ITS323 – Assignment 1 Answers

Question 1 Answers

Consider the Satellite path first.

A message must be transmitted from Bangkok to Satellite, and then from Satellite to London. Therefore there are two links (with transmission and propagation delays) as well as processing delay within the satellite.

$$d_{sat} = transmission_{up} + propagation_{up} + proccessing + transmission_{down} + propagation_{down}$$

The transmission and propagation delays on the uplink and downlink will be identical (since the message size, data rate, distance and speed are the same for both links. Lets assume the message size is 1500 Bytes.

$$transmission = \frac{(MsgSize + Header) \times 8\,bits}{20\text{Mb}/s}$$
$$= \frac{1600 \times 8}{20\text{Mb}/s}$$
$$= 0.64\text{ms}$$

$$propagation = \frac{36000\text{km}}{3 \times 10^8\,m/s}$$
$$= 120\text{ms}$$

Assuming a processing delay of 20ms, the total satellite delay is: 261.28ms.

The throughput for the satellite path is:

$$thr_{sat} = \frac{1500}{1600} 20\text{Mb}/s$$
$$= 18.75\text{Mb}/s$$

Now consider the Fibre path.

A message is sent across a network with dedicated rate of 10Mb/s. The total propagation delay is 50ms and processing/queuing delay 0.01ms.

$$d_{fib} = \frac{(1500 + 80) \times 8\,bits}{10\text{Mb}/s} + 50\text{ms} + 0.01\text{ms}$$
$$= 51.274\text{ms}$$

The throughput for the fibre path is:

$$thr_{sat} = \frac{1500}{1580} 10\text{Mb}/s$$
$$= 9.49\text{Mb}/s$$

To compare the two paths we consider a the weighted sum to determine the cost (assume the weights for throughput and delay are 53 and 60 respectively):

$$F_{sat} = \frac{weight_t}{thr_{sat}} + \frac{d_{sat}}{weight_d}$$
$$= \frac{53}{18.75} + \frac{261.28}{60}$$
$$= 7.18$$

Using the same approach we can find the cost of the fibre path is 6.62. Therefore we choose the path with the lowest cost: in this case, Optical Fibre.

Question 2 Answers

As an example, I will use the group members: Steven Gordon, Bill Gates and Linus Torvalds. I will use NRZ-Level as the codec, with default data rate of 10Mb/s.

A detailed explanation of the steps (especially the Scilab) was provided to you as hints at:
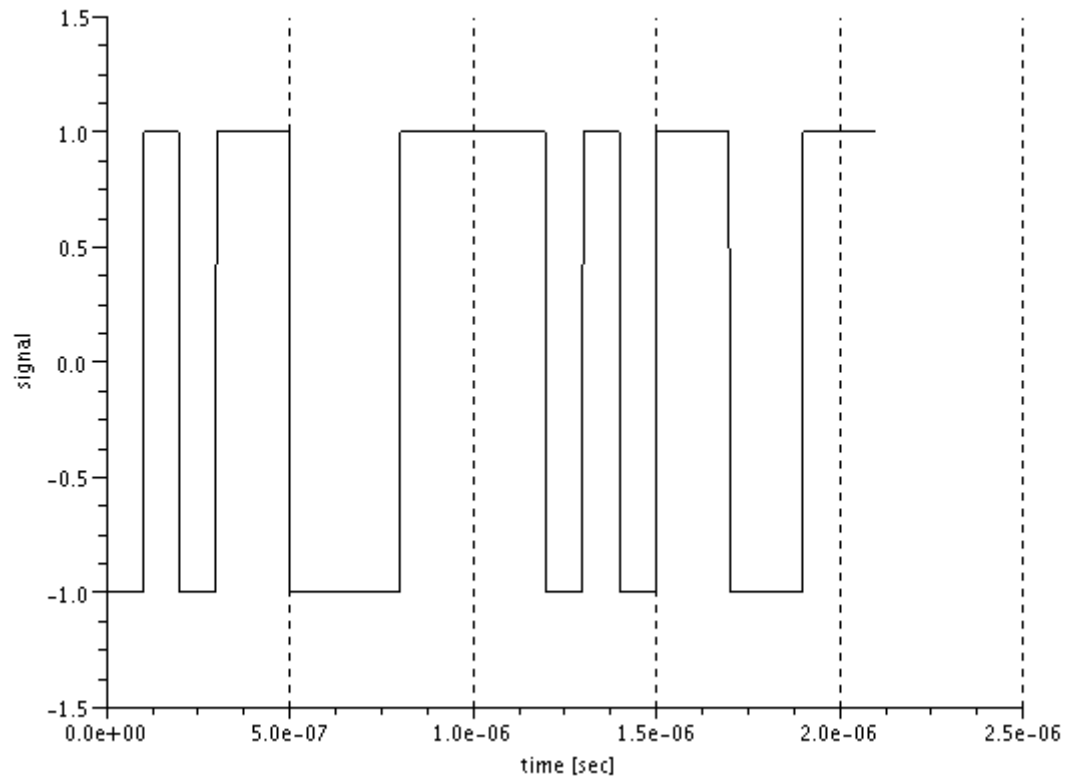
http://ict.siit.tu.ac.th/~steven/its323/assignment1-hints.html

Part (a)

The Scilab code for producing the signal is:

```
-->M1 = 'SBL';
-->M1_bin = text2bin(M1)
 M1_bin  =
        column  1 to 12
    1.    0.    1.    0.    0.    1.    1.    1.    0.    0.    0.    0.
        column 13 to 21
    1.    0.    1.    0.    0.    1.    1.    0.    0.
-->num_bits = size(M1_bin,2)
 num_bits  =
    21.
-->[out_signal,signal_rate] = ds_nrzl_encode(M1_bin,10000000);
-->as_plot(out_signal);
```
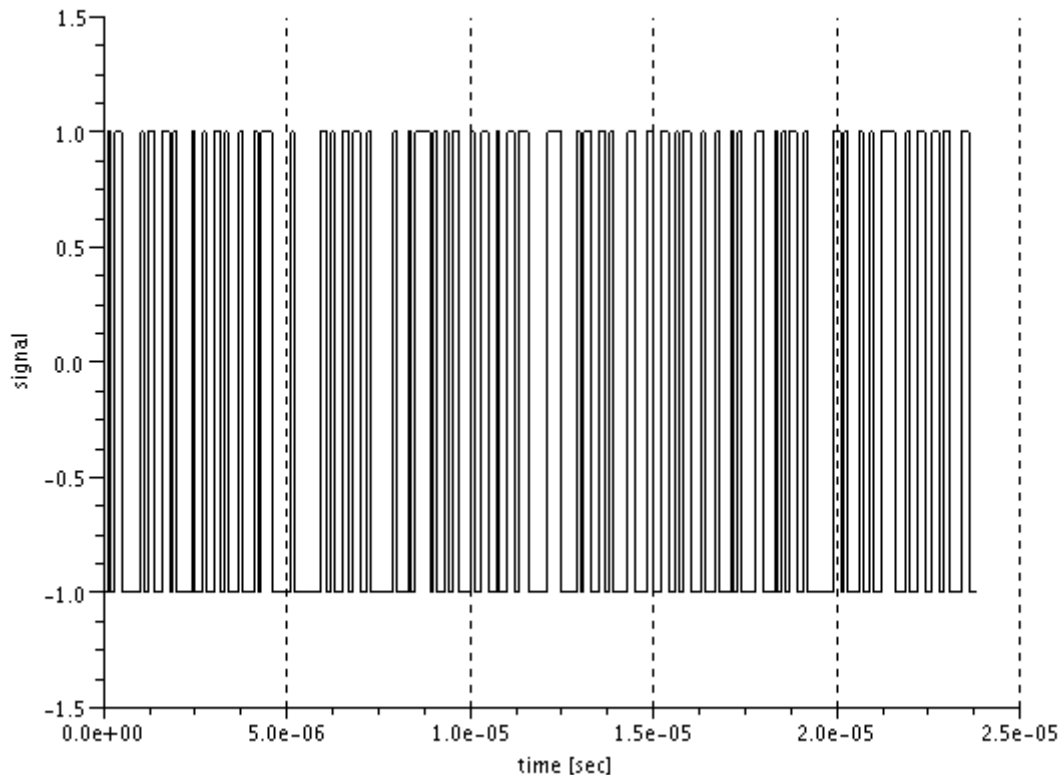
The output signal is:

Part (b)

The transmitted signal:

First considering uniform noise. I measured the Bit Error Rate for different values of Data Rate and Noise Amplitude. I repeated the measurements multiple times. The reason for this is that the exact amplitude (and location of impulse) noise is random (within some range). Therefore different results may be obtained. Taking multiple measurements and determining the average gives a more accurate representation of performance.

Figure 1 shows the BER as the noise amplitude is increased for uniform noise. The data rate is maintained at 10Mb/s. Note that the noise amplitude A means random noise is generated between the values of +A and -A.
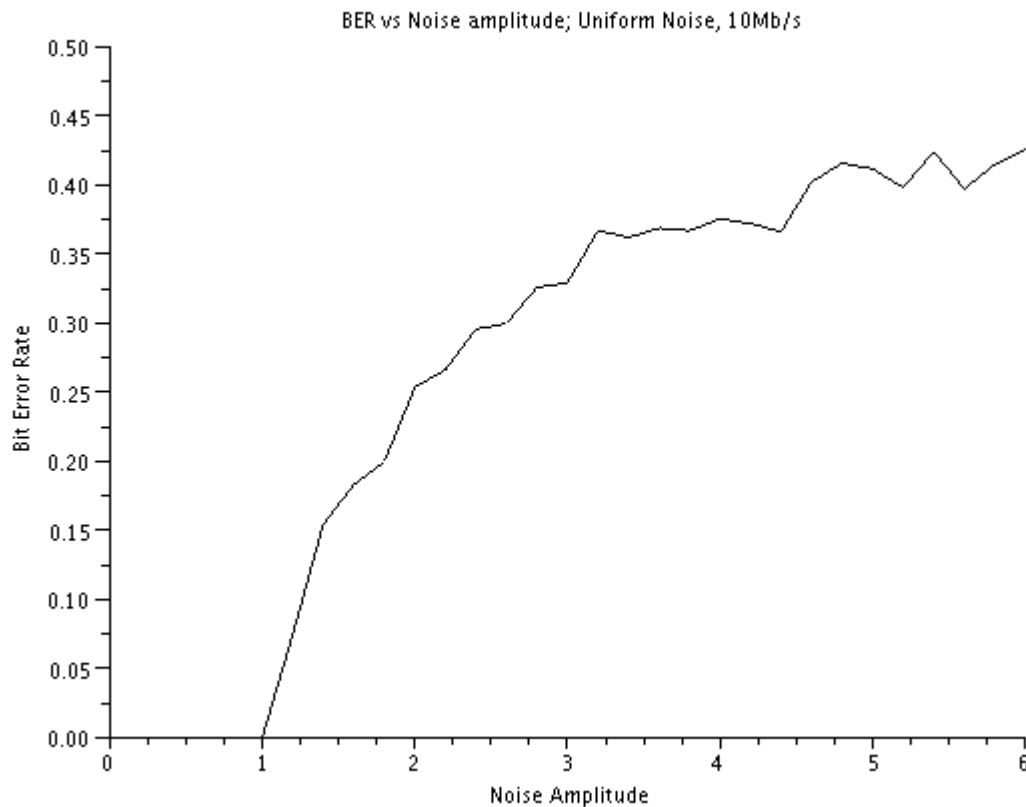
Figure 1

To explain the results seen in Figure 1 lets consider how errors occur. In the decoding function, for every signal element received, if the value measured at the middle of the element is positive, then the signal is assumed high (and if negative, the signal is assumed low). The mapping of high/low signals to bits depends on the encoding scheme (NRZ-L or NRZ-I).

The transmitted signal element either has a value of +1 or -1. After adding the noise, the received signal element will be different. For example, if the noise amplitude is 0.5, then we have the following cases:

- A transmited +1 can have the receive value between +0.5 and +1.5. No matter which value, it will always be positive, which is the same as the transmitted signal element, and hence no error occurs.

- A transmitted -1 can have the receive value between -0.5 and -1.5. As with the above case, the received signal is always negative, and hence no errors.

In summary, with a noise amplitude of 0.5, the sign (positive or negative) of the received value will always be the same as the transmitted value. Hence no bit errors. This is the same when the Noise Amplitude ranges from 0 to 1.

Now consider Noise amplitude greater than 1, for example 2:

- A transmitted +1 can have the receive value of between +1+2 = 3 and +1-2=-1. As the noise level is uniform between -2 and +2, there is 75% chance the received value is positive (0 to

3) and 25% chance it is negative (0 to -1). Therefore, in this case there is 25% chance of a bit error.

The same logic can be applied for a transmitted -1, leading to a 25% chance of error.

In summary, with a noise amplitude of 2, there is a 25% chance that the sign of the received value will be different from the transmitted value. Hence a BER of 25% (see the graph – it is measured at about 0.25 when noise amplitude is 2).

If the Noise Amplitude was 4, a transmitted +1 can be received as between -3 and +5, leading to 37.5% chance of error (3 out of 8). The measured results closely match this theoretical value.

Figure 2 shows the BER as the data rate increases for uniform noise (with amplitude 2). As the uniform noise affects all bits in the signal, changing the data (and hence signalling) rate should not change the number of bits affected by noise. Hence we see the BER is approximately constant as the data rate changes.
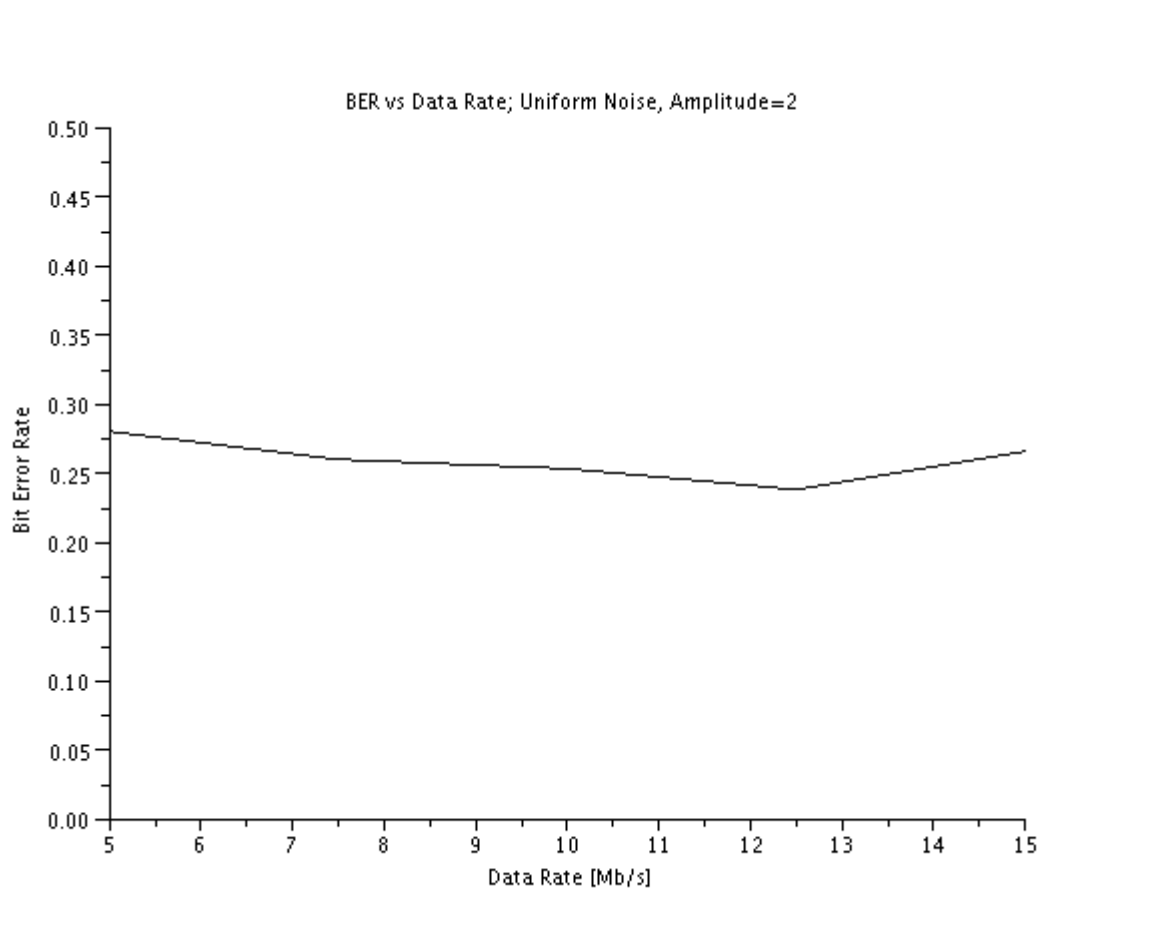

Figure 2

Now consider the impact of Impulse Noise, where the impulse has an amplitude of 2 and is of a duration shorter than the total signal duration. The impulse may be +2 or -2 (the sign of the impulse is chosen randomly). Figure 3 shows the BER as the impulse duration increases (for a fixed data rate of 10Mb/s). Note in the following cases there are 238 bits in total.
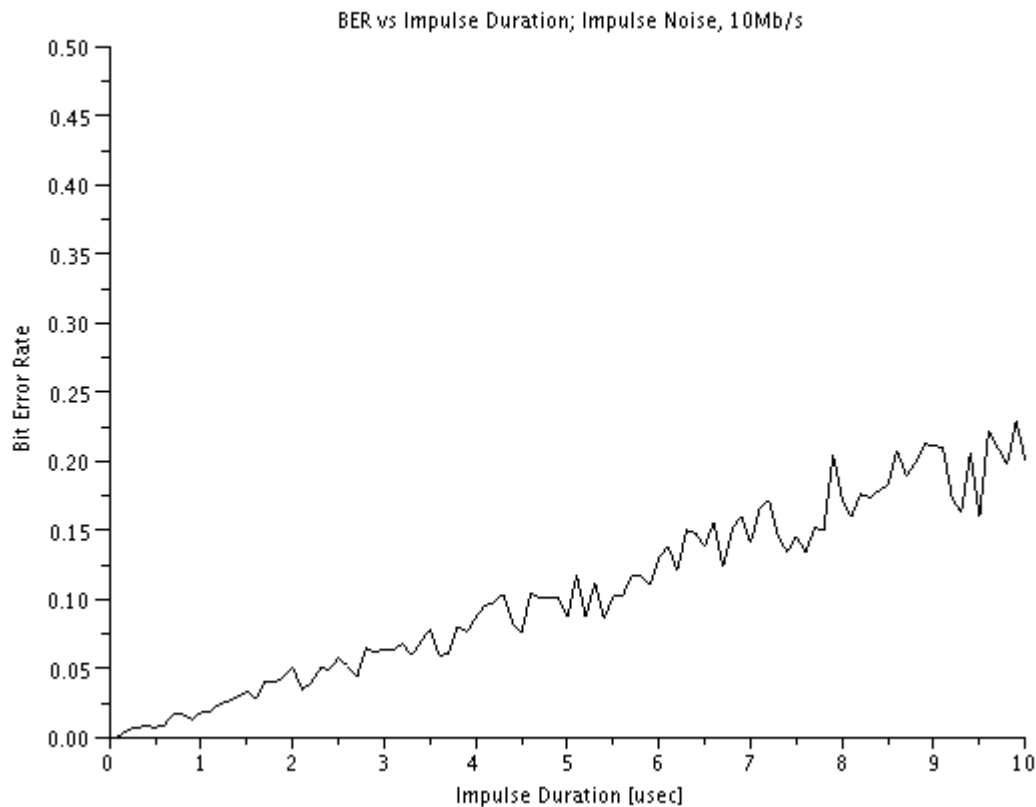
Figure 3

With a data rate of 10Mb/s, each signal element is 0.1us in length. Therefore an impulse duration of 1us can impact a maximum of 10 bits. Consider what the impulse may do to these ten bits:

- Transmitted value of +1: impulse of +2 will change to received value of +3. This is no change in the sign of the value, and hence no bit error.

- Transmitted value of -1: impulse of +2 will change to received value of +1. This is a bit error.

Of the 10 bits affected by the impulse, on average half of the bits will be in error (5 bits). 5 bit errors out of 238 bits is a BER of 0.02 (approximately the same as the measured value).

A linear increase in the impulse duration results in a linear increase in the bits affected, and hence linear increase in BER. If the duration is 10us, a maximum of 100 bits are covered by the impulse, leading to an average of 50 bit errors. 50/238 = 0.21, which is close to the measured value.

Fnially, Figure 4 shows the BER as the data rate increases with a fixed impulse duration of 4us. The same logic as increasing the impulse duration (with fixed data rate) applies. The higher the data rates, the more bits an impulse of 4us can impact, leading to more bit errors.
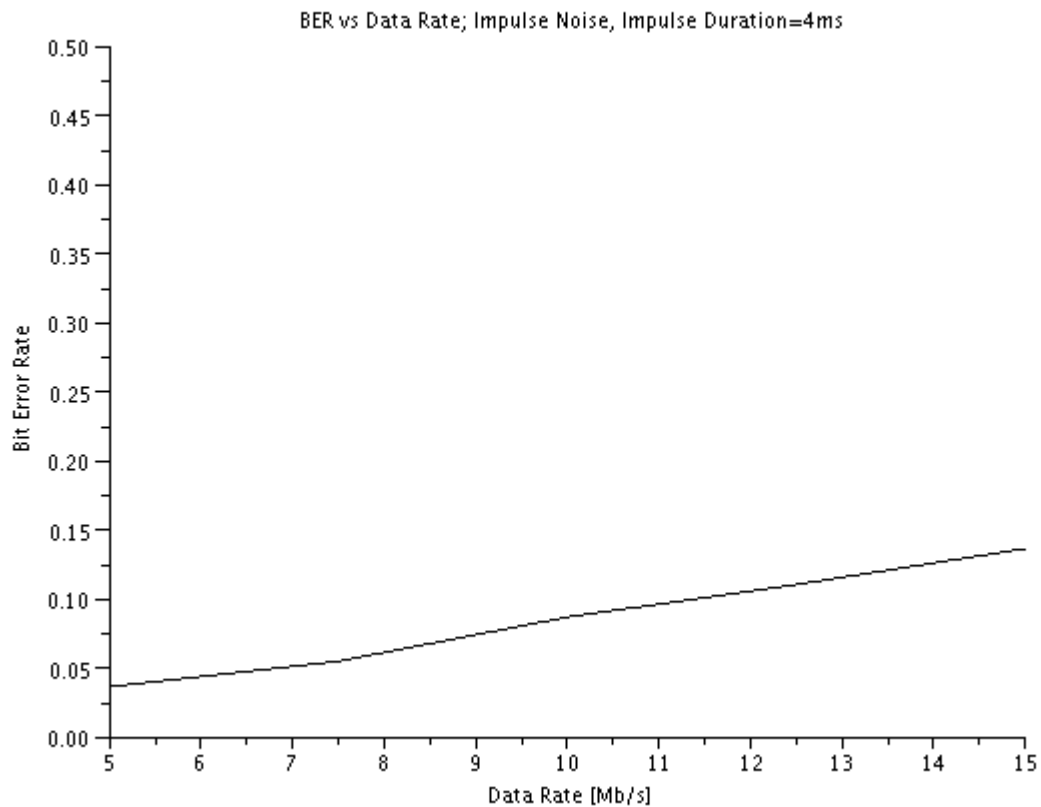
Figure 4

Question 3 Answers

This question required you to generate analog data, convert it to digital data using PCM encoder, then convert back to analog data using PCM decoder. By using different sampling rates and different number of code levels you can analyse the quality and required bit rate for the received analog data.

Consider the following analog data at the source.
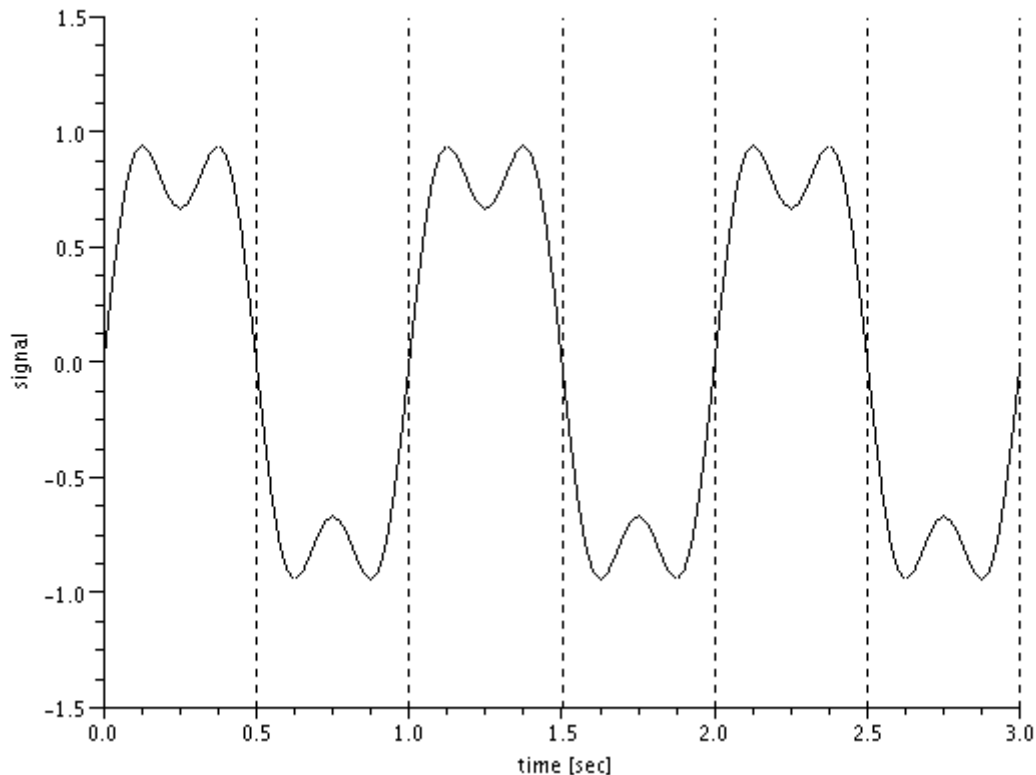
After encoding with PCM the digital data is sent to the destination. The destination then decodes and outputs the analog data. The quality of the received analog data (from the users point of view) can be compared by seeing how close the output analog data is to the transmitted analog data. Figures 6 to 9 show the output analog data as the sampling rate increases.

You can see how the quality (closeness to the original data) increases as the sampling rate increases. Similar results can be observed as the number of code levels increases. In summary we can say:

- Increasing the sampling rate and code levels, increases the output data quality (GOOD)

The disadvantage is the data required to send the data. For example, with a sample rate of 2/s and 32 levels, each level is represented by 5 bits, therefore 10b/s must be sent. Note that if a lower data rate was sent (say 6b/s), then the receiver would not be able to reconstruct the analog data within the same time as the original analog data was produced. If the original analog data was voice, then if the data is not sent at 10b/s, then the received voice would not contain everything said by the source user. Therefore, we must transmit at 10b/s.

Increasing the sample rate to 4/s means 20b/s must be sent.

So if we had a link data rate of 100b/s between source and receiver, with sampling rate of 2/s (and 32 code levels) we use 10b/s (or 10% of the link) for sending the analog data. But with a sample rate of 4/s (and 32 code levels) use 10b/s (or 20% of the link). We want to use as little of the link as possible to send the analog data (so we can use the rest of the link for sending other data). In summary:

- Increasing the sampling rate and code levels, increases the data rate required to send the analog data (BAD)
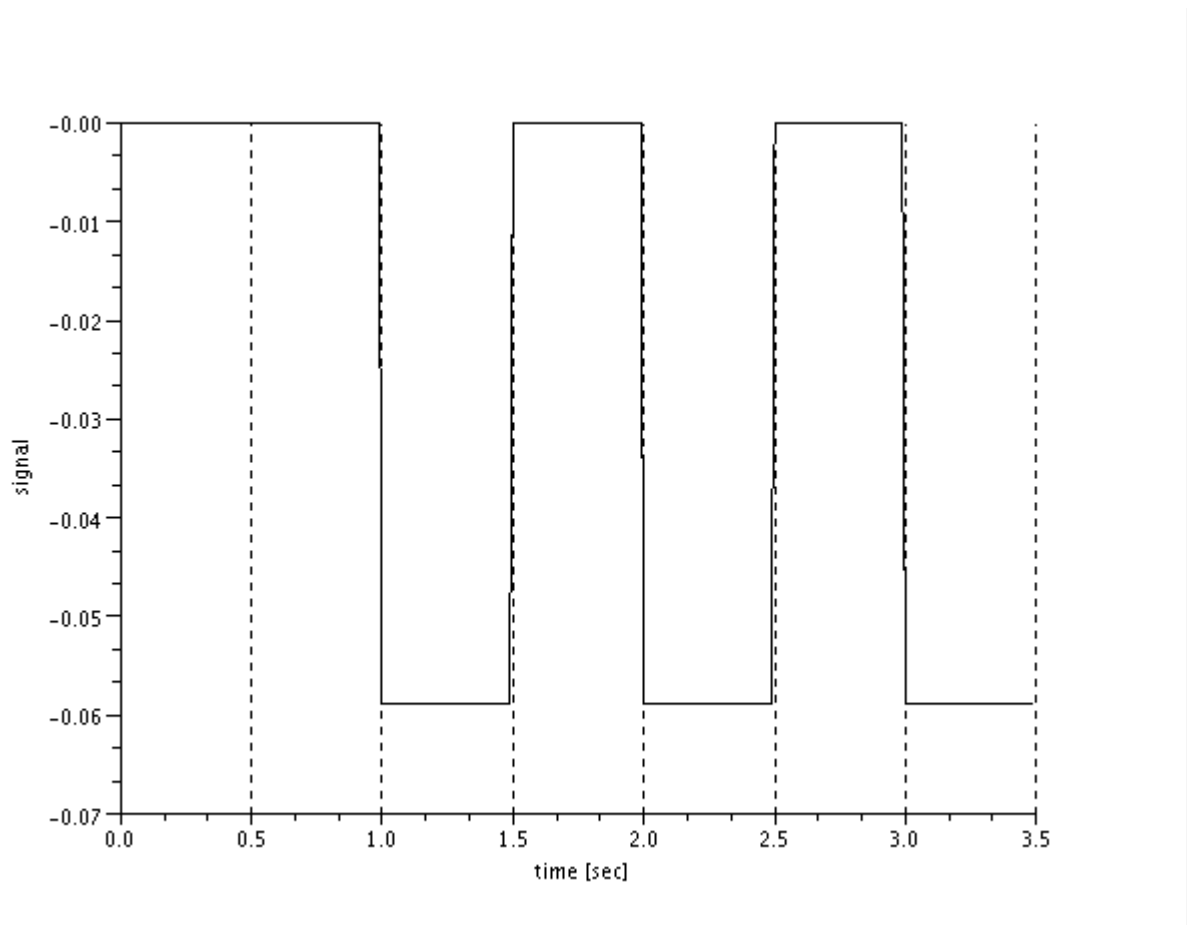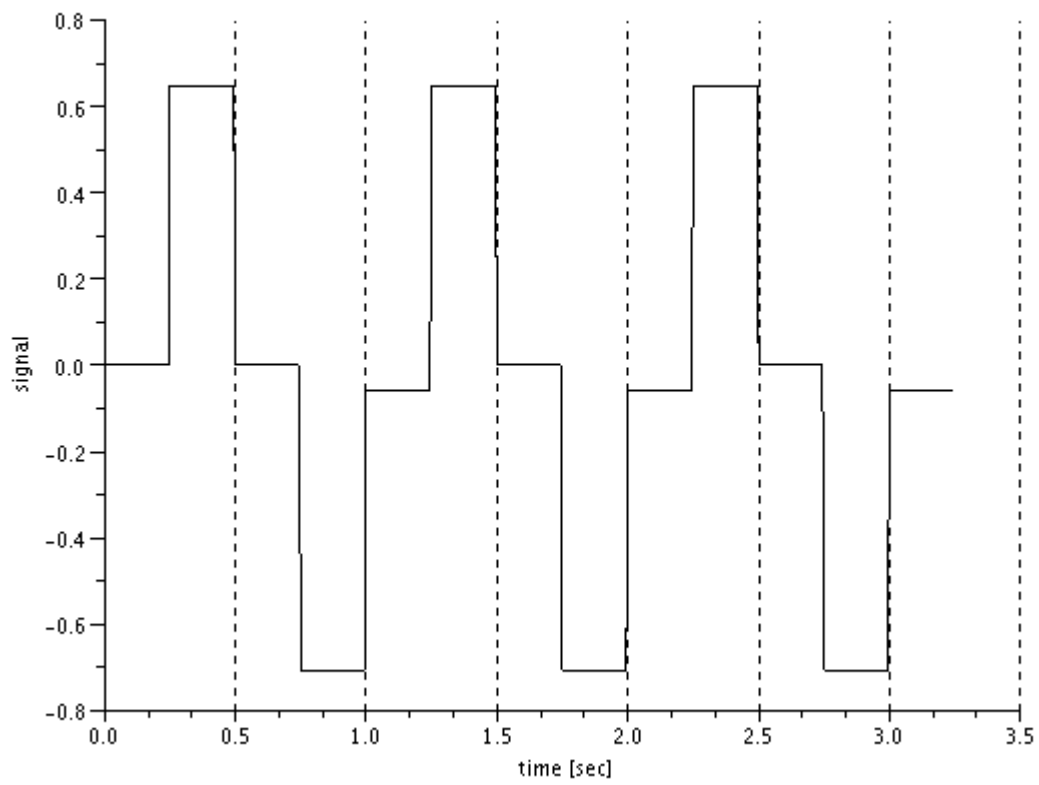
Figure 5: Rate = 2/s, Code levels = 32
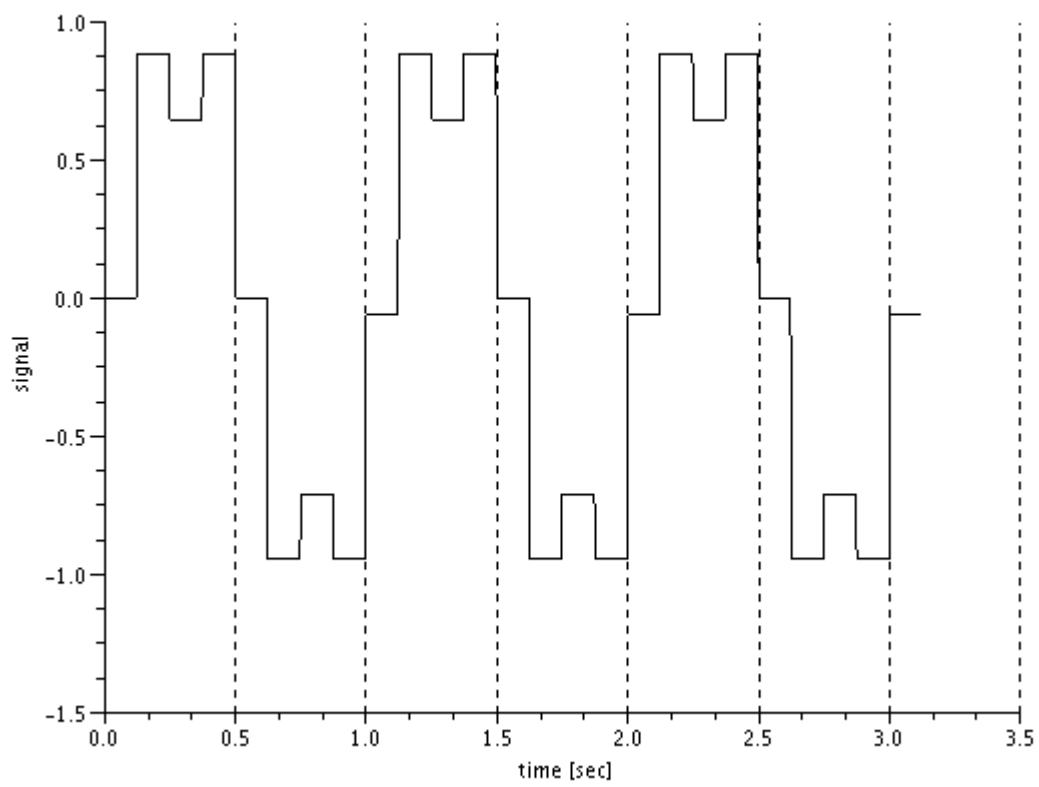
Figure 6: Rate = 4/s, Code levels =32

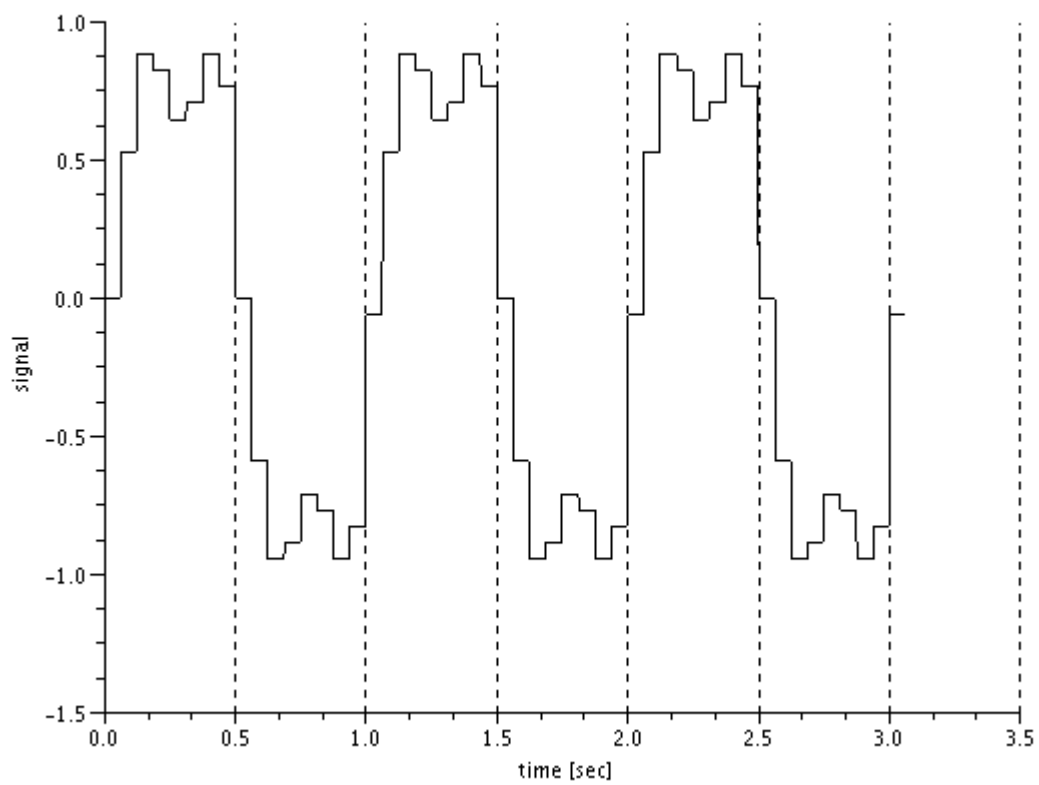*Figure 7: Rate = 8/s, Code levels = 32*

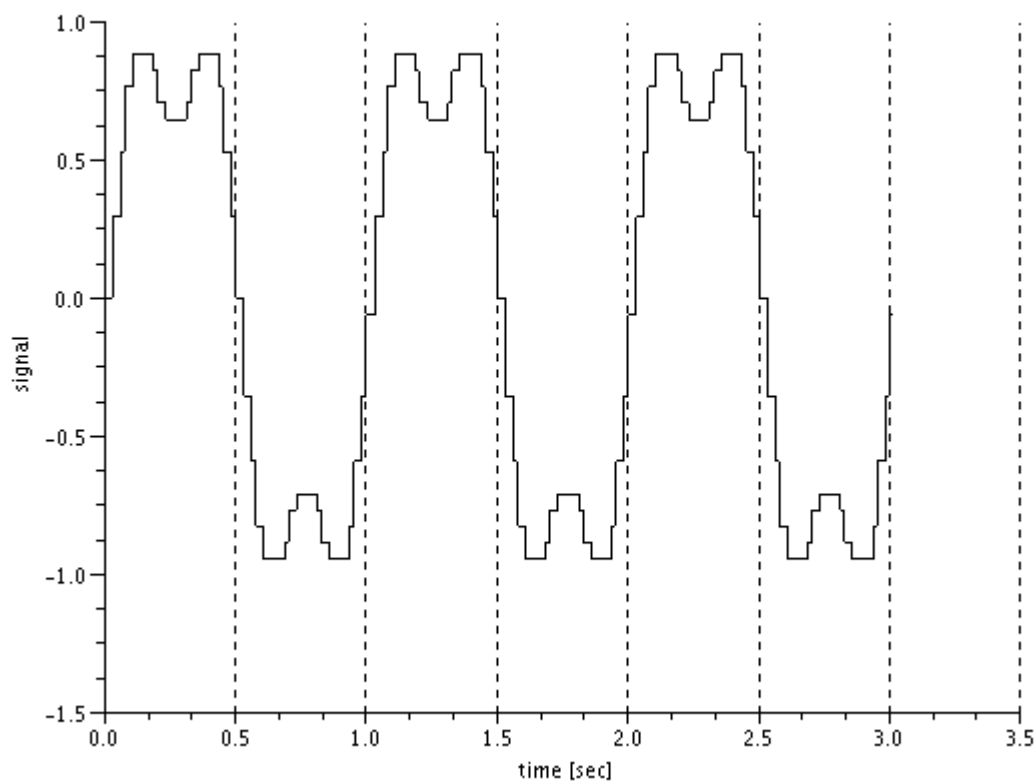*Figure 8: Rate = 16/s, Code levels = 32*

Figure 9: Rate = 64/s, Code levels = 32

Question 4 Answers

As covered in the lectures and quizzes, you must recognize the relationship between transmit power and receive power. Mathematically we have described it as:

$$P_r = \frac{P_t \times G_t \times G_r}{L}$$ when using absolute values.

Or alternatively we can consider in dB. Remember:

$$dB = 10 \log\left(\frac{P1}{P2}\right)$$

$$dBW = 10 \log\left(\frac{P}{1W}\right)$$

$$dBi = 10 \log(G)$$ (since G is relative to an isotropic antenna)

So take the logarithm of both sides of the above received power equation and multiply by 10:

$$10 \log(P_r) = 10 \log\left(\frac{P_t \times G_t \times G_r}{L}\right)$$

Using properties of logarithms this can be rearranged as:

$$10\log(P_r)=10\log(P_t)+10\log(G_t)+10\log(G_r)-10\log(L)$$

And using the definitions of dB we get:

$$P_{r_{dBW}}=P_{t_{dBW}}+G_{t_{dBi}}+G_{r_{dBi}}-L_{dB}$$

From the question, for each wireless device you know the available transmit power, antenna gains and required received power (i.e. the threshold of the receiver). You want to find the maximum distance possible give the office environment. The path loss is given by:

$$L=20\log(f)+N\log(d)+P_f(n)-28$$

where you know f (e.g. 900 if using 900MHz device), N from the tables provided and $P_f(n)$ depends on the number of floors (using the tables provided).

$$P_{r_{dBW}}=P_{t_{dBW}}+G_{t_{dBi}}+G_{r_{dBi}}-\left(20\log(f)+N\log(d)+P_f(n)-28\right)$$

From the above equation you know all variables except d, and a simple re-arrangement will allow you to find d.

Example:

Device 1: f = 900, G = 3dBi, P_t = 30mW, P_r = -80dBm, n = 2, N = 33.

We need to convert all values to the dB (and specifically dBW for P_t and P_r – although you could use dBm for both and get the same answer).

$$\begin{aligned}P_t&=30\text{mW}\\&=10\log\left(\frac{30\text{mW}}{1\text{W}}\right)dBW\\&=10\log(0.03)\,dBW\\&=-15\,dBW\end{aligned}$$

$$\begin{aligned}P_r&=-80\,dBm\\&=10^{\frac{-80}{10}}\,mW\\&=10^{-8}\,mW\\&=10^{-11}\,W\\&=10\log\left(10^{-11}\right)dBW\\&=-110\,dBW\end{aligned}$$

Noting that 20 log (900) is approx. 59, we get:

$$-110=-15+3+3-\left(59+33\log(d)+9-28\right)$$

Solving for d gives 70 metres.