Telecommunications I Tutorials

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Tutorials from [1]

1. 3.2-1, 3.2-2, 3.2-3, 3.2-4, 3.2-5, 3.2-6, 3.2-7, 3.2-8, 3.2-9, 3.2-10, 3.2-11, 3.2-12, 3.3-1, 3.3-2, 3.3-3, 3.3-4, 3.3-5, 3.3-6, 3.3-7, 3.3-8, 3.3-9

Tutorials from [2]

1. 3.3 (a), 3.4, 3.5, 3.6, 3.7, 3.8, 3.11(a), 3.12, 3.14, 3.16, 3.18, 3.19, 3.21, 3.22, 3.23, 3.24, 3.25, 3.26, 3.27, 3.28, 3.29, 3.35, 3.36, 3.37, 3.38, 3.39, 3.40, 3.41, 3.42, 3.44, 3.45, 3.49

2. 4.7, 4.8, 4.10(b), 4.11, 4.12, 4.13, 4.15, 4.18, 4.22, 4.25, 4.26

More to follow ....

References

[1] A.B. Carlson, P.B. Crilly and J.C. Rutledge, *Communication Systems, An Introduction to Signals and Noise in Electrical Communication*, McGraw Hill, fourth edition, 2002.

[2] S. Haykin, *Digital Communications*, John Wiley & Sons, third edition, 1988.

Hints to solving questions

Haykin 3.3

(a) Assume $vi(t) = A_c cos(\pi f_c t) + m(t)$. Find the output current using the nonlinear equation given. It will then be easy to see that to extract a DSB-SC component at the output, we need a BPF. What do you think the ideal specifications of filter should be? Hence, determine the exact analytical expression at the output of the filter.

Haykin 3.4

(a) Simply substitute $v_1(t)$ in $v_2(t)$. Expand the result to get the desired AM term plus undesired terms.

(b) Assuming that the input signal is bandlimited to |f|<=W, sketch the resulting spectrum. Hence, determine the frequency specifications of the tuned circuit. Bear in mind that the circuit is tuned to fc as given in Figure P3.2.

(c) Use Equation (3.2) to determine the amplitude sensitivity of the modulator.

Haykin 3.6

1. Simply substitute the given $v_1(t)$ in $v_2(t)$.

2. Determine the lowpass components!!. There are two lowpass component. A desired one $m(t)$ and the other one is undesired $m^2(t)$. From your expression determine the required condition.

Haykin 3.7

This question is straightforward. Just follow Figure P3.3 to determine the output $v_3(t)$ and hence the message signal $m(t)$.

Haykin 3.8

Sketch the DSB-SC spectra for cases (a) and (b). You will see that there is aliasing for case (b); the carrier frequency is not high enough. From your results, determine the lowest allowable carrier frequency.

N.B. This problem is only for educational purposes and has no practical significance.

Haykin 3.11

(a) Straightforward. Show that the resulting message signal is modulated by a sinusoidal signal of low frequency δf.

Haykin 3.14

The local oscillator output signal has a phase offset ϕ, i.e. $\cos(2\pi f_c t + \phi)$. It is easy to show that there is crosstalk at the receiver output.

Haykin 3.16

(a) Manipulate $s(t)$ and expand it into its in-phase and quadrature components. Remember that from $\cos(2\pi f_c t)$ you can determine the in-phase component, and from $\sin(2\pi f_c t)$ you can determine the quadrature component.

(b) Add the carrier $A_c \cos(2\pi f_c t)$. Keep the in-phase and quadrature components separate. Determine the envelope. You need to do some manipulation to determine the expression for distortion in using envelope detectors.

(c) From the expression in (b), determine for what value of a the distortion is maximum.

Haykin 3.19

(a) Tabulate the results for each stage for positive and negative frequency bands.

(b) From the table above, determine the required filterig specifications in order to have SSB-USB at the output.

Haykin 3.21

(a) Use the spectrum given in Figure 3.22. You can assume an ideal case. I suggest you take $f_a = 1$ and $f_b = 3$. Therefore $f_0 = 2$ and the cutoff frequency $(f_b - f_a)/2 = 1$. Plot the spectrum at different points of the Weaver modulator. Deduce that the output signal is SSB-USB.

(b) The same as above.

(c) simply change the sign of one of *sin*( ) terms in the quadrature channel and repeat plotting the spectrum at different points of the Weaver modulator. Deduce that the output signal is SSB-LSB.

Haykin 3.22

(a) Multiply the expressions of $s_u(t)$ and $\hat{s}_u(t)$ by $\cos(2\pi f_c t)$ and $\sin(2\pi f_c t)$, respectively, and add the results together to get $m(t)$. Similar approach is used to deduce the expression for $\hat{m}(t)$.

(b) Follow the steps in (a) to determine expressions for $m(t)$ and $\hat{m}(t)$.

(c) By looking carefully at the expressions found for $m(t)$ and $\hat{m}(t)$, it can be conveniently seen that no matter we transmit SSB-USB or SSB-LSB, one receiver is used; our receiver multiply the input and its hilbert transform by $cos(2\pi f_c t)$ and $\sin(2\pi f_c t)$, respectively, and adds them together to extract the message signal. Don't forget that a scaling by $2/A_c$ is also needed.

Haykin 3.23

Find the expression for the envelope of $s(t)$. We want the output of the detector to be a good approximation of $m(t)$. By doing simple manipulation, it will be easy to determine the required conditions. You must determine some sort of relationship between A_c and $m(t)$, and A_c and $\hat{m}(t)$.

Haykin 3.27

(a) Multiply $v_1(t)$ by $v_2(t)$. Pass the result from the LPF. Do frequency division by $(n+2)$. Remember that the frequency and the phase of the signal are divided by $(n+2)$. Multiply the output of the divider by $v_2(t)$. Substitute the expressions given for f_1, f_2, n and ϕ_2. It will then be easy to deduce that the output of the BPF is proportional to the carrier wave.

(b) Swap the place of the upper and the lower narrow-band filters. Repeat the method given above. Use the new expressions for f_1 and f_2. From your results the new relationship between ϕ_1 and ϕ_2.

Haykin 3.28

By mixing, you get the sum and the difference of the input frequencies f_1 and f_2, i.e. $f1 \pm f2$. Tabulate all the frequencies at the output of the synthesizer and, hence, determine the frequency resolution.

Haykin 3.29

It is an FDM sytem. Have a look at Figure 3.27. Use the same carrier waves in the receiver. Notice the typing mistake; β_3 in the first brackets must be α_3.

Haykin 3.36

(a) Specify the in-phase and out of phase (quadrature) components. Write the expression for envelope, i.e. envelope is $\sqrt{(\text{in-phase component})^2 + (\text{quadrature component})^2}$. Hence, determine the maximum and minimum values of the envelope and, therefore, their ratio. Finally, sketch this ratio versus different values of beta in the specified range.

(b) Here expand $s_i(t)$ into cosine terms. Find the total average power by averaging each term and adding them together. Find the ratio of this total average power to the average power of unmodulated carrier. Finally, plot this ratio versus beta in the specified range.

(c) Here we have to prove that the angle of $s_i(t)$ is as given in the body of the question. Well, again we use the expansion of $s_i(t)$ as found in part (b). Call the angle of $s_i(t)$ as $\theta_i(t)$. The desired form for the representation of a bandpass signal is

$$s(t) = \text{ in-phase component } \cos(2\pi f_c t) - \text{ quadrature component } \sin(2\pi f_c t)$$

The phase angle is $2\pi f_c t + \arctan(\text{quadrature component}/\text{in-phase component})$. Do the substitution and expand the arctan term up to the third harmonic. You will see that the resulting expression is the same as the one asked to find. *N*th harmonic distortion is the ratio of the amplitude of Nth harmonic to the amplitude of the first harmonic. Ignore the sign. In the part, simply expand the phase angle and find the harmonic distortion.

Haykin 3.37

(a) Write the expression for the phase-modulated (PM) signal, i.e.

$$s(t) = A_c \cos[2\pi f_c t + k_p A_m \cos(2\pi f_m t)]$$

Expand $s(t)$. Now take into consideration that phase deviation is small so you can simplify $\cos()$ and $\sin()$ terms. Hence an approximate expression to $s(t)$ can be found. The expression found is in the time domain. To find the spectrum of the PM signal, simply take the Fourier transform. Although the question hasn't asked for the spectrum to be plotted, I suggest you to do the plotting.

(b) Have a look at Figure 3.32 on page 161 of Haykin's book. The AM phasor diagram is given there. Now, sketch the PM phasor diagram.

Haykin 3.39

(a) Use table A4.1 to find for what values of beta $J_0(\beta)$ becomes zero. The table is by no means exhaustive exhaustive. So you just have to some sort of guesswork to find your answer. Accurate answers can be found using, say, Matlab.

(b) $\beta = \Delta f / f_m$ or $\beta = k_f A_m / f_m$. You found betaŠs from part (a). Use the first one here. Am and fm are given. So k_f, i.e. the frequency sensitivity, can be found. To solve the last part, we use the second beta found in part (b) to find the desired Am.

Haykin 3.40

From table A4.1 find different values of $J_n(1)$. Write and expand the FM expression. Pass it through the BPF and determine the analytical expression of the signal at the output of the filter. Bear in mind that the passband of the filter is $f_c - 2.5f_m$ to $f_c + 2.5f_m$ for positive frequencies. Although Haykin hasn't asked the spectrum to be sketched, but it is a good idea to do so for both positive and negative frequencies.

Haykin 3.41

(a) Find frequency deviation and, hence, β. Finally, use the Carson's rule to find the approximate BW.

(b) Use β in Figure 3.36 to find the corresponding $B_T/\delta f$ on the vertical axis. Knowing what δf is, it is easy to calculate the transmission BW.

(c) Same as part (a)

(d) Same as part (b)

Haykin 3.42

(a) Very easy. Review Example 5.3 before doing this question. It is straightforward to show that BW varies linearly with f_m.

(b) Again it is easy to prove that BW for FM is independent of f_m.

Haykin 3.44

Just follow Figure 5.10 on page 230 of Lathi's book. How about frequency separation?

Haykin 3.45

Simply write down the FM equation and substitute it in the given expression. Following expansion and passing the result through a bandpass filter centred at 2f1, you will get a new FM signal whose frequency and frequency deviation is twice those of the original FM signal.

Haykin 3.49

This concept of this question is straightforward, but the solution is long-winded. The purpose of this question is to recover the message signal from the received FM signal. Bear in mind that here we have a tone modulation. Find the signal at the input to the envelope detector. Do the given simplifications. From the simplified expression, determine the envelope and, hence, show that it is proportional to the sinusoidal message.

Haykin 4.7

(a) Not included.

(b) Determine the autocorrelation function and the mean of $Z(t)$ and, hence, show that this random process is WSS.

Haykin 4.8

(a) Assume $X(t) = A + Y(t)$. Find ACF and, hence, show that it contains a constant component equal to A^2.

(b) Assume $X(t) = A_c \cos(2\pi f_c t + \Theta) + Z(t)$. Find ACF and, hence, show that it contains a sinusoidal component if the same frequency as $X(t)$.

Haykin 4.10

(a) Not included.

(b) Either use the table of the Fourier transform to prove the required PSD. Find the average power by integrating the PSD. Or, you can substitute $t = 0$ in ACF to determine the average power.

Haykin 4.13

(a) Simply use Eq. (4.86) with $h(t)$ as the convolution of $h_1(t)$ and $h_2(t)$.

(b) Write down the expression for CCF, i.e. $E[V(t+t)Y(t)]$. Substitute integral convolution for $Y(t)$, and find the CCF wrt ACF of $V(t)$.

Haykin 4.15

(a) The PSD consists of two components. Find the inverse FT of each component and add them together.

(b) From (a) you can determine the dc (constant) value.

(c) From (a) substitute $t = 0$ in the AC component to determine the AC power.