Chapter 6. Effect of Noise on Analog Communication Systems

Essentials of Communication Systems
Engineering

Introduction

- Angle modulation systems and FM can provide a high degree of noise immunity
- This noise immunity is obtained at the price of sacrificing channel bandwidth
- Bandwidth requirements of angle modulation systems are considerably higher than that of amplitude modulation systems
- This chapter deals with the followings:
 - Effect of noise on amplitude modulation systems
 - Effect of noise on angle modulation systems
 - Carrier-phase estimation using a phase-locked loop (PLL)
 - Analyze the effects of transmission loss and noise on analog communication systems

EFFECT OF NOISE ON AMPLITUDE-MODULATION SYSTEMS

- Effect of Noise on a Baseband System
- Effect of Noise on DSB-SC AM
- Effect of Noise on SSB-AM
- Effect of Noise on Conventional AM

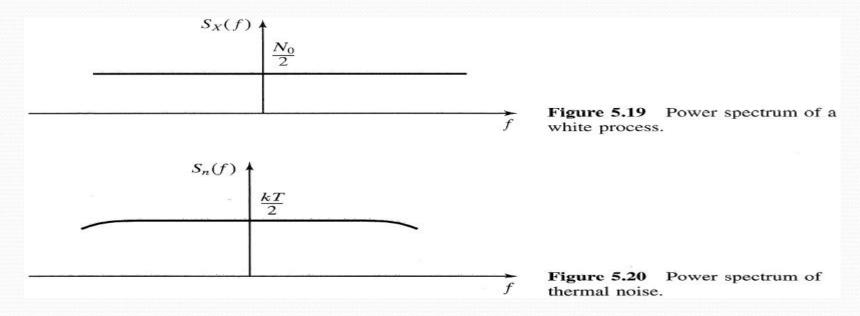
Effect of Noise on a Baseband System

- Since baseband systems serve as a basis for comparison of various modulation systems, we begin with a noise analysis of a baseband system.
- In this case, there is no carrier demodulation to be performed.
- The receiver consists only of an ideal lowpass filter with the bandwidth *W*.
- The noise power at the output of the receiver, for a white noise input, is $P_{n_0} = \int_{-W}^{W} \frac{N_0}{2} df = N_0 W$
- If we denote the received power by P_R , the baseband SNR is given by

$$\left(\frac{S}{N}\right)_b = \frac{P_R}{N_0 W} \qquad (6.1.2)$$

White process (Section 5.3.2)

- White process is processes in which all frequency components appear with equal power, i.e., the power spectral density (PSD), $S_x(f)$, is a constant for all frequencies.
- the PSD of thermal noise, $S_n(f)$, is usually given a $S_n(f) = \frac{kT}{2}$ (where k is Boltzmann's constant and T is the temperature)
- The value kT is usually denoted by N_{o_i} Then $S_n(f) = \frac{N_0}{2}$



- Transmitted signal : $u(t) = A_c m(t) \cos(2f f_c t)$
- The received signal at the output of the receiver noiselimiting filter: Sum of this signal and filtered noise
- Recall from Section 5.3.3 and 2.7 that a filtered noise process can be expressed in terms of its in-phase and quadrature components as

$$n(t) = A(t)\cos[2ff_c t + _{"}(t)] = A(t)\cos_{"}(t)\cos(2ff_c t) - A(t)\sin_{"}(t)\sin(2ff_c t)$$
$$= n_c(t)\cos(2ff_c t) - n_s(t)\sin(2ff_c t)$$

(where $n_c(t)$ is in-phase component and $n_s(t)$ is quadrature component)

 Received signal (Adding the filtered noise to the modulated signal)

$$r(t) = u(t) + n(t)$$

$$= A_c m(t) \cos(2f f_c t) + n_c(t) \cos(2f f_c t) - n_s(t) \sin(2f f_c t)$$

- Demodulate the received signal by first multiplying r(t) by a locally generated sinusoid $cos(2\pi f_c t + \phi)$, where ϕ is the phase of the sinusoid.
- Then passing the product signal through an ideal lowpass filter having a bandwidth *W*.

• The multiplication of r(t) with $\cos(2\pi f ct + \phi)$ yields $r(t)\cos(2f f_c t + w)$

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 = u(t)\cos(2f f_c t + W) + n(t)\cos(2f f_c t + W) 
 = A_c m(t)\cos(2f f_c t)\cos(2f f_c t + W) 
 + n_c(t)\cos(2f f_c t)\cos(2f f_c t + W) - n_s(t)\sin(2f f_c t)\cos(2f f_c t + W) 
 = \frac{1}{2}A_c m(t)\cos(W) + \frac{1}{2}A_c m(t)\cos(4f f_c t + W) 
 + \frac{1}{2}[n_c(t)\cos(W) + n_s(t)\sin(W)] + \frac{1}{2}[n_c(t)\cos(4f f_c t + W) - n_s(t)\sin(4f f_c t + W)]
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• The lowpass filter rejects the double frequency components and passes only the lowpass components.

 $y(t) = \frac{1}{2} \dot{A_c} m(t) \cos(\mathbf{W}) + \frac{1}{2} \left[n_c(t) \cos(\mathbf{W}) + n_s(t) \sin(\mathbf{W}) \right]$

- In Chapter 3, the effect of a phase difference between the received carrier and a locally generated carrier at the receiver is a drop equal to $cos^2(W)$ in the received signal power.
- Phase-locked loop (Section 6.4)
 - The effect of a phase-locked loop is to generate phase of the received carrier at the receiver.
 - If a phase-locked loop is employed, then ϕ = 0 and the demodulator is called a coherent or synchronous demodulator.
- In our analysis in this section, we assume that we are employing a coherent demodulator.
 - With this assumption, we assume that $\phi = 0$

$$y(t) = \frac{1}{2} \left[A_c m(t) + n_c(t) \right]$$

• Therefore, at the receiver output, the message signal and the noise components are additive and we are able to define a meaningful SNR. The message signal power is given by

$$P_o = \frac{A_c^2}{4} P_M$$

- power P_M is the content of the message signal
- The noise power is given by

$$P_{n_0} = \frac{1}{4} P_{n_c} = \frac{1}{4} P_n$$

• The power content of n(t) can be found by noting that it is the result of passing $n_w(t)$ through a filter with bandwidth B_c .

• Therefore, the power spectral density of n(t) is given by

$$S_n(f) = \begin{cases} \frac{N_0}{2} & |f - f_c| < W \\ 0 & otherwise \end{cases}$$

The noise power is

$$P_n = \int_{-\infty}^{\infty} S_n(f) df = \frac{N_0}{2} \times 4W = 2WN_0$$

Now we can find the output SNR as

$$\left(\frac{S}{N}\right)_0 = \frac{P_0}{P_{n_0}} = \frac{\frac{A_c^2}{4}P_M}{\frac{1}{4}2WN_0} = \frac{A_c^2P_M}{2WN_0}$$

• In this case, the received signal power, given by Eq. (3.2.2), is

$$P_R = A_c^2 P_M / 2$$
.

The output SNR for DSB-SC AM may be expressed as

$$\left(\frac{S}{N}\right)_{0_{DSR}} = \frac{P_R}{N_0 W}$$

- which is identical to baseband SNR which is given by Equation (6.1.2).
- In DSB-SC AM, the output SNR is the same as the SNR for a baseband system
 - ⇒ DSB-SC AM does not provide any SNR improvement over a simple baseband communication system

SSB modulated signal :

$$u(t) = A_c m(t) \cos(2f f_c t) \mp A_c \hat{m}(t) \sin(2f f_c t)$$

Input to the demodulator

$$\begin{split} r(t) &= A_c m(t) \cos(2f \ f_c t) \mp A_c \hat{m}(t) \sin(2f \ f_c t) + n(t) \\ &= A_c m(t) \cos(2f \ f_c t) \mp A_c \hat{m}(t) \sin(2f \ f_c t) + n_c(t) \cos(2f \ f_c t) - n_s(t) \sin(2f \ f_c t) \\ &= \left[A_c m(t) + n_c(t) \right] \cos(2f \ f_c t) + \left[\mp A_c \hat{m}(t) - n_s(t) \right] \sin(2f \ f_c t) \end{split}$$

- Assumption: Demodulation with an ideal phase reference.
- Hence, the output of the lowpass filter is the in-phase component (with a coefficient of ½) of the preceding signal.

$$y(t) = \frac{1}{2} \left[A_c m(t) + n_c(t) \right]$$

• Parallel to our discussion of DSB, we have

$$P_{n_0} = \frac{1}{4} P_{n_c} = \frac{1}{4} P_n$$

$$P_n = \int_{-\infty}^{\infty} S_n(f) df = \frac{N_0}{2} \times 2W = WN_0$$

$$P_R = P_U = A_c^2 P_M$$

$$P_0 = \frac{A_c^2}{4} P_M$$

$$P_0 = \frac{A_c^2}{4} P_M$$

$$P_0 = \frac{P_0}{P_{n_0}} = \frac{A_c^2 P_M}{WN_0}$$

• The signal-to-noise ratio in an SSB system is equivalent to that of a DSB system.

- DSB AM signal : $u(t) = A_c[1 + am_n(t)]\cos(2f f_c t)$
- Received signal at the input to the demodulator

$$\begin{split} r(t) &= A_c [1 + a m_n(t)] \cos(2f \ f_c t) + n(t) \\ &= A_c [1 + a m_n(t)] \cos(2f \ f_c t) + n_c(t) \cos(2f \ f_c t) - n_s(t) \sin(2f \ f_c t) \\ &= \left[A_c [1 + a m_n(t)] + n_c(t) \right] \cos(2f \ f_c t) - n_s(t) \sin(2f \ f_c t) \end{split}$$

- *a* is the modulation index
- $m_n(t)$ is normalized so that its minimum value is -1
- If a synchronous demodulator is employed, the situation is basically similar to the DSB case, except that we have $1 + am_n(t)$ instead of m(t).
- After mixing and lowpass filtering

$$y(t) = \frac{1}{2} \left[A_c a m_n(t) + n_c(t) \right]$$

Received signal power

$$P_{R} = \frac{A_{c}^{2}}{2} \left[1 + a^{2} P_{M_{n}} \right]$$

- Assumed that the message process is zero mean.
- Now we can derive the output SNR as

$$\left(\frac{S}{N}\right)_{0_{AM}} = \frac{\frac{1}{4}A_{c}^{2}a^{2}P_{M_{n}}}{\frac{1}{4}P_{n_{c}}} = \frac{A_{c}^{2}a^{2}P_{M_{n}}}{2N_{0}W} = \frac{a^{2}P_{M_{n}}}{1+a^{2}P_{M_{n}}} \frac{\frac{A_{c}^{2}}{2}\left[1+a^{2}P_{M_{n}}\right]}{N_{0}W}$$

$$= \frac{a^{2}P_{M_{n}}}{1+a^{2}P_{M_{n}}} \frac{P_{R}}{N_{0}W} = \frac{a^{2}P_{M_{n}}}{1+a^{2}P_{M_{n}}} \left(\frac{S}{N}\right)_{b} = y\left(\frac{S}{N}\right)_{b}$$

- y denotes the modulation efficiency
- Since $a^2 P_{M_n} < 1 + a^2 P_{M_n}$, the SNR in conventional AM is always smaller than the SNR in a baseband system.

- In practical applications, the modulation index a is in the range of 0.8-0.9.
- Power content of the normalized message process depends on the message source.
- Speech signals : Large dynamic range, P_M is about 0.1.
 - The overall loss in SNR, when compared to a baseband system, is a factor of 0.075 or equivalent to a loss of 11 dB.
- The reason for this loss is that a large part of the transmitter power is used to send the carrier component of the modulated signal and not the desired signal.
- To analyze the envelope-detector performance in the presence of noise, we must use certain approximations.
 - This is a result of the nonlinear structure of an envelope detector, which makes an exact analysis difficult.

- In this case, the demodulator detects the envelope of the received signal and the noise process.
- The input to the envelope detector is

$$r(t) = [A_c[1 + am_n(t)] + n_c(t)]\cos(2f f_c t) - n_s(t)\sin(2f f_c t)$$

• Therefore, the envelope of *r* (*t*) is given by

$$V_r(t) = \sqrt{\left[A_c[1 + am_n(t)] + n_c(t)\right]^2 + n_s^2(t)}$$

• Now we assume that the signal component in r(t) is much stronger than the noise component. Then

$$P(n_c(t) << A_c[1 + am_n(t)]) \approx 1$$

• Therefore, we have a high probability that

$$V_r(t) \approx A_c[1 + am_n(t)] + n_c(t)$$

After removing the DC component, we obtain

$$y(t) = A_c a m_n(t) + n_c(t)$$

- which is basically the same as y(t) for the synchronous demodulation without the $\frac{1}{2}$ coefficient.
- This coefficient, of course, has no effect on the final SNR.
- So we conclude that, under the assumption of high SNR at the receiver input, the performance of synchronous and envelope demodulators is the same.
- However, if the preceding assumption is not true, that is, if we assume that, at the receiver input, the noise power is much stronger than the signal power, Then

$$\begin{split} V_{r}(t) &= \sqrt{\left[A_{c}[1 + am_{n}(t)] + n_{c}(t)\right]^{2} + n_{s}^{2}(t)} \\ &= \sqrt{A_{c}^{2}[1 + am_{n}(t)]^{2} + n_{c}^{2}(t) + n_{s}^{2}(t) + 2A_{c}n_{c}(t)[1 + am_{n}(t)]} \\ &\stackrel{a}{\longrightarrow} \sqrt{\left(n_{c}^{2}(t) + n_{s}^{2}(t)\right)\left[1 + \frac{2A_{c}n_{c}(t)}{n_{c}^{2}(t) + n_{s}^{2}(t)}\left(1 + am_{n}(t)\right)\right]} \\ &\stackrel{b}{\longrightarrow} V_{n}(t)\left[1 + \frac{A_{c}n_{c}(t)}{V_{n}^{2}(t)}\left(1 + am_{n}(t)\right)\right] \\ &= V_{n}(t) + \frac{A_{c}n_{c}(t)}{V_{n}(t)}\left(1 + am_{n}(t)\right) \end{split}$$

- (a): $A_c^2[1+am_n(t)]^2$ is small compared with the other components (b): $\sqrt{n_c^2(t)+n_s^2(t)}=V_n(t)$; the envelope of the noise process Use the approximation

$$\sqrt{1+V} \approx 1 + \frac{v}{2}$$
, for small V , where $V = \frac{2A_c n_c(t)}{n_c^2(t) + n_s^2(t)} (1 + am_n(t))$

Then

$$V_r(t) = V_n(t) + \frac{A_c n_c(t)}{V_n(t)} (1 + am_n(t))$$

- We observe that, at the demodulator output, the signal and the noise components are **no longer additive**.
- In fact, the signal component is multiplied by noise and is no longer distinguishable.
- In this case, no meaningful SNR can be defined.
- We say that this system is operating below the threshold.
- The subject of threshold and its effect on the performance of a communication system will be covered in more detail when we discuss the noise performance in angle modulation.