UNIT II
ANGLE MODULATION

FREQUENCY & PHASE MODULATION

Introduction

Besides using the amplitude of carrier to carrier information, one can also use the angle of a carrier to carrier information. This approach is called angle modulation, and includes frequency modulation (FM) and phase modulation (PM). The amplitude of the carrier is maintained constant. The major advantage of this approach is that it allows the trade-off between bandwidth and noise performance. he other type of modulation in continuous-wave modulation is the **Angle Modulation**. Angle Modulation is the process in which the frequency or the phase of the carrier varies according to the message signal. This is further divided into frequency and phase modulation.

- Frequency Modulation is the process of varying the frequency of the carrier signal linearly with the message signal.
- Phase Modulation is the process of varying the phase of the carrier signal linearly with the message signal.

Frequency Modulation

In amplitude modulation, the amplitude of the carrier varies. But in Frequency Modulation (FM), the frequency of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal. The amplitude and the phase of the carrier signal remains constant whereas the frequency of the carrier changes. This can be better understood by observing the following figures Figure 2.1.1, Figure 2.1.2, Figure 2.1.3 represents Base band Signal , Carrier Signal and FM Signal.

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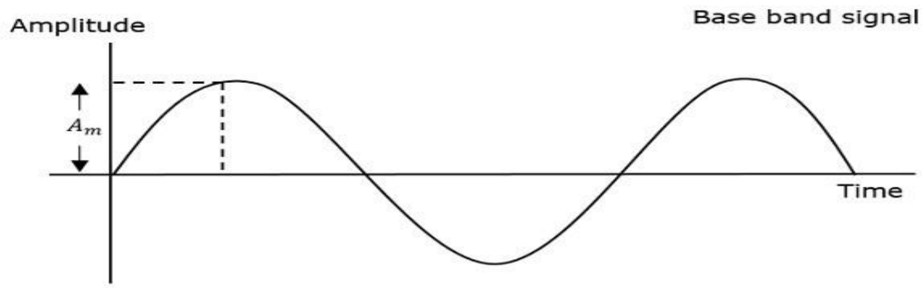

Figure 2.1.1 base Band Signal

Diagram Source Electronics Tutorials

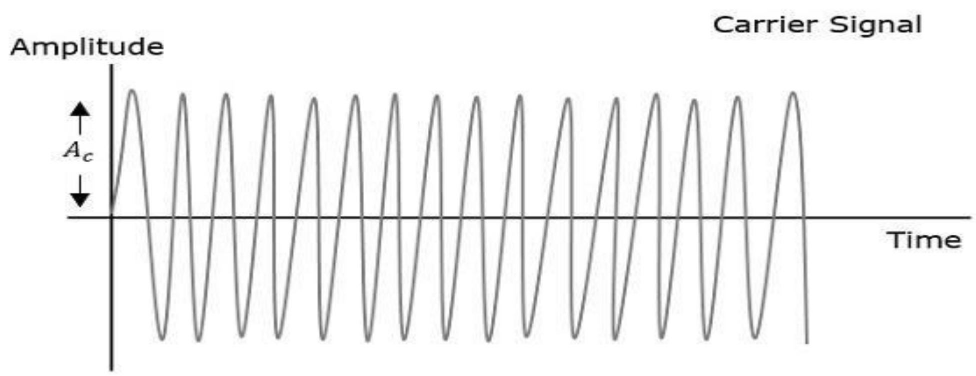

Figure 2.1.2 Carrier Signal

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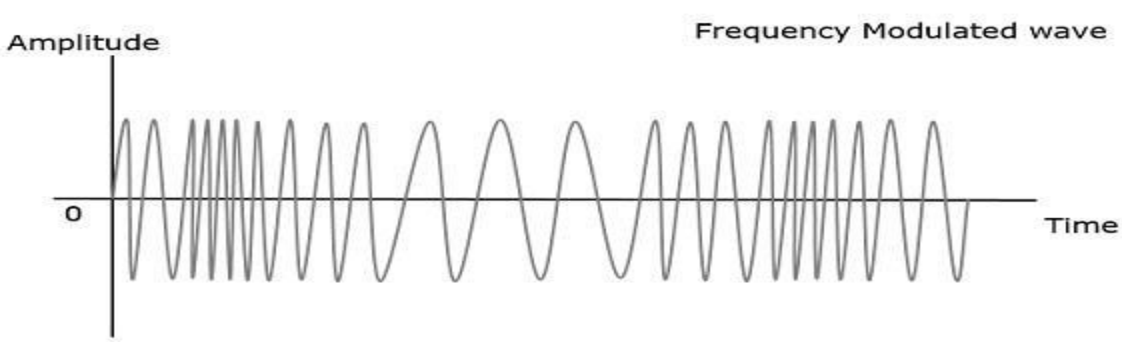

Figure 2.1.3 FM Signal

Diagram Source elprocus.com

EQUATIONS OF FM AND PM WAVES

An angle modulated signal can be written as

$$s(t) = A\cos\theta(t) \quad (1)$$

where $\theta(t)$ is usually of the form $\theta\ t = 2\pi f_c t + \phi(t)$ and fc is the carrier frequency. The signal $\phi(t)$ is derived from the message signal m(t) . If $\phi\ t = k_p m(t)$ for some constant kp ,the resulting modulation is called phase modulation. The parameter kp is called the phase sensitivity.In telecommunications and signal processing, frequency modulation (FM) is the encoding of information in a carrier wave by varying the instantaneous frequency of the wave. (Compare with amplitude modulation, in which the amplitude of the carrier wave varies, while the frequency remains constant.) Frequency modulation is known as phase modulation when the carrier phase modulation is the time integral of the FM signal.

Modulation index:

As in other modulation systems, the value of the modulation index indicates by how much the modulated variable varies around its unmodulated level. It relates to variations in the carrier frequency. The modulation index of FM is defined as the ratio of the frequency deviation of the carrier to the frequency of the modulating signal

$$m_f = \text{Modulation Index of FM} = \Delta f/f_m \quad (2)$$

The FM equation include the following

$$v = A\sin[\omega_c t + (\Delta f / f_m)\sin\omega_m t]$$

$$= A\sin[\omega_c t + m_f \sin\omega_m t] \quad (3)$$

A = Amplitude of the FM signal. Δf = Frequency deviation

mf = Modulation Index of FM

$m_f = \Delta f/f_m$

mf is called the **modulation index** of frequency modulation.

$m = 2\pi\ fm\ \omega c = 2\pi\ fc$ (4)

The Bandwidth of Frequency Modulation Signal

Bandwidth is one of the main elements of FM signal. In FM signal, the sidebands will extend either side which will extend to infinity; however, the strength of them drops away. Auspiciously, it is the potential to restrict the BW of an FM signal without changing its value excessively. Recall, the bandwidth of a complex signal like FM is the difference between its highest and lowest frequency components, and is expressed in Hertz (Hz). Bandwidth deals with only frequencies. AM has only two sidebands (USB and LSB) and the bandwidth was found to be 2 fm. In FM it is not so simple. FM signal spectrum is quite complex and will have an infinite number of sidebands as shown in the Figure**.** This figure 2.1.4 gives an idea, how the spectrum expands as the modulation index increases. Sidebands are separated from the carrier by fc ± fm, fc ± 2fm, fc ± 3fm, and so on.

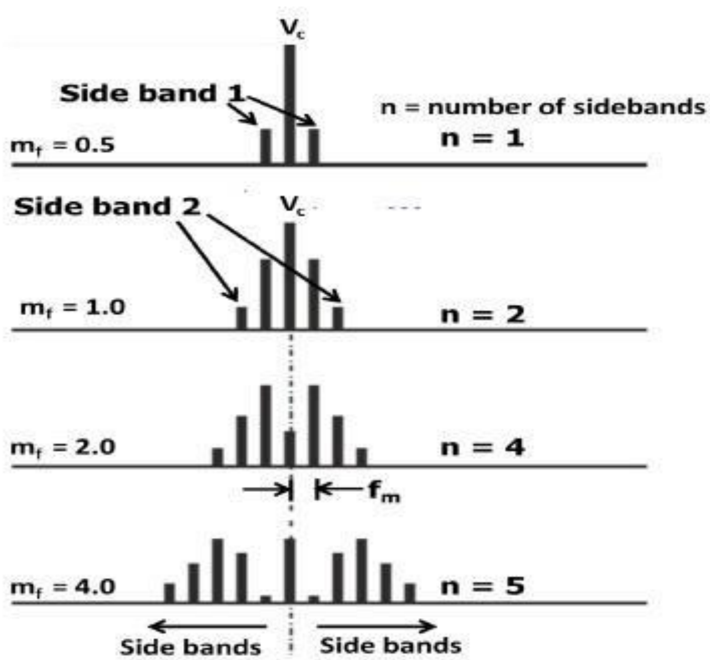

Figure 2.1.4 The Band width of FM Signal

Diagram Source elprocus.com

Only the first few sidebands will contain the major share of the power (98% of the total power) and therefore only these few bands are considered to be significant sidebands.
As a rule of thumb, often termed as Carson's Rule, 98% of the signal power in FM is contained within a bandwidth equal to the deviation frequency, plus the modulation frequency-doubled.

Carson's rule: Bandwidth of FM BWFM

$$= 2\,[\,\Delta f + fm\,].$$

$$= 2\,fm\,[\,mf + 1\,]$$

$$BT = 2(\delta + fm_{(max)}) \quad (5)$$

With a tone-modulated FM wave, if the modulation frequency is held constant and the modulation index is increased, the (non-negligible) bandwidth of the FM signal increases but the spacing between spectra remains the same; some spectral components decrease in strength as others increase. If the frequency deviation is held constant and the modulation frequency increased, the spacing between spectra increases.

Frequency modulation can be classified as narrowband if the change in the carrier frequency is about the same as the signal frequency, or as wideband if the change in the carrier frequency is much higher (modulation index >1) than the signal frequency. [6] For example, narrowband FM is used for two way radio systems such as Family Radio Service, in which the carrier is allowed to deviate only 2.5 kHz above and below the center frequency with speech signals of no more than 3.5 kHz bandwidth. Wideband FM is used for FM broadcasting, in which music and speech are transmitted with up to 75 kHz deviation from the center frequency and carry audio with up to a 20-kHz bandwidth.

The frequency of the modulated wave remains constant as the carrier wave frequency when the message signal is at zero. The frequency increases when the message signal reaches its maximum amplitude.

Which means, with the increase in amplitude of the modulating or message signal, the carrier frequency increases. Likewise, with the decrease in the amplitude of the modulating signal, the frequency also decreases.

Mathematical Representation

Let the carrier frequency be fc

The frequency at maximum amplitude of the message signal = $f_c + \Delta f$

The frequency at minimum amplitude of the message signal = $f_c - \Delta f$

The difference between FM modulated frequency and normal frequency is termed as Frequency Deviation and is denoted by Δf.

The deviation of the frequency of the carrier signal from high to low or low to high can be termed as the Carrier Swing.

$$\text{Carrier Swing} = 2 \times \text{frequency deviation}$$
$$= 2 \times \Delta f$$

Equation for FM WAVE

The equation for FM wave is

$$s(t) = A_c \cos[\omega_c t + 2\pi k_f m(t)] \qquad (6)$$

Where,

A_c = the amplitude of the carrier

ω_c = angular frequency of the carrier = $2\pi f_c$

$m(t)$ = message signal

FM can be divided into Narrowband FM and Wideband FM.

Narrowband FM

The features of Narrowband FM are as follows −

- This frequency modulation has a small bandwidth.
- The modulation index is small.
- Its spectrum consists of carrier, USB, and LSB.

This is used in mobile communications such as police wireless, ambulances, taxicabs, etc.

Wideband FM

The features of Wideband FM are as follows −

- This frequency modulation has infinite bandwidth.
- The modulation index is large, i.e., higher than 1.
- Its spectrum consists of a carrier and infinite number of sidebands, which are located around it.
- This is used in entertainment broadcasting applications such as FM radio, TV, etc.

Phase Modulation

- In frequency modulation, the frequency of the carrier varies. But in Phase Modulation (PM), the phase of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.
- The amplitude and the frequency of the carrier signal remains constant whereas the phase of the carrier changes. This can be better understood by observing the following Figure 2.1.5 Base Band Signal, Figure 2.1.6 , Carrier Signal and Figure 2.1.67 Phase Modulated Sigal

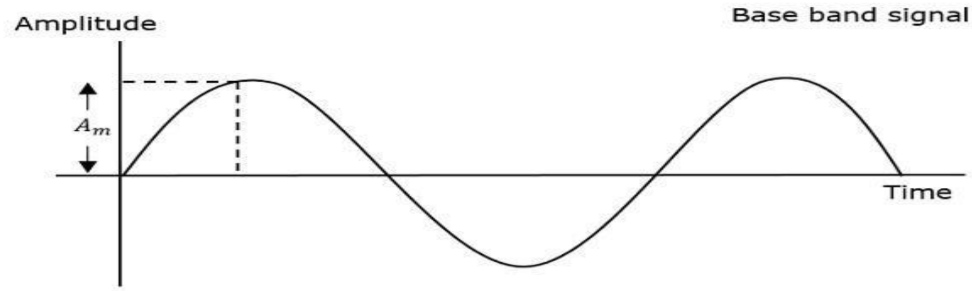

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Figure 2.1.5 Base Band Signal

Diagram Source Electronics Tutorials

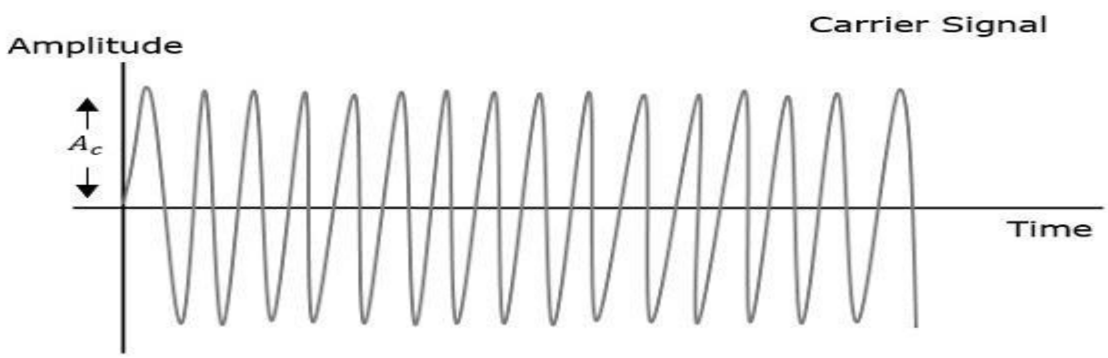


Figure 2.1.6 Carrier Signal

Diagram Source Source Brain Kart

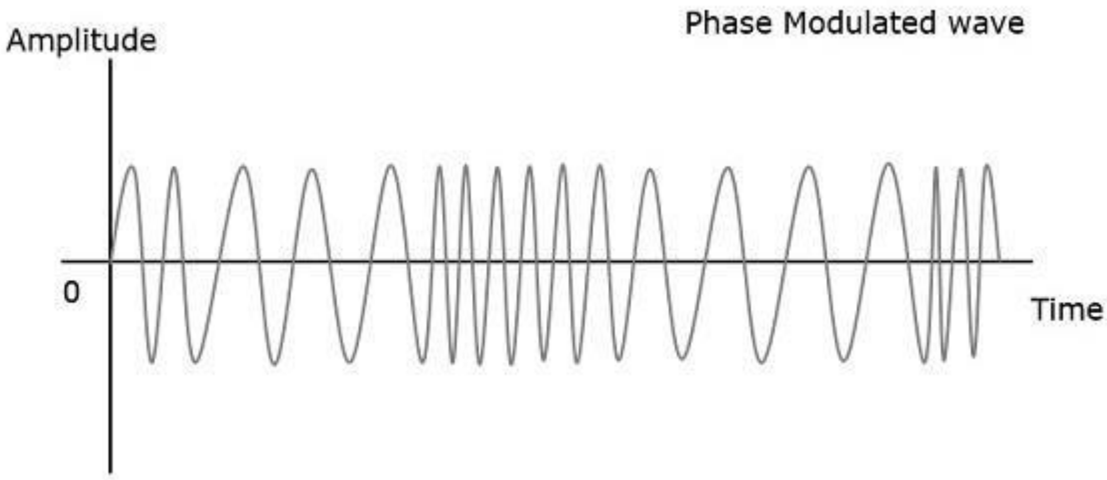


Figure 2.1.7 Phase Modulated Signal

Diagram Source Brain Kart

The phase of the modulated wave has got infinite points where the phase shift in a wave can take place. The instantaneous amplitude of the modulating signal, changes the phase of the carrier. When the amplitude is positive, the phase changes in one direction and if the amplitude is negative, the phase changes in the opposite direction.

- In frequency modulation, the frequency of the carrier varies. Whereas, in Phase Modulation (PM), the phase of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.
- So, in phase modulation, the amplitude and the frequency of the carrier signal remains constant. This can be better understood by observing the following figures.

The phase of the modulated wave has got infinite points, where the phase shift in a wave can take place. The instantaneous amplitude of the modulating signal changes the phase of the carrier signal. When the amplitude is positive, the phase changes in one direction and if the amplitude is negative, the phase changes in the opposite direction.

Mathematical Representation

The equation for instantaneous phase ϕiϕi in phase modulation is

$$\phi i = kpm(t) \qquad (7)$$

Where,

- kp is the phase sensitivity
- m(t) is the message signal

The standard equation of angle modulated wave is

$$s(t)=Accos(2\pi fct+\phi i) \qquad (8)$$

Substitute, ϕi value in the above equation.

$$s(t)=Accos(2\pi fct+kpm(t)$$

This is the **equation of PM wave**.

If the modulating signal, m(t)=Amcos(2πfmt), then the equation of PM wave will be

$$s(t)=Accos(2\pi fct+\beta cos(2\pi fmt) \qquad (10)$$

Where,

- β = **modulation index** = Δϕ=kpAm
- Δϕ is phase deviation

Phase modulation is used in mobile communication systems, while frequency modulation is used mainly for FM broadcasting.

Relation between PM and FM

The change in phase, changes the frequency of the modulated wave. The frequency of the wave also changes the phase of the wave. Though they are related, their relationship is not linear. Phase modulation is an indirect method of producing FM. The amount of frequency shift, produced by a phase modulator increases with the modulating frequency. An audio equalizer is employed to compensate this.

The other type of modulation in continuous-wave modulation is Angle Modulation. Angle Modulation is the process in which the frequency or the phase of the carrier signal varies according to the message signal.

The standard equation of the angle modulated wave is

$$s(t)=Accos\ \theta i(t)$$

Where,

Ac is the amplitude of the modulated wave, which is the same as the amplitude of the carrier signal

$\theta i(t)\theta i(t)$ is the angle of the modulated wave

Angle modulation is further divided into frequency modulation and phase modulation.

- **Frequency Modulation** is the process of varying the frequency of the carrier signal linearly with the message signal.
- **Phase Modulation** is the process of varying the phase of the carrier signal linearly with the message signal.
- In amplitude modulation, the amplitude of the carrier signal varies. Whereas, in Frequency Modulation (FM), the frequency of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.
- Hence, in frequency modulation, the amplitude and the phase of the carrier signal remains constant. This can be better understood by observing the following figures.
- The frequency of the modulated wave increases, when the amplitude of the modulating or message signal increases. Similarly, the frequency of the modulated wave decreases, when the amplitude of the modulating signal decreases. Note that, the frequency of the modulated wave remains constant and it is equal to the frequency of the carrier signal, when the amplitude of the modulating signal is zero.

Mathematical Representation

The equation for instantaneous frequency fifi in FM modulation is

$fi=fc+kfm(t)$ Where, fc is the carrier frequency

kt is the frequency sensitivity

m(t) is the message signal

We know the relationship between angular frequency ωi and angle θi(t) as

$$\omega i = d\theta i(t)/dt$$

$$\Rightarrow 2\pi fi = d\theta i(t)/dt \qquad (11)$$

$$\Rightarrow \theta i(t) = 2\pi \int fidt \qquad (12)$$

Substitute, fi value in the above equation.

$$\theta i(t)=2\pi\int(fc+kfm(t))dt$$

$$\theta i(t)=2\pi fct+2\pi kf\int m(t)dt \tag{13}$$

Substitute, θi(t) value in the standard equation of angle modulated wave.

$$s(t)=Accos(2\pi fct+2\pi kf\int m(t)dt) \tag{14}$$

This is the **equation of FM wave**.

If the modulating signal is m(t)=Amcos(2πfmt), then the equation of FM wave will be

$$s(t)=Accos(2\pi fct+\beta sin(2\pi fmt)$$

Where,

β = **modulation index** =Δf/fm

$$=kfAm\ /\ fm \tag{15}$$

The difference between FM modulated frequency (instantaneous frequency) and normal carrier frequency is termed as Frequency Deviation. It is denoted by Δf, which is equal to the product of kf and Am. FM can be divided into Narrowband FM and Wideband FM based on the values of modulation index β.

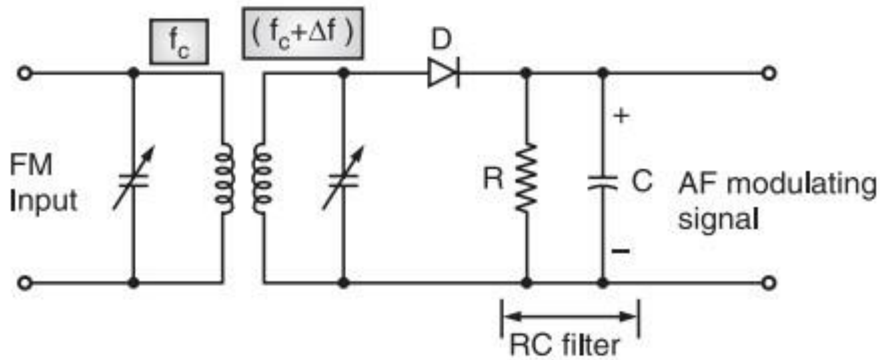
FM DEMODULATORS

The following two methods demodulate FM wave.

(1) Frequency discrimination method (Balanced FM Slope Detector (Balanced Frequency Discriminator)
(2) Phase discrimination method (Foster Seeley FM Demodulator)

FM Slope Detector / Simple FM Slope Detector (Frequency Discriminator)

The circuit diagram of a simple slope detector is as shown in figure 2.5.1,



Simple Slope Detector

Figure 2.5.1 Simple slope Detector

Diagram Source Electronics Post

The output voltage of the tank circuit is then applied to a simple diode detector of an RC load with proper time constant. This detector is identical to the AM diode detector. Even though the slope detector circuit is simple it has the following drawbacks.

- To overcome the drawbacks of the simple slope detector, a Balanced slope detector is used.

Balanced FM Slope Detector (Balanced Frequency Discriminator)

- The circuit diagram of the balanced slope detector is shown in Figure 2.5.2.

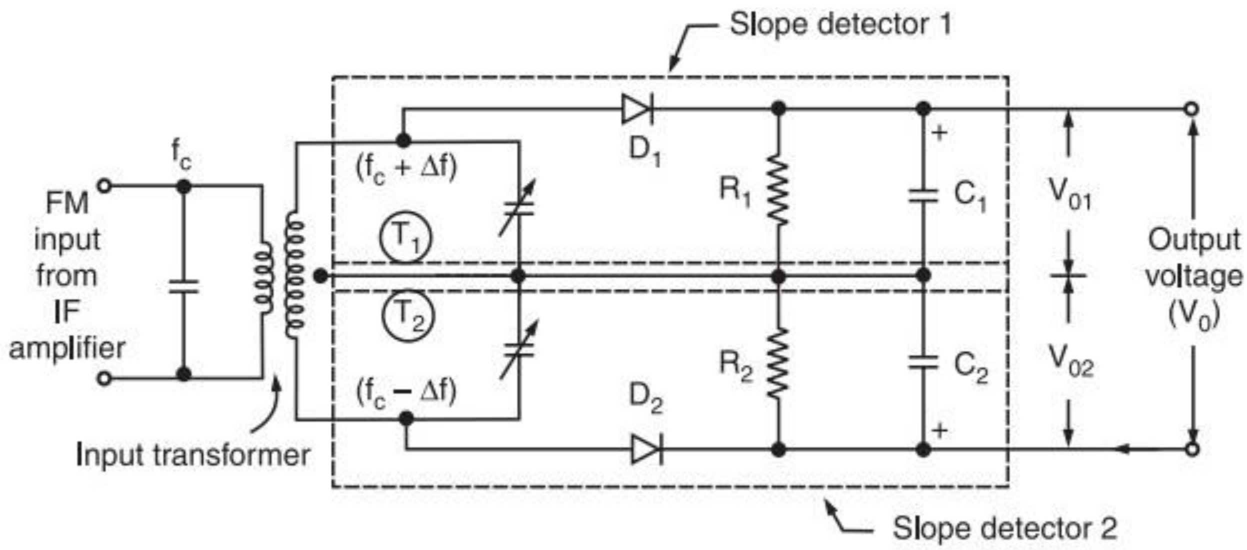


Figure 2.5.2 Balanced slope Detector

Diagram Source Electronics Post

As shown in the circuit diagram, the balanced slope detector consists of two slope detector circuits.

The input transformer has a center tapped secondary. Hence, the input voltages to the two slope detectors are 180° out of phase. There are three tuned circuits,Out of them, the primary is tuned to IF i.e., fc .

The upper tuned circuit of the secondary (T1) is tuned above fc by Δf i.e., its resonant frequency is (fc+ Δf). The lower tuned circuit of the secondary is tuned below fc by Δf i.e., at (fc – Δf). R1C1 and R2C2 are the filters used to bypass the RF ripple. Vo1 and Vo2 are the output voltages of the two slope detectors. The final output voltage Vo is obtained by taking the subtraction of the individual output voltages, Vo1 and Vo2, i.e.,

$$V_0 = V_{01} - V_{02} \quad (1)$$

We know that the equation of FM wave is

$$s(t)=A_c\cos(2\pi f_c t+2\pi k_f\int m(t)dt) \quad (2)$$

Differentiate the above equation with respect to 't'.

$$ds(t)/dt=-A_c(2\pi f_c+2\pi k_f m(t))\sin(2\pi f_c t+2\pi k_f\int m(t)dt) \quad (3)$$

We can write, $-\sin\theta$ as $\sin(\theta-180^0)$

$$ds(t)/dt=Ac(2\pi fc+2\pi kfm(t))\sin(2\pi fct+2\pi kf\int m(t)dt-180^0$$

- In the above equation, the amplitude term resembles the envelope of AM wave and the angle term resembles the angle of FM wave. Here, our requirement is the modulating signal m(t)m(t). Hence, we can recover it from the envelope of AM wave.
- The following figure 2.5.3 shows the block diagram of FM demodulator using frequency discrimination method.

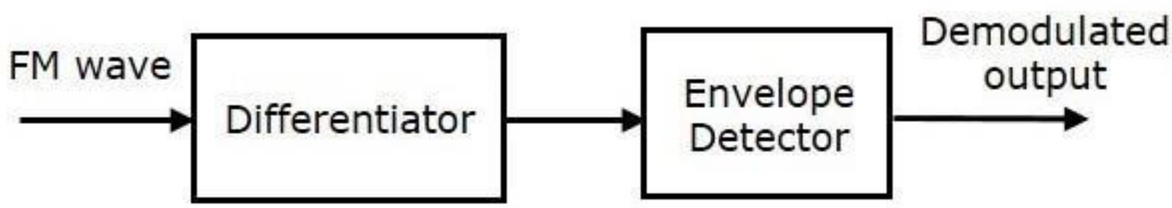


Figure 2.5.3 FM Demodulator using Frequency Discrimination method
Diagram Source Brain Kart

This block diagram consists of the differentiator and the envelope detector. Differentiator is used to convert the FM wave into a combination of AM wave and FM wave. This means, it converts the frequency variations of FM wave into the corresponding voltage (amplitude) variations of AM wave. We know the operation of the envelope detector. It produces the demodulated output of AM wave, which is nothing but the modulating signal.

Working Operation of the Circuit

The circuit operation can be explained by dividing the input frequency into three ranges as follows:

(i) **fin = fc:** When the input frequency is instantaneously equal to fc, the induced voltage in the T1 winding of secondary is exactly equal to that induced in the winding T2.

Thus, the input voltages to both the diodes D1 and D2 will be the same.

Therefore, their dc output voltages Vo1 and Vo2 will also be identical but they have opposite polarities. Hence, the net output voltage Vo = 0.

(ii) **fc < fin < (fc + Δf):** In this range of input frequency, the induced voltage in the winding T1 is higher than that induced in T2.

Therefore, the input to D1 is higher than D2. Hence, the positive output Vo1 of D1 is higher than the negative output Vo2 of D2. Therefore, the output voltage Vo is positive. As the input frequency increases towards (**fc + Δf**), the positive output voltage increases as shown below Figure. 2.5.4 .

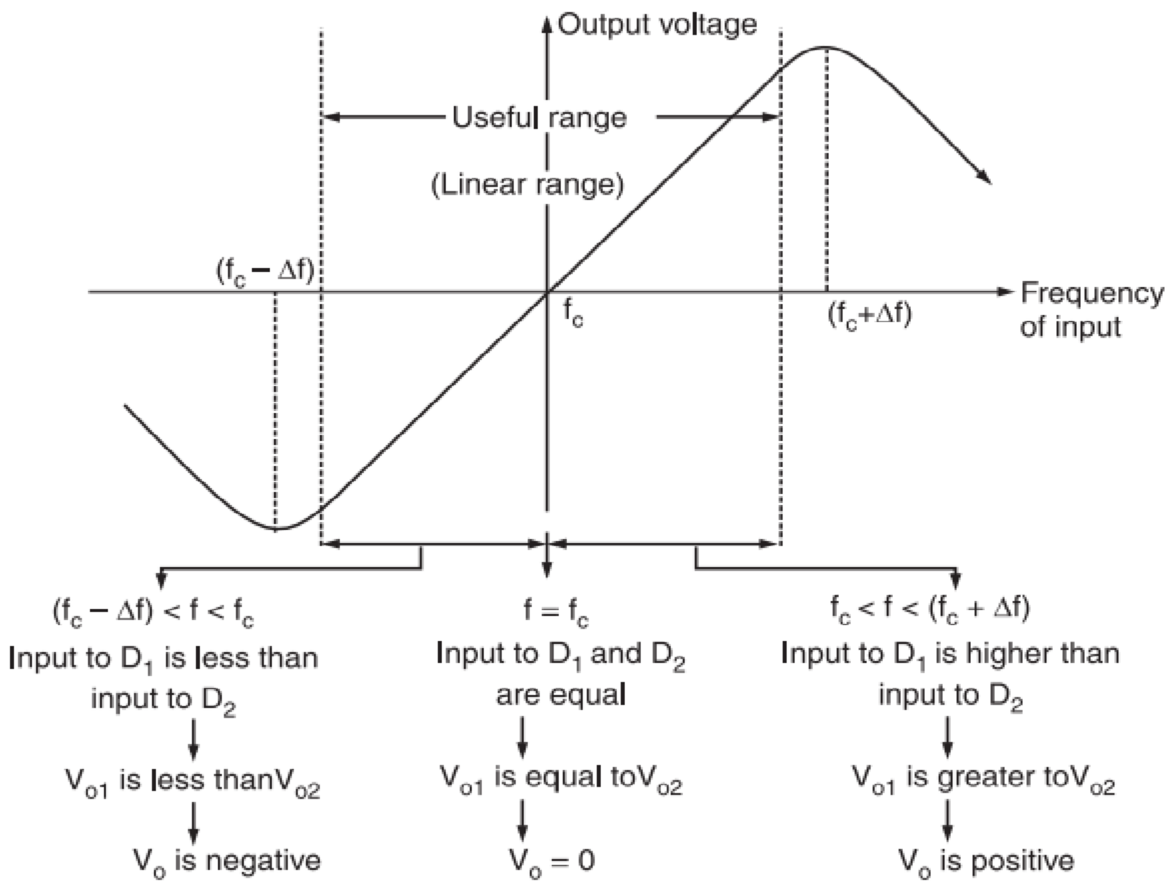


Fig. 2.5.4 Characteristics of the balanced slope detector

Diagram Source Electronics Post

If the output frequency goes outside the range of (fc – Δf) to (fc + Δf), the output voltage will fall due to the reduction in tuned circuit response.

Drawbacks of Slope Detector

- (i) It is inefficient.
- (ii) It is linear only over a limited frequency range.
- (iii) It is difficult to adjust as the primary and secondary winding of the transformer must be tuned to slightly different frequencies.

Advantages of Slope Detector

- The only advantages of the basic slope detector circuit is its simplicity.

Phase Discrimination Method (Foster Seeley FM Demodulator)

The following figure 2.5.5 shows the block diagram of FM demodulator using phase discrimination method.

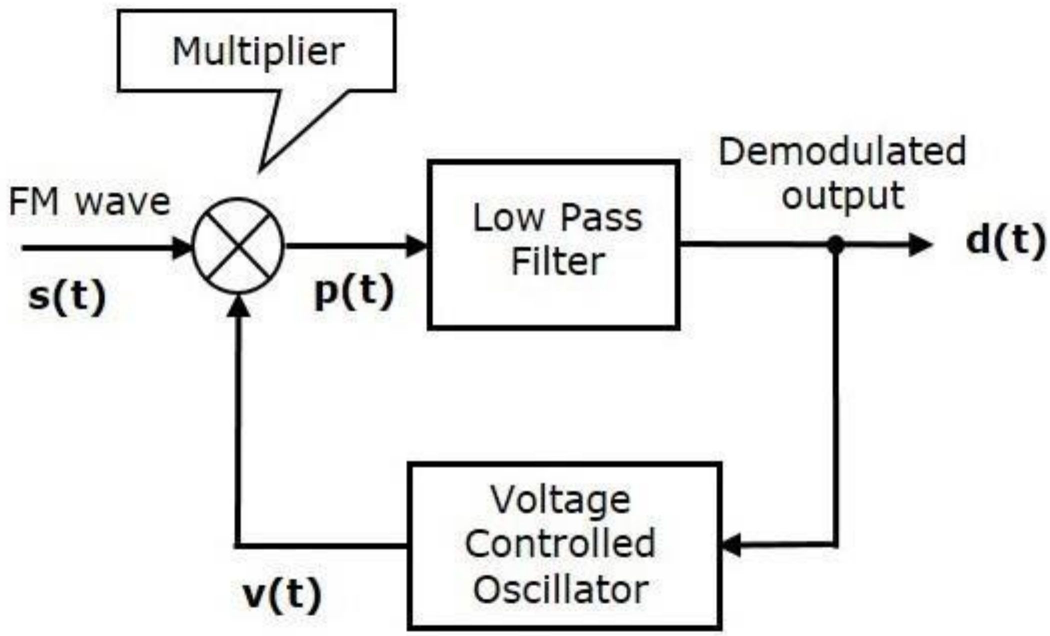


Figure 2.5.5 FM Demodulator using phase Discrimination method

Diagram Source Brain Kart

This block diagram consists of the multiplier, the low pass filter, and the Voltage Controlled Oscillator (VCO). VCO produces an output signal v(t)v(t), whose frequency is proportional to the input signal voltage d(t)d(t). Initially, when the signal d(t)d(t) is zero, adjust the VCO to produce an output signal v(t)v(t), having a carrier frequency and -90^0phase shift with respect to the carrier signal.

FM wave s(t)s(t) and the VCO output v(t)v(t) are applied as inputs of the multiplier. The multiplier produces an output, having a high frequency component and a low frequency component. Low pass filter eliminates the high frequency component and produces only the low frequency component as its output.

This low frequency component contains only the term-related phase difference. Hence, we get the modulating signal m(t) from this output of the low pass filter.

he Foster Seeley circuit is probably most commonly called the Foster Seeley discriminator. This is really a hang-over from early days of FM, and today the terms detector or probably better demodulator would probably be used.

The Foster Seeley discriminator circuit is characterised by the transformer, choke and diodes used within the circuit that forms the basis of its operation.

This FM demodulator circuit was invented by Dudley E. Foster and Stuart William Seeley in 1936. Although it was originally intended as a circuit to provide automatic frequency control, it was more widely used as an FM demodulator, whilst also being able to provide a voltage for automatic frequency control.

The Foster Seeley circuit was widely used until the 1970s when ICs using other techniques that were more easily integrated became widely available.

The circuit was widely used for all forms of radio communications applications from broadcasting to two way radio communications.

Foster-Seeley FM discriminator basics

The Foster Seeley detector or as it is sometimes described the Foster Seeley discriminator is quite similar to the ratio detector at a first look. It has an RF transformer and a pair of diodes, but there is no third winding - instead a choke is used.

FM Foster Seeley discriminator / detector circuit

In many respects the Foster Seeley FM demodulator resembles the circuit of a full wave bridge rectifier - the format that uses a centre tapped transformer, but additional components are added to give it a frequency sensitive aspect.

The basic operation of the circuit can be explained by looking at the instances when the instantaneous input equals the carrier frequency, the two halves of the tuned transformer circuit produce the same rectified voltage and the output is zero. If the frequency of the input changes, the balance between the two halves of the transformer secondary changes, and the result is a voltage proportional to the frequency deviation of the carrier.

Looking in more detail at the circuit, the Foster-Seeley circuit operates using a phase difference between signals. To obtain the different phased signals a connection is made to the primary side of the transformer using a capacitor, and this is taken to the centre tap of the transformer. This gives a signal that is 90° out of phase.

When an un-modulated carrier is applied at the centre frequency, both diodes conduct, to produce equal and opposite voltages across their respective load resistors. These voltages cancel each one another out at the output so that no voltage is present. As the carrier moves off to one side of the centre frequency the balance condition is destroyed, and one diode conducts more than the other. This results in the voltage across one of the resistors being larger than the other, and a resulting voltage at the output corresponding to the modulation on the incoming signal.

The choke is required in the circuit to ensure that no RF signals appear at the output. The capacitors C1 and C2 provide a similar filtering function.

Both the ratio detector and Foster-Seeley detectors are expensive to manufacture. Any wound components like the RF transformers are expensive to manufacture when compared with integrated circuits produced in vast numbers. As a result the Foster Seeley discriminator as well as the ratio detector circuits are rarely used in modern radio receivers as FM demodulators.

Foster Seeley circuit for frequency control

Prior to the introduction of very stable local oscillators within superhet radios - the universal format for radios receiving FM, local oscillators had a tendency to drift. Drift was a major factor in domestic radio receivers, although it was present in all radios.

When receiving FM signals the drift meant that the incoming FM signal might drift away from being at the centre of the FM detector slope onto the non-linear portions. This meant that the signal would become distorted.

To overcome this, radio receivers would incorporate a facility known as automatic frequency control was implemented. Using this, the DC offset from the FM demodulator is used to tune the receiver local oscillator to bring it back on frequency.

FM demodulator curve produces

A DC offset is produced when the centre frequency of the carrier is not on the centre of the demodulator curve. By filtering off the audio, only a DC component remains. Typically a long time constant RC combination is used to achieve this was shown in the figure 2.5.6. The time constant of this RC network can be quite long as the drift of the oscillator occurs gradually over a period of seconds, and it must also be longer than that of the lowest frequency of the audio.

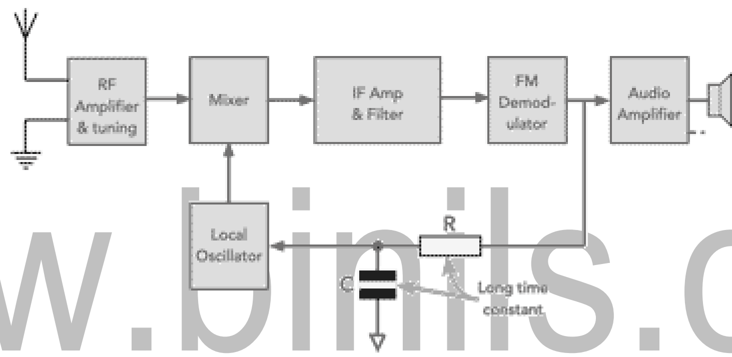


Fig 2.5.6 AFC circuitry for a super heterodyne radio receiver

Diagram Source Electronics Post

The filtered voltage is applied to a varactor diode within the local oscillator such that it causes the local oscillator to remain on tune for the FM signal being received. In this way the receiver can operate so that the signal being received is demodulated within the linear region of the FM demodulator.

Essentially the effect of the AFC circuitry is to create a form of negative feedback loop that seeks to keep the centre of the FM signal at the centre of the FM demodulation S curve. It is essentially a frequency locked loop.

Most radios used for FM reception that have free running local oscillators incorporate an automatic frequency control, AFC circuit. It uses only a few components and it provides for a significant improvement in the performance of the receiver, enabling the FM signal to be demodulated with minimum distortion despite the drift of the local oscillator signal.

Prior to the widespread introduction of frequency synthesizers, AFC was not always used in radios such as walkie talkies and handhelds radios aimed at for two way radio communications applications as they tended to use crystal controlled oscillators and these did not drift to any major degree. Hence there was less requirement for an AFC. It uses a double-tuned RF transformer to convert frequency variations in the received fm signal to amplitude variations. These amplitude variations are then rectified and filtered to provide a dc output voltage. This voltage varies in both amplitude and polarity as the input signal varies in frequency.

The output voltage is 0 when the input frequency is equal to the carrier frequency (FR). When the input frequency rises above the center frequency, the output increases in the positive direction. When the input frequency drops below the center frequency, the output increases in the negative direction. The output of the Foster-Seeley discriminator is affected not only by the input frequency, but also to a certain extent by the input amplitude. Therefore, using limiter stages before the detector is necessary.

Circuit Operation of a Foster-Seeley Discriminator

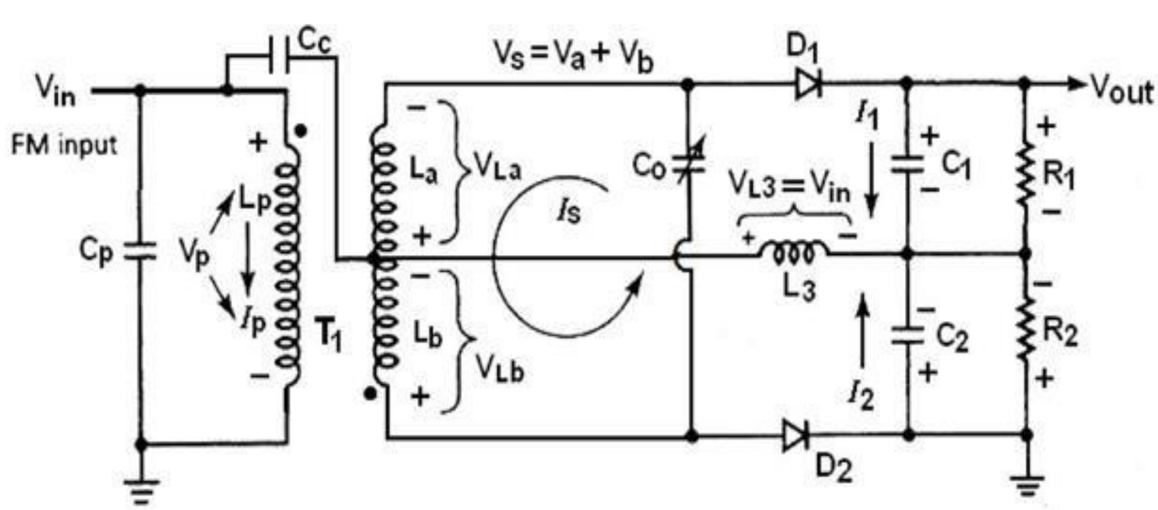


Fig 2.5.7 Circuit Operation of a Foster-Seeley Discriminator

Diagram Source Electronics Post

- Fig. shows a typical Foster-Seeley discriminator. The primary tank circuit consists of C1 and L1. C2 and L2 form the secondary tank circuit. Both tank circuits are tuned to the center frequency of the incoming fm signal.
- Choke L3 is the dc return path for diode rectifiers D1 and D2. Resistors R3 and R4 are the load resistors and are bypassed by C3 and C4 to remove rf.
- To obtain the different phased signals a connection is made to the primary side of the transformer using a capacitor, and this is taken to the centre tap of the transformer. This gives a signal that is 90° out of phase.
- When an un-modulated carrier is applied at the centre frequency, both diodes conduct, to produce equal and opposite voltages across their respective load resistors. These voltages cancel each one another out at the output so that no voltage is present.
- As the carrier moves off to one side of the centre frequency the balance condition is destroyed, and one diode conducts more than the other. This results in the voltage across one of the resistors being larger than the other, and a resulting voltage at the output corresponding to the modulation on the incoming signal.
- The choke is required in the circuit to ensure that no RF signals appear at the output. The capacitors C1 and C2 provide a similar filtering function.
- The operation of the Foster-Seeley discriminator can best be explained using vector diagrams fig. 3 that show phase relationships between the voltages and currents in the circuit. Let's look at the phase relationships when the input frequency is equal to the center frequency of the resonant tank circuit.

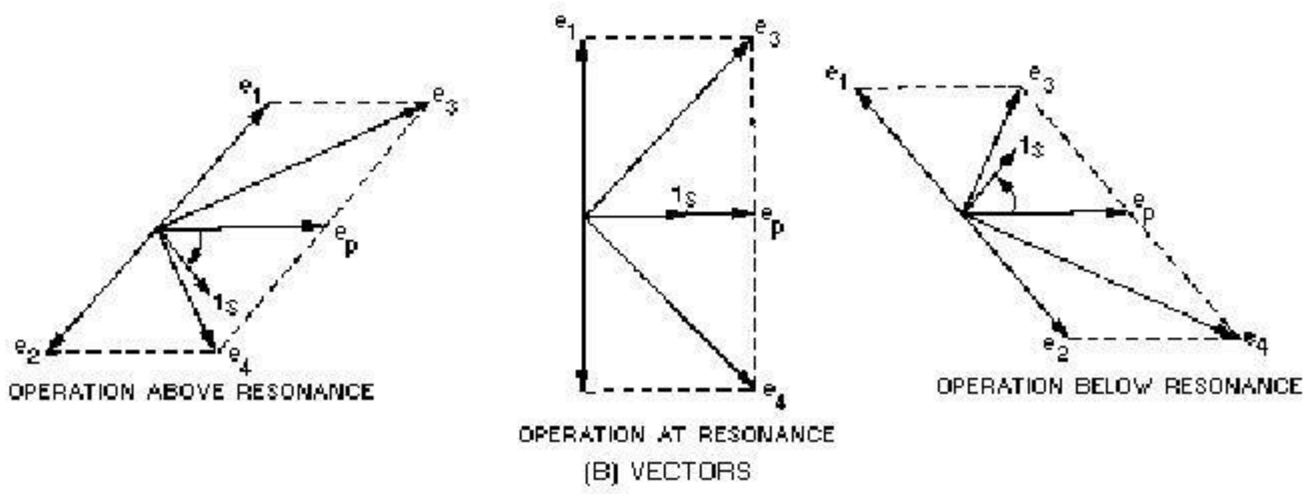


Fig.2.5.8 Phasor diagram of foaster seeley discriminator - Typical discriminator response
Diagram Source Electronics Post

Advantages of Foster-Seeley FM discriminator:

- Offers good level of performance and reasonable linearity.
- Simple to construct using discrete components.
- Provides higher output than the ratio detector
- Provides a more linear output, i.e. lower distortion than the ratio detector
- Offers good level of performance and reasonable linearity.
- Simple to construct using discrete components.
- Provides higher output than the ratio detector
- Provides a more linear output, i.e. lower distortion than the ratio detector
- This circuit is more efficient than simple slope detector.
- It has better linearity than the simple slope detector.

Disadvantages of Foster-Seeley FM discriminator:

- Does not easily lend itself to being incorporated within an integrated circuit.
- High cost of transformer.
- Narrower bandwidth than the ratio detector
- Does not easily lend itself to being incorporated within an integrated circuit.
- High cost of transformer.

- Narrower bandwidth than the ratio detector
- The circuit is sensitive to both frequency and amplitude and therefore needs a limiter before it to remove amplitude variations and hence amplitude noise.
- (i) Even though linearity is good, it is not good enough.
- (ii) This circuit is difficult to tune since the three tuned circuits are to be tuned at different frequencies i.e., fc, (fc+Δf) and (fc – Δf).
- (iii) Amplitude limiting is not provided.

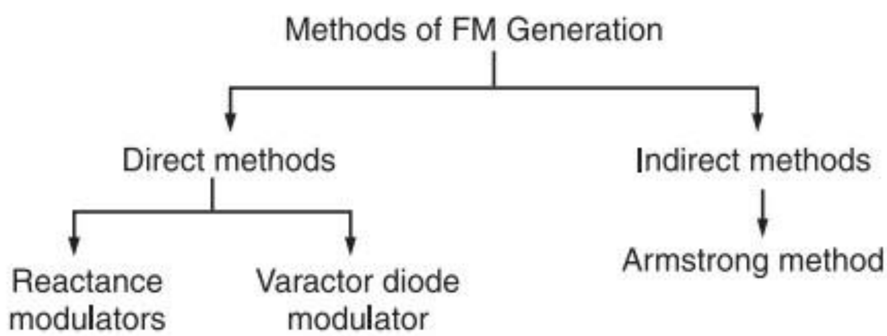
Generation of FM Wave

The FM modulator circuits used for generating FM signals can be divided into two categories such as:

(i) The direct method or parameter variation method

(ii) The Indirect method or the Armstrong method

The classification of FM generation methods is shown below :



The Direct Method or Parameter Variation Method

In direct method or parameter variation method, the baseband or modulating signal directly modulates the carrier. The carrier signal is generated with the help of an oscillator circuit. This oscillator circuit uses a parallel tuned L-C circuit. Thus the frequency of oscillation of the carrier generation is governed by the expression:

$$\omega_c = \frac{1}{\sqrt{LC}}$$

Now, we can make the carrier frequency ωc to vary in accordance with the baseband or modulating signal x(t) if L or C is varied according to x(t). An oscillator circuit whose frequency is controlled by a modulating voltage is called voltage controlled oscillator (VCO). The frequency of VCO is varied according to the modulating signal simply by putting a shunt voltage variable capacitor with its tuned circuit.

This voltage variable capacitor is called varactor or varicap. This type of property is exhibited by reverse biased semiconductor diodes. Also the capacitance of bipolar junction transistors (BJT) and field-effect transistors (FET) is varied by the Miller-effect. This miller capacitance may be utilized for frequency modulation. In addition to this, the electron tubes may also provide variable reactance (either it is inductive or capacitive) which is proportional to modulating or baseband signal. This type of tubes are called reactance tubes and may be used for FM generation. The inductance L of the tuned circuit may also be varied in accordance with the baseband or modulating signal x(t). The FM circuit using such inductors is called saturable reactor modulator.Frequency modulation can also be achieved from voltage controlled devices such as PIN diode, Klystron oscillators and multivibrators.

Reactance Modulator

In direct FM generation shown in figure 2.4.1, the instantaneous frequency of the carrier is changed directly in proportion with the message signal. For this, a device called voltage controlled oscillator (VCO) is used. A VCO can be implemented by using a sinusoidal oscillator with a tuned

circuit having a high value of Q.

The frequency of this oscillator is changed by changing the reactive components involved in the tuned circuit. If L or C of a tuned circuit of an oscillator is changed in accordance with the amplitude of modulating signal then FM can be obtained across the tuned circuit as shown in figure 2.4.1 below.

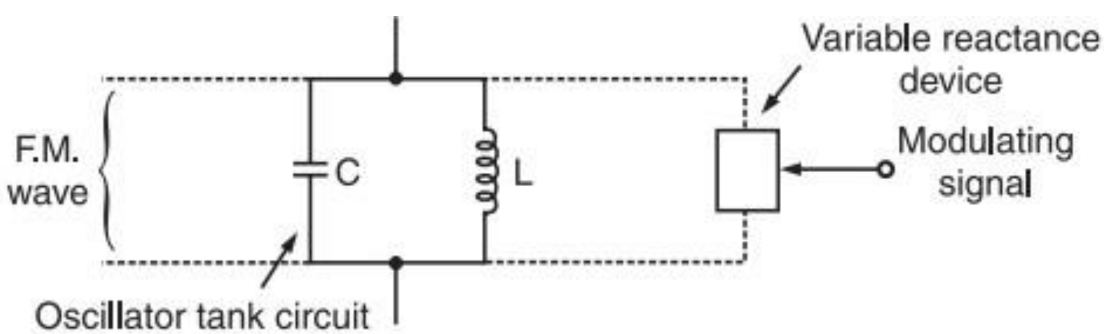


Figure.2.4.1 Principle of Reactance Modulator

Diagram Source Electronics Tutorials

A two or three terminal device is placed across the tuned circuit. The reactance of the device is varied proportional to modulating signal voltage. This will vary the frequency of the oscillator to produce FM. The devices used are FET, transistor or varactor diode. An example of direct FM is shown in figure 1 which uses a Hartley oscillator along with a varactor diode.

The varactor diode is reverse biased. Its capacitance is dependent on the reverse voltage applied across it. This capacitance is shown by the capacitor C(t) in figure 2.4.2

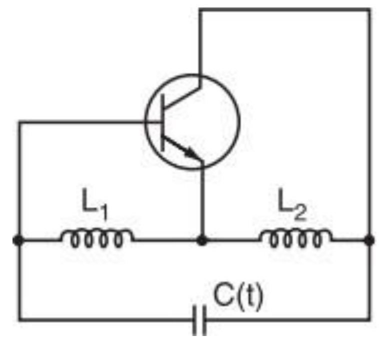


Figure.2.4.2 Hartley Oscillator

Diagram Source Electronics Post

The FM transmitter has three basic sections.

1. The exciter section contains the carrier oscillator, reactance modulator and the buffer amplifier.
2. The frequency multiplier section, which features several frequency multipliers.
3. The power output section, which includes a low- level power amplifier, the final power amplifier, and the impedance matching network to properly load the power section with the antenna impedance.

The essential function of each circuit in the FM transmitter may be described as follows.

The Exciter

1. The function of the carrier oscillator is to generate a stable sine wave signal at the rest frequency, when no modulation is applied. It must be able to linearly change frequency when fully modulated, with no measurable change in amplitude.

2. The buffer amplifier acts as a constant high-impedance load on the oscillator to help stabilize the oscillator frequency. The buffer amplifier may have a small gain.

3. The modulator acts to change the carrier oscillator frequency by application of the message signal. The positive peak of the message signal generally lowers the oscillator's frequency to a point below the rest frequency, and the negative message peak raises the oscillator frequency

to a value above the rest frequency. The greater the peak-to-peak message signal, the larger the oscillator deviation.

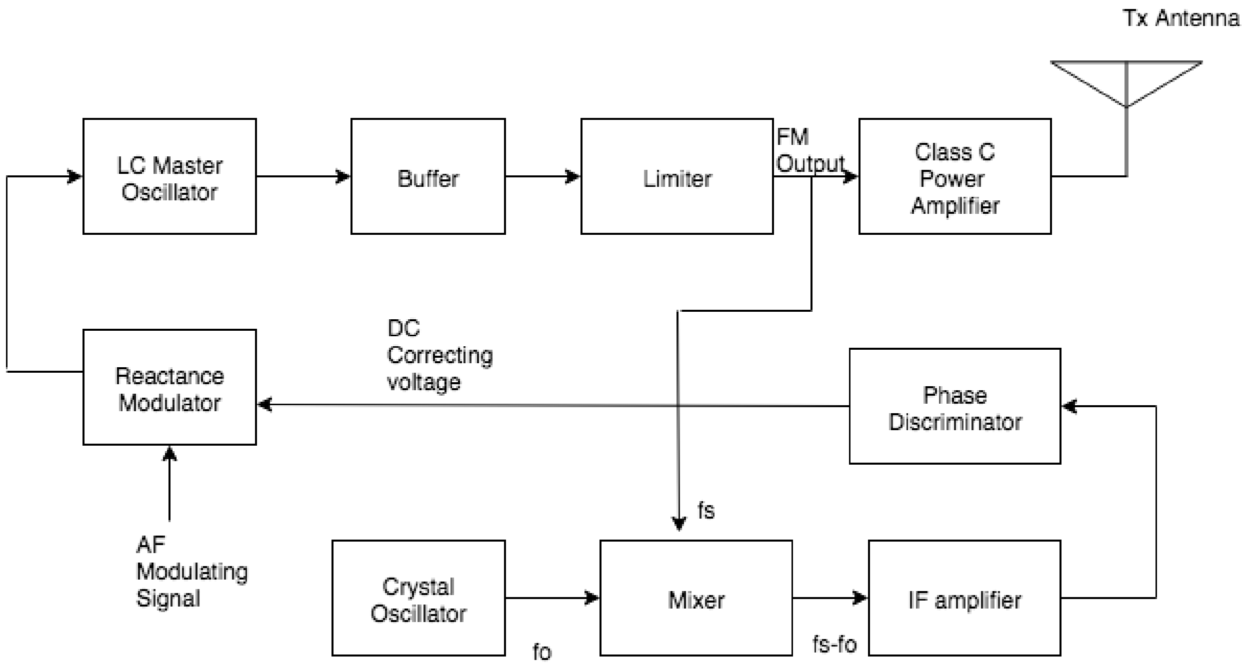


Figure.2.4.3 Reactance Modulator

Diagram Source Electronics Post

Frequency multipliers are tuned-input, tuned-output RF amplifiers in which the output resonant circuit is tuned to a multiple of the input frequency in the above figure 2.4.3. Common frequency multipliers are 2x, 3x and 4x multiplication. A 5x Frequency multiplier is sometimes seen, but its extreme low efficiency forbids widespread usage. Note that multiplication is by whole numbers only. There can not a 1.5x multiplier, for instance. The final power section develops the carrier power, to be transmitted and often has a low-power amplifier driven the final power amplifier. The impedance matching network is the same as for the AM transmitter and matches the antenna impedance to the correct load on the final over amplifier.

Frequency Multiplier

A special form of class C amplifier is the frequency. multiplier. Any class C amplifier is capable of performing frequency multiplication if the tuned circuit in the collector resonates at some integer multiple of the input frequency.

For example a frequency doubler can be constructed by simply connecting a parallel tuned circuit in the collector of a class C amplifier that resonates at twice the input frequency. When the collector current pulse occurs, it excites or rings the tuned circuit at twice the input frequency. A current pulse flows for every other cycle of the input.

A Tripler circuit is constructed in the same way except that the tuned circuit resonates at 3 times the input - frequency. In this way, the tuned circuit receives one input pulse for every three cycles of oscillation it produces Multipliers can be constructed to increase the input frequency by any integer factor up to approximately 10. As' the multiplication factor gets higher, the power output of the multiplier decreases. For most practical applications, the best result is obtained with multipliers of 2 and 3.

Another way to look the operation of class C multipliers is .to .remember that the non-sinusoidal current pulse is rich in harmonics. Each time the pulse occurs, the second, third, fourth, fifth, and higher harmonics are generated. The purpose of the tuned circuit in the collector is to act as a filter to select the desired harmonics.

In many applications a multiplication factor greater than that achievable with a single multiplier stage is required. In such cases two or more multipliers are cascaded to produce an overall multiplication of 6. In the second example, three multipliers provide an overall multiplication of 30. The total multiplication factor is simply the product of individual stage multiplication factors.

Reactance Modulator

The reactance modulator takes its name from the fact that the impedance of the circuit acts as a reactance (capacitive or inductive) that is connected in parallel with the resonant circuit of the Oscillator. The varicap can only appear as a capacitance that becomes part of the frequency determining branch of the oscillator circuit. However, other discrete devices can appear as a capacitor or as an inductor to the oscillator, depending on how the circuit is arranged. A colpitts oscillator uses a capacitive voltage divider as the phase-reversing feedback path and would most likely tapped coil as the phase-reversing element in the feedback loop and most commonly uses a modulator that appears inductive.

Frequency of oscillations of the Hartley oscillator is given by :

$$f_i(t) = \frac{1}{2\pi\sqrt{(L_1 + L_2)C(t)}}$$

where C(t) = C + Cvarector

This means that C(t) is the effective capacitance of the fixed tuned circuit capacitance C and the varactor diode capacitance C varector.

Let the relation between the modulating voltage x(t) = 0 and the capacitance C(t) be represented as under:

$$C(t) = C - k_c\, x(t)$$

where C = total capacitance when x(t)

kc is the sensitivity of the varactor capacitance to change in voltage

Substituting expression for C(t) in equation(1) , we get

$$f_i(t) = \frac{1}{2\pi\sqrt{(L_1 + L_2)(C - k_c\, x(t))}} = \frac{1}{2\pi\left[\sqrt{(L_1 + L_2)C - (L_1 + L_2)k_c\, x(t)}\right]}$$

or $$f_i(t) = \frac{1}{2\pi\sqrt{(L_1 + L_2)C}\left[1 - \frac{k_c\, x(t)}{C}\right]^{1/2}}$$

But, let

$$\frac{1}{2\pi\sqrt{(L_1 + L_2)C}} = f_0$$

which is the oscillator frequency in absence of the modulating signal [x(t) = 0]. Therefore, we have,

$$f_i(t) = f_0 \left[1 - \frac{k_c}{C} x(t)\right]^{-1/2}$$

If the maximum change in the capacitance corresponding to the modulating wave is assumed to be small as compared to the unmodulated capacitance C then equation (2) for fi (t) can be approximated as under:

$$f_i(t) = f_0 \left[1 + \frac{k_c}{2C} x(t)\right]$$

or

$$f_i(t) = f_0 + \frac{f_0 k_c}{2C} . x(t)$$

Now, let us define

$$\frac{f_0 k_c}{2C} = k_f$$

Therefore, we have

$$f_i(t) = f_0 + k_f x(t)$$

where kf is called as the frequency sensitivity of the modulator.

Varactor Diode Modulator

- Varactor diode modulator is the direct method of FM generation wherein the carrier frequency is directly varied by the modulating signal.
- A varactor diode is a semiconductor diode whose junction capacitance varies linearly with applied voltage when the diode is reverse biased.

- Varactor diodes are used along with reactance modulator to provide automatic frequency correction for an FM transmitter. The varactor diode modulator circuit is shown in Figure5. for generation of FM wave.

The varactor diode FM modulator has been shown below in figure 2.4.4.

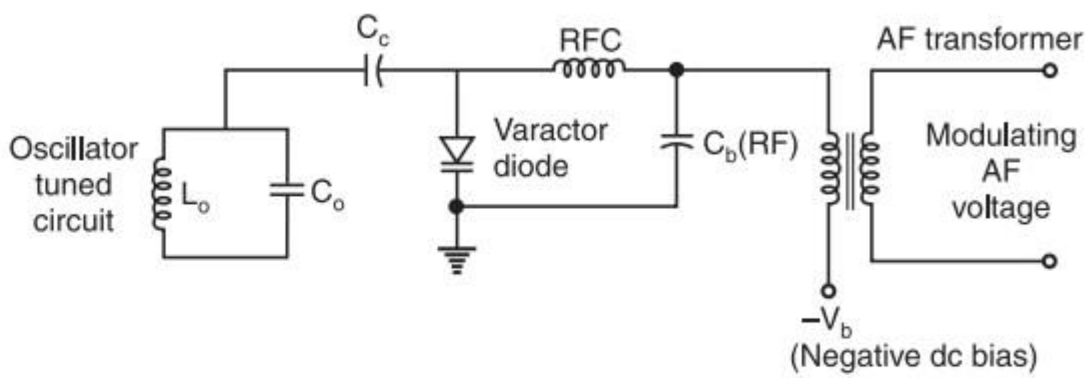


Figure.2.4.4 Varactor Diode Modulator

Diagram Source Electronics Post

A varactor diode is a semiconductor diode whose junction capacitance varies linearly with the applied bias and the varactor diode must be reverse biased.

Working Operation

Varactor diode is arranged in reverse bias to offer junction capacitance effect. The modulating voltage which is in series with the varactor diode will vary the bias and hence the junction capacitance, resulting the oscillator frequency to change accordingly.The external modulating AF voltage adds to and subtracts from the dc bias, which changes the capacitance of the diode and thus the frequency of oscillation. Positive alternations of the modulating signal increase the reverse bias on the varactor diode, which decreases its capacitance and increases the frequency of oscillation.

Conversely, negative alternations of the modulating signal decrease the frequency of oscillation. The RFC and capacitor C b act as a filter which transmits only the AF variations to the varactor diode and blocks high frequency RF voltage from reaching the AF stage. The varactor diode FM modulators are widely accepted because they are simple to use, reliable and have the stability of a crystal oscillator. This method of FM generation is direct because the oscillator frequency is varied directly by the modulating signal, and the magnitude of frequency change is proportional to the amplitude of the modulating signal voltage.

Varactor diode modulator is used for automatic frequency control and remote tuning. The drawback of varactor diode modulator is that since it uses a crystal, the peak frequency deviation is limited to relatively small values. Thus they are used mostly for low index applications such as two way mobile radio. Also since they are a two terminal device, the applications are quite limited. The varactor diode is reverse biased by the negative dc source –Vb.

The modulating AF voltage appears in series with the negative supply voltage. Hence, the voltage applied across the varactor diode varies in proportion with the modulating voltage .This will vary the junction capacitance of the varactor diode. The varactor diode appears in parallel with the oscillator tuned circuit. Hence the oscillator frequency will change with change in varactor diode capacitance and FM wave is produced. The RFC will connect the dc and modulating signal to the varactor diode but it offers a very high impedance at high oscillator frequency. Therefore, the oscillator circuit is isolated from the dc bias and modulating signal.

Indirect Method of WBFM Generation

This method is called as Indirect Method because we are generating a wide band FM wave indirectly. This means, first we will generate NBFM wave and then with the help of frequency multipliers we will get WBFM wave. The block diagram of generation of WBFM wave is shown in the following figure.

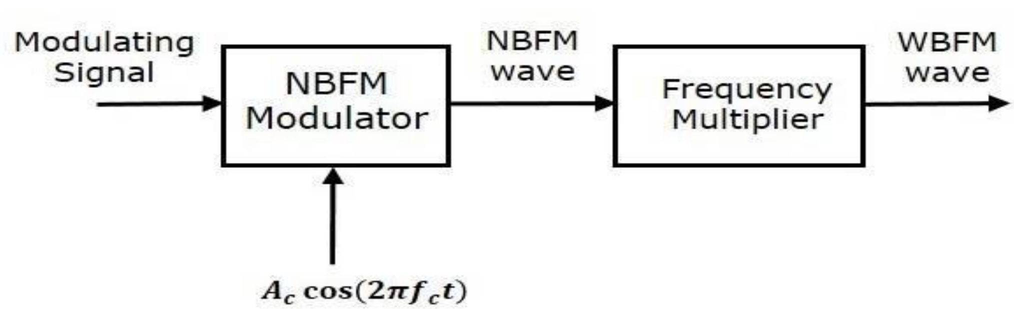


Figure.2.4.5 Varactor Diode Modulator

Diagram Source Electronics Post

This block diagram shown in figure 2.4.5 contains mainly two stages. In the first stage, the NBFM wave will be generated using NBFM modulator. We have seen the block diagram of NBFM modulator at the beginning of this chapter. We know that the modulation index of NBFM wave is less than one. Hence, in order to get the required modulation index (greater than one) of FM wave, choose the frequency multiplier value properly. Frequency multiplier is a non-linear device, which produces an output signal whose frequency is 'n' times the input signal frequency. Where, 'n' is the multiplication factor. If NBFM wave whose modulation index ββ is less than 1 is applied as the input of frequency multiplier, then the frequency multiplier produces an output signal, whose modulation index is 'n' times ββ and the frequency also 'n' times the frequency of WBFM wave.Sometimes, we may require multiple stages of frequency multiplier and mixers in order to increase the frequency deviation and modulation index of FM wave.

The part of the Armstrong FM transmitter (Armstrong phase modulator) which is expressed in dotted lines describes the principle of operation of an Armstrong phase modulator. It should be noted, first that the output signal from the carrier oscillator is supplied to circuits that perform the task of modulating the carrier signal. The oscillator does not change frequency, as is the case of direct FM. These points out the major advantage of phase modulation (PM), or indirect FM, over direct FM. That is the phase modulator is crystal controlled for frequency.

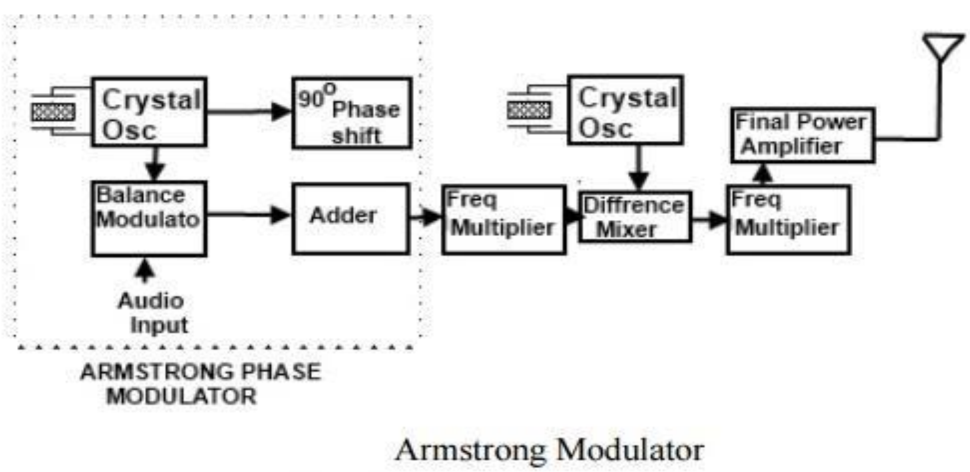


Figure.2.4.6 Armstrong Modulator

Diagram Source Electronics Post

The crystal-controlled carrier oscillator signal is directed to two circuits in parallel. This signal (usually a sine wave) is established as the reference past carrier signal and is assigned a value 0°.The balanced modulator is an amplitude modulator used to form an envelope of double side-bands and to suppress the carrier signal (DSSC). This requires two input signals, the carrier signal and the modulating message signal shown in te figure 2.4.6. The output of the modulator is connected to the adder circuit; here the 90° phase-delayed carriers signal will be added back to replace the suppressed carrier. The act of delaying the carrier phase by 90° does not change the carrier frequency or its wave-shape. This signal identified as the 90° carrier signal.

The carrier frequency change at the adder output is a function of the output phase shift and is found by. $f_c = \Delta\theta f_s$ (in hertz) .When θ is the phase change in radians and fs is the lowest audio modulating frequency. In most FM radio bands, the lowest audio frequency is 50Hz. Therefore, the carrier frequency change at the adder output is 0.6125 x 50Hz = ± 30Hz since 10% AM represents the upper limit of carrier voltage change, then ± 30Hz is the maximum deviation from the modulator for PM. The 90° phase shift network does not change the signal frequency because the components and resulting phase change are constant with time. However, the phase of the adder output voltage is in a continual state of change brought about by the cyclical variations of the message signal, and during the time of a phase change, there will also be a frequency change.

NARROW BAND FM MODULATION:

Types of FM

The FM systems are basically classified into following two types :

Narrow band FM

Wide band FM / Broadband FM

Narrow Band FM

A narrow band FM is the FM wave with a small bandwidth , The modulation index mf of narrow band FM is small as compared to one radian . Hence, the spectrum of narrow band FM consists of the carrier and upper sideband and a lower sideband .

For small values of mf , the values of the j coefficients are as under :

$J0(mf) = 1$,

$J1(mf) = mf/2$

$Jn(mf) = 0$ for $n > 1$

Practically, the narrow band FM systems have mf less than 1 . The maximum permissible frequency deviation is restricted to about 5 kHz . This system is used in FM mobile communications such as police wireless, ambulances, taxicabs etc . This frequency modulation has a small bandwidth when compared to wideband FM. The modulation index β is small, i.e., less than 1.Its spectrum consists of the carrier, the upper sideband and the lower sideband. This is used in mobile communications such as police wireless, ambulances, taxicabs, etc.

Generation of Narrow band Frequency Modulated Wave (NBFM)

We know that the standard equation of FM wave is

$$s(t)=Accos(2\pi fct+2\pi kf\int m(t)dt) \qquad (1)$$

$$s(t)=Accos(2\pi fct)cos(2\pi kf\int m(t)dt) - Acsin(2\pi fct)sin(2\pi kf\int m(t)dt) \qquad (2)$$

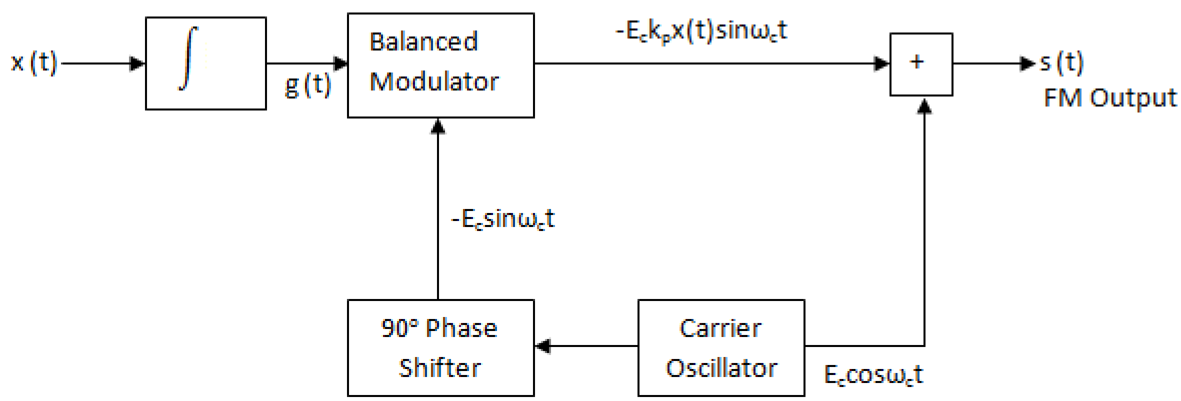


Figure 2.2.1 Generation of Frequency

Modulated Signal, Diagram Source Brain Kart

For NBFM,

$$|2\pi k_f \int m(t)dt| << 1 \tag{3}$$

We know that cosθ≈1 and sinθ≈1 when θ is very small.

By using the above relations, we will get the NBFM equation as

$$s(t)=A_c\cos(2\pi f_c t)-A_c\sin(2\pi f_c t)2\pi k_f\int m(t)dt \tag{4}$$

The block diagram of NBFM modulator is shown in the following figure 2.2.1. Here, the integrator is used to integrate the modulating signal m(t)m(t). The carrier signal Accos(2πfct) is the phase shifted by -90^0 to get Acsin(2πfct) with the help of -90^0 phase shifter. The product modulator has two inputs ∫m(t)dt∫m(t)dt and Acsin(2πfct). It produces an output, which is the product of these two inputs.

This is further multiplied with 2πkf by placing a block 2πkf in the forward path. The summer block has two inputs, which are nothing but the two terms of NBFM equation. Positive and negative signs are assigned for the carrier signal and the other term at the input of the summer block. Finally, the summer block produces NBFM wave as shown in the figure 2.2.2.

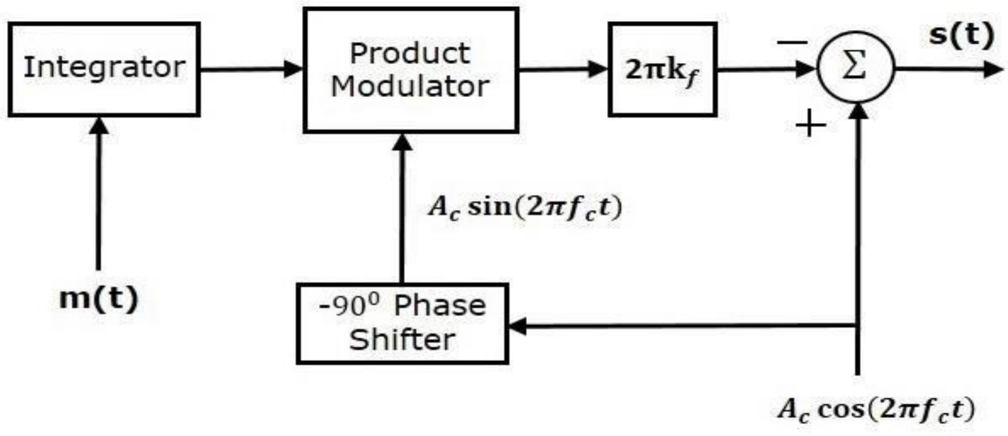


Figure 2.2.2 Generation of Frequency Modulated Signal

Diagram Source Brain Kart

Analysis of Narrow band FM

As we know, the expression for instantaneous frequency of FM wave is given as :

$$f_i = f_c + k_f x(t) \tag{5}$$

Where, x(t) is the modulating signal .

The term kf x(t) represents the frequency deviation . The constant kf will control the deviation . For small values of kf , the frequency deviation is small and the spectrum of FM signal has a narrow band . Hence, it is called as the narrow band FM .

Let us consider the expression for FM wave as under :

$$s(t) = E_c \cos[2\pi f_c t + 2\pi k_f \int x(t)dt]$$

Expressing it in terms of ω, we have

$$s(t) = E_c \cos[\omega_c t + 2\pi k_f \int x(t)dt]$$

We can represent this in the exponential manner as under:

$$s(t) = E_c \cos\theta(t) = E_c e^{j\theta(t)}$$

This has been written by considering only the real part of $E_c e^{j\theta(t)}$

Therefore,

$$s(t) = E_c e^{j\theta(t)} = E_c e^{j[cos\omega_c t + k_f \int x(t)dt]}$$

Let $\int x(t)dt = g(t)$

Thus,

$$s(t) = E_c e^{j[cos\omega_c t + k_f g(t)]}$$

If $k_f g(t) << 1$ for all values (which is the case for narrow band FM), then, the expression for FM will be

$$\hat{s}(t) = E_c[1 + jk_f g(t)]e^{j\omega_c t}$$

Also,

$$s(t) = R_c[\hat{s}(t)] = \underbrace{E_c \cos\omega_c t}_{\text{Carrier}} - \underbrace{E_c k_f g(t)\sin\omega_c t}_{\text{Side band}} \qquad (6)$$

This is the expression for narrow band FM

Mathematical Expression for Single-tone Narrow Band FM

As we know the expression for instantaneous frequency of FM wave is given by :

$$f_i = f_c + k_f x(t) \qquad (7)$$

where, x (t) is the modulating signal and the term kf x (t) represents the frequency deviation.The constant kf will control the deviation. For small values of kf, the frequency deviation is small and the spectrum of FM signal has a narrow band. Hence, it is called as the narrow band FM.

Let us consider the expression for FM wave as under:

$$s(t) = E_c \cos\left[2\pi f_c\, t + 2\pi k_f \int x\,(t)\, dt\right] \tag{8}$$

Expressing it in terms of ω, we have:

$$s(t) = E_c \cos\left[\omega_c\, t + 2\pi k_f \int x\,(t)\, dt\right] \tag{9}$$

We can represent this in the exponential manner as under:

$$s(t) = E_c \cos\theta(t) = E_c\, e^{j\theta(t)}$$

This has been written by considering only the real part of Ec ejθ(t)

Therefore,

$$s(t) = E_c\, e^{j\theta(t)}. = E_c\, e^{j[\cos\omega_c t + k_f\, x\,(t)\, dt]} \tag{10}$$

Let

$$\int x\,(t)\, dt = g(t)$$

Thus,

$$s(t) = E_c e^{j[\cos\omega_c t + k_f\, g\,(t)]} \tag{11}$$

If kf g (t) < < 1 for all values (which is the case for narrow band FM), then, the expression for FM will be

$$\hat{s}(t) = E_c\left[1 + jk_f\, g(t)\right]e^{j\omega_c t}$$

Also,

$$s(t) = R_c\left[\hat{s}(t)\right] = \underbrace{E_c \cos\omega_c t}_{\text{carrier}} - \underbrace{E_c k_f\, g(t) \sin\omega_c t}_{\text{side band}}$$

This is the expression for narrowband FM. Hence, a narrow band FM wave can be expressed mathematically as under, The (-) sign associated with the LSB represents a phase shift of 180o.

$$e_{FM}(t) = s(t) = \underbrace{E_c sin\omega_c t}_{\text{Carrier}} + \frac{m_f E_c}{2}\sin\underbrace{(\omega_c + \omega_m)}_{\text{USB}}t - \frac{m_f E_c}{2}\sin\underbrace{(\omega_c - \omega_m)}_{\text{LSB}}t \qquad (12)$$

Narrowband FM Applications:

This frequency modulation has a small bandwidth when compared to wideband FM. The modulation index β is small, i.e., less than 1. Its spectrum consists of the carrier, the upper sideband and the lower sideband. This is used in mobile communications such as police wireless, ambulances, taxicabs, etc.

Phase locked loop FM detector (PLL FM demodulator)

Phase locked loop, PLL FM detectors can easily be made from the variety of phase locked loop integrated circuits that are available, and as a result, PLL FM demodulators are found in many types of radio equipment ranging from broadcast receivers to high performance communications equipment. The PLL technology started to be used when integrated circuits took over for many radio functions. The PLL could easily be integrated into the radio IC by simply adding a little extra circuitry to the IC. This added very little cost and only required a few external components - normally just resistors and capacitors which are cheap.

The PLL technology eliminates the costly RF transformers needed for circuits like the ratio FM detector and the Foster Seeley circuit. Typically a phase locked loop FM demodulator does not require the use of an inductor, let alone a transformer which is even more costly to manufacture.

PLL FM demodulation basics

The way in which a phase locked loop, PLL FM demodulator works is relatively straightforward. It requires no changes to the basic phase locked loop, itself, utilising the basic operation of the loop to provide the required output.

PLL Phase locked Loop FM demodulator

To look at the operation of the PLL FM demodulator take the condition where no modulation is applied and the carrier is in the centre position of the pass-band the voltage on the tune line to the VCO is set to the mid position. However if the carrier deviates in frequency, the loop will try to keep the loop in lock. For this to happen the VCO frequency must follow the incoming signal, and in turn for this to occur the tune line voltage must vary.

Monitoring the tune line shows that the variations in voltage correspond to the modulation applied to the signal. By amplifying the variations in voltage on the tune line it is possible to generate the demodulated signal. Although no basic changes to the phase locked loop are required for it to be able to demodulate FM, a buffer amplifier is typically provided from the tune line to prevent the tune line being loaded by other sections of the receiver. It provides a lower output impedance and as a result, this prevents loading from the audio amplifier from upsetting the loop in any way.

PLL FM demodulator performance

The PLL FM demodulator is normally considered a relatively high performance form of FM demodulator or detector. Accordingly they are used in many FM receiver applications.

The PLL FM demodulator has a number of key advantages:

Linearity: One of the advantages of the PLL FM demodulator is its high degree of linearity. This is governed by the voltage to frequency characteristic of the VCO within the phase locked loop.Normally the phase locked loop will be able to operate over a wide bandwidth - normally this is much wider than the bandwidth of the FM signal or even the IF stages of the FM receiver. As the frequency deviation of the incoming FM signal covers only a small portion of the PLL bandwidth the overall conversion is very linear.The VCO voltage to frequency curve is the main determining factor and this can be made to be very linear for the range needed for FM demodulation. Distortion levels for PLL FM demodulators are normally very low and are typically of the order of a tenth of a percent. This makes the PLL FM demodulator a very good option for high fidelity tuners as well as for many other applications including radio communications, etc.

Insensitive to amplitude noise: In general the phase locked loop FM demodulator is very insensitive to amplitude noise. As the phase locked loop will track the frequency of the incoming signal, it provides a relatively high degree of AM noise immunity. Obviously it can help if the IF amplifier of the radio is run into saturation such that the signal level is limited and noise is removed, but even on its own the PLL FM demodulator provides good noise immunity.

Ease of incorporation into ICs: Phase locked loops are very easy to implement in an integrated circuit. PLLs have long been available as ICs and this has meant that the technology is easy to implement. Also the PLL FM demodulator blocks are available for IC designers, and therefore many radio IF amplifier ICs have demodulators for AM and FM built in. Often the FM demodulator can be a phase locked loop demodulator.

Manufacturing costs: As the phase locked loop FM demodulator lends itself to integrated circuit technology, only a few external components are required to complete the FM demodulator. One particular advantage is that often no inductor is required for the VCO circuit.

As inductors are relatively expensive components, this can considerably reduce overall component costs and make this approach very attractive for large scale manufacture. These facts make the PLL FM demodulator particularly attractive for modern applications.

PLL FM demodulator design considerations

When designing a phase locked loop system for use as an FM demodulator, one of the key considerations is the loop filter. This must be chosen to be sufficiently wide that it is able to follow the anticipated variations of the frequency modulated signal. Accordingly the loop response time should be short when compared to the anticipated shortest time scale of the variations of the signal being demodulated.

A further design consideration is the linearity of the VCO. This should be designed for the voltage to frequency curve to be as linear as possible over the signal range that will be encountered, i.e. the centre frequency plus and minus the maximum deviation anticipated.

In general the PLL VCO linearity is not a major problem for average systems, but some attention may be required to ensure the linearity is sufficiently good for hi-fi systems. Phase locked loop FM demodulators are used in many radio receivers both domestic and professional for the demodulation of FM signals. The PLL FM demodulator provides a very attractive option in many instances, offering exceedingly low levels of distortion, and the ability to be incorporated into integrated circuit technology.

PLL FM Demodulator

A Phase-Locked Loop (PLL) is basically a negative feedback system. It consists of three major components such as re multiplier, a loop filter and a voltage controlled oscillator (VCO) connected together in the form of a feedback loop. A VCO is a sine wave generator whose frequency is determined by the voltage applied to it from an external source. It means that any frequency modulator can work as a VCO. A phase-locked loop (PLL) is primarily used in tracking the phase and frequency of the carrier component of an incoming FM signal. PLL is also useful for synchronous demodulation of AM-SC (i.e., Amplitude Modulation with Suppressed carrier) signals or signals with few cycles of pilot carrier. Further, PLL is also useful for demodulating FM signals in presence of large noise and low signal power.

This means that, PLL is most suitable for use in space vehicle-to-earth data links or where the loss along the transmission line or path is quite large. Recently, it has found application in commercial FM receivers.

The block diagram of a PLL is shown in Figureure.2.6.1 below.

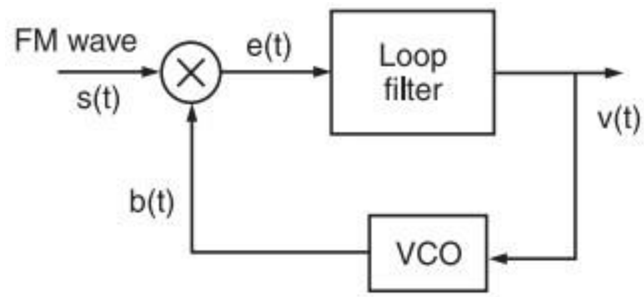


Figure. 2.6.1 The block diagram of a Phase-Locked Loop (PLL)

Diagram Source Brain Kart

Working Operation

The operation of a PLL is similar to any other feedback system where the feedback signal tends to follow the input signal. If the signal fed back is not equal to the input signal, the error signal will change the value of the fed back signal until it is equal to the input signal. The difference signal between s(t) and b(t) is called an error signal.A PLL operates on a similar principle except for the fact that the quantity feedback is not the amplitude, but a generalized phase Φ(t). The error signal or difference signal e(t) is utilized to adjust the VCO frequency in such a way that the instantaneous phase angle comes close to the angle of the incoming signal s(t). At this point, the two signals s(t) and b(t) are synchronized and the PLL is locked to the incoming signal s(t).

Mathematical Explanation

Here, we have assumed that the VCO is adjusted initially so that when the control voltage comes to zero, the following two conditions are satisfied:

(i) The frequency of the VCO is precisely set at the unmodulated carrier frequency fc
(ii) The VCO output has a 90° phase-shift w.r.t. the unmodulated carrier wave.

Let the input signal applied to the PLL be an FM wave. It is defined as ,

$$s(t) = A \sin [\omega_c t + \phi_1(t)] \quad (1)$$

where A is the unmodulated carrier amplitude and $\omega c = 2\pi fc$ = Angular carrier frequency and

$$\phi_1(t) = 2\pi k_f \int_0^t x(t)\, dt \quad (2)$$

where x(t) is the message or baseband signal or modulating signal

and kf = frequency sensitivity of frequency modulator. Let the VCO output be defined by,

$$b(t) = A_v \cos[\omega_c t + \phi_2(t)] \quad (3)$$

there Av = Amplitude of VCO output when the control voltage applied to the VCO is denoted

$$\phi_2(t) = 2\pi k_v \int_0^t v(t)\, dt$$

by v(t), then, we have, kv is the frequency sensitivity of VCO, measured in Hertz/volt.
It may be observed from equations (1) and (3) that the VCO output and the incoming signals are 90° out of phase, while the VCO frequency in absence of v(t) is precisely equal to the unmodulated frequency of the FM signal. The incoming FM have s(t) and the VCO output b(t) are applied to a multiplier. The output of the multiplier has the following components:

(i) A high frequency component represented by,

$$k_m A A_v \sin[2\omega_c t + \phi_1(t) + \phi_2(t)] \quad (5)$$

(ii) A low frequency component represented by,

$$k_m A A_v \sin[\phi_1(t) - \phi_2(t)] \quad (6)$$

where km = Multiplier Gain measured in per volt.

The high frequency component can be eliminated by using a filter. Hence, discarding the high frequency component, the effective input to the low pass filter (LPF) will be given by,

$$\phi_e(t) = \phi_1(t) - \phi_2(t)$$

This means that

$$\phi_e(t) = \phi_1(t) - 2\pi k_v \int_0^t v(t)\, dt \tag{7}$$

The loop filter operates on error signal e(t) to produce the output v(t). It is given by,

$$v(t) = \int_{-\infty}^{\infty} e(\tau)\, h(t-\tau)\, d\tau$$

where h(t) = Impulse response of the low-pass filter (LPF).

$$\phi_e(t) = \phi_1(t) - 2\pi k_m k_v A\, A_v \int_0^t \int_{-\infty}^{\infty} \sin[\phi_e(\tau)]\, h(t-\tau)\, d\tau dt$$

$$\phi_e(t) = \phi_1(t) - 2\pi k_o \int_0^t \int_{-\infty}^{\infty} \sin[\phi_e(\tau)]\, h(t-\tau)\, d\tau dt$$

Where $k_o = k_m\, k_v\, A\, A_v$

Now, differentiating both sides of equation , we get,

$$\frac{d\phi_e(t)}{dt} = \frac{d\phi_1(t)}{dt} - 2\pi k_o \int_{-\infty}^{\infty} \sin[\phi_e(\tau)]\, h(t-\tau)\, d\tau \tag{8}$$

Here, k_o has the dimension of frequency. On the basis of equation we can construct and equivalent model of PLL as shown in Figure. 2.6.2 given below.

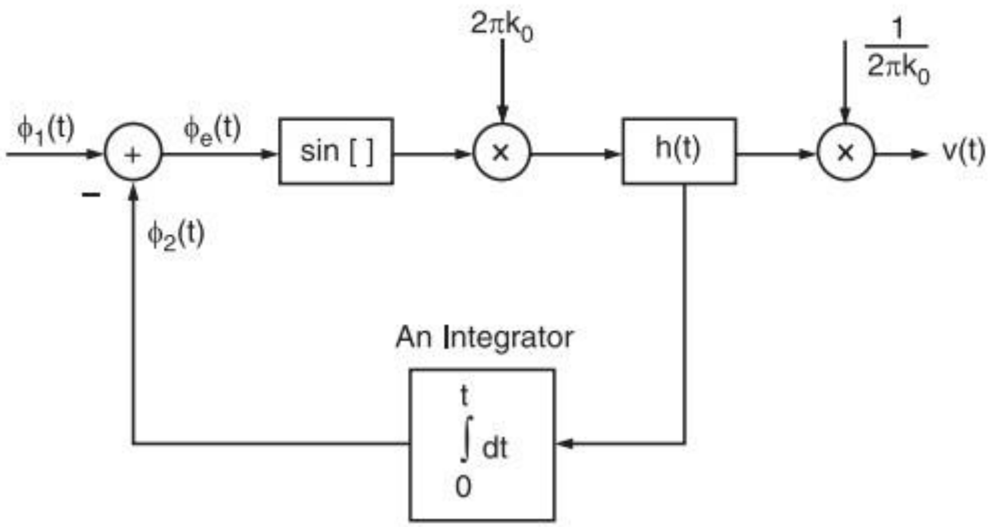


Figure 2.6.2 A non-linear equivalent model of PLL

Diagram Source Brain Kart

In this model, v(t) and e(t) are also included utiliZing the relationship between them as given in equations,

$$e(t) = k_m A A_v \sin[\phi_e(t)]$$

and

$$v(t) = \int_{-\infty}^{\infty} e(\tau)]\, h(t-\tau)\, d\tau$$

we can see that they are similar except for the fact that the multiplier in the equivalent model has been replaced by a subtractor and a sinusoidal non-linearity and the VCO by an integrator. When the phase error Φe(t) is zero, then PLL is said to be phase-locked. When the phase error Φe(t) at all times is small compared to 1 radian, then we can approximate sin[Φe(t)] as Φe(t), i.e.,

$$\sin[\phi_e(t)] \cong \phi_e(t) \tag{9}$$

It is almost accurate as long as Φe(t) is less than 0.5 radian. In this case, PLL is said to be Near-Lock Condition and the sinusoidal non-linearity can be discarded.

The linearized model of PLL is valid under above-mentioned condition as shown in Figure 2.6.3.

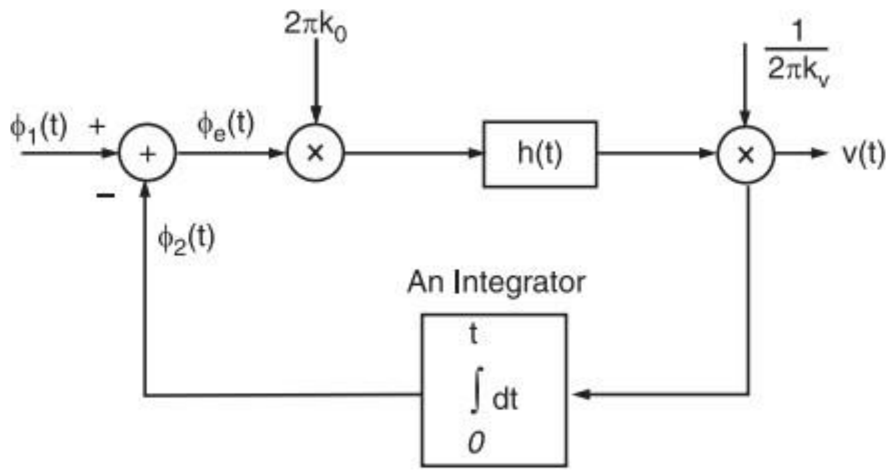


Figure. 2.6.3 Equivalent model of PLL

Diagram Source Brain Kart

In this model, phase error Φe(t) is related to the input phase Φ1(t) by the Integro-differential equation. It is expressed as,

$$\frac{d\phi_e(t)}{dt} + 2\pi k_o \int_{-\infty}^{\infty} \phi_e(\tau)\, h(t-\tau)\, d\tau = \frac{d\phi_1(t)}{dt}$$

Taking the Fourier transform of both sides of equation (10), we get,

$$\Phi_e(f) = \frac{1}{1 + k_o \dfrac{H(f)}{jf}} \Phi_1(f)$$

where Φe(f) and Φ1(f) are the Fourier transform of Φe(t) and Φ1(t), respectively and H(f) is the Fourier transform of impulse response h(t) and is known as transfer function of the loop filter. The quantity ko(H(f)/ jf is called the open loop transfer function of the PLL.

$$L(f) = \frac{k_o H(f)}{jf}$$

Substituting L(f) in the previous equation , we get,

$$\Phi_e(f) = \frac{1}{1 + L(f)} \Phi_1(f)$$

Now, let us consider that for all values of frequency f inside the baseband signal, we make the magnitude of L(f) very large compared to unity. Thus, from equation (10) we get,

$$\Phi_e(f) \to 0 \text{ as } L(f) >> 1 \tag{10}$$

Under above-mentioned condition, the phase of the VCO becomes asymptotically equal to the phase of the incoming wave and the phase lock is thereby established.

WIDE BAND FM MODULATION:

For large values of modulation index m_f, greater than 10 the FM wave ideally contains the carrier and an infinite number of sidebands located symmetrically around the carrier. Such a FM wave has infinite bandwidth and hence called as wideband FM. The modulation index of wideband FM is higher than 1. The maximum permissible deviation is 75 kHz and it is used in the entertainment broadcasting applications such as FM radio, TV etc.

$$s(t) = A_C \operatorname{Cos}(2\pi f_c t + \varphi(t)) \quad (1)$$

Finding its FT is not easy: $\phi(t)$ is inside the cosine. To analyze the spectrum, we use complex envelope. Consider single tone FM s(t) can be written as,

$$s(t) = ACcos(2\pi fct + \beta sin2\pi fm(t)) \quad (2)$$

Wideband FM is defined as the situation where the modulation index is above 0.5. Under these circumstances the sidebands beyond the first two terms are not insignificant. Broadcast FM stations use wideband FM, and using this mode they are able to take advantage of the wide bandwidth available to transmit high quality audio as well as other services like a stereo channel, and possibly other services as well on a single carrier.

The bandwidth of the FM transmission is a means of categorizing the basic attributes for the signal, and as a result these terms are often seen in the technical literature associated with frequency modulation, and products using FM. This is one area where the figure for modulation index is used.

GENERATION OF WIDEBAND FM SIGNALS:

The following two methods generate WBFM wave.

(1) Direct method

(ii) Indirect method

Direct Method

This method is called as the Direct Method because we are generating a wide band FM wave directly. In this method, Voltage Controlled Oscillator (VCO) is used to generate WBFM. VCO produces an output signal, whose frequency is proportional to the input signal voltage. This is similar to the definition of FM wave. The block diagram of the generation of WBFM wave is shown in the following figure 2.3.1.

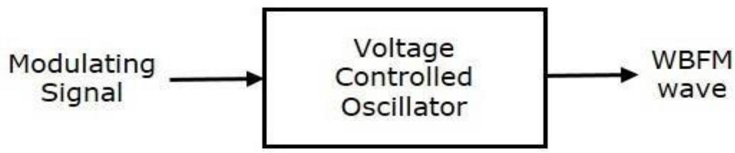


Figure 2.3.1 Block diagram of the generation of WBFM wave

Diagram Source Electronics Post

Here, the modulating signal m(t)m(t) is applied as an input of Voltage Controlled Oscillator (VCO). Voltage Controlled Oscillator produces an output, which is nothing but the WBFM.

$$Fi \alpha \ m(t)$$

$$fi=fc+kfm(t) \quad (3)$$

Where, fi is the instantaneous frequency of WBFM wave.

Indirect Method

This method is called as Indirect Method because we are generating a wide band FM wave indirectly. This means, first we will generate NBFM wave and then with the help of frequency multipliers we will get WBFM wave. The block diagram of generation of WBFM wave is shown in the following figure 2.3.2.

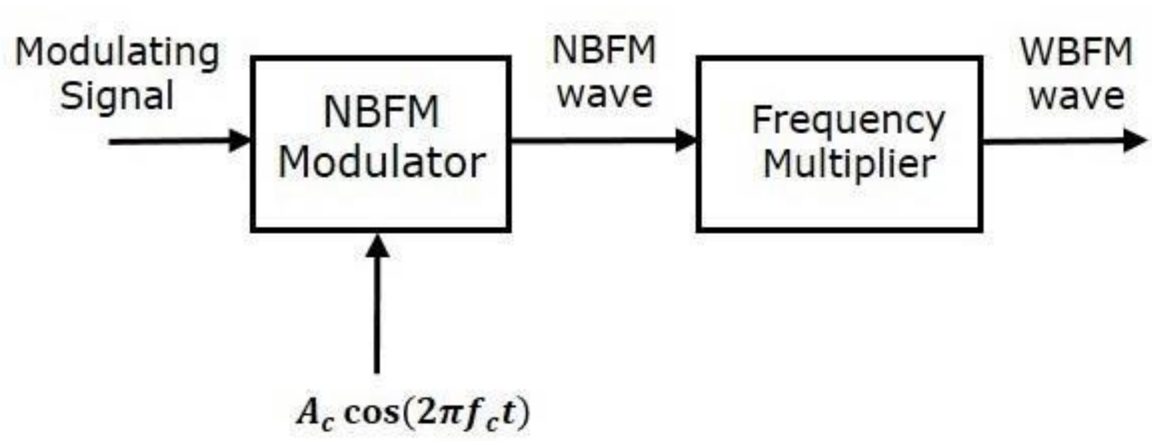


Figure 2.3.2 Block diagram of indirect method of generation of WBFM wave

Diagram Source Electronics Post

This block diagram in figure 2.3.2 contains mainly two stages. In the first stage, the NBFM wave will be generated using NBFM modulator. We have seen the block diagram of NBFM modulator at the beginning of this chapter. We know that the modulation index of NBFM wave is less than one. Hence, in order to get the required modulation index (greater than one) of FM wave, choose the frequency multiplier value properly. The generated narrowband FM signal can be converted to a wideband FM signal by simply passing it through a non–linear device with power P. Both the carrier frequency and the frequency deviation Df of the narrowband signal are increased by a factor P. Sometimes, the desired increase in the carrier frequency and the desired increase in Df are different. In this case, we increase Df to the desired value and use a frequency shifter (multiplication by a sinusoid followed by a BPF) to change the carrier frequency to the desired value.

Frequency multiplier is a non-linear device, which produces an output signal whose frequency is 'n' times the input signal frequency. Where, 'n' is the multiplication factor. If NBFM wave whose modulation index ββ is less than 1 is applied as the input of frequency multiplier, then the frequency multiplier produces an output signal, whose modulation index is 'n' times β and the frequency also 'n' times the frequency of WBFM wave. Sometimes, we may require multiple stages of frequency multiplier and mixers in order to increase the frequency deviation and modulation index of FM wave. For large values of modulation index mf , the FM wave ideally contains the carrier and an infinite number of sidebands located symmetrically around the carrier. Such a FM wave has infinite bandwidth and hence called as wideband FM.

The modulation index of wideband FM is higher than 1. The maximum permissible deviation is 75 kHz and it is used in the entertainment broadcasting applications such as FM radio, TV etc.

Frequency Spectrum of a Wideband FM wave

The expression for the wideband FM is complex since it is sine of sine function. The only way to solve this equation is by using the Bessel functions. By using the Bessel functions the equation for wideband FM wave can be expanded as follows :

$$eFM = s(t) = E_c\{J_0(m_f)sin\omega_c t + J_1(m_f)[sin(\omega_c + \omega_m)t - \sin((\omega_c - \omega_m)t] + J_2(m_f)[sin(\omega_c + 2\omega_m)t - \sin((\omega_c - 2\omega_m)t + J_3(m_f)[sin(\omega_c + 3\omega_m)t - \sin((\omega_c - 3\omega_m)t + J_4(m_f)[sin(\omega_c + 4\omega_m)t - \sin((\omega_c - 4\omega_m)t \quad (4)$$

Looking at equation (4), we can conclude the following points: The FM wave consists of carrier. The first term in equation(1) represents the carrier. The FM wave ideally consists of infinite number of sidebands. All the terms except the first one are sidebands. The amplitudes of the carrier and sidebands is dependent on the J coefficients. As the values of J coefficients are dependent on the modulation index mf , the modulation index determines how many sideband components have significant amplitudes as shown in fig.2 below.

The reason for this is that the amplitude of the FM signal i.e. Ec is always constant. AND the power transmitted is given by,

$$P_t = \frac{\left(\frac{E_c}{\sqrt{2}}\right)^2}{R} = \frac{{E_c}^2}{2R} \quad (5)$$

Where Ec = peak amplitude of FM wave. Therefore,

$$P_t = \frac{{E_c}^2}{2R} \quad If\, R = 1\Omega \quad (6)$$

Some of the J coefficients can be negative. Therefore, there is a 180o phase shift for that particular pair of sidebands. The carrier component does not remain constant. As J0(mf) is varying the amplitude of the carrier will also vary. However, the amplitude of FM wave will remain constant. For certain values of modulation index, the carrier component will disappear completely. These values are called Eigen values. In FM, the total transmitted power always remains constant. It is not dependent on the modulation index.