University *of* Alexandria
Faculty of Engineering

Analog Communication Theory

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Lab Manual

2023-2024

CONTETNTS :

Introduction

• Software Defined Radios

The NI-USRP (National Instrument - Universal Software Radio eripheral) is one of many Software-defined radios in the marketer. Software-defined radios is an emerging technology which has gained a lot of publicity in the past few years. The reason is the uniqueness of being able to model and control complicated analog RF tasks, such as modulation and demodulation, simply by using software and programming environments. Beforehand these tasks required extensive knowledge within the analog world and expensive tools to build and test designs, whereof multiple design-spins were necessary before everything was right.

Software developers have always had the advantage of easily being able to test the software for bugs or flaws before deployment. Software Defined Radio describes the technique of using a universal hardware frontend to receive and transmit RF signals with waveforms defined in software applications. The software-defined radio there-fore enables
developers to do the same thing with RF designs, as the developers are enabled to change carrier frequencies, modulation schemes and data to be transmit-ted on the fly. A benefit that is useful in both the research and development stage as well as in the deployment stage.

Software-defined radios, SDR in short, are widely used today. An SDR can be programmed to any RF specific task such as GSM or LTE networking, FM transmission, GPS tracking, WiFi or Bluetooth communication. In the recent satellite by GOMSpace they have installed
an SDR to be able to update the satellite on the fly for mission specific RF tasks. When a task finishes, they can reprogram the satellite to another task, even though the communication or tracking is using another frequency or band.

• Complex signal representation

From signal processing and communication theory we know that a physical signal can be represented as the real part of its' corresponding complex signal:

$$s(t) = Re\, a(t)e^{j\theta(t)} = a(t)cos(\theta(t)) \qquad 1\text{-}1$$

Where $a(t)$ is the time varying amplitude of the signal and $\Theta(t)$ is the time varying phase of the signal. A constant phase change over time corresponds to a sine-wave signal with a given frequency:

$$\theta(t) = 2\pi f_c t + \varphi(t) \tag{1-2}$$

Where *fc* is the constant frequency, such as the carrier frequency of an RF signal, and $\varphi(t)$ is any time varying changes done to the phase of this constant frequency signal.

In general an RF signal corresponds to a modulated carrier wave, where the modulation consists of time dependent changes of the amplitude and time dependent changes phase of the carrier frequency. The actual message signal to be transmitted and the desired modulation scheme defines these time dependent amplitude and phase changes. Let $a(t)$ denote the amplitude changes and $\varphi(t)$ the phase changes, both due to the message signal,

and let *fc* denote the carrier frequency, then the combined RF signal can be written in its complex form:

$$s(t) = a(t)cos(2\pi f_c t + \varphi(t)) = Re(a(t)e^{j\varphi(t)}\ e^{j2\pi f_c t}) \tag{1-3}$$

Notice how the carrier frequency is just a complex multiplication. This allows us to remove the carrier frequency part and extract the complex envelope also known as the baseband signal or complex baseband.

$$\tilde{s}(t) = a(t)e^{j\varphi(t)} \tag{1-4}$$

The baseband signal is generated from the message signal and therefore contains a modulated version of the message signal which can be extracted. The actual relationship between the modulation amplitude, $a(t)$, and modulation phase, $\varphi(t)$, depends on the chosen modulation scheme. When the baseband signal is combined with the carrier frequency the actual RF signal to be transmitted is denoted as:

$$s(t) = Re(\tilde{s}(t)e^{j2\pi f_c t}) \tag{1-5}$$

So to sum up, a baseband signal contains all required information about a message signal and the chosen modulation scheme. The baseband signal can be combined with any desired carrier frequency to generate signal to be transmitted, or the baseband signal can be extracted from a received signal to demodulate the message content.

- ## I and Q samples:

Software-defined radios represent the baseband signal as two real numbers, by splitting the signal into its' real and imaginary part. These are called the I and Q samples. The I samples being the real part and the Q samples being the imaginary, also known as the quadrature samples

$$\tilde{s}(t) = a(t)e^{j\varphi(t)} = I(t) + jQ(t) \tag{1-6}$$

Using the angle-addition identity of cosine with the equation for the RF signal, equation 1-3 , the RF signal can be rewritten into:

$$s(t) = a(t)cos\,(2\pi f_c t + \varphi(t)) \qquad 1\text{-}7$$
$$= a(t)cos(2\pi f_c t)cos(\varphi(t)) - a(t)sin(2\pi f_c t)sin(\varphi(t)) \qquad 1\text{-}8$$

From which we can separate the carrier frequency part and the baseband signal part, given by its' I and Q samples

$$s(t) = I(t)cos(2\pi f_c t) - Q(t)sin(2\pi f_c t) \qquad 1\text{-}9$$

From this we see that the I and Q samples correspond to the amplitude of the cosine and sine part of the phase changes to the signal, and that the I and Q samples are therefore 90 degrees apart. This agrees with the complex equation of the baseband signal, equation , as the real and imaginary part is 90 degrees apart

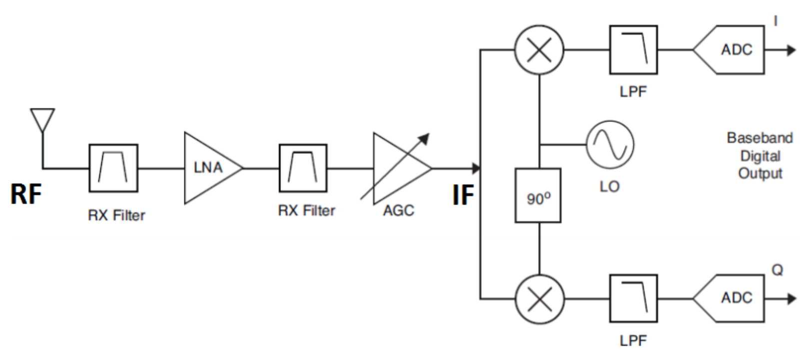
$$I(t) = a(t)cos\big(\varphi(t)\big) = Re(a(t)e^{j\varphi(t)}) \qquad 1\text{-}10$$
$$Q(t) = a(t)sin(\varphi(t)) = Im(a(t)e^{j\varphi(t)}) \qquad 1\text{-}11$$

Because of this 90-degree separation of the I and Q samples, they are also referred to as the "in-phase", *I*(*t*), and "quadrature", *Q*(*t*), components of the signal.

The actual rate of change in the message signal is usually many factors less than the rate of change in the combined RF signal, due to the high carrier frequency. In other words, the bandwidth of the message signal is usually in a size of megahertz while the carrier frequency is in region of sub-gigahertz or gigahertz. The I and Q samples allow us to sample or generate samples at a lot lower rate as the high-frequency part, due to the carrier, has been removed. This exact feature is used within software-defined radios, to allow a small bandwidth message signal to be generated and modulated in software and transmitted over the air with a high-frequency carrier.

- ## SDR hardware

To allow a software application to change the characteristics of an RF communication channel, the hardware has to be designed for dynamic adjustments and the processing of the I and Q samples. In principle, a universal hardware serves as an interface between the baseband and the RF. The waveform of the baseband signal that has to be transmitted, is fully generated through software, as well as a received baseband signal is fully processed and demodulated within software algorithms. In SDR, the processing power required for signal processing, including modulation, encryption, channeling, quadrature encoding etc., is outsourced to a universal host, eg. a PC



Software-defined radio RF frontend for reception.

Depending on whether the SDR is used for transmission or reception, the analog part, also known as the RF frontend, takes care of combining the baseband signal with a carrier frequency or extracting the baseband signal from the carrier frequency. An RF frontend for reception is shown in the previous figure. The frontend includes two tun-able RF filters, to filter away any unnecessary nearby frequencies, and a Low Noise Amplifier (LNA) and an Automatic Gain Control (AGC). The right part of the RF frontend takes care of the conversion from the IF signal, taken from the output of the AGC, to the complex baseband signal, given by its' I and Q signals. This is done by using a quadrature demodulation technique with a Local Oscillator, tuned to the desired carrier frequency. This quadrature demodulation implements the extraction of the I and Q samples from the complex equation, equation 1-9 . These I and Q signals are then sampled by the two ADC's in the SDR hardware.
Due to the quadrature conversion and splitting happening in hardware, the ADC's within the SDR can sample the I and Q samples at a lot lower rate than if the actual RF signal with the carrier frequency had to be sampled. For transmission this is the same, as the baseband signal to be transmitted can be generated and sent to the SDR hardware at a lower sample rate than the actual carrier frequency. Thereafter the RF frontend takes care of quadrature modulating the baseband signal with the desired carrier frequency.

- ## Modulation scheme summary

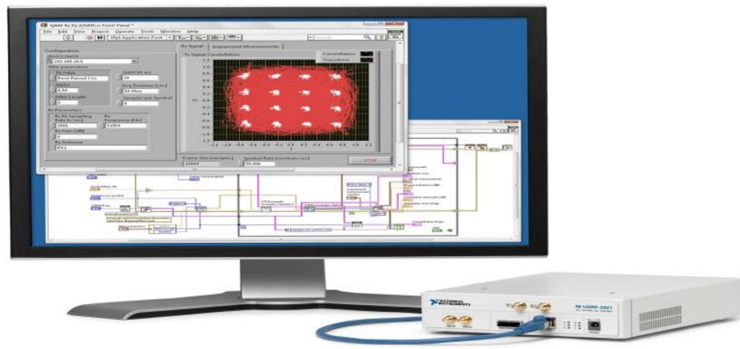
  As described, a specific modulation scheme is used to generate the baseband signal, given by its I and Q samples. When defining the I and Q samples for a specific message signal and modulation scheme, you have to consider the following two equations:

$$\tilde{s}(t) = a(t)e^{j\varphi(t)} = I(t) + jQ(t) \qquad 1\text{-}12$$

$$s(t) = I(t)cos(2\pi f_c t) - Q(t)sin(2\pi f_c t) \qquad 1\text{-}13$$

- ## RF Hardware in Wireless Communication Lab

In this course you will use software reconfigurable RF hardware from National Instruments to build a digital communication system. This hardware can be easily configured using LabVIEW. The RF Hardware used in the lab is the National Instruments USRP-2920. The following figure shows a USRP connected to a PC (running LabVIEW). This PC controls the USRP through the gigabit ethernet cable connecting the two together. In this section you will learn more about these modules.



1.1 NI USRP (Transmit)

Let's begin by taking a look at the NI USRP when acting as a transmitter. The USRP receives a waveform from the host PC with 16 bits of resolution sampled at up to 25MSamples/second. This signal is upconverted to a radio frequency (RF) before being sent to an amplifier and then transmitted over the air. For more information on the NI USRP, please refer to the help files associated with the USRP (**Help⇒NI-USRP Help⇒Devices⇒NI USRP-292x Specifications**)

1.2 NI USRP (Receive)

The NI USRP is also capable of receiving and does so in the following manner. The received signal is mixed with a desired carrier frequency in order to down-convert it to a complex IQ baseband signal sampled at 100 MSamples/second. The digital signal is then downsampled to a rate specified by the user and passed to the host PC for processing. When the ADC of the NI USRP samples at the full digitizer rate (100 MSamp/sec), it can acquire a bandwidth of 20 MHz. Sampling at such a high rate with high resolution (14 bits) produces a large amount of data. However, to acquire a signal with smaller bandwidth (a narrowband signal), it is sufficient to sample at a rate twice that of the signal bandwidth.

Decimation is the process of reducing the sampling rate of a discrete-time signal. In practice, this entails applying an anti-aliasing filter and throwing away some samples. The NI USRP features a digital down converter (DDC),
which allows it to perform decimation in hardware instead of in software (which can be considerably slower). Using the DDC, the USRP can acquire narrowband signals at sample rates much less than the full digitizer rate (100 MSamp/sec), thus reducing the memory required to store a waveform. For example, a waveform sampled at 100MSamp/sec for 1 second produces 100 times more data than the same waveform sampled at 1 MSamp/sec. The effective sample rate of the USRP (including DDC) ranges from a maximum of 25 MSamp/sec down to 200kSamp/sec and is directly configurable by the user.

Experiment 1
Introduction to LabVIEW

Objective

To be familiar with LabVIEW software and start to program using it

Background & procedure

● LabVIEW is a graphical language for programming math and signal processing applications

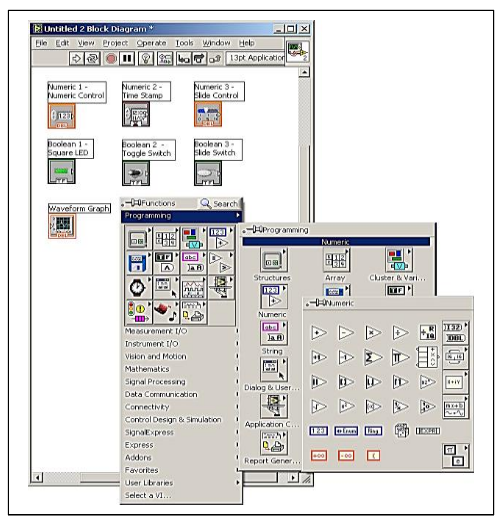
● "Code" in LabVIEW is in the form of a Virtual Instrument (VI)

● A VI consists of

– Front Panel: Top-level (user) interface

– Block Diagram: Actual structure of the code

● Terms: sub-vi, terminal, etc

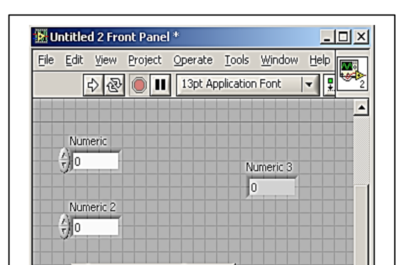


Front Panel

● User interface to the code (in block diagram)

– Controls (Input): Boolean controls, Numerical controls, etc.

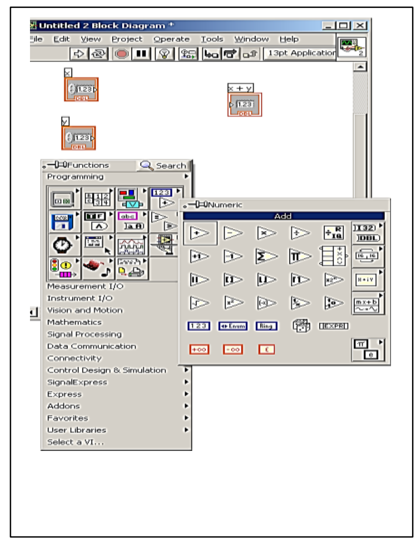
– Indicators (Output):Graphs, Charts, Numerical indicators, etc.



– Other

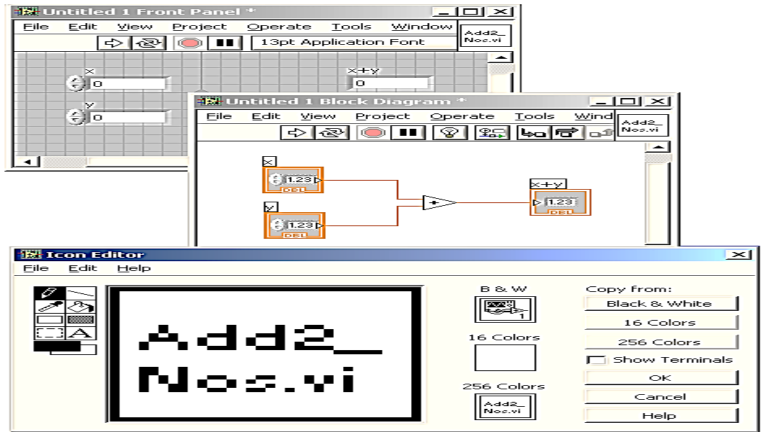
Block Diagram

Structure of code constructed of
– Inputs/Outputs: controls (indicators) from Front Panel are set as inputs (outputs) on the Block Diagram
– Function Pallete: Math, Signal processing, Loop structures, Arrays, etc

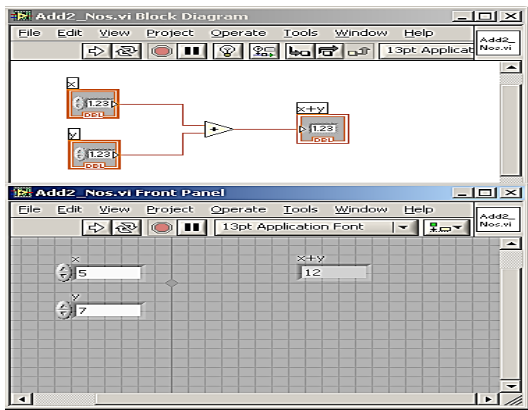


Example 1 : Construct a VI to add two numbers

–Step 1: Start >> LabVIEW >> Blank VI (from “Getting Started‟ screen.
– Step 2: On the Front Panel, add two “Numeric Controls‟ (inputs) and one “Numeric Indicator‟ (output).
– Step 3: Name them “x‟, “y‟ and “x+y”.
– Step 4: Switch to the Block Diagram – note the inputs and outputs (same names as Front Panel)
– Step 5: Right-click on the Block Diagram and choose “Add‟ under Numeric Palette – drag and drop on the Block Diagram.
– Step 6: Connect the inputs and outputs using the mouse for wiring.
– Step 7: Save the VI as *Add2_Nos.vi*
– Step 8: Modify the icon (top right corner) by right clicking on the icon and choosing “Edit icon‟
– Step 9: After editing the icon, right click on the icon again and choose “Show connector‟ and connect the inputs and output appropriately.



The final VI should look like this



● To run the VI

– enter x and y values in the Front Panel

– Click the icon in the top left corner of the Front Panel

Programming Structures

●LabVIEW (like C or Matlab) supports the following structures:

– If / Else

– For loop

– While loop

● These can be found in "Structures" in the Programming Palette, by right clicking on the Block Diagram

1- *If /Else Structure*

● LabVIEW is graphical, so code of the form below is written graphically

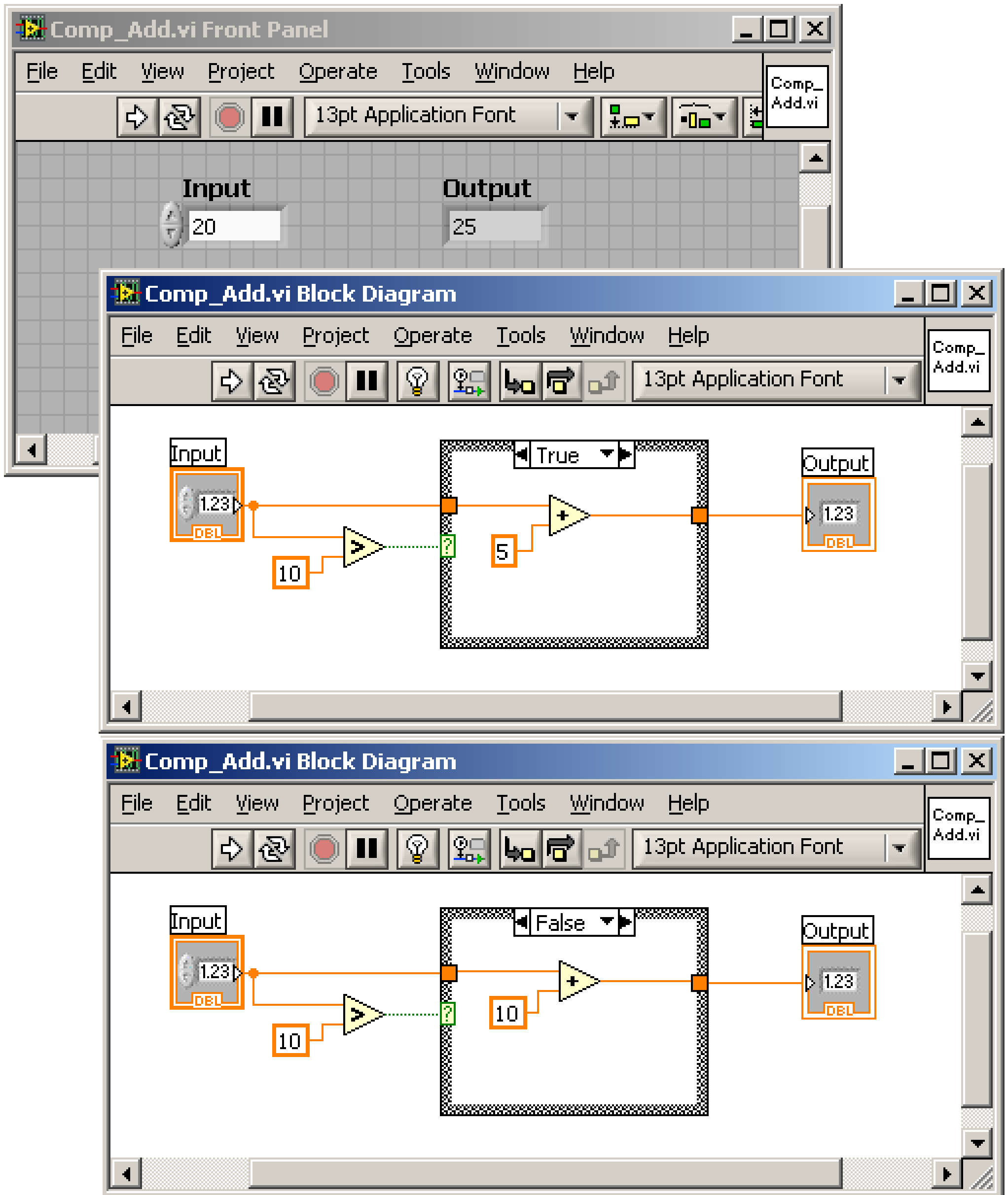
if{condition = true}
    {Program for true condition}
else (if{condition = false})
    {Program for false condition}

● Basic Steps

– Use a comparison to produce a true or false, then wire that result to the Case Structure
– Place code in both the True and False parts of the case structure selected at the top of the structure
– When the code runs, a true boolean value will run the True case, while a false will run the False case

Example 2 : Create a VI to add "5" to a number if it is greater than "10", else add "10".

– Step 1: Open a blank VI
– Step 2: On the Front Panel, insert a "numeric control" to obtain the input number and a "numeric indicator" for the ouput
– Step 3: In the Block Diagram, go to „Structures" in the Programming Palette (right click for the Palette)
– Step 4: Click on „Case structure", drag and drop it onto the Block Diagram. Adjust the size of the structure as needed
– Step 5: Insert a "*Greater?.vi*" from "Comparison" in the Programming Palette
– Step 6: Compare the input to a "constant" (Numeric << Programming Palette) set to 10.
– Step 7: If the condition (number > 10) = True, set the case structure to "True" and "Add" 5 to the number
– Step 8: If the condition (number > 10) = False, set the case structure to "False" and "Add" 5 to the number
– Step 9: Edit the icon and the connector, after saving the VI



2- <u>For Loop</u>

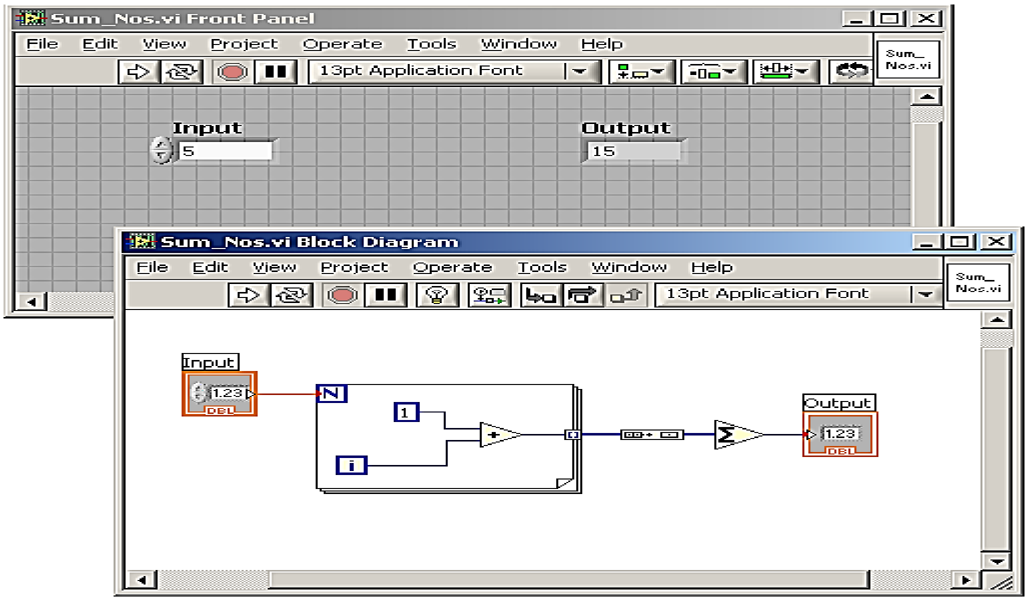
● ” For loop‟ in LabVIEW is the same as in other languages

– Each for loop has a “Loop Count‟ input to the loop and a “Loop Iteration‟ output inside the loop.
– Simply wire up the number of loops you require to Loop Count and put appropriate code inside loop.

Example 3: Create a VI to output the sum of numbers from 1 to the number input.

– Step 1: Open a Blank VI
– Step 2: In the Front Panel, insert a „numeric control‟ and name it “input‟ and a “numeric indicator‟ and name it “output.
– Step 3: On the Block Diagram, drag and drop a “For Loop‟ from Structures in the Programming Palette
– Step 4: Connect the “Loop Count‟ to “input‟ and the “Loop Iteration‟ to an “*Add.vi*‟ (the loop

iteration starts from 0 to (N-1)) and build an array with the results.
– Step 6: Sum the array elements.
– Step 7: Edit the icon and connector.

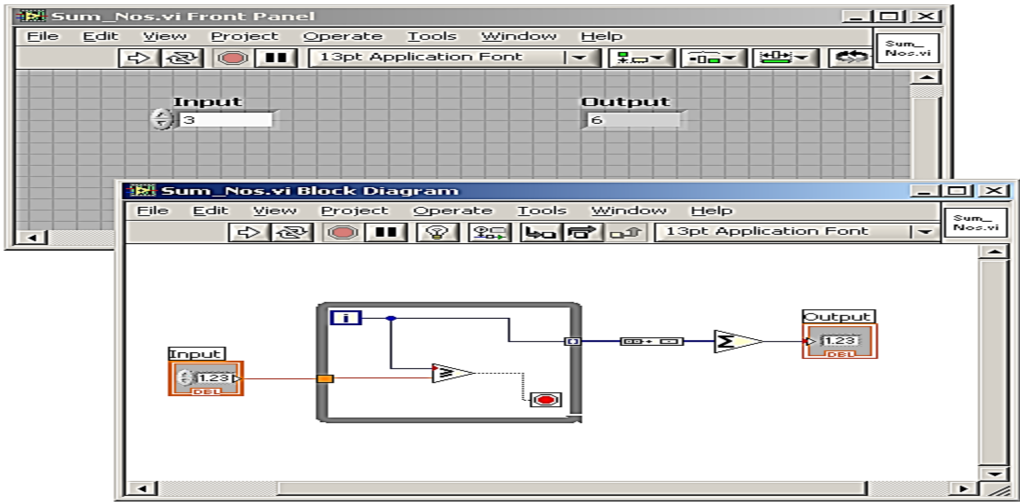


3- *<u>While Loop</u>*

- The While Loop in LabVIEW is similar to "For Loop"

– Loop runs until the stop condition is met.

Example 4 : Repeat the previous exercise with a While Loop that stops when iteration count is ≥input number.



Data Structures

- Arrays

– Multidimensional collections of like data

– Vectors, matrices, array of booleans, etc.

- Clusters

– Collections of unlike data used for conveniently transporting the data from one place to another

– Similar to the idea of a Struct in C or Matlab.

1- *Arrays*

- You have already been introduced to arrays – when talking about the "For" and "While" loops
- Can have arrays of virtually anything

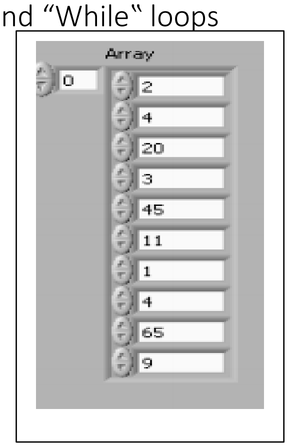
– Controls, indicators, numerics, booleans, etc.

- Can specify many dimensions
- Make an array – new VI

– Right-click on the Front Panel and choose Array under "Array, Matrix..."

– This is a shell array, place numeric control inside

– Add values inside array to use in Block Diagram



- Useful Array functions

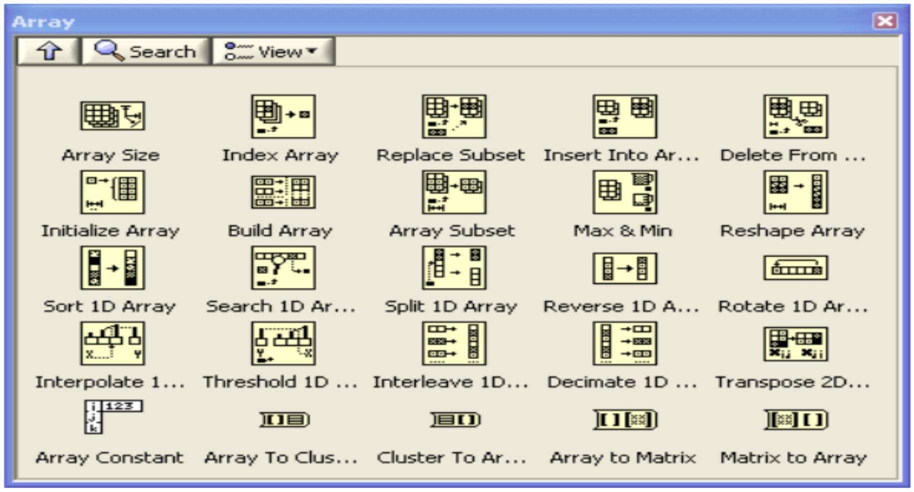
– Array Size – gives you an integer of the array size

– Build Array – allows you to concatenate arrays and other data together into one array

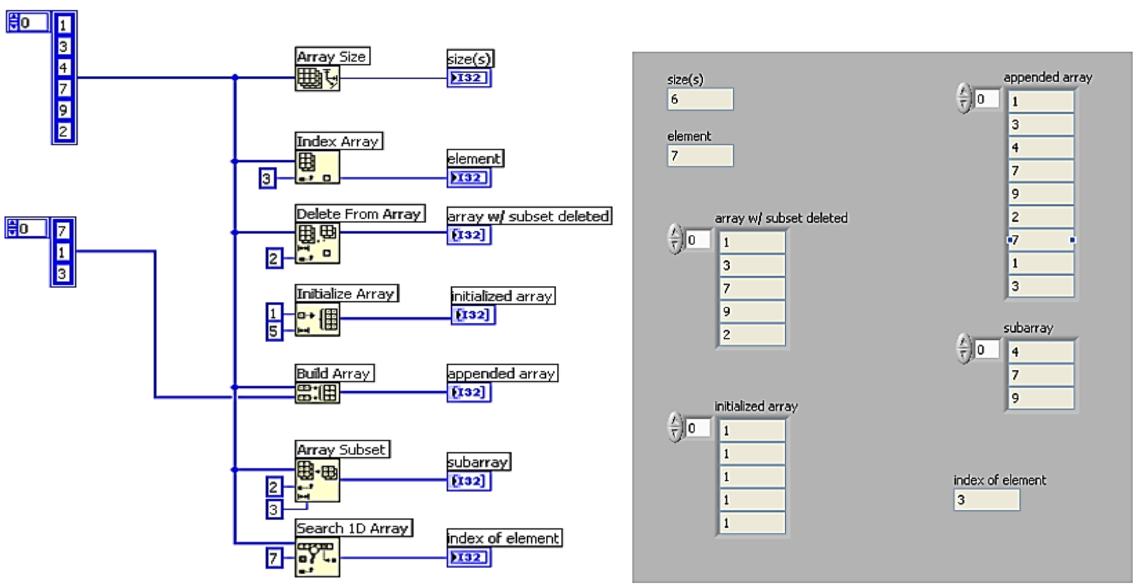
– Max & Min – gives you value and index of max/min

– Array Subset – allows you to resize an array given new dimensions and starting index

– Many more...

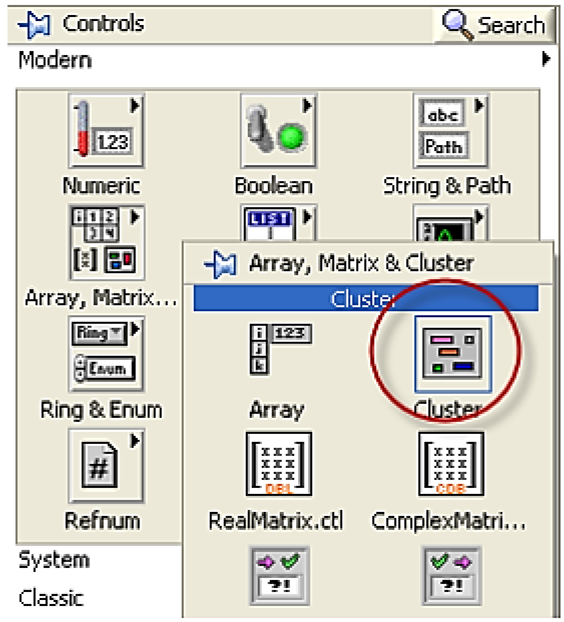


Example 5 : in this example we show how we can use these array functions and what they do

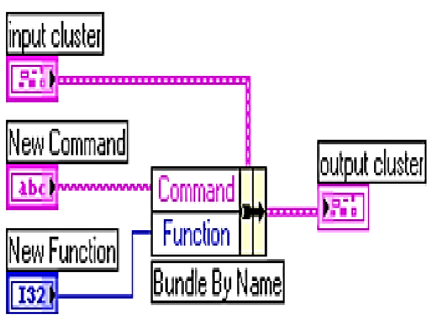


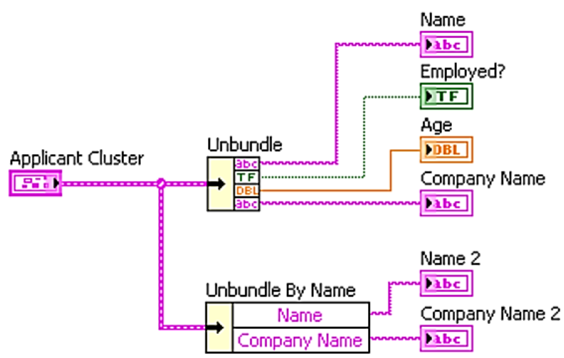
2- <u>Clusters</u>

- You can mix various types of data into a single cluster – mostly for passing to/from subVIs.



- In the Block Diagram, use Bundle and Unbundle to access elements of the cluster





Experiment 2
USRP as Sine Generator/ RF Analyzer

Objective

In this the lab you will learn about the NI RF hardware (USRP) to be used in the Wireless Communications Lab. Our goal is to generate a Sine wave using (USRP) , transmit it and receive it using another (USRP).

Background

When the baseband signal $m(t)$ is combined with the carrier frequency using complex multiplication, the actual RF signal to be transmitted is denoted as $s(t)$ as following:

$$m(t) \ = \ a(t)e^{j\varphi(t)}$$

$$s(t) = \ m(t) \ e^{j2\pi f_c t}$$

Where $a(t)$ and $\varphi(t)$ are the amplitude and phase of signal as a function of time

But if we use a constant signal $(t) = 1$, then the transmitted signal only contains the carrier part a follow

$$s(t) = e^{j2\pi f_c t}$$

Which can be written as

$$s(t) = \cos\left(2\pi f_c t\right) + j \sin(2\pi f_c t)$$

So to generate a cosine signal all we need is to generate a constant signal and apply it to the USRP modulator (multiply it by $e^{j2\pi f_c t}$)

Procedures

1- Transmitter

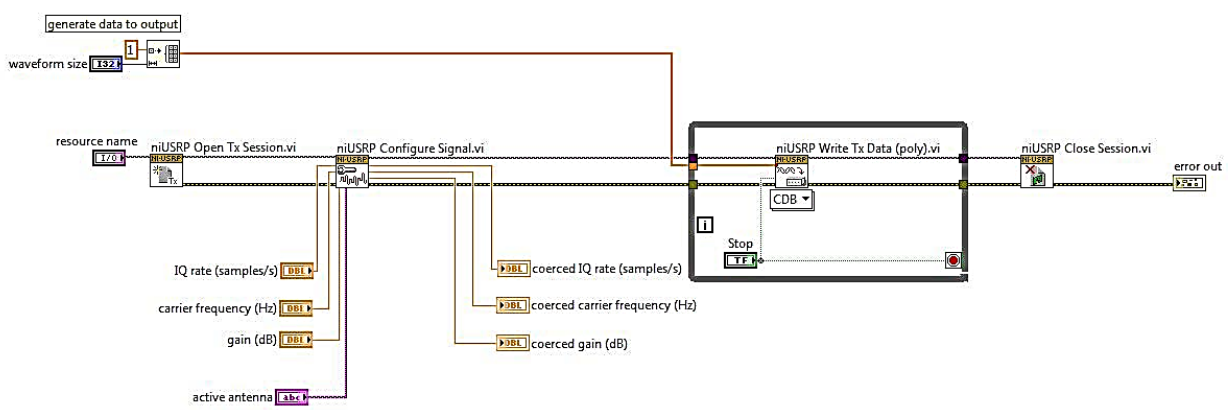
Use the following procedure to begin using the NI USRP to generate a signal:

1. Select **Start⇒Programs⇒National Instruments⇒LabVIEW 2013⇒LabVIEW** to launch LabVIEW. The LabVIEW dialog appears.

2. Click **New**, choose **Blank VI**, and click **OK** to create a blank VI.

3. Display the block diagram by clicking it or selecting **Window⇒Show Block Diagram** or using the shortcut (ctrl + E).

4. Navigate to the NI USRP VIs on the **Functions⇒Instrument I/O⇒ Instrument Drivers⇒NI-USRP⇒TX** palette. With LabVIEW running, click here to fin the **NI USRP TX** LabVIEW palette
5. Create the block diagram shown in the following figure by placing the four core VIs on the block diagram in the order they appear in the NI USRP RX functions palette.



6. Hover the cursor over the **device name** terminal on the niUSRP Open TX Session VI and right-click. Select **Create⇒Control** to create a front panel fiel where you specify the NI USRP device name.

7. Hover the mouse tool over the **IQ rate**, **carrier frequency**, and **gain** terminals of the niUSRP Configur Signal VI.

8. Right-click each terminal and select **Create⇒Control** from the shortcut menu to create frequency, IQ rate, and gain controls.

9. Display the front panel by clicking it or selecting **Window⇒Show Front Panel**. Fields are displayed in which you can specify a frequency, IQ rate, and gain.

Continuous waveform generation is controlled by means of a STOP button. A STOP button is typically used within a While Loop. This example places a While Loop around the Write TX Data VI so that signal generation continues until you click STOP.

Build a While Loop and STOP button by completing the following steps:

1. Display the block diagram by clicking it or selecting **Window⇒Show Block Diagram**.

2. Select the While Loop on the**All Functions⇒Structures** palette. With LabVIEW running, click here to fin the While Loop on the **Structures** palette.

3. Enclose the niUSRP Write TX Data (Poly) VI in the While Loop.

4. Hover the mouse tool over the **Loop Condition** terminal of the While Loop.

5. Right-click the **Loop Condition** terminal and select **Create Control** from the shortcut menu to create a **STOP** button on the VI front panel.

6. Create an error indicator by right-clicking on the **error out** terminal of the niUSRP Close Session VI and selecting **Create⇒Indicator**.

Now we need to produce a waveform to be transmitted. We will be transmitting a sinusoid at a carrier frequency that is specified by the **carrier frequency** control. This means that the baseband signal we will be passing to the NI USRP is simply a constant signal.

To generate this signal, complete the following steps:

1. Using the quick-drop functionality (ctrl + space) place the **Initialize Array** function down on the block diagram.

2. Right-click the **element** input of the **Initialize Array** function and select **Create⇒Constant**. Double-click on the constant that was created and change it to 1.

3. Right-click on the **dimension size** input and select **Create⇒Control**.

4. Wire the output of the **Initialize Array** function block to the **data** input of the niUSRP Write Tx Data (poly) VI.

5. Finally, wire the STOP button that was created for the While Loop to the **end of data?** input of the niUSRP Write Tx Data (poly) VI and select **CDB** from that VIs drop down menu.

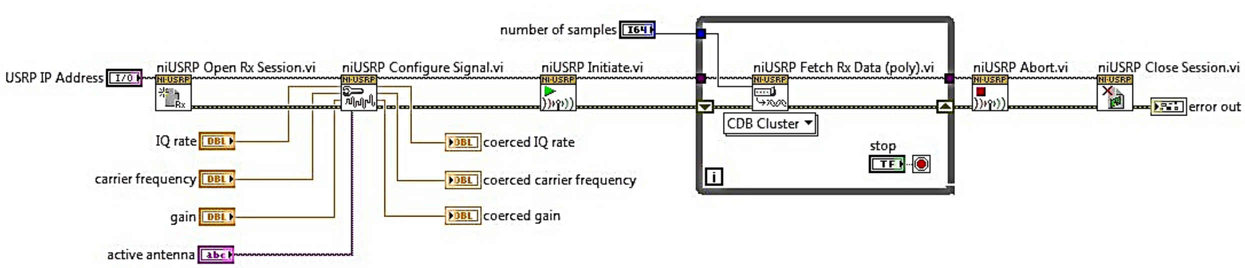
Remember that in order to operate your sine generation VI, you must properly configur the *USRP IP Address* for the USRP (an input to *niUSRP Initialize.vi*). You can find the IP addresses for all of the USRP devices connected to your PC by using the **NI-USRP Configuration Utility**

Now you need to set the transmitter parameters as following

A.	carrier frequency	2GHZ
B.	IQ sampling rate	200000 sample/sec
C.	gain (dB)	20
D.	active antenna	TX1
E.	**Waveform size**	10000

2- Receiver

build a Continuous Acquisition VI and it should look like the Continuous IQ Acquisition VI shown in the following figure. Add all of the appropriate controls and indicators necessary to control the sub-VIs in your code



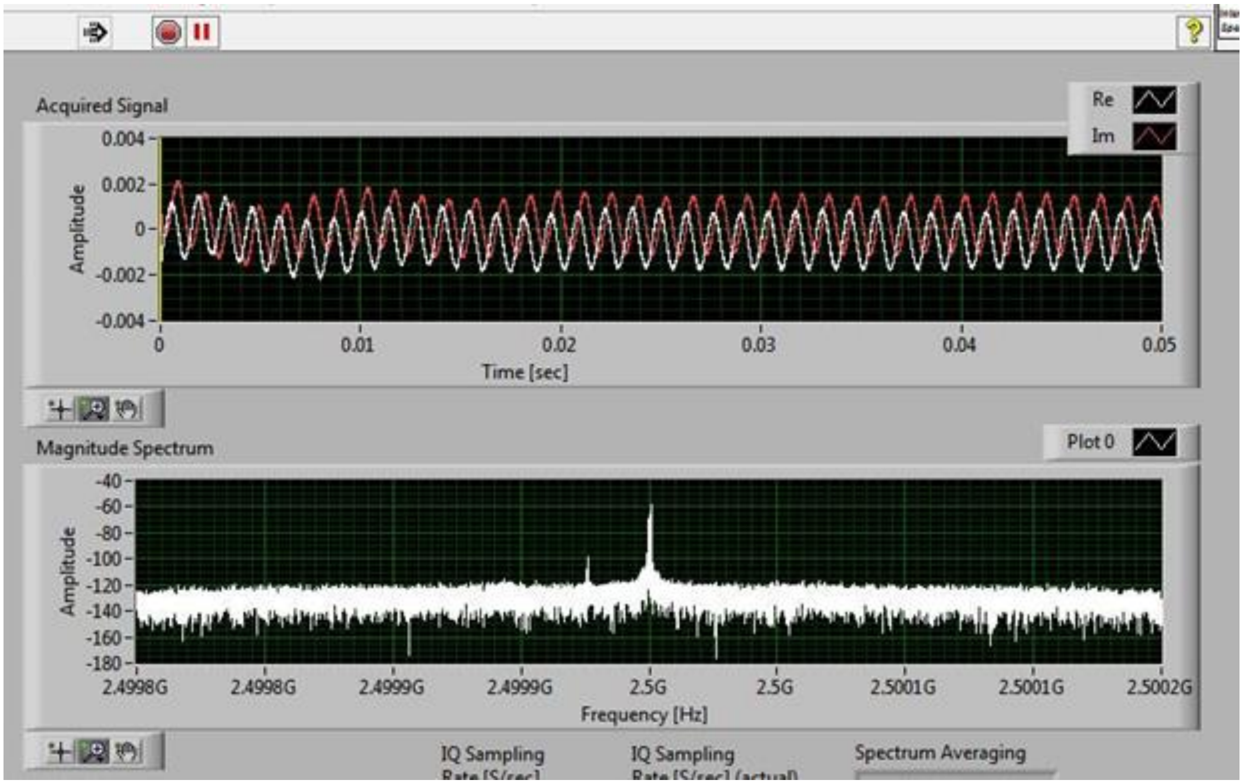
You can now get the real and imaginary parts of the received RF signal and its spectrum before getting back to baseband, by inserting **niUSRP EX Spectral Monitoring (Interactive).vi**. to your VI and set the parameters as that given for RX (shown in the next table)

This can be accessed from **Functions Palette⇒Instrument I/O⇒Instrument Drivers⇒ NI-USRP⇒Examples⇒niUSRP EX Spectral Monitoring (Interactive).vi**

You need to set the receiver parameters as following

A	**carrier frequency**	2GHZ
B	**IQ sampling rate**	200000 sample/sec
C	**gain (dB)**	20
D	**active antenna**	RX1
E	**Waveform size**	10000

Your result has to be something like that (***but with center frequency 2GHZ not 2.5GHZ***)


Acquired Signal
Re
Im
0.004
0.002
0
-0.002
-0.004
Amplitude
0
0.01
0.02
0.03
0.04
0.05
Time [sec]
Magnitude Spectrum
Plot 0
-40
-60
-80
-100
-120
-140
-160
-180
Amplitude
2.4998G
2.4998G
2.4999G
2.4999G
2.5G
2.5G
2.5001G
2.5001G
2.5002G
Frequency [Hz]
IQ Sampling
IQ Sampling
Spectrum Averaging

Experiment 3
Amplitude Modulation using USRP

Objective

To learn how to perform Amplitude modulation using LabVIEW & USRP

Background

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

<u>**Part 1: transmitter side**</u>

If

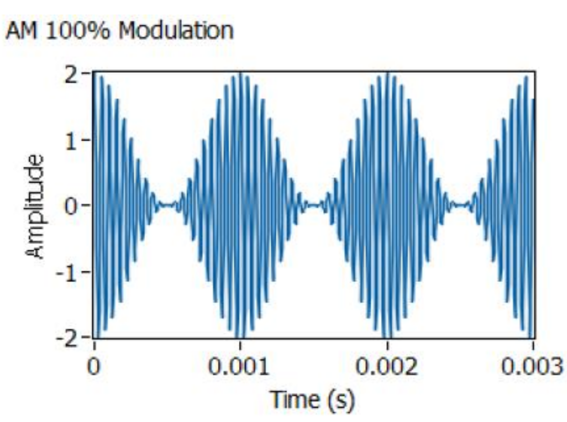
- $m(t)$ is a baseband "message" signal with a peak value m_p
- $A_c \cos(2\pi f_c t)$ is a "carrier " signal at carrier frequency f_c
- μ is the modulation index & takes the value in the range 0< μ <1

  $then$,The AM signal can be written as

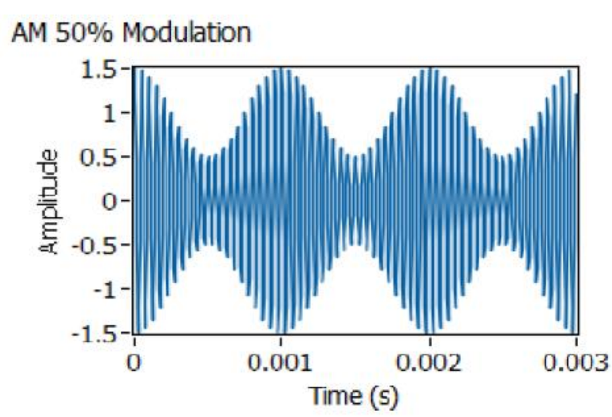
$$g(t) = A_c\,[1 + \mu \frac{m(t)}{m_p}]\cos(2\pi f_c t)$$

If $m(t) = m_p \cos(2\pi f_m t)$ then, the AM signal is

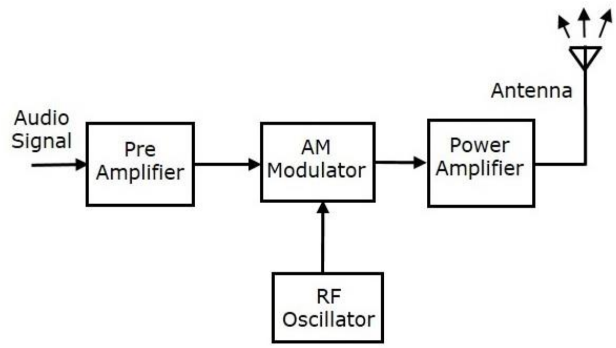
$$\begin{aligned} g(t) &= A_c[1 + \mu\cos(2\pi f_m t)]\cos(2\pi f_c t) \\ &= A_c\left[\cos(2\pi f_c t) + \frac{\mu}{2}[\cos(2\pi[f_c - f_m]t) + \cos(2\pi[f_c + f_m]t)]\right] \end{aligned}$$



AM Signal: Modulation Index = 1



AM Signal: Modulation Index = 0.5



<u>Part 2: Receiver side</u>

In general there is a phase & frequency offset between transmitter & receiver. The signal can be written as

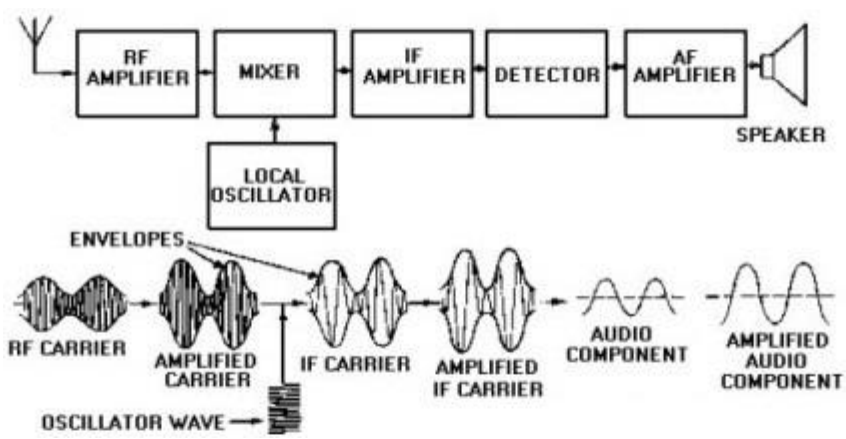
$$r_1(t) = A_c\,[1 + \mu\frac{m(t)}{m_p}]\cos(2\pi f_{IF} t + \theta)$$

Where $f_{IF}$is the intermediate frequency which is $f_{IF} = f_c - f_o$

The signal $r_1(t)$can be passed through a bandpass filter to remove interference from unwanted signals on frequencies nearf_c.

Demodulation of the signal 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 $r(t)$ is given by

$$A(t) = A_c\,[1 + \mu\frac{m(t)}{m_p}]$$

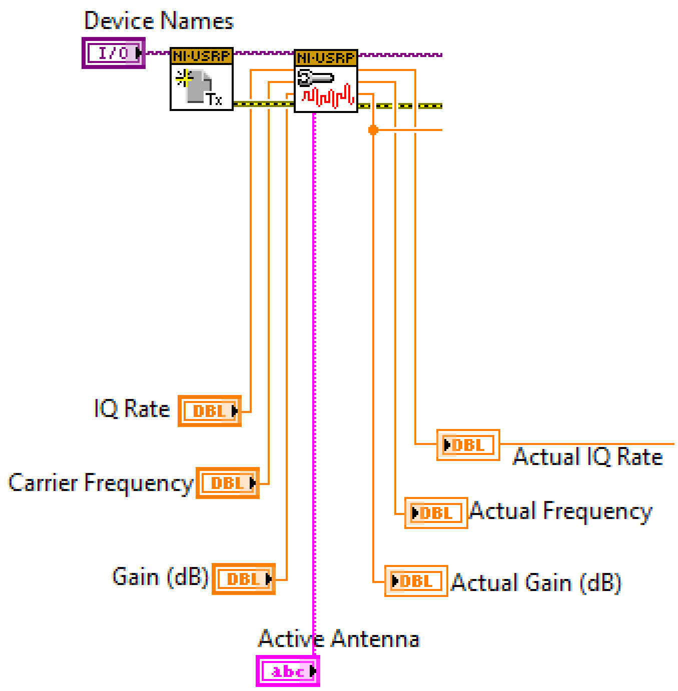


Procedure

<u>PART (1): The transmitter side</u>

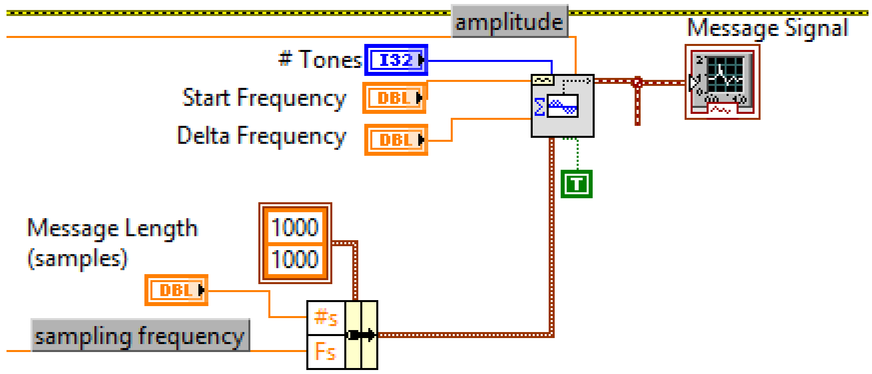
1. Create the following USRP parameters

	carrier frequency	915.1 MHz
	Message Length	100000 sample
	gain (dB)	20
	active antenna	TX1



2. In this part we want to generate the message signal as in the following table

IQ rate	200,000 sample
Modulation Index	1
Start Frequency	1 KHz
Delta Frequency	1 KHz
Number of Tones	1
Sampling frequency	= IQ rate

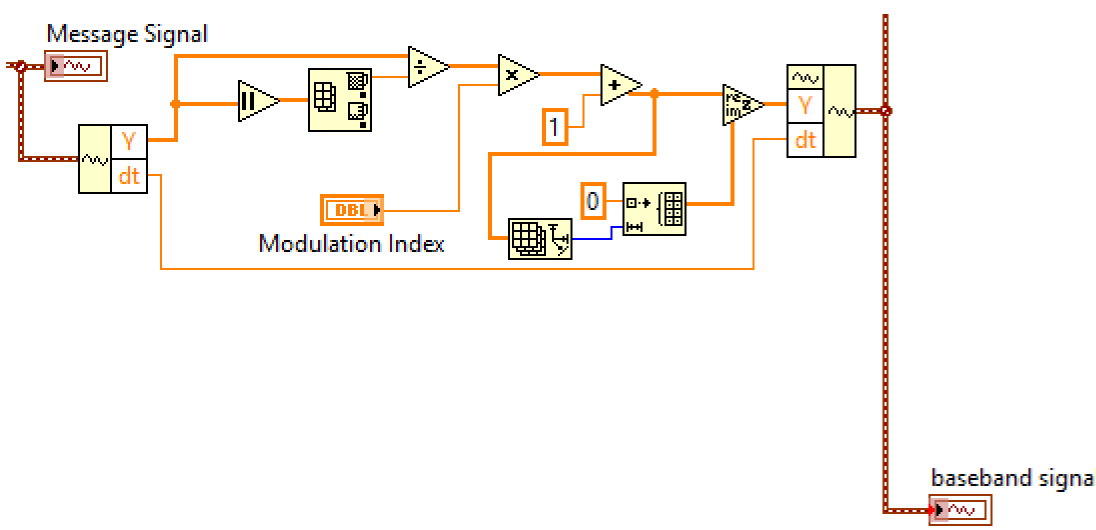


The implementation of AM message is as following

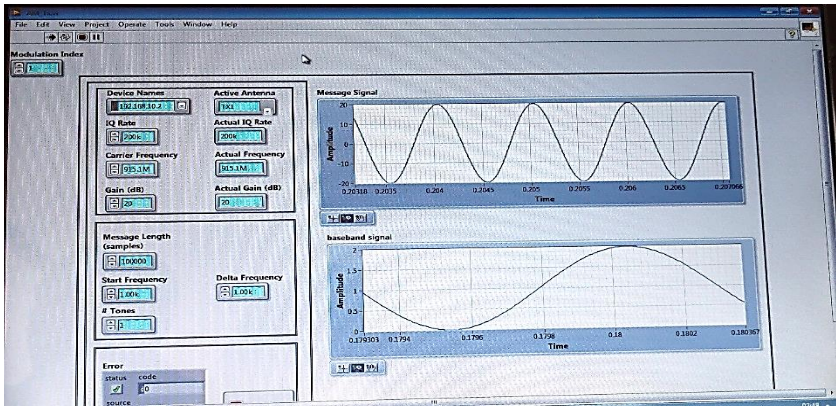
1. Based on the given template **AM_Tx_Template.VI** The message generator creates a signal that is the sum of a set of sinusoids of equal amplitude. You can choose the number of sinusoids to include in the set, you can choose their frequencies, and you can choose their common amplitude.
2. Get the data values of the generated signal by using the “Get Waveform Components” VI
3. You compute the peak value of the message by using both **Absolute Value VI & Array Max and Min VI**
4. Divide the signal by its absolute value then multiply by the modulation index then add 1

$$baseband\ signal = [1 + \mu \frac{m(t)}{m_p}]$$

5. The USRP is designed to transmit using a quadrature modulation approach. So in order to use the radio to transmit an AM signal, it is necessary to represent the signal as a complex sequence



The transmitter results have to be like the following figure.



PART (2): The Receiver side

1) Create the following USRP parameters

carrier frequency	915.05 MHz
Message Length	100000 sample
gain (dB)	0
active antenna	RX1

The implementation of AM message is as following

1) Based on the **AM_Rx_Template.VI** Get the data values of the signal received from the "niUSRP Fetch Rx Data" VI by using a "**Get waveform components" VI** so as to perform filtering operations
2) To remove unwanted interferences around carrier frequency, design a fifth order "**Chebyshev**" band-pass filter

Order	2
Type	Bandpass filter
high cutoff frequency	55 KHz
Low cutoff frequency	45 KHz
Ripple	**0.1 Db**
sampling frequency	equal to the "actual IQ rate"

3) Extract the real part of the complex filtered signal from the output of the Chebyshev band-pass filter using the **"Complex to Real/Imaginary" VI**

4) Use **"Absolute Value" VI** to take the absolute value of the real part of the filtered signal for full-wave rectification

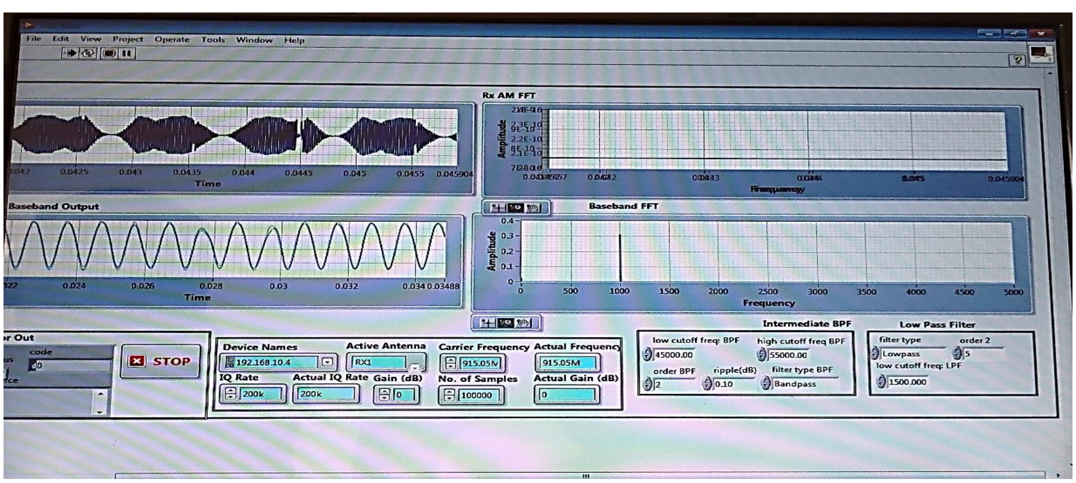
5) To filter out high frequencies to complete envelope detection, design a second order "Butterworth" low-pass filter

Order	5
Type	Low pass filter
Low cutoff frequency	1.5 KHz
sampling frequency	equal to the "actual IQ rate"

6- to graph the signal you have to get the waveform back from its data values so you have to use "**Build Waveform**" VI and to normalize the signal waveform you can use "**Normalize Waveform** "VI .

7- Plot both the demodulated signal waveform and spectrum.

Your results have to be like the following figure



Experiment 4
DSB using LabVolt instruments

NEW TERMS AND WORDS

1. **audio**

Signals that a person can hear.

2. **electromagnetic waves**

The radiant energy produced by oscillation of an electric charge.

3. **message signal**

Any signal that contains information

4. **Audio Frequency (AF)**

Frequencies that a person can hear. AF signals range from about 20 Hz to 20 kHz.

5. **Radio Frequency (RF)**

The transmission frequency of electromagnetic (radio) signals. RF frequencies are from about 300 kHz to the 1,000,000 kHz range.

6. **carrier signal**

A single, high-frequency signal that can be modulated by a message signal and transmitted.

7. **Modulation**

The process of combining the message signal with the carrier signal that causes the message signal to vary a characteristic of the carrier signal.

8. **demodulation**

The process of recovering or detecting the message signal from the modulated carrier frequency.

9. **Amplitude Modulation (AM)**

The process of combining the message signal with the carrier signal and the two sidebands: the lower sideband and the upper sideband.

10. **Frequency Modulation (FM)**

The process of combining the message signal with the carrier signal that causes the message signal to vary the frequency of the carrier signal.

11. **Phase Modulation (PM)**

    The process of combining the message signal with the carrier signal that causes the message signal to vary the phase of the carrier signal.

12. **angle modulation**

    The process of combining the message signal with the carrier signal that causes the message signal to vary the frequency and/or phase of the carrier signal.

13. **Double-Sideband (DSB)**

    an amplitude modulated signal in which the carrier is suppressed, leaving only the two sidebands: the lower sideband and the upper sideband.

14. **mixer**

    an electronic circuit that combines two frequencies.

15. **phase detector**

    an electronic circuit whose output varies with the phase differential of the two input signals.

16. **Envelopes**
17. the waveform of the amplitude variations of an amplitude modulated signal.
18. **sidebands**

    the frequency bands on each side of the carrier frequency that are formed during modulation; the sideband frequencies contain the intelligence of the message signal.

19. **AM**

    an amplitude modulated signal that contains the carrier signal and the two sidebands: the lower sideband and the upper sideband.

20. **bandwidth**

    the frequency range, in hertz (Hz), between the upper and lower frequency limits.

21. **harmonics**

    signals with frequencies that are an integral multiple of the fundamental frequency.

Description of the equipment

1) **Function generator A and B:**

   These two function generator is used to generator message signals.

2) **AM/DSB/SSB GENERATOR:**

   It is used to for amplitude modulation and generator either double side band or single side band waveforms for transmission.

3) **RF Noise Generator:**

   This can be used to generate RF noise.

4) **Power Supply:**

   It is used to supply power to equipment

5) **Dual Audio Amplifier:**

   Use for amplifier signals from the receiver to a proper listening level.

6) **AM/DSB Receiver.**

   This equipment receives the amplitude modulated signals and demodulate it before sending it to audio amplifiers

7) **Frequency Counter:**

   This equipment can be used for monitoring frequency of signals.

8) **Spectrum Analyzer:**

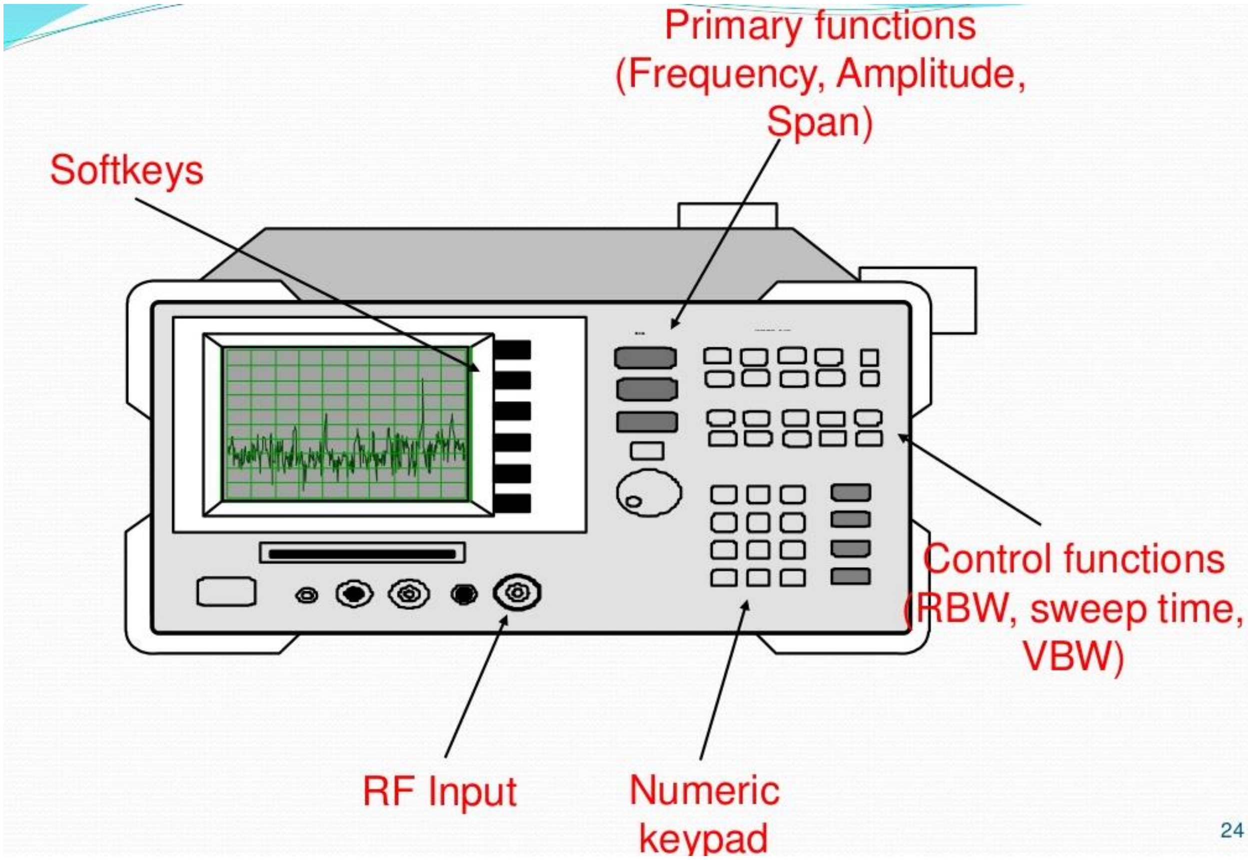
   This equipment is used for displaying signals in frequency base for analysis.

9) **Oscilloscope:**

   Use to display waveforms in time domain.

Spectrum Analyzer

A spectrum analyzer is a laboratory instrument that displays signal amplitude (strength) as it varies by signal frequency. The frequency appears on the horizontal axis, and the amplitude is displayed on the vertical axis. To the casual observer, a spectrum analyzer looks like an oscilloscope and, in fact, some lab instruments can function either as oscilloscopes or spectrum analyzers.



How to use a spectrum analyzer - the basics

There are a number of different controls and interfaces on a spectrum analyser. Although these instruments may appear to be complicated, it is possible to make good use of them after a little practice as it is necessary to use the controls correctly.

- ***The display***

  When looking at how to use a spectrum analyzer, one of the main elements of the unit is the display. The display has a graticule which typically has ten major horizontal and ten major vertical divisions.

The horizontal axis of the analyzer is linearly calibrated in frequency with the higher frequency being at the right hand side of the display. The vertical axis is calibrated in amplitude. This scale is normally logarithmic, although it is often possible to have other scales including linear ones for specialised measurements.

A logarithmic scale is normally used because it enables signals over a very wide range to be seen on the spectrum analyser - signals of interest may vary by 70dB, 80dB or more. Typically a value of 10 dB per division is used. This scale is normally calibrated in dBm (i.e. decibels relative 1 milliwatt) and therefore it is possible to see absolute power levels as well as comparing the difference in level between two signals.

In addition to the display of the spectrum, modern analyzers using digital technology often have soft keys to provide various functions around the edge of the display.

- ***Setting the frequency***

To set the frequency of a spectrum analyser, there are two selections that can be made. These selections are independent of each other and on different controls or entered via a keypad separately:

- ***Centre frequency:***

The centre frequency selection sets the frequency of the centre of the scale to the chosen value. It is normally where the signal to be monitored would be located. In this way the main signal is in the centre of the display and the frequencies either side can be monitored.

- ***Span:***

The span selection is the extent of the frequency coverage that is to be viewed or monitored when using the spectrum analyzer. The span may be given as a bandwidth per division on the graticule, or the total span that is seen on the calibrated part of the screen, i.e. within the maximum extents of the calibrations on the graticule. Another option that is often available is to set the start and stop frequencies of the scan. This is another way of expressing the span as the difference between the start and stop frequencies is equal to the span.

- ***Gain and attenuation adjustments***

There are other controls to use on a spectrum analyser. Most of these fall into one of two categories. The first is associated with the gain or attenuation of sections within the spectrum analyzer. If sections are overloaded, then spurious signals may be generated within the instrument. If this occurs then false readings will be given. To prevent this happening it is necessary to ensure that the input stages in particular are not overloaded and an RF attenuator is used. However if too much attenuation is inserted, additional gain is required in the later stages (IF gain) and the background noise level is increased and this can sometimes mask lower level signals. Thus a careful choice of the relevant gain levels within the spectrum analyzer is needed to obtain the optimum performance.

- ***Scan rate***

  The spectrum analyser operates by scanning the required frequency span from the low to the high end of the required range. The speed at which it does this is important. Obviously the faster it scans the range the faster the measurement can be made. However the rate of scan of the spectrum analyzer is limited by two other elements within the instrument. These elements are the filter that is used in the IF, and the video filter that may also be used to average the reading. These filters must have time to respond otherwise signals will be missed and the measurements rendered useless. Nevertheless it is still essential to keep the scan rate as high as is reasonably feasible to ensure that measurements are made as quickly as possible. Normally the scan rate, span and the filter bandwidths are linked within the instrument to ensure the optimum combination is chosen.

objective

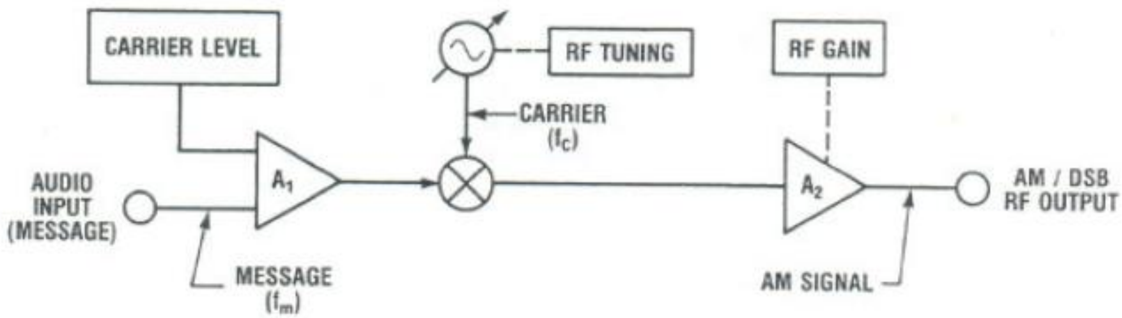
- Demonstrate the operations of the spectrum analyzer
- Demonstrate the transmission of the AM signals

Theoretical background

Amplitude Modulation (AKA AM) was the first modulation type to impress audio on an RF carrier. Prior to this, information was transmitted via on/off keying of a continuous wave transmitter using Morse code or some equivalent.

The block diagram of Figure 1 shows how an AM signal is produced by the AM / DSB / SSB Generator. A dc level (for the carrier level) is added to the message signal.

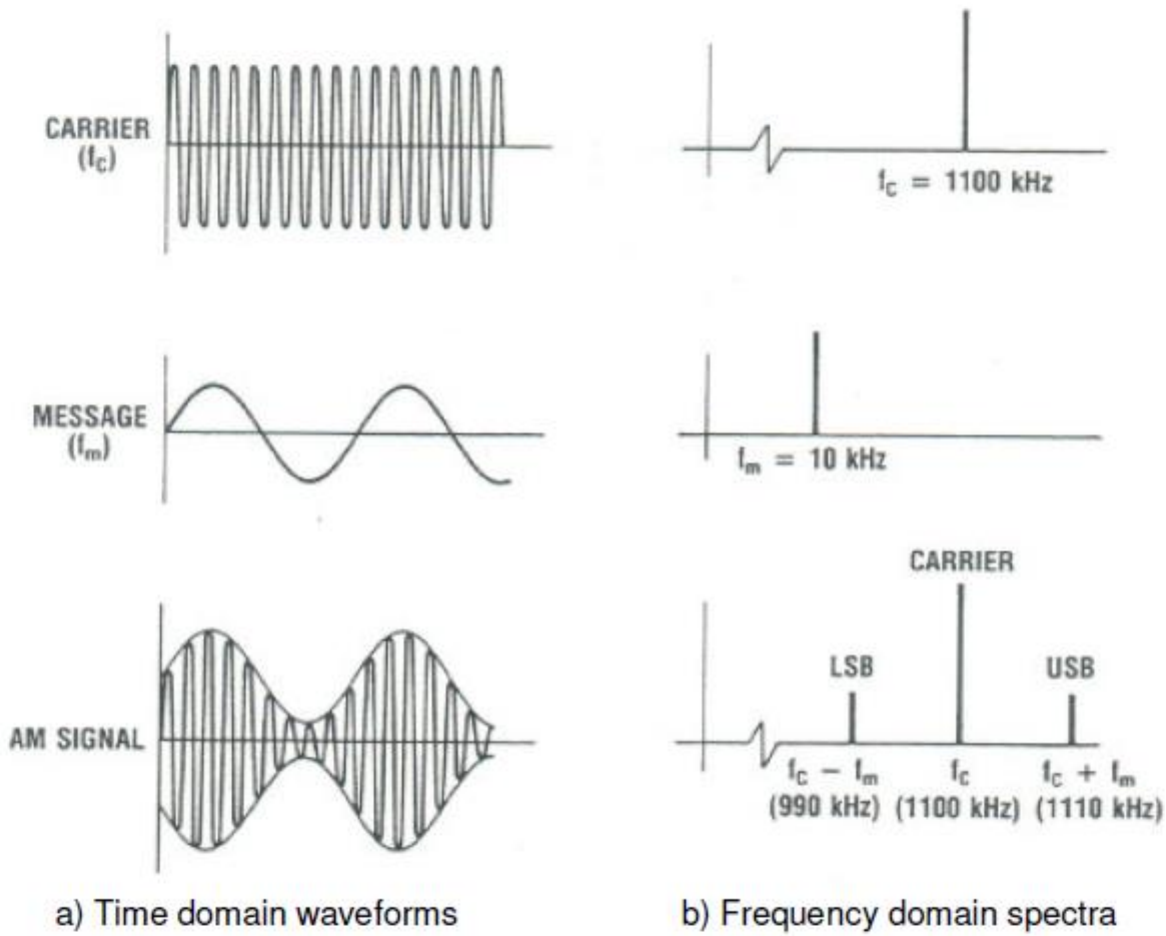
The resulting signal is mixed with the RF carrier to frequency translate the message signal, and is then amplified with the RF amplifier.



. Block diagram tor generating an AM signal.

Figure 1

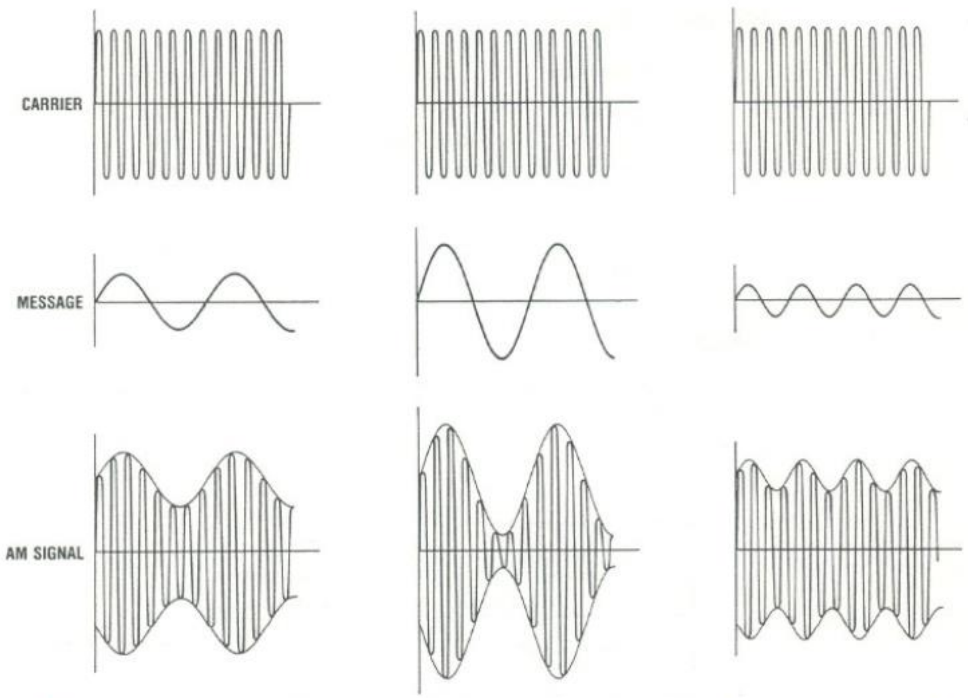
Figure1-2 shows the waveforms and frequency spectra for an 1100-kHz carrier modulated by a 1 0-kHz sine-wave.



Waveforms and spectra tor the AM signal

Figure 2

Notice that the information (message) has been impressed onto the carrier and that the envelope of the AM signal is an exact copy of the message signal. Also, the envelope varies at the same frequency as the message signal. The effect on the AM signal produced by different message signal amplitudes and frequencies is shown in Figure 3.



a) AM modulation index = 0.40 b) AM modulation index = 0.80 c) AM modulation index = 0.20

AM waveforms for different message signal conditions.

Figure 3

Experiment setup

AM/DSB/SSB GENERATOR	FREQUENCY COUNTER
FUNCTION GENERATOR A	SPECTRUM ANALYZER
FUNCTION GENERATOR B	OSCILLOSCOPE
POWER SUPPLY	DUAL AUDIO AMPLIFIER

Steps

Step 1:

This step is to set up the equipment in a manner where it will be easy to counter the appropriate modules together. We need to make sure all the level and gain control is set to the minimum to avoid any incidents.

Step2:

Adjust the function generator A to produce the following signal: A. set the signal to a sine wave B. Adjust the frequency to 10kHz C. Set the output level to get a signal of 400mVp-p

Step 3:

Set the carrier frequency to 1100kHz. This can be accomplished by connect the AM/DSB Output terminal to the frequency counter. Adjust the RF Tuning knob to get the frequency to be 1100kHz display on the frequency counter.

Step 4:

Connect the output of the AM/DSB Output terminal to channel 1 of the oscilloscope.

Step 5:

Display the signal from function generator A to the oscilloscope and connect it to the Audio Input on the AM/DSB/SSB Generator. This can be done by splitting the signal from function generator A using a T connector. Connect one to the oscilloscope and the other one to the Input on the AM/DSB/SSB Generator.

Step 6:

Change the signal of function generator A from a sine wave to a square wave and observe the waveform.

Step 7:

Used different forms of message signal from function generator an observe the result waveform such as triangular and saw-tooth.

Step 8:

Set the message signal back to a sine wave, and observe the change in the modulated waveform to the change in frequency to the message signal.

Step 9:

Observe the change in the modulated waveform as the amplitude of the message signal is varied.

Step 10:

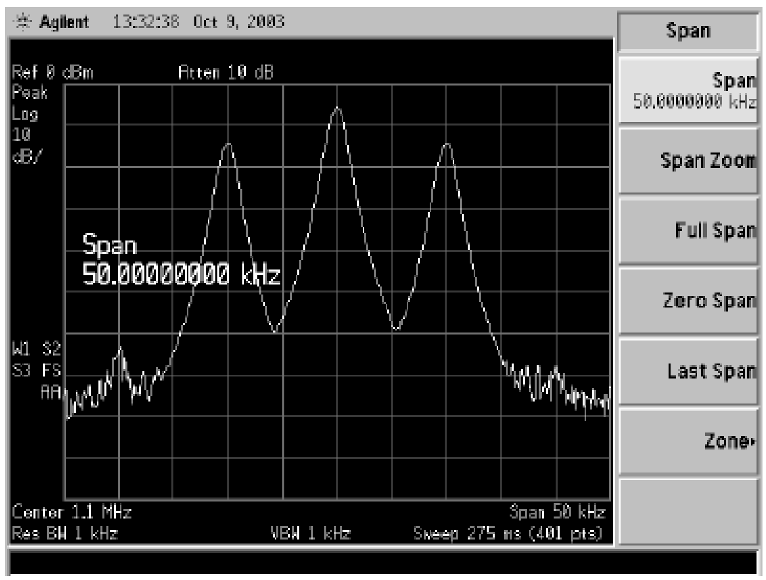
The change of the amplitude level of the information signal in step 9 corresponds to the variation of the modulation index.

Step 11:

Disconnect the oscilloscope and set up the spectrum analyzer. Set the center frequency of the spectrum analyzer to 1.1MHz. Span. 50-200 KHz

Step 12:

Display the modulated signal to the spectrum analyzer. This can be accomplished by connecting the AM/DSB Output to he Input of the Spectrum Analyzer.



Step 13:

Determine the difference between the display of the frequency response shown in step 12

Step 14:

Observe the changes of the modulated signal in the spectrum analyzer as the frequency of the message signal is varied.

Step 15: Observe the changes of the modulated signals in the spectrum analyzer as the amplitude of the message signal is varied.

Step 16:

Determine the effects of changes in the modulation index to the frequency spectrum.

Step 17:

Observe the change in the frequency spectrum when the message signal is changed from a sine wave to a square wave.

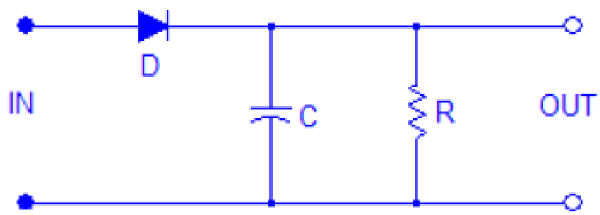
Reception of AM signals

Objective

- Demonstrate the envelop detector
- Demonstrate the sync detector
- Demonstrate the DSB modulation

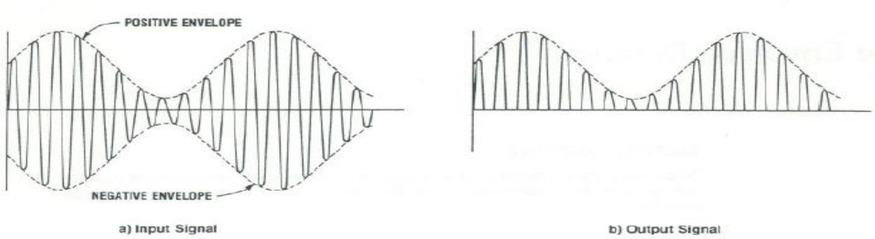
Theoretical background

Any circuit whose output follows the envelope of an AM signal can serve as a detector, and be used to demodulate the RF wave. one of the most widely used and simplest detectors is the non-linear charging circuit formed by a diode in series with the parallel RC network shown in Figure This kind of envelope detector is also known as a diode detector.



The Envelope Detector.

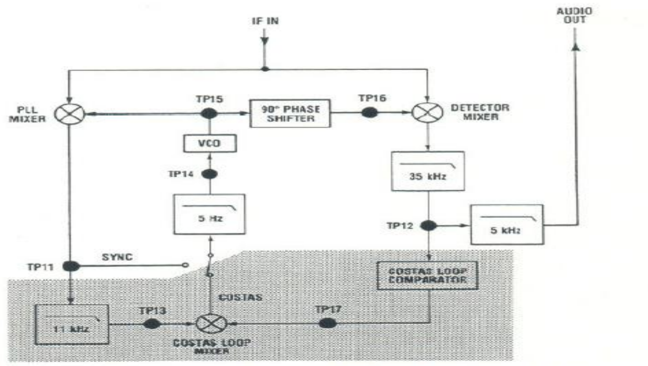
The circuit is designed to have a fast charge time and a slow discharge time, with the resistor controlling the discharge time constant.



Theoretical input and output signals of the envelope detector.

Among the other types of detectors available is the PLL (phase-locked loop) Synchronous detector. The functional block diagram of the SYNC detector used in the AM / DSB Receiver is shown in Figure below. This type of detector provides better detection of AM signals, and allows the RF TUNING to vary over a larger frequency range before the station is lost.

Coupled with the AGC circuit, it also allows greater variations in RF signal levels, and permits .modulation levels approaching 100%



Functional block diagram of the SYNC DETECTOR (shaded).

Experiment setup

AM/DSB/SSB GENERATOR	FREQUENCY COUNTER
FUNCTION GENERATOR A	SPECTRUM ANALYZER
FUNCTION GENERATOR B	OSCILLOSCOPE
POWER SUPPLY	DUAL AUDIO AMPLIFIER

Steps

Step 1

Set up the equipment in a manner where it will be easy to counter the appropriate modules together. We need to make sure all the level and gain control is set to the minimum to avoid any incidents.

Step 2:

Set up Function Generator A to send out a message signal of 1.0kHz and 200mV. If the modulation index (m) is equal to 1.0, the optimum value of the RC time constant for the 1.0kHz message signal is given by: RCoptimum=1/(m*2*π*fm)

Step 3:

Set up a carrier frequency of 1000kHz. This can be accomplished by using the AM/DSB/SSSB Generator. Open the top panel and connect TP13 to the frequency counter to continuously monitor the frequency. Set the RF Gain to IA cw and make sure the CARIER LEVEL knob is pushed in and at the MAX position.

Step 4:

Determine the LO frequency of the AM/DSB Receiver to receive the 1000kHz signal. Result: fLO = fIF + fc

Step 5:

Adjust the RF TUNING control on the AM/DSB Receiver to 1455kHz and connect the AM/DSB RF OUTPUT to the AM receiver. Select the ENV DETECTOR and place the AGC switch in the 1 (active) position.

Step 6:

Connect the IF Output of the AM/DSB Receiver to channel 1 of the oscilloscope and connect the Audio Output to channel 2. Set the time base control to .5ms/DIV and set the oscilloscope to trigger on the Audio signal.

Step 7:

Set up the measurement of the oscilloscope. Both the channel should be set on the dc coupling mode with channel 1 at 1V/DIV and channel 2 at .5V/DIV. Set the reference level line for channel 2 on the 2nd graticule line of the oscilloscope and position

Step 8:

Verify the carrier frequency at 1000kHz and describe the signals displayed on the oscilloscope. Step 10: Observe what happen when the CARRIER LEVEL control on the AM/DSB/SSB Generator between MAX and MN.

Step 9:

Observe the difference on the demodulated signal between the SYNC DETECTOR AND THE ENV DETECTOR. This can be done by increase the amplitude of FUNCTION GENERATOR A until the modulation index is approximately 100%. Then switch between) the SYNC DETECTOR AND THE ENV DETECTOR and observe what happens.

Step 10:

Repeat the previous steps using the DSB – SC modulation & observe the difference

Experiment 5

SSB using LabVolt Instruments

Objective

- Demonstrate the SSB generations
- Demonstrate the SSB receiver

Theoretical background

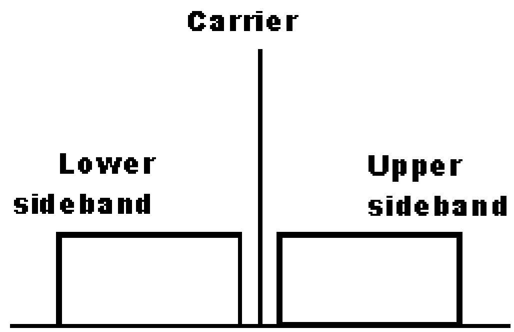
Single sideband modulation is a form of amplitude modulation. As the name implies, single sideband, SSB uses only one sideband for a given audio path to provide the final signal.

Single sideband modulation, SSB, provides a considerably more efficient form of communication when compared to ordinary amplitude modulation. It is far more efficient in terms of the radio spectrum used, and also the power used to transmit the signal.

In view of its advantages single sideband modulation has been widely used for many years, providing effective communications, as well as forms being used for some analogue television signals, and some other applications.

Single sideband modulation basics

Single sideband modulation can be viewed as an amplitude modulation signal with elements removed or reduced. In order to see how single sideband is created, it is necessary to use an amplitude modulated signal as the starting point.

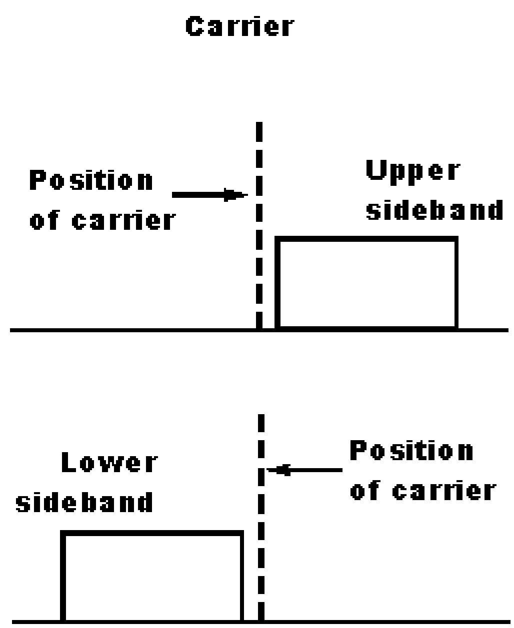


An amplitude modulated carrier

showing sidebands either side of the carrier

From this it can be seen that the signal has two sidebands, each the mirror of the other, and the carrier. To improve the efficient of the signal, both in terms of the power and spectrum usage, it is possible to remove the carrier, or at least reduce it, and remove one sideband - one is the mirror image of the other.

A single sideband signal therefore consists of a single sideband, and often no carrier, although the various variants of single sideband are detailed below.

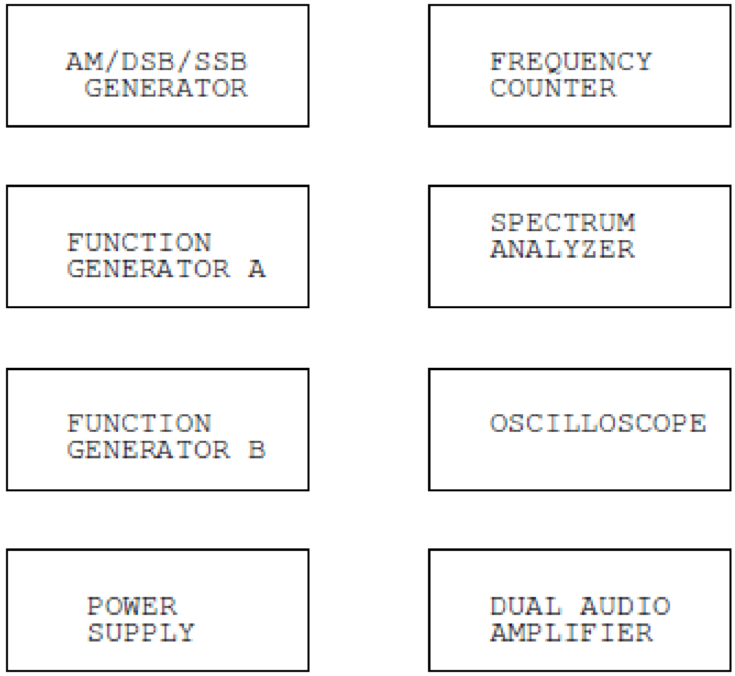


Single sideband modulation

showing upper and lower sideband signals

It can be seen that either the upper sideband or lower sideband can be used. There is no advantage between using either the upper or lower sideband. The main criterion is to use the same sideband as used by other users for the given frequency band and application. The upper sideband is more commonly used for professional applications.

Experiment setup



Steps

Step 1 :

Set up the equipment in a manner where it will be easy to counter the appropriate modules together. We need to make sure all the level and gain control is set to the minimum to avoid any incidents.

Step 2:

Generate a 2.5 kHz, 200mV sine wave from Function Generator A, and connect it to the AUDIO INPUT of the AM/DSB/SSB Generator.

Step 3:

Setup the AM/DSB/SSB Generator. The CARRIER LEVEL should be at the minimum position and push in the LINEAR OVERMODULATION position. Set the SSB RF GAIN (amplifier A3) at V2 turner cw.

Step 4:

Adjust the BFO Output to 452.5 kHz using the BFO TUNIGN knob.

Step 5:

Adjust the FVO TUNING control to measure a frequency of 4355 kHz at VFO OUTPUT

Step 6:

Prepare the Spectrum Analyzer ready for measurement. The center frequency should be center around ,5MHz.

Step 7:

Connect the SSB section to the input of the spectrum analyzer. Use the Tuning knob to place the spectrum analyzer of 455 kHz in the center of the screen.

Step 8:

Find the best setting to place the display in the center of the spectrum analyzer.

Step 9:

Sketch out the frequency spectrum.

Step 10:

Observe the frequency spectrum of the signal at the IF OUTPUT (terminal 4) and describe what happened.

Step 11:

Observe and explain what happened as fBFo approaches 455 kHz.

Step 12:

Observe what happens when the message signal frequency is increased to 3.5 kHz.

Step 13:

Readjust the message signal to 2.5 kHz and tune the BFO to obtain 457.5 kHz. Observe what happens as faro is increased from 455 to 457.5 kHz.

Step 14:

Of this step is to vary faro between 453 and 457 kHz and observe what happens in the spectrum response as faro varies between 453 and 457 kHz.

Step 15:

Set the spectrum analyzer up for the next measurement. Select the span to 1 MHz. Make sure SSB RF OUTPUT is at 455 kHz.

Step 16:

Select the right setting to make sure the image is centered.

Step 17:

Observe and describe the frequency spectrum using frequency span of 2 kHz/V.

Step 18:

Vary fBFo between 453 and 457 kHz and observe what happens.

Step 19:

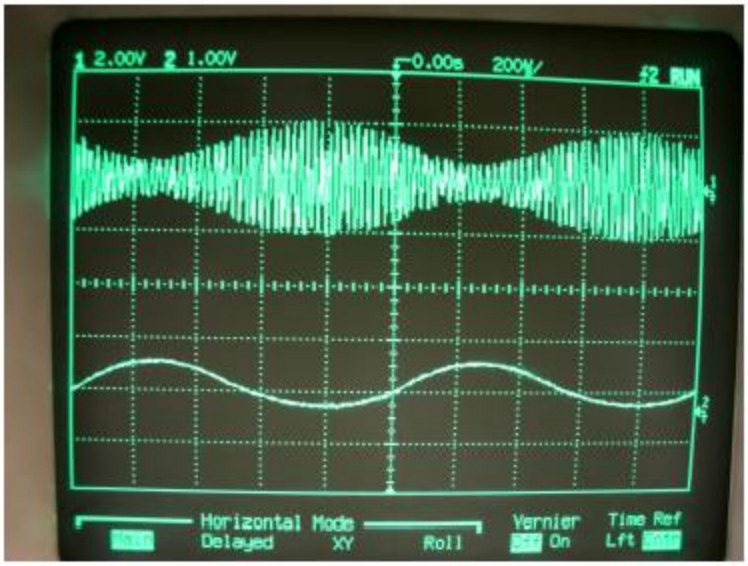
Conclude the effect caused by adjusting fBFo at 452.5 or 457.5 kHz.

Step 20:

Determine the time waveform for an SSB signal modulated by a single-tone sine wave.

Step 21:

Observe the waveform of the SSB signal at the SSB RF OUTPUT of the AM/DS13/SSB Generator when feFo = 452.5 kHz. Compare the results with the predictions of step 20.



Time waveform from SSB Generator

Experiment 6
Frequency Modulation

Objective

To provide you an experience in programming the USRP to act as an FM transmitter or a receiver.

Background

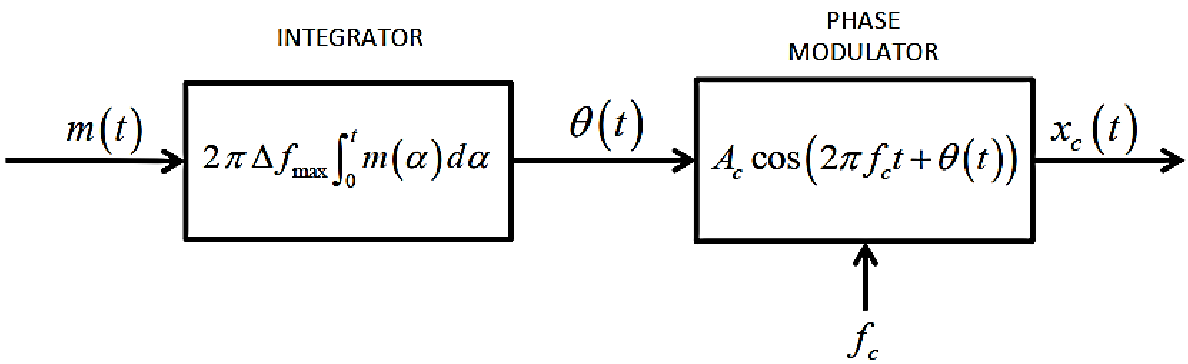
1- Frequency modulation

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.

Of the two forms of angle modulation- frequency modulation (FM) and phase modulation (PM), this lab would focus on FM modulation, and provide you with a nice illustration of the utility of the easier to implement software-defined radio approach. For angle modulated signal as following

$$x_c(t) = A_c \cos(2\pi f_c t + \theta(t)) \qquad 6\text{-}1$$

Where A_c is the amplitude of the carrier signal and f_c is the carrier frequency.

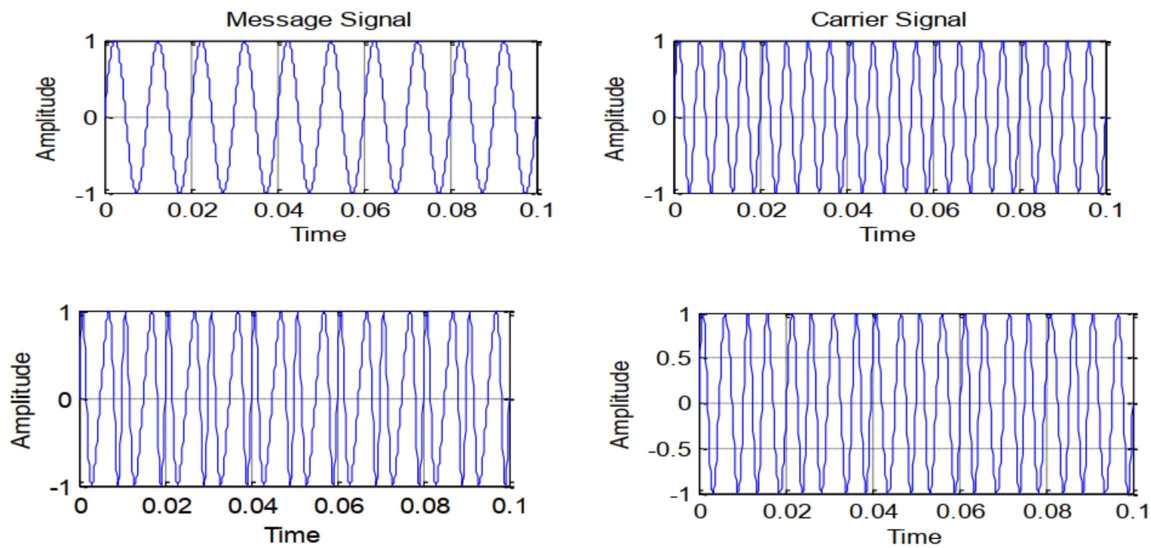


Generating a FM Signal Using a Phase Modulator

In FM, the instantaneous phase $\theta(t)$ of the carrier is varied linearly with the message signal (t) . Thus, the instantaneous frequency $f_i(t)$ of the carrier signal varies linearly with the message signal $m(t)$ and can be written as

$$f_i(t) = f_c + k_f m(t) \qquad 6\text{-}2$$

Where k_f is the "frequency sensitivity" of the FM modulator.



$k_f = 1$ $\qquad$ $k_f = .5$

The maximum instantaneous frequency change from the carrier frequency produced by the modulating signal is given by

$$\Delta f_{max} = \max|f_i(t) - f_c(t)| = k_f \ \max\ |m(t)| \qquad 6\text{-}3$$

Given the instantaneous frequency, we can find the total instantaneous angle $\emptyset(t)$, of the carrier by integrating the instantaneous frequency as

$$\emptyset(t) = 2\pi \int_0^t f_i(\alpha)\, d\alpha = 2\pi f_c(t) + 2\pi\Delta f_{max} \int_0^t m_n(\alpha)\, d\alpha \qquad 6\text{-}4$$

Where $m_n(t)$ is the normalized message signal defined as

$$m_n(t) = \frac{m(t)}{\max|\, m(t)|} \qquad 6\text{-}5$$

Using Equation (6-4), the FM signal can be expressed in terms of its instantaneous frequency and we can re-write Equation (6-1) for the FM signal, in terms of its instantaneous frequency

$$x_c(t) = A_c \cos(2\pi f_c\, t + 2\pi \Delta f_{max} \int_0^t m_n(\alpha)\, d\alpha) \qquad 6\text{-}6$$

Note that the second term in (6-6) represents of the FM signal. Thus, the instantaneous phase is the integral of a normalized message multiplied by $2\pi\Delta f_{max}$

To create an FM signal using the USRP, the complex-valued signal is formed, where

$$\breve{g}(t) = A_c\, e^{j(2\pi f_c\, t + 2\pi \Delta f_{max} \int_0^t m_n(\alpha)\, d\alpha)} \qquad 6\text{-}7$$

2- Frequency demodulation

An FM demodulator recovers the message signal from the received FM waveform. This requires a circuit that produces an output that is linearly proportional to the instantaneous frequency of the input FM signal. FM demodulation is much easier to carry out using the USRP rather than conventional hardware.

Consider that the signal $x_r(t)$ received ideally is the same as the transmitted signal is the same as $\breve{g}(t)$,

$$x_r(t) = A_c\, e^{j(2\pi f_c\, t + \theta(t))} \qquad 6\text{-}8$$

Now to extract the message from el modulated signal we need first to *differentiat*e the real component of the received signal

$$e(t) = -K_D \left[2\pi f_c + \frac{d\theta}{dt}\right] \sin(\, 2\pi f_c\, t + \theta(t)) \qquad 6\text{-}9$$

Where $K_D = 2\pi A_c$ is the frequency discriminator gain constant.

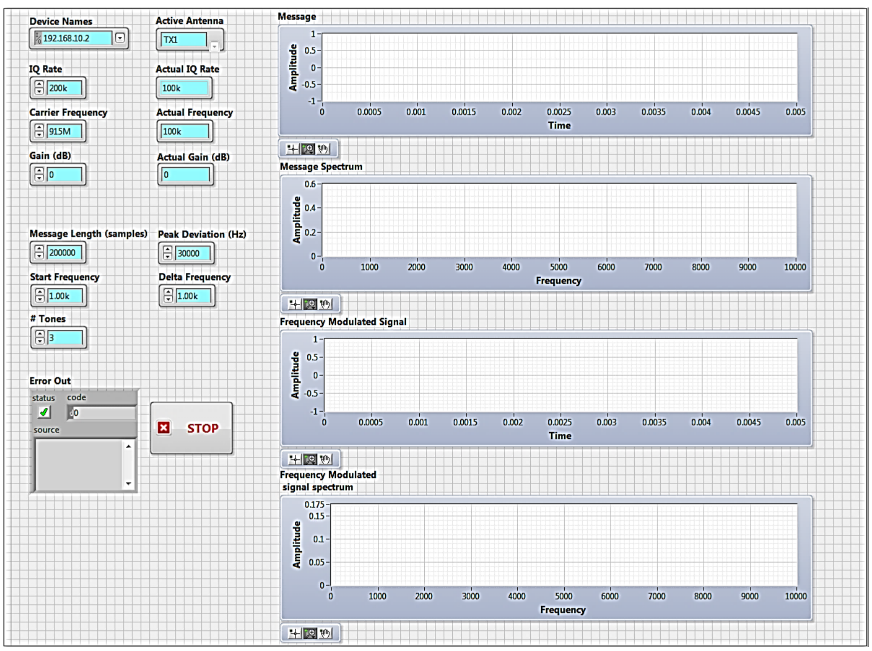
Now it’s clear that to get the normalized message we have to use envelope detector and divide the output by $2\pi\Delta f_{max}$.

Procedure

Transmitter

A template for the transmitter has been provided in **FM_Tx_Template.vi**. The Basic Multi-tone VI is used as the "message generator" to produce a message signal consisting of a chosen number of tones, start frequency (frequency of first tone) and delta frequency (increment in frequency for next tones w.r.t. frequency of first tone). These values can be selected using front-panel controls.

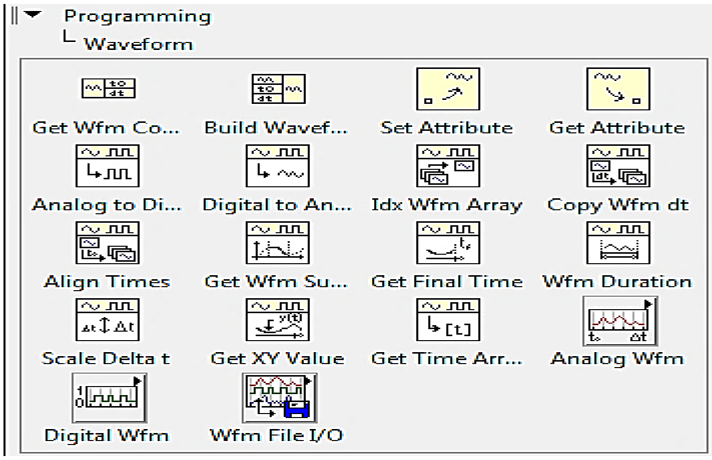
The following figure shows the front panel of **FM_Tx_Template.vi**



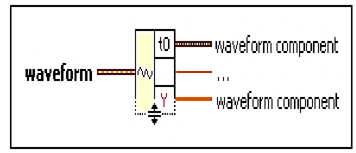
Your task is to add blocks as needed to produce the frequency modulated signal $\breve{g}(t)$ (of Equation 6-7 , and then to pass this signal to the "Write Tx Data block".

Step-by-step LabVIEW setup to create a FM modulated signal using USRP:

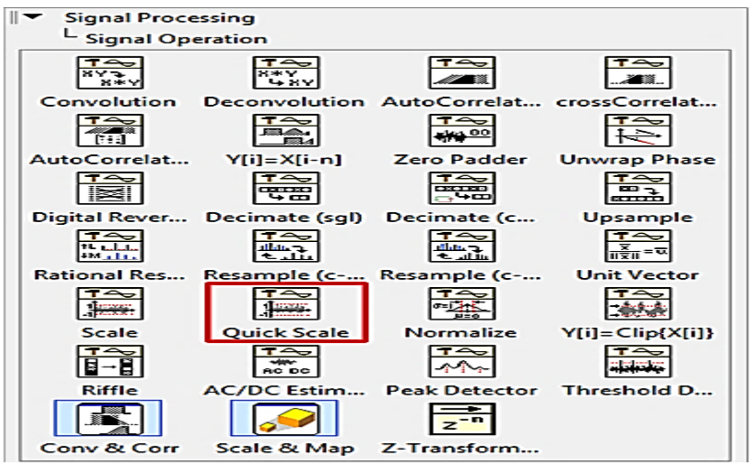
1- Obtain the data values and sampling time interval of the analog message signal using "Get Waveform Components" VI, and get the normalized message signal by using the "Quick Scale" VI as shown in the following figures

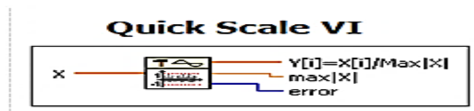


Get Waveform Components (Analog Waveform) Function



Get Waveform Components VI

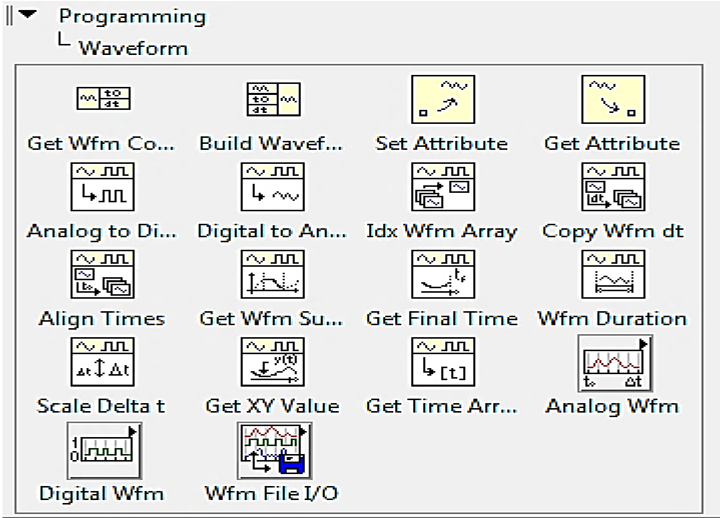




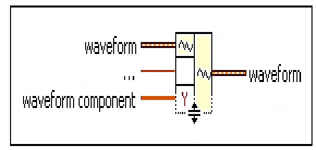
Quick Scale VI

2- Convert the data values of the scaled message signal into a waveform using “Build waveform” VI (as shown in figure) by setting the sampling time same as that of the original message signal. Plot the scaled message waveform using the waveform chart provided. This would allow you to do a comparison with the demodulated signal waveform charts in the receiver.

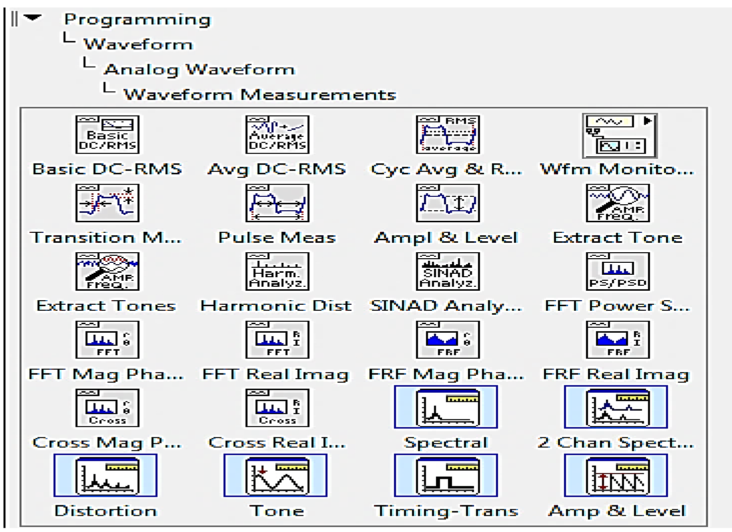
3- You may do a similar comparison for the spectrum of the message waveform and its demodulated version. To get the message spectrum, connect the message waveform (original or scaled) to the FFT Power Spectrum and PSD VI found in Signal Processing palette, and then connect the output to a waveform graph.



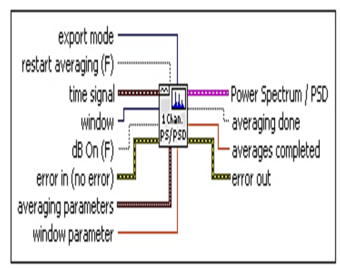
Build Waveform (Analog Waveform) Function



Build Waveform VI



FFT Power Spectrum and PSD VI



FFT Power Spectrum and PSD VI

4- Next, you have to integrate the normalized signal $m_n(t)$before multiplying by $2\pi\Delta f_{max}$ in equation (6-7). but we deal now with discrete data values so the integration transforms to summation using the following equation.

$$\int_0^t m_n(\alpha)d\alpha = \sum_{k=0}^{n} m_n(nT)T \qquad \text{6-10}$$

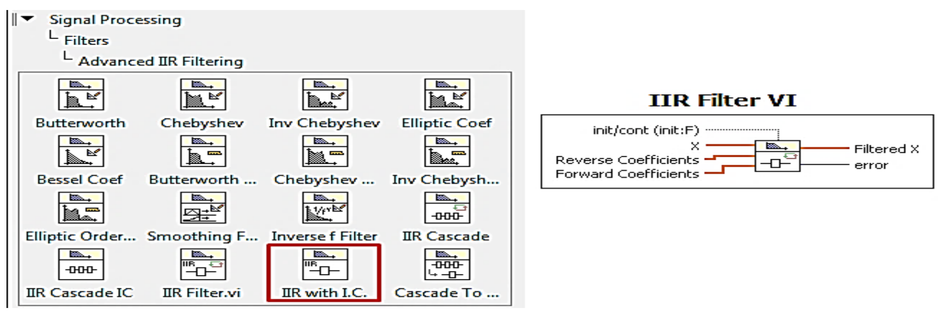
where is the reciprocal of the IQ sample rate. If we consider the equation

$$y[n] = \sum_{k=0}^{n} m_n(nT)T \qquad \text{6-11}$$

Then

$$y[n] - y[n-1] = \sum_{k=0}^{n} m_n(nT)T - \sum_{k=0}^{n-1} m_n(nT)T = x(nT)T \qquad 4\text{-}12$$

Equation (6-12) is a recurrence relation, or a difference equation of an IIR filter with “forward coefficients” array of and a “reverse coefficients” array of, and can be implemented using “IIR filter with initial conditions” VI .

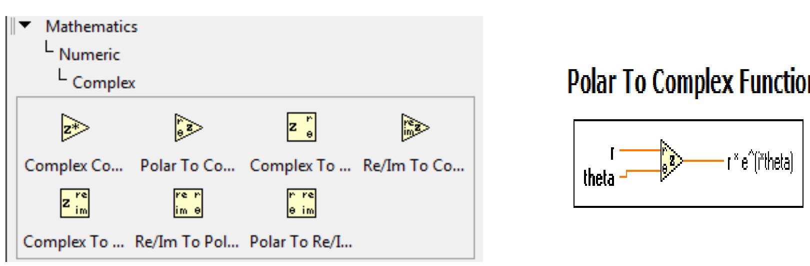


IR Filter with Initial Conditions V

5- Multiply the discrete filtered by $2\pi\Delta f_{max}$, which produces

$$\theta(t) = 2\pi\Delta f_{max} \sum_{k=0}^{n} m_n(nT)T \qquad 6\text{-}13$$

6- Form the complex value frequency modulated signal $\breve{g}(t)$ by utilizing the “Polar to Complex” VI . In this case, inputs to the polar to complex VI “**r”** and **theta** are A_c and $\theta(t)$ respectively.



Polar To Complex VI

7- Remember that the output signal generated by the IIR filter is a sequence. You need to convert it back into an analog waveform before sending it to the buffer, “Write Tx Data” VI. Use “Build Waveform” VI to convert the data values into a waveform by setting in the sampling interval of the frequency modulated signal as the product of the sampling interval of the original message signal and the term $2\pi\Delta f_{max}$.

8- Write the frequency modulated waveform onto the buffer, “niUSRP Write Tx Data” VI.

9- Plot the frequency modulated waveform using the waveform chart provided. Get the frequency modulated signal spectrum by connecting the waveform to the “FFT Power Spectrum and PSD” VI and then connect the output to the waveform graph provided.

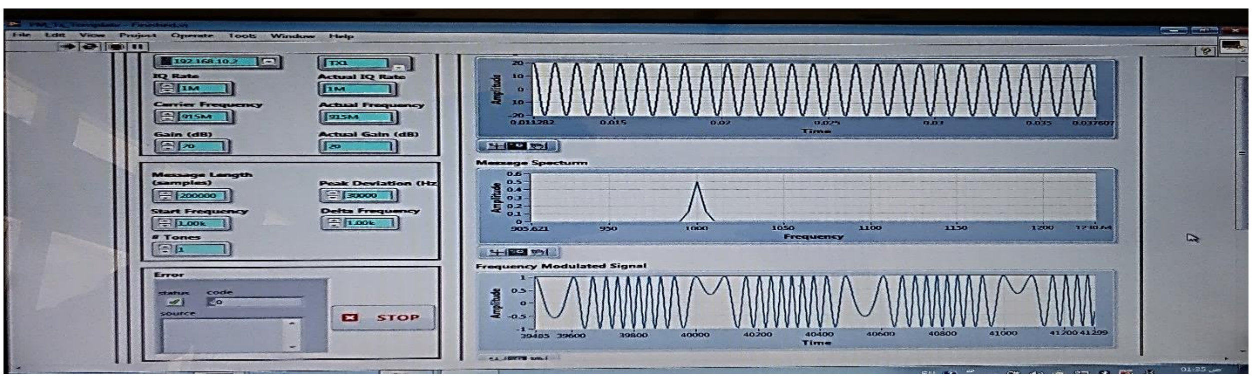
10- Save your transmitter as a file whose name includes the letters “FM_Tx” and your initials.

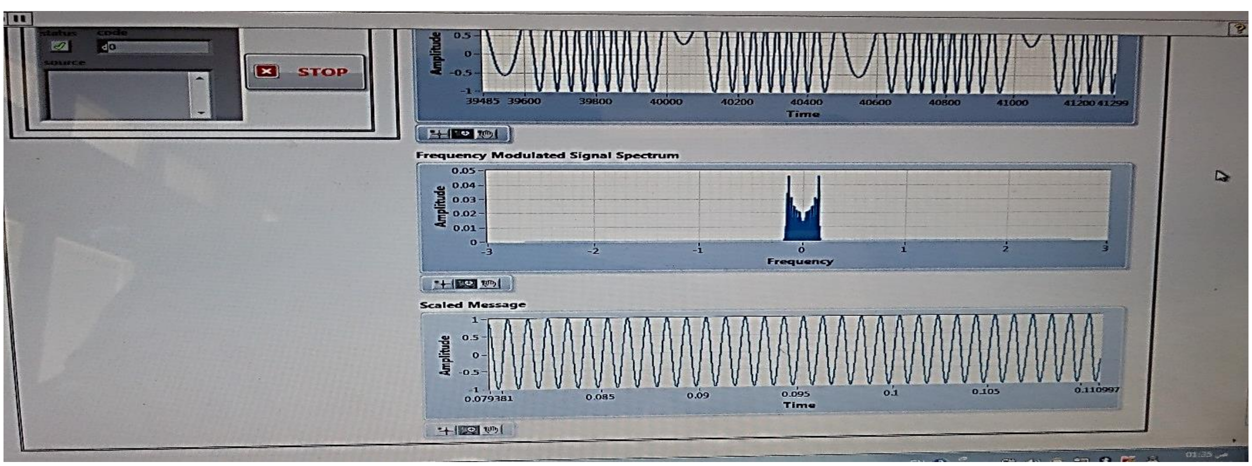
You need to set your message and transmitter parameters as following

Peak frequency deviation	30 KHZ
Start frequency	1 KHZ
Delta Frequency	1 KHZ
Number of tones	1

Carrier frequency	915 MHZ
IQ rate	1 MHZ
gain	20 db
Active antenna	TX1
Message length	200,000 sample

Your results have to be like that

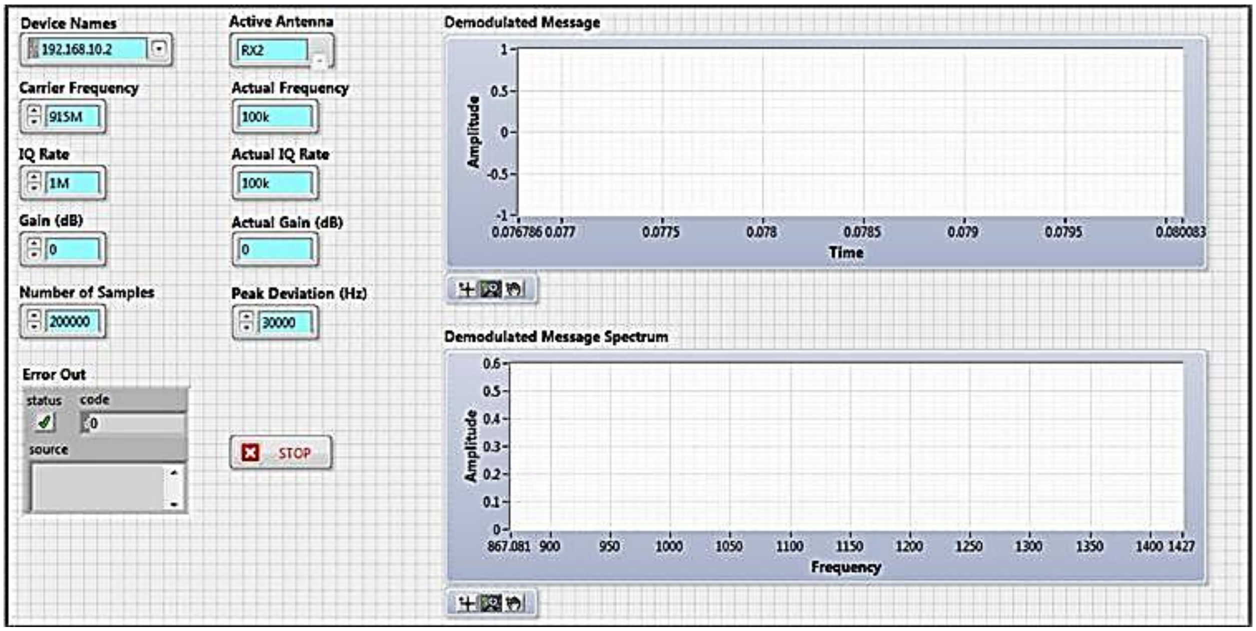




Receiver

Your task is to add blocks as needed to the given receiver VI template “FM_Rx_Template.vi” to demodulate the complex array $\breve{g}(t)$ retrieved by the “Fetch Rx Data” VI to get the quick scaled message signal $m_n(t)$.

the following figure shows the front panel of FM_Rx_Template.vi.

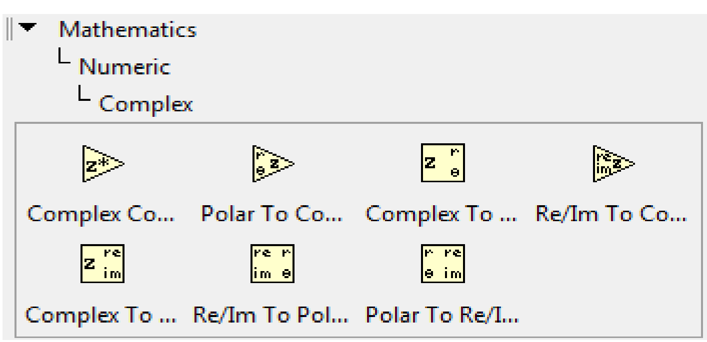


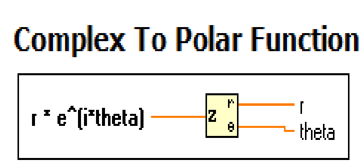
Front panel of FM_Rx_Template.vi

Step-by-step LabVIEW setup to demodulate the FM signal using USRP:

1. After fetching from the “niUSRP Fetch Rx Data VI”, get data values and sampling time interval of this received waveform using the “Get Waveform Components” VI.

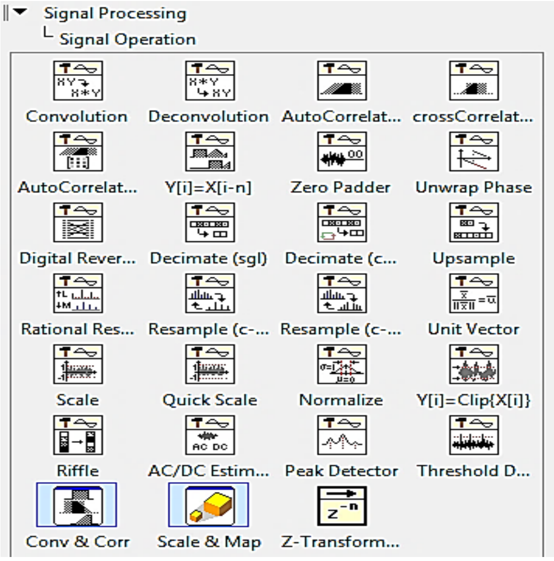
2. Extract the angle of using the “Complex to Polar” VI.

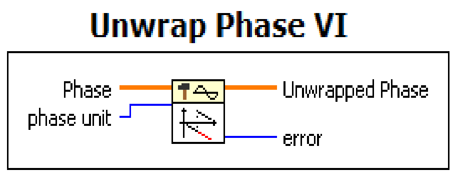




Complex to Polar V

3- Unwrap the angle with the “Unwrap Phase” VI



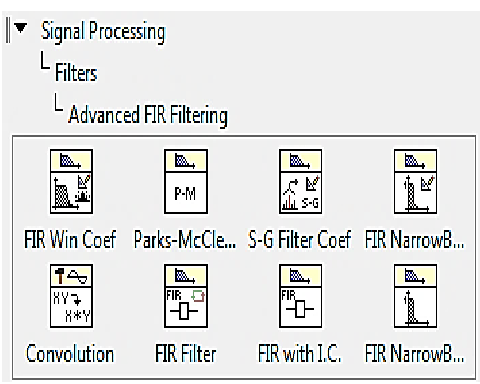


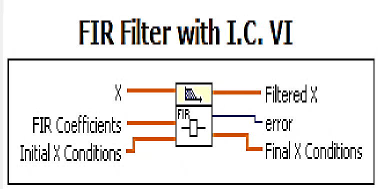
Unwrap Phase VI

4- The unwrapped sequence of angles is then passed into a differentiator. We recognize that in discrete time

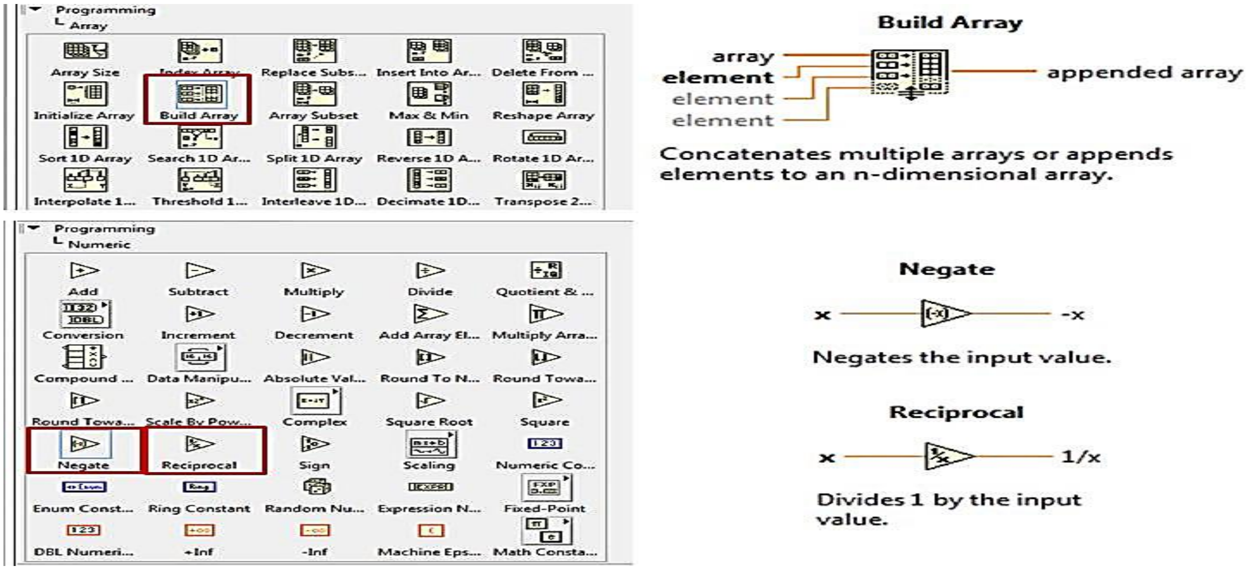
$$\frac{d\theta}{dt} = \frac{\theta[n]-\theta[n-1]}{T} \qquad 6\text{-}14$$

where T is the sampling time interval. The differentiator can be implemented using one of the “FIR Filter” Vis .For the “FIR coefficients” array use$(\frac{1}{T}, \frac{-1}{T})$ Generate this array by using the “Build Array” VI, reciprocal block and negate block.



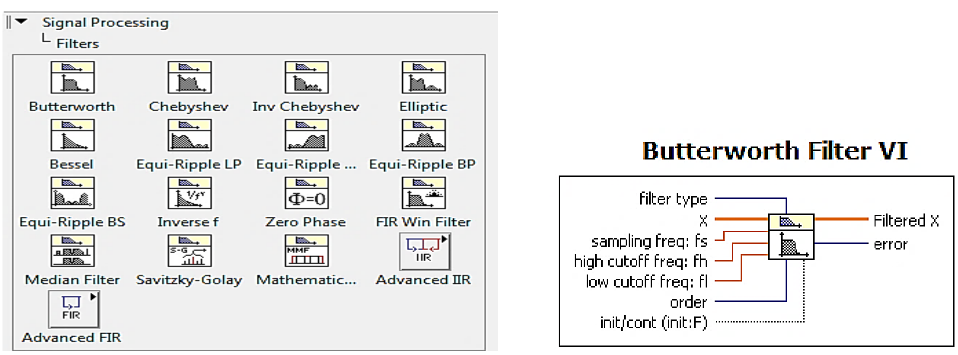


FIR Filter with Initial Conditions VI



Top: Build Array VI; Bottom: Negate and Reciprocal functions

5- The output from the differentiator is then passed into an envelope detector; a low-pass filter would be best since differentiation tends to enhance high-frequency noise. In this example, we use the Butterworth Filter VI , but any low-pass filter will produce usable results. Set the filter type as low-pass, filter order as 5, low-cut off frequency as 5000 Hz, and the sampling frequency as reciprocal of sampling time.



Butterworth Filter VI

6. Connect the output of the LPF to the “Build Waveform” VI to get the analog waveform. Set the sampling time interval to be the same as that of the FM signal fetched by the buffer.

7. Downscale the amplitude of the waveform by a factor of $2\pi\Delta f_{max}$. to get the message signal.

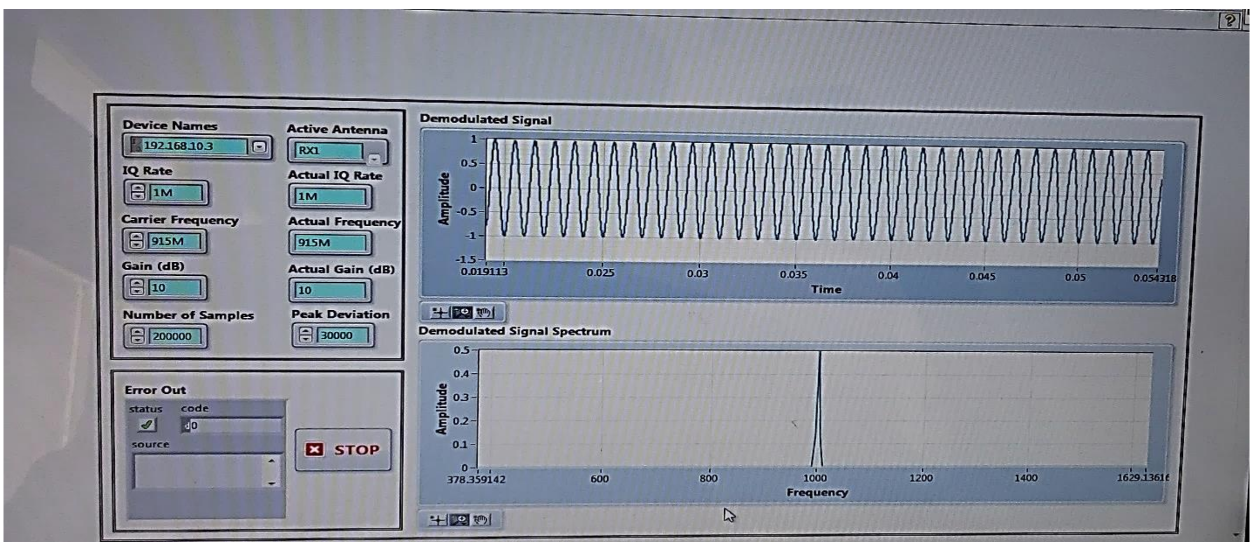
8. Plot the downscaled waveform and compare it with the original message signal you

modulated and transmitted. Plot the spectrum of the demodulated signal using the "FFT Power Spectrum and PSD" VI , and then connecting the output thereof to a waveform graph.

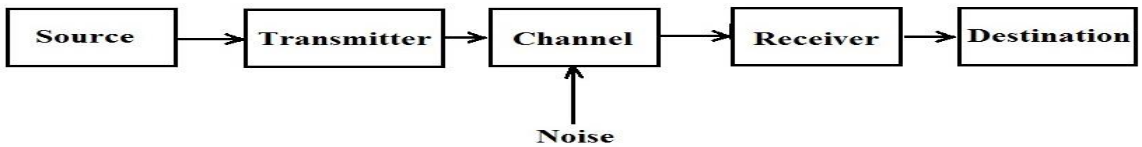
9. Save your receiver as a file whose name includes the letters "FM_Rx" and your initials. You have to set the receiver parameters as following

Carrier frequency	915 MHZ
IQ rate	1 MHZ
Gain	10 db
Active antenna	RX1
Message length	200,000 sample
Peak frequency deviation	30 KHZ

Your results have to be like that



MATLAB EXPERIMENT ONE: DOUBLE SIDEBAND MODULATION



1.1 INTRODUCTION

Double Sideband modulation is the easiest and most direct type of analog modulation. In this scheme, the modulated signal is obtained using a direct multiplication of the modulating signal (i.e. the message) by a cosine carrier. This multiplication results in shifting the entire spectrum of the message to a center frequency defined by the carrier frequency. The modulation is said to be double sideband transmitted carrier (DSB-TC) when the carrier is transmitted along the modulation term. If the carrier term is omitted, the modulation is termed double sideband suppressed carrier (DSB-SC). DSB-TC has a significant advantage in the receiver design (i.e. the envelop detector). Also transmitting the carrier independently enables us to extract useful information such as the carrier frequency which can be helpful for carrier synchronization. However, the DSB-TC loses to the other variant (i.e. the SC) in terms of power efficiency.

1.2 AIM

In this experiment, you're required to achieve the following:

1. Get familiar with the concept of DSB modulation, and its parameters.
2. Study the performance of the DSB modulation.
3. Examine different detectors (coherent detector, envelope detector).
4. Study the performance of coherent detection in the presence of frequency or phase mismatch.

1.3 PROCEDURE

1. Use Matlab to read the attached audio file, which has a sampling frequency Fs= 48 KHz. Find the spectrum of this signal (the signal in frequency domain). [*audioread, fft , fftshift , plot*]

2. Using an ideal Filter, remove all frequencies greater than 4 KHz.

3. Obtain the filtered signal in time domain and frequency domain, this is a band limited signal of BW=4 KHz. [*ifftshift ,ifft*]

4. sound the filtered audio signal (make sure that there is only a small error in the filtered signal) [*sound*]

5. Modulate the carrier with the filtered signal you obtained, you are required to generate both types of modulation (DSB-TC and DSB-SC). Choose a carrier frequency of 100 KHz. For the DSB-TC take the DC bias added to message before modulation to be twice the maximum of the message (modulation index =0.5 in this case).

   Note: You will also need to increase the sampling frequency of the filtered audio signal, the sampling frequency must be at least 2 times the carrier frequency, In this

simulation use $\quad Fs = 5\ F_c$.
[*resample]*

You have to sketch the modulated signal of both DSB-TC & DSB-SC in frequency domain.

6. For both types of modulations (DSB-SC & DSB-TC), use envelop detector to receive the message (assume no noise). Note: to obtain the envelope you can use the following matlab command.

$$envelope = abs(hilbert(modulated\ signal))$$

7. After the reception of both modulation types using envelope detector, sketch the received signal in time domain, and Play the received signal back (Note: to sound signal after demodulation process you have to decrease the sampling frequency again). What observation can you make of this or which type of modulation the envelope detector can be used with?

For DSB-SC, perform steps 9-11.

8. Use coherent detection to receive the modulated signal with SNR=0, 10, 30 dB then sound the received signals and plot them in both time and frequency domain.
9. Repeat the coherent detection with frequency error, F=100.1 KHz instead of 100 KHz and Find the error. Do you have a name for this phenomenon?
10. Repeat the coherent detection with phase error = 20^0.

1.4 USEFUL MATLAB FUNCTIONS

audioread,fft, fftshift, ifft, ifftshift , plot, awgn, resample, sound, hilbert, abs, max.

1.5 HINT

You are not allowed to use built in functions like **ammod, amdemod**

MATLAB EXPERIMENT TWO: SINGLE SIDEBAND MODULATION

1.6 INTRODUCTION

The bandwidth inefficiency stemming from the DSB transmission was the main reason why the single sideband (SSB) was developed. In SSB modulation, the bandwidth required for band pass transmission is equal to the bandwidth of that of the baseband. In other words, the band requirement for SSB is halved with respect to that of DSB which requires twice the baseband bandwidth. This reduction in transmission bandwidth is possible since the sideband are replicated twice over the positive and negative frequencies. However, the bandwidth reduction doesn't come entirely free. In fact, SSB suffers from several disadvantages that we hope to cover in the following simulation

1.7 AIM

In this experiment, you're required to achieve the following:

1. Familiarize students with the SSB modulation scheme.
2. Study the performance of SSB with different detectors.
3. Investigate the disadvantages of the SSB

1.8 PROCEDURE

1. Use Matlab to read the attached audio file which has a sampling frequency Fs= 48 KHz. Find the spectrum of this signal. *(same as previous experiment)*
2. Using an ideal Filter, remove all frequencies greater than 4 KHz. *(same as previous experiment)*
3. Obtain the filtered signal in time domain, this is a band limited signal of BW=4 KHz. You could play the sound back, to make sure only small distortion was introduced. *(same as previous experiment*
4. Generate a DSB-SC modulated signal and plot its spectrum. Choose the carrier frequency to be 100kHz. Remember to set the sampling frequency to Five times the carrier frequency Fs = $5F_C$.
   (same as previous experiment)
5. Obtain the SSB by filtering out the USB (we need to get LSB only) of the DSB-SC modulated signal using an ideal filter then Plot the spectrum again.
6. Use coherent detection with no noise interference to get the received signal (to demodulate the SSB-SC) and play the file back also sketch the received waveform and spectrum.
7. Repeat steps 5 and 6, only this time ▪ Use a practical 4th order Butterworth filter. *[butter, filter]*
8. For the ideal filter case, get the received signal again but when noise is added to SSB-SC with SNR = 0, 10, and 30 also play the received sound and sketch the received waveform/spectrum in each case.
   [awgn]

9. For the ideal filter case, generate a SSB-TC. As with experiment one, set the DC bias to twice the maximum of the message. Use envelope detector to demodulate the message (without noise) . Play back the received message and sketch its waveform.

1.9 USEFUL MATLAB FUNCTIONS

audioread ,fft, fftshift, ifft, ifftshift , plot, awgn, resample, sound. , hilbert, abs, max, butter,filter.

MATLAB EXPERIMENT THREE: FREQUENCY MODULATION

3.1 INTRODUCTION

Frequency modulation (FM) is a modulation type in which the instantaneous frequency of the carrier is changed according to the message amplitude. The motive behind the frequency modulation was to develop a scheme with inherent ability to combat noise. The noise, being usually modeled as additive, has a negative effect on the amplitude by introducing unavoidable random variations which are superimposed on the desired signal. Unlike the amplitude, frequency has a latent immunity against noise. Since it resides “away” from the amplitude, any changes in the amplitude would be completely irrelevant to the frequency. In other words, there is no direct correlation between the variation in amplitude and frequency, thus making FM a better candidate over AM with respect to noise immunity. However, what FM gains in noise immunity lacks in bandwidth efficiency. Since FM usually occupies larger bandwidth, AM is considered more bandwidth wise.

3.2 AIM

In this experiment, we investigate the narrowband frequency modulation when the SNR is varied.

Students are expected to:

1. Develop an appreciation of FM ability to counteract noise.
2. Be able to simulate the generation and the demodulation of NBFM using matlab.
3. To be able to tell the similarities and differences between AM and NBFM.

3.3 PROCEDURE

1. Repeat steps 1 through four in experiment 1,2.
2. Generate the NBFM signal. Use a carrier frequency of 100kHz and a sampling frequency of Fs =5Fc. Plot the resulting spectrum. What can you make out of the resulting plot?
3. what is the condition we needed to achieve NBFM.
4. Demodulate the NBFM signal using a differentiator and an ED. For the differentiator, you can use the following command: diff. Assume no noise is introduced.

3.4 USEFUL COMMANDS

audioread, fft, fftshift, ifft,ifftshift, awgn, upsample,resample, sound,hilbert, abs,mean,cumsum,diff.