Chapter 4: Effect of Noise on Analog Communication Systems

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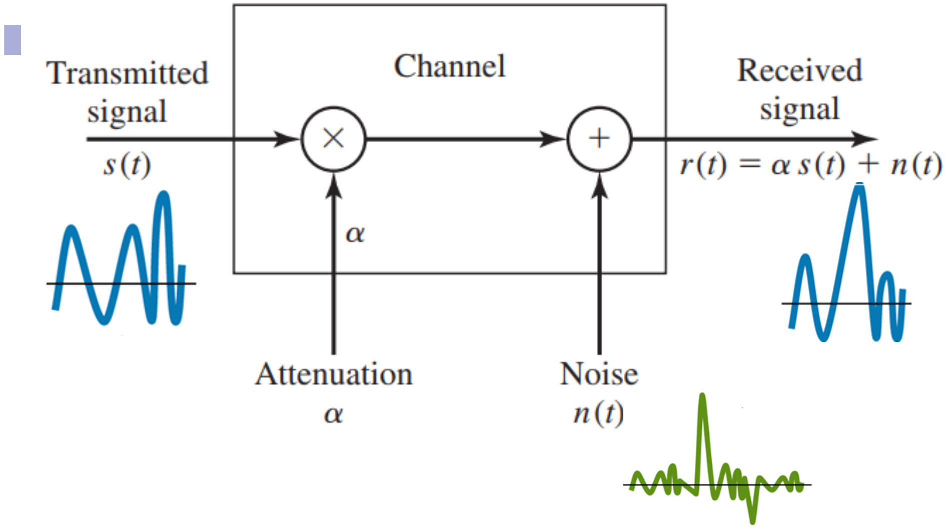
Review: Transmission Media

Information Source and Input Transducer → Transmitter → Channel → Receiver → Output Transducer

- Means used to carry information signal to the destination/receiver
- Characterized by
  - Physical properties — Bandwidth
  - Signaling method(s) — Sensitivity to noise
- *Wireline channels* : a *guided medium* over which the information will be transmitted from the transmitter to the receiver
- *Wireless Channels*: an *unguided medium* where information transmission is via electromagnetic waves from antenna to antenna.

Transmission Losses and Noise

Information Source and Input Transducer



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Output Transducer

- Two dominant factors that limit the performance of the system are *Attenuation* and *Additive Noise.*
  - Lossy channels
  - Electronic devices that are used to filter and amplify, wired media

Transmission Losses

- The amount of signal attenuation generally depends on
  - The physical medium,
  - The frequency of operation,
  - The distance between the transmitter and the receiver

$$\mathcal{L} = \frac{P_t}{P_r}$$

or, in decibels, as

$$\mathcal{L}_{dB} = 10\log_{10}\mathcal{L} = 10\log_{10}P_t - 10\log_{10}P_r$$

- Example 1:

  Determine the transmission loss for a 10-Km and a 20-Km coaxial cable if the loss/Km is 2 dB at the frequency operation.

Cont.…

- Example 2:

In line of site wireless systems the transmission loss is given as

$$\mathscr{L} = \left(\frac{4\pi d}{\lambda}\right)^2$$

: also known as free-space path loss

Determine the free-space path loss for a signal transmitted at f = 900MHz over distances of 1 Km and 2 Km.

- *How will the increases in distance affect the received signal strength?*
- *If the frequency is changed to 1800MHz what will be the new transmission loss at 1 Km and 2 Km? how does it change?*

<u>Challenge:</u>

If a mobile tower antenna is transmitting at 16dBw, at what maximum distance will your phone still can be able to receive a signal? (take the transmission loss = free-space path loss, GSM transmission at 900MHz)

Noise

- Noise is undesired or unwanted signal
  - Thermal noise, which is due to the random motion of electrons in a wire which creates an extra signal not originally sent by the transmitter.
  - Induced noise comes from sources such as motors and appliances. These devices act as a sending antenna, and the transmission medium acts as the receiving antenna.
  - Crosstalk is the effect of one wire on the other. One wire acts as a sending antenna and the other as the receiving antenna.
  - Impulse noise is a spike (a signal with high energy in a very short time) that comes from power lines, lightning,…

Review : Random Processes

- Important concept in modeling the randomness of noise.
- Random process (signal) $X(t)$ can be viewed as collection on random variables $\{X(t_1), X(t_2), X(t_3)\ldots\}$ at $t_1, t_2, t_3 \ldots.$ All $t \in \mathbb{R}$
- Can we quantity it?
  - Yes, with Statistical descriptions
- Can it be filtered?
  - Yes , LTI filters

Cont.…

- Autocorrelation Function

$R_{xx}(\mathrm{t},\tau)$ is defined by $R_{xx}(\mathrm{t},\tau) = E[X(\mathrm{t})X(t+\tau)]$

Where the expectation $E[X(\mathrm{t})] = \int_{-\infty}^{\infty} x f_{X(t)}(x) dx$

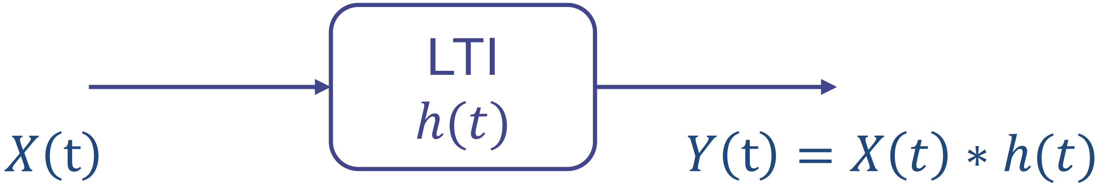
- If $R_{xx}(\mathrm{t},\tau) = R_{xx}(\tau)$ and $E[X(\mathrm{t})] = m_x$.. *(the autocorrelation is dependent on τ & mean is constant) then the process is* ***Wide sense stationary.***

Power Spectral density (w\Hz): $S_{xx}(f) = \int_{-\infty}^{\infty} R_{xx}(\tau)\, e^{-j2\pi f\tau}\, d\tau$

Signal Power: $\int_{-\infty}^{\infty} S_{xx}(f) d(f)$

Cont…

- For WSS Process X(t),



- Proof: Ex!!

$$R_{yy}(\tau) = h(\tau)R_{xx}(\tau)h(-\tau)$$

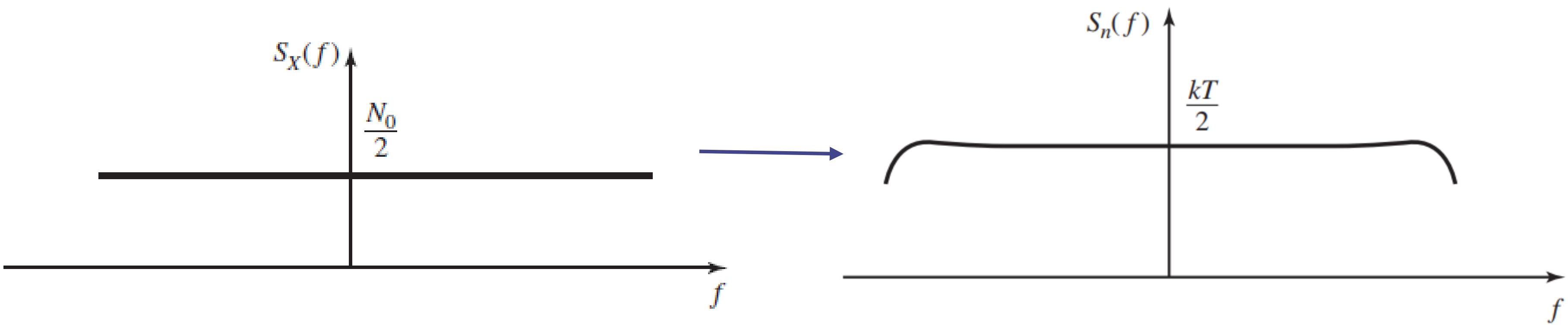
$$S_{yy}(f) = \int_{-\infty}^{\infty} R_{yy}(\tau)\, e^{-j2\pi f\tau}\, d\tau = S_{xx}(f) * |H(f)|^{2}$$

Ex!

- For a given random signal $X(t) = Acos(2\pi f_c +$

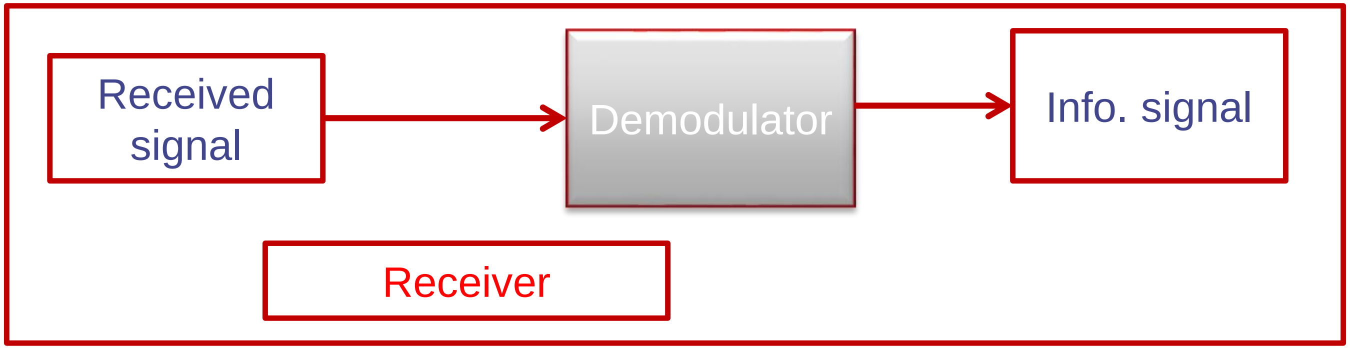
Noise characterization

- Thermal noise can be closely modeled as Gaussian Process.
- Noise process exist in all frequency components
  - appear with equal power; i.e., the power-spectral density is a constant for all frequencies→ white noise
- Thus, we can refer it as ***Additive white Gaussian Noise***
- The spectral density of AWGN where $N_o = \kappa . T$



Effect of Noise at the Receiver

- Main function: to recover the message from the received signal
  - *Somewhat inverse of the transmitter function*
- Demodulate, decode and extract the information content of the received signal.
- Operates in the presence of noise, interference, attenuation
  - Hence, some distortions are unavoidable
- Some other functions: filtering, suppression of noise and interference
- Error detection and correction.



Signal to Noise Ratio (SNR)

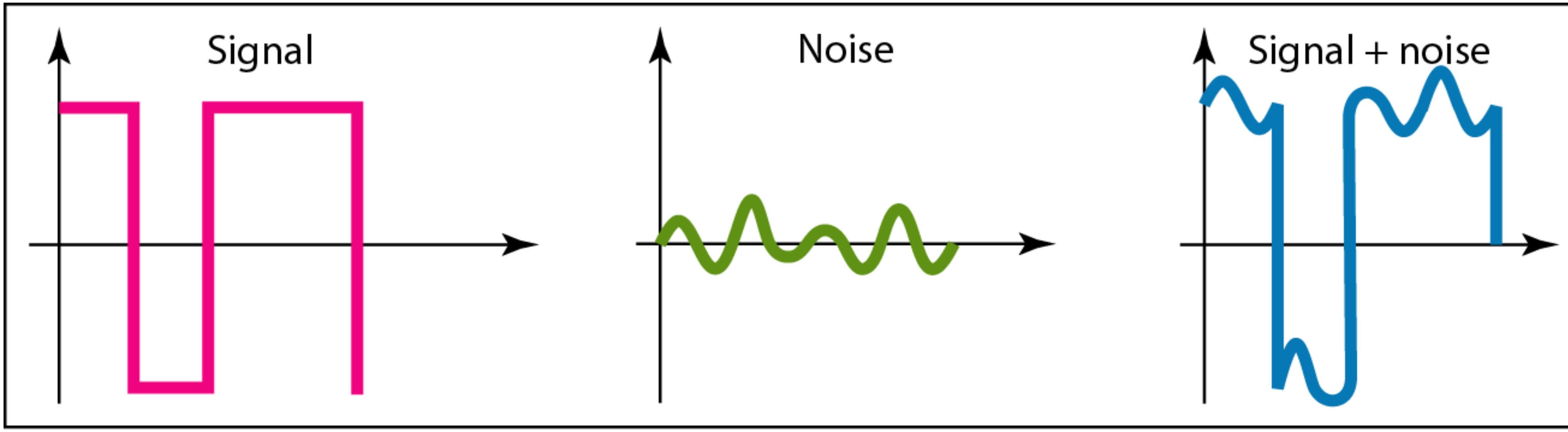
- To measure the quality of a system the SNR is often used.
- It indicates the strength of the signal w.r.t. the noise power in the system.

$$SNR = \frac{P_s}{N}$$

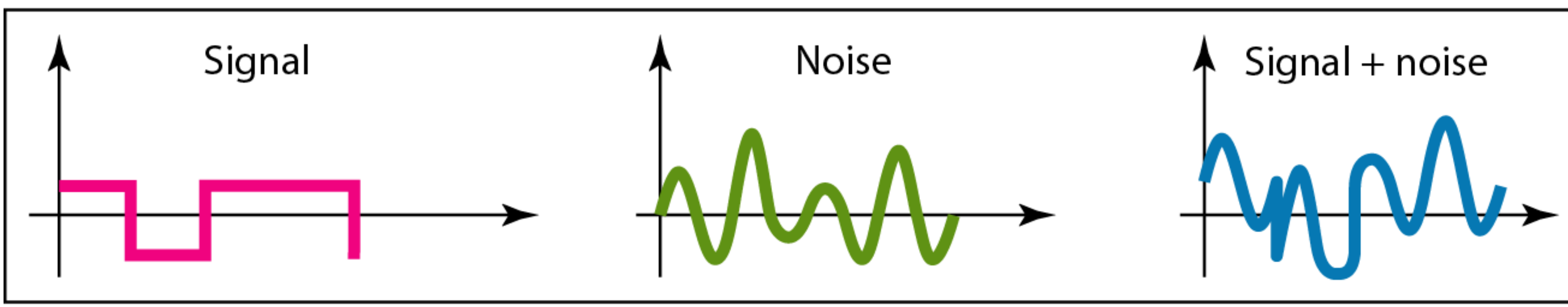
- It is usually given in dB and referred to as SNR_{dB}.

$$SNR_{dB} = 10\log_{10} SNR = 10\log_{10} P_s - 10\log_{10} N_o$$

Where $N = \kappa . Bw . T = Thermal\ Noise\ power$



a. Large SNR



b. Small SNR

Effect of Noise on a Baseband System

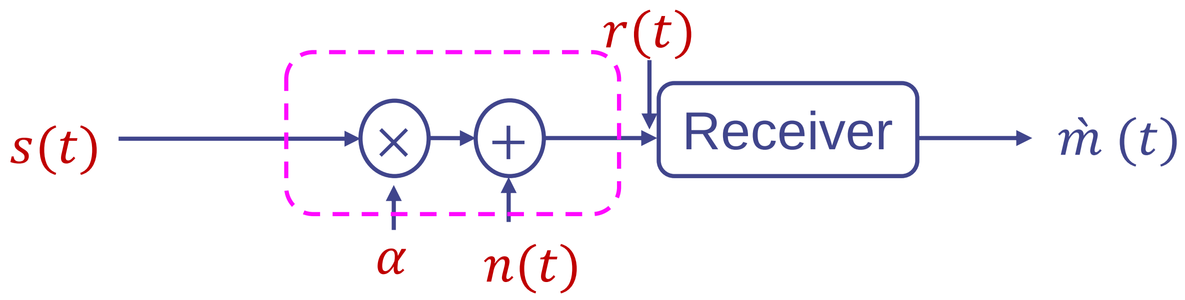
- Since baseband systems serve as a basis for comparison of various modulation systems, we begin with a noise analysis of a baseband system.
- In this case, there is no carrier demodulation to be performed.
- The receiver consists only of an ideal lowpass filter with the bandwidth W.
- The noise power 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 PR, the baseband SNR is given by

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

Effect of Noise on Linear-Modulation Systems



- The transmitted signal, $s(t) =$

$A_c m(t) Cos 2\pi f_c t$ … **DSB-SC**

$A_c (1 + m(t)) Cos 2\pi f_c t$ … **C. AM**

$A_c m(t) Cos 2\pi f_c t \mp A_c \widehat{m}(t) Sin 2\pi f_c t$… **SB-SC**

- The received signal at the output of the receiver noise-limiting filter : Sum of this signal and filtered noise

$$r(t) = \alpha S(t) + n(t)$$

Cont….

- The filtered noise process can be expressed in terms of its in-phase and quadrature components as

$$n(t) = A(t)\cos[2\pi f_c t + \theta(t)] = A(t)\cos\theta(t)\cos(2\pi f_c t) - A(t)\sin\theta(t)\sin(2\pi f_c t)$$
$$= n_c(t)\cos(2\pi f_c t) - n_s(t)\sin(2\pi f_c t)$$

(where nc(t) is in-phase component and ns(t) is quadrature component)

Effect of Noise on DSB-SC AM

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

$$\begin{aligned} r(t) &= \alpha S(t) + n(t) = u(t) + n(t) \\ &= Am(t)\cos(2\pi f_c t) + n_c(t)\cos(2\pi f_c t) - n_s(t)\sin(2\pi f_c t) \end{aligned}$$

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

Cont.…

- The multiplication of r(t) with cos(2πfct + ϕ) yields

$$
\begin{aligned}
r(t)\cos(2\pi f_c t+\phi) \\
&= u(t)\cos(2\pi f_c t+\phi)+n(t)\cos(2\pi f_c t+\phi) \\
&= Am(t)\cos(2\pi f_c t)\cos(2\pi f_c t+\phi) \\
&\quad + n_c(t)\cos(2\pi f_c t)\cos(2\pi f_c t+\phi)-n_s(t)\sin(2\pi f_c t)\cos(2\pi f_c t+\phi) \\
&= \tfrac{1}{2}Am(t)\cos(\phi)+\tfrac{1}{2}A_c m(t)\cos(4\pi f_c t+\phi) \\
&\quad + \tfrac{1}{2}\left[n_c(t)\cos(\phi)+n_s(t)\sin(\phi)\right]+\tfrac{1}{2}\left[n_c(t)\cos(4\pi f_c t+\phi)-n_s(t)\sin(4\pi f_c t+\phi)\right]
\end{aligned}
$$

- The lowpass filter rejects the double frequency components and passes only the lowpass components.

$$
y(t)=\tfrac{1}{2}Am(t)\cos(\phi)+\tfrac{1}{2}\left[n_c(t)\cos(\phi)+n_s(t)\sin(\phi)\right]
$$

Cont.…

- The effect of a phase difference between the received carrier and a locally generated carrier at the receiver is a drop equal to $\cos^2(\phi)$ in the received signal power.
- If we assume that $\phi = 0$

$$m`(t) = \tfrac{1}{2}\left[A\, m(t) + n_c(t)\right]$$

Cont.…

- 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

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

  - 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 W.

Cont.…

- 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

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

- The received signal power, given by

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

Cont.…

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

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

  - which is identical to baseband SNR

- In DSB-SC AM, the output SNR is the same as the SNR for a baseband system

$\Rightarrow$ DSB-SC AM does not provide any SNR improvement over a simple baseband communication system

Example 3:

In a broadcasting communication system the transmitter power is 40 KW, the channel attenuation is 80 dB, and the noise power-spectral density is 10^{-10} W/Hz. The message signal has a bandwidth of 10^4 Hz.

- Find the output SNR if the modulation is DSB.
- *Find the pre-detection SNR (SNR in r(t) = ku(t) + n(t))*

Effect of Noise on SSB AM

- Input to the demodulator

$$
\begin{aligned}
r(t) &= A\, m(t)\cos(2\pi f_c t) \mp A\, \hat{m}(t)\sin(2\pi f_c t) + n(t) \\
&= Am(t)\cos(2\pi f_c t) \mp A\, \hat{m}(t)\sin(2\pi f_c t) + n_c(t)\cos(2\pi f_c t) - n_s(t)\sin(2\pi f_c t) \\
&= \left[Am(t) + n_c(t)\right]\cos(2\pi f_c t) + \left[\mp A\hat{m}(t) - n_s(t)\right]\sin(2\pi f_c t)
\end{aligned}
$$

- Assumption :
  - Demodulation with an ideal phase reference ($\phi = 0$).
- Hence, the output of the lowpass filter is the in-phase component (with a coefficient of ½) of the preceding signal.

$$
m`(t) = \tfrac{1}{2}\left[A\, m(t) + n_c(t)\right]
$$

Cont…

- Parallel to our discussion of DSB, we have

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

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

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

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

$$P_R = P_U = A^2 P_M$$

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

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

Effect of Noise on Conventional AM

- Received signal at the input to the demodulator

$$r(t) = A[1 + am_n(t)]\cos(2\pi f_c t) + n(t)$$
$$= A\,[1 + am_n(t)]\cos(2\pi f_c t) + n_c(t)\cos\left(2\pi f_c t\right) - n_s(t)\sin\left(2\pi f_c t\right)$$
$$= \left[A\,[1 + am_n(t)] + n_c(t)\right]\cos(2\pi f_c t) - n_s(t)\sin\left(2\pi f_c t\right)$$

  - a is the modulation index
  - $m_n(t)$ is normalized
  - 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

$$m`(t) = \tfrac{1}{2}\left[A\,am_n(t) + n_c(t)\right]$$

Cont…

- Received signal power

$$P_R = \frac{A^2}{2}\left[1 + a^2 P_{M_n}\right]$$

- Now we can derive the output SNR as

$$\left(\frac{S}{N}\right)_{0_{AM}} = \frac{\frac{1}{4}A^2 a^2 P_{M_n}}{\frac{1}{4}P_{n_c}} = \frac{A^2 a^2 P_{M_n}}{2N_0 W} = \frac{a^2 P_{M_n}}{1 + a^2 P_{M_n}} \frac{\frac{A^2}{2}\left[1 + a^2 P_{M_n}\right]}{N_0 W}$$

$$= \frac{a^2 P_{M_n}}{1 + a^2 P_{M_n}} \frac{P_R}{N_0 W} = \frac{a^2 P_{M_n}}{1 + a^2 P_{M_n}} \left(\frac{S}{N}\right)_b = \eta \left(\frac{S}{N}\right)_b$$

- η 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.

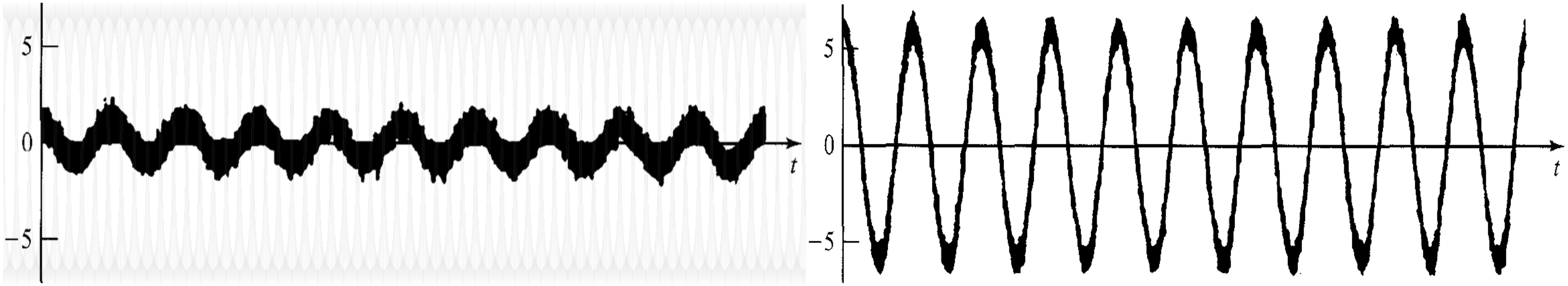
EX:

From Example 3;

- Find the output SNR if the modulation is conventional AM with a modulation index of 0.85 and normalized message power of 0.2.

Effect of Noise on Angle Modulation

- A figure shown in below is the effect of additive noise on zero crossings of two FM signals, one with high power and the other with low power.
- From the previous discussion and also from the figure it should be clear that the effect of noise in an FM system is different from that for an AM system.
- We also observe that the effect of noise in a low-power FM system is more severe than in a high-power FM system.
    - In a low power signal, noise causes more changes in the zero crossings.
- The analysis that we present next verifies our intuition based on these observations.

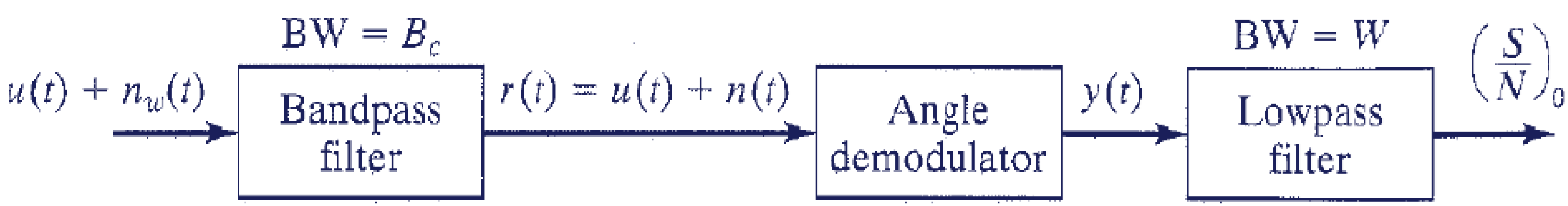


Cont….

- The receiver for a general angle-modulated signal is shown in below
- The angle-modulated signal is represented as

$$u(t) = A_c \cos\left(2\pi f_c t + \phi(t)\right) = \begin{cases} A_c \cos\left(2\pi f_c t + 2\pi k_f \int_{-\infty}^{t} m(\tau)\,d\tau\right) & FM \\ A_c \cos\left(2\pi f_c t + k_p m(t)\right) & PM \end{cases}$$

- The AWGN $n_w(t)$ is added to $u(t)$, and the result is passed through a noise-limiting filter whose role is to remove the out-of-band noise.
- The bandwidth of this filter is equal to that of the modulated signal
- Therefore, it passes the modulated signal without distortion.
- However, it eliminates the out-of-band noise.
- Hence, the noise output of the filter is a filtered noise denoted by n(t).



Cont.…

- The output of this filter is

$$r(t) = u(t) + n(t) = u(t) + n_c(t)\cos\left(2\pi f_c t\right) - n_s(t)\sin\left(2\pi f_c t\right)$$

- A precise analysis is complicate due to the nonlinearity of demodulation .
- Let us assume that the signal power is much higher than the noise power.
- Then, the bandpass noise is represented as

$$n(t) = \sqrt{n_c^2(t) + n_s^2(t)}\cos\left(2\pi f_c t + \arctan\frac{n_s(t)}{n_c(t)}\right) = V_n(t)\cos\left(2\pi f_c t + \Phi_n(t)\right)$$

  - where $V_n(t)$ and $\Phi_n(t)$ represent the envelope and the phase of the bandpass noise process, respectively.

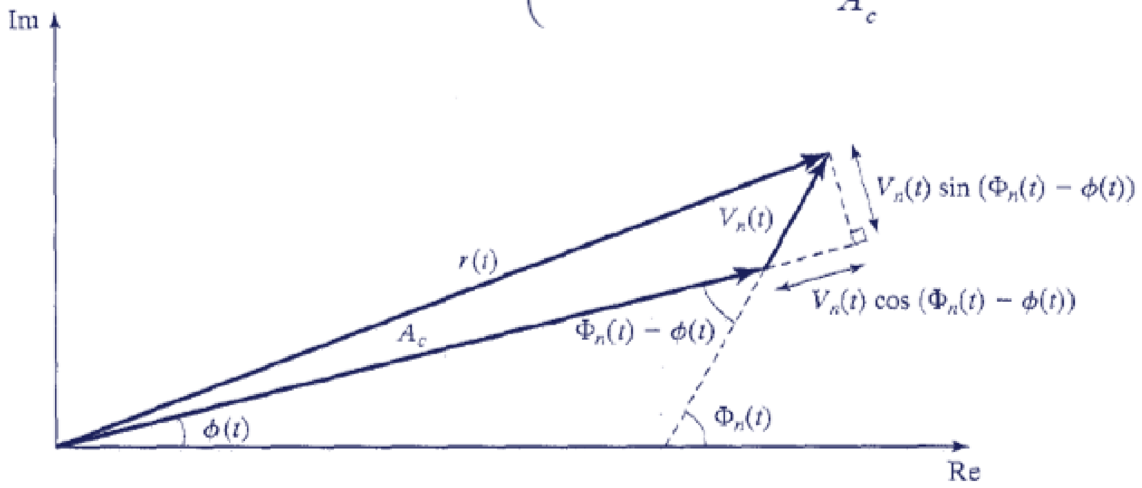
- Assume that the signal is much larger than the noise, that is,

$$P\left(V_n(t) << A_c\right) \approx 1$$

- The phasor diagram of signal and noise are shown in below.
- From this figure, it is obvious that we can write

$$r(t) \approx \left[A_c + V_n(t)\cos\left(\Phi_n(t)-\phi(t)\right)\right]\cos\left(2\pi f_c t + \phi(t) + \arctan\frac{V_n(t)\sin\left(\Phi_n(t)-\phi(t)\right)}{A_c + V_n(t)\cos\left(\Phi_n(t)-\phi(t)\right)}\right)$$

$$\approx \left[A_c + V_n(t)\cos\left(\Phi_n(t)-\phi(t)\right)\right]\cos\left(2\pi f_c t + \phi(t) + \frac{V_n(t)}{A_c}\sin\left(\Phi_n(t)-\phi(t)\right)\right)$$



- Noting that $\phi(t) = \begin{cases} k_p m(t), & PM \\ 2\pi k_f \int_{-\infty}^{t} m(\tau)\,d\tau, & FM \end{cases}$
- We see that the output of the demodulator is given by

$$y(t) = \begin{cases} \phi(t) + \dfrac{V_n(t)}{A_c}\sin\left(\Phi_n(t) - \phi(t)\right) & PM \\ \dfrac{1}{2\pi}\dfrac{d}{dt}\left[\phi(t) + \dfrac{V_n(t)}{A_c}\sin\left(\Phi_n(t) - \phi(t)\right)\right] & FM \end{cases}$$

$$= \begin{cases} k_p m(t) + \dfrac{V_n(t)}{A_c}\sin\left(\Phi_n(t) - \phi(t)\right) & PM \\ k_f m(t) + \dfrac{1}{2\pi}\dfrac{d}{dt}\dfrac{V_n(t)}{A_c}\sin\left(\Phi_n(t) - \phi(t)\right) & FM \end{cases} = \begin{cases} k_p m(t) + Y_n(t) & PM \\ k_f m(t) + \dfrac{1}{2\pi}\dfrac{d}{dt}Y_n(t) & FM \end{cases}$$

  - where we define

$$Y_n(t) \underline{\underline{def}} \frac{V_n(t)}{A_c}\sin\left(\Phi_n(t) - \phi(t)\right)$$

Cont.…

$$y(t) = \begin{cases} k_p m(t) + \dfrac{V_n(t)}{A_c} \sin\left(\Phi_n(t) - \phi(t)\right) & PM \\ k_f m(t) + \dfrac{1}{2\pi} \dfrac{d}{dt} \dfrac{V_n(t)}{A_c} \sin\left(\Phi_n(t) - \phi(t)\right) & FM \end{cases} = \begin{cases} k_p m(t) + Y_n(t) & PM \\ k_f m(t) + \dfrac{1}{2\pi} \dfrac{d}{dt} Y_n(t) & FM \end{cases}$$

- The first term in above equation is the desired signal component.
- The second term is the noise component.
- The noise component is inversely proportional to the signal amplitude A_c.
- Hence, the higher the signal level, the lower the noise level.

$$S_{Y_n}(f) = \left(a^2 + b^2\right) S_{n_c}(f) = \frac{S_{n_c}(f)}{A_c^2}$$

- $S_{nc}(f)$ is the power spectral density (psd) of the in-phase component of the filtered noise given :
- Therefore

$$S_{Y_n}(f) = \begin{cases} \frac{N_0}{A_c^2} & |f| < \frac{B_c}{2} \\ 0 & otherwise \end{cases}$$

Cont…

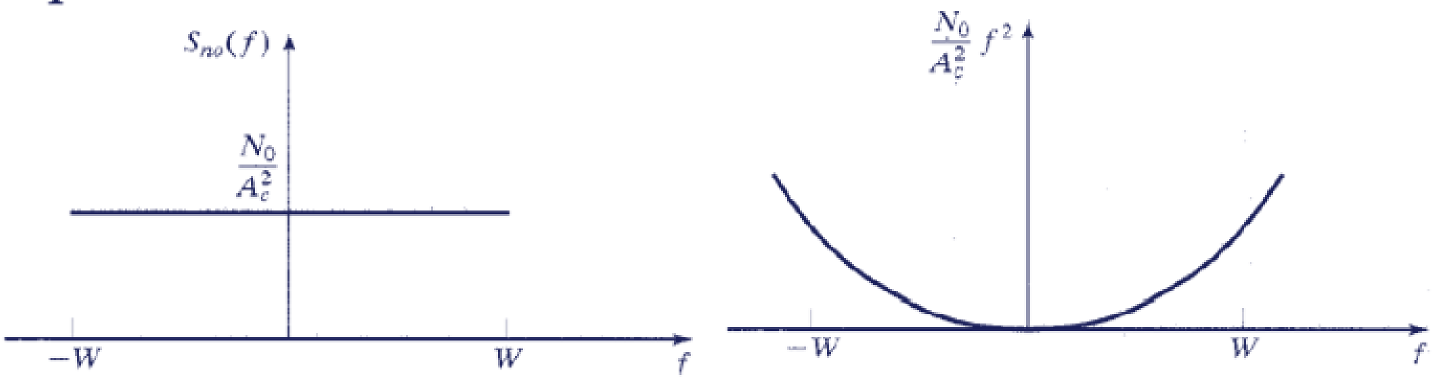
- This equation provides an expression for the power spectral density of the filtered noise at *the front end of the receiver.*
- After demodulation, another filtering is applied; this reduces the noise bandwidth to W, which is the bandwidth of the message signal.
- Note that in the case of FM modulation,
  the process $Y_n(t)$ is differentiated and scaled by $1/2\pi$.
- The PSD of the process $(1/2\pi)\,(dY_n(t)/dt)$ is given by

$$\frac{4\pi^2 f^2}{4\pi^2} S_{Y_n}(f) = f^2 S_{Y_n}(f) = \begin{cases} \frac{N_0}{A_c^2} f^2 & |f| < \frac{B_c}{2} \\ 0 & otherwise \end{cases}$$

- Hence, for $|f|<W$

$$S_{n_0}(f) = \begin{cases} \frac{N_0}{A_c^2} & PM \\ \frac{N_0}{A_c^2} f^2 & FM \end{cases}$$

- Fig. *6.4* shows the power spectrum of the noise component at the output of the demodulator for PM and FM.



Noise power spectrum at demodulator output for $|f|<W$ in (a) PM (b) and (b) FM.

- It is interesting to note that **PM has a flat noise spectrum and FM has a parabolic noise spectrum**.
- Therefore, the effect of noise in FM for higher frequency components is much higher than the effect of noise on lower frequency components.
- The noise power at the output of the lowpass filter is the noise power in the frequency range [W, +W].
- Therefore, it is given by

$$P_{n_0} = \int_{-W}^{W} S_{n_0}(f)\,df = \begin{cases} \int_{-W}^{W} \frac{N_0}{A_c^2}\,df \\ \int_{-W}^{W} \frac{N_0}{A_c^2} f^2\,df \end{cases} = \begin{cases} \frac{2WN_0}{A_c^2} & PM \\ \frac{2N_0W^3}{3A_c^2} & FM \end{cases}$$

SNR

- the output SNR in angle modulation.
- First, we have the output signal power

$$P_{S_o} = \begin{cases} k_p^2 P_M & PM \\ k_f^2 P_M & FM \end{cases}$$

- Then the SNR, which is defined as

$$\left(\frac{S}{N}\right)_o \stackrel{def}{=} \frac{P_{S_o}}{P_{n_o}} \longrightarrow \left(\frac{S}{N}\right)_o = \begin{cases} \frac{k_p^2 A_c^2}{2} \frac{P_M}{N_0 W} & PM \\ \frac{3k_f^2 A_c^2}{2W^2} \frac{P_M}{N_0 W} & FM \end{cases}$$

- Noting that $A_c^2/2$ is the received signal power, denoted by P_R, and

$$\begin{cases} \beta_p = k_p \max\left[|m(t)|\right] & PM \\ \beta_f = \frac{k_f \max\left[|m(t)|\right]}{W} & FM \end{cases} \longrightarrow \left(\frac{S}{N}\right)_o = \begin{cases} P_R \left(\frac{\beta_p}{\max\left[|m(t)|\right]}\right)^2 \frac{P_M}{N_0 W} & PM \\ 3P_R \left(\frac{\beta_f}{\max\left[|m(t)|\right]}\right)^2 \frac{P_M}{N_0 W} & FM \end{cases}$$

NOTE:

- **Observations**
  - In both PM and FM, the output SNR is proportional to β^2. Therefore, increasing β increases the output SNR.
  - Increasing β increase the bandwidth (from Carson's rule).

    So **angle modulation provides a way to trade off bandwidth for transmitted power**.

Quiz

1. Briefly explain advantage of angle modulation over AM modulation /list only two.
2. Why do we need to modulate /list only two.
3. An FM modulating signal has 500Hz frequency, 3.2volt amplitude and 6.4 KHz frequency deviation.
   a. If the baseband signal voltage is now increased to 8.4volt, determine the new frequency deviation, modulation index and Carson's bandwidth.
   b. If the message signal voltage is raised to 20volts while the audio frequency is dropped to 200Hz, determine the frequency deviation, modulation index and Carson's bandwidth.

Quiz

1. For a given random signal $X(t) = Acos(2\pi f_c +$