Part-I

Experiment- 4 Double Sideband Suppressed

Carrier(DSB_SC).

1. Introduction

1.1 Objectives

This experiment deals with the basic of various amplitude modulation techniques for analog communication. The student will learn the basic concepts of amplitude modulation and examine the using of various modulation schemes. Upon completion of the experiment, the student will:
* Recognize a mixer as practical multiplier.
* Understand amplitude modulation and the difference between suppressed and transmitted carrier modulation
* Learn how to construct $DSB-SC$ modulators.
* Know how to construct a synchronous demodulators.
* Possess the necessary tools to evaluate and compare the performance of systems.

1.2 Prelab Exercise

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1.Evaluate equation 6 for the Product detector shown in Fig.-7 .
2. Suppose that you have to calibrate (synchronize) the phase between the LO with the carrier of the DSB signal, What is the desired phase? what is the worst case phase between the signals?
3.Draw $AM\ DSB-SC$ signal using $MATLAB$ or other mathematics software with the following parameters: Carrier sinewave frequency 10Hz amplitude 1volt, modulating frequency sinewave 1 Hz amplitude 1 volt. Draw 4 subplots on the same graph . The first plot modulating frequency as a function of time. The second plot carrier frequency as a function of time. The third plot $AM\ DSB-SC$ signal as a function of time. The fourth plot Fourier transform of $AM\ DSB-SC$ signal.
4. Prove that $DSB-SC$ signal can be generated from two AM modulator as shown in Fig.-1, using mathematics describe signal at each point?

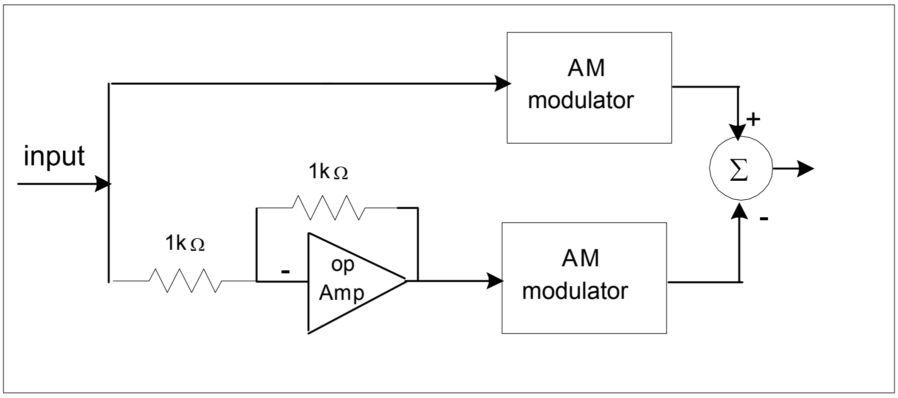

Figure - 1 - : DSB_SC Modulator

1.3 Necessary Background

The student is required to have an understanding of Fourier transforms. He will need to know systems analysis in order to be in a position to design modulators and demodulators. A working knowledge of random processes and probability is needed to be able to evaluate system performance.

2. Background Theory

2.1 Modulation Concept

Suppose you (engineer) were given the job of transmitting signal- speech or simple sinus as baseband data through an ideal channel (air). The first question you should ask yourself is whether the signals must be modified before transmitting them into the channel. If the answer is no, your job is very simple: You must simply decide how to couple (connect) the signal into the channel .
For many channels, the answer will be yes, and the signal will have to be modified. The Fourier transform of a typical baseband sinewave is sketched in Figure -2.

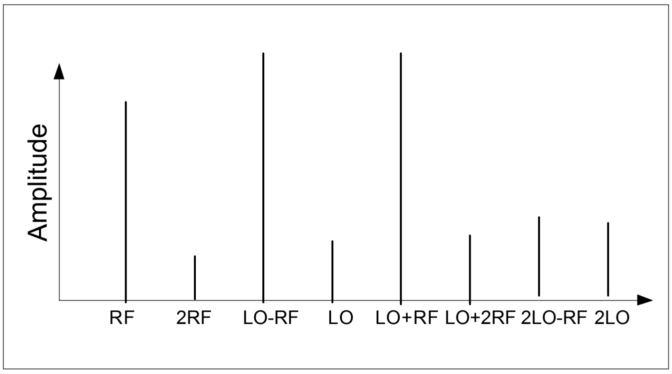

Figure - 2 - : Frequency domain of baseband signal

Suppose we take an audio signal and attempt to transmit it through the air. Let us choose a typical audio frequency of 10 kHz. The wavelength of a 10-kHz signal in air is approximately 30 Km. A quarter-wavelength antenna would then have to be 7.5 Km long, and erecting such antennas in backyards of homes would be impractical! But even if we were willing to erect them, we would still be left with two very serious problems. The first is related to the characteristics of air at audio frequencies: While propagation does occur at frequencies below 10 kHz, these frequencies are not efficiently transmitted through air. Even more serious is the second problem: interference. Often, it is desirable to transmit more than one analog signal at a time. For example, many local radio stations transmit broadcasts simultaneously. If they used quarter-wavelength antennas, they would each have antennas 7.5 Km long on top of their studios (or on mountaintops), and they would transmit into the air many audio signals. The listener would erect an antenna 7.5-Km high and receive a weighted sum of all of the audio signals (depending on relative distances and antenna patterns from the different transmitting antennas to the receiving antenna). Since the only information the receiver would have about the signals is that they would all be bandlimited to the same upper cutoff frequency, there would be absolutely no simple way to separate the signal from one station from all of the others.
Given the above scenario, it is important to modify a low-frequency signal before sending it from one point to another. An added bonus arises from the modified signal is less susceptible to noise than is the original signal.
The most common method of accomplishing the modification is to use the low-frequency signal to modulate (to modify the parameters of) another, higher frequency signal. Most commonly, this other signal is a pure sinusoid.
We start, then, with a pure sinusoid $S_C(t)$ called the carrier waveform. It is given this name because it is used to carry the information signal from the transmitter to the receiver. Mathematically,

$$S_C(t) = A\cos(2\pi f_c t + \theta) \tag{1}$$

If f_c is properly chosen, this carrier waveform can be efficiently transmitted. For example, suppose you were told that frequencies in the range between 1 MHz and 3 MHz propagate in a mode that allows them to be reliably sent over distances up to about 200 Km.. If you chose the frequency f, to be in this range, then the pure sinusoidal carrier would transmit efficiently. The wavelength of transmission in the range of 1 MHz to 3 MHz is on the order of 100 meters and antennas of reasonable length can be used.
We now ask the question whether the preceding pure sinusoidal carrier waveform can somehow be altered in a way that (a) the altered waveform still propagates efficiently and (b) the information we wish to send is somehow superimposed on the new waveform in a way that it can be recovered at the receiver. In other words, we are asking whether there is some way that the sinusoid can carry the information along. The answer is yes, as we now illustrate.

The right-hand side of Equation-1 contains three parameters that may be varied: the amplitude A, the frequency f_c. ,and the phase θ . using the information signal to vary A, f_c, or θ leads to amplitude modulation, frequency and phase modulation, respectively.
We will show that efficient transmission is achieved for each of these three cases. We will also show that if more than one signal is simultaneously propagated through the channel, separation of the signals at the receiver is possible. In addition, we will find it critical to illustrate a third property: The information signal $s(t)$ must be uniquely recoverable from the received modulated waveform; it would not be of much use to modify a carrier waveform for efficient transmission and station separability if we could not reproduce $s(t)$ accurately at the receiver.

2.2 Double Balanced Mixer as Modulator

Mixer converts radio-frequency energy at one frequency to a second frequency. While the most common use for mixers is in the front end of radio receivers (See Figure 3), where they convert input signals frequencies to a lower intermediate frequency, mixers are also used in up-converters,AM modulators, phase detectors, frequency synthesizers, etc.

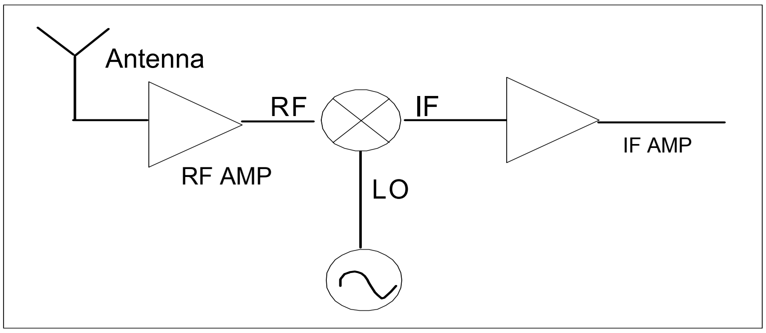

Figure - 3 - : Simplified front end of Superhetterodyne receiver

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2.3 Multiplying action.

The mixing action of a mixer arises from two distinct processes acting in tandem. The input signal (designated RF) is multiplied with a locally generated signal (the local oscillator, LO), thus generating two output signals at the sum and difference frequencies. The difference frequencies are referred to as the intermediate frequency (IF) frequency. In a receiver, the sum frequency is normally rejected by a bandpass IF filter leaving only the difference. Multiplication, however, is effected using non-linear elements (diodes) and these non-linearities are responsible for the generation of many additional frequencies other than the pure sum and difference frequencies.

2.4 Sum and Image Frequencies

We have seen above that an input RF signal and a local oscillator LO signal are multiplied to generate sum and difference IF frequencies. If another input frequency is found that, when mixed with the local oscillator, the correct IF frequency will be generated, then signal or noise power at this frequency will be passed to the mixer IF terminals
A frequency of $2xLO - RF$ is such an input frequency. This particular frequency is called the image frequency (See Figure 2).

2.5 Conversion Loss

We have seen above that half the converted power is inevitably lost in the mixing process. Hence this loss (the Single Sideband Conversion Loss) between RF input power and IFoutput power will have a minimum value of 3 dB. In practice, extra losses due to the generation of spurious products. Resistive losses in the diodes, mismatches at the mixer ports, etc., will combine to increase this figure. Careful selection of local oscillator power to bias the diodes at their optimum operating points will minimize mixer conversion loss. All mixers have been designed , with optimum diode/LO drive power combinations. Accordingly, our devices always should be operated with the LO drive power specified in each data sheet.

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2.6 Double Sideband Suppressed Carrier

If we modulate the amplitude of the carrier of Equation-. 2 the following modulated waveform results:

$$S_m(t) = A(t)\cos(2\pi f_c t + \theta) \tag{2}$$

The frequency f_c and the phase θ is constant. The amplitude $A(t)$ varies somehow in accordance with the baseband signal $s(t)$ - the signal we want carried through the channel.
We simplify the expression by assuming that$\theta = 0$. This will not affect any of the basic results, since the angle actually corresponds to a time shift of $\theta/2\pi f_c$, A time shift is not considered distortion in a communication system.
If somebody asked you how to vary $A(t)$ in accordance with $s(t)$, the simplest answer you could suggest would be to make $A(t)$ equal to $s(t)$. This would yield a modulated signal of the form

$$S_m(t) = s(t)\cos 2\pi f_c t \tag{3}$$

Such a signal is given the name double-sideband suppressed carrier $(DSB - SC)$ amplitude modulation for reasons that will soon become clear.

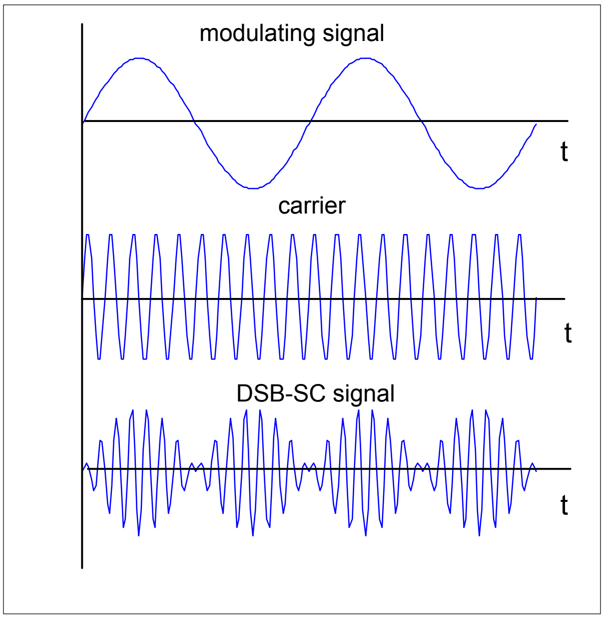

Figure - 4 - : DSB_SC modulation process

This simple equating of $A(t)$ with $s(t)$ does indeed satisfy the criteria demanded of a communication system. The easiest way to illustrate this fact is to express $S_m(t)$ in the frequency domain, that is, to find its Fourier transform.

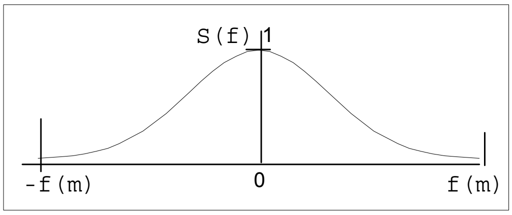

Figure - 5 - : Spectrum of baseband signal

Suppose that we let $S(f)$ is the Fourier transform of $s(t)$. We require nothing more of $S(f)$ than that it be the Fourier transform of a baseband signal. That is, $S(f)$ must equal zero for frequencies above some cutoff frequency f_m,. (The subscript $m-$stands for maximum.) Figure 5 gives a representative sketch of $S(f)$.It does not mean to imply that $S(f)$must be of the shape shown; the sketch is meant only to indicate the transform of a general low-frequency bandlimited signal. The modulation theorem is used to find $Sm(f)$:

$$S_m(f) = \Im\left[s(t)\cos 2\pi f_c t\right] = \frac{1}{2}\left[s(f+f_c) + s(f-f_c)\right] \tag{4}$$

This transform is sketched as Figure 6 Note that modulation of a carrier with $s(t)$ has shifted the frequencies of $s(t)$ both up and down by the frequency of the carrier. This is analogous to the

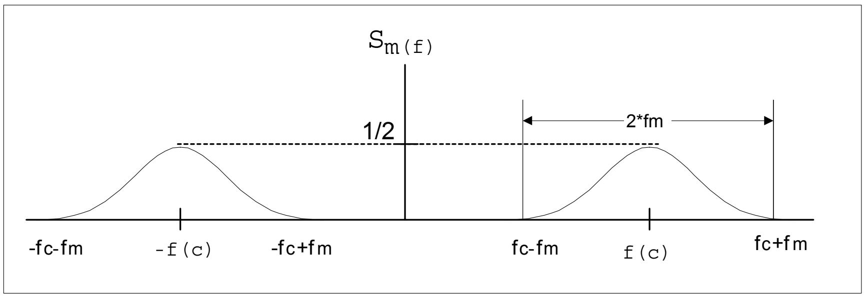

Figure - 6 - : Frequency domin of DSB-SC signal

trigonometric result that multiplication of a sinusoid by another sinusoid results in sum and difference frequencies. That is,

$$\cos A \cos B = \frac{1}{2}\cos(A+B) + \frac{1}{2}COS(A-B) \tag{5}$$

If $cosA$ is replaced by $s(t)$, where $s(t)$ contains a continuum of frequencies between 0 and f_m the trigonometric identity can be applied term by term to yield a result containing all sums and differences of the frequencies. Equation 4 indicates that the modulated waveform $S_m(t)$ contains components with frequencies between f_c- f_m and $f_{c+}f_m$ As long as signals in this range of frequencies transmit efficiently and an antenna of reasonable length can be constructed, we have solved the first of the two problems. Let us plug in some typical audio numbers. Let f_m, be 5 kHz and f_c,. be 1 MHz. Then the range of frequencies occupied by the modulated waveform is from 995,000 to 1,005,000 Hz.

2.7 Transmitted signal power and spectra.

The bandwidth required for transmitting a message signal $s(t)$ of bandwidth f_m is twice the bandwidth of baseband signal (see FIG. 5) .

$$B_T = 2f_m$$

Another important parameter is the average power needed to transmit the modulated signal $S_m(t)$ DSB_SC signal, to compute the average power S_T assume $s(t)$ is power signal thus

$$\begin{aligned} S_T &= \left\langle (S_m(t))^2 \right\rangle = \lim_{T\to\infty} \frac{1}{T} \int_{-\frac{T}{2}}^{\frac{T}{2}} \frac{A^2}{2} s^2(t)\cos^2(2\pi f_c t)dt \\ &= \lim_{T\to\infty} \frac{1}{T}\left[\lim_{T\to\infty} \frac{1}{T} \int_{-\frac{T}{2}}^{\frac{T}{2}} \frac{A^2}{2} s^2(t) + \int_{-\frac{T}{2}}^{\frac{T}{2}} \frac{A^2}{2} s^2(t)\cos 2(2\pi f_c t)dt\right] \end{aligned}$$

The Band Pass Filter null the second integral, and if we define the average power of $s(t)$ as

$$S_x \triangleq \lim_{T\to\infty} \frac{1}{T} \int_{-\frac{T}{2}}^{\frac{T}{2}} s^2(t)dt$$

$$S_T = \frac{A^2}{2} S_x = S_c S_x = 2P_{sb} \tag{6}$$

where $S_c = \frac{A^2}{2}$ is the average carrier,P_{sb} = average power of each sideband. According to equation 6 we see that $DSB-SC$ modulation makes better (relative to $DSB-TC$) usage of transmitter power, in this case all the power transferred into the sidebands.

2.8 Synchronous Detector- Product Detector

A product detector (Fig.-7) is a mixer circuit that down-converts the (bandpass signal plus noise) to baseband. The output of the multiplier is

$$V_1(t) = S_m(t)A_{LO} = \frac{A_{LO}}{2}\left[K_u s(t) + K_u s(t)\cos 4\pi f_c t\right]$$

where the frequency of the oscillator is fc and the phase between LO and modulated signal $Sm(t)$ is 0 . Therefore the LO is synchronized with the carrier of transmitted signal, in both phase and frequency .If we assume $s(t)$ is a real signal and LPF reject high frequency component $(cos4\pi ft)$, then V_{out} equal to:

$$V_{out} = \frac{A_{LO}}{2}K_u s(t) = K_d s(t) \tag{7}$$

From Equation- 7 we see that the output of the is equal to modulation signal $s(t)$ multiply by constant K_d where it is stand for the detection constant.

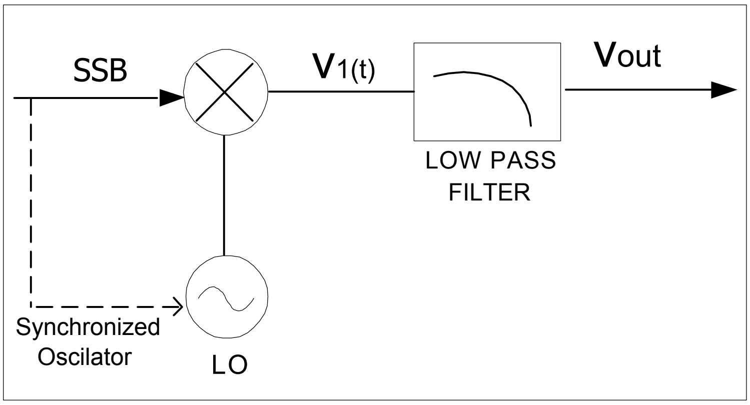

Figure - 7 - : Product Detector

2.9 Effect Of Non Ideal LO On Synchronous Detector.

Any practical oscillator has inaccuracy due to frequency drift and in our case phase error relative to the transmitted carrier. Let represent it in mathematical form as $LO = \cos(2\pi f_c t + 2\pi f' t + \theta')$, where $f\prime$ and θ'represents frequency and phase errors. In our case $(DSB-SC)$ with single tone modulation ,refer to Fig.-8 the detected signal $V_1(t)$ in our case $(DSB-SC$ modulation) is

$$V_1(t) = S_m(t)A_{LO} = \frac{A_{LO}\cos(2\pi f' t + \theta')}{2}\left[K_u s(t) + K_u s(t)\cos 4\pi f_c t\right]$$

If we assume $s(t)$ is a real signal and LPF reject high frequency component $(cos4\pi ft)$, then V_{out} equal to

$$V_{out} = \frac{A_{LO}}{2}K_u s(t)\cos 2\pi f' t + \theta') = K_d s(t)\cos 2\pi f' t + \theta') \tag{8}$$

if we assume that $f\prime \rightarrow 0$ during detection we get

$$V_{out} = K_d s(t)\cos\theta'$$

which mean that it's very important to synchronizes the frequency and phase of the LO to the modulated signal.

3. Experiment Procedure

3.1 Required Equipment

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1.Spectrum Analyzer $(SA)\ HP - 8590L$.
2.Oscilloscope $HP - 54600A$.
3.Signal Generator $(SG)HP - 8647A$.
4. Arbitrary Waveform Generator $(AWG)HP - 33120A$.
5.Double Balanced Mixer Mini-Circuit $ZAD - 2(ZP - 2)$.
6. Band Pass Filter 10 MHz Mini-Circuit $BBP - 10.7$

3.2 Multiplying Signals

Here we want to look at the basic component Double Balanced Mixer as frequency converter or frequency multiplier, and the properties of the signals which they generate ,in frequency domain using spectrum analyzer. This experiment will realize the non-ideal operation of frequency multiplication, and the limitation of our Test and Measurement equipments .
1. Connect the double balanced mixer according to Fig. 8 .
2. Adjust the test equipment as follow:
$LO - AWG$- Frequency 10.7 MHz amplitude 7dbm.
$RF - AWG$- Frequency 100 kHz amplitude -10dbm.
3. Set the spectrum analyzer to
Center Frequency 10.7 MHz, Span 1 MHz, Resolution bandwidth 10 kHz.
4. Observe the signal at Spectrum-Analyzer spectrum analyzer use the marker function to define the frequency domain around 10 MHz, try to identify another major component explain how each component generated.
5. Using the markers of spectrum analyzer, measure the amplitude of multiplication products, why they are lower then input signal (-10dbm)?

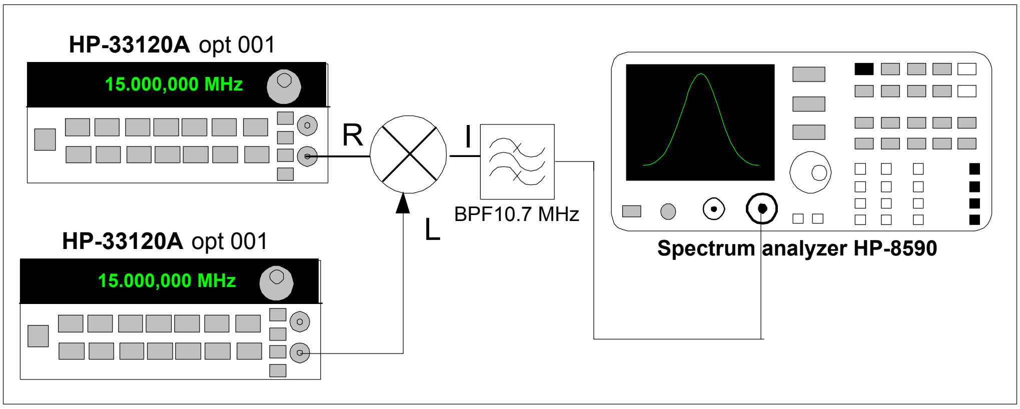

Figure - 8 - : Mixer products

6. Press MKR key on spectrum analyzer, and place 4 markers on the 4 major multiplication products.
7. Press MKR FCTN and $MK\ TABLE\ ON\ OFF$, now you operate the split screen , and you have the frequency and amplitude of each marker, print the results.
8. Turn off the markers and table.
9. Change only the baseband frequency to 3 kHz observe the frequency domain around 10.7 MHz, what happen. Why can't you see the spectrum as before?
10. Change the baseband frequency to 10 kHz observe the frequency domain around 10.7 MHz what happen, Why can't you see the spectrum as you saw it in pharagraph 3?

3.3 Sum, Harmonics and Image Frequencies

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1. Disconnect the IF filter from the above system, and change the baseband frequency to 100 kHz.

2. Change the span of spectrum analyzer to 500 MHz, now you can see all the spurious of the mixer, print the results.
3. Choose the 32.1 MHz harmonic, make zoom, by setting the span to 1 MHz around the product, print the results.

3.4 Double Side Band-Suppress Carrier ($DSB - SC$) modulation/detection.

3.4.1 DSB modulator

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1. Connect the Test and Measurement test equipment according to Fig.-9.
2. Adjust the test equipment as follow:
Signal Generator- Frequency 10.7 MHz amplitude 7dbm.
AWG- Frequency 100 kHz amplitude -10dbm.
3. Use the marker function of the spectrum analyzer to compute the bandwidth of DSB signal. How the spectrum of this signal related to the baseband signal? print the results.

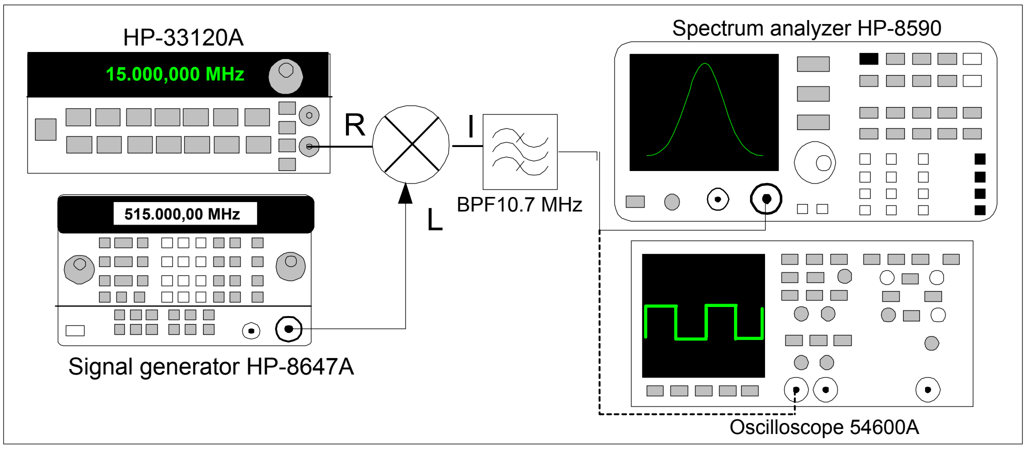

Figure - 9 - : DSB-SC modulator

4. Replace the spectrum analyzer with oscilloscope ,now try to stabilize the signal on the screen, using the cursors or other function, what is the frequency of envelope of the modulated carrier and how this frequency relate to the baseband signal? what is the exact mathematics relation between the two signals, print the results
5. Reconnect the spectrum analyzer change the amplitude of the $AWG(100KHz)$ between -10 dBm to +20dbm, now explain what happen to the spectrum of the modulated signal? print the results.
6.Change the amplitude of the AWG (modulating frequency) to -10dbm and the amplitude of the Signal Generator (LO) between 7 dBm to -80dbm, and watch the display of the spectrum analyzer and try to explain what happen? print the results.

3.4.2 Product Detector.

In this part of the experiment you will analyze the operating concept of product detector. We choose to demonstrate the Product Detector it with $DSB - SC$ modulation, and later with other modulation systems.
1. The AWG of the detector have to be connected as slave to the signal generator $HP - 8647A$ according to Appendix 1.
2.Connect the system as indicated in Fig.-10 to implement product detector.
3. Set the $T\&M$ equipment as follow:
$HP - 8647A$ -Signal Generator - Frequency 10.7 MHz ,amplitude 7 dbm .
$HP - 33120A$ -Baseband generator frequency 100 kHz ,amplitude -10 dBm.

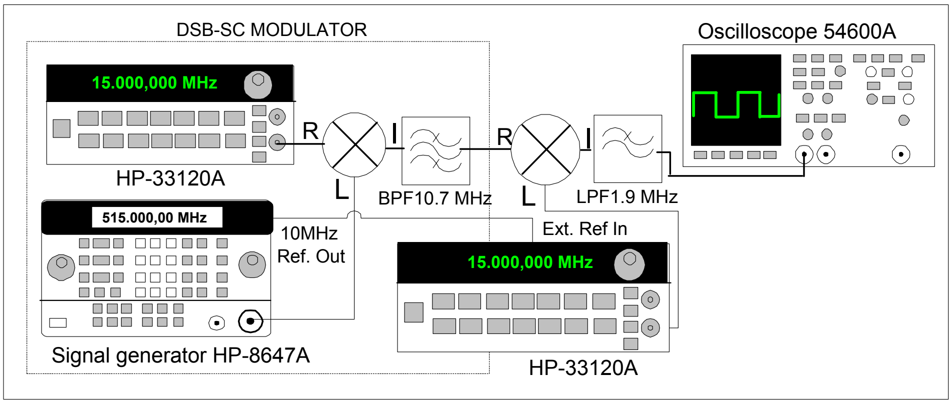

Figure - 10 - : Product detector

$HP - 33120Aopt. - 001$ - Frequency 10.7 MHz ,amplitude 7 dbm .
4. Connect AWG-of the demodulator as slave refer to appendix ,1 set the phase between the two instruments to 0○.
5 Observe the detected signal at oscilloscope , change the phase between the two generators to 10°,30°,45°90° at which phase the detection is worst?print the best and worst detected signals. $DSB - SC$ Demodulator.

4. Final Report

Attached all the print results, short description, answers and your analyzes to the following question.
1.An $DSB - SC$ modulated signal has the form:

$$s_m(t) = 25[5\cos 2\pi 750t + 5\cos 2\pi 250t]\cos 2\pi 20000t \tag{9}$$

* Using Matlab or other software, draw $s_m(t)$ in time domain and frequency domain
* Find the average power content of each spectral component.

5. Appendix 1

5.1 To phase lock to an exrernal 10 MHz signal

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1. Connect rear- panel $Ext.$ 10MHz output terminal of the master Signal Generator HP-8647A to $Ref\ in$ on the rear panel of the slave HP-33120A(or other 10MHz clock) as indicated in Fig-11. Connect the two signals to input 1 & 2 of the scope, observe the phase between the signals.
2. Turn on the menu by pressing $\boxed{\text{shift}}\boxed{\text{Menu On/Off}}$ then the display look like $\boxed{\text{A: MOD MENU}}$.
3. Move to $\boxed{\text{G: PHASE MENU}}$ by pressing the ◁ button.
4. Move down a level to the ADJUST command, by pressing ▽ the display look like $\boxed{\text{1: ADJUST}}$
5. Press ▽ a level and set the phase offset, choose any value between -360 and 360 degrees. Then you see a display like $\boxed{\wedge 120.000\ DEG}$.
6. Turn off the menu by pressing $\boxed{\text{ENTER}}$ .You are then exited the menu.
Important
1. At this point, the function generator HP-33120A is phase locked to another HP-33120A or external clock signal with the specified phase relationship. The two signals remain locked unless you change the output frequency.
2. If you adjust the phase between two function generator, there is a phase difference between the output at the BNC connector, and the end of the cable, therefore always measure the phase difference at the end of the coax cable.
3. It is strongly recommended to set a zero phase reference according to the next section in order to easily set new phase between the signals.

5.2 Setting a zero phase reference

After selecting the desired phase relationship as described on the previous section, you can set a zero-phase point at the end of the coax cable. The function generator then assume that its present phase is zero and you can adjust the phase relative to this new ”zero”

1. Turn on the menu by pressing shift Menu On/Off then the display look like A: MOD MENU .
2. Move across to the PHASE MENU choise on this level by pressing ◁ the display look like G: PHASE MENU
3. Move down a level and then across to the SET ZERO by pressing ▽ and ▷ buttoms, the display show the message 2: SET ZERO .

4.move down a level to set the zero phase reference. Press ▽ buttom, . The displayed message indicates PHASE = 0

5. Press Enter , save the phase reference and turn off the menu.