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Тема: «Амплітудна демодуляція цифрових сигналів змінної частоти»

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NATIONAL AVIATION UNIVERSITY

Faculty of Aeronautics, Electronics and Telecommunications

Aerospace Control Systems Department

APPROVED FOR DEFENCE

Head of the ACS Department

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“____” ______________ 2023

QUALIFICATION PAPER

(EXPLANATORY NOTE)

FOR THE ACADEMIC DEGREE OF BACHELOR

Subject: “Amplitude demodulation of variable frequency digital signals”

Submitted by: student of group CS-402 Kateryna GORSHKOVA

Supervised: Valeriy CHIKOVANI

Norm control: Valeriy CHIKOVANI

Kyiv 2023

НАЦІОНАЛЬНИЙ АВІАЦІЙНИЙ УНІВЕРСИТЕТ

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Спеціальність 151 Автоматизація та комп’ютерно-інтегровані технології

ЗАТВЕРДЖУЮ

Завідувач кафедри АКСУ

_____________ Юрій МЕЛЬНИК

«_____»___________2023 р.

ЗАВДАННЯ

на виконання кваліфікаційної роботи

Горшкової Катерини Андріївни

1. **Тема кваліфікаційної роботи** «Амплітудна демодуляція цифрових сигналів змінної частоти».

Затверджено наказом ректора від «13» квітня 2023 р. №507/ст.

2. **Термін виконання проекту (роботи): 10.03.23 – 10.06.23**

3. **Вихідні дані до проекту (роботи):** Амплітуда модульованого сигналу до 50 В, частота несучий до 100 Гц, дисперсія шуму до 10 B^2

4. **Зміст пояснювальної записки (перелік питань, що підлягають розробці):** Розділ 1. Мета та сфера дослідження; Розділ 2. Застосування синтезованого сигналу; Розділ 3. Амплітудна демодуляція сигналів змінної частоти; Розділ 4 Результати Симуляції; Висновки; Список літератури;

5. **Перелік обов’язкового графічного матеріалу:** Графіки результатів моделювання та розрахунків. Матеріали презентації в Power Point.

6. Календарний план-графік

№ пор.	Завдання	Термін виконання	Відмітка про виконання
1	Отримання завдання	10.03.2023 – 11.03.2023	Виконано
2	Формування мети та основних завдань дослідження	12.03.2023 – 15.03.2023	Виконано
3	Аналіз існуючих методів	15.03.2023 – 17.03.2023	Виконано
4	Теоретичний розгляд рішення проблеми	17.03.2023 – 10.04.2023	Виконано
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NATIONAL AVIATION UNIVERSITY

Faculty of Aeronautics, Electronics and Telecommunications

Department of Aerospace Control Systems

Specialty 151 Automation and computer-integrated technologies

APPROVED BY

Head of the ACS Department

_____________ Yurii MELNYK

“_____”_______________2023

Qualification Paper Assignment for Graduate Student

Gorshkova Kateryna

1. **Theme of the project:** «Amplitude demodulation of variable frequency digital signal»

Approved by rector order from «13» April 2023 No 507/st

2. **The term of the project (work): 10.03.23 – 10.06.23**

3. **Output data to the project (work):** The amplitude of the modulated signal is up to 50 V, the carrier frequency is up to 100 Hz, the noise dispersion is up to 10 V^2 .

4. **Contents of the explanatory note (list of questions to be developed):** Section 1. The aim and scope of implementation; Section 2. Application to the synthesized signal; Section 3. The amplitude demodulation of the variable frequency signals; Section 4. Simulation results; Conclusions; List of references.

5. **The list of mandatory illustrations:** Graphs of simulation and calculation results. Presentation materials in Power Point.

6. Timetable:

№	Assignment	Dates of completion	Completion mark
1	Task receiving	10.03.2023 – 11.03.2023	Executed
2	Purpose formation and describing the main research tasks	12.03.2023 – 15.03.2023	Executed
3	Analysis of existing methods	15.03.2023 – 17.03.2023	Executed
4	Theoretical consideration of the problem solution	17.03.2023 – 10.04.2023	Executed
5	Developing a structure and solving demodulation error	11.04.2023 – 15.05.2023	Executed
6	Making an explanatory note	16.05.2023 – 25.05.2023	Executed
7	Preparation of presentation and handouts	26.05.2023 – 07.06.2023	Executed

7. Date of task receiving: «10» March 2023y.

Diploma thesis supervisor__________________Valeriy CHIKOVANI.

(signature)

Issued task accepted_____________________Kateryna GORSHKOVA.

(signature)

ABSTRACT

Text part of the work: 66 pages, 36 graphs, 3 tables, 1 page appendix.

The object of study is the methods of amplitude demodulation suitable for sensors' applications.

The subject of research is synchronous demodulation errors under modulated frequency change.

Purpose: to study the principles of amplitude demodulation in frequency digital signals.

Research methods: Theoretical analysis and experimental studies using simulation in the MATLAB environment.

Summary of the work: This work provides an overview of amplitude demodulation methods. Examples of amplitude modulation and demodulation signals for various purposes, including sensor signals, are given. The Simulink block diagram of signal modulation and demodulation is built. The research was conducted to study the demodulation error of modulated signals under the influence of variable amplitude and frequency, along with the presence of noise. The method of synchronous demodulation was used in this work.

Keywords: modulation, demodulation, synchronous demodulation, demodulation errors, Simulink block diagram.

РЕФЕРАТ

Текстова частина роботи: 66 сторінок, 36 графіків, 3 таблиці, 1 сторінка Додаток.

Об'єктом дослідження: методи амплітудної демодуляції, придатні для сенсорних застосувань.

Предметом дослідження: похибки синхронної демодуляції при модульованій зміною частоти.

Мета роботи: дослідження принципів амплітудної демодуляції частотних цифрових сигналів.

Методи дослідження: Теоретичний аналіз та експериментальні дослідження з використанням моделювання в середовищі MATLAB.

Короткий зміст роботи: В роботі наведено огляд методів амплітудної демодуляції. Наведено приклади амплітудної модуляції та демодуляції сигналів різного призначення, в тому числі сенсорних сигналів. Побудовано блок-схему Simulink модуляції та демодуляції сигналів. Проведено дослідження похибки демодуляції модульованих сигналів під впливом змінних амплітуди та частоти, а також присутності шуму. В роботі використано метод синхронної демодуляції.

Ключові слова: модуляція, демодуляція, синхронна демодуляція, похибки демодуляції, блок схема Simulink..

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GLOSSARY

AM – Amplitude modulation

FM – Frequency modulation

RSM – root mean square

LS – least squares

PTC – Parametric Technology Corporation

APM – amplitude pulse modulation

FPM – frequency-pulse modulation

PWM – Pulse Width Modulation

FOG – fiber optic gyroscopes

MEMS – Micro-Electro-Mechanical Systems

PLL – phase-locked loops

INTRODUCTION

Undoubtedly, we present an introductory discourse for a diploma thesis concerning the amplitude demodulation of variable frequency digital signals. The present study aims to provide a comprehensive analysis of the topic at hand. This introductory section serves to offer a glimpse into the study's overall objective and intended approach, in an effort to articulate the relevance and importance of the topic to the reader. By contextualizing the subject matter within existing scholarly literature and established theoretical frameworks, this study seeks to contribute new insights and perspectives to the discourse. In contemporary times, in light of advanced communication systems and the emergence of digital signal processing, precise demodulation of variable frequency digital signals is of utmost significance. Amplitude modulation (AM) is a modulation technique that is extensively employed for the transmission of information through a carrier signal. The process entails modulation of the amplitude of the carrier signal in accordance with the underlying digital data intended for transmission. Nonetheless, the demodulation of signals featuring varying carrier frequencies presents noteworthy complexities that arise from the dynamic nature of the carrier frequency. The process of demodulation is a vital aspect in the retrieval of the original digital information from the modulated signal. The process of amplitude demodulation involves the extraction of amplitude variations imposed by the modulation process, with the specific aim of recovering the underlying digital signal. The signal that has been demodulated is a true and accurate indication of the initial digital information, thereby facilitating subsequent examination, interpretation, and application. The objective of this thesis is to explore and advance methods pertaining to the process of amplitude demodulation of digital signals that exhibit variable frequency characteristics. The primary aim is to develop algorithmic solutions that are efficient and dependable in accurately demodulating signals characterized by fluctuating carrier frequencies, notwithstanding the presence of noise, interference, and other transmission obstacles. The current investigation aims to undertake a comprehensive examination of prevailing techniques for amplitude demodulation while taking into account their limitations when applied to digital signals of fluctuating frequencies. The present investigation aims to evaluate the efficacy, intricacy, and suitability of diverse demodulation methods such as envelope detection, synchronous detection, square law detection, and others with respect to their ability to handle signals with varying frequencies. This study aims to explore the intricacies surrounding the conception and execution of innovative algorithms distinctly customized for demodulating digital signals of variable frequencies. The present study seeks to capitalize on signal processing techniques of a sophisticated nature by leveraging adaptive filtering, frequency tracking, and noise suppression to intensify demodulation accuracy and resilience. The efficacy of the developed algorithms will be verified through comprehensive simulations and practical experiments utilizing both synthetic and acquired variable frequency digital signals. The evaluation of the effectiveness of the proposed methods in comparison to existing techniques will be conducted through the analysis of performance metrics such as demodulation error rate, signal-to-noise ratio, and computational efficiency. It is anticipated that the results obtained from this diploma thesis will make a notable contribution to the domain of amplitude demodulation with respect to digital signals that exhibit variable frequency. The research outcomes and algorithms formulated within this study possess considerable potential for practical utilization in a range of domains comprising wireless communication systems, sensor networks, digital audio broadcasting, and software-defined radios.

In summary, the overarching objective of this thesis is to confront the obstacles that arise in the context of amplitude demodulation of digital signals with modulating frequencies that vary over time, and to put forth novel approaches that can heighten the degree of precision and dependability of the demodulation procedure. The present study aims to enhance the current demodulation techniques, thereby making a notable contribution towards the uninterrupted transmission and extraction of digital information from signals possessing carrier frequencies that are subject to dynamic fluctuations.

SECTION 1
THE AIM AND SCOPE OF IMPLEMENTATION

1.1 PURPOSE AND SCOPE OF APPLICATION

One of the most patterns within the advancement of arrange advances is the transmission of both discrete and analog information over a single organize. The sources of discrete information are computers and other computing gadgets, whereas the sources of analog information are phones, video cameras, sound and video playback hardware. At the early stages of tackling this issue, all sorts of information were transmitted in analog shape in nearby systems, whereas discrete computer information were changed over to analog shape utilizing modems.

With the evolution of methods for acquiring and relaying analog information, it became evident that transmitting such information in its analog form does not enhance the quality of the data received at the receiving end of the line, especially if it has been significantly distorted during transmission. The analog flag, in and of itself, does not provide indicative evidence of a waveform being altered or aid in remedying such alterations, as there exists a possibility for waveform divergence, inclusive of the waveform received by the intended recipient. Elevating the quality of lines, with a particular emphasis on regional ones, necessitates substantial efforts and investments. Thus, the traditional approach of analog technology in recording and transmitting sound and images has been replaced by modern advanced technologies. This approach utilizes the discrete alteration of the yield of sustained analog configurations, commonly referred to as «discrete tweak».

The main advantages of amplitude demodulation are:

1. Simplicity: The process of amplitude demodulation is considered relatively straightforward in comparison to other demodulation methods employed for different modulation schemes. The utilization of fewer components and simpler circuitry often results in heightened cost-effectiveness and simplification of implementation.

2. Compatibility: The demodulation of amplitude modulation (AM) signals can be readily achieved through the utilization of fundamental and easily attainable components. The aforementioned characteristic renders the technology suitable for a broad spectrum of recipient devices, encompassing receivers of lower cost. The aforementioned modulation technique enables seamless retrieval and isolation of the unadulterated modulating signal from the amplitude modulated carrier.

3. Robustness: The process of AM demodulation exhibits a considerable degree of resiliency against specific types of noise and interference. The demodulation process is inherently straightforward, thereby facilitating the retrieval of the initial signal even in the presence of distortions or ambient noise in the received amplitude-modulated (AM) signal.

4. Efficiency in Power Usage: The process of AM demodulation is characterized by its ability to operate without any supplementary power-consuming elements or intricate algorithms, thus promoting the efficient utilization of power. The demodulation procedure does not necessitate noteworthy power consumption, thus rendering it appropriate for low-power implementations.

5. Signal Reconstruction: The technique of amplitude demodulation enables precise restoration of the initial modulating signal. Provided that the carrier signal and modulation index are discerned, the demodulation procedure can accurately recreate the source signal with minimal degradation or distortion.

6. Compatibility with Analog Signals: The process of AM demodulation is primarily intended for the purpose of demodulating analog signals, and it has demonstrated a high degree of proficiency in extracting the initial analog information. This renders it appropriate for usage in scenarios where analog signals are pervasive, such as in traditional broadcasting and audio transmission applications.

The disadvantages of this modulation are:

1. Sensitivity to Amplitude Variations: The process of AM (amplitude modulation) demodulation is known to exhibit a high level of sensitivity toward alterations in the amplitude of the signal being received. Anomalies in the intensity of signals caused by factors such as signal attenuation, signal interference, or signal noise have the potential to significantly impact the integrity and precision of the demodulated signal.

2. Limited Noise Immunity: The process of AM (amplitude modulation) demodulation is known to exhibit a high level of sensitivity toward alterations in the amplitude of the signal being received. Anomalies in the intensity of signals caused by factors such as signal attenuation, signal interference, or signal noise have the potential to significantly impact the integrity and precision of the demodulated signal.

3. Inability to Recover Carrier Phase Information: The process of AM demodulation solely extracts the modulating signal's envelope without restoring the original carrier phase information. The foregoing constraint curtails its efficacy in demodulating signals predicated on phase data transmission, such as phase-shift keying (PSK) or quadrature amplitude modulation (QAM).

4. Inefficient Bandwidth Utilization: The process of amplitude modulation (AM) demodulation inherently demonstrates suboptimal utilization of the bandwidth at hand. The utilization of amplitude modulation (AM) necessitates a broader range of frequency allocation in contrast to alternative modulation techniques, such as frequency modulation (FM) or digital modulation, culminating in reduced spectral efficiency.

5. Limited Demodulation Accuracy: The process of AM demodulation may result in signal distortion or inaccuracies, which can be attributed to deficiencies in the applied demodulation techniques. The occurrence of collinearities within the demodulation circuitry or deviations in the carrier frequency may result in signal distortions, thereby influencing the precise reproduction of the demodulated signal.

6. Incompatibility with Digital Signals: The AM demodulation technique is essentially intended to demodulate analog signals. Direct demodulation of digital

signals is deemed unsuitable. Extra techniques, such as pulse modulation or digital demodulation schemes, would be necessary to derive digital data from an amplitude modulation (AM) signal in an academic context.

In radio engineering, amplitude demodulation is used to transmit information over a distance in radio broadcasting, acoustic localization, etc. For example, in radio broadcasting, sound vibrations are converted into a low-frequency electrical signal (modulating signal) that changes (modulates) the amplitude of a high-frequency (carrier) signal. In the resulting modulated radio signal, the amplitude changes in accordance with the strength of the audio signal.

The process of demodulation involves the retrieval of data from a carrier signal, and the resultant output of a demodulator may take the form of either a video signal or a digital signal, represented in binary format. Nevertheless, the most prevalent utilization entails audio.

Similar to how modulation is employed for radio transmission, demodulation serves the identical objective. Nevertheless, it is crucial to note that several alternative systems employ demodulation. One widely cited instance is the modem, which functions as both a modulator and demodulator. The purpose of this technology is to extract digital data from a modulated signal.

There are various terms for demodulation, such as diode detector, synchronous detector, integral, detector, etc. However, demodulation is much more commonly used, especially when extracting signals from a carrier wave.

Amplitude demodulation is widely used in:

- AM Broadcasting;
- Two-Way Radio Communication;
- Radio Navigation;
- Public Address Systems;
- Audio Transmission;
- Industrial and Scientific Equipment.

1.2 RESEARCH OBJECT

The concept of a function appeared in the eighteenth century in the theory of mathematics as a certain dependence of any value «y» on another value - the independent variable «x» with the mathematical representation of such a dependence in the form $y(x)$. Soon, the mathematics of functions became the main basis of the theory of all natural and technical sciences. Functional mathematics became especially important in audio, video, and communication technologies, where signals were called time functions of the form $s(t)$, $v(f)$, etc., which were used to describe the process of information transmission. In technical fields, the term «signal» is often used in a generalized way without adherence to strict terminology. This term is used in different ways. For example, it can be used to understand the technical means of transmitting, converting and using information - an electrical, magnetic, optical signal; or to describe a physical process that is the material embodiment of the message information - a change in any environmental parameter (voltage, frequency, power of electromagnetic oscillations, light intensity, etc.) in time, space or depending on changes in the value of any other arguments (independent variables); or to understand the meaning of a certain physical state or process, such as a traffic light, a communication system, etc. All these concepts are united by the ultimate purpose of signals.

In this sense, signals are certain information, messages, information about any processes, states, or physical quantities of material objects in the world, expressed in a form convenient for the transmission, processing, storage, and use of this information. The term «signal» is often identified with the terms «data» or «information». Indeed, these concepts are interdependent and do not exist without each other, but belong to different categories. The concept of information has many definitions, the broadest (information is a formalized the real world) to the practical (information and data that are the object of storage, transmission, transformation, perception and management). Nowadays, world science is increasingly inclined to believe that this information, along with matter and energy, belongs to the main philosophical categories of science and is one of the properties of the objective world, although not very specific. As for data, it is a set of facts, results of observations, measurements of any objects, phenomena or processes of the material world, which are presented in a formalized form - quantitative or qualitative. This is not information itself, but only its attribute - raw material for obtaining information through appropriate processing and interpretation (interpretation). The term «signal» in the world practice is a general term to describe the form of data representation resulting from certain measurements of the parameters of the object of study in the form of a sequence of scalar values (analog, numerical, graphical) depending on changes in any variable (energy, temperature, spatial coordinates, etc.). Nevertheless, the term «signal» in the narrow sense of the word will mean a certain orderly reflection of changes in the physical state of any object - a material carrier of the signal, certain data about the nature of the change in space or time. Since the data contain information about the main purpose of the object under study, as well as about various related factors and various interferences, in a broad sense of the word they can be considered a signal carrier of general measurement information.

It should be noted that the material form of signal carriers (mechanical, electrical, magnetic, acoustic, optical and any other), as well as the form of reflection in certain physical parameters, are not important. An information parameter of a signal can be any parameter of a signal carrier functionally related to the value of information data. In the general sense, a signal is the dependence of one quantity on another, and from a mathematical point of view, it is a function. A signal in the general sense is the dependence of one quantity on another, and in mathematical terms, a function.

The most common is the representation of signals in electrical form as a voltage versus time $U(t)$. A signal is an information function that carries a message about the physical properties, state or behavior of any physical system, object or environment, and the purpose of signal processing can be considered to be to obtain certain information reflected in these signals (in short, useful or target information) and to transform this information into a form convenient for perception and further use. When we talk about «analyzing» signals, we are talking not only about purely mathematical transformations, but also about drawing conclusions about the characteristics of the relevant processes and objects based on these transformations.

The purpose of signal analysis is usually:

– determining or estimating the numerical parameters of signals (energy, average power, root mean square, etc.)

– decomposition of the signal into elementary components, comparison of the properties of different signals;

– comparing the degree of proximity, «similarity», «affinity» of different signals, including certain quantitative estimates.

– The mathematical apparatus of signal analysis is quite extensive and is widely used in practice in all fields of science and technology without exception.

1.3 SIMULATION PLATFORMS

The concept of information modeling and its various manifestations presuppose the existence of an information model that functions within a precisely delineated information process. Constructing a model layout is a fundamental principle of physical modeling that cannot be violated. This phenomenon is correlated with incurred expenses related to the procurement of materials and necessitates supplementary, occasionally laborious, investigation.

The distinctive feature of the informational processing method utilized in modeling software platforms lies in the synthesis of the critical attributes of both analog and digital circuits in a virtual format, which are subsequently analyzed with the active involvement of human participants, akin to the traditional process of physical modeling.

An exemption from the norm occurs in cases whereby demonstration models are constructed, primarily for the purpose of advertising, where the anticipated outcome of the «experimentation» is predetermined, thereby rendering any analysis unnecessary. There exists a noteworthy assemblage of computer programs that constitute comprehensive platforms for the purpose of modeling electronic devices.

The Simulink interactive computer modeling system holds a significant position within the virtual laboratory of students, engineers, and researchers. The Simulink software possesses ample capabilities to effectively compute, assess, and devise digital signals and systems.

1.3.1 MATHCAD

MathCAD is a powerful and flexible platform that offers a consistent solution for addressing diverse problems in scientific and technological spheres, financial and economic domains, and mathematical and statistical disciplines. The utilization of this approach is applicable in all domains where mathematical techniques are employed. MathCAD was initially formulated and authored by Allen Razdov, one of the co-founders of Mathsoft Inc. , using his intellectual prowess and expertise in the field. The software in question, which was first introduced in 1986 and received its most recent update in 2015, became a subsidiary of PTC (Parametric Technology Corporation) in 2006.

MathCAD provides a convenient graphical milieu for carrying out numerous mathematical computations and documenting their execution in adherence to established norms. This approach facilitates the development of certified payment instruments for corporate and industry-based applications in diverse fields of science and technology, thereby enabling a consistent methodology for all establishments within a corporation or industry. The primary distinction of MathCAD, along with its notable benefit, lies in its graphical mode for inputting expressions, as opposed to a text-based approach.

MathCAD can be regarded as a software tool that facilitates programming tasks without requiring actual programming knowledge or expertise. In essence, it presents an accessible programming platform that eliminates the need for conventional coding approaches. It is plausible that an individual can comprehend the written code of a program with ease, without necessitating any supplementary information.

1.3.2 MATLAB

MATLAB is a high-level language for technical calculations, an interactive environment for developing algorithms, and a modern data analysis tool.

MATLAB in comparison with traditional programming languages (C/C ++, Java, Pascal, FORTRAN) allows to reduce the time for solving typical tasks by an order of magnitude and greatly simplifies the development of new algorithms.

At the same time, special attention was paid to increasing the speed of calculations and adapting the system to solve various user tasks.

MATLAB provides the user with mathematical and computational systems, as well as a large number (several hundred) of data analysis functions that cover almost all areas of mathematics, including: matrices and linear algebra - matrix algebra, linear equations, eigenvalues and vectors. Key Points and Interpolation - roots of polynomials, operations on polynomials and their differentiation, interpolation and extrapolation of curves, etc.

MATLAB contains a large number of graphing functions, including three-dimensional, visual analysis, and animation clips. The integrated development environment allows you to create a graphical user interface with various controls, such as buttons, input fields, etc. Thanks to the MATLAB compiler, GUIs can be converted into separate programs.

MATLAB provides convenient tools for developing algorithms, including high-level ones, using object-oriented programming concepts. It has all the tools you need to integrate your development environment, including a dehydrator. Functions for working with integer data types make it easy to create algorithms for microcontrollers and other applications where needed.

1.3.3 Simulink

Simulink could be a exceptionally supportive computer program made by a company called MathWorks. It can do a lot of diverse things. It could be a program that combines MATLAB and a way to form models. It’s a extraordinary put for doing computer work like programming and making pretend versions of things. This program can be utilized by diverse individuals to study complicated frameworks with numerous parts. The most way to use it is through a picture device where you’ll be able move pieces around. There are moreover a bunch of diverse pieces you’ll be able include to create it your claim. This thing has cool stuff like controlling how items see, keeping track of things, and analyzing how it gets utilized.

Simulink may be a tool that makes a difference you make reenactments utilizing MATLAB. It looks decent and it’s simple to utilize. This tool helps individuals make imagine models of things they need to plan. It’s simple to utilize and lets clients attempt out parcels of different ideas without much work. You do not need to be an master to utilize this stage. Utilize pre-made squares and drag and drop them to make pictures of frameworks effectively.

Understudies learn way better when they get input frequently. Simulink makes a difference you learn by letting you test. You'll alter things rapidly and see what happens, which makes a difference you inquire «what in the event that» questions. Finally, the vital thing is that Simulink and MATLAB are connected together so that data can be effortlessly shared between them.

1.3.4 LabVIEW

LabVIEW is a development environment and platform for executing programs created in the graphical programming language «G» by National Instruments (USA). The first version of LabVIEW was released in 1986 for the Apple Macintosh, and now there are versions for Unix, Linux, Mac OS, etc. The most developed and popular version is for Microsoft Windows.

LabVIEW is also used in data acquisition and processing systems, as well as for managing technical facilities and technological processes. From an ideological standpoint, LabVIEW shares similarities with SCADA systems. However, it is noteworthy that LabVIEW distinguishes itself from SCADA systems by prioritizing the resolution of issues within the realm of ASNA, as opposed to automatic control systems.

LabVIEW is a virtual device that consists of two parts:

– a block diagram that describes the logic of the virtual device; opponents can use partial virtual devices to create other virtual devices.

– The front panel of a virtual device contains input/output devices: buttons, switches, LEDs, nonius, scales, information displays, etc. They are used by humans to control virtual devices as well as other virtual devices for data exchange.

– Components can use partial virtual devices to create other virtual devices. The front panel of a virtual device contains input and output devices: buttons, switches, LEDs, nonius, scales, information displays, and so on. They are used by a person to control virtual devices, as well as other virtual devices for data exchange.

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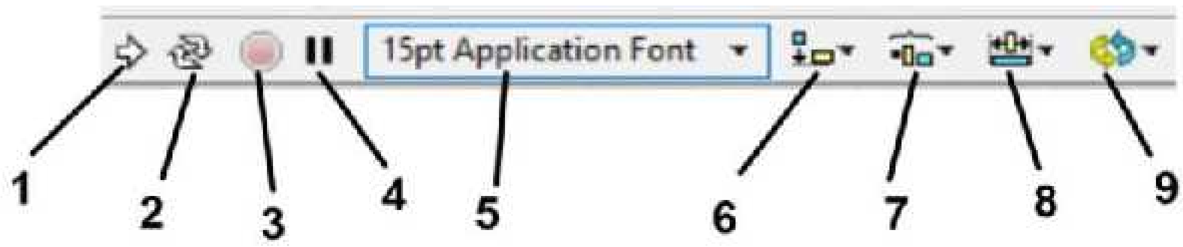

Figure 1.1 - LabView toolbar

The front panel contains tools that contain service buttons and status indicators designed to control virtual instruments (see Figure 1.1):

– button to start the program for execution, the icon changes its appearance during the program operation;

– button to start the program in a cyclic mode, the icon changes its appearance during the program operation;

– during the program operation, this button is in the active state, it is used to stop the program execution;

– The «pause» button stops the program execution until you press this button again;

– menu for editing font properties: type, size, style and color;

– a menu that allows you to align and position objects, there are options such as aligning selected objects to the left edge, to the right;

– a menu containing commands for spatial distribution of graphic objects on the front panel;

– a menu that contains commands for resizing components on the front panel

– if you place an object on top of another, the lower one may be blocked and inaccessible, using this menu you can place the object above or below the desired one.

LabVIEW supports a huge range of equipment from different manufacturers and includes (or allows you to add to the basic package) numerous component libraries:

– connect external equipment via the most common interfaces and protocols (RS-232, GPIB-488, TCP/IP, etc.)

– for remote control of the experiment;

– to control robots and machine vision systems;

– to generate and digitally process signals;

– apply various mathematical methods of data processing;

– visualize data and the results of their processing (including 3D models);

– to model complex systems;

– store information in databases and generate reports;

– interact with other programs within the concept of COM / DCOM/OLE.

1.4. PRINCIPLES OF DIGITAL SIGNAL PROCESSING

The field of study related to the manipulation and analysis of data in a digital format, commonly referred to as Digital Signal Processing (DSP), is of significant interest within academic literature.

The process of translating signals into a digital format is known as digital conversion.

Continuous analog signals, denoted as $s(t)$, have the potential to be discretized through a time-sampling process and subsequently represented numerically in digital format. If the sampling frequency, denoted as F_d, satisfies the condition $F_d \geq 2F_m$ where F_m represents the maximum frequency contained within the signal spectrum, the resultant discrete signal, denoted as s(k), is equivalent to the original continuous time signal denoted as $s(t)$, when the least squares (LS) method is employed.

Through the utilization of mathematical algorithms, the signal $s(k)$ can be subjected to a series of transformations resulting in the generation of $s_1(k)$ that exhibits the desired properties. The act of modifying signals is referred to as filtering, while the contraption responsible for carrying out this action is commonly known as a filter. Given that the signal values are received at a uniform rate denoted by F_d, it is essential that the filter allows for sufficient time to process the current signal within the series prior to the arrival of the subsequent one (typically, prior to the arrival of the next n samples, whereby n represents the filter delay). Specifically, the filter is expected to operate in real-time for processing the signal. In real-time processing of a signal, a specialized computing device known as a digital signal processor is employed to perform filtering operations. This phenomenon is applicable not solely to persistent signals, but also to intermittent signals and signals that have been recorded on a digital storage medium. The processing speed holds no significant impact in the latter scenario.

Various techniques exist for signal processing in both the temporal and spectral domains. The determination of the equivalence between the time-frequency transform and the Fourier transform is exclusively established by the latter.

In contemporary electronic oscilloscopes and digital oscilloscopes, the utilization of time domain signal processing is prevalent. Digital spectrum analyzers are commonly utilized for the purpose of accurately depicting signals in the domain of frequency. In order to investigate the mathematical underpinnings of signal processing, computer-based mathematical software extensions, including widely recognized platforms such as MATLAB, Mathcad, Mathematica, and Maple, are frequently employed. The original text is not provided. Please provide the text to be rewritten in academic writing.

In contemporary times, there exists a novel mathematical framework employed for encapsulating signals via «short pulses» known as wavelets that have gained significant traction in the field of signal and image processing. The application of the proposed methodology facilitates the processing of signals that exhibit characteristics such as fixed values, discontinuities, as well as other relevant features. The principal uses of this technology include:

Linear filtering involves the process of extracting signals in the frequency domain. Furthermore, it encompasses the synthesis of filters that are compatible with signals, as well as the frequency separation of channels. An additional means of signal extraction through Hilbert digitization (L^n (a, b)) and differentiation is also employed. Additionally, linear filtering includes the ability to alter and modify channel characteristics.

The practice of diagnostic examination and pattern recognition through the processing of audio signals, including those of speech, seismic events, and sound, is known as spectral analysis.

Time-frequency analysis is a significant area of study that finds numerous applications in image compression, radar technology, and detection of various signals. The technique employs multiple operations to identify radars and accurately detect signals.

Adaptive filtering finds utility in a multitude of applications such as speech processing, image processing, pattern recognition, and noise mitigation through implementation of adaptive antenna arrays.

The utilization of nonlinear processing techniques in signal and image processing has garnered considerable interest in recent years. Such techniques include correlation calculations, median filtering, synthetic amplitude, phase and frequency detection, speech processing, and vector coding. These advancements in nonlinear processing hold significant promise for enhancing the efficacy of signal processing systems.

In telecommunication systems and audio systems that utilize multiple rates, sample rate interpolation (i.e, increase) and decimation (i. e, decrease) are employed as part of the data processing strategy.

The process of acquiring convolutions.

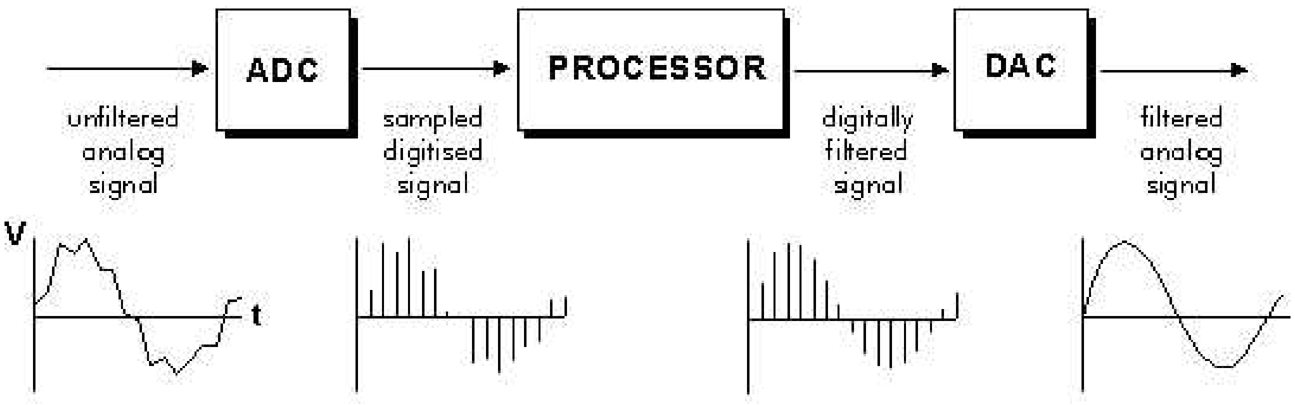

Figure 1.2. Digital signal processing

1.5 DIFFERENCE BETWEEN TIME AND FREQUENCY DOMAIN

A signal represented in the time domain is a set of instantaneous values set in the time interval for which the sample was obtained. However, in many cases, it is necessary to know the harmonic components of the signal rather than the values of individual readings in the sample.

The Fourier theorem posits that a periodic signal of any form in the time domain can be represented as a sum of sines and cosines, and their frequencies are determined by the period of the analyzed signal. The same signal can be expressed pairwise in the frequency domain, where each harmonic component is characterized by its amplitude and initial phase value.

In the frequency domain, you can conceptually form the harmonic components that need to be added to synthesize complex signals in the time domain. shows simple harmonic components that are a function of time and individual lines in the frequency domain. The height of each line is the amplitude of the harmonic component in time.

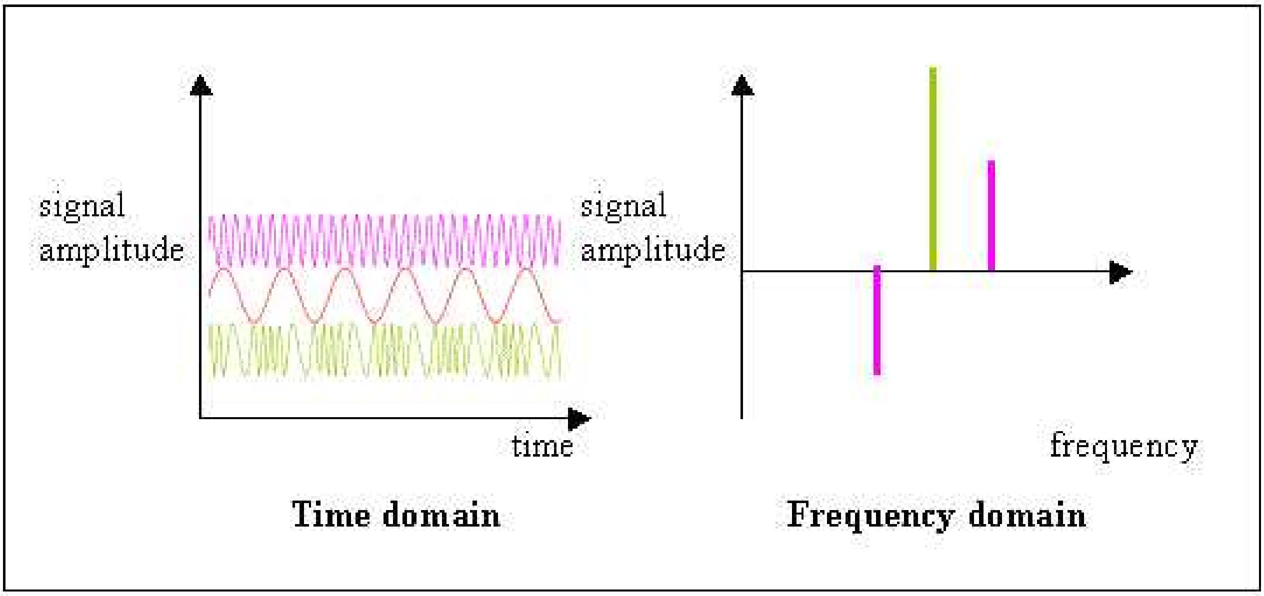

Figure 1.3. Difference between time and frequency domain

A signal expressed in terms of individual harmonic components is its representation in the frequency domain, which allows you to better understand the signal and how it was generated.

Samples obtained from a data acquisition device - data acquisition is the process of obtaining digital data from an ADC or digital data input connector - represent a time-domain representation of the signal. With the time-domain variables on the oscilloscope screen, certain measurements, such as harmonic distortion, are difficult to quantify. By using a FFT analyzer (also called a dynamic signal analyzer) to display the same signal in the frequency domain, you can easily measure harmonic frequency and amplitude and evaluate the nonlinear distortion of amplifiers.

1.6 MODULATION AND DEMODULATION

Modulation refers to the systematic manipulation of the parameters of a physical process over a specified duration according to a predetermined law or function.

The parameters of a physical process refer to the specific variables or factors that exert an influence on the behavior and outcome of said process. One of the crucial physical processes in radio engineering is an oscillation, referred to as a carrier frequency. This harmonic oscillation is responsible for transmitting the information over the airwaves.

The frequency of reference is a term commonly used in academic writing. The parameters of a physical process refer to the specific variables or factors that exert an influence on the behavior and outcome of said process. One of the crucial physical processes in radio engineering is an oscillation, referred to as a carrier frequency. This harmonic oscillation is responsible for transmitting the information over the airwaves.

The frequency of reference is a term commonly used in academic writing. The use of modulation allows you to:

– match the signal parameters with the line parameters;
– increase signal stability;
– increase the signal transmission range;
– organize multichannel transmission systems (SMEs with DCC).

Modulation is carried out in modulator devices. The conventional graphical designation of a modulator is as follows:

During modulation, signals are applied to the modulator input:

1) $u(t)$ is a modulating signal, this signal is informative and low-frequency (its frequency is denoted by W or F);

2) $s(t)$ is a carrier, this signal is uninformative and high frequency (its frequency is denoted by w_0 or f_0);

3) $s_{\text{:}}\ (t)$ is a modulated signal, the signal present is informative and high frequency.

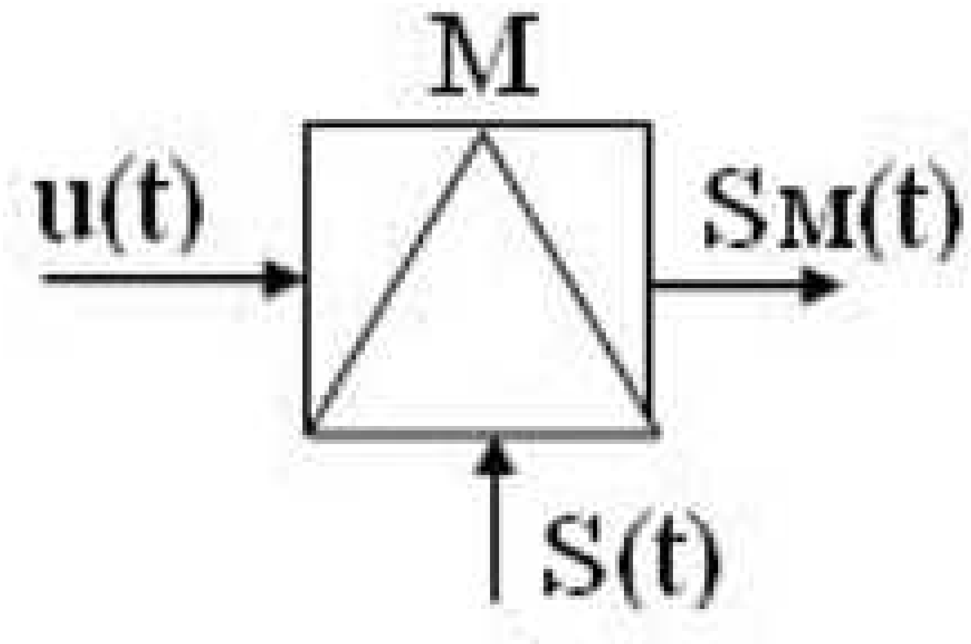

Figure 1.2 - Conventional graphic designation of the modulator

As a carrier signal, you can use:

– a harmonic oscillation, in which case the modulation is called analog or continuous;

– a periodic sequence of pulses, in which case the modulation is called pulse;

– direct current, while the modulation is called noise-like.

Since the modulation process changes the information parameters of the carrier oscillation, the name of the type of modulation depends on the variable parameter of this oscillation.

As a carrier signal, you can use:

– a harmonic oscillation, in which case the modulation is called analog or continuous;

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– direct current, while the modulation is called noise-like.

Since the modulation process changes the information parameters of the carrier oscillation, the name of the type of modulation depends on the variable parameter of this oscillation.

1. Types of analog modulation:

– Amplitude modulation (AM), which changes the amplitude of the carrier oscillation;

– frequency modulation (FM), which changes the frequency of the carrier oscillation;

– Phase modulation (FM), changes the phase of the carrier oscillation;

2. Types of pulse modulation:

– amplitude pulse modulation (AIM), changes the amplitude of the carrier signal pulses;

– frequency-pulse modulation (FPM), there is a change in the frequency

– of the carrier signal pulses;

– Pulse Width Modulation (PWM), which changes the phase of the carrier signal pulses;

– pulse width modulation (PWM), which changes the duration of the carrier signal pulses.

– Demodulation is the process of restoring the base signal (information signal) that was modulated to the carrier signal during the modulation process.

The process of modulation involves the modification of the characteristics of the carrier signal, specifically its amplitude, frequency, or phase, in accordance with the attributes of the information signal. This measure guarantees the propagation of data through radio frequency, along with other modes of communication.

To recover an information signal from a modulated signal, different demodulation methods are used, depending on the type of modulation. For example, for amplitude modulation (AM), a simple detector circuit is used to extract the information signal from fluctuations in the amplitude of the carrier signal. For frequency modulation (FM), an FM demodulator is used to recover the information signal from changes in the frequency of the carrier signal.

Demodulation is an important part of any communication system, as it allows you to extract useful information from a modulated signal.

SECTION 2
APPLICATION TO THE SYNTHESIZED SIGNAL

2.1 AMPLITUDE DEMODULATION

Amplitude demodulation (AM demodulation) is the process of recovering an information signal from an amplitude modulated (AM) signal.

In amplitude modulation (AM), the information signal modulates the amplitude of the carrier signal, resulting in an amplitude-modulated signal. To recover the information signal from the AM signal, an AM demodulator is used.

A simple AM demodulator can be built from a diode and a low-pass filter. The diode acts as a rectifier, converting the input signal into a pulsating DC voltage. The low-pass filter removes the high-frequency component, leaving only a low-frequency information signal. Other AM demodulation techniques include the use of a synchronous detector that uses two phase-locked circuits operating in different phases to detect the information signal. Other methods use the principles of phase locking or resonant oscillation to recover the information signal.

AM demodulation is used in radios, television receivers, and other communication systems to recover an information signal from an AM signal.

In Fig. 2.1 shows an arbitrary amplitude spectrum of a modulating signal and the corresponding amplitude spectrum of an AM signal, which consists of a harmonic oscillation of the carrier frequency, the upper sideband (UBS) and the lower sideband (NBS). At the same time, the UBS is a scale copy of the modulating signal spectrum shifted in frequency by the value f_0. The NBS is a mirror image of the VBS with respect to the carrier frequency f_0.

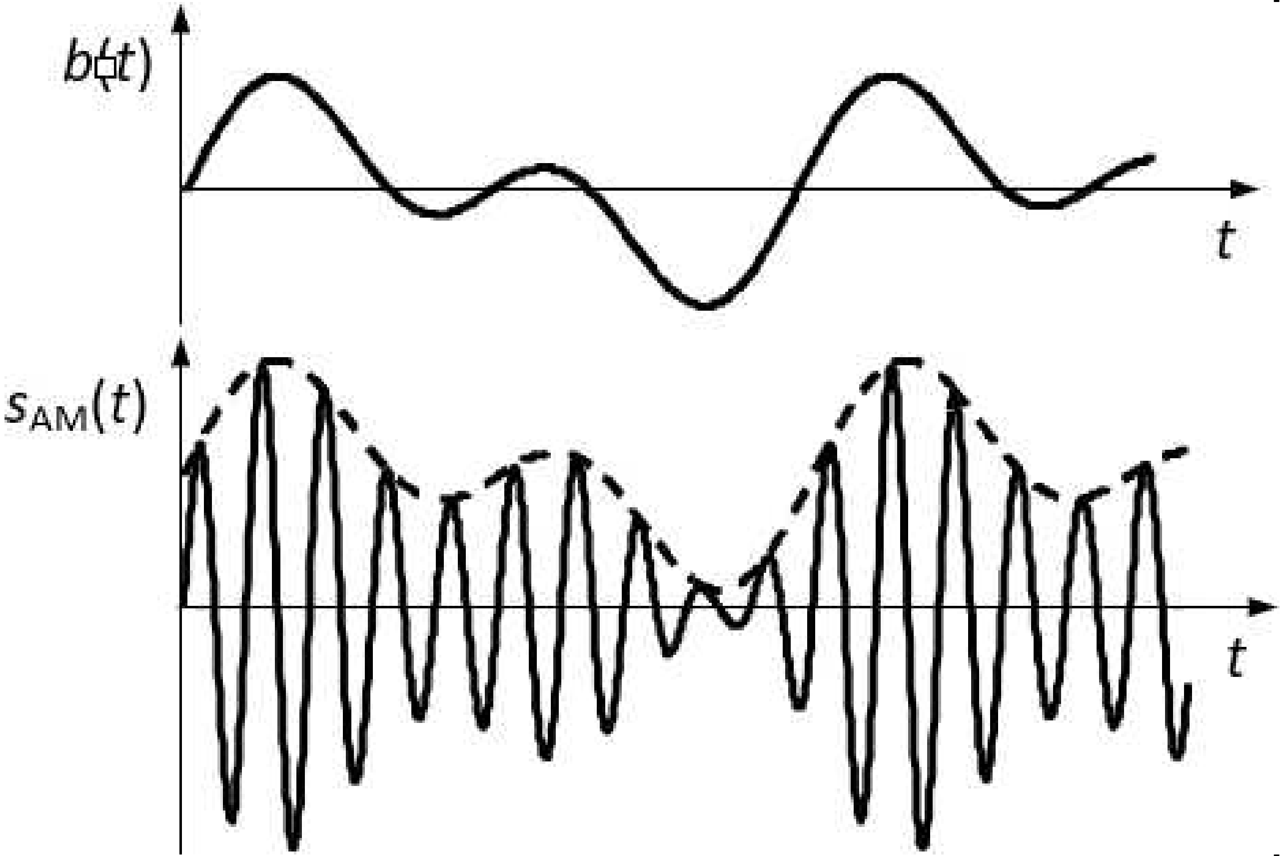

Figure 2.1 - Modulated b(t) and demodulated SAM(t) signal

From Fig. 2.1 shows an important result: the spectral width of the AM signal FAM is equal to twice the maximum frequency of the modulating signal spectrum, i.e., FAM = 2Fmax.

Calculations show that when modulating signals are the primary telecommunication signals, the sideband power share is only a few percent of the modulated signal power. Therefore, it is advisable to generate a signal with a spectrum consisting of only two sidebands (no carrier frequency fluctuation), such a signal is a balanced modulation signal.

2.2 AMPLITUDE DEMODULATION IN ANGLE RATE RESPONSE

The method of amplitude demodulation is not conventionally employed in the context of measuring angular velocity response. The concept of angle rate response pertains to the quantification of the rate at which an angular factor, such as angular velocity or rotation speed, undergoes change. The measurement of angular velocity response is frequently conducted through the utilization of devices, denoted as gyroscopes, which serve as sensors.

Gyroscopes are a type of sensor that are specifically engineered to quantify the angular rate or rotation. They utilize principles such as the Coriolis effect or the Sagnac effect in their operations. The sensors offer a means of measuring angular velocity or rotation in a straightforward manner, without requiring any amplitude demodulation.

Amplitude demodulation is a widely used technique for retrieving the modulating signal from an amplitude-modulated carrier signal. It is frequently employed in various applications, including AM radio communication. The measurement technique under consideration does not possess immediate relevance to the quantification of angle rate response.

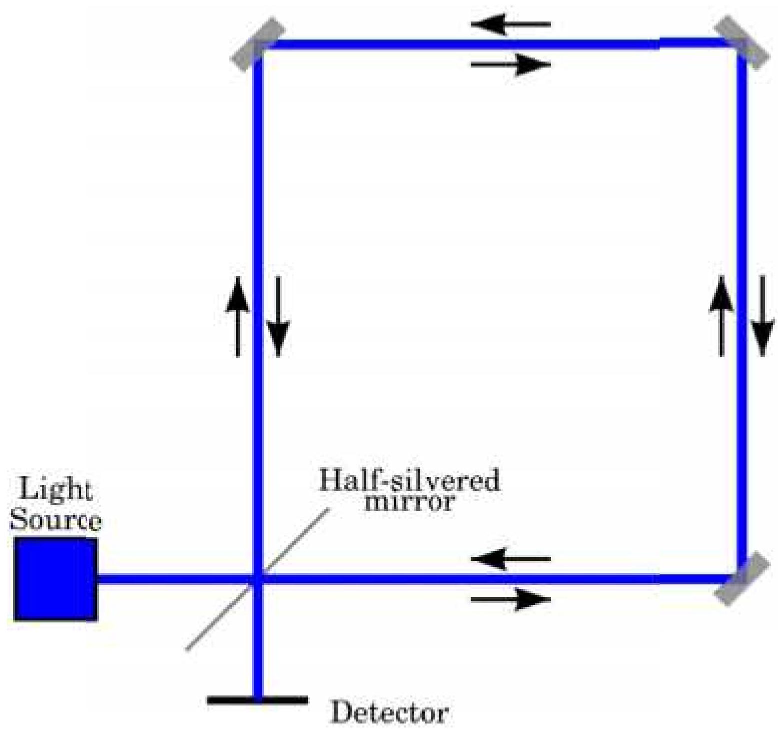

Figure 2.2. Sagnac effect

There are diverse approaches and devices leveraged to evaluate the angle rate response, including the utilization of a range of techniques and sensors.

Mechanical gyroscopes employ mechanical components, including oscillating masses or rotating disks, to measure angular velocity in a direct manner.

Vibrating gyroscopes are instruments that harness the Coriolis effect to detect and measure rotational motion. This phenomenon is manifested as deflection of a vibrating structure, resulting from the Coriolis force induced by rotation. The measurement of the deflection serves to ascertain the angular velocity.

The present discourse concerns optical gyroscopes, specifically the fiber optic gyroscopes (FOGs), which can ascertain angular displacement through the application of the Sagnac effect. The detection of rotation is achieved through the application of light waves that interfere with one another while travelling in opposing directions.

Micro-Electro-Mechanical Systems (MEMS) gyroscopes employ microfabrication methodologies to produce compact and integrated gyroscopic detectors. These devices are grounded in the tenets of micro-mechanical structures and are capable of precise measurement of angular velocity.

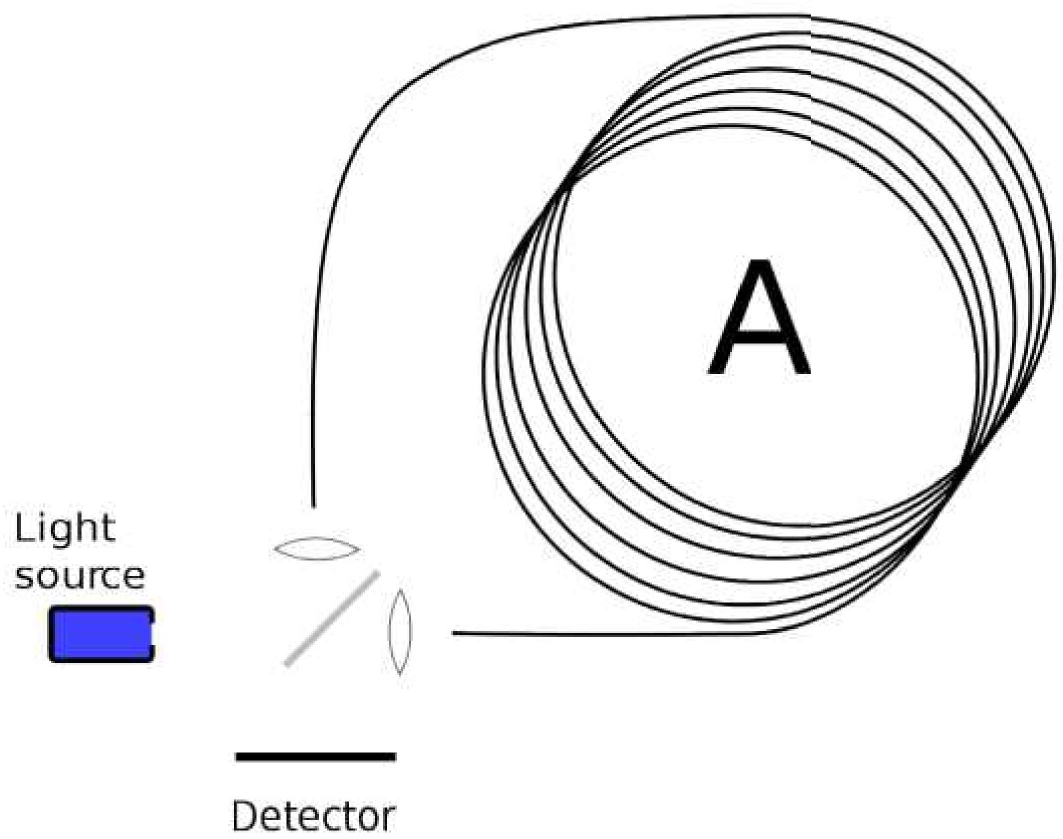

Figure 2.3. FOG

2.3 ALGORITHMS AND THEIR APPLICATION IN SENSORS

The underlying concept of amplitude demodulation is reliant on the identification and manipulation of amplitude fluctuations within the carrier signal of an AM transmission, with the objective of retrieving the modulating signal. The present discourse outlines the fundamental stages entailed in amplitude demodulation.

1. Reception of the AM Signal: The amplitude modulated (AM) signal, comprising a carrier wave that is modulated through changes in its amplitude by a modulating signal, is detected by means of an antenna or receiver.

2. Rectification: The AM signal received undergoes rectification to convert the alternating current (AC) signal into a unidirectional signal.

3. Low-Pass Filtering: The rectified signal encompasses the intended modulating signal as well as superior frequency components brought about during modulation. A low-pass filter is utilized to isolate the modulating signal. The utilization of a low-pass filter results in the elimination of high-frequency elements,

consequently permitting solely the envelope of the amplitude-modulated (AM) signal- the component with a relatively slow rate of change- to be transmitted.

4. Envelope Detection: The output of the low-pass filter is indicative of the envelope of the amplitude modulation (AM) signal, reflecting the fluctuations in the carrier signal's amplitude due to the presence of the modulating signal. The circuit designed for envelope detection successfully isolates and acquires said variations, thereby restoring the primary modulating signal in an efficient manner.

5. Amplification and Signal Conditioning: The demodulated modulating signal is often amplified and conditioned to ensure appropriate levels and fidelity for further processing or utilization. This may involve amplification, filtering, and additional signal processing steps as required for the specific application.

SECTION 3
THE AMPLITUDE DEMODULATION OF THE VARIABLE FREQUENCY SIGNALS

3.1 PRINCIPLES OF AMPLITUDE DEMODULATION OF THE VARIABLE FREQUENCY SIGNALS

The term «amplitude demodulation» denotes the operation of retrieving the modulating signal from a carrier signal that has been subjected to amplitude modulation (AM), which is subject to temporal oscillations in carrier signal frequency. The fundamental concept of amplitude demodulation remains consistent with that of conventional amplitude modulation (AM) demodulation, with the added caveat of accommodating the fluctuating frequency component. The following is a holistic summary of the underlying principle:

The reception of the variable frequency amplitude modulation (AM) signal involves the receipt of a carrier wave that has been subjected to modulation by the modulating signal. The signal is acquired through the use of an antenna or receiver.

The process of demodulation involves a crucial step of accurately tracking and synchronizing with the carrier frequency, which is subject to variations. In academic literature, it is frequently denoted as «carrier tracking». The attainment of this objective can be realized through the application of methodologies such as the utilization of phase-locked loops (PLL) or frequency tracking algorithms.

Following carrier frequency tracking and synchronization, the AM signal is subjected to envelope detection. to extract the fluctuations within the amplitude of the carrier signal.

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Consultant		V. Chikovani				FAET 402		
S. controller.		V. Chikovani						
Dep. head		Yu. Melnik						

The process of envelope detection entails the rectification of the received signal, which converts it into a unidirectional signal. Subsequently, a low-pass filter is utilized

This technique is crucial for signal processing applications.

The process of demodulating the modulating signal involves the derivation of the envelope of the variable frequency AM signal from the output of the envelope detection process. It is postulated that by performing demodulation on the aforementioned envelope, it is plausible to retrieve the modulating signal in its original form. The aforementioned objective may be achieved by employing a range of methodologies, such as peak detection, synchronous detection, or coherent detection.

Signal conditioning is a crucial step that may involve amplification, filtering, or other processing in order to improve the quality of the demodulated modulating signal. This process is necessary to prepare the signal for subsequent utilization or processing.

One of the primary difficulties in the amplitude demodulation of signals of varying frequency is the precise tracking and synchronization of the carrier frequency as it fluctuates. The maintenance of proper demodulation is of critical importance to prevent distortion or loss of the modulating signal. Sophisticated tracking algorithms and carrier recovery techniques are frequently utilized to mitigate this challenge.

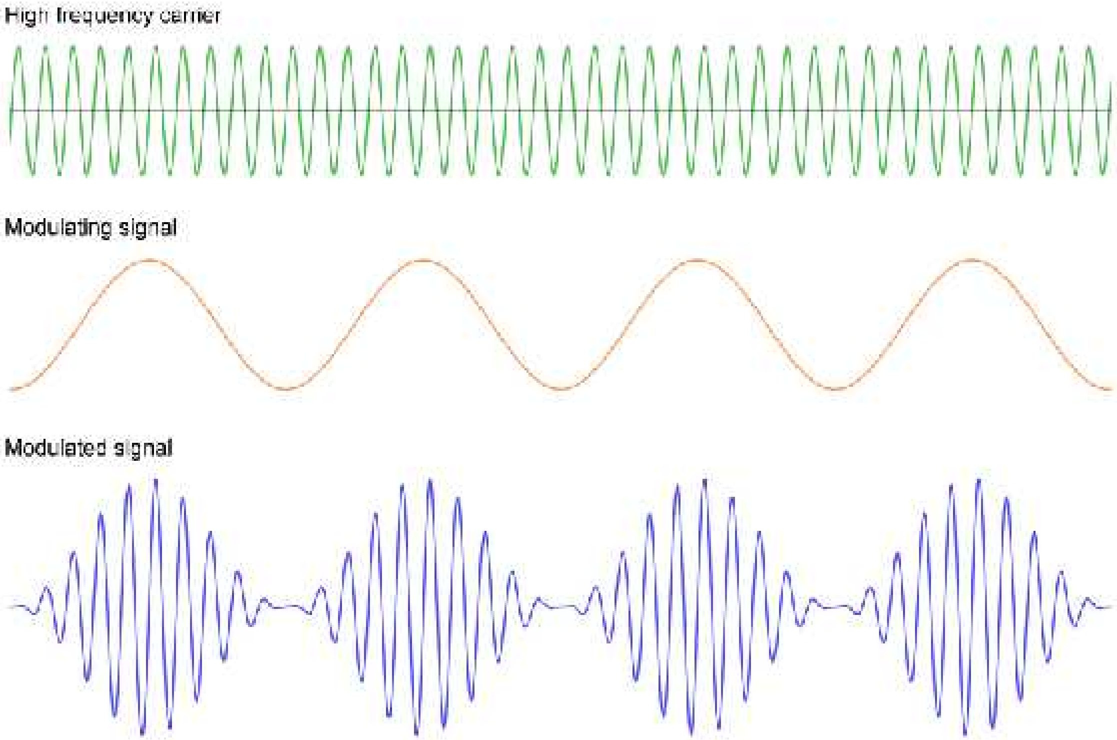

Figure 3.1. Amplitude modulation

3.2 EXISTING TECHNIQUES

The process of demodulation refers to the systematic extraction of the primary signal, which carries essential information, from the carrier wave. The process of extracting information from a modulated wave is accomplished through the use of an electronic device commonly referred to as a demodulator in academic writing.

Demodulation refers to the process of extracting information from a modulated signal. The output of a demodulator may consist of either video signals or digital signals in the binary data format. The foremost and prevalent utilization of this is typically observed in the realm of audio applications.

Modulation and demodulation represent two key processes utilized in radio transmission, whereby the former is utilized for signal modulation while the latter is responsible for signal demodulation. Nevertheless, numerous alternate systems incorporate demodulators in their operations. A prevalent illustration is that of a modem, which functions as both a modulator and a demodulator. The process of retrieving digital information from a designated carrier signal is commonly referred to as extraction.

Instead of utilizing demodulation, a variety of terms such as diode detector, synchronous detector, and product detector are employed in signal processing. Demodulation is a ubiquitous process that finds extensive application in the signal extraction from carrier waves.

A multitude of techniques are employed for the process of demodulating amplitude modulation signals in various applications. The existence of varied categories is attributable to the diverse utilization contexts related to both financial expenditure and output efficacy.

3.2.1 Synchronous detection

The synchronous detector, also known as demodulator, can be considered a derivative of the product detector circuit, and as such, it delivers quality outcomes for the demodulation of AM signals. The deployment of multiple components is a distinguishing feature of the demodulator in question, compared to a rudimentary diode detector. Nonetheless, given the widespread adoption of integrated circuit technology, it is highly feasible to seamlessly integrate this aforementioned demodulator variant into a myriad of radio receivers with a negligible increase in cost.

The demodulation of synchronous amplitude modulation entails the utilization of a mixer or product detector in conjunction with a local oscillator signal. The local oscillator signal is synchronized with the incoming signal carrier to ensure the absence of a beat note between them. The modulation scheme of amplitude modulation (AM) involves the generation of sidebands in the frequency domain. To retrieve the original audio signal, the sidebands are subsequently demodulated.

Considering its significantly enhanced operational efficiency and the facile integration with integrated circuits, this type of demodulator is commonly employed in a multitude of amplitude modulation (AM) broadcasting receivers, alongside sophisticated AM-based radio communication apparatus and handheld transceivers, among others. On figure 3.1. it’s easy to notice the principle of work of synchronous detector.

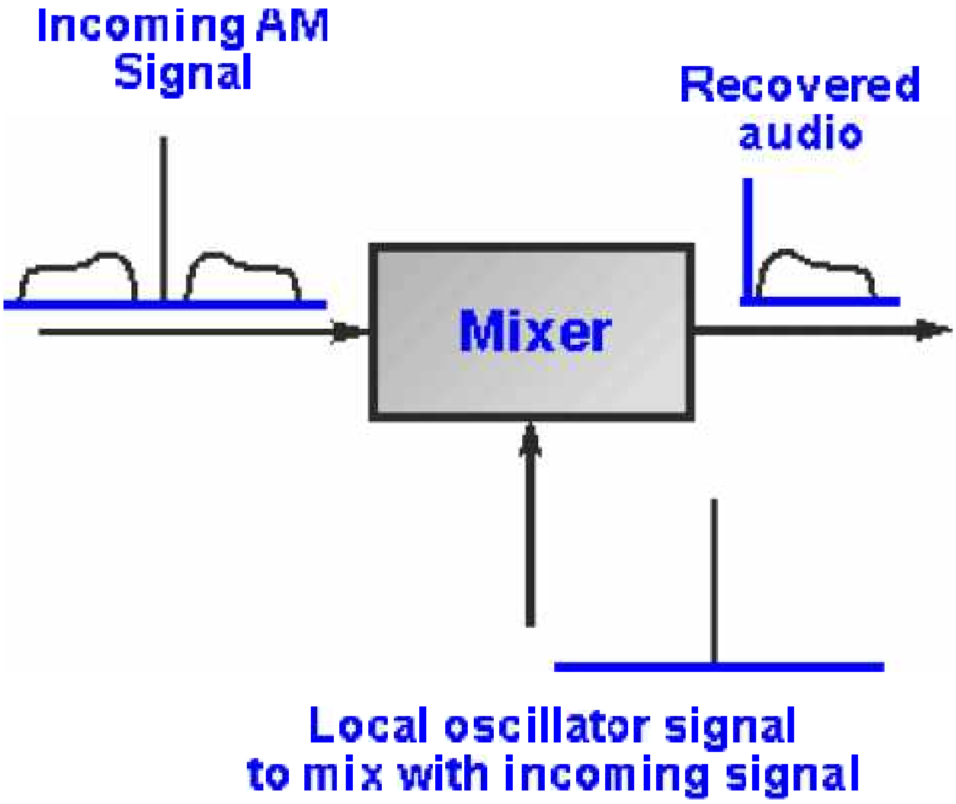

Figure 3.2 – Principle of work of synchronous detector

The primary obstacle with synchronous detection is the need for precise synchronization between the local oscillator and carrier signal. The occurrence of phase or frequency disparity can lead to erroneous demodulation and a decrease in fidelity.

3.2.2 Product detector

Amplitude modulated signals can be demodulated through the utilization of a receiver containing a product detector comprised of a mixer and either a local beat frequency oscillator or a carrier injection oscillator.

In the field of radio communication, it is common practice to utilize a product detector to facilitate the reception of single sideband, which is a variation of the amplitude modulation technique. In the process of demodulating Single Sideband (SSB) signals, a circuit commonly referred to as a product detector finds application. Single sideband is a modulation technique that removes both carrier and one sideband, leaving only a single sideband. This approach is a distinct form of amplitude modulation with unique characteristics. To restore the signal, a beat frequency oscillator or carrier insertion oscillator is employed to reintroduce the carrier wave that has been extracted from the AM signal. The resulting mixture is then subjected to mixing, generating a product that features both signals, which ultimately results in the restoration of the initial modulating signal.

The aforementioned circuit can be utilized for the purpose of actively receiving Morse code signals. The deployment of a sine wave generator is employed in order to generate an acoustic pulsation amidst the intermittent carrier, thereby rendering the Morse code audible.

To implement amplitude modulation (AM) demodulation, the receiver must be finely adjusted such that the carrier signal of the AM is synchronized with the beat frequency oscillator, therefore resulting in a zero beat. The audio that has been demodulated is subsequently observed at the output of the product detector. In order for this system to function appropriately, it is imperative for the receiver to sustain its frequency so that the Beat-Frequency Oscillator (BFO) frequency aligns precisely with that of the incoming carrier signal. Failure to do so will result in the persistent audible detection of a disquieting beat note. The product detector poses multiple challenges, encompassing synchronization, non-linear distortion, sensitivity to amplitude fluctuations, carrier leakage, as well as susceptibility to noise and interference.

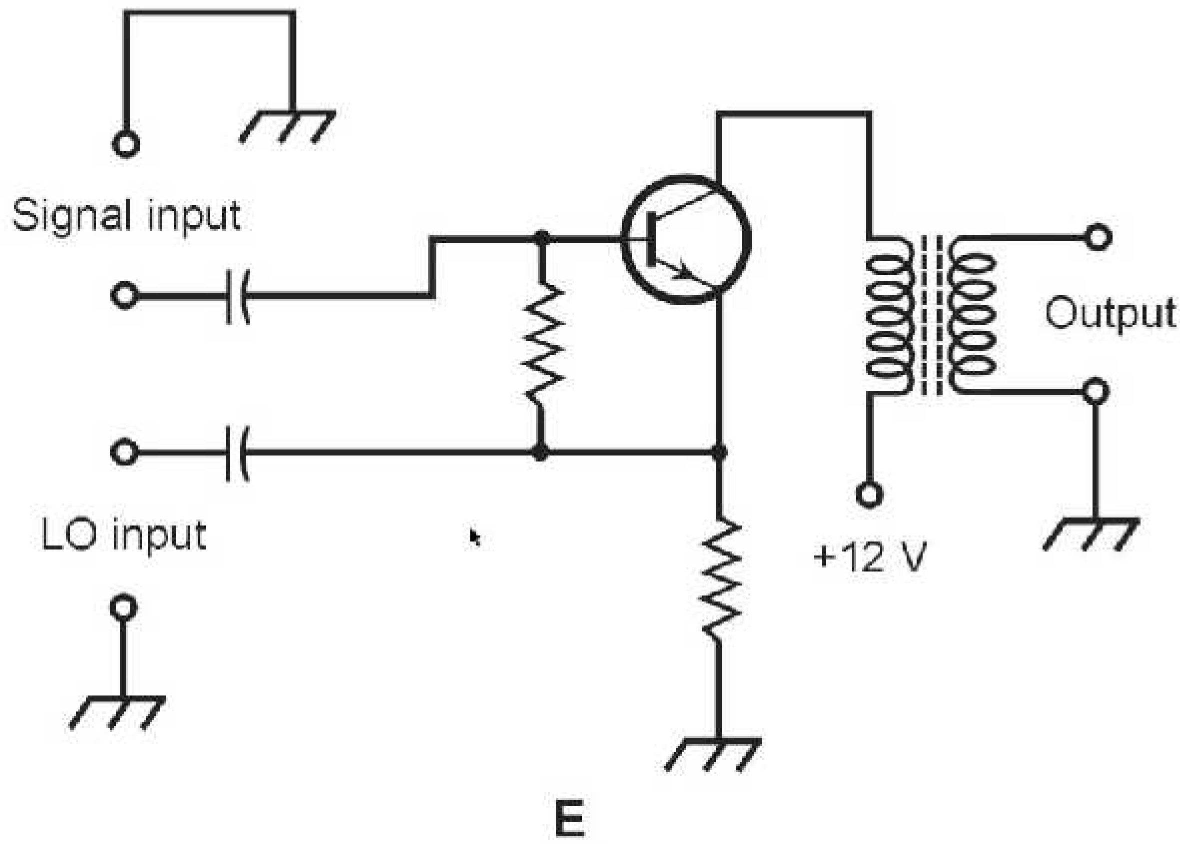

Figure 3.3 – Principle of work of product detector

3.2.3 Diode rectifier envelope detector

This particular type of detector adopts a basic configuration, necessitating only a solitary diode and a few supplementary, inexpensive components. The performance of low-cost AM broadcast radios is found to be acceptable; however, it fails to adhere to the benchmarks established by alternative forms of demodulation.

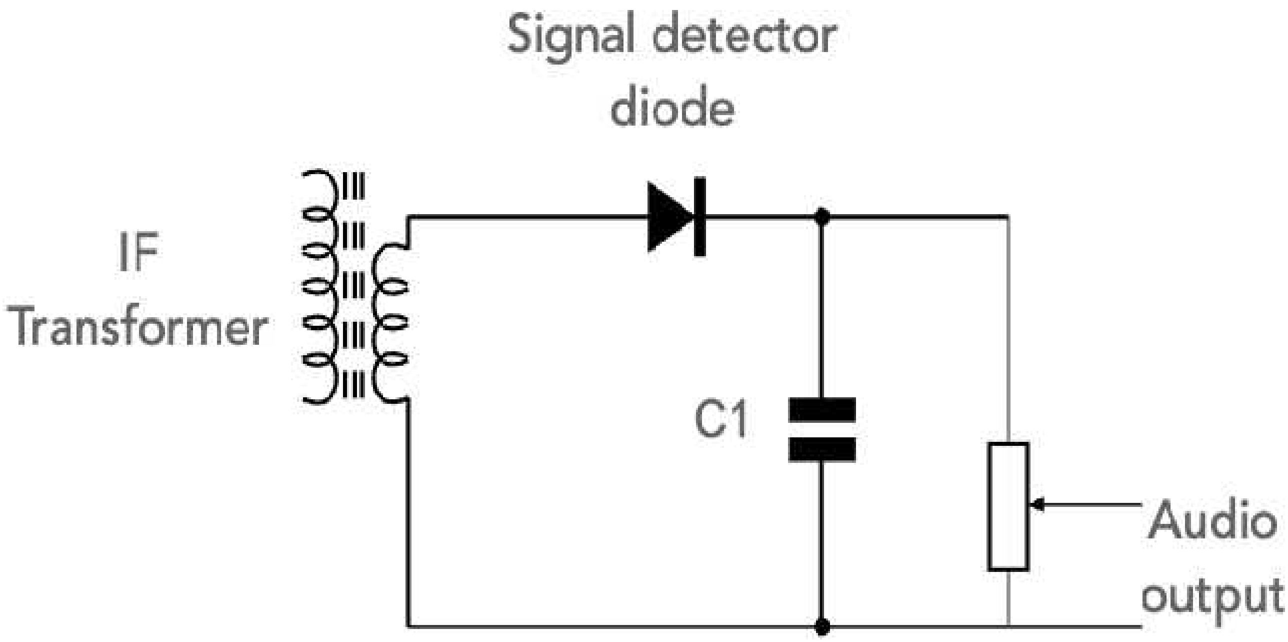

Figure 3.4 – Principle of work of diode rectifier envelope detector

The present article contends that the observed phenomenon is characterized by a considerable degree of distortion and lacks optimal performance when subjected to selective fading. This is particularly evident in the context of medium and short-wave bands. The diode detector has been employed for a considerable span of time, as previously indicated. The implementation of simplistic diode detectors became increasingly prevalent upon the replacement of valves with semiconductors in both domestic and professional valve or tube radio applications.

SECTION 4
SIMULATION RESULTS

4.1 SYNCHRONOUS DEMODULATION ALGORITHM

Amplitude demodulation process implements multiplication of input signal by reference one and low pass filtration to suppress high frequencies components arising after multiplication to separate low frequency envelope (see fig. 4.1).

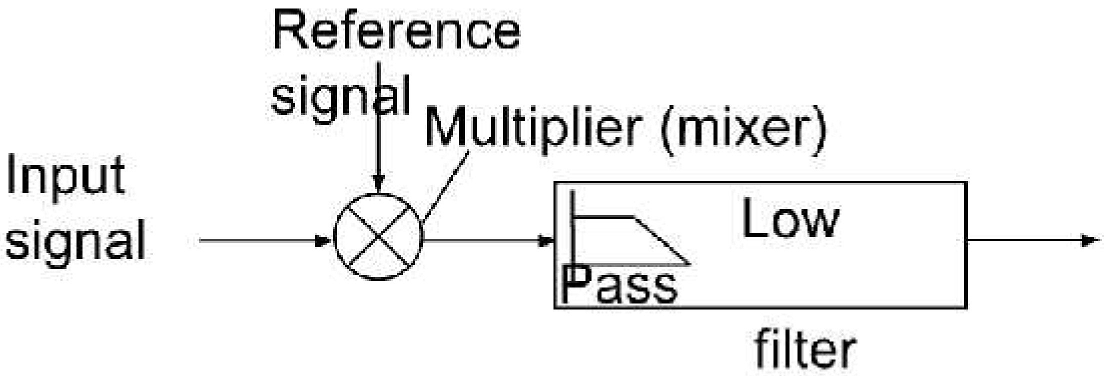

Figure 4.1. Amplitude demodulator

Input signal is an amplitude-modulated signal as shown in fig. 4.2.

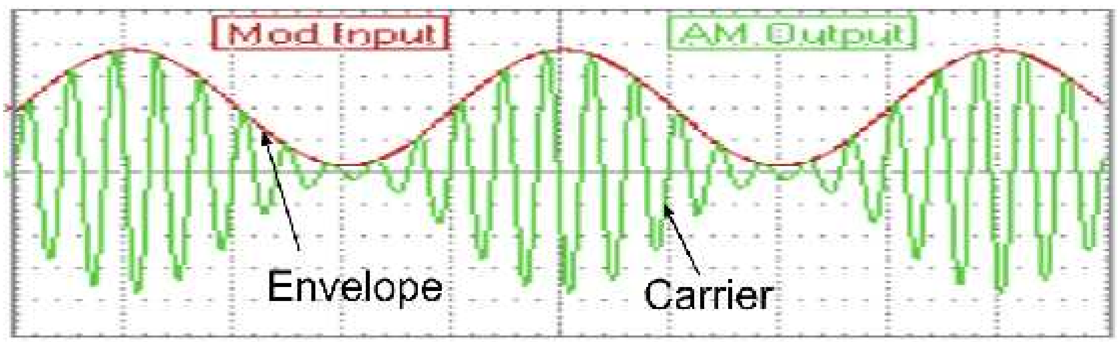

Figure 4.2. Amplitude signal

In general case the amplitude modulated signal can mathematically be described as follows: $y(t)= [A+m(t)sin(\omega_m t+\phi_m)]sin(\omega_c t+\phi_c)$; $t=i\Delta t$; $i=1\ldots N$,

where ω_m – modulating frequency; ω_c – carrier frequency; ϕ_m, ϕ_c – modulating and carrier signals phases, respectively. In expression (4.1) the next inequality should be valid $\omega_m<<\omega_c$ and $m(t)\leq A$. The ratio max($m)/A$ is called modulation index and expressed in percent.

For example, in case of angle rate sensor $A=0$, because in nodal point where angle rate arises the signal in absent of angle rate is equal to zero. When the resonator is rotating about its axis of symmetry with angle rate $\Omega_o sin(\omega_\Omega t+\phi_\Omega)$,where Ω_o is angle rate amplitude, ω_Ω is angle rate frequency, ϕ_Ω is angle rate phase, the nodal signal is: $y(t)= \Omega_o sin(\omega_\Omega t+\phi_\Omega)sin(\omega_x t)$; here ω_x is resonant frequency along X axis.

Demodulation process is that to separate the low frequency envelope $\Omega_o sin(\omega_\Omega t+\phi_\Omega)$, that is angle rate, from the high frequency carrier $sin(\omega_x t)$ by multiplication the signal $y(t)$ by the carrier frequency, which is reference signal (normalized antinode sense signal):

$y(t)sin(\omega_x t)= \Omega_o sin(\omega_\Omega t+\phi_\Omega)sin^2(\omega_x t)= \Omega_o sin(\omega_\Omega t+\phi_\Omega)[1-cos(2\omega_x t)]/2= \Omega_o sin(\omega_\Omega t+\phi_\Omega)/2-\Omega_o sin(\omega_\Omega t+\phi_\Omega) cos(2\omega_x t)]/2=\Omega_o sin(\omega_\Omega t+\phi_\Omega)/2-\Omega_o\{ sin[(2\omega_x+\omega_\Omega) t+\phi_\Omega]-sin[(2\omega_x-\omega_\Omega) t-\phi_\Omega]\}/2;$

So, the first summand

$\Omega_o sin(\omega_\Omega t+\phi_\Omega)/2$ multiplied by 2 is envelope (for gyroscopes it is angle rate) and is concentrated at the low frequency ω_Ω (usually $\omega_\Omega\leq$100Hz), the two others summands are concentrated at the double resonance frequencies $2\omega_x+\omega_\Omega$ and $2\omega_x -\omega_\Omega$ (usually $2\omega_x\approx$10 kHz). The power spectrum of the signal $y(t)sin(\omega_x t)$ is shown in fig. 4.3:

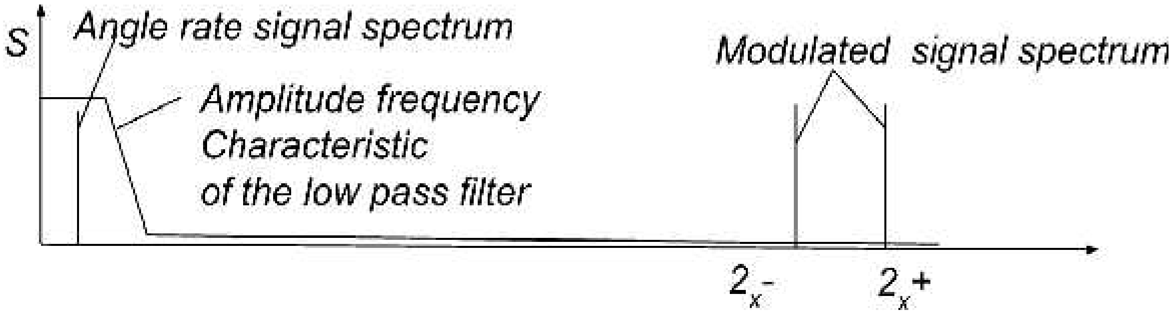

Figure 4.3. Power spectrum off modulated and demodulated signals

When this signal is passing through the low pass filter with cut frequency a little more than ω_Ω, on the output of this filter only envelope $\Omega_o \sin(\omega_\Omega t+\phi_\Omega)/2$ can be obtained.

4.2 AMPLITUDE MODULATION – DEMODULATION SIMULINK BLOCK DIAGRAM. SIMULATION RESULTS

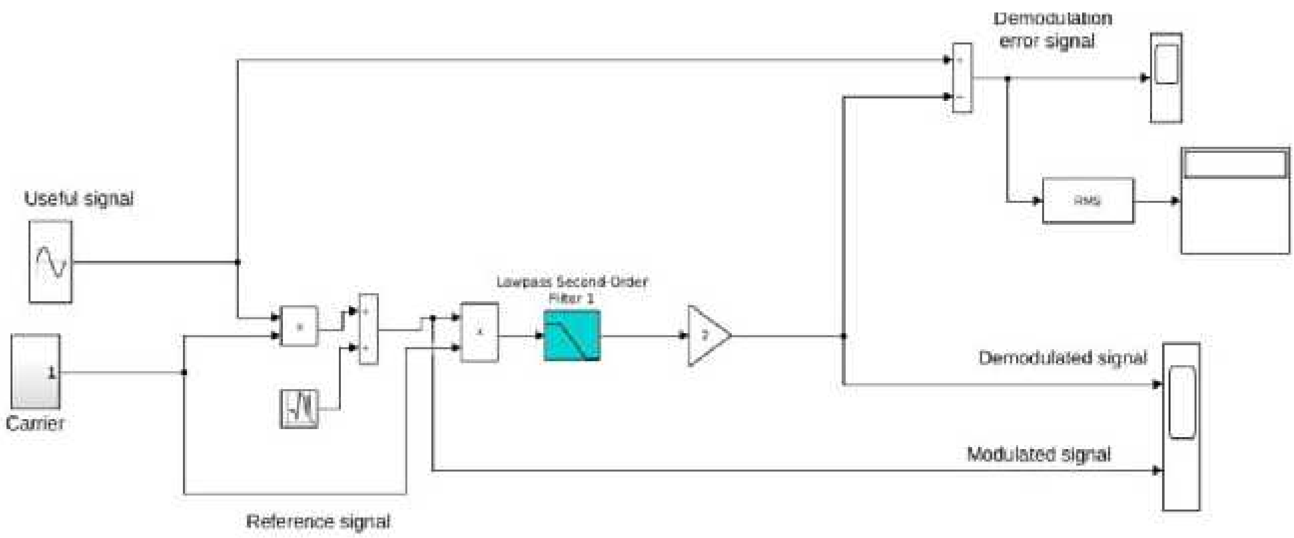

Figure 4.4. – Simulink block diagram to investigate Amplitude demodulation error.

In order to demonstrate the functioning of amplitude demodulation, a series of tasks must be carried out during the modulation process. This investigation aims to examine the effects of changing Amplitude, Frequency, and Variance on demodulation. The initial methodology depicts snow accumulation on a fig 4.4. The results obtained in this study with the use of data and values that were collected and recorded during the experimental process. Amplitude demodulation error scheme consist of three main blocks:

– Useful signal. Within this block, it is feasible to modify the values associated with Amplitude and Frequency.

– Noise. In the present block, the modifiable aspect pertaining to the Variance variable is discussed.

– RMS. RMS is a block which performs calculation of root mean square for the demodulation error.

Moreover, we do have an additional block called carrier where we can see one more scheme in which possible modulation is done (fig. 4.5) and block called display where we are able to see results of modulation. The process of demodulation is confined within a Lowpass Filter Block.

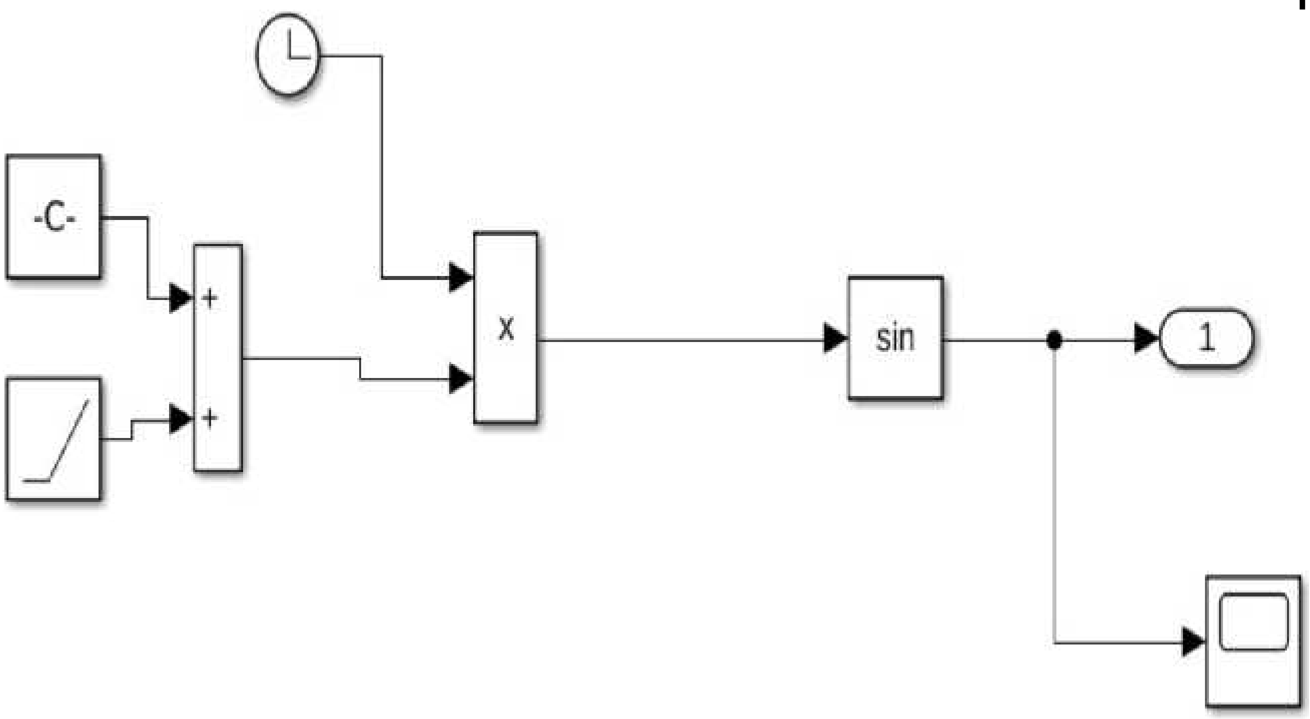

Figure 4.5. Carrier simulation Simulink block diagram

1. To plot the dependence of the root-mean-square error of the amplitude demodulation on the amplitude of the useful signal (RMS value of the demodulation error signal versus useful signal amplitude) for the following amplitude values: 10, 20, 30, 40, 50 B.

Each distinct meaning will be assigned its own corresponding definition. In order to access the intended significance, one must engage the option denoted as "useful signals" and adjust the variable labeled as "Amplitude". The value of 10 serves as the initial starting point and is subject to modification throughout the course of this modulation endeavor.

1) For Amplitude value 10B we will get a block diagram with RMS equal 0.629.

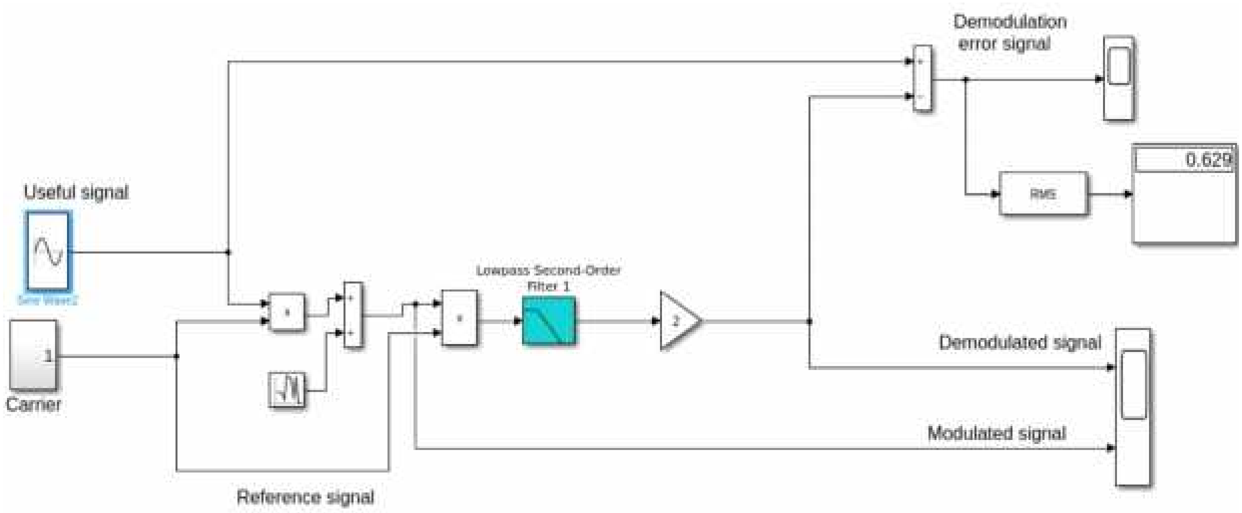

Figure 4.6. RMS of error at modulation Amplitude 10V

2) For Amplitude value 20B we will get a block diagram with RMS equal 5.124

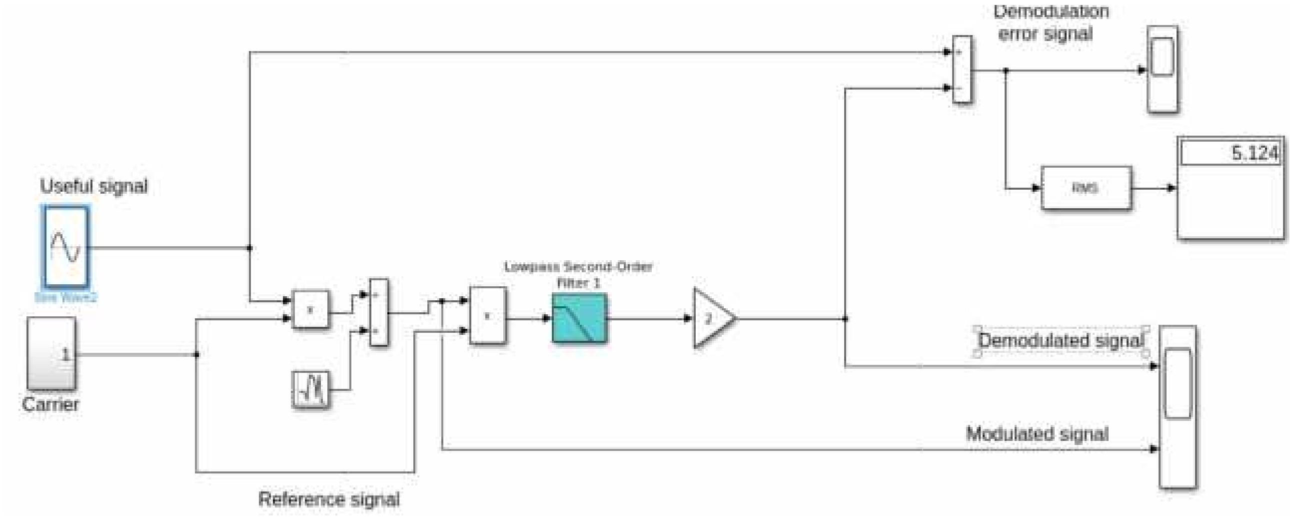

Figure 4.7. RMS of error at modulation Amplitude 20 V

3) For Amplitude value 30B we will get a block diagram with RMS equal 0.8174

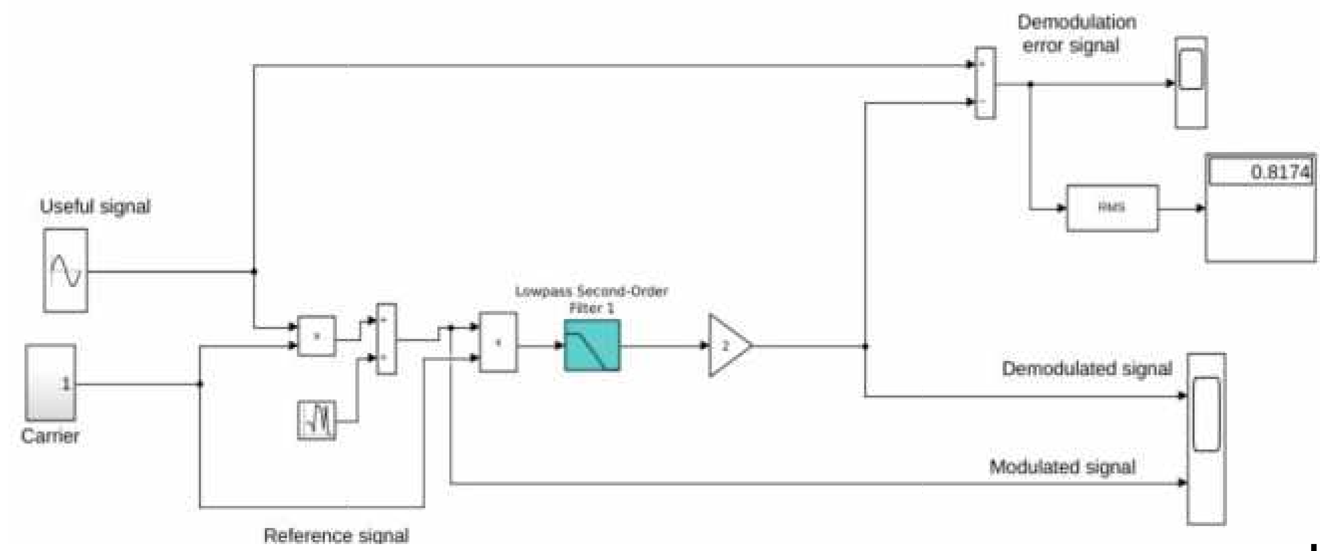

Figure 4.8. RMS of error at modulation Amplitude 30 V

4) For Amplitude value 40B we will get a block diagram with RMS equal 2.319

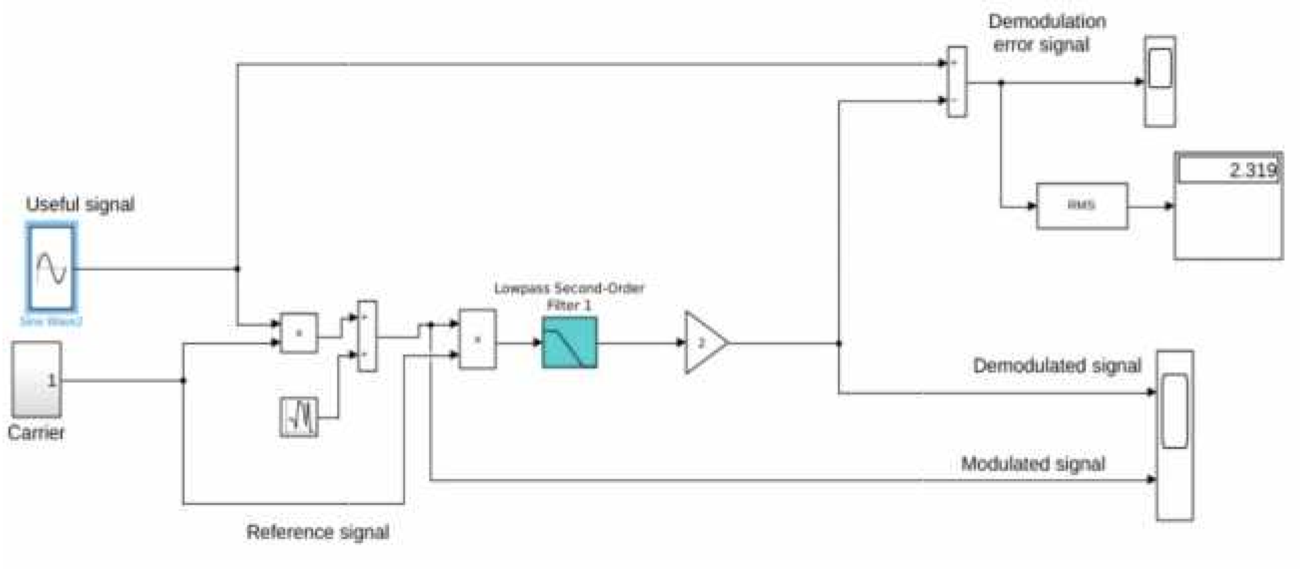

Figure 4.9. RMS of error at modulation Amplitude 40 V

5) For Amplitude value 50B we will get a block diagram with RMS equal 1.316

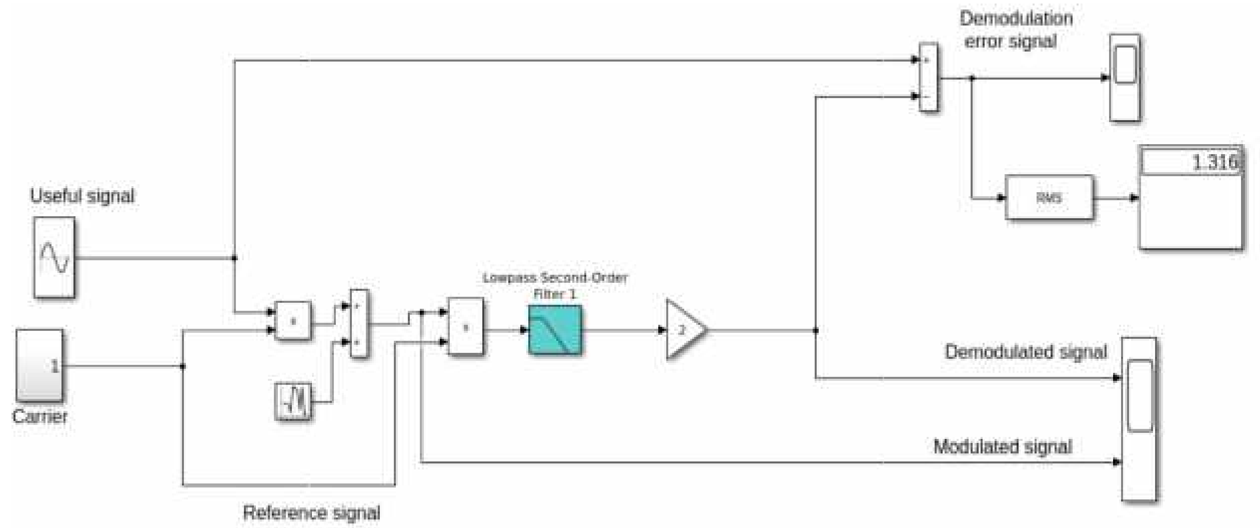

Figure 4.10. RMS of error at modulation Amplitude 50 V

In the modulation analysis undertaken earlier, the alteration in the root mean square (RMS) value was examined. The non-linearity of the data indicates that the modulation process exhibits a non-linear behavior. The result you can see in the table (table 4.1.) represented below. Upon completion of data collection pertaining to amplitude demodulation during fluctuations in amplitude, a graph (fig. 4.11.) has been produced, which is visible below.

Table 4.1.

RMS error versus amplitude

Amplitude	RMS
10B	0.629
20B	5.124
30B	0.8174
40B	2.319
50B	1.316

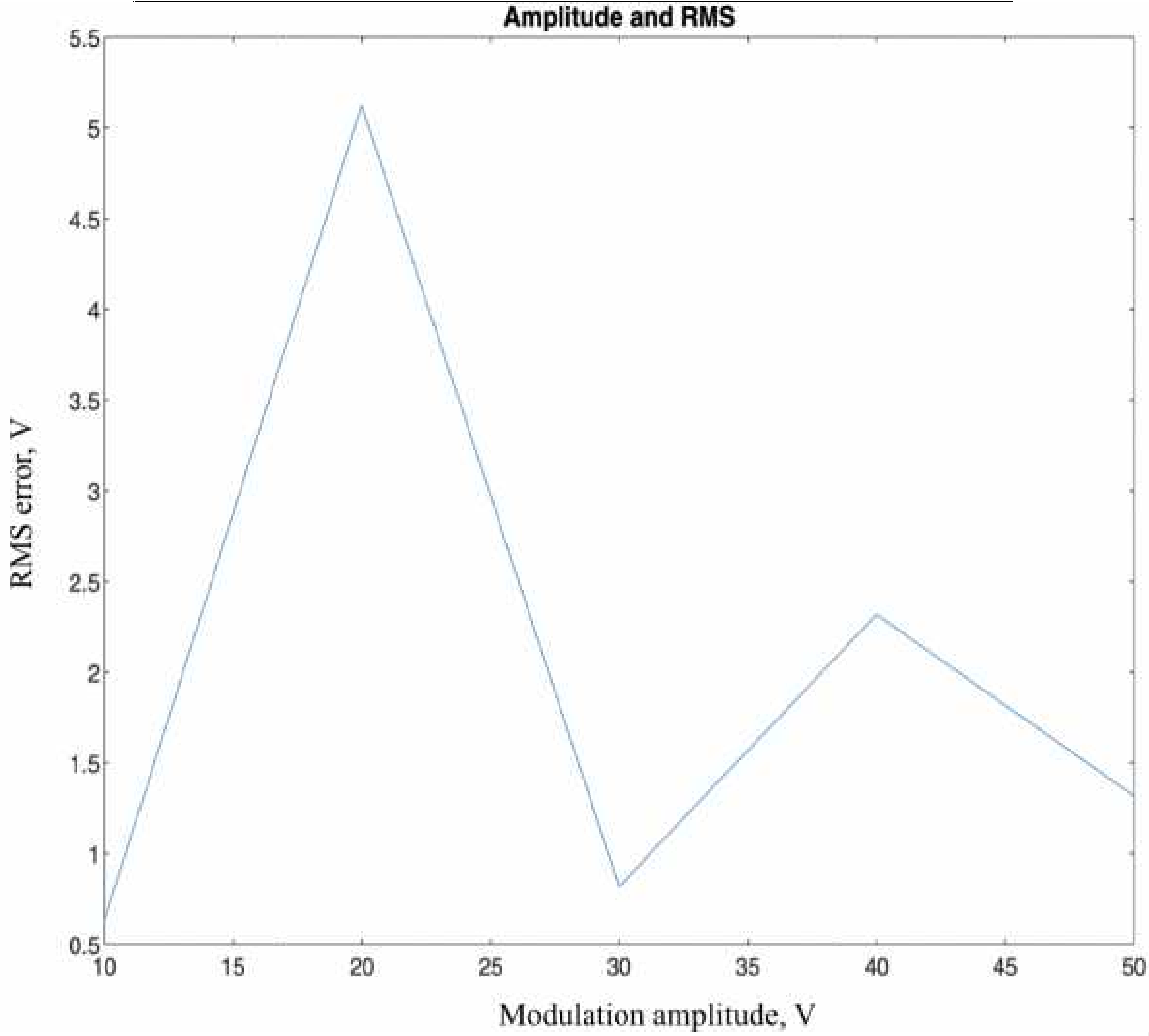

Figure 4.11. The final plot of RMS error versus amplitude

2. To plot the RMS value of the demodulation error signal versus the useful signal frequency for the following frequency values: 10, 30, 50, 80, 100 Hz.

Each distinct meaning will be assigned its own corresponding definition. In order to access the intended significance, one must engage the option denoted as «useful signals» and adjust the variable labeled as «Frequency». The value of 10 serves as the initial starting point and is subject to modification throughout the course of this modulation endeavor.

1) For Frequency value 10HZ we will get a block diagram with RMS equal 0.629.

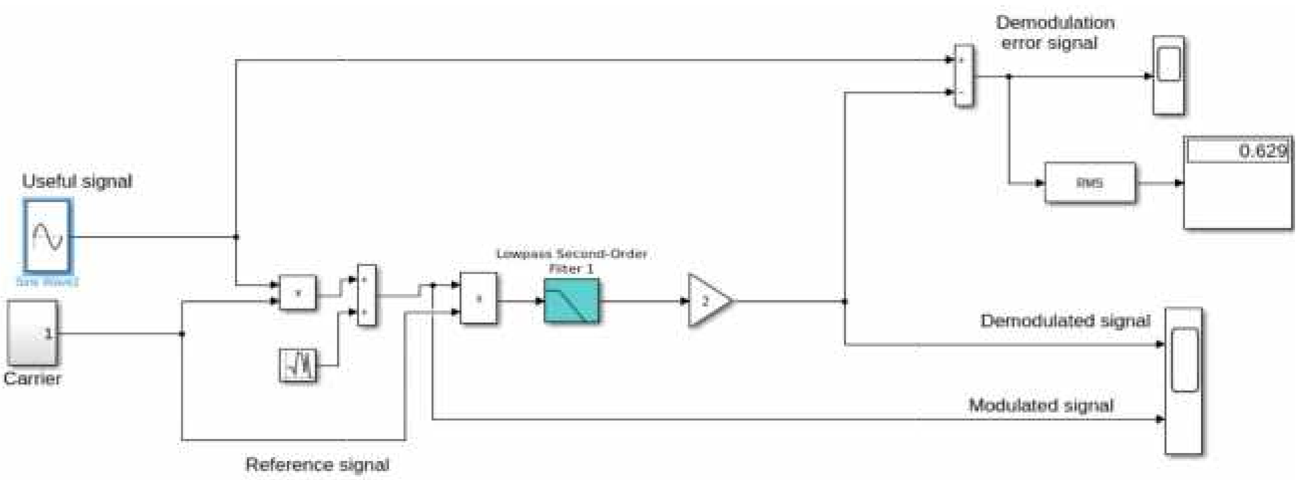

Figure 4.12. RMS of error at modulation Frequency 10 Hz

2) For Frequency value 30HZ we will get a block diagram with RMS equal 1.032

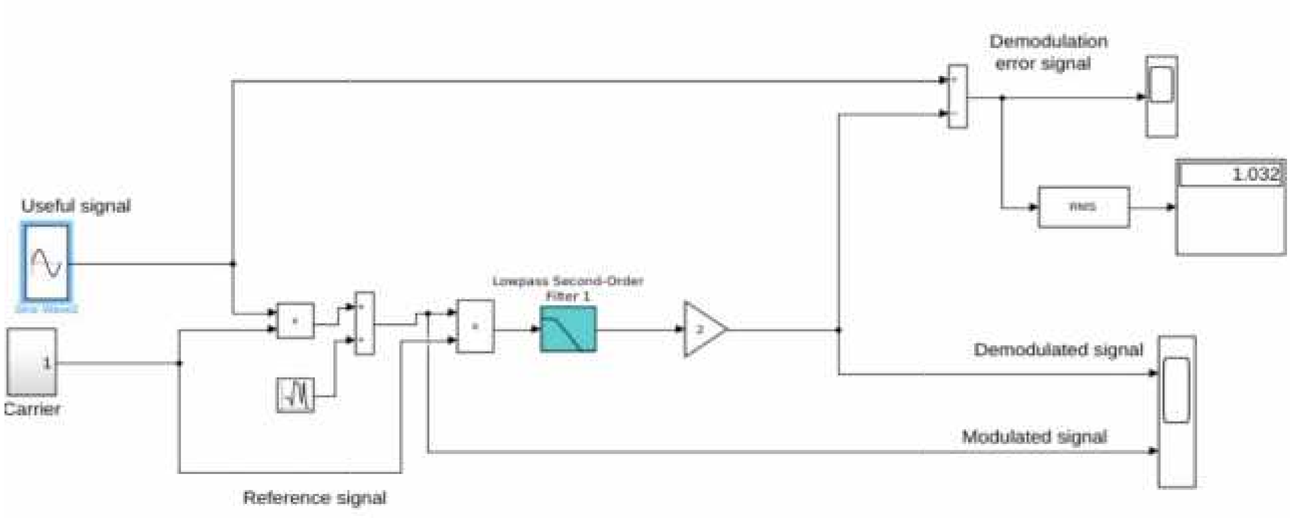

Figure 4.13. RMS of error at modulation Frequency 30 Hz

3) For Frequency value 50HZ we will get a block diagram with RMS equal 0.3135.

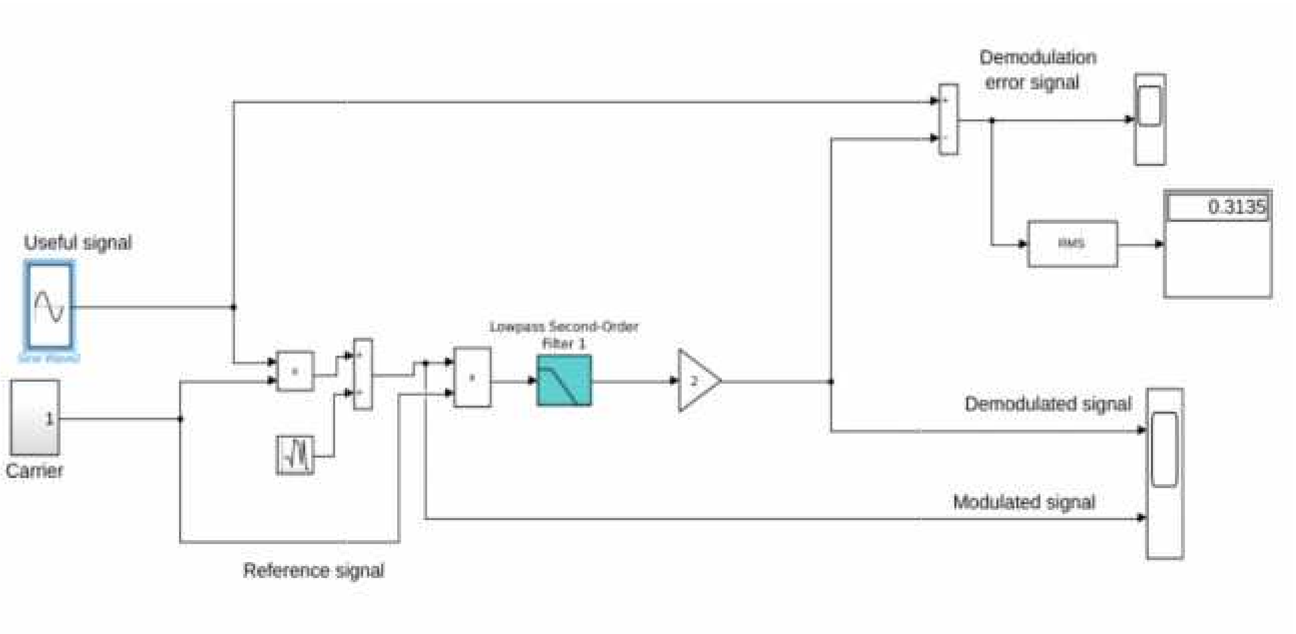

Figure 4.14. RMS of error at modulation Frequency 50 Hz

4) For Frequency value 80HZ we will get a block diagram with RMS equal 2.58.

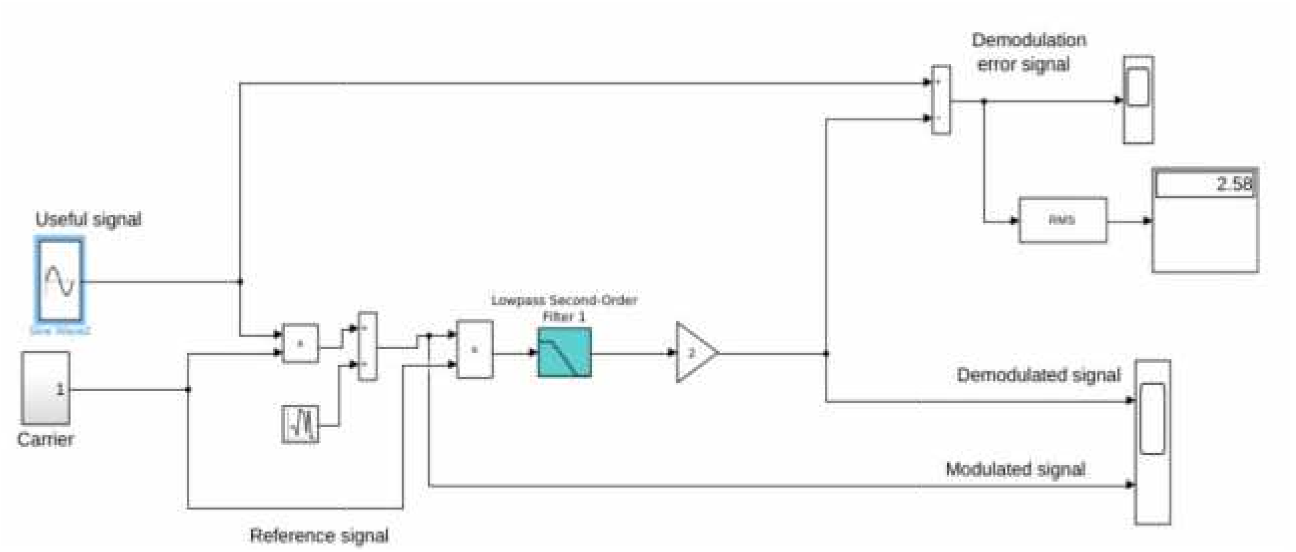

Figure 4.15. RMS of error at modulation Frequency 80 Hz

5) For Frequency value 100HZ we will get a block diagram with RMS equal 1.295

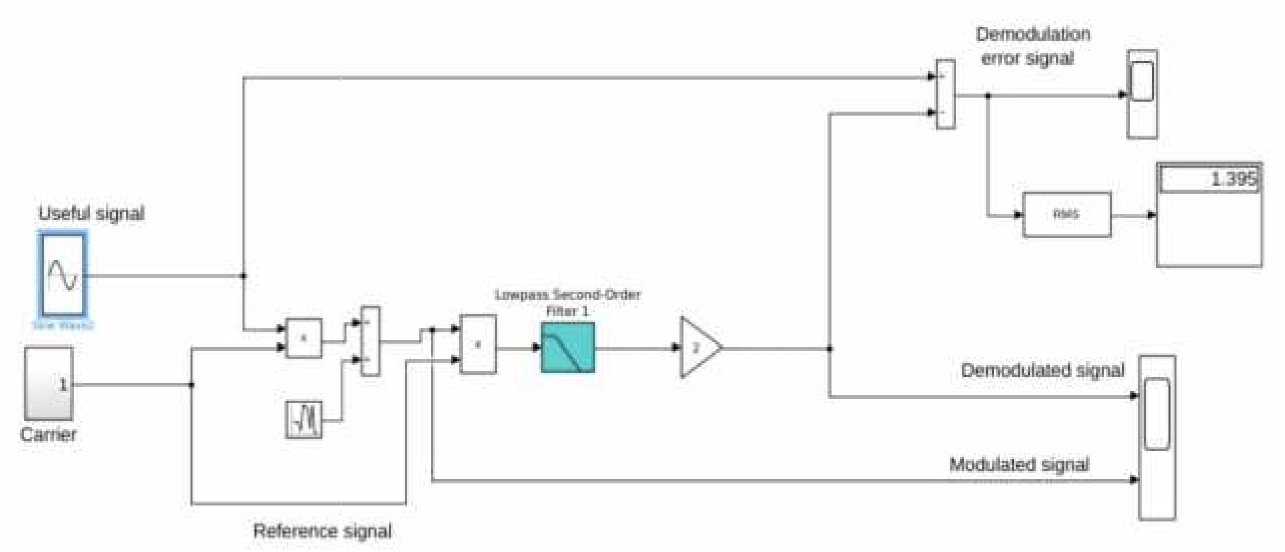

Figure 4.16. RMS of error at modulation Frequency 100 Hz

In the modulation analysis undertaken earlier, the alteration in the root mean square (RMS) value was examined. The non-linearity of the data indicates that the modulation process exhibits a non-linear behavior. The result you can see in the table (table 4.2.) represented below. Upon completion of data collection pertaining to amplitude demodulation during fluctuations in frequency, a graph (fig. 4.17.) has been produced, which is visible below.

Table 4.2.

RMS error versus frequency

Frequency	RMS
10Hz	0.629
30Hz	5.124
50Hz	0.8174
80Hz	2.319
100Hz	1.295

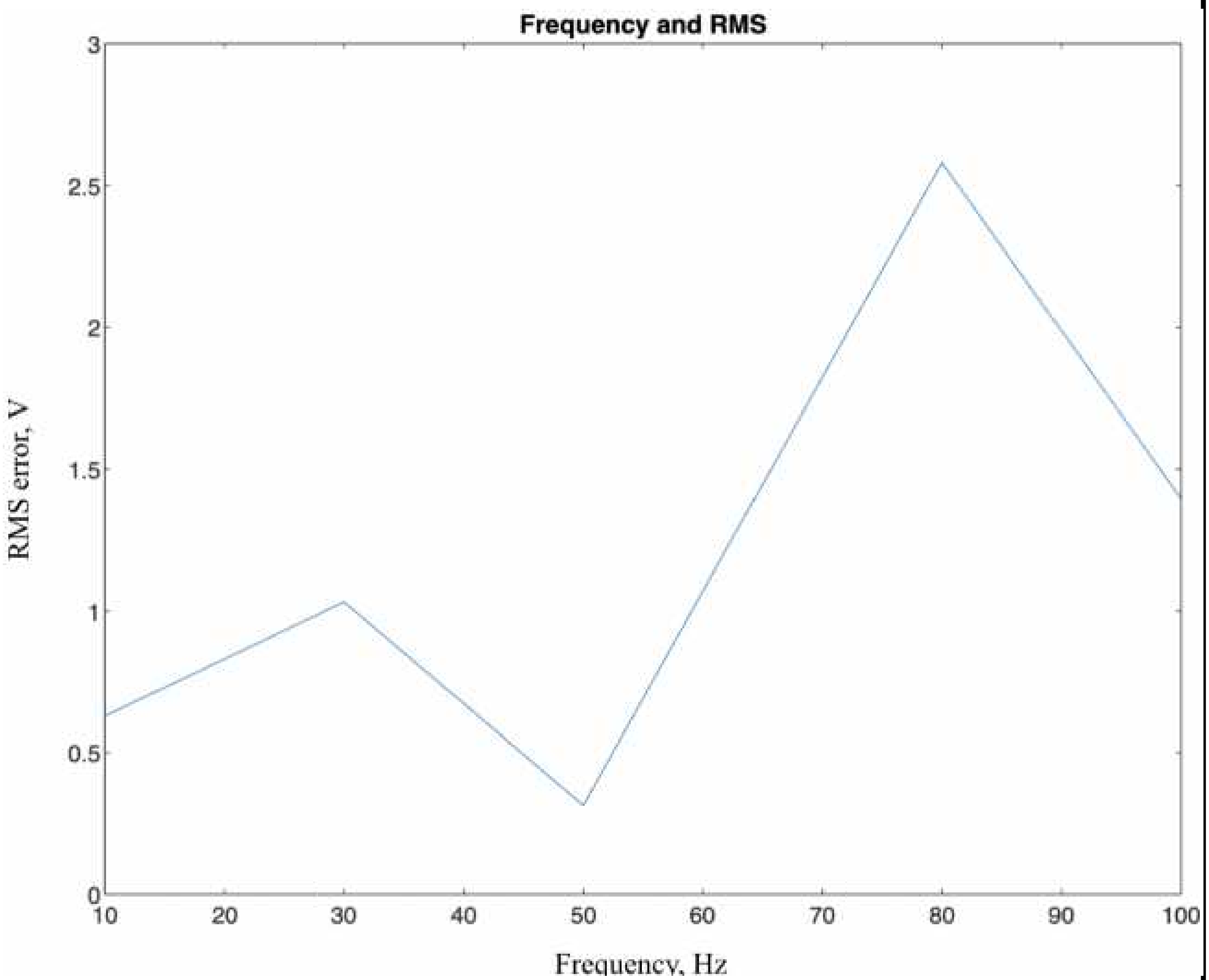

Figure 4.17. The final plot of RMS error versus frequency

3. To plot the RMS value of the demodulation error signal versus noise variation for the following values of the noise variation: 1, 3, 5, 7, 10 V^2

Each distinct meaning will be assigned its own corresponding definition. To access the intended significance, one must engage the option denoted as «noise» and adjust the variable labeled as «Variance». The value of 10 serves as the initial starting point and is subject to modification throughout the course of this modulation endeavor.

1) For Variance 1V^2 we will get a block diagram with RMS equal 0.03144.

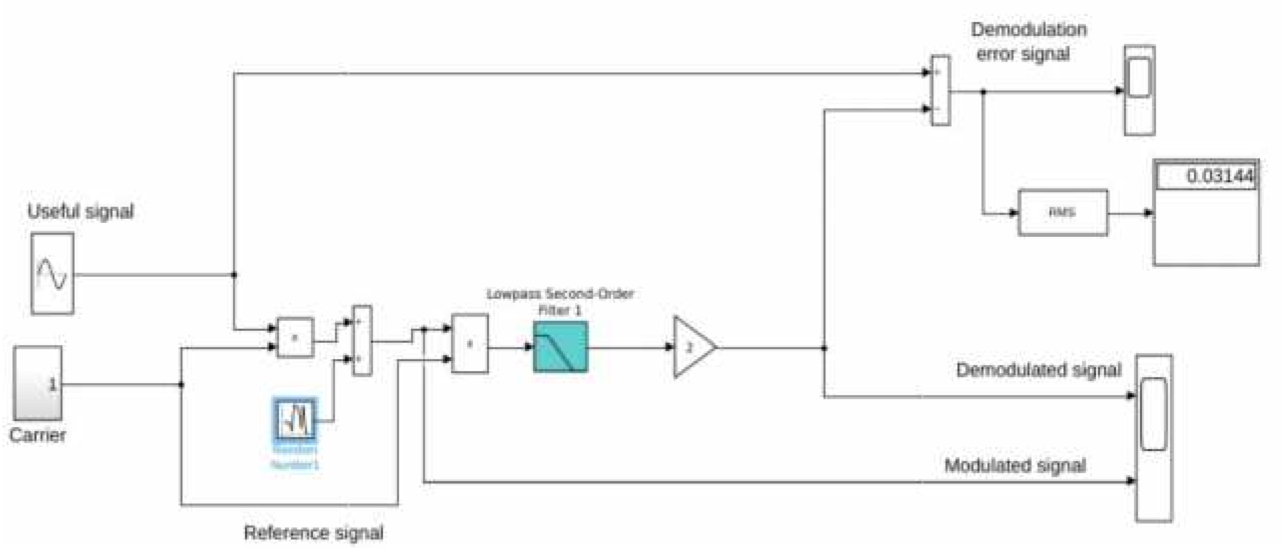

Figure 4.18. RMS error at input noise Variance 1

2) For Variance $3V^2$ we will get a block diagram with RMS equal 0.2251.

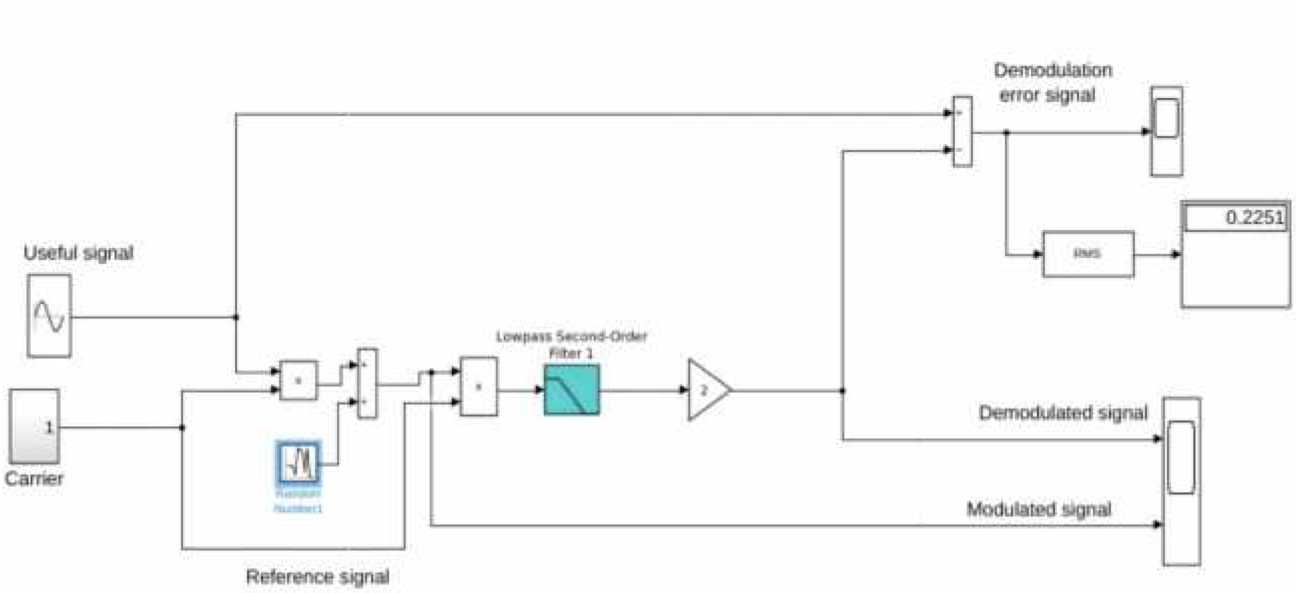

Figure 4.19. RMS error at input noise Variance 3

3) For Variance $5V^2$ we will get a block diagram with RMS equal 0.08666.

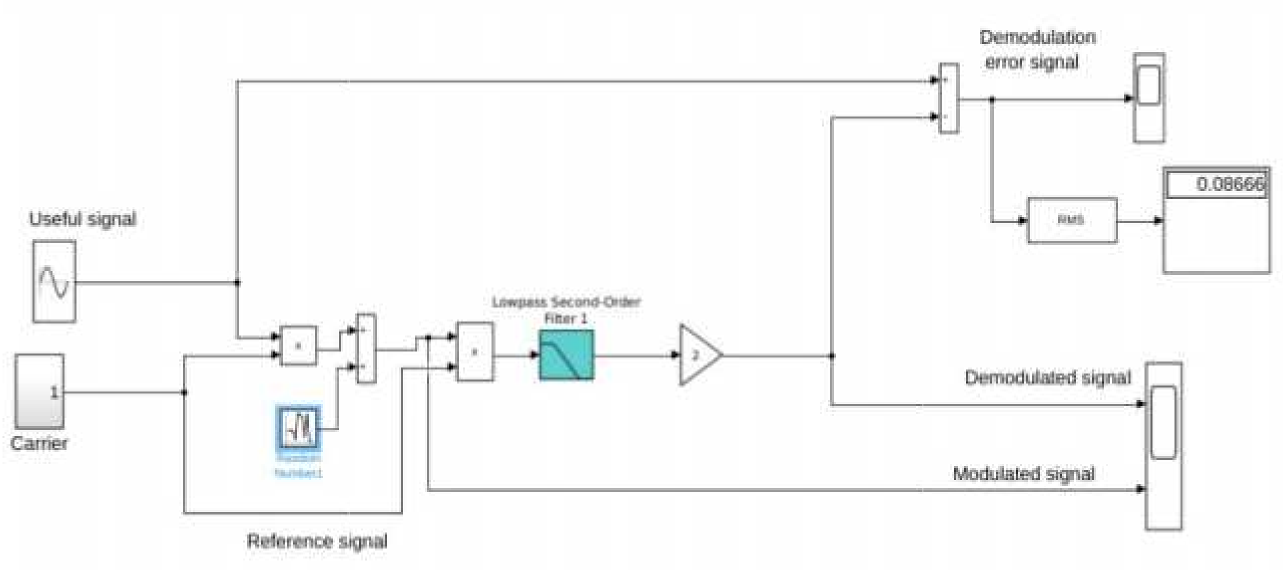

Figure 4.20. RMS error at input noise Variance 5

4) For Variance $7V^2$ we will get a block diagram with RMS equal 0.7603.

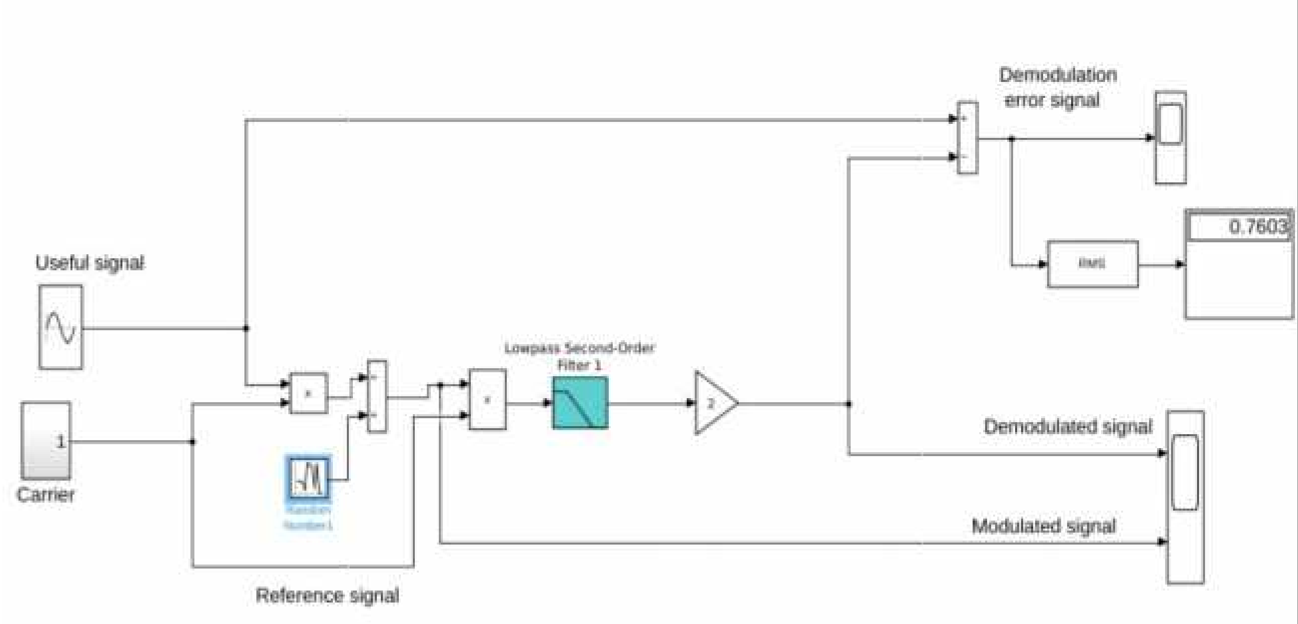

Figure 4.21. RMS error at input noise Variance 7

5) For Variance $10V^2$ we will get a block diagram with RMS equal 0.629.

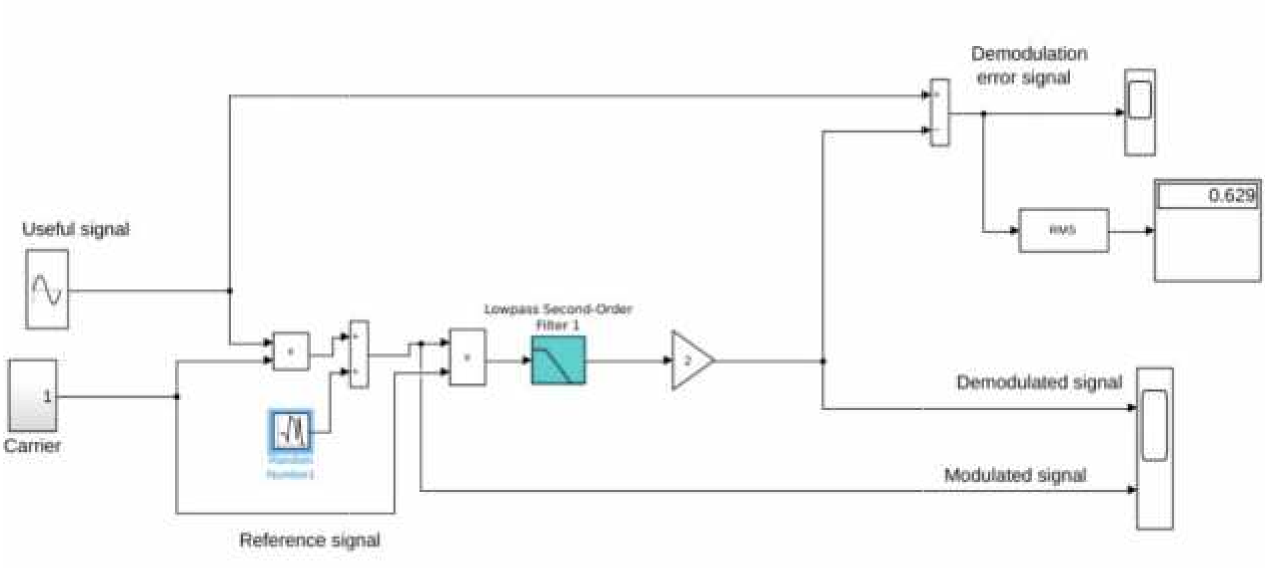

Figure 4.22. RMS error at input noise Variance 10

In the modulation analysis undertaken earlier, the alteration in the root mean square (RMS) value was examined. The non-linearity of the data indicates that the modulation process exhibits a non-linear behavior. The result you can see in the table (table 4.3.) represented below. Upon completion of data collection pertaining to amplitude demodulation during fluctuations in variance, a graph (fig. 4.23.) has been produced, which is visible below.

Table 4.3.

RMS error versus variance

Variance	RMS
$1V^2$	0.03144
$3V^2$	0.2251
$5V^2$	0.08666
$7V^2$	0.7603
$10V^2$	0.629

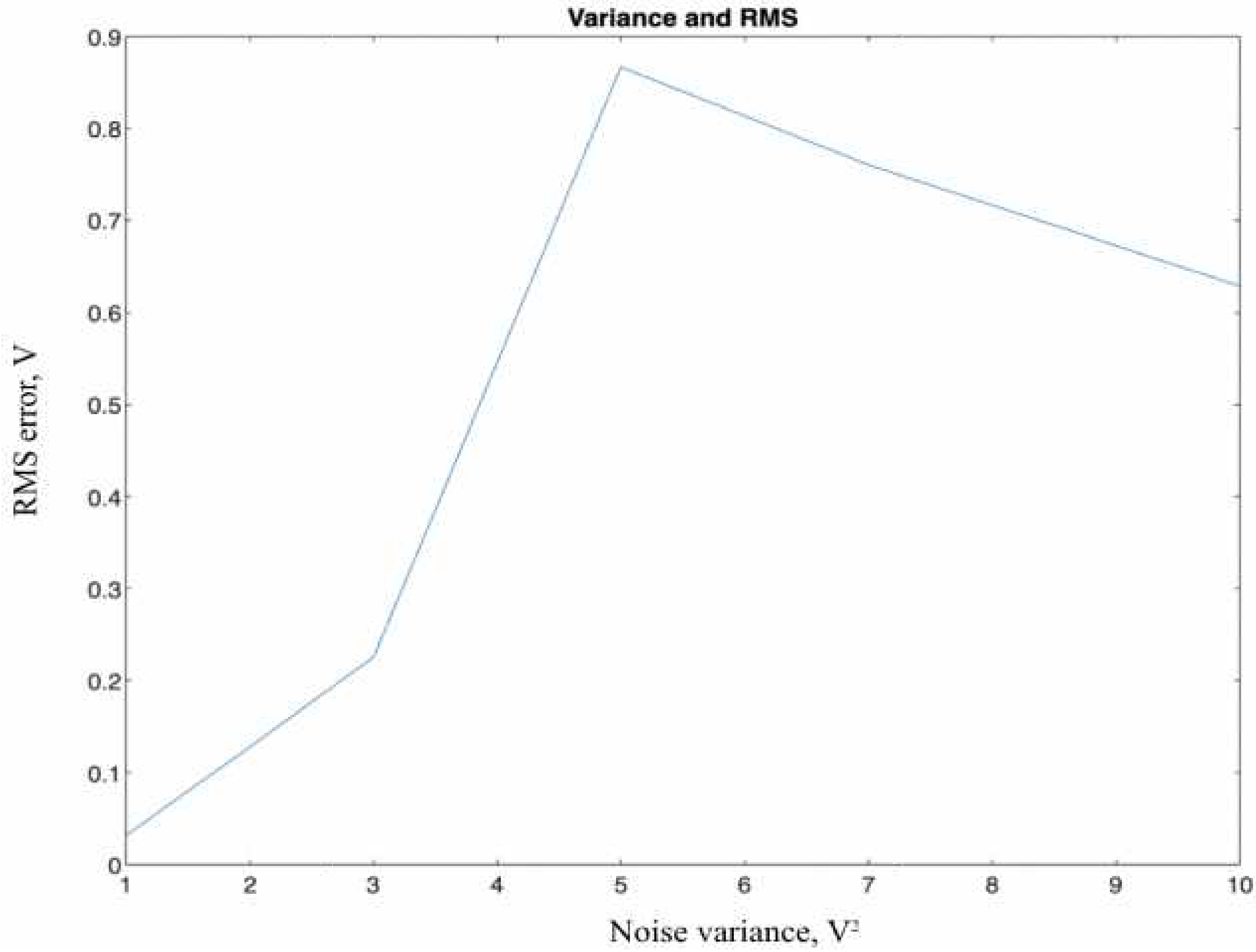

Figure 4.23. The final plot of RMS demodulation versus noise variance

4. For each item (1, 2, 3), present graphs showing Scope and Scope1 for the first values of the parameters (i.e., for 10V amplitude, for 10 Hz frequency, and for 1V^2 variation).

In regard to the current unit of study, it is imperative to utilize the outcomes obtained from the Scope and Scope1 modules. The ensuing illustrations depict the interplay between demodulation and modulation when considering the initial values of amplitude, frequency, and variance. The results of Scope we can see on figure 4.25 and results of Scope1 is on figure 4.26.

Fig. 4.25 shows graph of the demodulator output signal for a 10 V of modulation amplitude at the carrier frequency of 4 kHz.

The upper graph of fig. 4.26 shows the demodulator output signal for a 10 Hz modulation frequency at the carrier frequency of 4 kHz.

The lower graph of fig. 4.26 shows the demodulator output signal for variance value of the input noise of a $1V^2$ at the carrier frequency of 4 kHz.

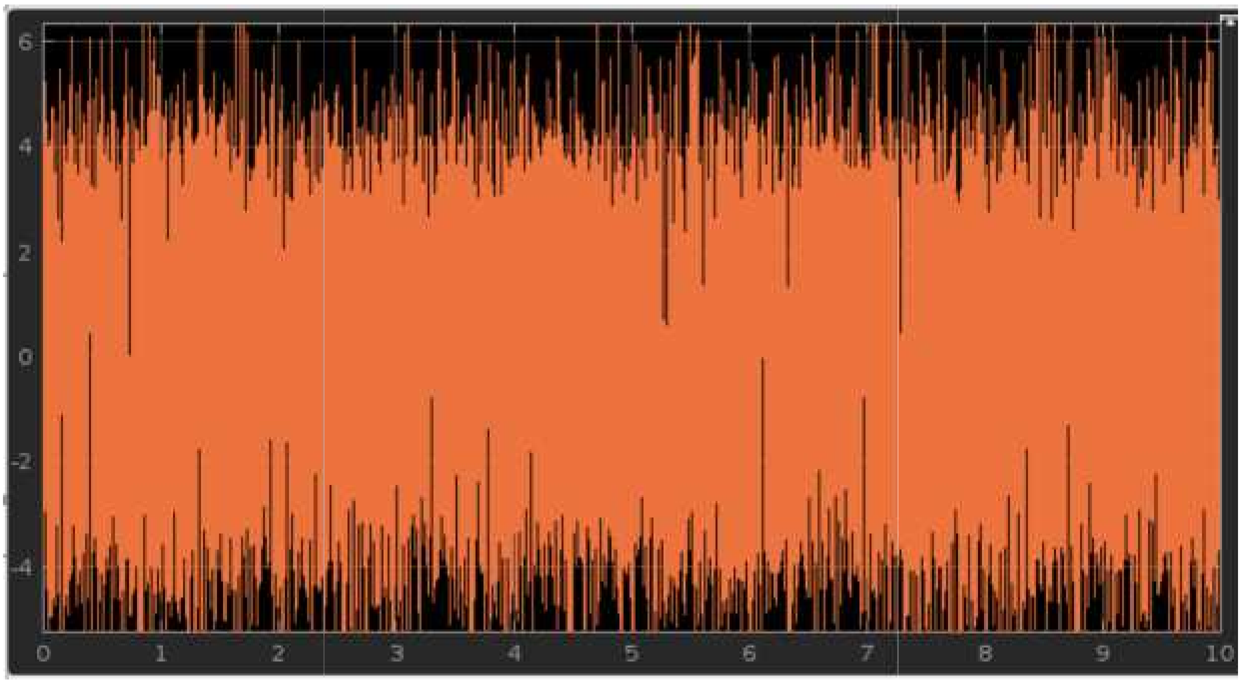

Figure 4.24. The demodulator output signal for a 10 V of modulation amplitude at the carrier frequency of 4 kHz

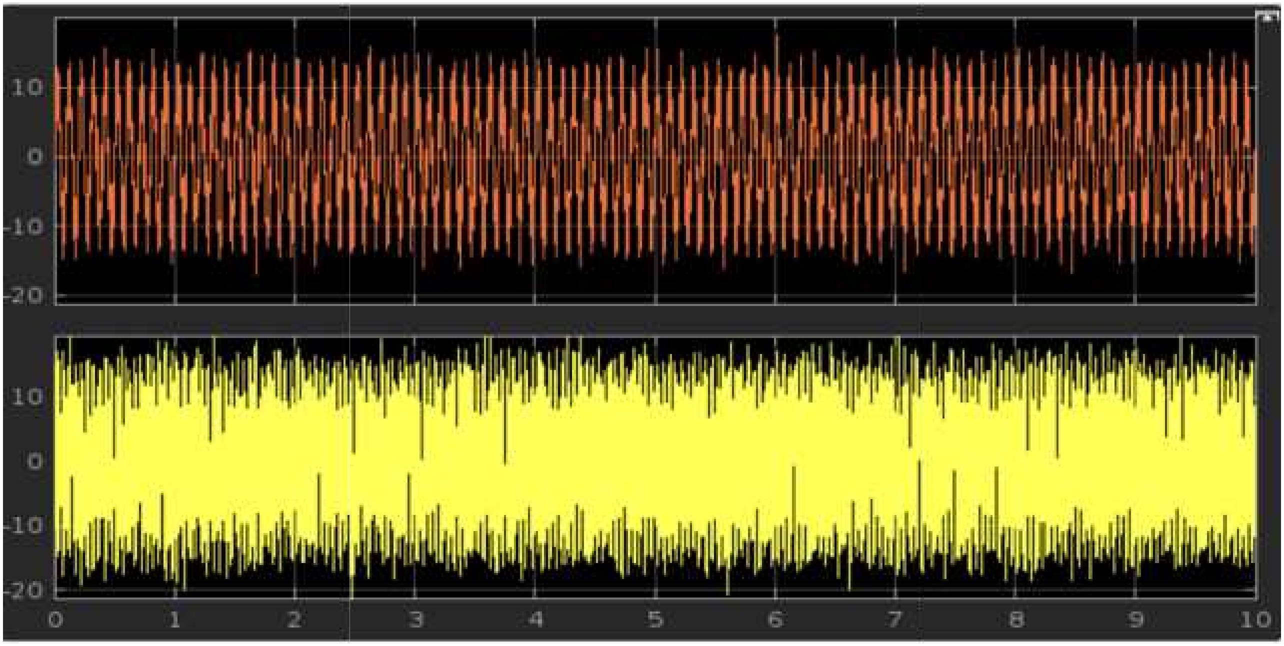

Figure 4.25. The Upper graph presents the demodulator output signal for a 10Hz modulation frequency at the carrier frequency of 4 kHz, and the lower graph presents the demodulator output signal for variance value of the input noise of a 1 V 2 at the carrier frequency of 4 kHz.

CONCLUSION

The present diploma thesis aims to investigate the complexities and methodologies related to signal amplitude demodulation within the context of variable frequency digital signals. This study aimed to create efficient and dependable algorithms that possess the ability to accurately demodulate signals with fluctuating carrier frequencies, despite the presence of noise, interference, and other impairments encountered during transmission.

The present study provides an elucidation of the procedures entailed in the execution of amplitude demodulation for digital signals that conform to predetermined standards. Moreover, the modulated signal corresponds to the visual portrayals, such as graphs and indicators, that are produced using the MATLAB software.

The model that has been successfully developed fulfills all the essential criteria and is considered appropriate for utilization in educational environments as well as the training of personnel involved in the amplitude demodulation of digital signals requiring a graphical representation.

Furthermore, the modernization of the package through the incorporation of contemporary Simulink graphical editor technology has resulted in several significant advancements. The utilization of Simulink provides a user-friendly and intuitive environment for designing and simulating complex systems, including amplitude demodulation of digital signals. The graphical nature of Simulink enables a visual representation of the demodulation process, allowing users to better understand and analyze the signal flow and system behavior.

An essential feature of the developed model is its ability to align the modulated signal with visual representations, including graphs and indicators, derived through MATLAB. This integration allows users to visualize the modulated signal, the demodulated output, and various intermediate signals at different stages of the demodulation process. The graphical representation enhances understanding, facilitates analysis, and aids in troubleshooting and fine-tuning the demodulation parameters.

In conclusion, the incorporation of Simulink graphical editor technology in the modernization of the amplitude demodulation package has resulted in a comprehensive and user-friendly model. The model not only satisfies the necessary prerequisites for accurate demodulation of digital signals but also provides a visual representation of the demodulation process. It holds great potential for educational settings, enabling effective teaching and learning, as well as for training personnel engaged in the demodulation of digital signals.

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APPENDIX

For this diploma thesis also exist addition MATLAB code that shows us how to create the plot of amplitude demodulation in Simulink.

```
>> amp = [10 20 30 40 50];
>> rms = [0.629 5.124 0.8174 2.319 1.316];
>>figure(1), plot(amp, rms), title('Amplitude and RMS'), xlabel('RMS error V), ylabel('Modulation amplitude, V')

>> freq = [ 10 30 50 80 100];
>> rms1 = [0.629 1.032 0.3135 2.58 1.395];
>> figure(2), plot(freq, rms1), title('Frequency and RMS'), xlabel('RMS error V), ylabel('Frequency, Hz')

>> var = [1 3 5 7 10];
>> rms2 = [0.03144 0.2251 0.8666 0.7603 0.629];
>> figure(3), plot(var, rms2), title('Variance and RMS'), xlabel('RMS error V'), ylabel('Noise variance V^2')
```