

Sam Harries - Signals and Comms Assignment 2

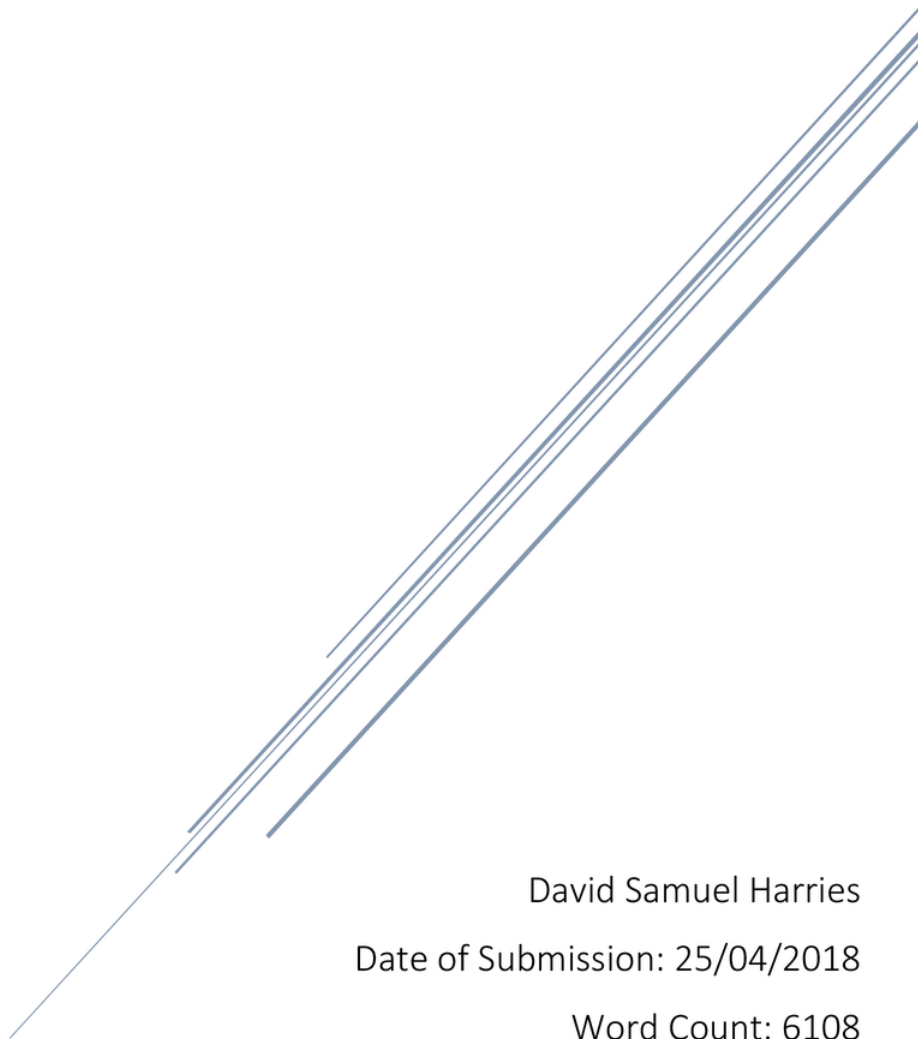
by David Harries

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Assignment 2: Digital Modulation Schemes



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1 – INTRODUCTION

“Modulation is the process of varying one or more parameters of a carrier signal in accordance with the instantaneous values of the message signal.” (www.tutorialspoint.com, 2018) The carrier signal is a high frequency signal which carries no data. This signal is superimposed onto a messenger signal which carries data, which allows for more efficient transmission over long distances.

Several different techniques have been developed to modulate messenger signals in various ways. Each of these techniques have advantages and disadvantages, which make them effective in various applications. These techniques will be researched, analysed and discussed in **Section 2**.

A practical will be carried out to investigate the limits of an ADC (Analogue to Digital Converter) effect the output modulated waveforms. Several parameters will be changed such as:

- The number of bits on the ADC.
- The input amplitude while keeping the frequency constant.
- The frequency while keeping the amplitude constant.

The setup of the practical, analysis of discussion of results can be found in **Sections 3 and 4** respectively.

The learning objectives of this assignment are to be able to analyse and manipulate a range of signals mathematically in both time and frequency domains for digital communication systems, and to demonstrate an understanding of the principles and requirements of digital communications systems from theory by practical testing.

2 – RESEARCH INTO PCM TECHNIQUES

2.1 PAM AND PCM TECHNIQUES

In signals and communication, digital signals are superior to analogue since they are less susceptible to noise and can easily be recovered, corrected and amplified. Because of these superiorities, it is common practise to convert analogue signals into digital data. There are many techniques used to convert analogue signals into digital data. This section will discuss PAM (Pulse Amplitude Modulation) and PCM (Pulse Code Modulation) which are two commonly used pulse modulation techniques in analogue to digital conversion.

2.1.1 PAM (Pulse Amplitude Modulation)

As its name suggests, Pulse Amplitude Modulation is a technique in which the instantaneous amplitude of the pulsed carrier signal is varied according the instantaneous amplitude of the message signal. This can be seen in **Figure 2-1**.

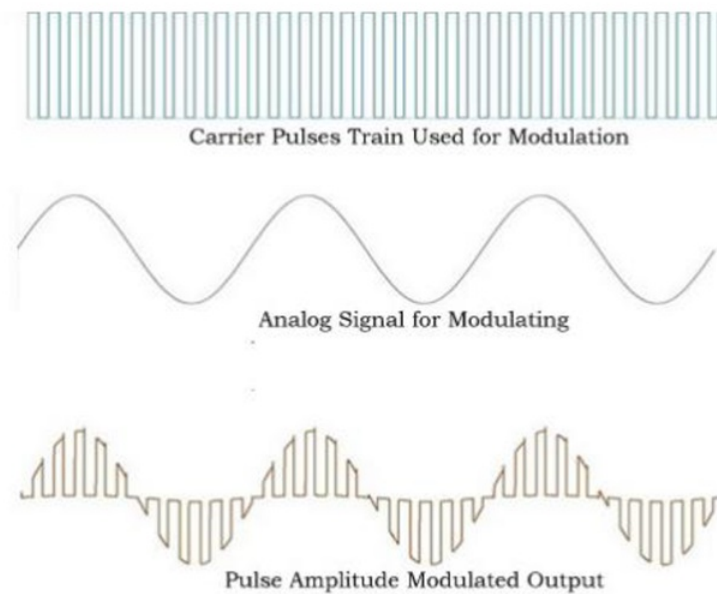


Figure 2-1

PAM can be further categorised into two categories. These are *Single Polarity PAM* and *Double Polarity PAM*. Single Polarity PAM is found when a fixed DC bias is superimposed onto the signal. By doing this all the pulses are made positive. Double Polarity PAM is when the pulses alternate between positive and negative. An example of a Single Polarity PAM waveform can be seen in **Figure 2-2** and an example of Double Polarity PAM can be seen in **Figure 2-3**.



Figure 2-2

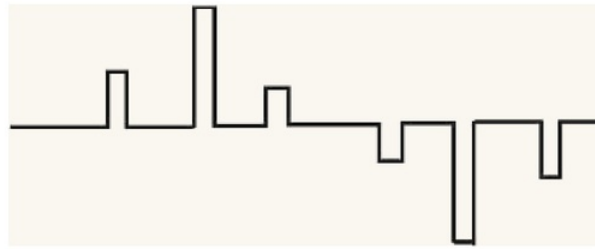


Figure 2-3

There are also two types of sampling techniques used for transmitting a signal using PAM. These techniques are *Natural PAM* and *Flat Top PAM*. Natural PAM has the instantaneous amplitude of each pulse equal that of the instantaneous amplitude of the messenger signal for the entire half cycle of the pulse. This creates a sloping effect for when the pulse is at a high. **Figure 2-4** shows an example PAM waveform using the natural sampling technique.

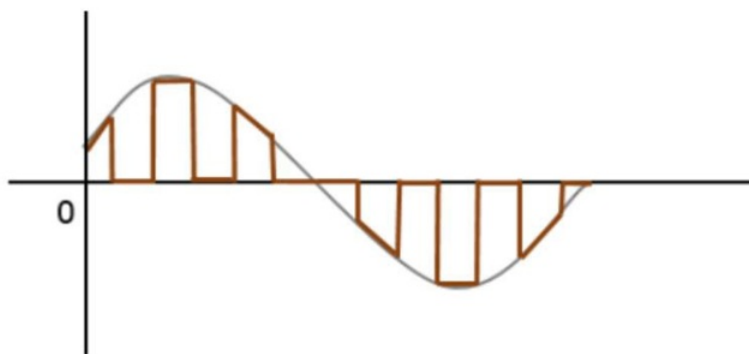


Figure 2-4

Flat top PAM sampling, as its name suggests has each pulse with a flat top. The amplitude of each pulse remains the same for the entire pulse half cycle. This can be seen in **Figure 2-5**.

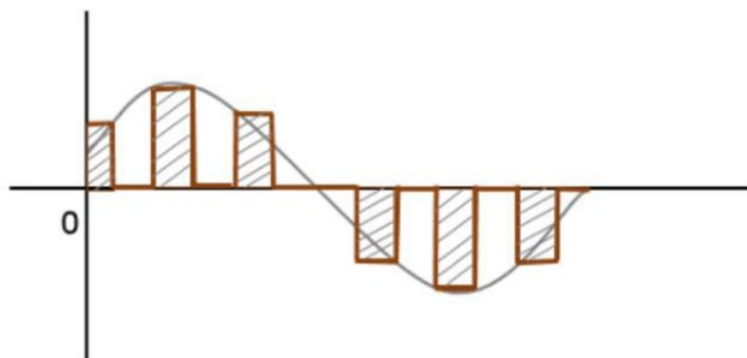


Figure 2-5

A PAM waveform is produced by a sine wave generator and a square wave pulse generator connected to an AND gate. The sine wave generator circuit can be seen in **Figure 2-6**.

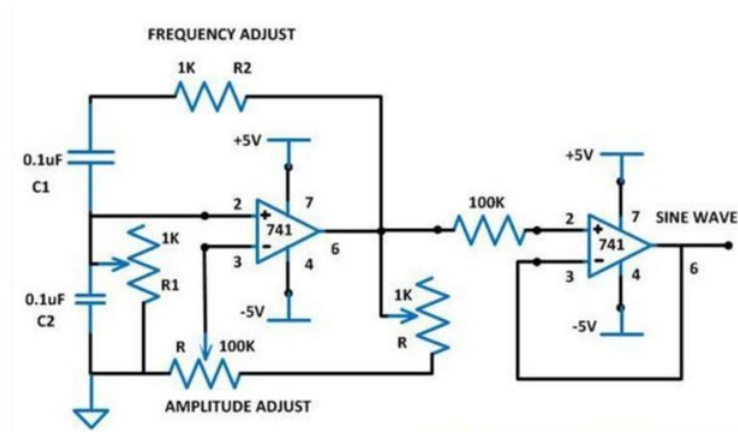


Figure 2-6

The frequency of the output sine wave which will be used as the messenger signal can be altered by changing the value of R1, and the amplitude can be altered by changing the value of R.

$$F = \frac{1}{2\pi\sqrt{R1R2C1C2}}$$

The square wave pulse generator circuit can be seen in **Figure 2-7**.

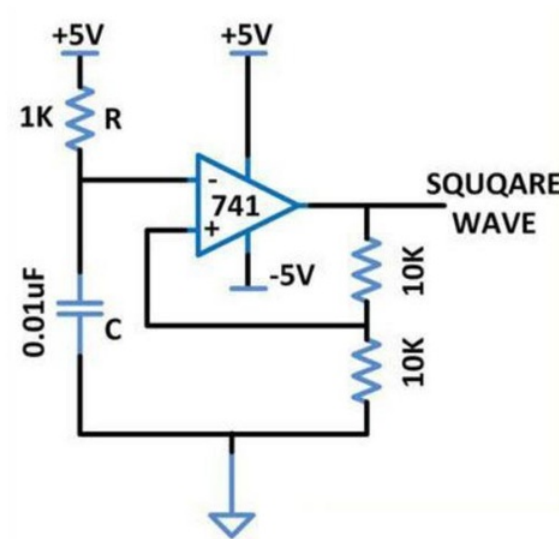


Figure 2-7

The time period of the pulses can be calculated using the following formula:

$$F = 2.2RC$$

The output of these two circuits is connected to an AND gate or PAM modulator circuit. Each time the square wave pulses, both terminals on the and gate become positive allowing the waveforms to become superimposed on each other at the output of the AND gate.

To demodulate a pulse amplitude modulated waveform, the modulated waveform is sent through a low pass filter. The purpose of the low pass filter is to filter out the high frequency ripples. The output of the filter is the demodulated waveform. This signal is then sent through an inverting amplifier which is used to amplify the waveform's amplitude so that it becomes equal to the modulating signal. **Figure 2-8** shows the demodulated output waveform of a PAM signal.

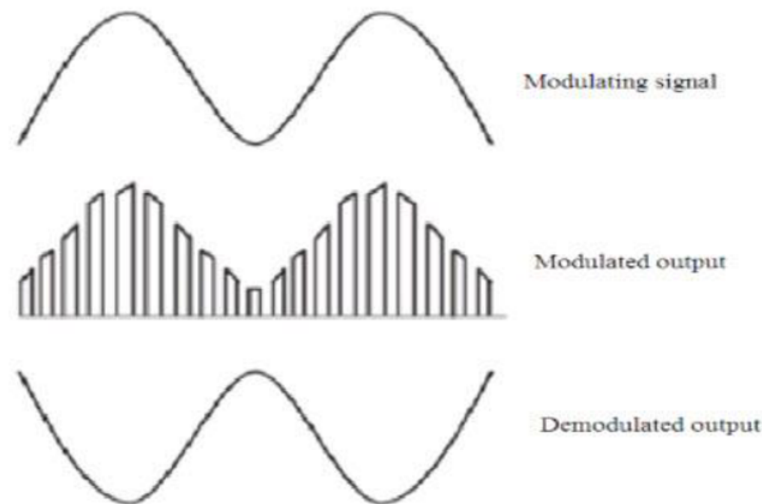


Figure 2-8

There are a few advantages and disadvantages of pulse amplitude modulation. Firstly, it has a simple process for modulating and demodulating. Secondly, PAM can generate other pulse modulated signals such as PWM or PTM and carry the message at the same time. Finally, the circuits for transmitting and receiving the PAM waveforms are simple and cheap.

However, a very large bandwidth is required to transmit PAM waveform. Since a pulse amplitude modulated waveform has varying amplitude, the modulated signal is not immune to noise, and requires more power to be transmitted.

Because of these factors, PAM is suited to a few applications. It is not as commonly used as other techniques due to efficiency and noise control. The applications PAM is used in include; as an electronic driver for LEDs, ethernet connections, generating control signals in micro-controllers and photo-biology.

(ElProCus - Electronic Projects for Engineering Students, 2018)

(Electricalvoice.com, 2018)

2.1.2 PCM (Pulse Code Modulation)

PCM is another technique of analogue to digital conversion which takes the conversion one step further and is in binary form. This means that there are only two possible states, 0 and 1. The

modulating process involves four steps; filtering, sampling, quantisation and coding. This is illustrated in **Figure 2-9**.

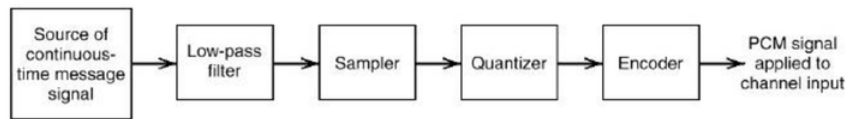


Figure 2-9

2.1.2.1 Filtering

To make the analogue messenger signal easier to convert into digital further into the process, the high frequencies are filtered out. For an audio file of speech, a bandwidth of 3kHz is adequate where the lower cut off frequency is around 200Hz and the upper cut off is roughly 3200kHz. All frequencies above this bandwidth can be filtered out to prevent aliasing when sampled.

2.1.2.2 Sampling

The second step in the above block diagram is the PAM sampler. This converts the filtered continuous sine wave messenger signal into a pulse amplitude modulated signal as discussed in **Section 2.1.1**. This step gets its name as each pulse is called a 'sample'. The frequency at which the samples are taken is called the Sampling rate. "According to the Nyquist Theorem, the sampling rate must be at least $2f_{max}$, or twice the highest analogue frequency component." (Miller, Vandome and McBrewster, 2010)

If the sample rate is much lower than that of the highest frequency of the messenger signal, an overlap between the frequency spectrum of the samples and the input messenger signal is created. When this signal is later passed through a low pass output reconstruction filter, this overlap cannot be recognised which creates a new signal which does not match the original messenger signal. When a false signal is created like this, it is called aliasing. Therefore, it is important to have the correct sample rate by following the Nyquist theorem.

2.1.2.3 Quantisation

The third step in **Figure 2-9** is quantisation which is the process of assigning each sample from the input signal a quantisation interval. Each quantisation intervals are equidistant from each other throughout the range of the input signal. Each sample is assigned to its closest interval based on its amplitude. If a sample's amplitude isn't exactly equal to the value of the interval then it is rounded up or down, and an error called quantisation error is introduced. This error is undesirable as it is the equivalent of reducing the SNR (Signal to noise ratio) of the original signal. The higher the SNR, the less noise is present which means higher quality audio. By introducing quantisation error, the quality of the audio is reduced.

One method of reducing quantisation error is by increasing the number of intervals. By doing this, the difference between the amplitude height of the input signal and the quantisation intervals decreases, meaning a lower error. However, the number of code words is proportional to the number of intervals which can introduce issues such as the capacity of code words a PCM system can handle.

Each interval is assigned an 8bit code word. If the sample rate of a system is 8kHz, and each sample is assigned an 8bit word, the maximum bit rate for this system is 64000 bps

2.1.2.4 Encoding

The final step of the modulation process is the encoder. The encoder encodes the quantised samples. Each sample is encoded using A-law into an 8bit word.

"A-law is an audio compression scheme (codecs) defined by Consultative Committee for International Telephony and Telegraphy (CCITT) G.711 which compress 16-bit linear PCM data down to eight bits of logarithmic data.

Limiting the linear sample values to twelve magnitude bits, the A-law compression is defined by this equation, where A is the compression parameter (A=87.7 in Europe), and x is the normalized integer to be compressed." (Cisco, 2014)

$$F(x) = \begin{cases} \frac{A * |x|}{1 + \ln(A)} & 0 \leq |x| < \frac{1}{A} \\ \frac{\text{sgn}(x) * (1 + \ln(A|x|))}{1 + \ln(A)} & \frac{1}{A} \leq |x| \leq 1 \end{cases}$$

The algorithm assigns each bit in the 8bit word a 1 or 0, each representing a specific property of the sample. Bit 1 being the most significant bit represents the polarity of the sample. It is assigned 1 if the sample is positive, and 0 if the sample is negative. Bits 2 to 4 represent the location of the sample by forming a linear curve for low level negative or positive samples. Bits 5 to 8 being the least significant bits represent a segments quantised value. Each segment quantised value is divided into 16 levels.

(ElProCus - Electronic Projects for Engineering Students, 2015)

PCM can be further categorised into two types; DPCM (Differential Pulse Code Modulation) and ADPCM (Adaptive Differential Pulse Code Modulation). DPCM works on the principle that the difference between a sample and the previous value is encoded. This difference is much smaller than the total sample value, so less bits are needed to be assigned to the sample. By reducing the number of bits for each sample, the bit rate is therefore reduced.

ADPCM uses typical characteristics of analogue signals to estimate the values of previous samples. The potential error in this process is derived using the same method as DPCM. This method reduces the bitrate required by only assigning 4 bits is used to determine the difference between the predicted value and the sample value. However, the method retains 15 quantum levels at half the bitrate.

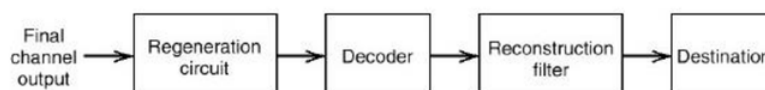


Figure 2-10

Figure 2-10 shows the receiver block diagram. The transmitted signal picks up noise during transmission which can be removed by being sent through the regeneration circuit. This circuit

recovers the signal by using a comparator to bring the received signal back to its original level. The demodulation process is the exact same as the modulation process only in reverse.

The next step is to decode the signal to obtain the original quantisation values of the digital signals. The signal is then sent through a reconstruction filter (low-pass) to remove any high frequency harmonics that might be included with the quantised digital signal.

The advantages of using PCM is that an analogue signal can be transmitted far distances over a high-speed digital system, the signal is much less likely to pick up noise during transmission and the probability of a reoccurring error is reduced using the A-law encoding algorithm.

PCM is used in a wide range of applications such as Telkom systems, digital video and radio control units for remote controlled devices.

2.2 ALIASING

As previously stated, aliasing happens when a signal is sampled at a low sample rate. This is because when a signal is sampled it is band-limited in frequency. This is because when a signal is being converted back from digital to analogue, some frequencies might not be represented as the converter will only operate within a specific frequency range determined by the sample rate.

The converter will interpret any frequencies that lie outside of this range as something else and will map it incorrectly. This is aliasing. As previously discussed the Nyquist theorem can be used to reduce the amount of aliasing by choosing the correct sample rate. The Nyquist frequency is half of the sample rate, and any signal that contains a frequency higher than the Nyquist frequency is mapped by the converter.

To avoid this, a component called an anti-aliasing filter is used to filter out frequencies that exceed the Nyquist frequency before the signal reaches the converter. This stops these frequencies from being incorrectly converted and mapped which causes aliasing.

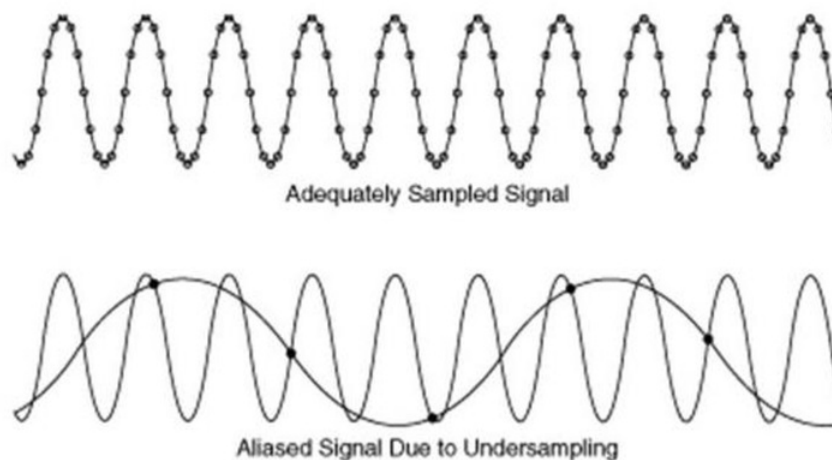


Figure 2-11

Figure 2-11 shows two waveforms, the top waveform has the correct sample rate which creates an output that resembles that of the original messenger signal while the bottom waveform has a much lower sample rate which creates an output that isn't even close to that of the input. This

is an example of severe under sampling. **Figure 2-12** shows more examples of under sampling causing aliasing.

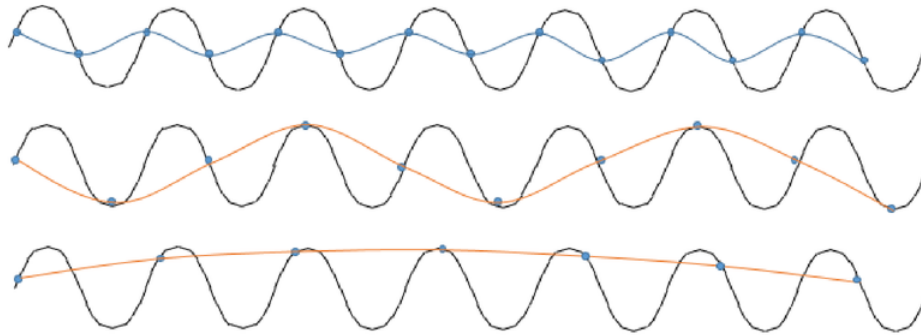


Figure 2-12

(Tarr, 2017)

2.3 MODULATION SCHEMES

The main objective of modulation in communication systems today is to contain as much data into a small spectrum as possible. By increasing the amount of data able to be sent within a certain spectrum, the “spectral efficiency” is increased. “Spectral efficiency is the measure of how fast data can be transmitted through a specific bandwidth” (Frenzel, 2018). Since there is a limited amount of spectrum available for data to be transmitted through, multiple methods of modulation have been developed to improve spectral efficiency. These include:

- ASK (Amplitude Shift Keying)
- OOK (On Off Keying)
- FSK (Frequency Shift Keying)
- PSK (Phase Shift Keying)
- BPSK (Binary Phase Shift Keying)
- QPSK (Quadrature Phase Shift Keying)
- QAM (Quadrature Amplitude Modulation)

The three most basic ways to modulate a carrier wave is by modulating the amplitude, frequency or phase. More advanced methods combine these methods to improve spectral efficiency.

2.3.1 ASK

ASK (Amplitude Shift Keying) is a method of modulation that shifts the amplitude of the waveform between various amplitudes. For a simple binary signal an ASK modulated waveform would have two levels. This can be seen in **Figure 4-16**.

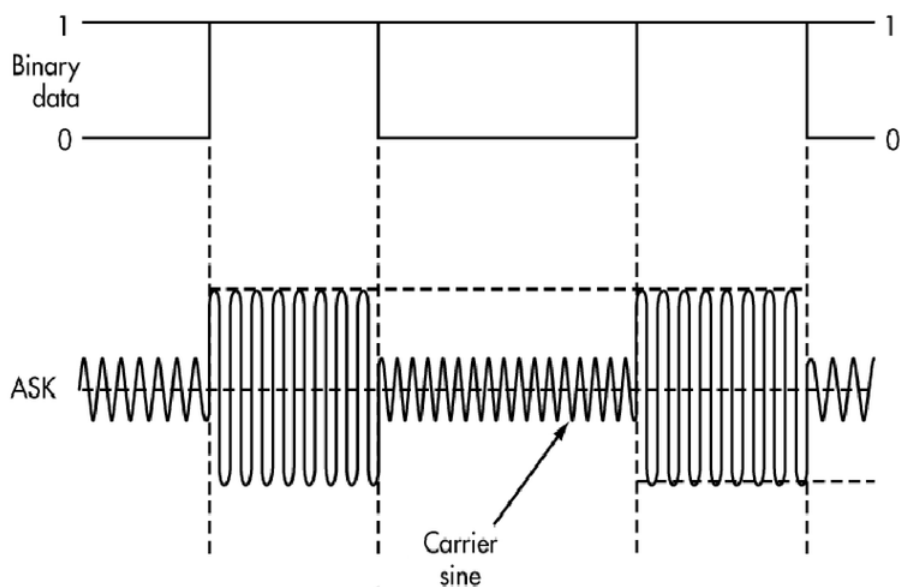


Figure 2-13

As **Figure 2-13** shows, when the binary level is 0, the amplitude of the carrier sine is low and when the binary level is 1 the amplitude of the carrier wave is high.

2.3.2 OOK

OOK (On Off Keying) is another method of amplitude modulation. It operates on the principle that when a binary level is 0, the waveform is in an 'off' state where it has 0 amplitude. When the binary data level is equal to 1 the waveform is in an 'on' state and has an amplitude equal to the carrier wave. This can be seen in **Figure 2-14**

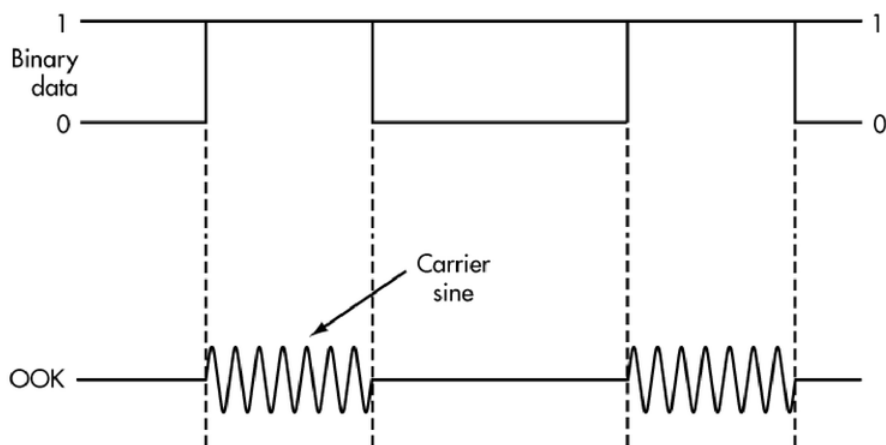


Figure 2-14

Amplitude modulation creates sidebands above and below the maximum frequency of the modulating signal on the carrier wave. This means that a bandwidth of two times the maximum frequency is required. The frequency remains the same during amplitude modulation.

2.3.3 FSK

FSK (Frequency Shift Keying) is the modulation method of shifting the carrier between different frequencies while the amplitude remains the same. This can be seen in **Figure 2-15**.

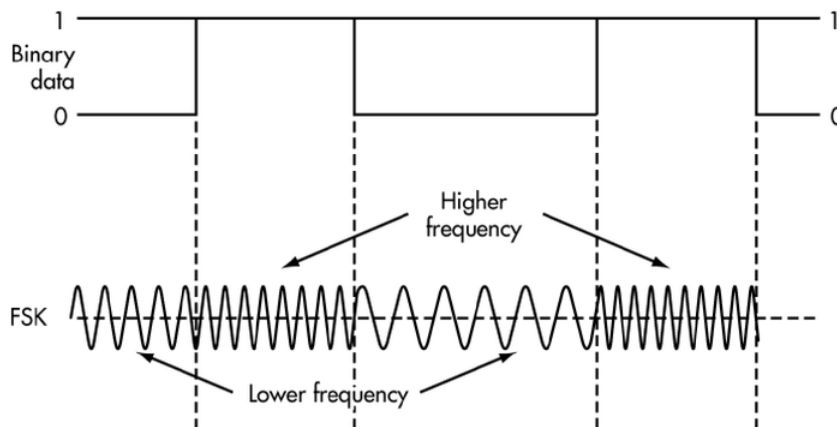


Figure 2-15

Frequency modulation creates sideband frequencies above and below the carrier frequency. The bandwidth that is created during frequency modulation is a function of the highest modulating frequency, including any harmonics and the modulation index. This can be seen below.

$$\text{Modulation Index} = \Delta f(T)$$

The variable T represents the bit time interval of the data, or the reciprocal of the data rate. This is measured in bit/s. Δf is the frequency deviation which is the shift between the two frequencies. These frequencies are called the mark and space frequencies, often labelled f_m and f_s respectively. This can be calculated as shown below.

$$\Delta f = f_s - f_m$$

By reducing the modulation index, the number of sidebands is also reduced. This has brought about a method of modulation called MSK (Minimum Shift Keying) which aims to use very small values for the modulation index.

One method of increasing spectral efficiency for ASK, OOK and FSK is called coherent operation. The objective of this method is to get the instantaneous changing of the binary value to line up with the sine wave crossing the horizontal axis making its value 0. To do this, the correct data rates, carrier frequencies and shift frequencies are needed to be selected.

By using coherent operation, fewer harmonics are generated and the bandwidth for all three modulation techniques is reduced. This increases spectral efficiency as a less of the frequency spectrum is required for transmission.

(Frenzel, 2018)

2.3.4 PSK BPSK

PSK (Phase Key Shifting) is the process of modulating a signal by changing its phase. BPSK (Binary Phase Shift Keying) is a method of modulation that shifts the carrier wave 180 degrees each time the binary state changes. This is illustrated in **Figure 2-16**.

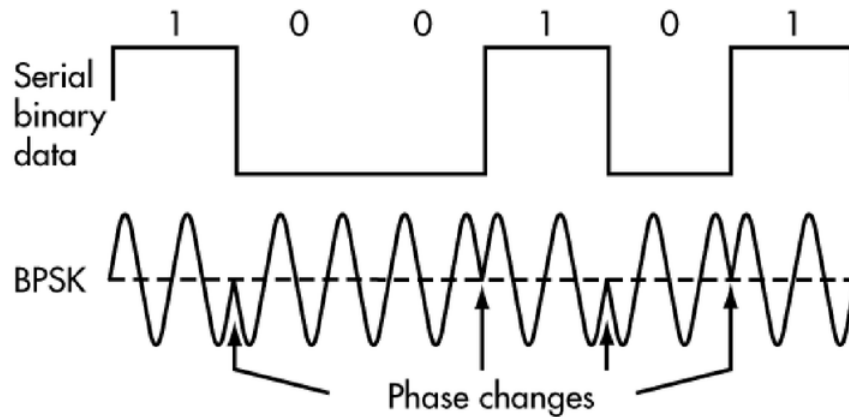


Figure 2-16

As shown, the phase changes when the binary state switches. This means that the signal is coherent and can later be demodulated by comparing the modulated signal to a sine carrier of the same phase. This process requires complex circuitry. BPSK is very spectral efficient as “data can be transmitted at a data rate equal to the bandwidth, or 1bit/Hz.” (Frenzel, 2018)

2.3.5 QPSK

QPSK (Quadrature Phase Shift Keying) is a variation of BPSK in which the modulator produces two carrier sine waves with a phase difference of 90 degrees from each other. When the two carrier waves are modulated by a binary signal, four unique sine signals are produced 45 degrees from each other. The two phases are then superimposed on to one another to produce the final signal. The table below shows the carrier phase shift for each pair of bits.

Bit Pairs	Phase (Degrees)
00	45
01	135
11	225
10	315

From this table a phasor diagram can be produced where the phasor represents the amplitude of the carrier wave, and its position is the carrier’s phase. This can be seen in **Figure 2-17**.

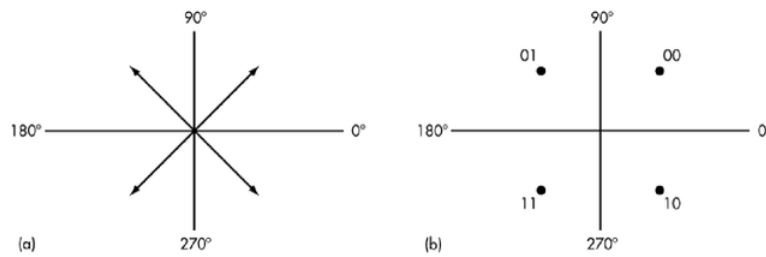


Figure 2-17

In comparison to BPSK, QPSK is much more spectral efficient as it can achieve twice the data rate in the same bandwidth as BPSK. This is because each carrier phase represents two bits of data rather than one.

2.3.6 QAM

QAM (Quadrature Amplitude Modulation) is a combination of amplitude and phase modulation. It operates similarly to QPSK, however instead of one point per quadrant of the phasor diagram, there are multiple which are determined by different phase carriers each with different amplitude levels. There are many variations of QAM such as 8QAM, 16QAM, 64QAM and 256QAM. The higher the number, the more bits can be transmitted. In 8QAM, there are four carrier phases and two amplitude levels. This allows 3 bits per symbol to be transmitted. **Figure 2-18** shows a phasor diagram for 16QAM.

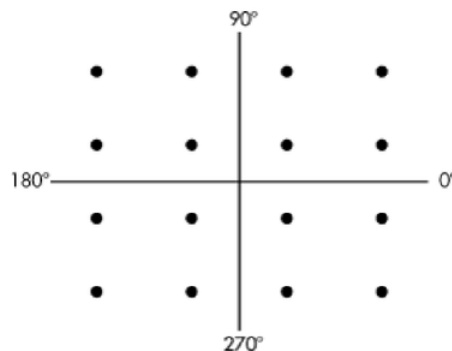


Figure 2-18

16QAM can achieve 4bits/Hz using twelve phase shifts and three amplitudes. QAM is extremely spectral efficient as high data rates can be achieved within a smaller bandwidth. However, due to the fact this method using amplitude modulation, it is susceptible to noise making it potentially very difficult to demodulate.

QAM is often used in modern communication systems such as TV, satellites, mobile phones, and internet as it can produce such high data rates in limited bandwidths.

3 - EXPERIMENTAL METHOD

This section will contain the setup for the practical. The aim of the practical is to investigate the limits of an ADC and the affect it has on the output waveform when certain parameters are changed. The sample rate of the ADC is 10000 samples/second and the range is 5Vp-p.

The experiment setup includes a function generator connected to an ADC, which is then connected to a Picoscope on the output. The function generator is used to send sine waves of various frequency and amplitude to the ADC. The ADC then converts the analogue signal into a digital one which then gets measured and displayed by the Picoscope.

The practical is split into three main sections; changing the number of bits of the ADC, changing the input amplitude of the sine wave and changing the frequency of the sine wave.

3.1 CHANGING THE NUMBER OF BITS OF THE ADC

This section of the practical sees the effect of changing the number of bits of the ADC has on the output waveform. The ADC should initially be set to 8 bits. The input frequency on the function generator should be set to 100Hz at 4.2Vp-p and the output should be recorded.

Following this, the ADC should be set to 4 bits and the input should remain the same. The output should again be recorded and compared with the prior results.

An expected result would be to see a less smooth curve with more 'steps' when the number of bits on the ADC is reduced. These results can be found in **Section 4.1**.

3.2 CHANGING THE AMPLITUDE OF THE SINE WAVE

This section of the practical sees the effect of changing the amplitude of the input sine wave when sent through the ADC at 4 bits, while keeping the frequency the same. The frequency is kept at 100Hz and the Vp-p is initially set to the minimum value the function generator can output. The output is then recorded. An expected result of this would be to see two quantisation levels with large quantisation error.

The amplitude is then gradually increased to exceed the maximum range of the ADC which is 5Vp-p. An expected result would be to see more quantisation levels appear as the amplitude is increased. Once the amplitude exceeds that of the ADC's range the output waveform should flat-line and no more quantisation levels are added.

Finally, the amplitude is once again reduced to show three quantisation levels and the peak to peak voltage is recorded. The amplitude is increased to show four quantisation levels and the peak to peak voltage is again recorded. This is done one more time for five quantisation levels. The resolution of the ADC should be calculated and compared to the measurements. An expected result would be the number of quantisation levels multiplied by the resolution should equal the peak to peak voltage.

3.3 CHANGING THE FREQUENCY OF THE SINE WAVE

The final section of the experiment involves changing the frequency of the input sine wave while keeping the amplitude the same. The ADC is to be set to 8 bit, the frequency is initially set to

100Hz and the amplitude is initially set to 4Vp-p. The frequency of the function generator is then increased in decades. The measurements are recorded each time the frequency is increased.

The Frequency is then increased to 6kHz and the output is recorded. The final measurement is when the frequency of the input sine wave is 10kHz. An expected result would be to have the output waveform accurately replicate the input at low frequencies, and as the frequency increases the output would become more distorted. The difference between the frequency of the output waveform and the frequency of the input waveform will increase. The output waveform will start to show aliasing as the frequency gets higher and eventually will not resemble the input waveform at all.

4 – TESTING AND RESULTS

This section will contain all the results and evidence from the practical experiment along with an analysis of each section.

4.1 RESULTS OF CHANGING THE NUMBER OF BITS ON THE ADC

The input sine wave for this section is set to 100Hz at 4.2Vp-p and the number of bits on the ADC is initially set to 8 bits.

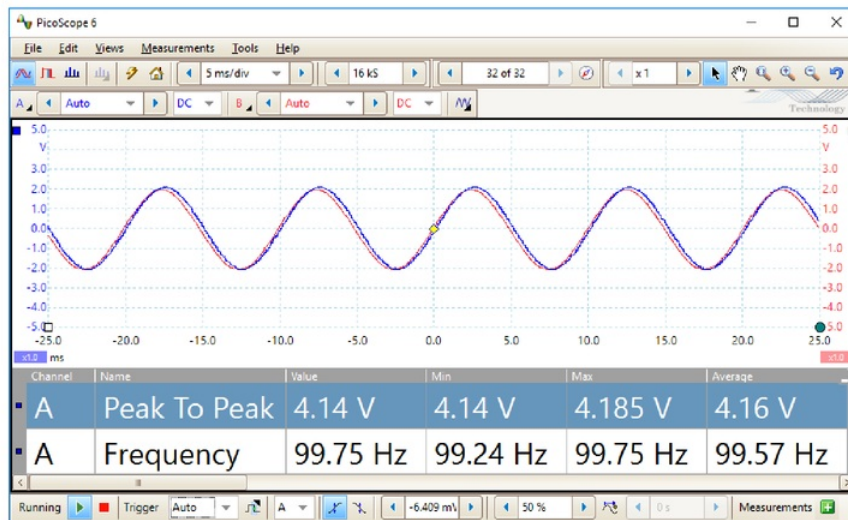


Figure 4-1

The number of bits is then changed to 4 bits.

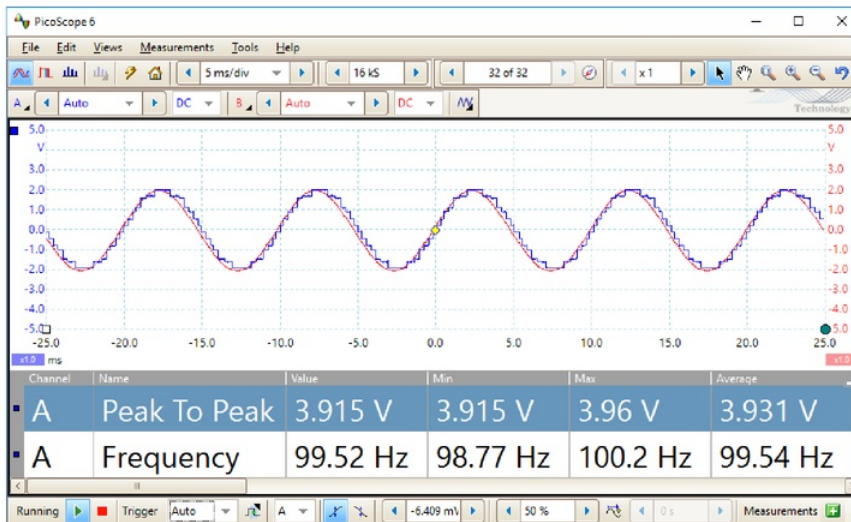


Figure 4-2

As expected, the output waveform gets a lot more distorted when the number of bits on the ADC is reduced. This is because the ADC has a lower resolution when the number of bits is reduced. A 4 bit ADC has a resolution of:

$$\frac{5V}{2^4} = 312.5mV$$

This means that there will be 16 levels equal to 312.5mV each. For an 8 bit ADC the resolution is:

$$\frac{5V}{2^8} = 19.5mV$$

The above calculation shows that instead of 16 levels, there are 256 each representing 19.5mV. These two calculations prove that as the number of bits is increased on the ADC, the precision is also increased. The amplitude and the frequency remain the same however.

4.2 RESULTS OF CHANGING THE AMPLITUDE OF THE INPUT SINE WAVE

The function generator is initially set to its minimum output voltage which happens to be 333mV. The frequency is kept at 100Hz and the ADC is set to 4 bits.

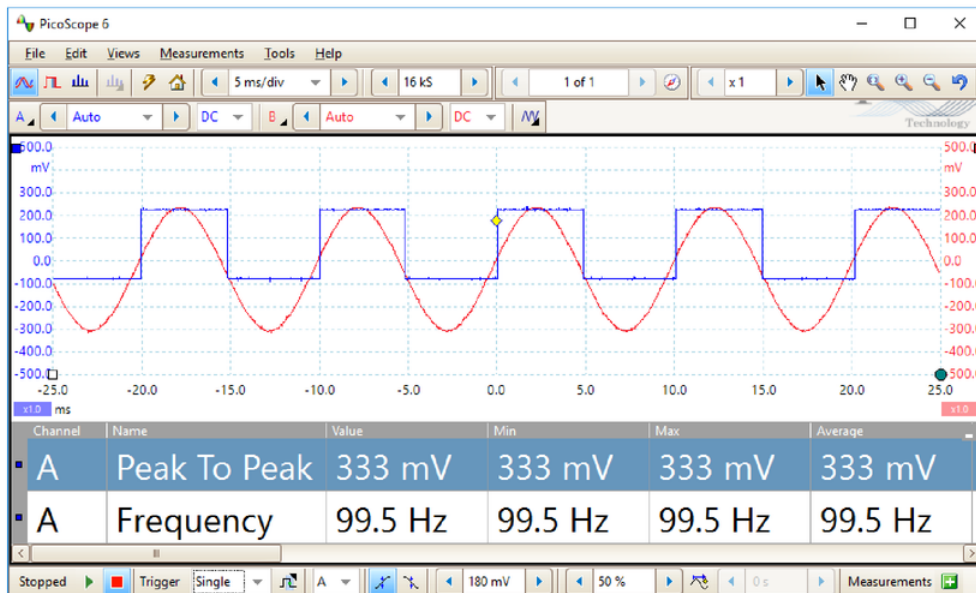


Figure 4-3

As expected there are only two quantisation levels and there is large inaccuracy caused by quantisation error. The output is no longer a sine wave and is instead a square wave. This is because the resolution of the ADC is 312.5mV per quantisation level, and the peak to peak voltage is 333mV. Therefore there are only two quantisation levels, 0V and 312.5mV.

The next part of this section involved increasing the amplitude to exceed the ADC's range of 5V. This can be seen in **Figure 4-4**.

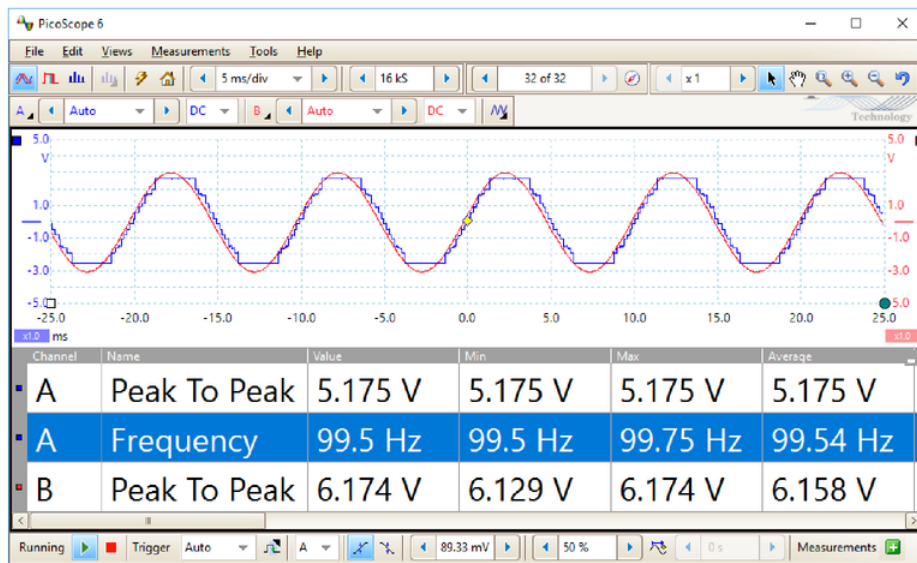


Figure 4-4

As expected, **Figure 4-4** shows many more quantisation levels when the amplitude is increased. The amplitude of the input is set to 6.174Vp-p as seen on channel B. However, the output waveform shows that the amplitude is cut off at 5.175V by the ADC. This causes a flat-line in the output waveform.

The final part of this section is to see the difference between the quantisation levels of the output. **Figure 4-5** shows the input amplitude set to 679.5mV which causes the output waveform to have three quantisation levels.

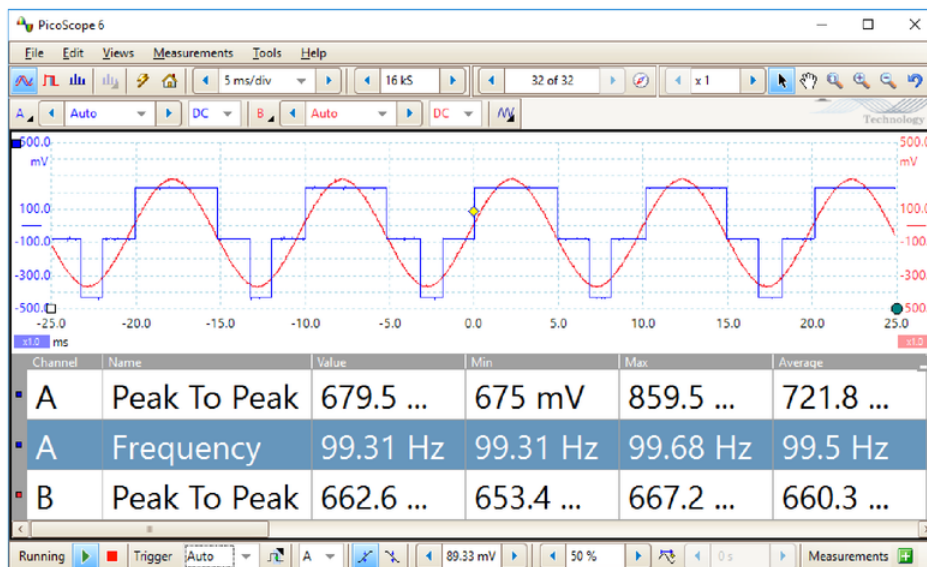


Figure 4-5

Figure 4-6 shows the output waveform with four quantisation levels after the input amplitude is increased.

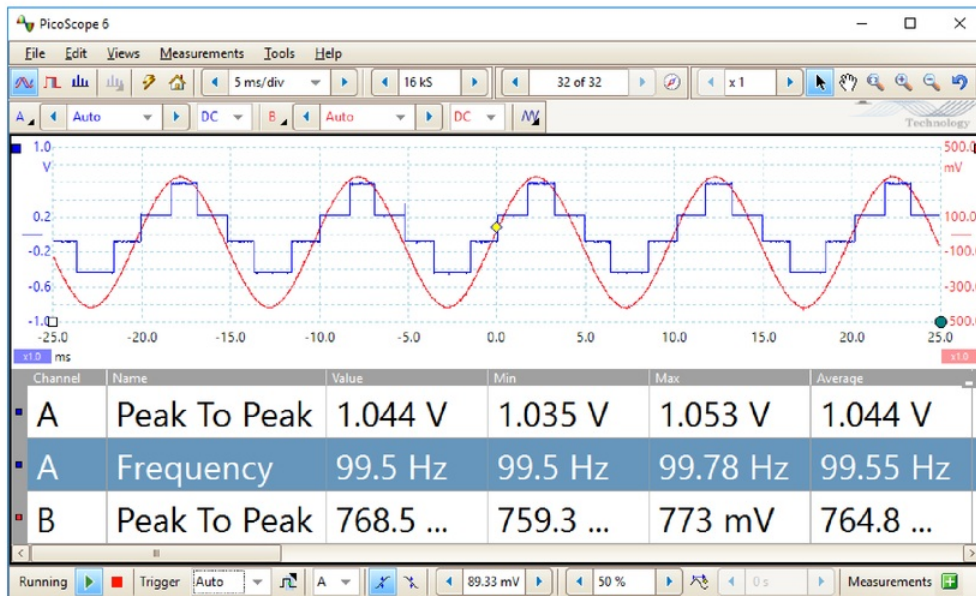


Figure 4-6

Figure 4-7 shows the output waveform with five quantisation levels after the input amplitude is increased.

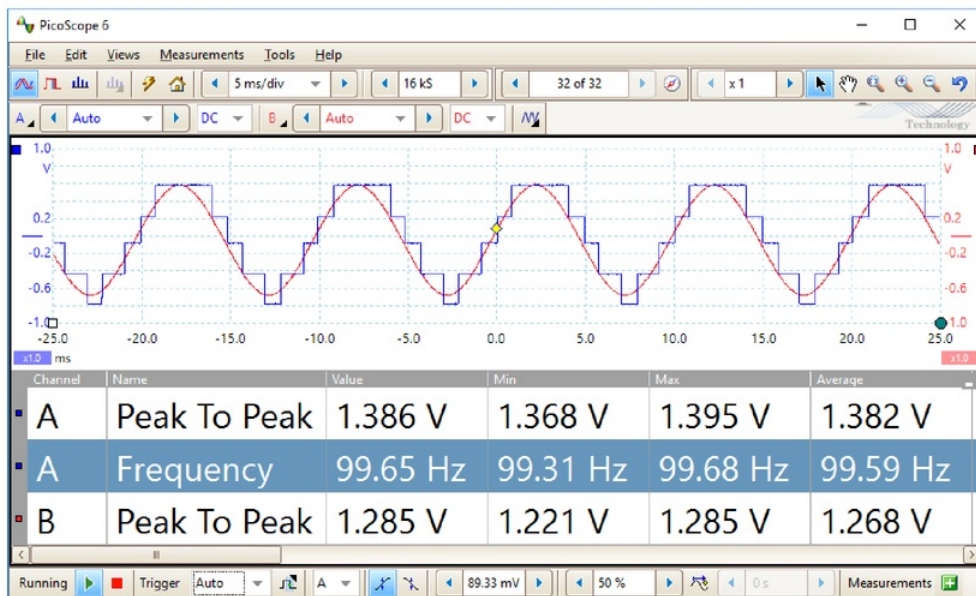


Figure 4-7

As previously calculated, the resolution of this 4 bit ADC is 312.5mV. Therefore each quantisation level should represent 312.5mV. For three quantisation levels, the input amplitude was set to 679.5m. The output from the ADC was 662.6mV. One of the quantisation levels will always represent 0V, meaning the other two represent 312.5mV each.

$$2 \times 312.5mV = 625mV$$

This is relatively close to the original amplitude of the sine wave.

For four quantisation levels the input amplitude is increased to 1.044V which creates an output voltage of 768.5mV. Right away it can be seen that this value is far from what it should be. Using the same method of calculating what the output should be for four quantisation levels:

$$3 \times 312.5mV = 937.5mV$$

The fact that the Picoscope is reading 768.5mV shows that there is something causing inaccuracy in the reading.

Finally, for five quantisation levels the input amplitude is increased to 1.386V which makes the output amplitude equal to 1.285V. Once again the sum of the quantisation level voltages is calculated:

$$4 \times 312.5mV = 1.250V$$

This is a relatively accurate value to get since the original amplitude was 1.386V. The difference between the original and the calculated value is 136mV.

The discrepancies between the measured values and the calculated values could be because of calibration error with the Picoscope or function generator, quantisation error due to the sample rate of the ADC or imperfections with the ADC.

4.3 RESULTS OF CHANGING THE FREQUENCY OF THE INPUT SINE WAVE

The first test is to see measure the output waveform at 100Hz which can be seen in **Figure 4-8**.

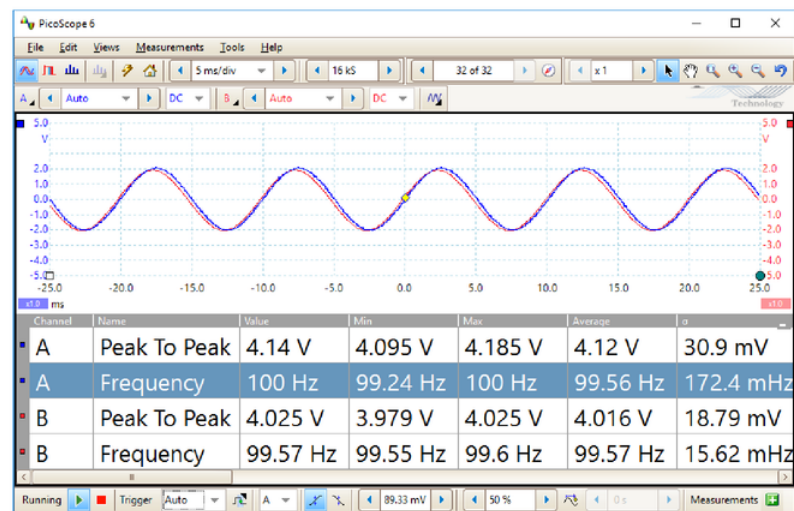


Figure 4-8

As expected the output waveform is accurately replicating the input. There is a slight phase shift due to the time taken to sample the input waveform. This could be fixed with a faster sample rate. The frequency of both input and output waveforms are the same.

Figure 4-9 shows the waveforms when the frequency is increased to 1 kHz.

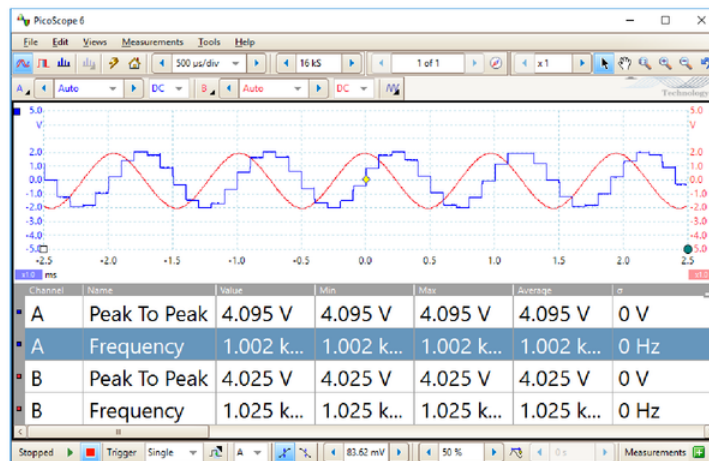


Figure 4-9

Right away a few differences can be noted. Firstly, the phase has shifted even further due to the sampling. The output waveform has become a lot less smooth and the quantisation levels are easily seen. This is because of the sample rate. At a sample rate of 10000 samples /s and a frequency of 1 kHz, a single cycle can only be sampled 10 times. This can be proved by this calculation and by counting the steps in 1 full cycle of the output waveform.

$$\frac{1}{1\text{kHz}} \times 10000 = 10$$

The frequency of the input has also drifted 23Hz from the original messenger wave. **Figure 4-10** shows the waveforms at 2 kHz.

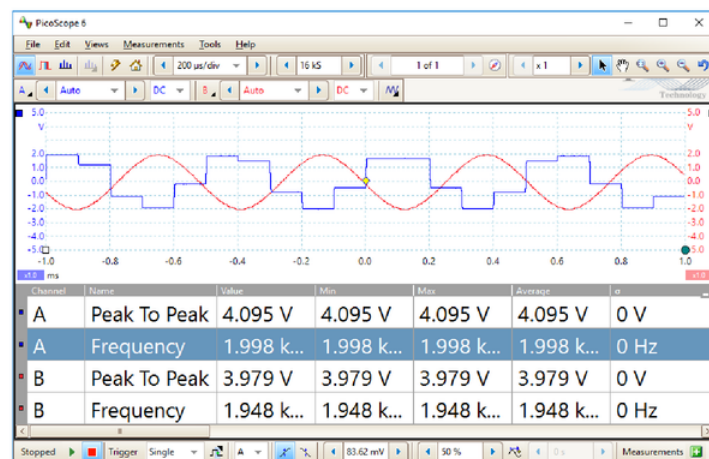


Figure 4-10

The new waveform has phase shifted 180 degrees from the original and has started to not resemble the original waveform. The frequency has deviated by 50Hz this time. Each cycle is only sampled 5 times which can be proved by:

$$\frac{1}{2\text{kHz}} \times 10000 = 5$$

The fact that the amount of samples per cycle decreases as the frequency increases proves that the integrity of the original waveform is being lost due to the sample rate.

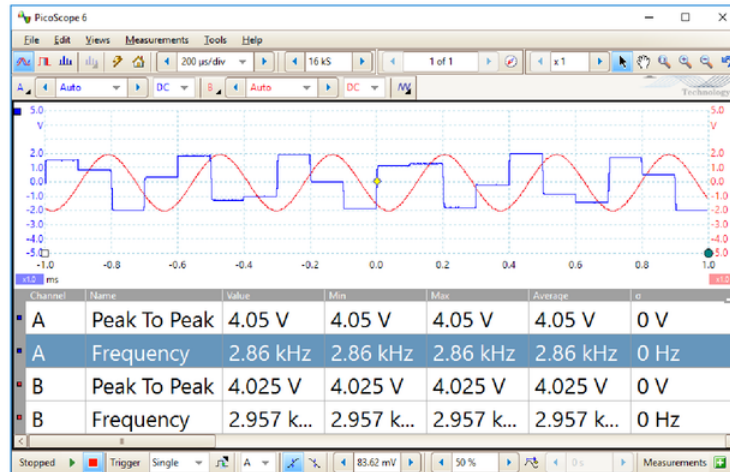


Figure 4-11

Figure 4-11 shows the waveforms at 3 kHz. The output waveform hardly resembles the original waveform anymore as there are only 3.3 samples per cycle. It does however still almost resemble a sine wave. The magnitude of the amplitude of the original sine wave is maintained, and the frequency is only slightly off.

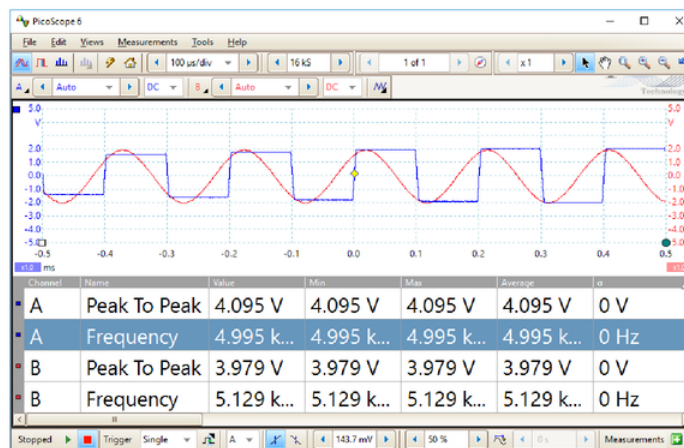


Figure 4-12

Figure 4-12 shows the frequency set to 5kHz. This is the Nyquist frequency of this system as the ADC operates at 10000 samples per second. This shows as there are only 2 samples per cycle of the waveform. The waveform has become a square wave instead of a sine wave, however the peak-peak voltage and frequency of the output is still relatively close to that of the original waveform. This is expected to be the final waveform that retains these qualities.

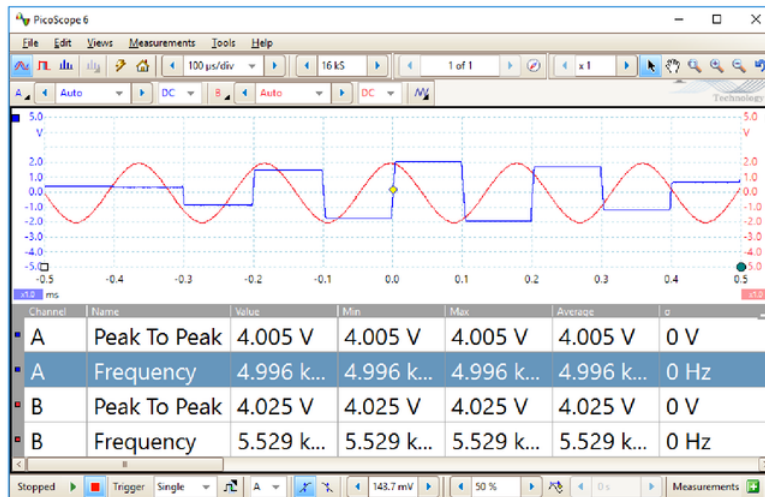


Figure 4-13

Figure 4-13 shows the results when the frequency is increased to the point where the picoscope can no longer accurately identify the frequency of the original input wave. The output waveform has become distorted from the previous square wave.

The input frequency is 5.5kHz however the picoscope reads it as 5kHz. As the frequency is further increased, the same trend in results is expected to happen.

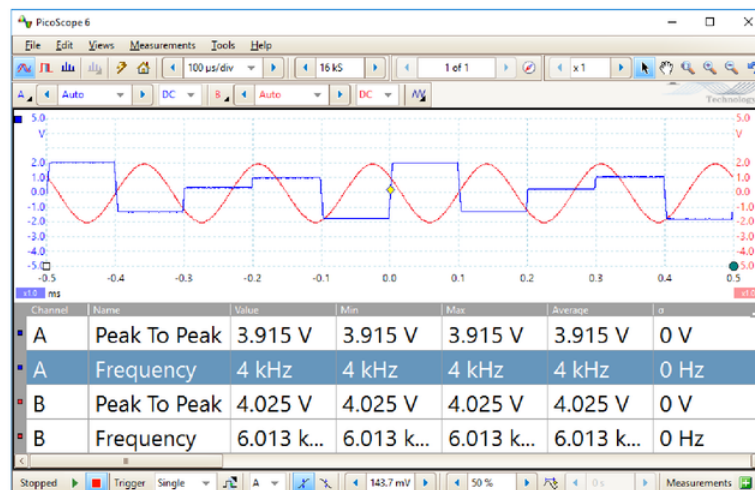


Figure 4-14

Figure 4-14 shows the waveform when the frequency is increased to 6kHz. The oscilloscope reads 4kHz which is a huge discrepancy. The waveform hardly even resembles an alternating square wave anymore.

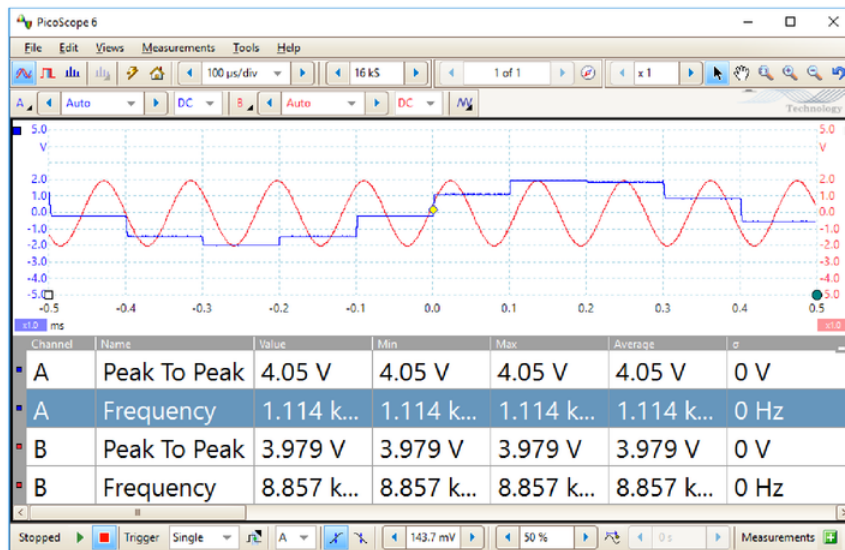


Figure 4-15

When the frequency is increased to 8.857kHz, the oscilloscope reads 1.114kHz on the output. This distortion has created a new sine wave which can be seen by the output waveform. This new sine wave has a frequency of 1.114kHz. Finally, the frequency is increased to 10kHz and can be seen in **Figure 4-16**.

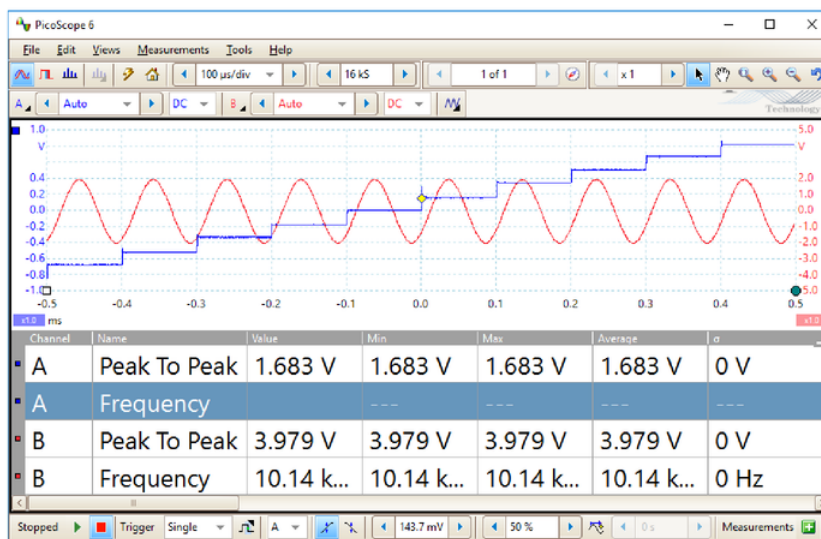


Figure 4-16

Figure 4-16 shows the waveforms at 10kHz. The output waveform is completely distorted as there is only 1 sample per cycle. The oscilloscope cannot determine the frequency of the input waveform at all and the amplitude can no longer be read. This waveform is completely aliased and is of no use.

These results were as expected as according to the research, the sample rate needs to be at least 2 times the f_{max} of the input (Nyquist frequency). After exceeding the Nyquist frequency the waveforms become aliased and lose the characteristics of the original waveform meaning they are no longer transmitting the correct data.

5 – CONCLUSIONS AND RECOMMENDATIONS

Since digital signals are widely used in modern communication systems, accurate analogue to digital conversion is a requirement. Using PCM analogue signals can be converted accurately into digital signals ready for transmission. These signals can be interpreted and converted back into analogue signals if the correct sample rate is selected for the frequency of the messenger signal.

If an incorrect sampling rate is used, distortion is introduced in the form of aliasing which is when a frequency is incorrectly mapped by a converted. This is unwanted and can be fixed by using a low-pass filter to filter out any unwanted frequencies.

Signal modulation is done using a wide variety of methods to improve what's called spectral efficiency. Good spectral efficiency is a requirement for communication systems as sending more data in a lower bandwidth requires less energy and has increased speeds. The three basic types of modulation are amplitude, frequency and phase. These variations can be combined to create much more spectral efficient methods of modulation such as QAM.

A practical experiment was carried out to see the effect of changing the number of bits on the ADC, amplitude of the input signal and frequency of the input signal would have on the output waveform.

Firstly, by increasing the number of bits on the ADC the resolution was increased. This made the output waveform much more precise as more quantisation levels were obtained. When increasing the amplitude, it could be seen that the resolution of the ADC becomes a lot more noticeable at lower frequencies as the output waveform became a square wave rather than a sine wave. Another effect of increasing the amplitude was that once the input signal exceeded the voltage rating of the ADC, the amplitude was cut off on the output and created a flat line. This is considered a form of quantisation error as there are no more quantisation levels available for the ADC to assign.

Finally, when changing the frequency, it was noted that as the frequency was increased, there were less samples per cycle. This is because the sample rate of the ADC was 10000 samples per second making it limited to the number of samples it could achieve. This made the quantisation levels a lot more noticeable and eventually cause aliasing when the frequency exceeded the Nyquist frequency.

An improvement to this experiment could be to have a higher resolution ADC which has a higher voltage rating and sample rate to test along with higher amplitudes and frequencies. The output waveform could also be reconstructed and tested to see where exactly the integrity is lost.

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7 – APPENDICES

Sam Harries - Signals and Comms Assignment 2

GRADEMARK REPORT

FINAL GRADE

GENERAL COMMENTS

Instructor

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GRADING FORM: CSG-RUBIC

DAVID HARRIES

87

RESEARCH-INFORMED LITERATURE



There was evidence of extensive further reading and research from highly credible and reliable sources. Your interpretation and understanding of the additional research was excellent and demonstrated a deep underlying appreciation of the subject. Your references were appropriate to the task and your reference list was of an appropriate length for this assignment.

9

KNOWLEDGE AND UNDERSTANDING

The knowledge and understanding of the subject



demonstrated by this assignment was generally excellent. There were very few gaps in your knowledge of the basic principles and the higher level concepts were very well explained. Your explanation of the design process and calculations was also excellent and it did extend your knowledge well beyond that contained in the course notes.

28

ANALYSIS



Your analysis of the task was excellent, both in class and as shown in this assignment. You have demonstrated very good analytical skills and your approach to the task was very logical. You carried out a detailed evaluation of the results and included an excellent conclusion relating your practical results to accepted theory.

18

PRACTICAL APPLICATION AND DEP.



Your application of practical skills for this task was generally excellent. You used the resources available appropriately when conducting the experiment and you used appropriate options for testing and evaluating the circuit performance. . Your results were very detailed, well presented and allowed for an excellent conclusion to be attempted.

24

SKILLS FOR PROFESSIONAL PRAC.



In this assignment you have demonstrated skills in working individually and also collaboratively. Your written report was very well structured and written in a coherent and logical way. The report was presented in a professional manner using appropriate language and formatting, good

8

diagrams in appropriate places, good
sentence/paragraph structure and clear calculations