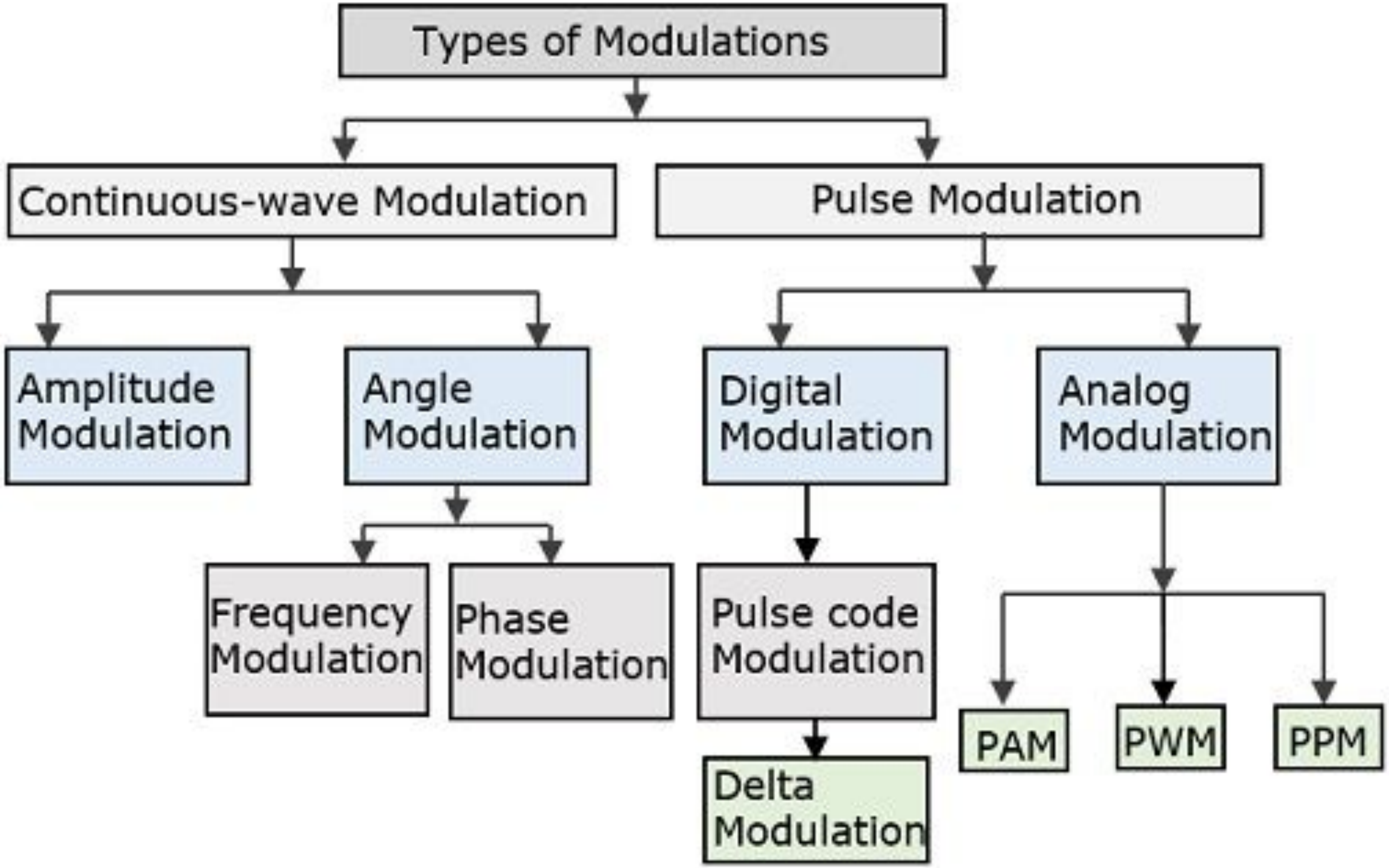
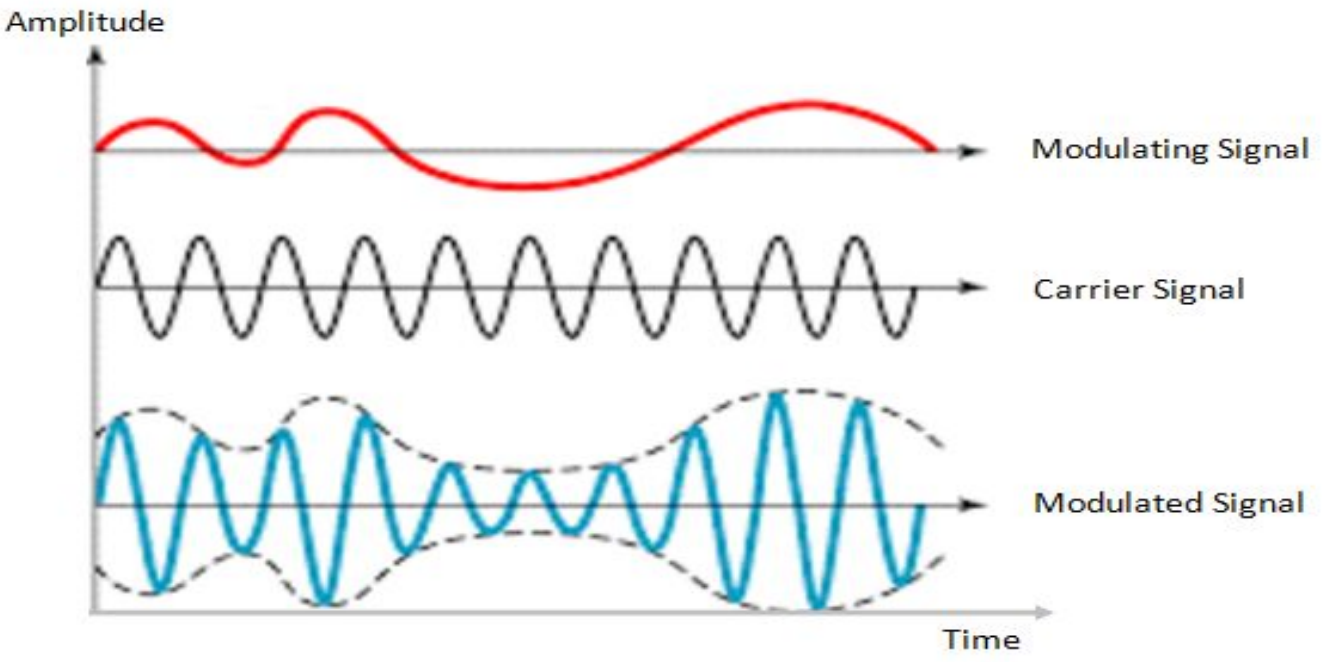
Chapter 4 Amplitude Modulation and Demodulation


Types of Modulations
Continuous-wave Modulation
Pulse Modulation
Amplitude Modulation
Angle Modulation
Digital Modulation
Analog Modulation
Frequency Modulation
Phase Modulation
Pulse code Modulation
Delta Modulation
PAM
PWM
PPM

Modulation

- The process of changing the characteristics (phase, frequency or amplitude) of high frequency carrier signal in accordance with the instantaneous value of message signal is modulation.
- When carrier signal is continuous type then such type of modulation is continuous wave modulation.



Continuous wave modulation

1) **Linear modulation (Amplitude modulation)**
2) **Non-linear modulation or Angle modulation**
    1. **Phase modulation(PM) and**
    2. **Frequency modulation(FM)**

Amplitude modulation:

The amplitude modulation is defined as the high frequency carrier signal is changed in accordance with the instantaneous value of message signal keeping phase and frequency constant.

Let us consider a carrier wave

$$C(t) = A_c \cos(2\pi f_c t + \phi_c)$$

$$C(t) = A_c \cos(2\pi f_c t)$$

Here $\phi = 0$ for phase of carrier wave=0

Let m(t) be the message signal or modulating signal

$$m(t) = A_m \cos(2\pi f_m t)$$

The standard time domain expression for **Amplitude modulated** signal is:

$$S(t) = A_c\, m(t) \cos(2\pi f_c t) + A_c \cos(2\pi f_c t)$$

$$= A_c [1 + K_a\, m(t)] \cos(2\pi f_c t)$$

Where K_a = amplitude sensitivity of AM modulation

A_c = carrier amplitude

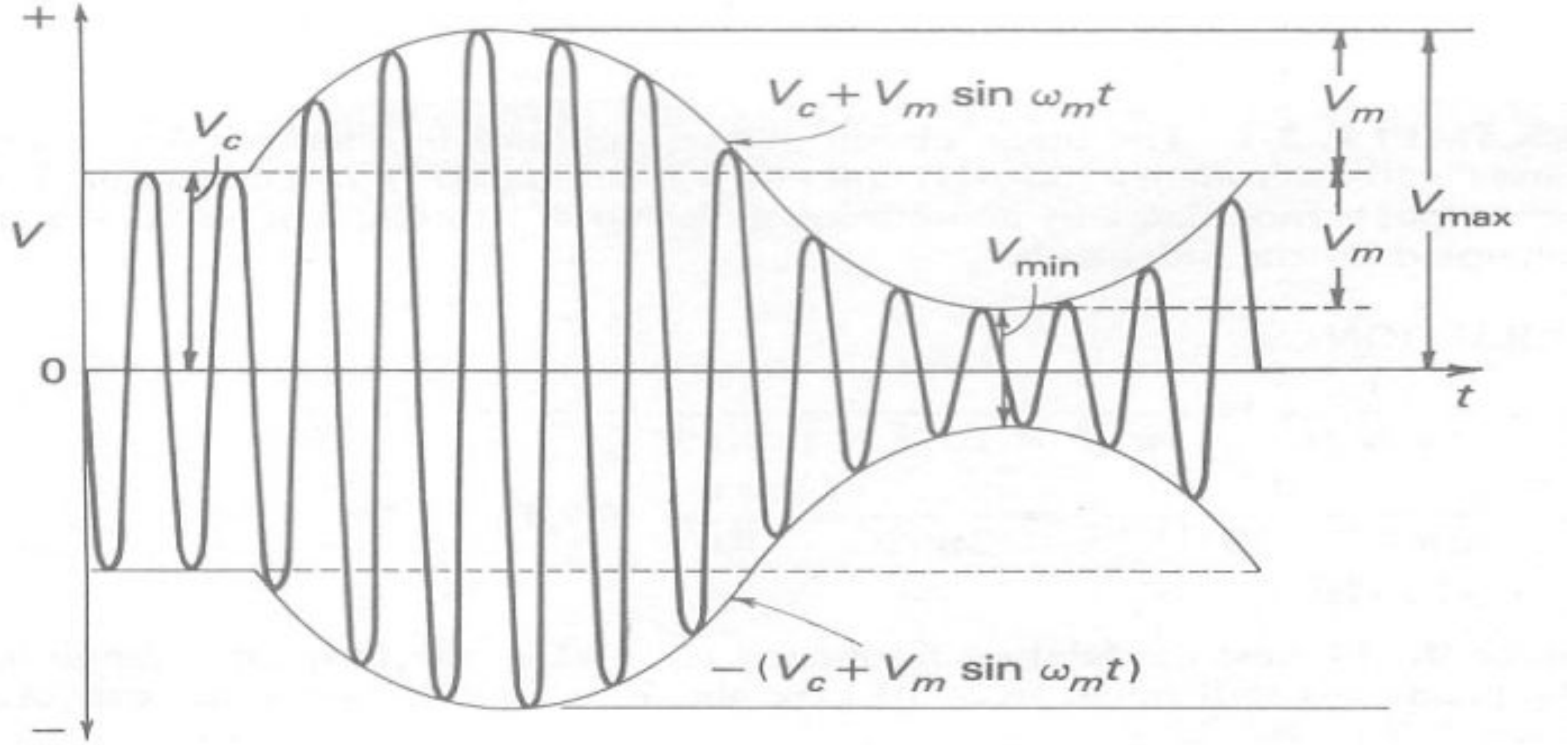
Modulation index or Amplitude Sensitivity:

- It is the measure of extent of amplitude variation about an unmodulated maximum carrier.
- It is the ratio of amplitude of message signal to the amplitude of carrier signal.
- It is given as:

'm' or 'a' or 'μ'= $\frac{\text{Maximum amplitude of message}}{\text{Maximum Carrier amplitude}}$

$$\mu = A_m / A_c \quad \text{or} \quad \frac{|m(t)|_{max}}{A_c}$$

- Generally expressed in % i.e. $a = (A_m / A_c) * 100\ \%$



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

Analysis of AM equation

In AM, amplitude is varied,

So, $A = A_c + m(t)$

$= A_c + A_m \text{Cos}(2\pi f_m t)$

$= A_c(1 + (A_m/A_c)\text{Cos}(2\pi f_m t))$

Therefore modulated signal is

$S(t) = A \text{ Cos } 2\pi f_c t$

$= A_c(1 + (A_m/A_c)\text{Cos}2\pi f_m t) \text{ Cos}2\pi f_c t$

$= A_c(1 + K_a\, m(t)) \text{ Cos}2\pi f_c t$

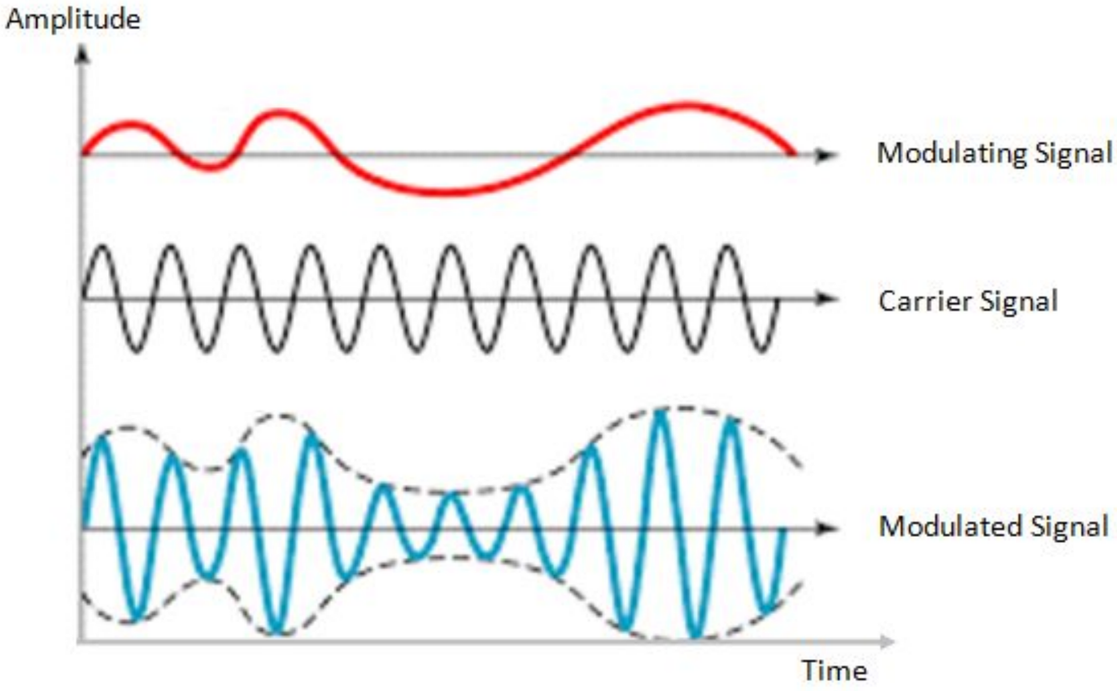
Where $k_a = 1/Ac$

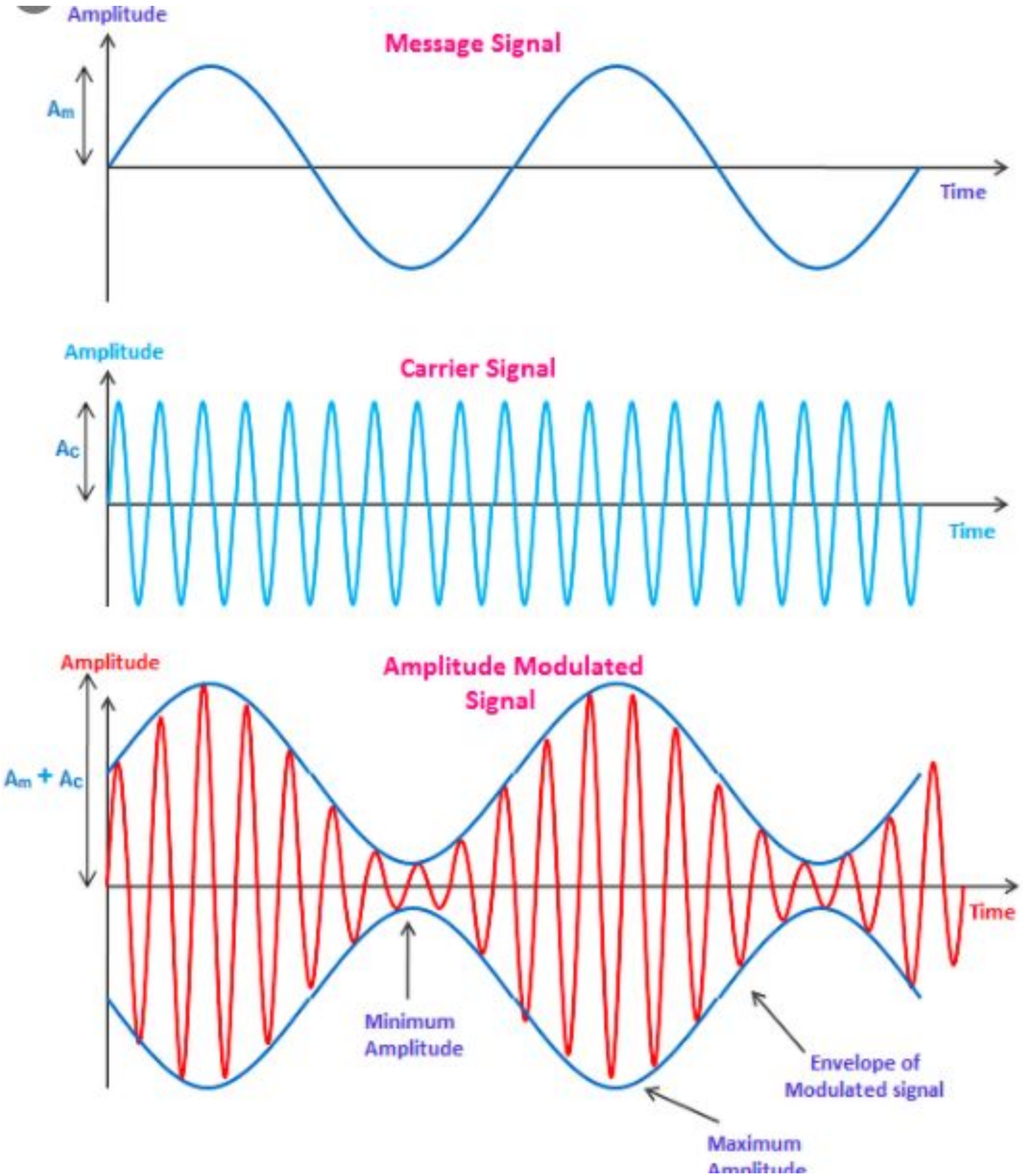
The term $A_c(1 + k_a\, m(t))$ is called **envelope** of AM wave.

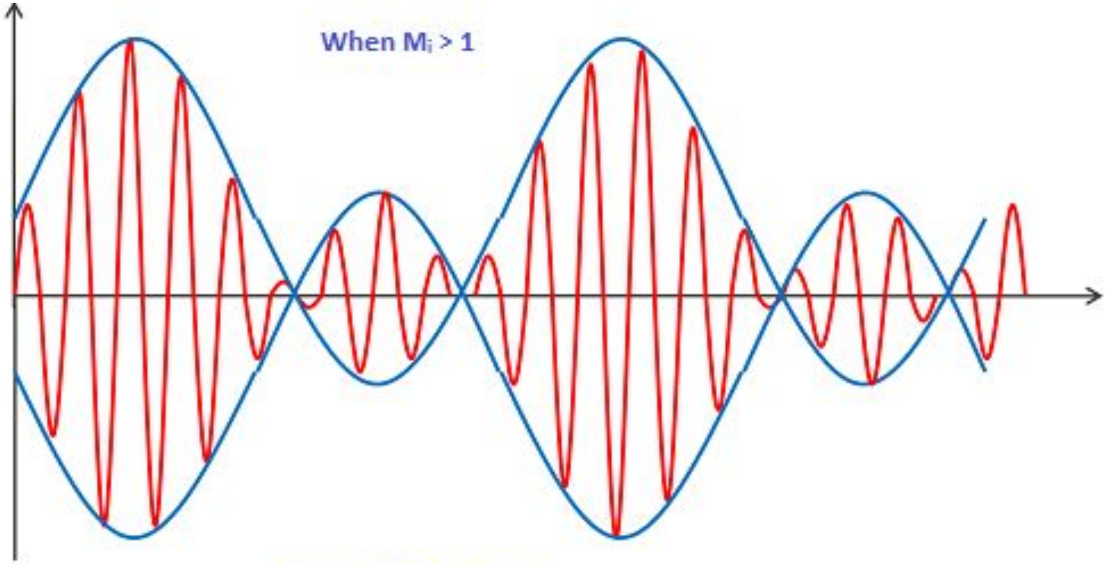
If $|k_a m(t)|$ or simply **modulation index(a) <1**, this case is **under modulation**.

If **$|k_a m(t)| > 1$**, this case is **overmodulation**.

If **$|k_a m(t)| = 1$**, this case is **100 % modulation or normal modulation**.

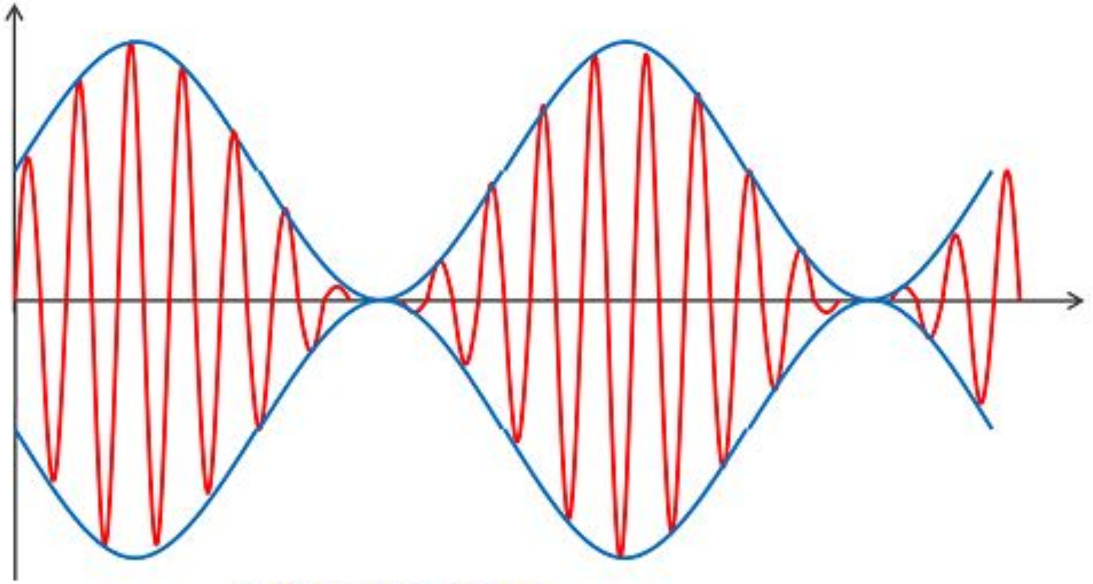






Over Modulation

When $M_i = 1$



Perfect Modulation

Frequency domain expression of standard AM

- The AM signal is written as

$s(t) = A_c\,[1 + K_a\, m(t)]\, \text{Cos}2\pi f_c t = A_c\, \text{Cos}2\pi f_c t + K_a A_c\, m(t)\, \text{Cos}2\pi f_c t$

For frequency domain analysis taking **Fourier transform** on both sides:

We know

$$s(t) \xleftrightarrow{\text{FT}} S(f)$$

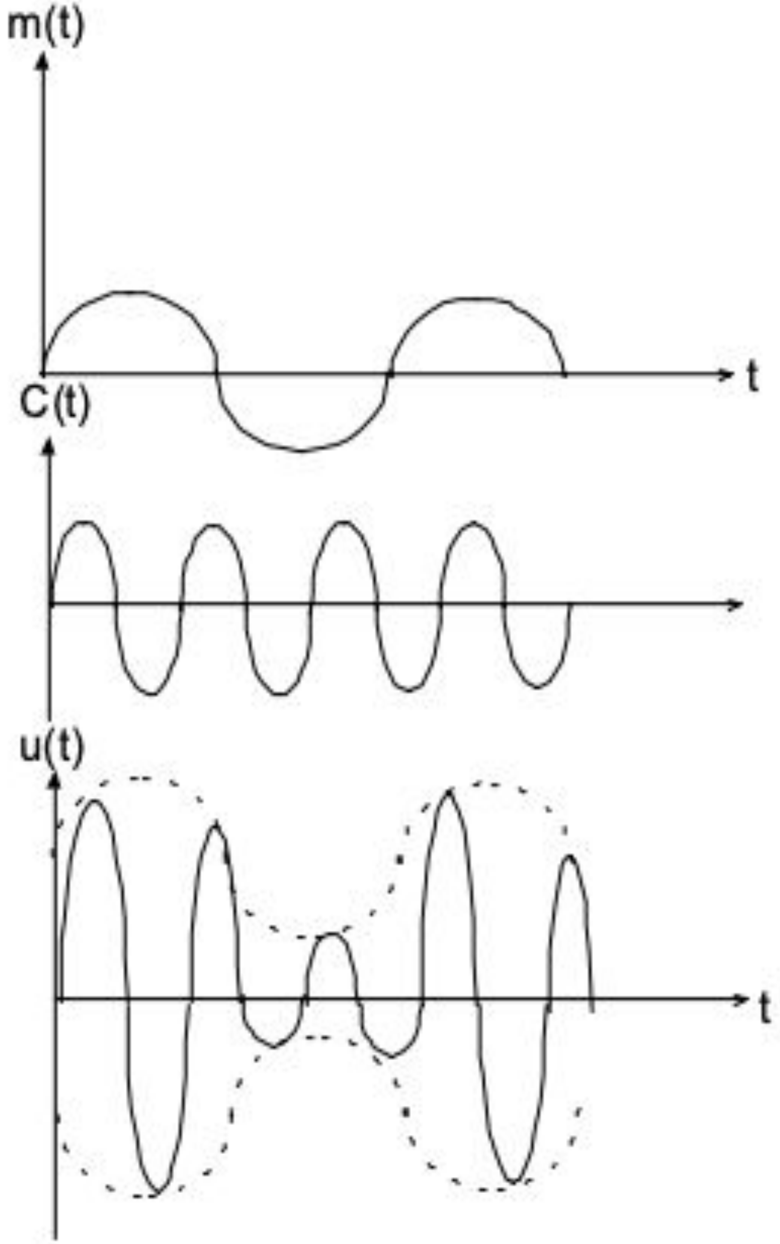
$$m(t) \xleftrightarrow{\text{FT}} M(f)$$

$$A_c \text{Cos}2\pi f_c t \xleftrightarrow{\text{FT}} A_c/2\,[\boldsymbol{\delta}(f-f_c)+\boldsymbol{\delta}(f+f_c)]$$

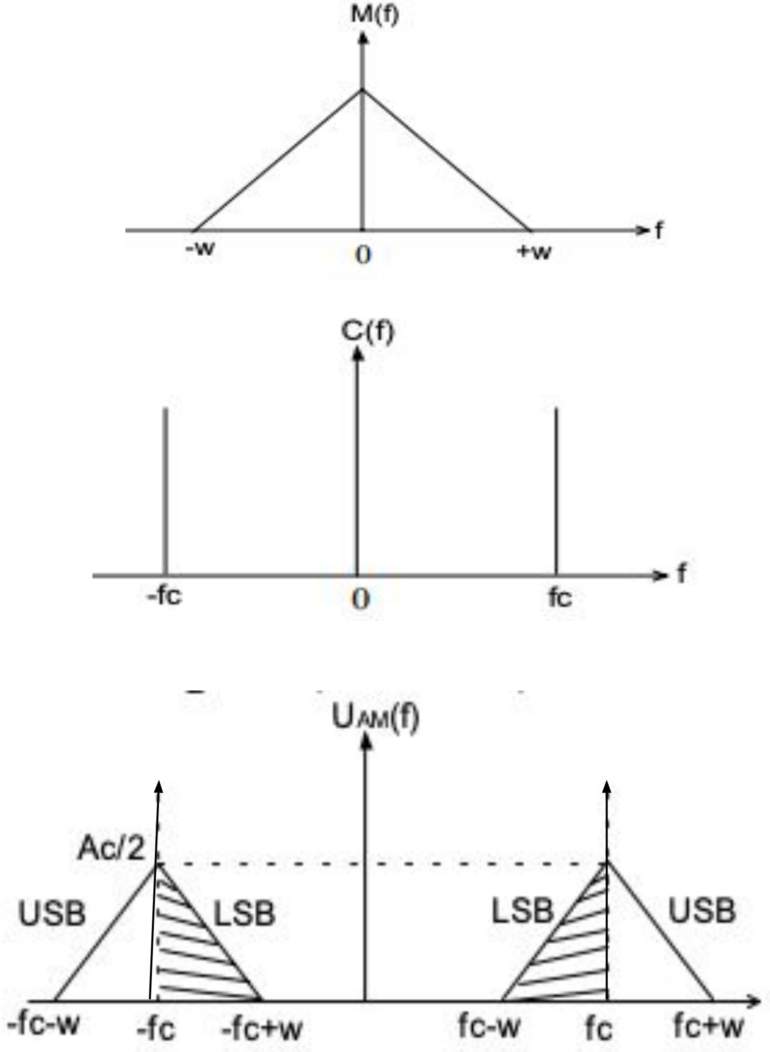
Using properties of Fourier Transform

$$K_a A_c m(t)\, \text{Cos}2\pi f_c t \xleftrightarrow{\text{FT}} \frac{K_a A_c}{2}\,[M(f-fc) + M(f+fc)]$$

$$\mathbf{S(f)} = \frac{A_c}{2}\,[\boldsymbol{\delta}(f-f_c\,) + \boldsymbol{\delta}(f+f_c)] + \frac{K_a A_c}{2}\,[\,M(f-fc) + M(f+fc)]$$



Fourier Transform



Transmission Bandwidth of AM

Bandwidth = Range of frequency

$$B = (f_c + f_m) - (f_c - f_m)$$

$$= 2f_m$$

So Bandwidth of Amplitude modulated signal is **twice** the frequency of message signal.

Single tone AM

When message signal contains **only one frequency** then this is called single tone signal and modulation is called single tone modulation.

Let us consider $\mathbf{m(t) = A_m \, Cos2\pi f_m t}$ And $\mathbf{c(t) = A_c \, Cos2\pi f_c t}$

Then, $\mathbf{s(t) = A_c \, [\, 1 + K_a \, m(t)] \, Cos2\pi f_c t}$

$$s(t) = A_c[\, 1 + K_a \, A_m \, Cos \, 2\pi \, f_m \, t \,] \, Cos2\pi f_c t$$

$$\mathbf{s(t) = A_c[\, 1 + \mu \, Cos \, 2\pi f_m t \,] \, Cos2\pi f_c t}$$

Where μ is the modulation index , $\mu = k_a A_m$

For **frequency domain** analysis

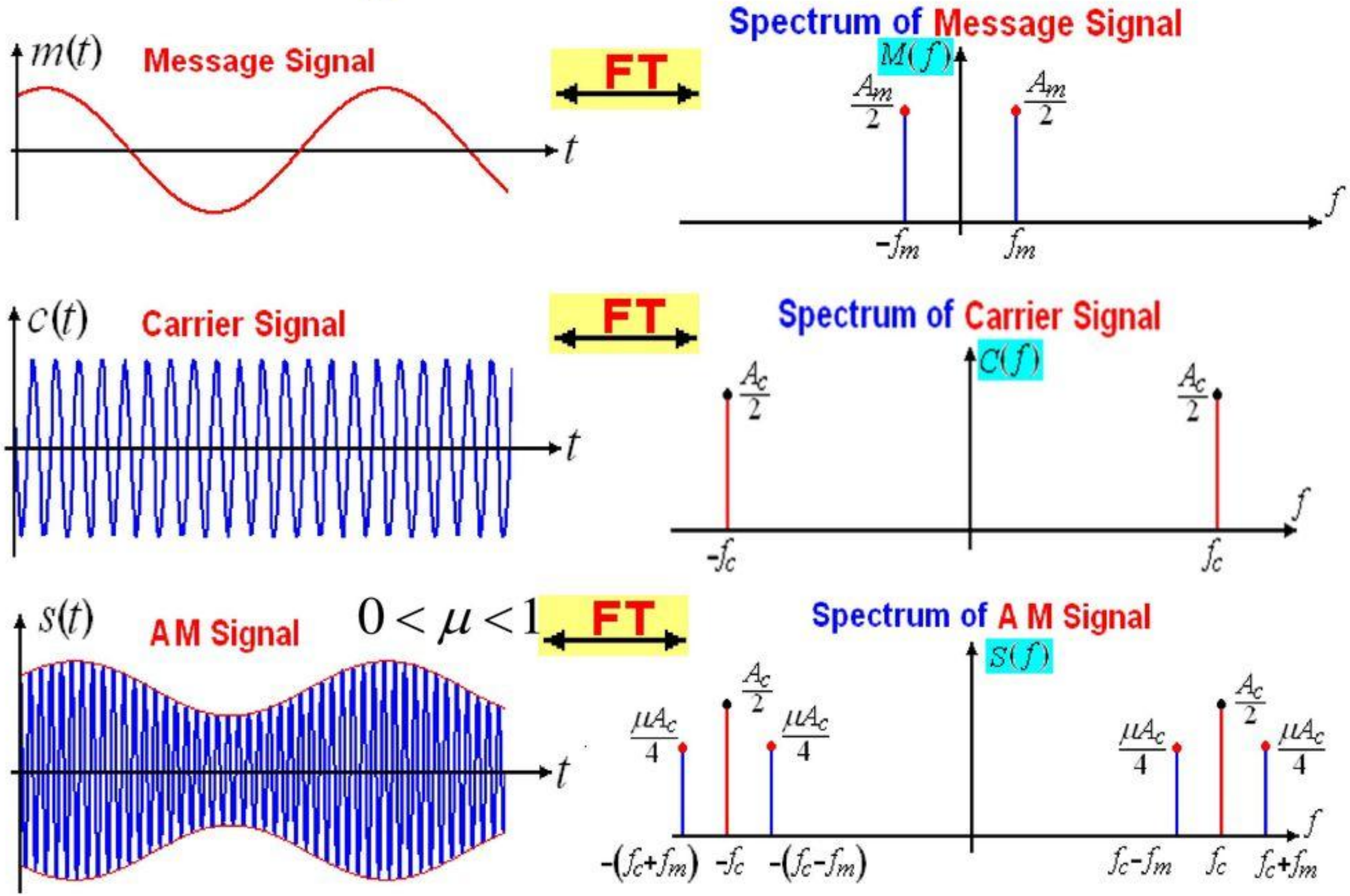
$s(t) = A_c \, Cos2\pi f_c t + A_c A_m \, Cos2\pi f_m t \, Cos2\pi f_c t$

$s(t) = A_c \, Cos2\pi f_c t + (A_c A_m/2) \, [Cos2\pi(f_c+f_m)t + Cos2\pi(f_c-f_m)t]$

Taking **Fourier transform**,

$S(f) = (A_c/2)[\delta(f-f_c) + \delta(f+f_c)] + (A_m A_c/4) \, [\delta(f-f_c-f_m) + \delta(f+f_c+f_m)] + (A_m A_c/4) \, [\delta(f-f_c+f_m) + \delta(f+f_c-f_m)]$

Single Tone Modulation



Power in DSB-FC AM

- Total power $P_t = P_c + P_{USB} + P_{LSB}$
- Carrier power = $P_c = A_c^2 / 2R$
- Upper side band power and Lower side band power is:

$$P_{USB} = P_{LSB} = (\mu A_c * 0.707/2)^2 = \mu^2 Ac^2/8$$

For R=1Ω

Therefore Total power $(P_t) = (A_c^2/2) + \mu^2 A_c^2/8 + \mu^2 A_c^2/8$

$$= (A_c^2/2)\,[1 + \mu^2/2]$$

$$= P_c\,(1 + \mu^2/2)$$

Transmission efficiency in DSB-FC AM

- Transmission efficiency (η%) = $\frac{Useful\ Power}{Total\ power} * 100\%$

$$= \frac{P_{USB} + P_{LSB}}{P_t} * 100\%$$

$$= \frac{\mu^2}{2+\mu^2} * 100\%$$

Maximum efficiency is obtained when μ=1 or 100% modulation

η=33.33%

Only one third of the total power is transmitted in AM wave

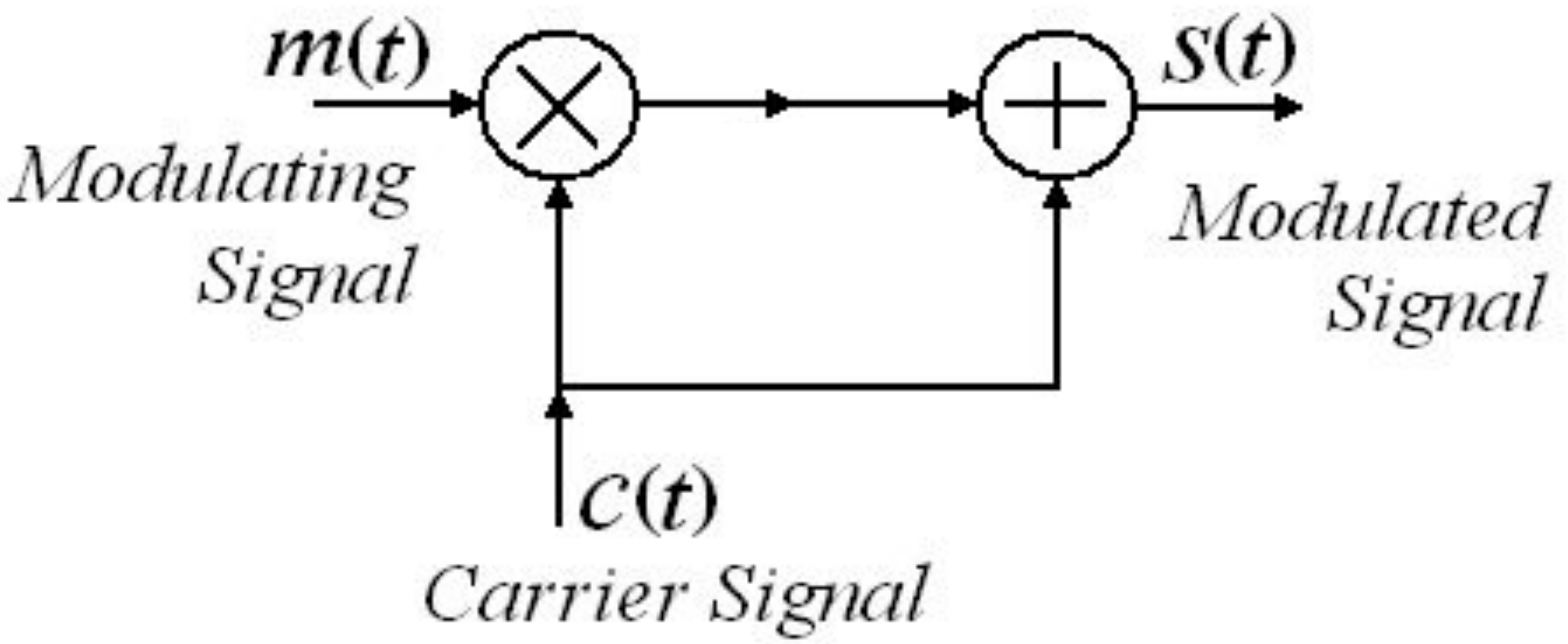
Characteristics of DSB-AM

1. Bandwidth occupied by this type of modulation is twice the BW of the message signal. Thus there is waste of the precious BW.
2. As the carrier is also transmitted there is wastage of power and therefore less efficient.
3. The modulation and demodulation processes are very simple and less expensive.
4. Therefore this type of modulation is extensively used in radio broadcasting where the cost of radio receivers are premium.

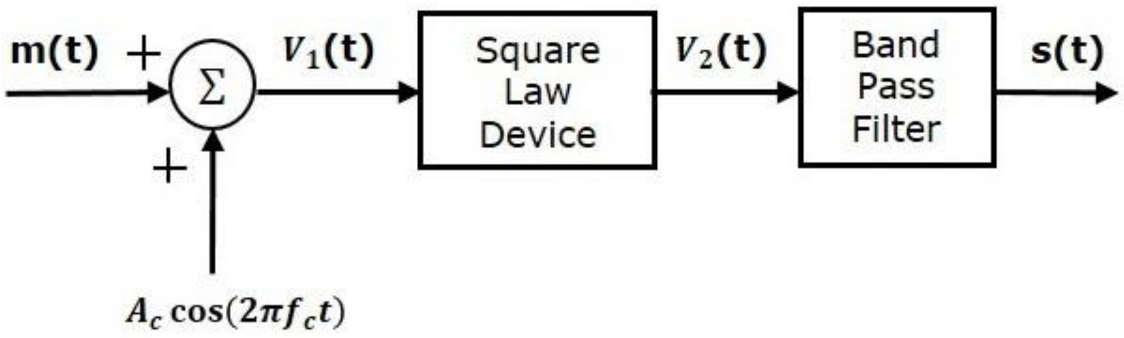
Generation of DSB-AM

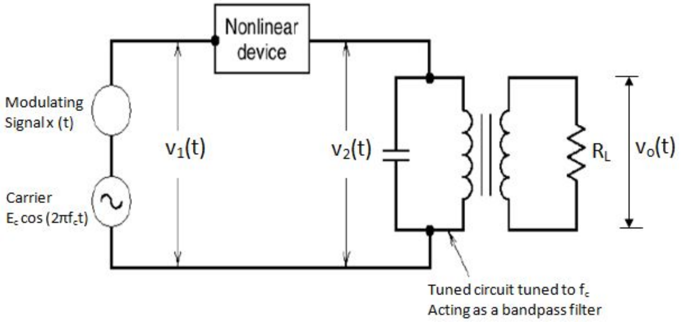
1) Direct method
2) Indirect method
   a) Non-Linear (Square law) modulator
   b) Switching (Chopper type) modulator

1. Direct method:



2. a) Non-linear or Square law:





Let the modulating and carrier signals be denoted as m(t) and $A_c \, Cos(2\pi f_c t)$

$$V_1(t)= m(t) +A_c Cos(2\pi f_c t)$$

$$V_2(t)=k1 \, V_1(t) + k2 \, V_1^2(t)$$

$$V_2(t)=k1 \, [m(t) + A_c Cos(2\pi f_c t)] + k2 \, [m(t)+ A_c Cos(2\pi f_c t)]^2$$

$$\Rightarrow V_2(t)=k1m(t) + k1 \, A_c Cos(2\pi f_c t) + k2 \, m^2(t) + k2 \, A_c^2 Cos^2(2\pi f_c t)+ 2k2m(t)A_c Cos(2\pi f_c t)$$

$$\Rightarrow V2(t)=k1m(t)+k1A_c \, Cos(2\pi f_c t)+ k2[m^2(t) + Ac^2 cos^2(2\pi f_c t)+ 2 \, m(t) \, Ac \, cos(2\pi f_c t)]$$

$$\Rightarrow V2(t)= k1m(t)+ k2 \, m^2(t) + k2 \, A_c^2 Cos^2(2\pi f_c t) + \mathbf{k1A_c[1+(2k2/k1)m(t)]cos(2\pi f_c t)}$$

- The last term of the equation represents the desired AM wave and the first three terms of the above equation are unwanted.
- So, with the help of band pass filter, we can pass only AM wave and eliminate the first three terms.
- Therefore, the output of square law modulator is:

$$\mathbf{s(t)=k1\ A_c[1+(2k2/k1)\ m(t)]\ Cos(2\pi f_c t)}$$

The standard equation of AM wave is:

$$s(t)=A_c[1+k_a m(t)]\ Cos(2\pi f_c t)$$

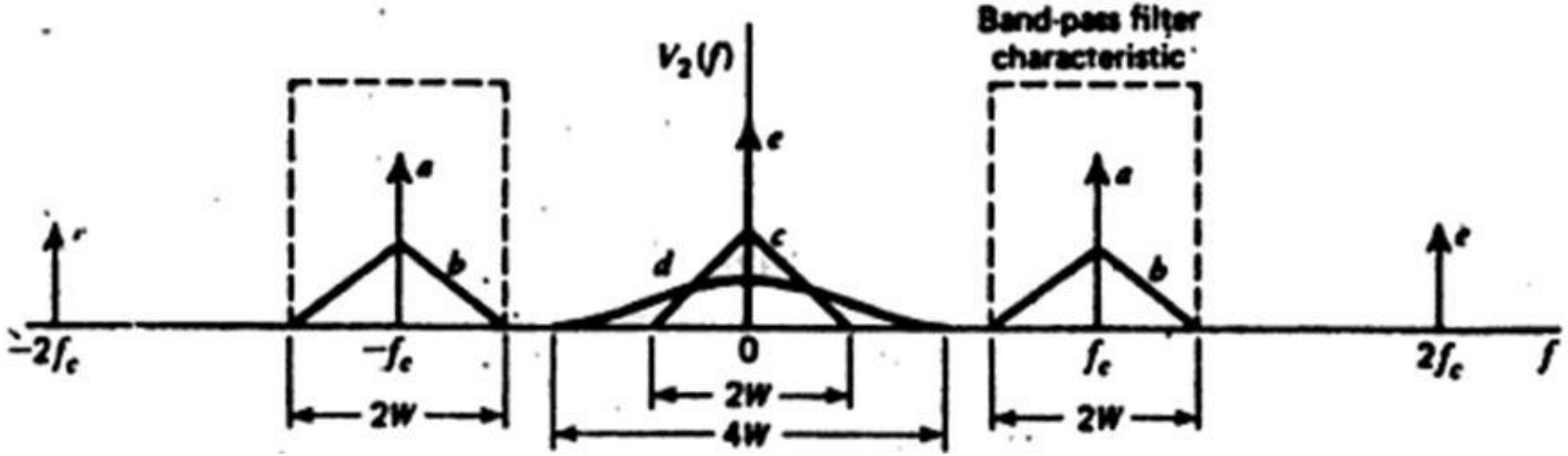
Where, Ka is the amplitude sensitivity

By comparing the output of the square law modulator with the standard equation of AM wave, we will get the scaling factor as k1 and the amplitude sensitivity k_a as 2k2/k1.

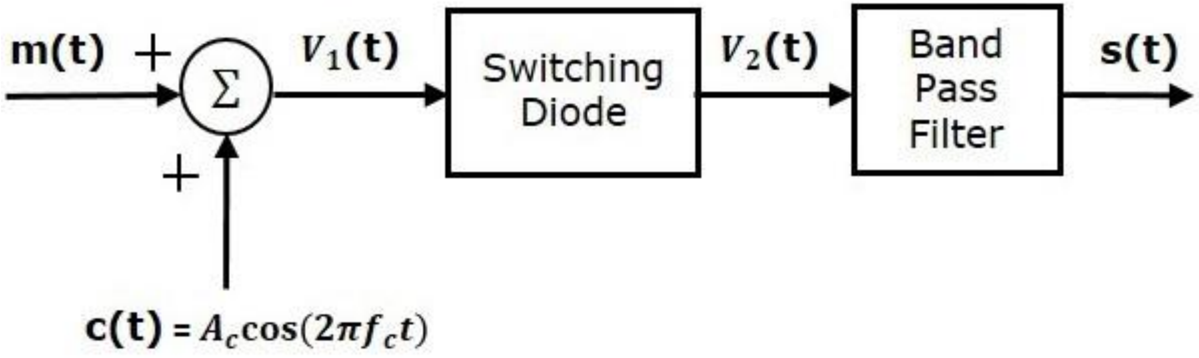
Spectrum of Square Law

$\Rightarrow$V2(t)= k1m(t)+ k2 $m^2(t)$ + k2 $A_c^{\ 2}Cos^2(2\pi f_c t)$ +

k1A_c[1+(2k2/k1)m(t)]cos(2πf_ct)



2. b) Switching or Chopper type modulator



- The diode has to operate as a switch.
- The message signal m(t) alone can not forward bias the diode i.e. turn it ON, because amplitude A_m is very low in comparison with carrier amplitude A_c .
- $$V2(t) = \begin{cases} V1(t) = c(t) + m(t) & \text{as} \quad c(t) > 0 \\ 0 & \text{as} \quad c(t) < 0 \end{cases}$$

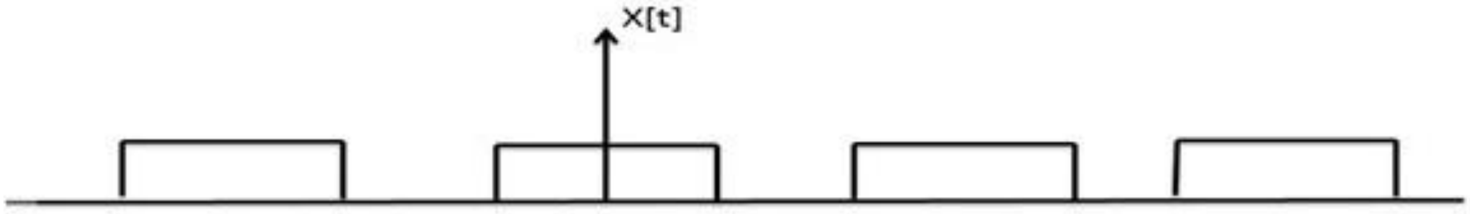
Let the modulating and carrier signals be denoted as m(t) and c(t)=Ac Cos(2πfct) respectively such that mathematically:

$$\mathbf{V1(t) = m(t) + c(t) = m(t) + Ac\ Cos(2\pi fct)}$$

We can approximate this as:

$$V2(t)=V1(t)\ x(t)$$

Where, x(t) is a periodic pulse train with time period $T=1/f_c$



The Fourier series representation of this periodic pulse train x(t) is:

$$x(t)=\frac{1}{2}+\frac{2}{\pi}\sum_{n=1}^{\infty}\frac{(-1)^n-1}{2n-1}\cos(2\pi(2n-1)f_ct)$$

$$\Rightarrow x(t)=\frac{1}{2}+\frac{2}{\pi}\cos(2\pi f_ct)-\frac{2}{3\pi}\cos(6\pi f_ct)+\ldots.$$

Substituting corresponding values of V1(t) and x(t) in V2(t) we get:

$$V_2(t) = [m(t) + A_c\cos(2\pi f_c t)]\left[\tfrac{1}{2} + \tfrac{2}{\pi}\cos(2\pi f_c t) - \tfrac{2}{3\pi}\cos(6\pi f_c t) + \ldots\ldots\right]$$

$$V_2(t) = \frac{m(t)}{2} + \frac{A_c}{2}\cos(2\pi f_c t) + \frac{2m(t)}{\pi}\cos(2\pi f_c t) + \frac{2A_c}{\pi}\cos^2(2\pi f_c t) -$$

$$\frac{2m(t)}{3\pi}\cos(6\pi f_c t) - \frac{2A_c}{3\pi}\cos(2\pi f_c t)\cos(6\pi f_c t) + \ldots\ldots$$

$$V_2(t) = \frac{A_c}{2}\left(1 + \left(\frac{4}{\pi A_c}\right)m(t)\right)\cos(2\pi f_c t) + \frac{m(t)}{2} + \frac{2A_c}{\pi}\cos^2(2\pi f_c t) -$$

$$\frac{2m(t)}{3\pi}\cos(6\pi f_c t) - \frac{2A_c}{3\pi}\cos(2\pi f_c t)\cos(6\pi f_c t) + \ldots\ldots$$

- The 1st term of the above equation represents the desired AM wave and the remaining terms are unwanted terms. Thus, with the help of band pass filter, we can pass only AM wave and eliminate the remaining terms.

Therefore, the output of switching modulator is:

$$s(t) = \frac{A_c}{2}\left(1 + \left(\frac{4}{\pi A_c}\right) m(t)\right) \cos(2\pi f_c t)$$

$$s(t) = A_c\left[1 + k_a m(t)\right] \cos(2\pi f_c t)$$

Where, ka is the amplitude sensitivity.
By comparing the output of the switching modulator with the standard AM wave, we will get the scaling factor as 0.5 and amplitude sensitivity ka as 4/πAc .

Spectrum of Switching modulator:

Class Assignment

$$V_2(t) = \frac{A_c}{2}\left(1 + \left(\frac{4}{\pi A_c}\right) m(t)\right)\cos(2\pi f_c t) + \frac{m(t)}{2} + \frac{2A_c}{\pi}\cos^2(2\pi f_c t) -$$

$$\frac{2m(t)}{3\pi}\cos(6\pi f_c t) - \frac{2A_c}{3\pi}\cos(2\pi f_c t)\cos(6\pi f_c t) + \ldots\ldots$$

Demodulation of DSB-AM waves

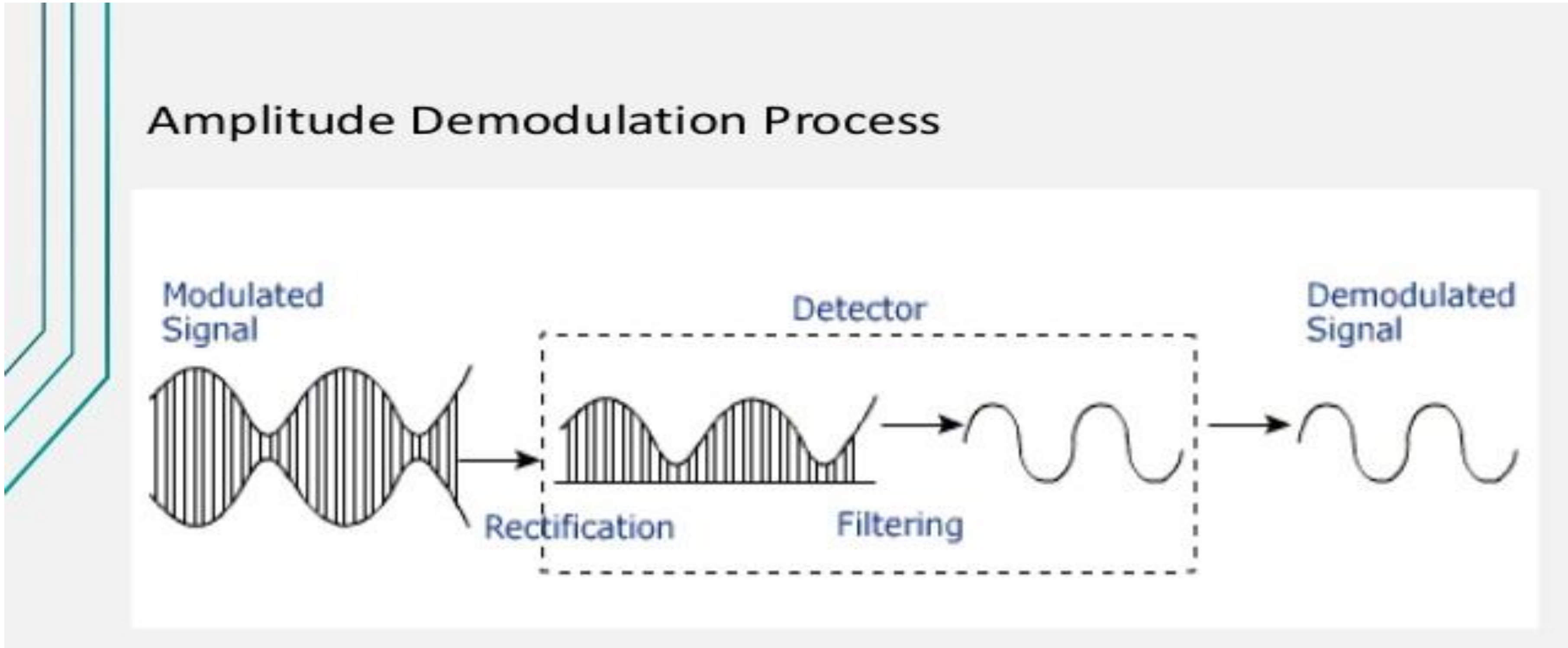
- The process of detection or demodulation provides a means of recovering the message signal from an incoming modulated wave.
- In effect, detection is the inverse process of modulation

We describe two technique for the detection of DSB-AM

1. **Square Law detector**
2. **Envelope detector**

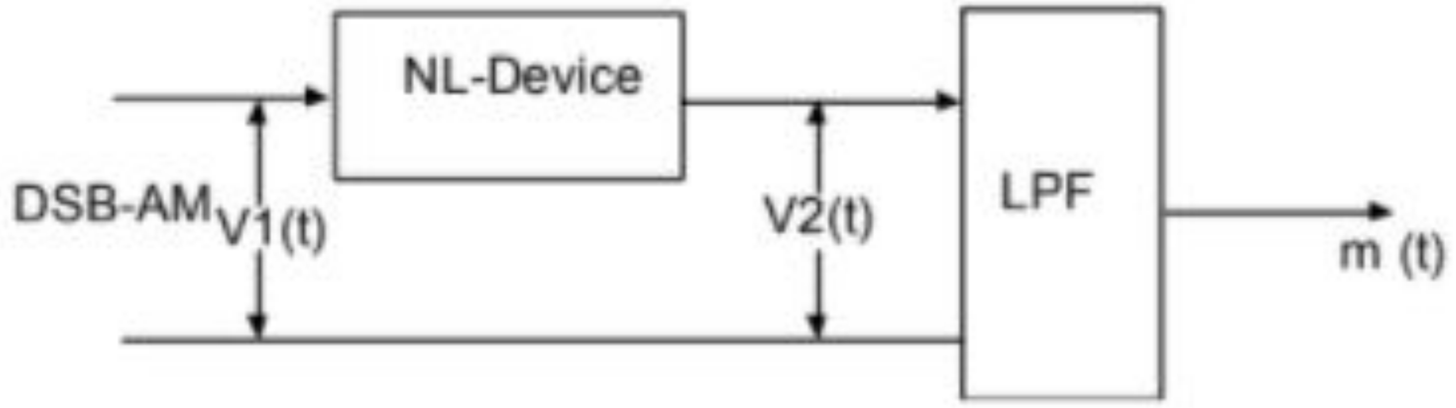
Demodulation of DSB-FC AM signal

- It is the reverse process of modulation.
- It converts the modulated signal into original form of message signal.
- It is the process of recovering the message signal.



1. Square law detector

- Demodulation of DSB FC Amplitude modulated wave
- A square law detector is obtained by using a square law modulator for the purpose of detection.
- If we supply the modulated signal to the non linear device followed by a LPF , we can recover message signal m(t).



- Consider the transfer characteristic of a non linear device.

$$\mathbf{V_2(t) = b1V_1(t) + b2V_1(t)^2 \text{ ....(i)}}$$

Where b1 and b2 are constant

The input to the NL device is the DSB-AM which can be written as

$$\mathbf{V_1(t) = A_c[1+ka.m(t)]Cos2\pi f_c t \text{ ....(ii)}}$$

From equation (i) and (ii)

$V_2(t) = b1\{A_c[1+ka.\ m(t)]Cos2\pi f_c t\} + b2\ A_c^2(1 + ka.m(t))^2\ Cos(2\pi f_c t)^2$

$= b1A_c[1+ka.m(t)]Cos2\pi f_c t + b2\ A_c^2\ ((1 + 2\ ka\ .m(t) + ka^2m^2(t)\ )[\frac{1 + Cos4\pi f_c t}{2}\ ]$

When this signal is passed to **low pass filter**,

$= \frac{1}{2}\ b2\ A_c^2\ + b2\ ka\ A_c^2m(t)\ \ + \frac{1}{2}\ b2\ .Ac^2.ka^2\ m^2(t)$

The first term is **dc term**, second term is message signal and third term is noise. All the other frequency above f_c are attenuated by low pass filter.

So recovered message signal is **b2 ka A_c^2m(t)** .

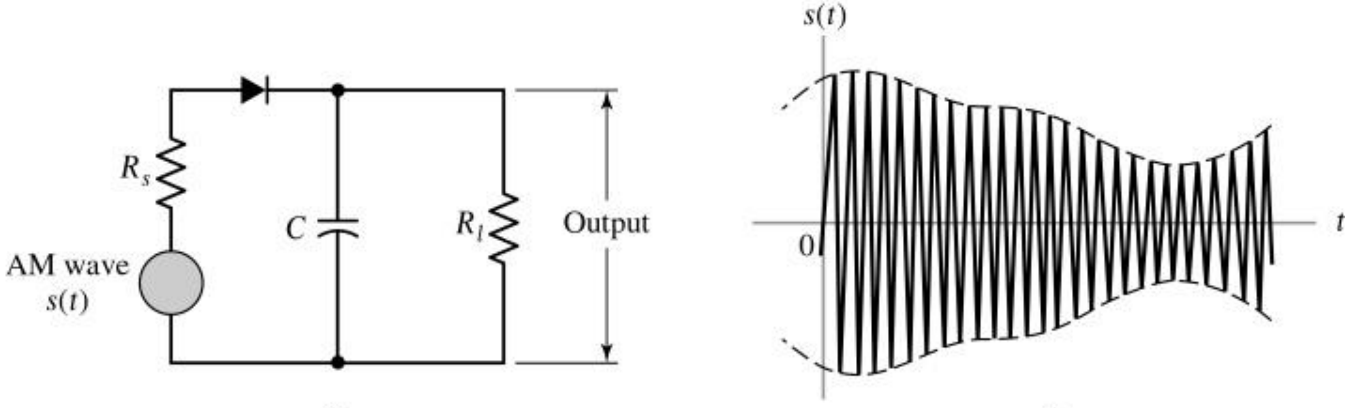
2. Envelope Detector

- The simplest and most popular method.
- Rectification and filtering of the AM wave will produce desired message signal.
- The charging time constant R_sC must be short compared with the carrier period $1/f_c$

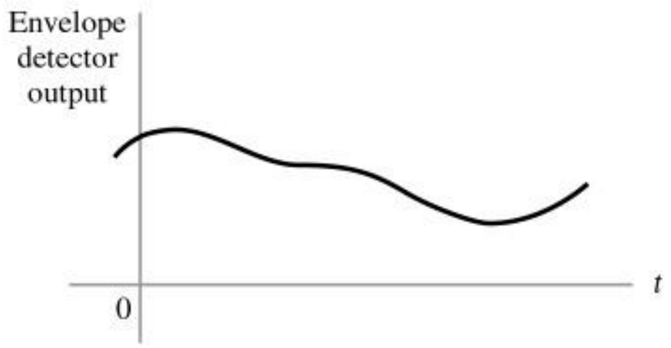
$$R_sC << \frac{1}{f_c}$$

- The discharging time constant R_LC must be long enough

$$\frac{1}{f_c} << R_LC << \frac{1}{\omega}$$



(a) (b)



Double Sideband Suppress Carrier DSB-SC

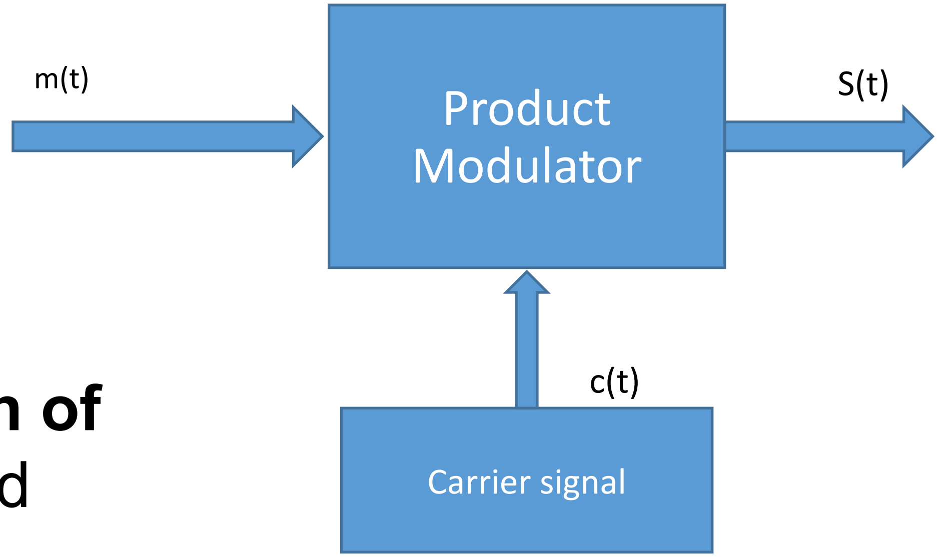
- In the process of AM, the modulated wave consists of the carrier wave and two sidebands. The modulated wave has the information only in the sidebands.
- **Sideband** is nothing but a band of frequencies, containing power, which are the lower and higher frequencies of the carrier frequency.
- The transmission of a signal, which contains a carrier along with two sidebands can be termed as **DSBFC**.
- However, such a transmission is inefficient. Because, two-thirds of the power is being wasted in the carrier, which carries no information.
- If this carrier is suppressed and the saved power is distributed to the two sidebands, then such a process is called as **DSBSC**.

Time Domain Description of DSBSC

Let us consider the same mathematical expressions for modulating and carrier signals as we have considered in the earlier chapters.

i.e. Modulating signal = $m(t)$ and
Carrier signal $c(t)=A_c\cos(2\pi f_c t)$

Mathematically, we can represent the **equation of DSBSC wave** as the product of modulating and carrier signals.



$s(t)=m(t)c(t)$
$\Rightarrow s(t)=A_c\cos(2\pi f_m t)\, m(t)$

- $S(t) = Ac \cos(2\pi\, fct) * m(t)$ ....(1)

Frequency Domain Description:

Taking Fourier transform of equation (1)

We know

$$s(t) \longleftrightarrow S(f)$$

$$m(t) \longleftrightarrow M(f)$$

$$A_c \mathrm{Cos} 2\pi f_c t \longleftrightarrow \frac{A_c}{2}\, [\boldsymbol{\delta}(f-f_c)+\boldsymbol{\delta}(f+f_c)]$$

Therefore:

$$S(f) = \frac{1}{2} A_c\, [M(f-fc) + M\,(f+fc)]$$

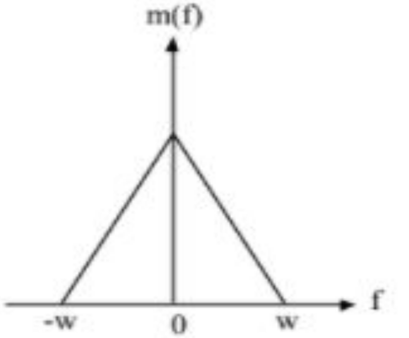


Fig. Spectrum of message

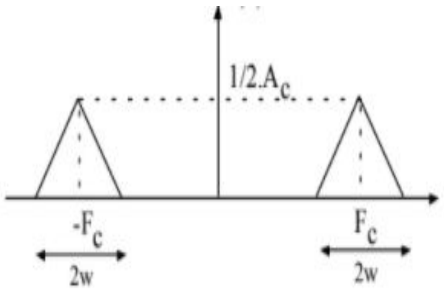
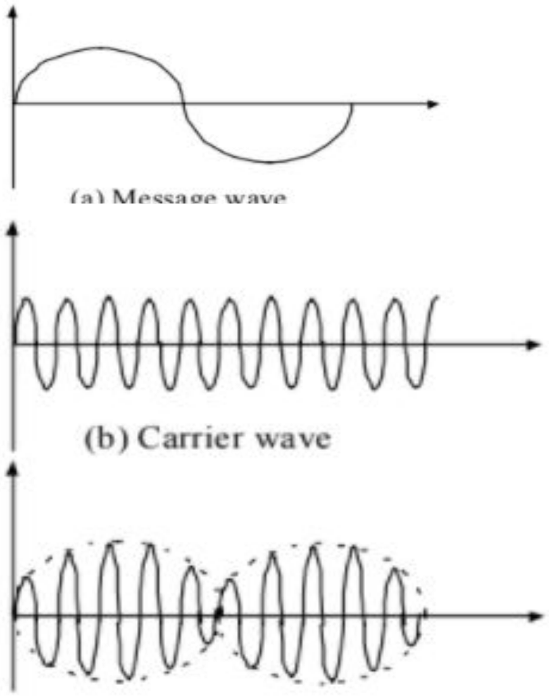


Fig. Spectrum of DSB-SC modulated wave


(a) Message wave

(b) Carrier wave

(c) DSB-SC modulated wave

Transmission **Bandwidth**:

BW = fc + fm – (fc- fm)

BW = fc + fm – fc +fm

BW = 2fm

That is twice the Bandwidth of message signal.

Transmission **Power**:

Pt = PLSB + PUSB

DSB –SC for single tone signal

- Let m(t) = Am Cos 2πfmt

C(t) = Ac Cos 2πfct

Then

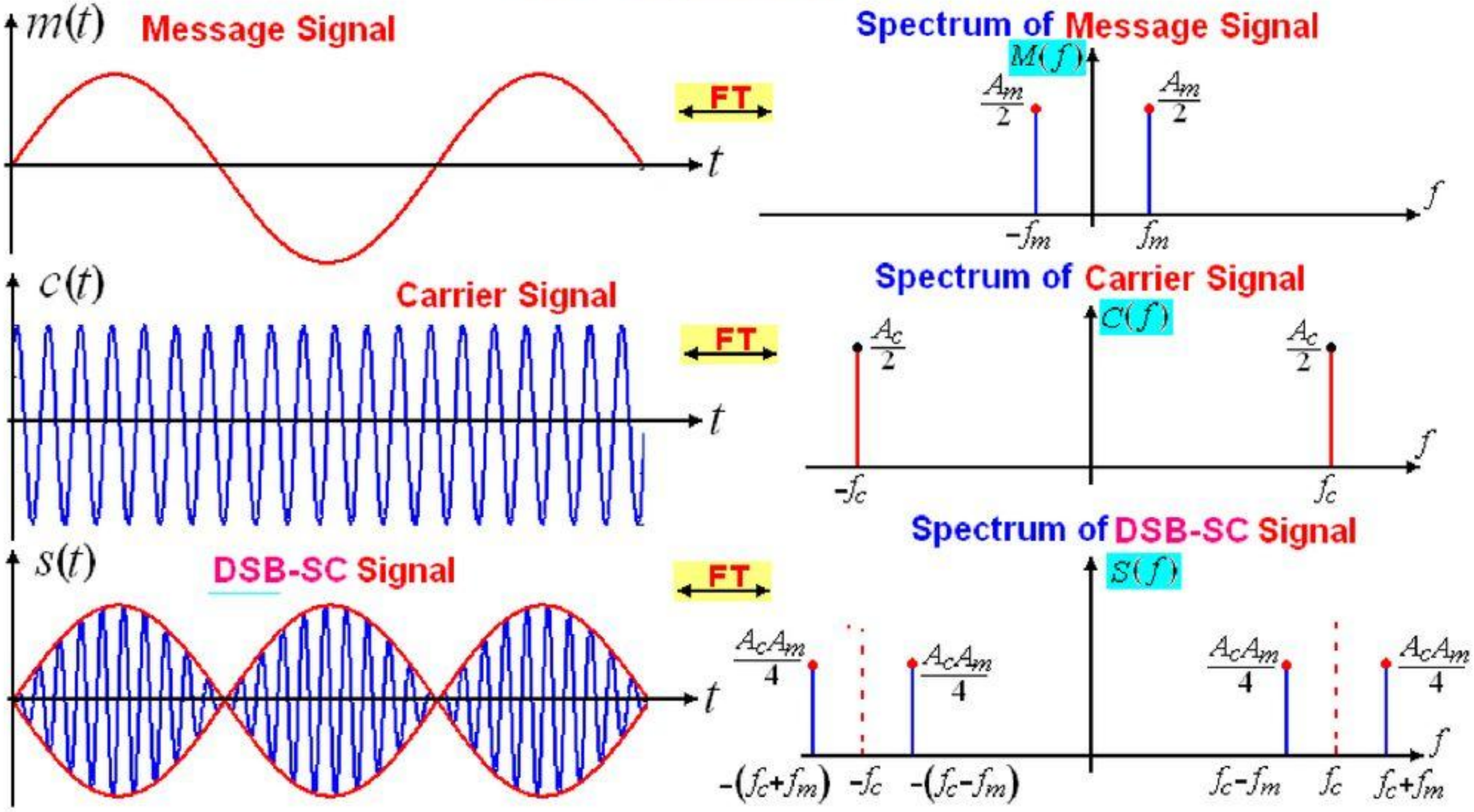
s(t) = m(t) * C(t)

= Am Ac cos 2πfmt cos2πfct

$= \frac{A_m A_c}{2}$ [cos 2π(fc +fm)t + cos2π(fc –fm)t]

Taking FT,

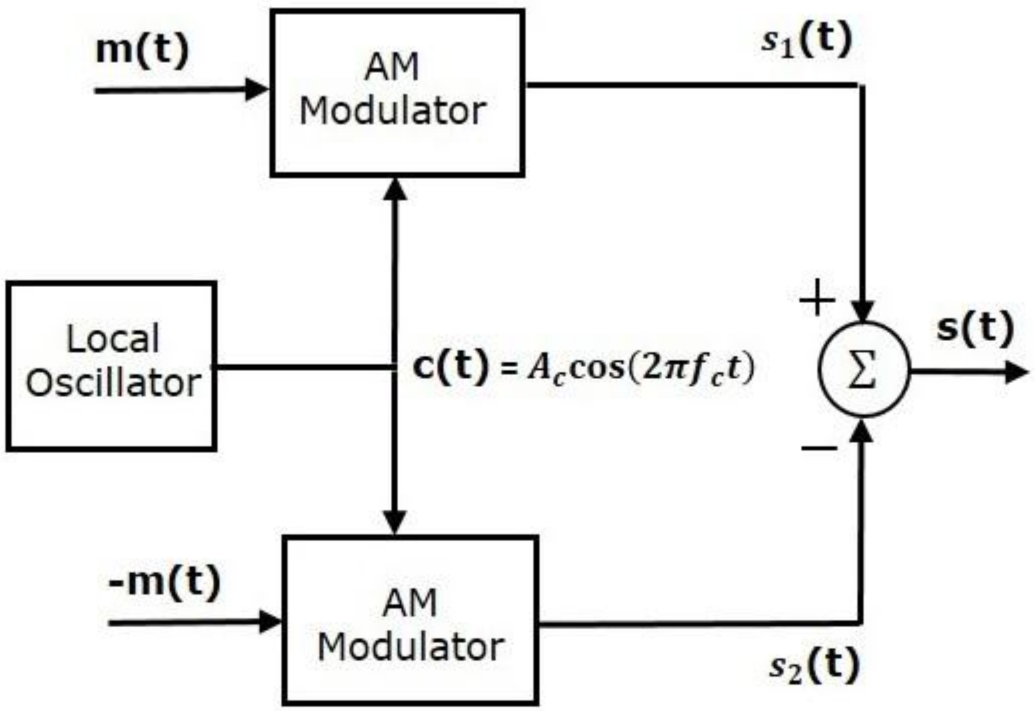
S(f) = $\frac{A_m A_c}{4}$ [δ (f-fc-fm) + δ(f +fm +fc) + δ(f-fc +fm) + δ(f +fc –fm)]



$s(t)$ undergoes a phase reversal whenever $m(t)$ crosses zero

Generation of DSCSC
1. Balanced Modulator

A balance modulator consists of two standard amplitude modulators arranged in a balance configuration so as to suppressed the carrier wave as shown in the block diagram given below:



We assume that two modulator are identical except for the sign reversal of the modulating wave apply to the input of one of them.

Thus the o/p of the two modulator may be expressed as:

$S1(t) = Ac[1+am_n(t)]\cos 2\pi fct$ ………………(i)

$S2(t) = Ac[1-am_n(t)]\cos 2\pi fct$ ………………(ii)

Then $S(t) = S1(t) - S2(t)$

$= Ac[1+amn(t)]\cos 2\pi fct + Ac[1-amn(t)]\cos 2\pi fct$

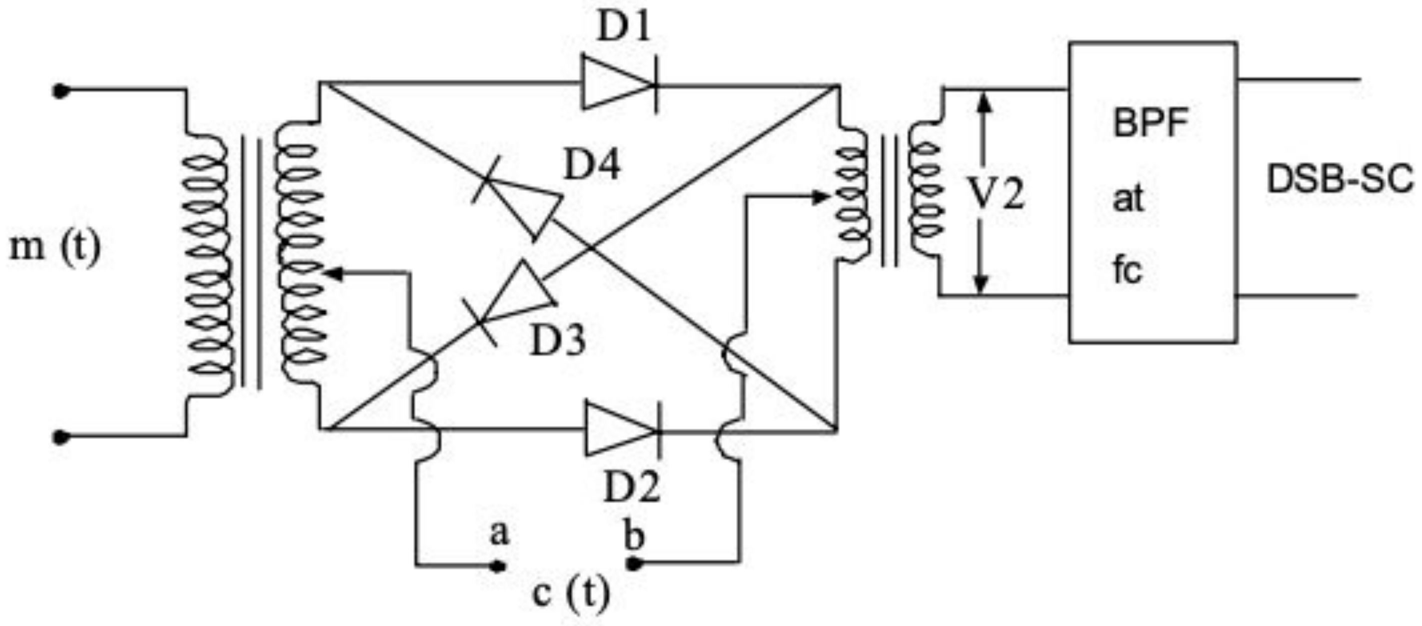
$\mathbf{S(t) = 2aA_c\, m_n(t)\cos 2\pi f_c t}$

Hence except for the scaling factor 2a the balance modulating factor is equal to the modulating wave and the carrier as required.

2. Ring Modulator

It is also know as a **lattice** or **double balance modulator**.

The four diode in figure below form a ring in which they all point in the same way.



The diodes are controlled by a square wave carrier **c(t)** of frequency fc which is applied by means of two centered tapped transformer.

We assume that

- diodes are ideal
- m(t) alone can not forward bias the diodes.
- C(t) is sufficient to forward bias the diodes.

In one half cycle of c(t) ,

terminal a become +ve and terminal b become –ve.

In this case diodes D1 and D2 are forward bias and D3 and D4 reverse bias.

As a result D1 and D2 pass the m(t) to the output as it is applied to the input.

Now in another half cycle of c(t) diodes D3 and D4 become forward bias and D1 , D2 reverse bias.

In this case m(t) is passed to the output in reverse polarity i.e – m(t) .

The net effect of this is equivalent to multiplying m(t) by a unit height square wave pulse train. g(t) .

The signal at the input of BPF is

$V2(t) = m(t) \times g(t)$ ……..(i)

Where,

$g(t) = 1$ if $c(t) > 0$

$= -1$ if $c(t) < 0$

- Expanding g(t) in Fourier series we get:

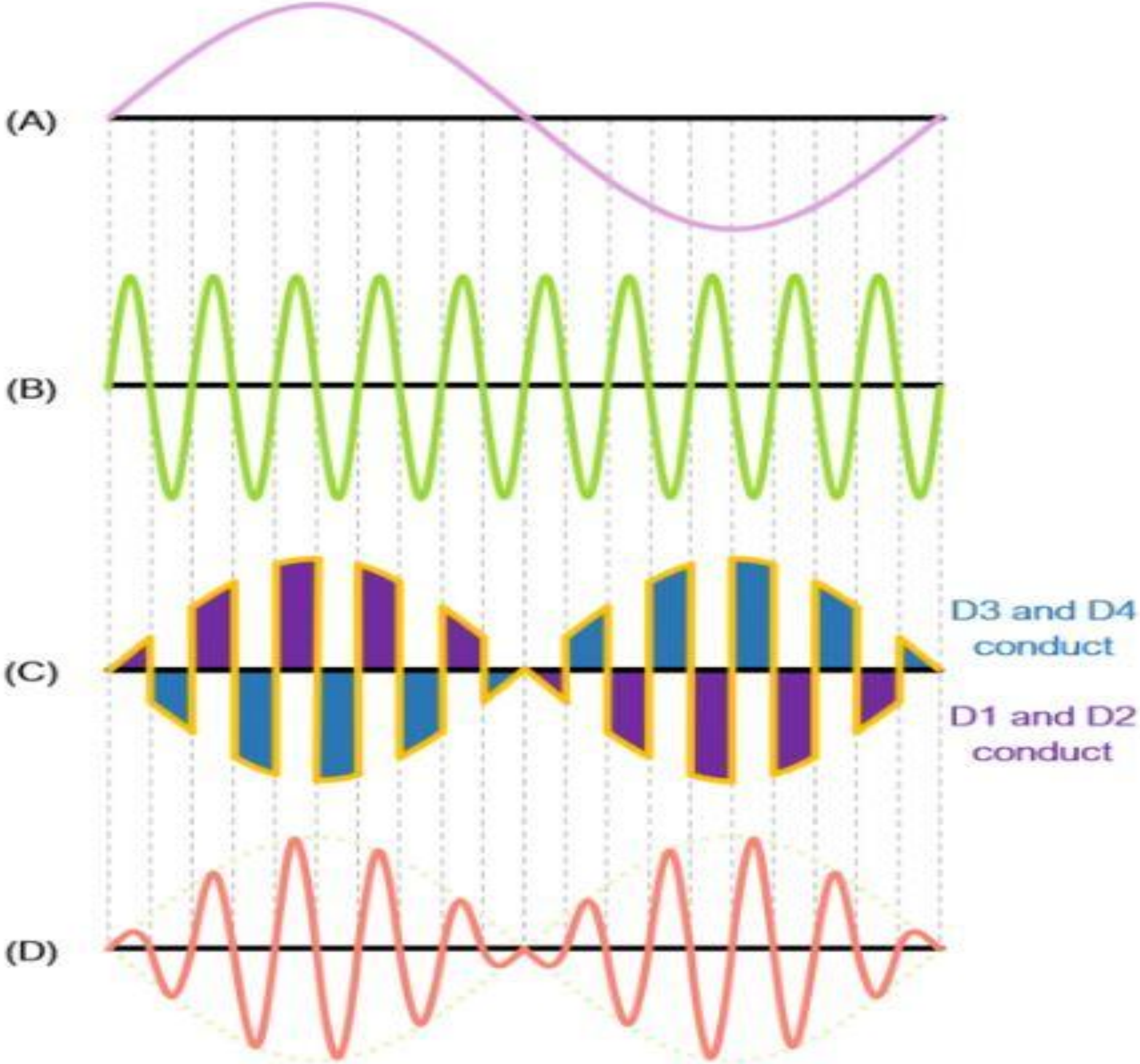
$$g(t) = \frac{4}{\pi} \; \mathrm{Cos} 2\pi f_c t + gh_o(t)$$

Where, $gh_o(t)$ = higher order harmonics.

The second term of above equation will be filtered out by BPF and the output of DSB-SC will be:

$$\mathbf{S(t) = \frac{4}{\pi} \, . \, m(t) \, . \, Cos2\pi fct}$$

Ring Modulator Waveform:



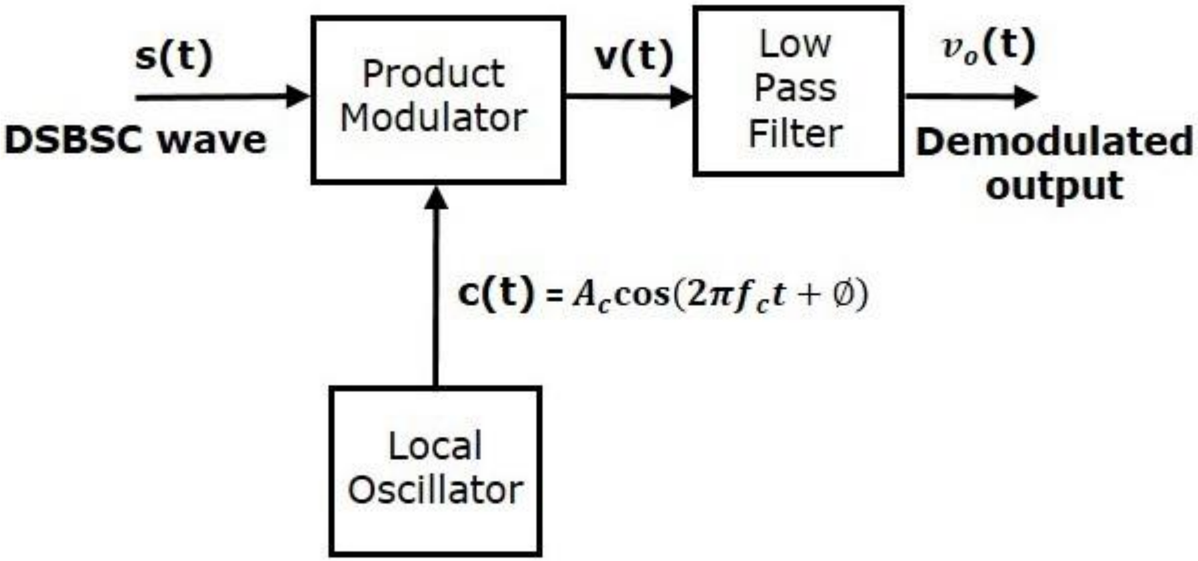
Demodulation of DSB-SC Signals

- The process of extracting an original message signal from DSBSC wave is known as detection or demodulation of DSBSC.
- Two types of detectors are used for demodulating DSBSC wave.
  - Coherent Detector or synchronous
  - Costas Loop

Synchronous Demodulation:

- The demodulation method in which received modulated signal is multiplied by locally generated carrier signal and then LPF is called coherent or synchronous detection.
- It is assumed that the frequency and phase of locally generated carrier signal and that of received signal are coherent or synchronized.

1. Coherent/Synchronous Detection of DSB-SC



Let modulated wave of DSB-SC be

$$s(t)=Ac\ Cos(2\pi fct).m(t)$$

$$c(t)=Ac\ Cos(2\pi fct+\varphi)$$

From figure: $v(t)=s(t)c(t)$

$$v(t)=Ac\ Cos(2\pi fct).m(t).Ac\ Cos(2\pi fct+\ \varphi)$$

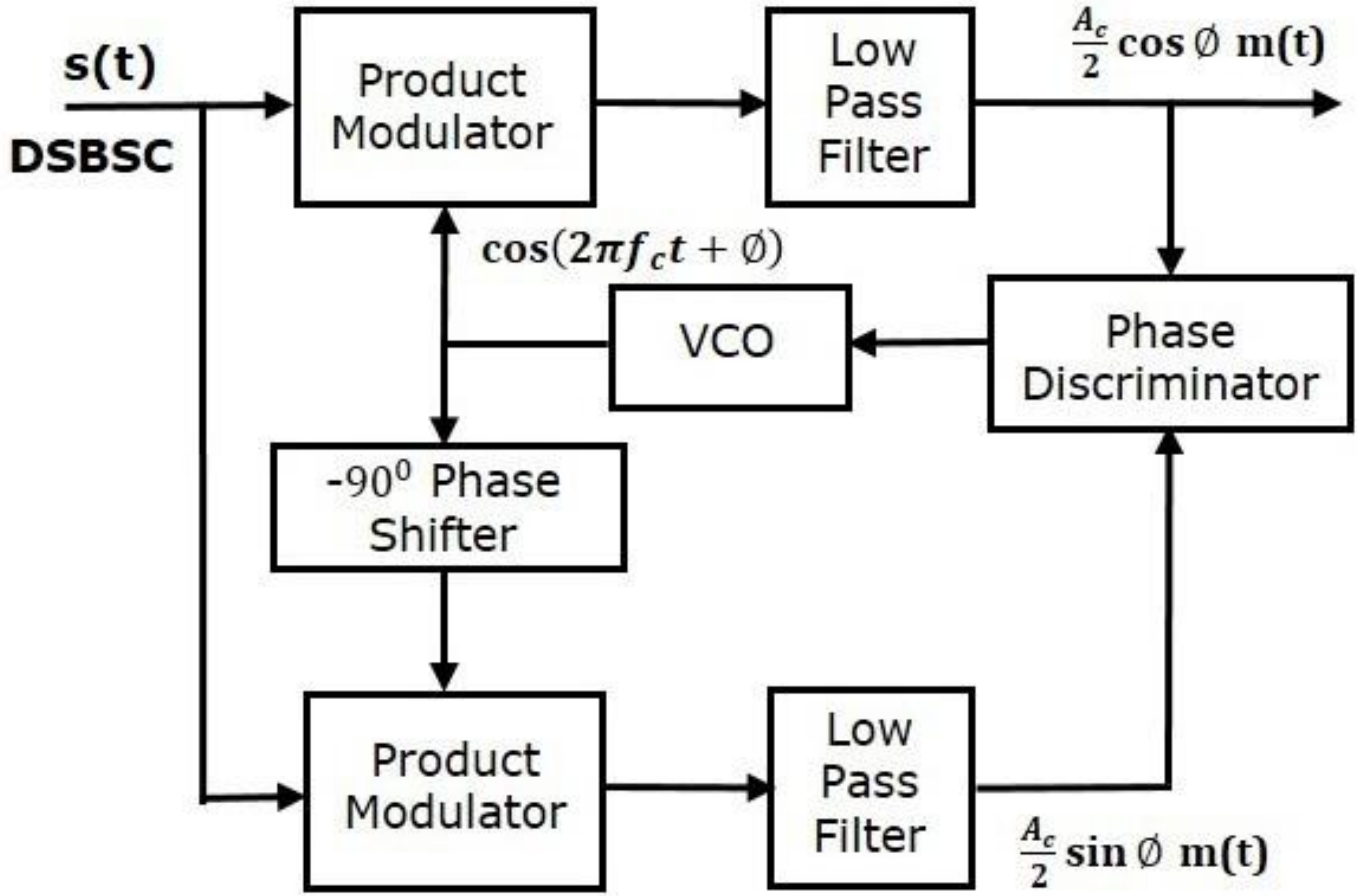
- $$= Ac^2 cos(2\pi fct)cos(2\pi fct+\phi)m(t)$$
  $$= Ac^2[cos(4\pi fct+\phi)+cos\phi]m(t)$$

$$v(t) \quad = \frac{Ac^2}{2} \; Cos\phi \; m(t) + \frac{Ac^2}{2} \; Cos(4\pi fct+\phi) \; m(t)$$

$$V_0(t) \quad = \frac{Ac^2}{2} \; cos\phi \; m(t)$$

- The demodulated signal amplitude will be maximum, when $\phi=0^0$. That's why the local oscillator signal and the carrier signal should be in phase, i.e., there should not be any phase difference between these two signals.
- The demodulated signal amplitude will be zero, when $\phi=\pm 90^0$.

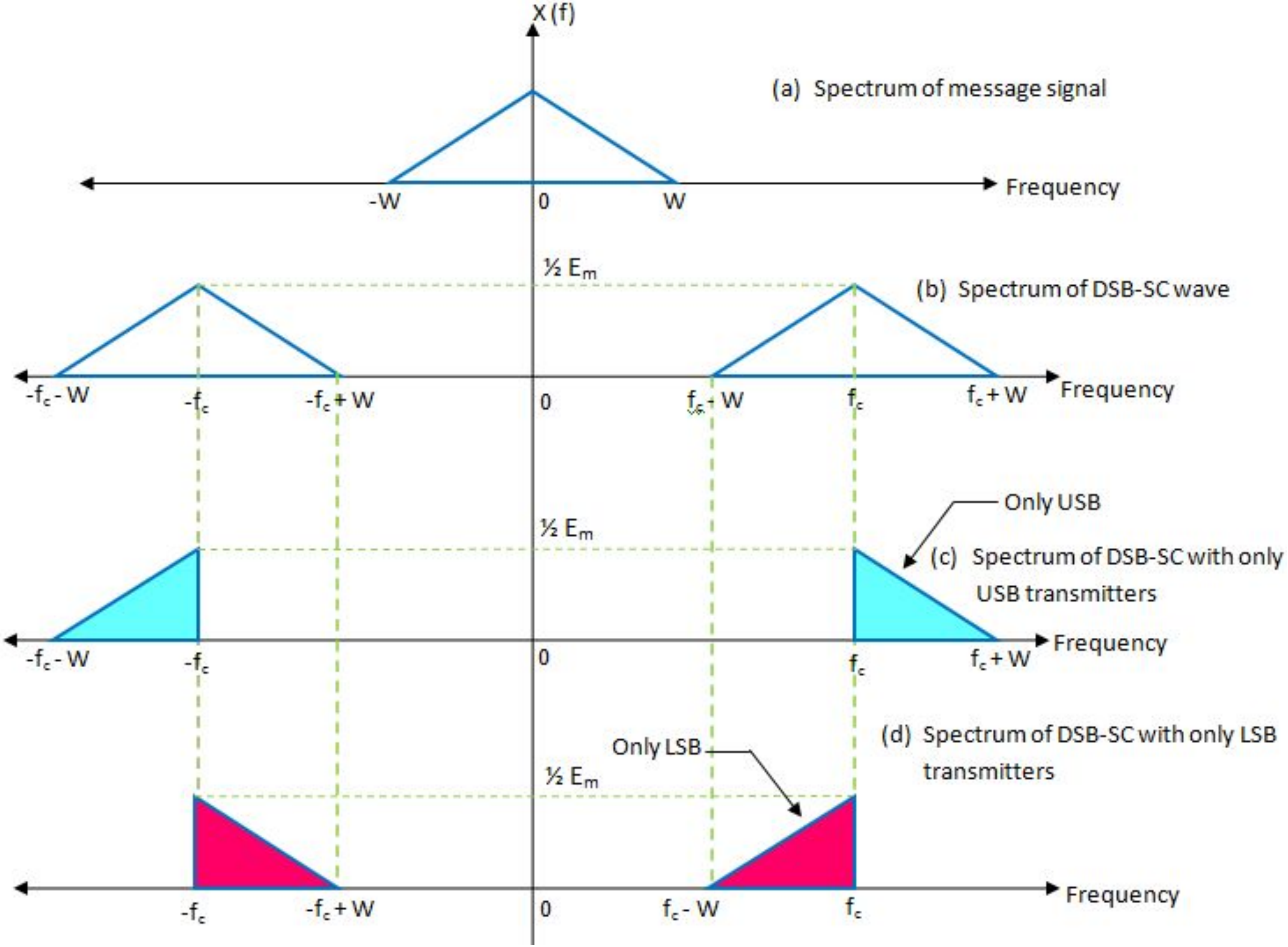
2. Costas loop:



SSB Amplitude Modulation

- DSB-AM and DSB-SC AM wave are wasteful of bandwidth since BW = 2fm.
- As far as the transmission of message is concerned only one sideband is necessary. Thus if carrier and one of the sideband is suppressed at transmitter , the message can be transmitted without wasting bandwidth.
- The modulation of this type which provides a single sideband with suppressed carrier is known as SSB-SC AM.
- It reduces transmission bandwidth by half compare to previous DSB FC AM and DSC SC AM

Frequency Domain Analysis



SSB Modulation

- Let us consider a standard AM signal in case of sinusoidal modulating signal which can be expressed as:

S(t) = Ac cos2πfct + AmAc/2 . cos2π(fc-fm)t + AmAc/2. cos2π(fc+fm)t

The standard equation of SSB-SC AM is

$$S(t) = \frac{Ac}{2} m(t) \cos 2\pi fct \pm \frac{Ac}{2} \widehat{m(t)} \sin 2\pi fct$$

Single Tone SSB modulation

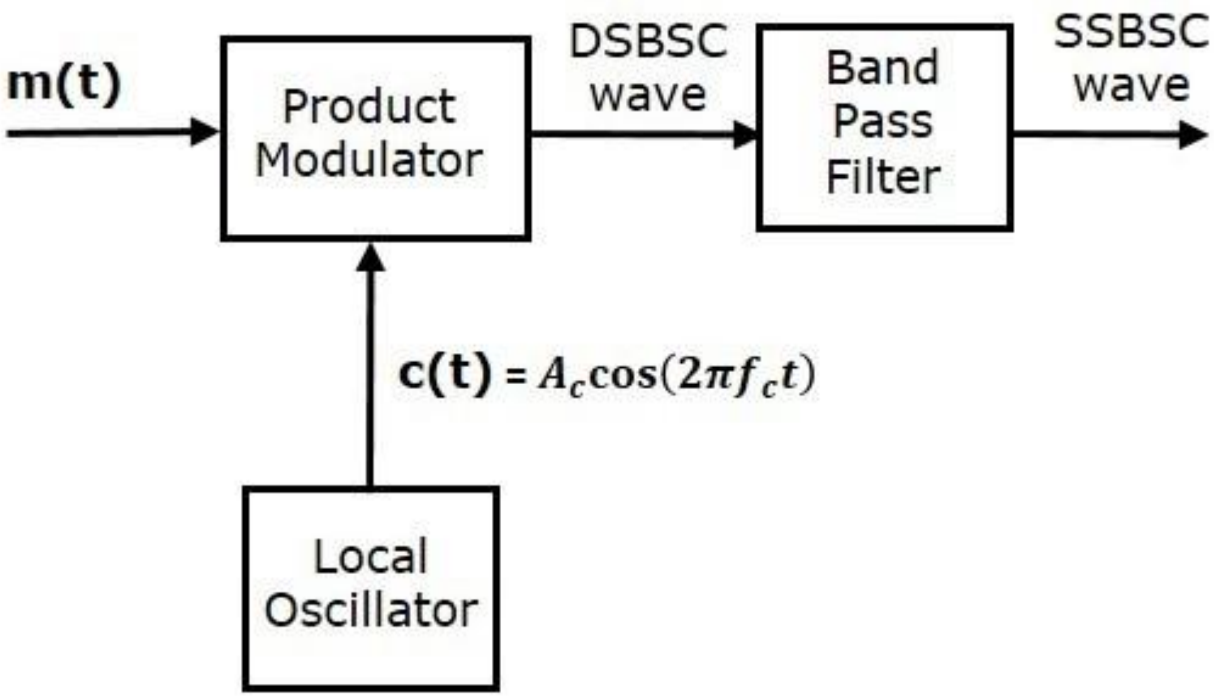
- **Class Assignment**
  - USB for odd roll number
  - LSB for Even roll number

Generation of SSB modulation

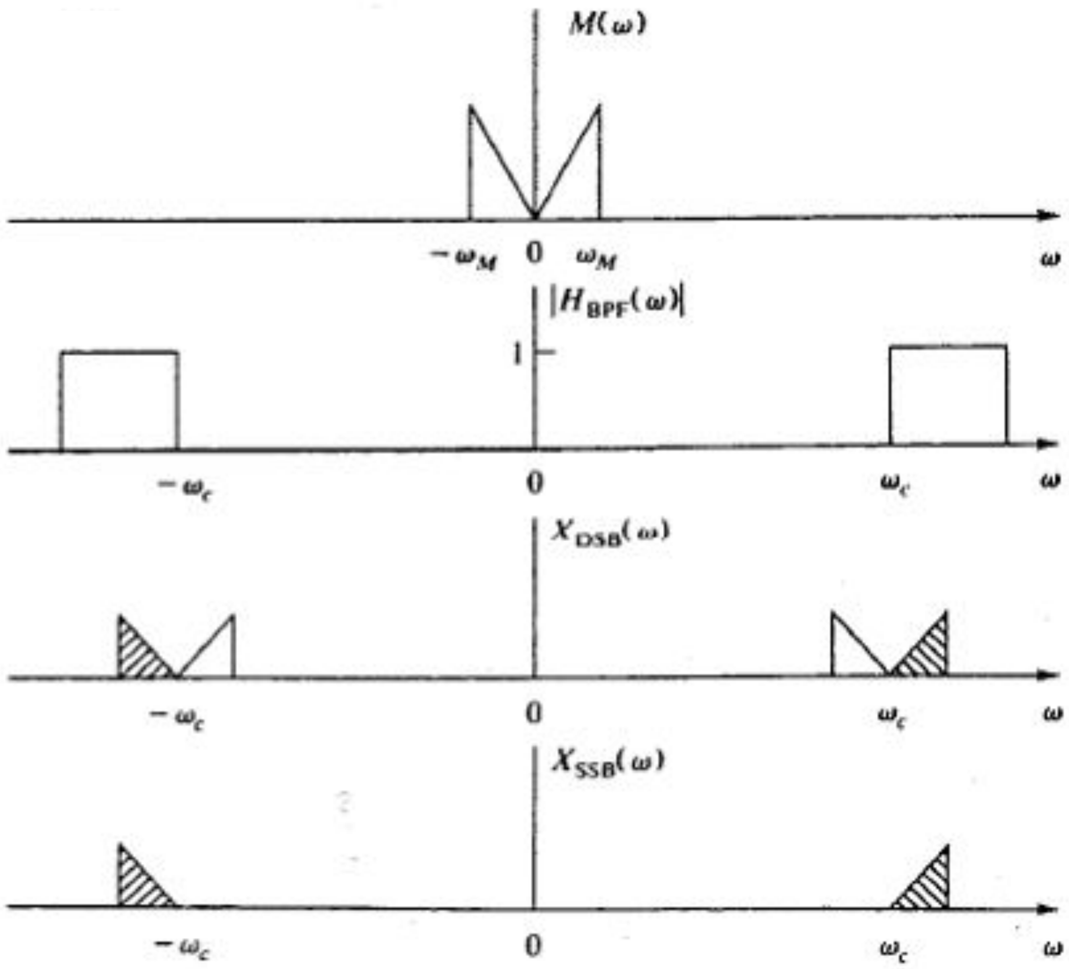
1. Filtering or Frequency Discrimination method
2. Phase Shift method

1. Filtering method

- In this method, first generate DSBSC wave with the help of the product modulator. Then, apply this DSBSC wave as an input of band pass filter. This band pass filter produces an output, which is SSBSC wave.
- Select the frequency range of band pass filter as the spectrum of the desired SSBSC wave. This means the band pass filter can be tuned to either upper sideband or lower sideband frequencies to get the respective SSBSC wave having upper sideband or lower sideband.

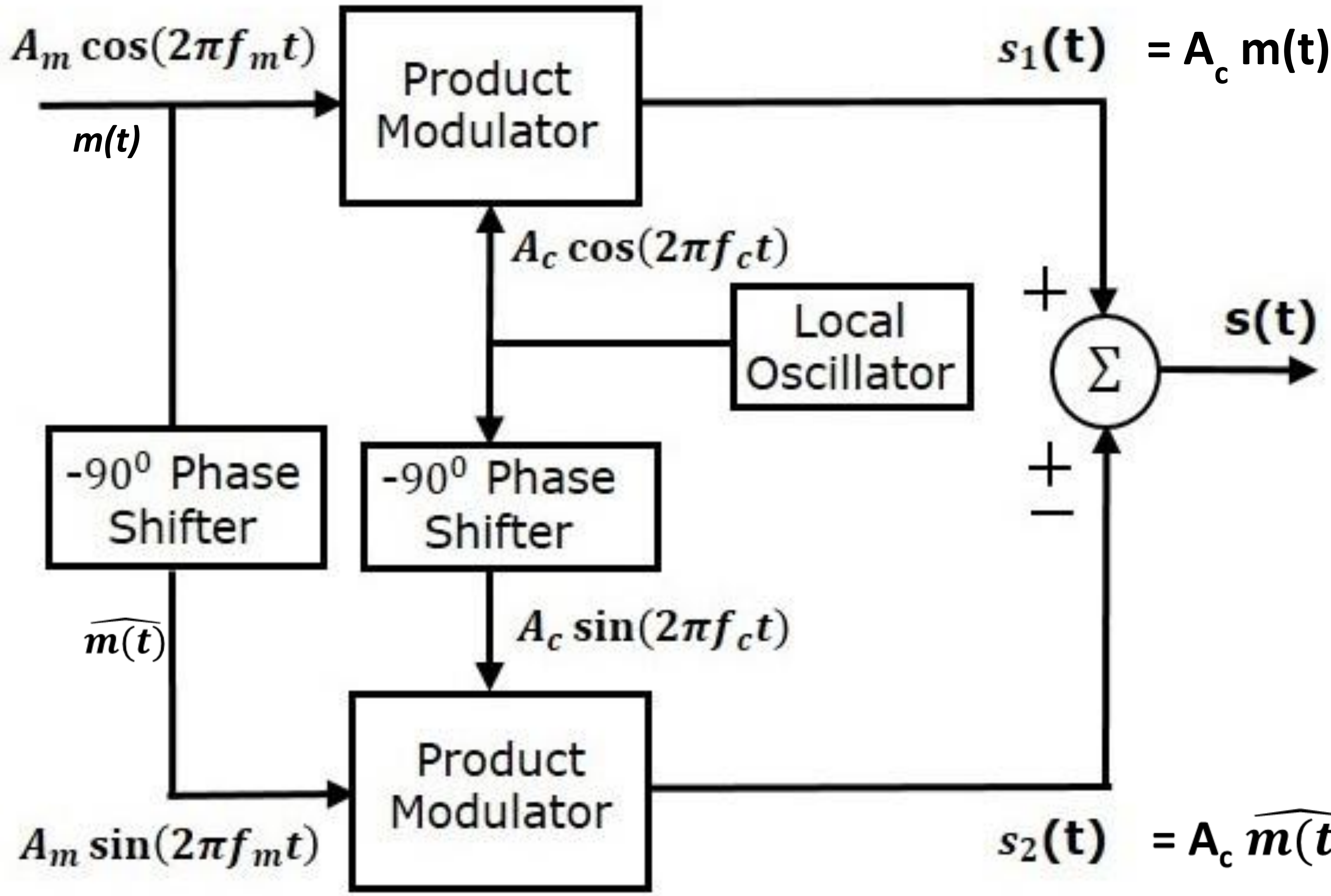


m(t)
Product Modulator
DSBSC wave
Band Pass Filter
SSBSC wave
Local Oscillator
c(t) = $A_c \cos(2\pi f_c t)$



$M(\omega)$
$-\omega_M$ 0 ω_M ω
$|H_{BPF}(\omega)|$
1
$-\omega_c$ 0 ω_c ω
$X_{DSB}(\omega)$
$-\omega_c$ 0 ω_c ω
$X_{SSB}(\omega)$
$-\omega_c$ 0 ω_c ω

2. Phase Shift method



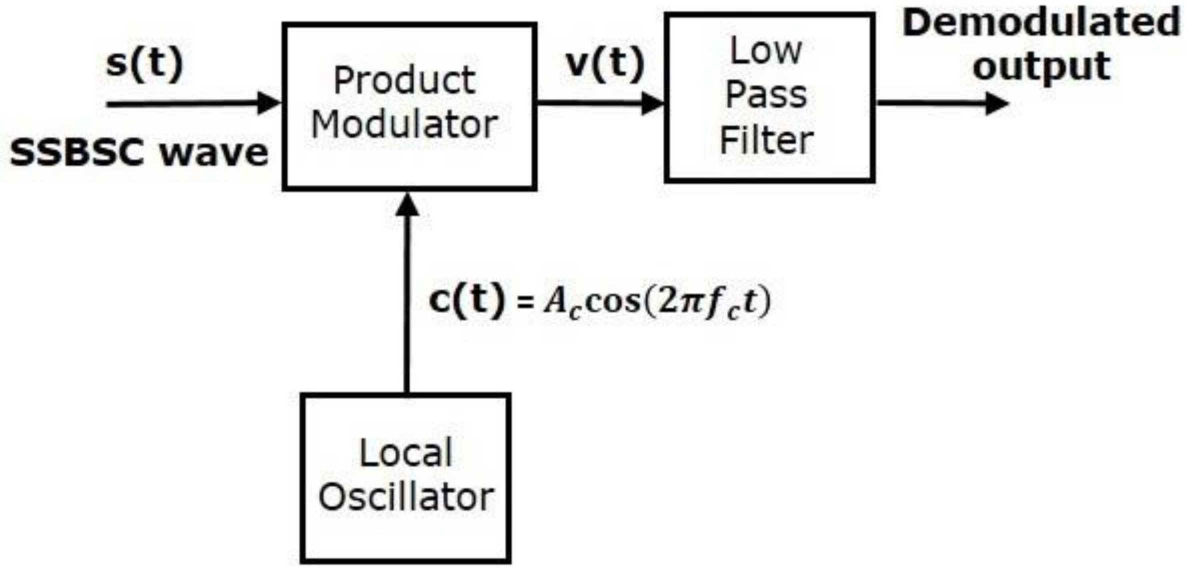
$= A_c\, m(t)\, \text{Cos}2\pi f_c t$

$= A_c\, \widehat{m(t)}\, \text{Sin}2\pi f_c t$

Demodulation of SSB waves:

Coherent or Synchronous detection

- Coherent means the same carrier signal (which is used for generating SSBSC wave) is used to detect the message signal.
- In this process, the message signal can be extracted from SSBSC wave by multiplying it with a carrier, having the same frequency and the phase of the carrier used in SSBSC modulation.
- The resulting signal is then passed through a Low Pass Filter. The output of this filter is the desired message signal.



- Let us consider the SSB modulated wave at the input is:

$$S(t) = \frac{Ac}{2} m(t) \cos 2\pi fct \pm \frac{Ac}{2} \widehat{m(t)} \sin 2\pi fct$$

Such that the output of product modulator is:

$$v(t) = c(t).s(t)$$

$$v(t) = \frac{Ac}{2} m(t) \cos^2 2\pi fct \pm \frac{Ac}{2} \widehat{m(t)} \sin 2\pi fct . \mathrm{Cos} 2\pi fct$$

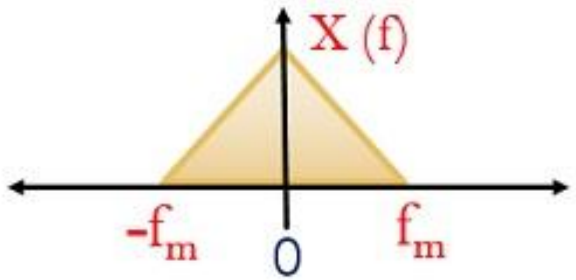
$$v(t) = \frac{Ac}{4} m(t) (1 + \cos 4\pi fct) \pm \frac{Ac}{4} \widehat{m(t)} \sin 4\pi fct$$

$$v(t)_{LPF} = \frac{Ac}{4} m(t)$$

Thus at the output of the filter we get the scaled message signal m(t).

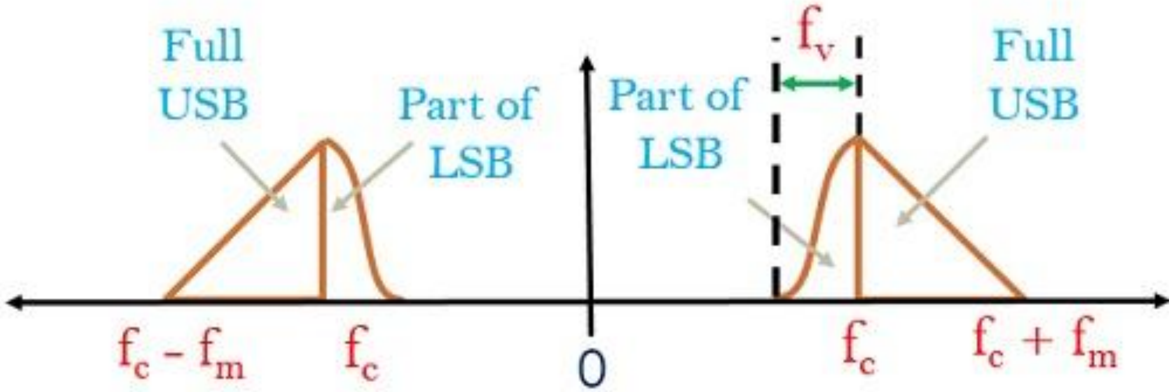
Vestigial Sideband (VSB) modulation:

- **VSB Modulation** is the process, where a part of the signal called as vestige is modulated along with one sideband.



Spectrum of message signal

***Bandwidth of VSB Modulated Wave* = f_m+f_v**

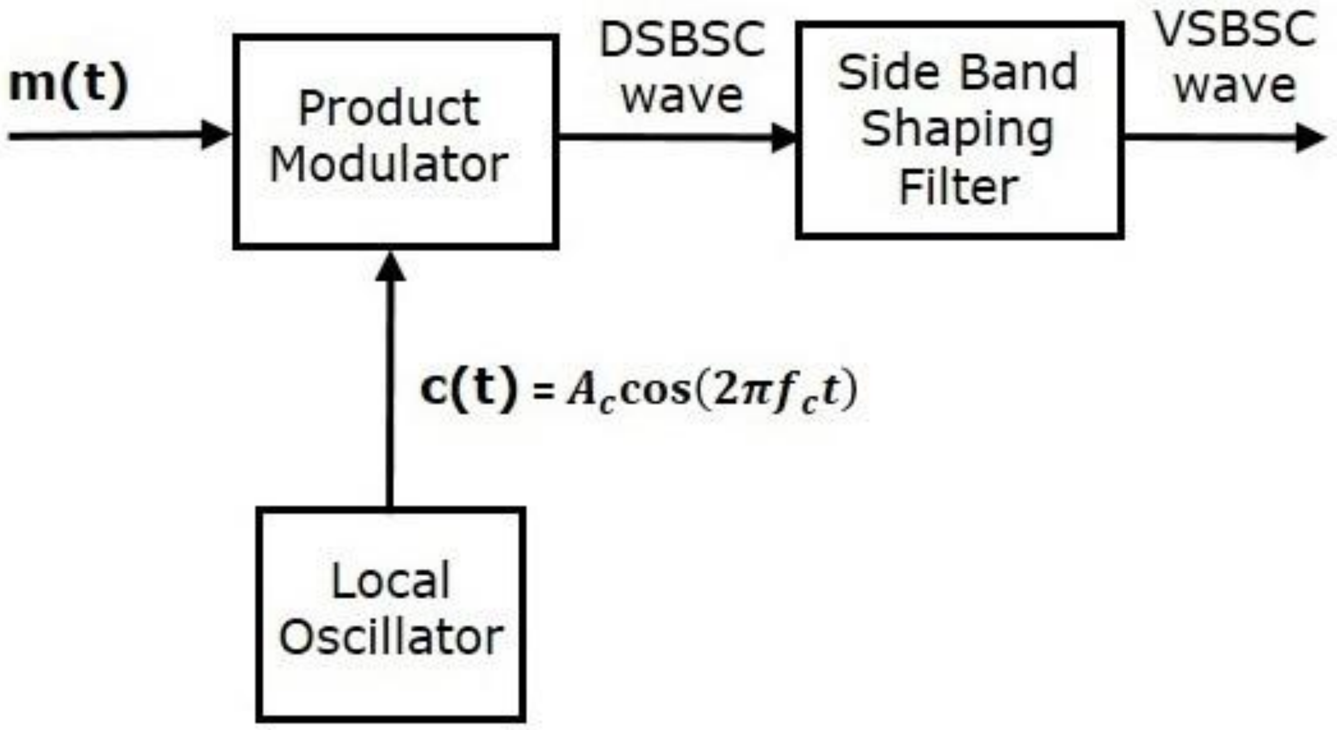


Spectrum of VSB signal

Generation of VSB:

- Generation of VSB wave is similar to the generation of SSB wave.
- The modulating signal m(t) and carrier signal $A_c \cos(2\pi f_c t)$ are applied as inputs to the product modulator.
- The product modulator produces an output, which is the product of these two inputs.
- The output of the product modulator is DSBSC signal

$$s1(t)=A_c \cos(2\pi f_c t)\, m(t)$$



- $$S1(f)=\frac{Ac}{2}\,[M(f-fc)+M(f+fc)]$$

Let the transfer function of the sideband shaping filter be H(f).

- This filter has the input s1(t) and the output is VSB modulated wave s(t).

The Fourier transforms of s1(t) and s(t) are S1(f) and S(f) respectively.
Mathematically,

$$S(f)=S1(f)H(f)$$

$$S(f)=\frac{Ac}{2}[M(f-fc)+M(f+fc)]H(f)$$

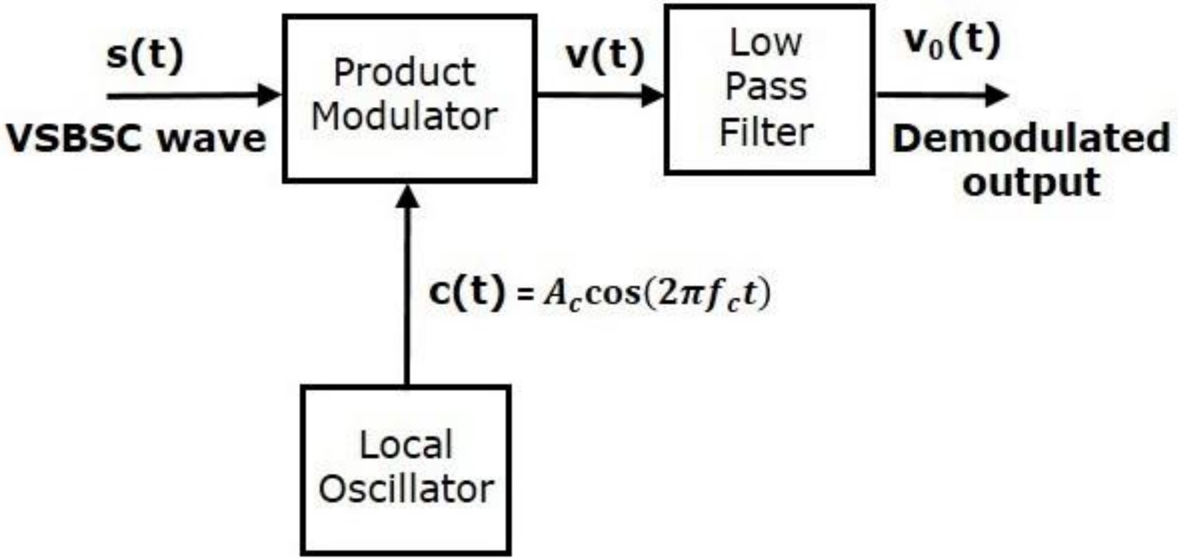
In time domain expression;

$$s(t) = s1(t) * h(t)$$

Demodulation of VSB:

- Let the VSB wave be s(t) and the carrier signal is Ac Cos(2πfct).
- From the figure, we can write the output of the product modulator as

v(t)=Ac Cos(2πfct) s(t)



Taking FT on both sides:

$V(f) = S(f) \cdot \frac{Ac}{2}\ [\boldsymbol{\delta}(f-f_c)+\boldsymbol{\delta}(f+f_c)]$

We know $\mathbf{S(f)=\frac{Ac}{2}[M(f-fc)+M(f+fc)]H(f)}$

So, $\mathbf{V(f) =\frac{Ac^2}{4}\ [M(f-fc)+M(f+fc)].}\ [\boldsymbol{\delta}(f-fc)+\boldsymbol{\delta}(f+fc)]\ \mathbf{.\ H(f)}$

$V(f) = \frac{Ac^2}{4}[M(f-2f_c)+M(f)]H(f-f_c) + \frac{Ac^2}{4}[M(f)+M(f+2f_c)]H(f+f_c)]$

$V(f) = \frac{Ac^2}{4}M(f)[H(f-f_c)+H(f+f_c)] + \frac{Ac^2}{4}[M(f-2f_c)H(f-f_c)+M(f+2f_c)H(f+f_c)]$

In the above equation, the first term represents the scaled version of the desired message signal frequency spectrum. It can be extracted by passing the above signal through a low pass filter.

$V_0(f)= \frac{Ac^2}{4}M(f)[H(f-f_c)+H(f+f_c)]$

Quadrature Amplitude Modulation (QAM)

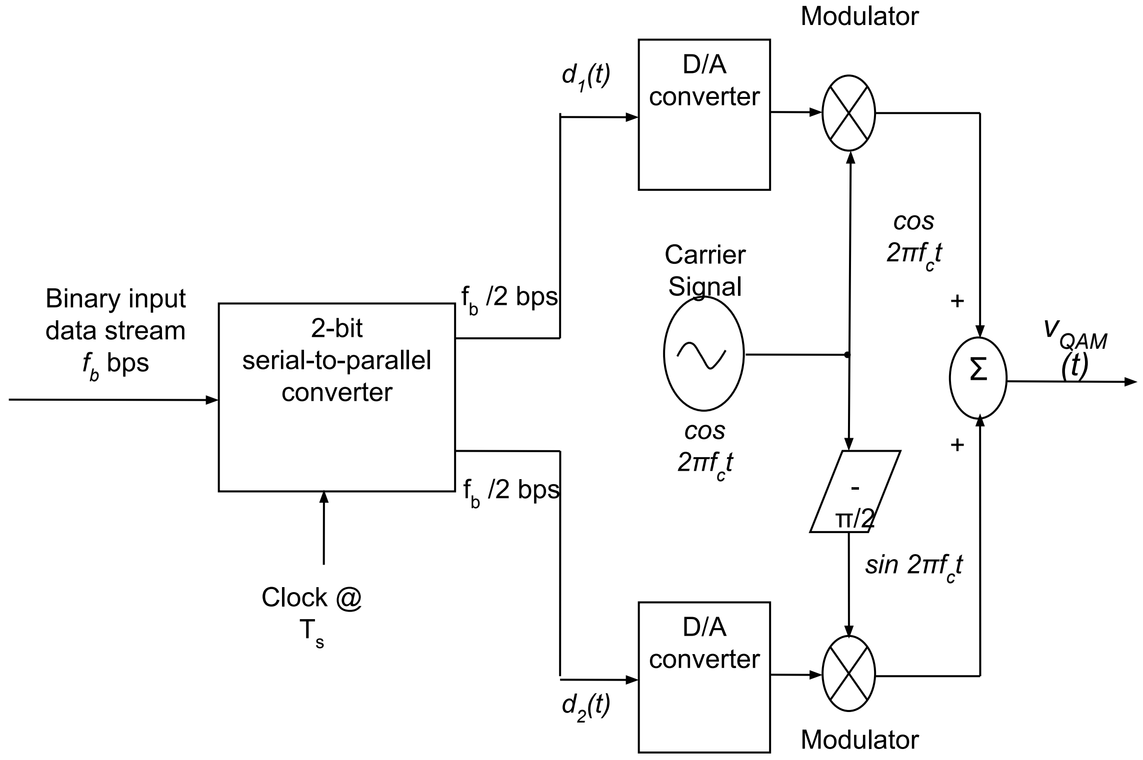
QAM - A form of digital modulation similar to PSK except the digital information is contained in both the amplitude and the phase of the modulated signal

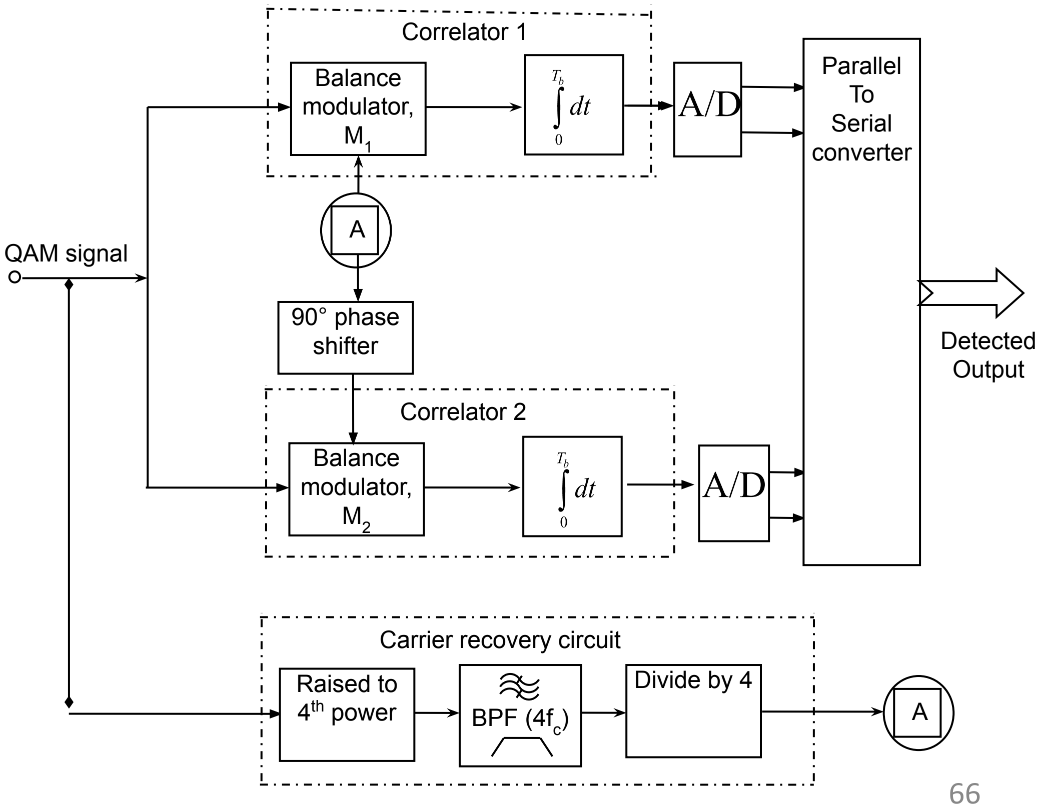
QAM can either be considered a logical extension of QPSK or a combination of ASK and PSK.

QAM is an efficient way to achieve high data rates with a narrowband channel by increasing the number of bits per symbol, and uses a combination of amplitude and phase modulation.

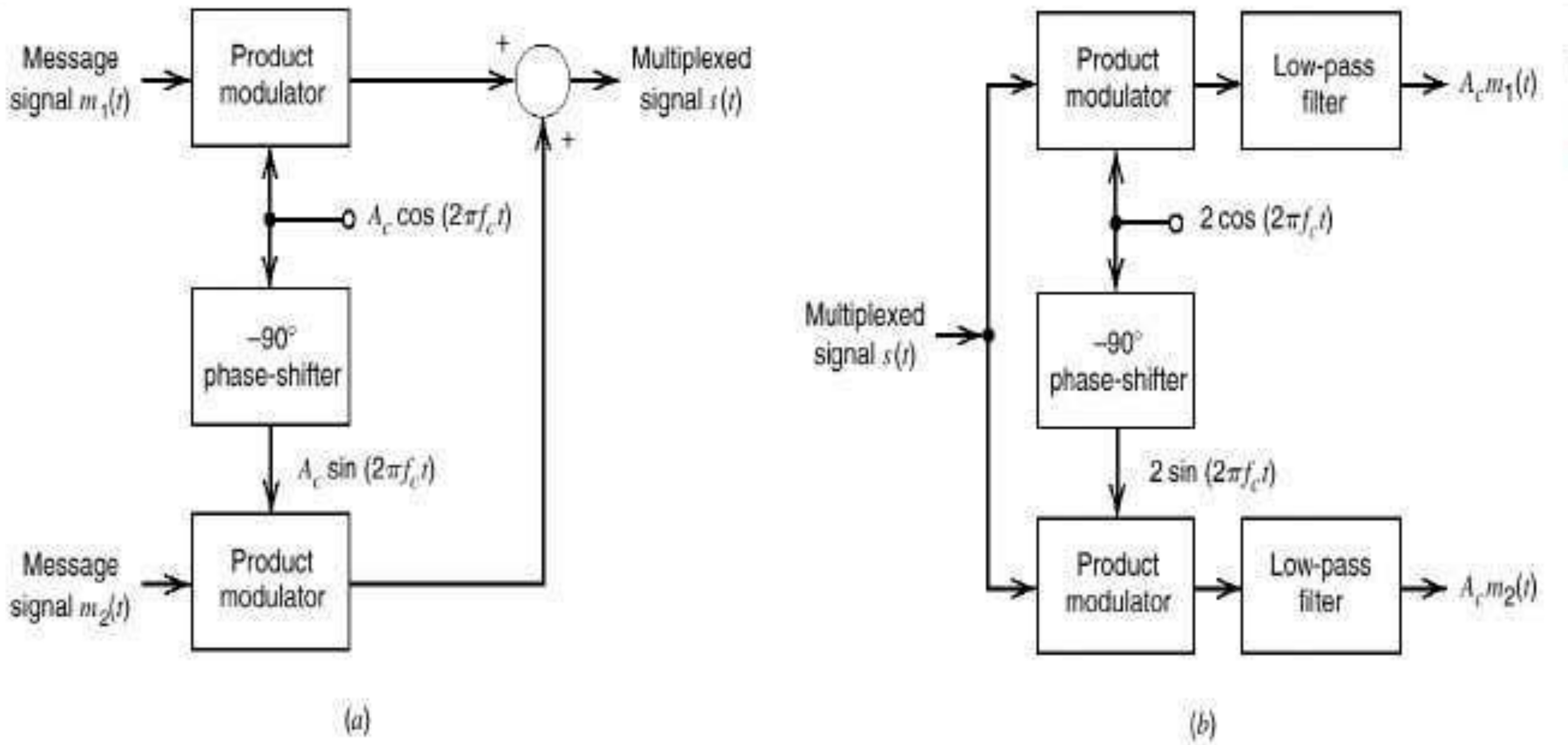
QAM Modulator and Demodulator

- QAM modulators and demodulators are required to provide the capability to modulate both the **in-phase** and **quadrature components** of the modulating signal onto the carrier.





Application of QAM: ISB (Independent Sideband)



Phase Locked Loop (PLL)

- Will be discussed later on chapter 6.

Thank You