Using GNU Radio Companion: Tutorial 4. Using Complex Signals and Receiving SSB

This tutorial is a guide to receiving SSB signals. It will also illustrate some of the properties of complex (analytic) signals and show why we use them in communications systems.

Complex/Analytic Signals

1. Open a new flow graph in GRC.

2. Create the simple flow graph shown in the figure below. Set the Type in each of the three blocks to Float as you have in the past. Other than that you can leave all of the values at their default settings.
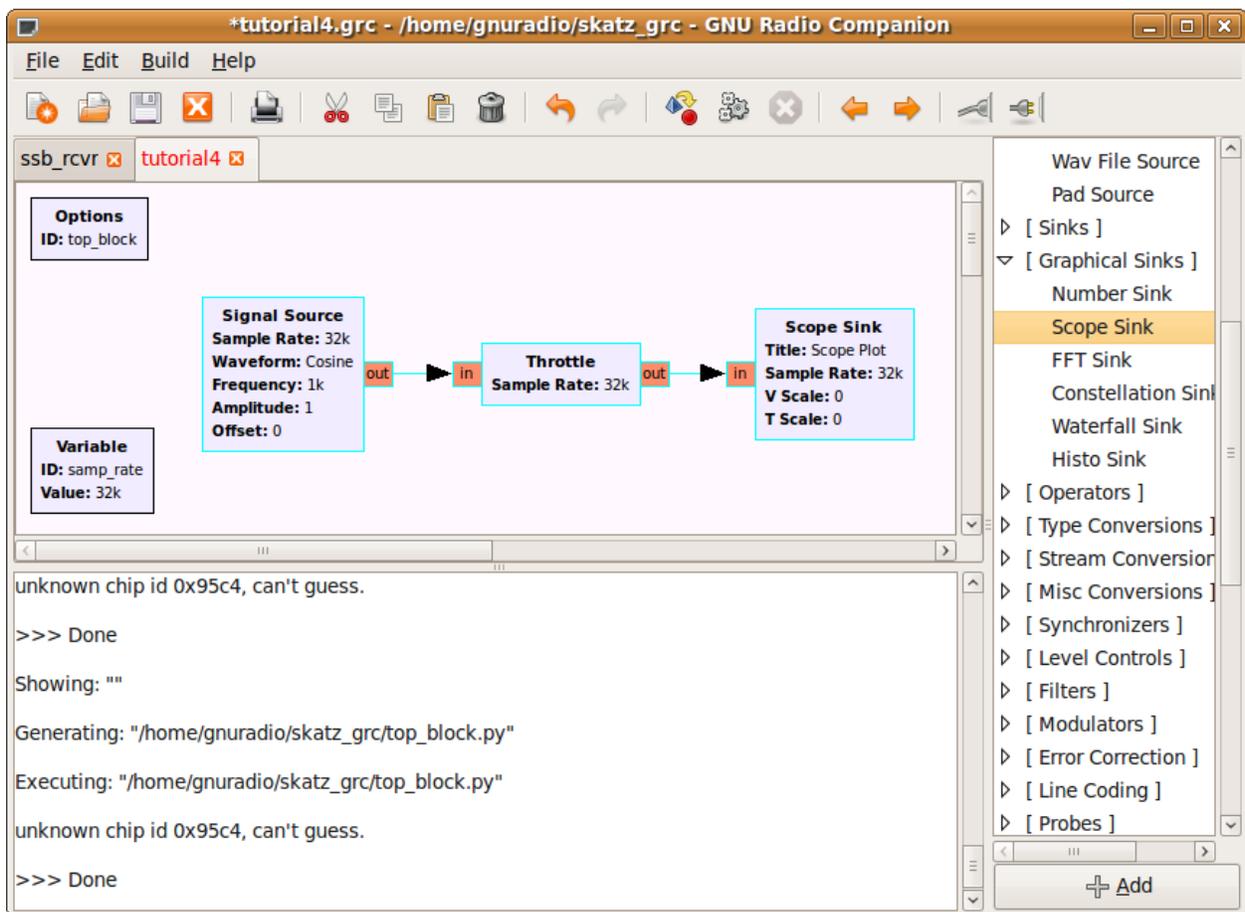
3. Execute the flow graph. The scope sink should open and display a sinusoidal signal. Convince yourself that this signal has the amplitude and frequency that you expect.

4. Modify the flow graph by changing the Type in each of the three blocks to Complex. Execute the flow graph. Your Scope Plot should now display two sinusoids that are 90° out

of phase with each other. The leading (channel 1) wave is the I or in-phase component of the complex signal and the lagging (channel 2) wave is the Q or quadrature component. When a signal source is set to Complex, it will output both the I and Q components.

5. Modify your flow graph as shown in the figure below. The Signal source should be set to output a Square wave with a Type of Float. Thus, the first Scope Sink and the Throttle must also be set to accept Float values.
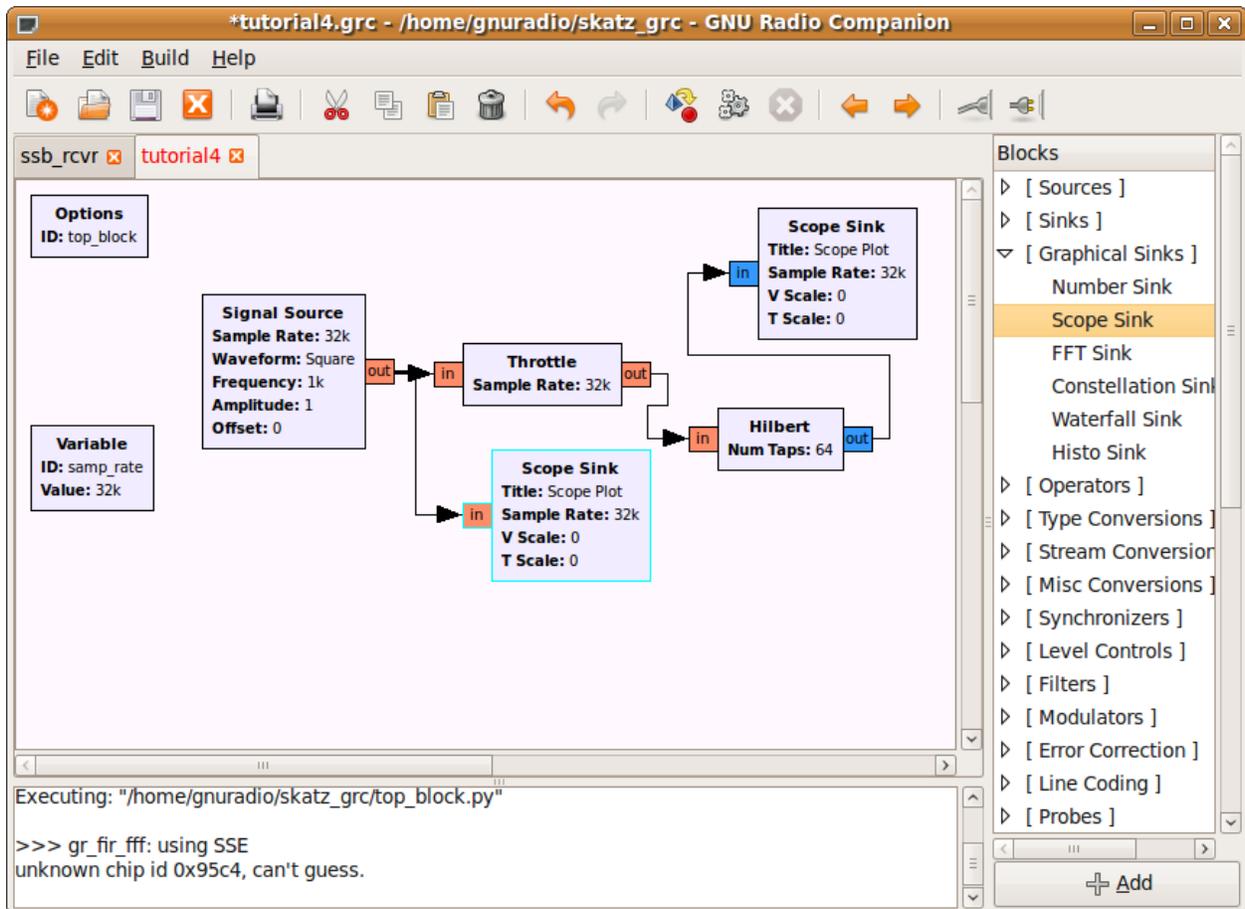
6. The Hilbert block is found in the Filters menu. This block outputs both the real input signal and the Hilbert transform of the input signal as a complex signal. Leave the number of taps at its default setting of 64. Since the output of this block is complex, the second Scope Sink must be set to accept Complex inputs.

7. Execute the flow graph. Two Scope Plots should open. One should contain the square wave output from the Signal Source only. The other should include both the original square wave and its Hilbert Transform. Do you understand why the Hilbert transform of a square wave looks this way?

8. As shown in step 4 above, the Signal Source can be set to output a Complex signal and display both the I and Q components. Modify the flow graph as shown below.
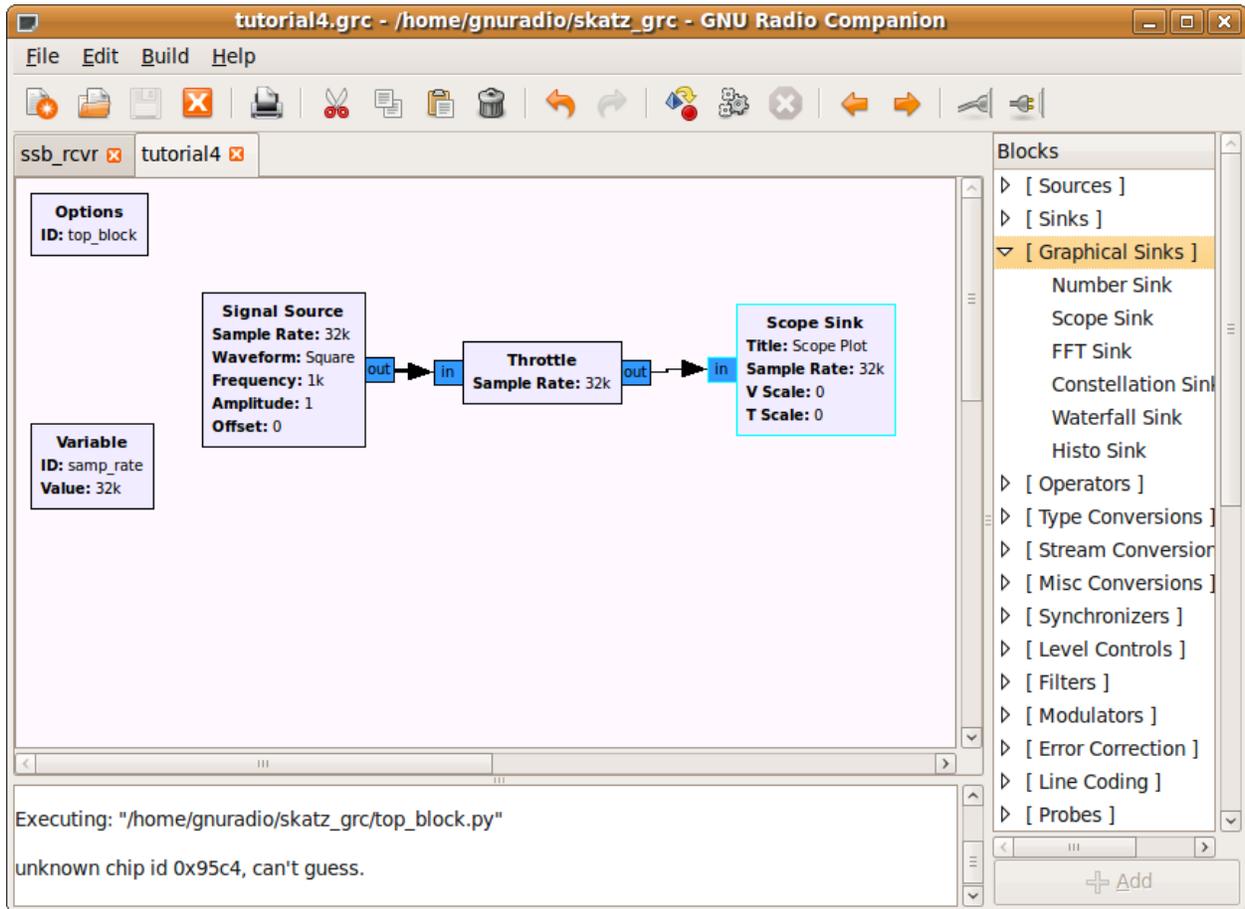
9. Set the Signal Source to output a Complex waveform. Make sure the Throttle and Scope Sink are also set to Complex. Execute the flow graph. Is the Complex waveform displayed here the same as the one obtained from the Hilbert transform? Your answer should be NO (assuming that you are using a version of GRC 3.2.2 or earlier). This is incorrect. GRC is NOT displaying the correct Q component of a complex square wave. The Hilbert transform did output the proper waveform.

10. Create the flow graph shown below. Make sure that all of the blocks are set to Type: Float. This flow graph takes two sinusoids, at frequencies of 1KHz and 10 KHz and multiplies them together. Using a trigonometric identity we know that the product of two cosines gives two cosines at the sum and difference frequencies of the original signals. In this case we expect outputs at 9 KHz (difference) and 11 KHz (sum).
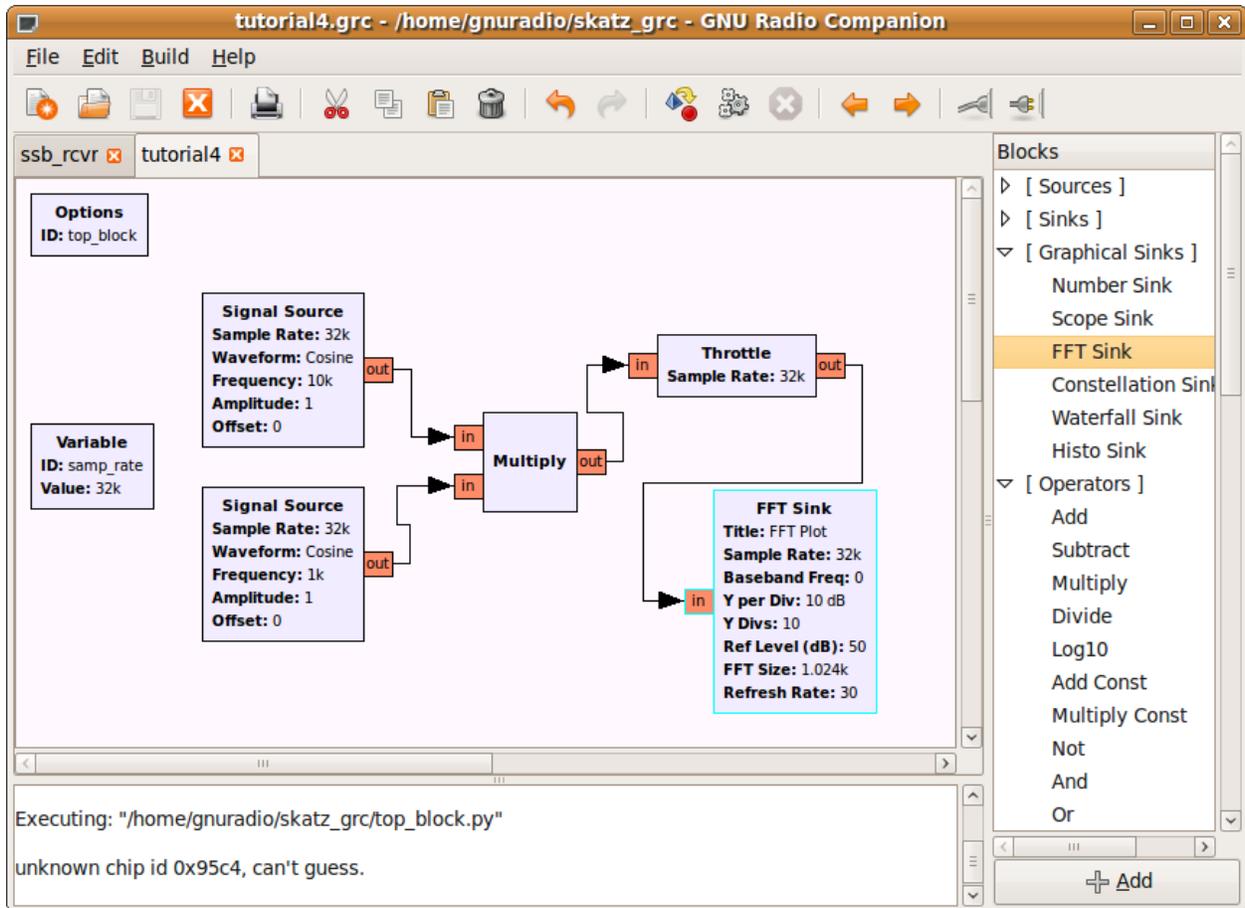
11. Execute the flow graph and confirm this result. Note that the FFT plot only shows the positive frequency spectrum when it is set to Type: Float. We know that for real inputs the negative frequency components are the same as positive frequency components.

12. Change ALL of the blocks to Type: Complex and execute the flow graph again. You should now observe a single output at 11 KHz. This is the original 10KHz signal shifted by the 1KHz signal. If we want to shift in the negative direction a frequency of -1000 can be used. Try this. From this example we see two of the primary advantages of using analytic signals. A signal can be shifted (sum) without creating an additional difference signal. Also, note that there are NO negative frequency components. Why is this?

Single Sideband (SSB) Demodulation

1. Download the zipped file: ssb_lsb_256k_complex2.dat and unzip it on your hard drive.

2. Create a new flow graph as shown in the figure below. The File Source should be set to the data file that you just downloaded. The Variable Block which sets the sampling rate

(samp_rate) should be set to 256000 as this is the data rate that the received signal was sampled at. The throttle and FFT Sink can be left at their default settings.
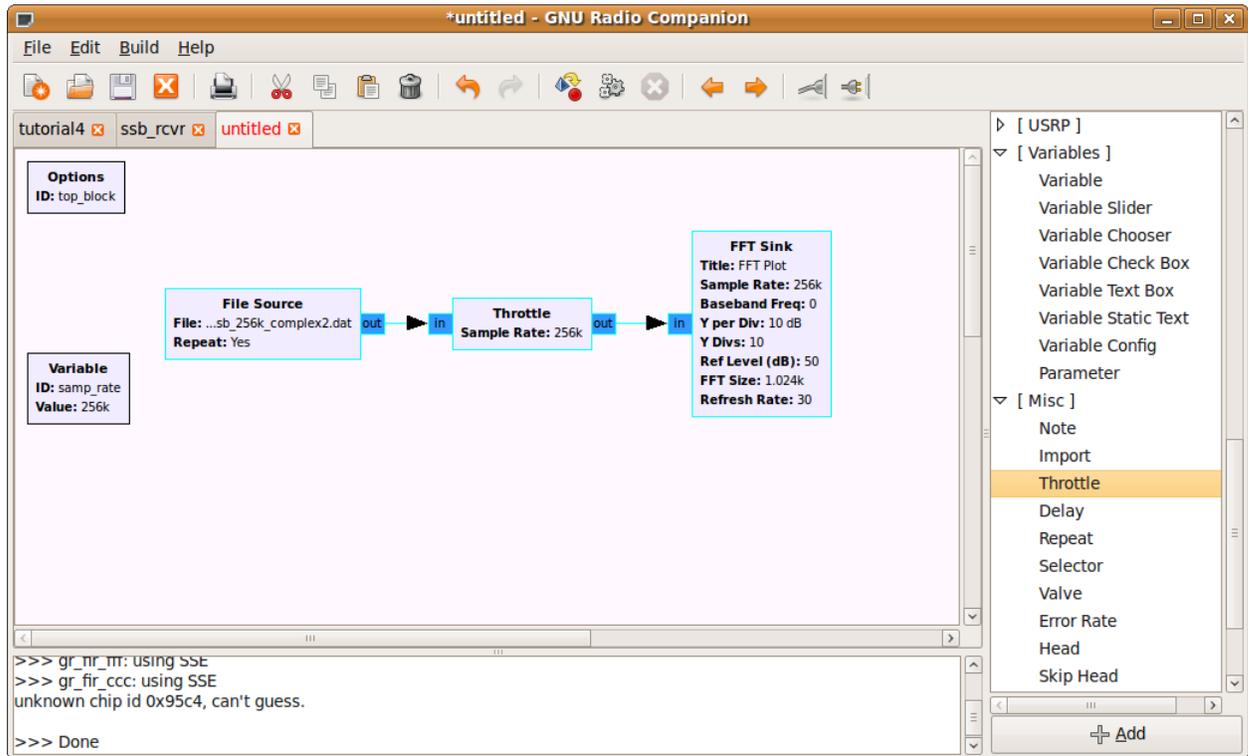
3. Execute the flow graph. After the FFT Plot window opens, adjust the Ref Level so that the Amplitude values start at 10 dB and set the Axis Options to 10dB/div. You should view a section of the spectrum that is 256 KHz wide (due to the sample rate). Note that there is one signal visible between 40 and 60 KHz.

4. When this signal was recorded, the USRP was set to a frequency of 50.3MHz. Thus, the 0 KHz point on the display corresponds to 50.3 MHz. While the FFT Plot is displayed move the cursor over the signal and note the frequency along its right edge. It should be approximately 53 KHz. Since this is a lower sideband (LSB) signal, this corresponds to the carrier frequency. Because the "zero" frequency corresponds to 50.3 MHz, the original carrier frequency of signal was 50.3MHz + 53 KHz = 50.353 MHz. However, now that the spectrum has been shifted down by 50.3 MHz, we use the carrier frequency of 53 KHz.

5. The first step in building a receiver is to use a channel filter to pass the signal of interest and filter out the rest of the signals in the band. The first step in this process is to shift the signal of interest down to zero frequency as shown below.
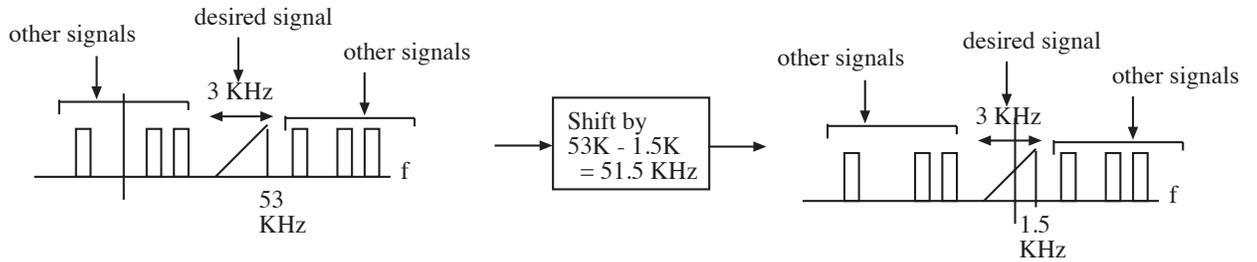
6. The second step is to apply a low pass filter so that the other signals will be filtered out.
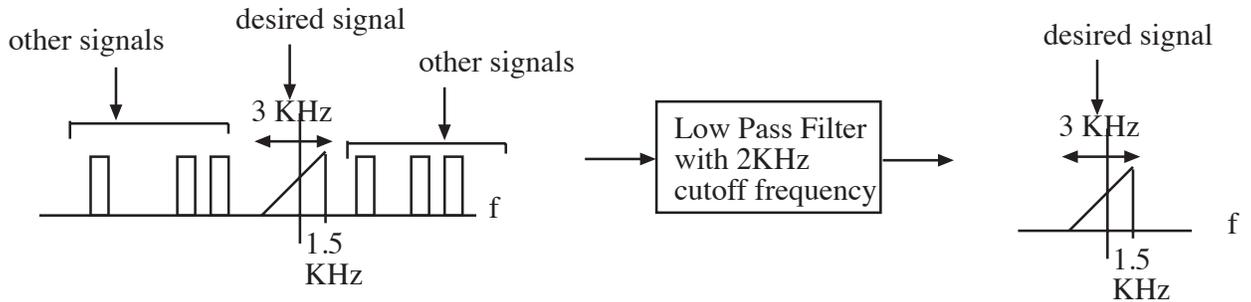
7. In GRC, the Frequency Xlating (translating) FIR Filter performs both of these operations. Insert this block between the File Source and the Throttle. Complete the properties window as shown below. The Center Frequency of -51500 will shift the entire spectrum down by 51500 Hz. The function indicated in the Taps parameter generates the taps for a low pass filter with a gain of 1 (in the pass band), a sampling rate equal to samp_rate (256K), a cutoff frequency of 2KHz and a transition width of 100.
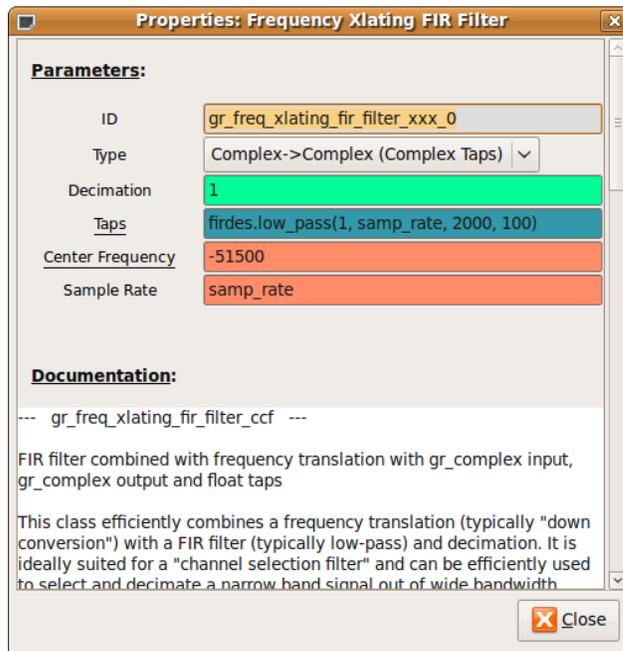
8. Execute the flow graph. You will see that your signal has now moved down to the origin and is the only signal present. Now that we have located the signal of interest, there is no reason that we need to be concerned with so much of the adjacent spectrum. We can reduce the range of frequencies that are being processed by reducing the sample rate (decimation). Re-open the Frequency Xlating FIR Filter block and change the Decimation parameter to 8. This will reduce the sample rate to 256000/8 = 32000. Change the sample rate of the Throttle and FFT sink to this new rate. What frequency range to you expect the FFT to display now?

9. Execute the flow graph again to see if you are correct. You should now observe an expanded version of your signal. Select Autoscale on the FFT Plot so that the peaks of the signal are observed. Notice that after a while, the signal level will be reduced for a few seconds. That occurs when the station stops transmitting.

10. Recall that the signal is a complex (analytic) signal. In order to demodulate SSB we need to operate on the real and imaginary parts of the signal separately. The Complex to Float block will take a complex signal and output its real (top) and imaginary (bottom) parts separately. Modify the flow graph to appear as shown in the figure below. The outputs of the complex to Float block are both real so the throttles and FFT Sinks need to be changed to Type: Float.
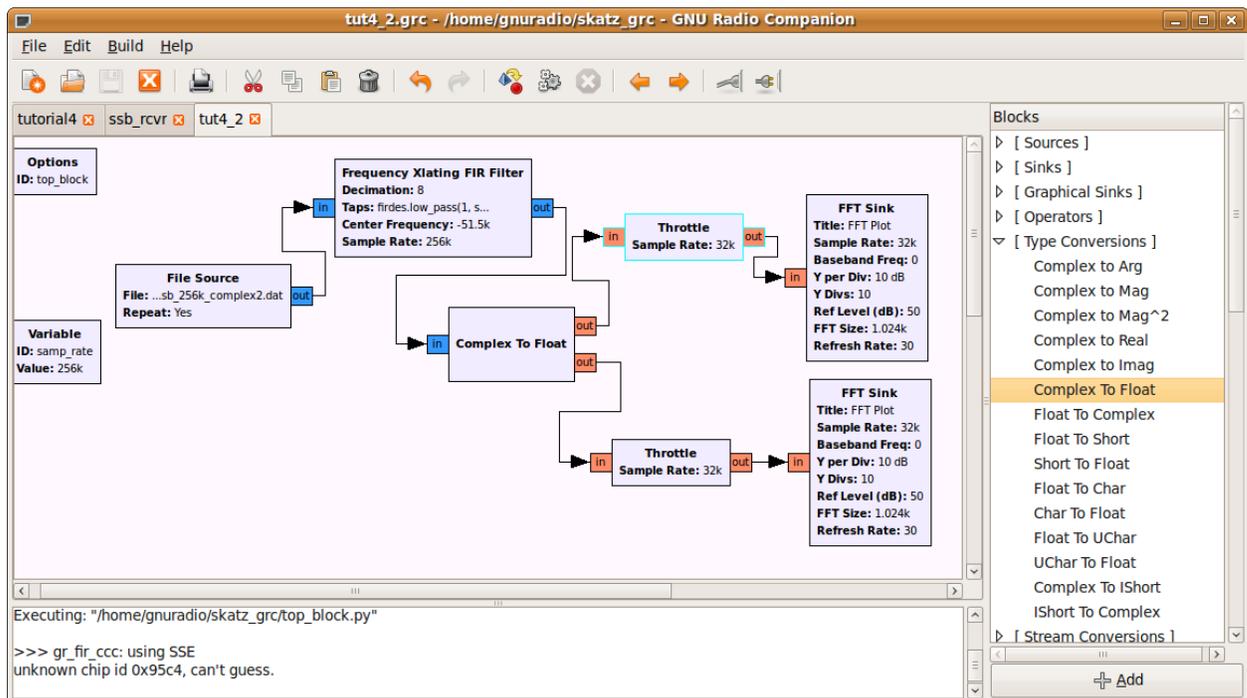
11. Execute the flow graph. You should now observe the spectra of the real and imaginary parts of the signal. Note that the signals extend out to 2KHz, the cutoff frequency of the filter.

12. One method of demodulating this SSB signal, known as Weaver's method, takes the real and imaginary part of the signal and processes them as shown below. Use the Signal Source in GRC to generate the Cosine and Sine needed to implement this demodulator. The multiply and add blocks are found in the operators menu.
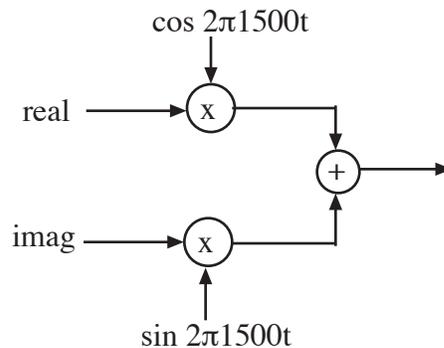
13. Observe the output of the adder on the FFT sink. This is the baseband signal that has been extracted from the modulated SSB signal.

14. The final step is to listen to the demodulated signal. Add an Audio Sink as demonstrated in tutorial 3. Recall that you will need a Rational Resampler to adjust the sampling rate to one that works with the Audio Sink. Also, you will need a volume control to reduce the amplitude of the signal before it enters the Audio Sink. Hint: try values between 0 and .0001 for your constant multiplier.