TRANSMITTER CIRCUITS

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1.1 INTRODUCTION

Transmitter circuits are responsible for generating the carrier; superimpose the information-carrying signal to be transmitted onto the generated carrier signal; amplifying the modulated carrier signal; band-limiting the modulated signal; couple it to the antenna and radiating it. Therefore, transmitters consists of RF oscillators for generating the carrier signal; modulators for superimposing the information-carrying signal, called modulating signal, onto the carrier; RF power amplifiers for amplifying the modulated carrier signal in order to meet the power requirements of the transmitter; band pass filter to remove any spurious modulator products that fall outside the required band; matching circuitry for ensuring correct impedance matching between the transmitter and the antenna to ensure maximum power transfer. The antenna is used in the transmitter to convert the output of the transmitter into radio waves for transmission.

1.2 FREQUENCY GENERATION AND CONVERSION

1.2.1 Carrier Generation

AM systems usually need a fixed carrier frequency while FM system need a carrier frequency that is variable in accordance with the modulating signal. The latter is usually accomplished by using a voltage tuned oscillator (VTO) or a voltage controlled oscillator (VCO). In order to ensure the stability of carrier frequency generated in those transmitters that need a fixed carrier frequency, a crystal oscillator or crystal oven is often used as the RF oscillator. To isolate the RF oscillator from the effects of the load changes, which can otherwise cause frequency shifts, buffer amplifiers with high input impedance are put between the oscillator and the load to ensure that the oscillator is not loaded.

1.2.2 Frequency Multiplication and Translation

If the frequency generated is not the required one, frequency multiplication and frequency translation can be used to obtain the required frequency. Frequency multiplication involves multiplying the frequency together with its side frequencies by a factor N while frequency translation involves moving the carrier from one value to another.

Frequency multiplication is used for both unmodulated and modulated carrier. It multiplies the frequency of the unmodulated carrier *N*-times to get the required frequency. It can also be used for frequency modulated carrier to increase both its carrier and amount of deviation, thereby increasing the modulation index. This is used in angle modulation to convert from narrow-band FM to wide-band FM. Class C amplifiers are commonly used as frequency multipliers. Their output collector current flows as a series of less than half sine wave pulses and contains components at the input signal frequency and at its harmonics, which are extending to many times of the input frequency. That is,

$$\begin{aligned} i_c &= I_{dc} + i_o + i_{2nd} + i_{3rd} + i_{4th} + \dots \\ &= I_c n_{dc} + I_c n_1 \cos wt + I_c n_2 \cos 2wt + I_c n_3 \cos 3wt + \dots \end{aligned} \qquad (1.1)$$

where I_c = peak value of the collector current. A tuned circuit is connected to the collector of the class C amplifier. This tuned circuit is tuned to be resonant at the required particular harmonic or multiple of the input frequency. For example, if the tuned circuit is tuned to be resonant at the third harmonic of the input frequency, then the input frequency will be tripled. A typical frequency multiplier using a class C amplifier is shown in Figure 1.1 below.

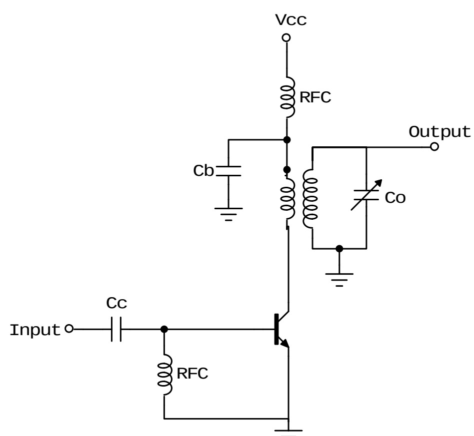


Figure 0.1 Frequency multiplier

Frequency translation, which is also known as *heterodyning*, on the other hand, is accomplished by using a mixer to mix the modulated output or input signal with another signal of higher frequency to produce a new set of frequencies that include the difference and the sum of the two frequencies, then out of the generated new set of frequencies the required frequency is selected while the unwanted frequencies are attenuated. Frequency translation is usually used after modulation, in transmitters, to up-convert the modulated signal to the required band in the spectrum before the signal is fed to the antenna. It does not change the deviation or the bandwidth of the modulated signal, instead it just simple change the carrier of the signal from one frequency to another, which is usually higher.

Frequency translation is also used in superheterodyne receivers to down-convert the selected incoming radio frequency to a lower frequency, which is known as *intermediate frequency* (IF).

For example, in Figure 1.2 the incoming frequency f_i and the output of the local oscillator f_o are fed to the mixer, which produces, among its output the sum and the different of the two frequencies. The output of the mixer is passed through a band pass filter that will allow only the difference of the two frequencies $f_o - f_i$, when down-converting or the sum $f_o + f_i$, when up-converting.

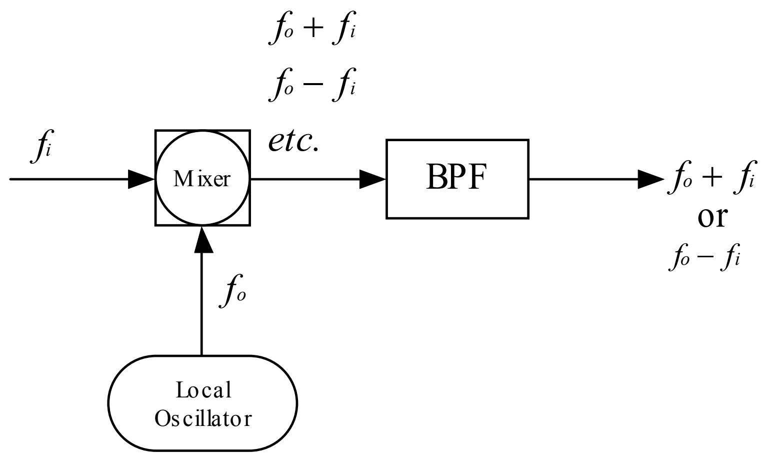


Figure 0.2 Frequency translation

1 ANALOGUE RADIO TRANSMITTERS

2.1 AM SYSTEMS

Amplitude modulation involves the varying of the amplitude of the carrier in accordance with the modulating signal. The resultant AM signal consists of consists of three frequency components: the carrier, the difference of frequencies of the carrier and the modulating signal, and the sum of the frequencies of the carrier and the modulating signal. That is,

$$v_{AM} = V_c \sin w_c t + m\frac{V_c}{2}\cos(w_c - w_m)t - m\frac{V_c}{2}\cos(w_c + w_m)t \quad (1.2)$$

where the sum is referred to as upper side frequency or band while the difference is called the lower side frequency or band. This is a default AM expression, which is known as a *double-sideband full carrier* (DSBFC) or just double-sideband (DSB) signal.

2.2 AM MODULATOR

An AM waveform can be generated by combining the modulating signal and the carrier through a non-linear device such as a diode or a transistor. Since the transistor is also providing the gain in addition to non-linearity, it is more popular in AM generation than the diode. However, for higher Wattages, the transistor is usually replaced by a vacuum tube because of its high plate voltage capability.

When using a transistor for AM generation, the most commonly used technique for producing an AM signal is to use a transistor to generate positive current pulses that are proportional to the modulating signal and then apply the generated series of current pulses to a tuned circuit whose Q is not too low. In the tuned circuit these pulses will generate damped oscillations with amplitudes that are proportional to the size of the current pulses. This phenomenon is referred to as the *flywheel effect* of the tuned circuit, and each pulse will result in a complete cycle of a sine wave with an amplitude that is proportional to the size of the pulse. This process is depicted in Figure 2.1 below.

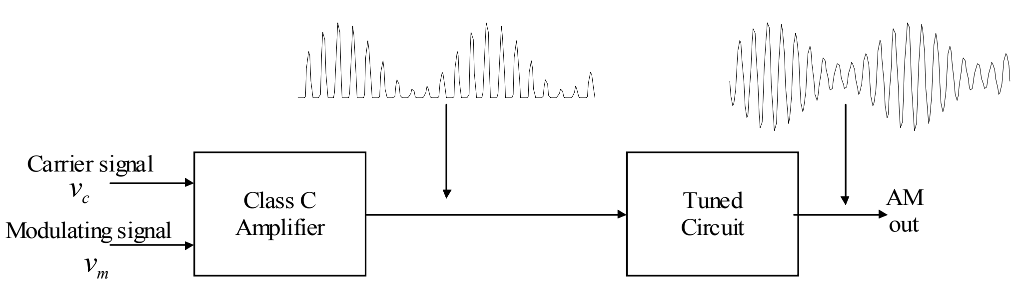


Figure 2.1 AM generation using class C amplifier

In order to generate the current pulses that are proportional to the modulating signal, a class C amplifier is normally used. In the class C amplifier the carrier is normally applied to the base while the modulating signal can be injected to any of the three transistor terminals, thus resulting in *base modulation*, *collector modulation* or *emitter modulation*. The tuned circuit is then tuned to resonate at the carrier frequency and its bandwidth is wide enough to also accommodate the upper and the lower sidebands.

2.2.1 Base Modulation

Base modulation is used mostly for low-level modulation. Figure 2.2 shows a base modulation AM modulator that is using a class C amplifier. The modulating signal is applied to the base together with the carrier, hence the name base modulation.

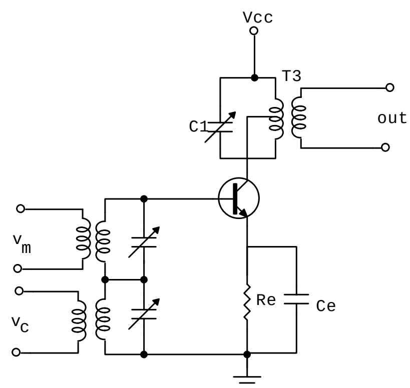


Figure 2.2 Amplitude modulator: Base modulation

The carrier signal and the modulating signal are summed up and their sum is applied as an input to the base of a class C amplifier. During the positive half-cycles the transistor will conduct whenever the input goes above 0.7 V, and will cutoff for the negative half-cycles. This will result in a series of positive current pulses at the collector that are having amplitudes that are proportional to the input voltage. Due to the non-linearity of the transistor, the output current will be of the form

$$i_o = a + bv_i + cv_i^2 \qquad (1.3)$$

Where v_i is the total input voltage, which is the sum of the carrier input and the modulating signal; that is,

$$v_i = v_c + v_m = V_c \sin w_c t + V_m \sin w_m t$$

Substituting for v_i in i_o we get

$$i_o = a + b(V_c \sin w_c t + V_m \sin w_m t) + c(V_c \sin w_c t + V_m \sin w_m t)^2$$

$$
\begin{aligned}
&= a + bV_c \sin w_c t + bV_m \sin w_m t + cV_c^2 \sin^2 w_c t + 2cV_c \sin w_c t V_m \sin w_m t + V_m^2 \sin^2 w_m t \\
&= a + bV_c \sin w_c t + bV_m \sin w_m t + cV_c^2 \sin^2 w_c t + cV_c V_m \cos(w_c - w_m)t \\
&- cV_c V_m \cos(w_c + w_m)t + V_m^2 \sin^2 w_m t
\end{aligned} \tag{1.4}
$$

This current is applied to a tuned circuit that is connected to the collector of the transistor. The tuned circuit restores the missing half-cycles of the collector current using fly-wheel effect, thus resulting in complete cycles at the output of the RF transformer T3.
The tuned circuit is also used to only allow the carrier component together with the upper and lower side bands to pass through while attenuating the other frequency components of the collector current. Therefore the final output will be

$$
bV_c \sin w_c t + cV_c V_m \cos(w_c - w_m)t - cV_c V_m \cos(w_c + w_m)t \tag{1.5}
$$

which is a DSBFC signal.

2.2.2 Collector modulation

The circuit for collector modulation is shown in Figure 2.3. The circuit is also using a class C amplifier. Unlike the base modulation circuit in Figure 2.2, only the carrier is applied to the base and the modulating signal is applied to the collector, hence the name collector modulation. Collector modulation is normally used for high-level modulation.

During the positive half-cycles the transistor conducts whenever the input goes above 0.7V, and will cut off for the negative half-cycles and voltages that are less than 0.7V. The modulating signal is fed in series with the supply voltage, V_{cc}. Since the modulating signal is a varying signal, it will make the collector supply voltage to vary in accordance with the modulating signal. This will result in a series of positive current pulses at the collector that are having amplitudes that are proportional to the input carrier voltage and the modulating signal. These current pulses are then applied to a tuned circuit that is connected to the collector of the transistor. The tuned circuit restores the missing half-cycles, thus resulting in an AM waveform at the output of the RF transformer T3.

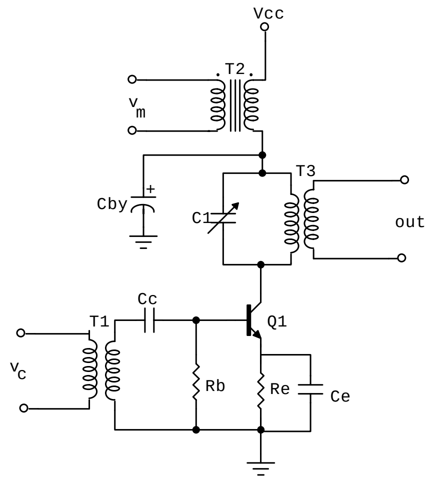


Figure 2.3 Amplitude modulator: Collector modulation

2.2.3 Emitter modulation

The circuit for emitter modulation is shown in Figure 2.4, and it is almost the same as the one for collector modulation except that in emitter modulation the modulating signal is connected to the emitter instead of the collector, hence the name emitter modulation. Like base modulation, emitter modulation is used for low-level modulation.

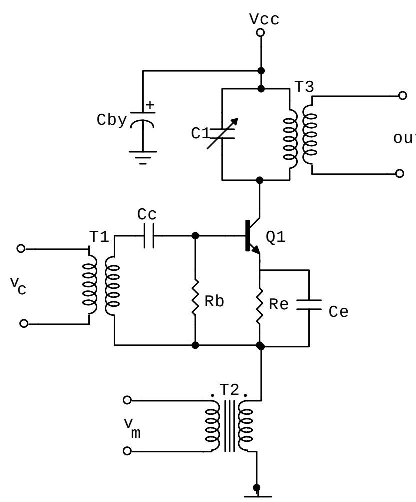


Figure 2.4 Amplitude modulator: Emitter modulation

The modulating signal that is connected to the emitter will cause the emitter voltage to vary. This will cause the emitter current to vary, which in turn will cause the gain and the collector voltage to vary with the modulating signal. This will result in a train of collector current pulses that are having amplitudes that are varying with modulating signal. These current pulses are then applied to a tuned circuit that is connected to the collector of the transistor. The tuned circuit restores the missing half-cycles, thus resulting in an AM waveform at the output of the RF transformer T3.

2.2.4 LIC AM generator

AM signals can also be generated using operational transconductance amplifier (OTA). The advantage of using OTA is that it does not need any inductor to generate high-quality AM waveforms. However, its use is limited to low-level amplitude modulation.

Unlike the ordinary OPAMP an OTA has a differential input voltage that produces an output current which is a linear function of the differential input voltage. That is,

$$I_o = \left(V_{in}^{+} - V_{in}^{-}\right) g_m = g_m \Delta v_{in} \tag{1.6}$$

where V_{in}^{+} and V_{in}^{-} are voltages at the non-inverting and inverting input respectively, and g_m is the transconductance of the amplifier. OTA has also an additional input for a control current, called amplifier bias current (I_{abc}), to control the transconductance of the amplifier. It is this feature that makes the OTA useful for generating AM. When used for AM generation, the modulating signal is used to produce the amplifier bias current which is used to vary the transconductance of the OTA. Since the transcoductance of the OTA is directly proportional to the amplifier bias current, any variations in the modulation signal will produce a varying amplifier bias current that will lead to variations in the transcoductance and hence the gain of the amplifier. That is, the amplitude of the output signal will vary in accordance with the modulating signal thereby producing an AM signal.

The commercially available OTAs include the classical CA3080, and dual OTAs such HA2735, CA3280, LM13600 and LM13700. Figure 2.5 shows a typical LIC AM modulator using the classical CA3080.

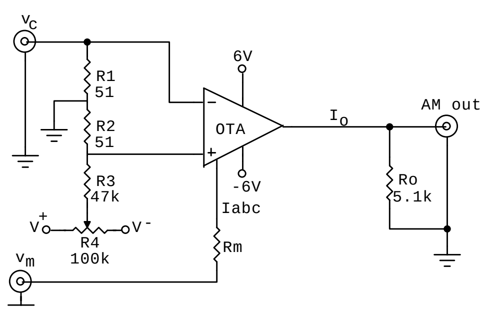


Figure 2.5 AM generation using CA3080 OTA

The variable resistor R4 is used to null the effects of the amplifier input offset voltage so as to set a symmetrical output voltage. The output voltage swing is the product of the output current I_o and output resistance R_o; that is,

$$v_o = I_o R_o \tag{1.7a}$$

$$= -g_m R_o v_c \tag{1.7b}$$

where the negative sign is due to the fact that the carrier is applied to the inverting input of the OTA. Modulation of the carrier occurs because the variation of the modulating signal forces a change in the amplifier bias current, which is supplied via resistor R_m. During the positive half-cycles of the modulating signal the amplifier bias current will increase which causes a corresponding increase in the transconductance of the amplifier. During the negative half-cycles of the modulating signal the amplifier bias current will decrease, and reduce the transconductance of the amplifier. The value of the transconductance is

$$g_m = 19.2 \times I_{abc} \tag{1.8}$$

And the amplifier bias current will be

$$I_{abc} = \frac{V_m - V^-}{R_m} \tag{1.9}$$

2.3 SINGLE-SIDEBAND SYSTEMS

Since both the upper and the lower sidebands contain all the information that was in the modulating signal, it is possible to transmit the same information using just one sideband. That is, the transmitted signal will then be

$$v_{AM_{(SSB)}} = V_c \sin w_c t + m\frac{V_c}{2}\cos(w_c - w_m)t \tag{1.10a}$$

Or

$$v_{AM_{(SSB)}} = V_c \sin w_c t - m\frac{V_c}{2}\cos(w_c + w_m)t \tag{1.10b}$$

This type of AM signal is called a *single-sideband* (SSB). The advantage of SSB over DSB is the reduction of the bandwidth to half of that of the DSB. Power will also be reduced. However, since the carrier does not contain any of the information that was in the modulating signal, it can also be left out without the loss of the information. Such a signal is known as a *single-sideband suppressed carrier* (SSBSC) signal. Thus making the transmitted signal to be

$$v_{AM_{(SSBSC)}} = m\frac{V_c}{2}\cos(w_c - w_m)t \tag{1.12a}$$

Or

$$v_{AM_{(SSBSC)}} = m\frac{V_c}{2}\cos(w_c + w_m)t \tag{1.12b}$$

Suppressing of one sideband and the carrier further reduce the total bandwidth required to transmit the signal and the power of the signal. For these reasons SSBSC is the most preferred single sideband (SSB) signal used in communication, especially in frequency-division multiplexing (FDM) systems.

The following sections describe how the carrier and one sideband are suppressed in order to produce a SSBSC signal.

2.3.1 Balanced modulator

At the heart of the single-sideband system is a balanced modulator (Bal. Mod). It is used in both single-sideband generation in the transmitter and demodulation in the receiver. The balanced modulator is an AM modulator circuit that produces only the two sidebands and the carrier frequency is suppressed; that is, it generates a DSBSC signal. There are three most commonly used balanced modulators: balanced ring modulator, dual-FET balanced modulator, and 1496/1596 LIC balanced modulator. The symbols of a balanced modulator are as shown in Figure 2.6.

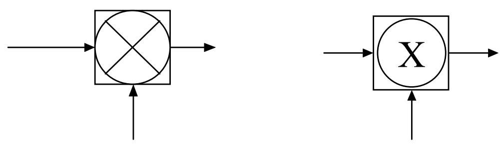


Figure 2.6 Balanced modulator symbols

2.3.1.1 Balanced ring modulator

A classical balanced modulator is shown in Figure 2.7. It is commonly known as a *ring modulator*. The circuit consists of input audio transformer T1, four diodes *D1* through *D4*, output RF transformer T2, and another RF transformer T3 for coupling the carrier.

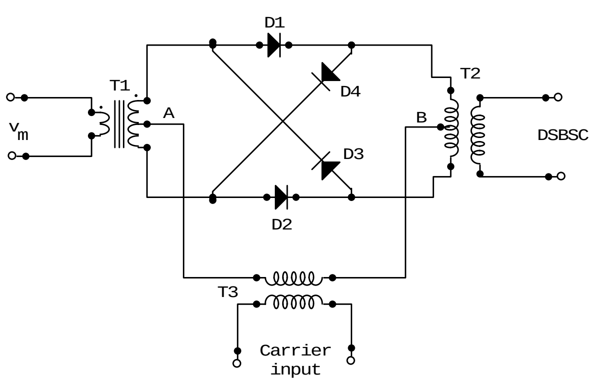


Figure 1.7 Balanced ring modulator

The circuit uses the four carefully matched diodes as the nonlinear devices to produce the two sidebands and to suppress the carrier. For the circuit to work properly, the amplitude of the carrier must be greater than that of the modulating signal. The carrier signal alternately switches on and off diode pairs *D1-D2* and *D3-D4*. During the positive half-cycles of the carrier, point A is positive relative to point B, thus making *D1* and *D2* to be forward biased while *D3* and *D4* are reversed biased. Current flows into the center-tap of transformer T1, then splitting and flowing through *D1* and *D2*, converging at the center-tap of T2 and back to carrier input transformer T3. Since the diodes are perfectly matched, the opposing currents through the output transformer T2 will be 180^0 out-of-phase and therefore will cancel. During the negative half-cycles of the carrier point B is positive relative to point A, thus making *D3* and *D4* to be forward biased while *D1* and *D2* are reversed biased. The current will flow into the center-tap of transformer T2, then splitting and flowing through *D3* and D4, converging at the center-tap of T1 and back to carrier input transformer T3. Again the current through the output transformer T2 will be 180^0 out-of-phase and therefore will cancel. The modulating signal v_m, which is fed through the audio transformer T1 either adds or opposes this conduction, thus upsetting the current balance in the primary of the output transformer T2 and resulting the production of the desired sidebands while continuing suppressing the carrier. The resulting sidebands are induced into the secondary of the output RF transformer T2; however, the same output transformer will offer low impedance to the modulating signal thereby attenuating it. Therefore, the output of the balanced ring modulator is a DSBSC signal.

2.3.1.2 Dual-FET balanced modulator

Like the balanced ring modulator, the dual-FET balanced modulator circuit consists of input audio transformer T1, output RF transformer T2, and another RF transformer T3 for coupling the carrier. However, instead of using diodes as nonlinear devices, it is using two identical field-effect transistors (FET) transistors as shown in Figure 2.8.

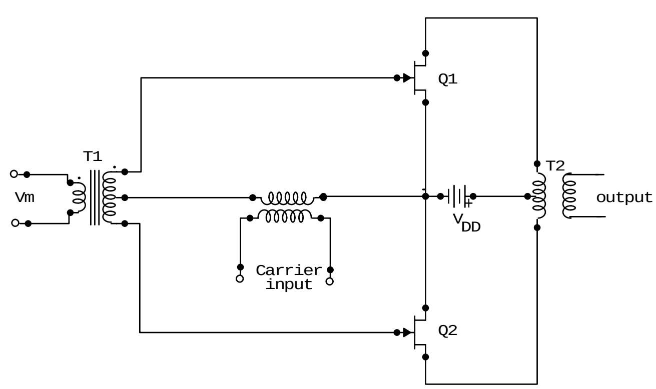


Figure 1.8 Dual-FET balanced modulator

The modulating signal is fed in a push-pull to the gates of a pair of identical FETs. The carrier signal is also applied to the two gates in phase. Therefore, the input voltage to the gate of *Q1* (v_{gs1}) will be the sum of the carrier voltage (v_c) and half of the modulating voltage (v_m) while that to the gate of *Q2* (v_{gs2}) will be the difference between the carrier voltage and half of the modulating signal. The transfer characteristics of the FETs are almost parabolic and can be approximated by

$$i_d = I_o + av_{gs} + bv_{gs}^2 \qquad (1.13)$$

Therefore the output currents of the two FET will be

$$i_{d1} = I_o + av_{gs1} + bv_{gs1}^2 \qquad (1.14a)$$

And

$$i_{d2} = I_o + av_{gs2} + bv_{gs2}^2 \qquad (1.14b)$$

These two output drain currents are combined in the primary of the output transformer. Since the two drain currents flow in opposite directions, the effective primary current i_p is

$$\begin{aligned} i_p &= i_{d1} - i_{d2} \\ &= I_o + av_{gs1} + bv_{gs1}^2 - I_o - av_{gs2} - bv_{gs2}^2 \\ &= av_{gs1} + bv_{gs1}^2 - av_{gs2} - bv_{gs2}^2 \\ &= a\left(v_{gs1} - v_{gs2}\right) + b\left(v_{gs1}^2 - v_{gs2}^2\right) \end{aligned}$$

Substituting $v_{gs1} = v_c + \frac{v_m}{2}$ and $v_{gs2} = v_c - \frac{v_m}{2}$ we get

$$\begin{aligned} i_p &= a\left[v_c + \frac{v_m}{2} - v_c - \frac{v_m}{2}\right] + b\left[\left(v_c + \frac{v_m}{2}\right)^2 - \left(v_c - \frac{v_m}{2}\right)^2\right] \\ &= a\left[v_m\right] + b\left[v_c^2 + v_c v_m + \frac{v_m^2}{4} - v_c^2 + v_c v_m - \frac{v_m^2}{4}\right] \\ &= av_m + 2bv_c v_m \end{aligned}$$

Substituting $v_m = V_m \sin w_m t$ and $v_c = V_c \sin w_c t$ we get

$$\begin{aligned} i_p &= aV_m \sin w_m t + 2bV_c \sin w_c t \times V_m \sin w_m t \\ &= aV_m \sin w_m t + bV_c V_m\left[\cos\left(w_c - w_m\right)t - \cos\left(w_c + w_m\right)t\right] \qquad (1.15) \end{aligned}$$

The output RF transformer will attenuate the low frequency component $aV_m \sin w_m t$, thereby allowing only $bV_c V_m\left[\cos\left(w_c - w_m\right)t - \cos\left(w_c + w_m\right)t\right]$ to pass through, which is the required DSBSC signal.

2.3.1.4 The integrated circuit balanced modulator

A DSBSC signal can also be generated using integrated circuits such as MC 1496, MC 1596, SL 640 and SL 641balanced modulators. The advantage of IC balanced modulators is that they do not need inductors or transformers like their discrete circuitry counterparts. Secondly, they offer exceptionally good carrier suppression and fully balanced input and output circuits. Thirdly, they are capable of operating over a wide range of frequencies. A typical circuit of an IC balanced modulator is shown in Figure 2.9.

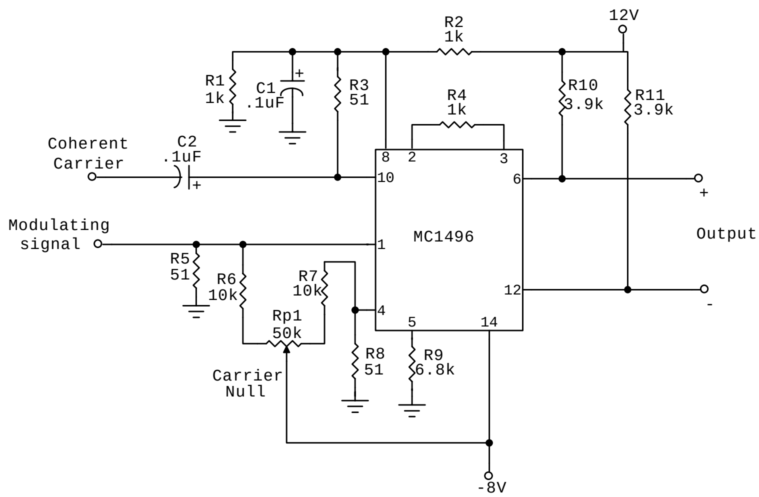


Figure 1.9 A MC 1496 Balanced Modulator

The circuit uses an MC 1496 balanced modulator, which is a double balanced modulator/demodulator. It produces an output voltage that is proportional to the product of the carrier signal (v_c) and the modulating signal (v_m).

2.3.1.5 Suppressing the unwanted sideband of the DSBSC

Since now the carrier has been suppressed, let us look at how to suppress the unwanted sideband so that we can get the required SSBSC signal. There are three techniques that can be used to suppress the unwanted sideband: *filter method; phasing or phase-shift method* and *third method*. The third method is very complex and is not often used commercially, hence for the discussion here only the filter method and phasing method will be considered. Depending on the system requirements and circuit configuration, either method will suppress either the lower or the upper sideband with ease.

Filter method of SSBSC generation

The filter method is the most commonly used technique of SSBSC carrier generation. It consists of a balanced modulator and a band pass filter as shown in Figure 2.10.

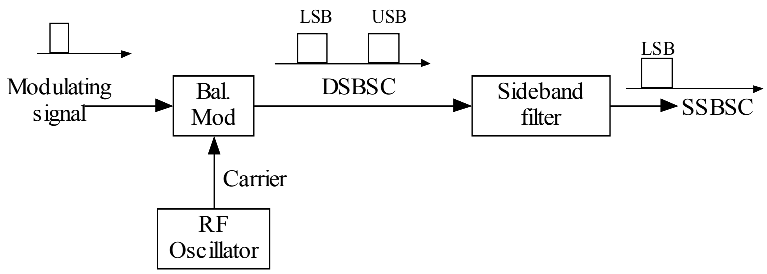


Figure 1.10 Filter method of SSBSC generation

Here, the modulating signal baseband and the carrier are applied to a balanced modulator. In the balanced modulator the carrier is modulated by the modulating signal to produce a DSBSC signal. The output of the balanced modulator is then passed through a sideband filter, which is a band pass filter that is designed to pass the required sideband and to block the unwanted sideband. The output of the sideband filter is a single sideband suppressed carrier signal. Though in the block diagram the resultant SSBSC signal is a lower sideband, the sideband filter can be configured to pass the upper sideband instead of the lower sideband. Since the two sidebands are close to each other, the sideband filter needs to be a high *Q* filter such as the crystal filter. These crystal filters are capable of producing sharply defined skirts that are required to properly filter out the unwanted sideband.

Phasing method of SSBSC generation

The phasing or phase-shift method uses the balanced modulators to suppress the carrier and phase-shifting networks to remove the unwanted sideband, thus eliminating the use of sideband filters and their high *Q* requirements. Figure 2.11 shows the block diagram of a phasing method of SSBSC generation.

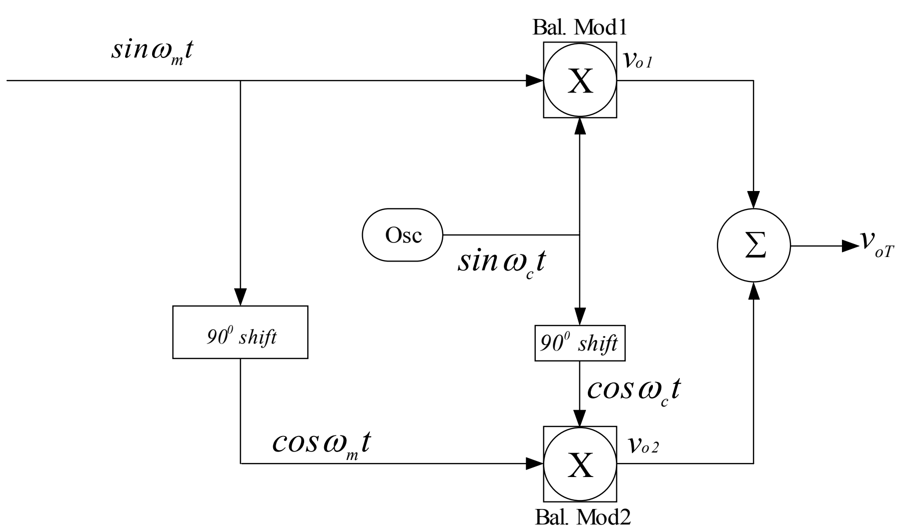


Figure 1.11 Phasing method of SSBSC generation

The phasing method uses two balanced modulators to produce two DSBSC signals that are then summed up to produce the required SSBSC signal. The modulating signal is split into two identical components. One of the components is applied directly to balanced modulator 1 while the other one is shifted by 90^0 before being applied to balanced modulator 2. The oscillator generates the carrier signal that is fed directly to balanced modulator 1. A sample of the same carrier signal is shifted by 90^0 before being fed to balanced modulator 2. In the balanced modulators the modulating signals are used to modulate the carrier signals. In each balanced modulators the carrier is eliminated to produce a DSBSC signal at each output. The two outputs are summed up to produced the final output v_{oT}; that is,

$$v_{oT} = v_{01} + v_{o2} \tag{1.16}$$

$$= sin\, w_c t\, sin\, w_m t + cos\, w_c t\, cos\, w_m t$$

$$= \frac{1}{2}\cos(w_c - w_m)t - \frac{1}{2}\cos(w_c + w_m)t$$

$$+ \frac{1}{2}\cos(w_c - w_m)t + \frac{1}{2}\cos(w_c + w_m)t$$

$$= cos(\, w_c - w_m\,)t \tag{1.17}$$

The resultant output is a SSBSC signal, which in this case is the lower sideband.

2.3.1.6 Vestigial Sideband Systems

The bandwidth saving of the SSBSC comes at the expense of a complicated demodulation at the receiver. The DSBFC, on the other hand, though it needs more bandwidth than SSBSC, it can be demodulated by a simple envelope detector circuit. To strike a compromise between the two, a vestigial sideband (VSB) system is used. Like in normal DSBFC, all the three output components of the AM signal are transmitted in VSB systems. However, before transmission, most of the lower sideband is removed. This is done to minimize the bandwidth of the signal while maintaining the characteristics of DSBFC for easy demodulation. Figure 2.12 shows a block diagram of a VSB system.

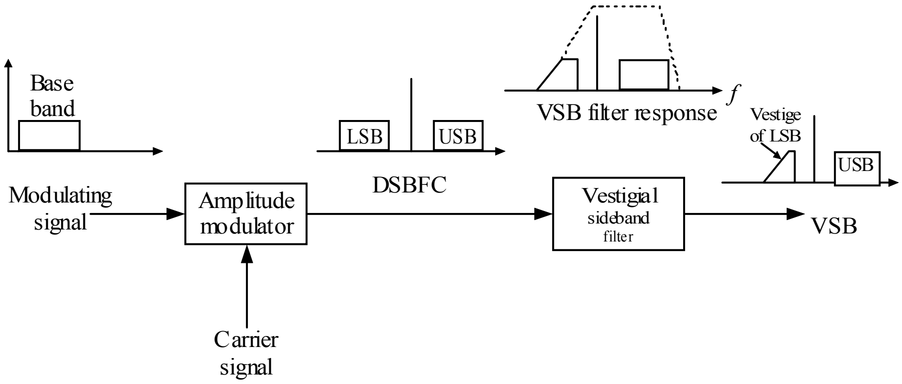


Figure 1.12 Vestigial sideband generation

The modulating signal and the carrier signal are applied to an amplitude modulator where the modulating signal is used to modulate the carrier. The output of the modulator, which is a DSBFC signal, is passed through a vestigial sideband filter that eliminates most of the lower sideband. The resultant output contains the full carrier, the whole of the upper sideband and a vestige or piece of the lower sideband, hence it is referred to as vestigial sideband.

VSB is used mostly in analog television broadcast systems such as NTSC, PAL and SECAM, and in digital transmissions such as ATSC to transmit the bandwidth-hungry video signals. After amplitude modulation of the video or picture carrier by the video baseband signal, the VSB filter is used to remove most of the lower sideband up to a certain frequency below the video carrier with a percentage of it being left undiminished. The frequencies to which the lower visual sideband is transmitted in full and undiminished below the video carrier depend on the standard being used. For example, in NTSC TV the first 1.25 MHz of the lower sideband is sent with its first 0.75 MHz being left intact, while in PAL CCIR system I, as used in South Africa, the first 1.75 MHz of the lower sideband is sent with its first 1.25 MHz being left intact. Figure 2.13 shows the video signal spectra for both NTSC and PAL CCIR system I. For completeness, the spectra also show the location of the frequency-modulated sound transmission that accompanies video. Please note that NTSC has a sound carrier that is 4.5 MHz above the video carrier while in South Africa the system being used has a sound carrier that is 6 MHz above the video carrier. The resultant total bandwidths are 6 MHz and 8 MHz for NTSC and PAL CCIR system I, respectively. These total TV signal bandwidths include sound of 180 to 200 kHz, which is centered around the sound carrier.

Though VSB uses smaller bandwidth than DSBFC and retains the DSBFC characteristics for easy demodulation, the main drawback of using VSB is that the receiver must compensate for the part of the lower sideband that was eliminated at the transmitter and if not the picture will suffer distortion. This distortion will cause the lower frequencies to be overemphasized in the picture and the fine details to be washed out. Compensation at the receiver can be accomplished by providing IF filtering before envelope detection. This is done by designing the IF amplifier such that its frequency response roll off linearly between from 1.25 MHz below the picture carrier to about 1.25 MHz above the picture carrier as shown in Figure 2.13b and Figure 2.13d. The reason for this is to attenuated the power in the lower sideband and partly attenuate the lower portion of the upper sideband so as to eliminate the overemphasis of the low frequency while the high frequency components are left unaffected so that they can be emphasized by the IF stage.

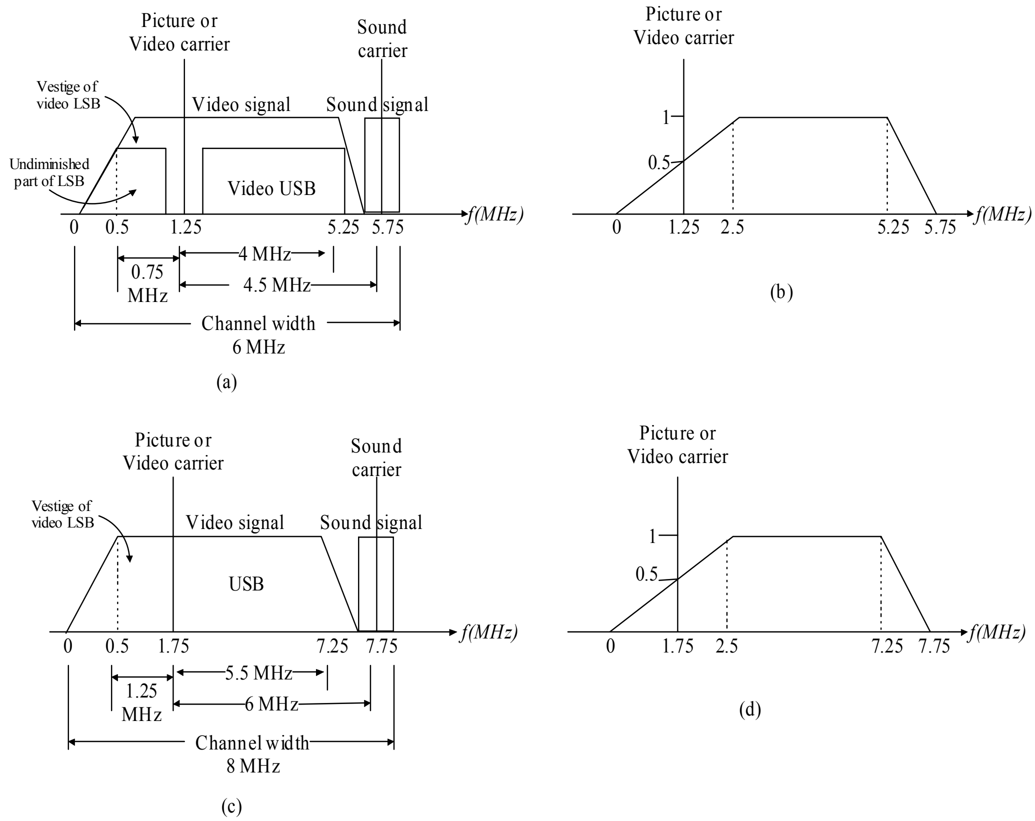


Figure 2.13 Television signal: (a) NTSC TV signal spectrum; (b) NTSC receiver video IF bandpass response; (c) CCIR PAL I TV signal spectrum; (d) CCIR PAL I receiver video IF bandpass response

2.4 FM SYSTEMS

FM systems use the changes in frequency of the carrier to carry the information from the transmitter to the receiver; that is, they use frequency modulation, which involves the varying of the frequency of the carrier in accordance with the modulating signal. These frequency shifts in the carrier is called *deviation*, and the broader the deviation the higher the percentage of modulation. The FM signal consists of the carrier and two or more side frequencies, and its expression is given by

$$v_{FM} = V_c \sin\left(w_c t - m_f \cos w_m t\right) \tag{1.18}$$

FM with low levels of deviation (i.e. $m_f < 0.25$) is referred to as *narrow-band frequency modulation* (NBFM) while that with higher levels of deviation (i.e. $m_f > 0.25$) is called *wide-band frequency modulation* (WBFM). NBFM is widely used for voice only communication systems such as taxicabs, aircraft, police, ambulances, marine, and security radio networks where high quality is not important. They have bandwidths ranging from 10 – 30 kHz. WBFM is widely used in FM broadcasting for entertainment to provide high quality sound. Unlike NBFM that has bandwidth of 10 to 30 kHz, WBFM has bandwidth that is about ten times that of NBFM. Equation 1.18 is usually used for WBFM while the equation for NBFM is obtained by expanding Equation 1.18 using trigonometric identity $\sin(a-b) = \sin a \cos b - \cos a \sin b$. That is,

$$v_{PM} = V_c \sin w_c t\left[\cos\left(m_f \sin w_m t\right)\right] - V_c \cos w_c t\left[\sin\left(m_f \sin w_m t\right)\right]$$

For narrow-band phase modulation (NBFM) or low-level deviation ($m_f < 0.25$) the angular variations of the sine and the cosine in brackets can be approximated as

$$\cos \Delta\phi \approx 1$$

and

$$\sin \Delta\phi \approx \Delta\phi$$

Therefore the NBFM can be approximated as

$$v_{PM} = V_c \sin w_c t - m_f V_c \cos w_c t \sin w_m t \tag{1.19}$$

2.4.1 Advantages and disadvantages of FM

There are two main advantages of using FM; immunity to noise and better signal quality. When the transmission is affected by noise, the noise is usually superimposed on the transmitted signal thereby altering its amplitude. Since in FM the demodulators are preceded by amplitude limiters and are also designed to have demodulators that only respond to frequency changes, noise has little or no effect in FM systems. Secondly, the bandwidth of the FM broadcast is in the order of 200 kHz as opposed to the 10 kHz used for AM broadcast, which allows for the whole of the music components to be transmitted thereby improving the sound quality in the FM radio receivers.

Amplitude modulation results in a signal that is only having two sidebands, while frequency modulation results in a signal that has two or more side frequencies hence the bandwidth required to transmit FM is usually wider than that required for AM. Therefore, if the signals are to be multiplex, you can be able to frequency-multiplex more AM signals than FM for any given bandwidth. It is for this reason why most of the telecommunication companies were using AM, instead of FM, in their FDM systems.

2.4.2 FM generation

A frequency generator needs to produce a varying frequency output signal, which has variations that are proportional to the instantaneous amplitude of the modulating signal. There are two methods of generating a frequency modulated signal: direct and indirect methods.

2.4.2.1 Direct FM generation

Direct FM generation involves varying the frequency of an oscillator using a modulating signal. The oscillators that are commonly used in direct FM generation include the voltage controlled oscillator (VCO) and tuned *LC* oscillators such as Clapp, Hartley and Colpitts oscillators. In VCOs the modulating signal is used to vary the control voltage, hence the output frequency of the oscillator while in tuned *LC* oscillators, the modulating signal is used to vary the reactance of one of the components that determine the output frequency.

FM generation using LC oscillators

Tuned LC oscillators employ a tuned circuit to generate frequency, and the generated frequency is given by

$$f_o = \frac{1}{2\pi\sqrt{LC}} \tag{1.20}$$

Any change in capacitance or inductance will result in a corresponding change in the output frequency. So to generate an FM signal either the capacitance or the inductance of the tuned circuit is varied by modulating signal. There are two way of achieving this, namely, using a varactor diode or using an active device such as [BJT or FET] transistor or a vacuum tube. The latter is usually referred to as a *reactance modulator*.

Varactor diode FM modulator

A varactor or varicap is a special type of a diode that is designed to have its capacitance varying with voltage; that is, it is a voltage-controlled capacitor. In a reverse biased condition no current flows through the diode, instead there are two conducting regions which are separated by a non-conducting region just like a normal capacitor. By increasing the reverse bias voltage the width of the non-conducting region will increase and the separation between the two conducting regions will also increase, thus causing the capacitance value to decrease. When the reverse bias voltage is decreased, the width of the non-conducting region will decrease and the separation between the two conducting regions will also decrease, thus causing the capacitance value to increase. In FM generation, these variations in the capacitance of the varactor diode are caused by the modulating signal, and the varactor diode forms part of the tuned circuit that determines the output frequency. So as the modulator signal varies, the capacitance of the varactor diode also varies, and hence the output frequency of the oscillator. Figure 2.14a shows a varactor diode with capacitance that is varied by the modulating signal v_m and Figure 2.14b shows a circuit of a Clapp oscillator.

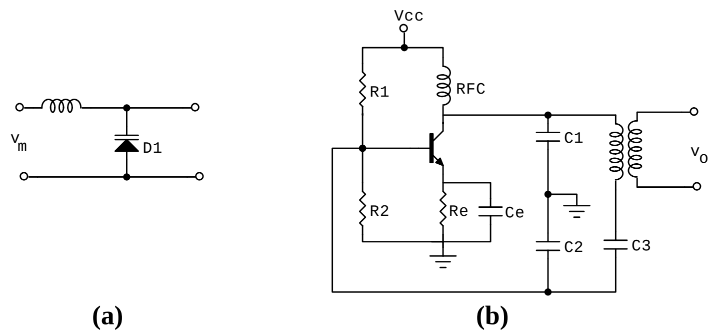


Figure 2.14 (a) Varactor diode; (b) Clapp Oscillator

Figure 2.15 shows a circuit of a FM modulator using a varactor diode and a Clapp oscillator. FM modulation is accomplished by replacing capacitor C_3 of the Clapp oscillator by a varactor diode, which is pre-biased to be at its center value by resistors R_3 and R_4. R_4 is a variable resistor to fine-tune the unmodulated carrier frequency. When the modulating signal is applied across the terminals of the varactor diode its capacitance varies with the modulating signal, thus causing the output frequency of the oscillator to also vary with the modulating signal. The radio frequency choke (RFC) that is connected in series with the modulating signal is used to block the generated RF frequency from entering the modulating signal and power supply circuits.

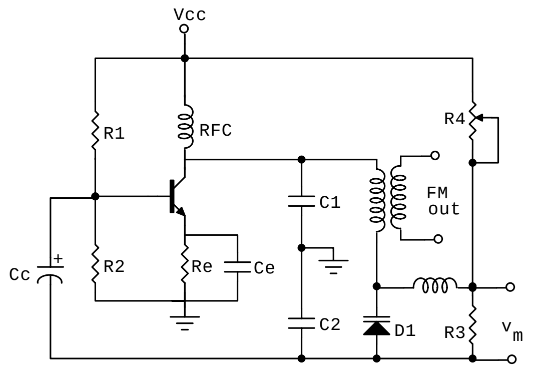
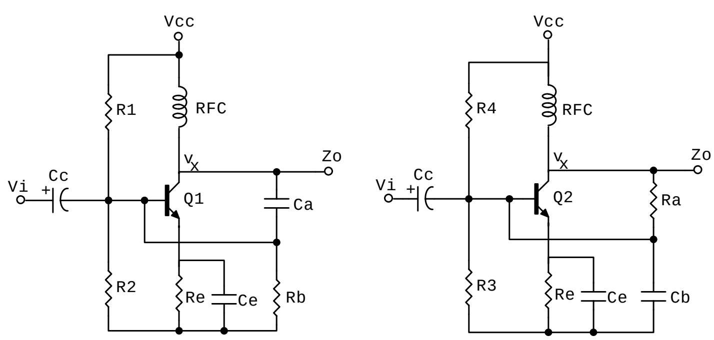


Figure 2.15 FM modulator using a varactor diode

Transistor reactance modulator

Transistors are used as reactance modulators to provide a voltage-controlled reactance. Unlike varactor diode, which only provides voltage-controlled capacitance, reactance modulators can be configured to provide either a voltage-controlled capacitance or a voltage-controlled inductance. The value of reactance produced is proportional to the transconductance of the transistor, which in turn is made to depend on the base (or gate) bias and its variations. Figure 2.16 and Figure 2.17 show the circuits for reactance modulators for providing voltage-controlled capacitance or voltage-controlled inductance.



(a) **(b)**

Figure 2.16 BJT reactance modulators: (a) Variable capacitance; (b) Variable inductance

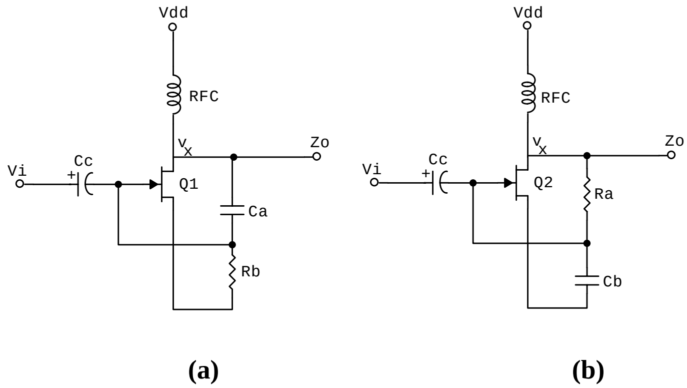


Figure 2.17 FET reactance modulators: (a) Variable capacitance; (b) Variable inductance

For the circuits to be reactive, the collector-to-emitter (drain-to-gate in FET) impedance must be greater than the base-to-emitter (gate-to-source in FET) impedance. That is, in Figures 2.16a and 2.17a the capacitive reactance of C_a must be much greater than resistance of R_b, and in Figures 2.16b and 2.17b the resistance of R_a must be much greater than capacitive reactance of C_b. In BJT reactance modulators, the additional resistors: R_1, R_2 and R_e are for biasing. In order to analyze these reactance modulator circuits, the collector-to-emitter (drain-to-gate in FET) impedance is represented by Z_a and the base-to-emitter (gate-to-source in FET) impedance is represented by Z_b, the voltage at the base or gate of the transistor is represented by v_m, and the voltage at the collector or drain of the transistor is represented by v_x. The analysis is as follows:

The voltage at the base or gate of the transistor is

$$v_i = \frac{Z_b}{Z_a + Z_b} v_x \tag{1.21}$$

The collector or drain current is

$$i_x = g_m v_i = g_m \frac{Z_b}{Z_a + Z_b} v_x \tag{1.22}$$

Therefore the output impedance of the reactance modulator is

$$Z_o = \frac{v_x}{i_x} = v_x \left(\frac{Z_a + Z_b}{g_m v_x Z_b} \right) = \frac{1}{g_m} \left(1 + \frac{Z_a}{Z_b} \right) \tag{1.23}$$

For Figures 1.17a and 1.18a Z_a is a capacitive reactance and Z_b is resistance, Z_o can be rewritten as

$$Z_o = \frac{1}{g_m} \left(1 - j \frac{X_c}{R} \right) \tag{1.24}$$

Since $Z_a >> Z_b$, $X_c >> R$, and Z_o will reduce to

$$Z_o = \frac{1}{g_m}\left(-j\frac{X_c}{R}\right) = -j\frac{X_c}{g_m R} \tag{1.25}$$

Which is a capacitive reactance, which may also be written as

$$X_{eq} = \frac{X_c}{g_m R} \tag{1.26}$$

Substituting for X_{eq} and X_c we get

$$\frac{1}{2\pi f C_{eq}} = \frac{1}{g_m R(2\pi f C)}$$

Re-arranging we get a pure capacitor that is given by

$$C_{eq} = g_m RC \tag{1.27a}$$

Substituting R with R_b and C with C_a, the equivalent capacitance becomes

$$C_{eq} = g_m R_b C_a \tag{1.27b}$$

For Figures 1.17b and 1.18b Z_a is resistance and Z_b is a capacitive reactance, Z_o can be rewritten as

$$Z_o = \frac{1}{g_m}\left(1 - \frac{R}{jX_c}\right) \tag{1.28}$$

Since $Z_a >> Z_b$, $R >> X_c$, and Z_o will reduce to

$$Z_o = \frac{1}{g_m}\left(-\frac{R}{jX_c}\right) = j\frac{R}{g_m X_c} \tag{1.29}$$

Which is an inductive reactance, which may also be written as

$$X_{eq} = \frac{R}{g_m X_c} \tag{1.30}$$

Substituting for X_{eq} and X_c we get

$$2\pi f L_{eq} = \frac{R(2\pi f C)}{g_m}$$

Re-arranging we get a pure inductor that is given by

$$L_{eq} = \frac{RC}{g_m} \tag{1.31a}$$

Substituting R with R_a and C with C_b, the equivalent inductance becomes

$$L_{eq} = \frac{R_a C_b}{g_m} \tag{1.31b}$$

From the above results it can be seen that the reactance modulator produces the capacitance or inductance whose value depends on the transconductance g_m of the transistor. During modulation the modulation the modulating signal varies the base or gate voltage which results in the varying of g_m, and hence the resultant reactance.

To generate FM, the reactance modulator is connected to the tank of a tuned circuit of an *LC* oscillator. For example, Figure 2.18 shows the most commonly used FM reactance modulator where a reactance modulator that is configured o provide a voltage-controlled capacitance is connected to the tank circuit of a Clapp oscillator. The modulating signal varies the capacitance of the reactance modulator, which in turn causes the capacitance of the tuned circuit to vary. Varying the capacitance of the tuned circuit will cause the output frequency to vary, and its variations will be proportional to the modulating signal, thus resulting in an FM output.

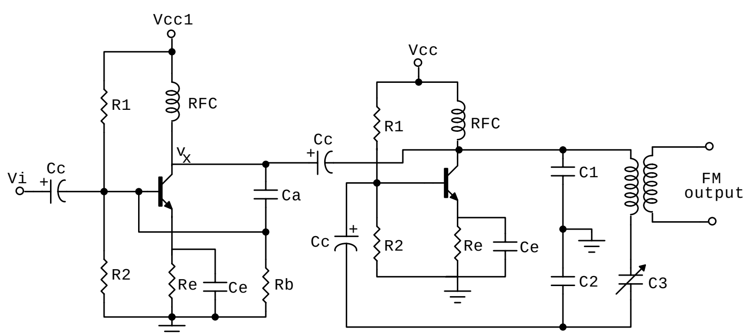


Figure 2.18 FM reactance modulator

FM generation using VCO

Voltage-controlled oscillator is an oscillator that generates a signal whose frequency is also determine by a control voltage, V_c, in addition to the normal *RC* network used for setting the oscillation frequency in *RC* oscillators. Using VCO as frequency modulators was formerly prohibitive on a discrete component basis, however, nowadays low cost monolithic linear integrated VCOs are available, thus making FM generation extremely simple. These LIC VCOs include LM566, MC1648, etc. Figure 2.19 shows a LM566 VCO and a LM566-based FM modulator.

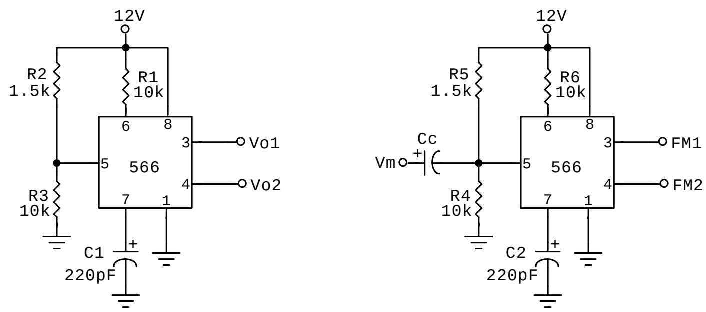


Figure 2.19 LM566: (a) Voltage controlled oscillator; (b) FM Modulator

A 566 VCO has two output of the same frequency: square-wave at Pin 3 and a triangular wave at Pin4. The free-running or center frequency of 566 is determined by resistor R_1 and capacitor C_1 together with the control voltage V_c at Pin 5, hence the name VCO. That is,

$$f_o = \frac{2}{R_1 C_1}\left(\frac{V^+ - V_c}{V^+}\right) \tag{1.32}$$

When used as an FM modulator, the modulating signal v_m is also applied to Pin 5 via a coupling capacitor C_c. During the positive half-cycles of the modulating signal v_m will add to V_c and during the negative half-cycles v_m will subtract from V_c. This will cause the voltage at Pin 5 to vary as modulating signal adds to and subtracts from V_c. Since V_c is involved in determining the output frequency, its variation will result in the output frequency varying. These frequency variations around center frequency will be on both the square-wave and triangular-wave outputs and they will vary proportional to the modulating signal. Either of these outputs can be applied to a tuned circuit that is resonant at VCO's center frequency in order to produce a standard sinusoidal FM waveform.

Sometimes the VCO within a phase lock loop (PLL) can be used, instead of a standalone VCO, as a FM modulator as shown below in Figure 2.20.

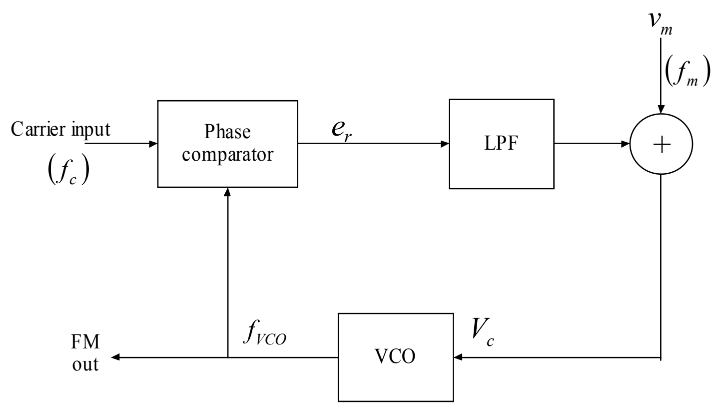


Figure 2.20 PLL FM Modulator

The carrier signal, which is within the capture range of the PLL, is applied to the phase comparator, and the VCO has a free-running frequency equal to the carrier. Without the modulating signal the VCO output frequency will be the same as that of the carrier. However, when the modulating signal is applied to the summer the control voltage of the VCO will be the sum of the error voltage e_r and modulating signal. During the positive half-cycle of v_m, the control voltage will increase resulting in a corresponding increase in the output frequency, and during the negative half-cycles of v_m the control-voltage will decrease resulting in a corresponding decrease in the output frequency. Thus the output of the VCO will vary around the carrier frequency thereby resulting in a FM output that has deviations that are proportional to the modulating signal. The advantage of this method is that it allows for the use of a crystal oscillator in order to bring stability to the FM signal generated, which is not possible with the previous methods. Commercial available PLL ICs that can be used include LM565, MC14046, NE564, NE567, NE568, etc. For example, Figure 2.21 shows a FM modulator using MC14046 PLL IC.

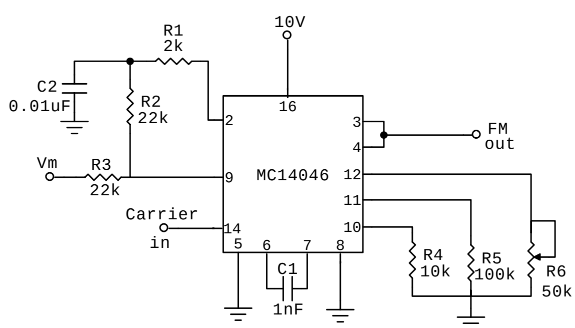


Figure 2.21 FM Modulator using MC14046 PLL IC

2.4.2.2 Indirect FM generation

Indirect method of FM generation uses a phase modulator to generate a FM signal. This allows for the use of a crystal oscillator to bring stability into FM generation. In order to generate FM using a PM modulator, the modulating signal is first passed through an integrator prior being used to phase modulate the output of the oscillator. This process is depicted in Figure 2.22.

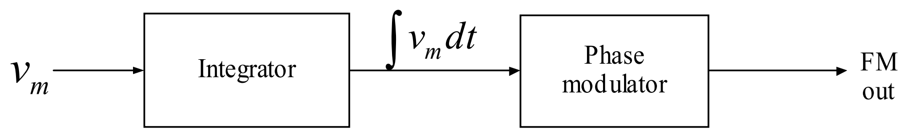


Figure 2.22 FM generation using phase modulator

2.5 PHASE MODULATION

Phase modulation is a type of angle modulation where the information transmitted is carrier by the phase of the carrier; that is, in phase modulation, the phase of the carrier is varied in accordance with the modulating signal, v_m. The phase-modulated signal is represented by

$$v_{PM} = V_c \sin(w_c t + m_p \sin w_m t) \qquad (1.33)$$

Using trigonometric identity $\sin(a+b) = \sin(b+a) = \sin a \cos b + \cos a \sin b$ to expand Equation (1.33) we get

$$v_{PM} = V_c \sin w_c t [\cos(m_p \sin w_m t)] + V_c \cos w_c t [\sin(m_p \sin w_m t)]$$

For narrow-band phase modulation (NBPM) or low-level deviation ($\Delta\phi < 0.25$) the angular variations of the sine and the cosine in brackets can be approximated as

$$\cos \Delta\phi \approx 1$$

and

$$\sin \Delta\phi \approx \Delta\phi$$

Therefore the NBPM can be approximated as

$$v_{PM} = V_c \sin w_c t + m_p V_c \cos w_c t \sin w_m t \qquad (1.34)$$

which means that the NBPM signal consists of a carrier and sidebands that are in quadrature with the carrier. Hence the diagram for generating NBPM is as shown in Figure 2.23.

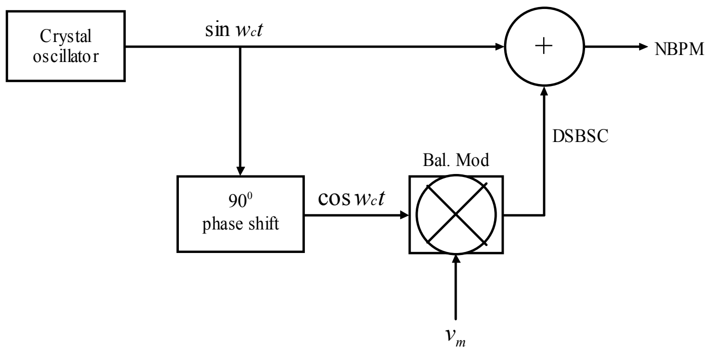


Figure 2.23 Phase modulator

The crystal oscillator generates a stable carrier frequency $v_c = V_c \sin w_c t$ that is fed directly to the summer. A sample of the carrier is phase-shifted by 90 degrees before being applied to a balanced modulator. In the balanced modulator the modulating signal is used to modulate the phase-shifted carrier to produce a DSBSC signal. This DSBSC is also applied to the summer where it is summed up with the original carrier. The output of the summer is

$$v_{PM} = V_c \sin w_c t + V_m V_c \cos w_c t \sin w_m t$$

which is similar to the NBPM in Equation 1.34.

Comparing the PM Equation 1.33 with the FM Equation 1.18, it is clear that PM and FM are very similar, in fact, if the modulating signal is passed through an integrator before phase modulation, the output of a phase modulator will be

$$v_{PM} = V_c \sin\left(w_c t - m_p \cos w_m t\right)$$

which is the same as FM Equation 1.18. Hence, in the indirect method of FM generation the modulating signal is first integrated before being fed to the phase modulator. Figure 2.24 shows the indirect method of generating FM using a phase modulator, which is commonly known as *Armstrong method*.

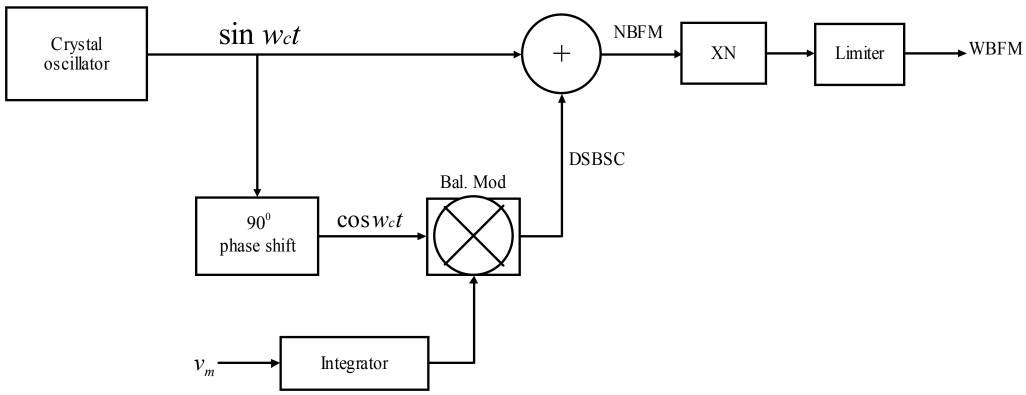


Figure 2.24 Armstrong method of FM generation

Like before, the crystal oscillator generates a stable carrier frequency $v_c = V_c \sin w_c t$ that is fed directly to the summer. A sample of the carrier is phase-shifted by 90 degrees before being applied to a balanced modulator. The modulating signal is passed through an integrator where it is integrated before being applied to the balanced modulator. In the balanced modulator the modulating signal is used to modulate the phase-shifted carrier to produce a DSBSC signal. This DSBSC is also applied to the summer where it is summed up with the original carrier. The output of the summer is

$$v_{PM} = V_c \sin w_c t + V_m V_c \cos w_c t \sin w_m t$$

which is similar to the NBFM in Equation 1.34. The generated NBFM is applied to a frequency multiplier to increase both the carrier frequency and frequency deviations. This is necessary in order to convert the generated low-index FM to high-index FM required for broadcasting. The high-index FM is passed through an amplitude limiter to eliminate any amplitude variations that are present in the signal due to the balanced modulator that forms part of the phase modulator. Please note, during conversion from NBFM to WBFM both the carrier and the frequency deviations are increased by factor N, so in order to get the required frequency after multiplication, the original carrier that is generated by the crystal oscillator must be f_c/N so that after multiplication it comes to the required value.

Typical circuits for generating PM and FM (indirect method) are shown in Figure 2.25. The circuits are exactly the same except that in Figure 1.26b there is a resistor R_i and capacitor C_i, which form an integrator that is required for indirect FM generation in order to integrate the modulating signal before being fed to the phase modulator.

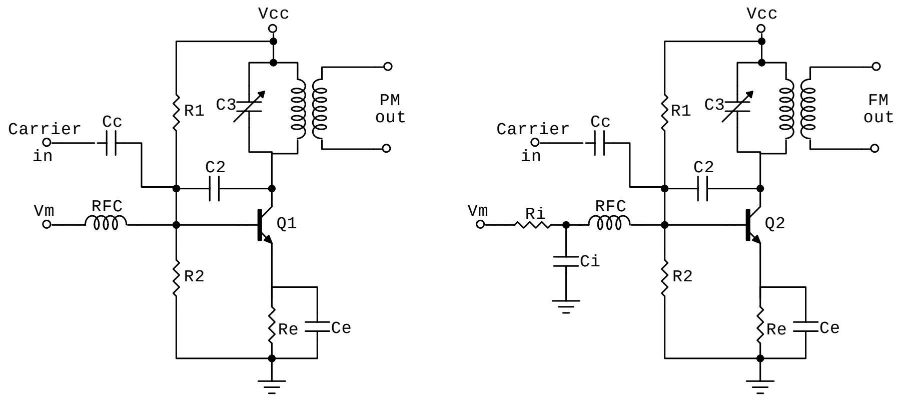


Figure 2.25 Phase Modulators: (a) PM generation; (b) FM generation using PM

In both circuits the carrier frequency from the crystal oscillator is applied simultaneously to both the base of the transistor and through capacitor C_2 to the tuned circuit that is connected to the collector of the transistor. C_2 will introduce a phase-shift that will cause the voltage across the primary of the tuned circuit to lead the base voltage. This phase-shifted carrier together with the voltage that is applied to the base will result in an out-of-phase collector current flowing through the primary of the output transformer. This will cause a voltage that is at some resultant phase to appear across the secondary of the output transformer. Upon application of the modulating signal to the base of the transistor it will add up to the base voltage that is already there from the crystal oscillator. This will result in the base voltage varying with modulating signal, which in turn will cause both amplitude variations and phase shifts at the output. The radio frequency chokes (RFCs) that are connected in series with the modulating signal are used to block the RF carrier frequency that is connected to the base from entering the modulating signal circuits.

2.6 BROADCAST FM

Standard broadcast FM uses WBFM with a maximum deviation of ±75 kHz around the station carrier and guard bands of 25 kHz at the lower and upper ends as shown in Figure 2.26. This makes the overall bandwidth of each station to be 200 kHz, which is very wide compared to the 10 kHz used for AM broadcast. The wider bandwidth allows for a true high-fidelity (Hi-Fi) modulating signal up to 15 kHz as opposed to 5 kHz used for AM.

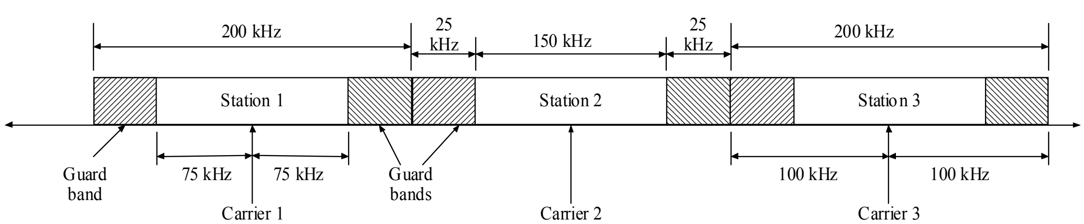
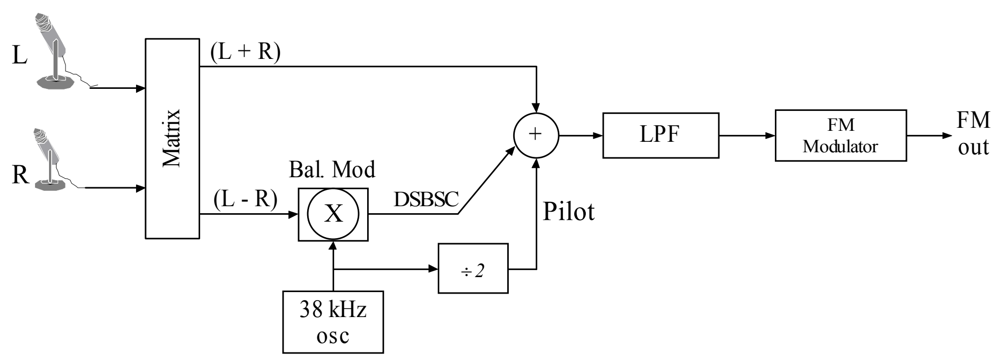


Figure 2.26 Broadcast FM bandwidth allocations for adjacent stations

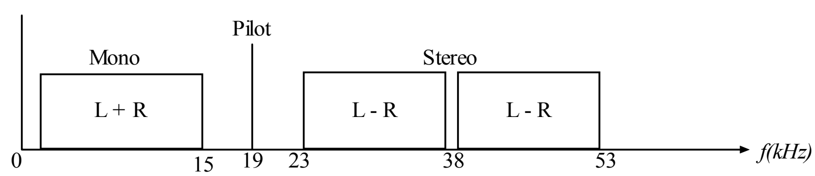
The first FM broadcasts were single-channel monophonic (mono). However, nowadays nearly all FM stations are transmitting two-channel stereophonic (stereo) programs. Furthermore, some FM stations are frequency-division multiplexing additional subcarrier channels on their carrier in order to provide auxiliary subcarrier services such as *radio data system* (RDS) and *subsidiary communication authorization* (SCA). These additional subcarrier channels are commonly known as subcarriers because they carry already modulated analog or digital signals that are used to modulate the main carrier of the station.

2.6.1 FM stereo

Stereo enables the left and the right channels to be simultaneously and independently transmitted. For compatibility with the monaural receivers, the sum of the left and right channels is also transmitted. Figure 2.27 shows the block diagram of a FM stereo generator together with the frequency spectrum of the resultant composite stereo baseband signal.



(a)



(b)

Figure 2.27 FM stereo: (a) Block diagram; (b) Composite baseband spectrum

The audio signals from the left and the right channel are applied to a linear matrix network to produced a L plus R (L + R) signal and L minus R (L – R) signal, where the L – R signal is obtained by inverting the right channel before adding it to the left channel. The L + R is sent as it is in order to provide monaural compatibility, hence it is called mono (M) signal. The L – R signal, which is also called stereo (S) signal, is DSBSC-modulated onto a 38 kHz carrier using a balanced modulator. The frequency translation of the L – R signal to a 38 kHz suppressed carrier is done to prevent spectral overlap since both L + R and L – R are occupying the same audio band. The 38 kHz carrier is also halved to produce a 19 kHz. The L + R signal, the L – R signal and the 19 kHz pilot are fed to a summer where they are added up to produce the composite stereo baseband signal. This composite stereo baseband signal is band-limited by a low pass filter before it is used to frequency-modulated the main station carrier. The transmitted 19 kHz pilot is used at the receiver to generate the coherent carrier that is required in order to demodulate the frequency-translated L – R signal. Sometimes the L + R is artificially delayed in order to maintain phase integrity between it and the L – R signal since there is a time delay introduced to L – R signal as it propagates through the balanced modulator in the transmitter and the product detector in the receiver.

2.6.2 FM subcarrier channels

In a FM broadcast system, each station is assigned a channel bandwidth of 200 kHz, which is wide enough for the transmission of frequencies up to 100 kHz. Since the normal stereo broadcast use frequencies up to 53 kHz, frequencies from 54 kHz up to 100 kHz are free. Subcarrier channels take the advantage of this extra free space in the spectrum to offer auxiliary services such as RDS and SCA for transmission of data and audio, respectively.

Generally, each FM broadcast can accommodate two subcarriers, which are transmitted together with the composite stereo baseband signal using the main station carrier. The amplitude of the subcarriers is kept low in order to avoid adjacent channel interference. The subcarrier used for RDS is 57 kHz while SCA uses a subcarrier of 67 or 92 kHz. These subcarrier frequencies used are chosen to ensure that the frequency ranges used by each subcarrier are beyond the human hearing range, so that there is no interference with the reception of the normal FM broadcast.

2.6.2.1 Radio data system (RDS)

The introduction of stereo around 1961 was followed by another development in FM broadcasting. However, this time it was the introduction of low-speed radio data service, which is aimed at making the radio receiver, especially car radios, to be more user friendly. RDS is a standard from European Broadcasting Union (EBU) for transmitting small digital data alongside the stereo baseband signal using a conventional FM radio broadcast. RDS enables the transmission of several types of information, such as identity of the station, artist/track information, time, radio text used for promotional information, and alternate frequencies to allow the [car stereo] receivers to retune automatically to a different frequency of the same station when the first signal becomes too weak. Additional to the data used by the station, there are other services that can be provided using RDS, such as radio paging, global positioning refinement data, traffic reporting and wide area local control.

The RDS signal is placed on a subcarrier of 57 kHz above the frequency-translated S-signal, as shown in Figure 2.28. Its bandwidth allows for the transmission of data at

1187.5 bps, which is equal to the 57 kHz RDS subcarrier divided by 48. The subcarrier used is equal to the third harmonic of the pilot carrier used for FM stereo.

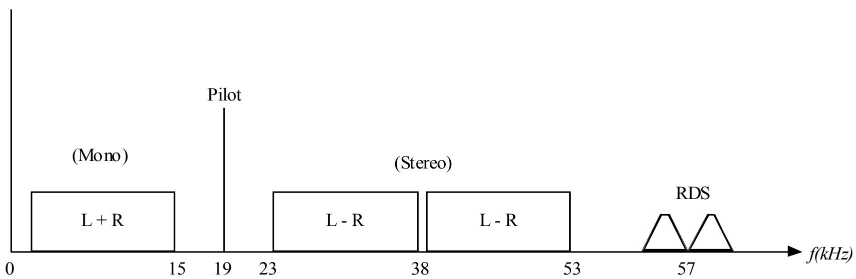


Figure 2.28 Frequency spectrum of the stereo baseband signal incorporating the RDS signal

Furthermore, the 57 kHz subcarrier is synchronized to be either in-phase or in quadrature with the third harmonic of the pilot carrier. This is done to prevent the RDS subcarrier from causing interference or intermodulation with 19 kHz pilot carrier or with the 38 kHz subcarrier used to transmit the S-signal. The data to be transmitted is phase-modulated into the 57 kHz subcarrier using either binary phase-shift keying (BPSK) with a phase deviation of 90 degrees or quadrature phase-shift keying (QPSK). The data is transmitted in groups, which are made up of four 26-bit wide blocks as shown in Figure 2.29 below.

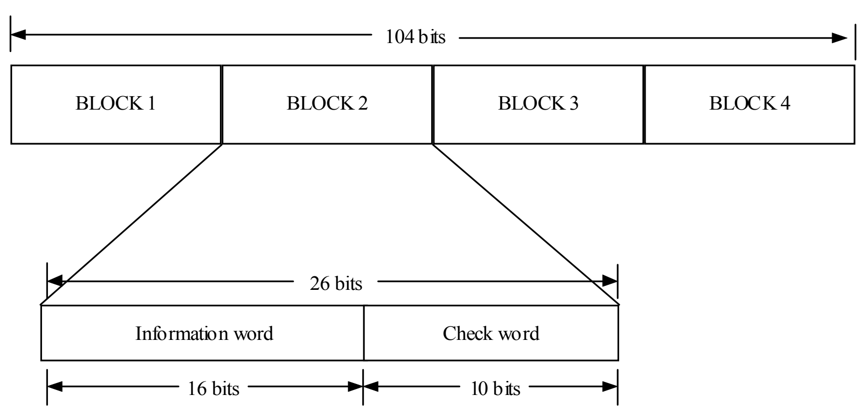


Figure 2.29 RDS group structure

Each block contains 16-bit information word and 10-bit check word. The check word is used for error detection and correction. It is also used for used to provide a method for synchronization. The RDS data normally contains the following fields, which are used for services that are offered by the RDS system:

AF	Alternate frequency	For retuning
CT	Clock time	For synchronizing the receiver clock
EON	Enhanced other networks	Allows the receiver to monitor stations for traffic broadcasts
PI	Program identification	For identifying the station

PS	Program service	8-character display for station name
PTY	Program type	For identifying the program (e.g. news, sport, jazz, etc.)
REG	Regional links	Allows the user to either lock-down the receiver to their region or open it to other regions
RT	Radio text	For transmitting text such as station slogans or artist/track info
TA, TP	Travel announcements Traffic program	Allows for the interruption of tape/CD, or volume increase to enable the listener to listen to travel news/report
TMC	Traffic message channel	Used by GPS navigational devices for dynamic re-routing

2.6.2.2 Subsidiary communication authorization (SCA)

Additional to RDS, other countries also have another auxiliary channel, called subsidiary communication authorization (SCA), which is also frequency-multiplexed on the FM modulating signal. This auxiliary channel is used to broadcast background music to shopping malls, stores, and public buildings. SCA is also used to provide radio reading services for the visually impaired (blind), and foreign language services. It uses a subcarrier frequency of 67 or 92 kHz and the information to be transmitted is frequency-modulated into the subcarrier with deviation of ±7.5 kHz. Since 92 kHz subcarrier is towards the end of the station bandwidth, it suffers from adjacent channel interference resulting in the quality of the signal transmitted using it being compromised, while the signals transmitted on the 67 kHz subcarrier are usually with less interference, hence more clearer than those on 92 kHz. This has resulted in 67 kHz to be the most popular subcarrier frequency for SCA than 92 kHz. The location of SCA in the FM modulating signal (without RDS) is shown in Figure 2.30 and the location of the SCA in the composite FM modulating signal with RDS is shown in Figure 2.31.

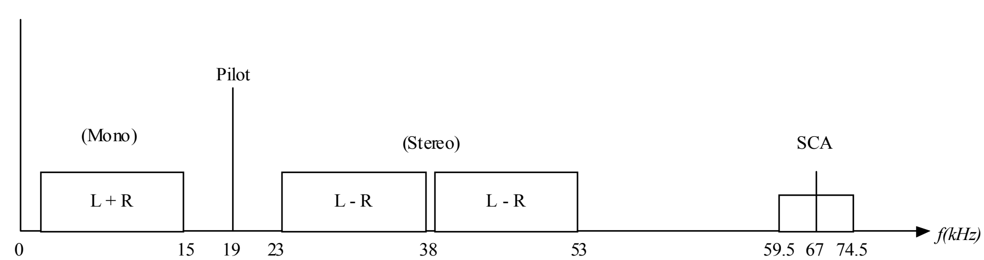


Figure 2.30 Frequency spectrum of the stereo baseband signal incorporating the SCA signal

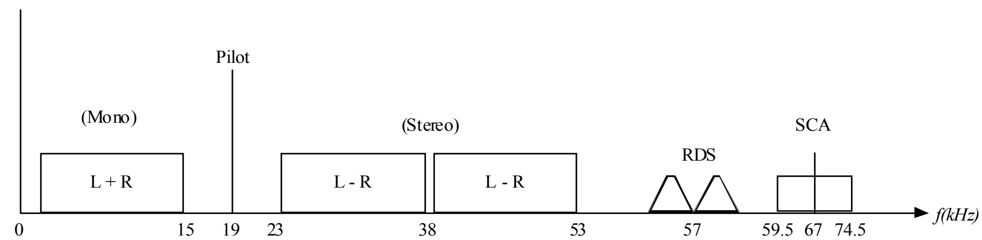


Figure 2.31 Frequency spectrum of stereo baseband signal incorporating the RDS and SCA signal

2.7 NOISE IN FM

Though FM is less susceptible to noise than AM its noise suppression capability decreases with higher frequency components of the modulating signal. This results in the output noise voltage increasing linearly with frequency. One of the causes of this is that high frequency components of the modulating signal tend to be of lower amplitude than the lower frequency components, which means that high frequency components result in low-level deviation and subsequently, low noise reduction capability.

To counter this problem, the high frequency components of the modulating signal are boosted before transmission, and then reduced by a corresponding amount in the receiver. This increase of the amplitude of high frequency components of the modulating signal before transmission is called *pre-emphasis*, and the corresponding reduction in the receiver is called *de-emphasis*. Pre-emphasis results in an increase of the modulation index for high frequency components of the modulating signal. This in turn will result in a high deviation and subsequently, improve noise reduction capability even at high frequencies. Figure 2.32 shows the circuit for pre-emphasis and de-emphasis, while Figure 2.33 shows the pre-emphasis/de-emphasis curve.

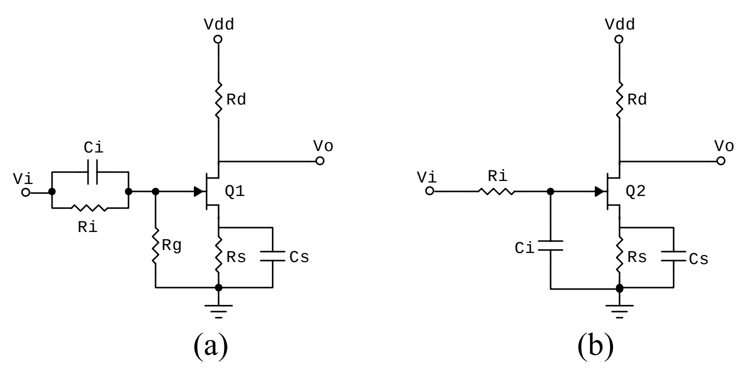


Figure 2.32 Pre-emphasis and (b) De-emphasis networks

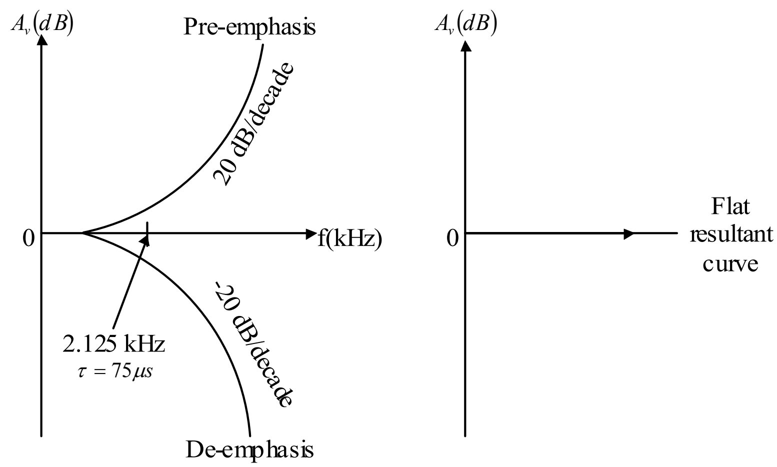


Figure 2.33 Pre-emphasis/de-emphasis curve

The amount of pre-emphasis and de-emphasis is determined by the RC network, which is chosen to give a time constant $\tau = R_iC_i = 75$ µs in South Africa and North America while for other parts of the world the time constant is 50µs. Pre-emphasis and de-emphasis are only applied to the composite stereo baseband not to RDS and SCA.

2 DIGITAL RADIO TRANSMITTER SYSTEMS

Digital radio systems use digital modulation techniques such as quadrature amplitude modulation (QAM), quaternary or quadrature phase-shift keying (QPSK) or Gaussian minimum-shift keying (GMSK). In these modulation schemes the analog carrier is modulated by a digital bit stream.

3.1 QAM MODULATOR

Quadrature amplitude modulation is a modulation method that enables the sending of two separate or uniquely different channels of information over the same frequency. This is accomplished by taking the generated carrier signal and shift it to create two carriers: sine and cosine, which are in quadrature with each other (i.e. 90^0 out-of-phase). The resultant inphase carrier (sine) is used to carry one signal and the quadrature (cosine) carrier is used to carry the other signal. That is, two channels: inphase (I) and quadrature (Q) channels are created to transmit two separate or uniquely different signals over the same frequency, thereby doubling the effective bandwidth. Since the two carriers are in quadrature, chances of intermodulation interferences are minimized. QAM is not only limited to digital modulation inputs, but can also be used to quadrature multiplex analog signals as well. Figure 1 shows a QAM with digital modulation inputs while Figure 2 shows a QAM with analog modulation inputs. The two QAM diagrams are almost the same except that when working with analog signals, there is no need for DAC and accompanying reconstruction LPFs since the signals are already in the analog format.

In Figure 3.1, the *I* and *Q* data streams are converted from digital to analog by digital-to-analogue (DAC) converters. Each output of the DAC is filtered using low-pass (reconstruction or interpolating) filters before being fed into a balanced modulator. The oscillator generates a carrier which is also applied to the balanced modulators; however, the carrier applied to the balanced modulator of the Q-channel is first phase-shifted by 90^0 using a phase-shifting circuit. In the balanced modulators the modulating inputs are used to modulate the carrier signals to produce two DSBSC signals. The outputs of the two balanced modulators are summed up to produce a QAM signal, which is a single signal that contains both the *I* and the *Q* information.

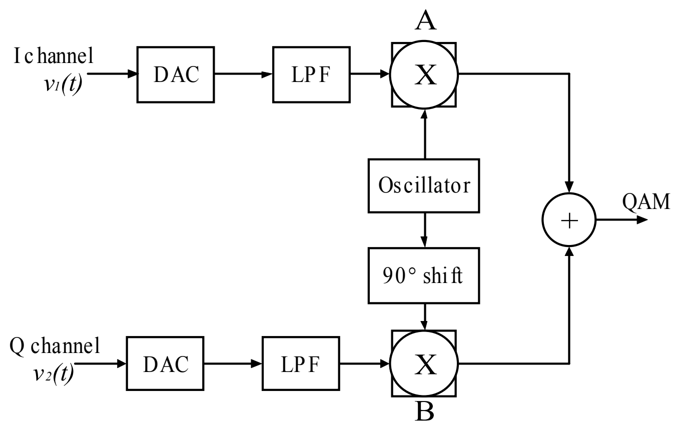


Figure 3.1 QAM Modulator with digital input signals

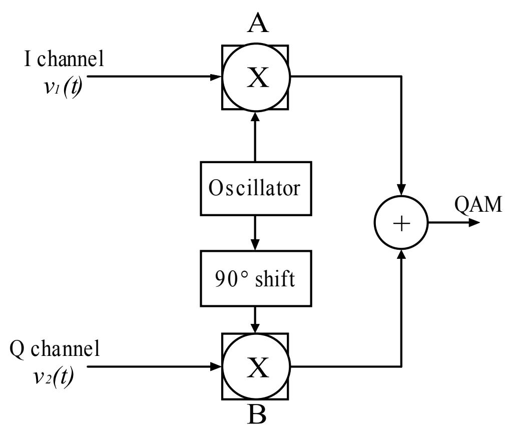


Figure 3.2 QAM Modulator with analog input signals

Analog quadrature multiplexing is used in color TV to transmit the color information.

On both cases the output QAM signal is

$$v_{QAM} = v_1(t)\sin w_c t + v_2(t)\cos w_c t \qquad (3.1)$$

3.2 QPSK MODULATOR

Quaternary or quadrature phase-shift keying (QPSK), which is also known as quadriphase PSK, is a multilevel modulation scheme which offers twice as many data bits per carrier change than BPSK. Since QPSK is a multilevel PSK, it conveys data by changing or modulating the phase of the carrier signal. However, in QPSK, the carrier undergoes four changes in phase during modulation as shown on the constellation diagram of Figure 3.3; that is, it uses four symbols and each symbol represents two binary bits or dibit. This enables the carrier to transmit double the number of bits per time, thus doubling the effective bandwidth of the system. Figure 3.3 shows the plot of the four symbols on the rectangular space; that is, the constellation diagram of QPSK.

The four points are equispaced around a circle; that is, the four symbols are spaced with a phase shift differing by $360^0/4 = 90^0$ on the circle.

The conceptual QPSK modulator is as shown in Figure 3.4. However, one NRZ encoder can be placed before the serial-to-parallel converter or the dividing circuit, instead of using two after it. Other modification that can be done to Figure 3.4 is to include pre-modulation LPFs (with root-raised cosine (RRC) response) between the NRZ encoders and the balanced modulators to shape the pulses so that we can reduce their bandwidth and thereby reducing the overall bandwidth required to transmit the resultant QPSK signal. The incoming digital data stream is split into two bit streams, which are the even and odd streams by a serial-to-parallel converter. The even bits are sent to the Q-channel while the odd bits are sent to the I-channel. These odd and even bits are further converted from unipolar to bipolar sequence by converting all 0 into -1 using the NRZ method before they are applied to a pair of balanced modulator. The oscillator generates a carrier frequency, which is also applied to the balanced modulators; however, the carrier applied to the modulator of the Q-channel is first phase-shifted by 90^0 using a phase-shifting circuit before it is applied to the balanced modulator. In the balanced modulators the odd and even sequence are used to modulate the carrier signals. The outputs of the two balanced modulators are summed up to produce a QPSK signal.

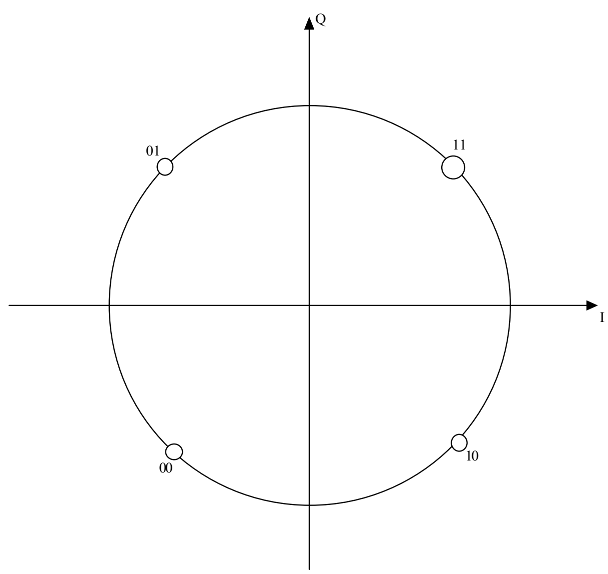


Figure 3.3 QPSK constellation diagram

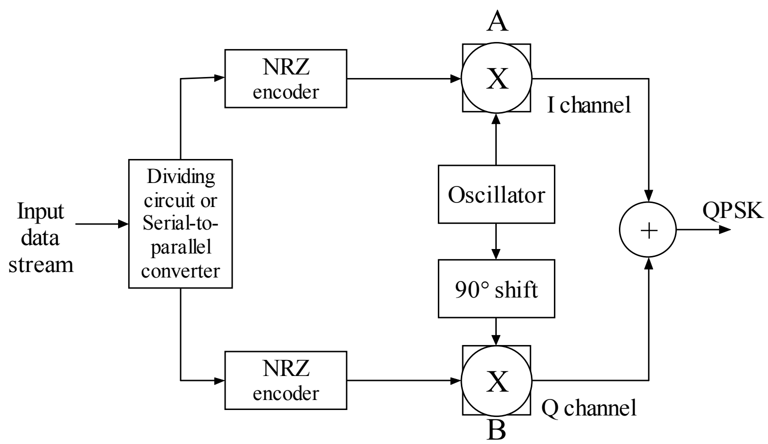


Figure 3.4 QPSK Modulator

Mathematical the QPSK can be analyzed as follows:

The output of the inphase modulator is

$$s_I(t) = \sqrt{\frac{2E}{T}} \sin\left(w_c t + \frac{\pi}{4} \right) \tag{3.2}$$

Where E is the symbol energy and T is symbol duration. And the output for quadrature modulator is

$$s_Q(t) = \sqrt{\frac{2E}{T}} \cos\left(w_c t + \frac{\pi}{4} \right) \tag{3.3}$$

Summing the two we get

$$s_{QPSK}(t) = s_I(t) + s_Q(t) \tag{3.4}$$

$$= \sqrt{\frac{2E}{T}} \sin\left(w_c t + \frac{\pi}{4} \right) + \sqrt{\frac{2E}{T}} \cos\left(w_c t + \frac{\pi}{4} \right)$$

$$= \sqrt{\frac{2E}{T}} \cos\left(w_c t + \frac{\pi}{4} + \phi(t) \right), \quad \phi = 0, \frac{\pi}{2}, \pi, -\frac{\pi}{2}. \tag{3.5}$$

Sometimes Equation (3.5) is written as follows

$$s_{QPSK}(t) = \sqrt{\frac{2E}{T}} \cos\left(w_c t + \frac{\pi}{4}(2i-1) \right), i = 1, 2, 3, 4. \tag{3.6}$$

Both Equations (3.5) and (3.6) yield four phases: π/4, 3π/4, 5π/4 (or -3π/4) and 7π/4 (or –π/4), as required.

3.3 GMSK MODULATOR

Gaussian Minimum Shift Keying (GMSK) is a form of digital modulation, which is a derived from minimum shift keying (MSK). Unlike QPSK that uses discrete phases to transmit data, both MSK and GMSK are continuous-phase frequency-shift keying (CPFSK) modulation techniques, and there are no phase discontinuities or discrete jumps at the edge of the symbols. Both MSK and GMSK encode each bit as half sinusoid with a frequency separation of one-half the bit rate; that is, the frequency difference between the logical 'high' and logical 'low' states is always equal to half the bit rate, and these frequency changes occur at carrier zero crossing points. Thus, the modulation index is 0.5. However, compared to other forms of phase-shift keying, GMSK modulation uses the spectrum efficiently, thereby making it to be one of the most popular modulation technique used in a number of radio communication applications such as Digital European Cordless Telephone (DECT), Global Speciale Mobile (GSM) cellular, Cellular Digital Packet Data (CDPD), etc. Another advantage of using GSMK is that, since there is no element of signal that is carried by amplitude variations, it can be amplified using non-linear amplifier without any distortion, which means less power consumption. Furthermore, not carrying any information in amplitude variations makes the GMSK signal to be less susceptible (or more immune) to noise.

GMSK is derived by passing the modulating signal through a low pass filter prior to the MSK modulation of the carrier as shown in Figure 3.5. The function of this pre-modulation LPF is to reduce the bandwidth required to transmit the information. This is accomplished by removing the sharp 'one-to-zero' and 'zero-to-one' transitions in the input binary data, which would otherwise result in a spectrum rich in harmonic content that will occupy a considerable bandwidth if they are allowed to modulate the carrier signal.

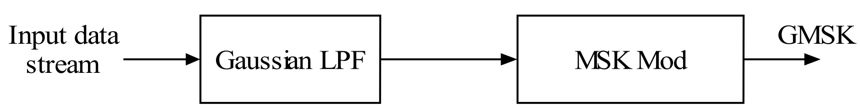


Figure 3.5 Derivation of GMSK from MSK

In order for the LPF to remove these sharp transitions in the input data, it must have a sharp cutoff, a narrow-bandwidth and an impulse response with no overshoots or ringing, and the filter that meets all these requirements is Gaussian filter. The Gaussian-shaped response of this filter is the one that makes the MSK modulator to produce a GMSK instead of a normal MSK output. The cutoff frequency of the pre-modulation Gaussian filter is determined by the bit rate and normalized bandwidth or *BT factor* (BT); that is,

$$f_{-3dB} = BT(Bitrate)$$

Re-arranging we get

$$BT = \frac{f_{-3dB}}{Bitrate} = T_b f_{-3dB} = T_b B_b \tag{3.7}$$

Where T_b is the bit duration or bit period and B_b is the signal bandwidth, which is determined by the cutoff frequency of the pre-modulation filter. Usually BT values of 0.3 and 0.5 are used; for example, GSM use $BT = 0.3$ while DECT and CDPD use $BT = 0.5$.

Basically there are two methods that can be used to generate GMSK signal: using a frequency modulator with a modulating index equal to 0.5, or using a quadrature modulator. The former is not normal used due to the drift of the tolerances of the components used for FM modulation, thus leaving the latter to be the most commonly used method for generating the GMSK signal, hence it will be the one considered for the discussion here. Figure 3.6 shows the generation of GMSK using a quadrature modulator.

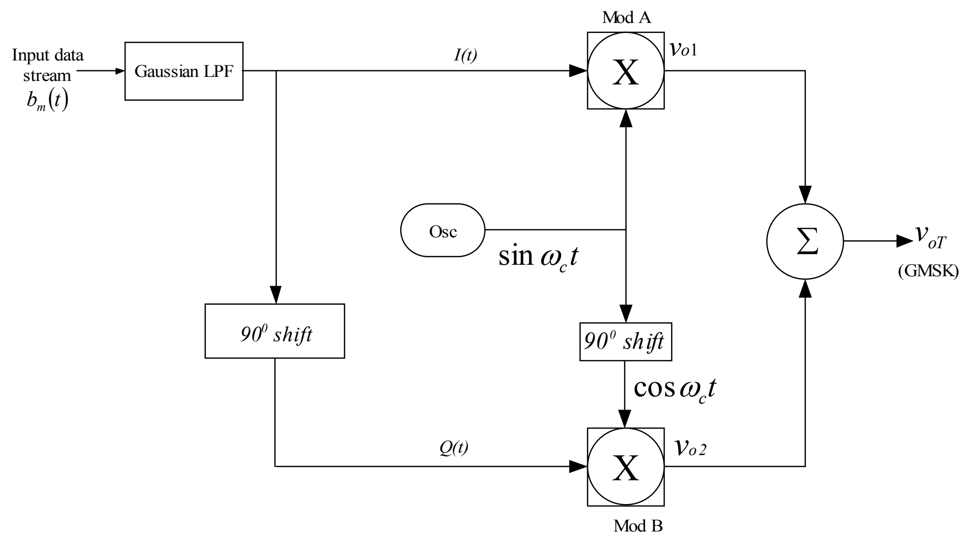


Figure 3.6 GMSK modulator

The Gaussian LPF is used for pre-modulation filtering in order to make the input data stream to be less sharp so that the resultant signal can occupy less spectrum after modulation. After filtering, the data is separated into inphase and quadrature components using a phase shifting network before it is applied to modulators A and B, respectively. The oscillator generates a carrier that is applied to modulator A inphase. A sample of the generated carrier is also passed through a 90 degrees phase-shift network prior being fed to the modulator B. In modulator A, the inphase input data is used to modulate the inphase carrier. Similarly, in modulator B, the phase-shifted input data is used to modulate the phase-shifted carrier. Then the outputs of the two modulators are summed up to produce a GMSK signal. The output of a GMSK modulator can be expressed as

$$\begin{aligned} GMSK(t) &= \cos w_c t \cos b_m t - \sin w_c t \sin b_m t \\ &= \cos[w_c t + b_m(t)] \end{aligned} \tag{3.8}$$

4. NOISE, INTERFERENCE AND DISTORTION IN RADIO COMMUNICATION

Radio communication systems use atmosphere as a communication channel to link the transmitter and the receiver. Atmosphere is a hostile environment, which is characterized by continuous changes in its parameters. These changes affect the radio signals as they propagate from the transmitter to the receiver through the channel. The result will be the signal being transmitted will affected by noise, interference and distortion.

4.1 NOISE

Noise is a random, unpredictable electrical signal, which obscures the required signal. Noise is predominantly generated from within the communications systems itself; that is, within the transmitter, repeater or receiver. Internal generated noise includes thermal noise from the resistors, partition or shot noise from transistors and diodes, and flicker noise. Additional to internal generated noise, noise can be also be generated from outside the system; that is, externally. External noise is mainly constituted by the atmospheric noise, which is noise arising from electromagnetic radiation from solar and galactic sources. Additional to atmospheric noise, external noise can also be man made.

Noise can be classified as either white or colored depending on the noise power spectral density versus frequency. Where white noise is a type of noise that has a flat power spectral density over all frequencies of interest, and the colored noise has only a nearly-flat spectral density over a limited bandwidth while the overall spectral distribution is non-uniform. Based on these characteristics, the noise is usually modeled as White Gaussian Noise (WGN) and Bandlimited White Gaussian Noise (BWGN), respectively. However, due to the multiplicity of noise sources in communication systems, noise is usually classified as Additive White Gaussian Noise (AWGN).

4.1.1 Thermal or Johnson Noise

Thermal noise is a result of random movement of electrical charge within a conductor or resistor. This movement produces a current through the resistor, which in turn produces voltage across the resistor. The voltage produced is directly proportional to the temperature and bandwidth, and is called thermal, Johnson or Nyquist noise. The value of the thermal noise voltage and the power are given by Equations (4.1) and (4.2), respectively.

$$e_n = \sqrt{4kTBR} \tag{4.1}$$

$$P_n = kTB \tag{4.2}$$

Where k = Boltzmann's constant = $1.38 \times 10^{-23} J/^0 K$

T = absolute temperature in degrees Kelvin (^{o}K = ^{o}C + 273^o)

B = Bandwidth of the frequencies to be measured

The frequency content of thermal noise is spread equally throughout the entire usable spectrum; hence, sometimes it is referred to as white noise.

4.1.2 Short Noise

Short noise, which is sometimes referred to as partition noise, is a kind of noise that is generated in semiconductor junctions when the electrons with an average current I_{dc} and charge q cross the potential barrier between the N-type and P-type materials that constitute the semiconductors. The value of short noise is given by

$$I_n = \sqrt{2qI_{dc}B} \tag{4.3}$$

4.1.3 Flicker noise

Flicker noise, which is sometimes referred to as excess noise or *1/f* noise, is a type of transistor noise that occurs at low or audio frequencies. It is inversely proportional to frequency and directly proportional to dc current levels and temperature. Flicker noise is having a spectrum that falls off steadily as the frequency increases; that is, it has a pink spectrum, hence it is also referred to as pink noise.

4.1.4 Signal-to-Noise Ratio and Noise Figure

The signal-to-noise ratio (SNR or S/N) and noise figure (NF) are used to express the performance of a system in the presence of noise. SNR is the ratio of the signal power to noise power at the same point. That is,

$$SNR = S/N = \frac{Signal\ Power}{Noise\ Power} = \frac{P_s}{P_n} \tag{4.4a}$$

And in decibel, SNR is

$$SNR = S/N = 10\log\frac{P_s}{P_n}\ \text{dB} \tag{4.4b}$$

Noise figure, which is sometimes referred to as noise factor, is the ratio of the SNR at the input of the receiver to that at the output; that is,

$$NF = \frac{Input\ SNR}{Output\ SNR} \tag{4.5a}$$

And in decibels, NF is

$$NF = 10\log\frac{Input\ SNR}{Output\ SNR}\ \text{dB} \tag{4.5b}$$

SNR is useful in identifying the noise content at a specific point, whereas NF is useful to specify exactly how much noise has the device introduced to the signal.

4.2 INTERFERENCE

Interference can be defined as any unwanted signal that hampers, hinders, obstructs or impedes the reception of the wanted signal. In some cases interference also results in the alteration or modification of the signal being transmitted. Though interference is often distinguished from noise, it is sometimes classified as a kind of man-made noise. Usually, interference occurs while the signal is traveling between the source and the receiver; however, there are cases where the interference takes place within the receiver itself. That is, interfering signal can either be generated within or outside the receiver. The internal interference, include interference from oscillators and power supply. External interference comes from natural electrical phenomena as well as other man-made sources such as electrical machines and other transmitting RF sources. Typical RF interference includes adjacent channel interference; co-channel interference, intersymbol interference, and image channel interference.

4.2.1 Adjacent Channel Interference

Adjacent channel is a channel immediately above or below the selected channel. If this channel is very close to the selected channel and the bandwidth of the RF stage and/or the IF stage is wide then part of adjacent channel will be allowed into the receiver and thus causing the two stations to be heard in your loudspeaker. Adjacent channel interference can also occur as a result of one channel occupying more than its allocated bandwidth, thereby overlapping into its adjacent channels.

4.2.2 Co-channel Interference

Co-channel interference is interference between two or more channels that are operating in the same area or proximity that are using the same carrier signal.

4.2.3 Delay Spread and Intersymbol Interference

Delay spread is the smearing or the spreading out of the received signal due to the multiple paths that it has followed during propagation from transmitter to the receiver. In digital communication systems, the delay spread causes one symbol slot to be spread out over neighbouring symbol slots thus resulting in part of the symbol energy in consecutive symbols to overlap with neighbouring symbols causing intersymbol interference. This overlap makes it difficult for the receiver to differentiate the current symbol from the overlapping adjacent symbols.

4.2.4 Image Channel Interference

An image channel is an unwanted channel that will interfere with the reception of the wanted channel. The RF stage is responsible for attenuating the image channel. However, if not attenuated it will find its way to the mixer and once it reaches the mixer and get mixed with the frequency from the local oscillator, it will also produce an frequency equals to the intermediated frequency, which when demodulated and amplified will be heard in the background while listening to the wanted station. The frequency of the image channel is given by

$$f_{image} = f_i = f_s \pm 2 f_{IF} \tag{4.6}$$

Where f_s = the selected frequency or the wanted channel
f_{IF} = intermediate frequency

And the image frequency rejection is given by

$$A_r = \frac{1}{\sqrt{1+\left(yQ''_p\right)^2}} \qquad \text{where} \quad y = \left(\frac{f_i}{f_o} - \frac{f_o}{f_i}\right) \tag{4.7}$$

4.3 DISTORTION

Imperfections in the processing hardware and the defects in the channel usually cause distortion. In radio communication, distortion includes phase shifts, gain variations, and frequency shifts.

4.3.1 Phase distortion

Phase distortion can be defined as a non-flat or nonlinear phase response in processing hardware or the channel in a communication system. It involves frequency-dependent phase variations as the signal is being processed by amplifiers, filters, mixers and other components of the systems. Phase distortion can also be due to the communication channel itself; especially where the signal travels different path from the transmitter to the receiver. These frequency-dependent phase variations usually give rise to differential group delay, which is the rate of change of phase shift with frequency.

4.3.2 Gain distortion

Gain distortion can be defined as nonlinear amplitude relationship between input and output of an amplifier, filter, mixer, or a channel in a communication system. This results in the power of some components of the transmitted signal being more emphasized than the others.

4.3.3 Doppler Shift

When either or both transmitter and receiver are moving relative to each other, the received signal's frequency is different from that of the transmitter. This change in the frequency of the transmitted signal is called *Doppler effect*, and the amount of change or shift in the frequency of the received signal depends on the relative motion between the transmitter and the receiver (i.e. direction of travel), the carrier frequency used and on the speed of the transmitter and/or the receiver. For example, when they are moving towards each other, the frequency of the received signal is higher than that of the transmitter. Conversely, when the two are moving away from each other, the frequency of the received signal is less than that of the transmitter. The Doppler shift is written as

$$\Delta f = \frac{v}{\lambda}\cos\theta = \frac{vf_c}{v_c}\cos\theta \tag{4.8}$$

Where v is the relative speed of the transmitter and the receiver, f_c is the carrier frequency, v_c is the speed of light, and θ is the angle between the signal arrival and the direction of travel.

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