CONTENTS

LEARNING OBJECTIVES

By the end of this lecture, students will be able to:

1. Understand the basic concept and purpose of communication systems in engineering applications.
2. Identify and describe the fundamental components of a communication system.
3. Explain different types of signals and their classifications used in communication systems.
4. Define key signal parameters such as frequency, phase, and wavelength.
5. Distinguish between baseband and passband communication systems.
6. Understand the basic concept of modulation and its importance in signal transmission.
7. Relate communication system principles to medical and biomedical applications.

1 INTRODUCTION

Communication systems play a fundamental role in modern engineering by enabling the transfer of information from one point to another through various transmission media. In engineering applications—particularly in medical instrumentation and biomedical systems—communication systems are essential for transmitting physiological signals such as ECG, EEG, EMG, blood pressure, and other vital parameters from sensors to monitoring and processing units.

A communication system generally consists of a **transmitter**, **channel**, and **receiver**. The transmitter processes and prepares the message signal for transmission, often through encoding and modulation. The channel provides the physical medium—wired or wireless—through which the signal travels. The receiver then demodulates and decodes the signal to recover the original information. During transmission, signals are often affected by noise and distortion, making proper system design critical for reliable communication.

Signals used in communication systems can be classified in several ways: continuous-time or discrete-time, analog or digital, deterministic or random, periodic or aperiodic. Understanding signal parameters such as **frequency**, **phase**, and **wavelength** is essential for analyzing system behavior. Furthermore, communication systems are categorized into **baseband** and **passband** systems. Baseband systems transmit signals in their original frequency range, while passband systems use modulation techniques to shift signals to higher frequencies suitable for long-distance transmission.

Modulation is a key concept in communication systems. In analog communication, common modulation techniques include:

- **Amplitude Modulation (AM)**
- **Frequency Modulation (FM)**
- **Phase Modulation (PM)**

Among these, Amplitude Modulation (AM) and its variations—such as Double Sideband Large Carrier (DSB-LC), Double Sideband Suppressed Carrier (DSB-SC), Single Sideband (SSB), and Vestigial Sideband (VSB)—are fundamental techniques that illustrate the trade-offs between power efficiency and bandwidth utilization.

In medical communication systems, these principles are applied in wireless patient monitoring, telemetry systems, medical imaging transmission, and remote healthcare technologies. A solid understanding of communication fundamentals therefore forms the foundation for advanced study in biomedical and medical communication systems engineering.

2 FUNDAMENTAL OF COMMUNICATION SYSTEMS

2.1 SIGNAL PARAMETERS

2.1.1 FREQUENCY

Frequency is measured in hertz (Hz), equivalent to cycles per second: thus 1 Hz is 1c/s. The electromagnetic spectrum consists of frequencies from just above 0 Hz to ∞Hz. The frequency spectrum is broken up into different segments which indicate the usage for particular frequency range as shown in Figure 2.

$$f = \frac{1}{T} \qquad (1)$$

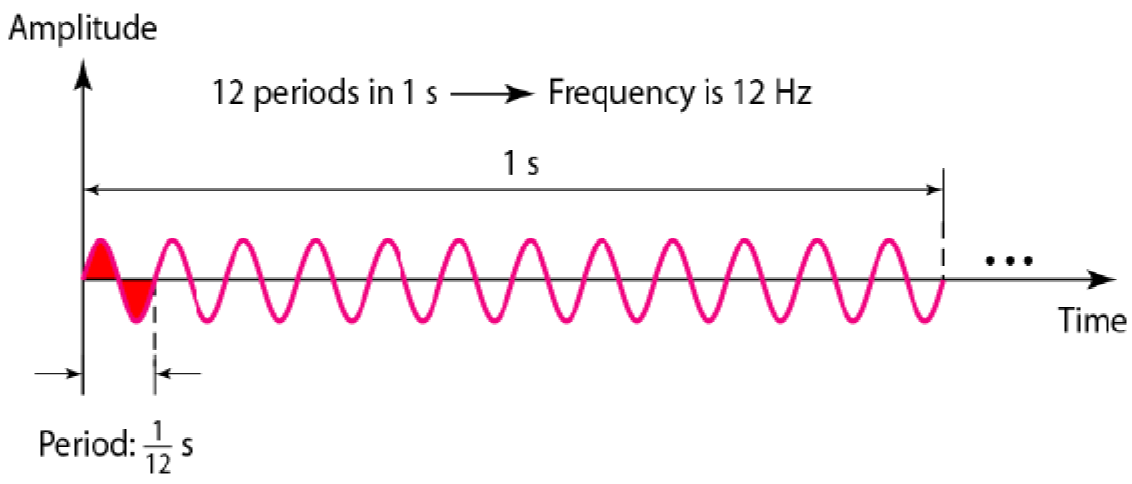


Figure 1: Signal with 12 Hz.

EXAMPLE 1 : The power we use at home has a frequency of 50 Hz . The period of this sine wave can be determined as follows:

Solution

$$T = \frac{1}{f} = \frac{1}{50} = 0.02 \text{ s} = 20 \text{ ms}$$

EXAMPLE 2 : The period of a signal is 100 ms . What is its frequency in kilohertz?

Solution

First we change 100 ms to seconds, and then we calculate the frequency from the period ($1\ \text{Hz} = 10^{-3}\text{kHz}$).

Class			Wave-length λ	Freq-uency f	Energy per photon E
Ionizing radiation	γ	Gamma rays	10 pm	30 EHz	124 keV
	HX	Hard X-rays	100 pm	3 EHz	12.4 keV
	SX	Soft X-rays	10 nm	30 PHz	124 eV
	EUV	Extreme ultraviolet	121 nm	3 PHz	10.2 eV
	NUV	Near ultraviolet	400 nm	750 THz	3.1 eV
		Visible spectrum	700 nm	480 THz	1.77 eV
Infrared	NIR	Near infrared	1 μm	300 THz	1.24 eV
	MIR	Mid infrared	10 μm	30 THz	124 meV
	FIR	Far infrared	100 μm	3 THz	12.4 meV
Micro-waves[11]	EHF	Extremely high frequency	1 mm	300 GHz	1.24 meV
			1 cm	30 GHz	124 μeV
	SHF	Super high frequency	1 dm	3 GHz	12.4 μeV
	UHF	Ultra high frequency	1 m	300 MHz	1.24 μeV
Radio waves[11]	VHF	Very high frequency	10 m	30 MHz	124 neV
	HF	High frequency	100 m	3 MHz	12.4 neV
	MF	Medium frequency	1 km	300 kHz	1.24 neV
	LF	Low frequency	10 km	30 kHz	124 peV
	VLF	Very low frequency	100 km	3 kHz	12.4 peV
	3	Band 3	1 Mm	300 Hz	1.24 peV
	2	Band 2	10 Mm	30 Hz	124 feV
	1	Band 1	100 Mm	3 Hz	12.4 feV

Figure 2: Electric Field Lines due to different charges.

2.1.2 PHASE OF SIGNALS

Phase describes the position of the waveform relative to time 0.

EXAMPLE 3 : A sine wave is offset $1/6$ cycle with respect to time 0 . What is its phase in degrees and radians?

Solution

$$\frac{1}{6} \times 360 = 60^{\circ} = 60 \times \frac{2\pi}{360}\text{rad} = \frac{\pi}{3}\text{rad} = 1.046\text{rad}$$

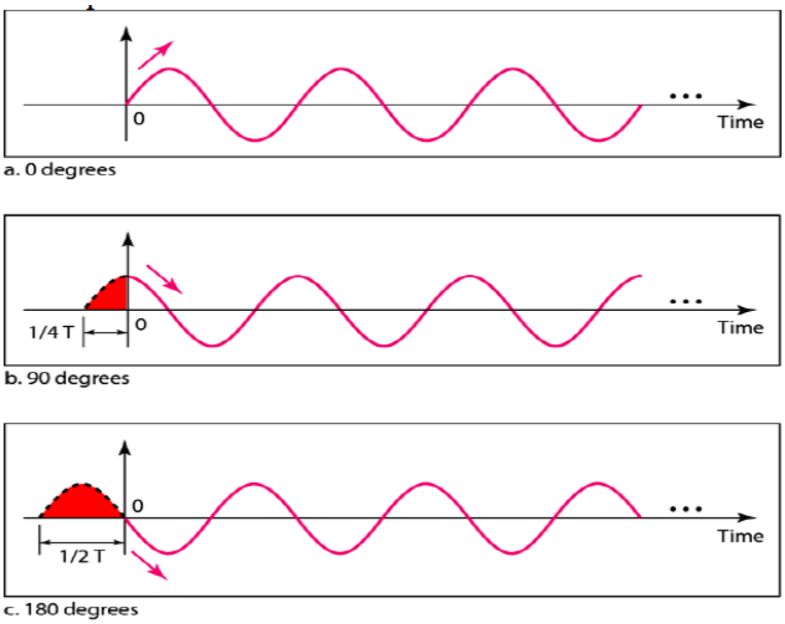


Figure 3: Different examples of signal phase.

2.1.3 WAVE LENGTH

Wavelength is the spatial period of a periodic wave, defined as the distance between consecutive corresponding points of the same phase, such as two adjacent crests, troughs, or zero crossings.

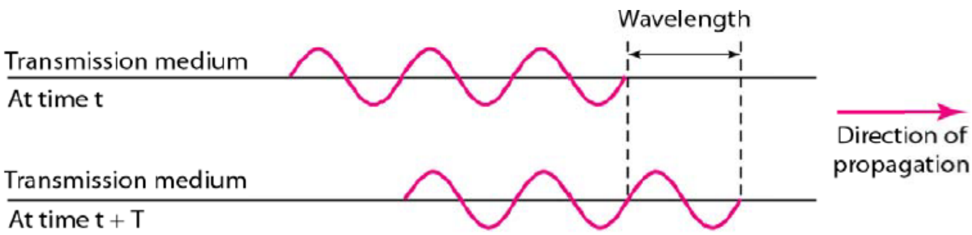


Figure 4: Wave length.

2.2 CLASSIFICATION OF SIGNALS

A signal is a function of one or more independent variables which contain some information.

For examples: Radio signal, TV signal, Telephone signal etc. For instance, in a RC circuit the signal may represent the voltage across the capacitor or the current flowing in the resistor. Mathematically, a signal is represented as a function of an independent variable(t). Usually (t) represents time. Thus, a signal is denoted by x(t).

2.2.1 CONTINUOUS-TIME AND DISCRETE-TIME SIGNALS

Continuous time (CT) signals are defined for all values of time. It is also called as an analog signal and is represented by x(t).

Discrete time (DT) signals are defined at discrete instances of time. It is represented by x[n].

Since a discrete-time signal is defined at discrete times, a discrete-time signal is often identified as a sequence of numbers, denoted by $\{x_n\}$ or $x[n]$, where n = integer.

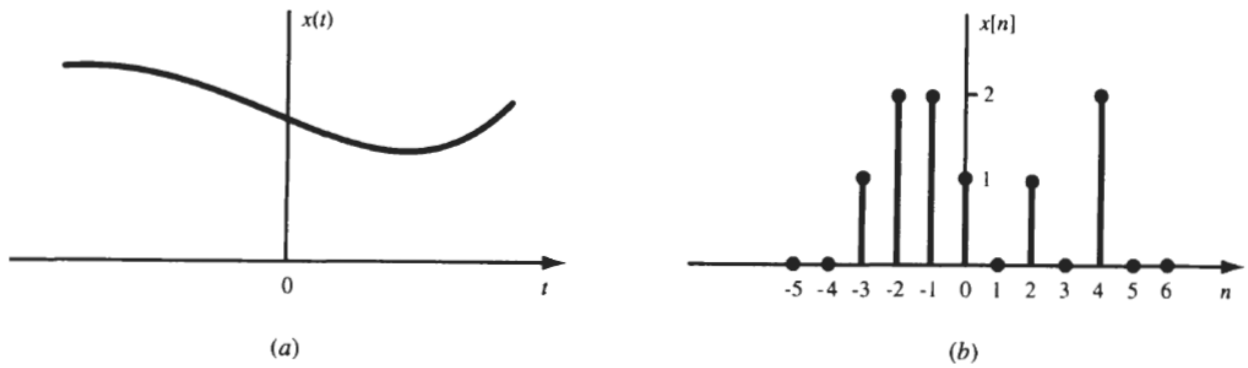
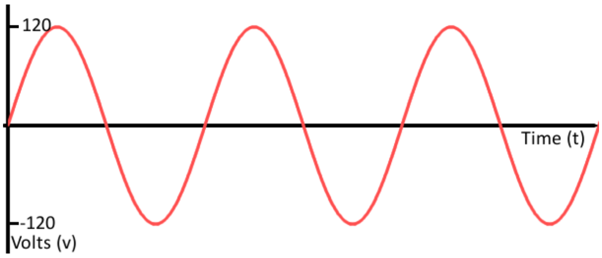


Figure 5: continuous-time signal x(t) and of a discrete-time signal x[n].

2.2.2 ANALOG AND DIGITAL SIGNALS

If a continuous-time signal $\boldsymbol{x}(\boldsymbol{t})$ can take on any value in the continuous interval (a, b), where **a** may be $-\infty$ and **b** may be $+\infty$, then the continuous-time signal $\boldsymbol{x}(t)$ is called an analog signal. If a discrete-time signal $\boldsymbol{x}[\boldsymbol{n}]$ can take on only a finite number of distinct values, then we call this signal a digital signal.



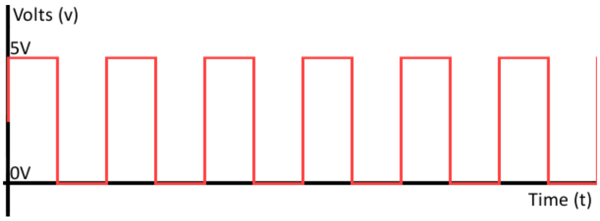


Figure 6: Analog vs Digital signals.

2.2.3 REAL AND COMPLEX SIGNALS

A signal $\boldsymbol{x}(\boldsymbol{t})$ is a real signal if its value is a real number, and a signal $\boldsymbol{x}(\boldsymbol{t})$ is a complex signal if its value is a complex number. A general complex signal $\boldsymbol{x}(\boldsymbol{t})$ is a function of the form

2.2.4 DETERMINISTIC AND RANDOM SIGNALS

Deterministic signals are those signals whose values are completely specified for any given time. Thus, a deterministic signal can be modeled by a known function of time t or A deterministic signal is one which can be completely represented by Mathematical equation at any time . For example:- $\boldsymbol{x}(\boldsymbol{t}) = \boldsymbol{coswt}.$ Random signals are those signals that take random values at any given time and must be characterized statistically or A random signal is one which cannot be represented by any mathematical equation. For example:- Noise generated in electronic components, transmission channels, cables etc.

2.2.5 PERIODIC AND NONPERIODIC SIGNALS

A signal is said to be ***periodic signal*** if it repeats at equal intervals. ***Aperiodic signals*** do not repeat at regular intervals. Periodic analog signals can be classified as ***simple*** or ***composite***. A simple periodic analog signal, a sine wave, cannot be decomposed into simpler signals. A composite periodic analog signal is composed of multiple sine waves.

2.3 BASEBAND VS. PASSBAND COMMUNICATION SYSTEMS

Communication systems can be broadly categorized into two types based on the frequency range used for transmitting information: **baseband systems** and **passband systems**. This classification depends on whether the information signal is transmitted directly or after frequency modification.

1. **Baseband Transmission:**
   - In a baseband system, the information signal is sent directly, without any modulation or frequency shifting. This means the signal retains its original frequency range throughout transmission.
   - Baseband signals are typically low-frequency signals, often close to zero. Common examples include:
     - **Human voice** signals, with frequencies ranging between 20 Hz and 5 kHz.
     - **Video signals** from TV cameras, which range from 0 Hz up to about 5.5 MHz.
   - Many systems use baseband transmission when transmitting signals locally, such as in-home or office settings. For instance, in a telephone system, local calls (e.g., calls within a neighborhood) transmit audio in its baseband form without modification.
2. **Passband Transmission:**
   - Passband transmission, on the other hand, involves shifting the original signal to a higher frequency before transmission. This frequency-shifting process, known as modulation, is necessary when the signal must travel long distances, especially over wireless mediums.
   - In passband systems, the transmitted signal is modulated to fit within a specific frequency band suitable for long-range transmission and is demodulated back to its original form at the receiver.

- A common application of passband transmission is in long-distance telephone calls. When a call is transmitted over microwave or satellite links, the audio signal is modulated to a higher frequency to suit the transmission medium. Similarly, transmitting a video signal from a camera to a television through wires might use baseband transmission, but if sent via satellite, the signal is converted to a passband frequency.

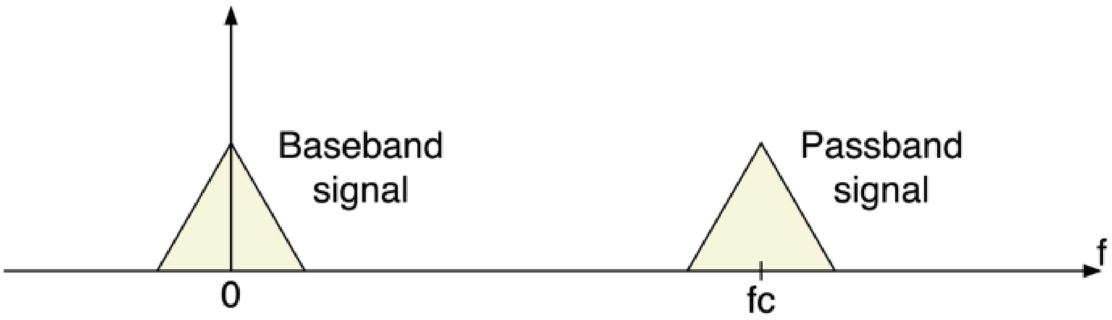


Figure 7: Baseband vs Passband.

2.4 MODULATION AND DEMODULATION

Modulation is the process of shifting a baseband signal to a higher frequency range, known as the passband, to prepare it for transmission. Conversely, demodulation is the process of converting this passband signal back to the baseband frequency range at the receiver end. During modulation, one or more properties of a carrier wave—typically a sinusoidal wave—are altered according to the information signal intended for transmission. The carrier's characteristics that can be modified include amplitude, frequency, or phase, leading to different types of modulation:

- ☒ Amplitude Modulation (AM)
- ☒ Frequency Modulation (FM)
- ☒ Phase Modulation (PM).

3 MAIN PARTS OF COMMUNICATION SYSTEMS

The main parts can be shown in Figure 8 and described as follows:

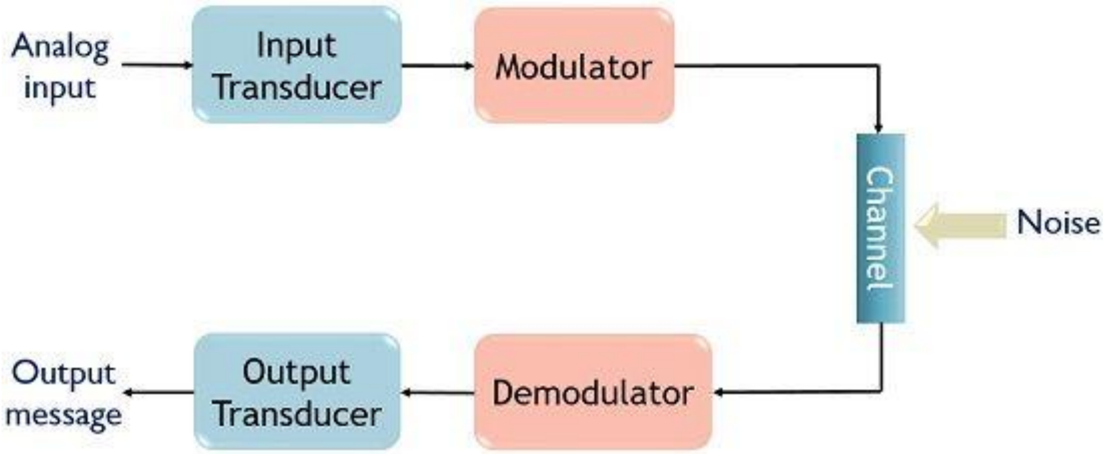


Figure 8: Simple diagram of analog communication system.

- **Transmitter:** A transmitter is a device which is used for transmission of signal in the communication channel. It includes many other processes like encoding, modulation, and amplification etc, basically it transmits the input data to the communication channel and its receiver.
- **Receiver:** The receiver is like a opposite of the transmitter as it receives the transmitted signal across the communication channel and then it processes it to extract the original message or information. This process includes many stages such as demodulation, decoding, and sometimes amplification to amplify or enhance the quality of the signal received.
- **Channel:** The channel is the path or we can say medium through which the signal or the information being transmitted travels from the transmitter to the receiver. A channel could be wired such cables or optical fibers or wireless like radio waves or microwaves. During this time some noise get into it and cause

many disturbances to the signal, its noise and some distortion in signal which should be reduced or eliminated for successful transmission.

- **Encoder**: Encoders are used for the process of encoding the original signal in simple words we can say that encoders are used to convert the message or information into a format which is suitable for transmission through the chosen communication channel, it is known as encoding of message.

- **Decoder**: Decoders are the opposite of encoders, they are useful at the receiving end to reverse the encoding process and get the original message from the received encoded signal by extracting from it, this process is known as decoding we get the original message by the end.

4 AMPLITUDE MODULATION (AM)

Amplitude Modulation (AM) is a technique in which the amplitude of a high-frequency carrier signal is varied in direct proportion to the instantaneous value of the information signal, while keeping the frequency and phase of the carrier constant. In essence, AM works by changing the carrier wave's amplitude based on the amplitude of the baseband or information signal, $m(t)$ around a constant average value.

The AM process requires two inputs:

1. Carrier Signal: A high-frequency, constant-amplitude signal, usually a sinusoidal wave, which serves as the primary medium for carrying the information.

2. Information Signal: A lower-frequency signal containing the actual information to be transmitted. This could be a simple sinusoidal wave or a complex waveform with multiple frequencies, depending on the nature of the data.

In AM, the carrier's amplitude is adjusted in a linear relationship with the amplitude of the information signal, effectively "embedding" the information within the variations of the carrier's strength. This method makes it possible to transmit the information signal across a communication channel using a high-frequency carrier that can travel greater distances and penetrate obstacles more effectively than low-frequency signals alone.

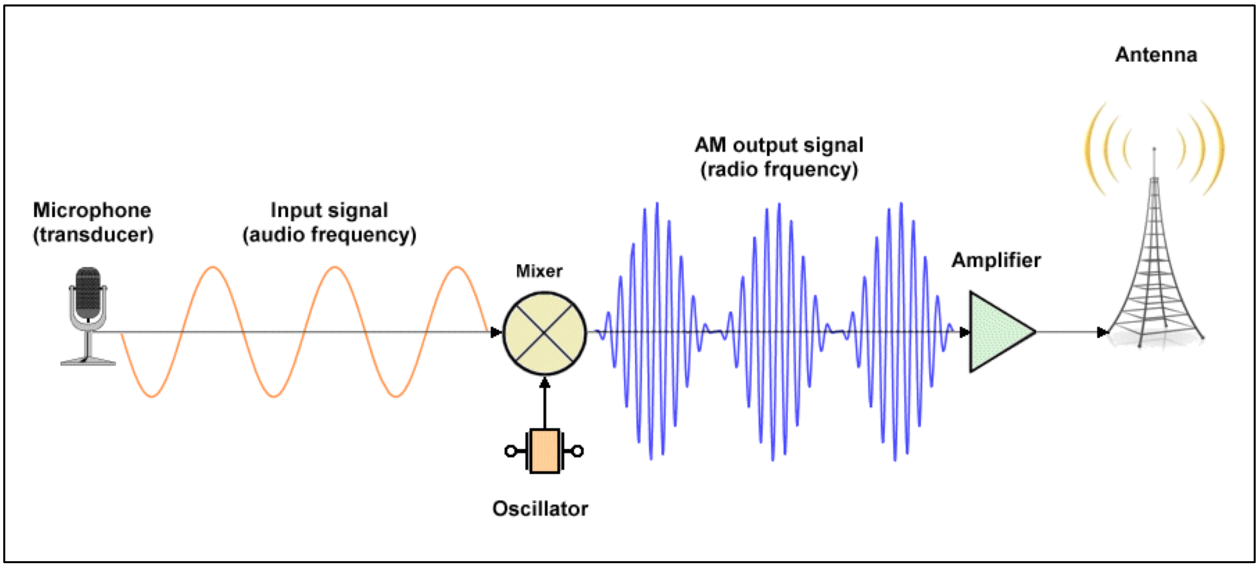


Figure 9: Block diagram of AM modulation.

4.1 TYPES OF AMPLITUDE MODULATION (AM)

AM can be further classified into various types:

1. **Double Sideband with Large Carrier (DSB-LC)**: This is the most commonly used form of AM, especially for AM radio broadcasting, where both sidebands and the full carrier signal are transmitted.
2. **Double Sideband Suppressed Carrier (DSB-SC)**: Similar to DSB-LC, but without transmitting the carrier signal, reducing power requirements.
3. **Single Sideband (SSB)**: In this type, only one of the two sidebands from the DSB-SC signal is transmitted, significantly saving bandwidth and power.

4. **Vestigial Sideband (VSB)**: A modified form of SSB that transmits a portion of the second sideband, simplifying the signal generation and reception process, commonly used in TV broadcasting.

4.1.1 DOUBLE SIDEBAND LARGE CARRIER (DSB-LC)

There are several types of amplitude modulation, with the most widely used being the Double Sideband Large Carrier (DSB-LC) scheme, also known as conventional AM. The figure below demonstrates the relationship between the carrier signal, the modulating signal, and the resulting modulated signal in conventional AM.

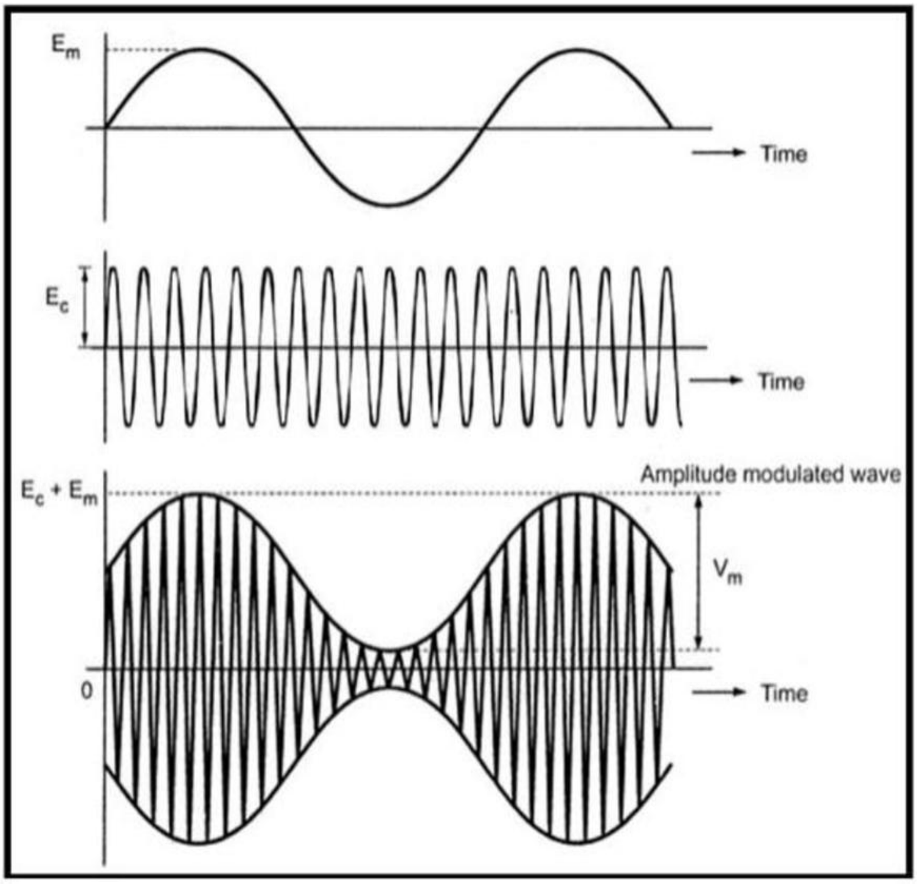


Figure 10: Waveform of AM modulation.

The expression of AM wave is given by:

$$E_{AM}(t) = [E_C + m(t)]cos\,(w_c t) \quad (2)$$

If

$$m(t) = E_m cos\,(w_m t) \tag{3}$$

Then

$$E_{AM}(t) = E_c\, cos(w_c t) + E_m\, cos(w_m t)\, cos\,(w_c t) \tag{4}$$

$$E_{AM}(t) = E_c \cos(w_c t) + \frac{E_m}{2} \cos(w_c - w_m)\, t + \frac{E_m}{2} \cos\,(w_c + w_m)t \tag{5}$$

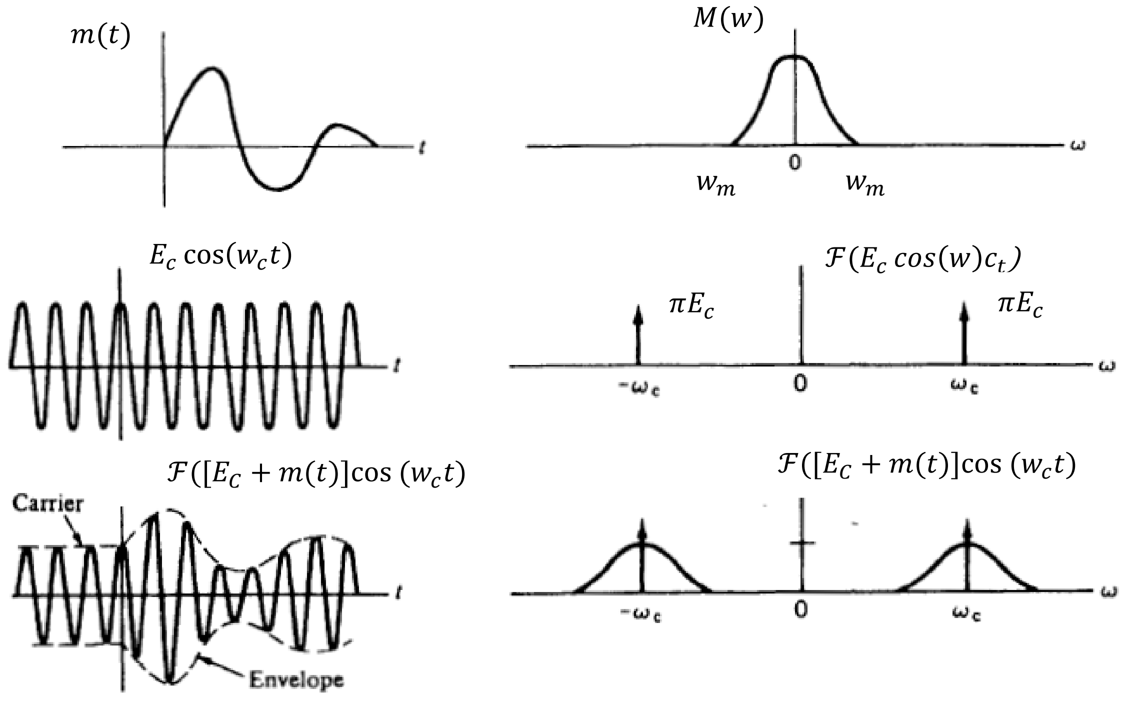


Figure 11: Waveform of DSB-LC modulation.

Where:

E_c : Carrier voltage

E_m: Modulating voltage

$m(t)$: Modulating signal

4.1.1.1 Generation of Double Sideband Large Carrier (DSB-LC):

Conceptually, the easiest way to generate a DSB-LC signal is to first generate a DSBSC signal and then add some carrier. This is shown in block diagram form in Figure 12 It turns out, however, that DSB-LC signals are generally easier to generate directly so that the system shown in Figure 12 has more analytical than practical value.

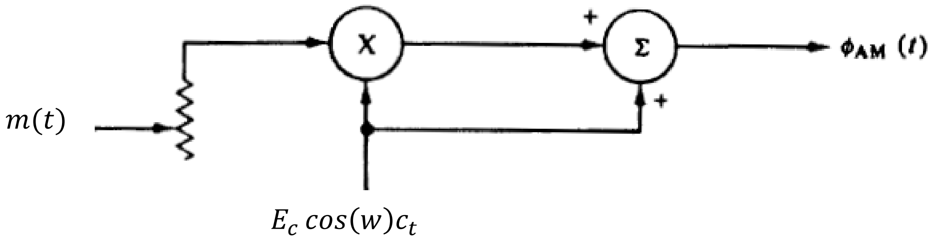


Figure 12: DSB-LC modulation.

The generation of Double Sideband Large Carrier (DSB-LC) signals can be achieved using a variety of techniques. The two main methods are **Nonlinear Modulation** and **Linear Modulation** techniques.

The first type, Nonlinear Modulation, is the most commonly used and will be the primary focus of this study. Nonlinear Modulation relies on the properties of nonlinear devices, like diodes or transistors, to produce the modulated signal. It includes methods like the Square-Law Modulator and Switching Modulator. These techniques are valued for their simplicity and cost-effectiveness, especially in low-power applications and basic communication systems.

The system shown in Figure 13 can be used to generate AM even if the diode is not operated as an ideal switch. In this case, the nonlinearities in the diode characteristic may be approximated with a power series of the form

$$i(t) = a_1 e(t) + a_2 e^2(t) + \cdots, i(t)R \ll e(t). \quad (6)$$

Retaining only the first two terms, we find that the voltage at the input to the bandpass filter (neglecting effects of any finite input impedance) is:

$$i(t)R = a_1R[f(t) + k\cos\,\omega_c t] + a_2R[f(t) + k\cos\,\omega_c t]^2 + \cdots. \qquad (7)$$

Expanding and collecting terms at the carrier frequency, we have:

$$v_o(t) = a_1Rk\cos\,\omega_c t + 2a_2Rkf(t)\cos\,\omega_c t \qquad (8)$$

Equation (8) is the desired result for a DSB-LC signal. A semiconductor diode operates as a combination of the rectifier modulator and the modulator using a nonlinear characteristic.

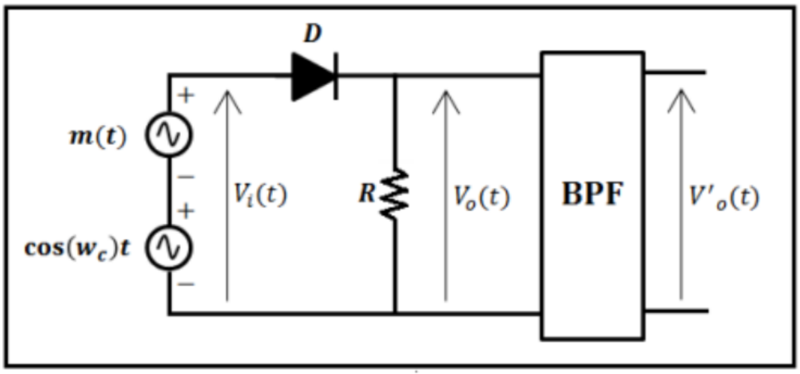


Figure 13: Nonlinear Modulator.

4.1.1.2 Demodulation (detection) of DSB-LC

In DSB-LC (AM) signals, the desired signal waveform $f(t)$ is available in the envelope of the modulated signal. The use of synchronous detection will, of course, yield the desired waveform, but it is possible to demodulate AM signals by much simpler techniques. The simplest and most popular method is one that detects the envelope of the modulated waveform directly and is called the envelope detector.

Any circuit whose output follows the envelope of the input signal waveform will serve as an envelope detector. The simplest form of an envelope detector is a nonlinear charging circuit with a fast charge time and a slow discharge time. It can easily be constructed using a diode in series with a capacitor, as shown in Figure 14.

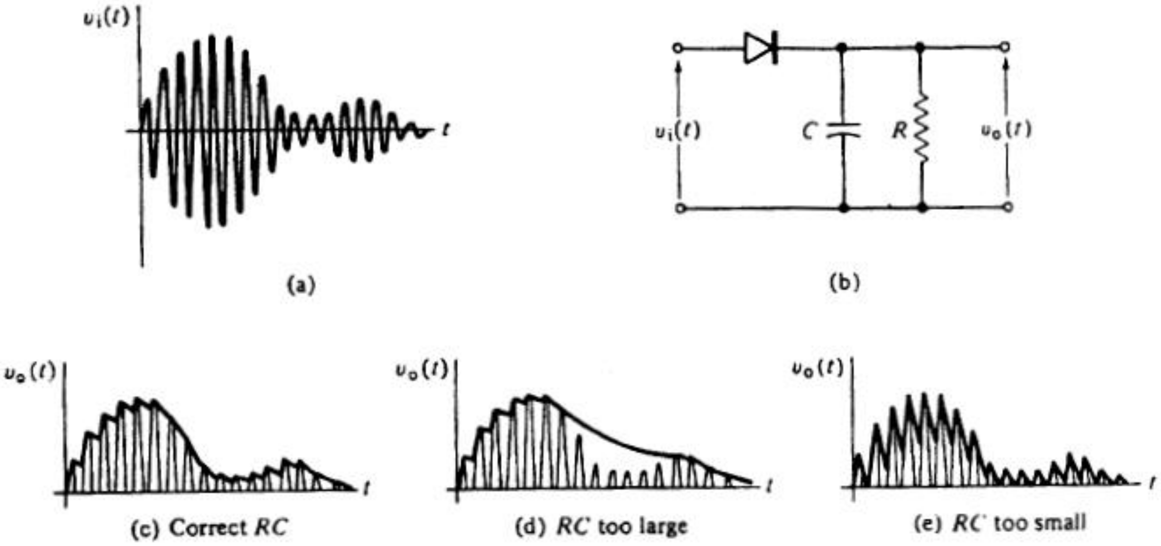


Figure 14: The envelope detector.

4.1.1.3 Modulation index

The modulation index (or modulation depth) in Amplitude Modulation (AM) quantifies the extent to which the carrier signal varies in response to the modulating (message) signal. It's an essential factor for ensuring signal fidelity and efficient transmission. The modulation index is typically denoted as m and defined as follows:

$$m = \frac{E_m}{E_c} \tag{9}$$

where:

- E_m is the amplitude of the modulating signal (message signal),
- E_c is the amplitude of the carrier signal.

The modulation index m in amplitude modulation (AM) describes the degree to which the carrier wave is modulated by the message signal, with ideal values typically in the range from (0 to 1):

- $\boldsymbol{m = 0}$: No modulation occurs. The carrier wave maintains a constant amplitude

E_c without carrying any information from the message signal.

- $\boldsymbol{0 < m < 1}$: This is the optimal range for modulation, where the carrier is modulated without distortion. As m increases toward 1, the modulation depth increases, enhancing the clarity of the transmitted signal.
- $\boldsymbol{m = 1}$: This represents the maximum modulation without distortion, where the carrier is fully modulated to the maximum amplitude without overlapping sidebands.
- $\boldsymbol{m > 1}$: Known as overmodulation, this range results in distortion due to the excessive amplitude of the message signal, causing overlapping of sidebands and making it difficult to recover the original signal without distortion.

4.1.1.4 Alternate Expression Using Maximum and Minimum Amplitudes

The modulation index can also be derived using the maximum (V_{max}) and minimum (V_{min}) envelope amplitudes of the modulated signal:

$$m = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \tag{10}$$

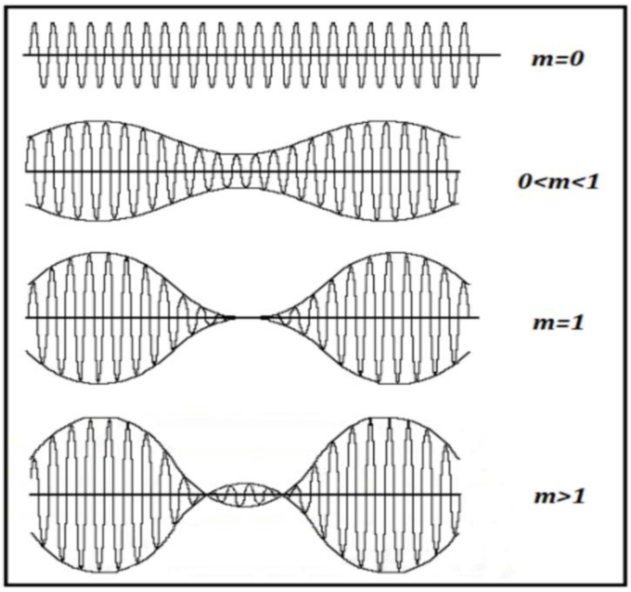


Figure 15: Modulation index of AM.

4.1.1.5 Power of DSB-LC

In AM signal waveforms, the carrier term does not contain any information about the modulating signal $m(t)$. Therefore, the power expended in this carrier is wasted for any transfer of information. It is the price one must be willing to pay in order to make cheap receivers available.

A general AM signal waveform can be described by

$$E_{AM}(t) = E_C cos\,(w_c t) + m(t) cos\,(w_c t) \tag{11}$$

For a 1 -ohm load, the total average power is given by the mean-square value of Eq. (11)

$$\overline{E_{AM}^2(t)} = E_C^2 \overline{cos^2\,(w_c t)} + \overline{m^2(t) cos^2\,(w_c t)} + 2E_C \overline{m(t) cos^2\,(w_c t)} \tag{12}$$

Where the bar indicates time averaging. We shall assume that $m(t)$varies slowly with respect to $\cos\omega_c t$. If we also assume that the average value of $m(t)$is zero (the usual case), then the last term in Eq. (12) is zero so that.

$$\begin{aligned} \overline{E_{AM}^2(t)} &= E_C^2\, \overline{cos^2(w_c t)} + \overline{m^2(t)\, cos^2(w_c t)} \\ &= \frac{E_C^2}{2} + \frac{\overline{m^2(t)}}{2} \end{aligned} \tag{13}$$

Thus, the total power P_t can be expressed as the sum of a carrier power P_c and a sideband power P_s:

$$P_t = \frac{E_C^2}{2} + \frac{\overline{m^2(t)}}{2} = P_c + P_s \tag{14}$$

That fraction of the total power contained in the sidebands, μ, is given by:

$$\mu = \frac{P_s}{P_t} = \frac{\overline{m^2(t)}}{E_C^2 + \overline{m^2(t)}} \tag{15}$$

Returning to the case in which $m(t)$ is a single sinusoid:

$$\begin{aligned}\phi_{AM}(t) &= E_C(1 + m\cos\ \omega_m t)\cos\ \omega_c t \\ &= E_C \cos\omega_c t + mE_C \cos\omega_m t \cos\omega_c t\end{aligned} \tag{16}$$

Then

$$\begin{gathered}P_t = P_c\left(1 + \frac{m^2}{2}\right) \\ P_s = \frac{m^2}{4}A^2 = P_c \cdot \frac{m^2}{2}\end{gathered} \tag{17}$$

And

$$\mu = \frac{m^2}{2 + m^2} \tag{18}$$

EXAMPLE 4 : An AM/DSB modulated signal is given by: $E_{AM}(t) = 3[1 + 0.5\,sin(12.566 * 10^3\ t)]\,sin(6.28 * 10^6 t)$ Calculate the following: a) Amplitude and frequency of each sideband. b) Carrier power, sideband power, total power, and efficiency

Solution

a)
$$e_{AM}(t) = [E_c + m(t)]\sin(w_c t)$$
$$e_{AM}(t) = [3 + 1.5\sin(12.566 * 10^3 t)]\sin(6.28 * 10^6 t)$$

From the equation, we have

$$E_c = 3\text{ V}$$
$$E_m = 1.5\text{ V}$$
$$f_m = \frac{12.566 * 10^3}{2\pi} = 2\text{KHz}$$
$$f_c = \frac{6.28 * 10^6}{2\pi} = 1000\text{KHz}$$

The amplitude of each sideband $= \frac{E_m}{2} = \frac{1.5}{2} = 0.75$ V

$$F_{LSB} = f_c - f_m = 1000 - 2 = 998\text{KHz}$$
$$F_{USB} = f_c + f_m = 1000 + 2 = 1002\text{KHz}$$

b)

$$P_c = \frac{(E_c)^2}{2R} = \frac{(3)^2}{2} = 4.5\text{ W}$$
$$P_{USB} = \frac{(E_m)^2}{8R} = \frac{(1.5)^2}{8} = 0.28125\text{ W}$$
$$P_{LSB} = \frac{(E_m)^2}{8R} = \frac{(1.5)^2}{8} = 0.28125\text{ W}$$
$$P_t = 4.5 + 0.28125 + 0.28125 = 5.0625\text{ W}$$
$$\eta = \frac{P_s}{P_t} \times 100\%$$
$$\eta = \frac{0.5625}{5.0625} \times 100\% = 11.11\%$$

4.1.2 DOUBLE SIDEBAND SUPPRESSED CARRIER (DSB-SC)

Double-sideband suppressed-carrier transmission (DSB-SC) is transmission in which frequencies produced by amplitude modulation (AM) are symmetrically spaced above and below the carrier frequency and the carrier level is reduced to the lowest practical level, ideally being completely suppressed. The expression of DSB-SC modulation is given by:

$$E_{DSB-SC}(t) = m(t)\cos(w_c t) \tag{19}$$

$$m(t) = E_m\, cos(w_m t) \tag{20}$$

$$E_{DSB-SC}(t) = E_m \cos(w_m t)\cos(w_c t) \tag{21}$$

$$E_{DSB-SC}(t) = \frac{E_m}{2}\cos(w_c - w_m)\,t + \frac{E_m}{2}\cos(w_c + w_m)t \tag{22}$$

4.1.2.1 Generation of Double Sideband Suppressed Carrier (DSB-SC):

The process of generating a DSB-SC is shown in Figure 16.

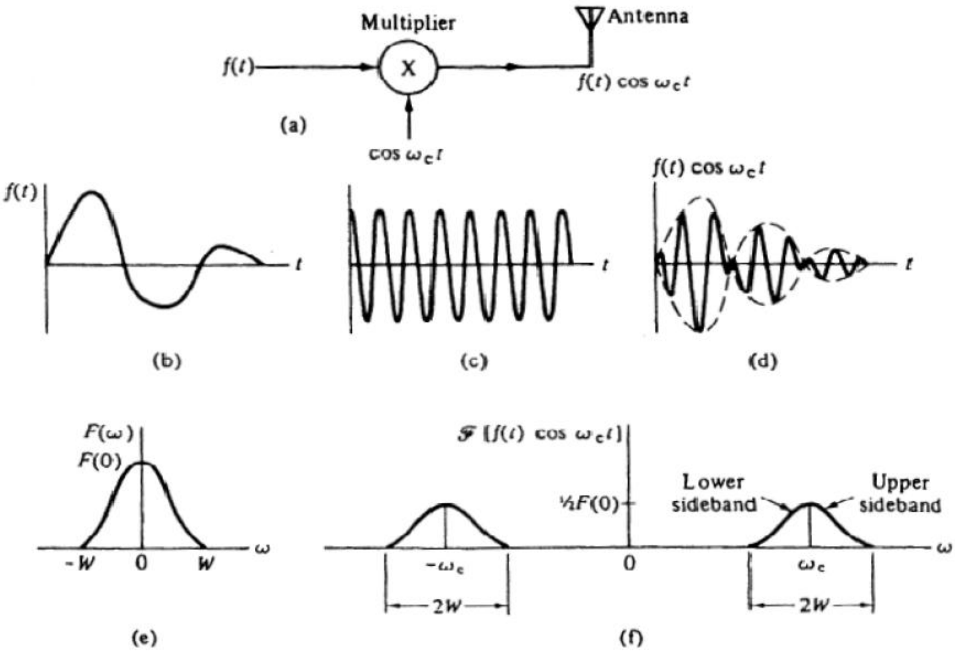


Figure 16: The Generation of DSB-SC.

Balanced modulator consists of two identical AM modulators. These two modulators are arranged in a balanced configuration in order to suppress the carrier signal. Hence, it is called as Balanced modulator. The same carrier signal $c(t) = A_c \cos(2\pi f_c t)$ is applied as one of the inputs to these two AM modulators. The modulating signal $m(t)$ is applied as another input to the upper AM modulator. Whereas, the modulating signal $m(t)$ with opposite polarity, i.e., $-m(t)$ is applied as another input to the lower AM modulator.

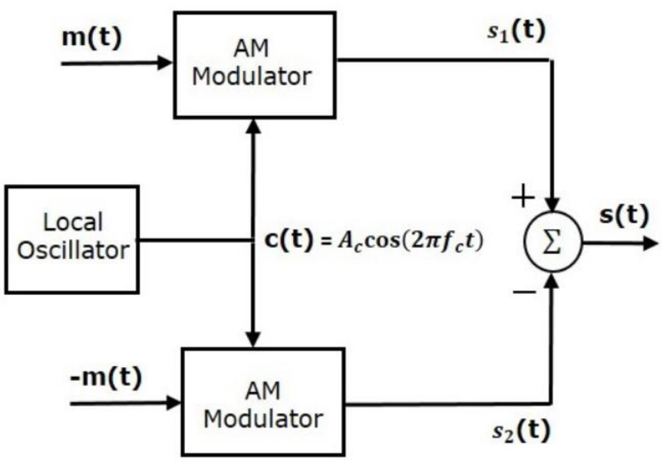


Figure 17: Balanced Modulator.

Output of the upper AM modulator is:

$$s_1(t) = A_c[1 + k_a m(t)]cos\,(2\pi f_c t) \tag{23}$$

Output of the lower AM modulator is:

$$s_2(t) = A_c[1 - k_a m(t)]cos\,(2\pi f_c t) \tag{24}$$

We get the DSBSC wave $s(t)$ by subtracting $s_2(t)$ from $s_1(t)$. The summer block is used to perform this operation. $s_1(t)$ with positive sign and $s_2(t)$ with negative sign are applied as inputs to summer block. Thus, the summer block produces an output $s(t)$ which is the difference of $s_1(t)$ and $s_2(t)$.

$$s(t) = A_c[1 + k_a m(t)]cos\,(2\pi f_c t) - A_c[1 - k_a m(t)]cos\,(2\pi f_c t) \tag{25}$$

$$s(t) = A_c m(t) cos\,(2\pi f_c t) \tag{26}$$

The double-sideband suppressed-carrier (DSB-SC) form of amplitude modulation is also generated using a ring modulator circuit. This circuit consists of two center-tapped transformers and four diodes arranged in a ring configuration to cancel out the carrier frequency, producing a double-sideband signal without the carrier.

Afterward, the resulting signal is passed through a band-pass filter to isolate the desired frequency range and enhance the quality of the modulated signal.

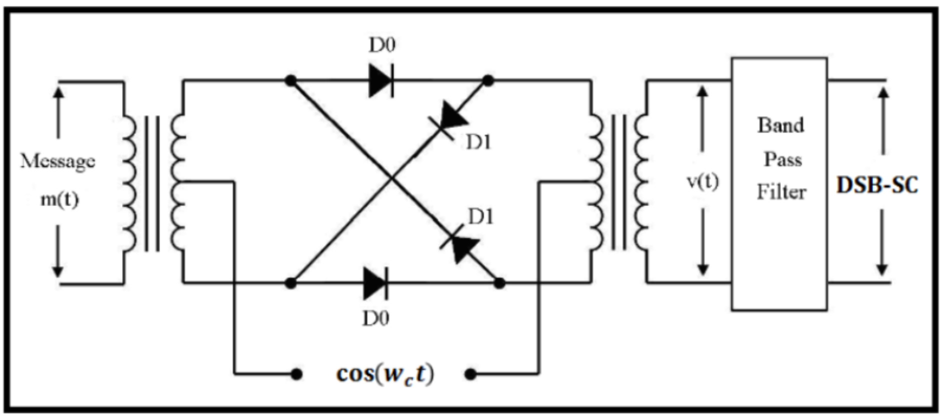


Figure 18: The ring modulator circuit.

4.1.2.2 Demodulation of DSB-SC:

In order to demodulate a double-sideband suppressed-carrier (DSB-SC) signal, the signal is first fed into a product modulator. In this modulator, the incoming DSB-SC wave is multiplied by a locally generated carrier signal, represented as $cos(w_c t)$. This locally generated carrier must be "coherent," meaning it is perfectly synchronized in both frequency and phase with the original carrier wave used during the generation of the DSB-SC signal.

This technique is known as ***coherent detection*** or ***synchronous detection*** because of the precise alignment needed between the locally generated carrier and the original carrier.

After modulation, the output from the product modulator contains both the message signal and various other frequency components. To isolate and retrieve only the message signal, the output is passed through a low-pass filter (LPF). The LPF removes all the unnecessary higher-frequency components, leaving only the original message signal as the final output.

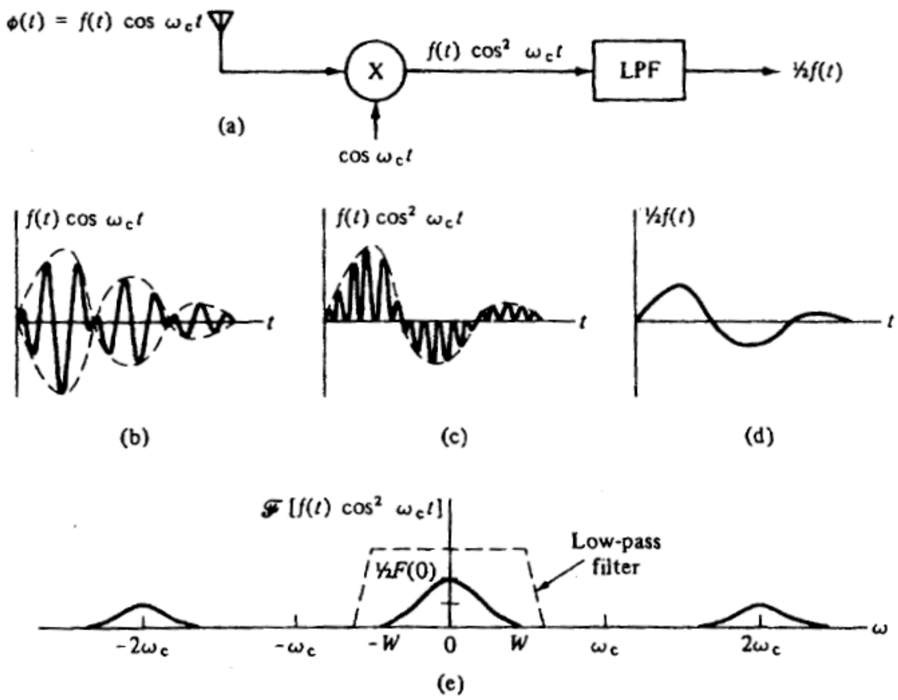


Figure 19: Demodulation of AM Suppressed-carrier.

4.1.3 SINGLE SIDEBAND (SSB)

In both double-sideband large carrier (DSB-LC) and double-sideband suppressed carrier (DSB-SC) modulation methods, the required transmission bandwidth is twice that of the original message signal $m(t)$. This is because both sidebands — the upper sideband (USB) and the lower sideband (LSB) — contain identical information about the message. However, since either sideband alone fully represents the message content, we can save bandwidth by transmitting only one of the sidebands, either the USB or the LSB. This approach is known as *Single Sideband (SSB) modulation*, and it is a more bandwidth-efficient method compared to DSB-LC and DSB-SC.

4.1.3.1 Generation of Single Sideband (SSB):

There are two common methods to generate a Single Sideband (SSB) signal.

1. **Filtering Method:** One common technique for generating a Single Sideband (SSB) signal is by using a band-pass filter (BPF) to remove one of the sidebands. This filtering process eliminates either the upper sideband (USB), which contains the higher frequency components, or the lower sideband (LSB), which contains the lower frequency components, leaving only one sideband to be transmitted. In SSB transmission, the carrier signal is often reduced or completely suppressed to further save bandwidth and power. This is known as *Single Sideband Suppressed Carrier (SSBSC)* modulation. By suppressing the carrier and transmitting only one sideband, SSBSC efficiently transmits the essential message information using minimal bandwidth.

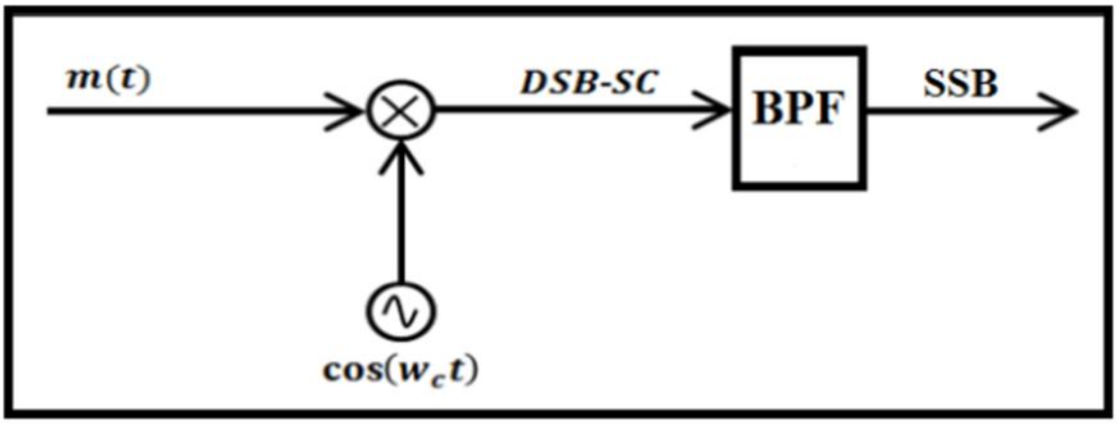


Figure 20: Generation of SSB signal using Filter method.

2. **Phasing Method:** An alternative method for generating a Single Sideband (SSB) signal is called the *Phasing Method*, commonly implemented using a Hartley modulator. This approach uses phase shifting to cancel out the unwanted sideband. In the phasing method, two versions of the original message signal are created, each shifted by 90° with respect to the other, for every frequency within the desired bandwidth. These two versions are then used to modulate two carrier signals, which are themselves 90° out of phase. By carefully adding or subtracting the resulting modulated signals, either the lower sideband (LSB) or the upper sideband (USB) can be isolated, depending on the configuration.

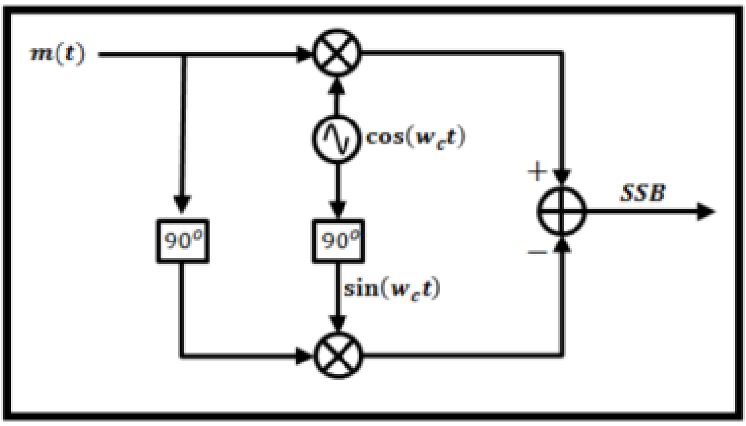


Figure 21: Generation of SSB signal using phasing method.

4.1.3.2 Demodulation of SSB:

Demodulation of SSB signals can be accomplished by using a synchronous detector as used in the demodulation of DSB-SC.

4.1.4 VESTIGIAL SIDEBAND (VSB)

Vestigial Sideband (VSB) modulation is a technique that combines one complete sideband with a portion of the other sideband, known as the "vestigial" sideband. In contrast, *Double Sideband (DSB)* modulation transmits both full sidebands, while *Single Sideband (SSB)* modulation transmits only one full sideband. VSB, however, transmits one full sideband along with a partial "vestige" of the opposite sideband.

The bandwidth required for VSB falls between the requirements of DSB and SSB. This makes VSB especially useful in cases where the input signal has a very wide frequency range, and it is essential to save spectrum space. The total bandwidth of a VSB signal ranges from 1 to 2 times the bandwidth of the original message signal, depending on the size of the transmitted vestigial sideband.

EXAMPLE 5 : For a particular DSB-SC modulator with a carrier frequency of 1000 rad/sec, a load resistance of 50 Ω, and a modulating signal $m(t) = 10\,cos(20t)$. Find the following:

1. The peak voltage (Em).
2. The modulating frequency.
3. The DSB-SC bandwidth.
4. The USB bandwidth and frequencies.
5. The LSB bandwidth and frequencies.
6. The total power.

Solution

$$E_m = 10 \text{ V}$$
$$f_m = \frac{w_m}{2\pi} = \frac{20}{2\pi} = 3 \text{ Hz}$$

The DSB-SC bandwidth is $2f_m = 6$ Hz.
The USB bandwidth = 3 Hz.
The USB start at $f_c = 159$ Hz and extends to $(f_c + f_m) = 162$ Hz.
The LSB bandwidth = 3 Hz.
The LSB start at $(f_c - f_m) = 156$ Hz and extends to $f_c = 159$ Hz.

$$P_t = \frac{(E_m)^2}{4R} = \frac{100}{200} = 0.5 \text{ W}.$$

EXAMPLE 6 : A signal $m(t)$ is band-limited to a frequency range of $0 - 10\ KHz$. This frequency is translated by multiplying it by the signal $c(t) = cos(2\pi fct)$. Find fc so that the bandwidth of the translated signal is 1% of the frequency fc.

Solution

Message Signal $m(t)$: Band-limited to a frequency range of $0 - 10\ kHz$.

Therefore, the bandwidth of $m(t) = 10\ kHz$.

The Bandwidth of the translated signal = $2f_m$

$BW = 2 \times 10k = 20kHz$

Thus,

$20k = 0.01 \times f_c$

$$f_c = \frac{20k}{0.01} = 2MHz$$

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