LABORATORY WORKBOOK

For the Course

COMMUNICATION SYSTEM

(TC-307)

Instructor Name: ______________________________

Student Name: ______________________________

Roll Number: ____________________ **Batch:** ______

Semester: ____________________ **Year:** ______

Department: ______________________________

Department of Electronic Engineering
NED University of Engineering & Technology

LABORATORY WORK BOOK

For The Course

<u>COMMUNICATION SYSTEMS</u>

<u>(TC-307)</u>

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CONTENTS

Lab No.	Dated	List of Experiments	CLO	Signature
01		To examine the main parameters of an AM signal. To check the operation of an Amplitude modulator		
02		To check the operation of the balanced amplitude modulator with suppressed carrier		
03		To examine the main parameters of the single sideband modulation and to check the use of filters to generate the SSB		
04		To investigate the demodulation of an AM signal with an Envelope Detector		
05		To observe the operation of Product Detector		
06		To investigate the demodulation of SSB signals using Product/ Synchronous detection		
07		To observe the characteristics of a Frequency Modulated wave in Time Domain and Frequency Domain.		
08		To analyze the Spectrum of an FM Signal with a Large Modulation Index		
09		To carry out FM demodulation using PLL detector		
10		To study the effects of a limiter and noise on the performance of PLL detector		
11		To examine the functioning of natural and flat sampling PAM modulator		
12		To examine the variations of reconstructed analog signal by changing (a) filter selectivity (b) sampling frequency		
13		To examine the working of PAM receiver		
14		To examine the operations of PWM and PPM modulators operation		

LAB SESSION 01

To examine the main parameters of an AM signal. To check the operation of an Amplitude modulator .

Student Name: ________________________________

Roll Number: ____________________ **Batch:** ____________

Semester: ____________________ **Year:** ____________

Total Marks	
Marks Obtained	

Remarks (If Any): ______________________________

Instructor Name: ________________________________

Instructor Signature: ____________________ **Date:** ____________

LAB SESSION 01

AMPLITUDE MODULATION

OBJECTIVE:

To examine the main parameters of an AM signal. To check the operation of an Amplitude modulator

EQUIPMENT REQUIRED:

Amplitude Modulation Work board 53-130 which comprises the following blocks:

- Signal Generation
- Modulation
- Filters
- Demodulation

THEORY:

Modulation:

The modulation is simply a method of combining two different signals and is used in the transmitter section of a communication system. The two signals that are used are the information signal and the carrier signal. The information signal is the signal that is to be transmitted and received and is sometimes referred to as the intelligent signal. The carrier signal allows the information signal to be transmitted efficiently through the transmission media. The carrier signal is normally generated by an **oscillator** and has a constant frequency and amplitude. The information signal that is fed into the transmitter modifies the carrier signal.

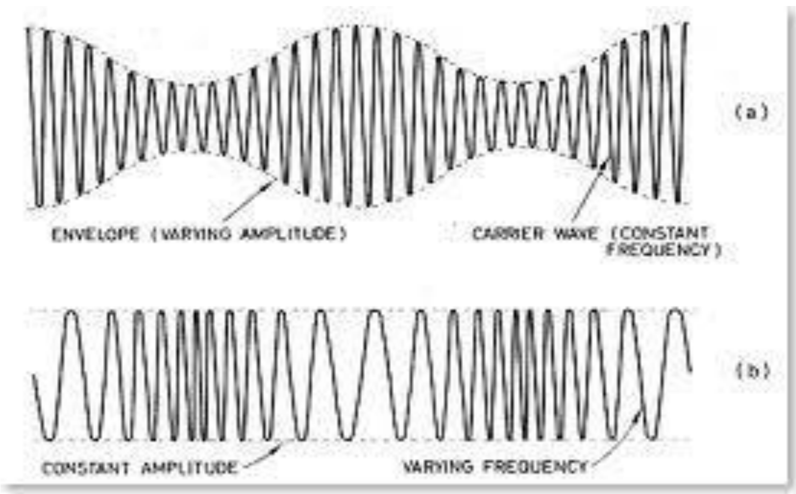

Figure 1.1 Modulation Techniques

Amplitude Modulation:

It is the simplest form of signal processing in which the **carrier amplitude** is simply changed according to the **amplitude of the information signal,** hence the name Amplitude Modulation. When the information signal's amplitude is increased the carrier signal's amplitude is increased and when the information signal's amplitude is decreased the carrier signal's amplitude is decreased. In other words, the **ENVELOPE** of the carrier signal's amplitude contains the information signal.

Modulation index "m" = $\frac{V_{max} - V_{min}}{V_{max} + V_{min}}$

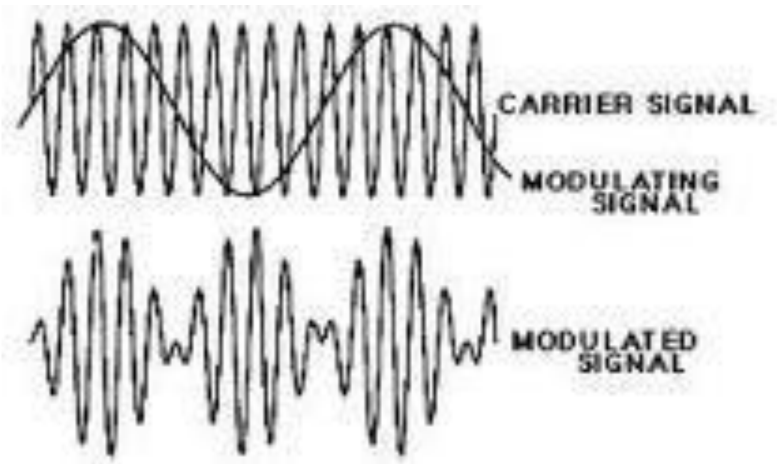

Figure 1.2 Amplitude Modulation

Modulation Mathematics:

The equation of a sinusoidal voltage waveform is given by: $v = V_{max} \sin(\omega t+Ø)$ where:

- v is the instantaneous voltage
- V_{max} is the maximum voltage amplitude
- ω is the angular frequency
- Ø is the phase

Amplitude modulation uses variations in amplitude (V_{max}) to convey information. The wave whose amplitude is being varied is called the **carrier wave**. The signal doing the variation is called the **modulating signal.** For simplicity, suppose both carrier wave and modulating signal are sinusoidal; i.e., $v_c = V_c \sin \omega_c t$ (*c* denotes carrier) and $v_m = V_m \sin \omega_m t$ (*m* denotes modulation)

We want the modulating signal to vary the carrier amplitude, V_c , so that:

$$v_c = (V_c + V_m \sin \omega_m t).\sin \omega_c t$$

where $(V_c + V_m \sin \omega_m t)$ is the new, varying carrier amplitude.

Expanding this equation gives:

$v_c = V_c \sin \omega_c t + V_m \sin \omega_c t. \sin \omega_m t$ which may be rewritten as:

Now:

$v_c = V_c\,[\sin \omega_c t + m \sin \omega_c t . \sin \omega_m t]$ where $m = V_m/V_c$ and is called the **modulation index**

$\sin \omega_c t . \sin \omega_m t = (1/2)\,[\cos(\omega_c - \omega_m)\,t - \cos(\omega_c + \omega_m)\,t]$

so, from the previous equation:

$v_c = V_c\,[\sin \omega_c t + m \sin \omega_c t . \sin \omega_m t]$

we can express vc as:

$v_c = V_c \sin \omega_c t + (mV_c/2)\,[\cos(\omega_c - \omega_m)\,t] - (mV_c/2)\,[\cos(\omega_c + \omega_m)\,t]$

This expression for v_c has three terms:

1. The original carrier waveform, at frequency ω_c, containing no variations and thus carrying no information.

2. A component at frequency $(\omega_c - \omega_m)$ whose amplitude is proportional to the modulation index. This is called the **Lower Side Frequency**.

3. A component at frequency $(\omega_c + \omega_m)$ whose amplitude is proportional to the modulation index. This is called the **Upper Side Frequency**.

It is the upper and lower side frequencies which carry the information. This is shown by the fact that only their terms include the modulation index *m*. Because of this, the amplitudes of the side frequencies vary in proportion to that of the modulation signal.

Sidebands:

If the modulating signal is a more complex waveform, for instance an audio voltage from a speech amplifier, there will be many side frequencies present in the total waveform. This gives rise to components 2 and 3 in the last equation being bands of frequencies, known as sidebands. Hence we have the upper sideband and the lower sideband, together with the carrier. Clearly, for a given carrier amplitude there are limits for the size of the modulating signal; the minimum must give zero carrier, the maximum gives twice the unmodulated carrier amplitude. If these limits are exceeded, the modulated signal cannot be recovered without distortion and the carrier is said to be **over-modulated.**

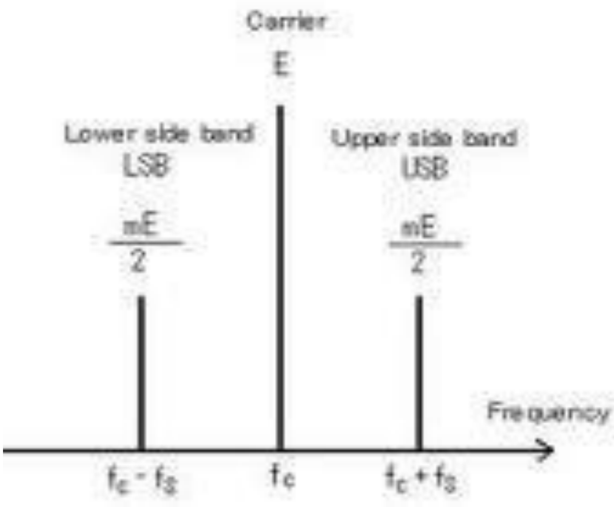

Figure 1.3 Frequency Spectrum of AM Signal

Experimental Determination of the Modulation Index

This is most easily done by measuring the maximum and minimum values which the instantaneous amplitude of the carrier reaches. Let us call these *x* and *y*. Taking our previous equation:

$v_c = V_c [\sin \omega_c t + m \sin \omega_c t. \sin \omega_m t]$

and re-arranging it yet again, we can express v_c

as: $v_c = V_c \sin \omega_c t [1 + m \sin \omega_m t]$

so that the instantaneous amplitude of the carrier

is: $V_c [1 + m \sin \omega_m t]$

Since $\sin w_m t$ can vary between +1 and -

1, $x = V_c (1 + m)$ and $y = V_c (1 - m)$

To get the value of modulation index m from x and y, we eliminate Vc between these equations by division, giving:

$y /x = (1 - m)/(1 + m)$.

Solving for *m* gives:

$\mathbf{m = (x - y)/(x + y)}$

PROCEDURE:

In this practical the hardware is configured as shown in figure 1.4.

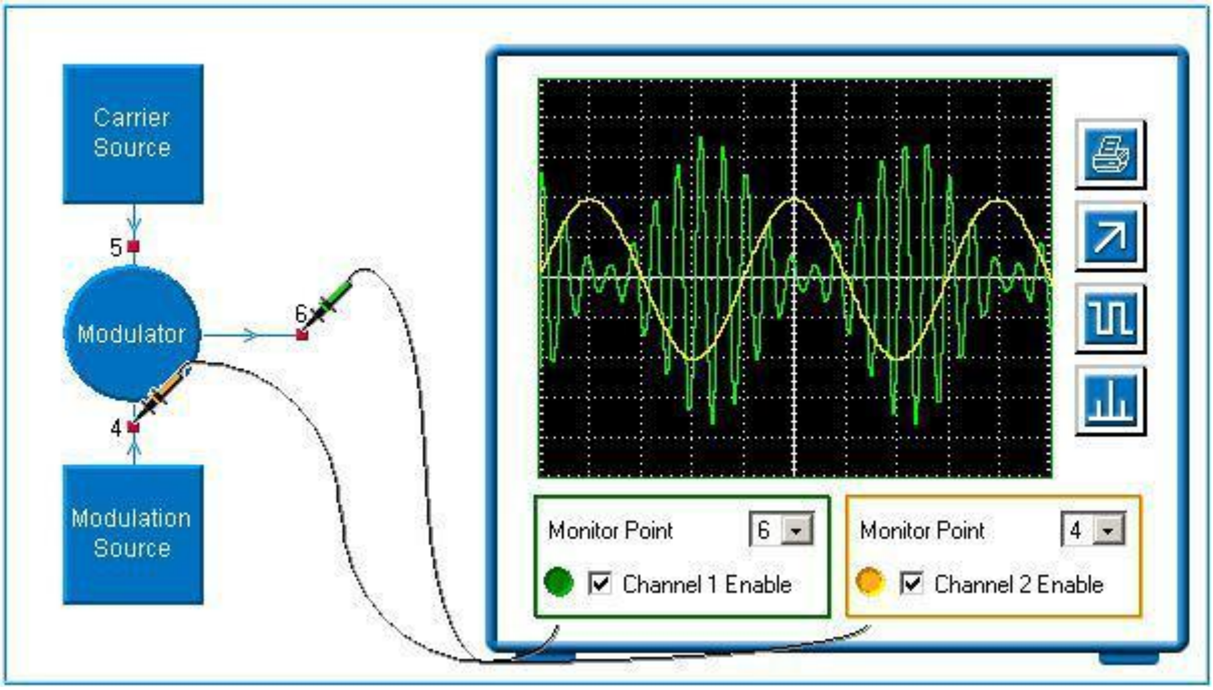

Figure 1.4 Experiment Setup

Set the ***carrier level*** to maximum. Set ***modulation level*** to zero. Observe the signals at all monitoring points. Write your observations.

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Now increase the ***modulation level*** and observe at monitor point 6

Increase the ***modulation level*** until the carrier amplitude just reaches zero on negative modulation peaks. This is 100% modulation. Observe the signals at all the monitoring points both with the oscilloscope and the spectrum analyzer at various modulation levels.

Also, with a fixed modulation level try adjusting the ***carrier level***.

RESULTS AND WAVEFORMS/SPECTRUM:

LAB SESSION 02

To check the operation of the balanced amplitude modulator with suppressed carrier

Student Name: ______________________________

Roll Number: ________________ **Batch:** ____________

Semester: ________________ **Year:** ____________

Total Marks	
Marks Obtained	

Remarks (If Any): ______________________________

Instructor Name: ______________________________

Instructor Signature: ________________ **Date:** ____________

LAB SESSION 02

DOUBLE SIDE BAND SUPPRESSED CARRIER

OBJECTIVES:

To check the operation of the balanced amplitude modulator with suppressed carrier

EQUIPMENT REQUIRED:

Amplitude Modulation Work board 53-130 which comprises the following blocks:

- Signal Generation
- Modulation
- Filters
- Demodulation

PRE-LAB THEORY:

Double sideband suppressed carrier modulation:

In AM modulation, two-third of the transmitted power appears in the carrier which itself conveys no information. The real information is contained within the sidebands. One way to overcome this problem is simply to suppress the carrier. Since the carrier does not provide any useful information, there is no reason why it has to be transmitted. By suppressing the carrier the resulting signal is simply the upper and lower sidebands. Such a signal is referred to as **a double-sideband suppressed carrier** (DSB-SC or DSB) signal. Double sideband suppressed carrier modulation is simply a special case of AM with no carrier. A circuit called **balanced modulator** generates double sideband suppressed carrier signals.

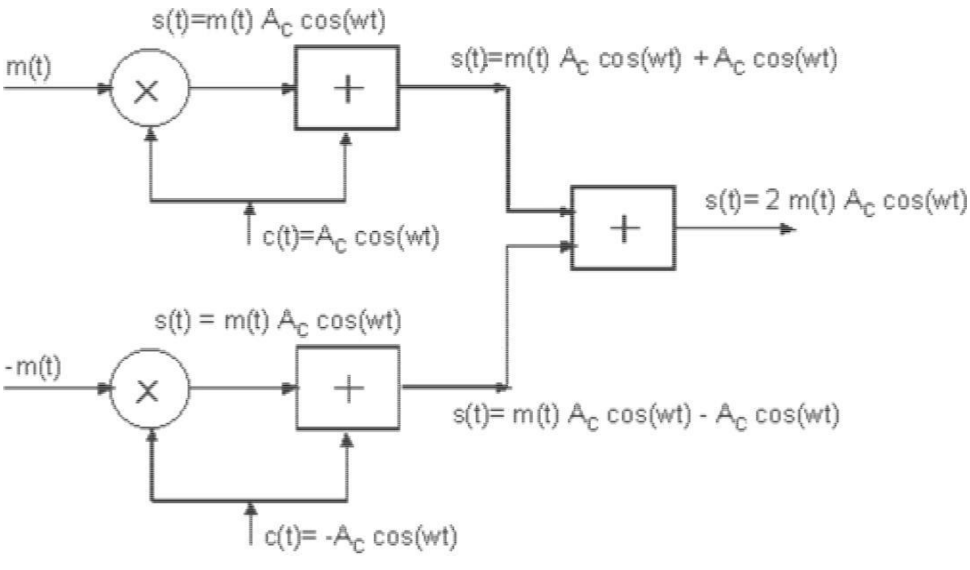

Figure 2.1 DSBSC Signal's Generation

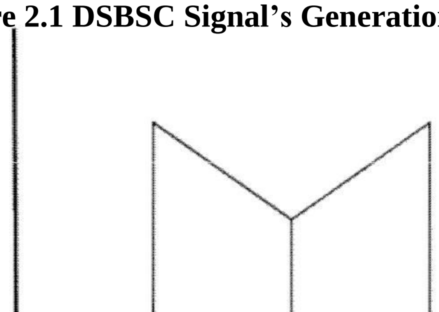

Figure 2.2 DSBSC Spectrum

In the theory for the Amplitude Modulation with Full Carrier assignment, it was established that the output signal of the AM Modulator circuit is:

$$v_C = V_C \sin \omega_c t + (mV_c/2)[\cos(\omega_c - \omega_m) t] - (mV_c/2)[\cos(\omega_c + \omega_m) t]$$

In DSB suppressed carrier modulation, the carrier term **Vc sin ωc t** is suppressed, leaving just:

$\boldsymbol{v_c = (mV_c/2)[\cos(\omega_c - \omega_m) t] - (mV_c/2)[\cos(\omega_c + \omega_m) t]}$ as the modulated signal.

The two cosine terms represent the lower and upper sidebands respectively

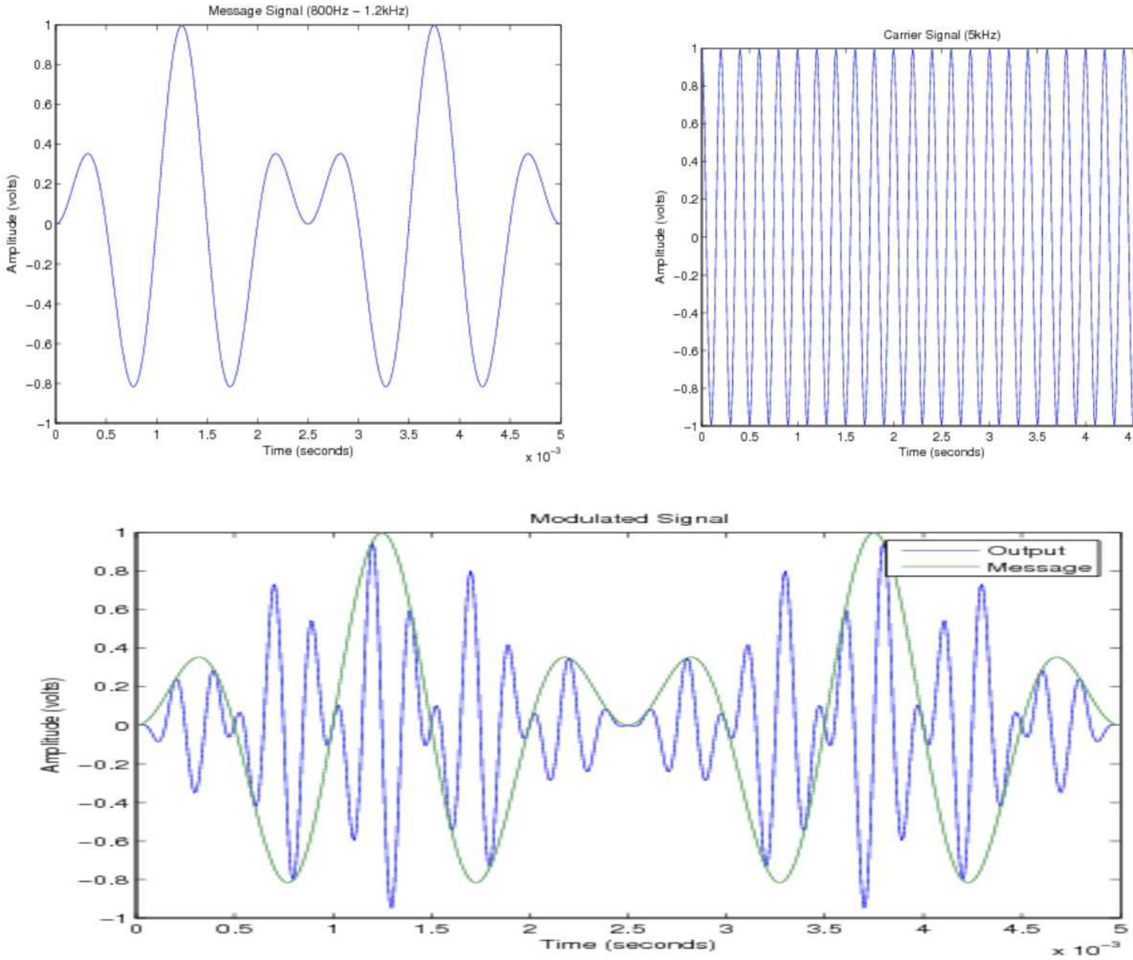

Figure 2.3 a. Message Signal b. Carrier Signal c. DSBSC modulator output

PROCEDURE:

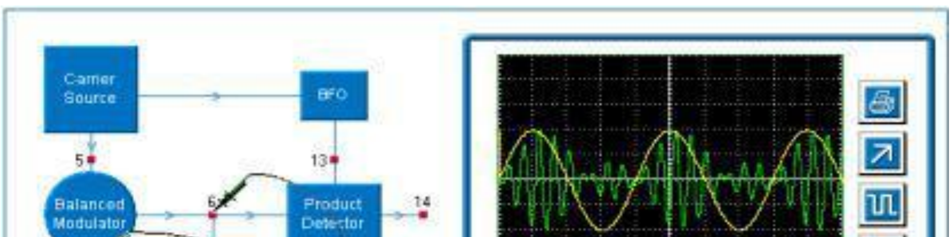

Figure 2.4 Experiment Setup

Use the oscilloscope and spectrum analyzer to examine the signals at monitor point **4** and monitor point **5**. Set the ***carrier balance*** to mid-scale. Note that they are the same as for simple AM. Now examine at monitor point **6** and observe the wave shape.

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Use the spectrum analyzer to observe that there are two sidebands but no carrier. Record your observations:

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Adjust the ***carrier balance*** and observe the effect on carrier amplitude

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Adjust ***modulation level*** and ***carrier level*** and observe the effects

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RESULTS AND WAVEFORMS/SPECTRUM:

LAB SESSION 03

To examine the main parameters of the single sideband modulation and to check the use of filters to generate the SSB

Student Name: ______________________________

Roll Number: ____________________ **Batch:** ____________

Semester: ____________________ **Year:** ____________

Total Marks	
Marks Obtained	

Remarks (If Any): ______________________________

Instructor Name: ______________________________

Instructor Signature: ____________________ **Date:** ____________

LAB SESSION 03

SINGLE-SIDE BAND SIGNAL

OBJECTIVES:

To examine the main parameters of the single sideband modulation and to check the use of filters to generate the SSB

EQUIPMENT REQUIRED:

Amplitude Modulation Work board 53-130 which comprises the following blocks:

- Signal Generation
- Modulation
- Filters
- Demodulation

PRE-LAB THEORY:

Single sideband modulation:

A modulation technique in which only one sideband out of the two is transmitted is known as **Single Side band transmission**. In double sideband transmission, the basic information is transmitted twice once in each sideband. Therefore, transmitting both signals is redundant. The information can be transmitted through one sideband by further suppressing the one sideband. The generated signal is termed as single sideband suppressed carrier.

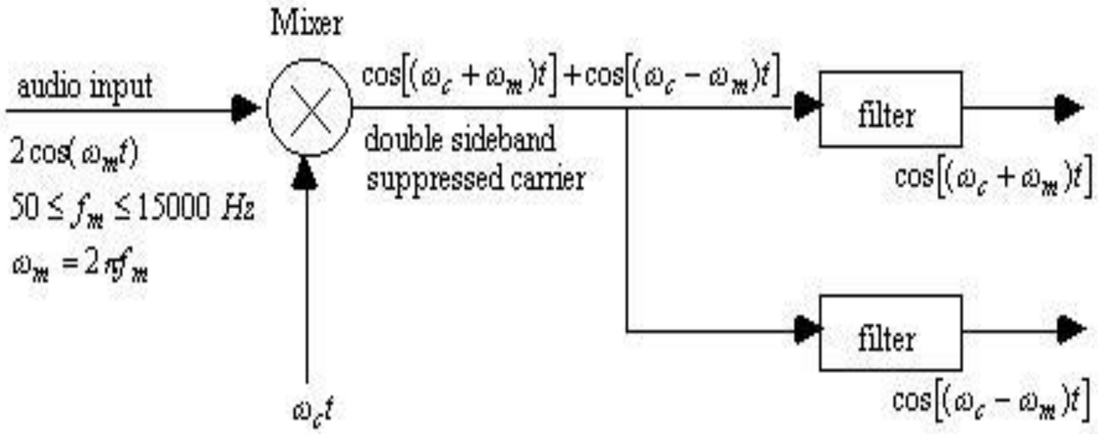

Figure 3.1 Generation of SSB signal

Generating SSB:

The generator in the practical is a balanced modulator, producing DSB, followed by a bandpass filter for the required sideband. There are other methods but this **filter method** is the simplest to understand and is in very common usage in communication systems. It may be necessary for the bandpass filter to have a very good shape factor because, at normal carrier and audio frequencies, the upper and lower sidebands are quite close in frequency.

Another consideration is that the sideband filter should offer significant attenuation to the carrier, so that the balanced modulator need not be so accurately balanced. In practice the balanced modulator might provide 30dB of carrier suppression and the filter a further 10dB. The other sideband would normally be about 30 to 40dB down on the wanted one. In order to achieve this, the SSB filter has several poles and is, in most cases a **ceramic filter** or **crystal filter**. Various filters are commercially available with different specifications depending on the application.

In the practical, a high modulating frequency is used, so one can see clearly the relationship between the various frequency components. This means that the filter specification can be relaxed and here a single tuned circuit is used. Separate filters are provided for upper and lower sidebands and the means is provided to monitor the output of both.

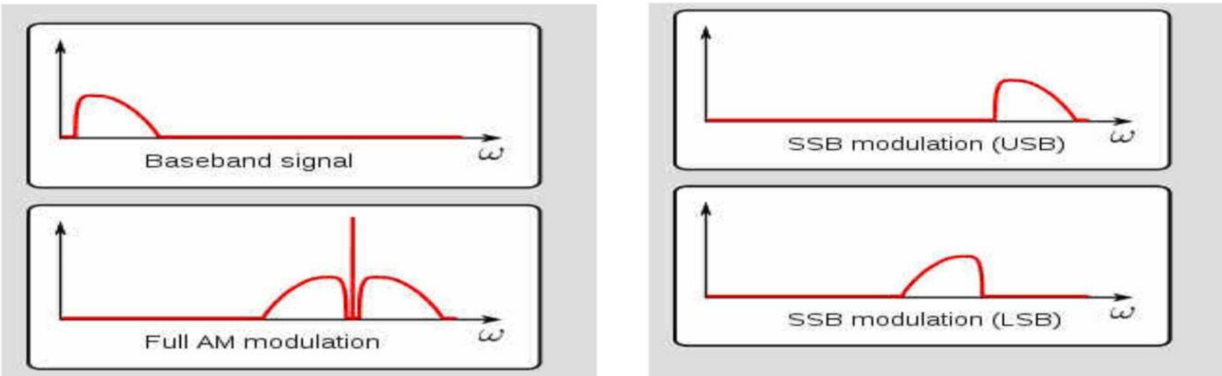

Figure 3.2 Frequency Spectrum Representations

Upper or Lower Sideband?

There is no reason why one sideband gives better results than the other, but general practice seems to favor the upper sideband. One convention is that with carrier frequencies below 10 MHz the lower sideband should be used, but this is not always the case. The result of this is that many pieces of communication equipment have to be able to deal with both.

<u>PROCEDURE:</u>

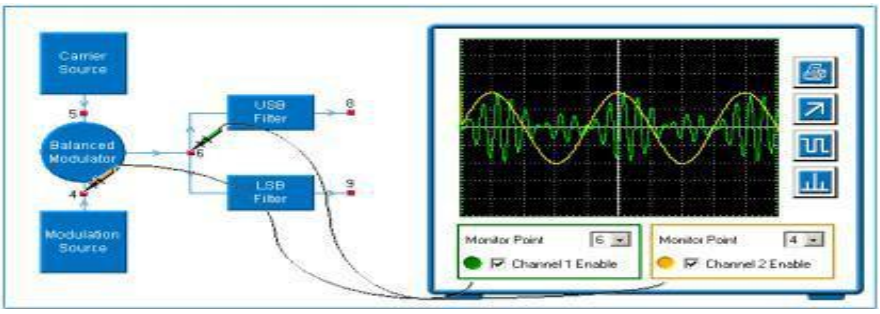

Figure 3.3 Experiment Setup

Use the spectrum analyzer and oscilloscope to observe at monitor point 6. Note that the signal is DSB. Adjust the ***carrier balance*** as before.

Use the spectrum analyzer and oscilloscope to observe at monitor point 8.

Use the spectrum analyzer and oscilloscope to observe at monitor point 9.

RESULTS AND WAVEFORMS:

LAB SESSION 04

To investigate the demodulation of an AM signal with an Envelope Detector

Student Name: ____________________________________

Roll Number: ______________________ **Batch:** ______________

Semester: ________________________ **Year:** ______________

Total Marks	
Marks Obtained	

Remarks (If Any): ________________________________

Instructor Name: ____________________________________

Instructor Signature: __________________ **Date:** __________

LAB SESSION 04

ENVELOPE DETECTOR

OBJECTIVE:

To investigate the demodulation of an AM signal with an Envelope Detector

EQUIPMENT REQUIRED:

Amplitude Modulation Work board 53-130 which comprises the following blocks:

- Signal Generation
- Modulation
- Filters
- Demodulation

PRE-LAB THEORY:

The purpose of any detector or demodulator is to recover the original modulating signal with the minimum of distortion and interference. The simplest way of dealing with an AM signal is to use a **simple half- wave rectifier circuit**. If the signal were simply passed through a diode to a resistive load, the output would be a series of half-cycle pulses at carrier frequency. So the diode is followed by a filter, typically a capacitor and resistor in parallel.

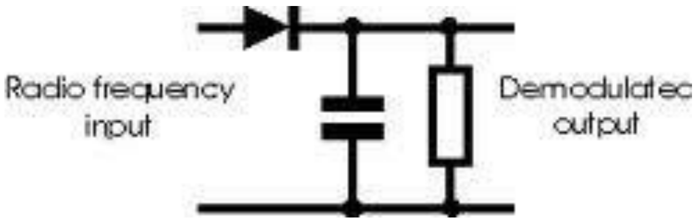

Figure 4.1 Envelope Detector Circuit

The capacitor is charged by the diode almost to the peak value of the carrier cycles and the output therefore follows the envelope of the amplitude modulation. Hence the term "**envelope detector**".

The **time constant** of the RC network is very important because if it is too short the output will contain a large component at carrier frequency. However, if it is too long it will filter out a significant amount of the required demodulated output.

In this practical the output of the AM generator that is fed to an envelope detector. The output can be monitored and compared with the original modulation source. The time constant of the filter following the detector can be adjusted. This filter is often called a **post-detection filter**. It also introduces a phase shift between the original signal and the output.

PROCEDURE:

Obtain an AM modulated signal from an AM modulator and apply it to the input of the envelope detector. Here the signal from the amplitude modulator from the AM Signal generator is demodulated using an envelope detector.

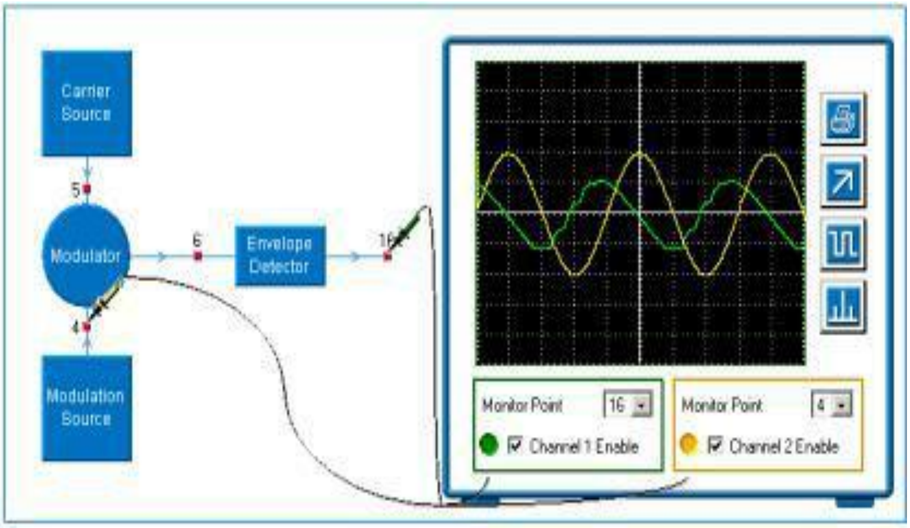

Figure 4.2 Experiment Setup

Use the oscilloscope to monitor the **detector output 16** and adjust the **time constant.** If it's too less and too large what will happen? Also state the reason.

Use the spectrum analyzer to observe the carrier spectral components. Record your observations:

Compare the original modulating signal with the detector output in both shape and phase at various time constants using the oscilloscope. Record your observations.

RESULTS AND WAVEFORMS/SPECTRUM:

LAB SESSION 05

To observe the operation of Product Detector

Student Name: ____________________________

Roll Number: __________________ **Batch:** ____________

Semester: __________________ **Year:** ____________

Total Marks	
Marks Obtained	

Remarks (If Any): ____________________________

Instructor Name: ____________________________

Instructor Signature: __________________ **Date:** ____________

LAB SESSION 05

PRODUCT DEMODULATOR

OBJECTIVE:

To observe the operation of Product Detector

EQUIPMENT REQUIRED:

Amplitude Modulation Work board 53-130 which comprises the following blocks:

- Signal Generation
- Modulator
- Filters
- Demodulator

PRE-LAB THEORY:

Product detector has certain advantages over the simple envelope detector but at the expense of some complexity. It is not often used for Amplitude Modulation but is the only type of detector that will demodulate the suppressed carrier amplitude. **It is important to appreciate that a product detector will demodulate all forms of AM.**

Product Detector:

If the AM signal is mixed with (i.e., modulated by) a frequency equal to that of its carrier, the two sidebands are mixed down to the original modulating frequency and the carrier appears as a dc level. The mathematics of the process shows that this will only happen if the mixing frequency is equal not only in frequency to that of the carrier, but also in phase; i.e., the two signals are **synchronous**. This is why a product detector when used for AM is sometimes called a **synchronous detector.** For AM, the effect is very similar to a full-wave rectifier rather than the half-wave of the envelope detector.

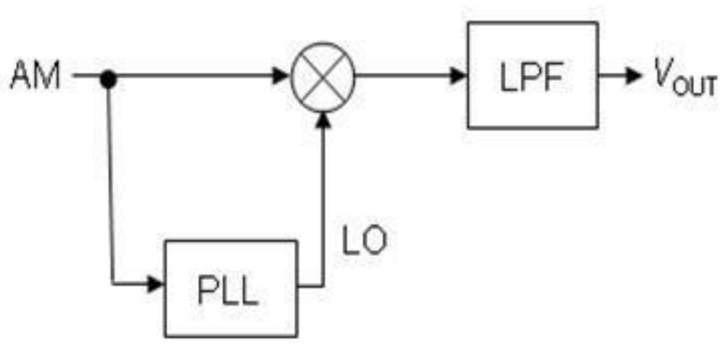

Figure 5.1 Block Diagram of Product Detector

The output still needs a post-detection filter to remove the residual ripple, but this time the ripple is at twice the carrier frequency and is therefore further away from the modulation and hence easier to remove. In general terms the product detector gives less distortion, partly because it uses both positive and negative peaks of the carrier.

Generating the Mixing Frequency:

This is produced by an oscillator which is usually referred to as a **Beat Frequency Oscillator** or BFO. This is because it is not at the same frequency as the carrier the output of the product detector. It works on a frequency equal to the difference between of the two, which is called a beat frequency. (You will be able to see this when you adjust the BFO for synchronism).

As previously described, it is vital that the BFO be synchronised to the carrier. In practice this is achieved with a special recovery circuit but here for simplicity a sample of the carrier is fed directly to the BFO and when the free running frequency of the BFO is near to that of the carrier it locks into synchronism.

PROCEDURE:

Follow required procedure to obtain product detector demodulating AM output.

The oscilloscope shows its input at **monitor point 6**, which is the output of the same modulator as before.

Monitor the BFO output with the oscilloscope and use the ***BFO frequency*** control to lock it to the carrier. This will be indicated by a stationary **TRACE.**

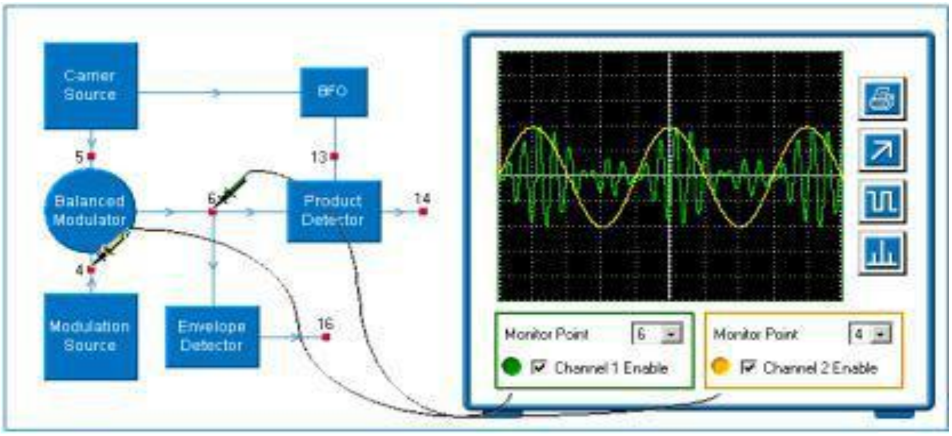

Figure 5.2 Experiment Setup

Use the oscilloscope to look at the output of the detector and compare it with original modulating signal.

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Use the spectrum analyzer to confirm this.

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Monitor the detector output with the oscilloscope, unlock the BFO with the ***BFO frequency*** control and observe the result. Repeat whilst using spectrum analyzer. Record your observations:

<u>RESULTS AND WAVEFORMS/SPECTRUM:</u>

LAB SESSION 06

To investigate the demodulation of SSB signals using Product/ Synchronous detection

Student Name: ____________________________________

Roll Number: ____________________ **Batch:** ______________

Semester: ______________________ **Year:** _______________

Total Marks	
Marks Obtained	

Remarks (If Any): ________________________________

Instructor Name: ________________________________

Instructor Signature: ____________________ **Date:** __________

LAB SESSION 06

DEMODULATION OF SINGLE-SIDE BAND SIGNAL

OBJECTIVE:

To investigate the demodulation of SSB signals using Product/ Synchronous detection

EQUIPMENT REQUIRED:

Amplitude Modulation Work board 53-130 which comprises the following blocks:

- Signal Generation
- Modulation
- Filters
- Demodulation

PRE-LAB THEORY:

Single sideband demodulation:

In the double sideband suppressed carrier practical, we saw how DSB is demodulated using the BFO to reinsert the carrier. In the case of DSB, the BFO must be in phase with the original carrier or the process will not work correctly. Since SSB is transmitted without a carrier it is not surprising that a similar method is employed.

The main difference is that, **for SSB, the BFO need not be in phase with the carrier**. It does need to be at the same frequency but even a small error in the frequency results only in a small error in the frequency of the demodulated output. This means that in non-critical applications, such as speech, a small overall frequency error does not make the system useless. The effect on speech is to raise or lower the tone of the voice, which within limits does not reduce intelligibility.

The fact that the BFO need not be locked, greatly simplifies the design of the receiver, and makes SSB one of the most powerful techniques for transmitting audio frequencies over radio links with its **narrow bandwidth and efficient use** of available transmitter power.

In the practical, you can use both upper and lower sidebands and see that with the BFO set correctly, near to the original carrier frequency, even though the two sidebands are at different frequencies the demodulated output is the same. You can also see that changing the BFO frequency causes the demodulated output to change in frequency by a similar amount.

PROCEDURE:

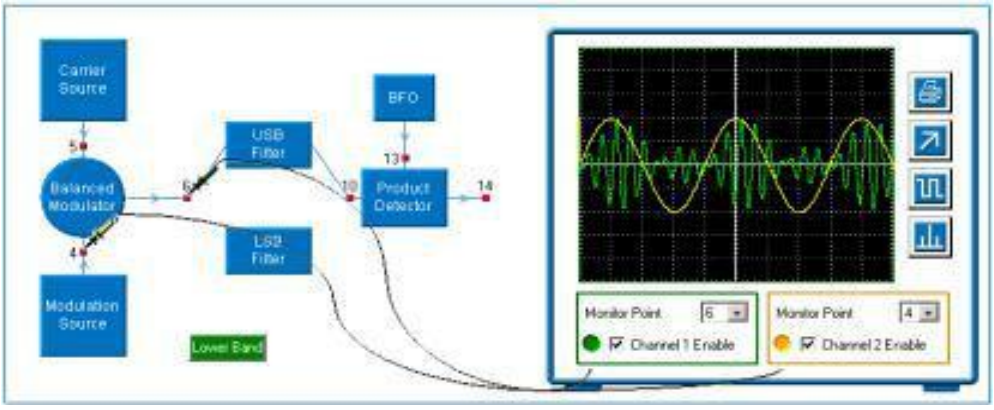


Figure 6.1 Experiment Setup

Monitor at monitor point **6**, and observe the DSB signal

Move to monitor point **10** and note the upper sideband signal

Change to lower sideband (by pressing the button) and note the lower sideband signal

Use either the oscilloscope or analyzer to set the ***BFO frequency*** to that of the carrier, by monitoring at monitor point **13**

Now monitor at point **14** and compare the output with the modulation input.

RESULTS AND WAVEFORMS/SPECTRUM:

LAB SESSION 07

To observe the characteristics of a Frequency Modulated wave in Time Domain and Frequency Domain.

Student Name: ________________________________

Roll Number: ____________________ **Batch:** ____________

Semester: ____________________ **Year:** ____________

Total Marks	
Marks Obtained	

Remarks (If Any): ______________________________

Instructor Name: ______________________________

Instructor Signature: ________________ **Date:** __________

LAB SESSION 07

FREQUENCY MODULATION

OBJECTIVES:

To observe the characteristics of a Frequency Modulated wave in Time Domain and Frequency Domain.

EQUIPMENT REQUIRED:

Feedback-Teknikit Console 92-300
Pentium 4 or equivalent computer available in lab

PRE-LAB THEORY:

Frequency Modulation Generation:
The circuits used to generate a frequency modulation must vary the frequency of a high frequency carrier signal as a function of the amplitude of low frequency signal (modulation signal). In practice there are two main methods used to generate FM:

- **Direct method:** A tank circuit is used in which the reactance of one of the elements of the resonant circuit depends on the modulating voltage. The most common device with variable reactance is the Varactor or Varicap. This is a particular diode and its capacitance varies according to the reverse bias voltage applied across it. The frequency of the carrier is established with Automatic Frequency Control (AFC) circuits or Phase Lock Loop (PLL).

- **Indirect method:** The FM is obtained in this case by a Phase modulation, after the modulating signal has been integrated. In the phase modulator the carrier can be generated by a quartz oscillator, and so its frequency stabilization is easier.

PROCEDURE:

Set *Carrier level* to about half scale (0.8 Vp-p). Monitor point **16** shows us the DC input voltage and monitor point **4** shows the output carrier which is frequency modulated. *Figure 7.1* shows the output signal when input voltage is 0 V.

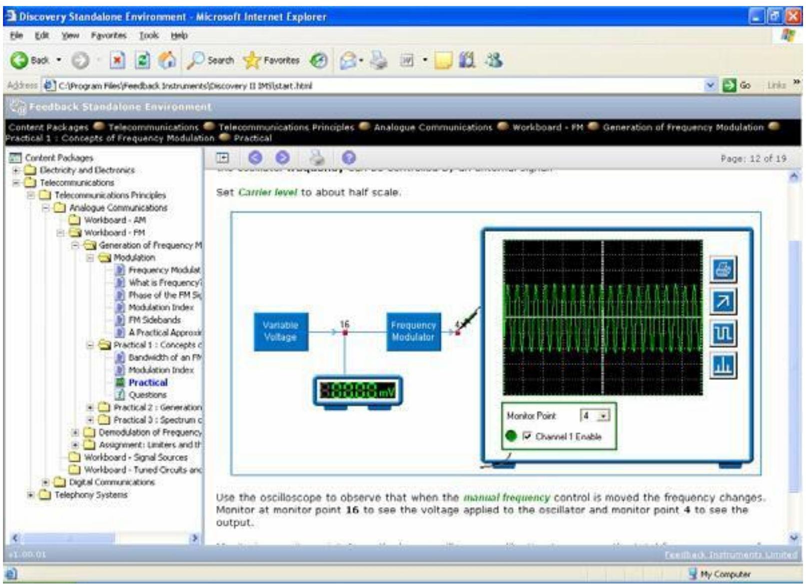


Figure 7.1 Experiment Setup

The Frequency corresponding to a zero input voltage is best observed by the spectrum analyzer as shown in *figure 7.2*. The left marker of the spectrum analyzer is utilized to measure the signal frequency.

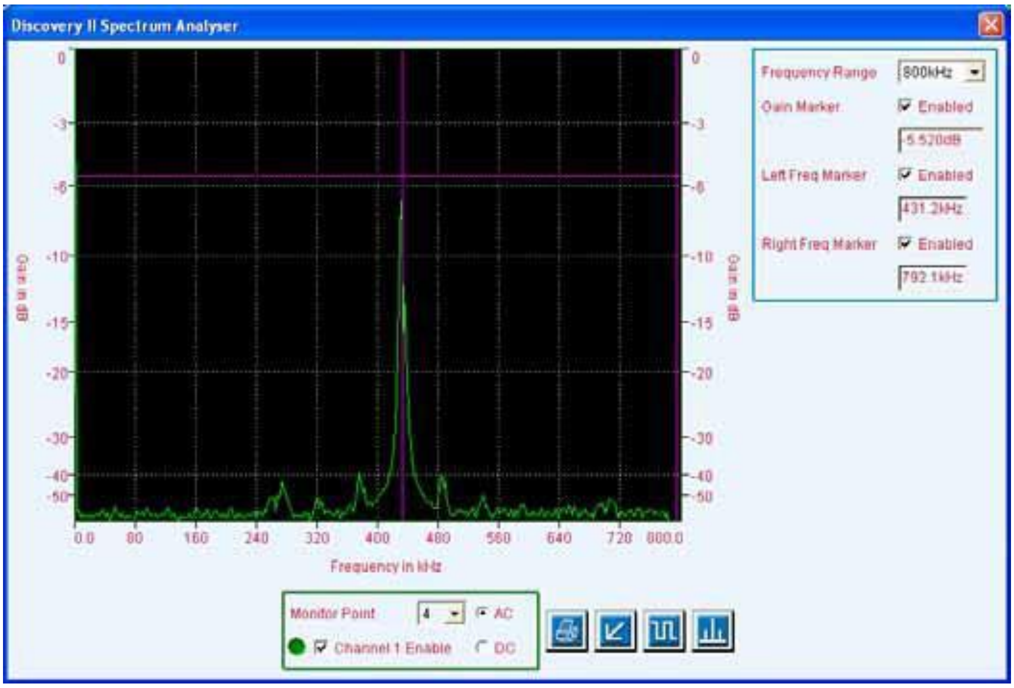


Figure 7.2: Spectrum analyzer output

A tedious way to measure the output signal frequency is by observing the signal in time domain. We take the inverse of the pulse time duration which is measured with the help of the left and right scope markers-*Figure 7.3*. Note, that the spectrum analyzer method is a bit more accurate.

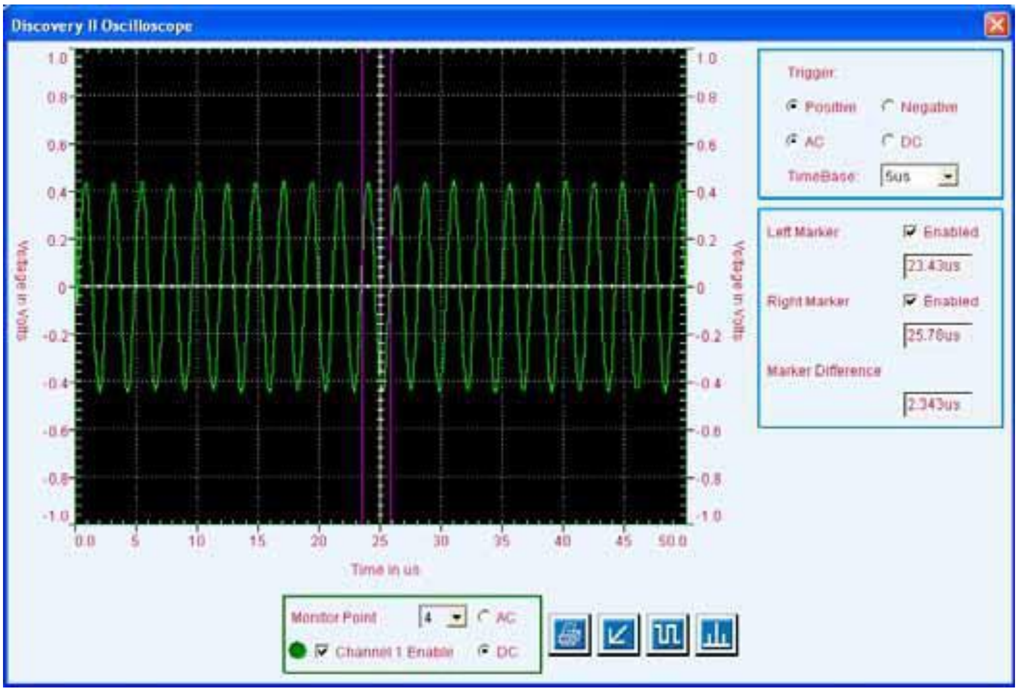


Figure 7.3: Oscilloscope output

OBSERVATIONS:

Set the carrier amplitude such that it is 2 divisions above and below the x-axis (approximately 0.8 Vp-p). Fill in the table below for DC input voltage vs. output carrier frequency. Plot a graph using the values you recorded in the table.

Input Voltage (V)	**Output Frequency (Oscilloscope) Hz**	**Output Frequency (Spectrum Analyzer) Hz**

RESULT:

LAB SESSION 08

To analyze the Spectrum of an FM Signal with a Large Modulation Index

Student Name: ____________________________________

Roll Number: ____________________ **Batch:** ______________

Semester: ______________________ **Year:** ______________

Total Marks	
Marks Obtained	

Remarks (If Any): ________________________________

Instructor Name: ____________________________________

Instructor Signature: __________________ **Date:** __________

LAB SESSION 08

EFFECT OF MODULATION INDEX

OBJECTIVE:

To analyze the Spectrum of an FM Signal with a Large Modulation Index

EQUIPMENT REQUIRED:

Frequency Modulation 53-140 module
Oscilloscope

PRE-LAB THEORY:

This is a simple practical where the **frequency modulator** is connected to the **spectrum analyzer**. The carrier frequency has been reduced to about 5 kHz so, since the maximum deviation is the same; the modulation index is much greater.

The **bandwidth** is:

$$\mathbf{B = 2\ (F_d + F_m)}$$

Where *B* is the bandwidth, F_d the deviation and F_m is the bandwidth of the modulation.

So if F_m is small compared with F_d, i.e. the **modulation index** is large, then

$$\mathbf{B = 2\ F_d}$$

On the analyzer the **spectrum** appears to be continuous but in reality it is made up of a large number of sidebands spaced at 5 kHz intervals from the carrier up to F_d.

This practical simply shows how when the modulation index is large the bandwidth is determined almost exclusively by the deviation.

PROCEDURE:

1. In this practical the modulation frequency has been set to 5 kHz. This means that the modulation index can be very high.
2. This enables you to see that under these conditions the bandwidth of an FM signal is almost equal to twice the deviation.
3. Set ***Carrier level*** to about half scale.
4. Turn the ***5 kHz level*** up and down and observe the bandwidth changing. Note that the bandwidth is almost proportional to the deviation.

OBSERVATIONS:

RESULT:

LAB SESSION 09

To carry out FM demodulation using PLL detector

Student Name: ____________________

Roll Number: ____________________ **Batch:** ____________

Semester: ____________________ **Year:** ____________

Total Marks	
Marks Obtained	

Remarks (If Any): ____________________

Instructor Name: ____________________

Instructor Signature: ____________________ **Date:** ____________

LAB SESSION 09

FREQUENCY DEMODULATION

OBJECTIVE:

To carry out FM demodulation using PLL detector

EQUIPMENT REQUIRED:

Frequency modulation work board 53-140 which comprises the following blocks

- Signal generation
- Modulator
- Limiter
- Quadrative demodulator
- VCO
- Phase comparator

PRE-LAB THEORY:

This practical introduces the **phase locked loop (or PLL) demodulator.** This type of detector offers some advantages over the quadrature detector when the signal to noise ratio is poor. Before trying to understand how a PLL can demodulate an FM signal it is necessary to understand what a PLL is. The concept is of an oscillator synchronized in phase to an external signal source using a feedback loop. As frequency is the same thing as rate of change of phase, once the phase of the local oscillator is synchronized to the external signal, the frequencies are automatically made identical

A phase locked loop consists of three main blocks:

1. An **oscillator,** the frequency of which is controlled by an external voltage or current source. A voltage controlled oscillator or VCO is used in this assignment.
2. A **phase detector,** which compares the phase of the oscillator with that of the external signal.
3. A **filter,** which smoothes the output from the detector to provide the control signal to the VCO, adjusting its frequency so as to reduce the phase difference.

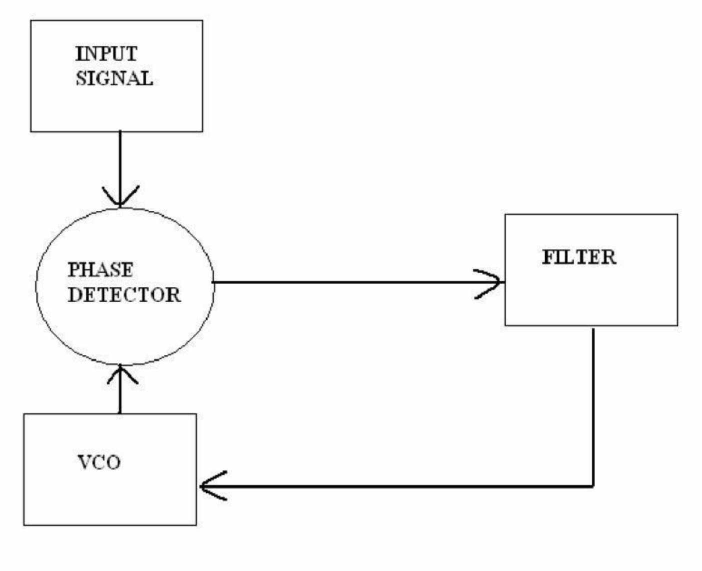
Operation of a PLL:
Imagine an incoming signal at a constant frequency within the range of the VCO.
Its phase is compared with that of the VCO and a voltage produced that alters the VCO frequency. The phase of the VCO therefore starts changing relative to the incoming signal, until eventually the phases match. Once they are equal, the control signal goes to zero and the system settles into equilibrium. Any drift of the VCO will be corrected by the control voltage which again appears. The two signals are said to be **phase locked.**

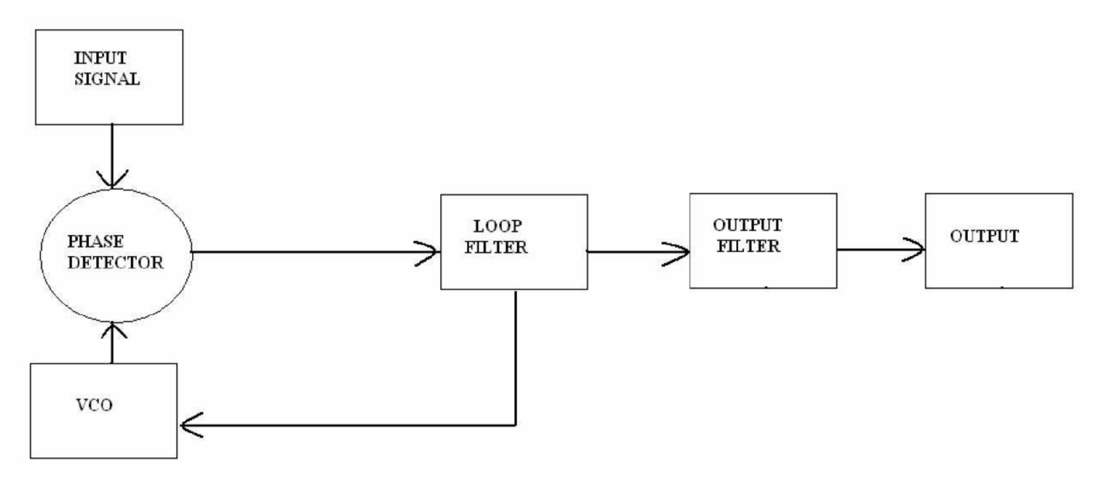
A filter is used in the control loop to keep the system stable and limit the maximum rate of change of oscillator frequency.
This whole description is a very simplified view, and the parameters that set the filter characteristics are very complex. An important factor in the design is the time before the two signals become locked.
Phase locked loops are used extensively in communications systems where it is necessary to produce a reference oscillator in phase with an incoming signal, also in special signal sources called frequency synthesizers

The PLL as an FM Detector:

Suppose that there was a PLL locked onto an incoming carrier which was unmodulated. The VCO would be at the same frequency as the carrier and the VCO control voltage would be constant. If the carrier were to change in frequency the VCO would follow the change by means of a change in control voltage. So the VCO control voltage varies with the carrier frequency, and if the carrier were frequency modulated the modulation would appear superimposed on the VCO control voltage.

When a post-detection filter is added to the simple PLL to remove all the frequency components above the maximum modulating frequency we now have a PLL FM detector.

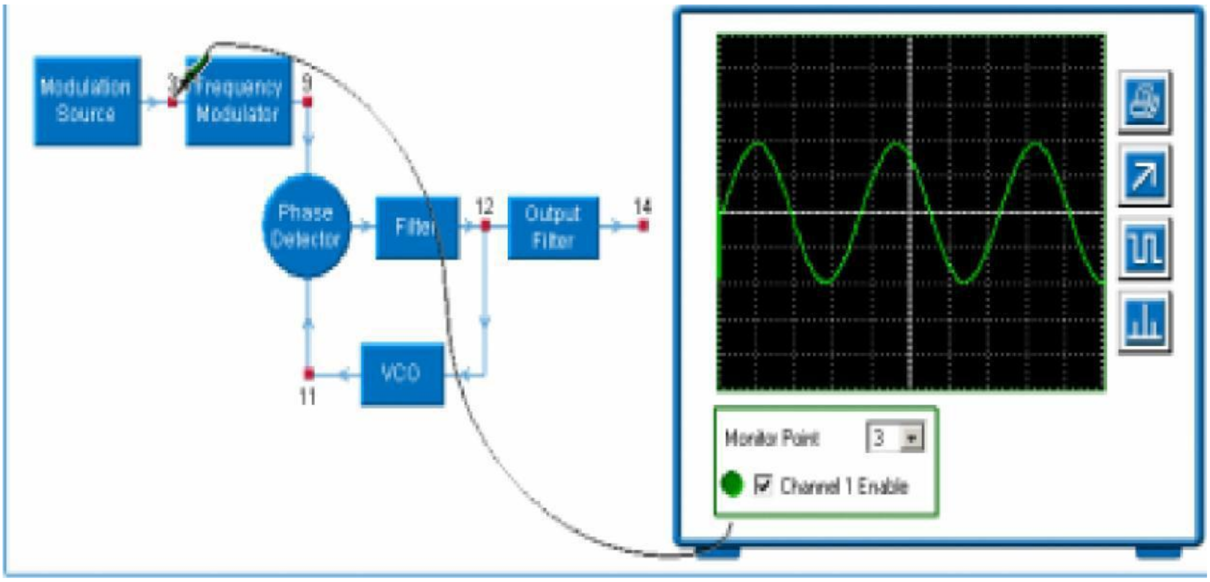




PROCEDURE:

In this practical we will see a PLL detector demodulating the same FM signal. The PLL is used when the ability to demodulate in the presence of noise is important. The distortion produced by this type of detector is determined mainly by the linearity of the VCO but this is often less important in noisy applications.

This practical shows a phase lock loop detector working. Monitor at **9** and observe the FM signal at different settings of ***modulation level.*** Examine the two signals at the input of the phase detector at **9** and the tracking VCO at *11*. Set ***carrier level*** to maximum. Observe the signal at the phase detector output **12** and then after the post detection filter at **14**



OBSERVATIONS:

Fc= fm= Vc= Vm=

Waveforms and Frequency Spectrum at:

Observation Point	Waveform	Frequency Spectrum
Point 9		
Point 11		
Point 12		
Point 14		

RESULT:

LAB SESSION 10

To study the effects of a limiter and noise on the performance of PLL detector

Student Name: ____________________

Roll Number: ____________ **Batch:** ________

Semester: ____________ **Year:** ________

Total Marks	
Marks Obtained	

Remarks (If Any): ____________________

Instructor Name: ____________________

Instructor Signature: ____________ **Date:** ________

LAB SESSION 10

EFFECT OF LIMITER AND NOISE

OBJECTIVE:

To study the effects of a limiter and noise on the performance of PLL detector

EQUIPMENT REQUIRED:

Frequency modulation work board 53-140 which comprises the following blocks

- Signal generation
- Modulator
- Limiter
- Quadrative demodulator
- VCO
- Phase comparator

PRE-LAB THEORY:

Effect of Limiter:
If the received signal is large enough, the PLL will lock the local oscillator phase to that of the received signal. Doubling the signal amplitude will not alter this situation, so will not affect the output signal. To this extent the PLL removes unwanted amplitude modulation of the received signal.

If the signal is small and the deviation is large, the phase detector cannot give enough output to move the VCO over a large enough range to track the deviation. This can be shown in the practical by reducing the carrier amplitude with no limiter in operation. Failure to track over the whole range of deviation shows as a distortion of the output signal. For small enough signals, the PLL fails to lock altogether.

The addition of a limiter means that the phase detector in the PLL has a constant amplitude signal to deal with. The gain of the phase lock control loop is therefore maintained as the signal level changes.

Effect of Noise:
Noise on the received signal causes both amplitude and phase changes. When a limiter is placed in the circuit, the amplitude changes are removed from the PLL input.

The principal effect on the PLL is that as the input signal tends to zero amplitude, there remains an adequate amplitude of signal to drive the phase lock loop. This continues to track the phase of the noisy received signal effectively, and with minimum error caused by noise amplitude variations.

Of course, the limiter cannot produce a signal from nothing, so as the carrier amplitude into the limiter falls, the noise from the limiter increases. This noise is faithfully detected by the PLL and degrades the output signal. This problem is due to fundamental noise problems and not due to any failing of the detector itself.

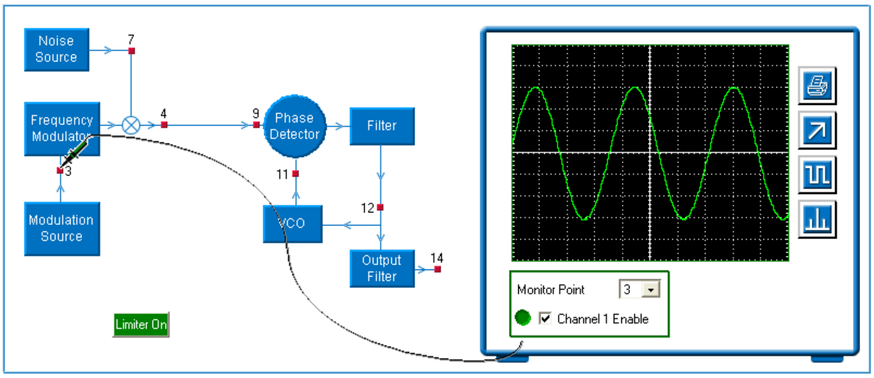


Figure 10.1 Experiment Setup

PROCEDURE:

Effect of Limiter:
Start with the limiter out (this is the default). Set the ***Carrier level*** and ***Modulation level*** to maximum. Set the ***Noise level*** to minimum. Observe at ***14*** that the demodulator is working correctly.

Reduce the ***carrier level*** and observe the output. Note that the detector loses track of the signal below a certain level, causing distortion of the detector output.

Note also that, when the modulation level is reduced, the carrier can be reduced further without distortion.

Now use the ***Limiter Button*** to switch in the limiter and repeat the tests. Note that the detector continues to work at much lower carrier levels. Use the other monitoring points to see how the system is operating.

Effect of Noise:
Set the ***Carrier level*** and ***Modulation level*** to maximum. The limiter should not be in use. Increase the ***Noise level*** and observe that the output becomes noisy.

Decrease the signal/noise ratio (SNR) further by reducing the ***carrier level*** until the signal becomes unrecognizable.

Now switch in the limiter using the ***Limiter Button***. Note that the detector keeps working at lower SNR when the limiter is in use.

RESULT:

LAB SESSION 11

To examine the functioning of natural and flat sampling PAM modulator

Student Name: ______________________

Roll Number: ______________ **Batch:** __________

Semester: ______________ **Year:** __________

Total Marks	
Marks Obtained	

Remarks (If Any): ______________________

Instructor Name: ______________________

Instructor Signature: ______________ **Date:** __________

LAB SESSION 11

PULSE AMPLITUDE MODULATION

OBJECTIVES:

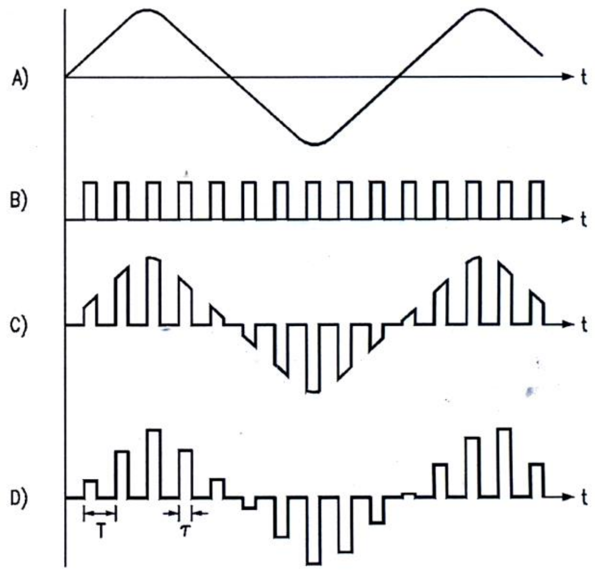
To examine the functioning of natural and flat sampling PAM modulator

EQUIPMENT REQUIRED:

Module T20A
Power supply
Oscilloscope

PRE-LAB THEORY:

PAM:
A PAM signal is a sampling signal made up by a series of pulses whose amplitude is proportional to analog signal amplitude. Sampling can be of normal & flat type. Flat sampling results in distortion of reconstructed signal as τ pulse duration increases. This sampling is used in PCM system.



a) **Analog Signal**
b) **Sampling Signal**
c) **Natural-sampling PAM Signal**
d) **Flat-sampling PAM Signal**

Figure 11.1
Natural & Flat Sampling PAM Signal

PAM modulator:
In *Natural sampling* block diagram mounted in model have an input analog signal that passes through a 3.5 kHz low pass filter which eliminates aliasing effect when sampling frequency is 8 or 12 kHz, then the signal goes to sampler. Sampling frequency in timing section can be selected at 4, 8, 12 kHz. Sampling pulse width is determined by pulse generator section.

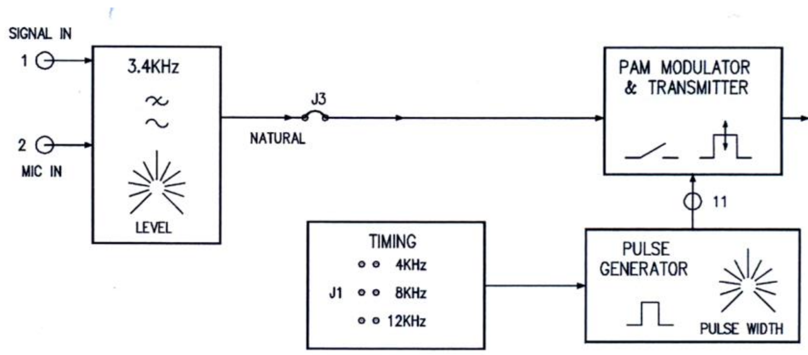


Figure 11.2 Natural-sampling PAM Modulator

In Flat sampling as compared with natural sampling modulator, a sample & hold circuit is added which fixes the output signal amplitude to keep it steady on the input value recorded in sampling. Sampler produces flat peak pulses whose width is proportional to analog signal width.

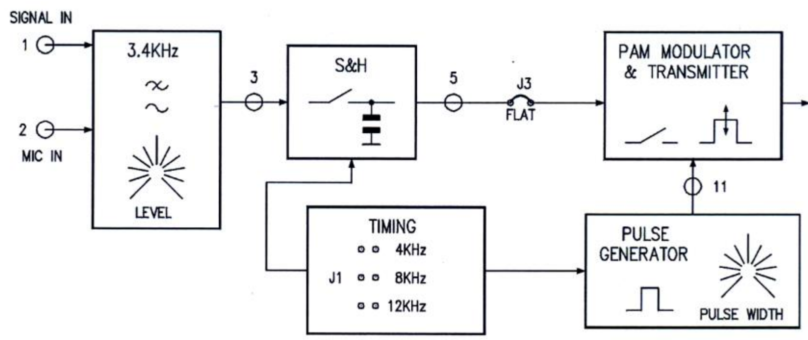


Figure 11.3 Flat-sampling PAM Modulator

a) Analog Signal
b) Sampling Signal
c) Sample & Hold Output
d) Flat-sampling PAM Signal

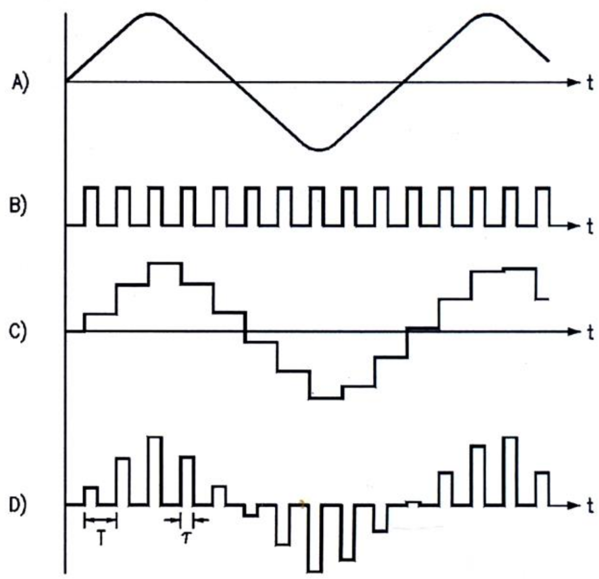


Figure 11.4 Flat-sampling PAM Signal

PROCEDURE & OBSERVATIONS:

Natural sampling

1. Supply ±12V power & carryout following pre-settings:
   Timing: J1=8 kHz, j3= natural sampling, pulse generator: completely turn pulse width clockwise
2. Connect TP13 to TP3 and check input analog signal of 1kHz at TP13.

3. Check that PAM signal at TP12 is formed by train of pulses having amplitude that reflects analog signal waveform.

4. Connect oscilloscope to the output of pulse generator at TP11 and check that sampling pulses tally with PAM signal at TP12.

5. Change sampling pulse width & observe corresponding variation of PAM signal and write the result.

Flat sampling

1. Now short J3 = flat sampling w/o changing previous settings.
2. Examine again waveform of input signal TP13, sample & hold output signal TP5 & of S/H sampling pulses TP4

3. Notice that signal is sampled at the beginning of sampling pulse & its amplitude is kept steady until next pulse. A step signal is obtained which approximates input analog signal.
4. Examine waveform at output of pulse generator at TP11 & output PAM signal TP12.
5. Notice that PAM pulse show constant amplitude over their whole duration.

6. Change sampling pulse width & observe corresponding variation of PAM signal

____
____
____
______

RESULT:

LAB SESSION 12

To examine the variations of reconstructed analog signal by changing (a) filter selectivity (b) sampling frequency

Student Name: ______________________________

Roll Number: ____________________ **Batch:** ______________

Semester: ____________________ **Year:** ______________

Total Marks	
Marks Obtained	

Remarks (If Any): ______________________________

Instructor Name: ______________________________

Instructor Signature: ____________________ **Date:** ____________

LAB SESSION 12

EFFECT OF FILTER SELECTIVITY & SAMPLING FREQUENCY

OBJECTIVE:

To examine the variations of reconstructed analog signal by changing (a) filter selectivity (b) sampling frequency

EQUIPMENT REQUIRED:

Module T20A
Power supply
Oscilloscope

PRE-LAB THEORY:

Analog signal reconstruction from samples is performed with a LPF. When sampling frequency equals 2B, an ideal LPF with a F/2 pass band can perfectly extract the same spectrum as original signal. If the filter is not an ideal one, there will be a section of spectrum S(f) centered around f which is super imposed on the section of spectrum to be extracted by filter which alters reconstruction of s(t). If sampling frequency is increased filtering becomes easier as repetitions of s(t) signal spectrum are spaced out. If sampling frequency is decreased, aliasing may occur.

Sampling a signal with a lower frequency than theoretical value or using a filter with an insufficient band to reconstruct original signal causes “aliasing effect.

PROCEDURE & OBSERVATIONS:

Effect of filter selectivity

1. Generate a flat sampling PAM signal by connecting TP 13 to TP3, jumper J3=flat, J1=8kHz.

2. Remove J8 jumper (if connected) & connect modulator output TP12 with 3.4kHz LPF input TP 24

3. At TP26, examine the waveform of reconstructed circuit. Check that this signal shows slight distortion due to faulty suppression of sampling frequency (8kHz)

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   ____
   ____
   ____

4. Cascadely connect 5 kHz & 3.4 kHz filter TP26 to TP 25 & check reconstructed output signal at TP27, this will increase overall filter selectivity. Check that distortion nearly disappears.

   ____
   ____

____
____

Effect of sampling frequency

1. Now maintain previous setting but select J1=12 kHz. At TP26 when 3.4 kHz filter is only selected examine the waveform of reconstructed signal. Check that signal show far low distortion in comparison with 8 kHz sampling.

   ____
   ____
   ____
   ____

2. Now select J1=4kHz & analyze the signal explaining the reason why it is considerably distorted.

   ____
   ____
   ____
   ____

RESULT:

LAB SESSION 13

To examine the working of PAM receiver

Student Name: ______________________________

Roll Number: __________________ **Batch:** ____________

Semester: __________________ **Year:** ____________

Total Marks	
Marks Obtained	

Remarks (If Any): ______________________________

Instructor Name: ______________________________

Instructor Signature: __________________ **Date:** ____________

LAB SESSION 13

PAM RECEIVER

OBJECTIVES:

To examine the working of PAM receiver

EQUIPMENT REQUIRED:

Module T20A
Power supply
Oscilloscope

PRE-LAB THEORY:

In order to demodulate PAM signal, a LPF is enough. This solution does not guarantee good connection quality & cannot be used in PAM- TDM. Therefore PAM receiver is constructed according to: PAM pulses coming from transmitter are sampled by sampling signal which is regenerated in receiver itself. Sampler output is kept at steady level until sample arrives, thereby generating a step signal. The signal reconstructed from step signal has wider amplitude than signal reconstructed directly from PAM pulses.

In Receiver**,** PAM signal coming from transmitter is amplified & applied to 2 sections: sampling pulse regenerator & demodulator (S/H). The demodulator output signal is filtered through LPF which produces demodulated analog signal.

Regeneration of sampling pulses for demodulator is carried out as follows: Amplified PAM signal passes through a limiting circuit which reduces signal amplitude variations. The next BPF (adjusted at 8 or 12 kHz) separates sampling frequency component. Such component gets to PLL which generates a synchronous sampling signal with PAM pulses it receives. The next circuit adjusts the phase of pulses coming from PLL.

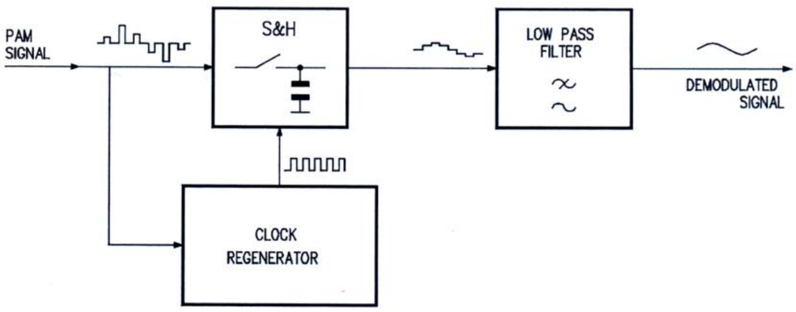


Figure 13.1 PAM Receiver

PROCEDURE & OBSERVATIONS:

1. Generate a flat sampling PAM signal presetting transmitter by connecting TP 13 to TP3 (1kHz input signal), jumper J3=flat, J1=8kHz.
2. Preset receiver J6=8kHz, J7=8kHz, J8= PAM
3. Connect transmitter output TP 12 with line input TP15 & line output TP16 to receiver input TP17. Bring line attenuation and noise to minimum & remove jumper (if connected) which selects line band pass.

4. Examine waveform at amplifier input & output (TP17 & 18 respectively). Output pulses have larger amplitude.

5. Examine signal after limiter TP19 & note considerable decrease of pulse amplitude variation.

6. At filter output TP20, an almost sinusoidal waveform is obtained having same frequency as PAM pulses at receiver input.

7. At PLL output TP21, if PLL locked a square waveform is obtained having same frequency as PAM pulses at receiver input.

8. Examine waveform at demodulator output (TP 24). Rotate phase adjust in order to obtain maximum step signal amplitude at demodulator output.

9. Examine signal waveform at reception filter output (TP26) & check if it is same as transmitted analog signal

RESULT:

LAB SESSION 14

To examine the operations of PWM and PPM modulators operation

Student Name: ______________________________

Roll Number: ________________ **Batch:** ____________

Semester: ________________ **Year:** ____________

Total Marks	
Marks Obtained	

Remarks (If Any): ______________________________

Instructor Name: ______________________________

Instructor Signature: ________________ **Date:** ____________

LAB SESSION 14

PULSE TIME MODULATION

OBJECTIVE:

To examine the operations of PWM and PPM modulators operation

EQUIPMENT REQUIRED:

Module T20A
Power supply
Oscilloscope

PRE-LAB THEORY:

A Pulse carrier can be modulated w.r.t. its amplitude or its timing.
This second case is usually defined as Pulse Time Modulation (PTM): two instances of PTM are Pulse Width Modulation (PWM) and Pulse Position Modulation (PPM).

A ***PWM signal*** is a pulse signal whose **pulse width** is proportional to the modulating analog signal amplitude. The PWM signal is also used to generate the PPM signal. ***PPM*** is a pulse signal whose **pulse position** is proportional to the modulating analog signal amplitude. The PPM pulses are usually generated by the descending front of the PWM pulses.

- **a) Analog Signal**
- **b) Sampling Signal**
- **c) PAM Signal**
- **d) PWM Signal**
- **e) PPM Signal**

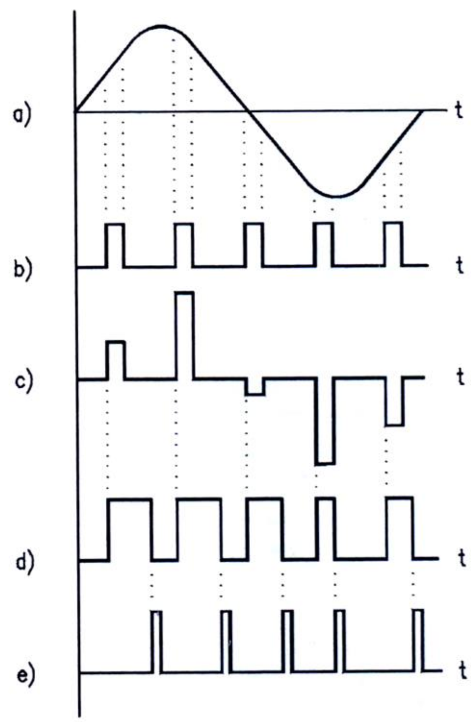
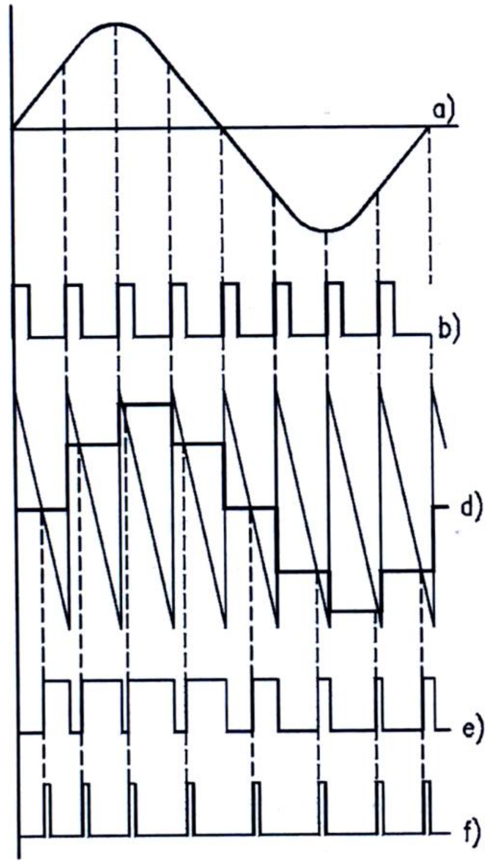


Figure 14.1 PAM/PWM/PPM Signals

PWM Modulator
The block diagram of the PWM modulator mounted on the module includes a stage comparator, which compares the respective amplitude of:

- a PAM signal obtained by sampling the input analog signal
- sawtooth generator (sampling-pulse-synchronous ramp signal).

The comparator switch the output when the PAM signal amplitude exceeds the ramp signal amplitude: this results into a PWM signal. From the modulator waveforms (figure.14.2) notice that the PWM pulse trailing edge corresponds to the sampling pulses, whereas the (variable) leading edge corresponds to the comparator switching.



- **a) Analog Signal**
- **b) Sampling Signal**
- **c) Ramp**
- **d) PAM Signal**
- **e) PWM Signal**
- **f) PPM Signal**

Figure 14.2 Modulator Waveforms

.

PPM Modulator

The PPM signal is obtained from the PWM signal, by generating fixed-duration pulse which corresponds to the leading edges of the PWM signal. This results into a train of pulses whose position depends on the input analog signal.

<u>PROCEDURE & OBSERVATIONS:</u>

<u>PWM</u>

1. Supply ±12V power & carryout following pre-settings:
   J1=8 kHz, J2= 8kHz, J4= PWM, Pulse width completely turned clockwise
2. Connect TP13 with TP1 and note the waveform at sampler output (TP5) and input signal TP1.
   ____
   ____
   ____
   ____
3. Observe that the sampled signal at TP5 is made up by a series of steps whose amplitude depends on the analog signal waveform.

4. At TP6 check that the SAWTOOTH GENERATOR supplies an approximate ramp of +3V to -3V for each sampling interval.

5. Compare PWM modulator output at TP8 with the PAM signal at TP5 and verify the following:
   • The trailing edge of the pulses corresponds to the sampling pulses
   • The leading edge (and the duration of the PWM pulses) varies according to the PAM signal amplitude and corresponds to the instant in which the PAM exceeds the ramp signal.

6. Vary the amplitude of the modulating analog signal and notice the corresponding variation of the PWM signal.

PPM

1. Supply ±12V power & carryout following pre-settings:
   J1=8 kHz, J2= 8kHz, J4= PPM, Pulse width completely turned clockwise
2. Re-examine the waveforms related to the PWM modulator (TP1,TP5,TP6,TP8)

3. Connect the oscilloscope with the PPM modulator output (TP9) and verify that the PPM signal (TP9) is made up by a train of generated pulses which correspond to the leading edges of the PWM pulses. Also notice that PPM pulses have a fixed duration and their position changes according to the modulating analog signal.

4. Change the amplitude of the modulating analog signal and notice the corresponding variation of the PPM signal.

RESULT: