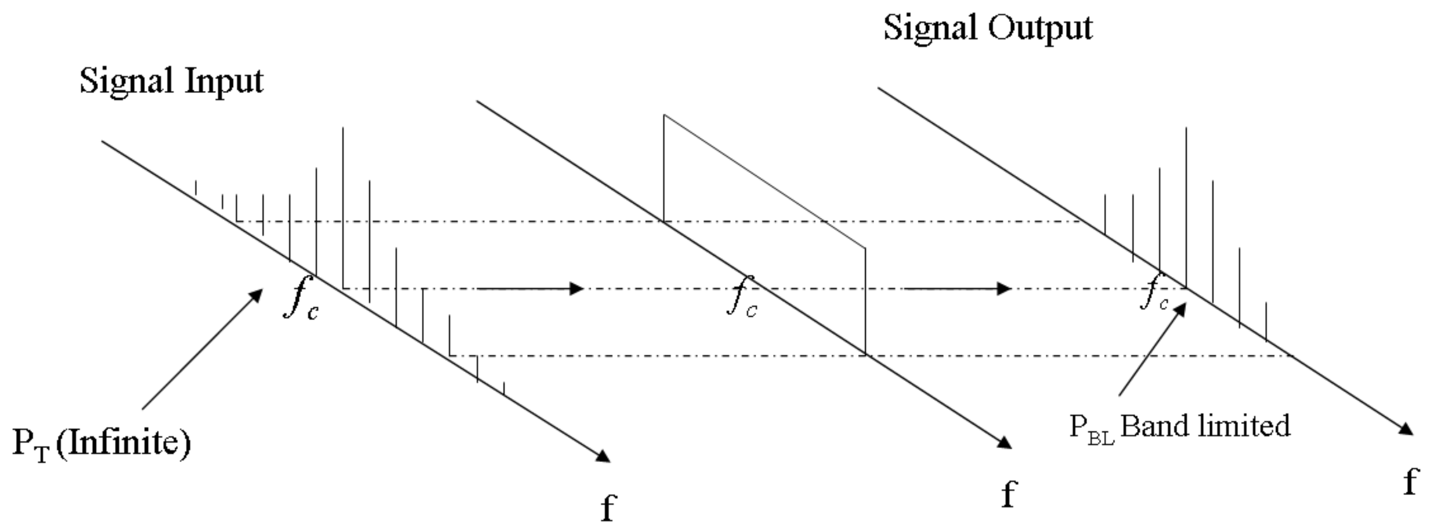
Power in FM Signals.

However, many signals (*e.g.* FM, square waves, digital signals) contain an infinite number of components. If we transfer such a signal via a limited channel bandwidth, we will lose some of the components and the output signal will be distorted. If we put an infinitely wide train through a tunnel, the train would come out distorted, the question is how much distortion can be tolerated?

Generally speaking, spectral components decrease in amplitude as we move away from the spectrum 'centre'.



Power in FM Signals.

In general distortion may be defined as

$$D = \frac{\text{Power in total spectrum - Power in Bandlimited spectrum}}{\text{Power in total spectrum}}$$

$$D = \frac{P_T - P_{BL}}{P_T}$$

With reference to FM the minimum channel bandwidth required would be just wide enough to pass the spectrum of significant components. For a bandlimited FM spectrum, let a = the number of sideband pairs, *e.g.* for $\beta = 5$, $a = 8$ pairs (16 components). Hence, power in the bandlimited spectrum P_{BL} is

$$P_{BL} = \sum_{n=-a}^{a} \frac{(V_c J_n(\beta))^2}{2}$$ = carrier power + sideband powers.

Power in FM Signals.

Since $P_T = \frac{V_c^2}{2}$

Distortion $D = \frac{\frac{V_c^2}{2} - \frac{V_c^2}{2}\sum_{n=-a}^{a}(J_n(\beta))^2}{V_c^2/2} = 1 - \sum_{n=-a}^{a}(J_n(\beta))^2$

Also, it is easily seen that the ratio

$$D = \frac{\text{Power in Bandlimited spectrum}}{\text{Power in total spectrum}} = \frac{P_{BL}}{P_T} = \sum_{n=-a}^{a}(J_n(\beta))^2$$ = 1 – Distortion

i.e. proportion p_f power in bandlimited spectrum to total power = $\sum_{n=-a}^{a}(J_n(\beta))^2$

Example

Consider NBFM, with β = 0.2. Let V_c = 10 volts. The total power in the infinite spectrum $\frac{V_c^2}{2}$ = 50 Watts, *i.e.* $\sum_{n=-a}^{a}(J_n(\beta))^2$ = 50 Watts.

From the table – the significant components are

n	$J_n(0.2)$	$\text{Amp} = V_c J_n(0.2)$	$\text{Power} = \frac{(Amp)^2}{2}$
0	0.9900	9.90	49.005
1	0.0995	0.995	0.4950125
			$P_{BL} = 49.5$ Watts

i.e. the carrier + 2 sidebands contain $\frac{49.5}{50} = 0.99$ or 99% of the total power

Example

Distortion = $\dfrac{P_T - P_{BL}}{P_T} = \dfrac{50 - 49.5}{50} = 0.01$ or 1%.

Actually, we don't need to know V_c, *i.e.* alternatively

Distortion = $1 - \sum_{n=-1}^{1} (J_n(0.2))^2$ (a = 1)

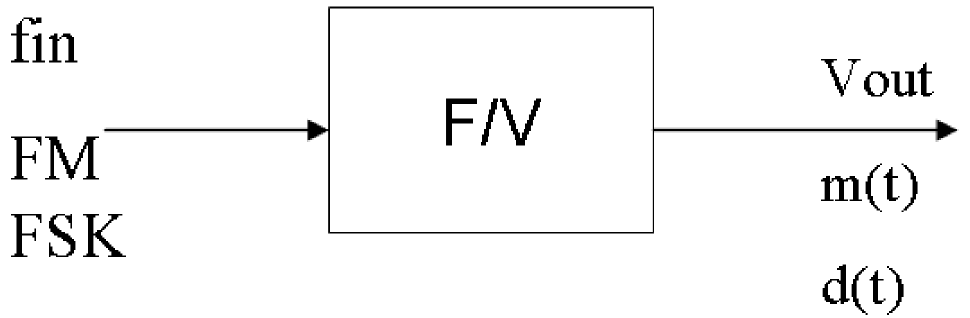
D = $1 - (0.99)^2 - (0.0995)^2 = 0.01$

Ratio $\dfrac{P_{BL}}{P_T} = \sum_{n=-1}^{1} (J_n(\beta))^2 = 1 - D = 0.99$

FM Demodulation –General Principles.

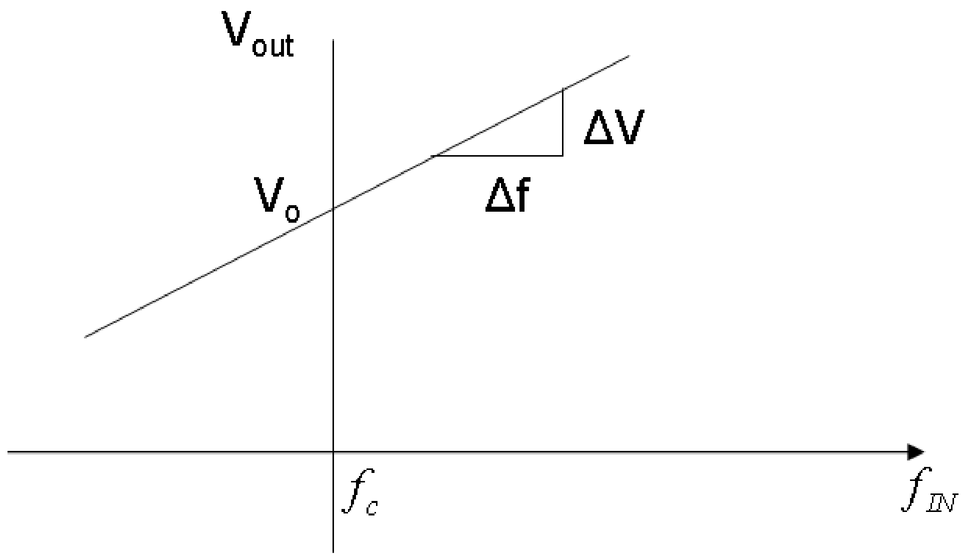
- An FM demodulator or frequency discriminator is essentially a frequency-to-voltage converter (F/V). An F/V converter may be realised in several ways, including for example, tuned circuits and envelope detectors, phase locked loops *etc.* Demodulators are also called FM discriminators.

- Before considering some specific types, the general concepts for FM demodulation will be presented. An F/V converter produces an output voltage, V_{OUT} which is proportional to the frequency input, f_{IN}.



FM Demodulation –General Principles.

- If the input is FM, the output is *m*(*t*), the analogue message signal. If the input is FSK, the output is *d*(*t*), the digital data sequence.
- In this case f_{IN} is the independent variable and V_{OUT} is the dependent variable (*x* and *y* axes respectively). The ideal characteristic is shown below.



We define V_o as the output when $f_{IN} = f_c$, the nominal input frequency.

FM Demodulation –General Principles.

The gradient $\frac{\Delta V}{\Delta f}$ is called the voltage conversion factor

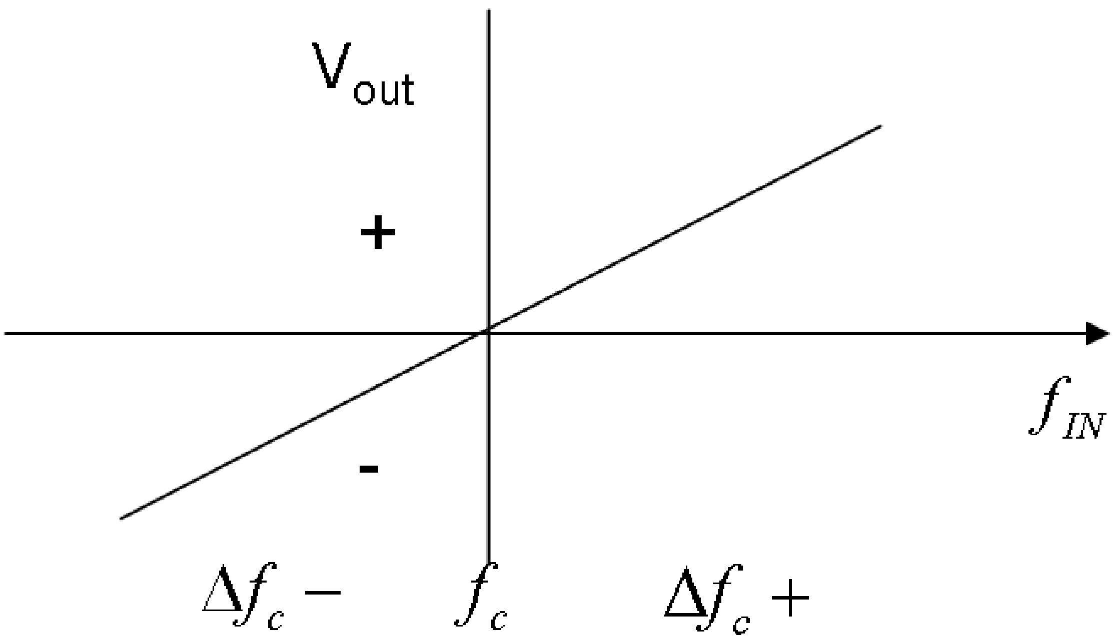
i.e. **Gradient = Voltage Conversion Factor**, *K* volts per Hz.

Considering $y = mx + c$ *etc*. then we may say $V_{OUT} = V_0 + Kf_{IN}$ from the frequency modulator, and since $V_0 = V_{OUT}$ when $f_{IN} = f_c$ then we may write

$$V_{OUT} = V_0 + K\alpha V_{IN}$$

where V_0 represents a DC offset in V_{OUT}. This DC offset may be removed by level shifting or AC coupling, or the F/V may be designed with the characteristic shown next

FM Demodulation –General Principles.



The important point is that $V_{OUT} = K\alpha V_{IN}$. If $V_{IN} = m(t)$ then the output contains the message signal $m(t)$, and the FM signal has been demodulated.

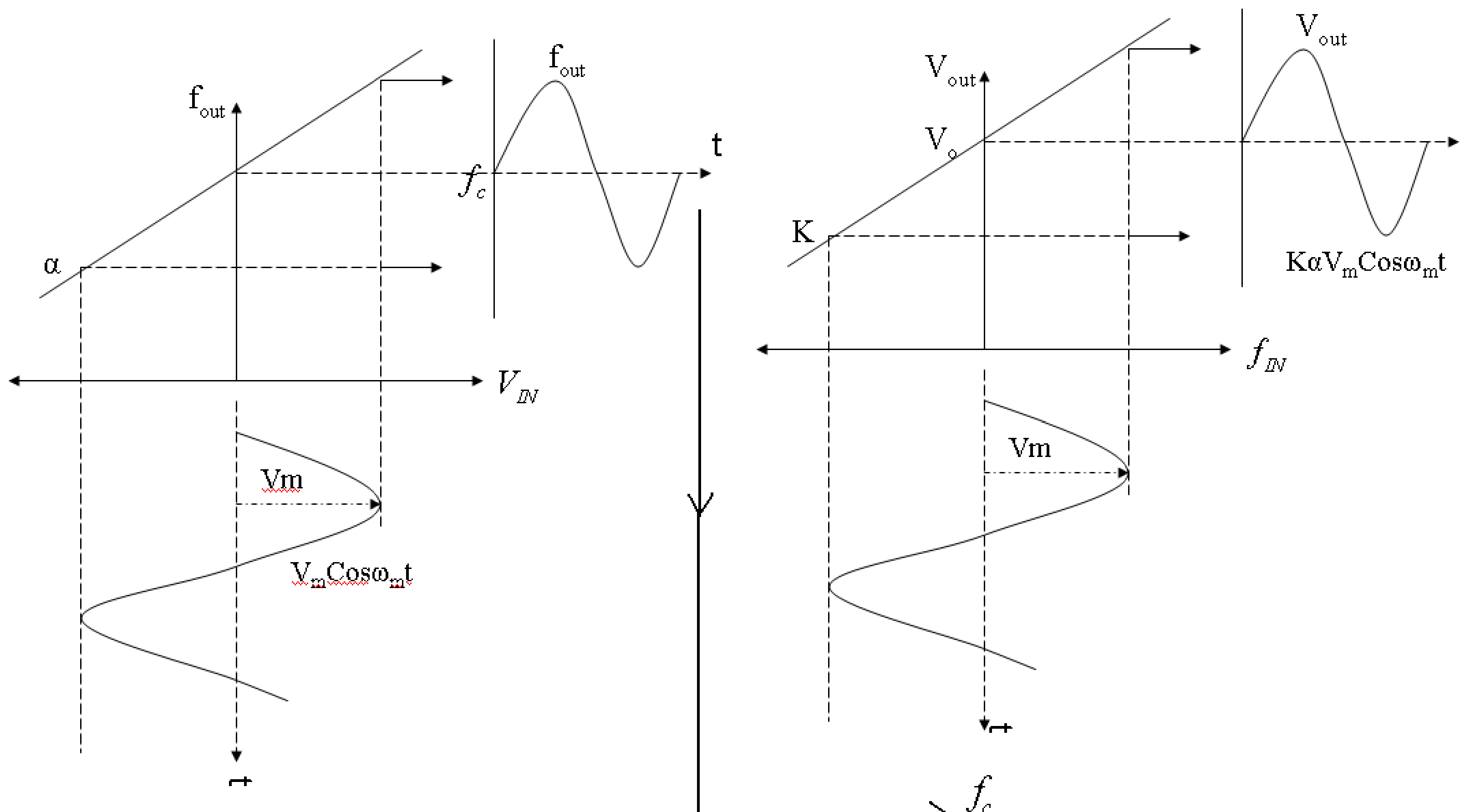
FM Demodulation –General Principles.

Often, but not always, a system designed so that $K = \frac{1}{\alpha}$, so that $K\alpha$ = 1 and $V_{OUT} = m(t)$. A complete system is illustrated.

Gradient = α Hz/Volt
α = Frequency conversion factor
$f_{OUT} = f_c + \alpha V_{IN} = f_{IN}$
$f_{OUT} = f_c + \alpha m(t) = f_{IN}$

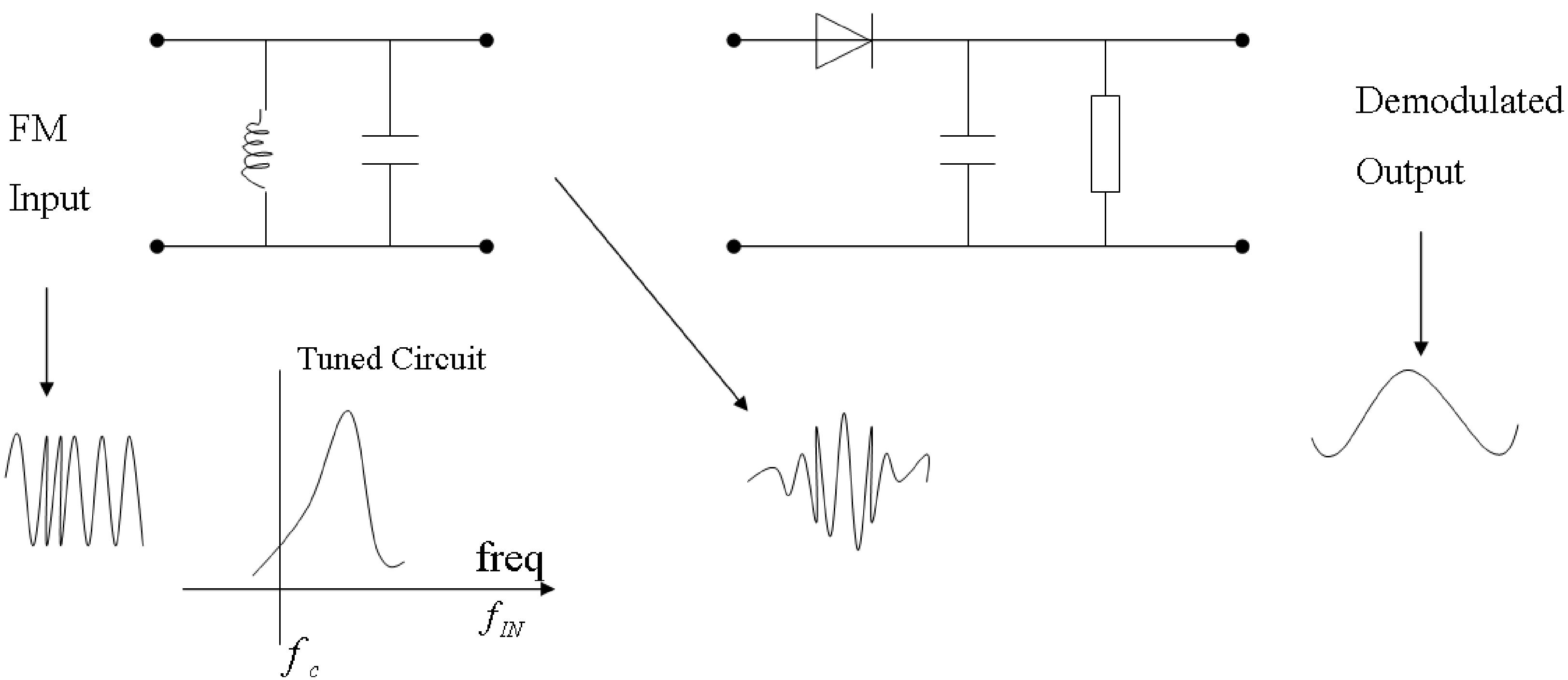
Gradient = K Hz/Volt
K = Voltage conversion factor
$V_{OUT} = V_0 + K\alpha V_{IN}$
$V_{OUT} = V_0 + K\alpha m(t)$

FM Demodulation –General Principles.



Methods

Tuned Circuit – One method (used in the early days of FM) is to use the slope of a tuned circuit in conjunction with an envelope detector.



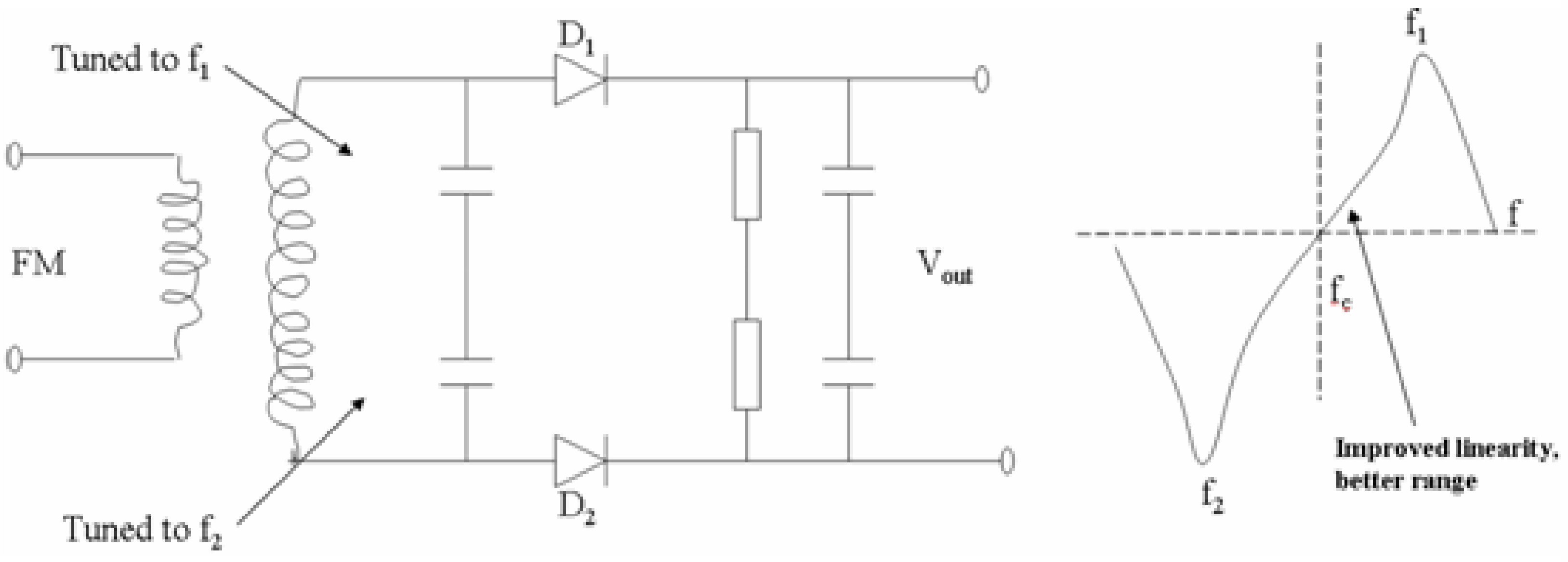
Methods

- The tuned circuit is tuned so the f_c, the nominal input frequency, is on the slope, not at the centre of the tuned circuits. As the FM signal deviates about f_c on the tuned circuit slope, the amplitude of the output varies in proportion to the deviation from f_c. Thus the FM signal is effectively converted to AM. This is then envelope detected by the diode *etc* to recover the message signal.

- Note: In the early days, most radio links were AM (DSBAM). When FM came along, with its advantages, the links could not be changed to FM quickly. Hence, NBFM was used (with a spectral bandwidth = $2f_m$, *i.e.* the same as DSBAM). The carrier frequency *fc* was chosen and the IF filters were tuned so that *fc* fell on the slope of the filter response. Most FM links now are wideband with much better demodulators.

- A better method is to use 2 similar circuits, known as a **Foster-Seeley Discriminator**

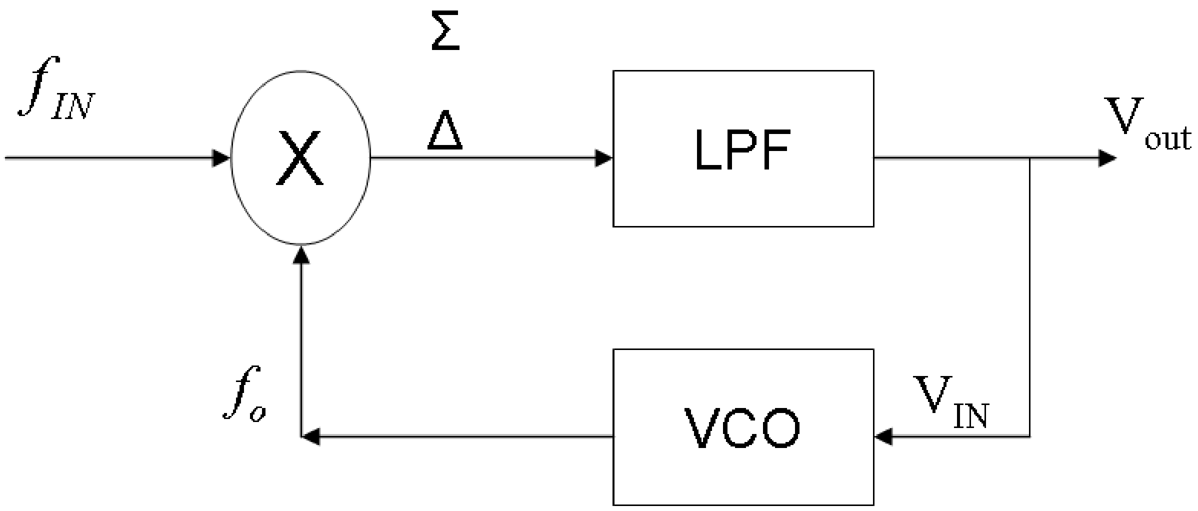
Foster-Seeley Discriminator



This gives the composite characteristics shown. Diode D_2 effectively inverts the f_2 tuned circuit response. This gives the characteristic *'S'* type detector.

Phase Locked Loops PLL

- A PLL is a closed loop system which may be used for FM demodulation. A full analytical description is outside the scope of these notes. A brief description is presented. A block diagram for a PLL is shown below.



- Note the similarity with a synchronous demodulator. The loop comprises a multiplier, a low pass filter and VCO (V/F converter as used in a frequency modulator).

Phase Locked Loops PLL

- The input f_{IN} is applied to the multiplier and multiplied with the VCO frequency output f_O, to produce $\Sigma = (f_{IN} + f_O)$ and $\Delta = (f_{IN} - f_O)$.
- The low pass filter passes only $(f_{IN} - f_O)$ to give *VOUT* which is proportional to $(f_{IN} - f_O)$.
- If $f_{IN} \approx f_O$ but not equal, $V_{OUT} = V_{IN}$, $\alpha f_{IN} - f_O$ is a low frequency (beat frequency) signal to the VCO.
- This signal, V_{IN}, causes the VCO output frequency f_O to vary and move towards f_{IN}.
- When $f_{IN} = f_O$, V_{IN} $(f_{IN} - f_O)$ is approximately constant (DC) and f_O is held constant, *i.e* locked to f_{IN}.
- As f_{IN} changes, due to deviation in FM, f_O tracks or follows f_{IN}. $V_{OUT} = V_{IN}$ changes to drive f_O to track f_{IN}.
- V_{OUT} is therefore proportional to the deviation and contains the message signal $m(t)$.