ITS 323 – Assignment 1 Answers

24 July 2008

10% of the final course mark

Question 0 (Preliminaries)

You must correctly complete this step, as answers in the remainder of the assignment depend upon it. You do not receive any marks for this question, *but you may be penalised if you make mistakes.*

Your ID is a 10 digit decimal number. Perform the following calculations, and use the answers in the remaining questions.

a) (ID mod 10) + 1

b) (ID mod 1000)

c) (ID mod 100)

d) Decimal2Binary(B mod 4096), where Decimal2Binary converts the decimal number to a 12 bit binary number, and B is the answer of part (b).

Answers

In this set of answers I will assume my ID is 4912345678. Each student has answers dependant on their ID – there is an accompanying Excel spreadsheet that includes the answers for each student.

A	=	(4912345678 mod 10) + 1
	=	**9**
B	=	4912345678 mod 1000
	=	**678**
C	=	4912345678 mod 100
	=	**78**
D	=	Decimal2Binary(B mod 4096)
	=	Decimal2Binary(678)
	=	**001010100110**

For part (d), remember it must be 12 bit binary number (pad the left with 0's). Also note that (B mod 4096) is the same as B, since B = ID mod 1000. This was a mistake in the question (I originally had (ID mod 4094), but changed "ID" to "B" and forgot to remove the mod 4096). However, the mistake makes no difference in your calculations.

Question 1 [5 marks]

Scenario:

You graduated from SIIT five years ago. After graduation you started an Internet Service Provider (ISP) company with your friends, and now it has become the number one ISP in Asia. From your hard work (and money earned) you have just purchased a new house on a remote island in the south of Thailand. To stay involved in the company activities while relaxing in your new home, you are planning a dedicated network connection between your home and the main office in Bangkok. Because of the remoteness of the island, satellite access is the option you are considering.

The network topology for connectivity between your laptop at home and server at the company office is shown in Figure 1. All links are full duplex (e.g. 10Mb/s from laptop and 10Mb/s to laptop at the same time).

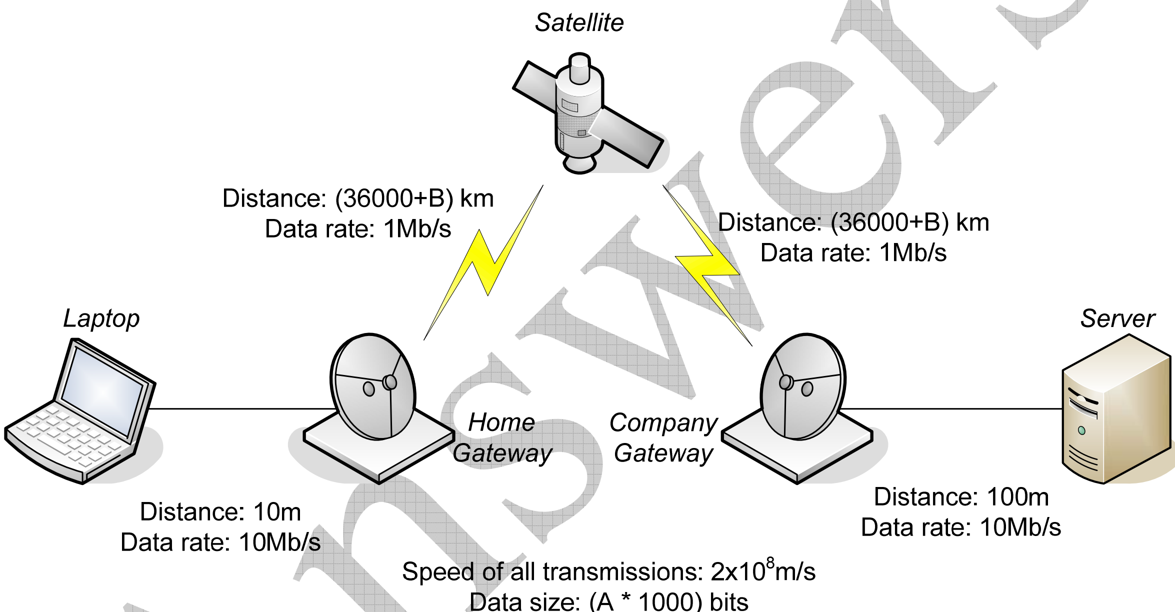


Figure 1

a) Assuming all processing and queuing delays are 0, calculate the one-way delay to send a packet of size (A x 1000) bits from your laptop to the server. (round to nearest millisecond) [5 marks]

Answer

You need to determine the transmission and propagation delays for each link, and then sum them to get the total one way (laptop to server) delay. For my ID, the data size is 9000 bits.

Link – Laptop to Home Gateway

Transmission time	=	9000 bits / 10Mb/s
	=	0.9 ms
Propagation time	=	10m / 2×10^8 m/s
	=	5×10^{-5} ms

Link – Home Gateway to Satellite

Transmission time	=	9000 bits / 1Mb/s
	=	9ms
Propagation time	=	36678 km / 2×10^8 m/s
	=	183.39 ms

Link – Satellite to Company Gateway

Same distance and rate as Home Gateway to Satellite, therefore same Transmission and Propagation times.

Link – Company Gateway to Server

Transmission time	=	9000 bits / 10Mb/s
	=	0.9ms
Propagation time	=	100m / 2×10^8 m/s
	=	5×10^{-4} ms

One way end to end delay	=	$0.9 + 5 \times 10^{-5} + 9 + 183.39 + 9 + 183.39 + 0.9 + 5 \times 10^{-4}$ ms
	=	386.58055 ms
	=	**387 ms** (rounded to nearest millisecond)

Question 2 [10 marks]

a) Now consider only the satellite link from Home Gateway to Satellite. Assume the link has a signal-to-noise ratio of C dB, where C is calculated in Question 0. In theory, what is the required bandwidth of the link? (round to the nearest Hz) [5 marks]

b) If the bandwidth calculated in part (a) is used for the Home Gateway to Satellite link, in theory what is the minimum number of signal levels needed to achieve the data rate? (must be a power of 2) [5 marks]

Answers

We know the following about Home Gateway to Satellite link:

Distance: 36678km

Data Rate: 1Mb/s

Signal to Noise Ratio: 78dB

We want to determine the bandwidth (in Hertz) required to support the 1Mb/s data rate. The signal-to-noise ratio suggests the use of Shannon's capacity theorem to be appropriate. Shannon's theorem gives an upper limit data rate (that is, capacity) that can be achieved on a link with some bandwidth and in the presence of noise.

Capacity = Bandwidth x $\log_2 (1 + SNR)$

Note that SNR in Shannon's theorem is an *absolute* value, not the dB value.

SNRdB = 10 $\log_{10}(SNR)$

Therefore SNR = $10^{78/10}$

= 63095734

Rearranging Shannon's theorem, and substituting the values for Capacity (1Mb/s) and SNR (63095734) gives:

Bandwidth = **38594 Hz** (rounded to nearest Hz)

Another way relationship between data rate (capacity) and bandwidth is given by Nyquist's theorem. This assumes no noise, and takes into account the number of signal levels (*M*) used to transmit the data. We can therefore use Nyquist's theorem to determine the *minimum* number of signal levels needed (of course in practice, a different number may be needed to cope with noise).

Capacity = 2 x Bandwidth x $\log_2 (M)$

Therefore, M = 7943.2 signal levels

However, we normally use 1 signal level to represent a set of bits, and therefore to be able to represent any combination of bits, we need 2^x signals levels (where each level represents *x* bits). If the minimum number is 7943, then the next highest power of 2 is **8192** signal levels.

Question 3 [40 marks]

Suppose an ARQ error control protocol is to be used over the Home Gateway to Satellite link. Assume each Data frame contains a 25 byte header, and each ACK frame is 25 bytes in length.

a) Calculate the maximum throughput that can be achieved for the following cases (round to the nearest bit per second):

   i. Stop and Wait ARQ; no errors. [10 marks]

   ii. Go-Back-N ARQ; Window field is A bits, where A is calculated in Question 0; no errors. [10 marks]

   iii. Selective Reject ARQ; Window field is A bits, where A is calculated in Question 0; no errors. [10 marks]

b) Compare the throughput (or efficiency) achieved for each of the above three ARQ protocols. For your particular case, explain the main reasons the throughput is as calculated (in other words, what are the main factors that result in the calculated throughput). Also, state which protocol is most appropriate for the link (and explain why you chose that protocol). [5 marks]

c) Consider Stop and Wait ARQ: if the processing time at the Satellite ranged from 2ms to 5ms (depending on the activities of the CPU on-board the satellite), then what is the minimum timeout value that the sender (Gateway) should use? (round to the nearest millisecond) [5 marks]

Answers

Part (a).

First, lets calculate some required information:

Propagation delay	=	183.39ms
DATA frame size	=	9000 bits + 25 bytes
	=	9200 bits
ACK frame size	=	200 bits
DATA transmission	=	9.2ms
ACK transmission	=	0.2ms

We want to calculate the throughput for the ARQ schemes. That is, the rate at which the original data can be received (or sent).

Now lets look at Stop and Wait ARQ (see the lecture notes for diagrams illustrating Stop and Wait).

The total time to transmit one frame is:

Total time	=	9.2 + 183.39 + 0.2 + 183.39 ms

With Stop and Wait ARQ, the next frame can only be transmitted once an ACK is received for the previous. This limits the rate at which original data can be sent.

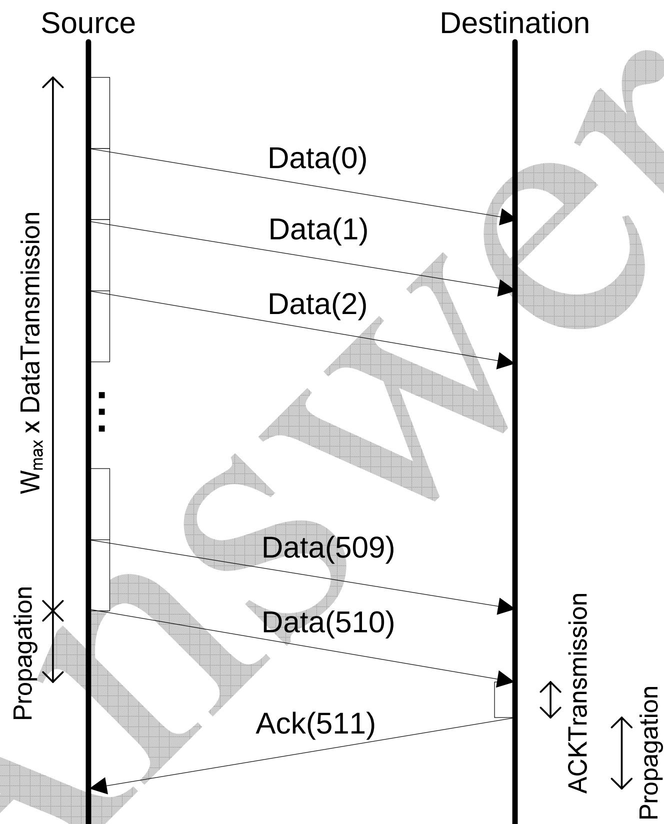
Throughput	=	9000 bits / 376.18ms
	=	**23925** bits per second

Now lets consider Go-Back-N ARQ. With Go-Back-N the source is allowed to transmit W frames at a time, W is the window size. To achieve maximum throughput, the source should send the maximum number of frames at a time, W_{max}. This depends on the number of sequence numbers available, which in turn is defined by the number of bits (k) used for the sequence number field in the header (referred to as Window field in original question). For our case, k = A = 9. We know that for Go-Back-N:

$$W_{max} = 2^k - 1$$

$$= 511$$

Another factor that will affect the throughput is how the destination responds with ACKs. In the lectures we saw an example of the destination sending a *single* ACK after receiving a full window of frames.



In that case, the throughput will be:

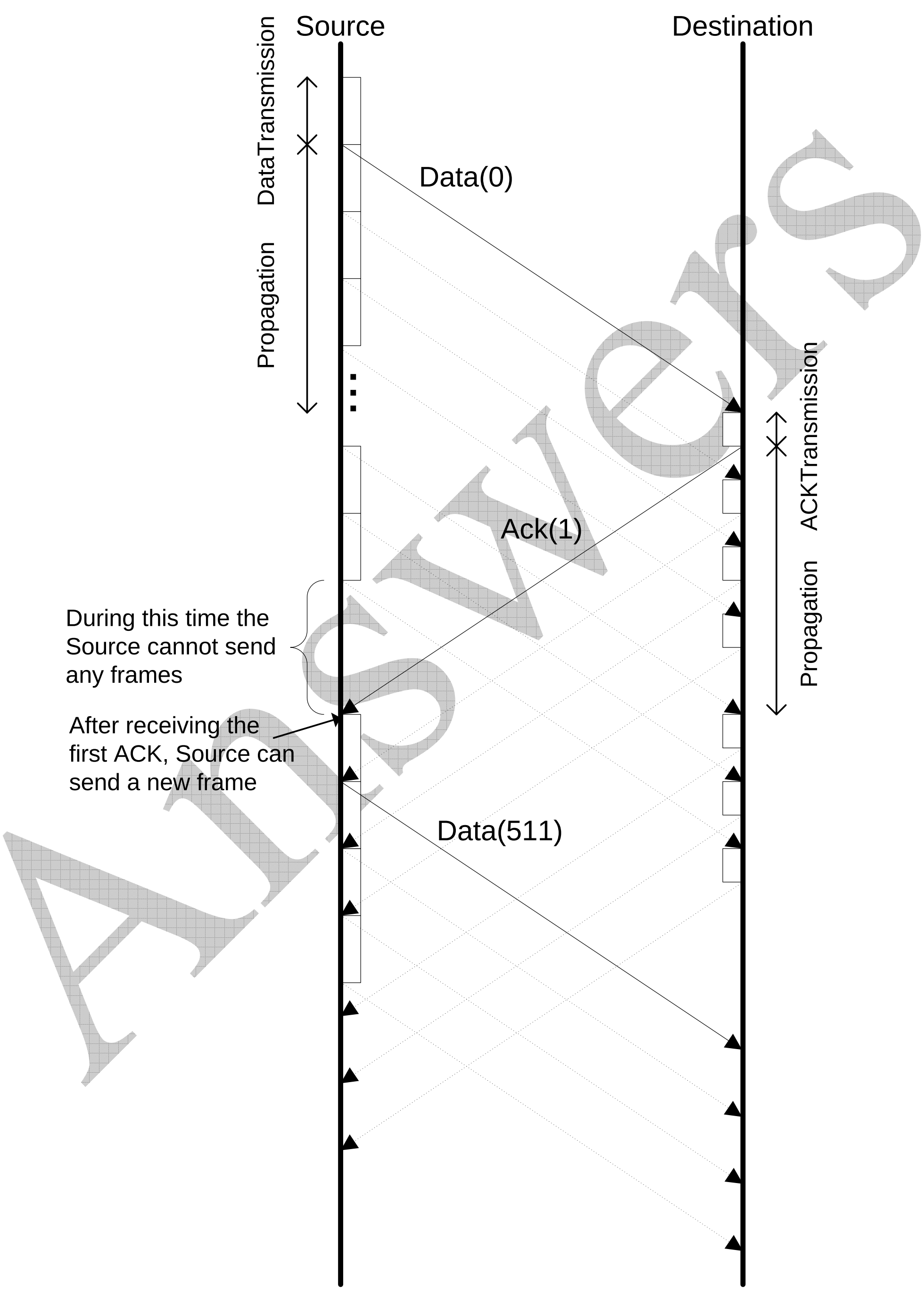
$$\text{Throughput} = \frac{W_{\max} \times OriginalData}{W_{\max} \times DataTransmission + ACKTransmission + 2 \times Propagation}$$

$$= 907426 \text{ bits per second}$$

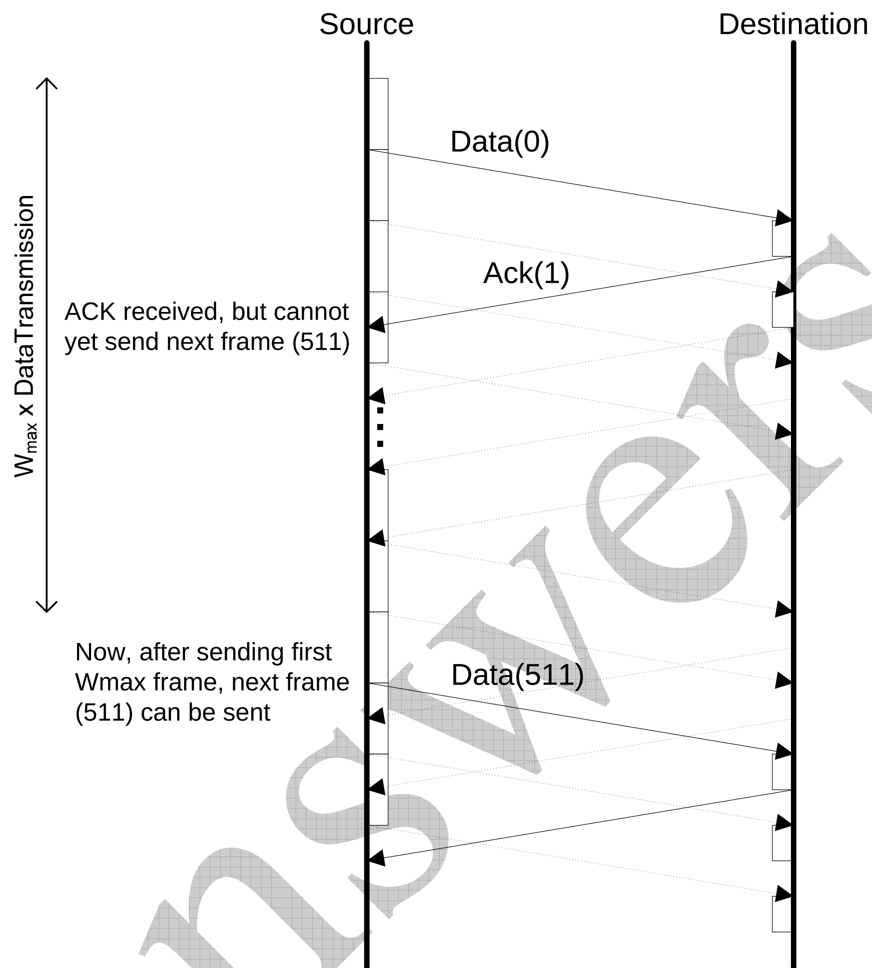
However, the destination may use a different approach: for example, sending one ACK for *every* DATA frame it receives. In this case, a higher throughput can be achieved because the source will receive an ACK earlier, therefore can transmit the next frame when ready (after the first W_{max} frames are transmitted).

But note there are two outcomes possible at the source in this case:

1. The source receives the ACK for the first transmitted DATA frame *after* it has finished transmitting the remaining W_{max} frames. In this case the source must wait for this first ACK before transmitting more frames.



2. The source receives the ACK for the first transmitted DATA frame *before* it has finished transmitting the remaining W_{max} frames. In this case the source must wait until it has finished transmitting these frames, after which it can immediately send subsequent frames.



The two cases above depend on how long it takes to receive the first ACK (compared to the time it takes to transmit the window of frames):

If $DataTransmission + 2 \times Propagation + ACKTransmission > W_{\max} \times DataTransmission$ then:

$$Throughput = \frac{W_{\max} \times OriginalData}{DataTransmission + ACKTransmission + 2 \times Propagation}$$

Else:

$$Throughput = \frac{W_{\max} \times OriginalData}{W_{\max} \times DataTransmission} = \frac{OriginalData}{DataTransmission}$$

For my ID, $DataTransmission + 2 \times Propagation + ACKTransmission = 376.18ms$ and $W_{\max} \times DataTransmission = 4701.2ms$ therefore (following case 2), Throughput = **978261** bits per second.

For selective reject ARQ, when no errors are present it becomes identical to Go-Back-N, except of the maximum window size:

$$W_{max} = 2^{k-1}$$
$$= 256$$

With this maximum window size, the first ACK is received before the window of frames are transmitted, and hence (following case 2), Throughput = **978261** bits per second. This is the same throughput as Go-Back-N.

Part (b).

In comparing the three ARQ protocols, we see that with a long propagation delay, the source in Stop-and-Wait will spend a lot of time waiting for the ACK before it can send the next frame. This is inefficient, and leads to a low throughput of around 23kb/s (efficiency of 2.3%). With Go-Back-N and Selective-Reject, W_{max} frames can be sent, therefore with a long propagation delay, frames can be transmitted while the previous frames (and corresponding ACKs) are propagating along the link. This is more efficient (the source is using the link more often). In fact, with a large W_{max} the ACK returns before the W_{max} frames are sent – as a result the source is transmitting all the time, leading to a high throughput of 978kb/s. This is the highest possible efficiency (97.8%) – the remaining 2.2% is used to send the header.

To generalise, we can say that the throughput/efficiency depends on the ratio between:

- Time spent transmitting (which is W_{max} x DataTransmission)
- Time spent propagating

The more time spent transmitting (versus propagation) the better. Since for our scenario the DataTransmission and Propagation times are constant for all three protocols, the comparison between Stop-and-Wait, Go-Back-N and Selective Reject depends on W_{max}:

- Stop and Wait: $W_{max} = 1$
- Go-Back-N: $W_{max} = 511$
- Selective-Reject: $W_{max} = 256$

Since Go-Back-N and Selective-Reject deliver the same high throughput, it is recommended to use Go-Back-N, as it is simpler than Selective-Reject and does not require a larger buffer to store out-of-order frames at receiver.

Part (c).

The timeout value should allow for enough time for an ACK to be returned from the destination. After transmitting the Data frame, the expected time for an ACK to be returned is:

Propagation of DATA + Processing at Receiver + Transmitting of ACK + Propagation of ACK

Hence, the minimum timeout value should be: 183.39 + 5 + 0.2 + 183.39 = **394** ms.

Question 4 [16 marks]

Under certain circumstances, the efficiency of the ARQ protocols is very low. It is even worse when there are errors. An alternative error control mechanism is to use Forward Error Correction. Assume the Gateway uses a Hamming FEC encoder/decoder for data transferred to the Satellite. The encoding scheme is shown in Table 1.

Data	*Codeword*	*Data*	*Codeword*
0000	010010	1000	101110
0001	101010	1001	001100
0010	001010	1010	100001
0011	000110	1011	001111
0100	110100	1100	010000
0101	010111	1101	101011
0110	010101	1110	000011
0111	110000	1111	110110

Table 1

a) If the 12 bits of data D (where D is calculated in Question 0) are received by the Satellite (in two blocks of 6 bits), explain the steps of the decoder at the satellite on each of the two blocks, including the final result. [8 marks]

b) What throughput can be achieved with the above FEC scheme? (round to nearest bit per second) [3 marks]

c) Increasing the codeword to 8 bits (instead of 6 bits) will reduce the efficiency of the FEC, but increase the error detection/correction capabilities. Explain how the error detection/correction capabilities are increased. [5 marks]

Answers

Part (a).

For a FEC there are different cases at the receiver, depending on the received codeword:

1. If received codeword is VALID, then the receiver assumes the transmitted data is the corresponding data for that received codeword. Note that this could be the correct assumption or incorrect assumption (if bit errors changed the codeword). For this question, we cannot determine whether or not the assumption is correct (since we don't know what the original data was).

2. If the received codeword is INVALID and there is a unique valid codeword with minimum Hamming distance from the received codeword, then the receiver detects an error, and corrects the error, assuming the transmitted data is the corresponding data for the minimum Hamming distance codeword. Again, depending on what was transmitted the error correction could be correct or incorrect.

3. If the received codeword is INVALID and there is two or more valid codewords with minimum Hamming distance, then the receiver can detect the error, but cannot correct it.

For my ID, the result is:

First 6 bits received: 001010. This is a valid codeword, so the receiver assumes the transmitted data was 0010.

Second 6 bits received. 100110. This is an invalid codeword, so the receiver detects an error. There are 3 valid codewords with a minimum Hamming distance of 1 (they codewords are 000110, 101110 and 110110), hence the receiver cannot correct the error.

Part (b).

With FEC there are no ACKs, and hence the source continually transmits bits. However the source transmits codewords, not the original data. With our scheme there are 4 bits of data and 6 bit codewords, hence the efficiency is 66%. As the satellite link is 1Mb/s, the throughput is **666666** bits per second. This assumes there are no errors! If there are errors, then the throughput will be less, because some of the data received cannot be used by the destination. The mathematical analysis of throughput in this case can get quite complex (and we don't cover it in this course).

Part (c).

Consider the ability of the receiver to detect errors. If a codeword is transmitted, and we assume a random number of bits may be in error, then the receiver may receiver anyone of the possible codewords. That is, with 6 bit codewords there are 64 possible codewords. The number of valid codewords depends on the size of the data. With 4 bits of data, there are 16 valid codewords and 48 invalid codewords. So if the receiver receives a codeword there are three cases:

1. The codeword is the same as transmitted. This represents the case of successful transmission with no bit errors.

2. The codeword is one of the 15 other valid codewords. This represents the case of an error occurring, but the receiver NOT detecting the error (a bad scenario!)

3. The codeword is one of the 48 invalid codewords. This represents the case of an error occurring, and the receiver detecting the error. (Depending on the minimum Hamming distance, the receiver may even correct the error).

So, 15 out of 64 codewords result in an undetected error: that's 23% chance of undetected error.

Now consider an increased codeword length: 8 bits. There are no 256 possible codewords, bit still 16 valid codewords. Following the same logic as above we find that 15 out of 256 codewords result in an undetected error. That's a 6% chance of undetected error.

Conclusion: increasing the codeword length (relative to the data length) reduces the chance of undetected errors (or in other words increases the chance of detecting errors).

Note that we have made some unrealistic assumptions (about probability of error). Detailed mathematical analysis of error detection/correction is not covered in this course.

Question 5 [9 marks]

After significant design effort, and several tests, instead of satellite, you finally decide to use a terrestrial microwave link from your home to the mainland (about 30km from your island), and then optical fibre to Bangkok.

a) If you use the ARQ protocols of Question 3 over the microwave link, do you think the throughput will increase (compared to satellite). Explain why for each ARQ protocol. (You do not need to perform calculations; explain your answer). [9 marks]

Answer

We saw in Question 3 that the throughput depends on the time spent transmitting, versus the propagation time. In Question 3 the satellite link contributed to a very high propagation delay. Transmitting up and down to the satellite gives a distance of about 72000km. With terrestrial microwave (that is wireless transmission between towers) and optical fibre, the transmission distance depends largely on the distance between the computer and server. Note the maximum length of Thailand is less than 2000km, so the distance between Bangkok and your island in the source is probably about 1000km. The propagation delay is going to very small in this new scenario (when compared to Question 3) – about 70 times smaller.

What about data rates? The satellite link was 1Mb/s. Although the data rates vary significantly, we can assume the terrestrial microwave and optical fibre will be at least 1Mb/s, possibly more.

For Stop and Wait ARQ, we may expect similar throughput to the satellite scenario, IF the data rate in the new scenario is around 70Mb/s. If the data rate is less than this, then the throughput in Question 4 will be higher than the throughput in Question 3.

With Go-Back-N and Selective Reject it also depends on W_{max}, since with a high value, the time spent transmitting is high. In my example, W_{max} was quite high (511 or 256), and so the throughput was independent of data rate – it is likely that with Question 5 the high throughput if 978kb/s will also be achieved because of W_{max}.