Frequency modulation

Sunday, February 13, 2005
2:10 PM

The modulating signal e_m, is used to vary the carrier frequency
e_m may be used to alter the capacitance of the carrier frequency oscillator circuit
Let the change in carrier frequency be $k \bullet e_m$,
k is a constant known as *the frequency deviation constant;* the instantaneous carrier frequency is

$$f_i = f_c + k \cdot e_m$$

with e_m a sine wave, The instantaneous carrier frequency becomes

$$f_i = f_c + k \cdot E_{m\max} \sin \omega_m t$$

The factor $k \bullet E_{mmax}$ is the peak-frequency deviation Δf. Thus Equation becomes

$$f_i = f_c + \Delta f \cdot \sin \omega_m t$$

In the more general case, e_m will be represented by $E_{m\max} \cdot m(t)$, where $m(t)$ is the modulating time function. Then

$$f_i = f_c + \Delta f \cdot m(t)$$

The unmodulated carrier is a sine wave, for which, E_{Cmax} may be set equal to unity:
The instantaneous frequency f_i is related to the modulation

$$e_c = \sin(\omega_c t + \phi)$$

The more general expression is

$$e = \sin \Theta(t)$$

The angular frequency of this general expression is the time rate of change of q*(t),*
When the frequency is varied, as in frequency modulation, an instantaneous angular frequency may be defined as

$$\omega_i = 2\pi f_i = \frac{d\Theta(t)}{dt}$$

$$\Theta(t) = \int \omega \, dt$$

For sinusoidal modulation,

$$\Theta(t) = \int 2\pi(f_c + \Delta f \cdot \sin \omega_m t)dt$$
$$= \omega_c t - \frac{\Delta f}{f_m} \cdot \cos \omega_m t + \phi \qquad \omega_m = 2\pi f_m.$$

Φ may be made equal to zero by appropriate choice of reference axis,
the equation for the sinusoidally frequency modulated wave obtained as

$$e = \sin\left(\omega_c t - \frac{\Delta f}{f_m} \cos \omega_m t\right)$$

The modulation index for frequency modulation is defined as $m_f = \frac{\Delta f}{f_m}$

The equation for the sinusoidally modulated carrier then becomes

$$e = \sin(\omega_c t - m_f \cos \omega_m t)$$

The modulation index for frequency modulation can be greater than unity.

Phase modulation results when the phase angle F of the carrier is a function of the modulating signal.
When phase modulated, Φ_c is replaced by Φ *(t),* where

$$\phi(t) = \phi_c + K \cdot e_m$$

K is the *phase deviation constant* (analogous to *k* for frequency modulation) and e_m is the modulating signal, Normally, Φ_c, can be dropped from the equation since it is a constant that does not affect the modulation
Also, letting $e_m = E_{mmax} \bullet m(t)$,

$$\phi(t) = \Delta\phi \cdot m(t)$$

$$\Delta\phi = K \cdot E_{m_{max}}$$

$\Delta\Phi$ is the peak phase deviation.
Substituting we have

$$e = \sin[\omega_c t + \Delta\phi \cdot m(t)]$$

For sinusoidal modulation

$$e = \sin(\omega_c t + \Delta\phi \cdot \sin\omega_m t)$$

Similar to sinusoidal frequency modulation, the peak phase deviation is termed the phase modulation index with symbol m_p giving

$$e = \sin(\omega_c t + m_p \sin\omega_m t)$$

EQUIVALENCE BETWEEN FM AND PM

The instantaneous angular frequency as defined earlier is

$$\omega_i = \frac{d\Theta(t)}{dt}$$

For a phase-modulated wave, the phase modulation is expressed by

$$\Theta(t) = \omega_c t + \phi(t)$$

Therefore, phase modulation has an equivalent angular frequency

$$\omega_{i_{eq}} = \omega_c + \frac{d\phi(t)}{dt}$$

Let f_{eq} *(t)* represent the equivalent frequency modulation then

$$\omega_c + 2\pi f_{eq}(t) = \omega_c + \frac{d\phi(t)}{dt}$$

$$f_{eq}(t) = \frac{1}{2\pi}\frac{d\phi(t)}{dt}$$

All angle-modulation receivers interpret angle modulation always as frequency modulation - actual or equivalent

Narrowband FM

Thursday, March 10, 2005
2:11 AM

- Expanding the FM equation, with modulation index denoted by β we get

$$s(t) = A_c \cos(2\pi f_c t)\cos[\beta \sin(2\pi f_m t)] - A_c \sin(2\pi f_c t)\sin[\beta \sin(2\pi f_m t)]$$

- Assuming that the modulation index is small compared to one radian, use the following approximations:

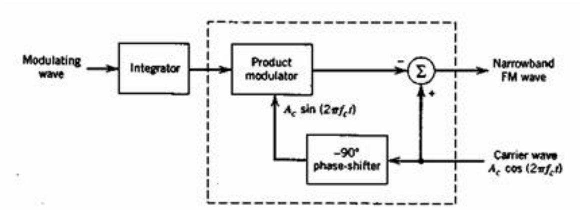
$$\cos[\beta \sin(2\pi f_m t)] \simeq 1$$

$$\sin[\beta \sin(2\pi f_m t)] \simeq \beta \sin(2\pi f_m t)$$

- Hence, Equation simplifies to

$$s(t) \simeq A_c \cos(2\pi f_c t) - \beta A_c \sin(2\pi f_c t)\sin(2\pi f_m t)$$

- The modulator shown in block diagram form in Figure can generate this FM.



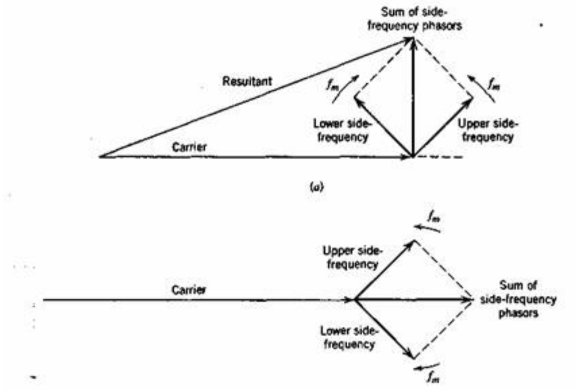
- Its output has two differences from ideal FM
    - The envelope contains a *residual* amplitude modulation and, therefore, varies with time.
    - For a sinusoidal modulating wave, the angle contains *harmonic distortion in* the form of third- and higher-order harmonics of the modulation frequency f_m.
- By restricting the modulation index to 0.3 radians, the effects of residual AM and harmonic PM are limited to negligible levels
- Equation for NBFM can be written as

$$s(t) \simeq A_c \cos(2\pi f_c t) + \frac{1}{2}\beta A_c\{\cos[2\pi(f_c + f_m)t] - \cos[2\pi(f_c - f_m)t]\}$$

- This is similar to the corresponding one defining an AM signal which is as follows:

$$s_{AM}(t) = A_c \cos(2\pi f_c t) + \frac{1}{2}\mu A_c\{\cos[2\pi(f_c + f_m)t] + \cos[2\pi(f_c - f_m)t]\}$$

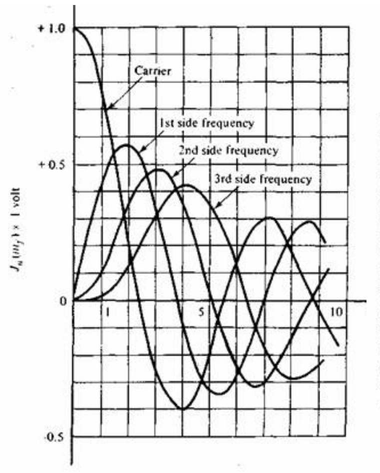
- In the case of sinusoidal modulation, the basic difference between an AM signal and a narrowband FM signal is that the algebraic sign of the lower side frequency in the narrowband FM is reversed.
- A narrowband FM signal requires the same transmission bandwidth (i.e., *2f_m)* as the AM signal.
- We can represent the narrowband FM signal with a phasor diagram as shown in Figure *a,*
  Compared to corresponding AM phasor diagram it is seen to have the assembly of vectors at tip of carrier rotated by 90 degrees.



- The spectrum consist of a carrier component, and side frequencies at harmonics of the modulating frequency even though no harmonics are present in the original modulating tone.

$$
\begin{aligned}
&= A \sin(\omega_c t + m \sin \omega_m t) \\
&= A \{J_0(m) \sin \omega_c t \\
&\quad - J_1(m)[\sin(\omega_c - \omega_m)t - \sin(\omega_c + \omega_m)t] \\
&\quad + J_2(m)[\sin(\omega_c - 2\omega_m)t + \sin(\omega_c + 2\omega_m)t] \\
&\quad - J_3(m)[\sin(\omega_c - 3\omega_m)t + \sin(\omega_c + 3\omega_m)t] \\
&\quad + \cdots \}
\end{aligned}
$$

- The amplitudes of the various spectral components are given by Bessel's Function of the First Kind.



Modulation Index m_f	Carrier	Side Frequencies											
		1st	*2nd*	*3rd*	*4th*	*5th*	*6th*	*7th*	*8th*	*9th*	*10th*	*11th*	*12th*
0.25	0.98	0.12	0.01										
0.5	0.94	0.24	0.03										
1.0	0.77	0.44	0.11	0.02									
1.5	0.51	0.56	0.23	0.06	0.01								
2.0	0.22	0.58	0.35	0.13	0.03	0.01							
2.4	0	0.52	0.43	0.20	0.06								
3.0	−0.26	0.34	0.49	0.31	0.13	0.04	0.01						
4.0	−0.40	−0.07	0.36	0.43	0.28	0.13	0.05	0.02					
5.0	−0.18	−0.33	0.05	0.36	0.39	0.26	0.13	0.05	0.02	0.01			
5.5	0	−0.34	−0.12	0.26	0.40	0.32	0.19	0.09	0.03	0.01			
6.0	0.15	−0.28	−0.24	0.11	0.36	0.36	0.25	0.13	0.06	0.02	0.01		
7.0	0.30	0	−0.30	−0.17	0.16	0.35	0.34	0.23	0.13	0.06	0.02	0.01	
8.0	0.17	0.23	−0.11	−0.29	−0.10	0.1°	0.34	0.32	0.22	0.13	0.06	0.03	0.01
8.65	0	0.27	0.06	−0.24	−0.23	0.03	0.26	0.34	0.28	0.18	0.10	0.05	0.02

- The spectrum component at the carrier frequency decreases in amplitude does *not* mean that the carrier wave is amplitude modulated.

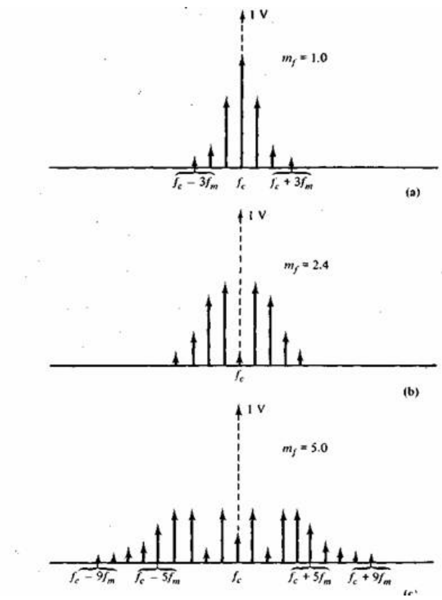
- The carrier wave is the sum of all the components in the spectrum, and these add up to give a constant amplitude carrier .

- The distinction is that the modulated carrier is not a sine wave, whereas the spectrum component at carrier frequency is. (All spectrum components are either sine or cosine waves.)

- for certain values (2.4, 5.5, 8.65), the carrier amplitude goes to zero. It is the sinusoidal component of the spectrum, at carrier frequency, which goes to zero, *not* the modulated carrier, which is nonsinusoidal and which varies from positive to negative peak

.

- The spectra for various values of m are shown in (a), (b), and (c). In each case the spectral lines are spaced by f_m, and the bandwidth of the spectrum is
  $B = 2 \cdot n \cdot f_m$
  where *n* is the highest order of side frequency for which the amplitude is significant.



- When the order of side frequency is greater than (m_f + 1), the amplitude is 5% or less of unmodulated carrier amplitude.

- Thus, Taking n = m_f +1
  $$B = 2(m_f + 1)f_m$$
  - $$B = 2(\Delta f + f_m)$$
- Example

$\Delta f = 75$ kHz, $f_m = 0.1$ kHz:
$$B = 2(75 + 0.1)$$
$$= \mathbf{150\ kHz}$$
$\Delta f = 75$ kHz, $f_m = 1.0$ kHz:
$$B = 2(75 + 1)$$
$$= \mathbf{152\ kHz}$$
$\Delta f = 75$ kHz, $f_m = 10$ kHz:
$$B = 2(75 + 10)$$

- In the frequency-modulation process, beat frequencies occur between the various side frequencies when the modulation signal is other than sinusoidal.

- The bandwidth requirements are determined by the maximum frequency deviation and maximum modulation frequency (harmonic) present in the complex modulating wave.
- The ratio of maximum deviation to maximum frequency component is termed the *deviation ratio.* Denoting this by M,
  $$M = \frac{\Delta F}{F_m}$$
  Where ΔF is the maximum frequency deviation and F_m the highest frequency component in the modulating signal.

- The bandwidth is then given by
  $$B_{max} = 2(M + 1)F_m$$
  $$= 2(\Delta F + F_m)$$

This is called *Carson's rule,*

Average Power

Average Power is proportional to amplitude, and amplitude is constant in FM

The average power in a sinusoidally frequency-modulated wave remains constant at the unmodulated value.

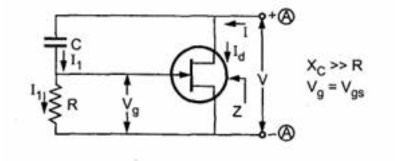
As modulation is applied power is redistributed over the various spectral components hence the amplitude of the carrier component decreases

Since the average power does not change with frequency modulation, the rms voltage, and current will also remain constant, at respective unmodulated values.

- There are a number of devices whose reactance can be varied by the application of voltage.
- These include FET and BJT, varactor diode etc.
- If such a device is placed across the tank circuit of the L-C oscillator, then FM will be produced when the reactance of the device is varied by the modulating voltage.
- At the carrier frequency, the oscillator inductance is tuned by its own capacitance in parallel with the average reactance to the variable reactance device.

FET Reactance Modulator

- Neglecting gate current, let the current through C and R be I_1,
- At the carrier frequency, the reactance of C is much larger than R



Then,

$$I_1 = \frac{V}{R + \frac{1}{j\omega C}}$$

$$\approx \frac{V}{\frac{1}{j\omega C}} = j\omega CV$$

$$V_g = I_1 R = j\omega CRV$$

$$I_d = g_m V_{gs} = g_m V_g$$

$$= j\omega CRg_m V$$

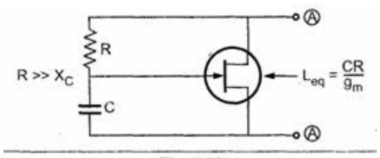
Then impedance of FET is,

$$Z = \frac{V}{I_d} = \frac{1}{j\omega[g_m CR]} = \frac{1}{j\omega[C_{eq}]}$$

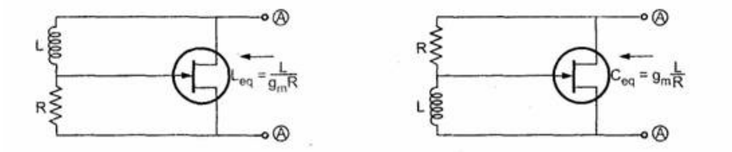
Thus impedance is seen to be capacitive reactance with ,

$$C_{eq} = g_m CR.$$

- By modulating voltage the operating point of FET i.e. g_m can be varied and hence equivalent capacitance changes
- Since the equivalent capacitance depends upon g_m, which in turn, is dependent on bias voltage of FET, C_{eq} can be varied by varying bias voltage with modulating signal.
- By selecting values of R and C, C_{eq} can be initially adjusted to the desired value, in unmodulated conditions.
- if Xc/R is not much greater than unity, then equivalent impedance has resistive component, resulting in certain amplitude modulation.
- By interchanging R and C, and selecting the value of R such that R is » X_c ; then inductive reactance can be obtained, as shown in Fig.,



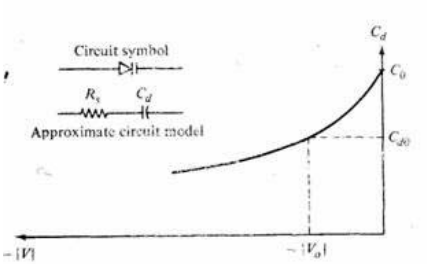
- The other two possible circuits are shown in the Fig. below using L and R

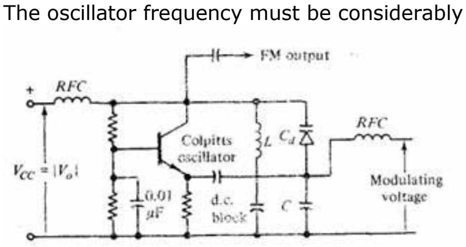


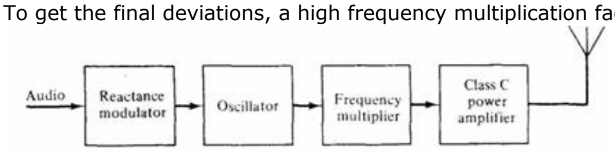
(a) $X_L >> R$

(b) $R >> X_L$

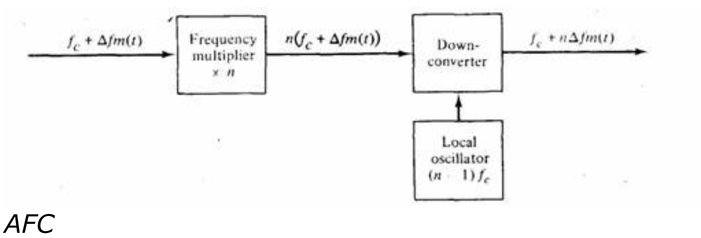
The Varactor Diode

- The depletion capacitance of the v*aractor diode* varies with reverse bias.
- $$C_d = \frac{C_0}{\left(1 + \frac{|V|}{\Phi}\right)^a}$$
- C_0 is the capacitance at zero bias
- ϕ is the contact potential of the junction (volts) for silicon, ϕ= 0.5 V.,
- The index *a* depends on the type of junction. For abrupt junctions, *a = 0.5* .
- 

- The diode is represented by the equivalent circuit of the depletion capacitance C_d in series with a resistance R_s, which represents the resistance of the bulk *pn* regions, and the lead resistance.
- Modulation voltage is applied as reverse bias,
- The resulting variation in C_d is used to vary either the frequency or the phase of an LC-tuned circuit.
- The oscillator voltage must produce negligible variation in C_d.
- The oscillator frequency must be considerably higher than the modulating frequency.
- 

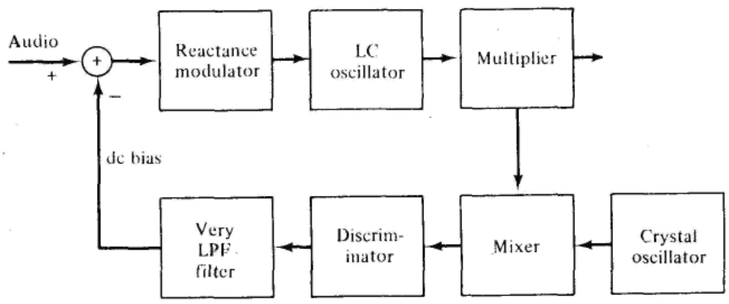
- A conflict exists between obtaining adequate frequency deviation while maintaining frequency stability.
- Crystal oscillators can be directly frequency modulated for small deviations since the crystal frequency can be "pulled" by a small amount,
- To get the final deviations, a high frequency multiplication factor is necessary
- 

- An index of 0.5 (narrowband FM) for 50 Hz frequency with a 1 Mhz carrier requires deviation of 25 Hz while the standard value of the deviation in broadcast FM is 75 KHz requiring a multiplication factor of 3000
- The factor of 3000 also multiplies the frequency of the NBFM carrier, taking the output carrier frequency to 3mhz x 3000 = 3 GHz - too high for commercial use which requires about 90MHz carrier.
- Conversely to get 90 MHz carrier, NBFM carrier should be 90Mhz/3000 = 3 KHz which is too low to design a relatively stable oscillator economically.
- This dilemma is solved by using downconvertor after multipliers to bring down the transmitted carrier frequency
- Class C power amplifiers can be used for FM transmitters because any small variations in the amplitude of the FM signal are usually removed in the receiver circuits by limiting amplifiers.
- Class C does not have any significant effect on the modulation itself, and interference from noise is greatly reduced.
- The result is that the FM transmitter is much more efficient than an equivalent AM transmitter.



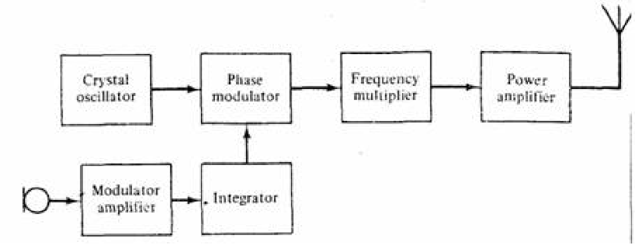
AFC

- If the main oscillator is *LC* oscillator, the direct-modulation scheme does not meet frequency stability regulations.
- Stability is improved by an AFC circuit.
- A sample of the final output signal is mixed with the signal from a stable crystal oscillator.
- The IF produced contains the difference frequency between the carrier and the fixed oscillator.
- A discriminator circuit generates a voltage which is proportional to this difference frequency.
- It also contains the modulation signal, and a low-pass filter is used to remove this,
- This leaves a varying dc level which is proportional to the difference between the carrier frequency and the oscillator.
- This voltage is added to the modulating audio signal and applied to the reactance modulator in a manner so as to correct any drift in the main oscillator frequency.



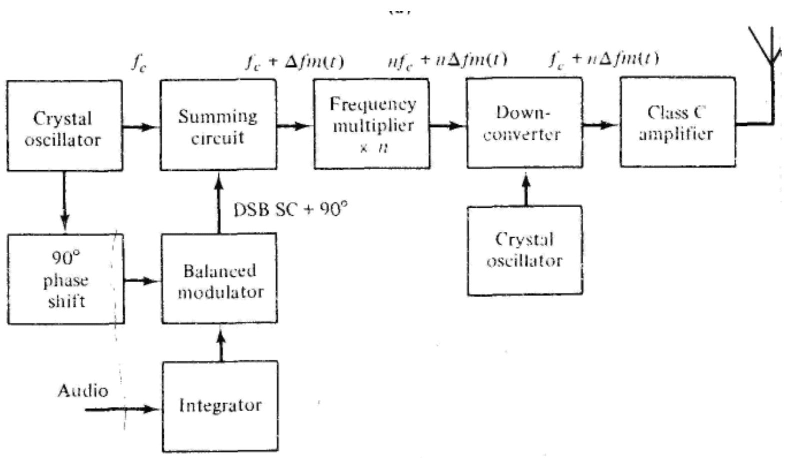
Indirect FM

- Phase modulation is used to achieve frequency modulation in the indirect method,
- It is necessary to integrate the modulating signal prior to applying it to the phase modulator,
- This transmitter is widely used in VHF and UHF radio telephone equipment.



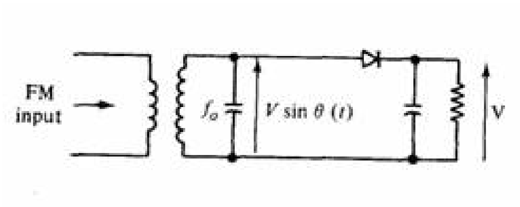
Armstrong method

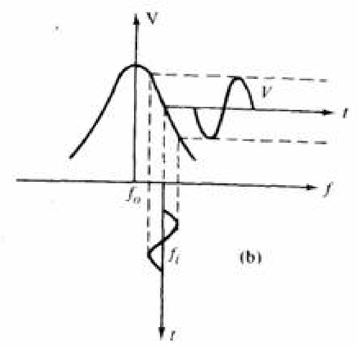
- The carrier source is a crystal oscillator with a low frequency , say 100 kHz, because of the need to do several stages of frequency multiplication.
- It is indirect as WBFM is produced from NBFM / Phase modulation
- A sample of the carrier is separated and shifted by 90° before application to a balanced modulator.
- The audio is passed through an integrator circuit before being applied to the modulator.
- The output of the balanced modulator is reduced in amplitude so that is very small compared to the oscillator output.
- This output is the phasor sum of the two sidebands, without any carrier and is displaced in phase by 90° from the oscillator output.
- The sidebands are added to the oscillator output so that phase modulation occurs.
- As the audio signal is integrated before the phase modulation this is effective narrow band frequency modulation by the audio
- Multipliers and down-convertor are used to adjust the frequency deviation and carrier frequency to required values to get WBFM



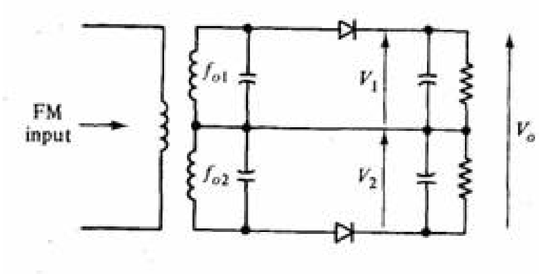
Detection of FM Signals

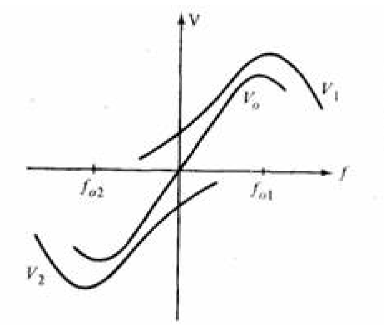
- To detect an FM signal, a circuit whose output voltage varies *linearly* with the frequency of the input signal is required.
- The L-R circuit and the slope detector are basic circuits with this property, although its linearity of response is not good.





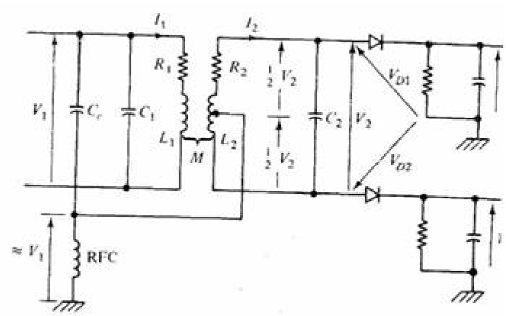
- By tuning the slope detector circuit to receive the signal on the slope of the response curve, carrier amplitude varies with frequency.
- The carrier is now both amplitude and frequency modulated.
- The modulation is recovered from the amplitude modulation by means of a normal envelope detector. However, the linear range on the voltage/ frequency-transfer characteristic is limited.
- Linearity can be improved by the arrangement of Figure below, the *Round-Travis detector,* or *balanced slope detector.*





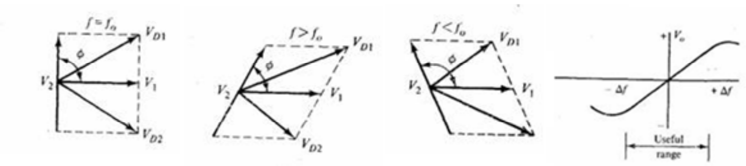
- One slope detector is tuned to resonance above the incoming carrier frequency, and the other to resonance below the carrier frequency,
- The envelope detectors combined to give a differential output.
- The output, which is $V_0 = |V_1| - |V_2|$, will have an S shape when plotted against frequency as shown in figure.
- This is the characteristic of FM detectors, and the linear range should be as large as possible.
- When the incoming signal is unmodulated, the output is balanced to zero
- When the carrier deviates toward f_{01} , V_1 increases while V_2 decreases, and the output goes positive; When it deviates toward f_{o2}, V_1 decreases while V_2 increases, and the output goes negative
- Linearity in the intervening range is better as the concavity of upper curve cancels convexity of the lower.

Foster-Seeley Discriminator



(a)

- The Foster-Seeley discriminator uses the phase-angle shift between primary and secondary voltages of a tuned transformer.
- The phase angle is a function of frequency,
- The phasor-sum and phasor-difference components of primary and secondary voltages are applied to two envelope detectors,
- The outputs of detectors are then combined, and demodulation is achieved.
- The circuit relies on phase-angle variation, hence it is also known as a *phase discriminator,*
- It converts a frequency variation to a circuit phase-angle variation, which in turn is converted to an amplitude variation.
- The full primary voltage, approximately, appears across the radio-frequency choke (RFC), at carrier frequency.
- The connection is such that the radio-frequency voltage applied to diode D_1 is $V_1 + 1/2\ V_2$, and to diode D2, $V_1 - 1/2V_2$.
- The phasor sum of $V_1 \pm 1/2\ V_2$ *is* then as shown in Figure for three different conditions of carrier frequency.



- The envelope detector D_1 will produce an output voltage proportional to $|V_{D1}|$. and that of D_2, an output voltage proportional to $|V_{D2}|$.
- The output of the detectors is

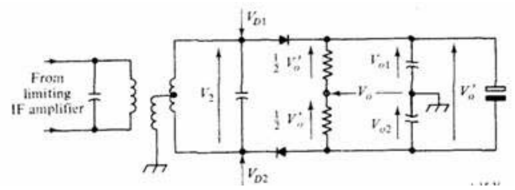
$$V_o = V_{o1} - V_{o2}$$
$$= K(|V_{D1}| - |V_{D2}|)$$

  where *K* is a constant of the detector circuits.
- As the frequency increases, the phase shift decreases and V_{01} increases while V_{02} decreases; hence, V_0 will increase.
- When frequency decreases phase shift increases and Vo will decrease

Ratio Detector

Monday, April 11, 2005
12:33 PM

Ratio Detector

- 

- Diodes D_1 and D_2 and loads *RC* form envelope detectors as before
- The frequency-to-phase-to-amplitude conversions occur as in the Foster-Seeley discriminator.
- However, the polarity of voltage in the lower capacitor is reversed, so the sum voltage appears across the combined loads (rather than the difference voltage as in the Foster-Seeley).
- Hence, as V_{01} increases, V_{o2} decreases and V'_0 remains constant (and, also, it remains constant as V_{01} decreases and $V_{0\,2}$ increases).
- Therefore, a large capacitor (electrolytic) can be connected across the V'_0 points without affecting the voltage .
- From the circuit of Figure equations can be written for the output voltage V_o ,

$$V_o = \tfrac{1}{2}V'_o - V_{o2}$$

and

$$V_o = -\tfrac{1}{2}V'_o + V_{o1}$$

Adding,

$$2V_o = V_{o1} - V_{o2}$$

Therefore,

$$V_o = \tfrac{1}{2}(V_{o1} - V_{o2})$$

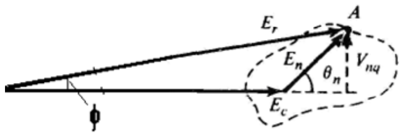
or

$$V_o = \tfrac{1}{2}K(|V_{D1}| - |V_{D2}|)$$

- The output voltage is one-half that of the Foster-Seeley circuit.
- Limiting action occurs as a result of variable damping on the secondary of the transformer.
- For example, if the input voltage amplitude were to suddenly increase, as would occur with a noise spike of voltage, the voltage V'_0 could not follow immediately, since it is held constant by means of the large capacitor.
- The voltage across the diodes in series is $V_2 - V'_0$ and since V_2 increases with V_1, the diodes conduct more heavily.
- This results in heavier damping of the secondary (which is also reflected into the primary), which reduces the *Q* factor.
- This, in turn, tends to cancel the increase in V_2 by reducing the gain of the limiting amplifier feeding the circuit.

NOISE IN FREQUENCY MODULATION

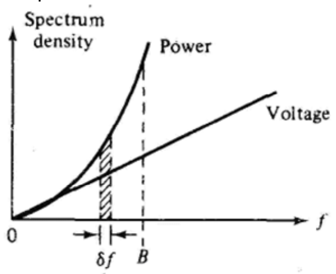
- Noise produces both amplitude and phase modulation.
- In FM receiver, the amplitude modulation is removed by limiters, while the phase modulation is detected as output noise.
- The noise voltage at the limiter input may be represented by a phasor at the carrier frequency but which has a randomly varying amplitude *E_n* and a randomly varying phase q_n with respect to the carrier phasor *E_c,* as shown in Figure.



- The tip of the resultant phasor *E_T,* the tip of this, point A, will trace out a random path shown dotted.
- For *E_c* > |*E_n*| the phase modulation is given approximately by

$$\Phi_n = \tan^{-1}\left(\frac{V_{nq}}{E_c}\right) \cong \frac{V_{nq}}{E_c}$$

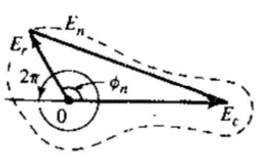
where V_{nq} is the quadrature component of the noise phasor,
- For phase detector the demodulated output voltage is proportional to the rate of change of the phase.
- Hence the amplitude spectrum density for the noise voltage is proportional to frequency.
- The power-spectrum density for the noise output is, therefore, proportional to frequency squared.
- The spectrum densities are sketched in figure.



- Since noise spectrum density increases with frequency, the signal-to-noise ratio is degraded at high audio frequencies.
- Good reproduction of these frequencies is required for good articulation efficiency of speech.
- The problem is solved by using Pre-emphasis of the high frequencies at the transmitter.
- A de-emphasis filter in the receiver audio section which removes the artificial pre-emphasis.
- Compared with 100% modulated AM, the S/N of FM can be shown to be given by:

$$S/N_{FM} = 4.5\left(\frac{\Delta f}{B}\right)^2 \cdot S/N_{AM}$$

- FM offers 20dB improvement over AM for index greater than 0.5 i.e. for wideband AM
- The above equation is valid under assumption that carrier is much greater than noise
- When carrier and noise are comparable as in out of range mobile receivers the phasor diagram shows that even 360 degree rotation of resultant is possible, resulting in spikes, and degrades performance.



(d)

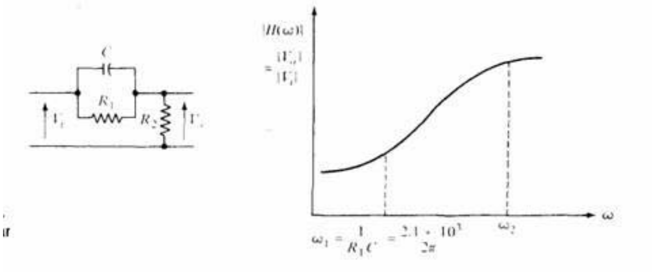
- The input signal level below which the s/n falls by 1 dB below the value given by the equation derived for case of Ec >>En

$$S/N_{FM} = \frac{3}{2kT} \cdot \frac{P_c(\Delta f)^2}{FB^3}$$

is called the FM Threshold

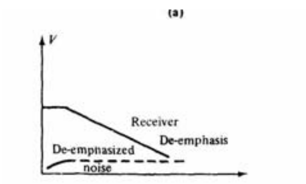
Preemphasis

- Pre-emphasis artificially boosts the power in the higher frequencies at which noise has a greater effect
- It is followed by de-emphasis introduced in the receiver to restore the frequency components to their natural values
- The net effect is to improve the signal-to-noise ratio.
- Pre-emphasis network is shown in figure along with the transfer function.
- Over the pre-emphasis range, the transfer function is
-
  - $$H(\omega) \cong \frac{R_2}{R_1 + R_2}\left(1 + j\frac{\omega}{\omega_1}\right)$$
- where $w_1 = (1/CR_1)$ and $f_1 = w_1/2\pi$ is standardized at 2.1 kHz.
- The frequency (w_2) at which pre-emphasis levels off is chosen to be above the bandpass.
- Phase modulation by itself provides the required pre-emphasis characteristic if the integrator is modified to be ineffective above the 2.1 kHz pre-emphasis cutoff frequency.

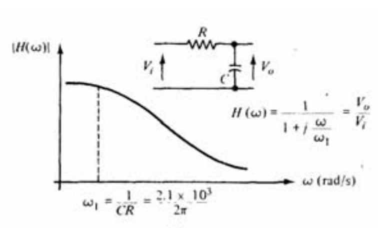


Deemphasis

- The noise voltage output resulting from noise phase-modulation increased directly in proportion to frequency, or at 6 dB per octave.
- A *de-emphasis network,* which attenuates at 6 dB per octave, levels the noise spectrum, thus improving the signal-to-noise ratio.



- Typical de-emphasis network, along with its transfer function, is shown in Figure.



- The transfer function is given by

$$H(\omega) = \frac{1}{1 + j\frac{\omega}{\omega_1}}$$

  where $w_1 = 1\,/CR$. The time-constant CR is standardized at 75 us, and thus f_l = 2.1 kHz.
- The reduction in average noise power is determined by the ratio of the area of the power output spectrum with and without de-emphasis, and is

$$\text{deemphasis improvement} = \frac{1}{3}\left(\frac{B}{f_1}\right)^2$$

  where B is the audio bandwidth. For f_1 =2.1 kHz and B = 15 kHz, the improvement is 12.3 dB.
- To compensate for de-emphasis of the modulating signal, a matching pre-emphasis network must be used at the transmitter.

FM receiver

Monday, April 11, 2005
6:15 PM