COMP3721 – Introduction to Data Communications
Assignment Two – Fall 2008

General Instructions

- You may work with one partner for this assignment. Your partner may be from your set or another full-time CST set.
- You and your partner may discuss any and all details of each question freely. You may also discuss questions in broad terms with others, particularly in lab, but ultimately your answers should show sufficient individuation from others' answers reflecting your work in answering the questions.
- All work submitted is subject to the standards of conduct as specified in BCIT Policy 5002.

Submissions

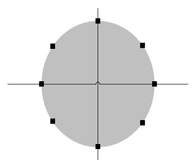
- This assignment is due Monday, October 27, 2008 by 1630 hrs at the latest. Late assignments will not be accepted.
- Submit your assignment to your lab instructor's assignment box in the SW2 connector.
- Your submissions must include a cover page clearly specifying your name, student number and set. If working with a partner, this information should be provided for each partner.

Marking

The assignment consists of 8 questions totaling 40 marks.

Questions

1. Does the following constellation diagram represent ASK, PSK, FSK or QAM modulation? How many levels does this system have? Given bandwidth *H*, what is the maximum data transfer rate possible using this modulation scheme? *[5 marks]*



The constellation diagram contains 8 points. Each point has the same distance from the origin, thus amplitude is not being varied. Each point is at a unique angle from the positive x-axis, thus there are 8 unique phase shifts. The system illustrated is 8-PSK.

Given eight possible ways to modulate the carrier for each baud, 8-PSK can encode $\log_2 8 = 3$ bits/baud. Assuming a maximum frequency of H, then the maximum baud rate is 2H (by Nyquist) and the maximum data transfer rate is:

Using Nyquist:

$2H * \log_2 V = 2H * \log_2 8 = 2H * 3 = 6H$ ***bits/second***

where H is specified in Hertz.

2. 24 voice signals are to be multiplexed and transmitted over a twisted pair.

a. What is the bandwidth required for FDM? *[2 marks]*

Assuming 4 kHz per voice signal, the required bandwidth for FDM is 24 × 4 = 96 kHz.

b. Assuming a bandwidth efficiency (ratio of data rate to the transmission bandwidth) of 1 bps/Hz, what is the bandwidth required for TDM using PCM? *[3 marks]*

With PCM, each voice signal requires a data rate of 64 kbps, for a total data rate of 24 × 64 = 1.536 Mbps. At 1 bps/Hz, this requires a bandwidth of 1.536 MHz.

3. You are browsing the Internet at home and click on a link to a server in Sydney, Australia. The HTTP request packet is 2 kB in size and the server response is 4 kB. Assume throughput is 64 kbps between the two sites and that the two sites are more or less directly connected by a fiber-optic cable circumnavigating the ocean floor between Vancouver and Australia (a distance of approximately 12500 km).
    a. Determine the time required before the entire response is received. *[4 marks]*

       Assuming no other delays (unlikely), we must account for the time required to place the data onto the channel (a function of data size and throughput) as well as the propagation delay (a function of the medium propagation rate and distance). Given that we are interested in the time from sending the request until receiving the entire response, we must calculate these values in both directions:

$$T_{req} = \frac{2kB * 8bits/byte}{64kbps} = 0.25s$$

$$T_{prop} = \frac{12500000m}{2*10^8 m/s} = 0.0625s$$

$$T_{resp} = \frac{4kB * 8bits/byte}{64kbps} = 0.50s$$

$$T_{total} = T_{req} + 2*T_{prop} + T_{resp} = 0.875s$$

    b. How does your answer change if the server is located in Toronto? *[1 mark]*

       The request and response times do not change, only the propagation delay (using a distance of 3366km to Toronto [*http://www.indo.com/cgi-bin/dist*]):

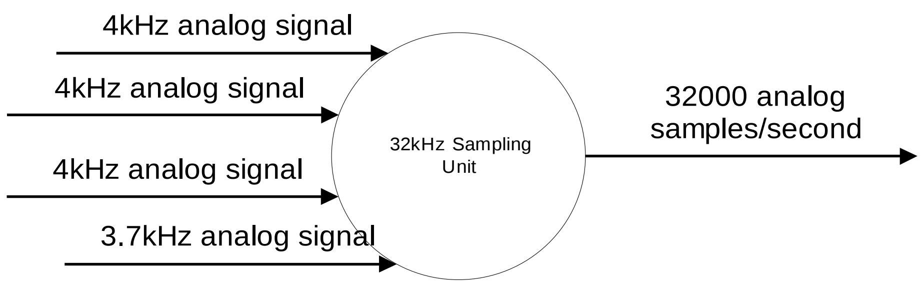
$$T_{prop} = \frac{3366000m}{2*10^8 m/s} = 0.01683s$$

$$T_{total} = T_{req} + 2*T_{prop} + T_{resp} = 0.784s$$

4. A system having four digital signals @ 19.2 kbps, three analog signals @ 4 kHz and one analog signal @ 3.7 kHz is to be TDM multiplexed. Analog samples are quantized using 14 bits/sample.

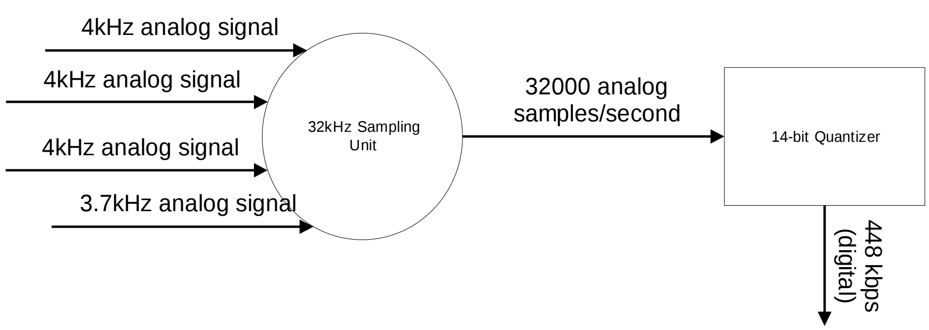
   c. Show a block diagram for the system specifying data format and bit rates at each point in the system. *[4 marks]*

      We start by digitizing all analog signals into a single digital bit-stream. This requires one sampling unit and one quantizing unit. As the sampling unit will have only a single sampling rate for each sampled channel. In this case we can use an 8kHz sampling rate for each analog signal (we are thus over-sampling the 3.7kHz signal slightly). With four signals, this then requires a total sampling rate of 4 * 8kHz = 32kHz or 32000 samples/second.

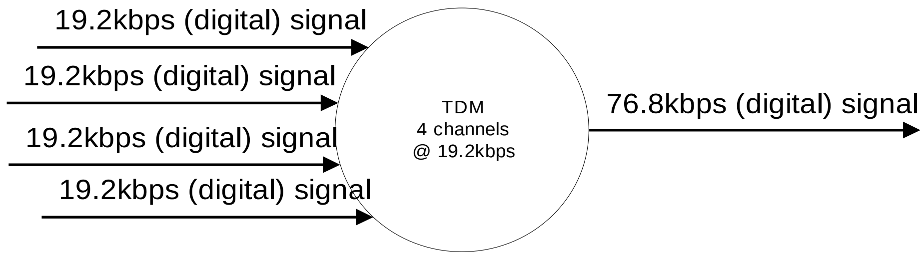


These analog samples must then be quantized. Using 14 bits/sample, we get a digital data rate of:

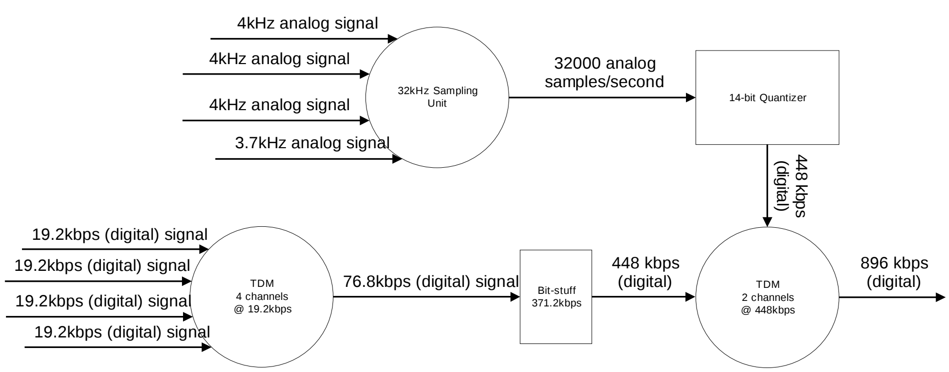
32 000 samples/sec * 14 bits/sample = 448kbps



The digital sources will also be combined into a single bit-stream through time-division multiplexing:



These two digital streams are them multiplexed into a single digital stream. However, to do so requires that both channels have the same data rate. So the 76.8kbps channel should be bit-stuffed up to 448kbps.



d. How much quantization error is introduced into the analog signals? *[1 mark]*

Signal strength in relation to quantization error is estimated by the equation:

$$S/N_Q = 6n - 1.25 \text{ dB}$$

where n is the number of bits/sample generated. With 14 bits/sample, we have:

$$S/N_Q = 6(14) - 1.25 \text{ dB} = 82.75\text{dB}$$

5. A way of visualizing the Nyquist theorem is in terms of periodic sampling of the second hand of a clock which makes one revolution around the clock every 60 seconds. The Nyquist sampling rate here should correspond to 2 samples per cycle, that is, sampling should be done at least every 30 seconds. (5 marks)

(a). Suppose we begin sampling when the second hand is at 12 o'clock and that we sample the clock every 15 seconds. Draw the sequence of observations that result. Does the second hand appear to move forward?

0 15 30 45 0 15 30 45 0 ...Yes, the second hand appears to move forward.

(b). Now suppose we sample every 30 seconds. Does the second hand appear to move forward or backward? What if we sample every 29 seconds?

0 30 0 30 0 30 0 30 0 30 ... No. It's hard to tell if it moves forward or backward because either direction will give the same sequence.

0 29 58 27 56 25 54 23 ... Yes, the second hand appears to move forward.

(c). Explain why a sinusoid should be sampled at a little more than twice its frequency.

When a sinusoid is being sampled at exactly twice its frequency, it is possible that all samples have value of zero. Therefore, the sampling rate should be a little more than twice its frequency.

(d). Now suppose that we sample every 45 seconds. What is the sequence of observations of the second clock hand?

One sampling every 45 seconds -> 0 45 30 15 0 45 30 15 0 … . The second hand appears to be moving backward.

(e). Motion pictures are made by taking a photograph 24 times a second. Use part (c) to explain why car wheels in movies often appear to spin backward while the cars are moving forward!

The spinning wheel is being sampled at 24 times per seconds. Depending on the rotation speed of the wheel, the wheel may appear either as spinning backward while actually spinning forward.

6. Consider the following parameters for communication over a channel: Character-synchronous transmission, 7-bit characters, 20 control characters per frame, 56 kbps. The data portion of the frame can be 50 or 400 characters. On the average 50 characters frames occur 40% of the time and 400 character frames occur 60% of the time. What is the effective data rate on the channel? *[5 marks]*

   The effective data rate measures the amount of actual data moving through the channel per time unit – it factors out the overhead required for control characters. Key to this is the ratio of actual data characters versus total characters (data and control) placed on the channel. For the two frames specified, we have:

$$R_{50} = \frac{50\,data\,characters}{50\,data\,characters + 20\,control\,characters}$$

$$R_{400} = \frac{400\,data\,characters}{400\,data\,characters + 20\,control\,characters}$$

   40and 60% of the time may be interpreted in one of two ways – if you interpret it to mean 40% of the time when you look on the network you observe a 50 data-character frame and 60% of the time you observe a 400 data-character frame, then the effective throughout is:

$$DTR_{effective} = (40\% * \frac{50}{70} + 60\% * \frac{400}{420}) * 56kbps = 48kbps$$

   Alternately, if you interpret it to mean that 40% of the time you place a 50 data-character frame on the channel and 60% of the time you place a 400 data-character frame on the channel, then the effective throughput is:

$$DTR_{effective} = \frac{40\% * 50 + 60\% * 400}{40\% * 70 + 60\% * 420} = 52kbps$$

7. Most digital transmission systems are “self-clocking” in that they derive the bit synchronization from the signal itself. To do this the systems use the transitions between positive and negative voltage levels. These transitions help define the boundaries of the bit intervals. For the following questions assume that we are transmitting a sequence of 4 consecutive 1s followed by 4 consecutive 0s.

1(a). The nonreturn-to-zero (NRZ) signaling method transmits a 0 with a +1 voltage of duration T, and a 1 with a −1 voltage of duration T. Plot the signal for the sequence n consecutive 1s followed by n consecutive 0s. Explain why this code has a synchronization problem. (2 marks)

Your diagram should show a sequence of 4 0s followed by 4 1s. A long sequence of 1s or a long sequence of 0s produces a long period during which there is no change in the signal level. Consequently, there are no transitions (“zero crossings”) that helps a synchronization circuit determine where the boundary of each signaling interval is located.

1(b). The Manchester signaling method transmits a 0 as a +1 voltage for T/2 seconds followed by a −1 for T/2 seconds; a 1 is transmitted as a −1 voltage for T/2 seconds followed by a +1 for T/2 seconds. Repeat part (a) and explain how the synchronization problem has been addressed. What is the cost in bandwidth in going from NRZ to Manchester coding? (3 marks)

With Manchester coding every T-second interval now has a transition in the middle, so synchronization is much simpler. However, the BW of the signal is doubled since pulses are essentially half as wide, that is, T/2 seconds.

8. A new broadcast service is to transmit digital music using the FM radio band. Stereo audio signals are to be transmitted using a digital modem over the FM band. The specifications for the system are the following: Each audio signal is sampled at a rate of 40 kilosamples/second and quantized using 16 bits; the FM band provides a transmission bandwidth of 200 kiloHertz.

   a. What is the total bit rate produced by each stereo audio signal? (2 marks)

      The bit rate for each signal is 40 ksamples/sec x 16 bits/sample = 640 kbps. The bit rate for the pair of signals is then 1.28 Mbps

1b. How many points are required in the signal constellation of the digital modem to accommodate the stereo audio signal? (3 marks)

A transmission bandwidth of 200 kHz allows 200 kilopulses/second. To obtain a bit rate of 1.28 Mbps, we need to send 1280/200 = 6.4 bits/pulse. If we use a 2^7 = 128 point constellation, we can then meet the desired bit rate.