International Conference on Communication Technology and System Design 2011

Analog and Digital Modulation Toolkit for Software Defined Radio

R.Gandhiraja, Ranjini Ram, K.P.Soman, a*

1Center for Computational Engineering and Networking, Amrita Vishwa Vidyapeetham, Coimbatore, India
2Communication Engineering Research Group (CERG), Department of ECE, Amrita Vishwa Vidyapeetham, Coimbatore, India

Abstract

This work is a small tutorial for the new users in the field of software defined radio. Applications are build up using graphical user interface called the GNU radio companion (GRC). The idea behind developing such a tool kit is to give practical exposure in the communication concepts like basic signal generations, signal operations, multi-rate concepts, analog and digital modulation schemes and finally multiplexing schemes with the help of GNU radio. Unlike MATLAB Simulink or Labview GNU radio is open source i.e. free of cost and the concepts can be easily reached to the normal people without much of programming concepts using the pre written blocks. And programmers also have the chance to write their own applications.

© 2011 Published by Elsevier Ltd. Selection and/or peer-review under responsibility of ICCTSD 2011

Keywords: Software Defined Radio (SDR); GNU Radio; USRP; Communication Tool Kit;

1. Introduction

Using the SDR a simple and easily interpretable demonstration of basic communication systems are illustrated in this paper. This paper is divided into different sections. First section deals with introduction to software defined radio. Second section deals with a brief introduction to commonly used blocks in GNU radio companion. Third section gives the implementation procedure and final section gives the conclusion and future work.

2. Software Defined Radio

Software defined radio is a radio communication systems wherein which the hardware like filters, amplifiers, modulators etc are implemented in the personal computers or some embedding devices. By doing so, hardware complexity can be widely reduced. In addition to software part there is a RF front end preceding this software section. By using such a design one can transmit and receive variety of signals based on the applications.

In SDR, signal is captured by an antenna which is further converted into digital samples with regular intervals. These digital values are then processed in software, where the required application is written. The resulting output can be then converted back into audio, video or required form. The next section describes the hardware and software platform used for this project.

3. GNU Radio Companion

The GUI termed GRC allows user to implement GNU radio signal processing blocks in a manner similar to Simulink and Labview. The entire interface consist of over more than 150 blocks. Blocks are manually integrated...
into GRC via descriptive python definitions. The definitions are very flexible, and allow multiple GNU Radio blocks to be grouped into a single GRC super-block. In order to start working with GNU Radio Companion, popularly known as GRC, type ‘grc’ in the terminal window and press enter.

Then the new window GRC will open up. GRC provides a graphical user interface to the user so that any hardware functions like mixers, oscillators, etc., can be implemented as a block and can be executed. These blocks use Python code to define its function (at the background) and xml code for creating GUI (in the foreground). A block is actually a class implemented in C++ language. SWIG is used as an interface between C++ and python. Thus python sees each block as a separate class defined in their respective module. Then the next step is to create suitable xml code for developing the code into a real block. Any operations that are being done using hardware can be executed in the same way using GRC also. But the constraint here is to place the blocks sequentially and making sure that all the parameters are defined properly. Several prewritten blocks are available which are developed mainly based on commonly used functions. But if the application demands for some additional blocks other than the available, change the parameters of the existing blocks according to the requirement and save as separate blocks. This type of blocks developed from the existing blocks is called the Hierarchical blocks. Or second method is to write a Python code based on our specification and suitably generating an XML code. This type of block is called as Custom block.

4. Using GNU Radio Companion

Here a list of communication experiments are going to be discussed which includes the very basic concept of signal generation which is pretty easy the GNU radio concept. In addition to signal generation signal operations, multi-rate concepts, analog and digital modulation schemes and finally multiplexing schemes experiments will be discussed.

4.1 Basic Signal Operation

Signal operation experiments include basic Addition, subtraction and Multiplication. Demonstration is quiet simple. In case of signal addition two signal sources of different frequencies are taken and is connected to a add block present in the operators list. Output of the add block is connected to the scope sink and FFT sink. Since two signals are added the same polarity will be added and opposite polarities get cancelled. This can be observed in the scope sink. Two spectrally separated peaks at the input frequencies can be observed in the FFT sink. Now in case of signal subtraction instead of connecting to the add block, connect the output of the input signal source to subtract block and the output plots can be observed.

Next, the important concepts in the communication systems signal multiplication. Connect input signals to a multiply block. Two signals at frequencies f_1 and f_2 are applied to a mixer, and it produces new signals at the sum f_1 + f_2 and difference f_1 - f_2 of the original frequencies. This can be observed in the frequency domain representation i.e. FFT sink. The corresponding frequency domain representation of the multiplication is also shown in Figure 1(b). Here the input frequencies are 1 KHz and 2 KHz. So the spectral plot will be at 3KHZ (f_1+f_2) and 1 KHz (f_1-f_2).

4.2 Multi-rate Operations

Sampling, interpolation and decimation are demonstrated here. Though theoretical knowledge is thee the physical meaning can be observed by this demonstration. Sampling can be explained very easily. Connect the input signal source with a frequency say f_1 to a scope sink. in case of grc when the window is opened two default block will be present in which one will variable block with parameter samp_rate set as 32KHz.in stead of fixed sampling rate set a variable slider so that a range of sampling rate can be chosen. Observe the plot when f_s=2f_m, f_s>2f_m (under sampling) and f_s<2f_m. According to the sampling theorem sampling frequency must be twice the maximum frequency. The demonstration is shown in Figure 1(a) with separate sampling rate for a single frequency. Here the demonstration is done in three separate blocks for better understanding. Time domain representation for three different sampling rates can be plotted. f_s>2f_m plot is shown in Figure 1(b).
Similarly interpolation is also demonstrated. Interpolation is the process of increasing the sampling rate of a signal. The interpolation factor is usually an integer or a rational fraction greater than unity. This factor multiplies the sampling rate or, equivalently, divides the sampling period. Here we insert zeroes in between the original samples thereby increasing the sampling rate. Here input signal block is connected to a Repeat block wherein the rate at which the signal is to be up-sampled can be set. The output of which is connected to low pass filter with suitable cut off frequency and transition width to satisfy Nyquist criteria. The output is then connected to scope sink. Vary the value in Repeat block to observe the changes in the output plots. The demonstration is shown in Figure 2(a). Similarly decimation also can be explained which is demonstrated in Figure 2(b).

Fig.1.(a) Sampling with three sampling rates; (b) Frequency domain representation of \( F_s > 2F_m \)

Similarly interpolation is also demonstrated. Interpolation is the process of increasing the sampling rate of a signal. The Interpolation factor is usually an integer or a rational fraction greater than unity. This factor multiplies the sampling rate or, equivalently, divides the sampling period. Here we insert zeroes in between the original samples thereby increasing the sampling rate. Here input signal block is connected to a Repeat block wherein the rate at which the signal is to be up-sampled can be set. The output of which is connected to low pass filter with suitable cut off frequency and transition width to satisfy Nyquist criteria. The output is then connected to scope sink. Vary the value in Repeat block to observe the changes in the output plots. The demonstration is shown in Figure 2(a). Similarly decimation also can be explained which is demonstrated in Figure 2(b).

Fig.2.(a) Interpolation; (b)Decimation

4.3 Analog Modulation Schemes

This is the most basic form of modulation. Here the amplitude of carrier signal is varied according to the variations in the amplitude of modulating signal. The variations of the amplitude modulation called Double side band suppressed carrier (DSB-SC) and single sideband (SSB) modulation and demodulation is demonstrated. The basic equation for modulated signal is given as

\[
s_{AM}(t) = A_c [1 + k_f m(t)] \cos(2\pi f_c t)
\]

where, \( A_c \) = amplitude of carrier signal, \( k_f \) = modulation index, \( m(t) \) = message signal and \( f_c \) = carrier frequency.

Signal source is baseband with frequency 1 KHz and amplitude 0.5 whose output is multiplied with a multiply constant. here multiply constant acts as modulation index(<1). The output of the multiply constant is added with add constant whose value is set as 1 which is then multiplied with carrier signal of frequency 8KHz and amplitude 1. Connect the output of the multiply block to scope sink and FFT sink. This is the modulation part. The output of modulation is again multiplied with signal source of same carrier frequency and is fed to a low pass filter with suitable cut off frequency which is then connected to the scope sink and FFT sink. Here ends the demodulation part. The demonstration is shown in Figure 3(a). Corresponding output plot is also shown in Figure 3(b).
4.3.1 DSB-SC Modulation and Demodulation

In the double-sideband suppressed-carrier transmission (DSB-SC) modulation, the wave carrier is not transmitted; thus, a great percentage of power that is dedicated to it is distributed between the sidebands, which imply an increase of the cover in DSB-SC, compared to AM, for the same power used. DSB-SC transmission is a special case of Double-sideband reduced carrier transmission. The name "suppressed carrier" comes about because the carrier signal component is suppressed. Here objective is to suppress the carrier. The only thing you need to do is remove the ‘add’ constant block from the flow graph of amplitude modulation represented as shown in Figure 4(a). You can see only two peaks at the sum and difference of the input signal with carrier being.

![Fig.3.(a)AM modulation and Demodulation; (b) AM Modulated signal](image1)

4.3.2 SSB Modulation and Demodulation

Transmission bandwidth of standard AM as well as DSB-SC modulated wave is 2W i.e. twice the message bandwidth W. Hence both the systems are bandwidth efficient. Both the systems, one half of the transmission bandwidth is occupied by the upper sideband (USB) and the other half is occupied by the lower sideband (LSB). But the information occupied in both these bands is same. So it’s enough if any one of the bands is transmitted suppressing one sideband completely. Since only one sideband is transmitted it also called as SSB or SSB-SC modulation. The DSB-SC signal when band pass filtered with appropriate high and low cut off the required SSB signal is obtained. Demodulation is same as AM. Demonstration and the plot are shown in Figure 4.

4.3.3 Angle modulation

In GRC, there are prewritten blocks for frequency and phase modulation. So a signal source of some frequency is fed into this frequency mod block wherein which sensitivity is the parameter to be given. When the sensitivity parameter is increased there is increase in the bandwidth of the modulated signal spectrum. Demodulator block is also present to demodulate the signal. The demonstration is shown in Figure 5.

![Fig.4.(a) SSB Modulation and Demodulation; (b)SSB Modulated spectrum](image2)
4.4 Digital and Pulse Modulation schemes

For Digital modulation we use digital signal which is binary. Our modulating signal will change according to our digital signal. Unlike analog signal which is continuous in time and amplitude, Digital signal is discrete in time and amplitude. It can be binary or multilevel signal. Therefore if our original data is analog like sound, it must be encoded into digital data.

4.4.1 Amplitude Shift Keying (ASK)

In Amplitude Shift Keying (ASK) the amplitude of the carrier signal is changed according to the modulating signal keeping frequency and phase constant. On-Off Keying (OOK) is an example of ASK where the binary signal is multiplied by the carrier signal. Since input is binary it’s enough to take a signal source with square as waveform. Multiply the same with a carrier of some frequency. Connect to scope sink to view the output plot. In the output plot there will be a shift in amplitude for all negative values and pulses for the positive values of input signal. This is the modulation part and demodulation part is similar to AM. The demonstration is shown in Figure 6(a) and the output plot in Figure 6(b).

4.4.2 Frequency shift Keying (FSK)

This method is also to transmit digital signal. Here the 0(low) is represented by one frequency and 1(high) is represented by another frequency. Here in GNU, logic high is represented by a square wave and logic zero by subtracting same square wave from 1. Now multiply each input signal with two different carrier frequency \( f_1 \) and \( f_2 \). Add the resultant output to view the frequency shifted output. Demonstration of FSK is shown below along with its output in Figure 7.
4.4.3 Phase shift Keying

Phase-shift keying (PSK) is a method of digital communication in which the phase of a transmitted signal is varied to convey information. There are several methods that can be used to accomplish PSK. The simplest PSK technique is called binary phase-shift keying (BPSK). Similar to FSK. Instead of taking two different carrier frequencies it uses two opposite signal phases (0 and 180 degrees). it’s enough to take single frequency but one at 180 degree phase shift from other carrier signal. Demonstration of PSK is along with its output is shown in Figure 8. Observing the output plot there is slight shift in the output waveform when the input bit changes from 1 to 0.

4.4.3 Pulse Amplitude Modulation

Pulse-amplitude modulation (PAM), is a form of signal modulation where the message information is encoded in the amplitude of a series of signal pulses. Pulse-amplitude modulation is now rarely used, having been largely superseded by pulse-code modulation and, more recently, by pulse-position modulation. Input signal is normal sine/cosine signal source with some frequency. This has to be multiplied with carrier signal with higher frequency with waveform type as square. When the output plot is observed the message signal will encoded in accordance with carrier signal. The demodulation part is similar to AM. Demonstration is shown below along with the output plot in Figure 9.

4.5. Frequency Division Multiplexer and Demultiplexer

Frequency-division multiplexing (FDM) is a form of signal multiplexing where multiple baseband signals are modulated on different frequency carrier waves and added together to create a composite signal. In this case the carrier signals are referred to as subcarriers: an example is stereo FM transmission. Here multiple inputs are taken which is applied to modulators that shift the frequency ranges of the signals so as to occupy mutually exclusive frequency intervals. The necessary carrier frequencies needed to perform these frequency translations are obtained from a carrier supply. For modulation any method above described can be used. At the receiving terminal band pass
filters are used to separate the message signals. Finally original signals are recovered using individual demodulators and low pass filters. Demonstration in GNU is shown below with corresponding multiplexed output.

5 Conclusion

The objective of this paper is to provide a small working platform on how to work with GRC open source software tool. The basic Demonstrations are done using the basic theories of communication systems. By doing so rather than theoretical knowledge some practical knowledge also is gained as the system deals with real signals. The various output plots can be interpreted and related to theoretical information. One single Universal Software Radio Peripheral (USRP) can implement all the modulations and multiplexing techniques in real time. The student community will get practical realization about the theoretical concepts.

6 References


