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POLITEKNIK MALAYSIA
SULTAN MIZAN ZAINAL ABIDIN

MODULATION TECHNIQUES

LIMITED ★★★★★ EDITION

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MODULATION
TECHNIQUES

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Bismillahirrahmanirrahim

First and foremost I am grateful to the Almighty God, the beneficence and the Most Merciful who makes all things possible. I appreciate my well-wishers for their support physically and morally in making this text success. Also acknowledge the support received from staff of Electrical Engineering Department, PSMZA.

This book is designed to assist Diploma in Electrical Engineering students in understanding modulation techniques, a concept vital to various forms of communication. The book follows the Communication System Fundamentals syllabus of the Department Polytechnic of Education and Community College (JPPKK), Ministry of Higher Education Malaysia. It contains notes, exercises, and practical implementation software. The authors express their appreciation to all who contributed to the publishing of the book and invite constructive comments from readers.

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Bismillahirrahmanirrahim

Congratulations to the writters for successfully publishing e-book '*Modulation Techniques*' as a reference book to semester 3 students in Diploma Electronics and Electrical Engineering and Diploma in Electronics (Communication) Engineering Polytechnic, KPTM. It is hoped that this effort can be used as a source of inspiration for lecturers to share knowledge through the field of writing.

As we all know, the publishing industry is always changing parallel to the development of technology. From manual production through handwriting in small quantities and takes time long to complete, publishing products are now available produced easily in more time short through *e-book*. Nowsdays, the implementation of PdP as well as digital programme have been applied successfully. Enabling digitization in Education Technical and Vocational is the new norm for Polytechnic and Kolej Komuniti citizens.

A book of excerpts from the author that touches on various modulation methods this can expose students to basic knowledge of communication elements and construct and test various applications of related communication assigned in practical work The production of scholarly books can be used as a reference source to academics as well as professional and administrative groups. Well done to the line of book writers and may it continue succeeded in producing books that became reference sources to all.

TS. MOHYIDDIN BIN SALLEH
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Executive Summary

SYPNOSIS

"Modulation Techniques is a book that introduces readers to communication systems. This e-book covers the principles of communications as well as analog and digital modulation techniques which is design to Diploma Electrical (Communication).

Course learning outcome

Upon completion of this book it is hoped that the readers will be able to apply the basic concept of communication system elements, various types of modulation techniques, and basic data communication in electronic communication by using appropriate diagram.

1. Apply the concept of electronic communication system by using appropriate diagram and standard formula
2. Assemble the related communication equipment systematically in performing the measurement of appropriate signals parameter
3. Demonstrate the ability to work in a team to complete the assigned tasks during practical work sessions

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01.MODULATION AND DEMODULATION

WHAT IS MODULATION?

Modulation is the process of modifying a carrier signal with the information-bearing signal to transmit the information over a distance. It involves altering one or more properties of the carrier signal, such as its amplitude, frequency, or phase, to encode the information signal.

The purpose of modulation is to enable long-distance communication and improve the efficiency of the transmission. Modulation allows the use of a wide range of frequencies to transmit the signal, and it reduces interference from other signals in the environtment.

Modulation plays a vital role in many forms of communication, including radio and television broadcasting, cellular telephony, satellite communications, and digital communication systems such as Wi-Fi and Bluetooth.

DEFINITION OF MODULATION

Modulation is a process of changing one or more properties of the high frequency analog carrier signal in proportion with the values of information signal. The information (modulating) signal modulates the carrier signal by changing either its amplitude, frequency, or phase to produce modulated signal .

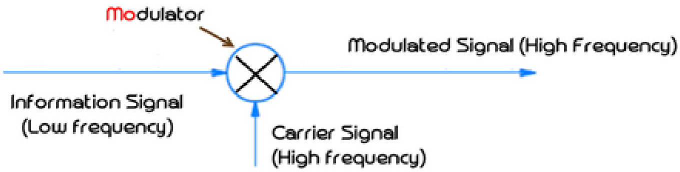


Figure 1.1 Modulation process

Modulation is performed in a transmitter by a circuit called a Modulator. The information can be in analog or digital form, and the modulator can perform either analog or digital modulator. The information signal combines with the carrier in the modulator to produce a high frequency modulated signal.

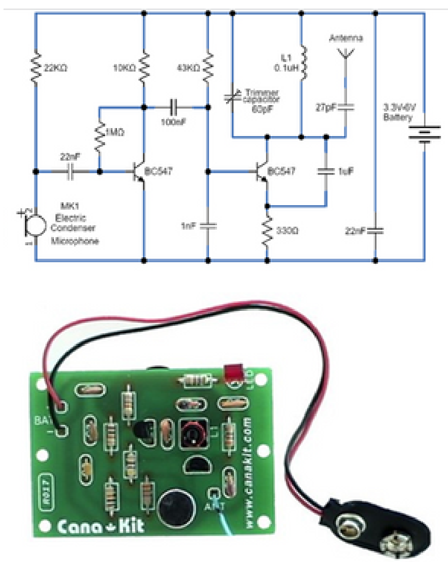


Figure 1.2 Multipurpose FM transmitter circuit

The FM (frequency modulation) transmitter is a low-power transmitter, in frequency modulation, the data is transferred by varying the frequency of carrier waves and it uses FM waves for transmitting the sound/data. This transmitter transmits the audio signals through the carrier wave by the difference of frequency. We design a simple multipurpose FM transmitter circuit by using two NPN transistors and a few easily available components. The performance and working of the wireless audio transmitter circuit depend on the induction coil & variable capacitor, this circuit oscillates up to 100 MHz and you can tune and adjust the oscillating frequency by varying Trim Capacitor.

DEMODULATION

Demodulation is the process of recovering the original information signal from a modulated carrier signal. In other words, it is the reverse process of modulation.

After modulation, the carrier signal has been altered in some way to carry the information signal. The demodulation process is necessary to extract the original signal from the modulated signal, so it can be understood by the receiving device.

The type of demodulation used depends on the type of modulation used in the transmission. For example, if the modulation is AM, the demodulation process involves detecting changes in the amplitude of the carrier signal. Similarly, in FM, the frequency changes of the carrier signal are detected to recover the original signal.

Demodulation is a crucial part of communication systems, as it allows the receiver to extract and interpret the original message that has been transmitted. Demodulation techniques are used in various applications, including radio and television broadcasting, wireless communication, and digital communication systems such as Wi-Fi and Bluetooth.

DEFINITION OF DEMODULATION

Demodulation is the reverse process of modulation. It is a process extracting the information signal from the modulated-carrier signal

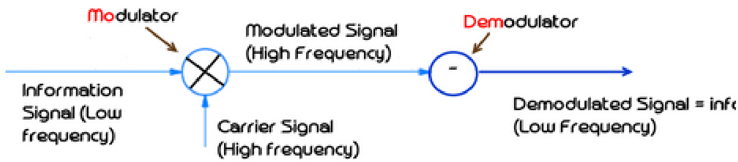


Figure 1.3 Demodulation process

Demodulation is performed in a receiver by a circuit called Demodulator. Demodulated signal is the process to generate the original Information Signal

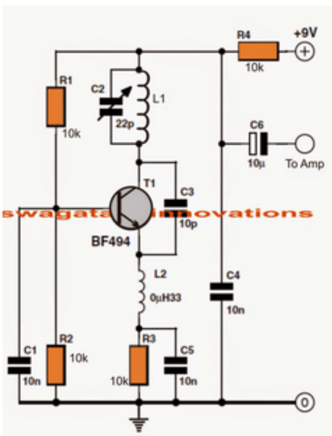


Figure 1.4 FM receiver circuit

The function of an FM receiver is to detect, demodulate, and amplify the FM signal transmitted from a radio station and convert it into an audio signal that can be heard through a speaker or headphones.

The FM receiver consists of several components, including an antenna, tuner, mixer, local oscillator, intermediate frequency (IF) amplifier, demodulator, audio amplifier, and speaker. Here's a brief explanation of each component's function in an FM receiver:

- Antenna: It captures the FM signal from the radio station and converts it into an electrical signal.

Tuner: It selects a specific frequency from the FM band that the user wants to listen to.

Mixer: It combines the selected FM signal with a local oscillator signal to produce a lower frequency signal, which is the intermediate frequency (IF) signal.

Local Oscillator: It generates a signal at a specific frequency that is mixed with the incoming FM signal to produce the IF signal.

IF Amplifier: It amplifies the IF signal to increase its strength and filter out unwanted frequencies.

Demodulator: It extracts the audio signal from the FM signal by separating the frequency variations in the signal.

Audio Amplifier: It amplifies the audio signal to a level that can drive the speaker or headphones.

Speaker: It converts the amplified audio signal into sound waves that can be heard by the listener.

The FM receiver's primary function is to receive, process and convert the FM signal into an audio signal that can be heard by the user. It is an essential component of FM radio broadcasting and is widely used in various applications, including communication systems, broadcasting, and entertainment.

Why Modulation is necessary?

It is extremely difficult to radiate low-frequency signals from an antenna in the form of electromagnetic energy. So we need to increase the frequency of information signal by doing the modulation process. Analog signals are continuous signals that vary over time and can take any value within a range. Digital signals, on the other hand, are discrete signals that take specific values at specific intervals of time.

To convert the analog signal to digital signal and vice versa for matching with communication medium and communication needs. Here are some reasons why:

- Bandwidth efficiency: A digital signal can carry more information per unit time than an analog signal, which means that it can transmit more data over a given channel. This is important for communication systems with limited bandwidth, such as wireless communication or satellite communication.

- Noise immunity: Digital signals are less susceptible to noise and interference than analog signals. They can be easily encoded and decoded using error-correction techniques, which makes them more robust and reliable in noisy environments.

- Compatibility: Digital signals are compatible with modern communication systems and devices, such as computers, smartphones, and the internet. Most modern communication systems use digital signals to transmit and receive data.

- Signal processing: Digital signals can be easily processed using digital signal processing (DSP) techniques. DSP allows for advanced signal filtering, modulation, and demodulation techniques that are not possible with analog signals.

You explained the importance of converting analog signals to digital signals and vice versa in order to match communication needs and characteristics. Digital signals offer advantages over analog signals such as being more bandwidth-efficient, noise-immune, and compatible with modern communication systems, and allowing for advanced signal processing techniques.

To reduce equipment complexity

The modulation process is essential to simplify the communication system's design and reduce costs. This can be achieved by using integrated circuits, software-defined radio, digital modulation techniques, frequency synthesizers, and system integration.

To reduce the antenna height and size

The height of the antenna must be a multiply of $\lambda/4$.

$h = \lambda/4$

where λ is wavelength ($\lambda = c/f$),

c is the velocity of light

and f is the frequency of the signal to be transmitted.

An example:

Let us consider two signals. One signal is modulated and the other is not modulated. The frequency of original baseband signal is taken as f = 10 kHz while the modulated signal is f = 1 MHz.

The height of the antenna required to transmit the original baseband signal of frequency f = 10 kHz:

$$h = \lambda/4 = c/4f = (3x10^8)/(4x10x10^3) = 7.5 \text{ km} \ldots\ldots \text{(i)}$$

The height of the antenna required to transmit the modulated signal of frequency f = 1 MHz:

$$h = \lambda/4 = c/4f = (3x10^8)/(4x10x10^6) = 75 \text{ m} \ldots\ldots\ldots \text{(ii)}$$

Comparing equation (i) and (ii), We can infer that it is practically feasible to construct an antenna of height 75 m while the one with 7.5 km is not possible. It clearly manifest that modulated signals reduce the antenna height and are required for long distance transmission.

Information signals often occupy the same frequency band. If signals from two or more sources are transmitted at the same time, they would interfere with each other. with each other, each station(source) converts its information to a different frequency band or channel by modulation process.

Modulation techniques can be used to avoid interfering with other frequency bands. Here are some ways to avoid interfering with other frequency bands using modulation techniques:

- Frequency Division Multiplexing (FDM): FDM is a modulation technique that allows multiple signals to share the same transmission medium by dividing the frequency band into smaller sub-bands. By using FDM, each signal can operate in its own sub-band, avoiding interference with other frequency bands.

- Time Division Multiplexing (TDM): TDM is a modulation technique that allows multiple signals to share the same transmission medium by dividing the transmission time into smaller time slots. By using TDM, each signal can operate during its own time slot, avoiding interference with other signals.

- Spread Spectrum Modulation: Spread Spectrum Modulation is a modulation technique that spreads the signal over a wider frequency band than is necessary for transmission. By spreading the signal over a wider frequency band, the signal becomes more resistant to interference and can avoid interfering with other frequency bands.

- Orthogonal Frequency Division Multiplexing (OFDM): OFDM is a modulation technique that divides the frequency band into many narrow sub-bands and modulates each sub-band separately. By using OFDM, each sub-band can operate independently, avoiding interference with other frequency bands.

In summary, modulation techniques can be used to avoid interfering with other frequency bands. This can be achieved through techniques such as FDM, TDM, Spread Spectrum Modulation, Adaptive Modulation, and OFDM.

TYPES OF MODULATION TECHNIQUES

Modulation is the process of modifying a signal to carry information from a source to a destination. In telecommunications, modulation techniques are used to modify the characteristics of an electromagnetic wave or signal in order to encode information for transmission over a communication channel.

There are several types of modulation techniques used in telecommunications. The most common types of modulation techniques are:

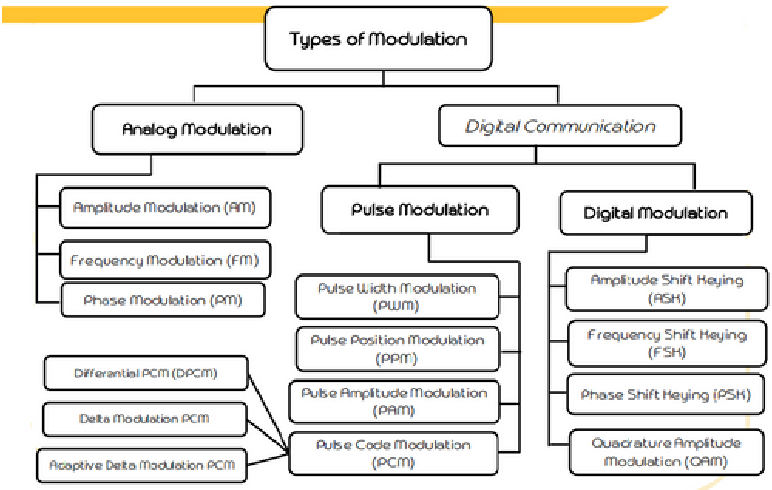


Figure 1.5 Types of modulation techniques

Analog Modulation

Analog modulation is the process of modifying an analog signal, such as a voice or music signal, to transmit it over a communication channel. In analog madulation, both information signal are in analog form

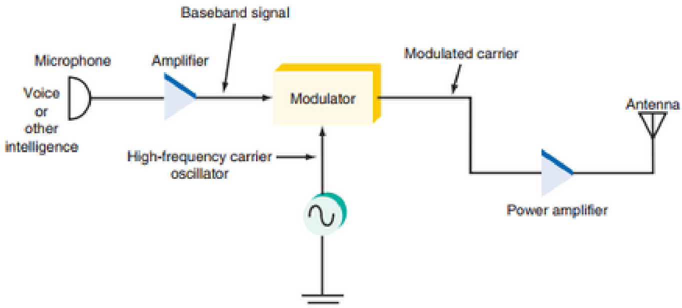


Figure 1.6 Analog Modulation tranmission techniques

Analog modulation is the process of modifying an analog signal, such as a voice or music signal, to transmit it over a communication channel. The most common types of analog modulation are:

- Amplitude Modulation (AM) - In AM, the amplitude of a high-frequency carrier wave is varied in proportion to the instantaneous amplitude of the message signal. The modulated signal can be demodulated by extracting the envelope of the carrier wave.

- Frequency Modulation (FM) - In FM, the frequency of the carrier wave is varied in proportion to the instantaneous amplitude of the message signal. The modulated signal can be demodulated by detecting the frequency deviations of the carrier wave.

- Phase Modulation (PM) - In PM, the phase of the carrier wave is varied in proportion to the instantaneous amplitude of the message signal. The modulated signal can be demodulated by detecting the phase deviations of the carrier wave.

Analog modulation techniques are used in many applications, such as broadcast radio and television, and mobile communication systems. However, they are susceptible to noise and interference, and have limited bandwidth efficiency compared to digital modulation techniques.

Advantages of analog modulation

Analog modulation has several advantages over digital modulation in certain applications:

- Compatibility: Analog modulation is compatible with the existing infrastructure of many communication systems, such as radio and television broadcast networks, and many legacy communication systems still use analog modulation.
- Simple implementation: Analog modulation is relatively simple to implement, as it requires only simple circuits and components, which can be more cost-effective than digital modulation systems.
- Natural representation of signals: Analog modulation techniques, such as amplitude modulation, provide a natural representation of analog signals such as voice and music, which are inherently continuous and vary smoothly over time.
- Robustness: Analog signals are generally more robust against signal degradation due to noise and interference compared to digital signals, as they can tolerate some level of noise and still remain intelligible.
- No quantization noise: Analog modulation does not suffer from quantization noise, which is a type of noise that can occur in digital modulation when the analog signal is quantized to discrete values.

Overall, analog modulation is still widely used in many applications where compatibility, simplicity, and natural representation of signals are important, although digital modulation is becoming increasingly popular due to its superior bandwidth efficiency and error-correcting capabilities.

Digital Modulation

Digital modulation is a technique used to transmit digital data over a communication channel. In digital modulation, the digital data is converted into a digital signal, which is then modulated onto a carrier wave for transmission. The modulated signal can then be transmitted over the communication channel and demodulated at the receiver to recover the original digital data.

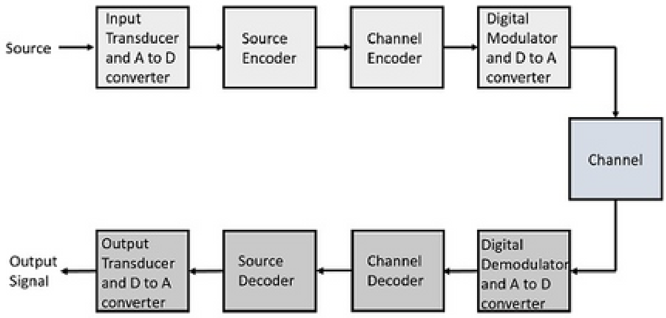


Figure 1.7 Basic Elements of Digital Communication System

There are several types of digital modulation techniques, including:

- Amplitude Shift Keying (ASK): In ASK, the amplitude of the carrier wave is varied to represent digital data.

- Frequency Shift Keying (FSK): In FSK, the frequency of the carrier wave is varied to represent digital data.

- Phase Shift Keying (PSK): In PSK, the phase of the carrier wave is varied to represent digital data.

- Quadrature Amplitude Modulation (QAM): In QAM, both the amplitude and phase of the carrier wave are varied to represent digital data.

The choice of digital modulation technique depends on factors such as the available bandwidth, the noise level, and the required data rate and reliability.

Advantages of Digital modulation

Digital modulation offers several advantages over analog modulation, including:

- Increased bandwidth efficiency: Digital modulation techniques can transmit more data in a given bandwidth compared to analog modulation techniques, making them more efficient.
- Improved error correction: Digital modulation techniques can use error-correcting codes to detect and correct errors that occur during transmission, which improves the reliability of the transmitted data.
- Greater security: Digital modulation techniques can use encryption to provide greater security for the transmitted data, which is important in many applications.

Applications of Digital modulation

Digital modulation techniques have a wide range of applications in modern communication systems, including:

- Wireless communications: Digital modulation techniques are widely used in wireless communication systems such as cellular networks, Wi-Fi, and satellite communication systems. They allow for high-speed data transfer and improved reliability over long distances.

- Digital broadcasting: Digital modulation techniques are used in digital broadcasting systems such as digital television (DTV) and digital radio (DAB) to provide better signal quality and improved compression of audio and video data.

- Computer networks: Digital modulation techniques are used in computer networks such as Ethernet and fiber-optic networks to transmit data between computers and other network devices.

- Internet of Things (IoT): Digital modulation techniques are used in IoT applications such as smart homes, smart cities, and industrial automation to transmit sensor data and control signals.

- Military communications: Digital modulation techniques are used in military communication systems to provide secure and reliable communication between military personnel and equipment.

- Medical devices: Digital modulation techniques are used in medical devices such as wireless implants and remote monitoring systems to transmit patient data and control signals.

Overall, digital modulation techniques are essential for modern communication systems, enabling high-speed data transfer, improved signal quality, and better reliability in a wide range of applications.

Pulse Modulation

Pulse modulation is a digital modulation technique that involves converting analog signals into a series of pulses for transmission. In pulse modulation, the analog signal is sampled at regular intervals, and each sample is represented by a pulse with a specific amplitude, width, and position in time. The pulses are then transmitted over a communication channel using a digital modulation technique.

There are several types of pulse modulation, including:

- Pulse Amplitude Modulation (PAM): In PAM, the amplitude of the pulses is proportional to the amplitude of the analog signal.
- Pulse Width Modulation (PWM): In PWM, the width of the pulses is proportional to the amplitude of the analog signal.
- Pulse Position Modulation (PPM): In PPM, the position of the pulses in time is proportional to the amplitude of the analog signal.
- Delta Modulation (DM): In DM, the difference between consecutive samples of the analog signal is represented by a series of pulses.

Advantages of Pulse modulation

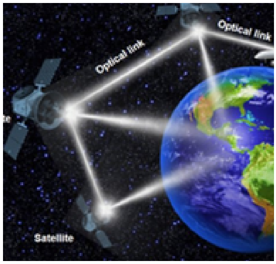
Pulse modulation has several advantages over analog modulation, including:

- Improved noise immunity: Pulse modulation techniques can use error-correcting codes and other signal processing techniques to reduce the effects of noise and interference on the transmitted signal.

- Higher bandwidth efficiency: Pulse modulation techniques can transmit more data in a given bandwidth compared to analog modulation techniques.

- Greater flexibility: Pulse modulation techniques can be easily adapted to different communication channels and data rates.

However, pulse modulation techniques can be more complex to implement than analog modulation techniques, and they may require more processing power and higher data rates to achieve the same level of quality as analog modulation techniques.


02 ANALOGUE MODULATION

Analog modulation is a process of modifying an analog signal, such as voice or video, by modulating the characteristics of a carrier signal to transmit the information over a communication channel. The analog signal is typically modulated by changing the amplitude, frequency, or phase of the carrier signal to create a new waveform that contains the information to be transmitted.

A summary of the various modulation technique is shown here:

$$\boxed{vc(t) = \overset{\text{AM}}{V_p} \sin(2\pi \overset{\text{FM}}{f} t + \overset{\text{PM}}{\theta})}$$

where :

V(ct) = time-varying sine wave of carrier signal voltage

A = peak amplitude (volts)

f = frrequency (hertz, Hz)

= phase shift (radians)

In Analog Modulation, both Information Signal and Carrier signal are in analog waveform. In ;

- Amplitude Modulation (AM) - the amplitude (Vp) of the analog carrier signal is varied proportional to the analog information signal.

- Frequency Modulation (FM) - the frequency (f) of the analog carrier signal is varied proportional to the analog information signal.
- Phase Modulation (PM) - the phase (Θ) of the analog carrier signal is varied proportional to the analog information signal

Amplitude Modulation

Definition: Amplitude Modulation (AM) is the process of changing the amplitude of analog carrier signal in proportion with the amplitude of the analog information signal.

In AM, the amplitude (V) of the carrier signal is varied proportional to the information signal. While the frequency (f) and phase (Θ) of carrier signal are remains unchanged. The carrier amplitude is simply changed according to the amplitude of the information signal. When the information signals amplitude is increased, the carrier signal amplitude also increased and vice versa.

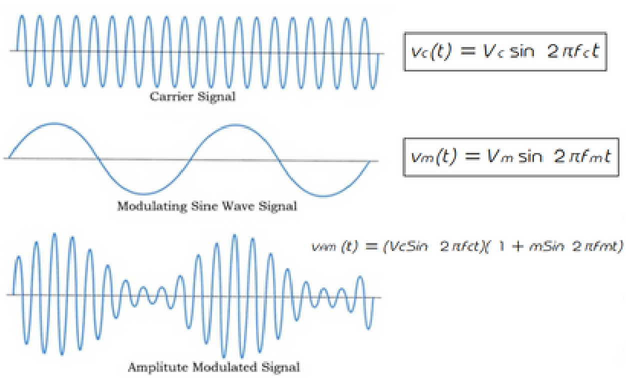


Figure 2.1 Amplitude Modulation

An AM (Amplitude Modulation) circuit is used to transmit information over a carrier signal by varying the amplitude of the carrier signal. The modulator circuit is the part of the transmitter responsible for producing the modulated signal. There are various types of modulator circuits, but here is a basic circuit diagram for a simple AM modulator using a transistor:

AM transmitter takes the audio signal as an input and delivers amplitude modulated wave to the antenna as an output to be transmitted. The block diagram of AM transmitter is shown in the following figure.

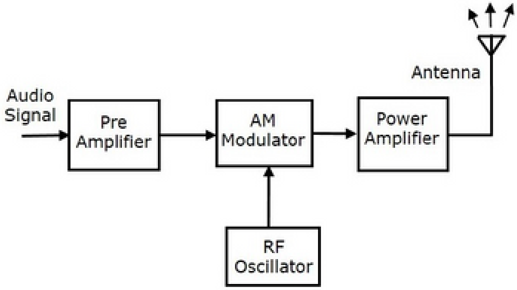


Figure 2.2 AM transmitter

The working of AM transmitter can be explained as follows.

- The audio signal from the output of the microphone is sent to the pre-amplifier, which boosts the level of the modulating signal.
- The RF oscillator generates the carrier signal.
- Both the modulating and the carrier signal is sent to AM modulator.
- Power amplifier is used to increase the power levels of AM wave. This wave is finally passed to the antenna to be transmitted.

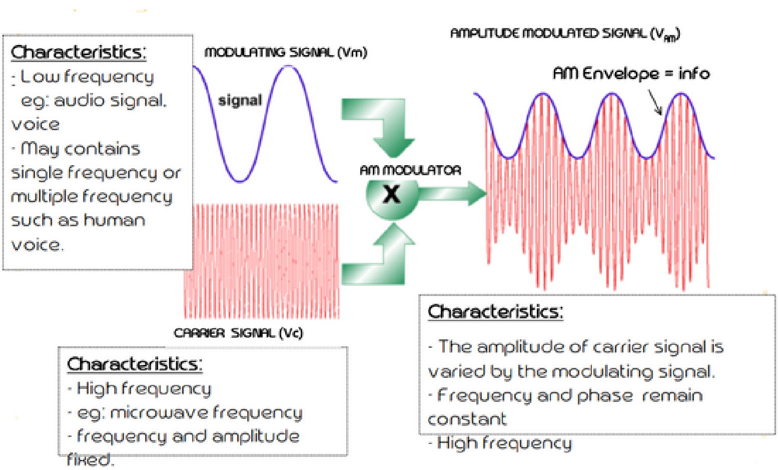


Figure 2.3 AM Process

Modulation Index and Percentage of Modulation

To compute the modulation index from measurements taken on the composite modulated. The modulation index can be computed from Vmax and Vmin,

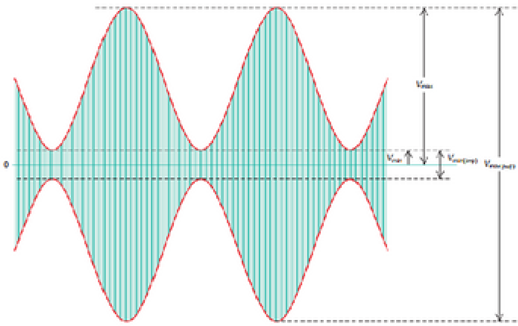


Figure 2.4 AM Waveform

The peak value of the carrier signal Vc is the average of the Vmax and Vmin values:

$$V_c = \frac{V_{\max} + V_{\min}}{2}$$

The modulation index is

$$m = \frac{V_{\max} - V_{\min}}{V_{\max} + V_{\min}}$$

The relationship, known as the modulation index m (also called the modulating factor or coeficient, or the degree of modulation), is the ratio

$$m = \frac{V_m}{V_c}$$

The peak values of the signals, and the carrier voltage is the unmodulated value. Multiplying the modulation index by 100 gives the percentage of modulation.

Example of Modulation Index and Percentage of Modulation

Suppose that onan Am signal, the Vmax(p-p) value read from the graticule on the oscilloscope screen is 5.9 divisions and Vmin(p-p) is 1.2 divisions.

a. What is the modulation index?

$$m\frac{V_{\max} - V_{\min}}{V_{\max} + V_{\min}} = \frac{5.9 - 1.2}{5.9 + 1.2} = \frac{4.7}{7.1} = 0.662$$

b. Calculate Vc, Vm, and m if the vertical scale is 2 V per division. (hint: Sketch the signal.)

$$V_c = \frac{V_{\max} + V_{\min}}{2} = \frac{5.9 + 1.2}{2} = \frac{7.1}{2} = 3.55 \ @ \ \frac{2\text{ V}}{\text{div}}$$

$$V_c = 3.55 \times 2\text{ V} = 7.1\text{ V}$$

$$V_m = \frac{V_{\max} - V_{\min}}{2} = \frac{5.9 - 1.2}{2} = \frac{4.7}{2}$$

$$= 2.35 \ @ \ \frac{2\text{ V}}{\text{div}}$$

$$V_m = 2.35 \times 2\text{ V} = 4.7\text{ V}$$

$$m = \frac{V_m}{V_c} = \frac{4.7}{7.1} = 0.662$$

A single-frequency sine wave modulating signal, generates two side bands i.e a whole range of frequencies modulate the carrier, and thus a whole range of side bands . The upper side band fUSB and lower side band fLSB are computed as ;

$$f_{\text{USB}} = f_c + f_m \quad \text{and} \quad f_{\text{LSB}} = f_c - f_m$$

The existence of sidebands can be demonstrated mathematically

$$v_{AM} = V_c \sin 2\pi f_c t + \frac{V_m}{2} \cos 2\pi t(f_c - f_m) - \frac{V_m}{2} \cos 2\pi t(f_c + f_m)$$

Observing an AM signal on an oscilloscope, you can see the amplitude variations of the carrier with respect to time. This time-domain display gives no obvious or outward indication of the existence of the sidebands, although the modulation process does indeed produce them, as the equation above shows.

Example of a simple side band

A standard AM broadcast station is allowed to transmit modulating frequencies up to 5 kHz. If the AM station is transmitting on a frequency of 980 kHz, compute the maximum and minimum upper and lower side bands and the total bandwidth occupied by the AM station.

$$f_{USB} = 980 + 5 = 985 \text{ kHz}$$
$$f_{LSB} = 980 - 5 = 975 \text{ kHz}$$
$$BW = f_{USB} - f_{LSB} = 985 - 975 = 10 \text{ kHz}$$
$$BW = 2(5 \text{ kHz}) = 10 \text{ kHz}$$

Exercise

A 400W, 1MHz carrier is amplitude-modulated with a sinusoidal signal 0f 2500Hz. The depth of modulation is 75%. Calculate the sideband frequencies, bandwidth in modulated wave.

Upper sideband Frequency

$$\therefore f_{USB} = f_c + f_m$$

$$\therefore f_{USB} = 1002.5 \ KHz$$

Lower sideband Frequency

$$\therefore f_{LSB} = f_c - f_m$$

$$\therefore f_{LSB} = 997.5 \ KHz$$

Bandwith

$$\therefore BW = 2f_m$$

$$\therefore BW = 2 \times 2.5KHz = 5KHz$$

EXPERIMENT : AMPLITUDE MODULATION (AM)

OBJECTIVES :

1. To understand and analyze the characteristics of modulating signal and carrier signal.
2. To identify, understand and analyze the characteristics of the amplitude Modulated Signal.

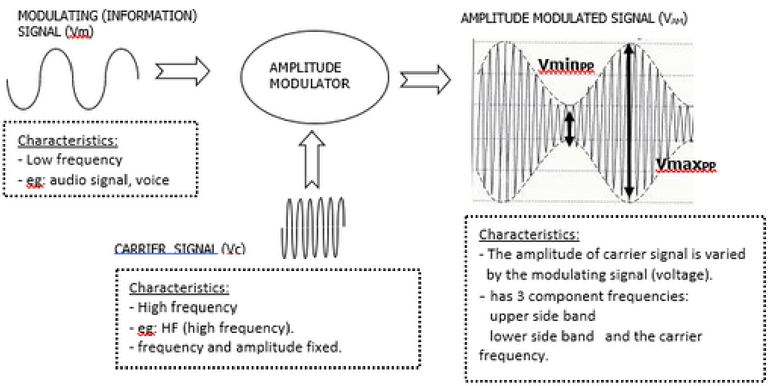
EQUIPMENTS :	Unit
Computer	1
Mouse	1

THEORY :

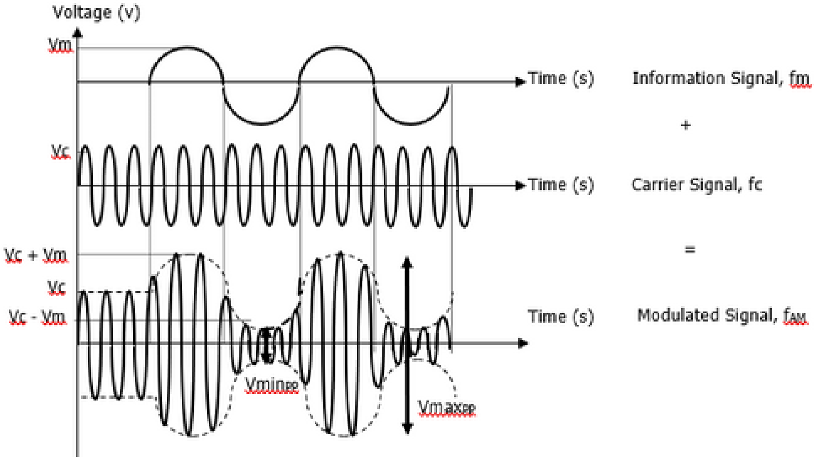
Amplitude Modulation

Modulation is simply the process of changing one or more properties of the analog carrier in proportion with the information signal.

Amplitude Modulation (AM) is the process of changing the amplitude of a relatively high frequency carrier signal in proportion with the instantaneous value of the information signal.



Below figure shows the process of amplitude modulation where the modulated signal is AM double-sideband full carrier (DSBFC).



In AM, the amplitude (V) of the carrier signal is varied proportional to the information signal. The carrier amplitude is simply changed according to the amplitude of the information signal. When the information signals amplitude is increased, the carrier signal amplitude also increased and vice versa.

The depth of the modulation is called Modulation Index, m where;

$$V_{max} = V_c + V_m$$

$$V_{min} = V_c - V_m$$

$$m = \frac{V_{max} - V_{min}}{V_{max} + V_{min}}$$

$$= \frac{(V_c + V_m) - (V_c - V_m)}{(V_c + V_m) + (V_c - V_m)} \qquad \textbf{OR} \qquad m = \frac{Vmax_{pp} - Vmin_{pp}}{Vmax_{pp} + Vmin_{pp}}$$

$$m = \frac{V_m}{V_c}$$

Depending on the value of m, the AM can be classified into three categories:

a. If m < 1, it is known as under modulation.

b. If m = 1, it is an ideal case.

c. If m > 1, it is known as over modulation.

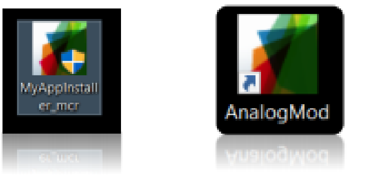
The values of the parameters can be extracted from the general equation of sinusoidal waveform or modulated equation that as stated below.

Modulating signal waveform, $V_m(t) = V_m \sin 2\pi f_m t$
Carrier signal waveform, $V_c(t) = V_c \sin 2\pi f_c t$
Amplitude Modulated Signal, $V_{AM}(t) = (V_c \sin 2\pi f_c t)(1 + m \sin 2\pi f_m t)$

PROCEDURE :

Setting of your computer

1. From the folder, click MyAppInstaller_mcr for installation. Please refer to Manual Installation AnalogMod for installation guidance.
2. After finish installation, click the AnalogMod.exe to run the program.



Procedure in AnalogMod.exe

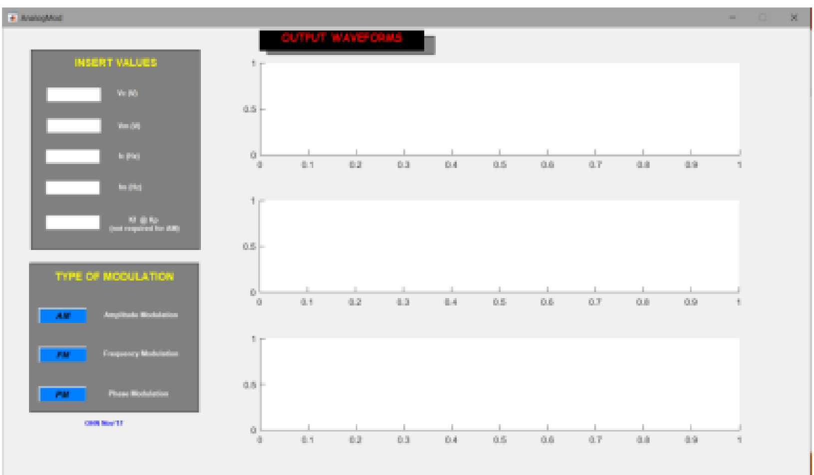


Figure 1

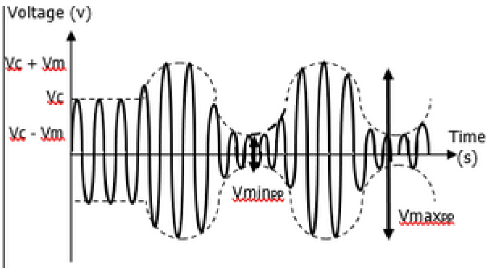
1.	Given an audio signal, v(t) = 6 sin 2π(3000)t modulates a high frequency carrier signal , v(t) = 10 sin 2π(10k)t. Determine each of the parameters below and record the result in Table 1:
	a) Carrier Amplitude
	b) Modulating
	c) Carrier Frequency
	d) Modulating Frequency
	e) Modulation Index
	f) AM Modulated Signal Expression
2.	The value of the parameters can be entered in each parameter boxes as shown in Figure 1 except the sensitivity value for Amplitude Modulation (AM).
3.	Click the AM button in Types of Modulation.
4.	Sketch the amplitude modulated signal waveform in Table 1.
5.	Repeat step 1 to 4 except Vm = 10V in Table 2 and Vm = 20V in Table 3.

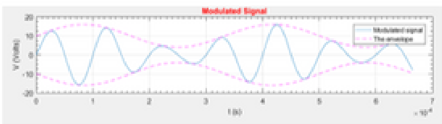
RESULT :

Modulating Depth < 1

AM waveform :

Table 1





Calculation :

a) Carrier Amplitude,
$V_c = 10V$

b) Modulating Amplitude,
$V_m = 6V$

c) Carrier Frequency,
$f_c = 10kHz$

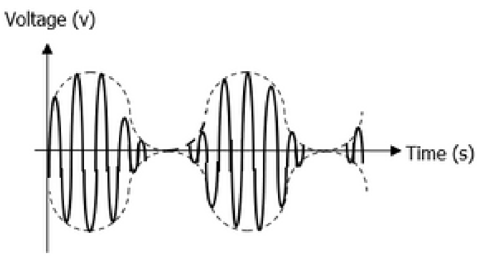
d) Modulating Frequency,
$f_m = 3kHz$

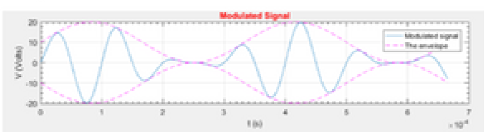
e) Modulation Index,
$m = Vm/Vc$
$= 6/10$
$= 0.6$

f) AM Modulated Signal Expression , VAM(t)
$= (VcSin2\pi fct)(1 + mSin2\pi fmt)$
$= (10 \text{ Sin } 2\pi(10k)t)(1 + 0.6 \text{ Sin}2\pi(3k)t)$

Modulating Depth = 1

Table 2





Calculation :

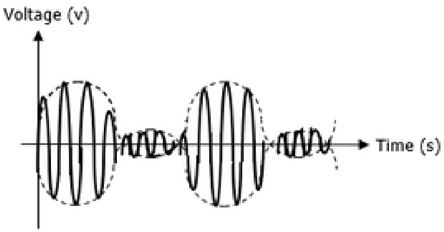
a) Carrier Amplitude, $V_c = 10V$

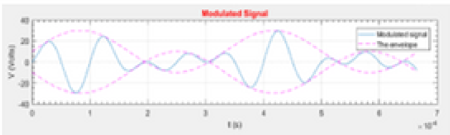
b) Modulating Amplitude,
$V_m = 10V$

c) Modulation Index,
$m = Vm/Vc$
$= 10/10$
$= 1$

Modulating Depth > 1

Table 3





Calculation :

a) Carrier Amplitude, Vc = 20V

b) Modulating Amplitude, Vm = 10V

c) Modulation Index, m = Vm/Vc
= 20/10
= 2

QUESTIONS & DISCUSSION:

1. Draw the block diagram of amplitude modulation process.
2. Can a modulation process be achieved if either one of above elements is do not exist? Why?
3. From the result in Table 1 to Table 3, what is the effect on the received signal?
4. Calculate the percentage of amplitude modulation index if the modulating signal is 3 Vpp and the carrier signal is 4 Vpp.

CONCLUSION:

1. What you can conclude with the elements that needed in amplitude modulation so the modulation could occur?
2. What you can conclude about the increasing of amplitude of modulating signal, Vm with the depth of modulation, m?
3. To reconstruct the information signal, what is the value of amplitude modulation index?
4. State the formula to find the modulation index, m?

Frequency Modulation

Frequency Modulation is the process of changing the frequency of analog carrier signal in proportion with the amplitude of the analog information signal. In Frequency Modulation, the carrier amplitude and phase remains constant while the carrier frequency is varied by the modulating signal. The amount of carrier frequency changes is proportional to the amplitude of the information signal. As the modulating signal amplitude increases, the carrier frequency increases and vice versa.

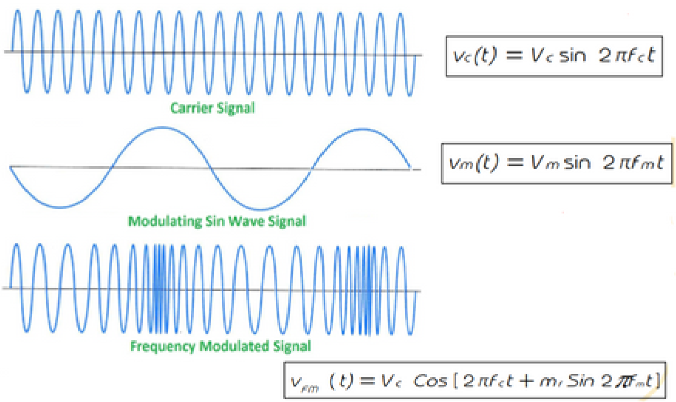


Figure 2.5 Frequency Modulation

In FM, the carrier amplitude remains constant and the carrier frequency is changed by the modulating signal. As the amplitude of the information signal varies, the carrier frequency shifts proportionately. As the modulating signal amplitude increases, the carrier frequency increases. If the amplitude of the modulating signal decreases, the carrier frequency decreases.

FM transmitter is the whole unit, which takes the audio signal as an input and delivers FM wave to the antenna as an output to be transmitted. The block diagram of FM transmitter is shown in the following figure.

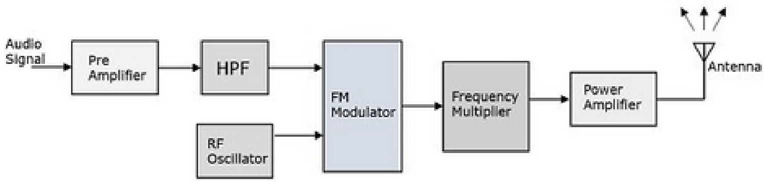


Figure 2.6 FM transmitter

The working of FM transmitter can be explained as follows.

- The audio signal from the output of the microphone is sent to the pre-amplifier, which boosts the level of the modulating signal.
- This signal is then passed to high pass filter, which acts as a pre-emphasis network to filter out the noise and improve the signal to noise ratio.
- This signal is further passed to the FM modulator circuit.
- The oscillator circuit generates a high frequency carrier, which is sent to the modulator along with the modulating signal.

Several stages of frequency multiplier are used to increase the operating frequency. Even then, the power of the signal is not enough to transmit. Hence, a RF power amplifier is used at the end to increase the power of the modulated signal. This FM modulated output is finally passed to the antenna to be transmitted.

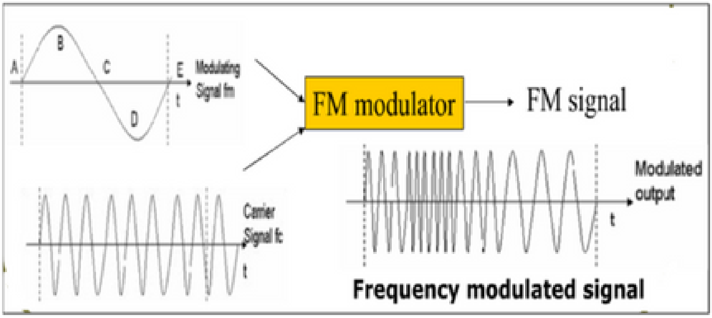


Figure 2.7 FM Modulator

An FM signal is illustrated in Fig. 2.7. Normally the carrier [Fig. 2.7(a)] is a sine wave, but it is shown as a triangular wave here to simplify the illustration. With no modulating signal applied, the carrier frequency is a constant-amplitude sine wave at its normal resting frequency.

The modulating information signal [Fig. 2.7(b)] is a low-frequency sine wave. As the sine wave goes positive, the frequency of the carrier increases proportionately. The highest frequency occurs at the peak amplitude of the modulating signal.

As the modulating signal amplitude decreases, the carrier frequency decreases. When the modulating signal is at zero amplitude, the carrier is at its center frequency point. When the modulating signal goes negative, the carrier frequency decreases. It continues to decrease until the peak of the negative half-cycle of the modulating sine wave is reached.

Then as the modulating signal increases toward zero, the carrier frequency again increases. This phenomenon is illustrated in Fig. 2.7(c), where the carrier sine waves seem to be first compressed and then stretched by the modulating signal.

Frequency modulation is used for sound broadcasting in the VHF band, for the sound signal of 625-line television broadcasting, for some mobile systems, and for multi-channel telephony systems operating in the UHF band. Frequency modulation (FM) refers to the process of varying the frequency of carrier signal with the change of amplitude modulating signal. The frequency of the carrier signal will become higher or lower according to the variation of input signal. If the modulating signal amplitude increases, the carrier frequency also increases or become higher and vice versa. During the modulation process, the amplitude of carrier signal is constant

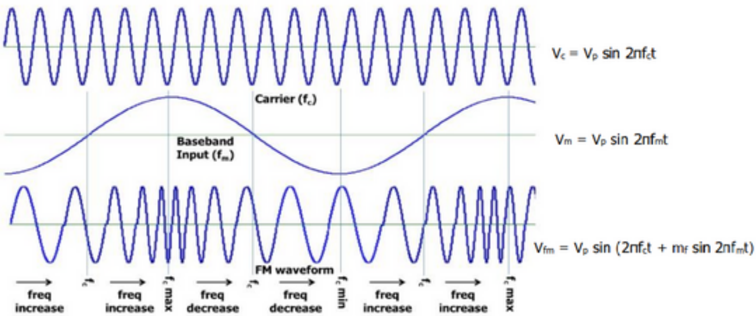


Figure 2.8 FM Modulation

The equation for modulating signal:

$$Vm = Vp \sin \omega mt = Vp \sin 2\pi fmt \ldots\ldots\ldots\ldots\ldots\ldots\ldots(1.1)$$

The equation for carrier signal:

$$Vc = Vp \sin \omega ct = Vp \sin 2\pi fct \ldots\ldots\ldots\ldots\ldots\ldots\ldots(1.2)$$

The equation for FM signal:

$$Vfm = Vp \sin (2\pi fct + mf \sin 2\pi fmt) \ldots\ldots\ldots\ldots\ldots\ldots(1.3)$$

where :

Vp = peak voltage

mf = Modulation index FM

$$mf = \Delta fd / fm = \ldots Kf\ Vm / fm \ldots\ldots\ldots\ldots\ldots\ldots\ldots(1.4)$$

fm – modulating signal frequency; Δfd – frequency deviation ;

$$\Delta fd = \frac{|fmax - fmin|}{2} \quad ; \text{@ } Kf\ Vm \text{ ....(1.5)}$$

Kf – Sensitivities
Vm – Modulating Signal Amplitude.

In FM, the frequency deviation is directly proportional to the amplitude of the modulating signal

Relationship Between the Modulating Signal and Carrier deviation

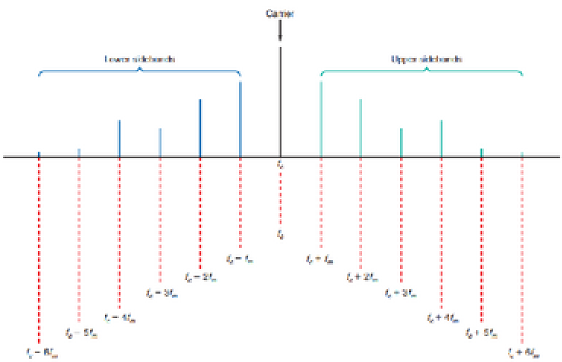


Figure 2.9 Frequency spectrum of an FM signal. Note that the carrier and sideband amplitudes shown are just examples. The amplitudes depend upon the modulation index mf.

Fig. 2.9 shows the frequency spectrum of a typical FM signal produced by modulating a carrier with a single-frequency sine wave. Note that the sidebands are spaced from the carrier fc and from one another by a frequency equal to the modulating frequency fm. If the modulating frequency is 1 kHz, the first pair of sidebands is above and below the carrier by 1000 Hz. The second pair of sidebands is above and below the first sidebands.

As the amplitude of the modulating signal varies, the frequency deviation changes. The number of sidebands produced, and their amplitude and spacing, depends on the frequency deviation and modulating frequency.

Theoretically, the FM process produces an infinite number of upper and lower sidebands and, therefore, a theoretically infinitely large bandwidth. However, in practice, only those sidebands with the largest amplitudes are significant in carrying the information.

Modulation Index

The ratio of the frequency deviation to the modulating frequency is known as the modulation index mf:

$$mf = fd/ fm$$

where fd is the frequency deviation
Fm is the modulating frequency.

Exercise

An FM wave is given by s(t)=20cos(8π×10^6t+9sin(2π×10^3t)). Calculate the frequency deviation, bandwidth, and power of FM wave.

Solution ;

Given, the equation of an FM wave as

$$s(t)=20\cos(8\pi\times106t+9\sin(2\pi\times10^3t))$$

We know the standard equation of an FM wave as

$$s(t)=Ac\cos(2\pi fct+\beta\sin(2\pi fmt))$$

We will get the following values by comparing the above two equations.

Amplitude of the carrier signal, Ac = 20VAc
= 20V

Frequency of the carrier signal,

$fc=4×106Hz=4MHzfc$

$=4×106Hz$

$=4MHz$

Frequency of the message signal,

$fm=1×103Hz$

$=1KHzfm$

$=1×103Hz$

$=1KHz$

Modulation index, $\beta=9\beta=9$

Here, the value of modulation index is greater than one.
Hence, it is Wide Band FM.

We know the formula for modulation index as

$$\beta=\Delta f/fm$$

Rearrange the above equation as follows.

$$\Delta f=\beta fm$$

Substitute β and fm values in the above equation.

$$\Delta f=9×1K=9KHz$$

Therefore, frequency deviation, Δf is 9KHz
The formula for Bandwidth of Wide Band FM wave is

$$BW=2(mf+1)fm$$

Substitute mf and fm values in the above formula.

$$BW=2(9+1)1K=20KHz$$

Therefore, the bandwidth of Wide Band FM wave is 20KHz

Bessel Functions

The FM equation

$$Vfm = Vc \sin[2\pi fct + mf \sin(2\pi fmt)],$$

where *Vfm* is the instantaneous value of the FM signal and mf is the modulation index

The term whose coeficient is mf is the phase angle of the carrier. Note that this equation expresses the phase angle in terms of the sine wave modulating signal. This equation is solved with a complex mathematical process known as Bessel functions.

Modulation Index	Carrier	Sidebands (Pairs)															
		1st	2d	3d	4th	5th	6th	7th	8th	9th	10th	11th	12th	13th	14th	15th	16th
0.00	1.00	—	—	—	—	—	—	—	—	—	—	—	—	—	—	—	—
0.25	0.98	0.12	—	—	—	—	—	—	—	—	—	—	—	—	—	—	—
0.5	0.94	0.24	0.03	—	—	—	—	—	—	—	—	—	—	—	—	—	—
1.0	0.77	0.44	0.11	0.02	—	—	—	—	—	—	—	—	—	—	—	—	—
1.5	0.51	0.56	0.23	0.06	0.01	—	—	—	—	—	—	—	—	—	—	—	—
2.0	0.22	0.58	0.35	0.13	0.03	—	—	—	—	—	—	—	—	—	—	—	—
2.5	−0.05	0.50	0.45	0.22	0.07	0.02	—	—	—	—	—	—	—	—	—	—	—
3.0	−0.26	0.34	0.49	0.31	0.13	0.04	0.01	—	—	—	—	—	—	—	—	—	—
4.0	−0.40	−0.07	0.36	0.43	0.28	0.13	0.05	0.02	—	—	—	—	—	—	—	—	—
5.0	−0.18	−0.33	0.05	0.36	0.39	0.26	0.13	0.05	0.02	—	—	—	—	—	—	—	—
6.0	0.15	−0.28	−0.24	0.11	0.36	0.36	0.25	0.13	0.06	0.02	—	—	—	—	—	—	—
7.0	0.30	0.00	−0.30	−0.17	0.16	0.35	0.34	0.23	0.13	0.06	0.02	—	—	—	—	—	—
8.0	0.17	0.23	−0.11	−0.29	−0.10	0.19	0.34	0.32	0.22	0.13	0.06	0.03	—	—	—	—	—
9.0	−0.09	0.24	0.14	−0.18	−0.27	−0.06	0.20	0.33	0.30	0.21	0.12	0.06	0.03	0.01	—	—	—
10.0	−0.25	0.04	0.25	0.06	−0.22	−0.23	−0.01	0.22	0.31	0.29	0.20	0.12	0.06	0.03	0.01	—	—
12.0	−0.05	−0.22	−0.08	0.20	0.18	−0.07	−0.24	−0.17	0.05	0.23	0.30	0.27	0.20	0.12	0.07	0.03	0.01
15.0	−0.01	0.21	0.04	0.19	−0.12	0.13	0.21	0.03	−0.17	−0.22	−0.09	0.10	0.24	0.28	0.25	0.18	0.12

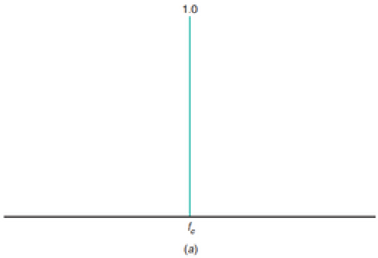
Figure 2.10 Carrier and sideband amplitudes for different modulation indexes of FM signals based on the Bessel functions.

Example

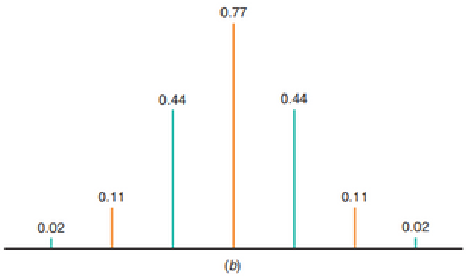
What is the maximum modulating frequency that can be used to achieve a modulation index of 2.2 with a deviation of 7.48 kHz?

$$f_m = \frac{f_d}{m_f} = \frac{7480}{2.2} = 3400 \text{ Hz} = 3.4 \text{ kHz}$$

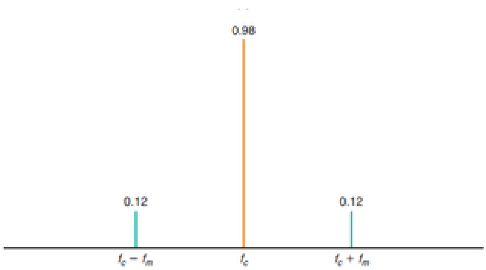
Examples of FM signal spectra.



(a) Modulation index of 0 (no modulation or sidebands)



(b) Modulation index of 1



(c) Modulation index of 0.25 (NBFM).

Figure 2.11 FM Signal Bandwidth

The higher the modulation index in FM, the greater the number of significant sidebands and the wider the bandwidth of the signal. When spectrum conservation is necessary, the bandwidth of an FM signal can be deliberately restricted by putting an upper limit on the modulation index.

The total bandwidth of an FM signal can be determined by knowing the modulation index.

For example, assume that the highest modulating frequency of a signal is 3 kHz and the maximum deviation is 6 kHz.

modulation index of mf ,

mf = 6 kHz/3 kHz

=2

The bandwidth can then be determined

BW = 2fmN

According to this formula, the bandwidth of our FM signal is

BW = 2(3 kHz)(4) = 24 kHz

Another way to determine the bandwidth of an FM signal is to use Carson's rule. This rule recognizes only the power in the most signiicant sidebands with amplitudes greater than 2 percent of the carrier.This rule is

BW = 2 [fd(max) + fm (max)]

According to Carson's rule, the bandwidth of the FM signal in the previous example would be

BW= 2[(6 kHz + 3 kHz)

= 2(9 kHz)

= 18 kHz

FM can be divided into Narrowband FM and Wideband FM based on the values of modulation index β
.

Narrowband FM

Following are the features of Narrowband FM.

- This frequency modulation has a small bandwidth when compared to wideband FM.
- The modulation index β is small, i.e., less than 1.
- Its spectrum consists of the carrier, the upper sideband and the lower sideband.
- This is used in mobile communications such as police wireless, ambulances, taxicabs, etc

Wideband FM

Following are the features of Wideband FM.

- This frequency modulation has infinite bandwidth.
- The modulation index β is large, i.e., higher than 1.
- Its spectrum consists of a carrier and infinite number of sidebands, which are located around it.
-
- This is used in entertainment, broadcasting applications such as FM radio, TV, etc

EXPERIMENT: FREQUENCY MODULATION (FM)

OBJECTIVES :

1. To understand and analyze the characteristics of a carrier signal.
2. To understand and analyze how frequency modulated signal (FM signal) is produced.
3. To understand the effect of varying amplitude of modulating signal on FM signal.

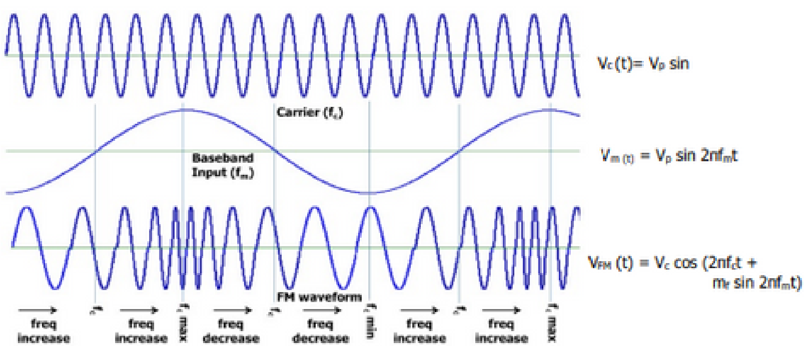
EQUIPMENTS : Unit

Computer 1

Mouse 1

THEORY :

Frequency Modulation (FM) Frequency modulation is used for sound broadcasting in the VHF band, for the sound signal of 625-line television broadcasting, for some mobile systems, and for multi-channel telephony systems operating in the UHF band. Frequency modulation (FM) refers to the process of varying the frequency of carrier signal with the change of amplitude modulating signal. The frequency of the carrier signal will become higher or lower according to the variation of input signal. If the modulating signal amplitude increases, the carrier frequency also increases or become higher and vice versa. During the modulation process, the amplitude of carrier signal is constant.



The equation for Modulating signal waveform,

$V_m(t) = V_m \sin 2\pi f_m t$ ……………………………..(1.1)

The equation for Carrier signal waveform,

$V_c(t) = V_c \sin 2\pi f_c t$ ………………………………..(1.2)

The equation for Frequency Modulated Signal,

v (t) Vc Cos[2πfct mf Sin2 fmt] FM = + π …….….….(1.3)

where :

Vp = peak voltage

mf = Modulation index FM

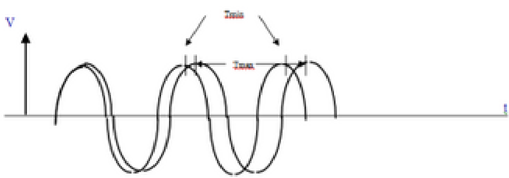
$$m_f = = \frac{\Delta f}{f_m} = \frac{K_f V_m}{f_m} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots \quad (1.4)$$

fm – modulating signal frequency

Δf – frequency deviation ; $\Delta f = \frac{|f_{max} - f_{min}|}{2}$ @ Kf Vm

Kf – Sensitivities

Vm – Modulating Signal Amplitude.



The amount of change in carrier frequency produced by the modulating signal is known as the frequency deviation Δf. Maximum frequency deviation occurs at the maximum amplitude of the modulating signal. Frequency deviation is typically given as a peak frequency shifts in hertz. The peak-to-peak frequency deviation (2Δfd) is sometimes called carrier swing.

PROCEDURE :

1. Click the icon AnalogMod.exe to run the program.

Figure 1

2. Extract the Vc and fc from Vc(t) = 7 sin 2π(3000)t and enter the following parameters value including
parameter values in Figure 2 into each parameter boxes as shown in Figure 3.

K_f	V_m	f_m
100	9	300

Figure 2

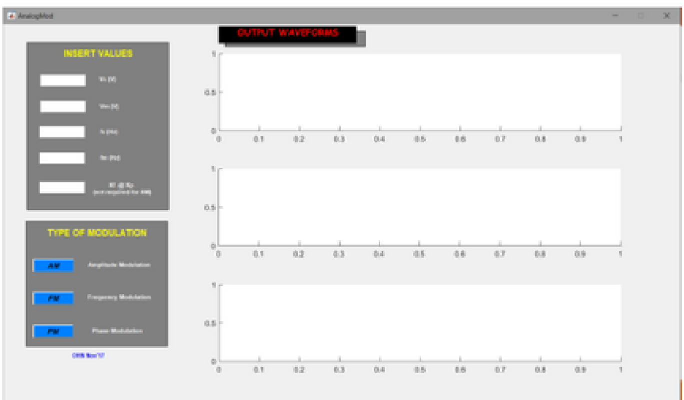


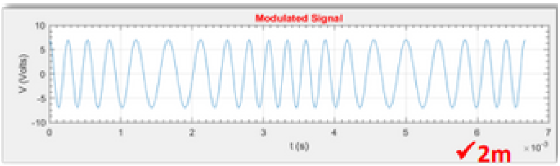
Figure 3

3. Choose FM as Type of Modulation.
4. Sketch the frequency modulated waveform in Table 1 and calculate the frequency deviation, Δfd and modulation index, mf .
5. Repeat step 2 to 4 except Vm = 15V in Table 2 and Vm = 21V in Table 3.

RESULT :

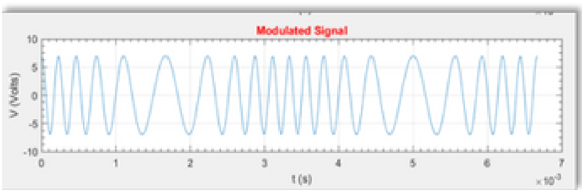
FM waveform :

Table 1



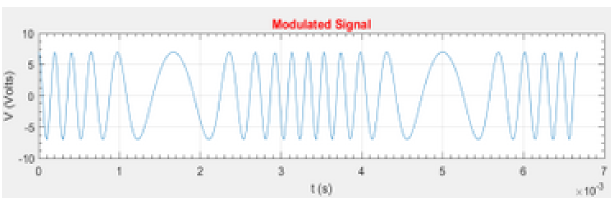
Calculation :
Δfd =Kf Vm
= (100) (9)
= 900
mf = Δfd /fm
= 900 / 300
= 3

Table 2



Calculation :
Δfd =Kf Vm
= (100) (15)
= 1500
mf = Δfd /fm
= 1500 / 300
= 5

Table 3



Calculation :
Δfd =Kf Vm
= (100) (21)
= 2100
mf = Δfd /fm
= 2100 / 300
= 7

QUESTIONS & DISCUSSION:

1. Name the parameter for the carrier signal that varies with the changes of the amplitude of the modulating signal
2. What are the differences between AM and FM?
3. When does maximum frequency deviation occur?

CONCLUSION:

1. Describe FM.
2. Relationship between mf and Δfd. if Δfd changes, will it affect to mf? if yes, describe it.
3. Define frequency deviation and its unit.

Phase Modulation

Phase Modulation (PM) is the process of changing the phase of analog carrier signal in proportion with the amplitude of the information signal.In PM, the carrier amplitude and frequency remains constant while thw carrier phase is varied by the mdulating signal. Frequency Modulation (FM) and Phase Modulation (PM) are both forms of Angle Modulation. This is because whenever the frequency of a carrier is varied, the phase is also varied and vice versa. Therefore, FM and PM must both occur whenever either form of angle modulation is performed.

The difference between FM and PM lies in which property of the carrier (the frequency or phase) is directly varied by modulating signal and which property is indirectly varied. If frequency is varied directly in accordance with modulating signal – FM. If phase is varied directly in accordance with modulating signal – PM

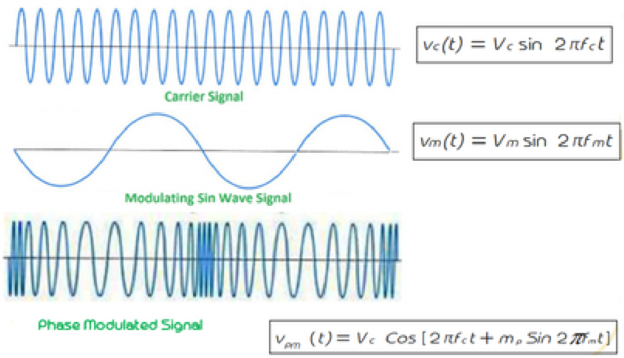


Figure 2.12 Phase Modulation

Principles of Phase Modulation

When the amount of phase shift of a constant-frequency carrier is varied in accordance with a modulating signal, the resulting output is a phase modulation (PM) signal [see Fig. 2.12(d)]. The greater the amplitude of the modulating signal, the greater the paste shift.

If a constant-amplitude, constant-frequency carrier sine wave is applied to the phase shifter whose phase shift is varied by the intelligence signal, the output of the phase shifter is a PM wave. As the modulating signal goes positive, the amount of phase lag, and thus the delay of the carrier output, increases with the amplitude of the modulating signal. The result at the output is the same as if the constant-frequency carrier signal had been stretched out, or had its frequency lowered. When the modulating signal goes negative, the phase shift becomes leading. This causes the carrier sine wave to be effectively speeded up, or compressed. The result is the same as if the carrier frequency had been increased.

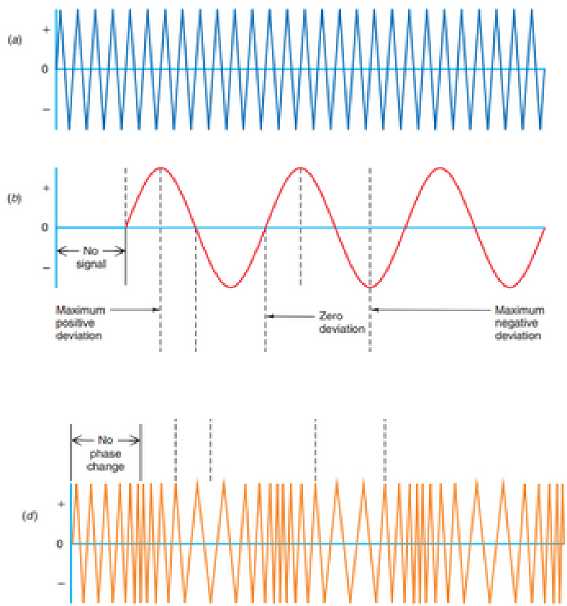


Figure 2.13 PM signal. The carrier is drawn as a triangular wave for simplicity, but in practice it is a sine wave. (a) Carrier. (b) Modulating signal. (c) PM signal.

Refer to the PM signal in Fig. 2.14(c). During increases or decreases in amplitude (t1, t3, and t5), a varying frequency is produced. However, during the constant-amplitude positive and negative peaks, no frequency change takes place. The output of the phase modulator is simply the carrier frequency that has been shifted in phase. This clearly illustrates that when a modulating signal is applied to a phase modulator, the output frequency changes only during the time that the amplitude of the modulating signal is varying. The maximum frequency deviation produced by a phase modulator occurs during the time when the modulating signal is changing at its most rapid rate. For a sine wave modulating signal, the rate of change of the modulating signal is greatest when the modulating wave changes from plus to minus or from minus to plus. As Fig. 2.14(c) shows,

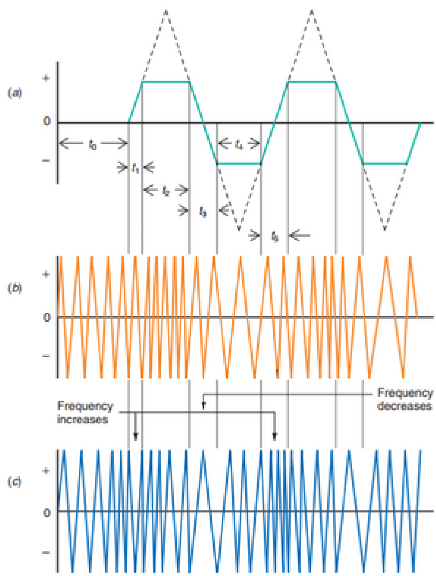


Figure 2.14 A frequency shift occurs in PM only when the modulating signal amplitude varies. (a) Modulating signal. (b) FM signal. (c) PM signal

In PM, the amount of carrier deviation is proportional to the rate of change of the modulating signal, i.e., the calculus derivative. With a sine wave modulating signal, the PM carrier appears to be frequency-modulated by the cosine of the modulating signal. Remember that the cosine occurs 90° earlier (leads) than the sine. Since the frequency deviation in PM is proportional to the rate of change in the modulating signal, the frequency deviation is proportional to the modulating signal frequency as well as its amplitude. This effect is compensated for prior to modulation.

Relationship Between FM and PM Modulation

In FM, the frequency deviation is directly proportional to the amplitude of the modulating signal. The maximum deviation occurs at the peak positive and negative amplitudes of the modulating signal. In PM, the frequency deviation is also directly proportional to the amplitude of the modulating signal. The maximum amount of leading or lagging phase shift occurs at the peak amplitudes of the modulating signal. This effect, for both FM and PM, is illustrated in Fig. 2.15(a)

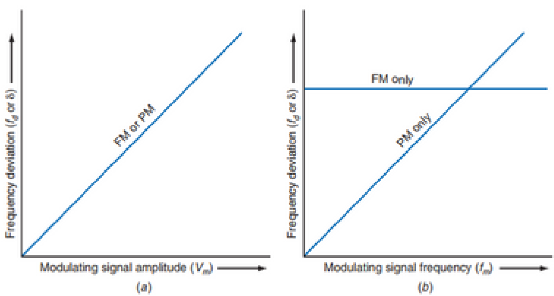


Figure 2.15 Frequency deviation as a function of (a) modulating signal amplitude and (b) modulating signal frequency

Fig. 2.11(b), which shows that the frequency deviation of an FM signal is constant for any value of modulating frequency. Only the amplitude of the modulating signal determines the amount of deviation.

In PM, then, the carrier frequency deviation is proportional to both the modulating frequency (slope of modulating voltage) and the amplitude. In FM, frequency deviation is proportional only to the amplitude of the modulating signal, regardless of its frequency.

Frequency Modulation Versus Amplitude Modulation

Noise Immunity.

The main benefit of FM over AM is its superior immunity to noise, made possible by the clipper limiter circuits in the receiver, which effectively strip off all the noise variations, leaving a constant-amplitude FM signal. Although clipping does not result in total recovery in all cases, FM can nevertheless tolerate a much higher noise level than AM for a given carrier amplitude. This is also true for phase-shift-induced distortion.

Capture Effect.

Another major benefit of FM is that interfering signals on the same frequency are effectively rejected. Because of the amplitude limiters and the demodulating methods used by FM receivers, a phenomenon known as the capture effect takes place when two or more FM signals occur simultaneously on the same frequency.

Transmitter Efficiency.

A third advantage of FM over AM involves efficiency. Recall that AM can be produced by both low-level and high-level modulation techniques. The most efficient is high-level modulation in which a class C amplifier is used as the final RF power stage and is modulated by a high-power modulation amplifier. The AM transmitter must produce both very high RF and modulating signal power.

FM and AM Applications

Here are some of the major applications for AM and FM.

Application	Type of Modulation
AM broadcast radio	AM
FM broadcast radio	FM
FM stereo multiplex sound	DSB (AM) and FM
TV sound	FM
TV picture (video)	AM, VSB
TV color signals	Quadrature DSB (AM)
Cellular telephone	FM, FSK, PSK
Cordless telephone	FM, PSK
Fax machine	FM, QAM (AM plus PSK)
Aircraft radio	AM
Marine radio	FM and SSB (AM)
Mobile and handheld radio	FM
Citizens band radio	AM and SSB (AM)
Amateur radio	FM and SSB (AM)
Computer modems	FSK, PSK, QAM (AM plus PSK)
Garage door opener	OOK
TV remote control	OOK
VCR	FM
Family Radio service	FM
Bluetooth radio	FSK

EXPERIMENT : PHASE MODULATION (PM)

OBJECTIVES :

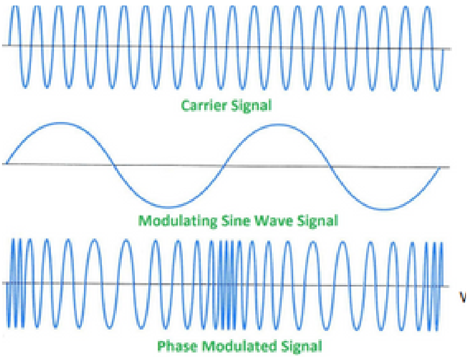
1. To understand and analyze the characteristics of a carrier signal.
2. To understand and analyze how phase modulated signal (PM signal) is produced.
3. To understand the effect of varying amplitude of modulating signal on PM signal.

EQUIPMENTS :	Unit
Computer	1
Mouse	1

THEORY :

Phase Modulation (PM)

In PM, the carrier amplitude and frequency remain constant while the carrier phase is varied by the modulating signal. As the modulating signal amplitude increases, the carrier phase increases and vice versa. Modulation Index for PM, mp = KpVm.



Vm(t) = Vp sin 2πfmt

Vc (t)= Vp sin 2πfct

VPM (t) =
Vc cos (2πfct + mp sin 2πfmt)

The equation for Modulating signal waveform,

Vm (t) = Vm sin 2πfmt ……………………….(1.1)

The equation for Carrier signal waveform,

Vc (t) = Vc sin 2πfct ……………………………….(1.2)

The equation for Phase Modulated Signal,

vPM (t) = Vc Cos[2πfct + mpSin2πfmt] ………(1.3)

where :

Vp – peak voltage

mp – Modulation index PM = Kp Vm

……………………………………………………………..………(1.4)

Vc – Carrier Signal Amplitude

Kp – Sensitivities

Vm – Modulating Signal Amplitude

PROCEDURE :

1. Click the icon AnalogMod.exe to run the program.

Figure 1

2. Extract the Vc and fc from Vc(t) = 7 sin 2π(3000)t and enter the following parameters value including parameter values in Figure 2 into e   each parameter boxes as shown in Figure 3.

K_p	V_m	f_m
1	3	300

Figure 2

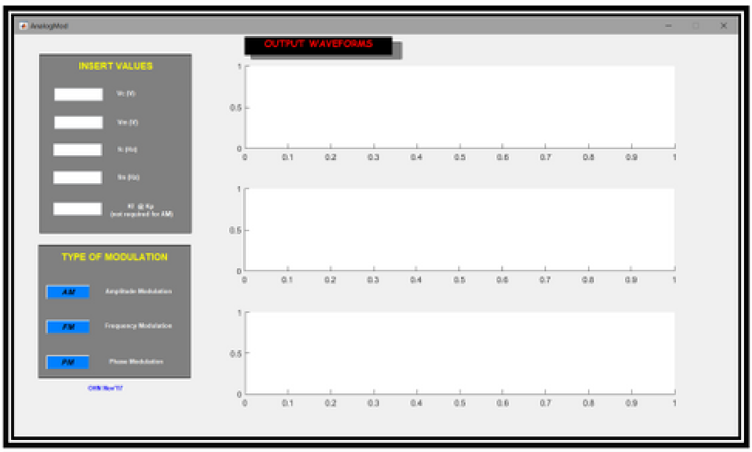


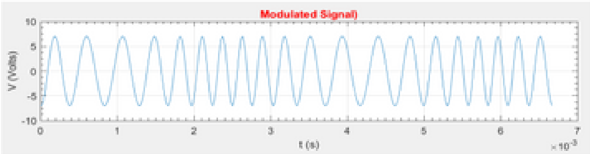
Figure 3

3. Choose PM as Type of Modulation.
4. Sketch the phase modulated waveform in Table 1 and calculate t the modulation index, mp.
5. Repeat step 2 to 4 except Vm = 5V in Table 2 and Vm = 8V in Table 3.

RESULT :

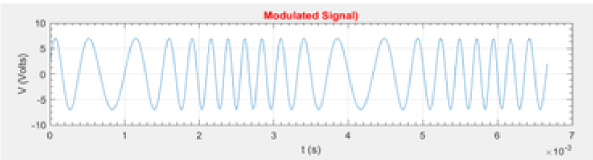
PM waveform :

Table 1



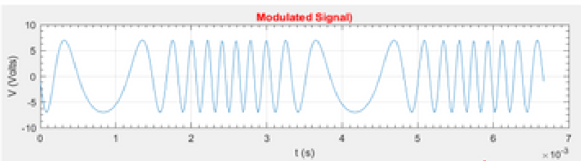
Calculation :
mp = Kp Vm
= 1 (3)
= 3

Table 2



Calculation :
mp = Kp Vm
= 1 (5)
= 5

Table 3



Calculation :
mp = Kp Vm
= 1 (8)
= 8

QUESTIONS & DISCUSSION:

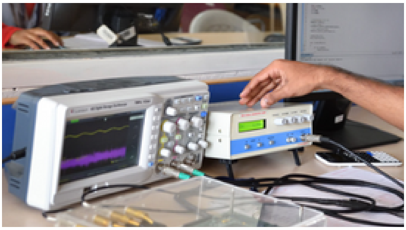
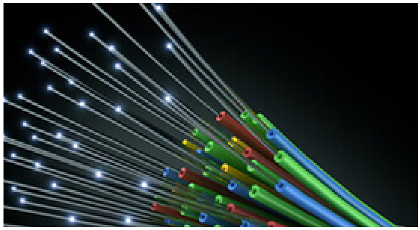
1. Is Phase Modulation (PM) one of the Analog Modulation or Digital Modulation? Name the TWO (2) others that are same type of modulation with PM.

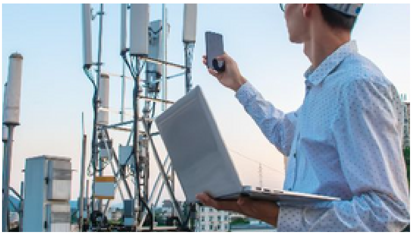
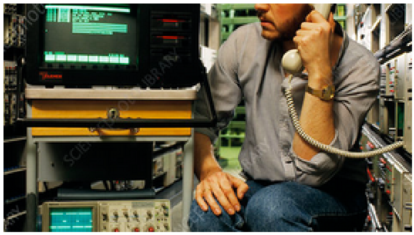
2. What can you conclude to the phase of modulated wave based to the result?

3. Based from the result in Table 1 to 3, what can you observed when the amplitude of modulating signal increase?

CONCLUSION:

1. Define PM.
2. Define phase modulation index.

03. DIGITAL MODULATION

Digital modulation is the process of converting digital data, which consists of a sequence of discrete symbols, into analog signals that can be transmitted over a communication channel. The process involves modifying a carrier wave, which is an analog signal with a specific frequency, amplitude, and phase, to encode the digital information.

Digital modulation is widely used in various communication systems such as satellite communication, wireless communication, and digital broadcasting. It plays a crucial role in enabling high-speed data transmission and reliable communication over long distances.

Digital transmission and digital radio modulation techniques are used to convert digital data into analog signals that can be transmitted over a communication channel. The term digital communication covers a broad range of communication techniques including digital transmission and digital radio.

Types of Digital Modulation

Both pulse and digital signals are widely used in modern digital communication systems. Pulse signals are used in applications that require high data rates, such as digital data transmission over copper and optical fiber cables. Digital signals are used in applications such as wireless communication, satellite communication, and digital broadcasting.

However, digital Signal cannot be transmitted through free space (wireless) medium but analog signal does. Therefore, digital data needs to be converted into analog signal by doing the Digital Modulation techniques. Digital Modulation is the process of changing one of the characteristics of an analog carrier signal based on the information in digital data.

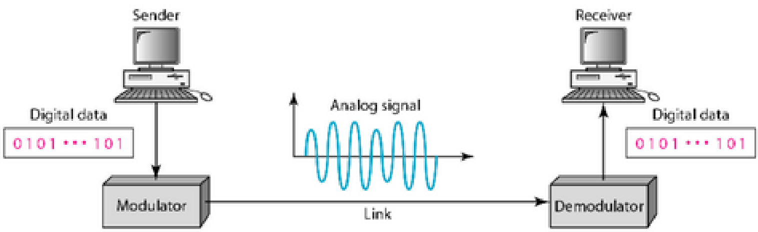
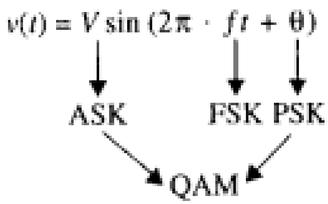


Figure 3.1 Digital-to-analog conversion

The digital signal is first converted into a series of pulses, which can then be used to modulate a carrier signal. The process of converting digital signals into analog signals involves digital-to-analog conversion (DAC). The digital signal is sampled at discrete time intervals, and the amplitude of each sample is quantized and encoded as a binary value. The resulting digital signal is then converted into an analog signal using a DAC, which generates a continuous analog waveform from the binary values of the digital signal.

The modulated analog signal is then transmitted over a communication channel using the chosen modulation technique. At the receiver end, the analog signal is demodulated to recover the original digital signal. The demodulated analog signal is then converted back into a digital signal using an analog-to-digital converter (ADC).

Equation 3.1 below shows a simplified block diagram for a digital modulation system.



...... (3.1)

Referring to Equation (3.1), if the information signal is digital and the amplitude (lV of the carrier is varied proportional to the information signal, a digitally modulated signal called amplitude shift keying (ASK) is produced.

If the frequency (f) is varied proportional to the information signal, frequency shift keying (FSK) is produced, and if the phase of the carrier (0) is varied proportional to the information signal, phase shift keying (PSK) is produced.

If both the amplitude and the phase are varied proportional to the information signal, quadrature amplitude modulation (QAM) results

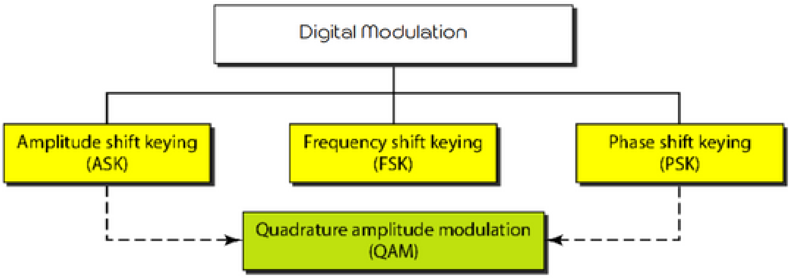


Figure 3.2 Types of Digital Modulation

In Digital Modulation, Information Signal is in digital waveform; while Carrier signal is in analog waveform.

- Amplitude Shift Keying (ASK) - the amplitude (Vp) of the analog carrier signal is varied proportional to the digital information signal.
- Frequency Shift Keying (FSK) - the frequency (f) of the analog carrier signal is varied proportional to the digital information signal.
- Phase Shift Keying (PSK) - the phase (Ɵ) of the analog carrier is varied proportional to the digital information signal.

Digital Modulation Transmitter

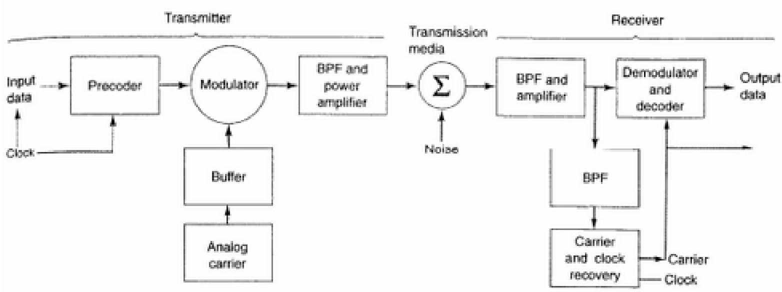


Figure 3.3 A simplified block diagram for a digital modulation system.

The block diagram of a typical digital modulation system consists of the following components:

In the transmitter, the precoder performs level conversion and then encodes the incoming data into groups of bits that modulate an analog carrier. Channel encoder: This block is responsible for adding redundancy to the digital signal to protect it from errors that may occur during transmission. Techniques such as forward error correction (FEC) and convolutional coding are commonly used for this purpose.

Modulator is responsible for converting the digital signal into a modulated waveform that can be transmitted over the communication channel. Common modulation techniques used in digital communication systems include amplitude shift keying (ASK), frequency shift keying (FSK), phase shift keying (PSK), and quadrature amplitude modulation (QAM).

The transmission medium represents the physical communication channel over which the modulated waveform is transmitted.The transmission medium can be a metallic cable, optical fiber cable, Earth's atmosphere, or a combination of two or more types of transmission systems. The channel introduces noise and other impairments that can cause errors in the received signal.

In the receiver, the incoming signals are filtered, amplified, and then applied to the demodulator and decoder circuits, which extracts the original source information from the modulated carrier. Demodulator: This block is responsible for converting the received modulated waveform back into a digital signal. The demodulator performs the inverse operation of the modulator, and must be designed to operate in the presence of noise and other impairments.

The clock and carrier recovery circuits recover the analog carrier and digital timing (clock) signals from the incoming modulated wave since they are necessary to perform the demodulation process. This block is responsible for converting the digital signal back into an analog signal that can be played back or displayed by the receiver.

Overall, the block diagram of a digital modulation system represents the process of converting an analog signal into a digital signal, modulating the digital signal, transmitting it over a communication channel, demodulating the received signal, decoding any errors, and converting the digital signal back into an analog signal.

Amplitude Shift Keying

The simplest digital modulation technique is amplitude-shift keying (ASK), where a binary information signal directly modulates the amplitude of an analog carrier. ASK is a simple digital modulation technique used in digital transmission to transmit binary data over a communication channel. In ASK, the amplitude of the carrier wave is varied to represent the digital data.

Mathematically, amplitude-shift keying is,

$$v_{(ask)}(t) = [1 + v_m(t)]\left[\frac{A}{2}\cos(\omega_c t)\right] \qquad (2.12)$$

where
vask(t) = amplitude-shift keying wave
vm(t) = digital information (modulating) signal (volts)
A/2 = unmodulated carrier amplitude (volts)
ωc = analog carrier radian frequency (radians per second, 2πfct)

Therefore, for a logic 1 input,

$$v_{(ask)}(t) = [1 + 1]\left[\frac{A}{2}\cos(\omega_c t)\right]$$

$$= A\cos(\omega_c t)$$

for a logic 0 input.

$$v_{(ask)}(t) = [1 - 1]\left[\frac{A}{2}\cos(\omega_c t)\right]$$

Thus, the modulated wave Vask(t), is either A cos(ωc t) or 0. Hence, the carrier is either "on"or "off," which is why amplitude-shift keying is sometimes referred to as on-off keying(OOK).

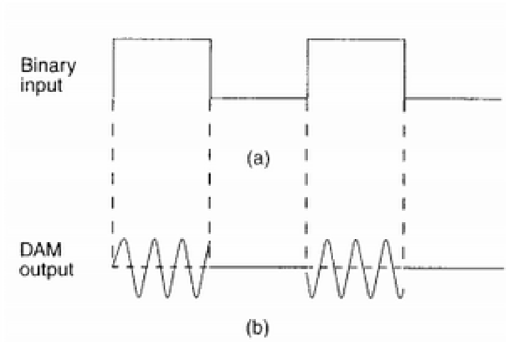


FIGURE 3-4 Digital Amplitude Modulation: (a) input binary; (b) output DAM waveform

The entire time the binary input is high, the output is a constant amplitude, constant-frequency signal, and for the entire time the binary input is low, the carrier is off.

E**XPERIMENT :** AMPLITUDE SHIFT KEYING (ASK) MODULATION

OBJECTIVES:

1. Generate Amplitude Shift Keying (ASK) Wave Using MultiSim Program.
2. Analyze the output waveform of ASK Modulation.

EQUIPMENTS:	Unit
MUltiSim Program	1
Laptop / Computer	1

THEORY:

Amplitude Shift Keying (ASK) is a special version of AM. It is used for transmitting digital data over an analog transmission medium. In this method two binary values are represented by two different amplitudes of the carrier signal as shown in Figure 5.1.

The carrier signal is a high frequency (RF) sinusoidal signal which will be amplitude modulated by the digital data stream. The presence of the carrier signal at the output represents logic 1 and its absence represents logic 0. Typical applications for ASK are limited to 1200 bps on voice grade lines and over optical fiber networks. The bandwidth of an ASK signal is the same as the bandwidth of a conventional AM signal.

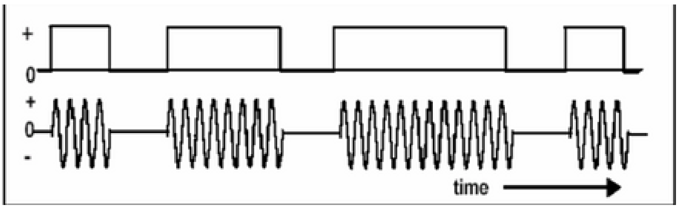


Figure 5.1: An ASK Signal (below) and the message (above)

PROCEDURE:

1. Start MultiSim Program at laptop or PC and draw the circuit below:

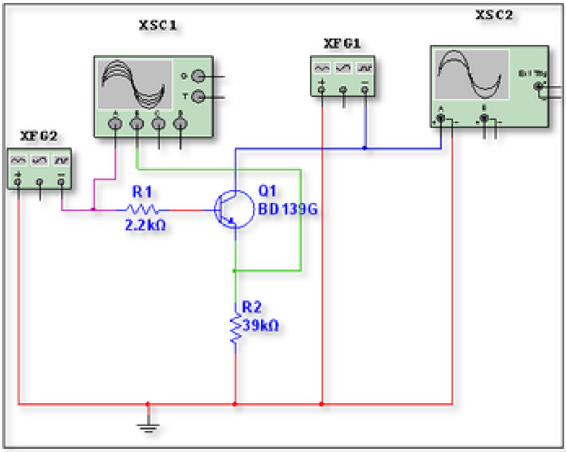


Figure 5.2

*** Observe Figure 5.1

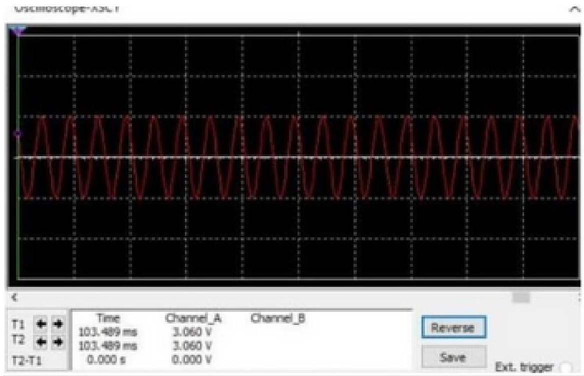
2. Complete your observation (Output waveform of ASK Modulation) at Graph 1.
3. Set f1 (Carrier Signal) to 10kHz and amplitude 5Vpp for XFG1.
4. For XFG2, set f2 (Modulating Signal) to 1kHz and amplitude 5Vpp.
5. Change the value of the f2 (Modulating Signal) to 500Hz and complete the result output of an ASK signal at Graph 2.

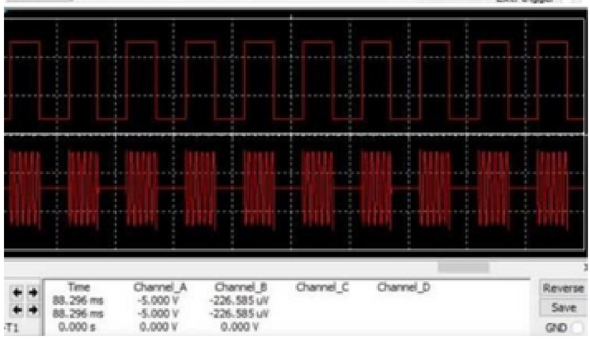
RESULT :

Carrier Signal :

Square Pulses:

ASK Signal:





QUESTIONS & DISCUSSION:

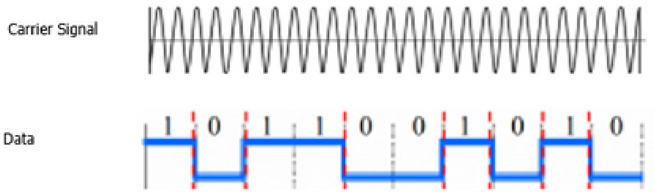
1. Digital Modulation can be grouped as:

i. ________________________________

ii. ________________________________

iii. ________________________________

2. Sketch each types output waveform of the Digital Modulation based on Carrier Signal and Information Signal (Data) below.



i. ASK

ii. FSK

iii. PSK

3. The carrier signal is a ____________________ sinusoidal signal which will be amplitude modulated by the digital data stream

4. Typical applications for ASK are limited to __________________ on voice grade lines and over optical fiber networks.

CONCLUSION:

1. What is parameter that varied and parameters that remain unchanged in ASK modulation technique?

2. What happen to ASK signal when the data is set to bit 0 and when the data is set to 1?

Frequency Shift Keying

FSK is a form of constant-amplitude angle modulation similar to standard frequency modulation (FM) except the modulating signal is a binary signal that varies between two discrete voltage levels rather than a continuously changing analog waveform.FSK is a digital modulation technique used in digital transmission to transmit binary data over a communication channel. In FSK, the frequency of the carrier wave is varied to represent the digital data.

Consequently, FSK is sometimes called binary FSK (BFSK).

The general expression for FSK is

$$v_{fsk}(t) = V_c \cos\{2\pi[f_c + v_m(t)\,\Delta f]t\}$$

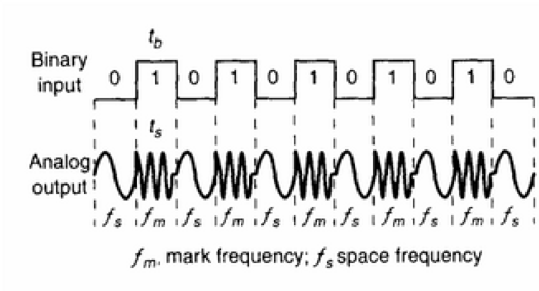
where
vfsk(t) = binary FSK waveform V
Vc = peak analog carrier amplitude (volts)
fc = analog carrier center frequency (hertz)
Δf = peak change (shift) in the analog carrier
f requency

Thus, for a logic I input

$$v_{fsk}(t) = V_c \cos[2\pi(f_c + \Delta f)t]$$

For a logic 0 input

$$v_{fsk}(t) = V_c \cos[2\pi(f_c - \Delta f)t]$$



binary input	frequency output
0	space (f_s)
1	mark (f_m)

(b)

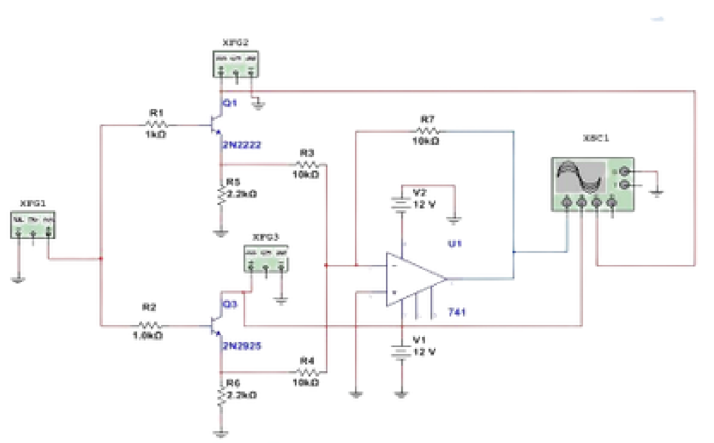
FIGURE 3.5 FSK in the time domain: (a) waveform: (b) truth table

EXPERIMENT : FSK Modulation and Demodulation

AIM : To design and verify the characteristics of frequency shift keying modulation and demodulation

APPARATUS REQUIRED: PC loaded with Multisim Software.

FSK Modulation circuit.



SIMULATION PROCEDURE:

1. Open MULTISIM Software
2. Click => New => Design1
3. Click save as in Desk Top rename the Design1 to your circuit name.
4.. Go to Component tool bar and select the components.
5. Draw the schematic of frequency shift keying modulator and demodulator as shown.
6. XFG1 F=150 H, XFG2 F=200 Hz, and XFG F = 2KHz
7. Specify the value of amplitude and frequency for the same value below mentioned.
8. Click simulate button or press F5 key ==> RUN
9. Design the comparator circuit using op-amp to restore the modulating signal.
10. Record the waveforms.

RESULT:

Thus the design of Frequency shift keying modulator and demodulator was completed and its output is verified by the frequency deviation and modulation index using simulation software.

Phase Shift Keying

Phase-Shift Keying (PSK) is a digital modulation technique used to modulate a carrier signal by varying the phase of the signal to represent digital data. In PSK, the phase of the carrier signal is changed to represent different symbols, such as 0 or 1.

In PSK, the carrier signal is a continuous sine wave, and the phase of the sine wave is changed according to the symbol to be transmitted. For example, if a "0" symbol is to be transmitted, the phase of the carrier signal may be set to 0 degrees, while if a "1" symbol is to be transmitted, the phase of the carrier signal may be set to 180 degrees.

PSK can be further classified into several types, including Binary Phase-Shift Keying (BPSK), Quadrature Phase-Shift Keying (QPSK), and Differential Phase-Shift Keying (DPSK).

Binary Phase-Shift Keying (BPSK)

The simplest form of PSK is binary phase-shift keying (BPSK),

where N = 1
and M = 2.

Therefore, with BPSK, two phases (21 = 2) are possible for the carrier.

One phase represents a logic 1, and the other phase represents a logic 0. As the input digital signal changes state (i.e., from a 1 to a 0 or from a 0 to a 1), the phase of the output carrier shifts between two angles that are separated by 180°.

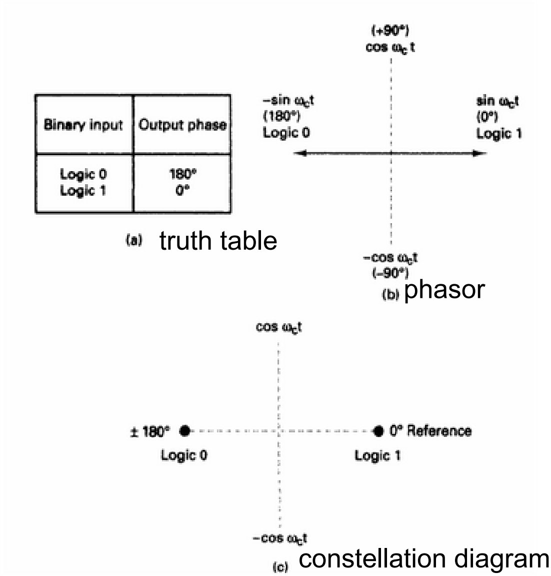


FIGURE 3.6 BPSK modulator

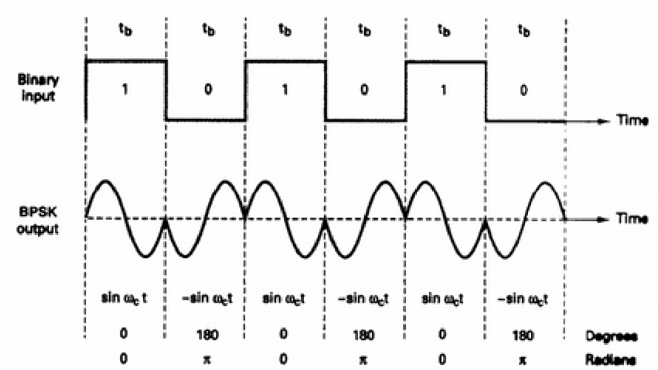


FIGURE 3.7 Output phase-vs-time relationship for a BPSK modulator

Quaternary Phase-Shift Keying

QPSK is an M-ary encoding scheme where N = 2 and M= 4 (hence, the name "quaternary" meaning "4"). A QPSK modulator is a binary (base 2) signal, to produce four different input combinations,: 00, 01, 10, and 11.

Therefore, with QPSK, the binary input data are combined into groups of two bits, called dibits. In the modulator, each dibit code generates one of the four possible output phases (+45°, +135°, -45°, and -135°).

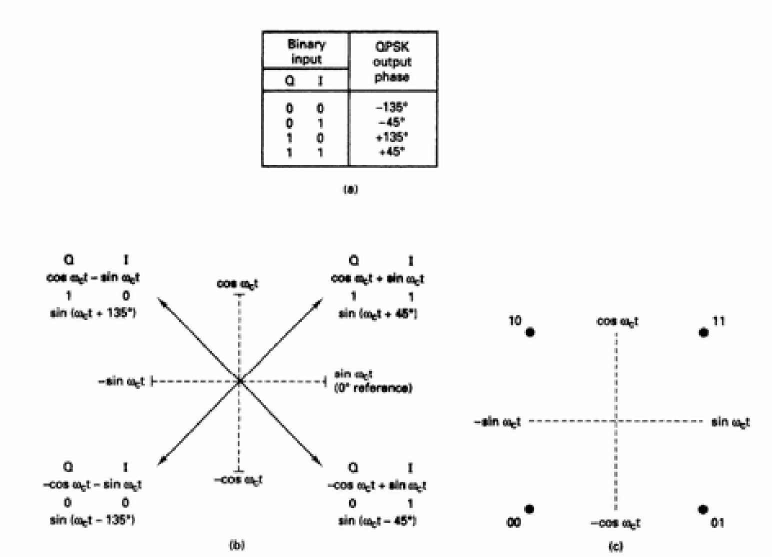


FIGURE 3.8 QPSK modulator: (a) truth table; (b) phasor diagram; (c) constellation diagram

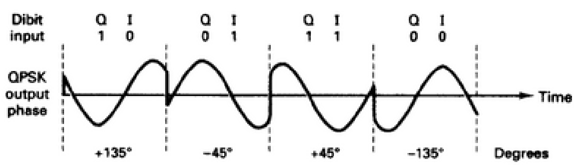


FIGURE 3.9 Output phase-vs-time relationship for a PSK modulator

8-PSK

With 8-PSK, three bits are encoded, forming tribits and producing eight different output phases. To encode eight different phases, the incoming bits are encoded in groups of three, called tribits (23 = 8).

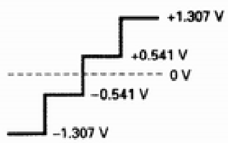
In QPSK, the carrier signal is a continuous sine wave, and the phase of the sine wave is changed according to the two bits of data to be transmitted. Specifically, each pair of bits is mapped to one of four possible phase shifts: 0, 90, 180, or 270 degrees. These four phase shifts are equally spaced around a circle and are known as the 0, 1, 2, and 3 states.

I	C	Output
0	0	−0.541 V
0	1	−1.307 V
1	0	+0.541 V
1	1	+1.307 V

(a)

Q	$\bar{C}$	Output
0	1	−1.307 V
0	0	−0.541 V
1	1	+1.307 V
1	0	+0.541 V

(b)



(c)

FIGURE 3.10 I- and Q-channel 2-to-4-level converters: (a) 1-channel truth table; (b) D-channel truth table; (c) PAM levels

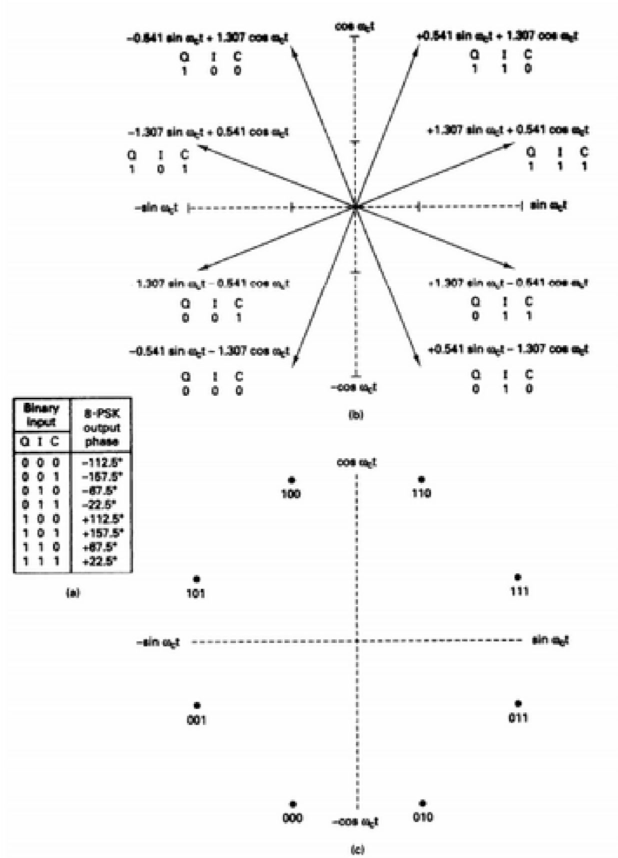


FIGURE 3.11 8-PSK modulator: (a) truth table; (b) phasor diagram; (c) constellation diagram

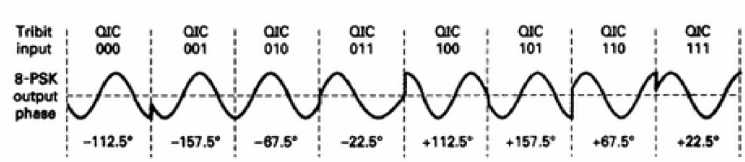


FIGURE 3.12 Output phase-vs-time relationship for a 8-PSK modulator

PSK is commonly used in digital communication systems, such as wireless and satellite communications, due to its ability to transmit data over long distances with low error rates. It is also widely used in applications such as RFID (Radio Frequency Identification) and magnetic storage systems.

In summary, Phase-Shift Keying is a digital modulation technique that varies the phase of a carrier signal to represent digital data. PSK has several variants, including BPSK, QPSK, and DPSK, and is widely used in communication and storage systems.

Quadrature Amplitude Modulation

Quadrature Amplitude Modulation (QAM) is a digital modulation technique used to transmit digital data over a radio frequency carrier signal.Quadrature Amplitude Modulation (QAM): QAM is a digital modulation technique used in digital radio to transmit high-speed digital data over a communication channel. In QAM, both the amplitude and phase of the carrier wave are varied to represent the digital data. QAM combines both amplitude and phase modulation techniques to transmit multiple bits of data simultaneously.

8-QAM is an M-ary encoding technique where M = 8. Unlike 8-PSK, the output signal from an 8-QAM modulator is not a constant-amplitude signal.

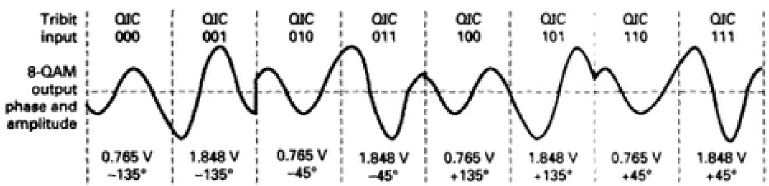


FIGURE 3.13 Output phase and amplitude -vs-time relationship for a 8-QAM

Binary input			8-QAM output	
Q	I	C	Amplitude	Phase
0	0	0	0.765 V	−135°
0	0	1	1.848 V	−135°
0	1	0	0.765 V	−45°
0	1	1	1.848 V	−45°
1	0	0	0.765 V	+135°
1	0	1	1.848 V	+135°
1	1	0	0.765 V	+45°
1	1	1	1.848 V	+45°

(a)

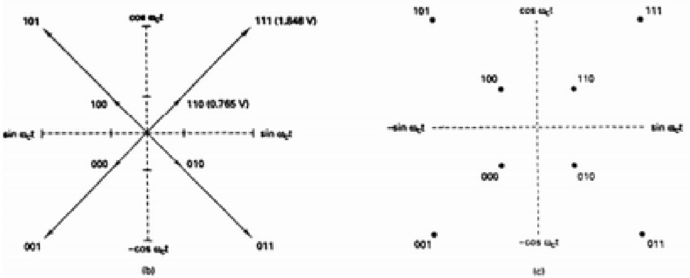


FIGURE 3.14 8-QAM modulator: (a) truth table (b) phasor diagram; (c) constellation diagram

QAM is used in a variety of applications, including digital cable TV, satellite communications, and wireless communication systems such as Wi-Fi and cellular networks. QAM provides a high data rate and spectral efficiency, making it an attractive modulation technique for high-speed digital communication systems.

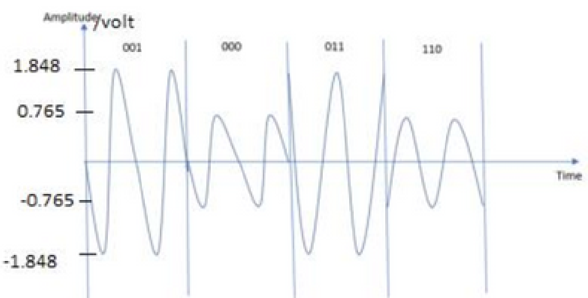
In summary, Quadrature Amplitude Modulation is a digital modulation technique that uses both amplitude and phase modulation to transmit multiple bits of data simultaneously. QAM can be rectangular or circular and is used in various communication systems for high-speed data transmission.

Exercise

By referring 8 QAM output truth table constellation diagram in Figure 3(a), identify the output of 8-QAM (3 bits) waveform for binary data 001000011110.

Binary input			8-QAM output	
Q	I	C	Amplitude	Phase
0	0	0	0.765 V	−135°
0	0	1	1.848 V	−135°
0	1	0	0.765 V	−45°
0	1	1	1.848 V	−45°
1	0	0	0.765 V	+135°
1	0	1	1.848 V	+135°
1	1	0	0.765 V	+45°
1	1	1	1.848 V	+45°

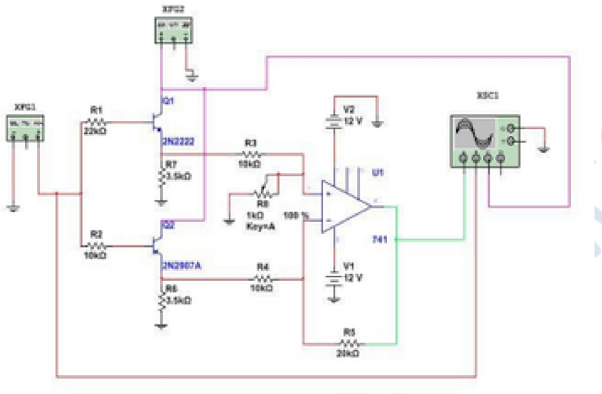
Input data = 0 0 1 | 0 0 0 | 0 1 1 | 1 1 0.



EXPERIMENT : PSK Modulation and Demodulation

AIM : To design and verify the characteristics of Phase shift keying Modulation and Demodulation.

APPARATUS REQUIRED: PC loaded with MULTISM software.



SIMULATION PROCEDURE:

1. Open MULTISIM Software
2. Click => New => Design1
3. Click save as in Desk Top rename the Design1 to your circuit name.
4.. Go to Component tool bar and select the components.
5. Draw the schematic of frequency shift keying modulator and demodulator as shown.
6. Specify the values of amplitude and frequency for the same values of the input signals.
7. Set frequency of XFG1 = 600M/12v, XFG2 = 2 KHz/10V.
8. Design the comparator circuit using op-amp to restore the modulating signal.
10. Record the output waveforms.

RESULT:

Thus the design of Phase shift keying modulator and demodulator was completed and its output is verified.

04. PULSE MODULATION

Pulse modulation is a technique used in electronics and communication systems to transmit analog signals.The basic idea behind pulse modulation is to convert the analog signal into a train of pulses, where the amplitude or duration of each pulse is proportional to the value of the original signal at that time.

Pulse modulation techniques are widely used in digital communication systems, including telecommunications, digital audio, and video recording and playback, and control systems. These techniques provide a reliable and efficient way to transmit analog signals over digital communication channels with minimal distortion and noise.

The term Digital Communication covers a broad range of communication techniques including digital transmission and digital radio.

- Digital transmission

Digital transmission is a true digital system where digital signals are transferred between two or more points in a communication system. With digital transmission, there is NO analog carrier and the original source information may be in digital or analog form.

- Digital Radio

Digital radio is a transmittal of digitally-modulated analog carrier signals between two or more points in a communication system. With digital radio, the information signal and demodulated signal are in digital form. While the carrier signal and modulated signal are in analog form. The digital pulses could be originated from computer-generated data or digital transmission system or digitally-encoded analog information signal

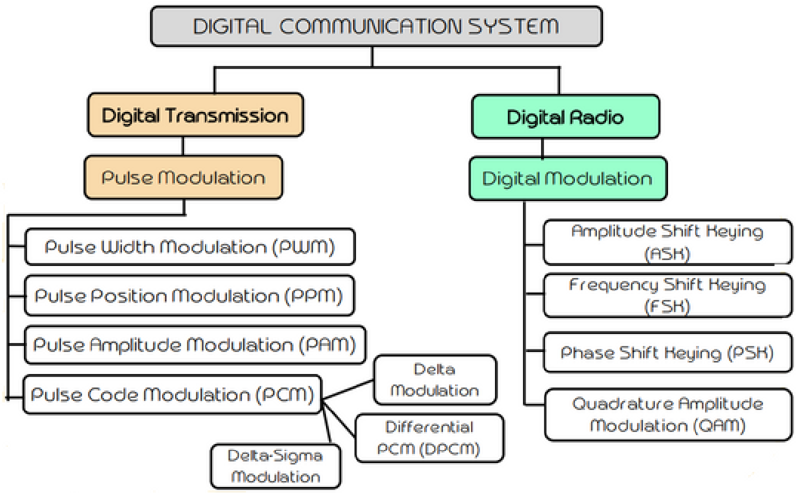


Figure 4.1 Types of Digital Modulation

In Pulse Modulation, information signal is in analog waveform. While Sampling signal is in digital waveform. (There is NO carrier signal in pulse modulation) .This modulation is necessary to convert the analog signal to digital signal for digital transmission. Usually used metallic cable and fiber optic cable. Cannot use free space as channel.

PULSE MODULATION TECHNIQUES

Definition:

Pulse Modulation (PM) is a process of sampling the analog information signals into sampled signal before converting those into digital signals. In Pulse Modulation the Information signal is in analog waveform. While Sampling signal is in digital pulses waveform. The properties of sampling pulses signal such as width, position and amplitude will be varied in proportion with amplitude of information signal. There are several types of pulse modulation techniques, including pulse amplitude modulation (PAM), pulse width modulation (PWM), and pulse position modulation (PPM).

Pulse modulation techniques are widely used in digital communication systems, such as pulse code modulation (PCM) used in telephone networks, and in digital audio and video systems. They are also used in various control systems, such as motor control and power electronics.

There are 4 predominant techniques for Pulse Modulation :

a) Pulse Width/Duration Modulation
   (PWM @ PDM)
b) Pulse Position Modulation (PPM)
c) Pulse Amplitude Modulation (PAM)
d) Pulse Code Modulation (PCM)==Digital PM

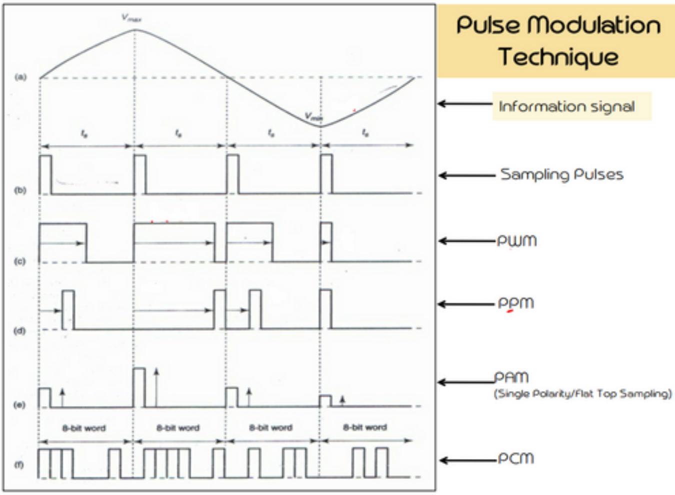


Figure 4.1 Pulse Modulation Techniques

PWM - the width of the pulses is varied proportional to the analog amplitude information signal..(The higher amplitude of Information signal, the wider of pulse.)

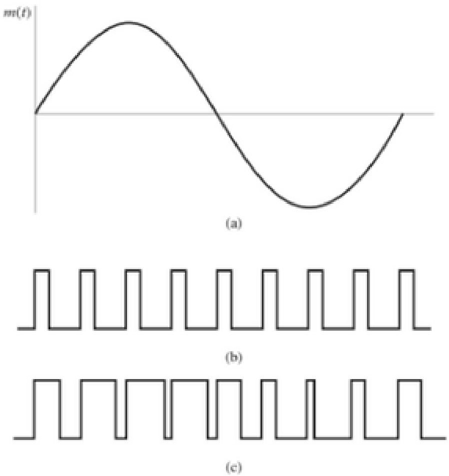
PPM – the position of the pulses is varied proportional to the analog amplitude information signal. (The higher amplitude of Information signal, the farther to the right the pulse is positioned).

PAM - the amplitude of the pulses is varied proportional to the analog amplitude information signal. (The higher amplitude of Information signal, the higher amplitude of pulse).

PCM – With PCM, the analog information signal is sampled into PAM signal and then converted to a serial n-bit binary code for transmission.

Pulse Width/Duration Modulation

In pulse-duration modulation (PDM), the samples of the message signal are used to vary the duration of the individual pulses. This form of modulation is also referred to as pulse-width modulation or pulse-length modulation. The modulating signal may vary the time of occurrence of the leading edge, the trailing edge, or both edges of the pulse.



- Width (W) of pulses is **varied** proportional to the amplitude of analog Information signal.
- Amplitude (A) and Position (P) of **pulses** are **constant**.
- The higher **amplitude** of Information signal, the **wider** of Pulses.

FIGURE 4.2 Illustration of two different forms of pulse time modulation for the case of a sinusoidal modulating wave. (a) Modulating wave. (b) Pulse carrier. (c) PDM wave.

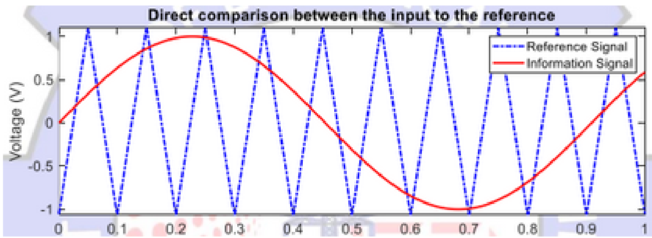
PDM is wasteful of power, in that long pulses expend considerable power during the pulse while bearing no additional information. If this unused power is subtracted from PDM, so that only time transitions are essentially preserved, we obtain a more efficient type of pulse modulation known as pulse-position modulation (PPM).

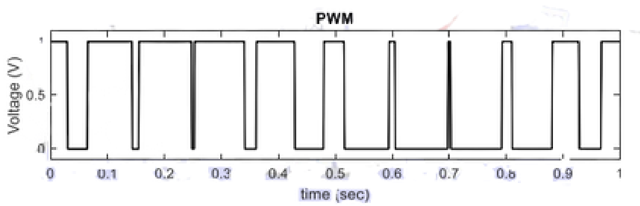
Experiment : PWM MODULATION USING MULTISIM

Objective: Experiment the PWM in Multisim

Theory:

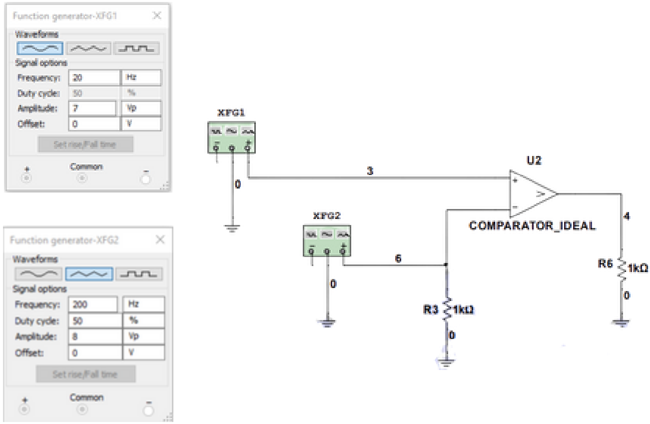
The PWM is achieved by comparing the input signal to a reference
signal (in this case is triangular waveform). The output is high whenever the information signal is greater than or equal to the reference signal. In the same way, the output becomes low when the
reference signal exceeds the input. The plots below illustrate the concept.





Procedure

1. Implement the following circuit in Multisim.
2. Examine and study the outputs.
3. Change the given values and compare

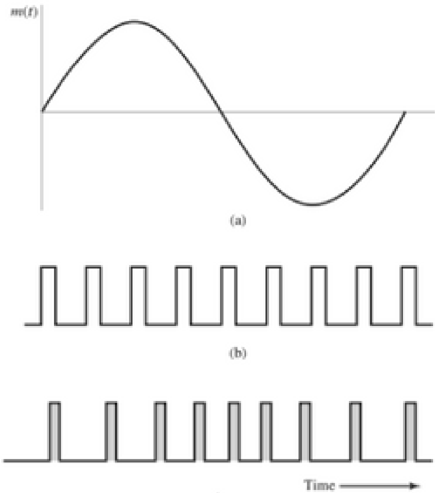


HOMEWORK

Although the given circuit above result in pulses whose widths are variable as a function to the input, their center location is the same within the cycle. Adjust the circuit to make the starting edge of the generated pulses is constant. This means the pulses are located at the beginning of each cycle, while the width is changing according to the input.
Implement PWM in Matlab

Pulse-Position Modulation

In PPM, the position of a pulse relative to its unmodulated time of occurrence is varied in accordance with the message signal, as illustrated in Fig. 5.8(d) for the case of sinusoidal modulation.



- **Position (P)** of pulses is **varied** proportional to the amplitude of analog Information signal.
- **Amplitude (A)** and **Width (W)** of **pulses** are **constant**.
- The higher **amplitude** of Information signal, the **farther to the right** the pulse is positioned.

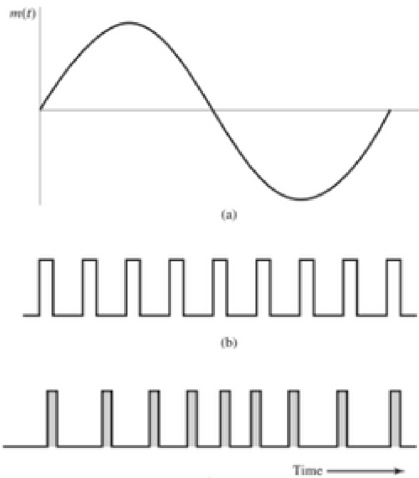
FIGURE 4.3 Illustration of two different forms of pulse-time modulation for the case of a sinusoidal modulating wave. (a) Modulating wave. (b) Pulse carrier. (c) PPM wave.

Experiment : PPM MODULATION USING MULTISIM

Objective: Experiment the PPM in Multisim

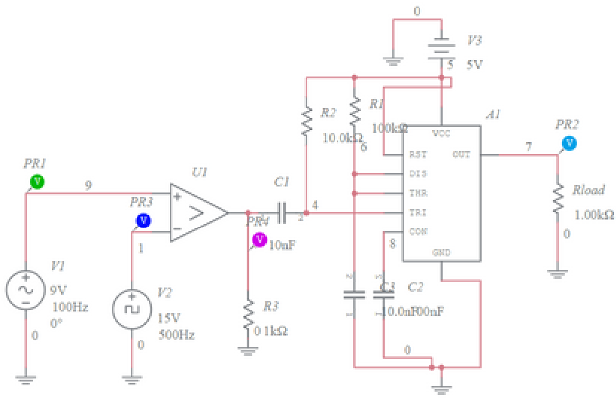
Theory:

The PPM is achieved by comparing the input signal to a reference signal. The position of the pulse output varied proportional to the amplitude of the information signal. The higher amplitude of the information signal the farther apart to the right the pulse is positioned. The plots below illustrate the concept.



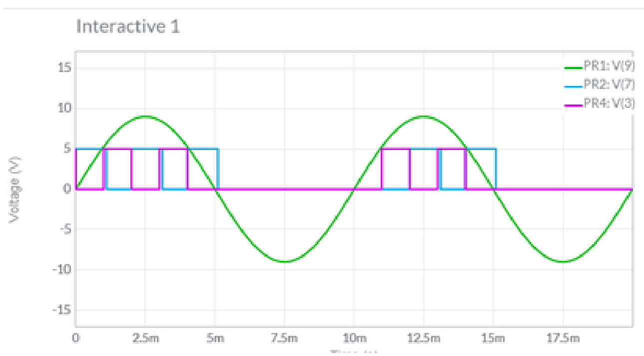
Procedure

1. Implement the following circuit in Multisim.
2. Examine and study the outputs.
3. Change the given values and compare



CIRCUIT FOR PPM EXPERIMENT

RESULT :



Pulse Amplitude Modulation :

In pulse-amplitude modulation (PAM), the amplitudes of regularly spaced pulses are varied in proportion to the corresponding sample values of a continuous message signal; the pulses can be of a rectangular form or some other appropriate shape. Pulse-amplitude modulation as defined here is somewhat similar to natural sampling, where the message signal is multiplied by a periodic train of rectangular pulses.

In digital circuit technology, these two operations are jointly referred to as “sample and hold.” One important reason for intentionally lengthening the duration of each sample is to avoid the use of an excessive channel bandwidth, since bandwidth is inversely proportional to pulse duration.

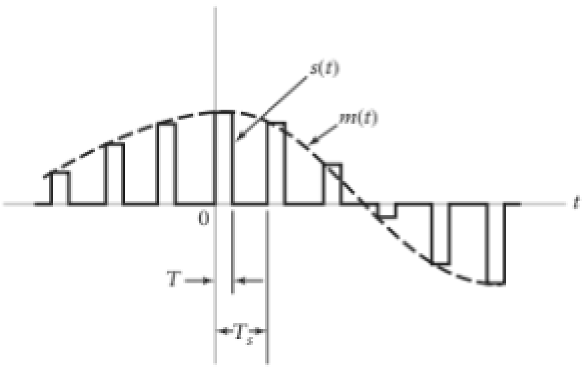


FIGURE 4.4 Flat-top sampling of a message signal.

The waveform of a PAM signal is illustrated in Fig. 4.4. The dashed curve in this figure depicts the waveform of the message signal and the sequence of amplitude modulated rectangular pulses shown as solid lines represents the corresponding PAM signal s(t). There are two operations involved in the generation of the PAM

1. Instantaneous sampling of the message signal every seconds, where the sampling rate is chosen in accordance with the sampling theorem.

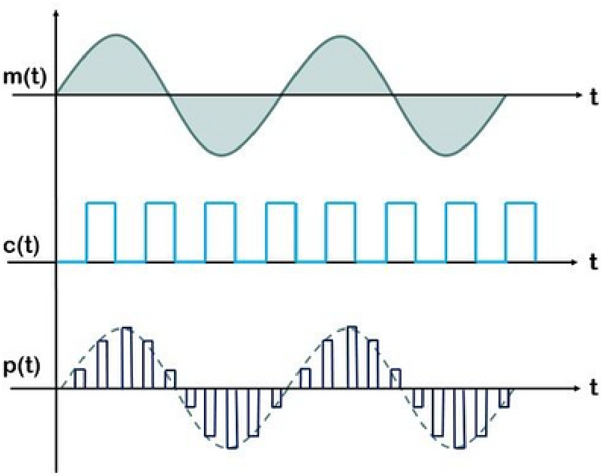
2. Lengthening the duration of each sample, so that it occupies some finite value T.

EXPERIMENT : PAM MODULATION USING MULTISIM

OBJECTIVE: EXPERIMENT THE PAM IN MULTISIM

Theory:

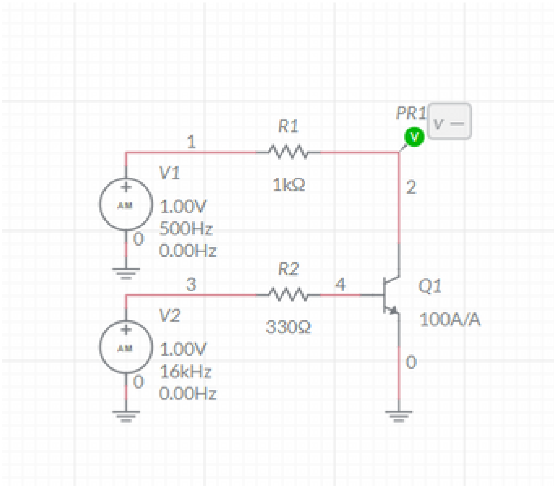
Pulse Amplitude Modulation (PAM) is an analog modulating scheme in which the amplitude of the pulse carrier varies proportional to the instantaneous amplitude of the message signal. Figure below illustrate the concept.



Waveform representation of PAM signal

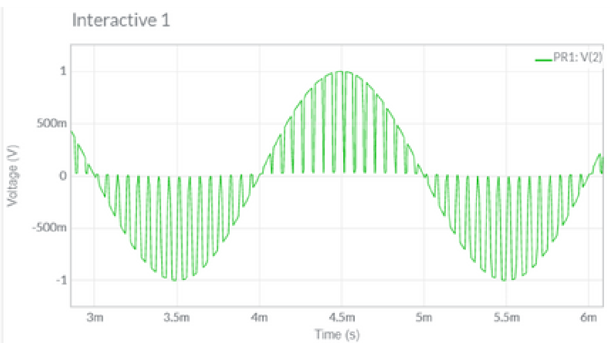
PROCEDURE

1. Implement the following circuit in Multisim.
2. Examine and study the outputs.
3. Change the given values and compare



CIRCUIT FOR PAM EXPERIMENT

RESULT :



Pulse Code Modulation

Pulse-code modulation (PCM) is known as a digital pulse modulation technique is quite complex as compared to the analog pulse modulation techniques i.e. PAM, PWM and PPM, in the sense that the message signal is subjected to a great number of operations . In PCM an analog signal or information is converted into a binary sequence, i.e.,' 1's and '0's. In Pulse Code Modulation, the message signal is represented by a sequence of coded pulses. This message signal is achieved by representing the signal in discrete form in both time and amplitude. The following figure 4.5 shows an example of PCM output with respect to instantaneous values of a given sine wave.

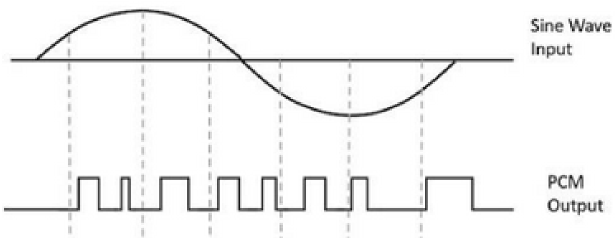


Figure 4.5 An example of PCM output

Basic Elements of Digital Communication System

The basic operations performed in the transmitter of a PCM system are sampling, quantization, and encoding, as shown in Fig. 4.6(a)

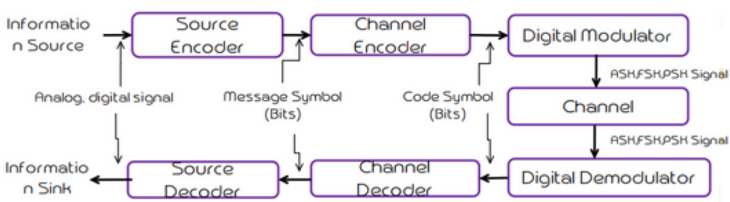


Figure 4.6(a) Basic Elements of Digital Communication System

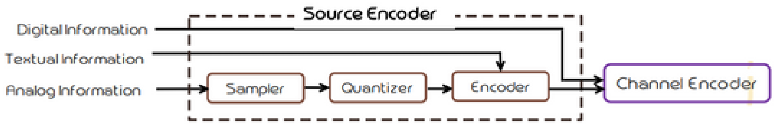


Figure 4.7 Source encoder

Information Source - The source of information can be analog or digital, e.g. Analog - audio, voice; Digital - teletype signal.

Source Encoder – to convert the information signal from source into digital signals (serial bits) by formatting the signals (refer Figure 4.7) and compressed that signal. Then, these bits are grouped to form message symbols. For example: PCM process, Character Encoding (ASCII code) process.

Channel Encoder – is used for error correction coding. It can reduces the probability of error by introduces some redundancy in the message symbols and transform it into code symbols(code words).

Digital Modulator - to convert the serial bits (digital waveform) into electric signals (analog waveform) so that we can transmit them on channel. For example ASK, FSK & PSK Modulation process.

Channel - The communication channel is the physical medium that s used for transmitting signals from transmitter to receiver. In wireless system, this channel consists of atmosphere. For telephony, this channel is wired like twisted pair cable & optical fiber.

Digital Demodulator – the reverse process of modulation and converts the electric signals back to the serial bits (code symbols).

Channel Decoder - to reconstruct the original serial bits (message symbols) from the code symbols used by the channel encoder and the redundancy contained in the received data. Example: Bit Error Rate (BER) process

Signal Decoder - to convert back the serial bits(message symbols) into original source information signal

In digital transmission, any analog information data should be changed into digital signal for the digital transmission.PCM is the only digitally encoded modulation technique that is commonly used for digital transmission. Therefore, PCM technique is needed to convert the discrete sampled signal(usually PAM) to serial bits.

PCM consists of three steps to digitize an analog signal:
i. Sampling
ii. Quantizing
iii.Encoding

RESULT :

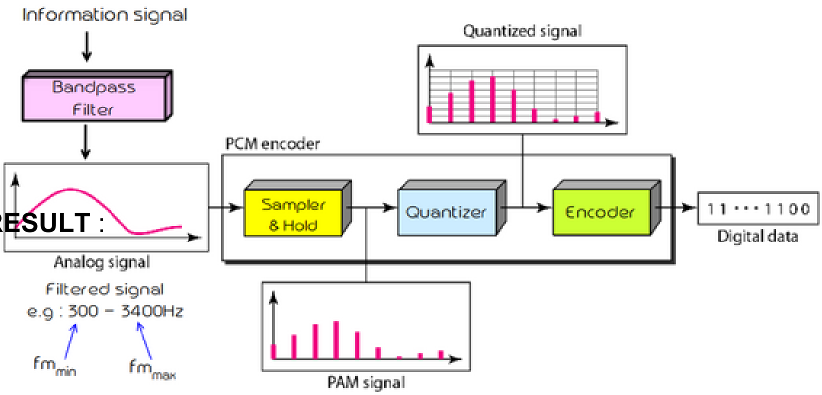


Figure 4.8 Block Diagram of PCM Encoder (Transmitter)

Sampling

Definition: Sampling is a process where the information signal (in analog signal) is sampled by sampling pulse signal which is generated at certain sampling rate,fs. Sampling process will converts an analog signal (in continuous-time signal) to a sampled signal (in discretetime signal) either in PAM, PWM or PPM. For PCM, the sampled signal is PAM signal. By this process, the amplitude of pulses signal is varied proportional to the analog amplitude of information signal

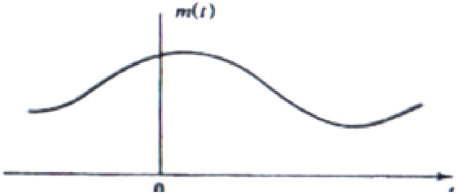


Figure 4.9 Analog information signal

According to Figure 4.9, Analog information signal is sampled every TS seconds. Ts is referred to as the Sampling Interval or Sampling Period.

fs = 1/Ts Sampling Rate or Sampling Frequency.

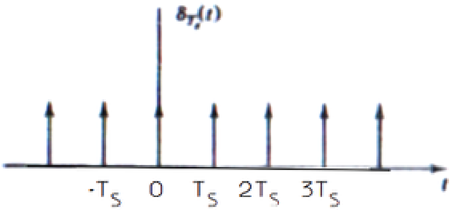


Figure 4.10 Sampling pulse signal

The amplitude of sampling pulse signal is varied ccording to amplitude of information signal which produce a PAM signal.

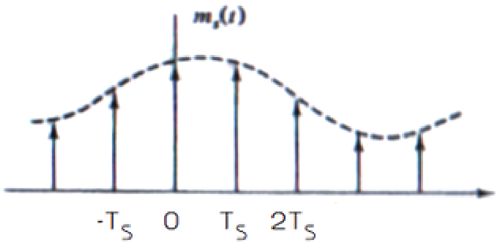


Figure 4.11 PAM signal

There are 3 methods of sampling which are

- Ideal Sampling
- Natural Sampling
- Flat-Top Sampling

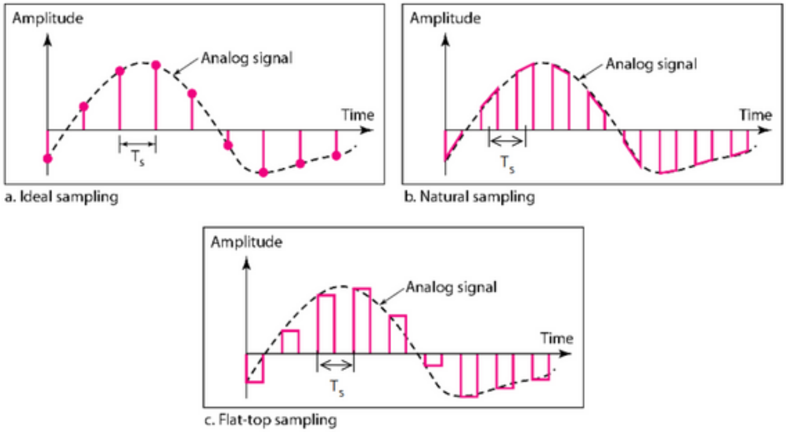


Figure 4.12 Three different sampling methods for PCM

- Ideal Sampling –

Ideal sampling refers to a theoretical process where an analog signal is converted into a digital signal by sampling it at regular intervals. the analog information signal is sampled instantaneously by pulses. This sampling is not practical and cannot be easily implemented.

- Natural Sampling

Unlike ideal sampling, natural sampling does not require a precise clock or a regular sampling interval, which can make it more flexible and adaptable to certain types of signals. However, natural sampling also introduces some challenges in terms of signal reconstruction, since the samples are not evenly spaced and may not be sufficient to accurately represent the original signal. The more practical sampling which is performed by high-speed switching circuits as shown in Figure 4.13.

- Flat-Top Sampling

The simplest and the most popular sampling method which is performed by Sample-and-Hold circuit as shown in Figure 4.14. However, this circuit creates a flat-top (staircase) sampled signal.

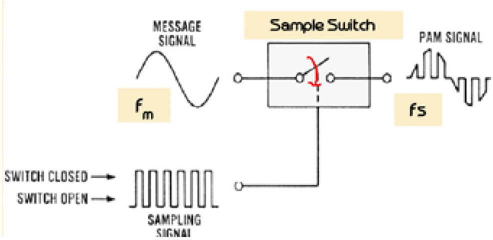


Figure 4.13 Natural sampling

A multivibrator switch is an electronic circuit that generates a square wave output that switches between two stable states. Multivibrators are often used in digital circuits, timing circuits, and power control circuits.

When the pulse is generated,the switch will CLOSED and the amplitude of information signal will be produced.When there is no pulse, the switch will OPEN and there is no output will produce.

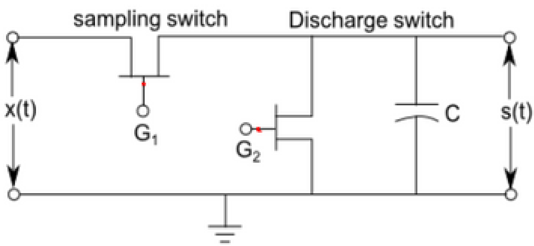


Figure 4.14 Flat-top sampling

Sampling theorem

The sampling theorem, also known as the Nyquist-Shannon sampling theorem, is a fundamental concept in digital signal processing that describes the conditions for accurately converting an analog signal into a digital signal.

According to the Nyquist Sampling theorem; to reconstruct the original signal, the sampling rate must be at least (minimum) two times the highest frequency contained in the info signal.

$$\boxed{f_s \geq 2f_{m(max)}}$$

From Figure 4.14, it could be seen that the higher the sampling rate, fs the smaller sampling interval, Ts, the closer the Recovered signal approaches the original signal.

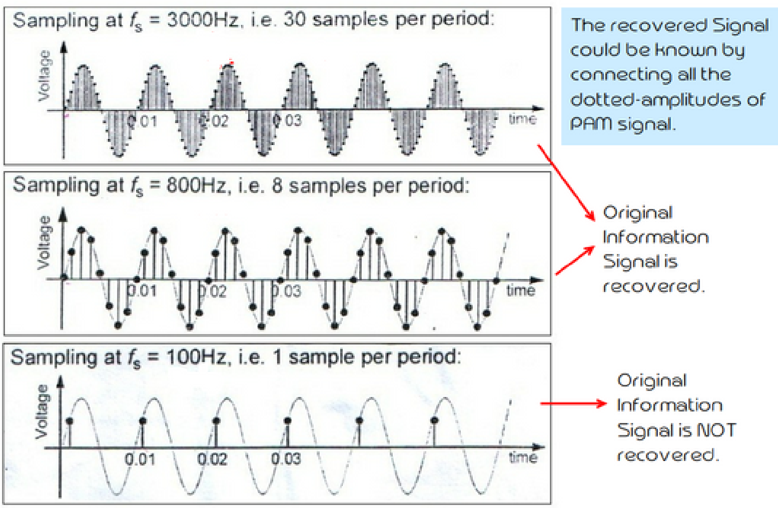


Figure 4.14 Three different Sampling Rate for PCM

For an intuitive example of the Nyquist Theorem, let us sample a simple sine wave at three sampling rates:
i) fs = 2fm (Nyquist rate),
ii) fs = 2(2fm) (2 times the Nyquist rate), and
iii) fs = ½(2fm) (one-half the Nyquist rate)

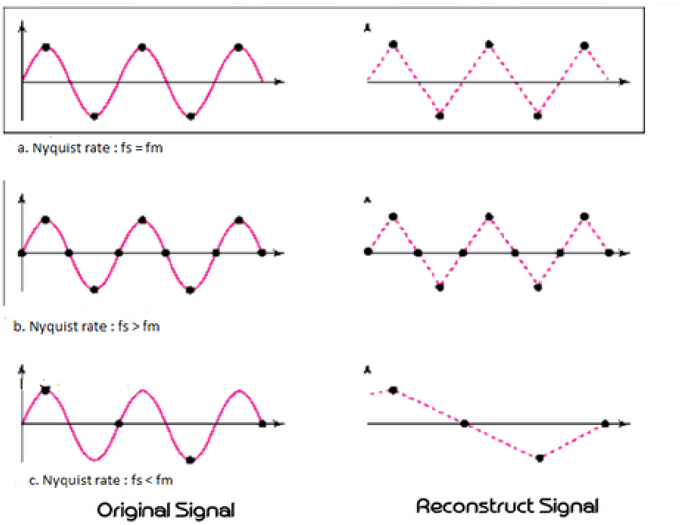


Figure 4.15 Three different Sampling Rate at Nyquist Rate for PCM

- For part (a) It can be seen that sampling at the Nyquist rate can create a good approximation of the original sine wave
- Oversampling in part (b) can also create the same approximation, but it is redundant and unnecessary.
- Sampling below the Nyquist rate (part c) does not produce a signal that looks like the original sine wave.

Nyquist-Shannon Sampling Theorem

If the sampling frequency is lower than the Nyquist rate, then the analog signal cannot be accurately reconstructed from the digital samples, and aliasing occurs. Aliasing is a distortion that occurs when high-frequency components in the analog signal are misrepresented as lower-frequency components in the digital signal. This can lead to loss of information and poor signal quality.

The sampling theorem states that in order to accurately reconstruct an analog signal from its digital samples, the sampling frequency must be at least twice the highest frequency component of the analog signal. This is known as the Nyquist rate or the Nyquist frequency, and it is equal to half of the sampling rate.

Harry Nyquist

Quantizing

Definition: Quantization is a process rounding off the amplitudes of sampled (PAM) signal to a countable number of quantization levels. We assume that the amplitude of the signal m(t) is confined to the range (-mp, +mp). This range (2mp) is divided into L levels, each of step size Δ, given by,

step size, $\Delta = 2\ mp / L$

A sample amplitude value is approximated by the midpoint of the interval in which it lies.

Analog signal has an infinite (uncountable) number of amplitude possibilities. By using the quantization process, the amplitudes of sampled PAM signal is rounding off to a finite(countable) set of quantization levels, L. The number of amplitude levels, L for the quantization depend on the number of bits, n used to code the signal. It use M-ary formula to determine the number of quantization levels, L.

$$\boxed{L = 2^n}$$

Where ;
n = number of bits per level
L = number of finite quantization level

For example, in digital signal processing, analog signals are first converted to digital signals through a process called analog-to-digital conversion (ADC). The resulting digital signal is a continuous stream of numerical values, which can be quantized by rounding each value to the nearest discrete value that can be represented by the system. This quantization process reduces the amount of data needed to represent the signal, which can be beneficial for storage, transmission, and processing.

There are two types of Quantizing method which are;

- Uniform Quantization – uniform step size, Δ
- Non-uniform Quantization – non-uniform step size, Δ

Uniform Quantization

Uniform quantization is a quantization technique where the quantization levels are equally spaced. In other words, the difference between each quantization level is constant. For example, if we have a 16-bit ADC (analog-to-digital converter), and we want to represent the analog signal with 8 bits, we can divide the full scale range into 256 equal intervals, and assign each interval to a unique 8-bit binary number. This is called uniform quantization.

In Uniform type, the quantization levels are uniformly spaced

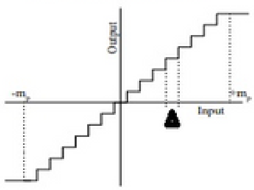


Figure 4.16 Uniform Quantization levels

Non-uniform Quantization

Non-uniform quantization, on the other hand, is a quantization technique where the quantization levels are not equally spaced. This technique is used to allocate more bits to the smaller amplitude signals, and fewer bits to the larger amplitude signals. Non-uniform quantization is used in applications where the dynamic range of the signal is large, and the number of bits used for quantization is limited.

One example of non-uniform quantization is called A-law and μ-law quantization, which are used in telecommunications systems. In A-law quantization, the quantization levels are closer together for smaller amplitude signals, resulting in better signal-to-noise ratio (SNR) for low amplitude signals. In μ-law quantization, the quantization levels are closer together for large amplitude signals, resulting in better SNR for high amplitude signals.

STEPS IN QUANTIZATION

The PCM signal is generated by carrying out following basic operations:

1. Band limiting (using LPF)
2. Sampling
3. Quantizing
4. Encoding

Two fundamental processes are involved in the generation of a PCM signal: sampling and quantization.

Sampling is time discretization

quantization is amplitude discretization.

In PCM, conversion of analog signal to digital signal is done in two steps ;

- Sampling
- Quantization

Below figure shows sampling step:

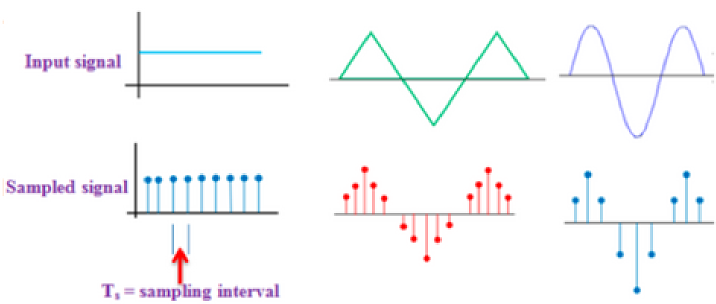


Figure 4.17 Sampling step

Quantization is the process of rounding of the sample value to the nearest quantization level.Number of quantization levels is predefined.

$$\boxed{L = 2^n}$$

STEPS IN QUANTIZATION

1. Total voltage range is divided into q equal intervals of step size, Δ

step size, Δ

$$= \frac{V_H - V_L}{q} = \frac{V_H - V_L}{2^n}$$

where VH = Max. voltage value
VL = Min. voltage value

2. Draw mid lines representing quantization levels

3. Assign binary codes (pre-defined) to each quantization level

4. Calculate quantization error

Quantization Example

A 2-bit PCM modulator is used with a 0-1 V signal. What is the binary digital value that will occur for the following inputs: 0.4 V, 0.78 V. What is the quantization error for these two samples?

n = number of bits = 2
L = No. of quantization levels
= 2n
= 4

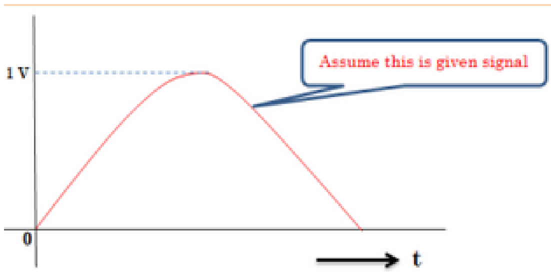


Figure 4.18 Information signal

STEPS IN QUANTIZATION

1. Total voltage range is divided into q equal intervals of step size S
2. Draw mid lines representing quantization levels
3. Assign binary codes (pre-defined) to each quantization level
4. Calculate quantization error

Step 1: Step size $S = \frac{V_H - V_L}{2^n} = \frac{1-0}{4} = \frac{1}{4} = 0.25$

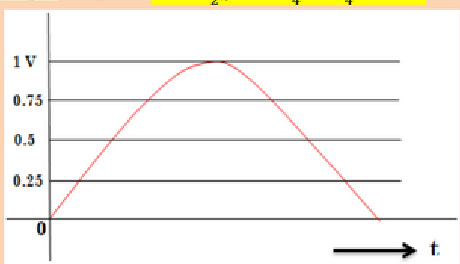
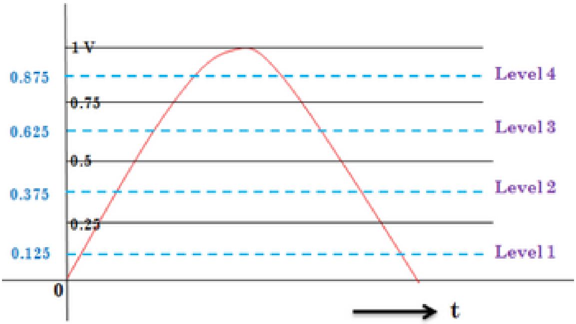
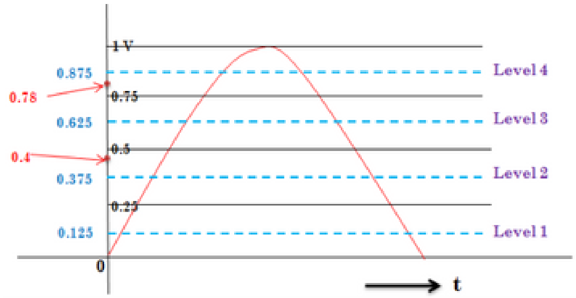


Figure 4.19 signal divided into equal interval of step S

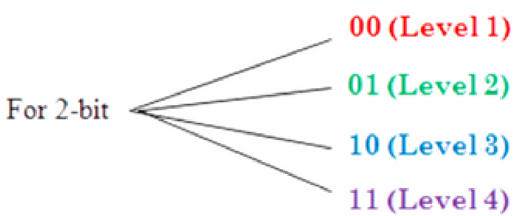
Step 2: Draw mid-lines, which represents quantization levels



Step 3: Denote given sample values i.e., 0.4 V & 0.78 V



Step 4: Represent each quantization level with predefined binary code



Quantizing 0.4 V sample value: See that level 2 is near to 0.4 V. So, for sample voltage 0.4 V, 01 code is transmitted.
Quantization error e = 0.4 – 0.375 = 0.025 V

NOTE:
Quantization process introduces a certain amount of error or distortion. This error known as quantization noise and is minimised by increasing the number of quantization levels. But increasing number of quantization levels increases number of bits to represents each sample and hence increases bit rate and cost of transmission.

Exercise

Pulse Code Modulation (PCM) is the process of converting technique from analog signal to digital signal. PCM consists of three steps to digitize an analog signal which are sampling, quantizing and encoding.The data output of sampling is given in Table B1 below.

Table B1

TIME INTERVAL (Ts)	5ms	10ms	15ms	20ms	25ms
VALUE SAMPLE	-0.25 V	3.1 V	1.5 V	- 2.4 V	- 3.7V

From the data shows in table B1, each sampled signal's amplitude is quantized (rounding-off) to the midpoint of the interval (quantization level) in which it lies. You are required to plot the quantizing signal graph which confines between two limits: Vmax = 4V and Vmin = -4V. Then, calculate the number of bit, the number of quantization level , and quantization step size for each sample for the PCM system created .

Answer

Quantization level, L $= 2^n$

$= 2^3$

$= 8$ level

Step size, $\Delta = 2$ Vmax/L

$= 2 (4)/8$

$=1$V

Δ = Step Size / Quantization Interval / Resolution / Quantum

$\Delta = 2$ Vmax /L

ZONE AREA	MID POINT ZONE
-4V to -3V	-3.5V
-3V to -2V	-2.5V
-2V to -1V	-1.5V
-1V to 0V	-0.5V
0V to 1V	0.5V
1V to 2V	1.5V
2V to 3V	2.5V
3V to 4V	3.5V

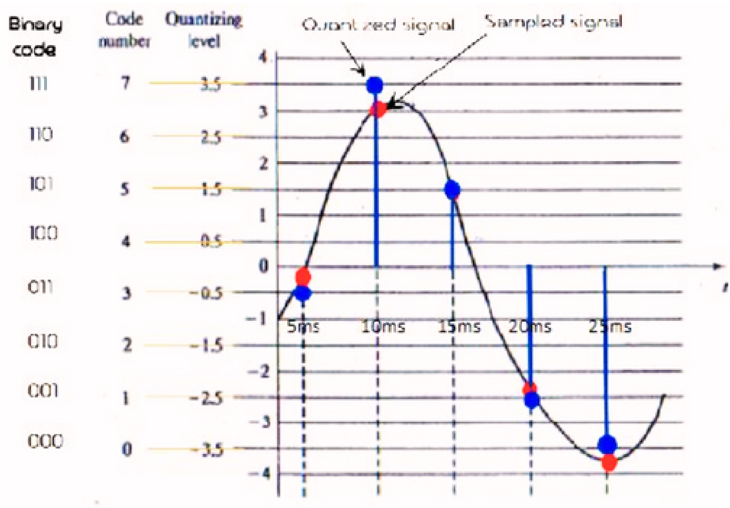


Figure 4.20 Quantized signal and sampled signal

TIME INTERVAL (Ts)	5ms	10ms	15ms	20ms	25ms
VALUE SAMPLE	-0.25 V	3.1V	1.5 V	-2.4 V	-3.7 V
Quantization level	-0.5V	3.5V	1.5V	- 2.5V	-3.5V
Decimal code	3	7	5	1	0
Binary code	011	111	101	001	000

EXPERIMENT : PCM MODULATION USING MULTISIM

OBJECTIVE: EXPERIMENT THE PCM IN MULTISIM

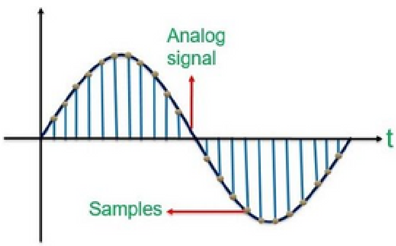
Theory:

Pulse Code Modulation (PCM) is a technique by which analog signal gets converted into digital form in order to have signal transmission through a digital network . It is abbreviated as PCM.

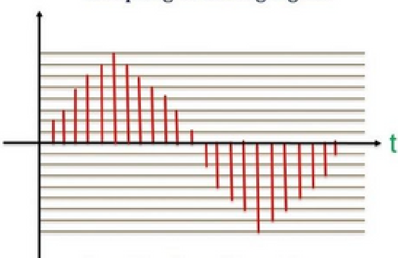
The analog message signal is first sampled, and then the amplitude of the sample is approximated to the nearest set of quantization level. This allows the representation of time and amplitude in a discrete manner. Thereby, generating a discrete signal.

This discrete signal is then converted into its binary form for the transmission of the signal.

It is to be noted here that, in PCM technique the signal gets transmitted in the coded format and must be decoded at the receiver in order to have the original message signal.



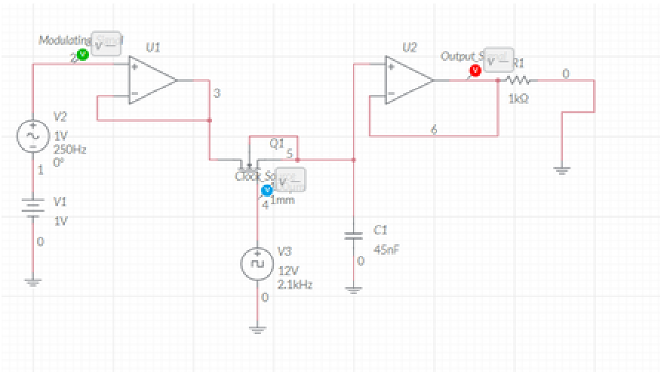
Sampling of analog signal



Quantization of samples

PROCEDURE

1. Implement the following circuit in Multisim.
2. Examine and study the outputs.
3. Change the given values and compare



CIRCUIT FOR PCM EXPERIMENT

RESULT :

EXERSISE :

An information signal in form $V_m(t) = 20\cos(60\pi \times 10^3 t)\,V$ to be transmitted through a PCM system. The signal is sampled at the rate of 15% higher than the minimum sampling frequency and quantization le used is 512. Calculate the sampling frequency that can be used, the number of bit per sample, the transmission rate, the resolution step

Answer

i. The sampling frequency that can be used,

Wm = $2\pi f_m = 60\pi \times 10^3$

Fm = $60\pi \times 10^3/2\pi$

= 30 kHZ

Fs = 2fm + (15% + 2fm)

= 2(30k) + (15% + 2(30k))

= 69Khz

ii. The number of bit per sample,

Bil bit = m ^2

Bil bit , $n = \frac{logL}{log2}$ = log 512 /log 2 = 9 bit

iii. The transmission rate,

Bit Rate = nfs = 9 (69 K) = 6210 bps

iv. The resolution step,

Resolution step , $\Delta = \frac{V_{pp}}{L} = \frac{2V_p}{L}$ = 2(20)/512 = 0.078 v

v. The quantization error

Qe = , $\Delta/2$ =0.078/2 =0.039 v

ENCODING

Definition : Encoding is a process of translating the quantized signal into a decimal code number. Encoding in PCM actually refers to the process of converting the quantized samples into digital code numbers. After quantization, each sample is represented by a specific numerical value that falls within a range of possible values determined by the number of bits used for quantization.The encoding process assigns a unique digital code number to each quantized sample. The code numbers are usually represented in binary form and are assigned based on the amplitude value of the quantized sample.The number of bits, n for each level of code number depends on the number of quantization level, L used to quantize the samples which can be determined using M-ary formula

$$n = \log_2 m$$

Where
n = number of bits,
M = number of bits

For example, in 8-bit PCM, there are 256 possible quantization levels, and each level is assigned a unique decimal code number ranging from 0 to 255. To convert these decimal code numbers to their binary sequence representation, each decimal number is represented as a series of 8 binary digits (bits), where each bit can be either 0 or 1.

Here is an example table showing the decimal code numbers and their corresponding binary sequences for 8-bit PCM:

Quantization Level	Decimal Code Number	Binary Sequence
0	0	00000000
1	1	00000001
2	2	00000010
3	3	00000011
...	...	...
253	253	11111101
254	254	11111110
255	255	11111111

Each decimal code number is represented by a unique binary sequence, with 8 bits used for encoding in this example. The binary sequence representation is used to represent the quantized samples in the digital audio signal.

DIGITAL PULSE MODULATION (DPM)

Pulse Code Modulation (PCM) is an extension of PAM wherein each analogue sample Analogue is quantized into a discrete value for representation as a digital code word.a PAM system can be converted into a PCM system by adding a suitable analogue-to-digital (A/D) converter at the source and a digital-to-analogue (D/A) converter at the destination.

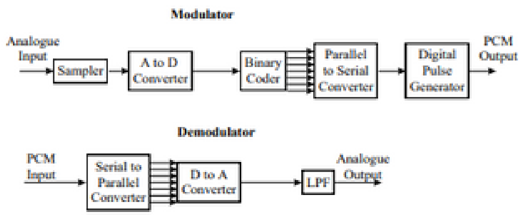


Figure 4.21 Block diagram of PCM

PCM is a true digital process as compared to PAM. In PCM the speech signal is converted from analogue to digital form

PCM is standardised for telephony by the ITU-T (International Telecommunications Union - Telecoms,

In DPM, a code is used to represent the amplitude of the samples that has been divided into various levels. The types of DPM:

- Delta Modulation (DM)
- Differential Pulse Code Modulation
- Adaptive Differential Pulse Code Modulation ADPCM
- Delta-Sigma Modulation

Delta Pulse Code Modulation

Delta Pulse Code Modulation (DPCM) is a variant of Pulse Code Modulation (PCM) that reduces the amount of data required to represent a signal. DPCM achieves this by only encoding the difference between adjacent samples rather than the absolute value of each sample.

Delta modulation converts an analogue signal, normally voice, into a digital signal.The analogue signal is sampled as in the PCM process. Then the sample is compared with the previous sample.

The result of the comparison is quantified using a one bit coder. If the sample is greater than the previous sample a 1 is generated. Otherwise a 0 is generated. The advantage of delta modulation over PCM is its simplicity and lower cost. But the noise performance is not as good as PCM.

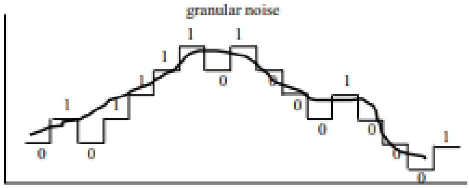


Figure 4.22 Delta Modulation Waveform

The reconstructed signal oscillates by 1 step size in every sample. It can be reduced by decreasing the step size. This requires that the sample rate be increased. Delta Modulation requires a sampling rate much higher than twice the bandwidth. It requires oversampling in order to obtain an accurate prediction of the next input, since each encoded sample contains a relatively small amount of information. Delta Modulation requires higher sampling rates than PCM.

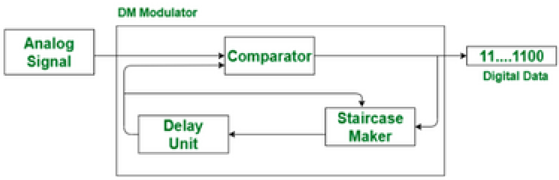


Figure 4.23 Block diagram of Delta Modulation

Differential PCM (DPCM)

DPCM is also designed to take advantage of the redundancies in a typical speech waveform. In DPCM the differences between samples are quantized with fewer bits that would be used for quantizing an individual amplitude sample. The sampling rate is often the same as for a comparable PCM system, unlike Delta Modulation.

In Differential Pulse Code Modulation, the decoder reproduces the signal by adding the decoded differences to the predicted value to reconstruct the original signal. In DPCM, the encoder first predicts the value of the next sample based on the previous sample, and then encodes the difference between the predicted value and the actual value of the next sample. This difference is then quantized and encoded using PCM.

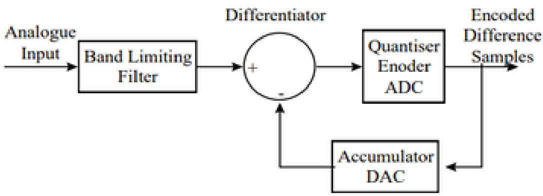


Figure 4.24 Differential PCM (DPCM) and ADPCM

One variant of Differential Pulse Code Modulation is called Adaptive Differential Pulse Code Modulation (ADPCM), which modifies the quantization step size to adjust to changes in the input signal. ADPCM can achieve better compression ratios than DPCM by varying the step size based on the properties of the input signal.

Adaptive Differential Pulse Code Modulation ADPCM

ADPCM is standardised by ITU-T recommendations G.721 and G.726. The method uses 32,000 bits/s per voice channel, as compared to standard PCM's 64,000 bits/s. Four bits are used to describe each sample, which represents the difference between two adjacent samples. Sampling is 8,000 times a second. It makes it possible to reduce the bit flow by half while maintaining an acceptable quality. While the use of ADPCM (rather than PCM) is imperceptible to humans, it can significantly reduce the throughput of high-speed modems and fax transmissions.

To implement ADPCM the original (audio) signal is sampled as for PCM to produce a code word. This code word is manipulated to produce the predicted code word for the next sample. The new predicted code word is compared with the code word of the second sample. The result of this comparison is sent to line. Therefore we need to perform PCM before ADPCM.

ADPCM is sometimes used by telecom operators to fit two speech channels onto a single 64 kbit/s link. This was very common for transatlantic phone calls via satellite up until a few years ago. Now, nearly all calls use fibre optic channels at 64 kbit/s.

Delta-Sigma Modulation (DSM)

Delta-Sigma Modulation (DSM) is a digital signal processing technique that converts analog signals into digital signals with high precision and low noise. It is commonly used in applications such as audio converters, digital filters, and sensors.

DSM works by oversampling the analog signal at a very high rate, typically several hundred or thousand times the Nyquist rate. The oversampled signal is then passed through a low-pass filter, which removes the high-frequency components and produces a stream of high-resolution, low-frequency samples.

The difference between each sample and the quantized value of the previous sample is then quantized using a one-bit quantizer, which produces a binary output (i.e., either 0 or 1). This binary output is then fed back into the low-pass filter to create a feedback loop. The loop adds noise shaping to the signal, meaning that the quantization noise is moved to higher frequencies, where it can be filtered out more easily.

The output of the feedback loop is a high-resolution, low-noise digital signal that can be used for further processing or conversion to other formats.

Delta-Sigma Modulation is particularly useful in applications that require high precision and low noise, such as audio and measurement systems. Its oversampling technique enables it to achieve high resolution with a low number of bits, while its noise-shaping technique reduces the impact of quantization noise on the final output.

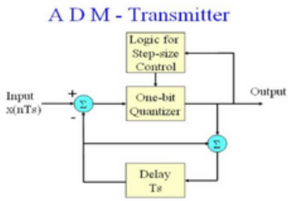


Fig 2.21 Block Diagram of ADM Transmitter

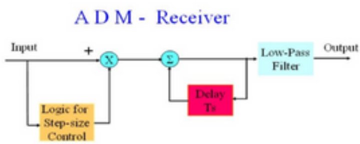


Figure 4.25 Block Diagram of ADM Receiver

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