Communication Systems Lab for Dummies

A Comprehensive Lab Manual for Communication Systems Course designed around Feedback's™ Communications Product Line

First Edition

by

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Disclaimer

I have compiled this document to the best my knowledge and abilities. However, I cannot guarantee the correctness of the information presented here. Use this document at your own risk. The document hasn't been proof read either, so there might be some typos and errors in this document. I would highly appreciate if you could bring any discrepancies in this document to my attention so that they can be rectified in the next edition of this manual. Feel free to contact me at *adnaniazi@gmail.com*

Message from the Author

Dear Students,

I have tried my utmost to make this document as easy to understand and comprehensible as possible. I may or may not have succeeded in doing so. But if you have any difficulties, whatsoever, in understanding this document, please do feel free to pop into the lab and ask for my help. I will be happy to assist you in whatever way I can. You can also drop me an email at adnaniazi@gmail.com. I will respond to it within 24 hours. Guaranteed!

I wish you all the best.

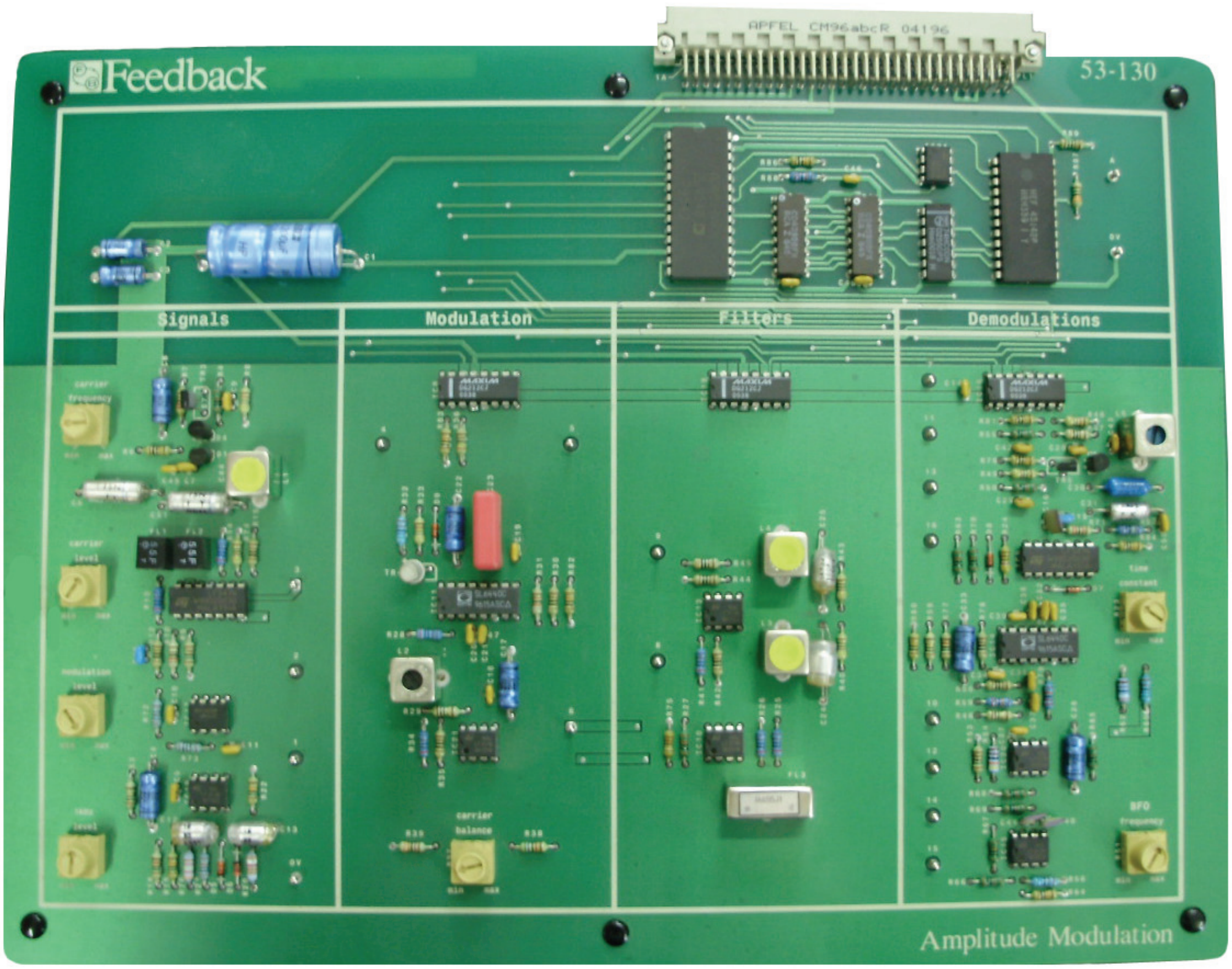
Kind Regards

Adnan Muhammad Niazi

Table of Contents

Equipments Required for Practical 01 to 06

1- Amplitude Modulation **Kit 53-130**
2- Data Acquisition System (DAQ) [1]**RAT 53-100**
3- PC with Feedback's **Discovery** Software installed
4- Triple trace Bench top Oscilloscope **TRIO CS-1040** with pincer probes



Amplitude Modulation Kit 53-130

[1] RAT is an acronym for Rapid Access Terminal. It's a data acquisition system capable of handling Analog inputs and Digital I/O

Triple Trace Bench Top Oscilloscope TRIO CS1040 40MHz

Pincer Probe with its pincer and the probe adjustment screw driver

Function of various knobs on Amplitude Modulation Kit 53-130

Carrier Frequcy
This knob is used to vary the carrier frequency used for modulation

Carrier Level
It changes the amplitude of the carrier signal

Modualtion Level
It changes the Peak value of the modulating signal. Therefore, this knob is used to change the modulation index of the AM.

Carrier Balance
This knob is used to remove carrier from DSB+C signal inorder to obtain DSB-SC signals.

Time Constant
This knob controls the RC time constnt of the envelope detector

BFO Frequency
This knob controls the frequecy of the output of the BFO.

Practical 01: Amplitude Modulation with Full Carrier

Objectives of the Experiment

To observe the effect of varying levels of *m(t)* on the modulated signal.

Background Information

Double side band with full carrier modulation is also known as AM signal and is given by

$AM = [A + m(t)] \cos(\omega_c t)$

In the Amplitude Modulation kit 53-130, 'A' is fixed and cannot be changed. The level of *m(t)* on the other hand, can be changed.

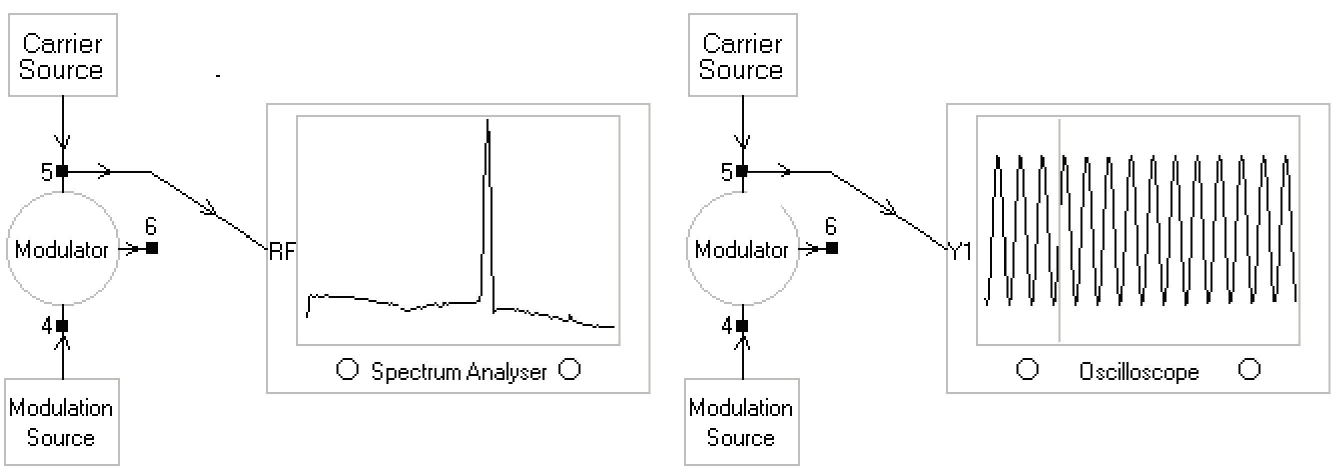
100% Modulation: If the peak value of *m(t)* i.e. m_p is equal to A then the *modulation is 100%*. In this case the two envelopes of the modulated signal will barely touch each other without crossing each other.

Over Modulation: If m_p is greater than A, then the envelopes in the modulated signal will cross each other. This is the case of *Over Modulation.*

Under Modulation: If m_p is less than A, then the modulation is called *Under Modulation*. In this case the envelopes won't touch each other.

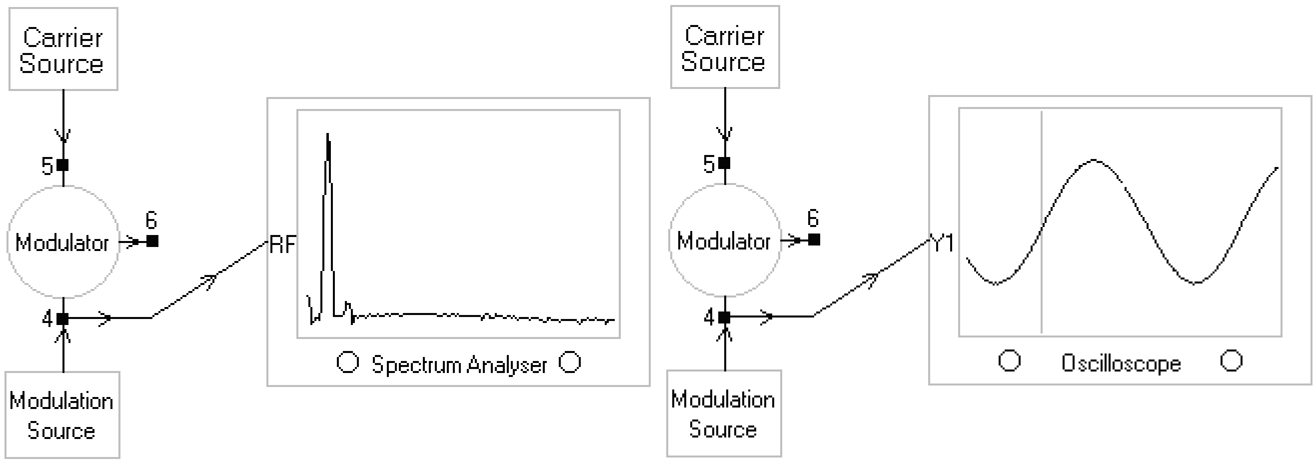
Procedure

1- Start the Discovery software.
2- Go to System->Index->1-Amplitude Modulation with Full Carrier
3- Select 'Yes' to Load Assignment 1.
4- Start Practical 01.
5- Observe the carrier in time and frequency domains



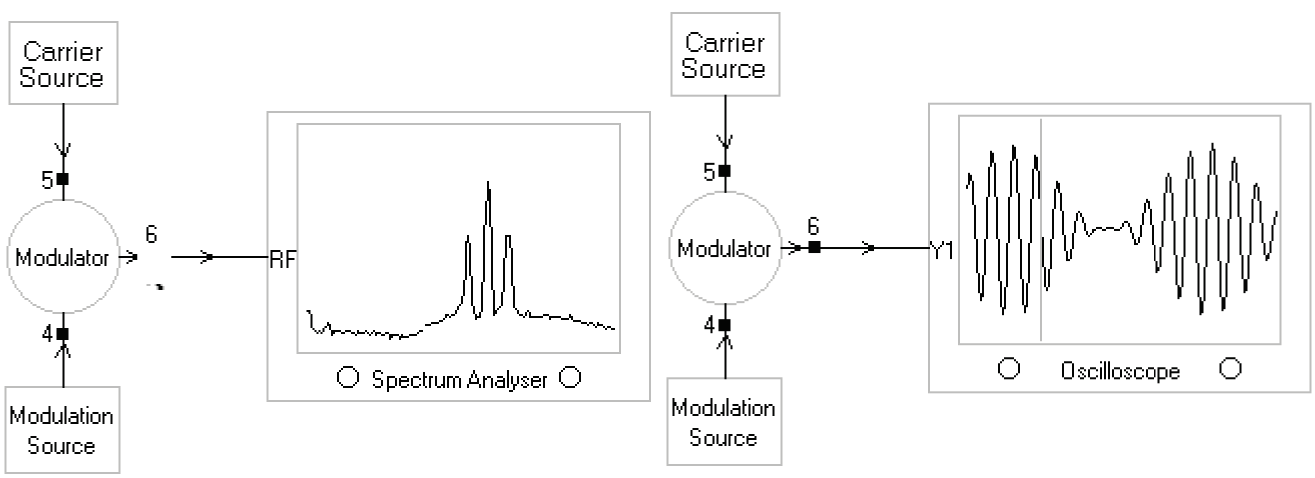
Carrier in Frequency and Time Domain

6- Observe the modulating signal in time and frequency domains



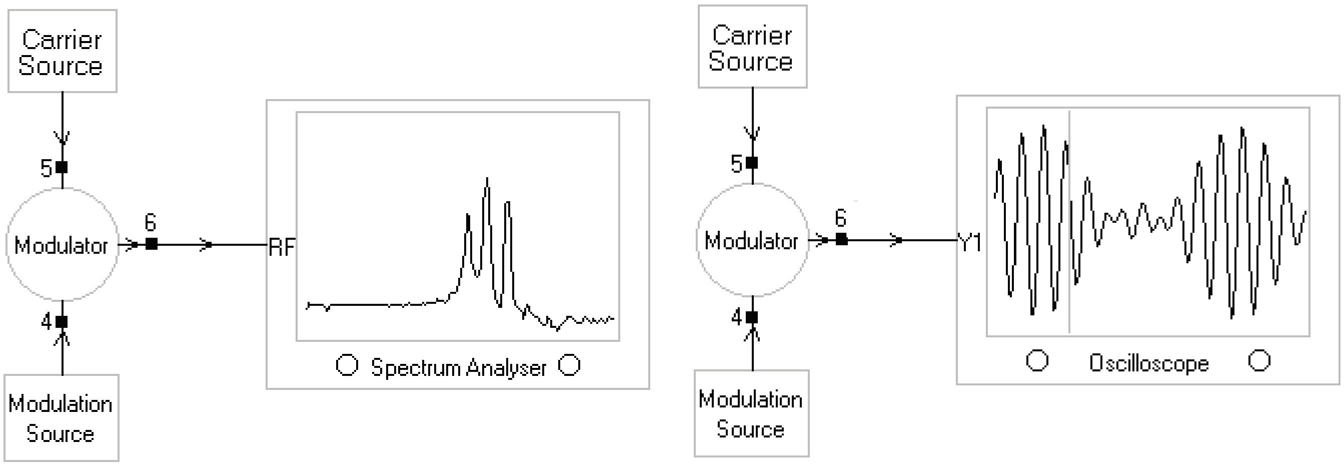
Modulating Signal m(t) in Frequency and Time Domain

7- Change the Modulation signals Level using 'Modulation Level' Knob until you see the 100% modulated signal. There will be no zero crossing in envelop in 100% modulation.

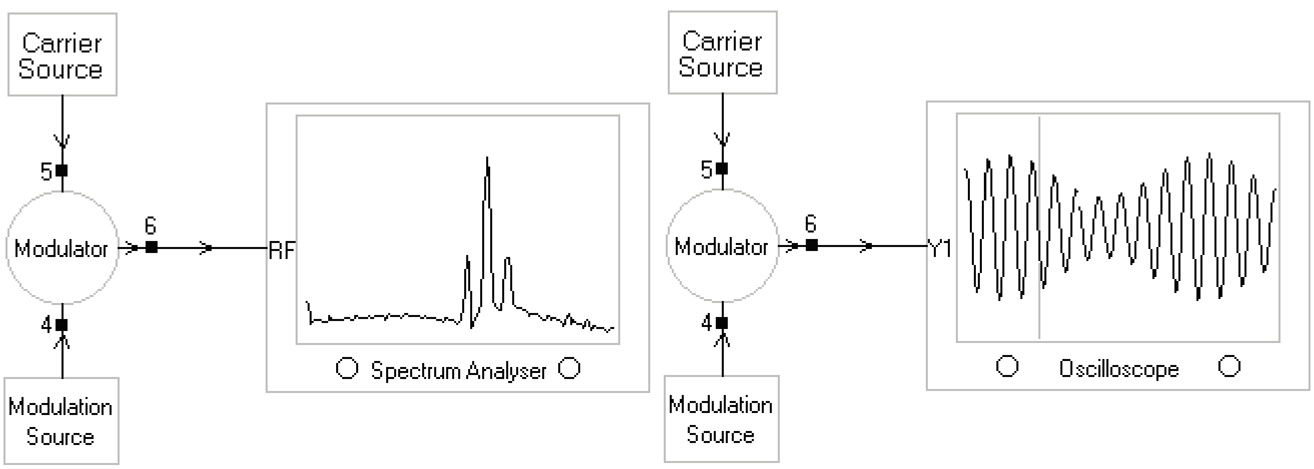


100% Modulated Signal in Frequency and Time Domain
(The two envelopes barely touch each other without crossing each other. In this case $m_p = A$)

8- Change the modulation level and observe Under Modulation and Over Modulation.



Over Modulated Signal in Frequency and Time Domain
(The two envelopes cross each other. In this case $m_p > A$)



Under Modulated Signal in Frequency and Time Domain
(The two envelopes cross are farther away from each other. In this case $m_p < A$)

Questions

1- What is the frequecy of the modualtion source and the carrier used in the practical?

2- Is it easy to measure the frequcey on spectrum analyzer or the oscilloscope?

Practical 02: Demodulation of DSB with Full Carrier using Envelop Detector

Objectives of the Experiment

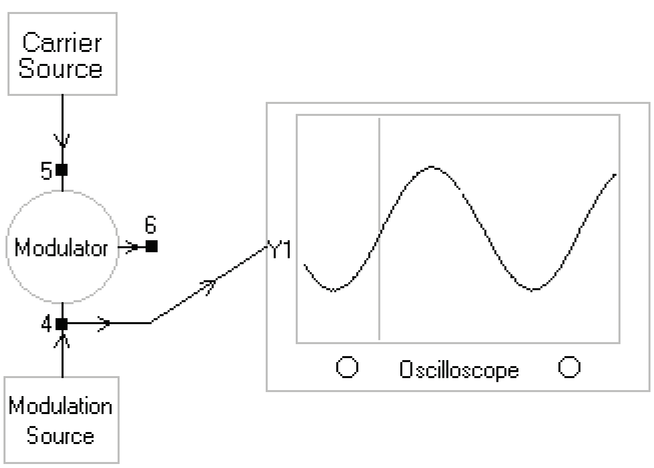
To observe the effect of RC time constant of Envelope Detector on the demodulated output.

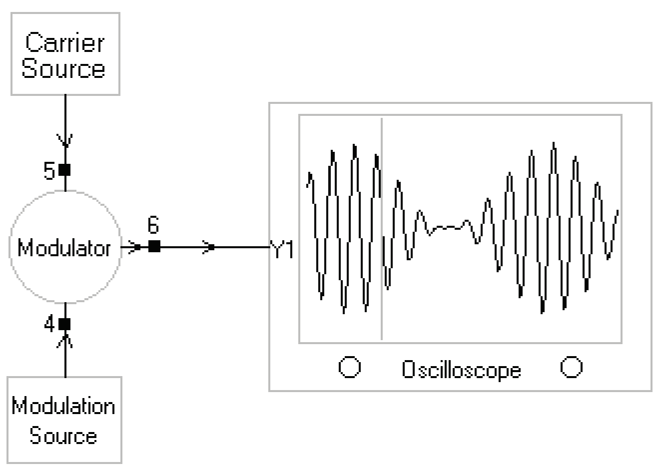
Background Information

Envelope detector contains a rectifier followed by an RC circuit. The rectifier's job is to isolate one envelope out of the two envelopes that a Double side band with full carrier modulated signal contains. The RC circuits job is to follow the envelope which contains the message signal *m(t)*. Selecting an optimum RC time constant for the envelope detector is one of the crucial steps in the design of the envelope detector. If the RC time constant is very small, then you will start seeing the carrier peaks in the demodulated output. This is because the capacitor will discharge too quickly when RC time constant is small. On the other hand, if the RC time constant is too large, then the capacitor in RC circuit once charged, will take too much time to discharge. Thus the detector will not be able to follow the envelope of the modulated signal during the discharge times. Somewhere in between these two extremes lies an optimum time constant at which you will see the best possible demodulated output.

Procedure

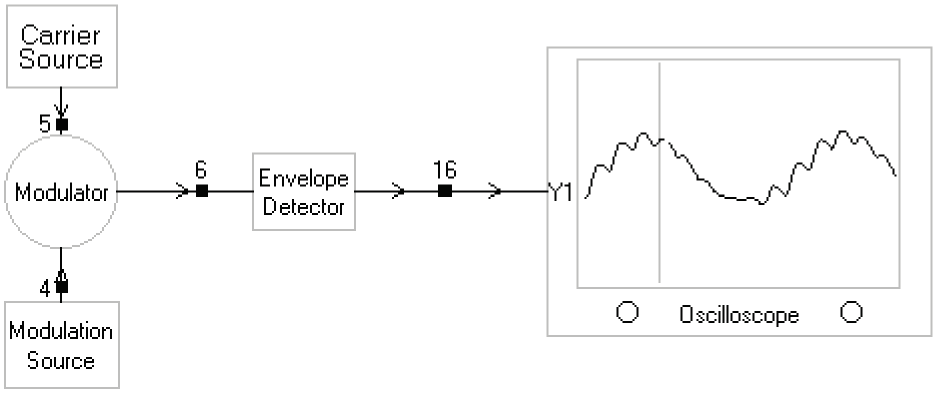
1- Start the Discovery software.
2- Go to System->Index->1- Amplitude Modulation with Full Carrier.
3- Select 'Yes' to Load Assignment 1.
4- Start Practical 02.
5- Set the modulation to 100% by adjusting the 'Modulation Knob'.



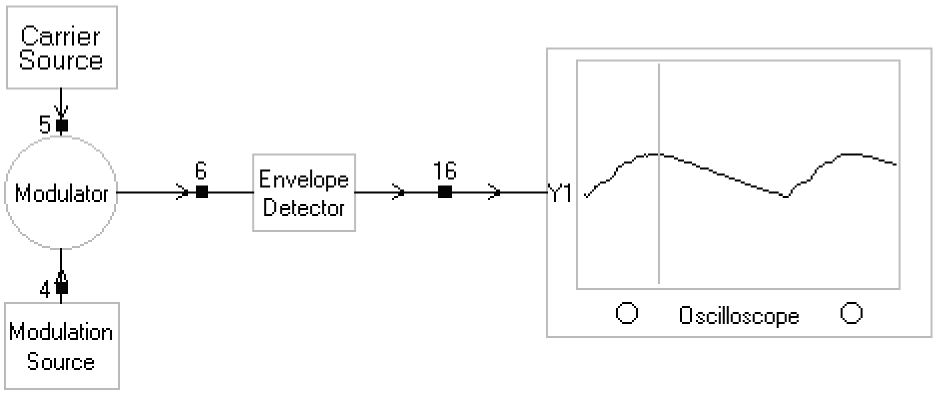


Modulating Signal and Modulated Signal in Time Domain
(The two envelopes barely touch each other without crossing each other. In this case $m_p = A$. So the modulation is 100 %)

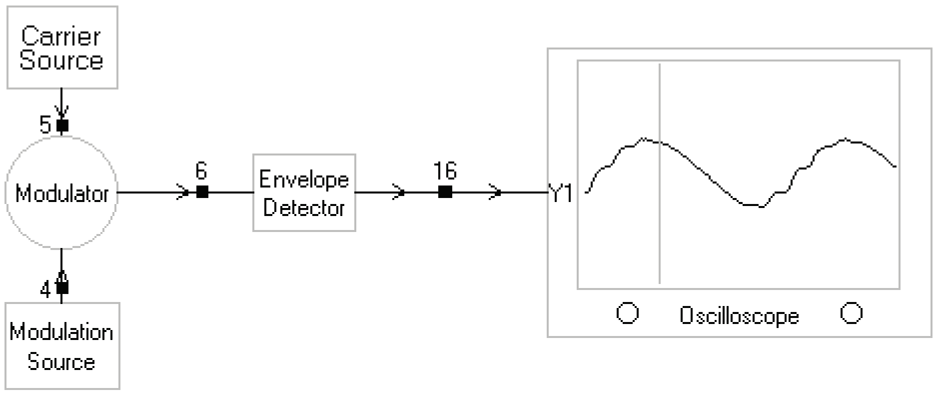
6- Now adjust the time constant using 'Time Constant' Knob to observe the effect of time constant on the demodulated output.



Time Constant is too small
(You can see the carrier peaks in the demodulated output as well. In this case, the capacitor is discharging at a very fast rate)



Time Constant is too large
(The capacitor is taking too much time to discharge)



Time Constant is Optimum

Question

Is the phase shift caused by envelope detector a lead or lag?

Practical 03: Demodulation of DSB with Full Carrier using Product detector

Objectives of the Experiment

To observe the effect of phase of the BFO on the demodulated output

Background Information

In product detector we multiply(Homodyne) the modulated signal with a pure tone that has the same phase and frequency as the original carrier that was used at the time of modulation. This signal can be extracted very easily from the modulated signal itself because it contains carrier. The device that is used to extract the carrier from the modulated signal is called BFO or Beat Frequency Oscillator. The output from BFO is multiplied with modulating signal to get baseband signal along with some high frequency components which are filtered out later.

$$AM = [A + m(t)]\cos(\omega_c t)$$

$$BFO\ output = B\cos(\omega_c t)$$

Output of the Product Detector

$$= AM \times BFO\ output$$

$$= [A + m(t)]\cos(\omega_c t)\, B\cos(\omega_c t)$$

$$= B[A + m(t)]\cos^2(\omega_c t)$$

$$= 0.5B\,[A + m(t)]\,[\cos(2\omega_c t)+1]$$

$$= 0.5AB\cos(2\omega_c t)+0.5AB+0.5B\,m(t)\cos(2\omega_c t)+0.5Bm(t)$$

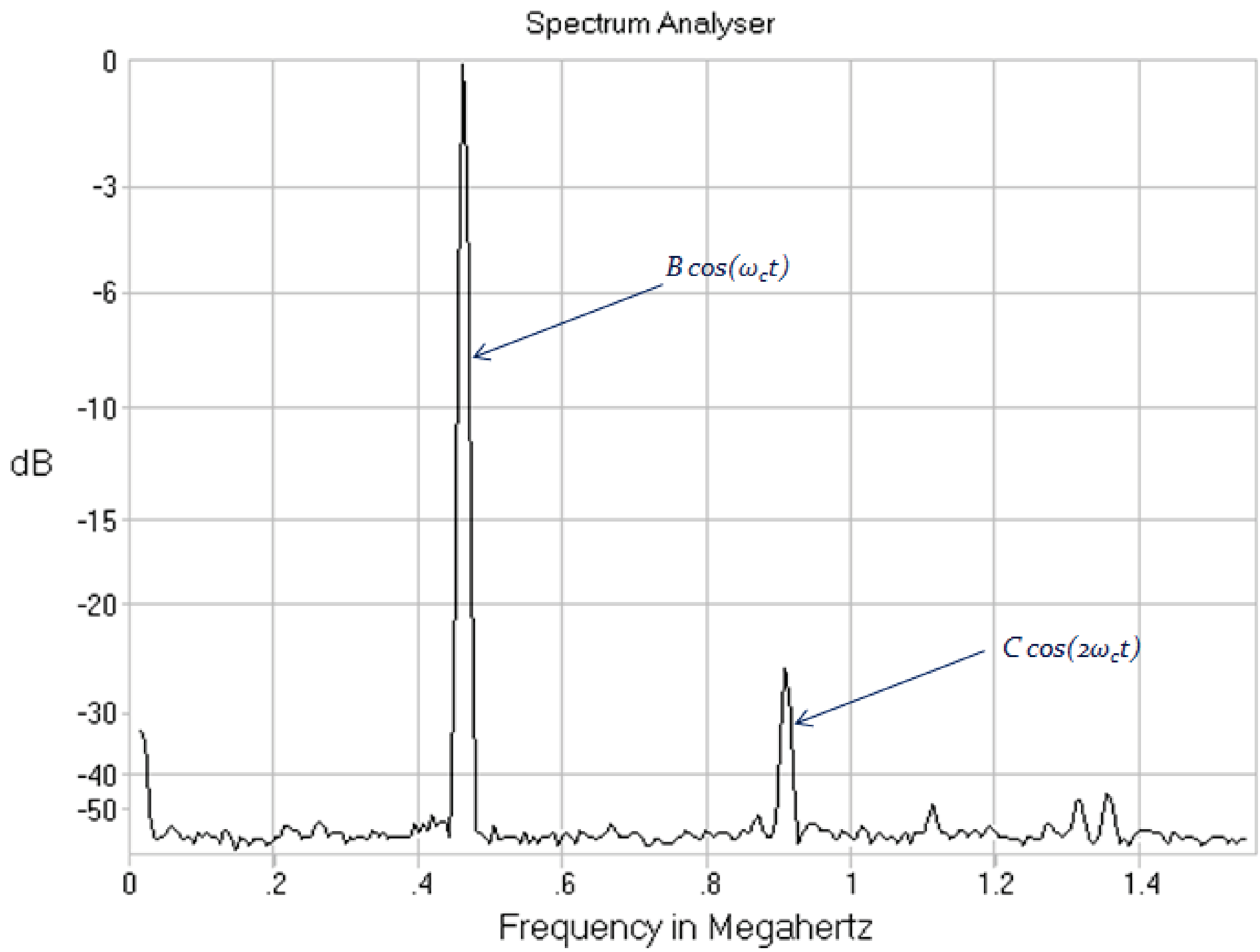
$$= 0.5AB+0.5B\,m(t)+0.5AB\cos(2\omega_c t)+0.5B\,m(t)\cos(2\omega_c t)$$

The last two terms are filtered out by the low pass filter and the DC shift of 0.5AB can be removed by a DC blocking capacitor. The above case happens *only* when the BFO output is a pure tone of $B\cos(\omega_c t)$.

However, the actual BFO used in the trainer kit in the lab has been designed so poorly that it contains a component at $2\omega_c$ as well. So the actual output of the BFO in our Lab is something like this:

$$BFO\ output = B\cos(\omega_c t) + C\cos(2\omega_c t)$$

where $C << B$



BFO output containing both $B\cos(\omega_c t)$ and $C\cos(2\omega_c t)$

For such a BFO, the output of the Product Detector is given by:

Output of the product detector

$= AM \times BFO\ output = [A + m(t)]\cos(\omega_c t)\ \{B\cos(\omega_c t)+C\cos(2\omega_c t)\}$

$=[A\cos(\omega_c t)+ m(t)\cos(\omega_c t)])\ \{B\cos(\omega_c t)+C\cos(2\omega_c t)\}$

$=AB\cos^2(\omega_c t)+AC\cos(\omega_c t)\cos(2\omega_c t)+B\,m(t)\cos^2(\omega_c t)+C\,m(t)\cos(\omega_c t)\cos(2\omega_c t)$

$=AB\cos^2(\omega_c t)+AC\cos(\omega_c t)[2\cos^2(\omega_c t)-1]+B\,m(t)\cos^2(\omega_c t)+C\,m(t)\cos(\omega_c t)[2\cos^2(\omega_c t)-1]$

$= AB\cos^2(\omega_c t)+2AC\cos^3(\omega_c t) - AC\cos(\omega_c t)+ B\,m(t)\cos^2(\omega_c t)+ 2C\,m(t)\cos^3(\omega_c t) - C\,m(t)\cos(\omega_c t)$

$=0.5AB[\cos(2\omega_c t)+1]+2AC[0.75\cos(\omega_c t)+0.25\cos(3\omega_c t)] + B\,m(t)[\cos(2\omega_c t)+1]+ 2C\,m(t)[0.75\cos(\omega_c t)+0.25\cos(3\omega_c t)] - C\,m(t)\cos(\omega_c t)$

$=0.5AB\cos(2\omega_c t) + 0.5+ 1.5AC\cos(\omega_c t)+0.5AC\cos(3\omega_c t)]+ B\,m(t)\cos(2\omega_c t) + B\,m(t) +1.5C\,m(t)\cos(\omega_c t) + 0.5C\,m(t)\cos(3\omega_c t) - C\,m(t)\cos(\omega_c t)$

$=0.5AB\cos(2\,\omega_c t) + 0.5 + 1.5AC\cos(\omega_c t) + 0.5AC\cos(3\omega_c t)] + B\,m(t)\cos(2\,\omega_c t) + B\,m(t) + 0.5C\,m(t)\cos(\omega_c t) + 0.5C\,m(t)\cos(3\omega_c t)$

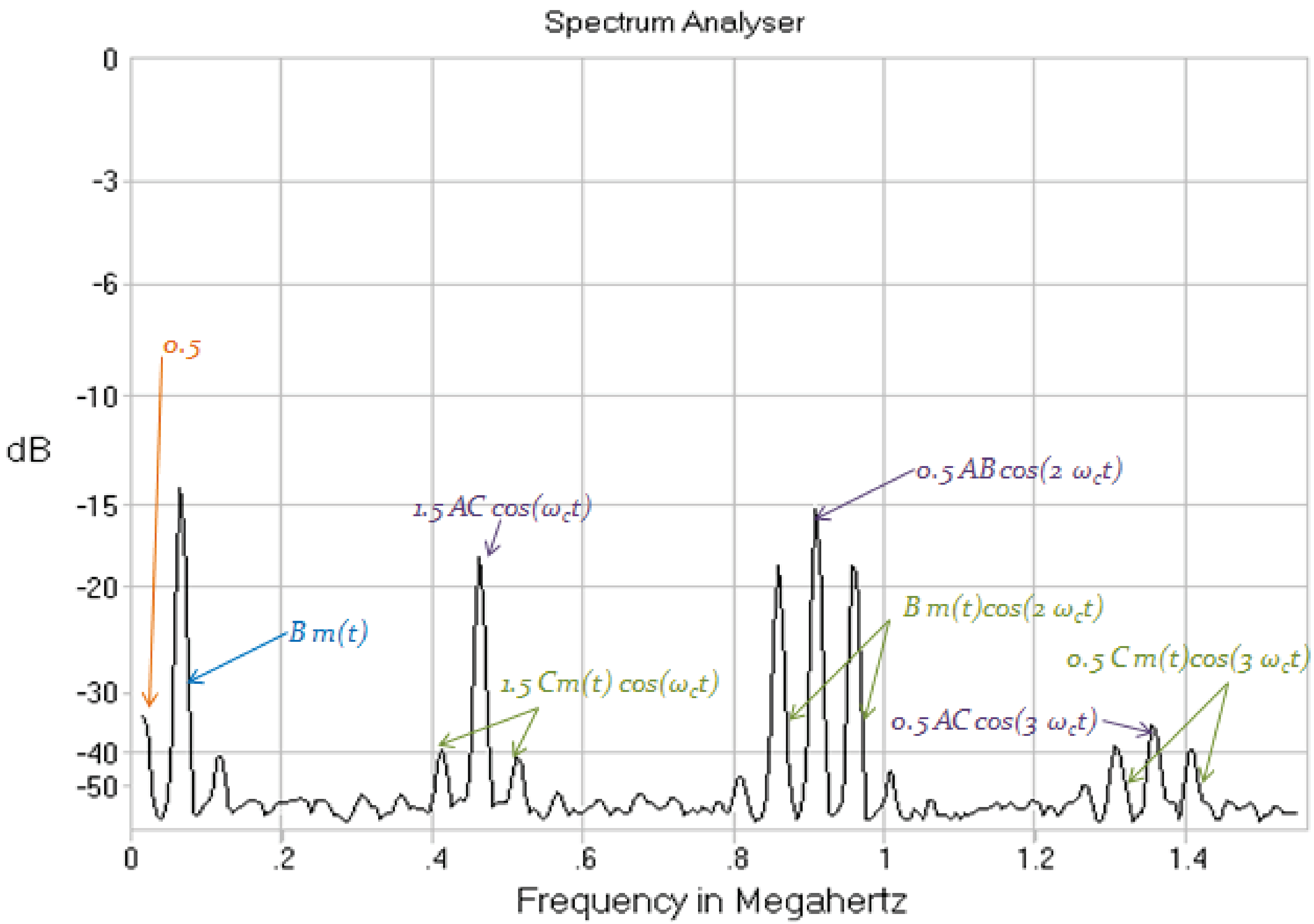
Rearranging the terms in the above equation, we get:

Output of the product detector

$$= 0.5 + 1.5\,AC\cos(\omega_c t) + 0.5\,AB\cos(2\,\omega_c t) + 0.5\,AC\cos(3\omega_c t) + B\,m(t) + 1.5\,C\,m(t)\cos(\omega_c t) + Bm(t)\cos(2\,\omega_c t) + 0.5\,Cm(t)\cos(3\omega_c t)$$

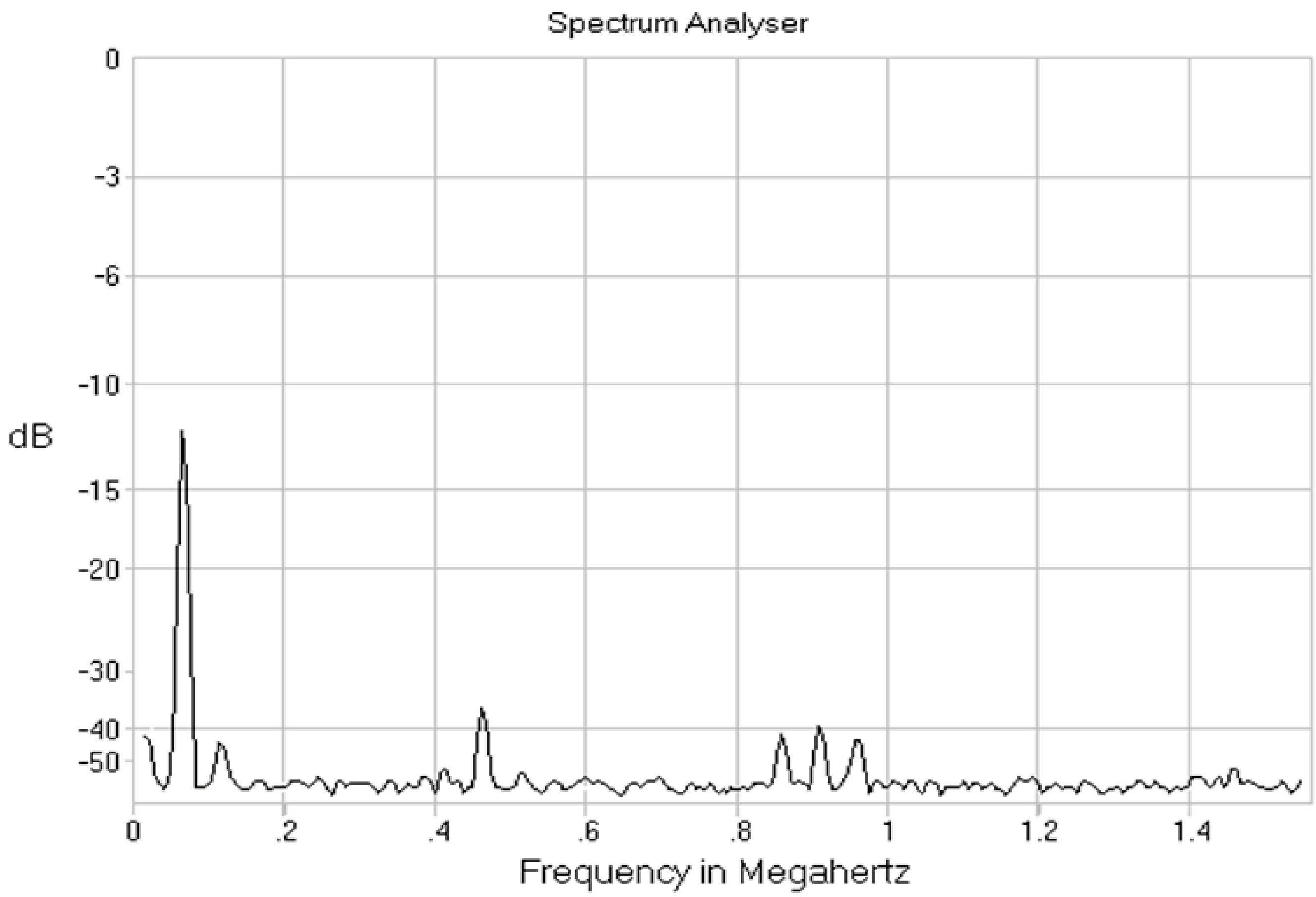
Each term in the above equation will contribute to the spectrum seen at the output of product detector.

The first term is a DC shift.
The next three terms are pure tones at ω_c, $2\omega_c$, & $3\omega_c$.
The fifth term is message signal spectrum
and the last three terms are spectrums of m(t) DSB-SC modulated at ω_c, $2\omega_c$, & $3\omega_c$.



Output of the Product Detector with all the Spectral components as derived in the equation above

By passing the output of product detector though a low pass the, *m(t)* signal's spectrum can be isolated from the rest of the spectrum. Obviously, there will be a DC shift in *m(t)* due to the 0.5 term which can be easily removed by passing the filter output through a DC blocking capacitor.



Spectrum of the product detector's output after passing it through a Low Pass Filter

Real world filters provide gradual attenuation to frequencies in the stop band. That's the reason why you can still see the component at ω_c and $2\omega_c$. The components at $2\omega_c$ are attenuated more than that at ω_c and the component at $3\omega_c$ is attenuated even more (you can't even see them). So real world filters provide a gradual roll off. Higher frequencies in the stop band suffer more attenuation compared to lower frequencies in the stop band.

Effect of phase difference between BFO and Carrier on the Demodulated Output

One other important thing to note is that carrier produced by the BFO must be in phase with the original carrier used for the modulation. The greater the Phase difference between the two the lesser will be the magnitude of the recovered *m(t)*. To see how phase difference between BFO and carrier affects the demodulated output, let's consider a BFO whose output is out of phase with the carrier by an amount 'δ'.

$$AM = [A + m(t)]\cos(\omega_c t)$$

$$\text{BFO output} = B\cos(\omega_c t + \delta)$$

Now

Output of the Product Detector

$= AM \times BFO\ output$

$= [\ A + m(t)\ ]\cos(\omega_c t)\ B\cos(\omega_c t + \delta)$

Using the identity $\cos(x)\cos(y) = 0.5\ [\cos(x+y) + \cos(x-y)]$ we get,

$= 0.5B[\ A + m(t)\ ]\ [\cos(2\omega_c t + \pi) + \cos(-\delta)]$

$= 0.5B[\ A + m(t)\ ]\cos(2\omega_c t + \pi) + 0.5B[\ A + m(t)\ ]\cos(\delta)$

The first term is removed by the low pass filter so we are left with the demodulated output, so

Filtered output $= 0.5B[\ A + m(t)\ ]\cos(\delta)$

Now if the phase difference 'δ' between the BFO and Carrier is 0, then

Filtered output $= 0.5B[\ A + m(t)\ ]\cos(0) = 0.5B[\ A + m(t)\ ]$

and if the phase difference 'δ' between the BFO and Carrier is $\pi/2$, then

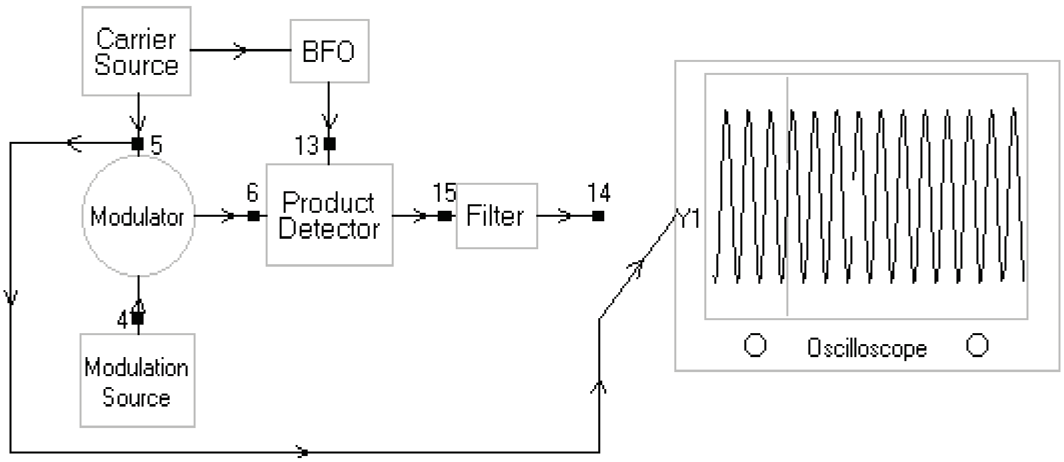
Filter output $= 0.5B[\ A + m(t)\ ]\cos(\pi/2) = 0$

So you can clearly see that as the phase difference between BFO and carrier varies from 0 to $\pi/2$, the recovered signals' amplitude falls from maximum to zero. So for best demodulated signal the BFO's output must match the carrier not only in frequency but in phase as well.

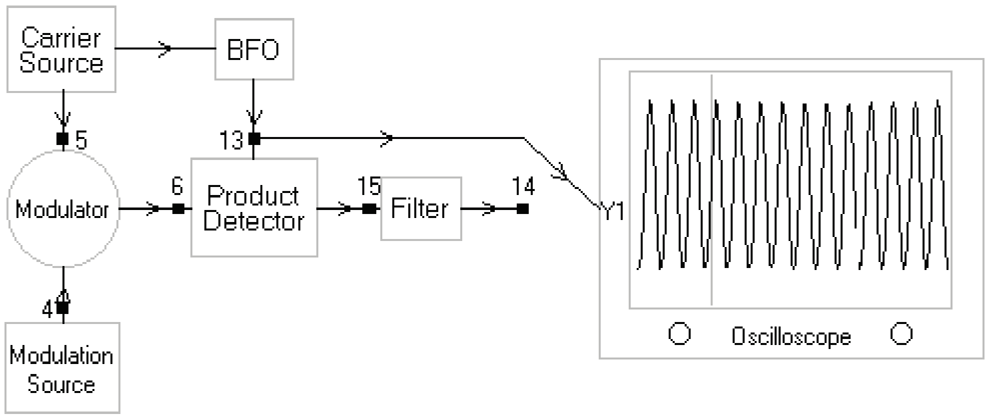
Procedure

1- Start the Discovery software
2- Go to System->Index->1- Amplitude Modulation with Full Carrier
3- Select 'yes' to Load Assignment 1
4- Start Practical 03
5- Adjust the phase of the BFO output such that it's in phase and frequency as the carrier. This is crucial for demodulation using product detector

Important: Don't rely on the software oscilloscope for phase measurement. It's very inaccurate and gives you the wrong indication of the phase. Always use the bench top oscilloscope for observing phase difference between BFO and carrier

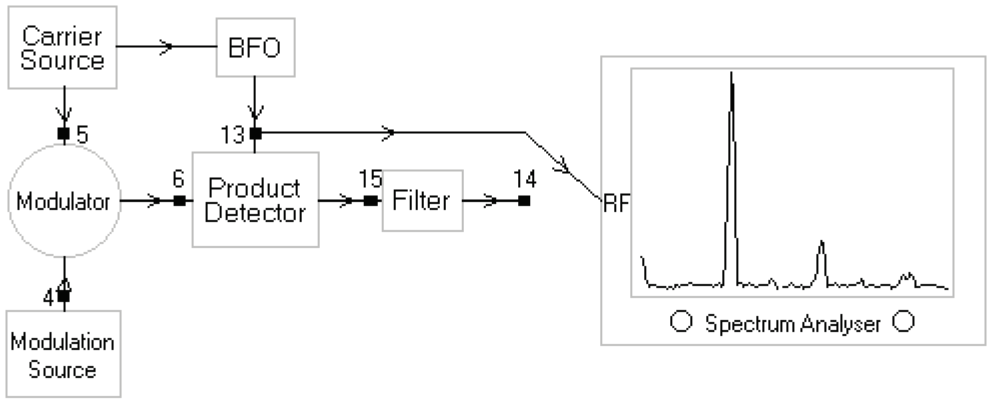


Oscilloscope's View of carrier
(Notice its phase)

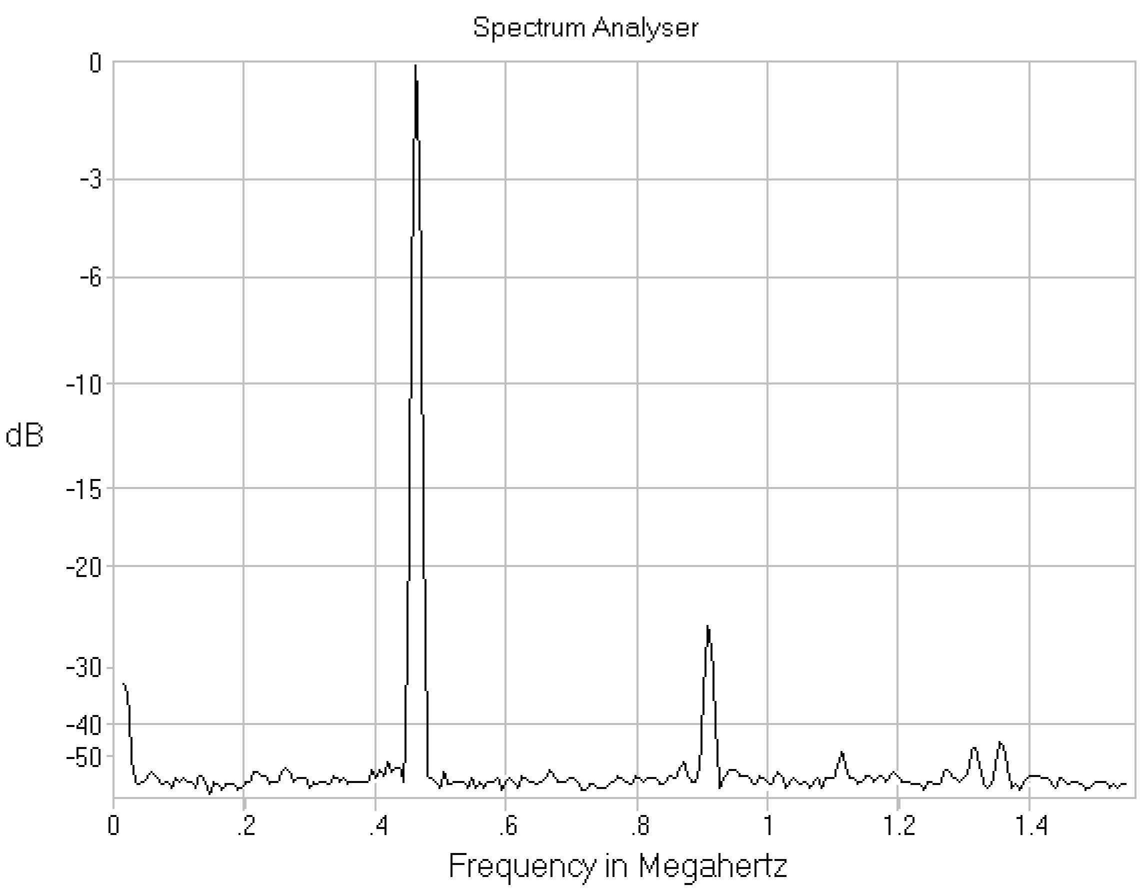


Oscilloscope's View of BFO's output
(Notice that the BFO's output is in phase with the carrier)

6- Observe the BFO on spectrum analyzer. Change the display size to see a zoomed version of it. You will see that the BFO output is not pure sinusoid but contains some harmonics as well, albeit, of very small amplitude compared to the fundamental.

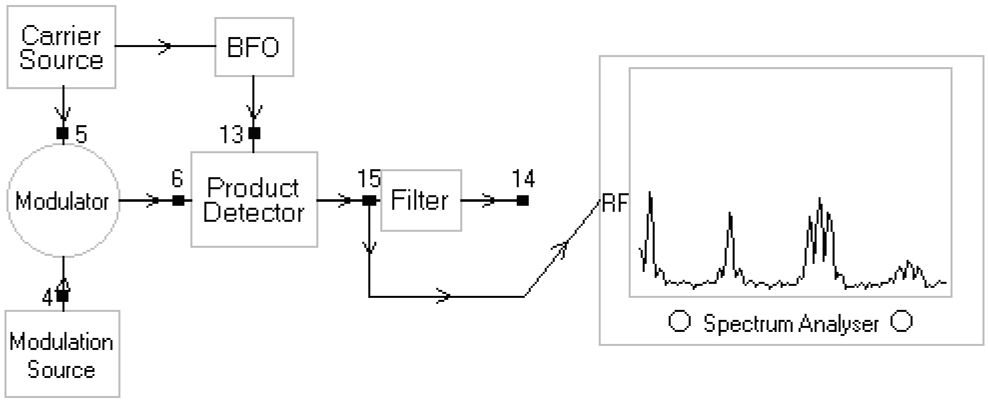


Spectrum Analyzer view of BFO's output

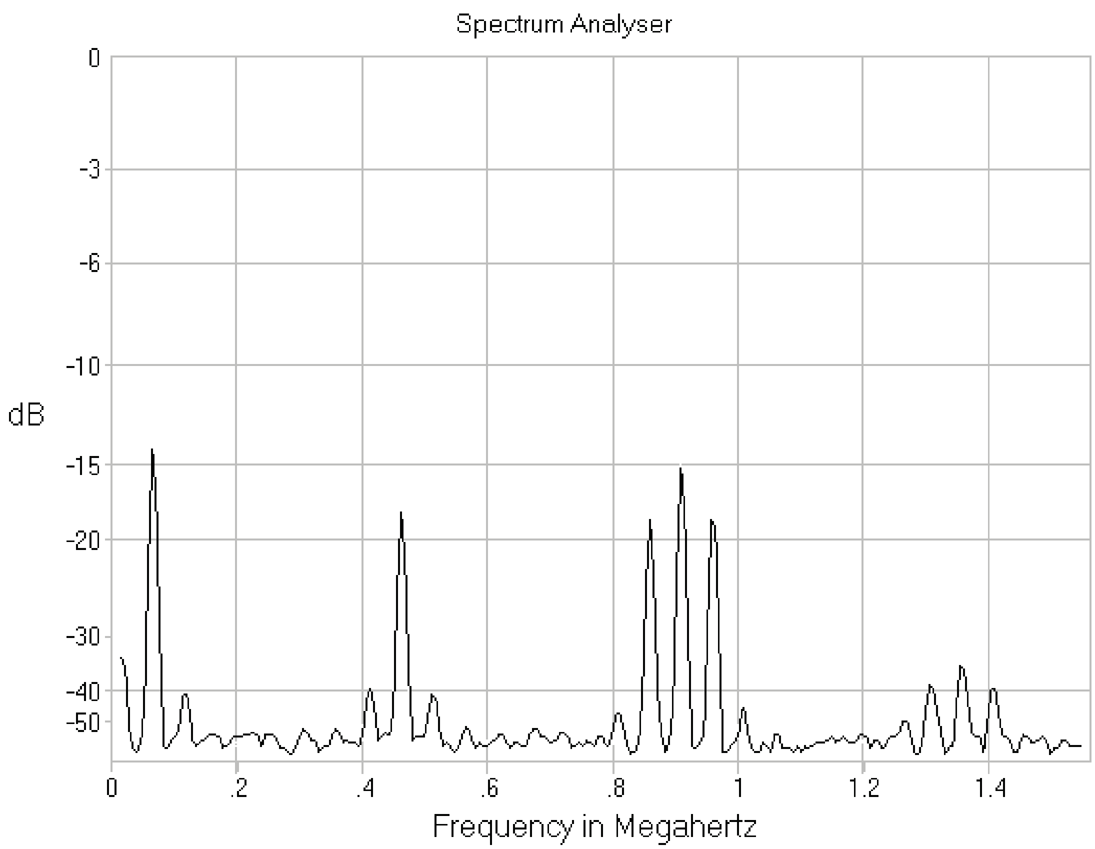


Zoomed spectrum Analyzer view of BFO's output
(You can clearly see that the BFO fails to produce a pure sinusoid. Its output contains harmonics as well)

7- Now observe the product detectors' output on spectrum analyzer. Change the display size to see a zoomed version of it.

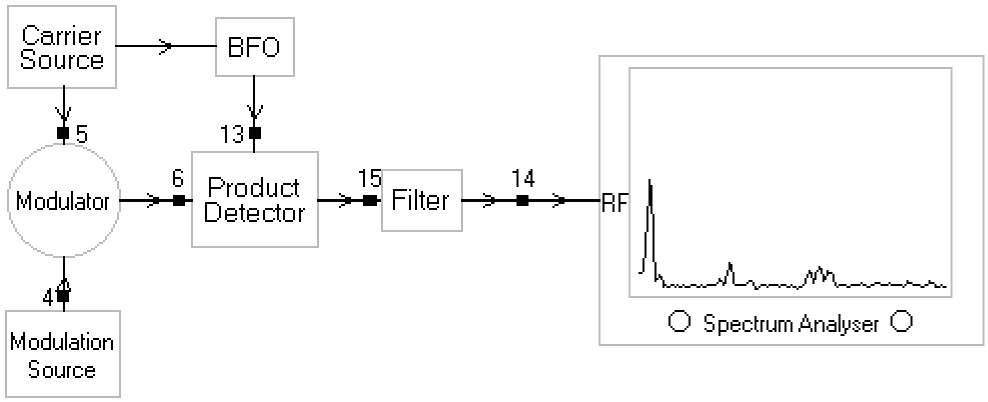


Spectrum Analyzer view of Product Detectors output

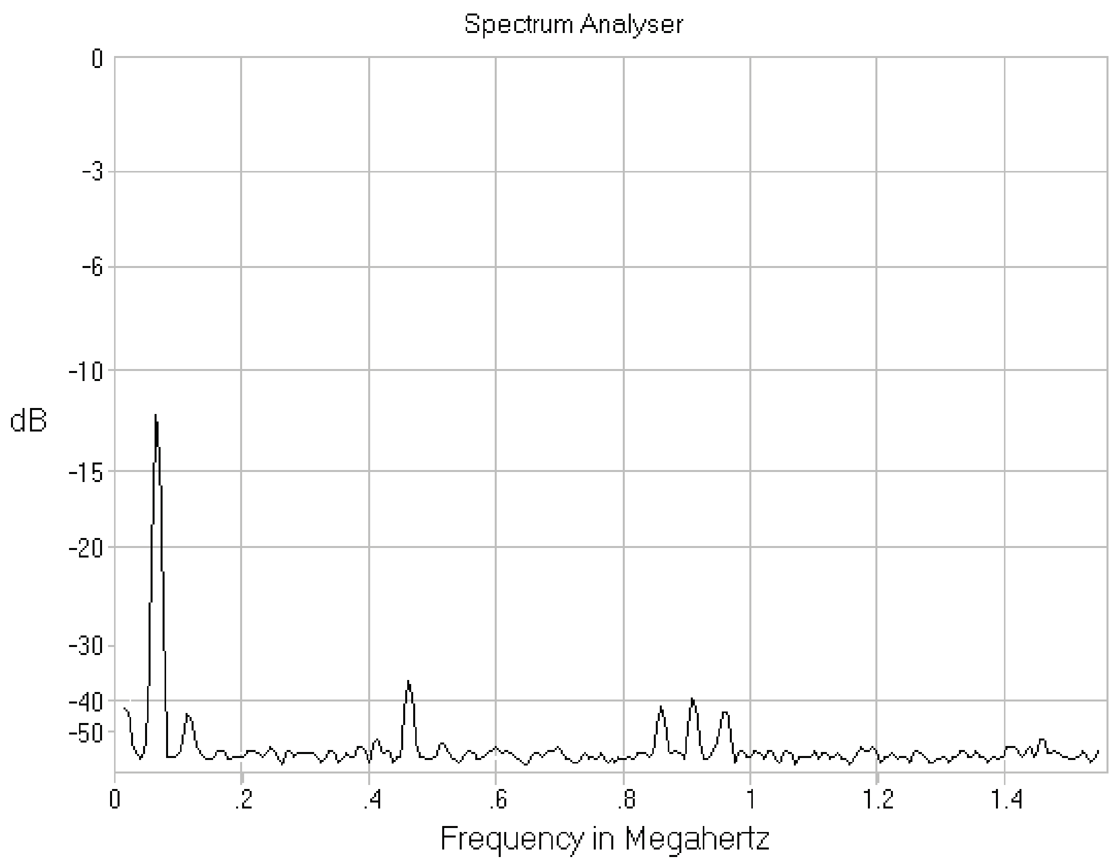


Zoomed Spectrum Analyzer view of Product Detectors output
(You can see all the components in the equation
$0.5 + 1.5\,AC\cos(\omega_c t) + 0.5\,AB\cos(2\,\omega_c t) + 0.5\,AC\cos(3\omega_c t) + B\,m(t) + 1.5\,C\,m(t)\cos(\omega_c t) + B\,m(t)\cos(2\,\omega_c t) + 0.5\,C\,m(t)\cos(3\omega_c t)$ *in the above spectrum)*

8- Now observe the output of the filter on spectrum analyzer. Change the display size to see a zoomed version of it.

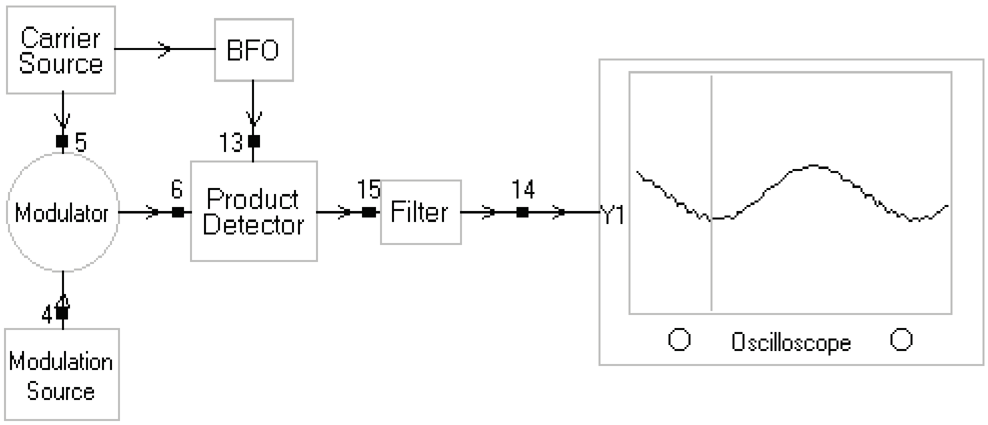


Spectrum Analyzer view of Filter's output



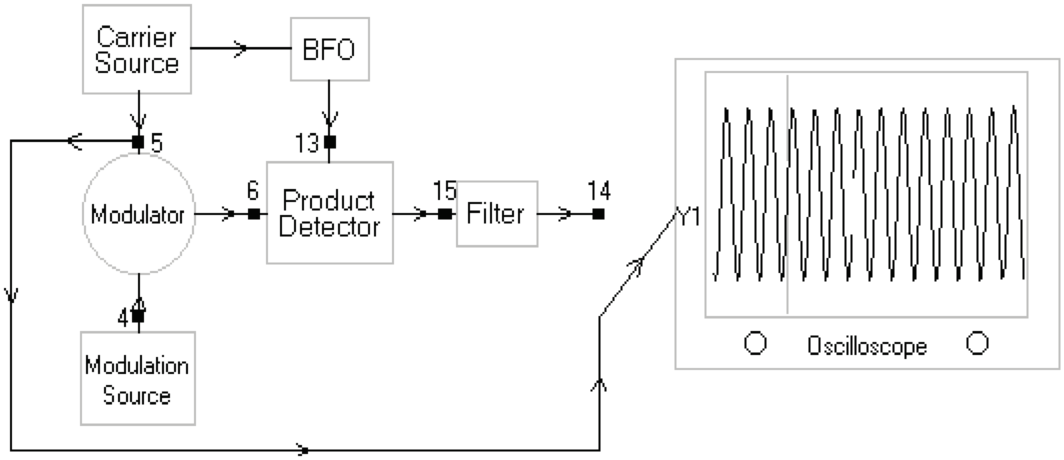
Zoomed Spectrum Analyzer view of Filter's output
(You can clearly see that the filter has attenuated the components at ω_c, $2\omega_c$, and $3\omega_c$ but failed to completely remove them. Furthermore, the filter has provided more attenuation to the frequency higher up in the stop band compared to the lower frequencies in the stop band. That's how real world filters perform)

9- Now observe the output of filter on oscilloscope. You will see that it resemble the original message signal

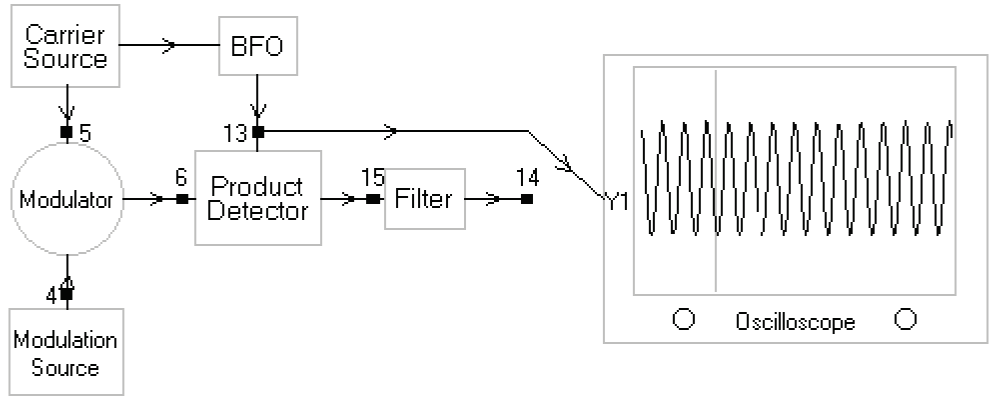


Oscilloscope's View of the recovered output
(When carrier and BFO's output are in phase with each other, maximum m(t) is recovered)

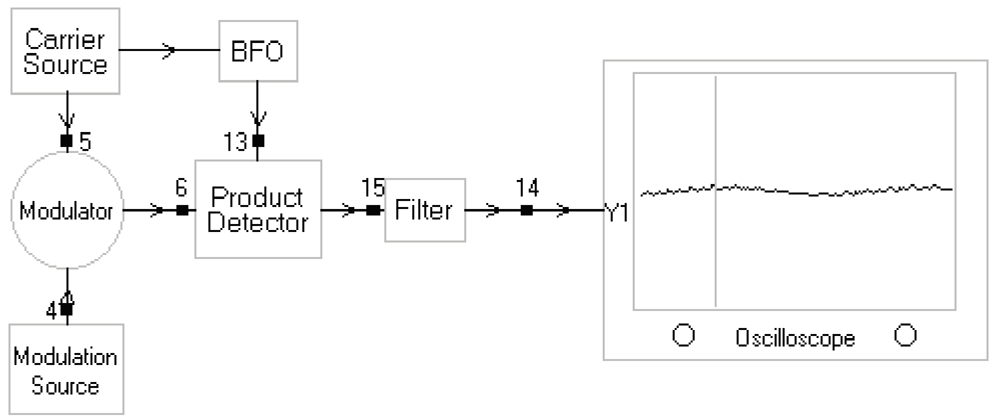
10- Now make the BFO out of phase w.r.t. the carrier using 'BFO Frequency' Knob. You will see that the filter's output will decrease. The larger that phase difference between the BFO and carrier, the smaller will be the recovered signal.



Oscilloscope's View of carrier
(Notice its phase)



Oscilloscope's View of BFO's output
(Notice that the BFO's output is out of phase with the carrier)



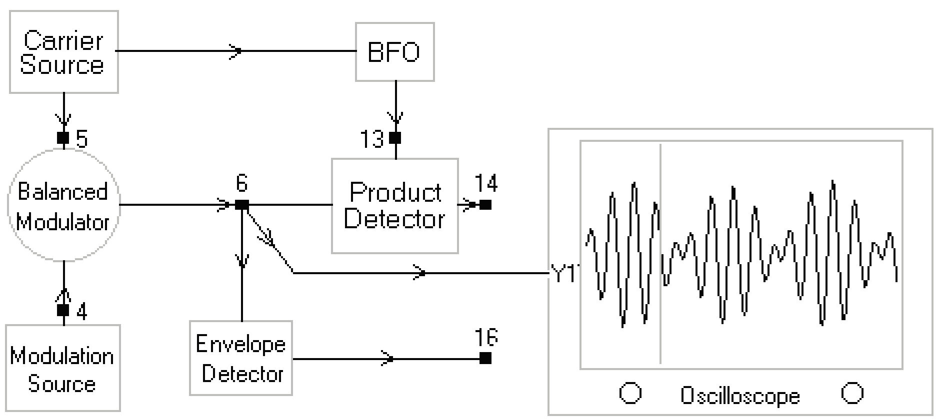
Oscilloscope's View of the recovered output
(When carrier and BFO's output are out of phase with each other, minimum m(t) is recovered)

Objectives of the Experiment

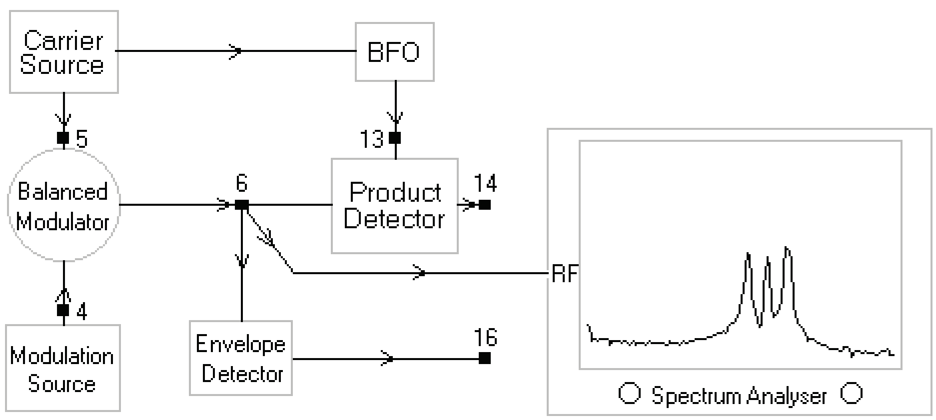
To observe the generation of DSB-SC modulated signals and its demodulation using product detector. We will also observe that DSB-SC cannot be demodulated using Envelope Detector.

Procedure

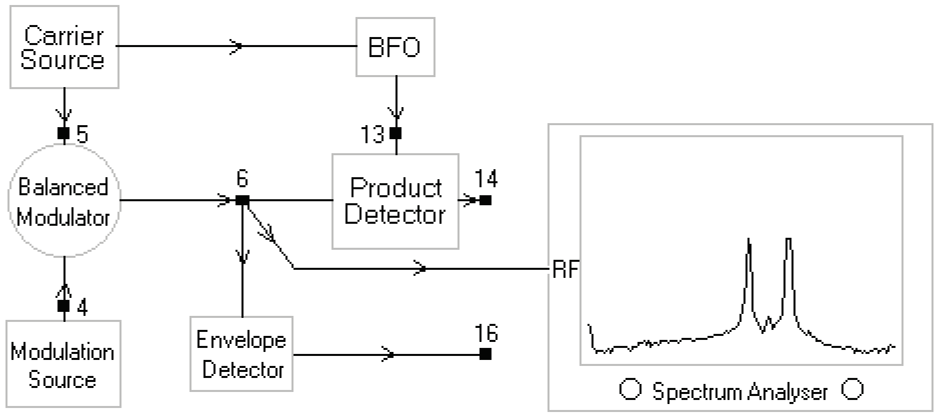
1- Start the Discovery software.
2- Go to System->Index->2-Suppressed Carrier AM
3- Select yes to Load Assignment 2
4- Start Practical 01
5- By removing the carrier from 'DSB with full carrier' modulated signal, we can get 'DSB-SC modulated' signals. To remove the carrier, observe the point 6 on spectrum analyzer. Now use the 'Carrier Balance' knob. Adjust the knob such that the carrier disappears and you see only two side bands. You have thus created a DSB-SC signal with no carrier and just two side bands



Modulated output in time domain

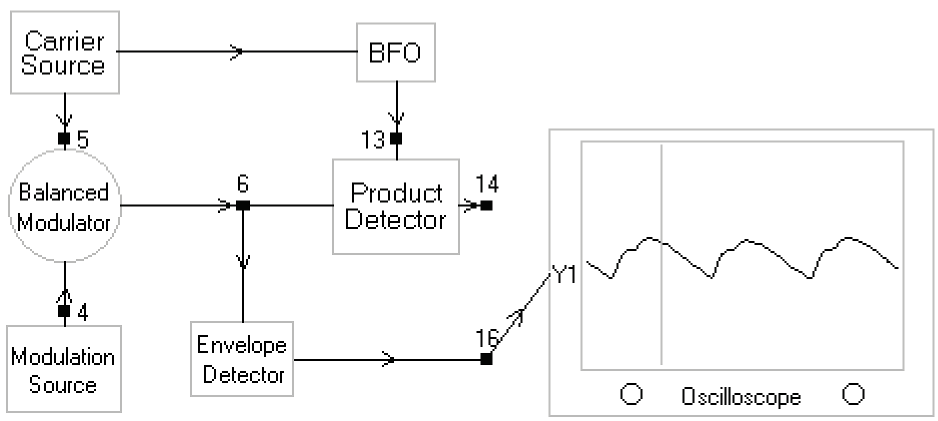


Spectrum of the Modulators' Output
(Note that carrier is present in the modulated output. To generate DSB-SC signal, this carrier component must be removed)



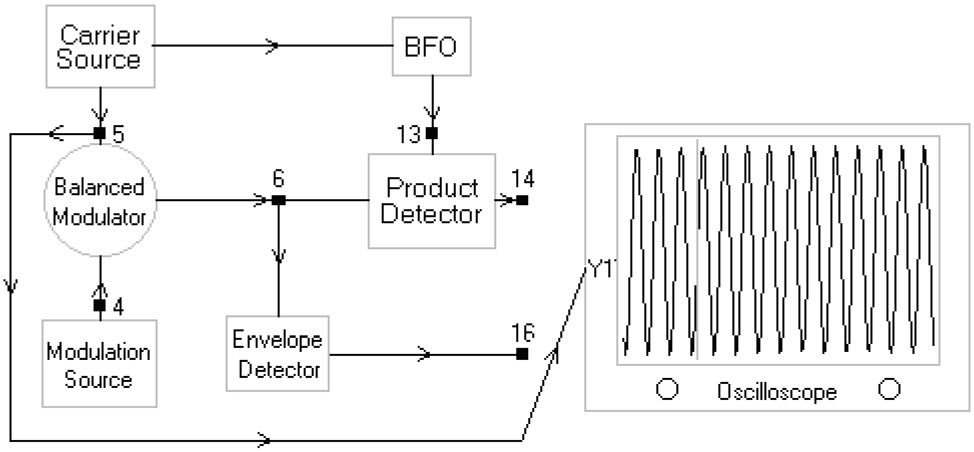
Spectrum of the Modulators' Output
(By turning the 'Carrier Balance' knob carrier can be removed completely. Once the carrier is removed you get only two side bands which constitute a DSB-SC modulated signal)

6- Now that we have produce DSB-SC signal, it's time to demodulate it. Because DSB-SC signals contain no carrier, therefore, its envelopes cross each other and half of the information gets rectified during the rectification process inside the envelope detector. That's why if you demodulate DSB-SC signals with an envelope detector, you won't be able to see the original *m(t)* at the output. Instead the output will be a full wave rectified version of it.

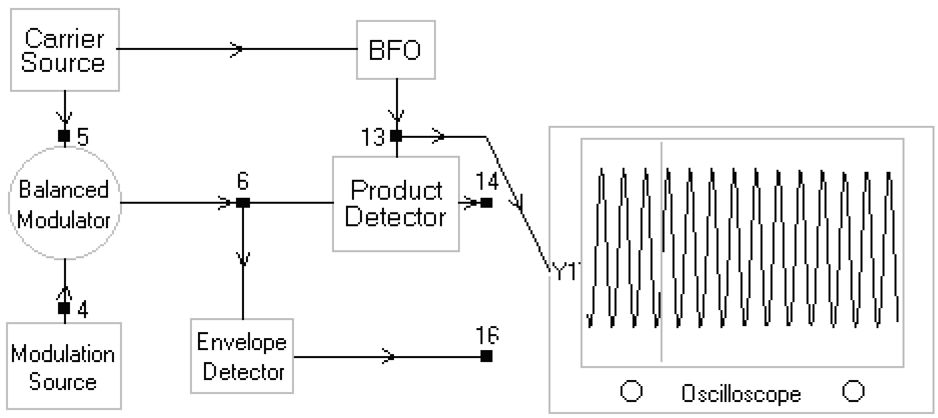


DSB-SC modulated signal Demodulated through Envelope Detector
(The recovered signal does not represent the original signal m(t) instead it presents a full wave rectified version of it)

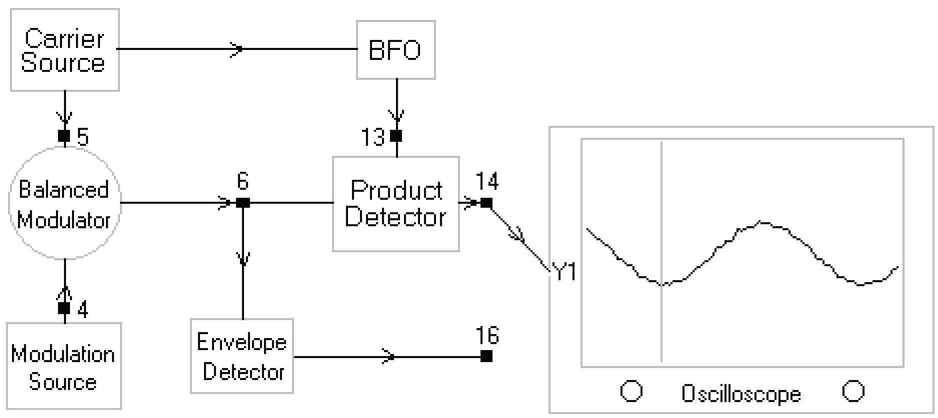
7- Product detectors on the other hand, work fine with DSB-SC signals. You can therefore, recover m(t) using product detectors. For product detection to work, the BFO and carrier must be in phase. Use 'BFO Frequency' knob to adjust the phase of BFO relative to the carrier.



Carrier in time domain



In-phase output of the BFO in time domain (Note that the output of BFO is in Phase with the original carrier which is crucial for proper demodulation of modulated signal)



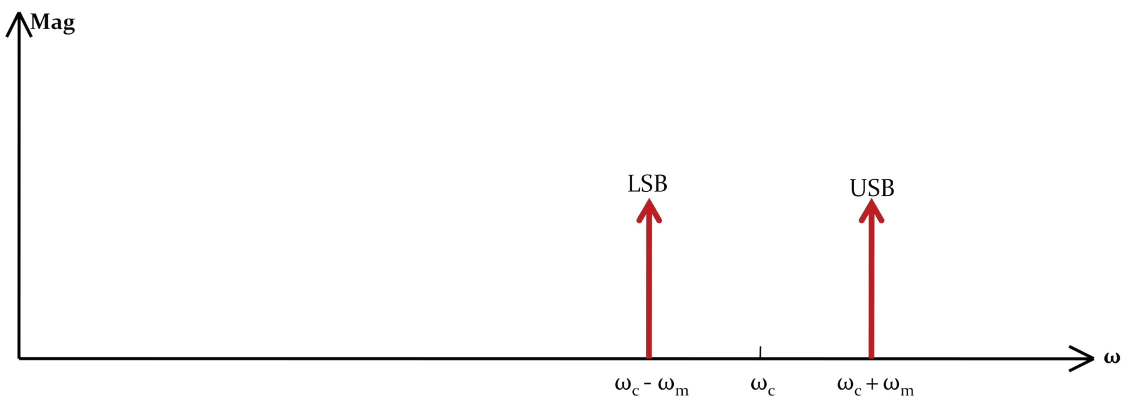
DSB-SC modulated signal Demodulated through Product Detector (The recovered signal represents the original m(t))

Objectives of the Experiment

To generate Upper Side Band and Lower Side Band modulated signals from Double Side Band Suppressed Carrier modulated signals

Background Information

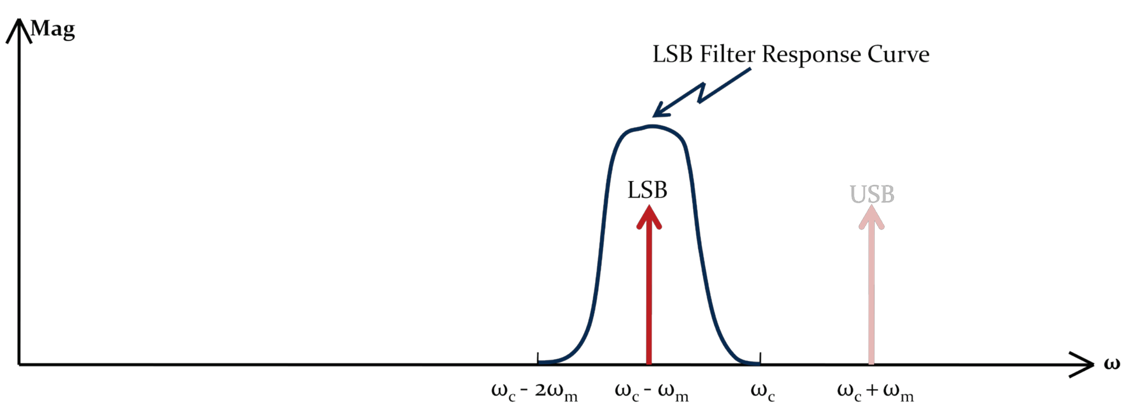
SSB signals can be generated using a number of different techniques. The technique used here is a very simple one. It involves generating DSC-SC modulated signals first. DSB-SC contains both LSB and USB. The DSB-SC signal is passed through LSB and USB filters.



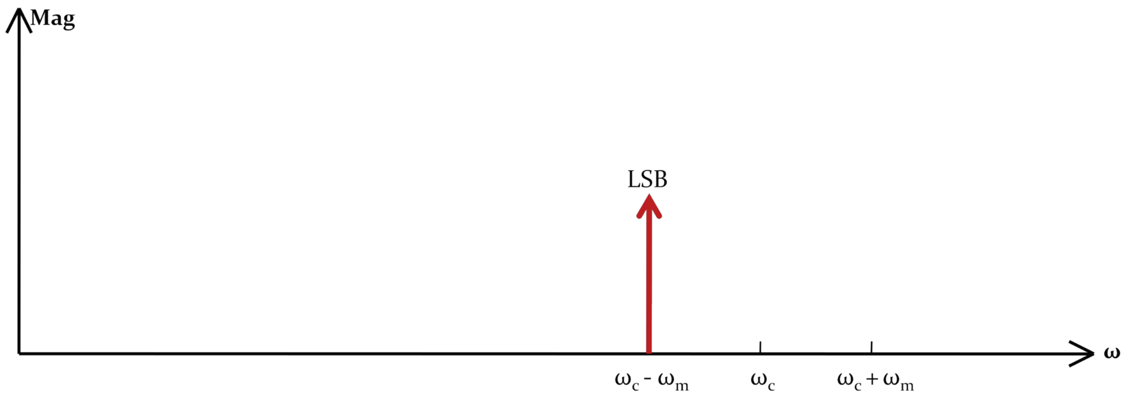
Spectrum of a pure tone ω_m Double Side Band modulated on a carrier ω_c

LSB Filter: LSB filter is essentially a Band pass filter centered at $\omega_c - \omega_m$ with stop band frequencies (also known as Corner Frequencies and denoted by f_{c1} and fc_2) at $\omega_c - 2\omega_m$ and ω_c. This filter rejects/attenuates all the frequencies that happen to be in its stop band. The USB which is present at $\omega_c + \omega_m$ is in the stop band of this filter and is thus rejected, thereby leaving us with only LSB modulated signal which is given by:

$$\psi_{LSB} = m(t)\cos(\omega_c t) + m_h(t)\sin(\omega_c t)$$



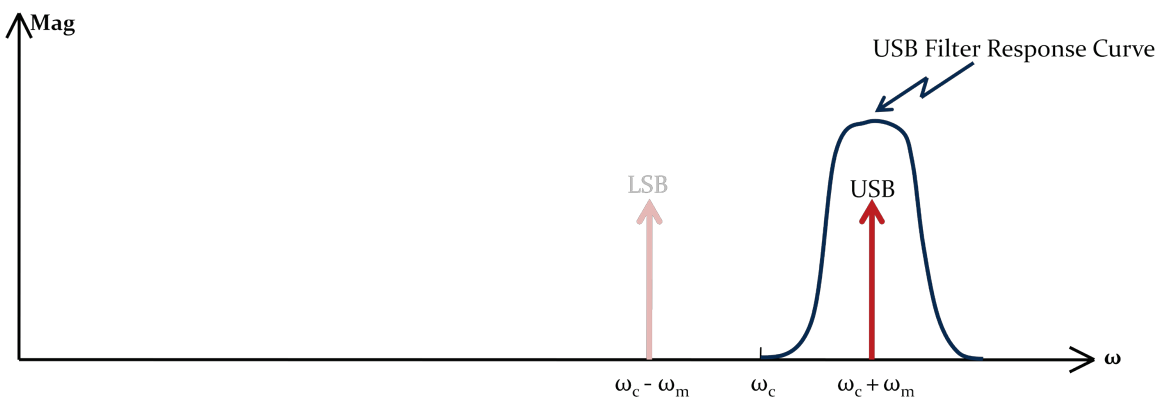
LSB Filter's Response superimposed on the Spectrum of DSB-SC modulated signal
(LSB Filter is a band pass filter centered at $\omega_c - \omega_m$ with stop band frequencies at $\omega_c - 2\omega_m$ and ω_c)



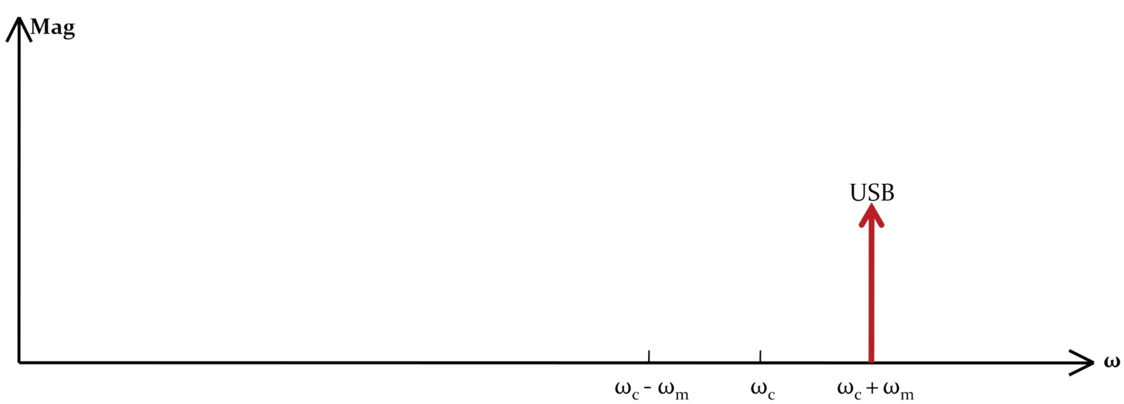
Spectrum of Lower Side Band modulated Signal $\psi_{LSB} = m(t)\cos(\omega_c t) + m_h(t)\sin(\omega_c t)$
(LSB Filter has rejected the Upper Side Band)

USB Filter: USB filter is also a Band pass filter but it's centered at $\omega_c + \omega_m$ with corner frequencies at ω_c and $\omega_c + 2\omega_m$. The LSB which is present at $\omega_c - \omega_m$ is thus rejected, thereby leaving us with only USB modulated signal which can be represented as:

$$\psi_{USB} = m(t)\cos(\omega_c t) - m_h(t)\sin(\omega_c t)$$



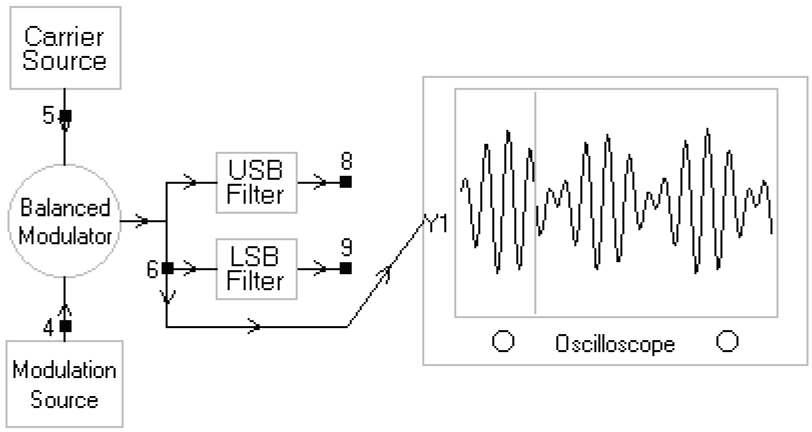
USB Filter's Response superimposed on the Spectrum of DSB-SC modulated signal
(USB Filter is centered at $\omega_c + \omega_m$ *with stop band frequencies at* ω_c *and* $\omega_c + 2\omega_m$*)*

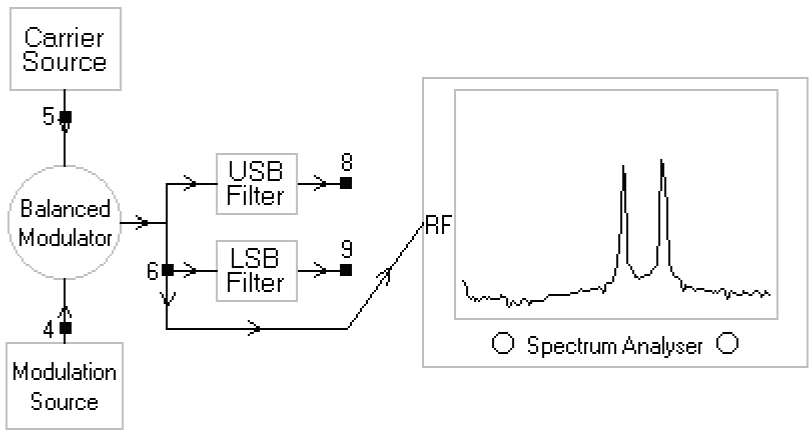


Upper Side Band modulated Signal $\psi_{USB} = m(t)\cos(\omega_c t) - m_h(t)\sin(\omega_c t)$
(USB Filter has rejected the Lower Side Band)

Procedure

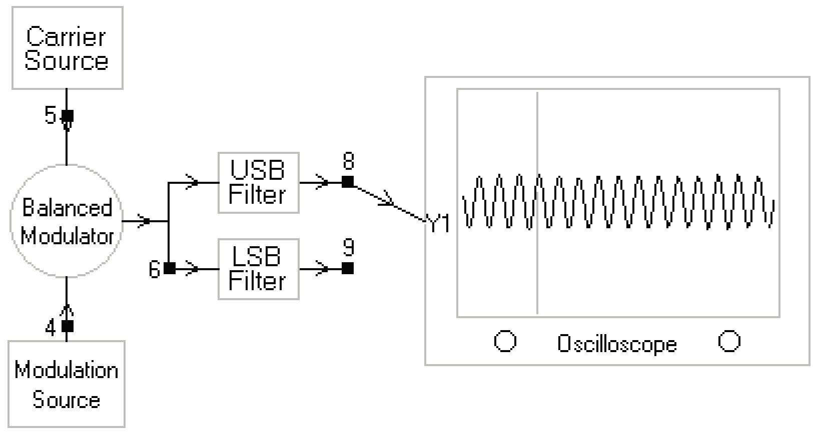
1- Start the Discovery software.
2- Go to system->Index->2-Suppressed carrier AM
3- Select yes to Load Assignment 2
4- Start Practical 02
5- Balance the output of the modulator so that its output contains no carrier. Use 'Carrier Balance' knob to remove carrier.

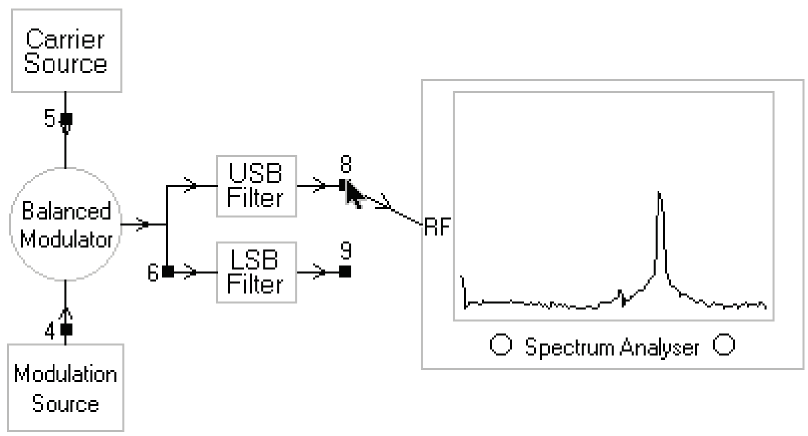




DSB-SC Modulated Signal as seen on an Oscilloscope and Spectrum Analyzer
(Generation of DSB-SC signal is the first step in the production of SSB signals. Note that in the above figure, there is no carrier in the spectrum. There are just two side bands. So our modulator is properly balanced)

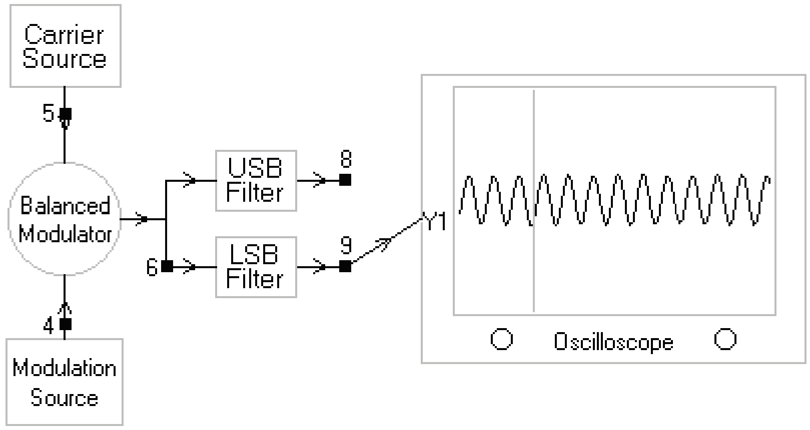
6- After the generation of DSB-SC signals, proper band pass filters are used to reject one of the sidebands
7- To reject LSB, USB Filter is used. Observe the output of USB filter in the time domain and then in frequency domain

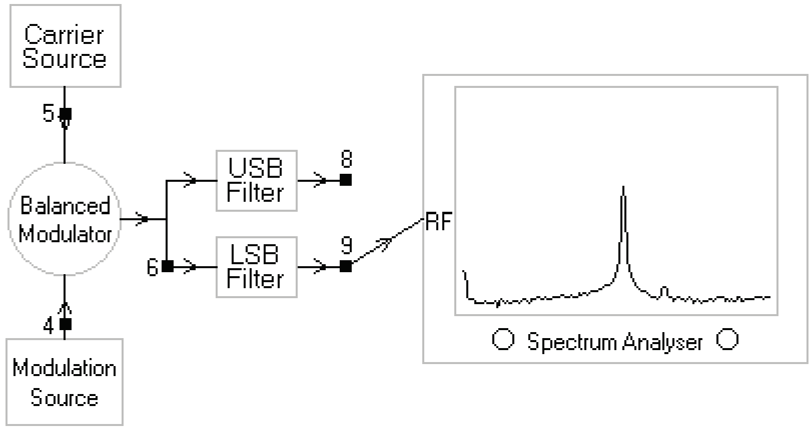




Upper Sideband Modulated Signal as seen on an Oscilloscope and Spectrum Analyzer
(A band pass filter centered at $\omega_c + \omega_m$ is used to remove the Lower side band*)*

8- To reject USB, LSB Filter is used. Observe the output of LSB filter in the time domain and then in frequency domain





Lower Sideband Modulated Signal as seen on an Oscilloscope and Spectrum Analyzer
(A band pass filter centered at $\omega_c - \omega_m$ is used to remove the upper side band*)*

Question

Explain what happens to the single side band signals when the output of the balanced modulator contains a carrier as well?

Practical 06: Demodulation of SSB

Objectives of the Experiment

To demodulate SSB signals using product detector and observing the effect of phase difference between the BFO and the carrier on the demodulated output

Background Information

SSB signals can be represented as:

$$\psi_{SSB}=m(t)\cos(\omega_c t) \pm m_h(t)\sin(\omega_c t)$$

where,

$$\psi_{LSB}=m(t)\cos(\omega_c t) + m_h(t)\sin(\omega_c t)$$

$$\psi_{USB}=m(t)\cos(\omega_c t) - m_h(t)\sin(\omega_c t)$$

Let's consider that the output of BFO is $\cos(\omega_c t)$ then,

Output of product detector

$$= \psi_{SSB} \times \cos(\omega_c t)$$

$$= m(t)\cos^2(\omega_c t) \pm m_h(t)\sin(\omega_c t)\cos(\omega_c t)$$

Using the identities $\cos^2(x) = 0.5\,[1 + \cos(2x)]$ and $\sin(x)\cos(y) = 0.5\,[\sin(x+y)+ \sin(x-y)]$, we get

$$=0.5\, m(t)\,[1+ \cos(2\omega_c t)] \pm m_h(t)\sin(2\omega_c t)$$

$$=0.5\, m(t) +0.5\, m(t)\cos(2\omega_c t) \pm m_h(t)\sin(2\omega_c t)]$$

The last two terms are rejected by the low pass filter, so we get modulating signal back at the output of the demodulator. The case discussed above happens only when the BFO output has the same frequency as the original carrier used for modulation.

What happens when BFO's output is different from Carrier in frequency

Let's consider the case when the frequency of the BFO's output differs from the carrier by an amount ω_d,

$$BFO\ output= \cos[(\omega_c + \omega_d)t]$$

Output of product detector

$$= \psi_{SSB} \times \cos[(\omega_c + \omega_d)t]$$

$$= [m(t)\cos(\omega_c t) \pm m_h(t)\sin(\omega_c t)]\ \cos[(\omega_c + \omega_d)t]$$

$$= m(t)\cos(\omega_c t)\cos(\omega_c + \omega_d)t \ \pm\ m_h(t)\sin(\omega_c t)\cos(\omega_c + \omega_d)t$$

Using the identities $\cos(x)\cos(y) = 0.5\,[\cos(x+y) + \cos(x-y)]$ and $\sin(x)\cos(y) = 0.5\,[\sin(x+y) + \sin(x-y)]$ we get, Output of product detector

$$= 0.5\, m(t) \cos(2\omega_c t + \omega_d t) + 0.5\, m(t) \cos(-\omega_d t) \pm 0.5\, m_h(t) \sin(2\omega_c t + \omega_d t) \pm 0.5\, m_h(t) \cos(-\omega_d t)$$

$$= 0.5\, m(t) \cos(2\omega_c t + \omega_d t) + 0.5\, m(t) \cos(\omega_d t) \pm 0.5\, m_h(t) \sin(2\omega_c t + \omega_d t) \pm 0.5\, m_h(t) \cos(\omega_d t)$$

$$= 0.5\, m(t) \cos(\omega_d t) \pm 0.5\, m_h(t) \cos(\omega_d t) + 0.5\, m(t) \cos(2\omega_c t + \omega_d t) \pm 0.5\, m_h(t) \sin(2\omega_c t + \omega_d t)$$

The last two terms are rejected by the low pass filter. So we get,

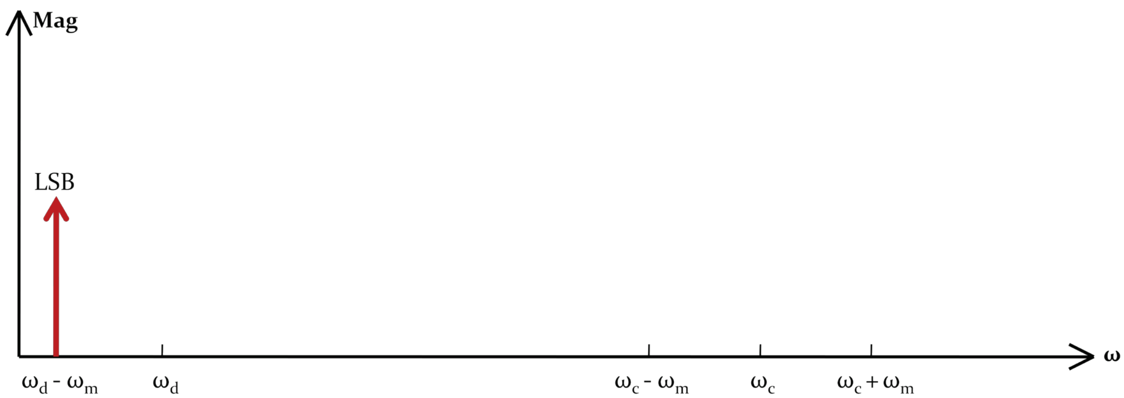
Filtered output= $0.5\, m(t) \cos(\omega_d t) \pm 0.5\, m_h(t) \cos(\omega_d t)$

So the output from LSB after product detection will be

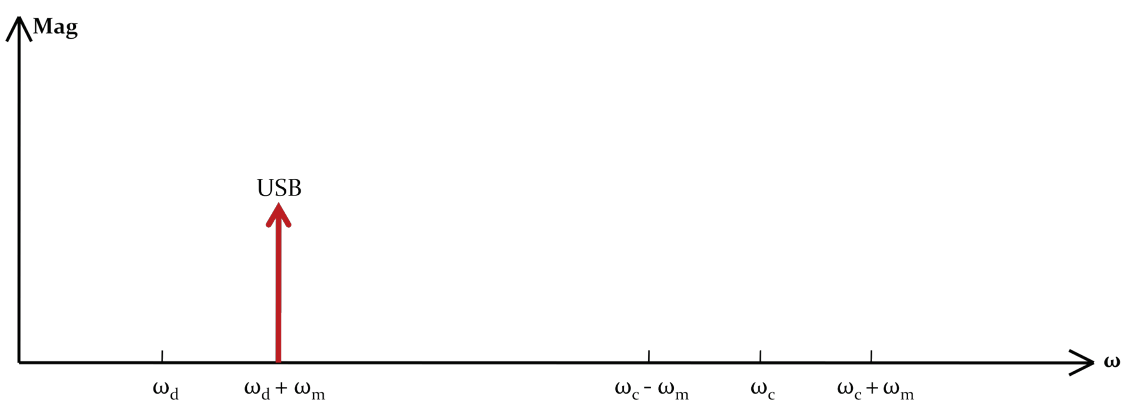
$$\psi'_{LSB} = 0.5\, m(t) \cos(\omega_d t) + 0.5\, m_h(t) \sin(\omega_d t)$$

and the output from USB after product detection will be

$$\psi'_{USB} = 0.5\, m(t) \cos(\omega_d t) - 0.5\, m_h(t) \sin(\omega_d t)$$



Spectrum of $\psi'_{LSB} = 0.5\, m(t) \cos(\omega_d t) + 0.5\, m_h(t) \sin(\omega_d t)$



Spectrum of $\psi'_{USB} = 0.5\, m(t) \cos(\omega_d t) - 0.5\, m_h(t) \sin(\omega_d t)$

Procedure

1- Start the Discovery software.

2- Go to system->Index->2-Suppressed carrier AM

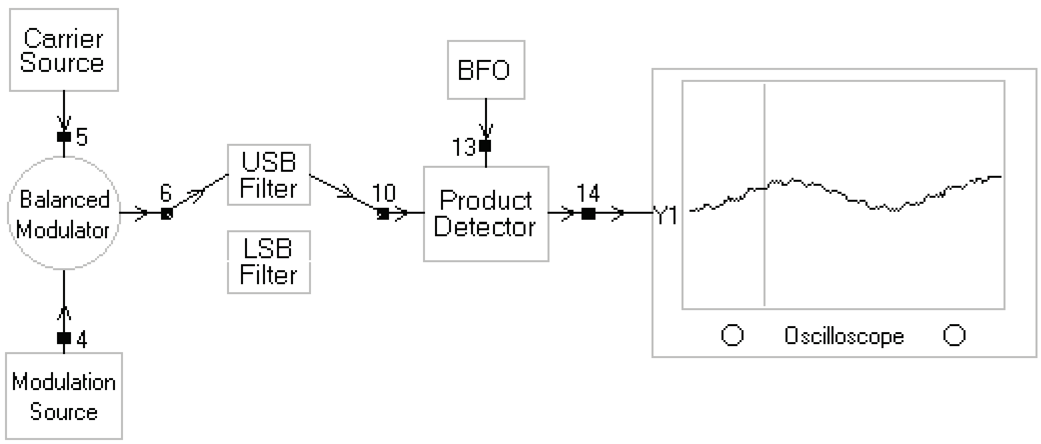
3- Select yes to Load Assignment 2

4- Start Practical 03

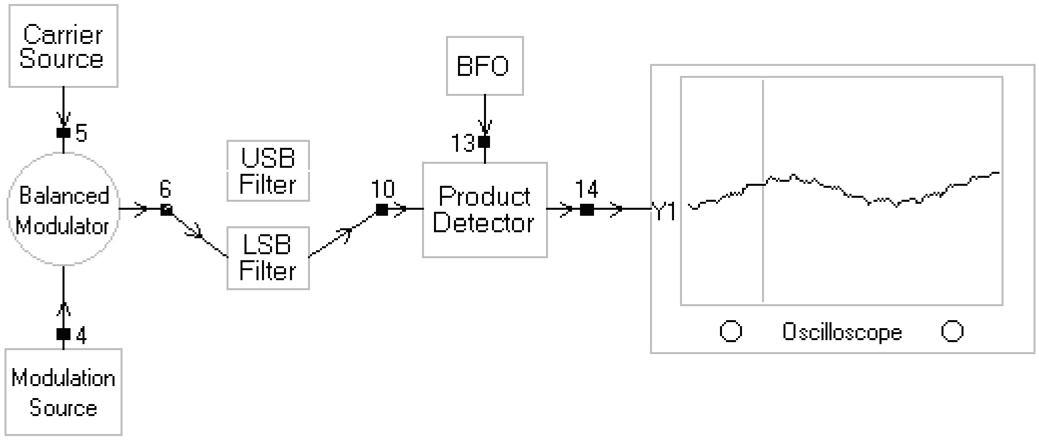
5- Now adjust the BFO output using 'BFO Frequency' knob so that its output has the same frequency(and hence phase) as the carrier

Note: For this experiment use Tri-trace bench top oscilloscope to observe the phase difference between BFO and carrier. This is because the oscilloscope in the software gives you incorrect indication of phase. So we cannot rely on software oscilloscope for phase comparison of carrier and BFO. That's why we are using the more reliable bench top oscilloscope to see the phase difference between the carrier and BFO output. Trigger mode should be set at VMODE.

6- Observe the output of the product detector. As both sidebands contain the same information, therefore, the demodulated output will remain the same no matter which sideband it's coming from. See the figures below.

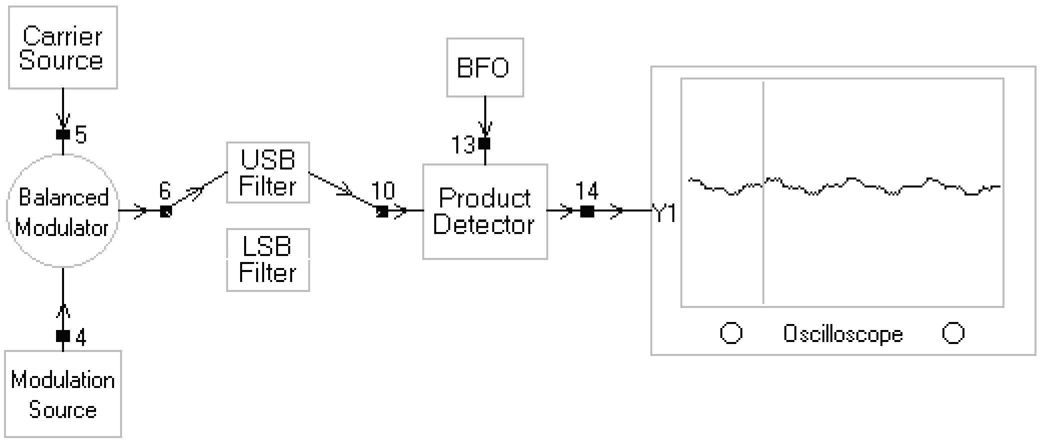


Demodulated output from the Upper Side band when BFO and carrier are in phase
(In this case the BFO and carrier are in phase and have the same frequency as well therefore, the demodulated output from both side bands is essential the same)

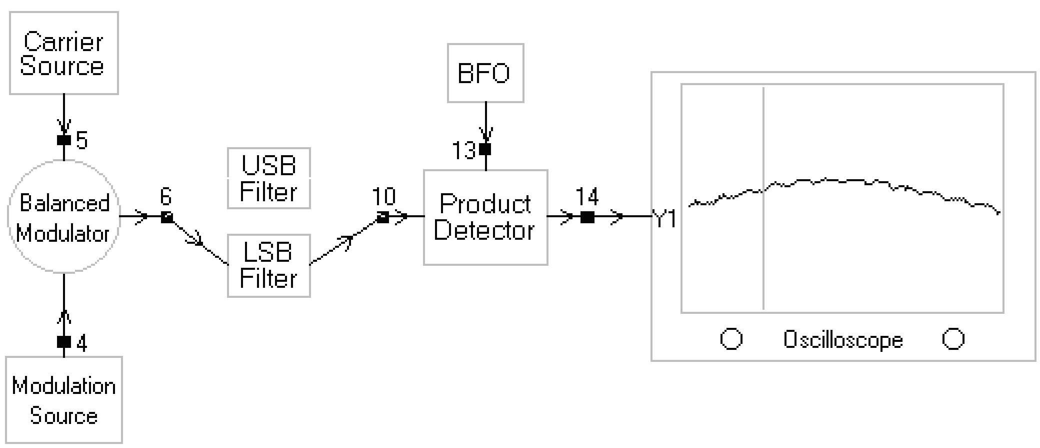


Demodulated output from the Lower Side band when BFO and carrier are in phase
(In this case the BFO and carrier are in phase and have the same frequency as well therefore, the demodulated output from both side bands is essential the same)

7- Now change the BFO frequency so that its frequency differs from carrier by an amount ω_d and observe the output from the product detector from each side band. See the figures below. The output of product detector from USB will be $\psi'_{USB} = 0.5\, m(t) \cos(\omega_d t) - 0.5\, m_h(t) \sin(\omega_d t)$ whose spectrum contain a peak at $\omega_d + \omega_m$ where as the output of product detector from LSB will be $\psi'_{LSB}=0.5\, m(t) \cos(\omega_d t) + 0.5\, m_h(t) \sin(\omega_d t)$ whose spectrum contain a peak at $\omega_d - \omega_m$. So that's why you see the high frequency output in the first figure and a low frequency output in the second figure below



Time Domain plot of $\psi'_{USB} = 0.5\, m(t) \cos(\omega_d t) - 0.5\, m_h(t) \sin(\omega_d t)$



Time Domain plot of $\psi'_{LSB} = 0.5\, m(t) \cos(\omega_d t) + 0.5\, m_h(t) \sin(\omega_d t)$

The above two plots shows demodulated output from the Upper and Lower Side bands when BFO and carrier differ in frequency by an amount ω_d *(Note that the demodulated output from both side bands is different. The spectrum of* ψ'_{USB} *is at* $\omega_d + \omega_m$ *and the spectrum of* ψ'_{LSB} is at $\omega_d - \omega_m$*. That's why the output* ψ'_{USB} *is at high frequency compared to* ψ'_{LSB}*)*

Question

Explain why SSB is more power efficient then either the AM or DSB?

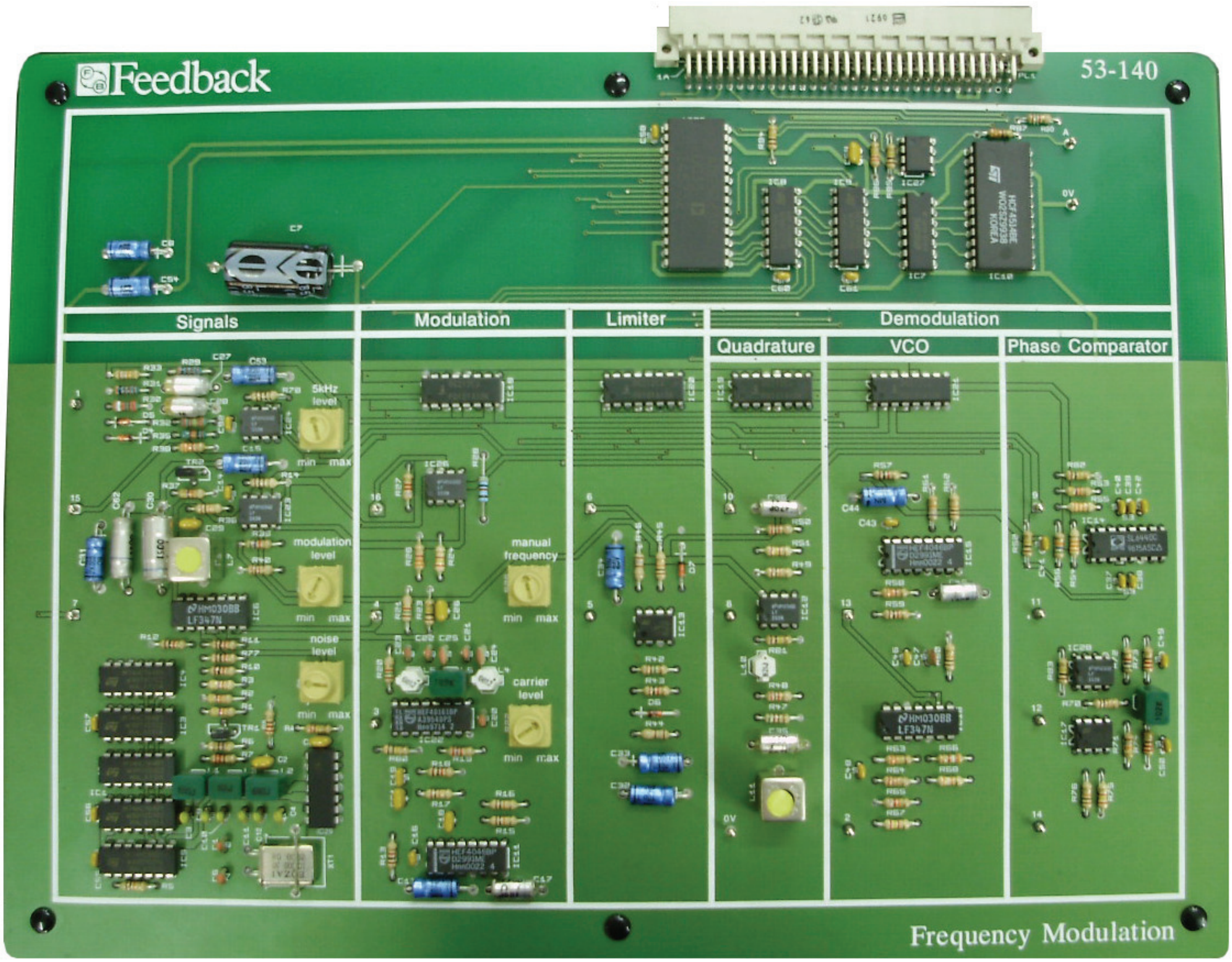
Equipments Required for Practical 07 to 10

1- Frequency Modulation **Kit 53-140**

2- Data Acquisition System (DAQ) **RAT 53-100**

3- PC with Feedback's **Discovery** Software installed

4- Tri-trace Bench top Oscilloscope **TRIO CS-1040** with pincer probes



Frequency Modulation Kit 53-140

Function of various knobs on Frequency Modulation Kit 53-140

Modualtion Level
It changes the peak value of the sinusoidal modulating signal. This knob is used to change the modulation index of the FM.

Niose
This knob is used to add Noise in modulated FM signal. This knob will not be used in any of the experiments discussed here

Manual Frequecy
This knob is only used in the Practical 7. It is used to vary the value of the varible DC supply applied to an FM modualtor

Carrier Level
It change the amplitude of the carrier signal

Practical 07: FM Modulation Concept

Objectives of the Experiment:

To find the Frequency sensitivity constant k_f of the FM Modulator

Background information

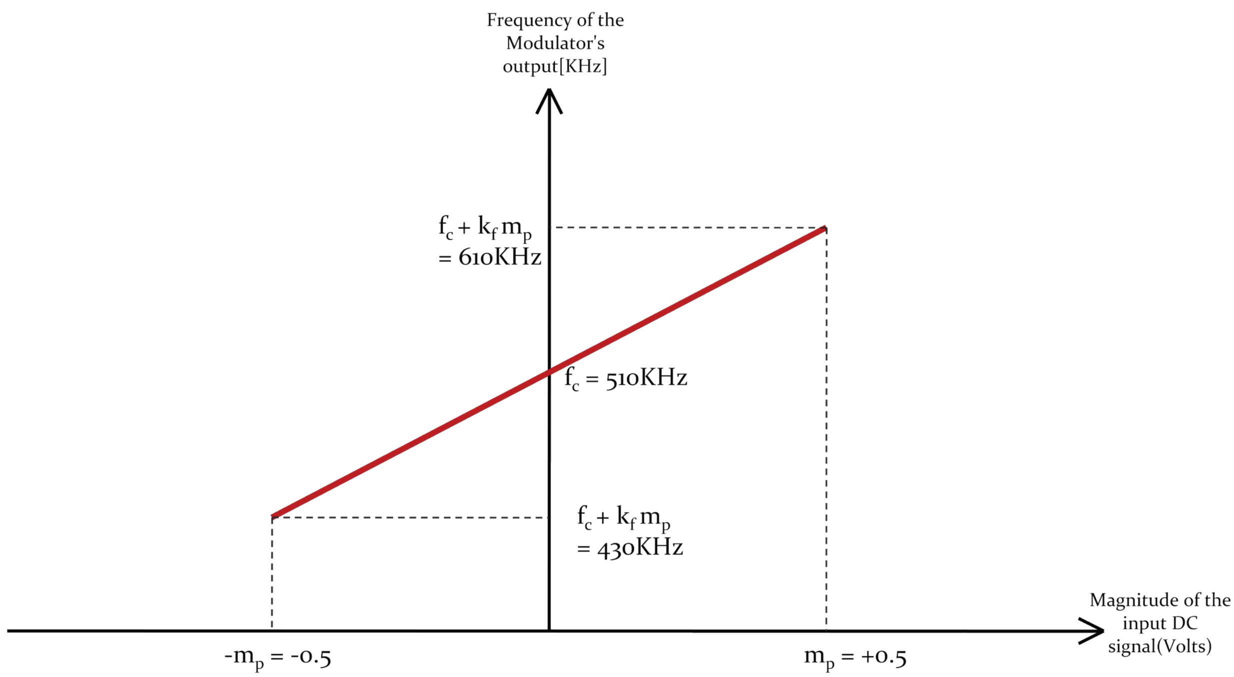
In this practical, the message signal is a variable dc source that can be set to any value from +0.5V to -0.5V. When this variable DC source is set to zero the output of the FM Modulator is a sinusoidal signal with carrier frequency. When the source is set to +0.5 volts the output of fm modulator will be a signal with frequency $f_c + k_f m_p$ and when the source is set at -0.5 V the output of FM will be a sinusoid of frequency $f_c - k_f m_p$. By plotting these three points on a graph you will get a straight curve whose slope equals the sensitivity constant k_f of the frequency modulator. As the name indicates k_f, is a constant and its value is set during the design of the modulator. Once the modulator has been designed k_f cannot be changed then.

k_f= Frequency Sensitivity of the FM modulator= Slope of the V-f Characteristic Curve

$= \Delta f/\Delta v = (f_2-f_1)/ (V_2-V_1)$

$= (610-430)/[0.5-(-0.5)]$

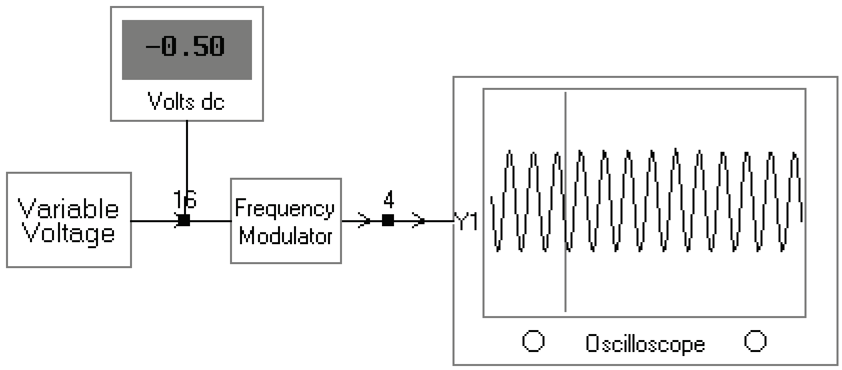
$= 180$ *KHz/V*

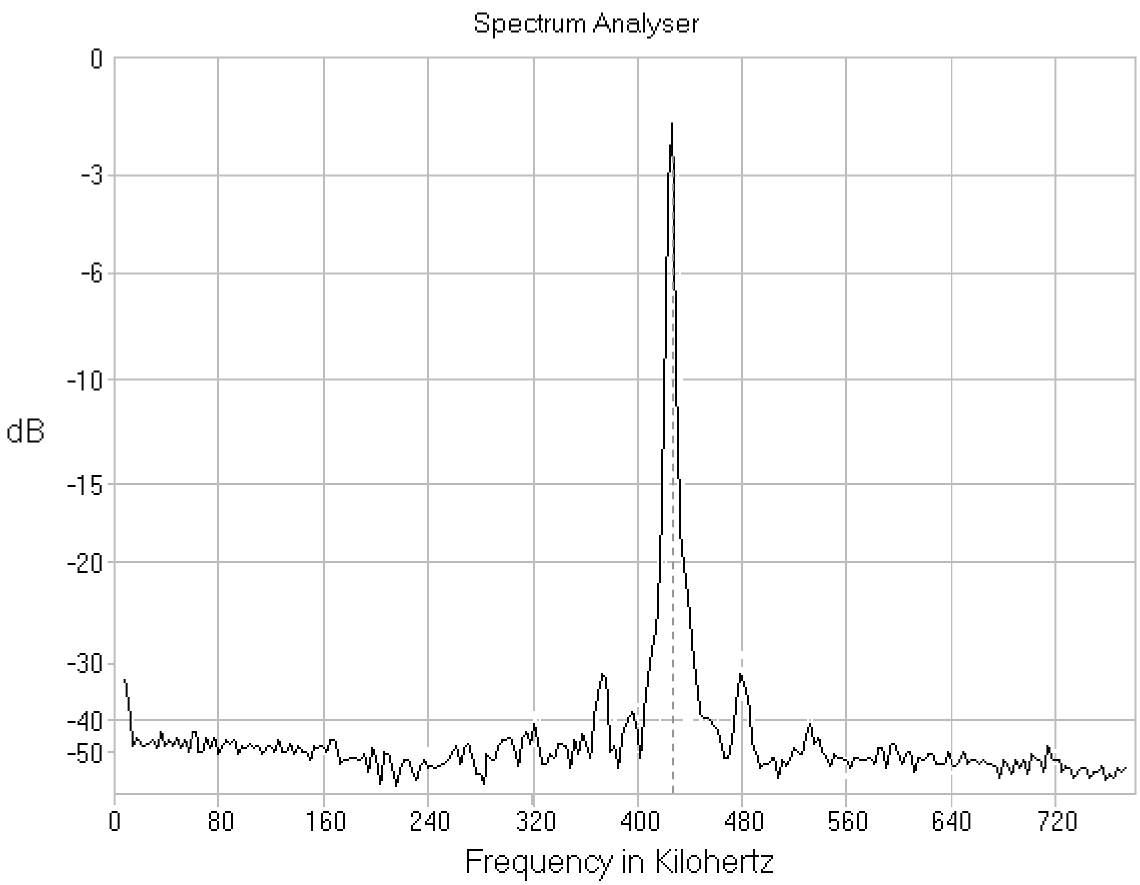


V-f Characteristic Curve of the FM Modulator

Procedure

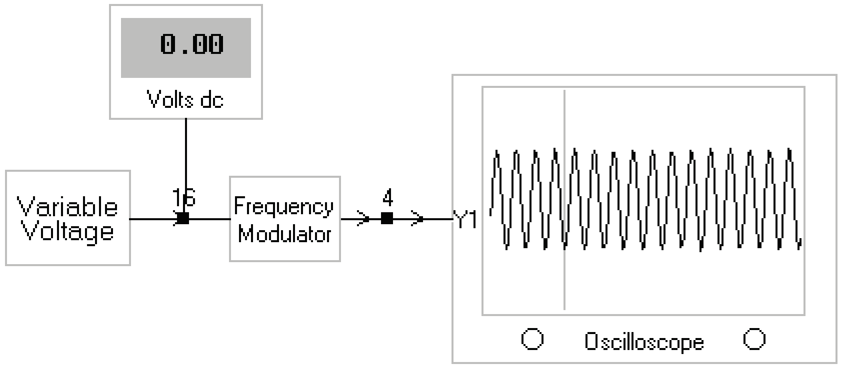
1- Start the Discovery software.
2- Go to system->index->3- Generation of Frequency Modulation
3- Select yes to Load assignment no 3
4- Start practical no 01
5- Use the *'Manual Frequency'* knob to change the output of the Variable DC source
6- First set the variable Voltage to -0.5V. Observe the output in oscilloscope and spectrum analyzer. From spectrum analyzer display calculate the frequency of the output.

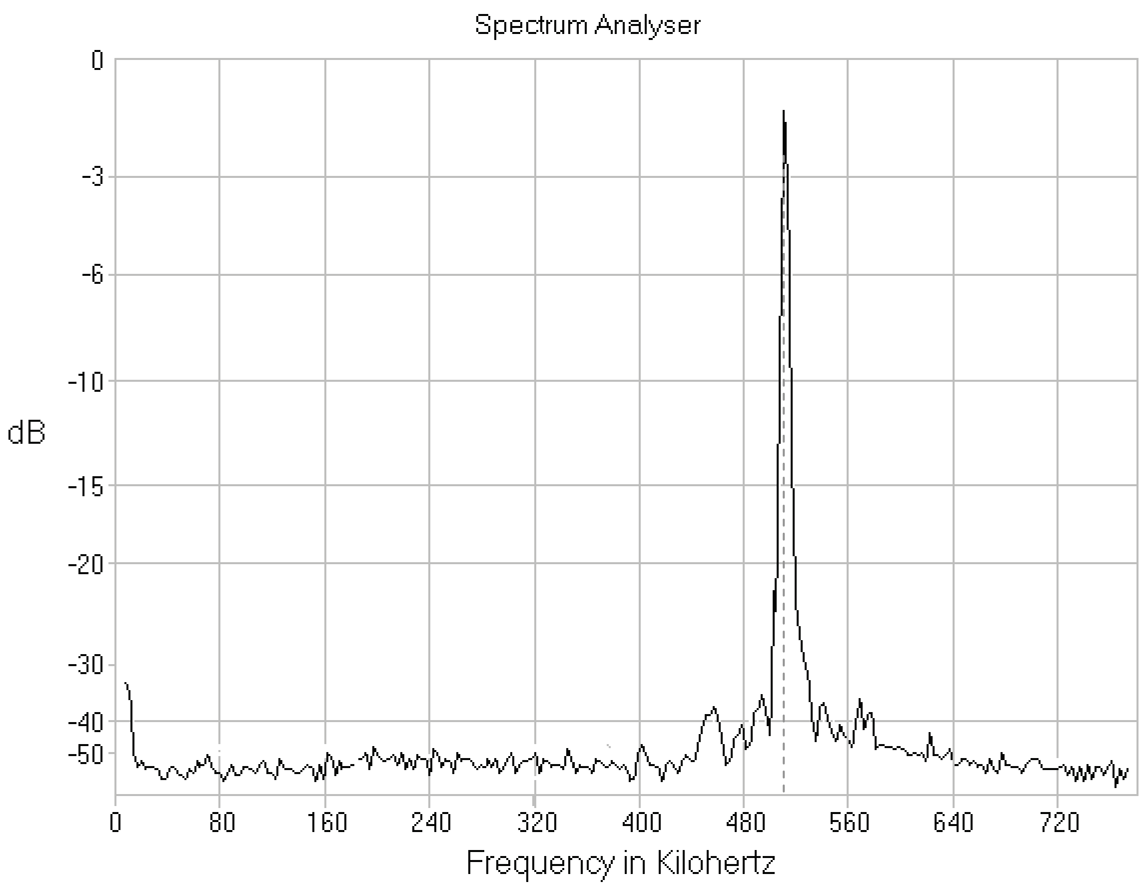




Output of FM modulator when m(t)=-0.5 V in both time and frequency domains
(From the Spectrum of FM modulators' output, you can clearly see that its peak occurs at 430 KHz)

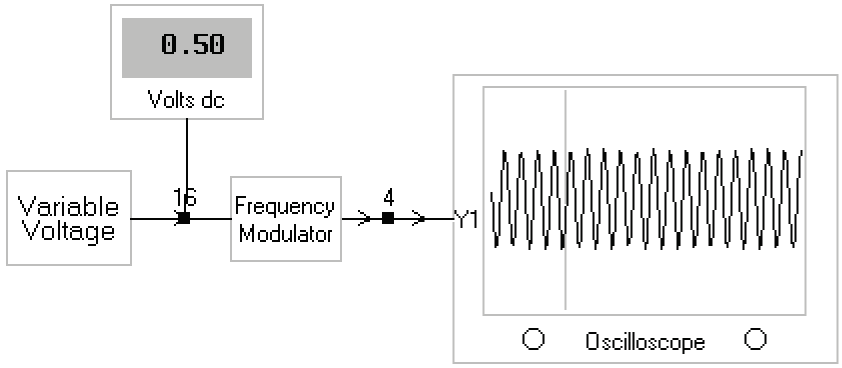
7- Now set the variable Voltage to 0.0 V. Observe the output in oscilloscope and spectrum analyzer. From spectrum analyzer display calculate the frequency of the output.

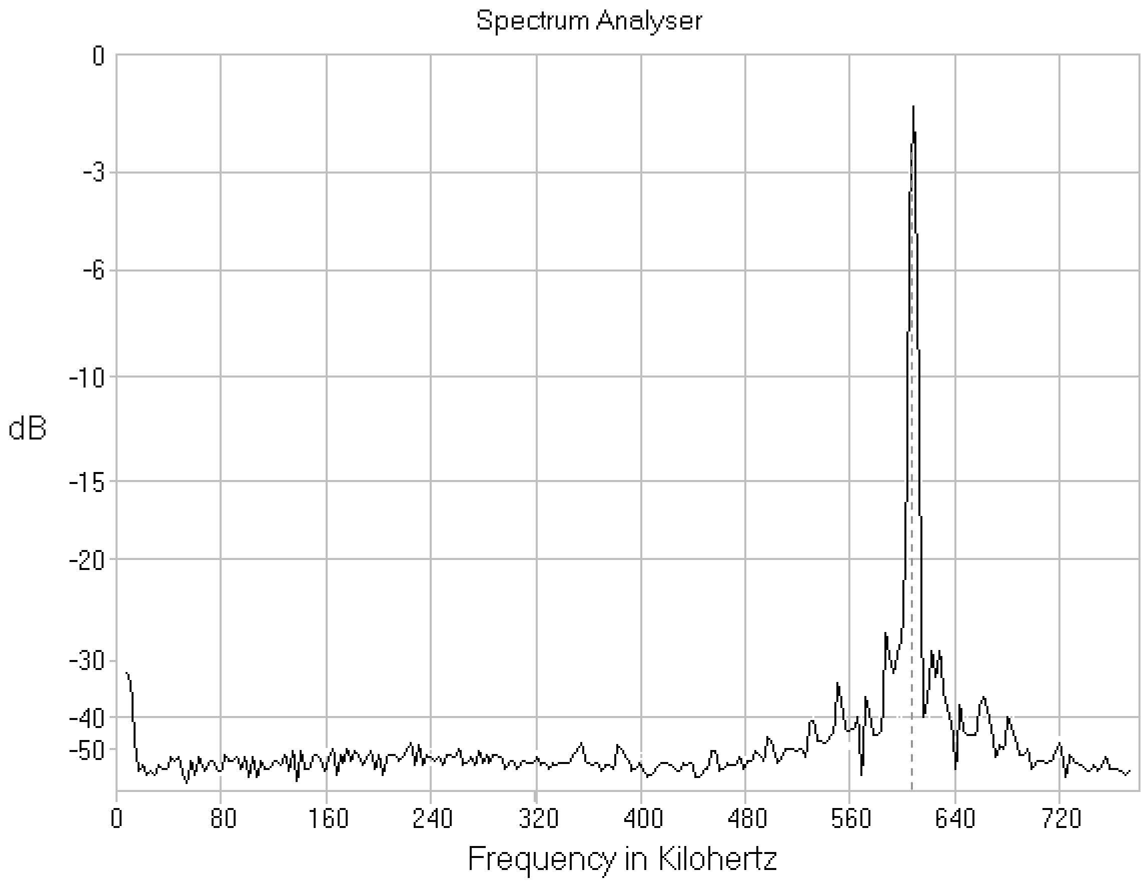




Output of FM modulator when m(t)=0.0 V in both time and frequency domains
(From the Spectrum of FM modulators' output, you can clearly see that its peak occurs at 510 KHz)

8- Now set the variable Voltage to 0.5 V. Observe the output in oscilloscope and spectrum analyzer. From spectrum analyzer display calculate the frequency of the output





Output of FM modulator when m(t)=+0.5 V in both time and frequency domains
(From the Spectrum of FM modulators' output, you can clearly see that its peak occurs at 610 KHz)

9- Plot the V-f characteristics curve of the modulator. Find its' slope to get the Frequency Sensitivity Constant k_f of the FM modulator used.

Question

What's the value of K_f for the FM Modulator used in this practical? Can it be changed?

Practical 08: A Frequency Modulator

Objectives of the Experiment

To find out the effect of varying the level of *m(t)* on the bandwidth of the FM Spectrum

Background information

In this experiment the Variable DC source is replaced by the sinusoidal source of 50 KHz (B=50 Hz). The carrier on which its being frequency modulated has a frequency of 510 KHz.

The Bandwidth of the FM signal is given by

$$BW=2(\Delta f+B)$$

where

$\Delta f=k_f\, m_p/2\pi$

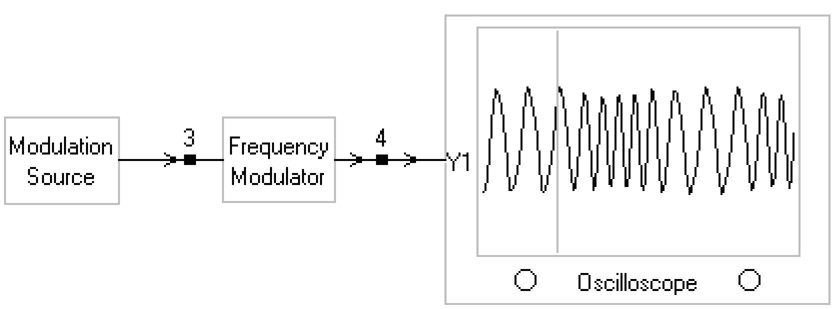
If $\Delta f >> B$ then the bandwidth is 2Δf and the FM is Wideband FM

and if $\Delta f<<B$ then the bandwidth is 2B and the FM is Narrowband FM.

Looking at the Equation $\Delta f= k_f\, m_p/2\pi$, you can see that it depends only on the peak value of the modulating signal. So that means the peak value of the message or modulating signal decides whether the FM signal is narrow band or wide band. By increasing the value of m_p slowly, you will clearly see that the FM changes from narrowband to wideband. You can easily observe it on the spectrum analyzer. You will also observe that the side bands don't increase monotonically but follow a peculiar pattern dictated by the Bessel Function.

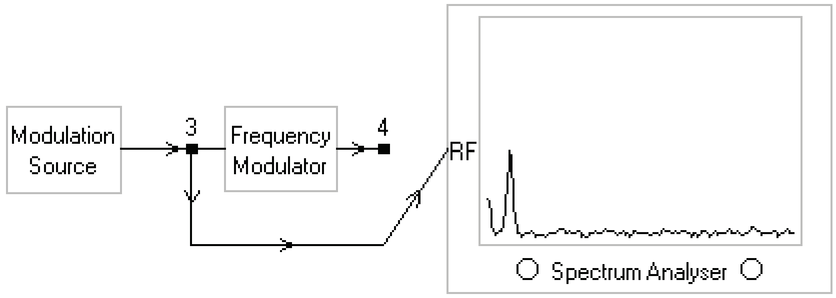
Procedure

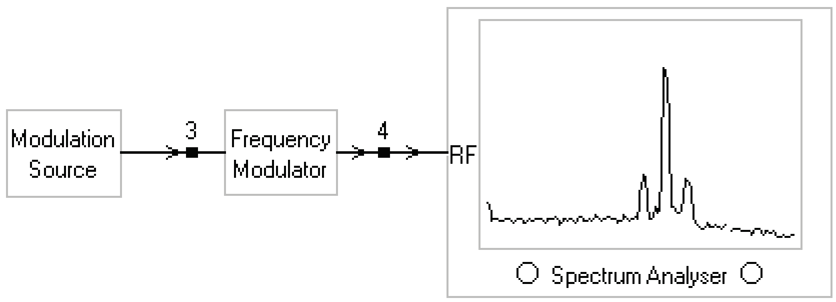
1- Start the Discovery software.
2- Go to System->Index->3- Generation of Frequency Modulation
3- Select yes to Load Assignment 3
4- Start Practical 02
5- Observe the modulated FM signal on oscilloscope



Frequency Modulated output when the modulating signal is a sinusoidal source

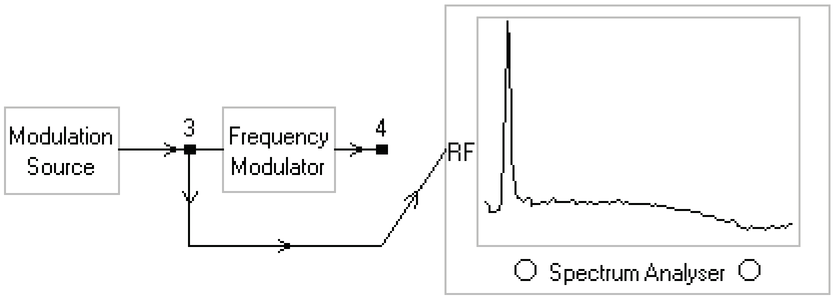
6- By turning the 'Modulation Level' knob you can change the peak value of the sinusoidal modulating signal. Observe the output of FM on Spectrum Analyze. Vary the Modulation Level. You will observe that for low modulation levels the FM will be Narrowband.

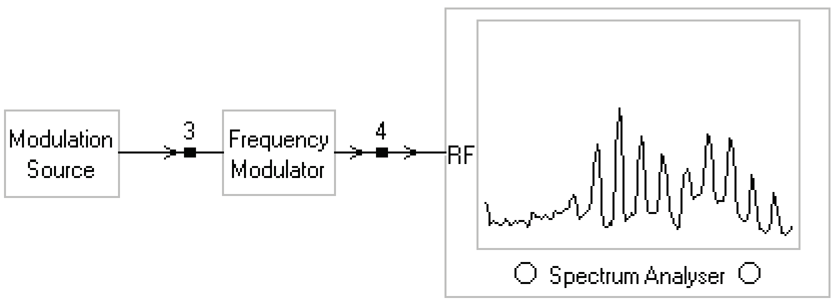




Modulation source and FM as seen on the Spectrum Analyzer
(In this particular case Δf <<B so therefore, the FM is NBFM with only two sidebands. m_p is small for NBFM as you can see in the spectral plot of m(t))

7- Now increase the modulation level. You will observe that for large values of modulation level the FM will be Wideband FM. Also observe that the FM bands don't increase monotonically but follow a peculiar behavior according to the Bessel Function.





Modulation source and FM as seen on the Spectrum Analyzer
(In this particular case Δf >>B so therefore, the FM is WBFM with many sidebands. m_p is Large for WBFM as you can see in the spectral plot of m(t))

Question

By looking at the spectrum of the FM modulated signal can you estimate the bandwidth of the modulating signal? Explain How?

Practical 09: The Quadrature detector

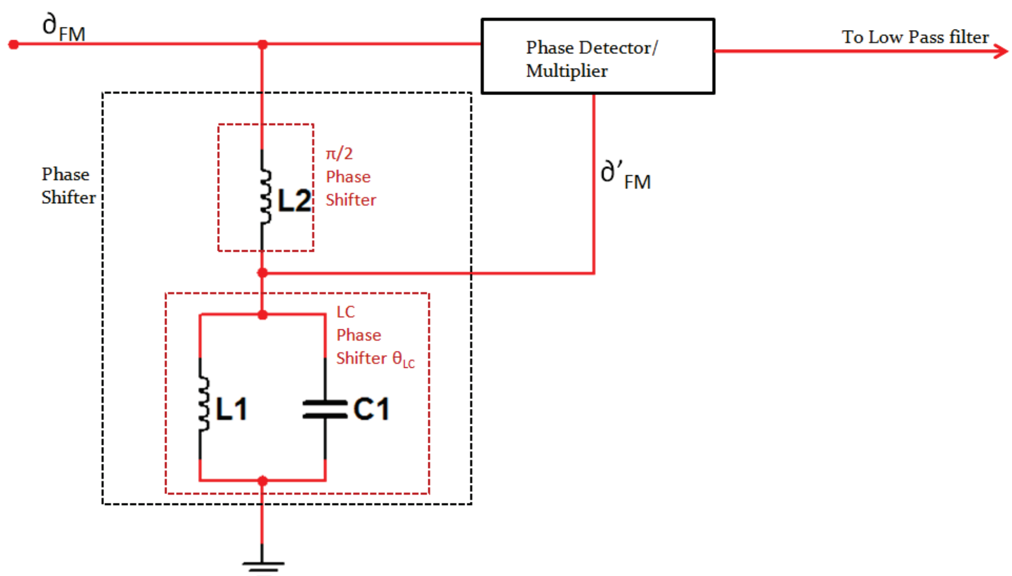
Objectives of the Experiment

To study the working of a quadrature detector and using it for demodulation of the FM signals

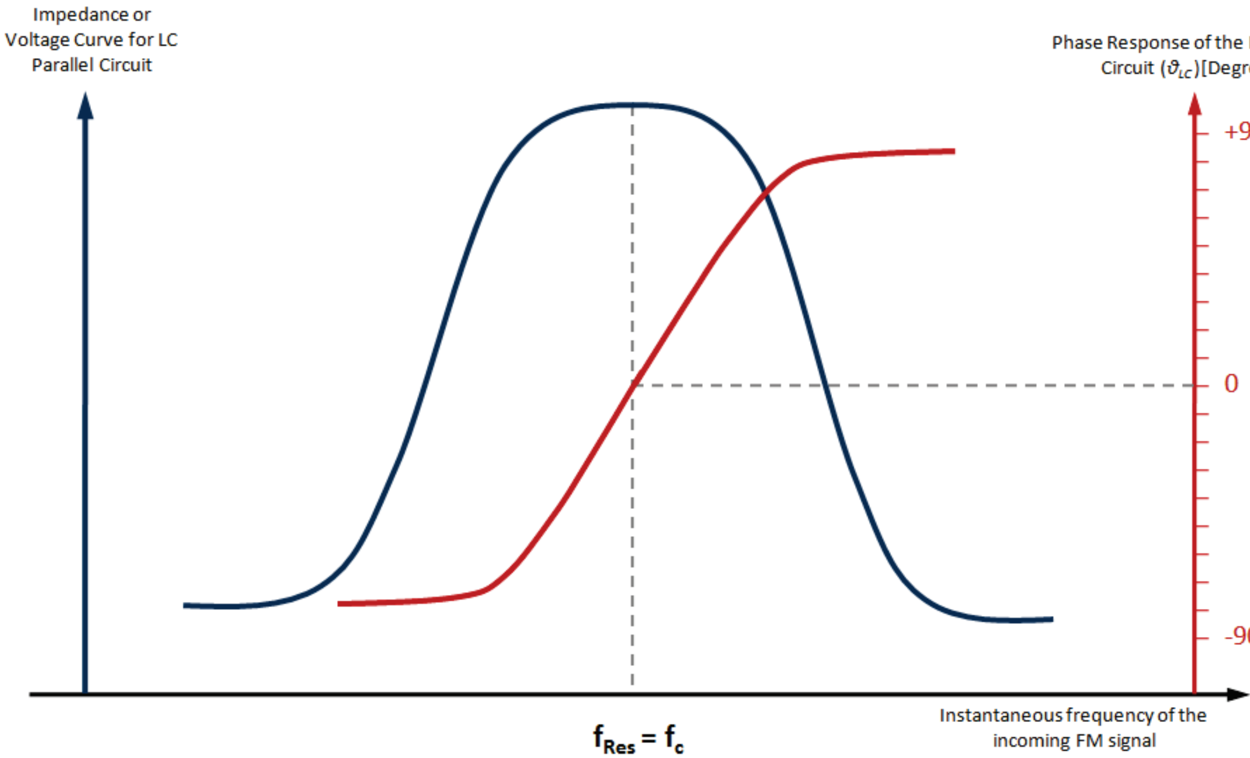
Background Information

In quadrature demodulator the FM signal and the signal passed through a phase shifter is given to a Phase detector. The output of the phase detector is signal that contains both modulating signal and a high frequency component which is removed by a low pass filtered

The phase shifter: The phase shifter contains two phase shifters inside it. One is a $\pi/2$ phase shifter and the other is an LC phase shifter that gives a phase shift of θ_{LC} to the FM signal. Thus the total phase shift given to the FM signal as it passes through the phase shifter is $\pi/2+\ \theta_{LC}$. The LC circuit is designed such that its resonance frequency is equal to the carrier frequency of the FM. The Phase shift θ_{LC} given by the LC parallel circuit is variable (ranging from +90 degrees to -90 degrees) depending on the frequency of the incoming FM signal. If the frequency of the FM signal is f_c (which means *m(t)* is zero) then the LC circuit will be in resonance and the phase shift θ_{LC} will be 0. But if the FM frequency is higher than f_c(this happens when *m(t)*is positive), then θ_{LC} will assume a positive value between 0 and 90 degrees(depending on the instantaneous frequency of FM signal). And if the FM frequency is lower than f_c, (this happens when *m(t)* is negative) then θ_{LC} will assume a negative value between 0 and 90 degrees(depending on the instantaneous frequency of FM signal). So in other words the θ_{LC} is directly proportional to *m(t)*. That's what helps us demodulate the incoming FM signals with a Quadrature Detector.



Actual circuit used in Quadrature Phase Detector
(Inductor L2 provides π/2 phase to every frequency in the incoming FM signal. L1 and C1 forming the parallel LC tank circuit provides a variable phase shift θ_{LC} depending on the instantaneous frequency of the incoming FM signals)



Phase response (shown in red) and Impedance Curve (shown in blue) of the Parallel LC tank circuit (Important to note here is that the resonance frequency f_{RES} is designed to be equal the carrier frequency f_c of the FM. As m(t)increases, FM's instantaneous frequency increases above f_c and therefore, θ_{LC} increases . As m(t) decreases, FM's instantaneous frequency decreases below f_c and consequently θ_{LC} decreases. This shows that θ_{LC} is directly proportional to m(t))

Let's consider the FM signal,

$$\partial_{FM} = \cos(\omega_c t + k_f \int m(\alpha)d\alpha)$$

After this FM signal has passed through the phase shifter it suffers a phase shift of $\pi/2 + \theta_{LC}$, so the output of phase shifter is

$$\partial'_{FM} = \cos(\omega_c t + k_f \int m(\alpha)d\alpha + \pi/2 + \theta_{LC})$$

Both ∂_{FM} and ∂'_{FM} are given to a [2]Phase detector which is essentially a multiplier. So,

Phase detector's output

$$= \partial_{FM} \times \partial'_{FM}$$

$$= [\cos(\omega_c t + k_f \int m(\alpha)d\alpha)][\cos(\omega_c t + k_f \int m(\alpha)d\alpha + \pi/2 + \theta_{LC})]$$

Using the identity $\cos(x) \times \cos(y) = 0.5[\cos(x-y) + \cos(x+y)]$ we get

$$=0.5[\cos(-\pi/2 - \theta_{LC}) + \cos(2\omega_c t + 2k_f \int m(\alpha)d\alpha + \pi/2 + \theta_{LC})]$$

[2] Phase detector, Phase Comparator, Multiplier & DSB-SC Modulator all refer to the same thing

After passing the output of product detector through the low pass filter the component at $2\omega_c$ is rejected and we are left with the

$$Filtered\ Output = 0.5\cos(-\pi/2 - \theta_{LC}) = 0.5\cos(\pi/2 + \theta_{LC}) = -0.5\sin(\theta_{LC})$$

When θ_{LC} is small $\sin(\theta_{LC}) \approx \theta_{LC}$

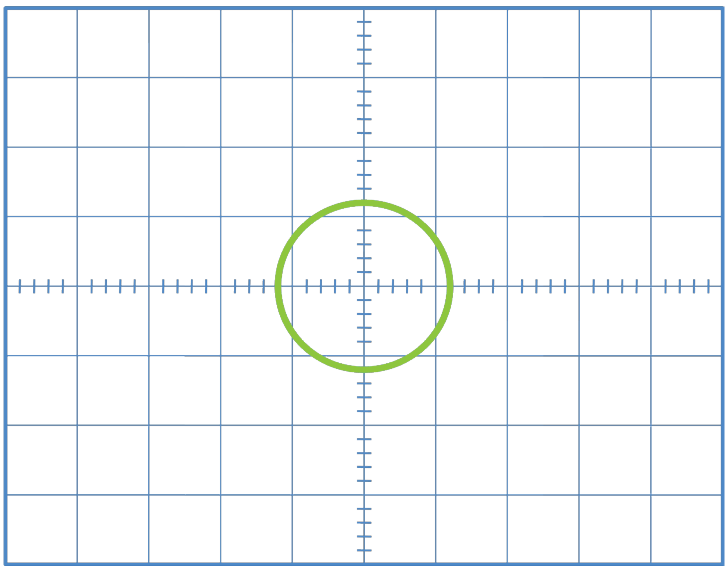
$$Output = -0.5\,\theta_{LC}$$

We know that $\theta_{LC}\ \alpha\ m(t)$ or $\theta_{LC} = k\,m(t)$

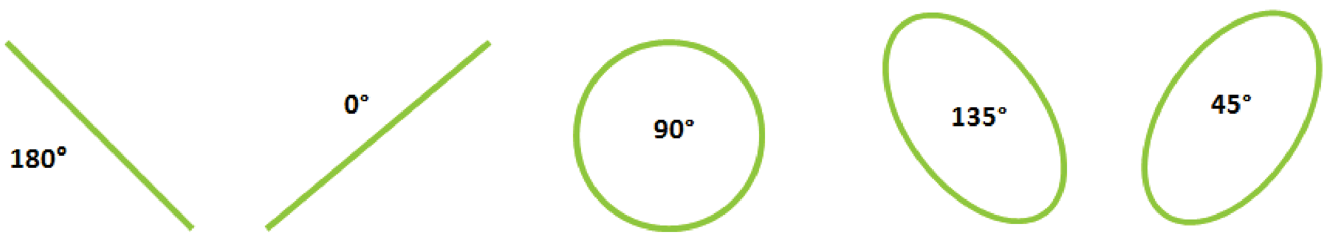
$$Output = -0.5\,k\,m(t)$$

So the output of the Quadrature phase detector will be the modulating or message signal

The quadrature detector is called so because when $m(t)=0$, then both ∂_{FM} and ∂'_{FM} will be 90° out of phase. You can easily observe the phase difference between the ∂_{FM} and ∂'_{FM} on the XY-Mode or Lissajous (*François*: Pronounced as *LEE-SA-ZHOO*) mode of the Oscilloscope. The Lissajous pattern of two same frequency, same amplitude but 90° out of phase signals will always be a perfect circle.



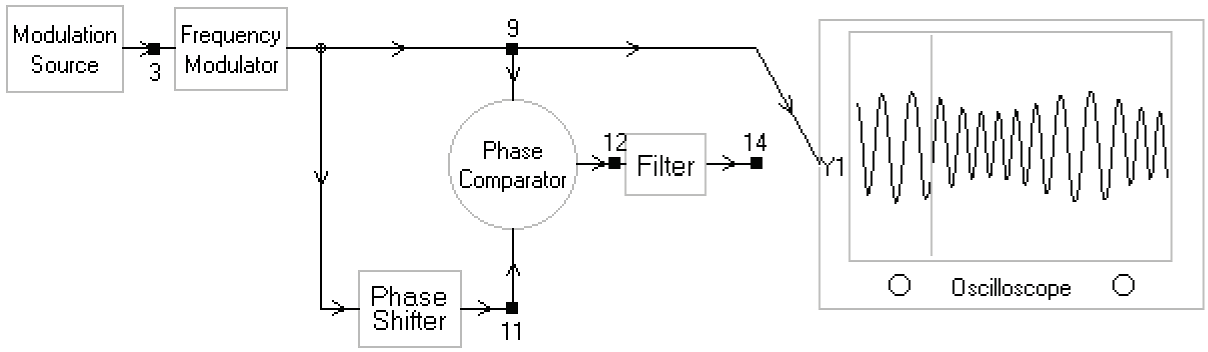
Lissajous Pattern of ∂_{FM} and ∂'_{FM} as seen on the oscilloscope screen in XY Mode when m(t) or Modulation level is set to zero



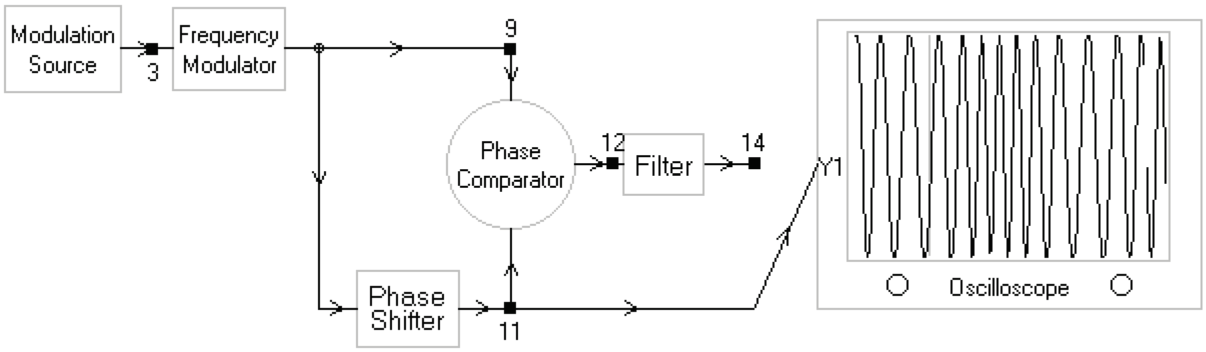
Lissajous Patterns for various phase shift between two signals with same frequency and amplitudes

Procedure

1- Start the Discovery software.
2- Go to System->Index->4-Demodulation of Frequency Modulated Signals
3- Click 'Yes' to Load Assignment 4
4- Start Practical 01
5- Observe the FM Modulated signal ∂_{FM} at test point 9 and ∂'_{FM} at test point 12.

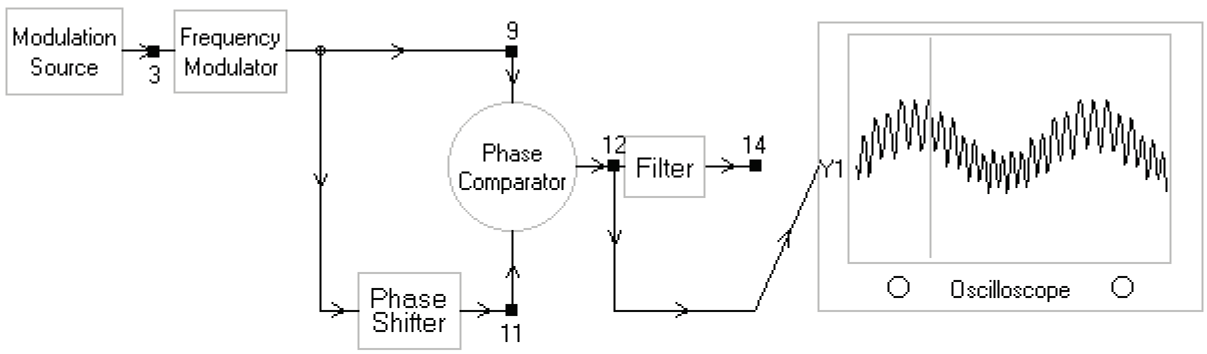


FM Modulated signal ∂_{FM} as seen on the Oscilloscope

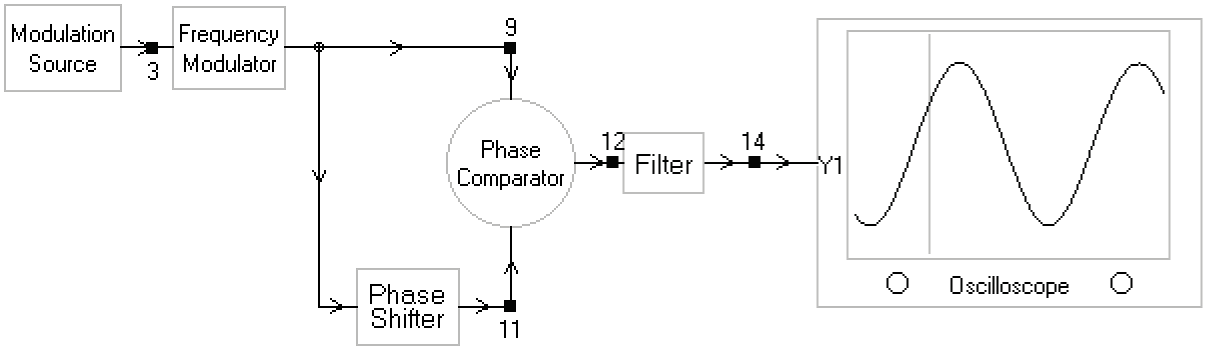


Phase shifted FM Modulated signal ∂'_{FM} as seen on the Oscilloscope

6- Observe the output of the Phase comparator. It contains the message signal as a well a $2\omega_c$ riding on it



Phase detector output showing high frequency component riding on the massage signal. This high frequency component needs to filtered out, 7- Now observe the output of filter. The filter rejects the high frequency components at and around $2\omega_c$ thereby, yielding the modulating signal m(t)



Demodulated Message signal as seen on the oscilloscope

8- Now set the modulating signal to zero by using the 'Modulation Level' knob. Observe the FM Modulated signal ∂_{FM} at test point 9 and ∂'_{FM} at test point 12 on the Bench top oscilloscope TRIO CS-1040. Switch to XY-Mode. You will observe a circle on the oscilloscope showing that both signals are 90° out of phase. In other words both ∂_{FM} and ∂'_{FM} differ in phase by quadrature (quadrature means one fourth of a cycle). That's precisely why this demodulator is called Quadrature Detector

Question

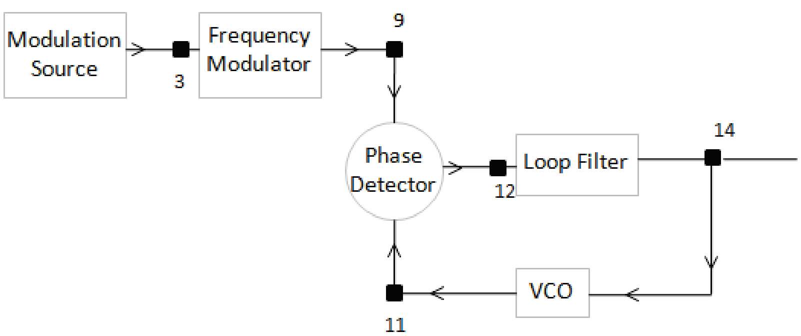
Locate the components used in the phase shifter i.e. L2, & C1 and L1 on the FM Modulation Kit 53-140.

Objectives of the Experiment

To study the working of a Phase Locked Loop and using it for demodulation of the FM signals

Note

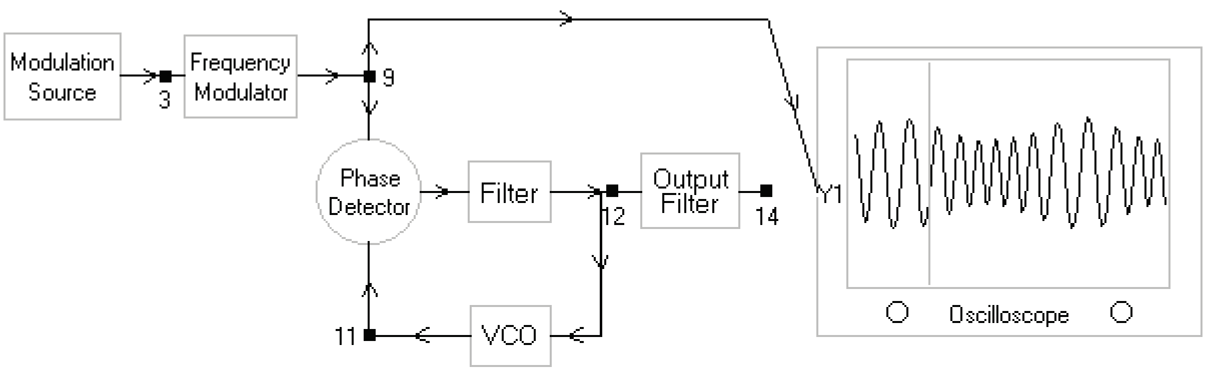
The diagram given below is the correct representation of a PLL detector. Unfortunately, this is not what you see in the actual software. The diagram in the software shows a filter between phase detector and point 12 which is totally wrong. Furthermore, the software also shows the VCO input coming from point 12. That's wrong as well. The input of VCO should be from point 14. So the diagram given below is the correct one. Whenever you see the PLL in software always bear this correct picture of PLL in mind.



Correct block diagram of a PLL detector

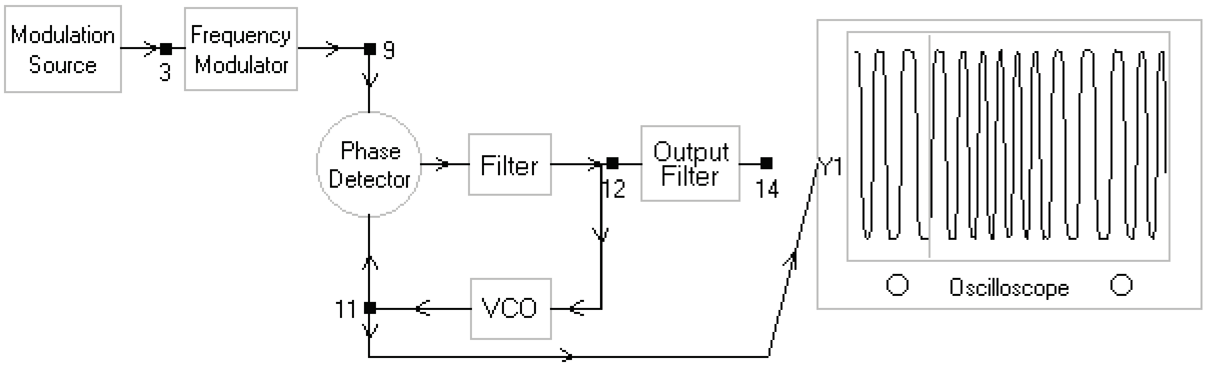
Procedure

1- Start the Discovery software.
2- Go to system->index->4-Demodulation of Frequency Modulated Signals
3- Select yes to Load Assignment 4
4- Start Practical 02
5- Observe the FM modulated signal at point 9



FM Modulated signal as seen on the Oscilloscope

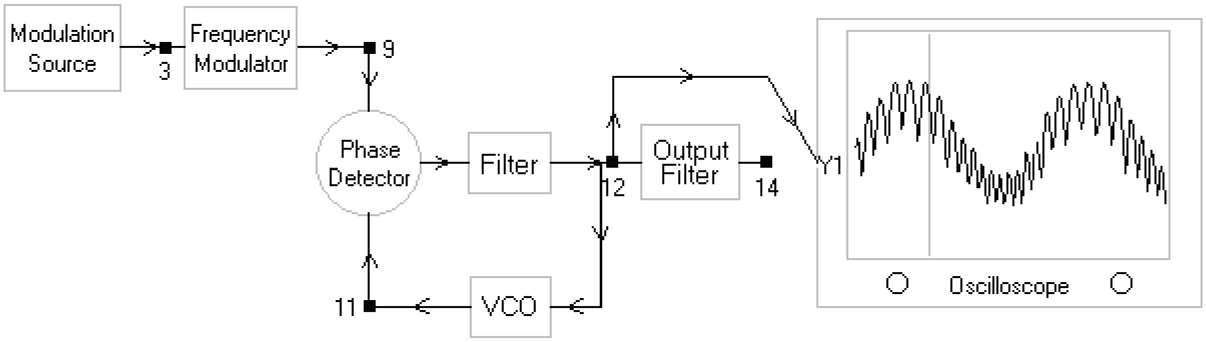
6- Now observe the output of VCO.



Output of the VCO as seen on Oscilloscope
(Note that the output of VCO follows the frequency variations frequency modulated signal)
(Note that this picture is not accurate. There is no Filter between point 12 and the phase detector output. Furthermore, the VCO input should be taken from point 14 instead of 12)

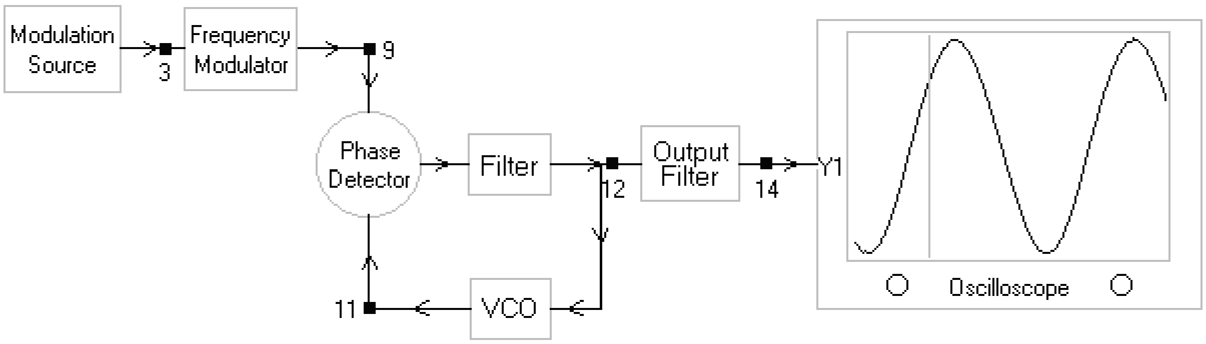
7- Also compare the output of both the VCO and FM modulator simultaneously on a bench top oscilloscope. You will observe that the VCO's output will lag behind the FM modulators' output. This is because the VCO is following the FM signal. The signal following will always be lagging compared to the signal being followed.

8- Observe the output of the Phase comparator. It contains the message signal as a well a $2\omega_c$ riding on it



Output of Phase Detector as seen on the Oscilloscope
(The phase detector output contains both the message signal and high frequency component
Note that this picture is not accurate. There is no Filter between point 12 and the phase detector output. Furthermore, the VCO input should be taken from point 14 instead of 12)

9- Now observe the output of the 'Output Filter'. The filter rejects the high frequency components at and around $2\omega_c$ thereby, yielding the modulating signal m(t)



Demodulated signal as seen on the Oscilloscope

(Note that this picture is not accurate. There is no Filter between point 12 and the phase detector output. Furthermore, the VCO input should be taken from point 14 instead of 12)

Equipments Required for Practical 11

1- AM Generator **AM2961A**

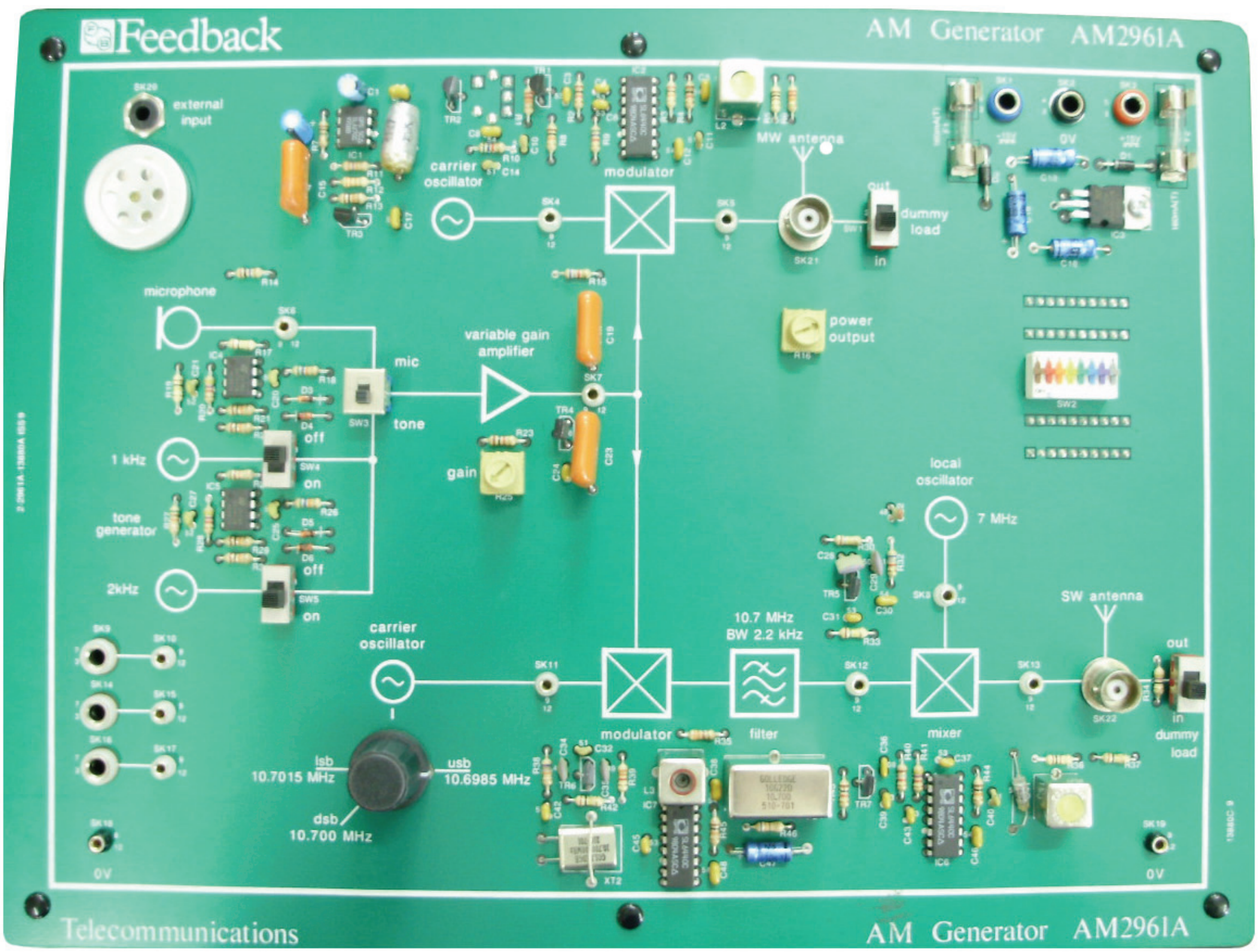
2- AM Receiver **AM2961B**

3- Dual Power Supply **PS 446**

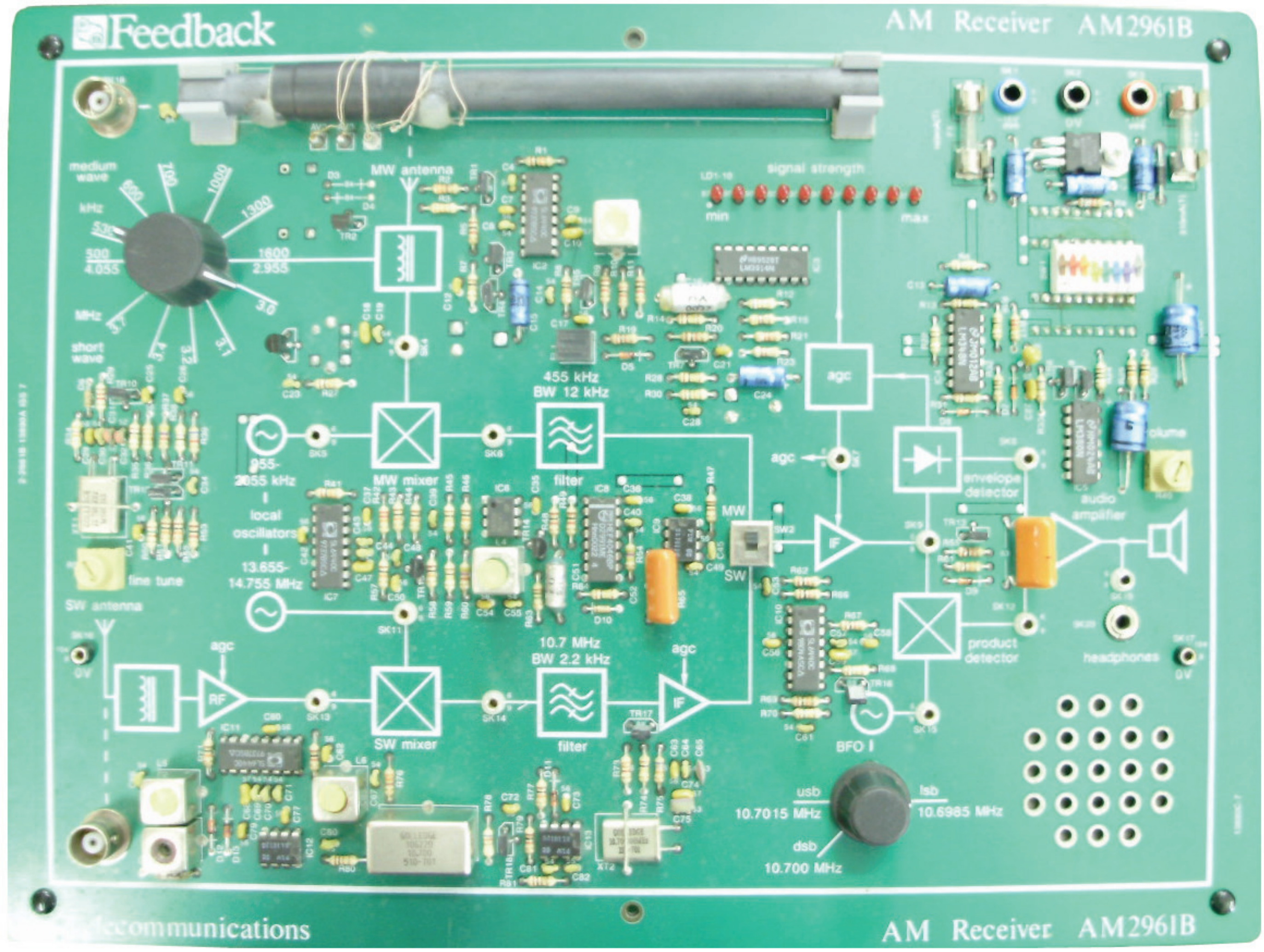
4- Tri-trace Bench top Oscilloscope **TRIO CS-1040** with pincer probes

5- Power Cables

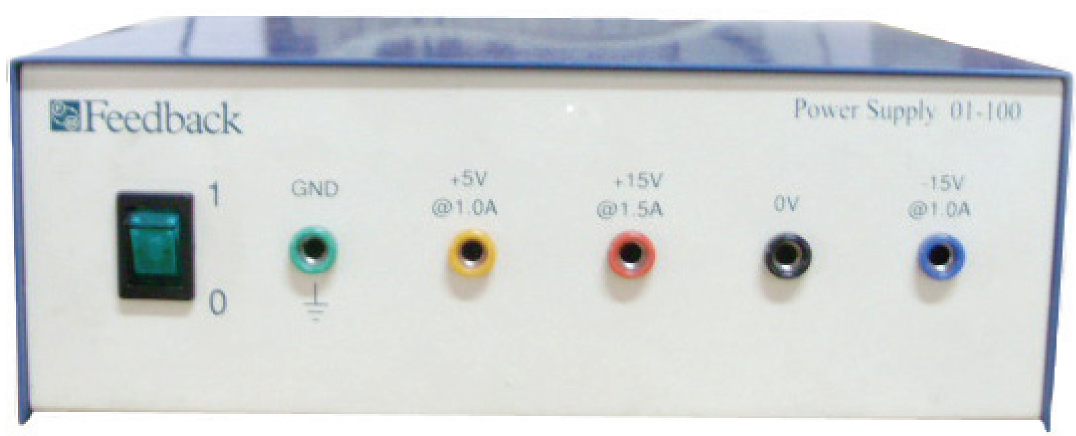
6- BNC-BNC RF Screened Cable



AM Generator AM2961A



AM Generator AM2961A



Dual Power Supply PS 446

Power Cable

BNC-BNC RF Screened Cable

Practical 11: AM Generator and Superheterodyne AM Receiver

Objectives of the Experiment

To study the AM transmitter and super heterodyne AM receiver

Background Information

AM Generator AM2961A: This AM Generator kit consists of an AM modulator and its' transmission related circuits. The modulating signal in this kit can be selected to either a Tone or a Mic. If you select tone as the modulating signal, then you have three further options. You can either transmit

- 1 KHz tone or
- 2 KHz tone or
- sum of 1 KHz and 2 KHz tones

Tones being periodic signals are very easy to see on oscilloscopes. The mic signal isn't periodic, so it can't be properly seen on analog oscilloscopes. That's precisely the reason why we use tone as the modulating signal so as to easily observe and study the AM transmission process. Other than that, there is no reason to transmit a pure tone via AM. In real life we would always be sending some music or voice from a mic via AM.

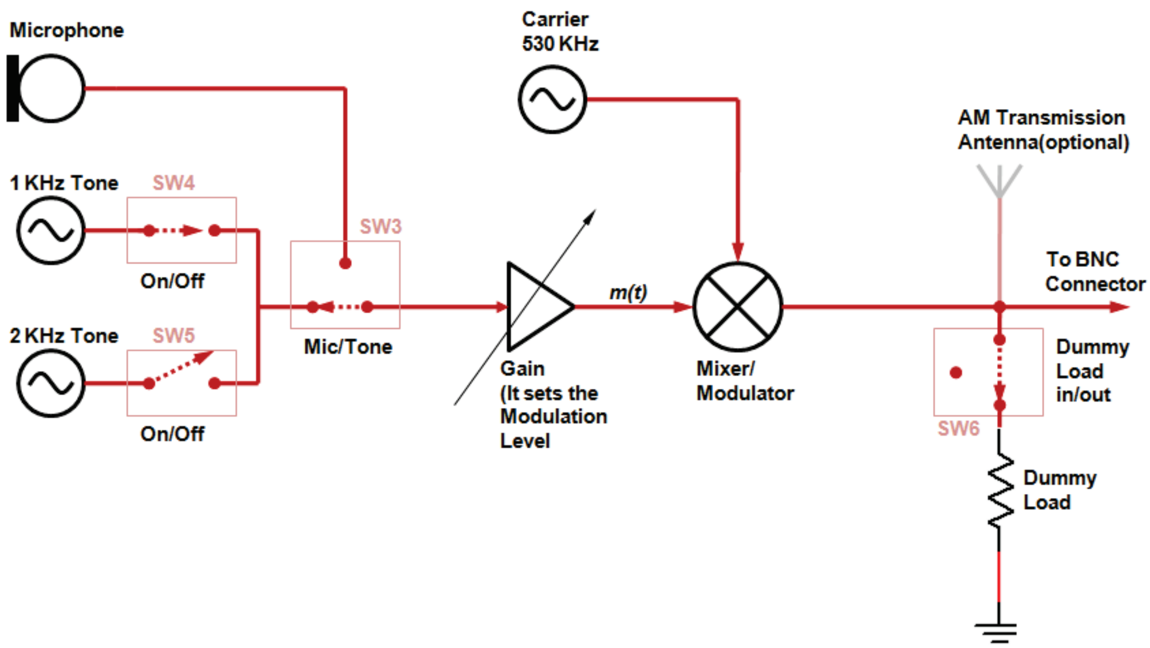
The modulating signal is then amplified. The amplification level is set using the 'Gain' Knob. This knob controls the modulation index of the AM. For our experiment we will be using 100% modulated signals (modulation index=1). So set the gain so that the output of the modulator has no zero crossings in the envelope. This particular AM transmitter kit modulates the modulating signal on a 530 KHz carrier, so this means that the AM receiver kit will receive the signal when we place its tuner dial near 530.

After the modulating signal has been modulated onto the carrier of 530 KHz, it is further amplified and sent to an antenna socket. However, for experimental purposes we do not use the antenna for AM transmission. There are two reasons for it

- The output power of AM transmitter is very low so very weak signal is received if we use antennas for transmission and reception of AM
- The medium between antennas i.e. the air, is very noisy. So although we would be sending a perfect AM signal at the transmitter side, we won't be able to receive a similarly perfect signal on the receiver side if we use antennas

For reasons discussed above, we don't use antennas for the experiment. Instead, we connect the transmitter and receiver directly by a BNC-BNC RF shielded cable. The advantage of using it is the noiseless & reliable transmission and reception of the modulated signal.

Important: The dummy load should be in 'In' setting when using BNC-BNC RF cable between transmitter and antenna. If the dummy load switch is in 'Out' position and the transmitter and receiver are connected by the cable, then a very large signal is sent to the receiver which might damage or saturate the electronics in the receiver. The dummy load absorbs some of the power thereby, sending a relatively small portion of the signal to the receiver so as not to damage or saturate the electronic components in the receiver. So it's very important that the dummy load should be in 'In' position when you are using direct cable connection between transmitter and receiver.



Simplified Block Diagram of AM Generator AM2961A

AM Receiver AM2961B: This kit is essentially the same as an ordinary AM Radio that one might buy in a store. The only difference is that the kit has all the components laid out so that you can see every stage of a Super Heterodyne Receiver and tap off the signal from its various stages. This AM receiver has a tuning range of 500-1600 which means that it can receive any AM channel that is being transmitted in 500 to 1600 KHz range.

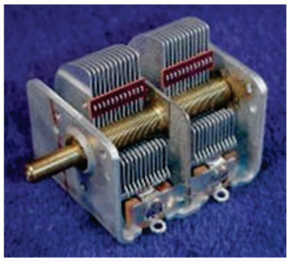
RF Filter and Local Oscillator: The tuning knob controls the center frequency of the RF Filter and the local oscillator. The RF Filter is a wide-band band pass filter centered at f_{Ch}, where f_{Ch} is the frequency of the channel being received or tuned to. This filter removes the image channel that might be present at $f_{Ch} + 2\ f_{IF.}$

The output of the RF Filter is amplified through the RF Amplifier and passed through an RF mixer where it is mixed with a sinusoid of frequency $f_{Ch} + f_{IF}$. The sinusoid of frequency $f_{Ch} + f_{IF}$ is produced by the local oscillator. The tuning knob that controls the center frequency of the RF Filter also controls the frequency of the local oscillator. So if the tuner knob is set to receive a channel of frequency f_{Ch}=530 KHz, then local oscillator will also produce a sinusoid of frequency $f_{Ch} + f_{IF}$ = 530 + 455 KHz. This controlling of two different circuits with one knob is known as Gang operation. A special variable capacitor is used for this purpose and is commonly known as Gang Capacitor.

Gang capacitor: Gang capacitor contains four set of plates. Two sets of these plates constitute a capacitor. So a gang capacitor essentially has two variable capacitors in it. All the plates are mounted on a single shaft. So when the shaft of the Gang Capacitor is turned, the capacitance of the both the capacitors changes. One capacitor of the Gang capacitor is used to vary the center frequency of RF Filter. The other variable capacitor of the Gang is used to vary the capacitance in the Colpitts oscillator that produces a sinusoid of frequency $f_{Ch} + f_{IF.}$ So both RF Filter's center frequency and Local Oscillator's oscillating frequency are controlled by a single knob. If the tuning knob of the gang capacitor is at 500 then, the center frequency of RF filter would be 500 KHz and the local oscillator will oscillate at 500 + 455 KHz. If the Tuning knob of the gang capacitor is at 530 then the center frequency of RF filter would be 530 KHz and the local oscillator will oscillate at 500 + 455 KHz. Thus the local oscillator will always be oscillating at a frequency 455 KHz above the channel frequency being tuned to.

Superheterodyning
Superheterodyning is the process of mixing the signal with a sinusoid of frequency that differs from the original carrier by an amount f_{IF}. Supperheterodyning centers the spectrum of modulated signal to Intermediate frequency $f_{IF.}$

Homodyning
Homodyning is the process of mixing the modulated signal with a sinusoid of frequency equal to the carrier frequency of the modulated signal. The process of homodyning shifts the spectrum back to baseband. It also produces modulated spectrum at twice the carrier frequency as well. Product detector is essentially a homodyner.



Gang Capacitor
(Image Courtesy of www.midnightscience.com)

Electrical Symbol of a Gang Capacitor

Gang Capacitor and its Electrical Symbol
(Gang capacitor contains four set of plates. Two sets form one variable capacitor. So the Gang capacitor has essentially two variable capacitors inside it. The capacitance increases when the two sets of plate interleave. The more interleaving there is between the two set of plates of a capacitor of a gang, the greater will be its capacitance)

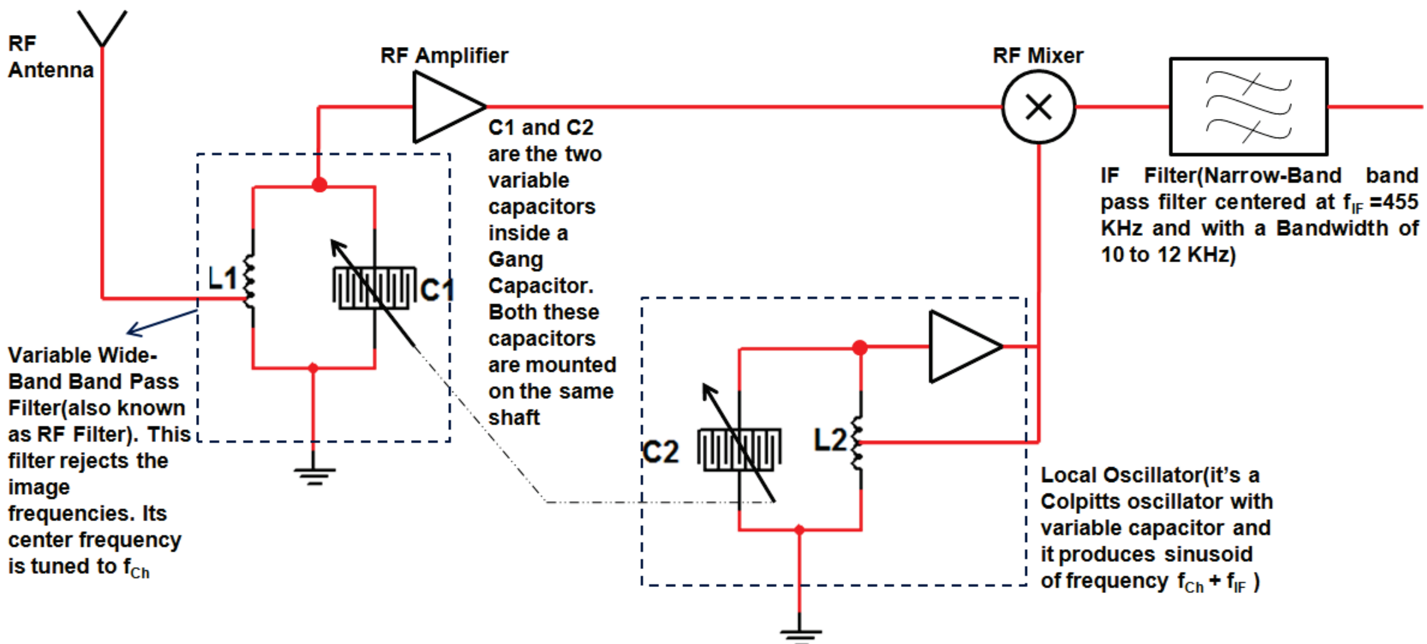
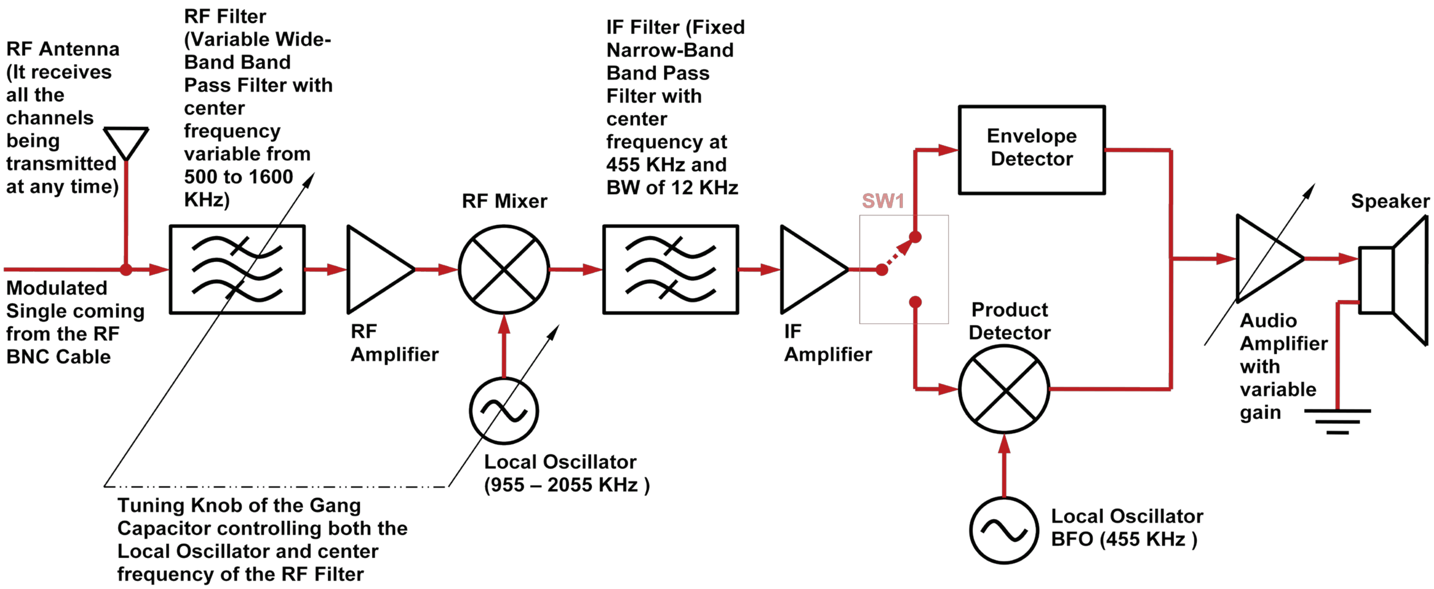


Figure 11.1 Circuit showing the RF filter and Local Oscillator in a Superheterodyne Receiver and how the Gang Capacitor is used to vary both the center frequency of the RF filter and the oscillation frequency of the Local Oscillator

IF Filter: After passing through the mixer, the AM spectrum gets shifted with the channel to be received now centered at 455 KHz. we now need to separate it from the rest of the channels. For this purpose we use a fixed Narrow-band band pass filter centered at 455 KHz and having a bandwidth of 10 or 12 KHz. (In the trainer kit the filter has a bandwidth of 12 KHz).

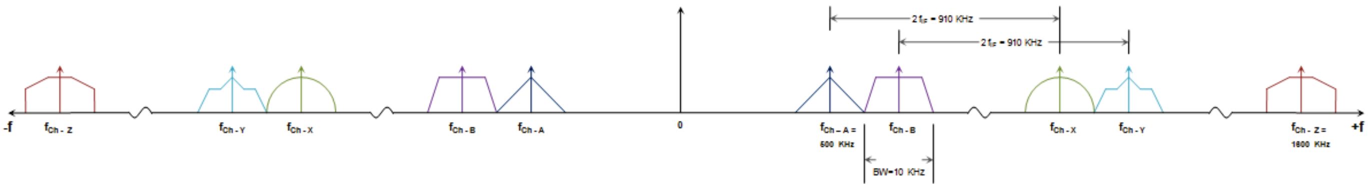
AM Demodulation: The output of the IF filter is an AM signal modulated at 455 KHz. All that's now required to get this information is to demodulate this AM signal. For this purpose the kit has provided two methods, one is envelope detector and the other is product detector (Standard AM radios contain only envelope detector because it's comparatively cheap compared to the product detector). Do remember that in the AM kit, at a time only one filter is engaged. So if product detector is engaged then envelope detector won't be working and if envelope detector is working then product detector won't be connected. The output of demodulator is amplified and then sent to an onboard speaker which makes the received channel audible.



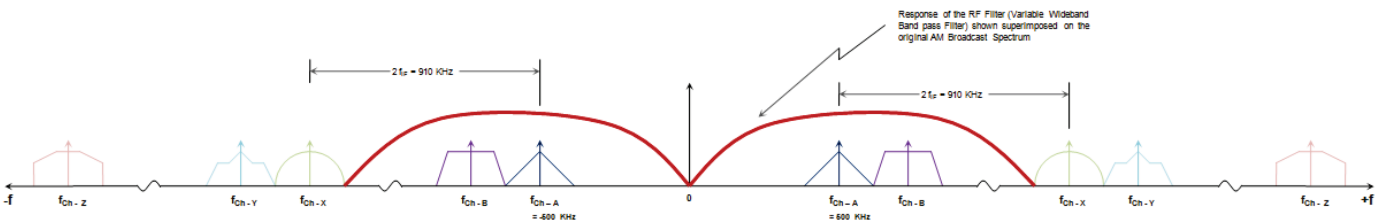
Simplified Block Diagram of AM Receiver AM2961B

Superheterodyne Receiver - A Spectral View of output from its various stages

The figures below show the spectrum of the signals coming from the various stages of a Superheterodyne AM Receiver. The user has tuned the receiver knob at 500 KHz in order to receive the Ch-A being transmitted on a carrier of 500 KHz. Carefully study the outputs from various stage of the superheterodyne receiver in frequency domain.

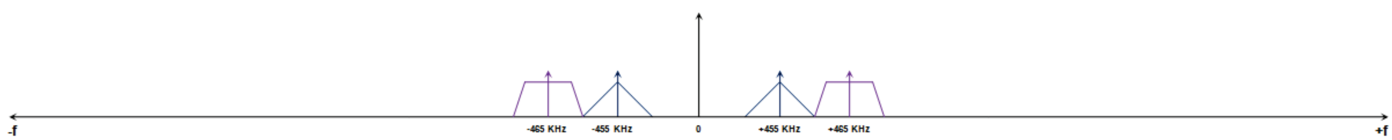


AM BROAD CAST SPECTRUM (First channel in the spectrum is f_{Ch-A} at 500 KHz and the last channel is f_{Ch-Z} at 1600 KHz. The ∿ sign shows that there are more channels present, but they haven't been shown for the sake of simplicity. Bandwidth of each Channel is 10 KHz. Ch-X is the image channel for Ch-A and Ch-Y is the image channel for Ch-B and so on. Image channel is always at a frequency 910 KHz higher than the channel being received. In he above plot both negative and positive frequencies have been shown.

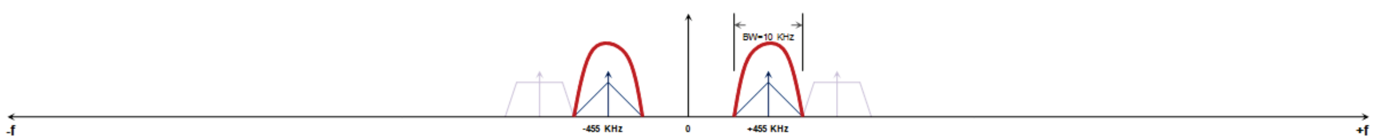


Response of the RF Filter (Variable Wideband Band pass Filter) shown in red, superimposed on the original AM Broadcast Spectrum. This filter whose center frequency is set to the frequency of the channel being received, rejects the image channel and those beyond it. In this particular case, the RF filter is centered at f_{Ch-A} = 500 KHz and it rejects the image channel f_{Ch-Z} = 500+910 KHz and the channels beyond this frequency.

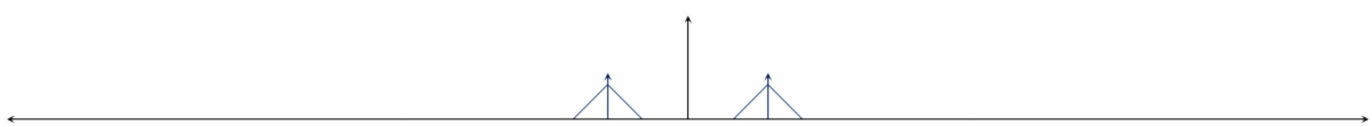
Spectrum of the RF Filter output. The image channel f_{Ch-Z} = 500+910 KHz and the channels beyond this frequency no longer exist



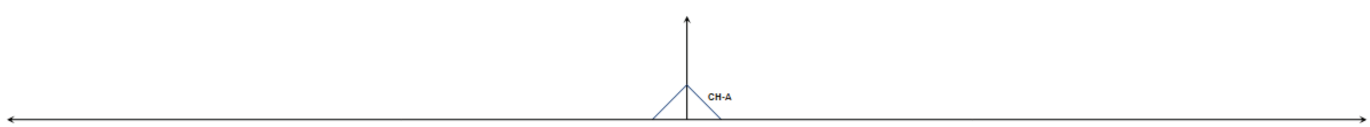
Spectrum after passing through the RF Mixer. The channel to be received (i.e. Ch-A) gets shifted from 500 KHz to the intermediate frequency of 455 KHz



Response of the IF Filter (Fixed Narrowband Band pass Filter) shown in red, superimposed on the mixer's spectrum. This filter whose center frequency is always 455 KHz and has a bandwidth of 10 KHz, rejects all the channels except the one that the user wishes to listen to. In this case, the desired channel is Ch-A, so the IF filter removes all channels except Ch-A.



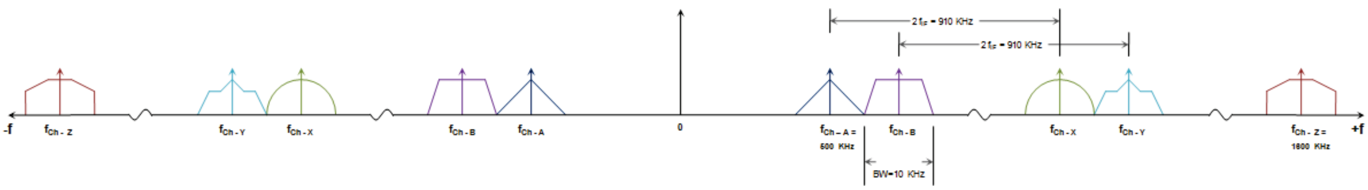
Spectrum of the output of the IF Filter. No channel remains except the channel the user wishes to listen to. In this case the desired channel is Ch-A. So this channel remains, while all others have vanished.



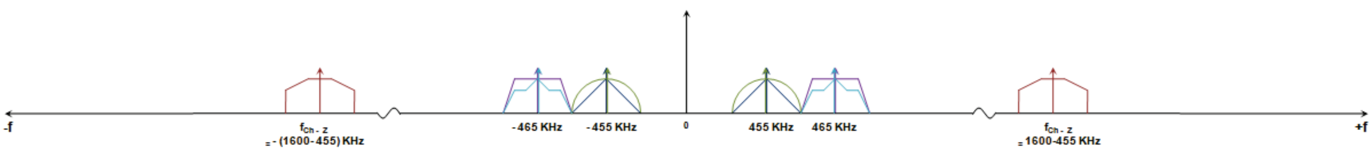
Spectrum of the output of AM Demodulator. The AM demodulator, whether its envelope detector or product detector, brings the CH-A spectrum back to its baseband state.

Superheterodyne Receiver without an RF Filter - A Spectral View of output from its various stages

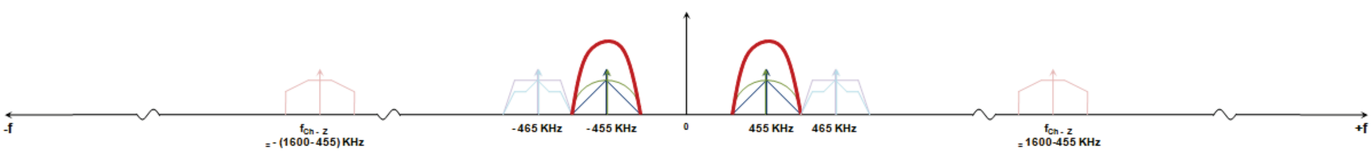
The figures below show the spectrum of the signals coming from the various stages of a Superheterodyne AM Receiver when we have removed the RF filter. The job of RF filter was to reject the image channel. Now that RF filter is gone, we not only receive Ch-A but Ch-X as well, where Ch-X is the image channel for Ch-A. Ch-X is located in the AM Broadcast spectrum at a frequency 2 f_{IF} higher than the channel user tunes to(in this case the user is tuning to Ch-A which is being broadcasted at 500 KHz). Study the spectral view of output from various stage of the receiver carefully in order to see what happens to the Channel being received when there is no RF filter.



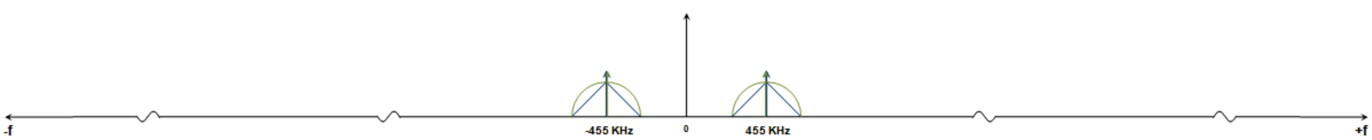
AM BROAD CAST SPECTRUM (First channel in the spectrum is f_{Ch-A} at 500 KHz and the last channel is f_{Ch-Z} at 1600 KHz. The ∿ sign shows that there are more channels present, but they haven't been shown for the sake of simplicity. Bandwidth of each Channel is 10 KHz. Ch-X is the image channel for Ch-A and Ch-Y is the image channel for Ch-B and so on. Image channel is always at a frequency 910 KHz higher than the channel being received. In he above plot both negative and positive frequencies have been shown.



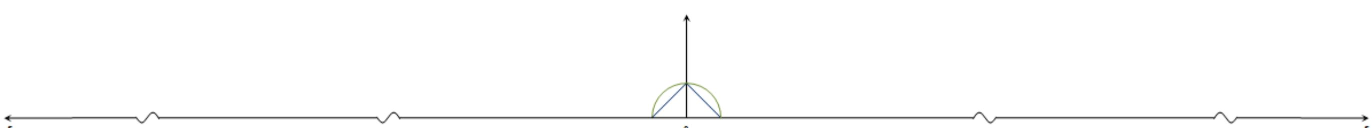
Spectrum of the output the RF mixer when there is no RF filter. You can clearly see that without this filter, the image channel's spectrum overlaps the spectrum of Ch-A which the user wishes to receive. Ch-A and its image channel Ch-X are occupying the same spot on spectrum. Similarly Ch-B and its image channel Ch-Y occupy the same spot on the spectrum



Response of the IF Filter(Fixed Narrowband Band pass Filter) shown in red, superimposed on the mixer's spectrum. This filter whose center frequency is always 455 KHz and has a bandwidth of 10 KHz, rejects all the channels except the one that the user wishes to listen to. Without any RF filter, the IF filter fails to remove the image channel Ch-X



Spectrum of the output of the IF Filter. The output contain both the channel to be received Ch-A and its' unwanted image channel Ch-X. Both channels occupy the same spot on the spectrum



Spectrum of the output of AM Demodulator. The AM demodulator, whether its envelope detector or product detector, brings both Ch-A and Ch-X spectrum back to its baseband state. Thus, without any RF filter, the user will be listening to two channel Ch-A and Ch-X at the same time. This is an annoyance because the user wont be able to understand either of the channels. That's why RF filter is so important. It's job is to remove the image channel whose frequency is 910 KHz higher than the frequency of the channel to be received

Procedure

Power up the transmitter and receiver and connect the RF out of the transmitter to the RF-in of the receiver using a BNC-BNC RF shielded cable. Dummy load should be In.

On the transmitter side:

1- Set the modulation source to 'Tone' and turn on the 1 KHz source and turn off the 2 KHz source. Using this setting, the modulating signal used will be a pure tone (sinusoid) of 1 KHz.
2- Observe the modulated output on Oscilloscope and Adjust the 'Gain' so that you get a 100% modulated AM signal.
3- You have now successfully configured the transmitter. Observe the modulating signal, carrier and modulated signal on the oscilloscope. To do this you will have to use all three channels of the oscilloscope in Tri- trace mode. The trigger should be from the channel whose input is the modulating signal. That way you will get a stable display on the oscilloscope.(Rule of thumb: if you are applying many signals to the oscilloscope at the same time, then the trigger source should always be lowest frequency and the most periodic signal of all. In the above case, modulating signal has the lowest frequency among all three of the applied signals so that's why we will trigger from the channel to which modulating signal is connected)

On the receiver side:

1- Observe the output of Local Oscillator on the scope. Its frequency will be f_{Ch} + 455 KHz. When you move the tuner knob from 500 to 1600, the local frequency output will increase from (500 + 455 = 1005) KHz to (1600 + 455 = 2055) KHz.
2- Now tune the receiver to 530. That's because our transmitter is sending AM signal at 530 KHz. Observe the output of Local Oscillator on the scope. Its frequency will be (530 + 455) KHz.
3- Now observe the output of the mixer on the scope. You won't be able to make any sense of the waveform seen on the scope. That's because the RF mixer output has many channels in it. So the output that you see on oscilloscope will be a composite signal containing all the channels
4- Now observe the IF filter's output on scope. Now you will see the 1 KHz tone (this is the modulating signal at the transmission side) riding on a 455 KHz carrier.
5- Observe the output of the demodulator on the scope and compare it with the original tone on the transmission side. You will see that both will be pretty much the same. The output will have a different phase compared to original. The phase difference is introduced by all the filters that were used in the reception of the signal. The phase shift introduced in the information signal is not an issue. Two out of phase tones/signals will always sound the same on a speaker. So as far as human ear is concerned, the phase difference between the original modulating signal and the recovered modulating doesn't matter.

Equipments Required For Practical 12

1- FM Generator **FM2962A**

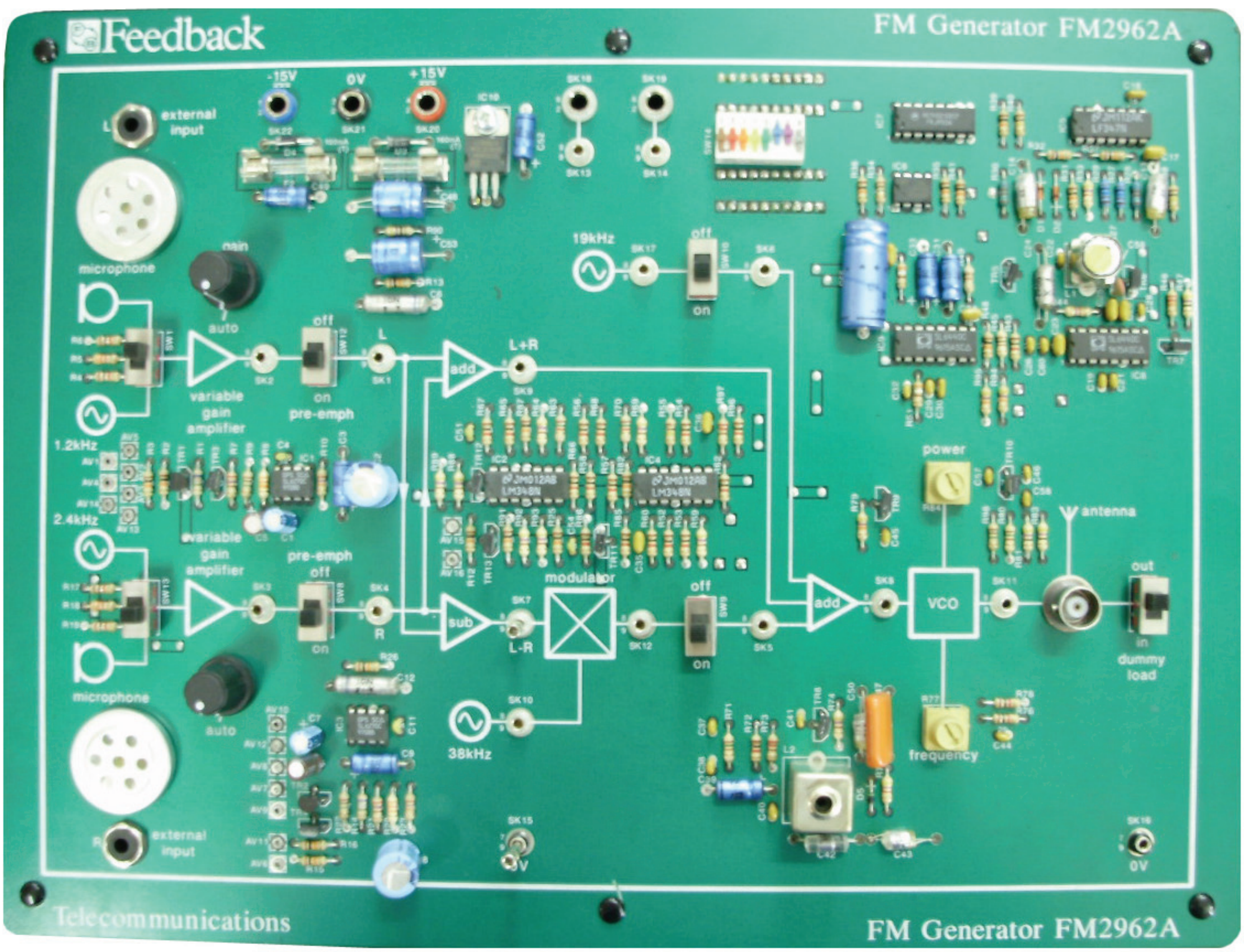
2- FM Receiver **FM2961B**

4- Dual Rail power supply **PS 446**

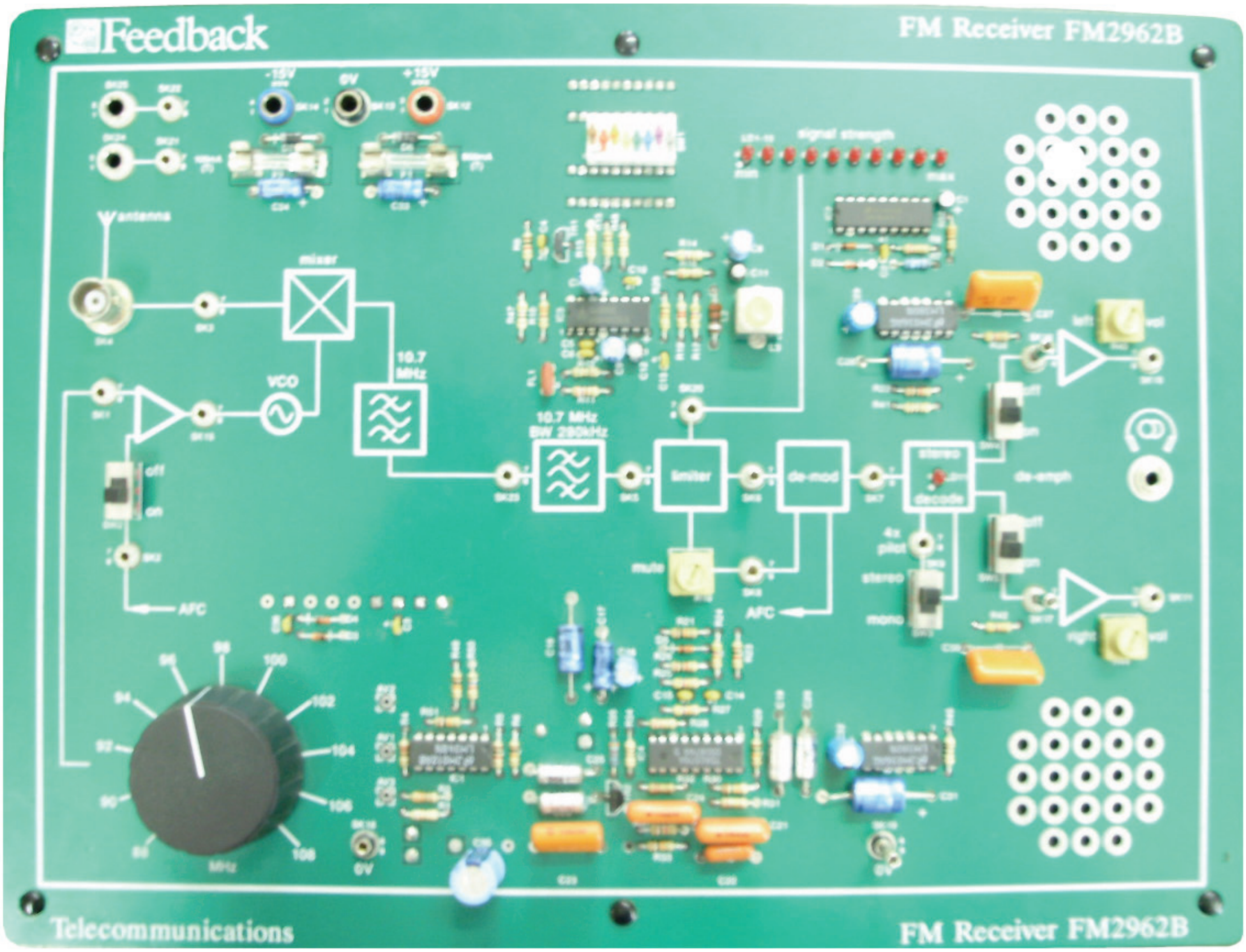
5- Kikusui **COS6100** 100 MHz Oscilloscope with pincer probes

6- Power Cables

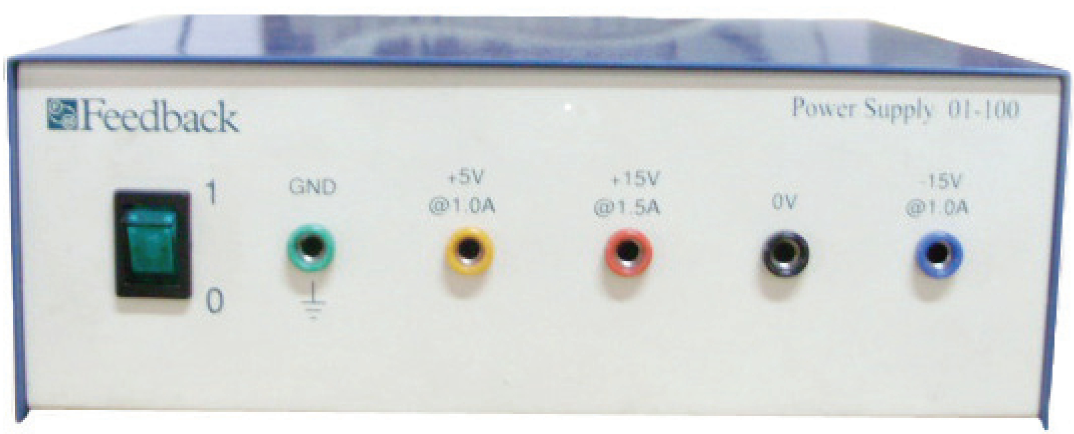
7- BNC-BNC RF Screened Cable



FM Generator FM2962A



FM Receiver FM2961B



Dual Power Supply PS 446

Kikusui COS6100 100 MHz Oscilloscope

Pincer Probe with its pincer and the probe adjustment screw driver

Power Cable

BNC-BNC RF Screened Cable

Practical 11: FM transmitter and Superheterodyne FM Receiver

Objectives of the Experiment

To study the FM transmitter and super heterodyne FM receiver

Background Information

FM transmitter FM2962A: The FM transmitter kit is a Stereo FM transmitter but it can work as a Mono FM transmitter as well. The modulating signal for Left Channel is a triangular wave of frequency 1.2 KHz and the modulating signal from Right Channel is a triangular wave of 2.4 KHz. Furthermore, you can also select the modulating signal to be from the Left Mic and Right Mic as well. Use switches SW1 and SW13 to switch between microphone and the triangular wave. But for sake of studying FM modulation, we prefer to use the periodic 1.2 KHz and 2.4 KHz triangular signals instead of the microphones. This way all the signals being studied would be periodic and hence easily observable on an oscilloscope.

The L and R signal can be amplified using the amplifiers. The amplifiers can be switched off by turning the 'Gain' knobs in fully counter-clockwise direction. After amplification, the signals are passed through the Pre-Emphasis filters. The Pre-Emphasis filters can be switched off, if needed, using SW12 and SW8 switches. The Pre-emphasized L' and R' signals are then matrixed to obtain (L' + R') and (L' - R') signals. The (L' - R') signal is DSB-SC modulated using a tone of 38 KHz which is obtained by passing a 19 KHz tone through a frequency doubler. The (L' + R') and DSB-SC (L' - R') and 19 KHz tone are added up in a summing amplifier to get a composite signal

Composite signal= $(L' + R') + (L' - R') \cos[(2\pi\ (38{,}000)t] + \cos[(2\pi\ (19{,}000)t]$

This composite is given as a modulating signal to a Voltage Controlled Oscillator (which is essentially a frequency modulator). The output VCO is an FM modulated signal. The carrier frequency of the FM modulated signal can be varied from 93 MHz to 95 MHz using the 'Frequency' Knob. The FM Modulated output from the VCO cannot be viewed on the oscilloscope available in the Lab. This is because to visualize a 93-95 MHz signal on oscilloscope, you will need an oscilloscope that has a bandwidth of at least 1 GHz. Unfortunately, we don't have an oscilloscope with such high specs in our lab. So we won't be able to observe this FM

Subsidiary Communications Authorization

In US, Canada, Australia and some other countries, the composite signal also contains an SCA (Subsidiary Communications Authorization) channel in the 58 - 75 KHz band

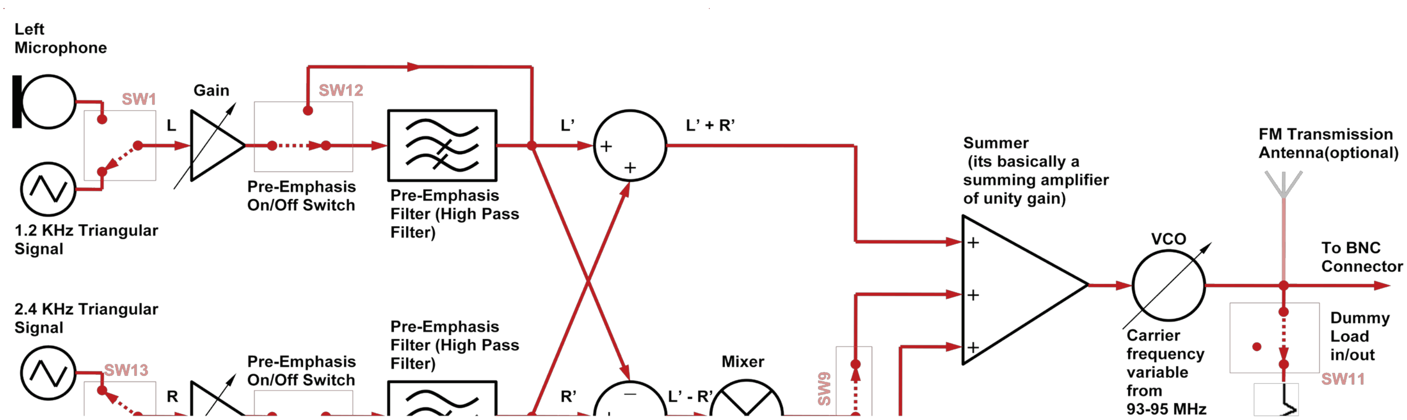
Composite signal = $(L' + R') + (L' - R') \cos[(2\pi\ (38{,}000)t] + \cos[(2\pi\ (19{,}000)t] + SCA$

All these countries have big superstore chains. US, for example, has superstore chains like Walmart and 7-Eleven. Studies have shown that shoppers relax and shop more while listening to some soothing music. So now every store has some kind of music playing in the background. The first thing you hear, when you step into a superstore is its background music. Each store must have one employee dedicated to changing music tracks all day long. There are 150,000 Walmart superstores in the US alone. So that would mean employing 150,000 employees just for changing the sound tracks. This is huge burden on the superstore chain. So they have come up with an ingenious plan. Why not broadcast this background music on the SCA channel. That way there need to be only one employee at the transmission station responsible for changing the music tracks. Each superstore that wants to play the background music will need an SCA receiver to receive the music being broadcasted in SCA band. So with the help of SCA channel, only one employee is required to change music tracks for all the superstores all day long.

modulated signal.

The Dummy load can be switched 'In' or 'Out' using the SW11 switch. The dummy load should be 'In' when using a direct cable connection between the transmitter and receiver and should be switched out when using antenna for transmission and reception.

The pilot and DSB-SC (L' - R') can turned off using SW10 and SW9 respectively. This is important when studying mono transmission.



Detailed block diagram of FM transmitter Trainer Kit FM2962A

FM Receiver FM2961B: Superheterodyne FM Receiver is exactly the same as superheterodyne AM receiver. The only difference is that for FM, the Intermediate frequency is 10.7 MHz, whereas for AM it's 455 KHz. The other difference is that the bandwidth of the IF Filter for FM is 200-220 KHz and the pass band of the filter is centered at 10.7 MHz, whereas for AM, the IF filter is centered on 455 KHz with a bandwidth of 10-12 KHz.

PLL is used as demodulator in the FM superheterodyne receiver. PLL based demodulators are standard on FM receivers because of their superior performance and because of their small size in IC form.

After demodulation through PLL we get the same composite signal back which was modulated using VCO on the transmitter side. So

PLL output= $(L'+R') + (L'-R')\cos[(2\pi(38{,}000)t] + \cos[(2\pi(19{,}000)t]$

All that is required now is to get back L and R channel.

To do this the PLL output is passed through

- A Low pass filter with cut-off near 15 KHz: This filter removes isolates the (L' + R') signals from the rest
- A band pass filter centered at 38 KHz and with corner frequencies at 23 KHz and 53 KHz: This filter isolates the DSB-SC modulated (L' - R') signal from the rest
- A Narrow-band Band pass filter centered at 19 KHz: this filter isolates the 19 KHz carrier so that it can be used to synchronously demodulate the DSB-SC modulated L' - R' signal

The 19 KHz carrier obtained is passed through a frequency doubler and then homodyned with the DSB-SC modulated (L' - R') signal to get just (L' - R'). Now that we have successfully obtained (L' + R') and (L' - R') signals, we matrix them together to get 2 R' and 2 L'. All that's left now is to deemphasize them. After De-emphasis, we get 2 L and 2 R signals that are further amplified and then fed to left and right speakers respectively. The De-Emphasis filters can be turned off as well using SW4 and SW5 to observe the effect of lack of De-Emphasis. Switch SW3 is used to simulate mono signal reception on a stereo receiver.

Detailed block diagram of FM Receiver Trainer Kit FM2962B

What is Pre-emphasis and De-emphasis & why use them

Let's consider that the input to an FM modulator in Figure 12.1 is set to zero i.e. m(t). The output of the FM modulator is sent via transmission antenna and received through receiver antenna on the receiver side.

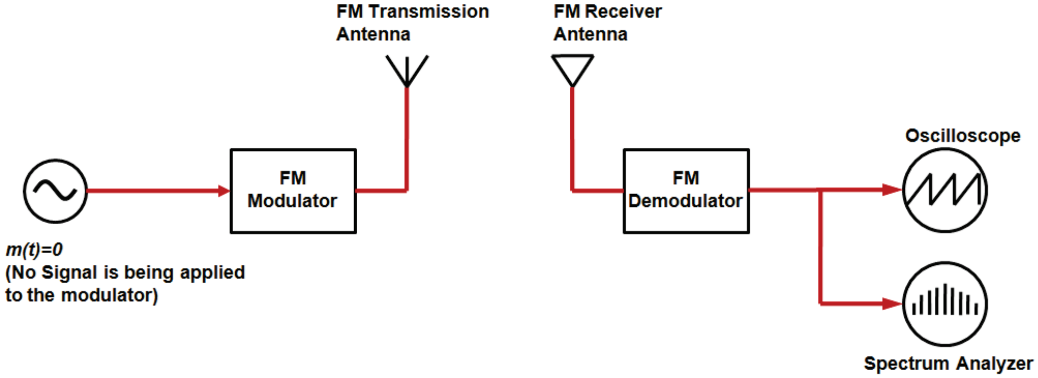


Figure 12.1. FM transmitter and Receiver without any pre-emphasis and de-emphasis

Even when the m(t) is zero at the transmitter side, we receiver a noise signal on the receiver side. The output of the FM demodulator as seen on the oscilloscope and the Spectrum analyzer is given below.

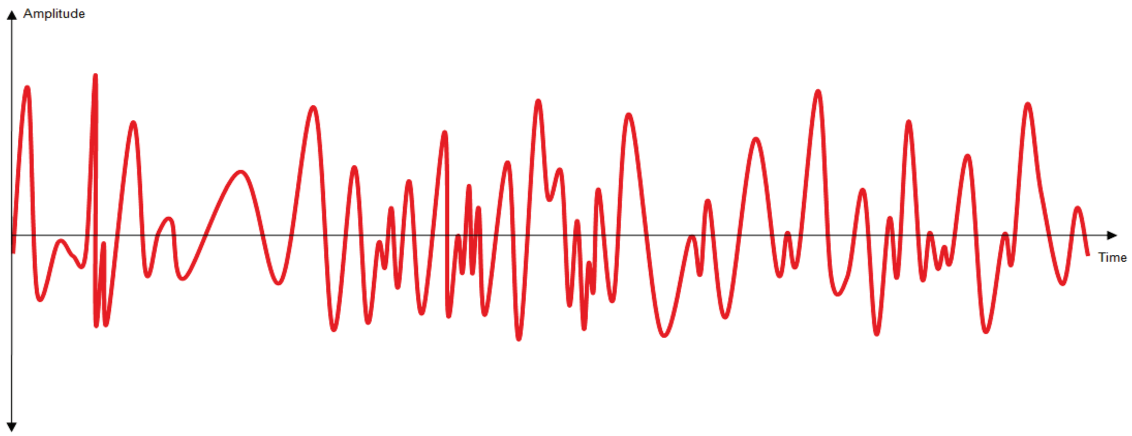


Figure 12.2 FM Demodulator's output as seen on the scope even when no modulating signal was present on the transmitter side

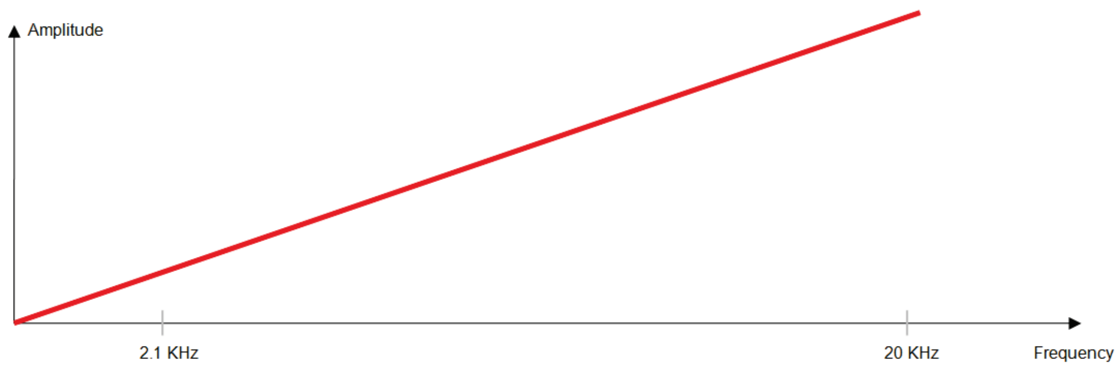


Figure 12.3 Demodulator's output as seen on a spectrum analyzer even when no modulating signal was present on the transmitter side

So even when we are sending no signal on the transmitter side, we will still be receiving noise on the receiver side. From Figure 12.3 it's clear that the noise is negligible in the 0 - 2.1 KHz range (let's called this the A range), but is pretty significant in the 2.1 - 20 KHz range (let's call this the B range). This high noise in the B range frequencies can severely corrupt the Audio signal frequencies that may lie in the B range. Let's denote the noise in B range of frequencies by $N_2(f)$. This noise is a function of frequency. Higher frequency in this noise has higher amplitude compared to the lower frequencies.

Let's say that the input to an FM modulator is an audio signal with A and B range spectral contents in it. So

Input to the FM Modulator = $[A] + [B]$ then,

Output of the FM Demodulator = $[A + N_1(f)] + [B + N_2(f)]$

Where $N_1(f)$ is the noise in A range frequencies and $N_2(f)$ is the noise in B range frequencies. As $N_1(f)$ is very small it can be neglected so,

Output of the FM Demodulator= $[A] + [B + N_2(f)]$

The noise $N_2(f)$ severely distorts the B range audio frequencies. We therefore, need a way to attenuate this $N_2(f)$ noise.

Let's pass our modulating signal first through a pre-emphasis filter and then use it as a modulating signal. The pre-emphasis filter will not amplify the A range frequencies but will give a gain of *Amp(f)* to frequencies in the B range. *Amp(f)* is a frequency dependent amplification factor. Higher frequencies are amplified more than the lower frequencies.

So now

Output of the pre-emphasis filter = Input to the FM Modulator = $A + [Amp(f)\ B]$

The modulated output is sent via antenna. Now when the signal is demodulated on the receiver side, $N_2(f)$ noise will get into the B range frequencies thereby corrupting it. So

Output of the FM Demodulator = $A + [Amp(f)\ B + N_2(f)]$

The output of the FM demodulator is passed through a de-emphasis filter which tries to remove the amplification *Amp(f)* provided by the pre-emphasis filter. So the de-emphasis filter essentially multiplies the B range frequencies with an inverse amplification (attenuation) of $1/Amp(f)$, so

Output of de-emphasis filter= $A + [Amp(f)\ B + N_2(f)]/Amp(f)$

$= A + [Amp(f)B\ /\ Amp(f) + N_2(f)\ /Amp(f)]$

$= A + [B + N_2(f)/Amp(f)]$

So with the help of pre-emphasis and de-emphasis we have hugely reduced the noise in B range frequencies from $N_2(f)$ to $N_2(f)/Amp(f)$. This is the whole concept behind pre-emphasis and de-emphasis.

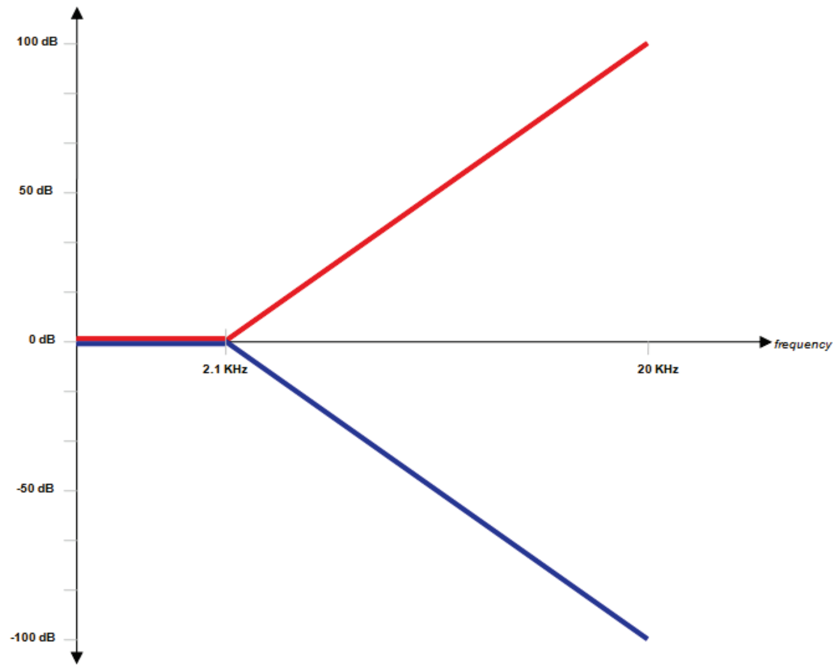


Figure 12.4 Response of the Pre-emphasis (in RED) and De-emphasis filters (in BLUE). The pre-emphasis filter amplifies the frequencies in 2.1 - 20 KHz range. Higher frequencies in this range suffer higher gains compared to the lower frequencies. The de-emphasis filter on the other hand, tries to cancel the effect of pre-emphasis. So it provides higher attenuation to higher frequencies in the 2.1 - 20 KHz range and lower attenuation to lower frequencies in this range. Bothe filters frequency response curves should have the same but opposite slopes for these filters to cancel the effect of each other.

Design of the Pre-emphasis Filter: Pre-emphasis filter is essentially a high pass filter with a gain of K to shift the frequency response so that it starts from 0 dB. To understand the above statement, let's consider a high pass circuit like the one shown in Figure 12.5.

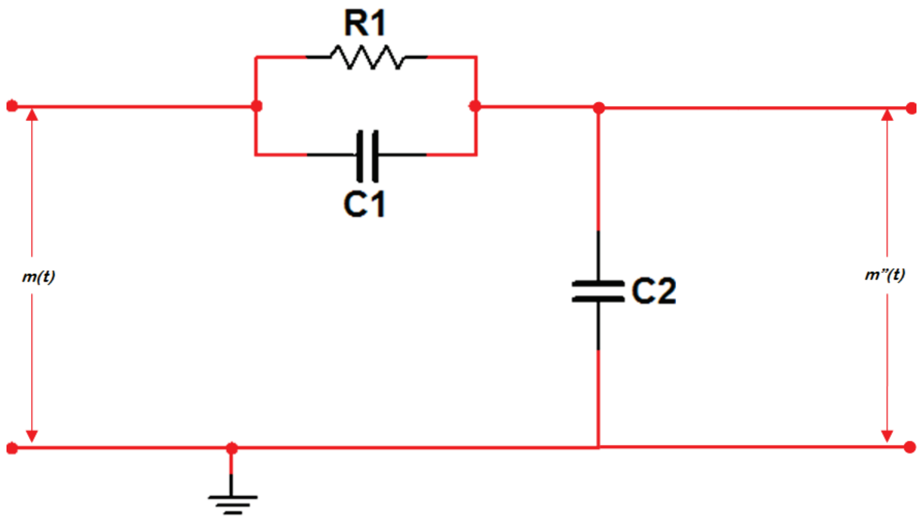


Figure 12.5 A high pass filter

The frequency response of this circuit is

$H(\omega) = m''(t)/m(t) = (j\omega + \omega_1)/(j\omega + \omega_2)$

When this equation is plotted, we get a frequency response as shown below:

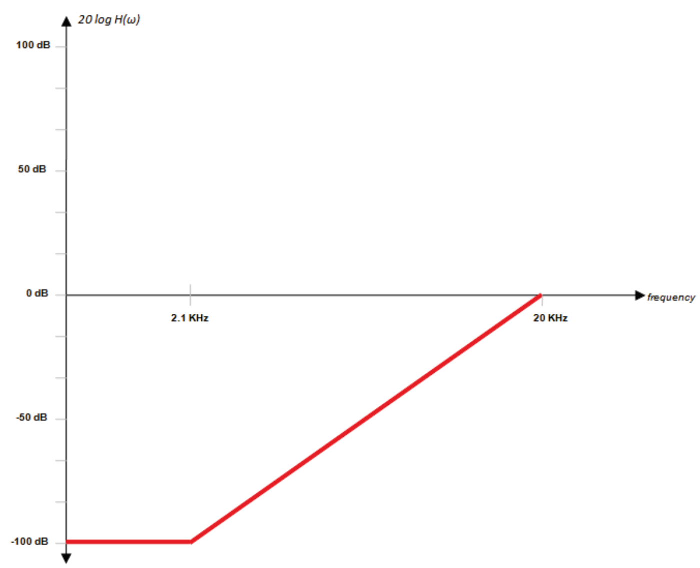


Figure 12.6 Frequency Response of the high pass filter in Figure 12.5. The frequencies between 0 to 2.1 KHz have been attenuated and the frequency between 2.1 KHz and 20 KHz suffer a gradually decreasing attenuation

To convert this high pass filter into a pre-emphasis filter which will provide 0 dB gain to A range frequencies and a gradual gain to frequencies in the B range, we need to shift the response of this filter so that it starts from 0 dB. To do this, we add an amplifier (providing an amplification K) in front of the high pass filter. See Figure 12.7

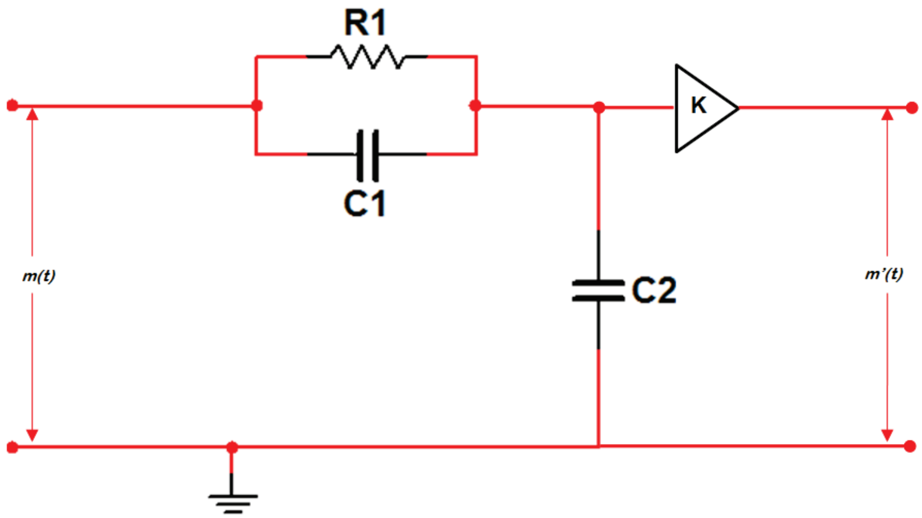


Figure 12.7 Pre-emphasis filter is high pass filter with a gain of K

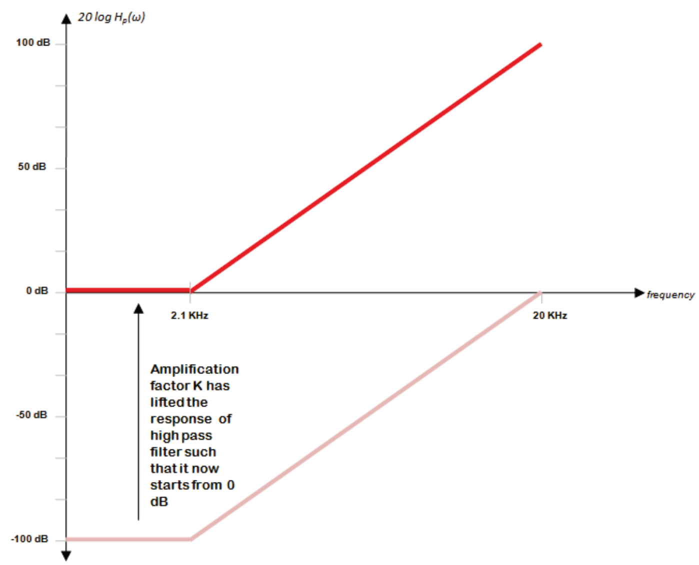


Figure 12.8 Frequency response of the high pass filter with a gain of K.
The amplification K has lifted the response of the high pass filter so that it now starts from 0 dB. The frequencies between 0 - 2.1 KHz pass without any attenuation or gain and the frequencies between 2.1 KHz and 20 KHz get a gradual amplification. Higher frequencies in B range get a higher gain.

Mathematically, the frequency response of this pre-emphasis filter is

$H_p(\omega) = m'(t)/m(t) = K(j\omega + \omega_1)/(j\omega + \omega_2)$

This response is plotted in Figure 12.8 in dark RED.

Design of a De-emphasis filter: A De-emphasis filter has a frequency response exactly opposite to that of the pre-emphasis filter. The de-emphasis filter is basically a low pass filter. It provides 0 dB gain to the A range frequencies but provides a gradual attenuation to the B range frequencies in-order to remove the effect of pre-emphasis.

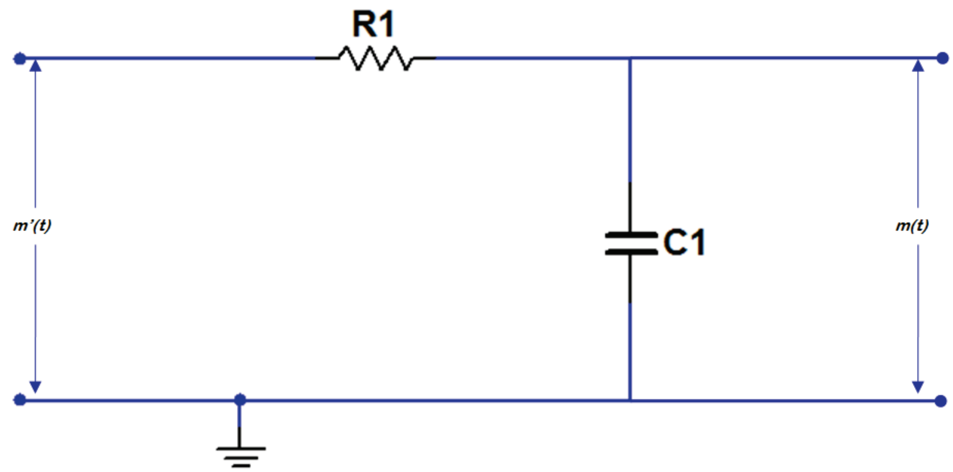


Figure 12.9 A low filter used for de-emphasis

The frequency response of this filter is given by

$H_d(\omega) = m(t)/m'(t) = \omega_1/(j\omega + \omega_2)$

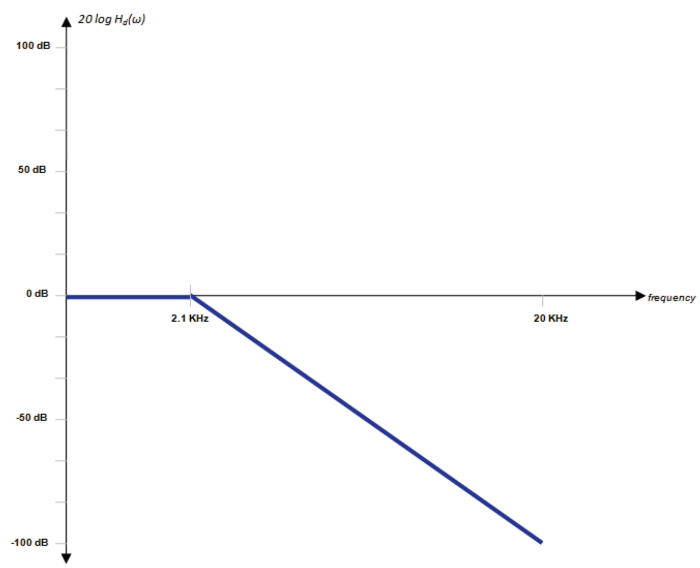


Figure 12.10 Frequency response of the De-emphasis filter
This filter provides no attenuation or gain to frequencies in the 0 - 2.1 KHz range and provides a gradual attenuation to frequencies in 2.1 - 20 KHz range. Higher frequencies suffer more attenuation compared to lower frequencies in this range

Procedure

Power up the transmitter and receiver and connect the RF out of the transmitter to the RF-in of the receiver using a BNC-BNC RF shielded cable. Dummy load should be 'In' when using direct cable connection between transmitter and antenna.

On the transmitter side:

1- Set the modulation source to 1.2 KHz on the left channel and 2.4 KHz on the right channel. Turn both the Gain knobs fully counter clock-wise to turn off the gain. Gain is important only when the modulating signals are coming from the microphones. Other than that, gain is not required.
2- Observe the signals before the pre-emphasis filter (L & R) and after the pre-emphasis (L' & R') on the oscilloscope. You will observe that the signal gets distorted after pre-emphasis. This is because high frequency components in the signal have been amplified.
3- Observe the L' + R' and L' - R' signals on the oscilloscope. Also observe the DSB-SC modulated L'-R' signal
4- Now observe the composite signal at the output of the summing amplifier.
5- Now connect the probe to output of the VCO. No matter how hard you try, you won't be able to see anything on the scope. This is because to see 93-95 MHz signals on an oscilloscope the bandwidth of the oscilloscope must be at least 1 GHz or more. Since we have 100 MHz oscilloscope at the very best therefore, we won't be able to see the output of VCO on the scope. So don't get confused if you don't see any signal at VCO output.

On the receiver side:

6- Turn both the left and right speaker to a high enough volume so that you can hear any channels being tuned to.
7- Now turn the Tuning dial from 93-95 MHz. you will start to receive the transmitted signal in this range. Set the tuning dial to get the best possible reception.
8- The output of the IF filter is an FM modulated signal at 10.7 MHz. This signal can be observed on a 100 MHz oscilloscope. So connect the probe to output of the IF Filter. The output of the IF filter is very small. If you connect he probe in 1X mode then the probe will eat away most of the IF filter's output. This would leave a negligible signal for the circuits ahead in the FM demodulation chain. This is called loading effect of the probe. To prevent this from happening, switch the probe to 10X mode. In this mode only 10% of the signal is given to probe while the rest 90% is available to the circuits ahead in the FM demodulation chain. You will see that the output of IF Filter will look like a pure sinusoid. But it's not a sinusoid it's actually an FM signal. It looks like a sinusoid because of two reasons:
    1- 10.7 MHz is still a very high frequency signal to correctly see on a 100 MHz oscilloscope. The oscilloscope is operating is at upper limit. You need a much higher bandwidth(about 500 MHz) oscilloscope if you want to see the output of IF very clearly.
    2- The frequency deviation of FM is 150 KHz. This deviation of 150 KHz is very small compared to 10.7 MHz carrier frequency. So it's very difficult to observe such as small frequency deviation on 10.7 MHz carrier.

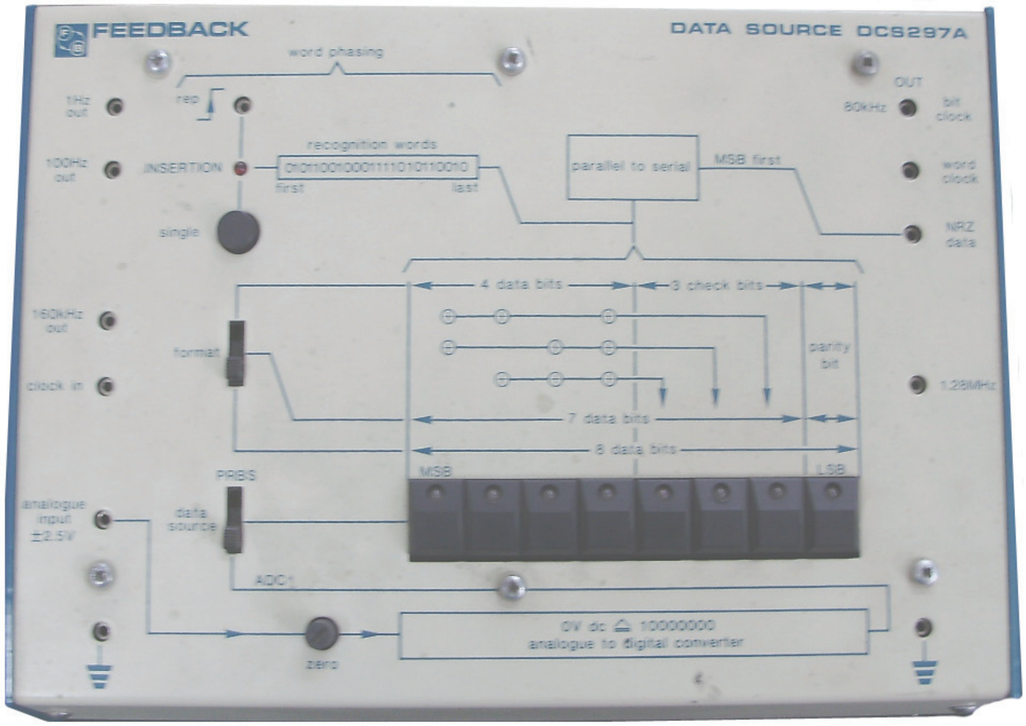
    So in short, you will see a sinusoid on the IF filter's output. But keep in mind that it's not a sinusoid instead it's the FM modulated signal.

9- Now observe the output of the PLL demodulator. It will be similar to the composite signal at the transmitter side.
10- Now that we have got the composite signal back at the receiver side, it's time to get back the left and right channel.
11- Observe the de-emphasized left and right signals on scope. You will also hear the demodulated on the left and right speaker as well. Left and right speakers will sound different. This is because the left signal is a 1.2 KHz triangular wave and the right signal is 2.4 KHz triangular wave.
12- Turn off the De-emphasis and pre-emphasis and see what happens to the demodulated output on the oscilloscope.

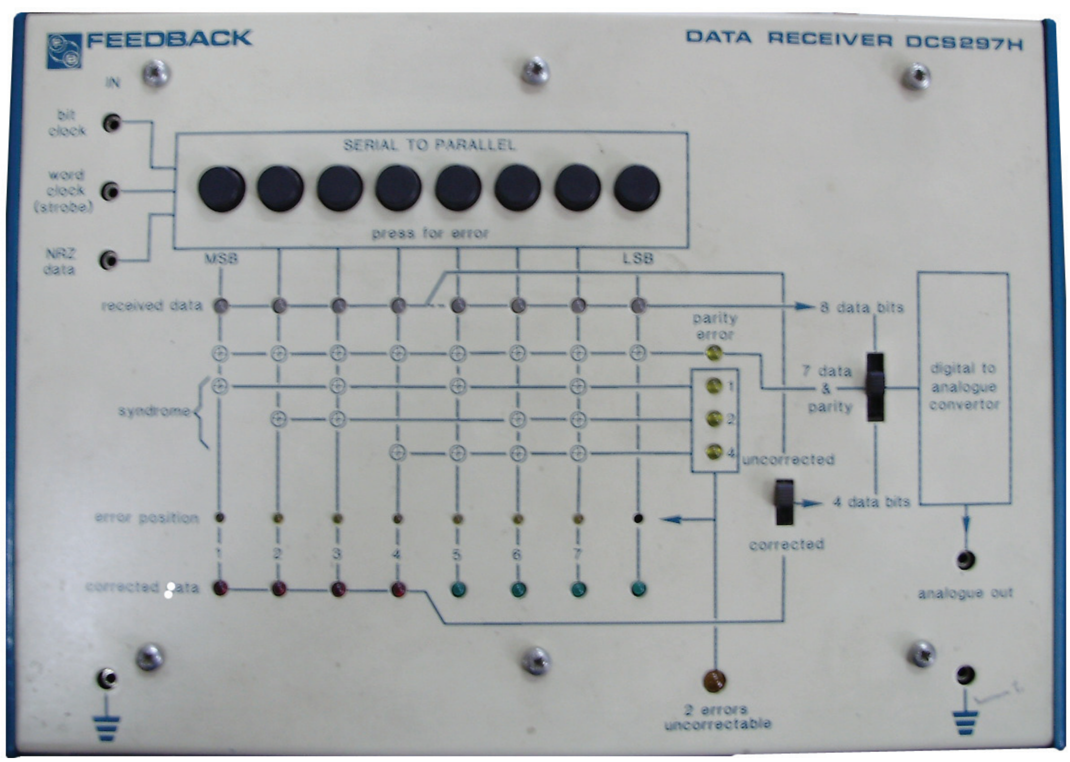
Practical 13-14 List of Equipments Required

1- Power Supply **DSC297M**
2- Data Source **DCS297A**
3- Data receiver **DCS297H**
4- **TRIO CS-1040** 40 MHz Oscilloscope with probes
5- Function Generator **FG601**
6- Frequency Counter **FC-756**

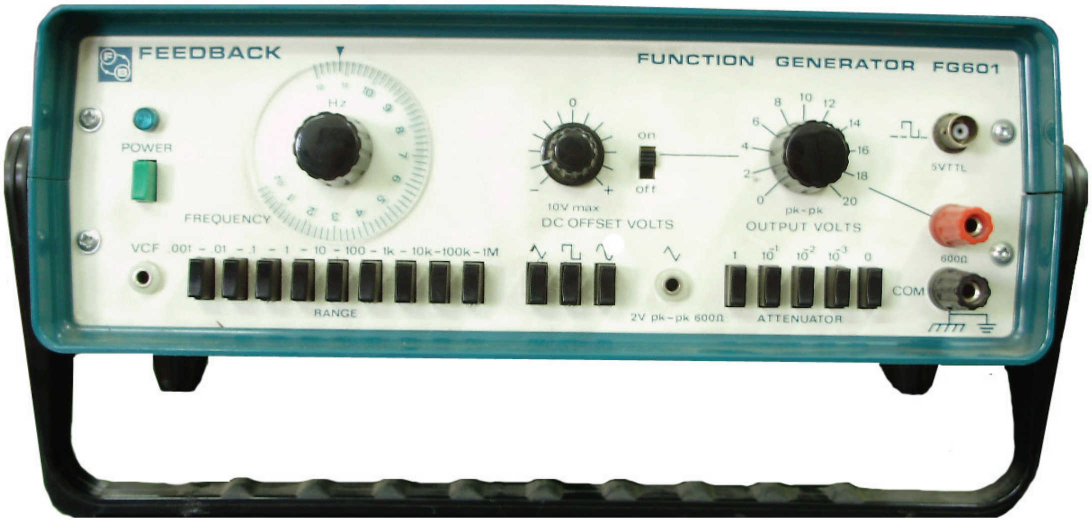
Power supply DSC297M



Data Source DCS297A



Data receiver DCS297H



Function Generator FG601

Frequency Counter FC-756

Practical 13: Sending and Receiving Binary Data

Background Information

In this practical, Binary Coded information is created with the help of 8 digital switches. The 8 bit word thus created, is transmitted serially over a serial link to the receiving unit where it is displayed on the LEDs. The data sent and the data received are essentially the same as the transmission is error free.

Bit Clock and Word Clock are also sent to the transmitter. The Word Clock helps the transmitter to locate the boundary of each word, whereas the Bit Clock helps the receiver in locating the boundary of each bit within a word.

Procedure

1- On the Data Source module connect '160 KHz clock' to 'Clock In'
2- Put (PRBS/Data Source/ADC1) switch to 'Data Source' position.
3- Set 'Format' switch position to '8 Data Bits'
4- Connect 'Bit Clock out' of data source module to 'Bit Clock In' of the data receiver module
5- Connect 'Word Clock Out' of data source module to 'Word Clock In' of the data receiver module
6- Connect 'NRZ Data Out' of data source module to 'NRZ Data In' of the data receiver module
7- Connect Grounds of both data source and data receiver modules
8- Now press the data switches on the data source module. The LEDs will light up according to the position of switches. The data pattern set using the switches will be converted to a serial data format. This serial data will be transmitted via the serial link. We are transmitting the word clock and bit clock from the data source module to the receiver module so as to assist the receiver in recovering the serial data being transmitted. The serial data received will be converted back to parallel and displayed on the LEDs in the receiver module

Practical 14: Analog to Digital conversion, the transmission of digital data and its conversion from Digital back to Analog

Background information

In this experiment, the analog data coming from a Signal Generator is converted to digital format by using an ADC. The ADC's sampling rate is 10 KHz or 10 KSps (Kilo Samples per second). The digital samples produced by the ADC are transmitted serially and received by the Data Receiver Module where it is converted back to analog using a DAC.

As the sampling rate of ADC is 10 KSps therefore, according to the Nyquist Sampling theorem the maximum frequency component in the analog signal being digitized should not exceed 5 KHz. If the analog signal's frequency increases beyond 5 KHz then Aliases would start to appear in the reconstructed output. For example if the Analog signal frequency is set to 11 KHz then the reconstructed output from the DAC will be a signal of frequency 1 KHz. The frequency of aliases can be easily calculated using the formula given below:

Alias Frequency = | Signal Frequency – Sampling Frequency |

Procedure

1- On the Data Source connect '160 KHz Clock' to 'Clock In'
2- Put ([3]PRBS/Data Source/ADC1) switch to 'ADC1' position.
3- Set 'Format' switch position to '8 Data Bits'
4- The ADC must be zero adjusted first. Turn 'Zero' knob so that the output pattern shown on LEDs is 1000 0000. Now the ADC is properly zero adjusted. It's now ready for use
5- Connect the output of the function generator to Analogue Input of the ADC in the Data Source module. Also connect the output of the signal generator to a Frequency Counter to accurately measure the Frequency of the Function Generators' output
6- Set the Function Generator to produce a sinusoid of frequency 1 KHz with a peak value of 2.5 Volts. The maximum input voltage range of the ADC is ±2.5 Volts, so the analog signal to the ADC must not exceed this range, otherwise the excess signal will be clipped and consequently in the output of the DAC will be a clipped version of the analog signal
7- Connect 'Bit Clock out' of data source module to 'Bit Clock In' of the data receiver module
8- Connect 'Word Clock Out' of data source module to 'Word Clock In' of the data receiver module
9- Connect 'NRZ Data Out' of data source module to 'NRZ Data In' of the data receiver module
10- Connect Grounds of both data source and data receiver modules
11- The reconstructed analog output from the DAC is taken from the 'Analogue Out' of the data receiver module.
12- Observe the input signal to ADC and output of the DAC on oscilloscope. Use TRIO CS-1040 Oscilloscope for this purpose. To properly see and compare both signals, 'Trigger Source' should be set to VMODE. The 'Trigger Mode' should be set to AUTO. Now adjust the 'Level' knob so that you see both the signals clearly.

[3] PRBS is an acronym for Pseudo-Random Bit Sequence.

13- Compare the analog signal from the function generator and the reconstructed analog signal. You will see that the reconstructed signal will not be a smooth one. Instead it will contain steps that are reminiscent of the quantization introduced by the analog to digital converter

14- Furthermore, you will observe that the reconstructed output will lag behind the original input. This time lag is mathematically given by

Total Lag Time = ADC Conversion Time + DAC Conversion Time

where 'ADC Conversion Time' is the time taken by the ADC to convert the analog sample into digital format. And 'DAC Conversion Time' is the time taken by the DAC to convert the digital sample into analog format

Observing Aliasing

15- To observe aliasing, set the frequency of the input to 11 KHz.

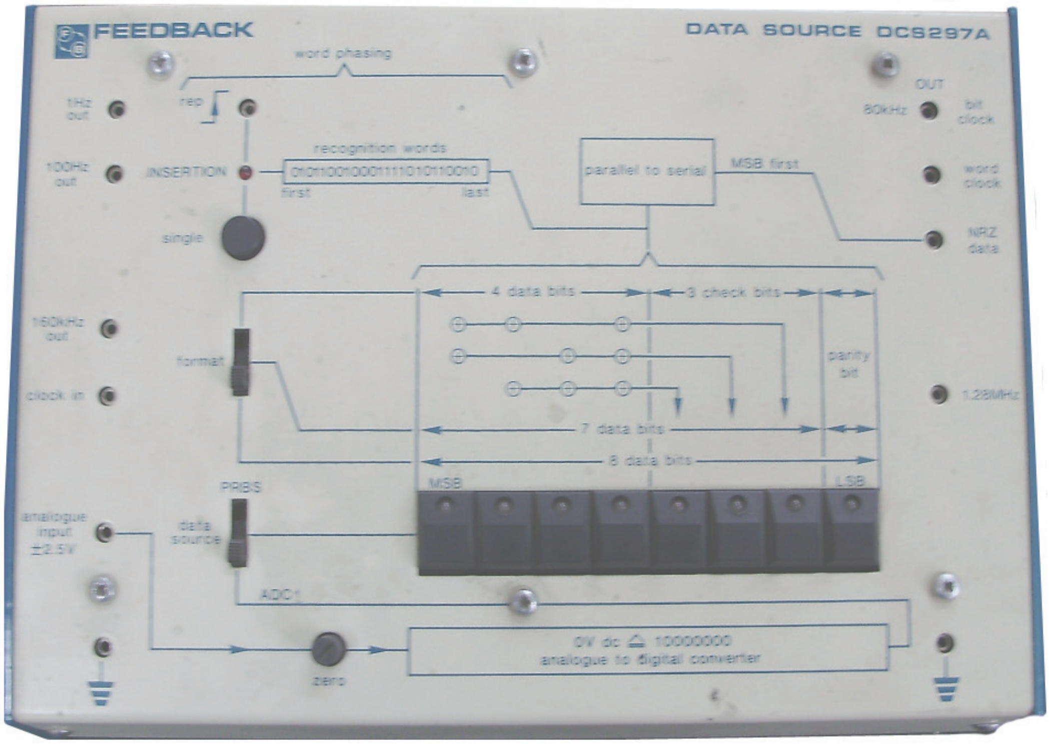
16- You will observe that the reconstructed output is 1 KHz sinusoid. This 1 KHz output is the Alias of the 11 KHz signal

17- Similarly when the input has a frequency of 12 KHz, the output will be a sinusoid of frequency 2 KHz

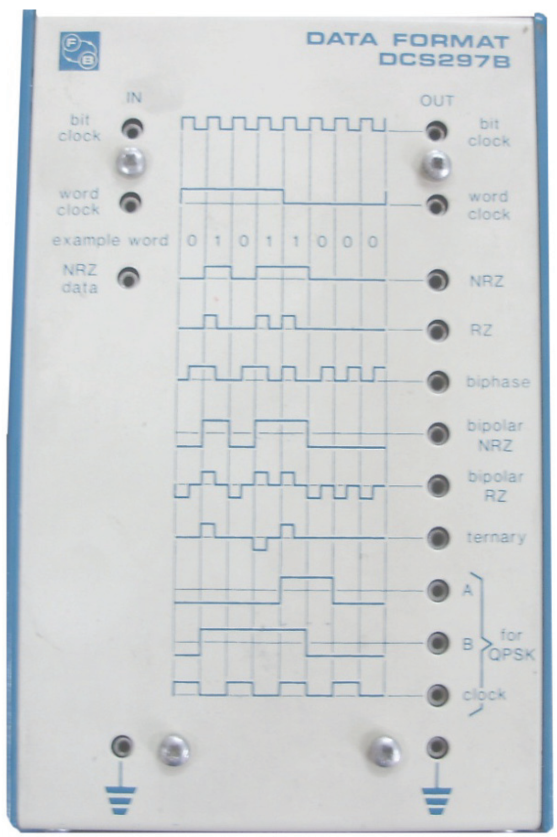
Equipments Required for Practical 15

1- Power Supply **DSC297M**
2- Data Source **DCS297A**
3- **TRIO CS-1040** 40MHz Bench top Oscilloscope
4- Data Format **DCS297B**

Power supply DSC297M



Data Source DCS297A

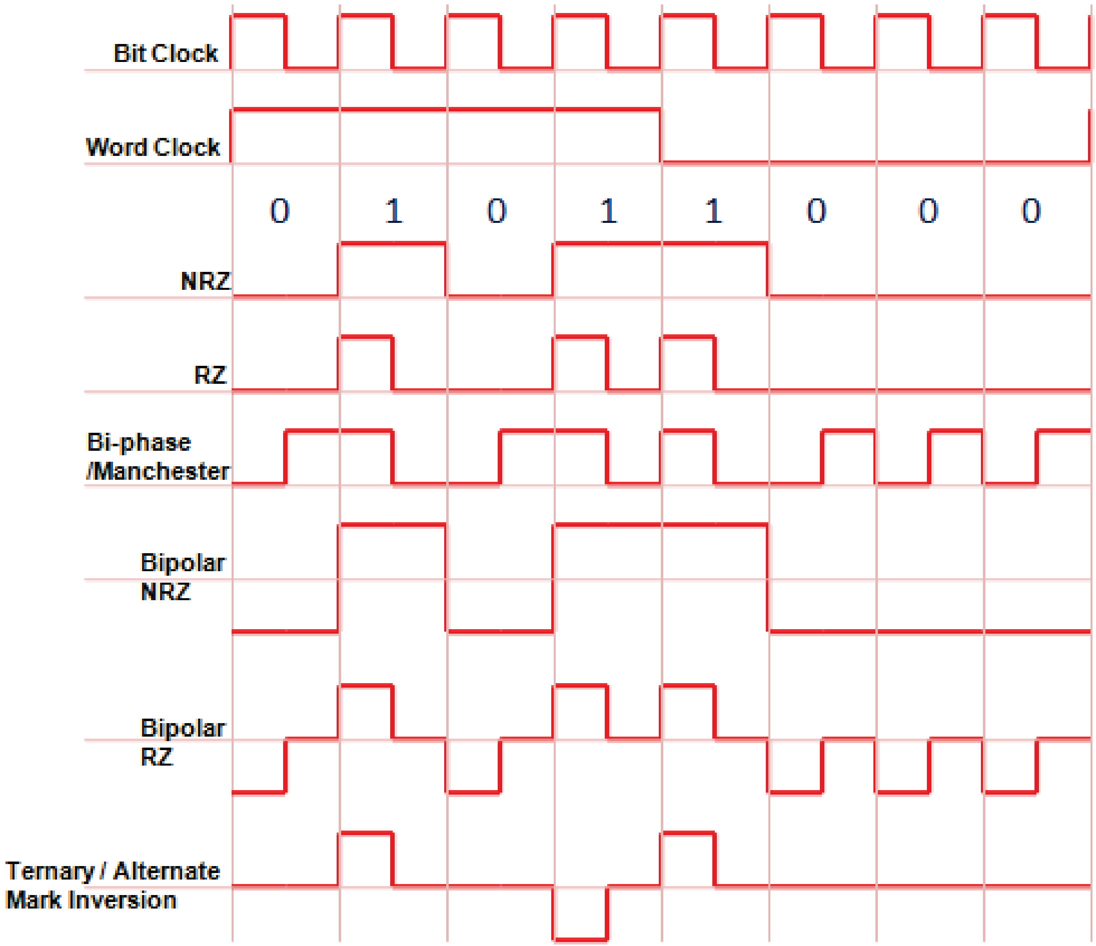


Data Format DCS297B

Practical 15: To observe different data formats

Objective

Digital data bits (0 and 1) can be encoded using different line encoding techniques. In this experiment you will some of these encoding techniques



6 Different encoding techniques

(The digital bits encoded in this example are 0101 1000*)*

Procedure

1- Connect ‘Bit Clock out’ of data source module to ‘Bit Clock In’ of the data format module
2- Connect ‘Word Clock Out’ of data source module to ‘Word Clock In’ of the data format module
3- Connect ‘NRZ Data Out’ of data source module to ‘NRZ Data In’ of the data format module
4- Connect Grounds of both data source and data receiver modules
5- Connect ‘Bit Clock Out’ and ‘Word Clock Out’ of the Data format module to the CH-1 and CH-2 of the oscilloscope.
6- Apply triggering from CH-2(i.e. from the channel to which word clock has been applied)
7- Apply a data of 0101 1000 from the data source module
8- Connect the CH-3 of the oscilloscope to the data format that you want to observe on the data format module
9- The data format seen on the oscilloscope will be similar to patterns shown on the Data Format Module

A primer to Analog Oscilloscopes

Oscilloscope is an electronics test and measurement gadget. It's one the most important tools in an Electrical Engineers' arsenal. Therefore, it's imperative that should understand the internals of an oscilloscope. Understanding how an oscilloscope works will help you in not only making correct and accurate measurements but would also make the learning process fun and enjoyable.

Types of Oscilloscopes

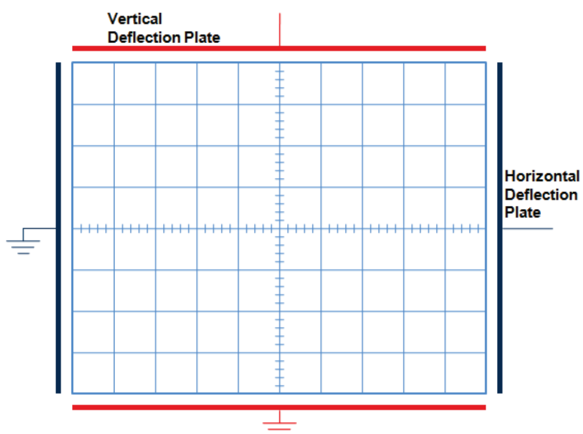
There are various types of oscilloscopes a few of which are mentioned below

1- Analog Oscilloscope or Cathode Ray Oscilloscope(CRO)
2- Digital Phosphor Oscilloscope or DPO
3- Digital Storage Oscilloscope or MSO

Because our lab has only Analog oscilloscopes therefore, I will confine this discussion to Analog Oscilloscopes.

Analog Oscilloscope

An analog oscilloscope contains a Cathode Ray Tube. The screen of this tube is coated with a chemical compound of phosphorus called *Phosphor*. When electron beam falls on the phosphor, it glows thereby making the electron beam visible. The phosphor continues to glow (for a very short time) even when the electron beam is no more there. This glowing of screen even when the electron beam vanished is called the Persistence and it's crucial for proper display of signals on the oscilloscope.



This figure shows the front of the CRT. The four plates shown are internal to the oscilloscope and are not normally visible to the user. Two of these plates are always grounded

Horizontal Deflection Plate

The Horizontal Deflection Plates control the movement of the electron in the horizontal or the X direction. When all the deflection plates are grounded, then the electron suffers no deflection and therefore, falls right in the middle of the tube thereby illuminating a spot on it.

All deflection plates are grounded. The electron beam suffers no deflection and therefore falls right in the middle of the tube

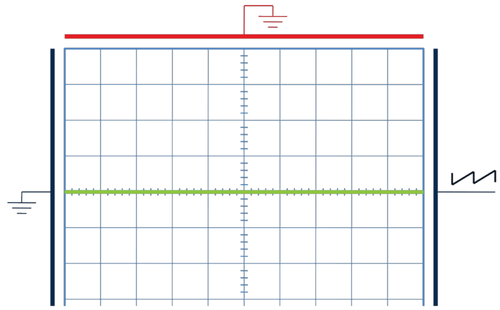
However, during the normal operation of an oscilloscope, a periodic Sweep signal is applied to the horizontal deflection plate. A sweep signal is a constantly increasing ramp signal. When sweep signal is negative the electron beam is repelled by the horizontal deflection plate to the extreme left of the screen and when the sweep signal is positive, the electron beam is attracted towards the deflection plate. The sweep signal is responsible for moving the electron beam from left to right. The time taken by the electron to sweep from left to right is controlled by the time period of the Sweep signal applied to the horizontal deflection plate. The smaller this time period, the faster the electron beam will sweep across the screen and vice versa.

When the sweep voltage is –V, the electron beam is pushed to the left corner of the CRT

,

After half the sweep time, the sweep voltage reaches 0, and therefore, the electron beam reaches the center of the screen

At the end of the sweep period, the sweep voltage reaches +V and the electron beam reaches the extreme right of the CRT. So observe that as the sweep voltage increases from –V to +V, the electron beam sweeps from left to right on the CRT. The time taken by the electron to sweep from left to right is equal to the time period of the Sweep signal and can be controlled by the time base knob of the oscilloscope



As sweep signal is periodic and is being continuously applied at the horizontal deflection plate, therefore the electron beam continuously moves left to right during each sweep. This usually happens at such a fast rate that due to the persistence of the phosphor, you only see a straight line. When you increase the sweep period to some value in seconds, only then you will be able to observe the electron beam slowly moving from left to right

Vertical Deflection Plate

Any signal a user wants to measure or observe on the oscilloscope is applied to the vertical deflection plate. This signal will cause the electron to move in vertical or y- direction at the same time when electron beam is moving from left to right under the influence of the sweep signal applied on the horizontal deflection plate. This combined x and y movement is what causes a signal shape to be formed on the screen. The figures below discuss a case when the user applies a sinusoidal signal to the oscilloscope for measurement. The RED waveform is the signal to be measured and is applied to the red colored vertical deflection plates. The blue signal is sweep signal is applied to the to the blue colored horizontal deflection plates and the Blue

The signal in Red is applied to the vertical deflection plate and it is the signal to be measured or viewed on the oscilloscope. The Blue signal is the sweep signal and is applied to the horizontal deflection plate. As the red signal is zero at point A, therefore there is only x deflection in the electron beam due to the sweep signal

As time progress, the red signal changes from A to B and therefore, causes a proportional y-deflection in the electron beam while its moving left horizontally under the influence of the sweep signal

As time progress, the red signal changes from B to C and therefore causes a proportional y-deflection in the electron beam while its moving left under the influence of the sweep signal

As time progress, the red signal changes from C to D and therefore causes a proportional y-deflection in the electron beam while its moving left under the influence of the sweep signal

As time progress, the red signal changes from D to E and therefore causes a proportional y-deflection in the electron beam while its moving left under the influence of the sweep signal

Because both the Sweep signal and the signal to be measured are periodic, therefore the electron retraces the same path ABCDE again and again during all the consecutive sweeps. Therefore, you will see a continuous trace of the signal on the oscilloscope screen

Triggering

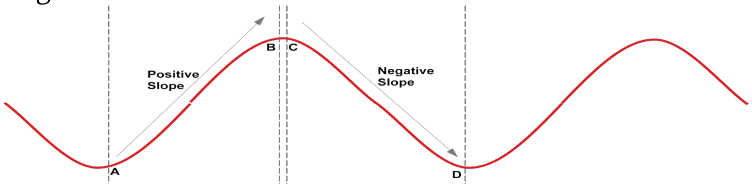
Triggering is the proper selection of the hold off time of the sweep signal, such that the electron beam traces the same path in every sweep. If triggering is not proper, you won't be able to view the signal properly on the oscilloscope. Discussed below diagrammatically are two cases. One is improperly triggered and the other is properly triggered.

This is the case of improper triggering. During the first sweep AB, the electron beam traces a path A'B' on the CRT. But during the sweep CD, the electron beam traces a path C'D'. Similarly, during the third sweep EF, the electron beam traces the path A'B and so on. So instead of following one path, the electron beam follows two paths A'B' and C'D' alternately. Therefore, you see two traces on the scope's screen. The display does not resemble the original signal applied. So that's why you can't view the signal if the oscilloscope is not properly triggered

This is the case of proper triggering. Now during sweep AB, the electron beam traces a path A'B' on the scopes' screen. During the time CD, the sweep is held back (i.e. it is held at -V). The electron beam just sits at the left corner of the screen during this time. During the sweep DE, the electron beam retraces its path A'B' again and so on. Using hold off time, we have suppressed the annoying C'D' path. Now the display of the oscilloscope resembles the original signal. Hold-off time knob is used to adjust the hold off time

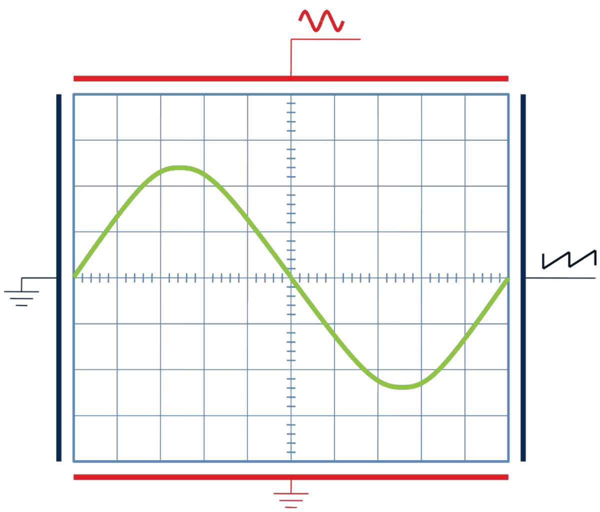
Trigger Slope and Level

Slope: Through Trigger Slope knob you can control whether the display starts from the positive going or negative going slope of the signal waveform

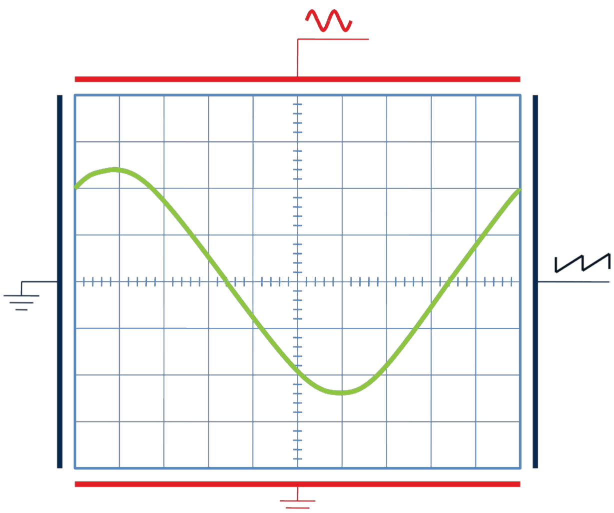


If positive slope is selected, then the oscilloscope display will start from some point in the AB region. If the Negative slope is selected then the oscilloscope display will start from some point in the CD region. The oscilloscope cannot be properly triggering in the BC region because it has neither positive nor negative slope

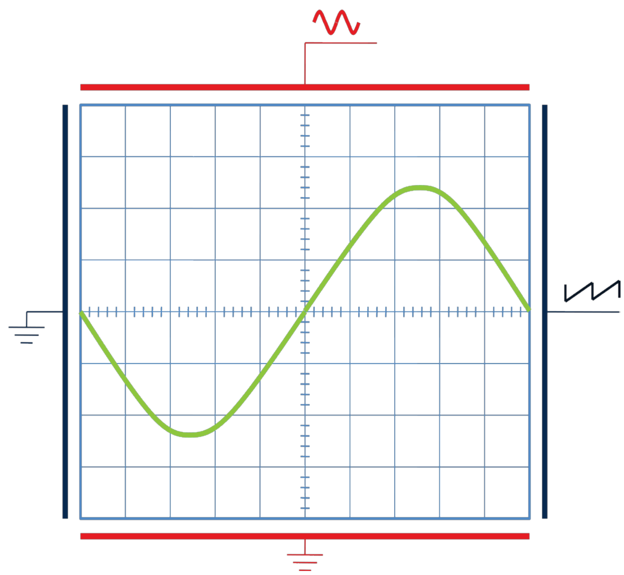
Level: The level knob controls at which point in the region AB /CD the waveform should start.



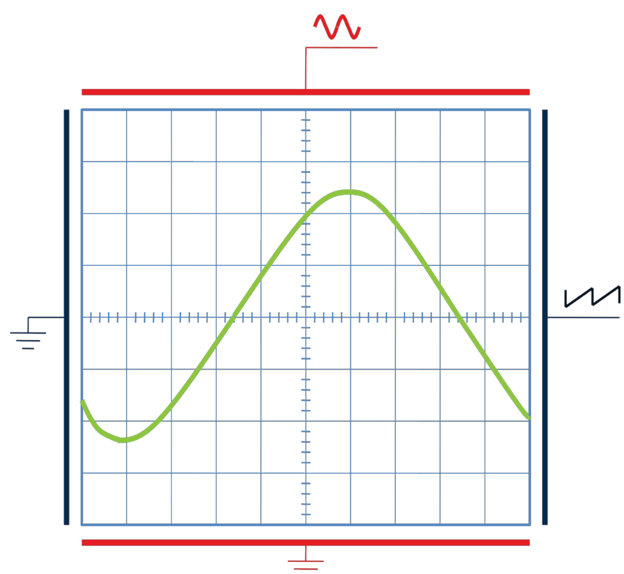
This is the display when Slope is set to Positive and the Level is set to zero



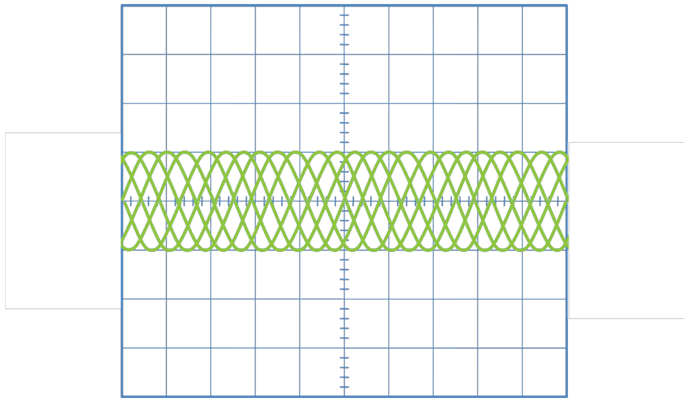
This is the display when Slope is set to Positive and the Level is set to some value greater than zero



This is the display when Slope is set to Negative and the Level is set to zero



This is the display when Slope is set to Negative and the Level is set to some value below zero



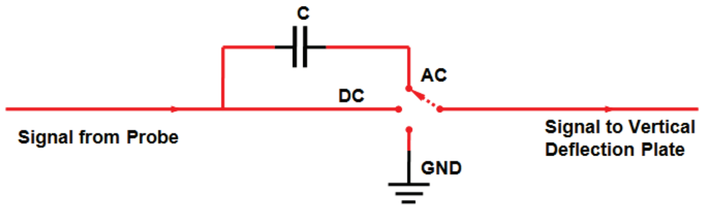
This is the case when you set the level in the BC region. As there is no slope in the BC region, therefore, the oscilloscope cannot be properly triggered and you won't be able to see the signal properly. You will just see a garbage waveform like the one shown in this figure

Signal or Input Coupling

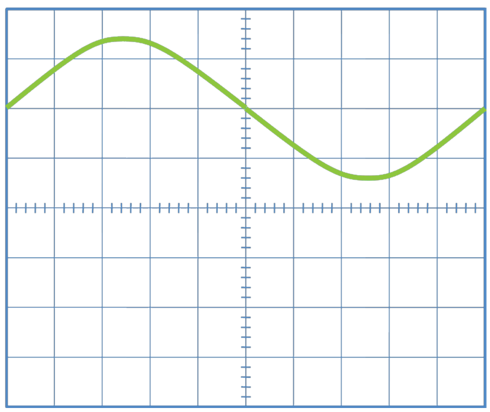
Oscilloscopes have an AC-GND-DC coupling switch. When its set at AC position, only AC portion of the signal is displayed on the oscilloscope and the DC portion is blocked. AC position is used, when an AC signal is riding on a very large DC offset. Such signals cannot be viewed properly unless the DC offset is removed. The AC position comes handy in such scenarios.

If the switch set at GND then a straight line is displayed on the oscilloscope as the vertical deflection plate gets grounded

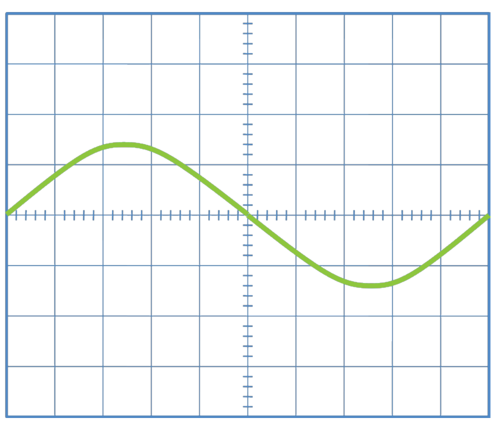
When set at DC position, the signal is displayed as it is. The DC portion of the signal is not blocked.



AC-GND-DC Switch



Signal with a DC offset. The coupling switch is set at DC position. Both AC and DC portion of the signal is displayed

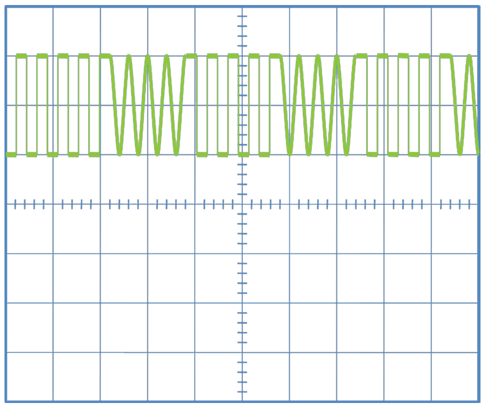


This figure shows the same Signal with a DC offset. But the coupling switch is set at AC position. Only AC portion of the signal is displayed. The DC portion of the signal gets blocked

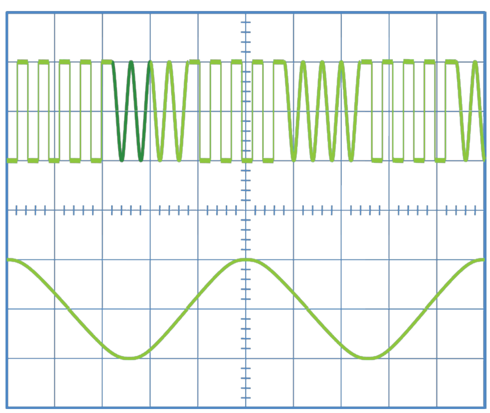
Delayed Sweep Mode

Sometimes you will need to see a signal and some of its portion magnified both simultaneously on the oscilloscope. In such scenarios, we use the delayed sweep feature of the oscilloscope. This feature is available only in advanced oscilloscopes.

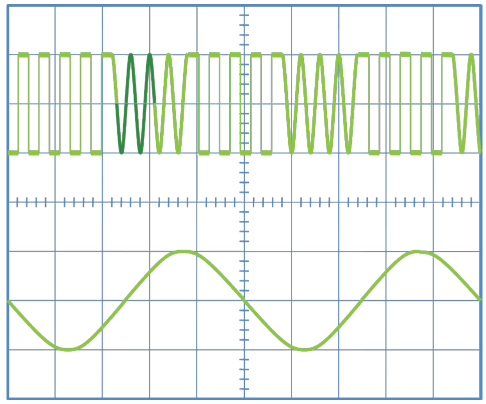
In this mode a bright window appears on the original signal. The portion of the signal inside the window is displayed zoomed.



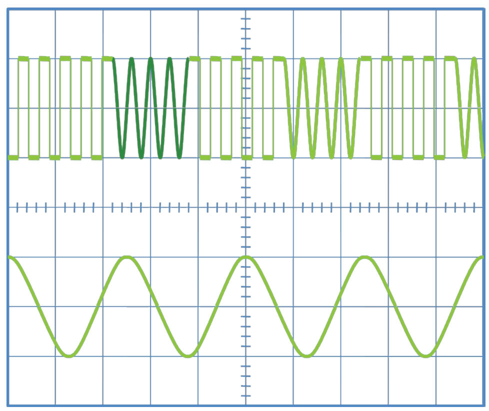
We need to see both the complete signal and some of its portion zoomed, both simultaneously on the oscilloscope. We therefore, need to activate the delayed sweep mode



Delayed Sweep is activated. The darker portion shows the window superimposed on the original signal. The contents of the window are zoomed and displayed at the bottom. The zoomed display is called B signal and the original signal above it is called A signal.



The window can be moved left or right using 'Delay Time Mult' knob. In this case, we have moved the window to the right. As the window moves the zoomed display updates as well to display the contents of the window



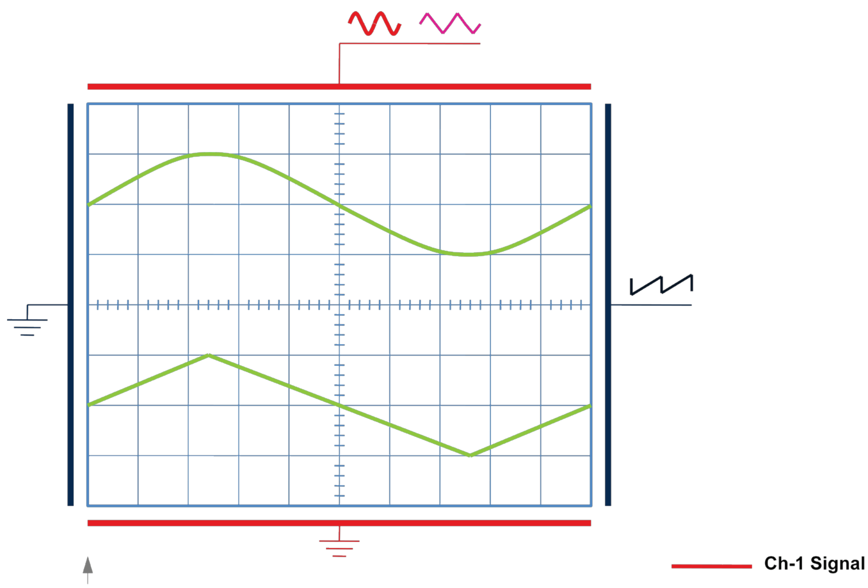
The size of the window can be changed using the 'B-Sweep' knob. Usually this knob is located on top of the Time base knob but its size is smaller than the time base knob. In this case, we have increased the window size. The zoomed view updates as well to display the contents of the window

ALT and CHOP Modes

These modes are used to display multiple waveforms on the oscilloscope. The problem with displaying multiple waveforms is that we have only one electron beam. So we can only plot one waveform at a time. To overcome this problem ALT and CHOP techniques are used

ALT Mode

In Alt mode the signal on vertical deflection plates switches between the multiple signals to be viewed. For the first sweep, the signal on the vertical deflection plate is from Ch-1. During the second sweep, the signal on vertical defection plate is from Ch-2. In third sweep, the signal on the vertical plate is again from Ch-1 and so on. So during each sweep the signal on the vertical deflection plates alternates between Ch-1 and Ch-2. This is the Alt or Alternate mode. ALT mode is used when the multiple signals to be viewed are high frequency signals around 500 KHz or above.

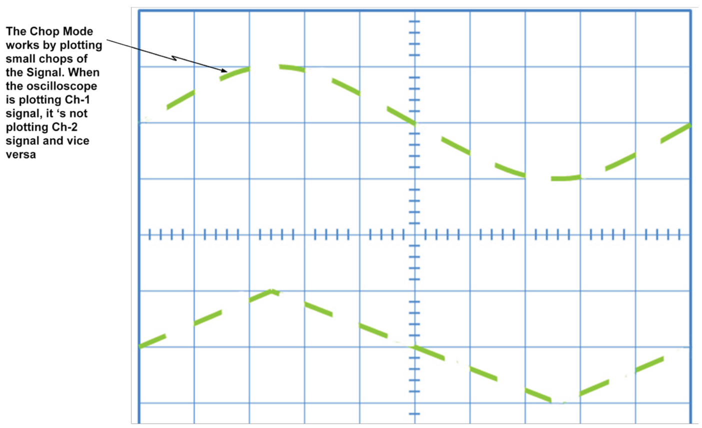


The oscilloscope is working in ALT mode to simultaneously displays a sine wave and a triangular wave. In first sweep the signal to vertical deflection plate is the sine wave. The triangular wave (shown in dotted light pink)is not applied during this time. In second sweep the signal to vertical deflection plate is the triangular wave. The sine wave shown in dotted red is not applied during this time. So the signal to vertical deflection plates alternates between the sine wave and the triangular wave during each consecutive sweep

Chop Mode

In chop mode, first a small portion of Ch-1 signal is plotted then a small portion of Ch-2 signal is plotted. After that again a small portion of Ch-1 signal is plotted and then a small portion of Ch-2 and so on. In contrast to ALT where one of the two channels is plotted during the complete duration of the sweep, in Chop mode only small chunks of the sweep time are dedicated to each channel alternately. That's way you will see chops in the signals being displayed. The chopping frequency depends on the make and model of the oscilloscope but is usually around 250 KHz.

Chop mode should be used when the multiple signals to be viewed are low frequency signals near 500 Hz or 1 KHz.



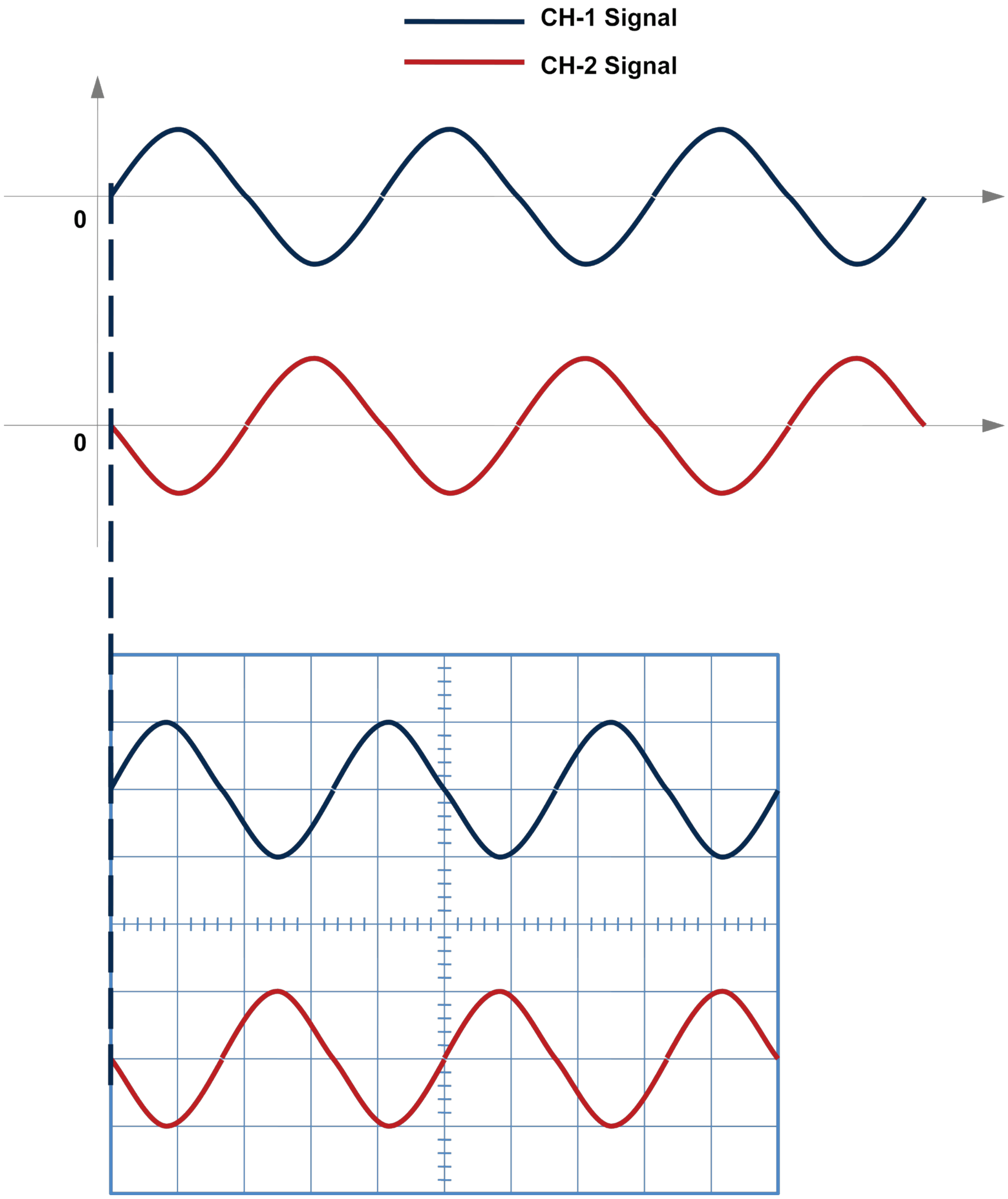
Oscilloscope displaying multiple signals using the Chop mode

Trigger Source

The trigger source switch specifies which signal is under the control of the Trigger Slope/Level knob

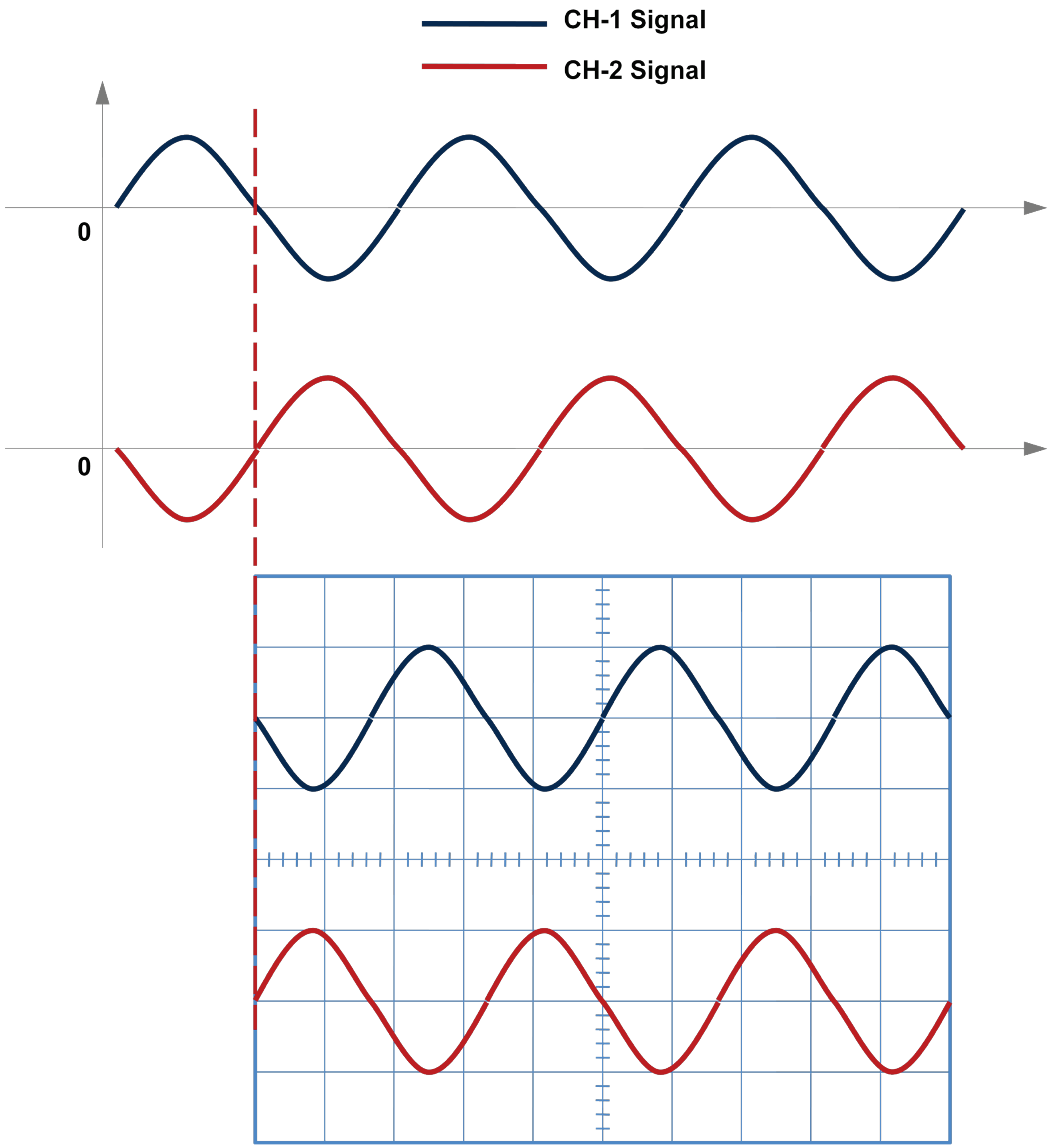
The trigger source switch has the five possible positions:

- CH-1: In this mode the slope/level knob controls the slope and level of only the CH-1 signal. Let's suppose that Level knob is set at 0 and Slope knob is set at positive. The oscilloscope will find 0 level point on the positive slope of the CH-1 signal and start plotting the CH-1 signal from that point. All other signals applied to the oscilloscope will start relative to CH-1 signal at any slope and any level.



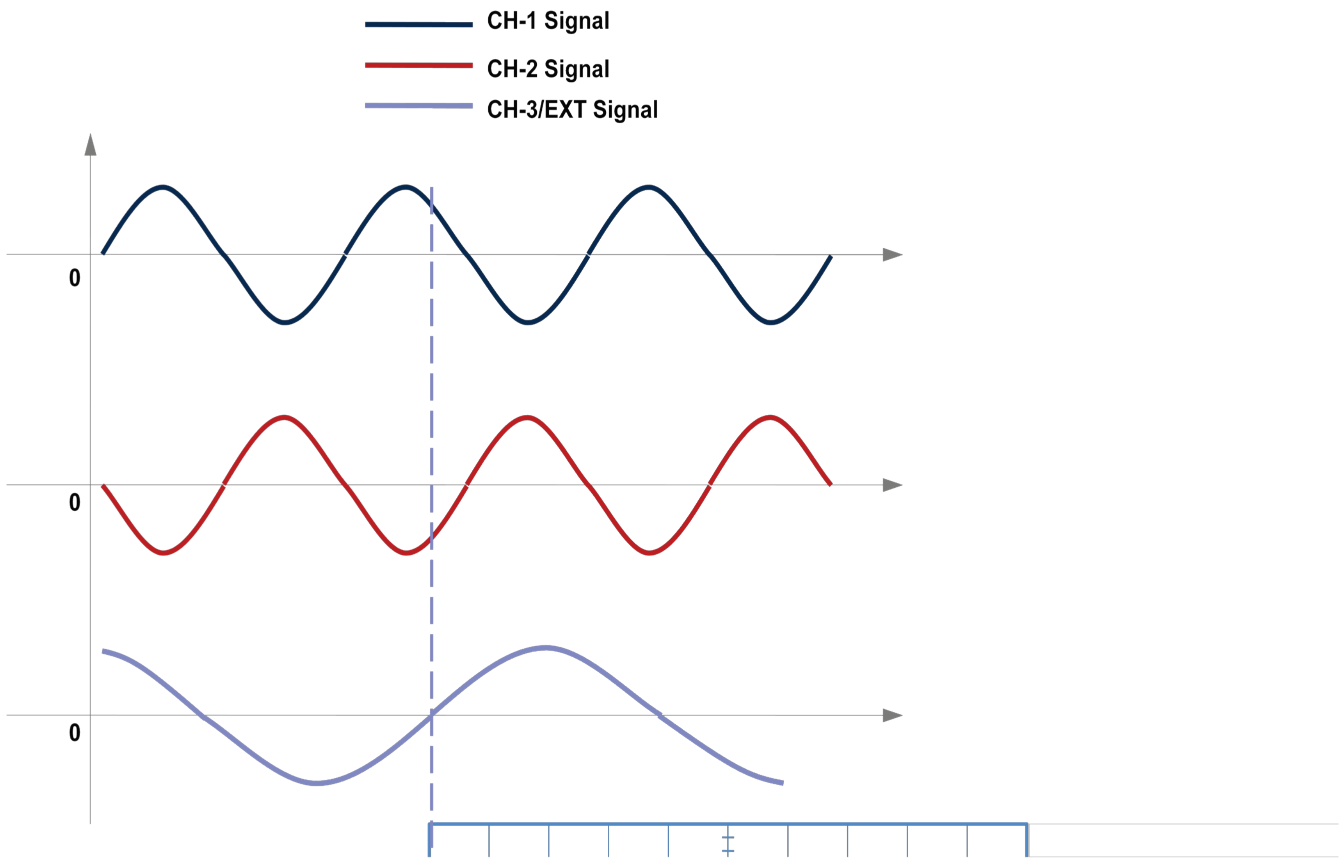
The Trigger Source switch is set at CH-1. The Slope knob is set at positive and the Level knob is set at 0 (Notice in this case the CH-1 signal will start from positive slope at level 0. The oscilloscope job is to make sure that when Trigger Source switch is set CH-1 then, the CH-1 signal should start from positive slope at level 0. All other signals, (in this case the CH-2 signal) can start relative to the starting point of the CH-1 signal at any slope and at any level)

- CH-2: In this mode the slope/level knob controls the slope and level of only the CH-2 signal. Let's suppose that Level knob is set at 0 and Slope knob is set at positive. The oscilloscope will find 0 level point on the positive slope of the CH-2 signal and start plotting the CH-2 signal from that point. All other signals applied to the oscilloscope will start relative to CH-2 signal at any slope and any level.



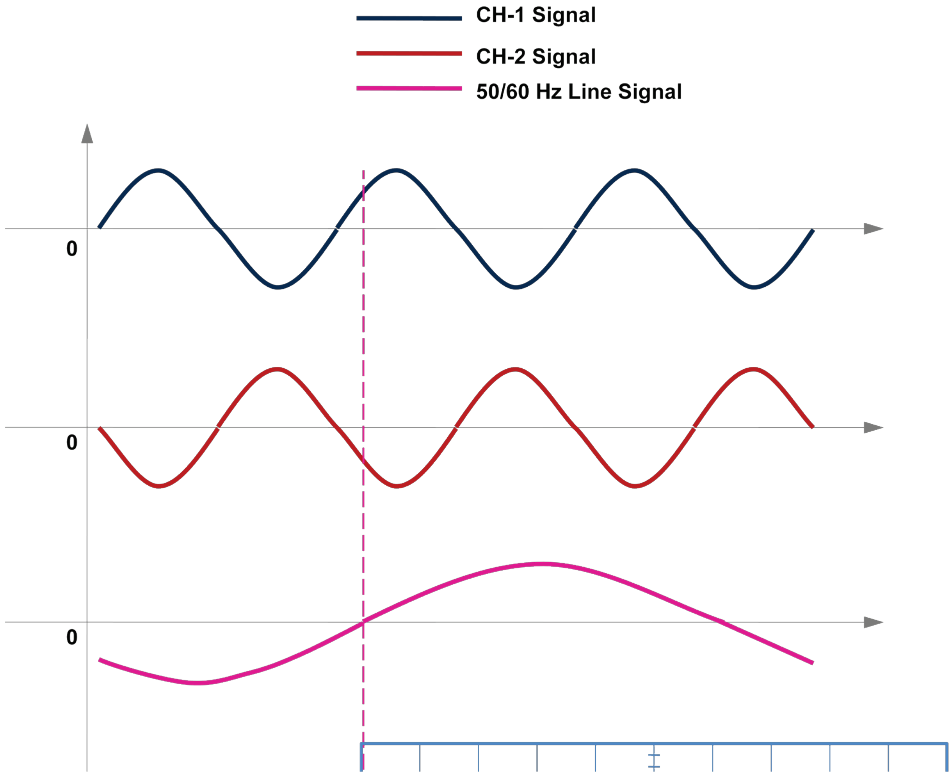
The Trigger Source switch is set at CH-2. The Slope knob is set at positive and the Level knob is set at 0 (Notice in this case the CH-2 signal will start from positive slope at level 0. The oscilloscope job is to make sure the when Trigger Source switch is set CH-2 then, the CH-2 signal should start from positive slope at level 0. All other signals, (in this case the CH-1 signal) can start relative to the starting point of the CH-2 signal at any slope and at any level)

- CH3/EXT: Let's suppose that Level knob is set 0 and Slope knob is set at positive. The oscilloscope will find 0 level point on the positive slope of the CH-3/EXT signal. All other signals applied to the oscilloscope will start relative to CH-3/EXT signal at any slope and any level.



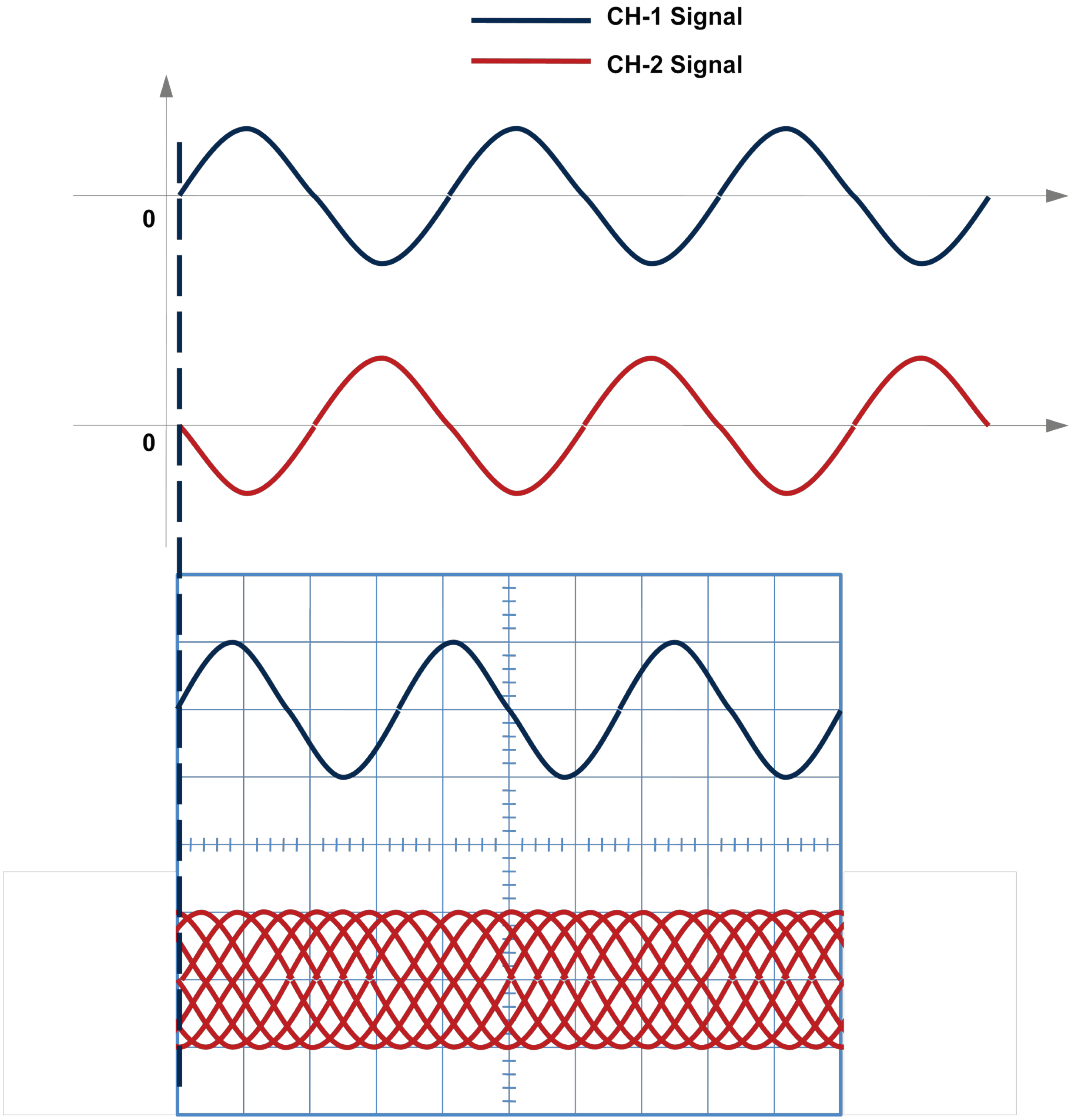
The Trigger Source switch is set at CH-3/EXT. The Slope knob is set at positive and the Level knob is set at 0 and the oscilloscope is working in dual trace mode and therefore, not displaying the CH-3/EXT signal
CH-1 and CH-2 signals will start relative to the Level 0 point on positive slope of the CH-3/EXT signal.

- Line: In this mode the slope/level knob triggers on the slope of the 50-60 Hz line voltages. Let's suppose that Level knob is set 0 and Slope knob is set at positive. The oscilloscope will find 0 level point on the positive slope of the Line signal. All other signals applied to the oscilloscope will start relative to the Line signal at any slope and any level. This mode is useful for observing signals from power line transformer and rectifiers



The Trigger Source switch is set at Line. The Slope knob is set at positive and the Level knob is set at 0 and the oscilloscope is working in dual trace
CH-1 and CH-2 signals will start relative to the Level 0 point on positive slope of the Line signal.

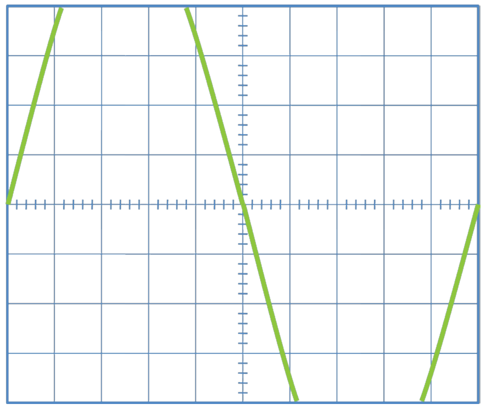
- VMODE To understand this mode, let's suppose that Level knob is set 0 and Slope knob is set at positive. In this mode the oscilloscope will find 0 level point on the positive slope of all the signals applied to oscilloscopes. So all signals will start from level 0 on their positive slope. If one of the signals doesn't have a positive slope at the same time while other signals have the positive slope, then that signal cannot be properly triggered and, therefore, will not be displayed properly on the scope. In figure below, you can see that when CH-1 has positive slope, CH-2 has negative slope. So the oscilloscope cannot find positive slope on both the signals simultaneously and therefore, only one of the signal is displayed properly



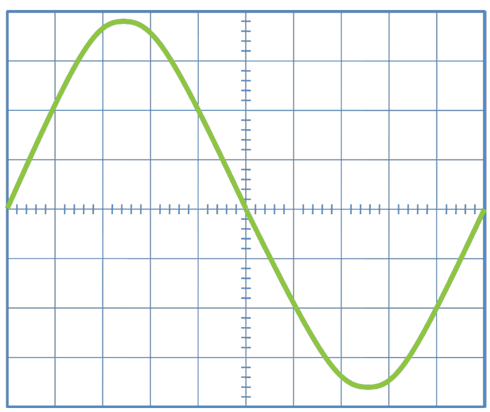
The Trigger Source switch is set at VMODE. The Slope knob is set at positive and the Level knob is set at 0 The CH-1 signal starts from level 0 on a positive slope. The CH-2 signal cannot be properly triggered because whenever CH-1 signal has positive slope, CH-2 signal has negative slope. The oscilloscope cannot find positive slopes on both CH-1 and CH-2 signals simultaneously when they are 180 degrees out of phase. So the CH-1 signal gets triggered on the positive slope but CH-2 signal cannot get triggered properly at the same time because it doesn't have a positive slope during that time. That's why if you want to observe both these signals properly you will have to set the trigger source switch to either CH-1 or CH-2

Uncalibrated Mode of the oscilloscope

Sometime the signal to be viewed on the oscilloscope is so large that you can't observe it completely on the oscilloscope screen even when you have turned the Volts/Div knob to the highest possible setting. For such cases, you can turn the 'CAL' knob to uncalibrated position. In uncalibrated position, the signal gets attenuated and therefore, squeezes into the available scope screen. As you have uncalibrated the scope therefore, any reading taken along the voltage axis would not be correct. This mode is used only when you need to see the shape of the waveform and you are not concerned about measuring its peak value



This figure shows are very large signal that cannot be displayed completely on the scope's screen even when the highest possible Volts/Div setting is engaged

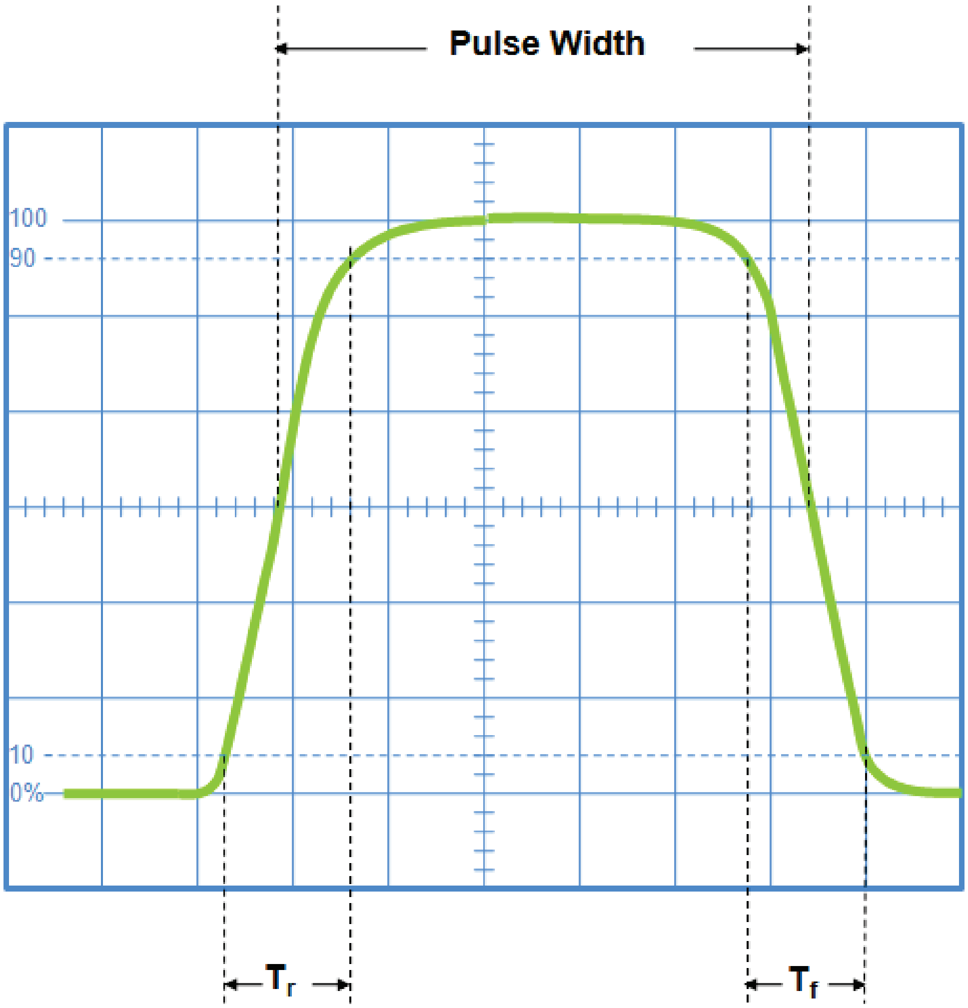


The same signal can be seen on the oscilloscope by turning the CAL knob in the uncalibrated direction. The signal is attenuated and therefore, fits in the viewing area of the oscilloscope

Rise time, Fall time and Pulse Width Measurement

Oscilloscope s can be used to measure the rise and fall time and the pulse width of a digital signal. For this purpose digital signal is applied to one of the oscilloscopes channel. The amplitude of the digital signal is adjusted such that it fits completely between 0 and 100% lines [this amplitude adjustment can be done using the Volts/Div knob (and the CAL knob if required)].

The rise time T_r is the time taken by a digital waveform to rise from 10% to 90%. The fall time T_f is the time required by the digital waveform to fall from 90% value to 10% value. Pulse width is the time taken by a waveform to go from 50% mark of it's on the positive slope to the 50% value on the negative slope



This figure shows the rise time, fall time and pulse width measurement of a digital pulse. The waveform should be exactly aligned to the 0% and 100% markers.

XY Mode or Lissajous Mode

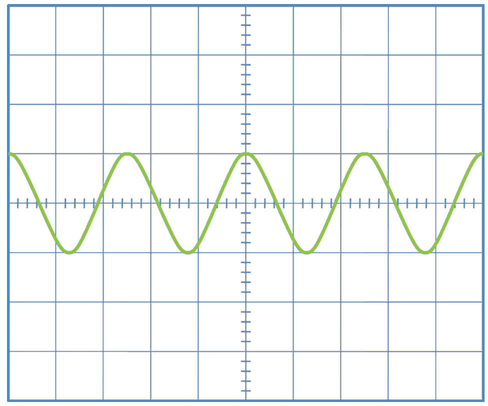
Oscilloscopes normally work in Voltage versus Time mode with time is along horizontal axis and voltage along vertical axis. However, oscilloscopes can also work in Voltage versus Voltage mode. This mode is called the XY mode or the Lissajous Mode. In XY- Mode, the sweep signal is disconnected from the horizontal deflection plate. CH-1 signal is applied to horizontal deflection plate and the CH-2 signal is applied to the vertical deflection plate. The plot formed in this mode is called a Lissajous Pattern.

This figure shows the Lissajous Pattern of two 90° out of phase sinusoids with same amplitude frequency. The red waveform is applied to the Y-Channel and hence controls the y deflection and the blue waveform is applied to X- Channel and therefore, controls the horizontal deflection of the electron beam. At point A on the waveform, the vertical deflection of the electron beam is positive and the horizontal deflection is zero. So the electron beam is present at point A'. At point B on the waveform, the horizontal deflection is positive and the vertical deflection is zero so the electron beam is at point B' on the oscilloscope. And so on. So as the waveforms goes through ABCDE, the electron beam follows the circular path A'B'C'D'E'. If the frequency of the two sinusoids is low enough then you can actually see the electron beam moving in the circle

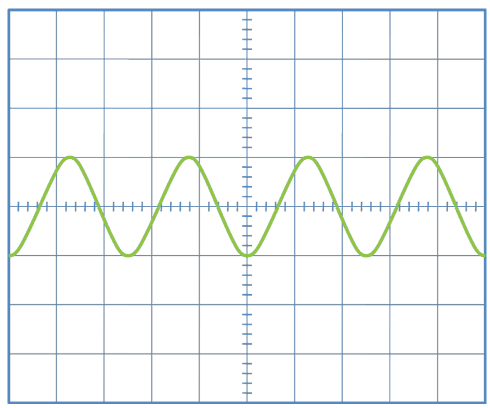
This figure shows the Lissajous Pattern of two 90° out of phase sinusoids with same amplitude and frequency. If the frequency of the two sinusoids is high enough, then instead of seeing an electron beam moving in a circle, you will only see a stable circular plot like the one in this figure

CH-2 INV

Most oscilloscopes have this button. When pressed the signal on CH-2 of the oscilloscope gets multiplied by -1.



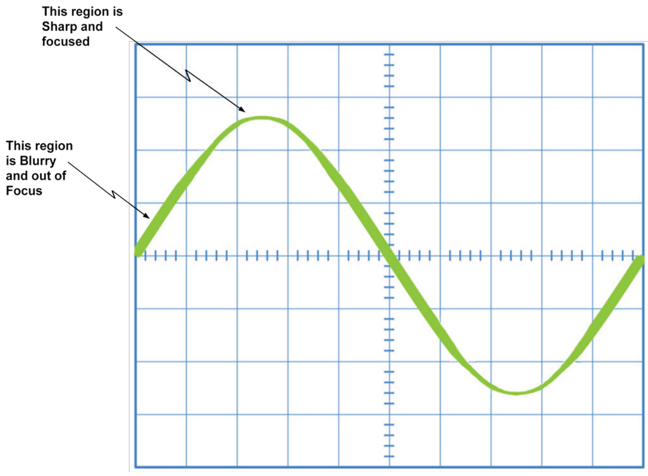
This is the signal applied on CH-2 of the oscilloscope



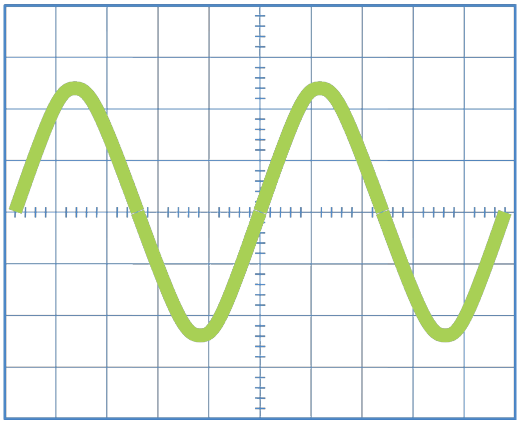
This is what you see when the CH-2 INV button is pressed. The waveform suffers a 180° Phase change.

Astigmatism

The oscilloscope is said to have astigmatism when some portions of its display are well focused while others are out of focus. ASTIG knob is used to remove the astigmatism error. With the help of this knob we can make all portions of the display to have a uniform focus.



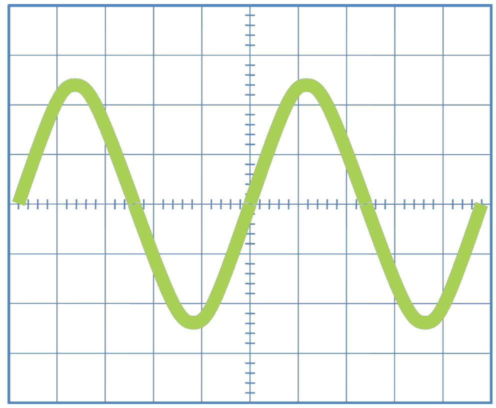
This figure shows an oscilloscope with astigmatism error.
Some portions of the display are well focused whereas others are out of focus



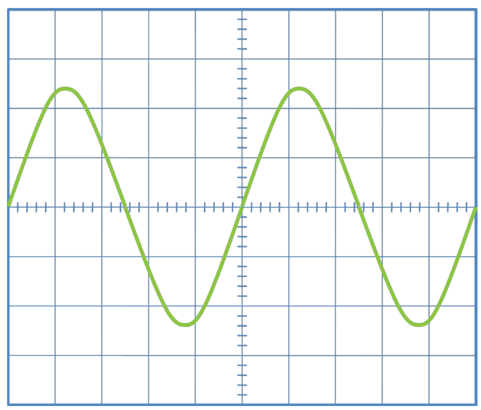
This is the oscilloscope display with no astigmatism error.
All portion of the waveform have a uniform focus and thickness

Focus

The focus knob control the sharpness of the display



The oscilloscope display is not properly focused

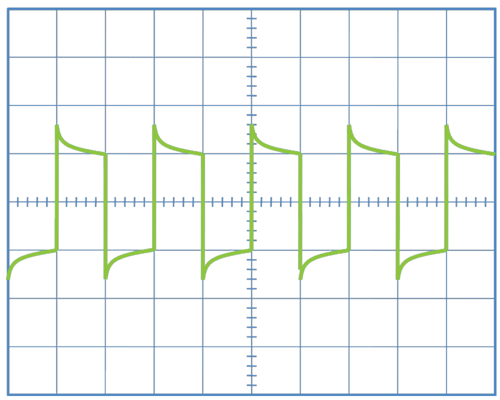


Now the oscilloscope display is properly focused and therefore looks very clear and sharp

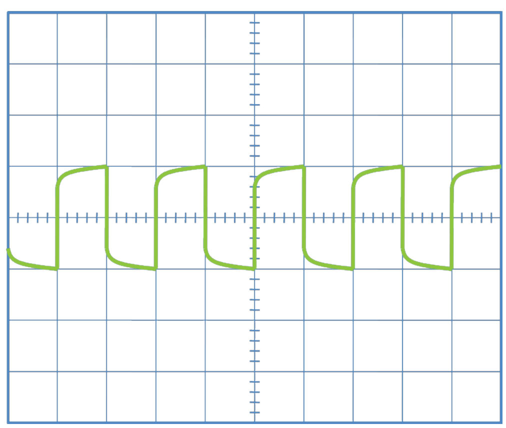
Probe and its calibration

Probe is the cable through which the signal to be measured is tapped off from a circuit and supplied to the oscilloscope for measurement. Most of the probes have a 1X - 10X switch. When the probe is set at 1X then the signal gets to the oscilloscope without any attenuation. But when the switch is set to 10X, then a 10 times attenuated signal reaches the oscilloscope for measurement. Therefore, if the probe is set at 10X mode then all the voltage measurements should be multiplied with 10 to get the original value of the signal voltage.

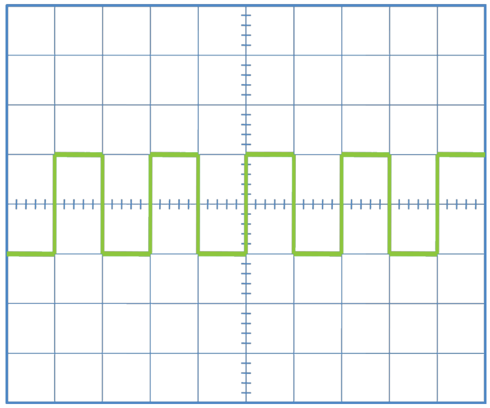
However, there is a problem which arises when using the probe in 10X mode. The signal gets distorted after it passes through the probe in 10X mode. To remove this distortion a procedure called *Probe Calibration* is used. The probe is hooked to a 1 KHz square wave that is generated internally in the oscilloscope. A screw driver is used to adjust a variable capacitor inside the probe. The capacitor is adjusted until the probe is properly compensated.



This is the case of an Over compensated probe. Overshooting occurs in the waveform



This is the case of an Undercompensated probe. Undershooting occurs in the waveform



This is the case of a properly compensated probe. In this case the probe provides no distortion to the signal being viewed

A probe is used in 10X mode whenever the signal to be viewed is very weak. If you connect the probe directly to such weak signals then 90% of the signal gets diverted to the oscilloscope leaving only 10% of the signal for the circuitry ahead. This is called loading effect of the oscilloscope. To reduce this loading effect the probe is switched to 10x mode. In this mode, 10% of the signal gets diverted to the oscilloscope and the remaining 90% is available for the circuitry ahead.

Bandwidth Consideration when using an oscilloscope

Generally speaking, if the highest frequency component in the signal to be viewed or measured on the oscilloscopes is X MHz, then you need an oscilloscope with a bandwidth of 10X MHz to be able to visualize and measure the signal.