6. Analog data → Analog signal : In general modulation is the process of combining input signal $m(t)$ and carrier signal (at f_c) to produce a signal $s(t)$ whose bandwidth is centered on f_c.

For digital data, the motivation is clear: when only analog transmission facilities are available, modulation is required to convert the digital data to analog form. The motivation when the data are already analog is less clean. After all, voice signals are transmitted over telephone lines at their original spectrum (baseband transmission). The principle reasons for analog modulation of analog signals are –

1. To achieve baseband transmission higher frequency may be needed for effective transmission. For unguided transmission, it is almost impossible to transmit baseband signal as the required antennas may have to be several kilometers in diameter.

2. Modulation permits frequency division multiplexing. modulation allows us to send a signal over a bandpass frequency range. If every signal gets its own frequency range, then we can transmit multiple signals simultaneously over a single channel, all using different frequency range.

The principle techniques for modulation using analog data are – Amplitude modulation (AM), Frequency modulation (FM) and phase modulation (PM).

* Bandpass is a range of frequencies which are transmitted through a bandpass filter which is a filter allowing specific frequencies to pass preventing signals at unwanted frequencies.

Amplitude modulation (AM) :– this is the simplest form of modulation and can be expressed as –

$$s(t) = [1 + n_a x(t)] \cos(2\pi f_c t).$$

~~that input signal $m(t) = n_a x(t)$ where n_a is known as modulation index and is the ratio of amplitude of the input signal to the carrier.~~

where $\cos(2\pi f_c t)$ is the carrier signal and $x(t)$ is the input signal both normalized to unity amplitude.

parameter n_a is known as modulation index and defined as the ratio of amplitude of the input signal to the carrier signal. the '1' in the equation is a dc component.

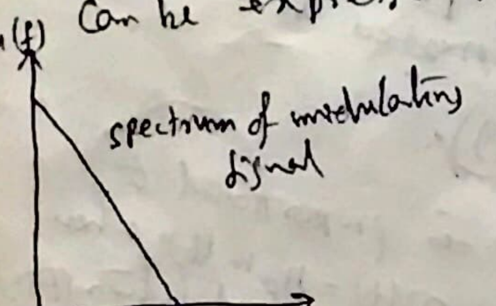
Consider an example — derive expression for $s(t)$ if $x(t)$ is the amplitude modulating signal $\cos(2\pi f_m t)$.

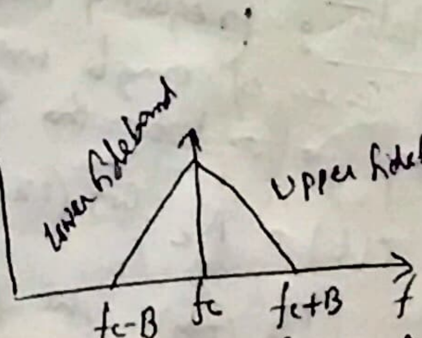
we have $s(t) = [1 + n_a \cos(2\pi f_m t)] \cos(2\pi f_c t)$.

$$= \cos(2\pi f_c t) + \frac{n_a}{2} \cos(2\pi (f_c - f_m) t) + \frac{n_a}{2} \cos(2\pi (f_c + f_m) t)$$

So, the resulting signal has a component at the original carrier frequency plus a pair of components each spaced f_m from the carrier. This is known as double sideband transmitted carrier (DSBTC).

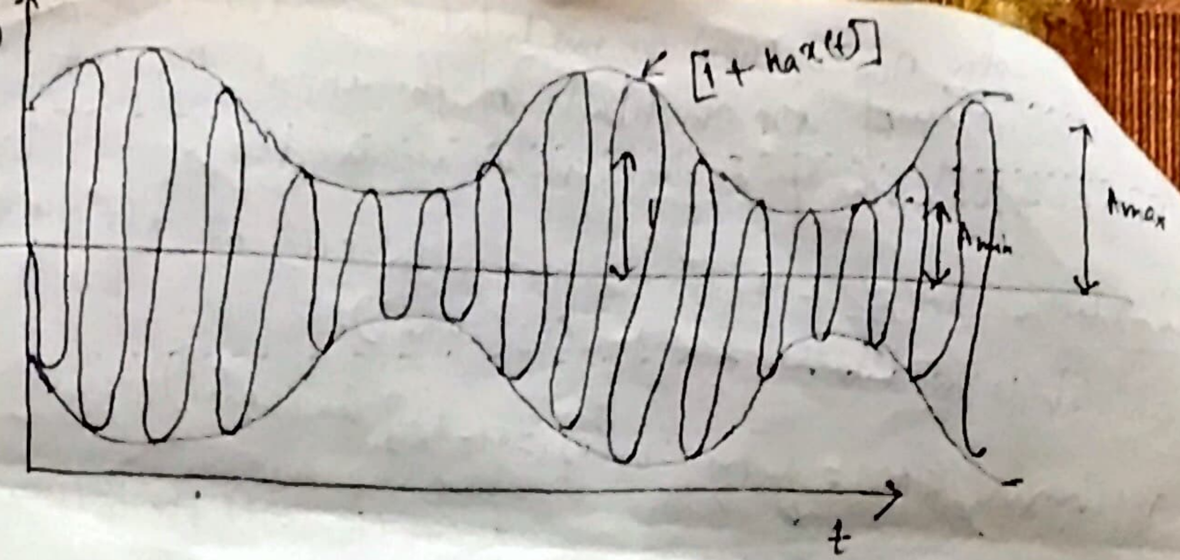
Basically, AM involves the multiplication of input signal by the carrier. The envelope of the resulting signal is then $[1 + n_a \cdot x(t)]$. Here, n_a should be < 1 to reproduce the original signal exactly. If $n_a > 1$, the envelope will cross the time axis and information is lost. The spectrum of resulting signal $S(f)$ can be expressed as below:





The spectrum of AM signal with carrier at f_c.

The portion of spectrum for $|f| > |f_c|$ is the upper sideband and the portion of spectrum for $|f| < |f_c|$ is the lower sideband. Both are replicas of the original spectrum $M(f)$, with the lower sideband being frequency reversed. The resulting signals will be of the following form if $x(t)$ is the sinusoidal wave.



Some variations of AM are — Single sideband (SSB) that sends only one sideband eliminating the other sideband and carrier. The principle advantage is that it requires half the bandwidth of the DSBTC. Also, lower power is required as no power is used to transmit the carrier or the other sideband. Another variation is double sideband suppressed carrier (DSBSC), which filters out the carrier frequency but sends both sidebands. This saves power but ~~a~~ uses as much bandwidth as DSBTC. ~~But~~ the disadvantage of suppressing carrier is that constant ~~for~~ carrier can be used for synchronization purpose.

Bandwidth of AM: AM creates a bandwidth twice the bandwidth of modulating signal and covers a range centered on f_c.

<u>Angle Modulation</u>: Frequency modulation and phase modulation are special cases of angle modulation. The modulated signal ~~is proportional to the~~ can be expressed as —

$$s(t) = A_c \cos(2\pi f_c t + \varphi(t)).$$

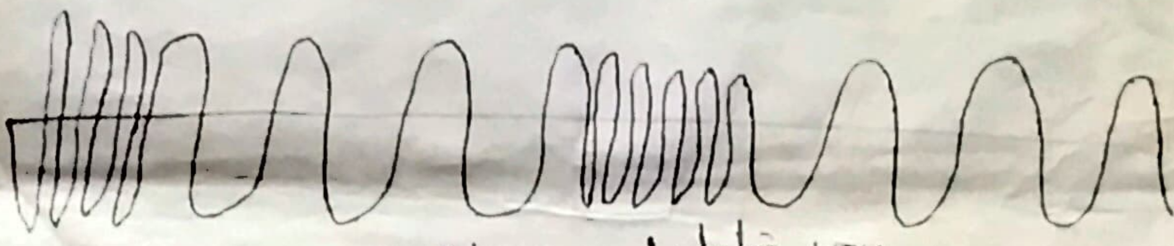
For phase modulation (PM), the phase $\varphi(t)$ is proportional to the modulating signal $m(t)$. Thus $\varphi(t) = n_p m(t)$ where n_p is the phase modulation index. For frequency modulation (FM), the derivative of phase $\left(\frac{d}{dt}\varphi(t) = \varphi'(t)\right)$ is proportional to the modulating signal $m(t)$. Thus, $\varphi'(t) = n_f m(t)$ where n_f is the frequency modulation index. [The phase of $s(t)$ at any instant is $[2\pi f_c t + \varphi(t)]$. In PM, this $\varphi(t)$ is proportional to $m(t)$. Since frequency can be defined as the rate of change of phase of a signal, the instantaneous frequency of $s(t)$

be expressed as $2\pi f_i(t) = \frac{d}{dt}[2\pi f_c t + \varphi(t)]$

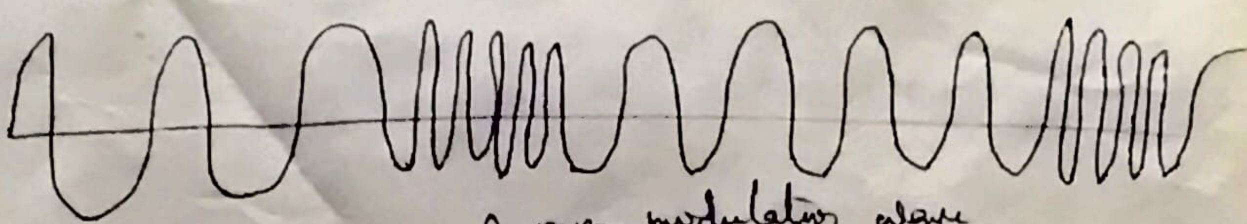
$f_i(t) = f_c + \frac{1}{2\pi}\varphi'(t)$ Here $\varphi'(t)$ is proportional to $m(t)$ for FM.

$\Rightarrow f_c + \frac{1}{2\pi} n_f m(t)$

peak deviation ΔF $\Rightarrow \Delta F = \frac{1}{2\pi} n_f . A_m$, when A_m is maximum value of $m(t)$. Thus, increase of magnitude of $m(t)$ will increase ΔF which in turn increase bandwidth B_T. But it will not increase the average power level. ~~Here~~, In AM, the modulation affects the power in AM signal but does not affect its bandwidth. But in all cases (AM, FM, PM) result in a signal whose bandwidth is centred at f_c. Though the magnitude of bandwidth is different for all these cases.



phase modulation wave



frequency modulation wave

the shapes of PM and ~~AM signals are very~~ similar.