Faculty of Computing, Engineering and Science

Assessment Cover Sheet and Feedback Form 2019-20

Module Code:	Module Title:	Module Team:
NG2S800	Analogue and Digital Communications	Sivagunalan Sivanathan, Leshan Uggalla

Assessment Title and Tasks:	Assessment No.
Various lab Experiments	1

Date Set:	Submission Date:	Return Date:
14-Oct-2019 16:00	16-Mar-2020 23:55	15-Apr-2020 23:55

IT IS YOUR RESPONSIBILITY TO KEEP RECORDS OF ALL WORK SUBMITTED

Marking and Assessment

This assignment will be marked out of 100%

This assignment contributes to 30% of the total module marks.

Part B: Marking and Assessment
(to be completed by Module Lecturer)

This assignment will be marked out of 100%

This assignment contributes to 30% of the total module marks.

This assignment is bonded / non- bonded. Details : All learning outcomes are covered in this assignment

Course Work Task:

Marking Scheme	Marks Available	Marks Awarded
Part A:		
• Synthesis of periodic waveform	20	
• Amplitude Modulation and Amplitude shift keying	20	
Part B:		
• Frequency Modulation and Frequency shift keying	25	
• Digital Modulation Schemes	25	
• Report structure	5	
• On time submission*	5	
Total	100	
*This marks will be awarded for the interim submission (Draft Assignment)		

Learning Outcomes to be assessed (as specified in the validated module descriptor https://icis.southwales.ac.uk/):

L01: To be able to analyse and synthesise a range of periodic and non-periodic signals in the time and frequency domains.

L02: To be able to understand the requirements of analogue and digital communications systems for designing and evaluating their system performance

Grading Criteria:

Performance Level	**Criteria**
Fail (< 40%)	Insufficient documentation and poor report writing with substantial flaws
3rd Class / PASS (40% - 49%)	Very basic documentation of experiment, with little background and insignificant conclusion
Lower 2nd Class / PASS (50% - 59%)	Some parts addressed well, but overall relatively basic or with significant flaws
Upper 2nd Class / MERIT (60% - 69%)	As 1st class, but with minor flaws in each section
1st Class / DISTINCTION (70% +)	Very good documentation of experiment, display of results in graphical form, in-depth

Learning Outcomes to be assessed (as specified in the validated module descriptor https://icis.southwales.ac.uk/):

L01: To be able to analyse and synthesise a range of periodic and non-periodic signals in the time and frequency domains.

L02: To be able to understand the requirements of analogue and digital communications systems for designing and evaluating their system performance

Provisional mark only: subject to change and / or confirmation by the Assessment Board

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1st Class / DISTINCTION (70% +)	Very good documentation of experiment, display of results in graphical form, in-depth

Analogue and Digital communication
Course Work Detailed Requirement

Introduction

There is one coursework assignment associated with this module. This course work assignment has four lab sessions required to be complete each in the lab environment and must critically analyse the results in the report.

The lab work must be carry out during the lab sessions and students may record the outputs/measurements. Measured results must be critically analysed outside the lab hours. Students are require to prepare a full Lab report for each lab sessions, these can be combined to a single Lab report for the final submission.

This assignment required to submit an electronic (preferably pdf) copy that needs to be uploaded into the blackboard. This copy will run through Turnitin, the University Plagiarism software.

Student are given opportunity to submit their draft assignment on or before 03/12/2019 and the informal feedback will be provided to the individuals.

Part A

Lab Experiment 1

- The synthesis of Periodic waveform.
- The Harmonics and fundamental frequency.
- The basic of Fourier series.
- Identify the elements of harmonic synthesiser circuit.

Lab Experiment 2

- The basic concepts and process of amplitude modulation (AM) and amplitude shift keying (ASK).
- The modulation Index.
- The concept of over-modulation & distortion.
- Identify elements of AM and ASK modulator circuit.

Part B

Lab Experiment 3

- The basic concepts and process of frequency modulation (FM) and frequency shift keying (FSK) with associated parameters.
- The Basic characteristics of voltage controlled oscillators (VCO).
- Identify elements of FM and FSK modulator circuits.
- Analyse the spectrum of an FM signal and compare observation with theoretical expectation.

Lab Experiment 4

- The Basic concepts of modulation schemes.
- Analyse the following schemes using MATLAB Simulink.
    - FM

        - BPSK
        - QPSK
        - QAM
    - Results analysis of individual schemes using different SNR scenarios.

Deliverables [1] Experiment result.

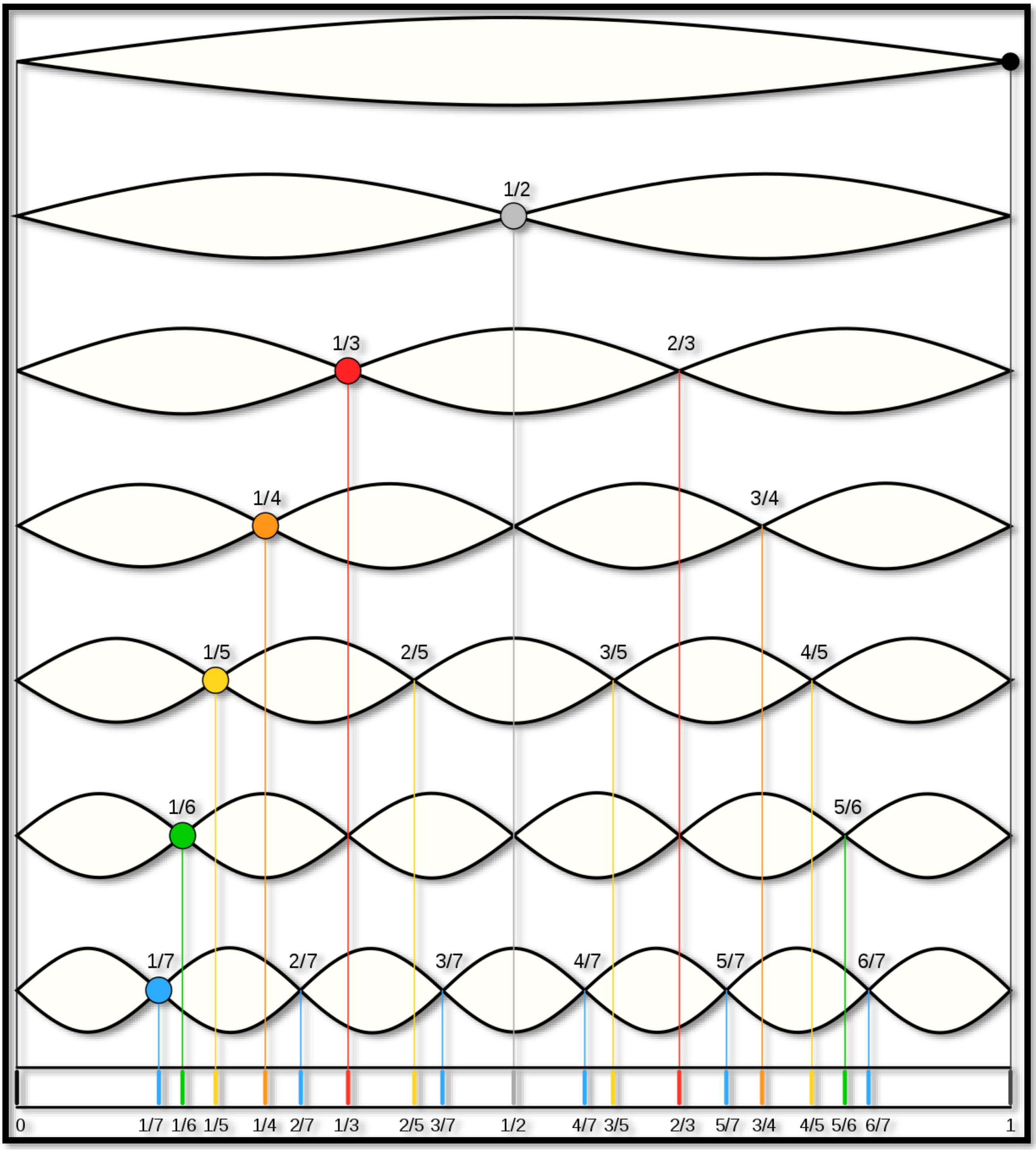
Marks will be awarded for:

- Appropriate experiment setup and procedures.
- Evidence of an expected lab results or observations.
- Comparison method used to verify the experiment result such as simulations.
- Critically analyse the Lab results against the theory or simulation.

Deliverables [2] Documentation.

- Table of contents.
- Introduction.
- Body.
- Discussion and Analysis.
- Conclusion.
- Reference List.

Harmonics



[1]

Ben Edwards: 18026826

Index:

Figures:

Square wave

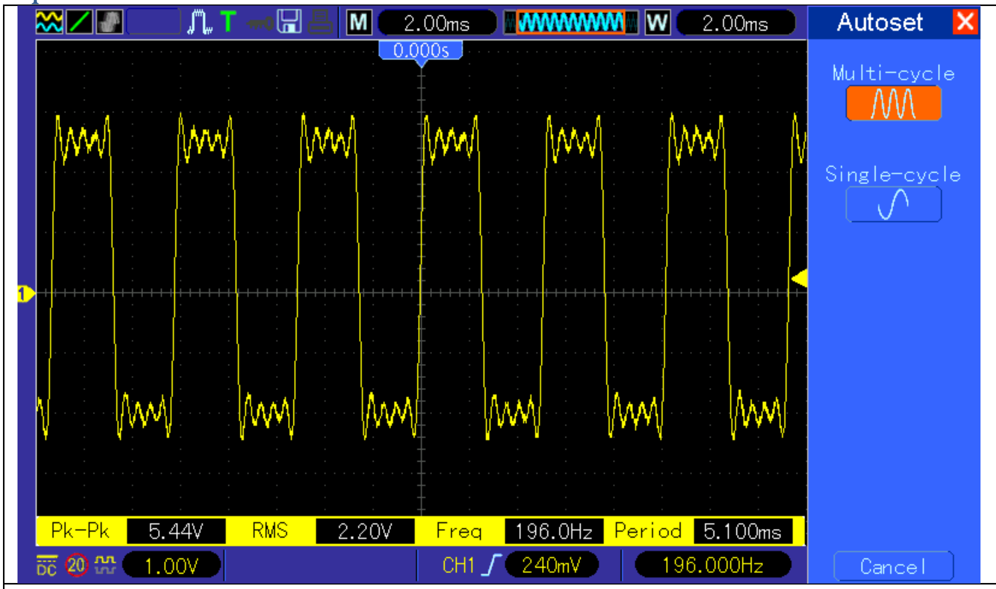


Fig 1.1

Harmonics 1, 3, 5, 7 & 9

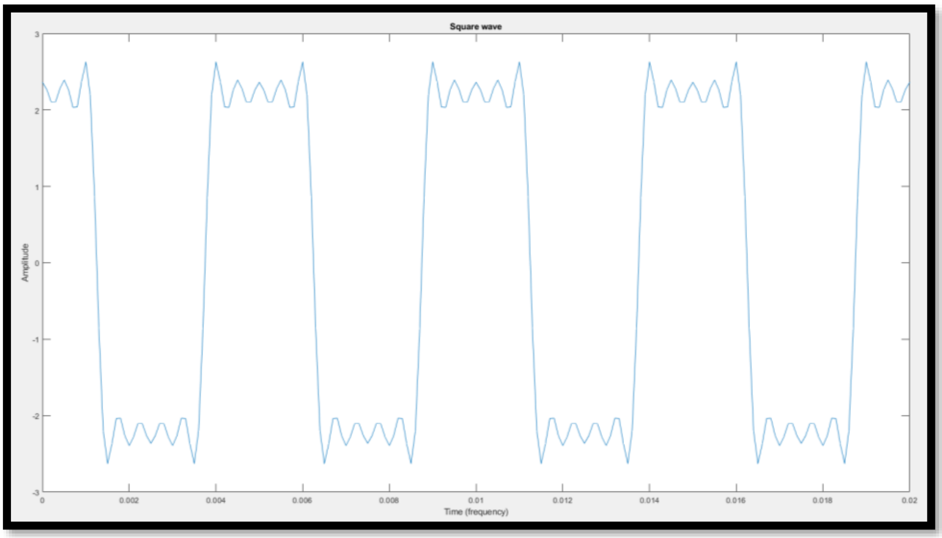


Fig 1.2

Simulated waveform

Triangle Wave:

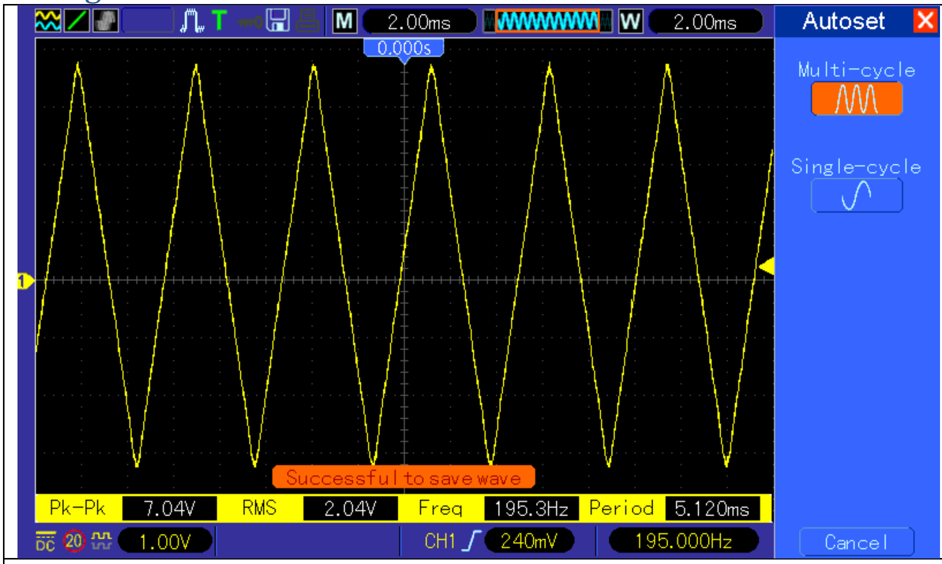


Fig 2.1

Harmonics 1, 3, 5, 7 & 9

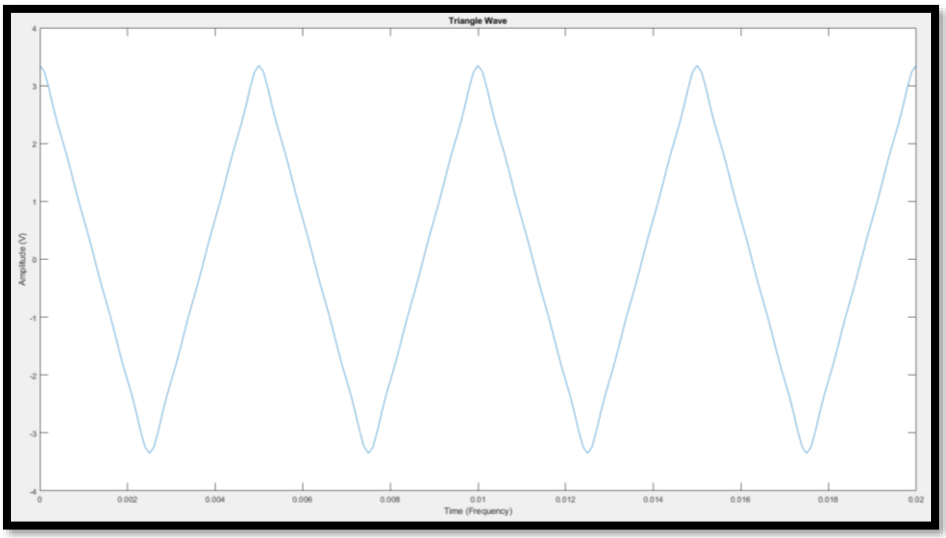


Fig 2.2

Simulated waveform

Sawtooth Wave:

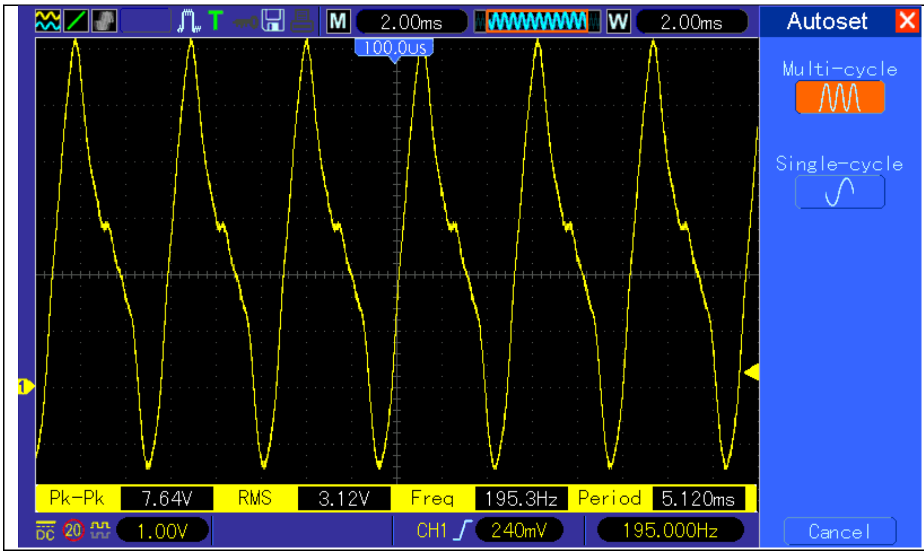


Fig 3.1

Harmonics 1, 2, 4, 6, 8 & 10

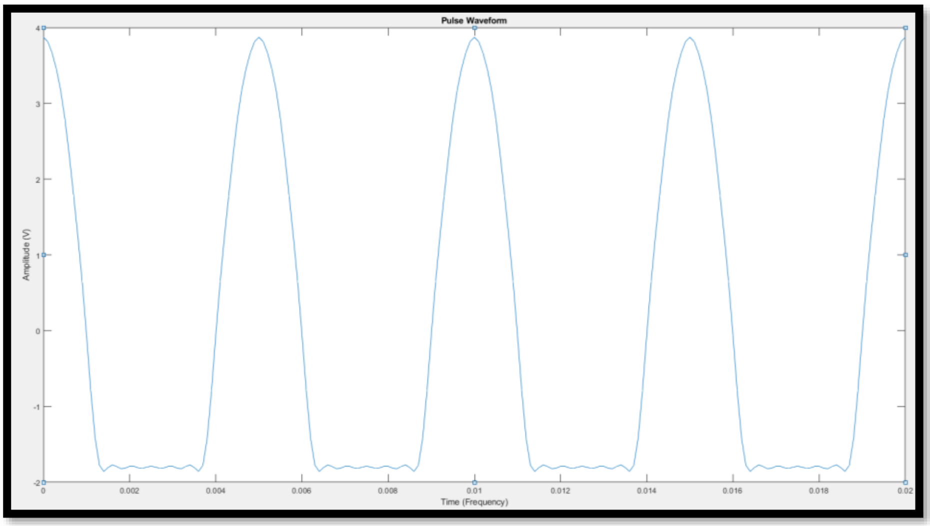


Fig 3.2

Simulated waveform

RNF wave:

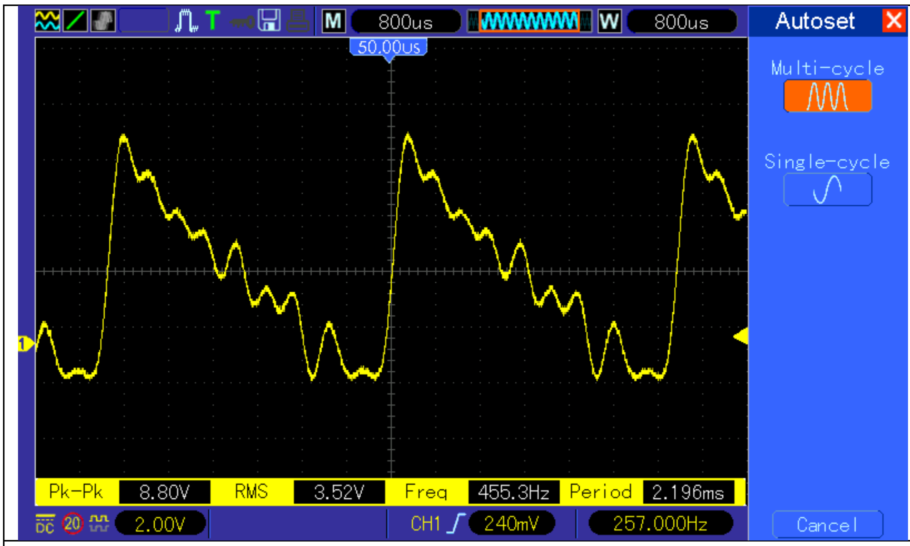


Fig 4.1

Harmonics 1, 2, 3, 4, 5, 6, 7, 8, 9 & 10

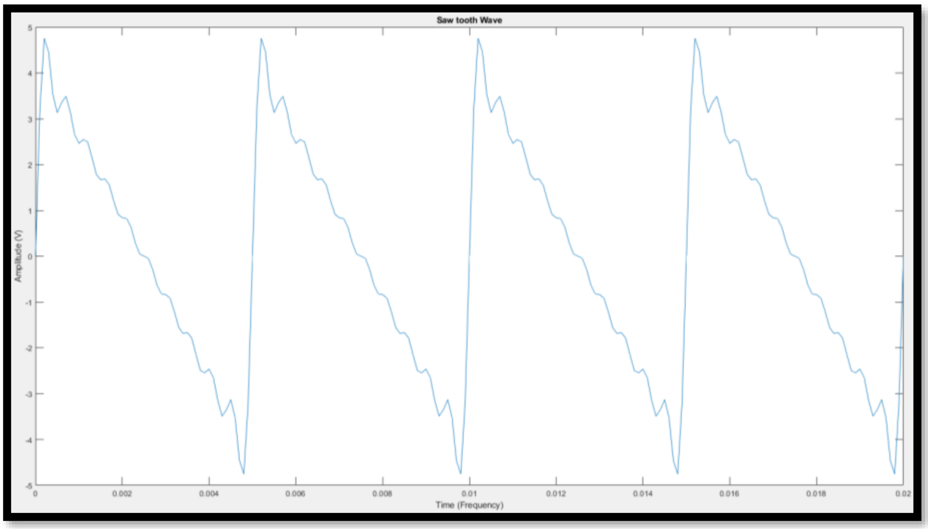


Fig 4.2

Simulated waveform

Complex Waveform:

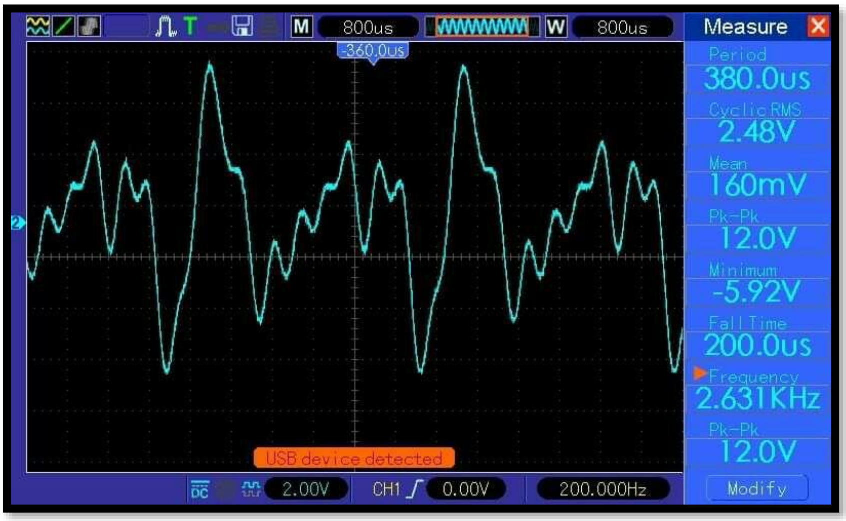


Figure 5.1
Practical waveform

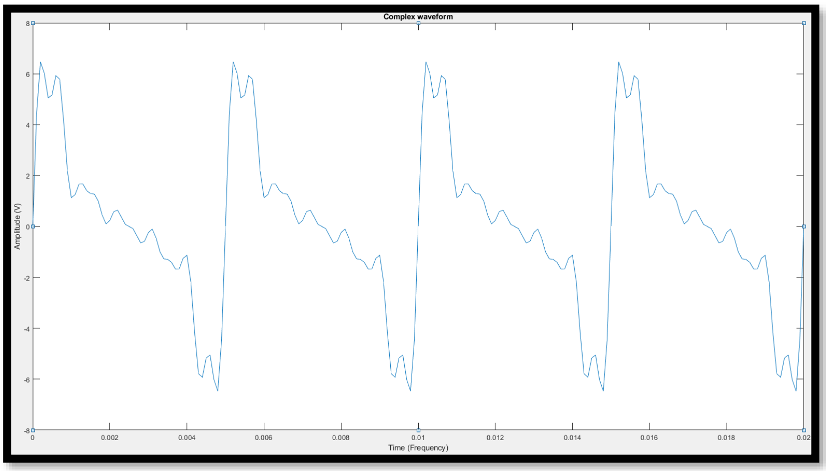


Figure 5.2
Simulated waveform

Tables:

Harmonic, n	1	3	5	7	9
Ampl, V	2.83	0.943	0.566	0.404	0.314
RMS, V	2	0.667	0.400	0.286	0.222
Phase, deg	0	180	0	180	0

Table 1.1

Harmonic, n	1	3	5	7	9
Ampl, V	2.83	0.314	0.113	0.058	0.035
RMS, V	2	0.221	0.079	0.041	0.024
Phase, deg	0	0	0	0	0

Table 2.1

Harmonic, n	1	2	4	6	8	10
Ampl, V	2.83	1.2	0.24	0.103	0.057	0.036
RMS, V	2	848m	169m	72.8m	40m	25m

Phase, deg	0	0	180	0	180	0

Table 3.1

Harmonic, n	1	2	3	4	5	6	7	8	9	10
Ampl, V	2.83	1.414	0.943	0.707	0.566	0.471	0.404	0.354	0.314	0.283
RMS, V	2	999m	666m	499m	400m	332m	285m	250m	221m	200m
Phase, deg	0	0	0	0	0	0	0	0	0	0

Table 4.1

Harmonic, n	1	2	3	4	5	6	7	8	9	10
Ampl, V	2.82	2.45	1.89	1.22	0.57	0	0.404	0.612	0.628	0.49
RMS, V	1.99	1.73	1.34	0.86	0.40	0	0.29	0.43	0.44	0.35
Phase, deg	0	0	0	0	0	180	180	180	180	180

Tabel 5.1

Introduction:

for this assignment the task is taking a fundamental frequency with respect to its harmonic frequencies, synthesizing them to create an array of different wave forms. These include the square wave, the triangle wave and a saw tooth wave.
To do this, the synthesiser board provided to the class by the university was used. Using this we could alter the amplitude, phase and which harmonic frequencies I wanted to use to create each waveform. Measuring the results on an oscilloscope, then screenshots of the created waveforms would be taken as proof of concept.
A simulation of the waveform would then be created using the MATLAB programming environment. Allowing for a comparison of both the practical and the simulated results. By comparing the two graphs and waveforms, it can be determined whether the practical result is giving a correct result. And if not, then an explanation should be given as to why it might be possible that they are different.

Harmonics:

Harmonics can be described as the component frequency or the multiples of the fundamental periodic frequency. This is referring to the fact that the harmonic is multiplied by the fundamental frequency to the number that the harmonic is in reference to. As an example, if the fundamental frequency was 100Hz, the 2nd harmonic would be 100Hz * 2 = 200Hz, 3rd would be 100Hz * 3 = 300, 4th is 100Hz * 4 = 400Hz etc. **[2]**
Harmonics can have a negative effect on the transmission of electronic signals. However, since transmitting signals can omit harmonic signals which can then interfere with other receiver devices, interfering with other signals at the frequency within the harmonics **[3]**. The International Telecommunications Union, ITU, are responsible for issues that involve the transmission of information and communication technology. They oversee assigning frequencies to communication companies which can include TV channels, radio stations and more importantly airport control. Any frequencies that disrupt this system will be investigated by the ITU and dealt with as soon as possible.

Square wave:

To create the periodic square wave, the fundamental, 3rd, 5th, 7th and 9th harmonics are put through an adder to create the desired square wave. However, first there had to be an alteration of the phase of the 3rd and 7th harmonic, shifted out by 180 degrees for the desired subtraction. The adjustment of the amplitude of the harmonics had to be the next priority. Converting the peak voltage value to Vrms values, using the formula **Vrms = Vpk * 0.7071,** which can be measured using a multimeter.
After adjusting the amplitude of the harmonics, they would then be added together to create the desired square waveform shown below in fig 1.1.

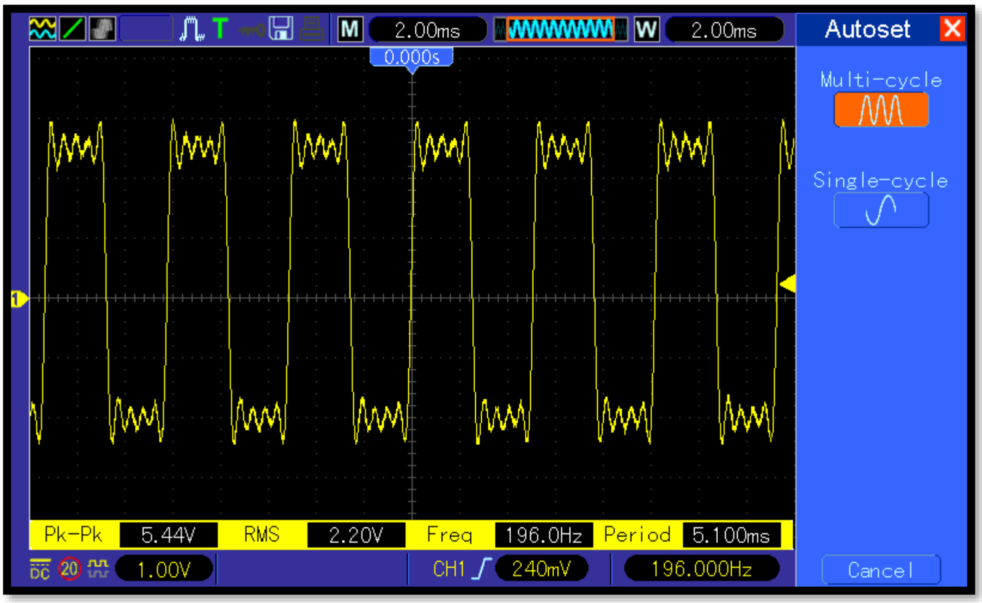


fig 1.1

From looking at the results of the synthesised harmonics, it can be seen that the number of spikes that appears at the peaks of the wave will equal to the highest harmonic that has been added to the fundamental frequency. It can also be seen that the rising and falling edge of the waveform has become more dramatic in comparison to the smooth periodic fundamental frequency. The wave is still far from a perfect square wave; however, it would still be suitable for basic applications. A very simple example of one of these uses would be this demonstration. For educational and testing purposes or for an application that doesn't require a perfect signal. To perfect this signal even further, many more harmonics must be added to the waveform following the same pattern to create the saturation like effect at the peak.

When the harmonics get added to each other it appears as if the signal is also being subtracted considering the signal reaches a saturation point instead of progressively increasing in amplitude. This is because every other harmonic in the formula is out of phase by 180 degrees. Because of this, when two signals are out of phase from each other, this acts as a subtraction of the two signal amplitudes **[4]**. In this example, with the FF being 2Vrms and the third harmonic being 0.667Vrms. this should then calculate out to a value of…
(2Vrms – 0.667Vrms = 1.793Vrms)

When connecting the output of the added harmonic signals to a speaker the created waveform can be heard from the speaker at 196Hz and 1.48Vrms when the speaker is on as recorded on a multimeter. When the speaker is off the rms voltage rises to a 2.20Vrms. the speaker would emit an audible buzzing noise when the signal is left untouched. However, when the phase shift of the waveform changed by 180, there was a noticeable change in its sound which is hard to describe, but almost as if it was playing at a higher pitch. The rms waveform is affected by the activation of the speaker from a 2.20Vrms and 1.48Vrms equalling to a 0.72Vrms drop when the speaker is switched on. The reason it does this is because the impedance of the speaker is equal to 8Ω, and so will consume power and dissipate it as heat.

The table for this can be shown below as…

Harmonic, n	1	3	5	7	9
Ampl, V	2.83	0.943	0.566	0.404	0.314
RMS, V	2	0.667	0.400	0.286	0.222
Phase, deg	0	180	0	180	0

Table 1.1

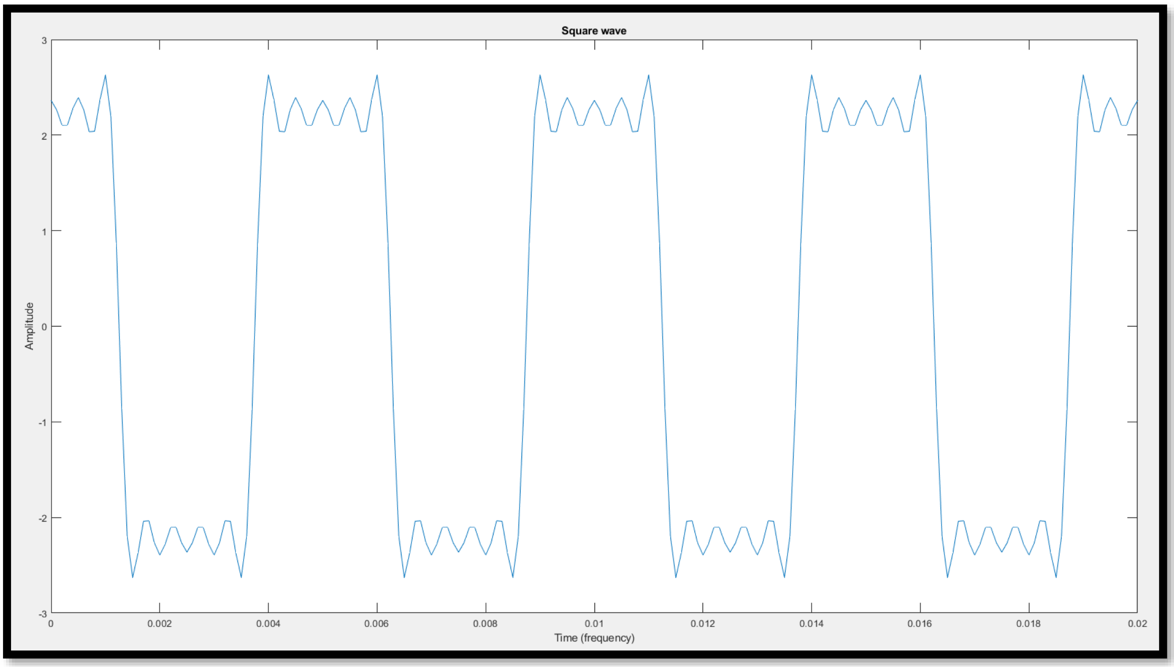


Figure 1.2

Using MatLab to simulate the effect of harmonic signals, it can be seen that the produced output is a square wave with spikes at the higher and lower peaks of the waveform. This result confirms that the practical result obtained was accurate to the expected result.

Triangle Wave:

To create the periodic triangle wave, the fundamental, 3rd, 5th, 7th and 9th harmonics was used similarly to the last square wave that had been constructed. For this periodic wave, none of the harmonics needed to be phase shifted by 180 degrees, so all the harmonic signals were in phase with each other. The formula…

Vrms = Vpk * 0.7071

…will be used again to calculate the rms values for each signal's amplitude, adding the signals together to create the desired triangle wave.

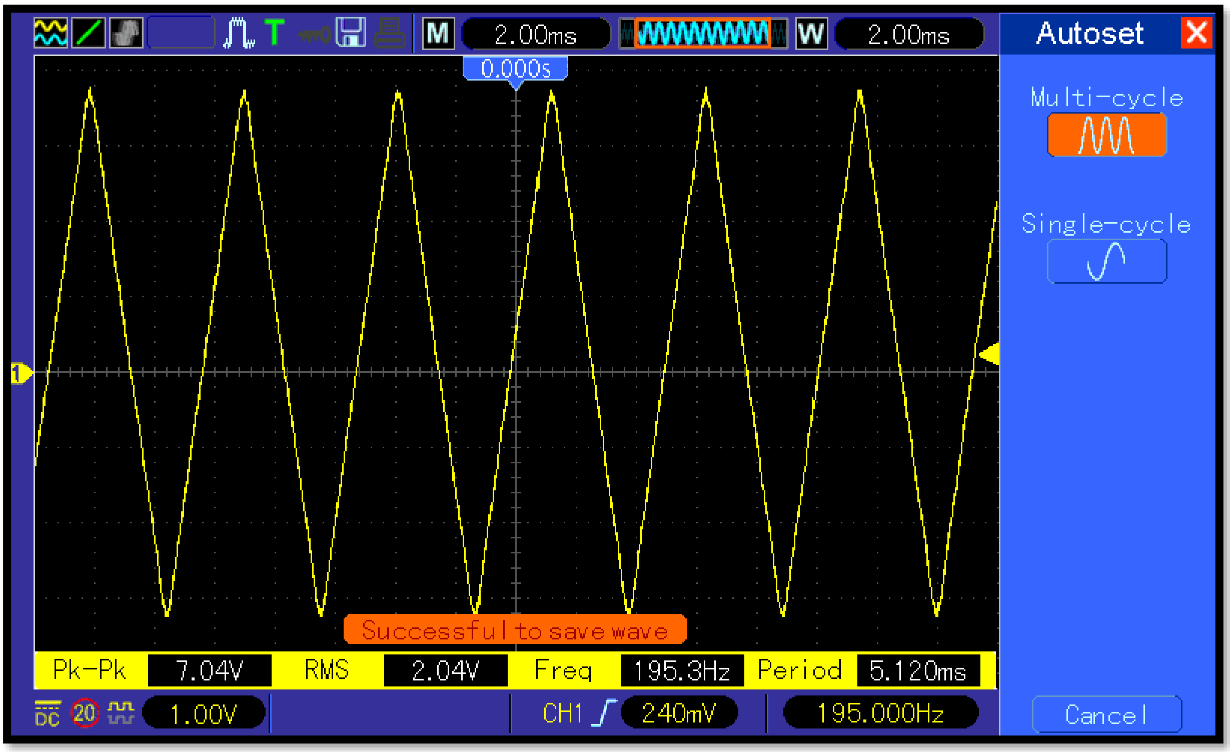


fig 2.1

From looking at the final signal off the bat, it can be seen that the signal looks very clean and is probably almost identical to the desired frequency wave shape. The running frequency is at 195.3Hz, almost the same frequency as the square wave above. The RMS voltage is recorded as a 2.04Vrms on the oscilloscope. However, after looking at the rms values for the other harmonic signals it can be seen that the 5^{th} 7^{th} and 9^{th} harmonics are at a rms value of 200mV while also having pk-pk values quadruple that of the intended amplitudes when they should be 79mV, 41mV and 24mV respectively. The intended values for these harmonics were set using a multimeter at those values, so there might be an issue with the oscilloscope or the testing board that was being used. Even with the 3^{rd} harmonic reading at a 320mV when it should be at a 221mV. My best guess is that the oscilloscope is having trouble with reading values lower than 200mVrms and reading an accurate pk-pk value.
The table for this can be shown as…

Harmonic, n	1	3	5	7	9
Ampl, V	2.83	0.314	0.113	0.058	0.035
RMS, V	2	0.221	0.079	0.041	0.024
Phase, deg	0	0	0	0	0

Table 2.1

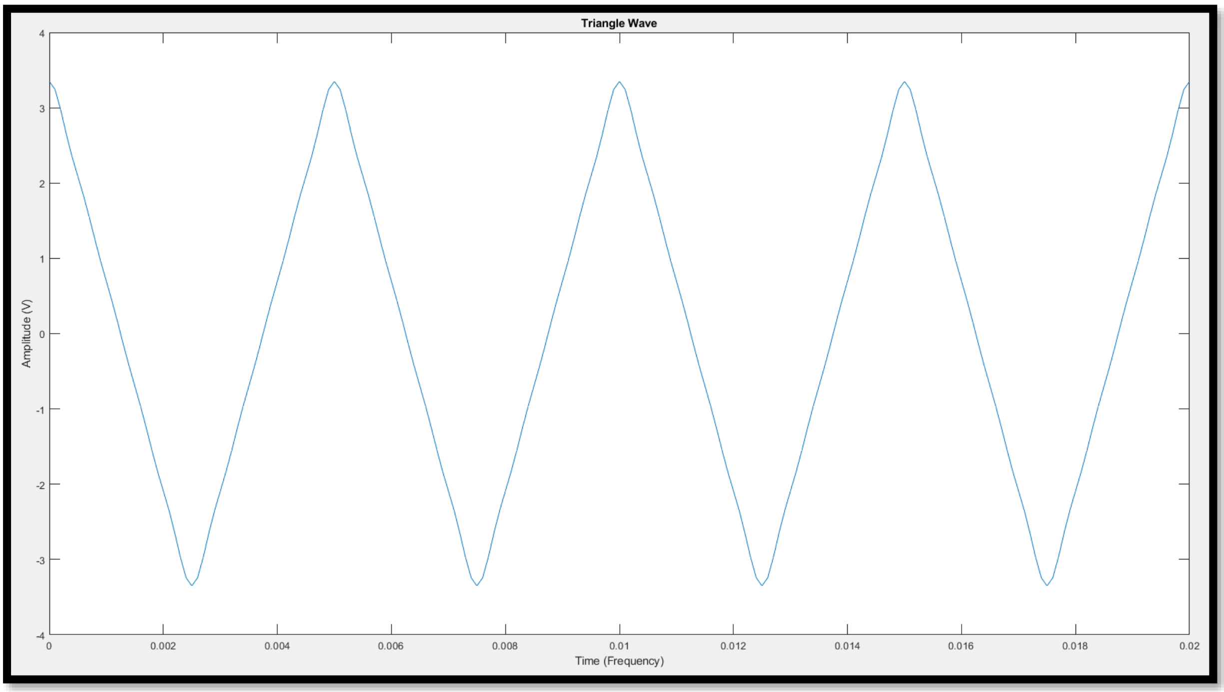


Figure 2.2
This is the simulation of the waveform above; a triangle waveform can clearly be shown on this graph. It can also be seen in the simulation that there are slight ridges and bumps along the waveform, showing that this is far from a perfect waveform. This proves that the practical result is also correct when testing it with the comms training board.

Pulse wave:

To create the pulse wave below the positive harmonics would need to be added to the fundamental frequency. The 4th and 8th harmonic have a phase shift of 180 degrees for this signal creation. Again, the Vpk to Vrms formula that has been specified above is used again to calculate the correct rms voltage to set the amplitude for each harmonic.

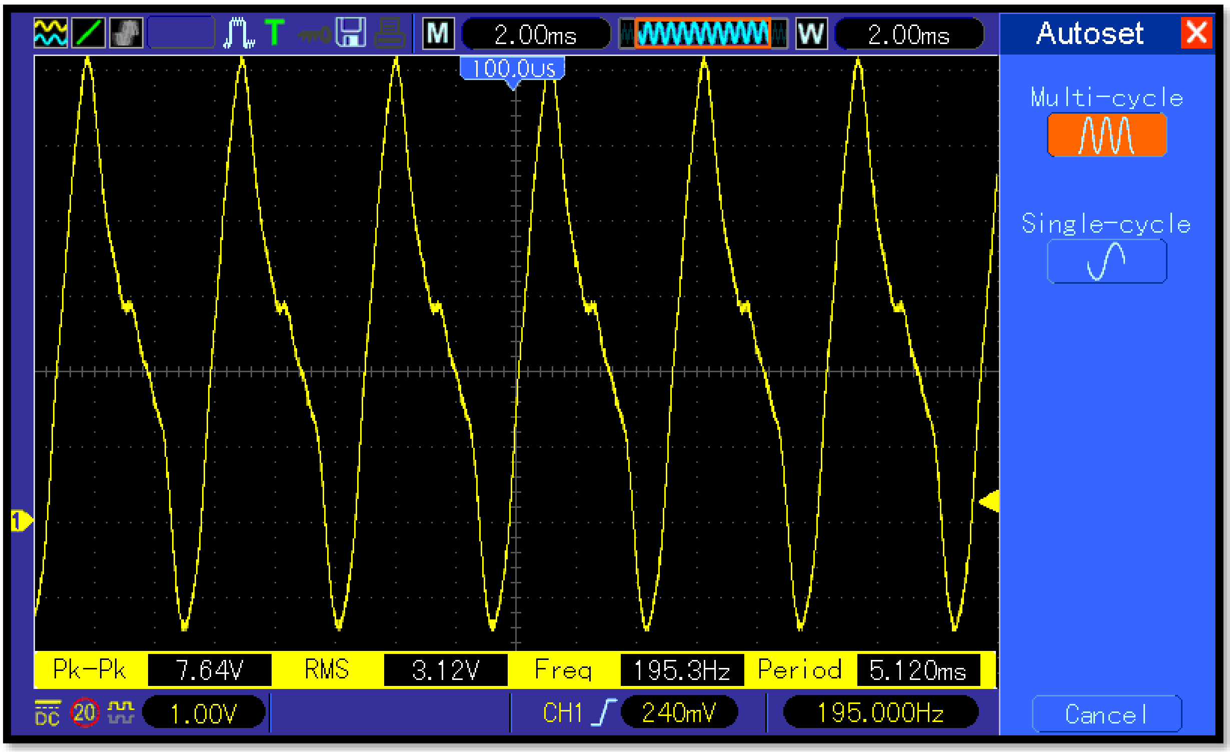


fig 3.1

From looking at these results, it can be seen that there is a sawtooth like wave. However, it is not a perfect waveform as there is an imperfection on the falling edge of the cycle. The frequency reads similarly to the previous signals with a 195.3Hz frequency. And looking through the expected rms voltages and the recorded rms values from the oscilloscope are nowhere near to being close, with a fundamental frequency Vrms value of 2V as the calculation and a value of 2.96Vrms recorded on the oscilloscope. The 6th harmonic is calculated to be 72.8mVrms but is measured as 2.24Vrms. this seems to be a recurring issue, so from now on the reading from the multimeter will be trusted for reading the voltage value of the harmonic's waves. There had been negligence to record these results regrettably. This is not a perfect waveform and I believe this can be caused by an irregularity with the phase difference set using the harmonics module.

This can be shown in the table below as…

Harmonic, n	1	2	4	6	8	10
Ampl, V	2.83	1.2	0.24	0.103	0.057	0.036
RMS, V	2	848m	169m	72.8m	40m	25m
Phase, deg	0	0	180	0	180	0

Table 3.1

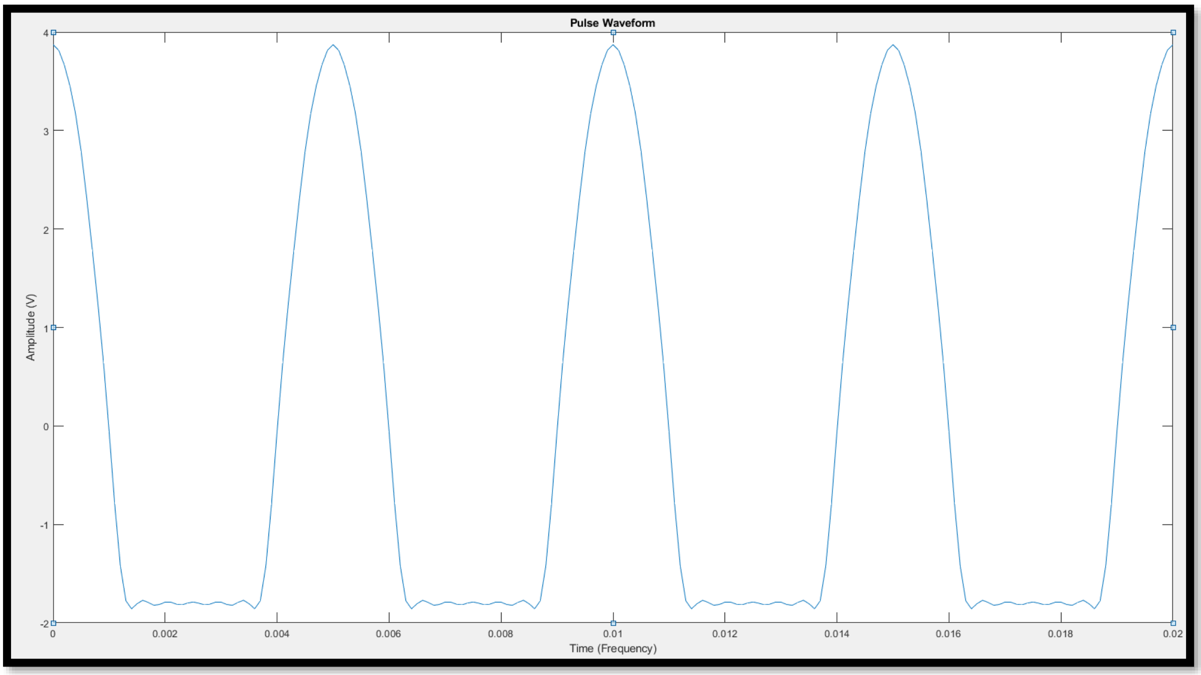


Fig 3.2

Looking at this result, it can be determined that the pulse response from the practical testing has some fundamental flaws. For the first thing, the pulse itself doesn't have a smooth rise and fall time, with the practical having a notch on the falling edge of the waveform. Secondly, the practical example doesn't have a constant low amplitude period. It instead falls and then immediately rises again. A reason for this could be down to the phase of the harmonics being set to the wrong value. Either due to the old components not being as accurate as they would have been in the past or we could have incorrectly set the phase for some phases when altering the waveform phase.

Sawtooth wave:

This next wave uses all the harmonics ranging from the fundamental frequency to the 10th harmonic. However, there is no phase shifting from any of the harmonic signals. By converting the Vpk values to rms values, a multimeter can be used to set the voltages of each harmonic signal.

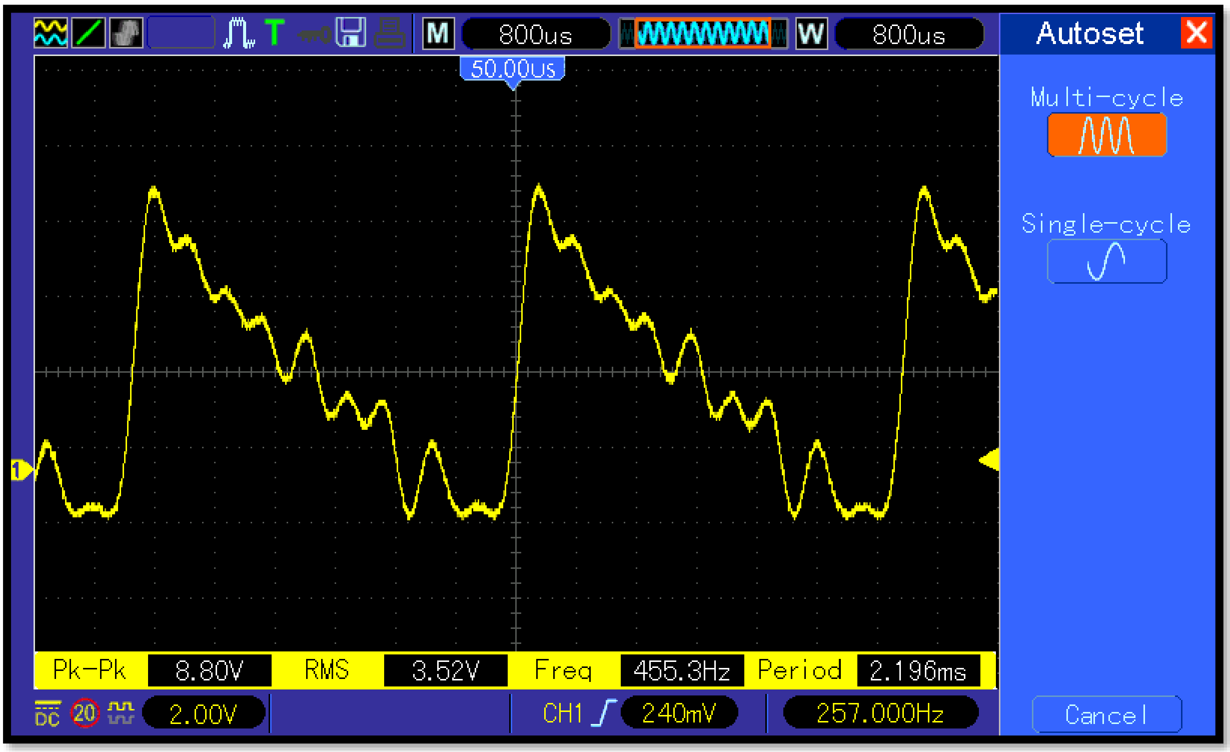


fig 4.1

By looking at the signal it can be seen that there is a significant amount of noise on the falling edge of the signal that is being generated. Using more harmonic signals can be believed to make this become an ideal, smooth, signal that will be more accurate for its decided purpose. The frequency recorded on the oscilloscope can be seen as 455.3Hz. This is a combination of all the waveforms at the same phase value.

This can be shown in the table below as…

Harmonic, n	1	2	3	4	5	6	7	8	9	10
Ampl, V	2.83	1.414	0.943	0.707	0.566	0.471	0.404	0.354	0.314	0.283
RMS, V	2	999m	666m	499m	400m	332m	285m	250m	221m	200m
Phase, deg	0	0	0	0	0	0	0	0	0	0

Table 4.1

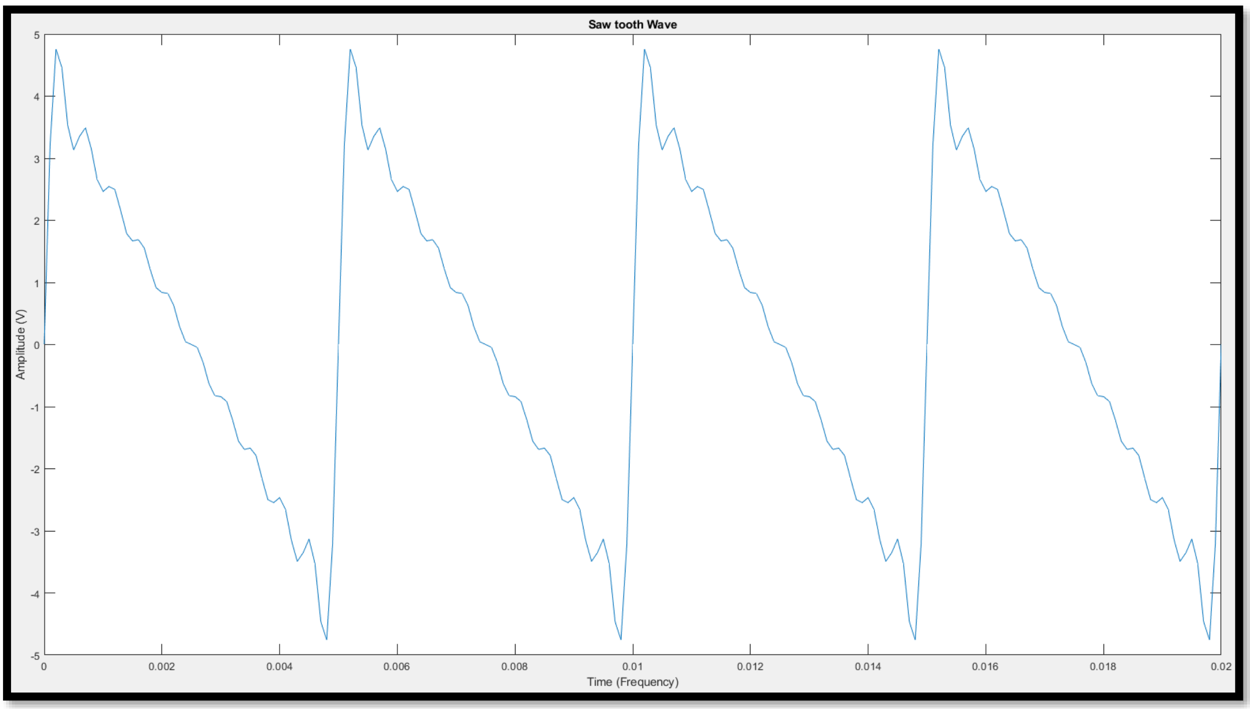


Fig 4.2

Looking at the simulated waveform above, it can be shown that this can be compared to the practical example in figure 4.1. comparing the two waveforms, it can be seen that the two graphs are very similar to each other showing that the practical result has given a correct result.

Complex waveform:

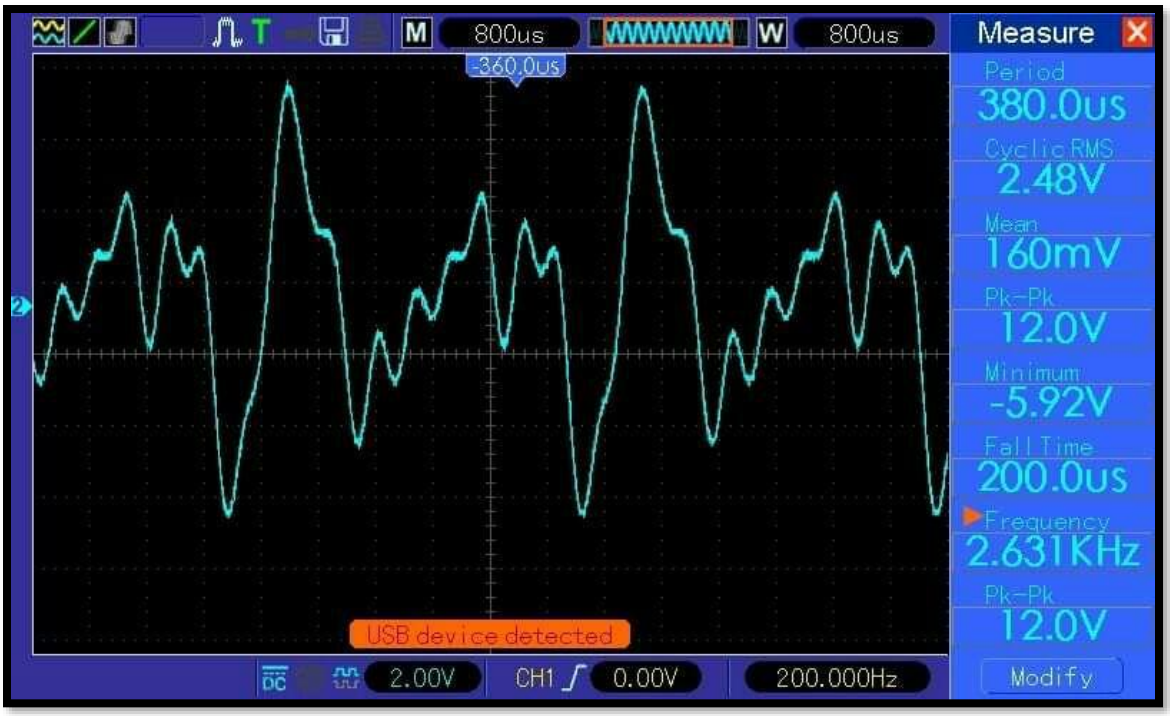


Figure 5.1

Looking at the waveform above that was projected on the oscilliscope, it can be seen that this complex waveform spikes rapidly at the beginning of the wave to the peak, then a jaggid rise before a sharp fall down to the negative peak. Mirroring the positive half of the waveform. This waveform is similar to the the saw tooth wave combined with the pulse waveform. The frequency is recorded as 2.6KHz and a voltage of 12Vpk-pk. This waveform is the combination of the fundimental, 2nd, 3rd, 4th, 5th, inverse 6th, inverse 7th, inverse 8th, inverse 9th and inverse 10th waveforms.

The table for this can be shown below as…

Harmonic, n	1	2	3	4	5	6	7	8	9	10
Ampl, V	2.82	2.45	1.89	1.22	0.57	0	0.404	0.612	0.628	0.49
RMS, V	1.99	1.73	1.34	0.86	0.40	0	0.29	0.43	0.44	0.35
Phase, deg	0	0	0	0	0	180	180	180	180	180

Tabel 5.1

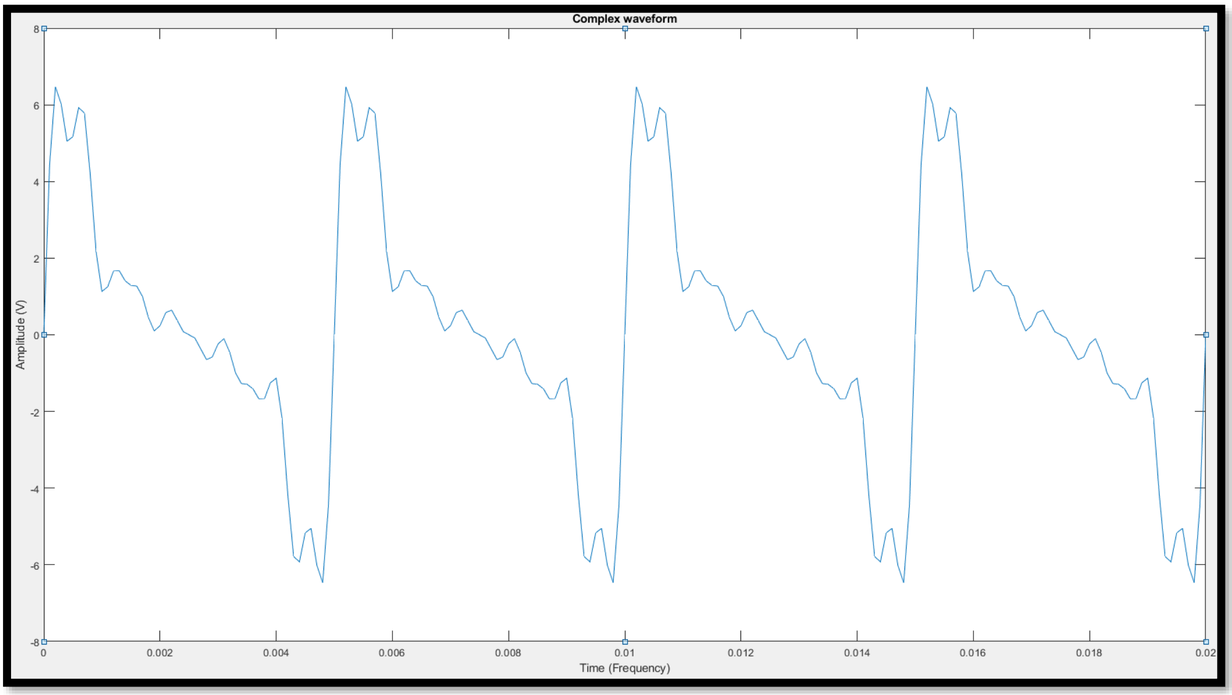


Figure 5.2

Looking at this simulated waveform graph created in MATLAB and comparing it to the practical waveform, it can be shown that there are similarities to the waveforms but with a difference. The amplitude of the points between the two peaks are at a downward sloping angle while the practical example had an upward slope. Looking at the two however, the two waveforms are similar enough that the practical waveform can be considered a successful waveform.

Effects of filters and harmonics:

Limiting the waveforms to just the few harmonics, synthesising them to create a waveform would be enough to create a simple shape of the desired signal. However, the resulting waveform would be imperfect and rough. Without the additional higher harmonics to continue to smooth the resulting waveform, the waveform would be jagged and would have different effects on sound output and when applied as a signal clock some errors could occur.

If these waveforms were to be but through a low pass filter at around a cut-off value of 200Hz, then what would be left would be the fundamental frequency waveform. This is because the filter would cut out all the other harmonics which have a frequency multiplied by the number of that harmonic. So, for example, the 5th harmonic would be the fundamental frequency (FF) times by the harmonic number, so…

(5th harmonic frequency = FF * 5)

Conclusion:

In conclusion, it has been discovered that the number of harmonics synthesised together with the fundamental can have an effect on the shape of the synthesised waveform and how smooth that waveform becomes. The phase of certain harmonics can also have an impact on the waveform, opening more options available to change how the waveform will be shaped. Altering these values will allow someone to create a custom waveform that can be applied to suit their current needs.

The Vrms values for the amplitude can be calculated by timing the amplitude voltage by 0.7071. This can be used to calculate the Vrms of the synthesised waveform. To shape the synthesised signal even further, it is possible to increase the peak voltage at higher harmonic frequencies or decrease the voltage with subtraction by changing the phase of that harmonic by 180 degrees.

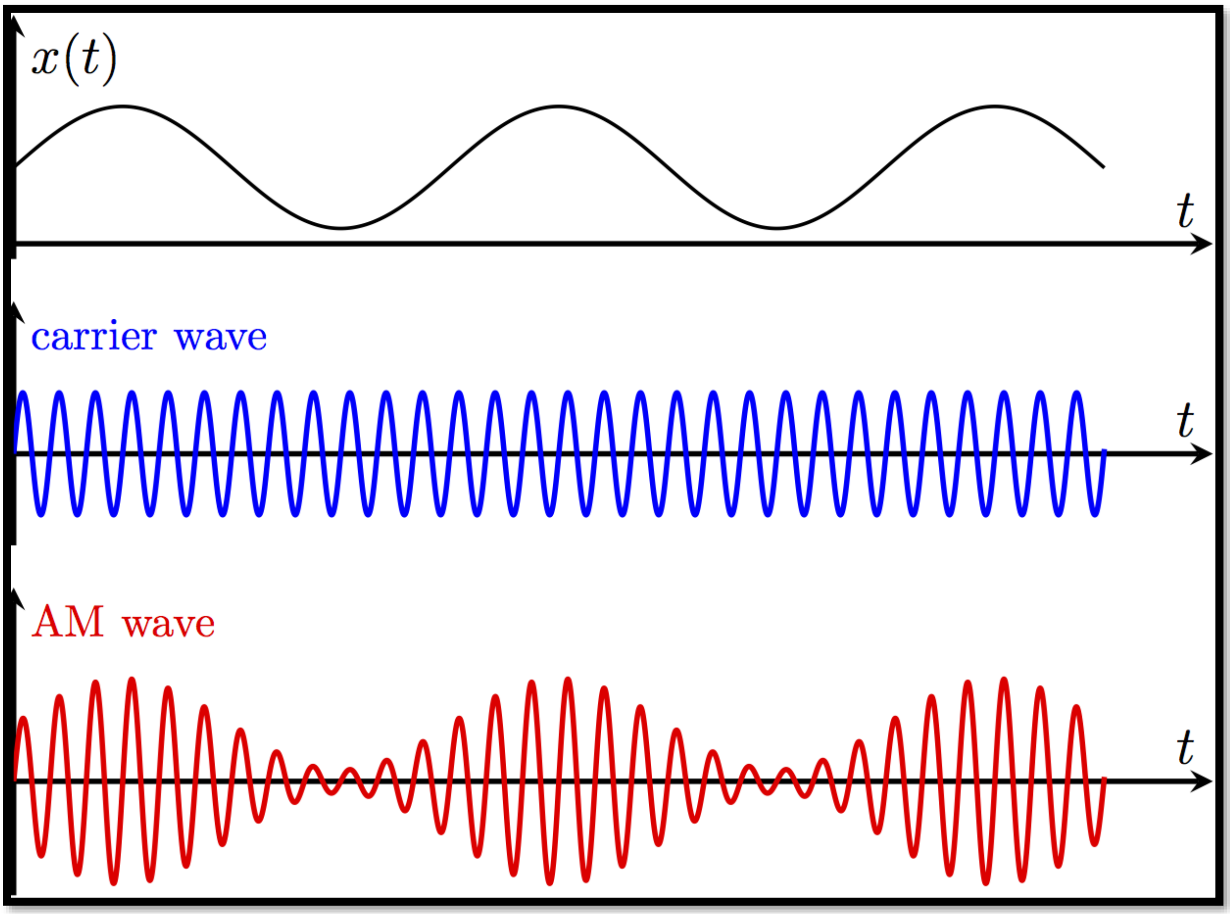
These waveforms can also be simulated in MATLAB to simulate the resulting synthesised waveform. All that would be required to do is to create a vector called ‘t’ for time and make the variable ‘y’ equal to

the formula for each FF and harmonic waveform, added or subtracted from each other. The result would then be plotted on a graph with t being the x axis and y being the y axis.
The simulated results have shown that the waveforms in the practical session can achieve a similar or accurate waveform but, would also require a certain amount of '*tuning*' to align all the signals in the right places due to the board being old and with failing components. The failing components would go on to then give a wrong phase shift at the specified rotation point. The need for the board to be calibrated has affected the experiment somewhat but, not enough to the point where there was no point to the practical in the first place. By comparing the simulated results to the practical ones, it could then be determined what part of the process would then be at fault. And it was determined that the phase shift control was the section of the board that had been causing the issues.

References:

1. https://en.wikipedia.org/wiki/Harmonic
2. Electronics-tutorials, 2020. Electronics-tutorialsws/accircuits/harmonicshtml. [Online].[02 April 2020]. Available from: https://www.electronics-tutorials.ws/accircuits/harmonics.html
3. Leshan Uggalla, L.U. 2020. UnilearnSouthwalesacuk. [Online].[02 April 2020]. Available from: https://unilearn.southwales.ac.uk/bbcswebdav/pid-3621915-dt-content-rid-4121281_1/courses/NG2S800_2019_v1/Lecture_2_NG2S800_2018_2019_Harmonics.pdf
4. Tracy V Wilson, T.V.W. 2020. Howstuffworkscom. [Online].[02 April 2020]. Available from: https://science.howstuffworks.com/hologram6.htm

Amplitude Modulation



[1]

Ben Edwards 18026826

Introduction:

For this lab session, a signal modulation training board, called the Emona Telecom Trainer 101, is used to create a modulated message signal using a carrier frequency. The purpose of this assignment is to study the stage of signal processing in the production of an AM (amplitude modulation) and ASK (amplitude shift keying) signal.

Using the 101 Telecom trainer kit, a series of modulated signal must be generated, displayed and compared to its original message signal pre-modulation. After completing this, a microphone input can be used to replace the message signal sine wave that will be used for the first modulation step. After this, the amplitude must be varied to observe how that will affect the modulated carrier. Finishing this, the modulation depth of the AM signal must be measured and recorded.

A series of equipment shall be used for the purpose of this experiment…

- Emona Telecom Trainer 101
- A Voltcraft DSO-1062D-VGA Oscilloscope
- Three Emona oscilloscope leads
- Assorted Emona patch leads.

Implementation:

Amplitude Modulation Sine Wave Signal with DC supply:

For this test, the variable DCV, Master Signals and the adder models on the Telecom trainer 101 board. Connecting VDC to the B input on the adder module and the 2KHz sine wave signal on the master signal module into the A input on the adder. The output of the adder would be connected to channel 1 on an oscilloscope. The VDC nob it turned almost completely clockwise, turn the g nob clockwise until the DC level increases by 1V and turn the G nob clockwise until a 1Vp-p sine wave is obtained. On the oscilloscope, the horizontal scale is set to 200us/div and the vertical scale is set to 500mV/div

Looking at the results for the output of the adder module, the difference that can be seen is that the DC offset of the signal has increased by under 0.5V, and the amplitude has increased up to a pk-pk voltage of 3.75V from the initial 1Vpk-pk. The frequency of the signal has stayed the same at the output of the adder module at 2KHz. This can be calculated by using the formula (1/Horizontal squares * time/div = f). this can be used to find the frequency by counting the squares on the time scale and multiplying it by the time division. This will give the formula of (1/2.4*200uS = 2083Hz). The frequency has been calculated because the oscilloscope has given a false reading of 4.7KHz. I believe this is due to the machine not being given enough time to calculate the frequency of the output before the screenshot could be taken.

Since the menu is obscuring the readings values as well, the voltage will also need to be calculated using a similar method. This method is as follows, (vertical blocks * V/div = Vout). So, the result calculated is (7.5*0.5V = 3.75Vpk-pk).

It should be noted that the frequency tends to stay the same throughout this process.

Amplitude Modulation Sine Wave Signal with 100KHz carrier:

Following the same process, the output of the adder is connected to the X DC input of the multiplier module and the 100KHz signal connected to the Y DC input. The output of the multiplier is then connected to the channel 2 of the oscilloscope.

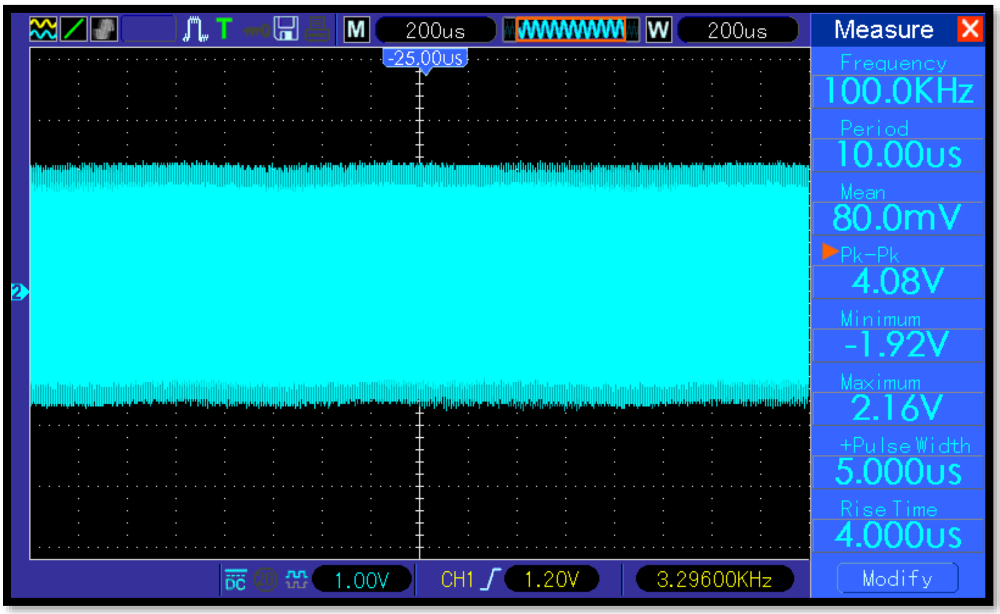


Fig 1.3

In fig 1.3, it can be shown that the voltage signal of the 100KHz carrier signal is at 4Vpk-pk.
Using this new setup, the results given can be seen in fig 1.4 and 1.5 below

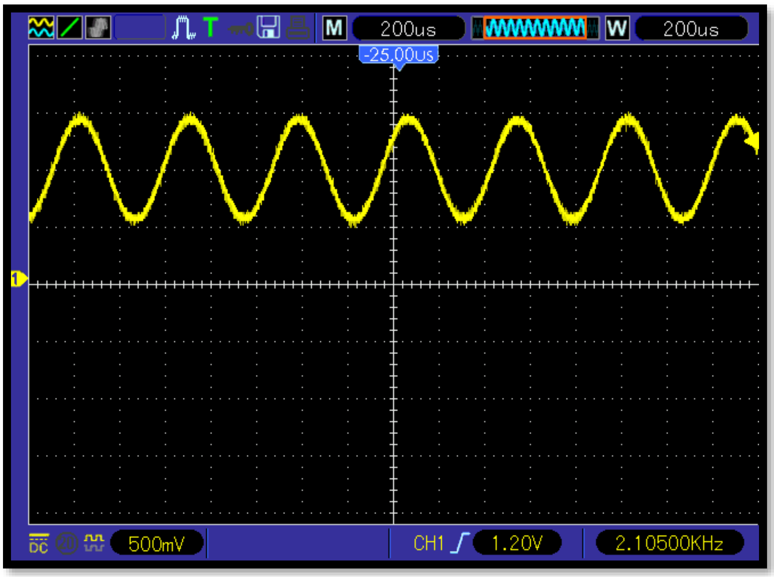


Fig 1.4

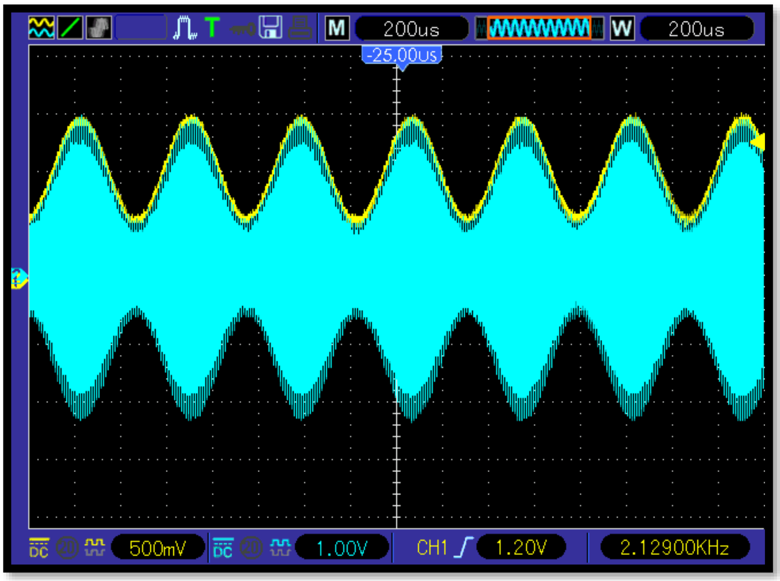


Fig 1.5

Comparing fig 1.3 and 1.5, it is shown that the carrier frequency has been multiplied with the message signal to create this new modulated waveform, taking a similar shape of the 2KHz message signal. This occurs because the amplitude of the message signal is then subtracted from the amplitude of the carrier signal. This will effectively leave an 'indentation' in the carrier signal which will then 'carry' the signal when transmitted.
The amplitude and shape of the carrier signal indicates that the amplitude of the message signal has been used to affect the amplitude of the carrier signal, so it can be ready for transmission.
For this modulated signal, to created it in this experiment, a total of two signals was used. The carrier signal of 100KHz modulated by the message signal of 2KHz
Below is the diagram of the training board layout in diagram 1.1.

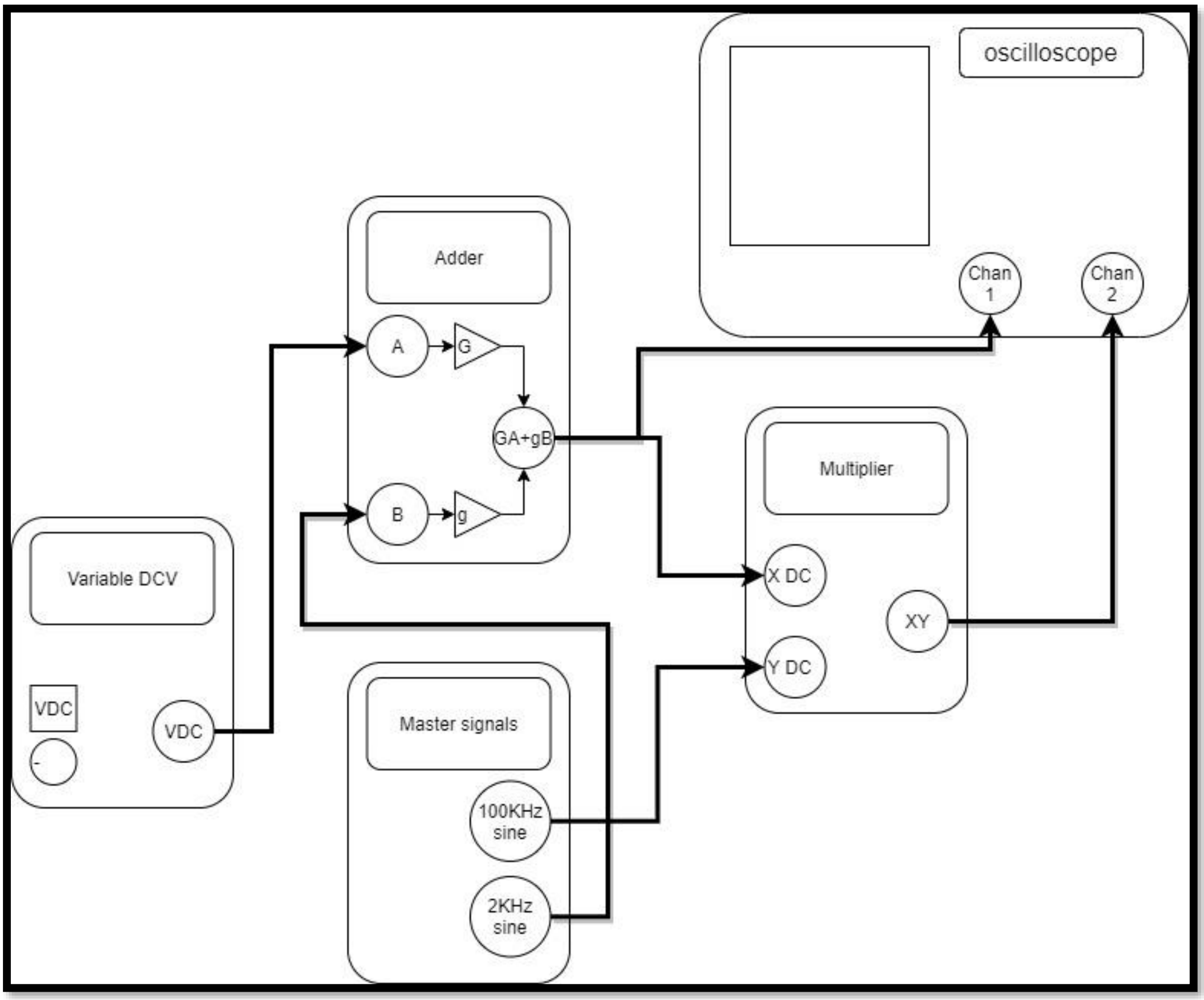


Diagram 1.1 (board arrangement for modulation)

Frequency Domain: -

For this procedure, the 2KHz sine signal is temporarily connected to channel 1. After turning off the channels 1 and 2 on the oscilloscope, the math button will be pressed to bring up the spectrum analyser mode. Pressing F1 to choose FFT, F3 to choose channel 1 and F4 to choose Hanning. After setting these options, the sec/Div nob is turned clockwise until the value 5KHz(50Ks/s) is showing on the screen. Pressing F6 to move onto the second page, page 2, of the scope. F5 is used to choose Vrms and F4 to increase the zoom by times 10 before using the horizontal positioning to bring the display to the centre of the screen. The channel 1 volt/div is set to 1V and then the curser button is pressed. Use F1 to choose time, F4 to choose S (start) and move that start position using the V0 knob to move that point as close to 0 as humanly possible. Using F4 again will choose the E (end) position point, moving it as close to the beginning of the peak as possible.
The dt value given on the oscilloscope was **2.023KHz**.
Pressing F1 again will change the type from time to voltage so now the value of dv can be given. F4 can be used again to move the S pointer to as close to 0V as possible, and again to move E as close to the beginning of the peak as possible.
The value of dv is given as 1.2V
To convert dv to Dbv, the formula **20log(dv) must be used. So, 20log(1.2) = 1.58 DbV**
This can be seen in the figures 1.6 and 1.7.

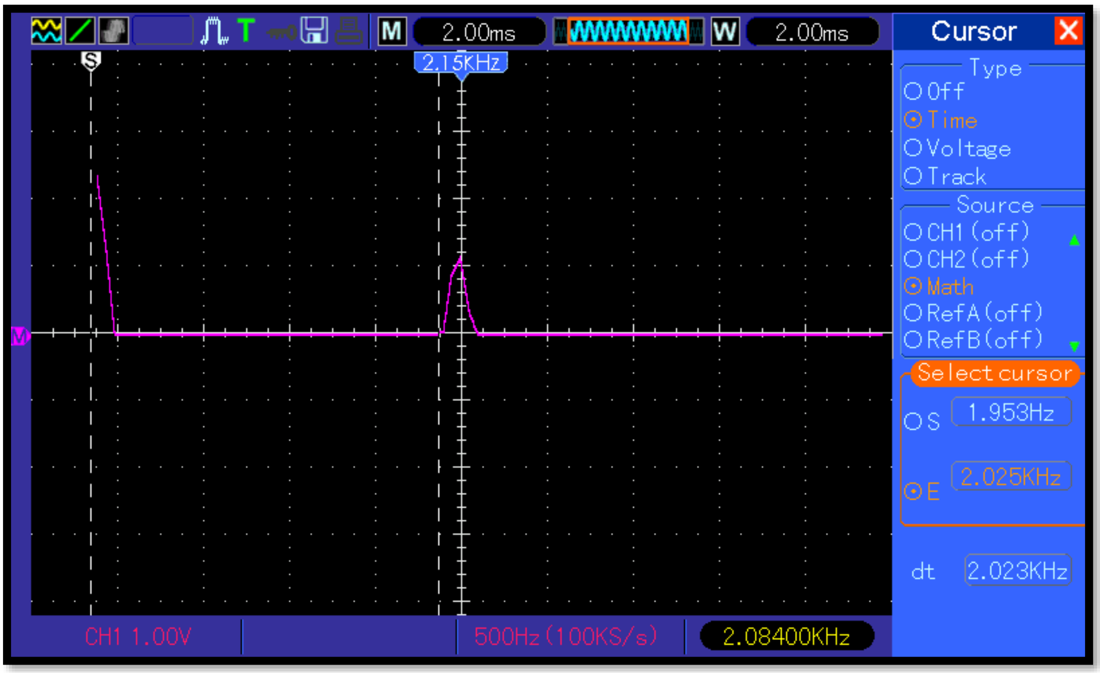


Fig 1.6

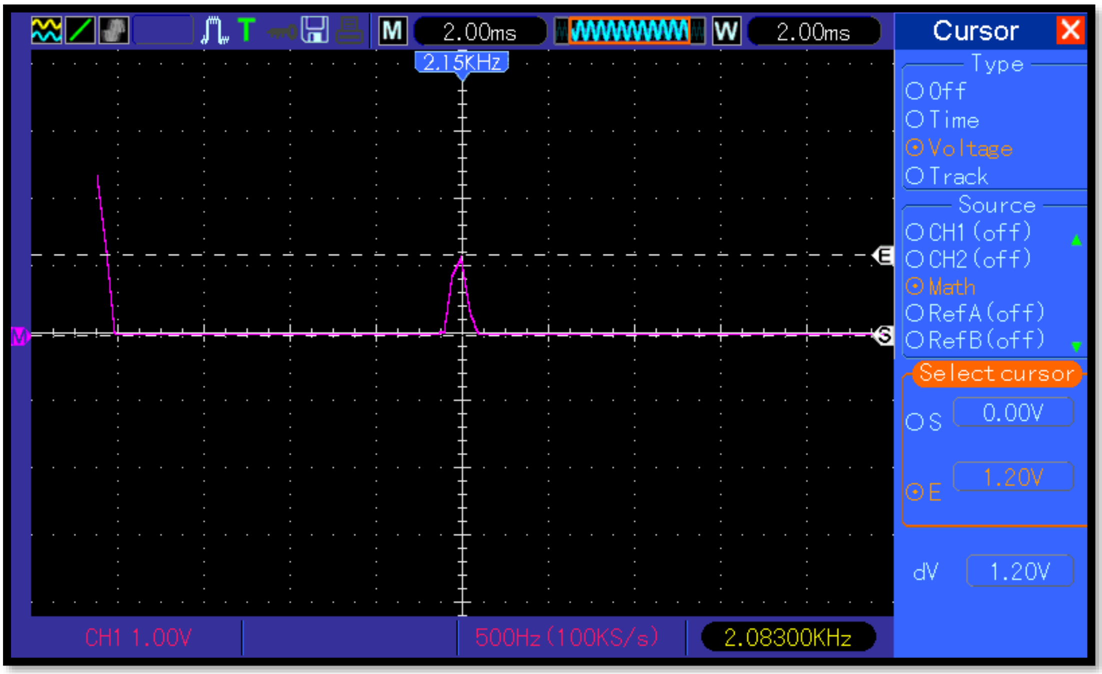


Fig 1.7

The above procedure is used again with some simple changes. By using the F3 button to change the source to channel 2 and changing the sample rate to a value of 125KHz(500KSa/s), and the value of v/div is set to 200Mv/div.

For the output of the modulation signal, after positioning the curser of the two side bands, the frequency could then be measured using the oscilloscope. The Lower sideband was 98.05KHz and the Upper sideband was 102.2KHz. this can be seen in the Figure 1.8

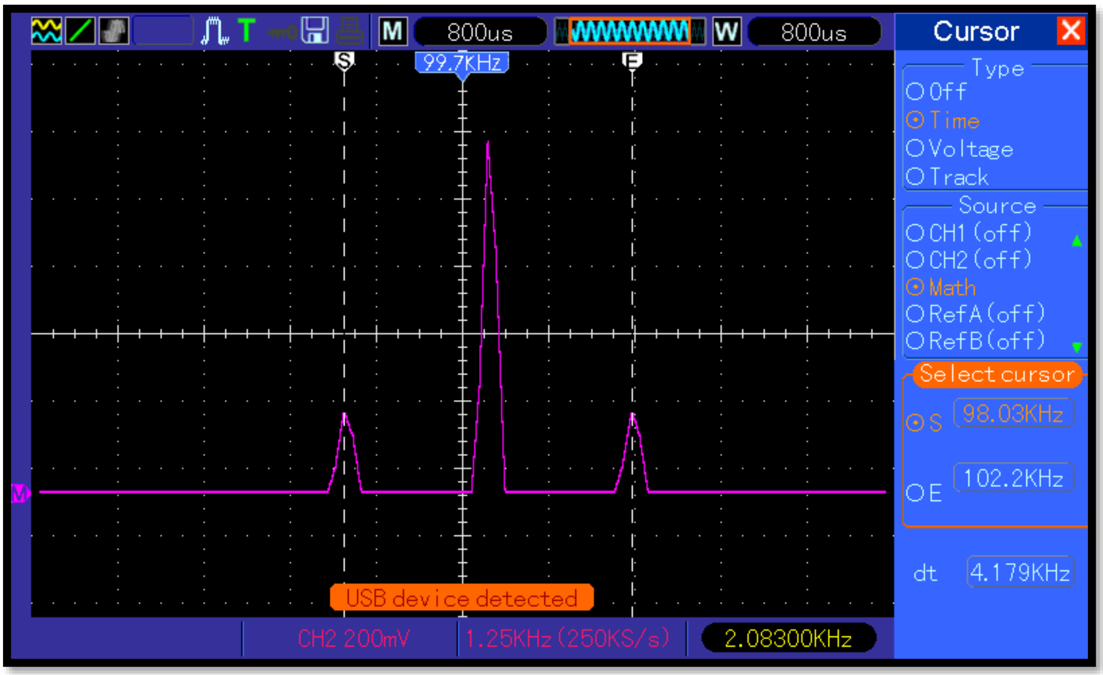


Fig 1.8

Positioning the S curser for voltage close to 0V, and then the E curser close to the peak of the sidebands as is possible will show what the amplitude of the waveform is. This can be shown in Figure 1.9

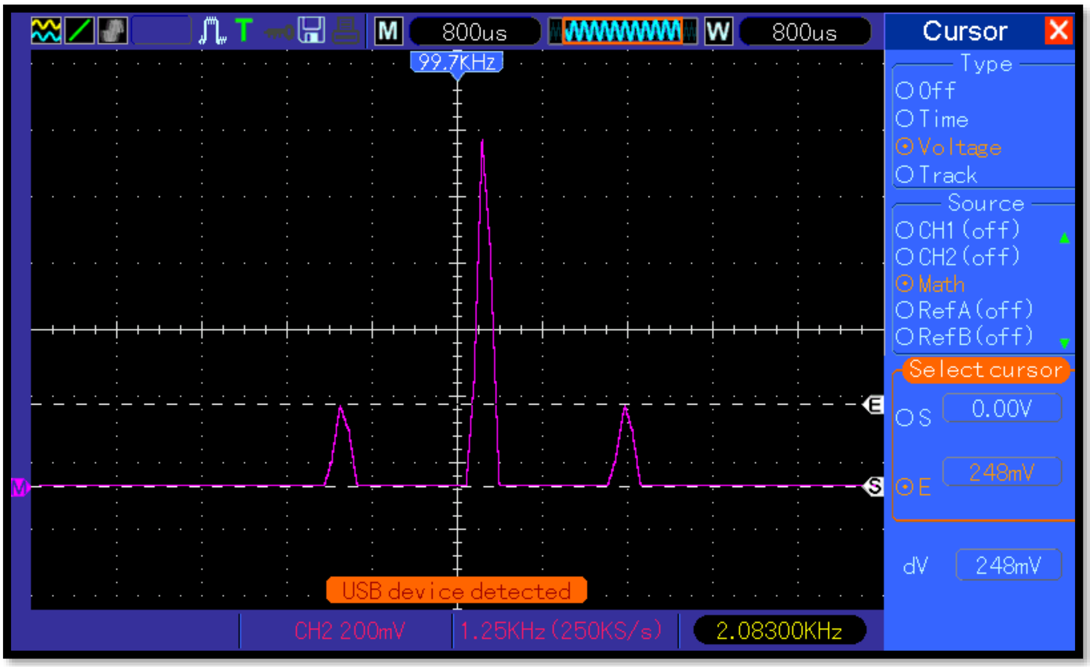


Fig 1.9

Looking at the figures 1.8 and 1.9, in terms of frequency and amplitude, it can be shown that the distance between the side bands is measured in dt from figure 1.8. the total distance is equal to double the frequency of the message signal. So, the bandwidth and distance of the side bands from the message signal frequency, is determined by the frequency of the message signal that is being modulated with the carrier frequency. Figure 1.9 shows the amplitude of the sidebands are at 248mV
The efficiency of an AM signal can depend on the modulation quality of the modulated signal. When the signal is under modulated then the efficiency of the signal is at its worst due to an amplitude being higher than necessary. When the modulation is at its most efficient is when the value of m is equal to 1/when the modulation is at 100%. Any higher than this however, and the signal would start to experience overmodulation resulting in crossover distortion. Crossover distortion would result in a loss of information from the original signal.

Generating an AM signal using speech:

For Part B of this assignment, speech will be captured using a microphone and applied to the signal modulation circuit, replacing the 2KHz message frequency. The output of the microphone will now become the message frequency intended for AM modulation.
First disconnecting the plug on the master signal module's 2KHz output connection and swapping that connection to the speech output module. The output of the speech module would then be connected to the B input of the adder module. On the oscilloscope, the horizontal scale is set to 0.2ms/div.
After completing these steps, the noise that gets picket up by the microphone will be converted into an electrical signal and modulated by the circuit. As if it would be ready to be transmitted as an AM signal. This can be shown in figures 1.10 and 1.11 Fig 1.10

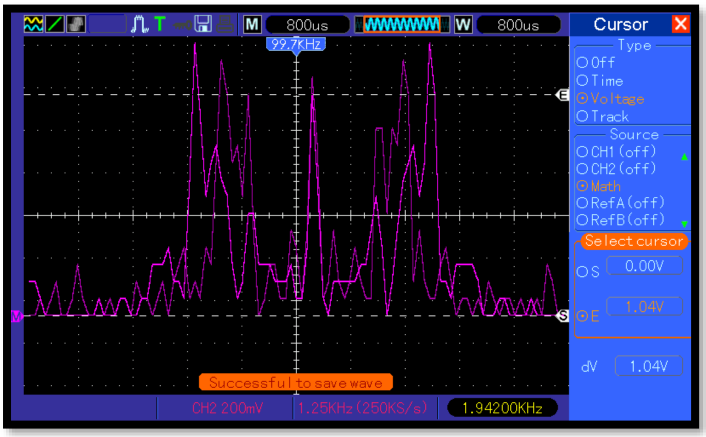


Fig 1.10

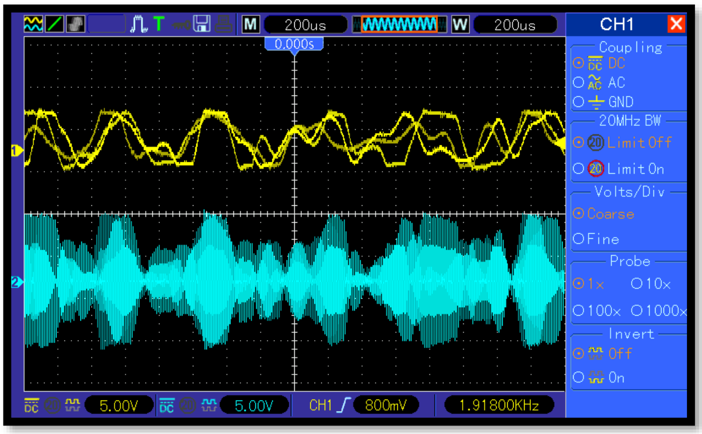


Fig 1.11

The circuit will still be modulating a signal however because the microphone will still be picking up noise from the surrounding area while the microphone is still being powered.
This results in a message signal still being generated and modulated with the carrier signal.

Investigating depth of modulation:

The depth of the modulated carrier can be calculated using the modulation Index, or modulation depth, formula is known as (**m = Vmax - Vmin/Vmax + Vmin**) **[2]**. This can be achieved using P and Q as the maximum and minimum peak to peak values of the waveform.
To set up this experiment, the scope must be set to the time domain by disabling the 'math' operation. The speech module must then be disconnected and the 2KHz from the master signal module must be reconnected to the adder module. If the G control is varied, then the modulation depth of the signal is also affected. This can increase the depth of modulation or reduce it to improve the efficiency of the circuit (reference). The P and Q values of the AM signal will be recorded and used to calculate the modulation depth using the formula specified above. The values that were recorded were **P = 3.2V and Q = 280mV.** The calculation preformed was…
(m = 3.2-280*10^-3/3.2+280*10^-3 = 0.889V or 889mV)
after next turning the G control to 1 third of its rotation, it is seen that when the value of m is more than 1 then the signal becomes overmodulated and when the value of m is less than 1, the signal becomes under modulated. **[2]**
Figures 1.12 and 1.13 shows an over-modulated signal when the value of m is more than 1.
This can be seen in the figures 1.12 and 1.13

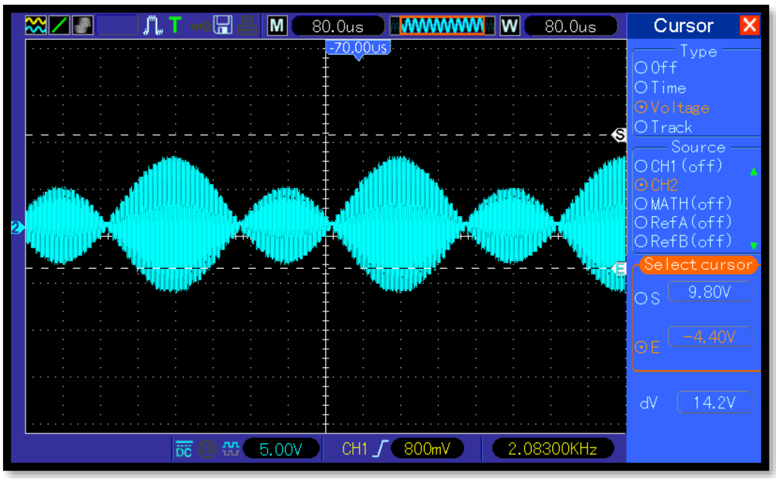


Figure 1.12 (m > 1)

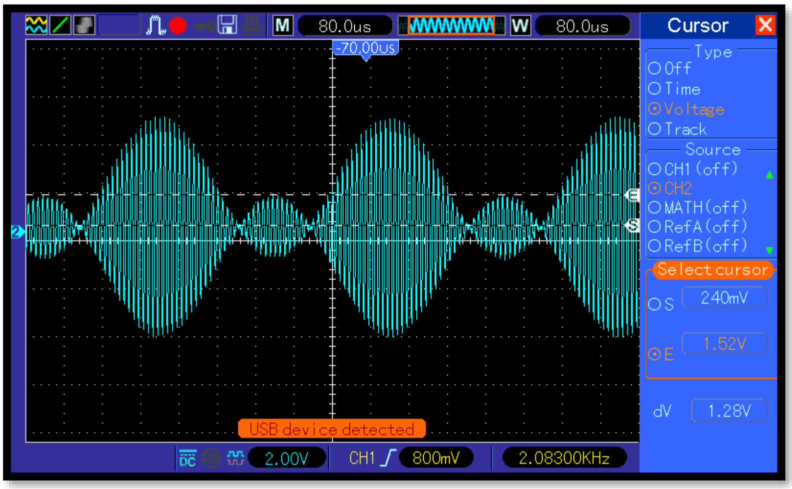


Figure 1.13 (m > 1)

To calculate the modulation depth from the examples above, the calculation can be shown in the table below.

Condition	Vmax	Vmin	Modulation depth, m
As in step 5	3.4V	0.2V	m = (3.4 - 0.2)/(3.4 + 0.2) = 0.8
As in step 8	4.72V	-1.52V	m = (4.72 - -1.52)/(4.72 + -1.52) = 1.95

Table 1.1

When an AM signal is overmodulated, then that will produce a crossover effect that will result in a loss of information being transmitted **[2]**. This is the least ideal situation due to the fact that a distorted signal is then being received by the antenna with a large loss of information.

Amplitude Key Shifting: -

This works similarly to the modulation, but instead uses digital data instead of an AC signal. The digital and carrier signal is then modulated to produce the ASK signal, ready for transmission. The upper and lower limits is considered the envelope of the ASK signal. This envelope follows the shape of the digital data stream, with the lower envelope inverted, allowing for a process that can be employed to recover the original data by tracking the envelope of the carrier.

($S_{ask}(t) = m(t)\cos \omega_c t$)

M(t) in this formula can either be a 1 or a 0. This can be shown in the oscilloscope screenshot below of this effect. This effect is called on-off keying or (OOK) as the binary value for m(t) determines when the

carrier frequency is effectively turned on and off. This represents the binary values of the digital signal. **[3]**
To set up this experiment, the scopes trigger source is set to external by pressing the button 'Trig menu'. press F1 to choose the edge. The scope is then set to dual display mod e by pressing button 1 and 2 so they are both lit up in the analogue section. Channel 1 and channel two are pressed so they are both set to DC coupling. The scope's horizontal scale is set to 800ns/div. the vertical scale for channel 1 is set to 2V/div and channel 2 is set to 1V/div.

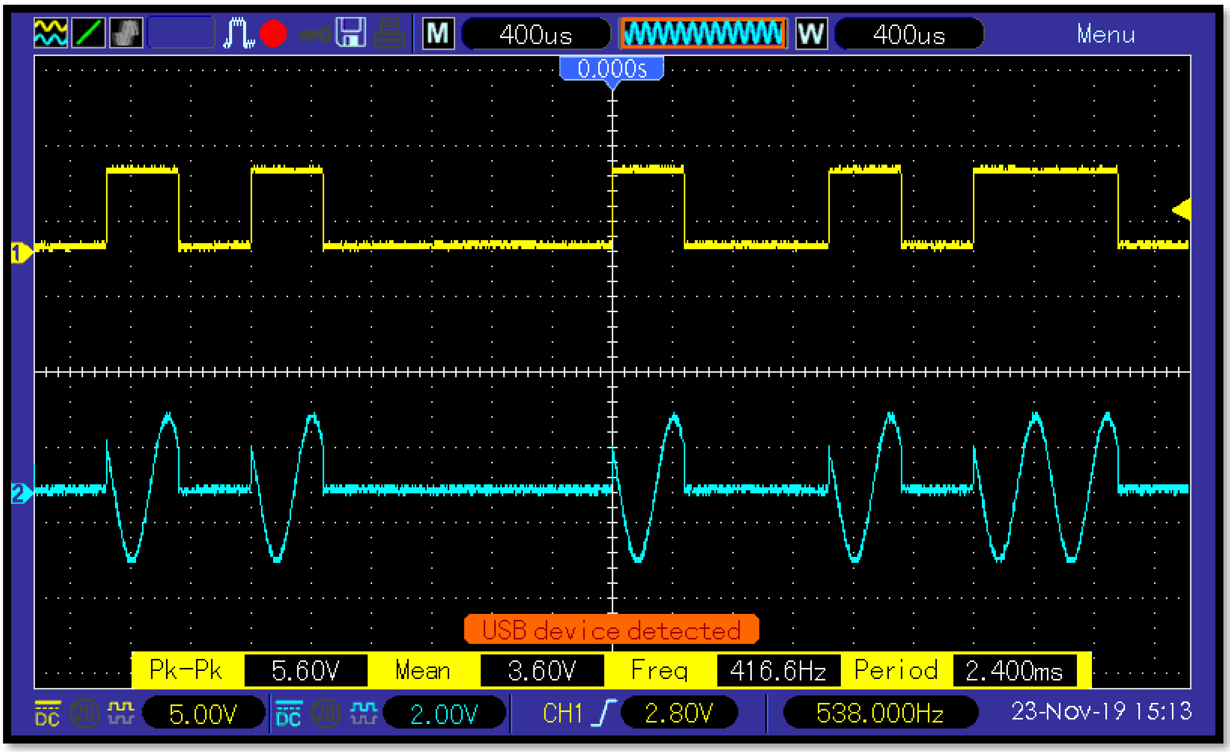


Fig 1.14

Looking at this screenshot the mirror of the signal can be seen in the ASK signal waveform with the original data signal being shown above it in the second channel.

Conclusion: -

In conclusion, the AM modulation method will work by using the amplitude of the message signal and subtracting that from the carrier signal in the time domain. This will allow the signal to be transmitted along long distances.
In the frequency domain however, the oscilloscope can record the frequency and amplitude voltage of the message signal and the modulated signal. The frequency and amplitude voltage of the signal can be recorded in the dt and dv values by measuring the distance between two points. For the screenshot in Figures 1.6-1.7 and figures 1.8-1.9 shows the values for the message signal and the modulated signal by measuring the distance between two points. The figures 1.6-1.7 shows the frequency and amplitude of the message signal when finding the values for dt and dv, while fig 1.8-1.9 shows the frequency domain for the modulated signal and its respective side bands. The side bandwidth for this signal can be measured this way by measuring the peak of both upper and lower sidebands to see the total bandwidth frequency. The amplitude would be measured as expected with one node at 0V and the other node at the peak amplitude of the side bands.
It has also been discovered that when a microphone has been applied to the input of the adder module instead of the 2KHz signal message being displayed, a signal would be generated from the soundwaves being picked up by the microphone. Because of this, the device would be constantly modulating a signal because the microphone would be able to pick up noise from the background.

And finally, in the amplitude shift keying, the resulting output would be a binary representation 1 or 0 depending on the amplitude of the ASK signal. This also means that there is no risk of overmodulation due to the signal being a digital signal, not relying on the varying amplitude signal of an AC signal.

Referencing:

1. Petar Veličković, P.V. 2016. Com/PetarV-/TikZ/tree/master/Amplitude%20modulation. [Online].[02 April 2020]. Available from: https://github.com/PetarV-/TikZ/tree/master/Amplitude%20modulation
2. Asif Shaik, A.S. 2020. Physics-and-radio-electronicscom. [Online].[02 April 2020]. Available from: https://www.physics-and-radio-electronics.com/blog/amplitude-modulation/
3. Tutorialspointcom, 2020. Tutorialspointcom. [Online].[02 April 2020]. Available from: https://www.tutorialspoint.com/digital_communication/digital_communication_amplitude_shift_keying.htm

Frequency Modulation

[3]

Ben Edwards 18026826

Introduction: -

For this assignment there are two sections. the first section is to be able to generate a real FM signal using an Emona Telecoms trainer 101 board using the VCO module. Digital square wave and speech signals will be frequency modulated, observing the effects this has on both message sources.
To demonstrate the effects amplitude has on the FM modulator, signal messages with varying amplitudes shall be put through the modulator system, observing the effects that this has on the resulting output waveform.
A sine wave will be used to observe the spectral composition of the FM signal in the Time and

Frequency domains. The modulation will also be used to monitor the effect that that will have on the modulator also.
The second part of the assignment will involve applying a digital input message to generate an FSK signal using the VCO module on the telecom's trainer 101 board. The type of FSK that will be used is called Binary FSK, which is the simplest method of FSK and is suitable for this application.

For this assignment, the following equipment will be used in conjunction with the…

- Telecoms-Trainer 101 board
- Digital storage Oscilloscope 60MHz (Voltcraft DSO-1062D-VGA)
- Three Emona oscilloscope leads
- And assorted Emona patch leads.

Main body:

Frequency Modulating a square wave:

For this procedure, the VCO module on the telecoms board should be located and the *Gain* control adjusted clockwise to about two thirds of its total travel.
The VCO modules *Frequency adjust* knob control should be adjusted to the middle of its travel.
The VCO's *range control* is set to the LO position before connecting the *SINE* output on the module to channel 1.
On the oscilloscope, the horizontal scale is adjusted to 40us/div using the SEC/DIV knob control in the vertical section. The VOLTS/DIV knob is then set on the scope to a Vertical scale of 1V/div.
Adjusting the *Frequency adjust* control on the VCO so that one cycle of its output is exactly 2 divisions (two squares on the x axis), the waveform will be displayed allowing for the frequency and the amplitude of the signal to be calculated and recorded.

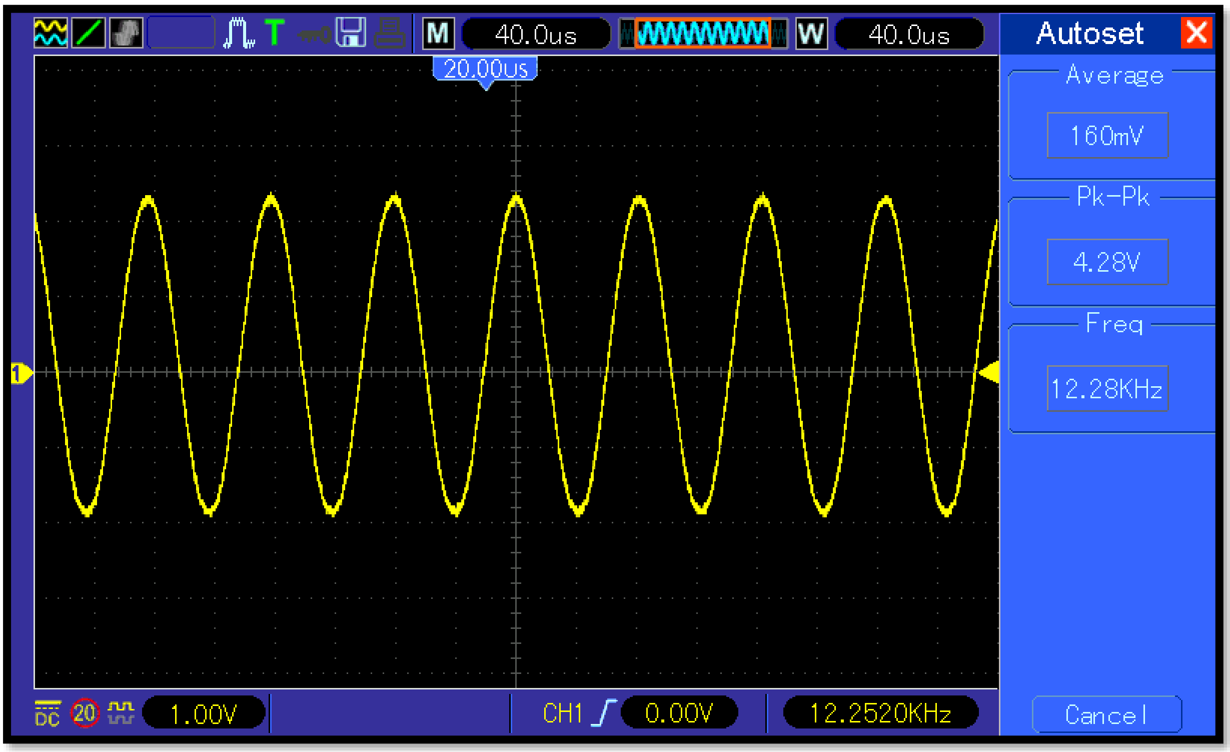


Figure 1.1

The calculated values are…
(Ac = 4.28/2 = 2.14Vpk)
(Fc = 12.25KHz)

Using the SEC/DIV knob in the horizontal section, the scopes horizontal scale is set to the value of 200us/div.
Modifying the setup so, that the 2KHz digital signal in the *master signal* module is connected to channel 1, switching places with the VCO output connection into channel 2, and the VCO input.
The scopes mode is set to dual by pressing channel button'1' and '2' so that both buttons are lit.
Channel 2's vertical scale is set in the vertical section to 1V/div.
The '*vertical position*' control is finally adjusted to overlay both channels to the message with the obtained FM signal. Pressing the Run/Stop button will allow for a snapshot of the oscilloscope display for an easier picture.
The result can be seen below in figure 1.2…

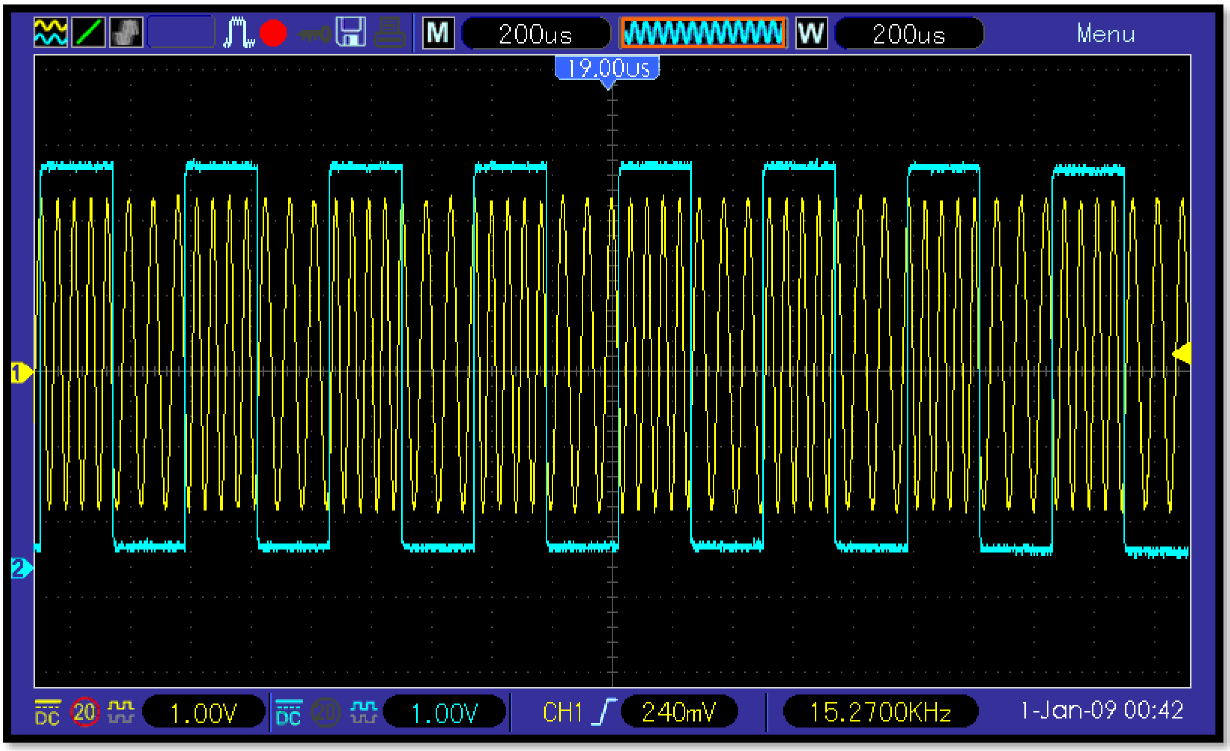


Figure 1.2

Looking at these results, it can be seen that the frequency of the carrier signal changes because it is being modulated with the 2KHz data signal. Using the amplitude of the data signal to change the frequency of the carrier signal during the alternating period.

Generating an FM signal using speech:

Now there will be an experiment for conducting the modulation using the signal generated from a microphone available on the telecoms board. This will allow for an accurate visual test of how an audio signal would be modulated before being transmitted via a radio tower.
The first step is to disconnect the plug to the '*Master signal*' module's '*2KHz Digital*' output, connecting it to the '*speech*' module's output connection.
The SEC/DIV knob in the horizontal section is set to a value of 200us/div on the horizontal scale.
The results from a quiet whistle and a loud whistle can be seen below in figures 1.3 - 4.

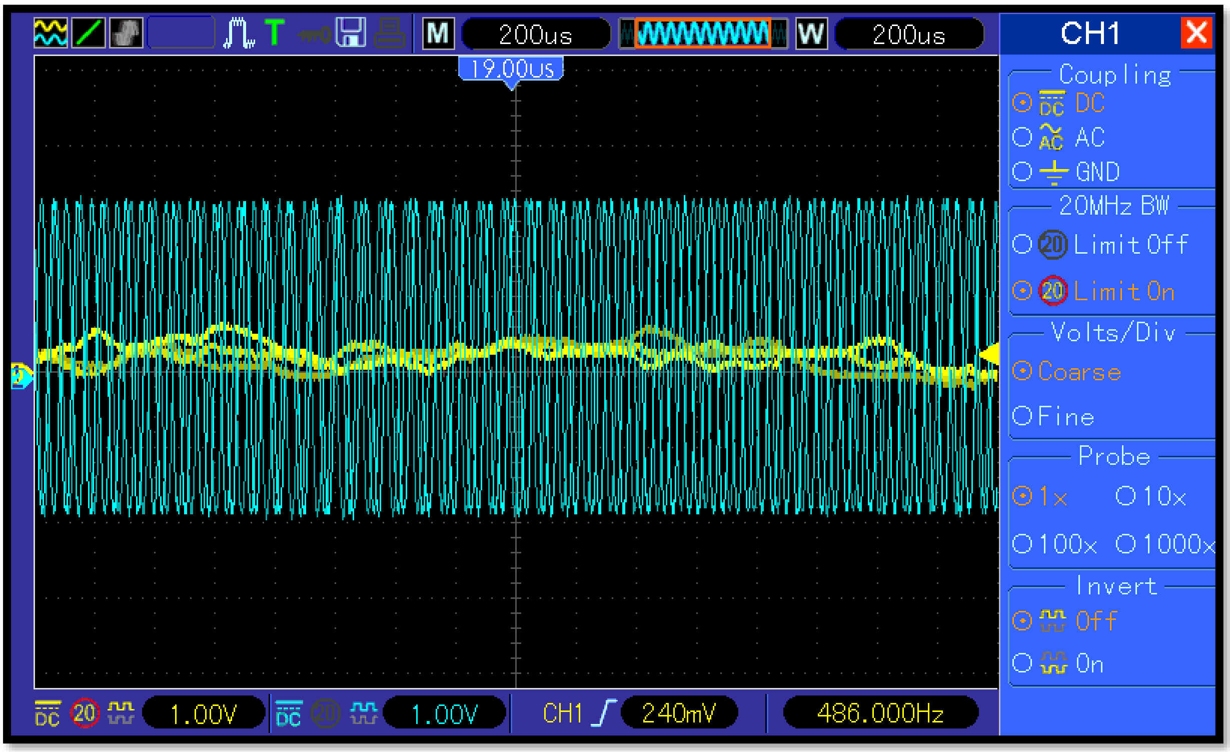


Figure 1.3 (quiet whistle)

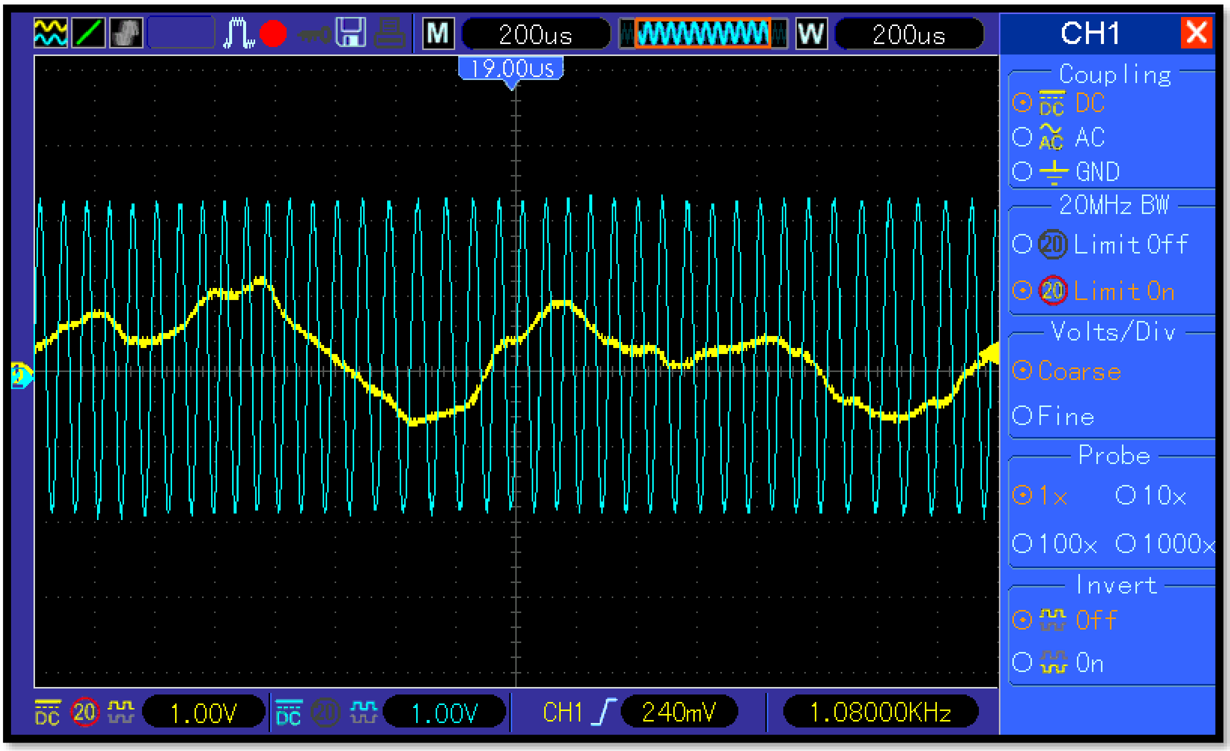


Figure 1.4 (loud whistle)

Looking at these results, it is clear to see that the amplitude of the message signal emitted from the microphone output, changes the frequency of the carrier signal when being modulated. The example shows that, in this scenario, there is a section of the modulated signal where the frequency is low when the amplitude of the message signal is low.

Spectral Composition of FM Signals:

For this task, the spectrum of the FM signal will be observed and demonstrated in the time and frequency domains.

First Display channel 2 making sure that the FM signal is the only signal being viewed.
Plugging the 2KHz signal back in to the VCO input and channel 1 on the oscilloscope.
Pressing trigger menu allows the user to press the F1 key to set '*Trigger type*' to '*Edge*', and F2 to set '*source*' to '*CH2*'
The SEC/DIV knob in the horizontal scale is set to a time base of 20us/div. the display wil now display what is known as the spectral composition of the FM signal. This can be seen in figure 1.5 below.

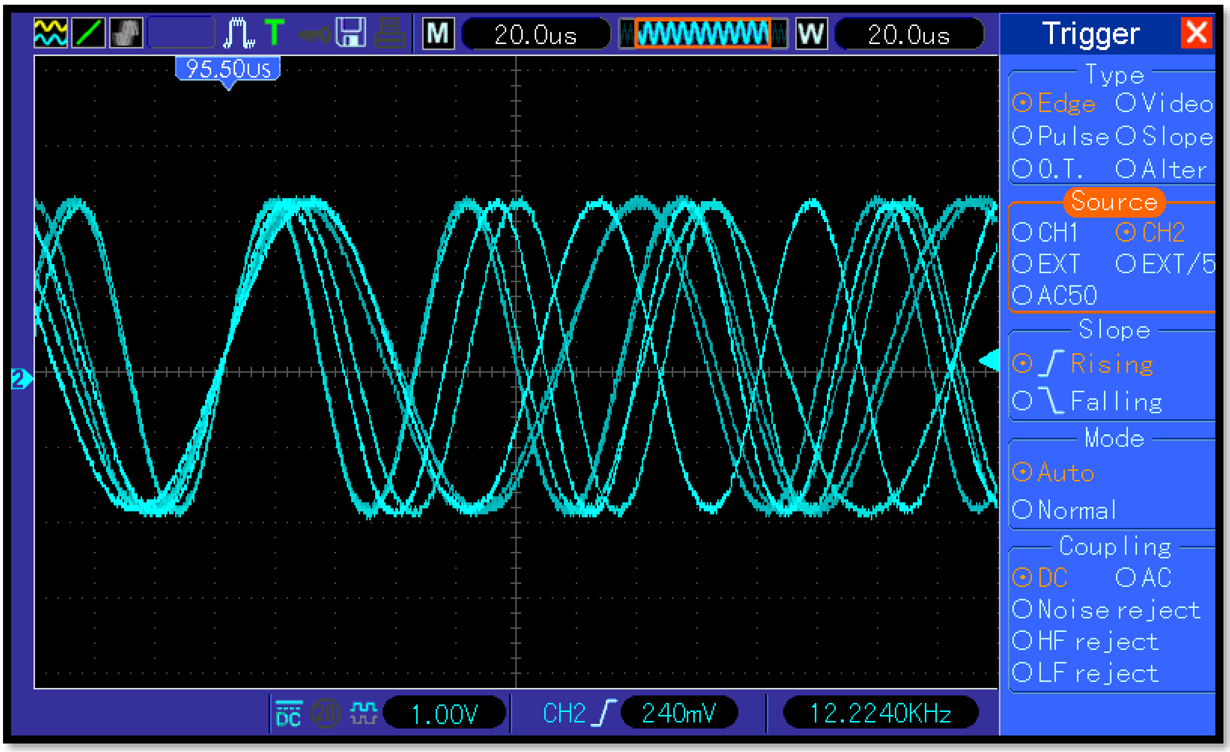


Fig 1.5 (FM Spectral composition)

Turning the gain control on the VCO module can be adjusted anti-clockwise for this result to be displayed.
In the results shown above, it can be seen that the side bands of the FM signal can be seen in edge mode. These side bands are a product of the frequency values ***fc±fm, fc±2fm, fc±3fm, fc±4fm***, ect. In this scenario ***fc*** = the carrier frequency and ***fm*** is equal to the frequency of the message signal. The amplitudes of the sidebands and how many of them are significant enough to be calculated are dependant on one value. That value is the modulation index (β) of the FM signal. The index value is defined as the ratio between the frequency deviation ***fd*** and message frequency ***fm***. The index value can be calculated by using the formula…

β = fd/fm

and the value of fd is equal to…

fd = Kf*Am

(where 'Kf' = is the frequency sensitivity of the modulator AND 'Am' is = to the maximum absolute amplitude value of the message signal.)

fd is the maximum value that the carrier frequency deviates from the unmodulated value of fc and is calculated using the formula above. This is showing the *Kf* of the modulator, being the change in fc per volt of modulating signal expressed in the form of Hz/V, and the absolute value of Am from the message signal. And so, both values are then combined together in a formula to calculate the value of ***fd***.
The image below shows an example of an FM signal spectrum.

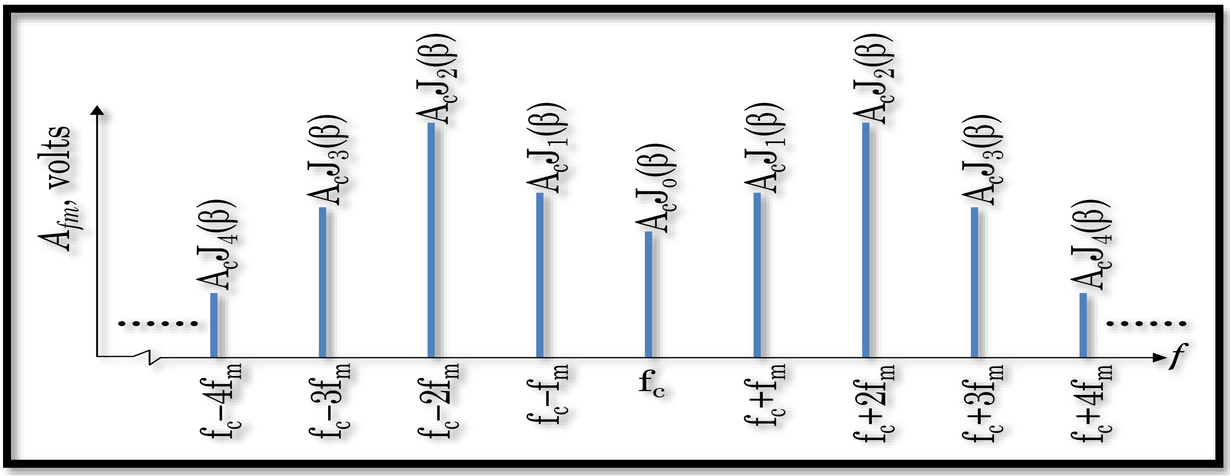


Figure 1.6 (FM signal spectrum diagram)

β	$J_0(\beta)$	$J_1(\beta)$	$J_2(\beta)$	$J_3(\beta)$	$J_4(\beta)$	$J_5(\beta)$	$J_6(\beta)$	$J_7(\beta)$	$J_8(\beta)$
0.01	1.0	0.005							
0.2	0.99	0.1							
0.5	0.94	0.24	0.03						
1	0.77	0.44	0.11	0.02					
2	0.22	0.58	0.35	0.13	0.03				
3	-0.26	0.34	0.49	0.31	0.13	0.04	0.01		
4	-0.4	-0.7	0.36	0.43	0.28	0.13	0.05	0.02	
5	-0.18	-0.33	0.05	0.37	0.39	0.26	0.13	0.05	0.02

Table 1.1 (Bessel Table)

To Theoretically find the amplitude and the number of sidebands of the FM signal, a table known as the '*Bessel Table'* is used to find these values depending on the calculated value of β. A higher value of β the higher number of side frequencies there are. The Bessel Table will be used later to calculate the theoretical sidebands for the signal recorded. **[1]**

To view the FM signal in the frequency domain, channel 1 and channel 2 are deactivated in the vertical section before pressing the 'math' button to be able to switch into the spectrum analyser mode. Pressing F3 will set the '*source*' to '*ch2*'. F4 will set '*windows*' to *'Hanning*'. F6 will then allow for the use of the F$ and F5 buttons to be used to set the '*FFT zoom'* to '2' and also to set '*Vertical base*' to '*dBrms*'. Adjusting the SEC/DIV knob in the horizontal section until the FFT Horizontal scale in freq per division, is 2.50KHz and the sample rate at 100KS/s.

The VOLTS/DIV knob in the horizontal section is then set to a scale of 10dBV/div.

The position knob is then used to set the '*offset*' value to 3.00div.

The value of Fc is then set to 10KHz using the '*frequency adjust control*' on the module. To do this first the input of the VCO is unplugged to just show the carrier frequency on the oscilloscope, and then after adjusting the control of the VCO module, the frequency of the carrier signal should then be 10KHz. The amplitude of the carrier, *Ac*, was then recorded as…

Ac = 67.2dBV

Pressing '*cursers*' in the menu section allows the user to be able to press F1 to switch between '*Frequency*' and '*Magnitude*', which can be adjusted with the V0 knob control, which will move the selected cursor along the frequency or amplitude of the displayed signal. This can be used to read the frequency and amplitude of each sideband.

The VCO input is then connected back again to measure the frequency and amplitude of the centre component and three sidebands by adjusting the frequency and amplitude cursors. Before taking any results however, the FM amplitude centre frequency is adjusted using the gain control to be between a value of 28.4dBV and 29.2dBV.

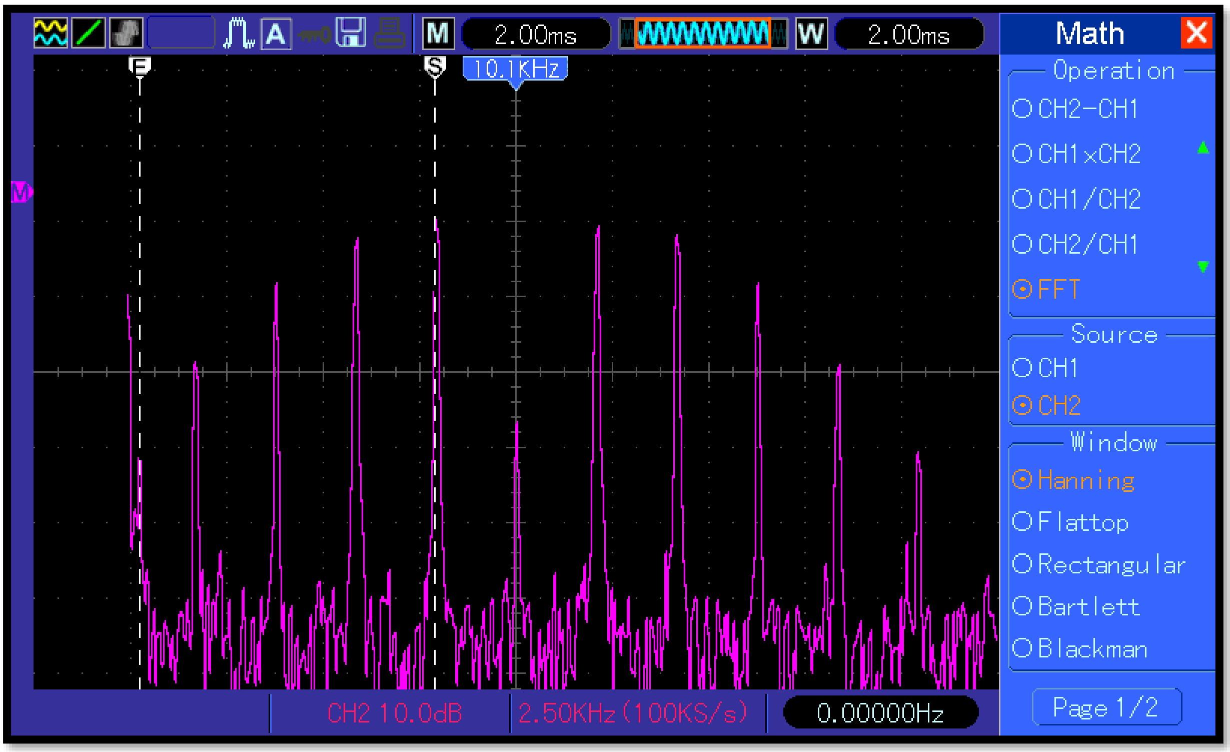


Figure 1.7 (FM spectrum)

The amplitude of the signal is measured in dBV shown as Vdbv which can be converted to a Vrms form using the following formula…

Vrms = 10^(Vdbv/20)

Using the measured results from the practical, they can then be compared to the results from the calculation below. The recorded results for J0, J1, J2 and J3 are as follows…

J0 = 24.4dB
J1 = 23.8dB
J2 = 18.4dB
J3 = 7.5dB

The calculations are as shown below…

Fc = 10KHz
Ac = 2.14Vpk
Fm = 2KHz
Am = 2Vpk
F = 9.62KHz + 7.05KHz = 16.7KHz
Kf = 7.05/5V = 1.41KHz/V
Fd = 1.41*2 = 2.82KHz
β = 2.82/2 = 1.41

When β = 1
J0 = carrier freq
If J0 = 0.77

Without input ➔ Fc = 31.7dB ***Vrms = 10^(31.7/20) = 38.5V*0.77 = 29.6V*** ****Theoretical****
Fc(J0) = 24.4dB ***Vrms = 10^(24.4/20) = 16.5V *Practical****

Fc+Fm(J1) = 23.8dB ***Vrms = 10^(23.8/20) = 15.49V*** ****Practical****
38.5*0.44 = 16.94V ****Theoretical****

Fc+Fm(J2) = 18.4dB ***Vrms = 10^(18.4/20) = 8.3V*** ****Practical****
38.5*0.11 = 4.24V ****Theoretical****

Fc+Fm(J3) = 7.5dB ***Vrms = 10^(7.5/20) = 2.37V*** ****Practical****
38.5*0.02 = 770mV ****Theoretical****

β = 1	Theoretical	Practical
J0	29.6V	16.5V
J1	16.94V	15.49V
J2	4.24V	8.3V
J3	770mV	2.37V

Table 1.2 (Theoretical and Practical results)

It Is also possible to calculate the '*Bandwidth'* of the FM signal using the following formula below…

B = 2(β + 1)*Fm

B = 2(1.41 + 1)*2KHz = 9.6KHz

The advantages and disadvantages of FM signals are seen as, FM being a more volatile method when being used at higher frequencies due to higher frequencies struggling to travel further distances when compared to the AM method, which is more stable at higher frequencies. However, the FM signal has a much higher signal quality compared to the AM signal due to the AM signal being more susceptible to accumulating noise in the form of amplitude during transmission, which would affect the message signal when the AM signal eventually becomes retrieved and demodulated. The AM demodulated signal would have to be filtered through a number of low pass filters to be able to make up for the noise level. FM on the other hand, will be able to minimise the signal noise due to the modulated signal being focussed on the frequency of the carrier frequency being altered depending on the amplitude of the message signal. FM also has a higher bandwidth compared to AM allowing for the inclusion of more frequency ranges, giving a higher quality signal as a result. **[2]**

The FM signal spectrum has a more complex design compared to the AM modulated signal because FM has a varying number of sidebands dependant on the value of the modulation index of the FM signal **[2]**. AM however, has only two sidebands in the example below because there is only the modulation of the amplitude of the message signal. The FM method has multiple sidebands because the harmonics can be described as an echoe in the modulation, leading to the extra sidebands being used. The distance between these harmonic sidebands are spaced out by a distance of the message signal.

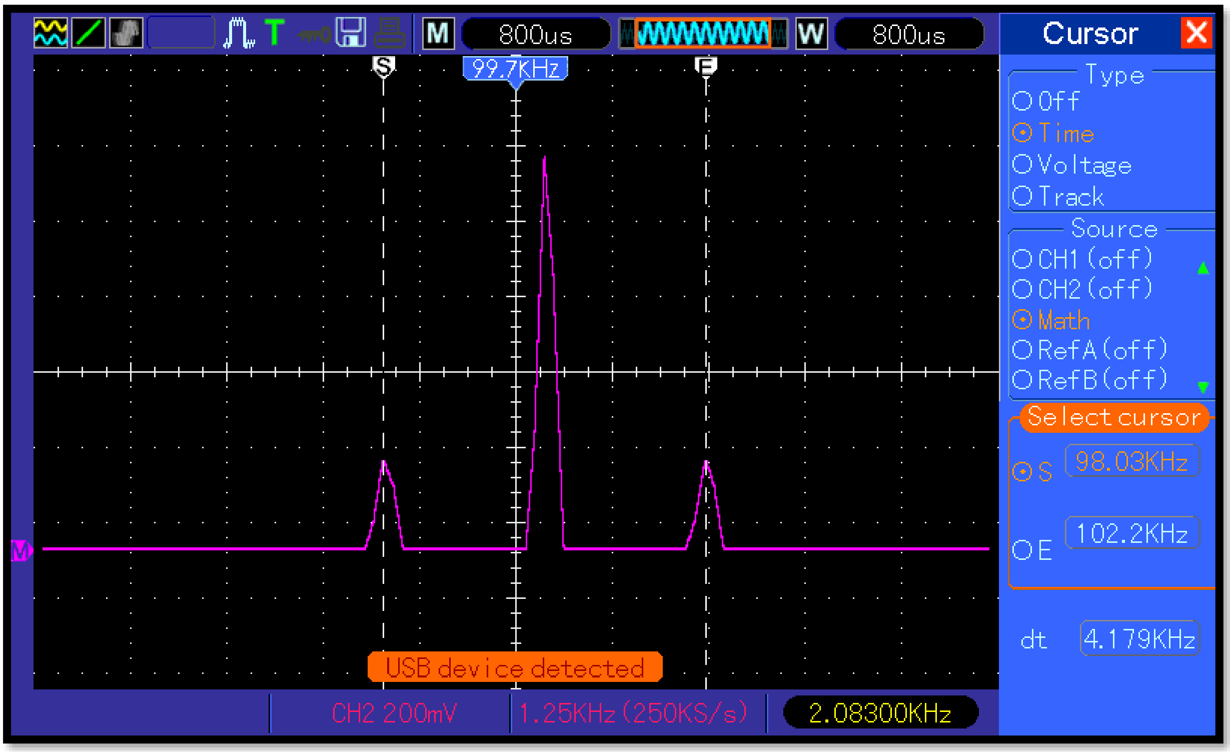


Fig 1.8 (AM spectrum)

Part 2: Frequency Shift Keying FSK:

Like in the last assignment, there is a different method besides using ASK. And this is called FSK, Frequency shift keying. Simply put, with this method, the frequency of the carrier signal changes depending on the amplitude of the message digital signal. For the example to show this, a single binary stream signal will be used to modulate the carrier signal. This will give rise to a similar type of shift keying called '*Binary Frequency Shift Keying*' or BFSK.

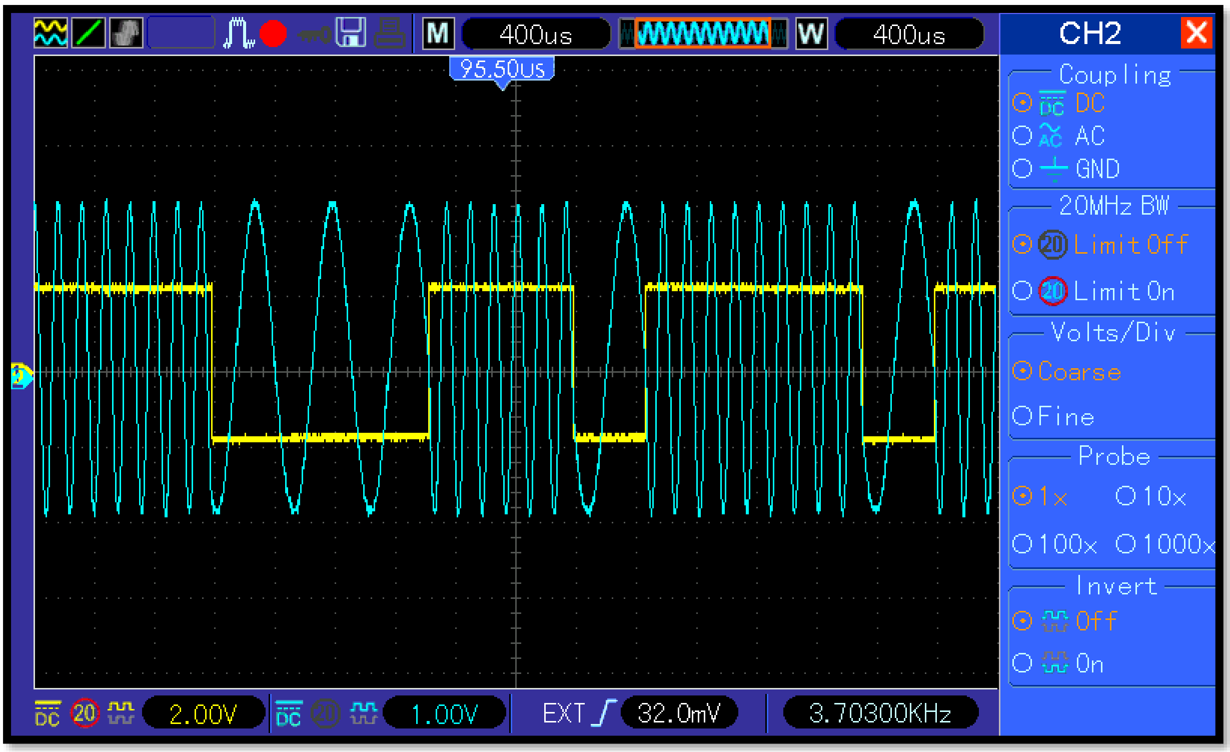


Fig 1.9 (BFSK practical results)

Looking at the results for this modulation method above, two sections can be seen. The high frequency period when the data signal is logic high called the mark frequency, '*Fm*', and the low frequency period when data signal is logic low is called the space frequency, or '*Fs*'.
'*Fm'* is normally higher than the carrier's original frequency and '*Fs*' being lower than the original carrier frequency. The modulator doesn't give an output signal while it is at its original carrier frequency, so the original carrier frequency has been dubbed as the '*nominal'* carrier frequency.

Using the VCO module on the telecoms board can be used to generate this result.
To do this, the trigger source is set to '*external*' by pressing the '*trig menu*' button, selecting '*EXT'* by pressing the '*F2*' button. Making sure to have both channel 1 and channel 2 to be displayed on the oscilloscope by pressing both chn1 and chn2 buttons until they are both lit up and displayed on screen.
Pressing 'CH1' in the vertical section of the scope and then pressing 'F1' button to select the 'Coupling DC' setting, doing the same method again for CH2 but pressing F2 instead to select 'Coupling DC'.
The SEC/DIV knob in the horizontal section is then used to adjust the horizontal scale to 400us/div.
The VOLT/DIV knob in the vertical section is then used to set the vertical scale for channel 1 to a value of 2V/div and channel 2 is set to a scale of 1V/div.
On the VCO module, the gain control is then set to half of its travel before then adjusting the frequency adjust control to a quarter of its travel. The range control of the VCO module is then set to the LO position.
The dip switches on the sequence generator module is set to the binary value of 00 before connecting the third oscilloscope lead to the 'SYNC' connection in the sequence generator module to the 'EXT TRIG' connection on the oscilloscope.
The results for this can be seen in figure 1.9.

When the FSK signal changes from '*mark*' to '*space*' and vice versa, there is an immediate shift in frequency changes in the carrier signal, never being at the nominal carrier frequency long enough for it to make any difference considering that the switch between high and low is so instantaneous.

Using the scopes cursors to measure the frequency of Fm and Fs and its periodic time between cycles. The results recorded are as follows…

Mark
Fm ➔ 920us – 750us = 170us
Fs ➔ 270us – 240us = 30us

Space
Fm ➔ 1.86ms – 1.71ms = 0.15ms
Fs ➔ 260ms – 230ms = 30ms

Frequency
Fm = 4.34KHz
Fs = 1.42KHz

Conclusion:

In conclusion, what has been discovered is that on a basic level, it can be seen that the frequency of the modulated carrier signal changes in proportion to the change in amplitude of the message signal. This allows for a basic understanding of how the FM method functions and how the waveform is affected or changed.
It was also seen that the amplitude of the sidebands could be calculated roughly by using the Bessel table. The way to use the Bessel table also involved calculating the modulation index from the recorded

values of the experiment. Using the calculated index and the rounded down index, we were able to see a difference between the two sets of results. Calculating the index and relating that value back to the Bessel table is also used to determine how many sidebands that modulated FM signal possesses.
The difference between the AM and FM signals can be simplified as the FM signal being more complex when compared to AM. FM having a varying number of sidebands when modulating using a single frequency signal at 2KHz.
Both AM and FM have their own set of positives and negatives. For example, FM struggles to penetrate through walls and becomes less stable at very high carrier frequencies but has a good signal quality in comparison to the AM signal. The AM signal also has a superb signal penetration.
The example of FSK shown above concludes that there are three possible states that there are three possible states the signal can be in. the first being '*mark*', where the signal frequency is higher than the modulators nominal signal frequency. The second being '*space*', where the signal is at a lower frequency compared to the modulators nominal signal frequency. And finally, the nominal modulator frequency. The nominal modulator frequency is the original carrier signal frequency that will not give an output signal at this frequency.

Referencing:

1. Electronics-notescom, 2020. Electronics-notescom. [Online].[02 April 2020]. Available from: https://www.electronics-notes.com/articles/radio/modulation/frequency-modulation-fm-sidebands-bandwidth.php
2. Rfwireless-worldcom, 2020. Rfwireless-worldcom. [Online].[02 April 2020]. Available from: https://www.rfwireless-world.com/Terminology/Advantages-and-Disadvantages-of-AM-and-FM.html
3. Com/radio-equipment-repairs-and-hire/, 2020. Com/radio-equipment-repairs-and-hire/. [Online].[02 April 2020]. Available from: http://wrootltd.com/radio-equipment-repairs-and-hire/

Analogue and Digital Modulations Systems

[8]

Ben Edwards 18026826

Introduction:

For this lab assignment, multisim will be used to simulate the process of varying modulating data transmission systems while being exposed to a simulating noise distortion. Using this, the effects of the noise on these different systems can then be researched, tested and recorded.
The methods that are going to be used in this assignment are…

- FM Modulation
- QPSK
- BPSK
- 16-QAM

These systems have their own characteristics that will determine how the process of data transmission will run during the simulation.

During the experiment, the effects of the bit rate per symbol of a system, the type of system used, and the SNR value will be used and altered to monitor the effects on the methods listed above.

FM Modulated Signal: -

Shown in the diagram below, a simulation of a Frequency modulation block diagram has been constructed in simulink. Using this block diagram, the effects of modulation and demodulation can be shown using the scope component.

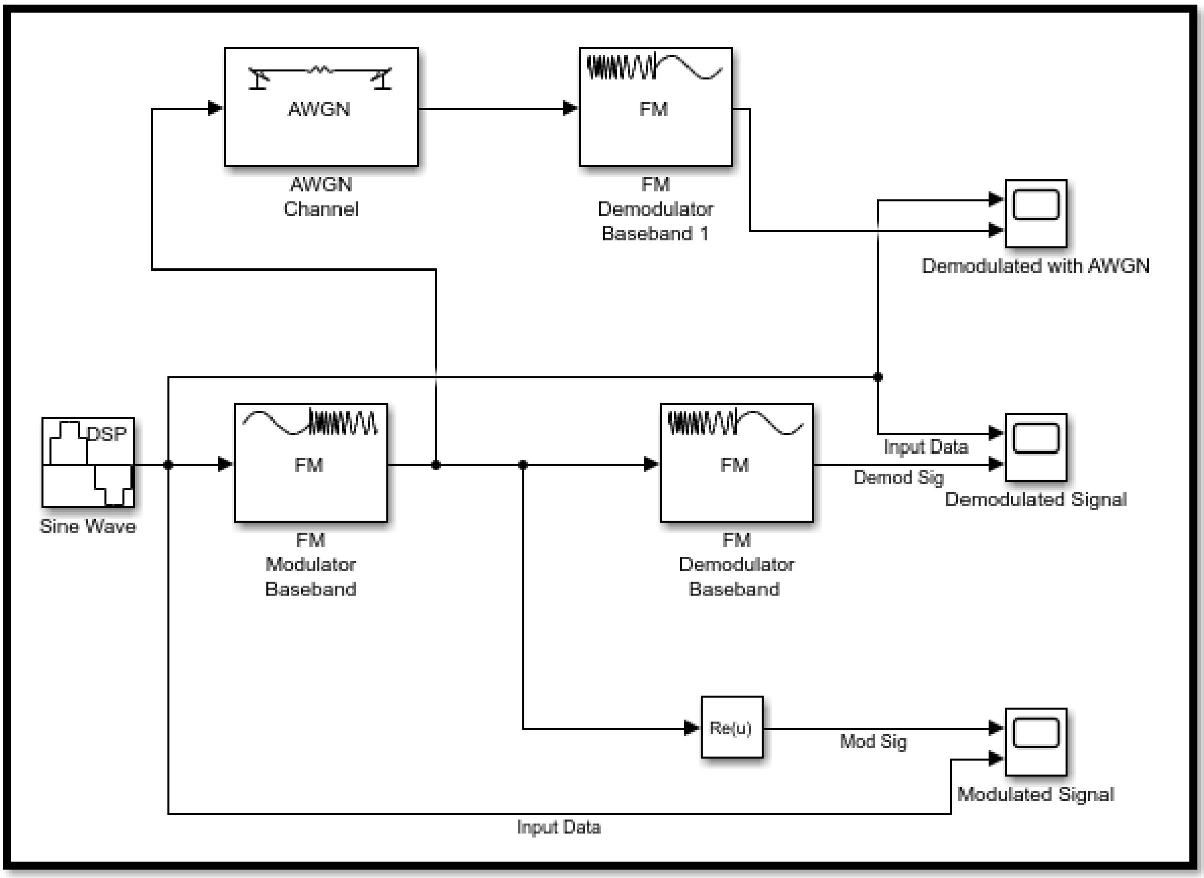


Fig 1.1(diagram for a FM modulated diagram in simulink)

The outputs measured for both the modulation and demodulation can be shown below in figure 2.2. the graph on the right being the modulated signal and the graph on the right being the demodulated signal. The modulated signal shows that it changes its frequency, depending on how fast the amplitude of the input signal changes within a given fraction of time. So, at either high or low peaks of the input signal, the modulated signal would have a high frequency, but then a low frequency between those two points of change. The input data in the modulated graph can be seen as a blue sine wave and the modulated signal in yellow.
When the signal becomes demodulated however, the demod signal becomes identical to the input signal. To the point that the two signals perfectly overlap each other, making it hard to see the two signals seperately on this graph.

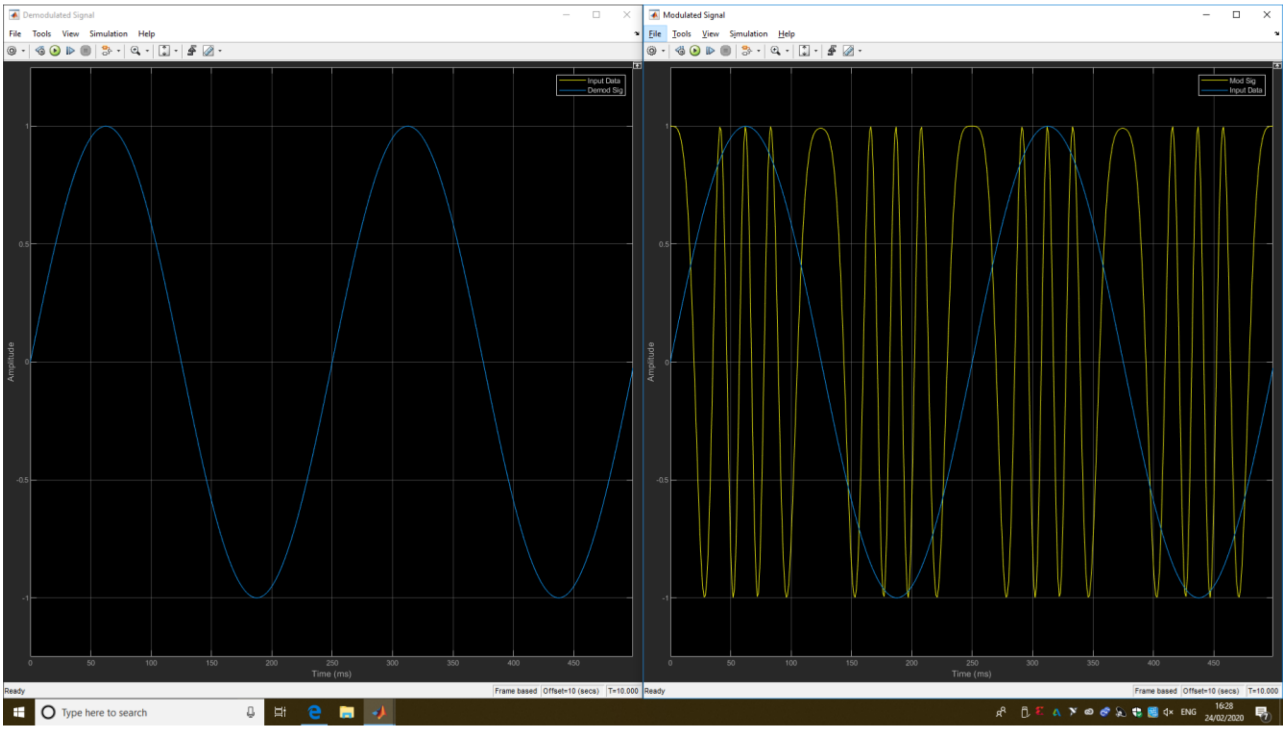


Fig 1.2 (demodulated and modulated signals)

Below is the graph of a demodulation graph while being put through a AWGN channel. An AWGN channel essentially is just a simulated white noise that would get picked up by the signal during transmission before demodulation. The signal to noise ratio in dB, called SNR, can effect the noise level of the final signal. Below is a graph showing the result of this using a ratio value of -10dB.

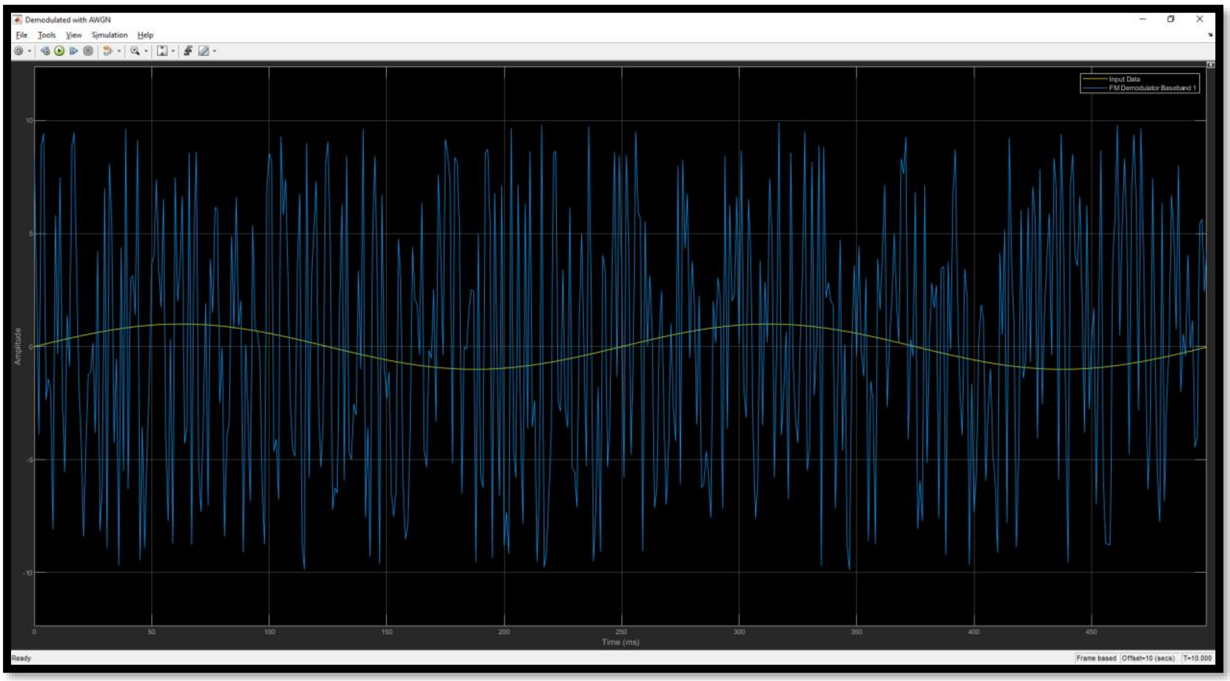
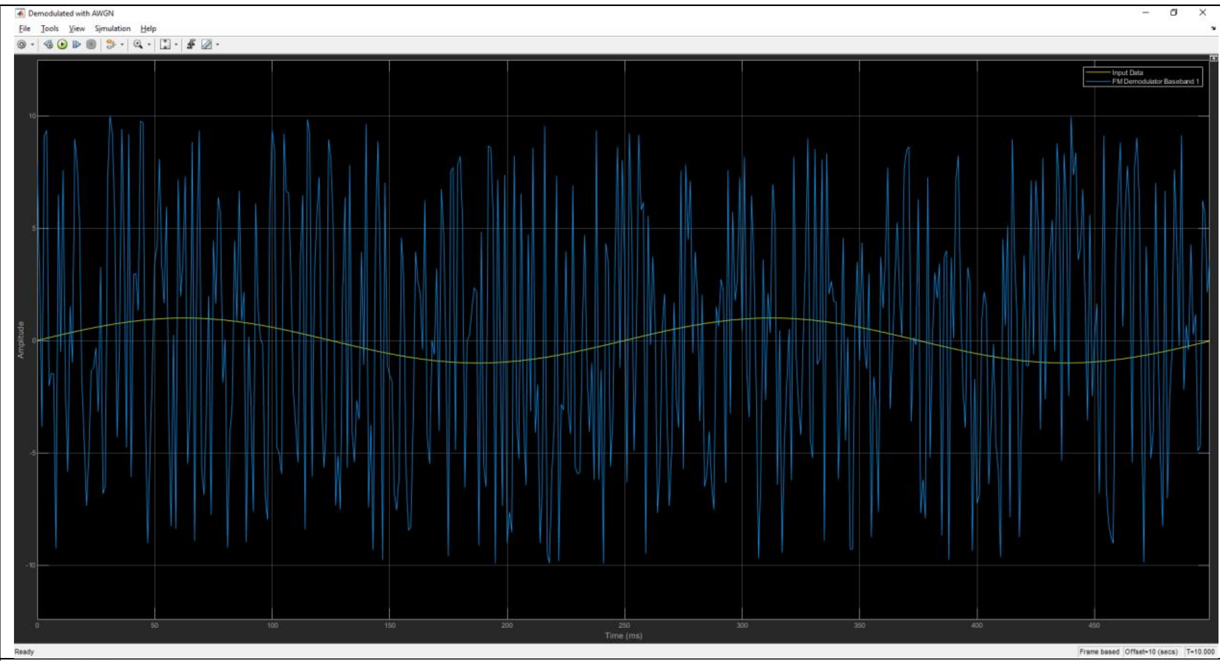
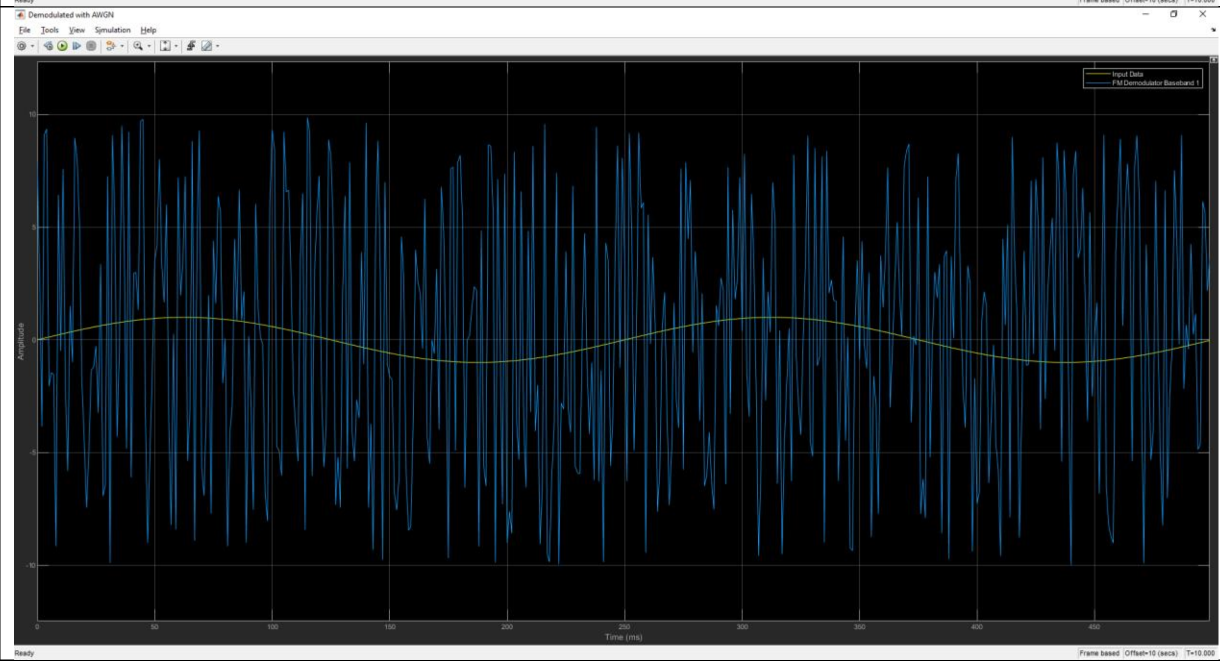
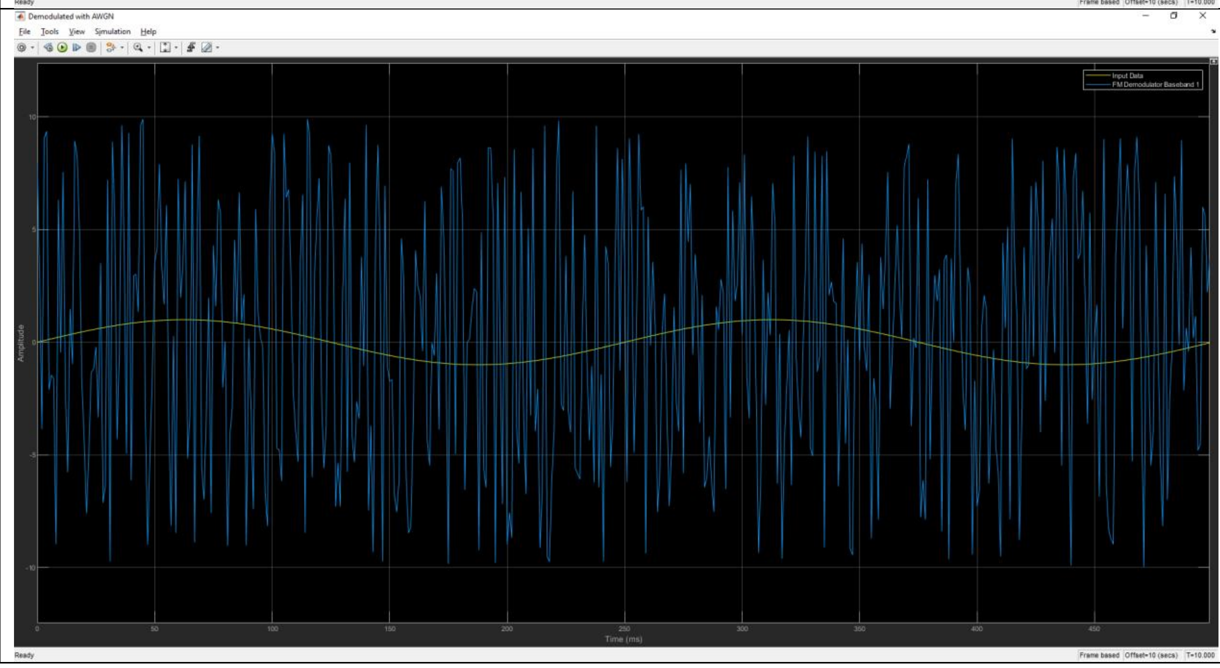


Fig 1.3 (demodulated signal with AWGN channel)

The table below demonstrates the effect of the AWGN channel using an array of incrementing dB values.

	-10dB
	-5dB
	0dB

Demodulated with AWGN
File Tools View Simulation Help
Input Data
FM Demodulator Baseband 1
Amplitude
Time (ms)
Ready
Frame based
Offset=10 (secs)
T=10.000

Demodulated with AWGN
File Tools View Simulation Help
Input Data
FM Demodulator Baseband 1
Amplitude
Time (ms)
Ready
Frame based
Offset=10 (secs)
T=10.000

Demodulated with AWGN
File Tools View Simulation Help
Input Data
FM Demodulator Baseband 1
Amplitude
Time (ms)
Ready
Frame based
Offset=10 (secs)
T=10.000

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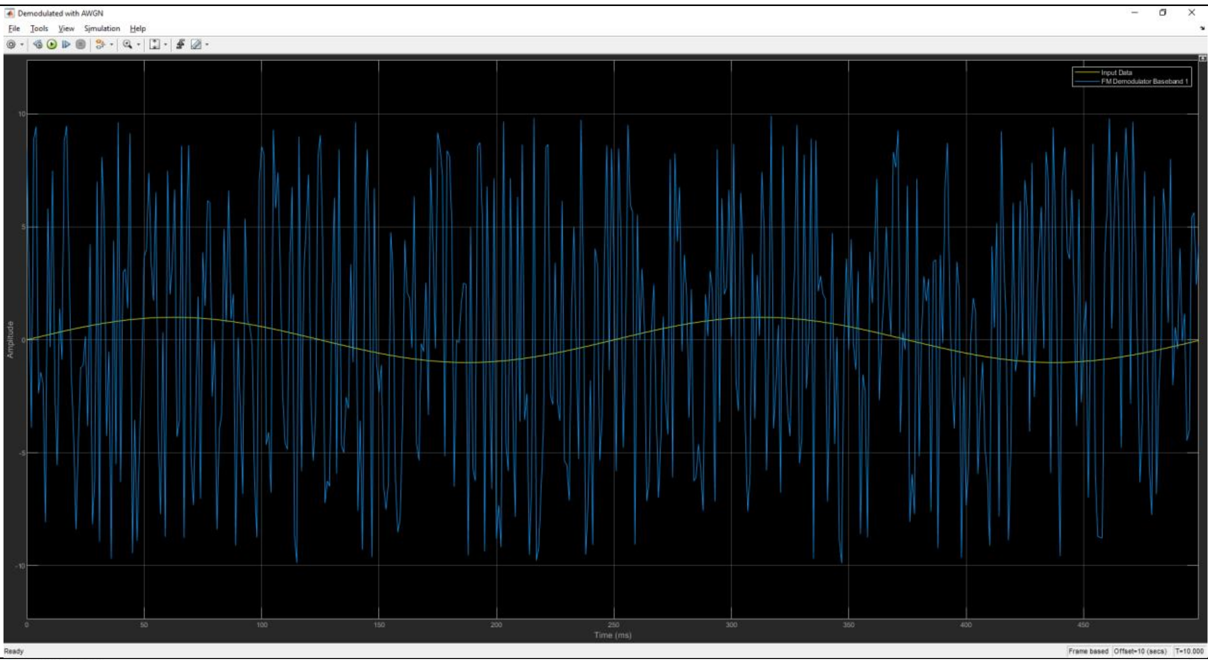
<tr><td>
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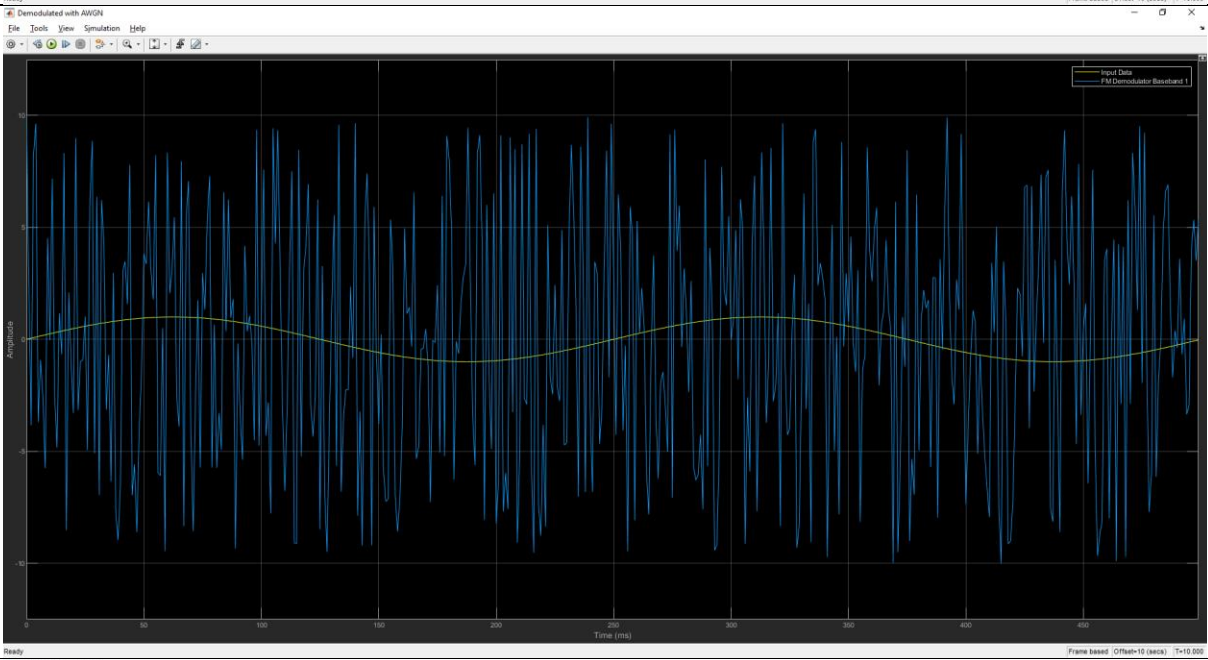
<tr><td>
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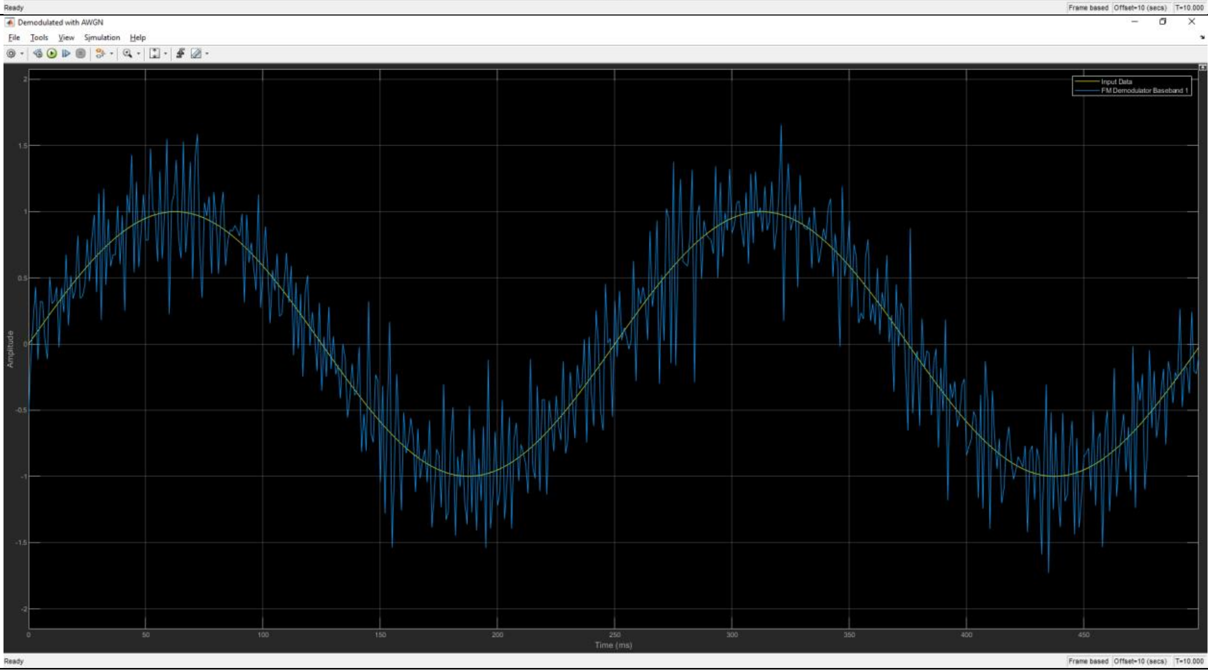
<tr><td>
</td><td>50dB</td></tr>
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Table 1.1 (demodulated signal using AWGN at -10, -5, 0, 10, 20 and 50dB)

Looking at these results, it shows that when the ratio value in dB increases, the more accurate the demodulation becomes. The effects of the dB values don't become as clear as to what's going on. At least not until reaching the value of 50dB. At 50dB the noisy sine wave can be seen following the same frequency of the input data in yellow. At lower dB values however, the noise seems to overpower the effects of the demodulation, rendering the signal unusable for any audio device.
The dB value effects the severity of the noise by using the formula **(SNRdB = Psignal-dB – Pnoise-dB)** so determining from this formula, when the SNRdB value is high then the dB value of Pnoise is significantly lower than the dB value of Psignal. This allows for a cleaner signal to be demodulated.

QPSK system: -

The Quadrature Phase-Shift Keying (QPSK) system that codes two digital bits by putting digital signals through a radio frequency carrier signal. Four phase states are used to perform the coding of the two digital bits. These phase states are 0, 90, 180 and 270 degrees. **[1]**
In the diagrams below, the left chart shows that there are four dot sections being displayed on the constellation graph. The number of dots for this transmission system is in direct correlation with the number of combinations of 2^n in bits. So, for example the first dot represents the binary value of 00, the second being 01, third 10, and the last being 11. These points are separate and have their own space of the graph. **[2]**
The values of 00,01,10, and 11 each represent a phase shifted sine signal at an angle of sections of 90*deg. Starting at a phase shift of 45*deg for 00 to 135* for 01, 225* for 10 and 315* for 11. Each signal wave that has been phase shifted, according to the information to be transmitted, will be sequenced one after the other to create this one transmission line for two bits of information at a time. **[3]**
This graph will be able to show the error rate for each section of data that has been transmitted. The experiment conducted on MATLAB's Simulink has been simulated using the decibal values of -10, -5, 0, 5, 10, 20, and 50dB. The results can be seen below in table 1.

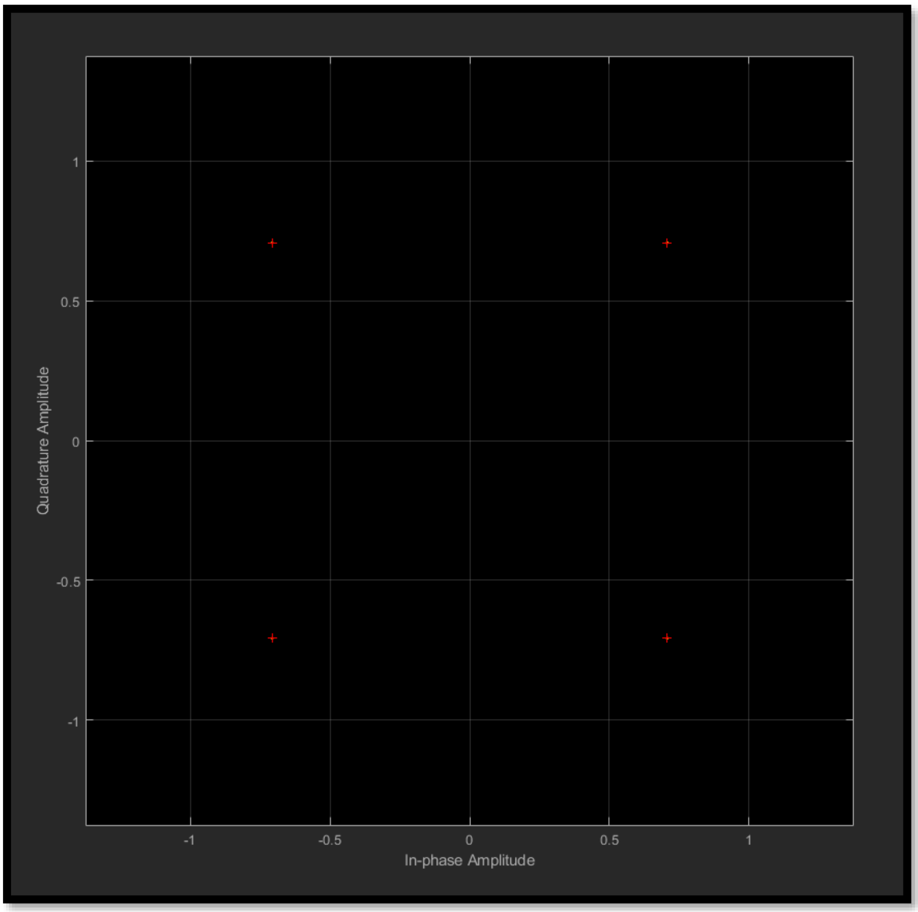

Figure 2.1

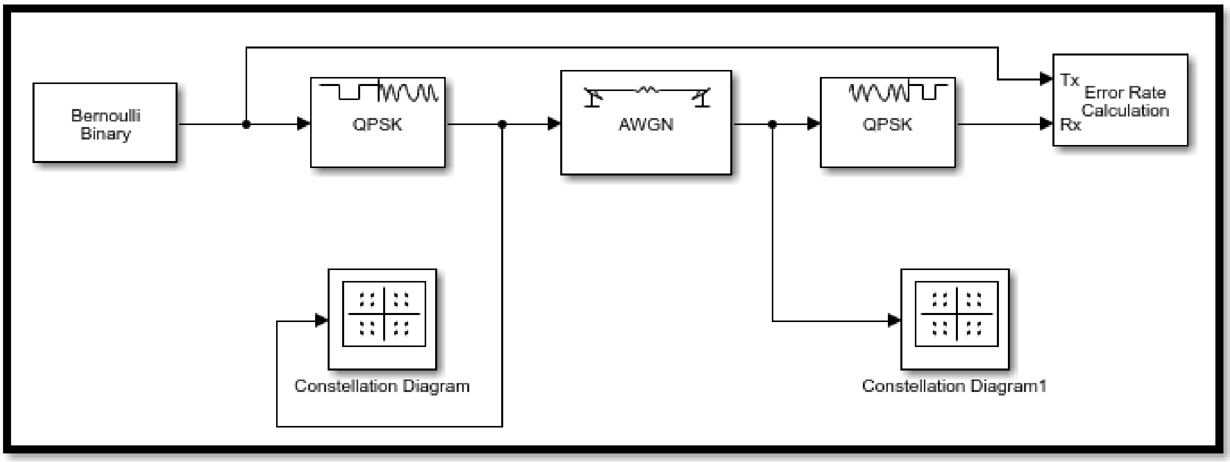

Figure 2.2

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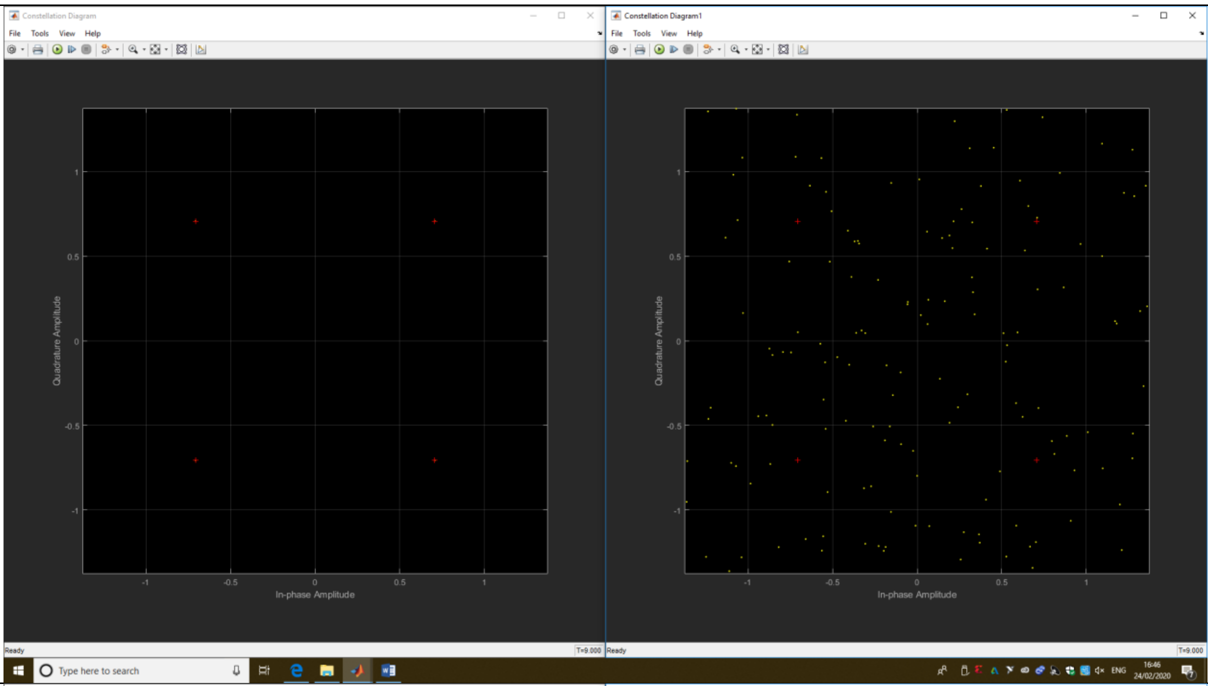
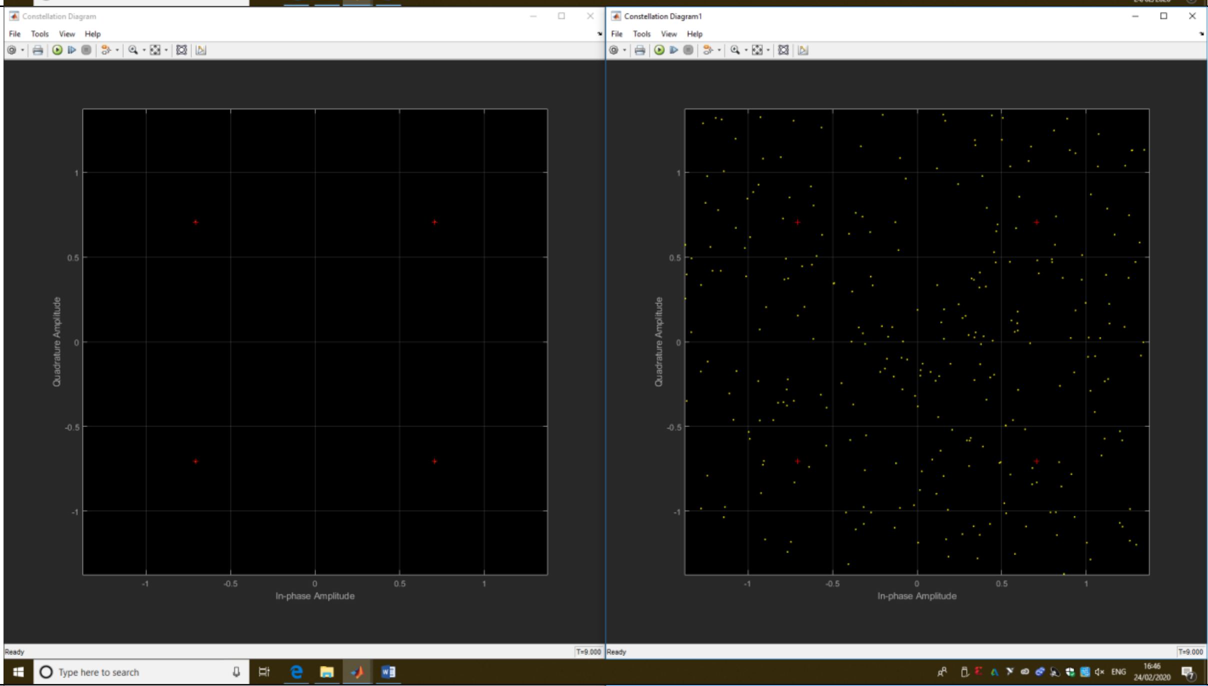
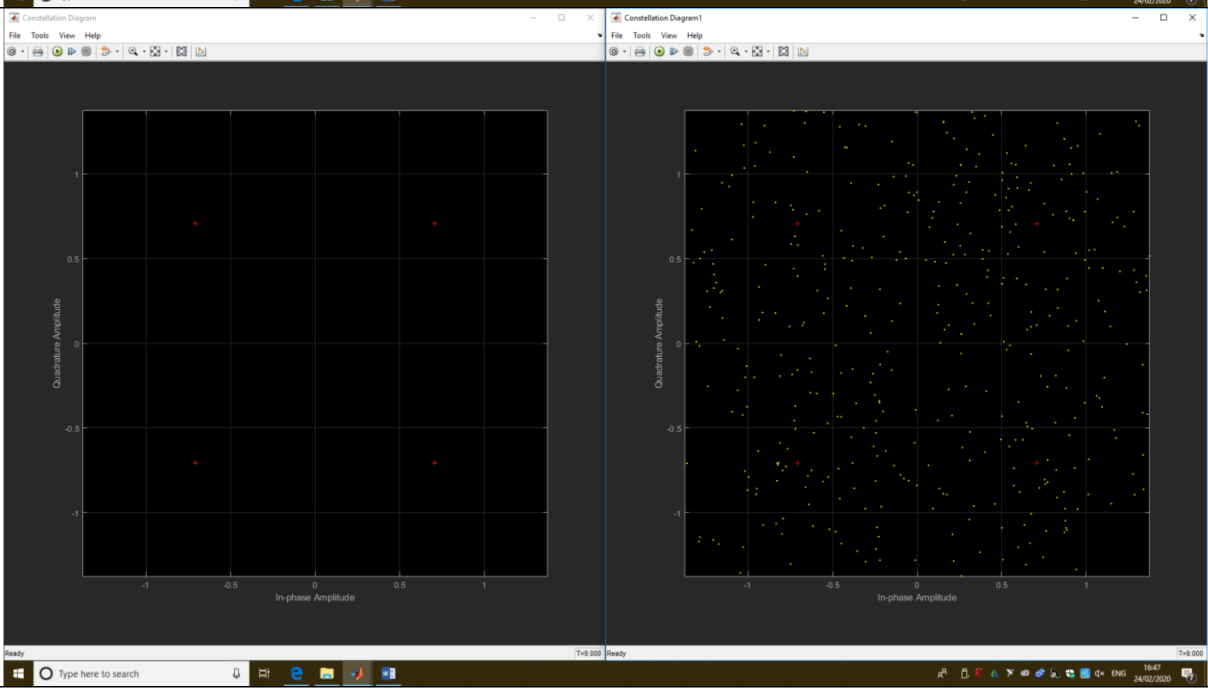
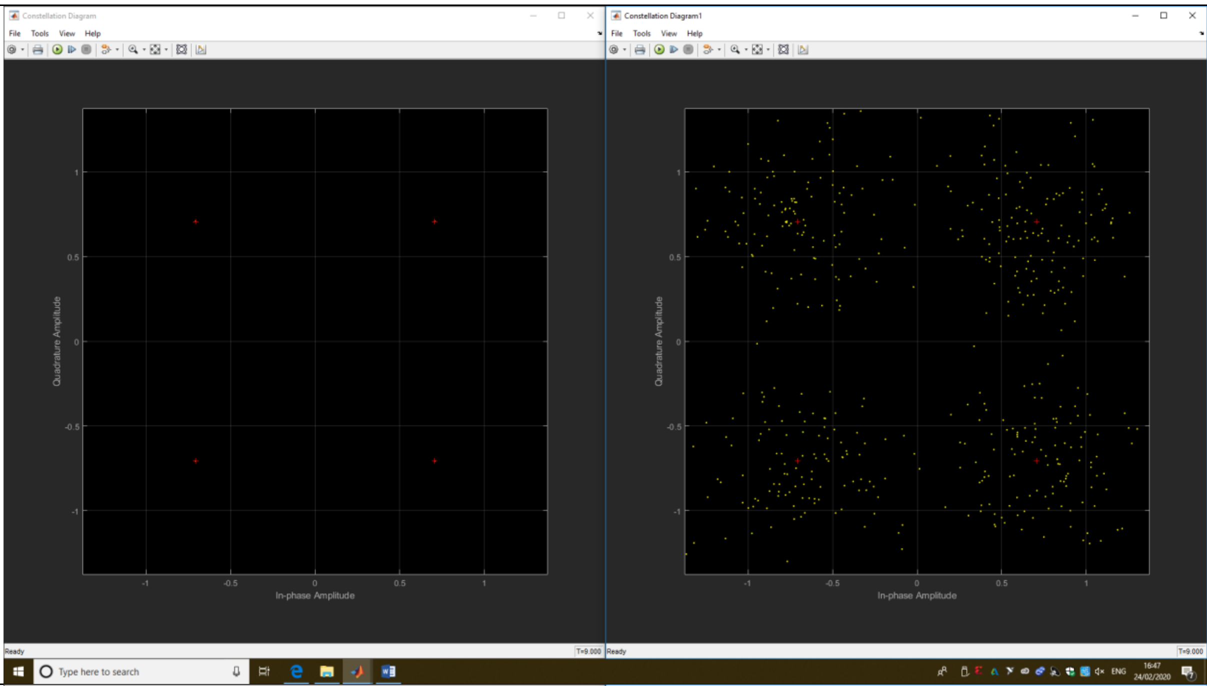
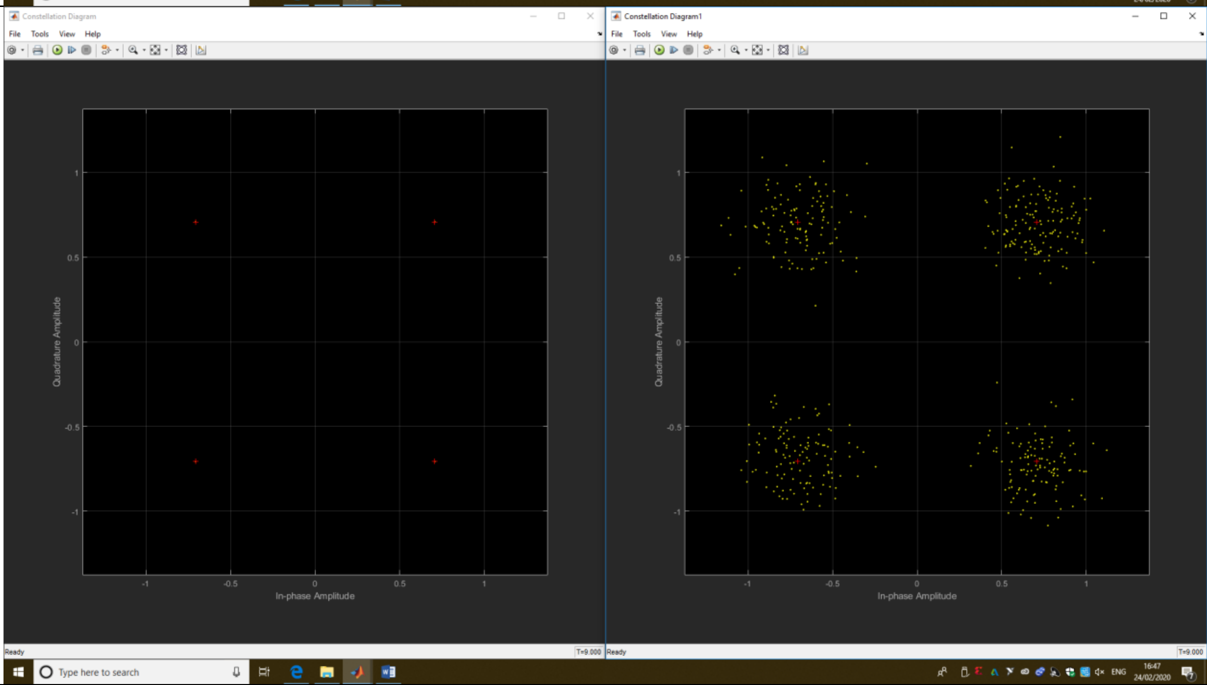
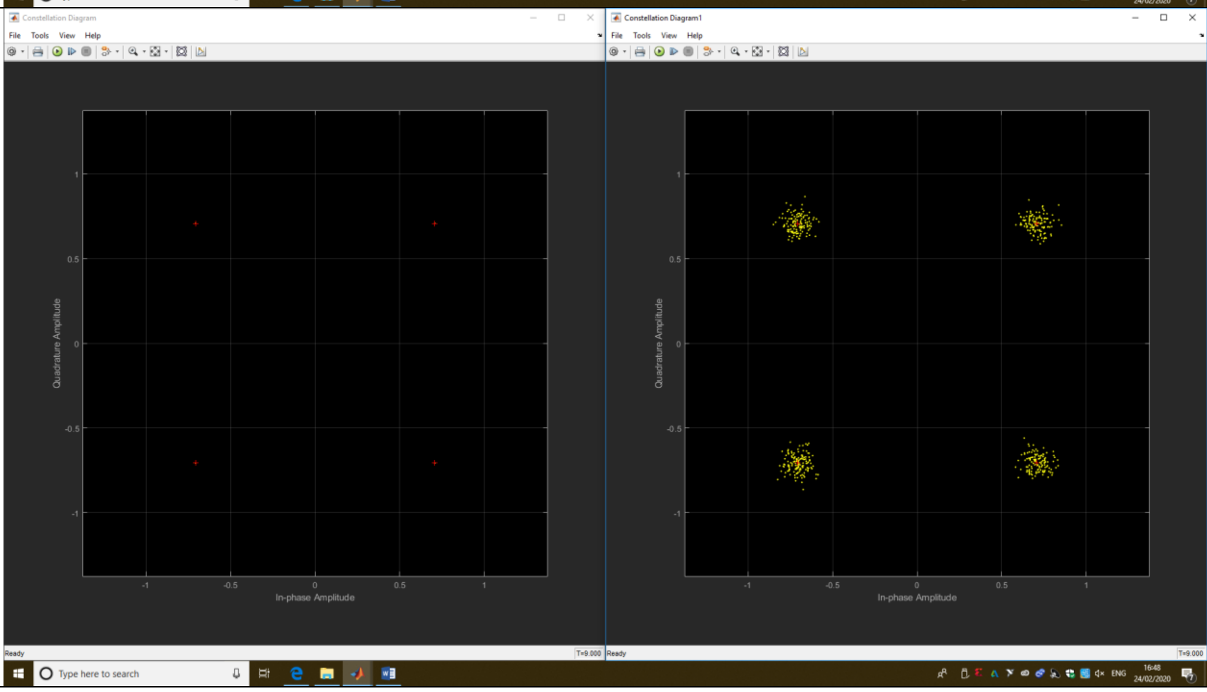
	-10dB
	-5dB
	0dB

	5dB
	10dB
	20dB

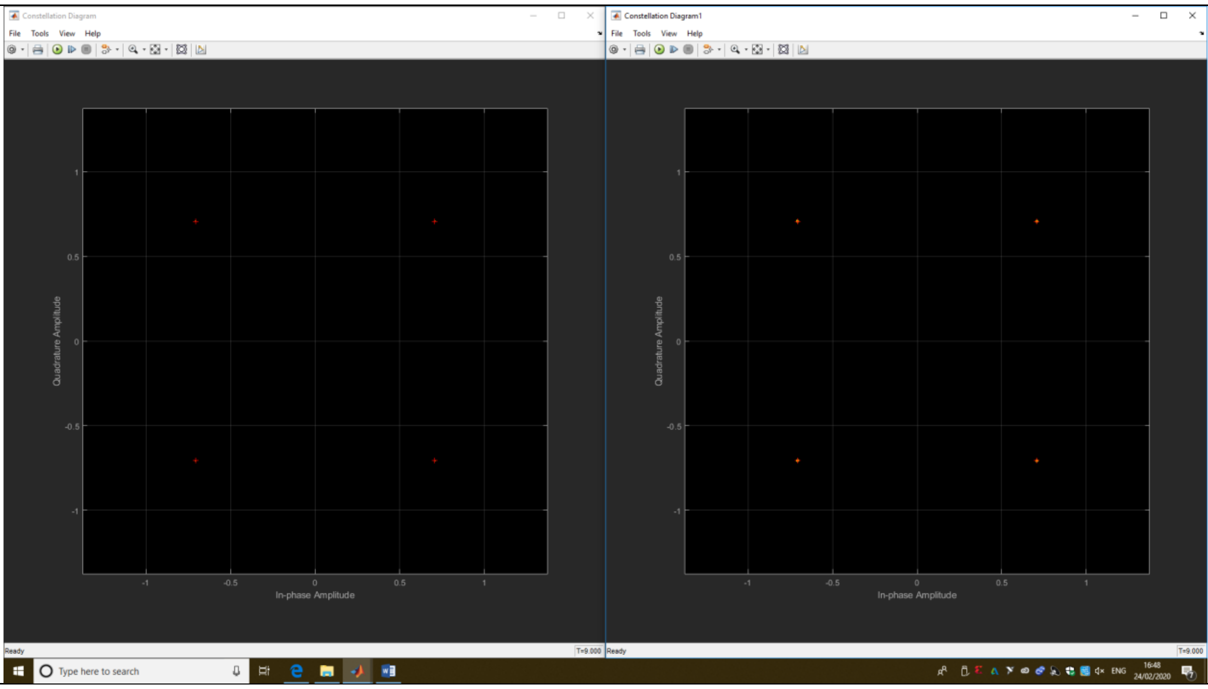
	50dB

Table 1

/***/

It can be seen that a high SNR value will reduce the error rate of the transmitted data. On the graph there are the four dots representing the 00, 01, 10 and 11 binary values of the phase shift signal. These dots are called symbols **[3]**. When the SNR value is at a value where the dB value of the signal is lower than the dB value of the noise, then the error rate for the signal becomes significant, not allowing for a reliable transmission of the modulated signal. When the SNR value increases to where the signal dB value is higher than the noise dB, then the error rate for the signal begins to reduce more and more the higher the ratio value. Eventually, it can be seen that the scattered points start to converge into its designated areas around the four symbols, each symbol representing a two-bit value.

BPSK System:

A BPSK System is called the '*Binary Phase Shift Keying*' system. It is also known as the '*phase reversal keying'* or '*2-PSK'* due to its characteristic of separating the phases by 180 degrees, or it can also be described as inverting the signal **[4]**. The maximum rate of transmission for this method is 1bit/symbol **[5]**. Because of this, this system functions similarly to the QPSK system but only at a binary value of 2^1. The bit that represents 0 can be seen on the right side at a coordinate of (1,0) for 0*deg phase shift, and the binary value for 1 is towards the left at a (-1,0) at 180*deg phase shift. So, looking at this graph, it is essentially an angle graph, rotating anticlockwise to 1 rotation until it reaches 360*deg. **[4]**
The block diagram for this system can be seen in figure 3.1 in Simulink.

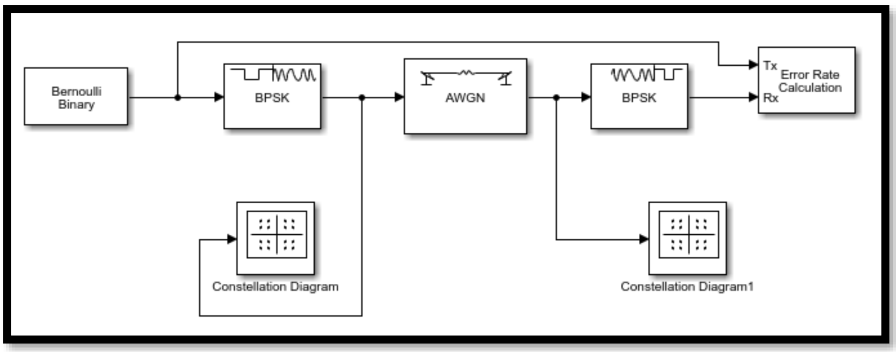


Figure 3.1

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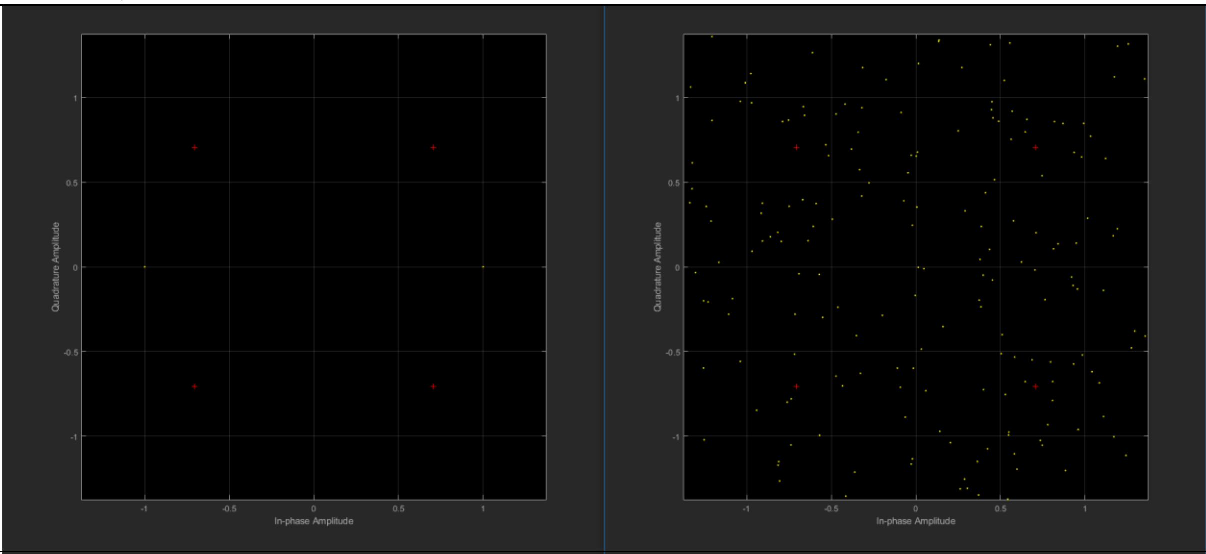
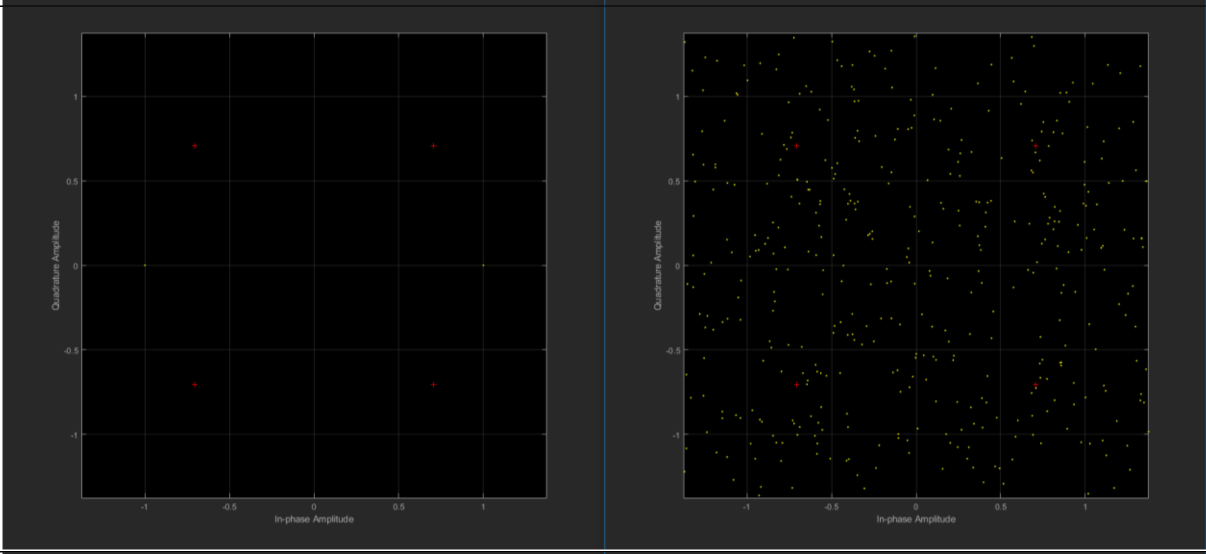
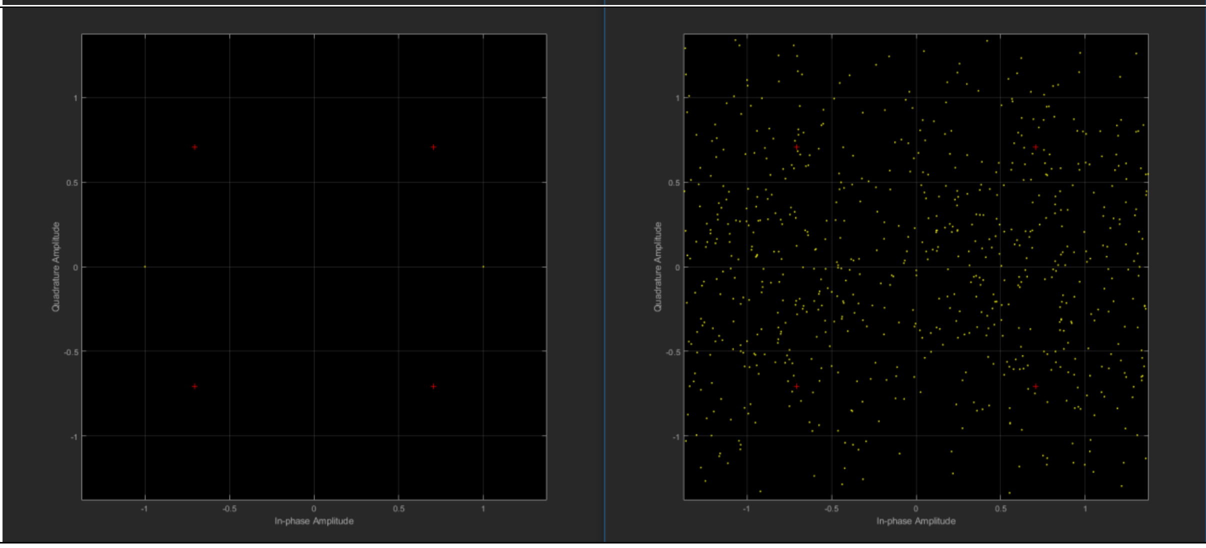
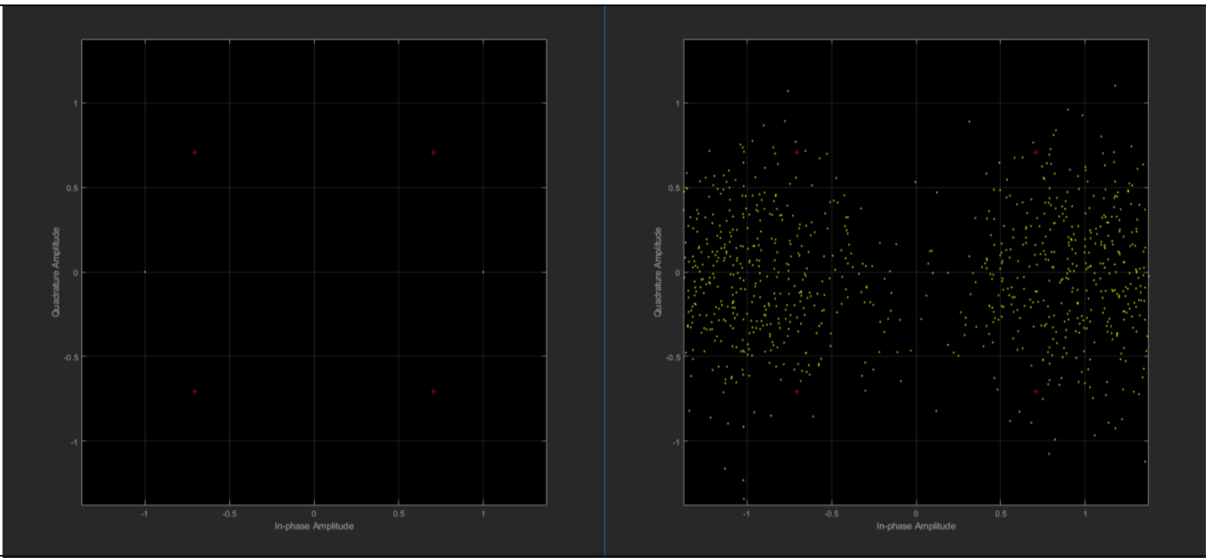
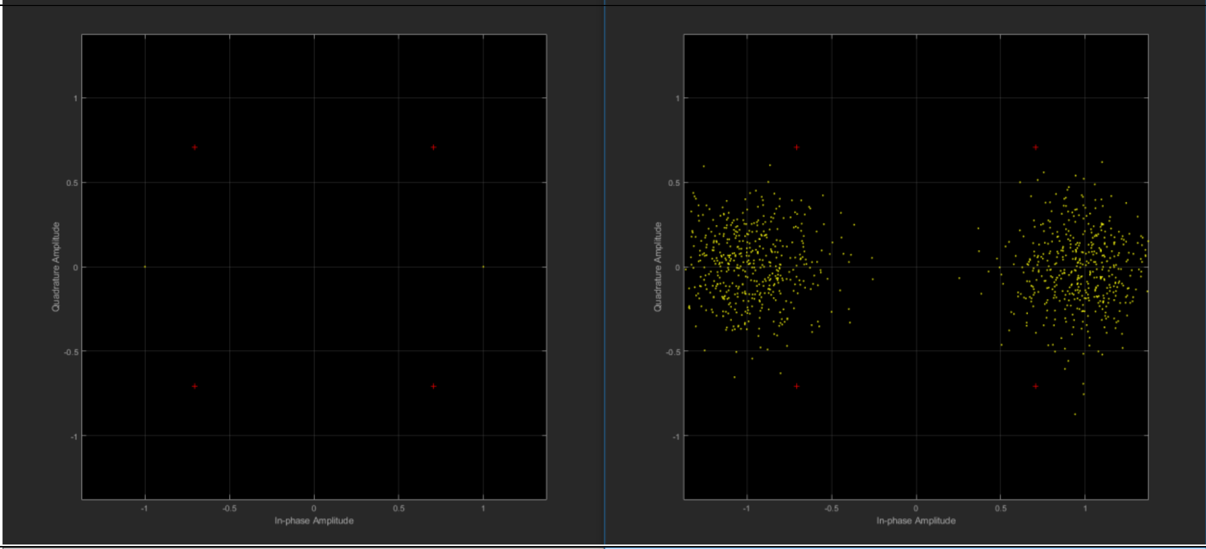
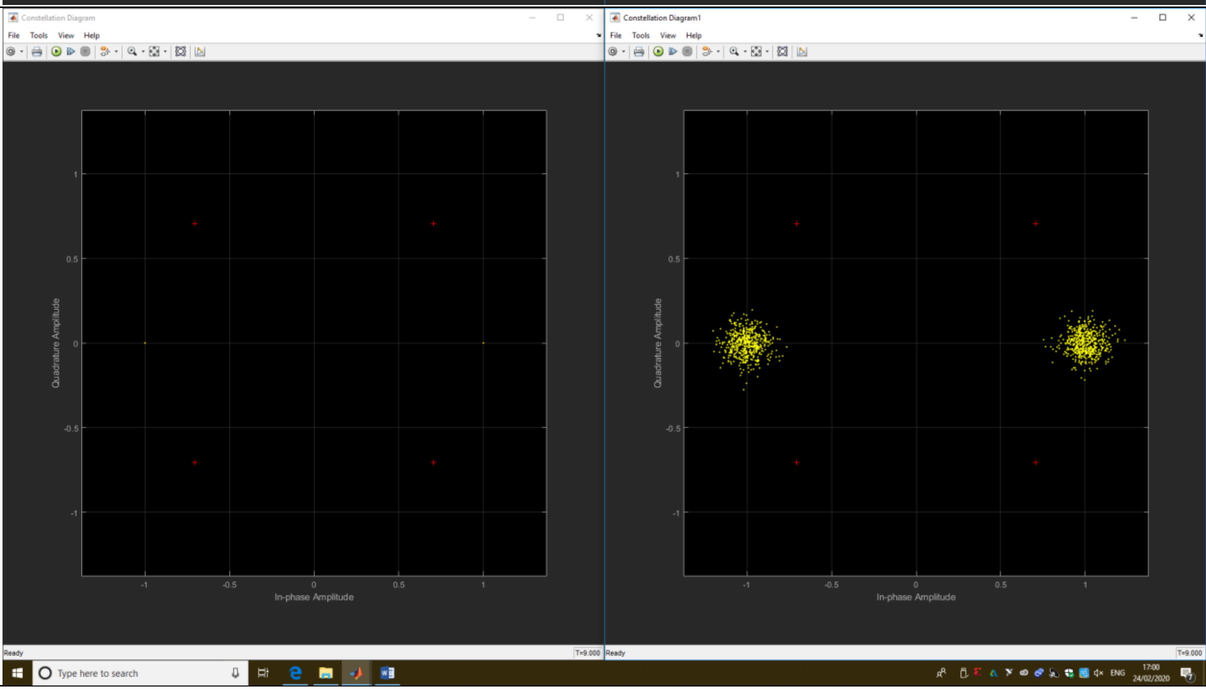
 Quadrature Amplitude In-phase Amplitude	-10dB
 Quadrature Amplitude In-phase Amplitude	-5dB
 Quadrature Amplitude In-phase Amplitude	0dB

	5dB
	10dB
	20dB

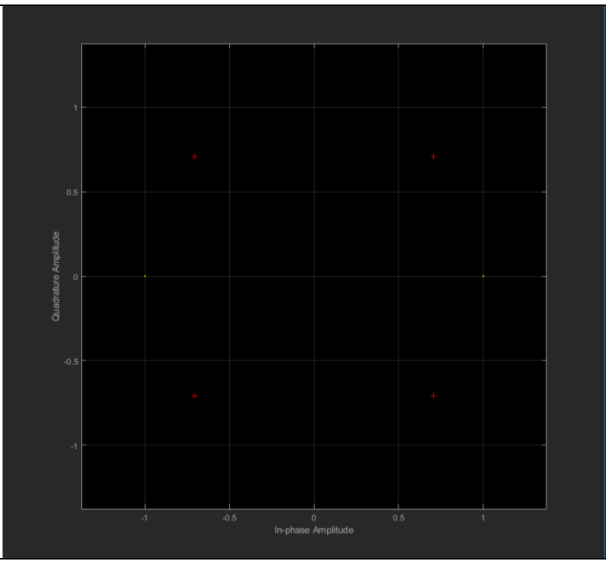
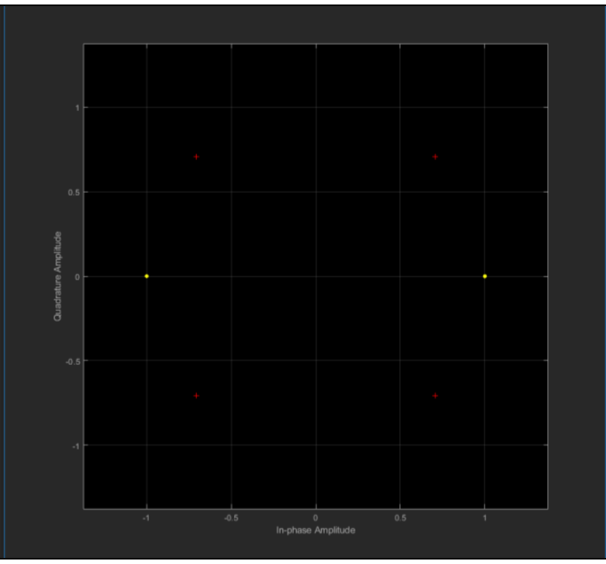
 	50dB

Table 2

/**/

Looking at the results for this simulation, the accuracy of the modulated transmission increases the higher the SNR value becomes. The graph also shows the two symbols that represent the binary numbers 0 and 1, and how the results have been received at the demodulator. The error rate at 50dB is slightly worse in comparison to the error results for the QPSK system. From this it is theorised that the lower the binary values there are, the less chance for errors in the system can occur from noise. However, when more bit values are added, then the system requires a higher SNR value to account for the number of bits that the system uses to transfer information.

16-QAM System:

A 16-QAM system, also called a '*16-bit Quadrature amplitude modulation system*' uses a combination of phase shifting and amplitude shifting **[5]**. Shifting the phase 4 times and having 4 different amplitudes will create a system that will allow for the use of 16-bit data transmission. This is because 4*4 is equal to 16, the maximum number of possibilities for a binary number made from 4 bits. Because of this, this system has a transmission rate of 4bits per symbol. Using this method will require the use of a higher amplitude than a QPSK system would usually require in its operation, along with a phase shift in steps of 45*deg.

The results for this and the block diagram can be seen below…

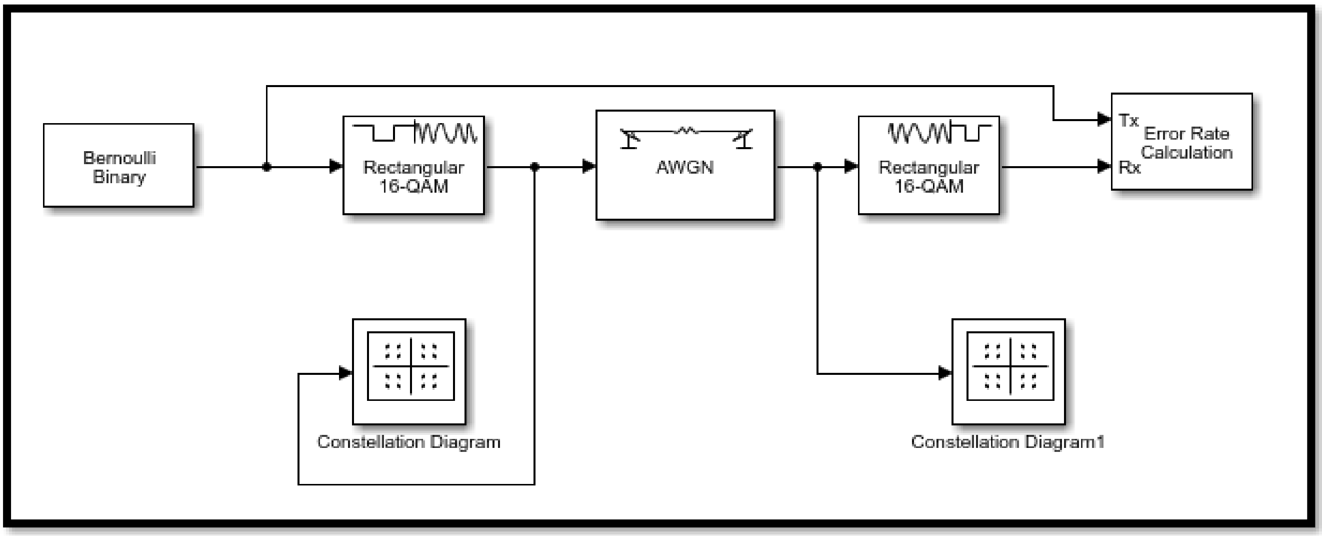


Figure 4.1

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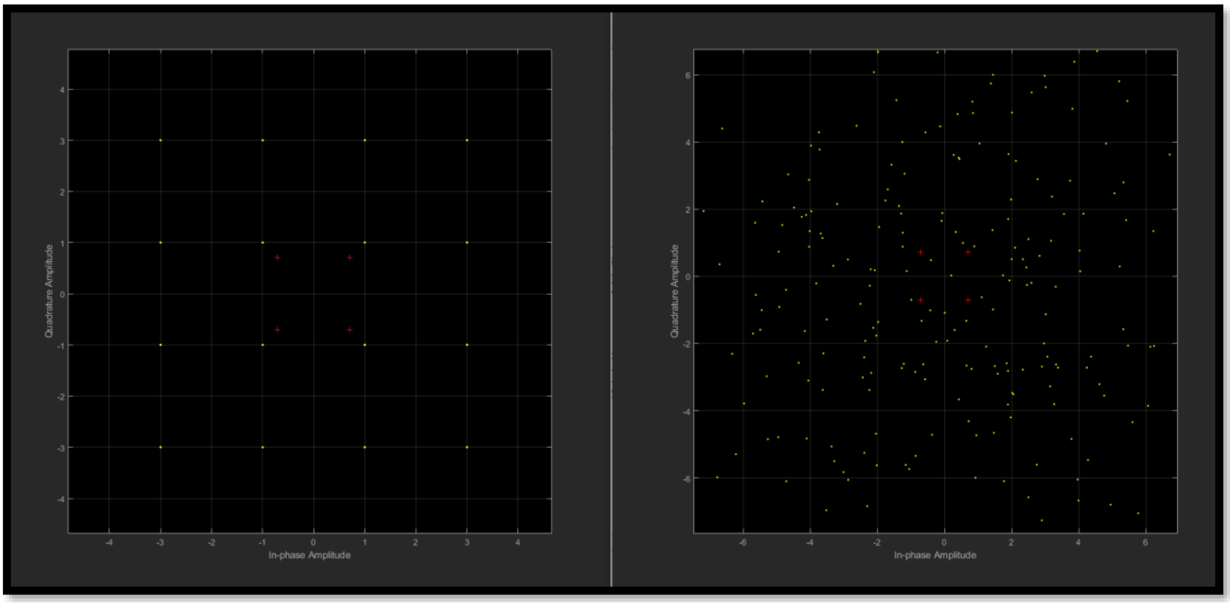
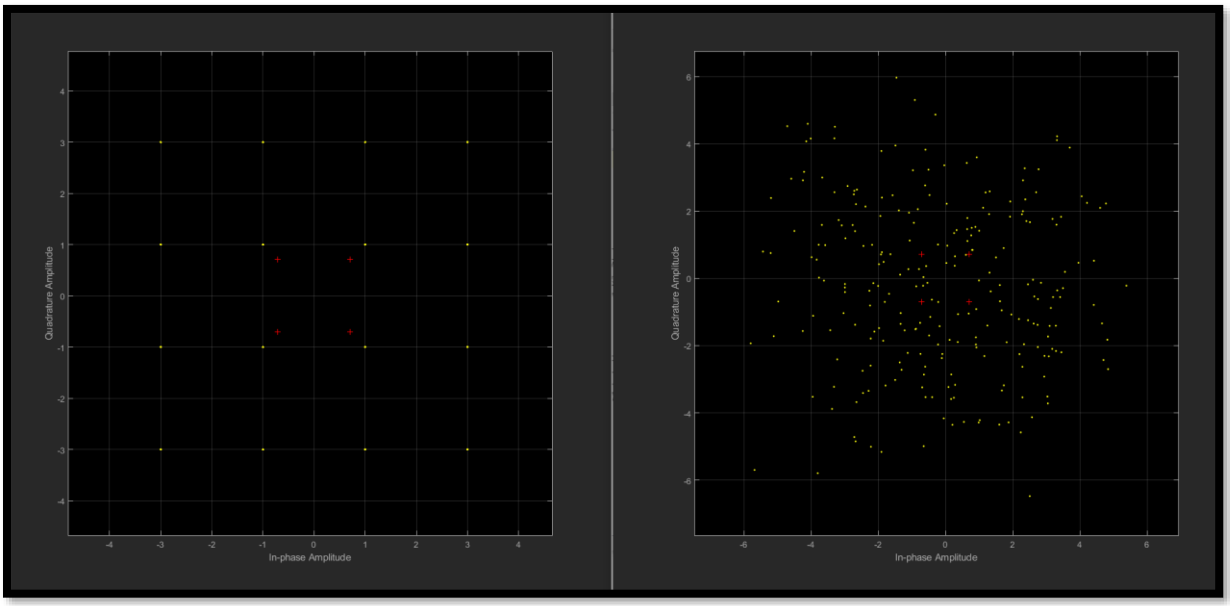
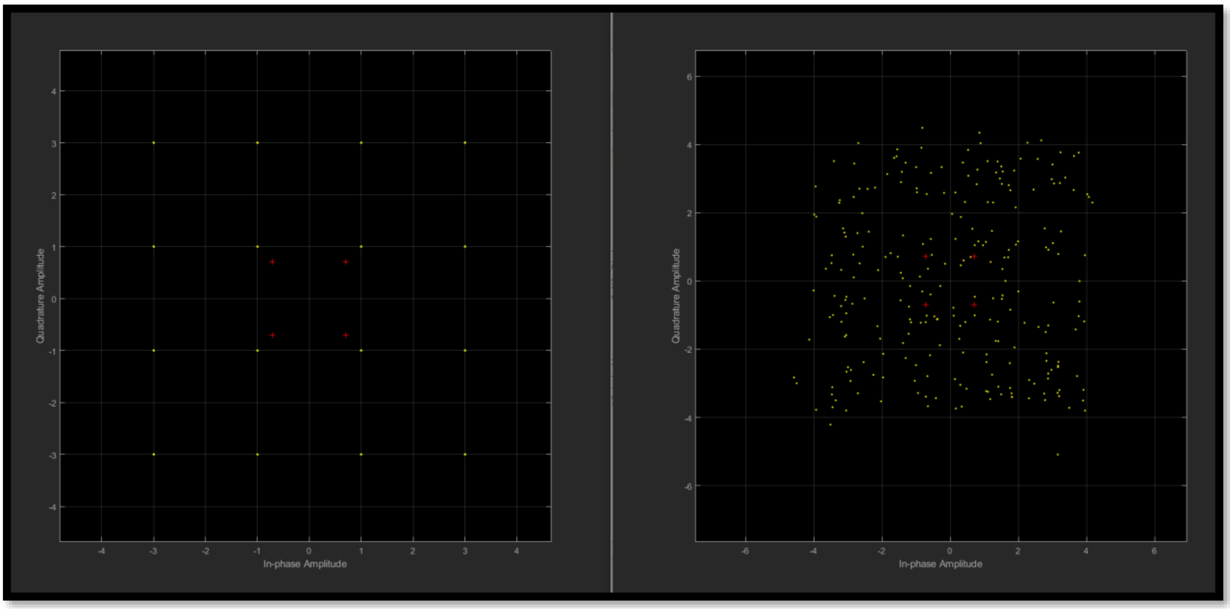
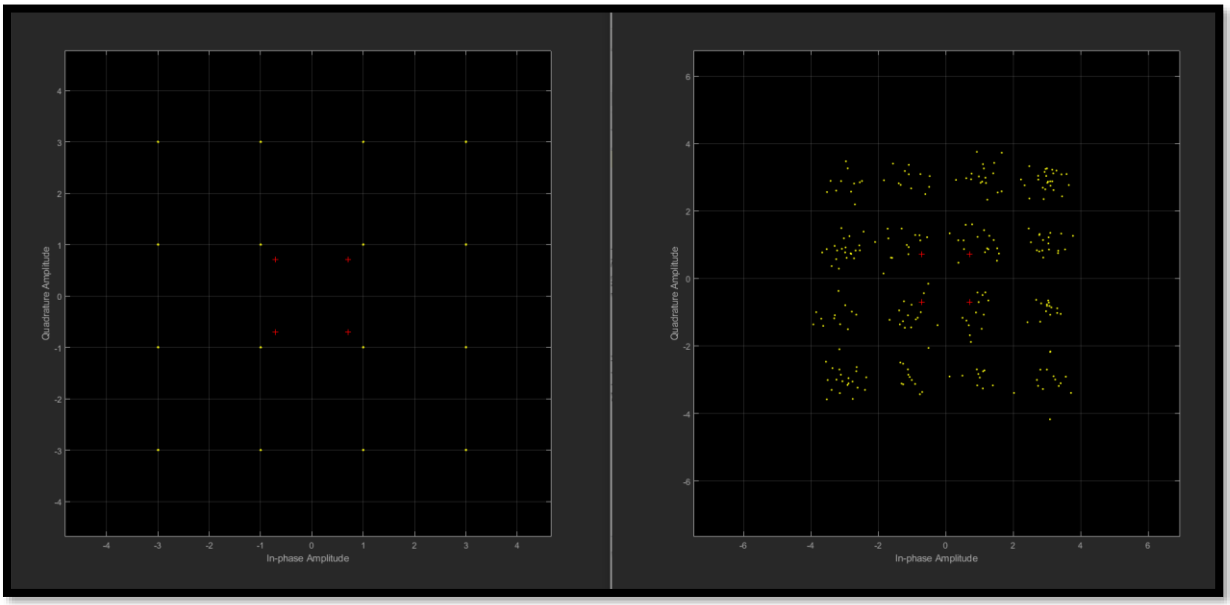
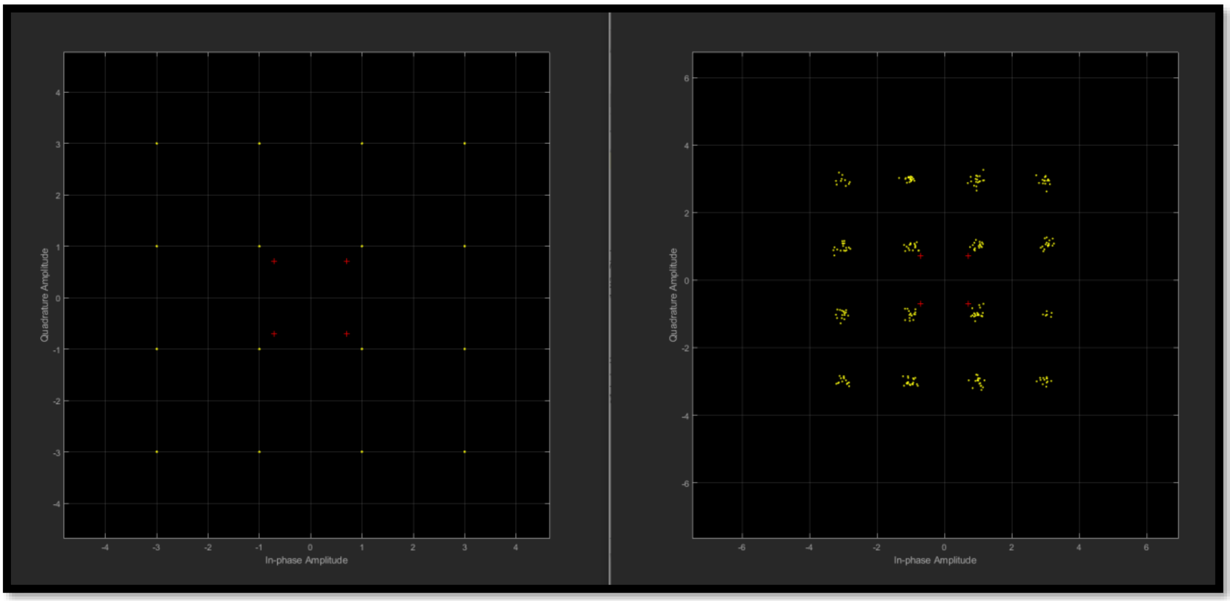
Quadrature Amplitude In-phase Amplitude Quadrature Amplitude In-phase Amplitude	-10dB
Quadrature Amplitude In-phase Amplitude Quadrature Amplitude In-phase Amplitude	0dB



5dB



10dB



20dB

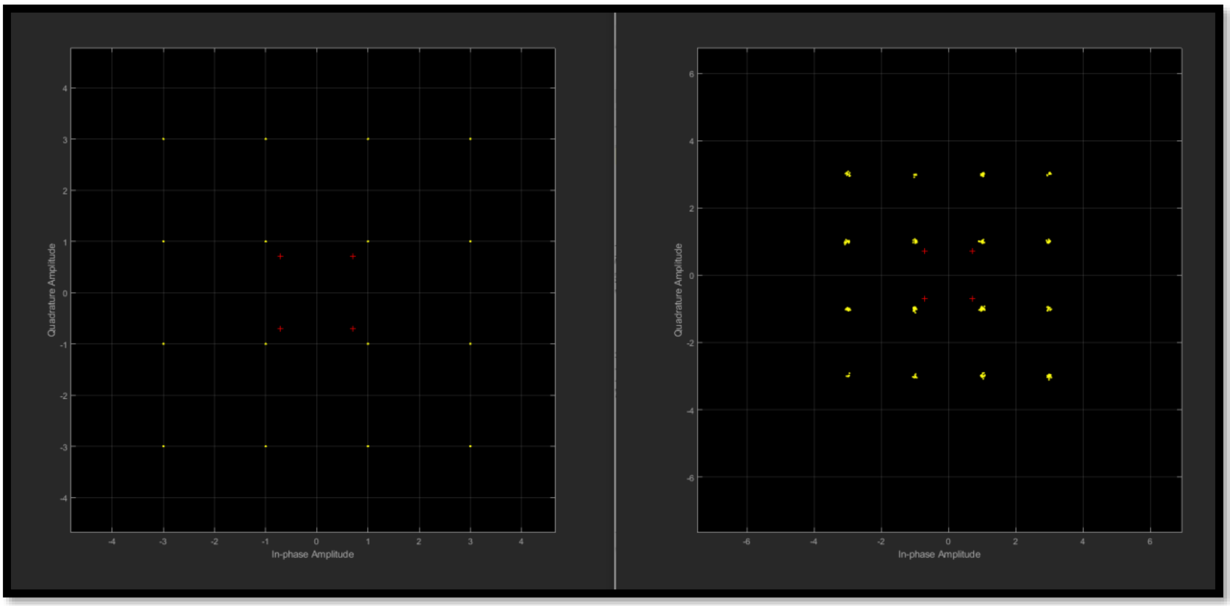
	30dB

Table 3

/***/

It can be seen that there are more symbols being used within the system for a faster data transfer rate. The more symbols there are then the higher the bit error rate will be. This can be seen in figure 4.2 **[6]**. This can be seen if the close up of the BPSK and the 16-QAM system results were displayed. This can be seen in figures 4.3 and 4.4 below.

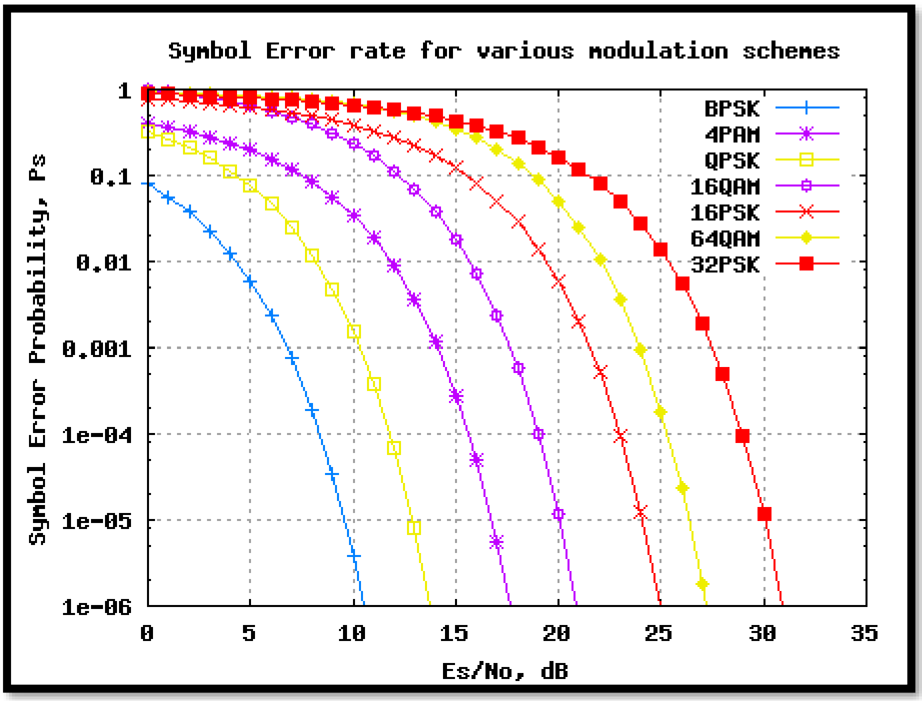


Figure 4.2

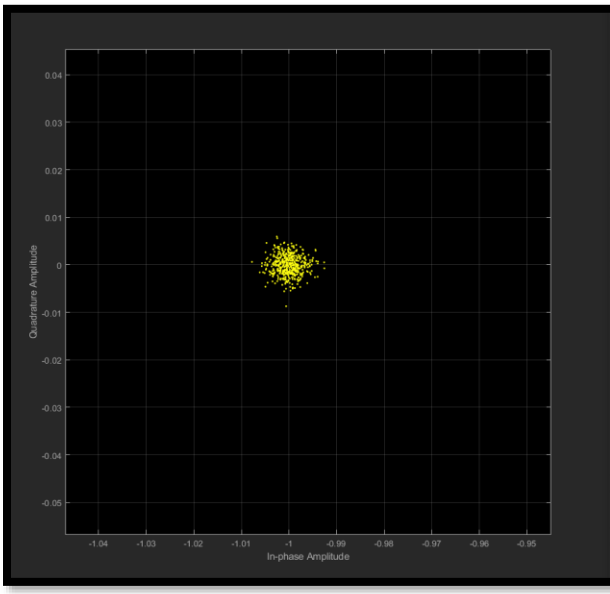


Figure 4.3 (BPSK 50dB)

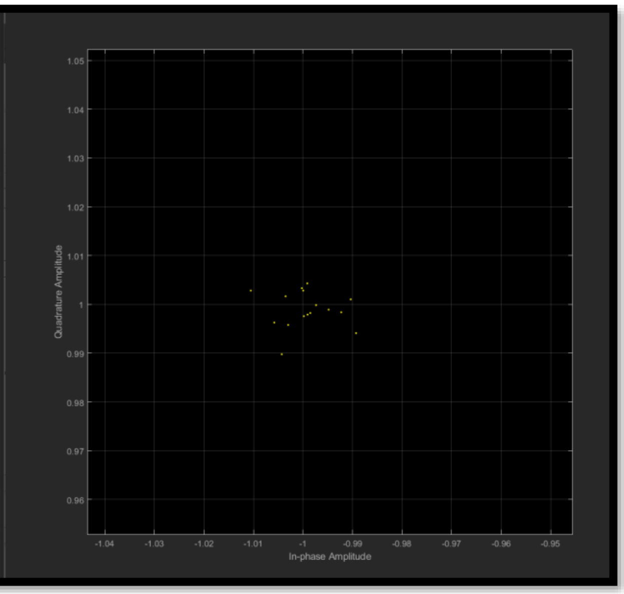


Figure 4.4 (16-QAM 50dB)

It can now be clearly seen that there is a slightly higher rate of error in the 16-QAM system in comparison to the BPSK system.
Because 16-QAM has a higher bit transmission per symbol, there is a higher probability that the error rate will occur more frequently at lower SNR values due to the symbols being so close to each other in a grid layout. Though because of this there is a trade-off of using less power in comparison to the PSK methods.
This is because in a PSK system, to achieve the same level of distance between the symbols as a QAM system, a higher amplitude would be needed. Leading to a higher consumption of power at the price of accuracy. **[7]**

Conclusion:

In conclusion, A system like a PSK and QAM uses the difference in amplitudes and phase shifted periods to differentiate between a specified binary value. The quantity of binary combination values assigned to each phase and amplitude is dependant on the number of bits per symbol. The higher the bit/symbol, the more symbols there are in the system. And the more symbols there are, the more the phase shift needs to be divided and segmented and the higher the amplitude is required to account for this.
It can also be seen that a higher SNR value will help to combat against the bit rate error created by any noise distortion, which has been simulated using Simulink. A higher number of symbols can affect the accuracy of a system, being more susceptible to bits having different values than intended due to noise distortion.

Reference:

1. Thomas BCross, T.B.C. 2012. telecomreseller. [Online].[02 April 2020]. Available from: https://telecomreseller.com/2012/08/16/quadrature-phase-shift-keying-explained/
2. Anthony Combs, A.C. 2020. Nuwavescom. [Online].[02 April 2020]. Available from: https://www.nuwaves.com/constellation-diagrams/
3. Robert Keim, R.K. 2016. Allaboutcircuitscom. [Online].[02 April 2020]. Available from: https://www.allaboutcircuits.com/technical-articles/quadrature-phase-shift-keying-qpsk-modulation/
4. https://www.youtube.com/watch?v=mfSWO3TpGq4 – 3:43 & 5:11 – tutorangel
5. https://www.youtube.com/watch?v=ZY_7FQk8CjU – 4:18 & 6:24 - Nayanatara Kolhapuri

6. EDN, 2008. EDNcom. [Online].[02 April 2020]. Available from: https://www.allaboutcircuits.com/technical-articles/quadrature-phase-shift-keying-qpsk-modulation/
7. https://www.youtube.com/watch?v=C7KXSG7QEsQ - 2:30 – Professor David S. Ricketts
8. Javier Diaz, J.D. 2020. Blogspotcom/2013/09/in-our-society-use-of-electronichtml. [Online].[02 April 2020]. Available from: http://bbdayz.blogspot.com/2013/09/in-our-society-use-of-electronic.html