Communication Theory
Project: Communication System Model

Samkit Jain
2022102062
IIIT Hyderabad
Hyderabad, India
samkit.jain@students.iiit.ac.in

Aryan Garg
2022102074
IIIT Hyderabad
Hyderabad, India
aryan.garg@students.iiit.ac.in

***Abstract*—In this project, we present a comprehensive MATLAB simulation of a communication model encompassing various stages crucial to digital communication systems. The model follows the journey of information through successive transformations, starting with an Analog-to-Digital (A/D) conversion, followed by encoding, line coding, modulation, introduction of channel noise, demodulation, and concluding with Digital-to-Analog (D/A) conversion.**

The encoding scheme adopted in this simulation is 4-ary Frequency Shift Keying (FSK), providing robustness and efficiency in transmitting digital data. Each stage of the communication model is meticulously implemented to ensure accuracy and fidelity to real-world communication systems.

This project aims to provide a deep understanding of the intricacies involved in digital communication systems, offering insights into the impact of various parameters such as noise, modulation techniques, and encoding schemes on the overall performance of the system. Through MATLAB simulation, we explore the behavior of the system under different conditions, facilitating analysis, optimization, and further research in the field of digital communications.

I. Introduction

Digital communication systems have become indispensable in modern society, serving as the backbone for the exchange of information across vast distances. These systems rely on intricate processes and technologies to ensure the reliable transmission of data in various environments and conditions. Understanding the inner workings of these systems is crucial for engineers and researchers alike to design, optimize, and innovate in the field of telecommunications.

The communication model under examination follows a structured approach, incorporating stages such as encoding, line coding, modulation, and demodulation. Each stage introduces specific techniques and algorithms tailored to optimize data transmission and reception while mitigating the effects of noise and other disturbances inherent in real-world communication channels.

Central to our simulation is the utilization of a 4-ary Frequency Shift Keying (FSK) encoding scheme, chosen for its ability to efficiently encode digital data into frequency variations suitable for transmission. FSK offers robustness against noise and interference, making it a suitable choice for a wide range of communication applications.

Throughout this report, we aim to provide a detailed exploration of the simulated digital communication model, elucidating the underlying principles, methodologies, and challenges associated with each stage.

II. Analog To Digital Converter

The Analog-to-Digital Converter (ADC) is a fundamental component in digital communication systems, responsible for converting continuous analog signals into discrete digital representations. This process involves several key steps:

1.Sampling: The analog signal is sampled at regular intervals, capturing its amplitude values at specific points in time. These samples represent discrete snapshots of the original analog waveform.

2.Quantization: Each sampled amplitude value is quantized, meaning it is rounded or truncated to fit within a finite set of discrete levels. This step effectively discretizes the continuous amplitude values into a finite number of levels.

3.Binary Encoding: The quantized amplitude values are then encoded into binary format. This conversion assigns a binary code to each quantized value, representing it as a sequence of bits. The number of bits used to represent each sample determines the resolution of the ADC.

4.Data Transmission: The binary representations of the samples are organized into a data stream suitable for transmission through the digital communication system. This data stream can be further processed, modulated, and transmitted over a communication channel.

Overall, the ADC stage plays a crucial role in digital communication systems by enabling the conversion of analog signals into a format compatible with digital processing, transmission, and storage. The accuracy and fidelity of the ADC directly impact the quality and reliability of the communication system.

Below is a plot of 100 bits from the digitised sound signal.

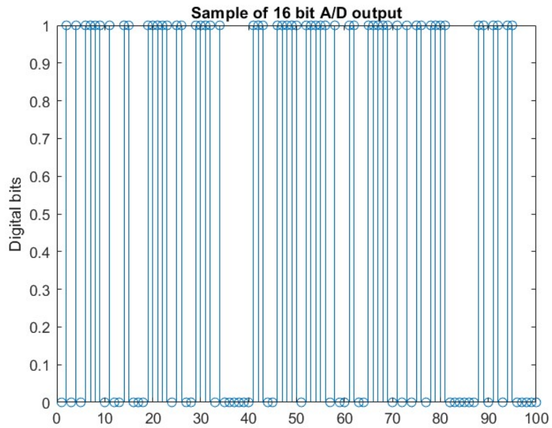

Fig. 1: Analog To Digital Converter Output

III. Encoding

Gray Code Mapping is a type of encoding scheme used in digital communication systems to map binary data to symbols with minimal transition errors. In Gray Code Mapping, adjacent symbols differ by only one bit, reducing the likelihood of errors caused by bit transitions.

Using the 4-ary FSK encoding scheme, we encode 2 bits with different carrier frequencies. The encoding is done according to gray coding.

Bits	Frequency shift
00	$0*\triangle f$
01	$1*\triangle f$
10	$3*\triangle f$
11	$2*\triangle f$

The value of $\triangle f$ can be calculated from binary FSK without loss of generality. In binary FSK,

$$1 : \sqrt{2}p'(t)cos(\omega_c + \triangle\omega/2)t$$

$$0 : \sqrt{2}p'(t)cos(\omega_c - \triangle\omega/2)t$$

Therefore,

$$p(t) = \sqrt{2}Acos(\omega_c + \triangle\omega/2)t$$

$$q(t) = \sqrt{2}Acos(\omega_c - \triangle\omega/2)t$$

$$E_{pq} = \int_0^{T_b} p(t)q(t)\,dt$$

$$= A^2T_b[\frac{sin(\triangle\omega)T_b}{(\triangle\omega)T_b} + \frac{sin(2\omega_c)T_b}{2(\omega_c)T_b}]$$

In practice $\omega_c T_b >> 1$, so the second term is negligible and hence,

$$E_{pq} = A^2T_b sinc(\triangle\omega Tb)$$

Similarly,

$$E_b = E_p = E_q = \int_0^{T_b} [p(t)]^2\,dt = A^2T_b$$

We know,

$$P_b = Q(\sqrt{\frac{E_b - E_b sinc(\triangle\omega T_b)}{2\eta}})$$

Finding the minimum of P_b, we get it at value,

$$\triangle f = \frac{0.715}{T_b}$$

So, the above frequency shift has been used to modulate the different symbols. Below is the plot of the encoded signal of the bits from the Analog to Digital converter.

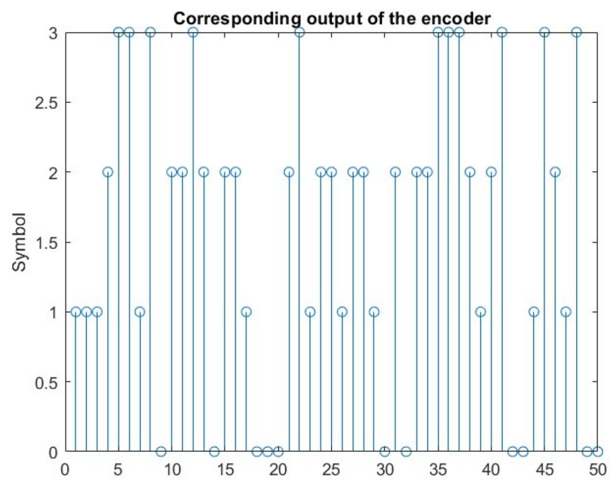

Fig. 2: Encoded samples

The cosine signals for each of the bit pairs is shown below,

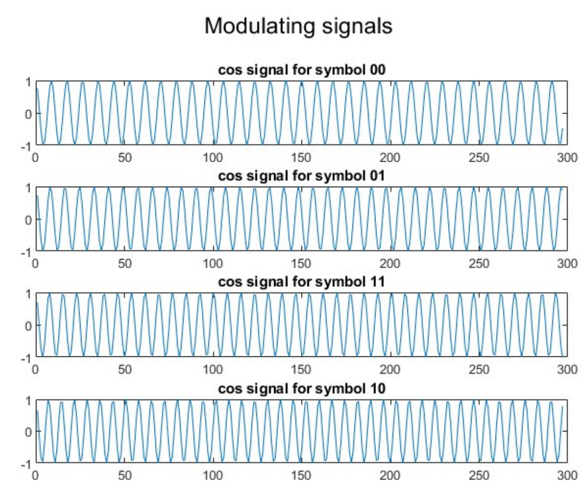

Fig. 3: Encoded samples

IV. Line Coding

The FSK encoding scheme does not have a line coding step. But to reduce the average signal power we use a Raised Cosine Pulse to shape the transmitting symbols.

The raised cosine pulse with roll off factor of 1 is shown below,

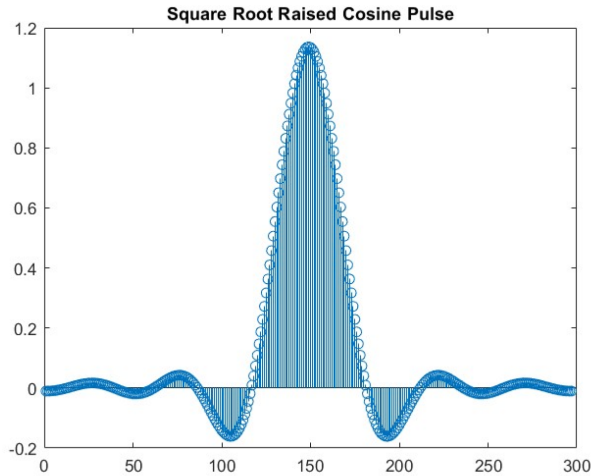

Fig. 4: Raised Cosine Pulse

Using the raised cosine pulse for pulse shaping has many benefits,

1. The bandwidth of the pulse is R_bHz and is a finite value at t=0.

2. It is zero not only at all the signaling instants but also at points midway between all the signaling instants.

3. It decays rapidly, as $\frac{1}{t^3}$. As a result, the full raised-cosine pulse is relatively insensitive to deviations of Rb, sampling rate and timing jitter.

We also used the rectangular pulse for the pulse shaping which is much more inefficient.

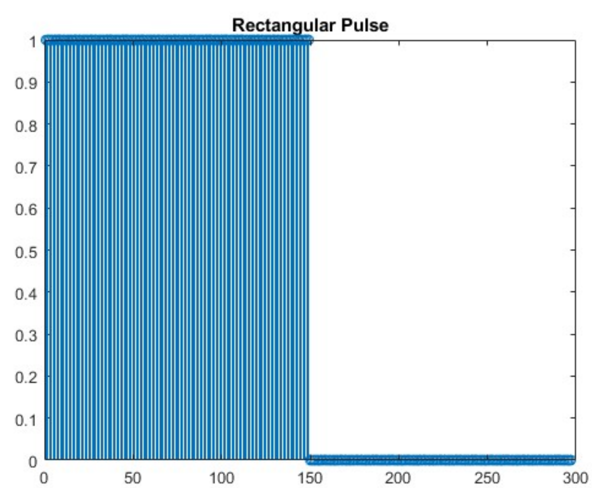

Fig. 5: Rectangular Pulse

V. Modulation

Using the frequency shift for each 2-bit pair, we modulate the symbols with cosine waves with frequencies of $f_c + m_i \triangle f$.

$$s_i(t) = p(t)cos(f_c + m_i \triangle f)$$

Below is the plot for the modulated symbols with the Raised Cosine pulse,

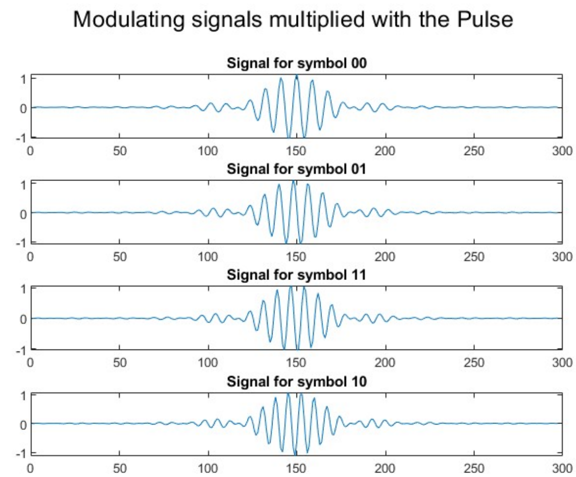

Fig. 6: Modulated pulse symbols with raised cosine

Multiplying the pulse with the cos waves of the symbols, we observe that the fft of the original Raised cosine shifts by the term $f_c + m_i \triangle f$.

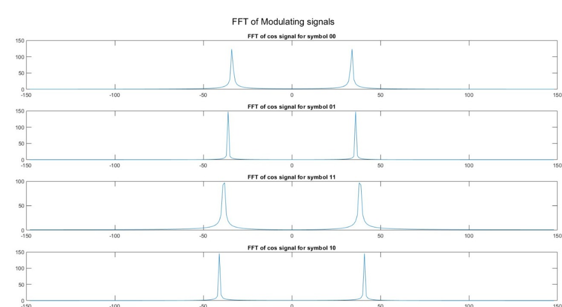

Fig. 7: FFT of the modulated pulse symbols

Following is a piece of the signal which has been modulated.

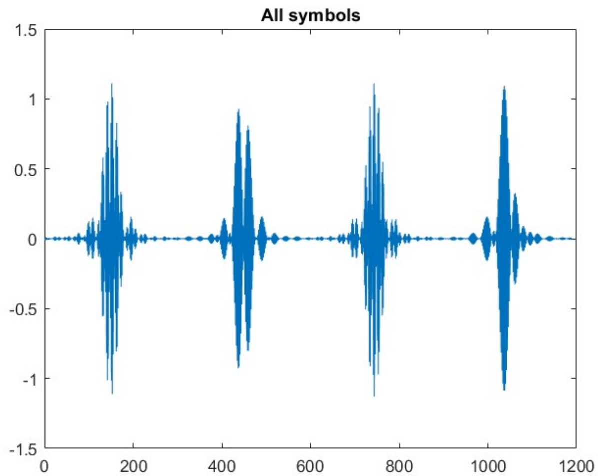

Fig. 8: Modulated signal with all 4 bit pairs

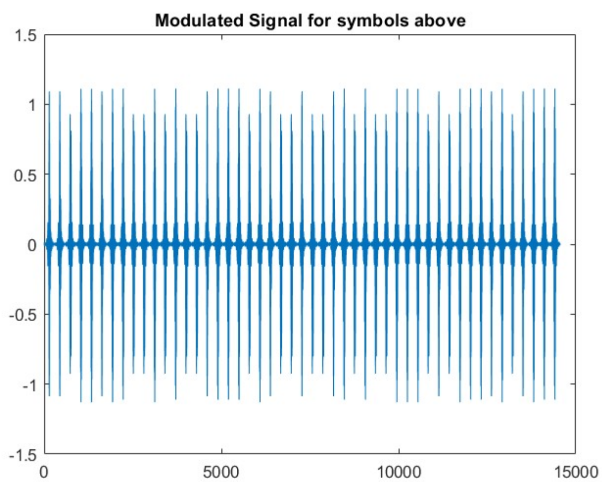

Fig. 9: Larger part of the modulated signal

Modulation with a half rectangular pulse gives the following output for each symbol,

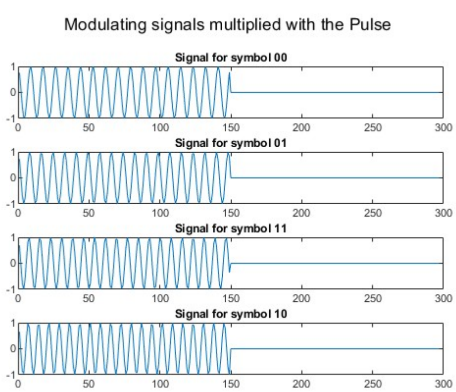

Fig. 10: Modulated pulse symbols with rectangular pulse

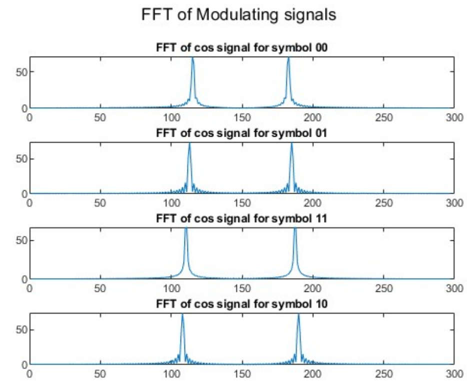

Fig. 11: FFT of the modulated pulse symbols

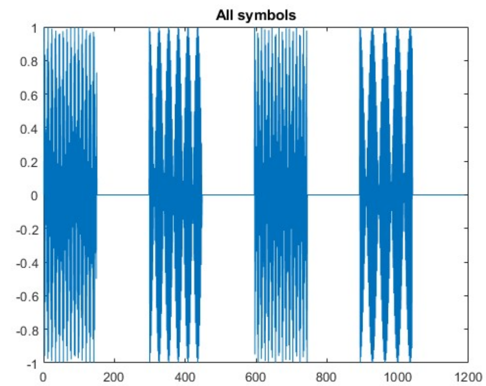

Fig. 12: Modulated signal with rectangular pulse with all 4 bit pairs

VI. Channel

After modulating the digital signal, we pass it through two channels:

1. Memoryless AWGN Channel: This channel is memoryless i.e the noise added to the signal is independent of previous symbols and their noise. This lack of correlation between symbols makes the channel memoryless. Mathematically it is represented by the following equation:-

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

Here r(t) represents the received signal, s(t) represents the transmitted signal and n(t) represents the Gaussian noise.

2. AWGN Channel with Memory: An AWGN channel with memory incorporates some form of dependence between the noise affecting different symbols. This can arise due to various factors such as intersymbol interference (ISI). Mathematically it is represented by the following equation:-

$$r(t) = h(t) * s(t) + n(t)$$

Here r(t) represents the received signal, s(t) represents the transmitted signal and n(t) represents the Gaussian noise and h(t) is:-

$$h(t) = a\delta(t) + (1-a)\delta(t - bT_b)$$

where T_b is the transmission time for one bit and a and b are constants.

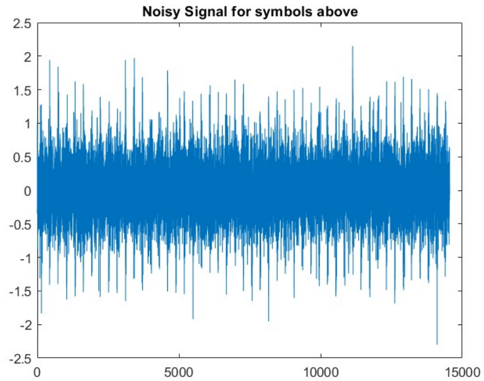

Fig. 13: Noisy modulated signal for SRRC pulse

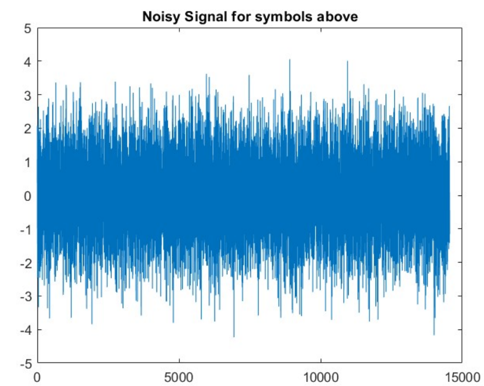

Fig. 14: Noisy modulated signal for rectangular pulse

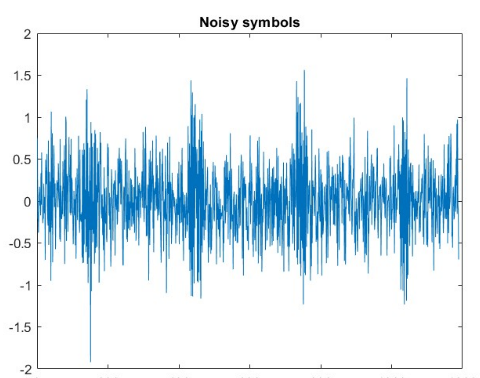

Fig. 15: Noisy symbols corresponding to the shown sample for SRRC pulse

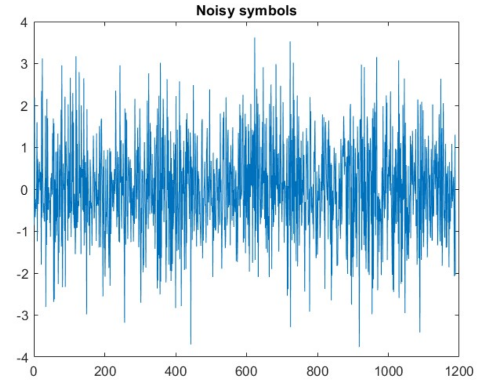

Fig. 16: Noisy symbols corresponding to the shown sample for rectangular pulse

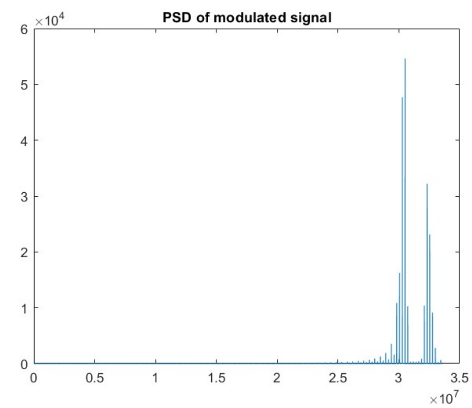

Fig. 17: PSD of modulated signal for SRRC pulse

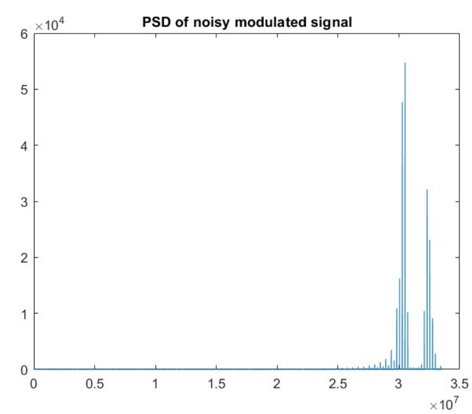

Fig. 18: PSD of Noisy modulated signal for SRRC pulse

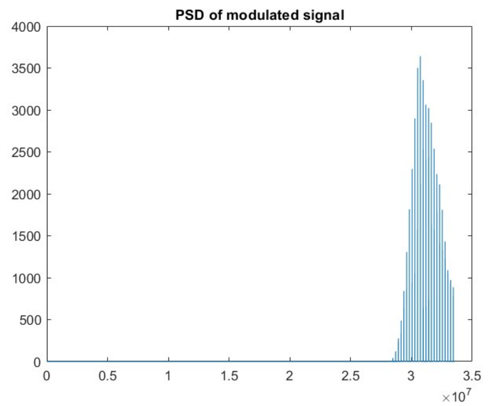

Fig. 19: PSD of modulated signal for rectangular pulse

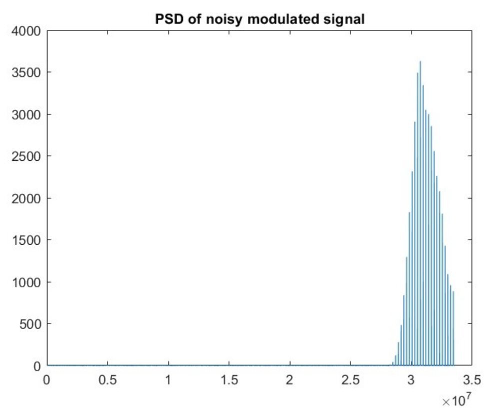

Fig. 20: PSD of Noisy modulated signal for rectangular pulse

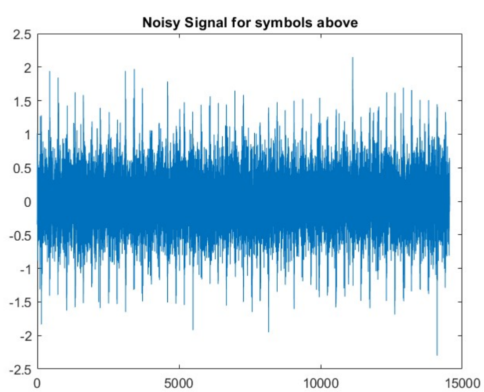

Fig. 21: Noisy modulated signal for SRRC pulse

VII. Demodulation

For FSK demodulation, we use a correlation detector. We correlate each modulating symbol to the received signal by integrating(correlate) them. Correlation measures the similarity between two signals therefore by comparing the results we can find make a decision on which symbol was modulated. It's relatively simple to implement and can provide robust demodulation performance even in the presence of noise and interference. Thus, a correlation detector is a widely used technique due to it's simplicity in implementation, robustness against noise and effectiveness in recovering data from the FSK modulated signal. The equation for correlator detector is as follows:-

$$r(T_m) = \int_0^{T_s} y(t) * cos(2\pi f t)\, dt$$

where f is the frequency of the different modulating symbols.+

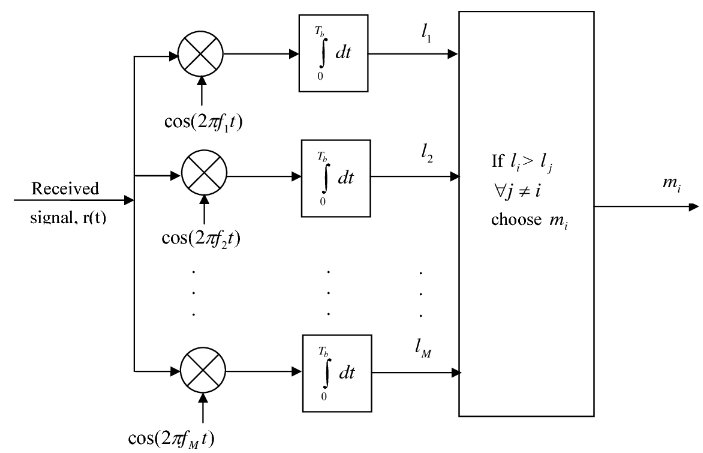

Fig. 22: Correlation Detector for Coherent Demodulation of FSK

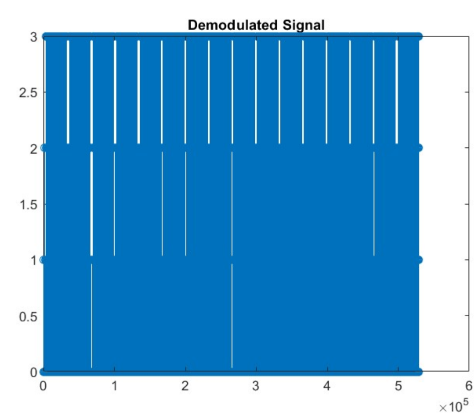

Fig. 23: Demodulated Signal after decision making for SRRC pulse

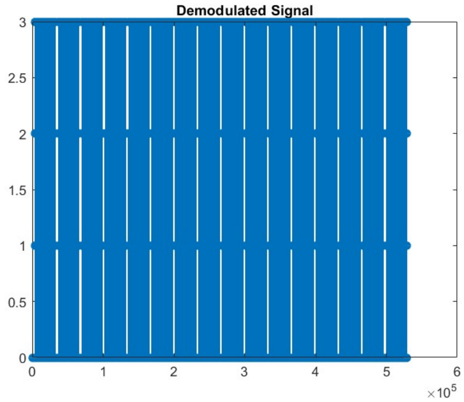

Fig. 24: Demodulated Signal after decision making for rectangular pulse

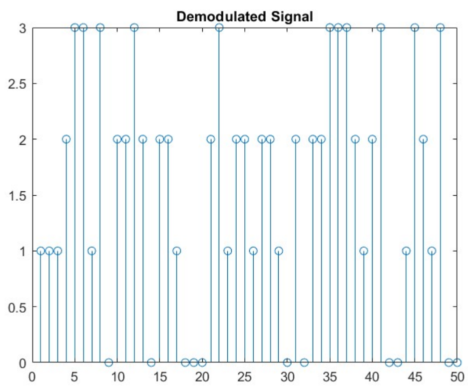

Fig. 25: Demodulated Signal for sampled version for SRRC pulse

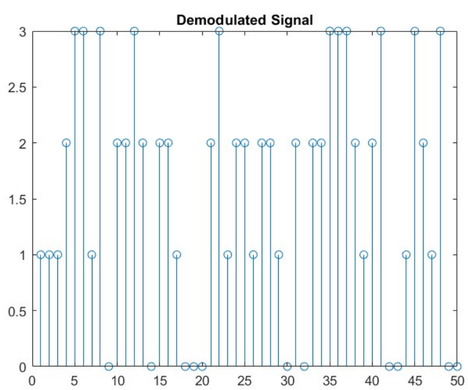

Fig. 26: Demodulated Signal for sampled version for rectangular pulse

VIII. Decoder

The decoder assigns 2 bits corresponding to each of the symbol chosen by the demodulator according to the Gray Code mapping and returns a bit stream.

Symbol Frequency	Bits
$0*\triangle f$	00
$1*\triangle f$	01
$3*\triangle f$	11
$2*\triangle f$	10

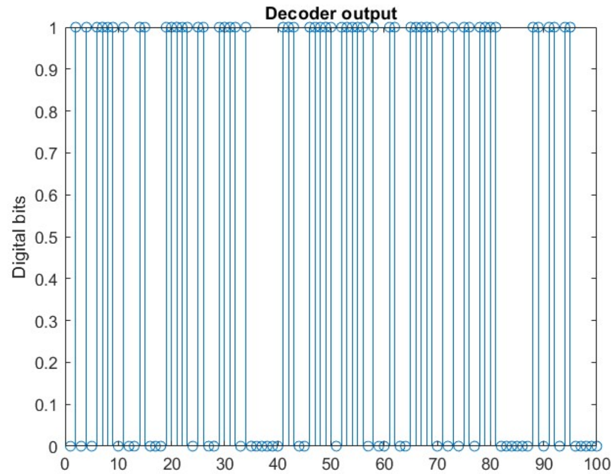

Fig. 27: Decoder output for the sampled version for SRRC pulse

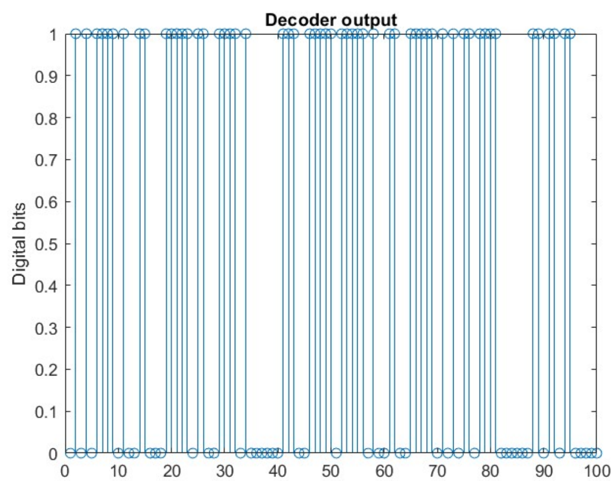

Fig. 28: Decoder output for the sampled version for rectangular pulse

As we can see that the bits are mapped in a Gray Code scheme as adjacent symbols are only one different from each other.

IX. D/A Converter

The Digital-to-Analog converter takes the output bit stream and segments it into 16 bits. Each of these 16 bits is converted to an integer value and normalised to be from [-1,1] by dividing it by a constant. These values are sequenced together and converted to an analog signal, thus reconstructing the original audio signal. Although the output signal is slightly distorted due to quantization noise.

X. Observations

We receive the same sound as given as input to the communication model albeit with some distortions due to errors obtained in bit-mapping from intersymbol interference (ISI) and the additive White Gaussian Noise (AWGN). The probability of error due to AWGN decreased when increasing the SNR, this is due to the decreasing power of the noise compared to the modulated signal. This can be seen from the figure below

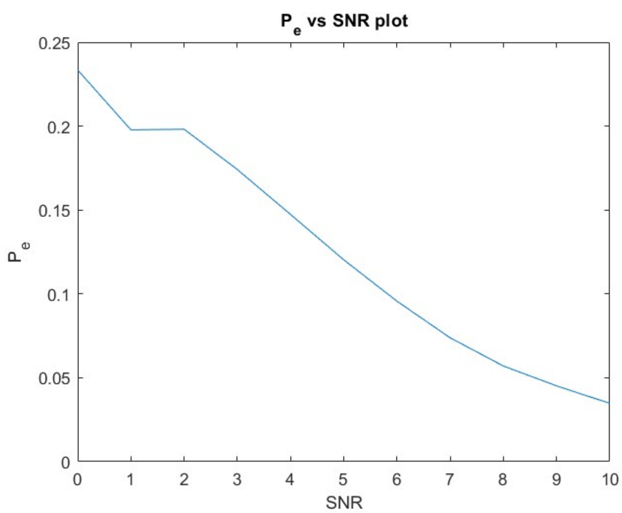

Fig. 29: P_e vs. SNR graph

We also observed how a memoryless channel performs better than the AWGN channel with memory. This is due to the output of the memory channel not being affected by previous symbols and their noise. The memory channel had elements of the previous symbols simulating ISI i.e interference from previous symbols resulting from errors in receiving the signals. This causes higher error in the channel with memory compared to the memoryless channel.

```
Probability of error is 0.29462 for a = 0.5 and b = 1
```

```
Probability of error is 1.8896e-06 for a = 0.75 and b = 1
```

```
Probability of error is 0.58355 for a = 0.25 and b = 1
```

```
Probability of error is 0.60508 for a = 0.25 and b = 2
```

```
Probability of error is 0.30422 for a = 0.5 and b = 2
```

```
Probability of error is 3.7793e-06 for a = 0.75 and b = 2
```

As we can see as we increase the value of a the error decreases caused by the lower interference of previous symbols. Similarly, for increasing b we see a decreasing error. This error is less than the error obtained for the memoryless channel.