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'LICENCE'

In Electrical and Electronic Engineering

Title:

Digital Transmission of Analog Signal

Presented By:

- **Nedjem Eddine AFFOUN**
- **Zineddine REZOUG**

Supervisor:

Mrs. N.DERRAGUI

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Abstract

Digital transmission is a way to transmit data from one point to another without consuming much bandwidth and hence power. It is also the only way to treat the data using a computer before transmission in order to encrypt or store it. In this project, we have introduced the digitizing process, designed and implemented a circuit to transmit an analog signal digitally and reconstruct it again at the receiver. The results of our circuit were satisfactory at many points whereas the reconstruction output was not perfect. As a channel, we were supposed to use the optical fiber which did not work, so we have used a copper wire to test our circuit.

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List of Abbreviations

BW	**B**AND **W**IDTH
PCM	**P**ULSE **C**ODE **M**ODULATION
LED	**L**IGHT **E**MMITING **D**IODE
ADC	**A**NALOG TO **D**IGITAL **C**ONVERTER
PISO	**P**ARALLEL **I**NPUT **S**ERIAL **O**UTPUT
SIPO	**S**ERIAL **I**NPUT **P**ARALLEL **O**UTPUT
DAC	**D**IGITAL TO **A**NALOG **C**ONVERTER
LPF	**L**OW **P**ASS **F**ILTER
SW	**SW**ITCH
GND	**G**ROU**ND**
REF	**REF**ERENCE
MSB	**M**OST **S**IGNIFICANT **B**IT
LSB	**L**EAST **S**IGNIFICANT **B**IT
NC	**N**O **C**ONNECTION
CLK	**CL**OC**K**
CLR	**CL**EA**R**

General Introduction

Due to a host of well-conceived ideas, indispensable discoveries, crucial innovations, and important inventions over the past two centuries, information transmission has evolved immeasurably. The technological advances in communications and their corresponding societal impacts are moving at an accelerating pace.

Throughout most of the twentieth century, a significant percentage of communication systems was in analog form. However, by the end of the1990s, the digital format began to dominate most applications. One does not need to look hard to witness the continuous migration from analog to digital communications: from audiocassette tape to MP3 and CD, from NTSC analog TV to digital HDTV, from traditional telephone to VoIP, and from VHS videotape to DVD. In fact, even the last analog refuge of broadcast radio is facing a strong digital competitor in the form of satellite radio.

Digital transmission is the sending of information over a physical communications media in the form of digital signals. It has many advantages comparing to analog transmission:

- Digital signals can withstand channel noise and distortion much better than the analog as long as the noise and the distortion are within limits.
- Digital hardware implementation is flexible and permits the use of microprocessors.
- Digital signals can be coded to yield extremely low error rates and high fidelity as well as for privacy.
- Digital signals have the viability of regenerative repeaters in the former.

Since signals in nature are always in the analog form, we have to find a solution to digitize them before transmission. In this project, we are designing and implementing a circuit to digitize analog signals, transmit them through a physical channel, and then reconstruct them in the original form. Firstly, we are going to introduce the digitizing process that is based on the Pulse Code Modulation (PCM) which is divided into three main parts: sampling, quantization, and encoding. In the same chapter, we are going to talk about data transmission especially using the optical fiber which is our transmission line in this work. Secondly, we are going to present our design to realize our project electronically using specific chips and boards. Finally, we are going to implement our project and introduce its results and all the problems we have faced.

The main objective of this work is to realize a digitizing and reconstruction process without the use of a microprocessor which makes it very interesting and challenging work that allows us to observe every single step in the process, study it, and find solutions for every occurred problem. So, our work is a beneficial project that is characterized by different features that enhance further research and gives us the chance to practice what we had as background knowledge.

Chapter 1

Digitizing Process and Data Transmission

Introduction

Digitization is the process of converting analog signals or information of any form into a digital format that can be understood by computer systems or electronic devices. The term is used when converting information or data into binary code. In this chapter, we are going to introduce general definitions of analog and digital signals, the main steps of digitizing process and finally the way to transmit the digitized signal.

1.1 Analog and Digital Signals

1.1.1 Analog Signals

An analog signal is one type of continuous time-varying signals, and these are classified into composite and simple signals. A simple type of analog signal is nothing but a sine wave, and that can't be decomposed, whereas a composite type analog signal can be decomposed into numerous sine waves. An analog signal can be defined by using amplitude, time period otherwise frequency, & phase. Amplitude streaks the highest height of the signal, frequency streaks the rate at which an analog signal is varying, and phase streaks the signal position with respect to time. An analog signal is not resistant toward the noise, therefore; it faces distortion as well as reduces the transmission quality. The analog signal value range cannot be fixed.

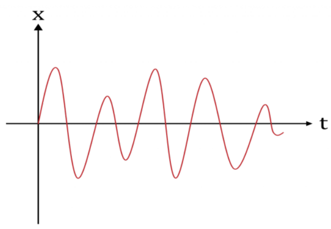

FIGURE 1.1: Analog Signal

1.1.2 Digital Signals

A digital signal represents information as a series of binary digits. A binary digit (or bit) can only take one of two values - one or zero. For that reason, the signals used to represent digital information are often waveforms that have only two discrete states.
In the signal waveform shown below, the signal alternates between two discrete states (0 volts and 5 volts) which could be used to represent binary zero and binary one respectively. If it were actually possible for the signal voltage to instantly change from zero to five volts (or vice versa), the signal could be said to be discontinuous. In reality, such an instantaneous transition is not physically possible, and a small amount of time is required for the voltage to increase from zero to five volts, and again for the signal to drop from five to zero volts. These finite time periods are referred to as the rise time and the fall time respectively.

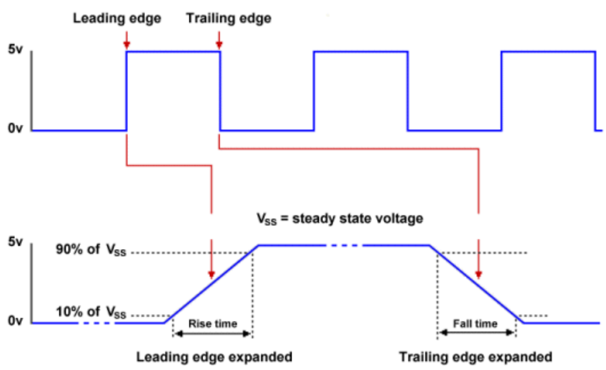

FIGURE 1.2: Digital Signal

1.2 Bandlimited Signals and Sampling Theorem

1.2.1 Bandlimited Signals

A bandlimited signal is a signal in which only some particular band of frequencies are present. The sinusoidal signal for example is band limited but it consists of a single frequency. For a better vision of a bandlimited signal, it's better to transform the signal to the frequency domain and observe its spectrum which must have its amplitude goes to zero for all frequencies beyond some threshold called the cutoff frequency.

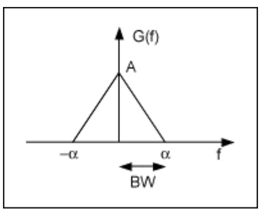

FIGURE 1.3: A spectrum of a bandlimited signal

1.2.2 Sampling Theorem

Mathematically, sampling a continuous signal is just multiplying it by a train of impulses:

$$\sum_{n=-\infty}^{\infty} \delta(t - nTs)$$

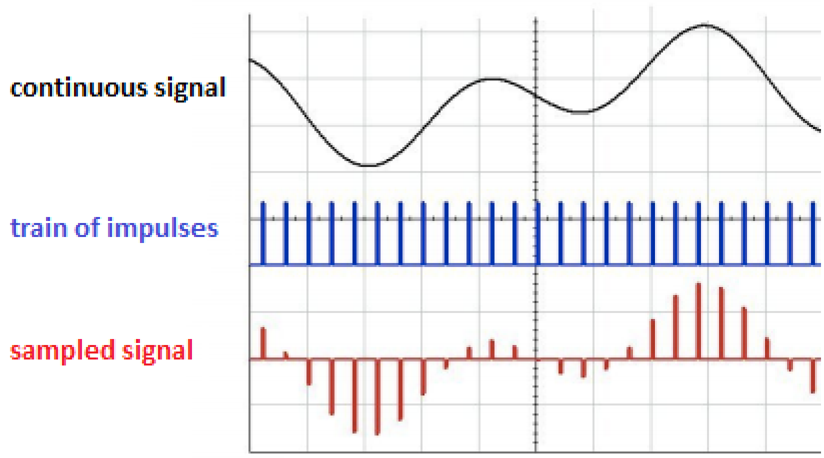

FIGURE 1.4: Sampling a continuous signal

The sampling theorem says that a bandlimited signal can be reconstructed completely if it is sampled at a rate at least twice the maximum frequency component in it. Given a Bandlimited signal $x(t)$ with a spectrum $X(f) = 0$ for $|f| > f_m$, the signal can be recovered from its samples $x(nTs)$ taken at a rate:

$f_s = 1/T_s$ with $f_s \geq 2f_m$. f_s: is the sampling frequency

$x(t)$ can be expressed as:

$$x(t) = \sum_{n=-\infty}^{+\infty} x(nT_s)\,\mathrm{sinc}\left(\frac{t - nT_s}{T_s}\right)$$

1.2.3 Aliasing

Aliasing is a phenomenon where the high frequency components of the sampled signal interfere with each other because of inadequate sampling $f_s < 2f_m$.

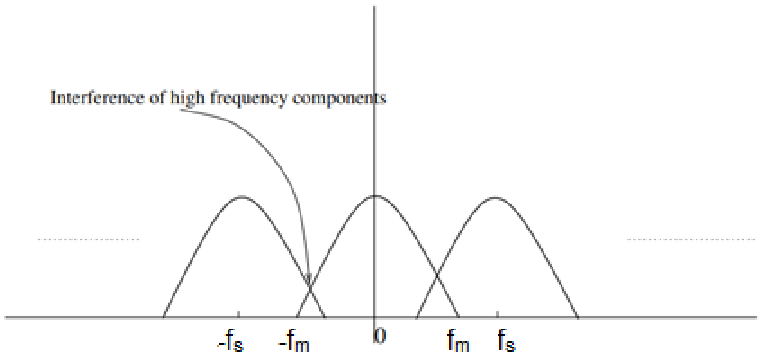

FIGURE 1.5: Aliasing due to inadequate sampling

Aliasing leads to distortion in recovered signal. This is the reason why sampling frequency should be at least twice the bandwidth of the signal.

1.2.4 Oversampling

In practice signal are oversampled, where fs is significantly higher than Nyquist rate to avoid aliasing.

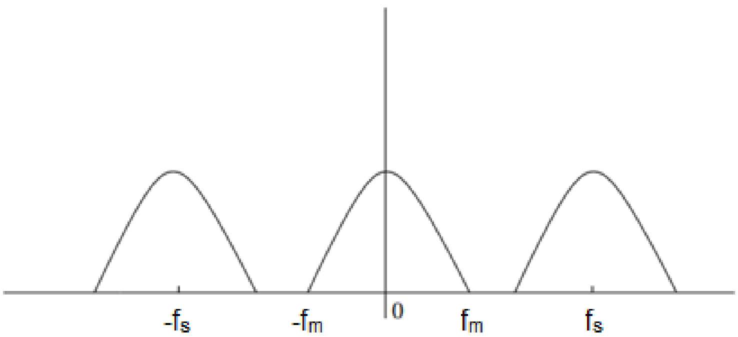

FIGURE 1.6: Oversampled signal-avoids aliasing

1.3 Quantization

The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal.
The following figure shows how an analog signal gets quantized. The blue line represents analog signal while the brown one represents the quantized signal.

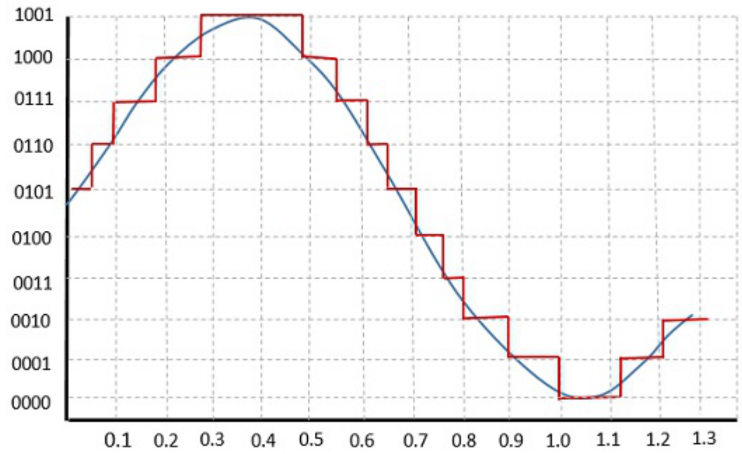

FIGURE 1.7: Quantization of analog signal

Both sampling and quantization result in the loss of information. The quality of a Quantizer output depends upon the number of quantization levels used. The discrete amplitudes of the quantized output are called as representation levels or reconstruction levels. The spacing between the two adjacent representation levels is called a quantum or step-size.

1.3.1 Types Of Quantization

There are two types of Quantization - Uniform Quantization and Non-uniform Quantization. The type of quantization in which the quantization levels are uniformly spaced is termed as a Uniform Quantization. The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a Non-uniform Quantization. There are two types of uniform quantization: Mid-Rise type and Mid-Tread type. The following figures represent the two types of uniform quantization.

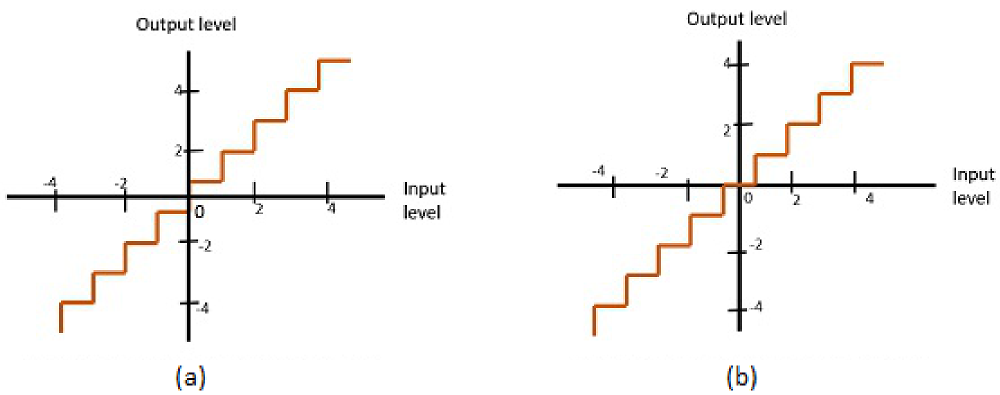

FIGURE 1.8: (a) Mid-Rise type , (b) Mid-Tread type

- The Mid-Rise type is so called because the origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number.
- The Mid-tread type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number.
- Both the mid-rise and mid-tread type of uniform quantizers are symmetric about the origin.

1.3.2 Quantization Error

For any system, during its functioning, there is always a difference in the values of its input and output. The processing of the system results in an error, which is the difference of those values. The difference between an input value and its quantized value is called a Quantization Error.
The following figure illustrates an example for a quantization error, indicating the difference between the original signal and the quantized signal.

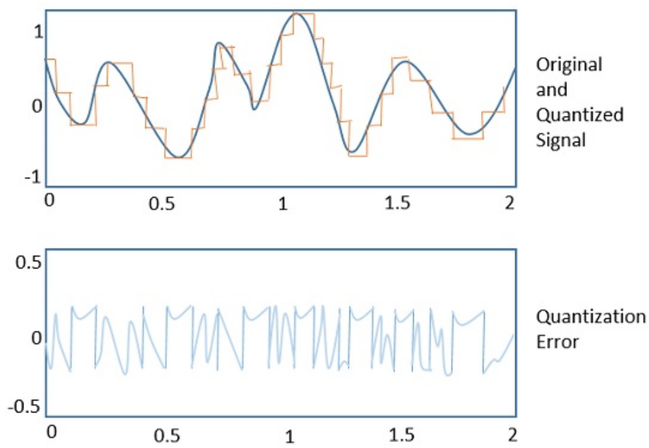

FIGURE 1.9: Quantization Error

1.3.3 Quantization Noise

It is a type of quantization error, which usually occurs in analog audio signal, while quantizing it to digital. For example, in music, the signals keep changing continuously, where regularity is not found in errors. Such errors create a wideband noise called as Quantization Noise.

1.4 Encoding

The encoding operation converts the quantized samples into a form that is more convenient for the purpose of transmission. It is a one-to-one representation of the quantized samples by using code elements or symbols of the required length per sample.

Binary Code

By far, the most popular from the point of view of implementation are the binary codes. With R-binary digits (bits) per sample, we can have 2^R distinct code words and we require $2^R \geq$ (number of quantization levels), so that it is possible for us to maintain a one-to-one relationship between the code words and the quantization levels.
Let us identify the R-bit sequence as $b_R\ b_{R-1} \ldots\ b_3\ b_2\ b_1$. In the natural binary code, this sequence represents a number (or level) N, where:
$N = b_R(2^{R-1}) + b_{R-1}(2^{R-2}) + ... + b_2(2^1) + b_1(2^0)$
Natural binary code results when the codeword symbols or digits are assigned to N, with N listed in an increasing or decreasing (decimal) order; that is, though the quantized samples could be either positive or negative, we simply label the quantized levels decimally without regard to the polarity of the samples.

Decimal	**Binary**		
	b2	b1	b0
0	0	0	0
1	0	0	1
2	0	1	0
3	0	1	1
4	1	0	0
5	1	0	1
6	1	1	0
7	1	1	1

TABLE 1.1: 3-bit binary code

1.5 Pulse Code Modulation and Demodulation

Pulse code modulation (PCM) is a method that is used to convert an analog signal into a digital signal, so that modified analog signal can be transmitted through the digital communication network. PCM is in binary form, so there will be only two possible states high and low (1 and 0). We can also get back our analog signal by demodulation.

1.5.1 Modulation

The Pulse Code Modulation process is done in three steps Sampling, Quantization, and Coding, which are discussed earlier in this chapter.

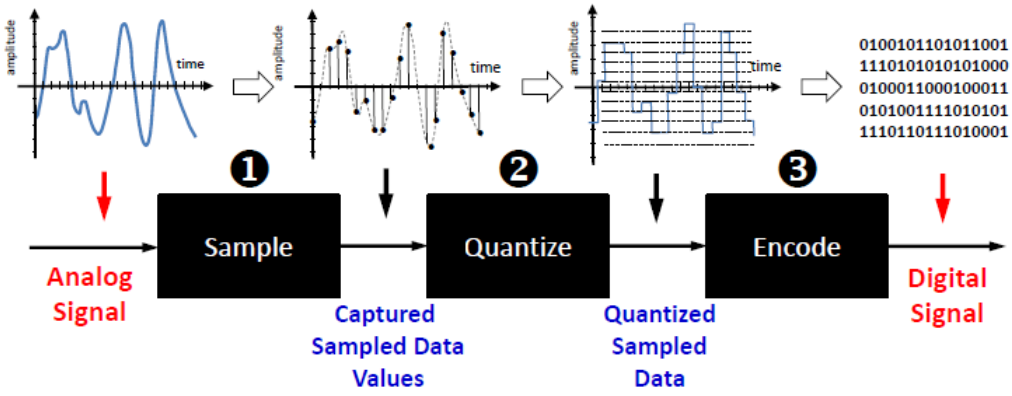

FIGURE 1.10: PCM Process

1.5.2 Demodulation

Pulse Code Demodulation will be doing the same modulation process in reverse. Demodulation starts with decoding process, during transmission the PCM signal will be affected by the noise interference. So, before the PCM signal sends into the PCM demodulator, we have to recover the signal into the original level for that we are using a comparator. The PCM signal is a series pulse wave signal, but for demodulation we need wave to be parallel.
By using a serial to parallel converter, the series pulse wave signal will be converted into a parallel digital signal. After that, the signal will pass through n-bits decoder; it should be a Digital to Analog converter. Decoder recovers the original quantization values of the digital signal. For avoiding unnecessary signals, we utilize a low-pass filter at the final part.

1.6 Data Transmission

Data transmission is the process of sending digital or analog data over a communication medium to one or more computing, network, communication or electronic devices. It enables the transfer and communication of devices in a point-to-point, point-to-multipoint and multipoint-to-multipoint environment.
It is also the physical passing of data over a communication channel which could be copper, wireless or optical fiber. In our project, we are concerned mainly by the Optical Fiber channel.

1.6.1 Optical Fiber

Fiber optics is a medium for carrying information from one point to another in the form of light. Unlike the copper form of transmission, fiber optics is not electrical in nature. A basic fiber optic system consists of a transmitting device that converts an electrical signal into a light signal, an optical fiber cable that carries the light, and a receiver that accepts the light signal and converts it back into an electrical signal.

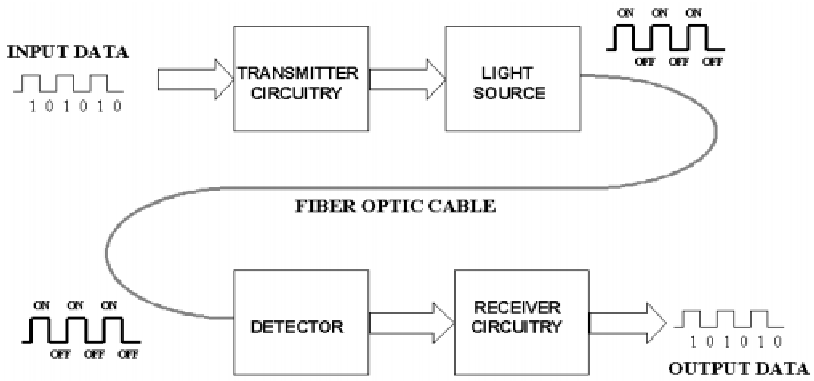

FIGURE 1.11: Basic optical fiber communication system

An optical fiber is a cylindrical dielectric waveguide made of low-loss materials such as silica glass. It has a central core in which the light is guided, embedded in an outer cladding of slightly lower refractive. Light rays incident on the core-cladding boundary at angles greater than the critical angle undergo total internal reflection and are guided through the core without refraction. Rays of greater inclination to the fiber axis lose part of their power into the cladding at each reflection and are not guided.

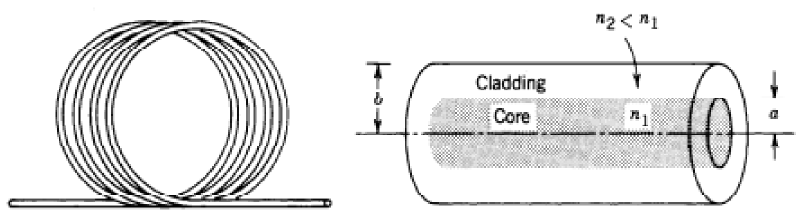

FIGURE 1.12: Optical Fiber

1.6.2 Optical Fiber Transmission Windows

Optical fiber transmission uses wavelengths that are in the near-infrared portion of the spectrum, just above the visible, and thus undetectable to the unaided eye. Typical optical transmission wavelengths are 850 nm, 1310 nm, and 1550 nm. Both lasers and LEDs are used to transmit light through optical fiber.
There are ranges of wavelengths at which the fiber operates best. Each range is known as an operating window. Each window is centered on the typical operational wavelength.

Window	Operating Wavelength
800-900 nm	850 nm
1250-1350 nm	1310 nm
1500-16600 nm	1550 nm

TABLE 1.2: Fiber Optic Transmission Windows

1.6.3 Fiber Optic Loss Calculations

Loss in a system can be expressed as the following:
$Loss = POUT/PIN$
where PIN is the input power to the fiber and POUT is the power available at the output of the fiber. For convenience, fiber optic loss is typically expressed in terms of decibels (dB) and can be calculated using the following equation:
$Loss_{dB} = 10log(POUT/PIN)$

Conclusion

The digitizing process is based on few main steps which are shown and discussed previously in this chapter. By now, we have the basic information that we need to digitize an analog signal and transmit it, and we are ready to go to the next step which is realizing the project.

Chapter 2

Project Realization

Introduction

Till now, we have discussed many important concepts as “the sampling theorem” and “Quantization” that represent the theoretical part which our project is based on. We are now aiming to realize the previous part so that we can observe it and test its results, so we have to find an electronic solution to equivalently represent the formulas using electronic components.

2.1 General Description

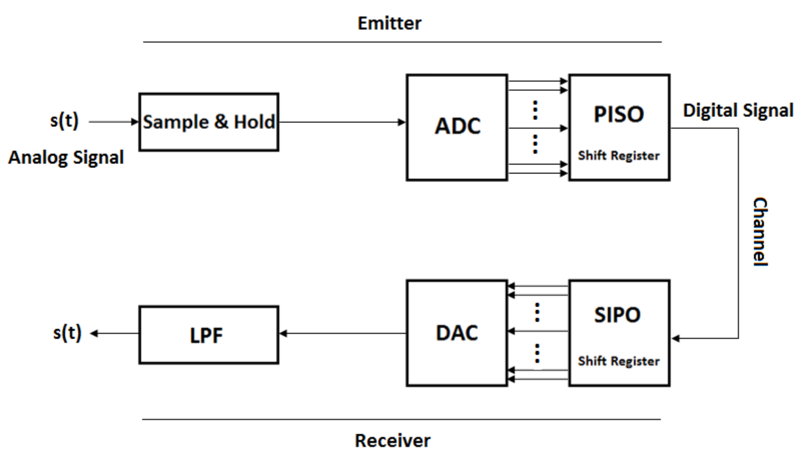

FIGURE 2.1: General Block Diagram

The purpose of our project is to digitize an analog signal, transmit it through a specific channel, then reconstruct the original signal. As the above figure shows, the original signal passes through 3 main steps:

- Emitter, where the analog signal is converted to a digital one to be transmitted.
- Channel, which is the transmission line of the signal.
- Receiver, where the digital signal is received and treated to reconstruct the original analog signal.

2.2 Emitter

In order to digitize our signal and transmit it in a digital form, it must pass through three main steps: sample and hold, analog to digital conversion, and parallel input parallel output shifting.

2.2.1 Sample and Hold

The sample and hold circuit is made using an electronic switch which is derived using a clock signal with a frequency equal to or greater than at least twice the one of the input analog signal. The above condition is obtained from the sampling theorem and it's really important for the reconstruction of the original signal.
The output of the electronic switch is then connected to a resistor and a capacitor to hold the values of the sampled signal for certain time in order for the process of conversion to be performed.

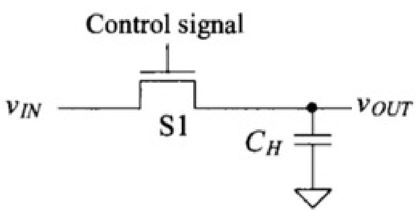

FIGURE 2.2: Sample and Hold

As an electronic switch, we have used the CD4016 chip as which is a quad bilateral switch. Some of its features are:

- Wide supply voltage range.
- Wide range of digital and analog switching.
- Extremely high control input impedance.

We choose one of the 4 switches to work with and we derive it from the appropriate control pin.

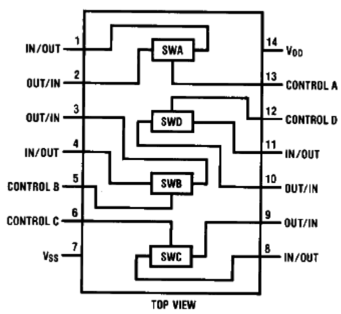

FIGURE 2.3: CD4016 chip

It's connected to a positive voltage from pin 14 and to the ground from pin 7, and the other pins are for the 4 switches which all have 2 pins for the input and the output and one for the control signal as the figure shows.

2.2.2 Analog to Digital Converter (ADC)

An Analog to Digital Converter (ADC) is a very useful feature that converts an analog voltage on a pin to a digital number. By converting from the analog world to the digital world, we can begin to use electronics to interface to the analog world around us.

As a chip, we have used the ADC0805 which is a CMOS 8-bit successive approximation A/D converter. It has the following features:

- Easy interface to all microprocessors.
- On-chip clock generator.
- 0V to 5V analog input voltage range with single 5V supply.

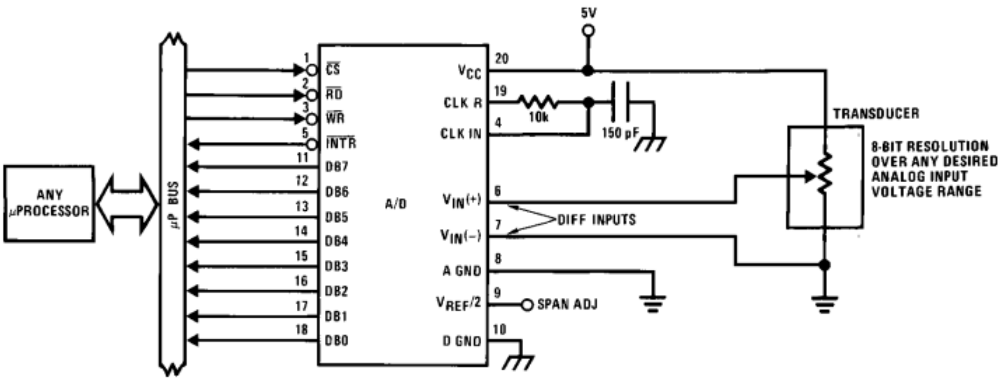

FIGURE 2.4: ADC0805 pins and typical application

The input of the ADC (pin 6) is connected to the output of the sample and hold process to be converted to parallel digital values (pin 11 to pin 18). The other pins are for powering up and controlling the process.

2.2.3 Parallel Input Serial Output (PISO) Shift Register

Since the output of the ADC is parallel digital signals, we need a way to convert it to a serial one to transmit it through one channel.
The parallel-in/ serial-out shift register stores data, shifts it on a clock by clock basis, and delays it by the number of stages times the clock period. In addition, parallel-in/ serial-out really means that we can load data in parallel into all stages before any shifting ever begins. This is a way to convert data from a parallel format to a serial format.
The chip we have used to perform this task is the SN54L/74LS166 which has the following features:

- Synchronous Load
- Direct Overriding Clear
- Parallel to Serial Conversion

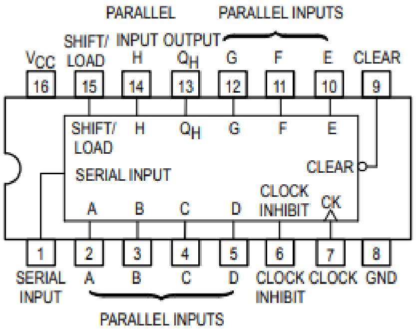

FIGURE 2.5: SN54L/74LS166

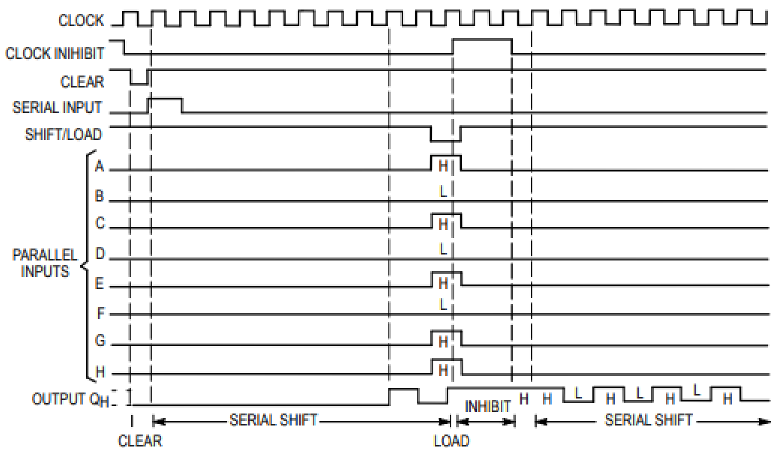

FIGURE 2.6: Timing diagram

We just have to power up the chip and connect the output of the ADC to the parallel input pins and use the other ones to control the process as the previous diagrams show.

2.3 Receiver

In order to reconstruct our analog signal, the signal entering the receiver after transmission must pass through 3 main parts: serial input parallel output shifting, digital to analog conversion, low pass filtering.

2.3.1 Serial Input Parallel Output (SIPO) Shift Register

Since the signal entering the receiver is a serial digital signal, we need a way to convert it to a parallel one to convert it to analog.
A serial-in, parallel-out shift register is similar to the serial-in, serial-out shift register in that it shifts data into internal storage elements and shifts data out at the serial-out, data-out, pin. It is different in that it makes all the internal stages available as outputs. Therefore, a serial-in, parallel-out shift register converts data from serial format to parallel format.
The chip we have used is the SN74LS164 which has the following features:

- Gated serial inputs.
- Fully buffered clock and serial inputs.
- Asynchronous clear.

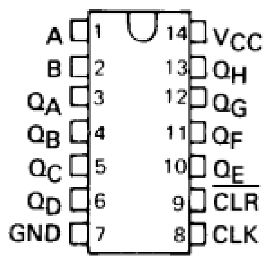

FIGURE 2.7: SN74LS164 chip

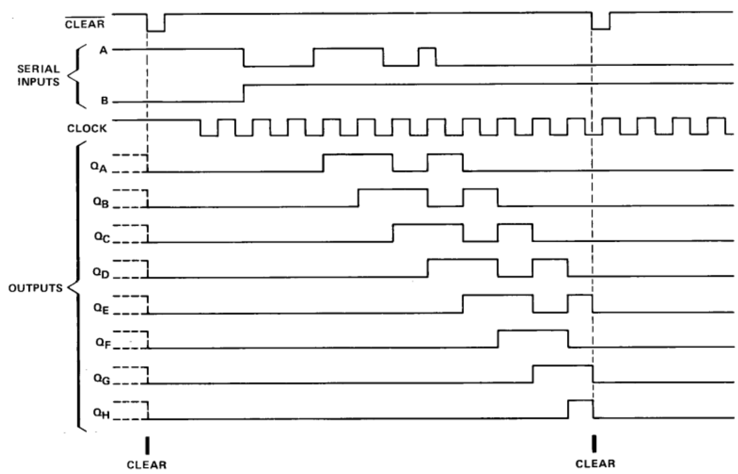

FIGURE 2.8: Time diagram

It is as simple as the PISO shift register, we just have to power up the chip and enter our serial date from A and B at pins 1 and 2 respectively and control it according to the previous figures to get the parallel output.

2.3.2 Digital To Analog Converter

A digital-to-analog converter (DAC) is a circuit that converts digital data (usually binary) into an analog signal. One important specification of a DAC is its resolution. It can be defined by the numbers of bits or its step size.
The chip we have used is the DAC0808 which has the following features:

- Fast settling time.
- Power supply voltage range: ±4.5V to ±18V
- Low power consumption

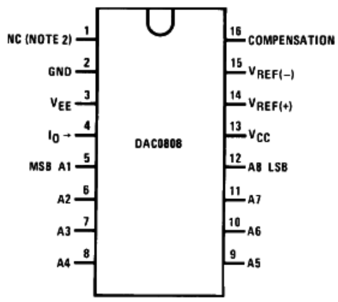

FIGURE 2.9: DAC0808

We power up the chip and connect the outputs of the SIPO shift registers to the inputs of the DAC (from A1 to A8) and get the output from Pin 4 as shown in the following diagram:

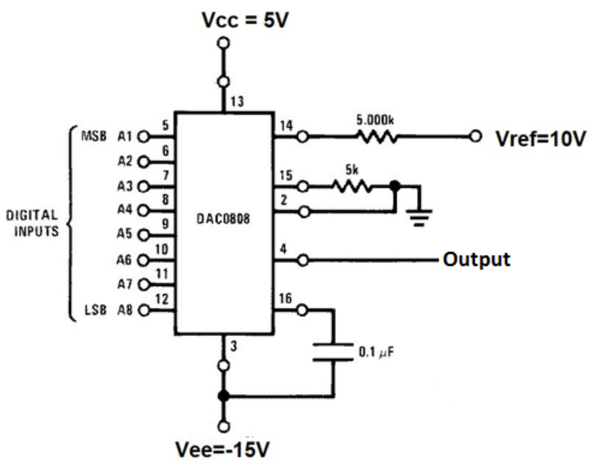

FIGURE 2.10: DAC0808 connection

2.3.3 Low Pass Filter (LPF)

The output of the DAC doesn't really represent the analog signal we want to reconstruct, it is similar to the output of the sample and hold process in the emitter. So,

to get our smooth sinusoidal signal back, we just have to filter the output signal using a Low Pass Filter.
A Low Pass Filter is a circuit that can be designed to modify, reshape or reject all unwanted high frequencies of an electrical signal and accept or pass only those signals wanted by the circuits' designer.
An ideal filter will separate and pass sinusoidal input signals based upon their frequency. In low frequency applications (up to 100kHz), passive filters are generally constructed using simple RC (Resistor-Capacitor) networks, while higher frequency filters (above 100kHz) are usually made from RLC(Resistor-Inductor-Capacitor) components.

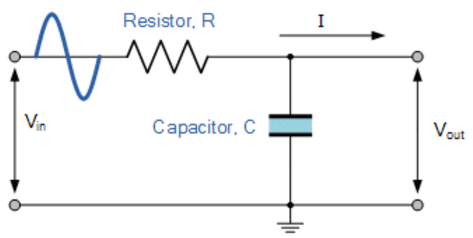

FIGURE 2.11: Low pass filter

2.4 Channel

In order to transmit the output of the emitter to the receiver, we have used the Optical Fiber as a transmission line.
We have used an optical transmitter board to convert the electrical digital signal to an optical one and then transmit it through an optical fiber cable and an optical receiver board to receive the optical signal and converts it back to an electrical digital signal.
The boards' circuits diagrams are shown below:

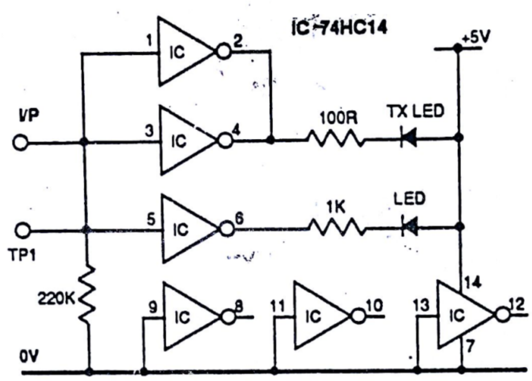

FIGURE 2.12: Optical transmitter board circuit diagram

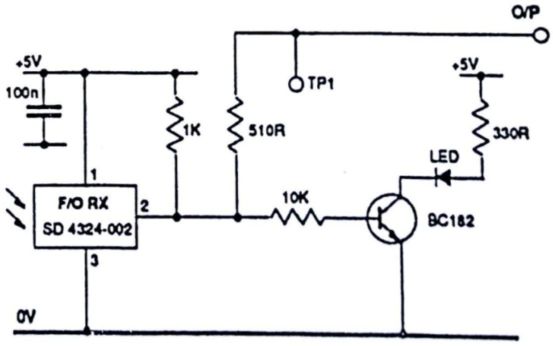

FIGURE 2.13: Optical receiver board circuit diagram

Both transmitter and receiver boards must be powered up using 2 power supply boards which have a circuit diagram as follows:

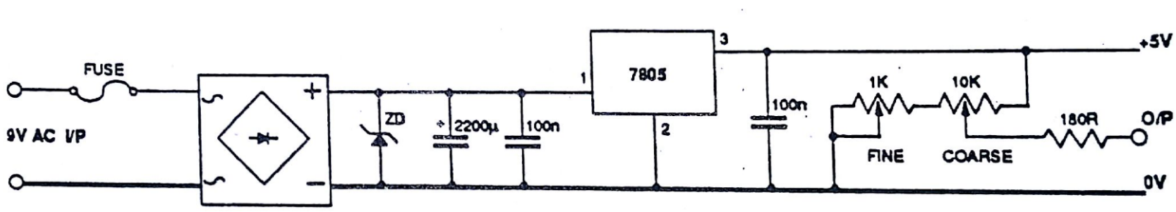

FIGURE 2.14: Power supply board circuit diagram

2.5 Clock Generator

In our project, we need a clock generator to derive the chips such as the CD4016 which is used in sample and hold process, and also the registers and the ADC chip. In order to generate the clocks, we have used the NE555 timer which is connected as follows:

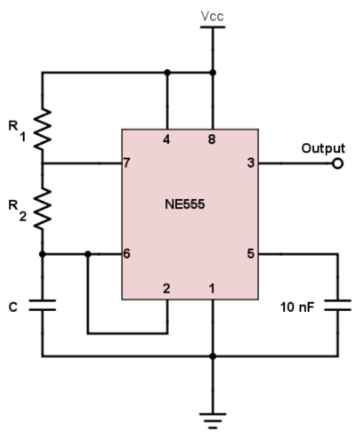

FIGURE 2.15: Clock generator using NE555

We choose the values of R1, R2, and C according to the frequency we need. The frequency of the output is computed as follows:
$f = 1/\{(R-1+2R_2)\times C\}$

Conclusion

In this chapter, we have introduced our electronic propositions to realize our project. We have shown the general diagram of our circuit, its main parts, and all the chips and boards needed to digitize an analog signal, transmit it, and finally recover it.

Chapter 3

Implementation and Results

Introduction

In this chapter, we are going to introduce our implementation of the project showing all the main parts. Also, we are going to show the result of each part, the problems we have faced and our suggested solutions to be done as a future work.

3.1 General Overview

As shown in the second chapter, we have used many chips to realize our project which is divided into three parts: emitter, channel and receiver. The figure below illustrates the full implementation of our circuit.

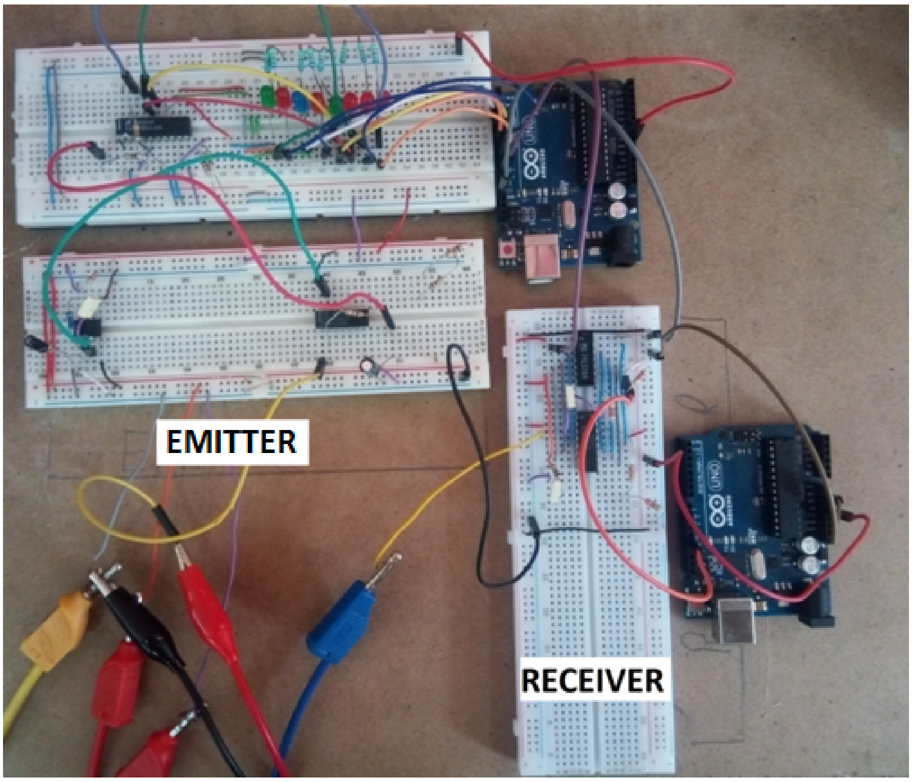

FIGURE 3.1: General overview of the implementation

We are aiming to sample our input signal into 8 portions to get an 8-bit digital signal at the output of the emitter. Then, we are going to transmit this signal and try to recover the original one at the receiver.

3.2 Emitter

In this portion of the circuit, we first have injected our signal, which has 800 mV peak to peak as an amplitude and 1.87 Hz as a frequency, to the CD4016 chip, which is derived by a clock of 15 Hz (approximately 8 times the one of the input signal) generated by the NE555 chip, with a resistor and a capacitor to sample and hold the input signal. The input and output signals of this part is shown in the figures below.

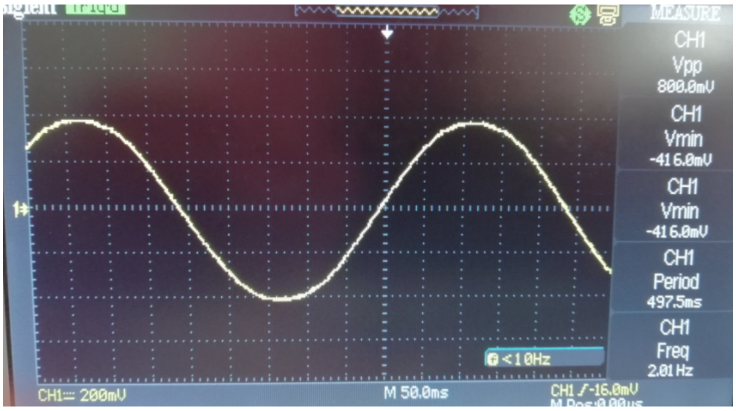

FIGURE 3.2: The input signal

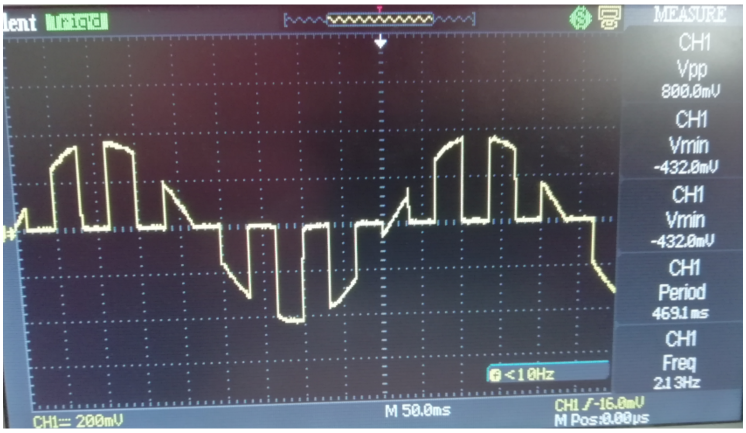

FIGURE 3.3: The output of the CD4016 chip

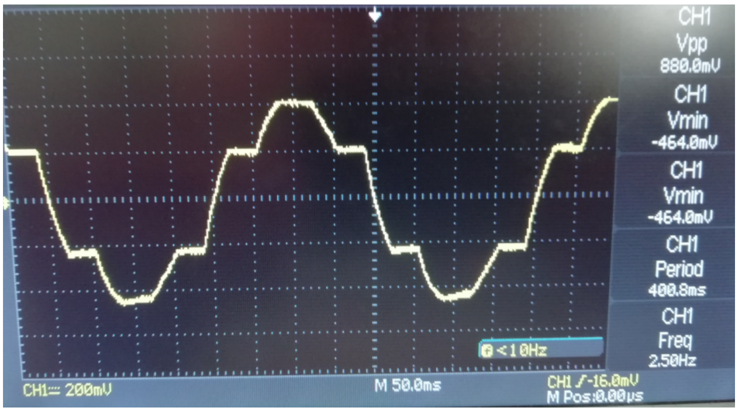

FIGURE 3.4: The output of the sample and hold process

As we can see from the previous figures, the output in general is good and can be used in the next step of digitizing the signal. We can also observe the instability of the signal frequency which is increasing with time. This previous error is caused by the function generator itself and will cause some trouble for us in order to get an 8-bit signal at the output of the emitter.

The next step is to take the output of the sample and hold process as an input of the ADC0805 chip which is used to convert the signal into eight digital parallel outputs. In order to have a serial digital output we wanted to use the parallel input serial output shift register chip SN54L/74LS166. Unfortunately, the chip didn't work because of its bad state, so we had to use the ATMEGA chip in an ARDUINO board to perform this task as show in the figure 3.1. In order to observe the output bits before converting them into serial signal, we have added 8 LEDs such that the most right LED represent the LSB and the most left one represent the MSB. The results of this process are shown in the figures below.

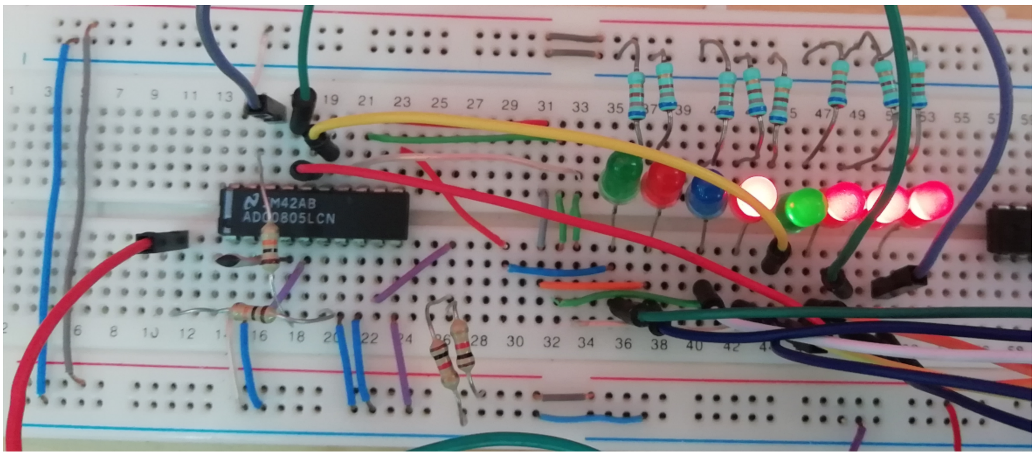

FIGURE 3.5: 8 LEDs representing the output bits of the ADC0805 at an instant of time

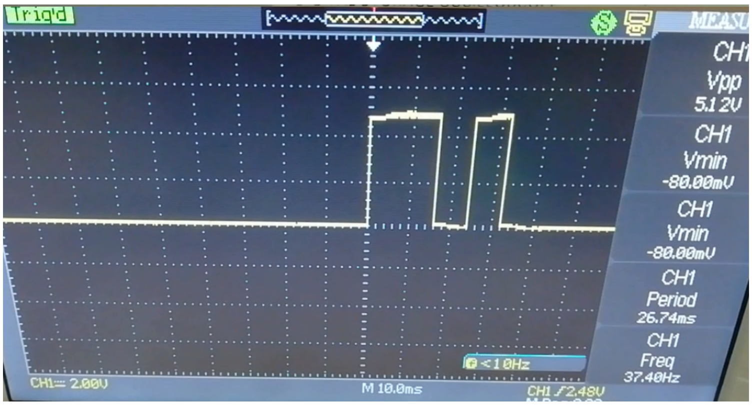

FIGURE 3.6: The output waveform of the emitter at an instant of time

3.3 Receiver

In this portion of the circuit, we are receiving the transmitted signal and reconstructing from it the original one. For that to be performed, we have to reverse the emitting process, so we have injected the received waveform signal into the serial input parallel output shift register chip SN74LS164. The obtained parallel signals are then injected into the DAC0808 chip which is supposed to convert them into an analog signal similar to the one obtained after the sample and hold process. In order to eliminate unwanted harmonics, we have added a low pass filter at the output of the ADC to reconstruct our smooth original signal. In this process, we have used the ATMEGA chip in an ARDUINO board to generate a clock with frequency of 120 Hz in order to derive the SN74LS164 (120 Hz because the SIPO chip is supposed to work 8 times faster than the PISO chip The output of the receiver which is the reconstructed signal is shown in the figure below.

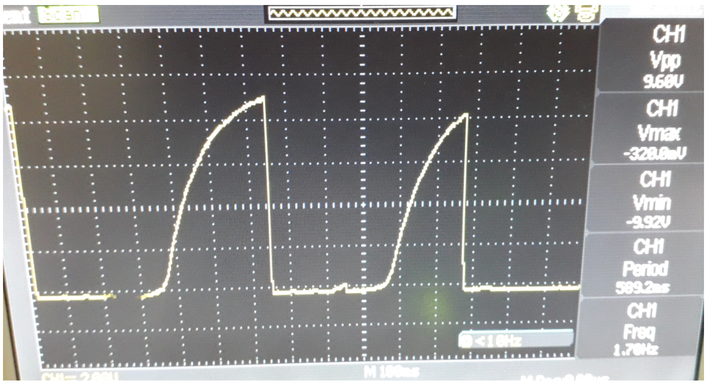

FIGURE 3.7: The reconstructed signal

Observing the previous figure, we can detect many problems at the reconstruction of our signal:

- The receiver reconstructs only less than the half of the original signal.

- There are some distortions in the signal (as we can see, the amplitude is varying).
- The signal is amplified comparing to the original signal.

Our analyses of these problems are as follows:

- The ADC0805 doesn't convert the negative values of the input to digital values. One of the solutions is to add an offset to the input, but this will damage the CD4016 which is limited by 0.5V as maximum positive amplitude. We can also generate a really small signal but it will be distorted by the offset.
- The distortions may be occurred because of the CD4016 which samples the input signals in different rates. Also, the use of the jumpers which are made of mixed materials can cause these distortions.
- The DAC0808 chip is powered by 5V as a positive voltage and -12 V as a negative one, and this caused the amplification of the signal and adding an offset. Also, the state of the chip may cause a wrong reconstruction of the signal comparing to the work of the ADC chip.

We suggest as solutions for the above problems in a future work the following:

- Replacing the CD4016 chip by another modern one that has higher limits for its input signals' amplitude.
- Connecting all our circuit using copper wires.
- To reconstruct our signal as it is at the output, we have to either replace the DAC0808 chip by another that does not have the problem of amplification, or we just add an attenuator at the output of the DAC.

3.4 Channel

As it was planned, we wanted to use optical fiber boards and cable for the transmission of the data from the emitter to the receiver. Unfortunately, the receiver board did not work and keeps showing a random signal as an output. The boards that we wanted to use to transmit our data optically are shown in the figure below.

FIGURE 3.8: The optical fiber boards

Our analysis of the problem of the receiver board is as follows:

- There is a short circuit in the receiving portion that prevents receiving the transmitted signal.
- The light detector in the board may be damaged.

Instead of using the optical fiber, we have used a copper wire as a channel to test our circuit.

Conclusion

Our implementation doesn't work perfectly, but we can observe that the analog signal is sampled and held, converted to binary and then transmitted. Our main problem is in the reconstruction of the original signal which is not well performed and can be enhanced by our suggested solutions as a future work.

General Conclusion

Digital transmission has become widely used all over the world because of its advantages comparing to the analog one. Nowadays, the need to transmit data safely and in a reliable way without consuming much bandwidth and power has a critical influence on experts' decision to choose digital transmission over the analog one.

In our project, we have been working on digitizing a simple analog signal (a sinusoidal signal), transmit it and then reconstruct it at the receiver. In order to achieve our objective, a theoretical part of digitizing process and data transmission has been discussed to clarify the basics of our work. After that, and to realize our project, we have designed and implemented a circuit to do sampling and holding, analog to digital conversion, data transmission, and signal reconstruction. After observing the results of the different parts of our circuit, we have got some satisfactory results especially the ones of the sample and hold portion. Other results were not as expected, especially the reconstructed signal which represented less than a half of the original one with some observable noise, and that is because of the fact that the ADC chip doesn't convert the negative values to the digital form and the limits of the sampling chip CD4016 that cannot be exceeded and that prevent us from adding an offset to the signal. Also, we were supposed to use an optical fiber as a transmission line using some specific optical transmission and receiving boards. Unfortunately, the receiving board did not work, so we had to test our circuit using a copper wire.

The fact that we did not make use of a microcontroller in one of our main portions of the circuit makes our project quite advantageous, so that it was so beneficial to apply our background knowledge that we have learnt from Digital Systems, Process Control, and Communication Principles courses in reality. However, since microcontrollers become nowadays the dominant tool that is used for building almost all circuits, implementing a circuit to perform a specific task becomes rarely carried and less preferable. Moreover, during this project we had the chance to be familiar with the optical fiber transmission line and its general features.

To conclude with, we suggest a further enhancement for this project by finding solutions to the problems we have faced and then adding a microprocessor to encrypt the digital data before transmission to provide a safe transmission system to the users.

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