MODERN COMMUNICATION

Courses COM-1 & COM-2
AM and FM COMMUNICATION CIRCUITS

THEORETICAL BACKGROUND MANUAL
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chapter 1

INFORMATION TRANSMISSION
& MODULATION

The problem of transmitting information from one place to another has always interested man and many different forms of transmission have been used over the centuries, i.e. drums, smoke signals, semaphore etc. The main limitations of all these methods is their limited range. During the 18th century, better and more far-reaching methods were being sought, mainly because the needs of that time began to demand faster and more reliable transfer of information. If we recall that to send a message from America to Europe took at least a month we can understand the motivation for developing faster means of communication. The breakthroughs started with the use of electricity. Firstly the telegraph, which enabled messages to be sent across continents instantly, and then the telephone. But still the problem of inter-continental communication existed. Then Marconi succeeded in sending a message through air without using wires and the wireless was born.

1.1 INFORMATION AND ITS TRANSMISSION

How can information be transmitted through space? When electricity flows in a wire it forms electrical and magnetic fields that vary as the current in the wire varies. These fields (jointly called the electromagnetic field) propagate through space at the speed of light. If we place a wire in the electromagnetic field, the field will cause a current to flow in the wire, which will vary in almost exactly the same way as the original current which created the electromagnetic field. In this way the presence of a current at one place causes a current at another place almost simultaneously (at least by human standards).

Let us now leave the subject of electromagnetic transmission and take a look at information from the electrical point of view.

If a pure sine wave is sent from A to B the only information an observer at B can obtain is that a sine wave was sent. This is because information sent as an electrical or electro-magnetic wave is contained in one or more parameters of the wave (amplitude or phase, for example) at a specific time. Since a pure sine wave is periodic it is impossible to recognize a particular point as having been sent at a particular moment in time, and therefore information cannot be sent by transmitting a pure sine wave.

So we must in some way change the pure sine wave so as to be able to send information. One way is to send different length bursts of the sine wave and to define a code - for instance the Morse Code. In the Morse code either dots (·) or dashes (−) are sent. A dot is a short burst of sine wave, while a dash is three times as long as a dot (see figure 1.1).

This is what is done in the telegraph. Another possibility is to send various sine waves with different amplitudes - this is what we hear as speech or music or sound.
Let us now look at electromagnetic waves. The velocity of electromagnetic waves in space is 300,000 km/sec and therefore the wavelength of a 1 kHz wave is:

\[
\lambda = \frac{300,000}{1,000} = 300 \text{ km}
\]

A wavelength of 300 km means that a wire which will receive or transmit the wave efficiently must be hundreds of kilometers long! Obviously this is impractical. Let us look at the problem from the other end, in other words what wavelength would be practical? The maximum wavelength should not be more than a few hundred meters, so for \(\lambda = 300 \text{ m} = 0.3 \text{ km}\) we obtain:

\[
f = \frac{v}{\lambda} = \frac{300,000}{0.3} = 1,000,000 \text{ Hz} = 1 \text{ MHz}
\]

So around 1 MHz is a practical frequency for electromagnetic wave transmission. Actually the frequency range of radio communication is roughly from a few hundred kilohertz to hundreds of megahertz and up.

1.2 AMPLITUDE MODULATION

The information we want to transmit in the case of speech or music has a frequency of at most 20 kHz which is a long way from 1 MHz. Also a pure sine wave at 1 MHz can carry no information, as we have already seen. So we must somehow "change" or modulate the 1 MHz wave so that it does carry information.
As has been already stated, there are three parameters that determine the information carried by a wave - its amplitude, its phase and its frequency. If we vary the amplitude of the high frequency wave in the same way that the amplitude of the low frequency information varies, we will obtain a wave at a frequency of 1 MHz carrying information in its amplitude. This is called Amplitude Modulation. Figure 1.3 shows the information wave, which is called the SIGNAL wave; the high frequency wave, which is called the CARRIER wave; and the MODULATED wave, which is called the AMPLITUDE MODULATED (AM) wave. It can be seen that the modulated wave is in effect the signal wave "riding" on the carrier wave.

From figure 1.3 it can be seen that the AM wave contains the information in its envelope or shape, which is carried by the carrier wave (hence the name) at a high frequency.

\begin{figure}[h]
\centering
\includegraphics[width=\textwidth]{amplitude_modulation}
\caption{SIGNAL, CARRIER and AM Waves}
\end{figure}
Therefore the AM wave enables information at low frequencies to be transmitted at much higher frequencies, as was required.

We shall now develop the mathematical expressions that represent AM signals.

If the modulating signal is a sine wave of frequency $f_m$ and the carrier wave is of frequency $f_c$ the expression representing the modulated wave is:

$$ v_{AM}(t) = (V_c + V_m \cos \omega_m t) \cos \omega_c t = A(t) \cos \omega_c t \quad (1-3) $$

where:

- $V_c$ - amplitude of carrier wave
- $V_m$ - amplitude of signal wave
- $\omega_c = 2\pi \cdot f_c$
- $\omega_m = 2\pi \cdot f_m$

Equation (1-3) describes a sine wave whose frequency $f_c$ is that of the carrier wave and whose amplitude $A(t)$ is determined by the signal wave.

By extracting $V_c$ from the brackets we obtain:

$$ V_{AM}(t) = V_c (1 + \frac{V_m}{V_c} \cos \omega_m t) \cos \omega_c t = \frac{V_c (1 + m \cos \omega_m t) \cos \omega_c t}{A(t)} \quad (1-4) $$

where:

- $m = \frac{V_m}{V_c}$ and is called the Modulation Index

The modulation index shows the relationship between the original carrier wave amplitude and the signal wave amplitude. Figure 1.4 shows AM waves with the same carrier and signal frequencies but with different modulation indices.

As can be seen the waveforms are true AM only for values of $m$ between 0 and 1. For $m > 1$ the envelope of the wave no longer resembles the signal waveform.

When $m$ is given as a percentage it is called the Modulation Percentage. For example an AM wave with $m = 0.3$ is 30% modulated.

1.3 FREQUENCY & POWER SPECTRUM OF AN AM WAVE

We shall now obtain the frequency spectrum of an AM wave.

By expanding equation (1-4) we obtain the following expressions:
FIGURE 1.4: AM Waveforms for Various Modulation Indices

\[ v_{AM}(t) = V_c \cos \omega_c t + mV_c \cos \omega_c t \cos \omega_m t = \]

\[ = V_c \cos \omega_c t + mV_c \left[ \frac{1}{2} \cos (\omega_c + \omega_m) t + \frac{1}{2} \cos (\omega_c - \omega_m) t \right] = \]

\[ = \frac{mV_c}{2} \cos (\omega_c - \omega_m) t + V_c \cos (\omega_c + \omega_m) t \]

(1-4)
Equation (1-5) shows that the AM wave is composed of three frequency components at frequencies $\omega_c - \omega_m$, $\omega_c$ and $\omega_c + \omega_m$. These are called the LOWER SIDE BAND, CARRIER and UPPER SIDE BAND respectively (see figure 1.5). Recalling that $mV_c = V_m$, we see that each of the sidebands contains the original information at a shifted frequency ($\omega_c \pm \omega_m$ instead of the original $\omega_m$), which represent the sum and difference of the carrier and signal frequencies.

Therefore amplitude modulation is in fact a process of shifting the information signal to two different frequencies equally spaced around the carrier frequency. Using the Fourier Transform it can be shown that this holds for any information signal, not only pure sine waves. Figure 1.6 shows the frequency spectrum of the information signal and the spectrum of the resulting AM wave.

Two important properties of AM can be seen in figure 1.6:

1. There is symmetry about the carrier frequency.
2. The bandwidth required to transmit an AM signal is exactly twice the bandwidth of the information signal.

The second property is very significant when comparing AM with other modulation techniques, as transmission bandwidth is always a characteristic parameter.
Another parameter is the average transmitted power, defined as:

\[ P = \frac{1}{T} \int_0^T [v_{AM}(t)]^2 dt \]  \hspace{1cm} (1-6)

where \( T \) is one or more periods (multiples of \( 2\pi \) radians).

The average power of a sine wave is:

\[ P = \frac{1}{2\pi} \int_0^{2\pi} \cos^2 \omega t \, dt = \frac{1}{2\pi} \int_0^{2\pi} \frac{1}{2} (1 + \cos 2\omega t) \, dt = \frac{1}{2} \]  \hspace{1cm} (1-7)

By substituting equation (1-4) into equation (1-6) and by using equation (1-7):

\[ P_{AM} = \frac{1}{2\pi} \int_0^{2\pi} V_c^2 (1 + m \cos \omega_m t)^2 \cos^2 \omega_c t \, dt = \]

\[ = \frac{V_c^2}{2\pi} \int_0^{2\pi} (1 + 2m \cos \omega_m t + m^2 \cos^2 \omega_m t) \cos^2 \omega_c t \, dt = \]

\[ = \frac{V_c^2}{2\pi} \int_0^{2\pi} [\cos^2 \omega_c t + 2m \cos \omega_m t \cos \omega_c t + m^2 \cos^2 \omega_m t \cos^2 \omega_c t] \, dt = \]

\[ = V_c^2 \left[ \frac{1}{2} + 0 + \frac{m^2}{4} \right] = \frac{V_c^2}{2} + \frac{m^2 V_c^2}{4} \]  \hspace{1cm} (1-8)

You can prove yourself that \( \frac{1}{2\pi} \int_0^{2\pi} \cos \omega_m t \cos^2 \omega_c t \, dt = 0 \) and that \( \frac{1}{2\pi} \int_0^{2\pi} \cos^2 \omega_m t \cos^2 \omega_c t \, dt = \frac{1}{4} \) by using trigonometric equalities and equation (1-7). The first expression derived in equation (1-8) is the Average Carrier Power which we shall call \( P_C \). The second expression is the Average Sideband Power which we shall call \( P_m \). Therefore:

\[ P_C = \frac{V_c^2}{2} ; \quad P_m = m^2 \frac{V_c^2}{4} = \frac{m^2 P_c}{2} \]

Hence:

\[ \frac{P_m}{P_C} = \frac{m^2}{2} \]  \hspace{1cm} (1-9)
Equation (1-9) shows the second characteristic of AM: the average side band power (which is the information power) is much less than the carrier power. For 30% modulation: 
\[
\frac{P_m}{P_c} = \frac{0.32}{2} = \frac{0.09}{2} = 0.045 = 4.5%. 
\]
What is more, the sideband power is divided equally between two sidebands.

If the total transmitted power is 1 W, then \( P_m + P_c = 1 \) W and substituting \( P_c = \frac{P_m}{0.045} \) gives: 
\[
P_m = \frac{1}{1 + \frac{1}{0.045}} = 0.043 \text{ W} = 43 \text{ mW}.
\]

Therefore, the carrier wave, which contains no information, is transmitted with an average power of 957 mW, while the information in each sideband, is transmitted with an average power of 21.5 mW!

This is a major disadvantage of AM, but it is counter-balanced by other considerations, mainly simplicity of electronic realization.

1.4 PHASE & FREQUENCY MODULATION

We have seen how information can be superimposed on a carrier wave by amplitude modulation. But, as was previously mentioned, a wave has other parameters apart from its amplitude - phase and frequency. We shall now discuss methods of modulation in which the information is contained in the phase or in the frequency of the carrier wave.

The basic equation for a phase or frequency modulated wave is:

\[
v(t) = V_c \cos(\omega_c t + \phi(t)) \tag{1-10}
\]

where:

\( V_c \) - carrier amplitude

\( \omega_c \) - carrier frequency

\( \phi(t) \) - time-varying phase (determined by the modulating signal)

In the case of phase modulation (PM) the phase \( \phi(t) \) is determined by the information signal:

\[
\phi(t) = K_p \cdot V_m \cos \omega_m t \tag{1-11}
\]

where:

\( V_m \cos \omega_m t \) - information signal

\( K_p \) - constant

The maximum value of \( \phi(t) \) is called the phase deviation and is represented by \( \beta = K_p \cdot V_m \) which is called the Modulation Index.
Therefore, in the case of PM, equation (1-10) can be written as:

\[ v_{PM}(t) = V_C \cos(\omega_C t + \beta \cos \omega_m t) \]  \hspace{1cm} (1-12)

It can be seen that the phase of \( v_{PM}(t) \) varies with the information, with the instantaneous phase deviation depending on both the amplitude and the frequency of the information wave.

Frequency is the derivative of phase: \( f = \frac{d\phi}{dt} \) and in frequency modulation (FM) it is the derivative of the phase that varies with the information:

\[ \frac{d\phi}{dt} = K_F V_m \cos \omega_m t \]  \hspace{1cm} (1-13)

where \( K_F \) is a constant.

By integrating both sides of equation (1-13) we obtain the phase:

\[ \phi(t) = \int_{-\infty}^{t} K_F V_m \cos \omega_m \lambda d\lambda \]  \hspace{1cm} (1-14)

where \( \lambda \) is the variable of integration.

By substituting equation (1-14) into equation (1-10) we obtain the FM wave:

\[ v_{PM}(t) = V_C \cos(\omega_C t + K_F V_m \int_{-\infty}^{t} \cos \omega_m \lambda d\lambda) \]  \hspace{1cm} (1-15)

To obtain the instantaneous frequency of \( v_{PM} \) we must take the derivative of the instantaneous phase [which is the argument of the cosine in equation (1-15)].

\[ \omega_{\text{inst}} = \frac{d}{dt} [\omega_C t + K_F V_m \int_{-\infty}^{t} \cos \omega_m \lambda d\lambda] = \omega_C + K_F V_m \cos \omega_m t \]  \hspace{1cm} (1-16)

It can be seen that the instantaneous frequency is composed of the carrier frequency, plus a frequency that varies with the information and is called the frequency deviation.

The maximum frequency deviation is: \( \Delta \omega = K_F V_m \). Therefore the instantaneous frequency of an FM wave varies around the carrier frequency at a rate determined by the information signal.

If we now evaluate the integral in equation (1-15), disregarding the lower limit of integration, and substituting \( \Delta \omega \), we obtain:

\[ v_{PM}(t) = V_C \cos(\omega_C t + \frac{\Delta \omega}{\omega_m} \sin \omega_m t) \]  \hspace{1cm} (1-17)
\( \frac{\Delta \omega}{\omega_m} \) is the ratio between the maximum frequency deviation and the modulating frequency and is defined as the Modulation Index (\( \beta \)).

Therefore equation (1-17) becomes:

\[
V_{FM}(t) = V_C \cos(\omega_{ct} + \beta \sin \omega_{mt})
\]  \(\text{(1-18)}\)

Figure 1.7 shows two information signals and their respective AM, PM and FM waveforms.

1.5 FM SPECTRUM*

Equation (1-18) describes an FM signal modulated by a sine-wave signal. By expanding the cosine we obtain:

\[
V_{FM}(t) = V_C \cos(\omega_{ct} + \beta \sin \omega_{mt}) = \\
= V_C \cos \omega_{ct} \cdot \cos(\beta \sin \omega_{mt}) - V_C \sin \omega_{ct} \cdot \sin(\beta \sin \omega_{mt})
\]  \(\text{(1-19)}\)

\( \cos(\beta \cdot \sin \omega_{mt}) \) and \( \sin(\beta \cdot \sin \omega_{mt}) \) can be expanded into Fourier series:

\[
\cos(\beta \cdot \sin \omega_{mt}) = J_0(\beta) + \sum_{n \text{ even}}^{\infty} 2J_n(\beta) \cos n \omega_{mt}
\]

\[
\sin(\beta \cdot \sin \omega_{mt}) = \sum_{n \text{ odd}}^{\infty} 2J_n(\beta) \sin n \omega_{mt}
\]  \(\text{(1-20)}\)

\* This section can be omitted with only the resulting spectrum studied.
where $J_0(\beta)$ are Bessel functions of the first kind, of order $n$ and argument $\beta$.

By substituting equations (1-20) into equation (1-19) and rearranging the expressions, we obtain:

$$v_{FM}(t) = V_C \cdot J_0(\beta) \cos \omega_C t +$$
$$+ \sum_{n \text{ odd}}^{\infty} V_C \cdot J_n(\beta) [\cos(\omega_C + n \omega_m) t - \cos(\omega_C - n \omega_m) t] +$$
$$+ \sum_{n \text{ even}}^{\infty} V_C \cdot J_n(\beta) [\cos(\omega_C + n \omega_m) t + \cos(\omega_C - n \omega_m) t] \quad (1-21)$$

or by again rearranging the expressions we obtain:

$$v_{FM}(t) = V_C J_0(\beta) \cos \omega_C t + \sum_{n=1}^{\infty} V_C J_n(\beta) \cos(\omega_C + n \omega_m) t +$$
$$+ (-1)^n \sum_{n=1}^{\infty} V_C J_n(\beta) \cos(\omega_C - n \omega_m) t \quad (1-22)$$

Equation (1-22) shows that the FM wave is composed of an infinite number of frequency components; Lower Side Bands of frequencies $(\omega_C - n \omega_m)$, a carrier frequency component, and Upper Side Bands of frequencies $(\omega_C + n \omega_m)$. The amplitude of each side band is determined by $V_C J_n(\beta)$ and is controlled by the carrier amplitude and the modulation index.

The frequency spectrum of the FM wave as given in equation (1-22) is shown in figure 1.8 (assuming $V_C = 1$).

A number of interesting characteristics of FM can be seen from figure 1.8. Since there are an infinite number of sidebands, the bandwidth required to pass an FM signal perfectly is infinite. The spectrum is not symmetrical about the carrier frequency (as in AM), and for certain values of $\beta$, $J_0(\beta) = 0$, and there is no carrier frequency component. This means that unlike in AM, the carrier frequency component contains part of the information.

*FIGURE 1.8: Frequency Spectrum of FM Wave Modulated by Sine Wave Signal*
To understand the practical bandwidth considerations for FM it is necessary to draw a graph of the Bessel Functions $J_n(\beta)$ for various values of $n$ (figure 1.9). The number of sideband lines that have appreciable amplitude depends on the modulation index $\beta$.

When $\beta << 1$ only $J_0(\beta)$ and $J_1(\beta)$ are significant, so that the spectrum will consist of three lines, similar to AM, but with a phase reversal of the lower sideband as shown in figure 1.10.

When $\beta >> 1$ there are many significant sideband lines, and this implies that a large bandwidth is necessary.

It can be shown that for $\beta >> 1$, the significant sideband lines are contained in the frequency range $\omega_c \pm \beta \omega_m$, while for $\beta << 1$ the significant lines are in the range $\omega_c \pm \omega_m$.

This means that the bandwidth required for an FM wave is: $BW = 2\beta \omega_m$ where $\omega_m = \frac{\omega_m}{2\pi}$.

Since $\beta = \frac{\Delta \omega}{\omega_m}$, the required bandwidth is equal to twice the maximum frequency deviation $2 \cdot \Delta \omega$, which means that in order to limit the bandwidth it is necessary to reduce the frequency deviation.

Another important result is that since the frequency deviation is dependent only on the modulating signal amplitude ($\Delta \omega = K \cdot \omega_m$) the bandwidth does not depend on the modulating frequency, though this frequency does determine the number of sideband lines contained in the bandwidth.
chapter 2
AM MODULATORS & DETECTORS

In chapter 1 we discussed AM, its spectrum, mathematical representation and parameters. In this chapter we shall see how AM is produced, how the information is recovered from the AM wave, and shall study some practical circuits.

There are four basic methods of amplitude modulating a carrier wave with an information signal:

1. Analog Multiplication
2. Chopping
3. Non-Linear Device Modulation
4. Tuned Circuit Modulation.

2.1 ANALOG MODULATION

An analog multiplier is a device with two inputs and one output, as shown in figure 2.1.

The output is proportional to the product of the input signals. To obtain AM it is necessary to feed one input with the carrier wave, and the other with the information signal (the signal, in short) to which a dc component has been added.

Therefore:

\[ v_1(t) = v_c(t) = V_1 \cos \omega_c t \]

\[ v_2(t) = V + v_m(t) = V + V_2 \cos \omega_m t \]

and

\[ v_o(t) = k \cdot v_1(V + V_2 \cos \omega_m t) \cos \omega_c t = kV_1V(1 + \frac{V_2}{V} \cos \omega_m t) \cos \omega_c t = v_c(1 + m \cos \omega_m t) \cos \omega_c t = v_{AM}(t) \]  \hspace{1cm} (2-1)

where:

\[ kV_1V = V_c \quad ; \quad \frac{V_2}{V} = m \]
The output is the desired AM wave.

Circuits that can operate as analog multipliers exist in integrated form, their great advantage being their capability of operating through a wide range of frequencies (for example the 1496 treated in DEGEM's System M).

2.2 CHOPPERS

A schematic diagram of a Chopper or Switching Modulator is shown in figure 2.2. It is basically a switch that opens and closes at the carrier frequency, through which the signal is transferred.

The output of the switch (before the filter) is shown in figure 2.3.

It can be seen that $v_S(t)$ can be obtained from $v_m(t)$ through multiplying the signal by a square wave of frequency $\omega_c$.

If we designate the square wave by $S(t)$, we can expand it into a Fourier series of the form:

$$S(t) = \frac{1}{2} + \frac{2}{\pi} \cos \omega_c t - \frac{2}{3\pi} \cos 3\omega_c t + \ldots \ldots$$  \hspace{1cm} (2-2)

If again, the signal contains a dc component and is given as:

$$v_m(t) = V + V_1 \cos \omega_m t$$

then the switch output will be:

$$v_S(t) = v_m(t) \cdot S(t) = \frac{1}{2}V + \frac{1}{2}V_1 \cos \omega_m t + \frac{2}{\pi}V \cos \omega_c t +$$

$$+ \frac{2}{\pi}V_m \cos \omega_m t \cos \omega_c t + [\text{higher frequency components}]$$ \hspace{1cm} (2-3)

By passing the expression of equation (2-3) through a band-pass filter whose center frequency is $\omega_c$; we obtain:

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FIGURE 2.3: Switch Output Signal (Chopped)

\[ v_o(t) = \frac{2}{\pi} V \cos \omega_c t + \frac{2}{\pi} V_m \cos \omega_m t \cos \omega_c t = \]

\[ = \frac{2V}{\pi} [1 + \frac{V_m}{V} \cos \omega_m t] \cos \omega_c t = \]

\[ = V_C [1 + m \cos \omega_m t] \cos \omega_c t = v_{AM}(t) \]  \hspace{1cm} (2-4)

where:

\[ V_C = \frac{2V}{\pi} \]

\[ \frac{V_m}{V} = m \]

The bandpass filter must attenuate the higher frequency harmonics \(2\omega_c - \omega_m\), \(3\omega_c - \omega_m\) etc., and the component containing \(\cos \omega_m t\). Therefore the lower side band frequency \((\omega_c - \omega_m)\) must be greater than the signal frequency \(\omega_m\), which means \(\omega_c > 2\omega_m\).

Therefore, a chopping modulator can only operate properly if the carrier frequency is more than twice the highest signal frequency component.
2.3 NON-LINEAR DEVICE MODULATORS

A non-linear device is one whose output is not a linear function of its input, such as is shown in figure 2.4.

Any non-linear function \( v_0 = f(v_1) \) can be expanded into a power series:

\[
v_0 = f(v_1) = a_0 + a_1 v_1 + a_2 v_1^2 + \ldots .
\]  \hspace{1cm} (2-5)

where:

\( a_0, a_1, a_2 \ldots \) are constants.

If the carrier and signal are added together and are then passed through a non-linear device the following result is obtained:

\[
v_1 = V_1 \cos \omega_c t + V_2 \cos \omega_m t
\]

\[
v_0 = a_0 + a_1 (V_1 \cos \omega_c t + V_2 \cos \omega_m t) + a_2 (V_1 \cos \omega_c t + V_2 \cos \omega_m t)^2 + \ldots =
\]

\[
= a_0 + a_1 V_2 \cos \omega_m t + a_1 V_1 \cos \omega_c t + 2a_2 V_1 V_2 \cos \omega_m t \cos \omega_c t +
\]

\[
+ a_2 V_1^2 \cos^2 \omega_c t + a_2 V_2^2 \cos^2 \omega_m t + \ldots .
\]  \hspace{1cm} (2-6)

The third and fourth expressions in equation (2-6) are the AM wave since:

\[
a_1 V_1 \cos \omega_c t + 2a_2 V_1 V_2 \cos \omega_m t \cos \omega_c t =
\]

\[
= a_1 V_1 (1 + \frac{2a_2}{a_1} V_2 \cos \omega_m t) \cos \omega_c t =
\]

\[
= V_C (1 + m \cos \omega_m t) \cos \omega_c t = v_{AM}(t)
\]  \hspace{1cm} (2-7)

where:

\( V_C = a_1 V_1 \); \( m = \frac{2a_2}{a_1} V_2 \)

![Figure 2.4: Non-Linear Device](image)
The other expressions in equation (2-6) are at frequencies either lower or higher than \( \omega_c \pm \omega_m \) and can be filtered out by a bandpass filter with a center frequency of \( \omega_c \).

2.4 TUNED CIRCUIT MODULATORS

Tuned Circuit Modulators use a tuned circuit fed by both the carrier and the signal with the resonant frequency set at the carrier frequency. A theoretical tuned circuit modulator is shown in figure 2.5.

We shall not analyse this circuit mathematically, leaving it for the enterprising student.

2.5 PRACTICAL MODULATORS

A practical analog modulator is shown in figure 2.6. A FET is used as a multiplier.

The current through the FET is proportional to the product of \( v_{DS} \) and \( v_{GS} \) (since the source of the FET is at virtual ground) and is converted by the operational amplifier into an output voltage.

The dc component which must be added to the signal \( [v_m(t)] \) is provided here by a battery \( (V) \).

One example of a chopping modulator is the diode bridge modulator shown in figure 2.7. The carrier amplitude must be sufficient to drive each pair of diodes on and off each cycle, so as to provide the necessary chopping.
FIGURE 2.7: Diode Bridge Chopping Modulator

When $v_c(t)$ has the polarity shown in figure 2.7, all the diodes conduct, the bridge acts as a low impedance between A and B, and the signal $v_m(t)$ does not reach the filter. When $v_c(t)$ reverses its polarity, all the diodes are cut off and the signal $v_m(t)$ reaches the filter. The bridge acts therefore as a parallel switch. Battery $V$ provides the necessary dc addition to the modulating signal.

An example of a non-linear device modulator is shown in figure 2.8 (the dc biasing circuitry has been left out).

The ac current through the transistor is determined by the signal and the carrier and is a non-linear function of them both. The tuned circuit in the collector composed of the primary of $T_1$ and $C_1$, must attenuate all the frequency components not in the range $\omega_c - \omega_{\text{max}}$ to $\omega_c + \omega_{\text{max}}$, where $\omega_{\text{max}}$ is the maximum modulating frequency, and must therefore have a fairly high $Q$ factor.

2.6 **AM DEMODULATION (DETECTION)**

There are two basic methods of detecting AM:

**SYNCHRONOUS DETECTION** and **ENVELOPE DETECTION**

FIGURE 2.8: Non-Linear Device Modulator
Synchronous detection involves multiplying the AM signal by a sine wave whose frequency is equal to the carrier frequency, and then filtering the resulting product voltage. If $v_{AM}(t) = V_c(1 + mf(t))\cos \omega_c t$ where $f(t)$ is the information signal (not necessarily a sine wave) then by multiplying by $\cos \omega_c t$ we obtain:

$$v_{AM}(t) \cdot \cos \omega_c t = V_c[1 + mf(t)]\cos^2 \omega_c t =$$

$$= V_c[1 + mf(t)]\left[\frac{1}{2}(\cos 2\omega_c t + 1)\right] =$$

$$= \frac{1}{2} V_c[1 + mf(t)] + \frac{1}{2} V_c[1 + mf(t)]\cos 2\omega_c t \quad (2-8)$$

The first expression in equation (2-8) is the information signal with an added dc component. The second expression is at a much higher frequency ($2\omega_c$) and can easily be filtered out by connecting a small capacitor in parallel with the output. Any analog multiplier can be used as a synchronous detector, though a chopping circuit can serve as well, since multiplying the AM signal by a square wave at the carrier frequency, produces the information signal plus higher frequency components [show this by multiplying the AM signal by equation (2-2)].

An example of a synchronous detector is given in figure 2.9.

The transistor is switched on and off by $V\cos \omega_c t$ and therefore $v_{AM}(t)$ is chopped; the high frequency components are filtered by $C_3$ leaving the demodulated information signal.

A simpler method of detection and by far the most common, is envelope detection. This method is based on the fact that the information is contained in the envelope of the AM waveform. The principle of envelope detection is shown in figure 2.10.

The AM signal is half-wave rectified (usually by a diode) and then the high-frequency carrier component is filtered out leaving the signal.

The important step in the detection is the filtering and a practical diode envelope detector (shown in figure 2.11) usually has a $\Pi$ RC filter.

![FIGURE 2.9: Synchronous Detector](image-url)
FIGURE 2.10: Principle of Envelope Detection

The electrolytic capacitor at the output (C3) filters the dc component of the signal and provides dc isolation for the next stage, which is the output amplifier.

FIGURE 2.11: Practical Diode Detector
In this chapter we shall describe some practical FM modulators and detectors, which are commonly used in FM circuits.

3.1 FM MODULATORS

There are many different practical methods of frequency modulation. Some are derived from a differential equation whose solutions are FM signals, while others involve modulating a square wave or a triangular wave and then filtering the result through an appropriate filter.

The method we shall discuss here is that using a voltage controlled oscillator (VCO).

A VCO is an oscillatory circuit, whose frequency of oscillation is determined by a dc or low frequency ac voltage or current. One way to obtain a VCO is to replace the capacitor in the resonant circuit of an oscillator with a voltage-variable capacitance diode (VVC), whose capacitance is a function of the voltage applied across it.

A VCO based on a Hartley oscillator is shown in figure 3.1.

If instead of controlling the frequency with a dc voltage, a low frequency signal is connected to the VVC, the frequency of the oscillator will vary according to the low frequency signal, and an FM signal will be obtained.

FIGURE 3.1:
Voltage Controlled
Hartley Oscillator
A better version of a Hartley VCO is shown in figure 3.2. Here, a differential stage (Q1 - Q2) with a transistor current source (Q3) serves as the amplifying element. P1 determines the carrier frequency and the modulating signal is connected to AF in.

3.2 LIMITERS

As was discussed in chapter 2, an AM detector need only retrieve the envelope shape of the AM wave. An FM demodulator, on the other hand, must retrieve information from the frequency and phase of the FM wave and must ignore variations in amplitude. Unfortunately most FM demodulators will also demodulate AM to some extent, and since the envelope of the FM signal is always affected by noise, nonlinearities in amplifier stages etc., these amplitude variations may be detected and will appear at the output as noise or distortion.

Nearly all FM receivers incorporate a circuit which removes envelope amplitude variations before the actual detection is done. This circuit is called a Limiter.

In many cases the Limiter is not a separate circuit but forms an integral part of the IF amplifiers.

The ideal limiter characteristic is shown in figure 3.3.

In the ideal limiter any input signal is converted to a constant amplitude output signal. In a practical limiter the input signal amplitude must be above a certain level - the limiting threshold. A diode limiter is shown in figure 3.4(a) with the characteristic in figure 3.4(b). When the input voltage is less than the diodes forward saturation voltage (0.7 V for silicon diodes) the diodes act basically as resistances and the output voltage follows the input voltage.

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When the input voltage increases above the diodes forward saturation voltage ($V_D$) the output voltage is limited to $V_D$.

Therefore this limiter operates well only for input signals that are greater than $V_D$.

A far better limiter which has a much lower limiting threshold is shown in figure 3.5.

The input signal $v_i$ switches the two differential stage transistors Q1 and Q2. When one of these transistors is fully on, the other is cutoff. In this state the output voltage is determined by the current provided by the source transistor Q3 (which is constant) and resistor $R_2$, and is not a function of the input voltage. Therefore the circuit operates as a dynamic limiter and can also amplify the input signal.

In a typical circuit which uses a 741 IC (which contains the transistors and diodes of figure 3.5) the input voltage required to switch one transistor on and the other off is about 115 mV, therefore an input amplitude of 250 mVpp will ensure good limiting.

Since this circuit amplifies as well as limits, it can serve as part of the IF amplifier, if resonant circuits are added to its inputs and output.

(a) Practical Diode Limiter

(b) Characteristic of Circuit in (a)

FIGURE 3.4
3.3 FM DEMODULATORS

Most FM Demodulators use a technique described in figure 3.6.

The amplitude-limited FM signal is differentiated and the resulting signal is passed to an envelope detector.

To understand this let us differentiate the FM signal given by equation (1-15):

\[
\frac{d}{dt} [v_{FM}(t)] = -V_C(\omega_C + K_FV_m \cos \omega_m t) \text{envelope} \cdot \sin[\omega_C t + K_FV_m \int_0^t \cos \omega_m \lambda d\lambda]
\]

(3-1)

By passing the signal of equation (3-1) through an envelope detector only \( V_C(\omega_C + K_FV_m \cos \omega_m t) \) remains which is proportional to the original information signal.

There are three basic methods of differentiating an FM signal:

(a) Direct Differentiation

(b) Frequency-Domain Differentiation

(c) Time-Delay Differentiation

\[
\begin{align*}
\text{Differentiator} & \quad v'(t) \\
\text{Envelope Detector} & \quad v_o(t)
\end{align*}
\]

FIGURE 3.6: The Principle of FM Demodulation by Differentiating

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3.4 DIRECT DIFFERENTIATION

An example of an FM Demodulator using direct differentiation is given in figure 3.7. Capacitor $C_1$ serves as the differentiating element. The emitter of $Q_1$ is ac grounded through diode $D_1$ and the base-emitter junction of the transistor. Therefore the current through the capacitor is: $i_c = C \frac{dv_i}{dt}$. Only when $i_c$ is in the direction shown in figure 3.7 does it flow through the transistor, therefore the circuit operates as a half-wave rectifier. $R_1$ and $C_2$ filter the rectified signal and the output is therefore the required information signal.

This demodulator has the disadvantage of distorting the output signal, because the diodes ($D_1$ and B-E of $Q_1$) are not ideal and therefore capacitor $C_1$ is not an ideal differentiator.

3.5 FREQUENCY-DOMAIN DEMODULATORS

A frequency-domain differentiator is a linear network whose transfer function has the shape shown in figure 3.8. It can be shown that any network with a similar transfer function operates as a differentiator in a range around the center of its linear slope.

For example, the transfer function shown in figure 3.8 has the form $H(s) = ks$. Therefore the inverse Laplace transform is $h(t) = k \cdot \frac{d}{dt}$ and this represents a differentiator.

One practical way of obtaining a differentiator is to use a resonant circuit, with the transfer characteristic similar to that shown in figure 3.9.
In the region between points A and B the transfer function is approximately linear and therefore the resonant circuit can operate as a differentiator around a frequency of $\omega_C$, when tuned to a resonant frequency of $\omega_0$ (the down-slope of the circuit can be used instead, giving a negative differential).

A practical frequency-demodulator using this principle is shown in figure 3.10.

The resonant circuit composed of coil L and capacitor C is followed by an envelope detector composed of resistor $R_0$ and capacitor $C_0$. The connection between $v_0(t)$ and the modulating frequency $\omega_m$ is shown in figure 3.11 and is called the Demodulator S Curve.

The disadvantage of this demodulator is that it will detect AM almost as well as FM, and if the FM input signal is not perfectly limited, the amplitude variation will appear at the output as noise. What is more, the detector output is not zero for a non-modulated input signal but is some dc voltage level.

A demodulator that solves these problems is the Balanced-Slope Demodulator shown in figure 3.12.

Each half of the demodulator consists of a single slope-demodulator tuned to different frequencies ($\omega_{r1}$ and $\omega_{r2}$). The left hand one operates as a positive-slope differentiator while the right-hand one operates as a negative-slope differentiator. The S curve of a balanced slope demodulator is shown in figure 3.13.
An AM signal or amplitude changes in an FM signal will be detected identically by each half of the demodulator and, since the output voltage is differential, will not interfere with the demodulated FM signal.

3.6 TIME DELAY DIFFERENTIATION

A time-delay differentiator operates by approximating the fundamental definition of a derivative:

\[
\frac{dv(t)}{dt} = \lim_{t_0 \to 0} \frac{v(t) - v(t-t_0)}{t_0}
\]
The block diagram of such a circuit is shown in Figure 3.14. The delay $t_0$ must be small compared with variations in $v(t)$.

Since a time delay can be seen as a phase shift, the same operation can be performed by a phase-shifting circuit.

A practical phase-shift demodulator is the Foster-Seely Demodulator shown in Figure 3.15.

The operation of the Foster-Seely demodulator is as follows: The secondary of transformer $T_1$ forms part of two resonant circuits, one consisting of the upper half of $T_1$ and $C_2$, and the other consisting of the lower half of $T_1$ and $C_3$. Both these circuits are tuned to the center frequency ($f_c$) of the FM signal which is to be detected.

Diodes $D_1$ and $D_2$ with their respective low pass filters ($R_1$ and $C_4$, $R_2$ and $C_5$) form envelope detectors.

When the input signal is a sine wave whose frequency is equal to $f_c$, the voltages on the secondary of $T_1$ are composed of two components. One is the voltage induced by the input signal in the primary of $T_1$, and the other is the input signal itself, which comes directly through $C_1$. The induced voltage is shifted in phase by $90^\circ$ in relation to the primary voltage as shown in Figure 3.16(a).

The voltage reaching $D_1$ and $D_2$ is the phasor sum of these two components $v_1$ and $v_2$. Since the envelope detectors detect the amplitudes of $v_1$ and $v_2$ (which are equal), the output voltage $v_o$, which is the difference between $v_1$ and $v_2$, is equal to zero.
FIGURE 3.16: Phasor Diagrams for (a) $f_{in} = f_c$ (b) $f_{in} < f_c$ (c) $f_{in} > f_c$

When the frequency of the input signal is less than $f_c$, the resonant circuits are reactive and the phasor diagram of figure 3.16(b) is obtained. Here $|v_2'| > |v_1'|$, therefore $v_o$ will be positive.

When the frequency of the input signal is greater than $f_c$, the resonant circuits are inductive and the phasor diagram of figure 3.16(c) is obtained. Here $|v_2''| < |v_1''|$, therefore $v_o$ will be negative.

It can be seen that $v_o$ depends on the frequency of the input signal, and therefore the Foster Seely demodulator can demodulate FM.

The main disadvantage of the Foster Seely demodulator is that it detects amplitude variations of the input signal, since the amplitudes of $v_1$ and $v_2$ at a particular frequency, depend on the input signal amplitude.

This disadvantage is overcome in the Ratio Detector shown in figure 3.17. The operation of the circuit is similar to that of the Foster Seely demodulator, and the same phasor diagrams apply.

If the diode output voltages are $v_a$ and $v_b$, as shown in figure 3.17, then $v_o = \frac{v_a + v_b}{2} - \frac{v_a - v_b}{2}$.

$v_a$ and $v_b$ are equal in amplitude to $v_1$ and $v_2$ in the Foster-Seely demodulator.

FIGURE 3.17: Ratio Detector
The voltage on capacitor $C_0$ is $v_a + v_b$. Since the time constant of $C_6$ and $R_1 + R_2$ is chosen to be very large compared with audio frequencies, the voltage on the capacitor is constant and therefore $v_a + v_b = \text{const}$. For an FM input signal, if $v_a$ increases $v_b$ decreases and vice versa as in the Foster Seeley demodulator. For an AM input signal both $v_a$ and $v_b$ should increase, but capacitor $C_6$ prevents this and therefore the AM input has no effect on the output voltage.

Another FM demodulator which operates on the phase shift principle is the Product Detector shown in figure 3.18.

The input resonant circuit composed of $R_1$, $C_3$ and $L_1$, produces a phase shift proportional to the frequency of the input signal. The phase shift for an input frequency equal to the resonant frequency is $90^\circ$. The phase-shifted signal is amplified by the differential stage composed of $Q_1$ and $Q_2$. The gain of this stage is determined by the current through $Q_3$ which is set by the input signal itself. Therefore this circuit serves as a multiplier.

By multiplying the phase-shifted signal by the input signal, a number of different frequency components are produced. The low frequency component is proportional to the information signal and therefore the output low pass filter removes all but this component.

FIGURE 3.18: Product Detector
We have seen that to send information by radio waves we must modulate a high-frequency carrier and then broadcast it from an antenna.

From now on we shall consider the other end of the story - the receiver.

4.1 RECEIVERS - GENERAL

Basically, a receiver must pick up the electromagnetic radio waves, convert them to electrical current and then remove the information from the other components (carrier wave, noise, other stations etc.).

Let us look at the different functions that must exist in a radio receiver. First of all there must be an antenna - to convert the electromagnetic waves into electrical current. External antennas come in all shapes and sizes depending on the frequency at which they operate and various other parameters.

In figure 4.1 the block diagram of a radio is given; each block representing a particular function.

The antenna is coupled to the receiver through an antenna circuit which can be just a wire or a complicated coupling circuit, depending on the type of receiver.

Since the antenna picks up a great number of radio frequencies it is necessary to select the desired frequency (or station). This is done by the tuner, which can be a bandpass filter, with a variable center frequency.

The signals picked up by the antenna are often very weak and must be amplified, so as to enable the information to be removed easily from the modulated wave. This amplification can be done before the tuner, after the tuner or both. In figure 4.1 it is done after the tuner by the RF amplifier.

The information can now be detected or in other words the radio wave can be demodulated. The resulting signal, which should resemble the modulating wave in the transmitter, is amplified by an audio amplifier (if it is an audio signal) and can then be fed to a loudspeaker.

This basic description of a receiver holds not only for AM receivers but for all types of radio receivers (FM, SSB etc.).

![Functional Block Diagram of Receiver](image)
4.2 THE AM RECEIVER

We shall now take a look at various practical realizations of an AM receiver. The very first commercial radio was the crystal or cat's whisker set. This basically contained a piece of crystal with a thin wire leading to it, which operated as a diode which detected the radio signal. To this an earphone and an antenna were connected (one at each end) and a radio broadcast could be heard.

The next stage in the development of the radio was the use of electronic components such as the diode and the triode for amplification and detection.

The great breakthrough in AM radio came with the perfection of the SUPERHETERODYNE receiver by E. H. Armstrong in 1918.

4.3 THE SUPERHETERODYNE RECEIVER

Before we explain how the Superheterodyne receiver works, it is first necessary to understand the disadvantages of the ordinary receiver.

The main problems arise in the tuner and the RF amplifier. Since the tuner is not ideal, not only the required frequency is passed by it, but also adjacent frequencies which must be removed in the RF amplifier, otherwise they will be detected and will distort the output audio signal. This means that the RF amplifier must have as narrow pass-band as possible, which will only pass the desired frequency and will suppress the additional frequencies, which were passed by the tuner. The center frequency of this tuned amplifier must be variable, so as to allow reception of different stations (or frequencies). Practically, it is difficult to build a variable bandpass amplifier with the required specifications, and it was this that led to the development of the superheterodyne receiver (or superhet, for short).

The basic idea is to have a very good narrow bandpass amplifier at a fixed frequency called the INTERMEDIATE FREQUENCY (IF), and then to convert the desired radio signal to this frequency. This is done by multiplying the RF signal by a sine wave produced by an oscillator in the receiver itself and called the LOCAL OSCILLATOR (LO). The product of two sine waves contains two frequency components, the sum and the difference of the two original frequencies.

Since the IF amplifier is tuned to a fixed frequency, if we determine the LO frequency so that the difference between it and the radio frequency is exactly the center frequency of the IF amplifier, in other words \( f_{IF} = f_{LO} - f_{RF} \), then the particular station will be received. The LO frequency must be variable in order to allow different stations to be received, but it is much easier to build a variable oscillator than a variable amplifier. Since the IF amplifier is a narrow bandpass amplifier the sum frequency is not passed by it.

A block diagram of a superhet receiver is shown in figure 4.2.

The antenna is the same whether the receiver is a superhet receiver or any other type since the differences occur after the RF signal has been received.

The RF amplifier is not essential and is often part of the mixer.

The mixer is the circuit that produces the IF signal by multiplying the LO signal
FIGURE 4.2: Block Diagram of an AM Superheterodyne Receiver

and the RF signal (by multiplying two sine waves at different frequencies $f_1$ and $f_2$ components at frequencies of $f_1 + f_2$ and $f_1 - f_2$ are produced). Most mixers produce the difference frequency, while attenuating the sum frequency.

The Local Oscillator is a variable frequency sine wave oscillator.

The AGC (Automatic Gain Control) circuit ensures that the output audio signal remains at a constant level, even if the RF signal changes its amplitude slightly.

The intermediate frequency for commercial AM radios is usually 455 kHz. The medium waveband for commercial AM is between 535 kHz - 1620 kHz. There are two possibilities of setting the local oscillator frequency band: $f_{LO}$ can be higher than $f_{RF}$ ($f_{LO} > f_{RF}$) or $f_{LO}$ can be lower than $f_{RF}$ ($f_{LO} < f_{RF}$).

Since $f_{IF}$ is the difference between $f_{RF}$ and $f_{LO}$, if $f_{LO} < f_{RF}$ the LO frequency band would be from 535 kHz - 455 kHz = 80 kHz to 1620 kHz - 455 kHz = 1165 kHz. The relative frequency change of the local oscillator would be 80 kHz: 1165 kHz which is about 1:15.

If $f_{LO} > f_{RF}$, the LO frequency band is from 535 kHz + 455 kHz = 990 kHz to 1620 kHz + 455 kHz = 2075 kHz. The relative frequency change of the local oscillator is 990 kHz : 2075 kHz, which is about 1:2.

Since it is much easier to design a variable oscillator with a 1:2 frequency ratio than with a 1:15 frequency ratio, the LO frequency is always higher than the RF. The RF and LO frequency bands are shown in figure 4.3.

4.4 CHARACTERISTIC PARAMETERS OF RECEIVERS

When a receiver is tuned to a particular station two things happen: the antenna circuit is tuned to the radio frequency ($f_{RF}$) and the local oscillator is tuned to $f_{RF} + 455$ kHz.

![IF LO frequency range](image)

FIGURE 4.3: AM: Medium and LO Wavebands
The need for a tuned antenna circuit arises because of the phenomena of IMAGE FREQUENCIES. When a radio frequency of $f_{LO} + 455\text{ kHz}$ enters the mixer, along with the desired frequency of $f_{LO} - 455\text{ kHz}$, both of them are converted to 455 kHz. This means that although the radio is tuned to one frequency, another frequency is also received.

Figure 4.4 shows a particular radio frequency and its image frequency.

The tuned antenna circuit before the mixer is meant basically to reject the image frequency, while the IF amplifier rejects frequencies adjacent to the desired station.

The IF amplifier is tuned to 455 kHz with a bandwidth of about 18 kHz, which is sufficient to pass the AM signal produced by standard transmitters.

There are a number of parameters which are of great importance in specifying AM receivers. They are:

- **SENSITIVITY** - input signal level required to produce certain audio power at the output.

- **SELECTIVITY** - the ability to receive separately two adjacent channels.

- **SIGNAL-TO-NOISE RATIO (S/N)** - the ratio of the signal at the output to the noise at the output.

- **FIDELITY** - the ability to reproduce accurately the information signal.

- **IMAGE FREQUENCY REJECTION** - see previous discussion of image frequency.

The following table shows how each stage of the receiver affects these characteristics.

These characteristics will be treated again when each stage of the receiver is discussed.
4.5 FM RADIO RECEIVERS

Most FM receivers use the superheterodyne principle, for the same reasons that it is used in AM receivers. In fact the basic block diagram of an FM superhet receiver is almost identical to that of an AM superhet receiver.

The AFC (Automatic Frequency Control) circuit ensures that the receiver remains tuned to a particular station, even if the station's frequency varies slightly.

The frequency range of commercial FM radio transmissions is between 88 MHz and 108 MHz, and therefore the required antenna length is about 1.5 m. The RF amplifier, mixer and local oscillator are basically the same as in the AM receiver, although various different tuning methods are used.

The intermediate frequency used in most FM receivers is 10.7 MHz, and in some cases the IF amplifier also serves as a limiter.

The characteristics of an FM receiver are defined and measured by the same parameters that apply to an AM receiver with a number of additions.

One of these is the AM Rejection Ratio, which measures the effect of amplitude variations of the input signal on the receiver output signal. Most of the AM rejection is provided by the limiter, and in certain cases by the detector.

FIGURE 4.5: Block Diagram of an FM Superheterodyne Receiver
5.1 ANTENNA

In chapter 1 we saw that for medium wave reception a fairly long antenna is required. This would not be practical in receivers for commercial use, and therefore a different kind of antenna is used. This is a ferrite-rod antenna as shown in figure 5.1(a). Ferrite is a ferromagnetic material which has the effect of increasing the currents produced in the antenna coil by the electromagnetic radio signals. In this way a short, normally ineffective antenna can operate satisfactorily. Often an additional coil is wound on the ferrite rod, which feeds the RF amplifier. In this way the ferrite antenna serves both as an antenna and as an input transformer.

Since the antenna must receive only a certain frequency, it must be tuned, and this is done by connecting a variable capacitor in parallel with the coil of the antenna as shown in figure 5.1(b).

In some cases the tuning circuit is part of the RF amplifier, and by designing the amplifier with high enough gain the ferrite-rod antenna can be replaced by a short piece of wire. As an antenna, the wire is very inefficient, but in conjunction with the RF amplifier it is sufficient.

5.2 RF AMPLIFIERS

In receivers which have an antenna with its own tuning capacitor, the RF amplifier is simply a linear amplifier with a bandwidth large enough to pass all the radio
frequencies received by the antenna. A discussion of such amplifiers can be found in any book on basic electronic circuits and they will not be treated here.

We shall look at an RF amplifier which contains the tuning circuit and is therefore a tuned amplifier.

One of the simplest tuned amplifiers is a single transistor stage with a resonant circuit in the output circuit. An example of such an amplifier appears in figure 5.2.

The circuit is basically a common emitter stage with a resonant load composed of the primary of T1 and variable capacitor C_{ant}.

The relative frequency range of the tuned amplifier (the ratio between the maximum frequency and the minimum frequency to which the amplifier can be tuned) is determined by the range of the variable capacitor. The absolute values of the maximum and minimum frequencies can be determined by tuning transformer T1.

5.3 MIXERS

The mixer converts the AF signal to a signal at the intermediate frequency, by multiplying it be a reference frequency produced by the local oscillator. Since the multiplication of two sine waves \( \cos(\omega_{L0} - \omega_{RF})t \) and \( \cos(\omega_{L0} + \omega_{RF})t \), and since only the difference frequency component is required, the mixer must filter out all but the desired component. This is done by tuning the output of the mixer to the intermediate frequency.

A basic transistor mixer is shown in figure 5.3. This mixer is based on the non-linear voltage current relationship of the transistor's B-E junction, which produces the product of the RF and LO signals along with other components. Note that both the inputs and output are connected through transformers.

This mixer has the disadvantage that the RF and the LO signals, as well as undesired harmonics produced by the mixer, reach the output.
An improved mixer which almost completely overcomes this disadvantage, is shown in figure 5.4.

This mixer can be constructed from an integrated circuit (371 for instance). The output is isolated from the inputs because of the balanced operation of the circuit. Capacitors C3 and C4 are bypass capacitors, while capacitors C1 and C2 tune transformer T3 to the intermediate frequency.

The RF input signal is amplified by the differential stage, whose gain depends on the current through Q3. This current is itself determined by the LO signal and therefore the output of the differential stage is the product of the RF and LO signals. The output transformer T3 attenuates the sum frequency component and leaves only the IF component.

Because of the balanced operation of this circuit the LO and RF signals do not appear at the output or are at least strongly suppressed.

5.4 LOCAL OSCILLATORS

The local oscillator is a sinusoidal oscillator whose frequency can be varied over a certain range. Since the difference between the LO frequency and the RF frequency must remain constant, the same control that alters the RF amplifier frequency must change the LO frequency.

This can be done with a double variable capacitor (two capacitors on the same axis) or by using VVC diodes instead of capacitors and controlling their dc voltages with the same potentiometer.

Most mixers require the amplitude of the LO signal to be greater than the amplitude of the RF signal. Therefore the local oscillator must produce a signal with sufficient amplitude.

The Local Oscillator which appears in system COM-1 is described and analyzed in the manual which accompanies system M.
5.5 IF AMPLIFIERS

The IF amplifier in an AM receiver must have certain specific characteristics. It must be tuned to the intermediate frequency, have a bandwidth wide enough to pass the modulated signal, but narrow enough to suppress undesired frequency components from the mixer, it must be linear and must amplify.

There are many different types of IF amplifiers, in fact any linear tuned amplifier can serve as an IF amplifier.

In most cases a single-stage amplifier is not sufficient to produce the required amplification and two or more stages are used.

An example of a two-stage IF amplifier is shown in figure 5.5.

Each transistor has a tuned LC circuit in its collector circuit and this serves both to tune the amplifier to the IF frequency and to provide the necessary bandwidth.

5.6 AUTOMATIC GAIN CONTROL (AGC)

The purpose of the AGC is to keep the detector output signal at a constant average level. There are a number of things that can cause this average level to vary, such as: variations in the RF signal amplitude, instability of the various amplifiers, etc.

The AGC monitors the average detector output level and adjusts the amplification of the IF amplifier in order to keep the level constant. This gain control is usually performed on the first IF stage only. An example of an AGC circuit that can operate with the IF amplifier shown in figure 5.5, appears in figure 5.6.
The circuit operates in the following manner: The dc current through Q1 (which determines the transistor's gain) depends on the voltage on resistor R6, which in its turn is determined by the base voltage of Q1. When QAGC conducts, the current through R3 increases thereby lowering the voltage on the base of Q1, which causes the dc current and the gain of Q1 to drop. Therefore the current through QAGC controls the gain of the 1st IF stage. This current is itself controlled by the average detector output voltage, thereby assuring that the detector output level remains constant. Since the AGC is a closed-loop control circuit it is necessary to stabilize it with capacitors as shown in figure 5.7, which in fact form a low-pass filter.
FIGURE 5.7: AGC Circuit with Stabilizing Capacitors

The other components that appear in an AM superheterodyne receiver are the detection circuits, which have already been discussed, and the audio amplifier, which will not be treated here.