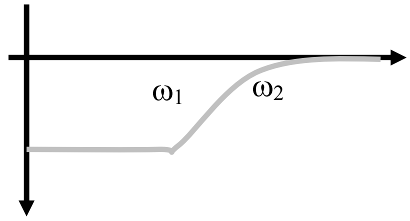
EEE 317 Tutorial answers – AM/FM modulation & Digital vs. Analogue

(1) $$\frac{S_o}{N_o} = \frac{3\pi\alpha^2\Delta\omega^2}{2\eta\omega_m{}^3}$$

(2)



Pre-emphasis filter characteristic

$\omega_2 > \omega_m$

(3) A DSB SC system will always have superior signal to noise ratio at the demodulator output compared to a DSB LC system. This is purely because a DSB LC system 'wastes' some signal power to transmit the carrier which carriers no message information. *However, we should remember that DSB LC systems provide a carrier which helps to demodulate the signal. So although some power is expended sending something other than the message, it makes the carrier recovery at the receiver far simpler.*

(4) $\omega_m = \Delta\omega/\beta$ = 75kHz/5 = 15kHz.

$$\frac{S_o}{N_o} = \frac{3\pi\alpha^2\Delta\omega^2}{2\eta\omega_m{}^3} = \frac{3\pi(0.1)^2\left(75\times10^3\right)^2}{2\left(1\times10^{-6}\right)\left(15\times10^3\right)^3} = 78.5$$

(5) The signal to noise ratio in an FM system is as follows $\dfrac{S_o}{N_o} = \dfrac{3\pi\alpha^2\Delta\omega^2}{2\eta\omega_m{}^3}$

Hence we can see that the signal to noise ratio will reduce with increasing message frequency ω_m. Most of the frequency content in an audio signal, that which is often transmitted by FM, is at low frequencies and very little occurs at these high frequencies that degrades the signal to noise ratio by so much. The pre-emphasis filter artificially increases the importance of the high frequencies at transmission. Noise is added during transmission, but the importance of the noise is reduced at the receiver by the use of the de-emphasis filter – thus enhancing the signal to noise ratio.

(6) $s_i(t) = g(t)\cos(\omega_c t) + \hat{g}(t)\sin(\omega_c t)$.

First calculate the input signal power,

$$S_i = \frac{1}{T}\int_{-T/2}^{T/2} s_i(t)^2\,dt = \frac{1}{2}\overline{g(t)^2} + \frac{1}{2}\overline{\hat{g}(t)^2} = \overline{g(t)^2}\,.$$

now calculate the output waveform,

$$s_o(t) = s_i(t)\times\cos(\omega_c t)$$

$$s_o(t) = \frac{1}{2}g(t)\left(1+\cos(2\omega_c t)\right) + \frac{1}{2}g(t)\left(\sin(2\omega_c t)+\sin(0)\right)$$'

which, after low pass filtering gives $s_o(t) = \frac{1}{2} g(t)$.

The corresponding output signal power can then be calculated

$$S_o = \frac{1}{T} \int_{-T/2}^{T/2} s_o(t)^2 \, dt = \frac{1}{4} \overline{g(t)^2} \,.$$

The noise power at the input is $N_i = \overline{n_i(t)^2}$ by definition

The noise at the output is given by,

$$n_o(t) = n_i(t) \times \cos(\omega_c t)$$

$$n_o(t) = \frac{1}{2} n_c(t)(1 + \cos(2\omega_c t)) + \frac{1}{2} n_s(t) \sin(2\omega_c t)$$

which, after low pass filtering gives, $n_o(t) = \frac{1}{2} n_c(t)$

The associated noise power is,

$$N_o = \frac{1}{T} \int_{-T/2}^{T/2} n_o(t)^2 \, dt = \frac{1}{T} \int_{-T/2}^{T/2} \left(\frac{n_c(t)}{2} \right)^2 dt = \frac{1}{4} \overline{n_c(t)^2} = \frac{1}{4} \overline{n_i(t)^2} \,.$$

The last step assumes that the noise is random.

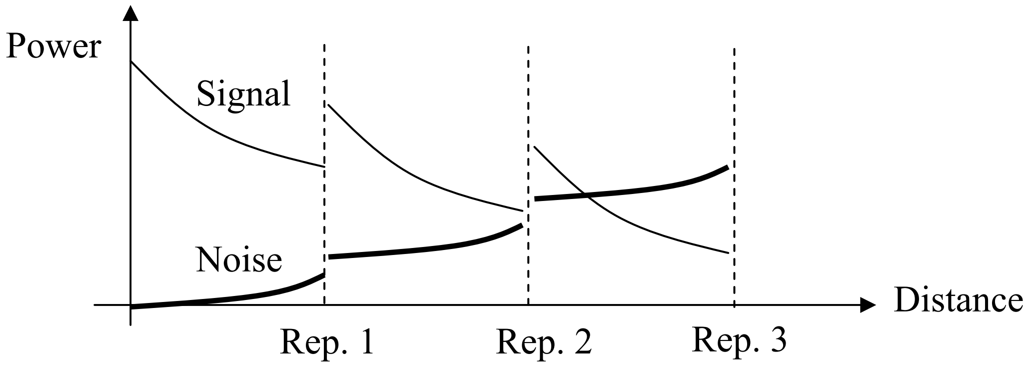
Therefore we can write the signal to noise ratios at the input and output of the demodulator.

$$\frac{S_o}{N_o} = \frac{1/4}{1/4} \frac{\overline{g(t)^2}}{\overline{n_i(t)^2}} \,, \quad \frac{S_i}{N_i} = \frac{\overline{g(t)^2}}{\overline{n_i(t)^2}} \,.$$

$$\therefore \frac{S_o}{N_o} = \frac{S_i}{N_i}$$

7) Bit stuffing is the process of sending redundant data through a data link in order when the transmitter has no useful data to transmit so that the receiver does not lose synchronisation for subsequent data packets. Often encountered in TDMA systems

8)



And for digital,