CHAPTER 7

PRACTICE SET

Questions

Q7-1. The period of a signal is the inverse of its frequency: T = 1/*f*.

Q7-3. A signal is periodic if its time domain plot repeats itself; a signal is non-periodic if its time domain plot does not repeat.

Q7-5. Attenuation and noise are two out of three causes of transmission impairment; distortion is the third one.

Q7-7. This is baseband transmission because there is no modulation.

Q7-9. Baseband transmission means sending a digital or an analog signal without modulation using a low-pass channel.

Q7-11. The Nyquist theorem defines the maximum bit rate of a noiseless channel.

Q7-13. In general, block coding changes a block of *m* bits into a block of *n* bits, where *n* is larger than *m*. Block coding provides redundancy to ensure synchronization and to provide inherent error detecting.

Q7-15. Normally, analog transmission refers to the transmission of analog signals using a band-pass channel. Baseband digital or analog signals are converted to a complex analog signal with a range of frequencies suitable for the channel.

Q7-17. ASK is more susceptible to noise because amplitude is more affected by noise than frequency.

Q7-19.

a. FM changes the frequency of the carrier.

b. PM changes the phase of the carrier.

Q7-21. In multiplexing, the word *link* refers to the physical path. One link can be divided into *n* channels.

Q7-23. In synchronous TDM, each input has a reserved slot in the output frame. This can be inefficient if some input lines have no data to send. In statistical TDM, slots are dynamically allocated to improve bandwidth efficiency.

Q7-25. The transmission media is located beneath the physical layer and controlled by the physical layer.

Q7-27. The three major categories of guided media are twisted-pair, coaxial, and fiber-optic cables.

Q7-29. In sky propagation radio waves radiate upward into the ionosphere and are then reflected back to earth.

Problems

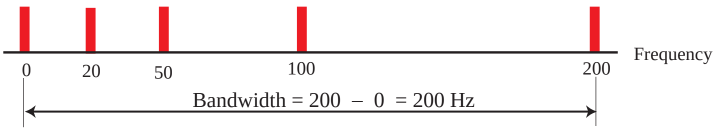
P7-1.

a. $T = 1/f = 1/(24\,\text{Hz}) = 0.0417\,\text{s} = 41.7 \times 10^{-3}\,\text{s}$ = **41.7 ms**

b. $T = 1/f = 1/(8\,\text{MHz}) = 0.000000125\,\text{s} = 0.125 \times 10^{-6}\,\text{s}$ = **0.125 μs**

c. $T = 1/f = 1/(140\,\text{kHz}) = 7.14 \times 10^{-6}\,\text{s} = 7.14 \times 10^{-6}\,\text{s}$ = **7.14 μs**

P7-3. See below:



P7-5. Each signal is a simple signal in this case. The bandwidth of a simple signal is zero. So the bandwidth is the same for both signals.

P7-7.

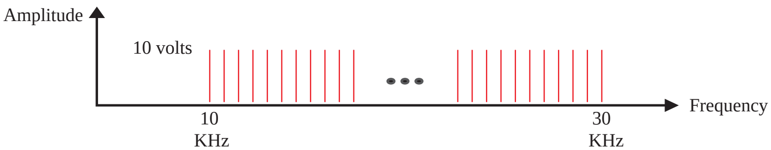
a. (10 / 1000) s = **0.01 s**

b. (8 / 1000) s = 0. 008 s = **8 ms**

c. ((100,000 × 8) / 1000) s = **800 s**

P7-9. The signal makes 8 cycles in 4 ms. The frequency is 8 /(4 ms) = **2 kHz**

P7-11. The signal is periodic, so the frequency domain is made of discrete frequencies with the bandwidth of 30 – 10 = 20 kHz. See below:



P7-13. We can calculate the attenuation as shown below:

$dB = 10 \log_{10} (90 / 100) =$ **–0.46 dB**

P7-15. The total gain is 3 × 4 = 12 dB. To find how much the signal is amplified, we can use the following formula:

$12 = 10 \log (P_2/P_1)$ → $\log (P_2/P_1) = 1.2$ → $P_2/P_1 = 10^{1.2} = 15.85$

The signal is amplified almost 16 times.

P7-17. Each cycle moves the front of the signal λ meter ahead (definition of the wavelength). In this case, we have

$1\ \mu m \times 1000 = 1000\ \mu m =$ **1 mm**

P7-19. SNR is the ratio of the powers. The power is proportion to the voltage square ($P = V^2/R$). Therefore, we have $SNR = (10)^2 / (10 \times 10^{-3})^2 = 10^6$. We then use the Shannon capacity to calculate the maximum data rate.

$C = 4{,}000 \log_2 (1 + 10^6) \approx$ **80 Kbps**

P7-21. To represent 1024 color levels, we need $\log_2 1024 = 10$ bits. The total number of bits are, therefore,

Number of bits = 1200 × 1000 × 10 = **12,000,000 bits**

P7-23. We have **SNR = (signal power)/(noise power).** However, power is proportional to the square of voltage. This means we have

$SNR = [(\text{signal voltage})^2] / [(\text{noise voltage})^2] = 20^2 =$ **400**

$SNR_{dB} = 10 \log_{10} SNR = 10 \log_{10} 400 =$ **26**

P7-25. We have (transmission time) = (packet length)/(bandwidth)

(transmission time) = (8,000,000 bits) / (200,000 bps) = 40 s

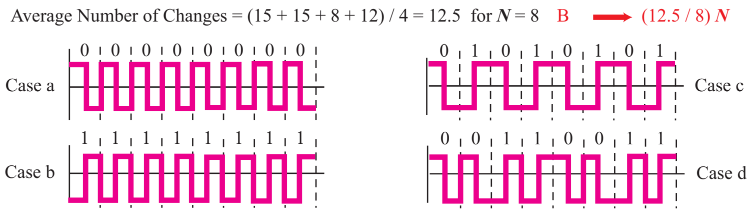
P7-27.

a. Number of bits = bandwidth × delay = 1 Mbps × 2 ms = **2000 bits**

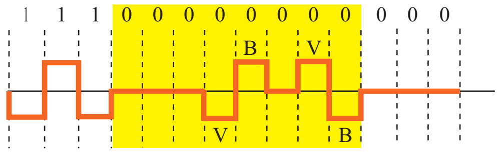
b. Number of bits = bandwidth × delay = 10 Mbps × 2 ms = **20,000 bits**

P7-29. The number of bits is calculated as (0.2 /100) × (1 Mbps) = **2000 bits.**

P7-31. Bandwidth is proportional to **(12.5/8)*N*.** The following figure shows the signal in each case.



P7-33. See the following figure. Since we specified that the last non-zero signal is positive, the first bit in our sequence is positive.



P7-35.

a.

$f_{max} = 0 + 200 = 200$ kHz → $f_s = 2 \times 200{,}000 =$ **400,000 samples/s**

$n_b = \log_2 1024 = 10$ bits/sample → $n = 400{,}000 \times 10 = 4$ Mbps

b.

$SNR_{dB} = 6.02 \times n_b + 1.76 = 61.96$

c.

$B_{PCM} = n_b \times B_{analog} = 10 \times 200$ kHz $= 2$ MHz

P7-37. We can first calculate the sampling rate (f_s).

$f_{max} = 0 + 4 = 4$ kHz → $f_s = 2 \times 4$ kHz = **8000 samples/s**

We then calculate the number of bits per sample.

$n_b = 30{,}000 / 8000 = 3.75$

We need to use the next integer $n_b = 4$. The value of SNR_{dB} is

$SNR_{dB} = 6.02 \times n_b + 1.72 =$ **25.8**

P7-39. We use the formula $S = (1/r) \times N$, but first we need to calculate the value of r for each case.

a. $r = \log_2 2 = 1 \rightarrow S = (1/1) \times (2000 \text{ bps}) =$ **2000 baud**

b. $r = \log_2 2 = 1 \rightarrow S = (1/1) \times (4000 \text{ bps}) =$ **4000 baud**

c. $r = \log_2 64 = 6 \rightarrow S = (1/6) \times (36{,}000 \text{ bps}) =$ **6000 baud**

P7-41. We use the formula $r = \log_2 L$ to calculate the value of r for each case.

a. $\log_2 8 = 3$ **b.** $\log_2 128 = 7$

P7-43. The number of points defines the number of levels, L. The number of bits per baud is the value of r. Therefore, we use the formula $r = \log_2 L$ for each case.

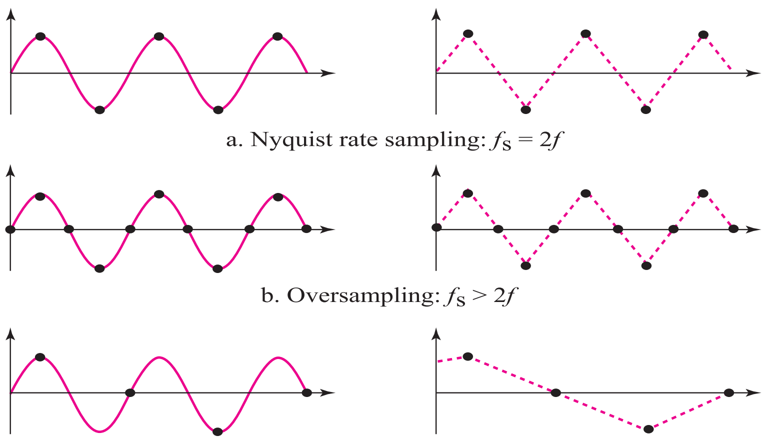
a. $\log_2 2 = 1$ **b.** $\log_2 4 = 2$ **c.** $\log_2 16 = 4$ **d.** $\log_2 1024 = 10$

P7-45. The bandwidth for each channel = (1 MHz) / 10 = 100 kHz. We then find the value of r for each channel:

$$B = (1 + d) \times (1/r) \times (N) \rightarrow r = N / B \rightarrow r = (1 \text{ Mbps}/100 \text{ kHz}) = 10$$

We can then calculate the number of levels: $L = 2^r = 2^{10} =$ **1024.** This means that we need a **1024-QAM** technique to achieve this data rate.

P7-47. The following figure shows the result. It can be seen that sampling at the Nyquist rate can create a good approximation of the original sine wave (part a). Oversampling in part b can also create the same approximation, but it is redundant and unnecessary. Sampling below the Nyquist rate (part c) does not produce a signal that looks like the original sine wave.



a. Nyquist rate sampling: $f_S = 2f$

b. Oversampling: $f_S > 2f$

c. Undersampling: $f_S < 2f$

P7-49. The bandwidth allocated to each voice channel is 20 kHz / 100 = 200 Hz. Each digitized voice channel has a data rate of 64 Kbps (8000 sample × 8 bit/sample). This means that our modulation technique uses

64,000/200 = 320 bits/Hz.

P7-51.

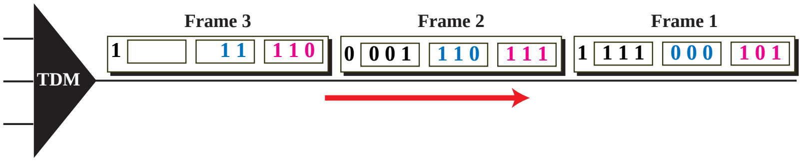
a. Frame size = 6 × (8 + 4) = **72 bits**.

b. We can assume that we have only 6 input lines. Each frame needs to carry one character from each of these lines. This means that the frame rate is **500 frames/s**.

c. Frame duration = 1 /(frame rate) = 1 /500 = **2 ms**.

d. Data rate = (500 frames/s) × (72 bits/frame) = **36 kbps**.

P7-53. See figure below:



P7-55. The Barker chip is 11 bits, which means that it increases the bit rate 11 times. A voice channel of 64 kbps needs 11 × 64 kbps = 704 kbps. This means that the bandpass channel can carry (10 Mbps) / (704 kbps) or approximately **14 channels.**

P7-57. $SNR_{dB} = 6.02 \times 8 + 1.76 =$ **49.92**.

P7-59. The loss of the cable for 10 Km is –10 dB = $10 \log_{10}(P_2/P_1)$. This means $\log_{10}(P_2/P_1) = -1$, or $(P_2/P_1) = 1/10$, resulting in $P_1 = 10 \times P_2 = 100$ mW.