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PERFORMANCE EVALUATION AND SIMULATION OF THE PHYSICAL LAYER OF AN EMBEDDED COMMUNICATIONS SYSTEM IN LABVIEW

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STUDENT DECLARATION

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Abstract

An important area of computer technology, particularly in the analysis of telecommunication networks and systems are modeling of complex radio systems and processes to evaluate and identify their performance and parameters. With the content of computer models being developed a new device can be much faster and more cost effective to identify weaknesses and pursue correction of system than to create a real prototype. Computer simulation tools allow to carry out experimental studies of real devices in a virtual environment that leads to the evaluation of possible improvements of the proposed devices prior to their practical implementation. Building a computer model is accomplished using specialized software packages, called a virtual environment modeling. The most powerful tools include virtual simulation software such as Matlab/Simulink LabVIEW [1]. In fact, we can quite easily measure and analyze the most important characteristics of wireless communication systems using a virtual environment, therefore its application for present thesis project is beneficial. Modeling those systems needs an integrated approach including modeling the transmission channels, digital signal processing techniques, transforming signals and performance evaluation.

Thus, I choose to work with LabVIEW to simulate the physical layer of an embedded wireless communication system, namely BPSK.

Összefoglaló

A számítógépes alkalmazások fontos területe, különösen az adatátviteli hálózatok és rendszerek analízisében az összetett rádiós rendszerek és folyamatok modellezése hogy meghatározzuk jellemző paramétereiket és kiértékeljük teljesítőképességeiket. A számítógépes modellek segítségével az új rendszerek hiányosságainak felismerése sokkal gyorsabbá és hatékonyabbá válik, mintha valódi prototípust került volna kifejlesztésre. A számítógépes szimulációs eszközök lehetővé teszik valós eszközök virtuális környezetben történő vizsgálatát anélkül, hogy azokat ténylegesen meg kellene valósítani. Egy számítógépes modell megalkotása egy specializált szoftvereszközzel történik, amelyben a virtuális környezet létrehozása lehetséges. Ilyen rendkívül hatékony szoftveres eszközök a Matlab/Simulink és a LabVIEW [1]. A szimulátorban a vezeték nélküli adatátviteli rendszer legfontosabb tulajdonságainak analízise és mérése egyszerűen megtehető, ezért jelen diplomamunka szempontjából ennek használata előnyös. Az ilyen rendszerek modellezése összetett megközelítést követel meg, amelynek elemei az adatátviteli csatorna modellezése, jelfeldolgozási megoldások, jeltranszformációk és teljesítőképesség kiértékelése.

Ennélfogva a BPSK beágyazott vezeték nélküli adatátviteli rendszer fizikai rétegének szimulációjához a LabVIEW-t választottam.

1 INTRODUCTION TO DIGITAL COMMUNICATION

This chapter discusses the digital communication system and its basics as it is the backbone of this thesis. In this chapter, we will deal first of all with the modules of digital communication system and we will describe their functionalities and their effects on the data. Furthermore, this chapter will enumerate the modulation schemes and their specifications. This general introduction will lead us to conclude on the need of this work and its objectives.

1.1 Digital communication

Nowadays, the communication occurs in the nature of signal. Almost all the messages commonly generated by the sources are analog signals by nature as the sound signals, electrical signals, mechanical and human speech. Analog communication over long distance suffers from interference, losses, distortion and security breach. Subsequently, the digital communication has evolved to overcome these problems as it is more immune toward external threats. In fact, digital communication consists on transferring sequence of bits physically over Point-To-Point or Point-To-Multipoint communication channels. In digital communication, the information is converted to an analog signal by modulation and sent as a sequence of bits. As a matter of fact, in baseband it is harder to separate noise from an analog signal than it is for digital signal [2]. As seen in Figure 1, we can notice that for the analog communication system the noise become part of the transmitted signal and the original information is lost hence the receiver cannot recover the noisy signal. In the other hand, for the transmitted analog signal shown in the right part of Figure 1. The high voltage and the low voltage still can be discerned and the receiver can decide if it is a bit 1 or bit 0. Therefore, the digitization of signals makes the communication more accurate and reliable.

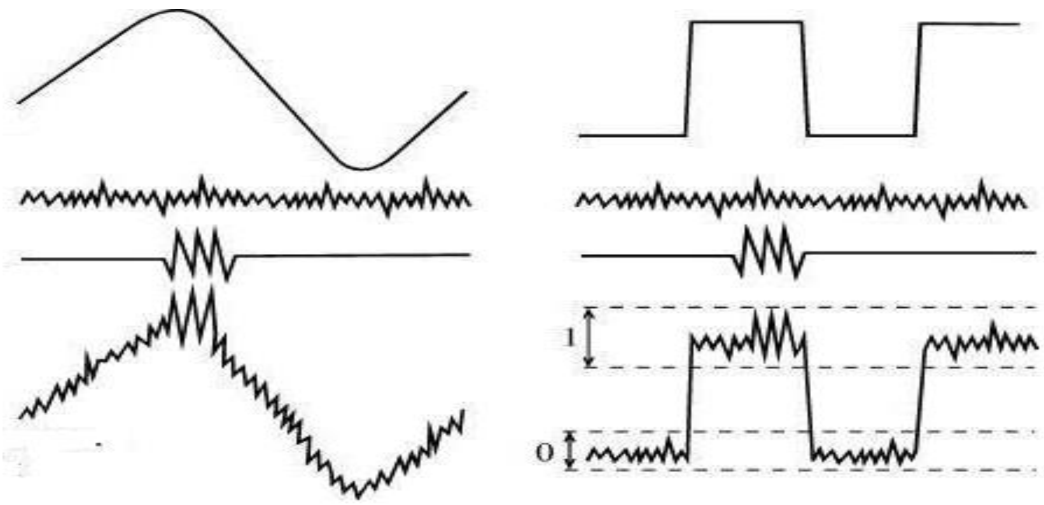

Figure 1. Transmitted and received signal in analog and digital communication

1.2 The components of digital communication system

The digital communication system, in its simplest form, involves a transmitter, a receiver and a communication channel. The transmitter is the interface between the information source and the channel. It translates the information symbols at its input into signals that are adequate for the channel and which can be physically transmitted. However, the transmitted signal is arisen by the noise due to the channel imperfections no matter what physical media the channel consists of. Hence, the receiver is responsible of recovering the distorted received signal to reconstruct the original message. In baseband, the transmitter is approached to a digital to analog converter and the receiver to an analog to digital converter as shown in Figure 2.

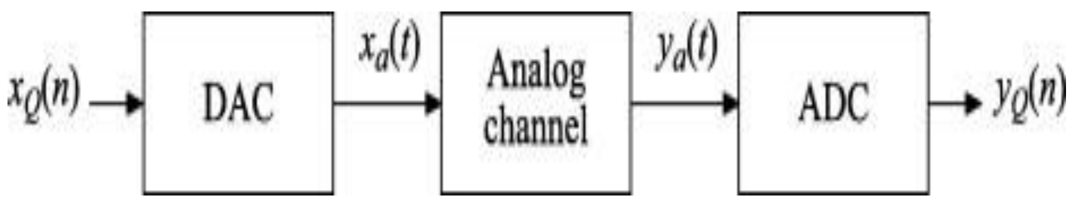

Figure 2. Simplified model of digital communication system [2]

$\chi_Q(n)$ Is the original digital information of the n^{th} bit.

$\chi_a(t)$ Is the analog information sent through channel.

$\Upsilon_a(t)$ Is the analog information that come to the receiver.

$\gamma_Q(n)$ Is the generated signal on the receiver side.

In more details, the Figure 3 shows the different modules of a typical digital communication system.

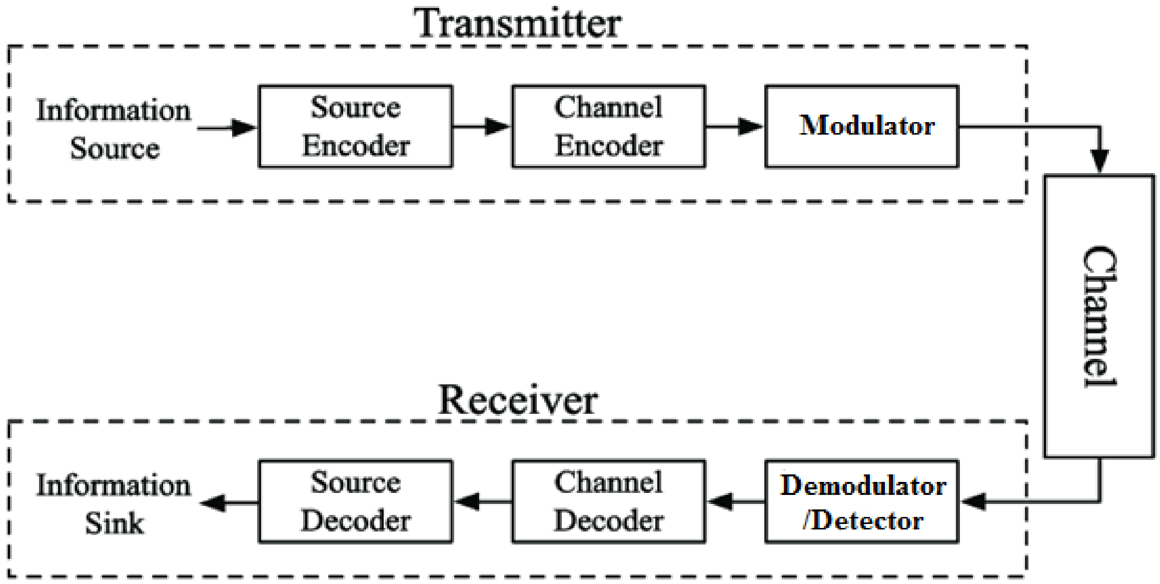

Figure 3. Block diagram of digital communication system

Source encoder: The source encoder is responsible for compressing the data to represent it with less redundancy and to remove the unnecessary bits [3].

Channel encoder: the channel encoder role is to introduce some redundancy to the compressed data. As the signal can be distorted during transmission, some unnecessary bits are added to the binary information sequence to prevent that and to increase reliability of the received signal. Thus, the channel encoder helps the receiver to decode the received signal later [3].

Modulator: the modulator plays first the role of interface that passes the channel encoded data from the encoder to the communication channel. Besides, the modulator serves also to encode the transmitted sequence using the suitable digital modulation techniques. This process serves to immune the transmitted signal toward noises, loss, interference and other distortions on the communication channel [3]. The modulator is also responsible for frequency translation to transform the baseband signal into the RF region in which the signals can be radiated.

Channel: The channel is the physical medium for the transmission of signal between the transmitter and the receiver. Noise and interferences in the communication signal can cause distortion of the transmitted. There is several types of communication channel. It may be air, wire, or optical cable [3]. In our case the channel is supposed to be air and characterized as an Additive White Gaussian Noise (AWGN) channel.

Detector/demodulator: The data retrieving process started by the detector. The received modulated signal incorporate some noise introduced by the channel. To remove noise filtering generally applied and then the received RF signal is transformed back into the baseband frequency region. Thus this is where the received signal is demodulated to obtain a sequence of channel encoded data in digital format [3].

Channel decoder: The channel decoder is responsible of the error corrections after the sequence detection. Using the redundant data inserted in the transmitter, the channel encoder try to recover the original signal by adding some additional bits [3].

Source decoder: the source decoder recover unaltered bit stream without information loss [3].

1.3 Modulation schemes

Modulation is the process of mapping the signal into a waveform, called a carrier signal by varying one or more parameters of it in accordance with the information bearing bit stream. The carrier signal is a non-data signal with high frequency that permits to transmit information over long distance. In communication system, there are three main modulation techniques: amplitude shift keying (ASK), frequency shift keying (FSK), and phase shift keying (PSK).

1.3.1 Amplitude shift Keying

Amplitude shift keying is the digital version of amplitude modulation where only the amplitude of carrier signal is modified among all the parameters. In ASK, the input signal having logic 1 is represented by transmitting a fixed-amplitude carrier wave and fixed frequency for the bit duration [4]. . The simplest version and most common of ASK is on–off keying (OOK). In OOK, for a binary input 1 a carrier wave are transmitted and for a binary input 0 nothing is transmitted. Other types of ASK use different (non-zero) amplitudes to represent 1 and 0.

Figure 4 (a) shows a binary bit sequence. Figure 4 1.2(b) illustrate the unmodulated carrier and Figure 4(c) and (d) illustrate the modulated waveforms using two versions of ASK. Figure 4 (c) uses OOK while (d) uses binary ASK version.

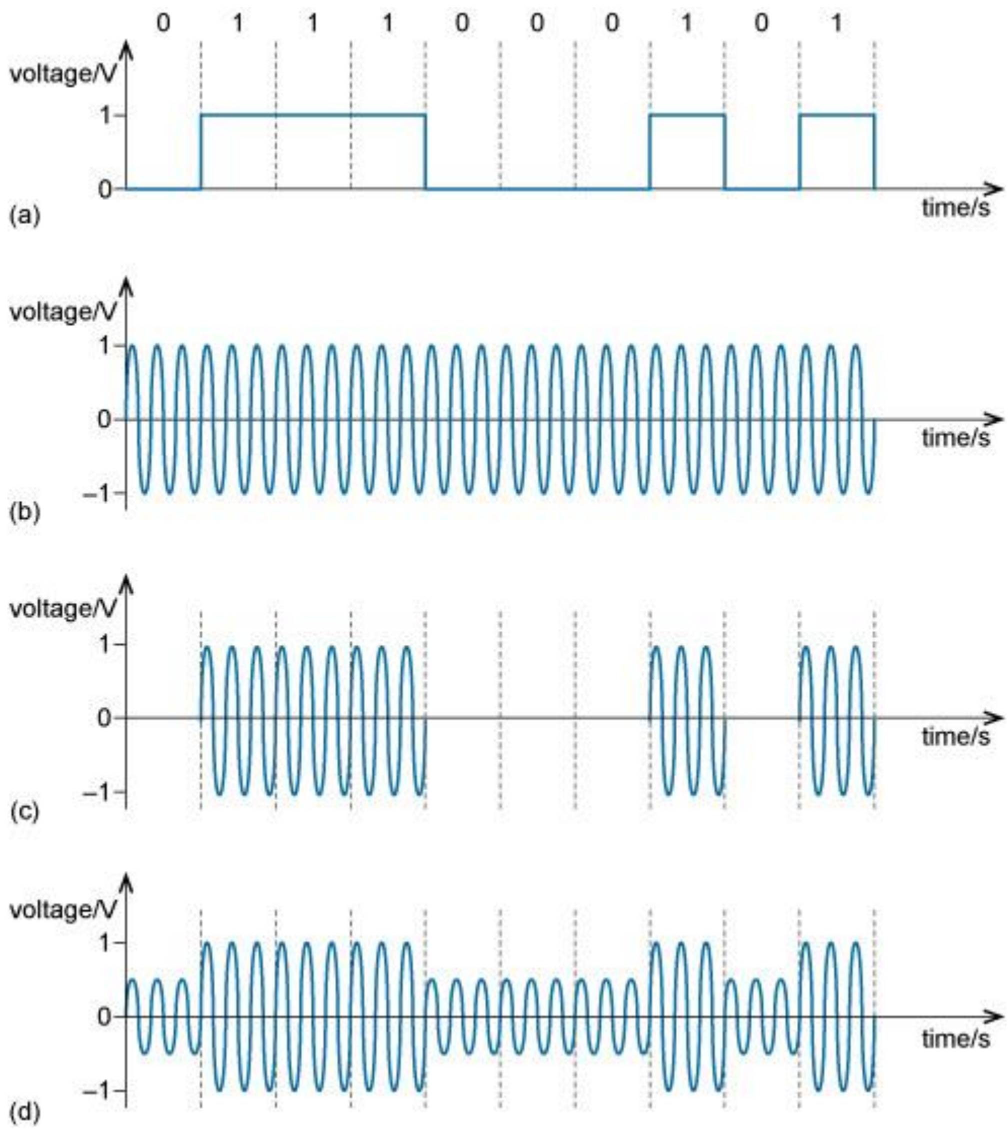

Figure 4. Modulated signal using OOK and binary ASK [4]

The ASK modulator, as we can see in Figure 5, contains a carrier signal generator, a unipolar binary sequence generator and the band-limited filter. The unipolar input can be either High or Low. In case of OOK if the input binary signal is logic High for certain time interval, then there is a waveform and if the input binary signal is Low no signal appears. On the other hand the logic of ASK modulator is to change the voltage of the input signal accordingly to the carrier signal. As a final step, the band-limiting filter shapes the waveform depending on the requirements of the communication system [5].

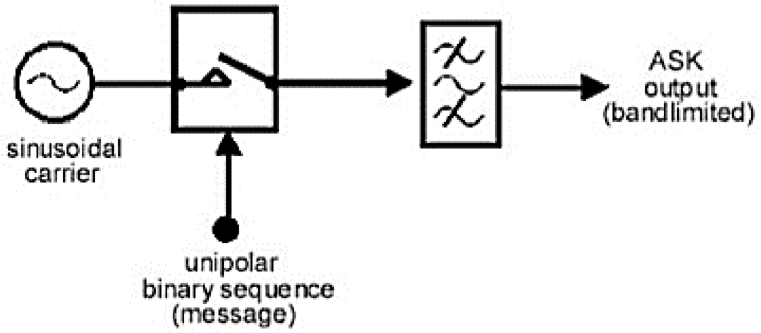

Figure 5. Block diagram of ASK modulator

The ASK demodulator can be asynchronous or synchronous. If the clock frequency at the transmitter and the clock frequency at the receiver matches each other, it is a synchronous demodulation, as the frequency gets synchronized. Otherwise, it is asynchronous.

The Figure 6 shows the block diagram of a synchronous ASK demodulator. In the synchronous demodulator, the signal is passed to the half-wave rectifier, which delivers a positive half output. The low pass filter then suppresses the higher frequencies. In the end, a comparator and a voltage limiter help to have a clean digital output.

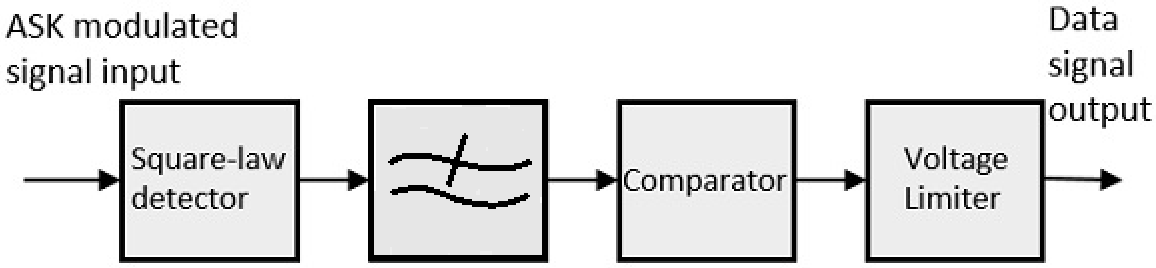

Figure 6. Block diagram of synchronous ASK demodulator

The Figure 7 shows the block diagram of an asynchronous ASK demodulator. In the asynchronous modulation, the demodulation process can be completed by using a square law device whose output voltage is proportional to the square of the amplitude modulated input voltage [5].

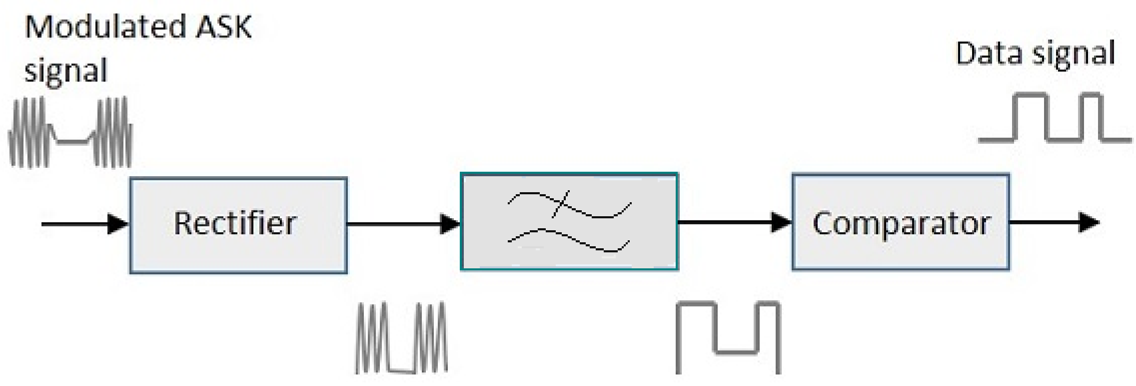

Figure 7. Block diagram of the asynchronous ASK demodulator

1.3.2 Frequency shift Keying

This type of modulation is commonly used for broadcasting music and speech, magnetic tape recording systems, two way radio systems and video transmission systems. When noise occurs naturally in radio systems, frequency modulation with sufficient bandwidth provides an advantage in cancelling the noise. In frequency-shift keying, there are two different carrier signals with two different frequencies: “mark frequency” that represent the bit 1 and “space frequency” that represent bit 0 [6]. These are the two carrier signals equations:

$$s_0(t) = \mathrm{A}_c \cos(2\omega_0 + \theta)t$$

$$s_1(t) = \mathrm{A}_c \cos(2\omega_1 - \theta)t$$

In this type of modulation, as seen in Figure 8 the frequency of the carrier signal varies in accordance with the message signal, and other parameters like amplitude and phase remain constant.

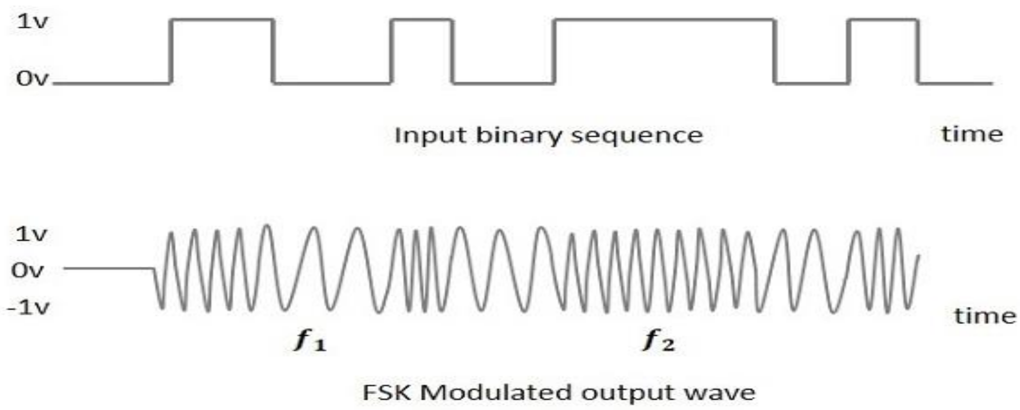

Figure 8. Modulated signal using FSK [6]

The FSK modulator block diagram includes two oscillators with a clock as seen in Figure 9. FSK modulator block diagram and the input binary sequence. Each oscillators generating its own signal in function of its frequency as it shown in the equations. The first oscillator is matched with $s_0(t)$ and the second is matched with $s_1(t)$. Hence, the FSK modulated signal is generated in function of according to the frequency of the oscillator.

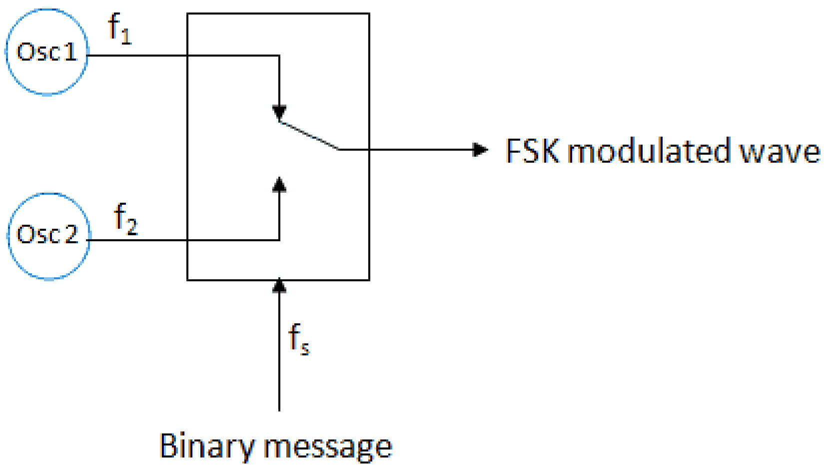

Figure 9. FSK modulator block diagram [6]

In the receiver side, the FSK modulated and transmitted wave must be demodulated. The demodulation in FSK can be done in two manners: Coherent FSK detection and Non-Coherent detection. The main difference between the two methods is the phase of the carrier signal. In the coherent detection case, the carrier signal of the transmitter and the receiver are in phase. Thus, the coherent detection is also known as the synchronous detection. As for the incoherent detection, the carrier signal of the transmitter and the receiver are not in the same phase. This method also is named asynchronous detection [6].

The FSK synchronous demodulator block diagram is shown in Figure 10. In the synchronous FSK detector, the carrier phase of the receiver and transmitter should match. However, the source of carrier phase often mismatches at the receiver due either to a propagation delay or to the generation of the carrier signal which is not phased locked to the transmitted carrier. It is therefore it is necessary to implement a circuit such as PLL (Phase Locked Loop) at the receiver for carrier phase estimation. Hence, the input signals are handed, first of all, to two PLLs where the output frequency of the oscillator is constantly adjusted until it matches with the input signal frequency. Then the signals are passed to a band pass filters that is assigned to cut off frequency which is equal to the binary input frequency. A decision circuit decide finally which output is more likely [6].

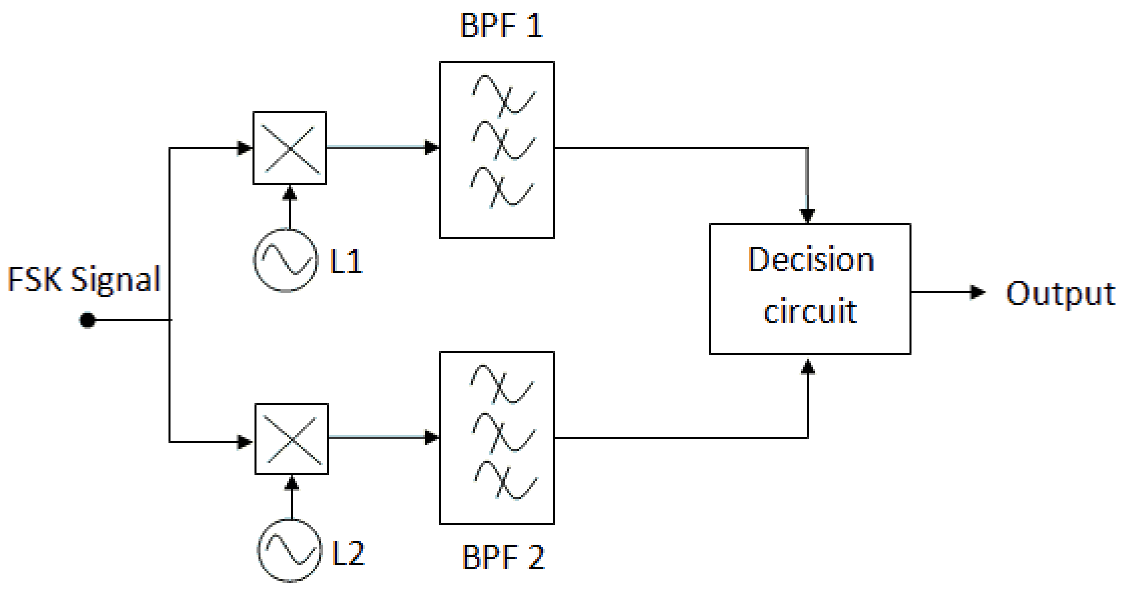

Figure 10. Block diagram of FSK synchronous demodulator [6]

In asynchronous data transmission, clock regeneration and the data re-timing are not required, it is sufficient to square the signal supplied by the filter. Thus, the two signals, with cut off frequencies equals to space and mark frequencies, are passed initially through two band passed filters to eliminate the noise. The input of the filters is then passed to two envelop detectors to be demodulated asynchronously. Finally, there is a decision circuit that choose which output is more probable from the envelope detectors [6].

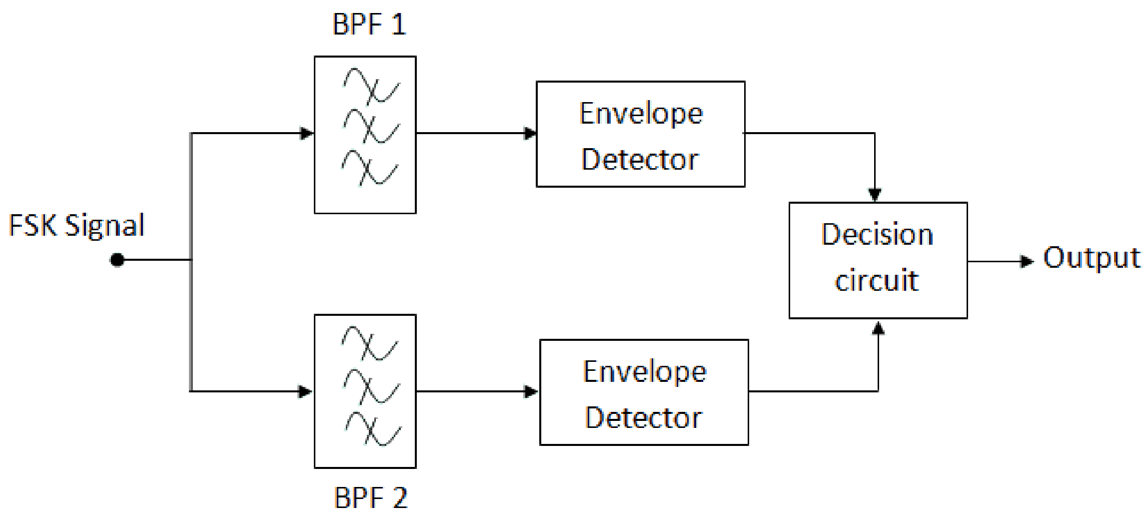

Figure 11. Block diagram of FSK asynchronous demodulator

1.3.3 Phase shift Keying

Phase shift keying is the digital modulation technique in which the phase of the carrier signal is changed by varying the sine and cosine inputs according to the information. This kind of modulation method uses a limited number of phases where each phase can be assigned with binary digits. Generally, every phase encodes an equivalent number of bits. Every bits pattern forms the symbol that is denoted by the exact phase. There are two variations of PSK which are QPSK and BPSK. QPSK stands for Quadrature Phase Shift Keying and it is done by a sine wave carrier that takes four phase reversals such as 0°, 90°, 180°, and 270°. The BPSK stands for Binary Phase Shift Keying and its carrier signal takes two reversals such as 0° and 180°. Comparing with FSK, PSK allows information to be carried with a radio communications signal more efficiently. Furthermore, comparing with ASK, PSK is less vulnerable to faults [7]. Therefore, Binary PSK (BPSK) modulation is chosen in this work to determine the performance of a wireless communication system applying BPSK modulation scheme in an AWGN channel via the implementation of a LabVIEW-based simulator.

1.3.4 M-ary Digital Modulation

This modulation technique consists of transmitting M bits at a time instead of only one bit. This enables reducing the bandwidth. There is several types of this technique: M-ary ASK, M-ary FSK, and M-ary PSK. M-ary ASK is a technique where the amplitude of the carrier signal takes M different level similarly M-ary FSK denotes the technique where the frequency of the carrier signal takes M different level. By the same logic, M-ary PSK stands for a method where data bits select one of M bits phase shifted versions of the carrier to transmit the data. In fact, it is commonly advantageous to encode at a level higher than binary where there are more than two conditions possible for example with 3 bits we have 8 possible combinations [8].

1.4 Binary Phase Shift Keying

Phase-shift keying (PSK) is a digital counterpart to phase modulation with the phase of the carrier wave taking on one of prescribed set of discrete values. The most straightforward type of PSK is called binary phase shift keying (BPSK). BPSK is a two phase modulation scheme, where the 0's and 1's in a binary message are represented by two different phase states in the carrier signal: phase 0 for binary 1 and phase π for binary 0. These methods are widely used in the field of radio and cellular telephony [9].

The equation of the general form for BPSK is:

$$s_n(t) = \sqrt{2 * \frac{E_b}{T_b}} \cos(2 * \pi * f * t + \pi * (1 - n))$$

As we have two cases: n=1 and n=0 thus we get the following equations:

$$s_0(t) = \sqrt{2 * \frac{E_b}{T_b}} \cos(2 * \pi * f * t + \pi) \text{ for binary 0}$$

and

$$s_1(t) = \sqrt{2 * \frac{E_b}{T_b}} \cos(2 * \pi * f * t) \text{ for binary 1}$$

In the equations above f is the carrier frequency, t is the instantaneous time, E_b the transmitted signal energy per bit and T_b is the bit period in seconds. The signal $s_0(t)$ stands for the carrier signal when information bit=0 was transmitted and the signal $s_1(t)$ denotes the carrier signal when information bit =1 was transmitted.

Most digital modulation schemes involve a discrete number of symbols which are used to convey information. These symbols are mapped to a discrete set of magnitude and phase values on the I/Q plane, which are referred to as constellation points. A Constellation diagram is a graphic representation of a digitally modulated signal consisting of the constellation points used to assess the quality of a transmitted signal [10].

As we can notice in the constellation diagram of BPSK in Figure 12, this modulation schemes has a variation of a two phases: +90 and -90. Therefore, it is immune to noise and interference therefore it improves as each modulation symbol represents a single phase. In fact, it needs an important distortion of the signal so that the demodulator gets the wrong received symbol. In this way, BPSK is the most robust PSK modulation technique [7].

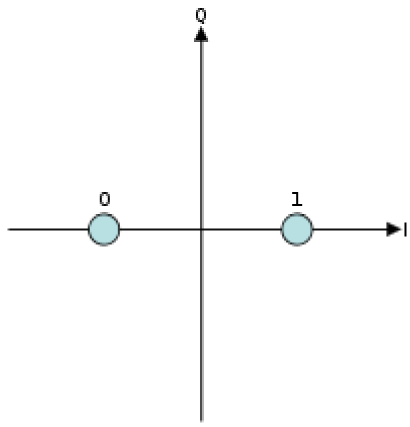

Figure 12. Constellation diagram of BPSK [7]

The block diagram of Binary Phase Shift Keying as illustrated bellow in Figure 13 consists of the balance modulator which has two inputs: the carrier sine wave and the binary sequence. Thus, the BPSK modulation by multiplying the two signals applied at the input using a balance modulator. The phase will be the 0° for a zero binary input and a reversal phase of 180° for a high input [9].

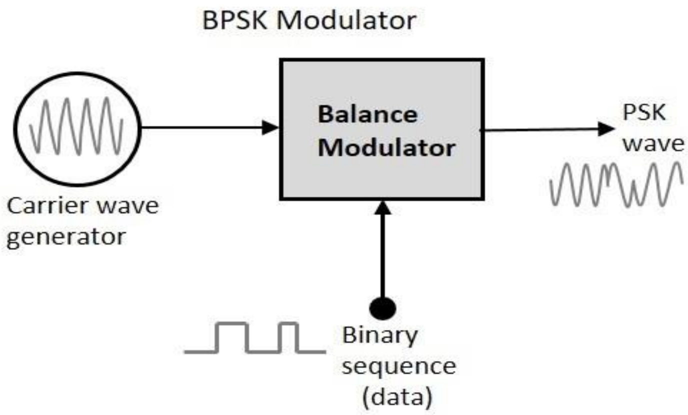

Figure 13. Block diagram of BPSK [9]

Following in Figure 14 is the representation of BPSK Modulated transmitted wave along with its given input. BPSK modulated output wave

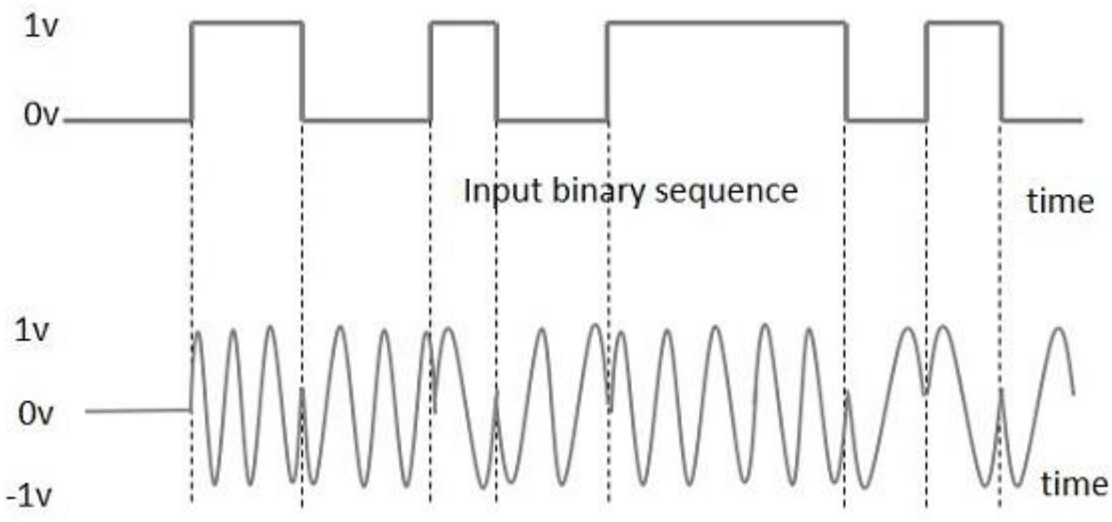

Figure 14. BPSK modulated output wave [9]

In the receiver end, BPSK demodulator consists of a mixer with local oscillator circuit, a bandpass filter, a two-input detector circuit. Its block diagram is shown in Figure 15. By recovering the band-limited message signal, with the help of the mixer circuit and the band pass filter, the first stage of demodulation gets completed. The base band signal which is band limited is obtained and this signal is used to regenerate the binary message bit stream [9].

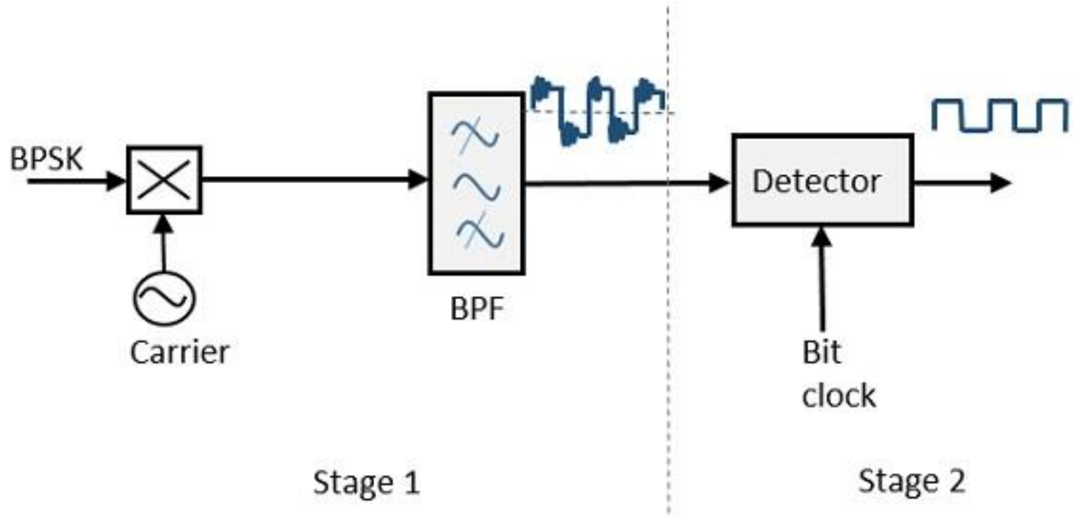

Figure 15. Block diagram of BPSK demodulator [9]

1.5 AWGN channel

The goal of the transmitter is to produce a signal waveform based on the bitstream of the message and to send it to the receiver over a communication channel which will inevitably distort the original signal. Communication channels usually suffer from impairments that lead to errors. These impairments include noise, fading, distortion, attenuation, and interference. Characteristics of a communication channel determine which impairments apply to that particular channel and which are the determining factors in the performance of the channel. Noise is the major impairment that is present in all communication channels.

The AWGN channel is named over the Additive White Gaussian Noise which is a basic model of noise that has its characteristics:

- *Additive*: it can be added to any noise or to the transmitted signal, where the noise is independent of the signal.
- *White*: it has flat power spectrum
- *Gaussian*: it has a Gaussian distribution

The AWGN channel model is a channel whose main impact is addition of a white Gaussian noise process to the transmitted signal. Although the AWGN channel model seems very limiting, its study is beneficial from two points of view. First, noise is the major type of corruption introduced by many channels. Therefore, isolating it from other channel impairments and studying its effect results in a better understanding of its effect on all communication systems. Second, the AWGN channel, although very simple is a good model for studying wireless communication channels in comparing their performance [11].

1.6 Evaluation of the performance of communications schemes – BER curve

When data is transmitted over a communication channel, the goal is that the information is efficiently transmitted with a certain degree of reliability yet there is always a possibility of errors being presented into the data due to channel distortion. If errors are introduced into the system, then the integrity of the system may be mislaid. As a result, it is primordial to evaluate the performance of the system, and bit error rate (BER), provides an ideal indicator in which this can be achieved. BER information can be determined for optic fiber links, ADSL, Wi-Fi, cellular communications, Internet of Things links and many more [11].

As for the analog communication system the optimum performance indicator is the distortion of the signal, for a digital communication system the optimum measure of system performance is BER which is calculated at the receiver output. As its name implies, a BER is defined as the probability at which errors occur in a communication system. This can be directly translated into the number of bit errors divided by the total number of transferred bits. Hence, the definition of bit error rate can be translated into a simple equation:

$$BER = \frac{Number\ of\ erroneous\ bits}{Total\ Number\ of\ Bits}$$

This approach gives identical BER with the real probability if an infinite number of the bits are transmitted. Unfortunately, this is not feasible, but as a rule of the thumb can be applied, namely, if the number of bits ten times greater than needed for 1 erroneous bit theoretically, the approximated BER will be close to the theoretical one.

Since BER quantifies the reliability of all the entire system, it depends on all the components of the communication system and may be affected by many factors: transmission channel noise, distortion, attenuation, fading, synchronization problems and interferences. However, it can also be improved by implementing error correction schemes, resource allocation mechanisms, power control or link adaptation schemes. In this thesis the focus is on the raw bit stream without any bit error correction techniques.

In general, the BER cannot be analytically calculated and need to be estimated. We can use the accurate BER estimates as meaningful feedback quality criteria. For example, power control mechanisms in digital communication systems typically use the bit error rate as a quality measure feedback to maintain transmitting power at minimum required levels to maintain a desired Quality of Service. Thus, the simulation of BER is very important not only as the most important criteria for digital communication but also as feedback that enables system level optimization [12, 13].

1.7 Summary of this chapter

This chapter was a general overview on the tools that I choose to use on this work with an explanation of each one. Actually, this chapter investigated the specification of a digital communication system along with its different modulation schemes and a detailed clarification of every technique which lead us to understand the reason of choosing the BPSK modulation schemes. Finally, this chapter presented the performance criteria that will be used further.

2 Simulator

This chapter gives the description of the simulation approach and discuss its adequacy in this work. Furthermore, I will introduce the simulator used in this thesis. Finally, I will deal with the parts of the model of digital communication system implemented in the simulator.

2.1 The simulation approach and its usefulness

Simulators are software that allows mimicking of the performance of a real system to investigate information about it. The usefulness of simulators consists first in the cost-effectiveness because the cost to repair a real system whenever an error is detected is dramatically high. Besides, interactive simulation instruments have an increasing role in the engineering education as it gives more flexibility and presents a safe and secure platform for learning and doing research. Simulators also provides interactivity with the implemented system. All those reasons make simulators a good tool for us to build our digital communication system and analyze its operation.

2.2 Simulation Platform – LabVIEW

As a simulator I choose to work with LabVIEW. The name LabVIEW is a shortened form of its description: Laboratory Virtual Instrument Engineering Workbench. LabVIEW is the software design framework of any engineering application that requires test, measurement or control. There are two things that makes LabVIEW uniquely valuable for engineers: the first is the fast and extensive hardware integration available for everything for bench type instruments or PC-based extension board to implement software defined radios, the second is that LabVIEW is a graphical approach to programming. LabVIEW recompile its code for every action which means you can detect unfixed errors at they happened rather than having to compile and fix things at the end. Besides, LabVIEW represent parallelism in the code naturally [13].

LabVIEW is essentially an environment that enables programming in G which is graphical programming language created by National Instruments. In fact, a LabVIEW program is a VI (Virtual Instrument). VI is composed by two windows represented in Figure 13:

Front panel: (the grey window of Figure 16 is the user interface in Labview program. The front panel has controls and indicators, which are the interactive input and output terminals, respectively, of the VI. It services anything to visualize Data for the program

Block diagram: (the white window of Figure 16) represents the functional/graphical code that we create. The concept of the block diagram is to separate the graphical source code from the user interface in a logical and simple manner. Front panel objects appear as terminals on the block diagram.

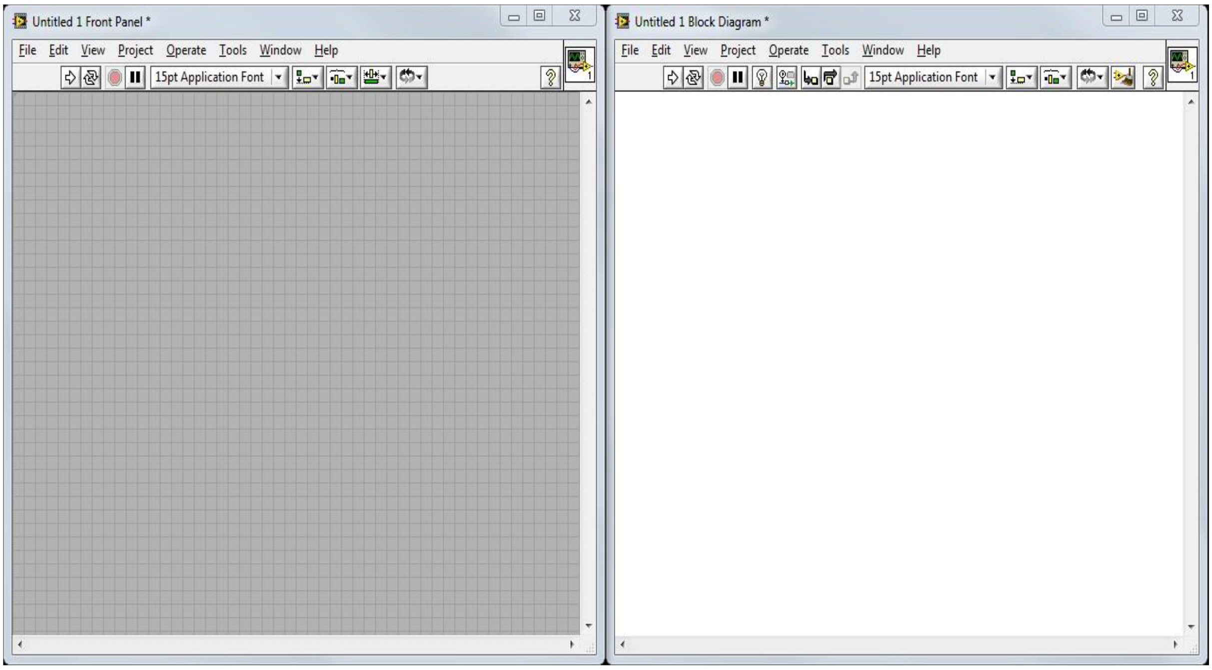

Figure 16. Interface of LabVIEW [13]

LabVIEW offers various display options and it is designed to facilitate promote data collection and analysis. LabVIEW involve several tools to help in troubleshooting the code. In addition, it offers a comprehensive set of VIs and functions for acquiring, analyzing, displaying and storing data. LabVIEW has many built-in features and may be used as for simulation and control. Other advantages of using LabVIEW is that it is possible to vary the input parameters and the corresponding results by setting changes in the appropriate block in the front panel. Within its built-in analysis capability, Labview contains functions to generate signals, analysis, visualization, and processing of standard and custom digital and analog modulation formats. LabVIEW also provides a Toolkit helps to rapidly develop custom applications for research, design, characterization, validation, and test of communications systems and components that are used to modulate and demodulate signals. Besides, there is a various Modulation Toolkit applications including analog and digital modulation schemes like AM, FM, PM, ASK, FSK, MSK, GMSK, PSK, QPSK, PAM, and QAM. These modulation schemes are the foundation of many digital communication standards found in 802.11a/b/g/n, ZigBee (802.15.4), WiMAX (802.16), RFID, satellite communications, and commercial broadcast among others [13].

In my thesis work I did not use pre-built Toolkits, instead I designed and implemented my own Vis to build up the simulator the BPSK system.

2.3 Elaborated BPSK simulator

The simulator focuses on the evaluation the performance of a physical layer of an embedded wireless communication system. Figure 17 shows the digital communication system implemented in LabVIEW. In the model shown in Figure 17 I generated a bit stream using *Bit_Stream_V03.vi* that imitate the information source that we already introduced in Figure 17. The components of digital communication system and this bits stream is processed in the subVI called *transmitter.vi* then it is sent through the channel called *channel.vi*. The subVI, named *receiver.vi*, that imitate the receiver of a real communication system treat the received signal and gives the recovered bit stream at its output that is most likely been transmitted by the transmitter unit.

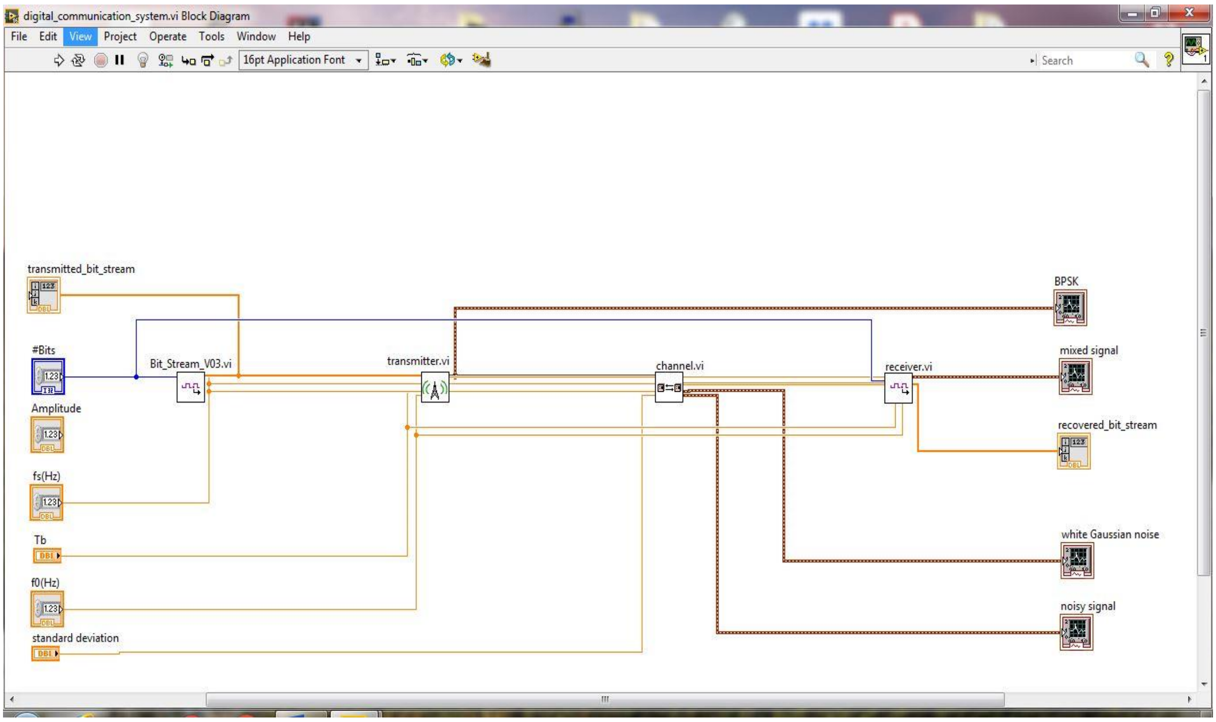

Figure 17. The simulated digital communication system in LabVIEW

2.3.1 Transmitter:

The transmitter consists of two subVI's as shown in Figure 18. the encoding *NRZ_signal_v03.vi* and the modulator *BPSK_modulator.vi*. The *NRZ_signal_v03.vi* in Figure 18 acts as the channel encoder and the *BPSK_modulator.vi* in Figure 18 acts as the modulator that we explained in The components of digital communication system. As for the encoding, I choose to work with the non-return to zero that returns the ones by a positive voltage and the zeroes to a negative voltage. Moreover, I choose to work with the bpsk modulation that we explained previously in 1.4. As for the Bpsk modulator, I generated a carrier signal and multiplied by our signal. The parameters of the modulator are:

T_b: The bit time

f_c : The carrier frequency

Array: the input bit stream which represent the information source in nature

And f_s is the sampling frequency for the simulator

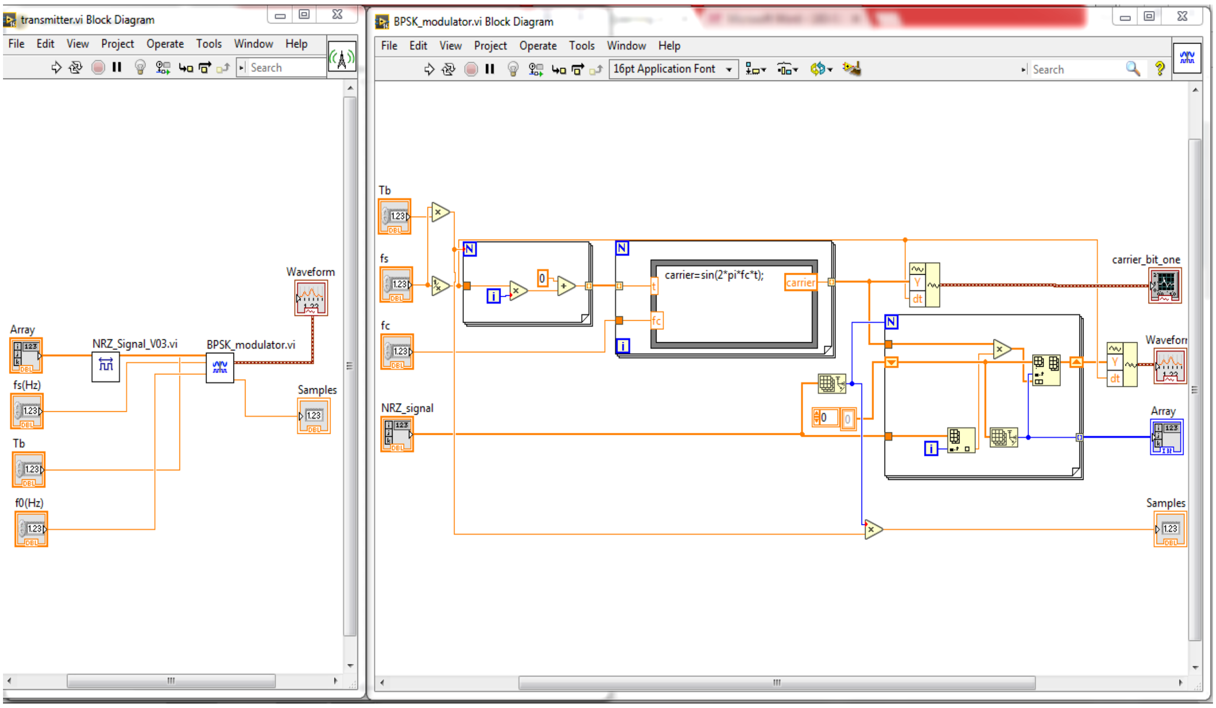

Figure 18. Block diagram of the transmitter and BPSK modulator

In Figure 19, I plotted the different signals that we have in the transmitter to insure that the work is accurate. As we can see, the generated bit stream the upper left corner is a sequence of 0 and 1 that we can chose its length by setting the parameter #bits and this bit sequence is represented in the Bit_stream_graph. This bit stream is then encoded and transformed to a square sequence using the NRZ encoding (Non Return to Zero). This encoding represents the ones by a voltage 1 and the zeros by a voltage -1 and it is displayed as an NRZ_Signal graph as shown in lower left corner of Figure 19. Within the *BPSK_modulator.vi*, we applied the BPSK modulation and we get the BPSK modulated signal as shown in upper left corner of Figure 19.

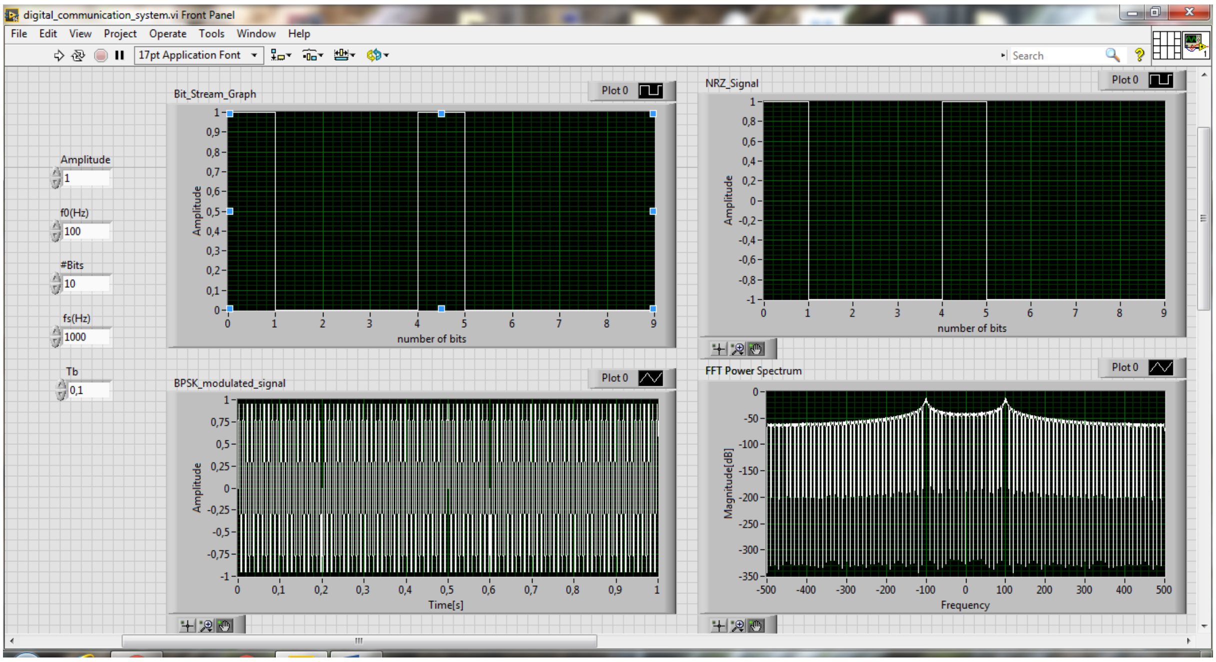

Figure 19. The transmitter part on the front panel of digital communication system

The Figure 20 is a zoomed version of the different graphs of the transmitter. The BPSK modulated signal is characterized by the reversing of the waveform in case of a bit 1 due to the change of the phase. Hence, when the 2th bit or the 5th bit is 1 in the bit stream graph in the upper left corner of Figure 20, the waveform is reversed at t=0.2s and t=0.5s in the BPSK modulated graph in the lower left corner of Figure 20 which marks the transition of the phase in BPSK modulation. Finally, to assure that the transmitter model is reliable, I ploted the frequency spectrum of the modulated signal. The fourier_transformation.vi displays the Fourier transformation of the BPSK modulated signal as shown in the lower right corner of the Figure 20. This yields us to check the frequency domain for example for the carrier frequency fc=100, the difference between the main lope and the next one is ~13 dB which prove that our spectrum is accurate when ZoH pulse shaping is used in the transmitter. Hence, this spectrum confirms that the transmitter model is accurate.

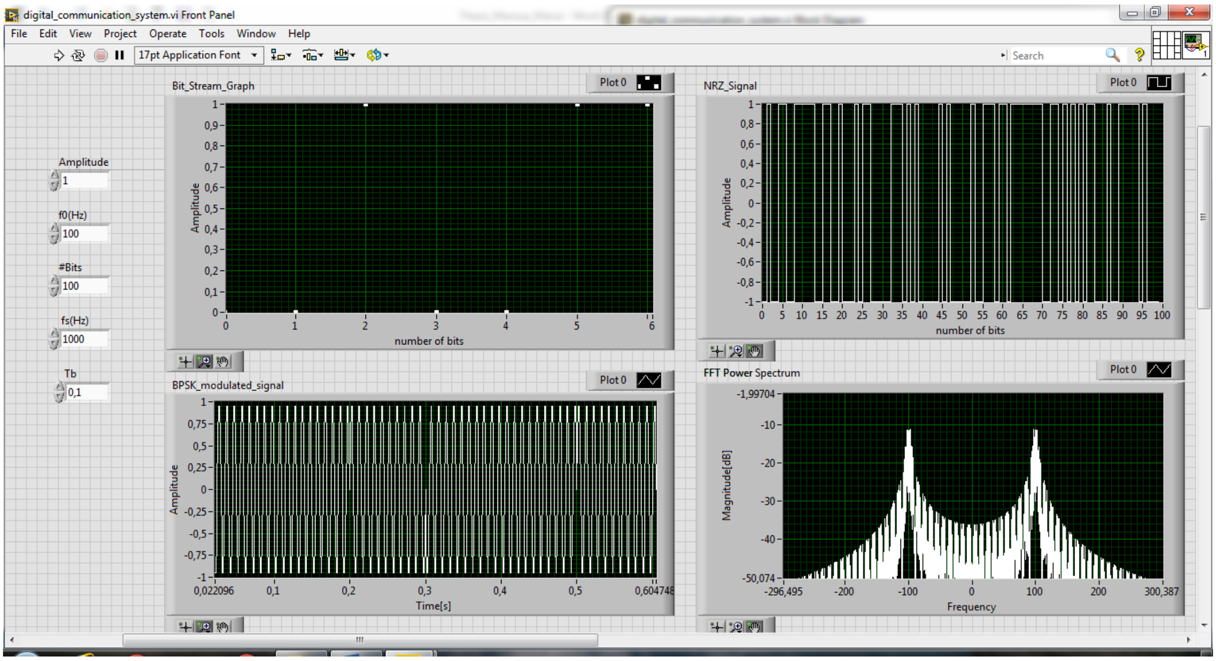

Figure 20. The zoom in of the transmitter graphs in LabVIEW

2.3.2 Channel

The channel is representing the physical medium of the digital communication system transferring the transmitted signal. As explained before the channel adds some noise that distorts the original transmitted signal. Therefore, I used the white Gaussian noise displayed in the channel to approach the effect of noisy signal in the real case. Additive White Gaussian Noise is a basic noise that its name encloses its characteristics. This noise is additive as it is added to any noise or to the transmitted signal while being statistically independent of the other signals. White refers to the fact that it has uniform power across the whole frequency band. As a consequence, the spectral density of white noise is ideally flat for all frequencies ranging from $-\infty-\infty$ to $+\infty+\infty$. Gaussian because it has a normal distribution in the time domain with an average time domain value of zero. The Additive White Gaussian is generated in function of the standard deviation as in the front panel in Figure 21. Block diagram of Channel subVI and front panel of digital communication system and it is been added as shown in the block diagram seen in the right side of Figure 21.

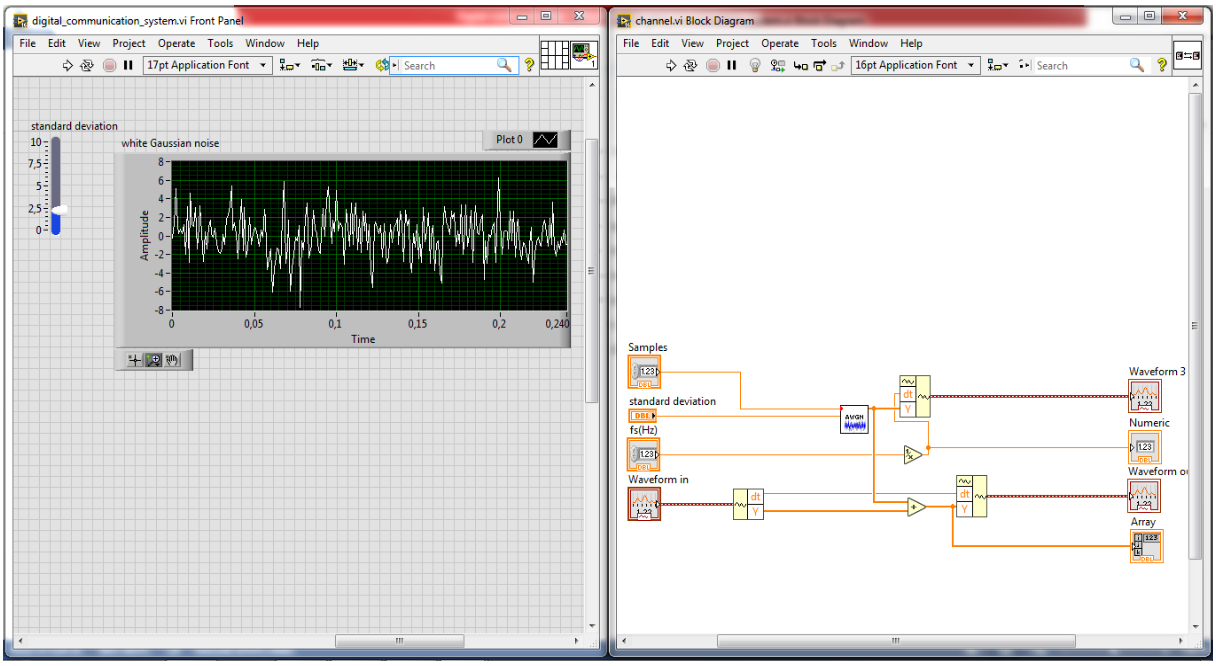

Figure 21. Block diagram of Channel subVI and front panel of digital communication system

As for the transmitter, I plotted the different graph of the channel to insure its right performance. As we already said, the white Gaussian noise is generated and added to the BPSK modulated signal. The result is a distorted and noisy signal as we can see in Figure 22. To check the reliability of the channel, I plotted the Fourier transformation. The spectrum of the noisy signal is a flat distribution as shown the noisy signal FFT power spectrum in the lower right corner of Figure 22. This shape proves that the channel model is good as we are working with white Gaussian noise characterized with its flat spectral density.

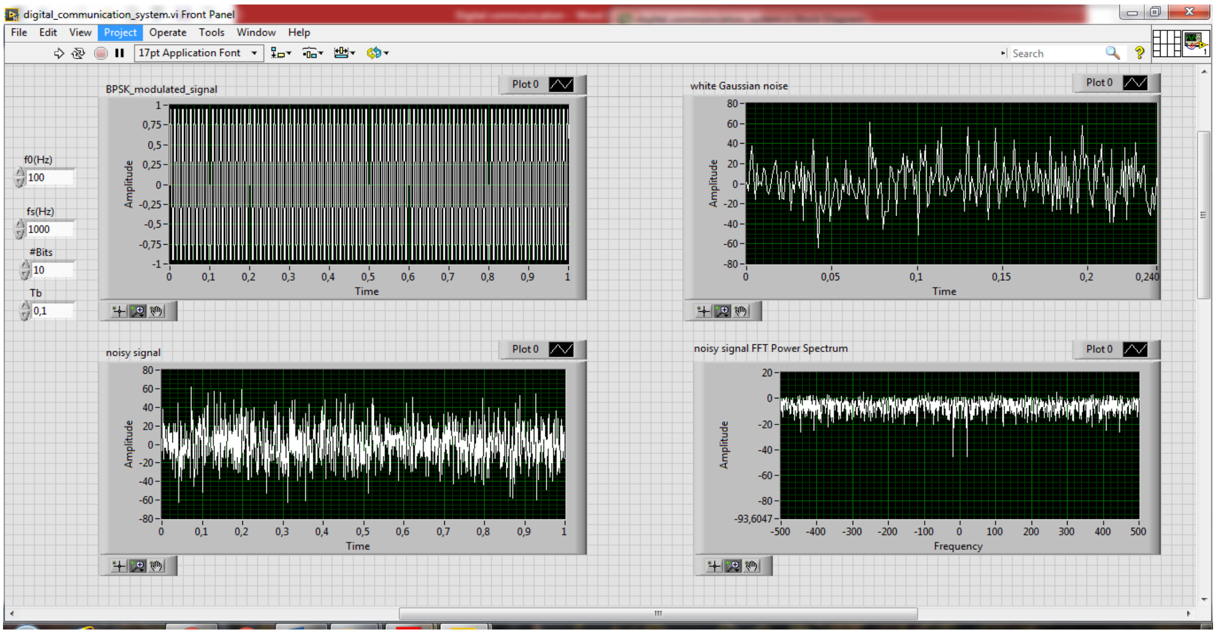

Figure 22. The Channel part on the front panel of the digital communication system

2.3.3 Receiver

After sending the transmitted signal through the channel, it arrives to the receiver. At the receiver end, the transmitted signal must be first demodulated and filtered. Actually, the received signal will be expressed as follows after mixing with the locally generated signal

$$(b_2 \cos(2\pi f_c\, t) + n(t)) * \cos(2\pi f_c t) = b_2 \frac{1 + \cos(2\pi 2 f_c t)}{2} + n(t) * \cos(2\pi f_c\, t)$$

Where the $n(t)$ is the noise and b_2 is the amplitude of the received signal. This signal is integrated over the bit time to determine an observation signal and be fed into a decision maker.

Thus, our receiver consist of a detector/demodulator and a decision maker as seen in the block diagram of the receiver in the right side of the Figure 23. In the left side of Figure 23 we can see the noisy signal and the mixed signal after been multiplied by the local carrier. We generated a recovered bit sequence as we can see it in the recovered bit stream array in the front panel of Figure 23. We display the recovered bit stream as an array to compare it with transmitted bit stream and to evaluate the performance of our system based on BER.

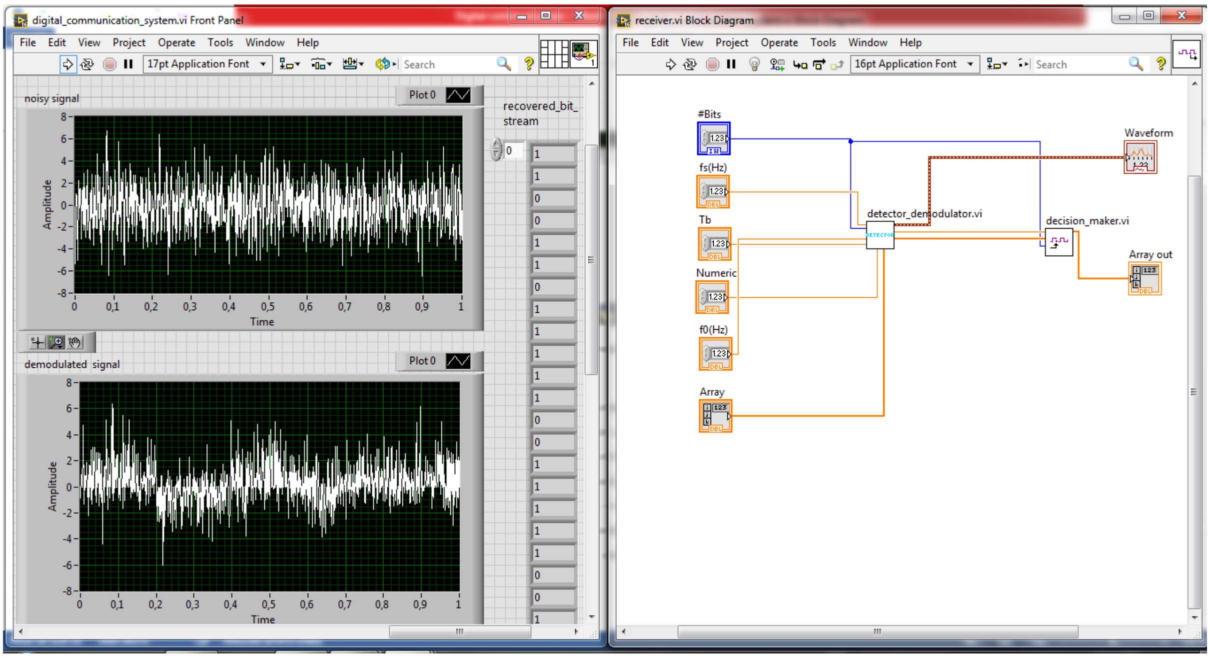

Figure 23. Block diagram of the receiver and its output in the front panel of the digital communication system

As mentioned when demodulating the signal, it is first multiplied by a locally generated carrier that is displayed in the mixed signal graph in upper right corner in Figure 24. Processing the observation signal, the decision maker determine wherever the signal is less than 0 or more to round it. The demodulated signal is then generated and as we can see it in the lower right corner of Figure 24. As we can notice, the input transmitted stream in the upper left corner and the received demodulated stream are not totally corresponding and this due to AWG noise.

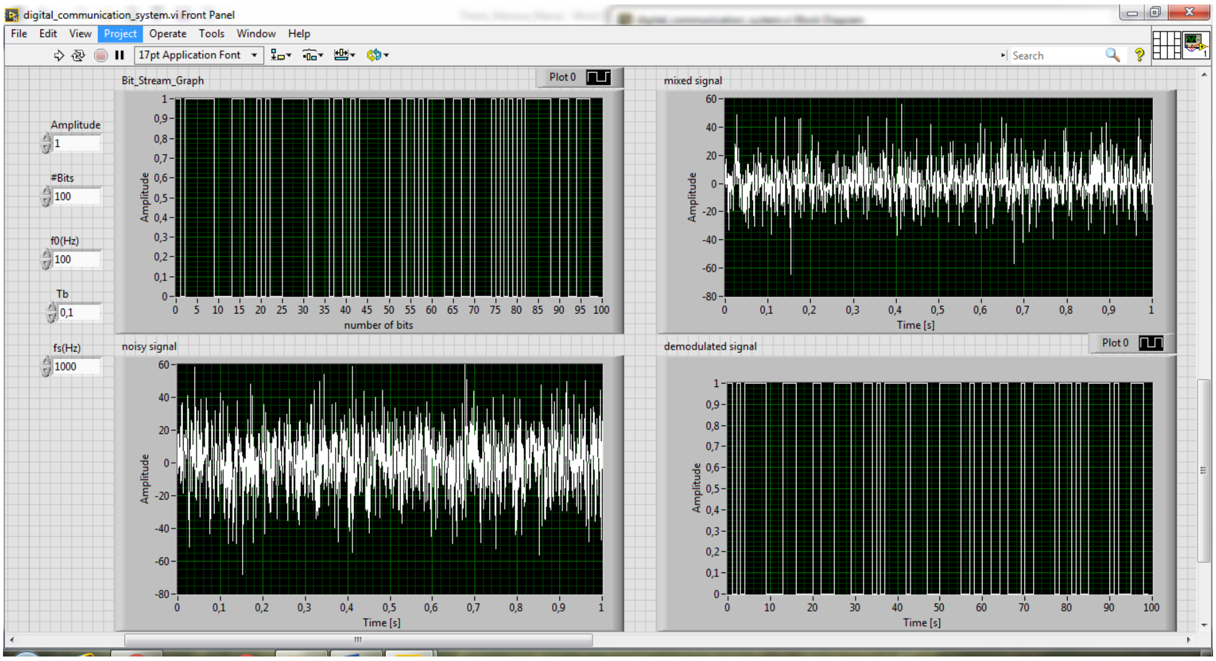

Figure 24. The receiver part on the front panel of digital communication system

In the absence of noise, furthermore, the received stream will be the same as the transmitted stream. Actually, we can see the noiseless version of the digital communication system in Figure 25 where the transmitted bit stream at the upper left corner of Figure 25 is the same as the demodulated signal in the lower left corner of Figure 25.

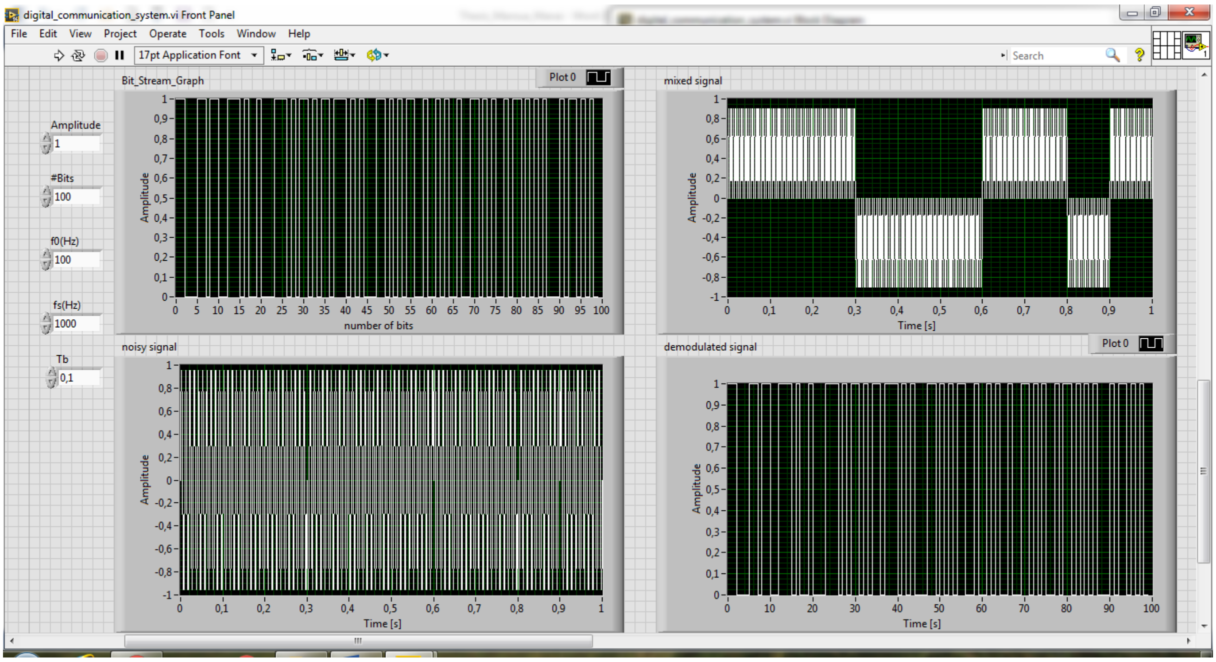

Figure 25. Front panel of a noiseless version of the digital communication system

2.4 Summary of this chapter

This chapter served first of all as a definition the simulation approach and its adequacy for this work. Secondly, I introduced the blocks of the simulator. Ultimately, I highlighted the model of digital communication system that is used to analyze the performance.

3 Results

This chapter devoted to the experiments that I did and the results. I will start by stating some theoretical essential and estimations. Then, I will analyze the results that I got and then I go through the different cases. Finally, I will discuss the cases and conclude whether the results are efficient and accurate or not.

3.1 Evaluation of the performance of the digital communication system:

3.1.1 Theoretical BER

The goal of this work is to evaluate the performance of the BPSK digital communication system. So the first step is to simulate the communication system by transmitting a BPSK signal through an AWNG channel and simulate the effects of the AWGN channel. The mathematical representation of an AWGN channel in which a signal is distorted is described by the following equation:

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

where s(t) is the transmitted signal, n(t) is the additive white noise and r(t) is the received signal. The receiver gets the received signal r(t) and try to make the optimal decision about which message symbol was transmitted. In the case of BPSK modulation, the received signal corresponds to a transmission of a 0 and 1 bit respectively:

$$r = s_0 + n$$

$$r = s_1 + n$$

The noise follows the Gaussian probability distribution function:

$$p(x) = \frac{1}{\sqrt{2\pi\sigma^2}} e^{-\frac{(x-y)^2}{2\sigma^2}}$$

where the mean μ is zero and the power spectral density of the noise σ [11].

So the conditional probability distribution in the case of BPSK is:

$$\mathrm{p}(\mathrm{r}|s_0) = \frac{1}{\sqrt{\pi N_0}} \mathrm{e}^{-\frac{\left(r+\sqrt{E_b}\right)^2}{N_0}}$$

$$p(r|s_1) = \frac{1}{\sqrt{\pi N_0}} e^{-\frac{(r-\sqrt{E_b})^2}{N_0}}$$

Where the transmitted signal energy per bit is:

$$E_b = A^2 T_b$$

If we assume that s_0 and s_1 are equally probable then $p(s_0)$ =$p(s_1)$ = ½ then the threshold will form the optimal decision boundary. Thus, if the received signal r is less than or equal to zero, then the receiver assumes s_0 was transmitted and if the received signal larger than zero the receiver assumed that it is s_1.

In the other hand, the AWGN model is defined by:

$$AWGN \sim N(0, \sigma^2)$$

Where σ is the standard deviation.

Let's define the random variable of the Gaussian Distribution Y is:

$$\sigma^2{}_Y = E[(Y + A)^2]$$

$$\sigma^2{}_Y = \frac{1}{{T_b}^2} E[\int_0^{T_b}\int_0^{T_b} w(t)w(t)dt\, du]$$

$$= \frac{1}{{T_b}^2} \int_0^{T_b}\int_0^{T_b} E[\, w(t)w(t)]\, dt\, du$$

$$= \frac{1}{{T_b}^2} \int_0^{T_b}\int_0^{T_b} R_W(t,u)\, dt\, du$$

Where $R_W(t,u)$ is the autocorrelation function of noise $w(t)$, which is white noise with a power density, $N_0/2$ thus, we get:

$$R_w(t) = \frac{N_0}{2}\, \delta(t-u)$$

Where $\delta(t-u)$ is the time-shifted delta function, hence the variance can be expressed as:

$$\sigma^2{}_Y = \frac{1}{{T_b}^2} \int_0^{T_b}\int_0^{T_b} \frac{N_0}{2}\, \delta(t-u)dt\, du$$

$$\sigma^2{}_{\gamma} = \frac{N_0}{2T_b}$$

Since the average probability of symbol error is

$$P_e = \frac{1}{2} erfc\,(\frac{A}{\sqrt{N_0/T_b}})$$

Then the theoretical bit error rate can be defined as:

$$BER_{theoritical} = \frac{1}{2} erfc(\frac{1}{\sqrt{2}\sigma})$$

Where $erfc(x)$ denotes the complementary error function defined as:

$$erf(x) = \frac{2}{\sqrt{\pi}} \int_0^x e^{-t^2}\, dt$$

$$erfc(x) = 1 - \mathrm{erf}(x) = \frac{2}{\sqrt{\pi}} \int_x^{+\infty} e^{-t^2}\, dt$$

The Figure 26 illustrates the theoretical BER for BPSK modulation achieved by an optimum receiver. This curve is expressed in function of ratio of bit energy to noise power spectral density.

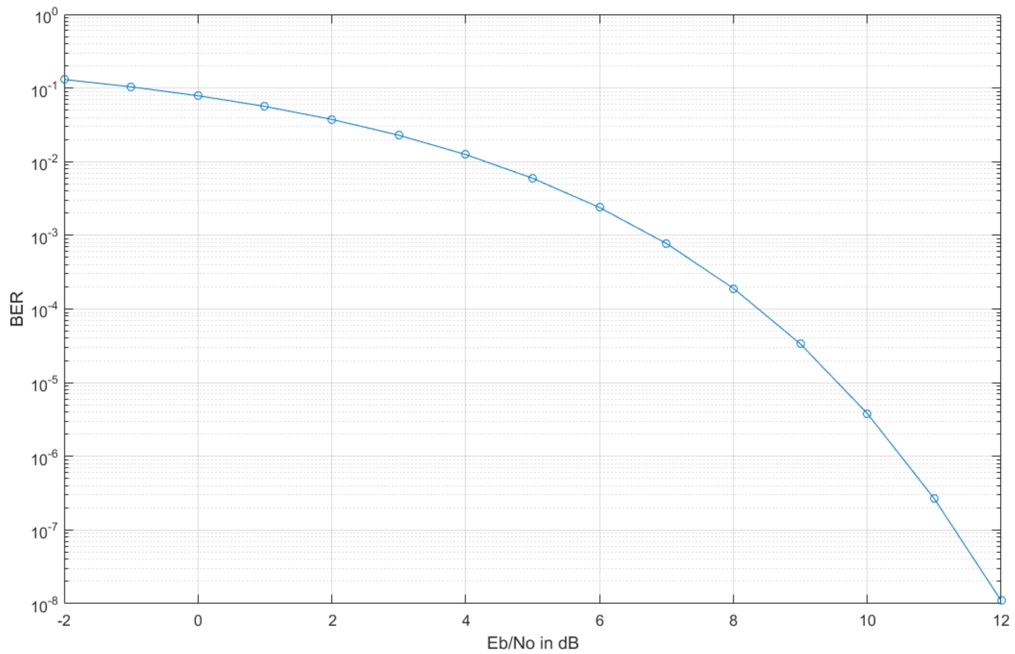

Figure 26. Theoretical BER curve

3.1.2 Simulated BER curve

To calculate BER the number of errors have to be determined that I got in the presence of the AWGN. That's why I applied a loop to evaluate the performance of the BPSK system for different noise levels of the channel and to automatically generate the BER curve. It has been carried out by generating a vector of containing the standard deviation of the noise and the bit error rate has been calculated for every value of the standard deviation of the noise. Then I plotted the BER curve as a function of the reciprocal value of the standard deviation squared, i.e., the variance, i.e., the power of the channel noise as shown in Figure 27.

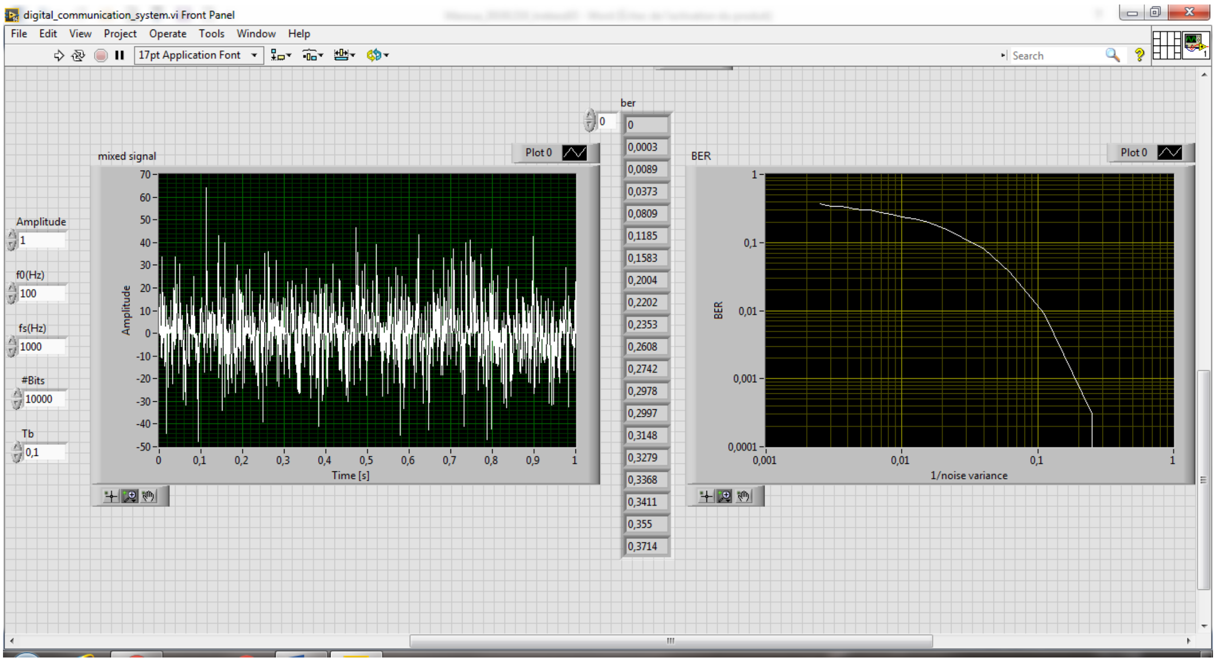

Figure 27. BER array and BER curve of digital communication signal for 10000 bits

The Figure 28 represents the BER curve for 1000 bits. We can observe that for a higher Number of bits the curve is accurate, that is explained by the fact that for a higher number of bits there is more probability to have errors, and the shape of the curve will be more close to the theoretical case. As an example, we plotted the curve for a number of bits= 1000 and we can see that it is not well shaped in Figure 28 since data samples are not sufficient to have a good precision but the results are better for a number of bits= 10000 shown in Figure 27.

In both cases, we notice that when the variance of the channel noise is high the bit error rate is equal to 0.5. This observation is explained by the fact that when the noise level is high the receiver can no longer demodulate and recover the original signal. As a consequence, it will just keep guessing which transmitted signal is sent and since the two signals are equally probable then the BER will be 0.5.

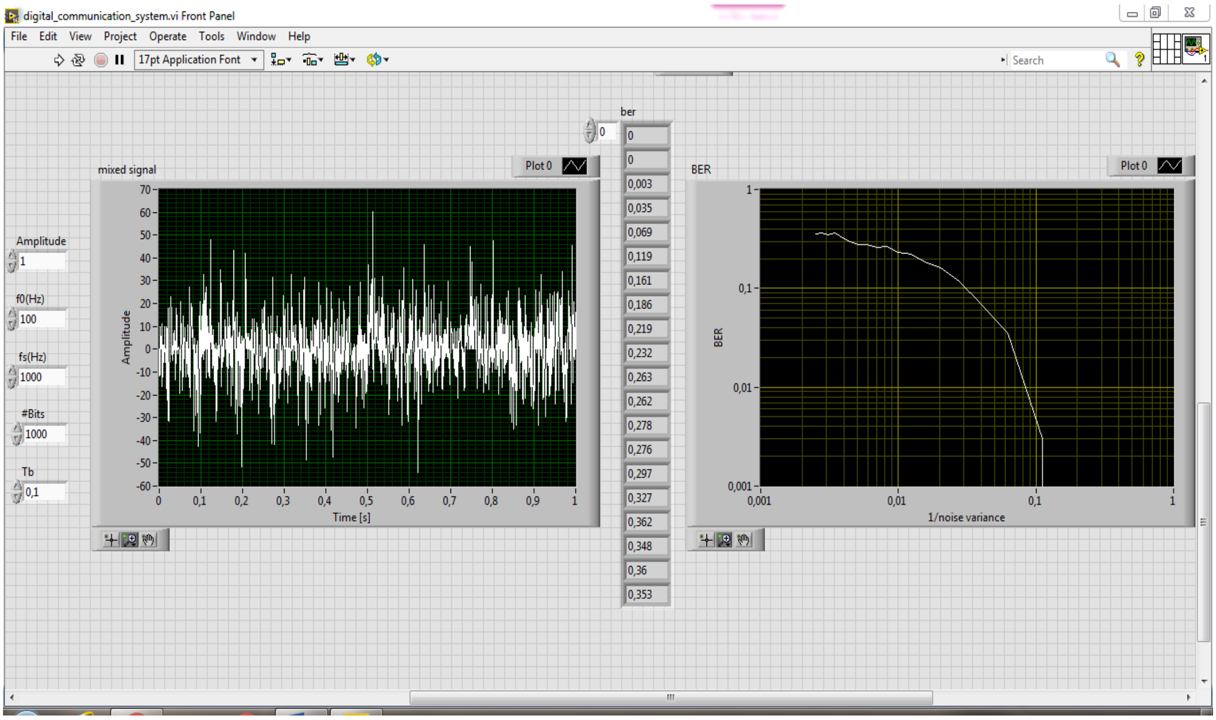

Figure 28. BER array and BER curve of digital communication system for 1000 bits

Based on our experiments, we can conclude that the BER for our digital communication system decreases monotonically with the decreasing noise variance and its curve has the shape in form of waterfall.

For BER= 10^ (-2), we have 1 error out of 100 bits.

From the experiments that we made and the results that we got we can resolve that the digital communication system that we simulated is transmission reliable for the transmission of a large number of transmitted symbols as an example we can see Figure 29.

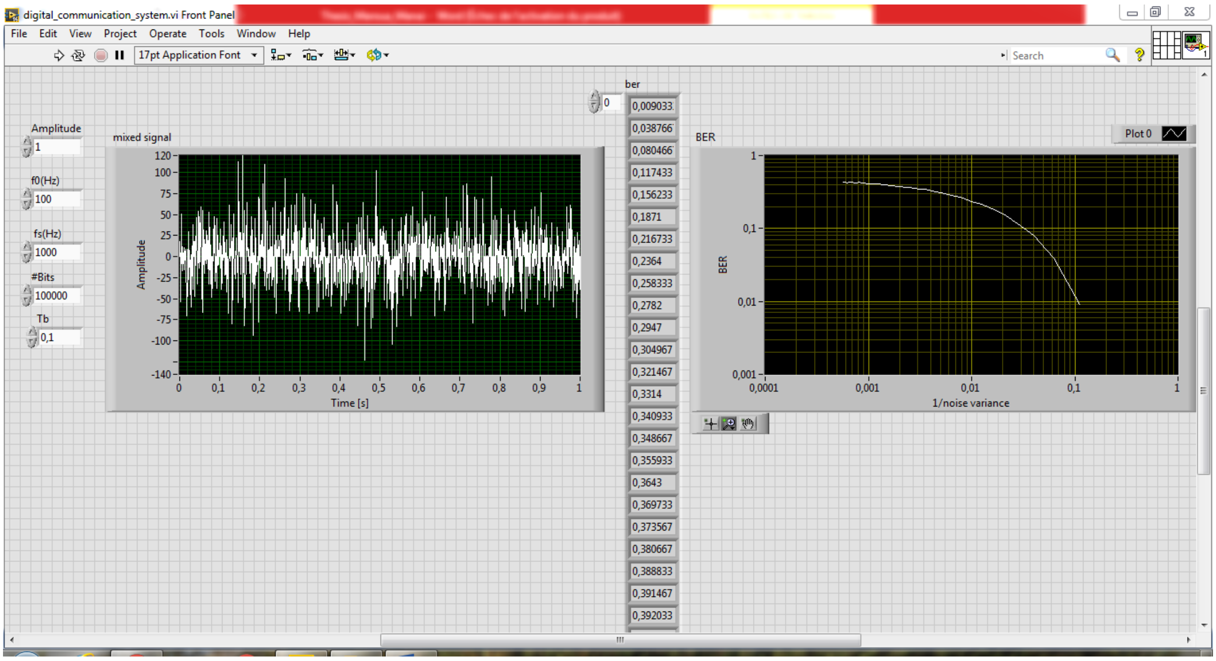

Figure 29. Front panel of Ber curve for 100000 bits

3.2 Summary of this chapter

In this chapter, we went through the theoretical and the simulated BER. We analyzed our simulated curve and made conclusions about it. Finally, this chapter was the summary of the results that we got.

4 Conclusion

In many communication systems, a real-time BER estimation or simulation would be efficient for evaluation the performance. In our case, we gave special interest to the physical layer of an embedded communication system as digital communication system has proliferated in big way recently. Thus, this work went through the simulation of a digital communication system that resembles the real case and then evaluated it performance using BER which is a key indicator to digital communication. In fact, this thesis was an overview and analysis of a digital communication system based on BPSK modulation scheme that shows its efficiency comparing to other modulation techniques. After the simulation and the experiments, the BER curve of our system was plotted. Finally, the BER curve of system confirms its efficiency since the typical BER shape developed and showed the qualitative performance of our BPSK digital communication system.

5 References

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