Faculty of Engineering and Technology
Department of Electrical and Computer Engineering

ENEE 4103- COMMUNICATIONS LAB
Lab Assignment

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May 15, 2022

Abstract

The aim of this assignment was to understand the pulse amplitude modulation (PAM) and demodulation systems along with transmitted noise.

The assignments included the generation of data using a random integer generator, transmitting them through a M-PAM modulator baseband unit, adding additive white Gaussian noise (AWGN) during this transmission, and finally, decoding the received corrupted signal using the M-PAM demodulator baseband unit. Moreover, the error between the generated and decoded data was calculated using an error rate calculation unit and the results were further analyzed.

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Theory

Digital Communication:

Signals are used to communicate in our daily lives. In general, these signals, such as sound signals, are analog in nature. When communication over a long distance is required, analog signals are transmitted through wire utilizing several ways for successful transmission.

There is a process called **digitization**, which is the digitization of signals and their conversion from analog to digital signal, and the importance of this process lies in reducing losses such as distortion, interference, and other losses, including security breaches, that can occur when utilizing traditional communication techniques such as analog signals for long-range communications, as this mechanism has been utilized to address these difficulties and by which signals. They are numbered in a variety of ways. Digital signals enable for clearer, more precise, and lossless communication.

Figure 1: Diffirence betwen analog and digital signal

Pulse Amplitude Modulation (PAM):

The amplitude of the pulse carrier is proportional to the instantaneous amplitude of the message signal in the Pulse Amplitude Modulation (PAM) method.

As the signal follows out the route of the entire wave, the pulse amplitude modulated signal will follow the original signal's amplitude. A signal recorded at Nyquist rate can be reconstructed using natural PAM by running it through an efficient Low Pass Filter (LPF) with specific cutoff frequency.

The Pulse Amplitude Modulation (PAM) is explained in the following figure below:

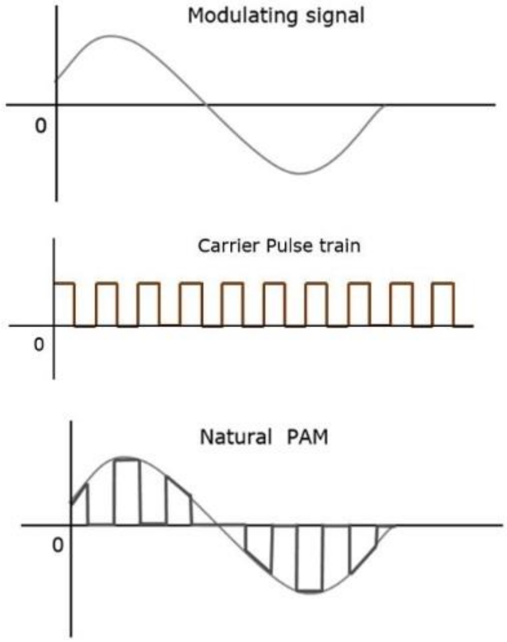


Figure 2: PAM Modulation

Despite the fact that the PAM signal is transmitted via an LPF, the signal cannot be recovered without distortion. As a result, employ flat-top sampling to avoid noise. The flat-top PAM signal is represented in the figure below:

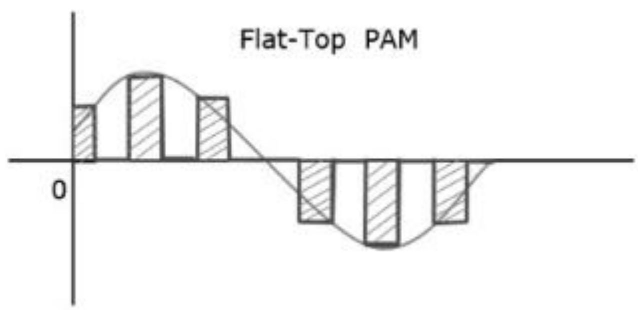


Figure 3: Flat-Top Modulation

Flat-top sampling is a method of representing a sampled signal in pulses in which the amplitude of the signal cannot be modified in relation to the analog signal being sampled. The amplitude tops stay flat. The circuit design is simplified as a result of this technique.

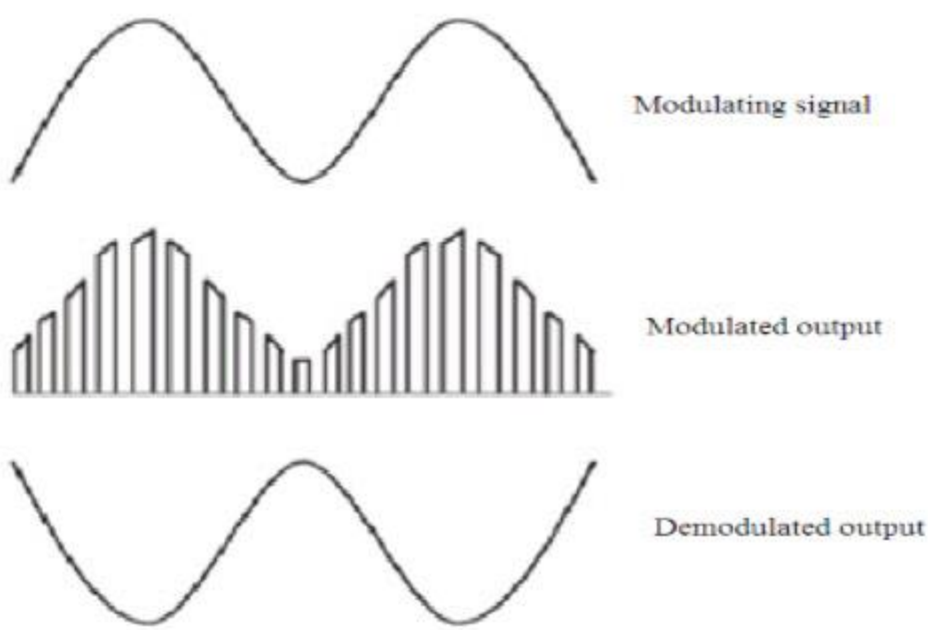
Generation of Pulse Amplitude Modulation (PAM) Signal:

PAM, or pulse amplitude modulation, is the most basic kind of pulse modulation, in which the signal is sampled at regular intervals and each sample is produced relative to the modulating signal's amplitude at the sampling point.

The PAM signal is produced by a sampling device with two inputs: the sampling or carrier signal and the modulation signal, with the amplitude of the signal being connected to the modulation signal by the data transmission location. As a result, this is the PAM signal. The carrier train of the signals is plotted using the waveform inside the time domain, where the spectrum of the PAM signal is created from the message and the sampling signals.

PAM Demodulation:

The PAM signal is fed to the low pass filter for demodulation. The demodulated signal is generated after the high-frequency ripples are removed by the low pass filter. This signal is then sent into an inverting amplifier, which amplifies it to the point where the demodulated output has almost the same amplitude as the modulating signal.



Demodulation of PAM signal

Figure 4: Demodulation of PAM signal

Additive white gaussian noise (AWGN):

A simple noise model used in information theory to mimic the effect of many random events that occur in nature is additive white Gaussian noise (AWGN).

Because of the following reasons, the term additive white Gaussian noise (AWGN) was formulated:

Additive: The noise is additive, which means that the received signal equals the transmitted signal plus the noise. In communication systems, this is the most widely utilized equality.

$$r(t) = s(t) + w(t)$$

as shown in the figure below. Therefore, statistically speaking, this noise is unrelated to the signal. Remember that the above equation is oversimplified since it ignores every single defect a Tx signal encounters, with the exception of noise.

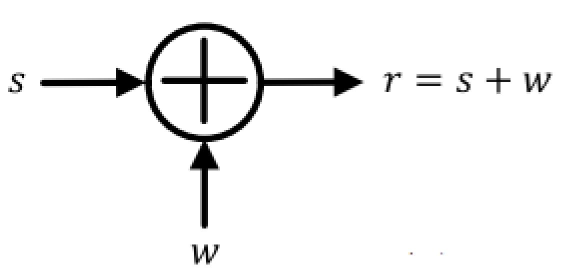


Figure 5: Adding noise to signal

White: White noise refers to the idea that it has consistent power over the entire frequency band, just as the white color, which is made up of all frequencies in the visible spectrum. As a result, white noise's Power Spectral Density (PSD) remains constant at all frequencies between $-\infty$ and $+\infty$, as seen in Figure below:

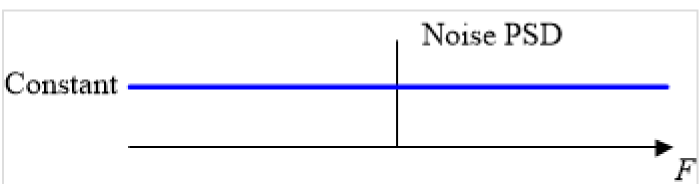


Figure 6: The power spectral density of the white noise

Nyquist studied thermal noise and discovered that its power spectral density is equal to k x T, where k is a constant and T is the temperature in Kelvin. As a result, the noise power at the receiver frontend is precisely proportional to the corresponding temperature, giving rise to the term thermal noise. This constant value, shown in Figure , has been denoted as N0 / 2 Watts/Hz in the past.

When seen as a rectangular series, the constant spectral density's iDFT must be a unit impulse. We also discovered that the spectral density's iDFT is the signal's auto-correlation function. Combining these two facts, a constant spectral density implies that the noise's auto-correlation in the time domain is a unit impulse, that is, it is zero for all non-zero time shifts. This is shown in the figure below:

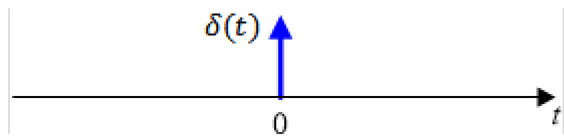


Figure 7: Representation of the constant spectral density in time domain

In words, every noise sample in a sequence is unrelated to every other noise sample in the same sequence. Therefore, the mean value of white noise is zero.

Gaussian: The noise samples' probability distribution is Gaussian with a zero mean, which means that in the time domain, the samples can acquire both positive and negative values, and that values close to zero have a higher chance of occurrence while values far away from zero have a lower chance of occurrence. This is shown in the figure below. As a result, a huge number of noise samples have a temporal domain average of zero.

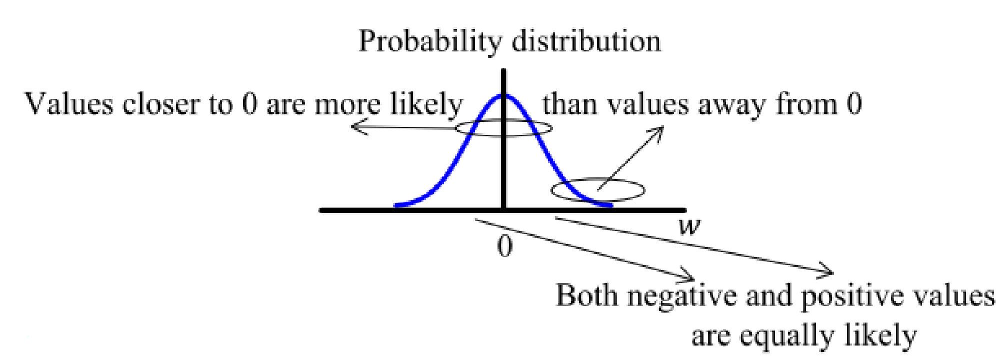


Figure 8: Probability distribution

In actuality, the ideal flat spectrum from $-\infty$ to $+\infty$ is true for wireless communications frequencies (a few kHz to hundreds of GHz), but not for higher frequencies. Nonetheless, every wireless communication system employs filtering to remove the majority of the noise energy that exists beyond the spectral region in which our intended signal is present. As a result, it's impossible to tell whether the spectrum was ideally flat or somewhat flat outside the band of interest after filtering. It can be considered to be flat before filtering to aid mathematical analysis of the underlying waveforms, culminating in closed-form expressions – the holy grail of communication theory.

For a discrete signal with sampling rate FS, the sampling theorem dictates that the bandwidth of a signal is constrained by a low-pass filter within the range ±FS/2 to avoid aliasing. For the purpose of calculations, this filter is an ideal lowpass filter with

$$H(F) = \begin{cases} 1 & -F_S/2 < F < +F_S/2 \\ 0 & \text{elsewhere} \end{cases}$$

The resulting in-band power is shown in red in the figure below, while the rest is filtered out.

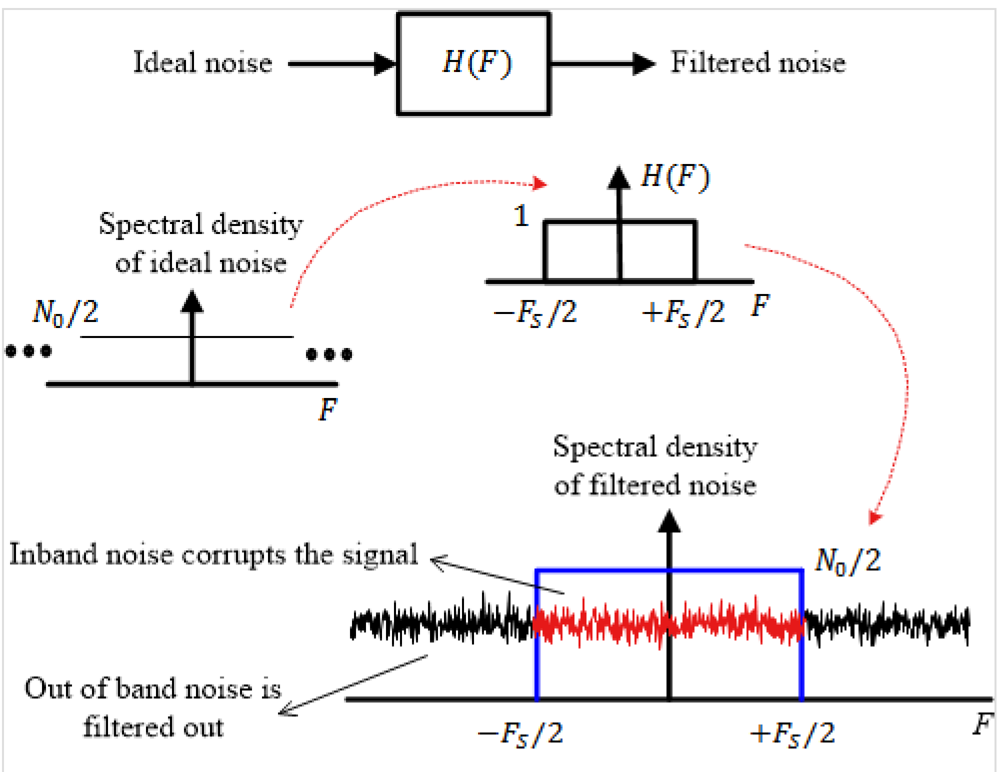


Figure 9: Filtering noise

As with all densities, the value N0 is the amount of noise power Pw per unit bandwidth BW

$$N_0 = \frac{P_w}{B}$$

For the case of real sampling, we can plug B=FS/2 in the above equation and the noise power in a sampled bandlimited system is given as

$$P_w = N_0 \cdot \frac{F_S}{2}$$

Thus, the noise power is directly proportional to the system bandwidth at the sampling stage.
The simplest fundamental model of a communication system is an AWGN channel. Space communications with highly directional antennas and some point-to-point microwave lines are two examples of systems that operate primarily in AWGN conditions. While some flaws remain (e.g., carrier and timing offsets), the basic signal structure is preserved, resulting in a simpler Rx design.

Procedure and Discussion

Part 1: Building Complete System Using Simulink

The aim of this part was to build the given system using Simulink, and following is our implemented system:

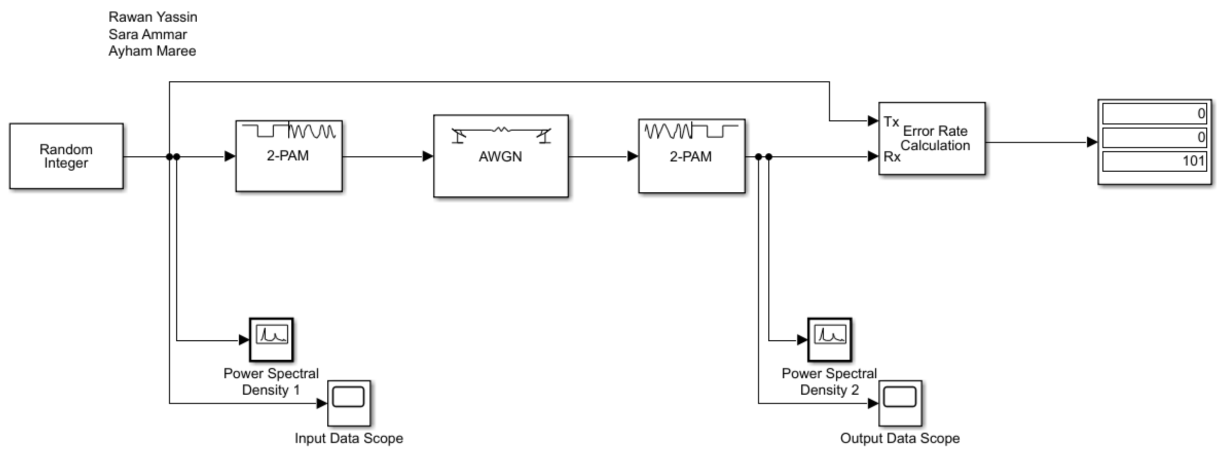
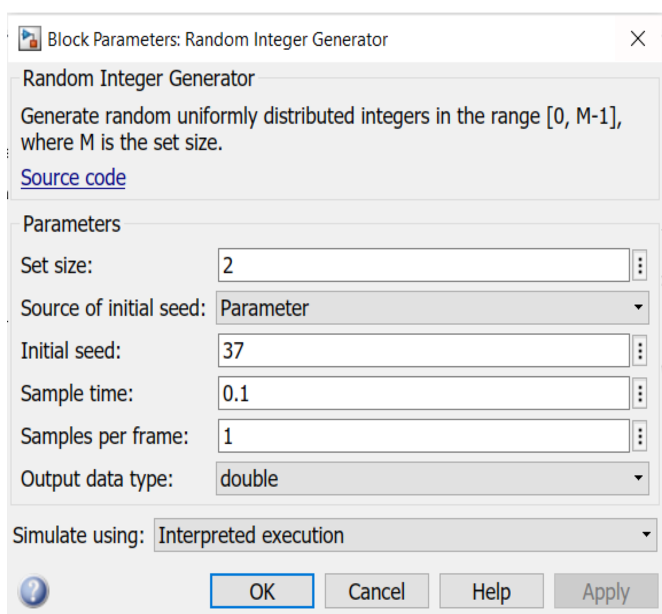


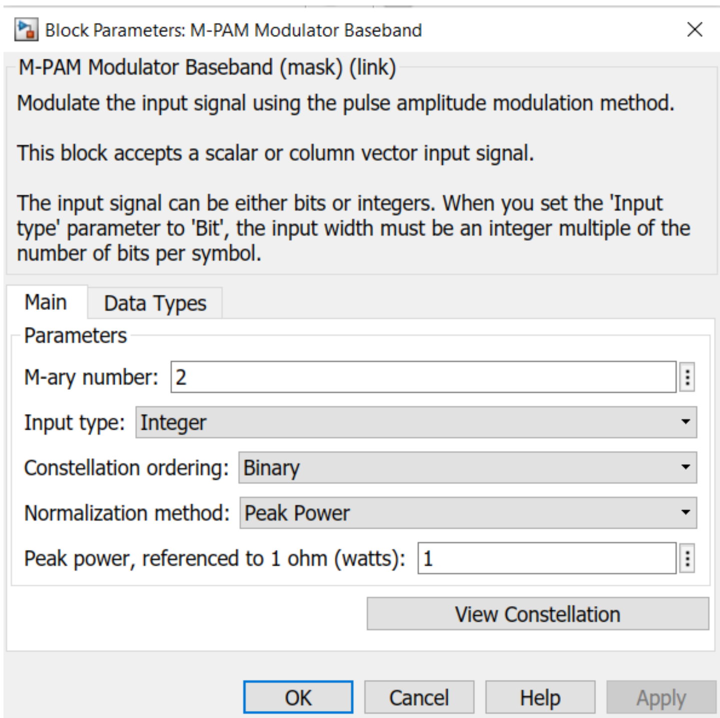
Figure 10: The complete system using Simulink

The above system was built with the following units:

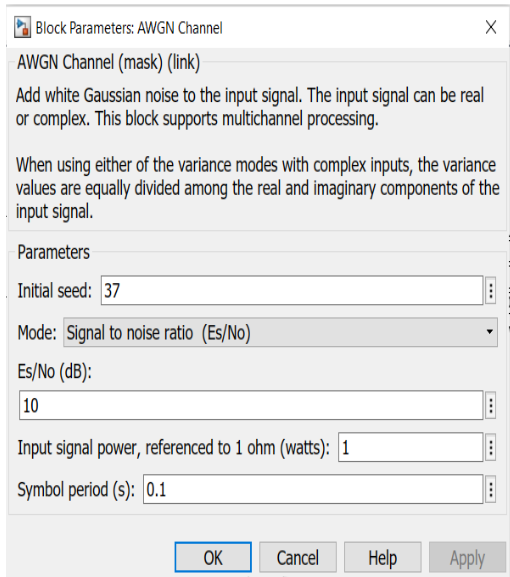
1- The random integer generator unit: creates evenly distributed random integers in the range [0, M-1], and M was set to 2. The initial seed was set to 37, the sample time was set to 0.1 with 1 sample per frame, following are the configurations:



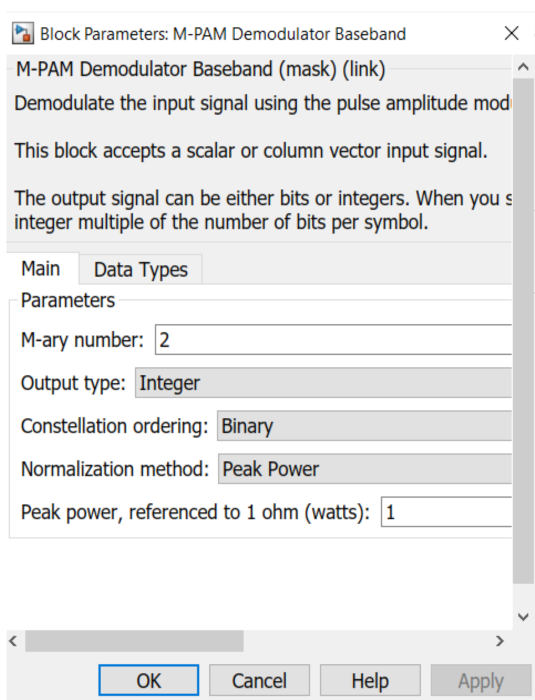
2- M-PAM Modulator Baseband unit: outputs a baseband representation of the modulated signal. It converts the sequence of (0's and 1's) generated by the random integer into a polar nonreturn to zero waveform. The number of points in the signal constellation is represented by M and set to 2 in our case. In addition, a binary constellation ordering was used, with a peak power normalization method- which means that the scaling condition for the default signal constellation is based on the maximum power of the symbols in that constellation.



3- Additive White Gaussian Noise unit: which added white Gaussian noise to the channel after inheriting the sampling time from the input. Our unit was set with the signal-to-noise ratio mode, with an tunable Es/No parameter, which is the ratio of information symbol energy per symbol to noise power spectral density in decibels. Following are our configurations for this unit.



4- M-PAM Demodulator Baseband unit: demodulates a signal that was modulated using M-ary pulse amplitude modulation. With the following parameters - similar to the ones used for the modulator:



5- Error Rate Calculation unit: compares data from a transmitter to data from a receiver By dividing the total number of unequal pairings of data items by the total number of input data elements from one source, the error rate is calculated as a running statistic. It returns a column vector of the form [R,N,S], with R as the error rate, N as the number of errors, and S as the number of samples compared.
6- Data Scopes: display their inputs with respect to simulation time.
7- Power Spectral Density units: display the frequency content of the buffer in a graph window.

Part 2: Comparing the Input and Output Data Sequences, with 1 Second Simulation and -5 db Es/No

In this part, the simulation time was set to 1 second through setting the simulation parameters. The power spectral density components were disconnected by simply removing their connecting wire. The energy per symbol to noise power spectral density was set to -5 db in the white Gaussian noise unit. Then the outputs of the two used scopes were captured as follows:
The one to the left was taken directly from the input scope and the second one was taken from the output of the demodulator scope.
It is noticed that the output differs from the input in almost 40% of the samples; this is due to the additive white Gaussian noise presented in the channel. The noise is additive, i.e, the received signal is equal to the transmitted signal plus noise. As Es/No is set to -5 db, it is expected to view the result of such noise in the output as a negative number here indicates that the noise level is greater than the signal level, $E_s/N_0(\text{dB})=10\log_{10}(0.5T_{sym}/T_{samp})+SNR$ (dB) for real input signals .

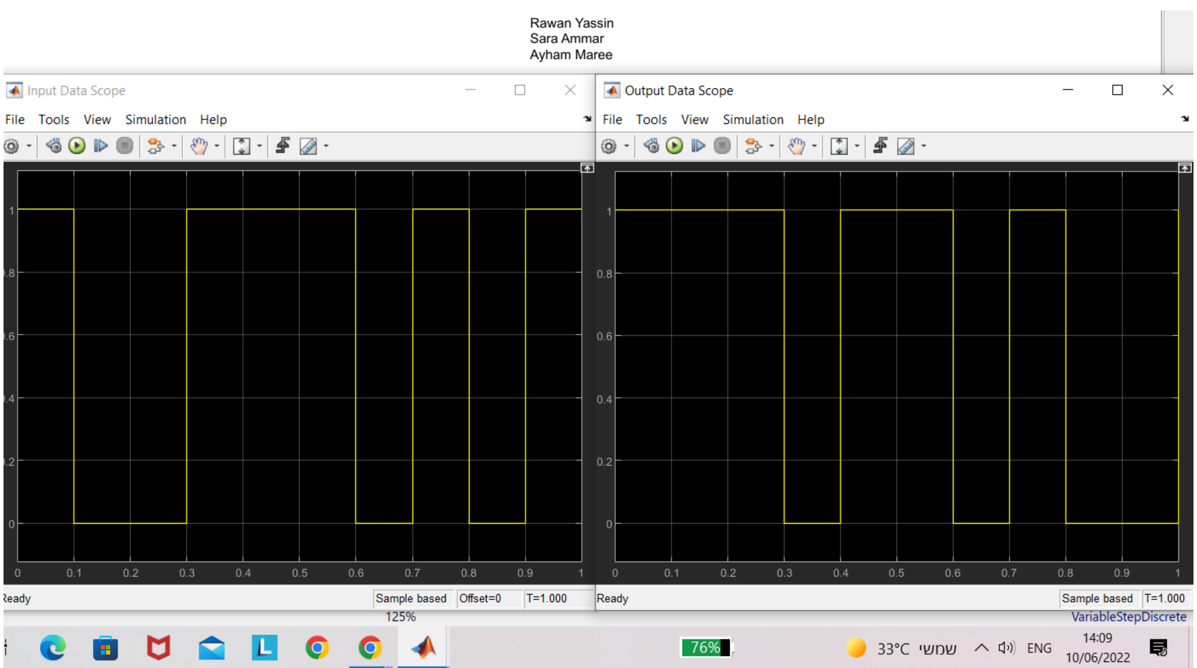


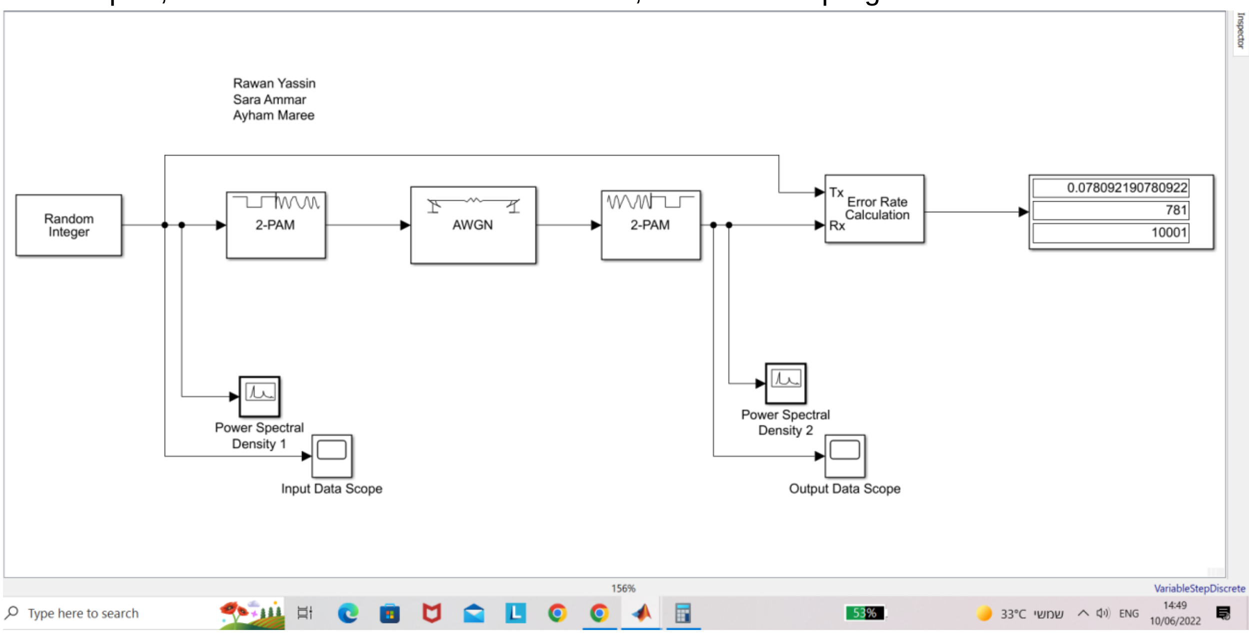
Figure 11: The input and output data sequenses

Part 3: Finding the 90% and 95% Power Bandwidth of the Baseband Signal in Hz

The symbol duration was set to 0.1 through the setting of the generator and it was verified by the above figure on the right, having this symbol duration (τ), leads to the following:
The 90% power bandwidth = 1/ τ = 1/0.1 = 10 Hz.
The 95% power bandwidth = 2 τ = 2/0.1 = 20 Hz.

Part 4: Finding the Probability of Error with Es/No= 0 dB and 1000 Second Simulation Time

For this part, the simulation time was set to 1000, and then the program was started:



In this part, it's obvious that the error rate equals 0.078092190, and what is required is Finding the Probability of Error with Es/No = 0 dB and 1000 Second Simulation Time.

Firstly we must find the value of E0/N0 to find the theoretical error rate , using this equation:

$$(Es\,/N0\,)\text{dB} = 10 \log\,(\,E0/N0\,)$$

Es/N0 = 0 db as given, then:

$$10 \log\,(\,E0/N0\,) = 0$$

⇒ E0 / N0 = 1
⇒ error rate value = 1

Hence, we have to find the probability of error rate E0/N0 using this equation:

$$Pb = Q\ (\sqrt{2E0/N0}\ )$$

⇒ Pb = Q(√2)
⇒ Pb = Q(1.41421) = 0.080757

Then the value of the probability of error rate = 0.080757, and the value of probability of error rate very close to the original value of the error value and even approximately equal to it.

Part 5: Compare the input and output power spectral densities

The aim of this part is to compare the input and output power spectral densities and explain the reason for the similarity or/ differences when Es/No = 0 dB (with the power spectral density units connected).

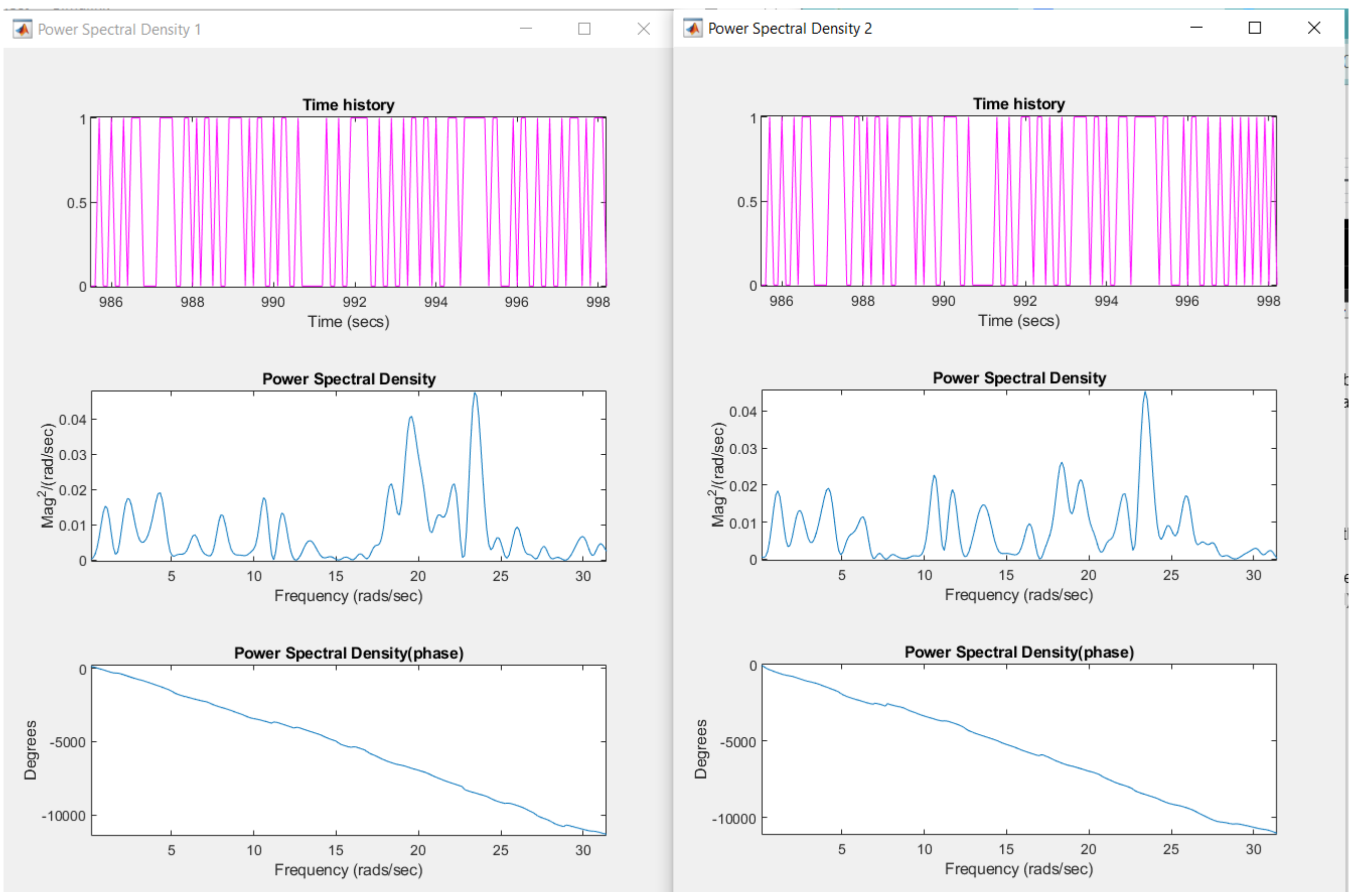


Figure 12: The input and output power spectral densities

The power spectral density should be the same, but there is a discrepancy because the demodulation process introduces error, causing the output to diverge somewhat from the input. By lowering the error value, we may bring the output closer to the input.

Part 6: Records the value of Es/No and the corresponding BER

In this part, the simulation time was set to 1000 sec. The value of the Es/No (dB) was adjusted in the AWGN block, starting from – 10 dB, incrementing by 1dB every step, and ending at 7dB. Then the error rate displayed in the Display block.

The table below records the value of Es/No and the corresponding BER:

-10 dB	0.31896810318968 3190 10001
-9 dB	0.2999700029997 3000 10001
-8 dB	0.28077192280772 2808 10001
-7 dB	0.25737426257374 2574 10001
-6 dB	0.23467653234677 2347 10001
-5 dB	0.20967903209679 2097 10001

-4 dB	0.18308169183082 1831 10001
-3 dB	0.15508449155084 1551 10001
-2 dB	0.12988701129887 1299 10001
-1 dB	0.1039896010399 1040 10001
0 dB	0.078092190780922 781 10001
1 dB	0.05979402059794 598 10001
2 dB	0.040595940405959 406 10001

3 dB	0.025197480251975 252 10001
4 dB	0.013198680131987 132 10001
5 dB	0.0068993100689931 69 10001
6 dB	0.0035996400359964 36 10001
7 dB	0.000999900009999 10 10001

Table 1

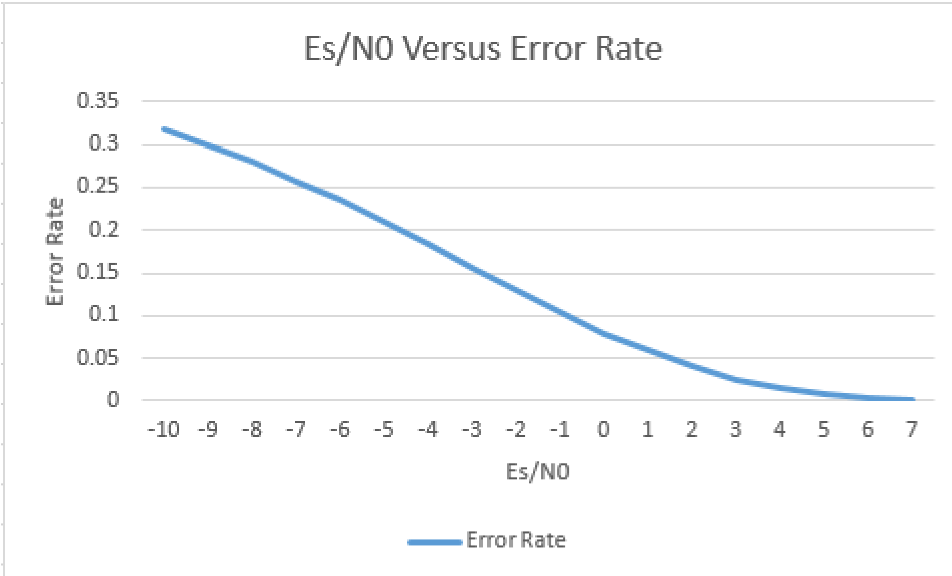


Figure 13: The error rate vs. Es/No in dB

Notice that the error rate decreases with the increase of the value of Es/No in the AWGN block, where when the value of Es/No = -10 DB, the value of the error rate was 0.32 then when the value of Es/No = 7 DB the value of the error rate was 0.0001, that it indicates that there is an inverse relationship between the error rate and the value of Es / No in the mass of AWGN.

Conclusion

We obtained a better knowledge of how baseband digital pulse amplitude modulation systems work with this project. We learned at first that the modulator and demodulator function as a transmitter and receiver, respectively, and that the Additive white Gaussian noise (AWGN) adds noise to the signal to simulate real-world noise created by the channel. Then we understood what the mistake did to the signal and how to reduce it by raising the value of Es/No.