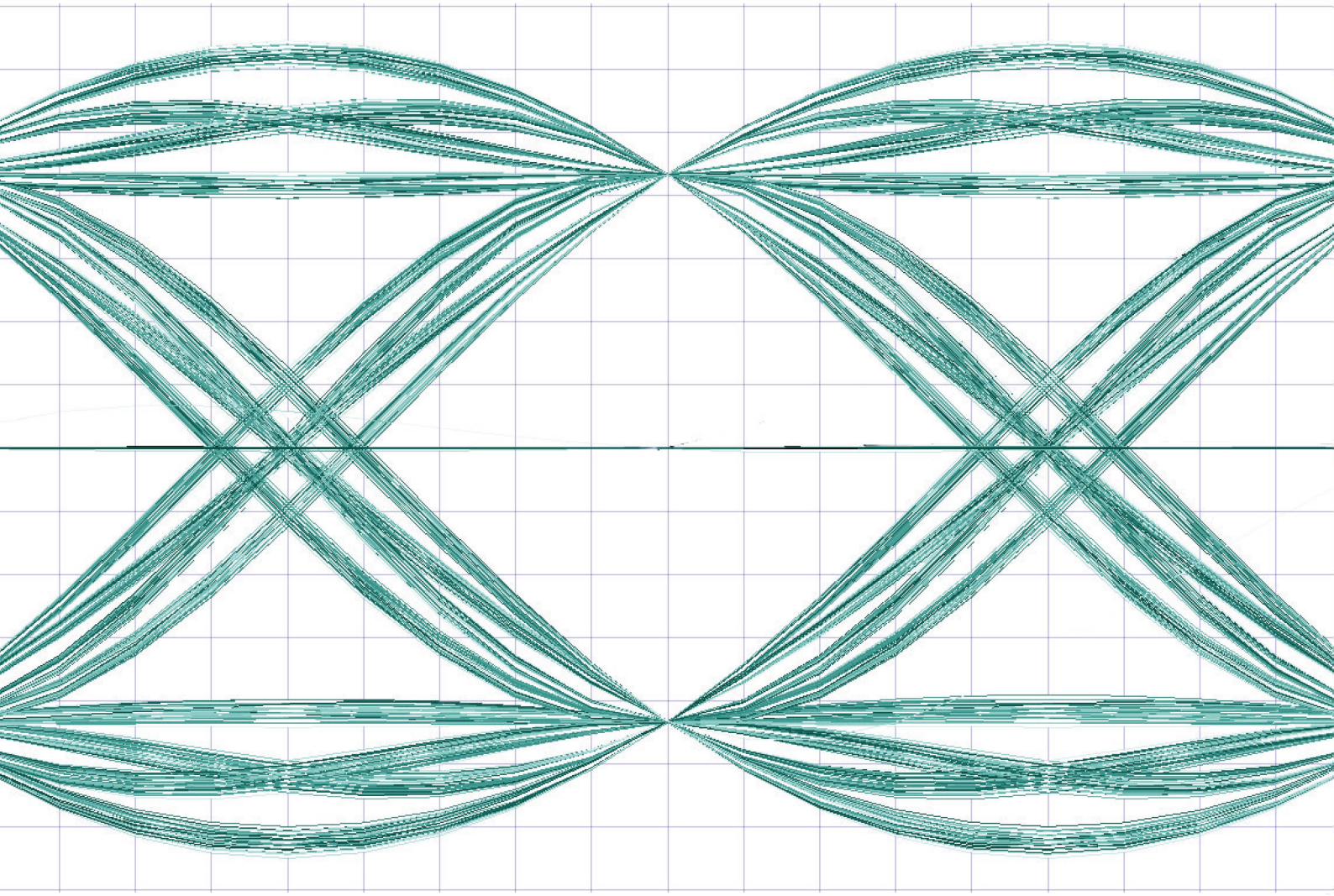

# INTRODUCTION TO COMMUNICATION SYSTEMS

## Lab Based Learning with NI USRP™ and LabVIEW Communications

Bruce A. Black

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# Introduction to Communication Systems

## Lab Based Learning with NI USRP and LabVIEW Communications

## Student Lab Manual

Bruce A. Black

NATIONAL INSTRUMENTS

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# Table of Contents

# Preface

The twelve lab exercises presented in this package are intended to accompany an introductory course in communication systems offered at the junior or senior level in an electrical or computer engineering program. The lab exercises use the NI USRP software defined radio platform; no additional laboratory equipment is needed, other than a computer to run LabVIEW Communications and to interface with the USRP. The USRP transceivers are operated in loopback mode with a coaxial cable and attenuator connecting the transmitter to the receiver.

The first lab exercise is an introduction to the USRP. This exercise is brief to allow time for an instructor to add an introduction to LabVIEW Communications for students who are unfamiliar with the programming environment. Labs two through six introduce various aspects of analog communication, including AM, frequency-division multiplexing, image rejection, double-sideband suppressed-carrier modulation, and FM. Labs six through twelve introduce various digital signaling techniques, including amplitude-shift keying, frequency-shift keying, binary phase-shift keying, the eye diagram, equalization, and quadrature phase-shift keying. Introducing analog modulation as a lead-in to digital techniques follows the approach used in most current communication systems textbooks.

These lab exercises are intended for students who have a basic background in signals and systems, and are beginning study of communication systems. Some prior experience with LabVIEW would be helpful, but this can be provided by the instructor at the beginning of the lab sequence. These lab exercises are intended to accompany a traditional communication systems course; they are not intended for self study. Each of the lab project descriptions begins with a background discussion, but these have been kept brief, and it is assumed that the accompanying course will provide detailed background as well as context for the lab projects.

Each of the lab projects after the introductory lab includes a prelab assignment, an in-lab exercise, and a lab report. Generally, each prelab assignment involves creating LabVIEW "virtual instruments" (VI's) to implement a transmitter and a receiver using a specified modulation method. Templates are provided to help students structure their programs and to assist in interfacing to the USRP. During the laboratory sessions, students will try out their virtual instruments on the USRP's, and correct errors. Some of the lab projects ask students to explore the effects of varying modulation parameters, other projects involve creating additional VI's to explore alternative methods such as differential phase-shift keying. Students are expected to submit their working programs and functions accompanied by documentation and measurement results after completing each lab exercise.

The twelve lab projects should be sufficient to support a one-semester course. For a one-quarter course spanning ten weeks, Labs 3 and 4 can be omitted without compromising continuity.

The USRP provides a powerful and flexible platform for learning about communication systems. They also provide a significant opportunity for advanced experimentation, beyond the scope of the lab projects presented here.

In order to complete labs 2 - 12, you will need to download the corresponding LabVIEW program files. Download these files from http://www.ni.com/white-paper/52344/en/ and unzip the files to a convenient location. You will access them as instructed by each lab.

Instructors, please visit www.ntspress.com/publications/usrp-labs/ for more information.

LAB 1

# Introduction to the USRP

# 1.1 Objective

The purpose of this introductory laboratory exercise is to ensure that students have a working installation of LabVIEW Communications on their computers and know how to connect to the USRP software defined radio.

# 1.2 Background

The Wireless Innovation Forum defines Software Defined Radio (SDR) as:

"Radio in which some or all of the physical layer functions are software defined." [1]

SDR refers to the technology wherein software modules running on a generic hardware platform are used to implement radio functions. By combining the NI USRP hardware with LabVIEW software you can create a flexible and functional SDR platform for rapid prototyping of wireless signals including  physical layer design, record and playback, signal intelligence, algorithm validation, and more.

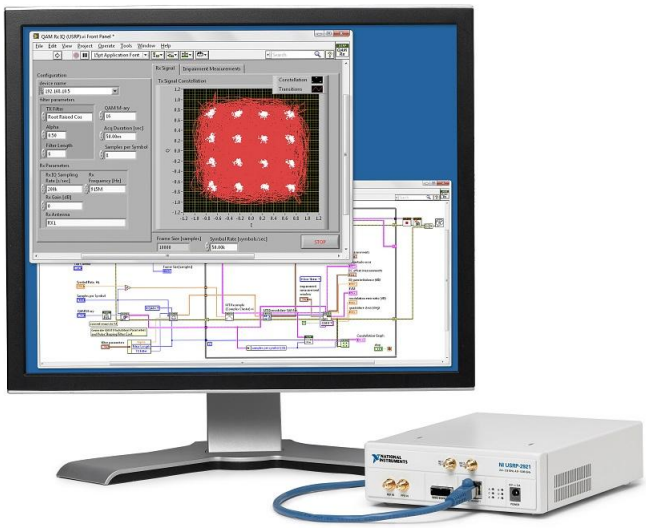


**Figure 1.  Hardware Setup in a Wireless Communications Lab**

## NI USRP Hardware

The NI USRP connects to a host PC creating a software defined radio. Incoming signals at the SMA connector inputs are mixed down using a direct-conversion receiver to baseband I/Q components, which are sampled by a analog-to-digital converter (ADC). The digitized I/Q data follows parallel

[1] http://www.wirelessinnovation.org/what_is_sdr

paths through a digital downconversion (DDC) process that mixes, filters, and decimates the input signal to a user-specified rate. The downconverted samples are passed to the host computer.

For transmission, baseband I/Q signal samples are synthesized by the host computer and fed to the USRP at a specified sample rate over Ethernet, USB or PCI express. The USRP hardware interpolates the incoming signal to a higher sampling rate using a digital upconversion (DUC) process and then converts the signal to analog with a digital-to-analog converter (DAC). The resulting analog signal is then mixed up to the specified carrier frequency.

More information about NI SDR hardware can be found in the respective Getting Started Guide[2] available in the start menu.

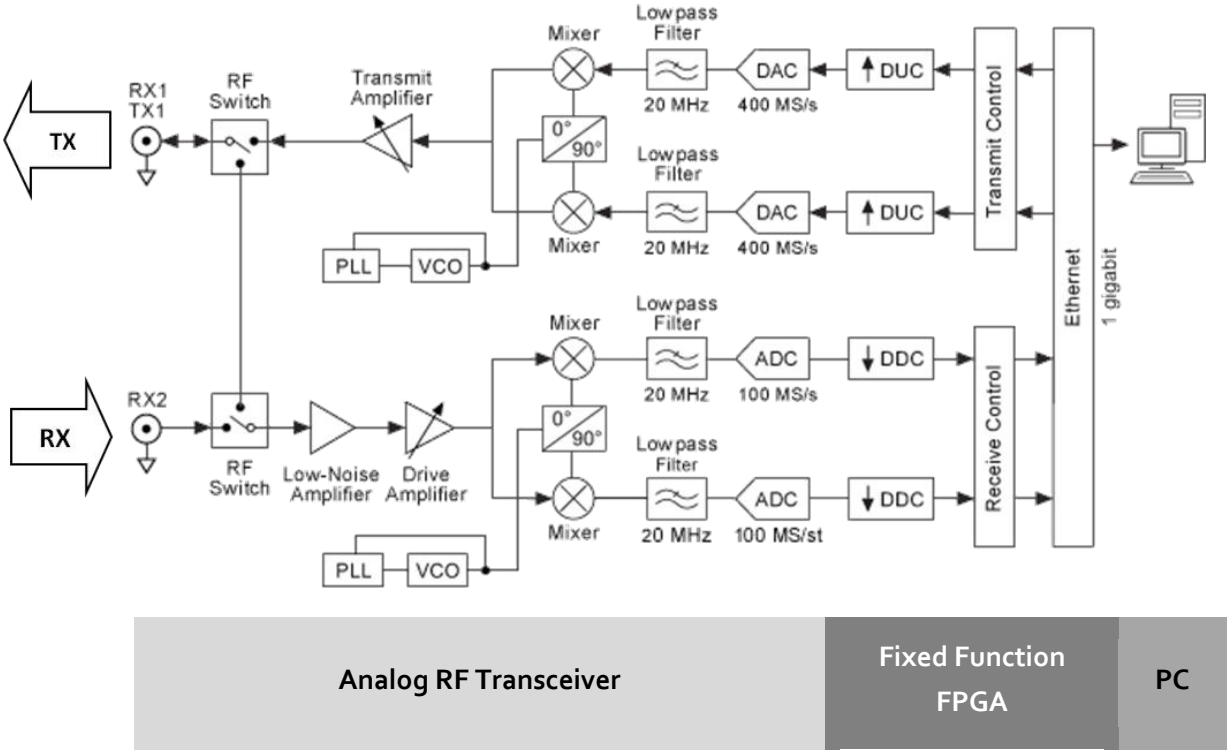



**Figure 2. Typical Block Diagram of an NI USRP**

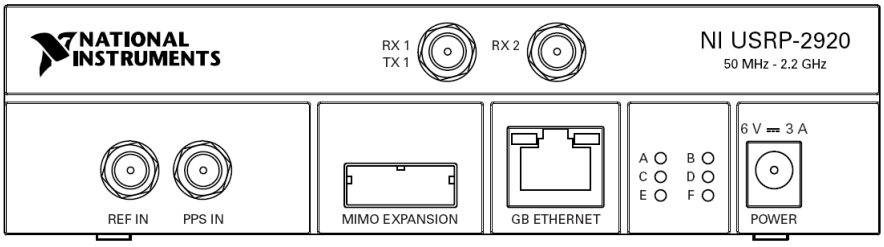



**Figure 3. Front View of an NI USRP-2920 Software Defined Radio**

[2] Available from the Start Menu→All Programs→National Instruments→NI-USRP→Documentation

## NI LabVIEW Communications System Design Software

LabVIEW is a graphical programming language developed by National Instruments. The basic building block of LabVIEW is the virtual instrument (VI). Conceptually, a VI is analogous to a procedure or function in conventional programming languages. Each VI consists of a *block diagram* and a *front panel*. The block diagram describes the functionality of the VI, while the front panel is a top level interface to the VI. The construct of the VI provides two important virtues of LabVIEW: code reuse and modularity. The graphical nature of LabVIEW provides another virtue: it allows developers to easily visualize the flow of data in their designs. NI calls this Graphical System Design. Also, since LabVIEW is a mature data flow programming language, it has a wealth of existing documentation, toolkits, and examples which can be leveraged in development.

In this course you will use National Instruments SDR hardware. LabVIEW provides a simple interface for configuring and operating various external I/O, including the NI SDR hardware used in lab. This is the main reason why you will use LabVIEW as the programming language to build an SDR in this course. You should realize that the algorithms considered here could also be programmed in optimized C/C++, assembly, or VHDL and implemented on a DSP, microcontroller, or an FPGA. The choice of hardware and software in this lab is mostly a matter of convenience.

In future labs you will need to be familiar with LabVIEW and the documentation/help available to you. This is the only lab in this course which will give you the opportunity to learn and practice LabVIEW programming; so it is important that you take this opportunity to ask the instructor any questions you might have about LabVIEW programming. The following tutorials and reference material will help guide you through the process of learning LabVIEW:

- LabVIEW Communications System Design Suite 1.0 Online Manual [3]
- LabVIEW Communications Guided Help tutorials
- Context help[4]

The context help window displays basic information about LabVIEW objects when you move the cursor over each object. To toggle the display of the context help window, select View » Context Help or press <Ctrl-H>.

[3] http://www.ni.com/documentation/en/labview-comms/1.0/manual/labview-comms-manual/
[4] Context help is available in LabVIEW Communications after opening a program from View→Context Help

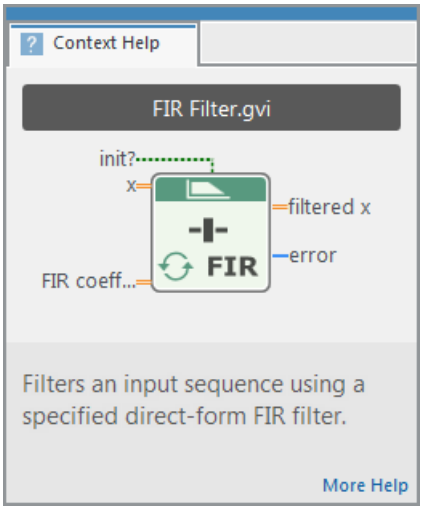



**Figure 3. Context Help Window**

The LabVIEW online help is the best source of detailed information about specific features and functions in LabVIEW. Online help entries break down topics into a concepts section with detailed descriptions and a how-to section with step-by-step instructions for using LabVIEW functions.

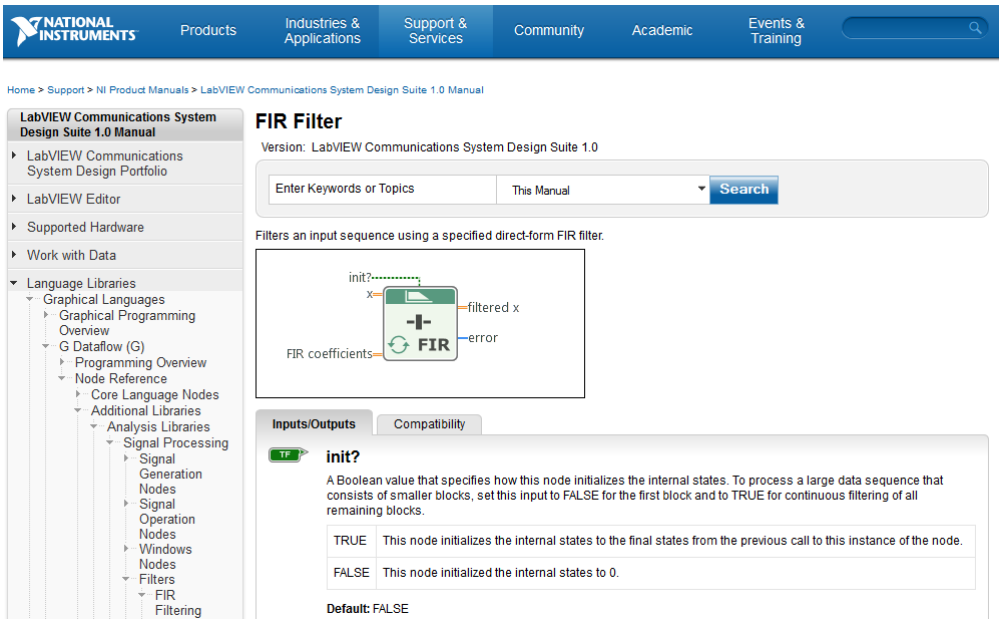



**Figure 4. Screenshot of LabVIEW Online Help**

# 1.3 Pre-Lab

In order to complete labs 2 - 12, you will need to download the corresponding LabVIEW program files. Download these files from http://www.ni.com/white-paper/52344/en/ and unzip the files to a convenient location. You will access them as instructed by each lab.

Instructors, please visit www.ntspress.com/publications/usrp-labs/ for more information.

1. Ensure LabVIEW Communications is installed
2. Open LabVIEW Communications
3. From the main window (also called the lobby), open and complete the guided help tutorial from Learn→Getting Started→Introduction to the LabVIEW Editor
4. To return to the lobby, select File→Close All
5. Complete 5 other guided help tutorials to learn more about LabVIEW Communications. From the lobby click Learn→Programming Basics to find the following:
   - Designing a User Interface
   - Debugging your VI
   - Basic Data Types
   - Arrays
   - While Loops

Bring any questions or concerns regarding LabVIEW or these tutorials to your instructor's attention. For the remainder of this lab you should be familiar with the basics of LabVIEW programming and where to look for help.

# 1.4 Lab Procedure

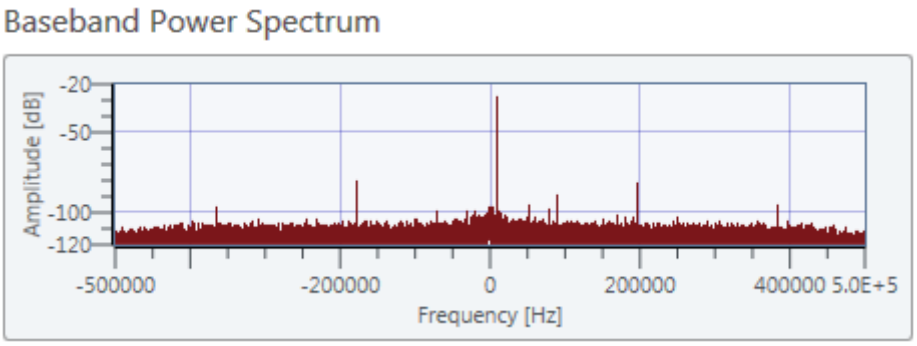
1. Connect the TX1 output to the RX2 SMA connector using a loopback cable and 30 dB attenuator provided.
2. Connect the USRP software defined radio to the computer as described in the Getting Started Guide[5] for your NI USRP transceiver.
3. Launch the NI USRP Configuration Utility[6] to find the *Device Name* for your NI USRP device.
4. From the lobby in LabVIEW Communications, open the following NI USRP example:

   Examples→Hardware Input and Output→NI USRP Host→Single Device→Single Channel→Continuous→RX Continuous Async

5. Give the example a project name and click Create.
6. From the example, open another NI USRP example:

   File→Examples→Hardware Input and Output→NI USRP Host→Single Device→Single Channel→Continuous→TX Continuous Async

7. Give the example a project name and click Create.
8. On the Tx Continuous Aysnc example (referred to as the transmitter program), enter the device name you found using the USRP configuration utility and note the value of the *tone frequency* control. This program generates a single frequency tone at baseband and sends it to the USRP.
9. Run the transmitter program.
10. On the Rx Continuous Async example (referred to as the receiver program) example window, enter the same device name as the transmitter and change the *Active Antenna* to RX2.
11. Run the receiver program and analyze the Baseband Power Spectrum Graph. You should see a spike near the center of the graph. This is the single tone that was generated by the transmitter.




12. Without changing the value of the *Carrier Frequency* control on the receiver or transmitter program, "move" the location of the single tone on the Baseband Power Spectrum graph to 150 kHz.

[5] Available from the Start Menu→All Programs→National Instruments→NI-USRP→Documentation
[6] Available from the Start Menu→All Programs→National Instruments→NI-USRP

Note: Changes made to the program do not apply while it is running. You should stop the program and start it for change to take effect.

Questions

1. On the *Diagram* of the receiver program, write the name of each node (sometimes called function or block) and its purpose on the block diagram. (Such as niUSRP Open Rx Session creates a session handle to the device for other functions).
2. On the *Diagram* of the transmitter program, write the name of each node and its function on the block diagram. (Such as niUSRP Open Tx Session creates a session handle to the device for other functions).
3. Describe how you changed the "spike" on the Baseband Power Spectrum graph to 150 KHz.

# 1.5 Report

Submit all of the answers to the *Questions* section above.

LAB 2

# Amplitude Modulation

Prerequisite: Lab 1 – Introduction to the USRP

# 2.1 Objective

This laboratory exercise has two objectives. The first is to gain a firsthand experience in actually programming the USRP to act as a transmitter and a receiver. The second is to investigate classical analog amplitude modulation and the envelope detector.

# 2.2 Background

## Amplitude Modulation

Amplitude modulation (AM) is one of the oldest of the modulation methods. It is still in use today in a variety of systems, including, of course, AM broadcast radio. In digital form it is the most common method for transmitting data over optical fiber.

If $m(t)$ is a baseband "message" signal with a peak value $m_p$ and $A\cos(2\pi f_c t)$ is a "carrier" signal at carrier frequency $f_c$, then we can write the AM signal $g(t)$ as

$$g(t) = A\left[1+\mu\frac{m(t)}{m_p}\right]\cos(2\pi f_c t), \tag{1}$$

where the parameter $\mu$ is called the "modulation index" and takes values in the range $0 \le \mu \le 1$ (0 to 100%) in normal operation. For the special case in which $m(t) = m_p \cos(2\pi f_m t)$, we can write

$$\begin{aligned} g(t) &= A\left[1+\mu\cos(2\pi f_m t)\right]\cos(2\pi f_c t) \\ &= A\cos(2\pi f_c t) + \frac{A}{2}\mu\cos\left[2\pi(f_c - f_m)t\right] + \frac{A}{2}\mu\cos\left[2\pi(f_c + f_m)t\right]. \end{aligned} \tag{2}$$

In the above expression the first term is the carrier, and the second and third terms are the lower and upper sidebands, respectively. Figure 1 is a plot of a 20 kHz carrier modulated by a 1 kHz sinusoid at 50% and 100% modulation.

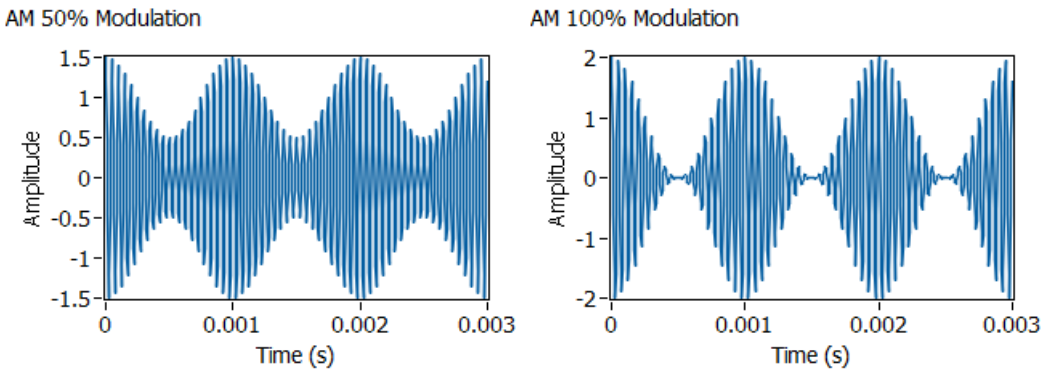



**Figure 1. Amplitude Modulated Signals**

When the AM signal arrives at the receiver, it has the form

$$r(t) = D\left[1 + \mu \frac{m(t)}{m_p}\right] \cos\left(2\pi f_c t + \theta\right), \tag{3}$$

where the carrier amplitude $D$ is usually much smaller than the amplitude $A$ of the transmitted carrier and the angle $\theta$ represents the difference in phase between the transmitter and receiver carrier oscillators. We will follow a common practice and offset the receiver's oscillator frequency $f_0$ from the transmitter's carrier frequency $f_c$. This provides the signal

$$r_1(t) = D\left[1 + \mu \frac{m(t)}{m_p}\right] \cos\left(2\pi f_{IF} t + \theta\right), \tag{4}$$

where the so-called "intermediate" frequency is given by $f_{IF} = f_c - f_0$. The signal $r_1(t)$ can be passed through a bandpass filter to remove interference from unwanted signals on frequencies near $f_c$. Usually the signal $r_1(t)$ is amplified as well.

Demodulation of the signal $r_1(t)$ is most effectively carried out by an envelope detector. An envelope detector can be implemented as a rectifier followed by a lowpass filter. The envelope $A(t)$ of $r_1(t)$ is given by

$$A(t) = D\left[1 + \mu \frac{m(t)}{m_p}\right] = D + \frac{D\mu}{m_p} m(t). \tag{5}$$

# Setting up the USRP

*Transmitter*

LabVIEW interacts with the USRP transmitter by means of four functions located on the block diagram's palette under Hardware Interfaces→NI-USRP→Tx. Figure 2 shows the basic transmitter structure. This structure is the starting point for all of the laboratory exercises in this series.

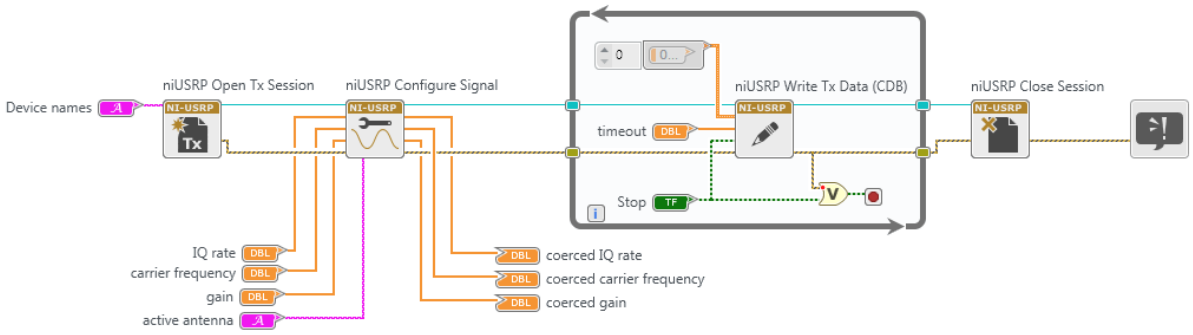



**Figure 2. Transmitter Template**

*Open Tx Session* initiates the transmitter session and generates a session handle and an error cluster that are propagated through all four functions. When you use this function, you must add a control called "device names" that you will use to inform LabVIEW of the IP address or resource name of the USRP.

*Configure Signal* is used to set parameter values in the USRP. Attach four controls and three indicators to this function as shown in the figure. To get started, set the IQ rate to 200 kSa/s (the lowest possible rate), the carrier frequency to 915.1 MHz, the gain to 0 dB, and the active antenna to TX1. When the function runs, the USRP will return the actual values of these parameters. These values will be displayed by the indicators you connected. Normally the actual parameter values will match the desired values, but if one or more of the desired values is outside the capability of the USRP, the nearest acceptable parameter value will be chosen, rather than returning an error.

*Write Tx Data* writes the baseband signal to the USRP for transmission. Placing this function in a *while* loop allows a block of baseband signal samples to be sent over and over until the "stop" button is pressed. Note that the *while* loop is programmed to terminate if an error is detected. Baseband signal samples can be provided to the *Write Tx Data* as either an array of complex

numbers or as a complex waveform data type. The Configure ribbon at the top of LabVIEW Communications allows you to choose the data type. If the baseband signal is expressed as

$$\tilde{g}(nT_x) = g_I(nT_x) + jg_Q(nT_x), \tag{6}$$

then the signal transmitted by the USRP is

$$g(t) = Ag_I(t)\cos(2\pi f_c) - Ag_Q(t)\sin(2\pi f_c t). \tag{7}$$

In this expression, the constant $A$ is set by the "gain" parameter and $f_c$ is the carrier frequency. The sampling interval $T_x$ is the reciprocal of the "IQ rate." Note that the signal $g(t)$ produced by the USRP is a continuous-time signal; the discrete-to-continuous conversion is done inside the USRP.

Observe that the baseband signal $\tilde{g}(nT_x)$ is actually two baseband signals. By long-standing tradition, the real part $g_I(nT_x)$ is called the "in-phase" component of the baseband signal and the imaginary part $g_Q(nT_x)$ is called the "quadrature" component of the baseband signal. The AM signal that we will generate in this lab project uses only the in-phase component, with

$$g_I(t) = A\left[1 + \mu\frac{m(t)}{m_p}\right], \tag{8}$$

and

$$g_Q(t) = 0. \tag{9}$$

We will explore other modulation methods in subsequent lab projects that use both components.


*Close Session* terminates transmitter operation once the *while* loop ends. Note that the function should be terminated using the STOP button rather than with "Abort Execution" on the toolbar. This is so that the Close Session function will be sure to run and will correctly close out the data structures that the function uses.

*Receiver*

LabVIEW interacts with the USRP receiver by means of six functions located on the block diagram in Hardware Interfaces→NI-USRP→Rx. Figure 3 shows the basic receiver structure. This structure is the starting point for all of the laboratory exercises in this series.

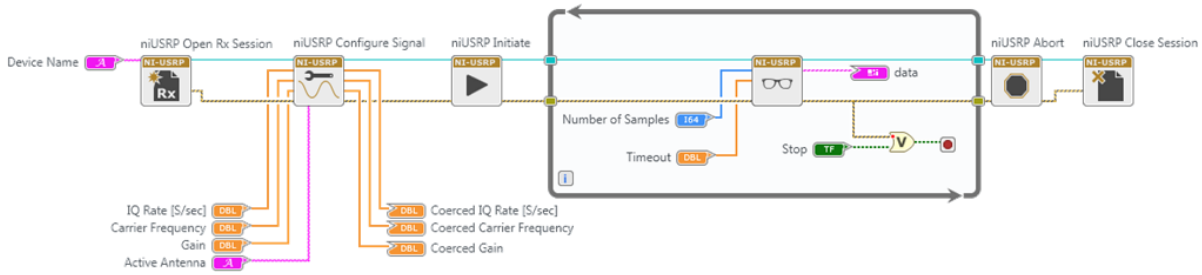



**Figure 3. Receiver Template**

*Open Rx Session* initiates the receiver session and generates a session handle and an error cluster that are propagated through all six functions. You must add a control called "device names" that you will use to inform LabVIEW of the IP address or resource name of the USRP.

*Configure Signal* has the same function as the corresponding function in the transmitter. Attach four controls and three indicators to this function as shown in the figure. This time, set the IQ rate to 1 MSa/s, the carrier frequency to 915.0 MHz, the gain to 0 dB, and the active antenna to RX2. When the function runs, the USRP will return the actual values of these parameters.

*Initiate* sends the parameter values you selected to the receiver and starts it running.

*Fetch Rx Data* retrieves the message samples received by the USRP. Placing this function in a *while* loop allows message samples to be retrieved one block at a time until the "stop" button is pressed. Note that the *while* loop is programmed to terminate if an error is detected. A "number of samples" control allows you to set the number of samples that will be retrieved with each pass through the *while* loop. In later lab projects in this series, we will not use the *while* loop, and will fetch only a single block of data from the receiver. *Fetch Rx Data* can provide message samples to the user as either an array of complex numbers or as a complex waveform data type. A pull-down tab allows you to choose the data type for the message samples.

*Abort* stops the acquisition of data once the *while* loop ends.

*Close Session* terminates receiver operation. As noted above, use the STOP button to terminate execution so that *Close Session* will be sure to run.

*File Hierarchy*

To access the files and functions specific to this course double click the *Communication Systems.lvproject.* After the project is loaded, you should see the Files Pane located on the top left of the screen, just below the ribbon. Files and functions can be accessed by expanding the folders and double clicking the file you would like. You can also drag and drop files from this pane to the block diagram to use it.

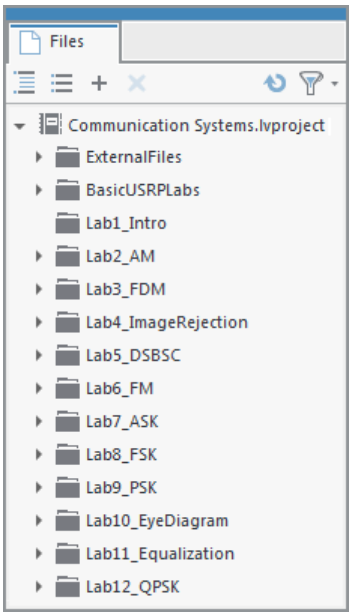



**Figure 4. Files Pane in LabVIEW Communications**

# 2.3 Pre-Lab

## Transmitter

1. A template for the transmitter has been provided in the file *Lab2TxTemplate.gvi*. This template contains the four interface functions described in the Background section above along with a "message generator" (Basic Multitone) that is set to produce a message signal consisting of three tones. The three tones are initially set to 1, 2, and 3 kHz, but these frequencies can be changed using front-panel controls. Your task is to add blocks as needed to produce an AM signal, and then to pass the AM signal into the *while* loop to the Write Tx Data function. The modulation index is to be user-settable in the range $0 \leq \mu \leq 1$, and a front-panel control has been provided.

Hint: The AM signal you generate will be $g_I(nT)$. For $g_Q(nT)$ set up an array the same length as $g_I(nT)$ containing all zeros. Then combine the two into a single complex array $\tilde{g}(nT) = g_I(nT) + jg_Q(nT)$. You can use the MathScript node to implement the AM Modulation formula easier if you know .m file syntax

Notes:

a. The message generator creates a signal that is the sum of a set of sinusoids of equal amplitude. You can choose the number of sinusoids to include in the set, you can choose their frequencies, and you can choose their common amplitude. The initial phase angles of the sinusoids are chosen at random, however, and will be different every time you run the program. This will make the message signal look somewhat different every time you run the program.

b. There is one practical constraint imposed by the D/A converters in the USRP: Scale the signals you generate so that the peak value of $\left|\tilde{g}(nT)\right|$ does not exceed +/-1. (+/-1 usually refers to full scale on the DAC in a device. Any value higher will result in clipping.)

c. Save your transmitter in a file whose name includes the letters "AMTx" and your initials (e.g., *AMTx_BAB.gvi*).

## Receiver

2. A template for the receiver has been provided in the file *Lab2RxTemplate.gvi*. This template contains the six interface functions described in the Background section above along with a waveform graph on which to display your demodulated output signal.

   Pass the complex array returned by *Fetch Rx Data* through a bandpass filter. Filters can be found in the ExternalFiles folder in the Files Pane. Use a fifth-order Chebyshev Filter (CDB) with a high cutoff frequency of 105 kHz and a low cutoff frequency of 95 kHz. The default passband ripple of 0.1 dB is acceptable. The sampling frequency input to the filter should be the "actual IQ rate" obtained from the Configure Signal function.

3. Using the *Complex to Re/Im* function, extract the real part of the Chebyshev bandpass output "Filtered X". The real part you obtain can be expressed as shown in Eq. (4). To extract the envelope, take the absolute value and pass the result through a lowpass filter. This acts as a full-wave rectifier. Then, to extract the message, pass the signal through the lowpass filter. For the lowpass filter, use a second-order Butterworth Filter (DBL) with a cutoff frequency of 5 kHz. As was the case for the bandpass filter, the sampling frequency input to the lowpass filter should be the "actual IQ rate" obtained from the *Configure Signal*. The output of your lowpass filter should be connected to the Baseband Output graph.

   Save your receiver in a file whose name includes the letters "AMRx" and your initials (e.g., *AMRx_BAB.gvi*).

## Questions

1. Suppose the message $m(t)$ is given by

$$m(t) = \cos(2\pi 1000t) + \cos(2\pi 2000t) + \cos(2\pi 3000t). \tag{10}$$

   Find and plot the power spectrum of $r_1(t)$ given by Eq. (4). Leave your answer in terms of $D$ and $\mu$.

2. For the message of Eq. (10), find and plot the power spectrum of $A(t)$ given by Eq. (10). Leave your answer in terms of $D$ and $\mu$.

3. Qualitatively, based on your answer to Question 1, what happens to the power in the carrier as the modulation index $\mu$ is varied? What happens to the power in the sidebands?

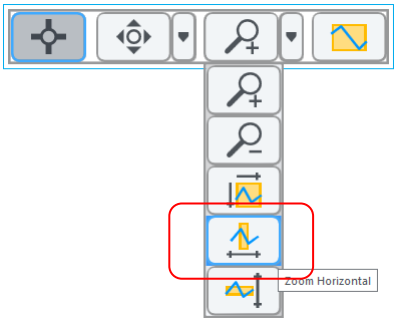
# 2.4 Lab Procedure

1. Connect a loopback cable and attenuator between the TX 1 and RX 2 connectors. Connect the USRP to your computer and plug in the power to the USRP. Run LabVIEW and open the transmitter and receiver functions that you created in the prelab.

2. Ensure that the transmitter is set up to use
   Carrier Frequency: 915.1 MHz
   IQ Rate: 200 kHz
   Gain: 0 dB
   Active Antenna: TX1
   Message Length: 200,000 samples gives a good block of data to work with.
   Modulation Index: Start with 1.0.
   Start Frequency, Delta Frequency, Number of Tones: Three tones seems to work well, but keep the highest frequency below 5 kHz.

3. Ensure that the receiver is set up to use
   Carrier Frequency: 915 MHz
   IQ Rate: 1 MHz
   Gain: Not critical. 0 dB
   Active Antenna: RX2
   Number of Samples: Same value as the transmitted message length.

   Run the transmitter, and then run the receiver. After a few seconds, stop the receiver using the STOP button, then stop the transmitter. Use the horizontal zoom feature on the graph palette to expand the "message" waveform in the transmitter and the "baseband output" waveform in the receiver.




   Both waveforms (transmit and receive) should be identical, except for scaling and possible DC offset.

4. *Power Spectrum*

   Drag the *FFT Power Spectrum for 1 Chan (CDB)* to your receiver block diagram from the ExternalFiles folder. To provide input to this power spectrum function, use the Cluster Properties function on the block diagram. Connect the "Y" input of *Cluster Properties* function to the output of your receiver's bandpass filter. Connect the "dt" input of *Cluster Properties* function to the "dt" obtained from the top left *Cluster Properties function*. Then connect the "cluster out" output of *Cluster Properties* function to the "time signal" input of *FFT Power Spectrum*. Only one other input of the power spectrum function needs to be connected: Wire a Boolean constant set to True to the "dB On" input. Now connect the "Power Spectrum/PSD" output of your power spectrum function to a waveform graph. On the waveform graph, make the "graph palette" visible so that you can use the zoom feature, and change the label on the horizontal axis to "Frequency."

   Set the transmitter to generate a message consisting of three tones starting at 1 kHz with a 1 kHz spacing. Set the modulation index to $\mu = 1$. Run the transmitter and then the receiver. Stop the receiver and then stop the transmitter. Zoom in on the power spectrum so that you can clearly see the components in the vicinity of 100 kHz. Take a screen shot of your power spectrum graph.

   Compare the power spectrum with the spectrum you predicted in Prelab Question 1. How many dB below the carrier are your sideband components?

   Change the modulation index to $\mu = 0.5$ and capture a new power spectrum. Take another screenshot of your power spectrum graph. How many dB below the carrier are your sideband components now?

5. The constant $D$ that represents the amplitude of the received carrier can be measured by passing the envelope of Eq. (5) through a lowpass filter. A measured value of $D$ is often used in practical receivers to adjust the gain of the receiver's output, providing an "automatic gain control" feature.

   Add a lowpass filter to your receiver such that the filter output is proportional to $D$. Run the transmitter and receiver, and measure the value of $D$. Increase the gain of the receiver to 20 dB and repeat the measurement of $D$. Is the change in $D$ consistent with a 20 dB change in receiver gain?

6. *Elegant receiver*

   The signal at the output of the bandpass filter has a real part that is given by Eq. (4). The imaginary part is given by

$$r_2(t) = A_r\left[1+\mu\frac{m(t)}{m_p}\right]\sin(2\pi f_{IF}t+\theta). \tag{11}$$

The complex signal at the output of the bandpass filter is therefore

$$\begin{aligned}\tilde{r}(t) &= r_1(t) + jr_2(t)\\ &= A_r\left[1+\mu\frac{m(t)}{m_p}\right]\cos(2\pi f_{IF}t+\theta) + jA_r\left[1+\mu\frac{m(t)}{m_p}\right]\sin(2\pi f_{IF}t+\theta)\\ &= A_r\left[1+\mu\frac{m(t)}{m_p}\right]\left[\cos(2\pi f_{IF}t+\theta) + j\sin(2\pi f_{IF}t+\theta)\right]\\ &= A_r\left[1+\mu\frac{m(t)}{m_p}\right]e^{j(2\pi f_{IF}t+\theta)}.\end{aligned} \tag{12}$$

The magnitude of $\tilde{r}(t)$ is $A_r\left[1+\mu\frac{m(t)}{m_p}\right]$, which is the desired demodulated output.

Connect the bandpass filter output directly to the absolute value block (bypass the *Complex to Re/Im* function). Connect the absolute value output directly to the Baseband Output graph (bypass the *Butterworth Filter*). Run the transmitter and receiver, and observe that the demodulated output is the same as it was in step 3. There is no need for the lowpass filter! (Note that since the modulation index $\mu$ is upper-bounded by one, the expression $A_r\left[1+\mu\frac{m(t)}{m_p}\right]$ is never negative, and is not affected by the absolute value).

# 2.5 Report

## Prelab

Hand in documentation for the functions you created for the transmitter and receiver. Also include documentation for any sub-functions you may have created. To obtain documentation, print out legible screenshots of the front panel and block diagram.

Answer the questions in the *Questions* section at the end of the prelab instructions.

# Lab

Submit the functions you created for the transmitter and receiver. Also submit any sub-functions you may have created. Be sure your files adhere to the naming convention described in the instructions above.
Resubmit documentation for any functions you modified during the lab.

Submit the spectrum graphs and answer all of the questions in Sections 4 and 5 of the Lab Procedure.

LAB 3

# Frequency-Division Multiplexing

Prerequisite: Lab 2 – Amplitude Modulation

## 3.1 Objective

In this laboratory exercise you will investigate sending multiple messages on a single carrier by frequency-division multiplexing. Each individual message is modulated onto a separate subcarrier and the modulated subcarriers are summed before being sent to the USRP. At the receiver, the individual messages are separated by filtering and then demodulated. The purpose of this exercise is to

- provide additional practice in programming the USRP,
- introduce the concept of frequency-division multiplexing, and
- explore the concept of intermediate-frequency filtering in the receiver.

## 3.2 Background

Frequency-division multiplexing is widely used in telemetry, in the satellite relaying of television signals, and, until the widespread adoption of fiber optics, was the standard transmission method for long-distance telephone signals. Frequency-division multiplexing also plays an important role in the OFDM technique used in DSL and in third-generation cellular telephone systems.

Suppose $m_1(t)$ and $m_2(t)$ are message signals. Let $f_1$ and $f_2$ be corresponding subcarrier frequencies. We can form the modulated subcarrier signals

$$\begin{aligned} g_1(t) &= A_1\left[1+\mu_1\frac{m_1(t)}{m_{1p}}\right]\cos(2\pi f_1 t), \text{ and} \\ g_2(t) &= A_2\left[1+\mu_2\frac{m_2(t)}{m_{2p}}\right]\cos(2\pi f_2 t). \end{aligned} \tag{1}$$

It is not necessary to use amplitude modulation to modulate the subcarriers. We are using AM in this lab exercise because it is familiar from Lab 2, and because it is easy to demodulate. Note that the subcarrier frequencies $f_1$ and $f_2$ must be spaced sufficiently apart in frequency so that the spectra of $g_1(t)$ and $g_2(t)$ do not overlap. The signals $g_1(t)$ and $g_2(t)$ are combined to give

$$g_I(t) = g_1(t) + g_2(t), \tag{2}$$

the in-phase component of the baseband signal. As in Lab 2 we will let the quadrature component $g_Q(t)$ equal zero, so that the signal sent to the USRP is

$$\tilde{g}(t) = g_I(t) + jg_Q(t) = g_I(t) = g_1(t) + g_2(t). \tag{3}$$

The signal actually transmitted by the USRP is

$$\begin{aligned} g(t) &= Ag_I(t)\cos(2\pi f_c t) - Ag_Q(t)\sin(2\pi f_c t) \\ &= A\left[g_1(t) + g_2(t)\right]\cos(2\pi f_c t), \end{aligned} \tag{4}$$

where $A$ is set by the "gain" parameter and $f_c$ is the USRP carrier frequency.

On the receiving side, the USRP receiver provides the output

$$\tilde{r}(t) = D\left[g_1(t) + g_2(t)\right]e^{j\theta}, \tag{5}$$

where $\theta$ is the phase difference between the transmitter and receiver oscillator signals and $D$ is a constant, usually much smaller than $A$. This complex-valued signal can be sent to a bank of two bandpass filters centered on the subcarrier frequencies $f_1$ and $f_2$. The filter outputs are the individual signals given in Eq. (1). These can now be demodulated using envelope detectors as in Lab 2.

## 3.3 Pre-Lab

### Transmitter

Create a sub-vi *AMonSubcarrier.gvi* to create the signals $g_1(t)$ or $g_2(t)$ as shown in Eq. (1). The required inputs and outputs are given in

Table 1.

**Table 1: AMonSubcarrier**

| **Inputs** | | | |
|---|---|---|---|
| | Message waveform in | $m_i(t)$ | waveform (double) |
| | Modulation index | $\mu_i$ | double |
| | Carrier level | $A_i$ | double |
| | Sampling information for subcarrier | | cluster |
| | Subcarrier frequency | $f_i$ | double |
| **Output** | | | |
| | Modulated signal out | $g_i(t)$ | array (double) |

Give your sub-vi a distinctive icon.

1. A template for the transmitter has been provided in the file *Lab3TxTemplate.gvi*. This template contains the four functions for interfacing with the USRP along with two message generators that will generate waveforms $m_1(t)$ and $m_2(t)$.
    a. Use two instances of your AMonSubcarrier sub-vi to create the signals $g_1(t)$ and $g_2(t)$.
    b. Add $g_1(t)$ and $g_2(t)$ together and normalize the peak value of the sum to one.
    c. Create another array of the same length as $g_1(t)$ or $g_2(t)$, but containing zeros using the *Re/Im to Complex* function. Form a complex array $\tilde{g}(t)$ as given in Eq. (3) above.
    d. Pass $\tilde{g}(t)$ into the *while* loop, and connect it to the Write Tx Data function.

2. Save your transmitter in a file whose name includes the letters "MuxTx" and your initials (e.g., *MuxTx_BAB.gvi*).

## Receiver

1. A template for the receiver has been provided in the file *Lab3RxTemplate.gvi*. This template contains the six functions for interfacing with the USRP along with two waveform graphs on which to display your demodulated output signals.
2. 
    a. Feed the array produced by *Fetch Rx Data* into two bandpass filters. Use Chebyshev (CDB) filters (ExternalFiles folder). Set one of the filters to have a high cutoff frequency of 505 kHz and a low cutoff frequency of 495 kHz. Set the other filter to have a high cutoff frequency of 515 kHz and a low cutoff frequency of 505 kHz. For both filters the default passband ripple of 0.1 dB is acceptable. The sampling frequency input to both filters should be the "actual IQ rate" obtained from *Configure Signal* (or reciprocal of dt from cluster properties).
    b. Pass the output of each bandpass filter through an envelope detector. Each envelope detector consists of an absolute value function followed by a lowpass filter. Set up the lowpass filters as you did in Lab 2. The lowpass filter outputs should be connected to the Build Waveform blocks that connect to the Message 1 Out and Message 2 Out graphs.

3. Save your receiver in a file whose name includes the letters "MuxRx" and your initials (e.g. *MuxRx_BAB.gvi*).

*Question*

Starting with Eq. (5), show analytically that the phase error $\theta$ does not affect either demodulated output signal. Note that in this lab project we do not take the real part of $\tilde{r}(t)$ prior to bandpass filtering.

# 3.4 Lab Procedure

1. Connect a loopback cable and attenuator between the TX 1 and RX 2 connectors. Connect the USRP to your computer and plug in the power to the USRP. Run LabVIEW and open the transmitter and receiver functions that you created in the prelab.

2. Ensure that the transmitter is set up to use:

   Carrier Frequency: 915 MHz

   IQ Rate: 2 MS/s.

   Gain: Not critical. 0 dB

   Active Antenna: TX1

   Message Length: 200,000 samples gives a good block of data to work with.

   Modulation Indices: Start with 1.0 for each subcarrier.

   Subcarrier frequencies: Use 500 kHz for $f_1$ and 510 kHz for $f_2$.

   Start Frequency, Delta Frequency, Number of Tones: Not critical, but keep the highest frequencies below 5 kHz. Three tones for each message seems to work well. The results are most dramatic if you use "low" frequencies for one message and "high" frequencies for the other, so that the messages look different when plotted.

3. Ensure that the receiver is set up to use:

   Carrier Frequency: 915 MHz

   IQ Rate: 2 MS/s

   Gain: Not critical. 0 dB

   Active Antenna: RX2

   Number of Samples: Same value as the transmitted message length.

   Note that there is no offset between the transmitter's carrier frequency and the receiver's carrier frequency in this lab project.

6. Run the transmitter, then run the receiver. After a few seconds, stop the receiver using the STOP button, then stop the transmitter. Examine the Message Out graphs to ensure that the receiver is correctly demodulating and displaying the two message signals.

7. *Power Spectrum*

   Add the *FFT Power Spectrum for 1 Chan (CDB)* from the ExternalFiles folder to your receiver. Obtain the "time signal" input from the waveform produced by *Fetch Rx Data*. Attach a Boolean constant set to True to the "dB On" input. Attach a waveform graph to the "Power Spectrum/PSD" output. Change the label on the horizontal axis of the waveform graph to "Frequency," and set the horizontal scale to display frequency components in the range 495 kHz to 515 kHz.

Run the transmitter and then run the receiver. Stop the receiver and then stop the transmitter. Take a screenshot of the spectrum for your report. Identify the two subcarriers on your print out.

8. *Crosstalk*

Crosstalk is the phenomena in which some of the signal in one channel "bleeds over" into an adjacent channel. As a preliminary step to observing crosstalk, we will measure the average power in the two demodulated output signals. The *AC & DC Estimator*, located in the ExternalFiles folder, can measure the RMS value of the AC component of a signal. If the *AC & DC Estimator* "AC estimate" output is squared, the result will be an estimate of the average power in a signal, exclusive of any DC offset. Attach an *AC & DC Estimator*, a squarer, and a numerical indicator to each of the lowpass filter outputs in your receiver.

Set up the transmitter so that the two message signals are identical. Set both modulation indices to one. Set the sampling rate ("IQ rate") at the receiver to 10 MSa/s. Run the transmitter and the receiver, and record the power in each demodulated signal. Now you are ready to measure crosstalk.

**Measurement A**: Disconnect one of your *AMonSubcarrier.gvi* blocks so that only message 1 is transmitted. Run the transmitter and the receiver and record the power in each demodulated signal. Label these measurements $P_{1A}$ and $P_{2A}$.

**Measurement B:** Now configure your transmitter so that only message 2 is transmitted. Run the transmitter and the receiver and record the power in each demodulated signal. Label these measurements $P_{1B}$ and $P_{2B}$. Compute the two crosstalk measurements $XT_{12} = 10\log_{10}\frac{P_{2A}}{P_{2B}}$ dB and $XT_{21} = 10\log_{10}\frac{P_{1B}}{P_{1A}}$ dB.

The amount of crosstalk present in a frequency-division-multiplexed signal depends to a great extent on the selectivity of the filters used to separate the channels in the receiver. To illustrate, change the order of both of the Chebyshev bandpass filters in the receiver from 5 to 2 and repeat the crosstalk measurements. Compare the values of $XT_{12}$ and $XT_{21}$ for the order-5 and order-2 cases.

# 3.5 Report

## Prelab

Hand in documentation for *AMonSubcarrier.gvi* and for the functions you created for the transmitter and receiver. Also include documentation for any additional sub-functions you may have created. To obtain documentation, print out legible screenshots of the front panel and block diagram.

Answer the question in the *Question* section at the end of the prelab instructions.

## Lab

Submit *AMonSubcarrier.gvi* and the functions you created for the transmitter and receiver. Also submit any additional sub-functions you may have created. Be sure your files adhere to the naming convention described in the instructions above.

Resubmit documentation for any functions you modified during the lab.

Submit the spectrum graph required in Step 4 and all of the crosstalk information required in Step 5 of the Lab Procedure.

LAB 4

# Image Rejection

**Prerequisite:** Lab 2 – Amplitude Modulation

# 4.1 Objective

This laboratory exercise illustrates the image problem in superheterodyne receivers. Image rejection is carried out using complex filtering. This lab introduces a processing technique that is straightforward in a software defined radio, but is virtually unavailable in a conventional hardware-based radio.

# 4.2 Background

## Frequency Conversion

As we saw in Lab 2, most communication receivers convert a received signal at carrier frequency $f_c$ to a signal at "intermediate" frequency $f_{IF}$ for amplification and filtering prior to demodulation. In the USRP, the frequency conversion can be carried out by offsetting the receiver's carrier frequency from the carrier frequency of the transmitted signal. To avoid confusion, the receiver's carrier oscillator is usually referred to as a "local" oscillator, and its frequency as the "local oscillator frequency" $f_{LO}$. In Lab 2 we set the transmitter's carrier frequency to $f_c = 915.1\ \text{MHz}$ and the receiver's local oscillator frequency to $f_{LO} = 915\ \text{MHz}$. These settings provided an intermediate frequency $f_{IF} = f_c - f_{LO} = 100\ \text{kHz}$.

In the USRP receiver, frequency conversion is carried out in hardware by multiplying the received signal by $\cos(2\pi f_{LO} t)$ and by $-\sin(2\pi f_{LO} t)$. For example, suppose the received signal is the AM waveform

$$r(t) = A_r\left[1 + \mu\frac{m(t)}{m_p}\right]\cos(2\pi f_c t + \theta). \qquad (1)$$

The receiver forms

$$
\begin{aligned}
r(t)\cos(2\pi f_{LO}t) &= A_r\left[1+\mu\frac{m(t)}{m_p}\right]\cos(2\pi f_c t+\theta)\cos(2\pi f_{LO}t) \\
&= \frac{1}{2}A_r\left[1+\mu\frac{m(t)}{m_p}\right]\cos\left[2\pi(f_c-f_{LO})t+\theta\right] \\
&\quad+\frac{1}{2}A_r\left[1+\mu\frac{m(t)}{m_p}\right]\cos\left[2\pi(f_c+f_{LO})t+\theta\right].
\end{aligned}
\tag{2}
$$

Of the two terms in Eq. (2), the first is at the intermediate frequency $f_{IF}=f_c-f_{LO}$, while the second is at a much higher frequency $f_c+f_{LO}$. The higher-frequency term is removed by filtering in the USRP, providing the "in-phase" signal

$$
r_I(t)=A_r\left[1+\mu\frac{m(t)}{m_p}\right]\cos(2\pi f_{IF}t+\theta). \tag{3}
$$

The receiver also forms a second signal,

$$
\begin{aligned}
-r(t)\sin(2\pi f_{LO}t) &= -A_r\left[1+\mu\frac{m(t)}{m_p}\right]\cos(2\pi f_c t+\theta)\sin(2\pi f_{LO}t) \\
&= \frac{1}{2}A_r\left[1+\mu\frac{m(t)}{m_p}\right]\sin\left[2\pi(f_c-f_{LO})t+\theta\right] \\
&\quad-\frac{1}{2}A_r\left[1+\mu\frac{m(t)}{m_p}\right]\sin\left[2\pi(f_c+f_{LO})t+\theta\right].
\end{aligned}
\tag{4}
$$

Again, the high-frequency term is removed, and the receiver provides the "quadrature" signal

$$
r_Q(t)=A_r\left[1+\mu\frac{m(t)}{m_p}\right]\sin(2\pi f_{IF}t+\theta). \tag{5}
$$

A conventional hardware-based receiver normally works with the in-phase signal given by Eq. (3). The USRP combines the in-phase and quadrature signals to form the complex IF signal $\tilde{r}(t)$ given by

$$
\begin{aligned}
\tilde{r}(t) &= r_I(t)+jr_Q(t) \\
&= A_r\left[1+\mu\frac{m(t)}{m_p}\right]\left[\cos(2\pi f_{IF}t+\theta)+j\sin(2\pi f_{IF}t+\theta)\right] \\
&= A_r\left[1+\mu\frac{m(t)}{m_p}\right]e^{j(2\pi f_{IF}t+\theta)}.
\end{aligned}
\tag{6}
$$

This complex signal is what is actually provided to the user by *Fetch Rx Data*.

In the frequency domain, the spectrum $R(f)$ of the received signal $r(t)$ given in Eq. (1) is shown in Figure 1.

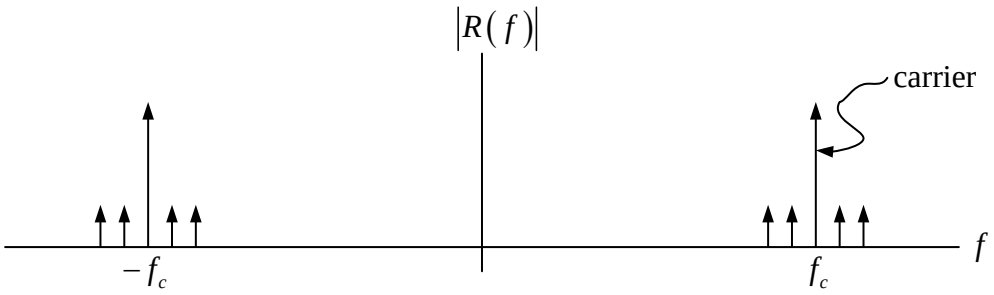



**Figure 1. Spectrum of Received AM Signal**

Since $r(t)$ is a real-valued signal, its spectrum contains both positive and negative-frequency components. After frequency conversion, the complex IF signal $\tilde{r}(t)$ has the spectrum $\tilde{R}(f)$ shown in Figure 2.

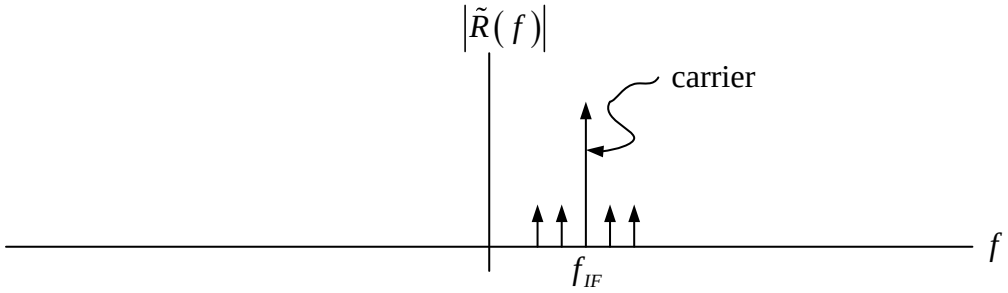



**Figure 2. Spectrum of Complex IF Signal**

Notice that $\tilde{r}(t)$ contains only positive frequency components.

In Lab 2 we passed the complex IF signal through a bandpass filter. Figure 3 shows the frequency response of the bandpass filter. It is intended that signals at carrier frequencies other than $f_{IF}$ will be rejected by the filter.

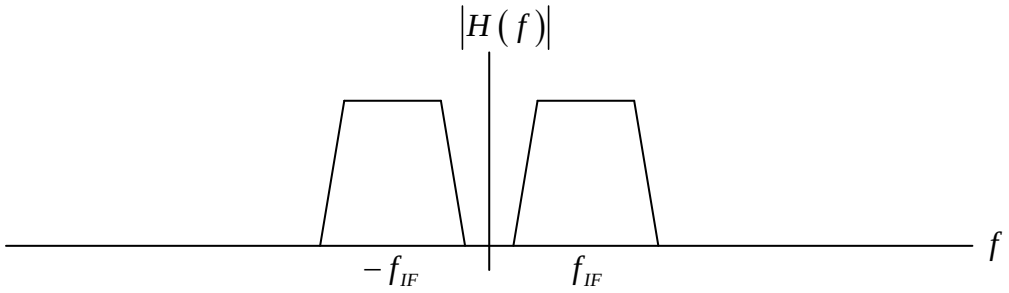



**Figure 3. Frequency Response of Intermediate-Frequency Filter**

## Image Signal

Suppose that there is a second signal received along with the signal of Eq. (1), and that this signal is given by

$$r_2(t) = A_{r2}\left[1+\mu_2\frac{m_2(t)}{m_{2p}}\right]\cos\left(2\pi f_{IM}t+\theta_2\right), \tag{7}$$

where the carrier frequency $f_{IM}$ happens to be given by

$$f_{IM} = f_{LO} - f_{IF}. \tag{8}$$

If we carry out the analysis of Eqs. (7) and (8), we find that this second signal produces the complex IF signal $\tilde{r}_2(t)$ given by

$$\tilde{r}(t) = A_{r2}\left[1+\mu_2\frac{m_2(t)}{m_{2p}}\right]e^{j(-2\pi f_{IF}t+\theta_2)}. \tag{9}$$

The spectrum $\tilde{R}_2(f)$ of this signal is shown in Figure 4.

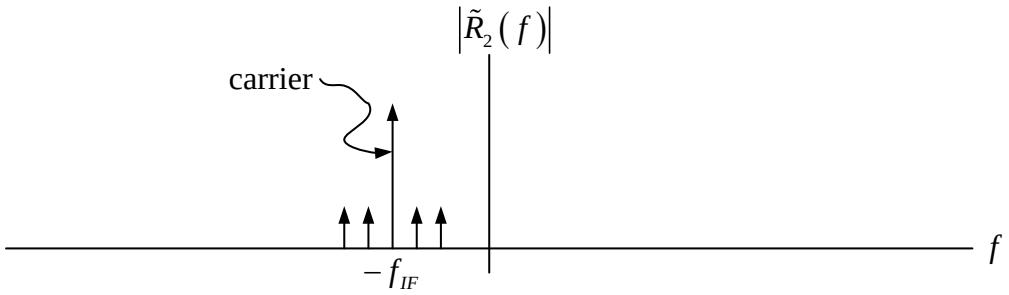



**Figure 4. Spectrum of the Image Signal**

The signals $r(t)$ and $r_2(t)$ are said to be "images" of one another. A glance at the frequency response shown in Fig. 3 shows that both $r(t)$ and $r_2(t)$ will pass through the IF filter, and the two signals will interfere with each other in the demodulator that follows the IF filter. The relationship between the frequencies of the two image signals is worth noting. One signal, $r(t)$, is at a carrier frequency of $f_c = f_{LO} + f_{IF}$, while the other signal, $r_2(t)$, is at a carrier frequency of $f_{IM} = f_{LO} - f_{IF}$. These carrier frequencies are symmetrically arranged about the receiver's frequency $f_{LO}$, the way a physical object and its image are symmetrically distant from the surface of a mirror.

## Image Rejection

In a conventional receiver, the unwanted image is normally removed before the frequency conversion step. This requires placing an analog filter in the receiver front end. Clearly, we will not have the option of adding analog hardware to the USRP, so we will take advantage of some clever signal processing to remove the unwanted image signal after frequency conversion. Notice in Figure 2 that the desired signal shows up after frequency conversion at the intermediate frequency $f_{IF}$, while in Figure 4 we see that the unwanted image signal shows up at the intermediate frequency $-f_{IF}$. A conventional bandpass filter passes both of these signals, but if we can construct a filter with a positive-frequency-only passband, we will be able to pass the desired signal while rejecting the unwanted image. Figure 5 shows the filter frequency response we have in mind.

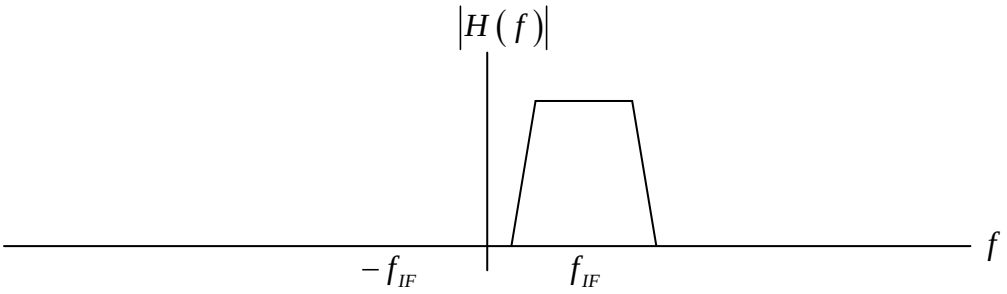



**Figure 5. Frequency Response of a Complex Image-Rejection Filter**

A frequency response such as that shown in Figure 5 lacks symmetry about the vertical axis, which implies that the impulse response of the filter is complex-valued. A filter with a complex-valued impulse response is difficult to produce in hardware, but, as we shall see, is easily produced in software.

On a practical note, an analog image rejection filter must be tunable if the receiver is to be capable of receiving signals at a range of carrier frequencies $f_c$. To make the filter tunable, it usually must be kept simple, which constrains the order of the filter to be low. Second-order filters are common in this application. A low-order filter cannot do a very thorough job of rejecting signals at the unwanted image frequency. In contrast, the complex image rejection filter of Figure 5 is centered at a fixed intermediate frequency, and does not have to be tunable. The quality of image rejection is limited only by the ability of the filter to reject signals at negative frequencies.

# 4.3 Pre-Lab

A complex filter has been provided in ChebyshevHilbert.gvi. Your task is to create a program to find the frequency response of this filter. There are several ways in which this can be done, and the method is left up to you.

1. Set the filter up with
    - a sampling frequency of 1MS/s
    - a high cutoff frequency of 105 kHz
    - a low cutoff frequency of 95 kHz
    - order 5
    - you can accept the default ripple value of 0.1 dB.

2. Plot the magnitude of the frequency response in decibels over the frequency range $-200\ \text{kHz}$ to $200\ \text{kHz}$. (The frequency axis must be a linear scale; this is not a Bode plot.)

Note:

The input to the filter is an array of complex numbers, as is the output. If you choose to measure the frequency response by inputting sinusoidal signals at a range of frequencies, you must use complex sinusoids $e^{j2\pi ft}$ rather than conventional sinusoids $\cos(2\pi ft)$ so that negative frequencies can be distinguished from positive frequencies.

# 4.4 Lab Procedure

1. Connect a loopback cable and attenuator between the TX 1 and RX 2 connectors. Connect the USRP to your computer and plug in the power to the USRP. .

2. Open the AM transmitter and receiver functions that you created for Lab 2.

   Ensure that the transmitter is set up to use

   Carrier Frequency: 915.1 MHz

   IQ Rate: Not critical. 200 kHz

   Gain: Not critical. 0 dB

   Active Antenna: TX1

   Message Length: 200,000 samples gives a good block of data to work with.

   Modulation Index: Start with 1.0.

   Start Frequency, Delta Frequency, Number of Tones: Not critical, but keep the highest frequency below 5 kHz. Three tones seems to work well.

   Ensure that the receiver is set up to use

   Carrier Frequency: 915 MHz

   IQ Rate: 1 MHz

   Gain: Not critical. 0 dB

   Active Antenna: RX2

   Number of Samples: Same value as the transmitted message length.

   Run the transmitter and receiver and verify that the demodulated message appears at the receiver output.

3. Modify the receiver functions by adding components to indicate the strength of the received carrier. To do this, recall that the output of an AM demodulator is

$$r_o(t) = D\left[1 + \mu \frac{m(t)}{m_p}\right]. \tag{10}$$

   Since the message $m(t)$ has an average value of zero, the average of $r_o(t)$ will be the received carrier $D$. You can average $r_o(t)$ by using a lowpass filter whose cutoff frequency is

below the lowest frequency component of $m(t)$, or by finding a suitable signal-averaging function.

Run the transmitter and receiver and record the value of the received carrier $D$.

4. *Power Spectrum*

   Add the *FFT Power Spectrum for 1 Chan (CDB)* from the ExternalFiles folder to your receiver. Obtain the "time signal" input from the waveform produced by *Fetch Rx Data*. Attach a Boolean constant set to True to the "dB On" input. Attach a waveform graph to the "Power Spectrum/PSD" output. Change the label on the horizontal axis of the waveform graph to "Frequency."

   Run the transmitter and the receiver. Take a screenshot of the spectrum of the received signal. The spectrum should correspond to the one shown in Figure 2.

5. Use Eq. (8) to determine the image frequency $f_{IM}$. Set the frequency of the transmitter to $f_{IM}$. Run the transmitter and receiver and record the value of the received carrier $D_2$. Compute the *image rejection ratio* (IRR) given by

$$\text{IRR} = 20\log_{10}\frac{D}{D_2}. \tag{11}$$

   (You should get a result near zero dB.)

   Take another screenshot of the spectrum of the received image signal. This time, the spectrum should correspond to the one shown in Figure 4.

6. Replace the bandpass filter in your receiver with ChebyshevHilbert.gvi. Set the transmitter's carrier frequency to $f_c = 915.1\ \text{MHz}$, run the transmitter and receiver, and measure $D$. Now set the transmitter's carrier frequency to $f_{IM}$, run the transmitter and receiver, and measure $D_2$. Calculate the IRR and compare with the result you obtained in Step 5.

7. Save your modified receiver in a file whose name includes the letters "AMImageRx" and your initials (e.g. *AMImageRx_BAB.gvi*).

8. *Challenge Question* (Familiarity with the Hilbert transform is assumed.)

   Suppose in Step 6 you want to receive the signal at carrier frequency $f_{IM}$ rather than the one at carrier frequency $f_c$. Modify *ChebyshevHilbert.gvi* to make this happen. (Hint: A single sign change in an appropriate place in the MathScript block is all that is required.)

# 4.5 Report

## Prelab

Hand in documentation for the program you created to measure the frequency response of *ChebyshevHilbert.gvi*. Hand in documentation for the modified receiver. Also include documentation for any additional sub-functions you may have created. To obtain documentation, print out legible screenshots of the front panel and block diagram.

Submit the frequency response plot from Step 2 of the Prelab.

## Lab

Submit the functions you created to measure the frequency response of *ChebyshevHilbert.gvi*. Submit the functions for the modified receiver. Also submit any additional sub-functions you may have created. Be sure your files adhere to the naming convention described in the instructions above.

Resubmit documentation for any functions you modified during the lab.

Submit the spectrum graphs required in Steps 4 and 5.

Submit your image rejection ratio computations and results from Steps 5 and 6.

If you completed the challenge question (Step 8), show how you modified *ChebyshevHilbert.gvi* and describe how you verified the result.

LAB 5

# Double-Sideband Suppressed-Carrier

**Prerequisite:** Lab 2 – Amplitude Modulation

# 5.1 Objective

This laboratory exercise introduces suppressed-carrier modulation. A simple scheme for phase and frequency synchronization is introduced in implementing the demodulator.

# 5.2 Background

## Double-Sideband Suppressed-Carrier

Amplitude modulation is inherently inefficient because the largest part of the transmitted power is contained in the carrier. In suppressed-carrier schemes the carrier is simply not transmitted. There are two common suppressed-carrier techniques in use, double-sideband suppressed-carrier (DSB-SC) and single-sideband (SSB). Double-sideband suppressed-carrier modulation is identical to AM, except that the carrier is omitted.

If $m(t)$ is a baseband "message" signal and $\cos(2\pi f_c t)$ is a "carrier" signal at carrier frequency $f_c$, then we can write the DSB-SC signal $g(t)$ as

$$g(t) = Am(t)\cos(2\pi f_c t). \tag{1}$$

For the special case in which $m(t) = m_p \cos(2\pi f_m t)$, we can write

$$\begin{aligned} g(t) &= Am_p \cos(2\pi f_m t)\cos(2\pi f_c t) \\ &= \frac{Am_p}{2}\cos\left[2\pi(f_c - f_m)t\right] + \frac{Am_p}{2}\cos\left[2\pi(f_c + f_m)t\right]. \end{aligned} \tag{2}$$

The two terms in Eq. (2) represent the lower and upper sidebands, respectively. There is no carrier term at frequency $f_c$. Figure 1 is a plot of a 20 kHz carrier modulated by a 1 kHz sinusoid using DSB-SC modulation.

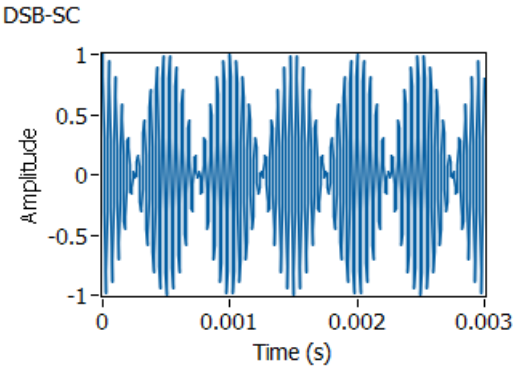



**Figure 1. Double-Sideband Suppressed-Carrier Modulation**

When the DSB-SC signal arrives at the receiver, it has the form

$$r(t) = Dm(t)\cos(2\pi f_c t + \theta), \tag{3}$$

where $D$ is a constant, usually much smaller than $A$, and the angle $\theta$ represents the difference in phase between the transmitter and receiver carrier oscillators. If the receiver's carrier oscillator (the "local" oscillator) is set to the same frequency as the transmitter's carrier oscillator, the USRP will generate the two demodulated signals

$$\begin{aligned} r_I(t) &= \frac{D}{2}m(t)\cos(\theta), \text{ and} \\ r_Q(t) &= \frac{D}{2}m(t)\sin(\theta). \end{aligned} \tag{4}$$

The *Fetch Rx Data* provides these demodulated signals as a single complex-valued signal $\tilde{r}(t)$ given by

$$\begin{aligned} \tilde{r}(t) &= \frac{D}{2}m(t)\cos(\theta) + j\frac{D}{2}m(t)\sin(\theta) \\ &= \frac{D}{2}m(t)e^{j\theta}. \end{aligned} \tag{5}$$

It is tempting to suppose that the message $m(t)$ can be extracted from $\tilde{r}(t)$ by taking the magnitude of the complex signal. Unfortunately, the magnitude of $\tilde{r}(t)$ is

$$|\tilde{r}(t)| = \frac{D_r}{2}|m(t)|, \tag{6}$$

where the absolute value represents unwanted distortion of the message signal. It is more productive to use the in-phase (real part) signal $r_I(t)$ given in Eq. (4). The $\cos(\theta)$ factor of $r_I(t)$

represents a gain constant. Unfortunately, the value of this gain constant is not under user control, and might be small if $\theta$ turns out to have a value near $\pm\pi/2$. Moreover, if the receiver's oscillator and transmitter's oscillator differ slightly in frequency, then the phase error $\theta$ will change with time, causing $r_I(t)$ to fade in and out. The next section discusses how we can compensate for the $\cos(\theta)$ term.

## Phase Synchronization

There are a number of techniques that can be used to eliminate the $\cos(\theta)$ phase-error term. The method we present here is simple and easy to implement in LabVIEW. The basic steps are

- Estimate $\theta$
- Multiply $\tilde{r}(t)$ by $e^{-j\theta}$ to produce $\tilde{r}(t)e^{-j\theta} = \frac{D}{2}m(t)e^{j\theta}e^{-j\theta} = \frac{D}{2}m(t)e^{j0}$
- Take the real part: $\mathrm{Re}\left\{\frac{D}{2}m(t)e^{j0}\right\} = \frac{D}{2}m(t)\cos(0) = \frac{D}{2}m(t)$.

Estimating $\theta$ requires several steps. Note first that the phase angle of $\tilde{r}(t)$ will jump by $\pm\pi$ whenever $m(t)$ changes sign. To eliminate these phase jumps, start by squaring $\tilde{r}(t)$:

$$\tilde{r}^2(t) = \frac{D^2}{4}m^2(t)e^{j2\theta}. \tag{7}$$

Since the squared message $m^2(t)$ never changes sign, the phase jumps are eliminated. The angle $2\theta$ can be extracted using a Complex to Polar function from the Data Types→Numeric→Complex palette. It turns out to be helpful at this point to smooth variations in $2\theta$ caused by noise. The *median filter* from the ExternalFiles folder does a good job. The default values can be accepted for the "left rank" and "right rank" parameters. Next the Unwrap Phase function from the Analysis→Signal Processing→Signal Operations palette will remove jumps of $\pm 2\pi$. Finally, dividing by two gives the desired estimate of the phase error $\theta$. The block diagram in Figure 2 shows the entire phase synchronization process.

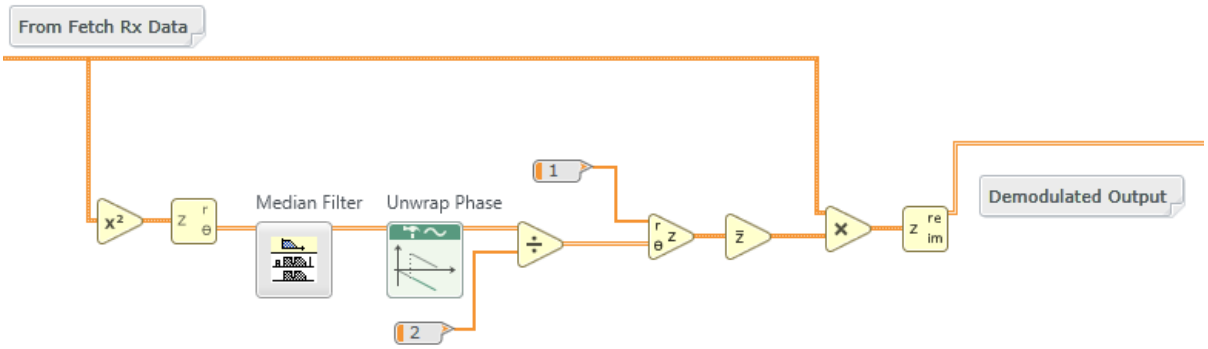



**Figure 2. Phase Synchronization**

# 5.3 Pre-Lab

## Transmitter

1. A template for the transmitter has been provided in the file Lab5TxTemplate.gvi. This template contains the four interface functions along with a "message generator" that is set to produce a message signal consisting of three tones. The three tones are initially set to 1, 2, and 3 kHz, but these frequencies can be changed using front-panel controls. Your task is to add blocks as needed to produce a DSB-SC signal, and then to pass the DSB-SC signal into the *while* loop to the Write Tx Data block.

   Hint: The DSB-SC signal you generate will be $g_I(nT)$. For $g_Q(nT)$ set up an array the same length as $g_I(nT)$ containing all zeroes. Then combine the two into a single complex array $\tilde{g}(nT) = g_I(nT) + jg_Q(nT)$.

   Notes:

   a. The message generator creates a signal that is the sum of a set of sinusoids of equal amplitude. You can choose the number of sinusoids to include in the set, you can choose their frequencies, and you can choose their common amplitude. In this template the message generator has been provided with a "seed." This causes the initial phase angles of the sinusoids to be the same every time you run the program. As a result, the same message will be generated every time, which is useful to aid debugging. To restore random behavior, set the seed to $-1$.

b. There is a practical constraint imposed by the D/A converters in the USRP: Scale the signals you generate so that the peak value of $\left|\tilde{g}\left(nT\right)\right|$ does not exceed +/- 1. (Check out the *Quick Scale 1Dfunction* in the ExternalFiles folder.)

c. Save your transmitter in a file whose name includes the letters "DSBSCTx" and your initials (e.g., *DSBSCTx_BAB.gvi*).

## Receiver

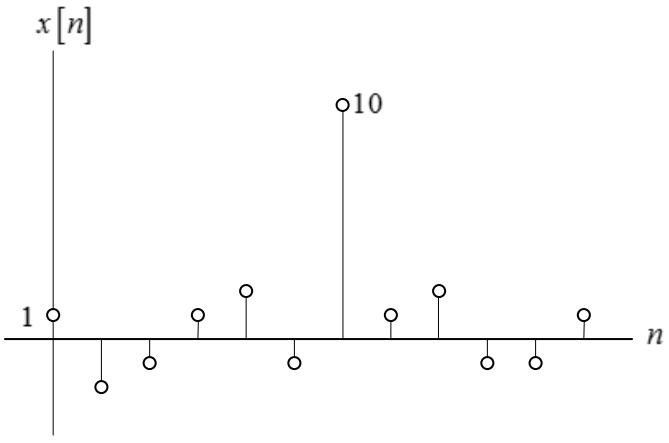
2. A template for the receiver has been provided in the file Lab5RxTemplate.gvi. This template contains the six interface functions along with a waveform graph on which to display your demodulated output signal.

   Complete the program to demodulate the complex array returned by the Fetch Rx Data function and display the result. Include the phase synchronization of Figure 2. Also, to help in debugging, include a graph to display the phase error $\theta$ vs. time.

   Save your receiver in a file whose name includes the letters "DSBSCRx" and your initials.

## Questions

1. The graph below shows a sequence of samples having values $x=\left[1 \quad -2 \quad -1 \quad 1 \quad 2 \quad -1 \quad 10 \quad 1 \quad 2 \quad -1 \quad -1 \quad 1\right]$ Assume that values of $x\left[n\right]$ at index values not shown are zero. Note that there is a single outlying sample at index $n=6$.



Suppose we have a simple *median filter* than produces an output $y[n]$ given by

$$y[n] = \text{median}\{x[n-2],\ldots,x[n+2]\}, \quad \text{n} = 0,\ldots,11.$$

Find $y[n]$ for the sample sequence $x[n]$ shown.

2. Suppose the receiver's carrier oscillator differs in frequency from the transmitter's oscillator by a small offset $\Delta f$. Modify Eqs. (4) and (5) to include the frequency offset.

# 5.4 Lab Procedure

1. Connect a loopback cable and attenuator between the TX 1 and RX 2 connectors. Connect the USRP to your computer and plug in the power to the USRP. Run LabVIEW and open the transmitter and receiver that you created in the prelab.

2. Ensure that the transmitter is set up to use

   Carrier Frequency: 915 MHz

   IQ Rate: Not critical. 200 kHz

   Gain: Not critical. 0 dB

   Active Antenna: TX1

   Message Length: 200,000 samples gives a good block of data to work with.

   Start Frequency, Delta Frequency, Number of Tones: Not critical, but keep the highest frequency below 5 kHz. Three tones seems to work well.

3. Ensure that the receiver is set up to use

   Carrier Frequency: 915 MHz

   IQ Rate: 200 kHz

   Gain: Not critical. 0 dB

   Active Antenna: RX2

   Number of Samples: Same value as the transmitted message length.

Run the transmitter, then run the receiver. After a few seconds, stop the receiver using the STOP button, then stop the transmitter (using the STOP button). Use the horizontal zoom feature on the graph palette to expand the "message" waveform in the transmitter and the "demodulated output" waveform in the receiver. Both waveforms should be identical, except for scaling.[7]

4. Modify your receiver to compute $r_I(t)$ without phase synchronization and $r_I(t)$ with phase synchronization. Plot both outputs on the same graph. Run the receiver several times and observe the outputs. Can you see the effect of the $\cos(\theta)$ term on the unsynchronized output?

5. Try using your AM receiver from Lab 2 to demodulate the DSB-SC signal. Note that you will need to offset the transmitter frequency to 915.1 MHz. Run the transmitter and receiver. Take a screenshot of both the transmitted message and the demodulated output. Be sure to expand the time base so that the waveforms can be clearly seen. Was the envelope detector in the AM receiver able to correctly demodulate the DSB-SC signal?

6. The phase synchronizer can also correct for modest frequency offsets. Use the DSB-SC transmitter and receiver, and offset the frequency of the transmitter by 10 Hz. Run the transmitter and receiver. Take a screenshot of the transmitted message, the unsynchronized demodulated output, and the synchronized demodulated output. Be sure to expand the time base so that the waveforms can be clearly seen. Verify that the synchronized demodulated output is correct, except possibly for being inverted.

   Repeat for frequency offsets of 100 Hz and 1 kHz. Can your phase synchronizer handle the 1 kHz case?

[7] The demodulated output may be inverted. This is a consequence of squaring the signal in the phase synchronization process. An error of $\pm 2\pi$ in the angle $2\theta$ is no error at all, but when the angle is divided by two, the error becomes $\pm\pi$.

# 5.5 Report

## Prelab

Hand in documentation for your transmitter and receiver. Also include documentation for any additional functions you may have created. To obtain documentation, print out legible screenshots of the front panel and block diagram.

Submit your answers to the Questions at the end of the Prelab section.

## Lab

Submit the program you created to implement the DSB-SC transmitter and receiver. Also submit any additional functions you may have created. Be sure your files adhere to the naming convention described in the instructions above. Resubmit documentation for any functions you modified during the lab.

Describe the effect on the demodulated signal of the $\cos(\theta)$ term, as described in Step 4.

Submit the graphs required in Step 5. Discuss whether DSB-SC can be properly demodulated using an envelope detector.

Submit the graphs required in Step 6. Comment on whether your phase synchronizer was able to compensate for frequency offsets of 100 Hz and 1 kHz.

LAB 6

# Frequency Modulation

# 6.1 Objective

This laboratory exercise introduces frequency modulation. This lab exercise is a nice illustration of the utility of the software defined radio approach, since the algorithms for creating and demodulating FM in software are much simpler than those used in the traditional hardware approach.

# 6.2 Background

## Frequency Modulation

Frequency modulation (FM) was introduced by E.A. Armstrong in the 1930's as an alternative to the AM commonly in use at the time for broadcasting. The advantage to frequency modulation is that, for a given transmitted power, the signal-to-noise ratio is much higher at the receiver output than it is for AM. The digital version of FM, frequency-shift keying, has been in use since an even earlier date.

In FM, the frequency of the carrier is modulated to follow the amplitude of the message signal. To be more specific, if $m(t)$ is a message signal with peak value $m_p$, then the *instantaneous frequency* $f(t)$ of the carrier is given by

$$f(t) = f_c + k_f m(t), \tag{1}$$

where $f_c$ is the carrier frequency and $k_f$ is a proportionality constant called the "frequency sensitivity." The term $k_f m(t)$ is called the "frequency deviation" of the instantaneous frequency from the carrier frequency, and the *peak frequency deviation* $\Delta f = k_f m_p$ is an important FM system parameter. Given the instantaneous frequency, we can find the total instantaneous angle $\theta(t)$ of the carrier by integrating the instantaneous frequency. That is,

$$\begin{aligned} \theta(t) &= 2\pi \int_0^t f(\alpha)\, d\alpha \\ &= 2\pi f_c t + 2\pi k_f \int_0^t m(\alpha)\, d\alpha + \theta(0). \end{aligned} \tag{2}$$

Since the initial angle $\theta(0)$ is of no consequence, we can simplify the equations by taking $\theta(0)=0$. The transmitted FM signal is then given by

$$\begin{aligned} g(t) &= A_c \cos\left[\theta(t)\right] \\ &= A_c \cos\left[2\pi f_c t + 2\pi k_f \int_0^t m(\alpha)\,d\alpha\right] \\ &= A_c \cos\left[2\pi f_c t + 2\pi \Delta f \int_0^t \left[m(\alpha)/m_p\right] d\alpha\right]. \end{aligned} \tag{3}$$

Figure 1 shows a 2 kHz carrier frequency modulated by a 200 Hz sinusoidal message.

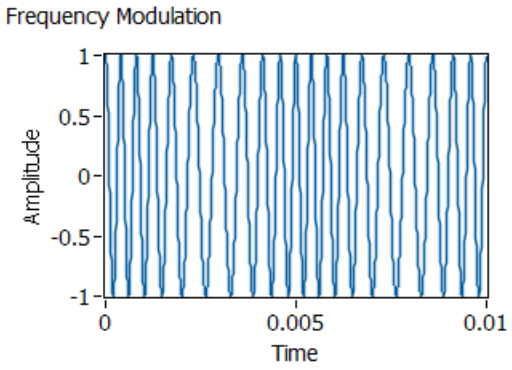



**Figure 1. Frequency Modulated Signal**

To create an FM signal using the USRP, the message signal is normalized to a peak value of one, multiplied by $2\pi\Delta f$ and integrated to give $2\pi\Delta f \int_0^t \left[m(\alpha)/m_p\right] d\alpha$. Next, the complex-valued signal $\tilde{g}(t)$ is formed, where

$$\tilde{g}(t) = A_c e^{j2\pi\Delta f \int_0^t \left[m(\alpha)/m_p\right] d\alpha}. \tag{4}$$

The complex-valued signal $\tilde{g}(t)$ is sent to the Write Tx Data function, and the USRP produces the FM signal.

The only tricky step in generating an FM signal is integrating the message. In discrete time we have

$$\int_0^t x(\alpha)\,d\alpha \cong \sum_{k=0}^{n} x(nT)T, \tag{5}$$

where $T$ is the reciprocal of the IQ sample rate. If we write

$$y[n] = \sum_{k=0}^{n} x(nT)T, \tag{6}$$

then

$$\begin{aligned} y[n] - y[n-1] &= \sum_{k=0}^{n} x(nT)T - \sum_{k=0}^{n-1} x(nT)T \\ &= x(nT)T. \end{aligned} \tag{7}$$

Equation (7) is the difference equation of an IIR filter. This filter can be implemented using the IIR Filter function found in the Analysis→Signal Processing→Filters→IIR Filtering palette. Use a "forward coefficients" array of $[T]$ and a "reverse coefficients" array of $[1 \quad -1]$.

## Demodulation

FM demodulation is much easier to carry out using the USRP than it is using conventional hardware. The signal provided by the Fetch Rx Data function is $\tilde{r}(t)$ given by

$$\tilde{r}(t) = A_r e^{j2\pi\Delta f \int_0^t [m(\alpha)/m_p]\,d\alpha}. \tag{8}$$

This is identical to the expression given by Eq. (4), except for the magnitude $A_r$ and the presence of noise (not shown in the expression). The angle of $\tilde{r}(t)$ is easily extracted using a *Complex to Polar function*.[8] Unwrap the angle before proceeding to the next step. Next, the unwrapped angle is differentiated, giving $2\pi\Delta f\left[m(t)/m_p\right] = 2\pi k_f m(t)$. This result should be passed through a lowpass filter, since the differentiation step tends to enhance high-frequency noise.

To implement the differentiator, we recognize that in discrete time,

[8] For those familiar with conventional FM demodulation, this step implements the limiter.

$$\frac{dx(t)}{dt} \cong \frac{x[n]-x[n-1]}{1}. \tag{9}$$

The differentiator can be implemented using an FIR Filter function from the Analysis→Signal Processing→Filters→ FIR Filtering palette. For the "FIR coefficients" array use $[1 \quad -1]$.

# 6.3 Pre-Lab

## Transmitter

1. A template for the transmitter has been provided in the file *Lab6TxTemplate.gvi*. This template contains the four interface functions along with a "message generator" that is set to produce a message signal consisting of three tones. The three tones are initially set to 1, 2, and 3 kHz, but these frequencies can be changed using front-panel controls. Your task is to add blocks as needed to produce the signal $\tilde{g}(t)$ of Eq.(4), and then to pass this signal to the Write Tx Data block. Set the carrier level $A_c$ to a constant value of 0.9. Figure 2 shows how to implement the FM modulator described by Eq. (4).

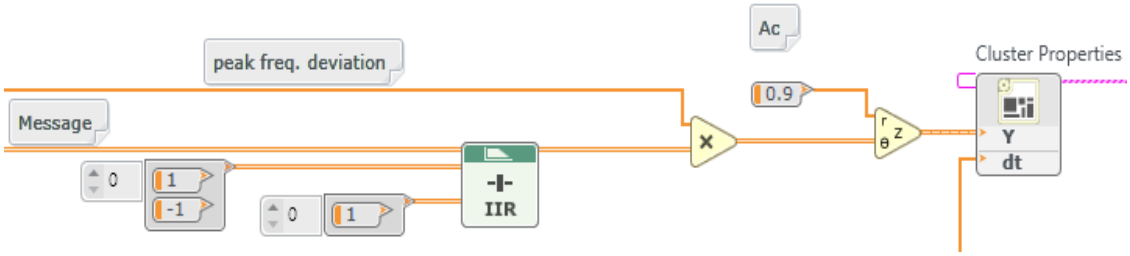



**Figure 2. FM Modulator**

Save your transmitter in a file whose name includes the letters "FMTx" and your initials (e.g., *FMTx_BAB.gvi*).

## Receiver

2. A template for the receiver has been provided in the file *Lab6RxTemplate.gvi*. This template contains the six interface functions along with a waveform graph on which to display your demodulated output signal.

Complete the program to demodulate the complex array returned by *Fetch Rx Data* and display the result. Figure 3 shows an implementation of the FM demodulator, including extracting the angle, unwrapping, differentiation, and lowpass filtering.

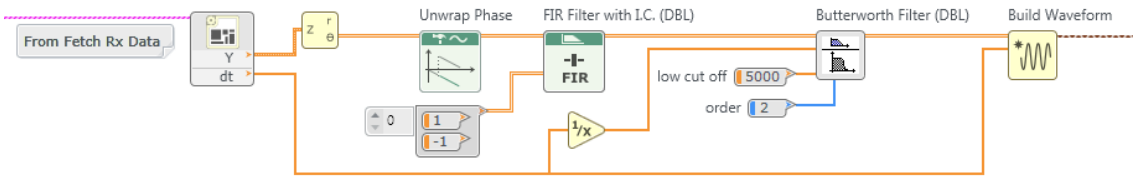



**Figure 3. FM Demodulator**

Save your receiver in a file whose name includes the letters "FMRx" and your initials (e.g., *FMRx_BAB.gvi*).

## Questions

1. Find the frequency response of the integrator given by Eq. (7). Compare with the frequency response of an ideal integrator. Is the discrete-time integrator more like an ideal integrator when the frequency of the input is low or when it is high?

2. Find the frequency response of the differentiator given by Eq. (9). Compare with the frequency response of an ideal differentiator. Is the discrete-time differentiator more like an ideal differentiator when the frequency of the input is low or when it is high?

# 6.4 Lab Procedure

1. Connect a loopback cable and attenuator between the TX 1 and RX 2 connectors. Connect the USRP to your computer and plug in the power to the USRP. Run LabVIEW and open the transmitter and receiver that you created in the prelab.

Note that all graphs in this lab are taken on a linear scale (dB on = False).

2. Ensure that the transmitter is set up to use

   Carrier Frequency: 915 MHz

   IQ Rate: Not critical; 1 MHz

   Gain: Not critical. 0 dB

   Active Antenna: TX1

   Message Length: 200,000 samples gives a good block of data to work with.

   Peak Frequency Deviation: 30 kHz seems a good value to start with.

   Start Frequency, Delta Frequency, Number of Tones: Not critical, but keep the highest frequency below 5 kHz. Three tones seems to work well, but you may wish to start with a single tone to verify operation.

3. Ensure that the receiver is set up to use

   Carrier Frequency: 915 MHz

   IQ Rate: 1 MHz

   Gain: Not critical. 0 dB

   Active Antenna: RX2

   Number of Samples: Same value as the transmitted message length.

   Run the transmitter, then run the receiver. After a few seconds, stop the receiver using the STOP button, then stop the transmitter (using the STOP button). Use the horizontal zoom feature on the graph palette to expand the "message" waveform in the transmitter and the "demodulated output" waveform in the receiver. Both waveforms should be identical, except for scaling.

4. The bandwidth of an FM signal is notoriously difficult to calculate analytically. J.R. Carson, writing in the 1920's, provided a rule of thumb for approximating the bandwidth:

$$B_{FM} \cong 2\left(\Delta f + B\right), \tag{10}$$

where $B_{FM}$ is the bandwidth of the FM signal, $\Delta f$ is the peak frequency deviation of the signal, and $B$ is the message bandwidth.

Add an *FFT Power Spectrum and PSD* (ExternalFiles folder) to your transmitter. Obtain the "time signal" input from the complex baseband waveform that is being sent to the *USRP Write Tx Data*. Connect the "Power Spectrum/PSD" output to a waveform graph. Change the label on the horizontal axis of the graph to "Frequency." The spectrum you obtain will be identical to the power spectrum of the actual transmitted FM signal, except that the carrier will appear at zero hertz, with the lower sideband on the negative-frequency side and the upper sideband on the positive-frequency side.

To obtain the classic textbook FM spectrum, set the message for a single tone at 1 kHz. Run the transmitter and obtain power spectra of the transmitted signal for peak frequency deviations of 1 kHz, 5 kHz, and 30 kHz. Take a screenshot of the power spectrum for each case. Be sure to scale the horizontal axis so that each spectrum is visible. Annotate your spectra to show the Carson's rule bandwidth, Eq. (10), for each case.

For a more realistic set of FM spectra, set the message for three tones at 1 kHz, 2 kHz, and 3 kHz. Run the transmitter and obtain power spectra of the transmitted signal for peak frequency deviations of 1 kHz, 5 kHz, and 30 kHz. Take another set of screenshots of the power spectrum for each case. Be sure to scale the horizontal axis so that each spectrum is visible. Annotate your spectra to show the Carson's rule bandwidth for each case.

5. One of the more curious, but also very useful, phenomena associated with FM is the so-called "capture effect." To observe this effect you will need to modify your transmitter to simultaneously generate two FM baseband signals at different carrier levels.

In your transmitter, duplicate the *Basic Multitone* to generate a second message signal. Your two *Basic Multitone* functions can share "sampling info," but each should have its own set of start frequency, stop frequency, and #tones. The constants "T" and "1" can be shared as well.

Inside the transmitter's *while* loop, create a second modulator, including a second integrator. Create two front panel controls, "Carrier 1" and "Carrier 2" to set the carrier levels $A_c$ for each modulator. For simplicity, the two modulators can share a common peak frequency deviation. Add the two FM baseband signals produced by your modulators together and send the sum to the *Build Waveform function* that feeds the *USRP Write Tx Data*.

Set up your first message generator for three tones at 1 kHz, 2 kHz and 3 kHz. Set up the second message generator for three tones at 100 Hz, 200 Hz, and 300 Hz. These two message signals should be easy to distinguish at the receiver. Set the peak frequency deviation to 30,000 Hz.

At this point test your system by setting carrier 1 to 0.9 and carrier 2 to zero. Run the transmitter and receiver and make sure that the receiver output matches message 1. Next, set carrier 1 to zero and carrier 2 to 0.9 and make sure that the receiver output matches message 2.

At the receiver, add an *FFT Power Spectrum and PSD* to view the spectrum of the baseband output. Label the horizontal axis "Frequency" and set the range to show 0 to 5000 Hz. It will be easier to distinguish message 1 from message 2 in the frequency domain than in the time domain.

Now you are ready to observe the capture effect! Start by setting carrier 1 to 0.4 and carrier 2 to 0.6. Run the transmitter and receiver. Take a screenshot of the receiver's baseband output spectrum. Repeat with carrier 1 set to 0.5 and carrier 2 set to 0.5. Repeat a third time with carrier 1 set to 0.6 and carrier 2 set to 0.4. You should find that in the first and third cases, the receiver demodulates (captures) only the stronger signal. This is the capture effect: If two FM signals are received at the same carrier frequency, the receiver will demodulate the stronger signal, even if the stronger carrier is only slightly stronger than the weaker one.

# 6.5 Report

## Prelab

Hand in documentation for your transmitter and receiver programs. Also include documentation for any additional functions you may have created. To obtain documentation, print out legible screenshots of the front panel and block diagram.

Submit your answers to the Questions at the end of the Prelab section.

## Lab

Submit the functions you created to implement the FM transmitter and receiver. Also submit any additional functions you may have created. Be sure your files adhere to the naming convention described in the instructions above. Resubmit documentation for any functions you modified during the lab.

Submit the graphs required in Step 4. Be sure to indicate on each graph the bandwidth estimated by Carson's rule. With reference particularly to the spectra for the three-tone message, does an FM signal always have sidebands that are symmetrical with respect to the carrier?

Submit the graphs required in Step 5. Explain briefly how this sequence of graphs demonstrates the capture effect.

LAB 7

# Amplitude-Shift Keying

**Prerequisite:** Lab 2 – Amplitude Modulation

# 7.1 Objective

Amplitude-shift keying (ASK) is the simplest form of digital modulation. We will use it to provide an introduction to digital communications, and as a vehicle to introduce some of the features that are common to digital communication systems, such as symbol mapping, pulse shaping, matched filtering, threshold detection, and pulse synchronization.

In this lab project, design of the transmitter and design of the receiver each present challenges and opportunities for investigation. The lab project is consequently divided into several parts. The transmitter part investigates creation of the ASK signal and the effect of transmitted pulse shape on the bandwidth of the transmitted signal. The receiver part investigates demodulation, matched filtering, and signal detection. There is a third part investigating alignment of the receiver and transmitter bit streams.

# 7.2 Part 1: Transmitter

## Background

ASK is simply AM with a binary message waveform. To illustrate, suppose $m(t)$ is a binary message represented in a polar, non-return-to-zero (NRZ) format, as shown in Figure 1.

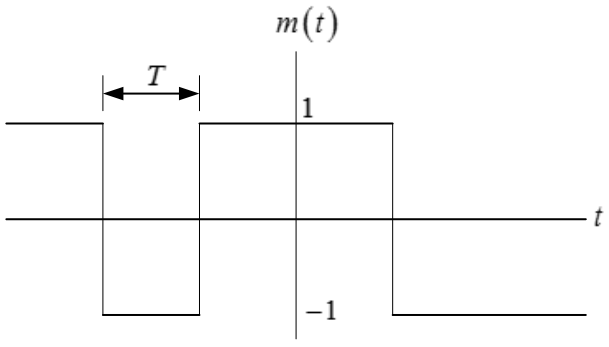



**Figure 1. Binary Message Waveform**

In the polar NRZ format, a binary 1 is represented by a pulse of amplitude $+1$ and a binary 0 by a pulse of amplitude $-1$. Each pulse has a duration of $T$ seconds. Let us define an AM signal as

$$g_{ASK}(t) = A\left[1 + m(t)\right]\cos(2\pi f_c t), \tag{1}$$

where $f_c$ is the carrier frequency and $A$ is the carrier amplitude. It is evident in Eq. (1) that either $g_{ASK}(t) = 2A\cos(2\pi f_c t)$ or $g_{ASK}(t) = 0$, depending on whether the corresponding message bit is a 1 or a 0. Thus the carrier is turned "on" to transmit a 1 and "off" to transmit a 0. This mode of ASK is sometimes referred to as "on-off" keying.

The pulse duration $T$ that appears in Figure 1 will be called the "symbol time" in this and subsequent lab projects. The "symbol rate" is then $1/T$. In a binary modulation method such as ASK, the symbol rate and the bit rate are identical. We will encounter other modulation methods, however, such as phase-shift keying, in which multiple bits can be transmitted on each symbol and the bit rate may therefore be faster than the symbol rate.

As straightforward as ASK is, several distinct steps are needed to actually produce a modulated signal. These are:

1. Symbol Mapping. The input data arrives as a stream of bits. Bits can be represented in any of a variety of formats. We will see below that the *MT Generate Bits* function produces an array of bytes (8-bit integers) containing the numbers 1 and 0. A bit stream can also be represented as a Boolean array. In the symbol mapping step, the bits are replaced by numerical values. For ASK we will represent a binary 1 by the complex double $1 + j0$ and a binary 0 by the complex double $-1 + j0$. Note that we are representing bits by complex numbers, even though the imaginary parts are zero. This is because the USRP requires a complex-valued input, and because in a future lab we will use the imaginary part to carry additional data. The complex numbers that represent the input bits are known as "symbols." Table 1 shows the ASK symbol mapping.

**Table 1. ASK Symbol Mapping**

| Bit Value | Symbol |
|---|---|
| 0 | $-1+j0$ |
| 1 | $1+j0$ |

2. Upsampling. We will see below that the symbols are carried on pulses whose shape is important in establishing the bandwidth of the transmitted signal. As a first step toward replacing symbols by pulses, we will place $L-1$ zeroes after each symbol. This produces a sample interval of

$$T_x = \frac{T}{L}, \tag{2}$$

or a sample rate of

$$\frac{1}{T_x} = L\frac{1}{T}. \tag{3}$$

A higher upsampling factor $L$ makes the D/A conversion in the transmitter easier, but requires faster digital processing. We will use $L = 20$ in this lab project. The *Upsample* function is made to order for implementing this step.

3. Pulse Shaping. If the upsampled signal is applied to a filter whose impulse response $g_{TX}[n]$ is a rectangular pulse of unit amplitude and length $L$ samples, then at the filter output, each symbol will be represented by a rectangular pulse. Figure 2 shows the effect of upsampling and filtering. Waveform (a) represents the symbol sequence, with symbols occurring every $T$ seconds. Waveform (b) shows the symbol sequence after upsampling. Waveform (c) shows the upsampled symbol sequence after filtering by the pulse-shaping filter. Note that waveform (c) is a discrete-time version of the polar NRZ message waveform shown in Figure 1. An important advantage to following this particular sequence of steps to generate the message waveform is that the impulse response $g_{TX}[n]$ of the pulse-shaping filter does not have to be a rectangular

pulse. We will see later in this lab project that there can be advantages to using other pulse shapes.

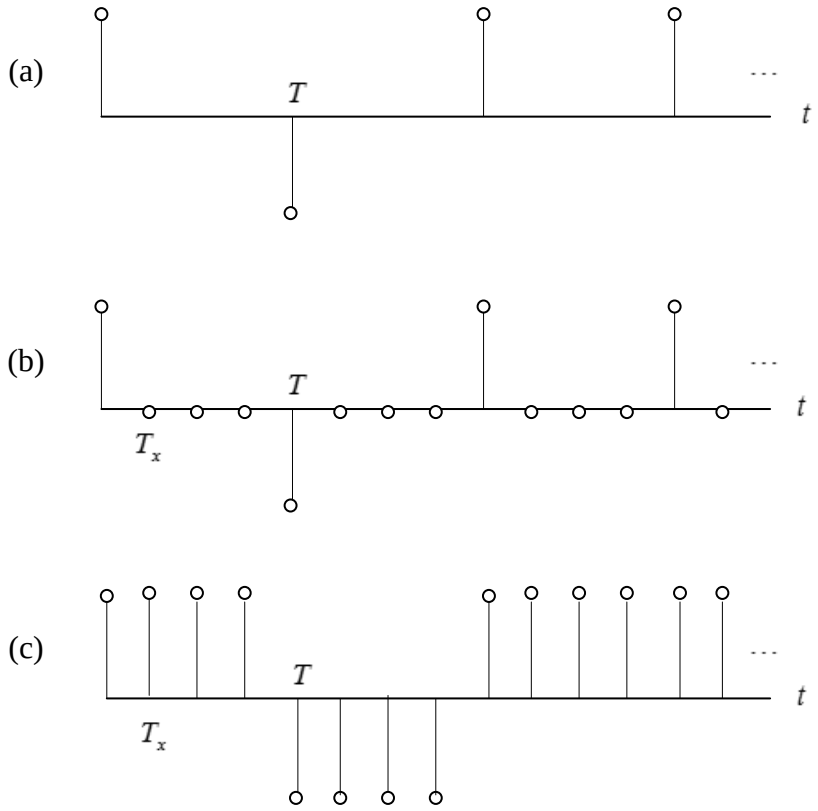



**Figure 2. (a) Symbol sequence, (b) Upsampled symbol sequence, (c) Filtered upsampled symbol sequence.**

4. Modulation. If $m[n]$ represents the message waveform at the transmitter filter output, then the final step is to apply Eq. (4):

$$\tilde{g}_{ASK}[n] = A(1+m[n]), \tag{4}$$

where the constant $A$ is chosen to keep the magnitude of $\tilde{g}_{ASK}[n]$ less than 1. This signal can be sent to the USRP transmitter. The USRP transmitter will perform the D/A conversion and multiplication by the carrier $\cos(2\pi f_c t)$.

# Prelab

1. Create a program to generate an ASK signal using the USRP. A template for the transmitter has been provided in the file *ASKTxTemplate.gvi*. This template contains the four functions for interfacing with the USRP along with *MT Generate Bits* from the Modulation Toolkit. *MT Generate Bits* will create a pseudorandom sequence of bits that can serve as a data sequence for testing your ASK system. Note that by default, *MT Generate Bits* will produce the same sequence of bits every time you run the program. This is useful for debugging, but if you would like to generate a different sequence of bits every time, wire a random number to the "seed in" input. The steps below contain details about how to create the required transmitter.

2. First do the symbol mapping, as shown in Table 1. This will convert the integers 0,1 from *MT Generate Bits* to complex doubles $\pm 1 + j0$. In future lab projects, where the symbol mapping may be more elaborate, this might be implemented as a sub-vi. In this lab project the symbol mapping is relatively simple, and a sub-vi implementation is optional.

3. Upsample using *Upsample* from the Analysis→Signal Processing→Signal Operation subpalette. In this lab project you are given control inputs to set the symbol rate $1/T$ and the IQ rate $1/T_x$. Set the symbol rate to 10,000 symbols/s and the IQ rate to $200 \times 10^3$ Sa/s. Use these two inputs to calculate the upsampling factor $L$.

3. Use *MT Generate Filter Coefficients* from the Modulation Toolkit to generate the pulse shaping filter. (*MT Generate Filter Coefficients* can be found on the Analysis→Communications→Digital→Utilities subpalette.) Set the modulation type to ASK, and the pulse-shaping filter "samples per symbol" to your calculated value of $L$. Create a front-panel control for "pulse shaping filter" and set this initially to "none." The setting of "none" will generate rectangular pulses. ("none" does not mean that there is no filter!) Wire the "pulse shaping filter coefficients" output to the "Y" input of a *Convolution*. The *Convolution* is available from the Analysis→Signal Processing→Signal Operation subpalette. Wire the output from your upsampler to the "X" input of the *Convolution*.

4. Normalize the amplitude of your filtered message signal to a maximum absolute value of 1. This will be important later when we investigate alternative pulse shapes. Check out the

*Quick Scale 1D* in the ExternalFiles folder for finding the maximum of the absolute value. To ensure that your scaled message remains complex-valued, use a separate division function to do the actual scaling. (That is, do not use the $Y[i] = X[i]/\mathrm{Max}|X|$ output of *Quick Scale 1D*.)

5. Implement Eq. (4). Let the constant $A$ be $1/2$, so that $\tilde{g}_{ASK}[n]$ varies between zero and one. Combine $\tilde{g}_{ASK}[n]$ with $T_x$ using the *Build Waveform function* provided in the template to produce the "Baseband Signal." Note that $1/T_x$ is available as the "actual IQ rate." Also connect $\tilde{g}_{ASK}[n]$ to the "data" input of the *Write Tx Data* function.

6. An *FFT Power Spectrum for 1 Chan* has been provided in the template to allow you to observe the spectrum of the transmitted signal.

This completes construction of the ASK transmitter. Save your transmitter in a file whose name includes the letters "ASKTx" and your initials (e.g. *ASKTx_BAB.gvi*).

## Lab Procedure

1. Connect a loopback cable and attenuator between the TX 1 and RX 2 connectors of the USRP. Connect the USRP to your computer and plug in the power to the USRP. Run LabVIEW and open the transmitter that you created in the prelab.

2. Ensure that the transmitter is set up to use

   Carrier Frequency: 915.1 MHz (Note: The 100 kHz offset from the receiver carrier frequency is deliberate.)

   IQ Rate: 200 kHz. Note: This sets the value of $1/T_x$.

   Gain: 0 dB

   Active Antenna: TX1

   Symbol rate: 10,000 symbols/s

Message Length: 1000 bits

Pulse shaping filter: None

Run the transmitter. Use the large STOP button on the front panel to stop transmission connectors.

3. After running the transmitter, observe the spectrum of the transmitted signal. You should be able to clearly see the carrier at frequency "zero." Two additional features are significant: the bandwidth and the rate of spectral rolloff. Using the Capture Data feature or using cursors on the Power Spectrum graph, measure the null-to-null bandwidth of the transmitted signal. Relate this bandwidth to the symbol rate $1/T$. The rate of spectral rolloff is a measure of the interference that your signal will cause to signals using nearby carrier frequencies. Print a copy of the spectrum for comparison with the spectrum you will obtain in Step 4 below.

4. Rectangular pulses are rarely used in practice because of the very gradual spectral rolloff they produce. Change the pulse-shaping filter to "Root Raised" for a root-raised-cosine filter. Run the program examine the spectrum again. Measure the null-to-null bandwidth of the transmitted signal. Print a copy of the spectrum and compare the rolloff rate with the spectrum you obtained using rectangular pulses.

*Questions*

1. In step 2 you are given $1/T = 10{,}000 \text{ symbols/s}$ and $1/T_x = 200 \times 10^3 \text{ Sa/s}$. Find the corresponding value for the number of samples per symbol $L$.

2. Relate the symbol rate to the null-to-null bandwidth of the ASK signal for (a) rectangular and (b) root-raised-cosine pulses.

3. Compare the rates of spectral rolloff of the transmitted signal for rectangular and root-raised-cosine pulses.

# 7.3 Part 2: Receiver

## Background

An ASK receiver begins as an analog AM receiver. We will offset the transmitter and receiver carrier frequencies by 100 kHz, so that the signal retrieved from the USRP receiver will be an AM signal having an "intermediate" carrier frequency of 100 kHz. The retrieved AM signal will be passed through a bandpass "intermediate frequency" filter and then demodulated using an envelope detector. The envelope detector is implemented by taking the magnitude of the bandpass filter output, and then lowpass filtering using a second filter. If you completed Lab 2, Amplitude Modulation, these steps should be familiar.

For digital communications, the lowpass filter should be designed to minimize the effects of noise and to also minimize the effects of intersymbol interference that can be caused when the filtered received pulses overlap. The best filter for eliminating noise is a so-called "matched" filter. A matched filter has a frequency response whose magnitude matches the magnitude of the frequency response of the transmitter's pulse-shaping filter. That is, if $g_{RX}[n]$ is the impulse response of the receiver's filter, then we want $\left|G_{RX}\left(e^{j\omega}\right)\right| = \left|G_{TX}\left(e^{j\omega}\right)\right|$. By using *MT Generate Filter Coefficients* at both the transmitter and receiver, we will ensure that the appropriate receiver filter is chosen to match the pulse-shaping filter at the transmitter.

To complete the digital receiver, several additional steps follow the AM demodulator.

1. Pulse Synchronization. The AM receiver output is an analog baseband signal that must be sampled once per symbol time, i.e. once every $T$ seconds. Because of filtering and propagation delays and distortion caused by the communication channel, it is necessary to determine the optimum time to take these samples. A function *PulseAlign(real)* has been provided in the *BasicUSRPLabs folder* to align the baseband signal so that the sample at index 0 is the correct first sample.

2. Sampling. The *Decimate* function will sample the aligned baseband waveform at index 0 and every $T$ seconds thereafter.

3. Detection. Once the baseband waveform has been sampled, each sample must be examined to determine whether it represents a symbol of value 1 or a symbol of value 0.

4. Symbol Mapping. The detected symbol values must be converted to bits. For ASK, this step is easily included in the detection step.

## Prelab

1. A template for the receiver has been provided in the file *ASKRxTemplate.gvi*. This template contains the six interface functions for interfacing with the USRP.

   We want the receiver to capture two frames of data each time it is run. Since the receiver's starting point is random, this will ensure that there will always be one complete frame in the captured data. Using the message length and symbol rate available from front panel inputs, and the "Actual IQ Rate" available from *Configure Signal*, have the receiver calculate the number of samples in a frame. Then double this number and provide the result as the "number of samples" input to the *Fetch Rx Data*.

   *Fetch Rx Data* returns a complex double cluster which makes the "Y" and "dt" components available. Pass the "Y" complex array through a bandpass filter. Filters can be found in the ExternalFiles folder in the Files Pane folder in the Files Pane folder. Use a fifth-order Chebyshev Filter (CDB) with a high cutoff frequency of 110 kHz and a low cutoff frequency of 90 kHz. The default passband ripple of 0.1 dB is acceptable. The "sampling frequency" input to the filter can be obtained as the reciprocal of the "dt" component of the waveform returned by *Fetch Rx Data*. Note that dt is the same as the "actual IQ rate" returned by *Configure Signal*. We also designate this parameter as $1/T_z$ .

2. Extract the real part of the complex array at the output of the Chebyshev bandpass filter "Filtered X". To obtain the envelope, take the absolute value and pass the result through a matched filter. The absolute value functions as a full-wave rectifier. For the matched filter, use *MT Generate Filter Coefficients* just as you did for the transmitter. Set the modulation type to

ASK, and calculate the "matched samples per symbol" $M$ from the "actual IQ rate ($1/T_z$)" and the symbol rate ($1/T$) obtained from the front-panel control. Create a front-panel control for "pulse shaping filter" and set this initially to "none." The setting of "none" will generate a matched filter with a rectangular impulse response (not the absence of a filter, as you might imagine). Wire the "matched filter coefficients" output to the "Y" input of a *Convolution*. The *Convolution* is available from the Analysis→Signal Processing→Signal Operation subpalette. The output of your matched filter should be connected to the *Cluster Properties function* provided in the template. The *Cluster Properties function* feeds the Baseband Output graph.

3. A convenient way to visualize the output of a digital demodulator is the so-called "eye diagram." An eye diagram is a plot of the baseband output signal with the horizontal axis scaled to be one or two symbol times long and successive symbols superimposed.

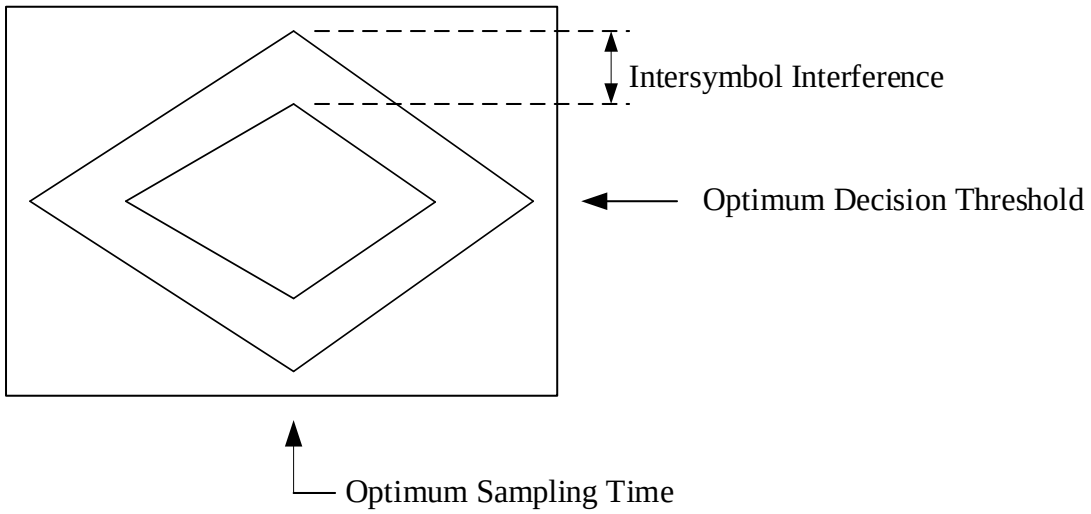



**Figure 3. Stylized Eye Diagram**

Figure 3 shows a stylized eye diagram. Some of the useful information that can be learned from the eye diagram is shown in the figure.

The Modulation Toolkit function *MT Format Eye Diagram* has been provided in the receiver template. Wire the baseband output waveform to the "waveform" input. The "symbol rate (Hz)" input value is available from the front panel control. Set the "eye length" parameter to 2.

4. Place the *PulseAlign(real)* on your block diagram and wire the baseband output waveform to the "input waveform" input and wire the calculated $M$ samples/symbol value to the "receiver sampling factor" input.

   Once the baseband waveform is aligned, it can be sampled. *Decimate (single shot)* can be obtained from the Analysis→Signal Processing→Signal Operation subpalette. The "decimating factor" is $M$.

5. To determine whether each received sample is more likely to represent a 1 or a 0, the sample must be compared with a threshold. Use the *Mean DBL* function from the ExternalFiles folder to compute the threshold. The result of the comparison is the receiver's digital output. The output of the comparison will be a Boolean array. You can convert this array to an integer array by using a *Boolean To Integer* function.

This completes construction of the ASK receiver. Save your receiver in a file whose name includes the letters "ASKRx" and your initials (e.g. *ASKRx_BAB.gvi*).

## Lab Procedure

1. Ensure that the receiver is set up to use

   Carrier Frequency: 915.0 MHz

   IQ Rate: 1 MHz

   Gain: 0 dB

   Active Antenna: RX2

   Symbol rate: 10,000 symbols/s

   Message length: 1000 bits

   Pulse shaping filter: none

2. Run the transmitter, then run the receiver. Once the receiver has acquired a block of data, you may stop the transmitter.

3. Use the horizontal zoom feature on the Baseband Output graph palette to expand the demodulated waveform so that you can see individual pulses. Ideally, rectangular pulses passed through the receiver's matched filter should produce triangular output pulses. Note whether the demodulated pulses have the expected shape.

4. Observe the eye diagram. Make note of the optimum sampling time and the presence of intersymbol interference.

5. Change the "pulse shaping filter" control at both the transmitter and the receiver to "Root Raised" for root-raised-cosine filters. Setting both the transmitter and receiver filters will ensure that the filters remain matched for optimum performance in the presence of noise. The cascade of two root-raised-cosine filters produces a raised-cosine pulse at the receiver filter output. The raised-cosine pulse is designed to minimize, or ideally eliminate, intersymbol interference.

   Run the transmitter and then run the receiver. Once the receiver has acquired a block of data, you may stop the transmitter. Observe the baseband output signal and the eye diagram. Comment on the changes to the eye diagram. Has intersymbol interference been reduced?

6. To see the effect of pulse synchronization, move the waveform input of *MT Format Eye Diagram* to the "aligned waveform" output of *PulseAlign(real)*. Run the transmitter and receiver again. Observe the eye diagram. What is the optimum sampling time now?

*Questions*

1. Give a formula showing how the IQ Sampling Rate, the symbol rate $1/T$, and the number of samples per symbol $M$ are related. Determine the value of $M$ for an IQ Sampling Rate of 1 MHz and a symbol rate of 10,000 symbols/s.

2. Note that the IQ sampling rate at the receiver is different from the IQ sampling rate at the transmitter. Using a higher IQ sampling rate requires faster digital processing. What is the advantage to using a higher IQ sampling rate at the receiver? (Hint: It has something to do with the action of the *PulseAlign(real)* and the value of $M$.)

3. Compare the rectangular and raised-cosine pulse shapes by examining the eye diagrams. What evidence for intersymbol interference do you see in each case?

4. Using root-raised-cosine pulses and receiver filtering, observe the eye diagram when the input of *MT Format Eye Diagram* is set to "baseband output" and when the input of *MT Format Eye Diagram* is set to the "aligned waveform" output of *PulseAlign(real)*. What function is *PulseAlign(real)* performing?

5. The transmitter is programmed to generate the same "frame" of 1000 symbols over and over. The receiver grabs a single block of 2000 symbols each time it is run. Can you identify, by examining the receiver's baseband output plot, where the symbol sequence ends and starts over? Frame synchronization of the receiver is an essential component of a digital communication system. We will examine frame synchronization in the next part of this lab project.

# 7.4 Part 3: Aligning the Received Bits

At this point you have a working transmitter and receiver, but it is hard to know whether the two are working together correctly as a system unless you can compare the received bits with the transmitted bits and verify that the bit patterns are the same. To allow this comparison, it is necessary that the receiver recognize the beginning of the transmitted sequence. The

*AddFrameHeader(real)*, available in the *BasicUSRPLabs folder*, inserts a specific 26-bit sequence at the start of transmission. At the receiver, the *FrameSync(real)*, also available in the *BasicUSRPLabs folder*, looks for this specific sequence and cuts off all bits received before this frame header. The *FrameSync(real)* also cuts off the frame header. This way, the bit sequence at the output of the receiver should match the bit sequence sent to the transmitter.

1. Add the *AddFrameHeader(real)* to the transmitter. Place *AddFrameHeader(real)* after the symbol mapping, but before conversion of the symbols to complex.

2. Add the *FrameSync(real)* to the receiver. Place *FrameSync(real)* immediately following *Decimate*. Wire the output of *Decimate* to the "Sampled Input" of *FrameSync(real)*. Leave the remaining inputs of *FrameSync(real)* unwired. Wire the "Aligned Samples" output of *FrameSync(real)* to the threshold comparison function and to *Mean* described in Receiver Prelab, Section 5.

   Wire the array of output bits from the threshold comparison to the "array" input of an *Array Subset function*. Set the "index" input to zero, and set the "length" input to the length specified by the "message length" control. Display the output of *Array Subset function* as "Output bits" on the receiver front panel.

   Note that the "Output Signal" and "max index" outputs of *FrameSync(real)* will not be used in this lab project.

3. Run the transmitter; then run the receiver; then stop the transmitter. Compare the first dozen or so received bits with the corresponding transmitted bits.

4. Measurement of the bit error rate (BER) can be automated using the *MT Calculate BER* from the Modulation Toolkit (Analysis→ Communications→ Digital→Measurements subpalette). From the Configure ribbon, select "PN Fibonacci." Set the "BER trigger threshold" to 0.4. Connect indicators to the "BER" and "trigger found?" outputs. When you run the program, "trigger found?" will be true whenever the measured BER is below the BER trigger threshold. Run the transmitter and receiver. The measured BER should be identically zero unless you add noise.

*Questions*

1. In step 3, do the received bits match the transmitted bits?

2. What would the measured BER be if there were something wrong with the receiver's bit alignment? Note that when the measured BER is higher than the BER trigger threshold, the "trigger found?" output will be false, and any BER reading shown will not be meaningful.

# 7.5 Report

## Prelab

Hand in documentation for the programs you created for the transmitter and receiver. Also include documentation for any functions you created. To obtain documentation, print out legible screenshots of the front panel and block diagram.

## *Lab*

Submit the programs you created for the transmitter and receiver. Also submit any functions you created. Be sure your files adhere to the naming convention described in the instructions above.

Resubmit documentation for any functions you modified during the lab.

Answer all of the questions in each of the sections marked *Questions* above.

LAB 8

# Frequency-Shift Keying

**Prerequisites:** Lab 6 – Frequency Modulation, Lab 7 – Amplitude-Shift Keying

# 8.1 Objective

In frequency-shift keying (FSK), a 1 is represented by a tone at a specific frequency, known traditionally as the "mark" frequency, while a 0 is represented by a tone at a different frequency, known as the "space" frequency. FSK owes part of its popularity to the fact that a tone is always being transmitted, even when the source generates a long string of zeros. This makes it easy for the receiver to distinguish between a transmitter that is idling and a transmitter that has stopped transmitting. FSK also has the property that the transmitted signal has a constant amplitude. This allows a very efficient nonlinear power amplifier to be used for transmission, a very important consideration when the transmitter is battery-powered.

FSK is the digital version of frequency modulation. Just as an ASK system is built around an AM transmitter and receiver, we will see that an FSK system is built around an FM transmitter and receiver. The additional features such as symbol mapping, pulse shaping, matched filtering, threshold detection, and pulse synchronization all apply to FSK as they do to ASK.

# 8.2 Background

## Transmitter

The bandwidth of the transmitted signal is an important property of any modulation method. Bandwidth has two components, the "main lobe" bandwidth determines the channel width needed to carry the transmitted signal, and also the bandwidth of the filter in the receiver front end. The rate of spectral rolloff determines the interference that a signal will cause to signals in adjacent channels. This rolloff determines how close in frequency similarly modulated signals can be placed. The bandwidth of an FSK signal is notoriously difficult to calculate analytically. J.R. Carson, writing in the 1920's, provided a rule of thumb for approximating the bandwidth:

$$B_{FSK} \cong 2\left(\Delta f + B\right), \tag{1}$$

where $B_{FSK}$ is the bandwidth of the FSK signal, $\Delta f$ is the peak frequency deviation of the signal (see below), and $B$ is the message bandwidth. Carson's rule is simple to apply, but it tends to provide a slight overestimate when applied to FSK signals.

The rolloff rate of an FSK signal is largely governed by the smoothness of the signal at the moments when the frequency changes from mark to space or vice versa. If the FSK signal is allowed to become discontinuous at these moments, the rolloff in the power spectrum will be proportional to $1/f^2$, which is comparable to ASK using rectangular pulses. *Continuous-phase* FSK has a spectral rolloff of $1/f^4$. Consequently, most modern applications only use the continuous-phase version. Even more rapid spectral rolloff rates can be achieved by smoothing the message signal before applying it to the FSK transmitter.

The generation of an analog FM signal is discussed at length in the background section of *Lab 6: Frequency Modulation*. It is strongly suggested that this material be reviewed at this point. Recall that given a message $m(t)$, the instantaneous frequency of an FM signal is defined as

$$f(t) = f_c + k_f m(t), \tag{2}$$

where $f_c$ is the carrier frequency and $k_f$ is the frequency sensitivity. The peak frequency deviation of the FSK signal is given by

$$\Delta f = k_f m_p, \tag{3}$$

where $m_p$ is the peak value of the message. To form a continuous-phase FSK signal, the message signal $m(t)$ is a binary message represented in a polar, non-return-to-zero (NRZ) format, as shown in Figure 1.

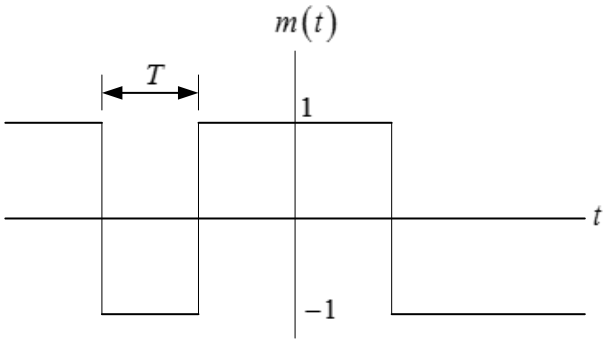



**Figure 1. Binary Message Waveform**

The pulse duration $T$ that appears in Figure 1 will be called the "symbol time," just as was the case in the ASK lab project. The peak value of the message is $m_p = 1$. Given the polar NRZ message format, we see from Eq. (2) that the mark frequency is $f_c + \Delta f$ and the space frequency is $f_c - \Delta f$. It is important to note, referring to Eq. (1), that the bandwidth of the FSK signal is not $2\Delta f$, but includes a contribution from the message bandwidth. Thus the bandwidth cannot be made arbitrarily small by reducing the peak frequency deviation.

As is shown in Lab 6, the signal that must be sent to the USRP to produce an FM output is

$$\tilde{g}(t) = A_c e^{j2\pi\Delta f \int_0^t m(\alpha)d\alpha}, \tag{4}$$

where $A_c$ is the carrier amplitude. The USRP will produce the FM signal given by

$$g(t) = \mathrm{Re}\left[\tilde{g}(t)e^{j2\pi f_c t}\right] = A_c \cos\left[2\pi f_c t + 2\pi\Delta f \int_0^t m(\alpha)d\alpha\right]. \tag{5}$$

It should be noted that the phase of the FM signal given by Eq. (5) is proportional to $\int_0^t m(\alpha)d\alpha$. For the polar NRZ message given in Figure 1, this integral yields a continuous function of time.

The steps needed, in addition to frequency modulation, to form an FSK signal should be familiar if you have completed the ASK lab project:

1. Symbol Mapping. The input data arrives as a stream of bits. Recall that the *MT Generate Bits* function produces an array of bytes containing the numbers 1 and 0. In the symbol mapping

step, the bits are replaced by numerical values. For FSK we will represent a binary 1 by the real double $1$ and a binary 0 by the real double $-1$. Table 1 shows the FSK symbol mapping.

**Table 1 FSK Symbol Mapping**

| Bit Value | Symbol |
|---|---|
| 0 | $-1$ |
| 1 | $1$ |

2. Upsampling. As a first step toward replacing symbols by pulses, we will place $L-1$ zeros after each symbol. This produces a sample interval of

$$T_x = \frac{T}{L}, \tag{6}$$

or a sample rate of

$$\frac{1}{T_x} = L\frac{1}{T}. \tag{7}$$

A higher upsampling factor $L$ makes the D/A conversion in the transmitter easier, but requires faster digital processing. We will use $L = 40$ in this lab project.

3. Pulse Shaping. If the upsampled signal is applied to a filter whose impulse response $g_{TX}[n]$ is a rectangular pulse of unit amplitude and length $L$ samples, then at the filter output, each symbol will be represented by a rectangular pulse. Steps 1 through 3 convert the input bit stream to a polar NRZ message signal as shown in Figure 1. Note that pulse shapes other than rectangular can be used simply by changing the shape of the impulse response $g_{TX}[n]$.

4. Modulation. Once we have the message signal $m(t)$, Eq. (4) is applied to produce the baseband signal to be sent to the USRP transmitter.

## Receiver

An FSK receiver begins with an FM demodulator. First, the phase is extracted from the complex baseband signal received from the USRP. Next, the phase is differentiated and filtered. These steps are described in more detail in the background section of Lab 6. To complete the digital receiver, several additional steps follow the FM demodulator:

1. Pulse Synchronization. The FM receiver output is an analog baseband signal that must be sampled once per symbol time, i.e. once every $T$ seconds. Because of filtering, propagation delays, and distortion caused by the communication channel, it is necessary to determine the optimum time to take these samples. A function called *PulseAlign(real)* has been provided in the *BasicUSRPLabs folder* to align the baseband signal so that the sample at index 0 is the correct first sample.

2. Sampling. The *Decimate* function will sample the aligned baseband waveform at index 0 and every $T$ seconds thereafter.

3. Detection. Once the baseband waveform has been sampled, each sample must be examined to determine whether it represents a symbol of value 1 or a symbol of value 0.

4. Symbol Mapping. The detected symbol values must be converted to bits. For FSK, this step is easily included in the detection step.

# 8.3 Pre-Lab

## Transmitter

1. Create a program to generate a continuous-phase FSK signal using the USRP. A template for the transmitter has been provided in the file FSKTxTemplate.gvi. This template contains the four functions for interfacing with the USRP along with *MT Generate Bits* from the Modulation Toolkit. *MT Generate Bits* will create a pseudorandom sequence of bits that can serve as a data sequence for testing your FSK system. Note that by default, *MT Generate Bits* will produce the same sequence of bits every time you run the program. This is useful for debugging, but if you would like to generate a different sequence of bits every time, wire a random number to the "seed in" input. The steps below contain details about how to create the required transmitter.

2. First do the symbol mapping, as shown in Table 1.

3. Upsample the array of symbols using *Upsample* from the Analysis→Signal Processing→Signal Operation subpalette. In this lab project you are given control inputs to set the symbol rate $1/T$ and the IQ rate $1/T_x$. Set the symbol rate to 10,000 symbols/s and the IQ rate to $400 \times 10^3$ Sa/s. Use these two inputs to calculate the upsampling factor $L$.

4. Use *MT Generate Filter Coefficients* from the Modulation Toolkit to generate the pulse shaping filter. (*MT Generate Filter Coefficients* can be found on the Analysis→Communications→Digital→Utilities subpalette.) Set the modulation type to FSK, and the pulse-shaping filter "samples per symbol" to your calculated value of $L$. Create a front-panel control for "pulse shaping filter" and set this initially to "none." As in the ASK lab project, the setting of "none" will generate rectangular pulses. (Remember: "none" does not mean that there is no filter.) Wire the "pulse shaping filter coefficients" output to the "Y" input of a *Convolution*. The *Convolution* is available from the Analysis→Signal Processing→Signal Operation subpalette. Wire the output from your upsampler to the "X" input of the *Convolution*.

5. Normalize the amplitude of your filtered message signal to a maximum absolute value of 1. This step will be important later when we investigate alternative pulse shapes. The *Quick Scale*

*1D* in the ExternalFiles folder is ideal for this task. (You may use the $Y[i] = X[i]/\mathrm{Max}|X|$ output of *Quick Scale 1D*, since the "X" input is real-valued in this lab project.) Connect the output of *Quick Scale* to the *Build Waveform* function that connects to the Message Signal graph provided in the template.

6. Now implement Eq. (4). To implement the integral use an IIR Filter from the Analysis→Signal Processing→Filters→ IIR Filtering palette. Use a "forward coefficients" array of $[1]$ and a "reverse coefficients" array of $[1 \quad -1]$. Multiply the integrated message by $2\pi\Delta f$ and also by $dt$. The peak frequency deviation $\Delta f$ is available from a front panel control; set the deviation initially to 5000 Hz. The sample interval $dt$ is the reciprocal of the "actual IQ rate" ($dt$ and $T_x$ refer to the same quantity). Use a *polar to complex* function to create the complex baseband signal. Let the carrier amplitude $A_c$ be $1$. The complex baseband signal connects to the "data" input of *Write Tx Data*.

7. To observe the spectrum of the transmitted signal, wire the complex baseband signal to the *Build Waveform* function that connects to the Baseband Power Spectrum graph provided in the template.

This completes construction of the FSK transmitter. Save your transmitter in a file whose name includes the letters "FSKTx" and your initials (e.g. *FSKTx_BAB.gvi*).

## Receiver

1. Create a program to implement an FSK receiver using the USRP. A template for the receiver has been provided in the file *FSKRxTemplate.gvi*. This template contains the six interface functions for interfacing with the USRP.

   Calculate the "number of samples" for the *Fetch Rx Data* to fetch using the message length and symbol rate front panel inputs. Double the number the number of samples in a frame so that the receiver will fetch two frames of data. Since the receiver's starting point is random, this ensures that there will be one complete frame of received data.

Pass the complex array returned by the *Fetch Rx Data* function through a *Complex to Polar* function to extract the phase. Unwrap the phase using *Unwrap Phase* (Analysis→Signal Processing→Signal Operation subpalette) to remove jumps of $2\pi$ before taking the derivative. To differentiate the phase, use an *FIR Filter*, with FIR Coefficients set to the array $[1 \quad -1]$. Differentiating the phase will create large spikes wherever there is a phase discontinuity. These occur at the beginning of the signal, and wherever the transmitter begins its transmitted sequence over again. To smooth out the spikes, use a *Median Filter* from the ExternalFiles folder. Set the "left rank" input to 5 and leave the "right rank" unwired.

2. To implement the receiver's matched filter, use *MT Generate Filter Coefficients* just as you did for the transmitter. Set the modulation type to FSK, and calculate the "matched samples per symbol" $M$ from the "actual IQ rate" ($1/T_z$) and the symbol rate ($1/T$) obtained from the front-panel control. Create a front-panel control for "pulse shaping filter" and set this initially to "none." (Remember: the setting of "none" will generate a matched filter with a rectangular impulse response.) Wire the "matched filter coefficients" output to the "Y" input of a *Convolution*. The output of your matched filter should be connected to the *Cluster Properties function* provided in the template. The *Cluster Properties function* feeds the Baseband Output graph.

3. Place the *PulseAlign(real)* on your block diagram and wire the baseband output waveform to the "input waveform" input and wire the $M$ samples/symbol to the "receiver sampling factor" input.

   Once the baseband waveform is aligned, it can be sampled. *Decimate (single shot)* can be obtained from the Analysis→Signal Processing→Signal Operation subpalette. The "decimating factor" is $M$.

4. The Modulation Toolkit function *MT Format Eye Diagram* has been provided in the receiver template. Wire the baseband output waveform to the "waveform" eye-diagram input. The "symbol rate (Hz)" input value is available from the front panel control. Set the "eye length" parameter to 2.

5. To determine whether each received sample is more likely to represent a 1 or a 0, the sample must be compared with a threshold. Because the message $m(t)$ is a polar signal, the threshold can be taken as zero. The result of the comparison is the receiver's digital output.

This completes construction of the FSK receiver. Save your receiver in a file whose name includes the letters "FSKRx" and your initials (e.g. *FSKRx_BAB.gvi*).

## Questions

1. In *Transmitter* step 3 you are given $1/T = 10{,}000 \text{ symbols/s}$ and $1/T_x = 400 \times 10^3 \text{ Sa/s}$. Find the corresponding value for the number of samples per symbol $L$.

2. Explain how an IIR filter having a forward coefficients array of $[1]$ and a reverse coefficients array of $[1 \quad -1]$ implements an integrator.

3. Explain how an FIR filter having a coefficients array of $[1 \quad -1]$ implements a differentiator, as used in *Receiver* step 1.

4. Explain what a median filter does. Refer to the LabVIEW 2014 online help for the *Median Filter* function for information.

5. In *Receiver* Step 2 the "actual IQ rate" $1/T_z$ may be different than the rate $1/T_x$ that was used at the transmitter. ( Note that the symbol rate $1/T$ must be the same at the transmitter and receiver.) The value of the receiver's IQ rate determines the receiver sampling factor $M$. What is the advantage to using a higher value of $M$? What is the advantage of using a lower value of $M$?

6. The FM receiver uses a differentiator to undo the integral in Eq. (4). What is the effect of the differentiator on any noise that might be present along with the signal? To answer this,

consider what a differentiator does in the frequency domain. What would be the effect of omitting the filter that follows the differentiator?

# 8.4 Lab Procedure

1. Connect a loopback cable and attenuator between the TX 1 and RX 2 connectors of the USRP. Connect the USRP to your computer and plug in the power to the USRP. Run LabVIEW and open the transmitter that you created in the prelab.

2. Ensure that the transmitter is set up to use

   Carrier Frequency: 915.0 MHz

   IQ Rate: 400 kHz. Note: This sets the value of $1/T_x$.

   Gain: 0 dB.

   Active Antenna: TX1

   Symbol rate: 10,000 symbols/s

   Message Length: 1000 bits

   Pulse shaping filter: None

   Peak frequency deviation: 5000 Hz

   Run the transmitter. Use the large STOP button on the front panel to stop transmission connectors.

3. After running the transmitter, observe the spectrum of the transmitted signal. The two significant features are the bandwidth and the rate of spectral rolloff. Measure the null-to-null bandwidth of the transmitted signal. Compare the measured bandwidth to the bandwidth predicted using Carson's rule, Eq. (1). The rate of spectral rolloff is a measure of the interference that your signal will cause to signals using nearby carrier frequencies. Set the vertical scale of your spectrum plot to the range -100 dB to 0 dB, and print a copy of the spectrum for comparison in Step 4 below.

4. The rate of spectral rolloff of an FSK signal is determined primarily by the smoothness of the transmitted signal. Continuous-phase FSK has no discontinuities when the frequency of the transmitted signal changes, but there can be "corners" where the slope of the transmitted signal changes abruptly. To smooth out these corners, the message $m(t)$ can be filtered before it is passed to the FM modulator. Change the pulse-shaping filter to "Gaussian" to create a very smooth pulse transition. Run the transmitter and observe the smoothed message signal and the power spectrum. Measure the bandwidth of the transmitted signal 60 dB below the spectral peak. Set the vertical scale of your spectrum plot to the range -100 dB to 0 dB and print a copy of the spectrum. Compare the rolloff rate with that of the spectrum you obtained using rectangular pulses.

5. Change the pulse-shaping filter back to "none." Set the peak frequency deviation to 20,000 Hz, 10,000 Hz, 5000 Hz, and 2500 Hz. In each case, measure the null-to-null bandwidth of the transmitted signal and compare to the values predicted by Carson's rule.

   For the given symbol rate, 2500 Hz is the "minimum" frequency deviation. Below this deviation, performance will suffer, as the mark and space signals are not sufficiently different.

6. Ensure that the receiver is set up to use

   Carrier Frequency: 915.0 MHz

   IQ Rate: 400 kHz. Note: This sets the value of $1/T_z$ .

   Gain: 0 dB

   Active Antenna: RX2

   Symbol rate: 10,000 symbols/s

   Message Length: 1000 bits

   Pulse shaping filter: None

7. Run the transmitter, then run the receiver. Once the receiver has acquired a block of data, you may stop the transmitter.

8. Use the horizontal zoom feature on the Baseband Output graph palette to expand the demodulated waveform so that you can see individual pulses. Ideally, rectangular pulses passed through the receiver's matched filter should produce triangular output pulses. Note whether the demodulated pulses have the expected shape. (You can also right click a graph and choose Capture Data then analyze the data from the Data pane)

9. Observe the eye diagram. Make note of the optimum sampling time and the presence of intersymbol interference. To see the effect of pulse synchronization, move the waveform input of *MT Format Eye Diagram* to the "aligned waveform" output of *Pulse_align(real)*. Run the transmitter and receiver again. Observe the eye diagram. What is the optimum sampling time now?

10. Set the transmitter's peak frequency deviation to 20,000 Hz, 10,000 Hz, 5000 Hz, and 2500 Hz. For each case, run the transmitter and then the receiver. Record the peak value of the baseband output.

11. Return the peak frequency deviation to 5000 Hz. Change the "pulse shaping filter" control at both the transmitter and the receiver to "Gaussian." In the Gaussian setting, the receiver filter is a wire, i.e., no filter at all. Run the transmitter and then run the receiver. Once the receiver has acquired a block of data, you may stop the transmitter. Observe the baseband output signal and the eye diagram.

## Questions

1. With the peak frequency deviation set at 5000 Hz, compare the rate of spectral rolloff with rectangular pulses and with Gaussian filtering. Also, by examining the plot of the transmitted message signal, can you find evidence of intersymbol interference when Gaussian filtering is used?

2. For each value of peak frequency deviation listed in Step 5, compare the null-to-null bandwidth of the transmitted signal with the bandwidth predicted by Carson's rule, Eq. (1). Compute the

percentage difference in each case. Is Carson's rule more accurate for large peak frequency deviation or for small peak frequency deviation?

3. Observe the eye diagram shown on the receiver front panel. Compare the display when the eye diagram shows the baseband output waveform and when the eye diagram shows the aligned baseband output waveform. Describe what function *PulseAlign(real)* is performing.

4. Make a plot of the amplitude of the receiver's baseband output waveform vs. the peak frequency deviation. What is the relationship between these two quantities?

5. As described in Lab 7: Amplitude-Shift Keying, one of the quantities easily seen on the eye diagram is the optimum decision threshold location. In the present lab project, you were instructed to set the decision threshold to zero. Run the transmitter and receiver several times for different values of peak frequency deviation and for filtering set to "none" and "Gaussian." Observe the eye diagram and comment on the appropriateness of using zero for the decision threshold.

## Optional

To verify the correctness of the received bit sequence, you can add *AddFrameHeader(real)* to the transmitter and add *FrameSync(real)* and *MT Calculate BER* to the receiver. Follow the instructions given in Lab 7: Amplitude-Shift Keying. The *AddFrameHeader(real)* and *FrameSync(real)* can be found in the *BasicUSRPLabs folder*.

# 8.5 Report

## Prelab

Hand in documentation for the programs you created for the transmitter and receiver. Also include documentation for any functions you created. To obtain documentation, print out legible screenshots of the front panel and block diagram.

## Lab

Submit the programs you created for the transmitter and receiver. Also submit any functions you created. Be sure your files adhere to the naming convention described in the instructions above.

Resubmit documentation for any functions you modified during the lab.

Answer all of the questions in each of the sections marked *Questions* above.

LAB 9

# Binary Phase-Shift Keying (BPSK)

**Prerequisites:** Lab 5 – Double-Sideband Suppressed Carrier, Lab 7 – Amplitude-Shift Keying

# 9.1 Objective

In phase-shift keying (PSK), information is encoded on the phase of the transmitted carrier, rather than on its amplitude (ASK) or its frequency (FSK). In binary phase-shift keying (BPSK) there are two phase values, $0°$ and $180°$, which means that an unmodified carrier is transmitted to represent one binary data value, while an inverted carrier is transmitted to represent the other binary data value.

BPSK is the digital version of double-sideband suppressed-carrier analog modulation. We will see that the BPSK transmitter and receiver have a DSB-SC transmitter and receiver at their core. The additional features such as symbol mapping, pulse shaping, matched filtering, threshold detection, and pulse and frame synchronization that are needed in ASK and FSK systems are also needed in PSK systems. Just as in DSB-SC, the absence of a transmitted carrier leads to a reduction in power needed for a given level of performance. BPSK is optimum among binary systems in providing the lowest average power needed for a given bit error rate. BPSK is easily extended to quadrature phase-shift keying (QPSK) and quadrature amplitude modulation (QAM), as we shall see in a subsequent lab exercise. These extensions allow transmission of multiple bits per pulse and can be very effective for transmitting data at high rates over a channel of limited bandwidth.

Demodulating a BPSK signal requires synchronizing the phase of the received signal. Phase synchronization was also encountered in the DSB-SC lab project. We will see that the difficulties of phase synchronization can be avoided, at a small performance penalty, by using differential encoding of the transmitted data. Differential binary phase-shift keying (DPSK) is a robust and efficient modulation method that is widely used in practice.

# 9.2 Background

The generation and detection of an analog DSB-SC signal is discussed at length in the background section of *Lab 7: Double-Sideband Suppressed-Carrier*. It is strongly suggested that this material be reviewed at this point.

## Transmitter

A binary phase-shift keyed signal is a train of pulses, each of the form

$$Ag_{TX}(t)\cos(2\pi f_c t+\theta). \tag{1}$$

In Eq. (1), $A$ is a constant that sets the transmitted power level, $g_{TX}(t)$ is a fixed pulse shape, $f_c$ is the carrier frequency, and $\theta$ takes a value of either $0°$ or $180°$ to carry the desired information. Note that we can also write Eq. (1) as

$$\pm Ag_{TX}(t)\cos(2\pi f_c t), \tag{2}$$

where the plus sign corresponds to $\theta = 0°$ and the minus sign to $\theta = 180°$. We will assume, as in the previous digital signaling lab projects, that a new pulse is transmitted every $T$ seconds, so that the symbol rate is $1/T$ symbols/s. For a binary scheme such as BPSK, the bit rate is the same as the symbol rate. Since the pulse $g_{TX}(t)$ does not carry information, its shape can be chosen to satisfy other criteria. As was the case with ASK, we desire a pulse shape that provides a rapid spectral rolloff and minimizes intersymbol interference. We will use the "root-raised-cosine" pulse shape, as we did in the ASK lab.

The steps needed to form a BPSK signal should be familiar if you have completed the ASK lab project:

1. Symbol Mapping. The input data arrives as a stream of bits. Recall that the *MT Generate Bits* function produces an array of bytes containing the numbers 1 and 0. In the symbol mapping step, the bits are replaced by numerical values. For BPSK we will represent a binary 1 by the complex double $1+j0$ and a binary 0 by the complex double $-1+j0$. Note that we are representing bits by complex numbers, even though the imaginary parts are zero. This is

because the USRP requires a complex-valued input, and because in a future lab we will use the imaginary part to carry additional data. Table 1 shows the BPSK symbol mapping.

**Table 1. BPSK Symbol Mapping**

| Bit Value | Symbol |
|---|---|
| 0 | $-1+j0$ |
| 1 | $1+j0$ |

2. Upsampling. As a first step toward replacing symbols with pulses, we will place $L-1$ zeros after each symbol. This produces a sample interval of

$$T_x = \frac{T}{L}, \tag{3}$$

or a sample rate of

$$\frac{1}{T_x} = L\frac{1}{T}. \tag{4}$$

A higher upsampling factor $L$ makes the D/A conversion in the transmitter easier, but requires faster digital processing.

3. Pulse Shaping. If the upsampled signal is applied to a filter whose impulse response $g_{TX}[n]$ is a root-raised-cosine pulse, then each symbol at the filter output will be represented by a root-raised-cosine pulse. Steps 1 through 3 convert the input bit stream to a polar message signal. Note that pulse shapes other than root-raised-cosine can be used simply by changing the shape of the impulse response $g_{TX}[n]$. The root-raised-cosine pulse has a very rapid spectral rolloff, so that the transmitted signal will not cause interference to signals at nearby carrier frequencies.

4. Modulation. The polar message signal, consisting of a train of pulses of the form $(\pm 1 + j0)\, g_{TX}[n]$, can be sent directly to the USRP transmitter. The USRP will convert the signal to continuous time and add the carrier as shown in Eq. (2).

## Receiver

A PSK receiver begins with a DSB-SC demodulator. When the transmitted BPSK signal arrives at the receiver it has the form of a train of pulses, each given by

$$r(t) = \pm D g_{TX}(t) \cos(2\pi f_c t + \phi), \tag{5}$$

where $D$ is a constant (usually much smaller than the constant $A$ in the transmitted signal) and the angle $\phi$ represents the difference in phase between the transmitter and receiver carrier oscillators. If the receiver's carrier oscillator is set to the same frequency as the transmitter's carrier oscillator, the USRP receiver will do most of the work in demodulating the BPSK signal. The receiver's *Fetch Rx Data* will provide a train of output pulses, each given by

$$\tilde{r}[n] = \pm \frac{D}{2} g_{TX}[n] e^{j\phi}. \tag{6}$$

The sampling rate in Eq. (6), $1/T_z$, is set by the receiver's "IQ rate" parameter. This rate is set to provide $M$ samples every $T$ seconds, where $1/T$ is the symbol rate.

We can remove the phase offset $\phi$ using any one of a variety of techniques. The technique used in the DSB-SC lab project works well. To summarize, we begin by squaring $\tilde{r}[n]$, giving

$$\tilde{r}^2[n] = \frac{D^2}{4} g_{TX}^2[n] e^{j2\phi}. \tag{7}$$

This step eliminates phase changes caused by the data, as well as phase changes caused by changes in the polarity of $g_{TX}[n]$. Next, the angle $2\phi$ can be extracted using a *Complex to Polar function* from the Data Types→Numeric→Complex palette. It turns out to be helpful at this point to smooth variations in $2\phi$ caused by noise. The *median filter* from the ExternalFiles folder does a good job. The default values can be accepted for the "left rank" and "right rank" parameters. Next the *Unwrap Phase* from the Analysis→Signal Processing→Signal Operations palette will remove

jumps of $\pm 2\pi$. Finally, dividing by two gives the desired estimate of the phase error $\phi$. The block diagram in Figure 1 shows the entire phase synchronization process.

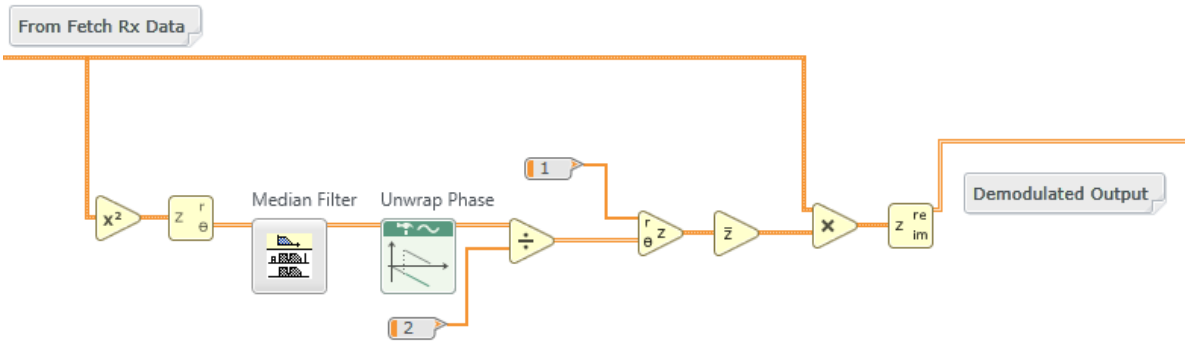



**Figure 1. Carrier Phase Synchronization**

The remaining steps carried out by the receiver should be familiar from the ASK and FSK labs. These steps are:

1. Matched Filtering. We will use a root-raised-cosine receiver filter. This filter's impulse response $g_{RX}[n]$ is matched to the pulse shape $g_{TX}[n]$ of the received pulses. The matched filter gives optimum performance in the presence of additive, white, Gaussian noise. Further, the cascade of the two root-raised-cosine filters $g_{TX}[n]$ and $g_{RX}[n]$ produce a raised-cosine pulse shape that is free from intersymbol interference.

2. Pulse Synchronization. The matched filter output is an analog baseband signal that must be sampled once per symbol time, i.e. once every $T$ seconds. Because of filtering, propagation delays, and distortion caused by the communication channel, it is necessary to determine the optimum time to take these samples. A function called *PulseAlign(real)* has been provided to align the baseband signal so that the sample at index 0 is the correct first sample.

3. Sampling. The *Decimate* function will sample the aligned baseband waveform at index 0 and every $T$ seconds thereafter.

4. Detection. Once the baseband waveform has been sampled, each sample must be examined to determine whether it represents a symbol of value 1 or a symbol of value 0.

5. Symbol Mapping. The detected symbol values must be converted to bits. For binary PSK, this step is easily included in the detection step.

# 9.3 Pre-Lab

## Transmitter

1. Create a program to generate a BPSK signal using the USRP. A template for the transmitter has been provided in the file *BPSKTxTemplate.gvi*. This template contains the four functions for interfacing with the USRP along with *MT Generate Bits* from the Modulation Toolkit. *MT Generate Bits* will create a pseudorandom sequence of bits that can serve as a data sequence for testing your BPSK system. Note that by default, *MT Generate Bits* will produce the same sequence of bits every time you run the program. This is useful for debugging, but if you would like to generate a different sequence of bits every time, wire a random number to the "seed in" input. The steps below contain details about how to create the required transmitter.

2. First do the symbol mapping, as shown in Table 1.

3. Upsample the array of symbols using *Upsample* from the Analysis→Signal Processing→Signal Operation subpalette. In this lab project you are given control inputs to set the symbol rate $1/T$ and the IQ rate $1/T_x$. Set the symbol rate to 10,000 symbols/s and the IQ rate to $200 \times 10^3$ Sa/s. Use the symbol rate and coerced IQ rate to calculate the upsampling factor $L$.

4. Use *MT Generate Filter Coefficients* from the Modulation Toolkit to generate the pulse shaping filter. (*MT Generate Filter Coefficients* can be found on the Analysis→

Communications→Digital→Utilities subpalette.) Set the modulation type to PSK, and the pulse-shaping filter "samples per symbol" to your calculated value of $L$. Create a front-panel control for "pulse shaping filter" and set this to "Root Raised." Wire the "pulse shaping filter coefficients" output to the "Y" input of a *Convolution*. The *Convolution* is available from the Analysis→Signal Processing→Signal Operation subpalette. Wire the output from your upsampler to the "X" input of the *Convolution*.

5. Normalize the amplitude of your filtered message signal to a maximum absolute value of 1. The *Quick Scale 1D function* in the ExternalFiles folder will find the maximum of the absolute value. To ensure that your scaled message remains complex-valued, use a separate division function to do the actual scaling. Connect the scaled message to the *Build Waveform* function that connects to the Baseband Waveform graph provided in the template. Also send the scaled message to the *Write Tx Data* function.

6. To observe the spectrum of the transmitted signal, wire the complex baseband waveform to the *FFT Power Spectrum function* that connects to the Power Spectrum graph provided in the template.

This completes construction of the BPSK transmitter. Save your transmitter in a file whose name includes the letters "BPSKTx" and your initials (e.g. *BPSKTx_BAB.gvi*).

## Receiver

1. Create a program to implement a BPSK receiver using the USRP. A template for the receiver has been provided in the file *BPSKRxTemplate.gvi*. This template contains the six interface functions for interfacing with the USRP.

   Calculate the "number of samples" for the *Fetch Rx Data* function to fetch using the message length, symbol rate front panel inputs and the coerced IQ rate. Double the number of samples the receiver will fetch to acquire two frames of data. Since the receiver's starting point is random, this ensures that there will be one complete frame of received data in the block of samples fetched.

Implement the carrier phase synchronization as shown in Figure 1.

2. To implement the receiver's matched filter, use *MT Generate Filter Coefficients* just as you did for the transmitter. Set the modulation type to PSK, and calculate the "matched samples per symbol" $M$ from the "actual IQ rate ($1/T_z$)" and the symbol rate ($1/T$) obtained from the front-panel control. Create a front-panel control for "pulse shaping filter" and set this to "Root Raised." Wire the "matched filter coefficients" output to the "Y" input of a *Convolution*. The output of your pulse shaping filter should be connected to the *Cluster Properties function* provided in the template. The *Cluster Properties function* feeds the Baseband Output graph.

3. Place the *PulseAlign (real)* function from the ExternalFiles folder on your block diagram and wire the baseband output waveform to the "input waveform" input and wire the $M$ samples/symbol to the "receiver sampling factor" input.

   Once the baseband waveform is aligned, it can be sampled. The *Decimate (single shot)* function can be obtained from the Analysis→Signal Processing→Signal Operation subpalette. The "decimating factor" is $M$.

4. The Modulation Toolkit function *MT Format Eye Diagram* has been provided in the receiver template. Wire the baseband output waveform to the "waveform" eye-diagram input. The "symbol rate (Hz)" input value is available from the front panel control. Set the "eye length" parameter to 2.

5. To determine whether each received sample is more likely to represent a 1 or a 0, the sample must be compared with a threshold. Because the message is a polar signal, the threshold can be taken as zero. The result of this comparison is the receiver's digital output. The output of the comparison will be a Boolean array. You can convert this array to an integer array by using a *Boolean To Integer function*.

This completes construction of the BPSK receiver. Save your receiver in a file whose name includes the letters "BPSKRx" and your initials (e.g. *BPSKRx_BAB.gvi*).

## Questions

1. In *Transmitter* Step 3 you are given $1/T = 10{,}000 \text{ symbols/s}$ and $1/T_x = 200 \times 10^3$ Sa/s. Find the corresponding value for the number of samples per symbol $L$.

2. Eq. (6) in the Background section above shows that the received baseband signal $\tilde{r}[n]$ includes a factor $e^{j\phi}$, where $\phi$ is any phase difference that may exist between the transmitter and receiver carrier oscillators. Explain what would happen if you omitted the phase synchronization step in the receiver. Specifically, what would be the receiver output if $\phi$ just happened to take the value $\pi/2$?

3. In *Receiver* Step 2 the "actual IQ rate" $1/T_z$ may be different from the rate $1/T_x$ that was used at the transmitter. ( Note that the symbol rate $1/T$ must be the same at the transmitter and receiver.) The value of the receiver's IQ rate determines the receiver sampling factor $M$. What is the advantage to using a higher value of $M$? What is the advantage of using a lower value of $M$?

4. Although BPSK is a suppressed-carrier version of ASK, we do not use an envelope detector to demodulate BPSK the way we did for ASK. Why is that? What would the receiver output be if we used an envelope detector for demodulation?

5. Bit error rate is a measure of performance of a digital communication system. What would the bit error rate be if the transmitter failed and the receiver received only noise? Explain your reasoning.

# 9.4 Lab Procedure

1. Connect a loopback cable and attenuator between the TX 1 and RX 2 connectors of the USRP. Connect the USRP to your computer and plug in the power to the USRP. Run LabVIEW and open the transmitter that you created in the prelab.

2. Ensure that the transmitter is set up to use

   Carrier Frequency: 915.0 MHz

   IQ Rate: 200 kHz. Note: This sets the value of $1/T_x$ .

   Gain: 0 dB.

   Active Antenna: TX1

   Symbol rate: 10,000 symbols/s

   Message Length: 1000 bits

   Pulse shaping filter: Root Raised

   Run the transmitter. Use the large STOP button on the front panel to stop transmission connectors.

3. After running the transmitter, observe the spectrum of the transmitted signal. Measure the "main lobe" bandwidth of the transmitted signal. Change the pulse shaping filter control to "none" to create rectangular pulses and run the transmitter again. Compare the spectrum of the transmitted signal with the spectrum for root-raised-cosine pulses. Return the pulse shaping filter control setting to "Root Raised."

4. Ensure that the receiver is set up to use

   Carrier Frequency: 915.0 MHz

   IQ Rate: 200 kHz. Note: This sets the value of $1/T_z$ .

   Gain: 0 dB

   Active Antenna: RX2

   Symbol rate: 10,000 symbols/s

   Message Length: 1000 bits

   Pulse shaping filter: Root Raised

5. Run the transmitter, then run the receiver. Once the receiver has acquired a block of data, you may stop the transmitter.

6. Observe the eye diagram. Make note of the optimum sampling time and the presence or absence of intersymbol interference. To see the effect of pulse synchronization, move the waveform input of *MT Format Eye Diagram* to the "aligned waveform" output of *PulseAlign(real)*. Run the transmitter and receiver again. Observe the eye diagram. What is the optimum sampling time now?

7. Modify the transmitter to include the *AddFrameHeader(real)*, available in the *BasicUSRPLabs folder*. Place *AddFrameHeader(real)* after the symbol mapping, but before conversion of the symbols to complex. Next, modify the receiver to include the *FrameSync(real)*, also available from the *BasicUSRPLabs folder*. Place *FrameSync(real)* immediately following *Decimate*. Wire the output of *Decimate* to the "Sampled Input" of *FrameSync(real)*. Leave the remaining inputs of *FrameSync(real)* unwired. Wire the "Aligned Samples" output of *FrameSync(real)* to the threshold comparison function.

   Wire the array of output bits from the threshold comparison to the "array" input of an *Array Subset function*. Set the "index" input to zero, and set the "length" input to the length specified by the "message length" control. Display the output of *Array Subset function* as "Output bits" on the receiver front panel.

   Note that the "Output Signal" and "max index" outputs of *FrameSync(real)* will not be used in this lab project.

8. Automate measurement of the bit error rate (BER) by using the *MT Calculate BER* from the Modulation Toolkit (Analysis→ Communications→Digital→Measurements subpalette). From the Configure ribbon, choose "PN Fibonacci." Set the "BER trigger threshold" to 0.4. Connect indicators to the "BER" and "trigger found?" outputs. When you run the program, "trigger found?" will be true whenever the measured BER is below the BER trigger threshold.

   Run the transmitter and receiver. If everything is working correctly, the BER should be 0 or 1.0. Run the transmitter and then run the receiver a dozen times or so. Verify that about half the time the BER is 0 and about half the time the BER is 1.0.

## Questions

1. How do the main lobe bandwidth and spectral rolloff rate for root-raised cosine pulses compare with the same quantities when rectangular pulses are used?

2. What does a BER of 1.0 signify? Explain why the BER is 0 half the time and 1.0 half the time.

## Differential Encoding

Suppose the data bits are passed through the logic circuit shown in Figure 2 before being sent to the transmitter. In this circuit, the exclusive-nor gate will produce a 0 whenever the current input differs from the previous output, and a 1 whenever the current input is the same as the previous output.

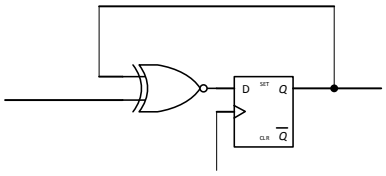



**Figure 2. Differential Encoder**

For example, an input of

$$1 \quad 0 \quad 1 \quad 1 \quad 0 \quad 0 \quad 0 \quad 1 \quad \ldots$$

produces an output of

$$0 \quad 0 \quad 1 \quad 1 \quad 1 \quad 0 \quad 1 \quad 0 \quad 0 \quad \ldots,$$

assuming that the flip-flop has an initial state of 0. Now suppose that the received bit sequence is passed through the circuit shown in Figure 3. The output will be a 1 whenever the current

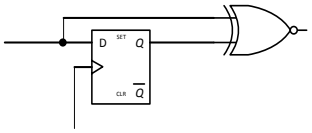



**Figure 3. Differential Decoder**

input and the previous input are the same, and the output will be a 0 whenever the current input and the previous input are different. For example, a received bit sequence of

$$0 \quad 0 \quad 1 \quad 1 \quad 1 \quad 0 \quad 1 \quad 0 \quad 0 \quad \ldots$$

produces an output of

$$1 \quad 0 \quad 1 \quad 1 \quad 0 \quad 0 \quad 0 \quad 1 \quad \ldots.$$

We see that the decoder of Figure 3 reverses the effect of the encoder of Figure 2. The important part of this is to notice what happens if the received bit pattern is inverted. If the received bit sequence is

$$1 \quad 1 \quad 0 \quad 0 \quad 0 \quad 1 \quad 0 \quad 1 \quad 1 \quad \ldots,$$

"same" and "different" are not altered, and so the decoder output will still be

$$1 \quad 0 \quad 1 \quad 1 \quad 0 \quad 0 \quad 0 \quad 1 \quad \ldots.$$

9. Add a differential encoder to your BPSK transmitter and a differential decoder to your receiver. Place the encoder immediately after *MT Generate Bits*, before the symbol mapping. Place the decoder after the threshold. Note that your transmitted sequence will have to be made one bit longer to include the initial state of the encoder flip-flop.

   Hint: The D flip-flop creates a one-sample delay. The delay is easily implemented in LabVIEW using a *Feedback Node function* from the Programming Flow palette.

10. Run the transmitter and then run the receiver a dozen times or so. You should find that the BER is always zero.

## Differential Encoding, Part 2

Differential encoding can be done after the symbol mapping.

**Table 2** is a truth table showing the correspondence between the exclusive-nor operation and ordinary multiplication. We will see below that the differential encoding allows the phase synchronizer to be eliminated from the receiver circuit.

**Table 2. Exclusive Nor and Numerical Multiplication**

| **Boolean** | | | **Numerical** | |
|---|---|---|---|---|
| Input | Output | | Input | Output |

| | | | | | | |
|---|---|---|---|---|---|---|
| 0 | 0 | 1 | | –1 | –1 | 1 |
| 0 | 1 | 0 | | –1 | 1 | –1 |
| 1 | 0 | 0 | | 1 | –1 | –1 |
| 1 | 1 | 1 | | 1 | 1 | 1 |

11. Implement the differential encoder numerically. Place the encoder in your transmitter following *AddFrameHeader(real)*. (This time the frame header also gets encoded.) Do not forget to add the initial state of the encoder to the beginning of the transmitted sequence.

    Save your transmitter in a file whose name includes the letters "DPSKTx" and your initials (e.g. *DPSKTx_BAB.gvi*).

12. In the receiver, remove the phase synchronizer and run the received data directly to the *Convolution* that implements the matched filter. Place the differential decoder immediately after *Decimate*, the receiver's sampler, and before *FrameSync(real)*. Since the received data are complex-valued at this point, design your decoder to form the product of the current sample and the *complex conjugate* of the previous sample. Then take the real part of the result. You may also need to replace *PulseAlign(real)* with *PulseAlign(Complex).*

    Save your receiver in a file whose name includes the letters "DPSKRx" and your initials (e.g. *DPSKRx_BAB.gvi*).

12. Run the transmitter and then run the receiver several times. Verify that the BER is always zero.

## Questions

1. Show that the differential encoder of Figure 2 and the differential decoder of Figure 3 continue to work properly as a system if the encoder's flip-flop has an initial state of 1.

2. Show, starting with Eq. (6), why the phase synchronizer is not needed in the DPSK receiver.

3. In the BPSK system, the eye diagram has the same amplitude every time you run the receiver. In the DPSK system, the amplitude of the eye diagram changes on every run. Explain why this happens. Explain whether, if noise were present, the BER would also change every time the DPSK receiver is run.

4. Differential phase-shift keying is intended to be robust in the presence of a phase difference between the transmitter and receiver oscillators. It is also possible for there to be a frequency difference between the oscillators. Suppose there is a small frequency difference of $\Delta f$ between the transmitter and receiver oscillators. Modify Eq. (6) to take this frequency difference into account. How will the output of the DPSK receiver change because of this frequency difference? Using the parameter values of this lab project, determine how large a frequency difference will be "significant." Be sure to specify what you mean by "significant."

# 9.5 Report

## Prelab

Hand in documentation for the programs you created for the transmitter and receiver. Also include documentation for any functions you created. To obtain documentation, print out legible screenshots of the front panel and block diagram.

Answer all of the questions in the Prelab section marked *Questions*.

## Lab

Submit the program you created for the transmitter and receiver. Include both the BPSK and DPSK programs. Also submit any functions you created. Be sure your files adhere to the naming convention described in the instructions above.

Submit documentation for the *DPSKTx* and *DPSKRx* programs. Resubmit documentation for any functions you modified during the lab.

Answer all of the questions in each of the Lab Procedure sections marked *Questions* above.

LAB 10

# The Eye Diagram

**Prerequisite:** Lab 9 – Binary Phase-Shift Keying (BPSK)

# 10.1 Objective

Although we have displayed an eye diagram in each of the digital labs so far, we have relied on the Modulation Toolkit to produce the display. In this lab project we will learn how to create an eye diagram from scratch. We will examine how the eye diagram changes when intersymbol interference (ISI) is present in the communication channel, and we will learn how to make quantitative measurements of the amount of ISI from the eye diagram.

In this lab project you will use the BPSK transmitter and receiver that you created for Lab 9 as a "test bed" system. To create ISI, a function called *Channel.gvi* has been provided. If a second USRP is available, however, you can try transmitting from one USRP to the other using antennas to see how much ISI is actually present owing to multipath propagation in your own physical environment.

# 10.2 Background

Viewed as a complex baseband signal, the waveform generated by the BPSK transmitter can be written

$$\tilde{x}(t) = A\sum_{n=-\infty}^{\infty} a_n g_{TX}(t - nT), \tag{1}$$

where $A$ is a constant that sets the transmitted average power, $a_n = \pm 1$ are the symbol values carrying the transmitted information, $g_{TX}(t)$ is the transmitted pulse shape, and $T$ is the time between symbols. The tilde over the $x$ signifies that this is the baseband signal; the actual transmitted signal is

$$\begin{aligned} x(t) &= \mathrm{Re}\left[\tilde{x}(t)e^{j2\pi f_c t}\right] \\ &= A\sum_{n=-\infty}^{\infty} a_n g_{TX}(t - nT)\cos(2\pi f_c t). \end{aligned} \tag{2}$$

Note that the pulse $g_{TX}(t)$ is assumed to be real valued.[9]

The transmitted signal passes through the communication channel and then is demodulated by the receiver. Let us model the communication channel as a linear time-invariant filter having baseband impulse response $h(t)$. When the signal arrives at the receiver and is demodulated, it is passed through the receiver filter having impulse response $g_{RX}(t)$. Let us designate the impulse response of the cascade of the transmitter filter, the channel, and the receiver filter as $g(t)$. That is,

$$g(t) = g_{TX}(t) * h(t) * g_{RX}(t). \tag{3}$$

The cascade of $g_{TX}(t)$ and $g_{RX}(t)$ is usually designed to be free of ISI. In this lab project we will use a cascade in which the combined filter has a raised-cosine spectral shape for that purpose. Unfortunately, the channel response $h(t)$ is not under the designer's control, and may contribute ISI to the received signal. After filtering at the receiver, the received signal can be written

$$\tilde{y}(t) = D\sum_{n=-\infty}^{\infty} a_k g(t - nT) + n(t), \tag{4}$$

where $D$ is an amplitude constant, $g(t)$ is the pulse shape given by Eq. (3), and $n(t)$ is white Gaussian noise that has been passed through the receiver filter.

Once the received signal has been filtered, it is sampled, at the rate of one sample every $T$ seconds. Assuming that pulse synchronization has been applied, so that the samples are taken at the correct times, the sampled signal is

$$\tilde{y}(kT) = D\sum_{n=-\infty}^{\infty} a_n g\left[(k-n)T\right] + n(kT). \tag{5}$$

These samples are compared with a threshold of zero to decide whether each received symbol is most likely to represent a $+1$ or a $-1$. Let us rewrite Eq. (5) as

[9] The signal $\tilde{x}(t)$ and the impulse responses $g_{TX}(t)$ and $g_{RX}(t)$ are all actually implemented in discrete time. Writing these as continuous time functions makes the discussion much easier to read, however.

$$\tilde{y}(kT) = Da_k g(0) + D\sum_{\substack{n=-\infty \\ n\neq k}}^{\infty} a_n g\left[(k-n)T\right] + n(kT). \tag{6}$$

The first term in Eq. (6) represents the desired sample at time $t = kT$, the second term represents ISI, and the third term represents noise.

To get a sense of how much ISI a given $g(t)$ might produce in the worst case, let us suppose that $a_k = 1$ and $a_n, n \neq k$, forms a pattern that makes every ISI term negative. That is, suppose the data pattern is

$$a_n = \begin{cases} -1, & g\left[(k-n)T\right] \geq 0 \\ 1, & g\left[(k-n)T\right] < 0. \end{cases} \tag{7}$$

Then Eq. (6) becomes

$$\begin{aligned} \tilde{y}(kT) &= Dg(0) - D\sum_{\substack{n=-\infty \\ n\neq k}}^{\infty} \left| g\left[(k-n)T\right] \right| + n(kT) \\ &= Dg(0) - D\sum_{n\neq 0} \left| g(nT) \right| + n(kT). \end{aligned} \tag{8}$$

Ignoring noise, the sample given by Eq. (8) represents a "worst case" value. We write

$$V_w = Dg(0) - D\sum_{n\neq 0} \left| g(nT) \right|. \tag{9}$$

We can also identify a "best case" sample value. If every ISI term enhances the desired sample we have

$$V_b = Dg(0) + D\sum_{n\neq 0} \left| g(nT) \right|. \tag{10}$$

Traditionally, the quantity of ISI present at the output of the receiver filter is measured by either of two parameters. The *peak distortion* is defined as

$$\text{peak distortion} = \frac{D\sum_{n\neq 0} \left| g(nT) \right|}{Dg(0)}. \tag{11}$$

Alternatively, the *eye opening* is defined as

$$\text{eye opening} = 1 - \text{peak distortion}. \tag{12}$$

An example of an eye diagram is shown in Figure 1.

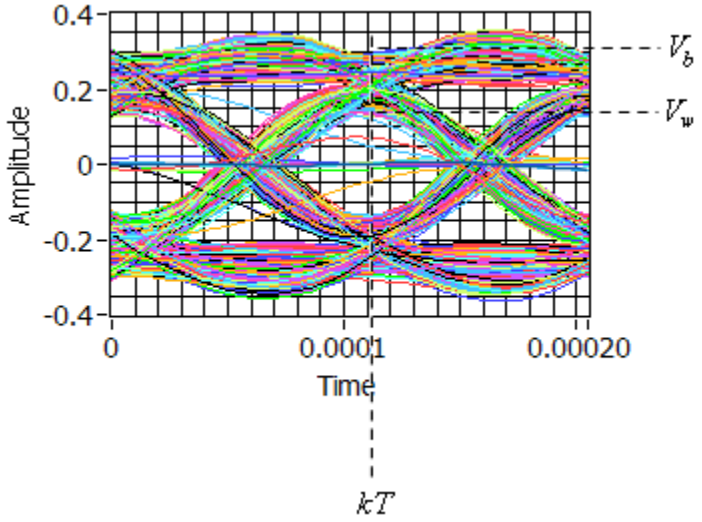



Figure 1. Eye Diagram

The sampling time $kT$ and the best-case and worst-case voltages, $V_b$ and $V_w$ respectively, are shown. Note that $V_b$ and $V_w$ must both be measured at the same time value, $kT$. As you can see from the figure, the quantities $D\sum_{n\neq 0}\left|g\left(nT\right)\right|$ and $Dg\left(0\right)$ needed to find the peak distortion or eye opening are not readily apparent. If we rearrange Eqs. (11) and (12), however, we can show that

$$\text{peak distortion} = \frac{V_b - V_w}{V_b + V_w} \tag{13}$$

and

$$\text{eye opening} = \frac{2V_w}{V_b + V_w}. \tag{14}$$

Thus, for example, we can measure $V_b$ and $V_w$ from the eye diagram, and use Eq. (14) to calculate the eye opening. Figure 1 shows $V_b = 0.296$ and $V_w = 0.124$, giving an eye opening of 0.59, or about 60%.

Intersymbol interference is of concern because it degrades the performance of the communication system. For a BPSK system using a matched receiver filter with additive white Gaussian noise and no ISI, the probability of error is given by

$$P_e = Q\left(\sqrt{\frac{2E_b}{N_0}}\right), \tag{15}$$

where $E_b$ is the average energy per bit at the receiver, $N_0/2$ is the power spectrum of the noise at the receiver input, and $Q$ is the Gaussian probability integral. If ISI is present, the probability of error is dominated by the worst-case situation. For an eye opening $\eta$ the worst-case probability of error is given by

$$P_e = Q\left(\eta\sqrt{\frac{2E_b}{N_0}}\right), \tag{16}$$

where $E_b$ is the energy per bit without including ISI, as in Eq. (15).

# 10.3 Pre-Lab

## Eye Diagram

1. Modify your BPSK receiver to include your own eye diagram functionality. Here is how:

   The required eye diagram is a plot of the aligned waveform at the output of *PulseAlign(real)*. The horizontal (time) axis has a duration of two symbol times. The plots corresponding to successive pairs of symbols are overlapped. To make such a plot, extract the Y-array from the aligned waveform. Also extract $dt$ ; you will need it as described below. Then use the *Reshape Array function* from the Data Types→Array subpalette to form a two-dimensional array for which each row contains two symbols of data. In the *Reshape Array function*, the first "dimension size" input is the number of rows and the second "dimension size" input is the number of columns. The number of rows is the number of curves to plot, and the number of columns is the number of samples in two symbols of data. You will need to calculate these numbers in your program. The output of *Reshape Array function* is a two dimensional array, which must be connected to the "2D DBL to 1D Cluster" function before it is ready for plotting. Use your value of $dt$ , and the output of *Reshape Array function*. Connect the resulting cluster to a *Waveform Graph indicator*. When the *Waveform Graph indicator* is given a two dimensional

array to plot, it plots each row of the array as a separate curve. This is exactly the behavior that you want to produce the eye diagram.

2. Save your modified BPSK receiver in a file whose name includes the letters "BPSKEyeRx" and your initials (e.g. *BPSKEyeRx_BAB.gvi*).

## Channel Model

To test your eye diagram, you will need to introduce some ISI. *Channel.gvi*, in the *BasicUSRPLabs folder* is provided for the purpose. Add *Channel.gvi* immediately after *Fetch Rx Data* in your receiver. Connect the "Use Channel" and "Propagation Delay" inputs to front panel controls. Connect the "Samples per Symbol" input to the samples per symbol calculated from the actual IQ rate and the symbol rate in your receiver. You may leave the other inputs of *Channel.gvi* unconnected.

*Channel.gvi* simulates a multipath channel. The default channel parameters create an impulse response given by

$$h(t) = \delta(t) + 0.2\delta(t-\tau) - 0.08\delta(t-2\tau), \quad (17)$$

where $\tau$ is the "propagation delay" set by the front panel control. *Channel.gvi* can also add noise to the signal, but we will not be using this feature in this lab project.

Multipath propagation tends to create peaks and dips in the frequency response of the communication channel. This creates the distortion of the received signal that leads to intersymbol interference. Some authors use the so-called *coherence bandwidth* as a measure of the irregularity of the channel frequency response. Roughly, the coherence bandwidth is the frequency interval over which the gain of the channel remains approximately constant. Intersymbol interference will become serious when the signal bandwidth exceeds the coherence bandwidth of the channel. Coherence bandwidth is inversely related to the spread in propagation delays between the first-arriving and last-arriving signal. For the default channel of Eq. (17), the coherence bandwidth is inversely proportional to the parameter $\tau$. The relation is

$$B_{coh} = \frac{0.816}{\tau}, \quad (9.1)$$

where $B_{coh}$ is the coherence bandwidth in Hz when $\tau$ is in seconds.

## Questions

1. Use Eq. (15) to find $E_b/N_0$ for a probability of error of $P_e = 10^{-6}$. Now use Eq. (16) to find $P_e$ for an eye opening of $\eta = 0.8$. Repeat for $\eta = 0.5$. Using the same $E_b/N_0$, find the value of eye opening $\eta$ that will give $P_e = 10^{-5}$.

2. Suppose $g(0) = 1$, $g(T) = 0.2$, $g(2T) = -0.08$, and $g(nT) = 0$ otherwise. Use Eqs. (9) and (10) to find $V_w$ and $V_b$. Then use Eq. (14) to find the eye opening.

3. Using Eq. (18), find the coherence bandwidth for $\tau = 50\ \mu s$, $\tau = 100\ \mu s$, and $\tau = 150\ \mu s$.

# 10.4 Lab Procedure

1. Connect a loopback cable and attenuator between the TX 1 and RX 2 connectors of the USRP. Connect the USRP to your computer and plug in the power to the USRP. Run LabVIEW and open the transmitter that you created in the prelab.

2. Ensure that the transmitter is set up to use

   Carrier Frequency: 915.0 MHz

   IQ Rate: 200 kHz. Note: This sets the value of $1/T_x$.

   Gain: 0 dB

   Active Antenna: TX1

   Symbol rate: 10,000 symbols/s

   Message Length: 1000 bits

   Pulse shaping filter: Root Raised

   Run the transmitter. Use the large STOP button on the front panel to stop transmission connectors.

3. After running the transmitter, observe the spectrum of the transmitted signal. Measure the "main lobe" bandwidth of the transmitted signal. The baseband signal bandwidth is half of the main lobe bandwidth, counting only the positive-frequency components.

4. Ensure that the receiver is set up to use

   Carrier Frequency: 915.0 MHz

   IQ Rate: 200 kHz. Note: This sets the value of $1/T_z$. Note that $T_z$ is the same parameter as $dt$.

   Gain: 0 dB

   Active Antenna: RX2

   Symbol rate: 10,000 symbols/s

   Message Length: 1000 bits

   Pulse shaping filter: Root Raised

   Use Channel: off

5. Run the transmitter, then run the receiver. Once the receiver has acquired its data, you may stop the transmitter. The receiver should show a BER of 0.0 or 1.0. Do not be concerned about a BER of 1.0 in this lab project.

6. Observe the eye diagram created by the program you created in step 1 of the prelab. Measure $V_w$ and $V_b$ and calculate the eye opening. Owing to filtering in the USRP, the eye opening should not be 100%.

   The most effective way to measure $V_w$ and $V_b$ from the eye diagram is to use cursors. Click on the eye diagram graph and in the ribbon, click the Graph Parts button. In the dialog window, click the Cursor Legend enable icon. To create a new cursor, click the New Cursor button in the Cursors dialog next to the Eye diagram.

   To measure $V_w$ and $V_b$, place the X cursor at a time at which the eye opening is widest. This will be near the center of the time axis. Then use the Y cursor to measure $V_w$ and $V_b$. Do not move the X position between measurements. You will get a more accurate pair of measurements if you zoom in on the relevant part of the eye diagram before positioning the Y cursor. (Graph Parts button, Graph Tools)

7. Set the propagation delay $\tau$ to $50\ \mu s$, set Use Channel to "on," and repeat steps 5 and 6. Repeat for $\tau = 100\ \mu s$ and $\tau = 150\ \mu s$. Prepare a table showing the eye opening for each value of $\tau$.

## Questions

1. Why in this lab project are we not concerned if the BER turns out to be 1.0?

2. Multipath propagation is common in the cellular telephone environment. Propagation delays can vary depending on the locations of the base station, the mobile unit, and nearby buildings, but these delays do not depend on symbol rate. However, with each new generation of cellular service the symbol rate increases. Based on your findings in this lab project, discuss the relationship between symbol rate, coherence bandwidth, eye opening, and BER. What happens as the symbol rate increases?

# 10.5 Report

## Prelab

Hand in documentation for your modified receiver including the eye diagram. Also include documentation for any functions you created. To obtain documentation, print out legible screenshots of the front panel and block diagram.

Answer all of the questions in the Prelab section marked *Questions*.

## Lab

Submit the modified receiver including the eye diagram. Resubmit documentation for any functions you modified during the lab. Submit the table required in Lab Procedure step 7 above. Answer the questions in the Lab Procedure section marked *Questions*.

LAB 11

# Equalization

**Prerequisite:** Lab 10 – The Eye Diagram

# 11.1 Objective

In most digital communication systems, the transmitter's pulse shaping filter and the receiver's matched filter are designed so that the pulses that emerge from the receiver's matched filter do not exhibit any intersymbol interference. As we discussed in the eye diagram lab project, however, the communication channel often introduces additional filtering that can result in intersymbol interference.

One way to deal with intersymbol interference created by the communication channel is to add an additional filter in the receiver. Ideally, this new filter will have a response that is "inverse" to the filtering caused by the channel. A filter intended to counter the adverse effects of channel filtering is called an *equalizer*.

In wireless systems, the channel distortion is often caused by multipath propagation. That is, the received signal may include reflections from buildings and other objects in the environment. The distortion that is observed in any particular wireless link will depend on the location of the transmitter, the location of the receiver, and the locations of nearby reflecting objects. Since the transmitter, the receiver, and even some of the reflecting object can move, channel distortion can change gradually with time. In this kind of application it is helpful to have an equalizer that is *adaptive*; that is, we would like the equalizer to be able to change its properties slowly with time to follow changes in the channel.

In this lab project you will use the BPSK transmitter and receiver that you created for Lab 9 as a "test bed" system. The *Channel.gvi* that you used in the eye diagram lab project will create the intersymbol interference. The equalizer that we will investigate is provided in the Modulation Toolkit.

# 11.2 Background

Equalizers, particularly adaptive equalizers, are nearly always implemented as FIR filters. There are two reasons for this. First, and most important, an FIR filter is always stable. As a result, we do not

have to be concerned that the equalizer might become unstable as it adapts. Second, it sometimes turns out that the "optimum" equalizer is not causal. A non-causal FIR filter can always be made causal by adding a finite delay. This easy fix cannot be applied to a non-causal IIR filter.

Experience shows that it is often possible to model the communication channel as an FIR filter as well. In a multipath environment successive reflections become weaker, and eventually drop below the noise floor. Commonly, only a few of the strongest reflections are significant causes of ISI. Unfortunately, the inverse of an FIR filter is always an IIR filter, and this means that an FIR equalizer will never be able to completely remove the ISI caused by an FIR channel. We will find that even a very long FIR equalizer always leaves a small amount of residual ISI.

Suppose that $\tilde{y}[k] = \tilde{y}(kT)$ represents the sampled output of the receiver's matched filter. The parameter $T$ represents the time between samples and also the time between transmitted symbols. Recall from the eye diagram lab project that each sample $\tilde{y}[k]$ is the sum of a desired sample value, ISI, and noise. The block diagram representation of an FIR equalizer is shown in Figure 1.

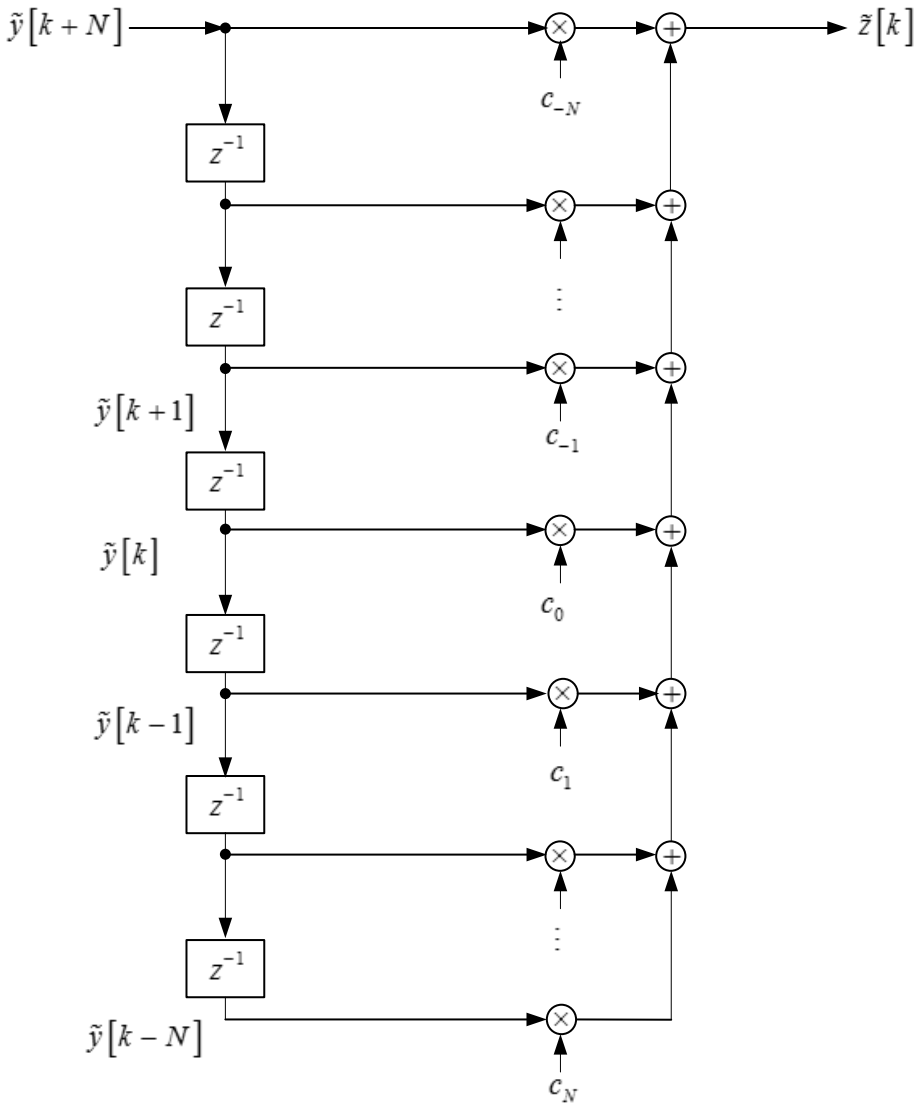



**Figure 1. An FIR Equalizer**

The output of the equalizer is denoted as $\tilde{z}[k]$. In the diagram, blocks designated $z^{-1}$ represent one-sample delay elements. Arrows leading to multiplication by coefficients $c_n$ are traditionally known as "taps," and the coefficients themselves are known as "tap gains." This block diagram represents the equation

$$\tilde{z}[k] = \sum_{n=-N}^{N} c_n y[k-n]. \tag{1}$$

Our goal in designing an equalizer is to find values for the tap gains $c_{-N},\ldots,c_N$ that will minimize any residual ISI. Even better, we would like to find a way to have the tap gains adjust themselves to reduce ISI as the equalizer runs.

To provide a basis for adjusting the equalizer tap gains $c_{-N},\ldots,c_N$, the receiver must know what data the transmitter is sending. Common practice is to begin transmission with a "training sequence" $b_k, k=0,\ldots,N_t-1$ that is known to the receiver. In this lab project we will use the same 26-symbol sequence that we are already using for frame synchronization. It is assumed that once the $N_t=26$ symbol training sequence has been received, the tap gains will have had time to adjust themselves to nearly their optimum values. Thus from this time on, the sequence $\tilde{z}[k]$ should be nearly ISI-free. Following the equalizer, the BPSK receiver performs detection and symbol mapping by comparing $\tilde{z}[k]$ with a threshold of zero. If $\hat{b}_k$ represents the output data, then

$$\hat{b}_k=\begin{cases}1, & \text{if } \tilde{z}[k]\geq 0\\ 0, & \text{if } \tilde{z}[k]<0.\end{cases} \tag{2}$$

What will the receiver use as a basis for adjusting the equalizer tap gains once the training sequence is complete? In a *decision-directed* equalizer, the decisions $\hat{b}_k$ are used to replace the training sequence bits $b_k$. Even when the probability of error is relatively high (e.g. 0.01), the decisions $\hat{b}_k$ are hardly ever wrong. It turns out that an occasional incorrect tap gain adjustment causes little change in the ISI output of the equalizer. Figure 2 shows the organization of the decision-directed equalizer.

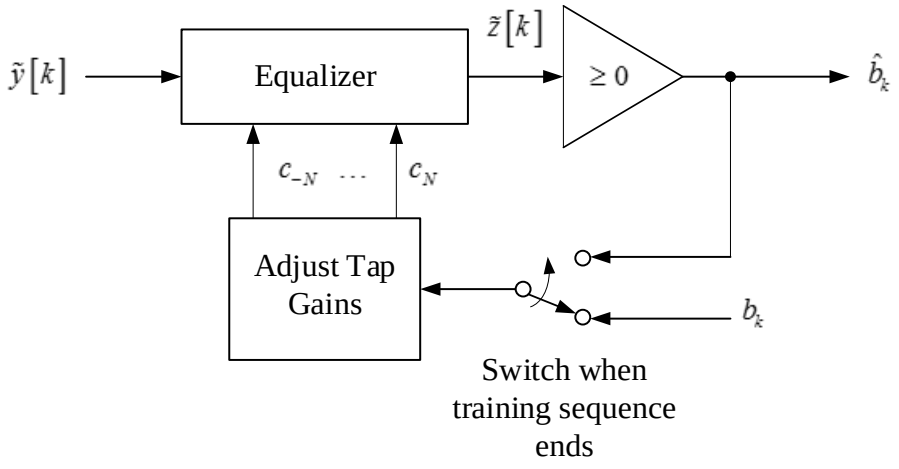



**Figure 2. Decision-Directed Equalizer**

There are a number of alternative criteria for deciding when the equalizer tap gains are optimally adjusted. One possibility is to adjust the tap gains to maximize the eye opening. It turns out, however, that equalizers have a tendency to amplify the noise that forms part of the input sequence $\tilde{y}[k]$. Maximizing the eye opening also tends to enhance the noise. An alternative criterion minimizes the mean squared error between the sequence $\tilde{z}[k]$ and the symbol sequence $a_k$ based on $b_k$. Since the mean squared error includes both ISI and noise, this criterion is less prone to noise enhancement than methods that ignore noise. Now if $a_k$ is the symbol sequence corresponding to the training sequence bits $b_k$, we can write the mean squared error $\mathrm{E}$ as

$$\mathrm{E} = E\left(\tilde{z}[k] - a_k\right)^2, \tag{3}$$

where $E$ is the expectation operator. Substituting Eq. (1) gives

$$\mathrm{E} = E\left(\sum_{n=-N}^{N} c_n \tilde{x}[k] - a_k\right)^2. \tag{4}$$

If we were designing a fixed equalizer, we would find the tap gains $c_{-N},\ldots,c_N$ that minimize $\mathrm{E}$ in Eq. (4). Instead, we want to find a way to let the tap gains adjust themselves. Suppose $c_{-N}^{(q)},\ldots,c_N^{(q)}$ represent the values of the tap gains after the $q$-th update. We can further update the tap gains according to the algorithm

$$c_n^{q+1} = c_n^q - \Delta\frac{\partial \mathrm{E}}{\partial c_n}, \quad n = -N,\ldots,N, \quad q = 0,1,\ldots. \tag{5}$$

This algorithm is an example of a *steepest descent* procedure, as it moves the coefficients in the direction of the negative of the gradient of the mean-squared error with respect to the coefficient values. This is the direction of the steepest descent down the slope to the minimum of the mean-squared error. The parameter $\Delta$ adjusts the step size. We can evaluate the derivative $\partial \mathrm{E}/\partial c_n$ by using Eq. (4):

$$\begin{aligned}\frac{\partial \mathrm{E}}{\partial c_n} &= E\left[2\left(\sum_{i=-N}^{N} c_i \tilde{y}[k-i] - a_k\right)\tilde{y}[k-n]\right] \\ &= 2E\left[\left(\tilde{z}[k]-a_k\right)\tilde{y}[k-n]\right] \\ &= 2E\left[e[k]\tilde{y}[k-n]\right], \quad n=-N,\ldots,N,\end{aligned} \tag{6}$$

where $e[k]=\tilde{z}[k]-a_k$ is the error signal (including both noise and ISI) at index $k$.

It turns out that the expectation operator in Eq. (6) poses something of a problem, since there is no way to evaluate it in an actual filter implementation. We can avoid the problem by approximating the expectation by the current value. That is,

$$E\left[e[k]\tilde{y}[k-n]\right] \cong e[k]\tilde{y}[k-n]. \tag{7}$$

Then the rule for updating the tap gains becomes

$$c_n^{q+1} = c_n^q - \Delta e[k]\tilde{y}[k-n], \quad n=-N,\ldots,N, \quad q=0,1,\ldots. \tag{8}$$

Normally we will update the equalizer tap gains after every received symbol. Thus the index $k$ and the index $q$ increment together. We have

$$c_n^{k+1} = c_n^k - \Delta e[k]\tilde{y}[k-n], \quad n=-N,\ldots,N, \quad k=0,1,\ldots. \tag{9}$$

The procedure expressed by Eq. (9) is called the *least mean square* (LMS) algorithm for adjusting the equalizer tap gains. We initialize the procedure by taking

$$c_n = \begin{cases} 1, & n=0 \\ 0, & \text{otherwise.} \end{cases} \tag{10}$$

A block diagram of the LMS adaptive equalizer is shown in Figure 3.

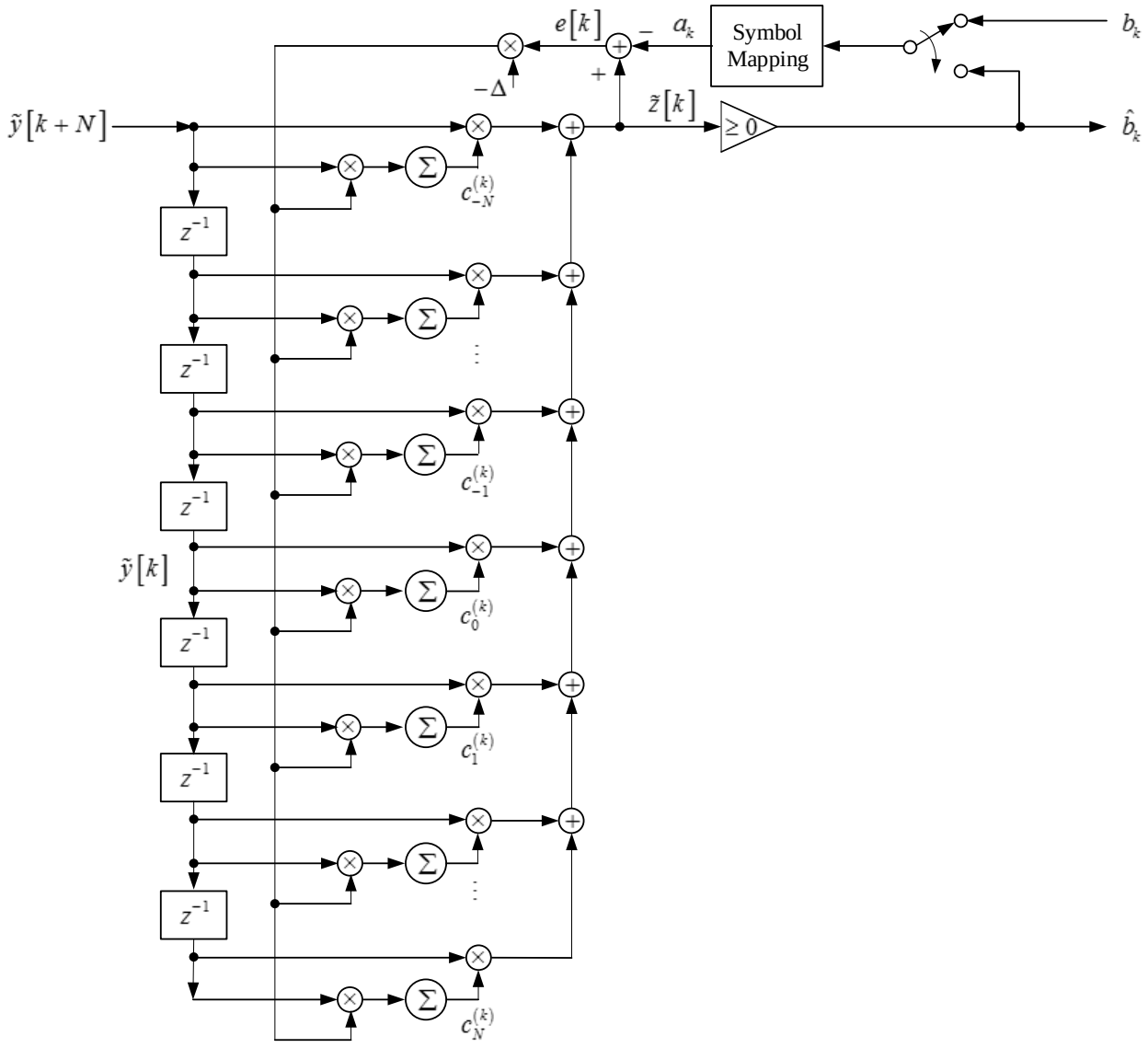



**Figure 3. LMS Adaptive Equalizer**

Note in Eq. (9) that if the step size $\Delta$ is small, the tap gain values will never change very much on a single update. This means that the tap gain values at any time are the accumulated sums of many small adjustments. Since accumulation is a form of averaging, Eq. (9) justifies our approximating the expectation operator by a single sample in Eq. (7). Further, since the tap gains change very little each time they are adjusted, a single symbol error does not push the equalizer very far out of adjustment when decision feedback is in use.

# 11.3 Pre-Lab

1. For this lab project you will use the BPSK transmitter and receiver you created for the phase-shift keying lab project. Be sure your receiver includes the *Channel.gvi* containing the channel model, and also includes the capability of displaying an eye diagram. You may use either *MT Format Eye Diagram* from the Modulation Toolkit, or the eye diagram program that you created for the eye diagram lab project. Wire the "ISI Channel Model" *Channel.gvi* input to a front panel control on your receiver.

2. Create a program to perform adaptive equalization. A template *EqualizerTemplate.gvi* has been provided to get you started. The inputs to your equalizer are to be connected as follows:

    a. Input Complex Waveform input to *Equalizer.gvi* will be connected to Output Complex Waveform on the *FrameSync(real)*. This waveform has been aligned to the start of the frame, but has not been downsampled. This waveform also includes the received header symbols that will be used to train the equalizer.
    b. Receiver Sampling Factor input to *Equalizer* will be connected to the number of samples per symbol parameter that is calculated in your BPSK receiver.
    c. Equalizer Length input to *Equalizer* should be connected to a control on your receiver front panel.

In addition to the inputs and outputs, the equalizer template contains a copy of the training sequence, an array containing the symbol map, and a cluster of "feedforward equalizer parameters." These parameters include the number of equalizer taps per symbol (one), and the values of step size $\Delta$ to use during training and during decision-directed operation (0.05 during training, 0.001 in decision-directed mode).

To complete the equalizer, you will need to add two functions from the Modulation Toolkit. These are *MT Generate System Parameters* from the Analysis→Communications→Digital→Utilities subpalette and *MT PSK Feedforward Equalizer* from the Analysis→Communications→Digital→Equalization subpalette.

Click on the MT PSK Feedfordward Equalizer function then, from the Configure ribbon, choose PSK(M) for the *MT Generate System Parameters*. Create a constant at the "PSK type" input and select "Normal." Wire your "Receiver Sampling Factor" to the "samples per symbol" input. Wire a constant of value 2 to the "M-PSK" input. The only output of the *MT Generate System Parameters* that you will use is the "PSK system parameters" cluster. Wire this to the *Cluster Properties function* provided in the template to replace the default symbol map with the symbol map provided. The modified PSK system parameters cluster will be used by the *MT PSK Feedforward Equalizer*.

From the Configure ribbon, choose Specify Length for the *MT PSK Feedforward Equalizer.gvi*. Wire up the "input complex waveform" and "PSK system parameters" inputs. Wire the "equalizer length" input to the appropriate control. Wire the "training bits" input to the training bits array provided in the template. Wire "feedforward equalizer (LMS) parameters" to the cluster provided in the template. The "reset" input can be left unwired, since it will default to the correct value of "true." Wire all of the outputs to the appropriate indicators and you are ready to go.

3. Save your equalizer in a file whose name includes the letters "Equalizer" and your initials (e.g. *Equalizer_BAB.gvi*).

4. Open the block diagram of your BPSK receiver. Wire up the remaining inputs of *FrameSync(real)*; that is, connect the "Input Signal" input to the "Aligned Signal" output of *PulseAlign(real)* and connect the "Receiver Sampling Factor" input to an appropriate location. The "Sampled Input" *FrameSync(real)* input should remain connected to the output of *Decimate*.

5. Add your equalizer to your BPSK receiver. Wire the inputs as described in Step 2 above. During the lab you will be asked to take data to compare the eye diagrams before and after equalization. Wire the "Equalized Complex Waveform" equalizer output to a second eye diagram display. Also, wire the "Squared Error" output to a waveform graph indicator so that you will be able to observe the error decrease as the equalizer adapts. Wire the "Output Bits" and "equalizer coefficients out" outputs to indicators.

## Questions

1. The default impulse response provided by *Channel.gvi* when the propagation delay is set at $100\ \mu s$ is $h[n]=\delta[n]+0.2\delta[n-1]-0.08\delta[n-2]$. Taking the z-transform gives channel system function $H(z)=1+0.2z^{-1}-0.08z^{-2}$. An ideal equalizer will have a system function that is the reciprocal of the channel system function. That is,

$$H_{eq}(z)=\frac{1}{1+0.2z^{-1}-0.08z^{-2}}\ . \tag{11}$$

   Find an expression for the ideal equalizer impulse response $h_{eq}[n]$. The coefficients of $h_{eq}[n]$ are the tap gains of an ideal (i.e. infinite length) equalizer. Find numerical values for the first five tap gains $c_0,\ldots,c_4$.

2. You should find that the ideal equalizer in Question 1 has a causal impulse response. The equalizer of Figure 3 can be non causal, if necessary. As a non-causal example, repeat Question 1 for $h[n]=0.1\delta[n]+\delta[n-1]+0.3\delta[n-2]$. Find numerical values for $c_{-4},\ldots,c_4$. (Hint: The ideal equalizer is an IIR filter. While it need not be causal, it must be stable.)

# 11.4 Lab Procedure

1. Connect a loopback cable and attenuator between the TX 1 and RX 2 connectors of the USRP. Connect the USRP to your computer and plug in the power to the USRP. Run LabVIEW and open the transmitter that you created in the prelab.

2. Ensure that the transmitter is set up to use

   Carrier Frequency: 915.0 MHz

   IQ Rate: 200 kHz. Note: This sets the value of $1/T_x$.

   Gain: 0 dB

   Active Antenna: TX1

   Symbol rate: 10,000 symbols/s

   Message Length: 1000 bits

   Pulse shaping filter: Root Raised

   Run the transmitter. Use the large STOP button on the front panel to stop transmission connectors.

3. Ensure that the receiver is set up to use

   Carrier Frequency: 915.0 MHz

   IQ Rate: 200 kHz. Note: This sets the value of $1/T_z$. Note that $T_z$ is the same parameter as $dt$.

   Gain: 0 dB

   Active Antenna: RX2

   Symbol rate: 10,000 symbols/s

   Message Length: 1000 bits

Pulse shaping filter: Root Raised

Use Channel: off

4. Run the transmitter, then run the receiver. Once the receiver has acquired its data, you may stop the transmitter. The receiver should show a BER of 0.0 or 1.0. Do not be concerned about a BER of 1.0 in this lab project.

5. Set the channel model for a propagation delay of $100\ \mu s$ with the default channel model. Set Use Channel to "on." Set the "Equalizer Length" to 11. Run the transmitter and then the receiver. Once the receiver has acquired data, you may stop the transmitter. Compare the eye diagrams before and after equalization. Use cursors to measure $V_w$ and $V_b$ and calculate the eye opening for each case.

6. Repeat Step 5 for propagation delays of $50\ \mu s$ and $150\ \mu s$. Compare the eye opening before and after equalization for each case. Prepare a table showing eye opening before and after equalization for each of the three propagation delays.

7. Set the propagation delay of the channel back to $100\ \mu s$ and run the transmitter and receiver. Observe the "Squared Error" equalizer output. Approximately what value does the squared error reach in steady state?

8. At a propagation delay of $100\ \mu s$ and an equalizer length of 11, record the values of tap gains $c_{-4},\ldots,c_4$. Compare with the values you computed in Prelab Question 1. (Do not expect extremely close agreement. The prelab calculations were for an infinite-length equalizer. Further, the adaptive equalizer tap gains are constantly being adjusted around their optimal values. Try running the receiver a few times to see how much variation there is in the steady-state tap gains.)

9. Change the ISI channel model to the model given in Prelab Question 2. Keep the propagation delay at $100\ \mu s$ and the equalizer length at 11. Run the transmitter and receiver. Check the

eye diagrams to verify that the equalizer is working. Record the values of tap gains $c_{-4},\ldots,c_4$. Compare with the values you computed in Prelab Question 2.

10. With the propagation delay set at $100\ \mu s$ and using either of the two channel models, measure the eye opening after equalization as the equalizer length is increased in odd-numbered steps from 1. At what equalizer length does the eye opening stabilize?

11. Phase Synchronization Revisited. You may recall from the BPSK lab project that any phase difference $\phi$ between the transmitter and receiver oscillators will produce a factor $\cos(\phi)$ in the demodulated signal. This factor affects the amplitude of the demodulated signal, and can also affect its polarity. It turns out that the equalizer can remove this phase error just as if it were a channel impairment. To see this happen, make the following changes to your receiver:

    a. The *MT PSK Feedforward Equalizer* includes a threshold comparator and symbol mapper. The output bits that the equalizer produces are available at the Output Bits output of *Equalizer.gvi*. Your receiver should have an *Array Subset function* following the threshold to limit the received bit sequence to the proper frame length. Replace the wire running from the threshold comparator to *Array Subset function* with a wire from the equalizer Output Bits.

    b. The output bit sequence from the equalizer includes the 26-bit frame header. To remove this header, set the index input to *Array Subset function* to 26 instead of 0.

       Run the transmitter and the receiver. You should get a BER of 0. Try running the receiver with Use Channel both off and on. You should get a BER of 0 every time, showing that the phase ambiguity caused by the phase correction algorithm has been removed.

    c. Now remove the phase synchronizer entirely. Figure 4 shows an expedient way to do this.

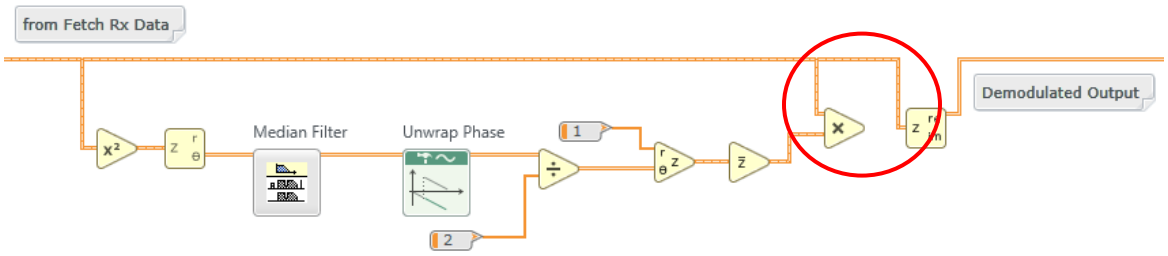



**Figure 4. Remove the Phase Synchronizer**

Run the transmitter and receiver again. Use Channel can be off or on. The BER should be zero every time.

## Questions

1. Answer the questions from Lab Procedure Steps 6 through 10.

# 11.5 Report

## Prelab

Hand in documentation for your equalizer program and your modified receiver that includes the equalizer and a second eye diagram. Also include documentation for any functions you may have created. To obtain documentation, print out legible screenshots of the front panel and block diagram.

Answer all of the questions in the Prelab section marked *Questions*.

## Lab

Submit your equalizer program and your revised receiver program. Also submit any functions you created. Be sure your files adhere to the naming convention described in the instructions above.

Resubmit documentation for any functions you modified during the lab.

Answer the questions in the Lab Procedure Steps 6 through 10. Comment on your success in using the equalizer as a phase synchronizer in Step 11.

LAB 12

# Quadrature Phase-Shift Keying

# 12.1 Objective

All of the modulation systems we have considered so far encode one bit of data into one symbol. For applications in which higher data rates are desired but channel bandwidth is limited, it is desirable to encode multiple bits of data into each symbol. In this lab project we introduce quadrature phase-shift keying (QPSK), a variation on binary phase-shift keying that encodes two bits of data into each symbol. Using QPSK, we can transmit data at twice the rate, without increasing the channel bandwidth. There will be some loss in performance, however, as somewhat more transmitted power will be needed to maintain a given bit error rate in the presence of noise.

The new concepts that are introduced in this lab project are the symbol mapping, which is more elaborate than the simple mapping used in BPSK, and the signal constellation, which is a visualization tool that is very useful for understanding high-efficiency modulation methods.

QPSK is easily extended to modulation methods that transmit even more than two bits per symbol. Some of these methods are known as "quadrature amplitude modulation" (QAM). For example, 16-QAM can transmit four bits of data per symbol. The more bits that are carried on each symbol, however, the more transmitted power will be needed to maintain a given bit error rate.

# 12.2 Background

As we saw in the BPSK lab project, a PSK signal can be represented as a train of pulses of the form

$$x(t) = Ag_{TX}(t)\cos(2\pi f_c t + \theta), \qquad (1)$$

where $A$ is a constant, $g_{TX}(t)$ is a fixed pulse shape designed to limit the signal bandwidth and control ISI, and $f_c$ is the carrier frequency. Information is carried on the phase angle $\theta$. For BPSK, $\theta$ can take one of two values, and each pulse carries one bit of information. For QPSK, $\theta$ can take four values. Since two bits are needed to specify one out of four possibilities, each QPSK pulse carries two bits of information. We will take the four possible phase angles to be $\theta = \pm\pi/4, \pm 3\pi/4$.

An alternative way of representing the pulse of Eq. (1) results from expanding the cosine. We can write

$$x(t) = A\cos(\theta)g_{TX}(t)\cos(2\pi f_c t) - A\sin(\theta)g_{TX}(t)\sin(2\pi f_c t). \qquad (2)$$

Substituting the four possible values of $\theta$ gives

$$x(t) = \pm\left(A/\sqrt{2}\right)g_{TX}(t)\cos(2\pi f_c t) \mp \left(A/\sqrt{2}\right)g_{TX}(t)\sin(2\pi f_c t). \qquad (3)$$

We can interpret Eq. (3) to imply that one data bit determines the polarity of the $\left(A/\sqrt{2}\right)g_{TX}(t)\cos(2\pi f_c t)$ term (the *in-phase* term), while the second data bit determines the polarity of the $\left(A/\sqrt{2}\right)g_{TX}(t)\sin(2\pi f_c t)$ term (the *quadrature* term).

To create the transmitted pulse represented by Eq. (1) or (3), we must provide the USRP with the complex baseband pulse

$$\tilde{x}(t) = Ag_{TX}(t)e^{j\theta}, \quad \theta = -3\pi/4, -\pi/4, \pi/4, 3\pi/4. \tag{4}$$

In this case the *symbol* is the complex number $Ae^{j\theta}$. The USRP will generate the transmitted signal of Eq. (1):

$$x(t) = \mathrm{Re}\left[\tilde{x}(t)e^{j2\pi f_c t}\right] = Ag_{TX}(t)\cos(2\pi f_c t + \theta). \tag{5}$$

A convenient way of visualizing the complex signal represented by Eq. (4) is to plot the possible symbol values $Ae^{j\theta}$ in the complex plane. A *signal constellation* for QPSK is shown in Figure 1.

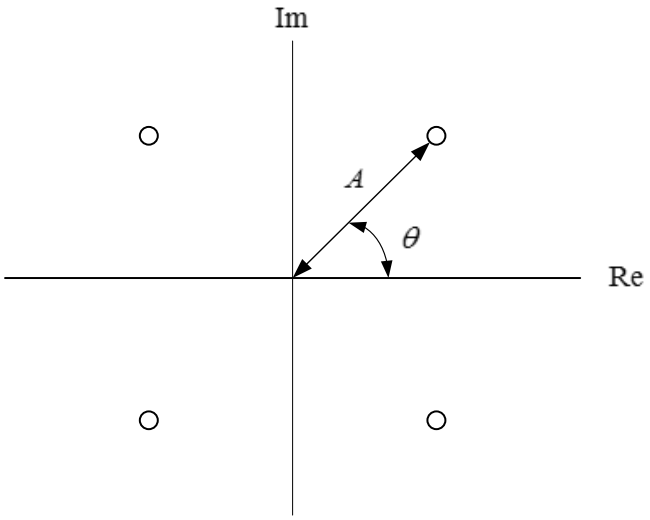



**Figure 1. QPSK Signal Constellation**

It is easy to see from the signal constellation that there are four possible transmitted signals (hence two bits per symbol), that each of the possible symbols has the same amplitude $A$, and that the four possible phase angles are $\theta = \pm\pi/4, \pm 3\pi/4$.

Let us now examine the relation between pairs of input bits and points in the signal constellation (i.e., symbol values). The mapping of bit pairs to points in the signal constellation is in fact arbitrary, as long as the transmitter and receiver use the same mapping. If we associate one bit with the polarity of the real part (in-phase term) and one bit with the polarity of the imaginary part (quadrature term) we obtain the mapping shown in Figure 2. As shown in the figure, the right-hand bit governs the polarity of the real part, while the left-hand bit governs the polarity of the imaginary part. It will be convenient later on to interpret the bit pairs as binary numbers. In doing this, we will interpret the left-hand bit as the *least* significant bit, and the right-hand bit as the *most* significant bit. This interpretation may seem contrary to intuition, but these bit sequences will be stored as arrays, and it is natural to write arrays with the index increasing to the right. The symbol mapping can also be shown in a table. Table 1 is the symbol mapping of Figure 2.

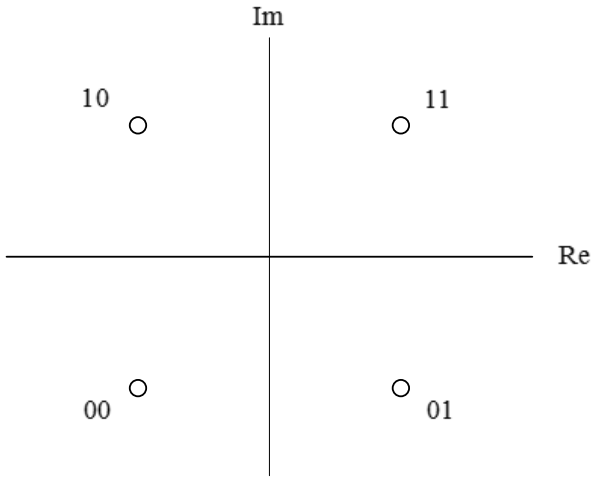



**Figure 2. A Possible Symbol Mapping**

**Table 1. Symbol Mapping**

| Index Number | Bit Pattern | Symbol Value |
|---|---|---|
| 0 | 00 | $Ae^{-j3\pi/4} = -A/\sqrt{2} - jA/\sqrt{2}$ |
| 1 | 10 | $Ae^{j3\pi/4} = -A/\sqrt{2} + jA/\sqrt{2}$ |
| 2 | 01 | $Ae^{-j\pi/4} = A/\sqrt{2} - jA/\sqrt{2}$ |
| 3 | 11 | $Ae^{j\pi/4} = A/\sqrt{2} + jA/\sqrt{2}$ |

A QPSK receiver begins with a DSB-SC demodulator. When the transmitted QPSK signal arrives at the receiver it has the form of a train of pulses, each given by

$$r(t) = Dg_{TX}(t)\cos(2\pi f_c t + \theta + \phi), \tag{6}$$

where $D$ is a constant (usually much smaller than the constant $A$ in the transmitted signal), the angle $\theta$ carries the information, and the angle $\phi$ represents the difference in phase between the transmitter and receiver carrier oscillators. If the receiver's carrier oscillator is set to the same frequency as the transmitter's carrier oscillator, the USRP receiver will do most of the work in demodulating the QPSK signal. The receiver's *Fetch Rx Data* will provide a train of output pulses, each given by[10]

$$\tilde{r}(t) = \frac{D}{2} g_{TX}(t) e^{j(\theta+\phi)}. \tag{7}$$

The pulse train represented by Eq. (7) is passed through a matched filter having impulse response $g_{RX}(t)$. If we write $g(t) = g_{TX}(t) * g_{RX}(t)$, then the matched filter output can be written

$$\tilde{y}(t) = \frac{D}{2} g(t) e^{j(\theta+\phi)}. \tag{8}$$

The matched filter output is then sampled, at the rate of one sample per symbol. Ideally $g(0) = 1$, so the sampler output is a train of complex numbers of the form

$$\tilde{y}[k] = \frac{D}{2} e^{j(\theta_k+\phi)}, \tag{9}$$

where $\theta_k$ is the phase value at the sample time $t = kT$ and $D$ and $\phi$ are constants. In practice, the samples $\tilde{y}[k]$ will be corrupted by ISI and noise. As in the previous lab project, ISI is removed by an equalizer filter. As we have seen, the equalizer will also remove the phase error $\phi$, leaving us with

$$\tilde{y}[k] = \frac{D}{2} e^{j\theta_k} + \text{ residual ISI} + \text{ noise}. \tag{10}$$

The remaining receiver steps are detection and symbol mapping. Signal detection is the process of examining each sample $\tilde{y}[k]$ and determining for that sample, which symbol is most likely to have been transmitted. It

[10] All of the baseband signals are actually represented in LabVIEW as discrete-time signals. Writing them as continuous-time signals makes the description given here much easier to read.

turns out that the algorithm for detection, called a *decision rule*, is surprisingly simple and has a nice geometric formulation. Figure 3 shows the signal constellation with the sample $\tilde{y}[k]$ also shown. Owing to residual ISI and noise, $\tilde{y}[k]$ does not correspond exactly to any of the possible symbol values.

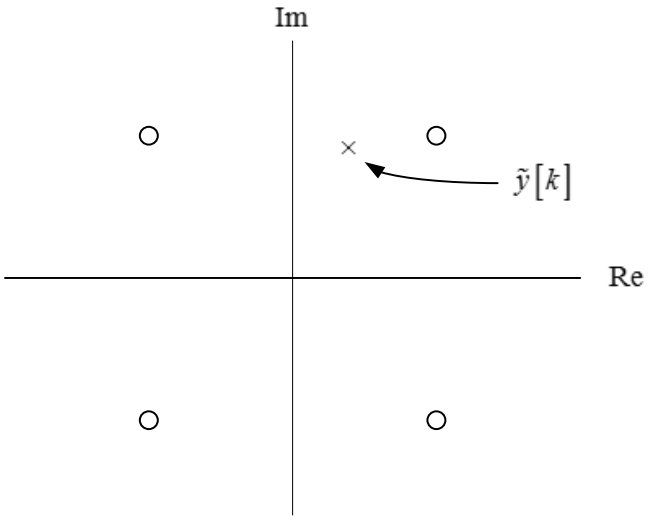



**Figure 3. Signal Constellation and Received Sample**

The decision rule for determining which symbol is most likely to have been transmitted given $\tilde{y}[k]$ is this: Calculate the distance between $\tilde{y}[k]$ and each possible symbol. The most likely symbol is the one at the smallest distance from $\tilde{y}[k]$. In Figure 3, the most likely symbol is the one in the first quadrant. Once the symbol has been determined, the symbol mapping of Table 1 can be used to convert the symbol to a bit sequence. For the example of Figure 3, the corresponding bit pattern is 11.

# 12.3 Pre-Lab

1. Create a function to perform the symbol mapping at the transmitter. (Detection and mapping at the receiver will be performed by *MT PSK Feedforward Equalizer*.) Table 2 gives the input and output specifications for your function.

**Table 2. Specifications for Map Data**

| Inputs | | | Output | | |
|---|---|---|---|---|---|
| Name | Type | | Name | Type | |
| Data In | 1-D array of 8-bit integer | Data bit stream | Symbols Out | 1-D array of double complex | Symbol stream to be transmitted |
| Symbol Map | 1-D array of double complex | Symbols indexed by bit-pairs represented as numbers | | | |

Your function must conform to the symbol mapping of Table 1 with $A=1$. A template *MapDataTemplate.gvi* has been provided with the inputs and outputs already wired. The symbol map array that you will need as one of the inputs is available in the template *QPSKTx.gvi* described below.

Save your symbol mapping function in a file whose name includes the letters "MapData" and your initials (e.g. *MapData_BAB.gvi*).

2. Create a program to implement the QPSK transmitter. A template *QPSKTxTemplate.gvi* has been provided to get you started. This template includes the inputs and outputs, the symbol map array, functions for signal and power spectrum display, and the functions needed to interface with the USRP.

You will need to add

- the symbol mapping function from Step 1 above,
- *AddFrameHeader(Complex)*
- Upsampling
- Root-raised-cosine pulse shaping
- Amplitude scaling

Tip: Use your BPSK transmitter as a model. The QPSK transmitter differs only in the symbol mapping.

Save your transmitter in a file whose name includes the letters "QPSKTx" and your initials (e.g. *QPSKTx_BAB.gvi*).

5. Create an equalizer function for QPSK. You can do this by making a copy of your BPSK equalizer function and then making three modifications as follows:

   - Change the "M-PSK" input to *MT Generate System Parameters* from 2 to 4.
   - Replace the symbol map with a copy of the QPSK symbol map from *QPSKTxTemplate.gvi*.
   - Replace the training sequence array constant with a new training sequence in which every bit is doubled. Figure 4 shows an easy way to do this with an *Interleave 1D Arrays function*.

   Save your equalizer in a file whose name includes the letters "QPSK_Equalizer" and your initials (e.g. *QPSK_Equalizer_BAB.gvi*).

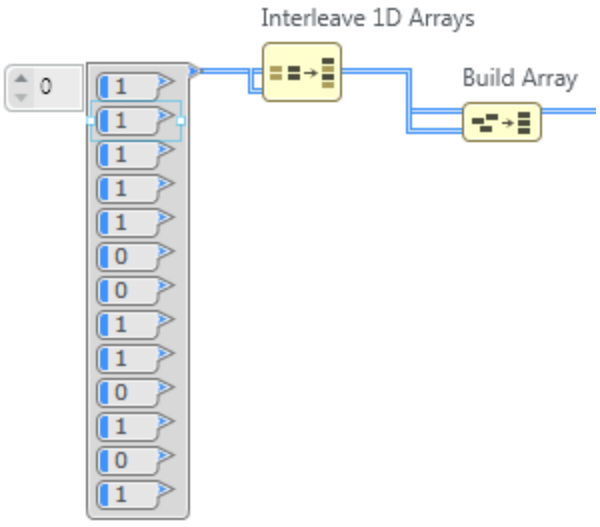



**Figure 4. Modifying the Training Sequence**

4. Create a program to implement the QPSK receiver. A template *QPSKRxTemplate.gvi* has been provided to get you started. The template includes the inputs and outputs, the functions needed to interface with the USRP, and several functions for configuring displays.

Tip: Use your BPSK receiver as a model. The QPSK receiver is very similar.

You will need to add

- *Channel.gvi*.
- Use a Cluster Properties function after the *Channel.gvi* to give access to its "Y" component, an array of double complex sample values.
- Root-raised-cosine matched filter and the *Convolution* function to implement the filter.
  Note: There is no phase synchronizer, since this function will be performed by the equalizer. Also, do not take the real part of the received sample values.
- Using a Cluster Properties function, create a complex double cluster from the output of *Convolution*.
- *PulseAlign (Complex)*.
- Use a Cluster Properties function after the *PulseAlign(Complex)* to give access to the "Y" component. Add *Decimate (single shot)* to sample the aligned pulses.
- *FrameSync(Complex)*.
- Your equalizer from Step 3 above comes next. The "Output Signal" from *FrameSync(Complex)* is the equalizer input. Do not use the "Aligned Samples" *FrameSync(Complex)* output.
- Connect the "Output Bits" equalizer output to an *Array Subset function*. Set the "index" to 52 to remove the training sequence header and set the "length" to the number of message bits. The output of *Array Subset function* can be displayed as the receiver's "Output Bits" and can also be sent to *MT Calculate BER*.
- There are two *MT Format Eye Diagram* functions in the template. Connect the first one to the "Output Signal" from *PulseAlign(Complex)*. This will display the eye diagram of the in-phase component of the received signal before equalization. Connect the second eye diagram to "Output Complex Waveform" from the equalizer. This will display the eye diagram of the in-phase component of the received signal after equalization.

Also connect the "Output Complex Waveform" from the equalizer to the "waveform" input of *MT Format Constellation*.

Save your receiver in a file whose name includes the letters "QPSKRx" and your initials (e.g. *QPSKRx_BAB.gvi*).

## Questions

1. The symbol rate of your transmitter and receiver is 10,000 symbols/second. What is the data rate in bits/second?

2. If the receiver is set for an IQ rate of 200 kHz, how many samples per symbol will there be at the receiver?

3. The *spectral efficiency* of a modulated signal is the data rate in bits/second divided by the bandwidth of the transmitted signal. Calculate the spectral efficiency of the BPSK signal using root-raised-cosine pulses that you generated in the BPSK lab project. Compare with the spectral efficiency of a QPSK signal, assuming that the bandwidth remains the same.

# 12.4 Lab Procedure

1. Connect a loopback cable and attenuator between the TX 1 and RX 2 connectors of the USRP. Connect the USRP to your computer and plug in the power to the USRP. Run LabVIEW and open the transmitter that you created in the prelab.

2. Ensure that the transmitter is set up to use

   Carrier Frequency: 915.0 MHz

   IQ Rate: 200 kHz. Note: This sets the value of $1/T_x$.

   Gain: 0 dB

   Active Antenna: TX1

   Symbol rate: 10,000 symbols/s

   Message Length: 1000 bits

   Pulse shaping filter: Root Raised

3. Run the transmitter. Use the large STOP button on the front panel to stop transmission connectors.

4. Using the power spectrum display on the transmitter front panel, measure the bandwidth of the complex baseband signal.

5. Ensure that the receiver is set up to use

   Carrier Frequency: 915.0 MHz

   IQ Rate: 200 kHz. Note: This sets the value of $1/T_z$. Note that $T_z$ is the same parameter as $dt$.

   Gain: 0 dB

   Active Antenna: RX2

   Symbol rate: 10,000 symbols/s

   Message Length: 1000 bits

   Pulse shaping filter: Root Raised

   Use Channel: off

6. Run the transmitter, then run the receiver. Once the receiver has acquired its data, you may stop the transmitter. The receiver should show a BER of 0.0.

7. Compare the eye diagram before and after equalization. Run the receiver a dozen times or so until you get a feel for what changes from run to run and what does not. Occasionally the "before" eye diagram will show four sample values rather than two. Take screenshots of the "before" and "after" eye diagrams for this case.

8. Set Use Channel to "on," with the default channel model, and a propagation delay of $100\ \mu s$. Run the transmitter and then run the receiver several times. Compare the eye diagrams before and after equalization. Measure the eye opening before and after equalization. Try to get a worst-case "before" scenario for your measurement.

9. Set Use Channel to "off," and run the transmitter and receiver again. This time observe the signal constellation. Take a screenshot of the constellation.

10. Change the wiring to *MT Format Constellation* so that the input is taken from the *FrameSync(Complex)* "Output Signal" instead of from the equalizer. Run the transmitter and then run the receiver several times. What happens to the angle of the signal constellation? What happens to the radius of the signal constellation?

## Questions

1. What is the measured bandwidth of the transmitted QPSK signal? (Note: Not the baseband signal bandwidth.) Compare with the bandwidth of the BPSK signal that you measured in an earlier lab project.

2. Report your eye opening measurement results from Step 6.

3. What is the cause of the signal constellation rotation that you observed in Step 8? What does the signal constellation look like when the "before" eye diagram shows four sample values rather than two?

Explain why the eye diagram appears as it does. Hint: Remember that the eye diagram plot displays only the in-phase component of the signal.

4. What would the signal constellation look like if the transmitter and receiver carrier frequencies were slightly different?

6. What would you expect to be the effect of rotation of the signal constellation on the BER? (Assume that the equalizer is not present.)

7. Observe the transition traces on the signal constellation graph. Notice that transitions between first and third quadrant signals pass through the origin of the graph, as do transitions between second and fourth quadrant signals. Describe what is happening to the amplitude of the received signal when these transitions occur.

# 12.5 Report

## Prelab

Hand in documentation for the four functions you created: the symbol mapping, the transmitter, the equalizer, and the receiver. Also include documentation for any functions you may have created. To obtain documentation, print out legible screenshots of the front panel and block diagram.

Answer all of the questions in the Prelab section marked *Questions*.

## Lab

Submit the symbol mapping, transmitter, equalizer, and receiver programs. Resubmit documentation for any functions you modified during the lab.

Submit the plots asked for in Steps 5 and 7.

Answer the questions in the Lab Procedure section marked *Questions*.