

UNIT-2

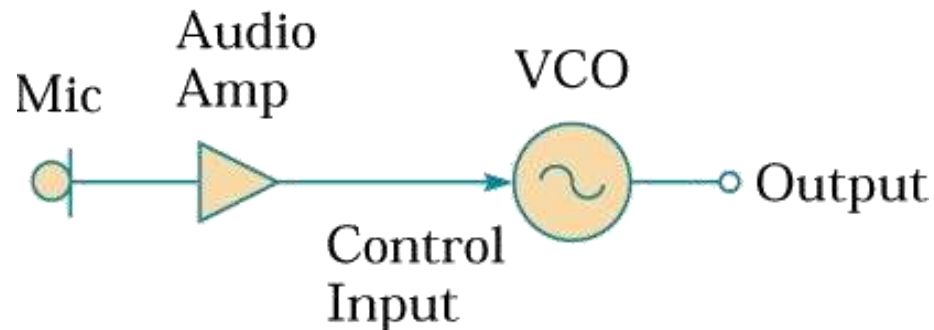
Angle Modulation System

Introduction

- There are three parameters of a carrier that may carry information:
 - Amplitude
 - Frequency
 - Phase

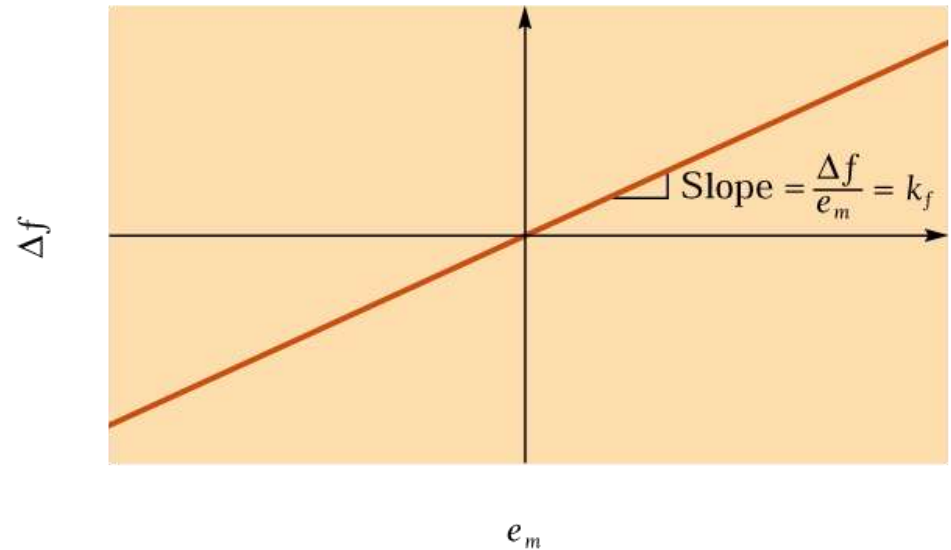
Frequency Modulation

- Power in an FM signal does not vary with modulation
- FM signals do not have an envelope that reproduces the modulation
- The figure below shows a simplified FM generator



Frequency Deviation

- Frequency deviation of the carrier is proportional to the amplitude of the modulating signal as illustrated



Frequency Modulation Index

- Another term common to FM is the *modulation index*, as determined by the formula:

$$m_f = \frac{\delta}{f_m}$$

Phase Modulation

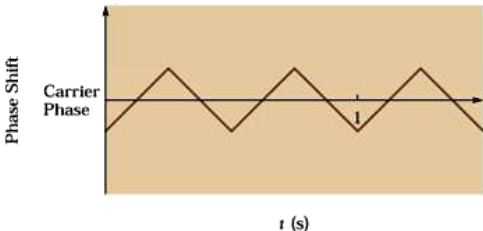
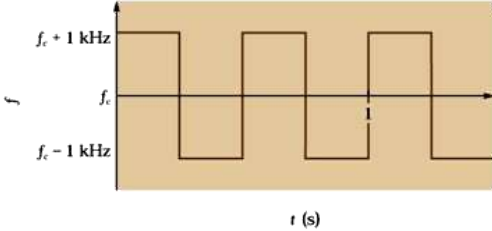
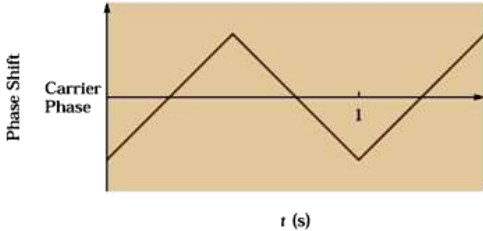
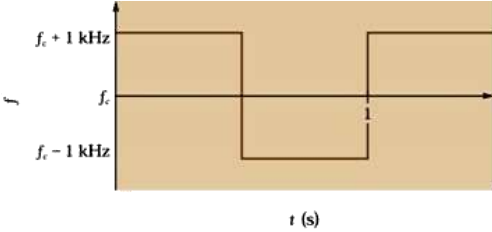
- In phase modulation, the phase shift is proportional to the instantaneous amplitude of the modulating signal, according to the formula:

$$k_p = \frac{\Phi}{e_m}$$

Relationship Between FM and Phase Modulation

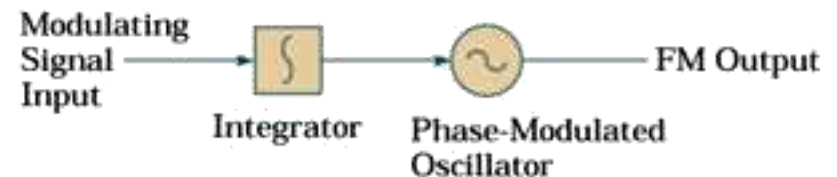
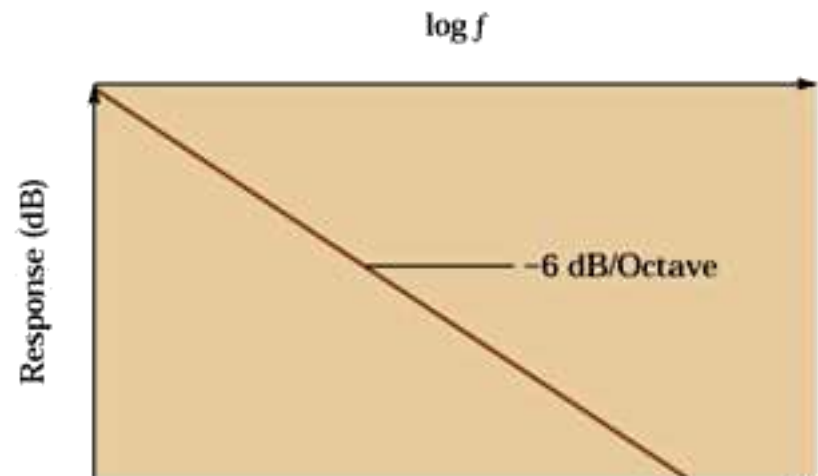
- Frequency is the derivative of phase, or, in other words, frequency is the rate of change of phase
- The modulation index is proportional to frequency deviation and inversely proportional to modulating frequency

Modulating Signal Frequency



Converting PM to FM

- An *integrator* can be used as a means of converting phase modulation to frequency modulation



The Angle Modulation Spectrum

- Angle modulation produces an infinite number of sidebands
- These sidebands are separated from the carrier by multiples of f_m
- For practical purposes an angle-modulated signal can be considered to be band-limited

Bessel Functions

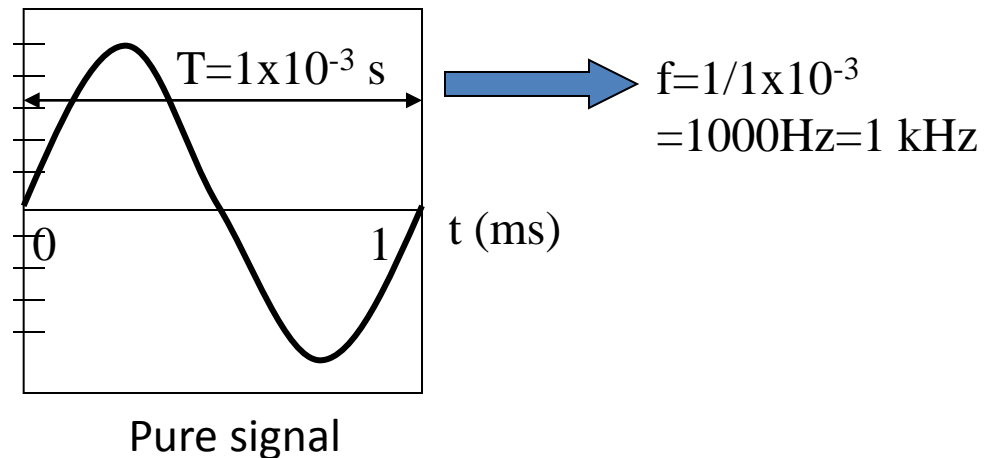
- FM and PM signals have similar equations regarding composition
- Bessel functions represent normalized voltages for the various components of an AM or PM signal

Bandwidth

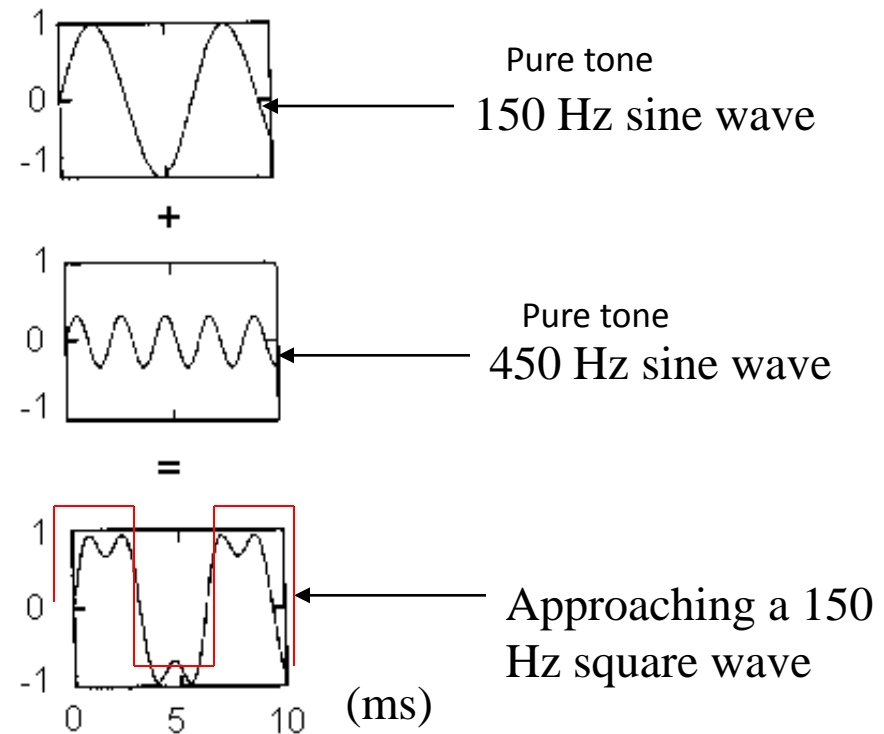
- For FM, the bandwidth varies with both deviation and modulating frequency
- Increasing modulating frequency reduces modulation index so it reduces the number of sidebands with significant amplitude
- On the other hand, increasing modulating frequency increases the frequency separation between sidebands
- Bandwidth increases with modulation frequency but is not directly proportional to it

❖ Bandwidth

- *Signal bandwidth:*
 - We can divide signals into two categories: The **pure tone** signal (the sinusoidal wave, consisting of one frequency component), and **complex** signals that are composed of several components, or sinusoids of various frequencies.

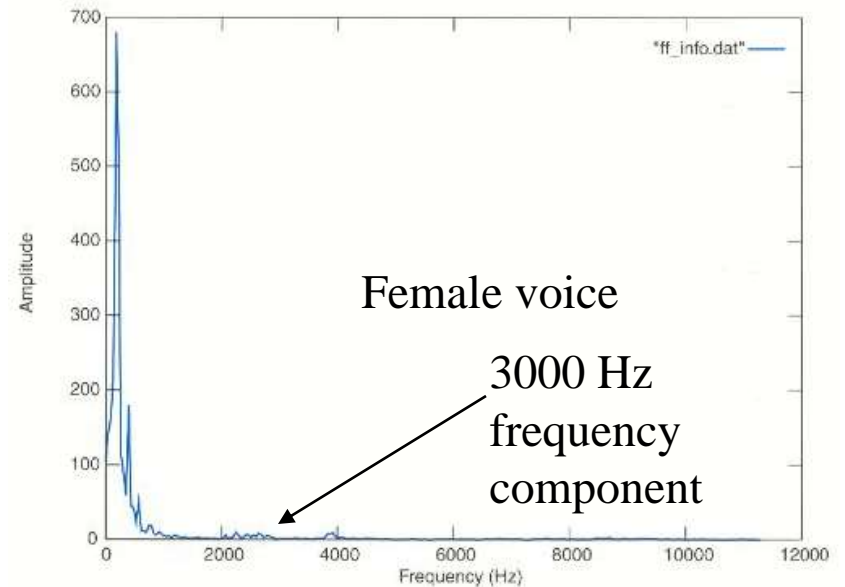
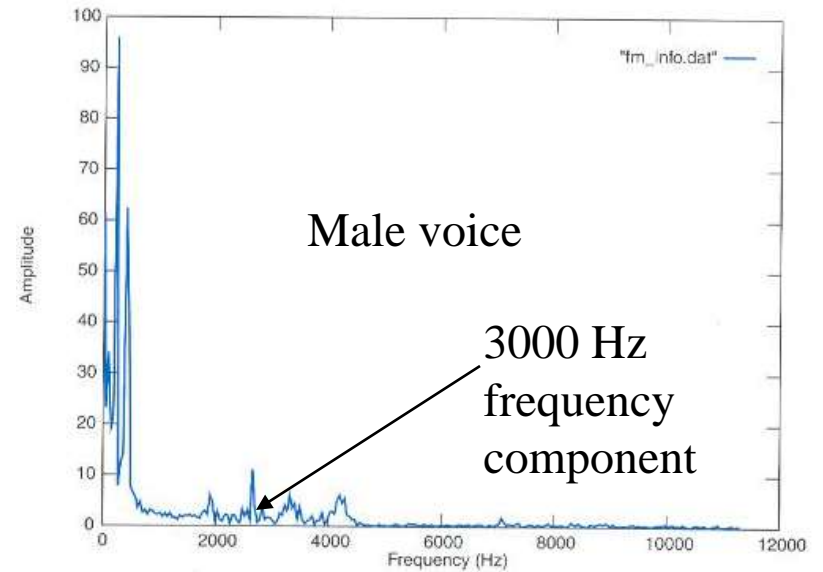


- The **bandwidth** of a signal composed of components of various frequencies (complex signal) is the **difference between its highest and lowest frequency components**, and is expressed in Hertz (Hz) - the same as frequency.
- For example, a square wave may be constructed by adding sine waves of various frequencies:



- The resulting wave resembles a square wave. If more sine waves of other frequencies were added, the resulting waveform would more closely resemble a square wave
- Since the resulting wave contains 2 frequency components, its bandwidth is around $450 - 150 = 300$ Hz.

- Since voice signals are also composed of several components (pure tones) of various frequencies, the bandwidth of a voice signal is taken to be the difference between the highest and lowest frequencies which are 3000 Hz and (close to) 0 Hz
- Although other frequency components above 3000 Hz exist, (they are more prominent in the male voice), an acceptable degradation of voice quality is achieved by disregarding the higher frequency components, accepting the 3kHz bandwidth as a standard for voice communications



- *channel bandwidth:*

- The bandwidth of a channel (medium) is defined to be the range of frequencies that the medium can support. Bandwidth is measured in Hz
- With each transmission medium, there is a frequency range of electromagnetic waves that can be transmitted:
 - Twisted pair cable: 0 to 10^9 Hz (Bandwidth : 10^9 Hz)
 - Coax cable: 0 to 10^{10} Hz (Bandwidth : 10^{10} Hz)
 - Optical fiber: 10^{14} to 10^{16} Hz (Bandwidth : $10^{16} - 10^{14} = 9.9 \times 10^{15}$ Hz)
- Optical fibers have the highest bandwidth (they can support electromagnetic waves with very high frequencies, such as light waves)
- The bandwidth of the channel dictates the information carrying capacity of the channel
- This is calculated using Shannon's channel capacity formula

Increasing
bandwidth



❖ Shannon's Theorem

(Shannon's Limit for Information Capacity)

- Claude Shannon at Bell Labs figured out how much information a channel could theoretically carry:

$$I = B \log_2 (1 + S/N)$$



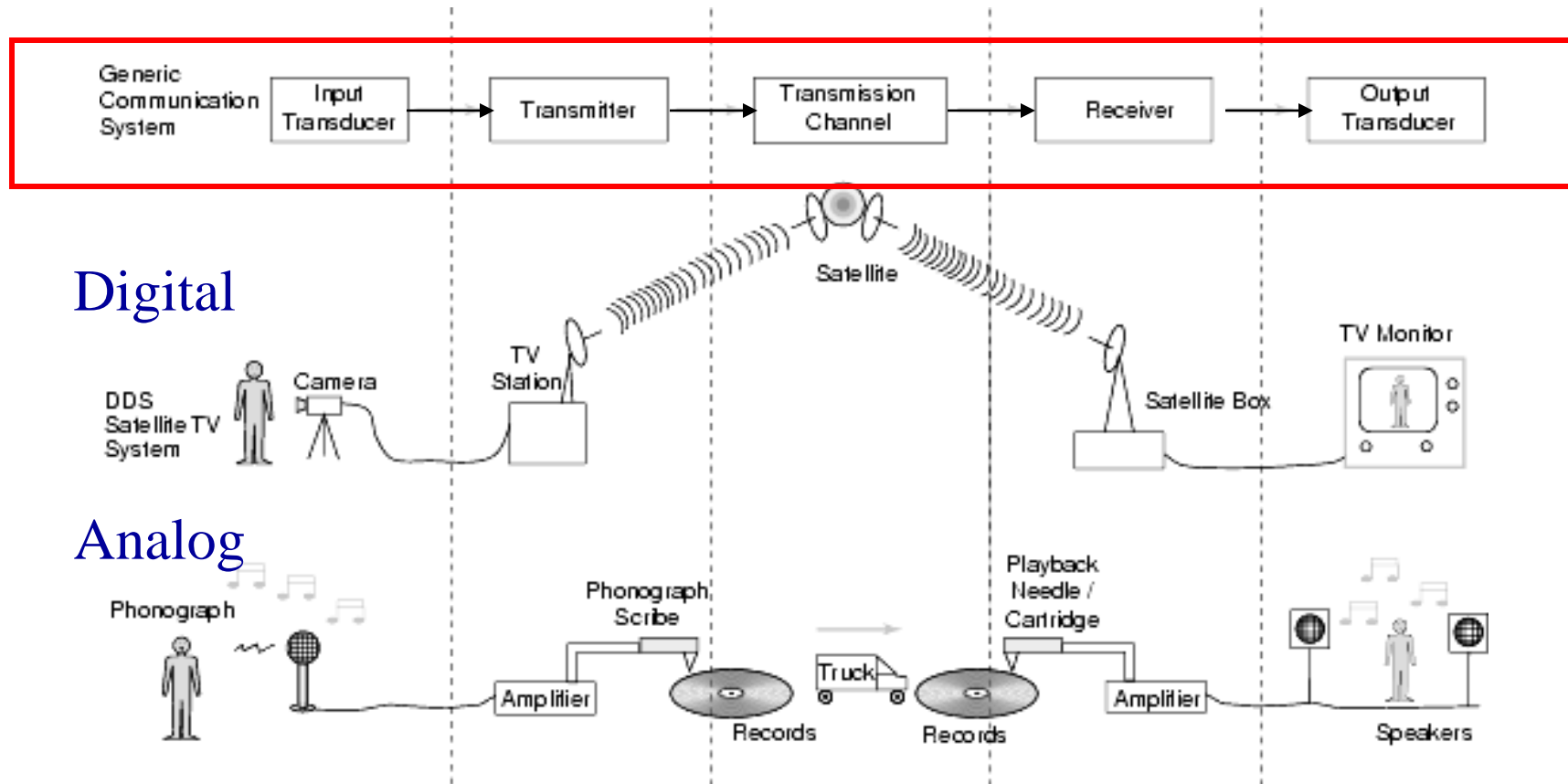
**Note that the log
is base 2!**

- Where I is Information Capacity in bits per second (bps)
- B is the channel bandwidth in Hz
- S/N is Signal-to-Noise ratio (SNR: unitless...don't make into decibel: dB)

Signal-to-Noise Ratio

- S/N is normally measured in dB (decibel). It is a relationship between the signal we want versus the noise that we do not want, which is in the medium.
- It can be thought of as a fractional relationship (that is, before we take the logarithm):
- 1000W of signal power versus 20W of noise power is either:
 - $1000/20=50$ (unitless!)
 - or: about 17 dB $\Rightarrow 10 \log_{10} 1000/20 = 16.9897$ dB

❖ Communication systems



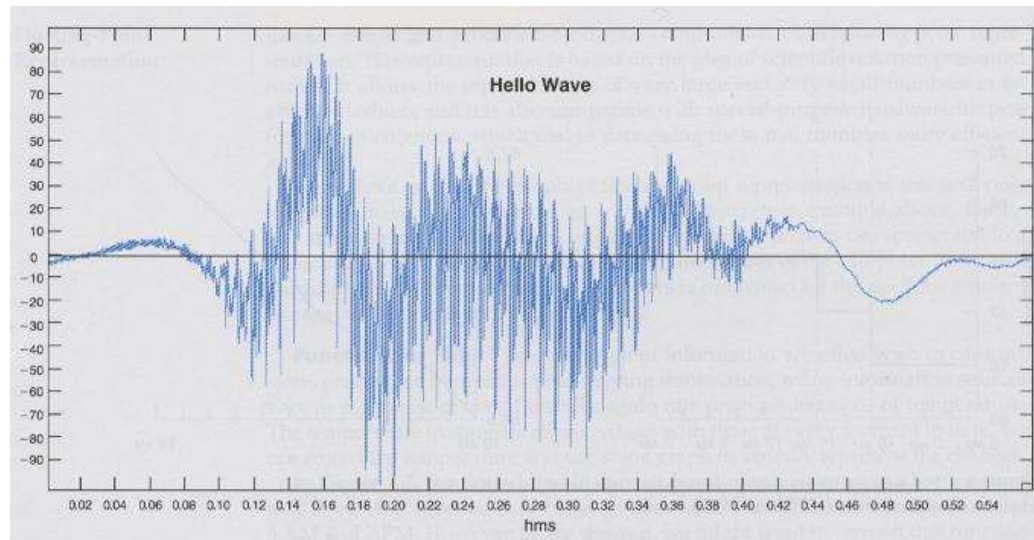
- The block diagram on the top shows the blocks common to all communication systems

We recall the components of a communication system:

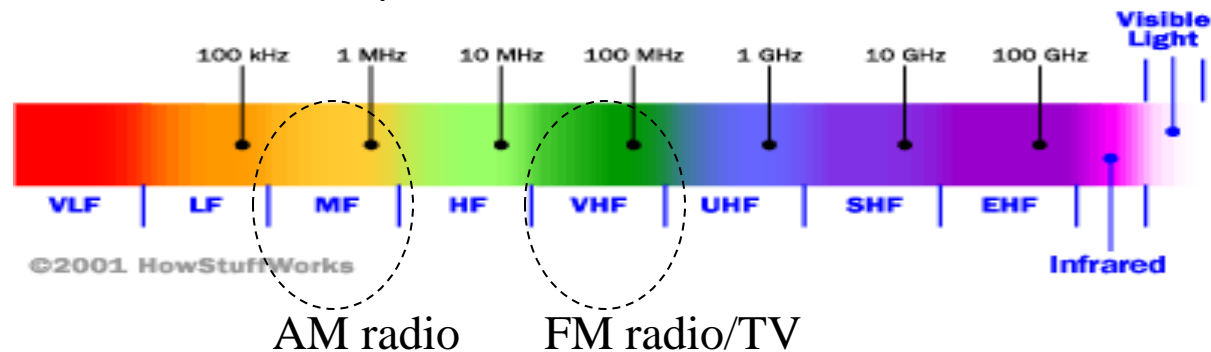
- *Input transducer*: The device that converts a physical signal from source to an electrical, mechanical or electromagnetic signal more suitable for communicating
- *Transmitter*: The device that sends the transduced signal
- *Transmission channel*: The physical medium on which the signal is carried
- *Receiver*: The device that recovers the transmitted signal from the channel
- *Output transducer*: The device that converts the received signal back into a useful quantity

❖ Analog Modulation

- The purpose of a communication system is to transmit information signals (baseband signals) through a communication channel
- The term *baseband* is used to designate the band of frequencies representing the original signal as delivered by the input transducer
 - For example, the voice signal from a microphone is a baseband signal, and contains frequencies in the range of 0-3000 Hz
 - The “hello” wave is a baseband signal:



- Since this *baseband* signal must be transmitted through a communication channel (such as air or cable) using electromagnetic waves, a procedure is needed to shift the range of *baseband* frequencies to other frequency ranges suitable for transmission; and, a corresponding shift back to the original frequency range after reception. This is called the process of *modulation* and *demodulation*
- Remember the radio spectrum:



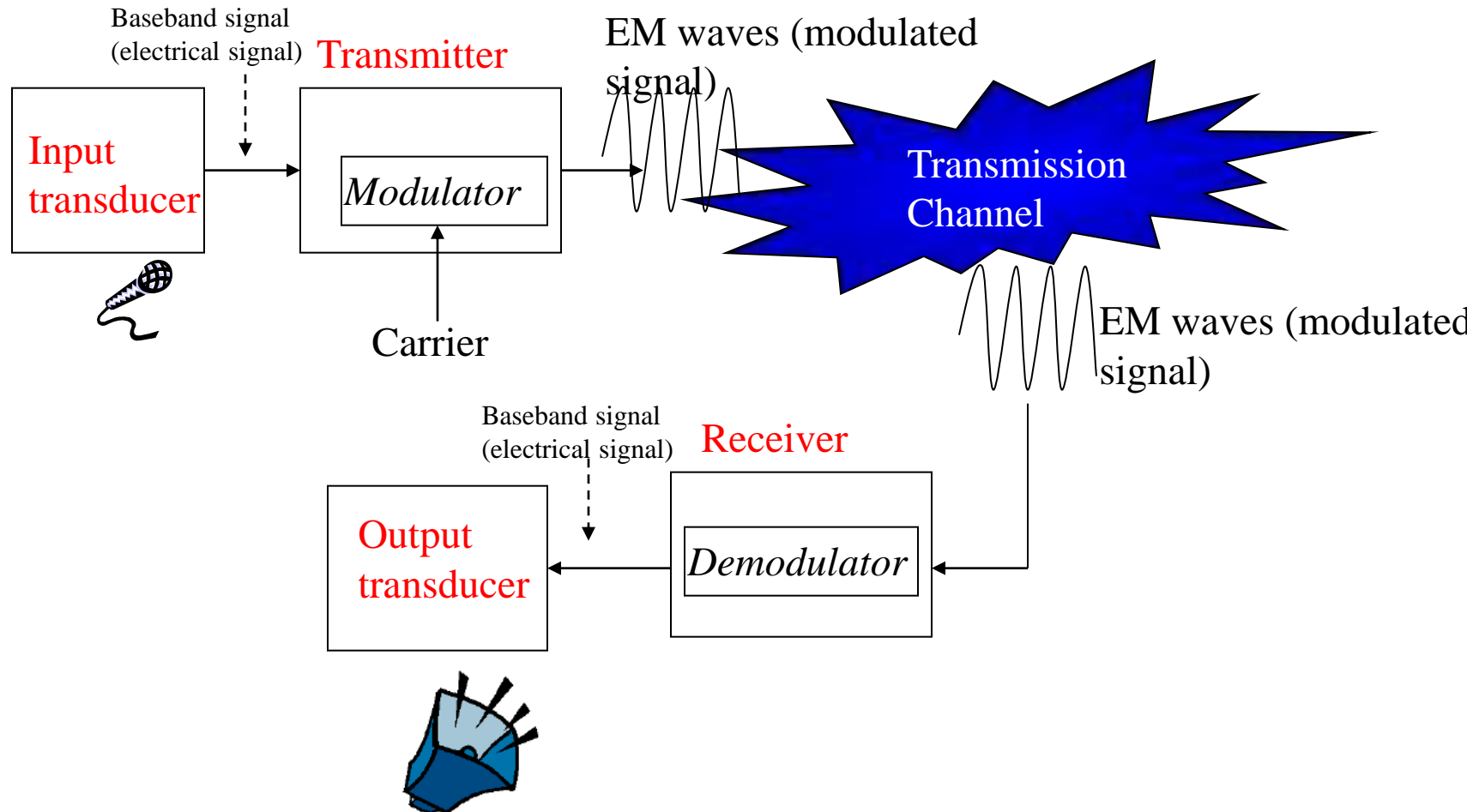
- For example, an AM radio system transmits electromagnetic waves with frequencies of around a few hundred kHz (MF band)
- The FM radio system operates with frequencies in the range of 88-108 MHz (VHF band)

- Since the *baseband* signal contains frequencies in the audio frequency range (3 kHz), some form of frequency-band shifting must be employed for the radio system to operate properly
- This process is accomplished by a device called a *modulator*
- The **transmitter** block in any communications system contains the *modulator* device
- The **receiver** block in any communications system contains the *demodulator* device
- The modulator **modulates** a *carrier wave* (the electromagnetic wave) which has a frequency that is selected from an appropriate band in the radio spectrum
 - For example, the frequency of a carrier wave for FM can be chosen from the VHF band of the radio spectrum
 - For AM, the frequency of the carrier wave may be chosen to be around a few hundred kHz (from the MF band of the radio spectrum)
- The demodulator extracts the original *baseband* signal from the received modulated signal

In Summary:

- Modulation is the process of impressing a low-frequency information signal (baseband signal) onto a higher frequency carrier signal

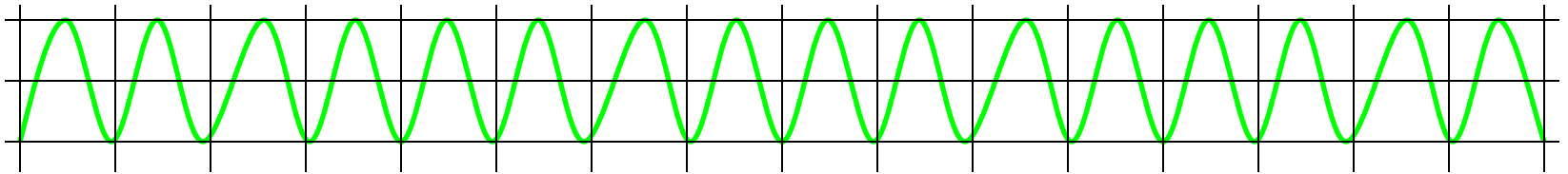
Basic analog communications system



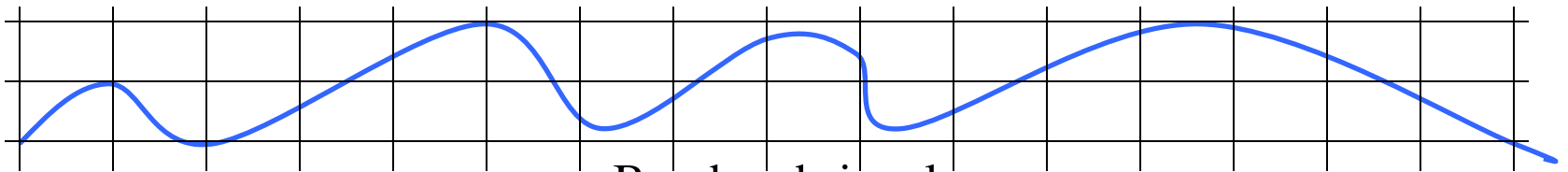
Types of Analog Modulation

- **Amplitude Modulation (AM)**
 - Amplitude modulation is the process of ***varying the amplitude of a carrier wave in proportion to the amplitude of a baseband signal***. The frequency of the carrier remains constant
- **Frequency Modulation (FM)**
 - Frequency modulation is the process of ***varying the frequency of a carrier wave in proportion to the amplitude of a baseband signal***. The amplitude of the carrier remains constant
- **Phase Modulation (PM)**
 - Another form of analog modulation technique which we will not discuss

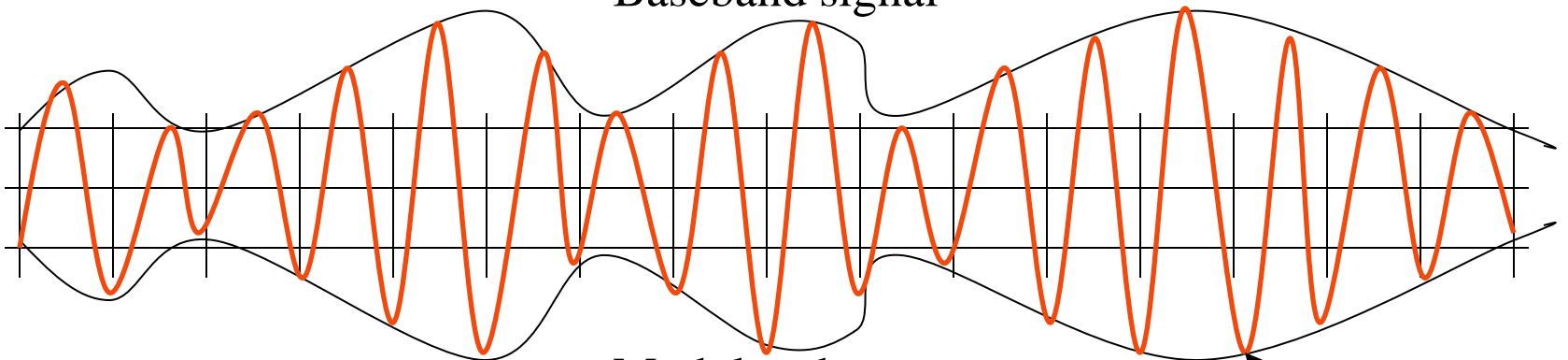
Amplitude Modulation



Carrier wave



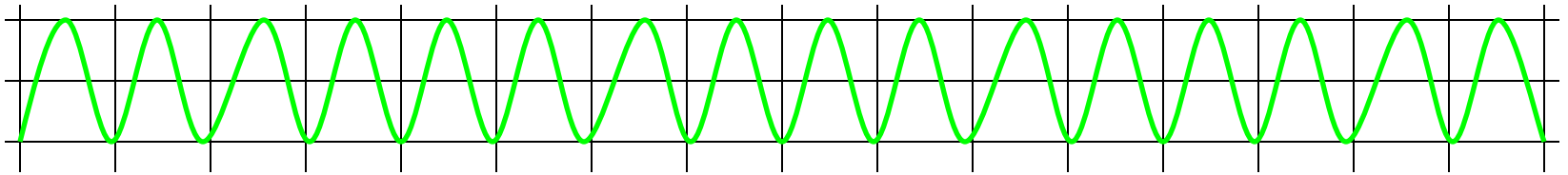
Baseband signal



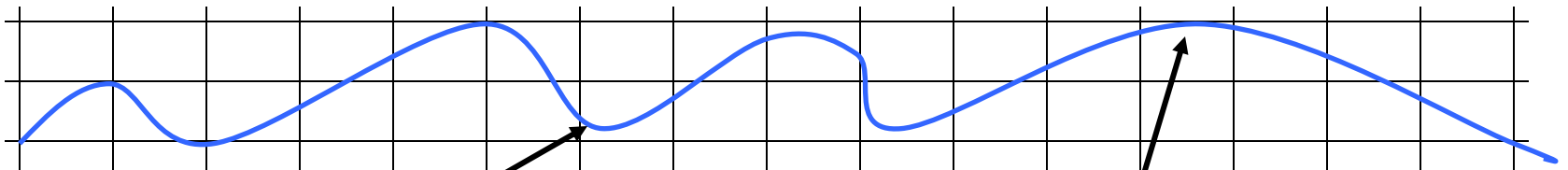
Modulated wave

Amplitude varying-
frequency constant

Frequency Modulation



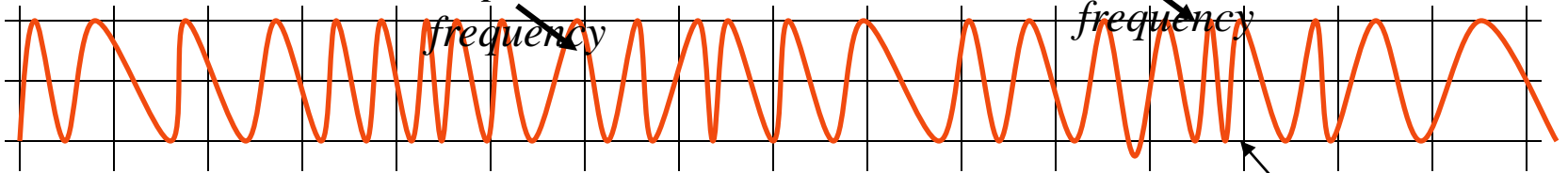
Carrier wave



Baseband signal

*Small
amplitude: low
frequency*

*Large
amplitude: high
frequency*



Modulated wave

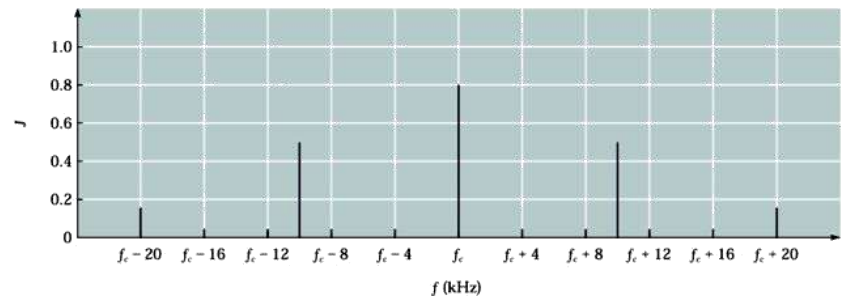
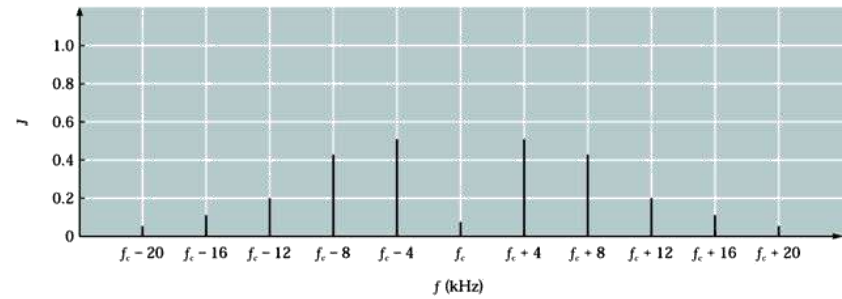
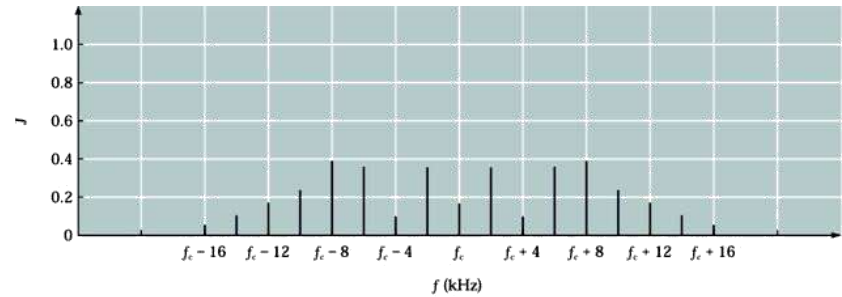
Frequency varying-
amplitude constant

Carson's Rule

- Calculating the bandwidth of an FM signal is simple, but tedious using Bessel functions
- Carson's Rule provides an adequate approximation for determining FM signal bandwidth:

$$B = 2\left(\delta_{\max} + f_{m(\max)}\right)$$

Variation of FM Signal



Narrowband and Wideband FM

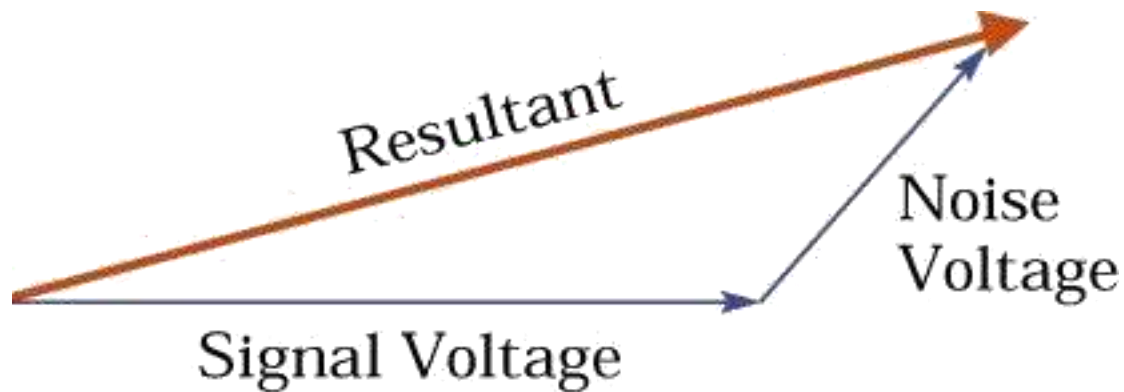
- There are no theoretical limits to the modulation index or the frequency deviation of an FM signal
- The limits are a practical compromise between signal-to-noise ratio and bandwidth
- Government regulations limit the bandwidth of FM transmissions in terms of maximum frequency deviation and the maximum modulation frequency

Narrow- and Wideband Signals

- Narrowband FM (NBFM) is used for voice transmissions
- Wideband FM (WBFM) is used for most other transmissions
- Strict definition of the term *narrowband FM* refers to a signal with m_f of less than 0.5

FM and Noise

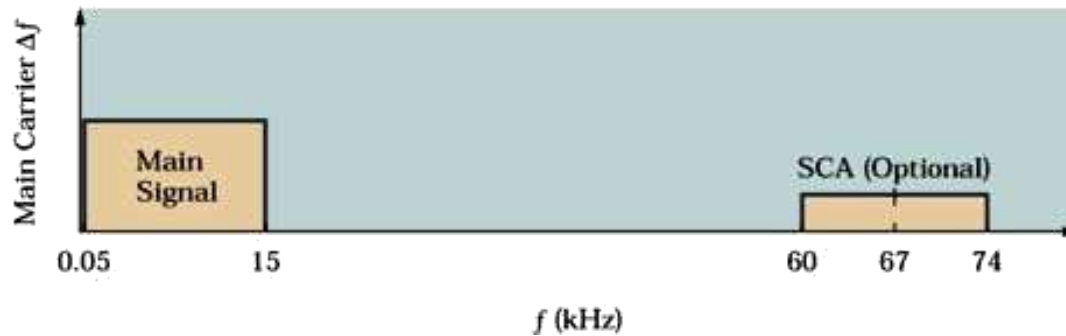
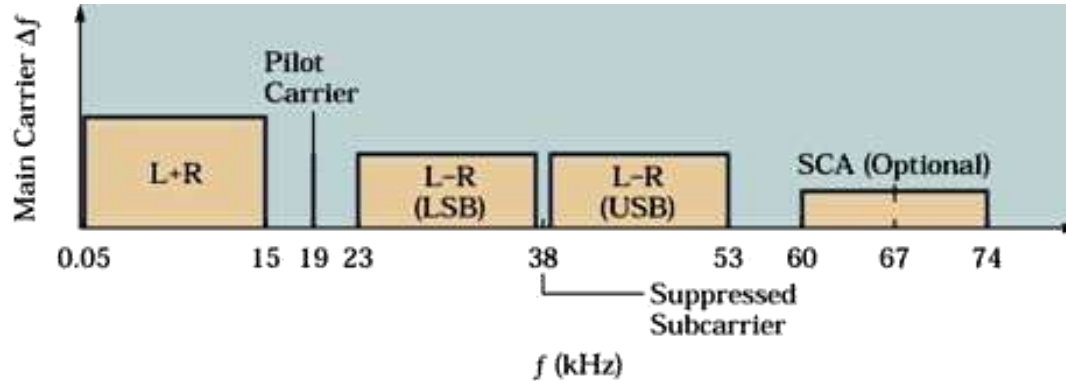
- One of the original reasons for developing FM was to give improved performance in the presence of noise, which is still one of the major advantages over AM
- One way to approach the problem of FM and noise is think of noise as a phasor of random amplitude and phase angle



FM Stereo

- The introduction of FM stereo in 1961 was accomplished in such a way so as to insure compatibility with existing FM monaural systems
- The mono FM receivers must be able to capture the L+R signal of a stereo transmitter

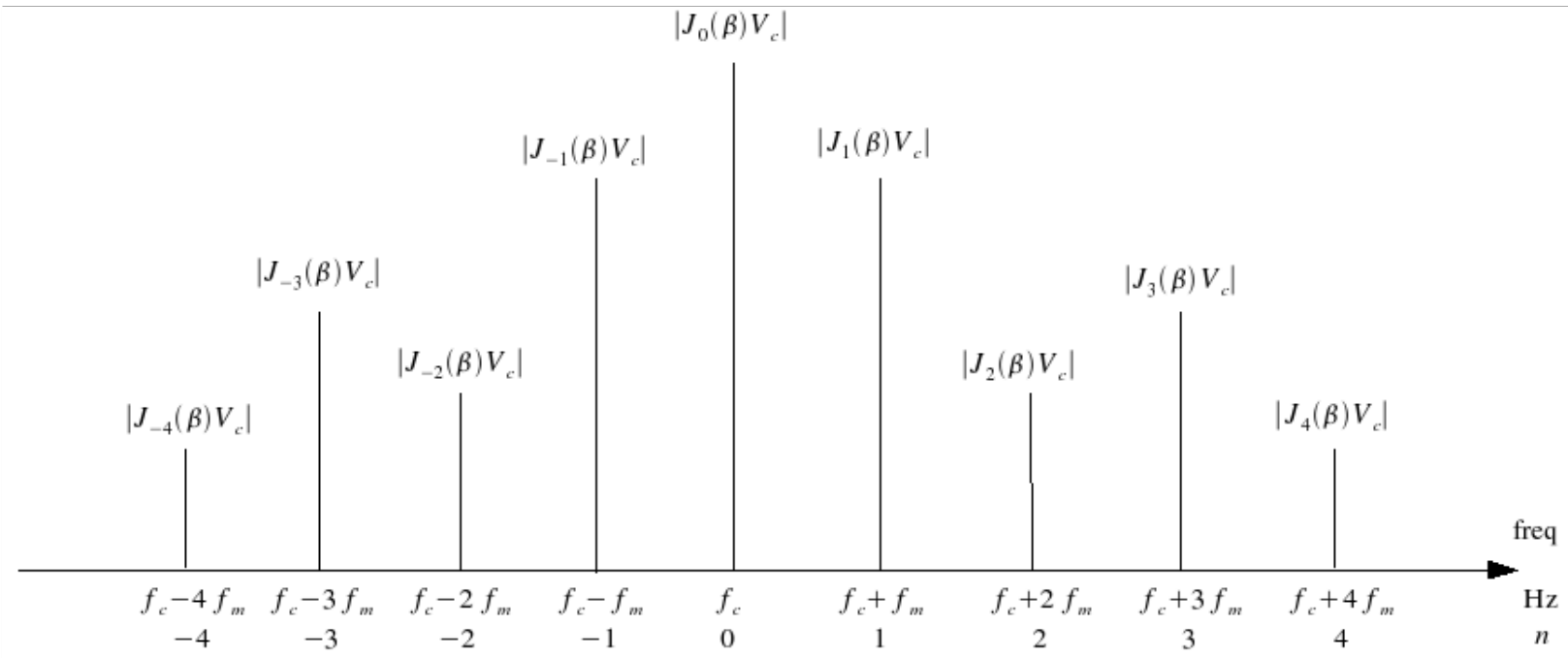
FM Broadcasting Spectra



FM Measurements

- The maximum frequency deviation of an FM transmitter is restricted by law, not by any physical constraint
- Traditional oscilloscope displays are not useful in analyzing FM signals
- A spectrum analyzer is much more useful in determining the qualities of an FM signal

FM Signal Spectrum.



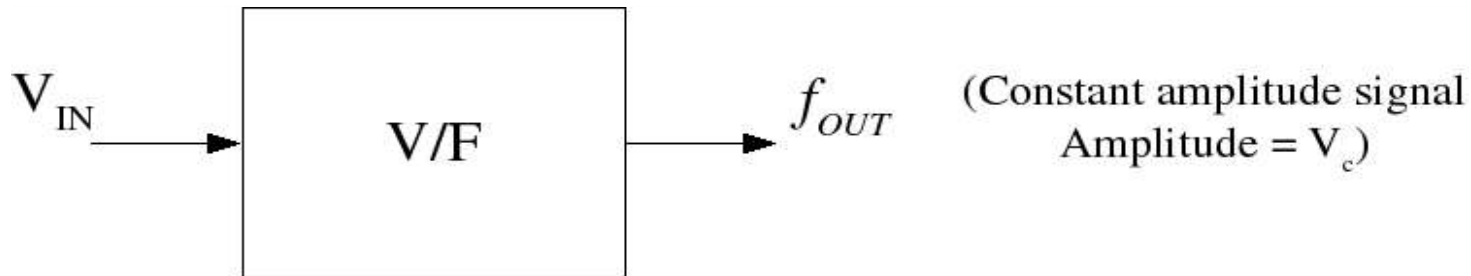
The amplitudes drawn are completely arbitrary, since we have not found any value for $J_n(\beta)$ – this sketch is only to illustrate the spectrum.

Generation of FM signals – Frequency Modulation.

An FM demodulator is:

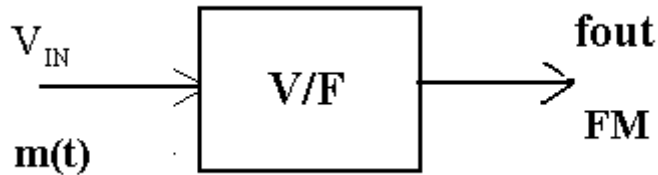
- a voltage-to-frequency converter V/F
- a voltage controlled oscillator VCO

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

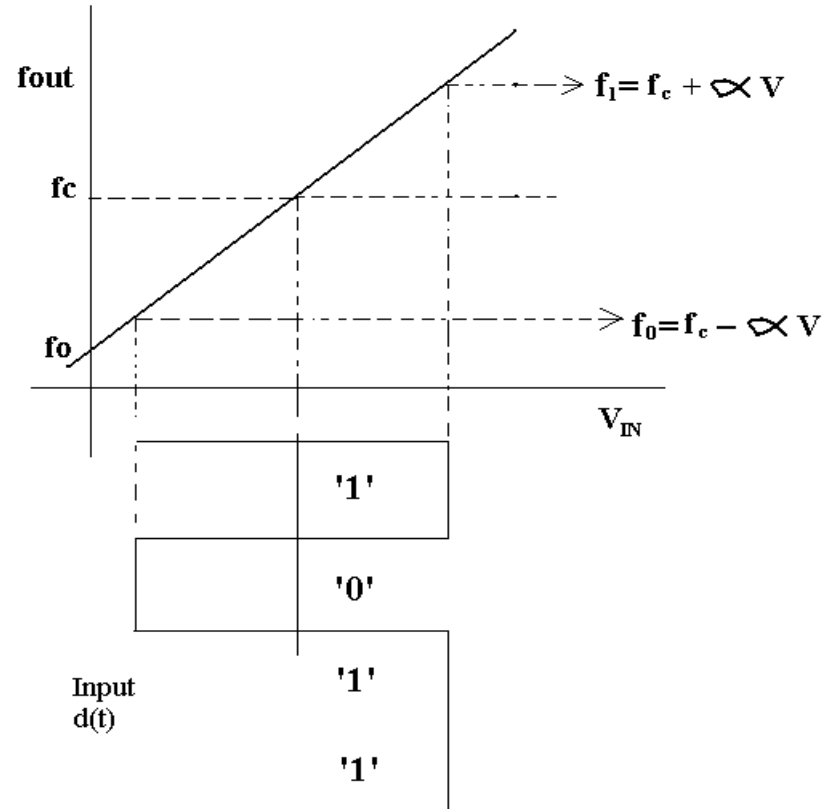


FM Signal Waveforms.

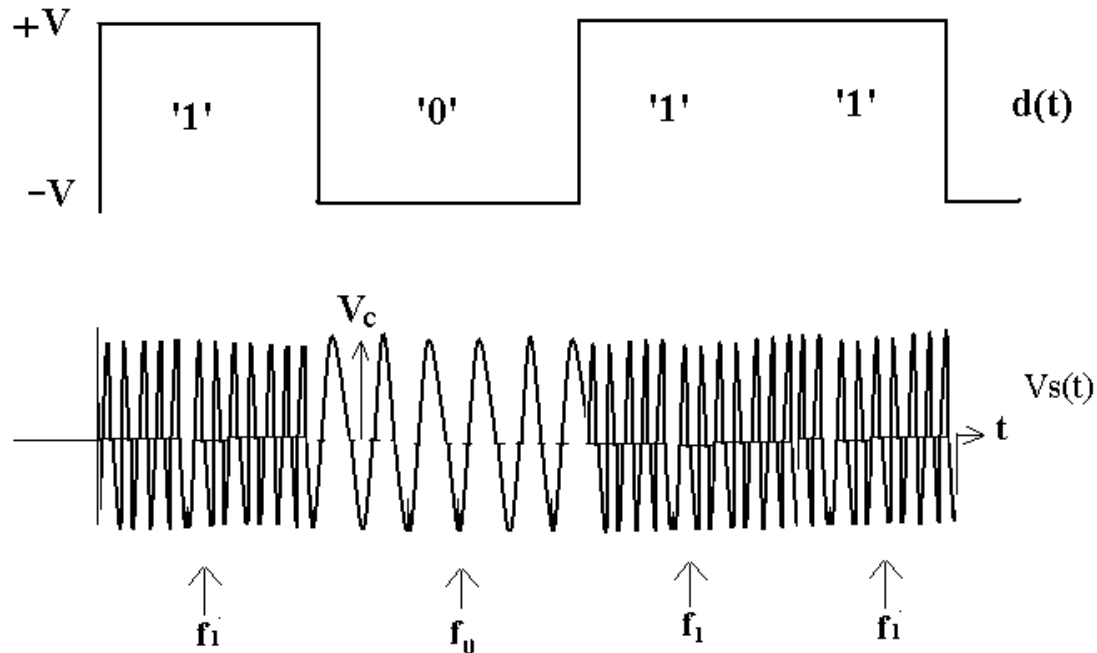
The diagrams below illustrate FM signal waveforms for various inputs



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



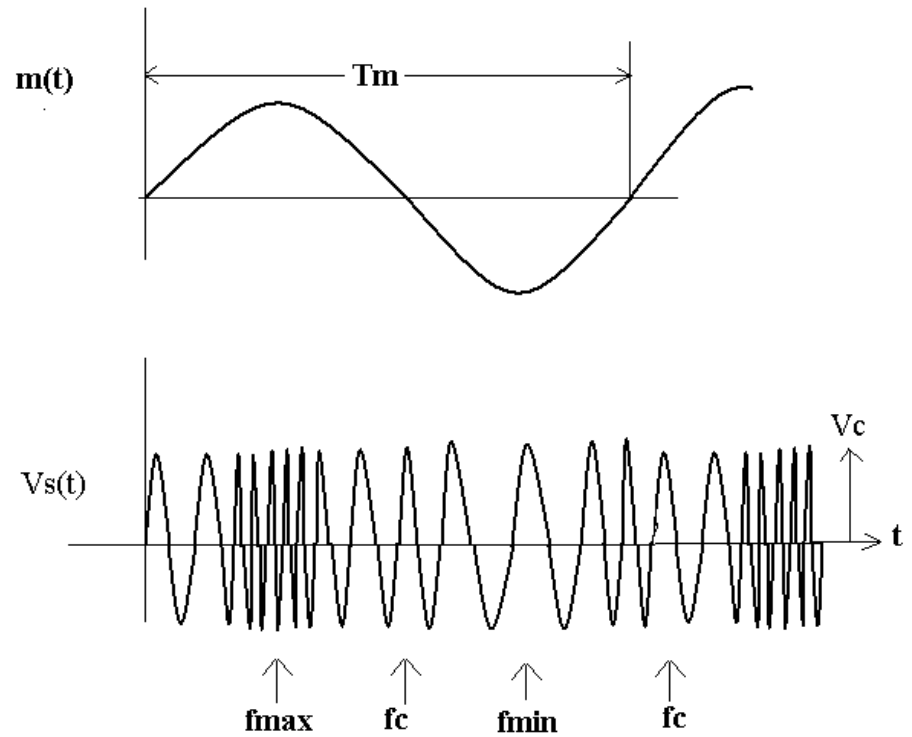
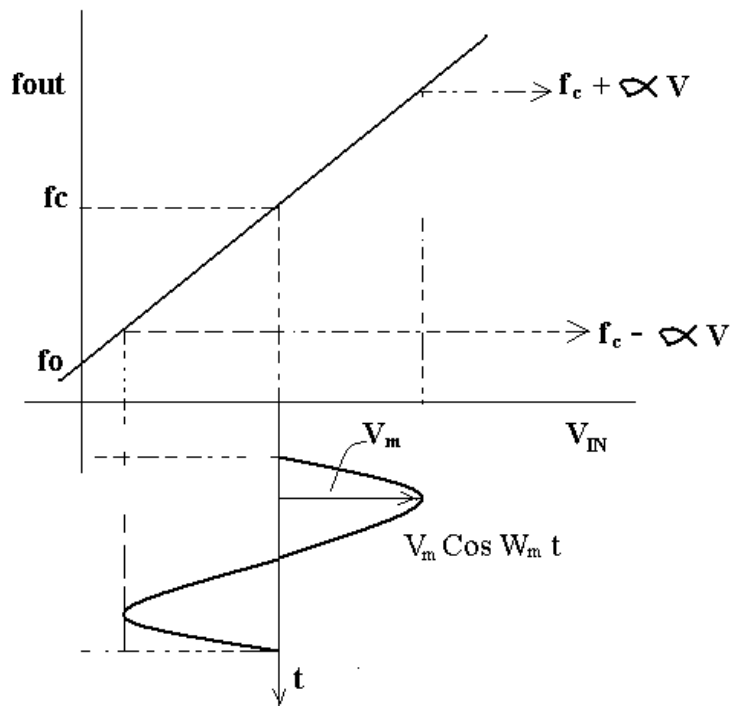
FM Signal Waveforms.



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

$= -V$ for 0's $f_{OUT} = f_0 = f_c - \alpha V$ for 0's } between f_1 and f_0 .

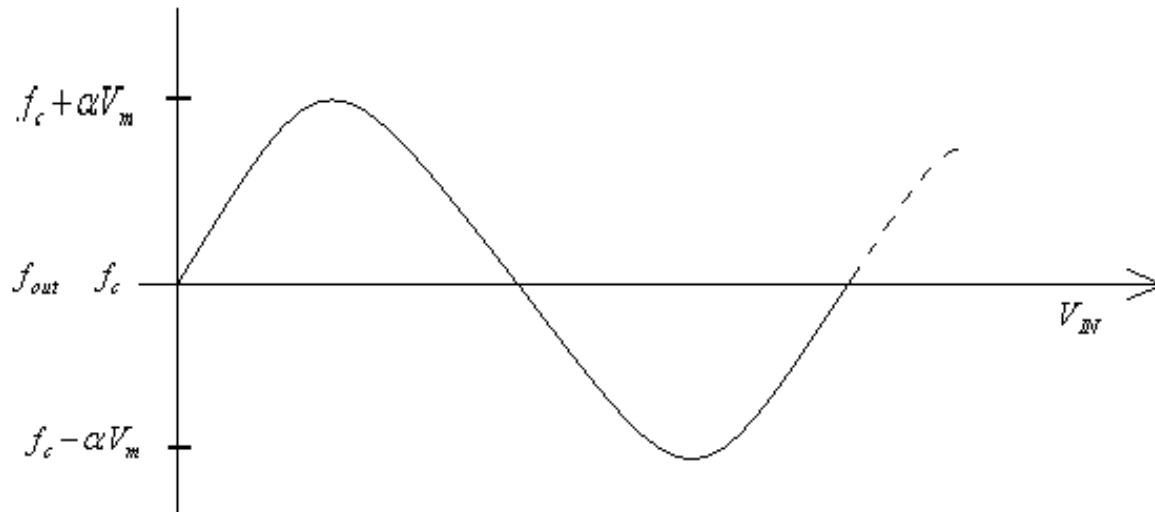
FM Signal Waveforms.



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} :

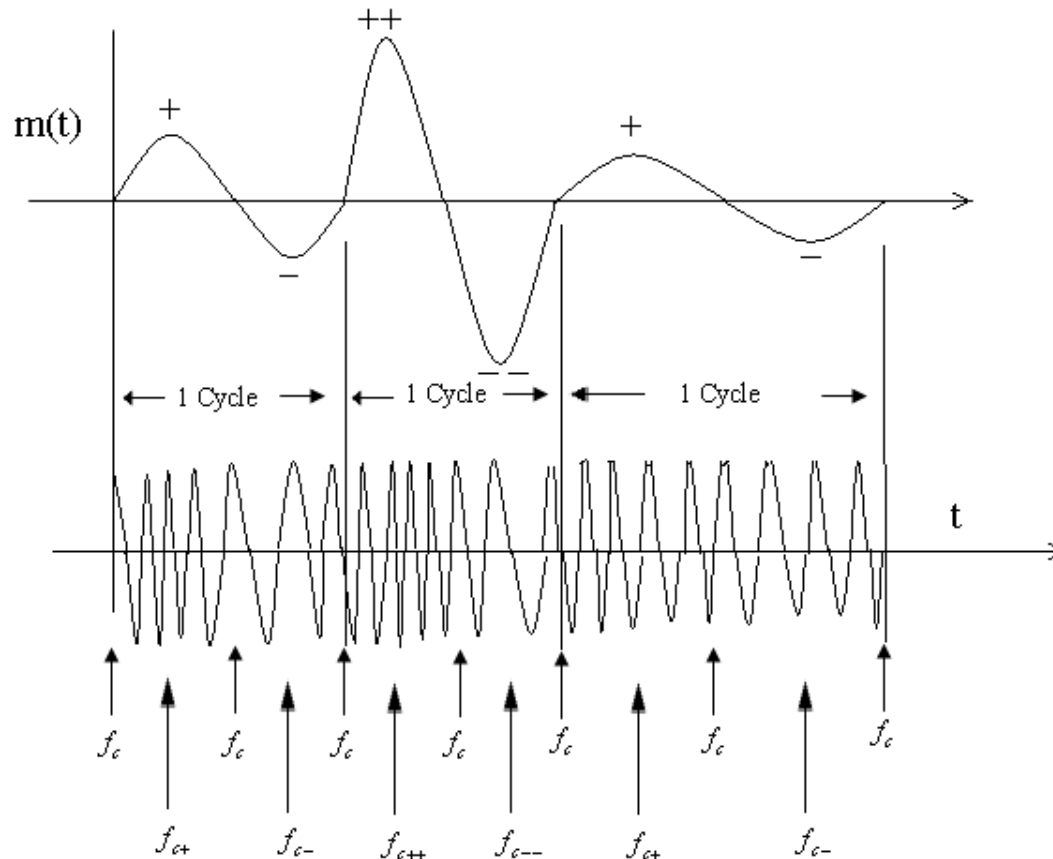


In general, $m(t)$ will be a 'band of signals', *i.e.* it will contain amplitude and frequency variations. Both amplitude and frequency change in $m(t)$ at the input are translated to (just) frequency changes in the FM output signal, *i.e.* the amplitude of the output FM signal is constant.

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:



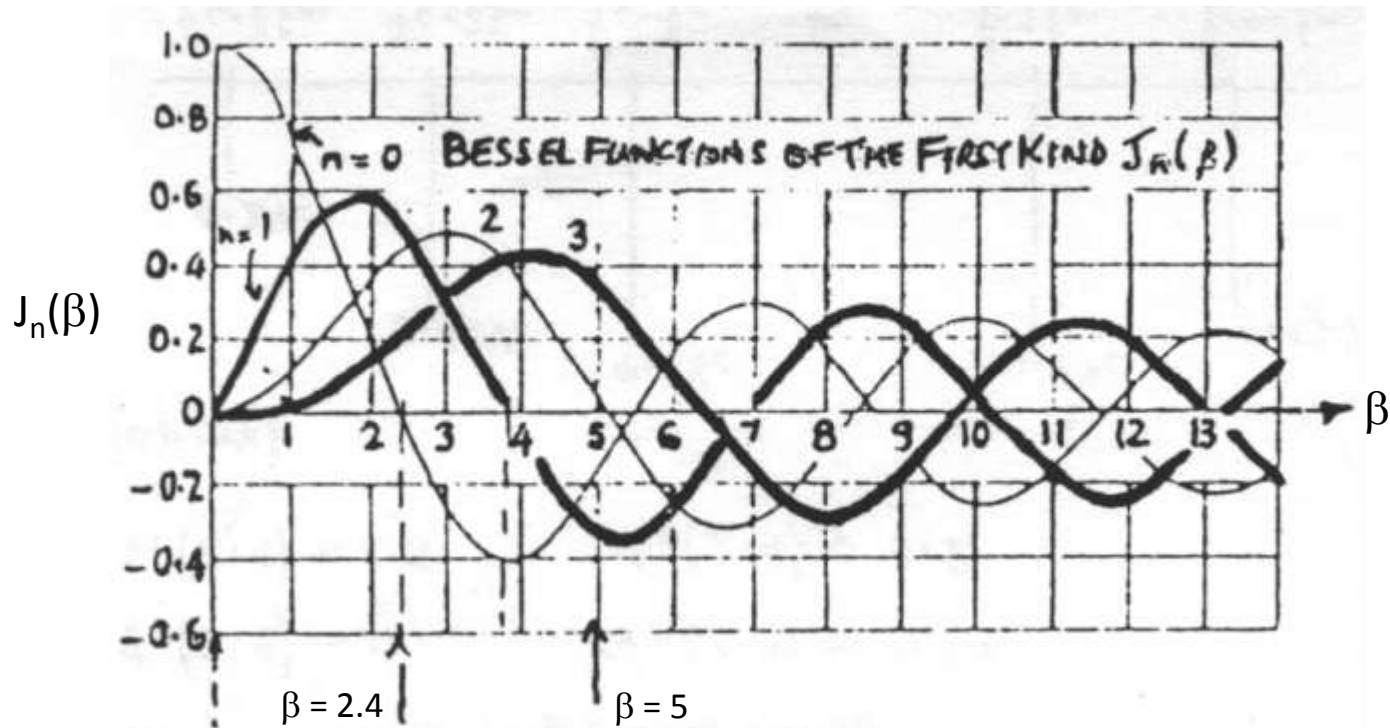
FM Spectrum – Bessel Coefficients.

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.

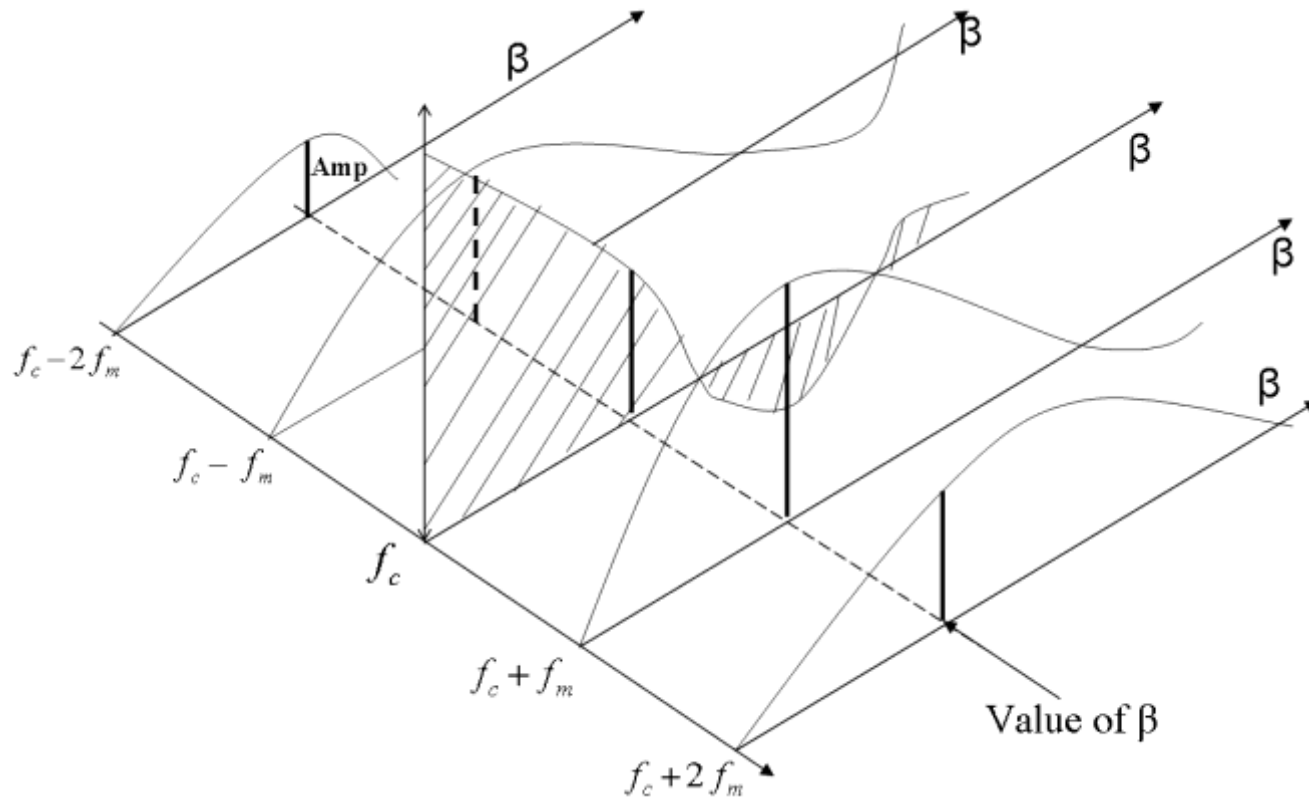


In the series for $v_s(t)$, $n = 0$ is the carrier component, *i.e.* $V_c J_0(\beta) \cos(\omega_c t)$, hence the $n = 0$ curve shows how the component at the carrier frequency, f_c , varies in amplitude, with modulation index β .

FM Spectrum – Bessel Coefficients.

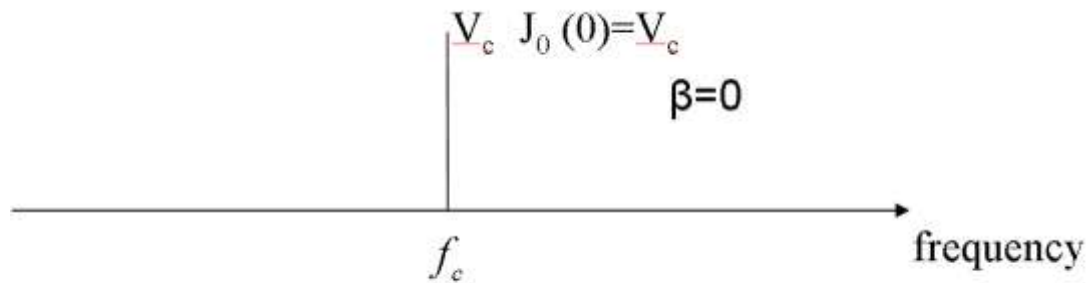
Hence for a given value of modulation index β , the values of $J_n(\beta)$ may be read off the graph and hence the component amplitudes ($V_c J_n(\beta)$) may be determined.

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



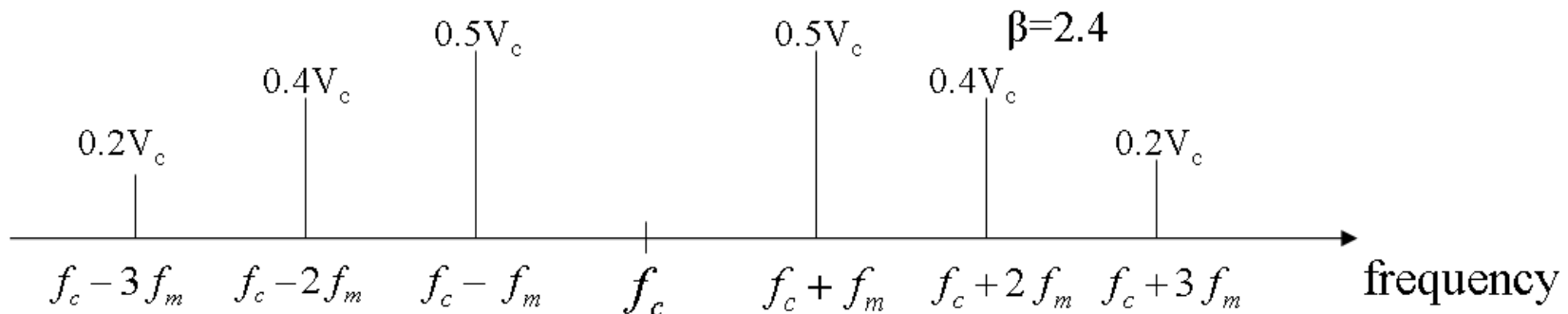
Examples from the graph

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



$\beta = 2.4$: 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.

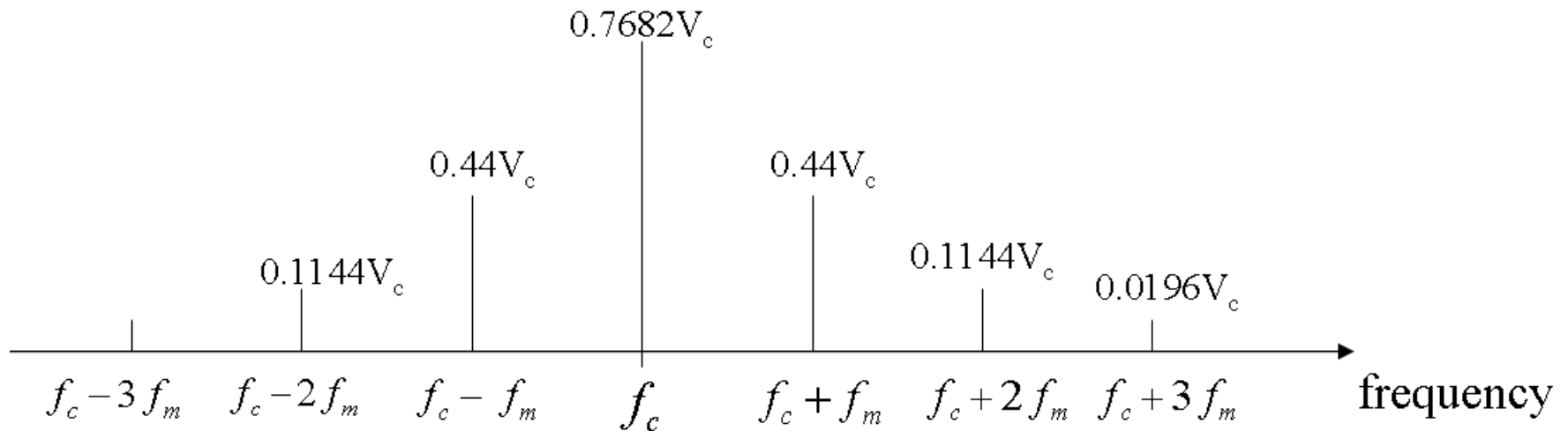
As may be seen from the table of Bessel functions, for values of n above a certain value, the values of $J_n(\beta)$ become progressively smaller. In FM the sidebands are considered to be significant if $J_n(\beta) \geq 0.01$ (1%).

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	$0.0196V_c$	$f_c + 3f_m$ $f_c - 3f_m$
4	0.0025	<i>Insignificant</i>	
5	0.0002	<i>Insignificant</i>	

Significant Sidebands – Spectrum.



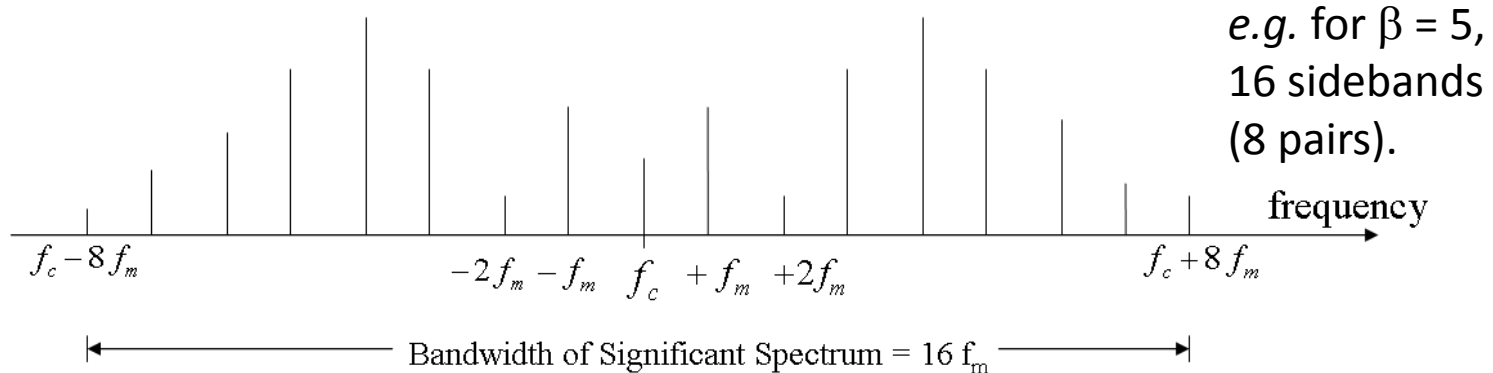
Bandwidth of Significant Spectrum = $6f_m$

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 $\geq 1\%$ 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	$28f_m$



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)

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

Narrowband and Wideband FM

Narrowband FM NBFM

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

FM with $\beta \leq 0.3$ is referred to as **narrowband FM** (NBFM) (Note, the bandwidth is the same as DSBAM).

Wideband FM WBFM

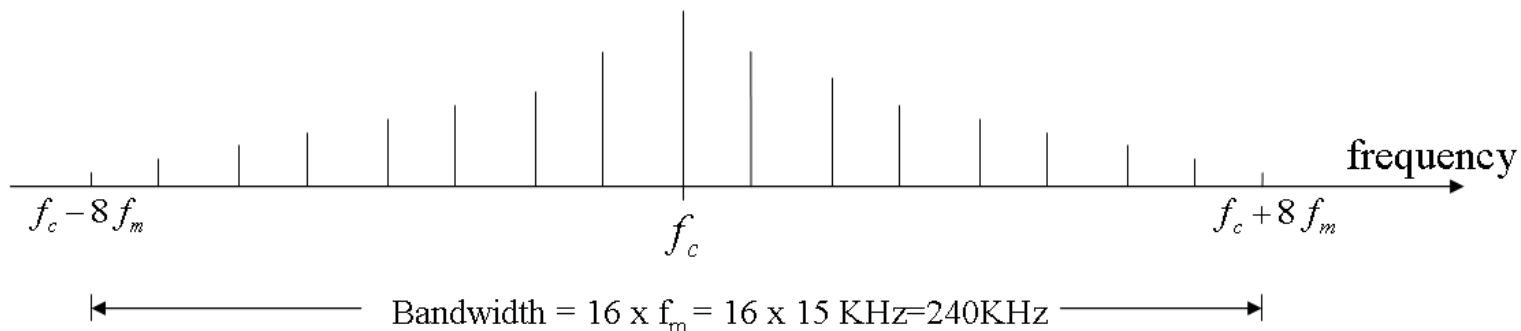
For $\beta > 0.3$ there are more than 2 significant sidebands. As β increases the number of sidebands increases. This is referred to as **wideband FM** (WBFM).

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 (<i>e.g.</i> music)	15kHz	f_m
Deviation	75kHz	$\Delta f_c = \alpha V_m$
Modulation Index β	5	$\beta = \frac{\Delta f_c}{f_m}$

For $\beta = 5$ there are 16 sidebands and the FM signal bandwidth is $16f_m = 16 \times 15\text{kHz} = 240\text{kHz}$. Applying Carson's Rule $\text{BW} = 2(75+15) = 180\text{kHz}$.



Comments FM

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

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

- In FM we refer to sideband pairs not upper and lower sidebands. Carrier or other components may not be suppressed in FM.
- The relative amplitudes of components in FM depend on the values $J_n(\beta)$, where $\beta = \frac{\alpha V_m}{f_m}$ 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, $f_m = 15$ kHz 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

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

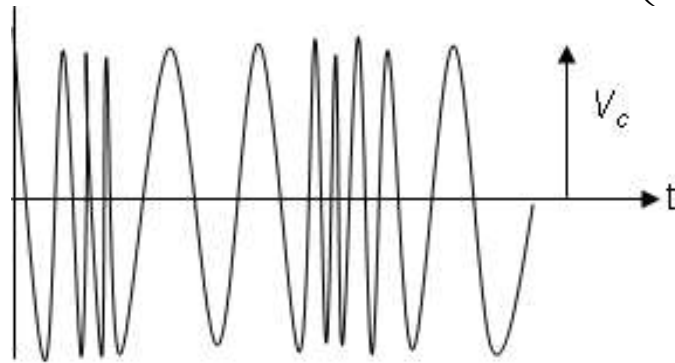
Hence, the total power in the infinite spectrum is

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

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 $\left(\frac{V_{pk}}{\sqrt{2}}\right)^2 = \frac{V_c}{2}$

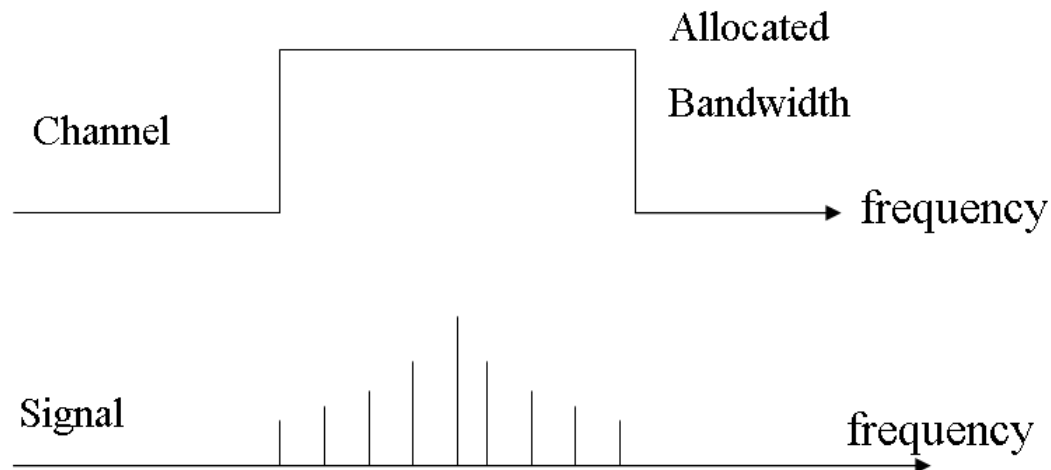


and power = $(V_{RMS})^2$ then we may deduce that $P_T = \left(\frac{V_c}{\sqrt{2}}\right)^2 = \frac{V_c^2}{2} = \sum_{n=-\infty}^{\infty} \frac{(V_c J_n(\beta))^2}{2}$

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.

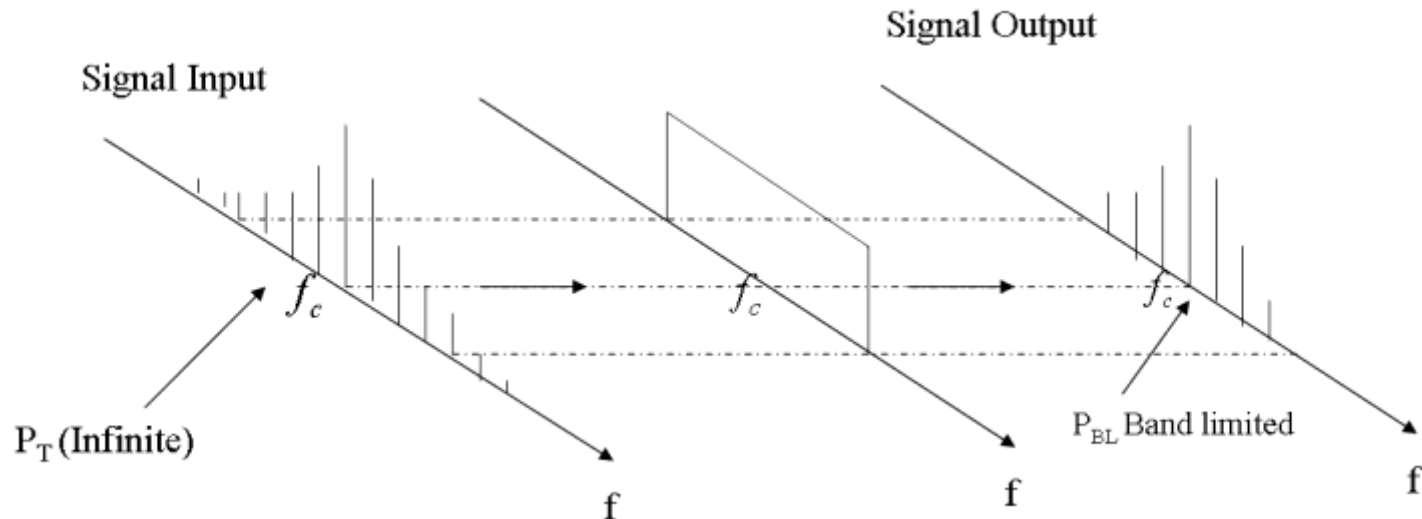
Now consider – if we generate an FM signal, it will contain an infinite number of sidebands. However, if we wish to transfer this signal, *e.g.* over a radio or cable, this implies that we require an infinite bandwidth channel. Even if there was an infinite channel bandwidth it would not all be allocated 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

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

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

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

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

Power in FM Signals.

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

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

Also, it is easily seen that the ratio

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

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

Example

Consider NBFM, with $\beta = 0.2$. 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

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

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

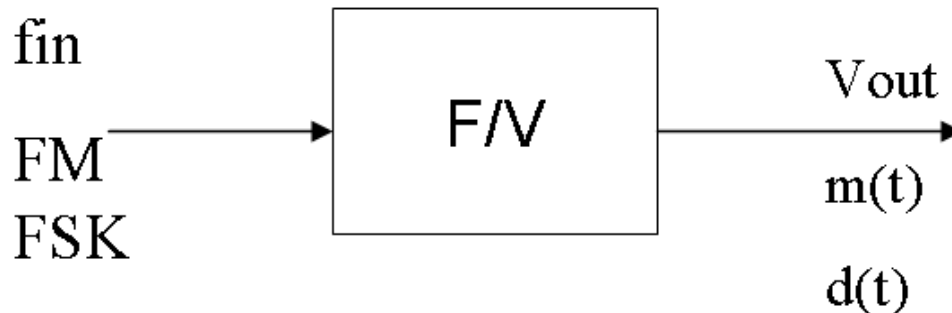
$$\text{Distortion} = 1 - \sum_{n=-1}^1 (J_n(0.2))^2 \quad (a = 1)$$

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

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

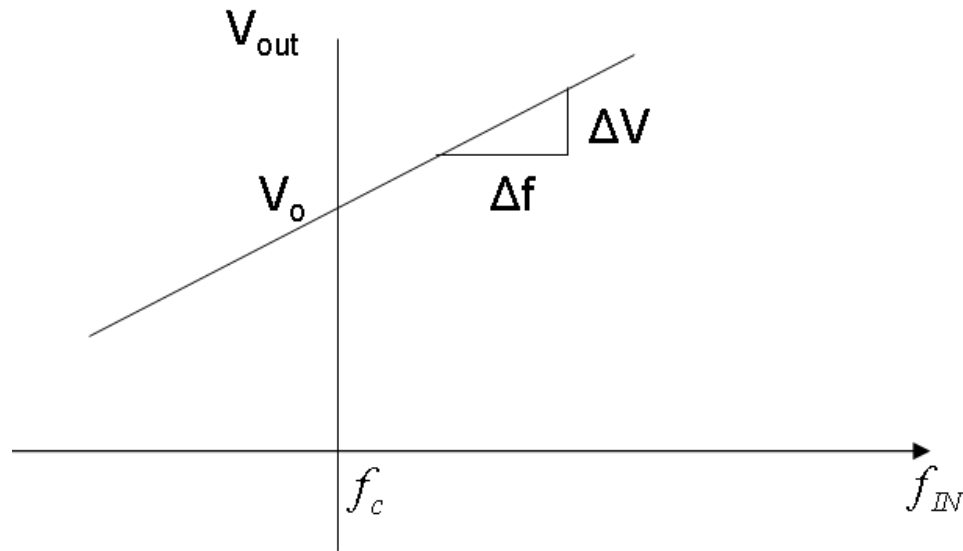
FM Demodulation –General Principles.

- 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

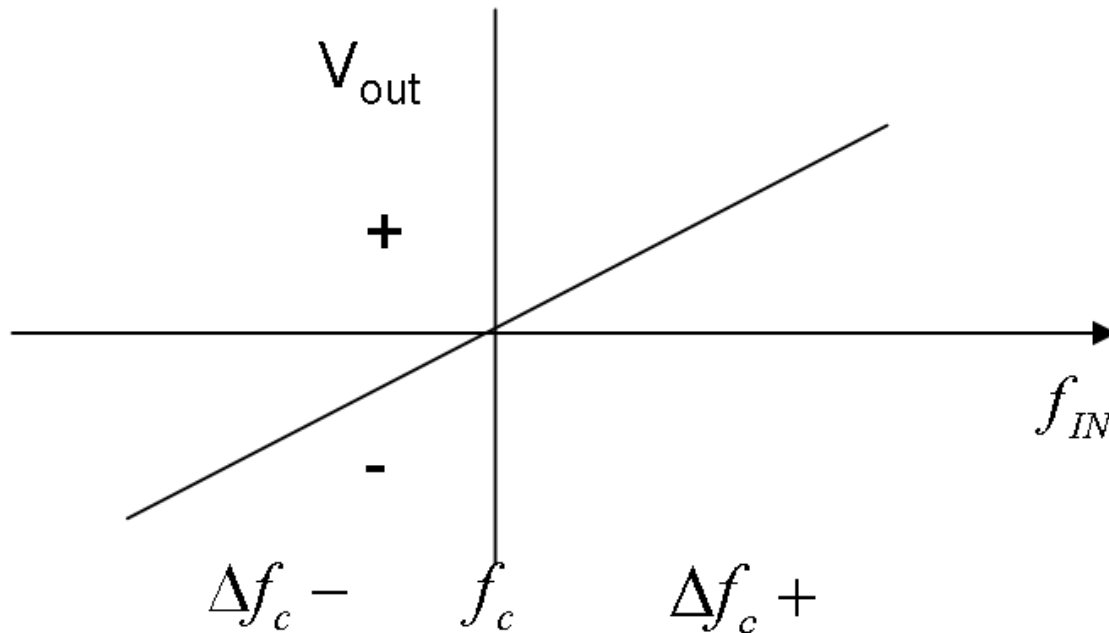
i.e. **Gradient = Voltage Conversion Factor**, K volts per Hz.

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 $V_{OUT} = m(t)$. A complete system is illustrated.

$$K = \frac{1}{\alpha}, \text{ so that } K\alpha = 1 \text{ and}$$



Gradient = α Hz/Volt
 α = Frequency conversion factor

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

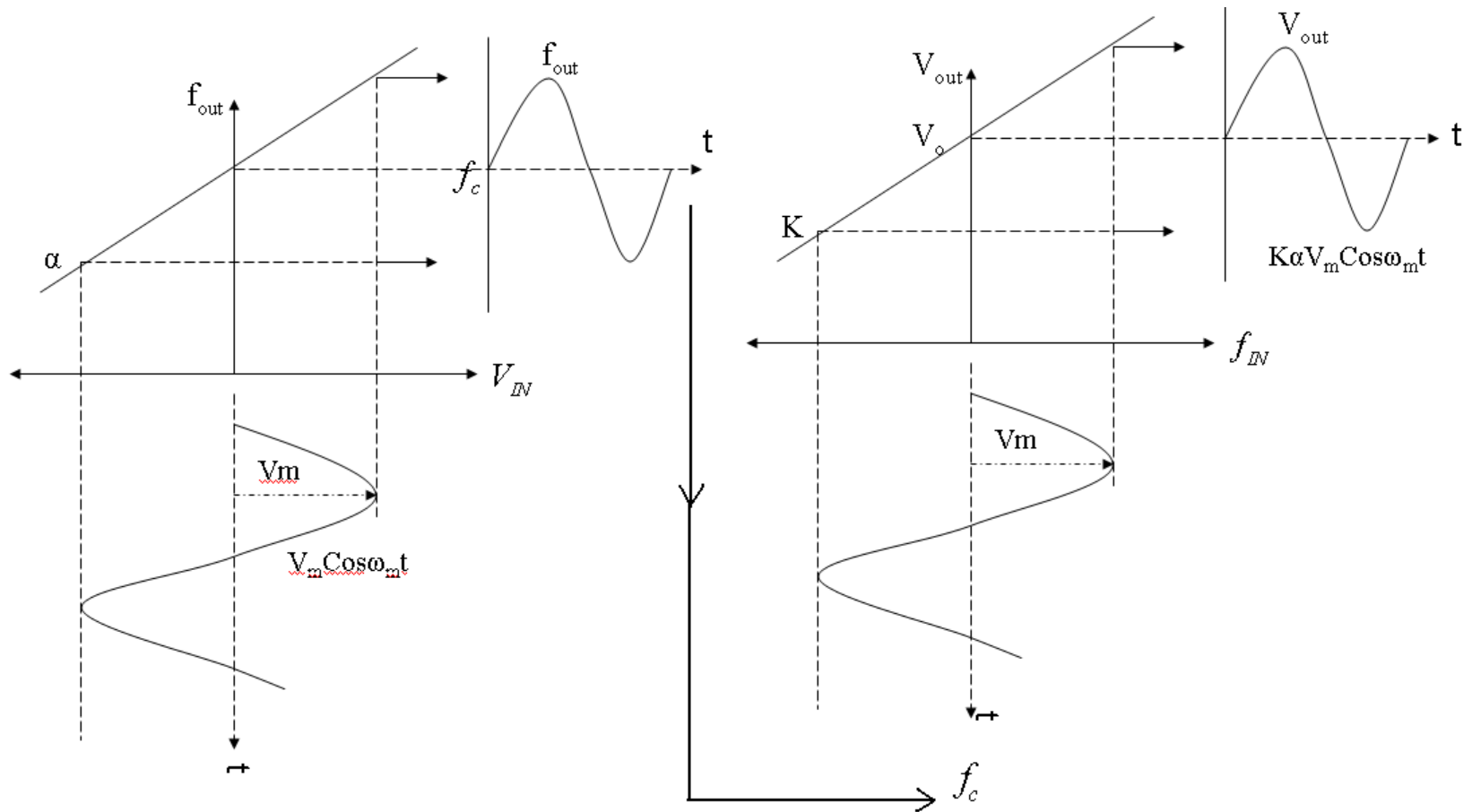
$$f_{OUT} = f_c + \alpha m(t) = f_{IN}$$

Gradient = K Hz/Volt
 K = Voltage conversion factor

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

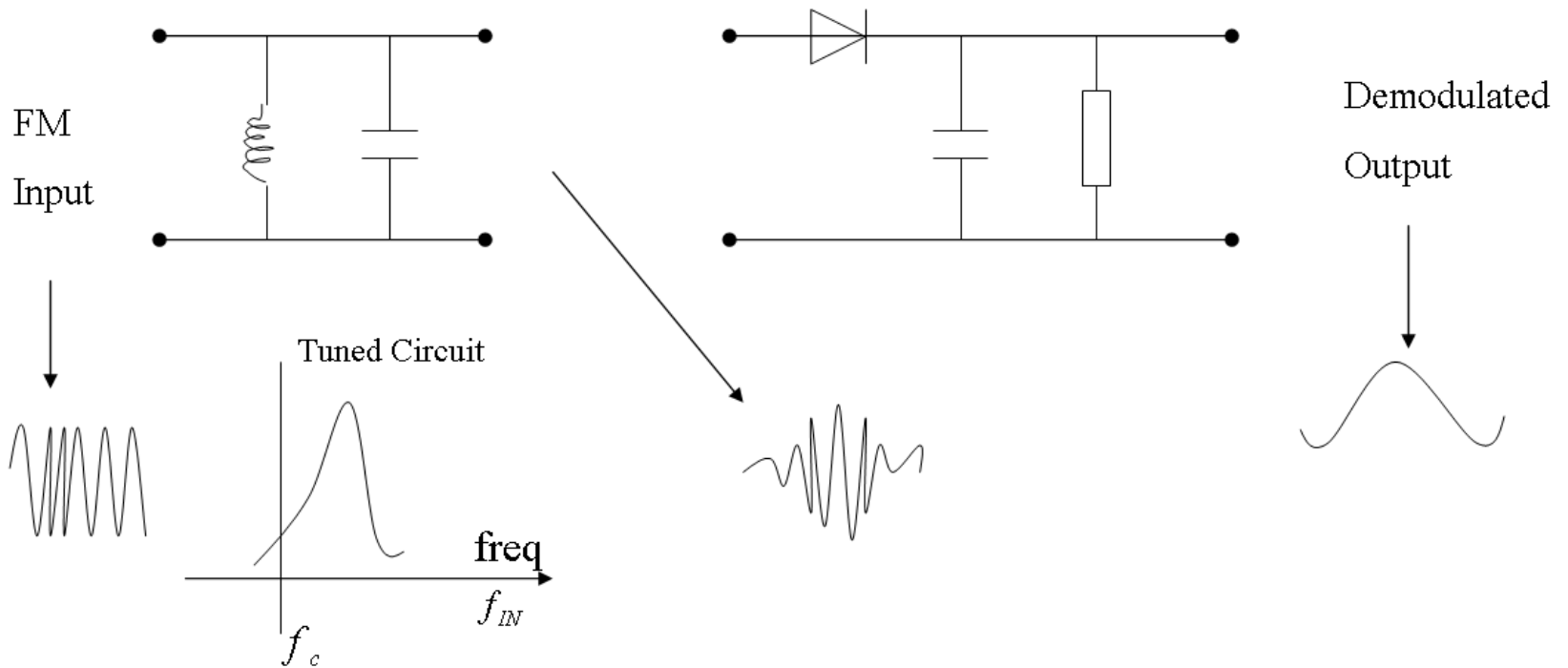
$$V_{OUT} = V_0 + K\alpha m(t)$$

FM Demodulation – General Principles.



Methods

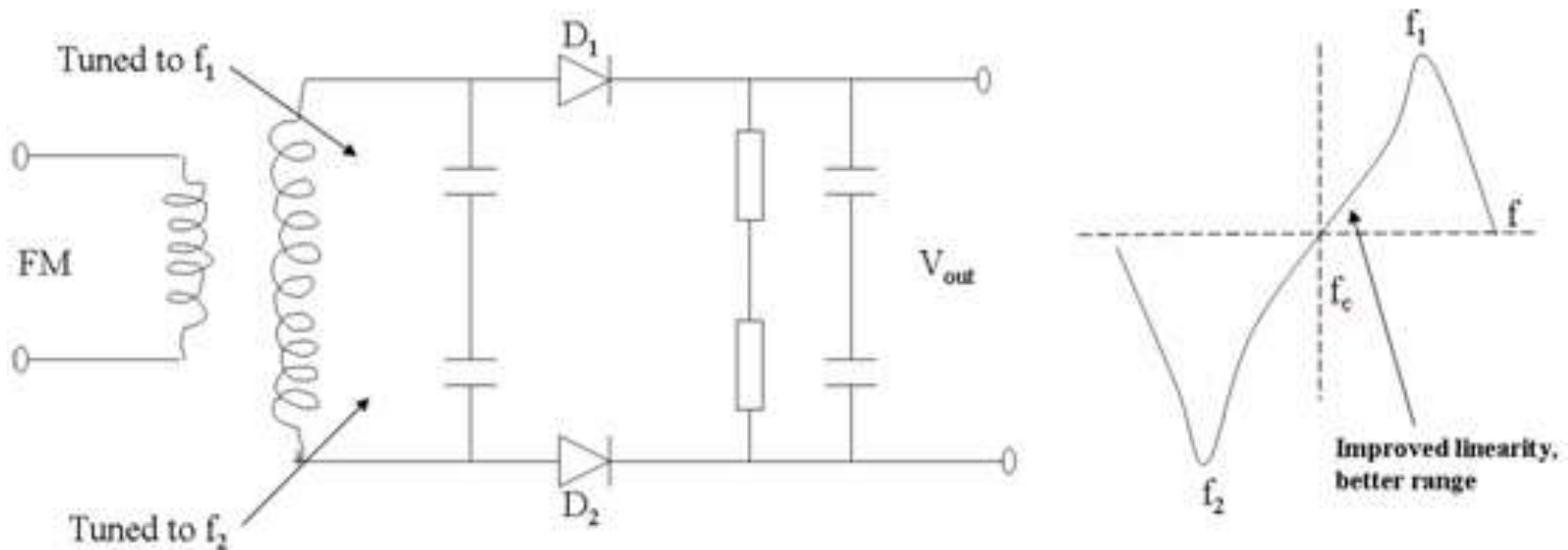
Tuned Circuit – One method (used in the early days of FM) is to use the slope of a tuned circuit in conjunction with an envelope detector.



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 = $2fm$, *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**

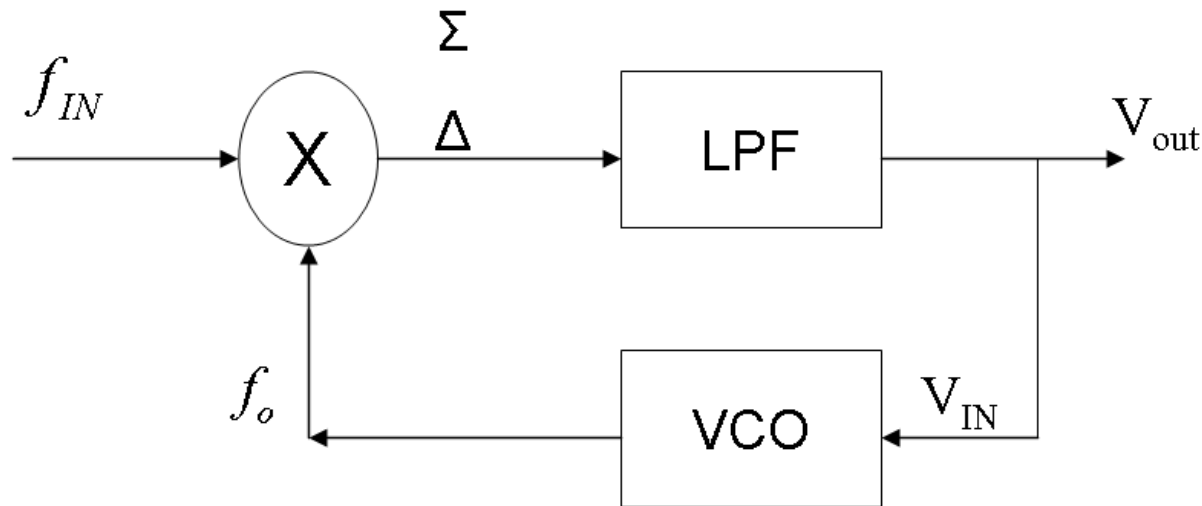
Foster-Seeley Discriminator



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 V_{OUT} 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 $m(t)$.

- Armstrong's Indirect FM

Outline

Two methods of generating FM waves:

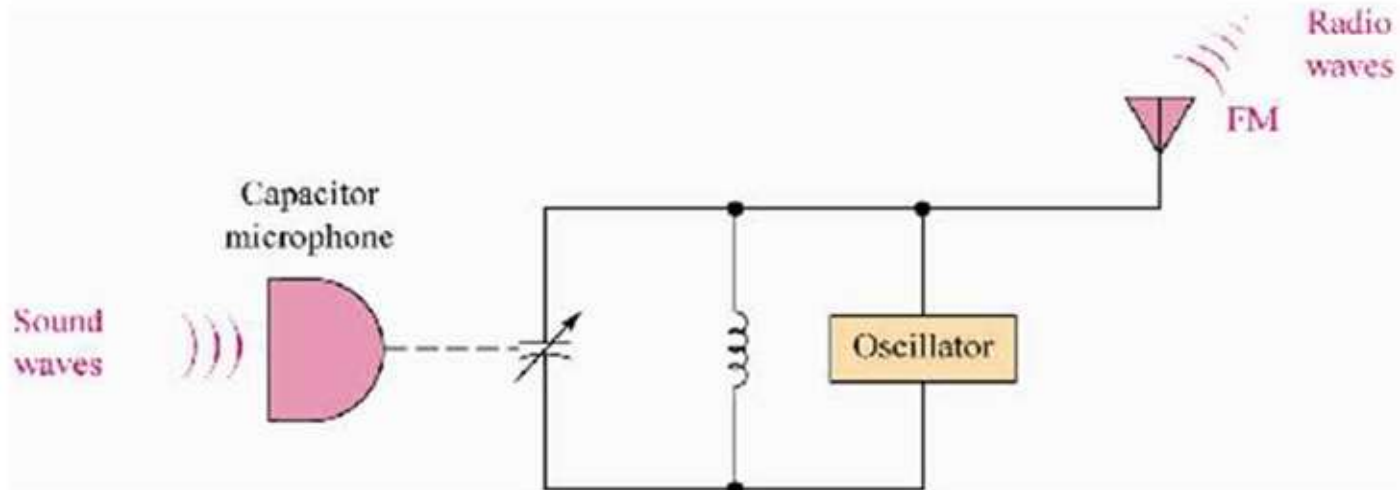
Direct method

Indirect Method

Armstrong's FM generator

Direct FM Generation

The carrier frequency is directly varied by the input signal. This can be accomplished by a Voltage-Controlled Oscillator (VCO), whose output frequency is proportional to the voltage of the input signal. A VCO example: implemented by a variable capacitor.



Problems of direct FM generator

The carrier freq of VCO tends to **drift** away.

(Crystal oscillator cannot be used in direct FM: its freq is too stable, and is difficult to change.)

Feedback freq stabilization circuit is required: The complexity is increased.

The **maximum freq deviation** with direct FM is around 5 KHz, too small for wideband FM:

Recall: the maximum freq deviation in commercial FM radio is 75kHz.

Indirect FM Generation

First obtain NBFM via a **NBPM** circuit with crystal oscillator

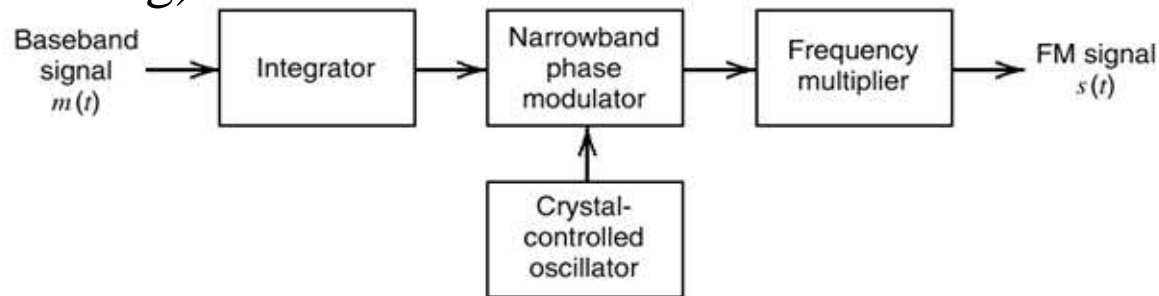
Change the phase of crystal oscillator is easier than changing its freq.

Then apply frequency **multiplier**

Increase both the carrier frequency and the freq deviation If necessary, use **mixer** to concatenate multiple multipliers

Change the carrier frequency, but not the deviation.

Indirect FM is preferred when the stability of carrier frequency is of major concern (e.g., in commercial FM broadcasting)

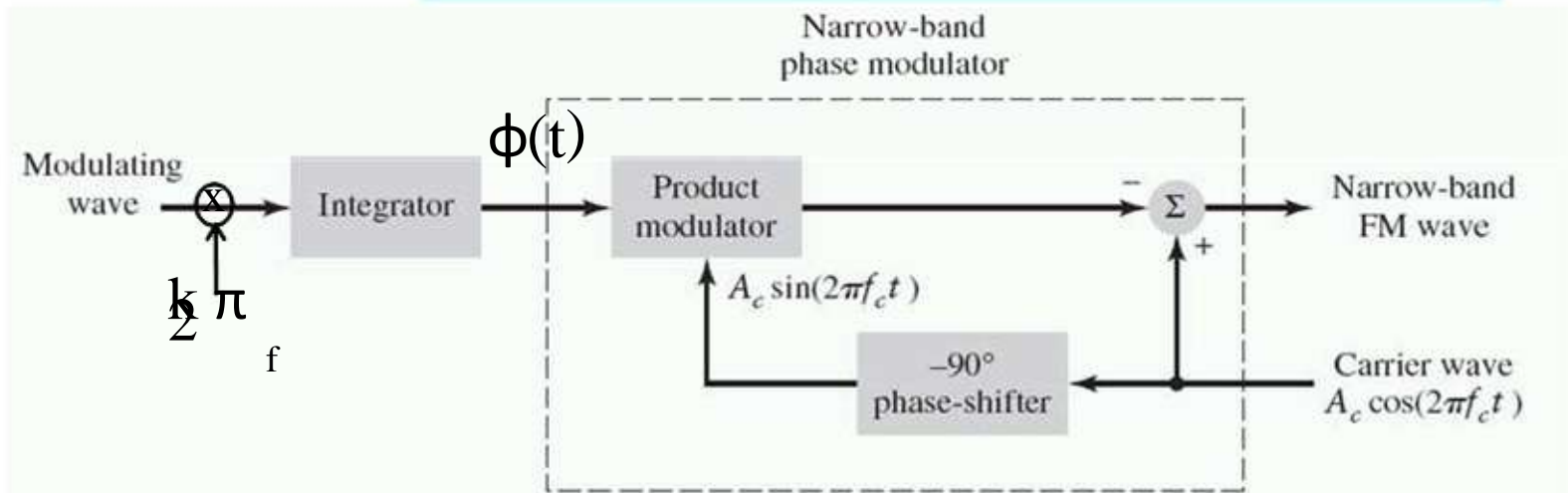


Recall: Narrow-band FM

if k_f is small: $s(t) = A_c \cos(2\pi f_c t + \phi(t))$



$$s(t) \approx A_c \cos(2\pi f_c t) - A_c \phi(t) \sin(2\pi f_c t)$$



Crystal oscillator can be used to get stable frequency (prevent drifting)

But frequency deviation of NBFM is small.

To get larger one, use freq multiplier...

Frequency Multipliers

How to increase the frequency deviation ?

Answer: **trigonometric identity!**

$$s(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int_0^t m(\tau) d\tau\right)$$

If we can get the squared signal:

$$s^2(t) = A_c^2 \cos^2\left(2\pi f_c t + 2\pi k_f \int_0^t m(\tau) d\tau\right)$$

=

Can use bandpass filter to get high freq component:

Both the **freq** and the **freq deviation** will be **doubled!**7

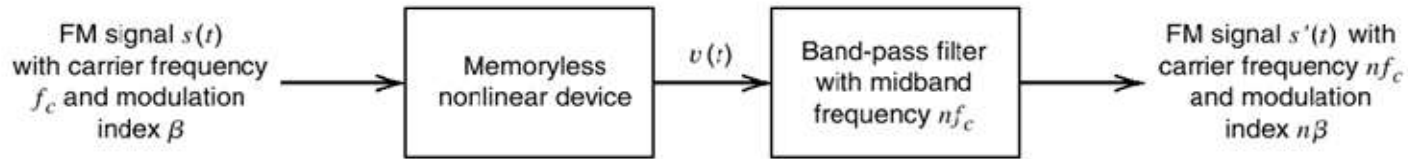
Frequency Multipliers

If we can get $s^3(t)$:

$$s^3(t) = s^2(t)s(t)$$

Can use bandpass filter to get the high freq component:
Both the **freq** and the **freq deviation** will be **tripled!**

Freq Multipliers via Nonlinear Circuit



A general **nonlinear circuit** produces

$$v(t) = a_1 s(t) + a_2 s^2(t) + \dots + a_n s^n(t)$$

The highest frequency and the freq sensitivity factor:
 $n f_c$ and $n k_f$, respectively.

The bandpass filter:

Center: $n f_c$.

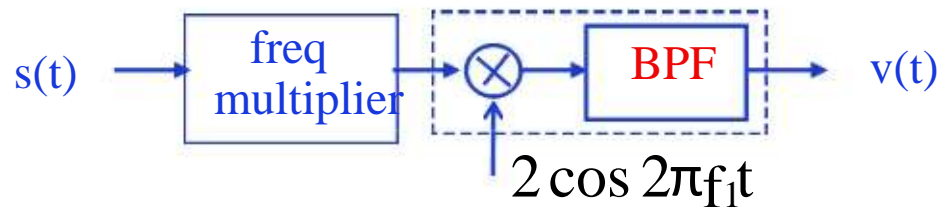
Passband width: determined by **Carson's rule**:

$2(f+W)$ In practice: $n = 2$, or 3 . Larger n is not

efficient.
 Can concatenate multiple stages to obtain higher orders.

Mixer & Frequency Multiplier

Frequency multiplier increases the freq and deviation together: How to adjust them individually to get more flexibilities? Use **mixer** after frequency multiplier.



Input: $s(t) = A_c \cos(2\pi f_c t + \phi(t))$

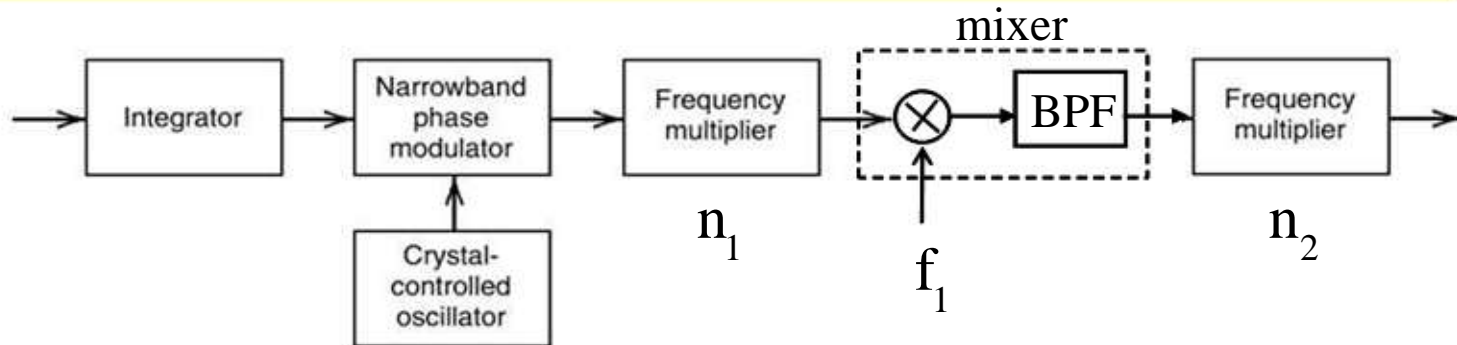
After freq multiplier:

After multiplying with local freq f_1 :

Only freq is changed! Freq deviation is untouched.

After BPF: only one freq component is kept.

Armstrong's Indirect FM

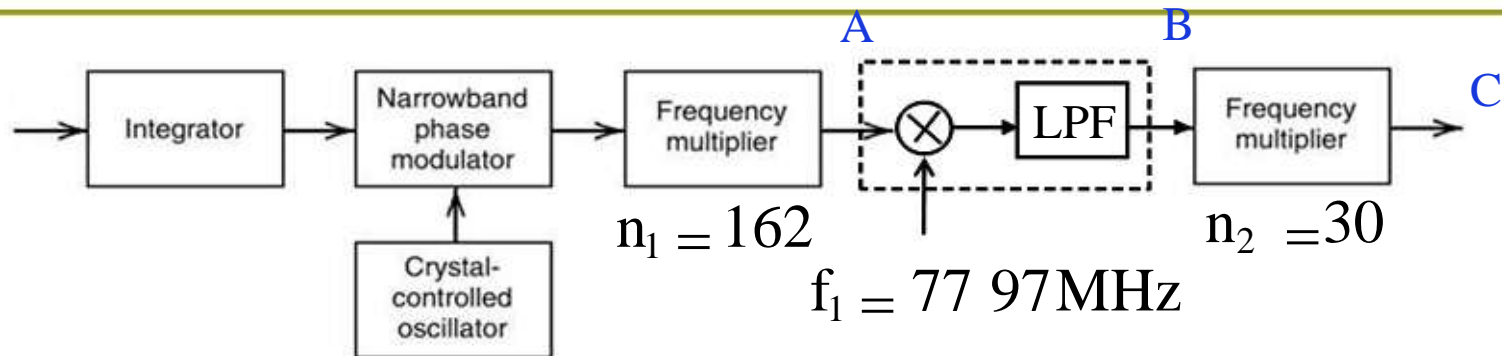


Two stages of multiplier and mixer give flexibility between freq and deviation:

The first stage multiplier amplifies both f_c and $9f$.

The mixer brings down the central freq. The second stage amplifies f_c and $9f$ again.

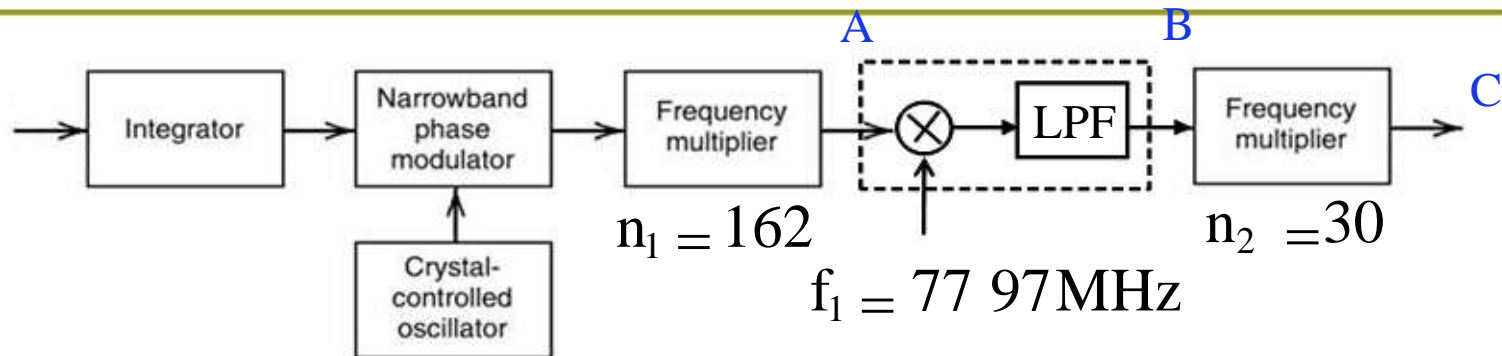
Example



NBPM output : $f = 500 \text{ kHz}$, $f = 15.432 \text{ Hz}$

Find f and f at A, B, C. (assume lowside tuning)

Example



NBPM output : $f = 500\text{kHz}$, $f = 15\,432\text{Hz}$

Point C:

Total multiplier for f :

This is the same as

The mixer does not affect the freq deviation.

Summary

Direct FM generation:

The carrier freq is directly varied by the input signal Frequency drifting is a problem

Max freq deviation: 5KHz

Indirect FM generation:

NBFM followed by freq multiplier Use nonlinear circuit to get multiplier Can

use mixer to change the carrier freq

Combination of mixer and multiplier provides flexibilities.

AM vs. FM

- AM requires a simple circuit, and is very easy to generate.
 - It is simple to tune, and is used in almost all short wave broadcasting.
 - The area of coverage of AM is greater than FM (longer wavelengths (lower frequencies) are utilized-remember property of HF waves?)
 - However, it is quite inefficient, and is susceptible to static and other forms of electrical noise.
-
- The main advantage of FM is its audio quality and immunity to noise. Most forms of static and electrical noise are naturally AM, and an FM receiver will not respond to AM signals.
 - The audio quality of a FM signal increases as the frequency deviation increases (deviation from the center frequency), which is why FM broadcast stations use such large deviation.
 - The main disadvantage of FM is the larger bandwidth it requires