UNIT I

AMPLITUDE MODULATION

Amplitude modulation or AM as it is often called, is a form of modulation used for radio transmissions for broadcasting and two-way radio communication applications. Although one of the earliest used forms of modulation it is still used today, mainly for long, medium and short wave broadcasting and for some aeronautical point to point communications.

One of the key reasons for the use of amplitude modulation was its ease of use. The system simply required the carrier amplitude to be modulated, but more usefully the detector required in the receiver could be a simple diode based circuit. This meant that AM radios did not need complicated demodulators and costs were reduced - a key requirement for widespread use of radio technology, especially in the early days of radio when ICs were not available.

Amplitude modulation history

The first amplitude modulated signal was transmitted in 1901 by a Canadian engineer named Reginald Fessenden. He took a continuous spark transmission and placed a carbon microphone in the antenna lead. The sound waves impacting on the microphone varied its resistance and in turn this varied the intensity of the transmission. Although very crude, signals were audible over a distance of a few hundred meters, although there was a rasping sound caused by the spark.

With the introduction of continuous sine wave signals, transmissions improved significantly, and AM soon became the standard for voice transmissions. Nowadays, amplitude modulation, AM is used for audio broadcasting on the long medium and short wave bands, and for two way radio communication at VHF for aircraft. However as there now are more efficient and convenient methods of modulating a signal, its use is declining, although it will still be very many years before it is no longer used.

Amplitude modulation applications

Amplitude modulation is used in a variety of applications. Even though it is not as widely used as it was in previous years in its basic format it can nevertheless still be found.

Broadcast transmissions: AM is still widely used for broadcasting on the long, medium and short wave bands. It is simple to demodulate and this means that radio receivers capable of demodulating amplitude modulation are cheap and simple to manufacture. Nevertheless, many people are moving to high quality forms of transmission like frequency modulation, FM or digital transmissions.

Air band radio: VHF transmissions for many airborne applications still use Amplitude Modulation. It is used for ground to air radio communications as well as two-way radio links for ground staff as well.

Single sideband: Amplitude modulation in the form of single sideband is still used for HF radio links. Using a lower bandwidth and providing more effective use of the transmitted power this form of modulation is still used for many point to point HF links.

Quadrature amplitude modulation: AM is widely used for the transmission of data in everything from short range wireless links such as Wi-Fi to cellular telecommunications and much more. Effectively it is formed by having two carriers 90° out of phase. These form some of the main uses of amplitude modulation.

Need for Amplitude modulation

In order that a radio signal can carry audio or other information for broadcasting or for two-way radio communication, it must be modulated or changed in some way. Although there are a number of ways in which a radio signal may be modulated, one of the easiest is to change its amplitude in line with variations of the sound. In this way the amplitude of the radio frequency signal varies in line with the instantaneous value of the intensity of the modulation. This means that the radio frequency signal has a representation of the sound wave superimposed in it.

In view of the way the basic signal "carries" the sound or modulation, the radio frequency signal is often termed the "carrier". A continuous-wave goes on continuously without any intervals and it is the baseband message signal, which contains the information. This wave has to be modulated. According to the standard definition, "The amplitude of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal." Which means, the amplitude of the carrier signal containing no information varies as per the amplitude of the signal containing information, at each instant. This can be well explained by the following figures 1.1.1, 1.1.2, 1.1.3 and figure 1.1.4.

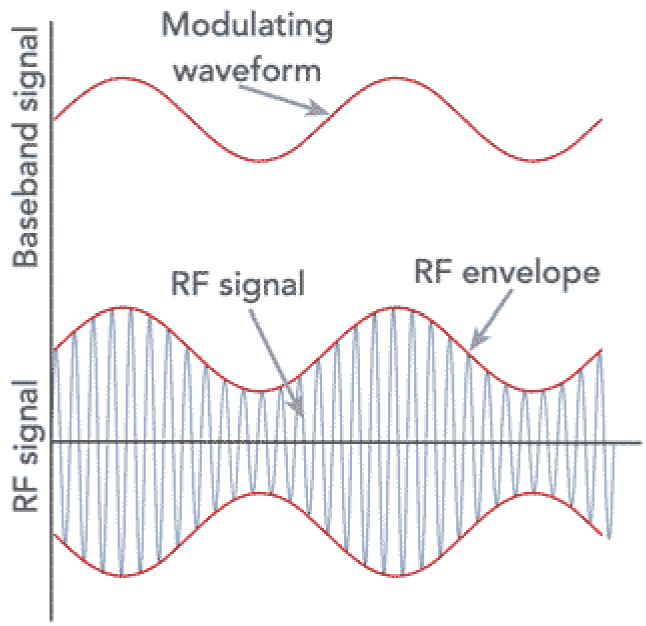

Figure 1.1.1 Modulating wave form and Modulated RF Signal

Diagram Source : Electronic Tutorials

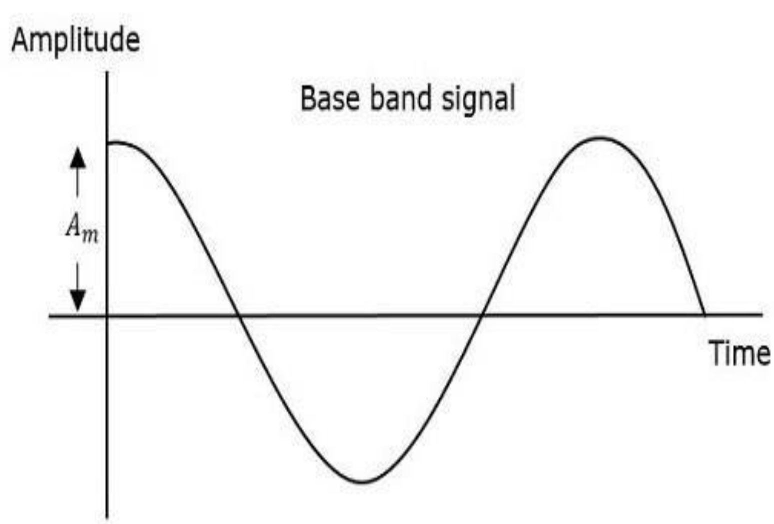

Figure: 1.1.2 Base Band Signal

Diagram Source : Electronic Tutorials

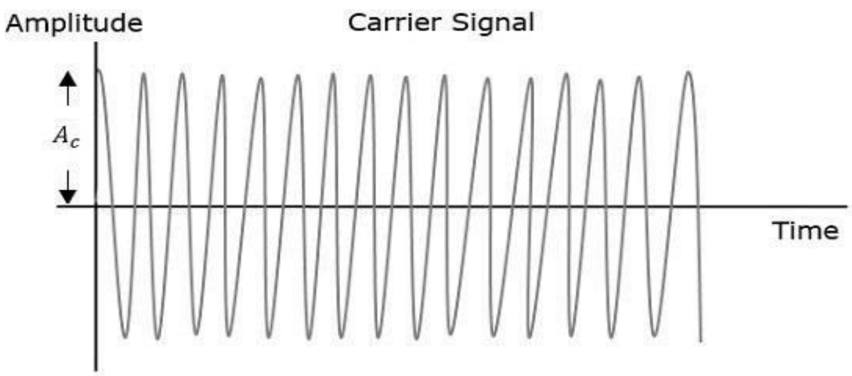

Fig1.1.3 Carrier Signal

Diagram Source : Electronic Tutorial

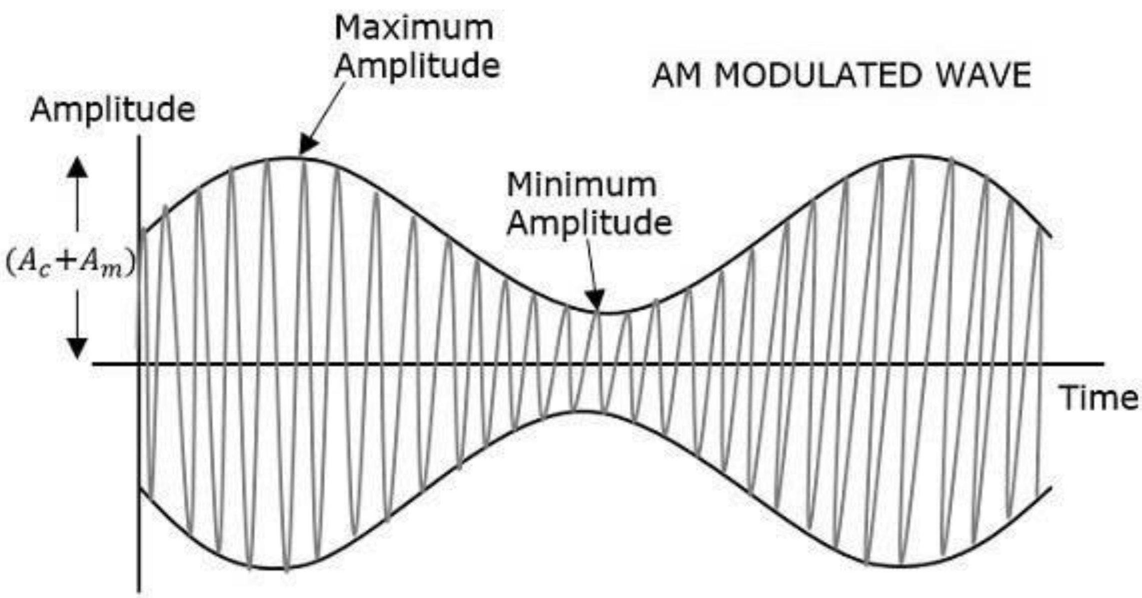

Figure 1.1.4 AM Modulated Signal

Diagram Source : Electronic Tutorial

The figure 1.1.4 shows the modulating wave, which is the message signal. The next one is the carrier wave, which is a high frequency signal and contains no information. While, the last one is the resultant modulated wave.

It can be observed that the positive and negative peaks of the carrier wave, are interconnected with an imaginary line. This line helps recreating the exact shape of the modulating signal. This imaginary line on the carrier wave is called as Envelope. It is the same as that of the message signal.

Mathematical Expressions

Following are the mathematical expressions for these waves. Time-domain Representation of the Waves Let the modulating signal be,

$$m(t)=A_m \text{Cos}(2\pi f_m t) \quad (1)$$

and the carrier signal be,

$$c(t)=A_c \text{Cos}(2\pi f_c t) \quad (2)$$

Where,

Am and Ac are the amplitude of the modulating signal and the carrier signal respectively.

fm and fc are the frequency of the modulating signal and the carrier signal respectively.

Then, the equation of Amplitude Modulated wave will be

$$s(t)=[A_c+A_m\cos(2\pi f_m t)]\text{Cos}(2\pi f_c t) \quad (3)$$

Modulation Index

A carrier wave, after being modulated, if the modulated level is calculated, then such an attempt is called as Modulation Index or Modulation Depth. It states the level of modulation that a carrier wave undergoes.

Rearrange the Equation as below.

$$s(t)=Ac[1+(A_m A_c)\text{Cos}(2\pi f_m t)]\text{Cos}(2\pi f_c t) \quad (4)$$

$$\Rightarrow s(t)=Ac[1+\mu\cos(2\pi f_m t)Cos(2\pi f_c t) \quad (5)$$

Where, μ is Modulation index and it is equal to the ratio of Am and Ac. Mathematically, we can write it as

$$\mu=Am/Ac \quad (6)$$

Hence, we can calculate the value of modulation index by using the above formula, when the amplitudes of the message and carrier signals are known. Now, let us derive one more formula for Modulation index . We can use this formula for calculating modulation index value, when the maximum and minimum amplitudes of the modulated wave are known.

Let Amax and Amin be the maximum and minimum amplitudes of the modulated wave.

We will get the maximum amplitude of the modulated wave, when cos(2πfmt)

$$\Rightarrow Amax=Ac+Am \quad (7)$$

We will get the minimum amplitude of the modulated wave, when cos(2πfmt)cos(2πfmt) is -1.

$$\Rightarrow Amin=Ac-Am$$
$$\Rightarrow Amin=Ac-Am$$

Adding

$$Amax+Amin=Ac+Am+Ac-Am=2Ac$$
$$Ac=(Amax+Amin)/2 \Rightarrow Ac=(Amax+Amin)/2 \quad (8)$$

Subtracting

$$Amax-Amin=Ac+Am-(Ac-Am)$$

=2AmAmax−Amin=Ac+Am−(Ac−Am)=2Am

Am=(Amax−Amin)/2⇒Am=(Amax−Amin)/2

μ=Amax−Amin / Amax+Amin

μ=Amax−Amin/ Amax+Amin (9)

Therefore, the above Equations are the two formulas for Modulation index. The modulation index or modulation depth is often denoted in percentage called as Percentage of Modulation. We will get the percentage of modulation, just by multiplying the modulation index value with 100.

For a perfect modulation, the value of modulation index should be 1, which implies the percentage of modulation should be 100%.

For instance, if this value is less than 1, i.e., the modulation index is 0.5, then the modulated output would look like the following figure. It is called as Under-modulation. Such a wave is called as an under-modulated wave. Fig 1.1.5 shows Under Modulated Wave.

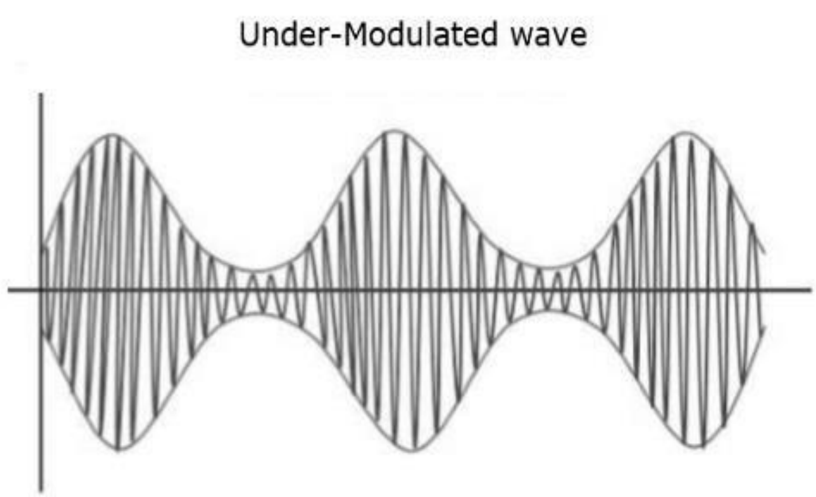

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Figure 1.1.5 Under Modulated Wave

Diagram Source : Brain Kart

If the value of the modulation index is greater than 1, i.e., 1.5 or so, then the wave will be an over-modulated wave. It would look like the following figure 1.1.6.

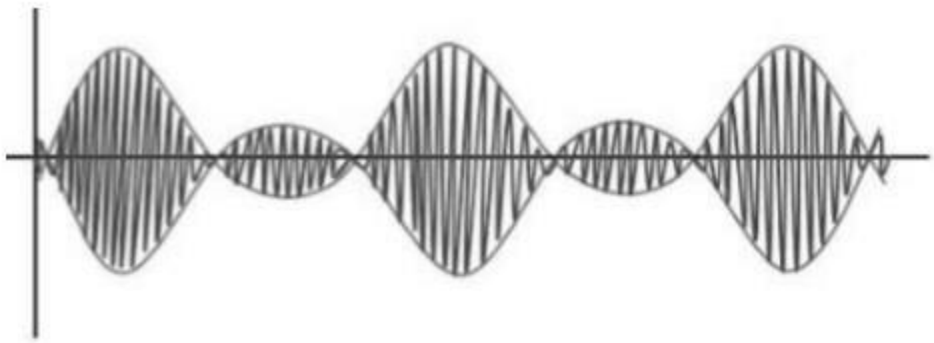


Figure 1.1.6 Over Modulated Signal

Diagram Source : Electronic Tutorials

As the value of the modulation index increases, the carrier experiences a 180^o phase reversal, which causes additional sidebands and hence, the wave gets distorted. Such an over-modulated wave causes interference, which cannot be eliminated.

Let us consider that a carrier signal A cos ωct is amplitude-modulated by a single-tone modulating signal

$$x(t) = Vm\ A \cos \omega_m t. \quad (9)$$

Then the unmodulated or carrier power

PC = mean square (rms value)

Power Content In Multiple-tone Amplitude Modulation

Mathematical Expression

Let us consider that a carrier signal A cos ωct is modulated by a baseband or modulating signal x(t) which is expressed as :

$$x(t) = V_1 \cos \omega_1 t + V_2 \cos \omega_2 t + V_3 \cos \omega_3 t \qquad (10)$$

We know that the general expression for AM wave is

$$s(t) = A \cos \omega_c t + x(t) \cos \omega_c t \qquad (11)$$

Putting the value of x(t), we get

$$s(t) = A \cos \omega_c t + [V_1 \cos \omega_1 t + V_2 \cos \omega_2 t + V_3 \cos \omega_3 t] \cos \omega_c t$$

The expression for AM wave can further be expanded as under:

$$s(t) = A \cos \omega_c t + m_1 A \cos \omega_c t \cos \omega_1 t + m_2 A \cos \omega_c t \cos \omega_2 t + m_3 A \cos \omega_c t \cos \omega_3 t$$

Now we know that the total power in AM is given as ,

Pt = carrier power + sideband power

Pt = PC + PS

The carrier power PC is given as considering additional resistance like antenna resistance R.

$$P_{carrier} = [(V_c/\sqrt{2})^2 / R] = V^2C/2R \qquad (12)$$

Each side band has a value of mVc /2 and r.m.s value of $(m_a V_c/2)/\sqrt{2}$. Hence power in LSB and USB can be written as

$PLSB = PUSB = (mVc/2)/\sqrt{2})^2/R = m^2V^2_C/8R = m2/4\ Pcarrier$

$P_{total} = V^2C/2R + [m^2V^2_C/8R] + [m^2V^2_C/8R]$

$= V^2C/2R + [m^2V^2_C/4R\}$

$= V^2_C/2R\ (1 + m^2/2)$

$$P_{total} = Pcarrier\ (1 + m^2/2) \qquad (13)$$

In some applications, the carrier is simultaneously modulated by several sinusoidal modulating signals. In such a case, the total modulation index is given as

$mt = \sqrt{}(m_1^2 + m_2^2 + m_3^2 + m_4^2 + \ldots..$

If Ic and It are the r.m.s values of unmodulated current and total modulated current and R is the resistance through which these current flow, then

$P_{total}/P_{carrier} = (I_{t.}R/Ic.R)^2 = (It/Ic)^2$

$Pt_{ota}l/P_{carrier} = (1 + m^2/2)$

$$It/Ic = 1 + m^2/2 \qquad (14)$$

Limitations of Amplitude Modulation:

1. Switching modulator Low Efficiency- Since the useful power that lies in the small bands is quite small, so the efficiency of AM system is low.
2. Limited Operating Range – The range of operation is small due to low efficiency. Thus, transmission of signals is difficult.
3. Noise in Reception – As the radio receiver finds it difficult to distinguish between the amplitude variations that represent noise and those with the signals, heavy noise is prone to occur in its reception.

4. Poor Audio Quality – To obtain high fidelity reception, all audio frequencies till 15 Kilo Hertz must be reproduced and this necessitates the bandwidth of 10 Kilo Hertz to minimize the interference from the adjacent broadcasting stations. Therefore in AM broadcasting stations audio quality is known to be poor.

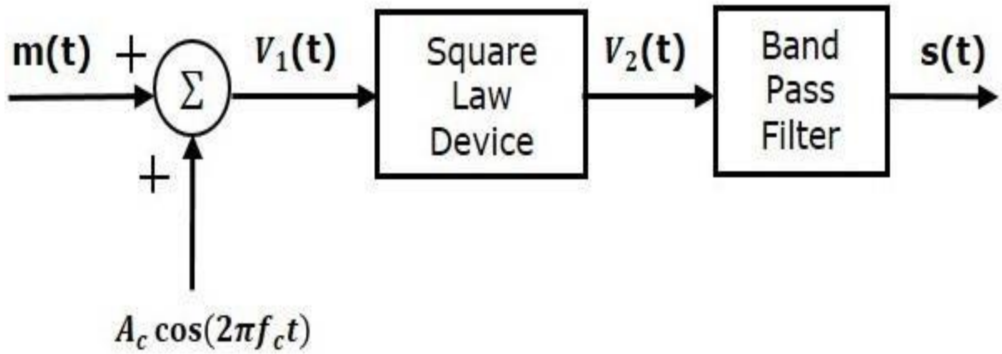
The following two modulators generate AM wave.

- Square law modulator
- Switching Modulator

Square Law Modulator

Following is the block diagram of the square law modulator sown in figure 1.1.7

Figure 1.1.7 Block diagram of the square law modulator



Let the modulating and carrier signals be denoted as m(t)m(t) and Acos(2πfct) respectively. These two signals are applied as inputs to the summer (adder) block. This summer block produces an output, which is the addition of the modulating and the carrier signal. Mathematically, we can write it as

$$V_1(t)=m(t)+A_c \mathrm{Cos}(2\pi fct) \quad (15)$$

This signal V1tV1t is applied as an input to a nonlinear device like diode. The characteristics of the diode are closely related to square law.

$$V_2(t)=k_1V_1(t)+k_2V_2(t)$$

$$V_2(t)=k_1V_1(t)+k_2V_2(t)$$

Where, k_1 and k_2 are constants.

Substitute V1(t) in V_2 (t),

$$V_2(t) = k_1[m(t) + A_c\cos(2\pi f_c t)] + k_2[m(t) + A_c\cos(2\pi f_c t)]^2$$

$$\Rightarrow V_2(t) = k_1 m(t) + k_1 A_c\cos(2\pi f_c t) + k_2 m^2(t) +$$

$$k_2 A_c^2\cos^2(2\pi f_c t) + 2k_2 m(t) A_c\cos(2\pi f_c t)$$

$$\Rightarrow V_2(t) = k_1 m(t) + k_2 m^2(t) + k_2 A_c^2\cos^2(2\pi f_c t) +$$

$$k_1 A_c\left[1 + \left(\frac{2k_2}{k_1}\right) m(t)\right]\cos(2\pi f_c t)$$

The last term of the above 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

$$s(t)=k_1 A_c[1+(2k_2 / k_1)m(t)]\cos(2\pi f_c t) \quad (16)$$

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 k1k1 and the amplitude sensitivity kaka as 2k /2k1

Switching Modulator

Following is the block diagram of switching modulator shown in Figure 1.1.8.

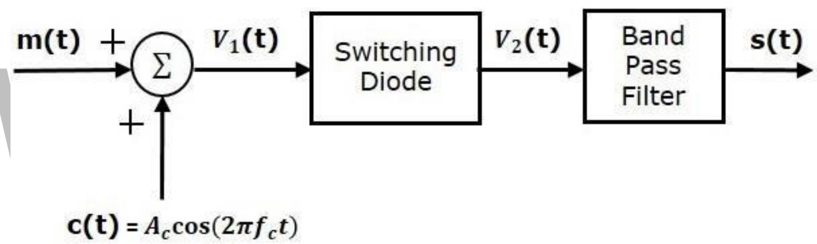


Figure 1.1.8 block diagram of the Switching modulator

Switching modulator is similar to the square law modulator. The only difference is that in the square law modulator, the diode is operated in a non-linear mode, whereas, in the switching modulator, the diode has to operate as an ideal switch.

Let the modulating and carrier signals be denoted as m(t)m(t) and $m(t)=A_c \cos(2\pi f_m t)$ and $c(t)=A_c\cos(2\pi f_c t)$ respectively. These two signals are applied as inputs to the summer (adder) block. Summer block produces an output, which is the addition of modulating and carrier signals. Mathematically, we can write it as

$$V_1(t)=m(t)+c(t)$$

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

This signal V1(t) is applied as an input of diode. Assume, the magnitude of the modulating signal is very small when compared to the amplitude of carrier signal Ac. So, the diode's ON and OFF action is controlled by carrier signal c(t). This means, the diode will be forward biased when c(t)>0 and it will be reverse biased when c(t)<0.

Therefore, the output of the diode is

$$V_2(t) = \begin{cases} V_1(t) & if \quad c(t) > 0 \\ 0 & if \quad c(t) < 0 \end{cases}$$

We can approximate this as

$$V_2(t) = V_1(t)\, x(t)$$

Where, $x(t)$ is a periodic pulse train with time period $T = \frac{1}{f_c}$

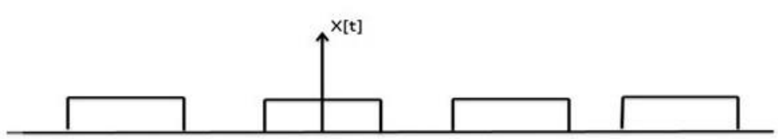


Figure 1.1.9 Periodic pulse train Diagram Source Brain Kart

Diagram Source : Brain Kart

The Fourier series representation of this periodic pulse train is in figure 1.1.9 is ,

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

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

Substitute, $V_1(t)$ and $x(t)$ √ 2.

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

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

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

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

$$s(t)=A_c/2(1+(4\pi A_c)m(t))\cos(2\pi f_c t) \qquad (17)$$

We know the standard equation of AM wave is

$$s(t)=Ac[1+k_a m(t)]\cos(2\pi fct)$$

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

The switching modulator using a diode has been shown in figures 1.1.10 (a) & (b) .

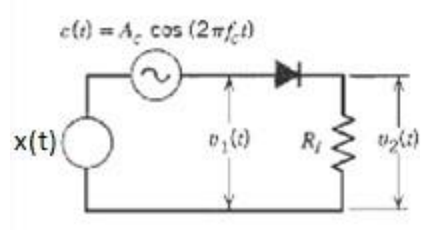


Figure 1.1.10 (a) Switching Modulator

Diagram Source : Brain Kart

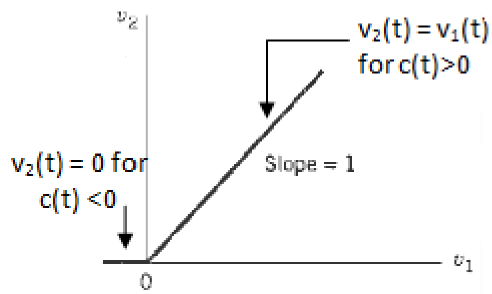


Figure1. 1.10 (b) Input and output voltage Characteristics

Diagram Source : Brain Kart

This diode is assumed to be operating as a switch . The modulating signal x(t) and the sinusoidal carrier signal c(t) are connected in series with each other. Therefore, the input voltage to the diode is given by

$$v_1(t) = c(t) + x(t) = E_c \cos(2\pi f_c t) + x(t) \qquad (18)$$

The amplitude of carrier is much larger than that of x(t) and c(t) decides the status of the diode (ON or OFF) .

Working Operation and Analysis

Let us assume that the diode acts as an ideal switch .Hence, it acts as a closed switch when it is forward biased in the positive half cycle of the carrier and offers zero impedance .Whereas it acts as an open switch when it is reverse biased in the negative half cycle of the carrier and offers an infinite impedance .Therefore, the output voltage v2(t) = v1(t) in the positive half cycle of c(t) and v2(t) = 0 in the negative half cycle of c(t) .

Hence ,

$$v_2(t) = v_1(t) \quad \text{for } c(t) > 0$$

$$v2(t) = 0 \quad \text{for } c(t) < 0 \qquad (19)$$

In other words , the load voltage v2(t) varies periodically between the values v1(t) and zero at the rate equal to carrier frequency fc .We can express v2(t) mathematically as under a pulse train. In figure 1.1.11 :

$$v_2(t) = v_1(t)\,.\,g_p(t) = [x(t) + E_c\cos(2\pi f_c t)]\, g_p(t)$$

where, gp(t) is a periodic pulse train of duty cycle equal to one half cycle period i.e. T0 /2 (where T0 = 1/fc) .

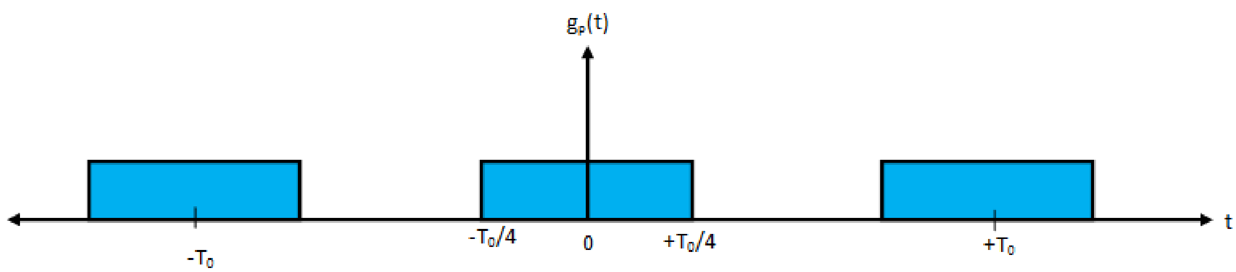


Figure 1.1.11 Periodic pulse train of duty cycle equal to one half cycle period

Diagram Source : Electronic Tutorial

Let us express $g_p(t)$ with the help of Fourier series as under :

$$g_p(t) = \frac{1}{2} + \frac{2}{\pi}\sum_{n=1}^{\infty}\frac{(-1)^{n-1}}{2n-1}\cos\,[2\pi f_c t(2n-1)]$$

$$g_p(t) = \frac{1}{2} + \frac{2}{\pi}\cos(2\pi f_c t) + odd\ harmonic\ components$$

Substituting $g_p(t)$ into $v_2(t)$ equation , we get

$$v_2(t) = [x(t) + E_c\cos((2\pi f_c t)]\left\{\frac{1}{2} + \frac{2}{\pi}\sum_{n=1}^{\infty}\frac{(-1)^{n-1}}{2n-1}\cos[2\pi f_c t(2n-1)]\right\}$$

Therefore,

$$v_2(t) = [x(t) + E_c\cos((2\pi f_c t)]\left\{\frac{1}{2} + \frac{2}{\pi}\cos(2\pi f_c t) + odd\,harmonics\right\}$$

The odd harmonics in this expression are unwanted, and therefore, are assumed to be eliminated , Hence,

$$v_2(t) = \underbrace{\frac{1}{2}x(t)}_{\text{Modulating Signal}} + \underbrace{\frac{1}{2}E_c\cos(2\pi f_c t) + \frac{2}{\pi}x(t)\cos(2\pi f_c t)}_{\text{AM Wave}} + \underbrace{\frac{2E_c}{\pi}cos^2(2\pi f_c t)}_{\text{Second harmonic of carrier}} \quad (21)$$

In this expression, the first and the fourth terms are unwanted terms whereas the second and third terms together represents the AM wav .Clubbing the second and third terms together , we obtain

$$v_2(t) = \frac{E_c}{2}\left[1 + \frac{4}{\pi E_c}x(t)\right]\cos(2\pi f_c t) + unwanted\,terms \quad (22)$$

This is the required expression for the AM wave with m=[4/πEc] . The unwanted terms can be eliminated using a band-pass filter (BPF) .

AM Spectra and Band Width

The AM signal has three frequency components, Carrier, Upper Sideband and lower side Band

Bandwidth of AM

The bandwidth of a complex signal like AM is the difference between its highest and lowest frequency components and is expressed in Hertz (Hz). Bandwidth deals with only frequencies.

As shown in the following figure 1.1.12

Bandwidth = **(fc – fm) – (fc + fm) = 2 fm**

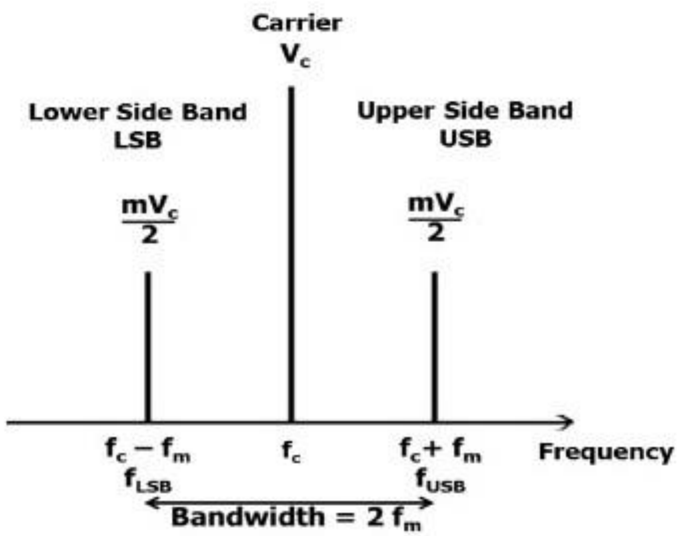


Figure 1.1.12 AM Spectra and Bandwidth

Diagram Source : elprocus.com

COMPARISION OF VARIOUS AM:

PARAMETER	VSB - SC	SSB - SC	DSB-SC
Definition	A vestigial sideband (in radio communication) is a sideband that has been only partly cut off or suppressed.	Single-sideband modulation (SSB) is a refinement of amplitude modulation that more efficiently uses electrical power and bandwidth.	In radio communications, asidebandis a band of frequencies hig her than or lower thanthe carrier frequency, containing power as a result of the modulation process.
Application	Tv broadcastings & Radio broadcastings	Tv broadcastings & ShortwaveRadio broadcastings	Tv broadcastings & Radio broadcastings Garage door opens keyless remotes
Uses	Transmits TV signals	Short wave radio communications	Two way radio communications.

AM APPLICATION & ITS USES:

- ✓ Radio broadcastings
- ✓ Tv broadcastings
- ✓ Garage door opens keyless remotes
- ✓ Transmits TV signals
- ✓ Short wave radio communications
- ✓ Two way radio communication.

Double-sideband suppressed-carrier transmission (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.

DSB-SC Spectrum

DSB-SC is basically an amplitude modulation wave without the carrier, therefore reducing power waste, giving it a 50% efficiency. This is an increase compared to normal AM transmission (DSB), which has a maximum efficiency of 33.333%, since 2/3 of the power is in the carrier which carries no intelligence, and each sideband carries the same information. Single Side Band (SSB) Suppressed Carrier is 100% efficient is shown in Figure 1.2.1 .

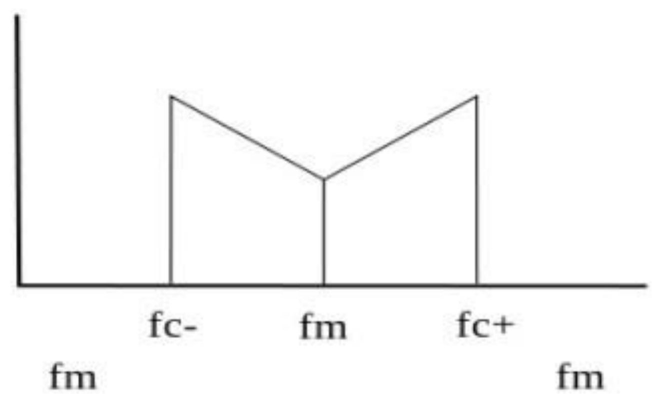


Figure 1.2.1 Spectrum plot of an DSB-SC Signal

Diagram Source : Electronic Tutorials

Generation:

DSB-SC is generated by a mixer. This consists of a message signal multiplied by a carrier signal. The mathematical representation of this process is shown below, where the product-to-sum trigonometric identity is used. Figure shows the 1.2.2 Generation of DSB Signals

$$\underbrace{V_m \cos(\omega_m t)}_{\text{Message}} \times \underbrace{V_c \cos(\omega_c t)}_{\text{Carrier}} = \underbrace{\frac{V_m V_c}{2}\left[\cos((\omega_m + \omega_c)t) + \cos((\omega_m - \omega_c)t)\right]}_{\text{Modulated Signal}}$$

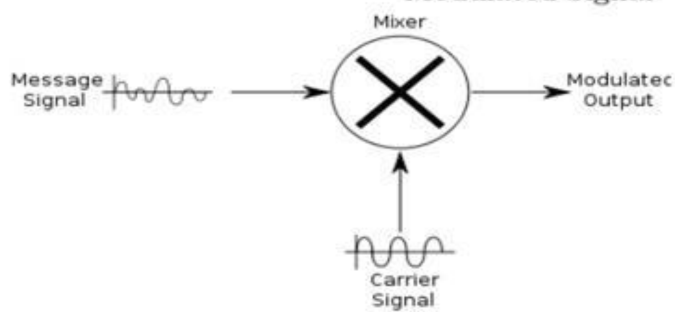


Figure 1.2.2 Generation of DSB Signals

Diagram Source : Electronic Tutorials

Demodulation:

Demodulation is done by multiplying the DSB-SC signal with the carrier signal just like the modulation process. This resultant signal is then passed through a low pass filter to produce a scaled version of original message signal. DSB-SC can be demodulated if modulation index is less than unity.

$$\overbrace{\frac{V_m V_c}{2}\left[\cos((\omega_m + \omega_c)t) + \cos((\omega_m - \omega_c)t)\right]}^{\text{Modulated Signal}} \times \overbrace{V_c' \cos(\omega_c t)}^{\text{Carrier}}$$

$$= \left(\frac{1}{2}V_c V_c'\right)\underbrace{V_m \cos(\omega_m t)}_{\text{original message}} + \frac{1}{2}V_c V_c' V_m \left[\cos((\omega_m + 2\omega_c)t) + \cos((\omega_m - 2\omega_c)t)\right]$$

The equation above shows that by multiplying the modulated signal by the carrier signal, the result is a scaled version of the original message signal plus a second term. Since , ωc>>ωm wthis second term is much higher in frequency than the original message. Once this signal passes through a low pass filter, the higher frequency component is removed, leaving just the original message.

Distortion and Attenuation:

For demodulation, the demodulation oscillator's frequency and phase must be exactly the same as modulation oscillator's, otherwise, distortion and/or attenuation will occur.

To see this effect, take the following conditions:

- Message signal to be transmitted: $f(t)$
- Modulation (carrier) signal: $V_c \cos(\omega_c)$
- Demodulation signal (with small frequency and phase deviations from the modulation signal): $V_c' \cos[(\omega_c + \Delta\omega)t + \theta]$

The resultant signal can then be given by

$$f(t) \times V_c \cos(\omega_c) \times V_c' \cos[(\omega_c + \Delta\omega)t + \theta]$$

$$= \frac{1}{2} V_c V_c' f(t) \cos(\Delta\omega \cdot t + \theta) + \frac{1}{2} V_c V_c' f(t) \cos[(2\omega_c + \Delta\omega)t + \theta]$$

$$\xrightarrow{\text{After low pass filter}} \frac{1}{2} V_c V_c' f(t) \cos(\Delta\omega \cdot t + \theta)$$

The $\cos(\Delta\omega \cdot t + \theta)$ terms results in distortion and attenuation of the original message signal. In particular, $\Delta\omega \cdot t$ contributes to distortion while θ adds to the attenuation.

In the process of Amplitude Modulation, 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 Double Sideband Full Carrier system or simply DSBFC. It is plotted as shown in the following figure 1.2.3.

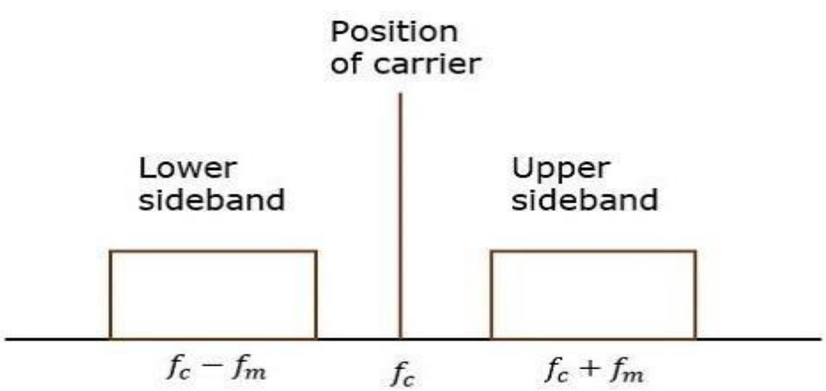


Figure 1.2.3 Spectrum of Double Sideband Full Carrier system

Diagram Source : Brain Kart

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 Double Sideband Suppressed Carrier system or simply DSBSC. It is plotted as shown in the following figure 1.2.4.

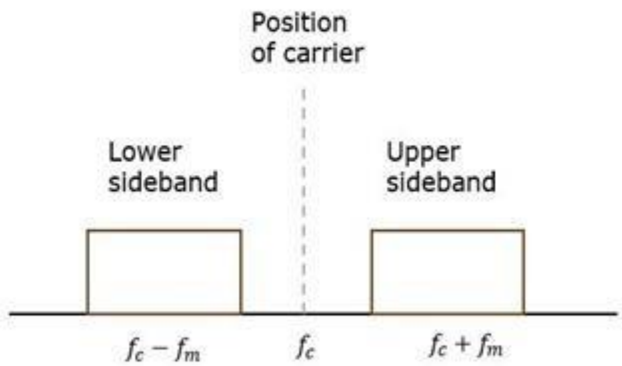


Figure 1.2.4 Carrier is suppressed and side bands are allowed for transmission

Diagram Source : Electronic Tutorials

Mathematical Expressions

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

i.e., Modulating signal

$$m(t)=Amcos(2\pi fmt)m(t) \quad (1)$$

Carrier signal

$$c(t)=Accos(2\pi fct) \quad (2)$$

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

$$s(t)=m(t)c(t)s(t)$$

$$s(t)=AmAccos(2\pi fmt)cos(2\pi fct) \quad (3)$$

Bandwidth of DSBSC Wave

We know the formula for bandwidth (BW) is

$$BW = fmax - fmin$$

$$BW = fmax - fmin$$

Consider the equation of DSBSC modulated wave.

$$s(t) = A_m A_c \cos(2\pi f_m t)\cos(2\pi f_c t) \quad (4)$$

$$s(t) = AmAc2\cos[2\pi(fc+fm)t] + AmAc2\cos[2\pi(fc-fm)t]$$

The DSBSC modulated wave has only two frequencies. So, the maximum and minimum frequencies are fc+fmfc+fm and fc−fmfc−fm respectively.

i.e.,

$f_{max} = f_c + f_m$ and

$f_{min} = f_c - f_m$

Substitute, fmax and fmin values in the bandwidth formula.

$$BW = fc + fm - (fc - fm)$$

$$BW = fc + fm - (fc - fm)$$

Band Width (BW) $= 2f_m$

Thus, the bandwidth of DSBSC wave is same as that of AM wave and it is equal to twice the frequency of the modulating signal.nConsider the following equation of DSBSC modulated wave.

$$s(t) = A_m A_c/2\cos[2\pi(fc+fm)t] + AmAc/2\cos[2\pi(fc-fm)t]$$

Power of DSBSC wave is equal to the sum of powers of upper sideband and lower sideband frequency components.

Pt=PUSB+PLSB

We know the standard formula for power of cos signal is

$$P = (Vrms/\sqrt{2})^2/R$$

$$= (Vm/\sqrt{2})^2 / R$$

First, let us find the powers of upper sideband and lower sideband one by one.

Upper sideband power

$$P_{USB}=(A_m A_c/2/\sqrt{2})^2/ R$$
$$=A_m^2 A_c^2/8R \quad (5)$$

Similarly, we will get the lower sideband power same as that of upper sideband power.

$$PUSB=Am^2Ac^2/8R$$

Now, let us add these two sideband powers in order to get the power of DSBSC wave.

$$P_t=Am^2Ac^2/8R+Am^2Ac^2/8R$$
$$Pt=Am^2Ac^2/4R \quad (6)$$

Therefore, the power required for transmitting DSBSC wave is equal to the power of both the sidebands.

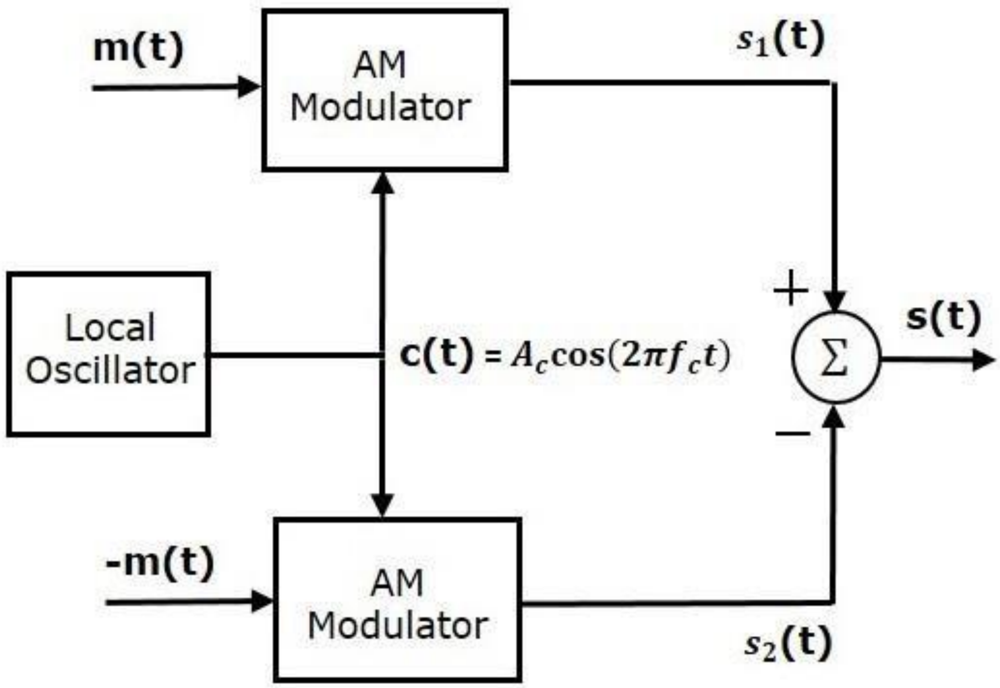
The following two modulators generate DSBSC wave.

- ✓ Balanced modulator
- ✓ Ring modulator

Balanced Modulator

Following is the block diagram of the Balanced modulator shown in figure. 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 $c(t)=A_c\cos(2\pi fct)$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)m(t) with opposite polarity, i.e., -m(t) is applied as another input to the lower AM modulator.



Output of the upper AM modulator is

$$s_1(t)=A_c[1+k_a m(t)]\cos(2\pi f_c t)$$

Output of the lower AM modulator is

$$s_2(t)=A_c[1-k_a m(t)]\cos(2\pi f_c t)$$

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 $s_2(t)$.

$$s(t)=Ac[1+kam(t)]Cos(2\pi fct)-Ac[1-kam(t)]Cos(2\pi fct)$$

$$s(t)=Accos(2\pi fct) + Ackam(t)Cos(2\pi fct)-AcCos(2\pi fct)+ Ackam(t)Cos(2\pi fct) \quad (7)$$

$$s(t)=2Ackam(t)cos(2\pi fct)$$

We know the standard equation of DSBSC wave is

$$s(t)=A_c m(t)\cos(2\pi f_c t) \quad (8)$$

By comparing the output of summer block with the standard equation of DSBSC wave, we will get the scaling factor as 2ka.

Ring Modulator

Following is the block diagram of the Ring modulator shown in Figure 1.2.5.

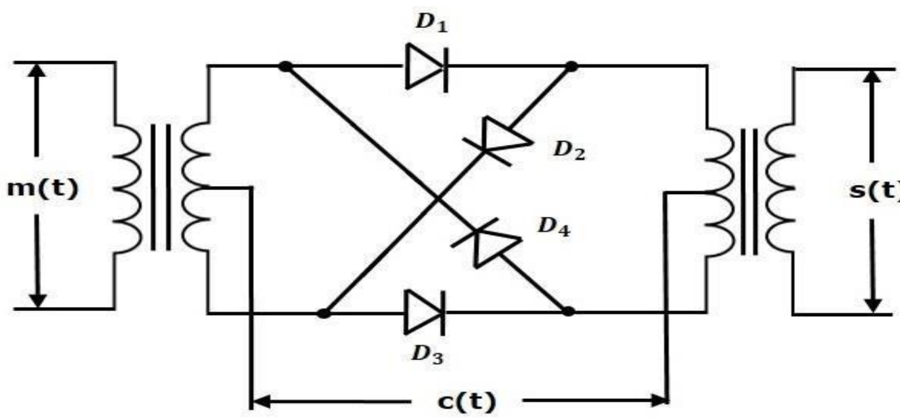


Figure 1.2.5 Ring Modulator

Diagram Source : Electronic Tutorials

In this diagram, the four diodes D1,D2,D3 and D4 are connected in the ring structure. Hence, this modulator is called as the ring modulator. Two center tapped transformers are used in this diagram. The message signal m(t)m(t) is applied to the input transformer. Whereas, the carrier signals c(t)c(t) is applied between the two center tapped transformers.

For positive half cycle of the carrier signal, the diodes D1 and D3 are switched ON and the other two diodes D2 and D4 are switched OFF. In this case, the message signal is multiplied by +1.

For negative half cycle of the carrier signal, the diodes D2D2 and D4D4 are switched ON and the other two diodes D1 and D3 are switched OFF. In this case, the message signal is multiplied by -1. This results in 180 degree phase shift in the resulting DSBSC wave. From the above analysis, we can say that the four diodes D1, D2, D3 and D4 are controlled by the carrier signal.

We will get DSBSC wave s(t)s(t), which is just the product of the carrier signal c(t)c(t) and the message signal m(t)m(t) i.e.,

$$s(t) = \frac{4}{\pi}\sum_{n=1}^{\infty}\frac{(-1)^{n-1}}{2n-1}\cos[2\pi f_c t(2n-1)]m(t) \quad (9)$$

The above equation represents DSB-SC wave, which is obtained at the output transformer of the ring modulator. DSBSC modulators are also called as product modulators as they produce the output, which is the product of two input signals.

HILBERT TRANSFORM

In Digital Signal Processing we often need to look at relationships between real and imaginary parts of a complex signal. These relationships are generally described by Hilbert transforms. Hilbert transform not only helps us relate the I and Q components but it is also used to create a special class of causal signals called analytic which are especially important in simulation. The analytic signals help us to represent band pass signals as complex signals which have specially attractive properties for signal processing. The role of Hilbert transform is to take the carrier which is a cosine wave and create a sine wave out of it. Now recall that the Fourier Series is written as

$$x(t) = \sum_{n=-\infty}^{n=+\infty} C_n e^{jn\omega t} \qquad (1)$$

Where $C_n = A_n + jB_n$ and $C_{-n} = A_n - jB_n$

Here An and Bn are the spectral amplitudes of cosine and sine waves. Now take a look at the phase spectrum. The phase spectrum is computed by

$$\Theta = \tan^{-1}(Bn/An) \qquad (2)$$

Cosine wave has no sine spectral content, so Bn is zero. The phase calculated is 90° for both positive and negative frequency from above formula. The wave has two spectral components each of magnitude A/2, both positive and lying in the real plane. Real plane is described as that passing vertically (R-V plane) and the Imaginary plane as one horizontally (R-I plane) through the Imaginary axis) Now compare Figure 1.4.1, in particular the spectral amplitudes. The cosine spectral amplitudes are both positive and lie in the real plane. The sine wave has spectral components that lie in the Imaginary plane and are of opposite sign. Now we wish to convert the cosine wave to a sine wave. There are two ways of doing that,

- ✓ Time domain
- ✓ Frequency domain.

To turn cosine into sine, we need to rotate the negative frequency component of the cosine by +90° and the positive frequency component by -90°. We will need to rotate the +Q phasor by-90° or in other words multiply it by -j. We also need to rotate the -Q phasor by +90° or multiply it by j. We can describe this transformation process called the Hilbert Transform as follows: All negative frequencies of a signal get a +90° phase shift and all positive frequencies get a -90° phase shift.

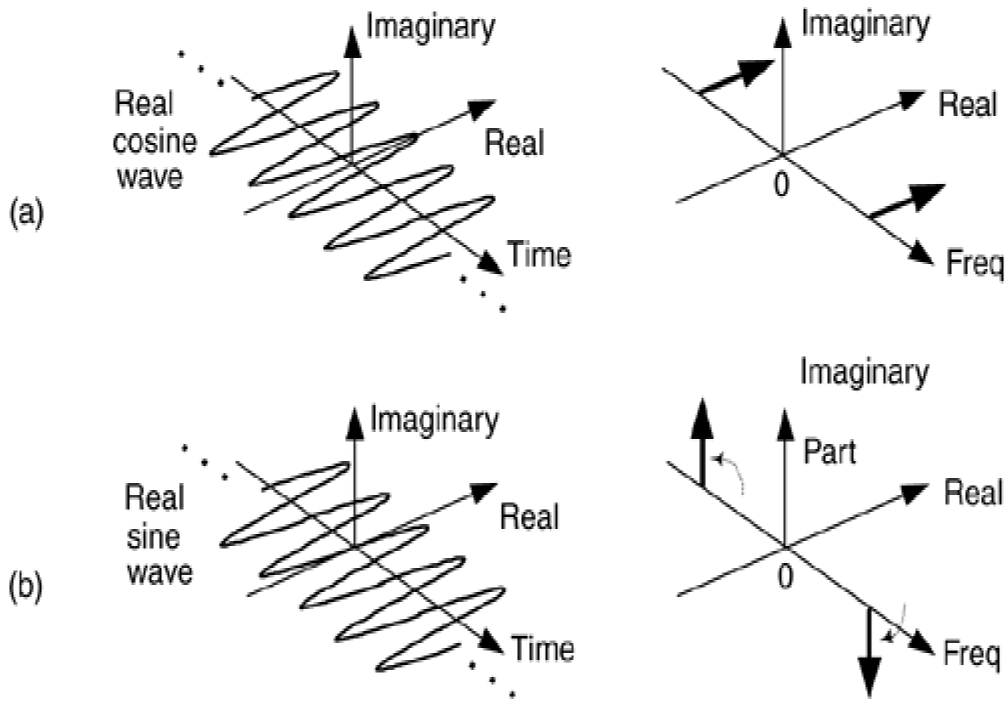


Figure 1.4.1 Hilbert Transform Cosine and Sine Spectral Amplitudes

Diagram Source Flylib.com

A negative cosine will come out a negative sine wave and one more transformation will return it to the original cosine wave, each time its phase being changed by 90°. For this reason Hilbert transform is also called a "quadrature filter".A peculiar sort of filter that changes the phase of the spectral components depending on the sign of their frequency. It only effects the phase of the signal. It has no effect on the amplitude at all.

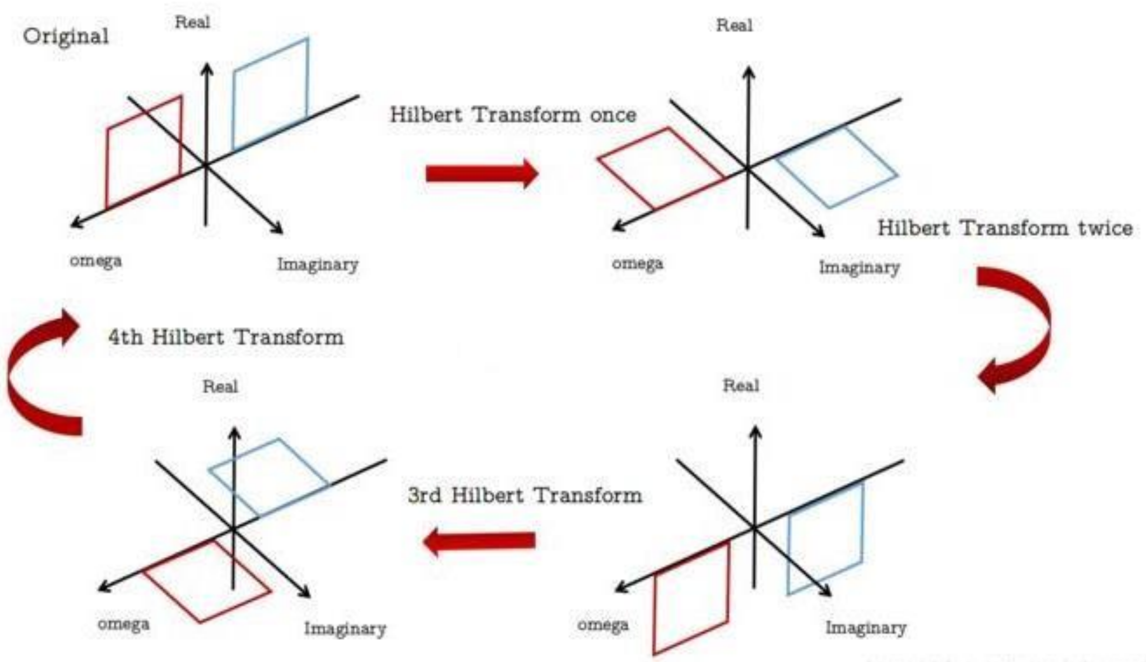


Figure 1.4.2 Hilbert Transform Process

Diagram Source Programmer sought

Hilbert transform of a signal x(t) is defined as the transform in which phase angle of all components of the signal is shifted by ±90o Hilbert transform of x(t) is represented with x^(t) shown in figure 1.4.2 it is given by,

$$\hat{x}(t) = \frac{1}{\pi} \int_{-\infty}^{\infty} \frac{x(s)}{t-s} ds, \quad (3)$$

where the integral is the Cauchy principal value integral. The reconstruction formula

$$x(t) = -\frac{1}{\pi} \int_{-\infty}^{\infty} \frac{\hat{x}(s)}{t-s} ds, \quad (4)$$

defines the Hilbert inverse transform. X(t), x^(t) is called a Hilbert transform pair.

Hilbert transformer:

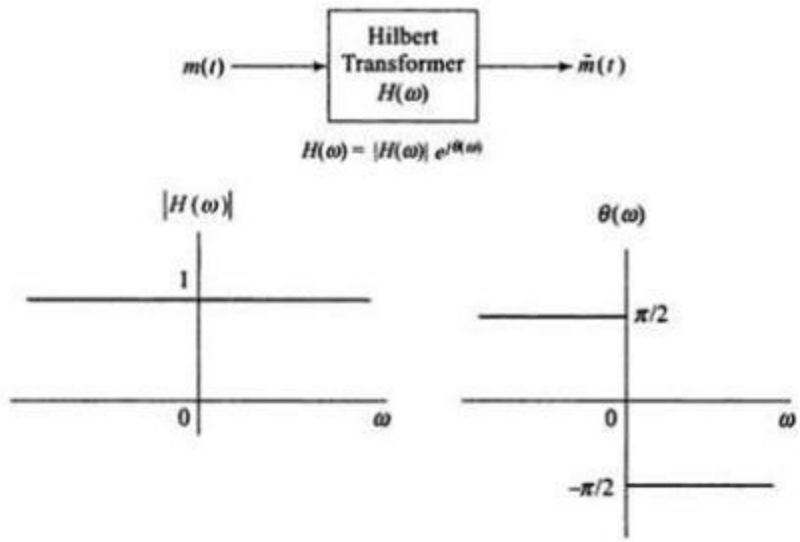


Figure 1.4.3 Magnitude and Phase Characteristics of Hilbert Transform

Diagram Source: Electronics Post

A Hilbert transformer produces a -90 degree phase shift for the positive frequency components of the input x(t), the amplitude doesn't change shown in figure 1.4.3.

Properties of the Hilbert transform:

A signal x(t) and its Hilbert transform $\hat{x}(t)$ have

1. the same amplitude spectrum
2. the same autocorrelation function
3. x(t) and $\hat{x}(t)$ are orthogonal
4. The Hilbert transform of $\hat{x}(t)$ is -x(t)

Pre envelope:

The pre envelope of a real signal x(t) is the complex function

$$x_+(t) = x(t) + j\,\hat{x}(t).$$

The pre envelope is useful in treating band pass signals and systems. This is due to the result

$$X_+(v) = \begin{cases} 2\,X(v), & v > 0 \\ X(0), & v = 0 \\ 0, & v < 0 \end{cases}$$

Complex envelope:

The complex envelope of a band pass signal x(t) is

$$\tilde{x}\ (t) = x_+(t)\, e^{-j2\pi fct}.$$

Properties of the Hilbert Transform

A signal x(t) and its Hilbert transform x^(t) have

- The same amplitude spectrum.
- The same autocorrelation function.
- The energy spectral density is same for both x(t) and x^(t).
- x(t) and x^(t) are orthogonal.
- The Hilbert transform of x^(t) is -x(t)
- If Fourier transform exist then Hilbert transform also exists for energy and power signals.

Single -sideband suppressed-carrier transmission (SSB-SC)

The DSBSC modulated signal has two sidebands. Since, the two sidebands carry the same information, there is no need to transmit both sidebands. We can eliminate one sideband.

The process of suppressing one of the sidebands along with the carrier and transmitting a single sideband is called as Single Sideband Suppressed Carrier system or simply SSBSC. It is plotted as shown in the following figure 1.3.1.

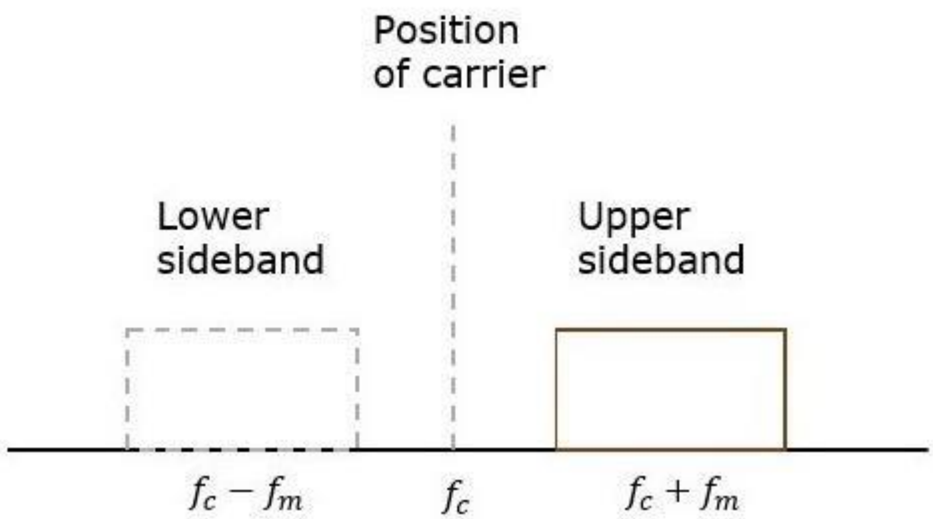


Fig 1.3.1 Carrier and Side band are Suppressed and a single side band is allowed for transmission,

Diagram Source Brain Kart

In the above figure, the carrier and the lower sideband are suppressed. Hence, the upper sideband is used for transmission. Similarly, we can suppress the carrier and the upper sideband while transmitting the lower sideband.

This SSBSC system, which transmits a single sideband has high power, as the power allotted for both the carrier and the other sideband is utilized in transmitting this Single Sideband.

Mathematical Expressions

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

$$m(t)=A_m\cos(2\pi f_m t) \qquad (1)$$

Carrier signal

$$c(t)=A_c\cos(2\pi f_c t) \quad (2)$$

Mathematically, we can represent the equation of SSBSC wave as

s(t)=AmAc2cos[2π(fc+fm)t] for the upper sideband (3)

s(t)=AmAc2cos[2π(fc−fm)t] for the lower sideband

Bandwidth of SSBSC Wave

We know that the DSBSC modulated wave contains two sidebands and its bandwidth is 2fm2fm. Since the SSBSC modulated wave contains only one sideband, its bandwidth is half of the bandwidth of DSBSC modulated wave.

i.e., Bandwidth of SSBSC modulated wave =2fm/2

$$=f_m$$

Therefore, the bandwidth of SSBSC modulated wave is fmfm and it is equal to the frequency of the modulating signal.

Power Calculations of SSBSC Wave

Consider the following equation of SSBSC modulated wave.

s(t)=AmAc2cos[2π(fc+fm)t]for the upper sideband Or

s(t)=AmAc2cos[2π(fc−fm)t]]for the lower sideband

Power of SSBSC wave is equal to the power of any one sideband frequency components.

Pt=PUSB=PLSB

We know that the standard formula for power of cos signal is

$$P=v_{rms}^2/R=(v_m/\sqrt{2})^2/R$$

In this case, the power of the upper sideband is

$$P_{USB}=(A_mA_c/2/\sqrt{2})^2/R$$
$$=A_m^2A_c^2/8R$$

Therefore, the power of SSBSC wave is

$$P_t=P_{USB}=P_{LSB}=A_m^2A_c^2/8R$$

SSB TRANSMISSION:

There are two methods used for SSB Transmission.

1. Filter Method
2. Phase Shift Method
3. Block diagram of SSB

Filter Method:

This is the filter method of SSB suppression for the transmission. Figure 1.3.2

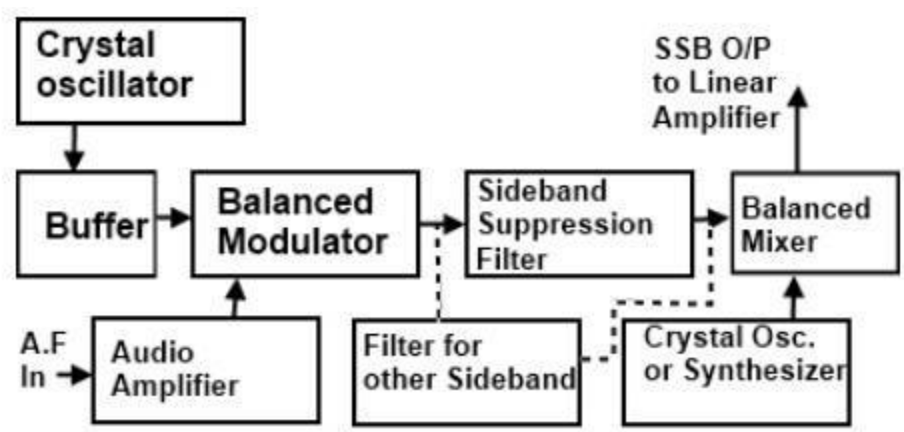


Fig 1.3.2 Filter Method

Diagram Source Electronic tutorials

1. A crystal controlled master oscillator produces a stable carrier frequency fc (say 100 KHz)
2. This carrier frequency is then fed to the balanced modulator through a buffer amplifier which isolates these two stages.
3. The audio signal from the modulating amplifier modulates the carrier in the balanced modulator. Audio frequency range is 300 to 2800 Hz. The carrier is also suppressed in this stage but allows only to pass the both side bands. (USB & LSB).
4. A band pass filter (BPF) allows only a single band either USB or LSB to pass through it. It depends on our requirements.

5. This side band is then heterodyned in the balanced mixer stage with 12 MHz frequency produced by crystal oscillator or synthesizer depends upon the requirements of our transmission. So in mixer stage, the frequency of the crystal oscillator or synthersizer is added to SSB signal. The output frequency thus being raised to the value desired for transmission.
6. Then this band is amplified in driver and power amplifier stages and then fed to the aerial for the transmission.

Phase Shift Method:

The phasing method of SSB generation uses a phase shift technique that causes one of the side bands to be canceled out. A block diagram of a phasing type SSB generator is shown in fig 1.3.3.

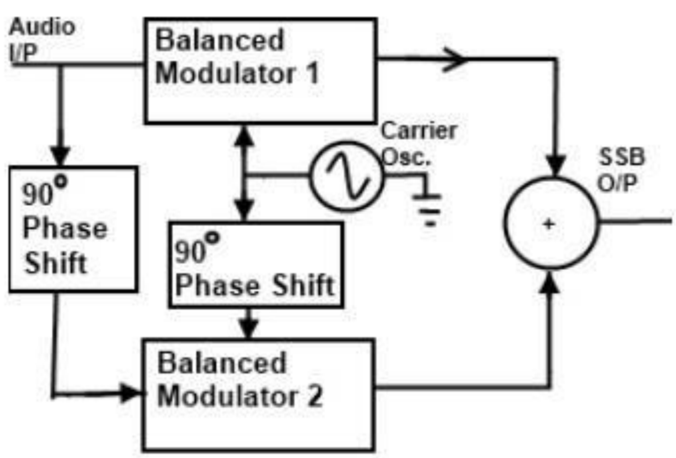


Fig 1.3.3 Phase Shift Method

Diagram Source Electronic tutorials

It uses two balanced modulators instead of one. The balanced modulators effectively eliminate the carrier. The carrier oscillator is applied directly to the upper balanced modulator along with the audio modulating signal. Then both the carrier and modulating signal are shifted in phase by 90o and applied to the second, lower, balanced modulator. The two balanced modulator output are then added together algebraically. The phase shifting action causes one side band to be canceled out when the two balanced modulator outputs are combined.

Block diagram of SSB:

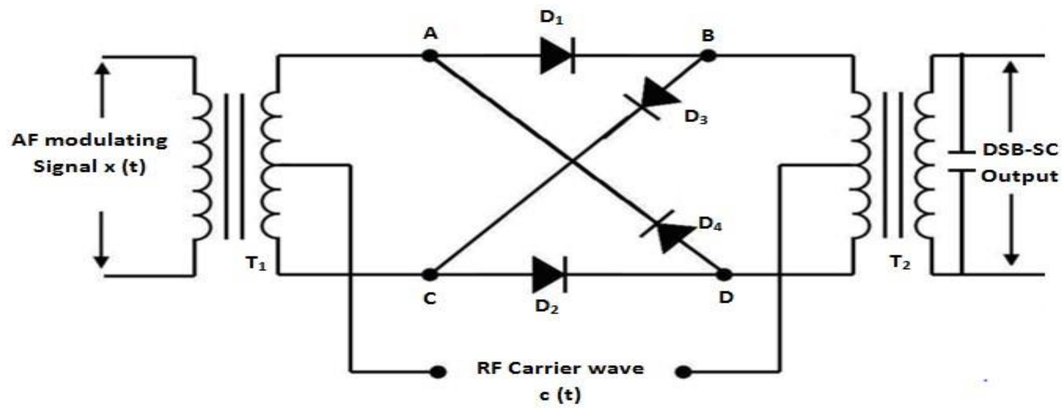


Fig 1.3.4 Balanced Ring Modulator

Diagram Source Electronic Post

Operation of Balance Ring Modulator:

Ring modulation is a signal-processing function in electronics, an implementation of amplitude modulation or frequency mixing, performed by multiplying two signals, where one is typically a sine-wave or another simple waveform is shown in figure 1.3.4. It is referred to as "ring" modulation because the analog circuit of diodes originally used to implement this technique took the shape of a ring. This circuit is similar to a bridge rectifier, except that instead of the diodes facing "left" or "right", they go "clockwise" or "anti-clockwise". A ring modulator is an effects unit working on this principle.

The carrier, which is AC, at a given time, makes one pair of diodes conduct, and reverse-biases the other pair. The conducting pair carries the signal from the left transformer secondary to the primary of the transformer at the right. If the left carrier terminal is positive, the top and bottom diodes conduct. If that terminal is negative, then the "side" diodes conduct, but create a polarity inversion between the transformers. This action is much like that of a DPDT switch wired for reversing connections. Ring modulators frequency mix or heterodyne two waveforms, and output the sum and difference of the frequencies present in each waveform. This process of ring modulation produces some signal rich in partials. As well, neither the carrier nor the incoming signal is prominent in the outputs, and ideally, not at all.

Two oscillators, whose frequencies were harmonically related and ring modulated against each other, produce sounds that still adhere to the harmonic partials of the notes, but contain a very different spectral make up. When the oscillators' frequencies are not harmonically related, ring modulation creates inharmonic, often producing bell-like or otherwise metallic sounds.

If the same signal is sent to both inputs of a ring modulator, the resultant harmonic spectrum is the original frequency domain doubled (if $f1 = f2 = f$, then $f2 - f1 = 0$ and $f2 + f1 = 2f$). Regarded as multiplication, this operation amounts to squaring. However, some distortion occurs due to the forward voltage drop of the diodes.

Some modern ring modulators are implemented using digital signal processing techniques by simply multiplying the time domain signals, producing a nearly-perfect signal output. Before digital music synthesizers became common, at least some analog synthesizers (such as the ARP 2600) used analog multipliers for this purpose; they were closely related to those used in electronic analog computers. (The "ring modulator" in the ARP 2600 could multiply control voltages; it could work at DC.)

Multiplication in the time domain is the same as convolution in the frequency domain, so the output waveform contains the sum and difference of the input frequencies. Thus, in the basic case where two sine waves of frequencies f1 and f2 ($f1 < f2$) are multiplied, two new sine waves are created, with one at $f1 + f2$ and the other at $f2 - f1$. The two new waves are unlikely to be harmonically related and (in a well designed ring modulator) the original signals are not present. It is this that gives the ring modulator its unique tones.

Inter modulation products can be generated by carefully selecting and changing the frequency of the two input waveforms. If the signals are processed digitally, the frequency-domain convolution becomes circular convolution. If the signals are wideband, this will cause aliasing distortion, so it is common to oversample the operation or low-pass filter the signals prior to ring modulation.

One application is spectral inversion, typically of speech; a carrier frequency is chosen to be above the highest speech frequencies (which are low-pass filtered at, say, 3 kHz, for a carrier of perhaps 3.3 kHz), and the sum frequencies from the modulator are removed by more low-pass filtering. The remaining difference frequencies have an inverted spectrum - High frequencies become low, and vice versa.

Advantages

- ✓ Bandwidth or spectrum space occupied is lesser than AM and DSBSC waves.
- ✓ Transmission of more number of signals is allowed.
- ✓ Power is saved.
- ✓ High power signal can be transmitted.
- ✓ Less amount of noise is present.
- ✓ Signal fading is less likely to occur.

Disadvantages

- ✓ The generation and detection of SSBSC wave is a complex process.
- ✓ The quality of the signal gets affected unless the SSB transmitter and receiver have an excellent frequency stability.

Applications

- ✓ For power saving requirements and low bandwidth requirements.
- ✓ In land, air, and maritime mobile communications.
- ✓ In point-to-point communications.
- ✓ In radio communications.
- ✓ In television, telemetry, and radar communications.
- ✓ In military communications, such as amateur radio, etc.

Super heterodyne Receiver

The non uniform selectivity of the TRF led to the development of the super heterodyne receiver near the end of World War I. Although the quality of the super heterodyne receiver has improved greatly since its original design, its basic configuration has not changed much, and it is still used today for a wide variety of radio communications services. The super heterodyne receiver has remained in use because its gain, selectivity, and sensitivity characteristics are superior to those of other receiver configurations. Heterodyne means to mix two frequencies together in a nonlinear device or to translate one frequency to another using nonlinear mixing. A block diagram of a non-coherent super heterodyne receiver is shown in figure 1.5.1. Essentially, there are five sections to a super heterodyne receiver: the RF section, the mixer/converter section, the IF section, the audio detector section, and the audio amplifier section.

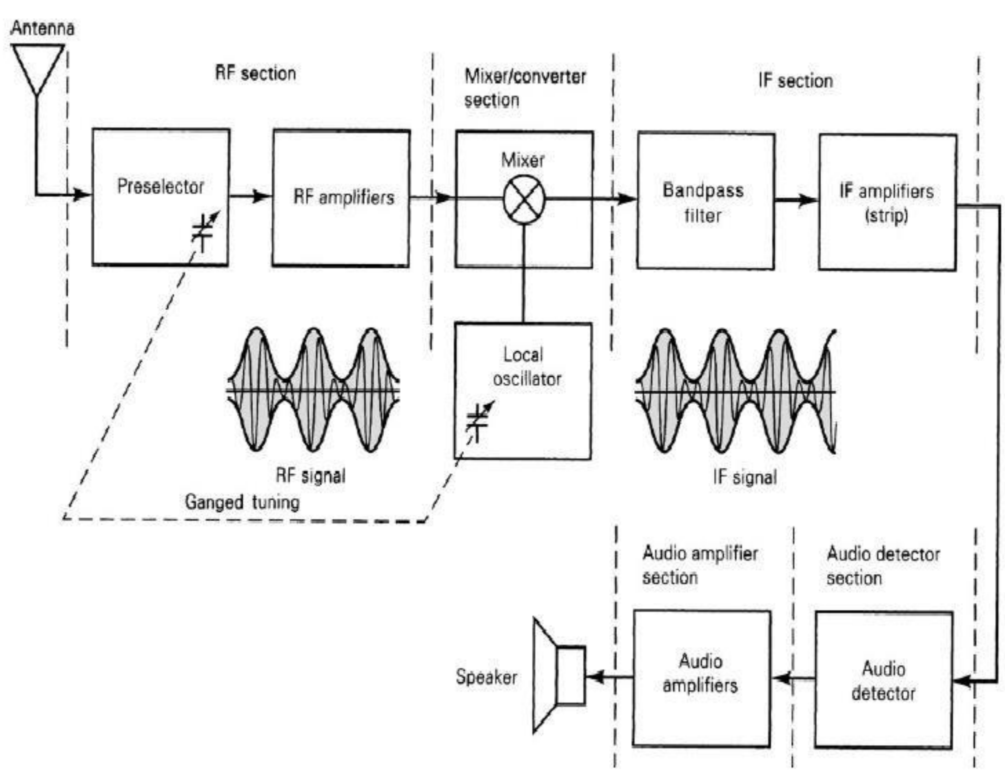


Fig 1.5.1 Superhetrodyne Receiver

Diagram Source Circuit Digest.com

Super heterodyne Principle

In the Super heterodyne Principle, the incoming signal voltage is combined with a signal generated in the receiver. This local oscillator voltage is normally converted into a signal of a lower fixed frequency. The signal at this intermediate frequency contains the same modulation as the original carrier, and it is now amplified and detected to reproduce the original information. The superhet has the same essential components as the TRF receiver, in addition to the mixer, local oscillator and intermediate-frequency (IF) amplifier.

A constant frequency difference is maintained between the local oscillator and the RF circuits; normally through capacitance tuning, in which all the capacitors are ganged together and operated in unison by one control knob. The IF amplifier generally uses two or three transformers, each consisting of a pair of mutually coupled tuned circuits. With this large number of double-tuned circuits operating at a constant, specially chosen frequency, the IF amplifier provides most of the gain (and therefore sensitivity) and bandwidth requirements of the receiver. Since the characteristics of the IF amplifier are independent of the frequency to which the receiver is tuned, the selectivity and sensitivity of the superhet are usually fairly uniform throughout its tuning range and not subject to the variations that affect the TRF receiver. The RF circuits are now used mainly to select the wanted frequency, to reject interference such as the image frequency and (especially at high frequencies) to reduce the noise figure of the receiver.

For further explanation of the Superheterodyne Principle, refer to Figure 6 -2. The RF stage is normally a wideband RF amplifier tunable from approximately 540 kHz to 1650 kHz (standard commercial AM band). It is mechanically tied to the local oscillator to ensure precise tuning characteristics. The local oscillator is a variable oscillator capable of generating a signal from 0.995 MHz to 2.105 MHz. The incoming signal from the transmitter is selected and amplified by the RF stage. It is then combined (mixed) with a predetermined local oscillator signal in the mixer stage. (During this stage, a class C nonlinear device processes the signals, producing the sum, difference, and originals.)

The signal from the mixer is then supplied to the IF (intermediate-frequency) amplifier. This amplifier is a very-narrow-bandwidth class A device capable of selecting a frequency of 0.455 kHz ± 3 kHz and rejecting all others.

The IF signal output is an amplified composite of the modulated RF from the transmitter in combination with RF from the local oscillator. Neither of these signals is usable without further processing. The next process is in the detector stage, which eliminates one of the sidebands still present and separates the RF from the audio components of the other sideband. The RF is filtered to ground, and audio is supplied or fed to the audio stages for amplification and then to the speakers, etc.

The following example shows the Super heterodyne Receiver tuning process:

1. Select an AM station, i.e., 640 kHz.
2. Tune the RF amplifier to the lower end of the AM band.
3. Tune the RF amplifier. This also tunes the local oscillator to a predetermined frequency of 1095 kHz.
4. Mix the 1095 kHz and 640 kHz. This produces the following signals at the output of the mixer circuit; these signals are then fed to the IF amplifier:

- 1.095-MHz local oscillator frequency
- 640-kHz AM station carrier frequency
- 445-kHz difference frequency
- 1.735-MHz sum frequency

Because of its narrow bandwidth, the IF amplifier rejects all other frequencies but 455 kHz. This rejection process reduces the risk of interference from other stations. This selection process is the key to the superheterodyne's exceptional performance, which is why it is widely accepted The process of tuning the local oscillator to a predetermined frequency for each station throughout the AM band is known as tracking. A simplified form of the Super heterodyne Principle is also in existence, in which the mixer output is in fact audio. Such a direct conversion receiver has been used by amateurs, with good results.

RF section

The RF section generally consists of a pre selector and an amplifier stage. They can be separate circuits or a single combined circuit. The pre selector is a broad-tuned band pass filter with an adjustable center frequency that is tuned to the desired carrier frequency. The primary purpose of the pre selector is to provide enough initial band limiting to prevent a specific unwanted radio frequency, called the image frequency, from entering the receiver. The pre selector also reduces the noise bandwidth of the receiver and provides the initial step toward reducing the overall receiver bandwidth to the minimum bandwidth required to pass the information signals. The RF amplifier determines the sensitivity of the receiver (i.e., sets the signal threshold). Also, because the RF amplifier is the first active device encountered by a received signal, it is the primary contributor of noise and, therefore, a predominant factor in determining the noise figure for the receiver. A receiver can have one or more RF amplifiers, or it may not have any, depending on the desired sensitivity. Several advantages of including RF amplifiers in a receiver are as follows:

1. Greater gain, thus better sensitivity
2. Improved image-frequency rejection
3. Better signal-to-noise ratio
4. Better selectivity

Mixer/converter section

The mixer/converter section includes a radio-frequency oscillator stage (commonly called a local oscillator) and a mixer/converter stage (commonly called the first detector). The local oscillator can be any of the oscillator circuits, depending on the stability and accuracy desired. The mixer stage is a nonlinear device and its purpose is to convert radio frequencies to intermediate frequencies (RF-to-IF frequency translation). Heterodyning takes place in the mixer stage, and radio frequencies are down-converted to intermediate frequencies. Although the carrier and sideband frequencies are translated from RF to IF, the shape of the envelope remains the same and, therefore, the original information contained in the envelope remains unchanged. It is important to note that although the carrier and upper and lower side frequencies change frequency, the bandwidth is unchanged by the heterodyning process. The most common intermediate frequency used in AM broadcast-band receivers is 455 kHz.

IF section

The IF section consists of a series of IF amplifiers and bandpass filters and is often called the IF strip. Most of the receiver gain and selectivity is achieved in the IF section. The IF center frequency and bandwidth are constant for all stations and are chosen .so that their frequency is less than any of the RF signals to be received. The IF is always lower in frequency than the RF because it is easier and less expensive to construct high-gain, stable amplifiers for the low frequency signals. Also, low-frequency IF amplifiers are less likely to oscillate than their RF counterparts. Therefore, it is not uncommon to see a receiver with five or six IF amplifiers and a single RF amplifier or possibly no RF amplification.

Detector section

The purpose of the detector section is to convert the IF signals back to the original source information. The detector is generally called an audio detector or the second detector in a broadcast-band receiver because the information signals are audio frequencies. The detector can be as simple as a single diode or as complex as a phase-locked loop or balanced demodulator.

Audio amplifier section

The audio section comprises several cascaded audio amplifiers and one or more speakers. The number of amplifiers used depends on the audio signal power desired.

Tuning:

Broadband tuning is applied to the RF stage. The purpose of this is to reject the signals on the image frequency and accept those on the wanted frequency. It must also be able to track the local oscillator so that as the receiver is tuned, so the RF tuning remains on the required frequency. Typically the selectivity provided at this stage is not high. Its main purpose is to reject signals on the image frequency which is at a frequency equal to twice that of the IF away from the wanted frequency. As the tuning within this block provides all the rejection for the image response, it must be at a sufficiently sharp to reduce the image to an acceptable level. However the RF tuning may also help in preventing strong off-channel signals from entering the receiver and overloading elements of the receiver, in particular the mixer or possibly even the RF amplifier.

Amplification:

In terms of amplification, the level is carefully chosen so that it does not overload the mixer when strong signals are present, but enables the signals to be amplified sufficiently to ensure a good signal to noise ratio is achieved. The amplifier must also be a low noise design. Any noise introduced in this block will be amplified later in the receiver.

Mixer / frequency translator block:

The tuned and amplified signal then enters one port of the mixer. The local oscillator signal enters the other port. The performance of the mixer is crucial to many elements of the overall receiver performance. It should eb as linear as possible. If not, then spurious signals will be generated and these may appear as 'phantom' received signals.

Local oscillator:

The local oscillator may consist of a variable frequency oscillator that can be tuned by altering the setting on a variable capacitor. Alternatively it may be a frequency synthesizer that will enable greater levels of stability and setting accuracy.

Intermediate frequency amplifier, IF block :

Once the signals leave the mixer they enter the IF stages. These stages contain most of the amplification in the receiver as well as the filtering that enables signals on one frequency to be separated from those on the next. Filters may consist simply of LC tuned transformers providing inter-stage coupling, or they may be much higher performance ceramic or even crystal filters, dependent upon what is required.

· **Detector / demodulator stage:** Once the signals have passed through the IF stages of the super heterodyne receiver, they need to be demodulated. Different demodulators are required for different types of transmission, and as a result some receivers may have a variety of demodulators that can be switched in to accommodate the different types of transmission that are to be encountered. Different demodulators used may include:

AM diode detector:

This is the most basic form of detector and this circuit block would simple consist of a diode and possibly a small capacitor to remove any remaining RF. The detector is cheap and its performance is adequate, requiring a sufficient voltage to overcome the diode forward drop. It is also not particularly linear, and finally it is subject to the effects of selective fading that can be apparent, especially on the HF bands.

Synchronous AM detector

This form of AM detector block is used in where improved performance is needed. It mixes the incoming AM signal with another on the same frequency as the carrier. This second signal can be developed by passing the whole signal through a squaring amplifier. The advantages of the synchronous AM detector are that it provides a far more linear demodulation performance and it is far less subject to the problems of selective fading.

SSB product detector:

The SSB product detector block consists of a mixer and a local oscillator, often termed a beat frequency oscillator, BFO or carrier insertion oscillator, CIO. This form of detector is used for Morse code transmissions where the BFO is used to create an audible tone in line with the on-off keying of the transmitted carrier. Without this the carrier without modulation is difficult to detect. For SSB, the CIO re-inserts the carrier to make the modulation comprehensible.

Basic FM detector:

As an FM signal carries no amplitude variations a demodulator block that senses frequency variations is required. It should also be insensitive to amplitude variations as these could add extra noise. Simple FM detectors such as the Foster Seeley or ratio detectors can be made from discrete components although they do require the use of transformers.

PLL FM detector:

A phase locked loop can be used to make a very good FM demodulator. The incoming FM signal can be fed into the reference input, and the VCO drive voltage used to provide the detected audio output.

Quadrature FM detector:

This form of FM detector block is widely used within ICs. IT is simple to implement and provides a good linear output.

Audio amplifier: The output from the demodulator is the recovered audio. This is passed into the audio stages where they are amplified and presented to the headphones or loudspeaker.