Experiment 1: Performance of Digital Modulation over Wireless Communications

NAME: shahd ahmed Mohamed

1 Introduction

We consider the performance of the digital modulation techniques over AWGN channels. The performance criteria of interest: the probability of error, de ned relative to either symbol or bit errors.

2 AWGN Channel Modeling

We de ne the signal-to-noise power ratio (SNR) and its relation to energy-per-bit (E_b) and energy-per-symbol (E_s). We then examine the error probability on AWGN channels for di erent modulation techniques as parameterized by these energy metrics.

1. Signal-to-Noise Power Ratio and Bit/Symbol Energy : In an AWGN chan-nel the modulated signal s(t) has noise n(t) added to it prior to recep-tion. The noise n(t) is a white Gaussian random process with mean zero and power spectral density N_0=2. The received signal is thus r(t) = s(t) + n(t).SNR is de ned as the ratio of the received signal power Pr to the power of the noise within the bandwidth of the transmitted sig-nal s(t). The received power P_r is determined by the transmitted power and the path loss. The noise power is determined by the bandwidth of the transmitted signal and the spectral properties of n(t). Since the noise n(t) has uniform power spectral density N0/2, the total noise power within the bandwidth 2B is $N = \frac{N_0}{2} 2B = N_0 B$. So the received SNR is given by

$$SNR = \frac{P_r}{N_0 B}$$

   The SNR is often expressed in terms of the signal energy per bit E_b or per symbolE_s as

$$SNR = \frac{P_r}{N_0 B} = \frac{E_s}{N_0 B T_s} = \frac{E_b}{N_0 B T_b}$$

   where T_s is the symbol time and T_b is the bit time.

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2. Monte Carlo Simulation: The objective of the Monte Carlo simulation is to estimate the symbol error rate our system can achieve. The idea behind a Monte Carlo simulation is simple:

   Simulate the system repeatedly.

   For each simulation count the number of transmitted symbols and symbol errors

   Estimate the symbol error rate as the ratio of the total number of observed errors and the total number of transmitted bits.

3 Statistical Validity

Performing a bit-error-rate simulation can be a lengthy process. We need to run individual simulations at each SNR of interest. We also need to make sure our results are statistically signi cant. When the bit-error-rate is high, many bits will be in error. The worst-case bit-error rate is 50 percent, at which point, the modem is essentially useless. Most communications systems require bit-error-rates several orders of magnitude lower than this. Even a bit-error-rate of one percent is considered quite high. We usually want to plot a curve of the bit-error-rate as a function of the SNR, and include enough points to cover a wide range of bit-error-rates. At high SNRs, this can become di cult, since the bit-error-rate becomes very low. If our test signal only contains bits, we will most likely not see an error at this bit-error-rate. In order to be statistically signi cant, each simulation we run must generate some number of errors. If a simulation generates no errors, it does not mean the bit-error-rate is zero; it only means we did not have enough bits in our transmitted signal. As a rule of thumb, we need about (or more) errors in each simulation, in order to have con dence that our bit-error-rate is statistically valid. At high SNRs, this can require a test signal containing millions, or even billions of bits.

4 Requirements

1. Figure 1 of the relation between the BER and SNR.
2. Calculation of the transmitted signal Power:

$$P_t = \text{norm}(\text{signal}; 2)^2 = \text{length}(\text{signal}) = 0{:}4994 \text{ watt} = \quad 3{:}0155 \text{ db}$$

3. Identify the meaning of 'measured' eld : It speci es that it measures the power of the signal before adding noise.
4. At which value of SNR the system is nearly without error : At SNR = 20 the BER is zero 2.

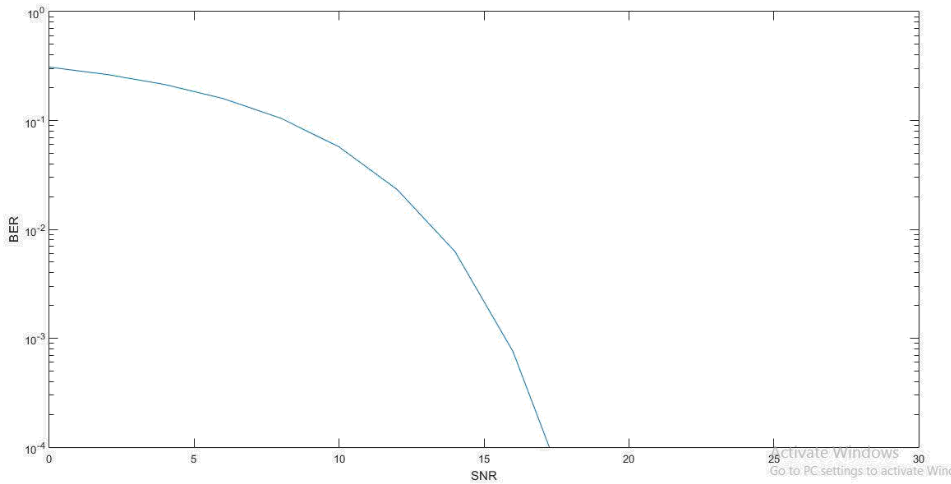

Figure 1: BER curve versus SNR

```
BER_sim =

  Columns 1 through 12

    0.3078    0.2637    0.2144    0.1594    0.1043    0.0568    0.0232    0.0060    0.0008    0.0000         0         0

  Columns 13 through 16

         0         0         0         0
```

Figure 2: The bit error rate values

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Experiment 2 Performance of Matched Filters and Correlators

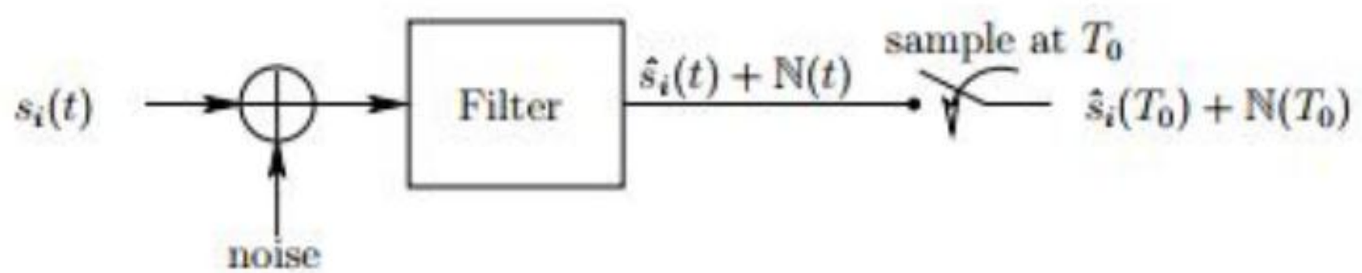
By: Marwan Nabil Mohammed Basiouny Alfiel
No. 189

1 Introduction

The matched filter is the maximum-likelihood receiver in the presence of additive white Gaussian noise. Thus, for equal prior symbol probabilities, it will yield optimum bit-error performance. This is equivalent on the AWGN channel to maximizing the signal-to-noise ratio.

2 Modeling Matched filter receiver

We will study the decision-making process in the digital communications receiver which was modeled as shown below in Fig. 1



1. It is well known, that the optimum receiver for an AWGN channel is the matched lter receiver. When the noise is absent, matched filter output is just signal energy i.e.
   - It matches the source impulse and maximize the SNR.
   - For non Gaussian noise, matched filter is not optimal.
   - Matched filter maximizes the SNR at the output of an FIR (also IIR) filter (even if the noise is non Gaussian).
   - Its operation depends on the symbols being used and the apriori probabilities .

3 Comparison between Simple detectors and Matched Filter and Correlators

For the Simple detector:

- At high SNR the BER is totally zero
- It's an ideal detector where we assume the probability of error is $P_e = \frac{1}{2} \times P(x \leq v_{th}) + P(y > v_{th})$
- Assuming that the threshold voltage: $V_t = \frac{S_1+S_2}{2}$
- Its BER vs SNR curve looks practically as shown in Fig. 2

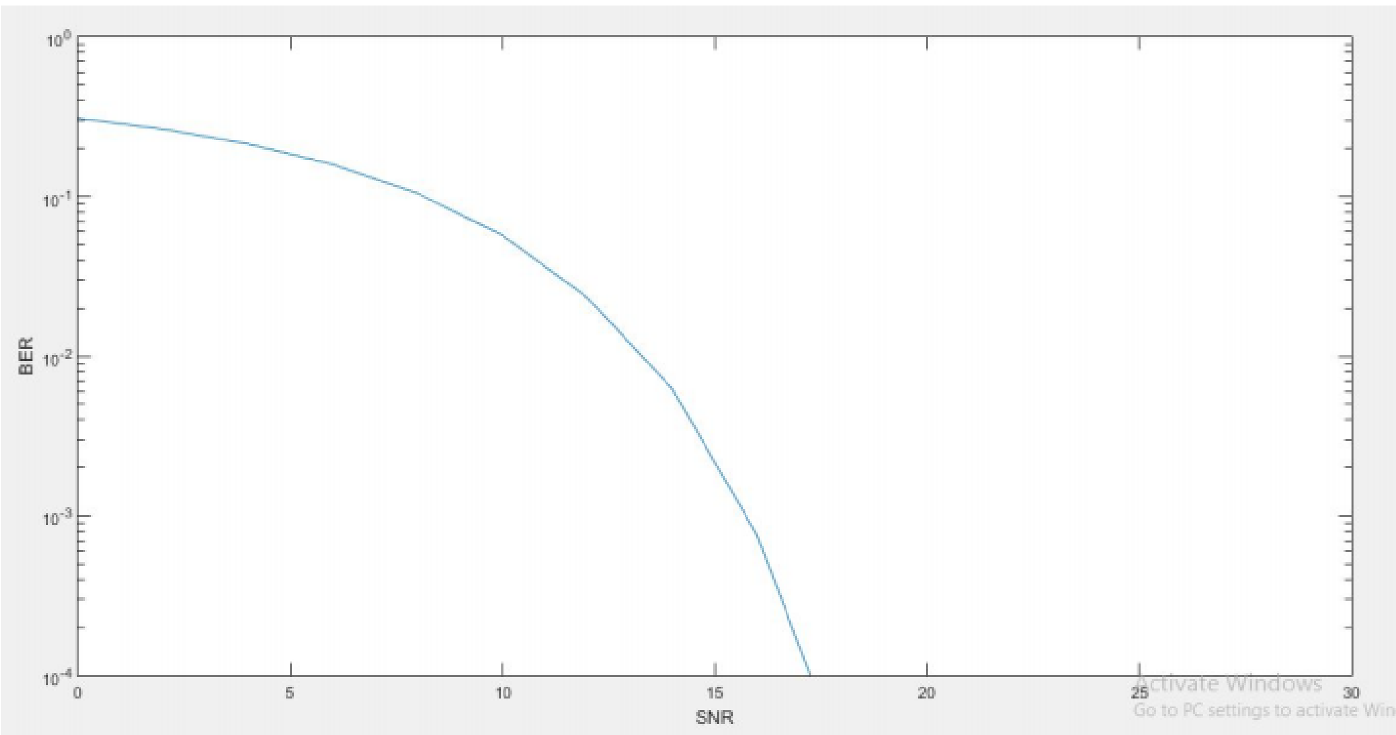


Figure 2: Simple Detector

For the Matched Filters and Correlators

- AT high SNR there is still BER not equal to zero.
- Probability of error takes the form of the P_e of the Gaussian leading to $P_e = \frac{1}{2} \times erfc(\frac{S1(10)-S2(10)}{2\sqrt{(2No)}})$
- As the erfc coefficient decrease the P_e will increase, therefore the least P_e will occur at the instant in which $(S1 - S2)$ is the maximum.

- Assuming that the threshold voltage: $V_t = \frac{S_1+S_2}{2}$

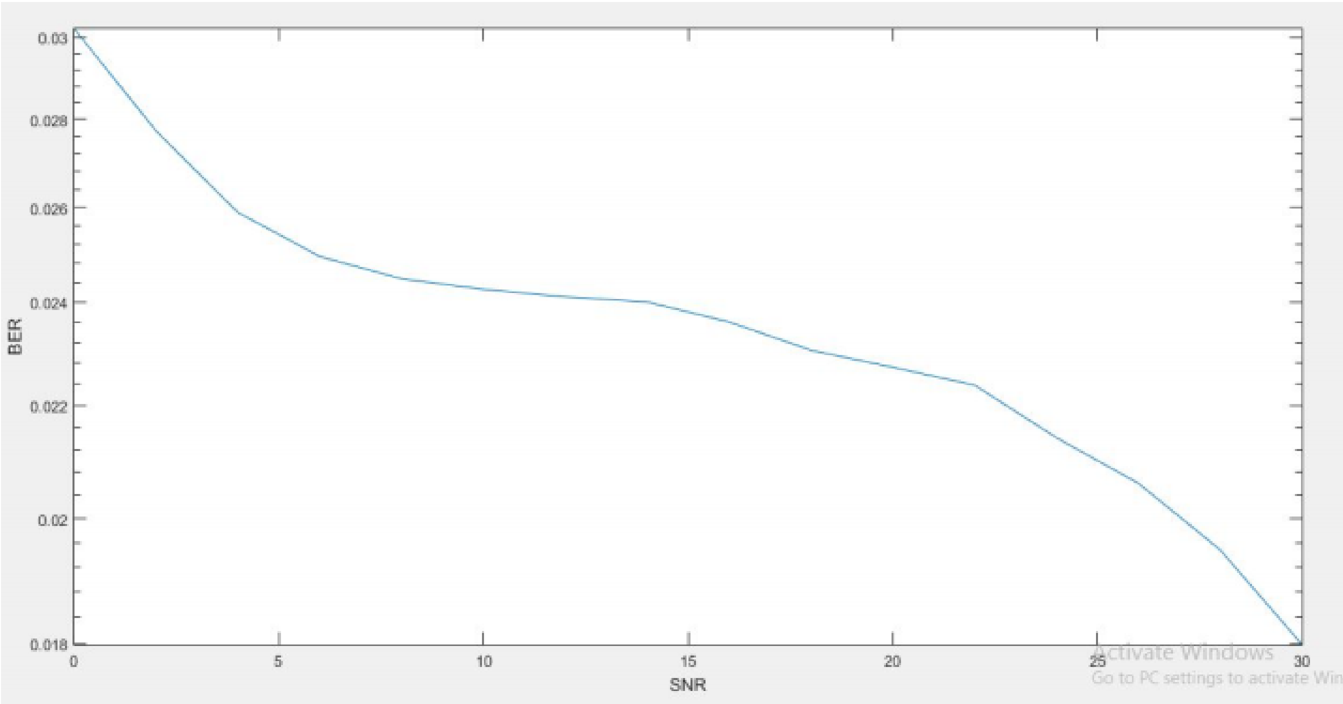


Figure 3: Matched Filter

4 Requirements

1. Calculation of the transmitted signal Power:

$$P_t = norm(signal, 2)^2/length(signal) = 0.5012 \texttt{ watt} = -3 \texttt{ db}$$

2. At which value of SNR the system is nearly without error : At SNR = 30 the BER is 0.0180 which is near zero as shown in Fig. 4

```
BER =

  Columns 1 through 12

    0.0302    0.0277    0.0259    0.0249    0.0245    0.0243    0.0241    0.0240    0.0236    0.0231    0.0227    0.0224

  Columns 13 through 16

    0.0214    0.0206    0.0195    0.0180
```


Figure 4: BER values for different SNR

3. Since the code should run for reasonable time (few minutes).We've tried 2 cases at which max-run=20 and 2 as shown in Figures 5 and 6

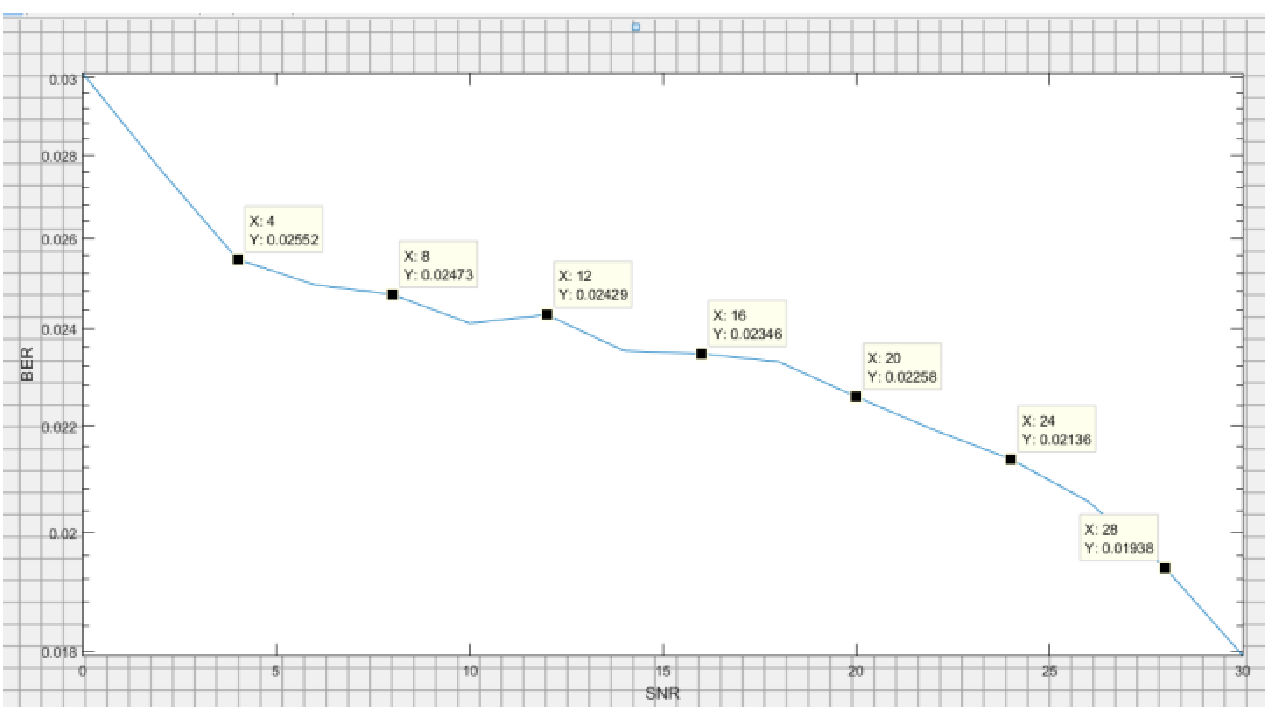


Figure 5: for max run= 20

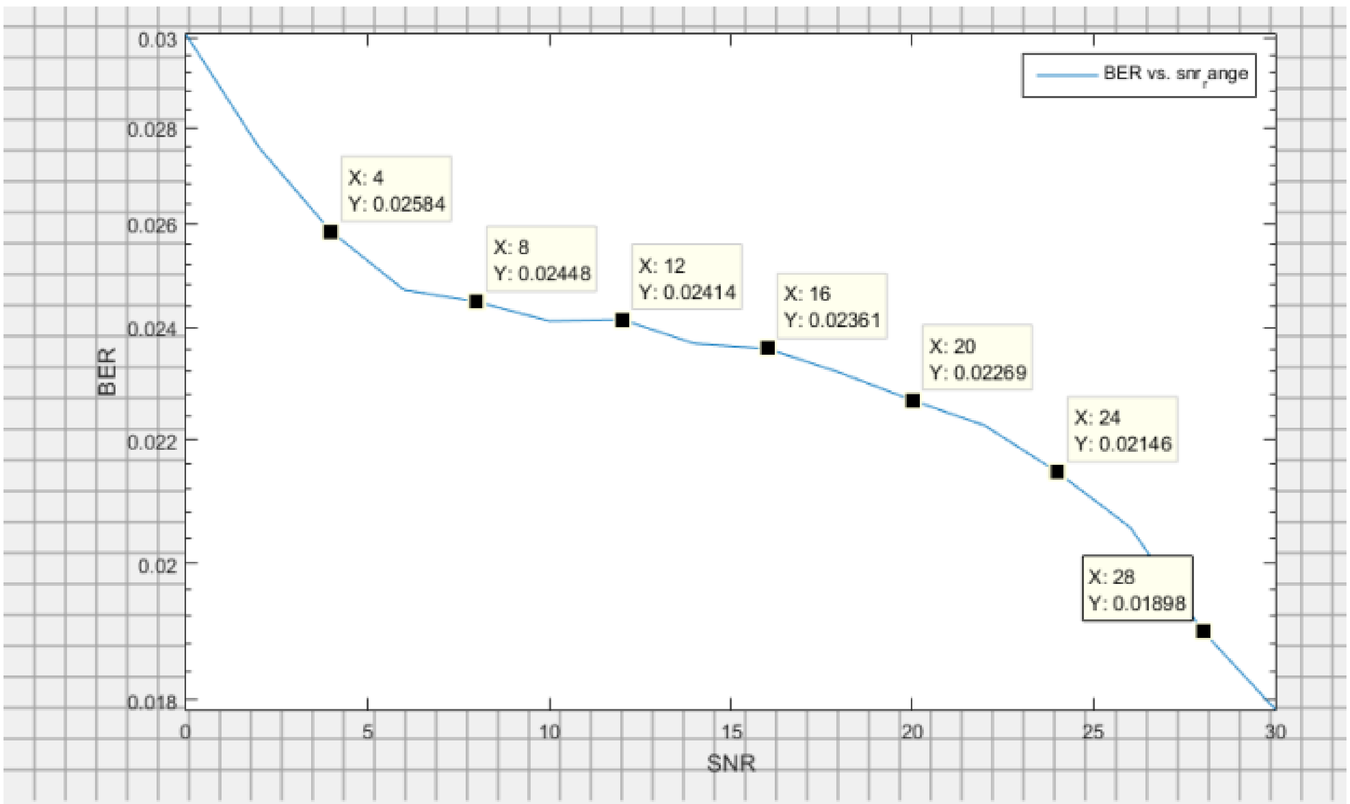


Figure 6: for max run= 2

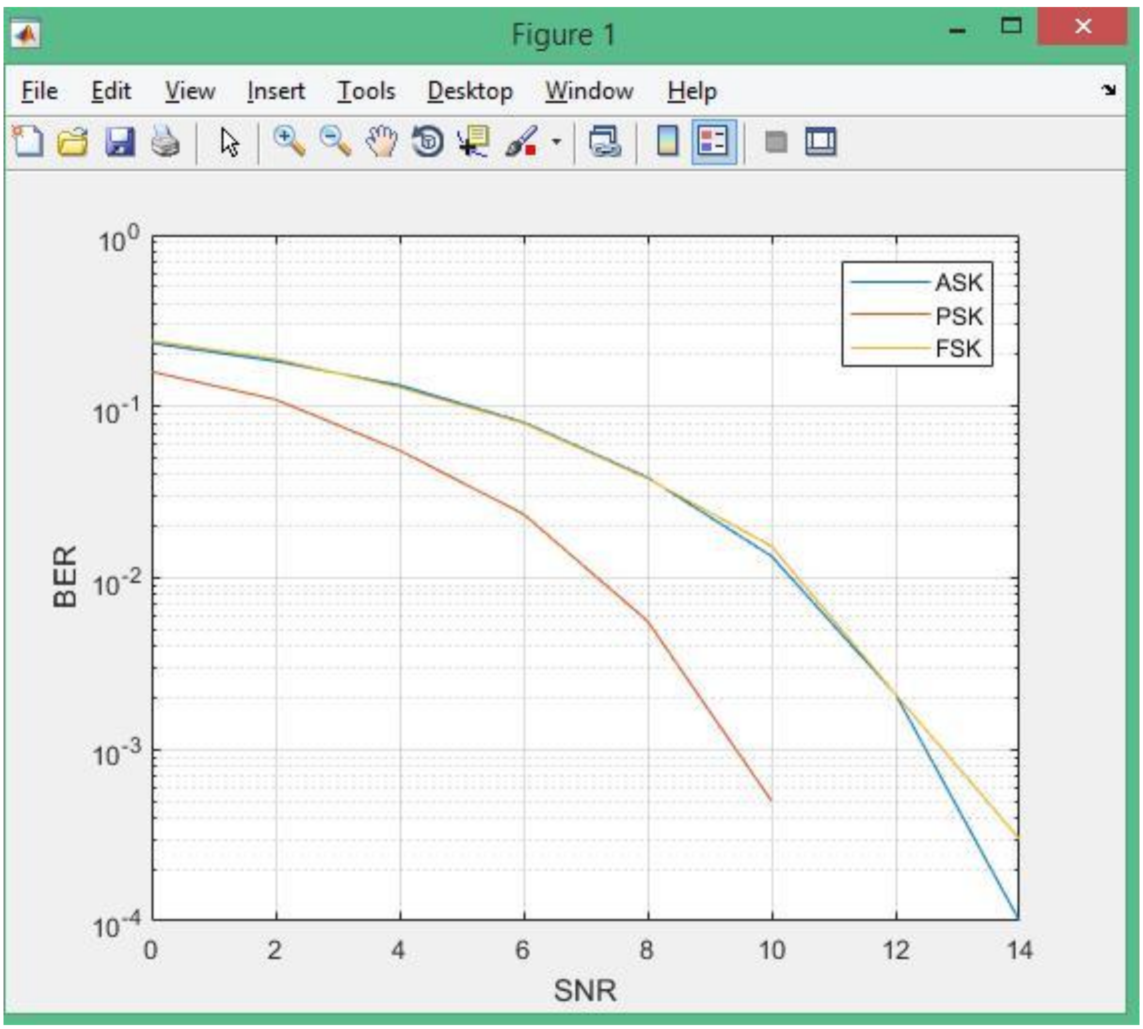
Experiment (3)

Performance of different modulation types
Name: Moustafa Raafat Moustafa

Report Requirements :

(1) M-files attached

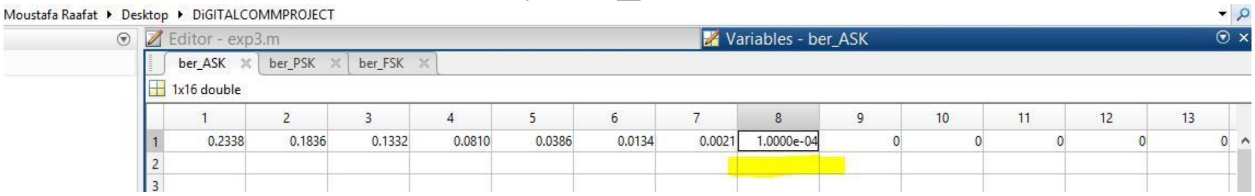
(2) Here we plot snr vs each BER together on the same figure to be able to compare.



(3) As we see BPSK(PRK) has the performance as it has the least BER among the 3 types, while OOK and orthogonal-FSK almost have the same performance.

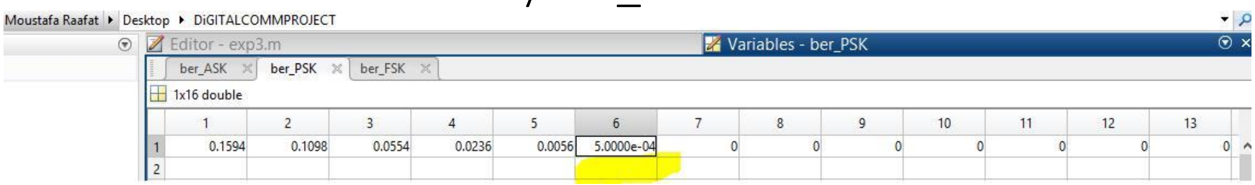
(4)The system has nearly no errors at these values of SNR:

-For OOK, it's about snr=16 dB we can see this more accurately from the values of the array ber_ASK



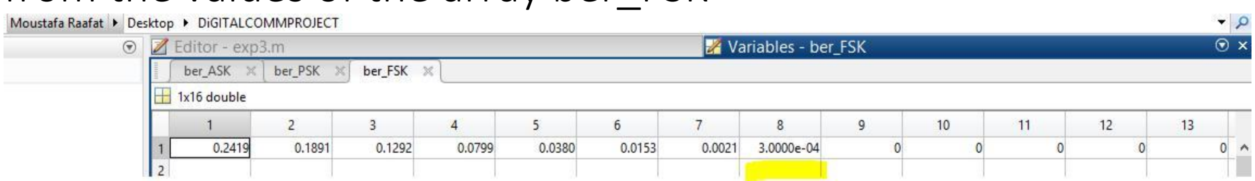
	1	2	3	4	5	6	7	8	9	10	11	12	13
1	0.2338	0.1836	0.1332	0.0810	0.0386	0.0134	0.0021	1.0000e-04	0	0	0	0	0
2													
3													

-For PRK, it's about snr = 12 dB we can see this more accurately from the values of the array ber_PSK



	1	2	3	4	5	6	7	8	9	10	11	12	13
1	0.1594	0.1098	0.0554	0.0236	0.0056	5.0000e-04	0	0	0	0	0	0	0
2													

-For FSK, it's about snr = 16 dB we can see this more accurately from the values of the array ber_FSK



	1	2	3	4	5	6	7	8	9	10	11	12	13
1	0.2419	0.1891	0.1292	0.0799	0.0380	0.0153	0.0021	3.0000e-04	0	0	0	0	0
2													

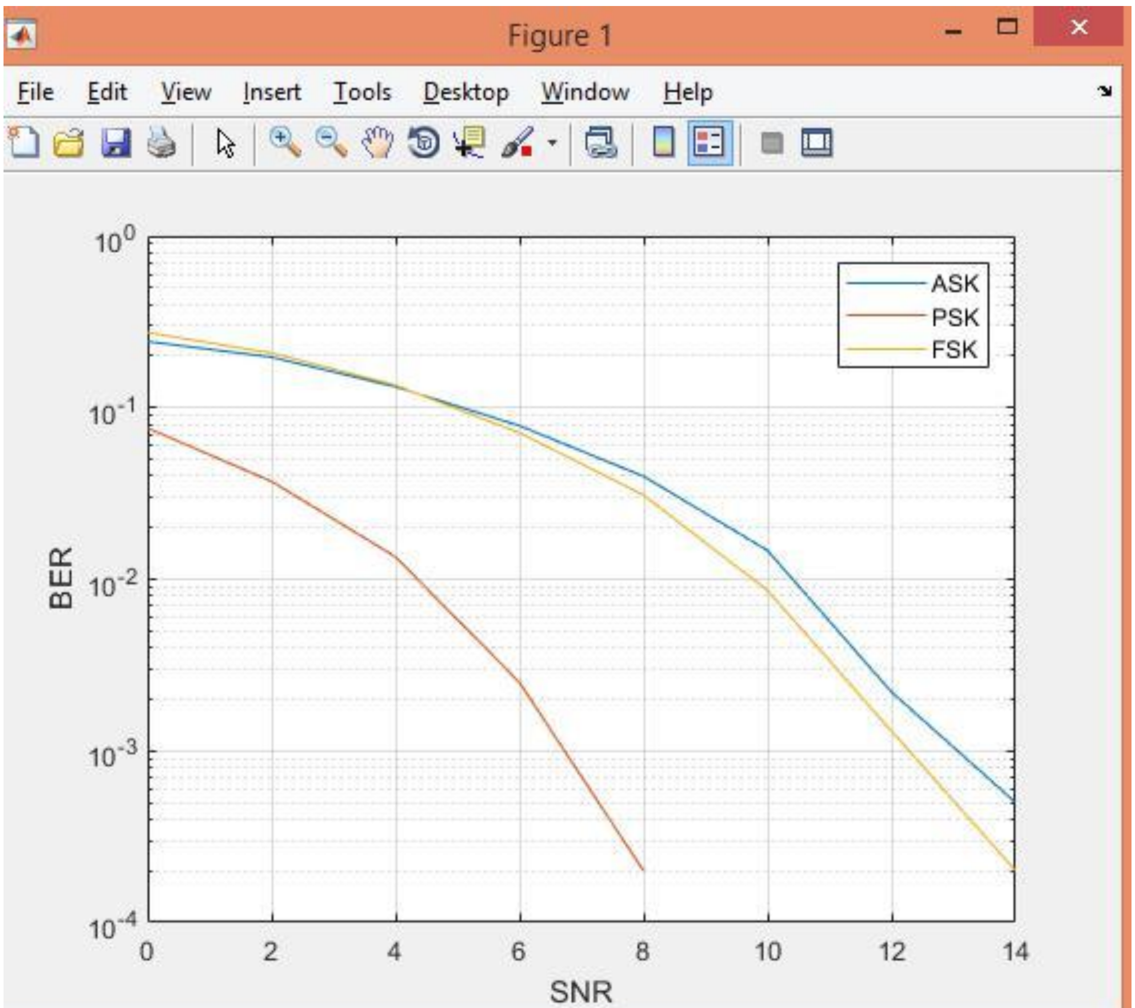
Also we could've see it directly from the graph when the values of each one vanishes, but it's less accurate.

Bonus of experiment (3)

(1) Evaluating the same curves using the MATLAB built-in functions.

*attached the m-file of it with the name “exp3bonuspart1.m”

Here also we plot snr vs each BER together on the same figure to be able to compare.

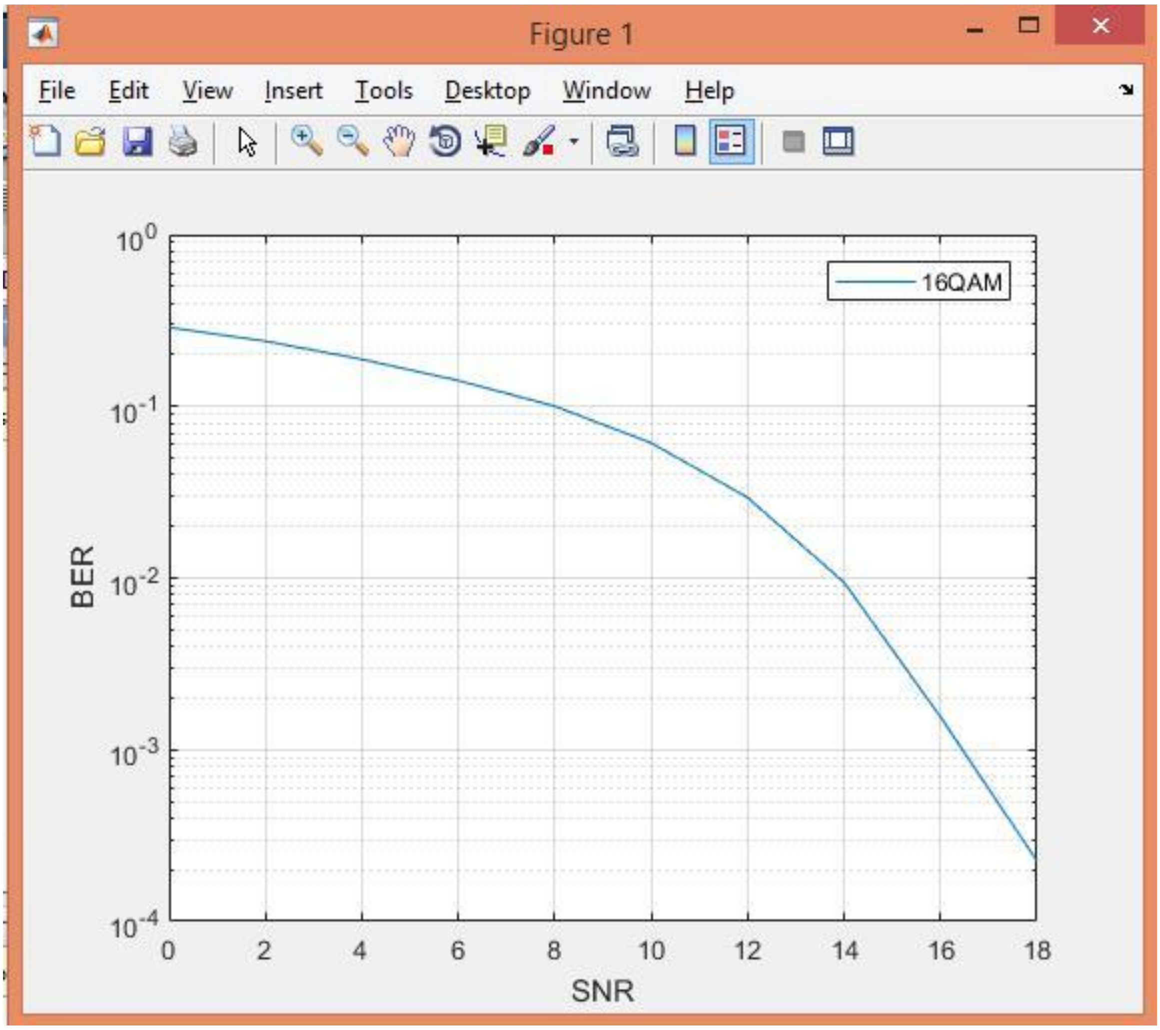


It’s also obvuios here that BPSK has the best performance as it has the lowest BER among the 3 types, while OOK and orthogonal-FSK almost have the same performance.

(2)Evaluating the probability of error of 16QAM modulation.

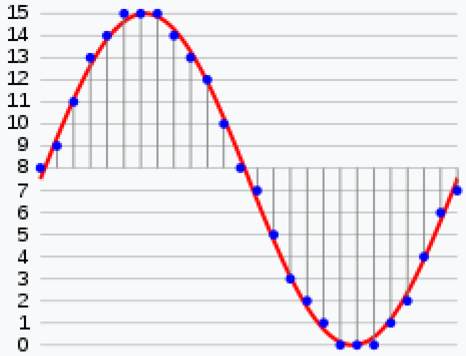
*attached the m-file of it with the name "exp3bonuspart2.m"

Here we plot SNR vs BER of 16QAM modulation type.



Experiment (4)
PCM ”pulse code modulation”
Name: Asmaa Mohamed Salah Eldien

Pulse-code modulation (**PCM**) is a method used to digitally represent sampled analog signals. It is the standard form of digital audio in computers, compact discs, digital telephony and other digital audio applications. In a PCM stream, the amplitude of the analog signal is sampled regularly at uniform intervals, and each sample is quantized to the nearest value within a range of digital steps.

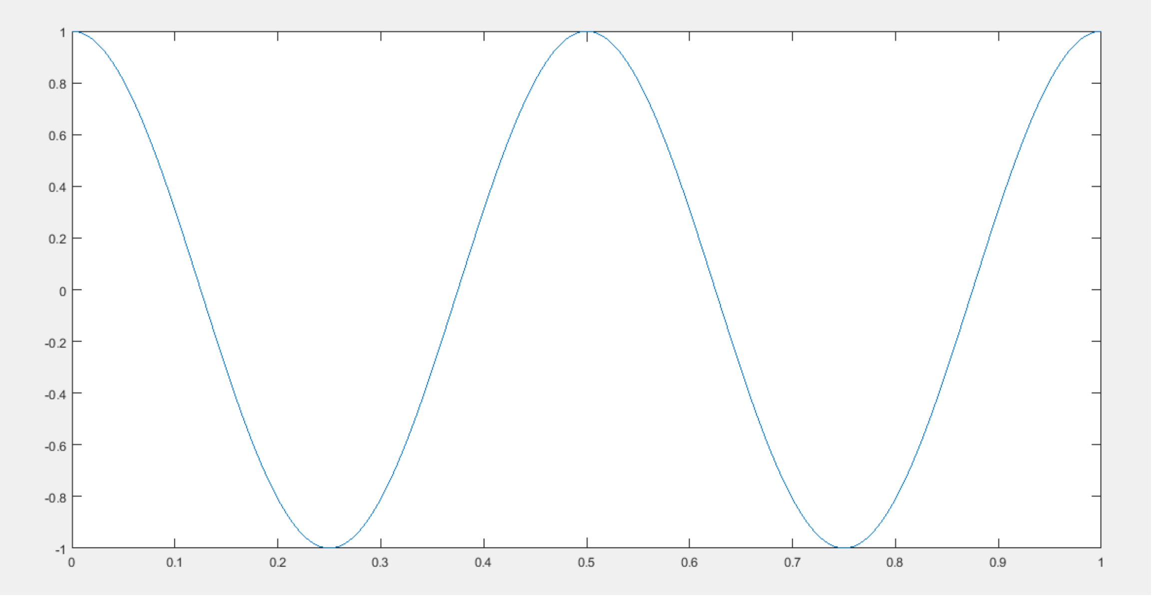


Sampling and quantization of a signal (red) for 4-bit LPCM

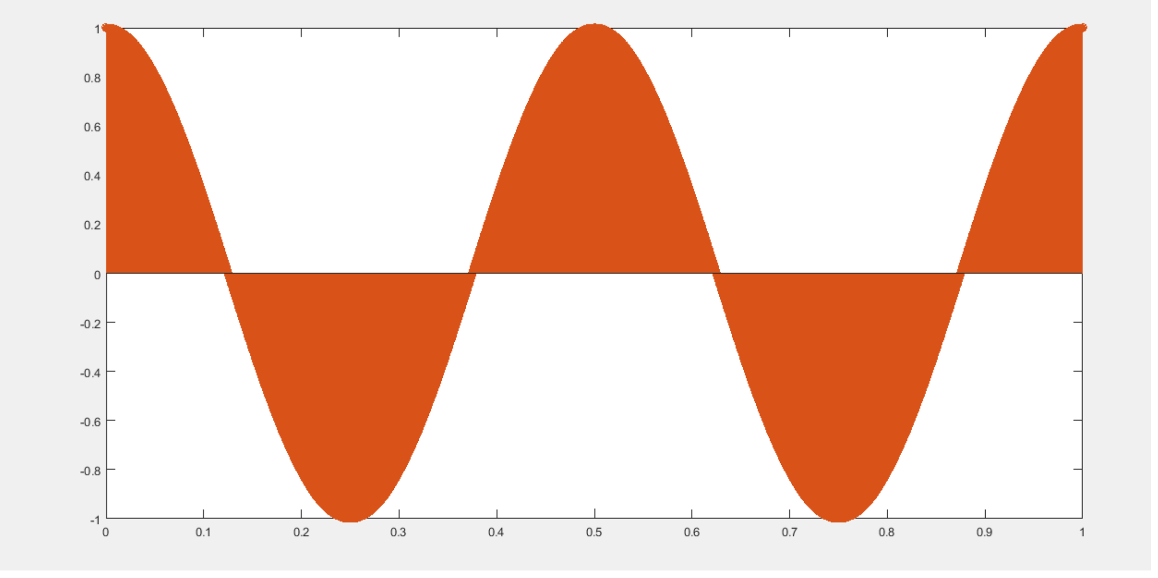
In the diagram, a sine wave (red curve) is sampled and quantized for PCM. The sine wave is sampled at regular intervals, shown as vertical lines. For each sample, one of the available values (on the y-axis) is chosen by some algorithm. This produces a fully discrete representation of the input signal (blue points) that can be easily encoded as digital data for storage or manipulation. For the sine wave example at right, we can verify that the quantized values at the sampling moments are 8, 9, 11, 13, 14, 15, 15, 15, 14, etc. Encoding these values as binary numbers would result in the following set of nibbles: 1000 ($2^3\times1+2^2\times0+2^1\times0+2^0\times0=8+0+0+0=8$), 1001, 1011, 1101, 1110, 1111, 1111, 1111, 1110, etc. These digital values could then be further processed or analyzed by a digital signal processor. Several PCM streams could also be multiplexed into a larger aggregate data stream, generally for transmission of multiple streams over a single physical link. One technique is called time-division multiplexing (TDM) and is widely used, notably in the modern public telephone system.

The PCM process is commonly implemented on a single integrated circuit generally referred to as an analog-to-digital converter (ADC).

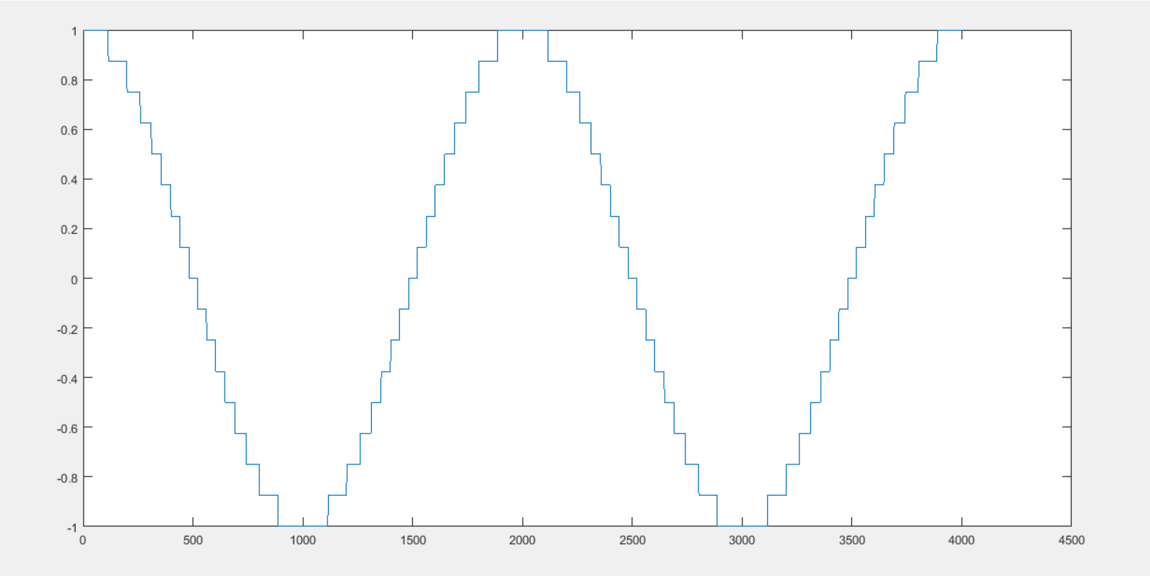
1) input signal at frequency =2 Hz



2)sampled signal at sampling frequency =4000 Hz



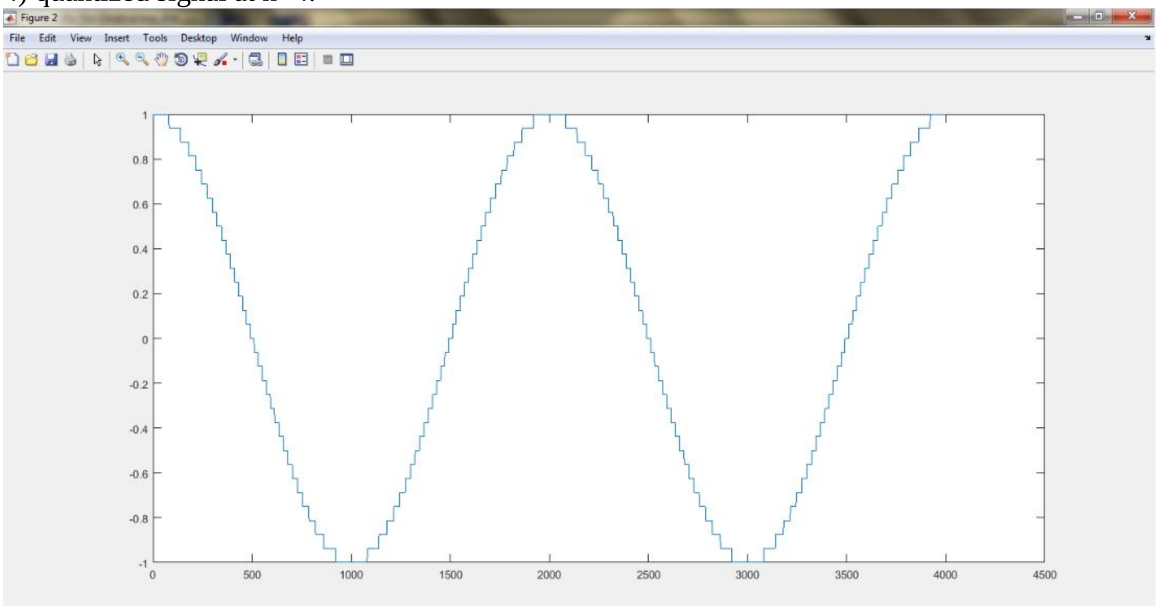
3)quantized signal at n=3:



MSE =

MSE: 0.0011

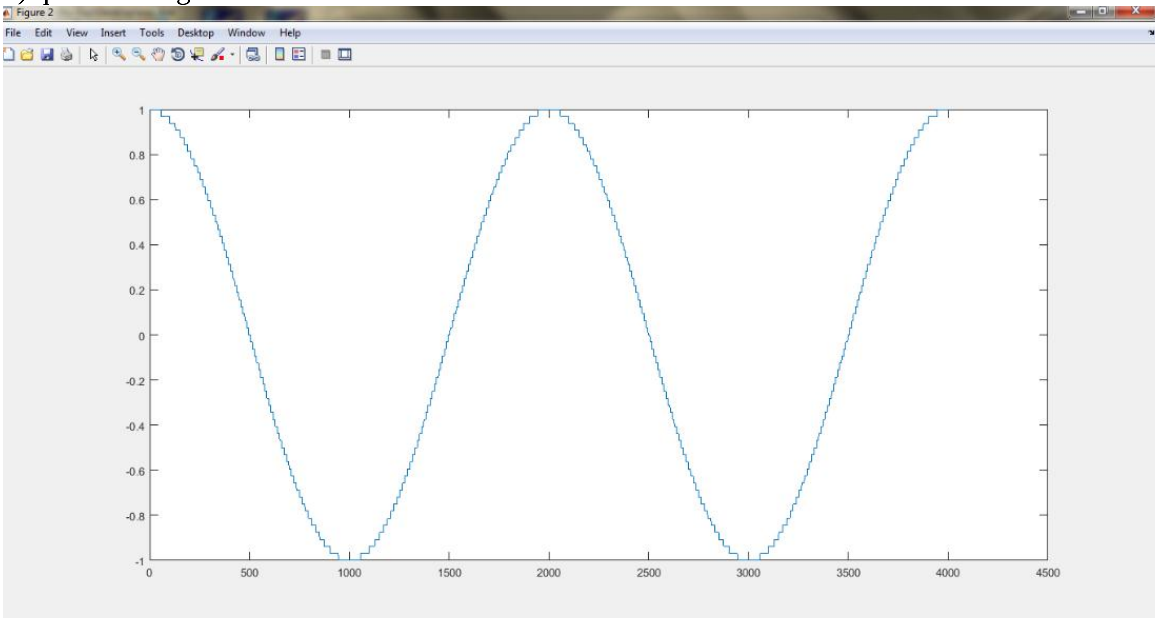
4) quantized signal at n=4:



MSE =

MSE: 7.6897e-08

4) quantized signal at n=5:

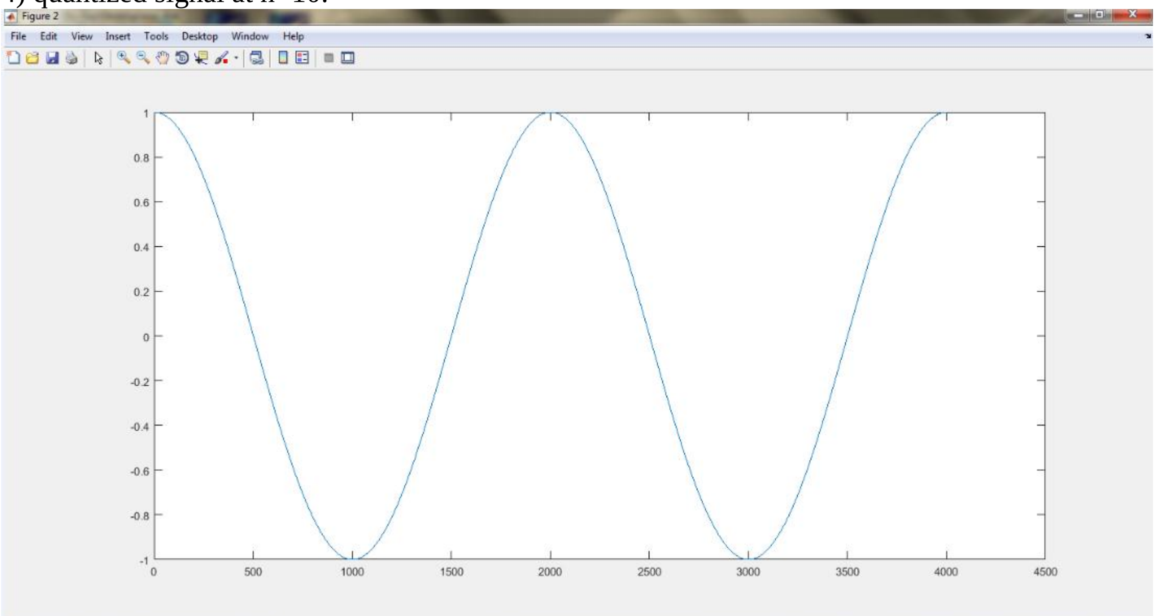



```
MSE =

    7.1783e-05
```

MSE:

4) quantized signal at n=10:

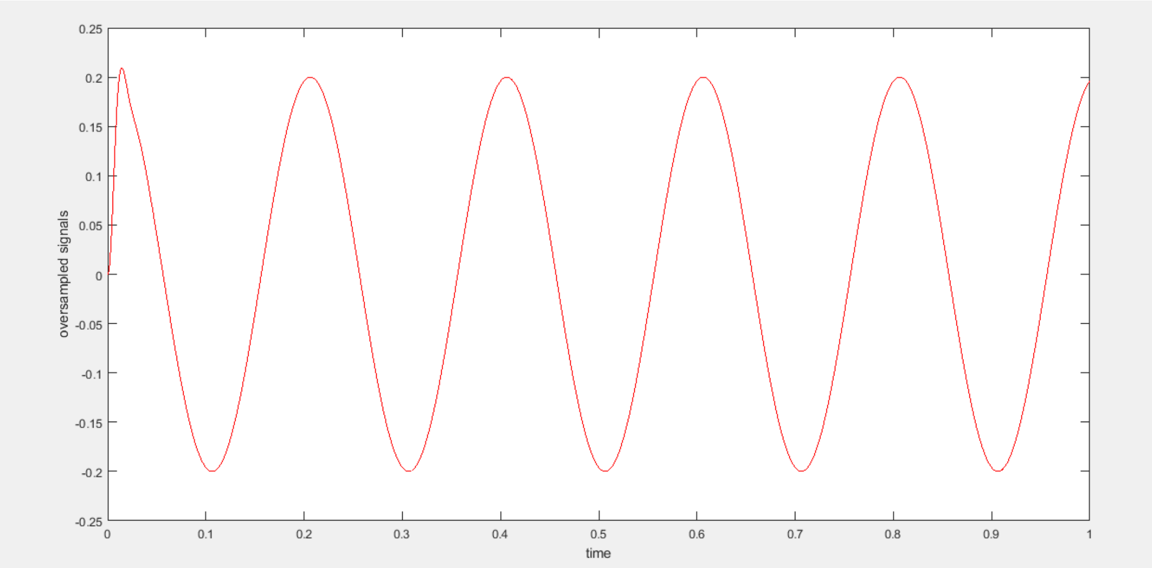



```
MSE =

    2.7318e-04
```

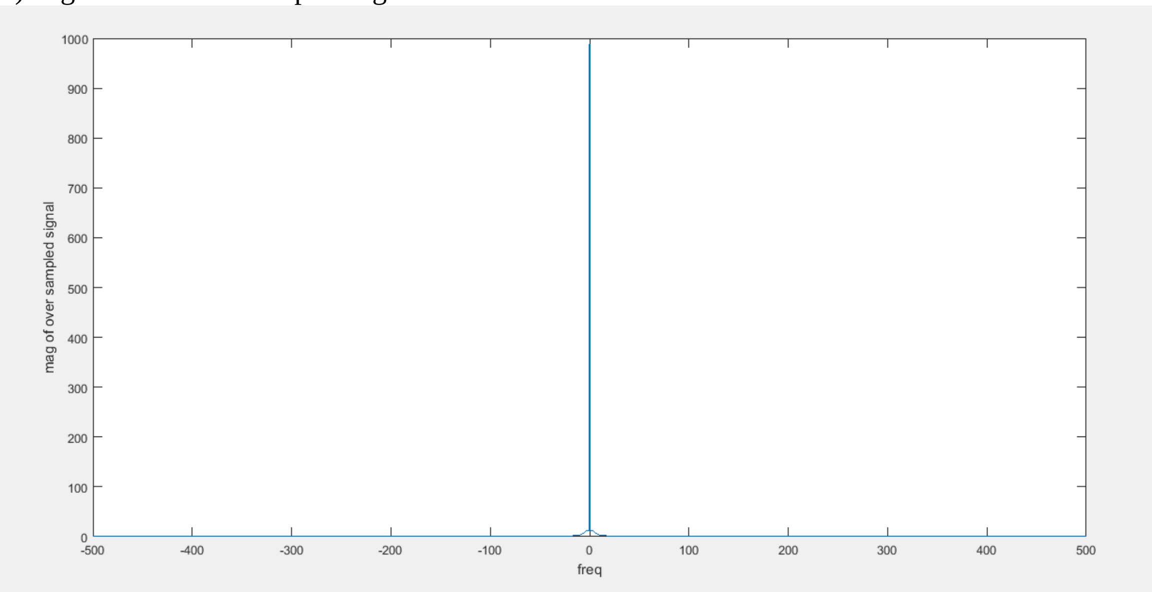
MSE:

5)over sampled signal:

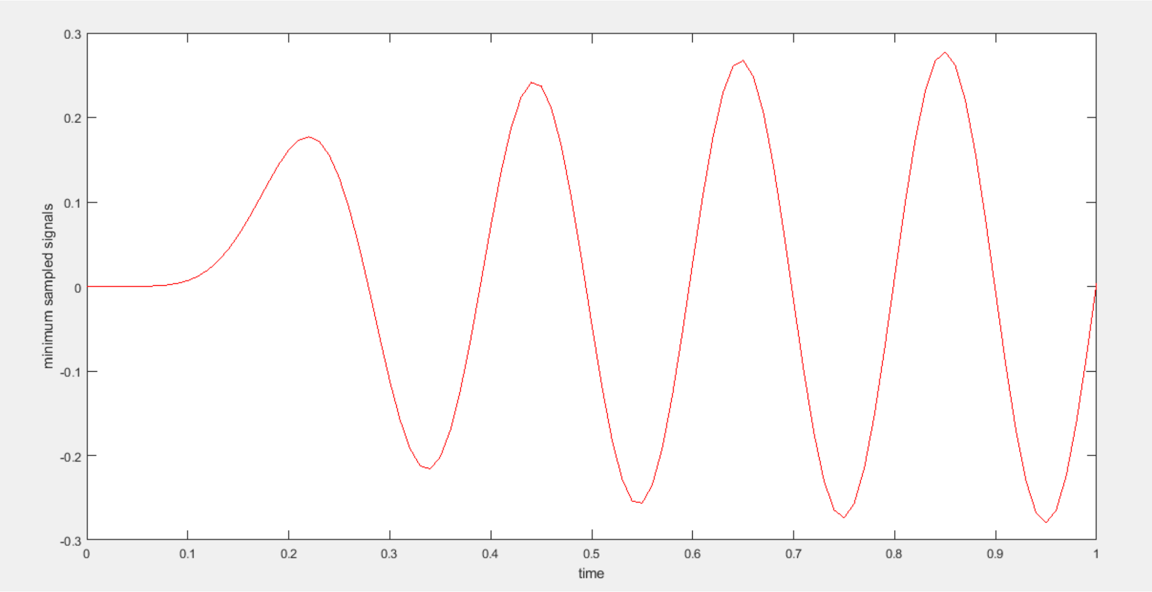


Signal can be easily reconstructed

6)magnitude of over sampled signal:

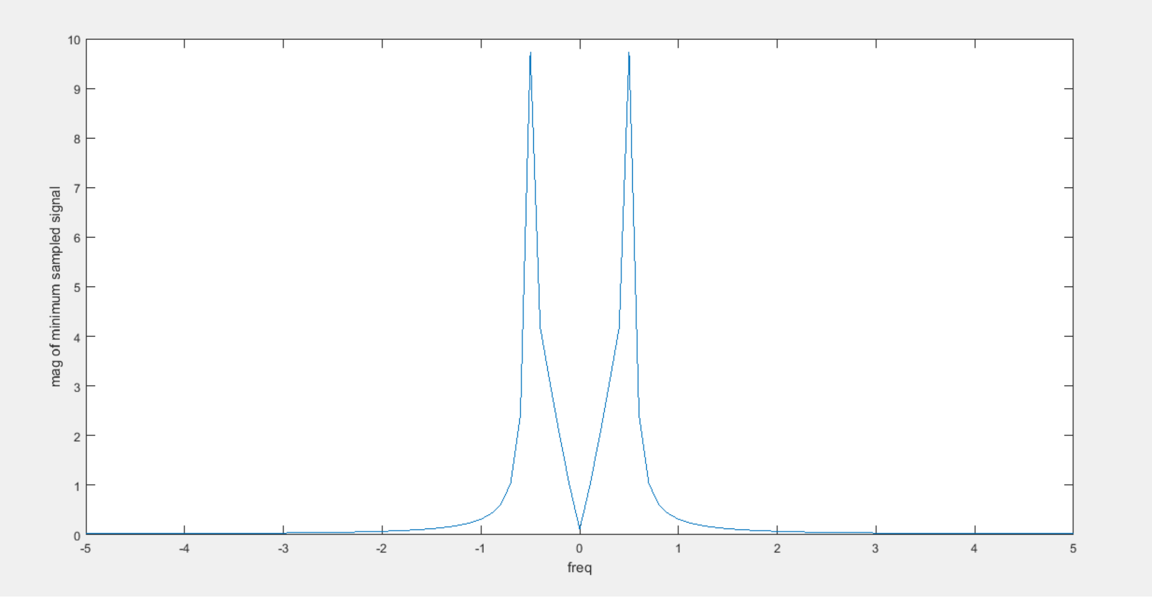


7)critical sampled signal:

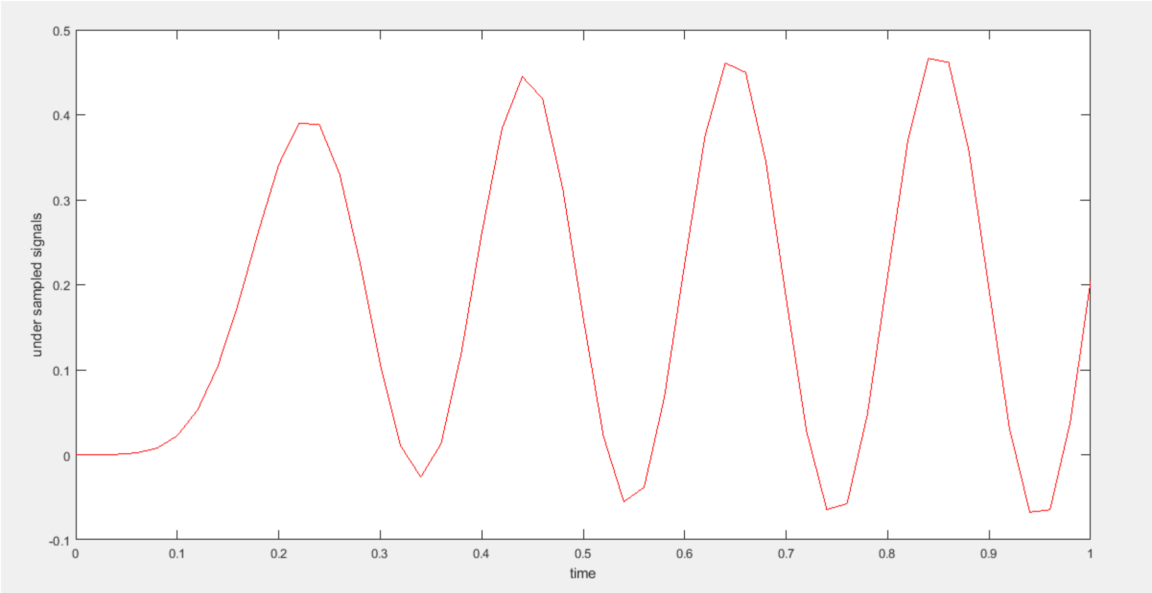


Signal is still close to the original one

8)magnitude of critical sampled signal:

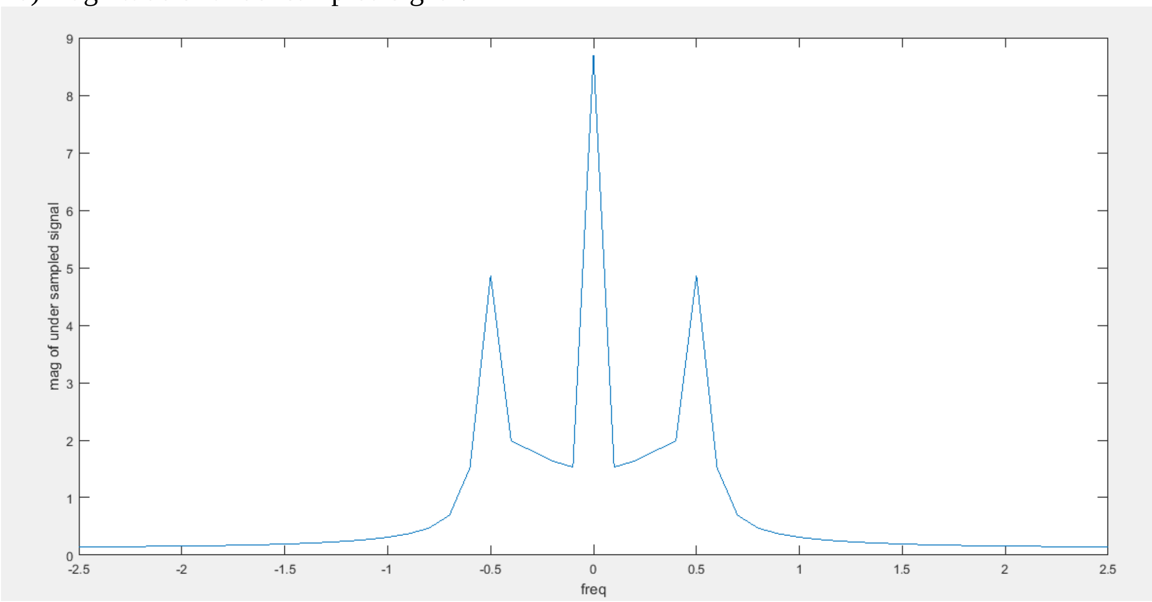


9)under sampled signal:



Signal can’t be reconstructed again

10) magnitude of under sampled signal:

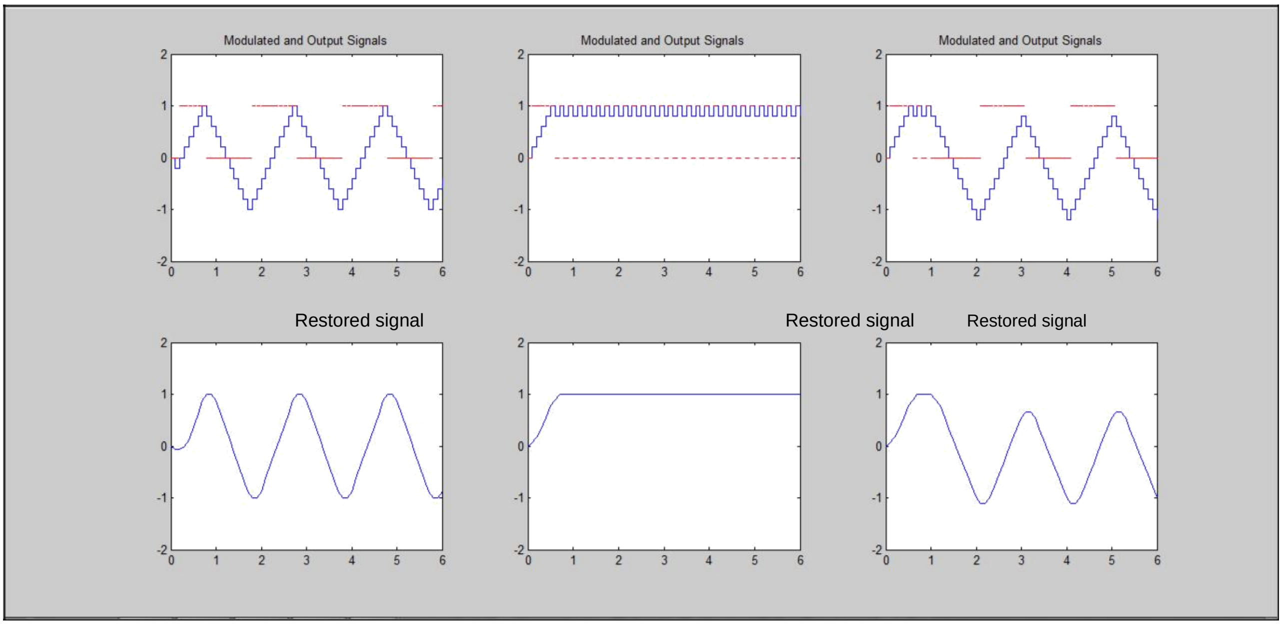


Delta Modulation Types

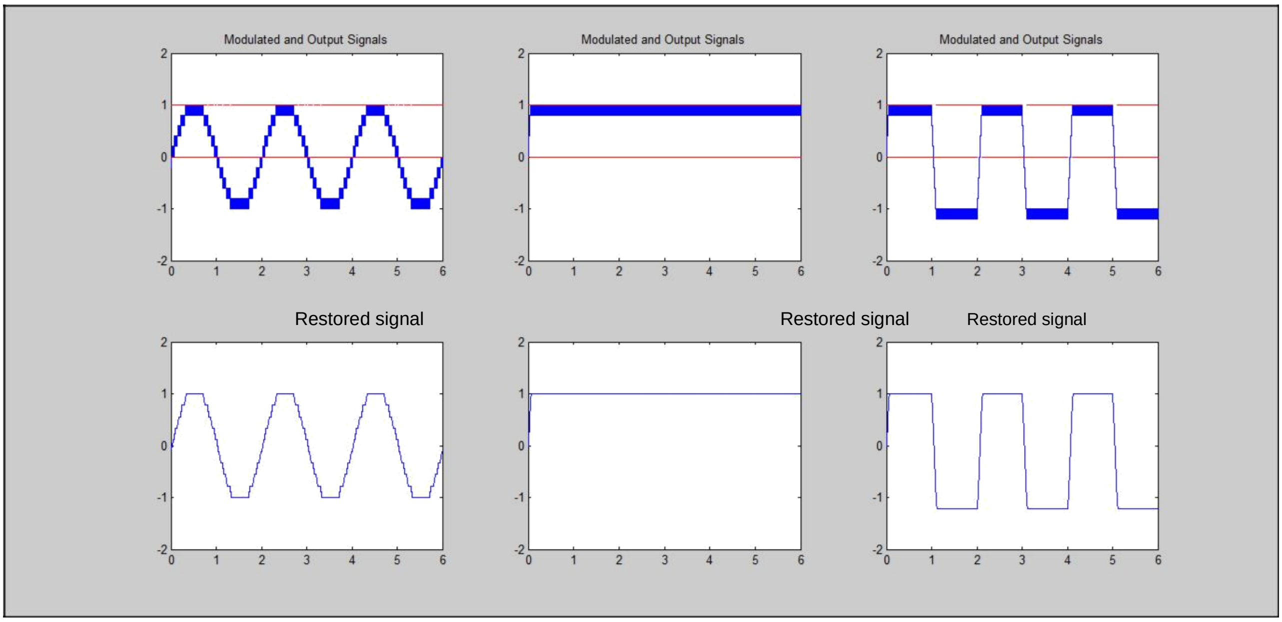
Name :Asmaa Moustafa Saad

I. <u>Delta-Modulated signal's, Output signal's, and Restored signal's plots</u>

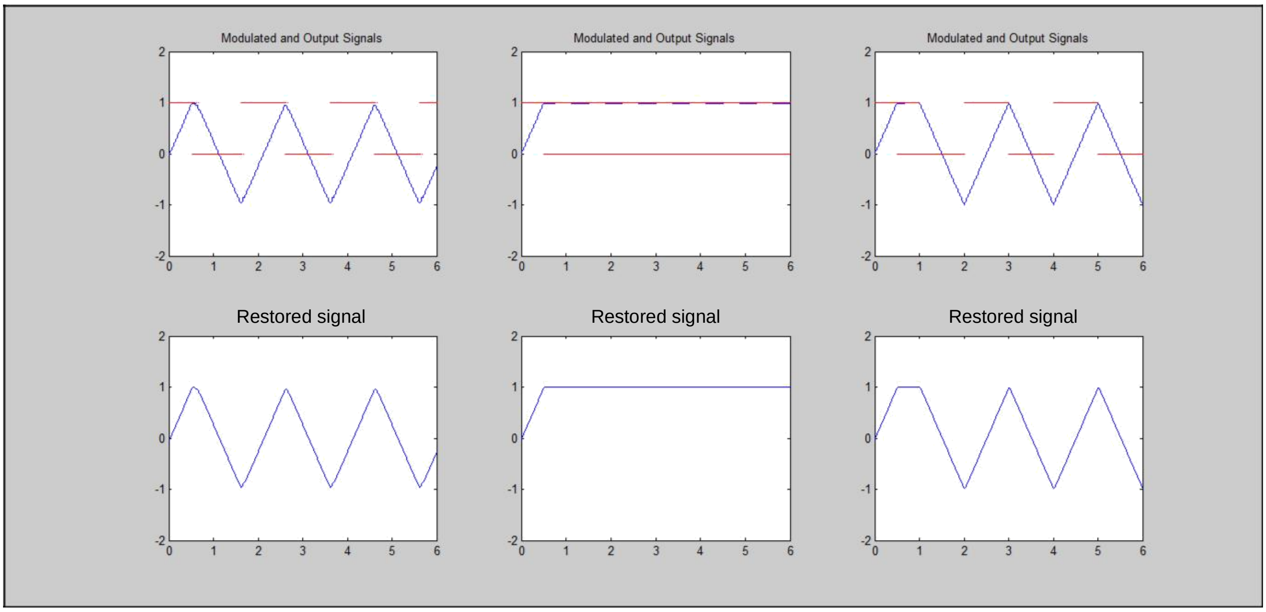
For chosen Ts = 0.1 ms and Step Size = 0.2 volt



3. For 0.1*Ts = 0.01 ms and Step Size = 0.2 volt



3 For Ts = 0.1 ms and 0.1*Step Size = 0.02 volt



4 <u>Mean Square Error</u>

	Ts = 0.1 ms *Step Size = 0.2 volt*	*Ts = 0.01 ms* *Step Size = 0.2 volt*	*Ts = 0.1 ms* *Step Size = 0.02 volt*
Sine Wave	0.4943	0.0097	0.0828
DC Level	**0.0539**	**0.0055**	**0.0307**
Square Wave	1.7736	0.2181	1.2175

We can see that the DC level signal always has the lowest Mean Square Error for different sampling times and step sizes.

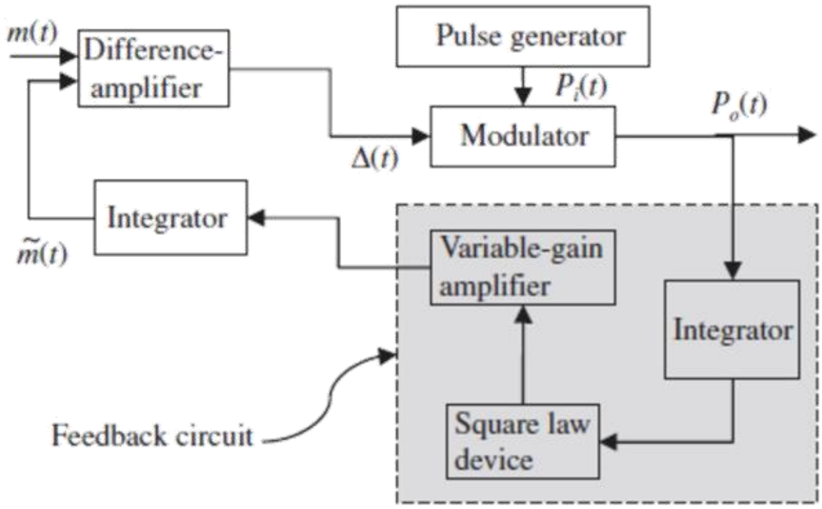
This due to the fact that the Delta Modulation detects the change in the signal and not the signal itself. Since the DC level signal is a constant signal, the delta modulated signal becomes similar to a transient response that takes small time to settle, which is very close to the actual signal.

5. <u>Reducing Slope Overload and Granular Noise</u>

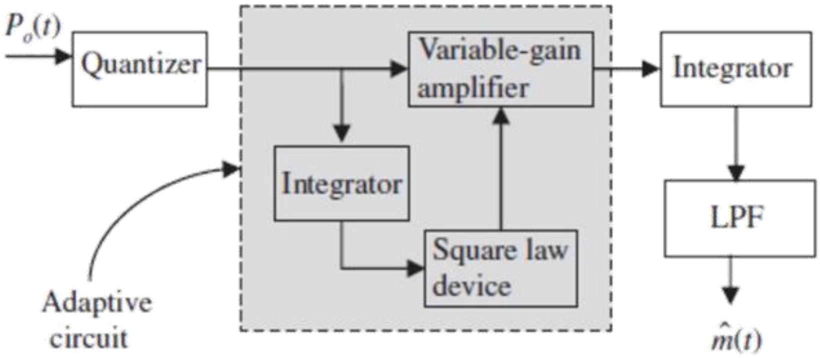
1) Slope overload occurs when the delta modulated signal is unable to follow the input message signal. Thus, the recovered signal gets distorted.

Solution: These drawbacks can be overcome by suitably changing the slop that can be changed either by changing the time period or the step size. Changing the time period will cause in changing the frequency and bandwidth.

So, a Delta Modulation system with variable step size, which is known as the *Adaptive Delta Modulation* (ADM), is used to overcome the limitations.
The ADM circuit is illustrated in the following block diagram:



(a) ADM transmitter

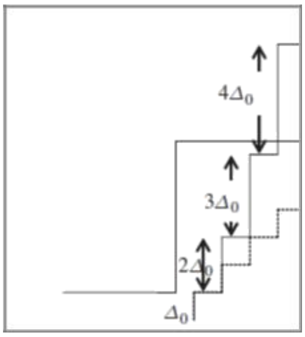


(b) ADM receiver

The Adaptive Delta Modulation can be achieved by two algorithms; Song Algorithm and Space Shuttle Algorithm.

For Song Algorithm,

- Positive slope of the modulated signal with respect to time results in the next value equal to the previous value added with a linear multiple of the first step size value ever as follows:



- The opposite to this applies to the negative slope case.

To sum up, the step size for this algorithm follows the following equations:

$$\Delta(n) = \Delta(n-1) + \Delta_0 \qquad if \qquad m(t) > \widetilde{m}(t)$$

$$\Delta(n) = \Delta(n-1) - \Delta_0 \qquad if \qquad m(t) < \widetilde{m}(t)$$

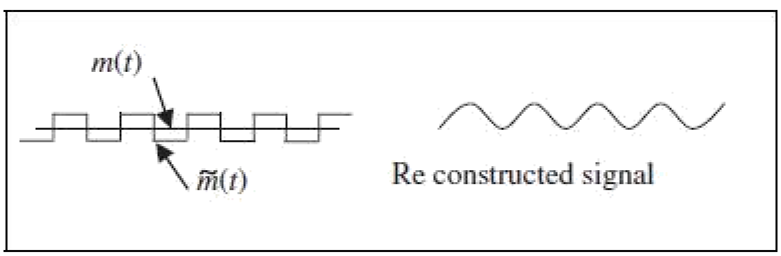
As for the Space Shuttle Algorithm, it's very similar to the Song Algorithm. However, instead of linear decrement space shuttle algorithm suggests constant decrement of amount Δ_0 to avoid the increase of the Granular Noise Effect. The step size for this algorithm follows the following equations:

$$\Delta(n) = \Delta(n-1) + \Delta_0 \qquad if \qquad m(t) > \widetilde{m}(t)$$

$$\Delta(n) = \Delta_0 \qquad if \qquad m(t) < \widetilde{m}(t)$$

2) Granular Noise

The error caused by this sort of noise can be overcome by reducing the step size to avoid large oscillations when restoring the signal.



Report On Experiment 6

Name : Mohamed Wagdy Nomeir

1 Introduction

Line codes, also called digital baseband modulation codes, are codes used in order to transmit a digital signal over a wired channel. Line codes are used to transmit a multi-level (digital) signal by a waveform that satisfy certain criteria depending on the channel state. There are di erent types of line codes, such as return to zero(RZ), non-return to zero(NRZ) and Manchester codes. In the following sections, we compare between each type of line code showing its advantages and disadvantages, we show a sample of the waveform of each type as well as the power spectral density.

1 Line Codes, A Comparison

In this section we will develop a comparison between six famous types of line codes; giving a brief description for each one.

2 NRZ : In this type a bit one is mapped to a voltage v, and a bit zero is mapped to a voltage v. NRZ is not a self clocking signal, so additional information is needed for clock synchronization. However it is very e cient for bandwidth where it uses half the bandwidth required by Machester encoding to transmit the same data.

2 Non Return to Zero Inverted(NRZ-I): This type the waveform depends on the current bit as well as the present value, beginning with initial condition the rst bit,1 or 0, is high(ie:v). if the next bit is zero it remains the same, else it is inverted to v. Clock recovery here is not suitable as well, additional information is needed to recover the transmitter clock. Its bandwith is also less than Manchester coding.

2 RZ: the rst half duration of each but resembles the NRZ. However the last half bot duration is always zero. The zero rest position is used as a guard to prevent ISI, although it requires double the bandwidth of NRZ to transmit the same data.

2 alternative Mark Inversion(AMI) : in this type the but zero is always at zero voltage. However the bit 1 is v if there are even number of ones before it and v if there are odd number of ones. with initial condition that the

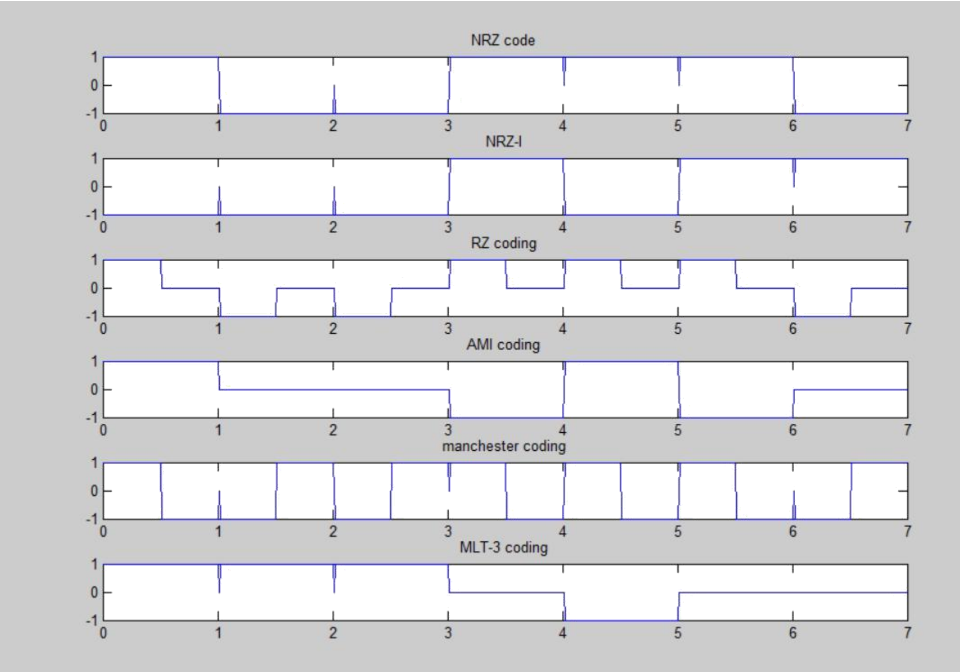


Figure 1: Di erent Waveforms for the same random input signal

rst one is high. In this type clock recovery can be done, however, long sequence of zeros will lose the synchronization.

3. Manchester coding : In this code if the bit is one the rst half bit duration is v and the second is v, and vice-versa for zero. It is very suitable for clock recovery of the transmitter at the receiver however is requires double the bandwidth of NRZ to transmit the same data.

4. MLT-3 coding: if bit zero is transmitted keep the previous level, if bit 1 then there are 4 transitions, high to neutral, neutral to low, low to neutral, neutral to high. It requires the four transitions (baud) to complete a full cycle. Thus it is suitable for copper wire transmission.The lack of transi-tion on a 0 bit means that for practical use, the number of consecutive 0 bits in the transmitted data must be bounded.

4. Figures and Comments

As shown in g. 1 di erent waveforms are generated for the same random binary input signal having the same bit duration. For description of each type refer to the previous section.

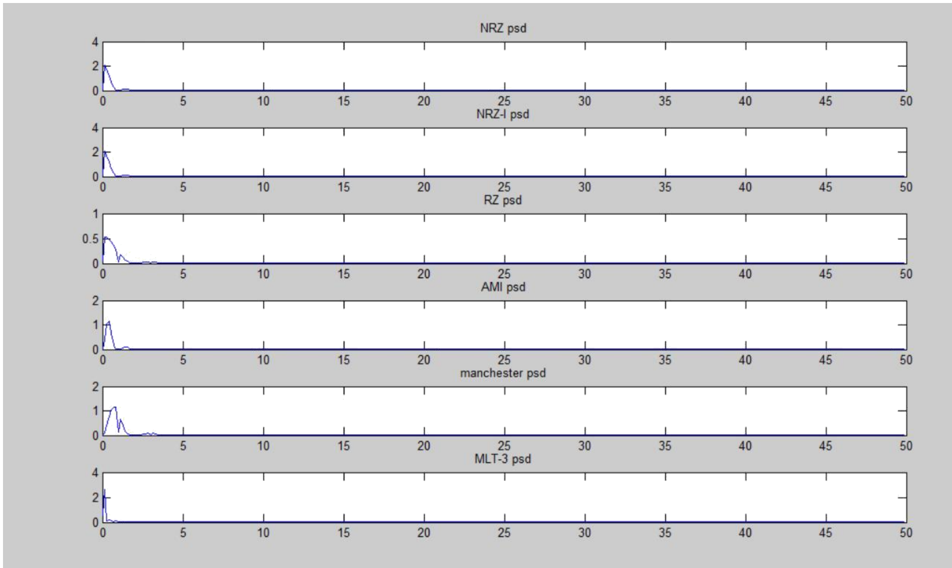


Figure 2: PSD of di erent code modulations for the same random input signal

Figure 2 shows the power spectral density of each modulation scheme for the same input binary random sequence. As shown manchester encoding has the largest bandwidth, this is due to its transition in half bit duration from v to v or vice versa, which makes it requires a larger BW compared to RZ which returns from v or v at half bit duration but to zero.

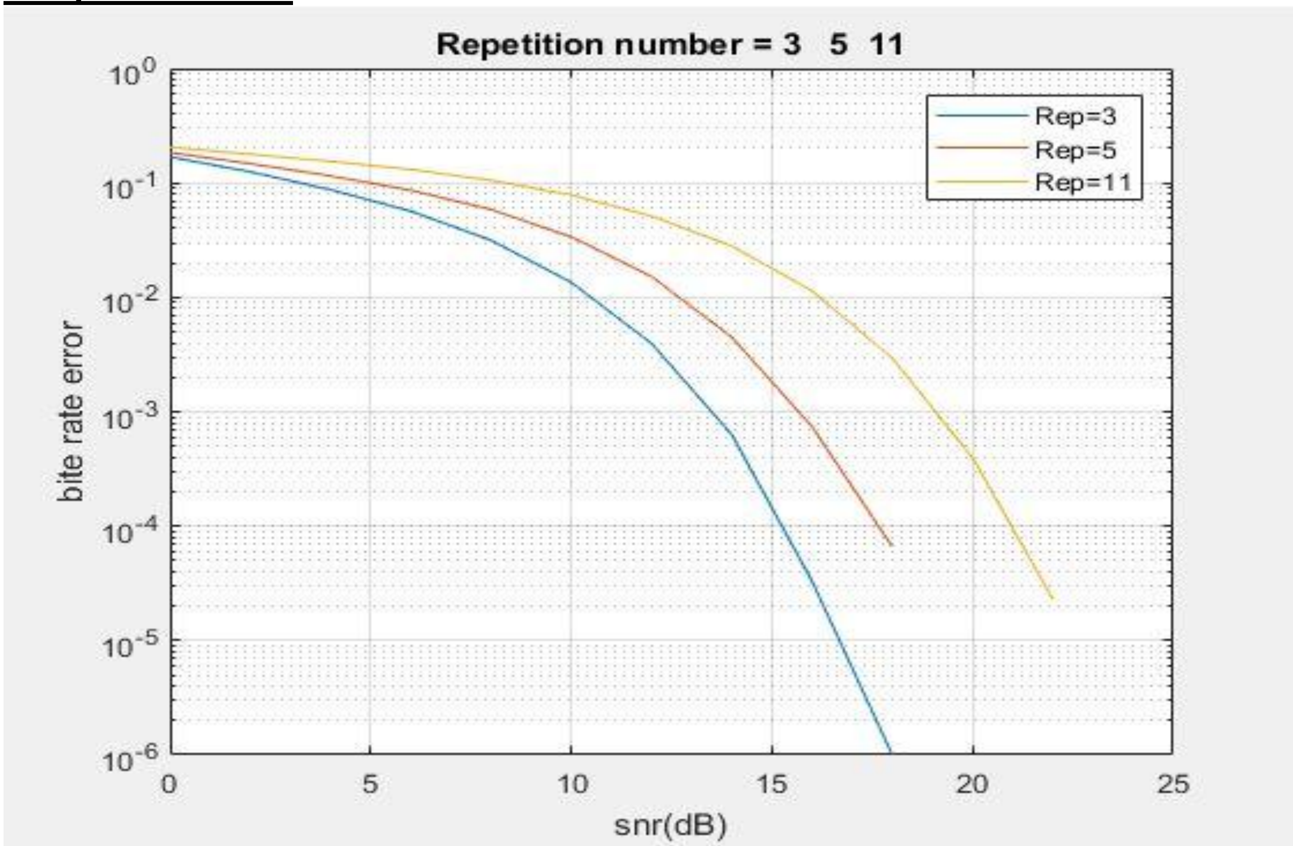
3 Other Types of Line Codes

Pulse-position modulation (PPM) : M-PPM collects every $\log_2$ M bits together, cuts their interval to M slots and sends a pulse at the slot index resembles their integer value. It can be considered as a generalization of the manchester coding. This line code is very e cient in bandwidth. However, it needs a high clock synchronization at the receiver.

Hamming codes : It collects a certain number of bitsn and adds a number of parity bitsk in order to detect and correct any error in the bits. Its main advantage is the error detecting as well as correcting property us-ing additional parity bits. However the rate of transmission decrease by increasing number of parity bits.

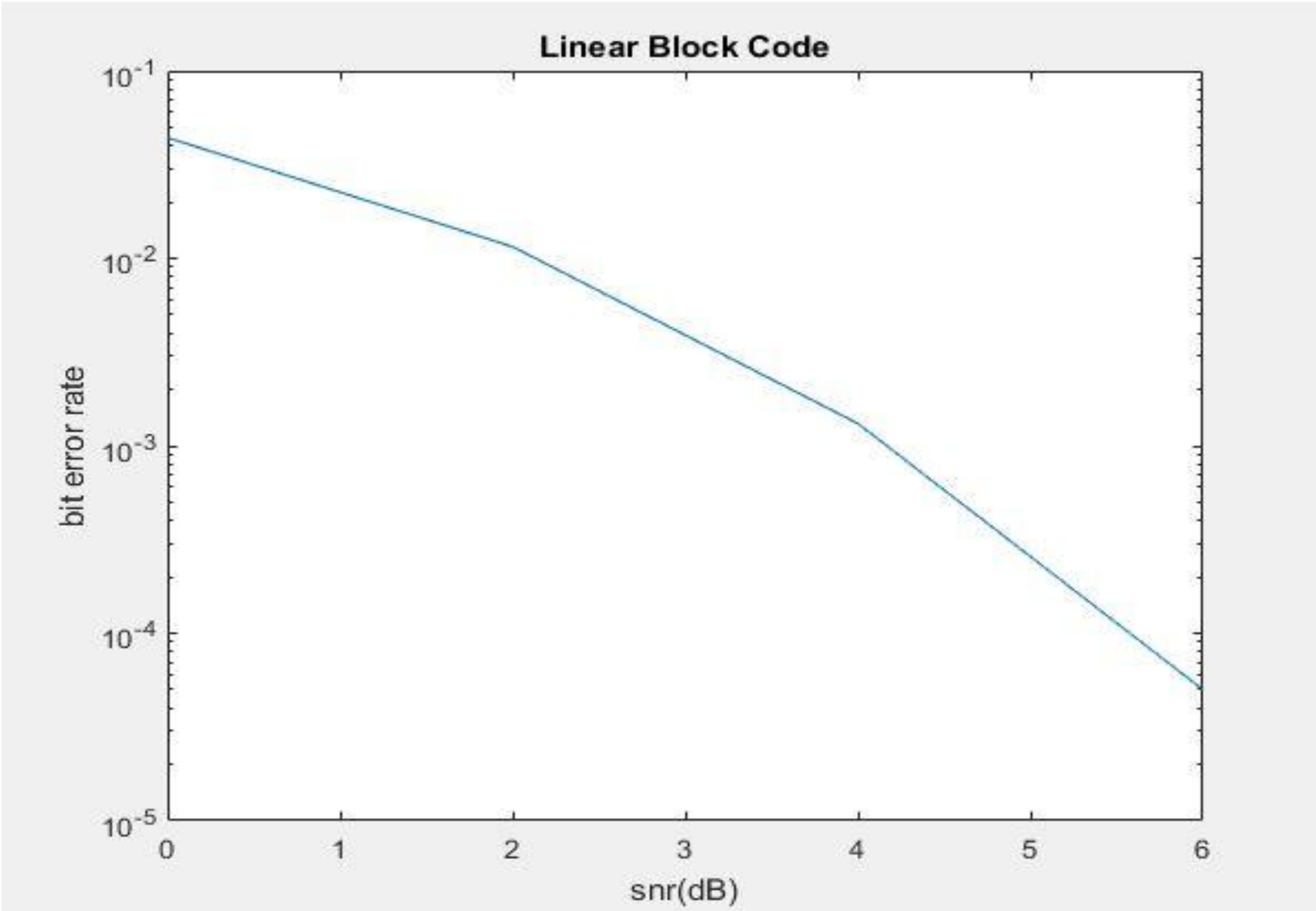
Channel coding
Exp 7
Name: Marwan Helmy Zaki

1-Repetition Code:

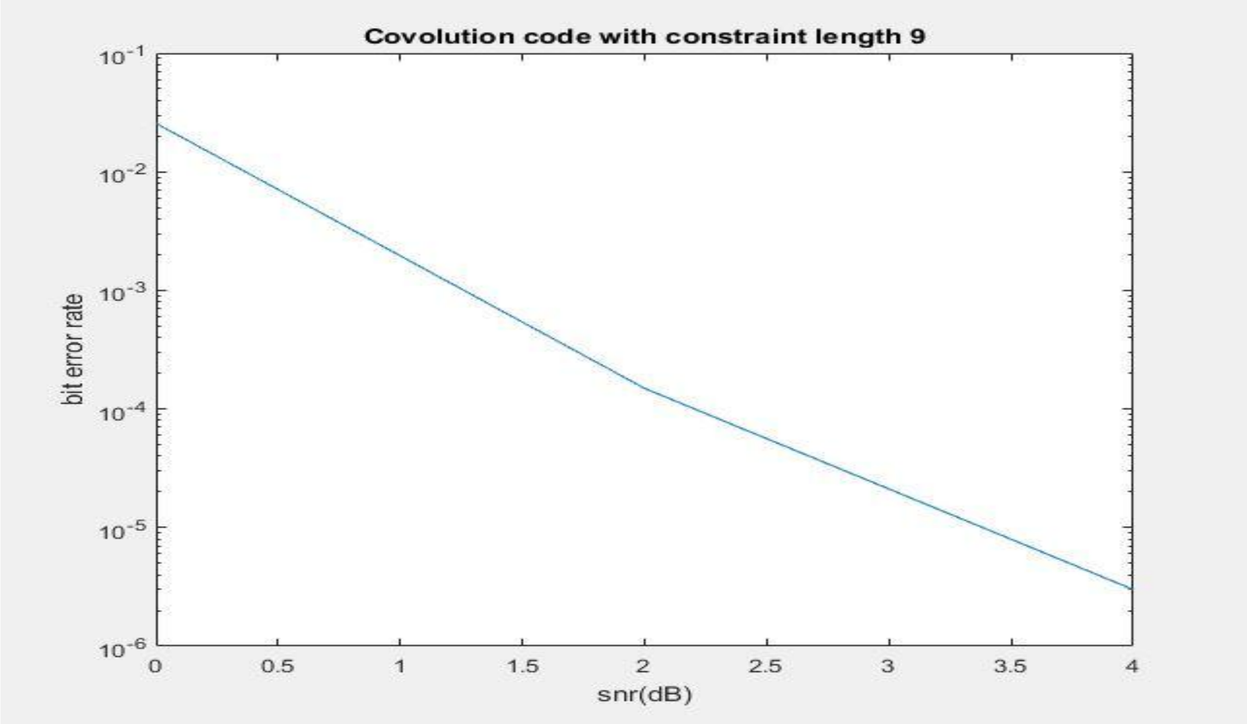


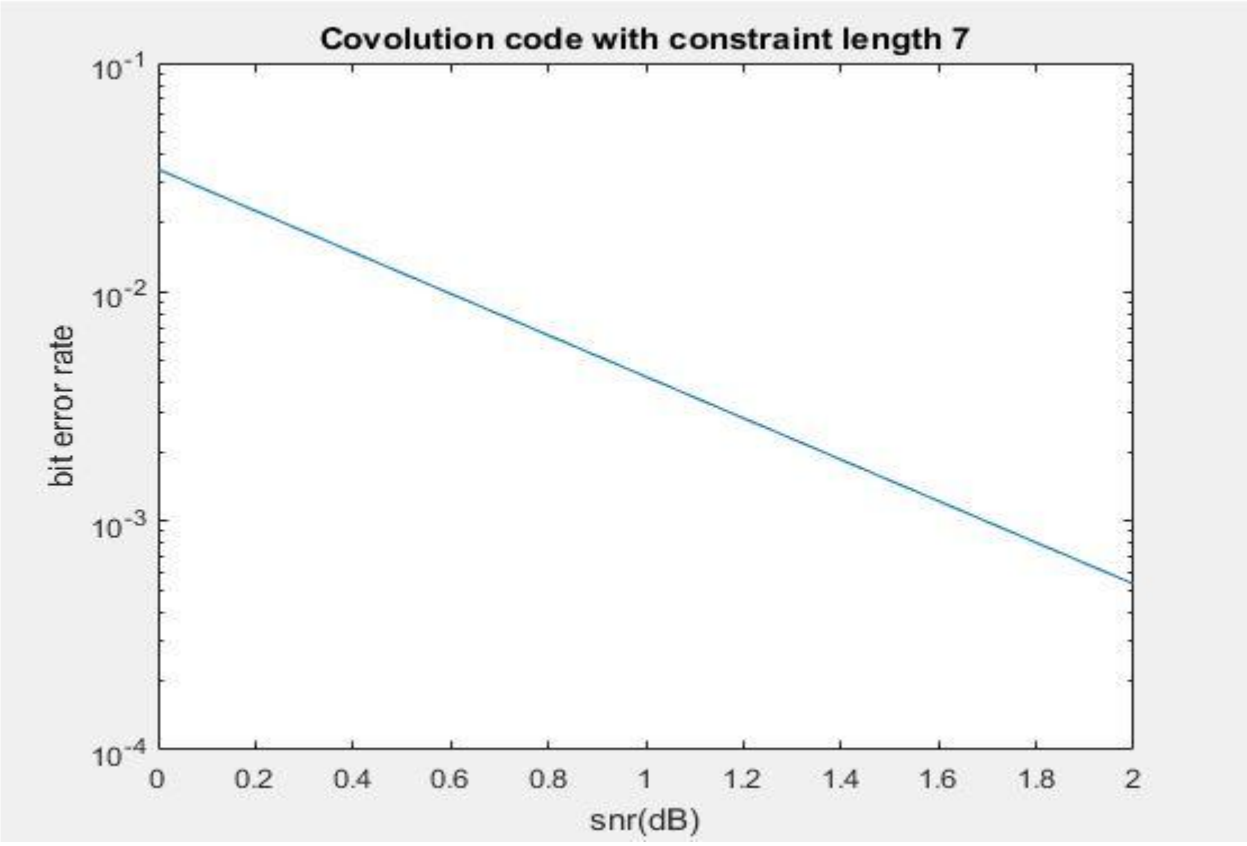
Repetition code channel is one of the worst channel coding due to the very high BER, in fading channels repetition codes has better BER with higher repetition factors, however, in our case here (AWGN) as the repetition factor increases the BERincreases, but the ease of its implementation is the reason we use it.

2-Linear Block Code:



3-Convolution Code:





Conclusion:

For AWGN channels, it’s not preferred to use repetition channel codding as it has the worst BER even with increasing the repetition factor, however, it can be used in other non-AWGN channels due to the ease in its implementation.

The Convolutional code has the least BER at higher SNR, then linear block code, then repetition code.