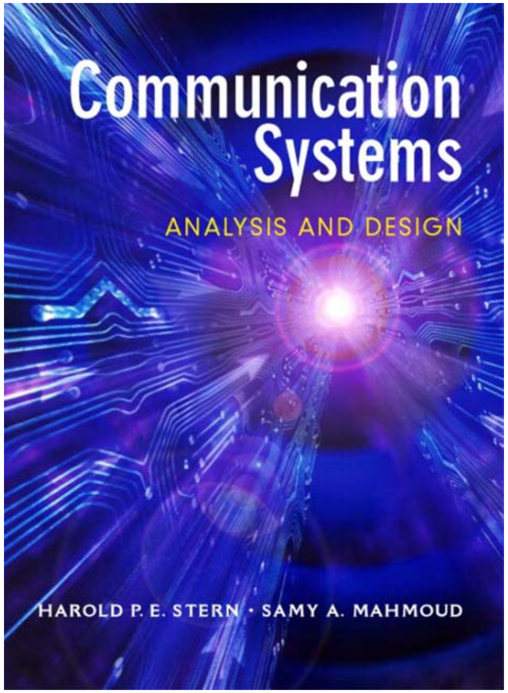
# Chapter 6

# Analog Modulation and Demodulation

# Chapter 6

# Analog Modulation and Demodulation

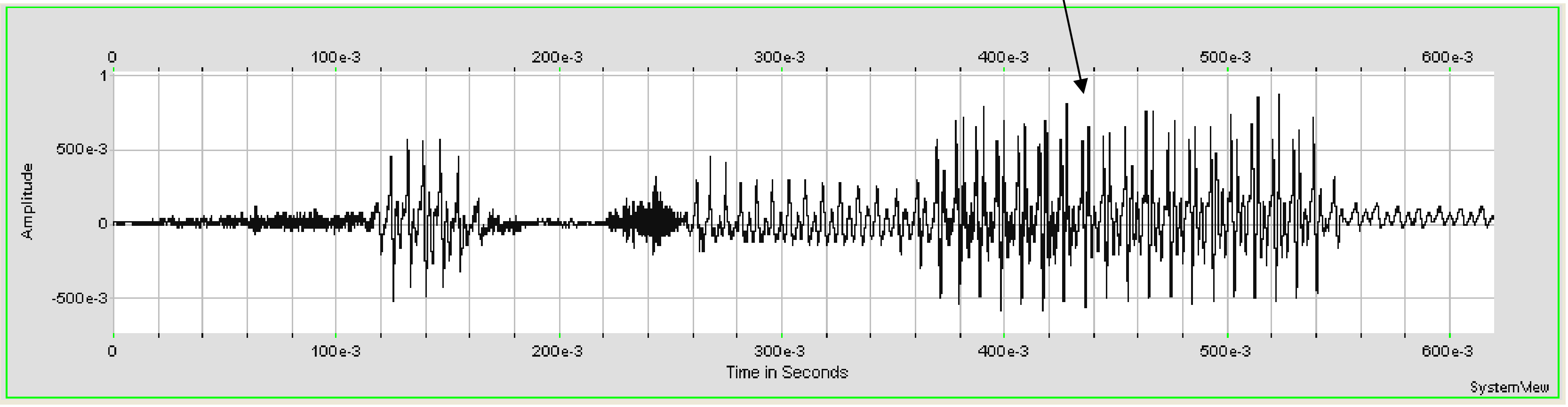
- *Amplitude Modulation*
- Pages 306-309



- The analytical signal for double sideband, large carrier amplitude modulation (DSB-LC AM) is:

$$s_{DSB\text{-}LC\ AM}(t) = A_C\,(c + s(t))\cos(2\pi f_C t)$$

where $c$ is the *DC bias* or *offset* and $A_C$ is the carrier amplitude. The continuous analog signal *s(t)* is a baseband signal with the information content (voice or music) to be transmitted.



- The baseband power spectral density (PSD) spectrum of the information signal s(t) or S(f) for voice has significant components below 500 Hz and a bandwidth of < 8 kHz:

$$S(f) = \mathcal{F}(s(t))$$

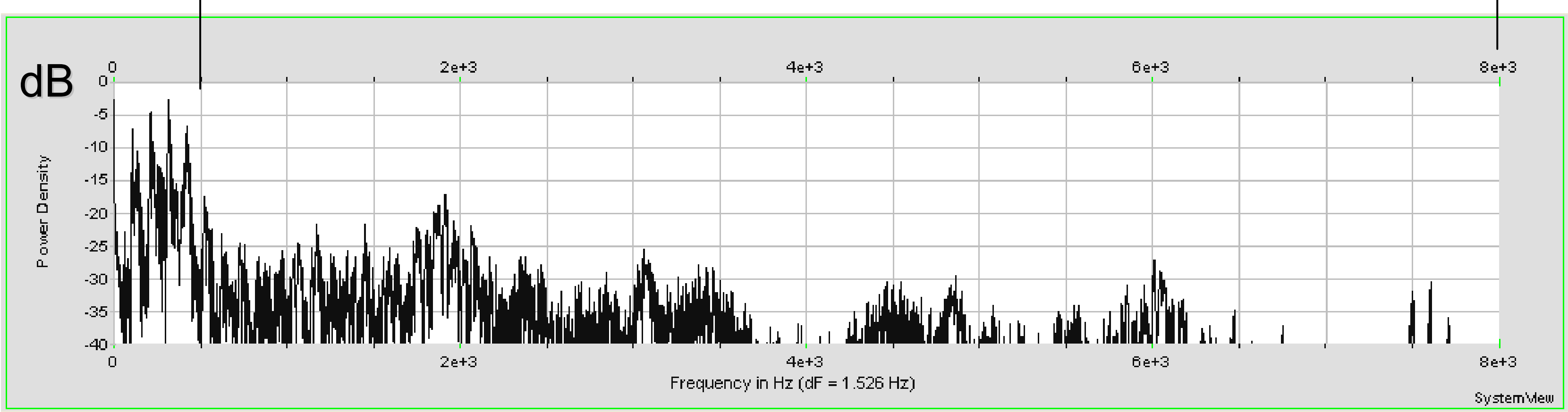
The *single-sided* spectrum of the modulated signal is:

$$\mathcal{F}(A_C\,(c + s(t))\cos(2\pi f_C\, t)) = S(f - f_C)$$

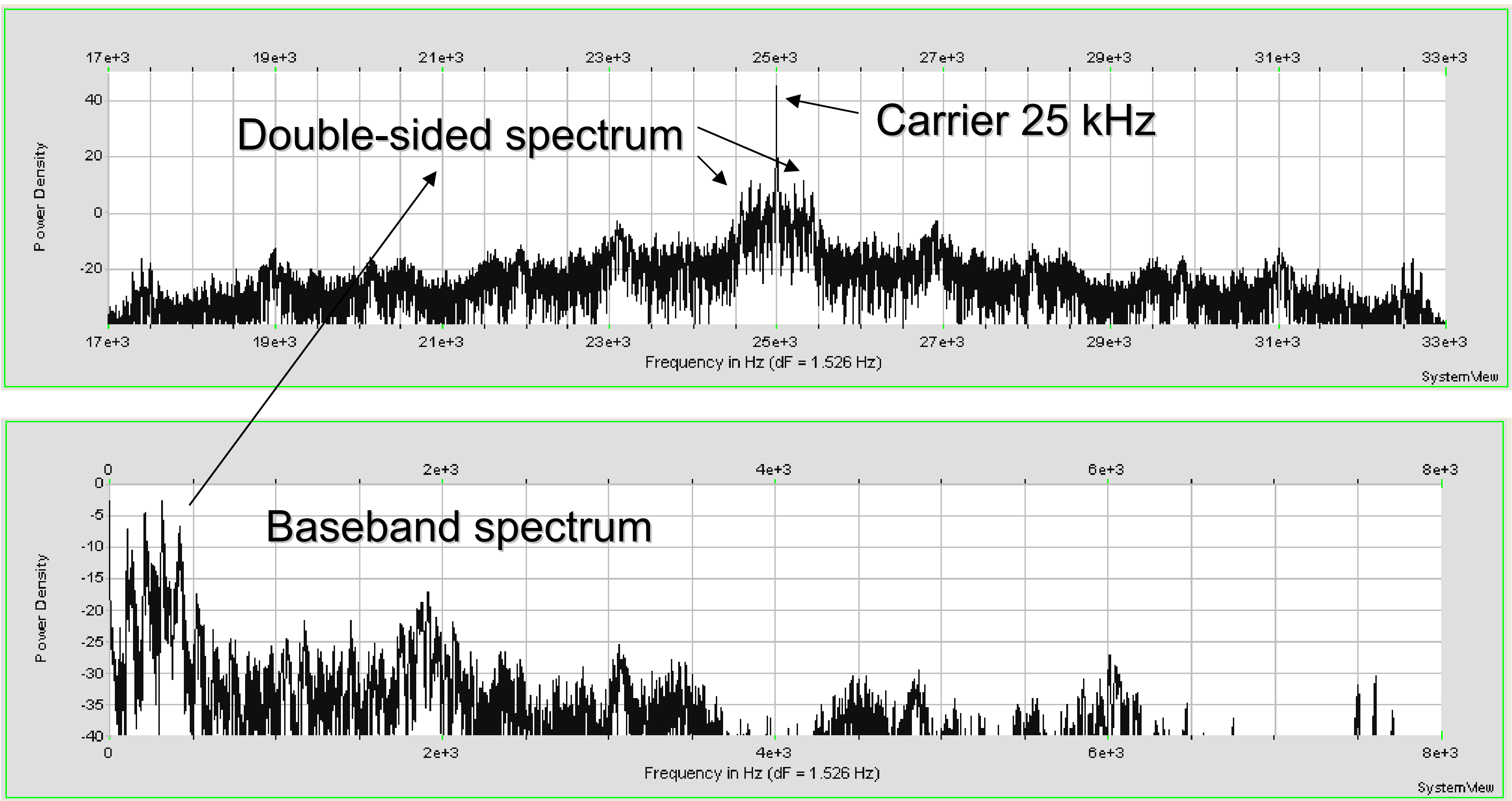
500 Hz

Power Spectral Density of s(t)

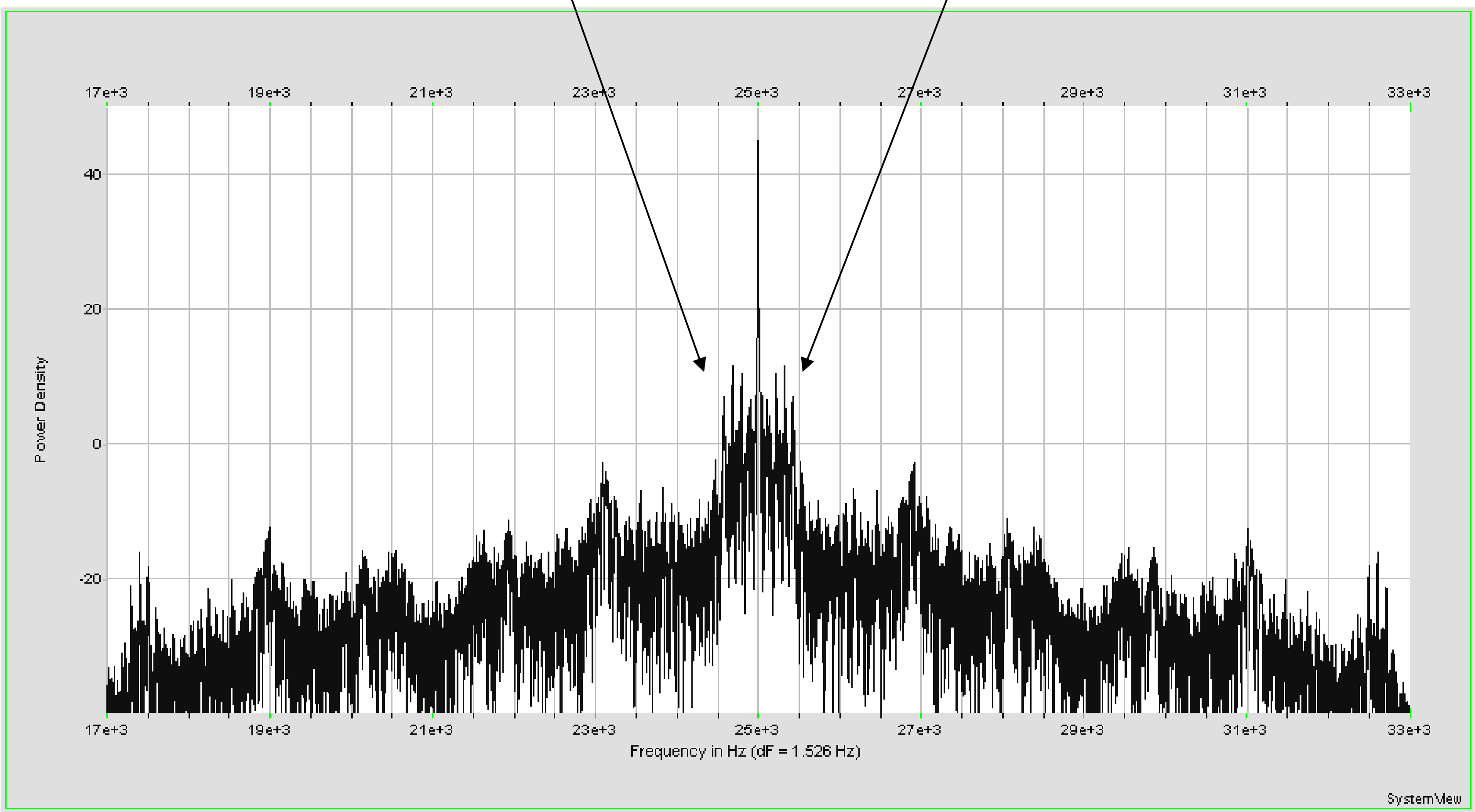
8 kHz



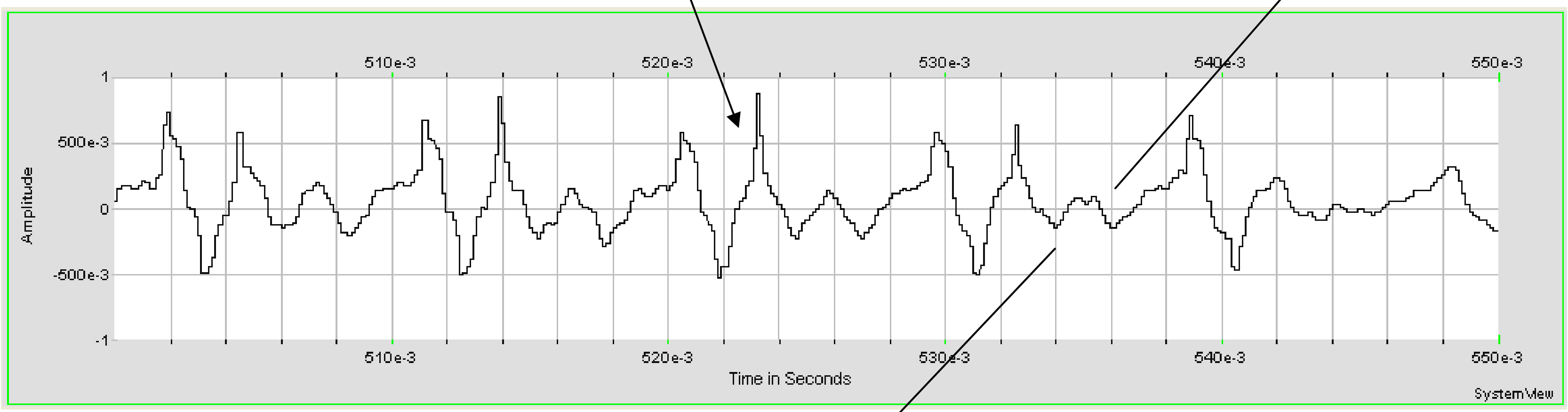
- The *single-sided* (positive frequency axis) spectrum of the *modulated signal* replicates the baseband spectrum as a *double-sided* spectrum about the carrier frequency.

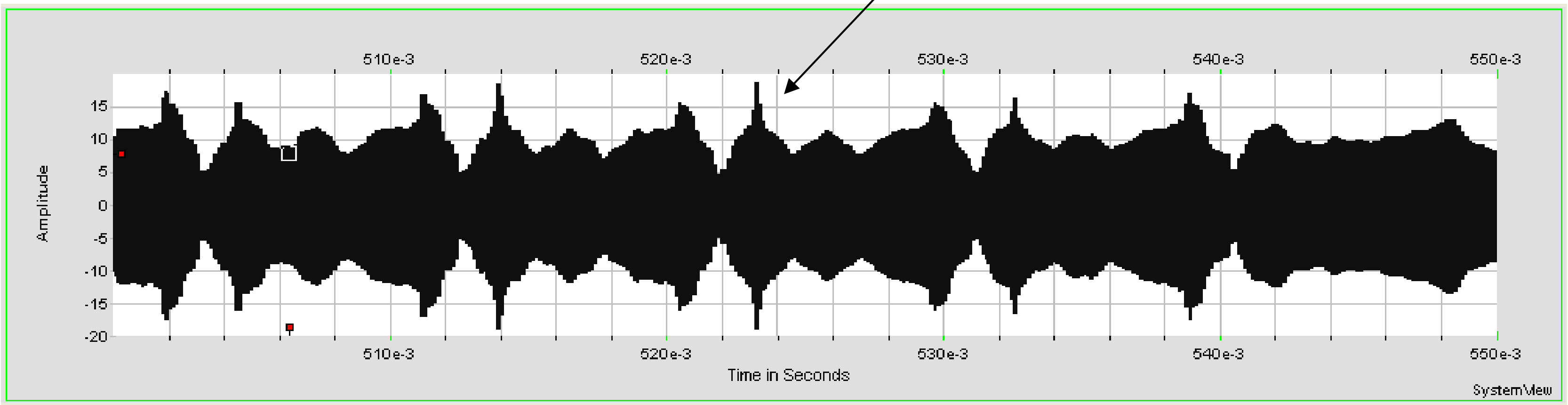


- The double-sided modulated spectrum about the carrier frequency has an *lower* (*LSB*) and *upper* (*USB*) sideband.



- The modulated DSB-LC AM signal shows an *outer envelope* that follows the polar baseband signal s(t).

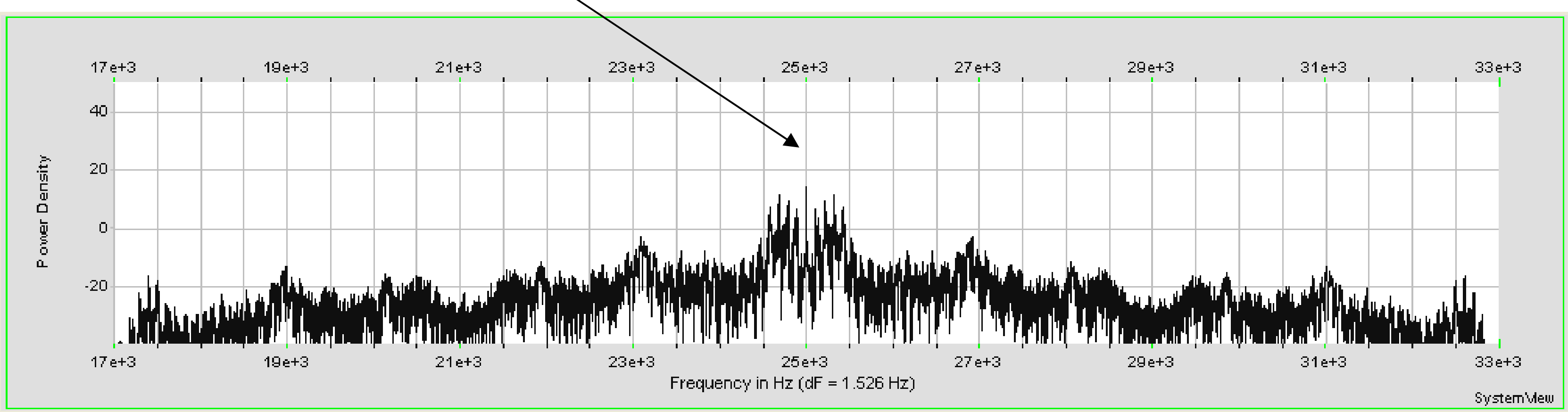


- The analytical signal for double sideband, suppressed carrier amplitude modulation (DSB-SC AM) is:

$$s_{DSB\text{-}SC\ AM}(t) = A_C\, s(t) \cos(2\pi f_C t)$$

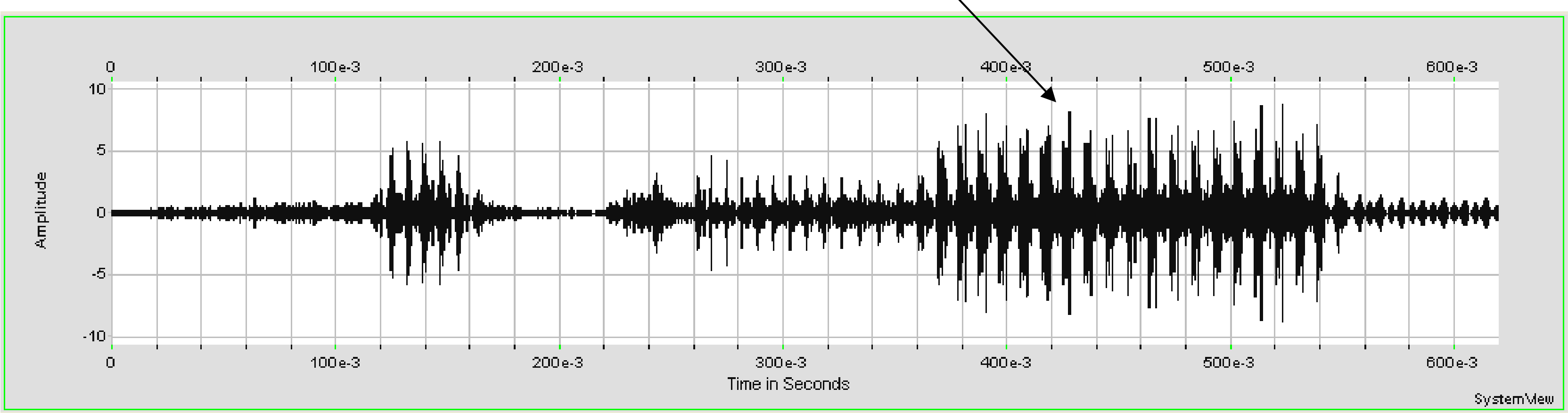
where $A_C$ is the carrier amplitude. The single-sided spectrum of the modulated signal replicates the baseband spectrum as a double-sided spectrum about the carrier frequency but *without* a carrier component.



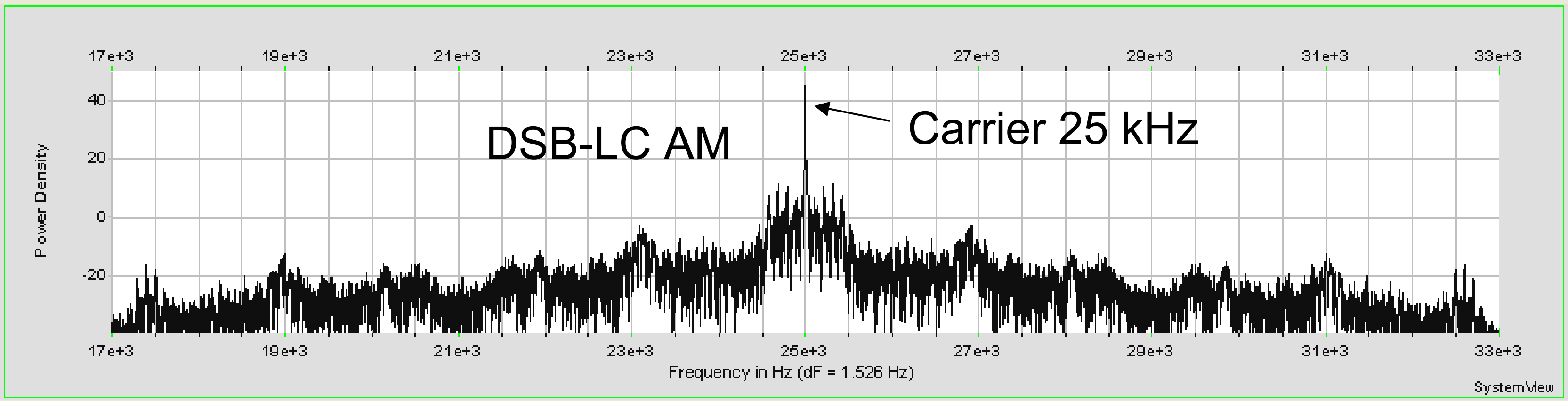
- The analytical signal for double sideband, suppressed carrier amplitude modulation (DSB-SC AM) is:

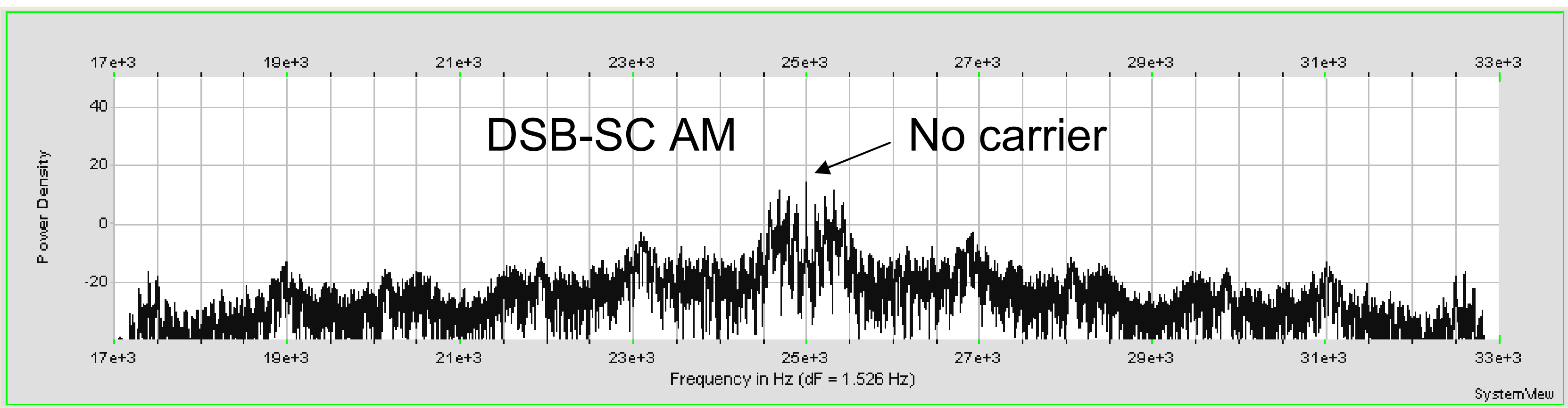
$$s_{\text{DSB-SC AM}}(t) = A_C\, s(t) \cos(2\pi f_C t)$$

where $A_C$ is the carrier amplitude. The modulated signal $s_{\text{DSB-SC AM}}(t)$ looks similar to s(t) but has a temporal but not spectral carrier component.

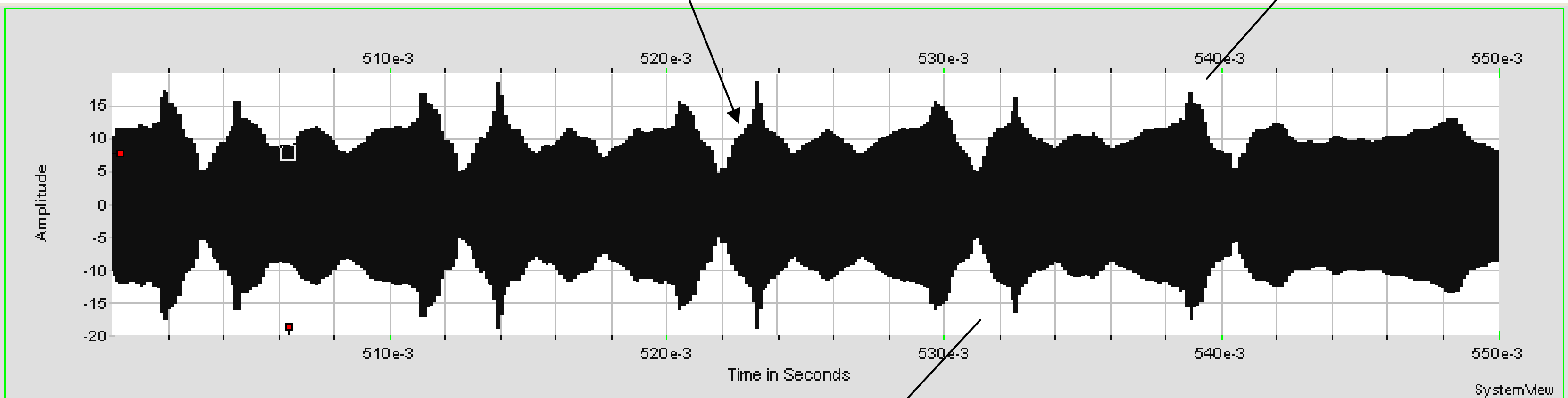


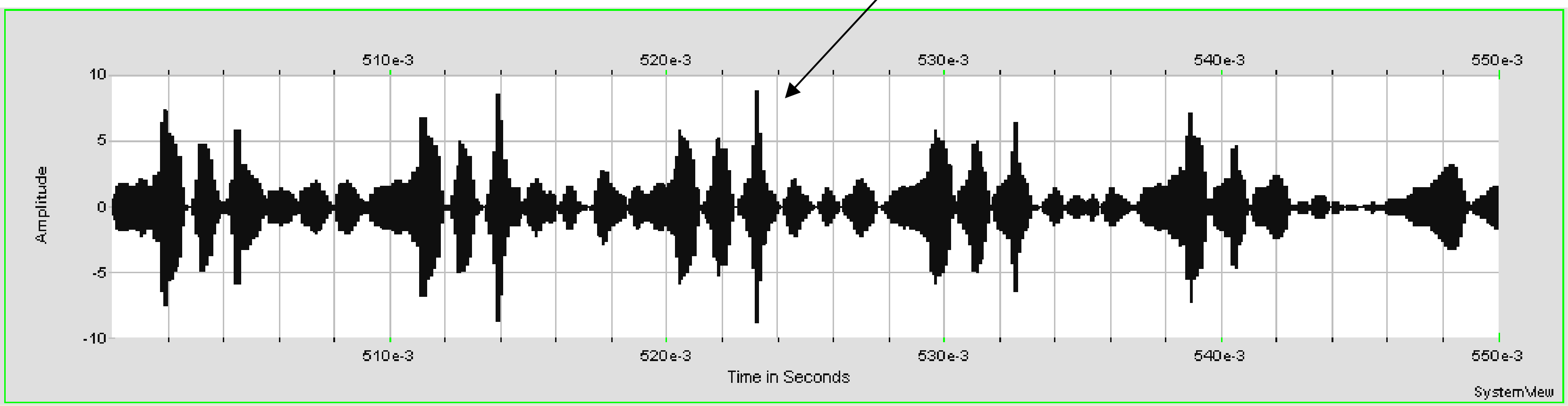
- The DSB-LC AM and the DSB-SC AM modulated signals have the same *sidebands*.



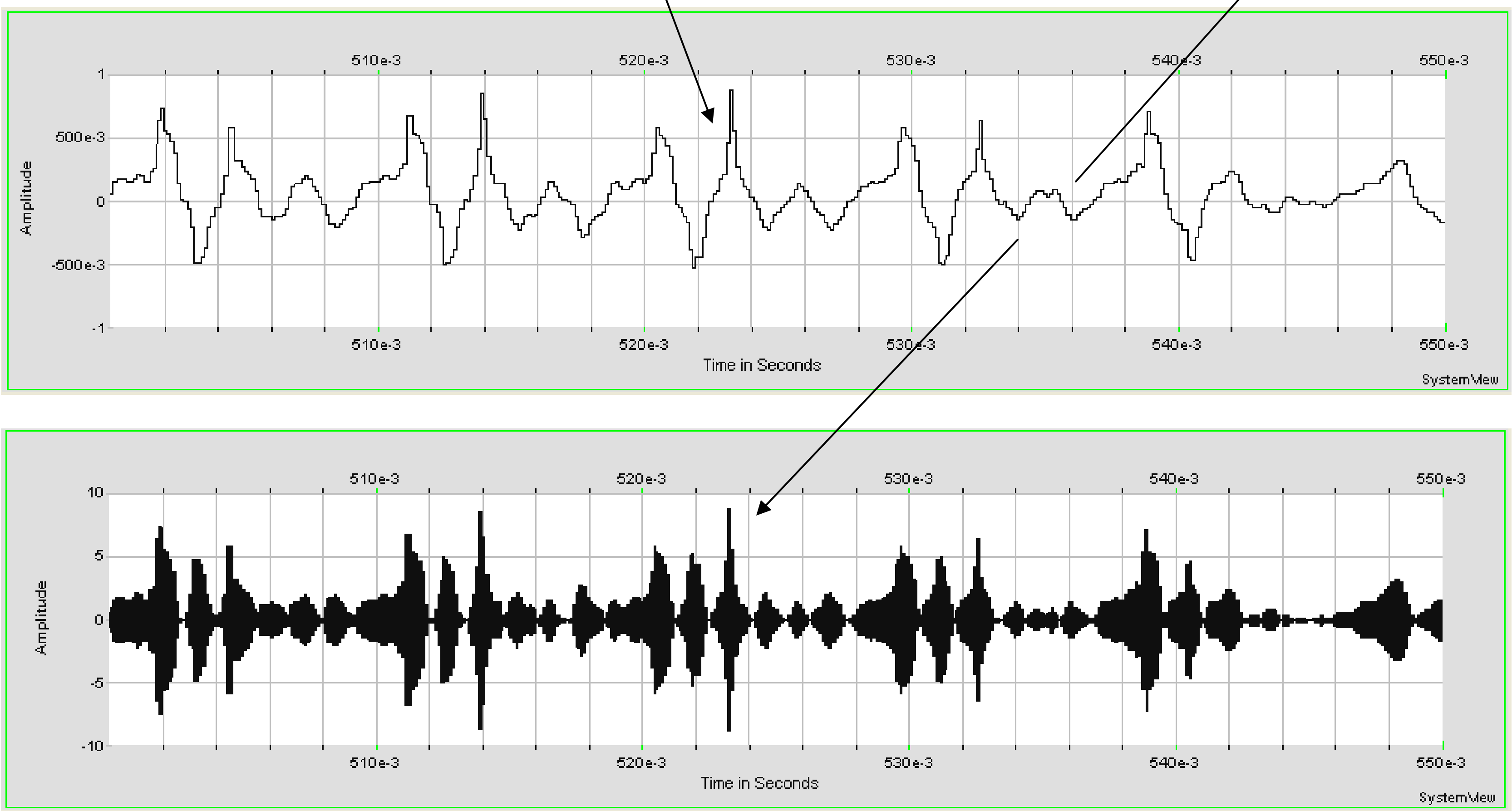



- The modulated DSB-LC AM and the DSC-SC AM signals are different.



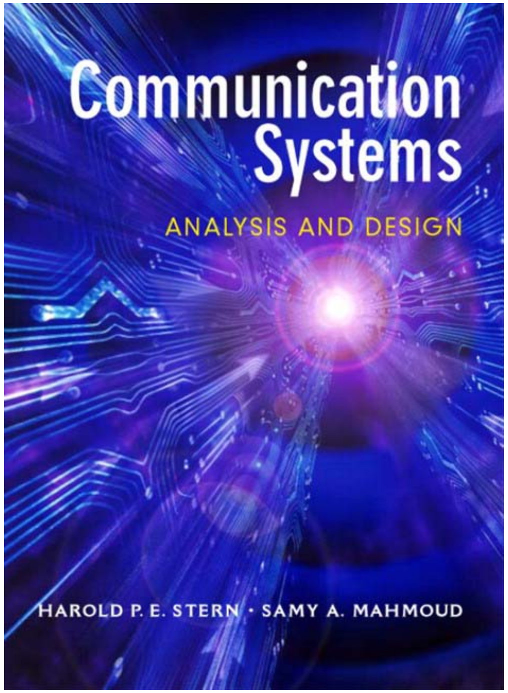
- The modulated DSB-SC AM signal has an *envelope* that follows the polar baseband signal s(t) but not an outer envelope.



# Chapter 6

# Analog Modulation and Demodulation

- *Coherent Demodulation of AM Signals*
- Pages 309-315



- The DSB-SC AM *coherent receiver* has a *bandpass filter* centered at $f_C$ and with a bandwidth of *twice* the bandwidth of s(t) because of the LSB and USB. The output of the multiplier is *lowpass filtered* with a bandwidth equal to the bandwidth of s(t).

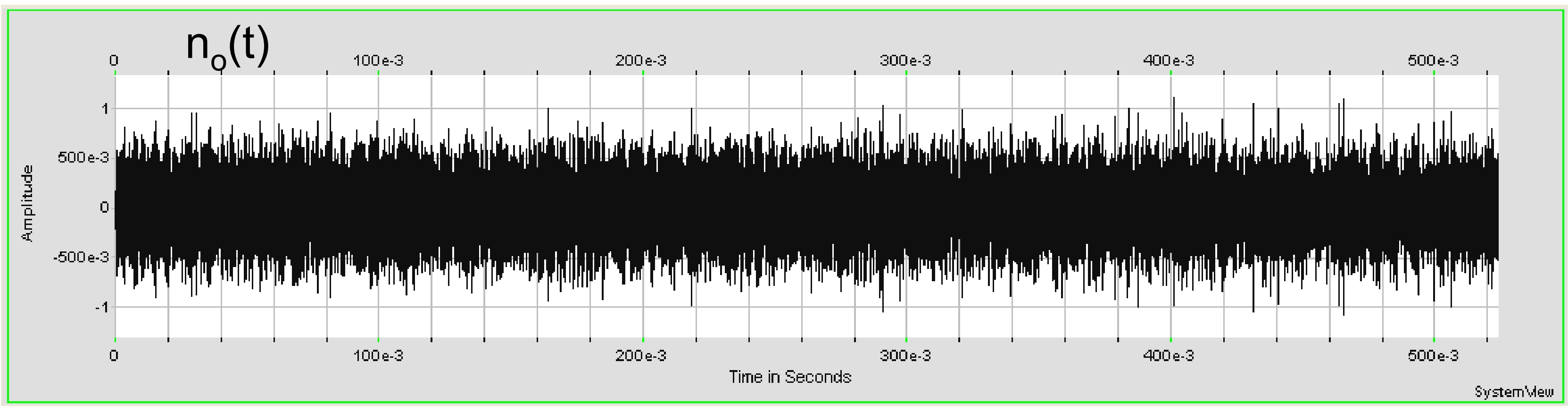
$$r(t) = \gamma\, s_{DSB\text{-}SC}(t) + n(t)$$

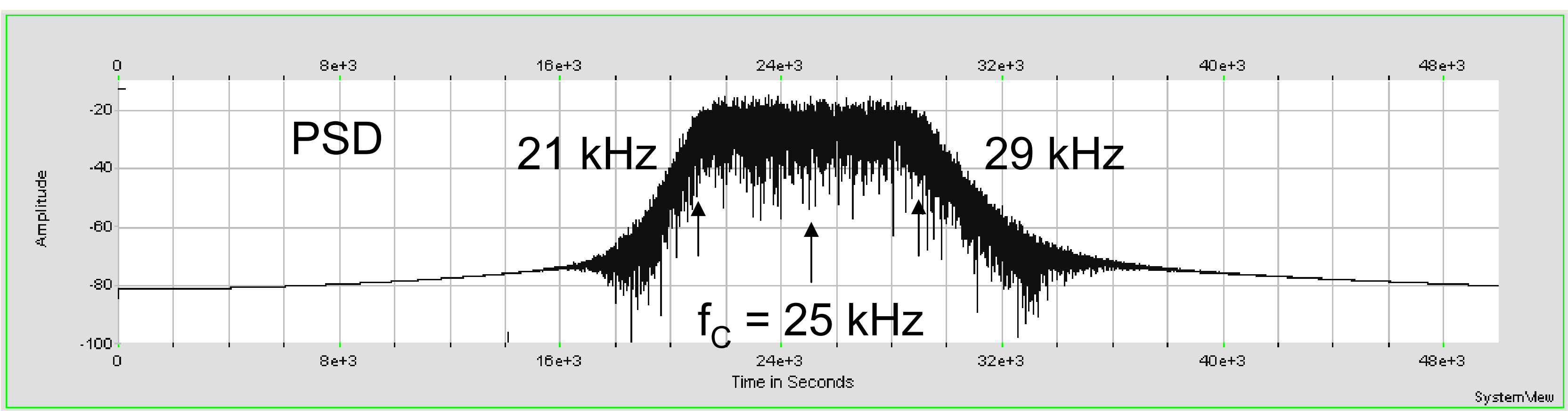
- The DSB-SC AM received signal is $r(t) = \gamma\, s_{DSB\text{-}SC}(t) + n(t)$. The bandpass filter passes the modulated signal but filters the noise:

$$z(t) = \gamma\, s_{DSB\text{-}SC}(t) + n_o(t) \qquad \text{S\&M Eq. 6.3}$$

$n_o(t)$ has a Gaussian distribution. The bandpass filter has a center frequency of $f_C$ = 25 kHz and a -3 dB bandwidth of 8 kHz (25 ± 4 kHz).

- The filter noise $n_o(t)$ has a *flat power spectral density* within the bandwidth of the bandpass filter:



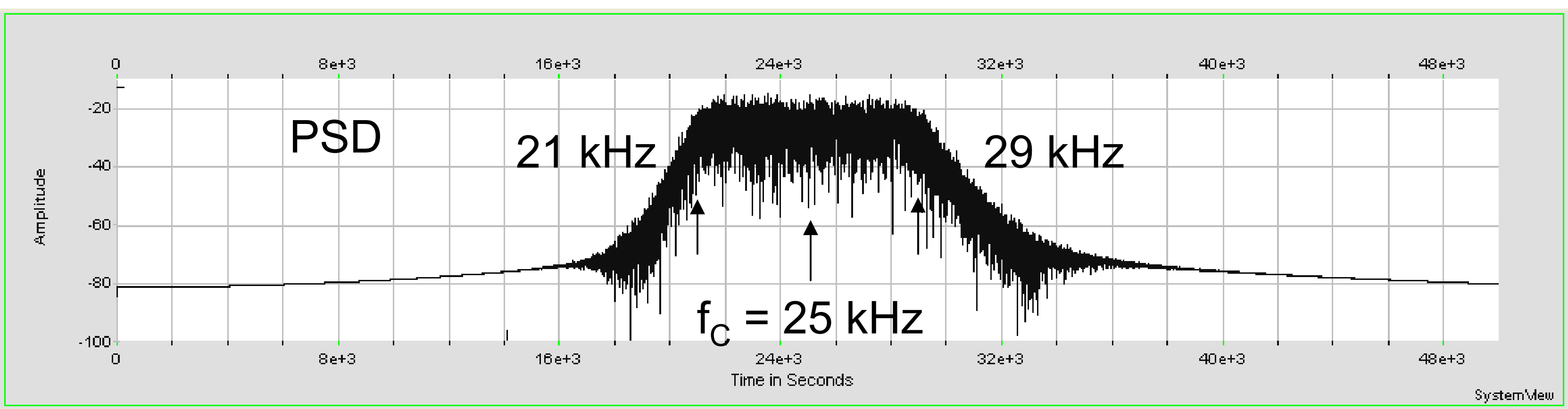



- The filter noise $n_o(t)$ can be described as a *quadrature* representation:

$$n_o(t) = \mathbf{W}(t) \cos(2\pi f_C t) + \mathbf{Z}(t) \sin(2\pi f_C t)$$ S&M Eq. 5.62R

In the coherent receiver the noise is processed:

$$n_o(t) \cos(2\pi f_C t) = \mathbf{W}(t) \cos^2(2\pi f_C t) + \mathbf{Z}(t) \cos(2\pi f_C t) \sin(2\pi f_C t)$$ S&M Eq. 6.5

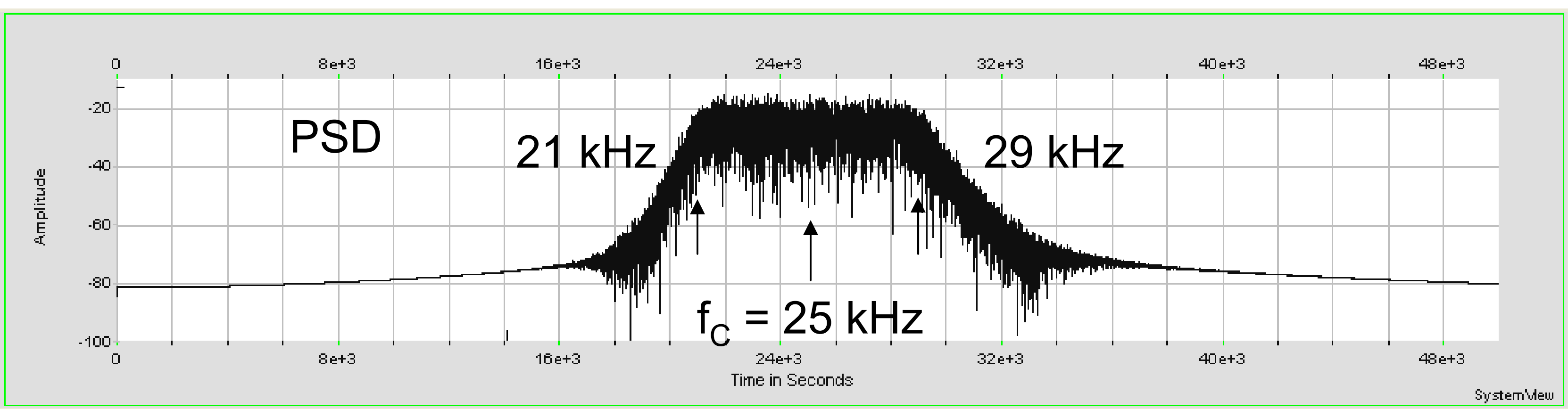


- Applying the *trignometric identity* the filter noise $n_o(t)$ is:

$$n_o(t) \cos(2\pi f_C t) = \tfrac{1}{2}\mathbf{W}(t) + \tfrac{1}{2}\mathbf{W}(t)\cos(4\pi f_C t) + \tfrac{1}{2}\mathbf{Z}(t)\sin(4\pi f_C t) \qquad \text{S\&M Eq. 6.5}$$

After the lowpass filter in the receiver the demodulated signal is:

$$s_{demod}(t) = \tfrac{1}{2}\gamma A_C\, s(t) + \tfrac{1}{2}\mathbf{W}(t) \qquad \text{S\&M Eq. 6.7}$$

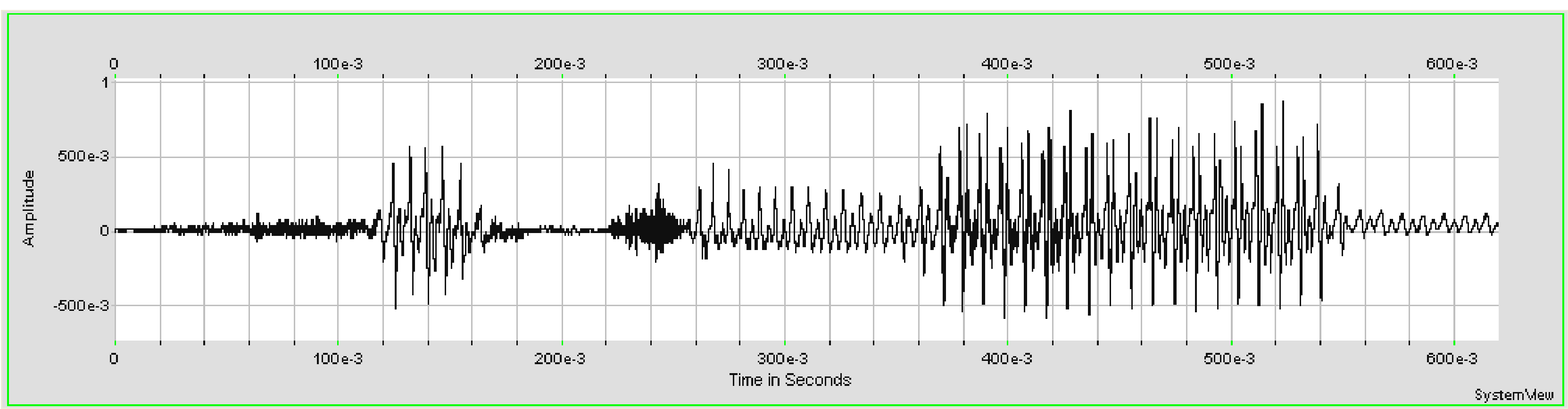


• The transmitted DSB-SC AM signal is:

$$s_{DSB\text{-}SC\ AM}(t) = A_C\, s(t) \cos(2\pi f_C t)$$

The average normalized bi-sided power of $s_{DSB\text{-}SC}(t)$ is found in the spectral domain with $S(f) = \mathbb{F}(s(t))$:

$$P_{trans} = A^2 \int \left| \frac{1}{2}\left[S(f - f_C) + S(f + f_C)\right] \right|^2 df \qquad \text{S\&M Eq. 6.8}$$

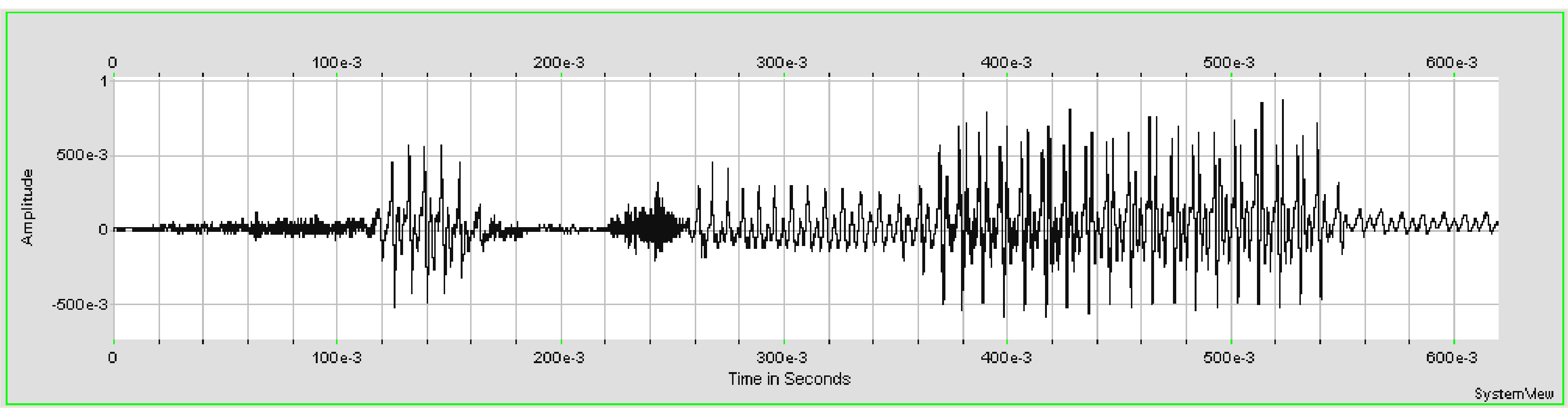


- The dual-sided spectral do not *overlap* (at zero frequency) and the *cross terms* are zero so that:

$$P_{trans} = A^2 \int \left| \frac{1}{2} \left[ S_{DSB\text{-}SC}(f - f_C) + S_{DSB\text{-}SC}(f + f_C) \right] \right|^2 df$$

$$P_{trans} = \frac{A^2}{2} P_s \qquad \text{S\&M Eq. 6.9}$$

where $P_s$ is the average normalized power of s(t).

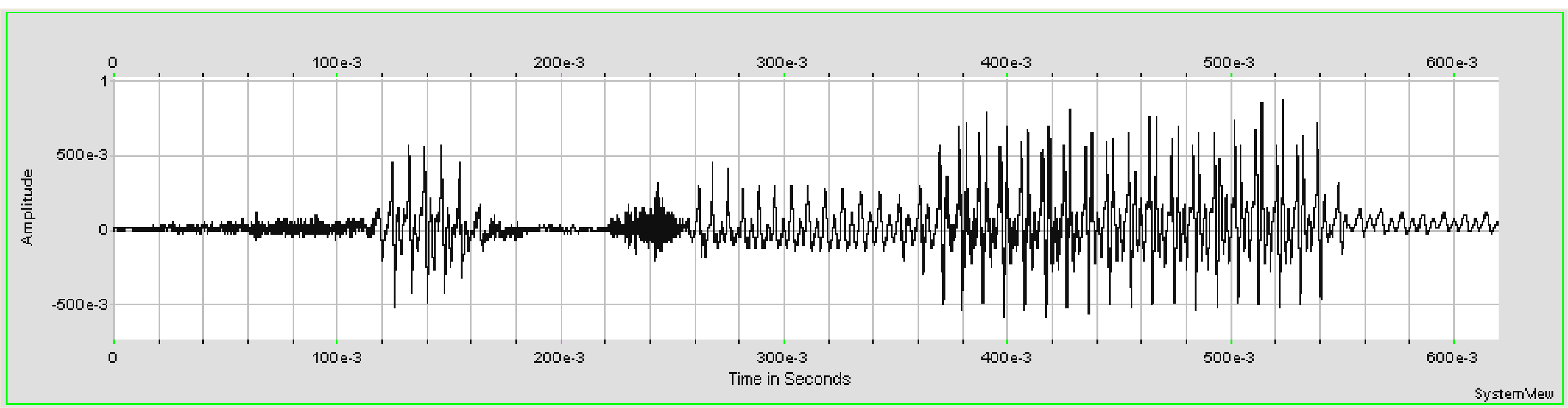


- The average normalized power of s(t) is found in the spectral domain:

$$P_s = \int | S(f) |^2 df = \int | S(f + f_C) |^2 df \qquad \text{S\&M Eq. 6.10}$$

In a *noiseless channel* the power in the demodulated DSB-SC AM signal is:

$$P_{demod,noiseless} = \frac{1}{4}\gamma^2 A^2 P_s = \frac{\gamma^2}{2} P_{trans} \qquad \text{S\&M Eq. 6.11}$$

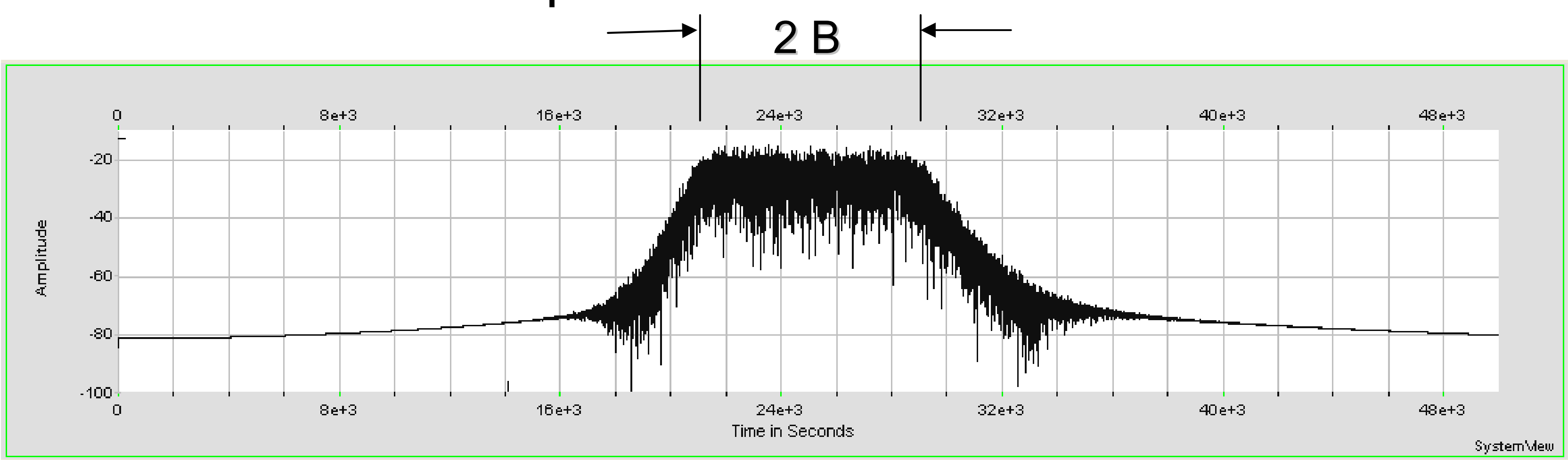


- The average normalized power of the processed noise is:

$$P_{\text{processed noise}} = \frac{1}{4} N_o (2\,B)$$

The signal-to-noise power ratio then is:

$$SNR_{\text{coherent DSB-SC}} = \frac{\dfrac{\gamma^2}{2} P_{\text{trans}}}{\dfrac{1}{4} N_o (2\,B)} = \frac{\gamma^2\, P_{\text{trans}}}{N_o\, B} \qquad \text{S\&M Eq. 6.12}$$



- The DSB-SC AM coherent receiver requires a phase and frequency synchronous reference signal. If the reference signal has a *phase error φ* then:

$$\mathrm{SNR}_{\text{coherent DSB-SC phase error}} = \frac{\gamma^2 \cos^2 \varphi \, P_{\text{trans}}}{N_o \, B}$$

S&M Eq. 6.17

- The DSB-SC AM coherent receiver requires a phase and frequency synchronous reference signal. If the reference signal has a *frequency error Δf* then:

$$
\begin{aligned}
S_{\text{demod frequency error}}(t) = {} & \tfrac{1}{2}\, \gamma\, A_C\, s(t) \cos(2\pi\, \Delta f\, t) \\
& + \tfrac{1}{2}\, \mathbf{X}(t) \cos(2\pi\, \Delta f\, t) \\
& + \tfrac{1}{2}\, \mathbf{Y}(t) \sin(2\pi\, \Delta f\, t)
\end{aligned}
$$

S&M Eq. 6.18

- Although the noise component remains the same, the *amplitude* of the demodulated signal varies with Δf:

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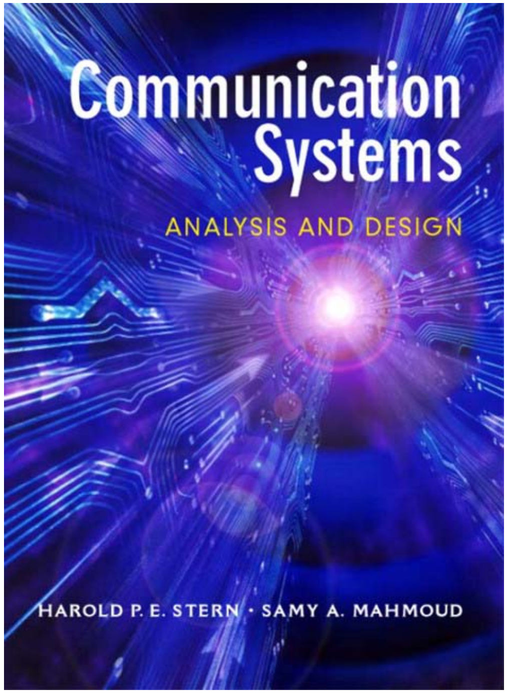
- *Non-coherent Demodulation of AM Signals*
- Pages 315-326



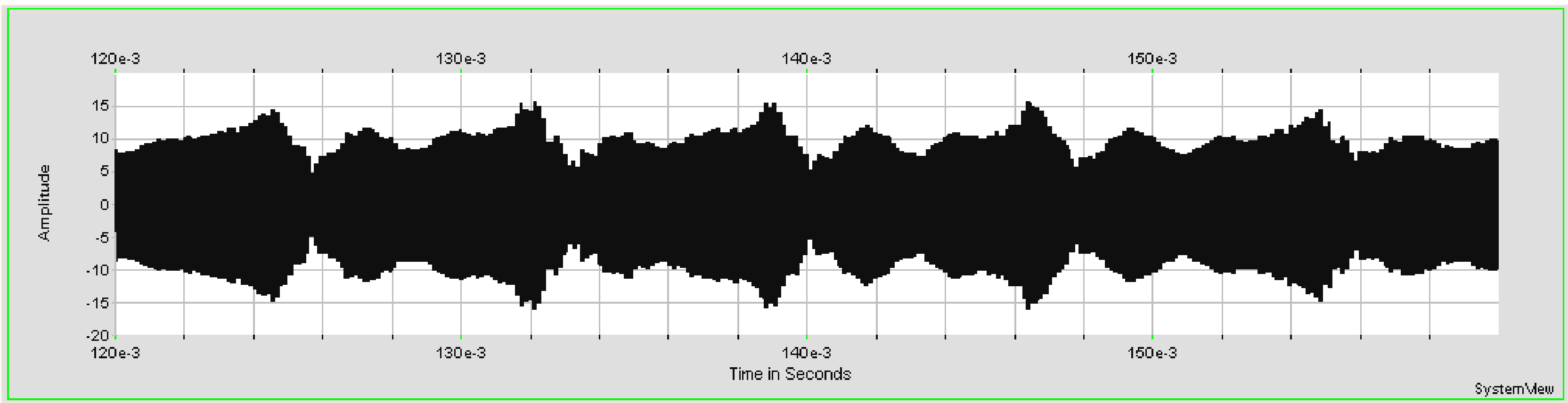
- The *non-coherent AM* (DSB-LC) receiver uses an *envelope detector* implemented as a *semiconductor diode* and a *low-pass filter*:

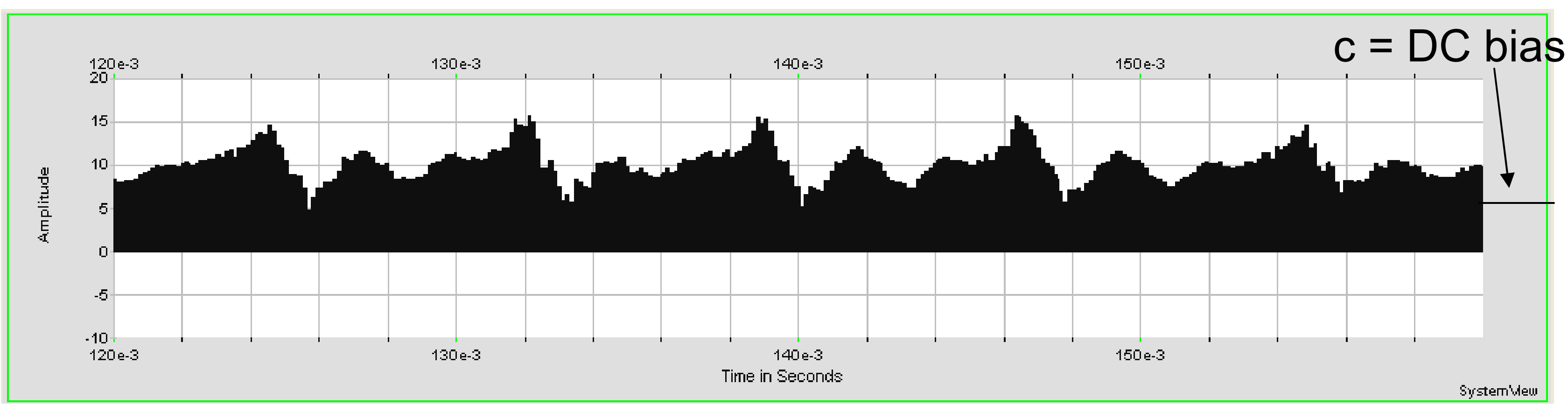
  The DSB-LC AM analytical signal is:

$$s_{\text{DSB-LC AM}}(t) = A_C\,(c + s(t))\cos(2\pi f_C t)$$

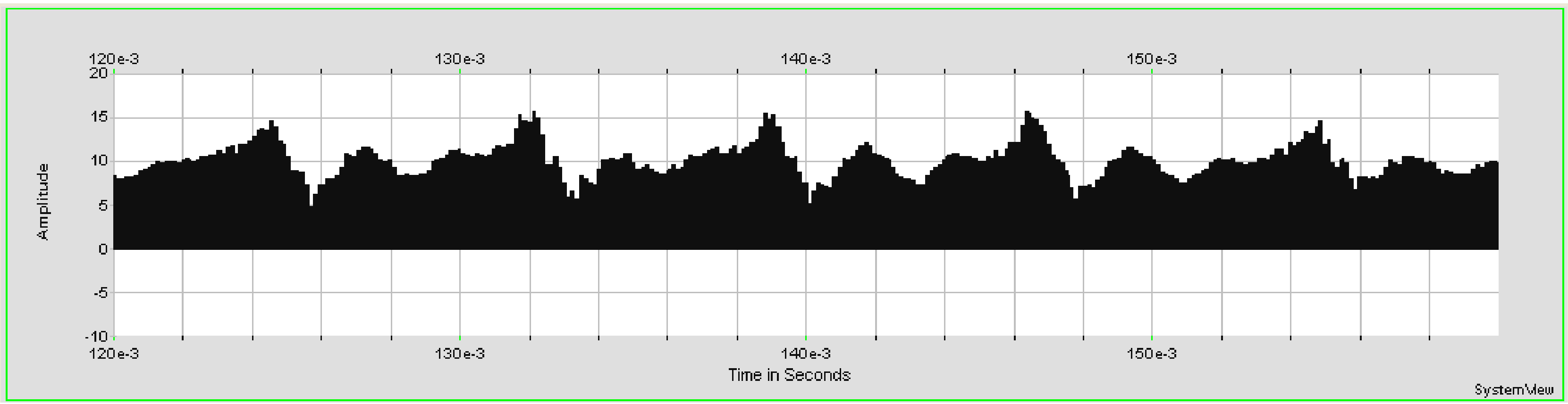
  where c is the DC bias (offset).

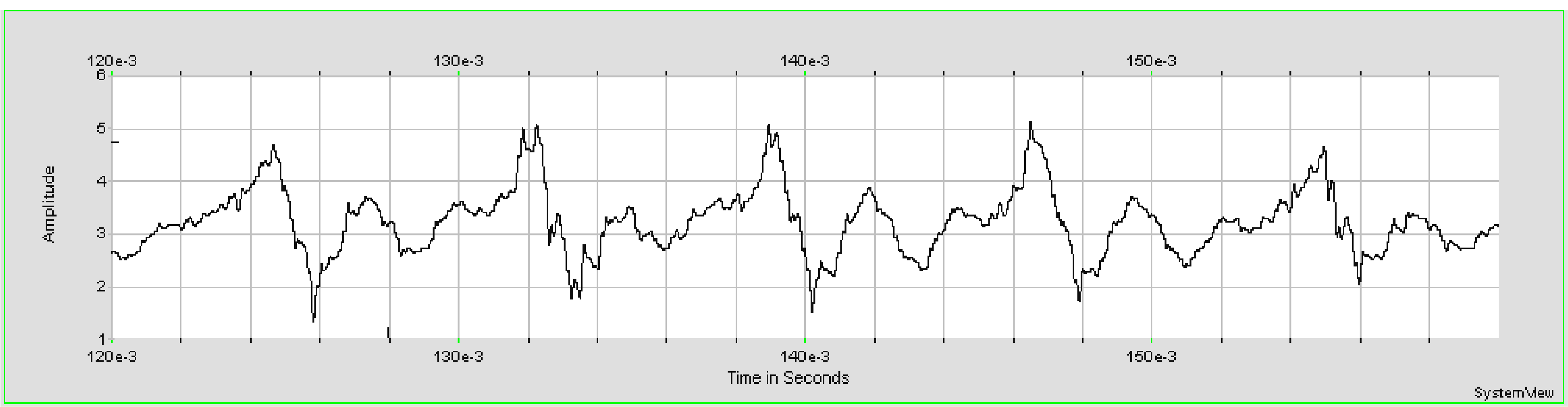
- The envelope detector is a *half-wave rectifier* and provides a *DC bias* ($c$) to the processed DSB-LC AM signal :



- The output of the half-wave diode rectifier is low-pass filtered to remove the carrier frequency and outputs the envelope which is the information:



- The DSB-LC AM signal can be decomposed as:

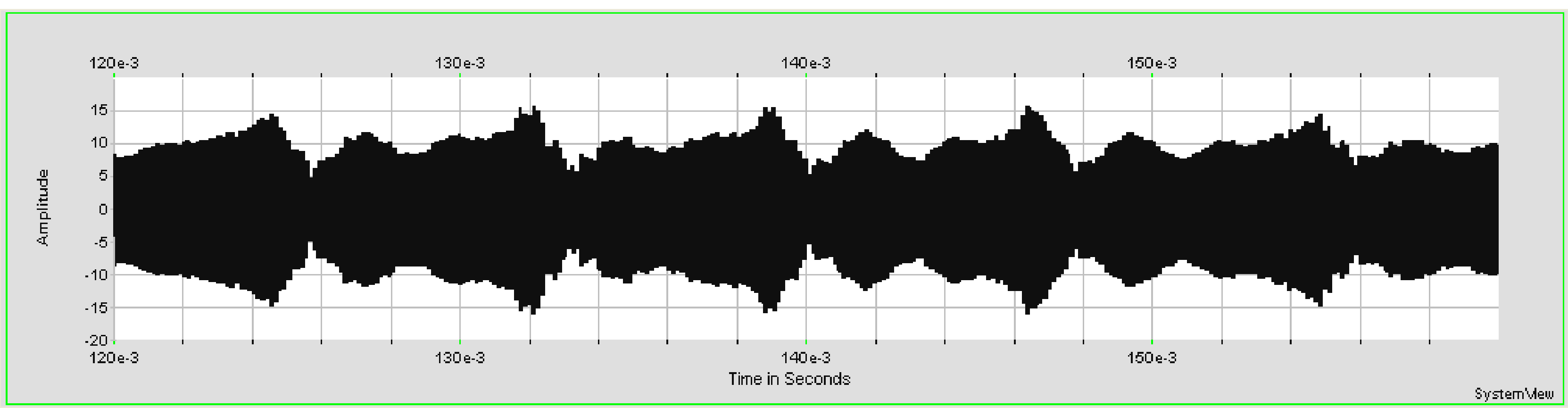
$$s_{DSB\text{-}LC\ AM}(t) = s(t) \cos(2\pi f_C t) + A_C c \cos(2\pi f_C t)$$

S&M Eq. 6.20R

The average normalized power of the information term:

$$P_{\text{info term}} = \frac{A_C^2}{2} P_S$$

S&M Eq. 6.23

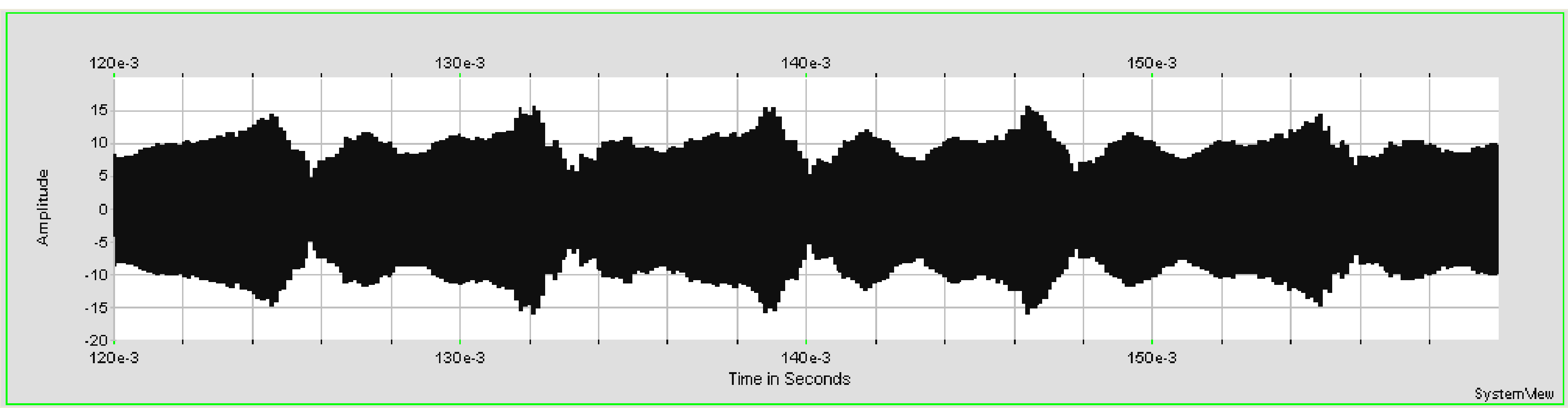


- The average normalized transmitted power is:

$$P_{\text{carrier term}} = \frac{1}{T}\int_0^T [A_C c \cos(2\pi f_C t)]^2\, dt$$

$$P_{\text{carrier term}} = \frac{A_C^2 c^2}{2} \qquad \text{S\&M Eq. 6.24}$$

Since s(t) + c must be >= 0 to avoid distortion in the DSB-LC AM signal: $c \geq |\min[s(t)]|$ or $c^2 \geq s^2(t)$ for all t.



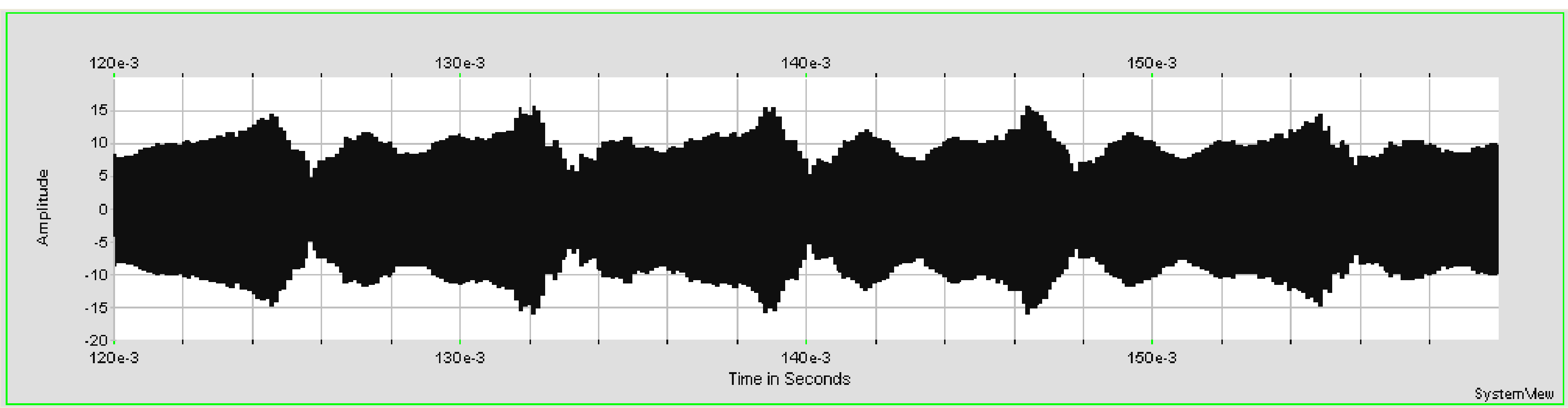
- Therefore $c^2 \geq P_s$ and for DSB-LC AM:

$$P_{\text{carrier term}} \geq P_{\text{info term}} \qquad \text{S\&M Eq. 6.28}$$

The power efficiency η of a DSB-LC AM signal is:

$$\eta = \frac{P_{\text{info term}}}{P_{\text{carrier term}} + P_{\text{info term}}} = \frac{P_{\text{info term}}}{P_{\text{trans DSB-LC AM term}}} \leq 0.5$$

S&M Eq. 6.29



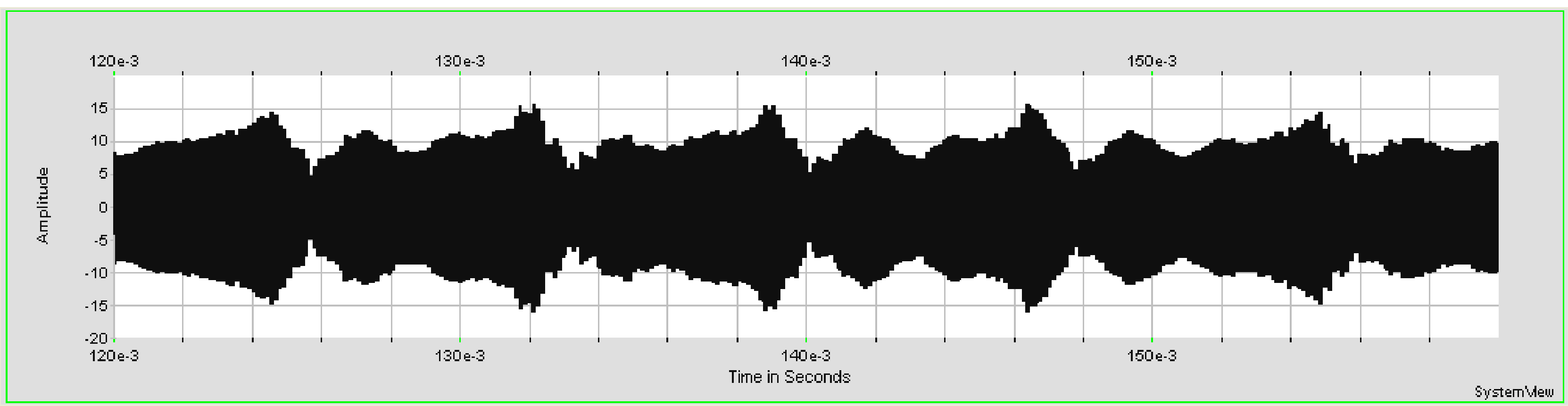
- The DSB-LC AM signal *wastes* at least half the transmitted power because the power in the carrier term has no information:

$$P_{\text{carrier term}} \geq P_{\text{info term}} \qquad \eta \leq 0.5$$

The *modulation index* m is defined as:

$$m = \frac{\max[s(t)+c]-\min[s(t)+c]}{\max[s(t)+c]+\min[s(t)+c]}$$

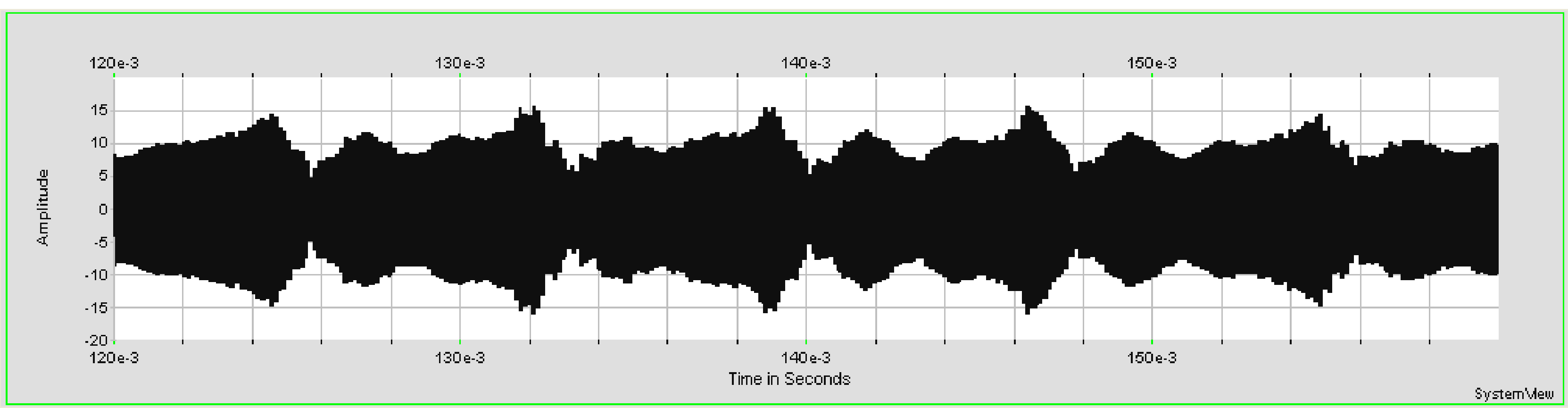
S&M Eq. 6.30



- The *modulation index* m defines the power efficiency but m must be less than 1. If m > 1 then min [s(t) + c] < 0 and distortion occurs.

$$m = \frac{\max\left[s(t)+c\right]-\min\left[s(t)+c\right]}{\max\left[s(t)+c\right]+\min\left[s(t)+c\right]}$$

S&M Eq. 6.30



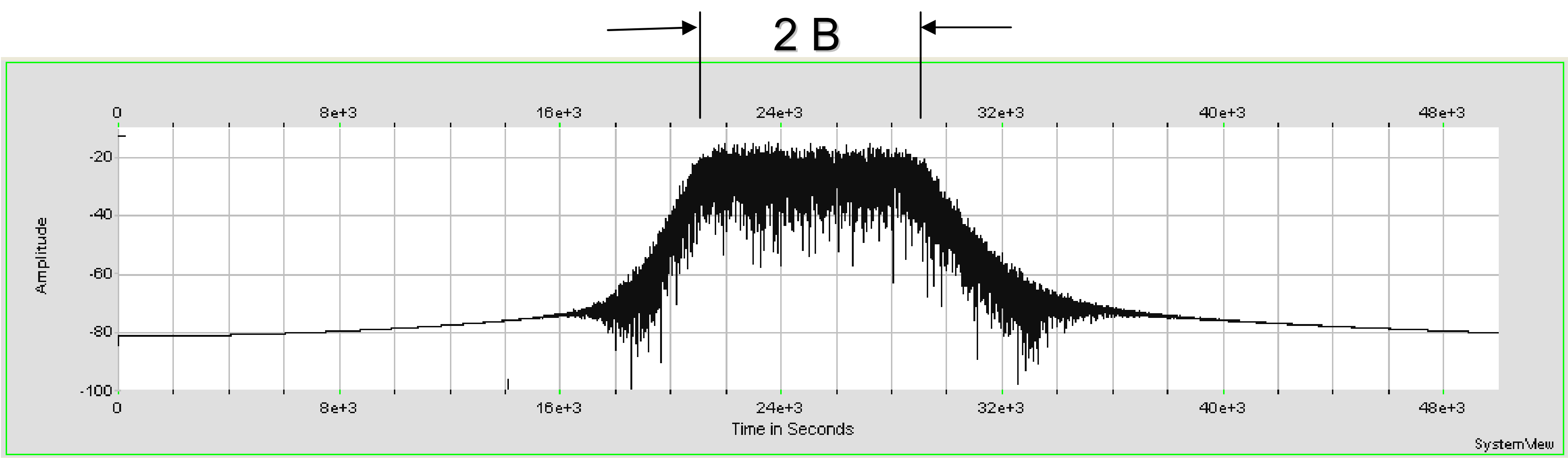
- The average normalized power of the demodulation noiseless DSB-LC AM signal is:

$$P_{\text{demod, noiseless}} = 2\,\gamma^2\,P_{\text{info term}}$$ S&M Eq. 6.39

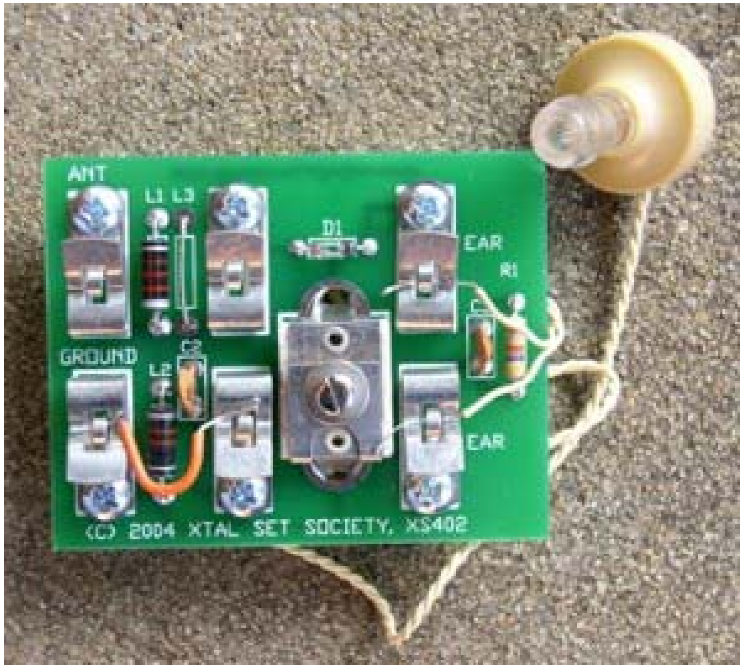
Then the signal-to-noise power ratio for the DSB-LC AM signal is:

S&M Eq. 6.40

$$SNR_{\text{noncoherent DSB-LC}} = \frac{2\,\gamma^2\,P_{\text{info term}}}{N_o\,(2\,B)} = \frac{\gamma^2\,P_{\text{trans DSB-LC}}}{N_o\,B}\,\eta$$



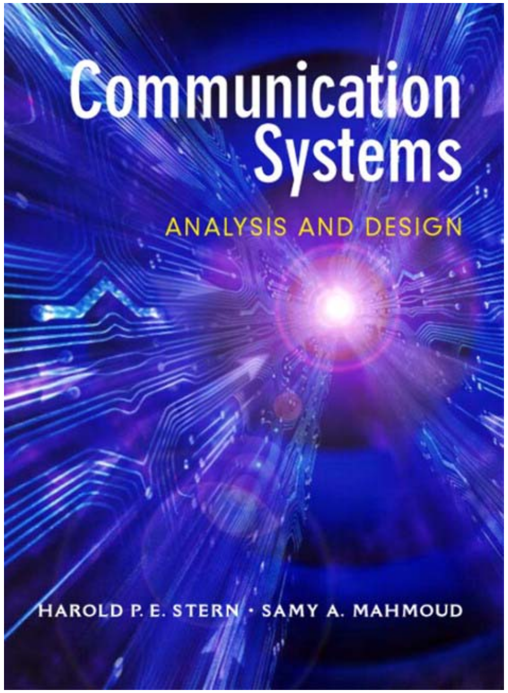
- The *non-coherent AM* (DSB-LC) receiver is the *crystal radio* which needs no batteries! Power for the high-impedance ceramic earphone is obtained directly from the transmitted signal. For simplicity, the RF BPF is omitted and the audio frequency filter is a simple *RC* network.



# Chapter 6

# Analog Modulation and Demodulation

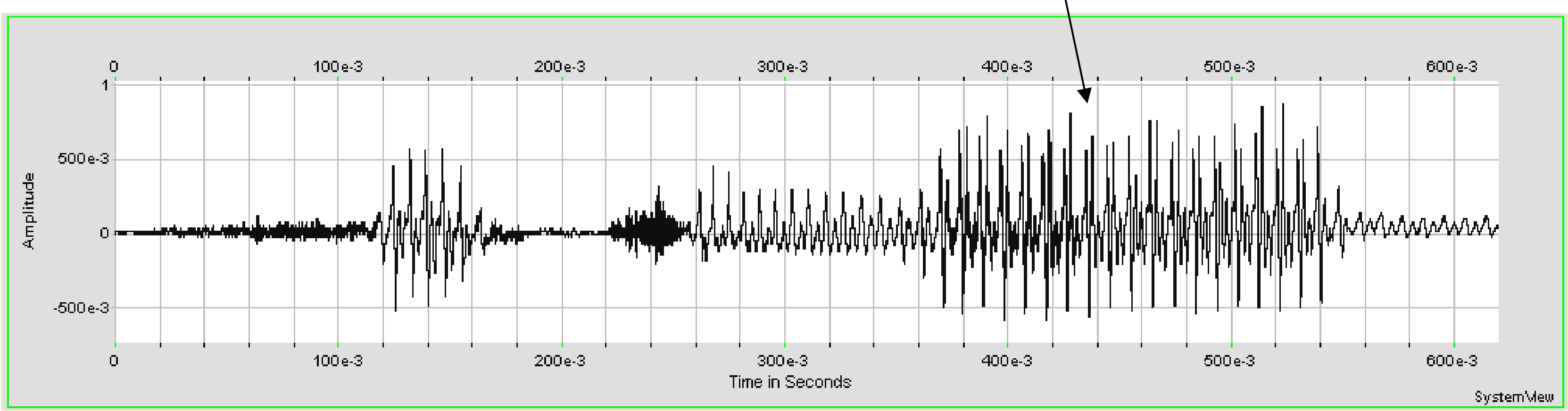
- *Frequency Modulation and Phase Modulation*
- Pages 334-343



- The analytical signal for an *analog phase modulated* (PM) signal is:

$$s_{PM}(t) = A_C \cos [2\pi f_C t + \alpha s(t)] \qquad \text{S\&M Eq. 6.53}$$

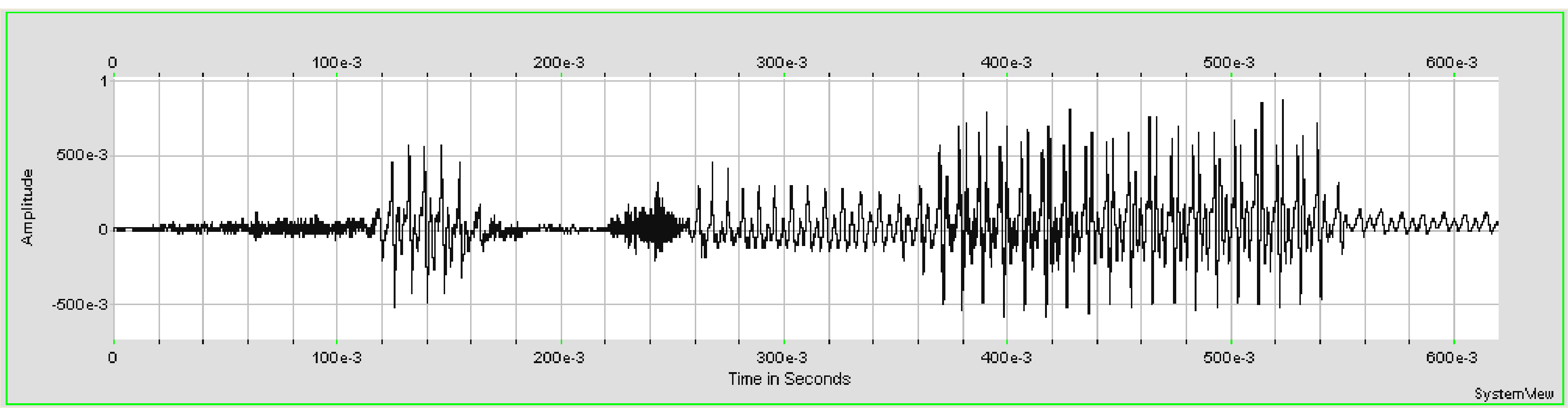
where $\alpha$ is the *phase modulation constant* rad/V and $A_C$ is the carrier amplitude. The continuous analog signal *s(t)* is a baseband signal with the information content (voice or music) to be transmitted.



- The analytical signal for an *analog frequency modulated* (FM) signal is:

$$s_{FM}(t) = A_c \cos\{\, 2\pi\, [f_C + k\, s(t)]\, t + \varphi]$$ S&M Eq. 6.53

where *k* is the *frequency modulation constant* Hz / V, $A_C$ is the carrier amplitude and φ is the initial phase angle at t = 0. The continuous analog signal *s(t)* is a baseband signal with the information content.



- The *instantaneous phase* of the PM signal is:

$$\Psi_{PM}(t) = 2\pi f_C t + \alpha s(t) \qquad \text{S\&M Eq. 6.56}$$

The *instantaneous phase* of the FM signal is:

$$\Psi_{FM}(t) = 2\pi [f_C + k s(t)] t + \varphi] \qquad \text{S\&M Eq. 6.57}$$

The instantaneous phase is also call the *angle* of the signal. The *instantaneous frequency* is the time rate of change of the angle:

$$f(t) = (1/2\pi) d\Psi(t) / dt \qquad \text{S\&M Eq. 6.58}$$

- The instantaneous frequency of the unmodulated carrier signal is:

$$f_{carrier}(t) = d\Psi_{carrier}(t) / dt = d/dt \{2\pi f_C t + \varphi\} \qquad \text{S\&M Eq. 6.59}$$

The instantaneous phase is also:

$$\Psi(t) = \int_{-\infty}^{t} f(\lambda)\, d\lambda = \int_{0}^{t} f(\lambda)\, d\lambda + \varphi \qquad \text{S\&M Eq. 6.60}$$

There are *practical limits* on instantaneous frequency and instantaneous phase. To avoid ambiguity and distortion in FM signals due to *phase wrapping*:

$$k\, s(t) \le f_C \quad \text{for all } t \qquad \text{S\&M Eq. 6.61}$$

- To avoid ambiguity and distortion in PM signals due to phase wrapping:

$$-\pi < \alpha\, s(t) \leq \pi \text{ radians for all } t \qquad \text{S\&M Eq. 6.61}$$

Since FM and PM are both change the angle of the carrier signal as a function of the analog information signal s(t), FM and PM are called *angle modulation*.

For example, is this signal FM, PM or neither:

$$x(t) = A_C \cos \{ 2\pi f_C t + \int_{-\infty}^{t} k\, s(\lambda)\, d\lambda + \varphi \} \qquad \text{S\&M Eq. 6.60}$$

- The instantaneous phase of the signal is:

$$\Psi_x(t) = 2\pi\, f_C t + \int_{-\infty}^{t} k\, s(\lambda)\, d\lambda + \varphi \qquad \text{S\&M Eq. p. 336}$$

which is *not* a linear function of s(t) so the signal is *not PM*. The instantaneous frequency of the signal is:

$$f_x(t) = (1/2\pi)\, d\Psi_x(t) / dt = f_C + k\, s(t) / 2\pi$$

and the frequency difference $f_x - f_C$ is a linear function of s(t) so the signal is FM.

The *maximum phase deviation* of a PM signal is max | αs(t) |. The *maximum frequency deviation* of a FM signal is Δf = max | k s(t) |.

- The spectrum of a PM or FM signal can be developed as follows: S&M Eqs. 6.64 through 6.71

$$v(t) = A_C \sin(2\pi f_C t + \beta \sin 2\pi f_m t)$$

$$v(t) = \mathrm{Re}\{\exp(j 2\pi f_C t + j \beta \sin 2\pi f_m t)\}$$

*now* $\exp(j 2\pi f_C t + j \beta \sin 2\pi f_m t) =$

$$\cos(2\pi f_C t + \beta \sin 2\pi f_m t) + j \sin(2\pi f_C t + \beta \sin 2\pi f_m t)$$

$$v(t) = \mathrm{Im}\{A_C \exp(2\pi f_C t + j\beta \sin 2\pi f_m t)\}$$

*now* $\exp(j \beta \sin 2\pi f_m t) = \sum_{n=-\infty}^{\infty} c_n \exp(j 2\pi n f_m t)$

*after further development*

Bessel function of the first kind

$$\exp(j \beta \sin 2\pi f_m t) = \sum_{n=-\infty}^{\infty} J_n(\beta) \exp(j 2\pi n f_m t)$$

- Bessel functions of the first kind $J_n(\beta)$ are tabulated for FM with single tone $f_m$ angle modulation (S&M Table 6.1):

β

n

**Table 6-1** Values of Bessel Function of the First Kind $J_n(\beta)$ for Various Values of $n$ and $\beta$

| $n$ | $\beta = 1$ | $\beta = 2$ | $\beta = 3$ | $\beta = 4$ | $\beta = 5$ | $\beta = 6$ | $\beta = 7$ | $\beta = 8$ | $\beta = 9$ |
|---|---|---|---|---|---|---|---|---|---|
| 0 | 0.7652 | 0.2239 | −0.2601 | −0.3971 | −0.1776 | 0.1506 | 0.3001 | 0.1717 | −0.0903 |
| 1 | 0.4401 | 0.5767 | 0.3391 | −0.0660 | −0.3276 | −0.2767 | −0.0047 | 0.2346 | 0.2453 |
| 2 | 0.1149 | 0.3528 | 0.4861 | 0.3641 | 0.0466 | −0.2429 | −0.3014 | −0.1130 | 0.1448 |
| 3 | 0.0196 | 0.1289 | 0.3091 | 0.4302 | 0.3648 | 0.1148 | −0.1676 | −0.2911 | −0.1809 |
| 4 | 0.0025 | 0.0340 | 0.1320 | 0.2811 | 0.3912 | 0.3576 | 0.1578 | −0.1054 | −0.2655 |
| 5 | 0.0002 | 0.0070 | 0.0430 | 0.1321 | 0.2611 | 0.3621 | 0.3479 | 0.1858 | −0.0550 |
| 6 | * | 0.0012 | 0.0114 | 0.0491 | 0.1310 | 0.2458 | 0.3392 | 0.3376 | 0.2043 |
| 7 | * | 0.0002 | 0.0025 | 0.0152 | 0.0534 | 0.1296 | 0.2336 | 0.3206 | 0.3275 |
| 8 | * | * | 0.0005 | 0.0040 | 0.0184 | 0.0565 | 0.1280 | 0.2235 | 0.3051 |
| 9 | * | * | 0.0001 | 0.0009 | 0.0055 | 0.0212 | 0.0589 | 0.1263 | 0.2149 |
| 10 | * | * | * | 0.0002 | 0.0015 | 0.0070 | 0.0235 | 0.0608 | 0.1247 |
| 11 | * | * | * | * | 0.0004 | 0.0020 | 0.0083 | 0.0256 | 0.0622 |
| 12 | * | * | * | * | 0.0001 | 0.0005 | 0.0027 | 0.0096 | 0.0274 |
| 13 | * | * | * | * | * | 0.0001 | 0.0008 | 0.0033 | 0.0108 |
| 14 | * | * | * | * | * | * | 0.0002 | 0.0010 | 0.0039 |
| 15 | * | * | * | * | * | * | 0.0001 | 0.0003 | 0.0013 |
| 16 | * | * | * | * | * | * | * | 0.0001 | 0.0004 |
| 17 | * | * | * | * | * | * | * | * | 0.0001 |
| 18 | * | * | * | * | * | * | * | * | * |
| 19 | * | * | * | * | * | * | * | * | * |

*The values designated by * are all $<10^{-4}$. The values in each column below the bar are all less than 0.1. Practically speaking, such values produce negligible spectral components.

- For single tone $f_m$ angle modulation the spectrum is periodic and infinite in extent:

$$v(t) = A_C \sum_{n=-\infty}^{\infty} J_n(\beta) \sin[2\pi (n f_m + f_c)\, t] \qquad \text{S\&M Eq. 6.72}$$

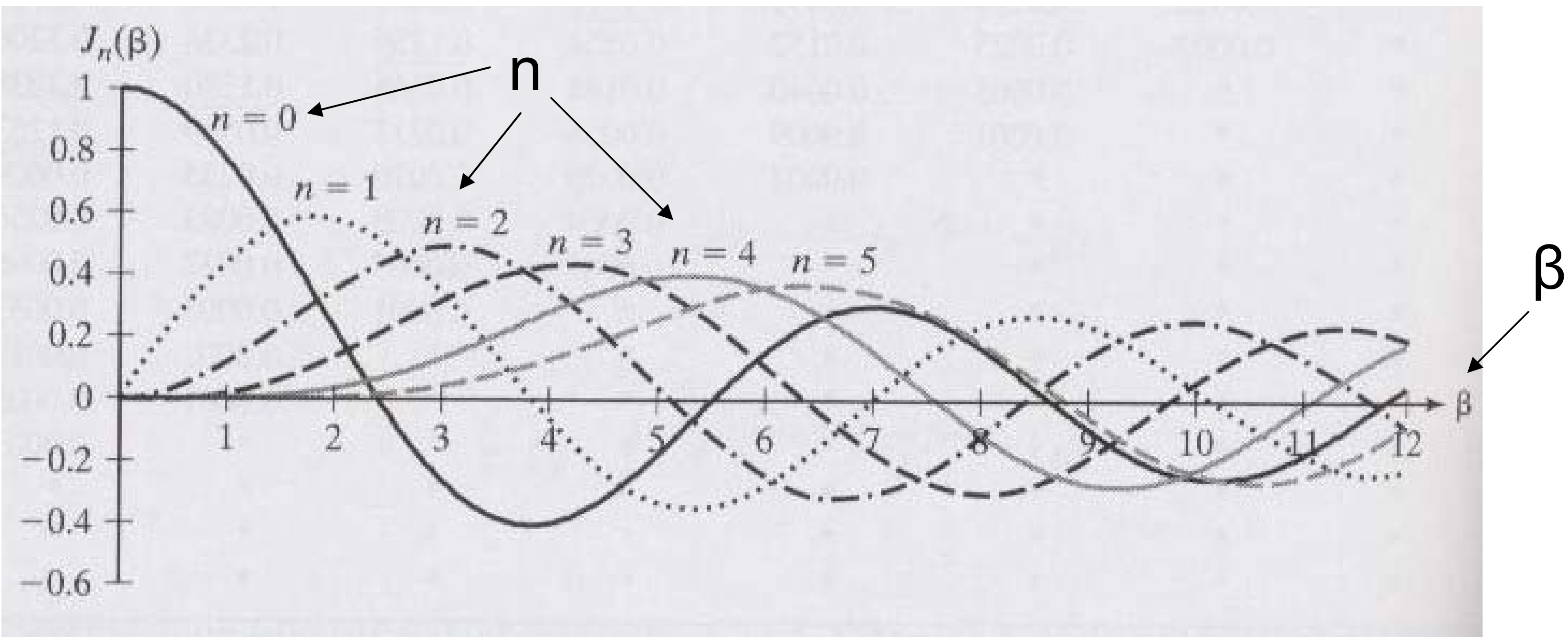



**Figure 6-21** Bessel function of the first kind for various values of $n$ (order).

- The *complexity* of the Bessel function solution for the spectrum of a single tone angle modulation can be simplified by the *Carson's Rule approximation* for the bandwidth *B*. Since $\beta = \Delta f / f_m$:

$$B = 2(\beta + 1) f_m = 2(\Delta f + f_m) \text{ Hz} \qquad \text{S\&M Eq. 6.74}$$

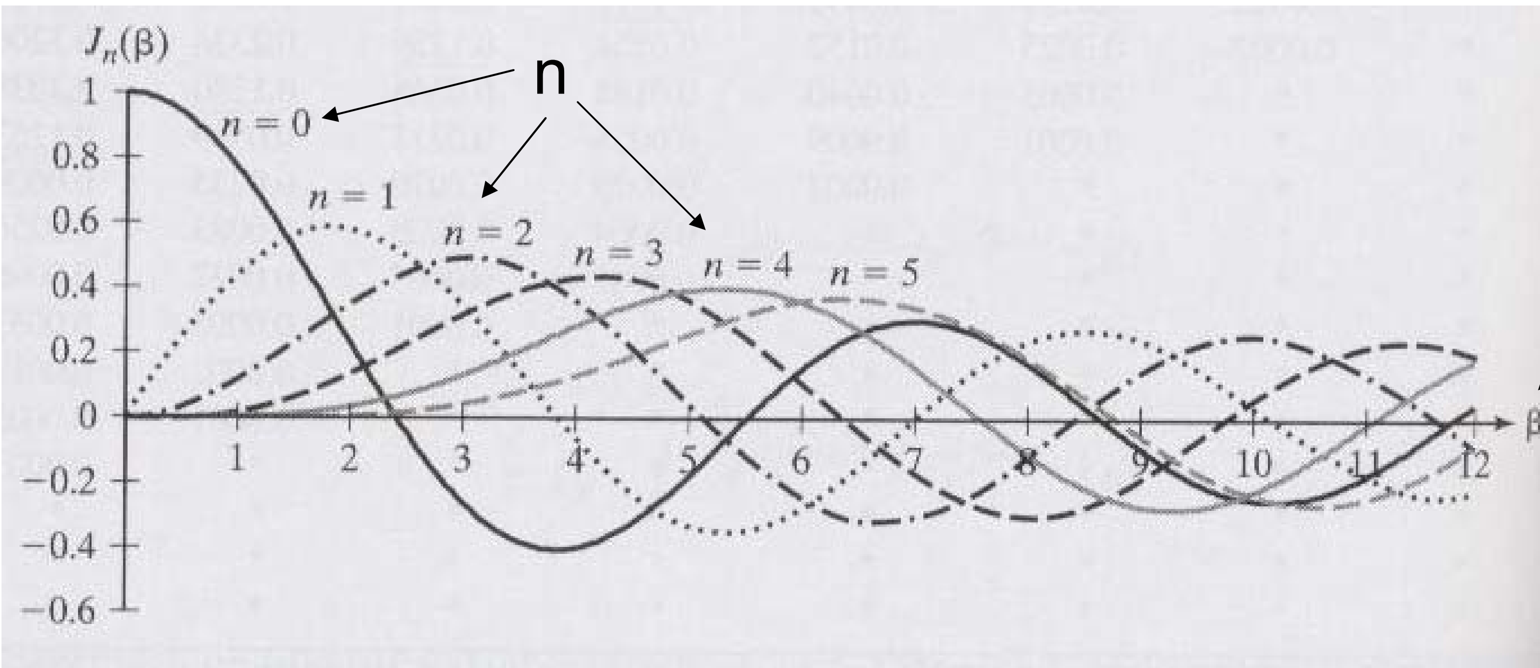
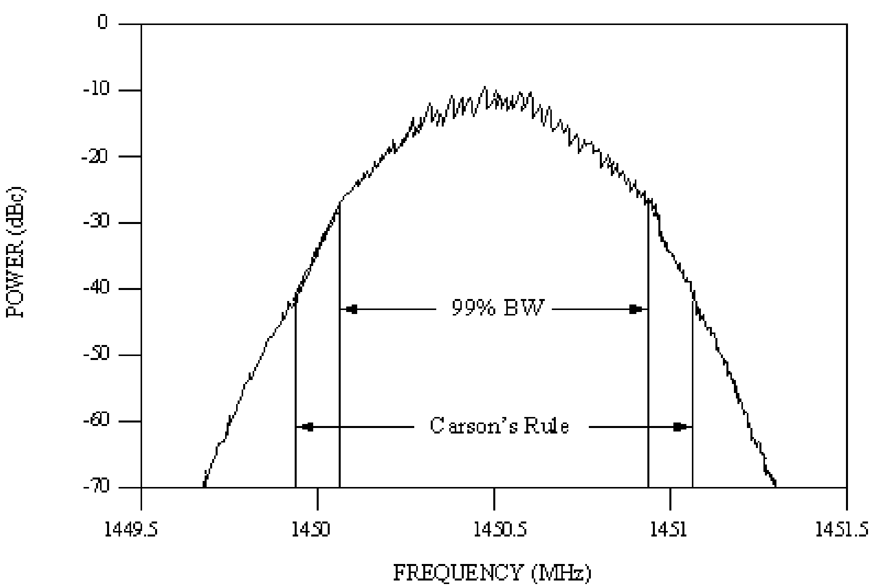



**Figure 6-21** Bessel function of the first kind for various values of $n$ (order).

- *Carson's Rule* for the approximate bandwidth of an angle modulated signal was developed by John R. Carson in 1922 while he worked at AT&T. Prior to this in 1915 he presaged the concept of *bandwidth efficiency* in AM by proposing the suppression of a sideband (see S&M p. 326-333):

$$B = 2\,(\beta + 1)\,f_m = 2\,(\Delta f + f_m)\ \text{Hz}$$

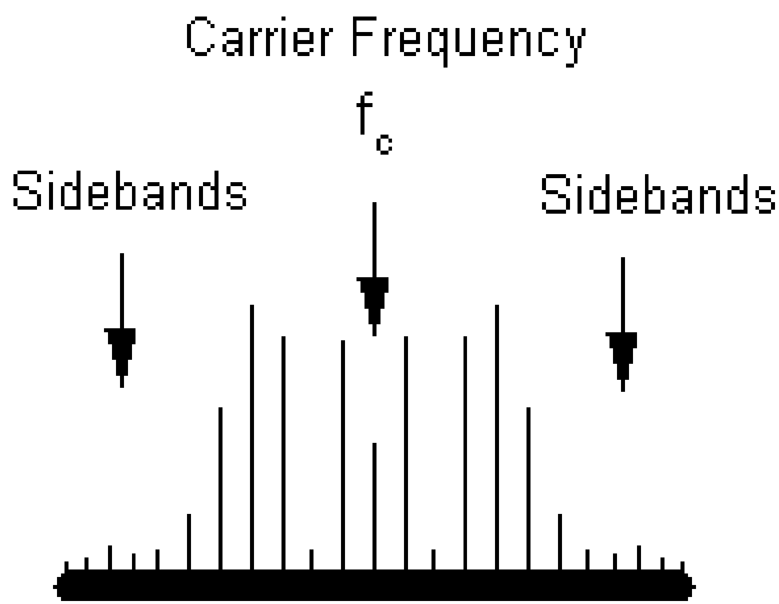



1886-1940

- The *normalized power* within the Carson's Rule bandwidth for a single tone angle modulated signals is:

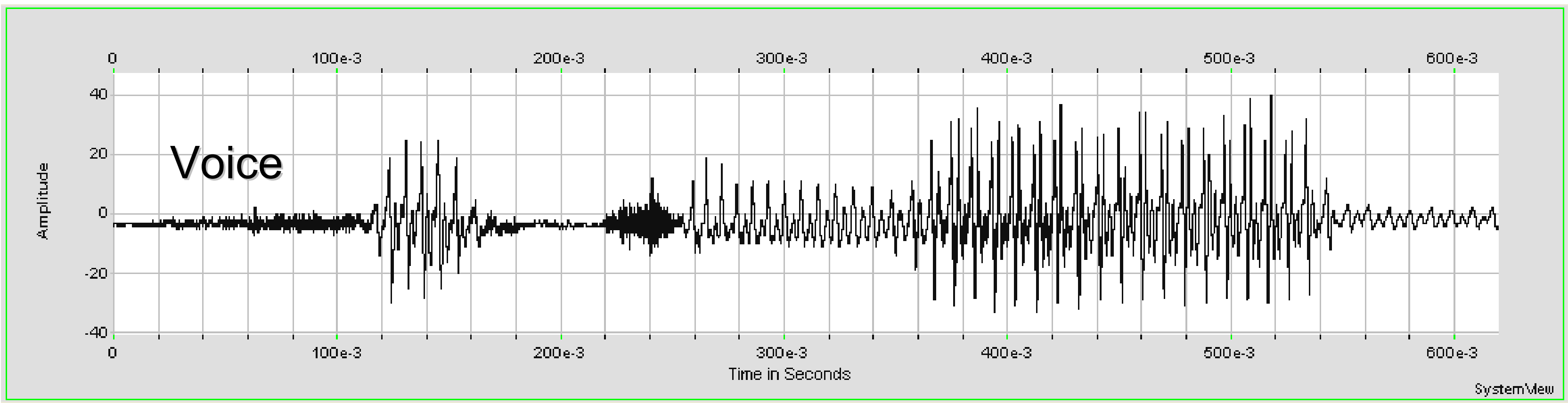
$$P_{\text{in-band, sinusoid}} = \frac{A_C^2}{2} \sum_{n=-(\beta+1)}^{\beta+1} J_n^2(\beta) \qquad \text{S\&M Eq. 6.75}$$

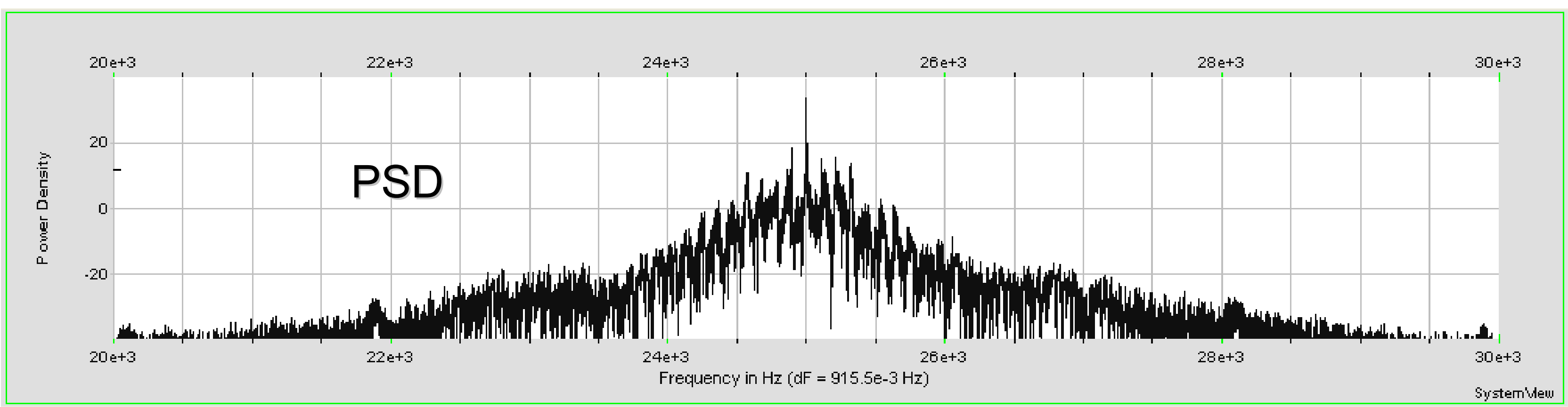
Note that $J_{-n}(\beta) = \pm\, J_n(\beta)$ so that $J_{-n}{}^2(\beta) = J_n{}^2(\beta)$ and for the normalized power calculation the sign of $J(\beta)$ is not used.



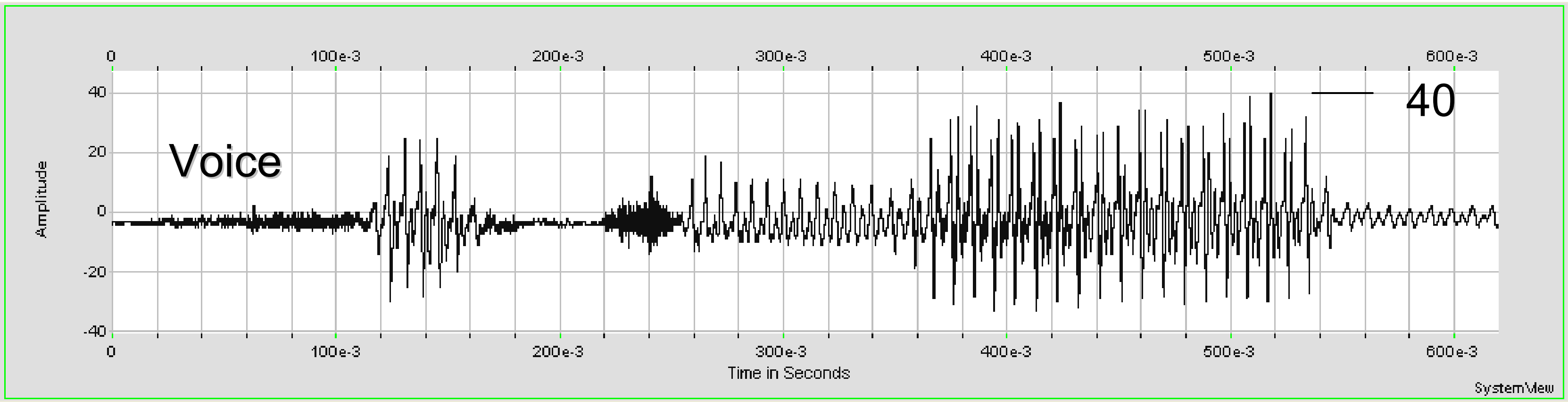

Spectrum of single tone FM modulation

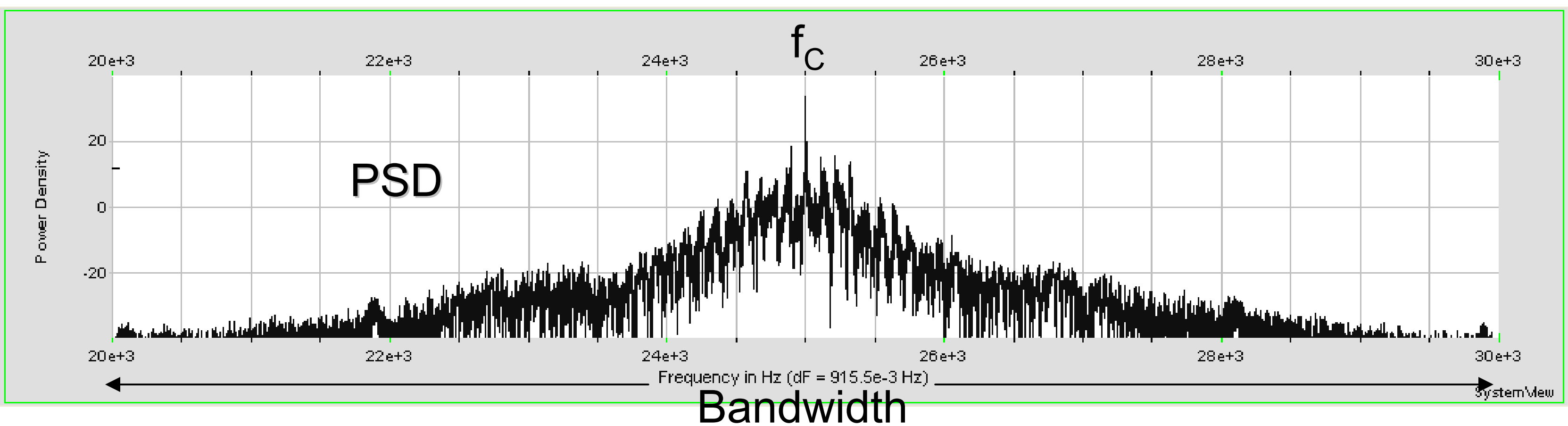
- The analog FM power spectral density PSD of the voice signal has a bandwidth predicted only by Carson's Rule since it is not a single tone.



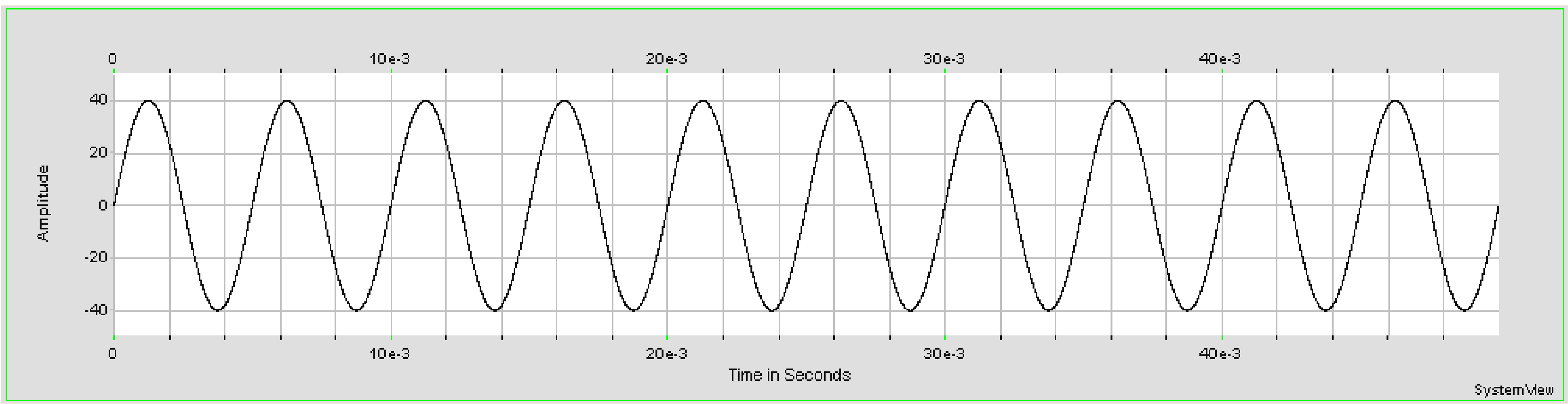



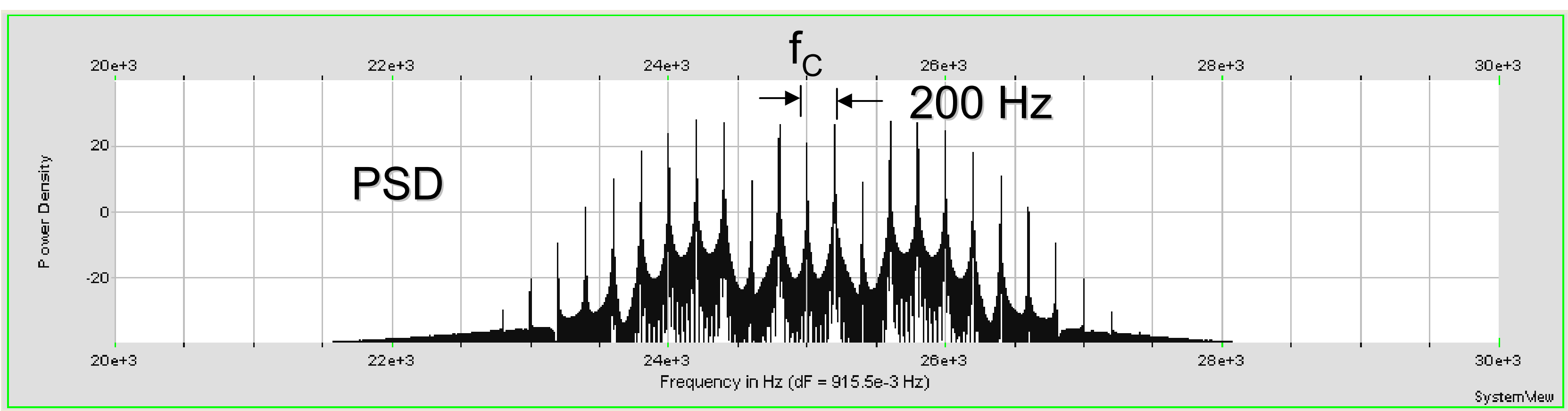
- Here $f_{max}$ = 4 kHz, k = 25 Hz/V and $\Delta f_{max}$ = 40(25) = 1 kHz. The Carson's Rule approximate maximum bandwidth B = 2 ($\Delta f$ + $f_m$) = 10 kHz or ± 5 kHz (but seems wrong!)



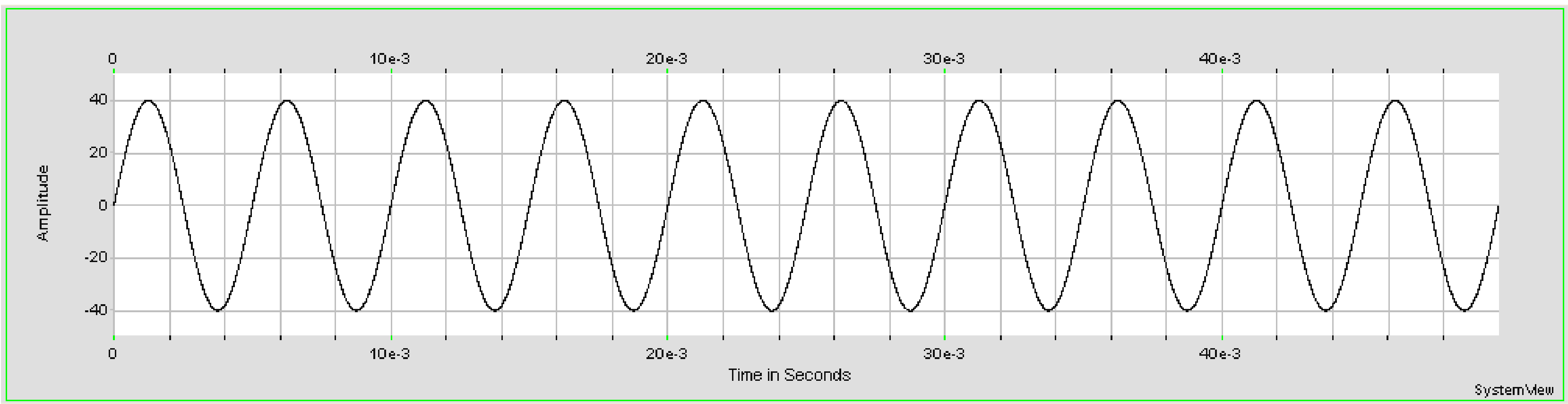



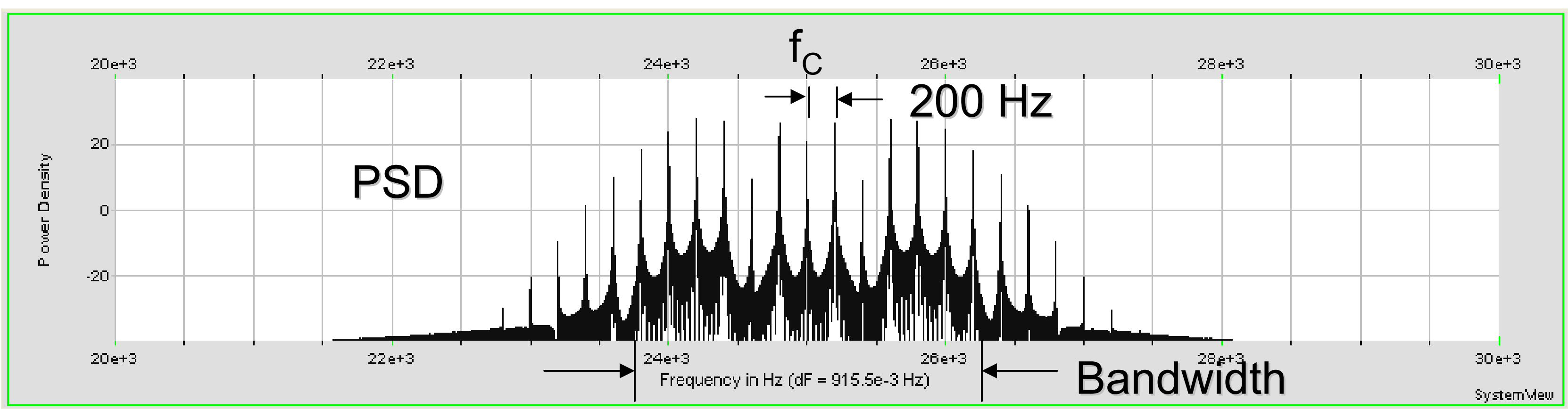
- A 200 Hz *single tone* FM signal has a PSD with periodic terms at $f_C \pm n\, f_m = 25 \pm 0.2\, n$ kHz.



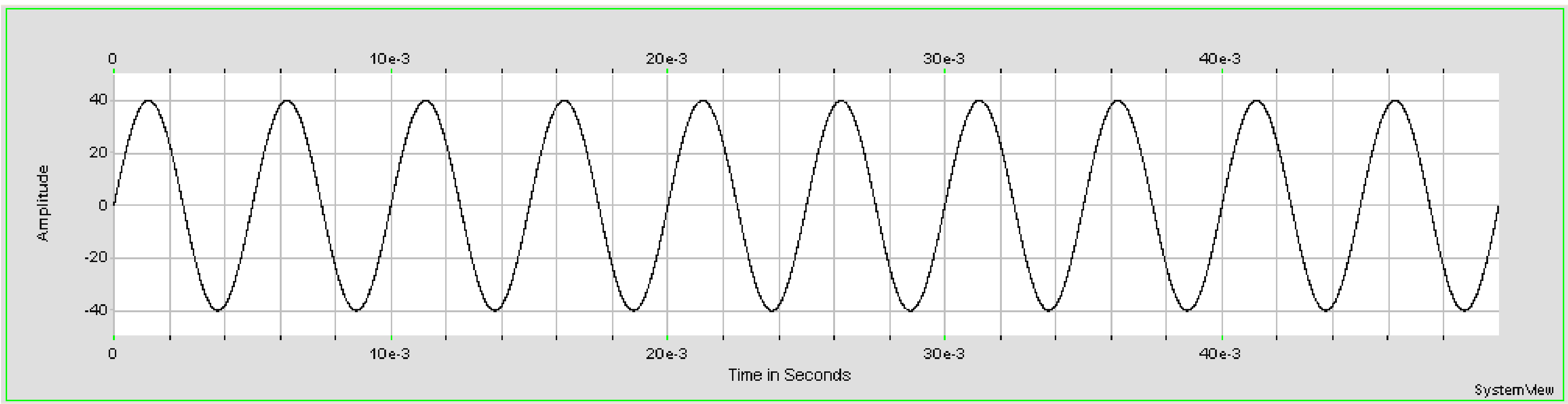



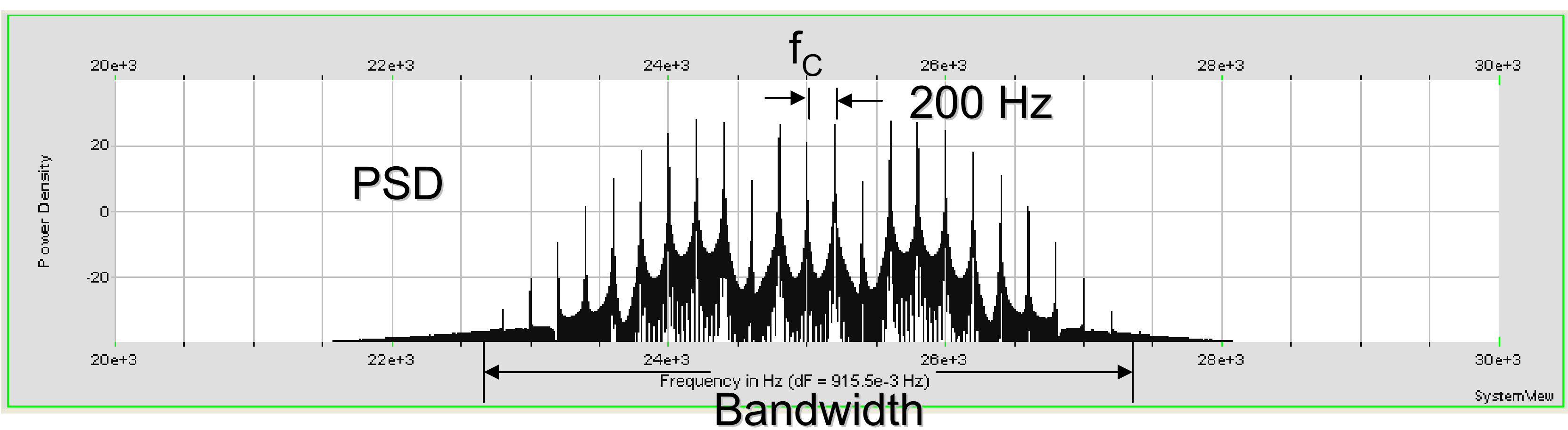
- Here $f_m$ = 200 Hz, k = 25 Hz/V and $\Delta f_{max}$ = 40(25) = 1 kHz. The Carson's Rule approximate maximum bandwidth B = 2 ($\Delta f + f_m$) = 2.4 kHz or ± 1.2 kHz:



- Since $\beta = \Delta f / f_m$ = 1 kHz / 0.2 kHz = 5 and the Bessel function predicts a bandwidth of $2\ n\ f_m$ = 2(12)(200) = 4.8 kHz (since n = 12 for $\beta$ = 5 from Table 6.1):

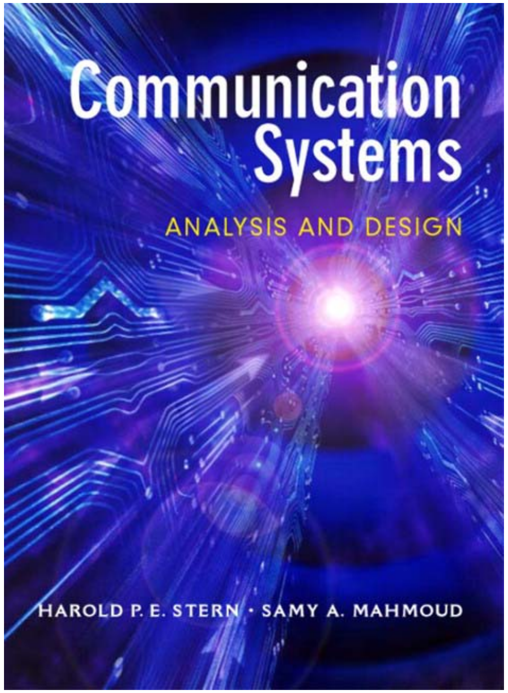


# Chapter 6

# Analog Modulation and Demodulation

- *Noise in FM and PM Systems*
- Pages 347-355

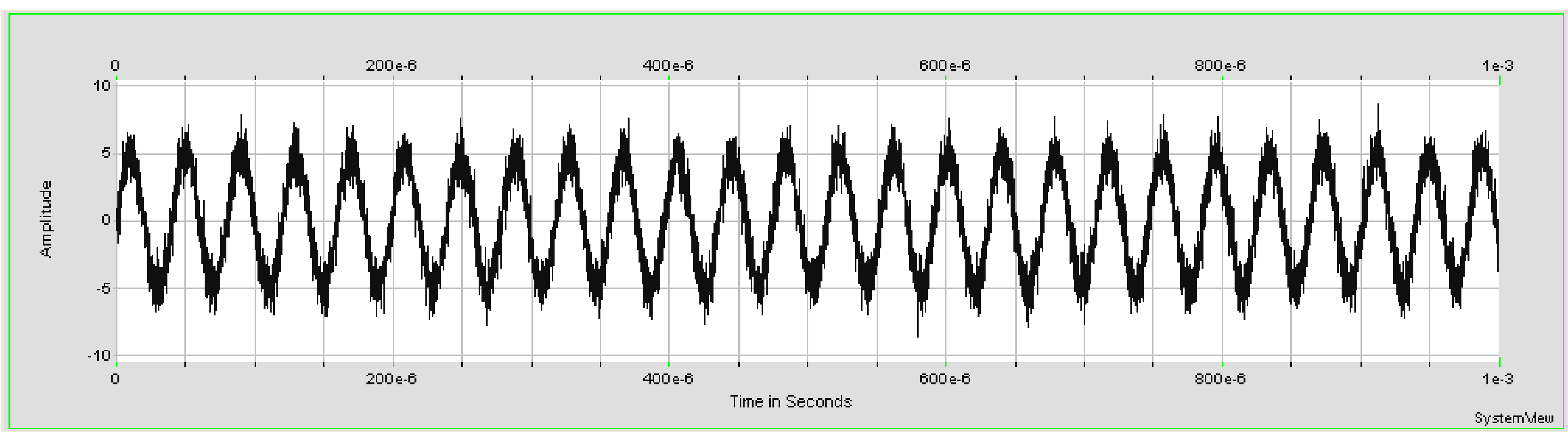


- A general angle modulated transmitted signal, where Ψ(t) is the instantaneous phase, is:

$$s_{\text{angle-modulated}}(t) = A_C \cos[\Psi(t)] \qquad \text{S\&M Eq. 6.86}$$

The received signals is:

$$r_{\text{angle-modulated}}(t) = \gamma A_C \cos[\Psi(t)] + n(t) \qquad \text{S\&M Eq. 6.87}$$



- The analytical signal for PM is:

$$s_{PM}(t) = A_C \cos[\Psi(t)] = A_C \cos[2\pi f_C t + \alpha s(t)]$$

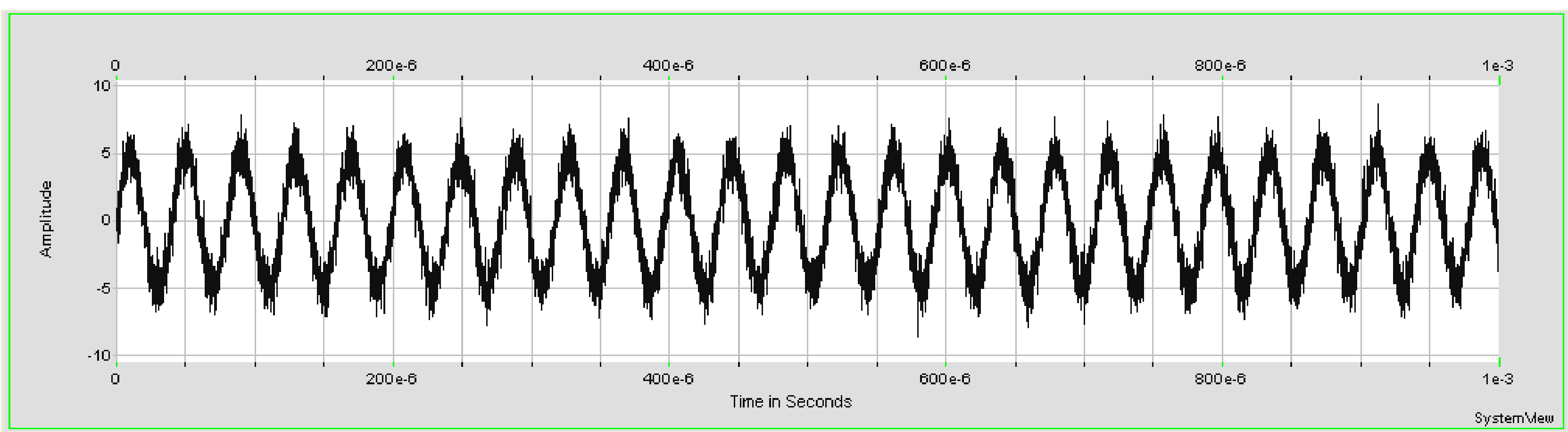
S&M Eq. 6.53

After development the SNR for demodulated PM is:

$$SNR_{PM} = (\alpha\gamma A_C)^2 P_S / (2 N_o f_{max})$$

S&M Eq. 6.98

where $-\pi < \alpha s(t) \leq \pi$ for all t.



- The analytical signal for FM is:

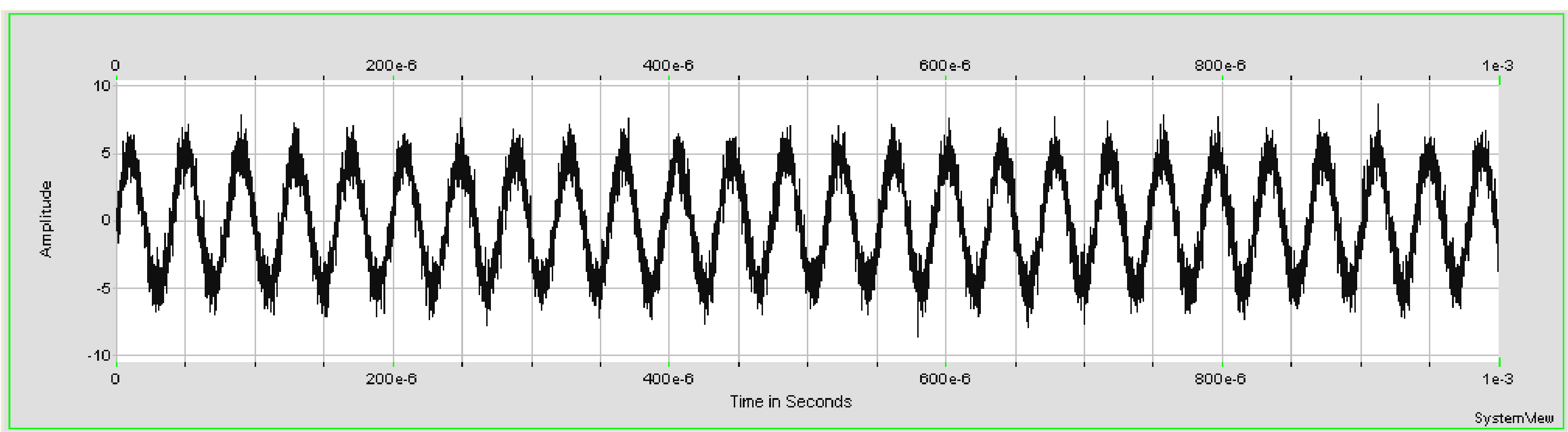
$$s_{FM}(t) = A_C \cos[\Psi(t)] = A_C \cos[2\pi f_C t + \int k\, s(\lambda)\, d\lambda]$$

S&M Eq. 6.53

After development the SNR for demodulated FM is:

$SNR_{FM} = 1.5\,(k\,\gamma\,A_C/(2\pi))^2\,P_S/(N_o\,f_{max}^{3})$ S&M Eq. 6.98

where $k\,s(t) \le f_C$ for all t.



# End of Chapter 6

# Analog Modulation and Demodulation