Angle Modulation – Frequency Modulation

Consider again the general carrier $$v_c(t) = V_c \cos(\omega_c t + \varphi_c)$$

$(\omega_c t + \varphi_c)$ represents the angle of the carrier.

There are two ways of varying the angle of the carrier.

a) By varying the frequency, ω_c – **Frequency Modulation.**
b) By varying the phase, ϕ_c – **Phase Modulation**

Frequency Modulation

In FM, the message signal *m*(*t*) controls the frequency f_c of the carrier. Consider the carrier

$$v_c(t) = V_c \cos(\omega_c t)$$

then for FM we may write:

FM signal $v_s(t) = V_c \cos(2\pi(f_c + \text{frequency deviation})t)$,where the frequency deviation will depend on *m*(*t*).

Given that the carrier frequency will change we may write for an instantaneous carrier signal

$$V_c \cos(\omega_i t) = V_c \cos(2\pi f_i t) = V_c \cos(\varphi_i)$$

where ϕ_i is the instantaneous angle = $\omega_i t = 2\pi f_i t$ and f_i is the instantaneous frequency.

Frequency Modulation

Since $\varphi_i = 2\pi f_i t$ then $\dfrac{d\varphi_i}{dt} = 2\pi f_i$ or $f_i = \dfrac{1}{2\pi}\dfrac{d\varphi_i}{dt}$

i.e. frequency is proportional to the rate of change of angle.

If f_c is the unmodulated carrier and f_m is the modulating frequency, then we may deduce that

$$f_i = f_c + \Delta f_c \cos(\omega_m t) = \frac{1}{2\pi}\frac{d\varphi_i}{dt}$$

Δf_c is the peak deviation of the carrier.

Hence, we have $\dfrac{1}{2\pi}\dfrac{d\varphi_i}{dt} = f_c + \Delta f_c \cos(\omega_m t)$,*i.e.* $\dfrac{d\varphi_i}{dt} = 2\pi f_c + 2\pi \Delta f_c \cos(\omega_m t)$

Frequency Modulation

After integration *i.e.* $\int \left(\omega_c + 2\pi \Delta f_c \cos(\omega_m t)\right) dt$

$$\varphi_i = \omega_c t + \frac{2\pi \Delta f_c \sin(\omega_m t)}{\omega_m}$$

$$\varphi_i = \omega_c t + \frac{\Delta f_c}{f_m} \sin(\omega_m t)$$

Hence for the FM signal, $v_s(t) = V_c \cos(\varphi_i)$

$$v_s(t) = V_c \cos\left(\omega_c t + \frac{\Delta f_c}{f_m} \sin(\omega_m t)\right)$$

Frequency Modulation

The ratio $\frac{\Delta f_c}{f_m}$ is called the **Modulation Index** denoted by β *i.e.*

$$\beta = \frac{\text{Peak frequency deviation}}{\text{modulating frequency}}$$

Note – FM, as implicit in the above equation for $v_s(t)$, is a non-linear process – *i.e.* the principle of superposition does not apply. The FM signal for a message $m(t)$ as a band of signals is very complex. Hence, $m(t)$ is usually considered as a 'single tone modulating signal' of the form

$$m(t) = V_m \cos(\omega_m t)$$

Frequency Modulation

The equation $v_s(t) = V_c \cos\left(\omega_c t + \frac{\Delta f_c}{f_m}\sin(\omega_m t)\right)$ may be expressed as Bessel series (Bessel functions)

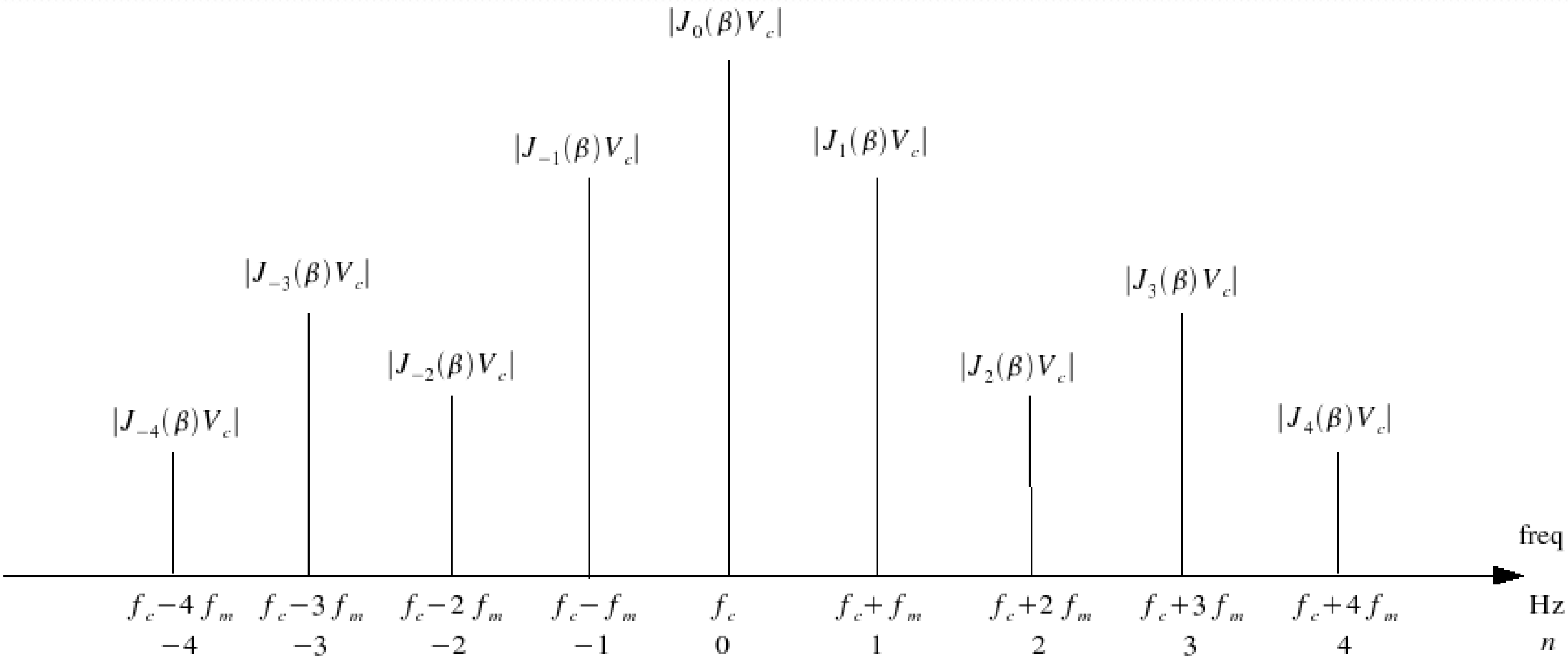
$$v_s(t) = V_c \sum_{n=-\infty}^{\infty} J_n(\beta)\cos(\omega_c + n\omega_m)t$$

where $J_n(\beta)$ are Bessel functions of the first kind. Expanding the equation for a few terms we have:

$$v_s(t) = \underbrace{V_c J_0(\beta)}_{\text{Amp}} \cos\underbrace{(\omega_c)}_{f_c}t + \underbrace{V_c J_1(\beta)}_{\text{Amp}} \cos\underbrace{(\omega_c + \omega_m)t}_{f_c + f_m} + \underbrace{V_c J_{-1}(\beta)}_{\text{Amp}} \cos\underbrace{(\omega_c - \omega_m)t}_{f_c - f_m}$$

$$+ \underbrace{V_c J_2(\beta)}_{\text{Amp}} \cos\underbrace{(\omega_c + 2\omega_m)t}_{f_c + 2f_m} + \underbrace{V_c J_{-2}(\beta)}_{\text{Amp}} \cos\underbrace{(\omega_c - 2\omega_m)t}_{f_c - 2f_m} + \cdots$$

FM Signal Spectrum.



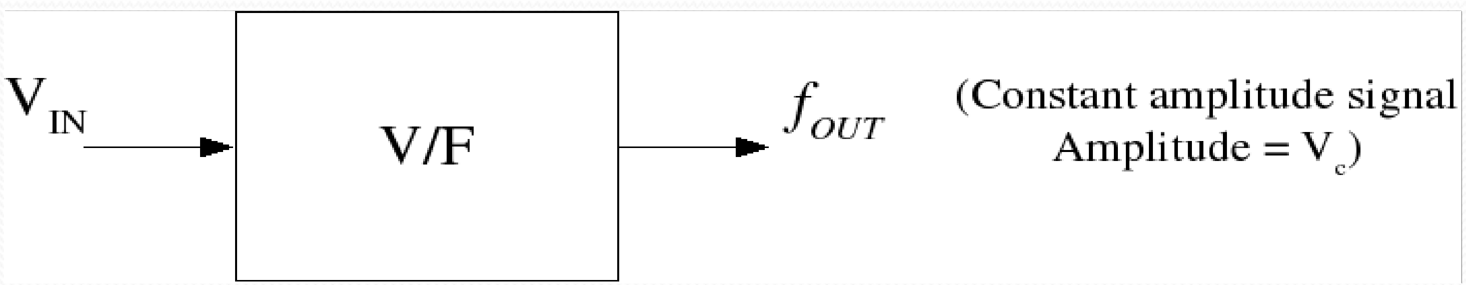
The amplitudes drawn are completely arbitrary, since we have not found any value for $J_n(\beta)$ – this sketch is only to illustrate the spectrum.

Generation of FM signals – Frequency Modulation.

An FM demodulator is:

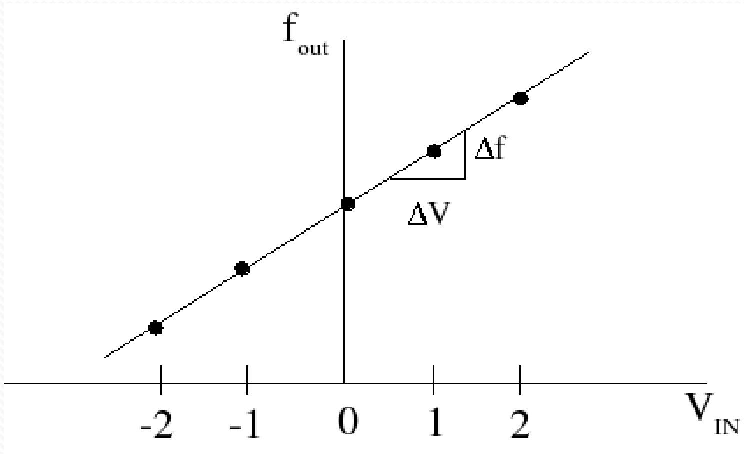
- a voltage-to-frequency converter V/F
- a voltage controlled oscillator VCO

In these devices (V/F or VCO), the output frequency is dependent on the input voltage amplitude.



V/F Characteristics.

Apply V_{IN} , *e.g.* 0 Volts, +1 Volts, +2 Volts, -1 Volts, -2 Volts, ... and measure the frequency output for each V_{IN} . The ideal V/F characteristic is a straight line as shown below.

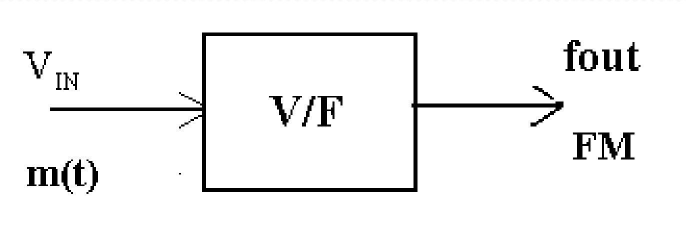


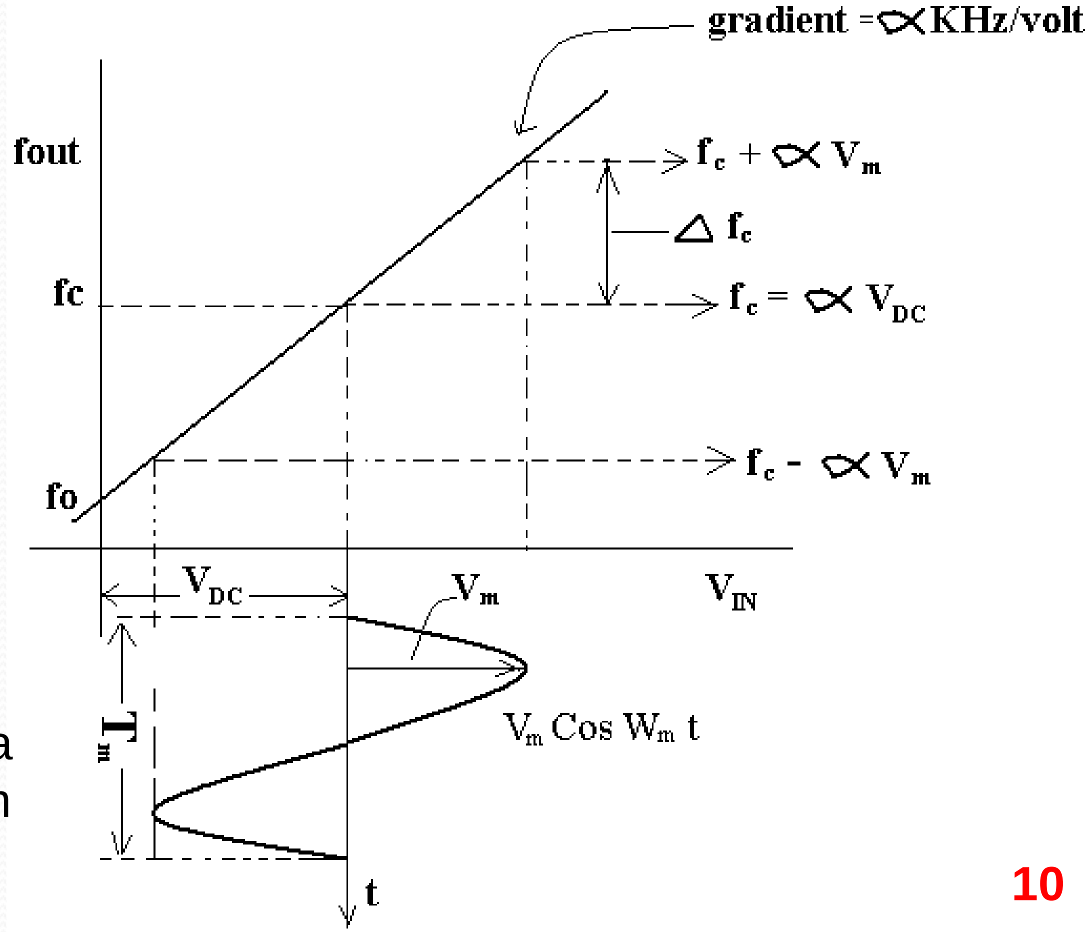
f_c, the frequency output when the input is zero is called the undeviated or nominal carrier frequency.

The gradient of the characteristic $\frac{\Delta f}{\Delta V}$ is called the **Frequency Conversion Factor**, denoted by α per Volt.

V/F Characteristics.

Consider now, an analogue message input, $m(t) = V_m \cos(\omega_m t)$





As the input *m*(*t*) varies from

$$+V_m \rightarrow 0 \rightarrow -V_m$$

the output frequency will vary from a maximum, through f_c, to a minimum frequency.

V/F Characteristics.

For a straight line, $y = c + mx$, where c = value of y when $x = 0$, m = gradient, hence we may say

$$f_{\mathrm{OUT}} = f_c + \alpha V_{\mathrm{IN}}$$

and when $V_{IN} = m(t)$ $f_{\mathrm{OUT}} = f_c + \alpha m(t)$ *,i.e.* the deviation depends on $m(t)$.

Considering that maximum and minimum input amplitudes are $+V_m$ and $-V_m$ respectively, then

$$f_{\max} = f_c + \alpha V_m$$

$$f_{\min} = f_c - \alpha V_m$$

on the diagram on the previous slide.

The peak-to-peak deviation is $f_{max} - f_{min}$, but more importantly for FM the peak deviation Δf_c is

Peak Deviation, $\Delta f_c = \alpha V_m$ Hence, **Modulation Index**, $\beta = \frac{\Delta f_c}{f_m} = \frac{\alpha V_m}{f_m}$

Summary of the important points of FM

- In FM, the message signal $m(t)$ is assumed to be a single tone frequency,

$$m(t) = V_m \cos(\omega_m t)$$

- The FM signal $v_s(t)$ from which the spectrum may be obtained as

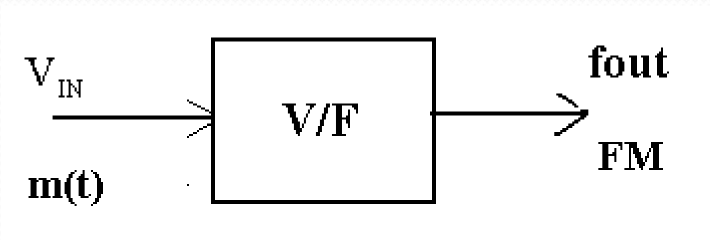
$$v_s(t) = V_c \sum_{n=-\infty}^{\infty} J_n(\beta) \cos(\omega_c + n\omega_m)t$$

where $J_n(\beta)$ are Bessel coefficients and **Modulation Index**, $\beta = \frac{\Delta f_c}{f_m} = \frac{\alpha V_m}{f_m}$

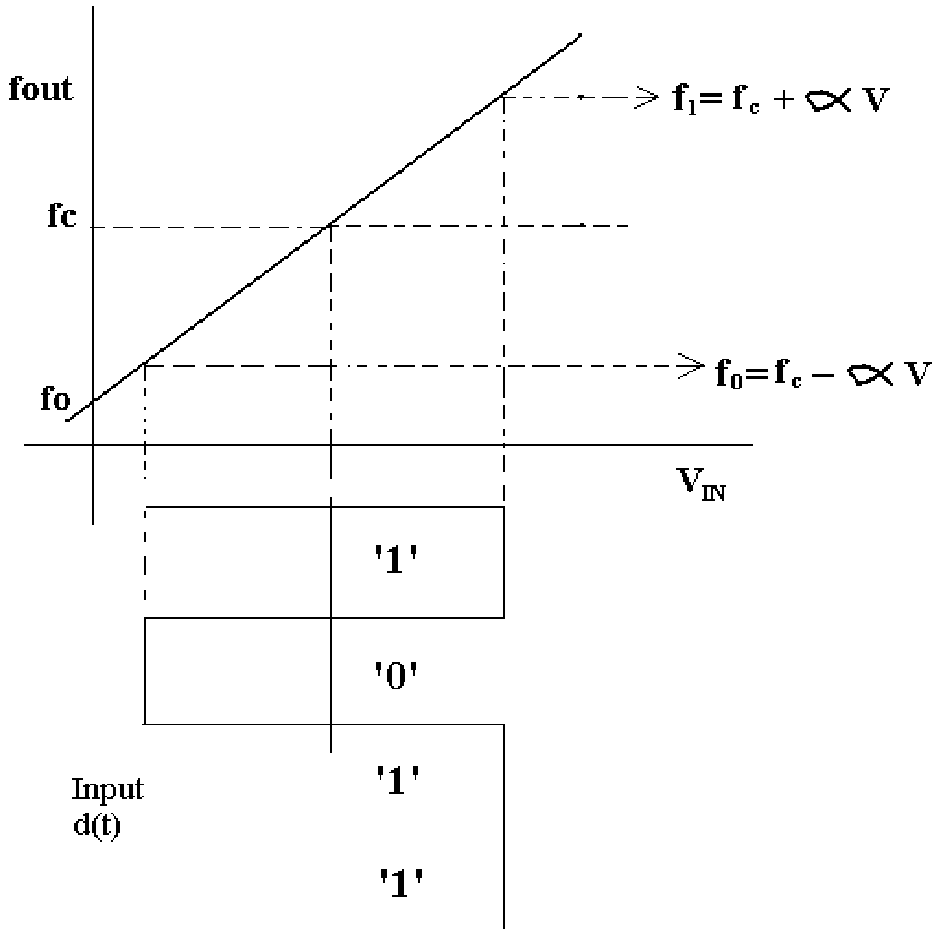
- α Hz per Volt is the V/F modulator, gradient or **Frequency Conversion Factor**, α per Volt
- α is a measure of the change in output frequency for a change in input amplitude.
- **Peak Deviation** (of the carrier frequency from f_c) $\Delta f_c = \alpha V_m$

FM Signal Waveforms.

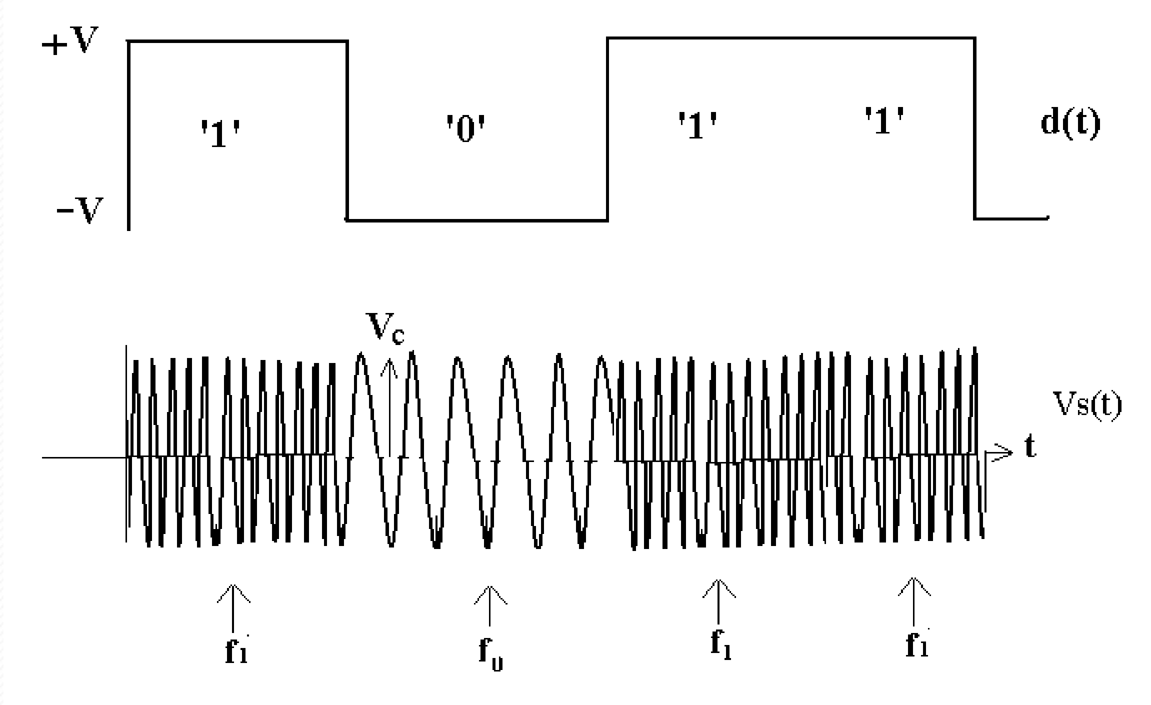
The diagrams below illustrate signal waveforms for various inputs



At this stage, an input digital data sequence, *d*(*t*), is introduced – the output in this case will be FSK, (Frequency Shift Keying).



FM Signal Waveforms.

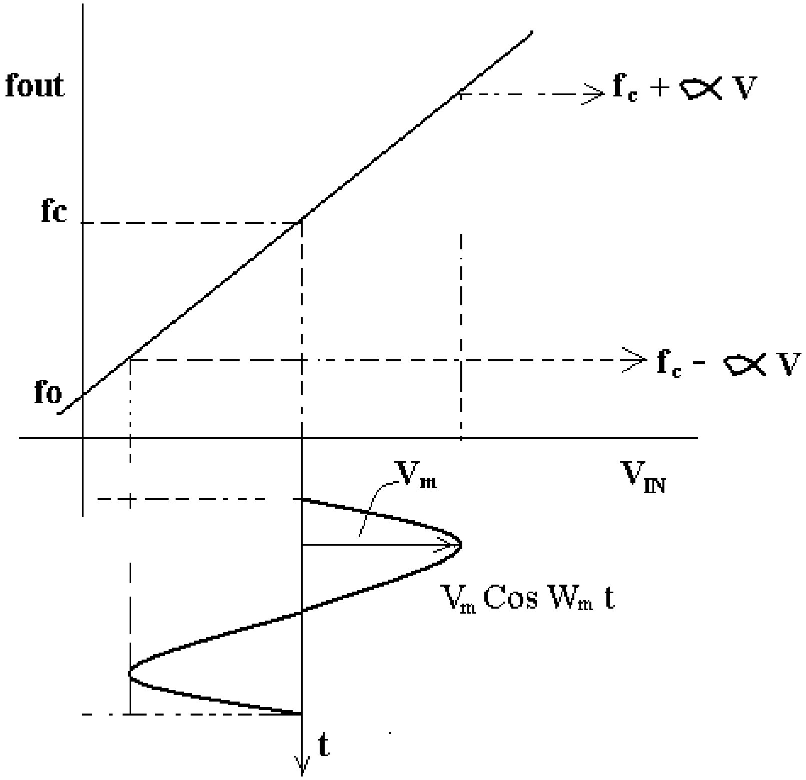


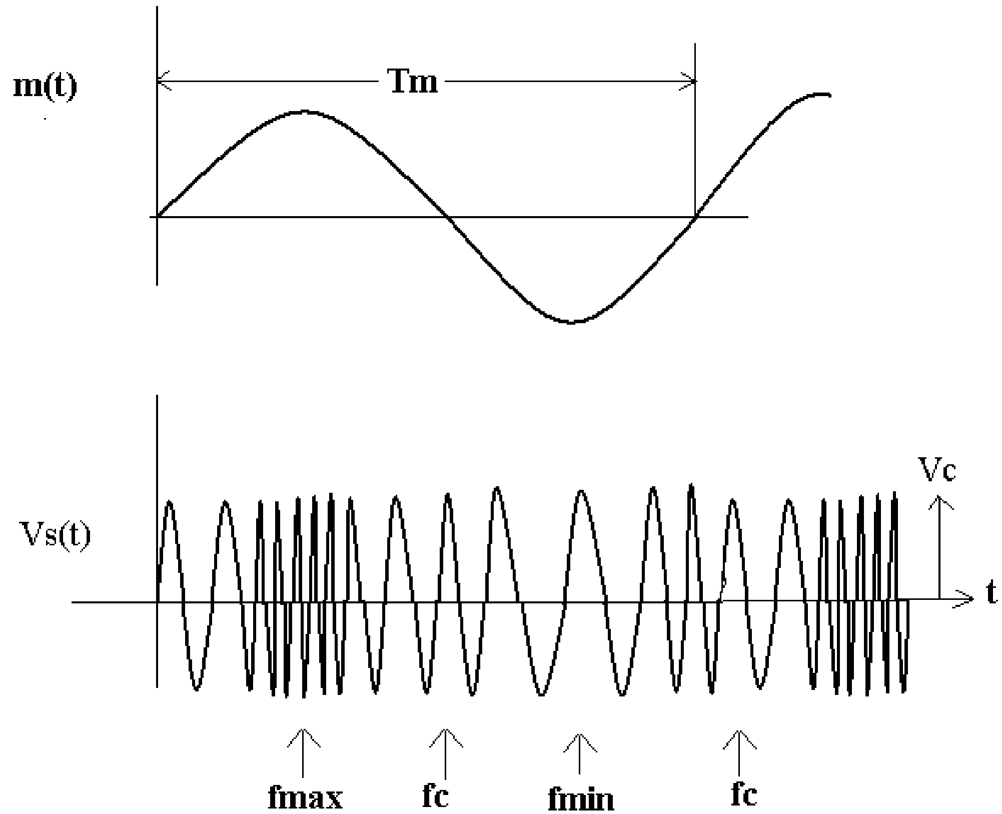
Assuming $d(t) = +V$ for 1's $\quad f_{OUT} = f_1 = f_c + \alpha V$ for 1's

$= -V$ for 0's $\quad f_{OUT} = f_0 = f_c - \alpha V$ for 0's

the output 'switches' between f_1 and f_0.

FM Signal Waveforms.

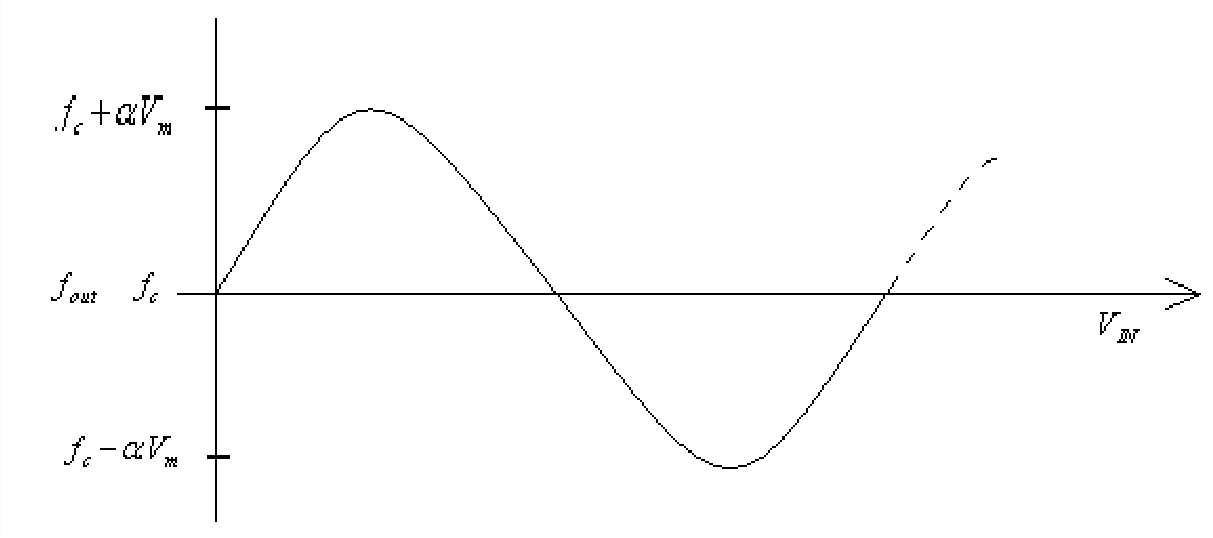




The output frequency varies ‘gradually’ from f_c to ($f_c + \alpha V_m$), through f_c to ($f_c - \alpha V_m$) *etc.*

FM Signal Waveforms.

If we plot f_{OUT} as a function of V_{IN}:

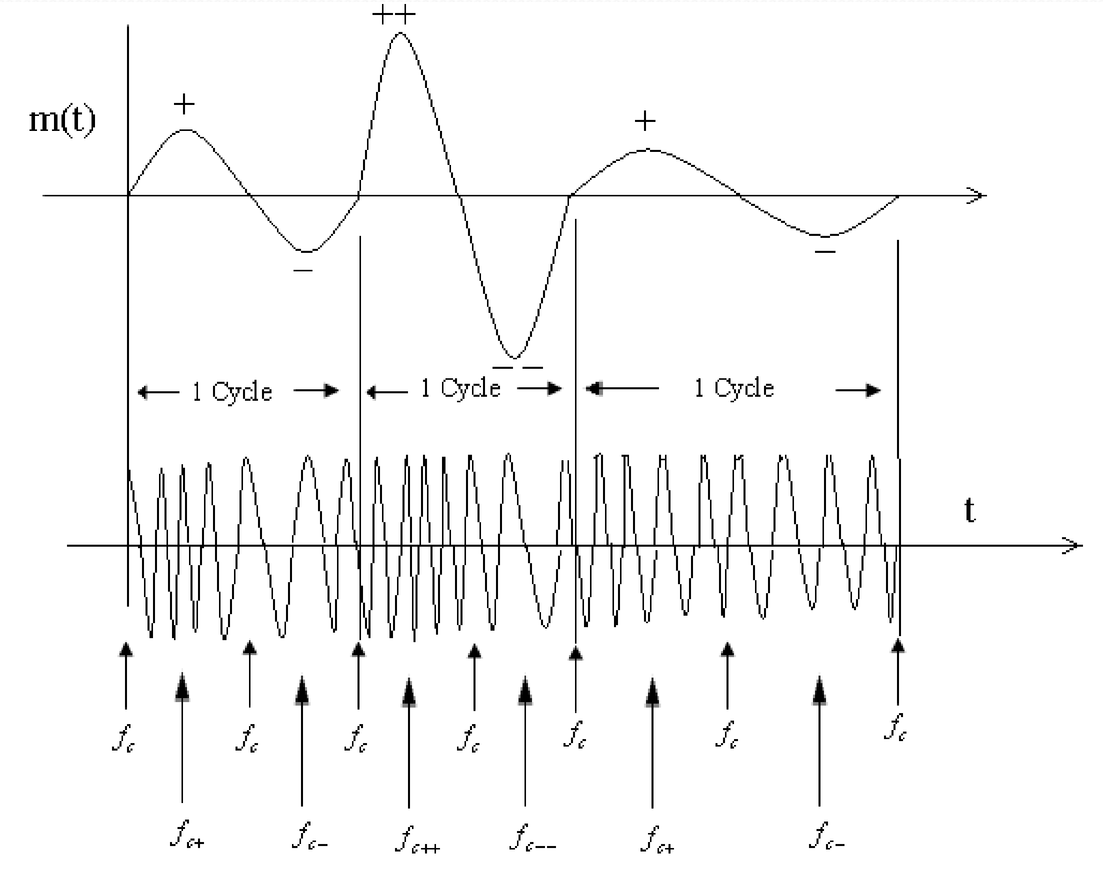


In general, *m*(*t*) will be a ‘band of signals’, *i.e.* it will contain amplitude and frequency variations. Both amplitude and frequency change in *m*(*t*) at the input are translated to (just) frequency changes in the FM output signal, *i.e.* the amplitude of the output FM signal is constant.

Amplitude changes at the input are translated to deviation from the carrier at the output. The larger the amplitude, the greater the deviation.

FM Signal Waveforms.

Frequency changes at the input are translated to rate of change of frequency at the output. An attempt to illustrate this is shown below:



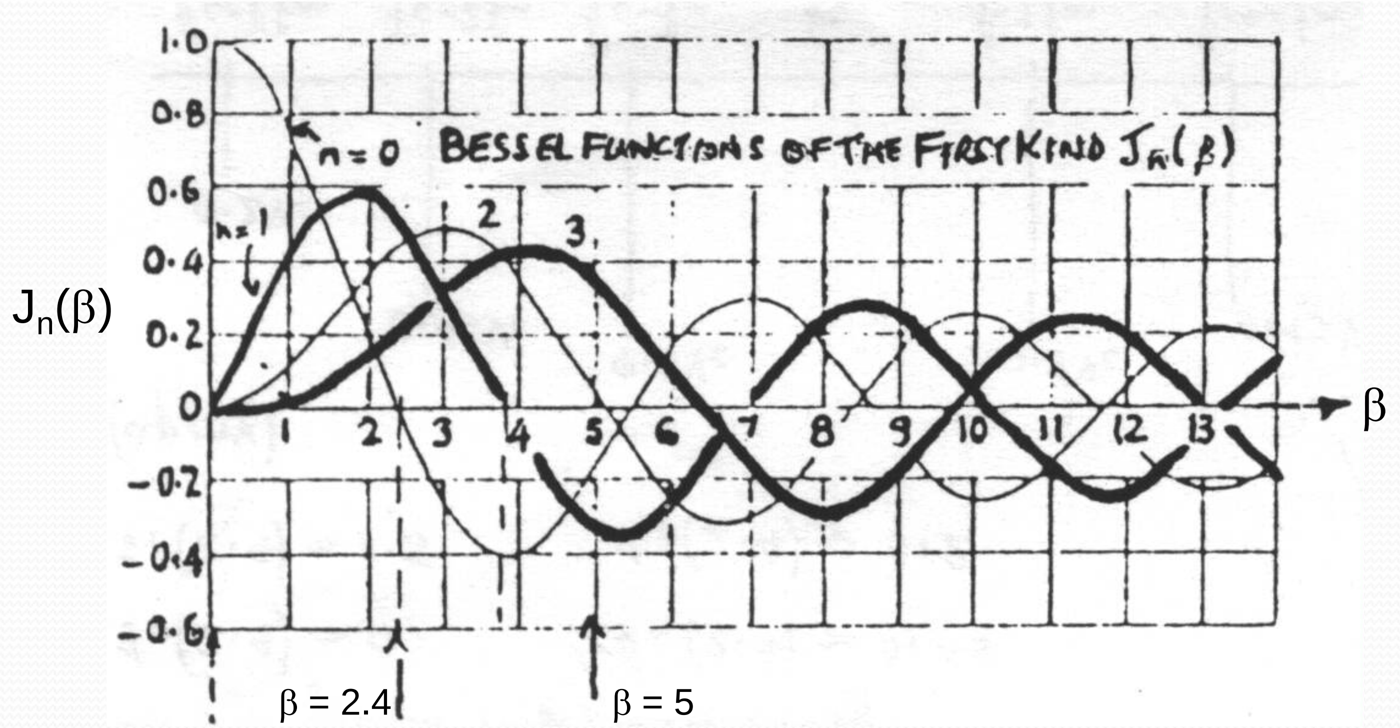
FM Spectrum – Bessel Coefficients.

The FM signal spectrum may be determined from

$$v_s(t) = V_c \sum_{n=-\infty}^{\infty} J_n(\beta)\cos(\omega_c + n\omega_m)t$$

The values for the Bessel coefficients, $J_n(\beta)$ may be found from graphs or, preferably, tables of 'Bessel functions of the first kind'.

FM Spectrum – Bessel Coefficients.

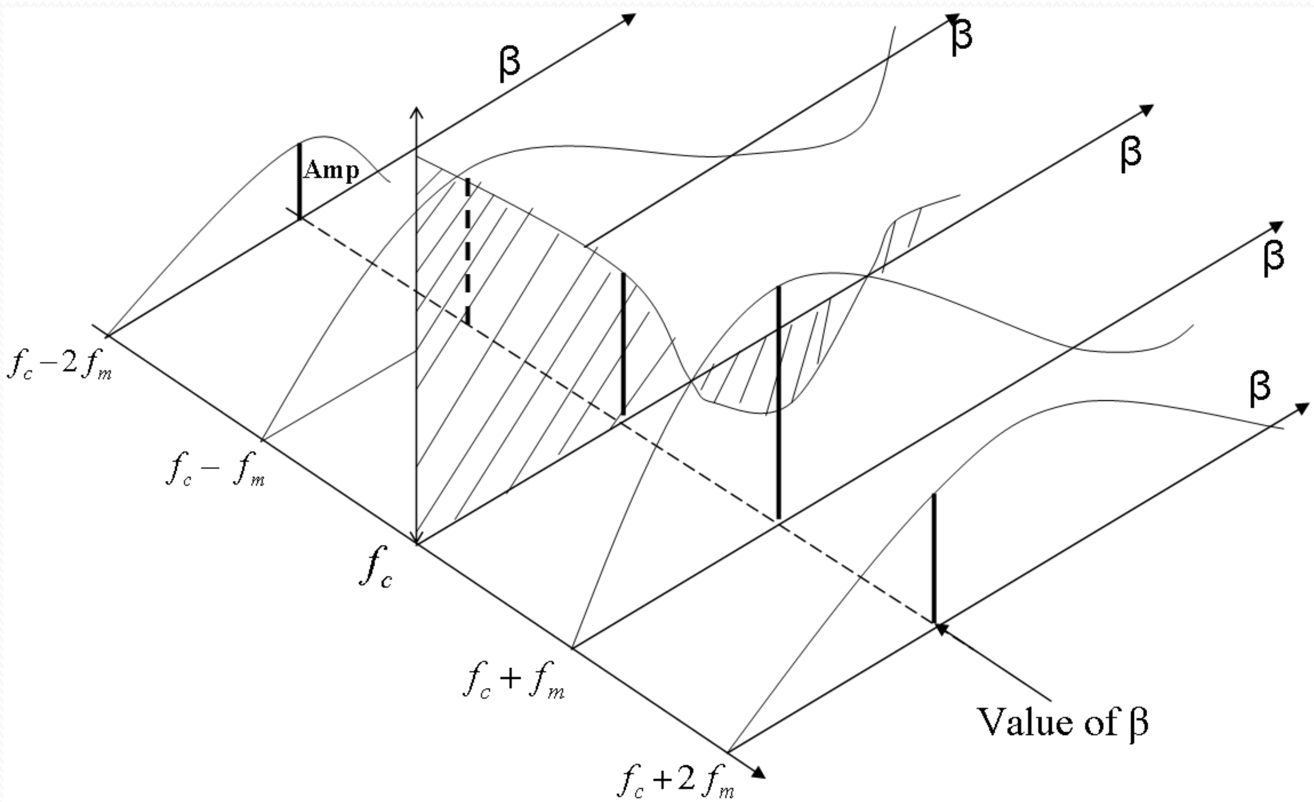


In the series for $v_s(t)$, n = 0 is the carrier component, *i.e.* $V_c J_0(\beta)\cos(\omega_c t)$, hence the n = 0 curve shows how the component at the carrier frequency, f_c, varies in amplitude, with modulation index β.

FM Spectrum – Bessel Coefficients.

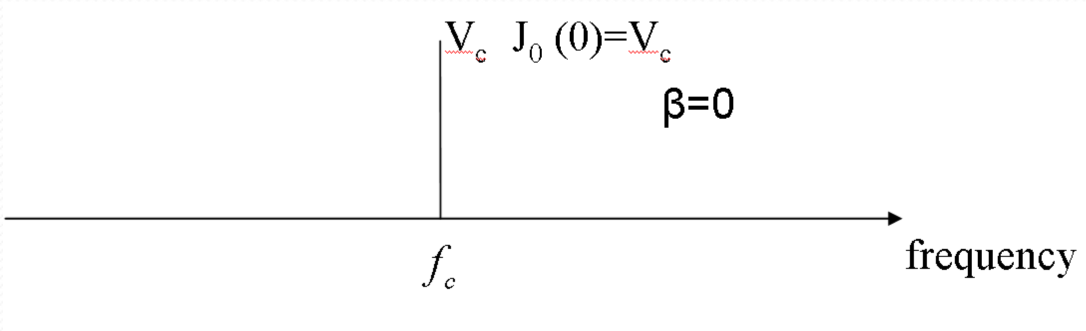
Hence for a given value of modulation index β, the values of $J_n(\beta)$ may be read off the graph and hence the component amplitudes ($V_c J_n(\beta)$) may be determined.

A further way to interpret these curves is to imagine them in 3 dimensions



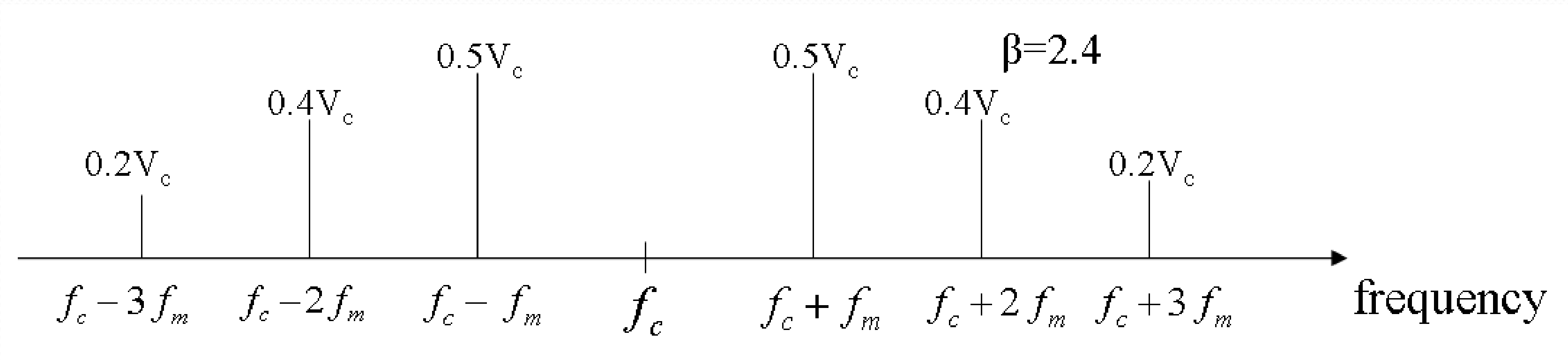
Examples from the graph

β = 0: When β **= 0 the carrier is unmodulated and $J_0(0)$ = 1, all other $J_n(0)$ = 0, *i.e.***



β = 2.4: From the graph (approximately)

$J_0(2.4) = 0$, $J_1(2.4) = 0.5$, $J_2(2.4) = 0.45$ and $J_3(2.4) = 0.2$



Significant Sidebands – Spectrum.

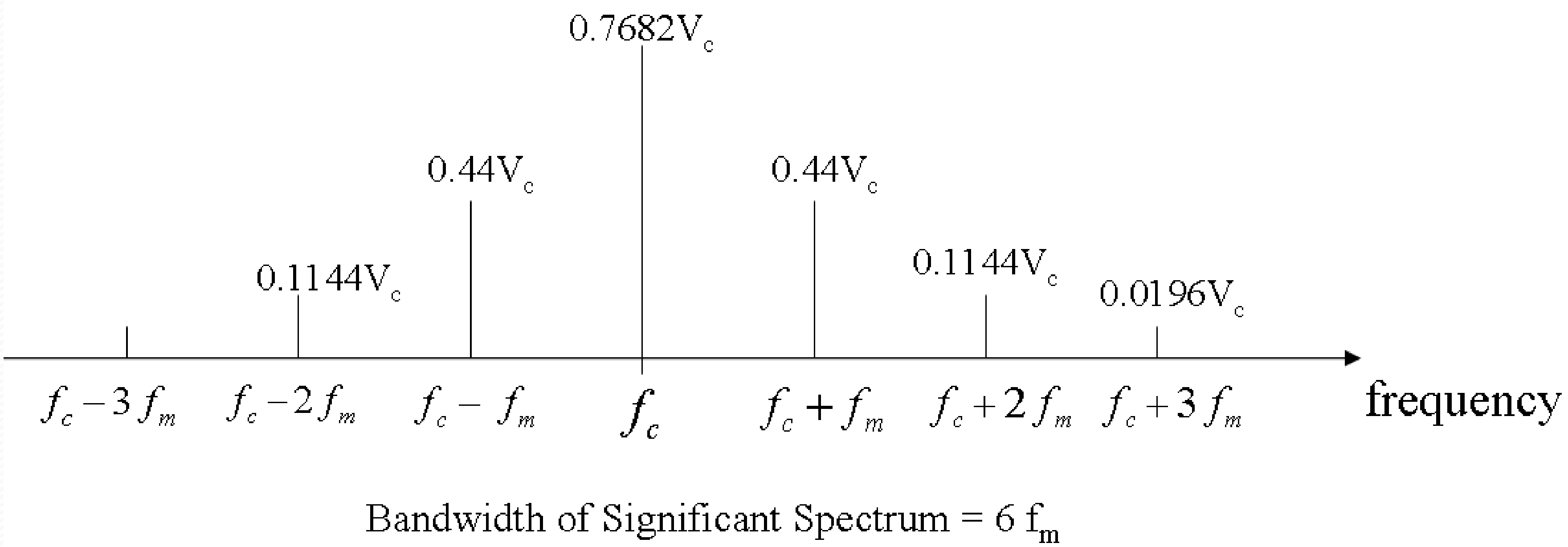
As may be seen from the table of Bessel functions, for values of n above a certain value, the values of $J_n(\beta)$ become progressively smaller. In FM the sidebands are considered to be significant if $J_n(\beta) \geq 0.01$ (1%).

Although the bandwidth of an FM signal is infinite, components with amplitudes $V_c J_n(\beta)$, for which $J_n(\beta) < 0.01$ are deemed to be insignificant and may be ignored.

Example: A message signal with a frequency f_m Hz modulates a carrier f_c to produce FM with a modulation index $\beta = 1$. Sketch the spectrum.

n	$J_n(1)$	Amplitude	Frequency
0	0.7652	$0.7652V_c$	f_c
1	0.4400	$0.44V_c$	f_c+f_m $f_c - f_m$
2	0.1149	$0.1149V_c$	f_c+2f_m $f_c - 2f_m$
3	0.0196	$0.0196V_c$	f_c+3f_m $f_c -3 f_m$
4	0.0025	*Insignificant*	
5	0.0002	*Insignificant*	

Significant Sidebands – Spectrum.

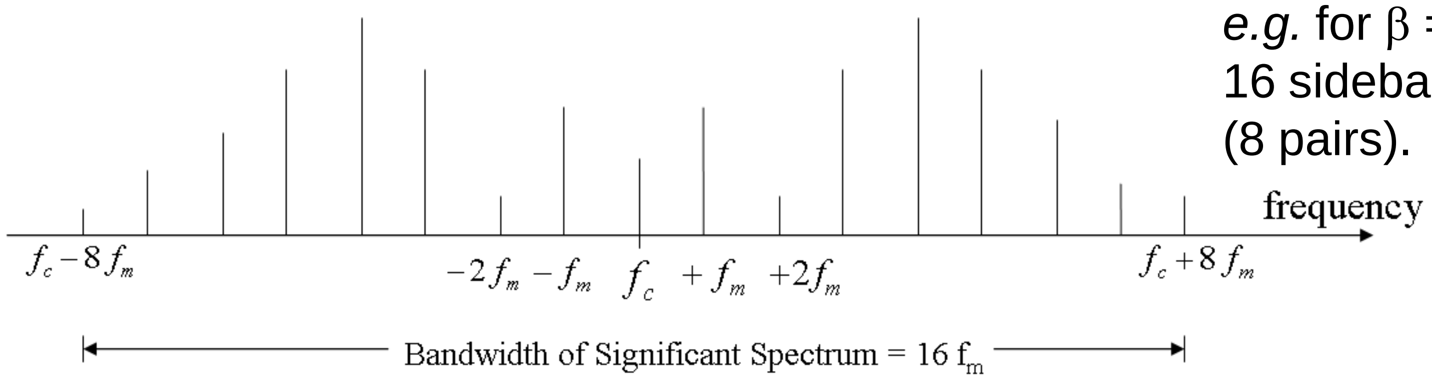


As shown, the bandwidth of the spectrum containing significant components is $6f_m$, for $\beta = 1$.

Significant Sidebands – Spectrum.

The table below shows the number of significant sidebands for various modulation indices (β) and the associated spectral bandwidth.

β	No of sidebands ≥ 1% of unmodulated carrier	Bandwidth
0.1	2	$2f_m$
0.3	4	$4f_m$
0.5	4	$4f_m$
1.0	6	$6f_m$
2.0	8	$8f_m$
5.0	16	$16f_m$
10.0	28	$28f_m$



e.g. for β = 5, 16 sidebands (8 pairs).

Carson's Rule for FM Bandwidth.

An approximation for the bandwidth of an FM signal is given by BW = 2(Maximum frequency deviation + highest modulated frequency)

$$\text{Bandwidth} = 2(\Delta f_c + f_m)$$ **Carson's Rule**

Narrowband and Wideband FM

Narrowband FM NBFM

From the graph/table of Bessel functions it may be seen that for small β, (β ≤ 0.3) there is only the carrier and 2 significant sidebands, *i.e.* BW = 2*fm*.

FM with β ≤ 0.3 is referred to as **narrowband FM** (NBFM) (Note, the bandwidth is the same as DSBAM).

Hence, NBFM have smaller bandwidth & modulation index is also smaller ~ one radian. e.g. entertainment – f.m radio broadcasting.

Wideband FM WBFM

For β > 0.3 there are more than 2 significant sidebands. As β increases the number of sidebands increases. This is referred to as **wideband FM** (WBFM).

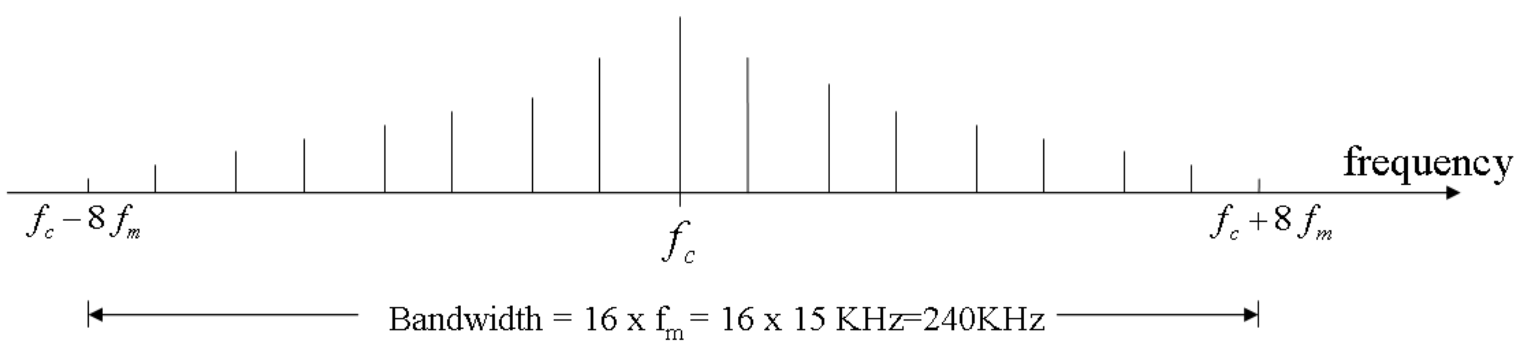
WBFM have larger valves of modulation index & contains carrier with infinite no of side bands. e.g. mobile communication.

VHF/FM

VHF/FM (Very High Frequency band = 30MHz – 300MHz) radio transmissions, in the band 88MHz to 108MHz have the following parameters:

Max frequency input (*e.g.* music)	15kHz	f_m
Deviation	75kHz	$\Delta f_c = \alpha V_m$
Modulation Index β	5	$\beta = \dfrac{\Delta f_c}{f_m}$

For β = 5 there are 16 sidebands and the FM signal bandwidth is 16*fm* = 16 x 15kHz = 240kHz. Applying Carson's Rule BW = 2(75+15) = 180kHz.



Comments FM

- The FM spectrum contains a carrier component and an infinite number of sidebands at frequencies $f_c \pm nf_m$ (n = 0, 1, 2, …)

FM signal, $$v_s(t) = V_c \sum_{n=-\infty}^{\infty} J_n(\beta)\cos(\omega_c + n\omega_m)t$$

- In FM we refer to sideband pairs <u>not</u> upper and lower sidebands. Carrier or other components <u>may not</u> be suppressed in FM.

- The relative amplitudes of components in FM depend on the values $J_n(\beta)$, where

$\beta = \dfrac{\alpha V_m}{f_m}$ thus the component at the carrier frequency depends on $m(t)$, as do all the other components and none may be suppressed.

Comments FM

- Components are significant if $J_n(\beta) \geq 0.01$. For $\beta \ll 1$ ($\beta \approx 0.3$ or less) only $J_0(\beta)$ and $J_1(\beta)$ are significant, *i.e.* only a carrier and 2 sidebands. Bandwidth is $2f_m$, similar to DSBAM in terms of bandwidth - called NBFM.

- Large modulation index $\beta = \frac{\Delta f_c}{f_m}$ means that a large bandwidth is required – called WBFM.

- The FM process is non-linear. The principle of superposition does not apply. When $m(t)$ is a band of signals, *e.g.* speech or music the analysis is very difficult (impossible?). Calculations usually assume a single tone frequency equal to the maximum input frequency. *E.g.* $m(t) \equiv$ band 20Hz $\rightarrow$ 15kHz, fm = 15kHz is used.

Power in FM Signals.

From the equation for FM $v_s(t) = V_c \sum_{n=-\infty}^{\infty} J_n(\beta)\cos(\omega_c + n\omega_m)t$

we see that the peak value of the components is $V_c J_n(\beta)$ for the n^{th} component.

Single normalised average power = $\left(\frac{V_{pk}}{\sqrt{2}}\right)^2 = (V_{RMS})^2$ then the n^{th} component is

$$\left(\frac{V_c J_n(\beta)}{\sqrt{2}}\right)^2 = \frac{(V_c J_n(\beta))^2}{2}$$

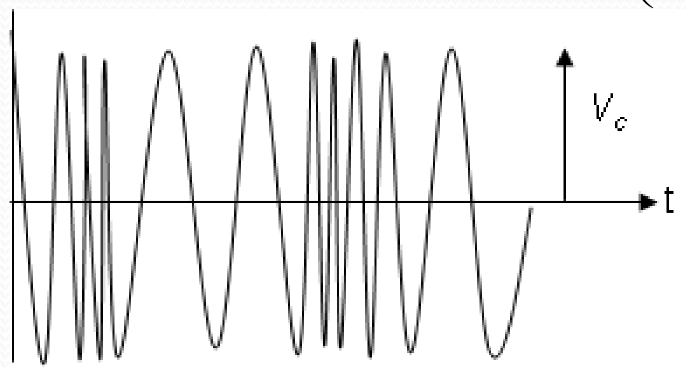
Hence, the total power in the infinite spectrum is

Total power $P_T = \sum_{n=-\infty}^{\infty} \frac{(V_c J_n(\beta))^2}{2}$

Power in FM Signals.

By this method we would need to carry out an infinite number of calculations to find P_T. But, considering the waveform, the peak value is V_{-c}, which is constant.

Since we know that the RMS value of a sine wave is $\left(\frac{V_{pk}}{\sqrt{2}}\right)^2 = \frac{V_c}{\sqrt{2}}$

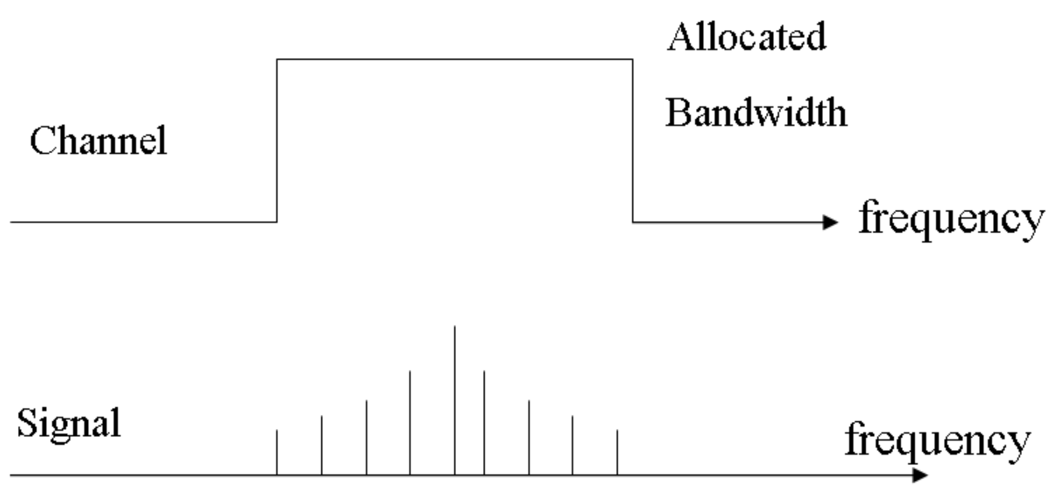


and power = $(V_{RMS})^2$ then we may deduce that $P_T = \left(\frac{V_c}{\sqrt{2}}\right)^2 = \frac{V_c^2}{2} = \sum_{n=-\infty}^{\infty} \frac{(V_c J_n(\beta))^2}{2}$

Hence, if we know V_c for the FM signal, we can find the total power P_T for the infinite spectrum with a simple calculation.

Power in FM Signals.

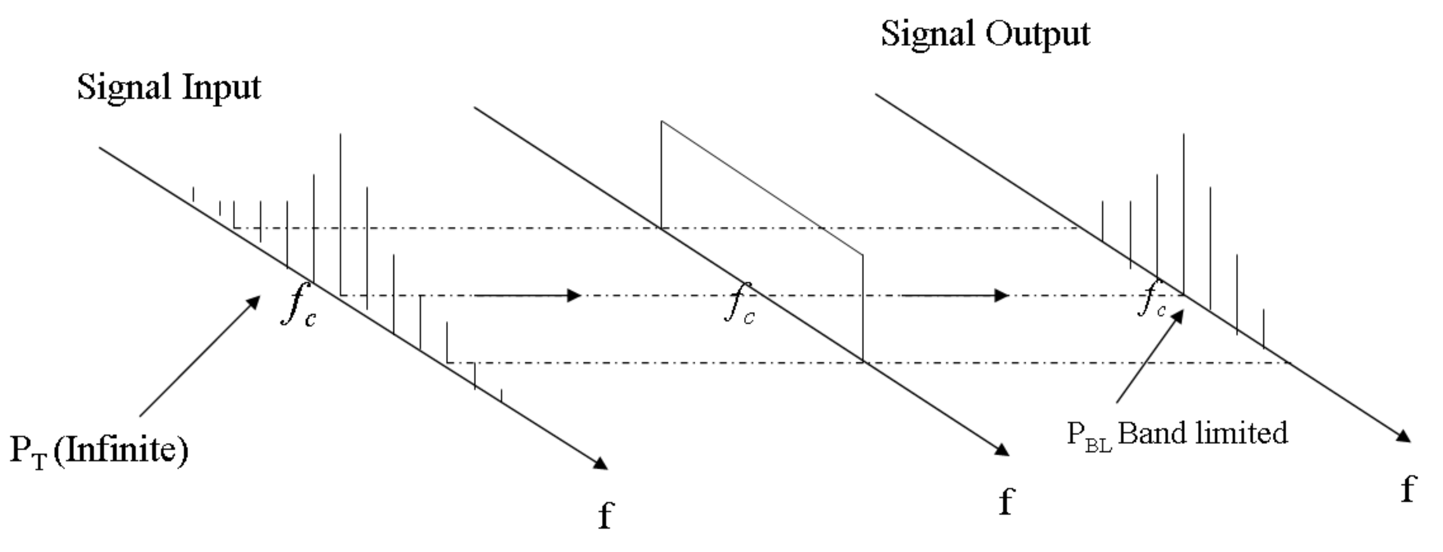
Now consider – if we generate an FM signal, it will contain an infinite number of sidebands. However, if we wish to transfer this signal, *e.g.* over a radio or cable, this implies that we require an infinite bandwidth channel. Even if there was an infinite channel bandwidth it would not all be allocated to one user. Only a limited bandwidth is available for any particular signal. Thus we have to make the signal spectrum fit into the available channel bandwidth. We can think of the signal spectrum as a 'train' and the channel bandwidth as a tunnel – obviously we make the train slightly less wider than the tunnel if we can.



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

Also, it is easily seen that the ratio

$D = \dfrac{\text{Power in Bandlimited spectrum}}{\text{Power in total spectrum}} = \dfrac{P_{BL}}{P_T} = \sum_{n=-a}^{a}(J_n(\beta))^2$ = 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	$J_n(0.2)$	Amp = $V_c J_n(0.2)$	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)

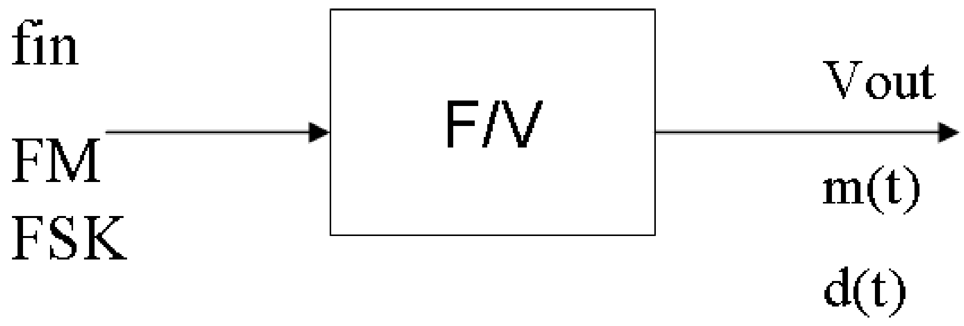
D = $1 - (0.99)^2 - (0.0995)^2 = 0.01$

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.

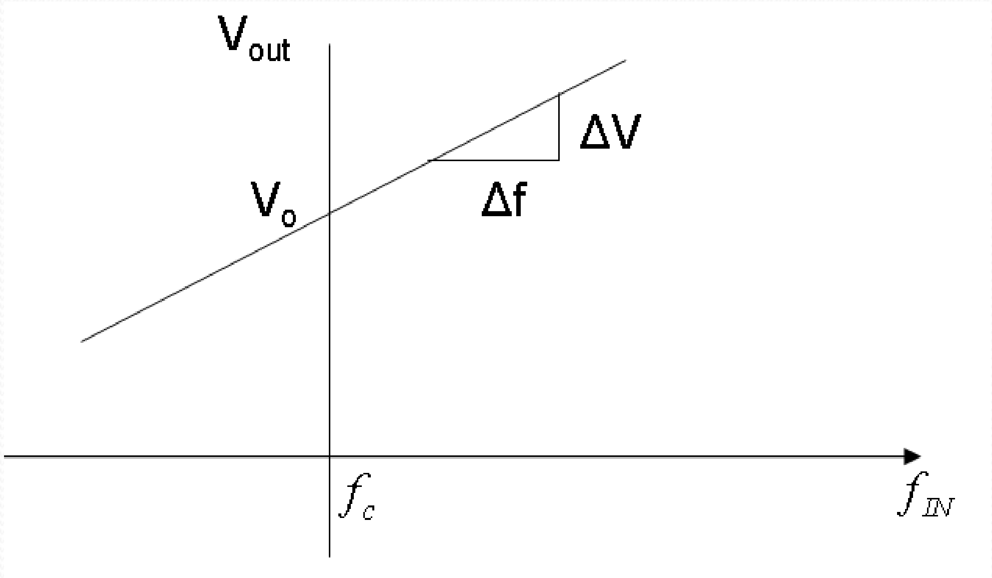
- Before considering some specific types, the general concepts for FM demodulation will be presented. An F/V converter produces an output voltage, V_{OUT} which is proportional to the frequency input, f_{IN}.



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 $\dfrac{\Delta V}{\Delta f}$ is called the voltage conversion factor

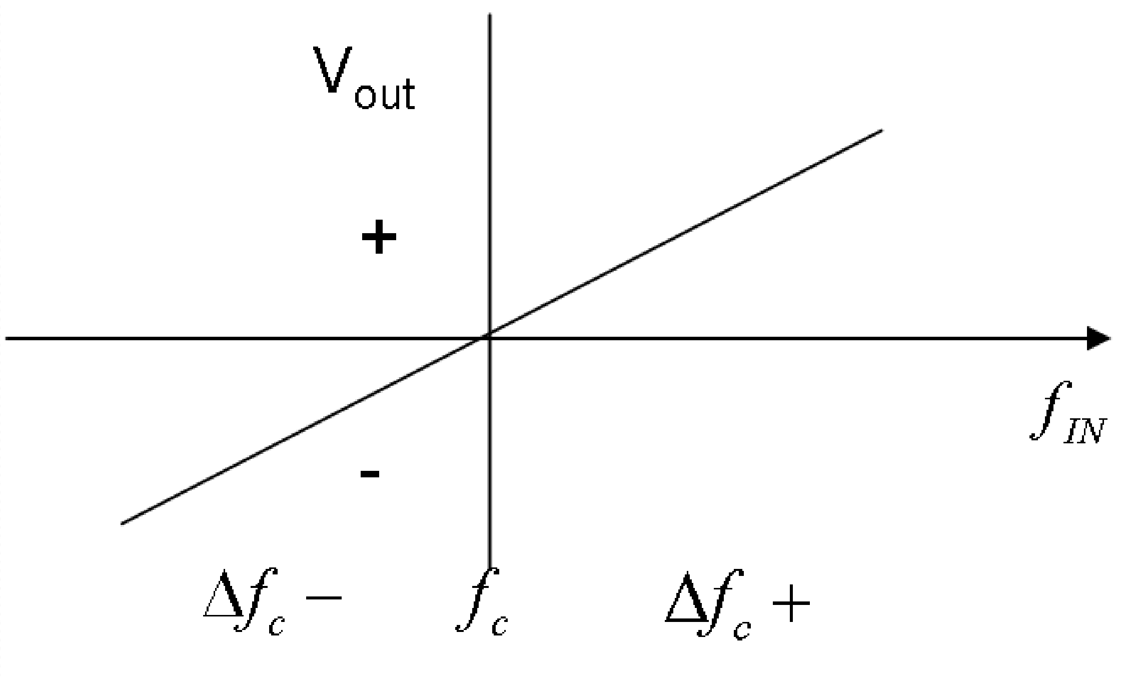
i.e. **Gradient = Voltage Conversion Factor**, *K* volts per Hz.

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

$f_{OUT} = f_c + \alpha V_{IN} = f_{IN}$

$f_{OUT} = f_c + \alpha m(t) = f_{IN}$

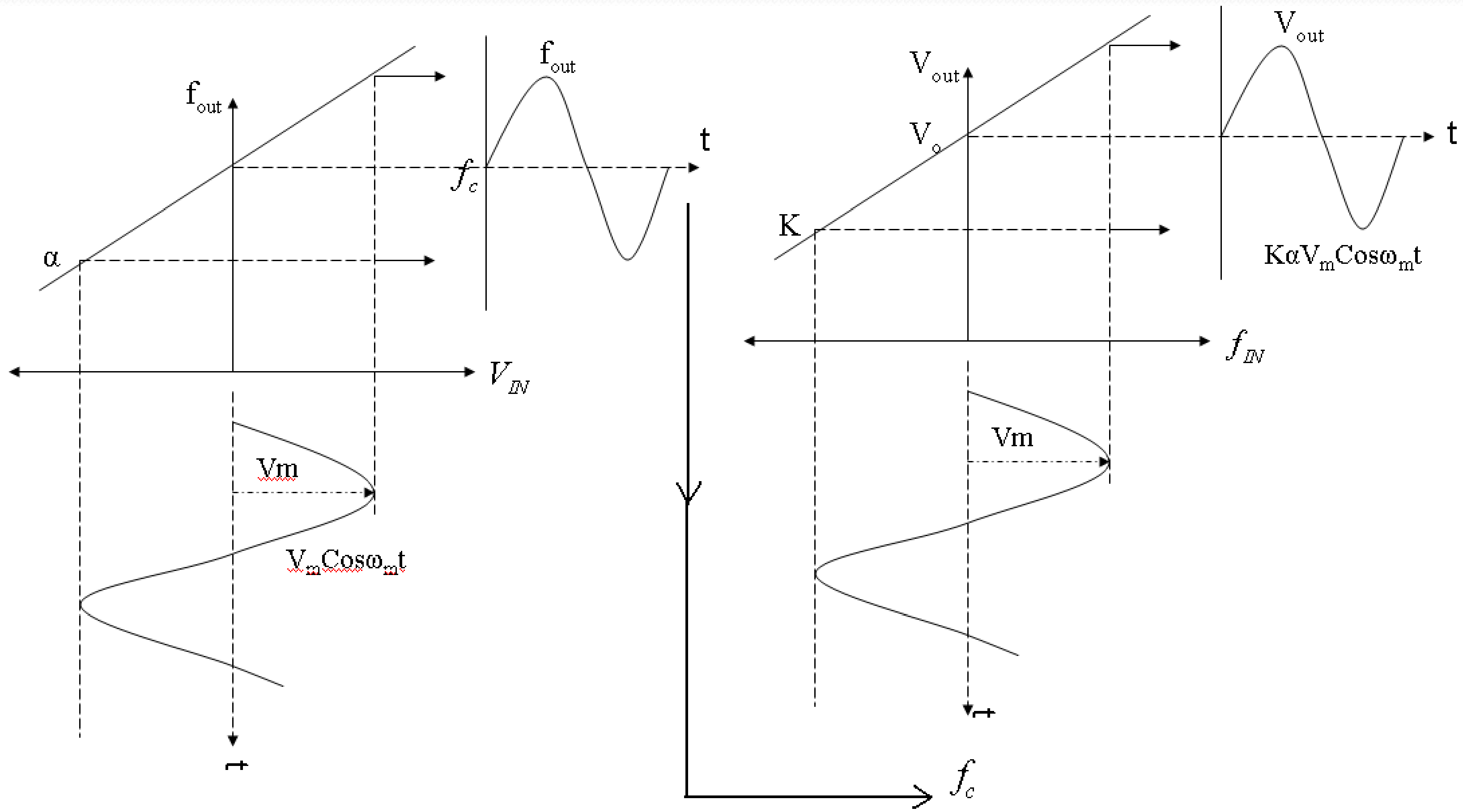
Gradient = K Hz/Volt

K = Voltage conversion factor

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

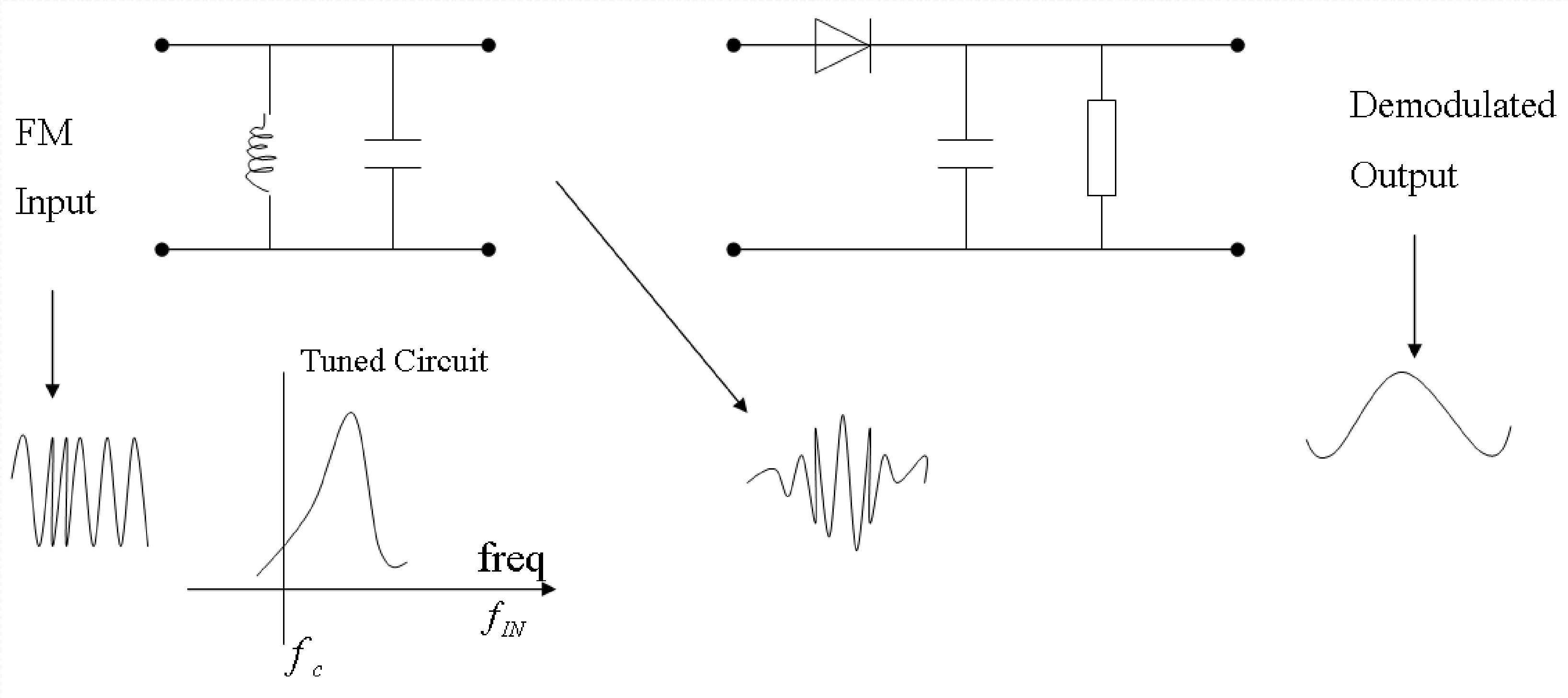
$V_{OUT} = V_0 + K\alpha m(t)$

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.



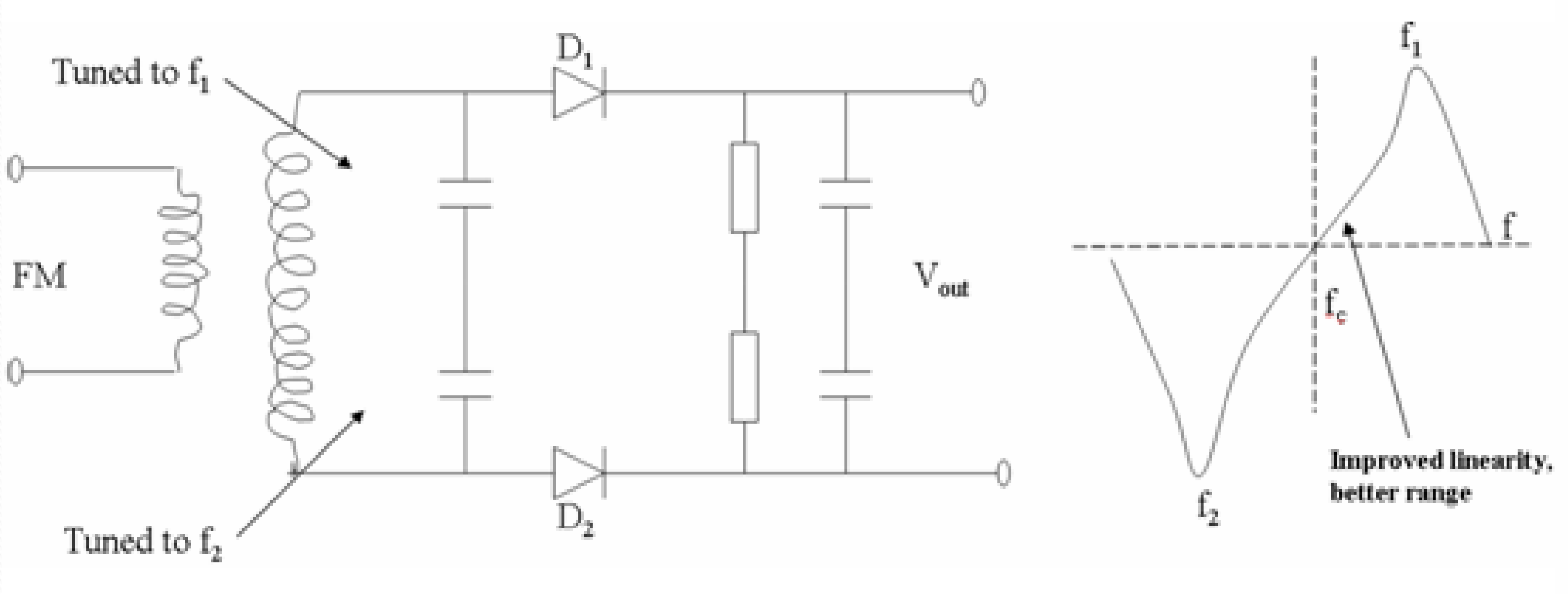
Methods

- 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 = 2*fm*, *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**

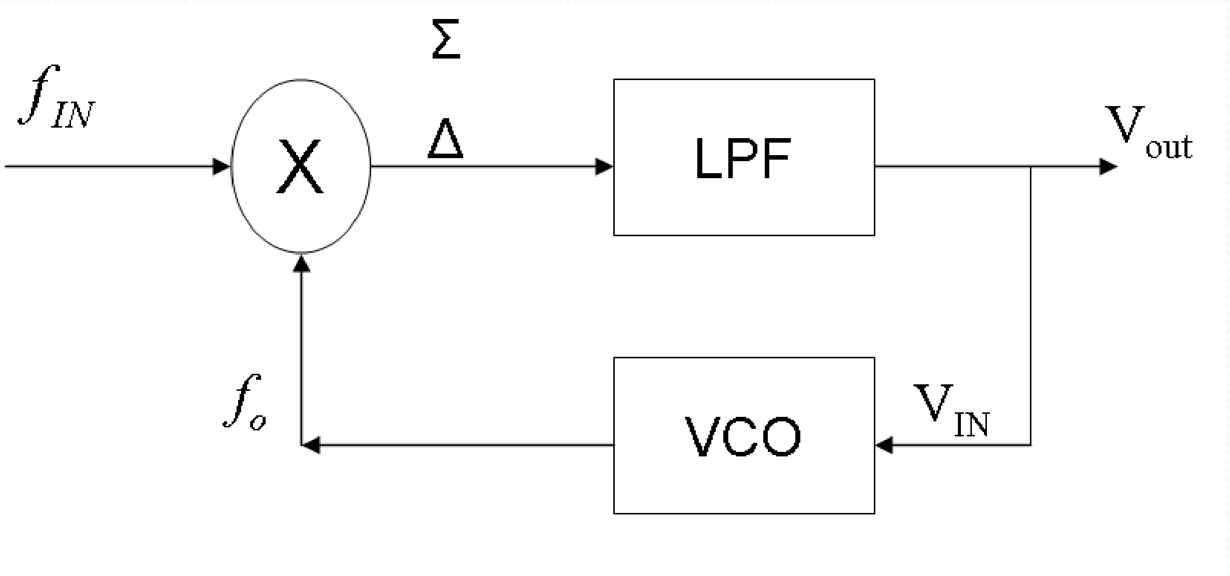
Foster-Seeley Discriminator



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 $m(t)$.