EE 453 – Software Defined Communications

Effects of AGC and Linear Block Coding Algorithms on PAM Radio with Noisy Channel

Final Project Report
16 June 2021

Objective

Software defined radio provides the transition from the hardware to the software implementation of the communication systems. Since, it is considered as milestone of the communication systems area. Applications becomes more flexible and cheaper within this change. Thus, the communication technologies also become easy to be improved. However, designing a reliable software defined communication system can be challenging. Especially the recovery of the transmitted message signal may be difficult without proper receiver design. Aim of this project designing a software defined radio that can smoothing away all of the corruptions during the transmission of a text massage. Normally, QAM and QPSK can provide improved

solutions for the noisy channel systems. However, to analyze the effects of the changed blocks properly, The most primitive system, the PAM system, was used. As it is known, the PAM system is the most vulnerable system when it comes to noise. Optimal actions need to be taken to properly transmit the signal. These operations were tried to be provided by making changes to the blocks.

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1. Background Information

1.1. Transmitter

1.1.1. File to Binary Converting and Linear Block Coding

Our concern in this project is to transmit a message from an analog medium (Air). However, with the developing technologies, more digital systems started to be used. In the transmitter part, our priority is to transfer the message to a digital environment. One of these transfer methods is ASCII encoding. In this encoding method, each letter is converted to its bit counterpart. Large documents have a lot of bitstreams. To combat this, digital modulation that can act as a grouping can be used.

The fundamental difficulty in the transmission is noise, interference or error, which compromises the communication system's efficiency. The coding process used to control the incidence of mistakes is known as error control coding. These methods aid in the detection and correction of errors. Depending on the mathematical concepts used, there are a variety of error-correcting codes. However, these codes have traditionally been divided into two categories: Linear Block Codes and Convolution Codes [55][58].

The practice of adding extra bits to a digital word end in order to increase transmission reliability is known as Block Coding. Any error-correcting code that works on a block of bits of k input data to create n bits of output data is referred to as a Block Code (n,k). k is the original bit length and n is the coded message length. In other words, bits are mapped with the blocks which has n bits in each block. Mapped blocks can be called as Codewords (C). Here n is greater than k [57]. The ratio of the k/n gives the Code Rate (R). For Block Code operations, Code Rates must be less than 1. A high rate indicates a big quantity of real message per transferred block. In this case, the rate represents the transmission speed, whereas the quantity represents the cost caused as a result of the block code encoding. Low Code Rates can be used to prevent mistakes [60].

The word is made up of message bits (also known as data or information) and code bits. Added new bits (n-k) can be called as Redundancy or Parity Bits [56]. Parity bits aid in mistake detection and correction, as well as data location. The message bits are represented by the leftmost bits of the code word, whereas the parity bits are represented by the rightmost bits of the code word [57].

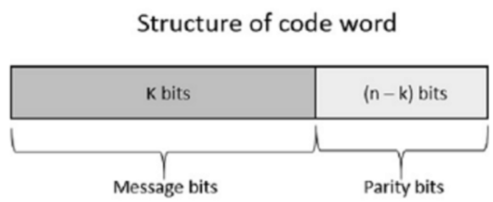

Figure 1: Block coded data structure

The parity bits and message bits in Linear Block Codes have a linear combination, meaning the final code word is the linear combination of any two code words. Linearity provides easier implementation and analysis [57]. The basic method of addition and multiplication of bits can be used to define the code construction in matrix form. These are modulo-2 addition and modulo-2 multiplication, respectively [59].

$$
\begin{array}{c|cc}
\oplus & 0 & 1 \\
\hline
0 & 0 & 1 \\
1 & 1 & 0
\end{array}
\qquad\qquad
\begin{array}{c|cc}
\cdot & 0 & 1 \\
\hline
0 & 0 & 0 \\
1 & 0 & 1
\end{array}
$$

Figure 2: Basıc linear operations

The Linear Block Coding process consists of encoding and decoding parts. 3 basic matrices are needed to complete the process. The first matrix is Generator Matrix. This matrix can be used to generate Codewords (C). The size of the matrix is only k x n bits [59]. The Generator Matrix and generated (coded) data are given below.

$$
G = \begin{bmatrix} 1 & 0 & 1 & 0 & 1 \\ 0 & 1 & 0 & 1 & 1 \end{bmatrix}
$$

00 ←→00000
01 ←→01011
10 ←→10101
11 ←→11110

Figure 3: Generator matrıx and coded data

Keep in mind that, the first 2 bits of the coded data is the same with the original data. After the linear operations (multiplication and addition), parity bits added the end of the original data bits. In the encoding part, generated (coded) data will be get.

To obtain the original data, decoder part should be placed to the receiver side. In the decoding part, the first step starts with Parity Check Matrix. This matris also known as Hadamard Matrix. The size of the Hadamard Matris is n x (n-k). Coded datas (incoming datas) will be multiplied with Hadamard Matrix. If there is no error, multiplication of 2 matrices will give the zero. In this case, original data can be obtained directly. If there are one or multiple errors, an extra matrix is needed. This matrix is called as Syndrome Matrix. The Syndrome Matrix includes all possible outcomes. According to Hamming Distance between incorrect data and possible correct data on the Syndrome Matrix, system tries to obtain original data. Linear Block Coding is good at solving a single bit error. However, for multiple errors, system may or may not (most probably) able to solve errors so for multiple errors, Linear Block Coding is not suitable[58].

$$
H^T = \begin{bmatrix} 1 & 0 & 1 \\ 0 & 1 & 1 \\ 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 1 \end{bmatrix}
$$

```
syn=[0 0 0 0 0;
     0 0 0 0 1;
     0 0 0 1 0;
     0 1 0 0 0;
     0 0 1 0 0;
     1 0 0 0 0;
     1 1 0 0 0;
     1 0 0 1 0];
```

Fig 4: Hadamard matrıx and coded data

It may also, as in the present case, contain a frame synchronization bit.Thus, its adds extra bits (redundancy bits) which helps in synchronization at receiver's and sender's end and also providing some kind of error detecting capability. However, this system useful for only one bit error detection and this would require an increased transmission bandwidth [55].

1.1.2. Digital Modulation Methods

The method of encoding a digital information signal into the amplitude, phase, or frequency of a broadcast signal is known as digital modulation. While the method was developed for antenna transmitting, it is now used in all forms of wired and wireless communications. Some digital modulation techniques are given below.

- **PAM (Pulse Amplitude Modulation)**

The message information is encoded in the amplitude of a sequence of signal pulses using pulse-amplitude modulation (PAM). Pulse-amplitude modulation is commonly used in non-baseband applications to modulate digital data transfer signals. The number of pulse amplitudes is reduced to a power of two with digital PAM. For instance, in 4-level PAM there are 2^2 possible discrete pulse amplitudes; in 8-level PAM there are 2^3 possible discrete pulse amplitudes. Figure 1 demonstrates the PAM graph. "1" refers to the original signal, "2" refers to PAM modulation and "a" refers to corresponding amplitude values of PAM. Level numbers cannot change constellation efficiency[1].

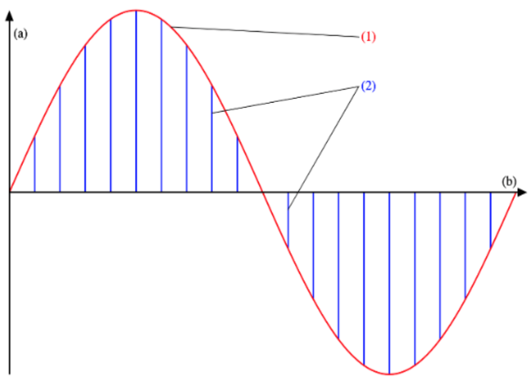

Figure 5: Pulse Amplitude Modulation[2]

- **FSK (Frequency Shift Keying)**

Baseband signal can change the frequency of carrier signal. We can assign more than one frequency for each symbol (each frequency represents different symbols). The amplitude of the carrier will be the same. It means the energy constellation of FSK in circular form. For high performance, we need more frequency bandwidth. When the M level increase, the error rate will decrease with the same SNR values[3].

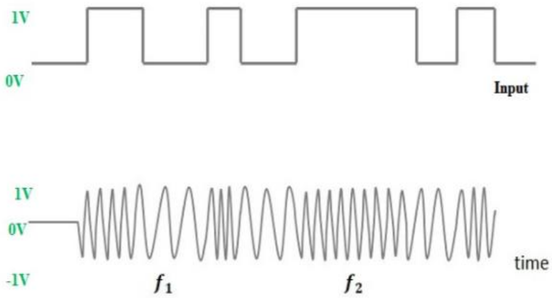

Figure 6: Frequency Shift Keying[4]

- ## PSK (Phase Shift Keying)

This modulation method is similar to FSK. However, the baseband signal just changes the phase (Qc) of the carrier signal. For different symbols, we can define different phase values. The main advantage of PSK is having more improved BER performance (3 dB). In PSK modulation, constellation points are 2-dimensional vectors (like FSK). There is no change in carrier amplitude so the PSK constellation has a circular diagram. For both FSK and PSK, if M levels increased within the same symbol energies, bit error rates also increase. In Figure 3, a PSK example is shown[5].

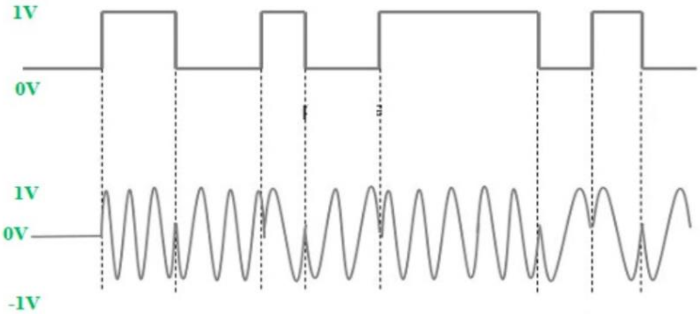

Figure 7: Phase Shift Keying[6]

- ## QAM (Quadrature Amplitude Modulation)

In this operation, there are two carriers that have phase shift (quadrature). Phase-shifting (90 degrees) prevents interferences even we sent two data at the same time without exceeding bandwidth. QAM method can be used for faster communication systems. In QAM, we can change both amplitudes of the carrier (like PAM) and the phase of the carrier (such as PSK). In other words, QAM is a mixed modulation type so its constellation diagram can be designed to increase performance at the same M level. QAM modulation is a more efficient method. However, the application of QAM is very difficult and also in demodulation, detection of the signal cannot be done with dual Correlators because of unstable constellation diagrams. For M = 4, the performance of both PSK and QAM is the same. RECTANGULAR FORM is using for the QAM constellation. This design, make the operation easier (than the other design) and provide more accurate demodulation. For rectangular design, QAM has a similar SNR graph with PSK. When the M level increase, the error probability will be an increase[7].

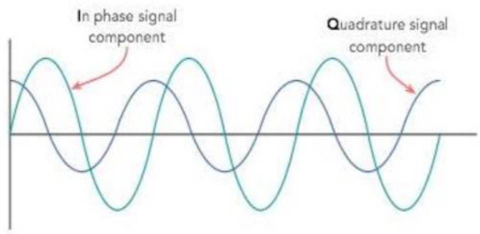

Figure 8: Quadrature Amplitude Modulation[8]

1.1.3. Pulse Shaping

Symbols created as a result of digital modulation were quantized as -3, -1, 1, and 3. these numbers are meaningless expressions in analog systems. To make them meaningful, we need to express them in certain forms of continuous signals. The critical point here is what the shape of the signal will be. Choosing the wrong signal forms may cause extra problems in the next steps.

Rectangular pulses give good results in high noise due to their flat shape. Rectangular pulses also give the best results in timing synchronization. They are very simple and easy to apply. However, since they cause extra ISI noise in the frequency domain, especially in multi-user spectrums, they create problems in the acquisition and detection of signals in the next steps[9].

Hamming pulse is the second choice (for now). It can be preferred in multi-user systems as it gives a limited bandwidth in the frequency spectrum. It is ineffective in high noise and timing should be done well while sampling in the receiver part. it is also possible to bring the signals closer together to obtain flat shapes (with under-sampling (fs should be chosen less than the Nyquist Rate)). To reduce its noise distortion, a proper Correlator can be used[10].

For more efficient pulse shaping, Raised Cosine also can be considered (We are thinking of choosing this method in the future). Raised-cosine is similar to Sinc, but with a slightly greater spectral width in exchange for narrower sidebands. Raised-cosine filters are simple to use and they are in wide use. They have configurable excess bandwidth, allowing transmission systems to select between a simplified filter and greater spectral performance. The other benefit of using this pulse shape is sideband effects (ISI Terms) can be reduced via some implementations[11].

1.1.4. Analog Upconversion

From the transmitter and pulse shaping blocks, the digital message has already been turned into an analog signal. In order to have an efficient transmission, the analog version the signal should also be shifted in frequency domain, to change the frequency for such aim is called modulation or up conversion. Further, once the signal reaches the receiver, the frequency should be shifted back down, this process is called demodulation or down conversion. Demodulation process includes one or two step processes. Analog demodulation process is done with one step, on the other hand an analog down conversion to the intermediate frequency and then the digital down conversion to baseband sums up as two steps. In the detailed block diagram below, two-step procedure is shown[12].

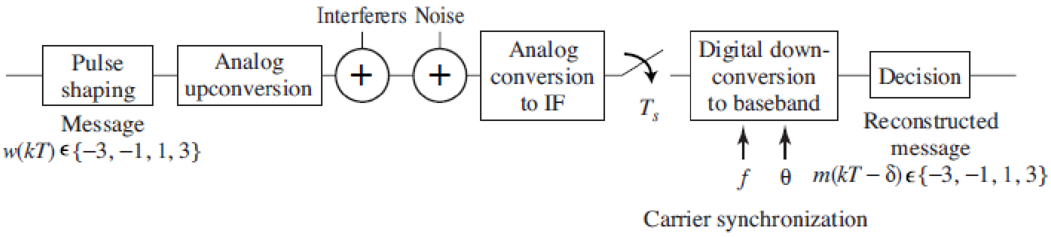

Figure 9: Block Diagram of Analog Upconversion[13]

In order to modulate a signal there are several ways, such as large-small carrier AM (amplitude modulation), single-sideband AM, vestigial sideband AM or quadrate modulation. In additional, it could be also employed by simple squaring and filtering. Demodulation also could be done by sampling.

- ## Amplitude Modulation with Large Carrier

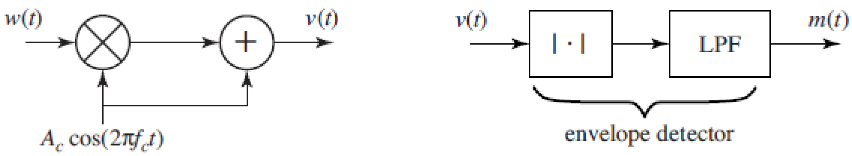

Figure 10: Block Diagram of Amplitude (De) Modulation with Large Carrier[14]

Amplitude modulation with large carrier process simply includes two steps: multiplication the message and the carrier signal; and then adding the product to the carrier signal.

$$v(t) = A_c w(\mathrm{t}) \cos(2\pi f_c t) + A_c \cos(2\pi f_c t) = A_c(w(t) + 1)\cos(2\pi f_c t) \ (1)$$

The process which multiplying the signal in time by a sinusoid is also called mixing. The Fourier transform of the amplitude modulation with large carrier is expressed as below.

$$V(f) = \mathcal{F}\{A_c(w(t) + 1)\cos(2\pi f_c t)\} = A_c \mathcal{F}\{(w(t) + 1)\} * \mathcal{F}\{\cos(2\pi f_c t\} \ (2)$$

As expected, the multiplication in time domain leads convolution in frequency domain[15].

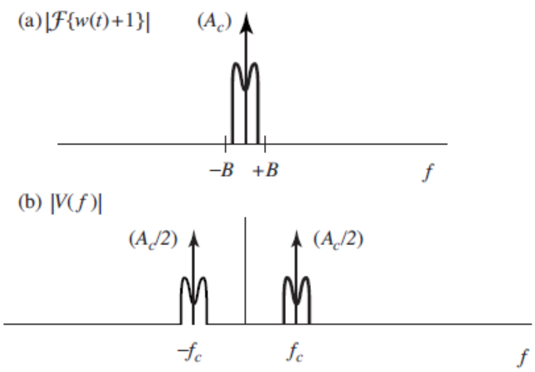

Figure 11: Frequency Representation of the Fourier Transform of the Signals
The vertical arrows in (b) represent the transform of the cosine carrier at frequency f_c. Also, the scaling by $A_c/2$ is indicated next to the arrowheads[16].

If $w(t) \geq -1$, the envelope of the message signal will be the same as *w(t)* and as demodulator, an envelope detector could be used. Filtering the absolute value of the signal with a low pass filter could help find the envelope of the signal.

Basic advantage of AM with large carrier is that there is no need to have an exact synchronization, in other words, the phase and the frequency should not be known one is reaches the receiver. Thus, the receiver circuitry becomes much simpler and cheaper without synchronization.

However, using AM with large carrier has also disadvantages. One of them is the more power needed to transmit as adding a carrier to the message signal, but not for the efficient information transmission[17].

- ## Amplitude Modulation with Suppressed Carrier

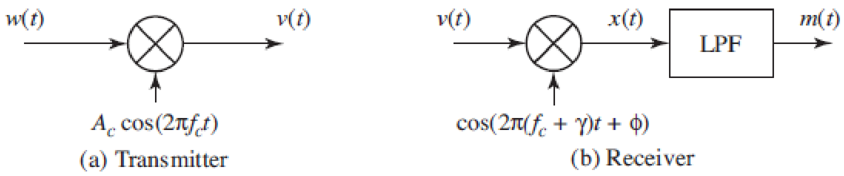

Figure 12: Block Diagram of Amplitude (De)Modulation with Suppressed Carrier[18]

Another way to use amplitude modulation is to choose not to add the carrier as shown in the figure above.

$$v(t) = A_c w(t)\cos(2\pi f_c t) \tag{3}$$

Additionally, direct application of the frequency-shift property of Fourier transform shows the frequency spectrum representation of the modulated signal.

$$V(f) = \frac{1}{2}A_c W(f + f_c) + \frac{1}{2}A_c W(f - f_c) \tag{4}$$

The issue about AM with suppressed carrier like the AM with large carrier, it has the twice of the original signal's bandwidth. For instance, if the message signal occupies between the frequencies $\pm B$ Hz, then the modulated signal will occupy between $f_c - B$ and$f_c + B$, so it will have a bandwidth of 2B.

While demodulation with suppressed carriers, mixing with a cosine with same frequency and phase as the modulating cosine, the message signal could be recovered by low pass filtering. However, the frequency and the phase of the cosine at the transmitter could never be known before exactly. The mixing signal is also known as local oscillator. In practice applications, accurate priori information is given for the carrier frequency. On the other hand, the phase could be anything since it depends on the distance between the transmitter and the receiver. Also, because of high frequencies, the wavelengths are small so that small motions could affect the phase significantly[19].

1.2. Channel

1.2.1. Modelling Corruption

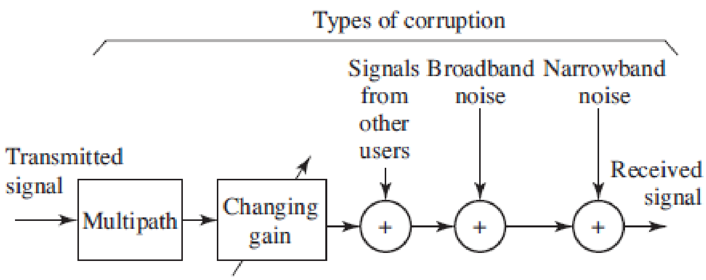

Figure 13: Block Diagram of Corruption Types[20]

While a signal travels from transmitter to the receiver, the path could be damaged because of some corruptions as shown above.

- ### Multipath Interference

When the radio communication between a spacecraft and some antenna on Earth is though, it is obvious that the signal from spacecraft will arrive more or less intact thanks to the vacuum of space. However the signals are mostly reflected diffracted, and when they arrived, so they are different from the initial sent signal.

It could be thought that the combination of scaled and delayed reflections of the message signal which occur while it travels the path, generate these distortions. For instance, between two transmission towers, the path could include reflections from atmosphere or nearby hills, and others that are scattered by multiple bounces of nearby buildings or etc[21].

- ### Broadband Noise

Once the signal arrives at the receiver, it should be amplified. It is possible to use high-gain amplifiers; however the noises and the inferences will also be amplified with the signal. Consequently, any noise in the amplifier itself will be increased as well. This noise is mostly modelled as “White (independent) broadband noise”[22].

- ### Narrowband Noise

Since all the noises do not have to be White, the spectrums should not also be flat. The signals with narrow spectrum such as stray sine waves, could also affect the receiver. Narrowband noises could be caused by errant transmitters or because they could be harmonics of a low frequency waves with nonlinear distortions. Generally, narrowband noises will automatically be attenuated by the bandpass filter, if they occur out of band[23].

- **Fading**

Another distortion could become because of the time-varying problems, which is called "fading." Fading simply means the change of the frequency response of the channel over time due to path changes[24].

1.3. Receiver

1.3.1. Band – Pass Filter (BPF)

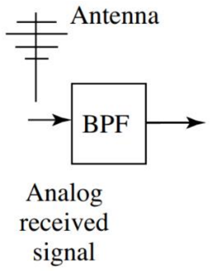

Figure 14: Band – Pass Filter on Receiver[25]

Band – pass filter allows to pass the certain frequencies inside of the specified bandwidth and attenuates the outside of the bandwidth.

In a communication system, many users are using the same channel at the same time. Frequency division multiplexing (FDM) allows using one common channel for transmission for a combined signal which contains the number of separate message signals from different users[26].

To be able to receive the message signal from the desired user, a band-pass filter should be used as in Figure 10. In the figure f_1, f_2, f_3 indicates the different carrier frequencies belong to different users, f is the frequency of the desired signal. There must be a gap between f_1, f_2, and f_3 more than 2f to prevent interference between them. When the second user is wanted to select, band - pass filter can remove the other users' frequency components (f_1 and f_3) which are outside of the band – pass filter bandwidth[27].

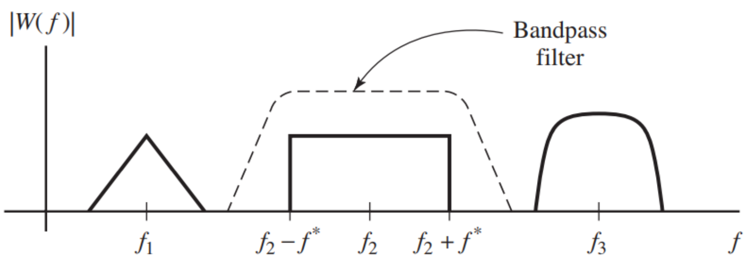

Figure 15: Selection of Desired User Frequency[28]

Besides, the BPF is used for to prevent the broadband and narrowband noise by improving the signal-to-noise ratio (SNR). After band – pass filtering, noise is decreased, but the signal power is same, so the SNR is increased.

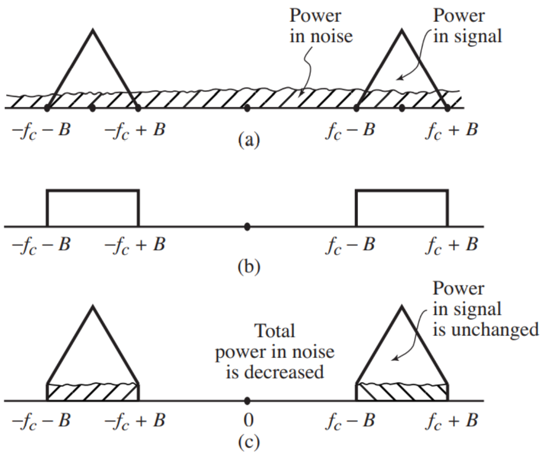

Figure 16: Removing Broadband Noise[29]

However, the narrowband noise cannot be removed by only band-pass filtering. After, band-pass filtering there should be a notch filter.

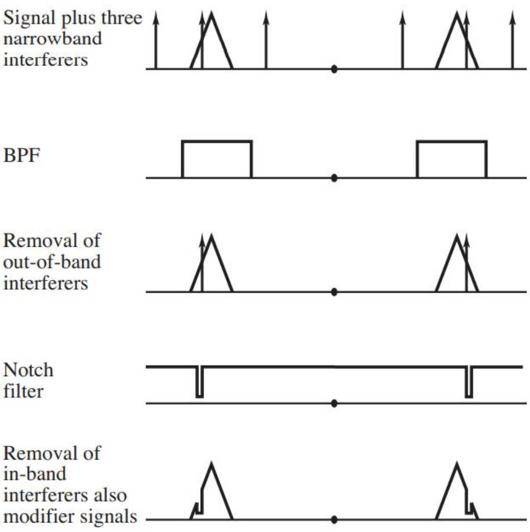

Figure 17: Removing Narrowband Noise[30]

1.3.2. Analog Conversion to Intermediate Frequency (IF) & Sampling

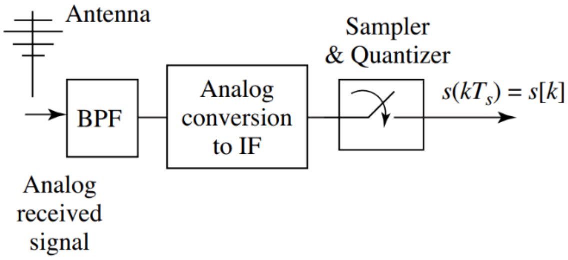

Figure 18: Analog Conversion to IF and Sampling[31]

Sampling and quantization are the backbones of digital communication systems. Since the information signal which comes from the transmitter is an analog signal. Thus, the transmitted signal ought to be converted to a digital signal by sampling. During the conversion to prevent any information loss, the Nyquist theorem should be taken into consideration. Nyquist theorem specifies that the sampling frequency has to be at least twice the maximum frequency component of the converted signal. This implies that the sampling rate should be fast enough to make the proper conversion.

On the transmitter side; during the modulation (analog up-conversion), a large carrier frequency component f_c is added to the message signal. Therefore, on the receiver side sampling rate should be very fast to implement the reliable conversion. However, a faster sampling rate may increase the cost of the sampler. Furthermore, all the frequency bands require different components like amplifiers, modulators, filters, etc. This means all of the components should be redesigned depending on the frequency band of the communication system. Taking into account all of these reasons, the received signal should be down-converted to a desired intermediate frequency (IF). In this way, the signal which is down-converted to an intermediate frequency (IF) can be sampled at the Nyquist rate of IF instead of the large frequency of transmission. In addition, components may become usable for frequency bandwidths after they are down-converted to IF[32].

1.3.3. Automatic Gain Control (AGC)

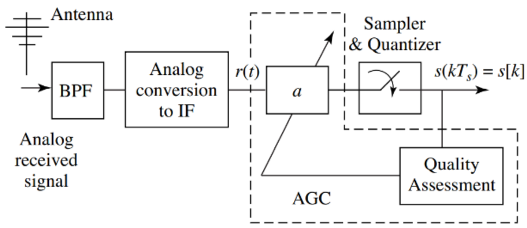

Figure 19: Automatic Gain Control[33]

The samplers have a certain amplitude range and quantization range (resolution). If the received signal has a very large amplitude which is out of the amplitude range of the sampler or the signal is very low from the quantization range; the sampler cannot sense the changes on the received signal. This implies information loss.

Automatic gain control endeavours to keep the received signal into the specified sampler range with quality assessment. In the quality assessment, the power of the received signal is compared with the predetermined signal power. Then, gain coefficient ‘a’ is increased or decreased based on this comparison. The ‘steepest descent’ algorithm is able to be used for the implementation of AGC, it updates the adaptive parameter by finding the minimum of the function which is wanted to be converged by going through the negative direction the the derivative[34],[35].

The automatic gain control also is used to handle the time-varying fading.

There are three main iterative algorithms to implement the automatic gain control as follows.

- **Heuristic Algorithm**

The heuristic algorithm based on the comparison between the square of the first arrival sample of the received signal ‘s’ and the square of predetermined ‘S’ value and it is implemented on the DSP side of the communication system. For the sample of the received signal r(t) is s[k] as follows (5)

$$s[k] = a*r(t) \tag{5}$$

‘r(t)’ cannot be known by the DSP side, because it is an analog signal. However, it can be defined as in (1) and used for the comparison. The algorithm starts with determining a positive ‘a’ value. After that, the comparison between ‘s^2’ and ‘S^2’ is done. Depending on the comparison ‘a’ is incremented or decremented[36].

$$a[k+1] = a[k] + \mu\{\mathrm{sign}(a[k])\}(S^2 - s^2[k]) \tag{6}$$

‘μ’ is the step-size. While it is increased the algorithm converges fast but it is selected very large; oscillations around the convergence may also increase. Besides, for very small values of ‘μ’ converges become very slow even if the oscillations are decreased. Thus, the sampler sensitivity has to be taken into account during the selection of the ‘μ’. If the sampler range is large enough, a bigger value of ‘μ’ would be chosen, but the range is too small it may not be possible to run a very fast algorithm for the sampler[37].

In this algorithm s^2 is expected to converge to S^2, however, if as in equation (1); ‘a’ and r(t) can be different from each other to make ‘s’ value converge. Thus, this algorithm behaves like a low – pass filter on the ‘a’ value.

- **Gradient Descent Algorithm**

Gradient descent algorithm brings (7) a more general perspective to automatic gain control. It is based on a cost function ‘J’ which is improved by taking an average on $S^2 - s^2$. Depending on the selection of the proper cost function, the algorithm converges the heuristic algorithm[38].

$$a[k+1] = a[k] - \mu\frac{\partial J\,(a)}{\partial a} \tag{7}$$

For the Least Square cost function J_{LS}

$$J_{LS} = \frac{1}{4} * \mathrm{avg}\{(s^2[k] - S^2)^2\} \tag{8}$$

Iterative algorithm becomes,

$$a[k+1] = a[k] - \mu\mathrm{avg}\{(s^2[k] - S^2)\frac{s^2[k]}{a[k]} \tag{9}$$

In this algorithm power of the ‘s’ is expected to converge to S^2.

For the Naive cost function J_N

$$J_N = \text{avg}\{|a|\left(\left(\frac{s^2[k]}{3}\right) - S^2\right)\} \quad (10)$$

Iterative algorithm becomes,

$$a[k+1] = a[k] - \mu\text{avg}\{\text{sign}(a)\ (s^2[k] - S^2) \quad (11)$$

In this algorithm a^2r^2 is expected to converge to S^2.

1.3.4. Carrier Synchronization

To have correct demodulation, the receiver must be able to accurately find both the phase and the frequency of the signal from the transmitter. This process is called carrier recovery.

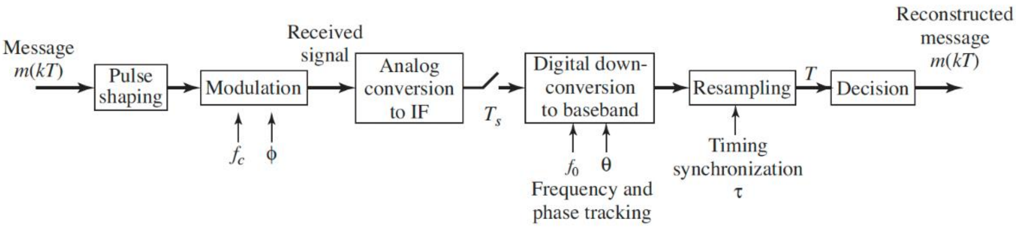

Figure 20: Diagram of a Communications System Emphasizing the Need for Synchronization[39]

The figure above includes one analog one digital down-conversion. The main problem of analog down-conversion is cost. Precise analog devices increase the cost of the receiver. The phase and frequency information of the received signal can be accessed using digital down-conversion FFT (Fast Fourier Transform). However, this method is also not useful because it requires high processing power.[40] The ideal solution is to use an adaptive element that predicts the optimal value rather than using FFT. Several methods are as follows;

- **Squared-Difference Method**

Squared difference loop is one of the adaptive methods used without finding the carrier phase. In this method, it is first estimated by assuming that the carrier frequency is known. The squaring generates the frequency components at twice as carrier frequency so it also increases the variance of phase error.[41]

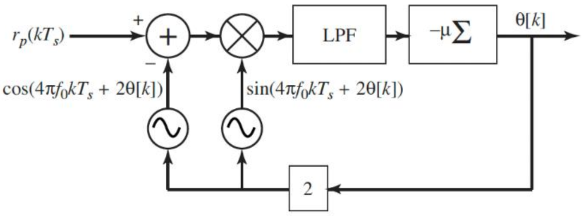

Figure 21: Model of the squared difference method[42]

To ensure that a lowpass filter set up. The output signal of the filter summed again and this process is feed itself. The output of sum is the phase of the carrier frequency.

- ## Phase-Locked Loop

PLL is one of the methods used for phase estimation consisting of multiplier, loop filter and voltage-controlled oscillator. The PLL circuit is as in the figure.

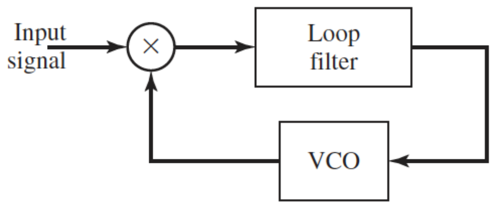

Figure 22: Model of the PLL[43]

Suppose that the input signal is cos ($4\pi f_c t + 2\varphi$) and the output of the VCO is sin ($4\pi f_c t + 2\hat{\varphi}$) where $\hat{\varphi}$ represents the estimate of φ.
The product of these two signals is transmitted to the loop filter. The loop filter is actually a low pass filter eliminates the high frequency components. Transfer function of the filter is

$$G(s) = \frac{1 + \tau 2s}{1 + \tau 1s} \quad (12)$$

Time constants $\tau 1$ and $\tau 2$ controls the bandwidth of the filter.
The output of the filter enables the voltage control. VCO is a simple signal generator. Phase estimation is made at the output of the VCO.

$$2\varphi = K_v \int_{-\infty}^{\tau} v(\tau)\, d\tau \quad (13)$$

where K_v is the gain constant.

- ## Costas Loop

In the costas loop, the input signal is multiplied by cos ($2\pi f_c t + \varphi$) and cos ($2\pi f_c t + \varphi$). This is also equal to the output of the VCO. Then the generated signal is filtered by the low pass filter. The error signal is generated by multiplying these filter outputs.The error signal is filtered by the loop filter whose output is the control voltage that manages VCO[44].

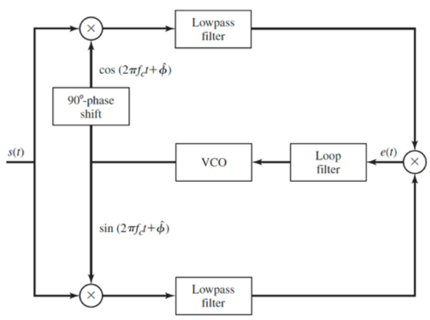

Figure 23: Model of the Costas Loop[45]

- **Decision Directed Method**

Decision directed method is modified version of costas loop. The algorithm behind the DD method is taking difference between the soft decisions and the hard decisions to drive the updates of the adaptive element[46].

1.3.5. Pulse Shaping and Filtering

In order to transmit digital messages, they must be converted into appropriate analog pulses. This conversion can be done with pulse shaping filters. After this conversion, the receiver filter recaptures the digital values from the pulses.

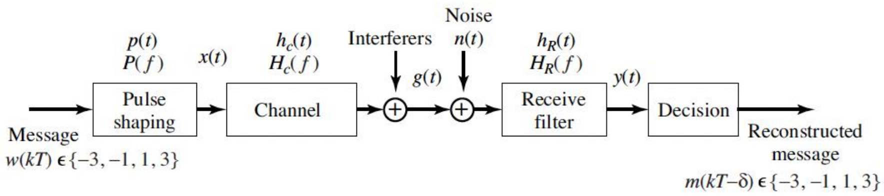

Figure 24: System diagram of a baseband communication system emphasizing the pulse shaping at the transmitter and the corresponding receive filtering at the receiver [47].

Communication systems are exposed to many external factors at every step. The matched filter is designed to reduce some of these external factors. Actually, the main goal of this part is to maximize the SNR (Signal to noise ratio).

1.3.6. Timing Synchronization

There may be time shifts in the signal reaching the receiver (these shifts are called tau). This results in sampling at wrong times and incorrect decision (or decoding) operations (even if the inter-symbol period is known exactly). Eye diagrams are a reference for correct decode operations. The situations where they are at the maximum opening are needed (closed eye situations occur in wrong timing operations). Without the use of adaptive components, The error between received and transmitted can be found only if the received data is known or the header on the receiver can be measured.

The alternative method is Cluster Variance Algorithm. This method takes the differences between the received data values and the nearest element of the source alphabet. The formula can be described below.

$$\text{avg}\{(\mathrm{Q}(x[k]) - x[k])^2\} \tag{14}$$

The second alternative way is measuring the power of the T-spaced output of the matched filter. Maximizing this power (by proper choice of tau) also leads to a good answer.

Error Surface Graph is used in the explanation of performance functions. The Error Surface graph of both Cluster Variance Minimization Method or Output Power Maximization can be seen below.

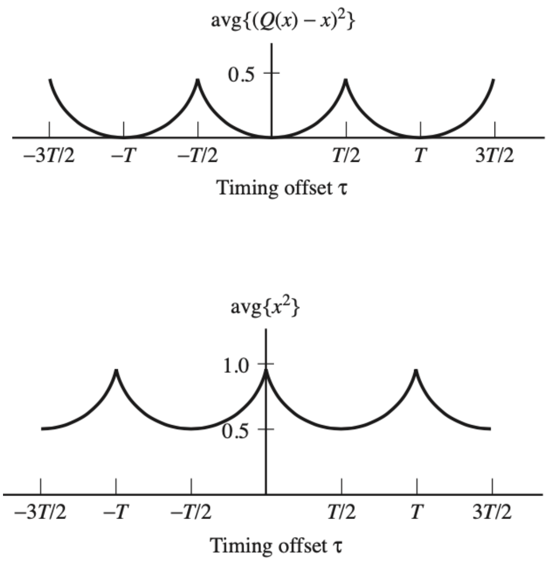

Figure 25: Performance Graph Representation[48]

In many cases, the error surface for the Cluster Variance has minima wherever the error surface for the output power has maxima.

In some cases, it is difficult to achieve optimization for both methods. These cases are noisy channel, multilevel source alphabet, chosen pulse shape, and ISI terms. To achieve optimization, it is necessary to change the design of adaptive elements[49].

1.3.7. Equalizer

- **Linear Equalization**

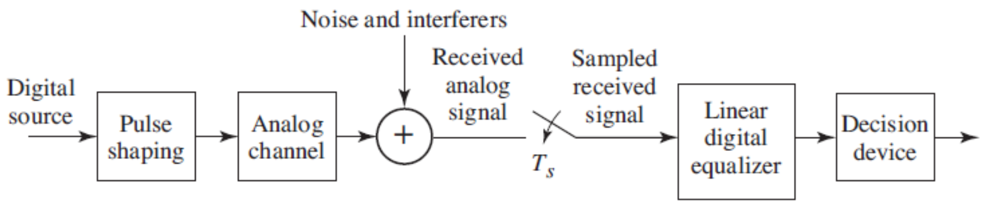

Figure 26: Block Diagram of the Linear Equalization[50]

In ideal cases, in the receiver, there is no interaction between successive symbols; each signal arrives independently from each other. However, when symbols interact or they corrupt each other, the received signal becomes distorted. In such cases, it is hard to decipher the desired message signal. This distortion generally called as “intersymbol interference” or ISI.

If there is no ISI from multipath or imperfect timing, etc., the impulse response of the system from the source to the recovered message has a single nonzero term or “spike”. The amplitude of this spike changes with respect to the transmission losses and the delay by the transmission time[51].

If there is no ISI caused by multipath channel, the spike is scattered or duplicated once for each path in the channel. Accordingly, the number of nonzero terms in the impulse response will be increase. Then the channel could be include a finite-impulse-response, linear filter C, and the delay will be equal to the total time interval which includes the reflections with significant energy arrival.

Hence, equalizers are built to model another filter in the receiver to counteract the effect of the channel. The main goal of the equalizers is to unscatter the impulse response so that the impulse response of combination of the channel and the equalizer has a single spike. Additionally, the equalizers are used to reject additive white noises with designing appropriate linear notch filters[52].

1.3.8. Decision

To decode the received signal, a decision mechanism finds the symbols which are contained in the received signal. This mechanism is based on finding the symbol in the received codeword either in the source alphabet or not. If the symbol is not in the source alphabet, it is quantized as a source alphabet element that has a minimum distance from the received symbol.

The symbol which is demodulated from the received signal is called 'soft decisions', and the alphabet element which has a minimum distance with the demodulated signal is called as 'hard decision'. The performance of the decision block depends on several measurements like 'symbol recovery error' which indicates the difference between transmitted massage and soft decision. Also, the quantized symbol may be different from the transmitted message and it is called as 'hard decision error'. Furthermore, the related error may occur between the soft decision and hard decision and it is also called as 'decision-directed error'[53].

1.3.9. Decoding

Decoding is simply the process of reversing the transactions made and outputting them in the format entered into the system. Decoding can basically be reduced to two steps. The structured redundancies of the signal reaching the receiver must be removed. This step may be called undo channel encoding. Reconstructing the message signal from the remaining signal after the redundancies are deleted can also be attempted to undo the source coding.

The method used to encode the message signal for this project will also be used to decode.

2. Methods and Proposed Solutions

2.1. Overview

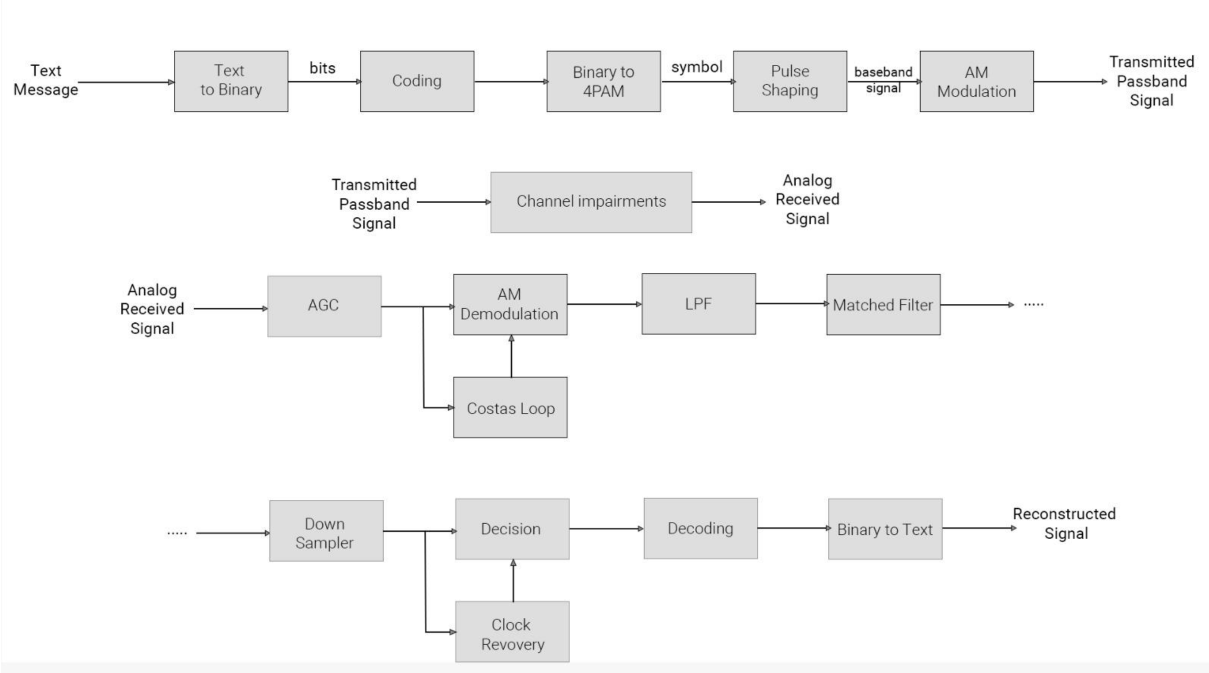

Figure 27: Block Diagram of the proposed solution

2.2. Transmitter

2.2.1. File Selection

It was thought to send an audio file as a message to be transmitted. 'gong.wav' was selected as the reference file. Sending an audio file is more complicated than sending a text file. Because, we need to obtain a suitable digital signal from the analog signal we have. As it is known from previous information, the analog to digital conversion process includes sampling, quantization (linear) and encoding operations. Sampling is provided by Matlab. For quantization, 8 bits (265 levels) were used to minimize the error rate. The quantization error and quantization step we found in our experiments are given below.

```
errquan =              quantization =

  2.2221e-06             0.0052
```

Fig 28: Quantıization error and step size

Our quantized signal is very close to our original signal. below is the signal quantized and ready to digitize

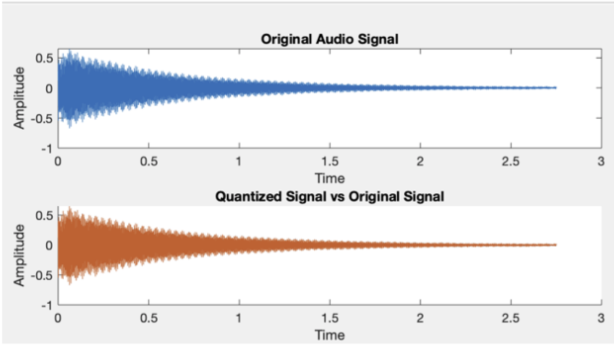

Fig 29: Quantized audio signal

As a final step, the levels of the audio signal need to be expressed in bits. Levels obtained in decimal can be converted to corresponding bit equivalents in ASCII code. Each level is expressed as 8 bits. The first step in the translation process is to express the signs of the levels as bits in order to express them more easily. As seen in the example below, the first two bits contain the level sign and the last eight bits contain the quantization level information.

```
sample_of_signal =

     1     0     0     0     0     1     0     1     1     1
```

Fig 30: Sample of sıgnal

Afterwards, it was almost impossible for us to get results from the tests we made with the sound signal. Since the code we are working on is not optimized, even a small audio file was found to be heavy. in this case the only thing left to do was to return the text file. The ready-made code (“text2bin”) we have was used to switch from text file to binary values. This code takes the file in decimal and converts it to 7-bit data according to ASCII code. The text file to be sent and the required function are given below

```
str = 'A0Oh well whatever Nevermindl ';

message=text2bin(str);
```

Fig 31: Text to bınary operatıon

2.2.2. Linear Block Coding

The first method used to protect our PAM system against noise was channel coding. There are 2 methods to be used as channel coding, these are Convolution Coding and Linear Block Coding. The word linearity indicates that addition and multiplication (as a matrix) are done linearly. This allows the Block coding event to be implemented more easily. Linear block Coding (5,2) means that when we apply a 2-bit input, a 5-bit output will be obtained. The 3-bit part added to the end is called the parity bit, it contributes to both a reduction in the error rate and an easier frame synchronization. The Encoding part of Linear Block Coding is on the transmitter and the Decoding part is on the receiver. In the encoding part, the parity bit is added using the Generator Matrix. The Generator matrix and coded data are given in figure 3.

In the receiving part, the Hadamard Matrix is used to eliminate parity bits and detect possible errors. If the product of the incoming signal and the Hadamard matrix is zero, the data can be detected without any problems. In case of one or more errors, the Syndrome Matrix containing all possibilities is checked. Error correction is made with the Hamming distance between the available data and the possibilities. It should be noted that the Linear Block Code method, which can easily detect one and zero errors, is useless in multiple errors. Figure 4 shows the Hadamard Matrix and the Syndrome Matrix.

We gave our original signal to the Linear Block Coda that we created to test it and got it back. As can be seen from the result, Linear Block Coding works properly.

```
data = 'A0Oh well whatever Nevermindl ';

message= text2bin(data);

encode = blockcode52_encode(message);

decode = blockcode52_decode(encode);

rec_data = bin2text(decode)
```

Command Window

```
>> deneme2

m =

   212


rec_data =

    'A0Oh well whatever Nevermindl '
```

Figure 32: Test Results

2.2.3. Modulation Method

When the Spectral Bit Rate graph is checked, it is understood that the FSK modulation technique is more effective in power-limited situations. Choosing FSK does not work for us now, as our primary goal is not to prevent energy consumption. QAM, PSK, and PAM are methods used in band-limited situations. We will use 4-level modulation so QAM and PSK represent the same performance. However their applications (especially on demodulation) a bit complicated. Finally, the method we decided on is 4-level PAM modulation. Another reason for preferring the PAM system is to try to see the effect of the changes in the blocks more clearly. As it is known, PAM systems are the most vulnerable system in case of noise. In this case, the effect and possible consequences of the modified blocks can be observed more easily.

$$
\begin{aligned}
01 &\rightarrow +3 \\
11 &\rightarrow +1 \\
10 &\rightarrow -1 \\
00 &\rightarrow -3
\end{aligned}
$$

Figure 33 : Binary to PAM

As we mentioned before, our concern is sending a message from the analog channel (Air). However, in the encoding section, we obtain some digital symbols. These symbols must be converted from digital form to analog form. This process is done in the pulse shape part.

2.2.4. Pulse Shape

We decided to use Raised Cosine pulses while making the signals analog. By using Raised Cosine and changing its roll-off factor, it is possible to create more flatten (such Rectangular pulse) structures. This process provides less bandwidth and less noise effect.As a result of our experiments, it was seen that the most appropriate roll-off factor level was 0.3. In the figure, the raised cosine function and its frequency domain equivalent, which are necessary to make our signal analog, are seen.

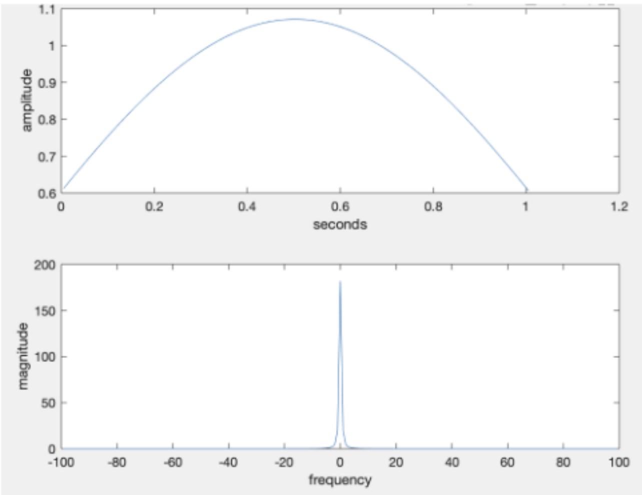

Figure 34: Raised Cosine Shape

Figure 35 shows the original signal oversampling with the raised cosine function.

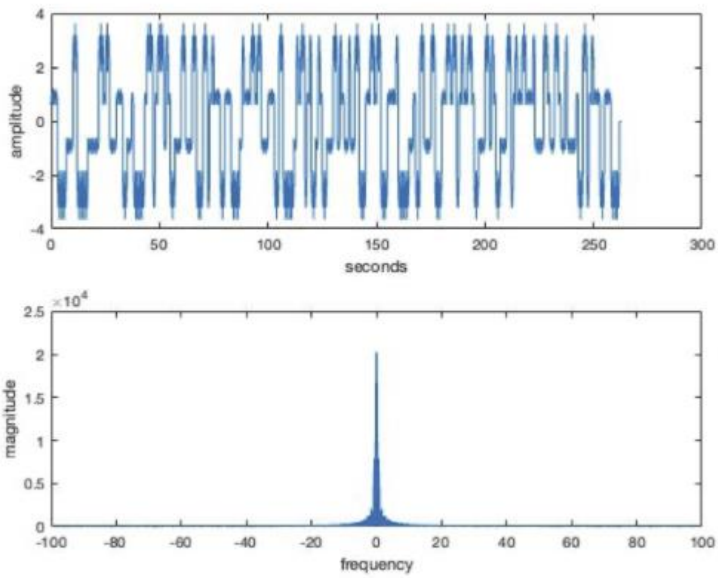

Figure 35: Oversampled signal

2.2.5 Analog Upconversion

It has been decided to use “amplitude modulation with a large carrier” for analog upconversion. Besides the other methods, AM with a large carrier method is cheaper and simpler to build . Also, it could be used for general types of data system, and has good performance at high S/N.

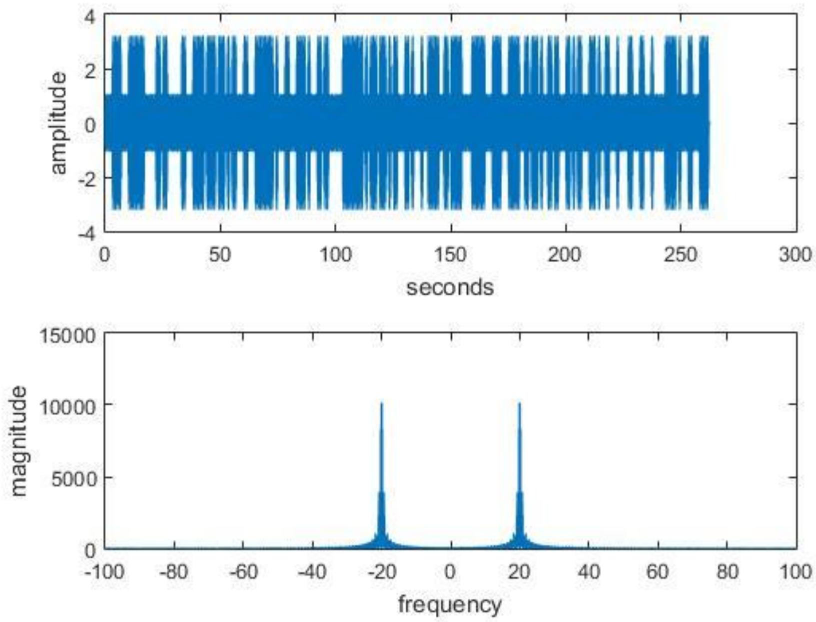

Figure 36: Amplitude and Magnitude Response of the Modulated Signal

During the amplitude modulation carrier frequency is selected as 20 Hz, the magnitude response of the signal proves the modulation is done properly.

2.2. Receiver

2.3.1. Noise, Other FDM Users – ISI

Unfortunately, such time-varying distortions or noises could not be corrected with a single filter. The filter should compensate differently with respect to time. "The ideal application of filters" could vary due to the system components.

In the project, the channel noise and the phase offset are added in order to make a more realistic scenario.

2.3.2. Automatic Gain Control Algorithm

To be able to implement the automatic gain control, the heuristic algoritm and gradient descent algorithm is planning to get together. Since for the gradient descent algorithm 'a' should not change over the range of averaging 'N' (eq (9) and eq (11)). Thus, the the new algorithm is to generated by narrow down the averaging range N as in (15).

$$a[k+1] = a[k] - \mu \mathrm{sign}(a[k]) \left(s^2[k] - S^2\right) \quad (15)$$

This algorithm is more simpler and more easy to implement and converges to the desired signal power. The averaging does not very effective on the iterative algorithm because the 'steepest descent algorithm' behaves like a low pass filter and the averaging operation in a certain window is also behaves like a low-pass filter. Thus, the avareging does not effect the performance of the iterative algorithm, so the averaging operation is completely removed[50].

In order to examine effect of the automatic gain control algorithms in (6) and (15) on software defined radio, different cases taken into account. In other words the channel impairments like noise, fading and phase off-set added to the system and behaviours of the two different algorithms has been observed based on these impairments.

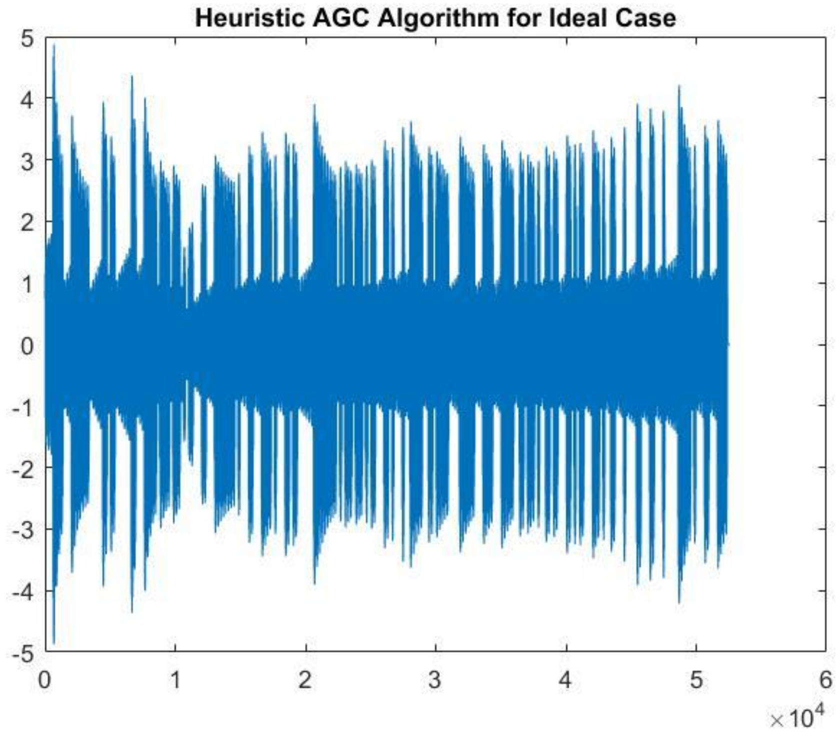

Figure 37: Heuristic Algorithm for Ideal Case

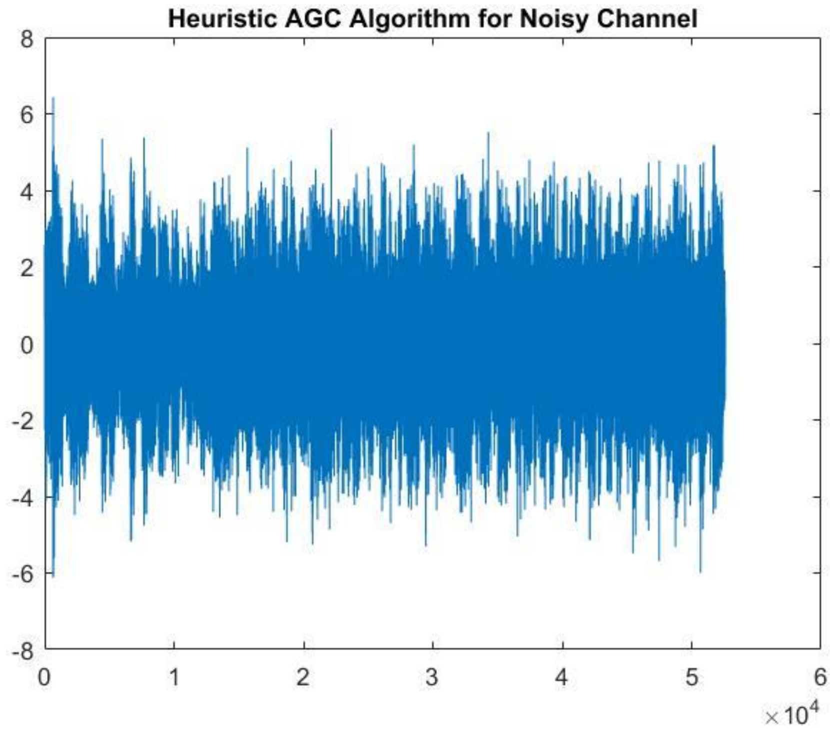

Figure 38: Heuristic Algorithm for Noisy Channel

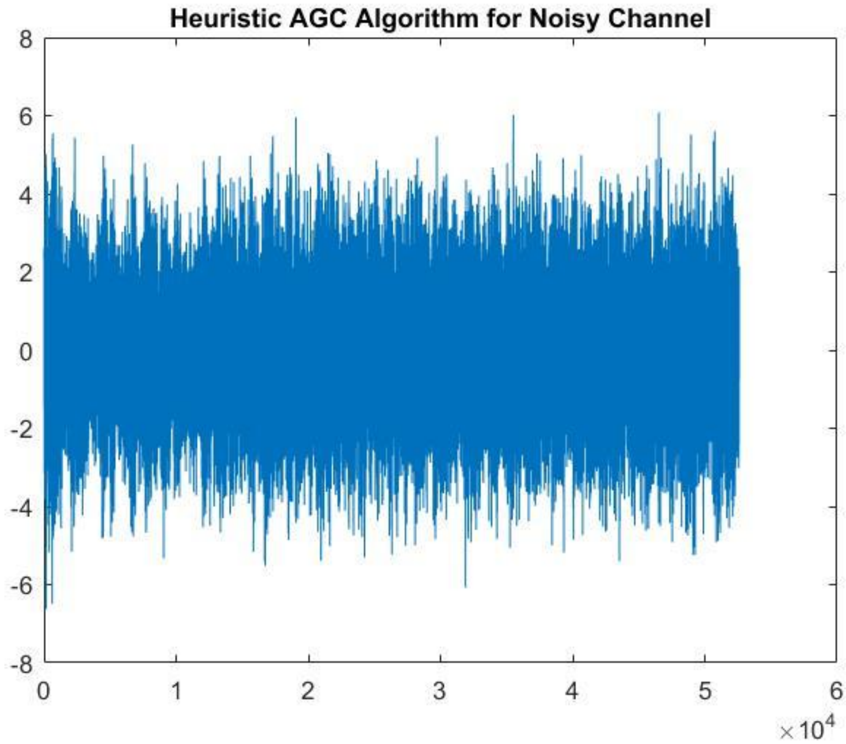

Figure 39: Heuristic Algorithm for Noisy Channel with High Noise Characteristic

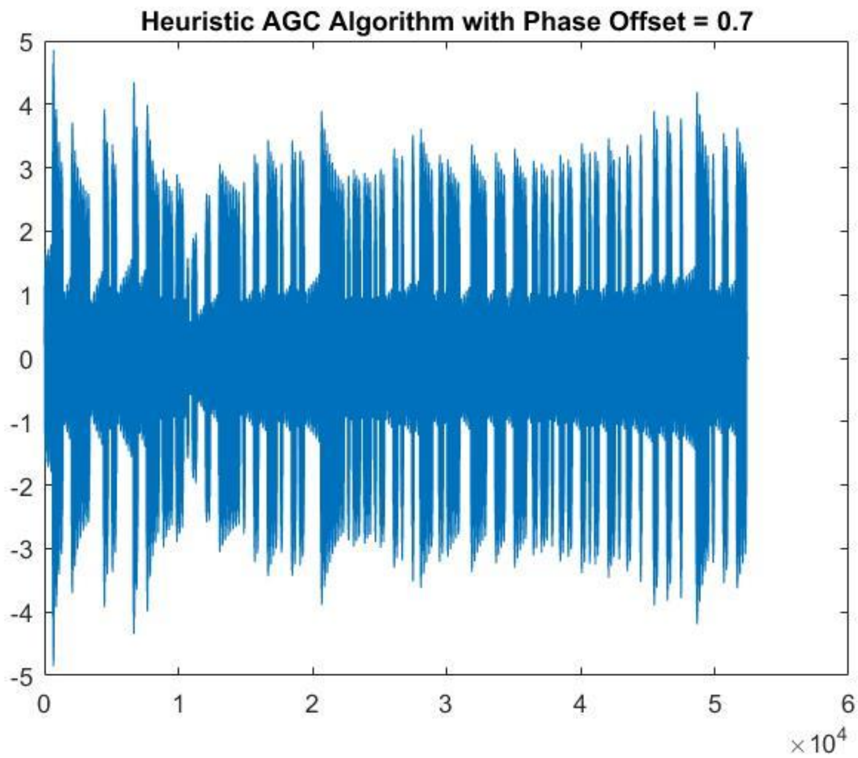

Figure 40: Heuristic Algorithm for Phase Offset

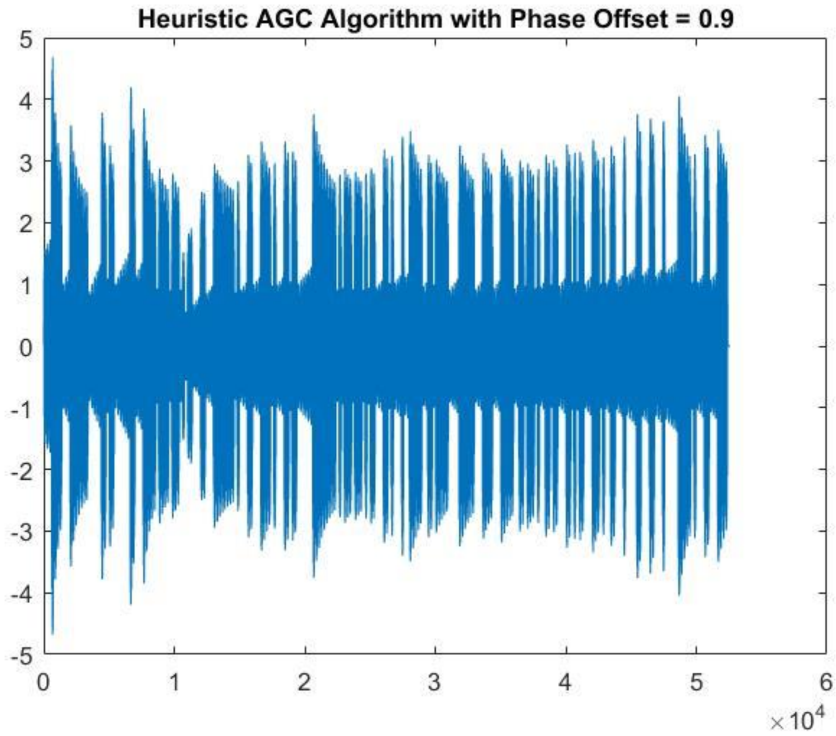

Figure 41: Heuristic Algorithm for Phase Offset

For the heuristic algorithm, as seen in Figure 25, if there is not any channel impairments AGC trying to keep the gain in a certain range successfully. Besides, the effect of the noise can easily observed in Figure 26 and Figure 27. Even though the noise is increased the AGC algorithm still tries to keep the gain. As for the phase offset, there is no change in algorithm behaviour as expected.

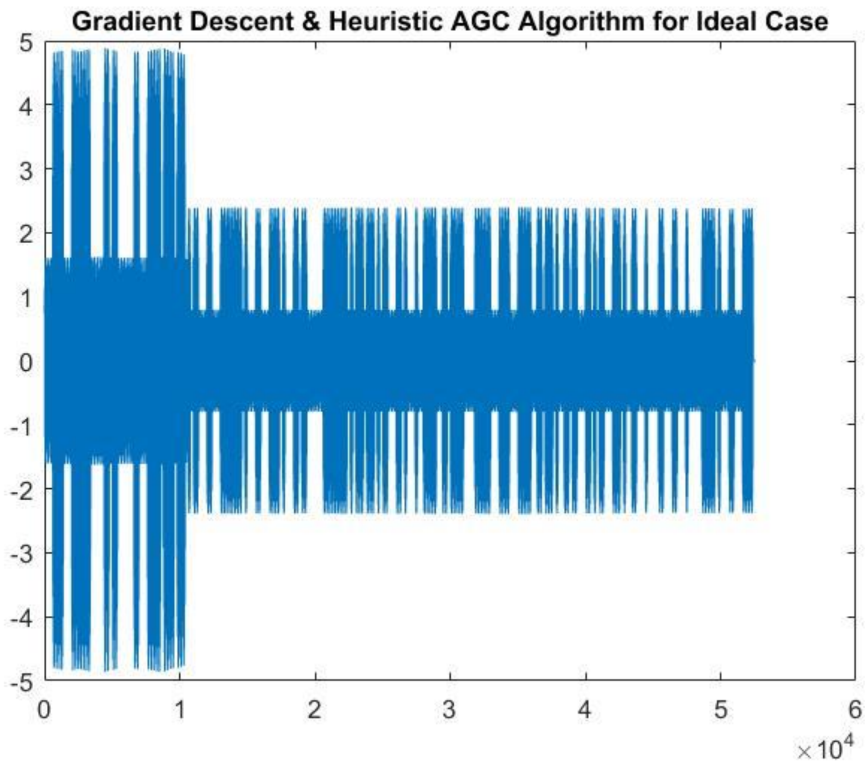

Figure 42: Heuristic & Gradient Descent Algorithm for Ideal Case

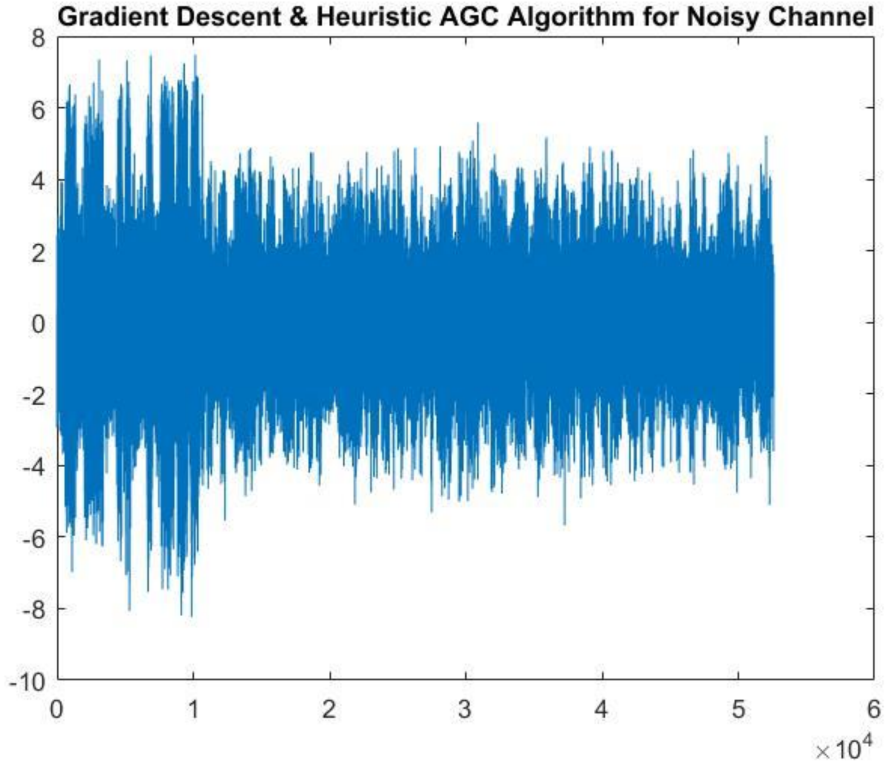

Figure 43: Heuristic & Gradient Descent Algorithm for Noisy Channel

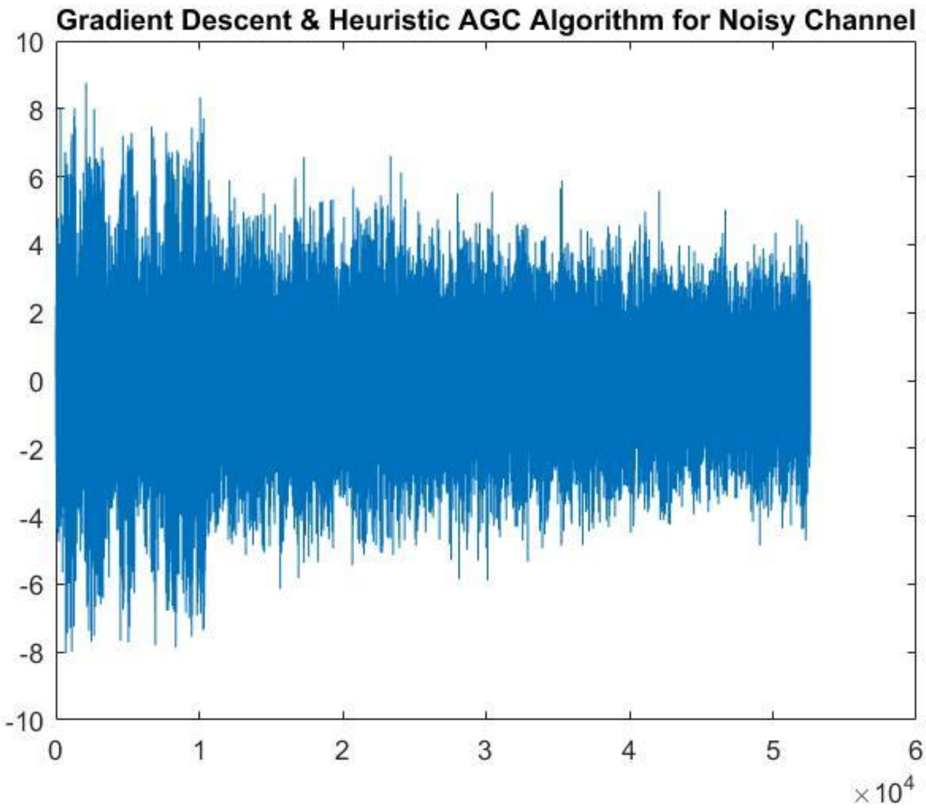

Figure 44: Heuristic & Gradient Descent Algorithm for Noisy Channel with High Noisy Characteristic

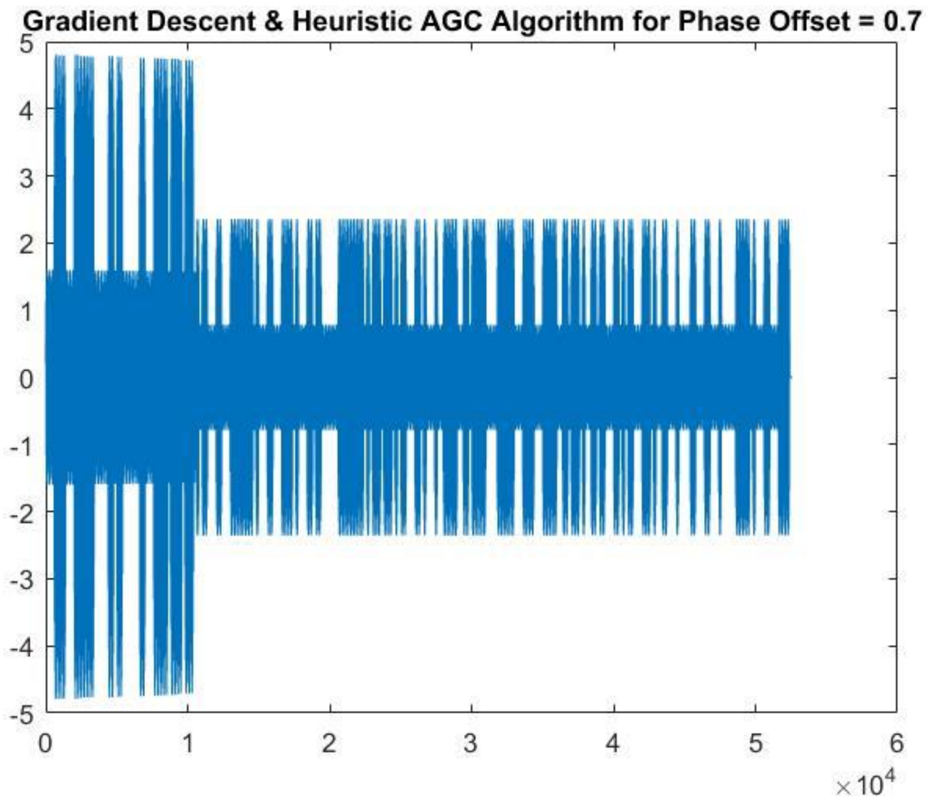

Figure 45: Heuristic & Gradient Descent Algorithm for Phase Off-set

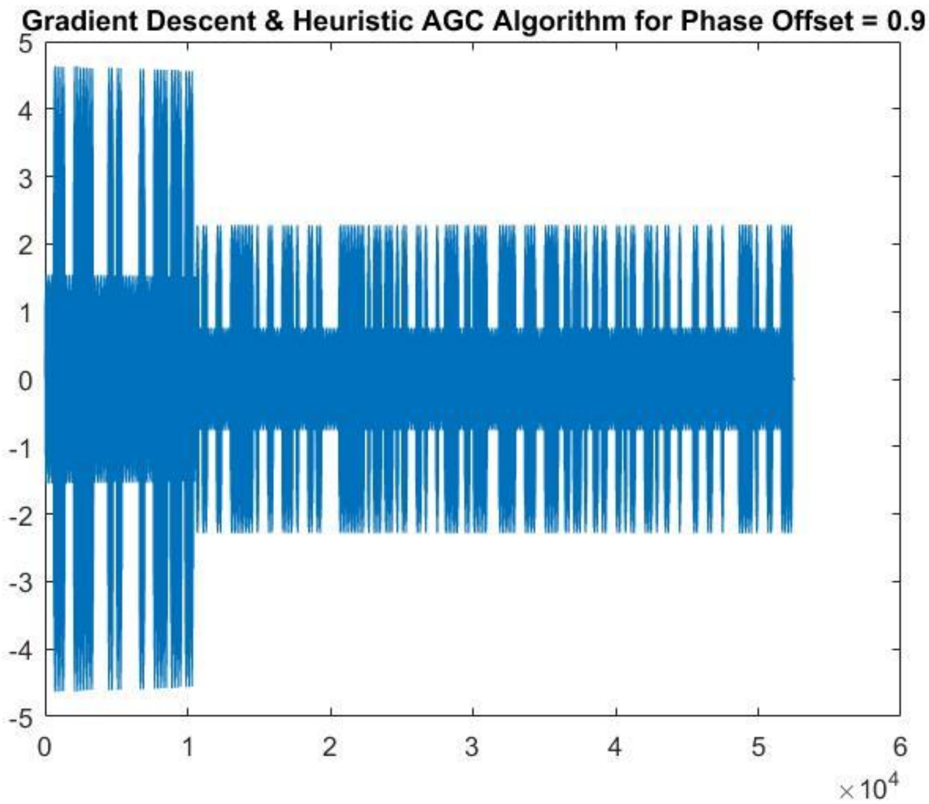

Figure 46: Heuristic & Gradient Descent Algorithm for Phase Off-set

For the combination of heuristic and gradient descent algorithm, similar kind of behavior has been observed with heuristic algoritm. It also tries to keep the gain in a certain range. However, the behaviour of the second algorithm is shaper than the first one. Even though there is an unstable state at the beginning of the algorithm.

The consequences of behaviours of these algorithms can be observed more clearly on text outputs. It is assumed the receiver sends the message 'A0Oh well whatever Nevermindl', and the message received by the receiver. Different cases are examined as in figure 47;

1. Heuristic Algorithm in Ideal Case
`ytext =`
`A0Oh wmll whatever Nevermindl`
	2. Heuristic & Gradient Descent Algorithm in Ideal Case
	`ytext =`
	`A0Oh well whatever Nevermindl`

3. Heuristic Algorithm with Noise
`ytext =`
`A0Oh wmll whatever Nevermindl`
	4. Heuristic & Gradient Descent Algorithm with Noise
	`ytext =`
	`A0Oh well whatever Nevermindl`

5. Heuristic Algorithm with Noise
`ytext =`
`A0Oh wm*])s(Smm6tj,jTn-6\i*d\)`
	6. Heuristic & Gradient Descent Algorithm with Noise
	`ytext =`
	`A0Oh well whatever Nevermindl`

7. Heuristic Algorithm with Phase offset = 0.7
`ytext =`
`A0Oh wmll whatever Nevermindl`
	8. Heuristic & Gradient Descent Algorithm with Phase offset = 0.7
	`ytext =`
	`A0Oh well whatever Nevermindl`

9. Output with Fading without AGC
`ytext =`
`A0Oh0xRU3^Lw-□Rk+□S□+□Ri3□U;3^`
	10. Output with Fading with AGC
	`ytext =`
	`A0Oh!7%ll whatever Nevermindl`

Figure 47: Received Text Messages for Different Scenarios

As seen in figure 47, the second algorithm which is in equation (15), has a more noise immunity. Receiver, can get the message properly even though the noise is very high. Even though there is any noise in the channel, message can be received with an error if the heuristic algorithm in equation (6) is implemented; it can be also seen easily from the figure 47. Moreover, the phase offset can occur in a channel but unfortunately AGC cannot handle the phase offset, different algorithms are used for recovering the phase offset as discussed in background information part.

In addition, the AGC may handle the channel fading. If there is not an AGC in the communication system; received message is meaningfull and corrupted. Fortunately, receiver can get more meaningful and proper message signal with AGC. There is still a distortion on the message but it is very likely to transmitted signal.

To considering the all of the behaviours and received text outputs. The AGC algorithm in (15) is more efficient and useful for the designed communication system. Thus, this algorithm has been used for the furter part of the project.

2.3.3. Carrier Synchronization – Costas Loop

The costas loop does not require signal preprocessing like SD or PLL. The incoming signal is used directly. The Costas loop works by reversing the sequence of operations. It converts it to DC first, applies a low pass filter and finally squaring it. It was decided to use the costas loop in this project because it can be directly processed and has simple steps.

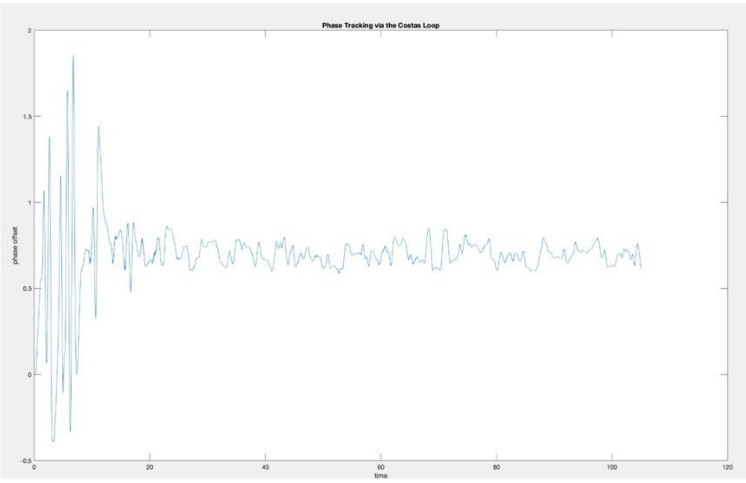

Figure 48 : Phase Tracking via the Costas Loop

Even if the costas loop algorithm behaves nonstable at certain points, it is successful in following the phase value. nonstable state is caused by noise.

2.3.4. Timing Synchronization Cluster Variance

It is important to make timing synchronization for the signal at the right time in order to get the correct information. Frequency offset can make the system sample at the wrong time.

For timing recovery, the Cluster Variance Algorithm is used.As it is understood from the figure below, since the algorithm is iterative, it is ensured that the error is low and the resulting constellations are distinct.

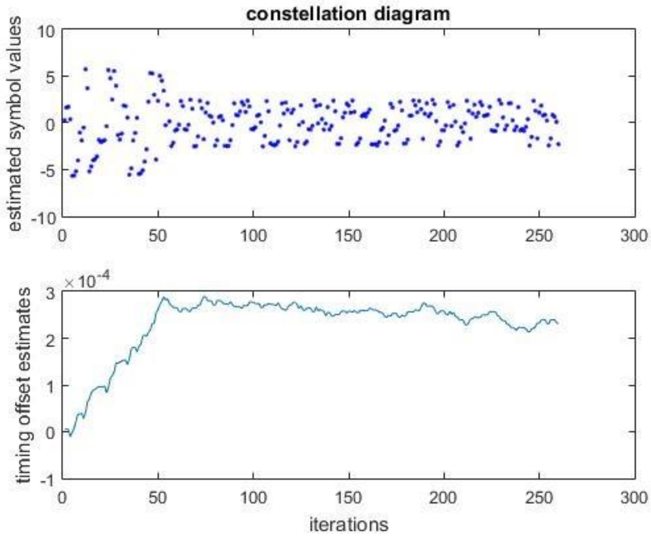

Figure 49: Constellation and Iteration

2.3.5. Equalizer – Least Squares

During this part of the project, it is decided to use "Least-Squares Equalization," since there will be a training sequence.

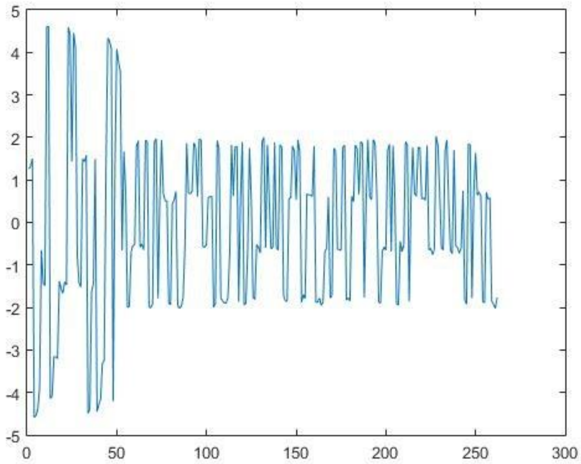
Figure 50: Equalized signal

After 10000th element, the signal will range between [-3,3]. When it is considered to the "mpam" graph, it is obviously seen that least square equalizer algorithm is suitable for the project.

After 50 samples, the signal will range between [-3,3]. When it is considered to the "mpam" graph, it is obviously seen that least square equalizer algorithm is suitable for the project.

2.3.6. Decision

The text massage which wanted be transmitted will be modulated with PAM. Thus on the receiver side; the decision part should recover the transmitted symbols to sent them to decoding part.

2.3.7. Decoding

The method used to encode the message signal for this project will also be used to decode. On the transmitter side (encoder part), Linear Block Coding was used. Necessary operations were also carried out on the receiver part and the original signal was tried to be obtained. Since we tried to see the effect of Linear Block Coding, both block coded (left) and unused block code (right) system results were shared.

```
ytext =

    'A0Oh well whatever Nevermindl '


percentage_symbol_errors =

    1.9084
```

```
ytext =

    'A0Oh weml uhate6Er"Jm&m"m)ldL'


percentage_symbol_errors =

   16.3462
```

Figure 51: The results

According to the results, Linear Block Code gives more reliable results. This proves our initial arguments.

References

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