

# Double side band suppressed carrier biology essay

[Science](#), [Biology](#)



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### **Double Side Band – Suppressed Carrier**

Consider a modulating signal  $m(t)$ , where  $f_m$  is the modulating frequency.

Consider also a proportional carrier signal where  $f_c$  is the carrier

frequency. Double Side Band Suppressed Carrier modulation occurs simply by multiplication of the carrier. Hence,  $s_{DSB-SC}(t) = m(t) \cos(2\pi f_c t)$ . By the trigonometric identity; ,

Recovery of the message signal  $m(t)$  is done by multiplying the modulated signal  $s_{DSB-SC}(t)$  by a local oscillator. This local oscillator signal should

ideally have the same exact phase and frequency as the carrier signal  $c(t)$ .

Therefore:

- By using the trigonometric formula; Which results in  $\frac{1}{2} m(t)$  and unwanted components which need to be filtered out by a low pass filter. These processes of Double Side band – suppressed carrier modulation and subsequent demodulation are illustrated in Figure 1. FigureAs shown above, DSB-SC modulation is the product of  $m(t)$  and  $c(t)$ . For DSB-SC demodulation, the modulated signal is multiplied by a local oscillator signal and filtered. The time domain signal plots obtained are shown in Figure 2. The modulating signal  $m(t)$  was set to a frequency of 1Hz and a peak to peak amplitude of 1. The carrier signal  $c(t)$  was set to a frequency of 10Hz and a peak to peak amplitude of 1. The local oscillator is equal to the carrier signal. The modulated signal is a signal of 10Hz enveloped within the modulating 1Hz signal. As expected the demodulated signal is  $\frac{1}{2} m(t)$ , with a peak to peak amplitude of 0. 5. Its frequency is 1Hz like the modulating signal, however it contains a slight phase shift which was not predicted in theory. time 1. PNGtime 2. PNGFigureLooking at the frequency domain, Figure 3 shows the spectrum of our modulating signal  $m(t)$  at the frequency of 1Hz and a peak amplitude of 0. 5. Similarly, Figure 4 shows the frequency spectrum of our carrier signal  $c(t)$  at 10Hz. FigureFigureUpon modulation, the frequency spectrum is as shown in Figure 5. There are two sidebands present; a lower sideband and an upper sideband at the frequencies of  $f_c - f_m(10 - 1 = 9\text{Hz})$  and  $f_c + f_m(10 + 1 = 11\text{Hz})$ . There is also no carrier present. Thus the name of Double Sideband- Suppressed Carrier. The magnitudes of these frequencies

should ideally be  $A_m/2$ . Figure 6. As explained in the theory above, the process of demodulating the spectrum shown above back a single frequency at 1Hz, is divided into two parts; mixing and filtering. When the modulated signal was multiplied by the local oscillator the resulting frequency response is as shown in Figure 6. As one can see, the result gives a frequency at 1Hz as required, however there exist also some other unwanted spectra at  $-2f_c$  and  $+2f_c$  which need to be removed. These frequencies are removed from the signal by using a low pass filter which has a  $f_{pass}$  at 4Hz and an  $f_{stop}$  at 6Hz. The final demodulated output signal in the frequency domain is shown in Figure 7. Ideally it should have a magnitude of  $A_m/2$ . Figure 7.

## Double Side Band – Large Carrier

DSB-SC requires a fairly complicated demodulator. This is because it requires a local oscillator which needs to be perfectly synchronised to the carrier signal used in the modulation process. Hence, an alternative approach is to design the modulation such that the modulating signal can be recovered purely from the envelope of the modulated signal. This can be done by making sure the modulated signal is always positive. Consider again a modulating signal  $m(t)$ , where  $m(t) \geq 0$  and also a carrier signal  $c(t) = A_c \cos(2\pi f_c t)$ . Double Side Band – Large Carrier modulation occurs by first multiplying the modulating signal with the carrier signal. Then, the carrier signal with some extra gain  $A_c$  is added to make the modulated signal. Hence, Let  $m_a$  be the modulation index such that  $m(t) = m_a \cos(2\pi f_m t)$ . By using trigonometric identities; From this equation, it can be deduced that the modulated signal should have a frequency spectrum of magnitude  $A_c$  at the carrier frequency  $f_c$ , a lower sideband of magnitude  $\frac{1}{2} m_a$  at  $(f_c - f_m)$ , and similarly an upper sideband of magnitude  $\frac{1}{2} m_a$  at

( $f_c + f_m$ ). Demodulation of DSB-LC signals can be implemented using a simple scheme called envelope detection, provided sufficient carrier power is transmitted. If the gain  $A_c$  is large enough, the amplitude of the modulated waveform given by  $s_{DSB-LC}(t)$  will be proportional to  $m(t)$ . Demodulation in this case simply reduces to the detection of the envelope of a modulated carrier with no dependence of the exact phase or frequency of the carrier. If  $A_c$  is not large enough, then the envelope of  $s_{DSB-LC}(t)$  is not always proportional to  $m(t)$ . Thus the condition for demodulation of double sideband large carrier by an envelope detector is for all  $t$ . In other words, the condition for demodulation by an envelope detector can also be expressed as  $A_c \geq A_m$ . The modulation index,  $m_a = 1$  when  $A_m = A_c$ . When  $m_a > 1$  and the carrier is said to be overmodulated, resulting in envelope distortion. Envelope detection can occur provided there is sufficient power from the carrier. In DSB-SC all the power is in the signal. However with DSB-LC; The average power  $P$  is given by  $P_T = P_c + P_s$ . When  $m_a$  is maximum, Power in Carrier  $P_c$  : Power in Signal  $P_s$ ,  $P_c : P_s = 2 : 1$  and  $P_s : P_T = 1 : (1+2) = 1 : 3$ . Hence, only 33% of the total transmitted power  $P_T$  is the signal power. The process of demodulation can occur in two different ways; either by squaring and then taking the square root of the received signal and passing the resulting signal through a low pass filter which eliminates the re-modulated components. Otherwise by using a rectifier and then passing the resulting signal through a filter. These types of demodulation are synchronous and asynchronous respectively. First, the DSB-LC setup will be analysed having an asynchronous demodulation technique. This setup is shown in Figure 8. Figure Next, time domain signal plots are shown in Figure 9. The modulating signal  $m(t)$  is set to have an

amplitude of 1 peak to peak and a frequency of 1Hz. The carrier signal  $c(t)$  is set to have a 1 peak to peak amplitude and a frequency of 10Hz. In this way amplitudes of the signals are equal such that  $m_a = A_m/A_c$  is at its maximum possible. This is the limiting condition because the envelope of the modulated signal touches 0 as shown in the 3rd time domain plot. For the process of demodulation used above, the signal  $s_{\text{DSSB-LC}}(t)$  is rectified and hence the part of the signal which falls below zero is cut off as shown in the plot. Then, a filter is used to retrieve the envelope of the signal and hence retrieving our demodulated message signal. FigureNow this technique will be analysed in the frequency domain. In Figure 10 and 11, one can see the spectra of our modulating and carrier signals at their respective frequencies. In Figure 12, the frequency response of the modulated signal is shown. As observed in theory, for the Double Sideband Large Carrier modulation, we have a spectrum  $A_c$  at  $f_c = 10\text{Hz}$ ,  $A_m/2$  at  $f_c + f_m = 11\text{Hz}$  and  $A_m/2$  at  $f_c - f_m = 9\text{Hz}$ . Upon rectifying our signal, the frequency response is as shown in Figure 13. All that is left is then to filter the signal in order to retrieve our message signal with a frequency response as shown in Figure 14. The filter used is a bandpass filter with  $f_{\text{pass}} = 5\text{Hz}$  and  $f_{\text{stop}} = 15\text{Hz}$ .

FigureFigureFigureFigureFigureLet us now consider what happens in the time domain when the gain of the carrier is increased to 2.  $A_c = 2$  so  $m_a = A_m/A_c = \frac{1}{2}$ . In this way, the condition required for demodulation from an envelope still holds because the modulation index is less than 1. What happens now is that the modulated signal will have its envelope be always greater than 0. Hence, the final demodulated recovered signal will be shifted upwards. This is shown in the time domain plots in Figure 15. FigureSubsequently let us

now analyse what happens when the gain of the carrier  $A_c$  is set to be less than the gain of the message signal  $A_m$ . Let us set  $A_c = 0.5$ . In this way  $m_a = A_m/A_c = 1/0.5 = 2$ .  $m_a$  is now greater than 1 and does not obey the condition required for envelope demodulation. This is because the envelope of the modulated signal is no longer proportional to  $m(t)$ ; resulting in a distorted output signal as shown in Figure 16. Figure 16 shows the time domain plots. The above plots were always retrieved by using asynchronous demodulation techniques. Let us now consider a synchronous demodulation technique. This is done by squaring the modulated signal, taking the square root of the received signal and afterwards applying a filter. Figure 17 shows this process. The gain of the carrier  $A_c$  was set again to 1 such that  $A_m = A_c$ . Figure 18 shows the time domain plots. As one can see, the process is very similar to that done above because prior to filtering, both time domain graphs are identical. The final retrieved signal is almost identical to that seen in Figure 9. Frequency and phase shift are the same; the only mismatch is in the amplitude. This is due to the fact that upon rectifying, half of the signal was lost, yet by squaring and then square rooting; the entire signal amplitude is left intact. Figure 18 shows the time domain plots.

## Single Side Band

Double Sideband modulation wastes bandwidth because it requires a transmission bandwidth equal to two times the message bandwidth. Since either the upper sideband or the lower sideband contains the entire message signal, only one sideband is required for information transmission. Hence there is single sideband modulation. A simple way to generate an SSB signal is to first generate a DSB signal and then suppress one of the sidebands by

filtering. This is known as the frequency discrimination method. In practice this method is not easy because the filter must have sharp cutoff characteristics. Another method for generating and SSB signal is known as the phase-shift method. The phase shifter delays the phase of every frequency component by  $\pi/2$ . An ideal phase shifter is almost impossible to implement exactly. But we can approximate it over a finite frequency band. This method is implemented by using two modulating signals and two carrier signals which are phase shifted by  $\pi/2$ , as shown in the model in Figure 19. Single Sideband Modulation has two possible outcomes upon modulation; either by using the Lower Sideband or by using the Upper Sideband. Both will result in very similar modulations. Equations are as follows:  $V_{SSB-LSB}(t) = A_m \cos(2\pi f_m t) \cos(2\pi f_c t) + A_m \sin(2\pi f_m t) \sin(2\pi f_c t) = A_m \cos[(2\pi f_c - 2\pi f_m)t]$  This results in a spectrum  $A_m$  at  $(2\pi f_c - 2\pi f_m)$ . Similarly for upper sideband modulation:  $V_{SSB-USB}(t) = A_m \cos(2\pi f_m t) \cos(2\pi f_c t) - A_m \sin(2\pi f_m t) \sin(2\pi f_c t) = A_m \cos[2\pi f_c + 2\pi f_m)t]$  This results in a spectrum  $A_m$  at  $(2\pi f_c + 2\pi f_m)$ . As shown in Figure 19, the product of the modulating signal and the carrier signal is added to the product of another pair of modulating signal and carrier signal. Hence, due to the addition instead of the subtraction, the modulated output obtained is Lower Sideband.

Demodulation of SSB signals can be achieved easily by multiplying the modulated signal by a local carrier and passing the resulting signal through a low-pass filter. Figure The time plots obtained from this setup are shown in Figure 20. The two sets of modulating and carrier signals are shown. It is evident that the two sets are identical except for a phase shift. The modulating signals have a frequency of 1Hz and the carrier signals have a



frequency of 10Hz. The fifth graph shows the modulated output in the time domain and the sixth graphs shows the demodulated output. FigureThe frequency spectra of the modulating and carrier signals are shown in Figure 21 and 22. FigureFigureAs explained before, Single sideband modulation results are similar to Double Sideband however only one sideband is obtained. In this case, the lower sideband is obtained; hence in Figure 23 there is a spectral line at  $10-1=9$ Hz. FigureThe modulated signal is multiplied by a carrier for demodulation and the resulting frequency spectrum is shown in Figure 24. This is then filtered using a low pass filter, so that the resulting demodulated output is similar to the initial modulating signal. FigureFigure

## **Quadrature Amplitude Modulation**

Quadrature Amplitude Modulation is a type of modulation which can carry two different modulating signals at the same time. This is done by changing the amplitude of two carrier signals using amplitude modulation. The two carrier signals are 90 degrees out of phase, hence the name quadrature. Each of the two modulating signals are multiplied by their respective carrier. These modulated waves are then added together to create one signal. Hence the QAM Modulated signal consists of both amplitude modulation and phase modulation. Phase is affected due to the fact that the carriers are not synchronized. Hence: Using trigonometric equations, the above derivation was done. It results that QAM has a spectrum of amplitude at frequencies  $(\omega_c + \omega_{m1})$  and  $(\omega_c - \omega_{m1})$  and another of amplitude at frequencies  $(\omega_c + \omega_{m2})$  and  $(\omega_c - \omega_{m2})$ . Then, demodulation takes place by multiplying the modulated signal by the carrier. In order to retrieve each signal back, the

modulated signal needs to be multiplied by their respective carrier signals. Then a simple low pass filter is used to remove unwanted frequencies and hence obtain our demodulated signals. These QAM modulation and demodulation techniques explained above may be seen in the diagram in Figure 26. Figure 27 shows clearly the Modulated signals, their respective carrier signals, the modulated signal and the demodulated outputs. Amplitude and Phase modulation can be clearly seen graphically on the modulated signal. As expected, the amplitudes at the demodulated output are half the amplitude of the inputs; however some phase shifting has occurred. The two figures above show the frequency response of Modulating Signal 1 and its Carrier Signal. The modulating signal has a frequency of 1Hz and the carrier is ten times the modulating signal. Similarly, the two figures above show the frequency response of Modulating Signal 2 and its Carrier Signal. As expected, in Figure 23 we can see that the modulated signal consists of at frequencies  $(f_c + f_{m1}) = 11\text{Hz}$  and  $(f_c - f_{m1}) = 9\text{Hz}$  and similarly another of amplitude at frequencies  $(f_c + f_{m2})$  and  $(f_c - f_{m2})$ . When the modulated signal was multiplied by carrier signal 1, the following frequency spectrum was obtained. This was then filtered to obtain the demodulated output as required. In a similar manner, the same was done to retrieve signal 2. The modulated signal was multiplied by the carrier signal 2 and then filtered as required; thus obtaining the demodulated spectrum of signal 2 as shown in Figure 35.

Figure

## FM

The basic definition of Frequency Modulation is that the instantaneous output frequency of a transmitter varies according to the modulating signal.

The equation of a sine wave may be written as: While amplitude modulation is achieved by varying, frequency modulation is obtained by varying with respect to the modulating signal.

The amount of frequency deviation a signal experiences is due to the change in transmitter output frequency from the rest frequency of the transmitter. The rest frequency of a transmitter is defined as the output frequency when no modulating signal is applied.

For a transmitter with linear modulation characteristics, the frequency deviation of the carrier is directly proportional to the amplitude of the applied modulating signal. Thus an FM transmitter has modulation sensitivity which is represented by a constant,  $k_f$ :

For a single modulating tone the amount of frequency deviation is given by:

where  $\Delta f$  is the instantaneous frequency deviation and  $f_m$  represents the modulating signal.

The peak deviation is given by where both  $\Delta f$  and  $f_m$  are peak values. Hence, an equation can be written for the FM modulated signal:

and substitution of  $\Delta f$  yields: An FM wave consists of three or more frequencies added together to give the appearance of a sine wave whose frequency is varying with time. Mathematical analysis gives an equation for the voltage of an FM wave of the form shown here: where  $f_c$  is the rest-

frequency peak amplitude,  $f_m$  and  $f_c$  represent the rest and modulating frequencies,  $m_f$  is the modulation index. The equation represents a single low-frequency sine wave,  $f_m$ , modulating another high-frequency sine wave,  $f_c$ .

The equation cannot be solved with algebra or trigonometric identities.

However, Bessel function identities may be used that will give a solution to the equation and allow us to

determine the frequency components of an FM modulated signal. The resulting equation is shown here. It singles out all the different signal components in an FM signal and their respective amplitudes.

What this equation shows is that there are an infinite number of sideband pairs for an FM signal.

Each sideband pair is symmetrically located about  $f_c$ , and separated from the rest frequency by integral multiples of the modulating frequency,  $n \times f_m$ , where  $n = 1, 2, 3, \dots$ .

The magnitude of the rest frequency and sideband pairs depends on  $m_f$ , and given by the Bessel function coefficients,  $J_n(m_f)$ ;

where  $n$  is the order

of the sideband pair and  $m_f$  is the modulation index. Therefore, one of the methods of constructing FM modulation and demodulation is shown in Figure

37. A discrete time VCO is used for modulation. A VCO is a Voltage

Controlled Oscillator wherein its output oscillation frequency varies according to the voltage being inputted. In this case the voltage at the input of the VCO is our modulating signal sine wave. For demodulation two filters are used.

Figure 38 depicts our output signals in the time domain. The first plot

shows the modulating signal which is our sine wave of frequency 0.2 Hz. Then there is the modulating signal when manipulated by FM. As one can see, the frequency deviation of our modulated signal varies according to the sine wave at the input. The third time domain plot is the output demodulated signal. This signal has one tenth of the gain of the modulating signal and a slight phase shift. Figure 39 to 42 show the signals obtained in the frequency domain. Figure 39 depicts the single spectrum of our sine wave. Figure 40 shows the spectra of our modulated signal. As expected, there are multiple spectra symmetrical around the frequency of 5 Hz due to Bessel functions explained above. This signal was then demodulated and filtered to retrieve the final signal whose spectrum is shown in Figure 42.

FigureFigureFigureFigureFigure