Double side band the suppressed carrier biology essay

Science, Biology

CCE3320: Communication Theory for Engineers

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3rd Year Electrical and Electronics Engineering 2012/2013

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Double Side Band – Suppressed Carrier

Consider a modulating signal , where and fm is the modulating frequency. Consider also a proportional carrier signal where and fc is the carrier frequency. Double Side Band Suppressed Carrier modulation occurs simply by multiplication of the carrier. Hence, . By the trigonometric identity; , Recovery of the message signal m(t) is done by multiplying the modulated signal sDSB-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:

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By using the trigonometric formula; Which results in ½ 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 ½ 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 fc-fm(10-1= 9Hz) and fc+fm(10+1= 11Hz). There is also no carrier present. Thus the name of Double Sideband- Suppressed Carrier. The magnitudes of these frequencies should ideally be Am/2. FigureAs 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 -2fc and +2fc which need to be removed. These frequencies are removed from the signal by using a low pass filter which has an fpass at 4Hz and an fstop at 6Hz. The final demodulated output signal in the frequency domain is shown in Figure 7. Ideally it should have a magnitude of Am/2. FigureFigure

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 , where and also a carrier signal where . 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 Ac is added to make the modulated signal. Hence, Let ma be the modulation index such that , By using trigonometric identities; From this equation, it can be deduced that the modulated signal should have a frequency spectrum of magnitude Ac at the carrier frequency fc, a lower sideband of magnitude ½ ma at (fc-fm), and similarly an upper sideband of magnitude ½ ma at (fc+fm). Demodulation of DSB-LC signals can be implemented using a simple scheme called envelope detection, provided sufficient carrier power is transmitted. If the gain Ac is large enough, the amplitude of the modulated waveform given by) will be proportional to . 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 Ac is not large enough, then the envelope of sDSB-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 . The modulation index, ma= 1 when Am= Ac. When 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 byWhen ma is maximum, Power in Carrier Pc : Power in Signal Ps, Pc : Ps = 2 : 1and Ps : PT = 1 : (1+2) = 1 : 3Hence, only 33% of the total transmitted power PT 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 then passing it 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. FigureNext, 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 ma= Am/Ac is at its maximum possible. This is the limiting condition because the envelope of the modulated signal touches 0 as shown the the 3rd time domain plot. For the process of demodulation used above, the signal sDDSB-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 Ac at fc= 10Hz, Am/2 at fc + fm = 11Hz and Am/2 at fc - fm= 9Hz. 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 fpass = 5Hz and fstop = 15Hz. FigureFigureFigureFigureFigureLet us now consider what happens in the time domain when the gain of the carrier is increased to 2. Ac= 2 so ma= Am/Ac= ½ . 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 Ac is set to be less than the gain of the message signal Am. Let us set Ac= 0. 5. In this way ma= Am/Ac = 1/0. 5 = 2. Ma 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. FigureThe 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 Ac was set again to 1 such that Am= Ac. 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. FigureFigure

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 π/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 π/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: VSSB-LSB(t) = Am. cos(2πfm t). cos(2πfc t) + Am. sin(2πfm t). sin(2πfc t) = Am. cos[(2πfc - 2πfm)t]This results in a spectrum Am at (2πfc - 2πfm). Similarly for upper sideband modulation: VSSB-USB(t) = Am. cos(2πfm t). cos(2πfc t) - Am. sin(2πfm t). sin(2πfc t) = Am. cos[2πfc + 2πfm)t]This results in a spectrum Am at (2πfc + 2πfm). 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 easily done by multiplying the modulated signal by a local carrier and then passing the resulting signal through a low-pass filter. FigureThe 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= 9Hz. 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 (ωc + ωm1) and (ωc - ωm1) and another of amplitude at frequencies (ωc + ωm2) and (ωc - ωm2). 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. FigureFigure 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. FigureFigureFigureThe 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. FigureFigureSimilarly, the two figures above show the frequency response of Modulating Signal 2 and its Carrier Signal. FigureAs expected, in Figure 23 we can see that the modulated signal consists of at frequencies (fc + fm1)= 11Hz and (fc - fm1)= 9Hz and similarly another of amplitude at frequencies (fc + fm2) and (fc - fm2). 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. FigureFigureIn 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. FigureFigure

Frequency Modulation

Let the message modulating signal be m(t) then in Frequency Modulation (FM) the instantaneous frequency fi(t) is varied linearly with m(t), kf is the frequency sensitivity factor, such that: Integrating this with respect to t and multiplying by 2π, we get: The frequency modulated carrier is then given by: Consider a sinusoidal modulating signal defined byΔf is called frequency deviation and is the maximum departure of the instantaneous frequency of the FM wave from the carrier frequency. Note that the instantaneous frequency is not a spectral component of the FM spectrum. By substituting:

We define β as the modulation index: Then the instantaneous angle is given by: And the FM wave itself is given by: It can be shown that: Where Jn(β) is an nth Bessel function, whose value is found from mathematical tables. Taking the Fourier transform of above yields the double sided spectrum S(f): Properties of Bessel Functions: a) Jn(β) = J-n(β) for n evenb) Jn(β) = -J-n(β) for n oddc) For β << 0J0(β) ≈ 1J1(β) ≈ β/2d)For arbitrary βWhat_this_means_is_that_there_are_an_infinite_number_of sideband pairs for an_FM_signal. _Each_sideband_pair_is_symmetrically_located_about_fC, _and_separated_fromthe_rest_frequency_by_integral_multiples_of_the_modulating_frequency, _n_×_fM, where_n= 1, 2, _3, _…_. _The_magnitude_of_the_rest_frequency_and_sideband_pairs_depends on mf, and_given_by_the_Bessel_function_coefficients, _Jn(mf); _where_the_n_of_is_the_orderof_the sideband_pair_and_mf_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. FigureFigure 38 depicts our output signals in the time domain. The first plot shows the modulating signal which is our sine wave of frequency 0. 2Hz. 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 5Hz 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.

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