Implementation of basic analog and digital modulation schemes using a SDR platform

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Abstract

Traditional radio communications systems (hardware based operation) has been displaced by radio systems whose functionality is totally (or almost) software dependent, this kind of systems are called *Software Defined Radios* (SDR). SDR systems are present in many commercial devices (cellular phones, tablets, notebooks, modems etc.), therefore it is important for people related with electronic communication systems to know how this systems works.

This paper describes fundamental concepts and basic mathematical theory used in SDR systems, and shows how to implement basic analog (amplitude and frequency) and digital (binary shift keying) modulation schemes using the *Universal Software Radio Peripheral* (USRP) platform.

Keywords

Software Defined Radio, USRP, Digital Radio, Software Radio.

An introduction to Software Defined Radios systems

The term *Software Defined Radio* was first used in 1991 by Joe Mitola and it refers to a radio communication system embedded in a programmable platform, Reed (2002), software running in this platform determines his operation. Flexibility to add new functions, easy and real time reconfiguration and multi mode operation are some important characteristics of SDR systems.

The architecture of both systems (software defined and traditional radio systems) is similar: stages that comprise it and processes performed on each one are the same. Data type driven, discrete data are used on SDR systems instead of continuous data, and development platform, SDR systems uses programmable platforms such as digital signal processors (DSP) or logic programmable arrays (FPGA), are the main differences among them, fig. 1, Lehr (2002).

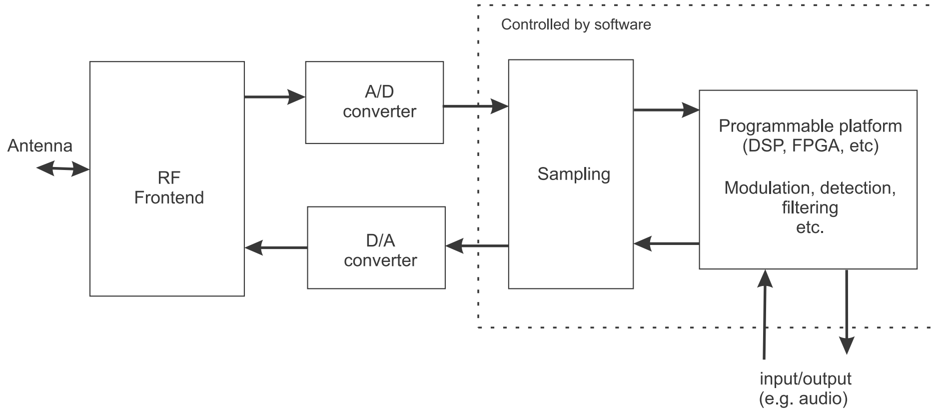


Figure 1: In software defined radios all processes (filtering, modulation, etc.) are performed by software, this software is running on a programmable platform.

In spite of his extensive use, SDR technology is unfamiliar to many people (even people concerned with electronic communications). Hence, theoretical and practical knowledge of this technology is highly important, it could be achieved using educational and research development platforms such as *Universal Software Radio Peripheral*, USRP, where electronic communications concepts and algorithms can be experimented.

USRP 1: a platform for developing and testing SDR systems

General characteristics

The USRP is an open source hardware platform consisting of a motherboard with four slots where we can connect daughter boards. An antenna can be connected to each daughter board to transmit or receive radio frequency (RF) signals. There are several types of antennas and daughter boards with a wide frequency operation range, 0 Hz to 6 GHz. Specific characteristics of antennas and daughter boards can be consulted in Ettus (2014).

The USRP 1 platform interfaces with a host computer via USB port. Host computer performs all base band signal processes such as modulation, demodulation, filtering, coding, etc., and USRP platform performs all pass band signal processes. Figure 2 shows, in a general form, the USRP platform architecture.

In receive path, daughter board converts RF received signal into intermediate frequency signal (IF), this IF signal is sent to an analog to digital converter (ADC), operating at 64 million samples per second (msps). Because of difference between sample rate of ADC and processing speed of host computer it is necessary data decimation, performed by a FPGA in the mother board. A dedicated USB controller chip receives decimated data from FPGA and sends it to host computer via USB port.

In transmit path, FPGA receives data from host computer via USB port and increases sample rate to match sample rates between data from host computer and a digital to analog converter (DAC), sampling at 128 msps. Daughter board translates analog signal from DAC into a signal with wanted transmission frequency.

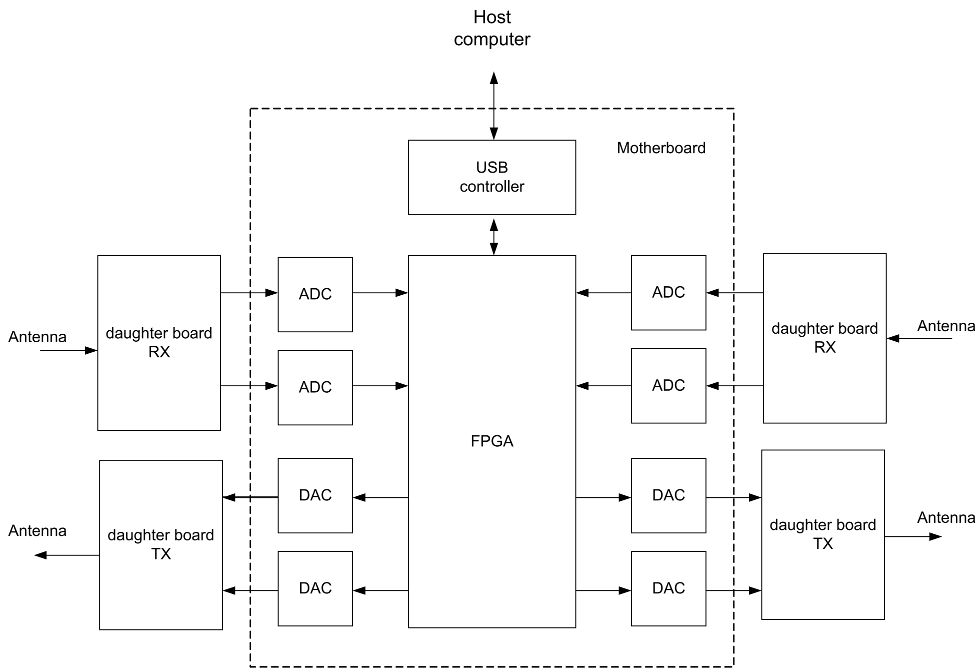


Figure 2: The USRP platform has a mother board where we can connect daughter boards. Digital/analog conversions, decimation and interpolation processes are implemented in the motherboard. RF/IF frequency conversions are performed in daughter boards.

Commercial availability, reasonable price, wide frequency range operation and compatibility with Windows and Linux operating systems are some characteristics that makes USRP feasible to experiment SDR systems.

Modulation schemes implemented in this work were done using USRP 1 and WBX daughter board ($60MHz$ to $2.2GHz$).

USRP driver and library installation

We used Windows and Matlab Simulink to implement mentioned modulation schemes. Neither of them supports the USRP platform, but Communications Engineering Research Group (Karlsruhe Institute of Technology, Germany) has developed drivers and libraries to use USRP on Windows and Simulink (USRP-Driver-1.2.zip and Simulink-USRP-2.1.3.zip), this files can be downloaded directly from Kit (2009). When extracting the files make sure to remember where you place the folder containing the drivers as you will need to point Windows and Matlab to this folder when installing the drivers.

USRP driver installation on Windows is like any other USB device driver installation: when you connect the device Windows detects it and ask for the path of driver file, once you set the path, press *Next* button and Windows will start installing the driver. After installation, USRP device must be appear in Device

Manager list from Control Panel.

To add the USRP library in Simulink you have to install a compiler (it can be Visual Studio C++ 2010, available for free download on internet) in your computer. Next, execute (from Matlab console) *mex –setup* command to set Visual C++ such as default compiler.

Finally, you have to set the path of Simulink-USRP-2.1.3 folder in Matlab, File Menu → Set Path → Add with subfolders..., and after execute (from Matlab console) *usrpBuildBinaries* command.

Concepts and mathematical theory associated with USRP platform operation

Baseband and passband signals, signal complex representation, and modulation are some things we have to review to operate and manipulate the USRP platform.

Signal complex representation

Analityc signal (also known as real signal complex representation) is composed by a real part, named *In-phase* component (I) and an imaginary part named *Quadrature* component (Q), fig. 3. This signal is widely used in electronic communications systems because it allows easy calculus, from I and Q components, of parameters such as instantaneous amplitude, (1), and phase, (2), Schoukens (2006), Youngblood (2002).

$$A = \sqrt{I^2 + Q^2} \tag{1}$$

$$\theta = \arctan \frac{Q}{I} \tag{2}$$

Analytic signal, $\hat{s}(t)$, of a real signal can be obtained from Hilbert transform, Lyons (2001). *In-phase* component is the original signal and *Quadrature* component is a 90 degrees phase shifted version of original signal. Figure 4 shows the analytic representation of a real signal $A\cos(2\pi ft + \theta)$.

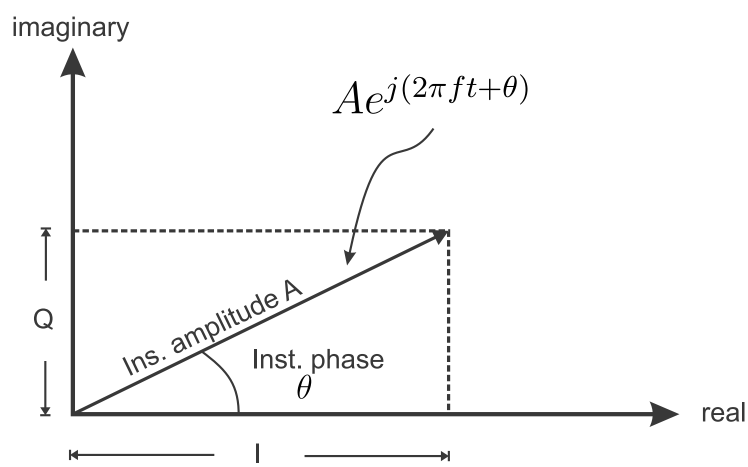


Figure 3: Graphic representation of analytic signal, it has a real component (I) and an imaginary component (Q).

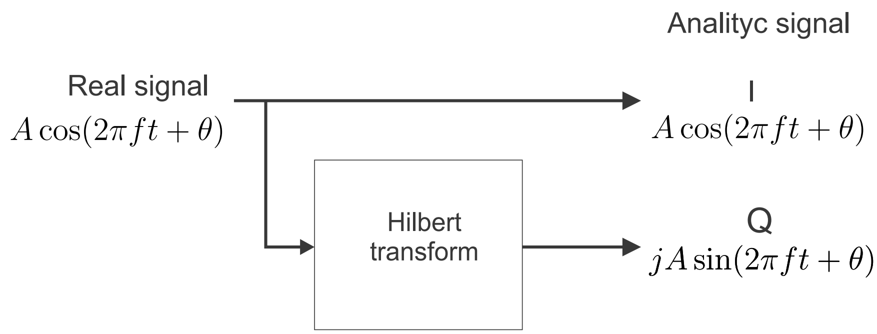


Figure 4: The Analityc signal can be obtained using Hilbert transform.

Euler's indentity can be used to represent an analytic signal, (3) shows analytic representation of a real signal $A\cos(2\pi ft+\theta)$ using Euler's indentity.

$$e^{j(2\pi ft)} = cos(2\pi ft) + jsen(2\pi ft) \tag{3}$$

An special characteristic of an analytic signal is the presence of only positive frequency components, fig. 5, Elfataoui (2004), Lyons (2000).

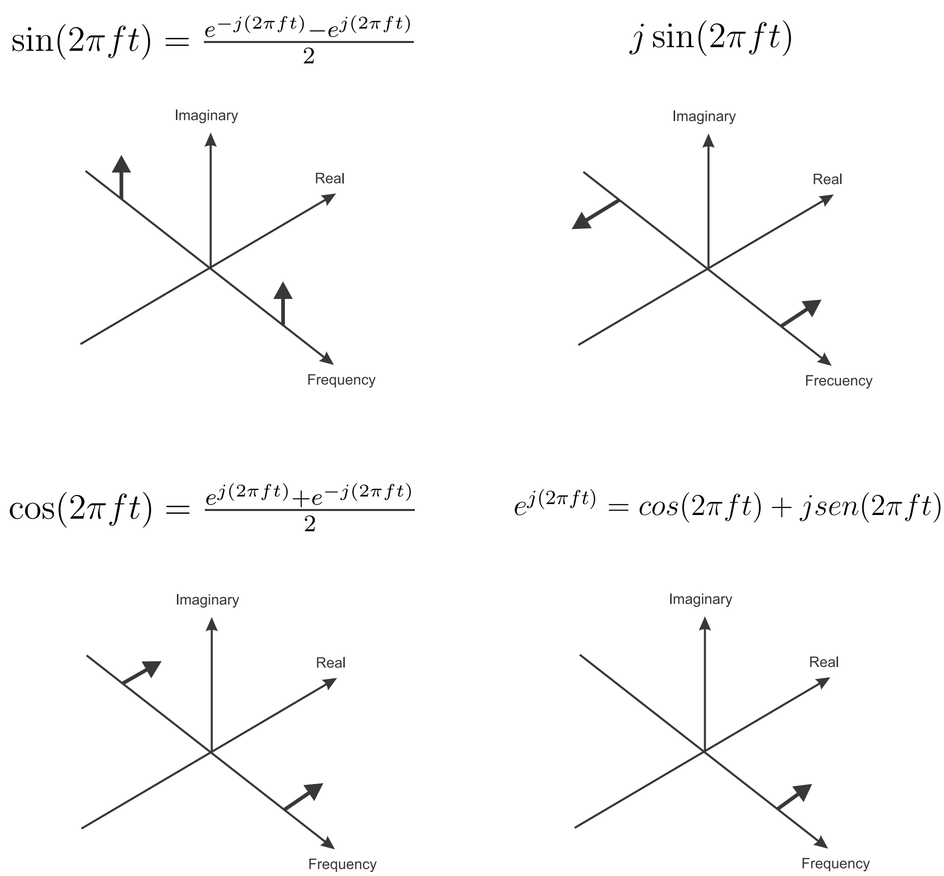


Figure 5: Only positive frequencies are present in analytic signal spectrum. We can find spectral components of an analytic signal using Euler's indentity.

Baseband and passband signals

A signal is named baseband signal when his spectral components are non zero close to 0 Hz, fig. 6. Because significantly lower sampling rate is needy when baseband signal is used, it is preferred to use it on digital processing.

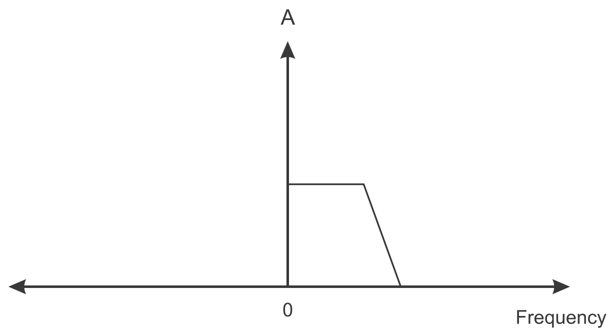


Figure 6: Spectral components of a baseband signal are close to 0 Hz.

Passband signal is a signal whose spectral components are non zero around some f_c frequency other than zero, fig. 7.

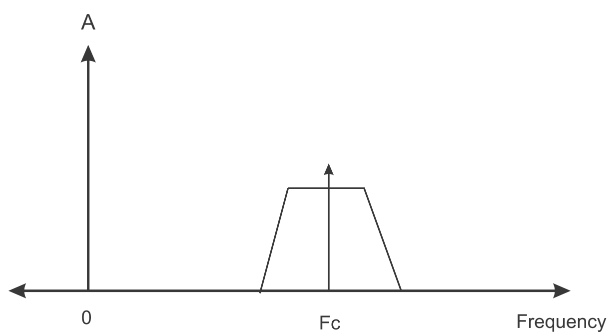


Figure 7: In a bandpass signal the spectral components are around some f_c frequency.

It is important to know how to convert between base band and pass band signals, since USRP platform carries out this conversion processes.

To convert from real passband signal to baseband signal we have to obtain her analytic equivalent and multiply it by $e^{-j(2\pi f_c t)}$. Extracting the real part of the product of this baseband signal and $e^{j(2\pi f_c t)}$ we can return to the original passband signal.

It is shown the conversion of passband signals into baseband signals of amplitude, frequency and binary phase shift keying modulated (AM, FM and BPSK respectively) signals. This modulation schemes are implemented later.

▷ Amplitude modulation

AM signal equation

$$s_{AM}(t) = (A_c + m(t))\cos(2\pi f_c t) \tag{4}$$

Analytic AM signal

$$\hat{s}_{AM}(t) = (A_c + m(t))\cos(2\pi f_c t) + j(A_c + m(t))\sin(2\pi f_c t) = (A_c + m(t))e^{j(2\pi f_c t)} \tag{5}$$

AM Baseband representation

$$(A_c + m(t))e^{j(2\pi f_c t)}e^{-j(2\pi f_c t)} = (A_c + m(t)) + j0 \tag{6}$$

Where $I = (A_c + m(t))$ and $Q = 0$

Original signal, (4)

$$Re\{[(A_c + m(t)) + j0]e^{j(2\pi f_c t)}\} \tag{7}$$

▷ Frequency modulated signal

FM signal equation

$$s_{FM}(t) = A_c\cos(2\pi f_c t + k_f \int m(t)) \tag{8}$$

Analytic FM signal

$$\hat{s}_{FM}(t) = A_c\cos(2\pi f_c t + k_f \int m(t)) + A_c\sin(2\pi f_c t + k_f \int m(t)) = A_c e^{j(2\pi f_c t + k_f \int m(t))} \tag{9}$$

FM Baseband representation

$$A_c e^{j(2\pi f_c t + k_f \int m(t))}e^{-j(2\pi f_c t)} = A_c e^{j(k_f \int m(t))} = A_c\cos(k_f \int m(t)) + A_c\sin(k_f \int m(t)) \tag{10}$$

Where $I = A_c\cos(k_f \int m(t))$ and $Q = A_c\sin(k_f \int m(t))$

Original signal, (8)

$$Re\{[A_c\cos(k_f \int m(t)) + A_c\sin(k_f \int m(t))]e^{j(2\pi f_c t)}\} \tag{11}$$

▷ BPSK modulated signal

BPSK signal equation

$$s_{BPSK}(t) = \begin{cases} \cos(2\pi f_c t) & \text{if input bit} \quad 1 \\ \cos(2\pi f_c t + \pi) = -\cos(2\pi f_c t) & \text{if input bit} \quad 0 \end{cases} \tag{12}$$

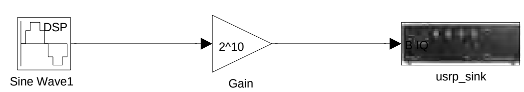
From (12) we can observe that BPSK modulated signal can be represented as an amplitude modulated signal where $m(t)$ is 1 or -1, thus the process to obtain baseband and passband signals is identical as in AM.

Modulation schemes on USRP 1

Basic Analog and digital modulations schemes were implemented to demonstrate how USRP 1 platform can be used to experiment, from the simplest (amplitude and frequency modulation/demodulation) to the most complex (channel estimation, carrier synchronization, etc.), electronic communications processes.
A simple test can be done to be sure if USRP 1 is working correctly, it consists in transmitting and receiving, with USRP, a known signal.

Checking that USRP works correctly

Figure 8 shows the model in Simulink to transmit an exponential complex signal $Ae^{j(2\pi ft)}$ with $f = 4kHz$ and $A = 1v$. Frequency Carrier signal is set to $1.7GHz$ (this frequency value is used in all Simulink models implemented here, unless specified otherwise). Theoretically, the same signal has to be present in the receiver.

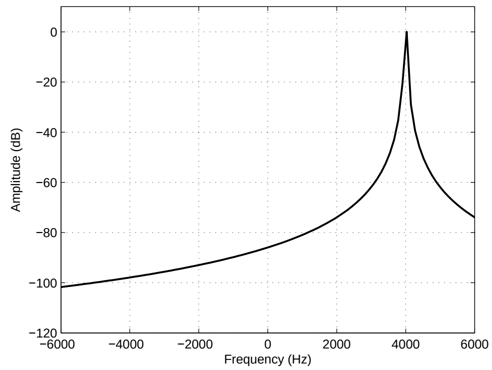


(a) Complex signal tranmitter simulink model

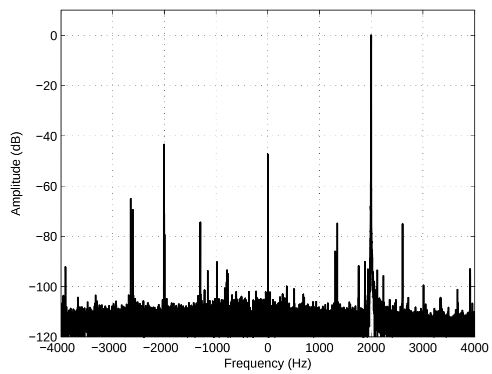
(b) Complex signal receiver simulink model

Figure 8: Complex signal transmitter and receiver Simulink models. Spectrum must be the same both in receiver and transmitter.

The transmitted and received complex signal spectral component is showed in Figure 9. It may be observed that there is a difference (2 KHz) between frequency values, this is because of *Carrier Frequency Offset*, Johnson (2003), and it should be corrected.



(a) Transmitted complex spectral component at 4 KHz.



(b) At receiver, the spectral component differs from original spectral component because of *Carrier Frequency Offset*.

Figure 9: Complex signal spectral components at transmitter and receiver. Ideally, both may be identical.

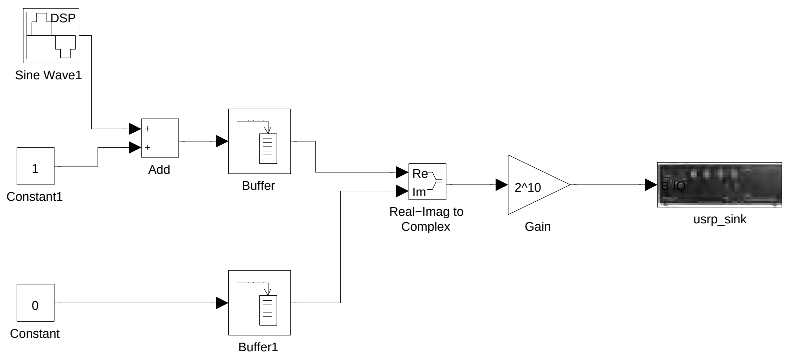
A gain block should be inserted before USRP block with the purpose of reach the dynamic range of DACs. We can ”play” a little with this gain value to achieve a good performance (values from 2^{10} to 2^{14} are appropriated).

Source signal and USRP must have the same sample rates, and it is therefore sometimes necessary to use interpolation and decimation blocks, in this case *Sine Wave* block is configured at the same sample rate that USRP block, so it is not necessary an interpolation or decimation block.

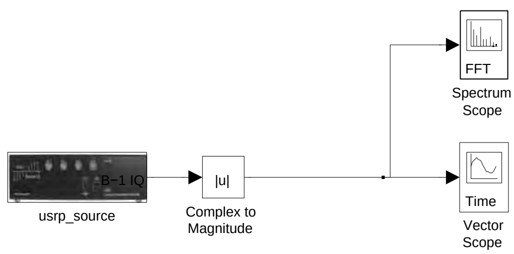
Data processing can be performed in two ways, sample and frame based. When sample processing is performed each sample is processed individually. In frame processing, samples are accumulated in a large group (frame) and after, all process are applied to this frame of data, resulting in a best efficiency of the system. USRP platform use frame based processing, and it is therefore sometimes necessary to use a *Buffer* block, otherwise configure each source signal block to provide a data frame output.

AM transmitter and receiver

Once we have verified that USRP works correctly, we implemented an Amplitude Modulation system using as reference (6). In this case, we transmit a sinusoidal signal with $A = 1v$ at $f = 2kHz$. Figure 10 shows the AM system implemented in Simulink and using USRP.



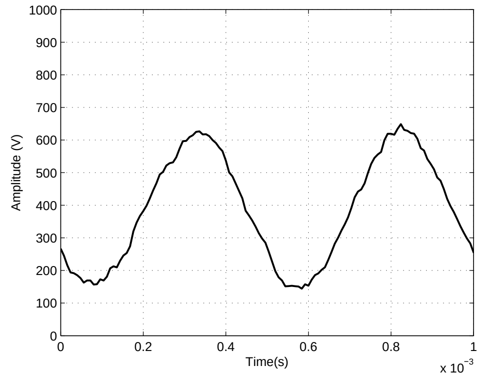
(a) AM transmitter



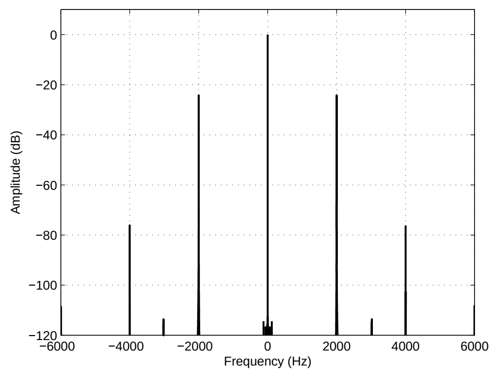
(b) AM receiver

Figure 10: Amplitude Modulation Simulink models.

Figure 11 shows time and frequency domain received signal, We can observe that frequency of received and transmitted signal are identical, Why there is not a difference between frequency values? The answer is simple: in an AM system the information is contained in carrier amplitude variations, so that received signal spectral component is not affected by *Carrier Frequency Offset*.



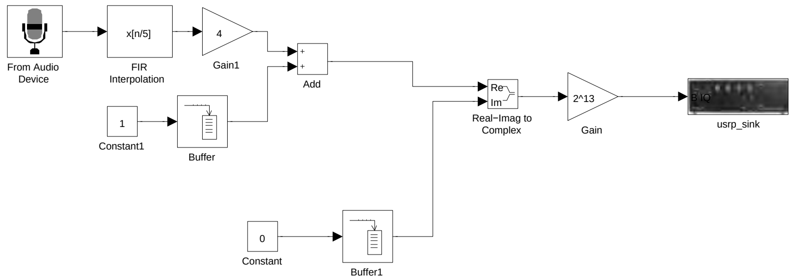
(a) Time domain received signal.



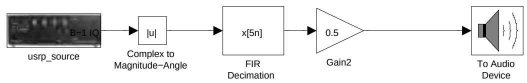
(b) Frequency domain recevived signal

Figure 11: Time and frequency domain AM demodulated signal .

It is possible to transmit and receive audio using some blocks whose function is to access to the computer audio card. Figure 12 shows the AM Simulink model modified to transmit and receive audio. Because audio card and USRP have different sample rates, *interpolation* and *decimate* blocks are needed.



(a) AM audio transmitter

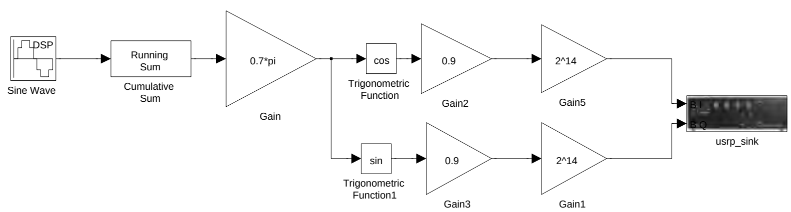


(b) AM audio receiver

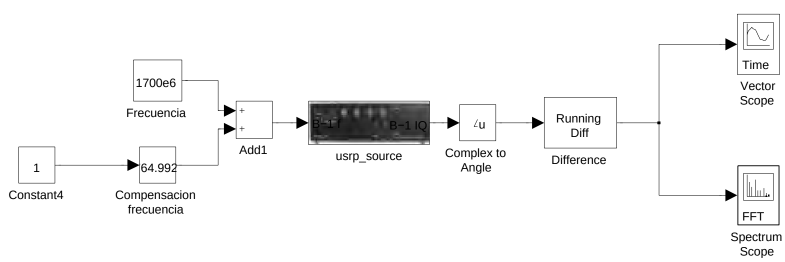
Figure 12: Simulink models used to transmit and receive audio.

FM transmitter and receiver

FM baseband representation, (10), is used to implement the FM system in Simulink, it is showed in figure 13. The transmitted signal is the same as above.



(a) FM transmitter

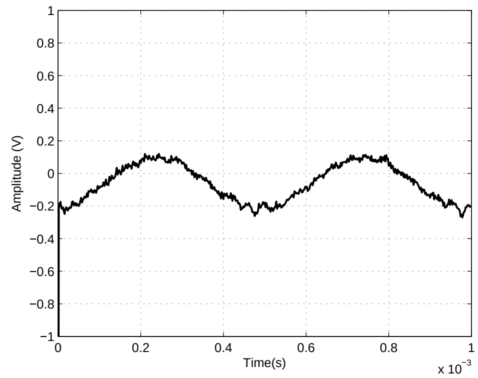


(b) FM receiver

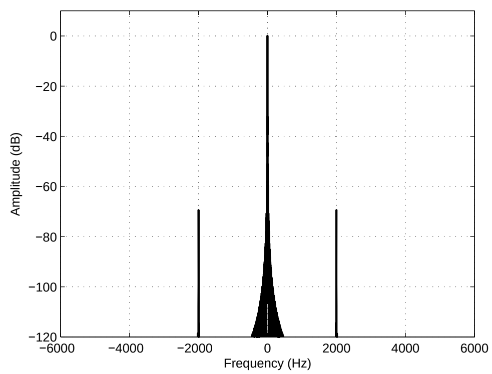
Figure 13: Frequency Modulation Simulink models

We can observe an extra port in *usrp_sink* block, this port allows to set and modify carrier signal frequency in real time through a *constant* block and a *slider gain* block, allowing an adjust of the *Carrier Frequency Offset* that in this case is around $765Hz$.

Received signal in time and frequency domain is showed in figure 14. Received spectral component has the same value that transmitted signal due to the adjustment to *Carrier Frequency Offset*.



(a) Time domain received signal.



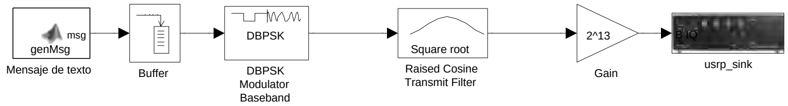
(b) Frequency domain received signal

Figure 14: Time and frequency domain Am demodulated signal .

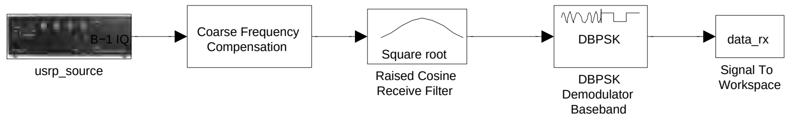
As with AM system, it is possible to transmit audio. Taking into account that frequency range operation of WBX daughter board ($60MHz$ to $2.2GHz$) covers commercial FM broadcast band ($88MHz$ to $108MHz$), transmitted audio from USRP can be received and heard in a portable radio.

DBPSK transmitter and receiver

As in previous cases, we used baseband signal representation to implement the system, but now, the modulation and demodulation process are carried out by *DBPSK Modulator Baseband* and *DBPSK Demodulator Baseband* blocks, figure 15 shows the DBPSK system implemented in Simulink.



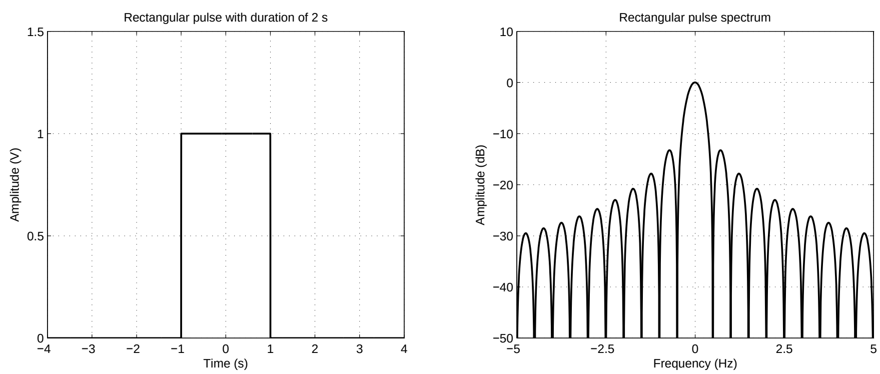
(a) DBPSK transmitter



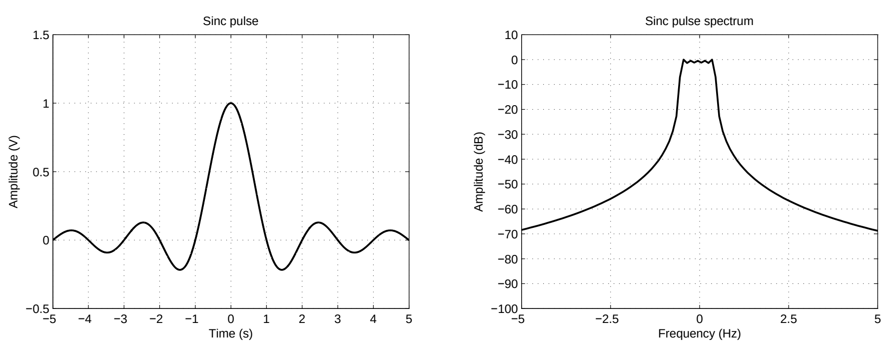
(b) DBPSK receiver

Figure 15: Dbpsk system simulink model

It is very important to consider that, in this case, baseband data is digital and is coded in 1 or -1 (for 1 or 0 respectively). Bandwidth of digital data is theoretically infinite, which is a problem in communication systems, to solve this situation it is necessary to convert this digital data into a signal with a reduced bandwidth, this conversion is performed by *Raised Cosine Transmit Filter* block, figure 16 shows rectangular and sinc signals spectrum.



(a) Rectangular pulse spectrum



(b) Sinc pulse spectrum

Figure 16: Sinc pulse have a finite bandwidth, for this reason it is preferable to send a sinc pulse (or ohter with similar characteristics) instead of a rectangular pulse.

To improve signal to noise ratio (SNR) is necessary to use a mathematical tool named correlation. Correlation provides a measure of the similarity between the received signal and a reference signal (in this case a raised cosine pulse), Ha (2010), increasing SNR in the sampling instant. A digital filter (*Raised Cosine Receive Filter* block) is used to calculate correlation because similarities between correlation and convolution (mathematical operation involved in digital filters), Proakis (1995).

Carrier Frequency Offset compensation is carried out automatically using an algorithm that calculates spectral components of the converted baseband signal, an spectral component with frequency value different of $0Hz$ represents frequency offset, this value is added algebraically to the received signal to eliminate it, Johnson (2003).

A Message Text and a wav file were sent using the DBPSK system. Start and stop marks were added to data in order to extract correct data from received information.

Conclusions

This paper has tried to explain some general concepts related to software defined radio in order to understand how the USRP works.

Basic communications systems were implemented using Simulink and the USRP. Some problems (such as Carrier Frequency Offset, SNR and bandwidth improvement, etc.) were faced. Solutions for this problems were experimented in real time and applied to solve it. With this, it is showed that USRP is a good tool to experiment algorithms and techniques related to software defined radio.

The USRP can be used in a electronics communication course as a didactic tool to demonstrate concepts that are hard to explain (improving student learning), or in a research laboratory to experiment channel estimation algorithms, diversity techniques, etc.

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