UNIT 9 AUDIO COMMUNICATION

Structure

9.1 INTRODUCTION

Communication of voice signal over a metal wire was proven by the invention of telephone. However wireless communication of audio signals remained a challenge. Beginning with our own Dr. Jagdish Chandra Bose and the Italian physicist Guglielmo Marconi, wireless communication of signals has come a long way now.

Dr. Jagdish Chandra Bose (1858-1937) worked in many different fields of sciences-from Physics to Botany! He was pioneer in designing the first semiconductor detector for detecting radio waves. He also invented many new commonplace microwave components.

The audio communication in daily life is experienced by us in the form of radio. Radio plays an important role in providing information and entertainment. You must have already experienced the use of radio as an educational tool in the form of IGNOU's radio phone-in programmes. The advances in technology have brought the size of a radio receiver from a table-mounted box to a tiny bead that could be fitted in a ring or wristwatch. Due to the use of semiconductor *transistor* device in these small radios, they are now generically known as transistors. You may remember that cellular mobile telephone communications are also sometimes referred to as mobile radio communication since, in essence, every mobile handset is a tiny radio transmitter and receiver. Since the early nineties, digital techniques are also being used in audio communication systems.

In this unit we will discuss various forms of radio communication like amplitude modulated (AM) and frequency modulated (FM) transmission. We discuss the block diagrams of transmitters and receivers for both these types of radio communication. In recent days, it is possible to gain access to radio signals from satellites. We will take a brief review of these new advances. There are also some higher sensitivity precision receivers mainly used by scientists or military personnel. A brief description of such special receivers is given as an Appendix to this Unit.

In Sec. 9.2 you will learn about the two main types of radio receivers viz. Tuned Radio Frequency (TRF) receiver and superheterodyne receiver. In Sec. 9.3 we discuss the basic blocks of an AM receiver along with some special circuits like mixers; intermediate frequency (IF) circuits and automatic gain control (AGC) circuit. The construction and working of AM transmitters is explained in Sec. 9.4. You have already learnt about frequency modulators and demodulators in Unit 5. In Sec. 9.5 we describe the FM communication system. Here we will also describe stereo FM communication. A brief overview of satellite radio broadcast is taken in Sec. 9.6.

Objectives

After studying this unit, you should be able to:

- discuss the construction and working of TRF receiver;
- describe the base blocks of superheterodyne receiver;
- list the advantages of superheterodyne method;
- explain with typical circuits working of mixer and automatic gain control;
- discuss the critical parameters determining the performance of AM receivers; and
- explain the working of FM stereo broadcast.

9.2 TYPES OF RADIO RECEIVERS

The function of any receiver is to select the desired signal from the other signals, amplify and demodulate it, and display it in the desired manner. The major difference between receivers of various types is essentially dependent on the way in which they demodulate the received signal. In turn, this depends on the type of modulation employed in transmission, be it AM or FM.

Of the various forms of receivers proposed over the years, only two have any real practical or commercial significance: *the tuned radio-frequency* (TRF) *receiver* and the *superheterodyne receiver*. Presently, only the second type is used to a large extent, but it is necessary to understand the operation of the TRF receiver first since it is the simplest form of a receiver.

9.2.1 Tuned Radio-Frequency (TRF) Receiver

Prior to 1945, most radio receivers were of the TRF type. A block diagram of a typical TRF receiver is shown in Fig. 9.1.

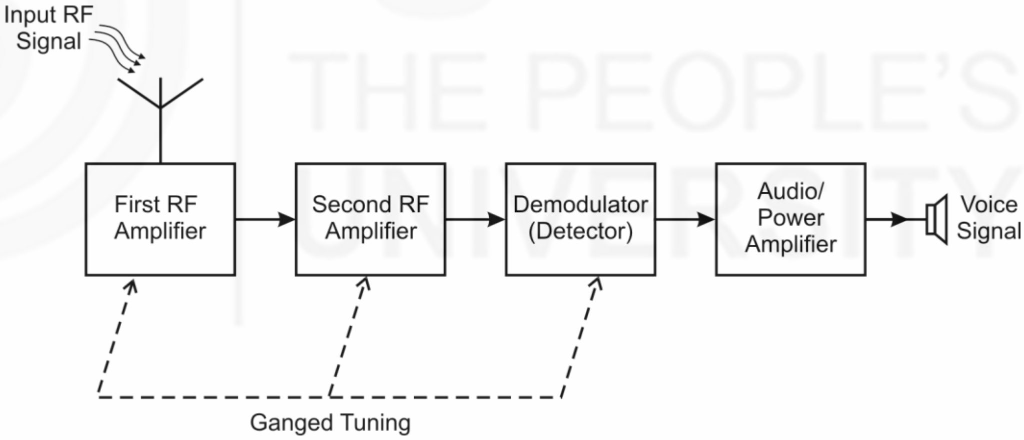

Fig. 9.1: The TRF receiver

The TRF receiver is a simple "logical" receiver with simplicity and high sensitivity. You must remember that when the TRF receiver was first introduced, it was a great improvement on the types of receiver in use previously: crystal, regenerative and super regenerative receivers.

In TRF receivers typically two or three RF amplifiers, all tuned to the same frequency, are employed to select and amplify the frequency of interest and simultaneously reject all others. After the signal is amplified to a suitable level, it is demodulated (detected) and fed to the loudspeaker after being passed through the appropriate audio amplifying stages. Such receivers are simple to design and work well at medium wave (MW) broadcast frequencies (535 to 1640 kHz), but they present difficulties at higher frequencies. This is mainly because of the risks of instability associated with

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high gain multistage amplifier. Consider an amplifier with a gain of 50,000. All that is needed is 1/50,000 of the output of the last stage to get fed back at the input of the first stage in phase with the input signal, to provide regenerative feedback and begin oscillations. This situation is almost inevitable at high frequencies and hampers the operation of a good receiver.

Another limitation faced by the TRF receiver is a variation in bandwidth over the tuning range. Consider a tuned circuit required to have a bandwidth of 10 kHz at a frequency of 535 kHz. The Q of this circuit must be $Q = f/\Delta f = 535/10 = 53.5$. At the other end of the broadcast band, i.e., at 1640 kHz, the inductive reactance (and therefore the Q) of the coil should in theory have increased to 164. In practice, however, various losses dependent on frequency will prevent so large an increase. Thus the Q at 1640 kHz is not likely to exceed 120 resulting in a bandwidth of $\Delta f = 1640/120 = 13.7$ kHz. It means that the receiver will pick up signals from an adjacent station as well as the one to which it is tuned.

Now consider a TRF receiver required to tune to 36.5 MHz, the upper end of the shortwave band. If the Q required of the RF circuits is again calculated, still on the basis of a 10 kHz bandwidth, we have $Q = 3650$! It is obvious that such a Q is impossible to obtain with ordinary tuned circuits and the bandwidth of tuning circuit is very large. The problems of instability, insufficient adjacent-frequency rejection, and bandwidth variation can all be solved by the use of a superheterodyne receiver.

Each radio broadcast station is allotted a fixed frequency from the range of broadcast band. The station has to transmit only at that frequency with assigned frequency bandwidth.

Table 9.1: Typical AM Radio Frequency Bands.

Band	Frequency
Medium Wave (MW)	520-1700 kHz
Short Wave-1 (SW-1)	4.39-5.18 MHz
Short Wave-2 (SW-2)	5.72-6.33 MHz

9.2.2 Superheterodyne Receiver

As you learnt in the case of a TRF receiver, the straightforward way to listen to any radio station would be to tune to the radio frequency (RF) of that station using a variable capacitor and then amplify the signal at that frequency. This is necessary because the signal caught by the antenna (aerial) of the radio set is very weak and cannot be used as it is.

Now, typically the AM radio at medium wave frequencies operates between 535 kHz to 1640 kHz and short wave bands from about 1700 kHz to 36 MHz. To build such a wide bandwidth amplifier, working linearly throughout the frequency range is almost impossible and if at all possible, is a very expensive affair. Instead, if an amplifier of narrow band, operating at low frequency is used then it can be quite cost effective and easy to design.

With this principle, the concept of superheterodyne (*superhet*) receiver was devised. Here the incoming signal of any above-mentioned frequency is mixed with a local oscillator frequency so as to produce *difference* or *beat* frequency. The frequency of the local oscillator is varied in such a way that for any selected input frequency of the receiver the output at the *beat frequency generator* or *mixer* (discussed in Sec. 9.3.2) is a constant frequency. Generally this frequency is kept at 455 kHz in the AM radio receivers. This frequency is called an **intermediate frequency (IF)**. Since the incoming signal is now at reduced frequency, its amplification can be done quite efficiently, using low frequency amplifiers. Such receivers using intermediate frequency are called the **superheterodyne receivers**.

You will appreciate that since most of the signal processing now is occurring at a fixed frequency of 455 kHz, the manufacturing of the electronic components can be standardised and hence their bulk production is quite cost effective and hence the cost of modern radio receivers is quite less. Use of semiconductor technology has also helped in miniaturisation and cost reduction of the radio sets in recent years.
The block diagram of basic superheterodyne receiver is shown in Fig. 9.2. In the superheterodyne receiver, the signal voltage is mixed with the local oscillator voltage such that the resultant component of difference frequency is the intermediate frequency. The IF signal contains the same modulation as the original carrier, which

can be amplified and demodulated to reproduce the original information. There is an additional block of automatic gain control in this diagram, about which we will be discussing in the next section. The *superhet* thus has the same essential components as the TRF receiver, with additional blocks of mixer, local oscillator and intermediate-frequency (IF) amplifier.

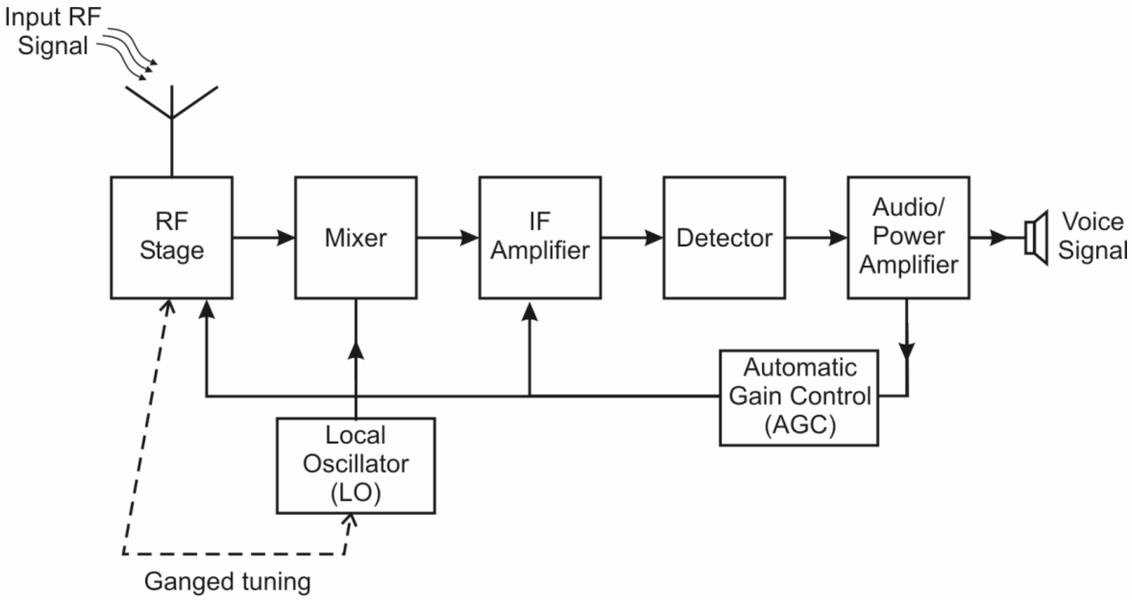

Fig. 9.2: The superheterodyne receiver

A constant frequency difference is maintained between the local oscillator and the RF circuits, normally through capacitance tuning, in which all the capacitors are *ganged* together and operated in unison by one control knob. The IF amplifier generally uses two or three transformers, each consisting of a pair of mutually coupled tuned circuits. With this large number of double-tuned circuits operating at a constant, specially chosen frequency, the IF amplifier fulfils most of the gain (and therefore sensitivity) and bandwidth requirements of the receiver. Since the characteristics of the IF amplifier are independent of the frequency to which the receiver is tuned, the selectivity and sensitivity of the superhet are usually fairly uniform throughout its tuning range and not subject to the variations as in the case of TRF receiver.

You may now like to attempt an SAQ.

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SAQ 1

Which of the following statements is *not* true?

The superheterodyne receiver replaced the TRF receiver because the latter suffered from:

i) gain variation over the frequency coverage range;

ii) insufficient gain and sensitivity;

iii) inadequate selectivity at high frequencies;

iv) instability.

Let us now discuss the various components of an AM receiver.

9.3 AM RECEIVER

The most popular form of radio communication in India over the last 50 years has been medium and shortwave AM broadcast. Recently FM broadcast is being used in some metropolitan cities however, AM continues to have the widest spread in rural India. In this section we will be discussing the functioning of an AM receiver we have in our home. All these radio sets use superheterodyne principle; hence we will focus our discussion on such receivers only.

9.3.1 RF Stage

A typical superheterodyne AM receiver has an RF section, which is a tuneable circuit connected to the antenna terminals. It selects the wanted frequency and rejects the unwanted frequencies picked up by the antenna. The signal picked up by the antenna is then passed on to an RF amplifier. The advantages of using an RF amplifier in the RF stage of the receiver are:

You have already learnt about the tuned RF amplifiers in Unit 4 of the course on Electrical Circuits and Electronics (PHE-10)

- better sensitivity;
- better selectivity; and
- improved image frequency rejection.

A typical transistorised transformer-coupled RF amplifier is shown in Fig. 9.3. This is a tuned amplifier circuit. Here the load impedance is supplied by a resonant circuit (C_4- primary of T_2), with parallel resonance, which is used to obtain the necessary high load impedance. Utilisation of resonance circuit makes the tuned amplifier frequency selective. Thus only the frequencies in the vicinity of the tuned load are amplified.

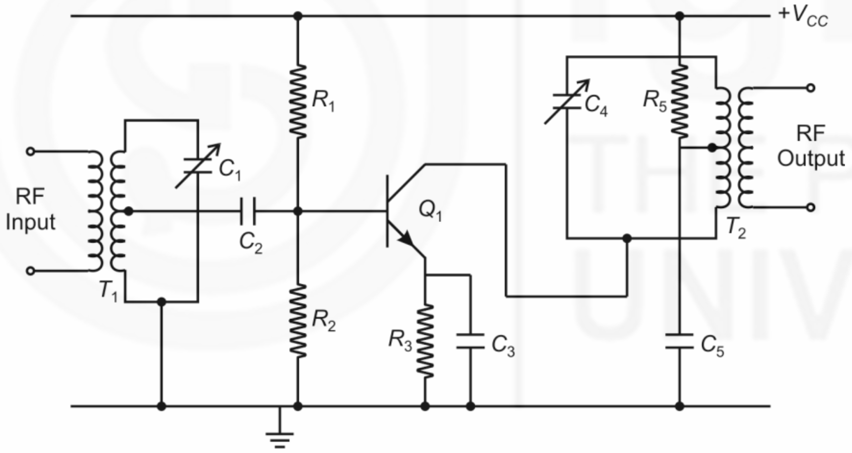

Fig. 9.3: Transistorised RF amplifier

In the present circuit, there are tuning circuits at the input and the output of the amplifier. Both these are designed to be resonant at the operating frequency of the amplifier. This tuning enhances the frequency selectivity of the circuits to a great extent.

Sensitivity

You have already learnt in the first unit (Sec. 1.3.2) that the sensitivity is defined in terms of the ratio of output signal and input signal. The characterisation of the system can be done by applying standard input signals.

For AM broadcast receivers, several of the relevant test parameters have been standardised. Typically a 30 percent modulation by a 400 Hz sine wave is used, and the signal is applied to the receiver through a standard coupling network known as a *dummy antenna*. The standard output is 50 mW, and for all types of receivers the loudspeaker is replaced by a load resistance of equal value as shown in Fig. 9.4a.

Sensitivity is often expressed in microvolts or in decibels below 1 V and is measured at three points along the tuning range while testing. A typical sensitivity curve is shown in Fig. 9.4b. It indicates that sensitivity varies over the tuning band. At 1000 kHz, this particular receiver has a sensitivity of 12.7 μV, or –98 dBV (dB below 1V).

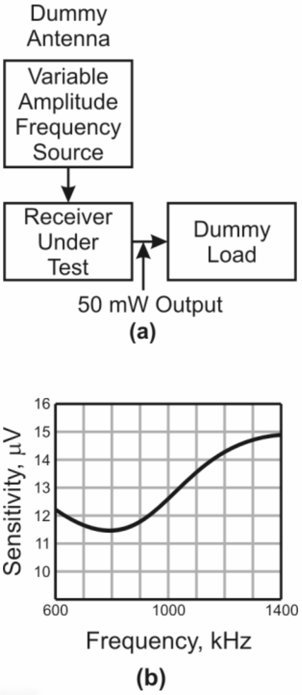

Fig. 9.4: a) Sensitivity measurement setup; and b) sensitivity curve for a typical radio receiver

For professional receivers, it is a practice to quote the sensitivity in terms of signal power required to produce a minimum acceptable output signal with a minimum acceptable signal-to-noise ratio. The measurements are made under the conditions described, and the minimum input power is quoted in dB below 1 mW or dBm. Thus, under the heading of *sensitivity* in the specifications of a receiver, you may find an entry like, "*–85 dBm 1 MHz signal,* 30 *percent modulated with a* 400 Hz *sine wave will, when applied to the input terminals of this receiver through a dummy antenna, produce an output of at least 50 mW with a signal-to-noise ratio not less than 20 dB in the output*".

The sensitivity plot in Fig. 9.4b belongs to a rather good domestic or car radio. Portable and other small receivers used only for the broadcast band might have a sensitivity in the vicinity of 150 μV, whereas the sensitivity of quality communication receivers may be better than 1 μV in the HF band.

The most important factors determining the sensitivity of a superheterodyne receiver are the gain of the IF amplifier(s) and that of the RF amplifier. You will agree that the noise figure also plays an important part in deciding the sensitivity.

Selectivity

The selectivity of a receiver is its ability to reject (adjacent) unwanted signals. Fig. 9.5 shows the attenuation that the receiver offers to signals at frequencies away from its tuned frequency. Selectivity is measured at the end of a sensitivity test with same conditions as for sensitivity, except that now the frequency of the generator is varied to either side of the tuned frequency. The output of the receiver falls as the input frequency goes away from the tuned frequency. Thus the input voltage must be increased until the output is the same as it was at tuned frequency. The ratio of the voltage required for off resonance frequency to the voltage required when the generator is tuned to the receiver's frequency is calculated at a number of points and then plotted in decibels to give a typical selectivity curve. In the curve of Fig. 9.5 you will find that at 10 kHz below the receiver tuned frequency, an input signal would have to be 20 dB greater than the resonant signal to result into the same amplitude. You will appreciate that the selectivity determines the *adjacent-channel rejection* of a receiver.

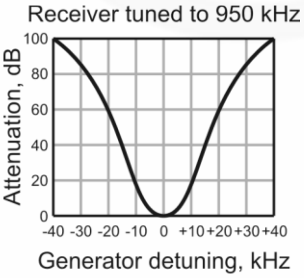

Fig. 9.5: Typical selectivity curve

There is one more characteristic of a receiver, which is called **fidelity**. It characterises the variation of output over the range of modulating frequency (typically 20-20kHz for audio signals). It is expected that over this width of frequency band, the output is linear. When very sharp selectivity is applied, the fidelity gets affected.

Image Frequency and Its Rejection

In a standard superheterodyne broadcast receiver the local oscillator frequency ω_o is kept higher than the incoming signal frequency, which is always equal to the signal frequency plus the intermediate frequency. Thus $\omega_o = \omega_s + \omega_i$ i.e. $\omega_s = \omega_o - \omega_i$, no matter what the signal frequency may be. When ω_o and ω_s are mixed, the difference frequency, which is one of the by-products, is equal to ω_i. As such, it is the only one passed and amplified by the IF stage.

If a frequency $\omega_x = \omega_o + \omega_i$, that is, $\omega_x = \omega_s + 2\omega_i$ manages to reach the mixer then this frequency will also produce ω_i when mixed with ω_o. As an effect, this spurious intermediate-frequency signal will also be amplified by the IF stage and will therefore cause interference. This has the effect of two stations being received simultaneously and is naturally undesirable. The term ω_x is called the **image frequency** and is defined as the signal frequency plus twice the intermediate frequency i.e. we have

$$\omega_x = \omega_s + 2\omega_i \tag{9.1}$$

or

$$f_x = f_s + 2f_i \tag{9.1a}$$

The rejection of an image frequency by a single-tuned circuit, i.e. the ratio of the gain at the signal frequency to the gain at the image frequency, is given by

$$\alpha = \sqrt{1 + Q^2\rho^2} \tag{9.2}$$

where

$$\rho = \frac{\omega_x}{\omega_s} - \frac{\omega_s}{\omega_x}; \tag{9.3}$$

and Q is the loaded quality factor of the tuned circuit.

If the receiver has an RF stage, then there are two tuned circuits, both tuned to ω_s. In such case, the rejection of each will be calculated by the same formula, and the total rejection will be the product of the two. Remember here that, whatever arithmetic applies to gain calculations also applies for rejection.

The image frequency rejection depends on the front-end selectivity of the receiver and must be achieved before the IF stage. Once the spurious frequency enters the first IF amplifier, it becomes impossible to remove it from the useful signal.

SAQ 2

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In a superheterodyne receiver having no RF amplifier, the loaded Q of the antenna coupling circuit (at the input to the mixer) is 100. If the intermediate frequency is 455 kHz, calculate (a) the image frequency and its rejection ratio at 1000 kHz, and (b) the image frequency and its rejection ratio at 25 MHz.

While solving the above SAQ, you must have realised that although image frequency rejection need not be a problem for a MW AM broadcast receiver without an RF stage, special precautions must be taken at HF.

Double spotting

This is a well-known phenomenon observed while tuning a radio set at short wave (SW) frequency. In this case the receiver picks up the same short wave station at two nearby points on the receiver dial. It is caused by poor front-end selectivity, i.e. inadequate image frequency rejection. Here, the front end of the receiver does not select different adjacent signals very well, but, of course, the IF stage takes care of eliminating almost all of them. It is obvious that the precise tuning of the local oscillator determines, which signal will be amplified by the IF stage.

Consider a receiver at HF, having an IF of 455 kHz. If there is a strong station at (say) 15 MHz, the receiver will naturally pick it up. Note that, when it does, the local oscillator frequency will be 15.455 MHz. However, the receiver will also pick up this strong station when it (the receiver) is tuned to 14.090 MHz. When the receiver is tuned to the second frequency, its local oscillator will be adjusted to 14.545 MHz. Since this is exactly 455 kHz below the frequency of the strong station, the two signals will produce 455 kHz when they are mixed, and of course the IF amplifier will not reject this signal. If there had been an RF amplifier in this receiver, the 15 MHz signal would be rejected before reaching the mixer, but in the absence of RF amplifier the 15 MHz signal cannot be adequately rejected when the receiver is tuned to 14.09 MHz.

Double spotting is harmful because a weak station may be masked by reception of a nearby strong station at the spurious point on the dial.

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SAQ 3

How can double spotting be used to calculate the intermediate frequency of an unknown receiver?

As expected, an improvement in image-frequency rejection will produce a corresponding reduction in double spotting.

9.3.2 Mixer

The **mixer** or **frequency changer**, also referred to as **beat frequency generator** is a circuit in the superheterodyne receiver which converts the input RF signal into intermediate frequency (IF) signal.

Since a new frequency is to be generated, a mixer has to contain a circuit component (device) which is working in the non-linear range of its operation. The most common choice of the devices are diodes, bipolar junction transistors (BJT) or field effect transistors (FET). Fig. 9.6 shows an input-output (or transfer) curve of a typical device.

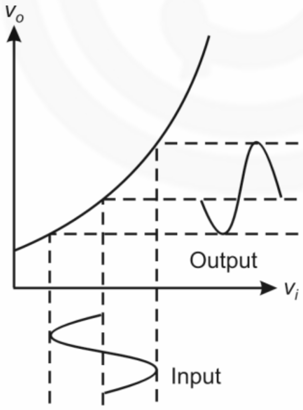

Fig. 9.6: Transfer curve of a non-linear device

The input-output relationship can be expressed as a linear equation if the signal swing is over a small part of the curve and we can write

$$v_o = A\,v_i \qquad (9.4)$$

However, when the signal swing is large the output can be expressed as power series:

$$v_o = Av_i + Bv_i^2 + Cv_i^3 + Dv_i^4 + ... \qquad (9.5)$$

In a mixer, we add the signal coming from the RF stage with the signal generated by the local oscillator (LO). The frequency of LO is varied in unison with the input RF tuning such that the difference between the two frequencies is always equal to IF. The amplitude of the incoming RF signal is quite small (of micro- or nano-volt level) and may not make the device operate in the non-linear range, however, if the LO output has enough amplitude, it provides a large swing in the input voltage to take the device into non-linear operating range.

It is advantageous to use FET as a mixer device since it can operate in the entire range from cut off to saturation with a square law relationship between the input and output. Hence we can rewrite Eq. (9.5) as

$$v_o = Av_i + Bv_i^2 \qquad (9.6)$$

in case of FET.

When two sine waves (RF input and LO output) are added, the resultant v_i is

$$v_i = v_x + v_y = V_x \sin \omega_x t + V_y \sin \omega_y t \qquad (9.7)$$

Hence the output will be of the form:

$$v_o = A\,(v_x + v_y) + B\,(v_x + v_y)^2 \qquad (9.8)$$

Simplifying, we obtain.

$$v_o = Av_x + Bv_x^2 + Av_y + Bv_y^2 + 2Bv_x v_y \qquad (9.9)$$

You will recollect that this is the same process of creating side bands in the amplitude modulation.

As we discussed in Sec. 1.4.1 (Eq. (1.9)), the last term in the above expression results into sum-difference of frequencies ω_x and ω_y.

$$2Bv_x\,v_y = B\,V_x\,V_y \cos\,(\omega_x - \omega_y)t - B\,V_x\,V_y \cos\,(\omega_x + \omega_y)t. \qquad (9.10)$$

Since the difference between the LO frequency and input RF tuned frequency is equal to IF, the term $(\omega_x - \omega_y)$ corresponds to IF. The radian frequency $(\omega_x - \omega_y)$ is equivalent to a cycle frequency of $(f_x - f_y)$. By tuning the IF amplifier to $(f_x - f_y)$, the other frequency components in Eq. (9.10) can be eliminated and only the signal corresponding to IF is amplified.

A typical mixer circuit using *n*-*p*-*n* transistor is shown in Fig. 9.7. Here, the LO output is connected to the base of the transistor and the RF signal to the emitter. The *LC* circuit (*C*-primary of T_1) is tuned at IF to give output corresponding to ($f_x - f_y$).

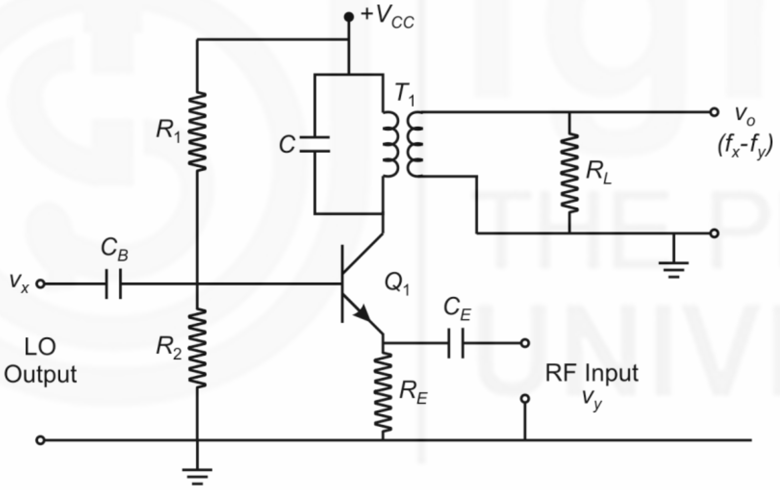

Fig. 9.7: Transistor mixer

A front end of a typical superheterodyne AM receiver is shown in Fig. 9.8.

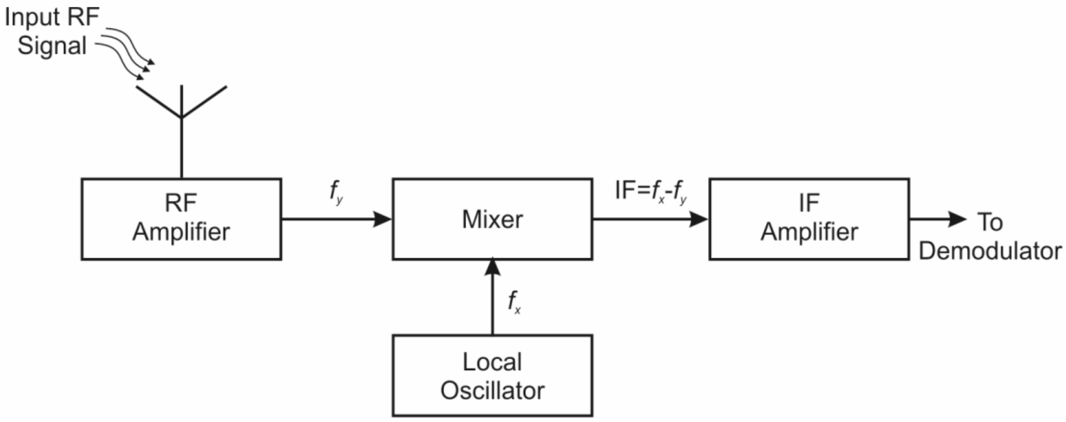

Fig. 9.8: Front end of AM receiver

A typical circuit of LO and mixer is shown in Fig. 9.9. Here, one device acts as a mixer while the other supplies the necessary oscillations. In this case, Q_1, the FET, is the mixer whose gate is connected to the output of Q_2, the bipolar transistor Hartley oscillator. As discussed earlier, the FET is well suited as a mixer because of the square-law characteristic of its drain current. The RF input is given to the source of Q_1.

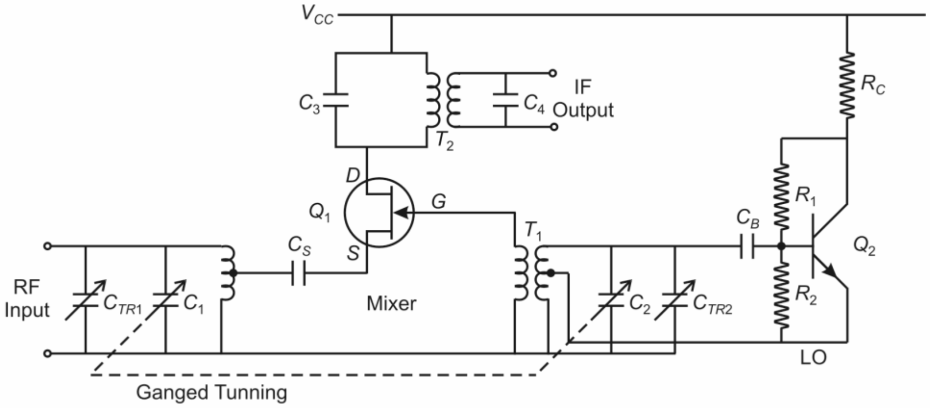

Fig. 9.9: FET mixer and transistorised local oscillator

You must have noticed the ganging together of the tuning capacitors across the mixer and oscillator coils. Both these circuits are provided in practice with trimmer capacitors (C_{Tr}) across them for fine adjustment. Note further that the output is taken through a double-tuned transformer (the first IF transformer) in the drain of the mixer and fed to the IF amplifier. This arrangement is most common at higher frequencies.

9.3.3 Intermediate Frequency (IF) Section

Choice of IF

The choice of intermediate frequency (IF) of a receiving system is usually a compromise, since there are reasons why it should be neither low nor high, nor in a certain range between the two. The following are the major factors influencing the choice of the intermediate frequency in any particular system.

- If the intermediate frequency is too high, poor selectivity and poor adjacent-channel rejection result unless sharp cut-off filters are used in the IF stages.
- As the intermediate frequency is lowered, image-frequency rejection becomes poorer. From Eq. (9.3) we can conclude that the rejection improves as the ratio of image frequency to signal frequency is increased, and this naturally requires a high IF.
- A very low intermediate frequency can make the selectivity too sharp, cutting off the sidebands. This problem arises because the Q must be low when the IF is low, and hence the gain per stage is low.
- If the IF is very low, the frequency stability of the local oscillator must be made correspondingly higher because any frequency drift is now a larger proportion of the low IF than of a high IF.
- The intermediate frequency must not fall within the tuning range of the receiver, or else instability will occur and heterodyne whistles will be heard, making it impossible to tune to the frequency band immediately adjacent to the intermediate frequency.

In all our discussion so far we consider 455 kHz as IF. This is the most commonly used IF for AM receivers. However, there are some other standard values of IF which emerged from above mentioned criteria for different applications. These are given in Table 9.2.

Table 9.2: IF frequencies for various application

S.No.	Application	Frequency of Operation	I.F.
1.	AM medium wave Radio	535-1640 kHz	455 kHz
2.	FM Radio	88-108 MHz	10.7 MHz
3.	TV Receiver	VHF: 54-223 MHz UHF: 470-940 MHz	26 MHz, 46 MHz
4.	Microwave and Radar Receiver	1-10 GHz	30 MHz, 60 MHz, 70 MHz
5.	Cellular Communication	450-2000 MHz	70 MHz, 140 MHz, 240 MHz

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SAQ 4

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What should be the rage of LO to operate in an FM radio receiver tuneable to 88 MHz -108 MHz band?

By and large, services covering a wide frequency range have IFs somewhat below the lowest receiving frequency, whereas other services, especially fixed-frequency microwave, may use intermediate frequencies as much as 40 times lower than the receiving frequency.

IF Amplifier

The IF signal coming from the output of a mixer is actually a frequency down converted AM signal. Since the intermediate frequency is quite lower than the RF signal, designing of IF amplifiers is much simpler than the RF amplifiers used in TRF receivers. A typical stage of IF amplifier is shown in Fig. 9.10. There can be multiple such stages to achieve high gain.

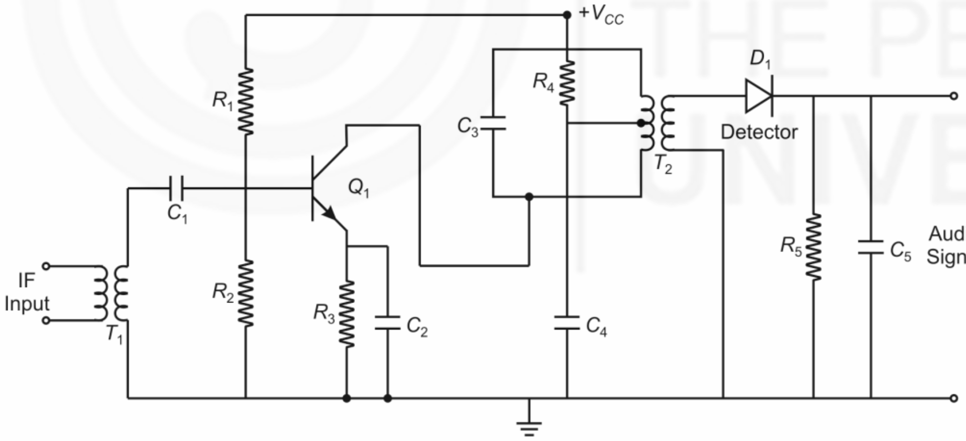


Fig. 9.10: IF amplifier with diode detector

You must have noticed a diode circuit at the output of this amplifier. This is the diode envelop detector for AM detection. You have learnt about the AM detectors in Unit 5 of this course. The detected signal is the audio signal, which is the actual intelligence. This signal is passed through audio amplifier stage to boost the power of voice signal before feeding it to the speaker. These amplifiers have to be good quality, less distortion types. Class *A* or *AB* amplifiers are preferred choices.

9.3.4 Automatic Gain Control (AGC)

The AM waves travel through atmospheric turbulences while reaching the receiver. Some transmitters have high power or they are nearer to the receiver than the other transmitters, and hence the signals received from them are stronger. Also atmospheric fade effects keep on altering the strength of the signal received from the same transmitter. If these variations in signal strength are faithfully reproduced at the output of the receiver, the audio signal we hear will have constant fluctuations and the listening experience will not be a pleasure. In order to compensate for the amplitude variations in the received signal, a circuit called **automatic gain control** (AGC) is used. In case of audio signals specifically, these circuits are referred to as **automatic volume control** (AVC).

In Fig. 9.11 we show a typical AGC circuit using a junction field effect transistor (JFET) Q_1 used as a voltage controlled resistance. For small-signal operation with drain voltage near zero, the JFET operates in the Ohmic region and offers resistance of $r_{ds_{(on)}}$ to ac signal. This resistance can be controlled by the gate voltage. More negative the gate-source voltage V_{GS} is, larger $r_{ds_{(on)}}$ becomes. For a typical JFET, the value of $r_{ds_{(on)}}$ may vary from 100 Ω to more than 10 MΩ. In our circuit we have chosen $R_3 = 100$ kΩ. R_3 and $r_{ds_{(on)}}$ form a voltage divider and depending on the value of $r_{ds_{(on)}}$ feed 0.001 v_{in} to v_{in} at the input P of amplifier A. Therefore the input of the amplifier can vary over a 60 dB range.

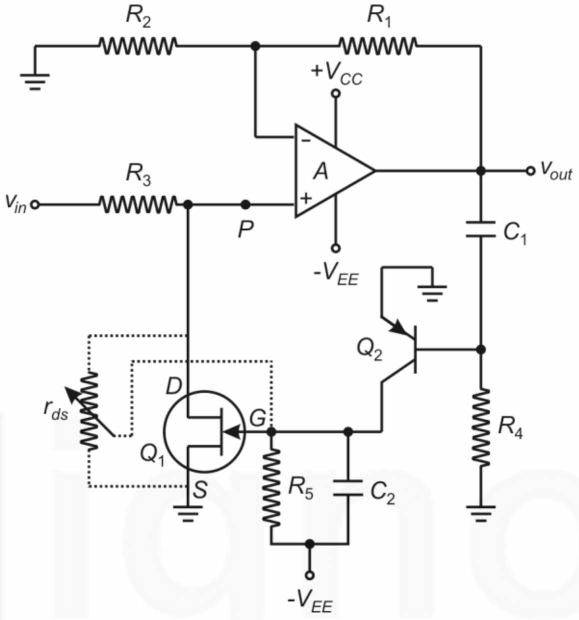


Fig. 9.11: Automatic Gain Control circuit using FET

The output voltage of the amplifier is given to the base of bipolar junction transistor Q_2 through coupling capacitor C_1. Since the emitter of *p-n-p* transistor Q_2 is at ground potential, it remains cut off, till the output voltage is not less than –0.7V. In such a condition C_2 remains uncharged and whole of $-V_{EE}$ appears through R_5 at the gate of Q_1. This cuts off the JFET. Under cut off, r_{ds} is maximum and entire v_{in} appears at the input of the amplifier A. Now, if the amplitude of v_{in} (received signal) increases suddenly, the amplifier amplifies it and large output voltage appears at the base of Q_2. A negative swing below –0.7V switches Q_2 *on* and C_2 charges through this conducting transistor. Due to the voltage on C_2, the gate voltage of Q_1 raises above the quiescent level of $-V_{EE}$. This reduces $r_{ds_{(on)}}$ and the output of voltage divider R_3- $r_{ds_{(on)}}$ at P. It means that the voltage at amplifier input reduces, effectively reducing output voltage.

In this way, the increase in the received signal is used to effectively reduce the total gain of the circuit and give stable output at AGC circuit. You must remember here that the control action taking place is valid over only a limited amplitude range of input signal. If there is a drastic change in the received signal amplitude, it will drive the amplifier into saturation and the circuit will not operate in its linear mode of operation. In such case there will not be any AGC action.

SAQ 5

Spend 2 Min.

Choose the proper statement:

In a radio receiver with AGC

i) the sensitivity is reduced at higher signal levels;
ii) the RF stage gain is normally controlled by the AGC;
iii) frequency selectivity is dependent on AGC performance;
iv) the AGC circuit operates device in non-linear region.

After understanding the details of various aspects of AM receivers, now we briefly discuss the AM transmitters.

9.4 AM TRANSMITTER

A block diagram of basic AM transmitter is shown in Fig. 9.12a. This simple type of transmitter is used in some special dedicated communication systems like wireless set ups but is not suitable for commercial broadcast transmission. The main limitations of these transmitters are i) limited output power and ii) poor frequency stability. It is not possible to get a great deal to power through an oscillator. Modulating the signal directly on the oscillator brings in instability in the operating frequency of the oscillator. To overcome these difficulties, the transmitter circuit is modified as shown in Fig. 9.12b. In this transmitter an additional stage of power amplifier is added and the modulating point is shifted from oscillator to this amplifier. Now the oscillator is running at a single predetermined fixed frequency and acts as a master oscillator (MO). Due to master oscillator and power amplifier in this system, it is referred to as MOPA transmitter. The power amplifier in this system provides higher power level of radiation. However there are some more modifications done in the sophisticated transmitters to make them more efficient and give better performance. A typical practical broadcast transmitter block diagram is shown in Fig. 9.13.

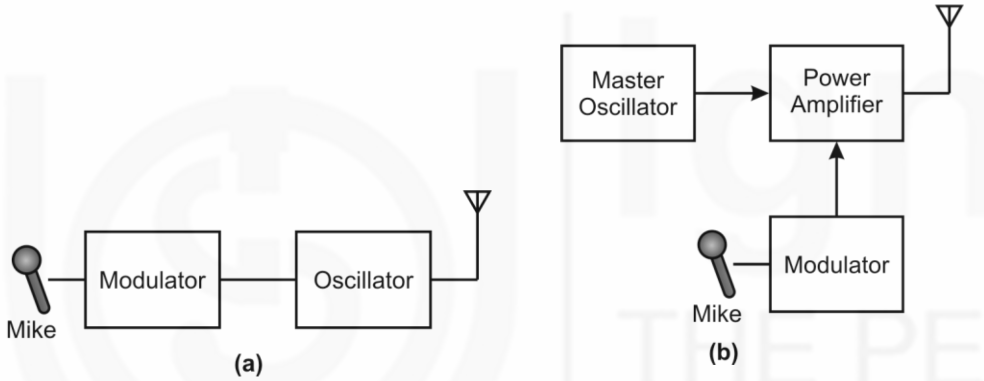


Fig. 9.12: a) Basic AM transmitter and b) MOPA transmitter

In this system a buffer amplifier is introduced between the master oscillator and the remaining circuit. This protects the oscillator from any stray feedback of modulating frequency from further circuit components and enhances the stability of the oscillator frequency. The oscillators used in typical AM broadcast transmitters can be Hartley, Colpitts or crystal controlled oscillators.

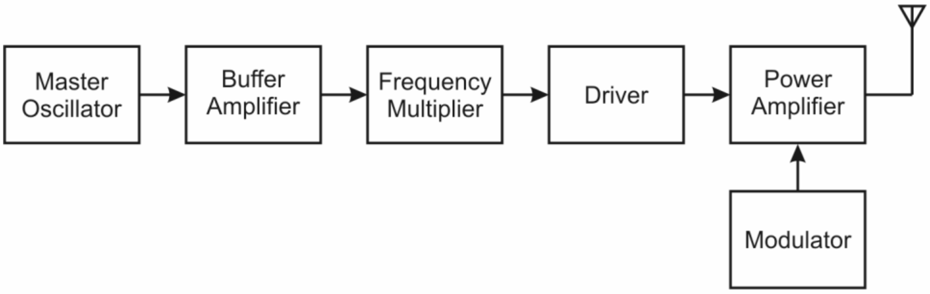


Fig. 9.13: Block diagram of a practical AM transmitter

In some cases the frequency of the oscillator is not adequate for using directly in radiation. In such cases, the oscillator is up converted by using frequency multipliers. In order to enhance the signal amplitude, usually some driver (amplifier) stages are added before letting the signal modulation in the power amplifier. All the amplifiers used here are of Class *C* type, which have advantage of higher power efficiency. You have already learnt about the amplitude modulators using transistorised circuits in Unit 5 of this course.

The last stage of a typical transmitter is a power stage. The output of this amplifier is fed to the radiating antenna for signal transmission. The coupling between the final stage and antenna is critical, so that only the desired frequencies should be transmitted.

For any broadcast transmitter the maximum allowed power that can be radiated is limited by the government regulations. Typically these powers are in the range of 500 W to 50 kW for radio transmitters. The limit assigned to a transmitter has to be observed; otherwise the signal from one transmitter may start interfering with another transmitter signal, which is prohibited.

Now, when there is restriction on maximum transmitter power, the use of this available power has to be carried out for communicating the signal with maximum possible efficiency. It means that the power is to be utilised only for the purpose it is allotted for and any undesired frequencies are not transmitted. Hence it is necessary to filter out any stray frequencies present in the signal before it is transmitted. This can be achieved by using coupling circuits before the antenna. A typical coupling arrangement between the final stage and the antenna is shown in Fig. 9.14.

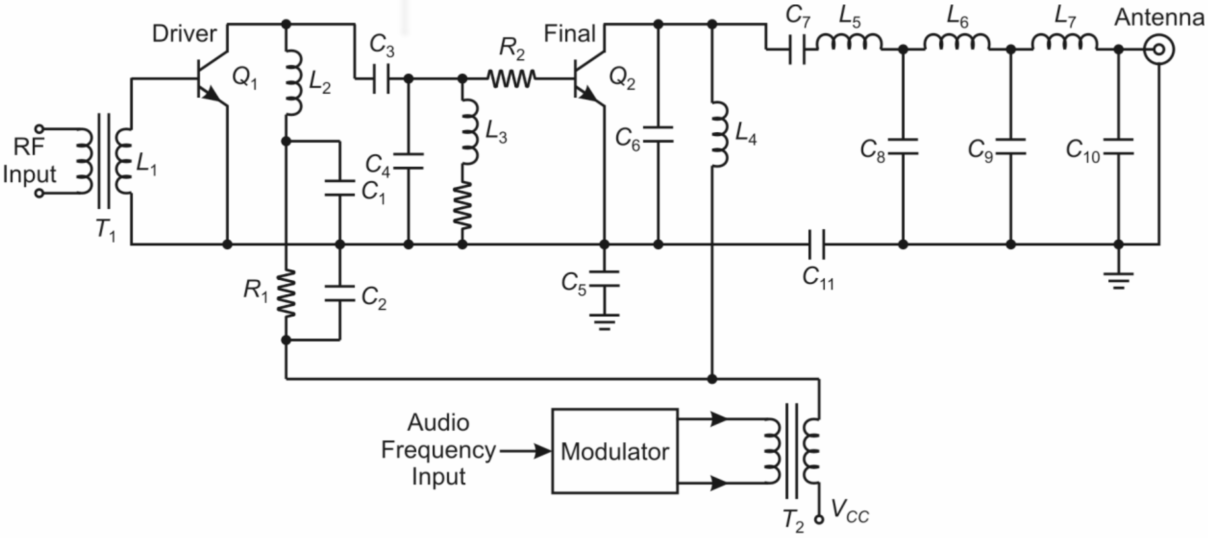


Fig. 9.14: Final stage of a typical commercial broadcast transmitter

The coupling circuit also performs the job of impedance matching between the output impedance of the amplifier and the load, which may be the input impedance by another amplifier or the antenna impedance.

In its function as a power transmitting unit, a coupling circuit is generally expected to *transform* load impedance in order to accomplish a particular purpose. For example, in inter-stage coupling, a typical requirement is the transfer of a specified amount of RF power from the driver to the input of the power output stage. Given the power specification, say P_{out}, the relationship between the RF voltage, V_{RF}, at the output of the driver stage and its (the driver's) load resistance, R_{load}, is

$$P_{out} = \frac{V_{RF}^2}{R_{load}} \qquad (9.11)$$

However, if the driver stage is class *C* amplifier, then, usually the peak-to-peak value of the RF voltage is equal to $2V_{CC}$, or, $V_{RFp} = V_{CC}$. Therefore,

$$V_{RF}\,(\text{effective}) = \frac{V_{CC}}{\sqrt{2}} \qquad (9.12)$$

and

$$V_{RF}^2 = \frac{V_{CC}^2}{2} \qquad (9.13)$$

More precisely, if the collector saturation voltage, $V_{CE(sat)}$, is known, then

$$V_{RF}^2 = \frac{(V_{CC} - V_{CE(sat)})^2}{2} \qquad (9.14)$$

Hence the amount of load resistance that must be presented to the driver stage in order to achieve the desired power level is

$$R_{load} = \frac{(V_{CC} - V_{CE(sat)})^2}{2P_{out}} \qquad (9.15)$$

If the input impedance of the output stage (the driver stage) is not equal to this value of R_{load}, the coupling circuit is designed to transform that impedance to *match* this required value. Please remember here that the match is not necessarily to achieve maximum power transfer, but to achieve the transfer of a specified amount of power.

The coupling circuit between the final stage of a transmitter and the transmitter load-usually an antenna at the end of a transmission line must serve a similar purpose. The circuit is designed to transform the transmission line/antenna load so that it matches the value of an R_{load}, as seen by the output stage, required to produce a specified amount of RF power.

Coupling circuits of many different forms can be devised to accomplish this. An "L" circuit (so-called because of its shape) composed of two reactances of opposite types is a simple and common coupling circuit used to transform the value of a component, or load, connected to its output terminals as shown in Fig. 9.15a. When we redraw the circuit as in Fig. 9.15b it becomes obvious that the overall circuit, including R_L, is parallel *LC* circuit. Let us assume that the resistance of inductance *L* is negligible compared to R_L. This is a reasonable assumption for most practical cases. Then, at f_o, the frequency that makes the circuit anti-resonant, the network will appear to be a pure resistance, R_{in}. The value of R_{in} is that of R_L transformed to its parallel equivalent (refer to Fig. 9.15c).

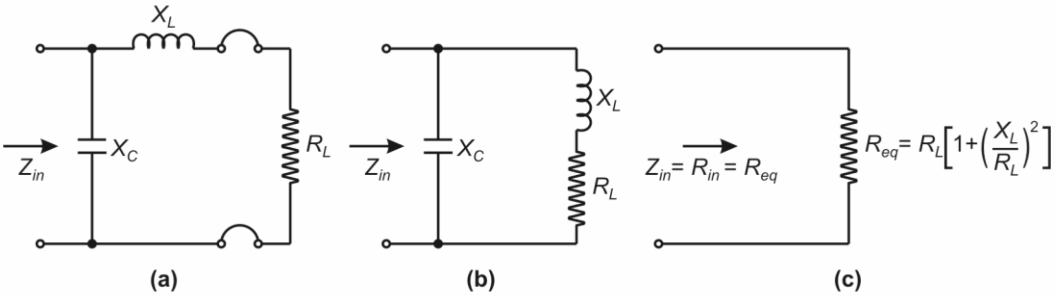


Fig. 9.15: a) Up transformation of impedance: L network; b) simplified L network; and c) equivalent impedance at resonance

This configuration makes the circuit a low-pass filter that can attenuate the second and higher-order harmonics of the carrier and other spurious signals. Formulae for calculating values for X_L and X_C to obtain desired R_{in}/R_L transformation ratio are:

If $n = R_{in} / R_L$ with $n > 1$, then

$$X_L = \sqrt{R_{in}R_L - R_L^2} = \frac{R_{in}}{n}\sqrt{n-1} \tag{9.16}$$

and

$$X_C = \frac{R_{in}R_L}{X_L} = \frac{R_{in}}{\sqrt{n-1}}\,, \tag{9.17}$$

with $$Q = \sqrt{\frac{R_{in}}{R_L} - 1} = \sqrt{n-1}\,. \tag{9.18}$$

This L network transforms the load resistance *up*, i.e. for a potential source (such as a transmitter output stage), a load resistance appears larger in value than it actually is, when seen *through* the network.

A *down* transformation can be obtained with only a slight change in the circuit. Connect X_C in parallel with R_L, instead of at the input to the network as shown in Fig. 9.16a. The X_C - R_L parallel circuit can now be transformed to an equivalent series circuit (see Fig. 9.16b). This circuit forms a series resonance. Thus, at f_o, the frequency at which the circuit is resonant, R_{in} for the network is equal to the transformed value of R_L ($R_{L\,series}$), if we neglect the resistance of L.

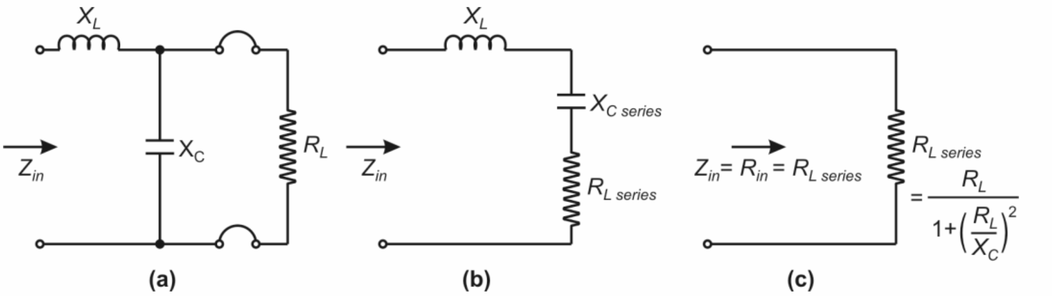


Fig. 9.16: L network for *down* transformation: a) L network for transforming an R_L to lower value; b) L network transformed to equivalent series circuit; and c) equivalent L network at resonance

The formulae for designing impedance down transforming circuit of this type are as follows:

Let $n = R_L / R_{in}$, with $n > 1$; then

$$X_L = \sqrt{R_{in} R_L - R_{in}^2} = R_{in} \sqrt{n-1}\,; \qquad (9.19)$$

and $$X_C = \frac{R_{in} R_L}{X_L} = \frac{n}{\sqrt{n-1}} R_{in}\,, \qquad (9.20)$$

with $$Q = \sqrt{\frac{R_L}{R_{in}} - 1} = \sqrt{n-1}\,. \qquad (9.21)$$

As can be seen from the equation for Q of the L network, the Q cannot be controlled by the design of the network ($Q = \sqrt{n-1}$), since it is a function of the transformation ratio only. This can be a distinct disadvantage. A simple modification of the circuit allows Q to be chosen by the designer. The modification is in the form of a capacitive reactance X_{C_1} connected in series with the inductance of the circuit as shown in Fig. 9.17. Design equations for this circuit then become:

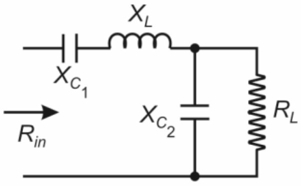


Fig. 9.17: L network modified to permit choice of *Q*

For a desired value of Q,

$$X_L = Q\,R_{in} \qquad (9.22)$$

$$X_{C_1} = X_L - R_{in}\sqrt{n-1} \qquad (9.23)$$

$$X_{C_2} = \frac{n}{\sqrt{n-1}} R_{in} \qquad (9.24)$$

The attenuation of harmonic and/or other spurious signals is an essential requirement of the output circuits of a transmitter. Coupling circuits used to transform impedances invariably provide significant attenuation of this type. It is generally true that each separate reactive component in a low-pass filter provides attenuation of 6 dB octave^{-1}.

Hence, if a coupling network contains two reactances, as with the L network, the attenuation is 12 dB octave^{-1}. When additional attenuation is required additional reactances must be incorporated in the network between the output stage and the antenna.

A signal power level is attenuated by approximately 6 dB when the power is cut to one-fourth of the input value.

Two frequencies are an octave apart when the higher frequency is twice that of the lower one.

SAQ 6

Spend 4 Min.

A transmitter power output stage is to drive a 50 Ω antenna load. The transmitter supply voltage, $V_{CC} = 12.5$ V. It is desired to match the stage to the antenna load so that P_{out}(RF) = 2.5 W. The operating frequency is 27.5 MHz.

a) Determine R_{in}, the equivalent load that the transistor must see in order to produce the desired amount of power assuming class C operation.

b) Design an L network that will provide the required match.

After discussing the amplitude modulation let us now discuss the frequency modulation used in commercial broadcasting.

9.5 FM COMMUNICATION

In recent days, the radio transmission is slowly getting converted from AM to FM mode. The FM transmission has some distinct advantages over the AM transmission and provides better sound quality. In India also we have started receiving FM radio signals in the last decade and they are becoming more and more popular day-by-day. The Government has begun to licence private broadcasters on FM bands and hence there are more players in the field of FM broadcast now. Even the educational channel of IGNOU (Gyan-Vani) has started FM broadcast transmission and you can receive it from the nearest transmitter.

The FM transmitter essentially contains an oscillator, where frequency is varied depending on the input audio signal. After enhancing the power of the frequency modulated signal (output of oscillator), it is fed to the transmission antenna.

There are some specific precautions taken like frequency stabilisation mechanism for the base frequency of the oscillator. These circuits are also termed as automatic frequency control (AFC) circuits. These are necessary because each FM transmitter is assigned a particular band of frequency and not allowed to drift away from it, since that may interfere with neighbouring frequency band signals.

In a communication system, it is observed that the effect of noise is more significant on the higher modulating frequencies than on the lower ones. Thus, if the signals corresponding to higher frequencies were artificially boosted at the transmitter and correspondingly attenuated at the receiver, it improves the noise immunity. This boosting of the higher modulating frequencies is termed as *pre-emphasis* and the compensation at the receiver is called *de-emphasis*.

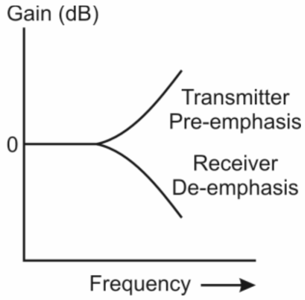


Fig. 9.18: Frequency response of pre-and de-emphasis circuits

These circuits are typically simple *R-C* filters. The high pass filter has lesser attenuation (higher gain) at higher frequencies as seen in the upper curve of Fig. 9.18. Hence it is used at the transmitter end as a pre-emphasis circuit. The lower curve in the figure corresponds to a low pass filter. Since it has higher attenuation at higher frequencies, it can be used as a de-emphasis circuit at the receiver end.

Spend
3 *Min.*

SAQ 7

Draw the circuits used for pre-and de-emphasis in the FM communication system.

We have extensively discussed about the transmitter used for FM communication in Sec. 5.7. Now we discuss some salient features of a FM receiver system.

9.5.1 FM Receivers

A number of blocks of the FM receiver correspond exactly to those of other receivers already discussed. For example, the same criteria apply in the selection of the intermediate frequency, and IF amplifiers are basically similar.

A block diagram of a typical FM receiver is shown in Fig. 9.19.

This is a superheterodyne receiver, which is quite similar to an AM receiver. The basic differences between the two are:

- operating frequency in FM are much higher;
- need for amplitude limiting and de-emphasis in FM; and
- different methods of demodulation.

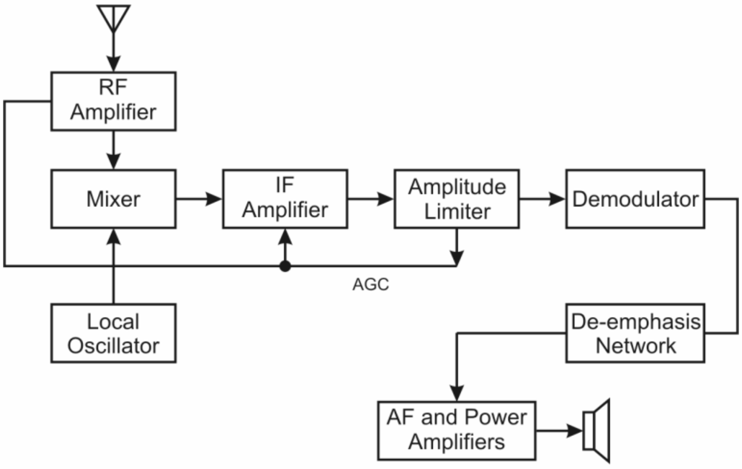


Fig. 9.19: FM receiver block diagram

You have already learnt the demodulation schemes for FM signals in Sec 5.8.

In order to make full use of the advantages offered by FM, an amplitude limiter must precede a demodulator. Since any amplitude changes in the signal fed to the FM demodulator are spurious, they must be removed if distortion is to be avoided. This point is significant, since most FM demodulators react to amplitude changes as well as frequency changes. You know that the limiter is a form of clipping device, a circuit whose output tends to remain constant despite changes in the input signal. Most limiters behave in this fashion, provided that the input voltage remains within a certain range. In Unit 5, while solving SAQ 7, you have drawn one such circuit.

9.5.2 Stereo FM System

The word *stereo* comes from a Greek word meaning solid or firm, in the sense of three-dimensional. The experience of listening to music reproduced on a *stereophonic sound system* consisting of multiple speakers is perceived as being more real or life-like than listening to the same music reproduced by a monophonic sound system. At the very minimum, a stereo system includes two speakers *with separation*.

Separation implies two elements: a *physical separation* of the speakers, from the listener's point of view; and a *separation of the signals* driving the speakers. The physical distance between the speakers must be significant compared to the distance of the listener from the speakers. If the listener is, say 5 m from the speakers, the speakers should be kept at least a meter apart.

To create the perception of realism the speakers must be driven by signals picked up using separate microphones at physically separated positions at the location where the music (or sound signal) is actually produced. If the stereo reproduction scheme is part of a radio broadcast system, the system must be capable of transmitting the two separate signals simultaneously. The two signals are universally designated the *L* (for left side) and *R* (for right side) signals.

The broadcast of signals for stereo reproduction has been a part of the FM broadcast system since the 1950s. A few AM stations are just beginning to broadcast signals for stereo reproduction. In each instance, FM and AM, the regulations make it mandatory that the stereo broadcasts should be receivable and reproducible by existing monophonic receivers. It means that, the introduction of stereo broadcasting is required to be *compatible* with the systems, as they existed previously. Though this is a simple, reasonable and logical requirement, technically it influenced significantly the choice of the form of signals to be broadcast, and thereby, the design of the equipment used to generate, transmit, and receive the signals.

In the FM system, a stereo broadcast consists of the multiplex transmission of two information signals and a pilot carrier. However, the two signals are not just simple *L* signal and an *R* signal. Such a broadcast would not be compatible with a monophonic receiver. The two signals are (*L*+*R*) component and (*L*–*R*) component. The (*L*+*R*) component is receivable in a normal way by a monophonic receiver. The reproduction of that signal by the monophonic receiver is indistinguishable from a monophonic broadcast. On the other hand, a receiver designed to receive and reproduce stereo broadcasts incorporates a special section that decodes the two signals (*L*+*R*) and (*L*–*R*); and recombines them so as to produce separate *L* and *R* signals. A stereo receiver will have a two-channel audio section (two complete audio amplifiers) which processes the signals and drives two separate speaker systems.

A block diagram of the stereo section of an FM transmitter is shown in Fig. 9.20. The process starts with the conversion of sound into electrical signals by means of two separate microphones, a left mike and a right mike. The signals from these two are processed by individual audio channels (amplifiers). The (*L*+*R*) signal is produced by adding these separate *L* and *R* signals, in phase, in an appropriate linear circuit called an *adder*. The FM transmitter is frequency modulated by this signal. The result, at this point, is indistinguishable from a monophonic transmission; it is the mono-compatible component of a stereo broadcast.

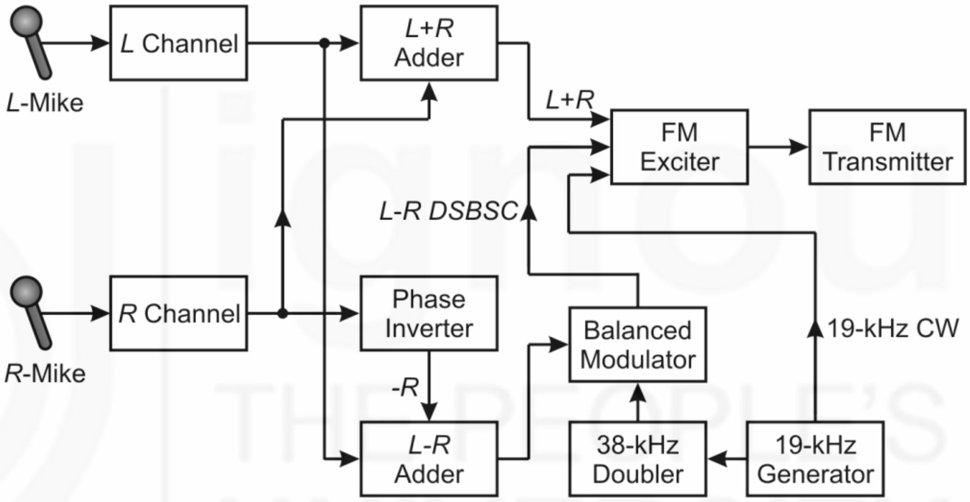


Fig. 9.20: Stereo section of FM transmitter

The *R* signal is also fed to a phase inverter which shifts its phase by 180°, producing a –*R* signal, which is combined with the *L* signal in a second adder to produce an *L*–*R* signal.

Simultaneously, 19 kHz signal, called a *pilot carrier* is continuously produced by a separate oscillator. It is called a pilot carrier because it is transmitted at relatively low amplitude and is not modulated directly. It is transmitted to provide a carrier at the receiver for the detection of the (*L*–*R*) signal. The frequency of the 19 kHz signal is doubled in the 38 kHz doubler and then fed to the *balanced modulator*. As you know, this modulator produces amplitude modulation of a carrier but suppresses the carrier in its output. It produces two AM-type sidebands with the carrier suppressed. The FM transmitter is modulated by 19 kHz pilot carrier and the 38-kHz double-sideband, suppressed carrier (DSBSC) signal.

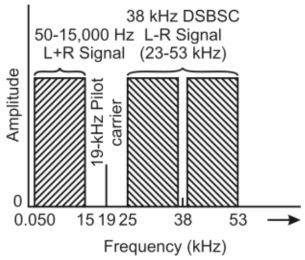


Fig. 9.21: Frequency spectrum of signals in FM stereo

The frequency spectrum of the signals modulating the stereo FM carrier is shown in Fig. 9.21. Remember that the 50 Hz to 15 kHz (*L*+*R*) signal is equivalent to a standard monophonic FM broadcast signal. The 19-kHz pilot carrier is transmitted as an aid in detecting the 38 kHz DSBSC (*L*–*R*) signal at the receiver. The (*L*–*R*) signal will be recombined with the (*L*+*R*) signal in the receiver to produce the separate *L* and

R signals. The concept of frequency division multiplexing (FDM) is incorporated into this process and there are three distinct groupings of frequencies: the 0 to 15 kHz band, the 19 kHz pilot carrier, and the 38 kHz DSBSC band. The FM carrier is modulated simultaneously by all three signals. Each produces its set of side frequencies; each can be separated from the other at the receiver by appropriate filtering.

Let us now discuss the working of the stereo decoding section of a receiver.

A typical scheme for decoding FM stereo broadcasts is shown in Fig. 9.22. FM stereo and mono receivers are identical up to the output of the FM detector. In a stereo receiver, the output of the detector is passed to the stereo decoder. The first function the decoder performs is to separate the three signal components. It performs this task by means of appropriate, frequency-selective filters: a low pass filter is used to separate the 0 to 15 kHz signal from the others; a narrow band filter selects out the 19 kHz pilot carrier; and a 23 to 53 kHz band-pass filter provides for the separation of the ($L-R$) signal.

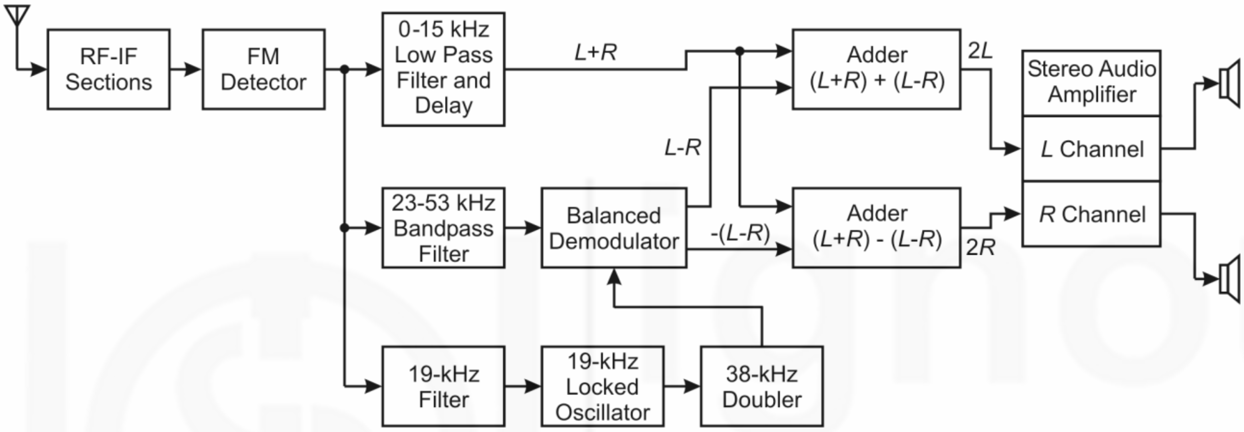


Fig. 9.22: Stereophonic FM receiver

The 38 kHz DSBSC signal requires a more involved demodulation process than the ($L-R$) signal. This processing takes up a very small but definite amount of time. Since the two signals must be recombined in the exact time phase with which they were created, the ($L+R$) filter incorporates a time delay to match the delay of the ($L-R$) processing.

The decoder section amplifies the 19 kHz pilot carrier and uses it to lock in a 19 kHz local oscillator. The 19 kHz local oscillator, when locked in, is in precise synchronism with the 19 kHz carrier at the transmitter. The frequency of the output of this oscillator is doubled and fed to a balanced demodulator, together with the ($L-R$) signal. By this means the ($L-R$) signal is recovered.

The recovered ($L-R$) signal is recombined with the ($L+R$) signal in two separate forms. It is combined in an adder in its normal form to produce a 2L signal: $(L+R) + (L-R) = 2L$. It is inverted to $-(L-R)$ and combined with ($L+R$) in a second adder to produce 2R signal: $(L+R) - (L-R) = 2R$.

These separate *L* and *R* signals are processed further in two separate audio amplifier channels and are used to drive separate *L* and *R* speakers, or speaker systems.

After studying about the FM broadcast in details, let us take account of another modern variant of audio communication, i.e. *satellite radio*.

9.6 SATELLITE RADIO

So far we have discussed about the audio wave communication using terrestrial transmission. Both AM and FM (including stereo) radio stations use terrestrial antennas to broadcast their signals. However, with the advent of technology, the radio broadcast has also slowly started using satellite communication as the medium of broadcasting. In this case, the signals are sent and received in digital format. Due to all the advantages of digital communication (like noise immunity, higher bandwidth and easier implementation of multiplexing), the quality of this broadcast is much better than analog transmission. The satellite radios are therefore commonly referred to as *digital radios.*

The audio signals are first converted into digital from by the techniques you have already learnt in earlier units. Since the frequency bands of satellite communication are typically in the microwave range, these digitised audio signals are modulated over the satellite uplink frequencies generally using frequency division multiplexing (FDM) techniques similar to the ones used to step-wise enhancement of carrier frequency in telephone systems.

The basic advantage of satellite radio is the wide area covered by the footprint of a satellite. This allows reaching of signals to remote areas, without installing any terrestrial transmitters.

A typical satellite radio receiver consists of an antenna for receiving signals from the satellite; a demodulator; a digital decoder that decodes the signal and digital to analog converter. The signal is amplified using certain amplifier circuits before feeding the signal to the speakers.

At present, the satellite radio systems are operated by private service providers and the user has to subscribe to these services. The receiver sets used by each service provider are proprietary items useful only for that particular provider's broadcast. The main hindrance in popularising satellite radio is the prohibitive cost of managing these systems. As you will learn in later units, the satellites are very costly, due to limited payload capacity and power constraints; the numbers of satellite channels available are restricted and hence cost more.

However, in India, there are efforts to regularise the access to satellite radio, through authorities like Telecom Regulatory Authority of India (TRAI). The following frequency bands are available to India for providing Satellite Radio Services:

'L' Band : 1452-1492 MHz
'S' Band : 2310-2360 MHz and
2535-2655 MHz.

Hence as you will observe, India is also gearing up for the new trends in audio communication and soon we can hope to use the new generation radio widely.

Let us now summarise the points we covered in this unit.

9.7 SUMMARY

- The audio receivers are usually of two types: Tuned Radio frequency (TRF) receiver and Superheterodyne receiver.
- TRF are not very popular due to need of broad band RF amplifier.

- Superheterodyne receivers reduce the incoming RF frequency to intermediate frequency (IF) by mixing a local oscillator frequency of incoming signal.
- IF frequency circuits are stable and low cost.
- AM receiver consists of RF stage, mixer, IF amplifier, demodulator, audio amplifier and speaker.
- RF stage provides sensitivity, frequency selectivity and improves image frequency rejection.
- Mixer operates in the non-linear operating region of the device. FET is an ideal mixer due to its square law characteristics.
- IF in AM receivers is 455 kHz while in FM radio receiver it is 10.7 MHz.
- Automatic Gain Control varies the gain of the circuit such that fluctuations in the incoming signal amplitude get smoothed out and audio amplitude is constant.
- AM transmitter needs to be coupled to the antenna circuit with proper impedance matching, in order to ensure fixed frequency (band) and power transmission; L networks can be used as couplers.
- In stereo FM systems, signal from left and right mike are collected separately and are sent over as $(L+R)$ and $(L-R)$ signals. These are decoded at the receiver using the pilot carrier signal sent along by the transmitter.
- Satellite or digital radio is the next generation of audio communication systems.

9.9 TERMINAL QUESTIONS

Spend 25 *Minutes*

1. When a superheterodyne receiver is tuned to 555 kHz, its local oscillator output frequency is 1010 kHz. What is the image frequency? The receiver is connected to a mixer via a tuned circuit with loaded Q of 40. What is the rejection ratio for the calculated image frequency?
2. Amongst the frequencies to be rejected by a superheterodyne receiver, why is image frequency the most significant? What steps could be taken to improve the image frequency rejection of a receiver?
3. How is frequency tracking achieved in a superheterodyne receiver?
4. Though AGC is a need of communication circuits, it is not preferred in the circuits collecting scientific data. Comment on this statement.
5. Why is it necessary to use proper coupling between the output stage of transmitter and the antenna?
6. In a stereo transmitter, why are the signals from L-mike and R-mike not sent out directly as L and R signals?

9.10 ANSWERS AND SOLUTIONS

Self Assessment Questions

1. (ii)

2. (a) $f_x = 1000 + 2 \times 455 = 1910$ kHz

$$\rho = \frac{1910}{1000} - \frac{1000}{1910} = 1.910 - 0.524 = 1.386$$

$$\alpha = \sqrt{1 + 100^2 \times 1.386^2} = \sqrt{1 + 138.6^2} = 138.6$$

This is 42 dB and is considered adequate for domestic receivers in the MF band.

(b) $f_x = 25 + 2 \times 0.455 = 25.91$ MHz

$$\rho = \frac{25.91}{25} - \frac{25}{25.91} = 1.0364 - 0.9649 = 0.0715$$

$$\alpha = \sqrt{1 + 100^2 \times 0.0715^2} = \sqrt{1 + 7.15^2} = 7.22$$

This is less than 20 dB and it is obvious that this rejection will be insufficient for a practical receiver in the HF band.

3. The spurious point on the dial showing double spotting is exactly at double the IF below the correct frequency.

4. The range of LO frequency should be 98.7 to 118.7 MHz.

5. (i)

6. (a) Assuming that $V_{CE(Sat)}$ is negligible,

$$R_{in} = \frac{V_{CC}^2}{2P_{out}} = \frac{(12.5)^2}{2 \times 2.5} = 31.25\Omega$$

(b) Since R_{in} is less than R_L, we select a network of the type shown in Fig. 9.16. Then,

$$n = \frac{R_L}{R_{in}} = \frac{50}{31.25} = 1.6$$

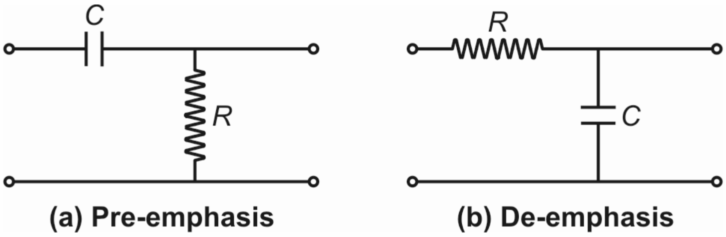
$$X_L = R_{in}\sqrt{n-1} = 31.25\sqrt{1.6-1} = 24.21\Omega$$

$$L = \frac{X_L}{2\pi f} = \frac{24.21}{2\pi \times 27.5(10^6)} = 0.14\mu H$$

$$X_C = \frac{n}{\sqrt{n-1}} R_{in} = \frac{1.6}{\sqrt{1.6-1}}(31.25) = 64.55\Omega$$

$$C = \frac{1}{2\pi f X_c} = \frac{1}{2\pi \times 27.5(10^6)(64.55)} = 89.66\text{pF}$$

7. Please refer to Fig. 9.23.



(a) Pre-emphasis

(b) De-emphasis

Fig. 9.23: a) Pre-emphasis; and b) De-emphasis circuits

Terminal Questions

1. Input frequency = f_s = 555 kHz.
   LO frequency = f_o = 1010 kHz
   $\therefore$ IF = f_i = 1010 - 555 = 455 kHz
   Image frequency $f_x = f_s + 2f_i$ = 555 kHz + 2(455 kHz) = 1465 kHz

$$\text{Rejection ratio } \alpha = \sqrt{1 + Q^2\,\rho^2}$$

$$= \sqrt{1 + (40)^2 \left[\frac{1465}{555} - \frac{555}{1465}\right]^2}$$

$$= \sqrt{1 + (1600)(5.11)} = 90.4 = 19.6\,\text{dB}.$$

2. Please refer Sec. 9.3.1.

3. The LO and input RF tuner are ganged to achieve a constant output from the mixer for different receiver tuned frequencies.

4. AGC circuit provides a gain control, where a constant output level is maintained irrespective of fluctuations in input signal. This is done for a good listening experience. However, in scientific data collection, every small fluctuation in the incoming signal may be a significant from the research point of view. Same is the case with the medical diagnostic techniques. Hence in such cases, the receiver circuit introducing a gain fluctuation (in response to fluctuation in incoming data) may result into wrong recording of data.

5. Please refer to Sec. 9.4.

6. Please refer to Sec. 9.5.2.

Reference Material:

1. *Electronic Communication Systems* by Kennedy, George; (III Edition) (Tata McGraw-Hill)
2. *Electronic and Radio Engineering* by Terman, F.E; (IV Edition) (Tata McGraw-Hill)