LJ Technical Systems

An Introduction to Analog Communications

Student Workbook
AT02

MT191/B

DiGiAC

Issue Number: MT191/B
Written by: J Crisp

LJ Technical Systems Ltd.
Francis Way
Bowthorpe Industrial Estate
Norwich. NR5 9JA. England
Telephone: (01603) 748001
Fax: (01603) 746340

LJ Technical Systems Inc.
85, Corporate Drive, Holtsville,
11742-2007, New York, USA.
Telephone: (631) 758 1616
Fax: (631) 758 1788

About this Student Workbook

Introduction

This Student Workbook has been designed to provide you with a record of your learning and achievement. It provides you with:

- A Pre-Test which assesses your understanding of the terms and definitions which are required to complete this AT02 learning program.
- A record of the theory you will learn.
- Grids on which to draw sketches and spaces to record results, as you work through the practical exercises in the Curriculum Manual.
- Notes pages for each chapter covered by the Curriculum Manual. This space allows you to record those personal notes that will help your understanding.

In short, your Student Workbook replaces the notes/handouts you would expect from a formal teaching session, providing the basis for future reference or revision. You should maintain it meticulously if you are to obtain maximum benefit from it during your studies and later.

Computerized Assessment of Student Performance

If your laboratory is equipped with the ClassAct computer managed learning system, then the system may be used to automatically monitor your progress as you work through the Pre-Test in this Student Workbook and the chapters of the Curriculum Manual.

If your instructor has asked you to use this facility, then you should key in your responses to questions at your computer managed workstation.

To remind you to do this, a  symbol is printed alongside questions that require a keyed-in response.

The following D3000 Lesson Module is available for use with the Pre-Test and the CT02 Volume 3 Curriculum Manual:

D3000 Lesson Module 20.12

Getting Started

- You should attempt the Pre-Test that appears opposite. Your tutor may wish to discuss with you the results of the Pre-Test, before you embark on your AT02 studies.

- When your instructor feels that you have the necessary pre-requisite knowledge to begin the AT02 Curriculum Manual, you should turn to Page 1 of the Curriculum Manual to begin your studies.

About the Pre-Test

The following Pre-Test has been designed to assess your understanding of basic electronics. You should undertake this Pre-Test before you embark on your study of the AT02 Curriculum Manual.

Please Note:

- Your Instructor may place a time constraint on this Pre-Test. If this is the case, then you should attempt to complete the Pre-Test within that time.

- You should **attempt all questions**.

- Do **not** begin the Pre-Test until told to do so by your Instructor.

- If you are working in a computer managed laboratory, you should load **Chapter 31** of **Module 20.12** using your computer managed workstation.

  This will allow you to key in your responses to the Pre-Test questions at your workstation.

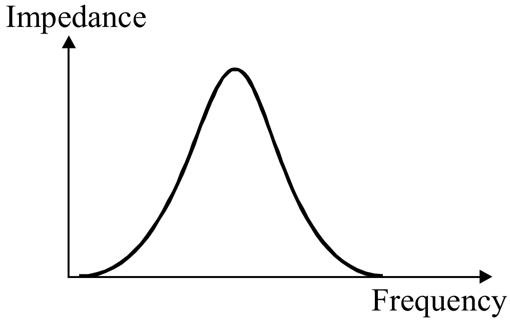
Pre-Test

For each question, select the correct option.

1. The output from an XOR gate is a logic 1 only when the two inputs are at:

- a logic 1.
- b different logic levels.
- c logic 0.
- d the same logic levels.

2. The characteristic shown below is that of:

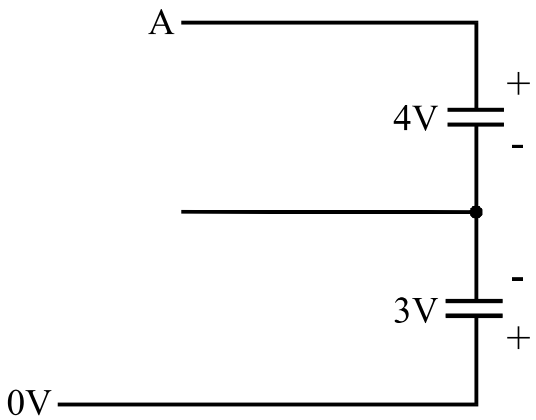


- a a parallel tuned circuit.
- b a C-R series circuit.
- c an L-R series circuit.
- d a series tuned circuit.

3. 450kHz is the same frequency as:

- a 45 000Hz
- b a wavelength of 666m.
- c 0.45MHz
- d 2.2µHz

4. The voltage at point A in the diagram below is:



- a -1V
- b +7V
- c +4V
- d +1V

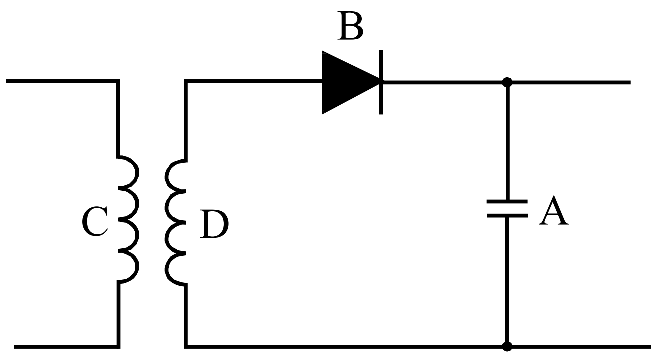
5. Modulation is the process of converting the information to be transmitted into:

- a a form that contains the least bandwidth.
- b a modular system.
- c a form suitable for transmission over the communication system.
- d a signal containing the least number of sidebands.

6. A filter is able to:

- [a] remove either the high frequency components or the low frequency components but not both.
- [b] remove or attenuate certain frequency components from a complex signal.
- [c] extract a square wave from a sinusoidal signal.
- [d] convert signals from an analog form to a digital form.

7. The transformer secondary in the diagram below is labeled:

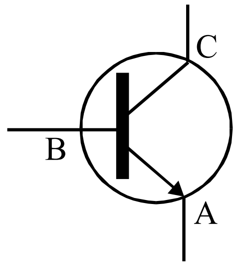


- [a] a
- [b] b
- [c] c
- [d] d

8. If the input signal had a frequency above the frequency of resonance, a parallel tuned circuit would appear to be:

- [a] capacitive.
- [b] resistive.
- [c] resonant.
- [d] inductive.

9. The collector, emitter and base are labeled:



[a] A, C, B respectively.

[b] A, B, C respectively.

[c] B, A, C respectively.

[d] C, A, B respectively.

10. In a series circuit consisting of a capacitor and a resistor, the:

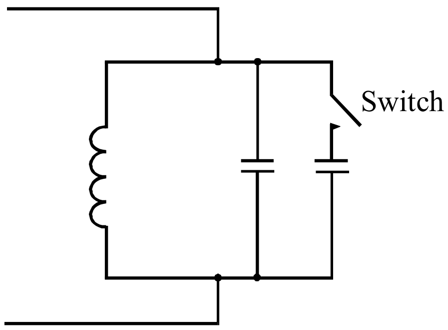
[a] voltage and current are equal.

[b] voltage leads the current in phase.

[c] voltage lags the current in phase.

[d] current is at its minimum value.

11. When the switch is closed, the resonance frequency will:



[a] decrease.

[b] become capacitive.

[c] may not change.

[d] increase.

12. Electrical noise:

a is always a sinusoidal signal.

b is only caused by electrical storms.

c is any unwanted signal present at the output of a system.

d can be eliminated by using a screened coaxial cable.

13. To measure a DC voltage level on an oscilloscope, the input selector can be set to:

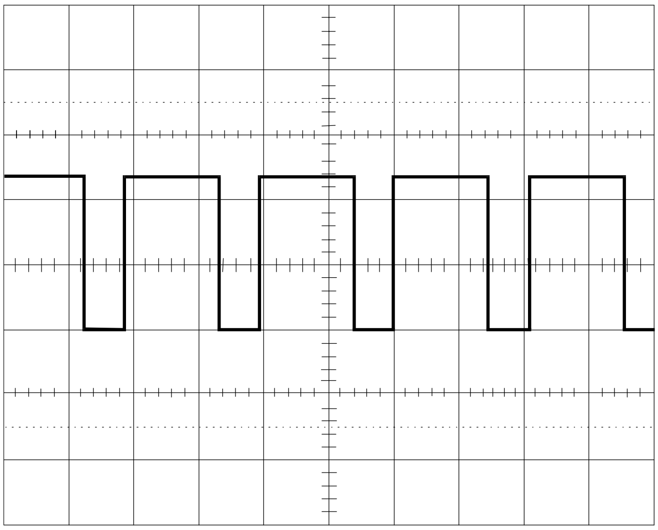
a either AC or DC if the trace line is first adjusted to the central position with the Y-POS. Control.

b DC

c GD

d AC

14. The frequency that is closest in value to that displayed on the oscilloscope is:



Oscilloscope control settings:

Y amplifier: 5mV/div.

trigger: internal +ve

timebase: 0.1ms/div.

a 210μHz

b 1kHz

c 5.4kHz

d 4.8kHz

15. Oscilloscope probes marked as X10 are used to:

- a ‘expand up’ low amplitude signals for easier observation.
- b reduce the loading effect of the oscilloscope on the circuit operation.
- c decrease the circuit noise by a factor of ten.
- d increase the oscilloscope timebase speed by a factor of ten.

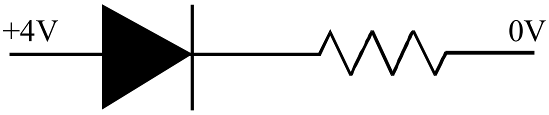
16. A magnetic field is always created by:

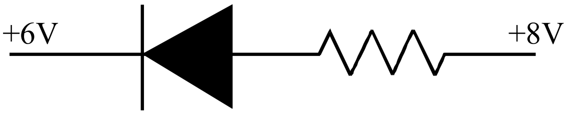
- a a difference in voltage between two points.
- b an electric current.
- c an open circuit.
- d a capacitor.

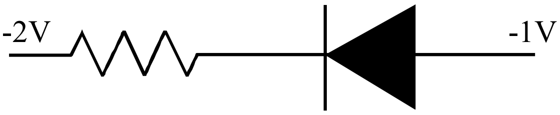
17. A bandstop filter has the effect of:

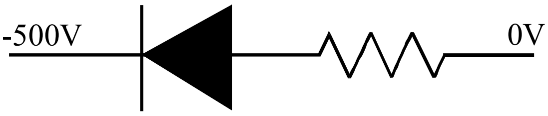
- a allowing only a single frequency to pass through the circuit.
- b attenuating the frequency of 455kHz.
- c attenuating a band of frequencies without affecting higher or lower frequencies.
- d preventing music from being played.

18. A current will flow in which of the following options:

a


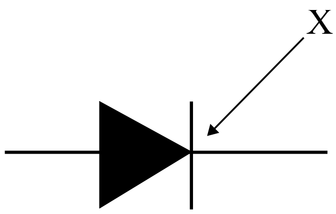
b


c


d


- a only in circuit a.
- b only in circuit b.
- c only in circuits c and d.
- d in all of these circuits.

19. The part of the symbol marked X is called the:



- a anode.
- b emitter.
- c cathode.
- d collector.

20. An oscillator:

a does not require an input signal.

b increases the amplitude of the input signal.

c removes some frequency components from the input signal.

d shifts the phase of an input signal.

21. Impedance is opposition to current flow caused by:

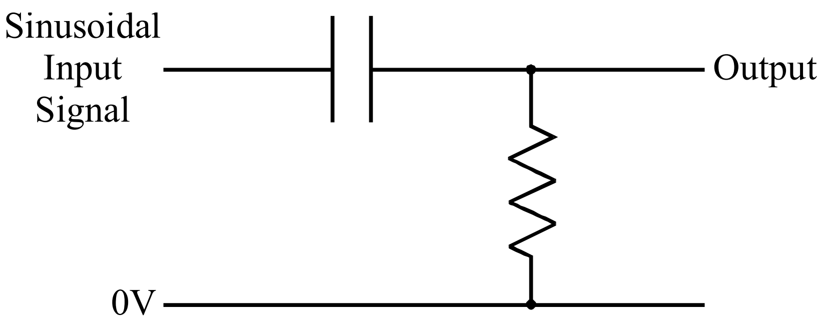
a reactance.

b capacitance.

c a combination of resistance and a reactance.

d resistance.

22. The output voltage in the diagram below will:



a lag the input voltage by an angle less than 90°.

b lead the input voltage by an angle less than 90°.

c be of greater amplitude than the input.

d be distorted.

23. Bandwidth is:

a the quality of the music.

b a broadcast signal.

c the length of the antenna.

d the range of frequency components within a signal.

24. An electric field is said to act:

a from positive to negative.

b from any object to an earth potential.

c in the space around any permanent magnet.

d from north to south.

25. A 10µF capacitor connected in series with a 20nF capacitor would result in a total capacitance:

a greater than 10µF.

b between 20nF and 10µF.

c exactly 10µF.

d less than 20nF.

26. Combining two 10V sinusoidal signals which are 90° out of phase would result in a signal of amplitude:

a 20V

b 14.14V

c 17.32V

d 0V

27. A screened cable:

[a] acts as a form of high pass filter to remove distortion of the waveform being broadcast.

[b] cannot be used for music signals as it would remove all the high frequencies.

[c] has a conducting layer around the signal carrying wire to shield it from interference.

[d] is a conductor that has been placed out of sight.

28. In an NPN bipolar transistor, the base voltage is normally:

[a] more positive than the emitter.

[b] at earth potential.

[c] more positive than the collector.

[d] less positive than the emitter.

29. Two 20Ω resistors connected in parallel would offer a total resistance:

[a] of 40Ω.

[b] of 20Ω.

[c] of 10Ω.

[d] of greater than of 10Ω but less than 20Ω.

30. 20% represents the same proportion as 1 in:

[a] 100

[b] 4

[c] 20

[d] 5

Chapter 1
The ANACOM 1/1 and ANACOM 1/2 Boards

1.1 Layout Diagram of the ANACOM 1/1 Board

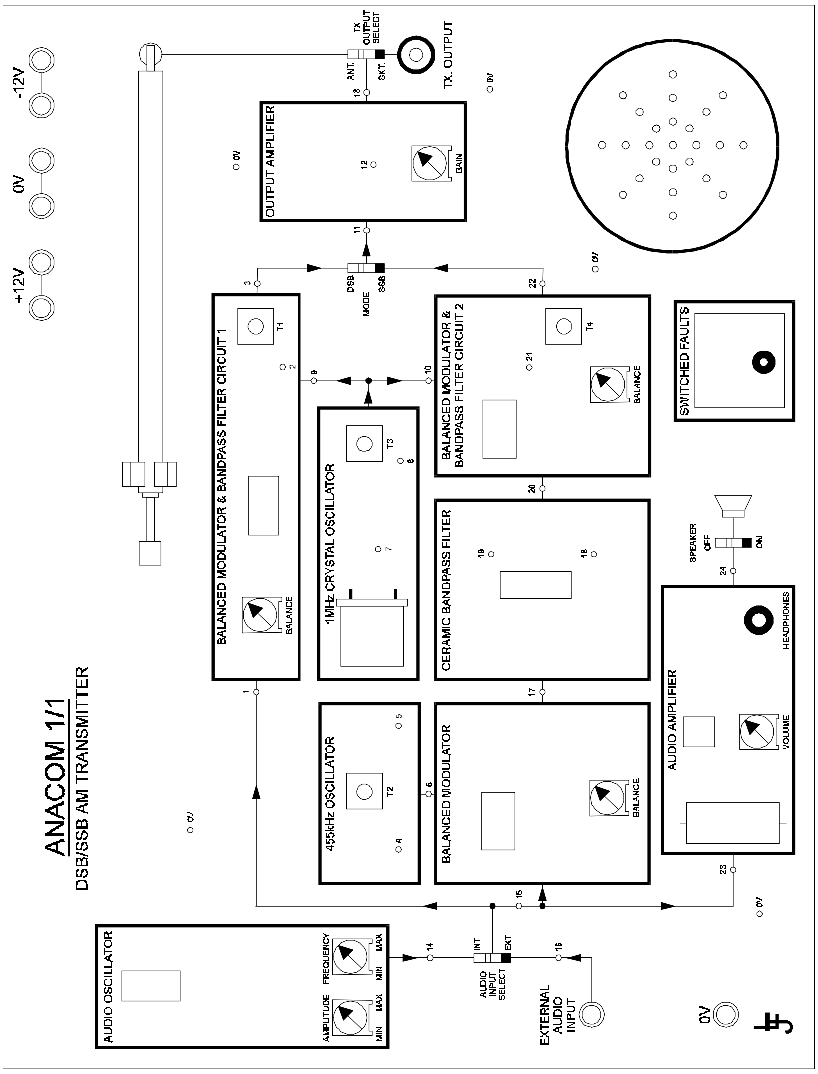


Figure 1

1.2 The ANACOM 1/1 Board Blocks

The transmitter board can be considered as five separate blocks:

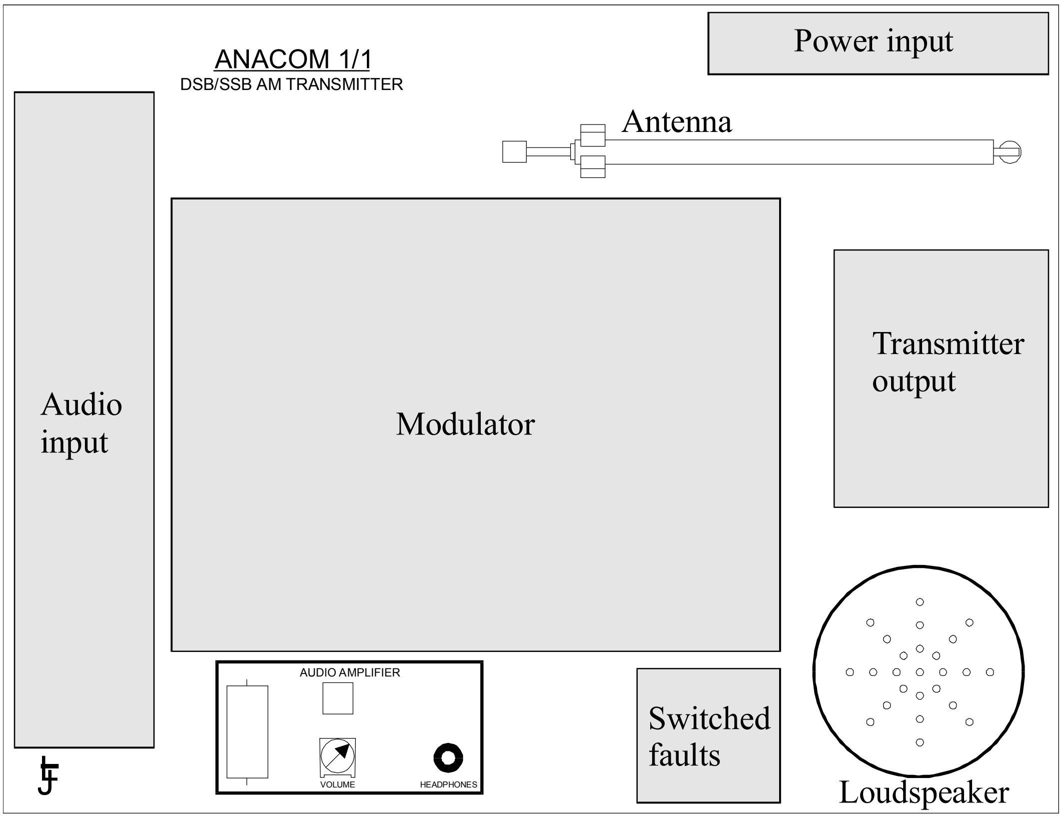


Figure 2

1.3 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.

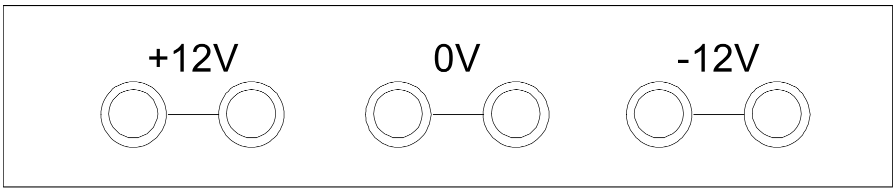


Figure 3

1.4 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.

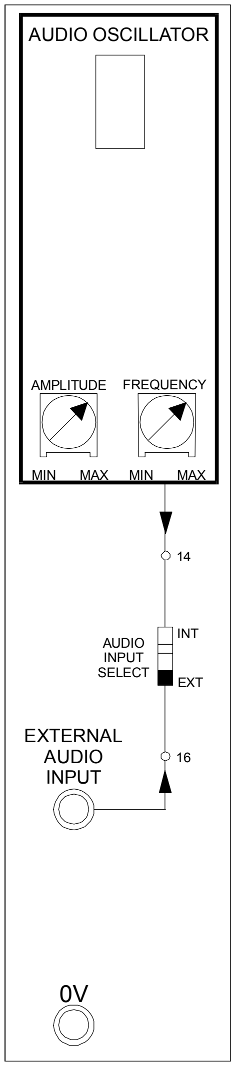


Figure 4

1.5 The Modulator

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

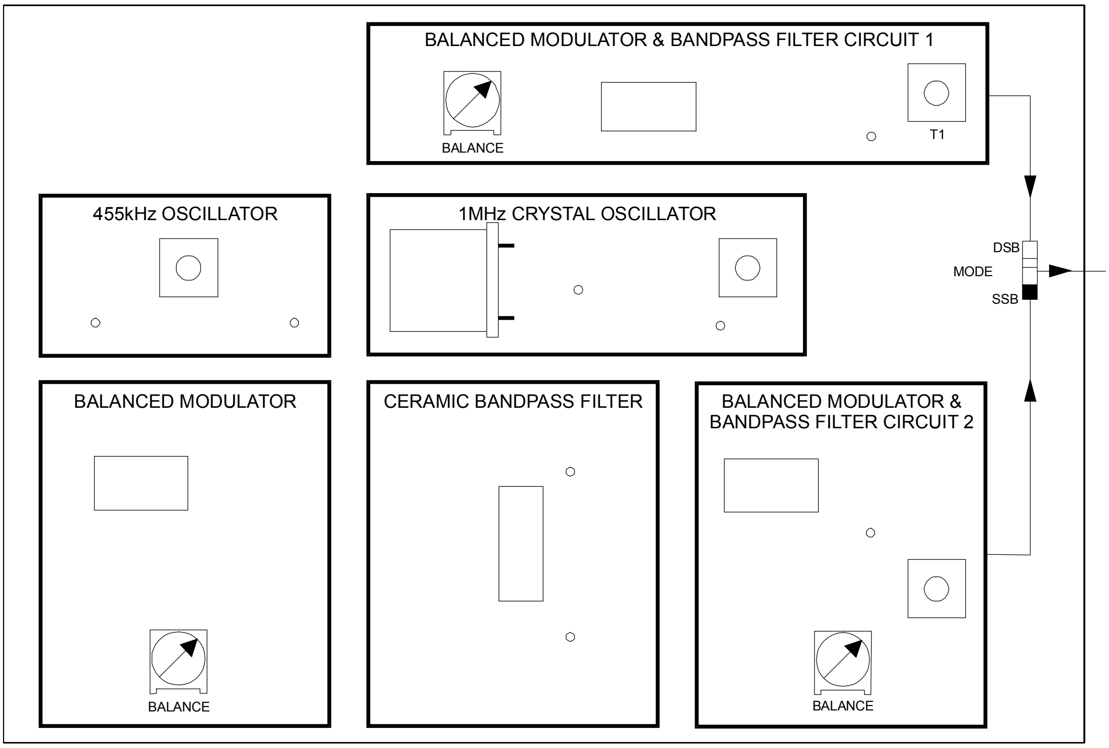


Figure 5

1.6 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.

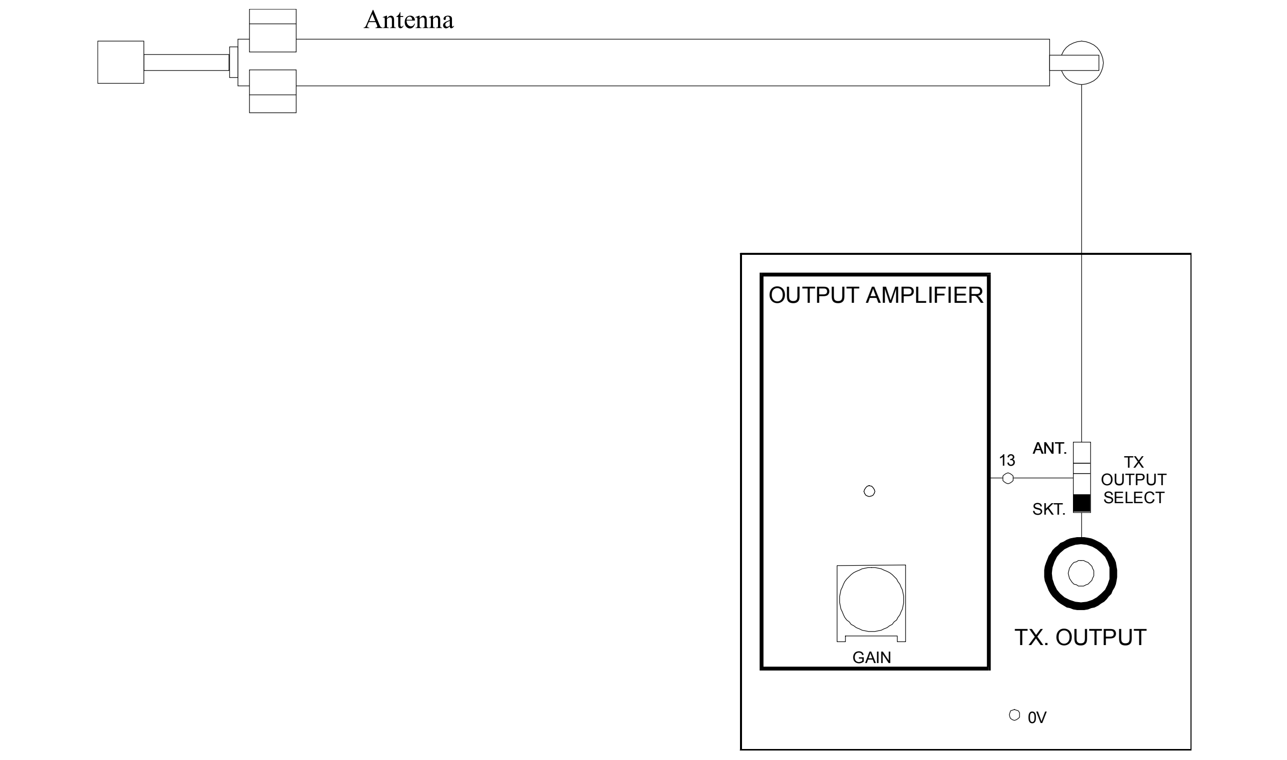


Figure 6

1.7 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.

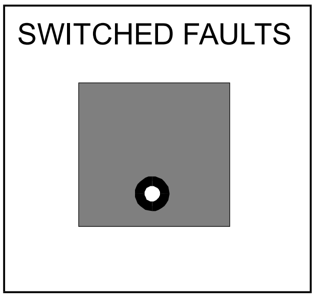


Figure 7

Notes:

....

....

....

....

....

....

....

....

....

....

1.8 Layout Diagram of the ANACOM 1/2 Board

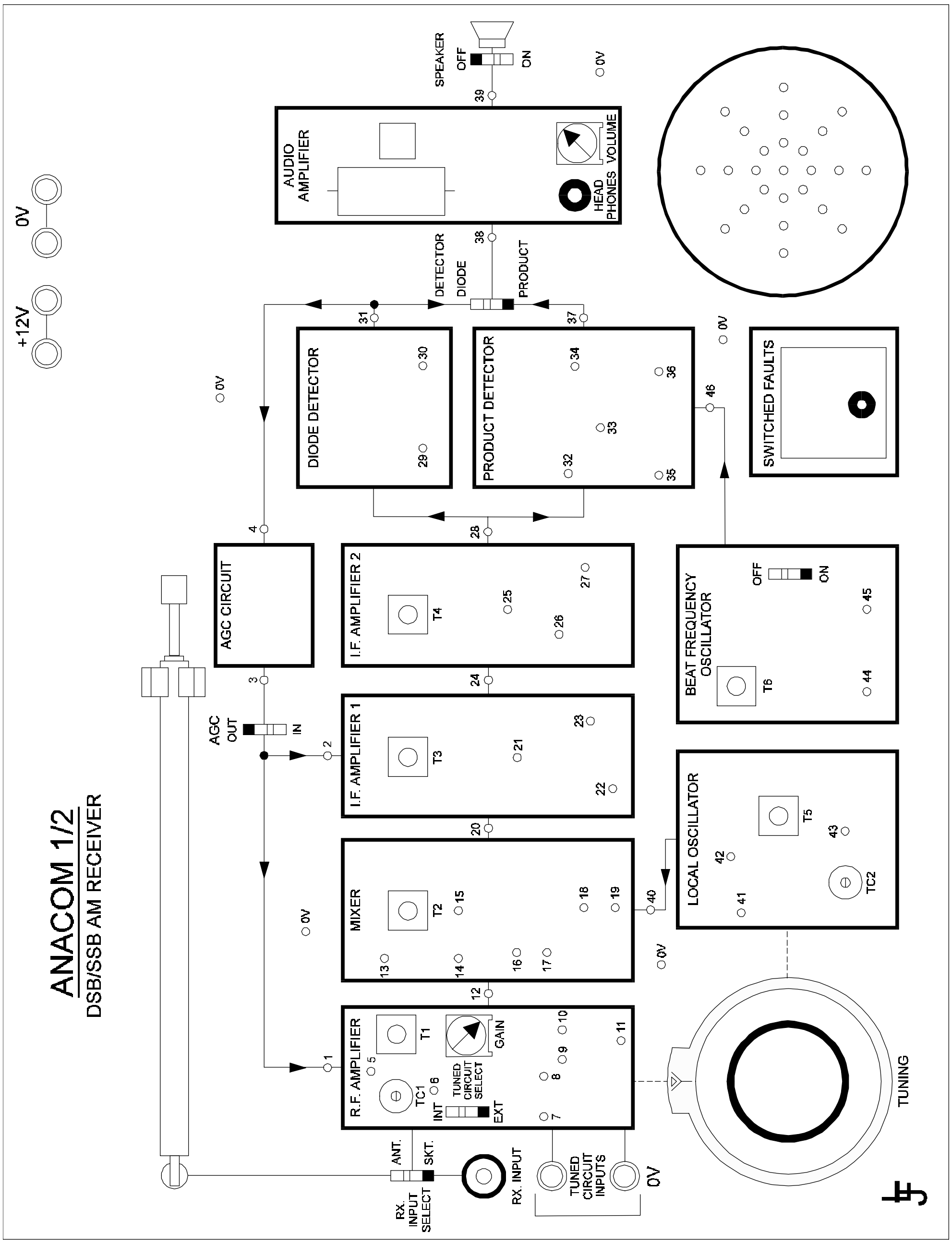


Figure 8

1.9 The ANACOM 1/2 Board Blocks

The receiver board can be considered as five separate blocks:

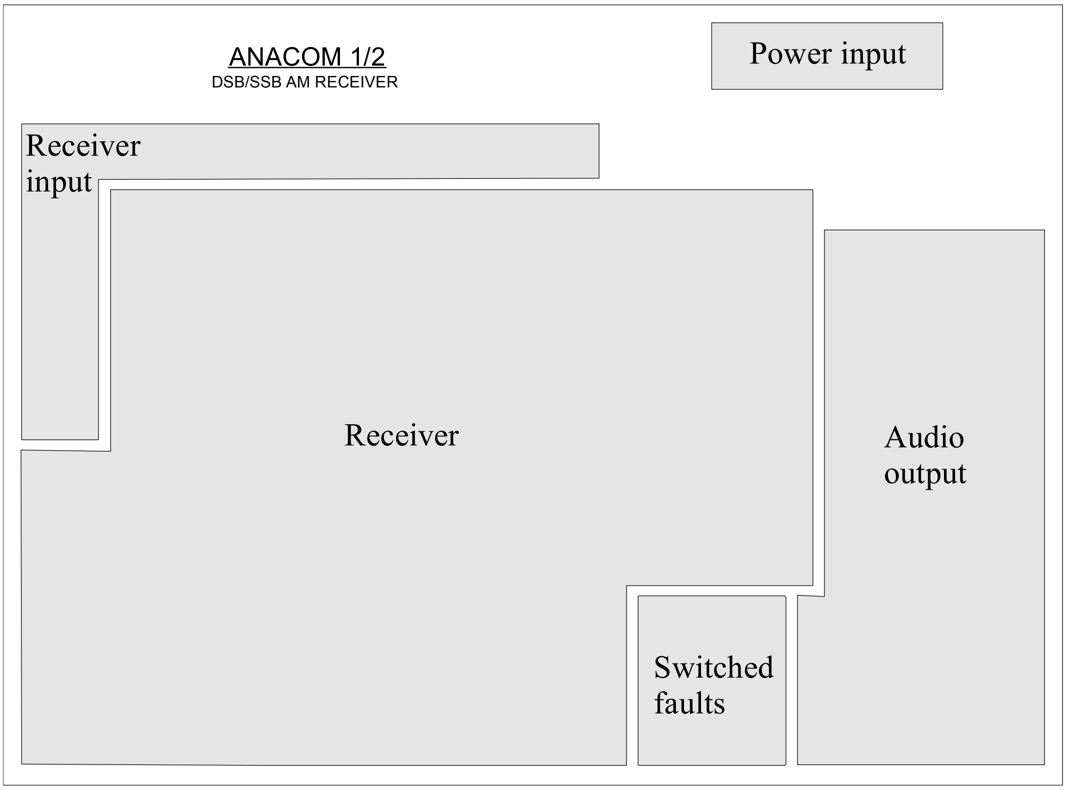


Figure 9

1.10 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.

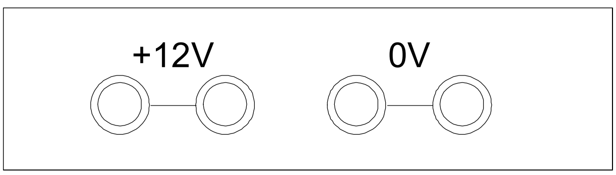


Figure 10

1.11 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.

Figure 11

Notes:

....

....

....

....

....

....

....

....

1.12 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.

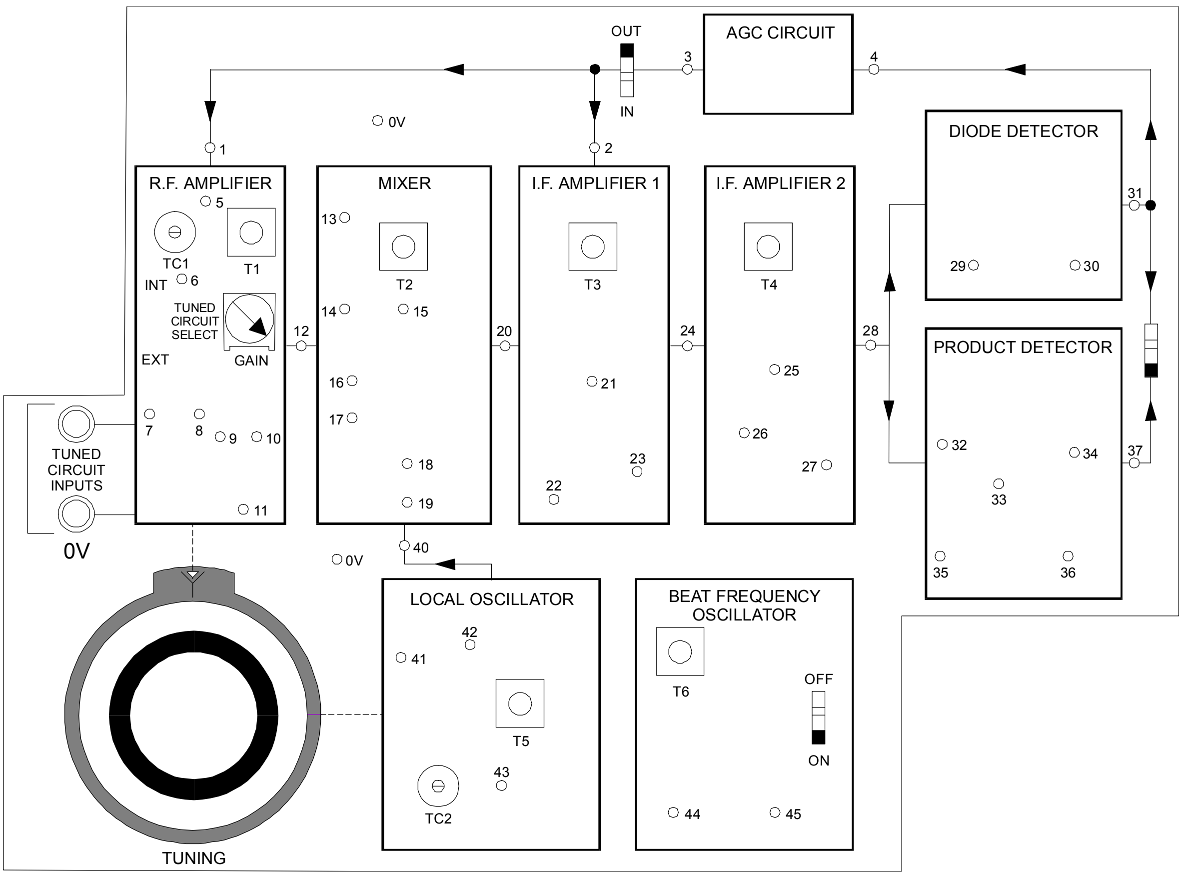


Figure 12

1.13 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.

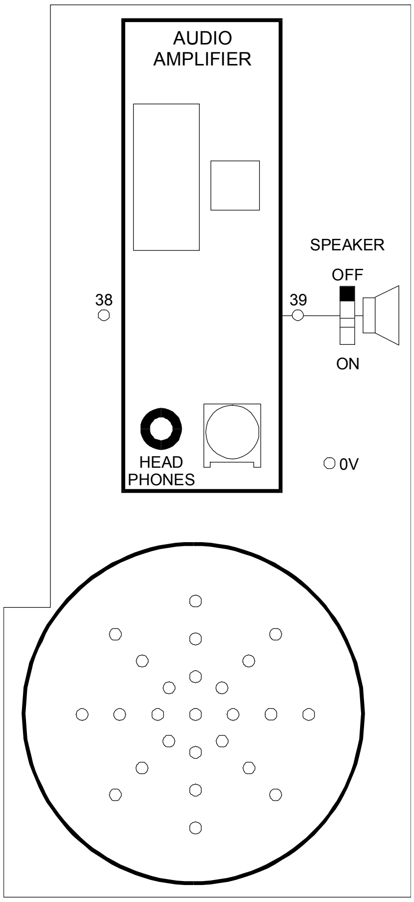


Figure 13

1.14 The Switched Faults

Under the 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.

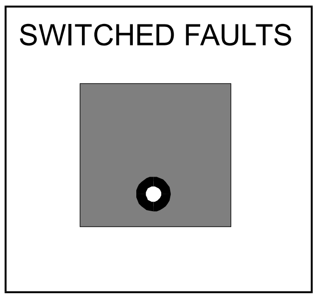


Figure 14

Notes:

....

....

....

....

....

....

....

....

....

....

Chapter 2
An Introduction to Amplitude Modulation

2.1 The Frequency Components of the Human Voice

When we speak, we generate a sound that is very complex and changes continuously so at a particular instant in time the waveform may appear as shown in Figure 15 below.

However complicated the waveform looks, we can show that it is made of many different sinusoidal signals added together.

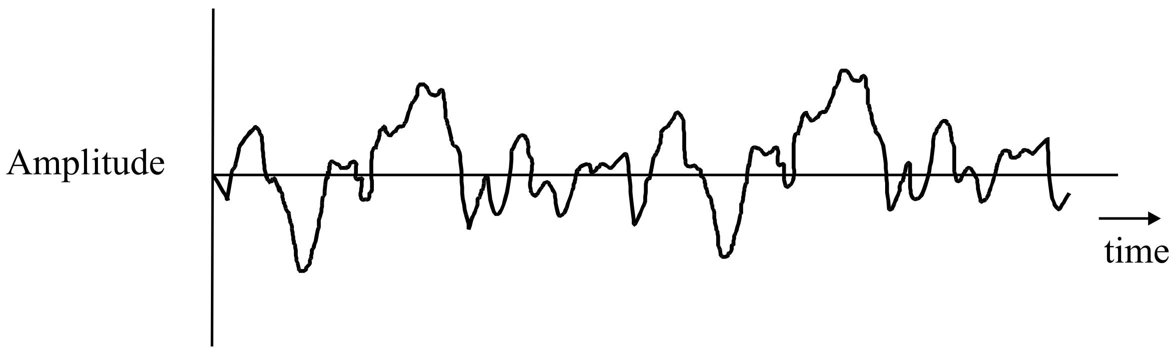
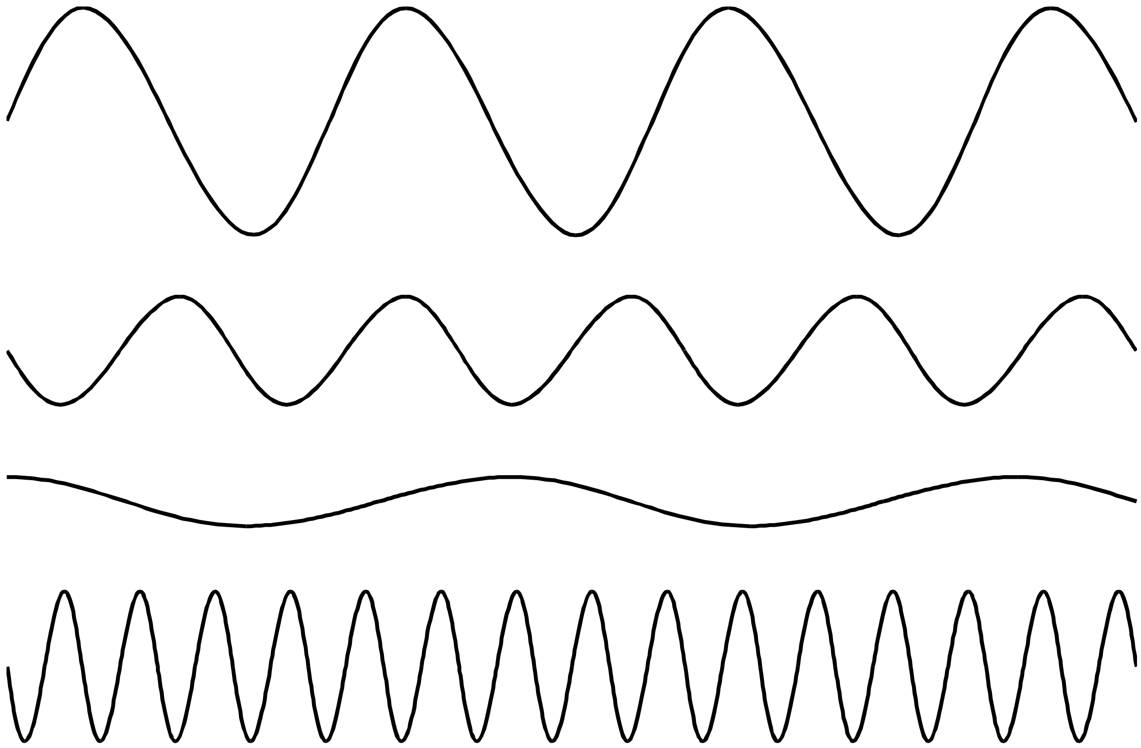


Figure 15

To record this information we have a choice of three methods. The first is to show the original waveform as we did in Figure 15.

The second method is to make a list of all the separate sinusoidal waveforms that were contained within the complex waveform (these are called 'components', or 'frequency components'). This can be seen in Figure 16 overleaf.



Only four of the components of the audio signal in Figure 15 are shown above. The actual number of components depends on the shape of the signal being considered and could be a hundred or more if the waveform was very complex.

Figure 16

The third way is to display all the information on a diagram. Such a diagram shows the frequency spectrum. It is a graph with amplitude plotted against frequency. Each separate frequency is represented by a single vertical line, the length of which represents the amplitude of the sinewave. Such a diagram is shown in Figure 17 opposite. Note that nearly all speech information is contained within the frequency range of 300Hz to 3.4kHz.

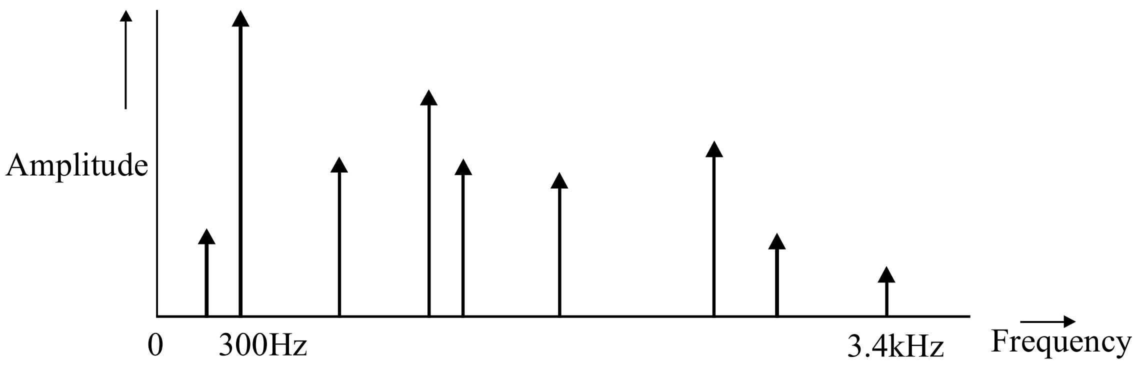


Figure 17 A Typical Voice-Frequency Spectrum

Although an oscilloscope will only show the original complex waveform, it is important for us to remember that we are really dealing with a group of sinewaves of differing frequencies, amplitudes and phases.

2.2 A Simple Communication System

Once we are out of shouting range of another person, we must rely on some communication system to enable us to pass information.

The only essential parts of any communication system are a transmitter, a communication link and a receiver, and in the case of speech, this can be achieved by a length of cable with a microphone and an amplifier at one end and a loudspeaker and an amplifier at the other.

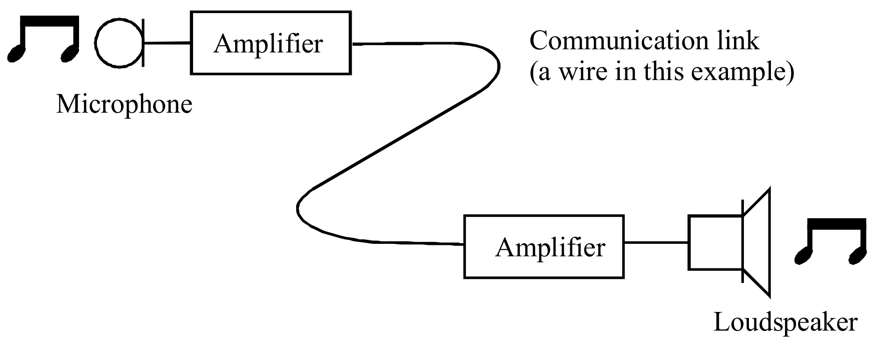


Figure 18 A Simple Communication System

For long distances, or for when it is required to send signals to many destinations at the same time, it is convenient to use a radio communication system.

2.3 The Frequency Problem

To communicate by radio over long distances we have to send a signal between two antennas, one at the sending or transmitting end and the other at the receiver.

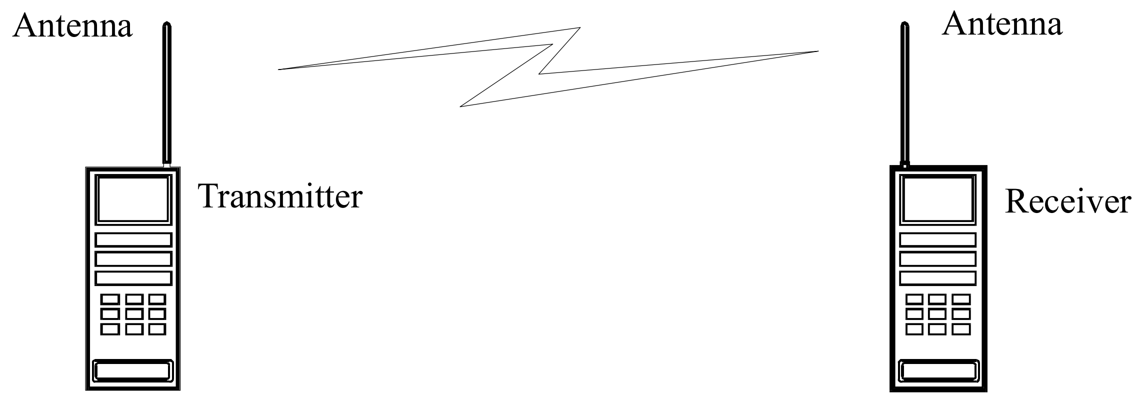


Figure 19

The frequencies used by radio systems for AM transmissions are between 200kHz and 25MHz.

A typical radio frequency of, say, 1MHz is much higher than the frequencies present in the human voice.

We appear to have two incompatible requirements. The radio system uses frequencies like 1MHz to transmit over long distances, but we wish to send voice frequencies of between 300Hz and 3.4kHz that are quite impossible to transmit by radio signals.

2.4 Modulation

This problem can be overcome by using a process called 'modulation'.

The radio system can easily send high frequency signals between a transmitter and a receiver but this, on its own, conveys no information.

Now, if we were to switch it on and off for certain intervals, we could use it to send information. For example, we could switch it on briefly at exactly one second intervals and provide a time signal (see Figure 20 below). Messages could be passed by switching it on and off in a sequence of long and short bursts and hence send a message by Morse Code. Figure 20 below shows the sequence that would send the distress signal SOS.

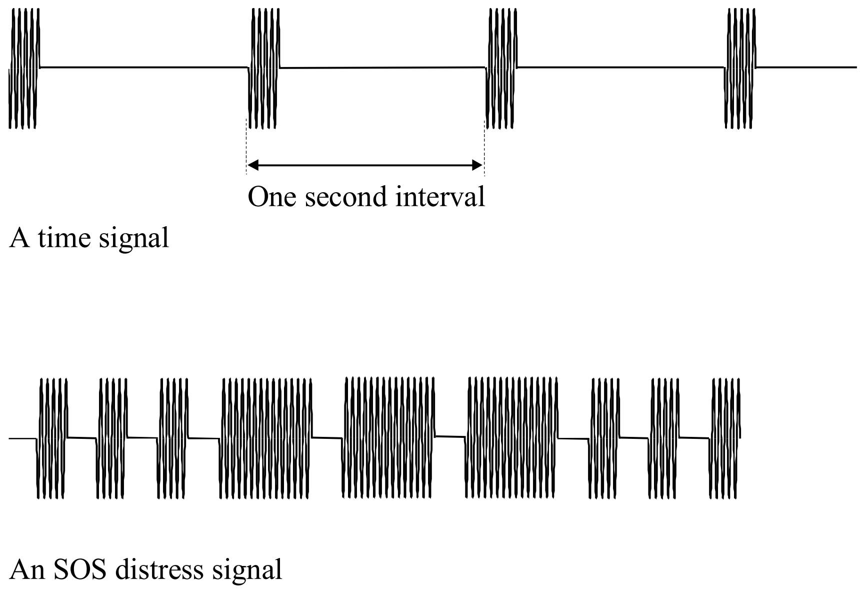


Figure 20

The high frequency signal that has been used to send or 'carry' the information from one place to another is called a 'carrier wave'.

The carrier wave must be persuaded in some way to convey the speech to the receiver. The speech signal represents the 'information' that we wish to send and therefore this signal is called the 'information signal'.

The method employed is to change some characteristic of the carrier wave in sympathy with the information signal and then, by detecting this change, be able to recover the information signal at the receiver.

2.5 Amplitude Modulation (AM)

The method that we are going to use is called Amplitude Modulation. As the name would suggest, we are going to use the information signal to control the amplitude of the carrier wave.

As the information signal increases in amplitude, the carrier wave is also made to increase in amplitude. Likewise, as the information signal decreases, then the carrier amplitude decreases.

By looking at Figure 21 below, we can see that the modulated carrier wave does appear to 'contain' in some way the information as well as the carrier. We will see later how the receiver is able to extract the information from the amplitude modulated carrier wave.

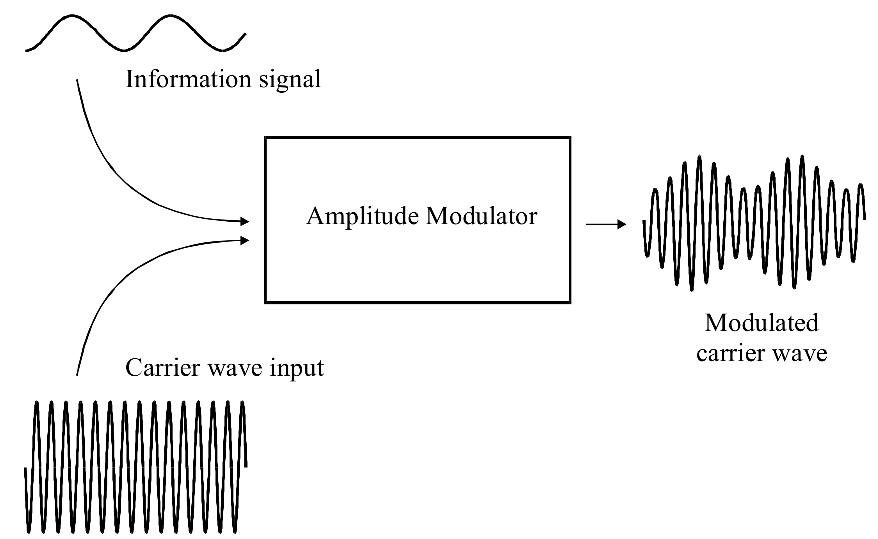


Figure 21

2.6 Depth of Modulation

The amount by which the amplitude of the carrier wave increases and decreases depends on the amplitude of the information signal and is called the 'depth of modulation'.

The depth of modulation can be quoted as a fraction or as a percentage.

$$\text{Percentage modulation} = \frac{V\,\text{max} - V\,\text{min}}{V\,\text{max} + V\,\text{min}} \times 100\%$$

Here is an example:

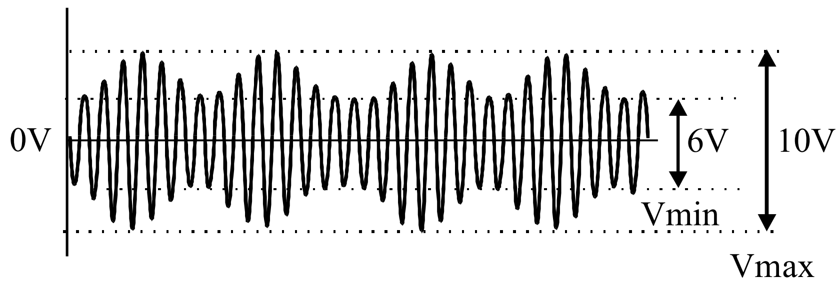


Figure 22 Depth of Modulation

In Figure 22 we can see that the modulated carrier wave varies from a maximum peak-to-peak value of 10 volts, down to a minimum value of 6 volts.

Inserting these figures in the above formula, we get:

$$\text{Percentage modulation} = \frac{10-6}{10+6} \times 100\%$$
$$= \frac{4}{16} \times 100\%$$
$$= 25\% \text{ or } 0.25$$

2.7 The Frequency Spectrum

Assume a carrier frequency (f_c) of 1MHz and an amplitude of, say, 5 volts peak-to-peak.

The carrier could be shown as:

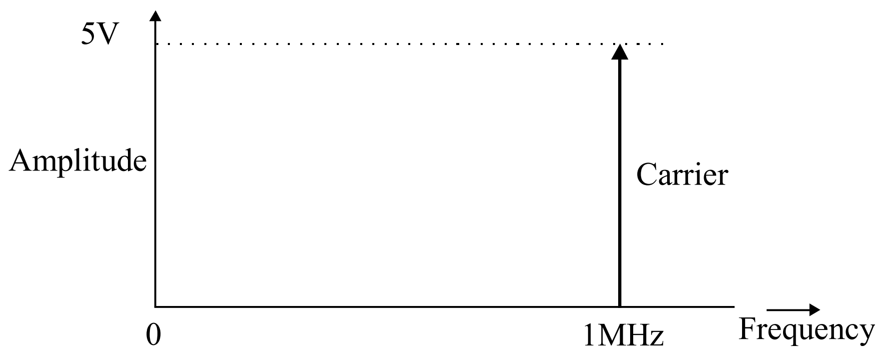


Figure 23 The Frequency Spectrum of a Carrier Wave

If we also have a 1kHz information signal, or modulating frequency (fm), with an amplitude of 2V peak-to-peak it would look like this:

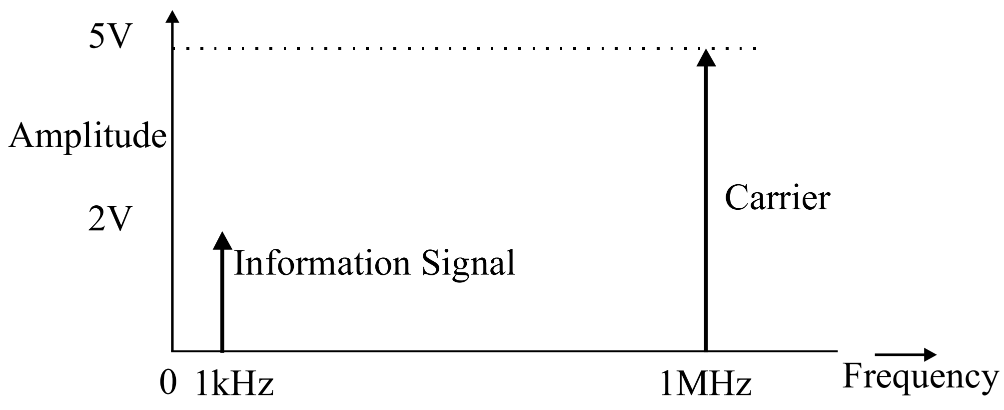


Figure 24 The Frequency Spectrum of a Carrier Wave and an Information Signal

When both signals have passed through the amplitude modulator they are combined to produce an amplitude modulated wave.

The resultant AM signal has a new frequency spectrum as shown in Figure 25 below:

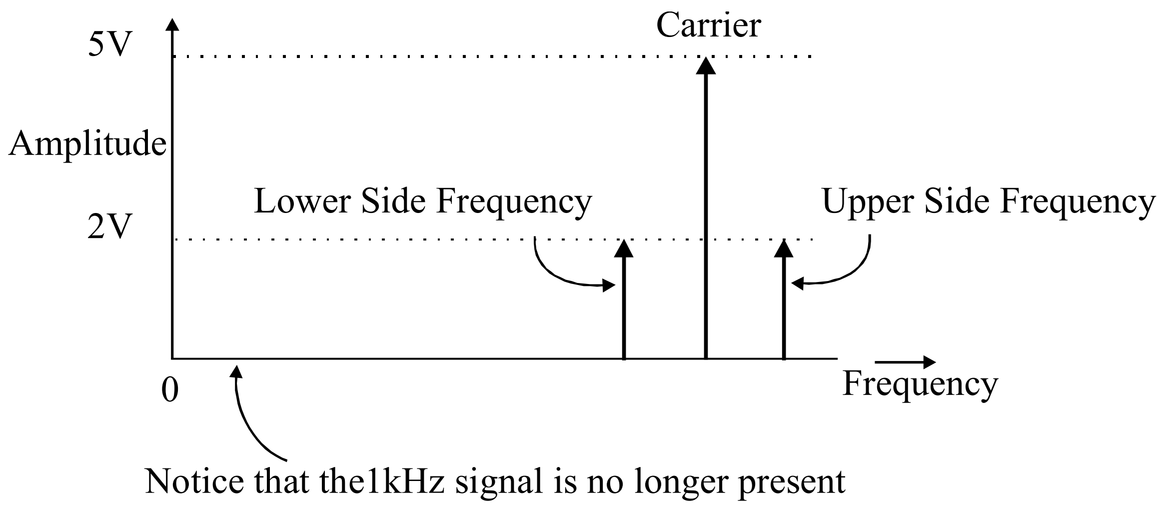


Figure 25 Frequency Spectrum of Resultant AM Signal

Some interesting changes have occurred as a result of the modulation process.

(i) The original 1kHz information frequency has disappeared.

(ii) The 1MHz carrier is still present and is unaltered.

(iii) There are two new components:

Carrier frequency (f_c) **plus** the information frequency, called the upper side frequency ($f_c + f_m$)
and
Carrier frequency (f_c) **minus** the information frequency, called the lower side frequency ($f_c - f_m$)

The resulting signal in this example has a maximum frequency of 1001kHz and a minimum frequency of 999kHz and so it occupies a range of 2kHz. This is called the bandwidth of the signal. Notice how the bandwidth is twice the highest frequency contained in the information signal.

Notes:

....

....

....

....

....

....

....

....

....

....

....

....

....

2.8 Constructing the Amplitude Modulated Waveform

It is often difficult to see how the AM carrier wave can actually consist of the carrier and the two side frequencies, all of which are radio frequency signals - there is no audio signal present at all. In appearance, the AM carrier wave looks more likely to consist of the carrier frequency and the incoming information signal.

Figure 26 shows this situation:

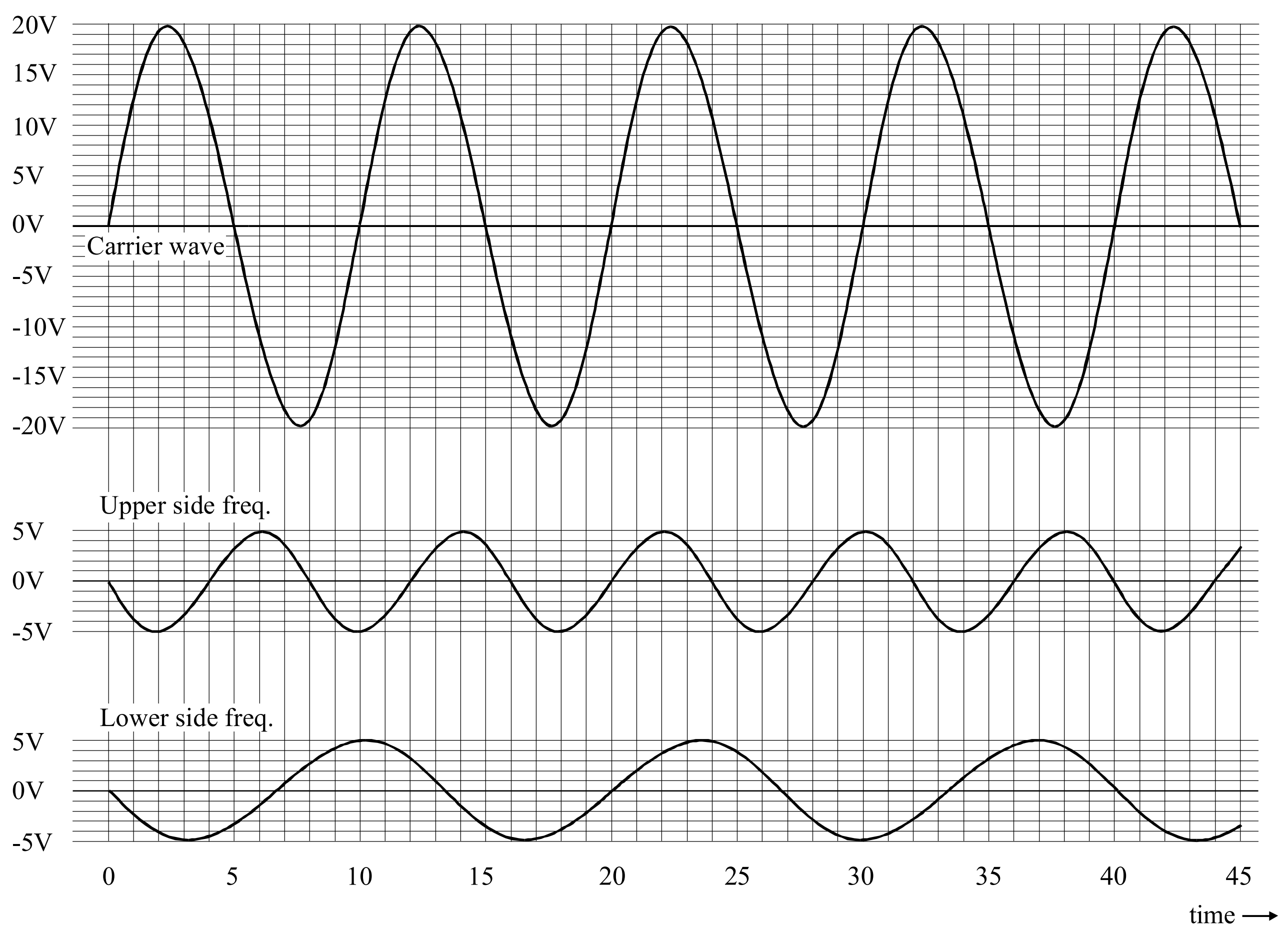


Figure 26

Here are the three radio frequency signals that form the modulated carrier wave. We are going to add the three components and (hopefully) reconstruct the modulated waveform.

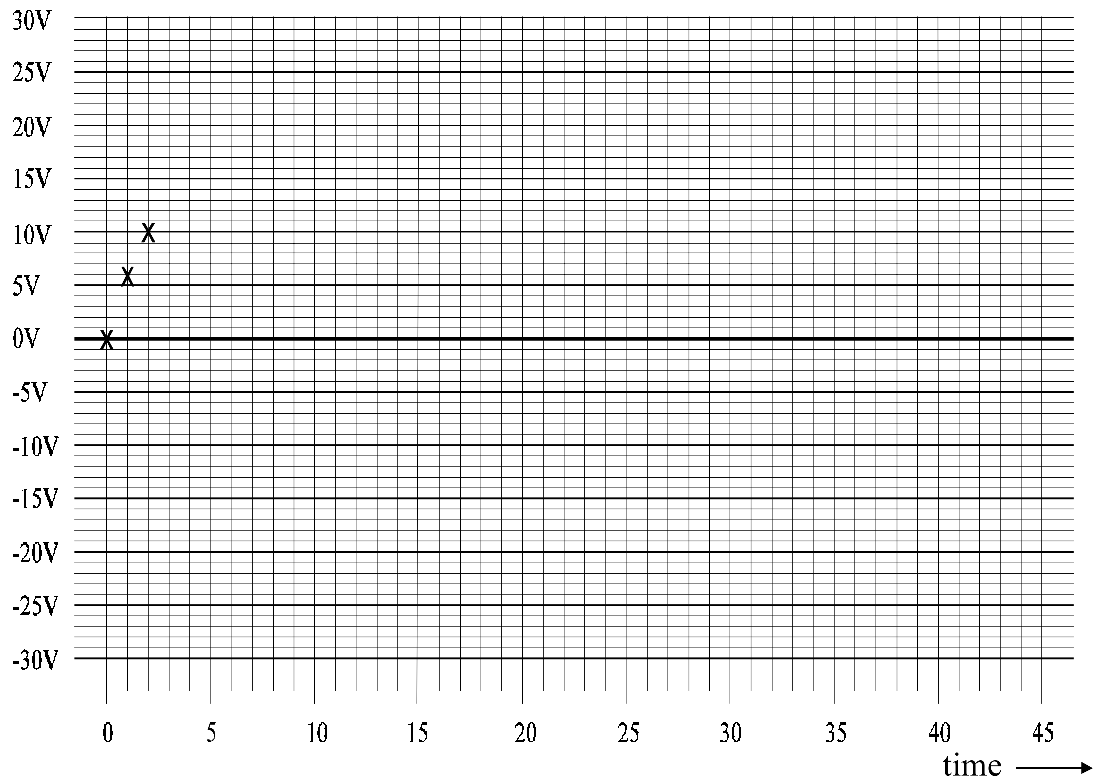


Figure 27 An Amplitude Modulated Wave

2.9 Sidebands

If the information signal consisted of a range of frequencies, each separate frequency will create its own upper side frequency and lower side frequency.

As an example, let us imagine that a carrier frequency of 1MHz is amplitude modulated by an information signal consisting of frequencies 500Hz, 1.5kHz and 3kHz.

As each modulating frequency produces its own upper and lower side frequency there is a range of frequencies present above and below the carrier frequency. All the upper side frequencies are grouped together and referred to as the upper sideband (USB) and all the lower side frequencies form the lower sideband (LSB).

This amplitude modulated wave would have a frequency spectrum as shown in Figure 28 below:

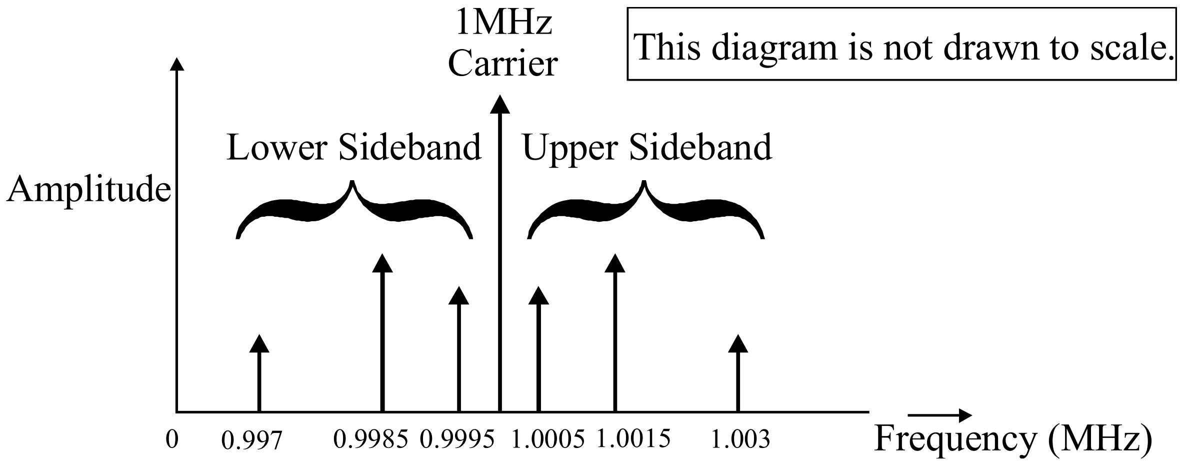


Figure 28 Frequency Spectrum Showing Upper and Lower Sidebands

Because the frequency spectrum of the AM waveform contains two sidebands, this type of amplitude modulation is often called a double-sideband transmission, or DSB.

2.10 Power in the Sidebands

The modulated carrier wave that is finally transmitted contains the original carrier and the sidebands. The carrier wave is unaltered by the modulation process and contains at least two-thirds of the total transmitted power. The remaining power is shared between the two sidebands.

The power distribution depends on the depth of modulation used and is given by:

$$\text{Total power} = (\text{carrier power})\left(1 + \frac{N^2}{2}\right) \quad \text{where N is the depth of modulation.}$$

Example:

A DSB AM signal with a 1kW carrier was modulated to a depth of 60%. How much power is contained in the upper sideband?

(i) Start with the formula:

$$\text{Total power} = (\text{carrier power})\left(1 + \frac{N^2}{2}\right) \quad \text{where N is the depth of modulation.}$$

(ii) Insert all the figures that we know. This is the 1000 for the carrier power and 0.6 for the modulation depth. We could have used the figure 60% instead of 0.6 but this way makes the math slightly easier.

$$\text{Total power} = (1000)\left(1 + \frac{0.6^2}{2}\right)$$

(iii) Remove the brackets.

$$\text{Total power} = (1000)\left(1 + \frac{0.36}{2}\right)$$

$$= 1000 \times (1 + 0.18)$$
$$= 1000 \times 1.18$$
$$= 1180\text{W}$$

(iv) The carrier power was 1000W and the total power of the modulated wave is 1180W so the two sidebands must, between them, contain the other 180W. The power contained in the upper and lower sidebands is always equal and so each must contain $\frac{180}{2} = 90\text{W}$.

The greater the depth of modulation, the greater is the power contained within the sidebands. The highest usable depth of modulation is 100% (above this the distortion becomes excessive).

Since at least twice as much power is wasted as is used, this form of modulation is not very efficient when considered on a power basis. The good news is that the necessary circuits at the transmitter and at the receiver are simple and inexpensive to design and construct.

Notes:

....

....

....

....

....

....

2.11 Practical Exercise: The Double Sideband AM Waveform

The frequency and peak-to-peak voltage of the carrier are: ....

....

The frequency and peak-to-peak voltage of the information signal are: ....

....

Record the AM waveform at tp3 in Figure 30 below.

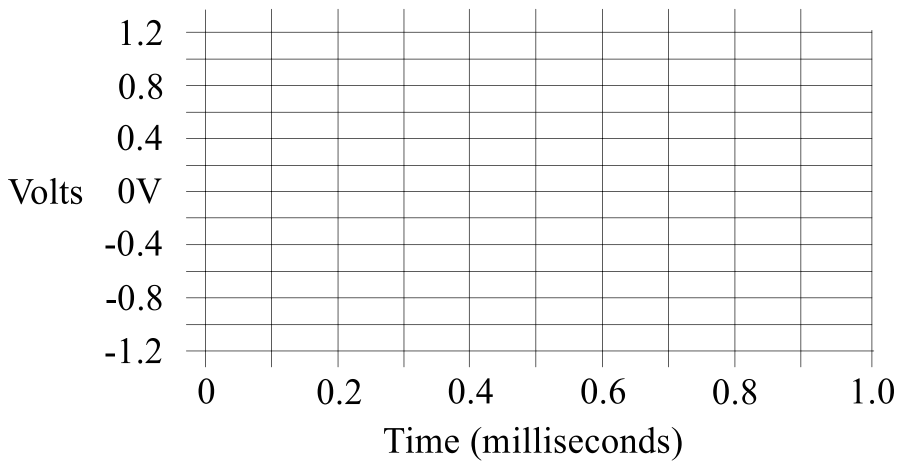


Figure 30 The AM Waveform at tp3 on ANACOM 1/1

The effects of adjusting the AMPLITUDE PRESET and the FREQUENCY PRESET in the AUDIO OSCILLATOR are: ....

....

....

....

Chapter 3
DSB Transmitter and Receiver

3.1 The Double Sideband Transmitter

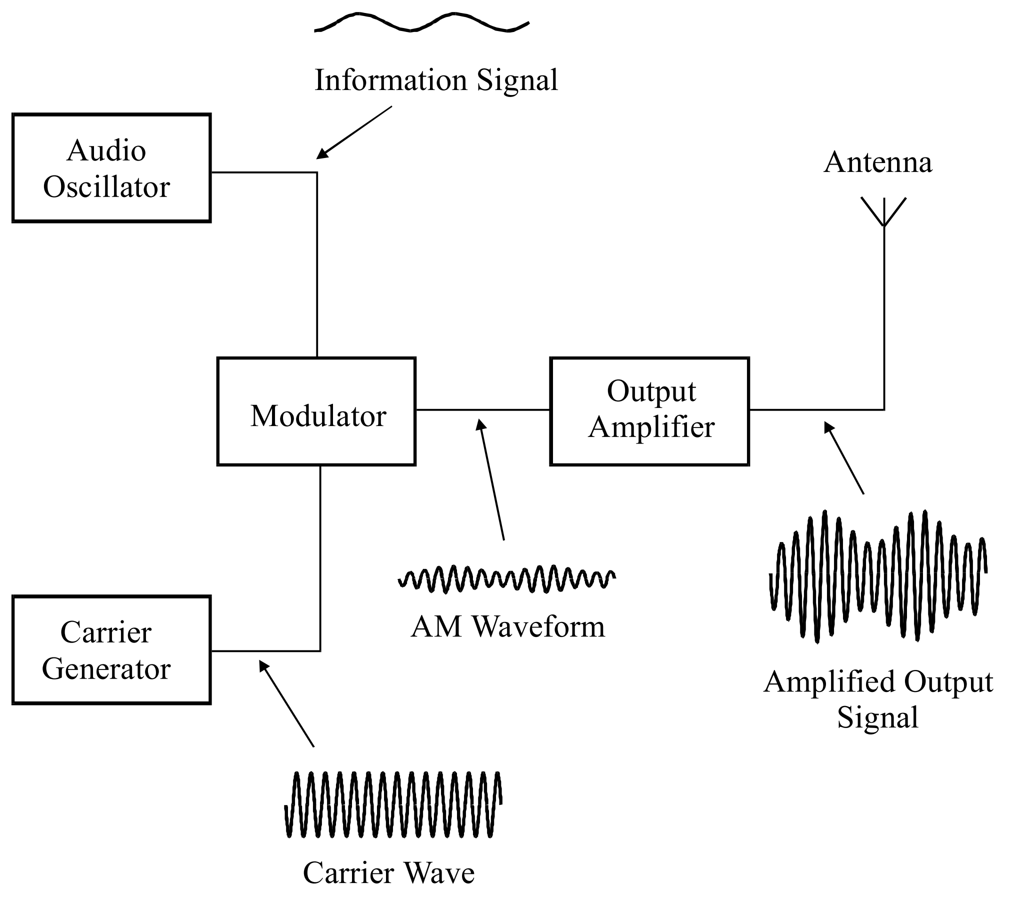


Figure 31 An Amplitude Modulated Transmitter

The transmitter circuits produce the amplitude modulated signals that are used to carry information over the transmission path to the receiver. The main parts of the transmitter are shown in Figure 31.

In Figures 31 and 32, we can see that the peak-to-peak voltages in the AM waveform increase and decrease in sympathy with the audio signal.

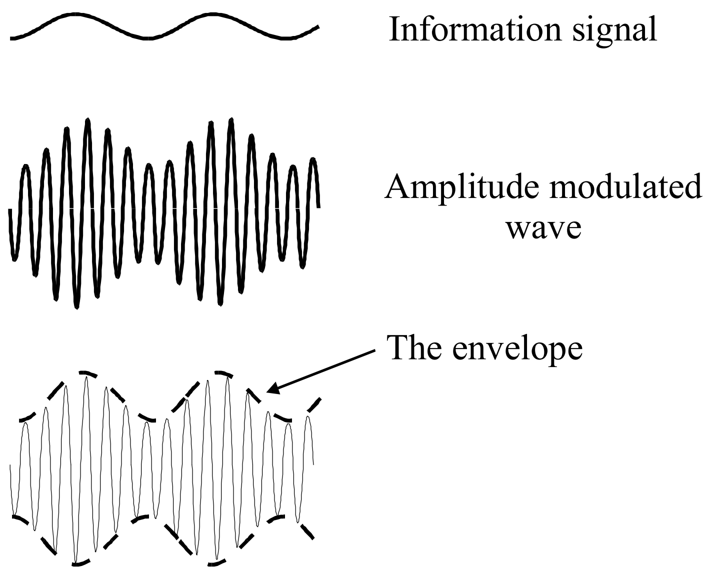


Figure 32 The Modulation Envelope

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 32. This shape, enclosed by a dashed line in our 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.

We will now consider the action of each circuit as we follow the route taken by the information that we have chosen to transmit.

The first task is to get hold of the information to be transmitted.

3.2 The Information Signal

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.

3.3 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 15kHz 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 15kHz 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 sinewave 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. In Figure 33 overleaf, a simple case is illustrated in which the carrier only contains three frequency components modulated by a single frequency component. Even so we can see that the overall bandwidth has been considerably increased.

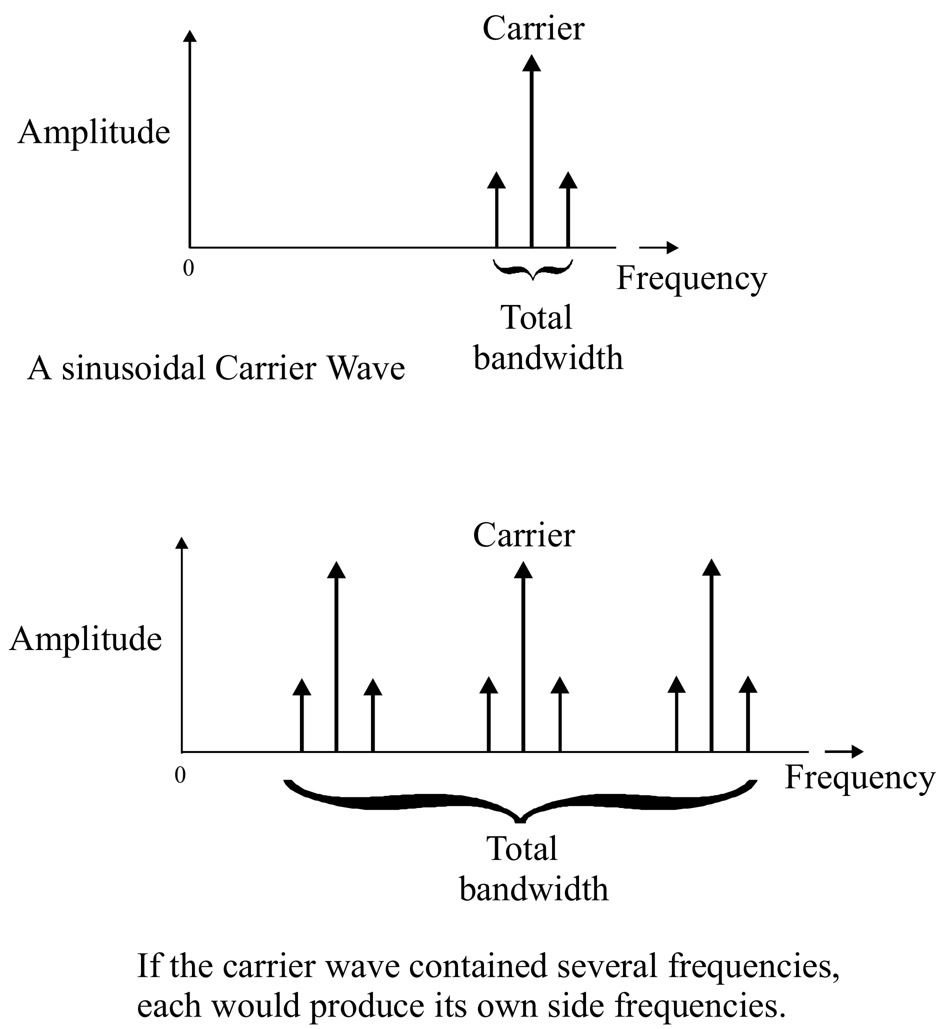


Figure 33

On ANACOM 1/1, the carrier wave generated is a sinewave of 1MHz.

Now we have the task of combining the information signal and the carrier wave to produce amplitude modulation.

3.4 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 as we saw in Chapter 2.

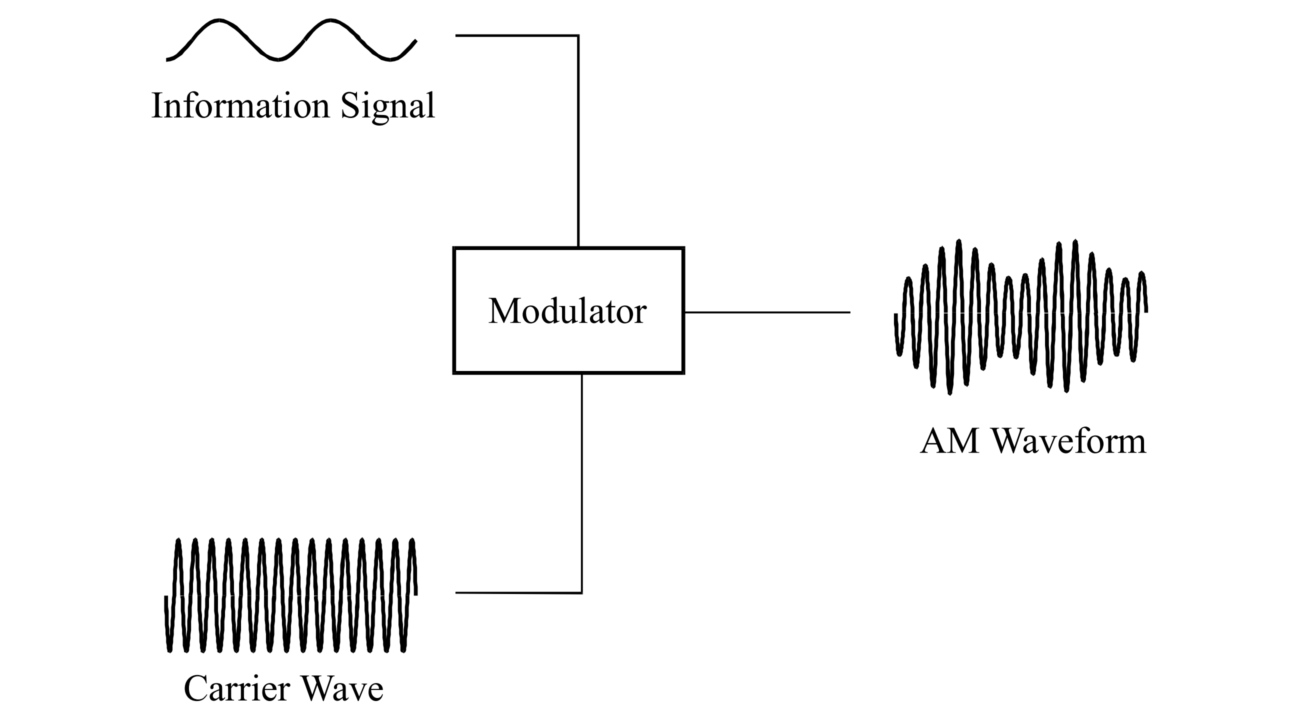


Figure 34 Modulation of Information Signal and Carrier Wave

The signal is now nearly ready for transmission.

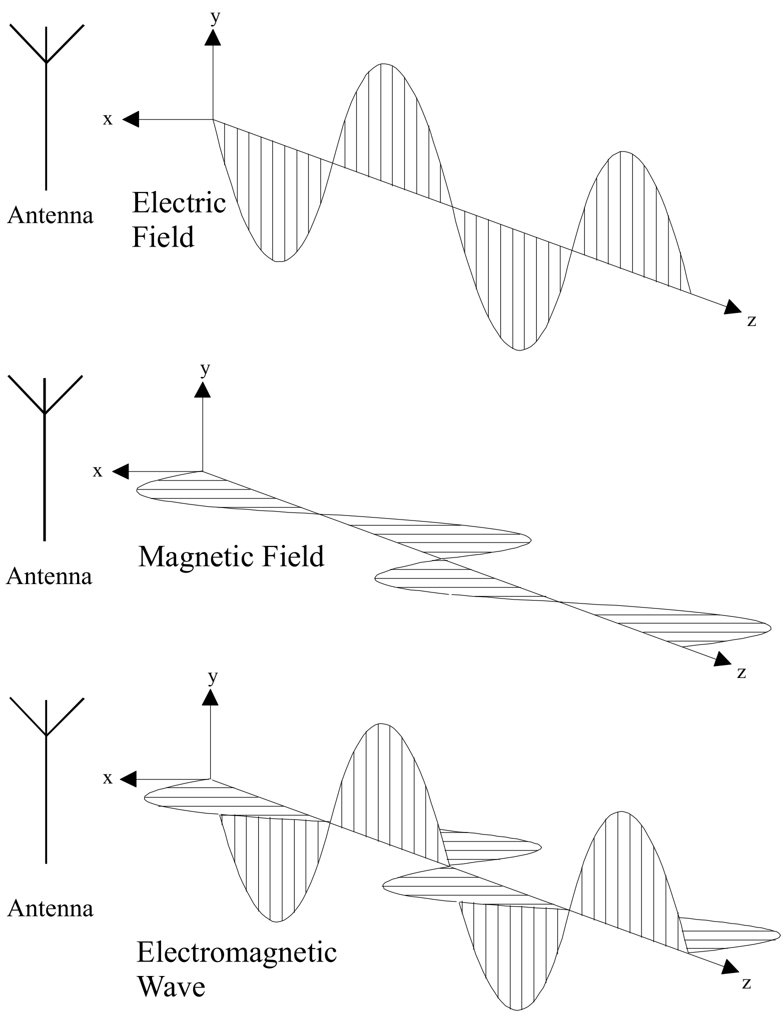
If the modulation process has given rise to any unwanted frequency components then a bandpass filter can be employed to remove them.

3.5 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.

3.6 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 that is at right angles to both the magnetic and the electric fields, this is shown in Figure 35.



This shows the electric field moving out from the antenna. In this example the electric field is vertical because the antenna is positioned vertically (in the direction shown by y).

The magnetic field is always at right angles to the electric field so in this case, it is positioned horizontally (in the direction shown by x).

In an electromagnetic wave both fields exist together and they move at the speed of light in a direction that is at right angles to both fields (shown by the arrow labeled z).

Figure 35 An Electromagnetic Wave

The antenna converts the power output of the Output Amplifier into an electromagnetic wave.

How does it do this?

The output amplifier causes a voltage to be generated along the antenna thus generating a voltage difference and the resultant electric field between the top and bottom. This causes an alternating movement of electrons on the transmitting antenna that is really an AC current. Since an electric current always has a magnetic field associated with it, an alternating magnetic field is produced.

The overall effect is that the output amplifier has produced alternating electric and magnetic fields around the antenna. The electric and magnetic fields spread out as an electromagnetic wave at the speed of light (3 x 10^8 meters per second).

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:

$$\lambda = \frac{3 \times 10^8}{1 \times 10^6}$$

$$\lambda = 300\text{m}$$

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!

3.7 Polarization

If the transmitting antenna is placed vertically, the electrical field is vertical and the magnetic field is horizontal (as seen in Figure 35). 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 em (electromagnetic) wave. A vertical transmitting antenna results in a vertically polarized wave, and a horizontal one would result in a horizontally polarized em wave.

3.8 The DSB Receiver

The em wave from the transmitting antenna will travel to the receiving antenna, carrying the information with it.

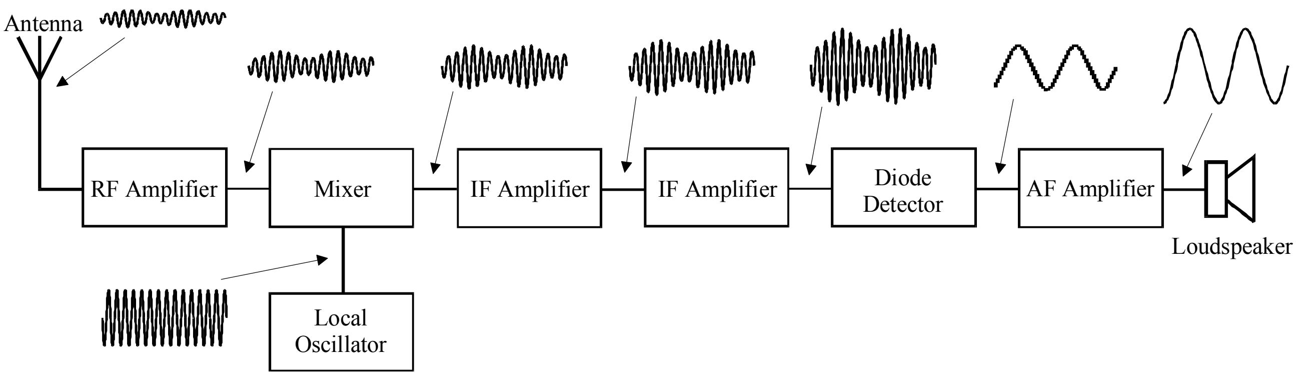


Figure 36 A Superheterodyne Receiver

We will continue to follow our information signal as it passes through the receiver.

3.9 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 millivolts and often only a few microvolts. The voltage supplied to the loudspeaker at the output of the receiver is up to ten volts.

We clearly need a lot of amplification.

3.10 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 any one transmission.

The receiver circuits generate noise signals that 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.

Notes:

....

....

....

....

....

....

....

....

....

....

....

....

....

....

3.11 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 that offers more gain at the frequency of resonance and reduced amplification above and below this frequency. This is called selectivity.

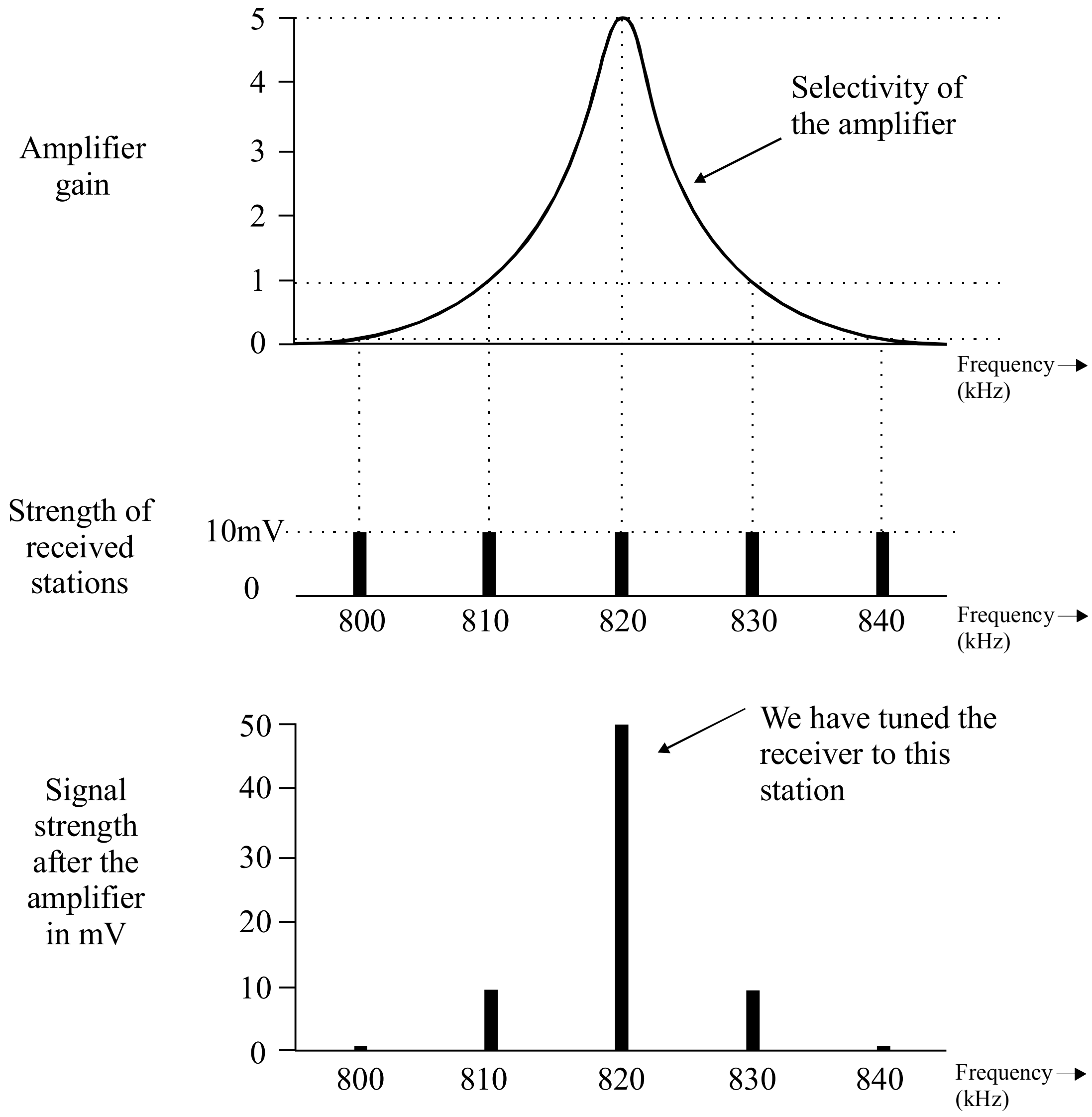


Figure 37

In Figure 37 we can see the effects of using an amplifier with selectivity.

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

3.12 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 that 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.

3.13 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:

(i) A 'difference' frequency of local oscillator frequency - RF signal frequency.

(ii) A 'sum' frequency equal to local oscillator frequency + RF signal frequency.

(iii) 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 above the audio spectrum, called 'supersonic' frequencies, the type of receiver is called a 'superheterodyne' receiver. This is normally abbreviated to 'superhet'. It is not a modern idea having been invented in the year 1917.

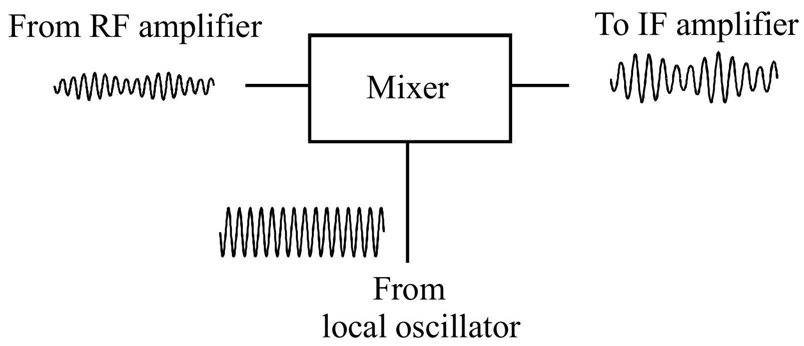


Figure 38 The Mixer

In Section 3.12, we saw 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 455kHz.

As an example, if the radio is tuned to receive a broadcast station transmitting at 800kHz, the local oscillator will be running at 1.255MHz. The difference frequency is 1.255MHz - 800kHz = 455kHz.

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

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

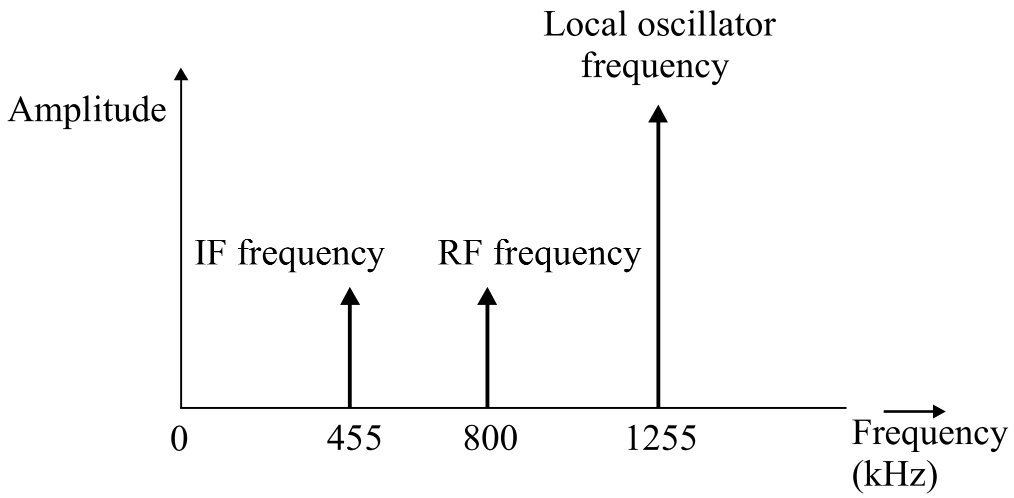


Figure 39 A Superhet Receiver Tuned to 800kHz

Note: In Figure 39, the local oscillator output is shown larger than the IF and RF frequency components, this is usually the case. However, there is no fixed relationship between the actual amplitudes. Similarly, the IF and RF amplitudes are shown as being equal in amplitude but again there is no significance in this.

3.14 Image Frequencies

In the last section, we saw we could receive a station being broadcast on 700kHz by tuning the local oscillator to a frequency of 1.155MHz thus giving the difference (IF) frequency of the required 455kHz.

What would happen if we were to receive another station broadcasting on a frequency of 1.61MHz?

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

Station 1:

Frequency 700 kHz, Local oscillator 1.155MHz, IF = 455kHz

Station 2:

Frequency 1.61MHz, Local oscillator 1.155MHz, IF = 455kHz

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 910kHz 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.

3.15 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 bandpass of all the stages must be carefully controlled. Each IF stage does not necessarily have the same bandpass characteristics, it is the overall response that is important. Again, this is something that is much more easily achieved without the added complication of making them tunable.

At the final output from the IF amplifiers, we have a 455kHz 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.

3.16 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 halfwave rectifier converting an AC input to a DC output.

Figure 40 shows a simple circuit diagram of the diode detector.

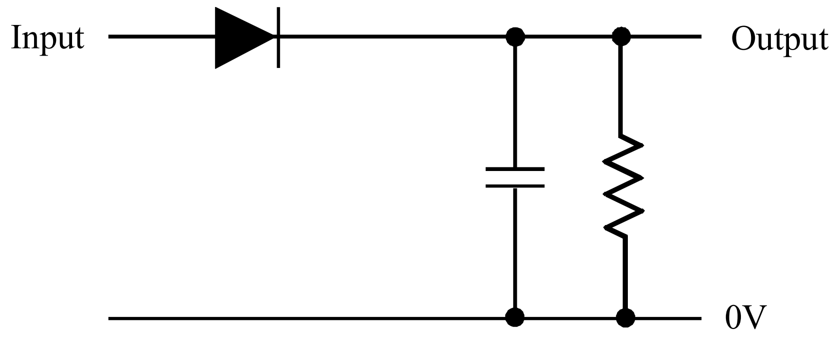


Figure 40 A Simple Diode Detector

In Figure 40, 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 41).

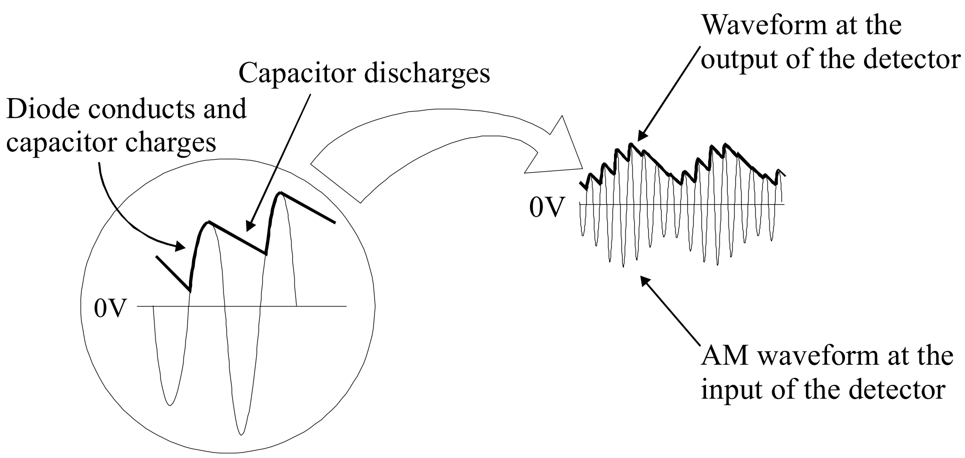


Figure 41

The result is an output that contains three components:

(i) The wanted audio information signal.

(ii) Some ripple at the IF frequency.

(iii) A positive DC voltage level.

3.17 The Audio Amplifier

At the input to the audio amplifier, a lowpass filter is used to remove the IF ripple and a capacitor blocks the DC voltage level. Figure 42 shows the result of the information signal passing through the Diode Detector and Audio Amplifier.

The input to the diode detector
from the last IF amplifier

Output of diode detector includes:
a DC level,
the audio signal,
ripple at IF frequency

Output after filtering

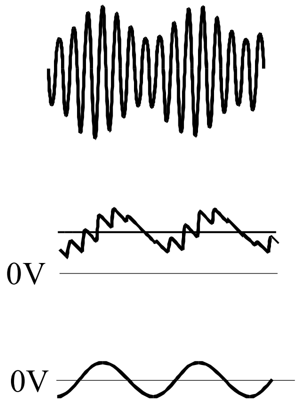


Figure 42

The remaining audio signals are then amplified to provide the final output to the loudspeaker.

3.18 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.

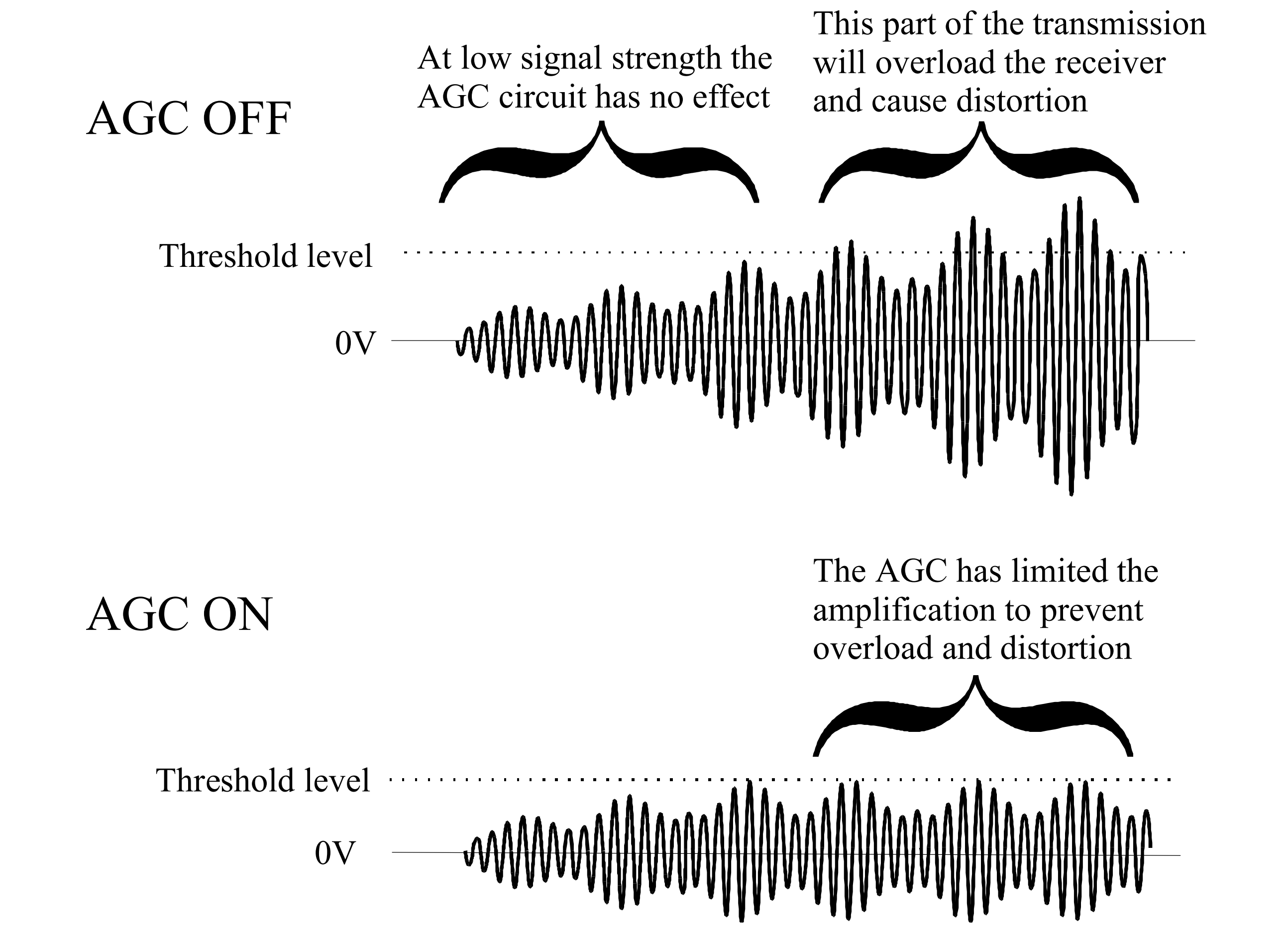


Figure 43

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'.

3.19 Practical Exercise: The DSB Transmitter and Receiver

The depth of modulation of the transmitter output at tp13 is: ....

....

Record the waveform at the output of the RF Amplifier (tp12).

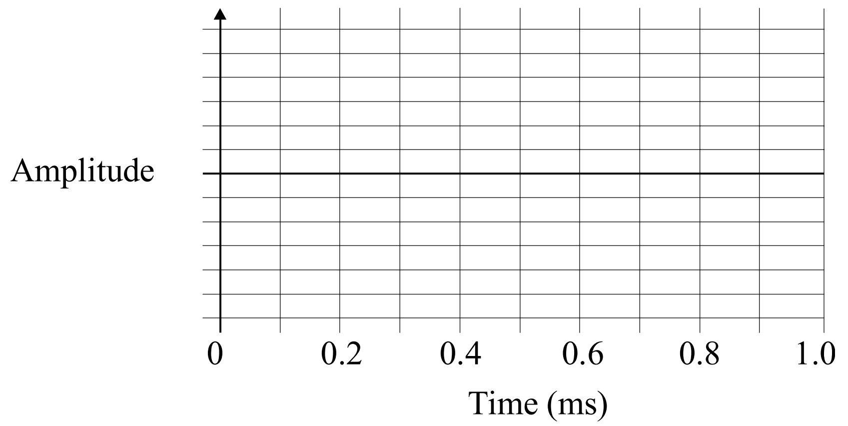


Figure 45 The Output of the RF Amplifier at tp12

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.

Record the waveform at the output of the Mixer (tp20).

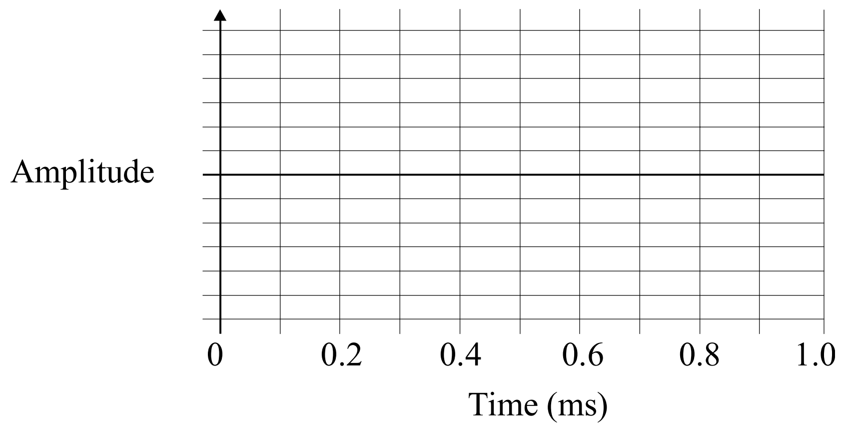


Figure 46 The Output of the Mixer Circuit at tp20

Record the waveform at the output of the first IF Amplifier (tp24).

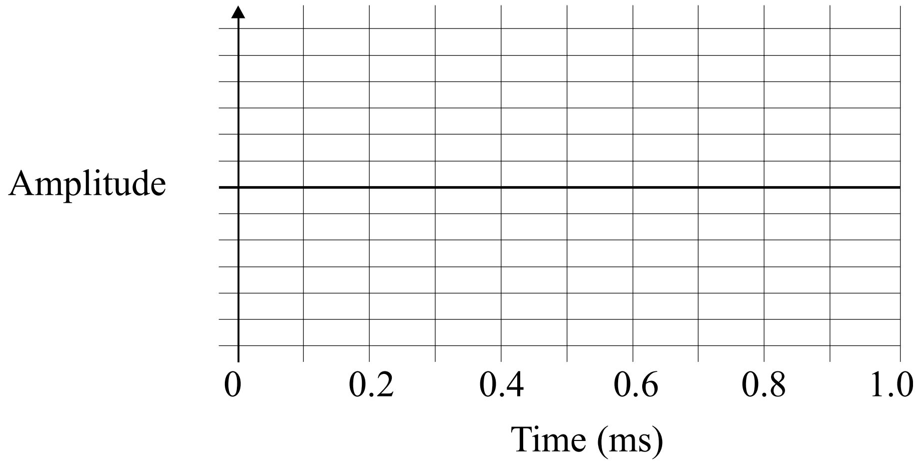


Figure 47 The Output of the First IF Amplifier at tp24

Record the waveform at the output of the final IF Amplifier (tp28).

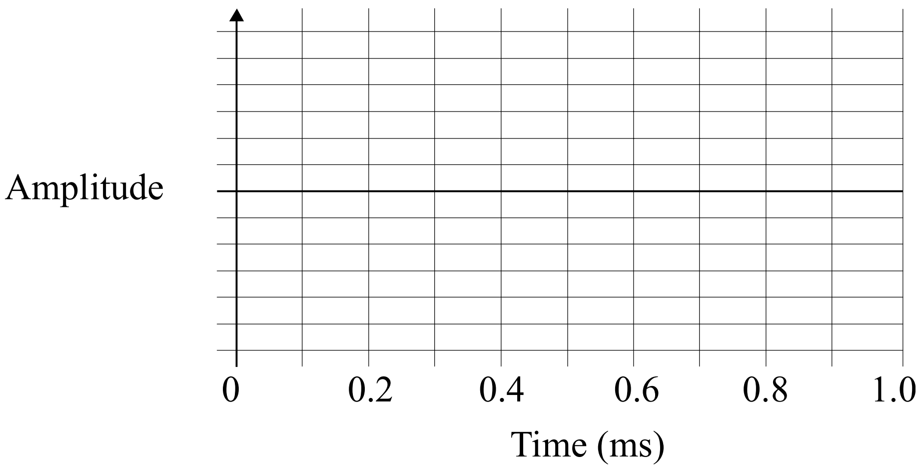


Figure 48 The Output of the Second IF Amplifier at tp28

■ By comparing the signal amplitude of tp24 and tp28, the gain of the second IF amplifier can be calculated.

The diode detector extracts the audio signal and removes, as nearly as possible, the IF signal.

Record the waveform at the output of the Diode Detector (tp31).

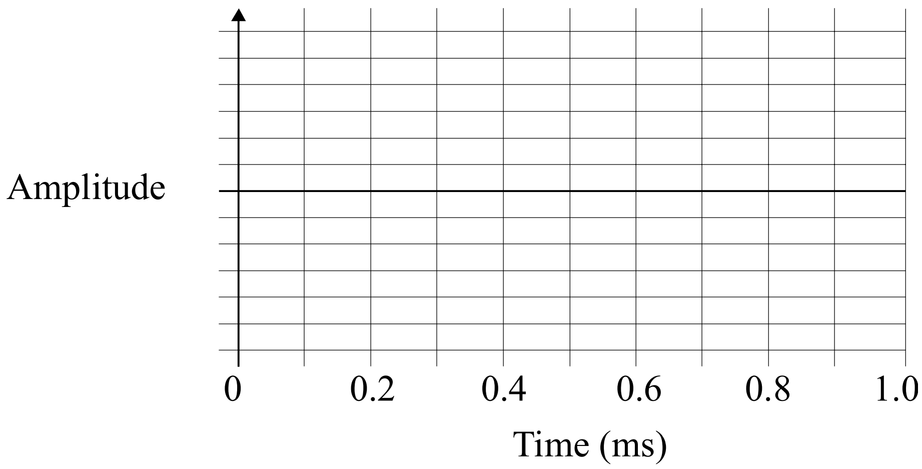


Figure 49 The Output of the Diode Detector at tp31

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

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.

3.20 Practical Exercise: Operation of the Automatic Gain Control Circuit (AGC)

AGC Practical Exercise Notes: ....

....

....

....

....

....

....

Chapter 4
Single Sideband Systems

4.1 A Final Look at DSB Transmissions

Double sideband transmissions were the first method of modulation developed and, for broadcast stations, are still the most popular. Indeed, for medium and long range broadcast stations it is the only system in use.

The reason for such widespread use is that the receiver design can be very simple and reliable. None of the characteristics are particularly critical so reception is still possible even in adverse conditions.

In this context, a broadcast is information transmitted for entertainment or information and available for use by anyone with a receiver. It never requires a response or acknowledgment for the receiving station. So in many ways it is similar to a newspaper or magazine which is published and distributed to anyone who is interested in reading a copy.

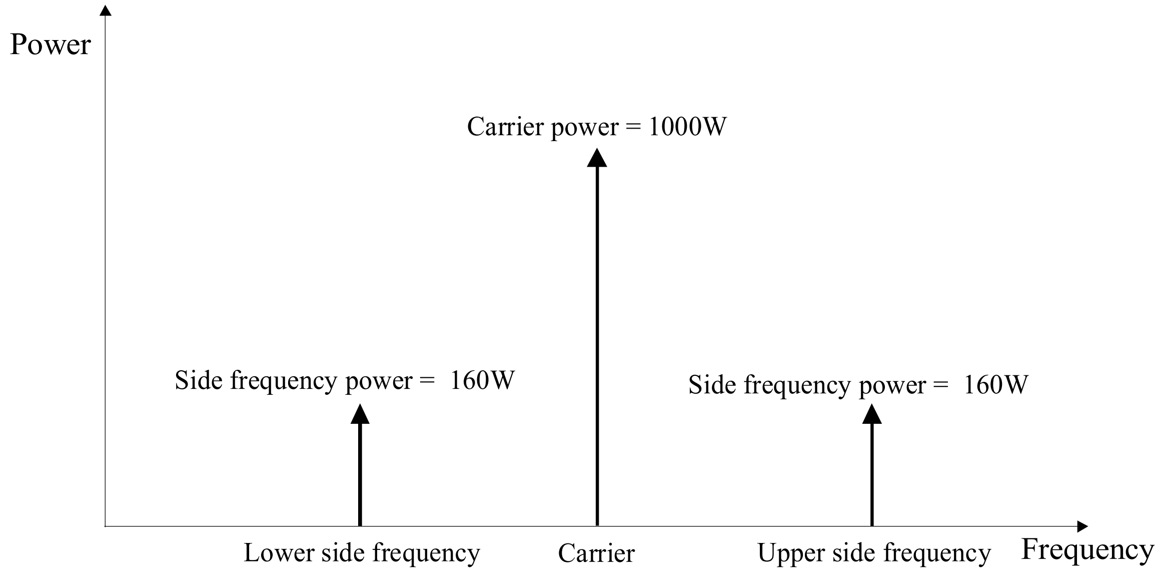
Radio is also used for communications in which the signal is addressed to a receiving station or a group of stations. Using the written word, this would correspond to a private letter or perhaps business or military information being exchanged. For this type of communication other systems are used, one of which is investigated in this chapter.

As we will see, there are two serious drawbacks to the DSB AM system.

4.2 DSB is Wasteful of Power

The first problem is to do with the power distribution in a DSB amplitude modulated wave.

To remind ourselves of the situation, try this calculation, having a glance at Chapter 2 for a quick reminder if necessary.



The total power being transmitted is $(1000) \cdot \left(1 + \frac{0.8^2}{2}\right) = 1320\text{W}$

Figure 51 Example DSB Spectrum

In Figure 51, we are transmitting a total power of 1.32kW. Of this power, the carrier contains 1kW and does not contain any of the information being transmitted. The side frequencies each have a power of 160W and each carries a copy of the same information signal. So, in this example, 1.32kW is being used in order to transmit only 160W.

4.3 DSB Has a Wide Bandwidth

When we amplitude modulated a carrier wave with a range of frequencies we generated an upper sideband consisting of the carrier frequency plus each of the components in the information wave together with the carrier wave minus each of the components.

4.4 How Much of the DSB AM Wave is Really Needed?

The whole purpose of the modulation system is to transfer information from one place to another. How efficiently does it achieve this?

We are transmitting two sidebands and a carrier.

The carrier contains no useful information at all and yet contains over half the total power. This is clearly a waste.

Even the sidebands can be improved. We can remember that combining the information signal and the carrier gave rise to an upper and a lower sideband, each of which contains a copy of the information being transmitted.

There is no necessity to send two copies of the same information. So this is a waste of power and bandwidth.

A waste of bandwidth?

There are more stations seeking permission to transmit than there are frequencies available. Within a band of say, 100kHz, we can transmit only 5 signals that occupy 20kHz, but 10 stations if they agree to limit their transmitted bandwidth to 10kHz. It is for this reason that we limit the highest frequency component within the information wave. High quality music transmissions on the medium waveband are therefore not allowed.

If so much of the transmitted wave is not required, then why transmit it?

We will now look at some of the alternatives.

4.5 Double Sideband Suppressed Carrier Transmission (DSBSC)

If we avoided using the carrier frequency shown in Figure 51, we would save ourselves 1kW of the transmitted power.

An example spectrum of the transmitted wave is shown in Figure 52.

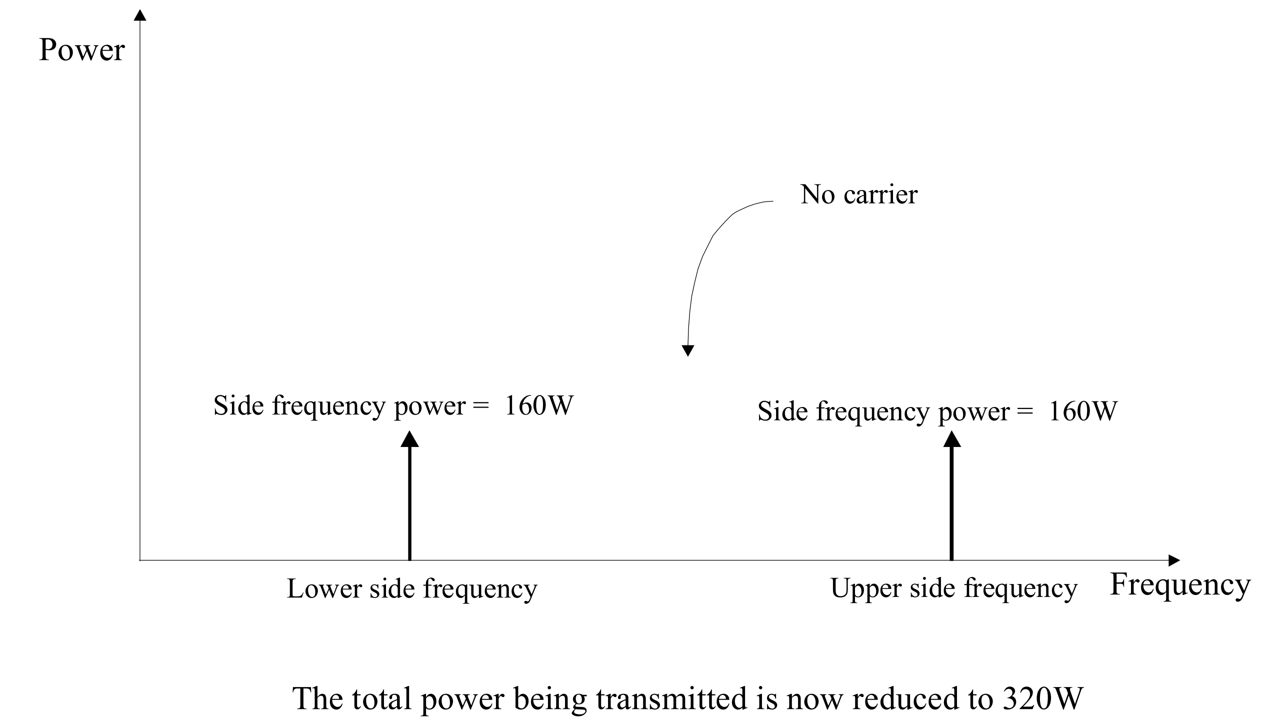


Figure 52 Example DSBSC Spectrum

You may be thinking 'that's a bit strange - how come the carrier can suddenly be removed when it was so important before'!

Well, the carrier has done its job - in the modulator. That is where we needed it to move or translate the audio signals up to radio frequency values that can be radiated by the antenna. This shifting, or translating of frequencies is the main function of a modulator.

At the transmitter, the carrier can easily be removed by a bandstop filter designed to eliminate the carrier frequency whilst allowing the two sidebands to be transmitted.

At the receiver, the carrier must be re-inserted to produce the modulation envelope to enable the detector to extract the information signal.

And here lies the problem.

The carrier has to be re-inserted at exactly the correct frequency to reproduce the original AM waveform (within a few Hertz). If it is not, there are serious problems with the reception.

Take a situation in which the upper and lower side frequencies are spaced 4kHz either side of the carrier at:

600 - 4 = 596kHz and 600 + 4 = 604kHz

Now, let's assume that the receiver carrier were to be re-inserted at an incorrect value of 601kHz. This would result in a spacing of only 3kHz between the carrier and the upper side frequency and 5kHz between the carrier and the lower side frequency.

What effect would this have?

Remembering our previous exercise in which we created an AM envelope by plotting a graph, we can see that these incorrect side frequency spacings will give rise to a badly deformed modulation envelope and hence a distorted output sound which makes speech sound like 'Donald Duck'.

With this type of transmission, the receiver would be carefully tuned in to the correct frequency and the station would be received. A few moments later, the re-inserted carrier frequency would drift slowly off tune and 'Donald Duck' would re-appear. We would have to reach over and retune the radio and settle back to enjoy the next few seconds of broadcast until the drift starts again.

The frequency control necessary to ensure that the re-inserted carrier stays at exactly the correct value regardless of changes of temperature, vibration etc. would make the receiver too complex and expensive for domestic use.

For this reason, DSBSC is very seldom used. Overall, the waste of transmitted power to send the carrier is less expensive than the additional cost of perhaps several million high quality receivers.

Such receivers are used for professional (and amateur) communications but are expensive, between ten and a hundred times the cost of a standard radio receiver.

4.6 Single Sideband Transmission (SSB)

This is just taking the previous reasoning to its ultimate conclusion. If we don't really need the carrier, we can leave it out and save power - this gives DSBSC transmission.

Just one step further and we can say that since both sidebands carry the same information, there is no point in transmitting both of them. It makes no difference which sideband is removed but in most systems the lower sideband is normally eliminated.

We can simply transmit a single sideband as shown in Figure 53 and by comparing the power use with Figure 51, we can see a considerable power saving.

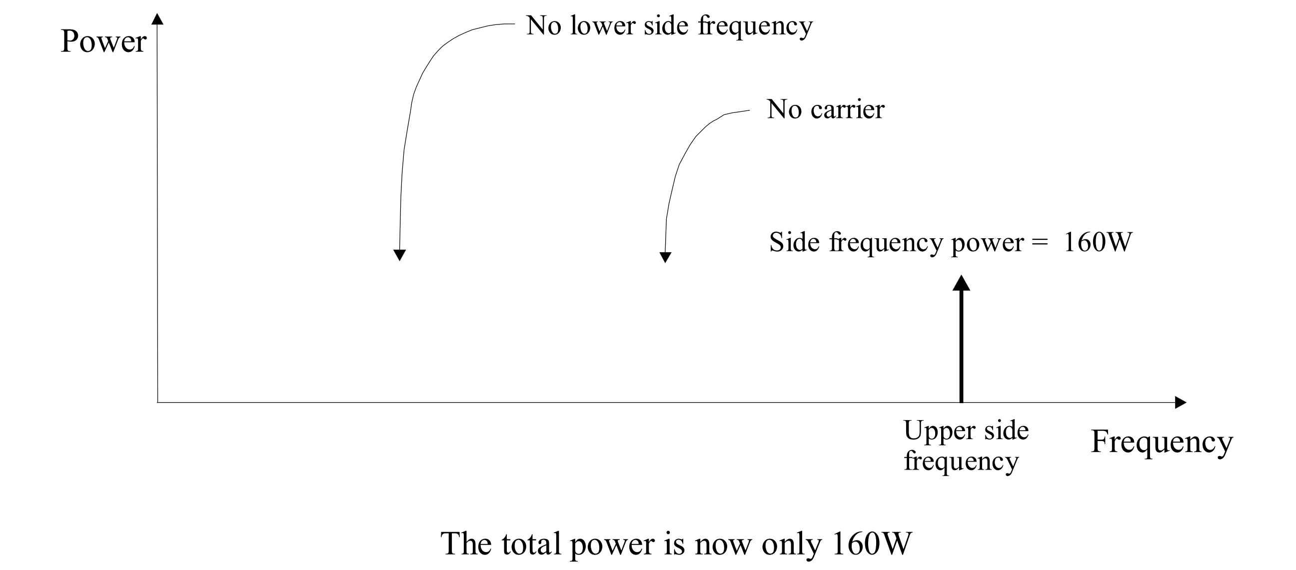


Figure 53 Example SSB Spectrum

The bandwidth of an SSB system is equal to the range of frequencies present in the information waveform whereas a DSB signal has a bandwidth twice as wide as the highest frequency component in the information signal. This also means a greatly reduced bandwidth for the system. In Figure 53, we are transmitting just a single frequency.

4.7 The SSB Transmitter

The design of the SSB transmitter is accomplished in two stages. First we generate a DSBSC signal and then remove the lower sideband to achieve the final SSB result.

4.8 Generating the DSBSC Signal

To do this, we use a Balanced Modulator. The principle of this circuit is shown in Figure 54.

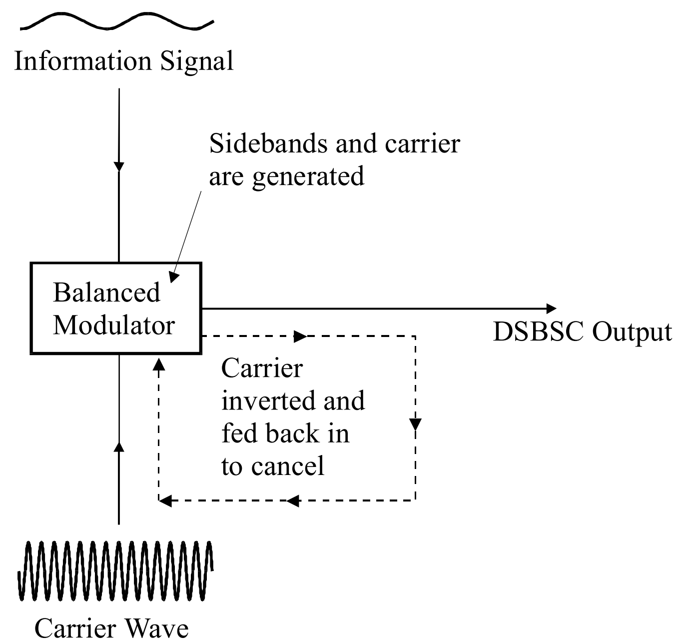


Figure 54 A Balanced Modulator

Internally, the balanced modulator generates the AM waveform, which includes the carrier and both sidebands. It then offers the facility to feed a variable amount of the carrier back into the modulator in anti-phase to cancel the carrier output. In this way we can balance out the carrier to suppress it completely leaving just the required DSBSC waveform.

4.9 Frequency Translating

One of the main uses of a modulator is that of frequency shifting.

Let us imagine that we had a 200Hz sinewave and required a 1500Hz sinewave. We could achieve this by using a modulator.

To do this, we could use a balanced modulator to amplitude modulate a 1300Hz signal with our 200Hz input. The DSBSC output would consist of the lower side frequency of 1100Hz and the upper side frequency of 1500Hz. By passing the signals through a bandpass filter with a center frequency of 1500Hz we can remove the unwanted frequency. We have used our modulator to change or translate the 200Hz to 1500Hz.

4.10 From DSBSC to SSB

This is basically the same pattern of events as we met in Question 4.9a. The DSBSC signal consists of the two sidebands, one of which can be removed by passing them through a bandpass filter.

On the ANACOM 1/1 this is achieved as shown in Figure 55.

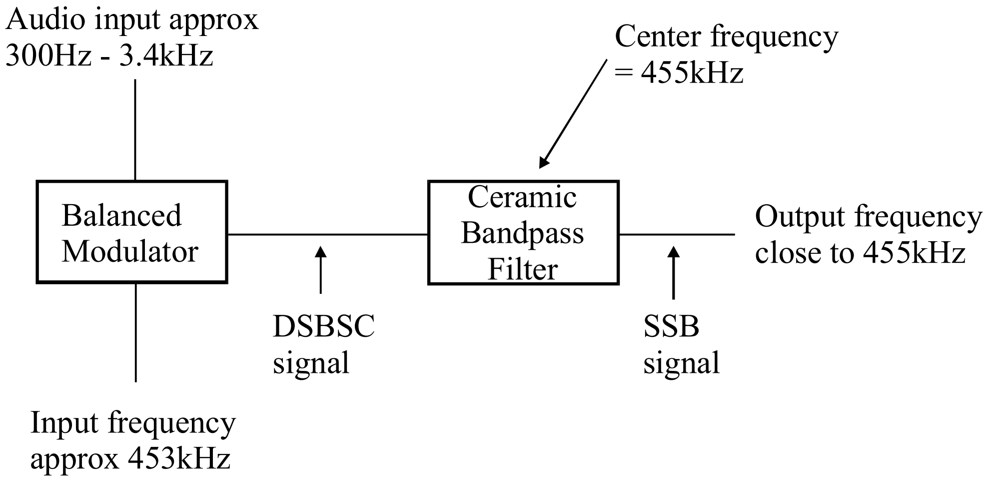


Figure 55

The inputs to the balanced modulator comprise the audio inputs from the audio oscillator, which extend from 300Hz to 3.4kHz, and the carrier input. On the ANACOM 1/1 board this carrier oscillator, although marked as ‘455kHz’, actually needs to operate at a frequency which is a little less than this, around 453kHz.

Why is this?

It is to ensure that the upper sideband can pass through the ceramic bandpass filter but the lower sideband cannot pass through.

In Figure 56 opposite, the upper sideband can be seen to be within the passband of the ceramic filter but the lower sideband is outside and will therefore be rejected. The sideband frequencies are quite close to each other and a good quality ceramic filter is required. A ceramic filter passes only a narrow range of frequencies with a sharp cut-off outside of its passband.

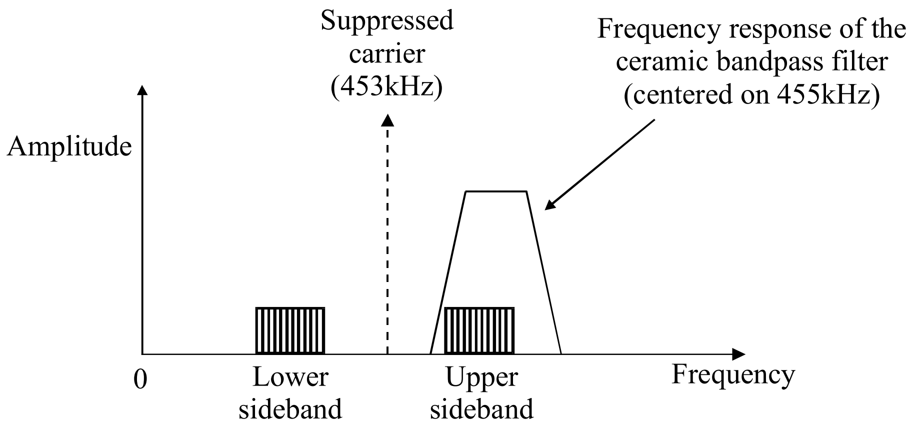


Figure 56

4.11 Transmitting the SSB Signal

So far, we have got an SSB signal but it is at a frequency around 455kHz. This is too low since we need it to be within the medium wave band if we are to hear it on our receiver.

We need to shift or translate the signal to a higher frequency. We know how to do this. We simply pass it through a balanced modulator and filter out the unwanted frequency. In the ANACOM 1/1 transmitter, we use the same 1MHz carrier that we used for the AM transmission in Chapter 3.

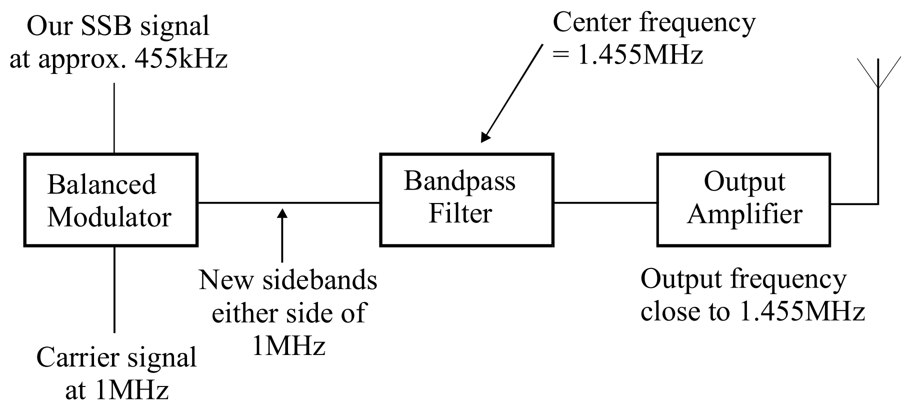


Figure 57

In Figure 57 our SSB signal that we have just generated is combined with a 1MHz carrier signal to produce a new DSBSC signal.

This signal will now have two new sidebands, one around 1MHz + 455kHz = 1.455MHz and the other at 1MHz - 455kHz = 545kHz. Since these two sidebands are separated by a wide frequency range, the filter design is not critical and a simple parallel tuned circuit is sufficient.

The 1.455MHz output signal then only requires amplification before transmission.

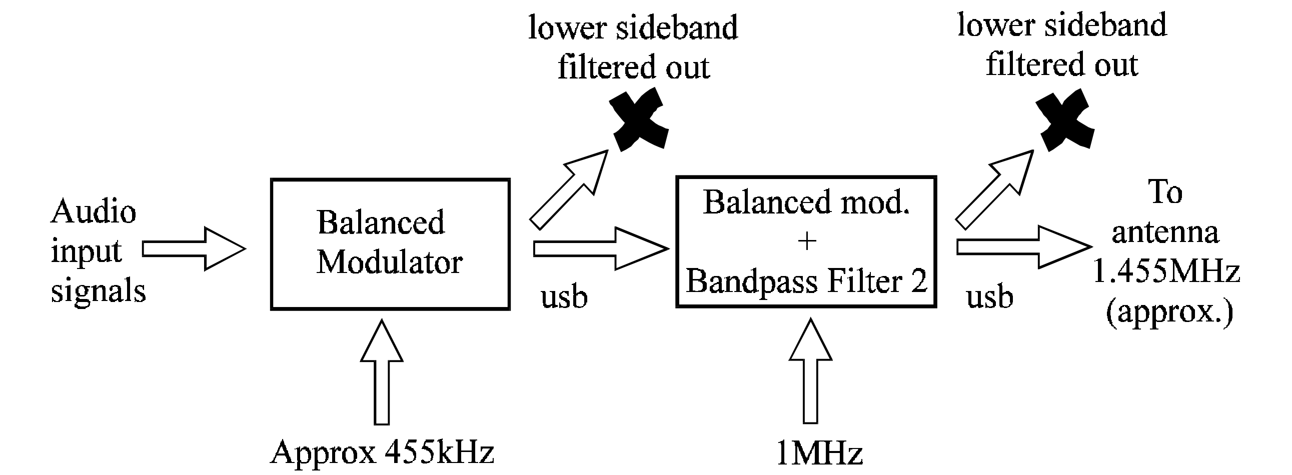


Figure 58 A Summary of the Frequency Changes That Have Taken Place

4.12 At the Receiver

The receiver is of the normal superhet design. The first stages are the same as we met in Chapter 3.

The incoming signal is amplified by the RF Amplifier and passed to the mixer. The other input to the mixer is the local oscillator that is running at 455kHz above the frequency to which the receiver is tuned. The mixer generates sum and difference signals and the lower of the two is the resulting IF signal occupying a range of frequencies around 455kHz.

The audio information must now be separated from these IF frequencies.

4.13 Recovering the Audio Signals

This is achieved by a circuit called an SSB AM decoder. It does the same job as a demodulator or detector in a DSB AM receiver. The SSB AM decoder is slightly more complicated when compared with the DSB equivalent (see Figure 59).

One way of extracting the audio signals is to use a mixer to shift the frequencies just as we have done several times already.

If a mixer combined an input of (audio + 455kHz) with another input of 455kHz the resultant outputs would be the usual 'sum' and 'difference' frequencies.

The 'sum' would be (audio + 455kHz) + (455kHz) = (audio + 910kHz) which is far too high a frequency to be of much interest to us.

The 'difference' frequency is just what we wanted (audio + 455kHz) - (455kHz) = (audio).

In the ANACOM 1/2, the mixer is called the product detector and the 455kHz input to the product detector is provided by an oscillator called a 'Beat Frequency Oscillator' or BFO.

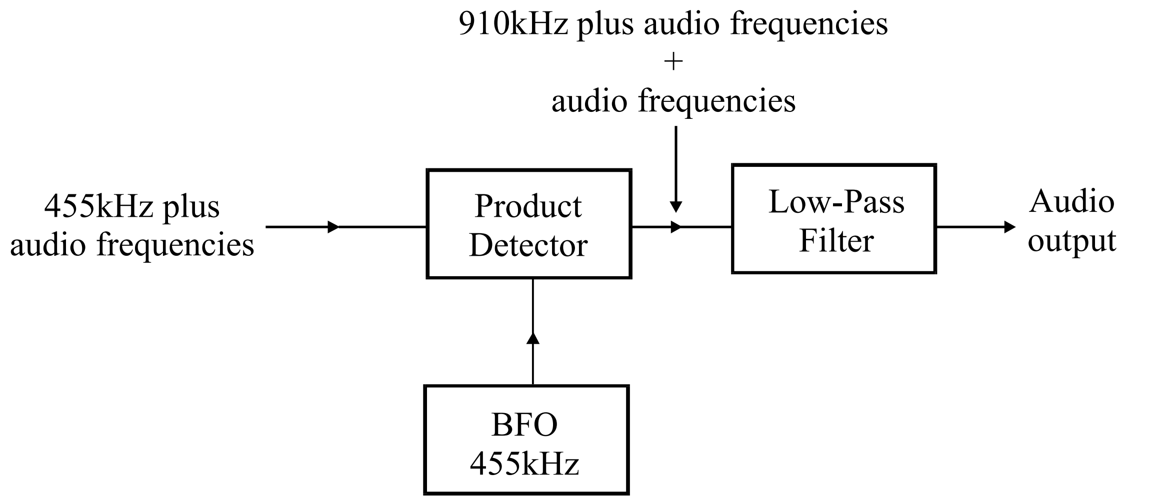


Figure 59 An SSB AM Decoder

The output amplifier and the loudspeaker perform in exactly the same way as we have seen previously.

4.14 Practical Exercise: A Single Sideband Transmitter

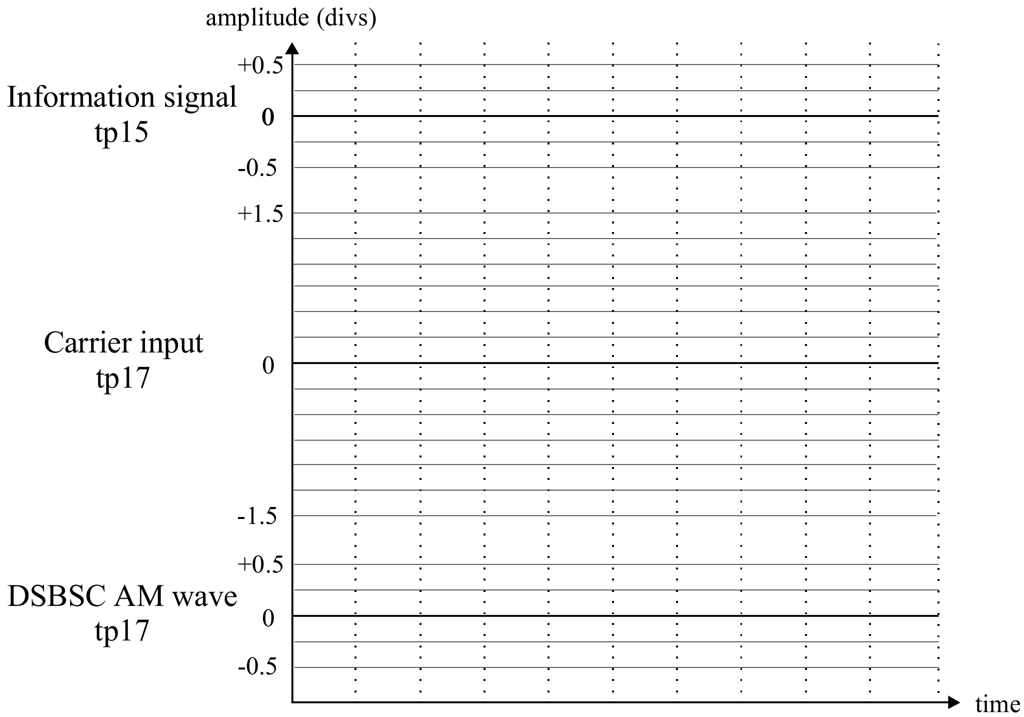


Figure 61

SSB Transmitter Practical Exercise Notes: ....

....

....

....

....

....

....

....

....

....

4.15 Practical Exercise: Receiving the SSB Signal

SSB Receiver Practical Exercise Notes: ....

....

....

....

....

....

....

....

....

....

....

Notes:

....

....

....

....

....

....

....

....

....

....

Notes:

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

Chapter 5
Locating Faults in DSB and SSB Transmission Systems

5.1 The Nature of Faults

Generally, faults occur one at a time. If several, or many, circuits fail suspect something common to all the failed circuits such as the power supplies.

5.2 General Approaches to Fault Finding

There are two methods which may be employed on the ANACOM 1/1 and ANACOM 1/2 boards and they are described in order of usefulness - the first being preferred.

5.3 The Half-Split Method

In this method, the suspect area is divided into two approximately equal sections by checking at the mid-point. Assume a system involving eight circuits as shown in Figure 65 below:

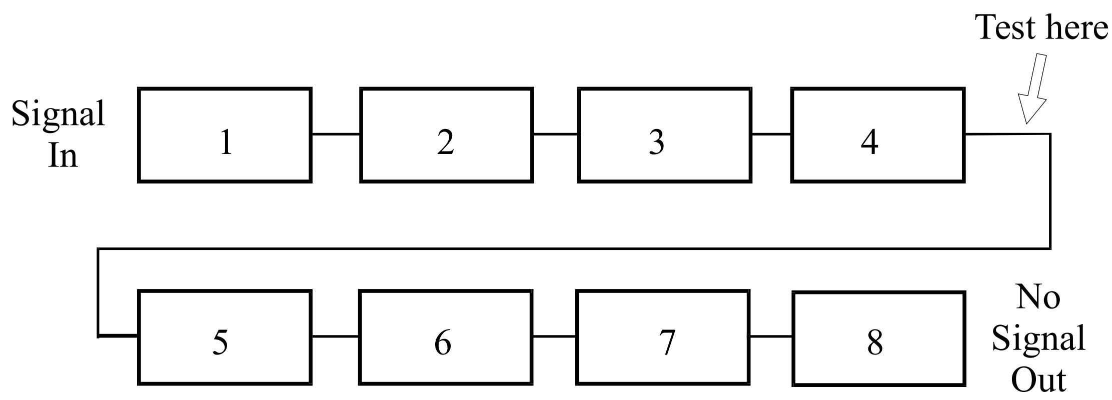


Figure 65

All we know at the moment is that a signal is being applied at the input and does not appear at the output.

The first step is to use an oscilloscope to test for a signal at the mid-point. Let us assume that the correct signal is present. This tells us that circuits 1, 2, 3 and 4 are working and therefore the fault must be in the remaining circuits 5, 6, 7 or 8.

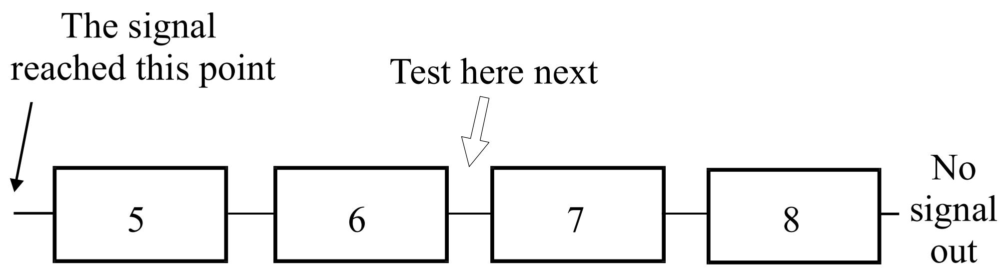


Figure 66

Testing between circuits 6 & 7 and finding no signal would tell us that the fault lies in either circuit 5 or circuit 6.

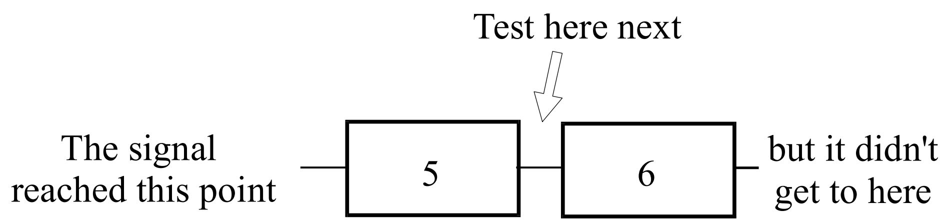


Figure 67

Testing between circuits 5 & 6 and finding a signal would tell us that circuit 6 must be the culprit.

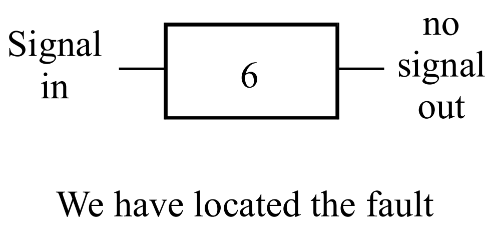


Figure 68

5.4 The Plod-Through Method

This method locates faults by starting at one end and checking the waveform at each stage all through the circuit until the fault is found. This works, but is generally a slow method. But any method is better than just taking random measurements.

We can work from the output back towards the input until the signal is located, or alternatively we could start from the beginning and work towards the end. The route taken is a matter of personal preference. As with the half-split method, it still relies on us understanding the system well enough to be know what to expect at each point.

Once you know which block is faulty, similar processes can be applied to home in on small circuits or even to continue to component level.

Notes:

....

....

....

....

....

....

....

....

....

....

....

....

....

5.5 Making Tests

Work slowly and methodically. Most fault finding time is spent sitting and thinking (not just sitting!).

Collect the symptoms and write them down.

Consider carefully which test would be most useful and the result you would expect from the test. Write down the test made and the results. This will allow you to gather all the relevant data together and prevent needless repetition, when you realize you have forgotten the previous result.

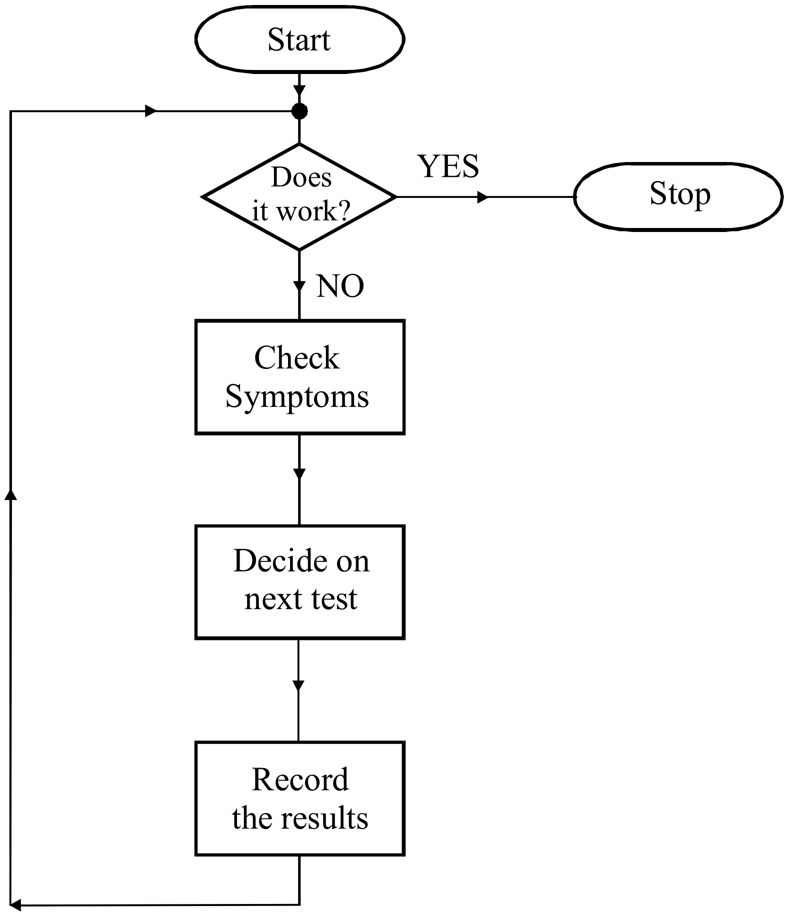


Figure 69 A General Fault Finding Procedure

5.6 A Few 'NEVERS' With Integrated Circuits (ICs)

(i) Never apply input signals to an IC when the power supply is OFF.

(ii) Never apply an input voltage which is greater in value than the supply voltage to the IC (well, almost never - there are a couple of special ICs that allow this).

(iii) Never reverse the supply polarity to an IC.

(iv) Never plug an IC into the board the wrong way round. One end of each IC has an indentation in it to tell you which end has pin number 1. On the board there is usually a corresponding mark, often a white dot or a figure 1. The pins are numbered counter-clockwise starting with pin 1 at the top left-hand corner of the IC.

(v) Never touch the pins of an IC. Any static electricity on your hands can easily damage it.

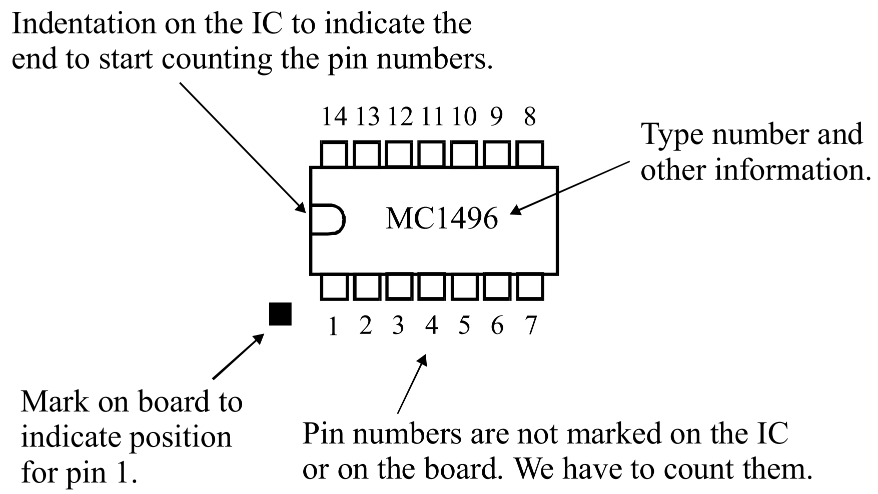


Figure 70

5.8 Practical Exercise: Fault Finding on an Amplitude Modulated Transmitter

Fault 1 - Tests Made and Results:

....

....

....

....

....

....

Fault 2 - Tests Made and Results:

....

....

....

....

....

....

Fault 3 - Tests Made and Results:

....

....

....

....

....

....

Fault 4 - Tests Made and Results:

....

....

....

....

....

....

Fault 5 - Tests Made and Results:

....

....

....

....

....

....

Fault 6 - Tests Made and Results:

....

....

....

....

....

....

Fault 7 - Tests Made and Results:

....

....

....

....

....

....

Fault 8 - Tests Made and Results:

....

....

....

....

....

....

Notes:

....

....

....

....

....

....

....

....

5.9 Practical Exercise: Fault Finding on an Amplitude Modulated Receiver

Fault 1 - Tests Made and Results:

....

....

....

....

....

....

Fault 2 - Tests Made and Results:

....

....

....

....

....

....

Fault 3 - Tests Made and Results:

....

....

....

....

....

....

Fault 4 - Tests Made and Results:

....

....

....

....

....

....

Fault 5 - Tests Made and Results:

....

....

....

....

....

....

Fault 6 - Tests Made and Results:

....

....

....

....

....

....

Fault 7 - Tests Made and Results:

....

....

....

....

....

....

Fault 8 - Tests Made and Results:

....

....

....

....

....

....

Notes:

....

....

....

....

....

....

....

....

Notes:

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

Chapter 6
The ANACOM 2 Board

6.1 Layout Diagram of the ANACOM 2 Board

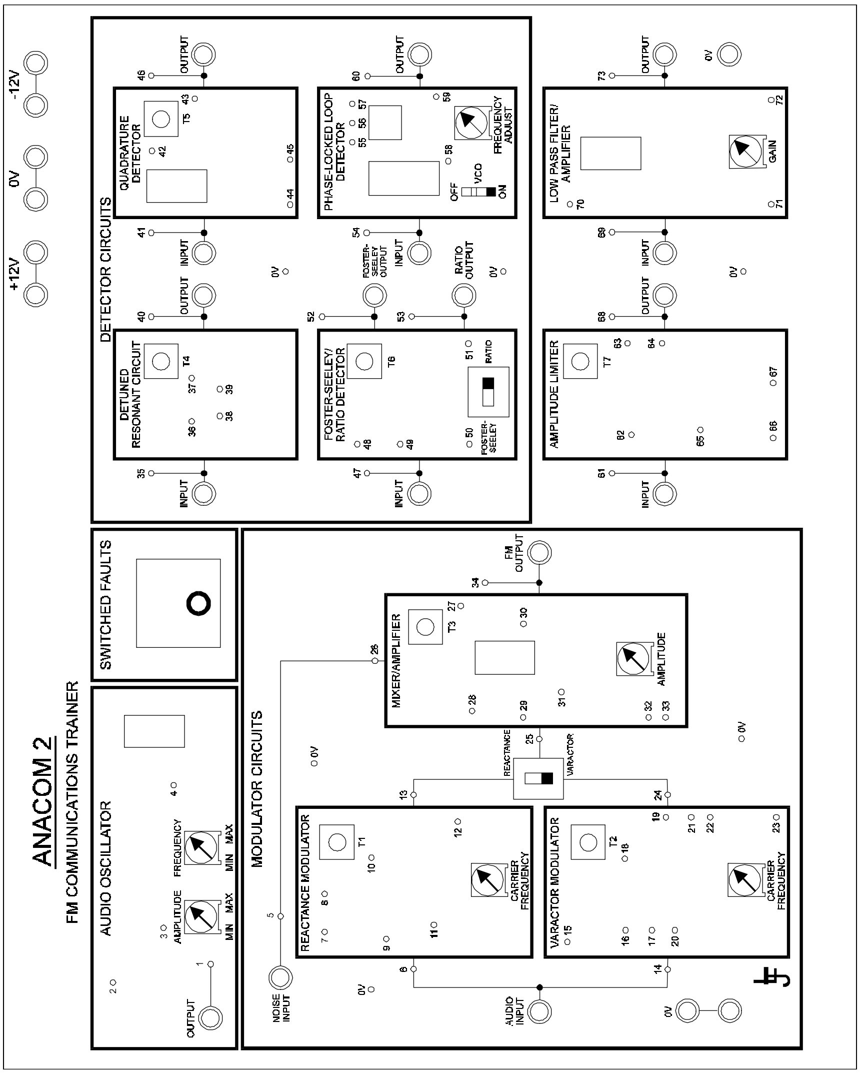


Figure 71

6.2 The ANACOM 2 Board Blocks

The board can be considered as five separate blocks:

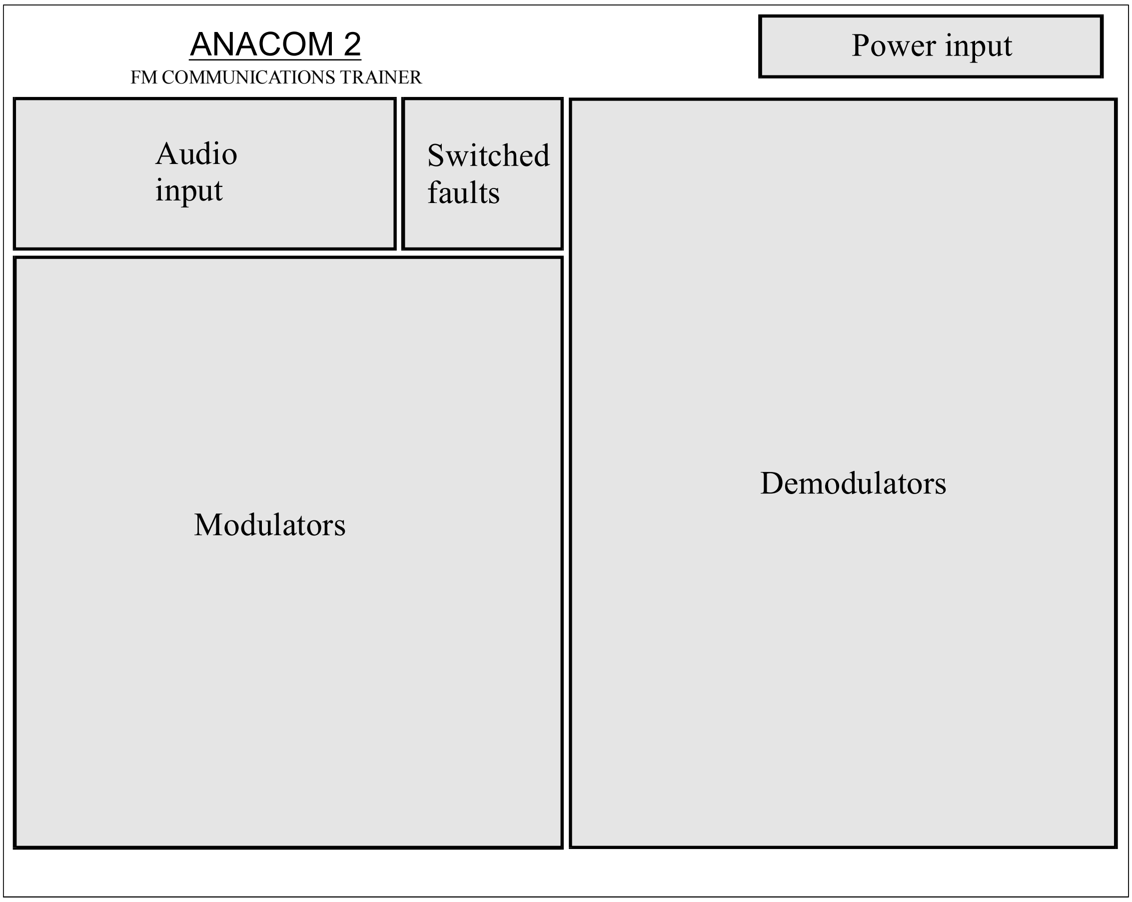


Figure 72

6.3 Power Input

We will start with the simplest block. 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.

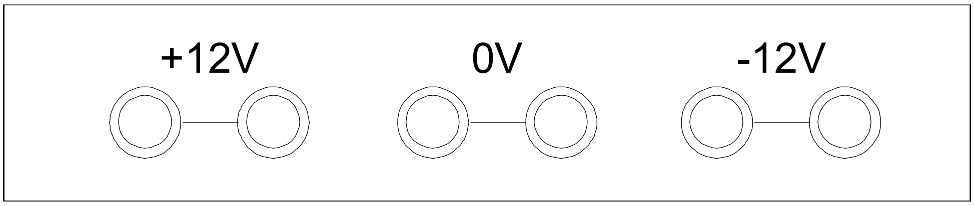


Figure 73

6.4 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 us to supply our own audio information signals.

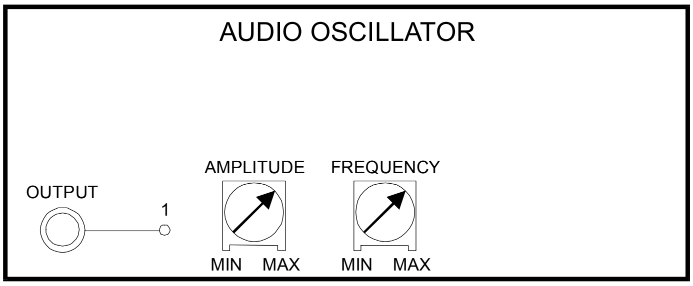


Figure 74

6.5 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).

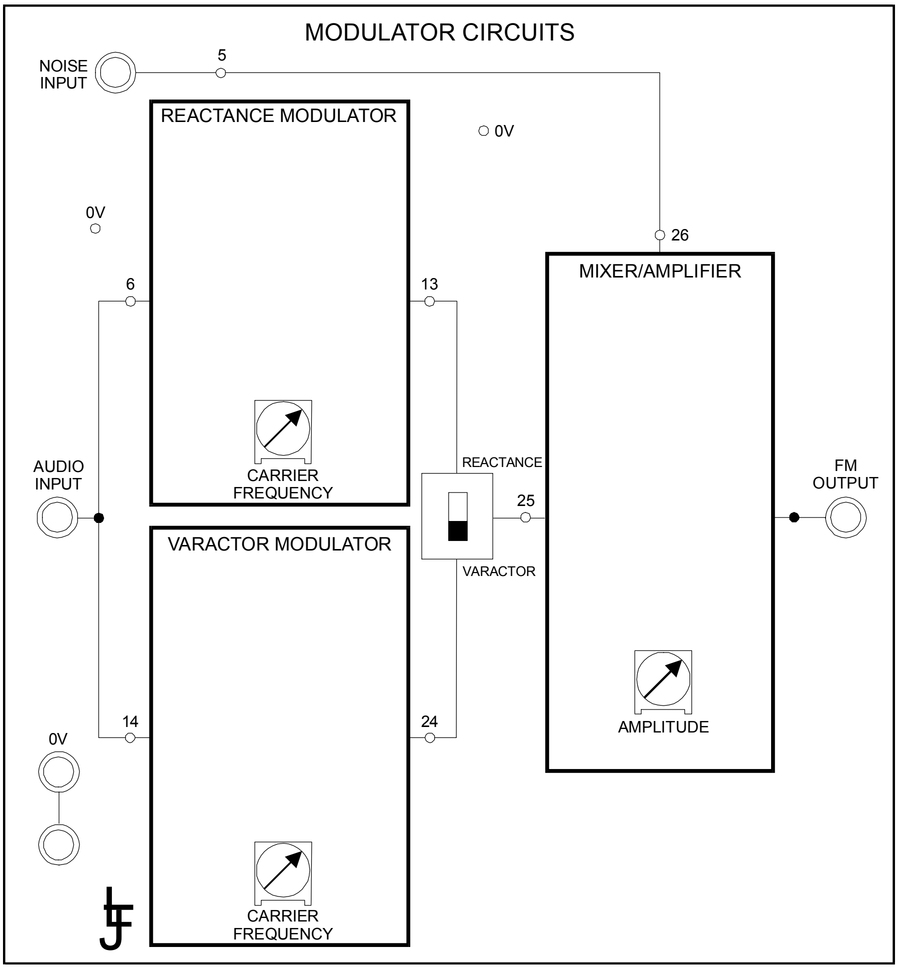


Figure 75

6.6 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 2 board is fully operational, all switches should be set to OFF before use. 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.

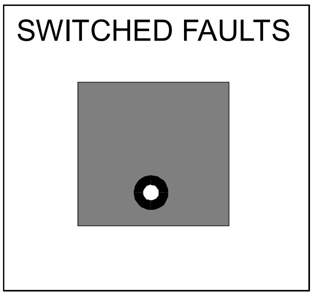


Figure 76

Notes:

6.7 The Detector Circuits

These circuits extract the incoming information signal from the FM signal generated by the modulator circuits. Put briefly, each detector undoes the work of the modulator. Detectors are also referred to as ‘demodulators’. Four different forms of detectors are available (Detuned, Quadrature, Foster-Seeley/Ratio and Phase-Locked Loop).

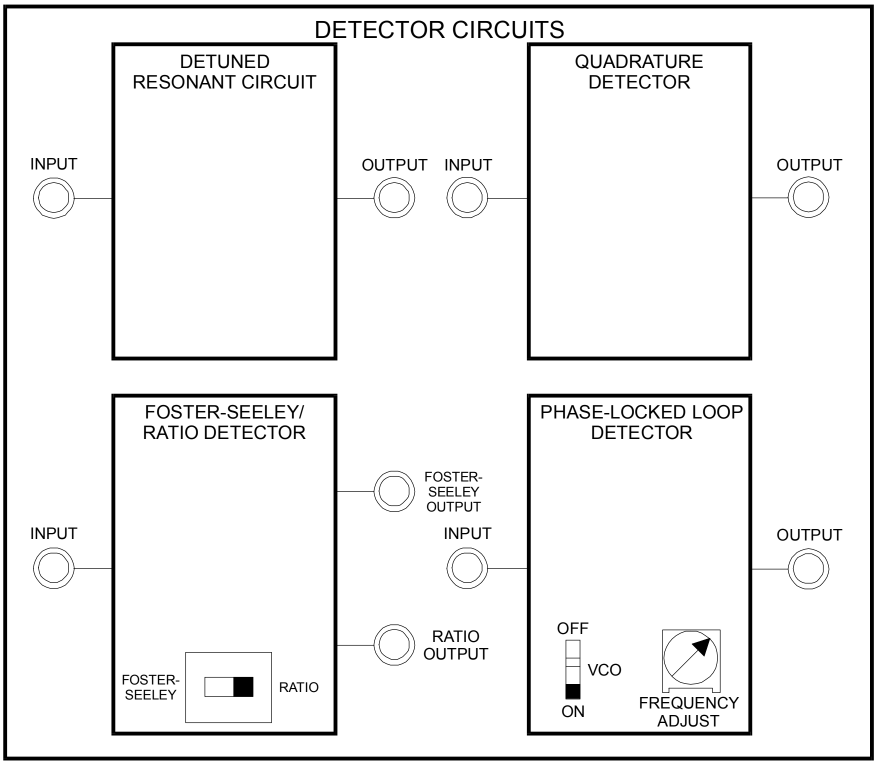


Figure 77

6.8 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.

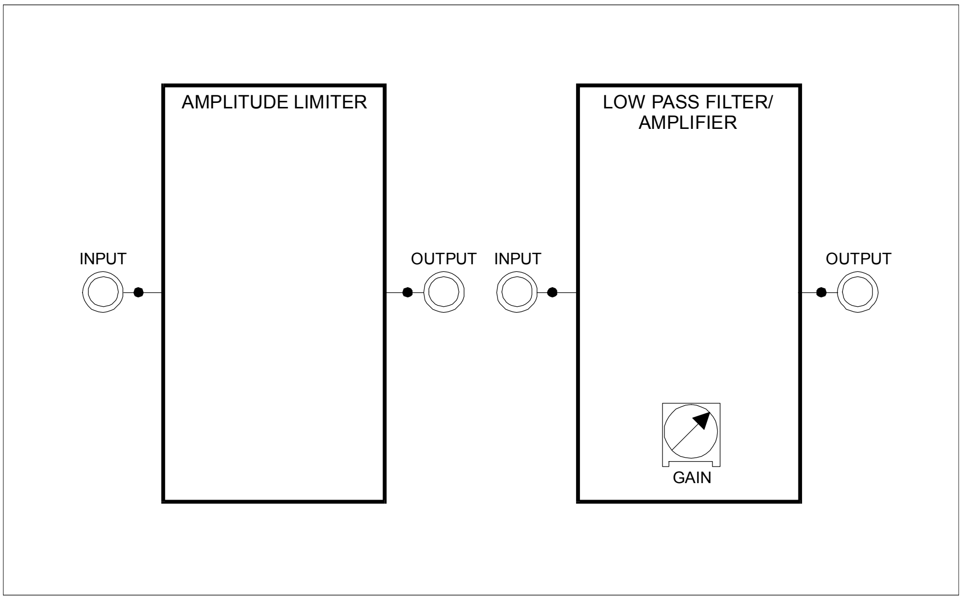


Figure 78

Notes:

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

Chapter 7
FM Modulators

7.1 Frequency Modulation (FM)

As we saw in Section 2.5, 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.

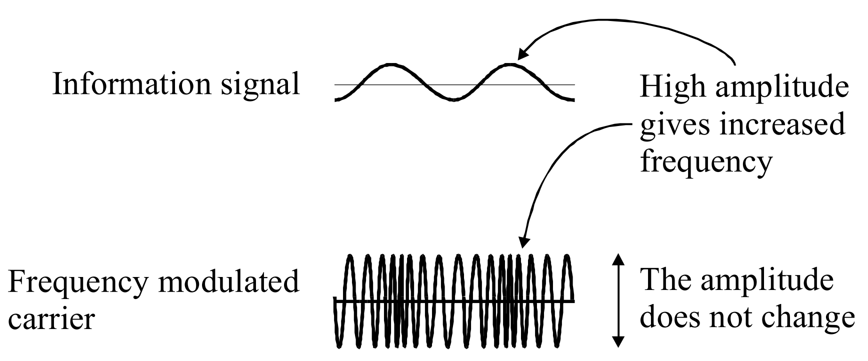


Figure 79

Notice in Figure 79 that the amplitude is not affected by the modulation process.

7.2 Frequency Deviation

How much the frequency is changed for each volt of information signal is called the 'Frequency Deviation' and a typical value is 15kHz/V with an upper limit of ±75kHz

As an example, an information signal of peak-to-peak voltage of 6 volts and a frequency of 10kHz with a frequency deviation of 15kHz/V would cause an FM carrier to change by a total of 90kHz (45kHz above and below the original carrier frequency). The carrier frequency would be swept over this range 10,000 times a second.

7.3 The Advantages of FM

There are three advantages of frequency modulation for a communication system.

(i) In the last section, we saw that the information signal controlled the frequency of the carrier but had no effect on its amplitude. Now, when any transmission is affected by electrical noise, the noise signal is superimposed on the transmitted signal as shown in Figure 80 below.

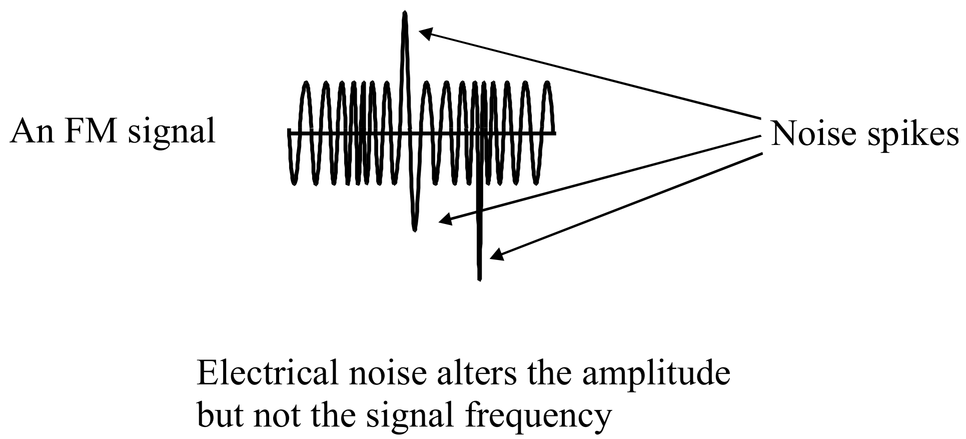


Figure 80

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.

(ii) The bandwidth of the FM signal is very wide compared with an AM transmission. Typical broadcast bandwidths are in the order of 250kHz. This allows a much better sound quality, so signals like music sound significantly better if frequency modulation is being used.

(iii) 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.

7.4 The Disadvantage of FM

This is the wide bandwidth of the transmission.

The medium frequency broadcast band extends from about 550kHz to 1,600kHz, and is therefore only a little over 1MHz in width. If we tried to use FM using a bandwidth of 250kHz 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 that extends from about 85MHz to 110MHz. This is a width of 25MHz and would hold many more stations.

7.5 The Bandwidth of an FM Signal

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 250kHz 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 that 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 25kHz but this is only for speech and the quality is not important.

These bandwidth figures bear no easy relationship with the frequency of the information signal nor with the frequency deviation - or, it seems, anything else. FM is unlike AM in this respect.

7.6 An FM Transmitter

As we can see from Figure 81 below, the FM transmitter is very similar to the AM transmitter that we met in Figure 31 of Chapter 3.

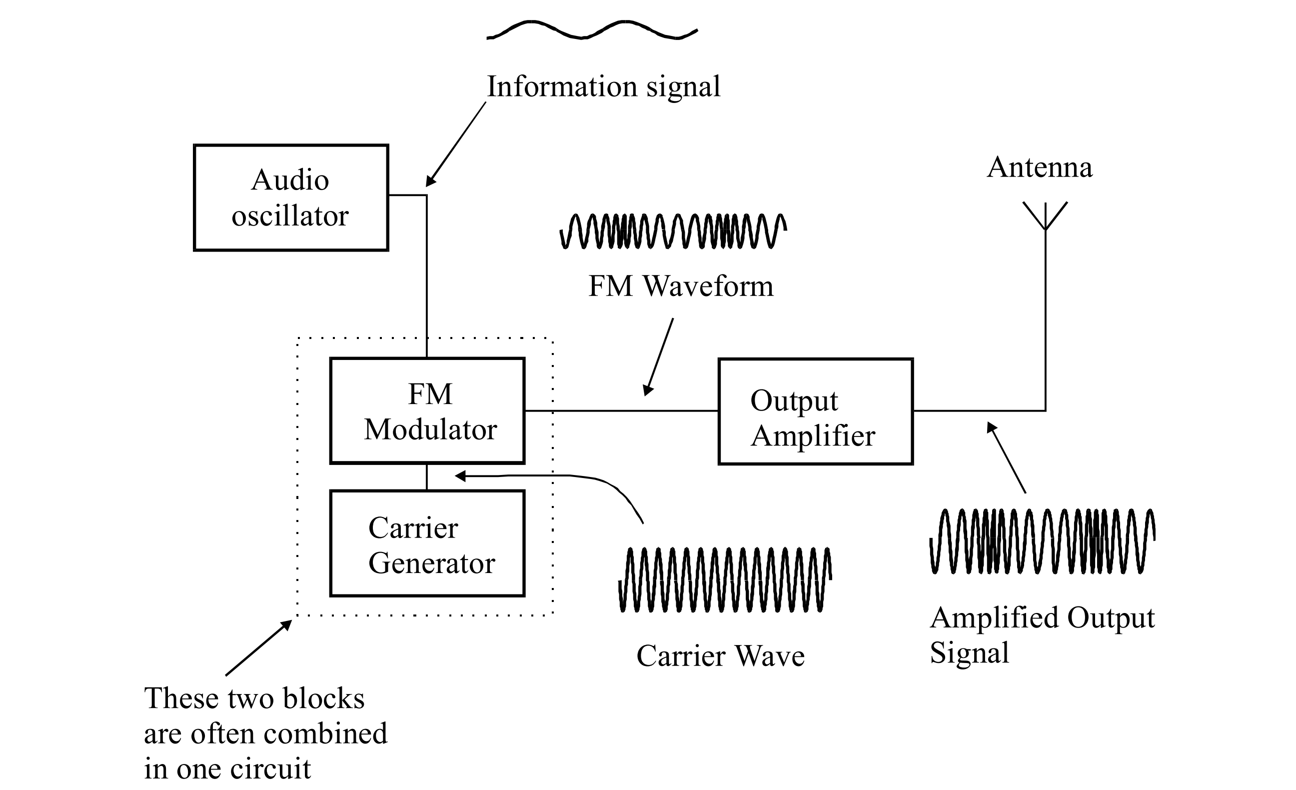


Figure 81 An FM Transmitter

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 sinewave signals that we are using with ANACOM 2.

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. It doesn't have to be, but in our case, it is.

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.

7.7 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.

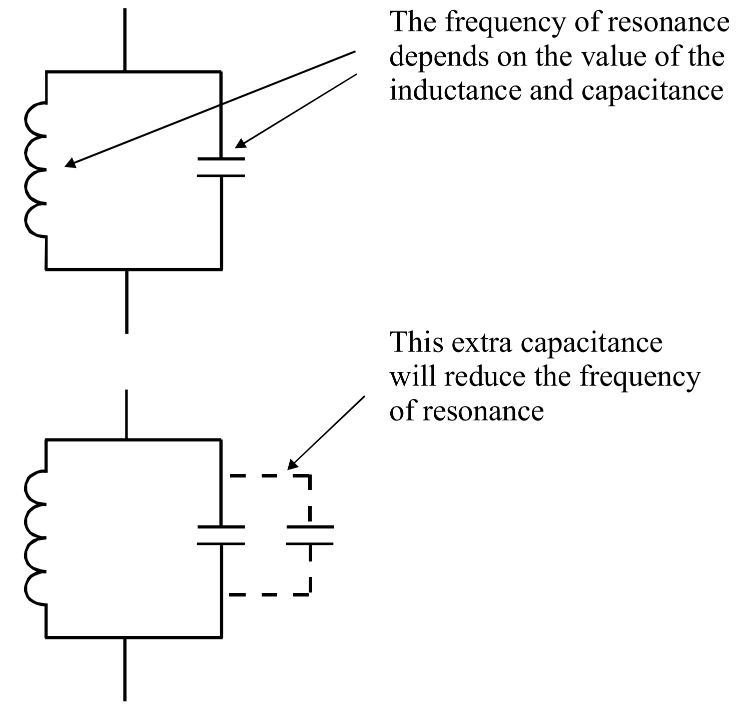


Figure 82

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:

$$f_o = \frac{1}{2\pi\sqrt{LC}} \text{Hz}$$

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 so will the carrier frequency. This is demonstrated in Figure 83 below:

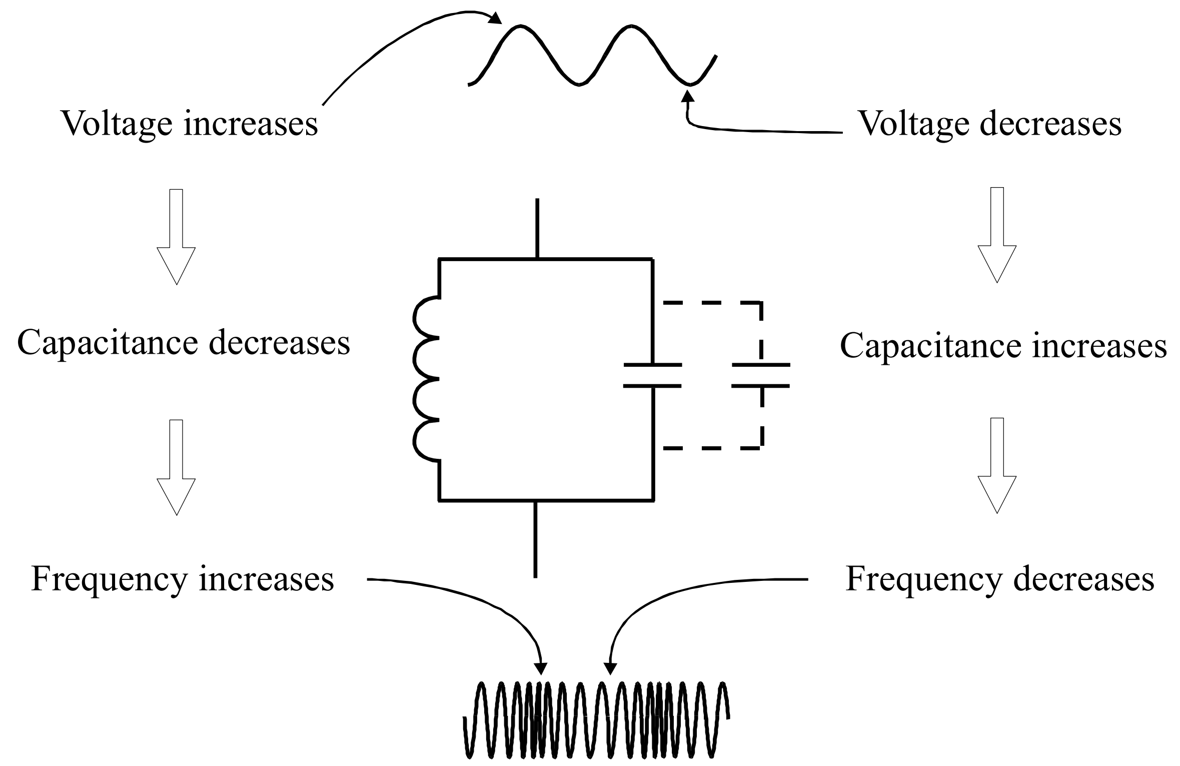


Figure 83

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.

7.8 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.

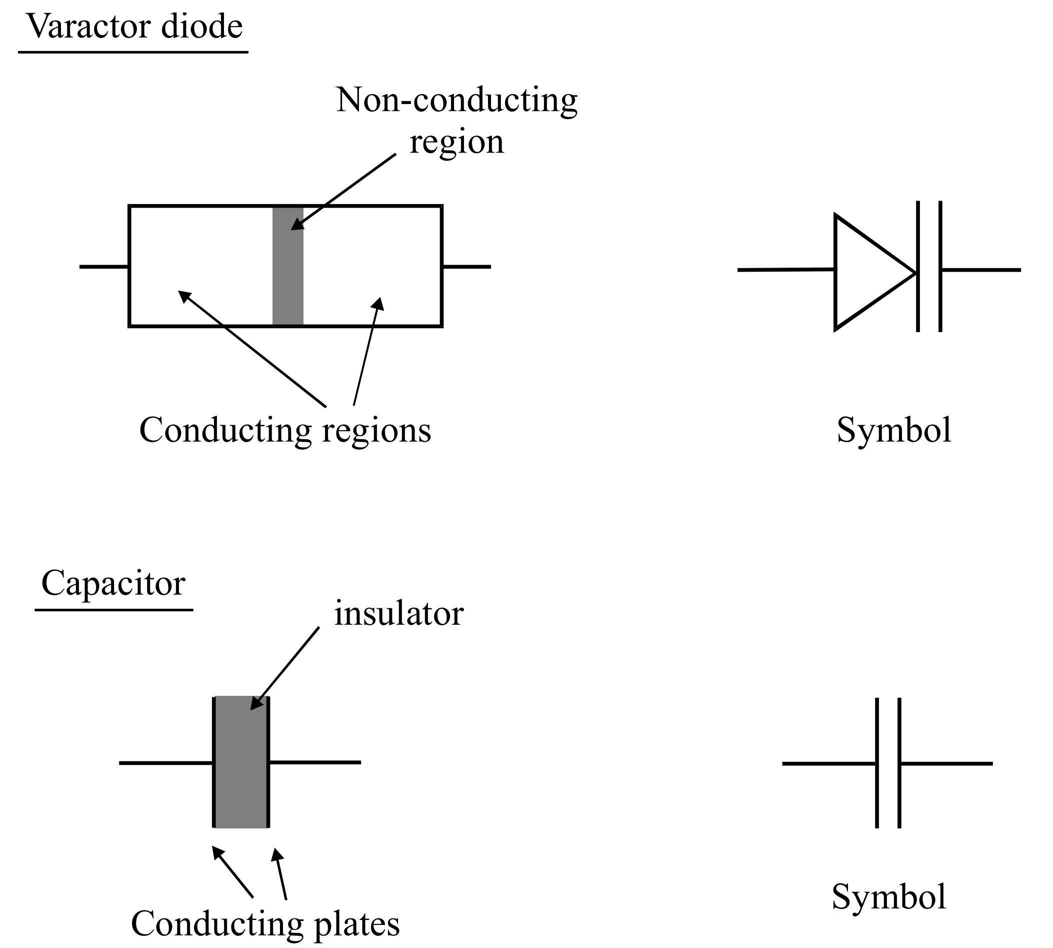


Figure 84

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 85.

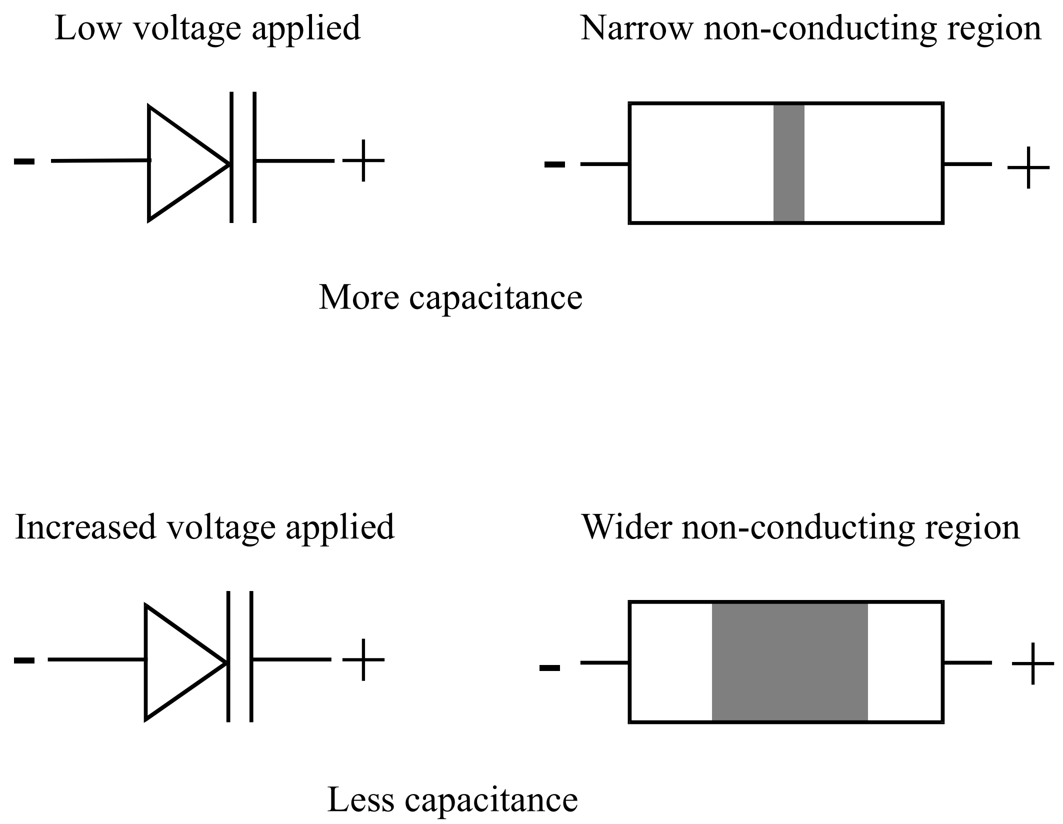


Figure 85 Operation of the Varactor Diode

If the information signal is applied to the varactor diode, the capacitance will therefore be increased and decreased in sympathy with the incoming signal.

Notes:

7.9 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 Figure 86 below.

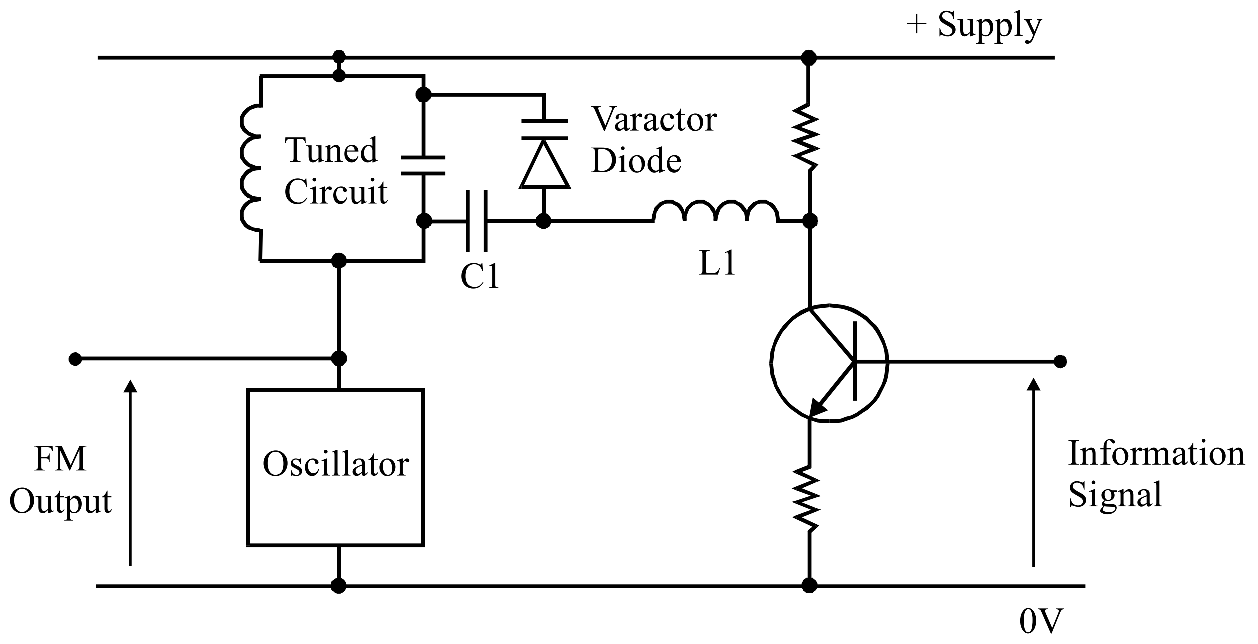


Figure 86 The Varactor Modulator Circuit

We can see the tuned circuit that 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 C1 and L1. C1 is a DC blocking capacitor to provide DC isolation between the Oscillator and the collector of the transmitter. L1 is an RF choke that allows the information signal through to the varactor but blocks the RF signals.

7.10 The Operation of the Varactor Modulator

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

(ii) This low frequency signal passes through the RF choke and is applied across the varactor diode.

(iii) 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.

(iv) 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.

7.11 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.

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 87 opposite.

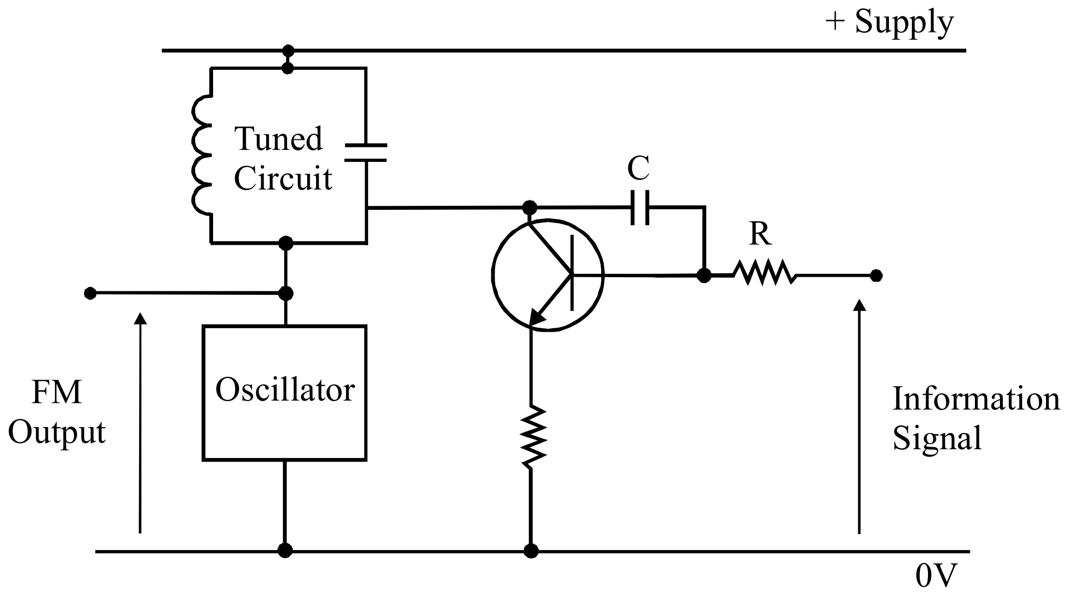


Figure 87 The Reactance Modulator Circuit

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.

7.12 Making the Capacitor Variable

Surprisingly enough, this part is very easy.

The size of the capacitance depends on the change in collector current that 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.

7.13 The Reactance Modulator Circuit

Figure 87 shows a complete reactance modulator. We can see that the left-hand half is the same as in the previous varactor modulator - simply an oscillator and a tuned circuit, which between them generates the unmodulated carrier.

The capacitor 'C' and the resistor 'R' are the two components used for the phase shifting described in Section 7.11 and, together with the transistor, form the voltage controlled capacitor.

This voltage controlled capacitor is actually in parallel with the tuned circuit. This is not easy to see but Figure 88 below may be helpful.

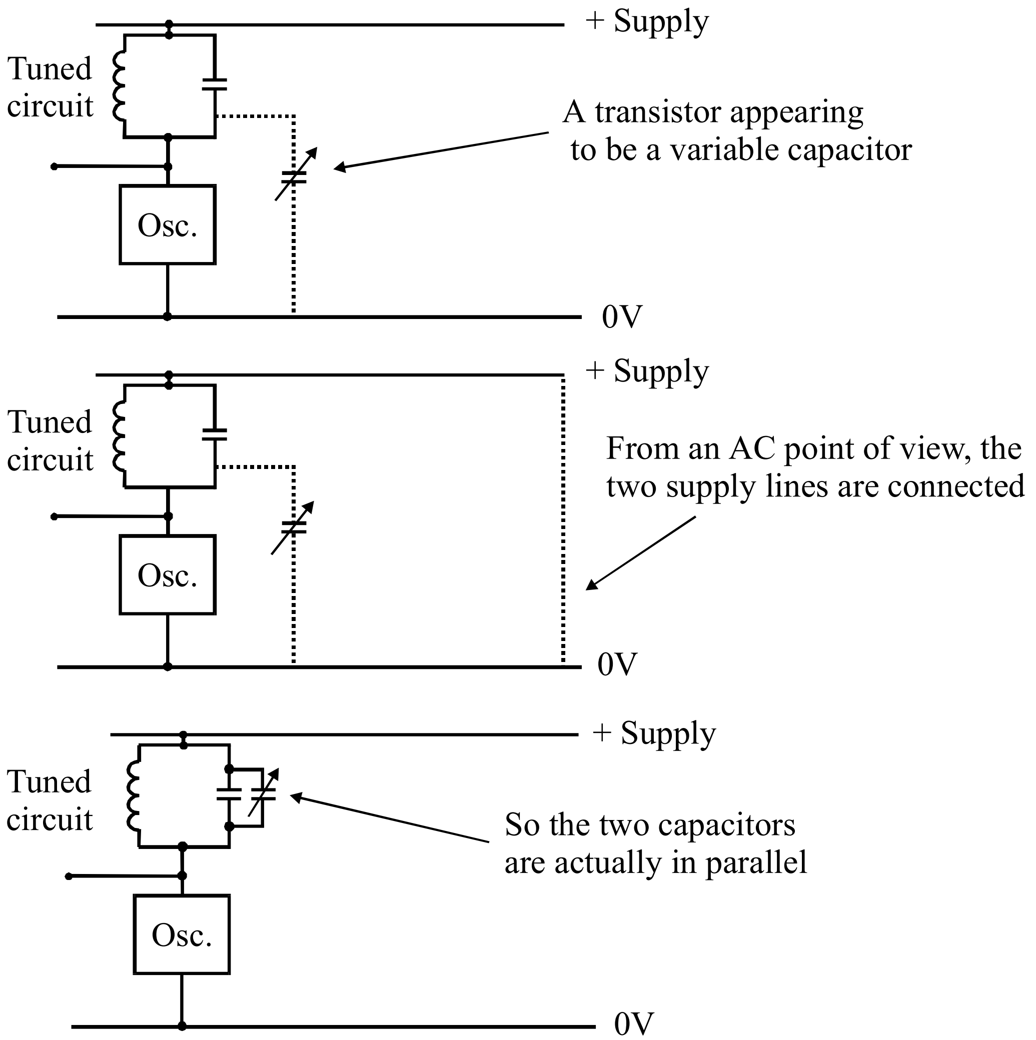


Figure 88

In the first part of the diagram, the capacitor and associated components have been replaced by the variable capacitor, shown dotted.

In the next part, the two supply lines are connected together. We can justify this by saying that the output of the DC power supply always includes a large smoothing capacitor to keep the DC voltages at a steady value. This large capacitor will have a very low reactance at the frequencies being used in the circuit - less than a milli-ohm. We can safely ignore this and so the two supply lines can be assumed to be joined together. Remember that this does not affect the DC potentials, which remain at the normal supply voltages.

If the two supply lines are at the same AC potential, the actual points of connection do not matter and so we can redraw the circuit as shown in the third part of the diagram.

7.14 The Operation of the Reactance Modulator

If required, reference can be made to Figure 87.

(i) The oscillator and tuned circuit provide the unmodulated carrier frequency and this frequency is present on the collector of the transistor.

(ii) The capacitor and the resistor provide the 90° phase shift between the collector voltage and current as described in Section 7.11. This makes the circuit appear as a capacitor.

(iii) The changing information signal being applied to the base has the same effect as changing the bias voltage applied to the transistor and, as we saw in Section 7.12, this would have the effect of increasing and decreasing the value of this capacitance.

(iv) 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.

7.15 Practical Exercise: The Varactor Modulator

The oscillator output frequency measured at tp34 and the DC input voltage measured at tp21 have values of: ....

....

With the Carrier Frequency preset in its maximum position (fully clockwise), the oscillator output frequency measured at tp34 and the DC input voltage measured at tp21 have values of: ....

....

The minimum and maximum values of the DC voltages and the corresponding values of output frequency are: ....

....

Record your results in Figure 91 below:

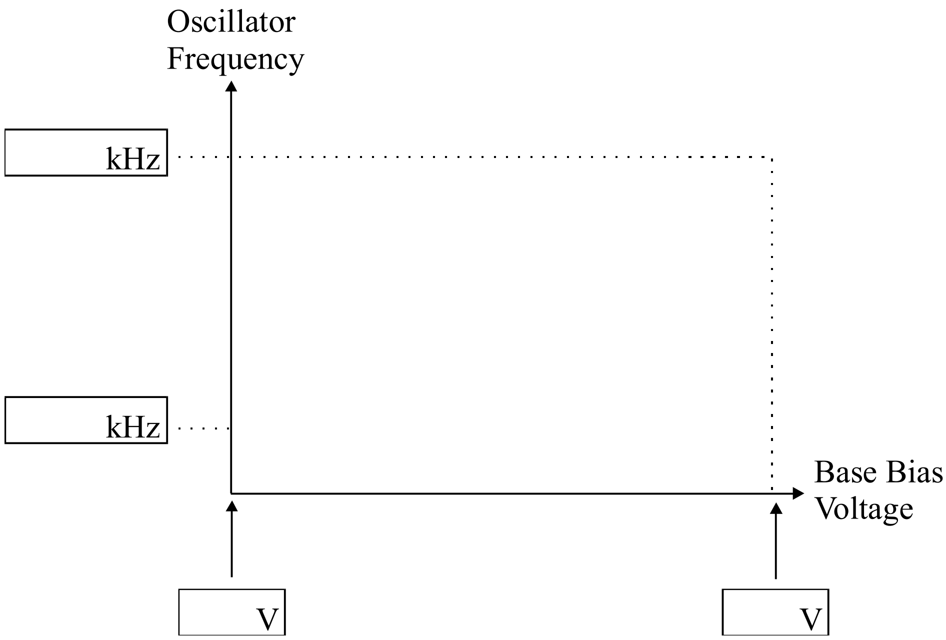


Figure 91 A Graph of DC Voltage/Frequency Output for a Varactor Modulator

7.16 Practical Exercise: The Reactance Modulator

The oscillator output frequency measured at tp34 and the DC input voltage measured at tp11 have values of: ....

....

With the Carrier Frequency preset in its maximum position (fully clockwise), the oscillator output frequency measured at tp34 and the DC input voltage measured at tp11 have values of: ....

....

The minimum and maximum values of the DC voltages and the corresponding values of output frequency are: ....

....

Record your results in Figure 95 below:

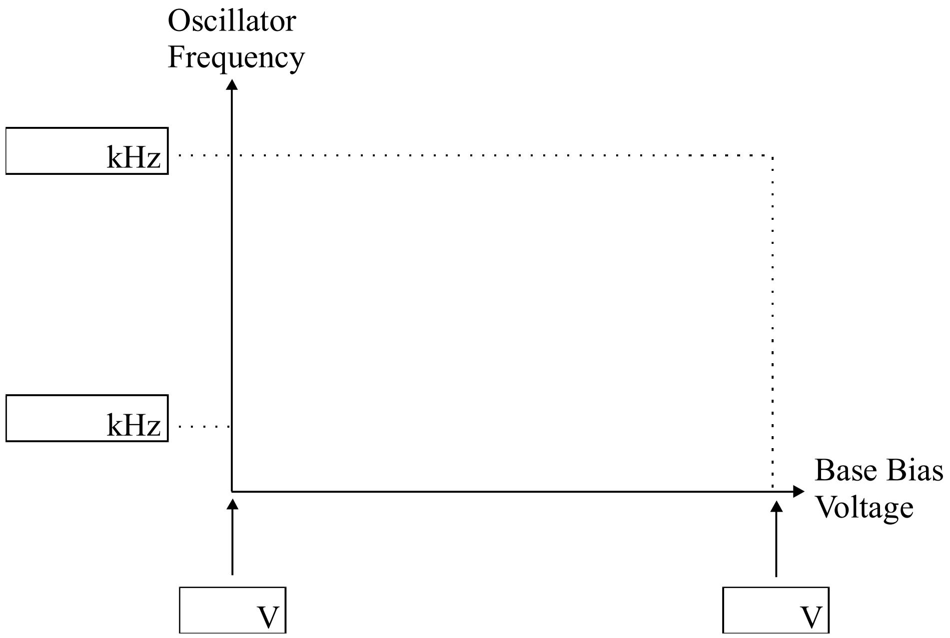


Figure 95 A Graph of DC Voltage/Frequency Output for a Reactance Modulator

Notes:

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

....

Chapter 8
FM Frequency Demodulators-1

8.1 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 an amplitude, modulated wave.

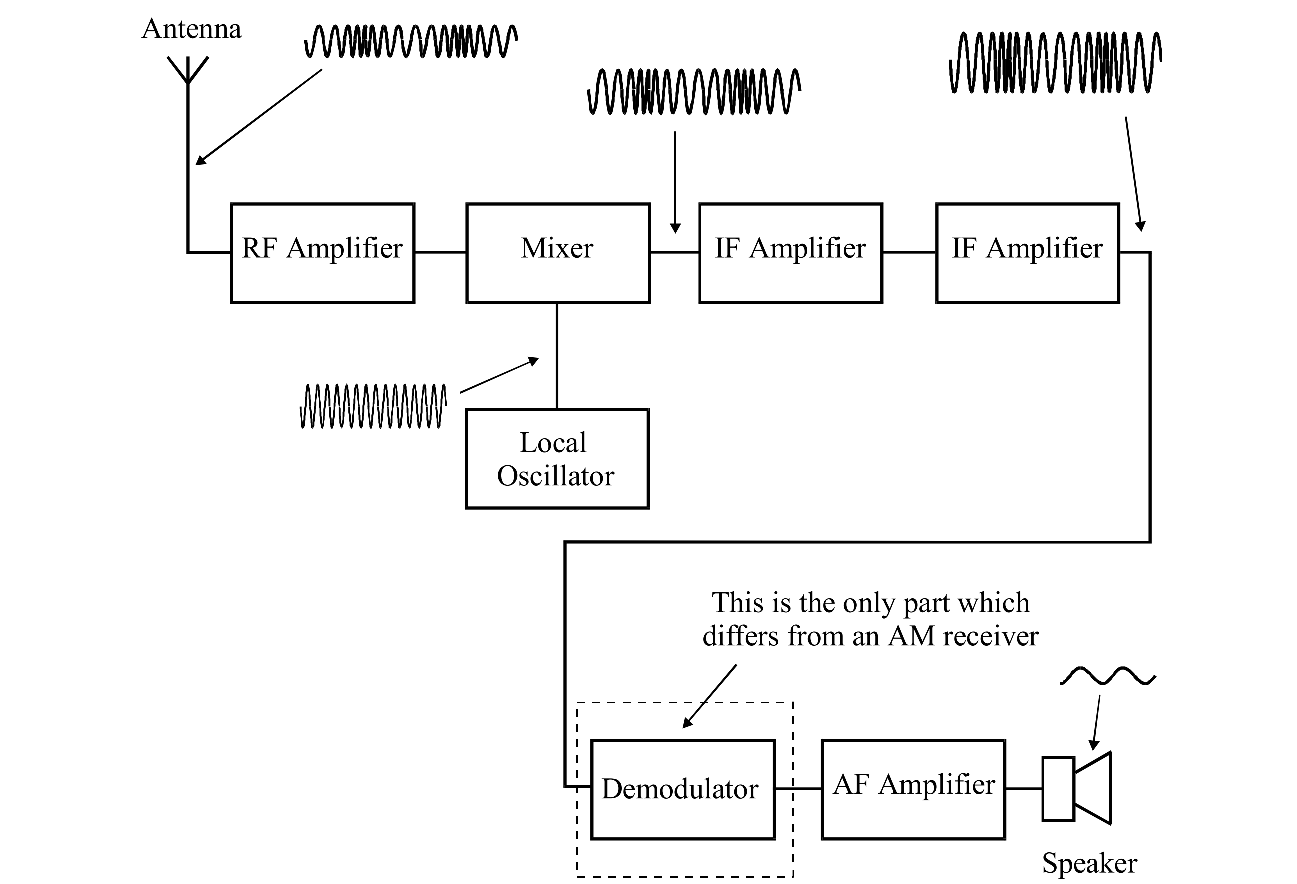


Figure 96 An FM Receiver

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 Figure 97 below:

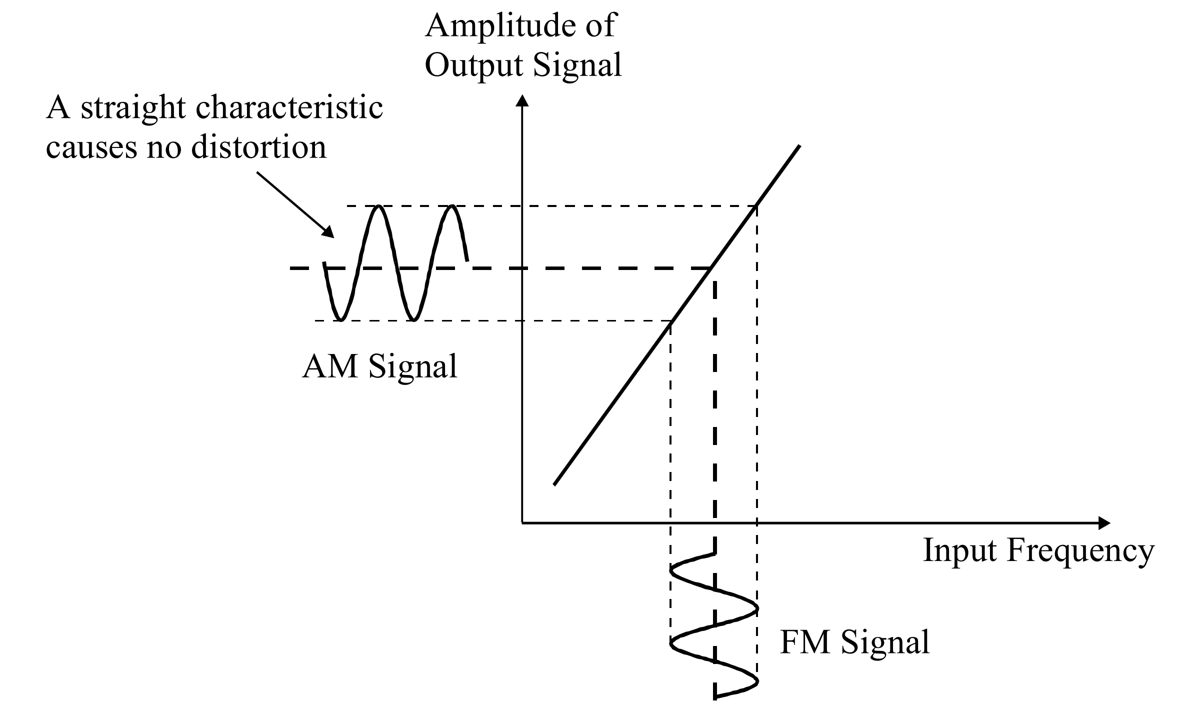


Figure 97 An 'Ideal' Linear Voltage/Frequency Characteristic

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 five 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?

8.2 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.

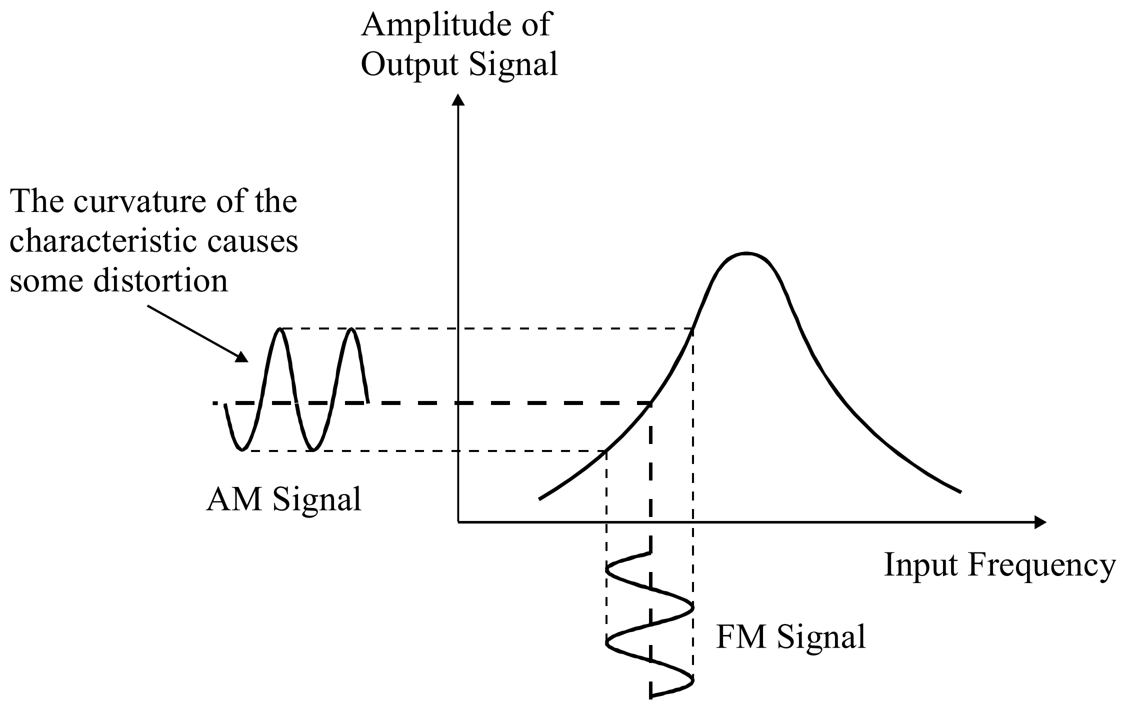


Figure 98

In Figure 98 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 99 below shows the circuit diagram of the Detuned Resonant Circuit Detector.

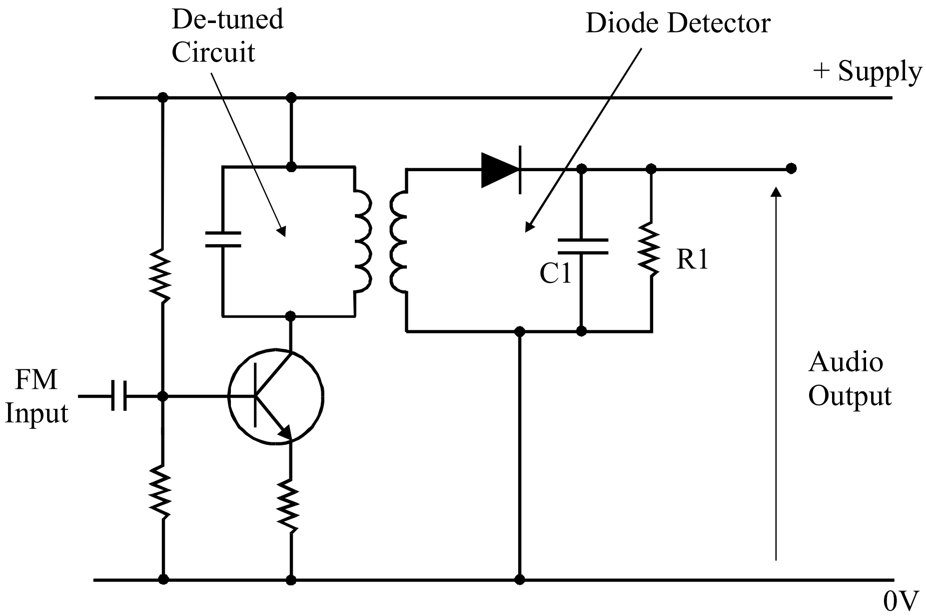


Figure 99 Detuned Resonant Circuit Detector

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 that is passed to the diode detector.

The diode detector was discussed in detail in Section 3.16 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.

8.3 The Amplitude Limiter

An amplitude limiter circuit is able to place an upper and lower limit on the size of a signal.

In Figure 100, the preset limits are shown by dotted lines. Any signal exceeding these levels is simply chopped off. This makes it very easy to remove any unwanted amplitude modulation due to noise or interference.

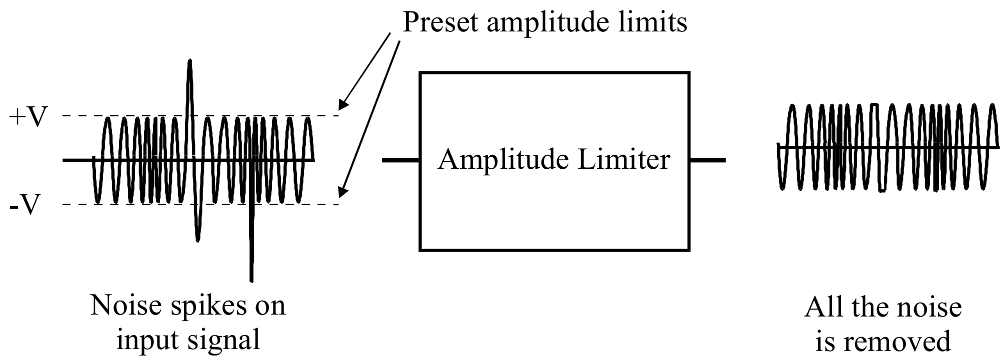


Figure 100 The Amplitude Limiter

8.4 Practical Exercise: The Detuned Resonant Circuit Detector

The peak-to-peak amplitude of the 'noise' output at tp73 is: ....

....

Practical Exercise Notes: ....

....

....

....

....

....

8.5 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:

(i) If the carrier is unmodulated, the phase shift is 90°.

(ii) If the carrier increases in frequency, the phase shift is GREATER THAN 90°.

(iii) 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 Figure 104 below:

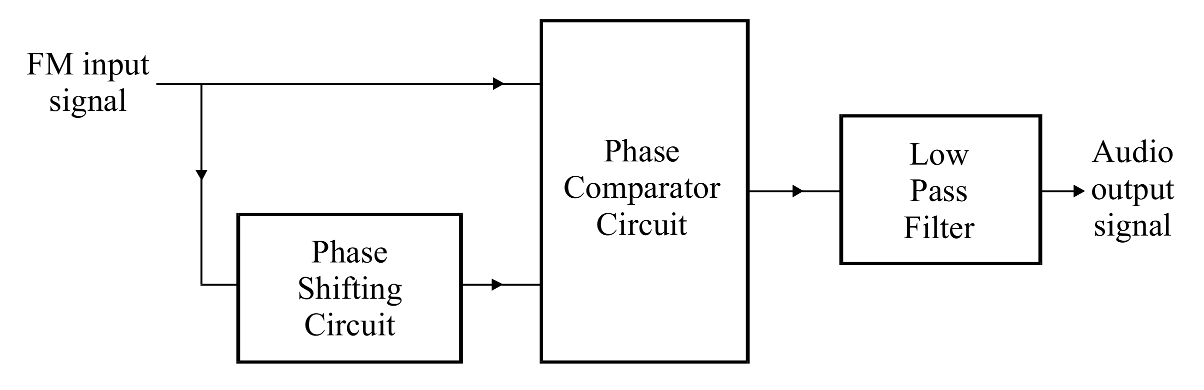


Figure 104

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 that depends on the result of the comparison according to the following rules:

(i) It provides no change in output voltage if the signal phase has been shifted by 90°.

(ii) Phases over 90° result in an increased DC voltage level.

(iii) Phases less than 90° result in a decreased DC voltage level.

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 105 is straight enough to cause very little distortion to the final audio output.

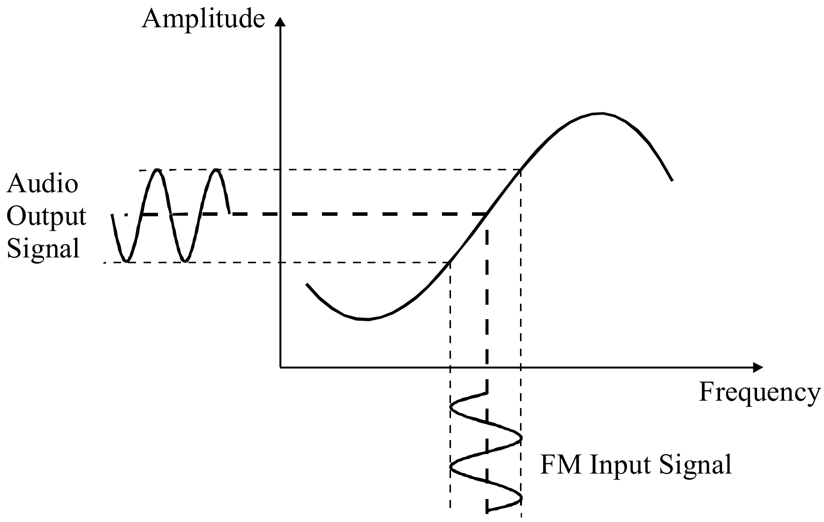


Figure 105 Quadrature Detector Characteristic

8.6 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. Altogether a very nice circuit.

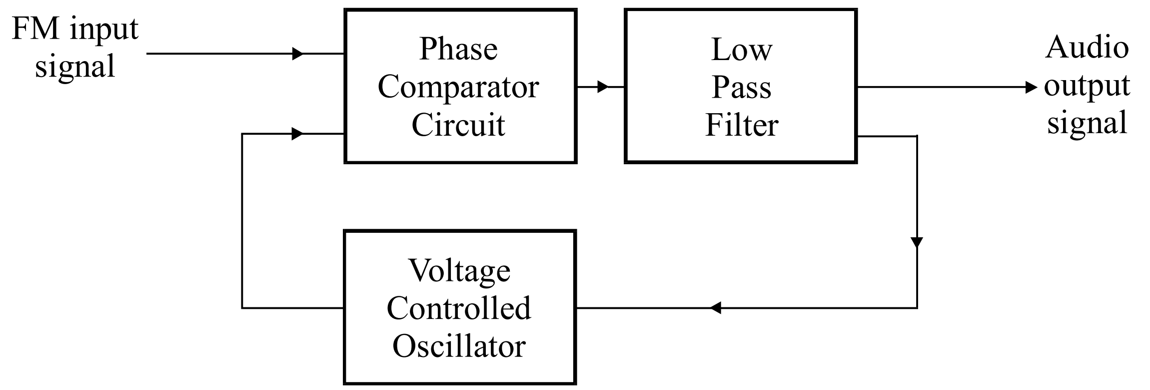


Figure 106 Block Diagram of the Phase-Locked Loop (PLL) Detector

The overall action of the circuit may, at first, seem rather pointless. As we can see in Figure 106 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.

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.

The frequency response is highly linear as shown in Figure 107 below:

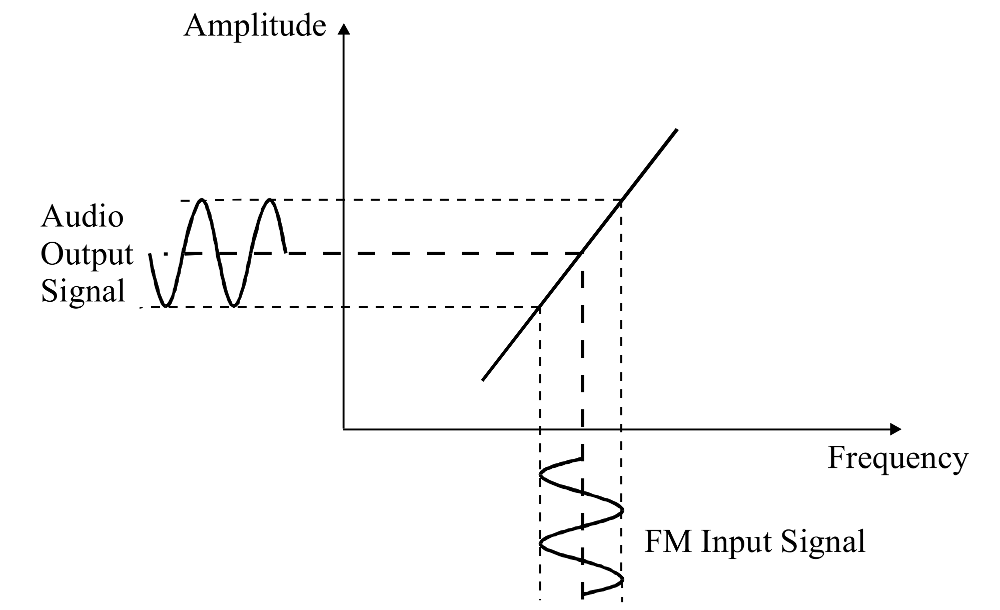


Figure 107 Phase-Locked Loop Detector Characteristic

8.7 Controlling the VCO

To see how the VCO is actually controlled, let us assume that it is running at the same frequency as an unmodulated input signal. The waveforms are given in Figure 108 below:

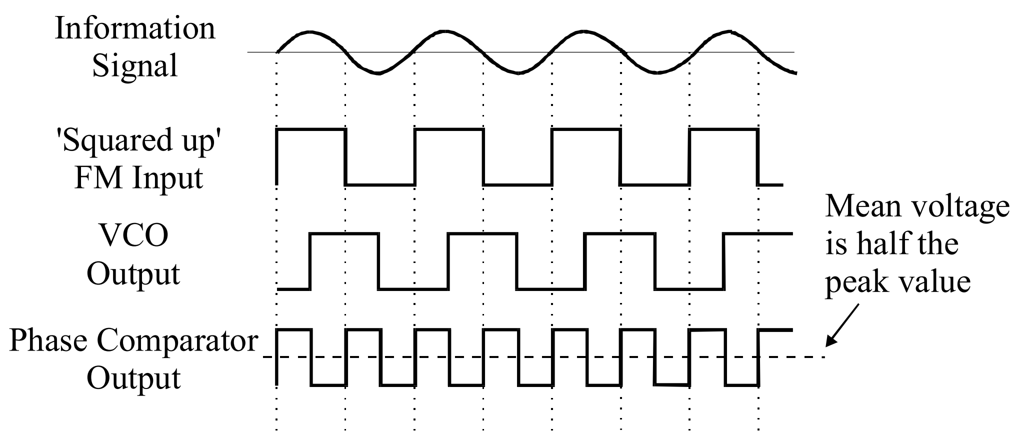


Figure 108

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 108 shows the situation when the FM input is at its unmodulated 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 (as shown).

Now let us assume that the FM signal at the input decreases in frequency (see Figure 109). 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.

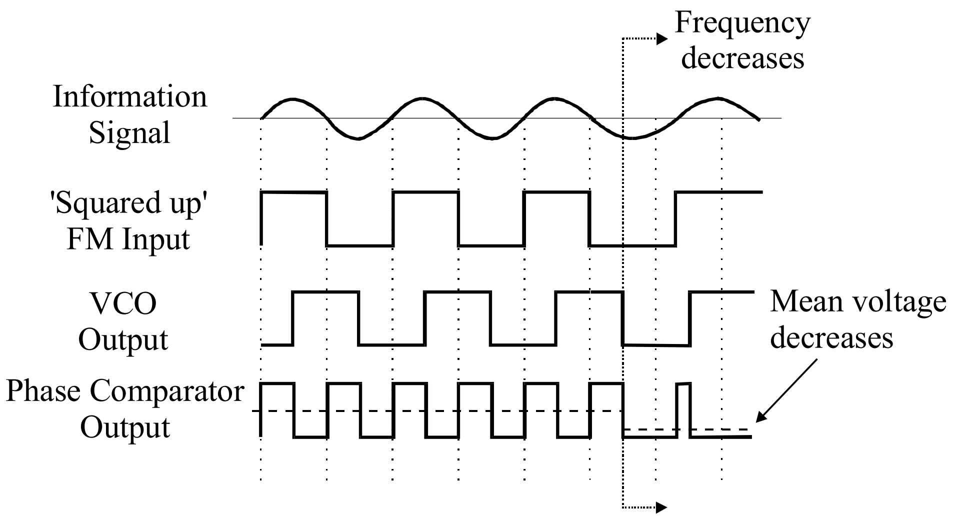


Figure 109

8.8 Practical Exercise: The Quadrature and Phase-Locked Loop Detectors

Record the DC voltage at tp40 against frequency on the grid provided in Figure 111 adding your own voltage scale.

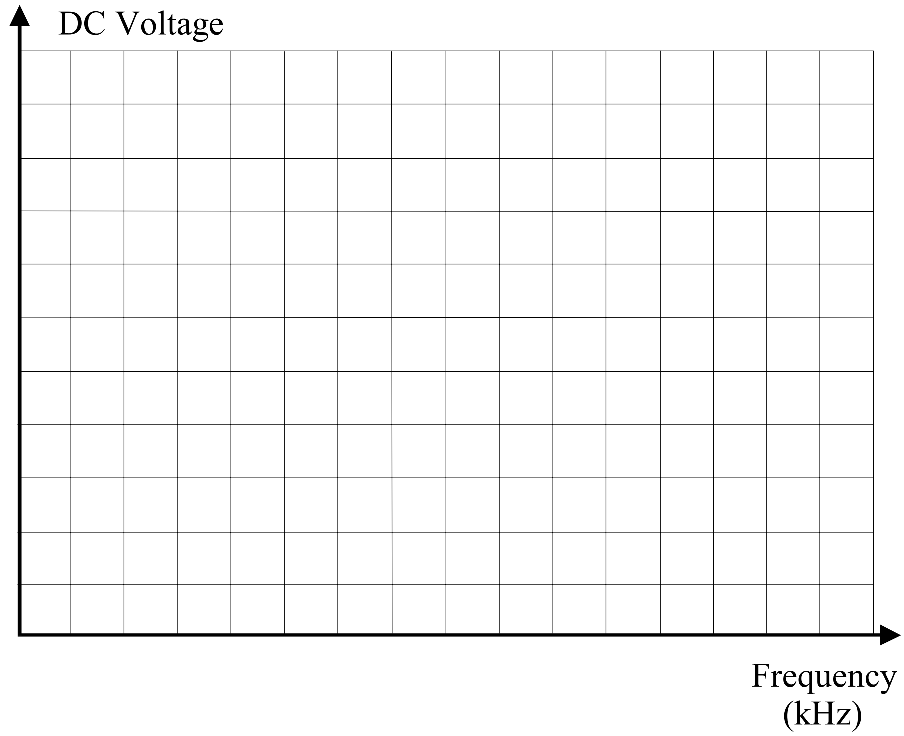


Figure 111 The Frequency Response of the Quadrature Detector

Peak-to-peak amplitude of the 'noise' output from the:

Quadrature Detector ....

Detuned Resonant Detector ....

Record the DC voltage at tp60 against frequency on the grid provided in Figure 114 adding your own voltage scale.

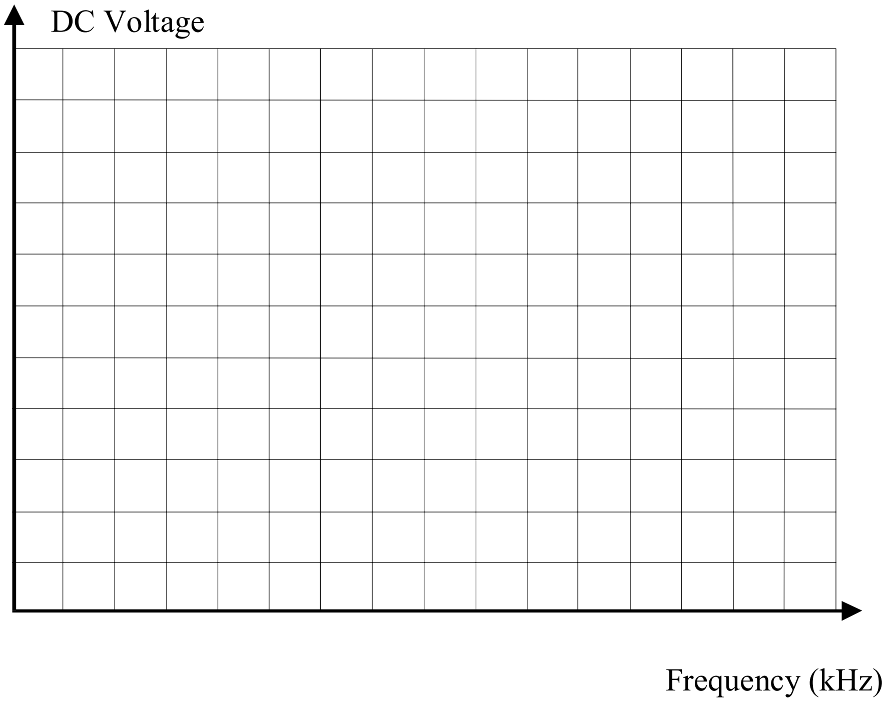


Figure 114 The Frequency Response of the PLL Detector

Notes:

....

....

....

....

....

....

....

Chapter 9
FM Frequency Demodulators-2

9.1 The Foster-Seeley Detector

The last two demodulators to be considered employ the phase shift that often accompanies a change in frequency in an AC circuit.

The Foster-Seeley circuit is shown in Figure 116. At first glance it looks rather complicated but it becomes simpler if we consider it a bit at a time.

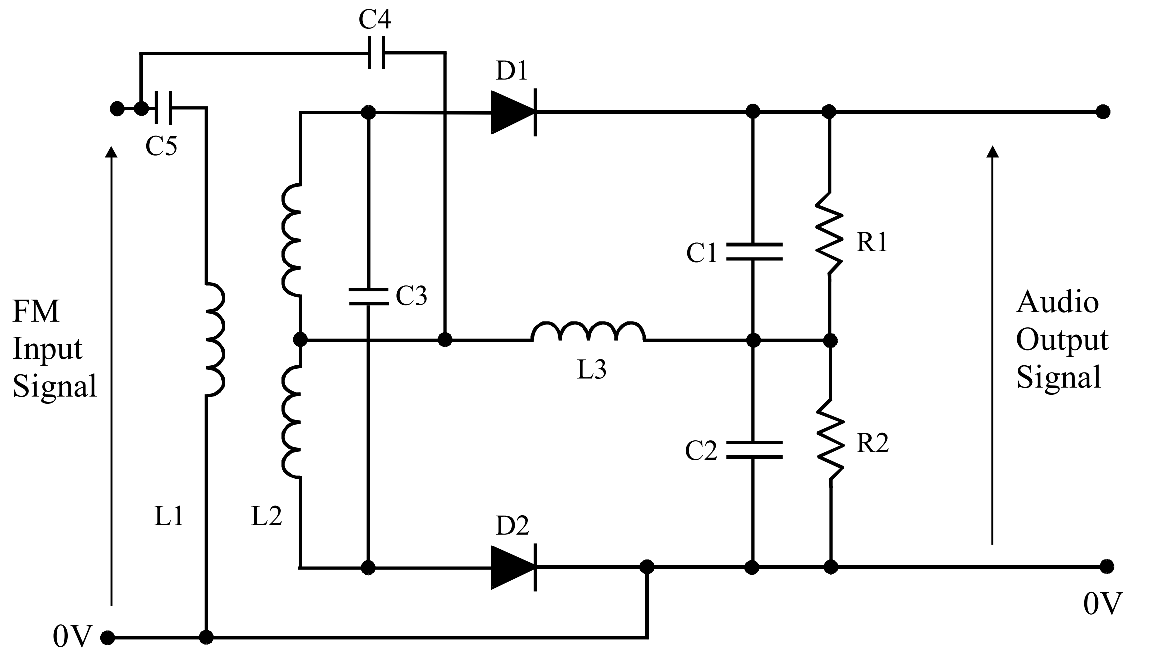


Figure 116 The Foster-Seeley Detector Circuit Diagram

9.2 When the Input Signal is Unmodulated

We will start by building up the circuit a little at a time. To do this, we can ignore many of the components.

Figure 117 below shows only the parts that are in use when the FM input signal is unmodulated.

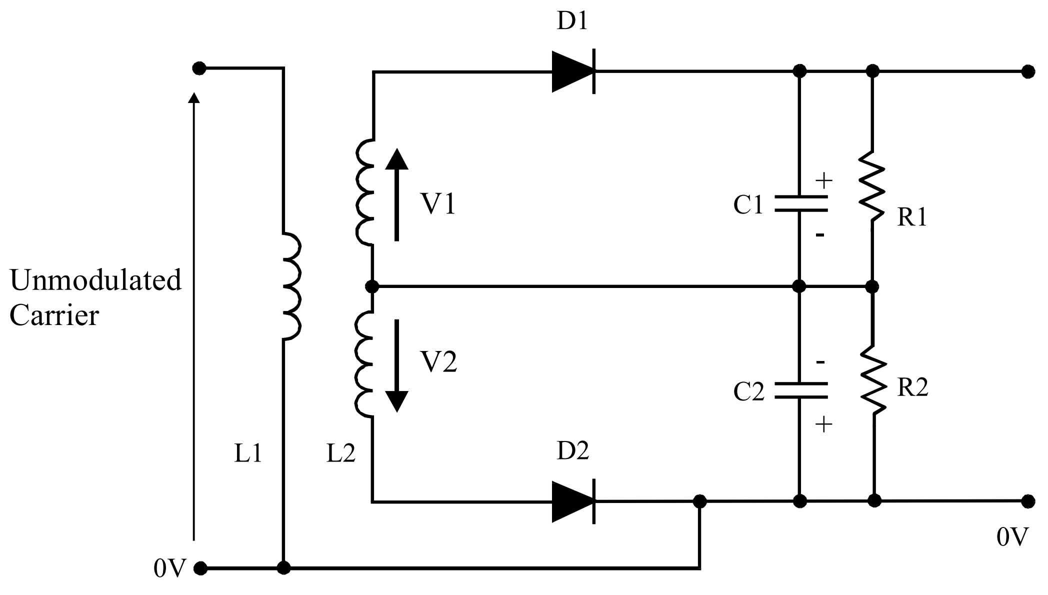


Figure 117

We may recognize immediately that it consists of two envelope detectors like halfwave rectifiers being fed from the center-tapped coil L2. With reference to the center-tap, the two voltages V1 and V2 are in antiphase as shown by the arrows. The output voltage would be zero volts since the capacitor voltages are in antiphase and are equal in magnitude.

9.3 Adding Two Capacitors

The next step is to add two capacitors and see their effect on the phase of the signals (see Figure 118).

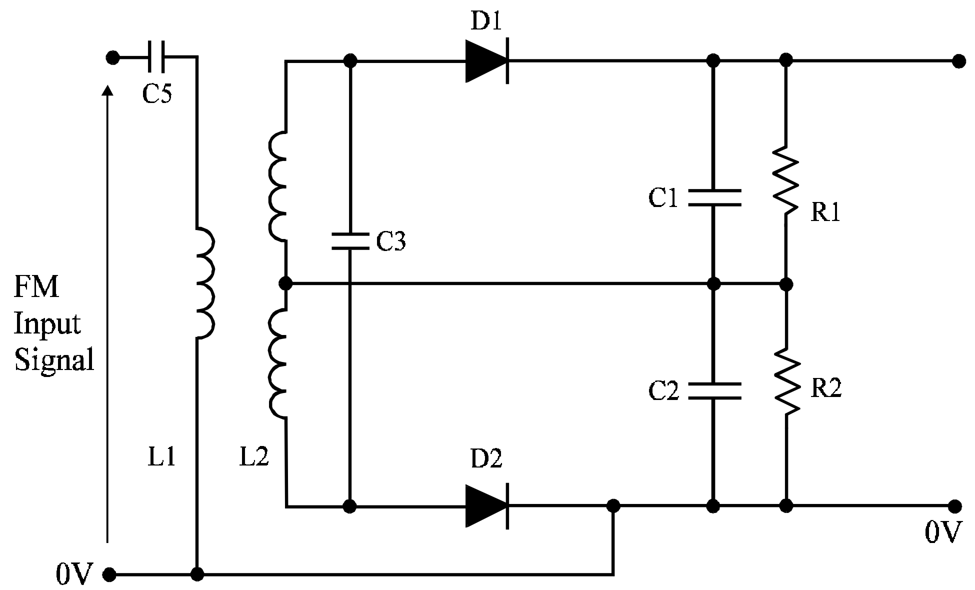


Figure 118

L1 and L2 are magnetically tightly coupled and by adding C3 across the center-tapped coil they will form a parallel tuned circuit with a resonance frequency equal to the unmodulated carrier frequency.

Capacitor C5 will shift the phase of the input signal by 90° with reference to the voltage across L1 (and L2). These voltages are shown as Va and Vb in the phasor diagram given in Figure 119.

Using the input signal Vfm as the reference, the phasor diagrams now look the way shown in Figure 119.

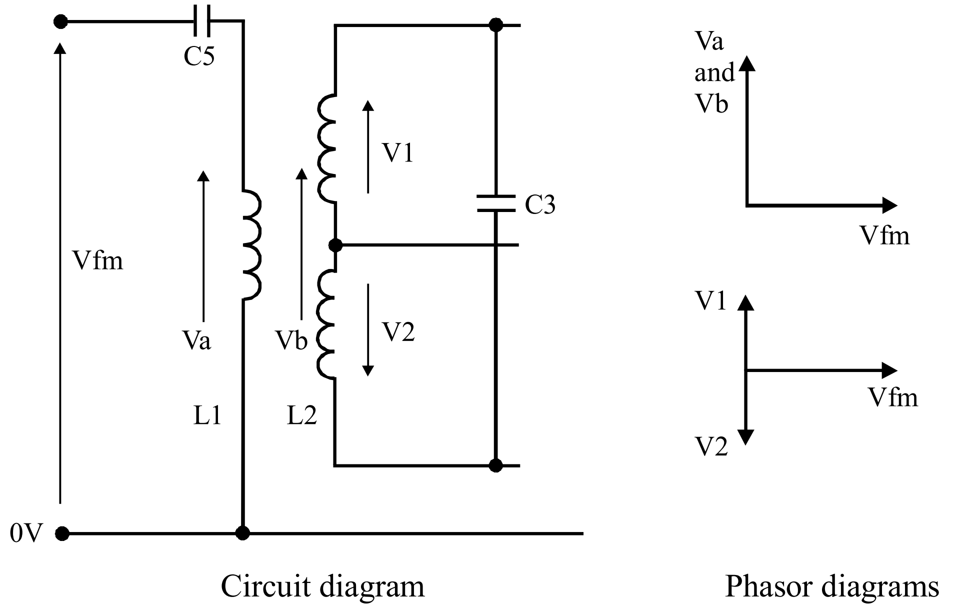


Figure 119

9.4 The Complete Circuit

By looking back at Figure 116, we can see that there are only two components to be added, C4 and L3.

C4 is not important. It is only a DC blocking capacitor and has negligible impedance at the frequencies being used. But what it does do is to supply a copy of the incoming signal across L3. All of the incoming signal is dropped across L3 because C1 and C2 also have negligible impedance.

If we return to the envelope detector section we now have two voltages being applied to each diode. One is V1 (or V2) and the other is the new voltage across L3, which is equal to Vfm.

This part of the diagram and the associated phasor diagram are shown in Figure 120 below.

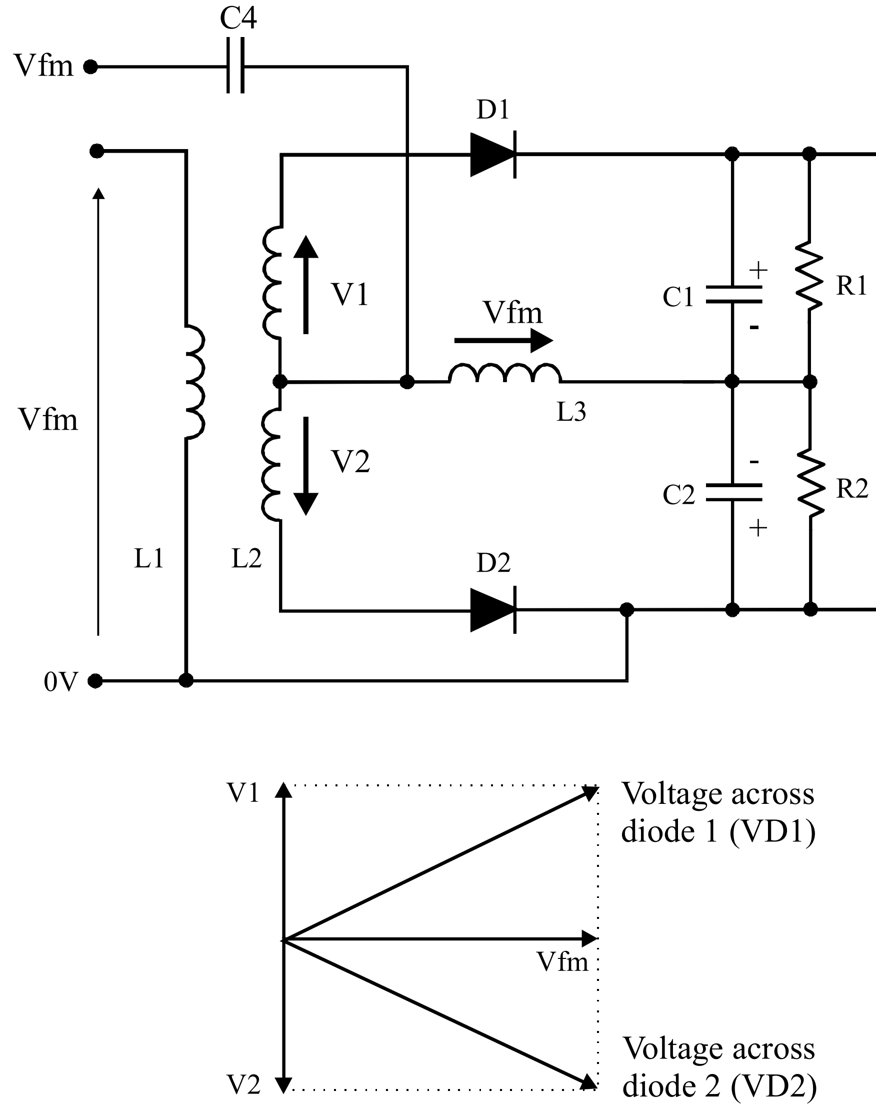


Figure 120

9.5 When the Input Frequency Changes

If the input frequency increased above its unmodulated value, the phase of Va would fall below 90° due to the parallel tuned circuit becoming increasingly capacitive.

The phasor representing V1 (and V2) would move clockwise as shown in Figure 121. This would result in a larger total voltage being applied across D1 and a reduced voltage across D2. Since the capacitor C1 would now charge to a higher voltage, the final output from the circuit would be a positive voltage.

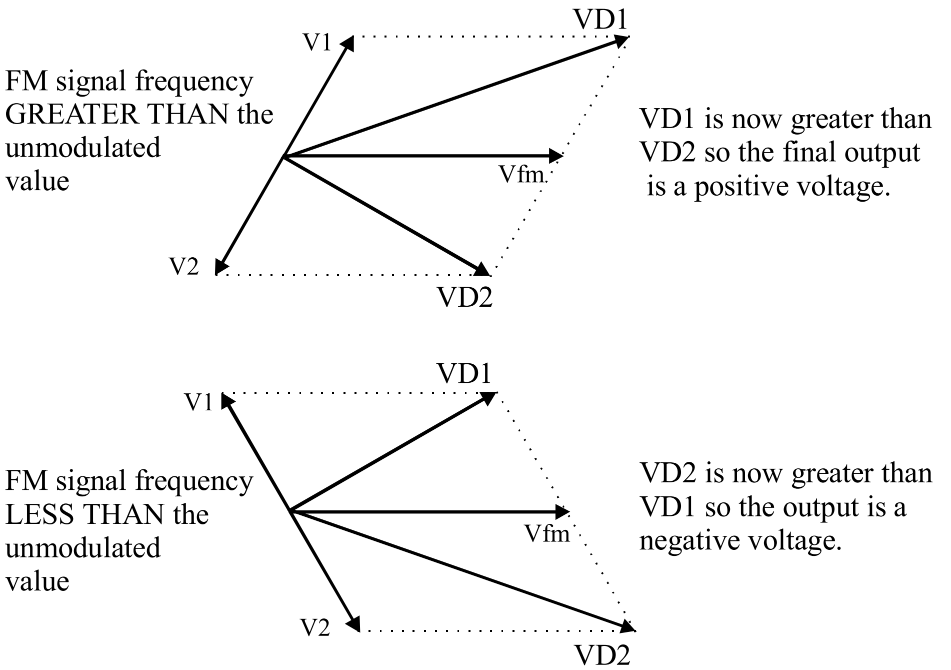


Figure 121

Conversely, if the frequency of the FM input signal decreased below the unmodulated value, the phase shift due to capacitor C5 increases above 90° as the parallel tuned circuit becomes slightly inductive. This causes the voltage across diode D2 to increase and the final output from the demodulator becomes negative.

The effect of noise is to change the amplitude of the incoming FM signal resulting in a proportional increase and decrease in the amplitudes of diode voltages VD1 and VD2 - and the difference between them. And since this difference in voltage is the demodulated output, the circuit is susceptible to noise interference and should be preceded by a noise limiter circuit.

9.6 The Ratio Detector

At first glance it appears to be the same as the Foster-Seeley Detector. There are a few modifications that have provided a much improved protection from noise. The circuit diagram is given in Figure 122 below.

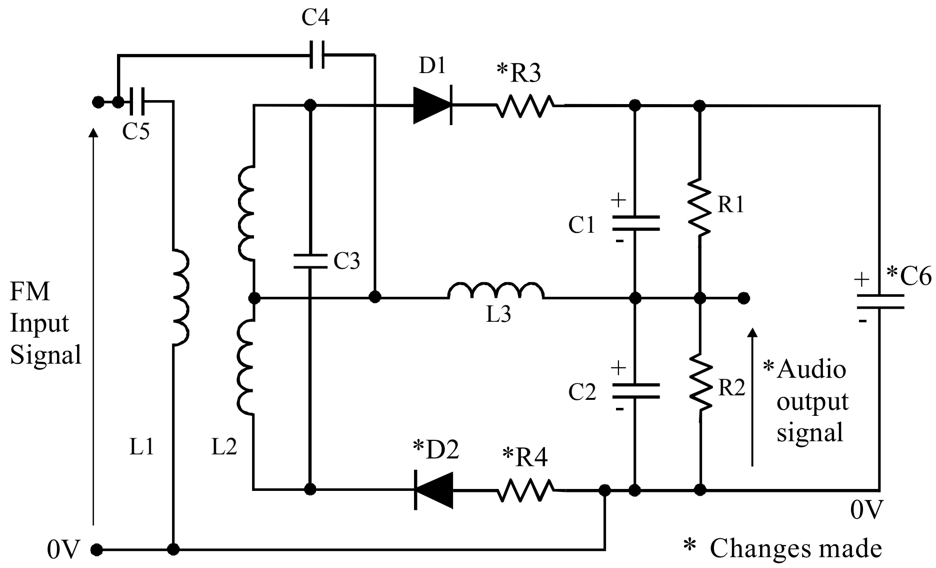


Figure 122 The Ratio Detector Circuit Diagram

Diode D2 has been reversed so that the polarity of the voltage across C2 will be as shown in the diagram. When the carrier is unmodulated, the voltages across C1 and C2 are equal and additive. The audio output is taken across C2 (or R2 of course).

Capacitor C6 is a large electrolytic capacitor. It charges to this voltage and, owing to the long time constant of C6, R1, and R2, the total voltage across it remains virtually constant at all times. In fact it just acts as a power supply or a battery. The important thing to note is that it keeps the total voltage of C1 + C2 at a constant value.

The generation of the voltage across the diodes D1 and D2 are by exactly the same process as we met in the Foster-Seeley Detector. Indeed even the changes in voltage occur in the same way and for the same reasons. If necessary, a quick read through Sections 9.1 to 9.5 may be helpful. For convenience, the resulting phasor diagrams are repeated here in Figure 123 overleaf.

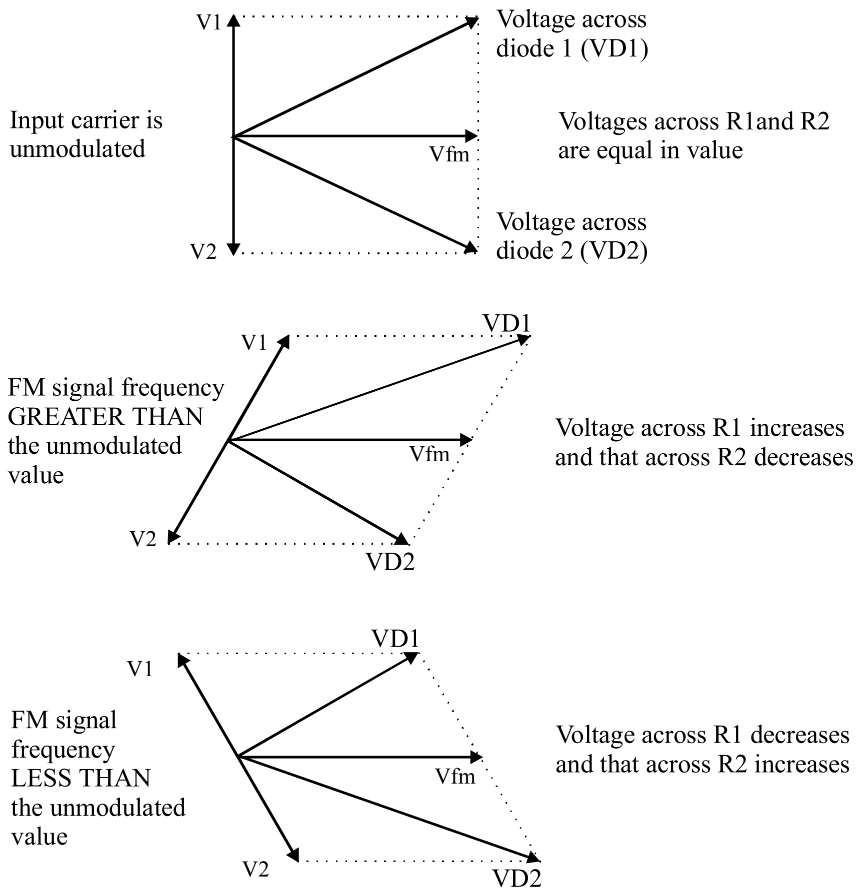


Figure 123

An unmodulated FM signal will result in equal voltages across R1 and R2. The voltage across R2 is the output from the circuit.

If the frequency of the FM signal increases, the voltage across R1 will increase and that across R2 will decrease.

Conversely, if the frequency of the FM signal decreases, the voltage across R1 will decrease and that across R2 will increase.

The final demodulated audio output voltage is taken across R2 and this voltage changes continuously to follow the frequency variations of the incoming FM signal.

Since the sum of the voltages across R1 and R2 remains constant, the ratio of the voltage across R2 to this total voltage, changes with the FM signal's frequency. It is this changing voltage ratio that gives the Ratio Detector its name.

9.7 Reducing the Effect of Electrical Noise

This is the real purpose of C6.

If the amplitude of the FM input signal suddenly increases, the voltages VD1 and VD2 will try to increase and these in turn will try to increase the voltages across both R1 and R2. However, since C6 is large, the overall voltage across R1 and R2 will not respond to the fast change in input amplitude. The result is that the demodulated audio output is unaffected by fast changes in the amplitude of the incoming FM signal.

R3 and R4 are current limiting resistors to prevent momentary high levels of current through the diodes, which would cause a brief fluctuation in the output voltage.

9.8 Practical Exercise: The Foster-Seeley and Ratio Detectors

Record the DC voltage at tp52 against frequency on the grid provided in Figure 125 adding your own voltage scale.

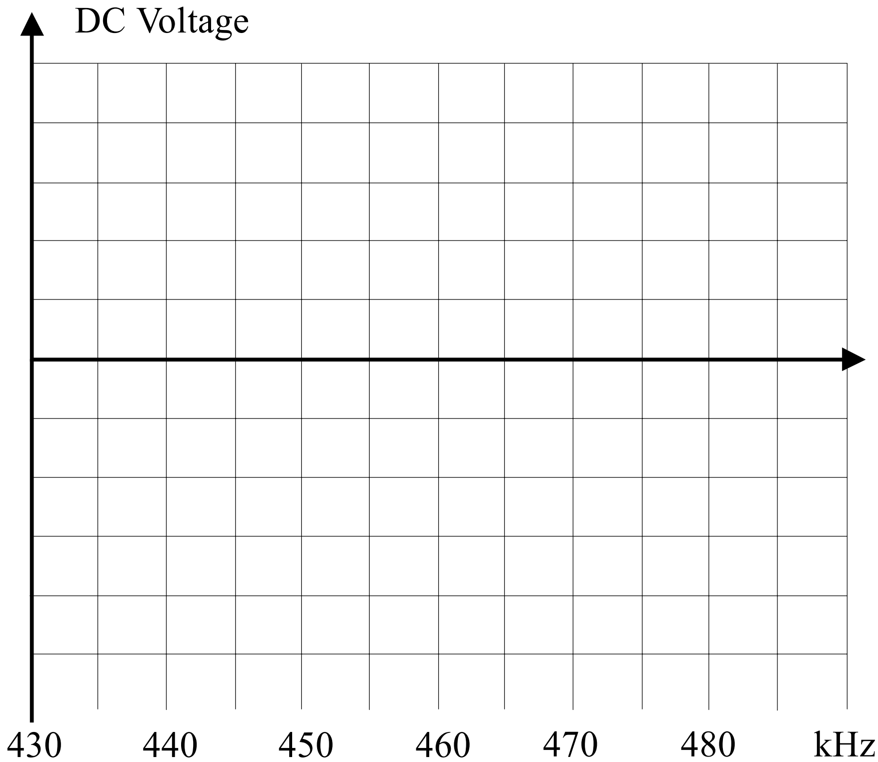


Figure 125 The Frequency Response of the Foster-Seeley Detector

Peak-to-peak amplitude of the 'noise' output from the:

Foster-Seeley Detector ....

Detuned Resonant Circuit Detector ....

Quadrature Detector ....

Record the DC voltage at tp53 against frequency on the grid provided in Figure 129 adding your own voltage scale.

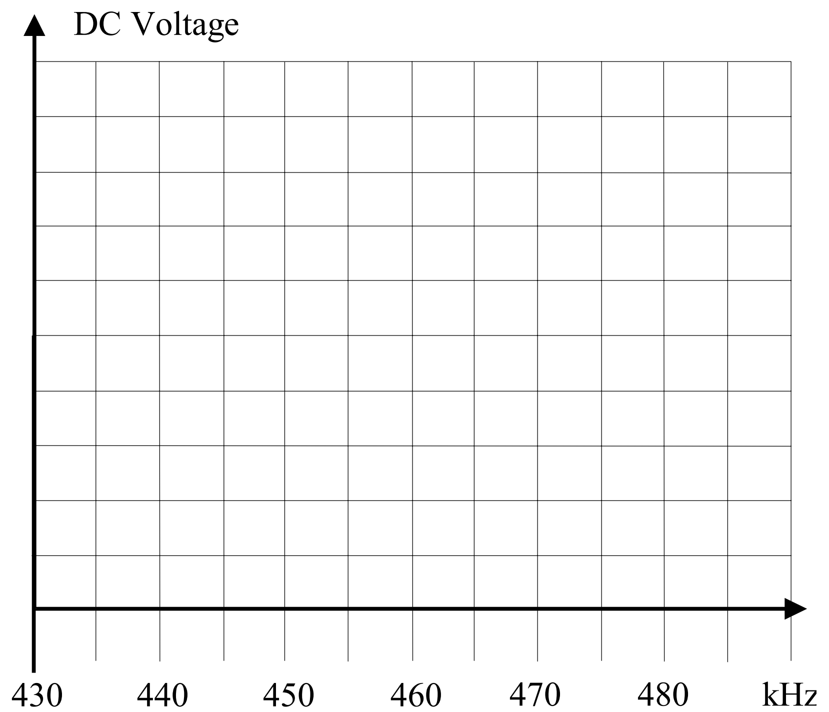


Figure 129 The Frequency Response of the Ratio Detector

Peak-to-peak amplitude of the 'noise' output from the:

Ratio Detector ....

Foster-Seeley Detector ....

Detuned Resonant Circuit Detector ....

Quadrature Detector ....

Chapter 10
Locating Faults in FM Transmission Systems

10.1 General Approaches to Fault Finding

There are two methods that may be employed on ANACOM 2 and they are summarized here as a reminder. If you feel at all unsure, it may be worth a few moments to read through the first part of Chapter 5 again.

10.2 The Half-Split Method

In this method, the suspect area is divided into two approximately equal sections by checking at the mid-point. The process is repeated until we finally home in on the faulty circuit.

10.3 The Plod-Through Method

This method locates faults by starting at one end and checking the waveform at each stage all through the circuit until the fault is found. This works, but is generally a slow method. But any method is better than just taking random measurements.

We can work from the output back towards the input until the signal is located, or alternatively we could start from the beginning and work towards the end. The route taken is a matter of personal preference. As with the half-split method, it still relies on us understanding the system well enough to be know what to expect at each point.

10.4 Making Tests

Work slowly and methodically. Make notes as you go along so that the tests do not have to be repeated.

10.6 Practical Exercise: Fault Finding on an FM System

Fault 1 - Tests Made and Results:

....

....

....

....

....

....

Fault 2 - Tests Made and Results:

....

....

....

....

....

....

Fault 3 - Tests Made and Results:

....

....

....

....

....

....

Fault 4 - Tests Made and Results:

....

....

....

....

....

....

Fault 5 - Tests Made and Results:

....

....

....

....

....

....

Fault 6 - Tests Made and Results:

....

....

....

....

....

....

Fault 7 - Tests Made and Results:

....

....

....

....

....

....

Fault 8 - Tests Made and Results:

....

....

....

....

....

....

Notes:

....

....

....

....

....

....

....

....