1010 indicating an error in bit 10 of the Hamming Word. Thus, the data word read from memory was 00011001.

10.16 Need $n - k$ check bits such that $2^{(n-k)} - 1 \geq 1024 + (n - k)$.
The minimum value of $n - k$ that satisfies this condition is 11.

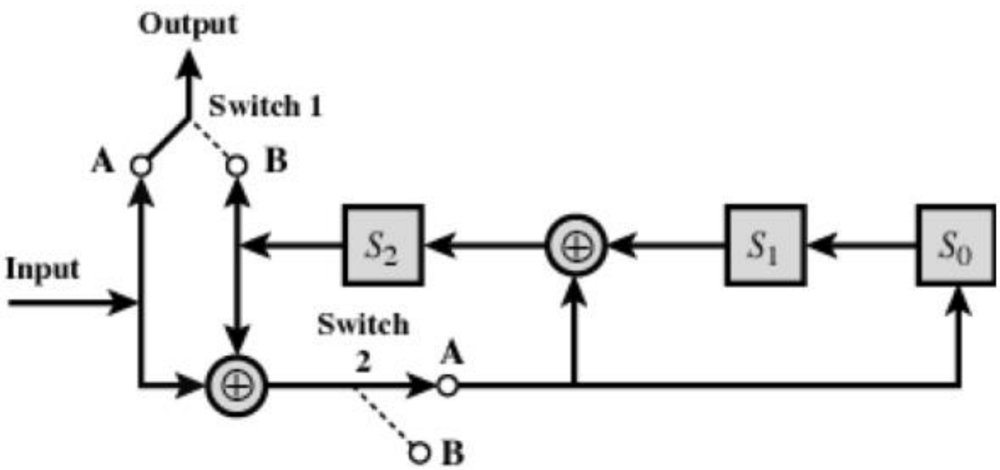
10.17 The calculation shows that g(X) divides f(X) with no remainder.

$$
\begin{array}{r|lllllll}
 & X^2 + X + 1 & & & & & & \\
\cline{2-8}
X^4 + X^3 + X + 1 & X^6 + & & & & & & 1 \\
 & X^6 + & X^5 + & & X^3 + & X^2 & & \\
\cline{2-8}
 & & X^5 + & & X^3 + & X^2 + & & 1 \\
 & & X^5 + & X^4 + & & X^2 + & X & \\
\cline{3-8}
 & & & X^4 + & X^3 & & + X & + 1 \\
 & & & X^4 + & X^3 & & + X & + 1 \\
\cline{4-8}
 & & & & & & & 0
\end{array}
$$

This result is verified by multiplying the quotient by g(X) to get back f(X) exactly:

$$
\begin{array}{lllllll}
 & & X^4 + & X^3 & & + X & + 1 \\
 & & & & X^2 & + X & + 1 \\
\hline
 & & X^4 + & X^3 & & + X & + 1 \\
 & X^5 + & X^4 + & & X^2 & + X & \\
X^6 + & X^5 & & + X^3 + & X^2 & & \\
\hline
X^6 + & & & & & & 1
\end{array}
$$

10.18 a.



b.

S_2	S_1	S_0	$S_2 \oplus S_1 \oplus I$	$S_2 \oplus I$	Input
0	0	0	1	1	1
1	0	1	1	1	0
1	1	1	1	0	1
1	1	0	0	1	0
0	0	1			

10.19

a.

1	2	3	4
5	6	7	8
9	10	11	12
13	14	15	16
17	18	19	20
21	22	23	24

b.

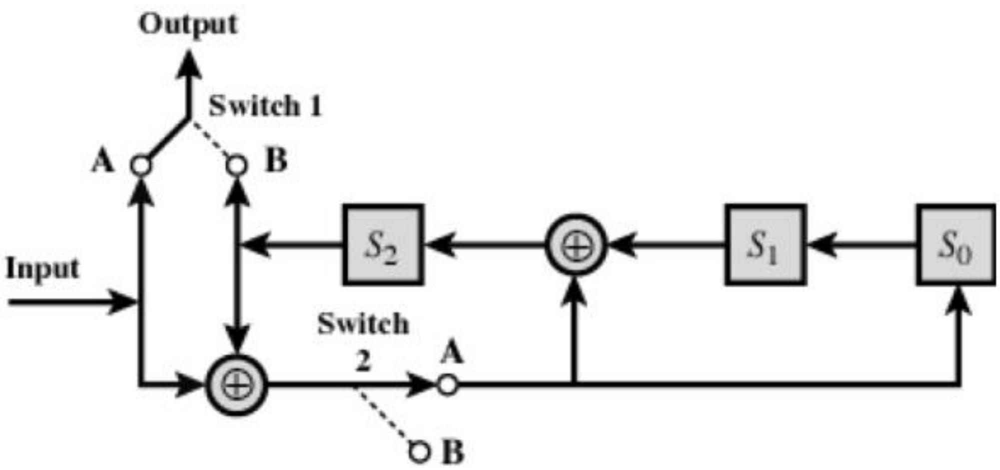
1	2	3	4
5	6	7	8
9	10	11	12
13	14	15	16
17	18	19	20
21	22	23	24

c.

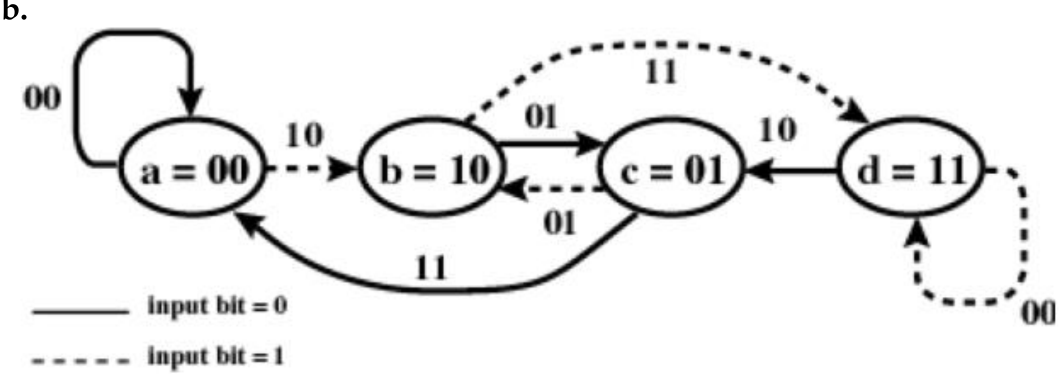
1	2	3	4
5	6	7	8
9	10	11	12
13	14	15	16
17	18	19	20
21	22	23	24

d. The first column is filled after 21 bits are read in. Similarly, 21 bits must arrive before deinterleaving. This confirms $2(n(m - 1) + 1) = 2(4(5) + 1) = 42$.
Source: {SKLA01].

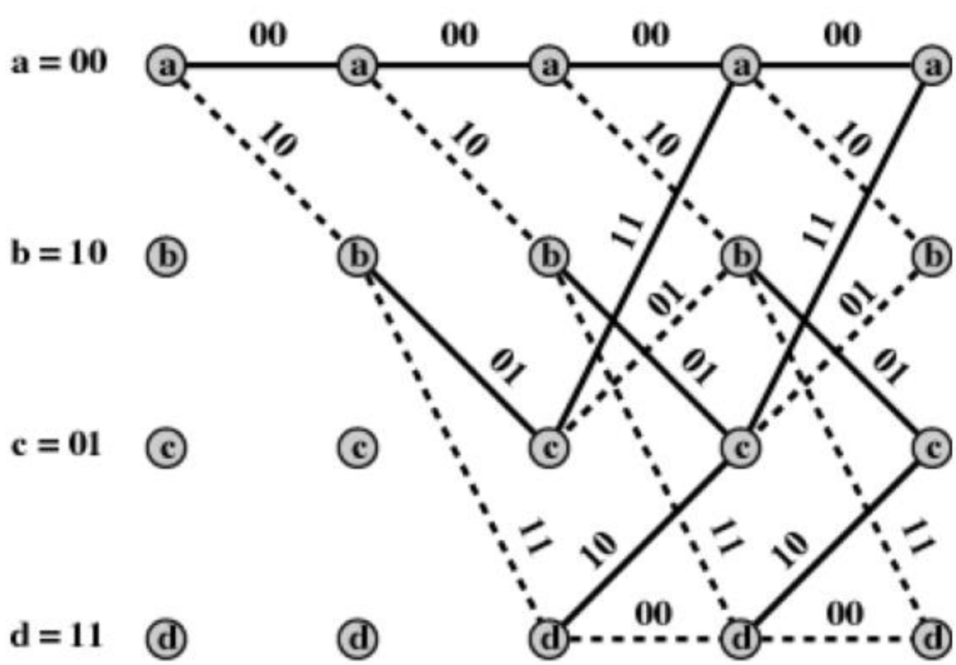
10.20 a.


Output
Switch 1
A
B
Input
S_2
S_1
S_0
Switch 2
A
B

b.


00
10
11
01
10
01
11
00
a = 00
b = 10
c = 01
d = 11
input bit = 0
input bit = 1

c.


a = 00
b = 10
c = 01
d = 11
00
10
11
01
00

10.21 **a.** This clears out the encoder, making it ready for use for the next transmission

b. The encoder is in state a = 00 before transmission and after transmission of the last information bit two zero bits are transmitted. The sequence of states traversed is abdcbcbdcb. The output sequence is 10 11 10 01 01 01 11 10 01 00

10.22 **a.** Because only one frame can be sent at a time, and transmission must stop until an acknowledgment is received, there is little effect in increasing the size of the message if the frame size remains the same. All that this would affect is connect and disconnect time.

b. Increasing the number of frames would decrease frame size (number of bits/frame). This would lower line efficiency, because the propagation time is unchanged but more acknowledgments would be needed.

c. For a given message size, increasing the frame size decreases the number of frames. This is the reverse of (b).

10.23 $A \rightarrow B$: Propagation time = $4000 \times 5\ \mu$sec = 20 msec

Transmission time per frame $= \dfrac{1000}{100 \times 10^3} = 10$ msec

$B \rightarrow C$: Propagation time = $1000 \times 5\ \mu$sec = 5 msec

Transmission time per frame = x = 1000/R

R = data rate between B and C (unknown)

A can transmit three frames to B and then must wait for the acknowledgment of the first frame before transmitting additional frames. The first frame takes 10 msec to transmit; the last bit of the first frame arrives at B 20 msec after it was transmitted, and therefore 30 msec after the frame transmission began. It will take an additional 20 msec for B's acknowledgment to return to A. Thus, A can transmit 3 frames in 50 msec.
B can transmit one frame to C at a time. It takes 5 + x msec for the frame to be received at C and an additional 5 msec for C's acknowledgment to return to A. Thus, B can transmit one frame every 10 + x msec, or 3 frames every 30 + 3x msec.

Thus:

$30 + 3x = 50$

$x = 6.66$ msec

$R = 1000/x = 150$ kbps

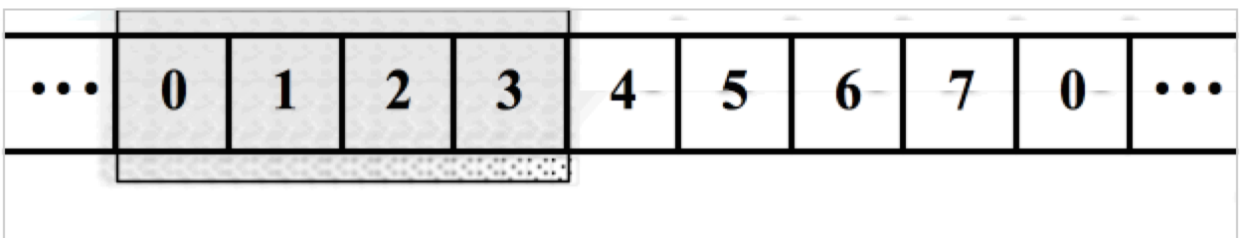
10.24 Round trip propagation delay of the link = 2 × L × t
Time to transmit a frame = B/R
To reach 100% utilization, the transmitter should be able to transmit frames continuously during a round trip propagation time. Thus, the total number of frames transmitted without an ACK is:

$N = \left\lceil \dfrac{2 \times L \times t}{B/R} + 1 \right\rceil$, where the brackets $\lceil X \rceil$ indicate the smallest integer greater than or equal to X (rounding up to the nearest integer).

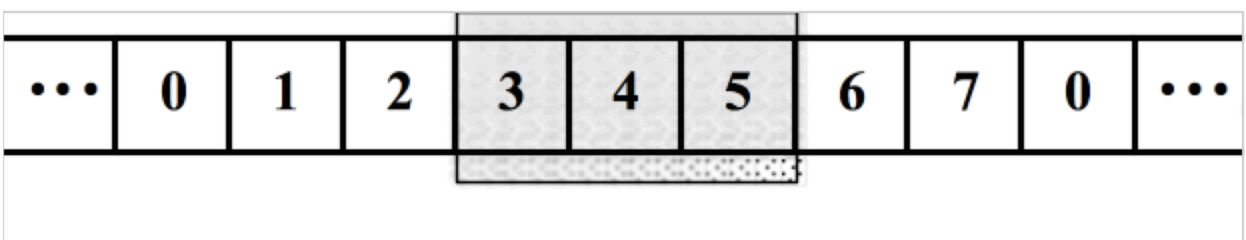
This number can be accommodated by an M-bit sequence number with:

$$M = \lceil \log_2(N) \rceil$$

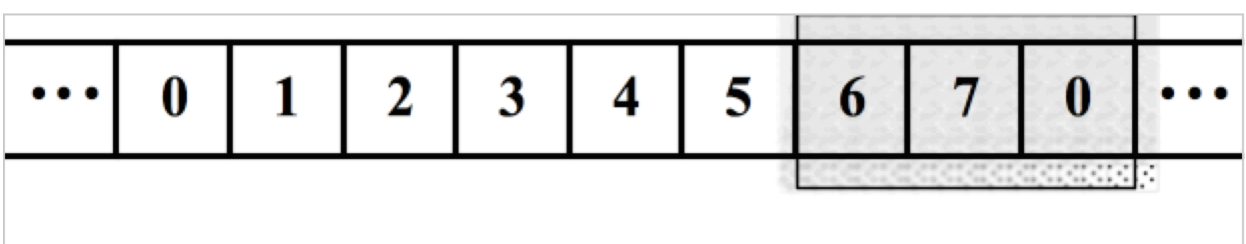
10.25 a.



b.



c.



10.26 Let t_1 = time to transmit a single frame

$$t_1 = \frac{1024 \text{ bits}}{10^6 \text{ bps}} = 1.024 \text{ msec}$$

The transmitting station can send 7 frames without an acknowledgment. From the beginning of the transmission of the first frame, the time to receive the acknowledgment of that frame is:

$$t_2 = 270 + t_1 + 270 = 541.024 \text{ msec}$$

During the time t_2, 7 frames are sent.

Data per frame = 1024 – 48 = 976

$$\text{Throughput} = \frac{7 \times 976 \text{ bits}}{541.024 \times 10^{-3} \text{ sec}} = 12.6 \text{ kbps}$$

Chapter 11 Wireless LAN Technology

ANSWERS TO QUESTIONS

11.1 **Logical link control (LLC):** provides an interface to higher layers and perform flow and error control; **medium access control (MAC):** provides addressing for physical attachment points to the LAN and provides medium access; **physical:** defines the topology, transmission medium, and signaling.

11.2 A Kiviat graph provides a pictorial means of comparing systems along multiple variables. The variables are laid out at equal angular intervals. A given system is defined by one point on each variable; these points are connected to yield a shape that is characteristic of that system.

11.3 A **MAC address** defines a physical point of attachment to the LAN. An **LLC address** identifies a particular LLC user (next higher layer above LLC).

11.4 It may or may not be.

11.5 **Association:** Establishes an initial association between a station and an AP. **Authentication:** Used to establish the identity of stations to each other. **Deauthentication:** This service is invoked whenever an existing authentication is to be terminated. **Disassociation:** A notification from either a station or an AP that an existing association is terminated. A station should give this notification before leaving an ESS or shutting down. **Distribution:** used by stations to exchange MAC frames when the frame must traverse the DS to get from a station in one BSS to a station in another BSS. **Integration:** enables transfer of data between a station on an IEEE 802.11 LAN and a station on an integrated IEEE 802.x LAN. **MSDU delivery:** delivery of MAC service data units. **Privacy:** Used to prevent the contents of messages from being read by other than the intended recipient. **Reassocation:** Enables an established association to be transferred from one AP to another, allowing a mobile station to move from one BSS to another.

11.6 **Mobility** refers to the types of physical transitions that can be made by a mobile node within an 802.11 environment (no transition, movement from one BSS to another within an ESS, movement from one ESS to another). **Association** is a service that allows a mobile node that has made a transition to identify itself to the AP within a BSS so that the node can participate in data exchanges with other mobile nodes.

11.7 **Single-cell wireless LAN**: all of the wireless end systems are within range of a single control module. **Multiple-cell wireless LAN**: there are multiple control modules interconnected by a wired LAN; each control module supports a number of wireless end systems within its transmission range.

11.8 **(1)** In order to transmit over a wired LAN, a station must be physically connected to the LAN. On the other hand, with a wireless LAN, any station within radio range of the other devices on the LAN can transmit. In a sense, there is a form of authentication with a wired LAN, in that it requires some positive and presumably observable action to connect a station to a wired LAN. **(2)** Similarly, in order to receive a transmission from a station that is part of a wired LAN, the receiving station must also be attached to the wired LAN. On the other hand, with a wireless LAN, any station within radio range can receive. Thus, a wired LAN provides a degree of privacy, limiting reception of data to stations connected to the LAN.

11.9 WI-FI Direct allows devices to easily connect with each other without an access point and transfer information at Wi-Fi speeds. Devices join a Group that can consist of a one-to-one connection or a larger group. Devices securely connect by use a push of a button, tapping two NFC-capable devices together, or by using a PIN.

To add to WLAN ad-hoc networking functionality, Wi-Fi Direct adds device and service discovery so Wi-Fi devices can find each other and exchange device/service capabilities. It also has a mechanism so a Wi-Fi Direct device can be in charge of a Group so that an AP is not needed.

ANSWERS TO PROBLEMS

11.2 **a.** (48 subcarriers per 20 MHz) × (2 spatial streams) × ($2/3 \times \log_2 16$) / (3.6×10^{-6}) = 71.1 Mbps

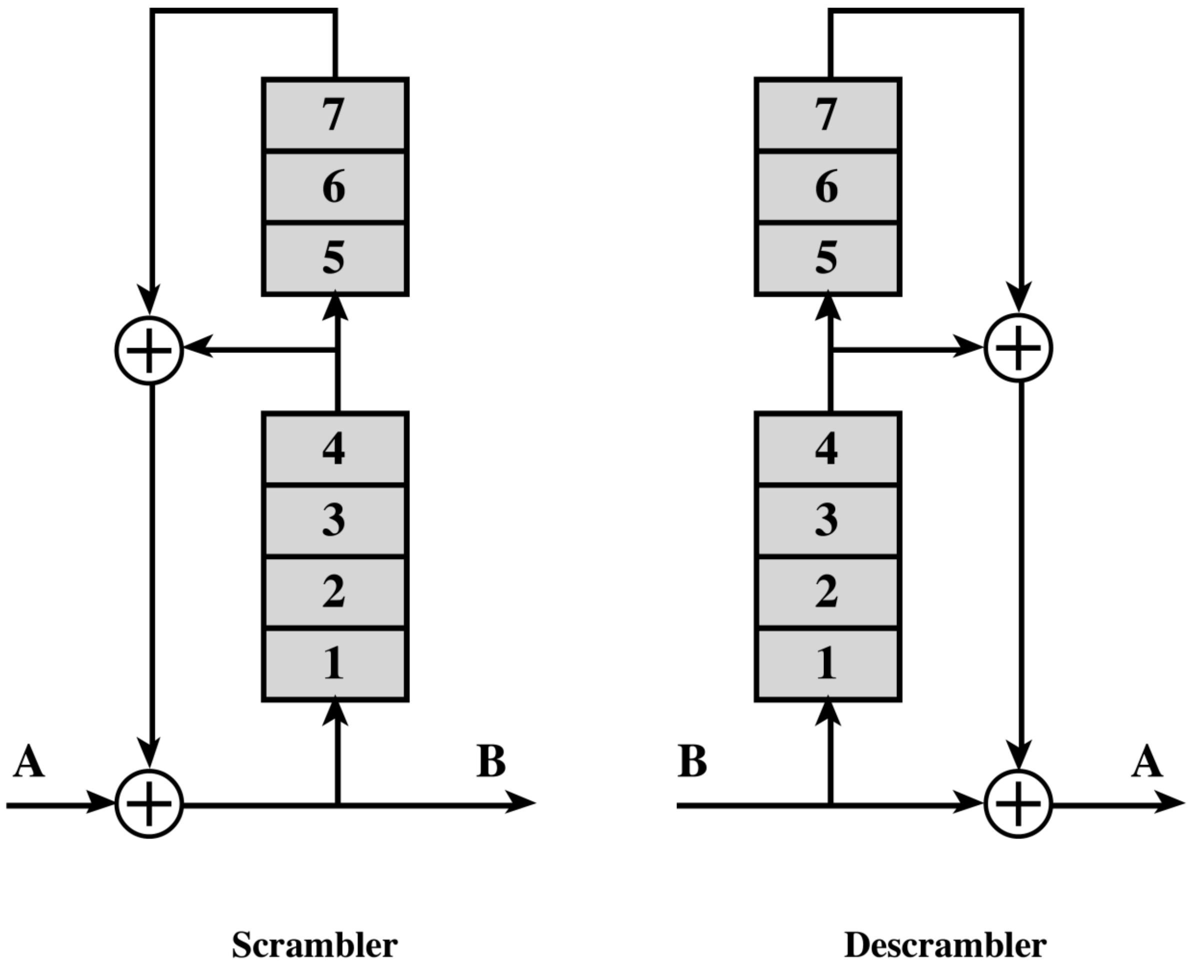
b. (108 subcarriers per 40 MHz) × (2 spatial streams) × ($2/3 \times \log_2 16$) / (3.6×10^{-6}) = 160 Mbps

11.3 (468 subcarriers per 160 MHz) × (8 spatial streams) × ($1/2 \times \log_2 64$) / (3.6×10^{-6}) = 4.16 Gbps

11.7 Wired LANs become part of a location's infrastructure and are therefore less flexible in maintenance and modification, however, the physical nature of the wired LAN makes it inherently more controllable. Wireless LANs are more flexible in implementation and modification, but require more complex maintenance mechanisms because of their more fluid characteristics. Unique concerns for wireless LAN designers include: device power consumption, quality and security of transmission medium, licensing and other concerns related to the mobile nature of the technology.

11.8 **a.** $B_m = A_m \oplus B_{m-4} \oplus B_{m-7}$

b.



Chapter 12 Bluetooth and IEEE 802.15

ANSWERS TO QUESTIONS

12.1 **Data and voice access points:** Bluetooth facilitates real-time voice and data transmissions by providing effortless wireless connection of portable and stationary communications devices. **Cable replacement:** Bluetooth eliminates the need for numerous, often proprietary, cable attachments for connection of practically any kind of communication device. Connections are instant and are maintained even when devices are not within line of sight. The range of each radio is approximately 10 m, but can be extended to 100 m with an optional amplifier. **Ad hoc networking:** A device equipped with a Bluetooth radio can establish instant connection to another Bluetooth radio as soon as it comes into range.

12.2 The **core specifications** describe the details of the various layers of the Bluetooth protocol architecture, from the radio interface to link control. The **profile specifications** are concerned with the use of Bluetooth technology to support various applications.

12.3 The radio designated as the master makes the determination of the channel (frequency hopping sequence) and phase (timing offset, i.e., when to transmit) that shall be used by all devices on this piconet. The radio designated as master makes this determination using its own device address as a parameter, while the slave devices must tune to the same channel and phase. A slave may only communicate with the master and may only communicate when granted permission by the master.

12.4 A frequency hopping defines a logical channel. This channel can be shared using TDD.

12.5 In FH-CDMA, the spreading sequence defines the sequence of frequencies to be used for frequency hopping. In DS-CDMA, the spreading sequence is used to map one data bit into multiple transmitted bits.

12.6 **Synchronous connection-oriented (SCO):** Allocates a fixed bandwidth between a point-to-point connection involving the master and a single slave. The master maintains the SCO link by using reserved slots at regular intervals. The basic unit of reservation is two consecutive slots (one in each transmission direction). The master can support up to three simultaneous SCO links while a slave can support two or three SCO links. SCO packets are never retransmitted **Asynchronous connectionless (ACL):** A point-to-multipoint link between the master and all the slaves in the piconet. In slots not reserved for SCO links, the master can exchange packets with any slave on a per-slot basis, including a slave already engaged in an

SCO link. Only a single ACL link can exist. For most ACL packets, packet retransmission is applied.

12.7 Bluetooth 3.0 achieves up to 24 Mbps by using 802.11 techniques.

12.8 Bluetooth Smart is also called Bluetooth Low Energy and is Bluetooth 4.0.

12.9 Both 802.15.3 and 802.15.4 have goals of low cost and low power compared to 802.15.1 and 802.11. 802.15.3 provides significantly higher data rates than 802.15.1, whereas 802.15.4 has a focus of very low cost and very low power, at the expense of having lower data rates than 802.15.1.

12.10 802.15.3 uses five modulation schemes in the 2.4 GHz band with 11 Mbaud to achieve rates 11 to 55 Mbps. It uses trellis-coded modulation. 802.15.3c utilizes 60 GHz frequencies to achieve rates over 1 Gbps.

12.11

- 868/915 MHz uses DSSS for data rates of 20, 40, 100, and 250 kbps
- 2.4 GHz DSSS achieves 250 kbps.
- Ultra wideband (UWB) operates in wide-frequency UWB bands using very short pulses that have high bandwidth. The PHY supports 851 kbps and optionally 110 kbps, 6.81 Mbps, and 27.234 Mbps
- 2.4 GHz chirp spread spectrum uses sinusoidal signals that change frequency to support 1 Mbps and optionally 250 kbps.

12.12 Active RFIDs have a transmitter, processor, and a power source. Passive tags consist only of a coil of wire or conducting material and they draw power from a reader, which sends electromagnetic waves that induce a signal to be transmitted from the passive RFID. Active tags can track high value goods or identify goods that must be scanned over longer ranges. Passive tags can be used when items can pass close to a scanner, such as on a conveyor belt.

12.13

- Network layer to provide routing.
- Application support layer for specialized services.
- ZigBee device objects (ZDOs) to keep device roles, manage requests to join the network, discover devices, and manage security.
- Application objects to allow customization from manufacturers.

ANSWERS TO PROBLEMS

12.1 **1.** The master sends message A to slave 1.
2. Slave 1 sends message F to the master, piggybacking the acknowledgement to A.
3. The master sends message B to slave 1, but it contains an error.

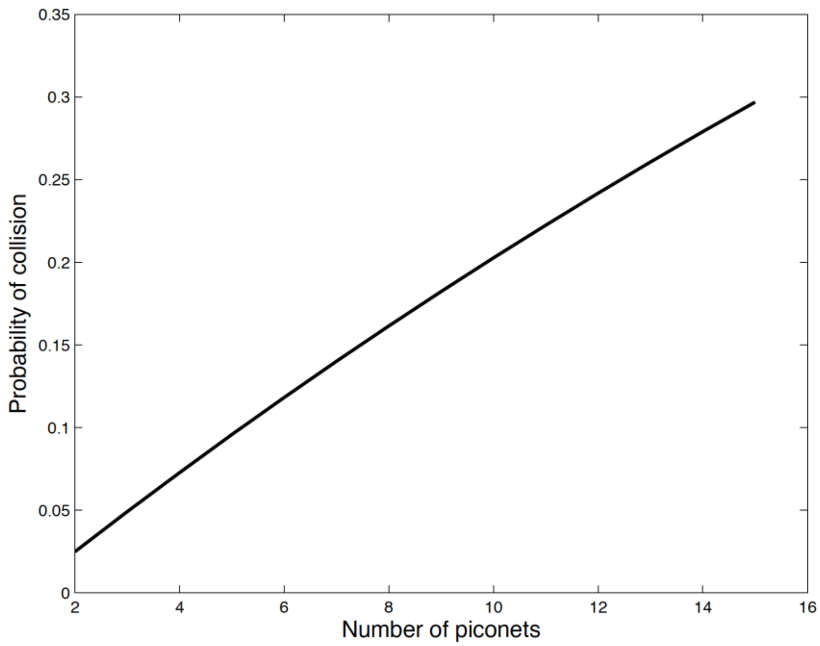
4. Slave 1 sends message G back, saying that the last message received contained errors.
5. The master retransmits message B, using the same SEQN bit, and acknowledges G.
6. Slave 1 sends message H, piggybacking the acknowledgement to B.
7. The master sends message X to slave 2.
8. Slave 2 sends message Z, piggybacking the acknowledgement to X. Message Z contains an error.
9. The master sends message C to slave 1, piggybacking the acknowledgement to H.
10. Slave 1 acknowledges C, but has no message to send.
11. The master sends a NAK to slave 2, indicating that its last message contained errors.
12. Slave 2 retransmits message Z.

12.2 Packets have a 79/80 probability of not colliding. The polling messages that are sent before them also have a 79/80 probability of not colliding.
The probability of success = $(79/80)^2$ = 0.9752
The probability of collision = 1 – 0.9752 = 0.0248

12.3 Use the binomial distribution to find the probability for the number of other piconets that will be using the same frequency. Use p=1/80 as the probability of choosing this frequency.
P = Pr{no other piconets are using this frequency} = $\binom{9}{0} p^0 (1-p)^9$
And the probability for this to be true for both the polling message and the packet
= $P^2 = \left(1 - \frac{1}{80}\right)^{18} = 0.7974$
The probability of collision is 1 – 0.7974 = 0.2026

As an aside, what is the probability all 10 piconets will be successful?
The proportion of possible choices for 10 piconets to all choose different frequencies is $\frac{\binom{80}{10}}{80^{10}} = 0.5564$
0.5564^2 = 0.3096 probability of all being successful

12.4



12.5 Probability of having no collision in the polling message slot then 5 slots without collisions = $\left(1 - \frac{1}{80}\right)^6 = 0.9273$

12.6 **a.** (28 mA*h) × (1.5 V) × (1 yr/365 days) × (1 day/24 h) × (1 A/1000 mA)/(2 yrs) = 2.40×10^{-6} W

b. (10% on) × (100% power) + (90% off) × (25% power) = 32.5% average power. 1/0.325 = 3.08 times longer life.

c. For 2 years, (1 mW) × 0.05 × (2 yrs) × (365 days/1 yr) × (24 H/1 day) / (1.5 V) = 584 mAh. This far exceeds the 28 mAh battery capacity already, no matter the other power consumption. The battery will not last 2 yrs.

12.7 These include light, sound, motion, heat, wave motion, wind.

12.8 L_{dB} = 20 log(f) +10n log (100) – 147.56

$L_{dB,60\ GHz} - L_{dB,2.4\ GHz}$ = (20 log(60×10^9) +10 log (d) – 147.56) – (20 log(2.4×10^9) +10 log (d) – 147.56) = 27.96 dB

27.96 dB more transmitter power is required.

For example, if originally 1 mW, now 623 mW is needed.

12.9 a. During a burst of S seconds, a total of MS octets are transmitted. A burst empties the bucket (b octets) and, during the burst, tokens for an additional rS octets are generated, for a total burst size of (b + rS). Thus,

$b + rS = MS$

$S = b/(M - r)$

b. $S = (250 \times 10^3)/(23 \times 10^6) \approx 11$ msec

Chapter 13 Cellular Wireless Networks

ANSWERS TO QUESTIONS

13.1 Hexagon

13.2 For frequency reuse in a cellular system, the same set of frequencies are used in multiple cells, with these cells separated from one another by enough distance to avoid interference.

13.3 **Adding new channels:** Typically, when a system is set up in a region, not all of the channels are used, and growth and expansion can be managed in an orderly fashion by adding new channels. **Frequency borrowing:** In the simplest case, frequencies are taken from adjacent cells by congested cells. The frequencies can also be assigned to cells dynamically. **Cell splitting:** In practice, the distribution of traffic and topographic features is not uniform, and this presents opportunities of capacity increase. Cells in areas of high usage can be split into smaller cells. **Cell sectoring:** With cell sectoring, a cell is divided into a number of wedge-shaped sectors, each with its own set of channels, typically 3 or 6 sectors per cell. Each sector is assigned a separate subset of the cell's channels, and directional antennas at the base station are used to focus on each sector. **Microcells:** As cells become smaller, antennas move from the tops of tall buildings or hills, to the tops of small buildings or the sides of large buildings, and finally to lamp posts, where they form microcells. Each decrease in cell size is accompanied by a reduction in the radiated power levels from the base stations and the mobile units. Microcells are useful in city streets in congested areas, along highways, and inside large public buildings.

13.4 To complete a call to a mobile unit, the base stations in a number of cells will send out a page signal in an attempt to find the mobile unit and make the connection.

13.5 **Cell blocking probability:** the probability of a new call being blocked, due to heavy load on the BS traffic capacity. In this case, the mobile unit is handed off to a neighboring cell based not on signal quality but on traffic capacity. **Call dropping probability:** the probability that, due to a handoff, a call is terminated. **Call completion probability:** the probability that an admitted call is not dropped before it terminates. **Probability of unsuccessful handoff:** the probability that a handoff is executed while the reception conditions are inadequate. **Handoff blocking probability:** the probability that a handoff cannot be successfully completed. **Handoff probability:** the probability that a handoff occurs before call termination. **Rate of handoff:** the number of handoffs per unit time. **Interruption duration:** the duration of time during a handoff in which a mobile is not connected to either base station. **Handoff delay:** the distance the mobile moves from the point at which the handoff should occur to the point at which it does occur.

13.6 As the mobile unit moves away from the transmitter, the received power declines due to normal attenuation. In addition, the effects of reflection, diffraction, and scattering can cause rapid changes in received power levels over small distances. This is because the power level is the sum from signals coming from a number of different paths and the phases of those paths are random, sometimes adding and sometimes subtracting. As the mobile unit moves, the contributions along various paths change.

13.7 **Open-loop power control** depends solely on the mobile unit, with no feedback from the BS, and is used in some SS systems. **Closed loop power control** adjusts signal strength in the reverse (mobile to BS) channel based on some metric of performance in that reverse channel, such as received signal power level, received signal-to-noise ratio, or received bit error rate.

13.8 The **mean rate of calls** is the number of calls attempted in a unit time, so its dimensions are calls per second or a similar dimension. **Traffic intensity** is a normalized version of mean rate of calls, and equals the average number of calls arriving during the average holding period. Thus, traffic intensity is dimensionless.

13.9 **Digital traffic channels:** The most notable difference between the two generations is that first generation systems are almost purely analog, whereas second generation systems are digital. In particular, the first generation systems are designed to support voice channels using FM; digital traffic is supported only by the use of a modem that converts the digital data into analog form. Second generation systems provide digital traffic channels. These readily support digital data; voice traffic is first encoded in digital form before transmitting. Of course, for second-generation systems, the user traffic (data or digitized voice) must be converted to an analog signal for transmission between the mobile unit and the base station. **Encryption:** Because all of the user traffic, as well as control traffic, is digitized in second-generation systems, it is a relatively simple matter to encrypt all of the traffic to prevent eavesdropping. All second-generation systems provide this capability, whereas first generation systems send user traffic in the clear, providing no security. **Error detection and correction:** The digital traffic stream of second-generation systems also lends itself to the use of error detection and correction techniques. The result can be very clear voice reception. **Channel access:** In first generation systems, each cell supports a number of channels. At any given time a channel is allocated to only one user. Second generation systems also provide multiple channels per cell, but each channel is dynamically shared by a number of users using time division multiple access (TDMA) or code division multiple access (CDMA).

13.10 **Frequency diversity:** Because the transmission is spread out over a larger bandwidth, frequency-dependent transmission impairments, such as noise bursts and selective fading, have less effect on the signal. **Multipath resistance:** The chipping codes used for CDMA not only exhibit low cross-correlation but also low autocorrelation. Therefore, a version of the signal that is delayed by more than one chip interval does not interfere with the dominant signal as much as in other multipath environments. **Privacy:** Because spread spectrum is obtained by the use of noise-like signals, where each user has a unique code, privacy is inherent. **Graceful degradation:** With FDMA or TDMA, a fixed number of users can simultaneously access the system. However, with CDMA, as more users simultaneously access the system, the noise level and hence the error rate increases; only gradually does the system degrade to the point of an unacceptable error rate.

13.11 **Self-jamming:** Unless all of the mobile users are perfectly synchronized, the arriving transmissions from multiple users will not be perfectly aligned on chip boundaries. Thus the spreading sequences of the different users are not orthogonal and there is some level of cross-correlation. This is distinct from either TDMA or FDMA, in which for reasonable time or frequency guardbands, respectively, the received signals are orthogonal or nearly so. **Near-far problem:** Signals closer to the receiver are received with less attenuation than signals farther away. Given the lack of complete orthogonality, the transmissions from the more remote mobile units may be more difficult to recover. Thus, power control techniques are very important in a CDMA system. **Soft handoff:** A smooth handoff from one cell to the next requires that the mobile acquire the new cell before it relinquishes the old. This is referred to as a soft handoff, and is more complex than the hard handoff used in FDMA and TDMA schemes.

13.12 **Hard handoff:** When the signal strength of a neighboring cell exceeds that of the current cell, plus a threshold, the mobile station is instructed to switch to a new frequency band that is within the allocation of the new cell. **Soft handoff**: a mobile station is temporarily connected to more than one base station simultaneously. A mobile unit may start out assigned to a single cell. If the unit enters a region in which the transmissions from two base stations are comparable (within some threshold of each other), the mobile unit enters the soft handoff state in which it is connected to the two base stations. The mobile unit remains in this state until one base station clearly predominates, at which time it is assigned exclusively to that cell.

13.13 Voice quality comparable to the public switched telephone network; 144 kbps data rate available to users in high-speed motor vehicles over large areas; 384 kbps available to pedestrians standing or moving slowly over small areas; Support (to be phased in) for 2.048 Mbps for office use; Symmetrical and asymmetrical data transmission rates; Support for both packet switched and circuit switched data services; An adaptive interface to the Internet to reflect efficiently the common asymmetry between inbound and outbound traffic; More efficient use of the available spectrum in general; Support for a wide variety of mobile equipment; Flexibility to allow the introduction of new services and technologies.

13.14 Wideband CDMA and CDMA2000.

13.15 Wideband CDMA: HSDPA, HSUPA, HSPA+
CDMA2000: 1xEV-DO Rel. 0, Rev. A, Rev. B

ANSWERS TO PROBLEMS

13.1 **a.** We have the number of clusters M = 16; bandwidth assigned to cluster B_{CL} = 40 MHz; bandwidth required for each two-way channel b_{ch} = 60 kHz. The total number of simultaneous calls that can be supported by the system is
$k_{SYS} = MB_{CL}/b_{ch}$ = 10,666 channels

b. Total number of channels available is K = B_{CL}/b_{ch} = 666. For a frequency reuse factor N, each cell can use $k_{CE} = K/N$ channels.
For N = 4, k_{CE} = 166 channels
For N = 7, k_{CE} = 95 channels
For N = 12, k_{CE} = 55 channels
For N = 19, k_{CE} = 35 channels

c. For N = 4, area = 64 cells; For N = 7, area = 112 cells; For N = 12, area = 192 cells; For N = 19, area = 304 cells.

d. From part b, we know the number of channels that can be carried per cell for each system. The total number of channels available is just 100 times that number, for a result of 16600, 9500, 5500, 3500, respectively. Source: [CARN99]

13.2 **a.** Assume every location is covered by a femtocell and a macrocell.
20 channels to every femtocell
180 channels left to be distributed among 4 cells per cluster and 3 sectors per cell, so 180/12 = 15 channels per sector.
Total of 35 channels available at every location.

b. 20 macrocells, 180/4 = 45 channels per macrocell, 20 × 45 = 900 channels
20 femtocells, 20 channels per femtocell, 20 × 20 = 400 channels
Total: 1300 channels

c. Only those channels from macrocells are available, so 15 channels at every location.

d. X channels per femtocell
(200 – X)/12 channels per sector
X + (200 – X) / 12 = 68
X = (68 – 200 / 12) × 12 / 11
X = 56 channels per femtocell
(200 – X) / 12 = 12 channels per macrocell sector

13.3 **a.** Steps a and b are the same. The next step is placing the call over the ordinary public switched telephone network (PSTN) to the called subscriber. Steps d, e, and f are the same except that only the mobile unit can be involved in a handoff.

b. Instead of steps a, b, and c, the process starts with a call coming in from the PSTN to an MTSO. From there, steps c, d, e, and f are the same except that only the mobile unit can be involved in a handoff.

13.4 This causes additional interference to co-channel users.

13.5 Suppose that a thermostat on a heating system is set to 20° C. Suppose the temperature in the room is greater than 20° C and falling. The heating system may not click on until, say, 19° C. As the temperature in the room rises, the thermostat may cause the heater to remain on until room temperature reaches 21° C.

13.6 $A = \lambda h = 1 \times (23/60) = 0.383$ Erlangs

13.7 **a.** For a given traffic level (A) and given capacity (N), what is the probability of blocking (P)?
$(10.5 - 10.07)/(11.1 - 10.07) = (P - 0.002)/(0.005 - 0.002)$; $P = 0.00325$

b. What traffic level can be supported with a given capacity to achieve a given probability of blocking?
$(0.015 - 0.01)/(0.02 - 0.01) = (A - 12.03)/(13.19 - 12.03)$; $A = 12.61$

c. For a given traffic level, what capacity is needed to achieve a certain upper bound on the probability of blocking ?
$(6 - 3.96)/(11.1 - 3.96) = (N - 10)/(20 - 10)$; $N = 12.857$

13.8 **a.** The total number of available channels is K = 33000/50 = 660. For a frequency reuse factor N, each cell can use $k_{CE} = K/N$ channels.
For $N = 4$, $k_{CE} = 165$ channels
For $N = 7$, $k_{CE} = 94$ channels
For $N = 12$, $k_{CE} = 55$ channels

b. 32 MHz is available for voice channels for a total of 640 channels.
For $N = 4$, we can have 160 voice channels and one control channel per cell
For $N = 7$, we can have 4 cells with 91 voice channels and 3 cells with 92 voice channels, and one control channel per cell.
For $N = 12$, we can have 8 cells with 53 voice channels and 4 cells with 54 voice channels, and one control channel per cell. Source: [RAPP96]

13.9 a. Number of 30-kHz channels = 12500/30 = 416
Number of voice channels = 416 – 21 = 395
Number of voice channels per cell = 395/7 = 56

b. $(56 - 40)/(70 - 40) = (A - 31)/(59.13 - 31)$; $A = 46$ Erlangs/cell

c. Number of calls/hour/cell = 46/(100/3600) = 1656
Number of calls/hour/km^2 = 1656/8 = 207

d. Number of users/hour/cell = 1656/1.2 = 1380
Number of users/hour/channel = 1380/56 = 24.6

e. The total number of cells is 4000/8 = 500
η = (46 Erlangs/cell × 500 cells)/(12.5 MHz × 4000 km^2)
= 0.46 Erlangs/MHz/km^2 Source: [GARG96]

13.10 $(12.5 \times 10^6 - 2(10 \times 10^3))/(30 \times 10^3) = 416$

13.11 a. The amount of bandwidth allocated to voice channels ($B_c N_t$) must be no greater than the total bandwidth (B_w). Therefore $\eta_a \leq 1$.

b. $x = (30 \times 10^3 \times 395)/(12.5 \times 10^6) = 0.948$

13.12 a. Number of subscribers = Traffic/0.03

Cell number	1	2	3	4	5	6	7
Subscribers	1026.7	2223.3	1620.0	1106.7	1273.3	1260.0	1086.7

b. Number of calls per hour per subscriber = $\lambda = A/h = 0.03/(120/3600) = 0.9$

c. Multiply results of part (a) by 0.9

Cell number	1	2	3	4	5	6	7
Calls per hour	924	2001	1458	996	1146	1134	978

d. The table in the problem statement gives the value of A. Use $P = 0.02$. Find N.

Cell number	1	2	3	4	5	6	7
Channels	40	78	59	43	48	48	42

e. Total number of subscribers = the sum of the values from part (a) = 9597

f. From (d), the total number of channels required = 358
Average number of subscribers per channel = 9597/358 = 26.8

g. Subscriber density = 9597/3100 = 3.1 subscribers per km^2

h. Total traffic = the sum of the values from table in the problem statement = 287.9

i. Erlangs per km^2 = 287.9/3100 = 0.09

j. The area of a hexagon of radius R is $A = 1.5R^2\sqrt{3}$. For A = 3100/7 = 442.86 km^2
We have R = 13 km
Source: [GARG96]

Chapter 14 Fourth Generation Systems and LTE-Advanced

ANSWERS TO QUESTIONS

14.1 High-speed, universally accessible wireless service, varieties of devices, 100 Mbps mobile data rates, 1 Gbps stationary data rates, all-IP packet switched network for voice, data, and video

14.2 3GPPP Releases 10 and beyond.

14.3

- Mobility Management Entity (MME): UE network connections, security
- Serving Gateway (SGW): Sending and receiving packet between the eNodeB and the core network.
- Packet Data Network Gateway (PGW): Connects the EPC with external networks.
- Home Subscriber Server (HSS): Database of user information.

14.4

- Radio Resource Control (RRC) performs control plane functions to manage resources. It supervises management of RRC connections, bearers, mobility, and measurement reporting.
- The Packet Data Convergence Protocol (PDCP) delivers packets to the UE from the eNodeB.
- Radio Link Control (RLC) segments or concatenates data units, performs ARQ, and delivers packets in sequence to higher layers.

14.5

- Logical channels provide data and control traffic delivery MAC services to the RLC.
- Transport channels provide PHY layer services to the MAC layer.
- Physical channels define the time and frequency resources used to carry information.

14.6 LTE uses 4.7 and 16.7 μs cyclic prefixes.

14.7 The LTE FDD frame structure supports normal and extended CP modes for 7 and 6 OFDM symbols respectively during a slot. The LTE TDD frame structure is compatible with 3GPP legacy systems and supports variable balance between uplink and downlink subframes. Also involved is a special subframe to accommodate the switch uplink-to-downlink, which reduces bandwidth efficiency somewhat.

14.8 A resource block is a collection of twelve 15 kHz subcarriers in a slot. This consists of either 6 or 7 OFDM symbols, resulting in 72 or 84 resource elements per resource block.

14.9 As seen in Table 14.7, LTE supports QPSK, 16QAM, and 64QAM.

14.10

1. Power on the UE
2. Select a network, PLMN, LTE, or a previous generation network.
3. Select a cell, based on suitable downlink and uplink frequencies.
4. Contention-based random access:
    a. The mobile transmits a random access preamble and keeps on retransmitting with increasing power until it receives a response or reaches a maximum number of retransmissions.
    b. The base station will provide a random access response. The base station provides a C-RNTI (cell radio network temporary identifier), timing advance, and resources on the PUSCH.
5. The mobile then sends an RRC Connection Request to move to RRC_CONNECTED state. The eNodeB responds with an RRC connection setup that configures the mobile's physical layer, MAC protocols, and signaling radio bearer. The mobile responds with the confirmation message *RRC Connection Setup Complete* that is also forwarded to the MME to serve as an EPS mobility management exchange with the MME.
6. Attach procedure: Four main objectives are accomplished. The UE registers its location with the MME, the network configures a radio bearer for non-access stratum signaling messages, the network gives an IP address, and the network sets up a default EPS bearer.
7. Packet Transmission: Then the UE transmits and receives data. It is now in EMM-REGISTERED, ECM-CONNECTED, and RRC_CONNECTED states and will stay there as long as it is communicating.
   For downlink transmission:
    a. The base station begins by sending a *scheduling command*.
    b. Then the base station uses the downlink shared channel (DL-SCH) and the PDSCH to send the data.
    c. In response, the mobile sends a hybrid ARQ acknowledgment on the PUCCH.

For uplink transmission, the mobile must first indicate to the BS that it wishes to send.

a. The base station begins by sending a *scheduling grant*.
b. The mobile sends data on the UL-SCH and the PUSCH.
c. If unsuccessful, the base station can respond with a simple NACK to request a retransmission, or the BS can respond with a new scheduling grant for resource block allocation or modulation.

8. Improve quality of service: If the user needs better QoS than the default bearer can provide, it sends an *ESM Bearer Resource Allocation Request* to the MME.

14.11 Carrier aggregation combines up to 5 component carriers. This can be accomplished with either intra-band contiguous (adjacent component carriers), intra-band noncontiguous (non-adjacent subcarriers within the same band), and inter-band noncontiguous (non-adjacent subcarriers within different bands) aggregation.

14.12 LTE-Advanced uses eight layer multiplexing for up to eight separate transmissions on the downlink to a single UE. If antennas are used for multiuser MIMO, up to four mobiles can receive signals simultaneously. LTE-Advanced uses downlink reference signals to provide this MIMO functionality. In the uplink, LTE-Advanced supports multiple UE antennas so up to four transmission layers can be used.

14.13 Relay nodes form smaller coverage areas to extend coverage of macro cells. They use the frequencies of the macro cell to retransmit the signals from a UE to eNodeB.

14.14 A femtocell is a low-power, short-range, self-contained base station designed for indoor coverage.

14.15

- Interference coordination provides indicators to neighboring eNodeBs to avoid interference possibilities. eICIC provides special indicators for small cells.
- Coordinated Multipoint Transmission and Reception (CoMP) is not an interference avoidance technique, but rather cooperation among eNodeB to increase power to mobiles at cell edges and reduce interference at cell edges. It may use techniques to steer antenna beam nulls and mainlobes, transmit data simultaneously from multiple transmission points to the same UE, and transmit from multiple transmission points but only one at a time.

ANSWERS TO PROBLEMS

14.1 4G achieves over 5 Mbps. 3G achieved 2 Mbps.

14.2 LTE-Advanced: Efficiency = 1000 Mbps/100 MHz = 10.0 bps/Hz
LTE: 100 Mbps/20 MHz = 5.0 bps/Hz
LTE-Advanced has improved efficiency by a factor of 2.

14.3 30 bps/Hz for LTE-Advanced downlink compared to 5 bps/Hz for LTE. This is a factor of 6 improvement in spectral efficiency.

14.4 For LTE-Advanced, spectral efficiency is 30 bps/Hz on the downlink and 15 bps/Hz on the uplink. This is a factor of ½ for spectral efficiency on the uplink versus the downlink. This same factor of ½ is true for LTE.

14.5 From Table 14.3, QCI 6, 8, or 9 can be used for buffered streaming video, all which have a 300 ms delay budget. See Section 3.5 for more discussion of this type of traffic.

14.6 Conversational traffic needs a lower round-trip delay so that users on both ends are able to interact in a natural manner. See Section 3.5 for discussion of QoS.

14.7 From 1/10 (configuration 5) to 6/10 (configuration 0) of the subframe can be used for uplink traffic. The special (S) subframes do not carry any actual uplink traffic.

14.8 15 MHz minus 1.5 MHz guard band (10%).
Also 900 subcarriers × 15 kHz per subcarrier = 13.5 MHz

14.9 From Table 14.7, CQI 9 achieves 2.4063 bps/Hz.
From Table 14.6, a 15 MHz channel has 13.5 MHz occupied bandwidth.
13.5 × 2.4063 = 32.49 Mbps

14.10 7 OFDM symbols: 160 samples for first CP, 144 samples for the next 6 CPs, 2048 samples for each OFDM symbol = $(160 + 6 \times 144 + 7 \times 2048) \times T_s = 15360/30720000 = 0.5$ μs.

6 OFDM symbols: 512 samples for each CP, 2048 samples for each OFDM symbol $= (6 \times 512 + 6 \times 2048) \times T_s = 15360/30720000 = 0.5$ μs.

14.11 CQI 6 can achieve 1.1758 bps/Hz.
$(3.0 \times 10^6 \text{ bps})/(1.1758 \text{ bps/Hz}) = 2.551$ MHz is required
$(2.551 \times 10^6 \text{ Hz})/(180 \times 10^3 \text{ Hz/RB}) = 14.17$ resource blocks
15 resource blocks are needed to meet this requirement.

14.12 CQI 13 can achieve 4.5234 bps/Hz.

$(3.0 \times 10^6 \text{ bps})/(4.5234 \text{ bps/Hz}) = 663.2$ Hz is required

$(663.2 \times 10^3 \text{ Hz})/(180 \times 10^3 \text{ Hz/RB}) = 3.68$ resource blocks

4 resource blocks are needed to meet this requirement.

14.13 Assume the system was designed for slow fading conditions so the system adapts more quickly than the channel changes. System adaptations occur every 1 ms, every subframe.

T_s = subframe time = 1 ms

Assume $T_c \gg T_s$

$T_c > 10T_s$

$T_c > 10$ ms

Chapter 15 Mobile Applications and Mobile IP

ANSWERS TO QUESTIONS

15.1 Mobile platforms require the same core systems facilities as desktops: memory management, process scheduling, device drivers, and security.

15.2 APIs typically allow the programmer to access
- GUI interaction with the user
- Sensor data available (GPS, acceleration, etc.)
- Authentication and account access
- Interaction with remote servers

15.3
- The **Applications and Framework** layer and APIs allow access to lower-layer services. It provides high-level building blocks that programmers use to create new apps. And it includes a core set of general-purpose applications.
- The **Binder Inter-Process Communication (IPC)** mechanism allows the application framework to cross process boundaries and call into the Android system services code.
- **Android system services** access the underlying hardware and kernel functions. System services include Activity Manager, Window Manager, Content Providers, View System, Notification Manager, and Power Manager.
- The **Android Runtime/Dalvik** layer provides the Android Runtime component and the Dalvik virtual machine .
- The **Hardware** Abstraction **Layer (HAL)** provides a standard interface to kernel-layer device drivers.
- The **Linux kernel** provides core system services such as security, memory management, process management, network stack, and driver model. The kernel also acts as an abstraction layer between the hardware and the rest of the software stack.

15.4 System services deal with system functions visible to the application.
- **Activity Manager:** Manages the lifecycle of applications. It is responsible for starting, stopping, and resuming the various applications.
- **Window Manager:** Allows applications to declare their client area, and use features like the status bar.
- **Content Providers:** Encapsulate application data that need to be shared between applications, similar to a database.

- **View System:** Provides the user interface primitives, such as buttons, list boxes, date pickers, and other controls, as well as UI events (such as touch and gestures).
- **Notification Manager:** Manages events, such as arriving messages and appointments.
- **Power Manager:** Provides an interface to features added to the Linux kernel to enhance the ability to perform power management.

15.5 Dalvik is a virtual machine for Android, which is similar to the Java virtual machine. It is implemented to optimize for the mobile platforms on which Android applications run by handling resources in a way tailored to mobile platforms.

15.6 Each component in the Android system structures its behavior around a state machine called a lifecycle. While an application comprises multiple activities, only one is on screen (active) at any time. Navigating between activities happens by the user performing actions (such as clicking the "back" button on the device) and the application sending requests (e.g., to start a new activity) via Intents (messages sent to the Android system that trigger certain action). The Activity class has methods (such as onCreate(), which is called when the activity is first created) that the developer extends to implement the desired application logic (such as displaying buttons on the screen).

15.7 A **mobile user** is connected to one or more applications across the Internet such that the user's point of attachment changes dynamically, and that all connections are automatically maintained despite the change. For a **nomadic user**, the user's Internet connection is terminated each time the user moves and a new connection is initiated when the user dials back in.

15.8 Tunneling is a process in which an IP datagram is encapsulated with an outer IP header so as to be transmitted across the Internet using the destination address and parameters of the outer header.

15.9 **Discovery:** A mobile node uses a discovery procedure to identify prospective home agents and foreign agents. **Registration:** A mobile node uses an authenticated registration procedure to inform its home agent of its care-of address. **Tunneling:** Tunneling is used to forward IP datagrams from a home address to a care-of address.

15.10 Discovery makes use of the existing ICMP (Internet control message protocol) by adding the appropriate extensions to the ICMP header.

15.11 The destination care-of address can either be that of a foreign agent, or it can be a co-located address that is associated physically with the node.

15.12 The foreign agent address is used when there is a foreign agent present and available on the foreign network. The co-located address is used if there is no foreign agent or all foreign agents on the foreign network are busy.

ANSWERS TO PROBLEMS

15.1 a.

MAC-H DA = MACDA(E-Z)	LLC-H DSAP = DSAP (E)	IP-H DA = IPDA(HA-E)	TCP-H	Data

b. MAC frame leaving D:

MAC-H DA = MACDA(R3-Z)	LLC-H DSAP = DSAP (R3)	IP-H DA = IPDA(HA-E)	TCP-H	Data

IP datagram leaving R3 (using header formats of Figure 12.7a):

IP-H DA = IPDA(CA-E)	IP-H DA = IPDA(HA-E)	TCP-H	Data

c. IP datagram leaving R3 (using header formats of Figure 12.7b):

IP-H DA = IPDA(CA-E)	IP-H DA = IPDA(HA-E)	TCP-H	Data

Legend: MAC-H = MAC header; LLC-H = LLC header; IP-H = IP header; TCP-H = TCP header; MACDA(E-Z) = MAC destination address of E on LAN Z; DSAP(E) = LLC DSAP for E; IPDA(HA-E) = E's home IP address; IPDA(CA-E) = E's care-of IP address.

15.2 a. IP datagram arriving at R1 from Internet (using header formats of Figure 12.7a):

IP-H DA = IPDA(R1)	IP-H DA = IPDA(HA-A)	TCP-H	Data

MAC frame leaving R1 onto LAN X:

MAC-H DA = MACDA(A-X)	LLC-H DSAP = DSAP (A)	IP-H DA = IPDA(HA-A)	TCP-H	Data

b. IP datagram arriving at R1 from Internet (using header formats of Figure 12.7b):

IP-H DA = IPDA(R1)	IP-H DA = IPDA(HA-A)	TCP-H	Data

MAC frame leaving R1 onto LAN X:

MAC-H DA = MACDA(A-X)	LLC-H DSAP = DSAP (A)	IP-H DA = IPDA(HA-A)	TCP-H	Data

c. IP datagram arriving at R1 from Internet (using header formats of Figure 12.7a):

IP-H DA = IPDA(A-X)	IP-H DA = IPDA(HA-A)	TCP-H	Data

MAC frame leaving R1 onto LAN X:

MAC-H DA = MACDA(A-X)	LLC-H DSAP = DSAP (A)	IP-H DA = IPDA(A-X)	IP-H DA = IPDA(HA-A)	TCP-H	Data

d. IP datagram arriving at R1 from Internet (using header formats of Figure 12.7b):

IP-H DA = IPDA(A-X)	IP-H DA = IPDA(HA-A)	TCP-H	Data

MAC frame leaving R1 onto LAN X:

MAC-H DA = MACDA(A-X)	LLC-H DSAP = DSAP (A)	IP-H DA = IPDA(A-X)	IP-H DA = IPDA(HA-A)	TCP-H	Data

15.3 Home address, care-of address, lifetime.

15.4 Home address, home agent address, MAC address, lifetime.

Chapter 16 Long Range communications

ANSWERS TO QUESTIONS

16.1 **Coverage area:** global, regional, or national. The larger the area of coverage, the more satellites must be involved in a single networked system. **Service type:** fixed service satellite (FSS), broadcast service satellite (BSS), and mobile service satellite (MSS). This chapter is concerned with FSS and BSS types. **General usage:** commercial, military, amateur, experimental.

16.2 **(1)** The area of coverage of a satellite system far exceeds that of a terrestrial system. In the case of a geostationary satellite, a single antenna is visible to about one-fourth of the earth's surface. **(2)** Spacecraft power and allocated bandwidth are limited resources that call for careful tradeoffs in earth station/satellite design parameters. **(3)** Conditions between communicating satellites are more time invariant that those between satellite and earth station or between two terrestrial wireless antennas. Thus, satellite-to-satellite communication links can be designed with great precision. **(4)** Transmission cost is independent of distance, within the satellite's area of coverage. **(5)** Broadcast, multicast, and point-to-point applications are readily accommodated. **(6)** Very high bandwidths or data rates are available to the user. **(7)** Although satellite links are subject to short-term outages or degradations, the quality of transmission is normally extremely high. **(8)** For a geostationary satellite, there is an earth-satellite-earth propagation delay of about one-fourth of a second. **(9)** A transmitting earth station can in many cases receive its own transmission.

16.3 **(1)** The orbit may be circular, with the center of the circle at the center of the earth, or elliptical, with the earth's center at one of the two foci of the ellipse. **(2)** A satellite may orbit around the earth in different planes. An **equatorial orbit** is directly above the earth's equator. A **polar orbit** passes over both poles. Other orbits are referred to as **inclined orbits**. **(3)** The altitude of communications satellites is classified as geostationary orbit (GEO), medium earth orbit (MEO), and low earth orbit (LEO).

16.4 LEO, GEO and HEO stand for low earth orbit, geostationary (or geosynchronous) orbit, and highly elliptical orbit, respectively. The traditional GEO satellite is in a circular orbit in an equatorial plane such that the satellite rotates about the earth at the same angular velocity that the earth spins on its axis. To accomplish this the satellite must be approximately 35,838 km above the earth’s surface at the equator. LEO satellites are satellites with much lower orbits, on the order of 700 to 1,400 km high. Finally, HEO satellites are characterized by an orbit that is an ellipse with one axis very substantially larger than the other. The height of the orbit can vary; it is the **shape** of the orbit that characterizes this type of satellite.

Because of the high altitude of the GEO satellite the signal strength is relatively weak compared to LEOs. Frequency reuse is more difficult because the antenna beam (all other things being equal) covers a much greater area from a GEO than from a LEO. The propagation delay for a GEO satellite is about 1/4th of second; that of a LEO satellite is much less. Because the GEO satellite must be over the equator, the coverage near the north and south poles is inadequate. For these regions, better communication can be achieved by LEO or HEO satellites. HEO satellites have the additional advantage that they spend most of their time at the "high" part of their orbit so that you get the most coverage for the longest time for this type of satellite. On the other hand, tracking and handoff is not necessary for GEO satellites because they appear stationary relative to the earth. LEO satellites, since they are so low travel very much faster, and cover less area than GEO so that tracking is more difficulty and passing off is frequent. HEOs require tracking and handoffs, as well. However, if the HEOs have high orbits the handoff frequency can be much less and the tracking easier than for LEOs.

16.5 You would use GEOs when the earth stations are not near the poles, when there is a premium on not having to steer the earth station antennas, and when broad earth coverage is important, for television broadcasting for instance. HEOs are primarily of use when coverage of areas near one of the poles is essential, such as the use of the Molniya satellites to cover the northern parts of the former Soviet Union. LEOs are useful for point-to-point communication, and for extensive frequency reuse. Since LEOs have much less propagation delay they are useful for interactive data services. They also can cover polar regions. Finally, while you need many more LEOs for broad coverage, each satellite is much less expensive than a GEO.

16.6 **(1)** Distance between earth station antenna and satellite antenna. **(2)** In the case of the downlink, terrestrial distance between earth station antenna and the "aim point" of the satellite. **(1)** Atmospheric attenuation.

16.7 Oxygen and water.

16.8 Thermal noise, intermodulation noise, and crosstalk.

16.9 Three satellites are used to determine two possible locations (latitude, longitude, and elevation), one of which is typically absurd so the other is chosen. The fourth satellite is used to detect timing offsets between GPS and receiver clocks to correct distance calculations.

16.10 Medium Earth Orbits (MEOs).

16.11 Fixed broadband wireless access provides broadband data rates between fixed transmitters and receivers. The U.S. FCC previously defined "broadband" as 4/1 Mbps download/upload, but in 2015 changed to 24/3 Mbps download/upload.

16.12

- The **unsolicited grant service (UGS)** is intended for real-time applications that generate fixed-rate data. A service flow with a data delivery service of UGS gets uplink resources assigned at uniform periodic intervals without requesting them each time.
- The **real-time variable rate (RT-VR)** downlink service is intended for time-sensitive applications. RT-VR applications transmit at a rate that varies with time. On the downlink, RT-VR is implemented by transmitting the available data at uniform periodic intervals.
- On the uplink, the real-time variable rate service is called the **real-time polling service (rtPS)**. The BS issues a unicast poll (poll directed at a SS station) at periodic intervals, enabling the SS to transmit a block of data in each interval.
- The **extended real-time variable rate (ERT-VR)** service is to support real-time applications with variable data rates, which require guaranteed data and delay, for example, VoIP with silence suppression.
- On the uplink, this service is called **extended rtPS**. The BS provides unicast grants of bandwidth in an unsolicited manner, thus saving the latency of a bandwidth request. However, in this case the allocations are variable in size.
- The **non-real-time variable-rate (NRT-VR)** service is intended for applications that have bursty traffic characteristics, do not have tight constraints on delay and delay variation.
- On the uplink, the service is called **non-real-time polling service (nrtPS)**. The BS issues polls at varying intervals, depending on how much data has so far been transferred, so as to keep up with the required flow.
- Unused capacity is available for the **best effort (BE)** service. This service is suitable for applications that can tolerate variable delays and rates.

16.13 WirelessMAN-SC – Single carrier in 10-66 GHz with required line of sight.
WirelessMAN-OFDM – Below 11 GHz, non-LOS, support mobile users.
WirelessMAN-OFDMA – Below 11 GHz, non-LOS, mobile users, using OFDMA.

ANSWERS TO PROBLEMS

16.1 **a.** Rearranging the equation, we have $a^3 = T^2\mu/(4\pi^2)$.
One sidereal day is T = 86,164.1 s
$a^3 = (86{,}164.1)2 \times (3.986004418 \times 10^5)/(4\pi^2) = 7.496020251 \times 10^{13}$ km^3
a = 42,164 km

b. h = 35,794 km

16.2 a. $a = 6378.14 + 250 = 6628.14$ km

$T^2 = (4\pi^2 a^3)/\mu = (4\pi^2) \times (6628.14)^3/(3.986004418 \times 10^5) = 2.88401145 \times 107\ s^2$

$T = 5370.3$ s = 89 min 30.3 s

b. The linear velocity is the circumference divided by the period

$(2\pi a)/T = (41645.83)/(5370.3) = 7.775$ km/s

16.3 The received signal is, essentially, the same. The received power will increase by a factor of 4

16.4 received_power = transmitted_power + transmitted_gain + received_gain – path_loss

From Equation (2.2):

path_loss = 20 log $(4\pi d/\lambda)$ = 20 log $[4\pi d/(c/f)]$

= 20 log $[(4\pi \times 4 \times 10^7)/(2.727 \times 10^{-2})$ = –205.3 dB

received_power = 10 + 17 + 52.3 – 205.3 = –126 dBW

16.5 The total bandwidth is 500 MHz. The channel bandwidth is 12 × 36 = 432 MHz. So the overhead is ((500 - 432)/500) × 100% = 13.6%

16.6 a. The data rate R = 2 × QPSK baud rate = 120.272 Mbps

The frame duration T = 0.002 s

The number of frame bits $b_F = R \times T = 1.20272 \times 10^8 \times 2 \times 10^{-3} = 240544$ bits

Overhead calculation: Overhead bits $b_o = N_R b_R + N_T b_p + (N_R + N_T) b_G$

where

N_R = number of participating reference stations = 2

b_R = number of bits in reference burst = 576

N_T = number of participating traffic stations

b_p = number of preamble bits =560

b_G = number guard bits = 24

$b_o = (2)(576) + 560\ N_T + (2)(24) + 24\ N_T = 1200 + 584\ N_T$

$b_F - b_o = 240544 - (1200 + 584\ N_T) = 239344 - 584\ N_T$

$N_T = (b_F - b_o)/16512 = (239344 - 584\ N_T)/16512$

$(16512 + 584)\ N_T = 239344$

Therefore, NT = 14

b. $(b_F - b_o)/b_F = (239344 - 584 \times 14)/240544 = 0.96$

Source: [GLOV98]

16.7 **a.** The time, T_d, available in each station burst for transmission of data bits is

$$T_d = [T_{frame} - N(t_g + t_{pre})]/N$$

That is, take the frame time, subtract out all the guard and preamble times, and divide by the number of stations N.

$$T_d = [2000 - 5(5 + 20)]/5 = 375\ \mu s$$

A burst transmission rate of 30 Mbaud is 30 million signal elements per second and QPSK signal elements carry 2 bits, so the transmitted bit rate in each burst is R_b = 60 Mbps

The capacity of each earth station, C_b, is the number of data bits transmitted in one burst divided by the frame time:

$$C_b = (375 \times 60)/2000 = 11.25\ \text{Mbps}$$

The number of 64-kbps channels that can be carried is:

$$(11.25 \times 10^6)/(64{,}000) = 175$$

b. 11.25 Mbps

c. The total available capacity is 60 Mbps.
The total data transmission rate is 5 × 11.25 = 56.25 Mbps
Efficiency = 56.25/60 = 0.9375

16.8 The transponder must carry a total data bit rate of 15 + 10 + 5 = 30 Mbps. Thus, each frame carries 30 Mbps × 0.001 s = 30 kb The three preamble and guard times take up 3 × (10 + 2) = 36 μs in each frame, leaving 1000 – 36 = 964 μs for transmission of the data. Therefore, the burst bit rate is

$$\text{Rbit} = 30\ \text{kb}/964\ \mu s = 31.12\ \text{Mbps}$$

For QPSK, the symbol rate is half the bit rate = 15.56 Mbaud

16.9 The UL-MAP in the downlink subframe gives instructions to the SS about how to perform its transmissions. Therefore, it must be sent on the downlink.