List of acronyms:

- AM = Amplitude Modulation
- PM = Phase Modulation
- FM = Frequency Modulation
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- LPF = Low Pass Filter
- CDMA = Code Division Multiplexing Access
- FDMA = Frequency Division Multiplexing Access
- PLL = Phase Locked Loop
- ...SK = ... Shift Keying
  - A... = Amplitude ...
  - BA... = Binary Amplitude ...
  - P... = Phase...
  - BP... = Binary Phase ...
  - F... = Frequency...
- PCM Pulse code modulation

BASK:

$$s(t) = \begin{cases} \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t + \phi_c), & \text{symbol '1'} \\ 0, & \text{symbol '0'} \end{cases}$$

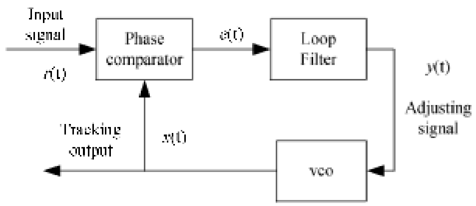
BPSK:

$$s(t) = \begin{cases} \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t) & \text{symbol '1'} \\ -\sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t) & \text{symbol '0'} \end{cases}$$

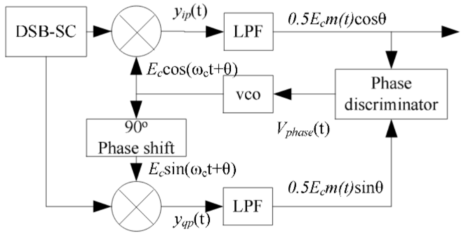
FSK:

$$s(t) = \begin{cases} \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_1 t) & \text{symbol '1'} \\ \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_2 t) & \text{symbol '0'} \end{cases}$$

Phase Locked Loops: Consist of 3 parts, low pass filter, a guessing part and an adjusting signal.



The guessing part is the VCO, which has the output:
$x(t) = \sin\left[\omega_F + K_v \int y(t)dt + \phi_0\right]$ The phase comparitor is made using a multiplier, XOR gate, etc. And the low pass filter is generally of the first order.
Input: $v_i(A_c \cos(\omega_c t + \phi_i))$ and VCO output: $v_o(t) = sin\left[\omega_F t + K_v \int v_c(\tau)d\tau + \phi_o\right]$ let, $\omega_o(t) = \omega_F t + K_v \int v_c(\tau)d\tau$ So the ouput of the phase comparitor (before the LPF) is $p(t) = \frac{A_c}{2}\{\sin((\omega_c t + \omega_o(t)) + (\phi_i + \phi_o)) + \sin((\omega_c t - \omega_o(t)) + (\phi_i - \phi_o))\}$
Using the low pass filter to remove the added components. It's output becomes, $v_c(t) = \frac{A_c K_0}{2} \sin((\omega_c - \omega_F)t + (\phi_i - \phi_o))$
In steady state, $v_c(t) = 2\frac{\omega_c - \omega_F}{K_v} = \frac{A_c K_0}{2} \sin(\phi_i - \phi_o)$ so the locking range is $-1 \leq 2\frac{\omega_c - \omega_F}{K_v A_c K_0} \leq 1$. Capture range less than the locking range according to the phase characteristics of the low pass filter.



Sampling:
$f_s \geq 2B$ ie the sampling rate needs to be at least double the highest frequency present in the input signal.
Pulse Code Modulation::
Midtread, where the value is in the middle of the sample time. **Midrise**, where the value is at the start of the sample time.
PCM and SNR: $E\left[q^2(t)\right] = \lim_{T\to\infty} \frac{1}{2BT} \sum_k q^2(kT_s)$ where q is the quantisation error. In an M-ary system this reduces to: $E\left[q^2(t)\right] = \frac{m_p^2}{3L^2}$ resulting in the signal to noise ratio, $\frac{S_0}{N_0} = 3L^2 \frac{E\left[m^2(t)\right]}{m_p^2}$