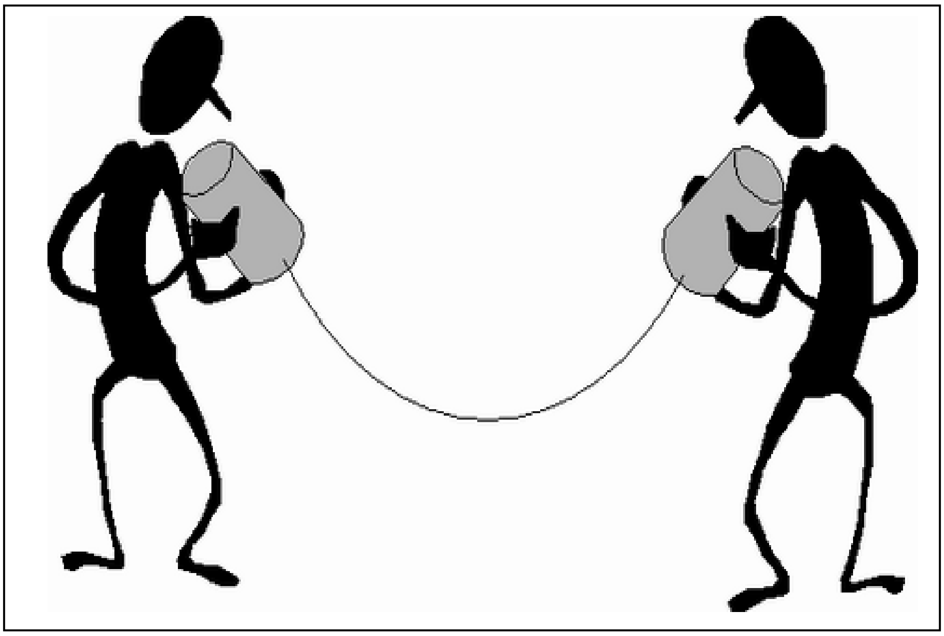
The Hashemite University

Faculty of Engineering

Electrical Engineering Department

Communications Lab Manual



Communications Lab Experiments:

Experiment 1: Spectral Analysis

Experiment 2: Introduction Amplitude Modulation

Experiment 3: Double Sideband (Transmitter and Receiver)

Experiment 4: FM Modulators/Demodulators

Experiment 5: Sampling and Reconstruction

Experiment 6: PCM Transmitter

Experiment 7: PCM Receiver

Experiment 8: ASK – FSK (Modulation & Demodulation)

General Lab Rules

- Be **PUNCTUAL** for your laboratory session.
- Foods, drinks and smoking are NOT allowed.
- The lab timetable must be strictly followed. Prior permission from the Lab Supervisor must be obtained if any change is to be made.
- Experiment must be completed within the given time.
- Respect the laboratory and its other users. Noise must be kept to a minimum.
- Workspace must be kept clean and tidy at all time. Points might be taken off on student/group who fails to follow this.
- Handle all apparatus with care.
- All students are liable for any damage to equipment due to their own negligence.
- At the end of your experiment make sure to switch off all the instruments.
- Students are strictly PROHIBITED from taking out any items from the laboratory without permission from the Lab Supervisor.
- Students are NOT allowed to work alone in the laboratory.
- Please consult the Lab Supervisor if you are not sure on how to operate the laboratory equipment.
- Report immediately to the Lab Supervisor if any injury occurred.
- Report immediately to the Lab Supervisor any damages to equipment, hazards, and potential hazards.
- Please refer to the Lab Supervisor should there be any concerns regarding the laboratory.

Grading Policy

The total mark for this lab is distributed as follows

Lab Report	20%
Quizzes	15%
Performance	5%
Mid-term Exam	20%
Final Exam	40%

Experiment One
Spectral Analysis

1. Objectives

- To demonstrate the frequency spectrum approach to signal analysis.
- To become familiar with the features and basic operation of the Spectrum Analyzer.

1.2 Basic information

1.2.1 Introduction

As signals and systems grow in complexity, their time domain representation becomes inadequate. The frequency domain representation forms a simple and straight - forward alternative. Sinusoidal functions occupy a unique position in engineering. They are easy to generate, and the steady state response of linear system to sinusoidal excitation can be found by using the impedance concept. Also, because periodic functions can be expanded into Fourier series of harmonic sinusoidal components, while transient non-periodic functions can be expressed as Fourier integrals. We can therefore discuss arbitrary waveforms in terms of their frequency spectra, and by superposition determine system response to an arbitrary excitations .in terms of its response to the various frequency components of the excitation.

If the given signal is periodic, the Fourier series expansion can be used to find the spectral content of the

signal:

$$g(t) = \sum_{\infty}^{-\infty} C_n e^{jnwot} \qquad (1.1)$$

Where $wo = 2\pi / To = 2\pi fo$, which is called the Fundamental angular frequency. The coefficients Cn

Are called the Fourier coefficients, and they are given by:

$$Cn = \frac{1}{To} \int_{To} g(t) e^{-jnwot} \qquad (1.2)$$

Where $n = 0, \pm 1, \pm 2, \ldots$

The representation of a periodic signal by its Fourier series expansion is equivalent to the resolution of the signal into its various harmonic components. The magnitude of the coefficients *Cn* give the power in each component.

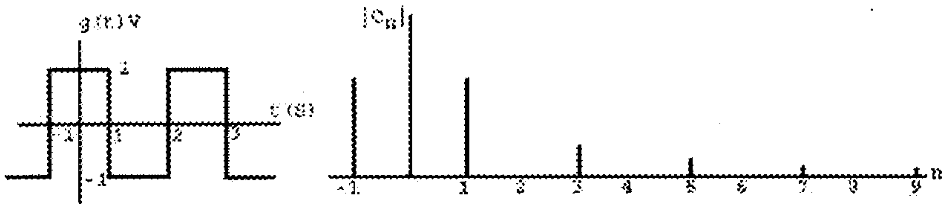


Fig.1: Square Wave and its Spectral Components

Example: The Fourier series expansion coefficients of the square wave shown in Fig. l are given by:

$$Cn = \frac{2}{n\pi}\sin\frac{n\pi}{2}$$

For n odd, $Cn = \frac{2}{n\pi}(-1)^{\frac{n-1}{2}}$

For n even, $Cn = 0$

1.2.2 Power and dB

Power is a common quantity measured in communication systems. In many measurement contexts, power is measured in Decibels dBW or dBm.

$$P_{dBW} = 10\log_{10} P_W$$

$$P_{dBm} = 10\log_{10}\frac{P_W}{1mW}$$

For example, if Pa = 5.012 mW, then:

$$Pa = 10\log_{10}(5.012) = 7dBm$$

Let Pb = 0.05012 mW then:

$$Pb = 10\log_{10}(5.012\times10^{-2}) = Pa - 20dB = -13dBm$$

Note that Pb is 20dB lower than Pa.

When applying RMS voltages to a 50 Ω.load, the above formula becomes:

$$dBm = 20\log_{10}\frac{Vrms}{0.224}$$

1.2.3 The Spectrum Analyzer

The traditional way of observing electrical signals is to view them in the time domain using an oscilloscope. The time domain is used to recover relative timing and phase information needed to characterize circuit behavior. However, not all circuits can be characterized from the time domain information. Circuit elements such as amplifiers, oscillators, mixers, modulators, detectors, and filters are best characterized by their response in the frequency domain. One instrument used to display the frequency domain is the spectrum analyzer. It graphically displays power or voltage as a function of frequency on a CRT. Like an oscilloscope, a spectrum analyzer produces a visible display on a CRT (some models uses a VGA screen rather that a CRT) unlike an oscilloscope, however, the. Spectrum analyzer has only one function-to produce a display of the frequency content of an input signal. Also like an oscilloscope, the spectrum analyzer will always produce a picture on the screen; but if you do not know how to properly use the spectrum analyzer, that picture may be complete gibberish.

CAUTION: *The input of the spectrum analyzer cannot tolerate large signals; before you connect a signal to the input, be sure you know that the signal will not exceed the maximum allowable input rating of the spectrum analyzer. (The maximum signal input is printed right on the front panel, near the input connector.)*

* *If you want to know signal Acquisition in a spectrum analyzer see appendix A*

1.2.3.1 Spectrum Analyzer Front Panel

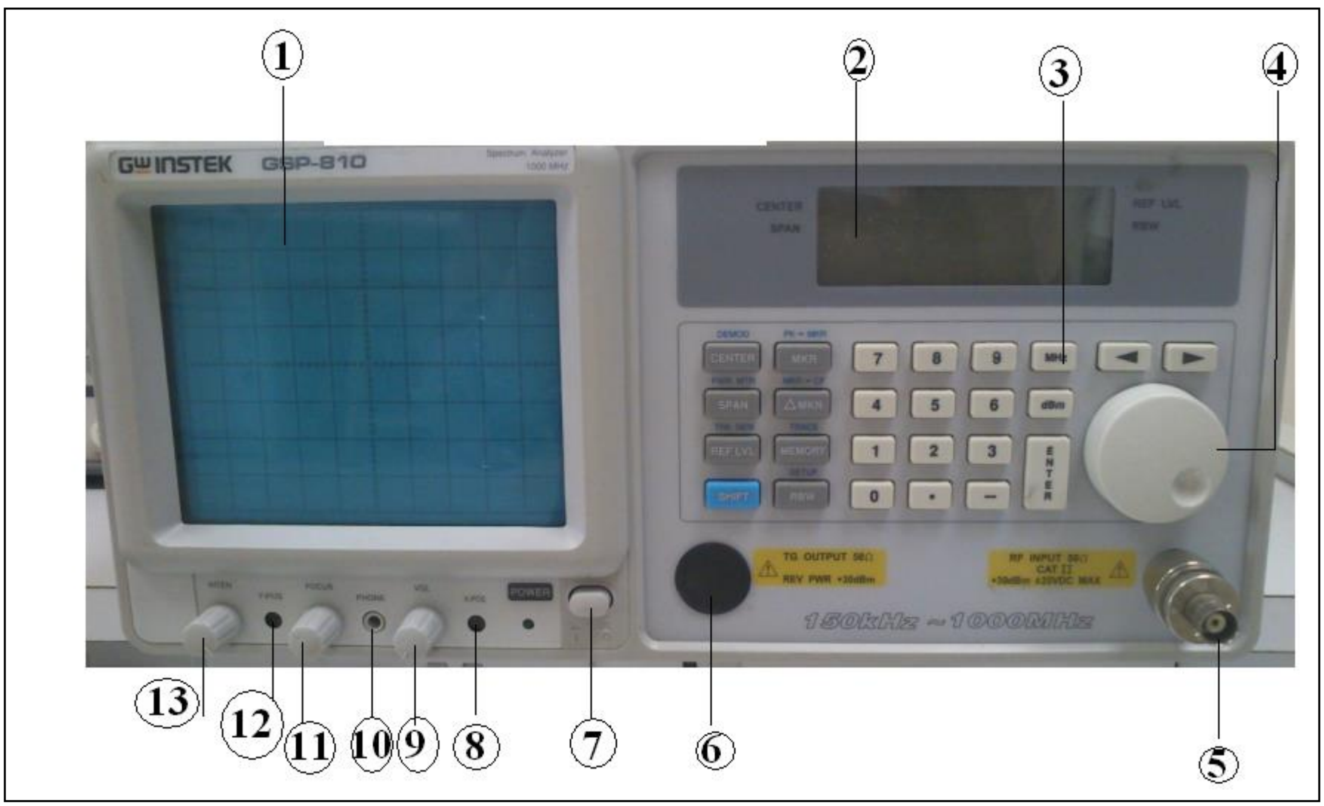


Fig.2: Spectrum Analyzer Front Panel

The spectrum analyzer front panel is shown in Figure-2, the description of the panel is:

Item	Description
1	Cathode Ray Tube (CRT) Display
2	Liquid Crystal Display (LCD)
3	Keypad, field selection and data entry
4	Spinner, field selection and data change
5	RF input
6	Tracking Generator output
7	Power Switch ON/OFF
8	RT Adjustment
9	Volume Control Knob 10 Phone Jack
11	CRT Focus Control Knob
12	CRT Y-axis Position Adjustments
13	CRT Intensity Control Knob

1.2.3.2 Spectrum Analyzer Controls

- The center frequency control, which slides the display in the horizontal axis. There by changing the display position relative to the full spectrum.

- The span control, which changes the horizontal scale allowing wider or narrower resolution of the display. The resolution of a spectrum analyzer is determined by its IF filter bandwidth.

- The marker control, used to find the exact amplitude and frequency of a certain point in the trace on the CRT.

- The bandwidth control, which selects one of several IF filters in the analyzer. Each filter has different width to allow higher resolution of adjacent signals. However, there is a limit to the narrowness of the filter bandwidth. This results in a pure sine wave to appear as a bell shaped trace instead of a sharp vertical line.

- The reference level control, spectrum analyzer vertical scale starts from the top of the display. The vertical display is in dB, where each vertical unit scale below the top represents 10dB.

1.2.4 Operation of Spectrum Analyzer

Figure 3 shows a typical spectrum display. This example shows the setting for a 20 MHz center frequency, a 1 MHz/division span and a reference level of -10 dBm. The 10 by 10 division display indicates RF level on the vertical or y-axis, and frequency on the horizontal or x-axis.

The reference level is the top line on the screen, and each vertical division down represents 10 dB. The center frequency is located at the center grid position of the display, and each division across the display represents 1MHz.

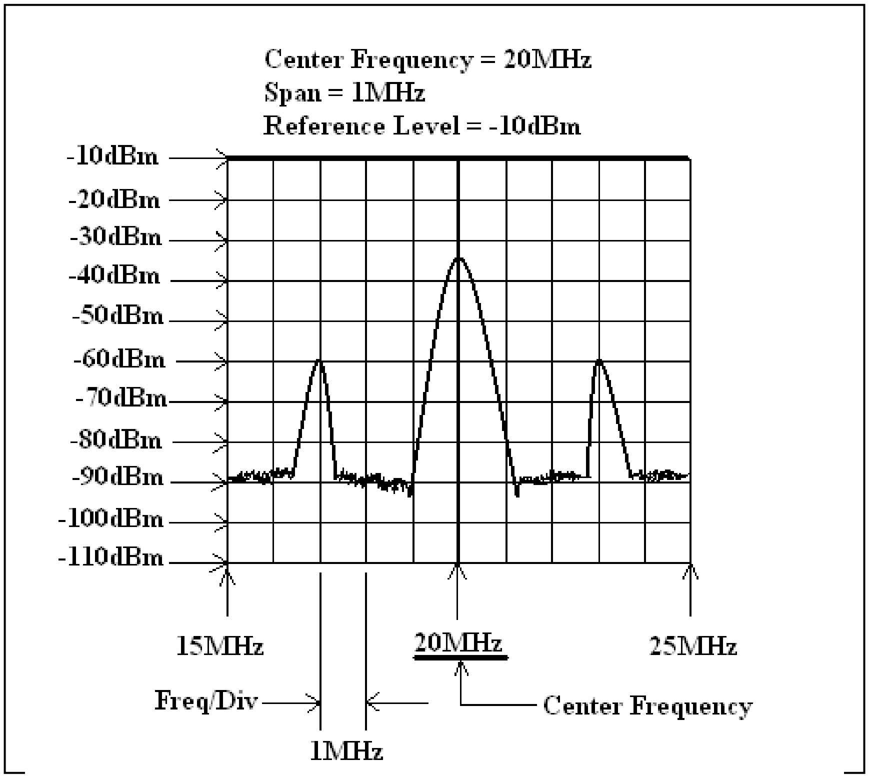


Fig.3: Spectrum Display, Example1

Another example of the spectrum analyzer display in shown in Fig. 4. This figure shows a single tone signal. If the center frequency is 10MHz and the span is 0.5MHz, then the tone is at 9MHz. Given the reference level at - 20dBm, then the tone power is -38dBm. Since the spectrum analyzer has a 500 termination, the tone is 2.8mVrms.

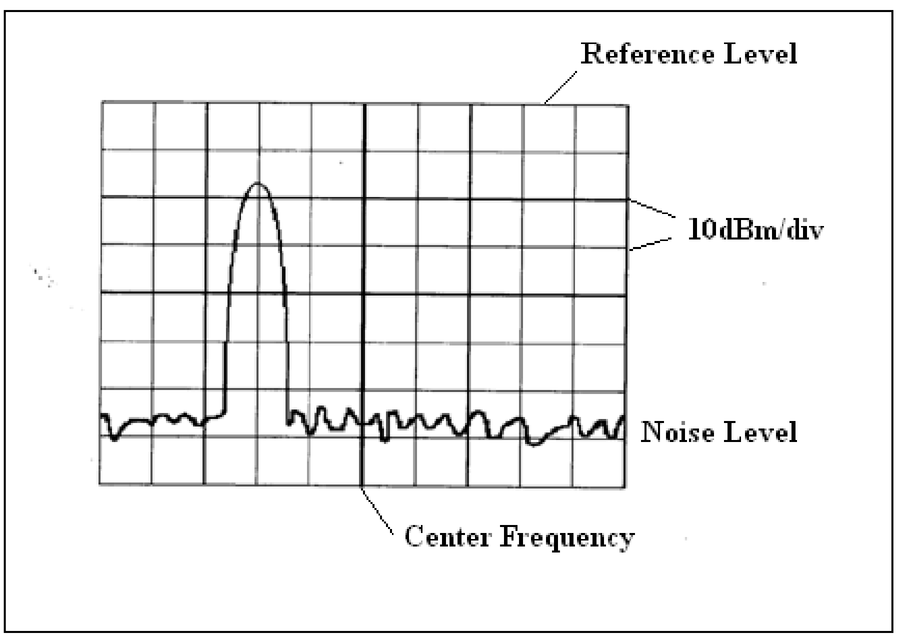


Fig.4: Spectrum Display, Example2

1.3 Procedure

Part1: Introducing the spectrum Analyzer

1. Turn the Spectrum. Analyzer on and let it to warm up.

2. Record the answers to the following questions:

- What is the frequency range that this spectrum analyzer will measure?
- What is the maximum DC level that can be applied to the RF input? What is the input impedance of the RF input?
- What is the maximum signal power, in dBm and in Watts, that can be applied to the RF input

Note: Before you connect any signal to the RF input, be sure that its amplitude or power does not exceed the maximum rated input. If you are unsure, measure the signal with the oscilloscope.

3. Given your answers to the questions in Item 2, calculate:

- The maximum amplitude sine wave (with zero DC offset) that can be applied to the RF input,
- The maximum amplitude square wave (with zero DC offset) having 50 % duty cycle that can be applied to the RF input, (When doing these calculations, don't forget what the input impedance of the analyzer is.)

Part2: Spectrum of a simple sinusoid

1. Use the function generator to produce a 1MHz sine wave of amplitude *200m Vp-p*. Use a BNC T splitter to connect this signal to the spectrum analyzer and the oscilloscope.

2. From the oscilloscope; record and explain the attenuation in the amplitude of the signal after connecting the spectrum analyzer. Do not increase the amplitude.

3. Set the center frequency of the spectrum analyzer to 5 MHz, the span of the display to 1MHz and set the RBW to 220kHz.

4. Record the spectrum on the display; use the marker to identify the different frequencies and amplitudes.

5. Compare your result with theoretical calculations.

6. Explain the presence of harmonics.

Part3: Spectrum of a square wave

1. Change the signal into the analyzer to a square wave 40mVP-P at 2MHz.

2. Record the display, identifying the fundamental frequency and the harmonics up to the 4th harmonic, (choose the best settings for the analyzer)

3. Compare these results with theoretical calculations.

4. Reduce the RBW, explain what happens.

He who makes no mistakes makes nothing

Good luck

Appendix A
Spectral Analysis

1. Signal Acquisition in a Spectrum Analyzer

Most spectrum analyzers are heterodyne spectrum analyzers (also called scanning spectrum analyzers). A heterodyne analyzer is essentially a radio receiver (a very sensitive and selective receiver).

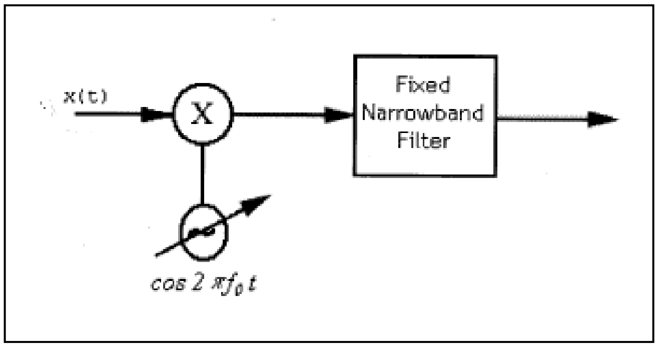


Fig.1: Frequency Mixing or Heterodyning

2. How Do Spectrum Analyzers Work?

The most common spectrum analyzers sweep the spectrum of a signal through a fixed band pass filter similar to the principle of super- heterodyne receiver. A block diagram of a spectrum analyzer is shown in Figure 2. The analyzer is basically a narrowband receiver, which is electronically tuned in frequency by applying a saw tooth voltage to a voltage-controlled oscillator (VCO). The same saw tooth is applied to the horizontal deflection plates of the CRT. The VCO frequency is mixed with the input to produce an intermediate frequency (IF).

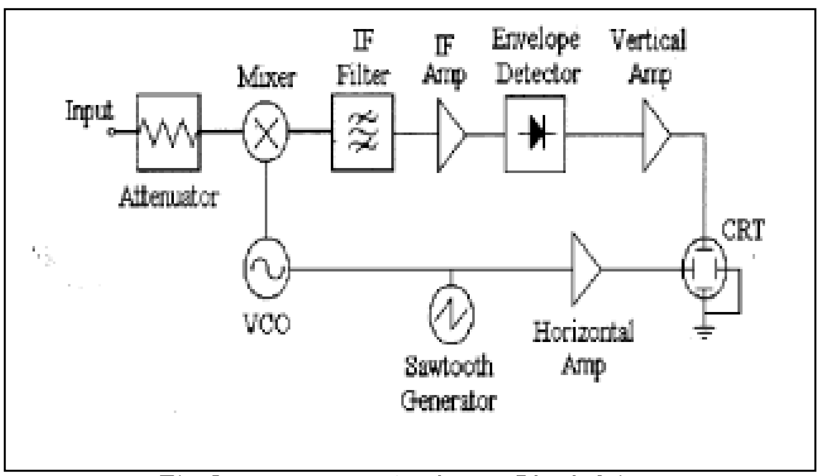


Fig.2: Spectrum Analyzer Block Diagram

The frequency component of the input equal to the difference between VCO and the IF is shifted to the IF, filtered and amplified and passed on to the detector, This produces a voltage proportional to the power in that frequency component of the input, which causes a comparable vertical deflection of the CRT beam at the relative horizontal position. This gives a plot of power vs. frequency. It is important to note that a peak always occurs at the zero frequency of the analyzer because of feeding from the VCO. This peak is called the zero frequency indicator and appears even when no signal is applied to the analyzer. The spectrum analyzer we will be using in the lab is "GW Instek Spectrum Analyzer" model GSP-810.

Experiment Two
Introd Introduction Amplitude Modulation

2.1 Objectives

- To recognize the functions of the main parts of the ANACOM 1/1 and 1/2 boards.
- Learn basic concepts of AM modulation.

2.2 Introduction to AM Modulation

In communication systems information is transmitted from one place to other using electrical signals (telephone, TV and radio broadcast etc.). The basic communications system has transmitter, channel and receiver as shown in Figure. 1.

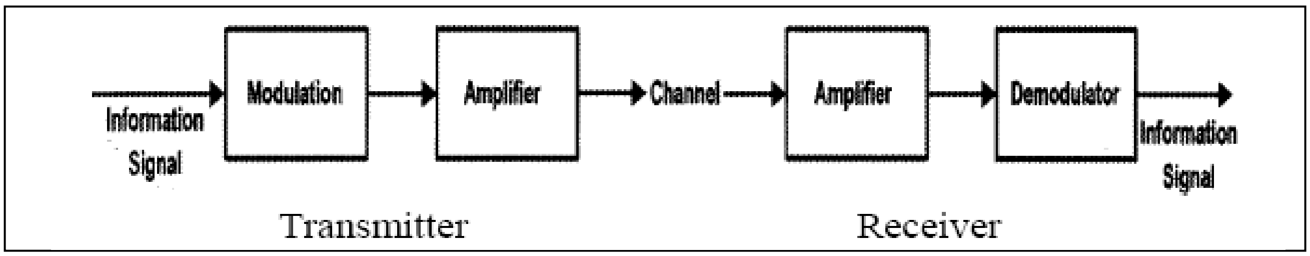


Fig.1: Basic Communication System

Usually the information bearing signals (message signals) are not suitable for transmission due to its propagation qualities (a large wavelength). Also, since these signals generally exist in the same frequency range it is necessary to transmit those using different frequency allocations to avoid interference.
One of the methods used to solve these problems is **linear modulation**, which is merely **the frequency translation of the spectrum of the information (or message) signal to a usually much higher frequency**.

The process of modulation means to systematically use the information signal to vary some parameter of the carrier signal. The carrier signal is usually just a simple; single- frequency sinusoid (varies in time like a sine wave). Any sinusoidal signal can be expressed as

$$x(t) = A \cos (2\pi f t+\phi) \qquad (2.1)$$

Where **x(t)** is the voltage of the signal as a function of time, **A** the amplitude of the signal , **f** the frequency of the signal, and **φ** the phase of the signal.

To modulate the signal just means to systematically vary one of the three parameters of the signal: amplitude, frequency or phase. Therefore, the type of modulation may be categorized as either amplitude modulation (AM), frequency modulation (FM) or phase modulation (PM).

2.2.1 Amplitude Modulation

If the amplitude of a high frequency sinusoid **A cos(2πf$_c$ t)** carrier, is forced to vary in proportion to a desired low frequency message signal **m(t)**, a modulated signal $x_{AM}(t)$ is generated whose frequency spectrum is concentrated in the vicinity of the carrier frequency. A typical amplitude modulated carrier has the form:

$$x_{AM}(t) = A\cos(2\pi f_c t) + m(t)\cos(2\pi f_c t) = [A + m(t)]\cos(2\pi f_c t) \qquad (2.2)$$

Two cases of AM signals are shown in Fig. 2. The spectrum of the modulated signal is given by

$$X_{AM}(f) = 1/2\,[M(f + f_c) + M(f - f_c)] + A/2\,[\delta(f + f_c) + \delta(f - f_c)] \qquad (2.3)$$

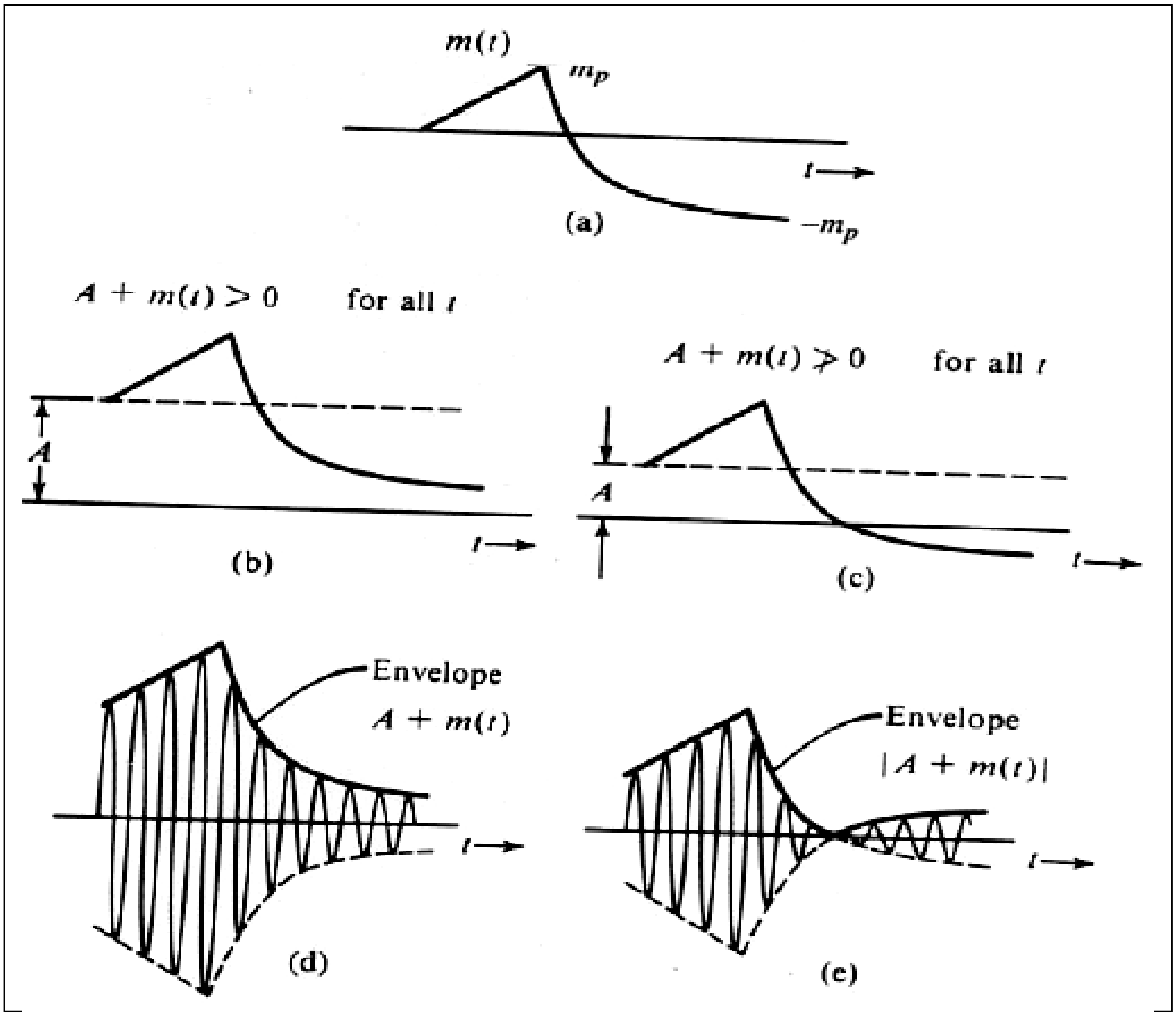


Fig. 2: AM Signals with its Envelope

The modulation index μ for AM signal is defined as:

$$\mu = \frac{\text{min peak vale of } m(t)}{A} \qquad (2.4)$$

When μ > 1 the envelope has no longer the shape of m (t) resulting in envelope distortion. This condition is referred to as over modulation.

Instead of using the envelope display to look at AM signals, an alternative is to use the trapezoidal pattern display as shown in Fig. 3-a. This is obtained by connecting the modulating signal to the **x** input of the 'scope and the modulated AM signal to the **y** input. The modulation index is measured as

$$\boxed{\mu = \frac{D-E}{D+E}}$$

Any distortion, over modulation, or non-linearity is easier to observe with this method, see Figure 3-b. This distortion is exhibited as a departure from straight lines for the upper and lower edges of the trapezoid.

The total power of the AM signal is the sum of the carrier power (Pc) and the sideband power (Ps). The sideband power is the useful power and the carrier power is the power wasted. Hence the power efficiency is.

$$\boxed{\eta = \frac{P_s}{P_c + P_s}} \qquad (2.6)$$

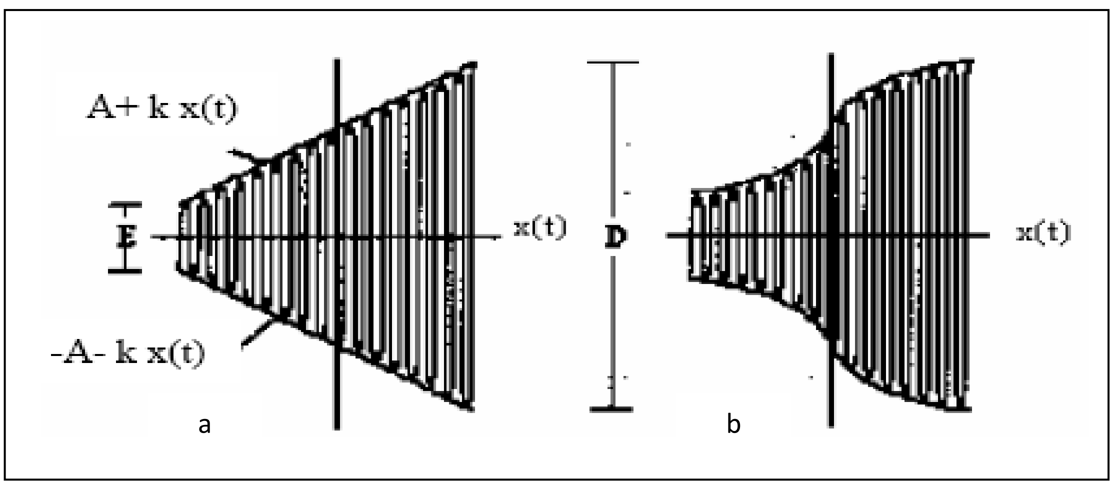


Fig.3: The Trapezoidal Pattern for:

a) Normal AM signal b) AM with Nonlinear Envelope.

For the special case of tone modulation with modulation index $0 \leq \mu \leq 1$

$$\boxed{\eta = \frac{\mu^2}{2+\mu^2}} \qquad (2.7)$$

Since a large amount of power carries no information, the carrier is sometimes removed before transmitting the AM signal.

The simplest modulation method to implement is the Double Sideband Suppressed Carrier Modulation (DSB-SC), in which the translated spectrum of the message signal is transmitted without further modification. The modulated signal is expressed as:

$$\boxed{x_{DSB}(t) = A\; m(t)\; \cos(2\pi f_c\, t)} \qquad (2.8)$$

And its spectrum given by:

$$\boxed{X_{DSB}(f) = \frac{A}{2}\left[M(f+f_c)+M(f-f_c)\right]} \qquad (2.9)$$

2.2.3 Experiment AM modulator/transmitter circuit

The transmitter circuits produce the amplitude-modulated signals which are used to carry information over the transmission path of the receiver. The AM modulator transmitter (Fig. 4) consists of; an audio oscillator; carrier generator; modulator and an output amplifier which connected to the transmitter antenna. The AM transmitter used in this experiment is rebuilt on a PCB. The purpose of the experiment is to observe the main signals and operations carried on them in a system approach. No detailed discussion of the electronics involved is intended.

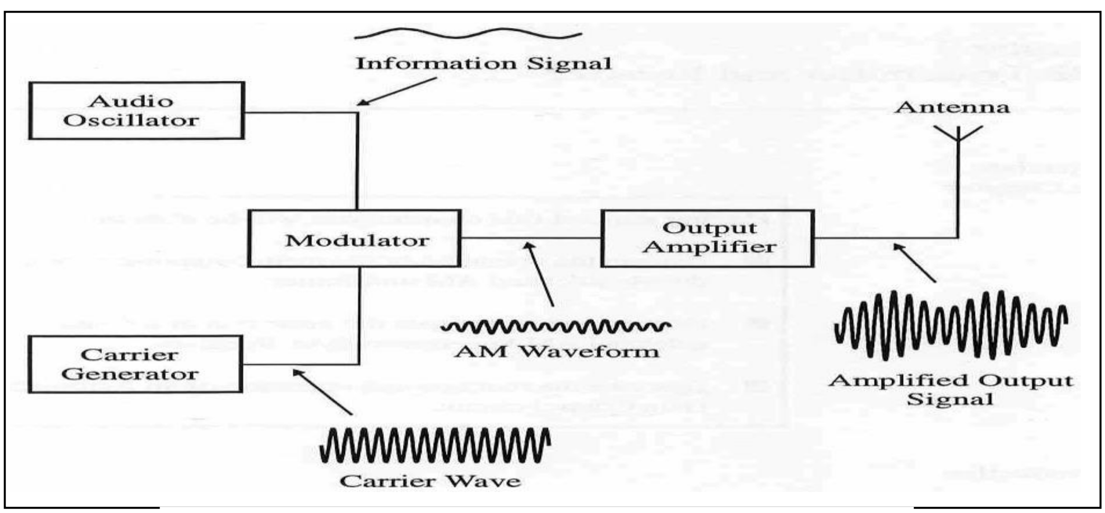


Fig. 4: An Amplitude Modulated Transmitter

The modulator in the experimental circuit is an analog multiplier IC (MC1496). This IC can provide either AM or DSBSC modulation. Internally, the IC is based on the Gilbert Cell principle. Figure-6 shows a simplified circuit diagram for operating the IC. The carrier and the audio signal inputs are single ended. The carrier is fed to pin 10 and the audio to pin 1.

The Modulation Level adjusts the modulation index. The output is available at pins 6 and 12.

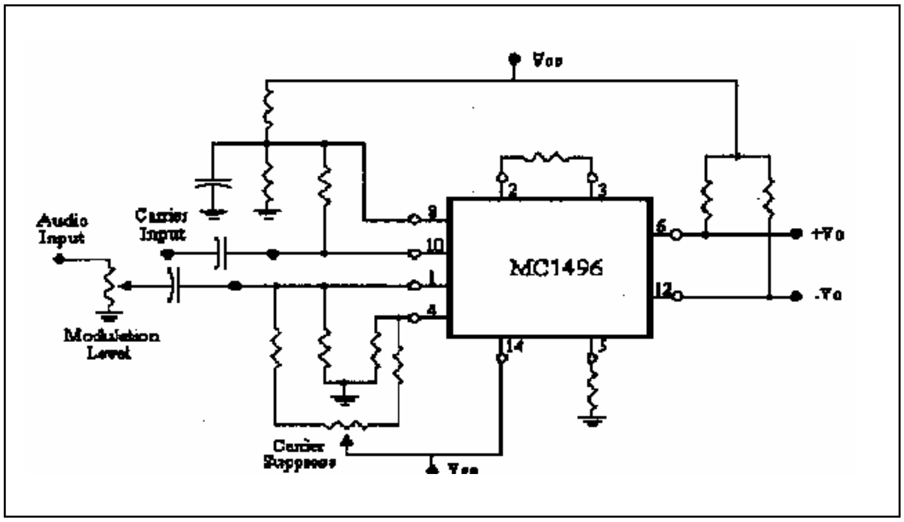


Fig. 5: AM Modulation Using The MC1496 IC.

2.3 The ANA COM 1/1 and ANA COM 1/2 Boards

This part describes the layouts of the ANACOM 1/1 and 1/2 boards and outlines the functions of the main circuits used. The aims of this section are recognizing the functions of the main components of the ANACOM 1/1 boards and the ANACOM 1/2 board and describe the main facilities offered by both boards. The layout diagram of the ANACOM 1/1 board and ANACOM 1/2 board are shown in Fig. 6 and Fig. 7.

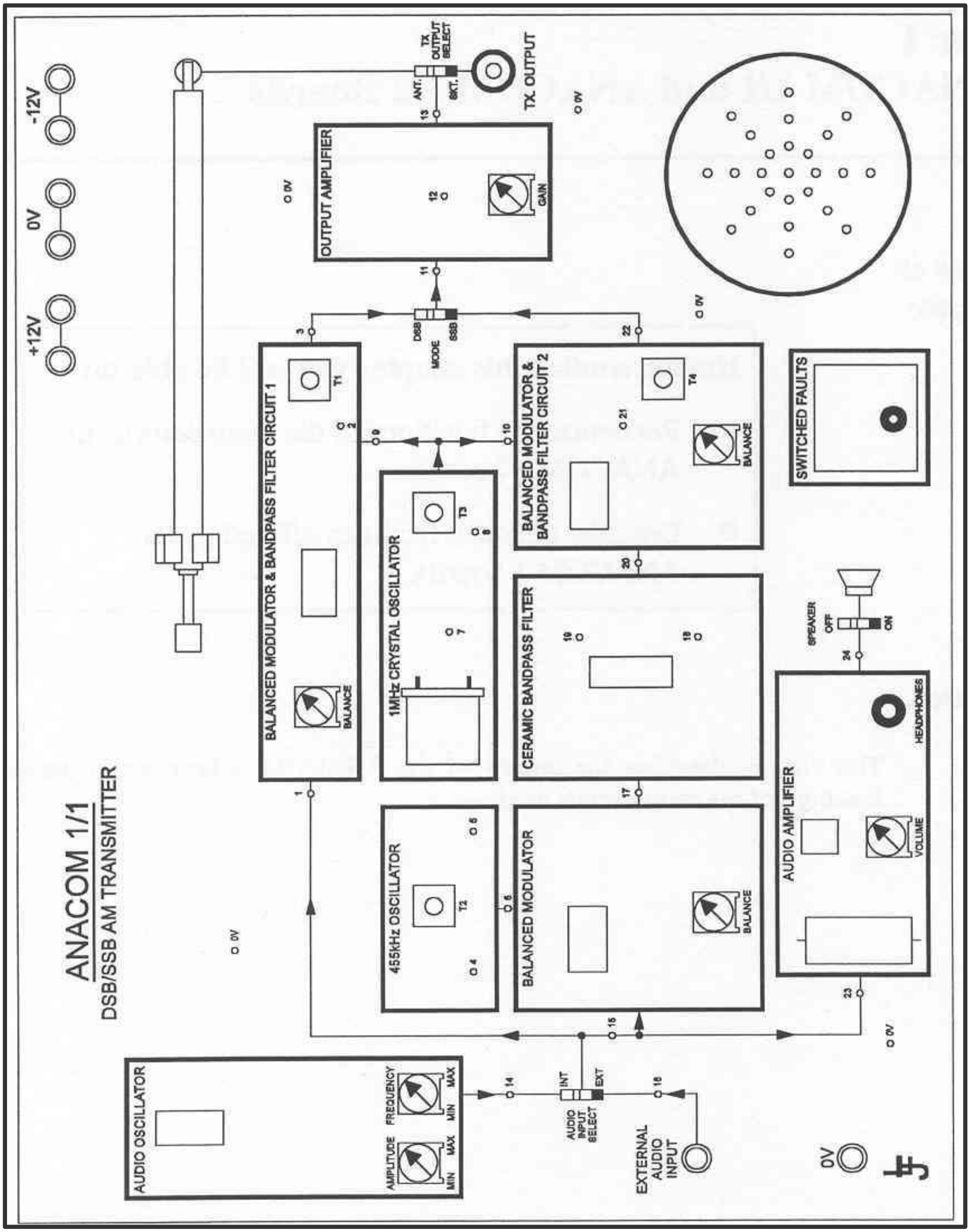


Fig. 6: ANACOM 1/1 DSB/SSB AM transmitter

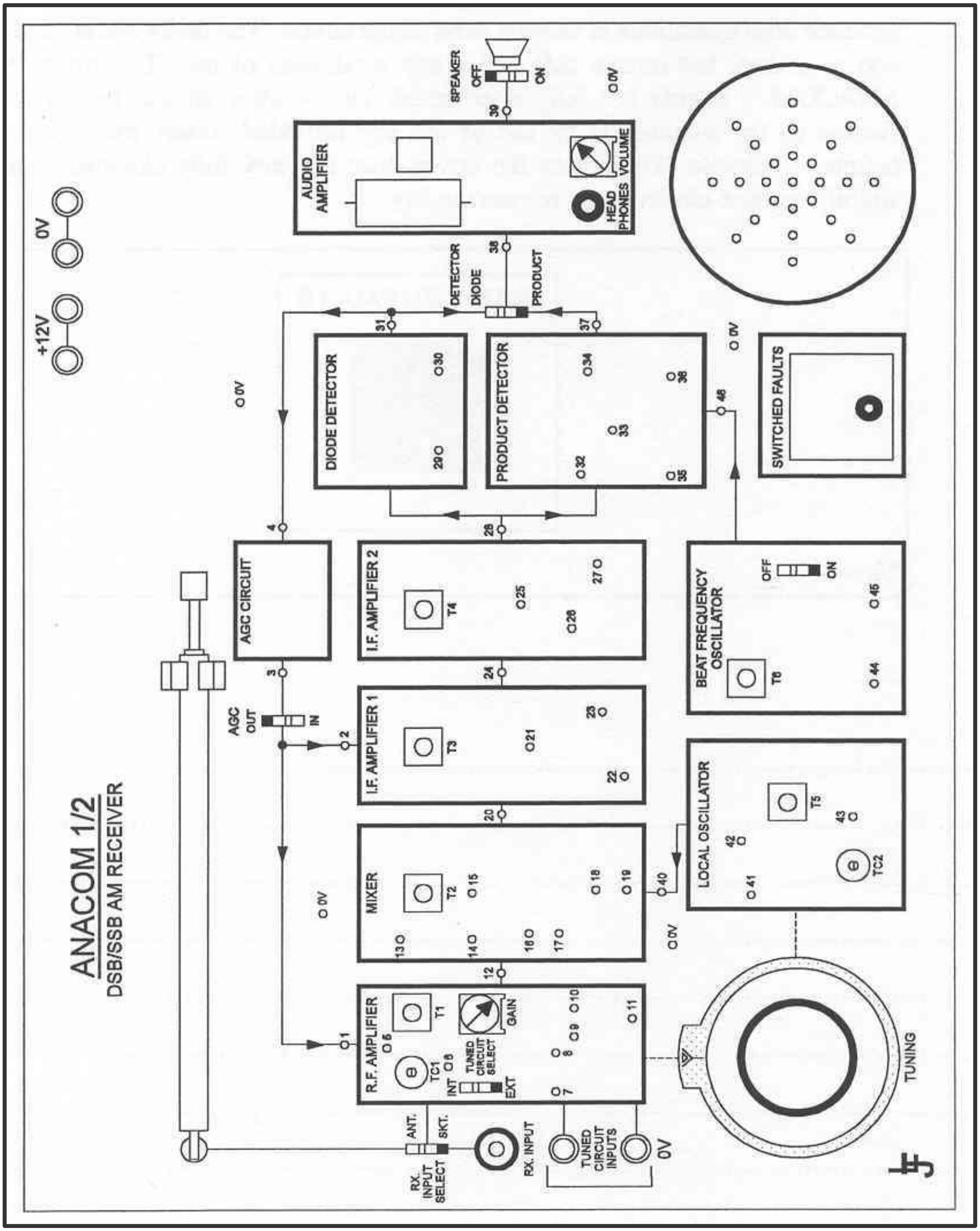


Fig. 7: ANACOM 1/2 DSB/SSB AM receiver

2.3.1 The ANA COM 1/1 Boards

The transmitter board can be considered as five separate blocks:

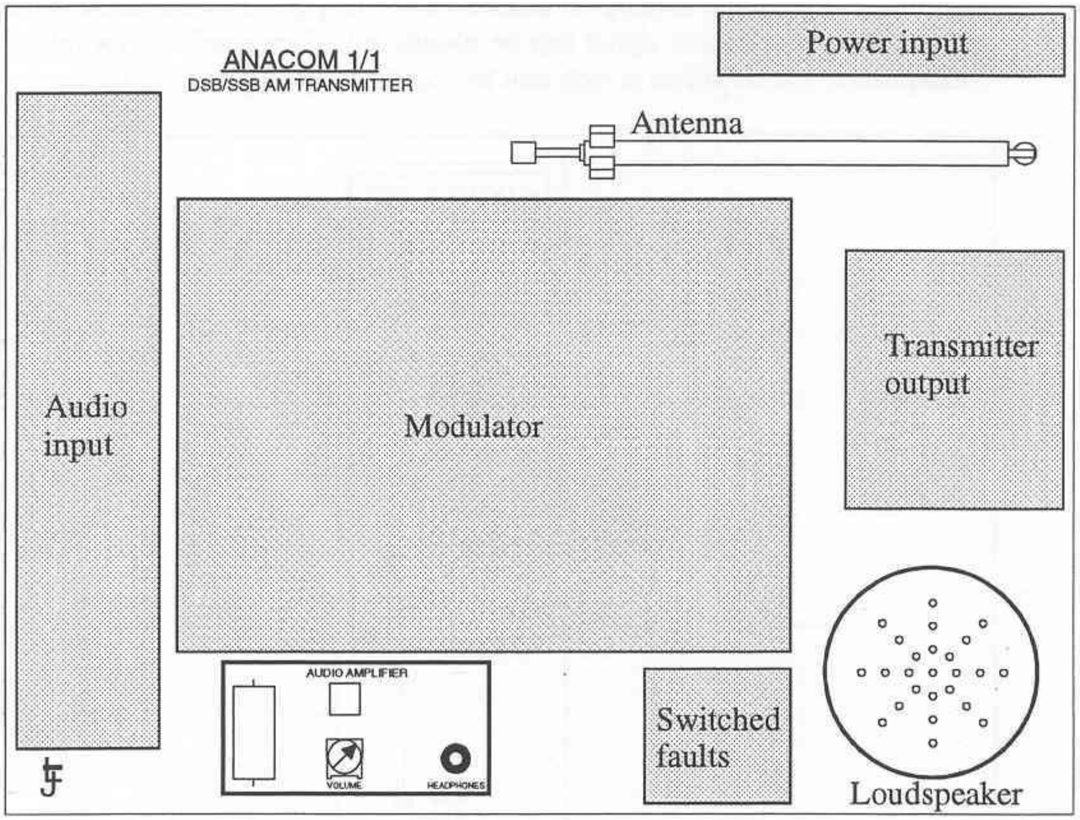


Fig. 8: ANACOM 1/1 Board Blocks

- Power Input

These are the electrical input connections necessary to power the module. The LJ Technical Systems "IC Power 60" or "System Power 90" are the recommended power supplies.

- The Audio Input and Amplifier

This circuit provides an internally generated signal that is going to be used as 'information' to demonstrate the operation of the transmitter. There is also an External Audio Input facility to enable us to supply our own audio information signals. The information signal can be monitored, if required, by switching on the loudspeaker. An amplifier is included to boost the signal power to the loudspeaker.

- The Modulator

This section of the board accepts the information signal and generates the final signal to be transmitted.

- The Transmitter Output

The purpose of this section is to amplify the modulated signal ready for transmission. The transmitter output can be connected to the receiver by a screened cable or by using the antenna provided. The on-board telescopic antenna should be fully extended to achieve the maximum range of about 4 feet (1.3m). After use, to prevent damage, the antenna should be folded down into the transit clip mounted on the ANACOM board.

- The Switched Faults

Under the black cover, there are eight switches. These switches can be used to simulate fault conditions in various parts of the circuit. The faults are normally used one at a time, but remain safe under any conditions of use. To ensure that the ANACOM 1 boards are fully operational, all switches should be set to OFF. Access to the switches is by use of the key provided. Insert the key and turn counter-clockwise.

To replace the cover: turn the key fully clockwise and then slightly counter-clockwise to release the key.

2.3.2 The ANA COM 1/2 Boards

The receiver board can be considered as five separate blocks:

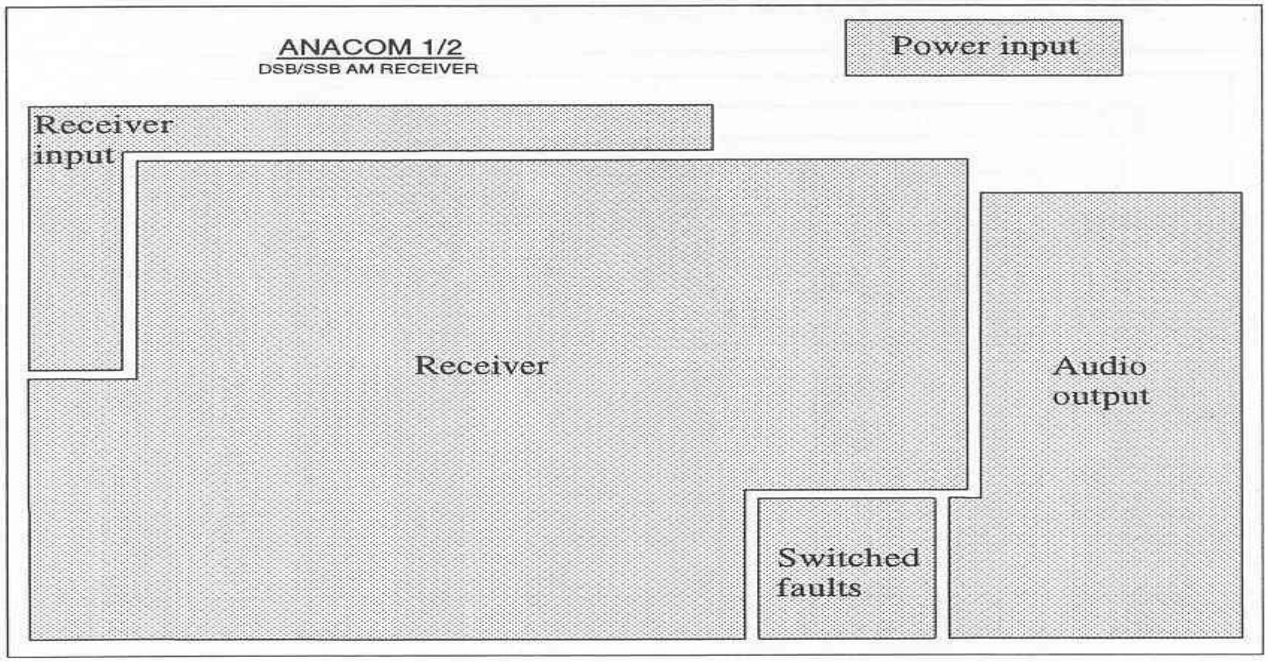


Fig. 9: ANACOM 1/2 Board Blocks

- Power Input

These are the electrical input connections necessary to power the module. The LJ Technical Systems "IC Power 60" or "System Power 90" are the recommended power supplies. If both ANACOM 1/1 and ANACOM 1/2 boards are to be used, they can be powered by the same power supply unit.

- The Receiver Input

In this section the input signals can be connected via a screened cable or by using the antenna provided. The telescopic antenna should be used fully extended and, after use, folded down into the transit clip.

- The Receiver

The receiver amplifies the incoming signal and extracts the original audio information signal. The incoming signals can be AM broadcast signals or those originating from ANACOM 1/1.

The Audio Output

The information signal from the receiver can be amplified and heard by using a set of headphones or, if required, by the loudspeaker provided.

4. Lab Procedure

- Connect the ANACOM 1/1 board to the power supply (see Figure 10).

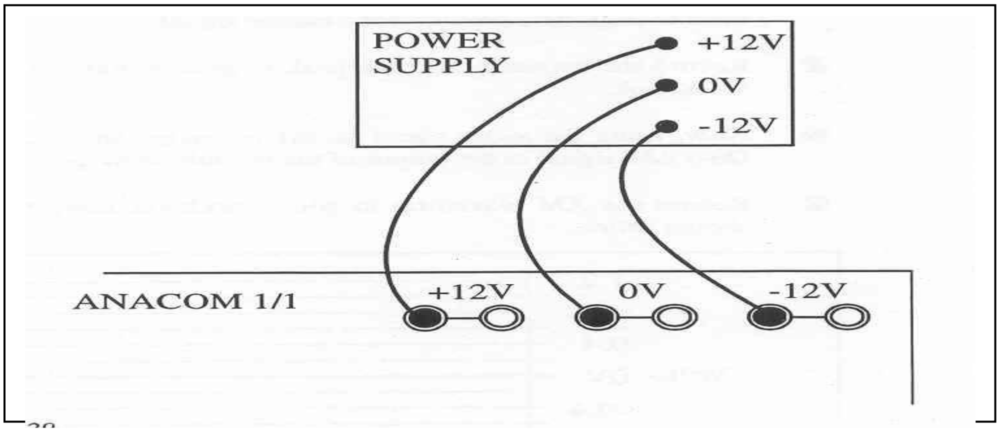


Fig. 10: Power Connection

- Ensure that the following initial conditions exist on the ANACOM 1/1:
  - AUDIO INPUT SELECT switch to INT position.
  - MODE switch to DSB.
  - SPEAKER switches to OFF.
  - In the AUDIO OSCILLATOR both the AMPLITUDE PRESET and the FREQUENCY PRESET should be set to maximum (fully clockwise).
  - In the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1, the BALANCE PRESET should be set to maximum (fully clockwise).
  - Switch on the power supply.

Part 1: AM Transmitter

1. Using CH1 oscilloscope examine the output of the AUDIO OSCILLATOR block at TP14. What is the range of the frequency and the amplitude you can get for the modulating signal?
2. In the AUDIO OSCILLATOR block turn the amplitude knob to its maximum and the frequency knob until you get a frequency of 850 Hz.
3. Connect CH2 to TP9, observe and record the carrier signal. What is the amplitude and frequency of the carrier?
4. Turn the BALANCE knob in the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT block to its fully clockwise position.

5. Connect CH1 to TP1 and CH2 to TP3. Observe the output signal. Always use CH1 as trigger channel.
6. Calculate the modulation index.
7. Switch the oscilloscope to XY mode. Record the trapezoidal pattern. Find the modulation index from the trapezoidal pattern.
8. Use the trapezoidal pattern change the BALANCE knob to get 100 % modulation. Record the resulting waveforms the modulated AM signal and the trapezoid pattern.
9. Reverse the X-Y inputs at the scope. Describe the effects that occur in relation to the trapezoidal pattern.
10. Change the message amplitude by varying the modulation level, and find the modulation index for the message amplitudes (0, 0.5, 1.5 VPP). Plot a graph of message amplitude vs. modulation index.

Note: turn the BALANCE knob in the BALANCED MODULATOR and BP- FILTER block to its fully clockwise position.

11. What can be concluded about this method for AM modulation measurements?

Part 2: DSB-SC Transmitter

1. Turn the BALANCE preset in the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT towards the central position, and observe the change in the shape of the modulated waveform.
2. Adjust the BALANCE knob where both peaks of the modulated signal are equal. Draw this waveform.
3. Switch the oscilloscope to XY mode. Record the trapezoidal pattern for the suppressed carrier case.
4. Measure the gain of the OUTPUT AMPLIFIER, by measuring the amplitude of the signal at TP 11 (the input to the OUTPUT AMPLIFIER), and the amplitude of the signal at TP 13 (the output of the OUTPUT AMPLIFIER), remember that the gain equal Vo / Vin.

A bad Workman Blames
His Tools

Good luck

Experiment 3
Double Sideband
(Transmitter and Receiver)

3.1 Objectives

- Explain the operation of the main components of a double sideband AM transmitter.
- Explain and investigate the operation of a double sideband AM Super-heterodyne Receiver.
- Describe the function and operation of an Automatic Gain Control circuit.

3.2 Double Sideband Transmitter and Receiver

This section explains the main components of a double sideband (DSB) AM transmitter and receiver.

3.2.1 DSB Transmitter

The transmitter circuits produce the amplitude modulated signals which are used to carry information over the transmission path to the receiver. The main parts of the transmitter are shown in Fig. 3.1.

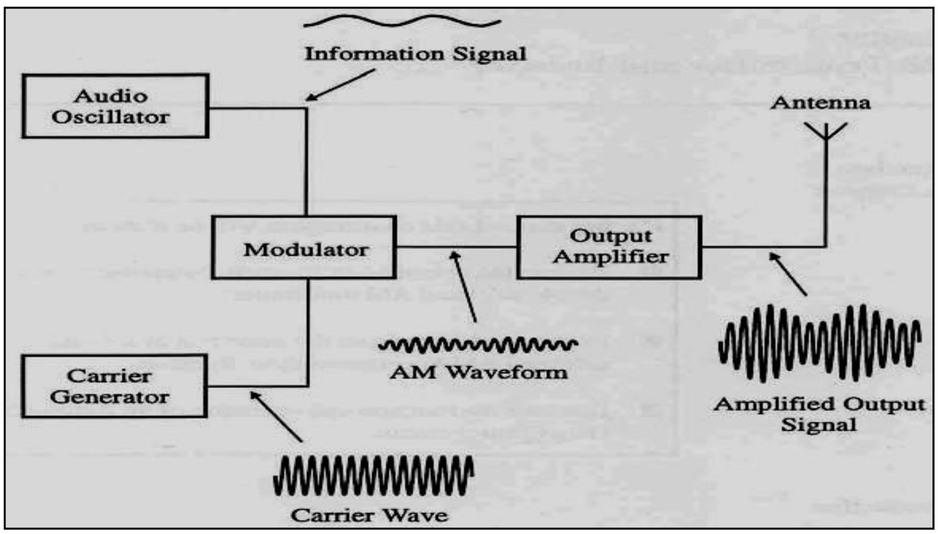


Fig.3.1: An Amplitude Modulated Transmitter

In Figures 3.1 and 3.2, it can be seen that the peak-to-peak voltages in the AM waveform increase and decrease in sympathy with the audio signal.

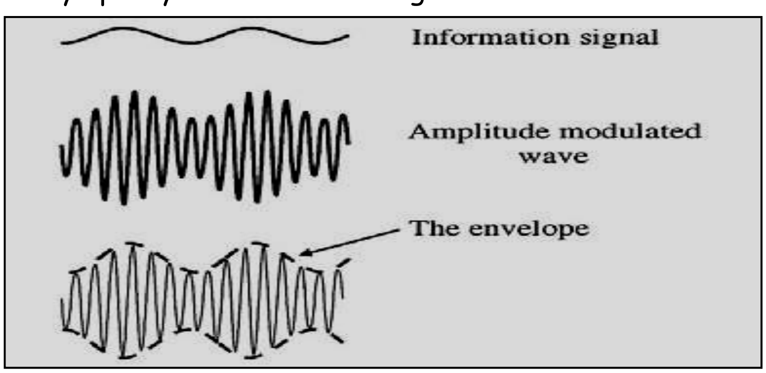


Fig.3.2: The Modulation Envelope

To emphasize the connection between the information and the final waveform, a line is sometimes drawn to follow the peaks of the carrier wave as shown in Figure 3.2. This shape, enclosed by a dashed line in the diagram, is referred to as an 'envelope', or a 'modulation envelope'. It is important to appreciate that it is only a guide to emphasize the shape of the AM waveform.

The action of each component will be discussed in the next subsections. The first task is to get hold of the information to be transmitted.

The Information Signal

There is a choice of information signals on ANACOM 1/1. We can use the audio signal provided in the audio oscillator or we can provide our own input signal from an external source. A convenient way of providing an external source is by using the optional AUDIO INPUT MODULE CT7 which allows us to plug in a microphone or a cassette recorder. In test situations it is more satisfactory to use a simple sinusoidal information signal since its attributes are known and of constant value. We can then measure various characteristics of the resultant AM waveform, such as the modulation depth for example. Such measurements would be very difficult if we were using a varying signal from an external source such as a broadcast station. The next step is to generate the carrier wave.

The Carrier Wave

The carrier wave must meet two main criteria. It should be of a convenient frequency to transmit over the communication path in use. In a radio link transmissions are difficult to achieve at frequencies less than 15 kHz and few radio links employ frequencies above 10GHz. Outside of this range the cost of the equipment increases rapidly with very few advantages.

Remember that although 15 kHz is within the audio range, we cannot hear the radio signal because it is an electromagnetic wave and our ears can only detect waves which are due to changes of pressure. The second criterion is that the carrier wave should also be a sinusoidal waveform. Can you see why?

A sinusoidal signal contains only a single frequency and when modulated by a single frequency, will give rise to just two side frequencies, the upper and the lower side frequencies. However, if the sine wave were to be a complex wave containing many different frequencies, each separate frequency component would generate its own side frequencies. The result is that the overall bandwidth occupied by the transmission would be very wide and, on the radio, would cause interference with the adjacent stations.

On ANACOM 1/1, the carrier wave generated is a sine wave of 1MHz. Now we have the task of combining the information signal and the carrier wave to produce amplitude modulation.

The Modulator

There are many different designs of amplitude modulator. They all achieve the same result. The amplitude of the carrier is increased and decreased in sympathy with the incoming information signal. If the modulation process has given rise to any unwanted frequency components then a band-pass filter can be employed to remove them.

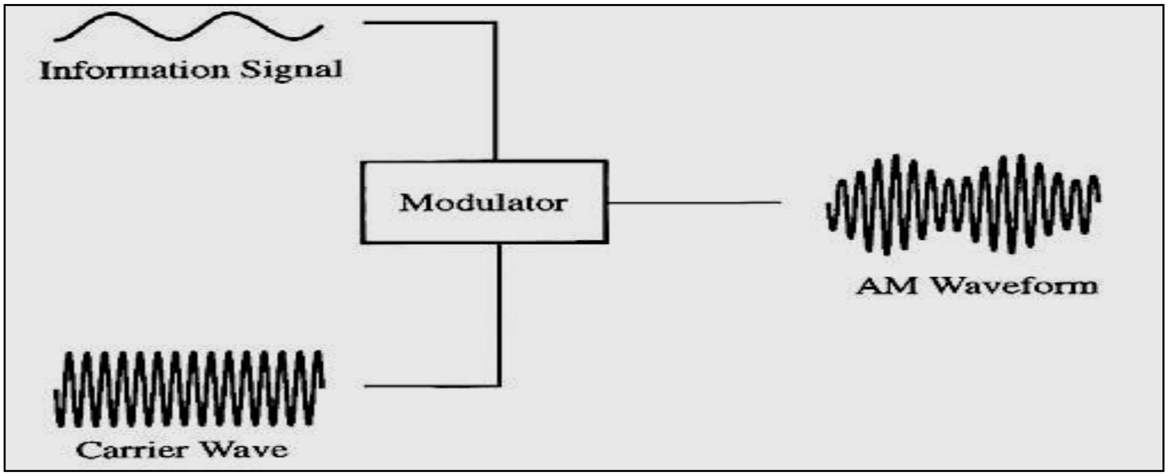


Fig. 3.3: Modulator

Output Amplifier (or Power Amplifier)
This amplifier is used to increase the strength of the signal before being passed to the antenna for transmission. The output power contained in the signal and the frequency of transmission are the two main factors that determine the range of the transmission.

The Antenna
An electromagnetic wave, such as a light ray, consists of two fields, an electric field and a magnetic field. These two fields are always at right angles to each other and move in a direction which is at right angles to both the magnetic and the electric fields; this is shown in Figure 3.4. The antenna converts the power output of the Output Amplifier into an electromagnetic wave. How does it do this?
For maximum efficiency the antenna should be of a precise length. The optimum size of antenna for most purposes is one having an overall length of one quarter of the wavelength of the transmitted signal. This can be found by

$$\lambda = \frac{v}{f}$$

Where **v**: speed of light, **λ**: wavelength and **f**: frequency in Hertz. In the case of the ANACOM 1/1, the transmitted carrier is 1MHz and so the ideal length of antenna is:
λ = 300m. One quarter of this wavelength would be 75 meters (about 245 feet).
We can now see that the antenna provided on the ANACOM 1/1 is necessarily less than the ideal size!

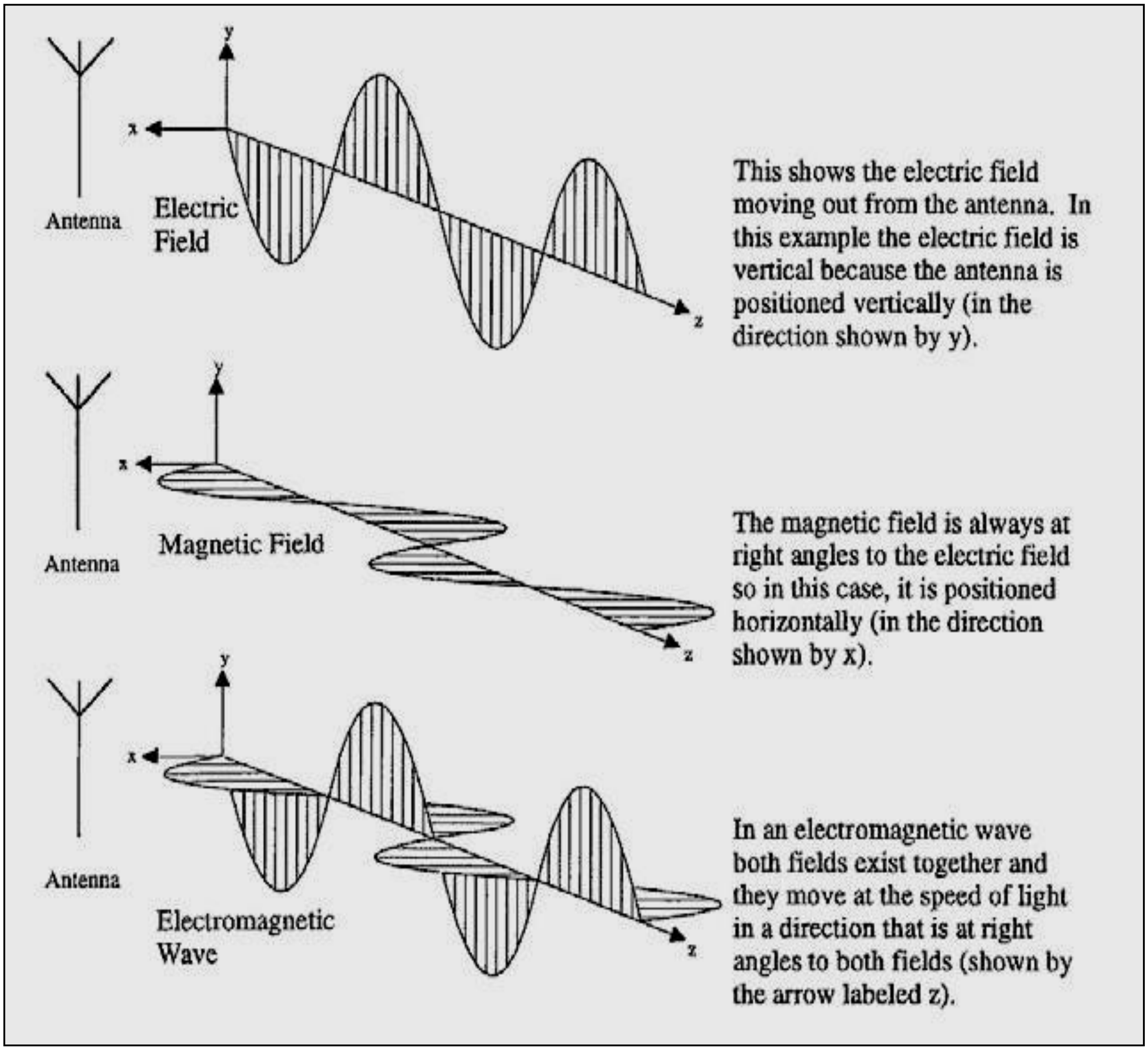


Fig.3.4: An Electromagnetic wave

Polarization

If the transmitting antenna is placed vertically, the electrical field is vertical and the magnetic field is horizontal (as seen in Figure 3.4). If the transmitting antenna is now moved by 90° to make it horizontal, the electrical field is horizontal and the magnetic field becomes vertical. By convention, we use the plane of the electric field to describe the orientation, or polarization, of the electromagnetic (EM) wave. A vertical transmitting antenna results in a vertically polarized wave, and a horizontal one would result in a horizontally polarized EM wave.

3.2.2 The DSB Receiver

The EM wave from the transmitting antenna will travel to the receiving antenna, carrying the information with it. Fig. 3.5 shows a typical block diagram for a receiver. We will proceed with the discussion of each block.

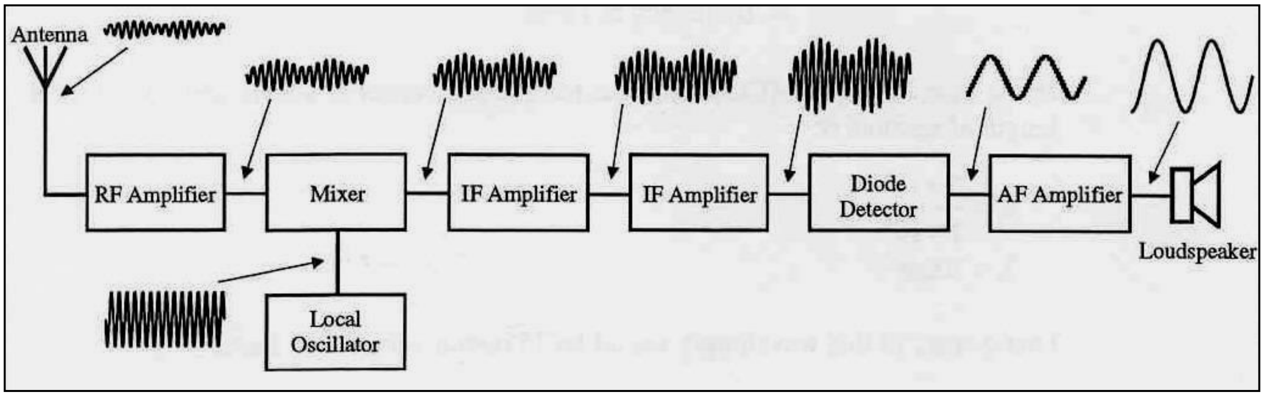


Fig. 3.5: A Super- heterodyne Receiver

The Receiving Antenna

The receiving antenna operates in the reverse mode to the transmitter antenna. The electromagnetic wave strikes the antenna and generates a small voltage in it. Ideally, the receiving antenna must be aligned to the polarization of the incoming signal so generally, a vertical transmitting antenna will be received best by using a vertical receiving antenna.

The actual voltage generated in the antenna is very small – usually less than 50 mV and often only a few microvolts. The voltage supplied to the loudspeaker at the output of the receiver is up to ten volts. Clearly, there is need a lot of amplification.

The Radio Frequency (RF) Amplifier

The antenna not only provides very low amplitude input signals but it picks up all available transmissions at the same time. This would mean that the receiver output would include all the various stations on top of each other which would make it impossible to listen to anyone transmission. The receiver circuits generate noise signals which are added to the wanted signals. We hear this as a background hiss and is particularly noticeable if the receiver is tuned between stations or if a weak station is being received. The RF amplifier is the first stage of amplification. It has to amplify the incoming signal above the level of the internally generated noise and also to start the process of selecting the wanted station and rejecting the unwanted ones.

Selectivity

A parallel tuned circuit has its greatest impedance at resonance and decreases at higher and lower frequencies. If the tuned circuit is included in the circuit design of an amplifier, it results in an amplifier which offers more gain at the frequency of resonance and reduced amplification above and below this frequency. This is called selectivity.

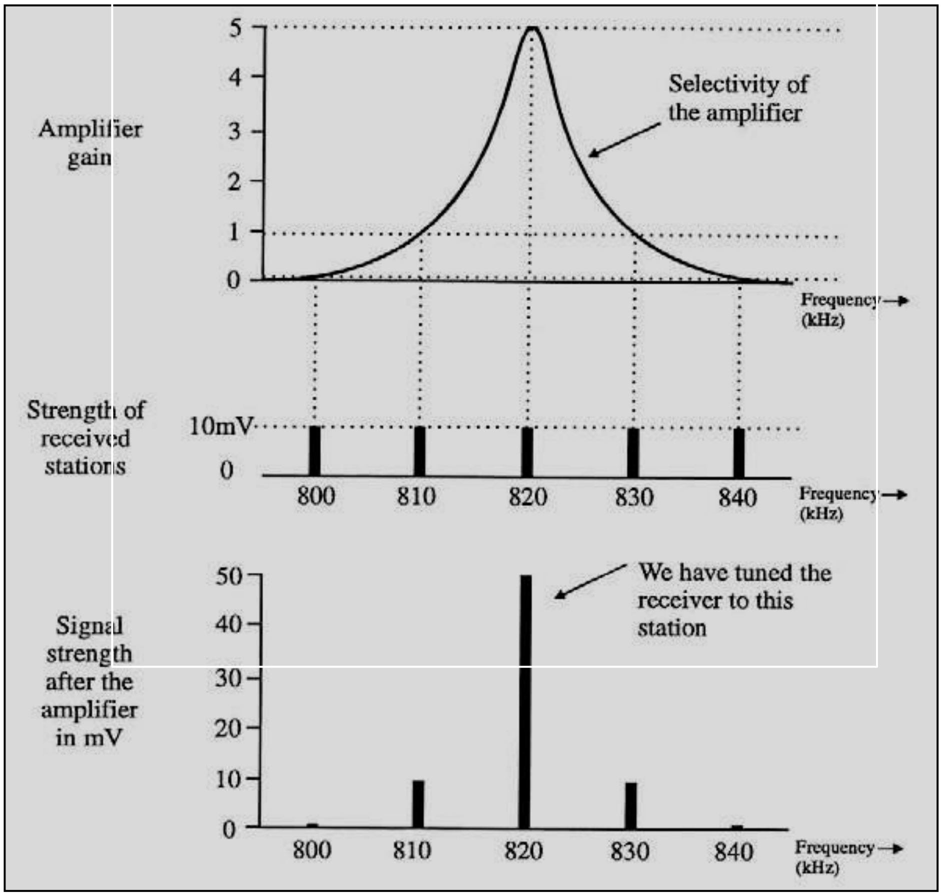


Fig. 3.6: Selectivity of the RF Amplifier

In Figure 3.6 we can see the effects of using an amplifier with selectivity. The radio receiver is tuned to a frequency of 820 kHz and, at this frequency; the amplifier provides a gain of five. Assuming the income signal has an amplitude of 10mV as shown, its output at this frequency would be 5 x 10m V= 50mV. The stations being received at 810 kHz and 830 kHz each have a gain of one. With the same amplitude of 10m V, this would result in outputs of 10mV. The stations at 800 kHz and 840 kHz are offered a gain of only 0.1 (approx.). This means that the output signal strength would be only 1mV.

The overall effect of the selectivity is that whereas the incoming signals each have the same amplitude, the outputs vary between 1mV and 50mV so we can select, or 'tune', the amplifier to pick out the desired station. The greatest amplification occurs at the resonance frequency of the tuned circuit. This is sometimes called the center frequency.

In common with nearly all radio receivers, ANACOM 1/2 adjusts the capacitor value by means of the TUNING control to select various signals.

The Local Oscillator

This is an oscillator producing a sinusoidal output similar to the carrier wave oscillator in the transmitter. In this case however, the frequency of its output is adjustable.

The same tuning control is used to adjust the frequency of both the local oscillator and the center frequency of the RF amplifier. The local oscillator is always maintained at a frequency which is higher, by a fixed amount, than the incoming RF signals. The local oscillator frequency therefore follows, or tracks, the RF amplifier frequency. This will prove to be very useful as we will see in the next section.

The Mixer (or Frequency Changer)

The mixer performs a similar function to the modulator in the transmitter.

We may remember that the transmitter modulator accepts the information signal and the carrier frequency, and produces the carrier plus the upper and lower sidebands.

The mixer in the receiver combines the signal from the RF amplifier and the frequency input from the local oscillator to produce three frequencies:

- ✓ A 'difference' frequency of local oscillator frequency – RF signals frequency.
- ✓ A 'sum' frequency equal to local oscillator frequency + RF signal frequency.
- ✓ A component at the local oscillator frequency.

Mixing two signals to produce such components is called a 'heterodyne' process. When this is carried out at frequencies which are above the audio spectrum, called 'supersonic' frequencies, the type of receiver is called a 'super-heterodyne' receiver.

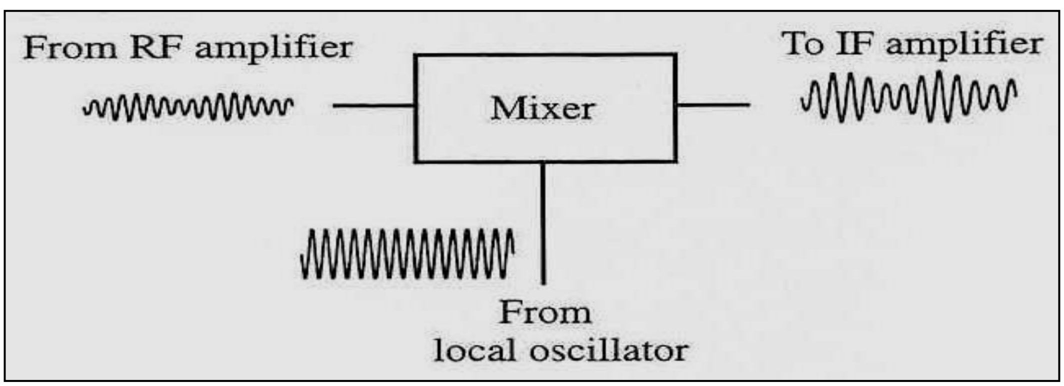


Fig. 3.7: The Mixer

We saw in the previous section how the local oscillator tracks the RF amplifier so that the difference between the two frequencies is maintained at a constant value. In ANACOM 1/2 this difference is actually 455 kHz. As an example, if the radio is tuned to receive a broadcast station which transmits at 800 kHz, the local oscillator will be running at 1.255 MHz. The difference frequency is 1.255 MHz – 800 kHz = 455 kHz.

If the radio is now retuned to receive a different station being broadcast on 700 kHz, the tuning control readjusts the RF amplifier to provide maximum gain at 700 KHz and the local oscillator to 1.155MHz. The difference frequency is still maintained at the required 455 KHz.

This frequency difference therefore remains constant regardless of the frequency to which the radio is actually tuned and is called the intermediate frequency (IF).

Fig. 3.8: A Super-heterodyne Receiver Tuned to 800 kHz

Image Frequencies

In the last section, we saw we could receive a station being broadcast on 700 kHz by tuning the local oscillator to a frequency of 1.155 MHz thus giving the difference (IF) frequency of the required 455 KHz. What would happen if we were to receive station broadcasting on a frequency of 1.61 MHz?

This would also mix with the local oscillator frequency of 1.155 MHz to produce the required IF frequency of 455 kHz. This would mean that this station would also be received at the same time as our wanted one at 700 kHz.

Station 1: Frequency 700 kHz, Local oscillator 1.155 MHz, IF = 455 kHz
Station 2: Frequency 1.61 MHz, Local oscillator 1.155 MHz, IF = 455 kHz

An 'image frequency' is an unwanted frequency that can also combine with the Local Oscillator output to create the IF frequency. Notice how the difference in frequency between the wanted and unwanted stations is twice the IF frequency. In the ANACOM 1/2, it means that the image frequency is always 910 KHz above the wanted station.

This is a large frequency difference and even the poor selectivity of the RF amplifier is able to remove the image frequency unless it is very strong indeed. In this case it will pass through the receiver and will be heard at the same time as the wanted station. Frequency interactions between the two stations tend to cause irritating whistles from the loudspeaker.

Intermediate Frequency Amplifiers (IF Amplifiers)

The IF amplifier in this receiver consists of two stages of amplification and provides the main signal amplification and selectivity. Operating at a fixed IF frequency means that the design of the amplifiers can be simplified. If it were not for the fixed frequency, all the amplifiers would need to be tunable across the whole range of incoming RF frequencies and it would be difficult to arrange for all the amplifiers to keep in step as they are re-tuned.

In addition, the radio must select the wanted transmission and reject all the others. To do this the band-pass of all the stages must be carefully controlled. Each IF stage does not necessarily have the same band-pass characteristics, it is the overall response that is important. Again, this

is something which is much more easily achieved without the added complication of making them tunable.
At the final output from the IF amplifiers, we have a 455 KHz wave which is amplitude modulated by the wanted audio information. The selectivity of the IF amplifiers has removed the unwanted components generated by the mixing process.

The Diode Detector
The function of the diode detector is to extract the audio signal from the signal at the output of the IF amplifiers. It performs this task in a very similar way to a half wave rectifier converting an AC input to a DC output. Figure 3.9 shows a simple circuit diagram of the diode detector

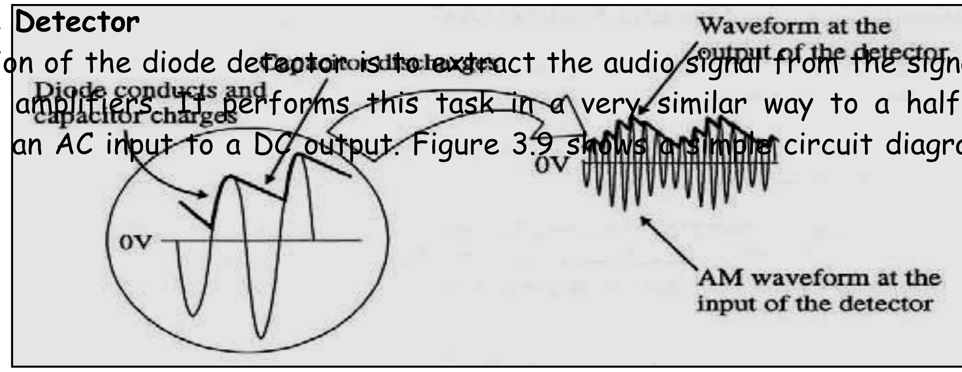


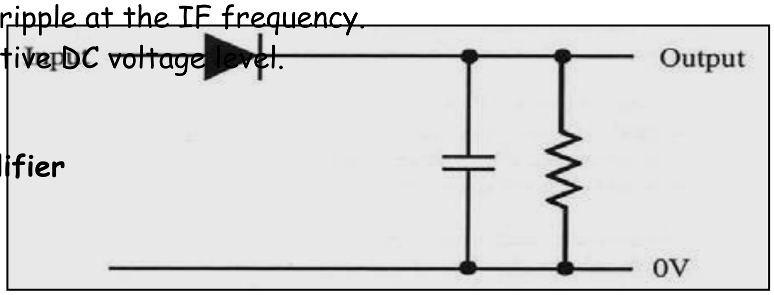
Fig. 3.9: A simple Diode Detector

In Figure 3.9, the diode conducts every time the input signal applied to its anode is more positive than the voltage on the top plate of the capacitor. When the voltage falls below the capacitor voltage, the diode ceases to conduct and the voltage across the capacitor leaks away until the next time the input signal is able to switch it on again (see Figure 3.10).

Fig. 3.10: Diode Product Operation

The result is an output which contains three components:

- ❖ The wanted audio information signal.
- ❖ Some ripple at the IF frequency.
- ❖ A positive DC voltage level.



The Audio Amplifier

At the input to the audio amplifier, a low-pass filter is used to remove the IF ripple and a capacitor blocks the DC voltage level. Figure 3.11 shows the result of the information signal passing through the Diode Detector and Audio Amplifier. The remaining audio signals are then amplified to provide the final output to the loudspeaker.

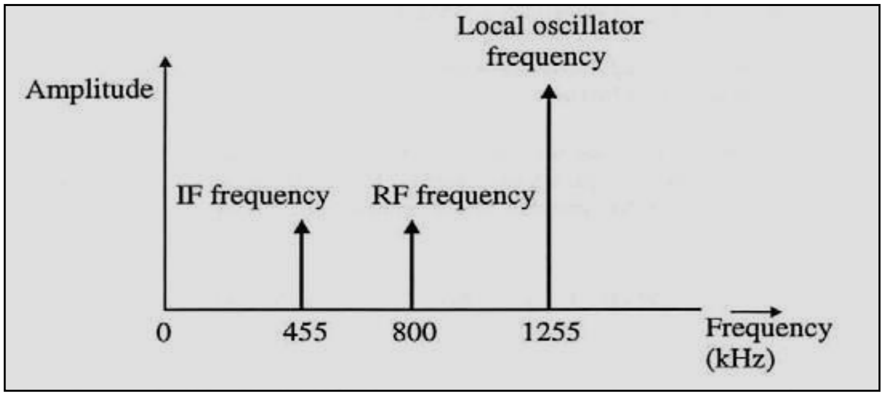
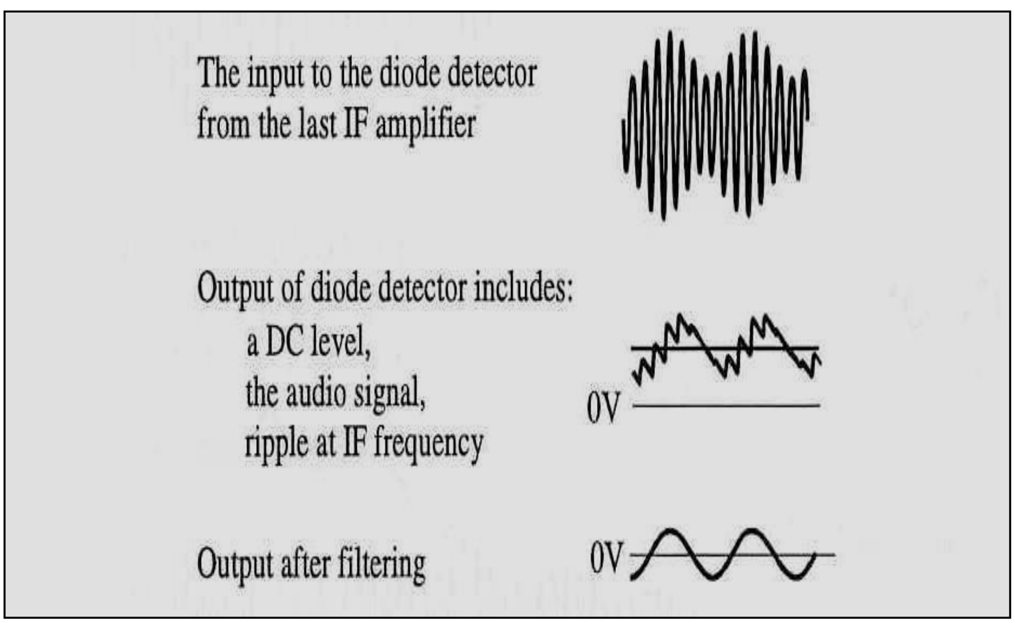


Fig. 3.11: Modulation Process



The Automatic Gain Control Circuit (AGC)

The AGC circuit is used to prevent very strong signals from overloading the receiver. It can also reduce the effect of fluctuations in the received signal strength. The AGC circuit makes use of the mean DC voltage level present at the output of the diode detector. If the signal strength increases, the mean DC voltage level also increases. If the mean DC voltage level exceeds a predetermined threshold value, a voltage is applied to the RF and IF amplifiers in such a way as to decrease their gain to prevent overload. As soon as the incoming signal strength decreases, such that the mean DC voltage level is reduced below the threshold, the RF and IF amplifiers return to their normal operation.

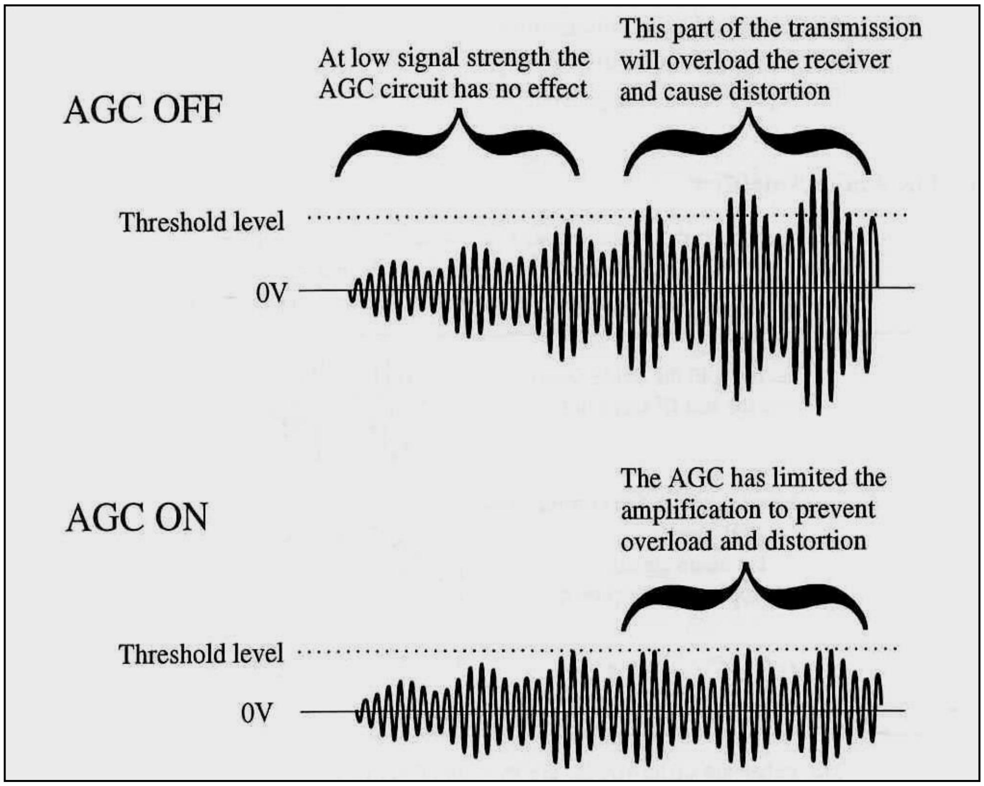


Fig. 3.12: AGC Operation

The mean DC voltage from the detector is averaged out over a period of time to ensure that the AGC circuit is really responding to fluctuations in the strength of the received signals and not to individual cycles.

Some designs of AGC circuit provide a progressive degree of control over the gain of the receiver at all levels of input signals without using a threshold level. This type is more effective at counteracting the effects of fading due to changes in atmospheric conditions. The alternative is to employ an AGC circuit as used in ANACOM 1/2. In this case the AGC action does not come into effect until the mean value reaches the threshold value, this type of AGC circuit is often referred to as 'Delayed AGC'.

3.3. Procedure

1. Connect the ANACOM 1/1 and ANACOM 1/2 boards to the power supply.

2. On ANACOM 1/1 set the following initial conditions:
   - AUDIO INPUT SELECT switch to INT position.
   - MODE switch to DSB.
   - SPEAKER switches to OFF.
   - In the AUDIO OSCILLATOR both the AMPLITUDE PRESET and the FREQUENCY PRESET should be set to maximum (fully clockwise).
   - In the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1, the BALANCE PRESET should be set to maximum (fully clockwise).
   - In the OUTPUT AMPLIFIER, increase the gain to its maximum value (fully clockwise).
   - The TX OUTPUT SELECT should be set to ANT (antenna) and the antenna should be in a vertical position and fully extended.

3. On ANACOM 1/2 set the following initial conditions:
   - In the AUDIO AMPLIFIER switch the SPEAKER to ON and decrease the VOLUME preset to its minimum value (fully counter-clockwise).
   - RX INPUT SELECT switch to ANT (antenna).
   - In the RF amplifier switch the TUNED CIRCUIT SELECT to INT (internal) position and increase the RF AMPLIFIER GAIN CONTROL to maximum (fully clockwise).
   - Set the AGC switch to the IN position.
   - Set the DETECTOR switch to the DIODE position.
   - Switch the BEAT FREQUENCY OSCILLATOR to the OFF position.
   - Fully extend the antenna and set in a vertical position.

4. Switch on the power supply.

5. Investigate the AM transmitted signal on tp13 in ANACOM1/1, Record the waveform (determine the amplitude and frequency).

6. Adjust the receiver tuning control until the audio signal from ANACOM 1/1 can be clearly heard. This should occur between 55 and 65 on the tuning scale (fine tuning for the strongest possible signal may be required).

7. On the receiver board increase the RF Amplifier gain preset to its mid-point and adjust the volume preset until the tone is just audible.

We will now follow the progress of the signals through the receiver starting from the RF Amplifier. On your dual channel oscilloscope:

8. Set the oscilloscope inputs to AC coupling.

9. Use Channel 2 as the trigger channel and use it to monitor the audio input signal at tp14 (the output of the audio oscillator on ANACOM 1/1).

   A note on oscilloscope triggering: *Leave Channel 1 connected to tp14 during this practical exercise. Use Channel 1 to trigger the oscilloscope whenever an AM wave is being observed. Use Channel 1 for triggering whenever we are looking at the RF or IF component of the waveform.*

10. Use channel 2 to monitor the RF Amplifier output at tp12 on ANACOM 1/2.

11. Record the waveform (determine the amplitude and frequency), adding your own choice of voltage scale to the figure. The waveform may drift up and down slightly on the oscilloscope display. This effect is due to the action of the AGC circuit and will be investigated further at a later stage.

12. Record the waveform at the output of the local oscillator at tp40 (determine the amplitude and frequency).

    The incoming RF amplitude modulated wave is mixed with the output of the local oscillator to provide an amplitude modulated waveform at the required IF frequency. The RF carrier and its sidebands have effectively been reduced in frequency to the required IF frequency.

13. Record the waveform at the output of the Mixer at tp20 (determine the amplitude and frequency).

    The first IF amplifier provides some amplification and a narrow band pass. It only allows the wanted frequencies to pass to the second IF amplifier.

14. Record the waveform at the output of the first IF amplifier at tp24 (determine the amplitude and frequency).

15. *The final IF amplifier provides some additional amplification and more selectivity.* This can be seen by observing the waveform at tp28. Record this signal (determine the amplitude and frequency).

16. By comparing the signal amplitude of tp24 and tp28, the gain of the second IF amplifier can be calculated.
    Record your calculated value for the gain of the second IF amplifier.

17. Observe the detector output at tp31 switching the oscilloscope input to DC coupling. Record the waveform (determine the amplitude and frequency).

We can see that the sine-wave appears thicker than the original audio input signal. This is because what appears to be a sine-wave is actually an envelope containing another frequency.

18. The output signal from the detector is now passed through a low pass filter that removes all the unwanted components to leave just the audio signals.

19. This can be seen at tp39 and if the signal is compared with the original input on the transmitter (at the output of the AUDIO OSCILLATOR tp14 on ANACOM 1/1), we can see that the information signal has been passed through the transmitter and the receiver with very little distortion. Draw both the transmitted signal on tp14 in ANACOM 1/1 and the o/p signal from the low-pass filter on tp39 in on ANACOM 1/2 in same time scale. What do you note?

Note that there is a slight delay between the output and input audio signals. This is due to the circuits of the Transmitter and Receiver.

20. Switch off the power supply.

Everybody makes mistakes; that's why they put erasers on pencils

Good luck

Experiment Four
FM Modulators/Demodulators

4.1 Objectives

- Learn the main characteristics of an FM transmission system.
- Understand the operation of a Varactor Modulator.
- Understand the operation of a Reactance Modulator.
- Understand the principles of FM demodulation.

4.2 Background Information

First we will introduce the main features and layout of the ANACOM 2 board. Then describe the main characteristics of an FM transmission.

4.2.1 The ANACOM 2 Board

The main idea behind this section is to recognize the functions of the main parts of the ANACOM 2 board, and to describe the main facilities offered by the ANACOM 2 circuits. The board can be considered as five separate blocks as shown in Fig. 1 and 2: Detailed description of some the main blocks will follow.

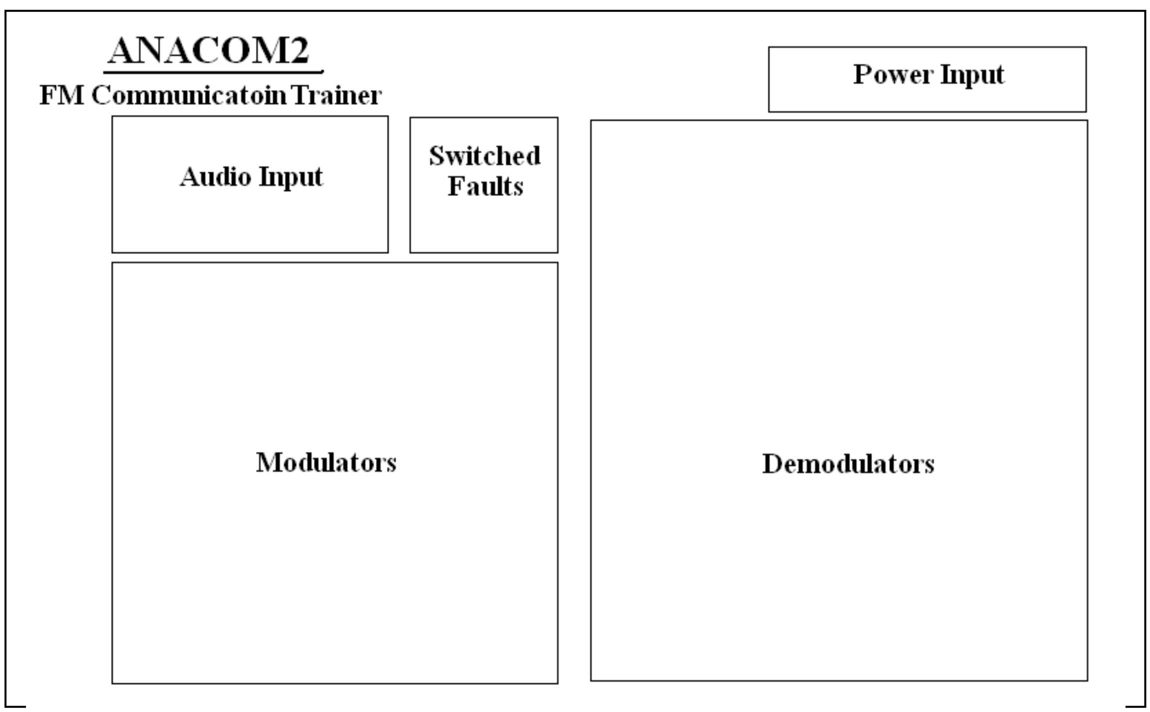


Figure 1: The ANACOM 2 Board Blocks

The Audio Oscillator: This circuit provides an internally generated signal that is going to be used as' information' to demonstrate the operation of the modulators and demodulators. There is also an external Audio Input facility to enable the supply of audio information signals.

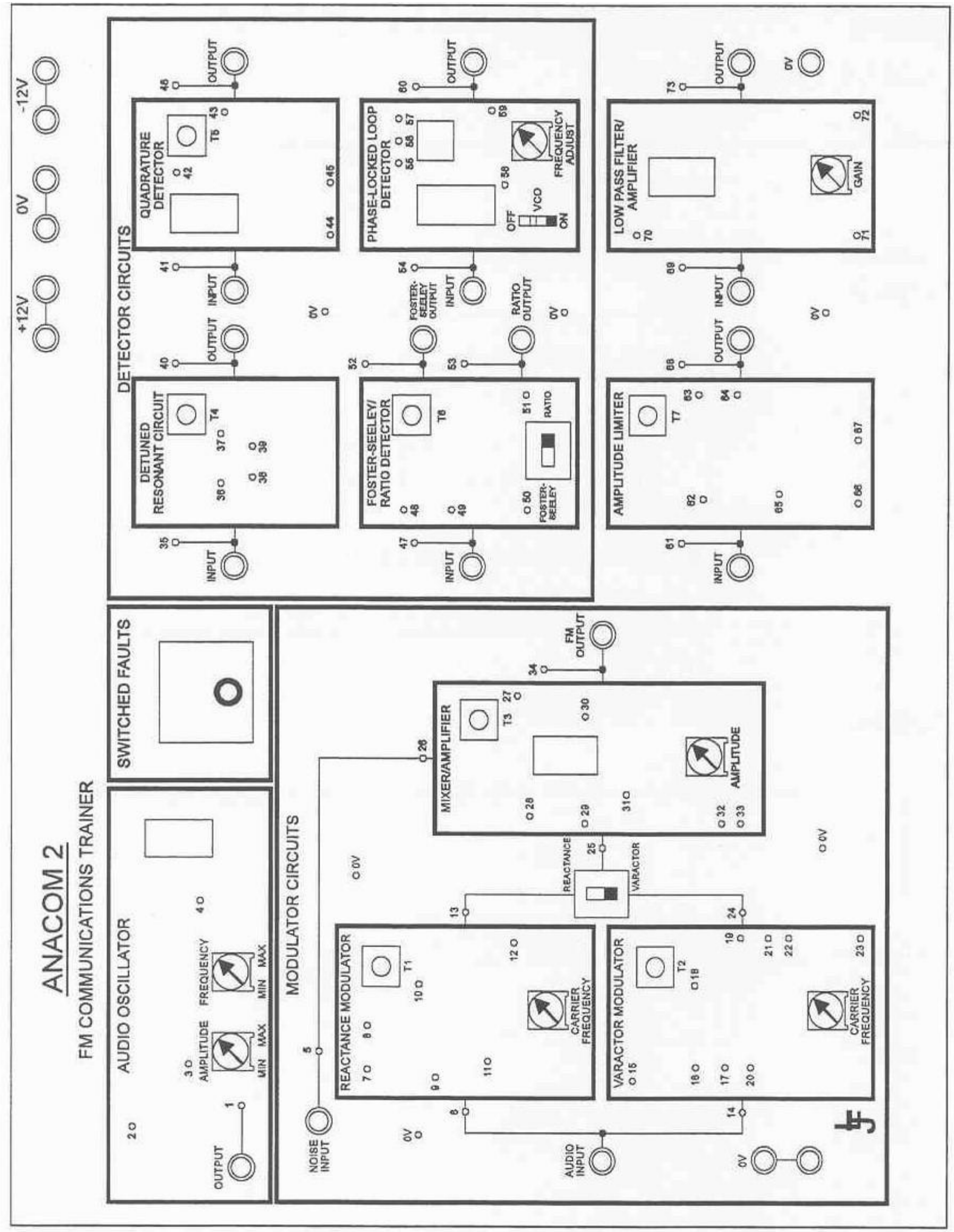


Figure 2: Layout Diagram of the ANACOM2 Board

The Modulator: This section of the board accepts the information signal and generates the final frequency modulated signal. Two different designs of modulator are provided (Reactance Modulator and Varactor Modulator).

The Demodulator: These are the detector circuits that are able to extract the incoming information signal from the FM signal which is available at the output of the modulator circuits. Four different forms of detectors are available (Detuned, Quadrature, Foster-Seeley/Ratio and Phase-Locked Loop).

Amplitude Limiter and Low Pass Filter/ Amplifier: The amplitude limiter and the low pass filter are additional circuits associated with an FM receiver whose function is to improve the quality of the output sound. They are described more fully in a later section. The Amplifier increases the output volume to a level set by the Gain preset control.

4.2.2 Frequency Modulation (FM)

One method of combining an information signal with a carrier wave was by amplitude modulation. In that case, we used the information signal to vary the amplitude of the carrier wave and then, at the receiver, these variations in the amplitude were detected and the information recovered. An alternative system is frequency modulation in which the information signal is used to control the frequency of the carrier wave. This works equally well, and in some respects, better than amplitude modulation. The frequency of the carrier is made to increase as the voltage in the information signal increases and to decrease in frequency as it reduces. The larger the amplitude of the information signal, the further the frequency of the carrier signal is shifted from its starting point. The frequency of the information signal determines how many times a second this change in frequency occurs. Notice in Figure 3 that the amplitude is not affected by the modulation process.

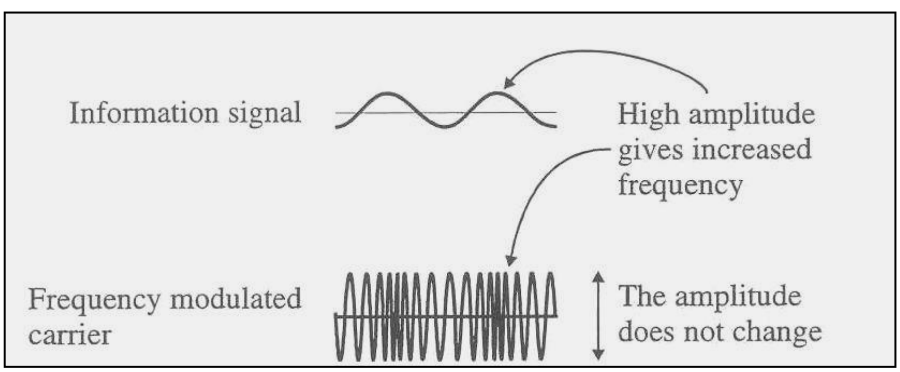


Figure 3 FM Modulation

There are three advantages of frequency modulation for a communication system. First, in an AM system, the demodulator is designed to respond to changes in amplitude of the received signal but in an FM receiver the demodulator is only watching for changes in frequency and therefore ignores any changes in amplitude. Electrical noise thus has little or no effect on an FM communication system. Second, the bandwidth of the FM signal is very wide compared with an AM transmission.

Typical broadcast bandwidths are in the order of 250 KHz. This allows a much better sound quality, so signals like music sound significantly better if frequency modulation is being used.

Lastly, when an FM demodulator is receiving an FM signal, it follows the variations in frequency of the incoming signal and is said to 'lock on' to the received transmission. This has a great advantage when two transmissions are received at the same time. The receiver 'locks on' to the stronger of the two signals and ignores the other. This is called the 'capture effect' and it means that we can listen to an FM station on a radio without interference from other stations.

FM transmission requires wide bandwidth and this is one of its disadvantages. The medium frequency broadcast band extends from about 550 kHz to 1,600 kHz, and is therefore only a little over 1 MHz in width. If we tried to use FM using a bandwidth of 250 kHz for each station, it would mean that no more than four stations could be accommodated. This wide bandwidth forces us to use higher carrier frequencies, usually in the VHF band which extends from about 85 MHz to 110 MHz. This is a width of 25 MHz and would hold many more stations.

In frequency modulation (FM), the phase angle of a carrier signal $A_c\cos(2\pi f_c t)$ is varied proportionally to the integral of the message signal x(t), the frequency modulated signal $x_c(t)$ is:

$$X_c(t) = A_c \cos\left[2\pi f_c t + k_f \int_{-\infty}^{t} x(\alpha)\, d\alpha\right] \qquad (1)$$

Where K_f is the frequency sensitivity of the FM modulator expressed in (rad/volt)

The instantaneous frequency of the FM signal is given by:

$$f_i = f_c + \frac{k_f}{2\pi} x(t) \qquad (2)$$

In the special case of tone-modulated FM, the message signal is a sinusoid:

$$\phi(t) = \frac{k_f A_x}{2\pi f_x} \sin(2\pi f_x t) \qquad (3)$$

And the modulated signal is given by:

$$x_c(t) = A_c \cos[2\pi f_c t + \beta \sin(2\pi f_x t)] \quad (4)$$

Where β is called modulation index defined as:

$$\beta = \frac{k_f A_x}{2\pi f_x} = \frac{\Delta f}{f_x} \qquad (5)$$

And represents the maximum phase deviation produced by the message signal.
The parameter Δf represents the peak frequency deviation of the modulated signal.
The modulation index is defined only for tone-modulation.
The calculation of the spectrum of an FM signal is difficult since frequency modulation is a nonlinear process. However, for the special case of tone-modulation an equivalent expression for Eq(4) is:

$$x_c(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta)\cos\left[2\pi(f_c + nf_x)t\right] \qquad (6)$$

This equation describes the modulated signal as a series of sinusoids whose coefficients $J_n(\beta)$ are given by the Bessel function of the first kind and nth order.

Bandwidth of an FM Signal
The bandwidth of the tone-modulated FM signal can be estimated by the Carlson's rule, and it depends on the modulation index and the freauency of the modulation tone:

$$\beta_{FM} = 2(\beta + 1) f_x \qquad (7)$$

(7)

From this equation it can be noticed that a small value of β(<<1) will result in a bandwidth of about $2f_x$.
Modulated signals with this condition are referred to a *narrowband* FM signals. Large values of the modulation index will produce signals with relatively large bandwidth, or wideband FM signals. The frequency modulation process generates a large number of side frequencies. Theoretically, the sidebands are infinitely wide with the power levels becoming lower and lower as we move away from the carrier frequency.
The bandwidth of 250 kHz was chosen as a convenient value to ensure a low value of distortion in the received signal whilst allowing many stations to be accommodated in the VHF broadcast band. Communication signals which do not require the high quality associated with broadcast stations can adopt a narrower bandwidth to enable more transmissions within their allotted frequency band. Marine communications for ship to ship communications, for example, use a bandwidth of only 25 kHz but this is only for speech and the quality is not important.

An FM Transmitter
As we can see from Figure4, the FM transmitter is very similar to the AM transmitter. The audio oscillator supplies the information signal and could, if we wish, be replaced by a microphone and AF amplifier to provide speech and music instead of the sine wave signals.

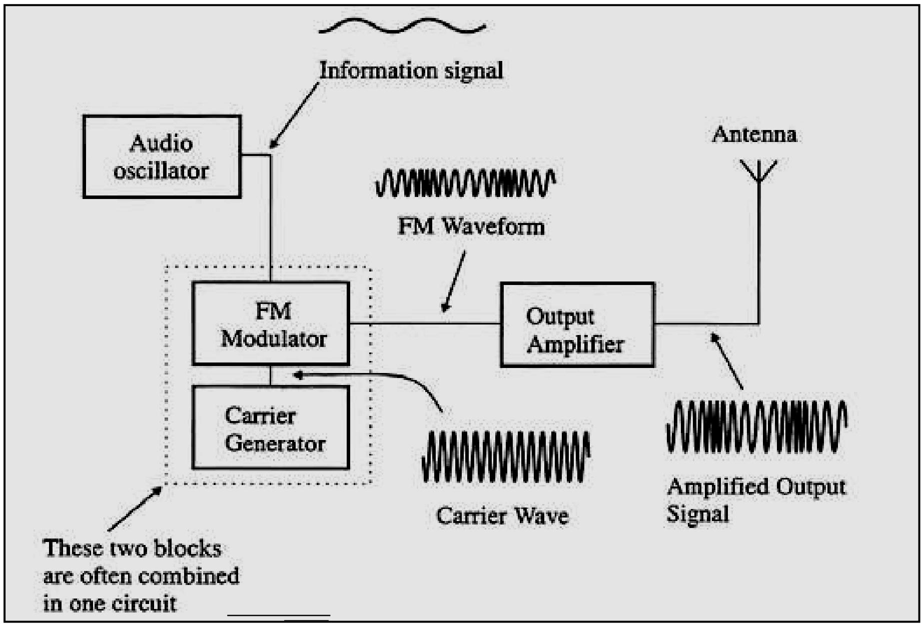


Figure 4: An FM Transmitter

The FM modulator is used to combine the carrier wave and the information signal in much the same way as in the AM transmitter. The only difference in this case is that the generation of the carrier wave and the modulation process is carried out in the same block.
The output amplifier increases the power in the signal before it is applied to the antenna for transmission just as it did in the corresponding block in the AM transmitter.
The only real difference between the AM and FM transmitters are the modulators, so we are only going to consider this part of the transmitter. We are going to investigate two types of modulator; they are called the VARACTOR MODULATOR and the REACTANCE MODULATOR.

How Do These Modulators Work?

The basic idea is quite simple and both modulators function in much the same way. They both include an RF oscillator to generate the carrier and these oscillators employ a parallel tuned circuit to determine the frequency of operation.
Adding an additional capacitor in parallel will cause the total capacitance to increase and this will result in a decrease in the resonance frequency. If you feel that a reminder of the formula may be helpful, the approximate frequency of resonance is given by:

$$f_O = \frac{1}{2\pi\sqrt{LC}} \qquad (8)$$

Where L is the inductance in Henrys and C is the capacitance in Farads.
The tuned circuit is part of the oscillator used to generate the carrier frequency so, if the capacitance changes, then the carrier frequency will change also. This is demonstrated in Figure 5 below:

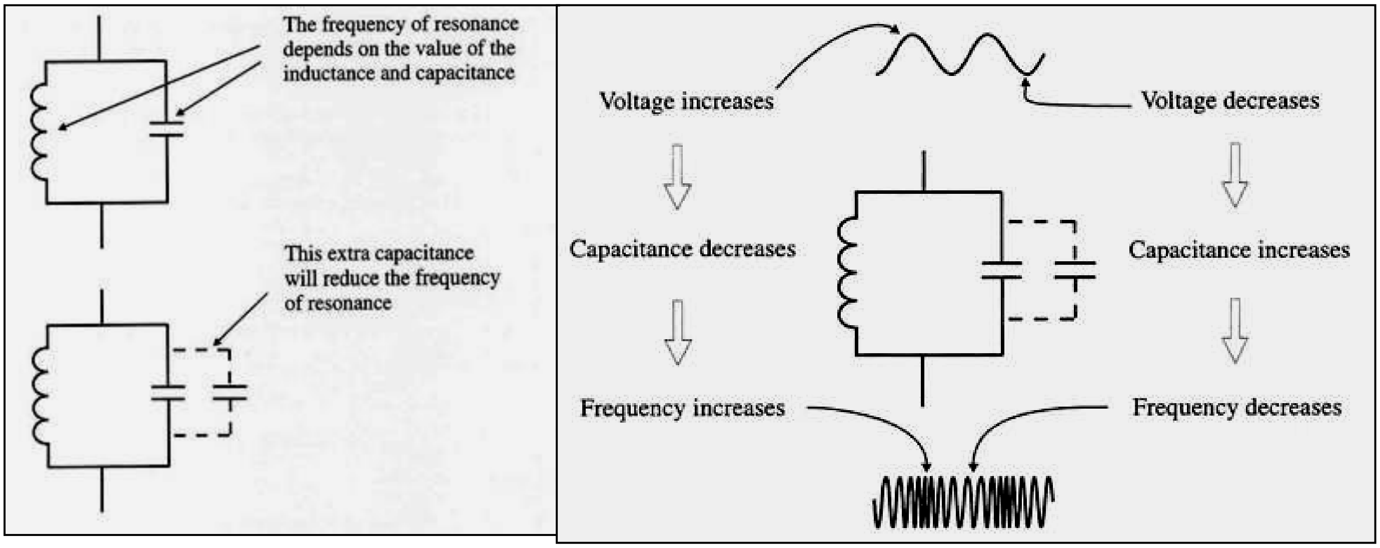


Figure 5: The Tuned Circuit Operation

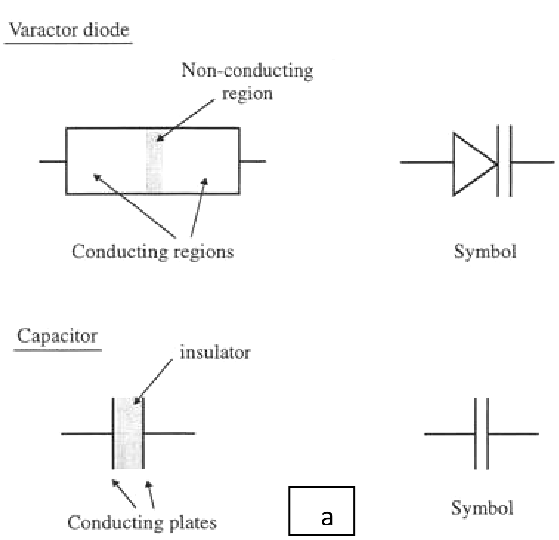
To produce a frequency modulated carrier, all we have to do is to find a way of making the information signal increase and decrease the size of the capacitance and hence control the carrier frequency.

In the following sections we will look to see two ways of achieving this. First by using a device called a Varactor Diode and then by using a transistor.

4.2.3 The Varactor Diode

The Varactor diode is a semiconductor diode that is designed to behave as a voltage controlled capacitor. When a semiconductor diode is reverse biased no current flows and it consists of two conducting regions separated by a non-conducting region. This is very similar to the construction of a capacitor (see Figure 6a). By increasing the reverse biased voltage, the width of the insulating region can be increased and hence the capacitance value decreased.

This is shown in Figure 6b. If the information signal is applied to the Varactor diode, the capacitance will therefore be increased and decreased in sympathy with the incoming signal.



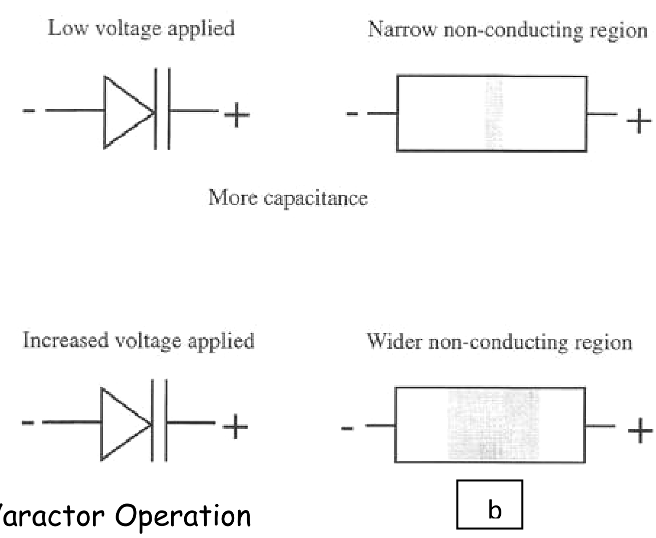


Figure 6 Varactor Operation

The Varactor Modulator Circuit

The variations in the capacitance form part of the tuned circuit that is used to generate the FM signal to be transmitted. Have a look at the Varactor modulator shown in Fig. 7.

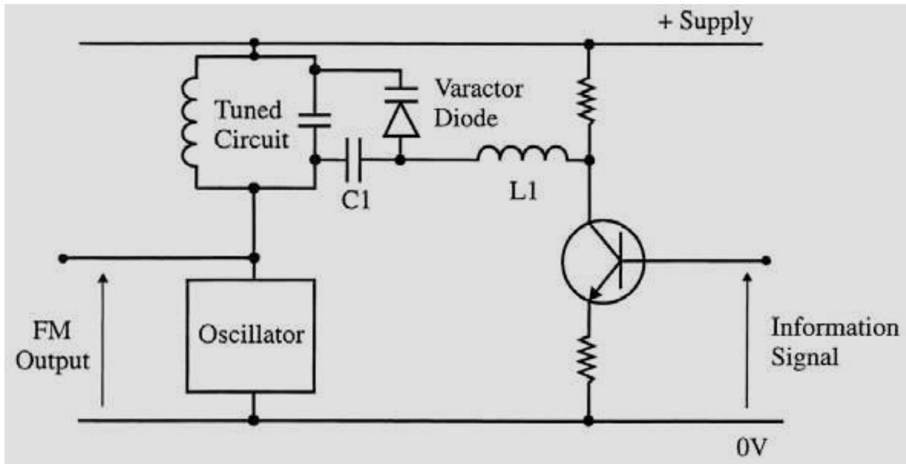


Figure 7: The Varactor Modulator Circuit

We can see the tuned circuit which sets the operating frequency of the oscillator and the Varactor which is effectively in parallel with the tuned circuit. Two other components which may not be immediately obvious are Cl and Ll. Cl is a DC blocking capacitor to provide DC isolation between the Oscillator and the collector of the transmitter. Ll is an RF choke which allows the information signal through to the Varactor but blocks the RF signals.

The Operation of the Varactor Modulator

1) The information signal is applied to the base of the input transistor and appears amplified and inverted at the collector.

2) This low frequency signal passes through the RF choke and is applied across the Varactor diode.

3) The Varactor diode changes its capacitance in sympathy with the information signal and therefore changes the total value of the capacitance in the tuned circuit.

4) The changing value of capacitance causes the oscillator frequency to increase and decrease under the control of the information signal. The output is therefore an FM signal.

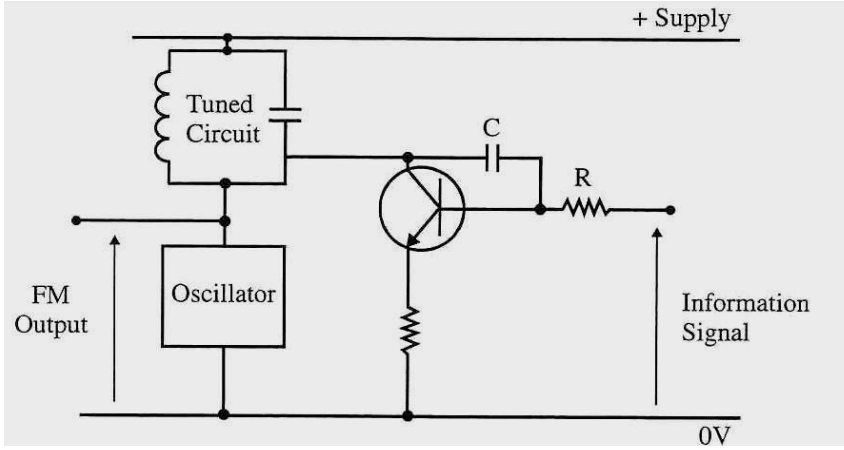


Figure 8: The Reactance Modulator Circuit

4.2.4 Using a Transistor as a Capacitor

In this section we will discover how we can persuade a transistor to behave like a capacitor. From previous work, we remember that when a capacitor is connected in series with a resistor, an alternating current flowing through the circuit will be out of phase with the voltage across the capacitor. The current will lead the voltage across the capacitor by 90° and will be in phase with the voltage across the resistor.

To make the transistor appear to be a capacitor, all we have to do is to find a way of making it generate a current that is leading an applied voltage. If it does this then it is behaving like a capacitor which is all we want. To achieve this effect, we connect a very small capacitor and a resistor in series between the collector and the input to the transistor labeled 'C' in Figure 8.
Now, if we use the voltage across the resistor as the input to the base of a transistor, the resulting collector current will be in phase with the base voltage and will lead the collector voltage by 90° just like a real capacitor.
The result is that the transistor now appears to be a capacitor.

Making the Capacitor Variable

The size of the capacitance depends on the change in collector current which occurs for a given change in the base voltage. This ratio, called the 'transconductance', is a measure of the amplification of the transistor and can be controlled by the DC bias voltage applied to the transistor. The larger the bias, the larger the value of the transconductance and the larger the capacitance produced.

The Reactance Modulator Circuit

Figure 5.8 shows a complete reactance modulator. We can see that the left-hand half is the same as in the previous
Varactor modulator is simply an oscillator and a tuned circuit, which between them generates the un-modulated carrier. The capacitor 'C' and the resistor 'R' are the two components used for the phase shifting, together with the transistor, form the voltage controlled capacitor.

Operation of the Reactance Modulator

If required, reference can be made to Figure 5.8.

1) The oscillator and tuned circuit provide the un-modulated carrier frequency and this frequency is present on the collector of the transistor.

2) The capacitor and the resistor provide the 90° phase shift between the collector voltage and current as described earlier. This makes the circuit appear as a capacitor.

3) The changing information signal being applied to the base has the same effect as changing the bias voltage applied to the transistor and, this would have the effect of increasing and decreasing the value of this capacitance.

4) As the capacitance is effectively in parallel with the tuned circuit the variations in value will cause the frequency of resonance to change and hence the carrier frequency will be varied in sympathy with the information signal input.

4.3 Demodulation of FM signals

An FM receiver is very similar to an AM receiver. The most significant change is that the demodulator must now extract the information signal from a frequency, rather than amplitude, modulated wave.

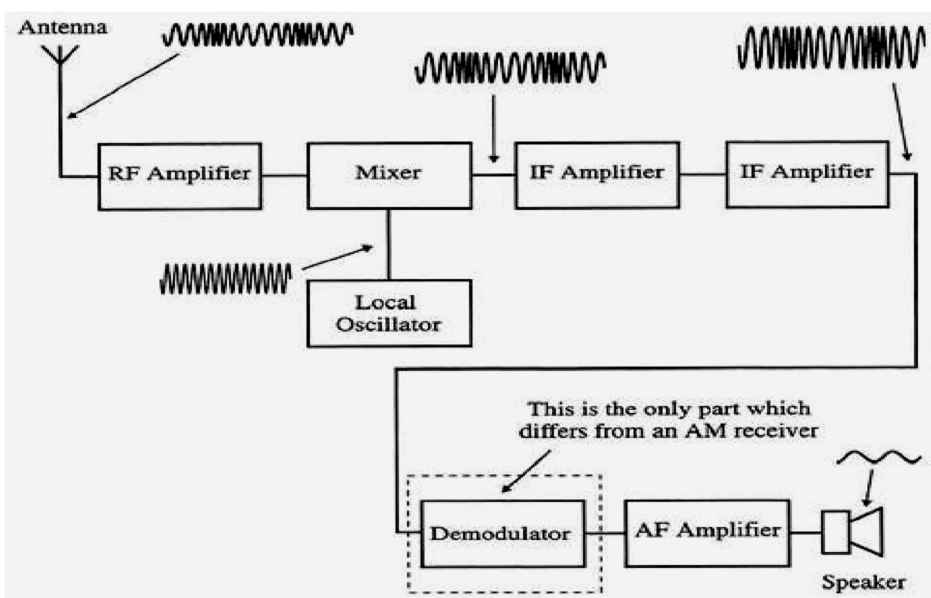


Figure 9: FM Receiver

The basic requirement of any FM demodulator is therefore to convert frequency changes into changes in voltage, with the minimum amount of distortion. To achieve this, it should ideally have a linear voltage/frequency characteristic, similar to that shown in Figure10.

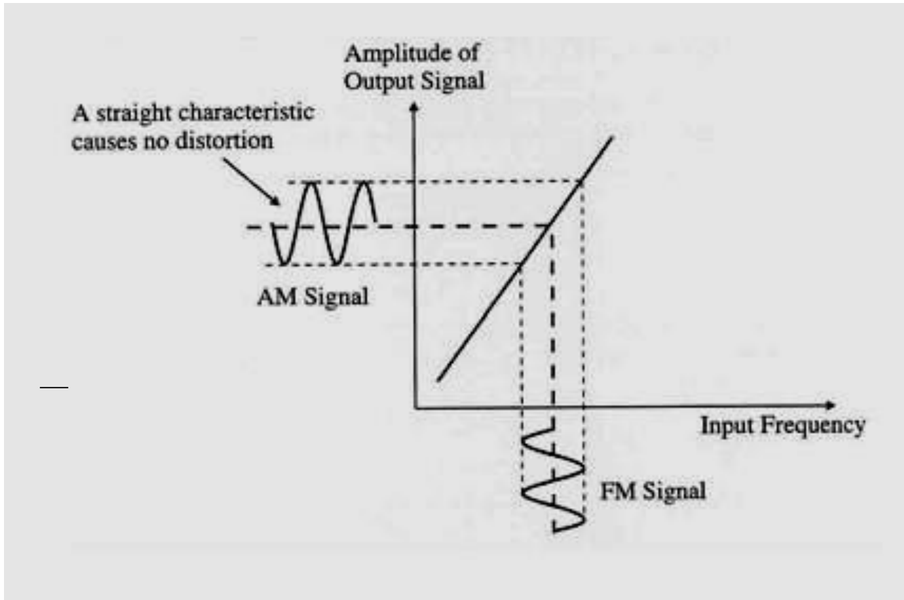


Figure 10: An 'ideal' Linear Voltage / Frequency Characteristic

The information in an FM signal resides in the instantaneous frequency $\omega_i = \omega_c + k_f\, m(t)$. Hence, a frequency-selective network with a transfer function of the form $|H(\omega)| = a\,\omega + b$ over the FM band would yield an output proportional to the instantaneous frequency. There are several possible networks with such characteristics. The simplest among them is an ideal differentiator with the transfer function $j\omega$. Figure 11 shows differentiator followed by an envelope detector is called a frequency discriminator.

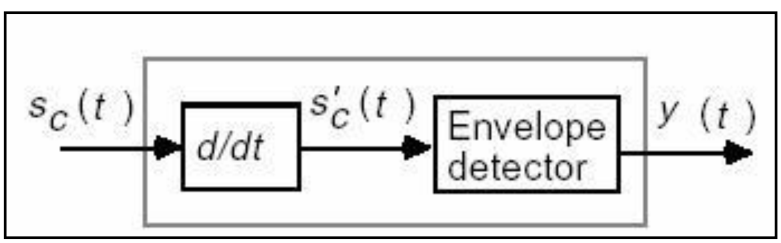


Fig.11 Frequency Discriminator

Let $S_c(t)$ be the FM signal, then:

$$S_c(t) = \frac{d}{dt}\left\{A\cos\left[\omega_c t + k_f \int_{-\infty}^{t} m(\alpha)d\alpha\right]\right\} = A\left[\omega_c + k_f m(t)\right]\sin\left[\omega_c t + k_f \int_{-\infty}^{t} m(\alpha)d\alpha\right]$$

Since $\omega_c + k_f\, m(t) > 0$ for all t, m(t) can be obtained by envelope detection as in AM demodulation.

In practice, channel noise and other factors may cause *A* to vary. If *A* varies, y(t) will vary with *A*. Hence, it is essential to maintain the amplitude of the input signal to the frequency discriminator using amplitude limiter.

A 'demodulator' can also be called a 'discriminator' or a 'detector'. Any design of circuit that has a linear Voltage/frequency characteristic would be acceptable and we are going to consider the three most popular types.

In each case the main points to look for are:

- How do they convert FM signals into AM signals?
- How linear is their response - this determines the amount of distortion in the final output?
- How good are they at rejecting noise signals?

4.3.1 Detuned Resonant Circuit Detector

This is the simplest form of demodulator. It works - but it does have a few drawbacks. A parallel tuned circuit is deliberately detuned so that the incoming carrier occurs approximately halfway up the left-hand slope of the response.

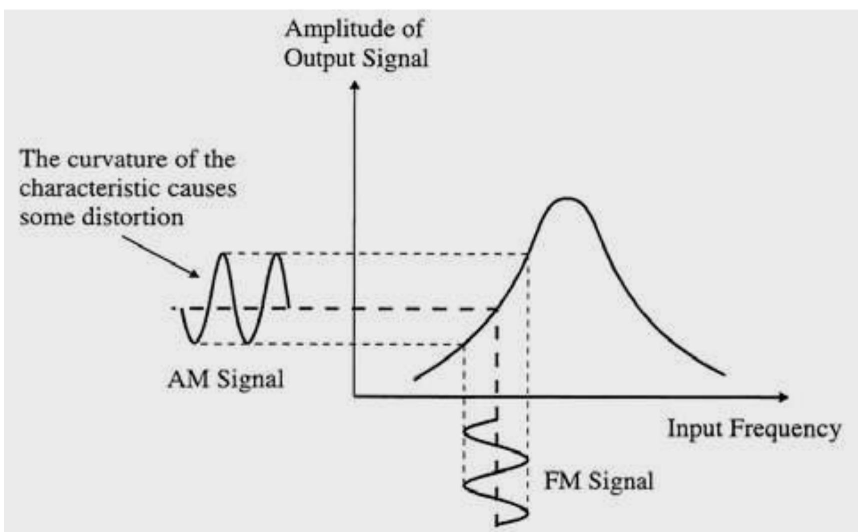


Figure 12: Detuned Resonant Circuit Detector Operation

In Figure 12 above, we can see that the amplitude of the output signal will increase and decrease as the input frequency changes. For example, if the frequency of the incoming signal were to increase, the operating point would move towards the right on the diagram. This would cause an increase in the amplitude of the output signal.

An FM signal will therefore result in an amplitude modulated signal at the output - it is really that simple!

Figure 13 below shows the circuit diagram of the Detuned Resonant Circuit Detector. If we break it down, the operation becomes very clear. The FM input is applied to the base of the transistor and in the collector there is the detuned resonant circuit that we have met earlier. In reality, it also includes the loading effect caused by the other winding which acts as a transformer secondary.

The signal at the collector of the transistor includes an amplitude modulated component which is passed to the diode detector. The diode detector was discussed but a brief summary is included here as a reminder.

In the diagram, the diode conducts every time the input signal applied to its anode is more positive than the voltage on the top plate of the capacitor. When the voltage falls below the capacitor voltage, the diode ceases to conduct and the voltage across the capacitor leaks away until the next time the input signal is able to switch it on again.

The output is passed to the Low Pass Filter/Amplifier block. The unwanted DC component is removed and the low-pass filter removes the ripple at the IF frequency. One disadvantage is that any noise spikes included in the incoming signal will also be passed through the diode detector and appear at the output. If we are going to avoid this problem, we must remove the AM noise before the input to the demodulator. We do this with an Amplitude Limiter circuit.

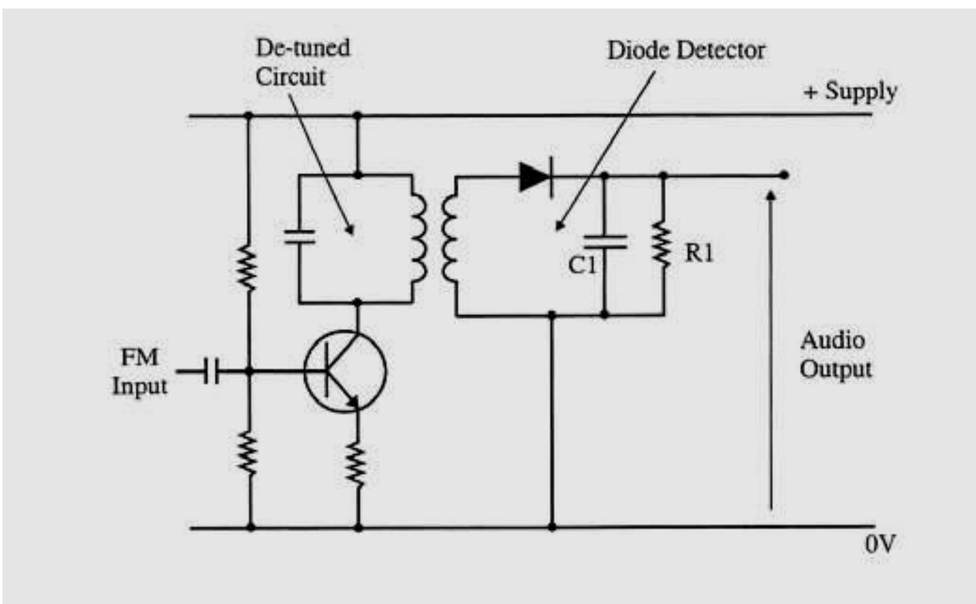


Figure 13: Detuned Resonant Circuit Detector

The Amplitude Limiter

An amplitude limiter circuit is able to place an upper and lower limit on the size of a signal. In Figure14, the preset limits are shown by dotted lines. Any signal which exceeds these levels are simply chopped off. This makes it very easy to remove any unwanted amplitude modulation due to noise or interference.

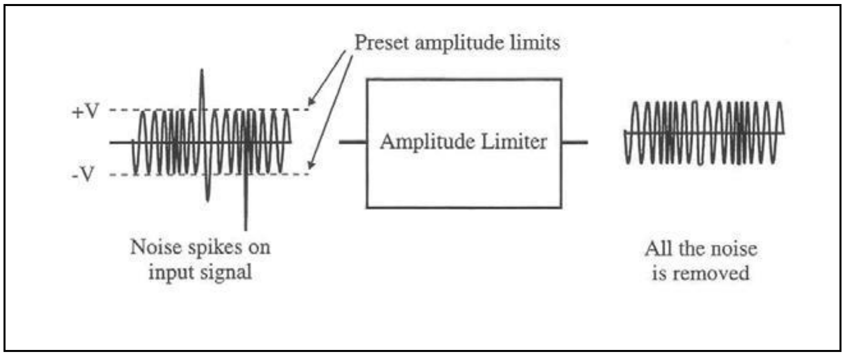


Figure 14: The Amplitude Limiter

4.4 Procedure

Part 1: The Varactor Modulator

1. Connect the ANACOM 2 board to the power supply.

2. On the ANACOM 2 board, set the following initial conditions:
   - AMPLITUDE and FREQUENCY presets in the Audio Oscillator block set to maximum (fully clockwise).
   - Set the Carrier Frequency preset in the Varactor Modulator to minimum (fully counter-clockwise)
3. Switch on the power supply.

4. Use your oscilloscope to measure the frequency and the peak-to-peak Amplitude of the output signal of the Audio Oscillator block at tp1.

5. Connect the output of the Audio Oscillator to the Audio Input socket of the Modulator Circuits block (see Figure15).

6. Set the Reactance/ Varactor switch to the Varactor position.

This switch selects the Varactor modulator and also disables the Reactance modulator to prevent any interference between the two circuits.

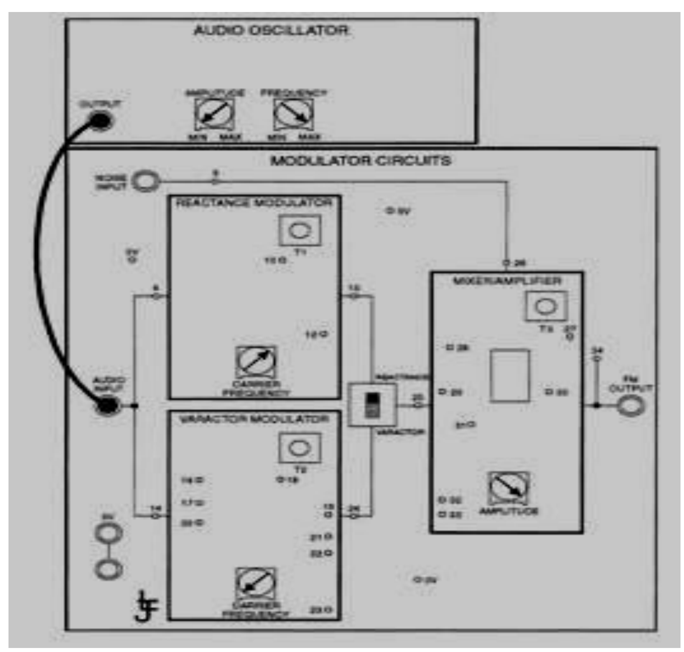


Figure15: Step Number 5

The FM signal at the output of the Varactor modulator has been amplified by the Mixer/Amplifier and appears at its output on tp34.

7. Set the Amplitude preset in the Mixer/Amplifier to maximum (fully clockwise).

8. Use your oscilloscope to measure the frequency and the peak-to-peak amplitude of the output signal at tp34.

9. Use X-expansion control on your oscilloscope, use this to 'expand up' the right hand cycles on the display for a closer look at how frequency modulation affects their appearance.

We can get an indication of the relationship between the input voltage and the carrier frequency by measuring the value of the change in the DC voltage applied to the base of the transistor in the Varactor Modulator circuit and recording the corresponding value of the output frequency at tp34.

10. Record output frequency measured at tp34 and the DC input voltage measured at tp21. Take 10 readings as the carrier frequency preset is increased to enable a graph of DC voltage/frequency output to the plotted. The result would be similar to that shown in Figure 16.

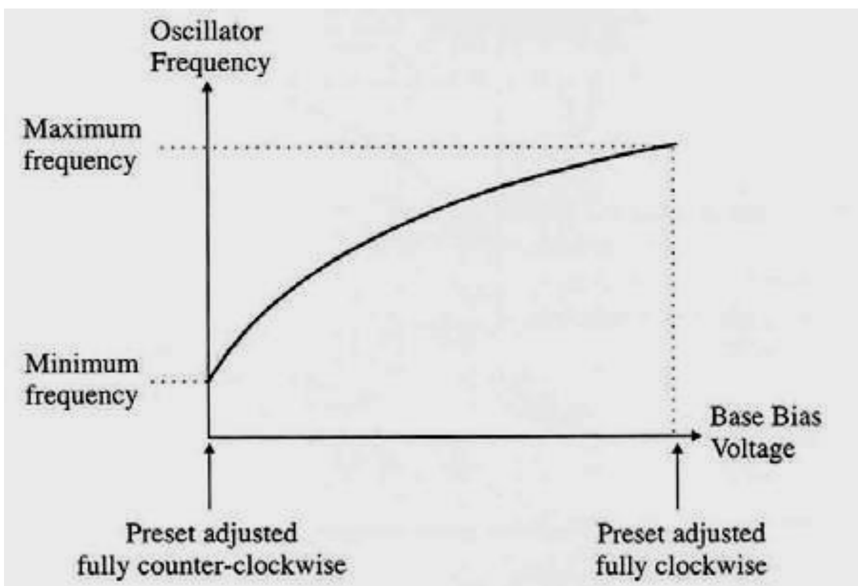


Figure16: A graph of DC Voltage / Frequency Output for a Varactor Modulator

11. Switch off the power supply.

Part 2: The Reactance Modulator

1. Set the Reactance/ Varactor switch to the Reactance position

2. Set the Carrier Frequency preset in the Reactance Modulator to its fully clockwise position.

The FM signal at the output of the Reactance modulator has been amplified by the Mixer/Amplifier and appears at its output on tp34.

3. Monitor the output waveform of the Mixer/Amplifier at tp34.
   Note that the carrier is being frequency modulated at the moment.

4. If you have an X-expansion control on your oscilloscope, use this to 'expand up' the right hand cycles on the display for a closer look at how frequency modulation affects their appearance.

We can get an indication of the relationship between the input voltage and the carrier frequency by measuring the value of the change in the DC voltage applied to the base of the transistor in the Reactance Modulator circuit and recording the corresponding value of the output frequency at tp34. It may be helpful at this time to have another look at Figure 13 or refer back to your investigation of the Varactor Modulator.

We are going to use the Carrier Frequency preset to vary the input voltage to the base of the modulating transistor. To avoid loading due to the oscilloscope, we are again going to use the amplified output at tp34 for the frequency measurement.

5. Turn the Carrier Frequency preset to its fully counter-clockwise position.

6. Again record the oscillator output frequency measured at tp34 and the DC input voltage measured at tp 11.Take 10 readings as the preset is adjusted to enable a graph of DC voltage/frequency output to the plotted. The result would be similar to that shown in Figure17:

7. Switch off the power supply.

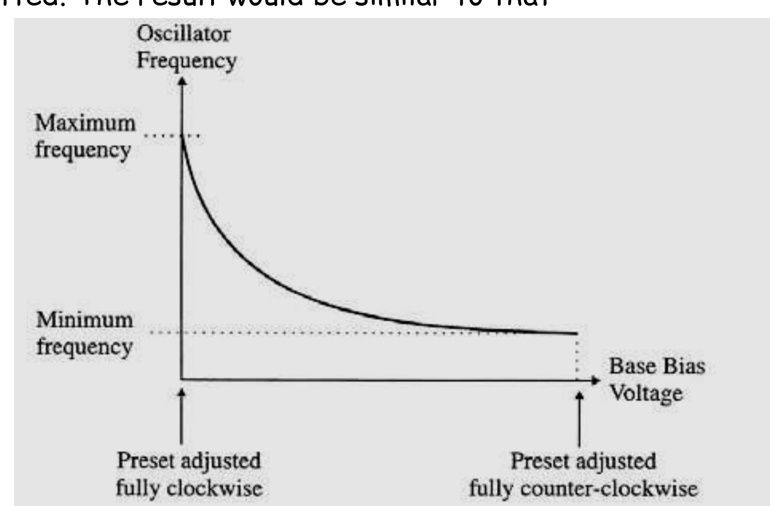


Figure 17: A group of DC Voltage/ Frequency Output for a Reactance Modulator

Part3: The Detuned Resonant Circuit Detector

1. Ensure that the following initial conditions exist on the ANACOM 2 board:
   - All the switched faults set to OFF.
   - FREQUENCY preset in the Audio Oscillator block set to minimum (fully counter-clockwise).
   - AMPLITUDE preset in the Audio Oscillator block set to maximum (fully clockwise).
   - AMPLITUDE preset in the Mixer! Amplifier block set to maximum (fully clockwise).
   - VCO switch in the Phase-Locked Loop Detector block set to OFF.

3. Switch on the power supply.

4. Set up a signal generator to have a sinusoidal output of 1 volt peak-to-peak and a frequency of approximately 455 kHz, and then connect it to the input socket of the Detuned Resonant Circuit (tp35).

5. Record the DC voltage at tp40 for incoming frequencies from 25 kHz below to 25 kHz above the resonance frequency of the tuned circuit. Take 10 readings each step increase frequency 5 KHz over the range from 430 KHz – 480 KHz.

6. Plot the DC voltage at tp40 against frequency and check that the response is similar to that shown in Figure12.
7. Disconnect the signal generator.

We will now use the Detuned Resonant Circuit Detector as part of a complete FM system.

9. Make the connections shown in Figure18.

10. Select the Varactor Modulator by sliding the Reactance/Varactor switch downwards. We are using the Varactor Modulator because it is ***the more linear*** of the two modulators that we have available.

11. Ensure that the Varactor Modulator's Carrier Frequency preset is in the midway position (arrowhead pointing to the top of the PCB) before continuing.

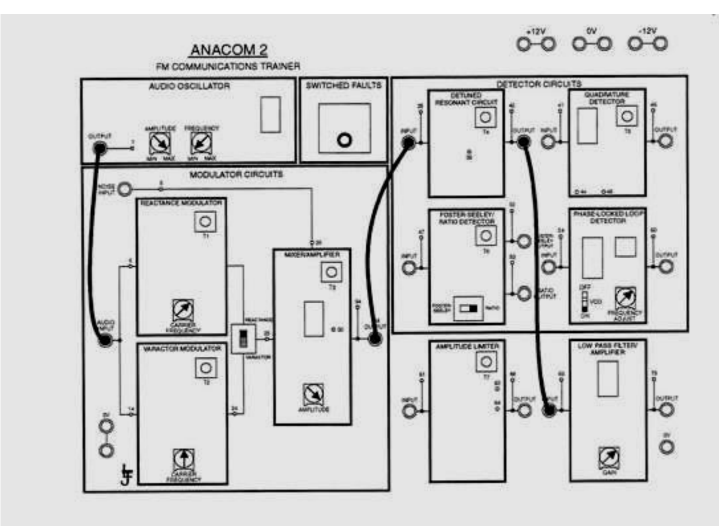


Figure 18: Detuned Resonant Circuit Detector Connections

12. Now monitor the audio input at tp14 and the output from the Detuned Resonant Circuit block at tp40, using tp14 to trigger the oscilloscope.

13. The high frequency ripple is now removed by passing the signal through the Low Pass Filter/Amplifier block.

14. Monitor the input tp69 and the output tp73 of the Low Pass Filter/Amplifier block (trigger by tp73) and note how the low pass filter has improved the quality of the output signal.

- We will now investigate the effect of noise on the system.

15. Connect signal generator to the Noise Input (tp5) and adjust it to provide a sinusoidal output of frequency 2 kHz and amplitude of 100m V peak -to- peak. This is going to be our 'noise' input.

16. Use your oscilloscope to monitor the noise input at tp5 and the FM output at tp34 (use tp5 for triggering the oscilloscope).

Note that the FM signal is being amplitude modulated by the noise signal and also being frequency modulated by the audio input from the Audio Oscillator block. This is a similar effect that would occur if an FM transmission were to suffer from noise degradation over the transmission route. We will now investigate how well the detector deals with this noise.

17. Observe the audio modulating signal at tp14 (trigger on this signal) and the output from the Detuned Resonant Circuit Detector at tp40. Also observe the input tp69 and the output tp73 of the Low Pass Filter/Amplifier block.

18. Connect the Amplitude Limiter block between the Mixer/Amplifier and the Detuned Resonant Circuit Detector blocks as shown in Figure 19 below.

As you will recall, the amplitude limiter removes amplitude variations from the FM output signal from the modulator so that the input to the detuned resonant detector has constant amplitude.

19. Use your oscilloscope to observe the waveform at the output of the Low Pass Filter/Amplifier block at tp73. Note that the amplitude of any remaining 'noise' component is now minimal.

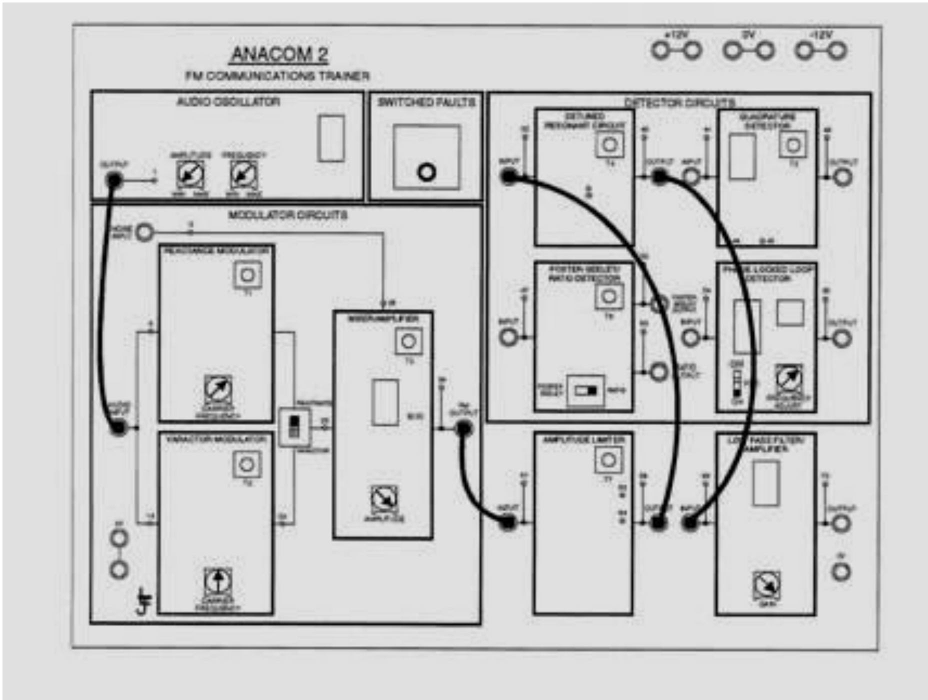


Figure 19: the Connection of Amplitude Limiter

19. Disconnect the signal generator and turn power off.

The key to evil is one word

Good luck

Appendix B
Another FM Demodulation Methods

1. The Quadrature Detector

This is another demodulator, again fairly simple but is an improvement over the previous design. It causes less distortion and is also better, though not perfect, when it comes to removing any superimposed noise. The incoming signal is passed through a phase-shifting circuit. The degree of phase shift that occurs is determined by the exact frequency of the signal at any particular instant.

The rules are:

1) If the carrier is un-modulated, the phase shift is 90°.
2) If the carrier increases in frequency, the phase shift is GREATER THAN 90°.
3) If the carrier decreases in frequency, the phase shift is LESS THAN 90°.

We now only require a circuit able to detect the changes in the phase of the signal. This is achieved by a phase comparator circuit as shown in Figure1. This circuit compares the phase of the original input signal with the output of the phase shifting circuit. It then produces a DC voltage level which depends on the result of the comparison according to the following rules:

1) It provides no change in output voltage if the signal phase has been shifted by 90°.
2) Phases over 90° result in an increased DC voltage level.
3) Phases less than 90° result in a decreased DC voltage level.

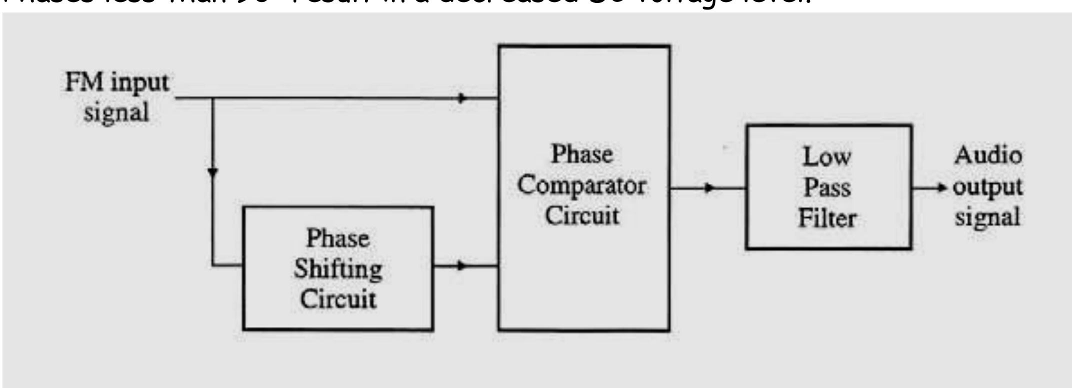


Figure 1: Quadrature Detector

As the phase changes, the DC voltage level moves up and down and re-creates the audio signal. A low pass filter is included to reduce the amplitude of any high-frequency ripple and also blocks the DC offset. Consequently, the signal at the output closely resembles the original input signal. The characteristic as shown in Figure 2 is straight enough to cause very little distortion to the final audio output.

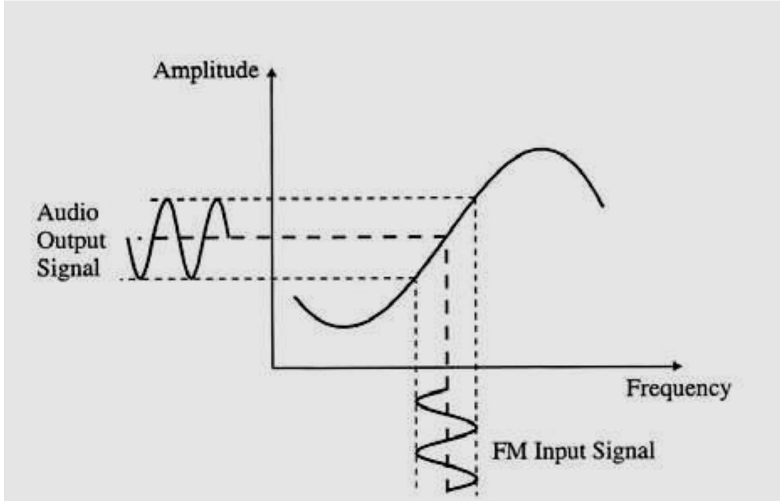


Figure 2: Quadrature Detector Characteristic

2. The Phase-Locked Loop (PLL) Detector

This is another demodulator that employs a phase comparator circuit. It is a very good demodulator and has the advantage that it is available as a self-contained integrated circuit so there is no setting up required - you plug it in and it works. For these reasons it is often used in commercial broadcast receivers. It has very low levels of distortion and is almost immune from external noise signals and provides very low levels of distortion. The PLL detector characteristic can be described by Fig.10. The overall action of the circuit may, at first, seem rather pointless. As we can see in Figure 3 there is a voltage controlled oscillator (VCO). The frequency of this oscillator is controlled by the DC output voltage from the output of the low pass filter. Now, this DC voltage keeps the oscillator running at the same frequency as the original input signal and 90° out of phase.

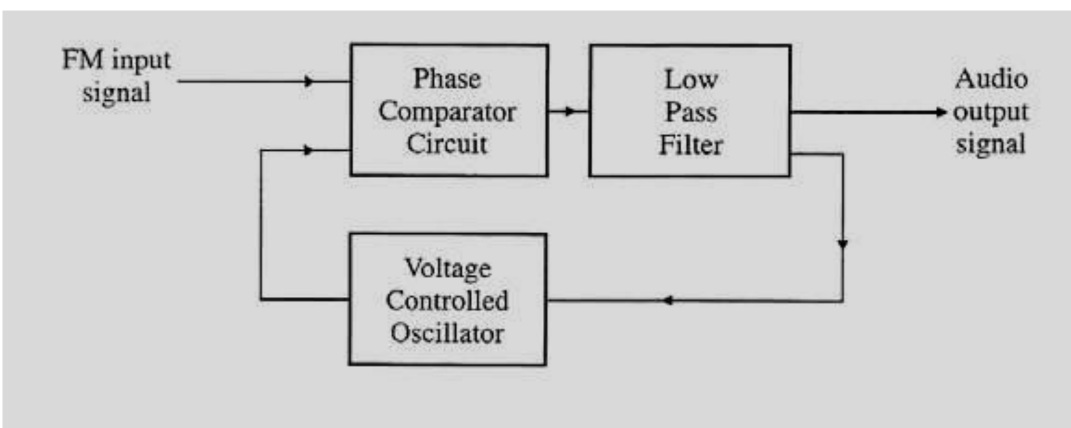


Figure 3: Block Diagram of the Phase-Locked Loop
(PLL) Detector

The question often arises as to why we would want the oscillator to run at the same frequency and 90° out of phase. And if we did, then why not just add a phase shifting circuit at the input to give the 90° phase shift? The answer can be seen by imagining what happens when the input frequency changes - as it would with an FM signal. If the input frequency increases and decreases, the VCO frequency is made to follow it. To do this, the input control voltage must increase and decrease. It is these changes of DC voltage level that form the demodulated signal.
The AM signal then passes through a signal buffer to prevent any loading effects from disturbing the VCO and then through an audio amplifier if necessary.

Controlling the VCO

To see how the VCO is actually controlled, let us assume that it is running at the same frequency as an un-modulated input signal. The waveforms are given in Figure 4 below:

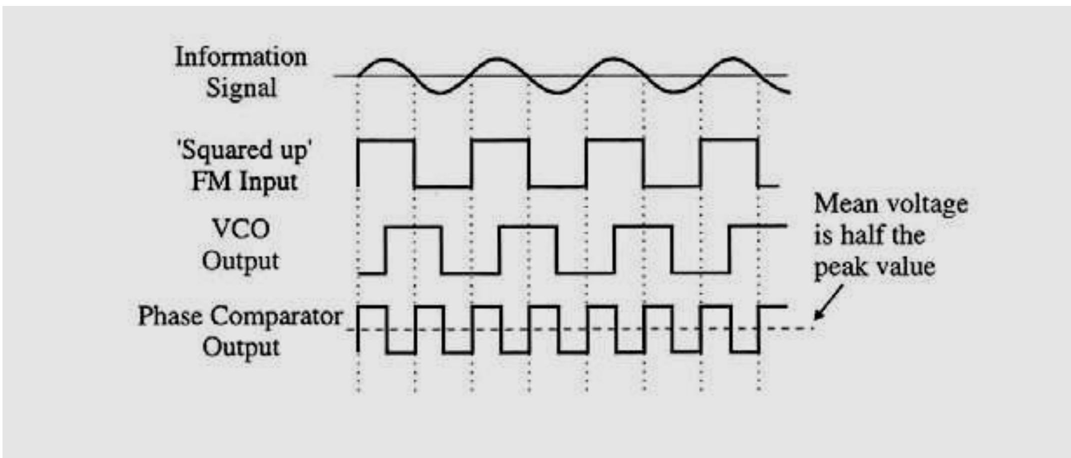


Figure 4: Controlling the VCO

The input signal is converted into a square wave and, together with the VCO output, forms the two inputs to an Exclusive-OR gate. Remember that the Exclusive-OR gate provides an output whenever the two inputs are different in value and zero output whenever they are the same.

Figure 4 shows the situation when the FM input is at its un-modulated carrier frequency and the VCO output is at the same frequency and 90° out of phase. This provides an output from the Exclusive-OR gate with an on-off ratio of unity and an average voltage at the output of half of the peak value.

Now let us assume that the FM signal at the input decreases in frequency (see Figure5). The period of the 'squared up' FM signal increases and the mean voltage level from the Exclusive-OR gate decreases. The mean voltage level is both the demodulated output and the control voltage for the VCO. The VCO frequency will decrease until its frequency matches the incoming FM signal.

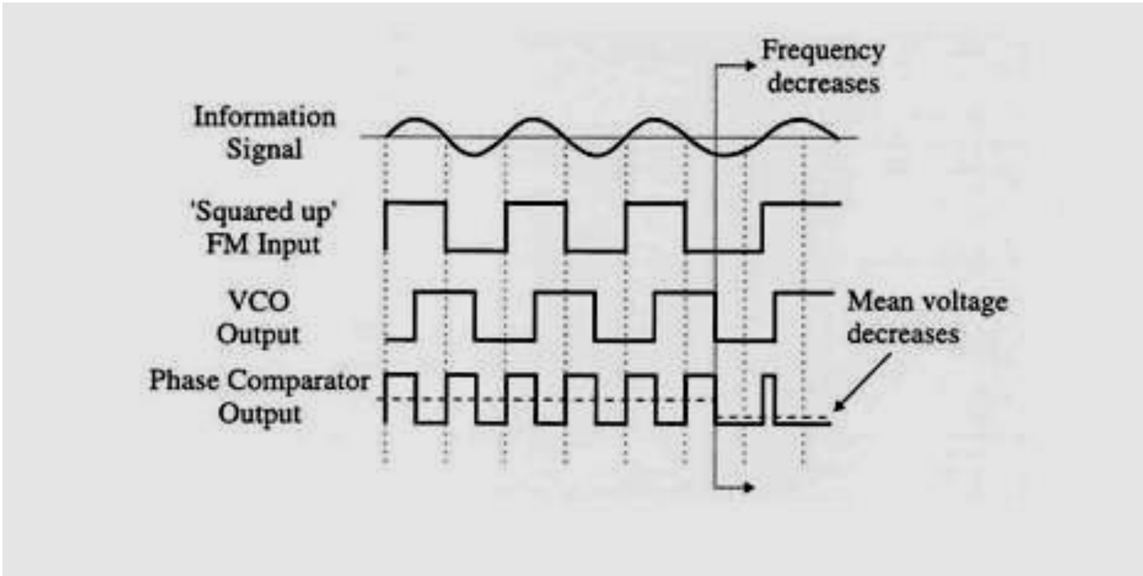


Figure 5: Controlling the VCO (2)

Experiment5

Sampling and Reconstruction

5.1 Objective

- Learn how an analog signal can be sampled, transmitted as a series of samples, and reconstructed at the receiver by low pass filtering.
- Examine the properties of sampled analog signals.

5.2 Introduction

There is an increase use of digital communication systems since digital communications offer several important advantages compared to analog communications such as higher performance, higher security and greater flexibility. To transmit analog messages such as voice and video signals by digital means, the signal has to be converted to a digital signal. This process is known as analog to digital conversion.

In the transmitter, the message or the information m (t) is first sampled, quantized, encoded. In the receiver, the signal is decoded and reconstructed using low pass filter (figure1). This experiment will focus on sampling.

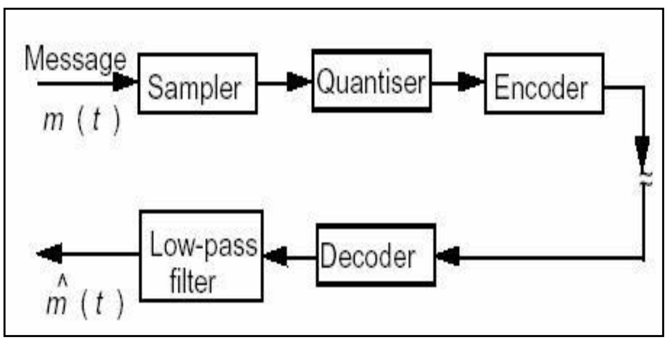


Figure 1: Simple Digital system

The basic sampling process is illustrated in figure2.

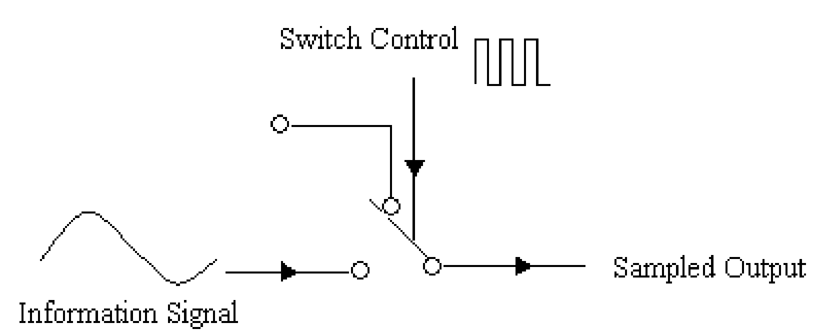


Figure2

A message is sent via switch which operates under control of a signal which is electrically isolated from the message. Consider the switch to be an ideal switch, that is to say it has negligible resistance, and it can be opened and closed without problem such as delay in operation or contact bounce.
When the switch is open in the position shown, the output to the next stage is zero volts; when closed the output voltage is equal to the message signal. The duration of the sample pulse depends on how long the switch is closed. The switch control signal is shown as a pulse train, because the switch can only have two conditions, open or closed. The final sampled output is illustrated in figure3 below.

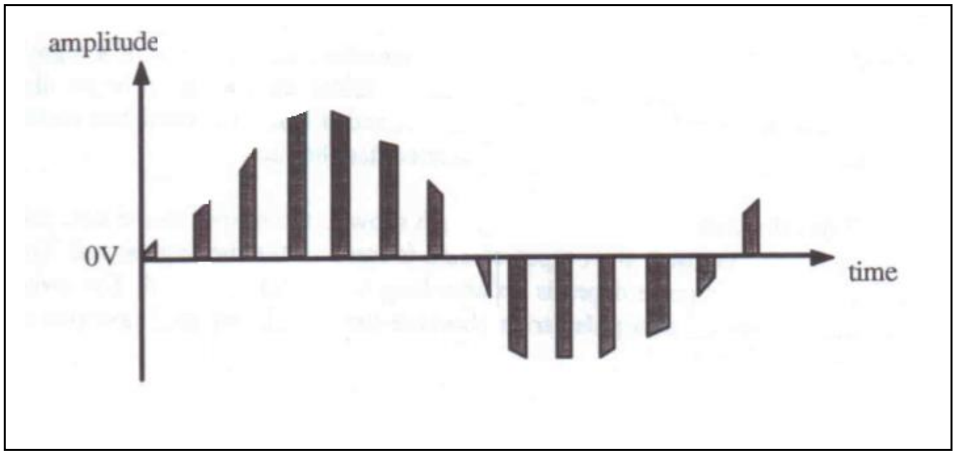


Figure3: Sampled output

This process called: Pulse Amplitude Modulation (PAM) where the heights of the transmitted pulse vary with the amplitude of the message.
We can think of pulse amplitude modulation as the amplitude modulation of a rectangular wave. Thus if the amplitude of a rectangular wave is modulated (to obtain PAM) we might expect that sidebands will be formed around the fundamental frequency, and each of the harmonics of the fundamental frequency that are contained in the rectangular waveform.
This is exactly what happened when we perform pulse amplitude modulation. The resulting PAM frequency spectrum is shown in figure 4, where fs (the fundamental frequency) is the sampling frequency.

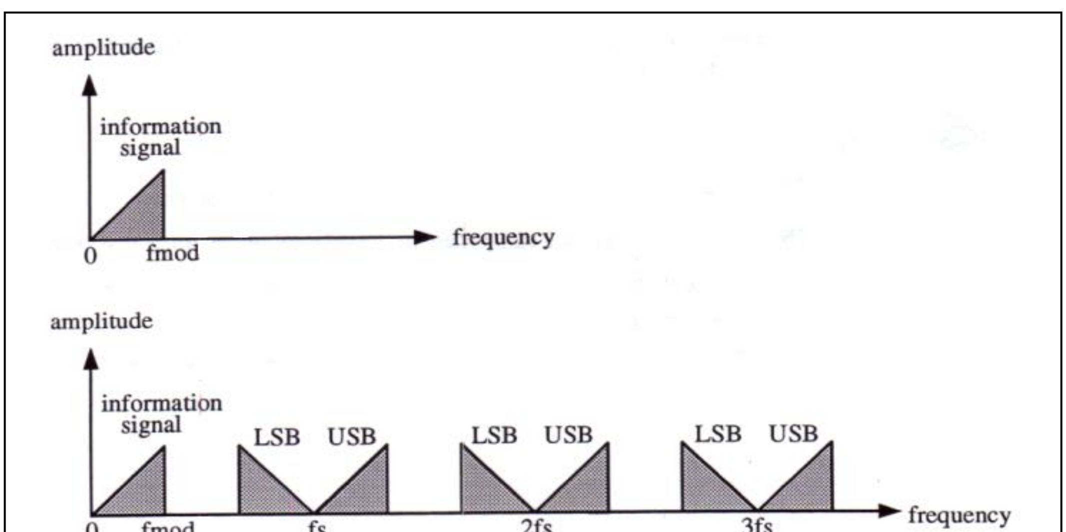


Figure4

As shown in figure below (figure 5-a the) lowest sampling frequency that could be used, without sidebands overlapping, is twice the highest frequency present in the information signal and this is the sampling theorem (Nyquist Criterion) *[any band limited analog signal x (t) having a bandwidth β can be reconstructed from its sampled values x (n T_s) with no distortion if the sampling frequency is: $f(s) \geq 2\beta$]*

If we reduce the sampling frequency still further (figure 5-b), the information signal's spectrum and sidebands overlap. This effect is known as fold-over distortion or aliasing. Note that the information signal can no longer be extracted from the PAM waveform simply by a low pass filter.

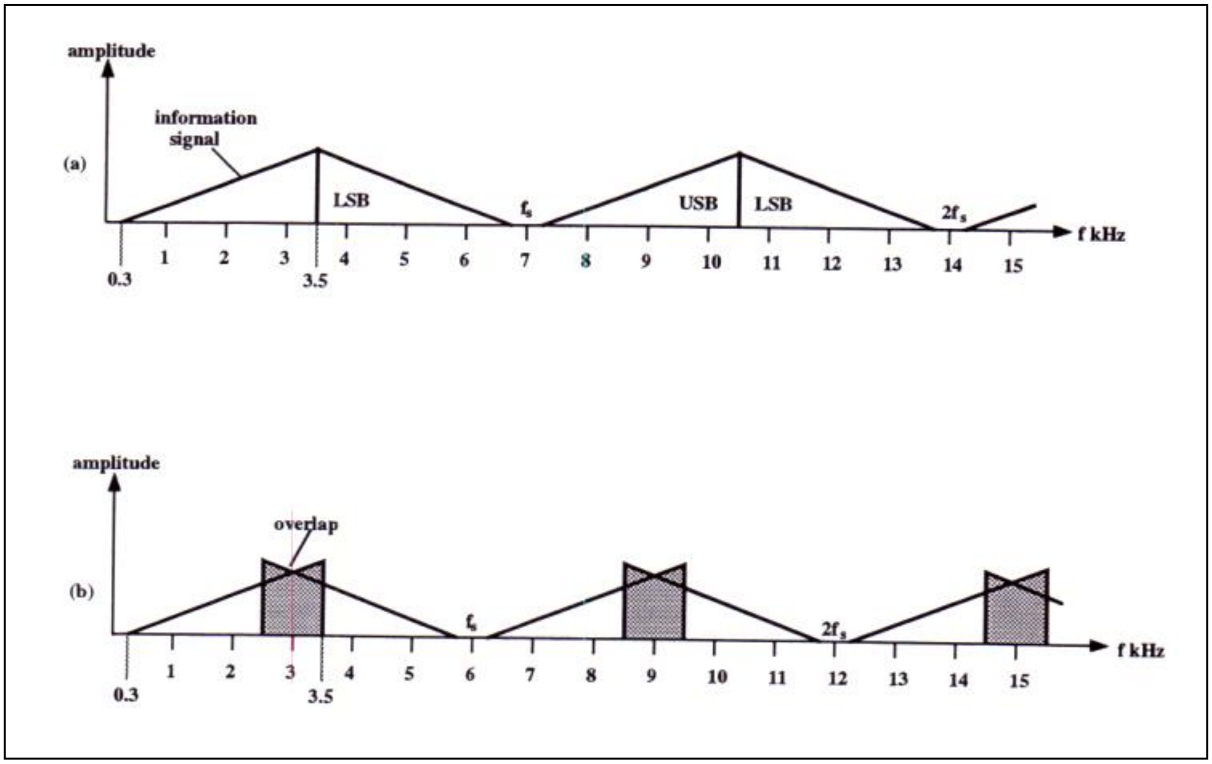


Figure 5: Sampling Frequency

In this experiment we will introduce two types of practical sampling: Natural Sampling (or Gating), and Instantaneous Sampling which is also known as flat-top sampling or sample-and-hold.

1. Natural sampling: If x(t) is an analog band-limited signal, the sampled signal is obtained by multiplying x(t) by a train of pulses of finite width.

$$x_s(t) = x(t)\sum_{n=-\infty}^{\infty} rect\left(\frac{t-nTs}{\tau}\right)$$

Where τ is the width of the rectangular pulse, and Ts is the sampling period. The ratio d= τ /Ts is called the duty cycle. The spectrum of $x_s(f)$ is:

$$Xs(f) = d\sum_{k=-\infty}^{\infty} \sin c\ (kd) X(f - kf_s)$$

2. Flat-top sampling: The sampled waveform, produced by practical sampling devices that are of sample and hold types, has the form:

$$x_s(t) = rect\left(\frac{t}{\tau}\right) \bullet \sum_{k=-\infty}^{\infty} x(t)\delta(t-kTs) = \sum_{k=-\infty}^{\infty} x(kT_s)rect\left(\frac{t-kTs}{\tau}\right)$$

The spectrum of the resulted signal is:

$$X_s(f) = \frac{1}{Ts} H(f) \sum_{k=-\infty}^{\infty} X(f-f_s)$$

$$H(f) = \tau \,.\, \text{Sinc}(\tau f)$$

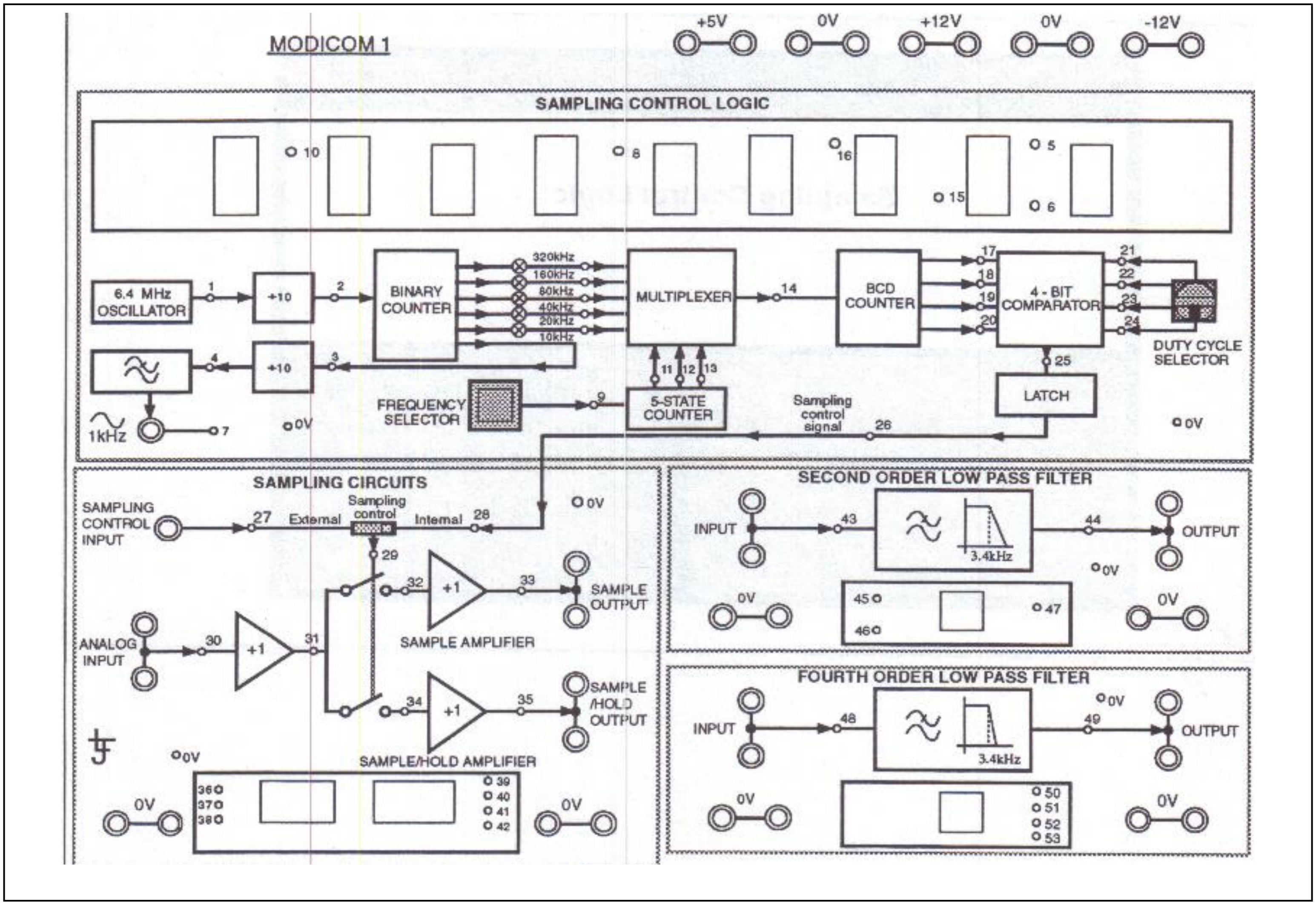


Fig.6: MODICOM 1

5.3 Description of the MODICOM1board

The purpose of the MODICOM 1 board is to illustrate how a signal may be sampled, transmitted as a series of samples, and reconstructed at the receiver by low pass filtering. A layout diagram for the board is shown in Figure6.

The board contains two low pass filters (each with a cut-off frequency of 3.4 kHz): a second order filter with a roll-off slope of -12dB/ octave, and a fourth order filter with the sharper roll-off slope of -24dB/ octave. Sampling frequencies of 2 kHz, 4 kHz, 8 kHz, 16 kHz and 32 kHz can be selected, and the sampling duty cycle can be varied in 10%steps from 0%to 90%.

The duty cycle of a digital waveform can be simply defined as the ratio of pulse duration to pulse repetition period. Duty cycle can be expressed as a ratio or more commonly, as a percentage. (i.e.: The duty cycle of a square wave with equal high (pulse) and low (pulse) duration is 0.5 or 50%

These features enable the student to examine the effect of different sampling frequencies and sampling duty cycles on the reconstructed waveform.

The transmitter provides a SAMPLE/HOLD output in addition to a SAMPLE output, so that comparisons can be drawn concerning the suitability of the two types of waveform as far as signal reconstruction is concerned. Although an external analog input may be used by the student, a unique feature is the on-board 1 kHz sine wave; this is synchronized to the sampling waveform and allows a 'static' oscilloscope display to be observed. Finally, the user has the option of supplying an external sampling control signal, to experiment with other sampling frequencies and duty cycles.

5.4 Procedure:

1. Connect supplies to board. The D.C. power requirements are; +5V, 1A, ±12V @ 1A.

2. On MODICOM1 set the following initial conditions:

- Ensure SAMPLING CONTROL switch is in 'INTERNAL' position.
- Put DUTY CYCLE SELECTOR switch in position '5'.
- Put sampling frequency to 32KHz.
- Link l kHz sine wave output to ANALOG INPUT.
- Turn on power to board.

~ (Natural Sampling and Reconstruction Process)

3. Display 1 kHz sine wave (tp. 7) and SAMPLE OUTPUT (tp. 33) on an oscilloscope. This shows the 1K Hz sine wave being sampled at 32 kHz; *so that there are 32 samples for every cycle of the sine wave* .

4. Link SAMPLE OUTPUT to input of FOURTH ORDER LOW PASS FILTER. Display SAMPLE OUTPUT (tp.33) and the output of the FOURTH ORDER LOWPASS FILTER (tp. 49) on the oscilloscope, to show how the originally 1 kHz sine- wave can be reconstructed from the samples by low pass filtering (Fig. 3b).

~ (The Effect of Sampling Frequency on Natural Sampling and the Reconstructed Signal)

5. By successive presses of the FREQUENCY SELECTOR switch, change the sampling frequency to 2kHz, 4kHz,8kHz, 16kHz and back to 32kHz (Sampling frequency is always 1/ 10 of the frequency indicated by the illuminated LED).

~ Observe how the SAMPLE OUTPUT (tp 33) changes in each case, and how the, sampling rates introduce distortion into the fitter's output waveform (tp 49). The sampled waveform contains components at fx, fs-fx, and fs +fx, 2fs-fx, etc, where fx = input sine wave frequency, fs = sampling frequency. Considering the characteristic of the filter circuit, can you explain why the distortion occurs?

~ (The Effect of Duty Cycle on Natural Sampling and the Reconstructed Signal)

6. The present position of the DUTY CYCLE SELECTOR switch (5) indicates that the duration of each sample is 50% of the sampling period (the time between the start of adjacent samples).

Variation of the switch setting allows this proportion (called the sampling duty cycle) to be changed from 0% to 90% in 10%steps.

7. *Using a 32 kHz sampling frequency*, vary the position of the DUTY CYCLE SELECTOR switch, observing how the SAMPLE OUTPUT (tp33) changes and how the amplitude of the fitter's output (tp49) waveform changes. This amplitude increases linearly as the sampling duty cycle increases from 10% to 90%.

~ (Comparison between Second-Order and Fourth-Order Low Pass Filters in Reconstruction the Signal after Natural Sampling)

8. Add a link from the SAMPLE OUTPUT to the input of the SECOND ORDER LOW PASS FILTER

9. Display the outputs of the SECOND ORDER and FOURTH ORDER low pass fitters (tp. 44 and tp. 49) on the oscilloscope. *With a sampling duty cycle of 50%* (DUTY CYCLE SELECTOR switch position S), step through the *8 kHz*, *16 kHz* and *32 kHz* sampling frequencies, comparing the filter outputs after each step.

~ Note that, for each sampling frequency, the fourth order output exhibits less distortion than the second order output. This is because the fourth order filter has a sharper 'roll-off' than the second order filter, and thus rejects more frequency components caused by the sampling signal.

~ (Comparison between Natural Sampling and Sample-Hold Circuits)

10. Remove the links between the SAMPLE OUTPUT and the inputs to the two filters. With a *sampling frequency of 32 kHz and a sampling duty cycle of 50%,* compare the SAMPLE OUTPUT (tp. 33) and the SAMPLE/HOLD OUTPUT (tp. 35) on the oscilloscope

~ (The Effect of Sampling Frequency and Duty Cycle on Sample-Hold)

11. a- Vary the sampling frequency to illustrate how each sample is held at SAMPLE/HOLD OUTPUT (tp 35).

b- Vary the duty Cycle to illustrate how each sample is held at SAMPLE/HOLD OUTPUT (tp 35).

~ Note how increasing the sampling duty cycle increases the proportion of time for which sampling occurs, and reduces the time for which these samples are held at the SAMPLE/HOLD OUTPUT.

~ ((Sample-Hold) Sampling and Reconstruction Process)

12. Link from SAMPLE/HOLD OUTPUT to input of FOURTH ORDER LOW PASS FILTER using a sampling frequency of 32 kHz and a duty cycle of 50%, display the SAMPLE/HOLD OUTPUT (tp35) and the output of the FOURTH ORDER LOW PASS FILTER (tp49) on the oscilloscope, to show once again that the original 1kHz sine wave can be reconstructed by low pass fi1tering.

~ (The Effect of Sampling Frequency on the Reconstructed Signal from Sample-Hold process)

13. By making successive presses of the FREQUENCY SELECTOR switch, note how the filter's output waveform (tp49) changes with sampling frequency.

~ (The Effect of Duty Cycle on the Reconstructed Signal from Sample-Hold process)

14. *Using a 32 kHz sampling frequency*, vary the sampling duty cycle and note the effect on the reconstructed signal (tp49)

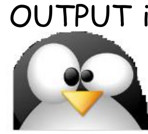
~ Note that, in contrast with the results of Step 7, the filter's output amplitude is now independent of the sampling duty cycle, and is equal to the amplitude of the original analog input. This is an important result - with the SAMPLE/HOLD OUTPUT, ***the proportion of sampling time to holding time has no effect on filter output amplitude, providing the sampling frequency is high enough to recover the original analog waveform.***

In a practical digital communications system, transmitted samples tend to be of short duration, so that as many samples can be transmitted as possible; at the receiver, each received sample is held in a sample/ hold circuit before low-pass filtering, in order that the filter's output has maximum amplitude.

~ (Comparison between Second-Order and Fourth-Order Low Pass Filters in Reconstruction the Signal after Sample-Hold process)

15. Add a link from the SAMPLE/HOLD OUTPUT to the input of the SECOND ORDER LOW PASS FILTER.

16. Display the outputs of the SECOND ORDER LOW PASS FILTER (tp. 44) and FOURTH ORDER LOW PASS FILTER (tp. 49) on the oscilloscope, and, for *a sampling duty cycle of 50%,* compare the outputs of the two filters for the *8 kHz*, *16 kHz* and *32 kHz* sampling frequencies. Are these waveforms more or less distorted than those obtained from the SAMPLE OUTPUT in Step 9, for each sampling frequency?



The tide has its ebb   Good luck

Experiment Six
PCM Transmitter

6.1 Objectives

- Summarize the principles of a PAM system.
- Describe how an analog signal can be converted to a serial transmission of digital pulses.
- Describe the quantization process and the cause of quantization noise.
- Explain the need for frame synchronization in a PCM ~ transmitter.

6.2 MODICOM 3/1 Board

The MODICOM 3/1 board provides a PCM transmitter function. The transmitter provides a 2-channel time division multiplexed, pulse code modulated output

The transmitter may be operated in FAST mode for real time operation, or it may be operated in SLOW mode to allow you to examine control signals and information flow.

Simulated information signals may be supplied externally or from the on-board Function Generator. Timing control for the board is provided by the Timing Control Logic unit

Initial sampling of the information signals is achieved by a Sample/Hold unit which feeds the Analog to Digital Converter. As on the MODICOM 2 board, sampling and time division multiplexing is achieved in one process.

The transmitter has the option of introducing error detection and correction codes through the inclusion of the Error Check Code Generator block and its associated switch unit

The output from the *A/D* converter is in a parallel format and therefore Parallel to Serial Conversion must take place before the PCM output waveform is passed to the transmission medium. This process is carried out by the Shift Register block.

The overall synchronization requirement for the system is provided by the Sync. Code Generator block.

The MODIC OM 3/1 board also provides a number of switched faults which simulate pre-designed transmitter faults.
The complete layout for the MODICOM 3/1 board is shown in Figure 1 opposite.

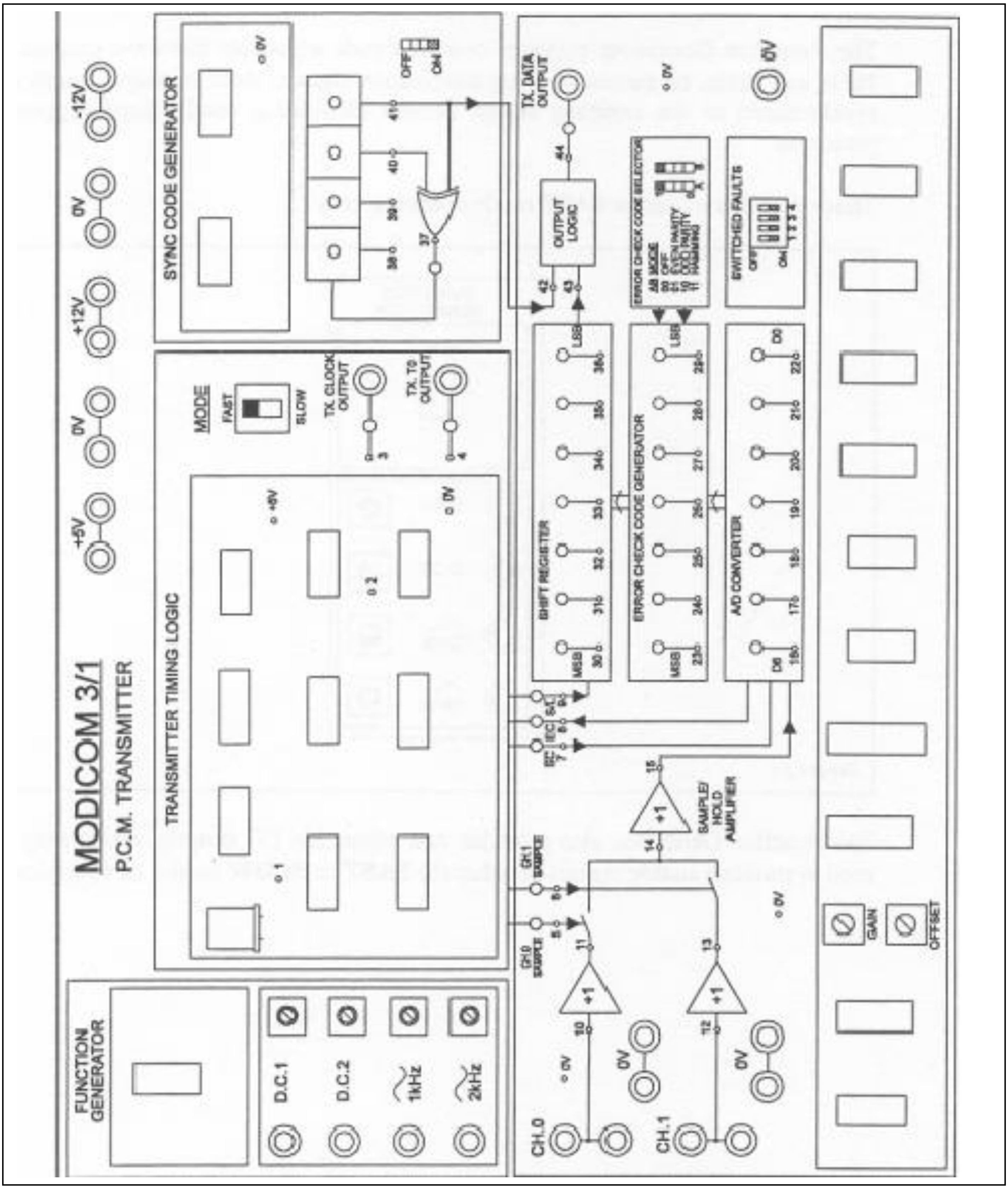


Fig.1: MODICOM3/1

6.2.2 Function Generator

The Function Generator provides two amplitude adjustable sine-wave outputs of 1kHz and 2kHz, to simulate analog information signals. Both of these signals are synchronized to the sampling signal, thereby eliminating oscilloscope triggering problems. These signals are used in **FAST mode operation only**.
The Function Generator also provides two adjustable DC outputs which may be used to simulate analog signals in either the **FAST or SLOW** modes of operation.

6.2.3 Timing Control Logic

As with all digital communication systems the timing of all operations is of vital importance. All internal timing control for the transmitter is derived from this unit

6.2.4 Analog to Digital (A/D) Converter

In this unit the time division multiplexed PAM samples, which are still analog in nature, are passed sequentially to the A-D Converter.

The converter is controlled by signals derived from the Transmitter Timing Logic, and produces as its output the binary coded number or word associated with the amplitude of the sample being processed.

This PCM word is then clocked into latched drivers which drive light emitting diodes (LEDs) indicating the bit state of the output word. The PCM word is simultaneously passed to the next processing stage.

6.2.5 Sampling and TDM Unit

This unit converts the analog information signals into PAM information samples which are then time division multiplexed. The unit also includes a sample-and-hold circuit.

6.2.6 Error Check Code Generator

The MODICOM 3/1 board provides you with the opportunity of investigating a number of error checking techniques.

Included in this unit is the facility to use a Hamming code format. This format is widely used in digital systems to allow error checking and correction.

6.2.7 Parallel to Serial Converter

As stated previously, the output from the A/D converter is in a parallel format. This signal must be converted to a serial format prior to transmission on the bearer (for example, cable).

This process is called parallel to serial conversion, and it is performed by two 4-bit shift registers.

6.2.8 Synchronization Code Generator

As in all digital systems, synchronization between receiver and transmitter is essential for correct operation and recovery of the original information signals. The more complex the system, the greater the need for more complex synchronization signals.

The Synchronization Code Generator on the MODICOM 3/1 board provides the synchronization signals necessary for the MODICOM 3 PCM/TDM system.

6.3 Sampling

The sampling of the analog information signals on the MODICOM 3/1 board follows the same principles which you learned during your studies on the MODIC OM 1 board. .

It is important that you realize that sampling must be carried out in accordance with the Nyquist Criterion; that the sampling frequency must be at least twice the r highest frequency present in the information signal.

As discussed previously, if samples are to be converted to PCM, then the sample duration must be small to ensure that near-instantaneous values are taken.

One of the circuits in the Sampling and TDM unit is a sample-and-hold circuit, essentially, this circuit samples the analog waveform and then instead of its output returning to zero volts, it holds this value until the next sample is taken. This process is achieved by charging a capacitor to hold the value until the next sample.

A major advantage of a sample-and-hold circuit over a simple switched-sampling circuit is that there is more power in a sampled- and-held waveform than in a stream of samples.

We have now seen that this circuit is also used immediately prior to A/D conversion, to allow conversion to take place in the finite time required by the A/D converter.

6.4 Quantization

The basic functions of a PCM transmitter are Sampling, Time Division Multiplexing, Quantization and Analog to Digital Conversion.

Quantization is the name given to the process of allocating discrete binary values to each of the samples taken in the initial PAM process. You should remember that PAM processing is the first stage in producing a PCM signal.

Once sampling has been achieved, then each sample is allocated a binary value. Binary values change in discrete steps and are not continuous variables as in analog waveforms. The range of binary values used is a design feature of the system and depends upon the amplitude range of the analog signal and the accuracy of conversion which can be tolerated.

Binary values, or words, may contain any number of binary digits, or bits. Most systems use 8-bit word lengths, which have been found to provide sufficient range and accuracy for speech signals.

Where a greater range of analog values is used, or better accuracy required. Then you should not be surprised to find systems using larger word lengths.

As with all electronic engineering processes, quantization produces its own problems and consequently compromise decisions have to be made.

One important problem caused by quantization is the problem of quantization noise, which unless carefully controlled leads to ambiguity and error in the reconstituted signal. Quantization noise arises because of the discrete, stepped, nature of binary values which are used to represent a continuously variable information signal.

6.5 Analog to Digital (A/D) Conversion

This describes the process of changing an analog signal to a digital word. Design constraints are placed upon us in terms of what devices are available and how they function. It is usually the case that A/D converter devices produce a parallel binary output code.

However, in order to aCH1eve PCM transmission, each binary code must be converted into a serial format. This requires the A/D conversion process to be followed by parallel to serial conversion, where binary codes presented in parallel format are converted to serial format prior to transmission through a medium.

6.6 Multiplexing

TDM (Time Division Multiplexing) is widely used in digital communication systems, to aCH1eve greater efficiency in the use of the transmission medium.

By electronically switching information samples, it is possible to interleave the samples from a number of information signals, sequentially, to allow those information channels to share the same transmission medium without mutually interfering with each other.

The major problem to be overcome in implementing TDM is that of synchronization. The timing of the receiver must be identical to that of the transmitter if the correct information samples are to be passed to the correct output channels for demodulation.

In the following practical exercise you will have the opportunity to investigate the time division multiplexing used on the MODICOM 3/1 board.

6.7 Parallel to Serial Conversion

You have learned that information signals can be sampled using a sample-and-hold circuit, time division multiplexed, and then digitized into coded binary numbers or words. A different binary word is used for each discrete analog value, a process known as quantization.

You further learned that the process of analog-to-digital (A/D) conversion used on the MODICOM 3/1 board produces a parallel output from the A/D CONVERTER block, in the fom1 of a 7-bit binary word.

For purely economic reasons, in any practical transmission system there should be a minimum number of transmission links between transmitter and receiver.

Now, the word appearing at the output of the A/D converter is in a 7-bit parallel format. To transmit this information to the receiver, the system would have to utilize 7 individual transmission media (for example, cables). This is clearly unacceptable in an efficiently managed communication system. Therefore our next step is to convert our digital word from parallel format (7 bits) into serial format for onward transmission.

This process is known as parallel to serial conversion.

On MODICOM 3/1, this operation is carried out using two 4-bit parallel access shift registers. It is essential that they are clocked at a rate which is sufficiently high to ensure that the input digital information, or data, loaded into the shift registers is completely shifted out to the transmitter output before the next valid data appears on the outputs of the A/D converter.

6.8 Procedure

a. Investigation of Sampling

1. Connect the power supply to the MODICOM 3/1 board.

2. Ensure that the following initial conditions exist on the MODIC OM 3/1 Transmitter board.

   (a) MODE switch in FAST position.
   (b) SYNC. CODE GENERA TOR "on/off" switch to OFF position.
   (c) ERROR CHECK CODE SELECTOR switches to OFF (00) position.
   (d) All four switched faults to OFF.
   (e) DCl and DC2 amplitude controls both fully clockwise.
   (f) 1kHz & 2kHz Levels Set to 10Vp-p

3. Connect the 1kHz output to the CHO input on the MODICOM 3/1 board

4. Turn **ON** the power supply.
5. Measure the frequency and amplitude with the oscilloscope, by placing a probe on test point 10 of MODICOM 3/1.

(This 1 kHz signal is the simulated analog information signal which is to be encoded into PCM format.)

6. Monitor **TP5** on MODICOM 3/1 using the oscilloscope and measure the frequency. (*This is the frequency at which the analog signal will be sampled).*

7. Connect CH1 oscilloscope probe to test point 15 on MODICOM 3/1 and CH2 oscilloscope probe to test point 10 and set the followings:
   - Set the oscilloscope time-base to 0.lms per division.
   - Set both oscilloscope volt division to 2V /division.
- Adjust the vertical positioning of both waveforms (Y -control) so that the two are superimposed.
   - Use the dual mode

8. Sketch the waveforms at tpl0 and tp15 (both on the same time scale).

9. Turn **OFF** the power supply

b. Investigation of Time Division Multiplexing

1. Set up the connections as in step a-2.
2. Connect the 1 kHz output to the CHO input on MODICOM 3/1.
3. Turn **ON** the power supply.
4. Check that the PAM pulses of the 1kHz sine-wave are on tp15 of MODICOM 3/1.
5. Connect: CH1 of the oscilloscope to tp5
   CH2 of the oscilloscope to tp6
   What do you notice about the timing of the sample control waveforms?

6. Turn **OFF** the power supply.

7. Connect the 2kHz signal to CH1 input
8. Connect: Oscilloscope channel1 to tp10 (CH0 input)
   Oscilloscope channel2 to tp15
9. Turn **ON** the power supply.
10. Using dual mode **sketch** and **explain** what you see.
11. Turn **OFF** the power supply.

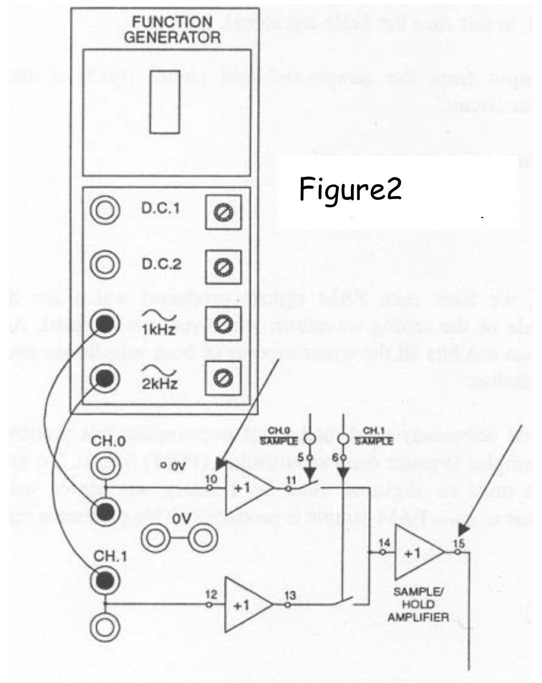


Figure2

Summary:

When sampling the 1 kHz signal on its own, we obtained a sampled-and-held waveform with 3lμs between samples. You saw from the output wave-form that there were unused intervals of 3lμs duration.

You have also seen, by comparing the wave-forms on tp5 and tp6, that the sampling pulses are approximately 3l μs apart. This leads to the conclusion that it is possible, during the unused intervals, to insert PAM samples from another analog input channel, in this case the 2 kHz test signal.

The output from the sample-and-hold circuit (tp15) is then fed to the A/D converter circuit.

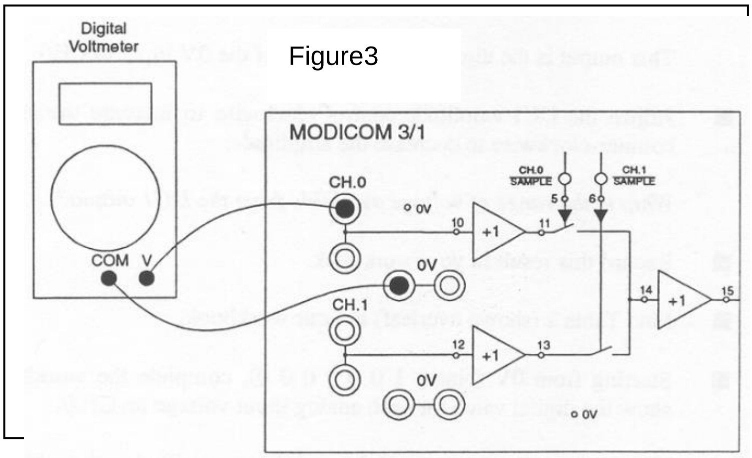


Figure3

c. Investigation of Quantization

1. Set up the connections as in step a-2.
2. Connect: DC1 output to CH0 input
   DC2 output to CH1 input

3. Connect a digital voltmeter to CH.O input on the MODICOM 3/1 board. SeeFig.3

4. Turn DC2 amplitude control, in the Function Generator block, fully counter- clockwise.
5. Adjust DCl amplitude control until DCl output measures OV on the meter (to within ±20mV).

In the A/D Converter block of MODICOM 3/1, identify the A/D output state LEDs (DO to D6).

These LEDs represent the bit state of the binary PCM word allocated to the PAM sample being processed. An illuminated LED represents a "1" state, while a non-illuminated LED indicates a "0" state. D6 is the MSB and DO the LSB.

6. Observe the parallel outputs DO to D6 from the ND converter. Ensure that the outputs are as follows:

D6	D5	D4	D3	D2	D1	D0
1	0	0	0	0	0	0

This output is the digital representation of the zero Volt input to CHO.

7. Adjust the DCl amplitude control clockwise to increase the amplitude and counter-clockwise to decrease the amplitude and fill table1 below.
   Take care to ensure that each input voltage specified in the table is set up to within ±20m Volts.

DC Input	D6	D5	D4	D3	D2	D1	D0
+Max							
+5V							
+4V							
+3V							
+2V							
+1V							
0V	1	0	0	0	0	0	0
-1V							
-2V							
-3V							
-4V							
-5V							
- Max							

Table1

Y
vo
T
e:
co
waveform. Therefore the numbers have to be allocated in steps, let us say every 0.5 volts.

This means that a unique binary value is allocated to 0 volts, another to 0.5 volts, and another to 1 volt and so on. This is [me providing that every time the analog waveform is sampled the amplitude lies on one of those discrete values. However, what happens if the analog value, at the time of sampling, is 0.8 volts? This is neither 0.5 volts which has one binary value, nor 1.0 volts which has a different binary value. The system has to allocate one *or* the other but neither is correct.

The difference between the analog value and the value represented by the binary number is known as the quantization error. This error is at its greatest when the original analog value falls midway between two quantization levels. The maximum possible value of quantization error is therefore half of one quantization step. In the above example the step size is 0.5 volts, so the maximum quantization error would be 0.25 volts.

The overall effect of the quantization errors over many samples is that when the signal is received at the far end, and demodulated, the information signal is found to be distorted. This distortion is known as quantization noise.

Quantization noise can be reduced by reducing the size of the quantization steps, but it can never be eliminated entirely. The complete process of converting an analog information signal to a series of coded binary values is known as analog to digital (A/D) conversion.

d. Parallel to Serial Conversion

1. Set up the connections as in step a-2
2. Make the following connections on the MODICOM 3/1 board
    - DC1 output to CHO input
    - DC2 output to CH1 input
    - Switch to FAST mode

3. Turn on the power supply.
4. Connect a digital multi-meter to CHO on the MODICOM 3/1 board.
5. Adjust DCl amplitude control until the output is equal to +3V.
6. Move the digital multi-meter to CH1 input on MODICOM 3/1.
7. Adjust DC2 amplitude control until the output is equal to -2V.

The LEDs on the A/D Converter and Shift Register are a combination of both input voltages. It is therefore impossible to determine what each code is, so we will switch the system to SLOW mode where each code can be easily seen.

8. Switch to SLOW mode and observe the LEDs on the shift register and A/D converter. Note that you may require approximately 10 seconds for the system to settle down before taking any readings.
9. Record your observations in your report.
10. .Compare your readings to those taken in Tables 1, you should find that they are the same.

Note: *You may have found that you results are not identical to those taken earlier. This is due to the characteristics of the sample/hold circuit of the MODICOM 3/1 Transmitter, switching between FAST and SLOW modes may cause a slight change in voltage at the sample/hold output. This may result in an increase or decrease in the A/D Converter code by one bit when the Transmitter is switched between FAST and SLOW modes.*

Your observations in this exercise should have indicated that at approximately 7-second intervals, a particular combination of LEDs is lit in the A/D Converter block. These LEDs represent the latched outputs from the A/D converter IC, and should have shown the two binary value words which relate to the voltages you set up at the inputs of CHO and CH 1,

For example: **CHO = 1 1 0 0 1 0 1** **CH1 = 0 1 0 0 1 1 0**

Observe that the quantized value for a particular channel is then parallel-loaded into the Shift Register circuit, which then shifts the bits along from left to right (LSD first) to the Output Logic block and hence to the transmitter output (Tx Data Output).

Note: *It is essential that one value is shifted out of the shift register circuit before the next value is output from the A/D converter. In fact, the speed at which the data is read out of the shift register is the actual signalling rate on the transmission medium.*

e. Timing Frame

Note the following:

The sequence of operations taking place at the transmitter is synchronized to the transmitter clock, which appears at the TX CLOCK output (tp3) on the MODICOM 3/1 transmitter board.
Each clock cycle (rising edge to rising edge) occupies a time period equal to the duration of one bit of the transmitted digital signal.
The sequence of operations at the transmitter is designed to repeat every 15 bits. These bits are numbered 0 to 14, and one complete cycle of 15 bits is known as a timing frame.
The transmitter's TX TO output (which stands for transmitter bit time 0) can be seen at tp4 and goes high during the bit 0 time element.
As shown in Figure 4, the transmitter Output Logic block has two inputs:

- Serial data from the Shift Register block.
- The output from the Sync. Code Generator

Data to be transmitted appears at the inputs to the transmitter Output Logic block at the start of each clock pulse.
The Output Logic block delays the data by half a bit time, so that the transmitted data on TX DATA OUTPUT changes half way through each transmitter clock cycle.

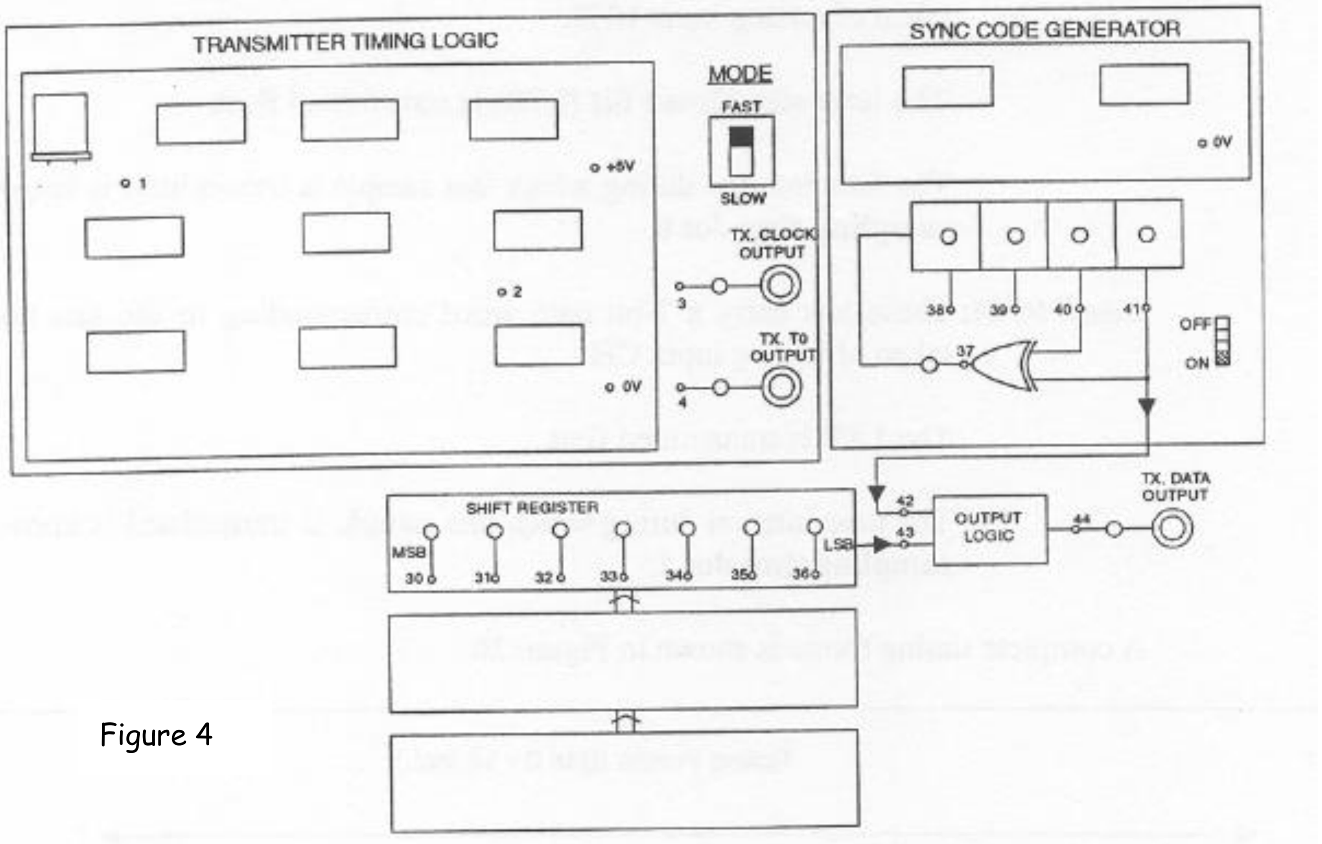


Figure 4

The data appearing at the transmitter's TX DATA OUTPUT, in the middle of each transmitter clock cycle, is as follows:
Bit 0: This bit is reserved for outputs from the Sync. Code Generator block, and since the Sync. Code Generator is currently off, a '0' is transmitted.
Bit 1 to 7: These bits carry a 7-bit data word corresponding to the last sample taken of analog input CH0
The least significant bit (LSB) is transmitted first.
The time interval during which this sample is transmitted is known as **sampling timeslot 0**.
Bits 8 to 14: These bits carry a 7-bit data word corresponding to the last sample taken of analog input CH 1.
The LSB is transmitted first
The time interval during which this sample is transmitted is known as **sampling timeslot 1**.
A complete timing frame is shown in Figure 5.

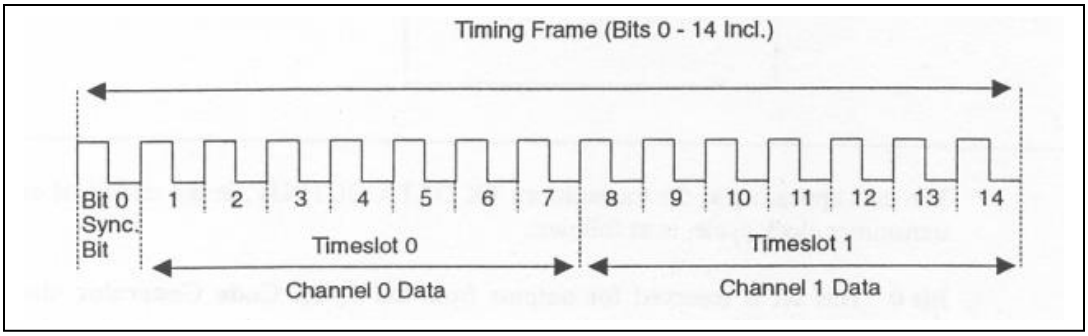


Figure 5

If you fail to plan you plan to fail
Good luck

Experiment Seven
PCM Receiver

7.1 Objectives

- Outline the principles of a PCM receiver.
- Describe the action of a Digital to Analog Converter. . Explain how the infornlation signals are recovered by demultiplexing and demodulating.
- Investigate the importance of system timing in a PCM receiver.

7.2 The Essential Features of a PCM System

In the last experiment you learned how MODICOM 3/1 transmitter demonstrated all of the essential processes of PCM transmission.

Essentially the receiver has to accept a binary-coded sequential data stream from the transmission medium. It then has to process this data and recover the original information signals.
Before it can even begin to achieve this, the system design must ensure that the receiver timing is synchronized to the transmitter timing and this can only be achieved if the receiver is in receipt of signals carrying two essential pieces of information:

1. Information regarding the Transmitter Clock, which allows the receiver to align its timing with that of the transmitter.
2. Information enabling the Receiver to identify each bit of the incoming data stream.

In this context, the identification of Bit 0 (sync. Bit) of each timing frame is of vital importance as this provides the receiver with positional reference.
The complete layout for the MODICOM 3/1 board is shown in Figure 1 opposite.

7.3 Receiver Processes

Essentially, the PCM receiver must provide the converse processes to those carried out at the transmitter.
The incoming PCM signal from the medium is in serial format and must first be converted to a parallel format before being processed further. This process is known as serial to parallel conversion.

Once the data is in a parallel format it may be presented, word by word, for digital to analog *(D/A)* conversion. This process converts the PCM signal back into the time division multiplexed information samples resulting from pulse amplitude modulation at the transmitter.
At this point in the study program you should remind yourself of the possibility of distortion due to quantization noise.
Once the PAM samples have been recovered then they must be demultiplexed back into their separate, original information channels.

The information samples for each channel may now be demodulated back into the original information signals.

MODICOM 3/2 PCM receiver (see Figure1) provides useful processing units facilities to recover the original signal.

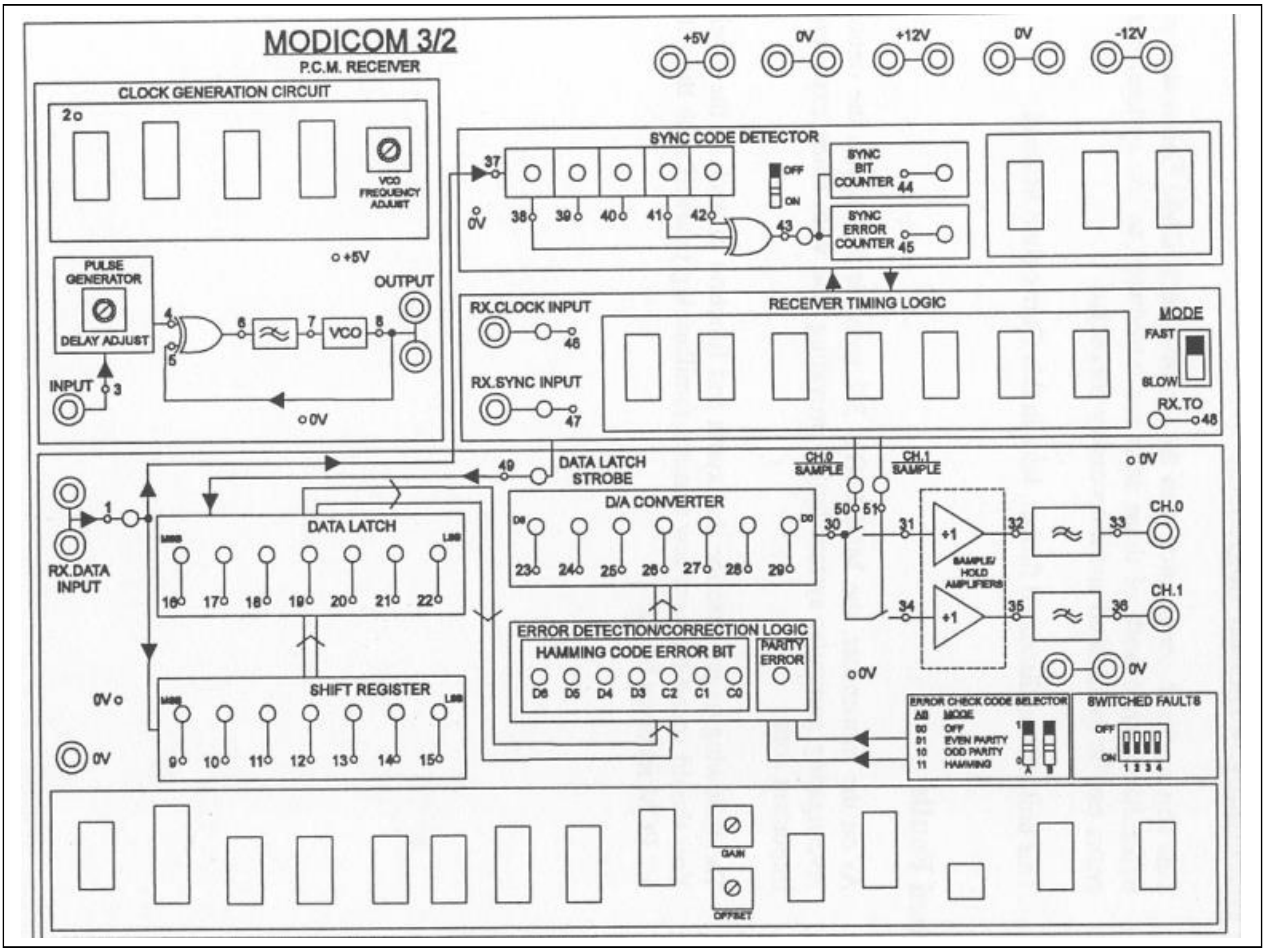


Figure1: MODICOM3/2 Receiver Board Layout

7.4 Receiver Facilities

MODICOM 3/2 PCM receiver (see Figure1) provides useful processing units facilities to recover the original signal.

7.4.1 Sync. Code Detector Unit

This unit works in conjunction with the transmitter Sync. Code Generator and its uses and operation will be discussed further, later(see Figure2).

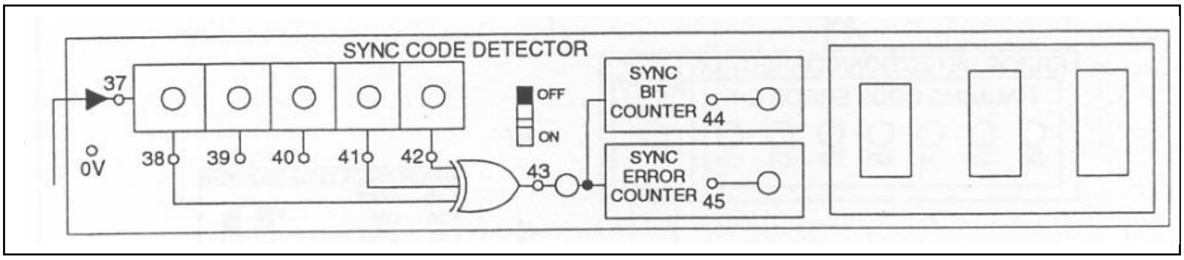


Figure2

7.4.2 Error Detection/Correction Logic Unit

This unit works in conjunction with the Error Check Code Generator unit on the transmitter (see Figure3). When both of these units are introduced to the system, transmission errors can be monitored and if necessary corrected.
This unit will be discussed further, later in this Curriculum Manual.

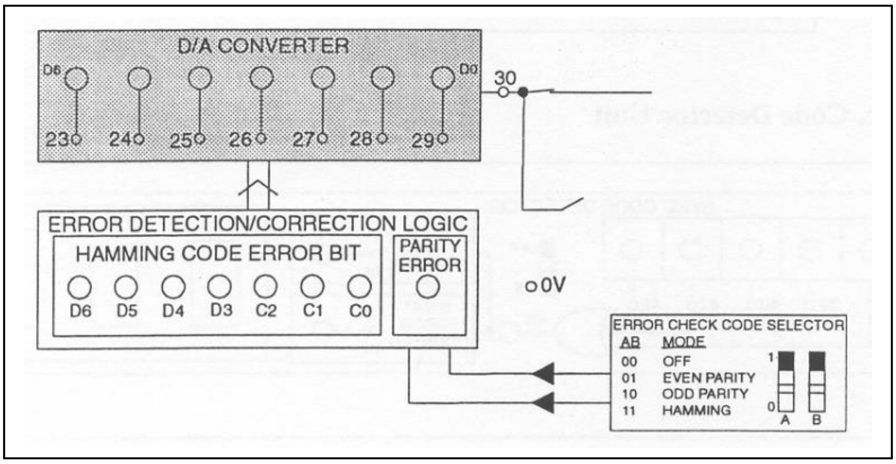


Figure3

7.4.3 Switched Faults

As on the transmitter, the MODICOM 3/2 receiver offers you the opportunity of investigating particular system faults, providing you with some experience in this important area (see Figure4).

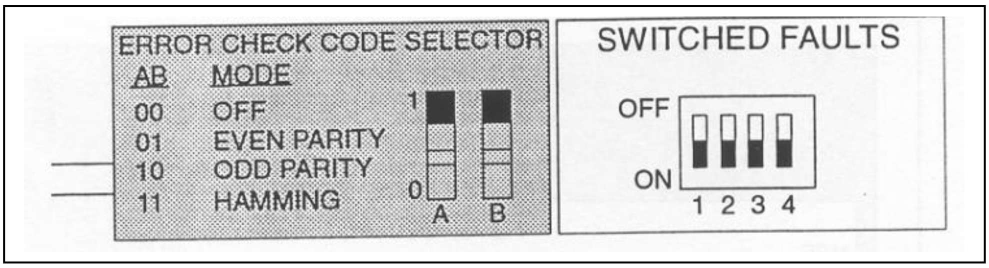


Figure4

7.4.4 Clock Regeneration Circuit

This circuit block provides the function of regenerating the Transmitter Clock, to ensure system timing compatibility (see Figure5).

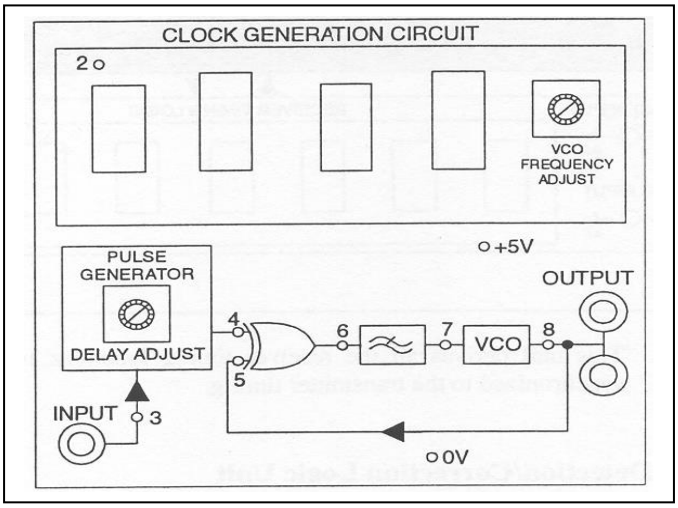


Figure5

7.4.5 Receiver Timing Logic Unit

This unit derives the entire receiver timing functions to ensure correct operation, synchronized to the transmitter timing (see Figure6).

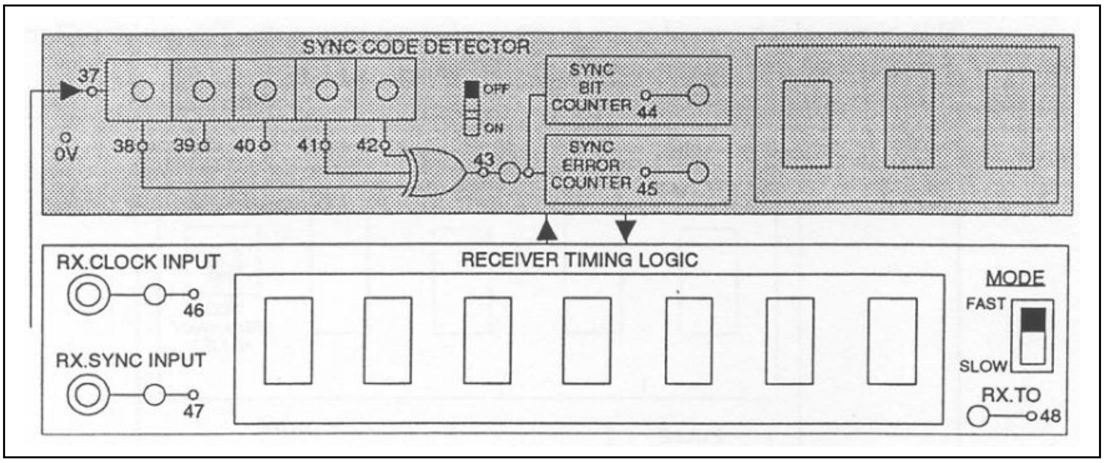


Figure6

7.4.6 Receiver Shift Register Block

This block provides the serial-to-parallel conversion required at the receiver. Incoming serial data is converted to a parallel binary word before further processing (see Figure7).

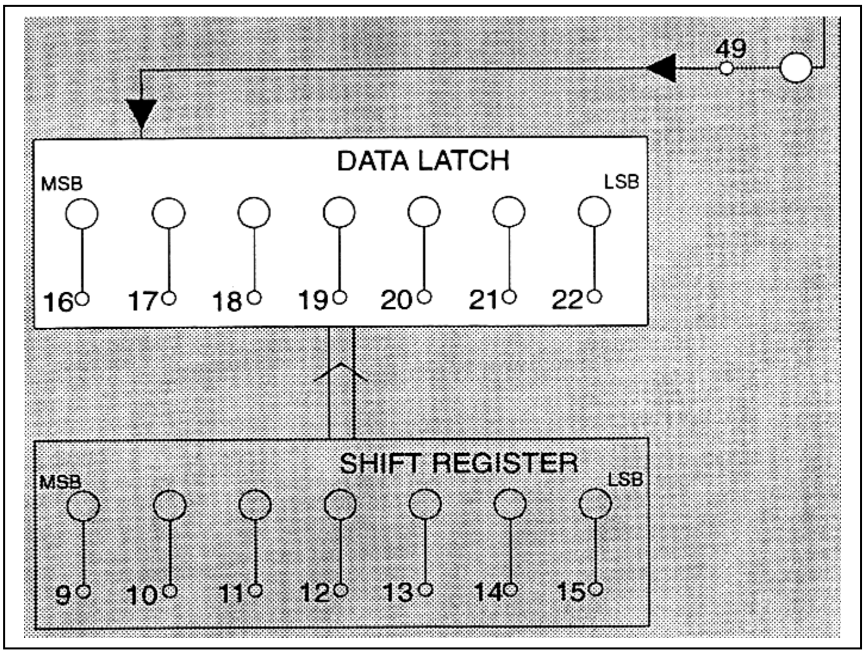


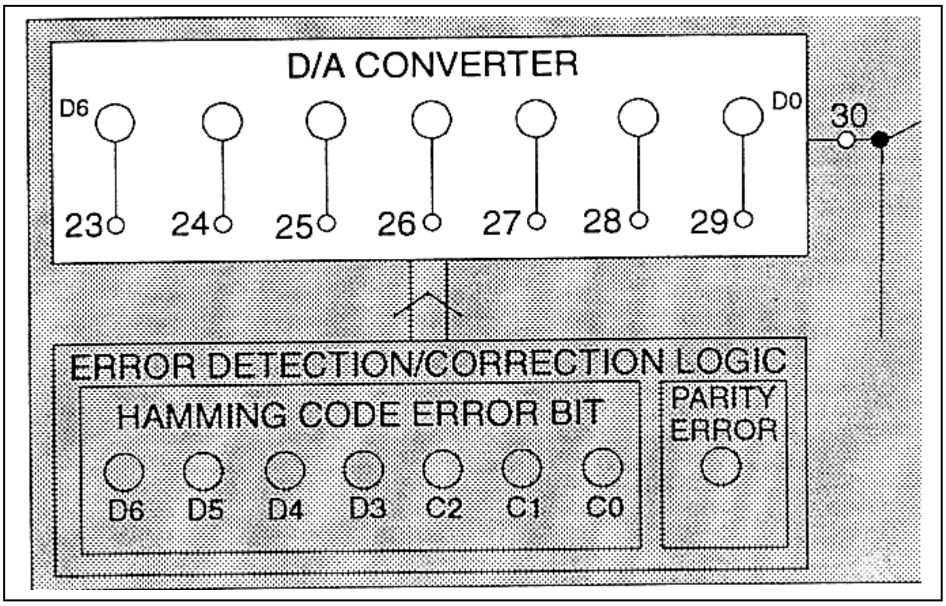
Figure7

7.4.7 Data Latch Block

This block latches and holds parallel output words from the Shift Register, in order to present each word to the D/A Converter at the correct moment in time (see Figure7).

7.4.8 Digital to Analog Converter

This block converts the binary coded PCM words back into PAM information samples (see Figure8).



7.4.9 Demultiplexer and Demodulator Unit

This unit demultiplexes the PAM samples back into their correct information channel. That is, CHO samples to the CHO demodulator and CH1 samples to the CH1 demodulator.

Demodulation (low pass filtering) recovers the original CHO and CH1 information signals (see Figure9).

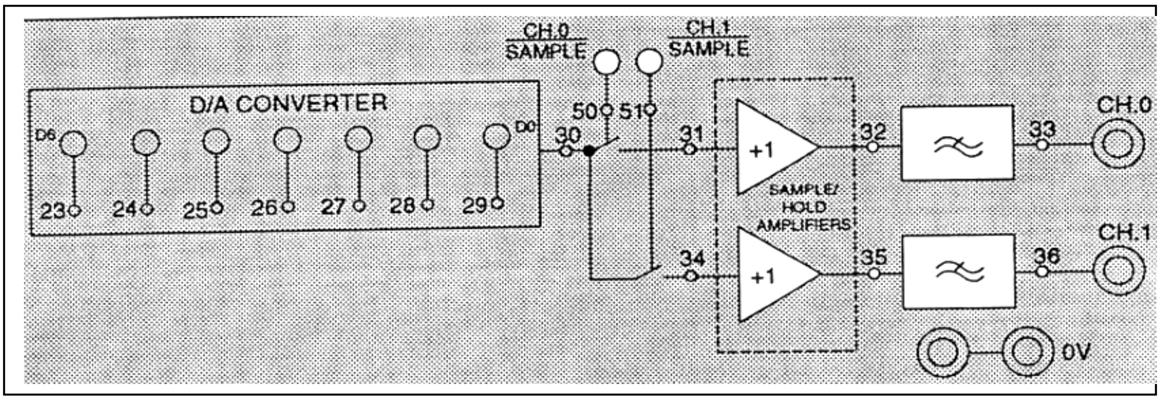


Figure9

7.5 Procedure

Part1: Validation of Receiver Board Operation

1. Connect the power supply to the MODICOM 3/1 and MODICOM 3/2 boards.

2. Ensure that the following initial conditions exist on the MODIC OM 3/1 Transmitter board.
   (a) MODE switch in FAST position.
   (b) SYNC. CODE GENERA TOR "on/off" switch to OFF position.
   (c) ERROR CHECK CODE SELECTOR switches to OFF (00) position.
   (d) All four switched faults to OFF.
   (e) DCl and DC2 amplitude controls both fully clockwise.
   (f) 1kHz & 2kHz Levels Set to 10V pk/pk.

3. Ensure that the following initial conditions exist on the MODIC OM 3/2 Receiver board.
   (a) MODE switch in FAST position.
   (b) SYNC. CODE DETECTOR "on/off" switch to OFF position.
   (c) ERROR CHECK CODE SELECTOR switches to OFF (A=0, B=0).
   (d) All four switched faults to OFF.
   (e) PULSE GENERATOR DELAY ADJUST control fully clockwise

4. Connect: 1 kHz output to CHO input
   2 kHz output to CH1 input

5. Connect MODICOM 3/1 to MODICOM 3/2 as shown in Table below:

MODICOM 3/1 (Transmitter)	MODICOM 3/2 (Receiver)
TX Clock Output tp3	RX Clock Input tp46
TX To Output tp4	RX SYNC Input tp47
TX DATA Output tp44	RX DATA Input tp1

Table 1

6. Turn **ON** the power supply.

7. Observe the display on the oscilloscope at the following test points on MODICOM 3/1:
   Transmitter CHO (1 kHz) at test point 10
   Transmitter CH1 (2 kHz) at test point 12

8. Display transmitter TX Data Output (tp44) on channel 2 of the oscilloscope.
(This clearly shows the data stream being sent from the transmitter to the receiver)

9. Vary the 1 kHz and 2 kHz amplitude controls (in the transmitter Function Generator block) and note that the transmitted data varies.

10. Display the receiver CHO and CH1 outputs (tp33 and tp36 on MODICOM 3/2

11. Check that the two sinew waves have been successfully sent from the transmitter to the receiver, and that each appears on the correct output channel

12. Vary the 1 kHz and 2 kHz amplitude controls on the transmitter, and observe the effect on the receiver outputs

13. Turn **OFF** the power supply.

Part2: Examination of Transmitter and Receiver Shift Registers

1. Set up the connections as in step 1, 2 and 3 in part1
2. On MODICOM 3/1 connect:
   CH0 input to DC1 output
   CH1 input to CH0 input

The same DC level will now be present on both CHO and CH1 inputs As shown in figure 10.

3. Turn **ON** the power supply.
4. Switch the MODE select keys on both MODICOM 3/1 and MODICOM 3/2 to FAST mode.

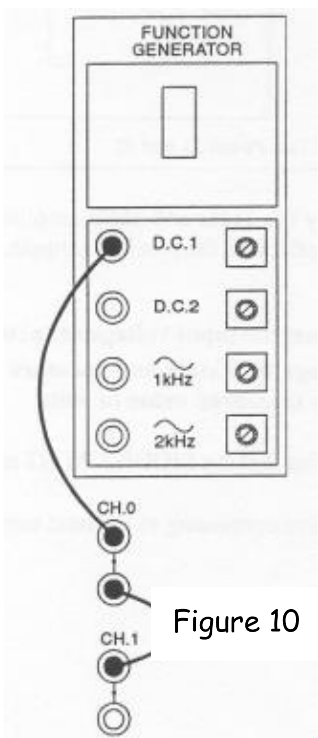
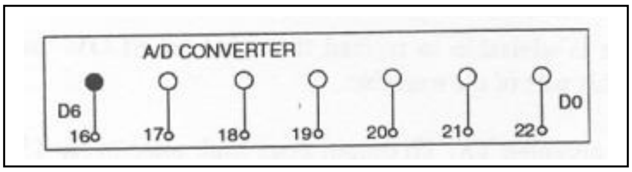
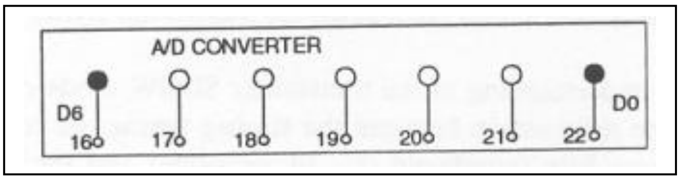


Figure 10

5. Adjust DCl amplitude control until OV is noted on the oscilloscope (or multi-meter) and the quantized value for OV is seen on the A/D Converter LEDs of MODICOM 3/1 (1000000) as shown below:



6. Carefully turn the DC1 amplitude control slowly clockwise until 1 00000 1 appears on the A/D Converter LEDs, as shown below:



7. Switch the MODE select keys on both MODICOM 3/1 and MODICOM 3/2 to SLOW.

9. Compare the display LEDs of both shift registers on MODIC OM 3/1 and MODICOM 3/2.

You should observe parallel data being loaded into the transmitter shift register. This data is then sent out serially to line and applied to MODICOM 3/2. Once the full CHO or CH1 7-bit word is loaded into the receiver shift register, it is then latched for use by the next stage of the receiver board. (See figure 11 below)

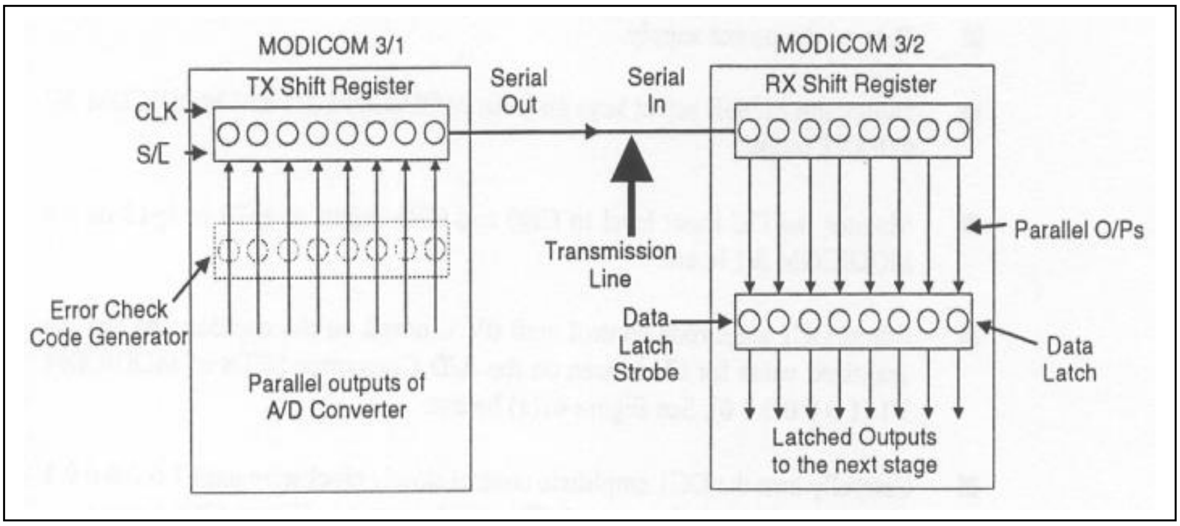


Figure 11: Data Shifting between Transmitter and Receiver

10. Identify TX. TO going high once every 15 seconds - this is bit 0 of the timing frame.

11. In your report, complete the top row of Figure 12 to show the status of both the transmitter and receiver shift register LEDs when bit 0 time period occurs. If an LED is lit, shade in that particular LED on the diagram.

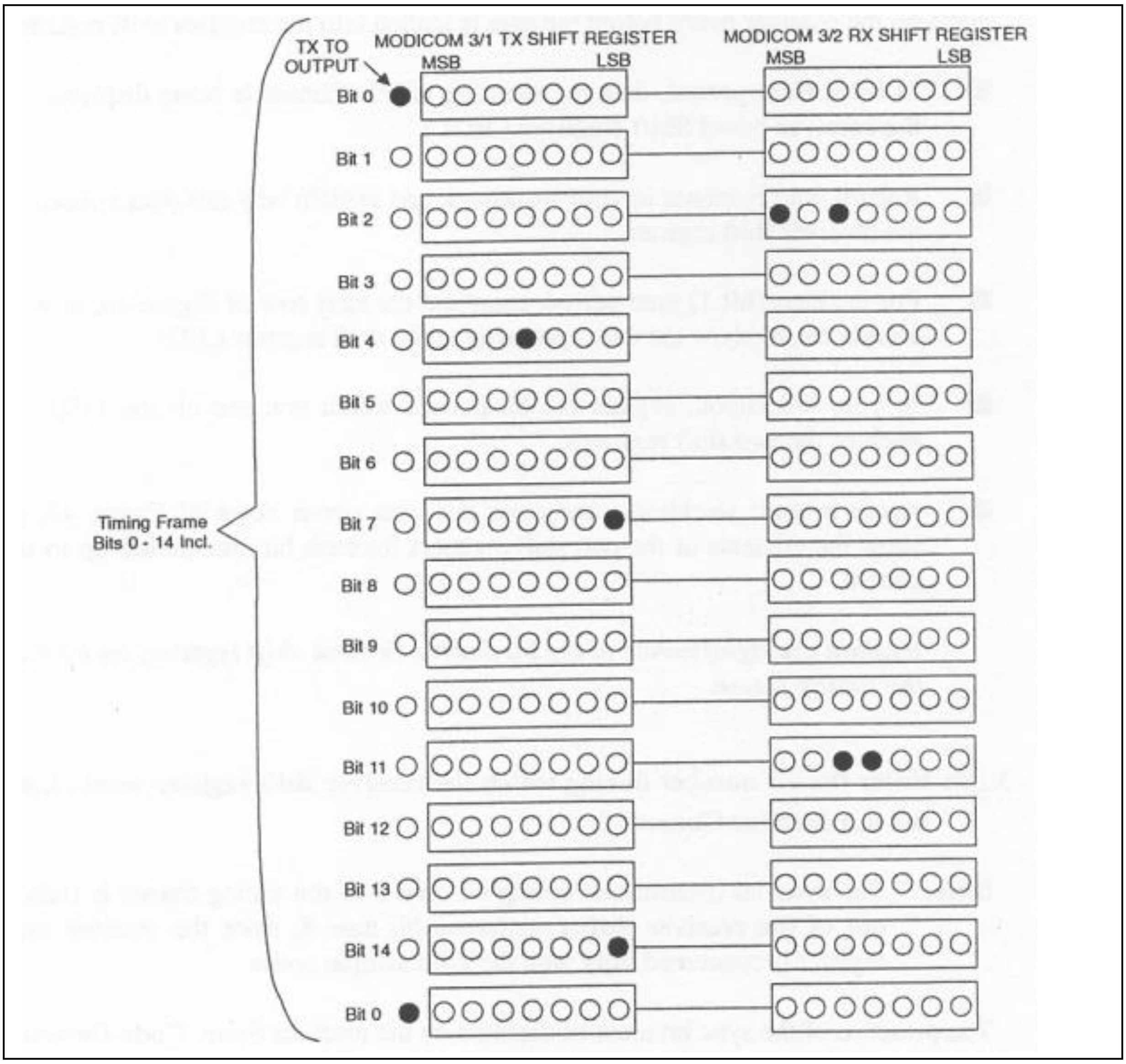


Figure 12: Relative Operations of Transmitter and Receiver Shift Registers

In bit 0 time period, no data is latched into the transmitter shift register since the transmitter board uses this time to send a logic '0' representing bit 0 to line.

Remember that bit 0 is the synchronization (sync) bit, which is zero when the Sync. Code Generator is off.
Serial data, output from the transmitter shift register, is delayed by half a clock cycle by the Output Logic block on the transmitter board, and also by half a clock cycle on the receiver board before the data is loaded into the receiver shift register.

12. For the next (bit 1) time period, complete the next row of Figure 12, in your report, to show the data appearing at the shift register LEDs.
13. In your report, explain the bit pattern which you see on the LEDs of each of the two shift registers.
14. Again in your report, complete the next seven rows of Figure 12, to show the contents of the two shift registers for each bit time period up to bit time 8.

Note: The sync bit (transmitted during bit time 0 of the timing frame) is shifted out of the receiver shift register on bit time 8, since the receiver shift register is concerned only with the 7-bit sample codes.
The presence of the sync bit must be detected by the receiver Sync. Code Detector to provide the receiver with a time reference, as part of the overall system synchronization.

15. In your report complete the next row of Figure 12, to show the data in the shift registers for bit 9 of the timing frame.

Note: No sync bit is inserted between CHO and CHI samples - the sync bit is only transmitted at the start of the timing frame.

16. In your report complete the remainder of Figure 49, up to bit 0 of the next timing frame.
17. Turn **OFF** the power supply.

Part3: Digital to Analog Conversion

1. Set up the connections as in step 1 in part 2

2. On MODICOM 3/1 connect:
   CH0 input to DC1 output
   CH1 input to CH0 input

3. Connect MODICOM 3/1 to MODICOM 3/2 as shown in Table 1 in part1.

4. Turn **ON** the power supply.

5. Set the MODE select switches on both MODICOM 3/1 and MODICOM 3/2 to FAST mode.

6. Connect the channel l input of the oscilloscope to tp30 of MODICOM 3/2.

7. Apply each of the following 7-bit digital codes to the *D/A* converter (by adjusting the transmitter DC! control) and measure the output voltage on the oscilloscope (or the multi-meter) and fill table 2 below.

8. Do your results agree with those obtained in the last experiment for A/D conversion?

DIGITAL CODE	MEASURED VOLTAGE
1 1 1 1 1 1 1	
1 1 1 0 0 1 1	
1 1 0 0 1 1 0	
1 0 1 1 0 0 1	
1 0 0 1 1 0 1	
1 0 0 0 0 0 0	
0 1 1 0 0 1 1	
0 1 0 0 1 1 0	
0 0 1 1 0 0 1	
0 0 0 1 1 0 0	
0 0 0 0 0 0 0	

Table2

~ T*he quantization step size in a PCM system is the difference between adjacent quantization levels.*
It is calculated by finding the voltage difference between the maximum and minimum quantization levels, and then dividing by the total number of quantization levels.
Assuming a difference of approximately lOV between the maximum measured voltage (corresponding to the digital code 1 1 1 1 1 1 1) and the minimum measured voltage (corresponding to the code 0 0 0 0 0 0 0), what is the quantization step size in the MODIC OM 3 PCM system?

Part4: Reconstruction of the Analog Waveform from PAM Samples

The receiver has now processed the incoming PCM serial bit stream into a stream of PAM samples. However, at this point in the receiver process, these samples are time division multiplexed. The next stage in the recovery of the original *information* signals is to demultiplex this waveform into two separate streams of PAM samples.

This is a process which you have investigated on the MODICOM 2 board.

Once demultiplexing has occurred, the two separated PAM signals can be passed to the demodulator circuits which consist of low pass filters.
You will recall from the spectrum analysis of a PAM signal that low pass filtering is a valid method of recovering the original information signals.

After the storm comes sunshine
Good luck

Experiment Eight
ASK – FSK (Modulation & Demodulation)

8.1 Objectives

- Describe the main characteristics of the NRZ forms of data conditioning.
- Recognize the operation of the basic components used in the construction of modulation systems.
- Outline the operation and characteristics of an Amplitude Shift Key modulator (ASK) and demodulator.
- Show how a Frequency Shift Key modulator (FSK) and demodulator can be achieved and account for its main characteristics.

8.2 Data Conditioning

MODIC OM 3/1 converts the input signal in use to a stream of digital data. The data must then be sent over some form of transmission path, and since transmission paths differ in their characteristics, the data must be presented in a form which makes the best use of the transmission path characteristics. For example, if a particular transmission method cannot pass a DC voltage level, then it is of no benefit to use a system of data transfer that relies on DC voltage levels.
It is due to problems like this that have resulted in the development of a selection of different ways of preparing the data before it is transmitted. This preparation process is called 'Data Conditioning'.

In this experiment we will discuss NRZ (L) Non Return –to- Zero (Level) method.

8.2.1 NRZ (L) Non Return –to- Zero (Level) method.

This is the simplest form of data representation. The signal goes high for one clock pulse to represent ***logic 1*** and goes low to represent ***logic 0***. See figure 1.

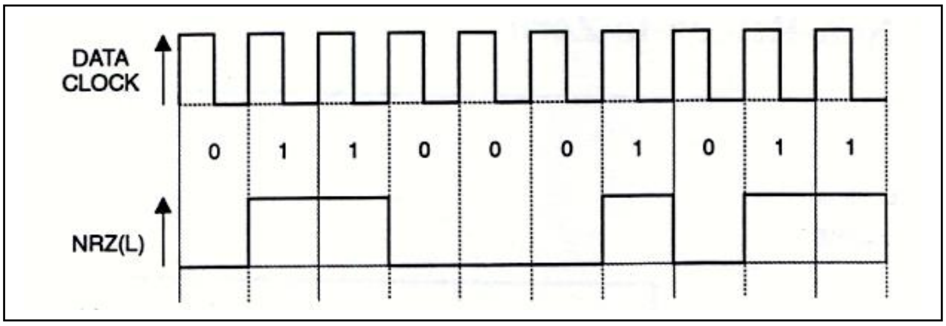


Figure 1: NRZ (L)

8.2.1.1 Receiver Clock

If the receiver has to derive its own clock and has to keep it in step with the transmitter clock, it must use the changes of data to provide some timing information. A data stream consisting of only O's or only 1 's will leave the receiver without these timing clues and will make the clock design slightly more complex.

8.2.1.2 Bandwidth

Imagine we wanted to send one complete 1, 0 cycle of data. We would need one complete clock pulse to send the logic 1 and another to send the logic O. Therefore two complete clock cycles are needed to send one cycle of data. So, the maximum rate of data transfer is half the rate of the clock pulse. You will also see, as we look at the other forms of conditioning, that the NRZ (L) method requires only a comparatively narrow bandwidth. The simplicity of the data waveform and the relatively narrow bandwidth, despite the drawbacks mentioned, make it a popular choice in many communication systems.

8.3 Modulation

If we wish to send our digital signals from one place to another, we have to choose a method of transmission.
What method have we used so far?
We have used the simplest possible method - we just connected the boards with apiece of wire. For very short connections there is nothing wrong with this. There is a lot to be said for using the simplest possible method - providing it will do what we want.
There are some situations where the simple wire would not be satisfactory connections to aircraft, or ships for example.
When the direct wire connection is not acceptable, we often use radio transmission. It is not possible to send the logic levels directly by means of a radio transmitter – it just would not work. The problem is that in order to make the radio work, we have to use very high frequencies, often much higher than the rate at which we send logic signals.

The answer is to combine two different signals - one to allow the radio system to operate which we call a carrier and the digital data that we wish to send.
The process of combining frequencies is called modulation. At the receiver we have to separate the two signals by a process called demodulation. We finish up with the high frequency 'carrier' signal which we discard and the digital information, which we keep.

We are going to study two methods of modulation which are of particular interest in the transmission of data signals: Amplitude Shift Keying and Frequency Shift Keying

Modulation with carriers is also used in telephone networks. It allows many different data streams to be transmitted down the same cable. The modulation and demodulation processes are exactly the same as for a radio link.

8.3.1 Amplitude Shift Keying (ASK) Modulation

This first method of modulation is very simple. *The carrier wave is switched ON every time the digital data is at logic 1 and it is switched OFF whenever the data is at logic O.* See Figure 2.

The modulator circuit has two inputs (See Figure 3):

- The data to be transmitted
- The high frequency carrier wave.

The carrier wave is a sine-wave since any other waveform would increase the bandwidth of the resulting transmission without, in any way, improving it.

The use of a carrier system effectively doubles the required bandwidth for the system.

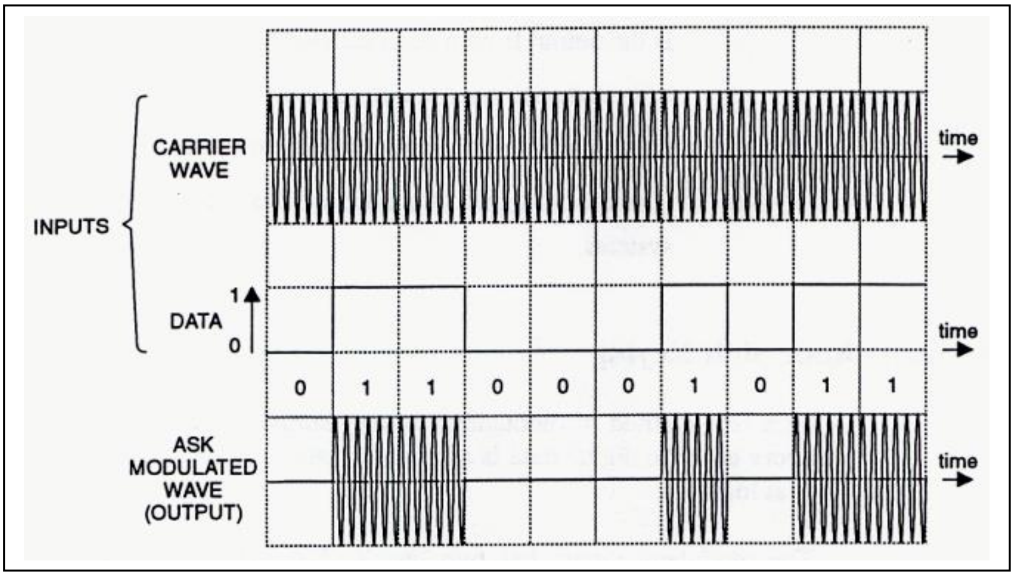


Figure 2: ASK Modulation

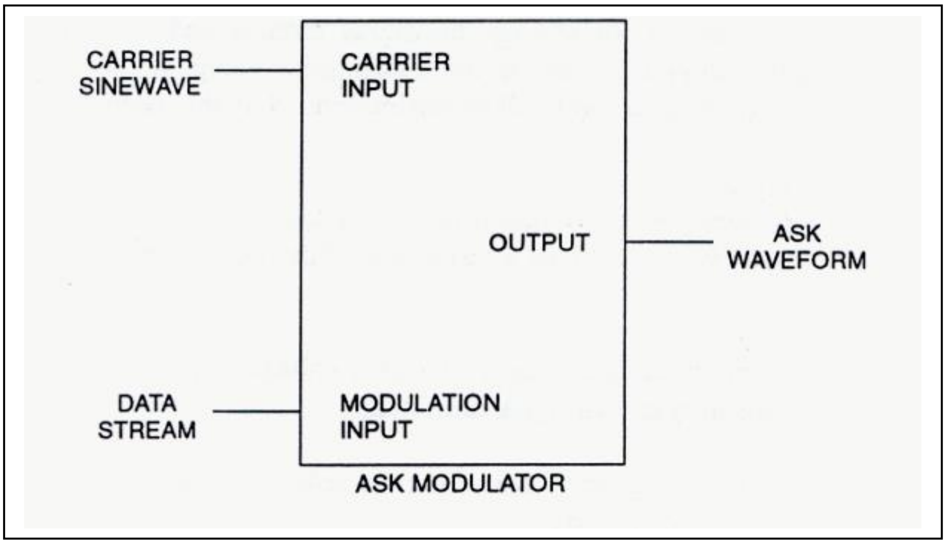


Figure 3: ASK Modulator

At the receiver, we have to extract the wanted data through the following steps:

- **Step 1:** The first step is to rectify the signal. This will give a positive going signal but it will still contain unwanted carrier wave components and the waveform will be rather too rounded for use by the digital circuits and of unreliable amplitude due to attenuation and noise in the transmission route. If the amplitude changes by two much, the data level will be misinterpreted by the receiver.

- **Step 2:** The next step is to pass it through a low pass filter to remove the remnants of the carrier wave. It is now a series of slightly rounded pulses of unreliable amplitude.

- **Step 3:** The next step is to pass it through a voltage comparator to square up the signal and to ensure known signal amplitude. It is now a square wave of known voltage levels in fact, hopefully, a true copy of the original input data.

Figure 4 shows the waveforms after each of the above steps.

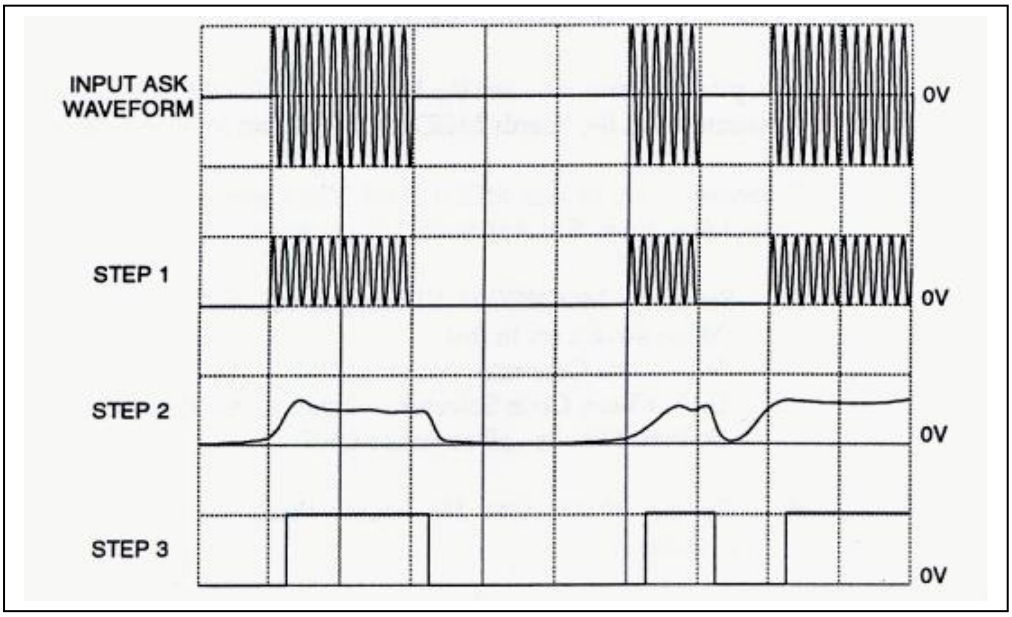


Figure 4: The Waveforms after Each Level of Reconstruction

The 'unreliable amplitude' mentioned in step 1 makes it difficult to be certain, under adverse conditions, about the correct decoding of the logic levels. For this reason, this system is not popular for use in telephone networks.

8.3.2 Frequency Shift Keying (FSK) Modulation

We have seen how the ASK modulation used the logic levels in the data to control the amplitude of carrier wave.

DATA = 1	AMPLITUDE HIGH
DATA = 0	AMPLITUDE LOW

Now, we look at another system, FSK, in which we use the logic levels to control the frequency of the carrier.

DATA = 1	FREQUENCY HIGH
DATA = 0	FREQUENCY LOW

This is what it looks like.

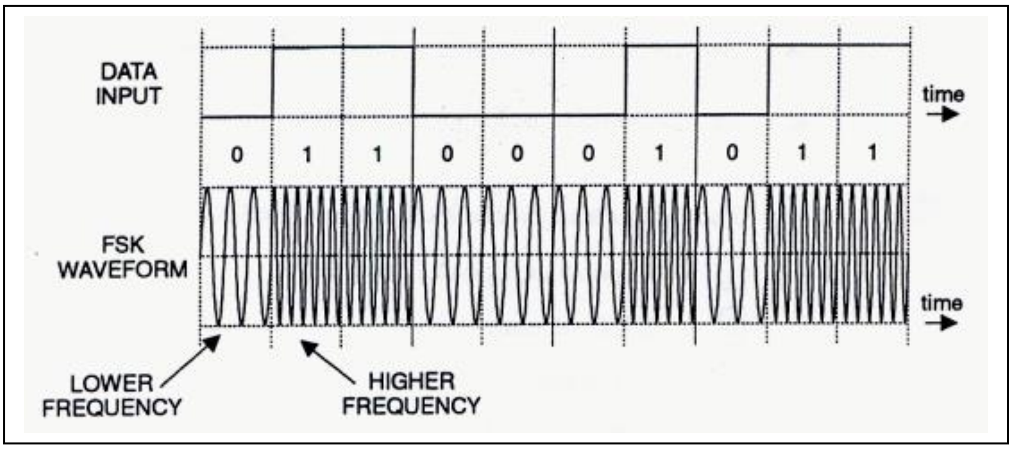
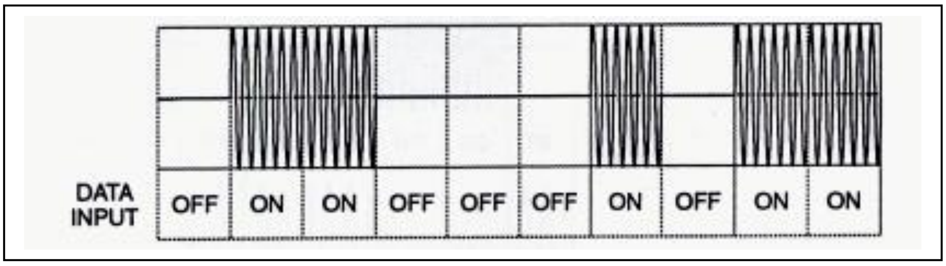


Figure 4: FSK Modulation

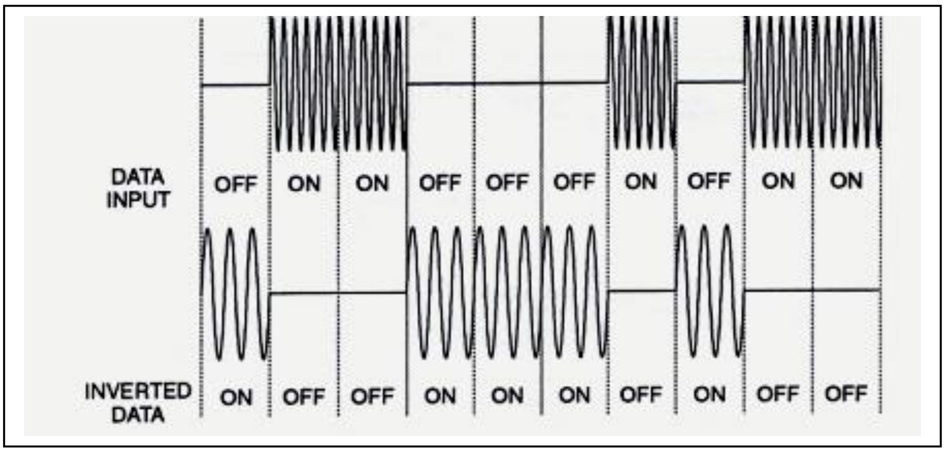
There are many different ways of generating an FSK waveform, one way is to treat it as the combination of two different ASK waves.
Let us assume that we apply the above data stream to an ASK modulator using the higher frequency as the carrier. This is the resulting output:



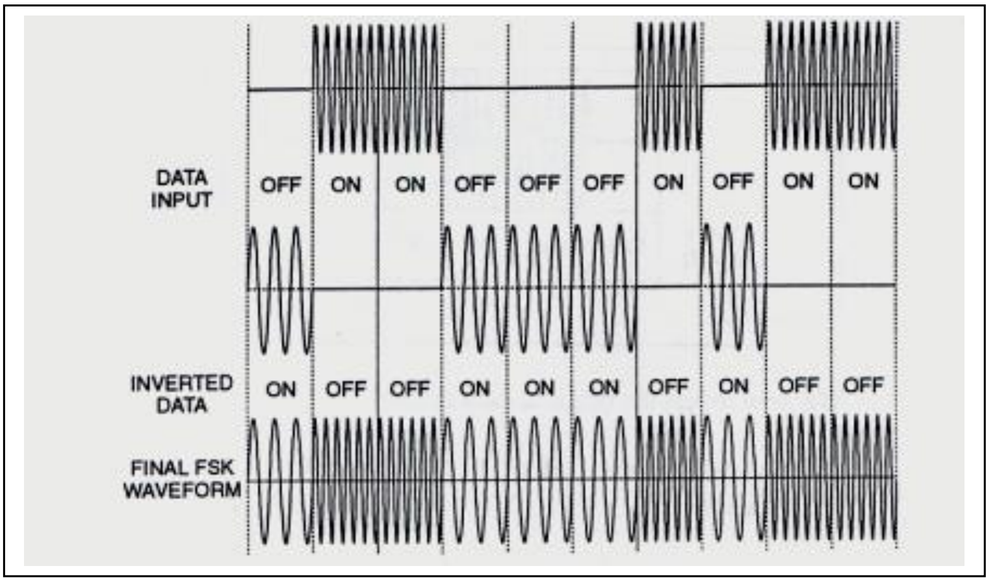
We now invert the original data stream.

Original data stream 0 1 1 0 0 0 1 0 1 1
Inverted stream   1 0 0 1 1   1 0 1 0 0

If we now apply the inverted data stream to an ASK modulator using a lower frequency for the carrier we will get the original data 0 filled with a lower frequency carrier.



The last stage in generating FSK waveform is very simple- we just have to add the two waveforms together.



The block diagram of an FSK modulator will not be a surprise. The summing amplifier is used to add the two ASK waveforms.

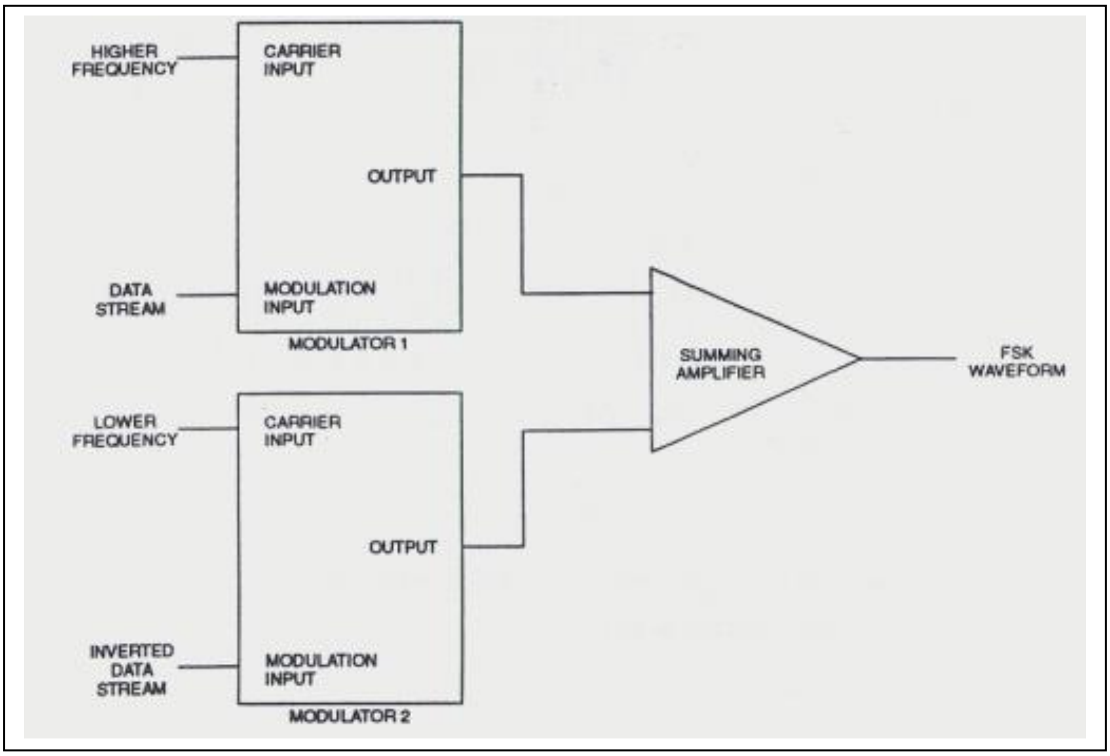


Figure 5: FSK Modulator

The benefit of FSK compared with ASK is that the reliability in terms of data accuracy at the receiver is higher but there is always a price.
The price in this case is that the bandwidth required is greater. The actual increase depends on the two frequencies used. For a given data rate, the higher the frequencies, and the more they differ from each other, the wider the required bandwidth. Compared with ASK the bandwidth is likely to be at least double. This means less communication channels in a given band of frequencies.

<u>At the receiver, we have to extract the wanted data through the following steps:</u>

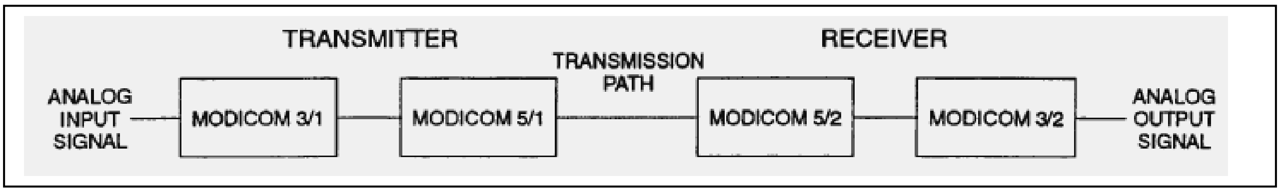
- **Step 1:** The first step is to convert the change of frequency into a change in DC voltage level using the phase- locked loop (PLL). This is just what we want. The original data generated two different frequencies to be transmitted. At the receiver the PLL can be used to recover the two different voltage levels contained in the data stream.

- **Step 2:** The next step is to pass it through a low pass filter to remove the remnants of the carrier wave. It is now a series of slightly rounded pulses of unreliable amplitude.

- **Step 3:** The next step is to pass it through a voltage comparator to square up the signal and to ensure known signal amplitude. It is now a square wave of known voltage levels in fact, hopefully, a true copy of the original input data.

8.4 An Overview of the System

The boards may look complicated and many of the words and initials will be unfamiliar to you. Try not to worry about it. We are only going to tackle one section at a time.



MODICOM 3/1 accepts an analog input and provides a PCM data stream as an output. The PCM data stream from the MODICOM 3/1 board is, in real life, sent over a variety of different transmission paths to the receiver on MODICOM 3/2. For example, it may be sent by a radio system or perhaps by a wire or fiber - optic link. These transmission paths often have different characteristics and so the PCM data stream has to be modified or 'Prepared for transmission' "as mentioned before" by coding the signal and modulating it. This preparation is the function of the MODICOM 5/1 board.
Whatever changes we have made in MODIC OM 5/1 have to be undone in MODICOM 5/2. This ensures that a copy of the original PCM data stream is available to the MODICOM 3/2 board.
The MODICOM 3/2 board is really just the inverse of the MODICOM 3/1 board and extracts the analog information from the PCM data stream.

8.4 MODICOM 5/1 Board

MODICOM 5/1 board can be considered as six separate blocks (figure 6):

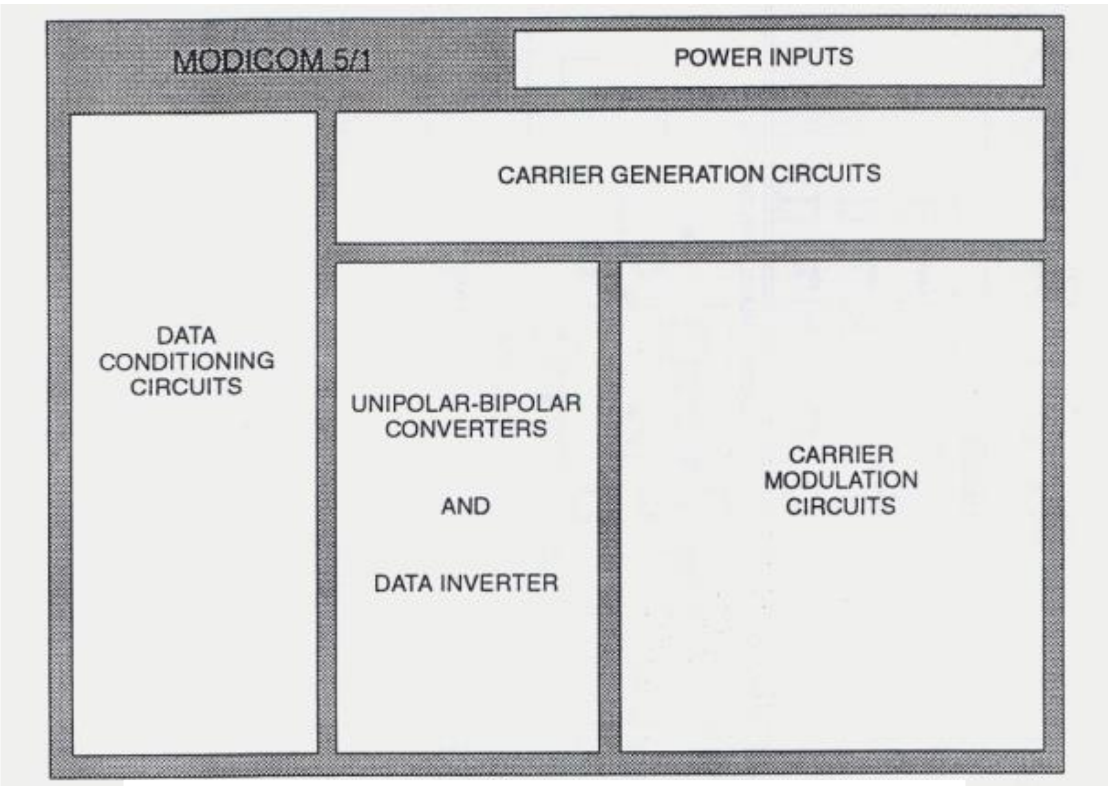


Figure 6: MODICOM 5/1 Board

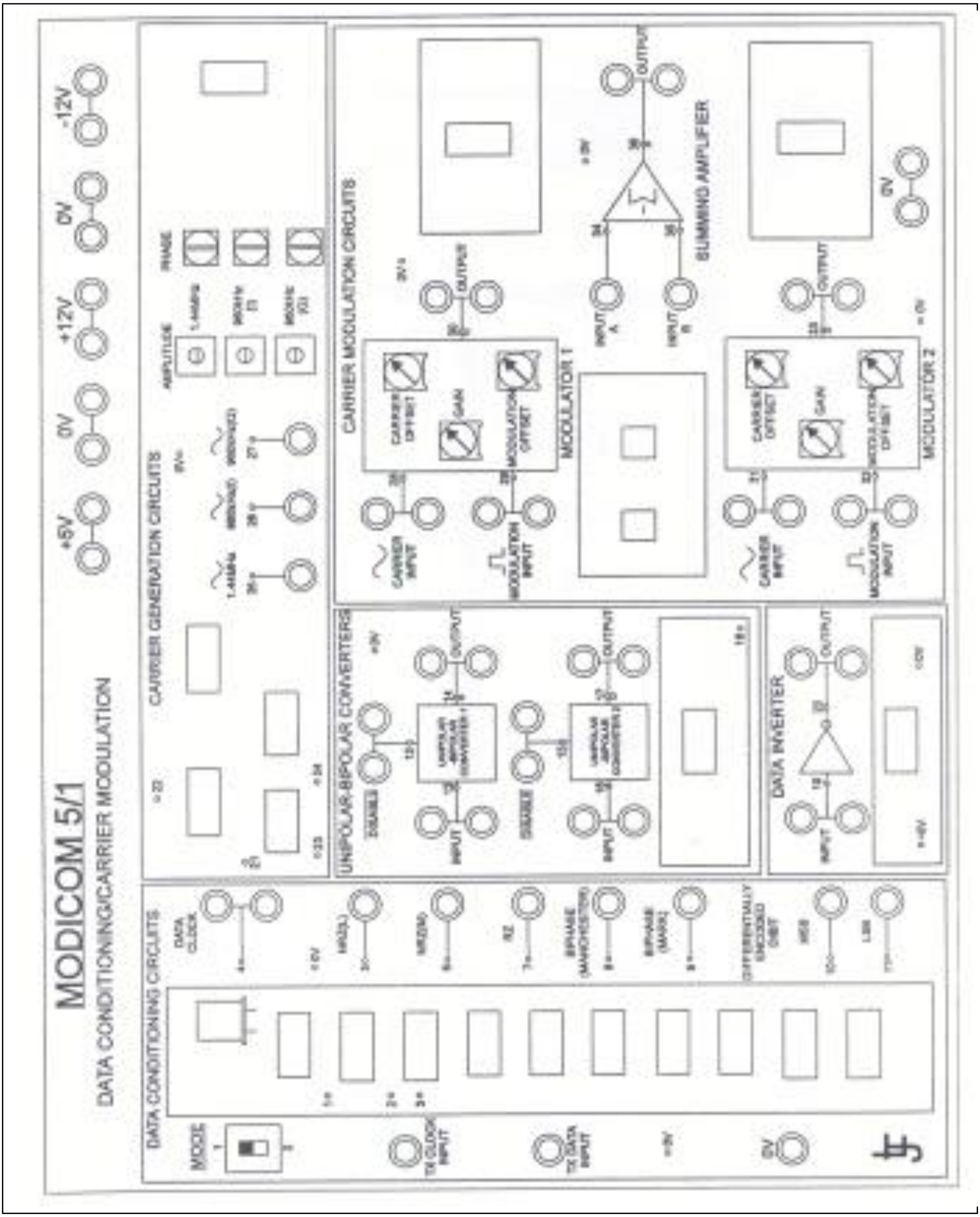
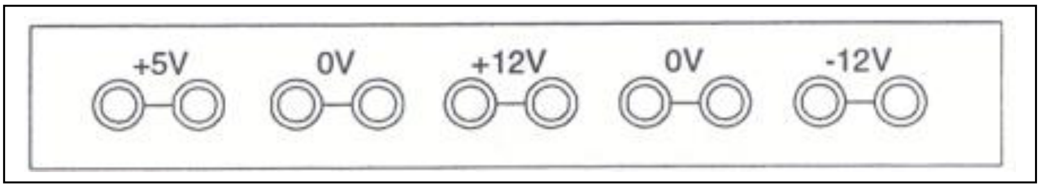


Figure 7: MODICOM 5/1 Board

8.4.1 Power input

These are the electrical input connections necessary to power the module.



8.4.2 Data Conditioning Circuits

The PCM data stream from MODICOM 3/1 board is converted by these circuits into the form best suited to the transmission system in use (line coding).

8.4.3 Unipolar-Bipolar Converter and Data Inverter Circuits

This area of the board simply chages the acual voltage levels of the digital signals

8.4.4 Carrier Generator Circuit

For some transmission system like radio links, it's necessary to have a high frequency carrier signal. This section produces two different frequency sine-waves for this purpose – 1.44 MHz and 960 KHz.

8.4.5 Carrier Modulation Circuits

To combine the carrier with the digital information to be transmitted, we must have a special circuit called a modulator. In this block we have two modulators which allow us, if we wish, to send two different signals along the same transmission path. This is achieved by time division multiplexing.

8.5 MODICOM 5/2 Board

MODICOM 5/2 board can be considered as five separate blocks (figure 8):

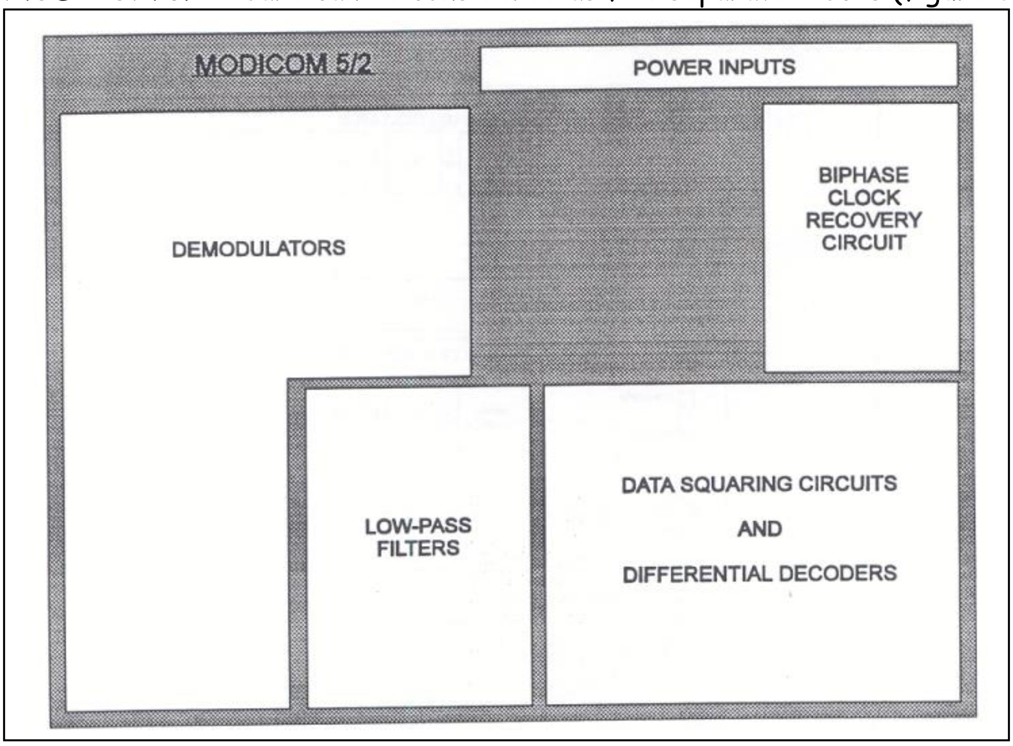


Figure 8: MODICOM 5/2 Board

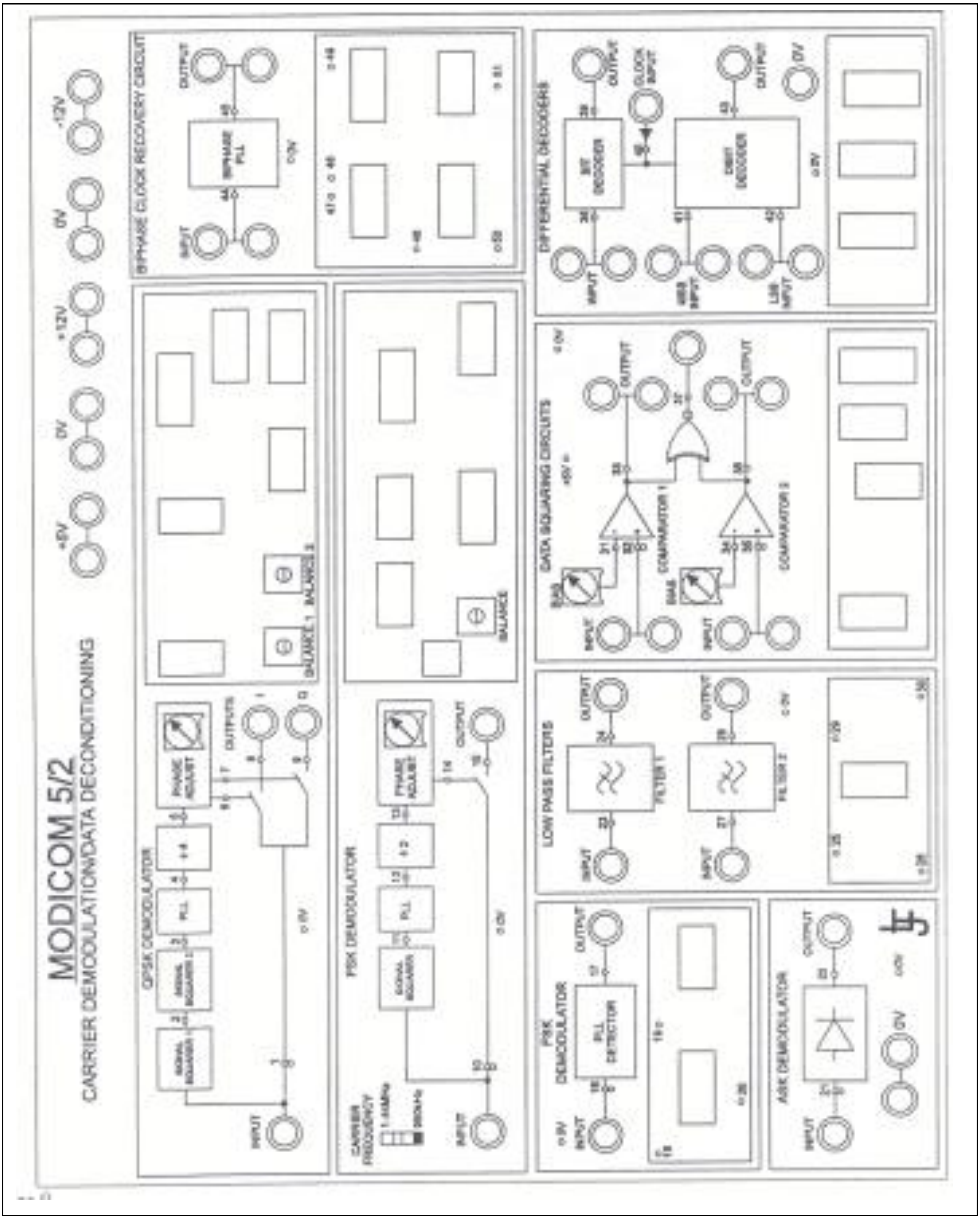
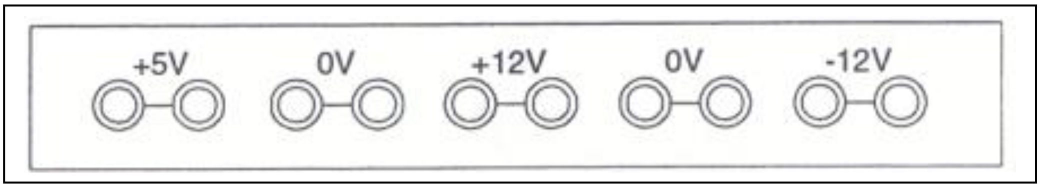


Figure 9: MODICOM 5/2 Board

8.5.1 Power input

These are the electrical input connections necessary to power the module.



8.5.2 Demodulators

In MODICOM 5/1 we combined the required signal with a carrier for transmission purposes. In fact, MODICOM 5 can demonstrate four different modulation methods. Having passed the signal along the transmission path, we have to recover the information at the other end. This is the job of the demodulators.

8.5.3 Low Pass Filters

The output from the demodulator includes the wanted signals and some unwanted high frequencies – these must be removed. This is the purpose of the low-pass filter.

8.5.4 Data Squaring and Differential Decoder Circuits

Following the demodulation process, there are one or two small jobs to be done to clean up and prepare the signals so that they are in standard PCM form to be acceptable to the MODICOM 3/2 board.

8.5.5 Biphase Clock Recovery Circuit

In the data conditioning circuit on MODICOM 5/1, you will see outputs called Biphase (Manchester) and Biphase (Mark). The significance of the biphase part is that these forms require two different phases of clock to decipher it. The purpose of the Biphase Clock Recovery Circuit is to generate this special form of clock.

8.6 Procedure

Part (1-a): Amplitude Shift Keying Modulation

This practical exercise uses the boards MODICOM 3/1 and MODICOM 5/1 as the transmitter and the boards MODICOM 5/2 and MODICOM 3/2 as the receiver.

1- Set up the MODICOM 3/1 board as follows:

- Mode switch set to fast
- Sync Code Generator switch to OFF
- Error Check Code Selector switches to A = 0 and B = 0
- Switched Faults - all switched OFF.

2- On the MODICOM 5/1 board: check that the MODE switch is set to position 1.

3- Set up the MODICOM 3/2 board as follows:

- Mode switch set to fast.
- Pulse Generator Delay Adjust - fully clockwise.
- Sync Code Detector switch to ON.
- Error Check Code Selector switches to A = 0 and B = 0.
- Switched Faults - all switched OFF.

4- Connect the +5 volt, +12 volt and the -12 volt supplies to the boards and link their zero volt connections.

5- Make the following connections:

- MODIC OM 3/1 Tx Clock Output tp3 to MODICOM 5/1 Tx Clock Input.
- MODICOM 3/1 Tx Data Output tp44 to MODICOM 5/1 Tx Data Input.

6- On MODICOM 3/1, connect the DC1 output on the function generator block to the Channel 0 input (tp10) and then connect a wire from this point to Channel 1 input (tp12).
We can now supply the same DC voltage level to each of the channels.
Remember that the two channels are going to be multiplexed by TDM.

7- Switch on the power supply.

8- Adjust the DC1 preset until the A/D Converter LEDs show the value:

D6	D5	D4	D3	D2	D1	D0
0	1	0	0	0	1	1

9- In MODICOM 5/1:

- Use your oscilloscope's CH1 to observe the Data Clock output at tp4.
- Use your oscilloscope's CH2 to observe the NRZ(L) waveform at tp5.

- Draw the two signals on the same graph.

Note:
- The least significant bit D0 is transmitted first and then through to D6.
- You will see that there is a logic 0 being transmitted ahead of the first bit so from left to right on the oscilloscope screen, the logic levels read: 0, then 1 1 0 0 0 1 0.

10- Switch off the power supply.

11- Make the following connections: (See figure 10)
<u>On MODICOM 5/1:</u>

- NRZ(L) output tp5 to ASK Modulator input tp29
- 1.44MHz Carrier tp25 to Carrier input tp28.

MODICOM 5/1 to MODICOM 5/2

- Modulator 1 output tp30 to ASK Demodulator input tp21.

On MODICOM 5/2:

- ASK Demodulator output tp22 to Low Pass Filter 1 input tp23
- Low Pass Filter 1 output tp24 Comparator 1 input tp32.

MODICOM 5/2 to MODICOM 3/2

- Comparator 1 output tp33 to Rx Data Input tp1.

On MODICOM 3/2:

- Rx Data input tp1 to Clock Regeneration Circuit input tp3
- Clock Regeneration Circuit output tp8 to Rx Clock input tp46.

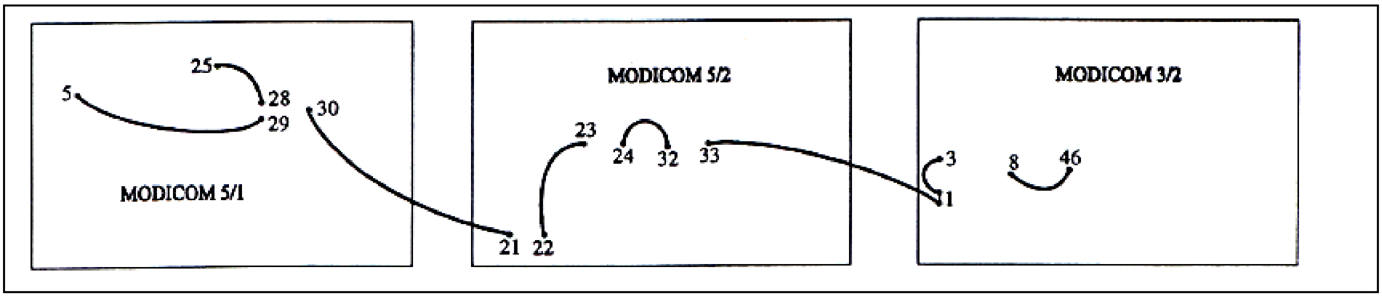


Figure 10: Connection of Step11

12- Draw the output waveform at tp30. You should now have an ASK waveform similar to that in Figure 2.

- There are three presets associated with this circuit that may need adjusting.
(Ask your lab-supervisor)

- Gain. Adjust this to give a 2 volt peak-to-peak voltage waveform.
- Modulation Offset. Adjust this to give minimum signal level during the OFF states.
- Carrier Offset. This ensures that the OFF level is halfway between the positive and negative peaks.

Part (1-b): Amplitude Shift Keying Demodulation

1 - To see the demodulation process, monitor the following waveforms and draw the signal at each stage:

- STEP 1: The ASK Modulator Output tp30.
- STEP 2: The ASK Demodulator Output tp22.
- STEP 3: The LOW Pass Filter 1 Output tp24.
- STEP 4: The Comparator 1 Output tp33.

Note: The COMPARATOR 1 bias level should be adjusted by the preset to give the correct pulse width at the output. **(Ask your lab-supervisor)**

~ The Complete ASK System

1- On MODICOM 3/1:

- Switch ON the Sync Code Generator and note that its A/D converter is now copied on the D/ A converter of MODICOM 3/2.
- Disconnect the inputs to Channel 0 and Channel 1 from the DC1 input and connect, instead, one channel to the 1 kHz signal and the other channel to the 2 kHz signal.

2- Use your oscilloscope to monitor both channel outputs on MODICOM 3/2 (tp33 and tp36). Notice that, once again, the two channels are quite independent - there is no interference between the two waveforms and the amplitudes are each controllable by turning the associated presets.

Part (2-a): Frequency Shift Keying Modulation

This practical exercise uses the boards MODICOM 3/1 and MODICOM 5/1 as the transmitter and the boards MODICOM 5/2 and MODICOM 3/2 as the receiver.

1- Set up the MODICOM 3/1 board as follows:

- Mode switch set to fast
- Sync Code Generator switch to OFF
- Error Check Code Selector switches to A = 0 and B = 0
- Switched Faults - all switched OFF.

2- On the MODICOM 5/1 board: check that the MODE switch is set to position 1.

3- Set up the MODICOM 3/2 board as follows:

- Mode switch set to fast.
- Pulse Generator Delay Adjust - fully clockwise.
- Sync Code Detector switch to ON.
- Error Check Code Selector switches to A = 0 and B = 0.
- Switched Faults - all switched OFF.

4- Connect the +5 volt, +12 volt and the -12 volt supplies to the boards and link their zero volt connections.

5- Make the following connections:

- MODIC OM 3/1 Tx Clock Output tp3 to MODICOM 5/1 Tx Clock Input.
- MODICOM 3/1 Tx Data Output tp44 to MODICOM 5/1 Tx Data Input.

6- On MODICOM 3/1, connect the DC1 output on the function generator block to the Channel 0 input (tp10) and then connect a wire from this point to Channel 1 input (tp12).

We can now supply the same DC voltage level to each of the channels.
Remember that the two channels are going to be multiplexed by TDM.

7- Switch on the power supply.

8- Adjust the DC1 preset until the A/D Converter LEDs show the value:

D6	D5	D4	D3	D2	D1	D0
0	1	0	0	0	1	1

9- Switch off the power supply.

11- Make the following connections: (See figure 11)

On MODICOM 5/1:

- NRZ(L) output tp5 to Modulation Input tp29
- Modulation input tp29 to Data Inverter input tp19
- Modulator output tp30 to Summing Amplifier input tp34
- 1.44MHz carrier tp25 to Carrier input tp28
- Data Inverter output tp20 to Modulation input of Modulator 2 tp32
- 960 MHz(1) Clock tp26 to Modulator 2 Carrier input tp31
- Modulator 2 output tp33 to Summing Amplifier input tp35.
- MODICOM 5/1 to MODICOM 5/2
- Summing Amplifier output tp36 to PLL Detector input tp16.

On MODICOM 5/2:

- PLL Detector output tp17 to Low Pass Filter 1 input tp23
- Low Pass Filter 1 output tp24 to Comparator 1 input tp32.
- MODICOM 5/2 to MODICOM 3/2
- Comparator 1 output tp33 to Rx Data Input tp1.

On MODICOM 3/2:

- Rx Data input tp1 to Clock Regeneration Circuit input tp3
- Clock Regeneration Circuit output tp8 to Rx Clock input tp46.

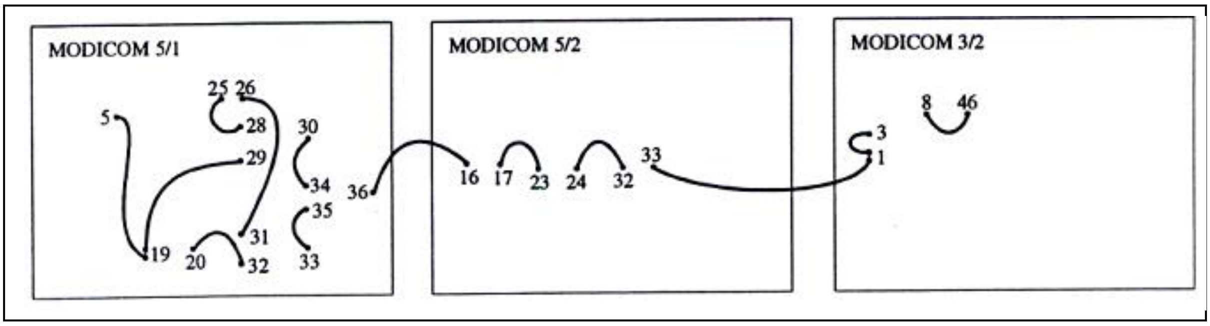


Figure 10: Connection of Step11

12- Use your oscilloscope to compare the data at input to Modulator 1 tp29 and the inverted version on the Data input to Modulator 2 tp32.

13- Use your oscilloscope to compare the ASK waveforms at:

- Modulator 1 output tp30 this is the data 1 code
- Modulator 2 output tp33 this is the data 0 code

Notice the two different frequencies.
There are three presets associated with this circuit that may need adjusting.
(Ask your lab-supervisor)

- Gain. Adjust this to give a 2 volt peak-to-peak voltage waveform.
- Modulation Offset. Adjust this to give minimum signal level during the OFF states.
- Carrier Offset. This ensures that the OFF level is halfway between the positive and negative peaks.

14- Use your oscilloscope to compare the NRZ (L) input tp5 with the Summing Amplifier output tp36 to see a complete FSK transmitter. Draw the two signals on the same graph. You should now have an ASK waveform similar to that in Figure 4.

Part (2-b): Frequency Shift Keying Demodulation

1 - To see the demodulation process, monitor the following waveforms and draw the signal at each stage:

- STEP 1: The FSK Modulator Output tp36.
- STEP 2: The PLL Output tp17.
- STEP 3: The LOW Pass Filter 1 Output tp24.
- STEP 4: The Comparator 1 Output tp33.

Note: The COMPARATOR 1 bias level should be adjusted by the preset to give the correct pulse width at the output. **(Ask your lab-supervisor)**

~ The Complete FSK System

1- On MODICOM 3/1:

- Switch ON the Sync Code Generator and note that its A/D converter is now copied on the D/ A converter of MODICOM 3/2.
- Disconnect the inputs to Channel 0 and Channel 1 from the DC1 input and connect, instead, one channel to the 1 kHz signal and the other channel to the 2 kHz signal.

2- Use your oscilloscope to monitor both channel outputs on MODICOM 3/2 (tp33 and tp36). Notice that, once again, the two channels are quite independent - there is no interference between the two waveforms and the amplitudes are each controllable by turning the associated presets.

The who desires the top must sit up many times
Good luck