Lab Working Guide

Introducció a les Comunicacions (ICOM)

February, 2022

Session 1. Basic operation of a spectrum analyzer based on the swept superheterodyne receiver principle

Objective	• Students will understand the basic behavior of communication receivers, both fixed frequency superheterodyne receivers and swept local oscillator receivers. • Students will be able to configure the spectrum analyzer to properly analyze basic signals in the frequency domain.
Background	• Students need to be familiar with general characteristics of systems in communication receivers: amplification/attenuation, frequency conversion, bandwidth, envelope detection and noise. • Students need concepts given in SIS of convolution, Fourier transform, spectral content and modulation of signals, and concepts of stochastic processes given in PPEE.
Manufacturer's guide to consult in the lab (Available at Atenea)	• R&S FSC Spectrum Analyzer Operating Manual (Chapter 4) • Agilent Application Note 150. Spectrum Analysis Basics

Homework

Question 1.1 Read document "How Spectral Analyzer works" (4 pages) available at Atenea.

Question 1.2 Solve the following exercise and bring it to class.

Exercise. Let's assume a signal generator with an internal impedance of 50Ω set to provide a 1MHz sinusoidal signal with an open-circuit amplitude of 200mV. Calculate the power measured by a power meter with an internal impedance of 50Ω connected to this signal generator. Express the result in dBm, where dBm is $10\cdot\log_{10}$[power in miliWatt]).

Activity in the lab:

Activity 1.1 Visualize the spectrum of a basic signal. Generate a 1MHz sinusoidal signal of 200mV amplitude with the PROMAX GF-856 signal generator, which has an input impedance of 50Ω. Connect it to the spectrum analyzer (SA) and observe its spectrum. Set the *Reference Level*, *Center Frequency* and *Span* to properly visualize the generated signal, with some of the higher order harmonics. Set the *Display Range* to 10dB per division. With the help of a *Marker* verify that the power of the fundamental harmonic is equal to the result of Question 1.2. Explain why the SA produces higher order harmonics.

Activity 1.2. Understanding information shown on the screen. Observe annotations shown on the SA screen. Use *FSCView* program in the PC to capture the screen, copy it in your report and explain the meaning of each acronym and annotation. Note that the *Resolution* and *Video Filter Bandwidth*, the *input RF Attenuation* and the *Swept Time* parameters are automatically set by the SA once you enter the primary parameters (i.e. *Center Frequency, Span and Reference Level*). However, you may modify them by setting the manual control mode.

Activity 1.3. Shape of spectral components. The spectrum of the fundamental harmonic of the sinusoid shown at the SA screen does not correspond to a Dirac delta function. Instead, a wider shape located around 1MHz can be observed. To gain more insight into this aspect of the SA, first adjust the configuration of the SA to visualize the spectrum shape around the principal harmonic and set the *Resolution Bandwidth* control to manual mode. Then, change the *Resolution Bandwidth* to 100kHz, observe the impact on the spectrum and compare this value with the -3dB bandwidth of that shape (use the *Display Line* and/or Markers to measure the bandwidth at -3dB). You may wish to get more familiar with this procedure by trying other *Resolution Bandwidth* values. Finally, explain why the spectrum of the principal harmonic adopts this particular shape.

Activity 1.4. Tuning with Zero Span. Set the *Resolution Bandwidth* control back to automatic mode and deactivate *Display Line*. Set *Span* to 200kHz and, if needed, reconfigure the SA to see the fundamental harmonic of the 1MHz signal at the center of the screen. Finally, select *Zero Span* and observe the trace on the screen. What does this trace represent? Observe what happens by slightly varying the *Center Frequency*. What is the relationship, if any, between that trace and the one you have for *Span* different from 0? At the end, press the *Last Span* key to return to the initial configuration.

Activity 1.5. Impact of noise on weak signals and noise rejection procedures in the SA. Set the *Resolution Bandwidth* control to automatic mode, and configure the SA to visualize the fundamental, second and third harmonic on the screen. Decrease the amplitude of the signal generator until the power of the main spectral component is -40dBm, so the higher harmonics are even lower and will be below the noise floor. Adjust the attenuation, RBW and VBW values to observe a clean trace that allows to measure accurately the frequency and power of the third harmonic. Include in your report a description of how these three parameters impact the measurement of weak signals.

Session 2. Characteristic parameters of a communications receiver

Objective	• Students will get familiar with the most relevant characterization parameters of communication receivers. • Students will learn techniques to measure these parameters. • Students will learn about the interference caused by the image frequency.
Background	• Students need to be familiar with the configuration of the Spectrum Analyzer and the procedure to make measurements. • Students need a basic knowledge of third order intermodulation products at the output of the mixer. • Students need to be familiar with the concepts of noise and Signal-to-Noise Ratio (SNR). • Students need to know the expression of a Gaussian function, the concept of variance and a basic use of MATLAB.
Manufacturer's guide to consult in the lab (Available at Atenea)	• Anritsu. Guide to Spectrum Analysis • Agilent Application Note 150. Spectrum Analysis Basics

Homework

Question 2.1. Measurement of the Third Order Intercept characteristic point. Radiofrequency (RF) communication receivers show a nonlinear effect when a high amplitude input signal saturates the RF input mixer. This nonlinear behavior is characterized in communication receivers by the so-called Third Order Intercept (TOI) point. TOI is defined as the power of two equal-level tones at the input of the mixer so that the power of the third order intermodulation products reaches the same level of the input tones. This parameter can be easily measured driving the communication receiver with two sinusoids of power P_i and frequency f_1 and f_2, and using

$$TOI\ (dBm) = P_i(dBm) + \Delta/2(dB) \tag{1}$$

where Δ is the ratio between P_i and the power of the third order intermodulation products.

In this exercise we ask you to read the document "Class_note_TOI.pdf" and to prove equation (1) using the following procedure. First assume that the powers of the linear component at the output and at the input of the mixer are equal. The power of the third order intermodulation products at the mixer output (that is, the signals with frequencies $2f_1 \pm f_2$ and $2f_2 \pm f_1$) is proportional to the cube of the input tones power P_i. Then express the power of the linear component and the power of the third order intermodulation products in dBm, and show that the intersection point fulfills (1). Note that the power of the third order intermodulation products versus the input power is a straight line with a slope equal to three, whereas the power of the linear component versus the input power has a unit slope.

Activity in the lab:

Activity 2.1. Measurement of Receiver Sensitivity. All receivers, including spectrum analysers, generate some internal noise that limits its capability to deal with small signals. The Sensitivity is the parameter that characterizes this feature, and is given by the noise floor in dBm of the instrument for a particular IF filter bandwidth, usually the smallest Resolution Bandwidth setting. In this activity we ask you to measure the sensitivity of your SA acting as a receiver of 10kHz bandwidth signals, with a carrier frequency between 500 and 1500 kHz and with a minimum Signal-to-Noise Ratio (SNR) of 30 dB at the demodulator output. Apply the following procedure to measure the Sensitivity:

- Configure the SA to operate with signals with a carrier frequency of your choice in the range of 500 to 1500 kHz and with a 10kHz bandwidth. Set the attenuation to 0dB.
- Without connecting any signal to the receiver input, decrease the Reference Level until you observe the noise floor.
- Measure the noise level setting Span to Zero and decrease the Video Filter bandwidth until the trace is almost flat. You may use Display Line.

The Receiver Sensitivity for a 30dB of SNR would then be the noise floor level plus 30dB. Write down the Receiver Sensitivity of your SA and justify the procedure followed.

Activity 2.2. Measurement of Third Order Intercept (TOI) point. The objective of this activity is to compute the TOI point of the SA using equation (1). For this purpose, generate two tones of 900kHz and 950kHz frequencies using the signal generators GF 856 and GF 232-B. Combine these signals using a "T-combiner" provided in the laboratory. Apply the sum of these tones to the receiver RF input. Please note that the TOI point is highly sensitive to the frequency range; therefore, the value measured in this exercise is valid only for the AM range. Then, configure the receiver in swept mode (i.e. spectrum analyzer mode) with an Attenuation of 0dB, and select the Center Frequency and Span to visualize both the two tones and the third order intermodulation products on the screen. Finally, adjust the level of the two input tones to - 20dBm.

The intermodulation products power is much smaller than -20dBm, so the intermodulation products might be masked by the noise floor. If needed, modify the RBW and VBW settings so that the two intermodulation products are clearly visible and you can measure their power accurately (remember that in the TOI point measurement the attenuation cannot be employed to reduce the noise floor).

As described in the document "Class_note_TOI.pdf", equation (1) can only be employed to compute the TOI point if the SA is working well below saturation and the two tones at 900kHz and 950kHz are not distorted due to the gain compression appearing for high power inputs. Verify that your set up corresponds to this regime by switching the RF Attenuator from 0 to 10 dB. If you observe a change in the input tones measured power it means that the SA is in saturation, so to measure the TOI point you have to decrease the input tones power below - 20dBm until this effect is not observed. Explain in your report why this procedure is useful to verify whether the SA is working in saturation or not.

Set back the attenuation to 0dB and compute the TOI point value using equation (1). Explain this activity in your report.

Activity 2.3. Measurement of the Receiver Selectivity. The capability of a communication receiver to reject adjacent channels is determined by the transition band of the Intermediate Frequency (IF) filter. This feature is characterized by the Receiver Selectivity parameter, defined as the ratio between the -60 dB bandwidth and the -3 dB bandwidth of the IF filter. Considering that the Resolution filter in the SA corresponds to the IF filter, configure the spectrum analyzer to depict the IF filter gain frequency response on the screen by following the procedure described next.

First use the PROMAX GF-856 signal generator to feed a 1MHz sinusoidal signal of -10dBm to the SA. Set the resolution bandwidth to 10kHz and specify the configuration of the SA (Center Frequency, Span, Reference Level, etc) to measure the Receiver Selectivity of the SA. Explain why we can observe the IF filter gain frequency response on the SA screen when the SA input signal is a single sinusoid.

Measure the bandwidth of the Resolution filter at -3dB, -5 dB, -10 dB, -20 dB, -30 dB and -60dB using the SA function "*N dB Down*". The -60dB bandwidth might be difficult to measure because of the noise floor, if so replace it by the -50dB bandwidth measurement. Plot the gain frequency response of the IF filter and include it in your report. What is the Receiver Selectivity of the SA in this set up?

Session 3. Principles and operation of a Vector Signal Analyzer

Objective	• Students will understand the basic functions performed by a vector signal analyzer (VSA), and the relation between time and frequency parameters of the VSA. • Students will be able to configure the VSA to visualize signals properly. • Students will use the VSA to study bandpass signals, equivalent lowpass signals, and in-phase and quadrature components.
Background	• Students need to know the relation between time and frequency parameters of discrete signals from the subject SIS (1A). • Students need to know the concept of signal windowing and FFT. • Students will use concepts of Lesson 2 of ICOM "Senyals I Sistemes Passabanda". Specifically, students need to know how to compute the lowpass equivalent signal of a bandpass signal.
Manufacturer's guide to consult in the lab	• Keysight VSA 89600 User's Guide • Online help of VSA 89600 software is of great help during lab activities.

Homework

Question 3.1. Compute the lowpass equivalent signal, and the in-phase and quadrature components of the following signal

$$s(t) = A_1 \cdot \cos(2\pi f_1 t + \varphi_1) + A_2 \cdot \cos(2\pi f_2 t + \varphi_2) \text{ with } f_1 < f_2$$

choosing as central frequency (a) $f_0 = f_1$, (b) $f_0 = f_2$ and (c) $f_0 < f_1 < f_2$. Represent the Fourier transform of the equivalent lowpass signal.

Question 3.2. Read sections 1-6 of the Keysight VSA 89600 User's Guide available at Atenea and answer the short questions (a)-(d) below. Note that the user's guide includes both hardware and software aspects of the system. However, in the lab class you will work with Keysight VSA 89600 software to visualize recorded signals. You may also want to have a look at sections 11-14 to make yourself more familiar with the software environment before starting the activity in the lab.

The system block diagram of section 4 shows how the signal is sampled and windowed before computing the FFT, from which the VSA makes measurements or visualizes the spectrum. Therefore, the VSA parameters frequency span, number of frequency displayed points (M), resolution bandwidth (RBW), and time record length (T) interrelate with each other. Sections 5-

6 of the user's guide explain these relations, which will help you improve your understanding of the VSA operation.

a) Signal recording files can come either from radiofrequency (RF) or baseband signals but they should be already in digital format. The recorded signal file must include information about the sampling frequency, the carrier (if any) and other data depending on the type of signal. In the lab you will have to select the recording file to be visualized. Into which stage of the system block diagram is the recording data fed?
b) What is the length of the time record (T) to achieve an effective resolution bandwidth (RBW) of 171kHz with a Flat Top window?
c) What is the sampling frequency (Fs) if the Span is equal to 36MHz?
d) If the length of the time record (T) and the sampling frequency (Fs) are the obtained in b) and c) respectively, compute the number of time samples (N) and the number of displayed frequency samples (M).

Activity in the lab: (check the User's Guide to complete the activities)

Activity 3.1. Acquisition and visualization of a recorded bandpass signal with Keysight VSA 89600. Start the VSA software and follow the procedure described in the last slide of the document "Lab_ICOM_Session3.pdf" to analyze the recorded file "*two_sinusoids_40dB.mat*" (available in Atenea). Please note that it is a *.mat* file. By default, the program displays 2 stacked subplots: the trace in the top subplot corresponds to the spectrum of the signal recorded in the file, and the trace in the bottom subplot is the magnitude of the complex envelope in logarithmic units. First of all, configure the display to show 3 stacked subplots with the spectrum, the in-phase signal component (that is, the real part of the lowpass equivalent signal, denoted *Real*) and the quadrature signal component (i.e., the imaginary part of the lowpass equivalent signal, denoted *Imag*). In order to select the trace to visualize in each subplot first select the subplot A, B or C. Then select the type of data in the *Trace/Data* menu, and finally go to *Trace/Format* to select the *Real(I)* or *Imag(Q)* signals. Stop the execution, and answer these questions regarding the spectrum trace.

- What is the maximum power level that can be represented in the figure?
- What is the power ratio between two consecutive horizontal lines?
- What are the minimum and the maximum frequencies represented in the figure?
- Configure the resolution bandwidth (RBW) mode as "*Arbitrary*" in the MeasSetup/RBW menu and compare observed parameters *RBW, WS, Span, Fs, T* and *M* to those obtained in question 3.2. Note that some of these parameter values are shown in the spectrum window and some others are indicated in the MeasSetup menu.

Activity 3.2. Measurements of frequency and magnitude. The file "*two_sinusoids_40dB.mat*" contains samples of a signal consisting of two sinusoids embedded in white noise:

$$s(t) = A_1 \cdot \cos(2\pi f_1 t + \varphi_1) + A_2 \cdot \cos(2\pi f_2 t + \varphi_2) + n(t)$$

Measure the amplitude and the frequency of each sinusoid with the help of markers. The procedure is as follows. First, in *Pause mode* place a marker at the first spectral component of the spectrum using the *Markers* functions. The screen will show the frequency f_1 and power at the marker's position. Then, compute the amplitude A_1 of the sinusoid by converting the power in dBm into a voltage in Volts, considering that the power is measured on an internal impedance of 50 Ω. Finally, repeat the process for the second spectral component and obtain the values of A_2 and f_2.

Compare the frequency separation of the two sinusoids with the resolution bandwidth value.

Activity 3.3. Visualization of the in-phase and quadrature components of a bandpass signal. In question 3.1, you computed the lowpass equivalent signal of $s(t)$ without noise, and its in-phase and quadrature components for different central frequencies. In this activity we ask you to compare these theoretical expressions with the practical ones plotted by the VSA. First of all, change the central frequency of the VSA to f_1 in the *MeasSeatup/Frequency* menu. Stop the execution and compare the *Real* and *Imag* traces with the theoretical in-phase and quadrature components you obtained in question 3.1 with $f_0 = f_1$. For this purpose, place two markers at the largest/smallest signal value in one period of the *Real* and *Imag* traces (four markers in total), one of the markers *Normal* and the other one of *Delta* type to provide differential measurements directly. Write down the amplitude/time value of the four markers and use them to evaluate the amplitudes A_1 and A_2, the frequency f_2 and the phase φ_1. *Hint*: Use the DC values of the *Real* and *Imag* traces. Verify that the value of A_1, A_2 and f_2 are consistent with the results in Activity 3.2.

Repeat the process with a central frequency $f_0 = f_2$ to evaluate amplitudes A_1, A_2, the frequency f_1 and the phase φ_2. Verify that the values of A_1, A_2 and f_1 are consistent with the results in Activity 3.2.

Once you have measured all the involved parameters, provide the temporal expression of the recorded signal, using the measured values of A_1, A_2, f_1, f_2, φ_1 and φ_2.

Session 4. Generation of digital signals with the LaVICAD Simulator, and their analysis using a Vector Signal Analyzer.

Objective	• Students will learn how to generate a QPSK digital signal with the LaVICAD simulator. They will also make themselves familiar with the square root raised cosine shaping pulse. • Students will learn how to configure the VSA to operate as a digital demodulator for the QPSK signal. With the help of the VSA, they will visualize the signals at the output of the IQ coherent digital demodulator, and will learn to interpret the received signal constellation and I/Q eye diagrams. • Students will better understand the impact of noise on the received signal constellation, and on the I/Q eye diagrams. They will learn the performance parameters of a digital demodulator and how they can be measured with the help of the VSA.
Background	• Students should know the expression of the power spectral density of a QPSK signal, and its equivalent lowpass signal. • Students need to know the architecture of a digital QPSK modulator and demodulator. • Students will use concepts of Lesson 3 of ICOM “Modulacions Digitals en canals AGWN”.
Manufacturer’s guide to consult in the lab (available at Atenea or online help)	• Keysight VSA 89600 User’s Guide. • Online help of Keysight VSA 89600.

Homework

Question 4.1. Obtain the power spectral density (PSD) of a QPSK (or 4 QAM) signal with a carrier frequency of 10 MHz, and equiprobable bits at a rate of 64 kbit/s. Draw the resulting PSD assuming a square root raised cosine (SRRC) pulse $p(t)$ of unity energy and with a roll-off factor of α=0.3. What is the bandwidth of this signal?

$$p(t)=\frac{\sin\left((1-\alpha)\pi rt\right)+4\alpha rt\cdot\cos\left((1+\alpha)\pi rt\right)}{\pi r^{1/2}t\left(1-(4\alpha rt)^2\right)}$$

$$S_p(f)=|P(f)|^2=\begin{cases}\dfrac{1}{r} & |f|<\dfrac{r}{2}(1-\alpha)\\ \dfrac{1}{r}\cos^2\left(\dfrac{\pi}{2\alpha r}\left(|f|-\dfrac{r}{2}(1-\alpha)\right)\right) & \dfrac{r}{2}(1-\alpha)\le|f|\le\dfrac{r}{2}(1+\alpha)\\ 0 & \dfrac{r}{2}(1+\alpha)<|f|\end{cases}$$

Question 4.2. The effective sampling frequency of the signal generated with LaVICAD depends on two parameters: the number of samples per symbol to be selected in stage 2, and the symbol rate in stage 3. On the other hand, the VSA 89600 requires a signal with a sampling frequency that satisfies

$$F_s=1.28\cdot Span$$

Choose the values of the number of "samples/symbol" and "symbol rate" for the QPSK signal defined in question 4.1 in order to visualize it properly with the VSA 89600 using a frequency Span of 400 kHz.

Use the LaVICAD simulator available at Atenea (Metacurs) to generate a modulated signal as the specified in question 4.1, using the samples/symbol parameter obtained in this question (4.2). Configure the parameters of the first three stages of LaVICAD to generate the QPSK signal defined above. Figure 4.1 shows a block diagram of the LaVICAD QAM Simulator. To complete this exercise, just choose one of the available carrier frequencies. Compare the PSD provided by the "QAM System" of the LaVICAD simulator after stage 3 with the one found in question 4.1.

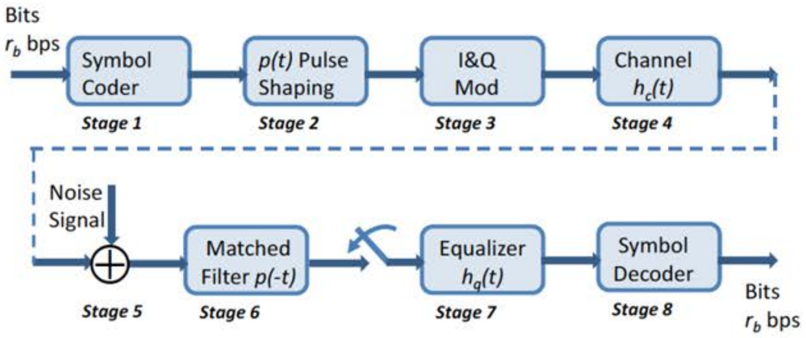

Figure 4.1. Functional block diagram of LaVICAD QAM&PAM Simulator.

Question 4.3. Draw a block diagram of a communication receiver of a QPSK digital signal that uses a coherent IQ demodulator.

<u>Activity in the lab:</u>

In this session, you must generate a QPSK signal with the LaVICAD simulator, save it into a file from the LaVICAD main menu and configure the Vector Signal Analyzer, or Keysight VSA 89600 software, to operate as a digital communication receiver for this signal.

Activity 4.1. Generation of a QPSK digital signal with LaVICAD.

To generate a digital signal with LaVICAD, first download the LavicadQAM software in your desktop (in Atenea you will find the link to download it). Then, run MATLAB and use the desktop as the current working directory (Note that this step is not necessary for students who have installed MATLAB and VSA in their own computer).

Execute LavicadQAM from MATLAB. Configure the LaVICAD simulator to generate a QPSK signal using equiprobable bits at a rate of 64 Kbit/s, a square root raised cosine (SRRC) shaping pulse and a carrier frequency of 10MHz. Use the results of Question 4.2 to select the number of samples per symbol in stage 2 and the bit rate in stage 3. Write down in your report the values you have selected for these parameters.
Then, select the option Adapt to AGILENT of the Session menu in LaVICAD to save the file that can be uploaded with the VSA.

Activity 4.2. Visualization of the signal spectrum with the Vector Signal Analyzer.
In order to visualize the QPSK signal with the VSA, execute the VSA software and upload the file saved in the previous activity. Remember that to upload a file to the VSA you may go to the menu "File/Recall/Recall Recording".
By default, one of the subplots should display the spectrum of the signal. Before proceeding, verify that the center frequency is 10 MHz and that the frequency Span is 400 KHz. The objective of this activity is that you compare this spectrum with the theoretical one obtained in Questions 4.1 and 4.2. To do so, first average 50 realizations of the spectrum using the *RMS (Video)* function in the "MeasSetup/Average" menu. Measure the signal bandwidth using the "Marker/OBW" menu and compare your result with the theoretical one. Comment the results in your report.

Activity 4.3. Configuration of the VSA as a demodulator.
First of all turn off the *RMS (Video)* function. Then, activate the demodulator selecting "MeasSetup/Measurement Type: Vector/General Purpose/Digital_Demod", and configure it through "MeasSetup/Digital Demod Properties". You will have to set up the following parameters according to the QPSK signal generated before (press Help button for comprehensive information):

a) *Format/Format* is the modulation used by the signal.
b) *Format/Symbol Rate* in number of symbols per second (i.e., 1 Hz = 1 baud)
c) *Filter/Measurement Filter* is the filter to be used in each branch of the IQ demodulator. It should be the matched filter for the shaping pulse used in the incoming modulated signal.
d) *Filter/Alpha/BT* is the roll-off factor for SRRC pulses.

Propose a set of suitable values for these parameters considering that the input signal is a QPSK signal with a SRRC pulse. Include them in your report and justify your choice.

Activity 4.4. Visualization of the Eye Diagram and Received Constellation.
Go to menu "Window/Trace Layout" and configure the display to show 4 subplots (grid 2x2) with the following traces: (A) Data "IQ Meas Time" and Format "Constellation", (B) Data "Spectrum" and Format "Log Mag" (dB), (C) Data "IQ Meas Time" and Format "Real (I)", and (D) Data "IQ Meas Time" and Format "Imag(Q)".
The signal shown in subplot (C) is the output of the matched filter of the in-phase component branch of the IQ demodulator. Similarly, the signal in (D) is the output of the matched filter of

the quadrature component branch of the IQ demodulator. Select a reduced number of symbols, for instance 10 symbols, in each subplot (C and D) to see the details of the signal properly (option "X Scale" in the "Trace" menu). The vertical lines represent the symbol clock times, that is, the time instants in which the output of the matched filter is sampled to make the decision on the transmitted symbols (decision strobe).

Subplot (A) shows the constellation of the received signal. This figure represents in a two-dimensional graph the samples at the output of the IQ demodulator in the symbol clock times, the horizontal axis corresponds to the sample of the in phase component branch and the vertical axis to the quadrature component branch. Do you observe any dispersion in the constellation? Why?

Now, change subplot (D) to show the data in "I-Eye" format. In this plot configure "X scale" as "Full scale" in the "Trace" menu. The eye diagram is the superposition of portions of the signal at the output of the matched filter of the in-phase component branch of the IQ demodulator with a duration of 2 symbol periods (it can be similarly defined for the output of the matched filter of the quadrature component branch of the IQ demodulator). A measure of the performance of a digital communication system is the relative aperture of the eye defined as *d/D%*, where *d* and *D* are shown in Figure 4.2. Measure the relative aperture of the eye diagram for the in-phase component in the VSA and explain the procedure in your report.

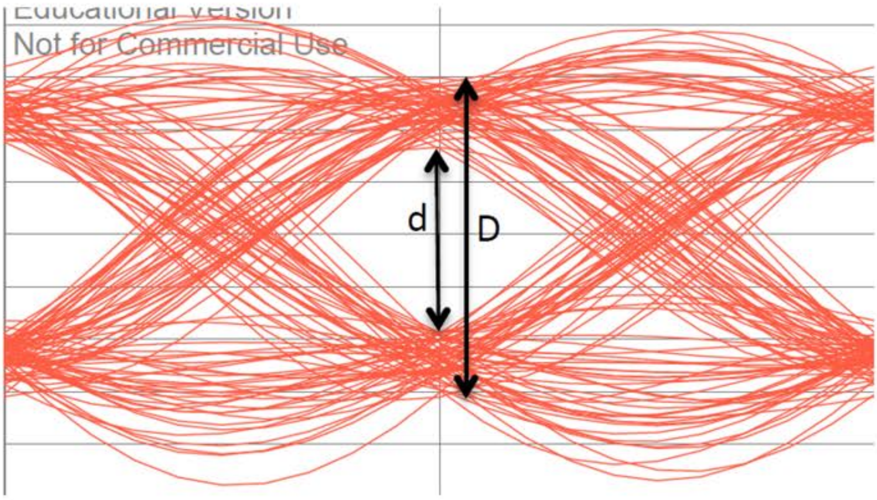

Figure 4.2. Eye Diagram with partial closure.

Activity 4.5. Performance parameters of the digital demodulator when there is a pulse mismatch.

The digital demodulator in Activity 4.4 performs perfectly because the digital demodulator is configured properly and the signal has no noise. The objective of this activity is to study the degradation observed in the demodulated signal when the demodulator filter is not matched to that one of the transmitter. For this purpose, change the roll-off factor of the measurement filter to 1. Observe the dispersion in the constellation and the closure of the eye diagram. Measure the relative aperture of the eye diagram of the in-phase component.

At the end of this activity return the matched filter roll-off factor to its original value (0.3).

Activity 4.6. Performance parameters of the digital demodulator in the presence of additive Gaussian noise.

In this activity we ask you to evaluate the degradation of digital demodulator performance in the presence of noise.

Using LaVICAD generate the QPSK signal with a SRRC pulse as in Activity 4.1 but now with an ideal channel and additive Gaussian noise with an Eb/No of 15 dB. Modify the signal using LaVICAD, save the signal in a file after stage 5, and load the saved file in the VSA. Observe the closure of the eye diagram and the dispersion in the constellation due to the noise. Comment these aspects in your report.

Session 5. Inter-Symbol Interference (ISI) and Baseband Equalization

<u>Objective</u>	• Students will better understand the effect of ISI on the received signal space • Students will have to design a linear equalizer.
<u>Background</u>	• Students should know the concepts of BER and Eb/No. • Students will use concepts in Lesson 3 of ICOM "Modulacions digitals sobre canals limitats en banda". Specifically, students should know how to design an FIR digital linear equalizer using the Zero Forcing criteria.
<u>Manufacturer's guide to consult in the lab (available at Atenea or online help)</u>	• Keysight VSA 89600 User's Guide. • Online help of Keysight VSA 89600.

Homework

Question 5.1. Solve the exercise. A sequence of independent bits at 2Mbps bit rate with $p_0 = p_1$ modulates a QPSK signal with symbols $a[k] = a_i[k] + ja_q[k]$ given in the following table

Symbol	$a_i[\cdot]$	$a_q[\cdot]$
00	*A*	*A*
01	- *A*	*A*
10	*A*	- *A*
11	- *A*	- *A*

and with a non-return to zero rectangular pulse of unit energy.

$$s(t) = \sum_{m=-\infty}^{+\infty} a_i[m]\, p(t-mT)\cos(2\pi f_c t) - \sum_{m=-\infty}^{+\infty} a_q[m]\, p(t-mT)\,\mathrm{sen}(2\pi f_c t)$$

Consider the communication system in Figure 5.2, where $h_c(t)$ is the channel impulse response, w(t) is a zero-mean white Gaussian noise with spectral density equal to $S_w(f) = N_0/2$ and uncorrelated with the useful signal, and the lowpass filter of the coherent IQ demodulator is the matched filter of the transmitted pulse, i.e. $h_{FA}(t) = p(-t)$.

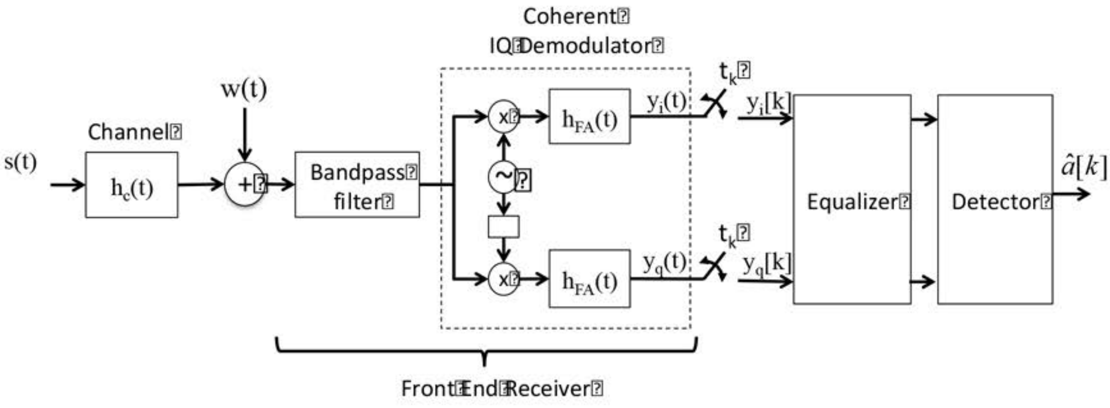

Figure 5.1 Bandpass communication system

Assuming that the decision regions of the detector are the four quadrants of the received constellation $\{y_i[k], y_q[k]\}$, answer the following questions.

a) Obtain the symbol rate, $r = 1/T$.

b) Considering $A = 1mVolts, N_0 = 5E-8$ W/Hz, obtain the Eb/No ratio in dB.

c) Assuming a channel $h_c(t) = \delta(t)$, prove that the Bit Error Rate (BER) is equal to

$$BER = Q\left(\sqrt{\frac{2E_b}{N_0}}\right)$$

where the Q-function is defined as

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_x^{+\infty} \exp\left(-\frac{\lambda^2}{2}\right) d\lambda .$$

Note that in this case the equalizer is not needed since the channel is ideal. Compute the BER for the Eb/No obtained in (b).

The equivalent lowpass system of Figure 5.1 is shown in Figure 5.2, where we have assumed the IQ demodulator demodulates the bandpass signal without phase or frequency synchronization errors. The numbers in red indicate the output of the corresponding LaVICAD stages.

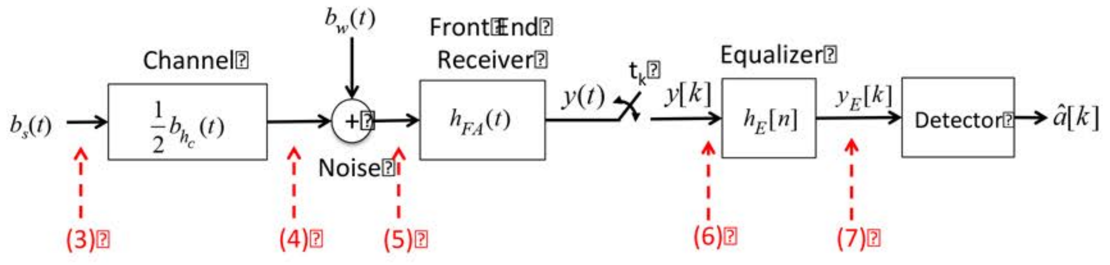

Figure 5.2 Lowpass equivalent communication system of Figure 5.1.

Assume hereafter the following equivalent lowpass signal of the channel impulse response $b_{h_c}(t) = 2\delta(t) - 0.4\delta(t-T)$. Answer the following questions.

d) Find the power spectral density of the received signal at stage (5).

e) Obtain $y(t)$, i.e. the output of the matched filter $h_{FA}(t)$.

f) Using the expression of the samples of the detected signal $y[k]=y_i[k]+jy_q[k]$, identify the terms corresponding to the useful signal, the Inter-Symbol Interference (ISI) and the noise. Plot the received constellation $\{y_i[k], y_q[k]\}$ in the absence of noise.

Question 5.2. Choose an appropriate value of the configuration parameters to simulate the communication system of Figure 5.2 with the LaVICAD simulator considering a sampling frequency equal to 16 MHz at stage (3) and a carrier frequency equal to 1GHz. In particular, select the following parameters:

- Stage 1: Modulation format, bits/symbol, Symbol distance (*mvolts*)
- Stage 2: Pulse, Rolloff, Duration/T, Samples/Symbol
- Stage 3: Bit rate, Carrier Frequency
- Stage 4: Channel impulse response
- Stage 5: EbNo
- Stage 6: You can use values by default
- Stage 7: Equalizer type, Number of taps (these parameters are provided in the activities when required)

Activity in the lab.

Activity 5.1. Visualization of the transmitted QPSK spectrum with the VSA software. Run LaVICAD from MATLAB until stage (3) to simulate the QPSK transmitted signal of the exercise solved in Question 5.2. Using the "Adapt to VSA" option save the signal generated in stage (3) and visualize it with the VSA. Recall the procedure to load files from LaVICAD to the VSA is explained in Session 4. Inside the VSA software, average 50 realizations of the spectrum using the *RMS (Video)* function and visualize the spectrum. Measure the symbol rate and the ratio in dB between the principal and secondary lobe; compare it with the theoretical ones and comment the results in your report.

Activity 5.2. Demodulation of the transmitted QPSK signal with VSA. Deactivate the *RMS (Video)* function and activate the Digital Demodulator through the *MeasSetup* menu to demodulate the transmitted QPSK signal. Recall this procedure is also explained in Session 4. Set up all the configuration parameters, i.e. format, symbol rate and measurement filter. Then, configure the screen layout to have a grid2x2 display with the traces of Spectrum (dB), IQ Constellation, I-Eye and Real(I). Identify the main differences between the eye diagram you observe now with a rectangular pulse and the one you obtained in Session 4 where a square root raised cosine was used.

Activity 5.3. Visualization of the received QPSK spectrum without noise with VSA. Execute again LaVICAD until stage (4) with an equivalent lowpass signal of the channel impulse response $b_{h_c}(t)=2\delta(t)-0.4\delta(t-T)$. Using the "Adapt to VSA" option save the signal generated in stage (4) and load it with the VSA software. Visualize the spectrum averaged with 50 realizations. What is the effect of the non-ideal channel on the spectrum? Compare it to the theoretical shape calculated in Question 5.1.

Activity 5.4. Demodulation of the received QPSK signal without noise with VSA. Deactivate the *RMS (Video)* function and activate the Digital Demodulator. Follow the steps as in Activity 5.2 and compare the results with the ones obtained there. Justify in your report the differences you observe.

Activity 5.5. Demodulation of the received QPSK signal with noise with VSA. Execute again LaVICAD until stage (5) to generate the received signal with noise,), with an Eb/No equal to the one computed in Question 5.1. Using the "Adapt to VSA" option save the signal generated in stage (5) and visualize it with the VSA. Activate the Digital Demodulator and follow the steps as in Activity 5.2. Compare the results with the ones obtained in Activity 5.4 and justify in your report the differences you observe.

Activity 5.6. Design of the equalizer. Several criteria for designing equalizers can be found in the literature. For instance, the Zero Forcing criterion, the MMSE criterion, and others more complex. You might be familiar with the Zero Forcing criterion if already seen in ICOM lectures. In this activity we ask you to compare both the received constellation (stage 6) obtained with LaVICAD without equalizer and with a Zero Forcing Equalizer with 3 coefficients in stage (7). Observe the signal space at stages (6) and (7) using the available options with and without noise. Comment the figures in your report.

Additional Lab Session. Measurement of the noise in superheterodyne receivers

Objective	• Students will learn how to measure the N_0 parameter of the noise power spectral density in a superheterodyne receiver.
Background	• Students need to be familiar with the operation and configuration of the spectrum analyzer. • Students need concepts of bandpass processes given in Chapter 3 of ICOM course, and of random variables given in PPEE.
Manufacturer's guide to consult in the lab (Available at Atenea)	• Agilent Application Note 1303. Measuring Noise and Noise-like Digital Communications Signals with Spectrum and Signal Analyzers • Slides of the Spectrum Analyzer of Session 1 in the lab ICOM course.

Homework

Question 1.1 Bandpass noise at the IF stage in a communication receiver. Assume that at the input of the spectrum analyzer (SA) IF filter you have a zero mean white Gaussian noise denoted by $w(t)$ with $S_w(f)=N_0/2$ in W/Hz. Under this assumption, when the SA works with the Zero Span mode, that is as a superheterodyne communication receiver, the signal at the output of the Intermediate Frequency (IF) filter is a bandpass noise that results from filtering $w(t)$ with the IF bandpass filter.

Considering a zero mean white Gaussian noise with $N_0(\text{dBm/Hz}) = 10\log_{10}(N_0(mW/Hz))$ = -140 dBm/Hz at the input of the spectrum analyzer IF filter, which is configured with a center frequency of 100 Mhz, a Zero Span mode, a RBW=1 Khz and VBW=10Khz, ATT=0dB show in a figure the Power Spectral Density (PSD) of the noise $n(t)$ at the output of the IF filter.

Based on this representation, provide an expression of the mean power of the noise $n(t)$, denoted by P_N(dBm), as a function of N_0(dBm/Hz) and the RBW. For this, take into account that the Noise Equivalent Bandwidth is equal to

$$B_N \triangleq \begin{matrix}\text{Noise Equivalent}\\ \text{Bandwidth}\end{matrix} = 1.128 \cdot B_{-3dB}^{IF}$$

where B_{-3dB}^{IF} is the bandwidth of the IF filter at -3dB. Calculate the noise mean power P_N(dBm) with the configuration of the SA indicated previously.

Activity in the lab:

Activity 1.1 Impact of ATT, RBW and VBW configuration parameters on the noise power. Leaving the input of the SA as an open circuit, configure the SA with the following values of the configuration parameters REF=-60 dBm, Center Frequency=100 Mhz, Attenuation = 0dB, Span=0, RBW=1 Khz and VBW=10Khz. With this configuration what you observe is a measurement somehow related to the power of the noise at the output of the IF filter. Use the Trace Average Mode functionality of the SA to average the trace and obtain a more reliable measurement. The objective of this activity is that you get familiar with the impact of the Attenuation (ATT), the RBW and VBW parameters on that measurement. Therefore, measure this value for different values of ATT, RBW and VBW and comment on the results. Important: please, note that the value you measure is not equal to P_N as explained hereafter.

Question 1.2 Noise mean power at the IF stage in a communication receiver. Indeed, the bandpass noise $n(t)$ at the output of the IF filter is a zero mean Gaussian wide sense stationary random process with variance equal to $\sigma_n^2 = E[|n(t)|^2]$. The mean power of $n(t)$ measured over an impedance of R=50Ω is equal to

$$P_N = \frac{1}{R} E\left[|n(t)|^2\right].$$

Show that P_N in dBm is equal to

$$P_N(dBm) = 10 \log_{10}\left(\sigma_n^2\right) + 10 \log_{10} 20 \tag{1}$$

Based on this equation and on the result of Question 1.1, observe that N_0(dBm/Hz) can be obtained if we measure σ_n^2. However, the SA does not measure σ_n^2; instead, as explained below, it measures the envelope of $n(t)$.

Activity 1.2 Measurement of the noise mean power P_N(dBm) in the SA. As before, leave the input of the SA as an open circuit, configure the SA with the following values of the configuration parameters REF=-60 dBm, Center Frequency=100 Mhz, Attenuation = 0dB, Span=0, RBW=1 Khz and VBW=10Khz. Since the output of the IF filter, $n(t)$, is the input of the envelope detector, in the screen you should observe the successive realizations of the envelope of the noise signal $n(t)$ at the output of the IF filter loaded over an impedance of R=50Ω. That is, each trace is

$$10 \log_{10}\left(\frac{1}{2R} e_n^2(t) 10^3\right) \text{dBm},$$

where $e_n(t)$ denotes the envelope of $n(t)$ and is equal to

$$e_n(t) \triangleq \begin{matrix}\text{Envelope}\\ \text{of } n(t)\end{matrix} = \sqrt{n_i^2(t) + n_q^2(t)} \tag{2}$$

with $n_i(t)$ and $n_q(t)$ the in phase and quadrature components of the bandpass signal $n(t)$.

Now, use the Trace Average Mode functionality of the SA to average the trace and obtain a more reliable estimation of this value. Select a number of averages until you observe a much more smooth line. Measure the averaged power to be denoted hereafter by $P_{meas}(dBm)$.

Question 1.3. Analytic expression of P_{meas}(dBm). Considering R=50Ω, $P_{meas}(dBm)$ is equal to

$$P_{meas}(dBm) \simeq 20 \cdot E\left[\log_{10}\left(e_n(t)\right)\right] + 10 \qquad (3)$$

where $E[\cdot]$ is the expectation operator. Please, note that equation (3) is diferent from equation in (1). Our objective is then to find the relation between $P_N(dBm)$ and $P_{meas}(dBm)$.

The stochastic processes $\{n_i(t)\}$ and $\{n_q(t)\}$ are known to be both wide sense stationary Gaussian random processes with zero mean and variance equal to σ_n^2, and uncorrelated between them. Therefore, the envelope given in equation (2) is known to be a wide sense stationary random process with a Rayleigh distribution, that is at each time instant t the envelope $e_n(t) \geq 0$ is a Rayleigh random variable with a probability distribution function (pdf) equal to

$$f_{e_n}(e_n) = \frac{e_n}{\sigma^2} \cdot e^{-\frac{e_n^2}{2\sigma^2}} \quad ; e_n \geq 0 .$$

with mean and second order moment equal to

$$E[e_n] = \sigma_n \sqrt{\frac{\pi}{2}} \quad ; \quad E[e_n^2] = 2\sigma_n^2 .$$

However, the measured power given in equation (3) computes the expectation of logarithm of the envelope, which is not equal to the logarithm of the expectation. Therefore, consider the following result of log-Rayleigh random variables.

Log-Rayleigh random variable. Given a Rayleigh random variable (rv) y with pdf

$$f_Y(y) = \frac{y}{\sigma^2} \cdot e^{-\frac{y^2}{2\sigma^2}} \quad ; y \geq 0,$$

the rv $z = \ln y$ is a log-Rayleigh rv with pdf shown in Figure 1 and equal to

$$f_Z(z) = \frac{e^{2z}}{\sigma^2} \cdot e^{-\frac{e^{2z}}{2\sigma^2}}$$

The mean and variance of a Log-Rayleigh random variable z are

$$E[z] = \ln\sigma + 0.058$$ [1],

$$E\left[(z - E[z])^2\right] = \frac{\pi^2}{24}.$$

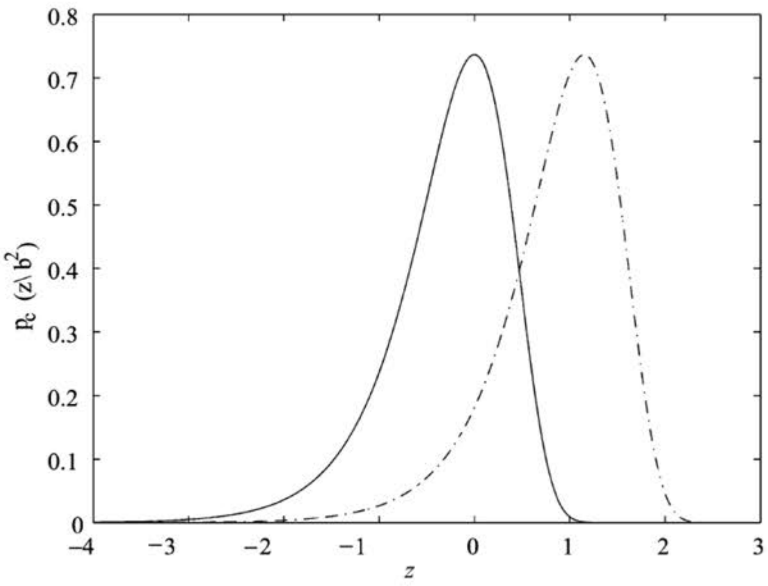

Figure 1. Log-Rayleigh pdf with $\sigma^2 = 1$ (solid line) and $\sigma^2 = 10$ (dashed line).

Taking into account all these results, prove that the measured power $P_{meas}(dBm)$ in (3) satisfies

$$P_{meas}(dBm) \simeq 10 \cdot \log_{10} \sigma_n^2 + \frac{20}{\ln 10} 0,058 + 10 \quad (4)$$

Activity 1.3 Measurement of the noise power spectral density N_0(dBm/Hz) of the SA. Using (4), (1) and the result of Question 1.1, prove the following expression, that is useful to estimate N_0(dBm/Hz) from $P_{meas}(dBm)$.

$$N_0(dBm/Hz) = P_{meas}(dBm) + 2{,}5dB - 10\log_{10} RBW - 0{,}52 \quad (5)$$

Calculate the noise power spectral density, N_0(dBm/Hz), with (5) measuring $P_{meas}(dBm)$. Compare this result with the value provided by the SA with a center frequency of 100MHz, and SPAN=10MHz. You may use the function *Marker Function/Noise* to check the value of N_0(dBm/Hz) provided by the SA.

[1] The constant 0.058 approximates $\frac{ln2}{2} - \frac{\gamma}{2}$, where γ is the Euler's constant.

Activity 1.4 Validation of the measured N_0(dBm/Hz). The purpose of this activity is to validate the estimation of the noise power spectral density, N_0(dBm/Hz) found in Activity 1.3 and given by equation (5). To do so, keep the same configuration as in the previous activity and obtain the measured noise mean power P_N(dBm) with the option *Meas/Channel Power*. You may select a 50 kHz channel bandwidth around the central frequency of 100 MHz. Compare this value provided by the SA with the one obtained using the results you obtained in Question 1.1 and the estimated N_0(dBm/Hz) in A1.4.

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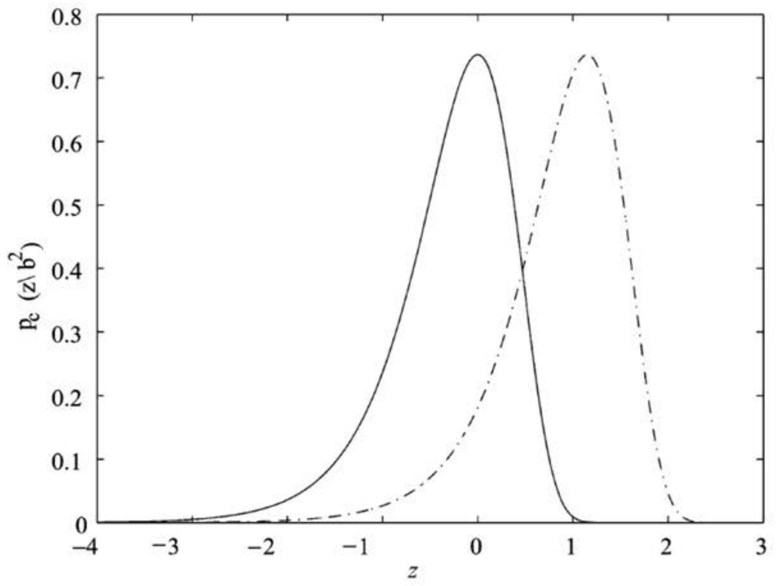

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◆

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