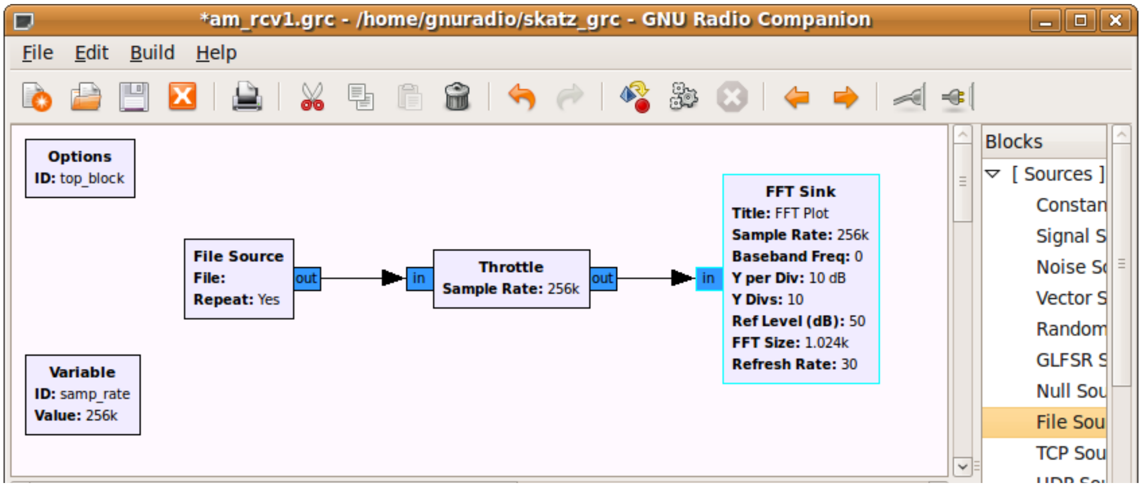
Using GNU Radio Companion: Tutorial 3. Receiving AM Signals

This tutorial is a guide to receiving AM signals. It uses a data file that contains several seconds of recorded signals from the AM broadcast band. This data file was obtained from the USRP. If you have a USRP available you could also use that as your input and receive live signals. The data file can be downloaded from the SDR Web page.

1. Before beginning this tutorial, download the file: am_usrp710.dat from the SDR Web page. Save it in a location that you can access later. If you download it as a zip file, unzip it.
2. Open a terminal window using: Applications > Accessories > Terminal. At the prompt type: `grc`
3. Construct the flow graph shown below consisting of a File Sink, Throttle, and FFT Sink. Set the Sample Rate in the variable block to 256000. This is the rate at which the saved data was sampled.



Options
ID: top_block

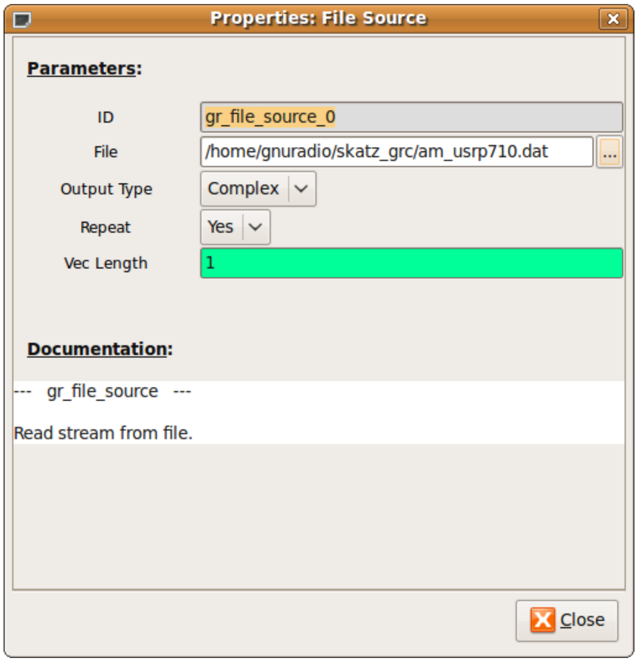
File Source
File:
Repeat: Yes

Throttle
Sample Rate: 256k

FFT Sink
Title: FFT Plot
Sample Rate: 256k
Baseband Freq: 0
Y per Div: 10 dB
Y Divs: 10
Ref Level (dB): 50
FFT Size: 1.024k
Refresh Rate: 30

Variable
ID: samp_rate
Value: 256k

4. Double click on the File Source block. Click on the ellipsis (…) next to the File parameter. Locate the am_usrp710.dat file that you saved in step 1 (unzipped). The path to your file will appear as shown in the figure below. Set the Output Type to Complex. The use of complex data to describe and process waveforms in SDR will be discussed next week. Set Repeat to Yes. This will cause the data to repeat so that you have a continuously playing signal.



Properties: File Source

Parameters:

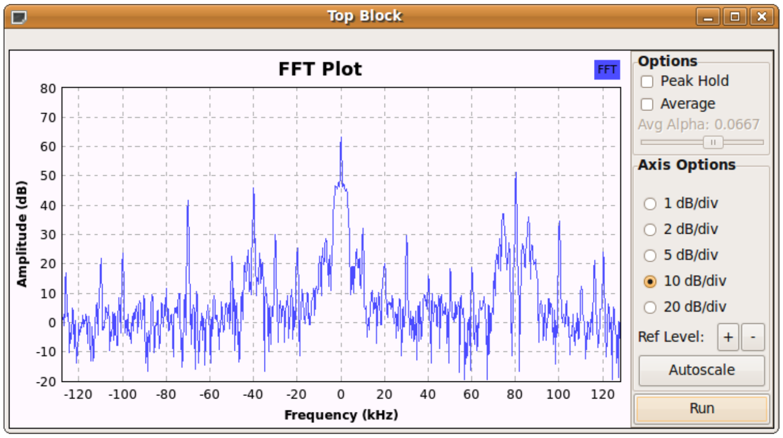
ID	gr_file_source_0
File	/home/gnuradio/skatz_grc/am_usrp710.dat
Output Type	Complex
Repeat	Yes
Vec Length	1

Documentation:

--- gr_file_source ---

Read stream from file.

5. Save and execute the flow graph. You should observe an FFT display similar to the one shown below. You may need to click on Autoscale button to scale the data as shown. Note the following:
    - This data was recorded with a USRP set to 710KHz. Thus, the signal you see at the center (indicated as 0 KHz) is actually at 710 KHz. Similarly, the signal at 80 KHz is actually at 710KHz + 80KHz = 790KHz.
    - The display spans a frequency range from just below -120KHz to just above 120KHz. This exact span is 256KHz, which corresponds to the Sample Rate that the data was recorded at.
    - The peaks that you observe on this display correspond to the carriers for AM broadcast signals. You should also be able to observe the sidebands for the stronger waveforms.



Top Block

Options

☐ Peak Hold

☐ Average

Avg Alpha: 0.0667

Axis Options

○ 1 dB/div

○ 2 dB/div

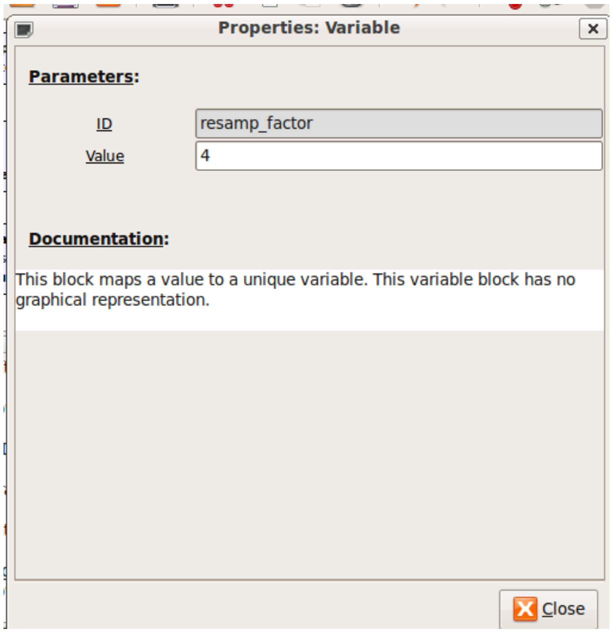
○ 5 dB/div

● 10 dB/div

○ 20 dB/div

6. In this step we will expand the frequency scale on the FFT display so that you can view the signals with greater resolution. Recall from step 5 above that the span of the frequency axis is determined by the sample rate (256K for this file). While we cannot change the original data, we can resample it to either increase or decrease the sample rate. We will decrease the sample rate by using decimation. Modify the flow graph as follows:

   - Add a Variable block (under Variables menu). Set the ID to resamp_factor and the Value to 4 as shown below.



Properties: Variable

Parameters:

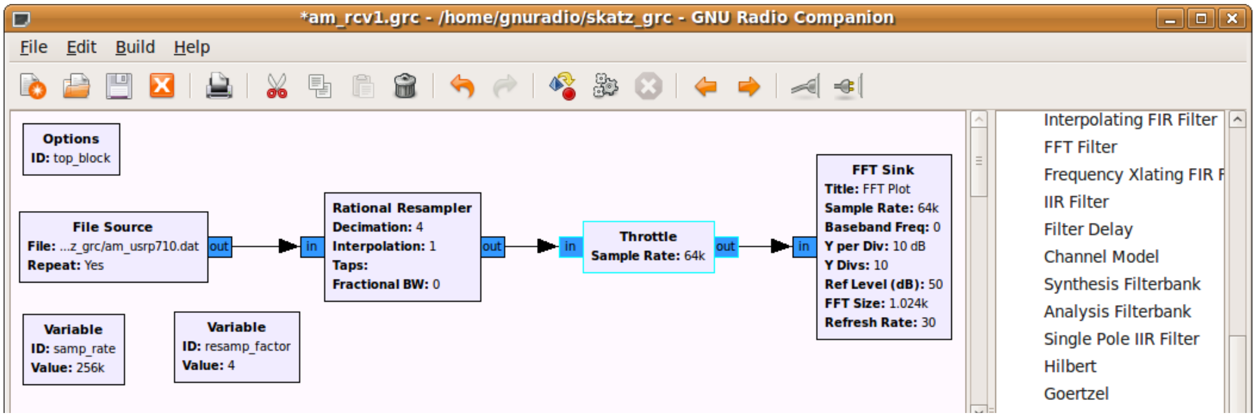
ID: resamp_factor

Value: 4

Documentation:

This block maps a value to a unique variable. This variable block has no graphical representation.

   - Add the Rational Resampler (NOT Rational Resampler Base) from the Filters menu as shown below. Set its decimation factor to resamp_factor. Thus, it will use the value of the variable set in the previous step (4) to decimate the incoming data. That means that it will divide the incoming data rate by the decimation factor. In this example, the incoming 256K samp/sec data will be converted down to 256K/4 = 64K samp/sec.
   - Note that the Throttle and FFT Sink now need their Sample Rates changed to correspond to this new rate. Change the Sample Rate in both of these blocks to: samp_rate/resamp_factor . Now we can change the decimation factor in the Variable block and it will be reflected in each of the other blocks automatically.
   - Your flow graph should now appear as shown below.



Options
ID: top_block

File Source
File: ...z_grc/am_usrp710.dat
Repeat: Yes

Rational Resampler
Decimation: 4
Interpolation: 1
Taps:
Fractional BW: 0

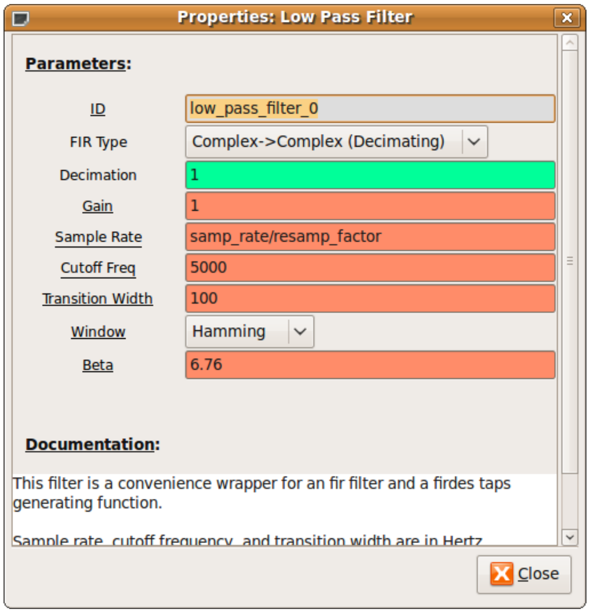
Throttle
Sample Rate: 64k

FFT Sink
Title: FFT Plot
Sample Rate: 64k
Baseband Freq: 0
Y per Div: 10 dB
Y Divs: 10
Ref Level (dB): 50
FFT Size: 1.024k
Refresh Rate: 30

Variable
ID: samp_rate
Value: 256k

Variable
ID: resamp_factor
Value: 4

7. Execute the new flow graph. You should now observe a frequency span of only 64KHz (-32KHz to +32KHz). What actual frequency range does this correspond to?
8. The bandwidth of an AM broadcast signal is 10KHz (+/- 5KHz from the carrier frequency). You may find it useful to click the "Stop" button on the FFT plot to see this more clearly. Also, note that many stations also include additional information outside of the 10KHz bandwidth. This will be discussed later.
9. In order to select the station at 710 KHz (or 0KHz) we need to insert a filter to eliminate all but the one station that we want to receive. This is often referred to as a channel filter. Since the station at 710 KHz has been moved to 0 KHz (in the USRP) we will use a low pass filter. The station bandwidth is 10 KHz, so we will use a low pass filter that cuts off at 5 KHz. Insert the Low Pass Filter (from the Filters menu) between the Rational Resampler and the Throttle. Set the parameters as shown below.



Parameters:

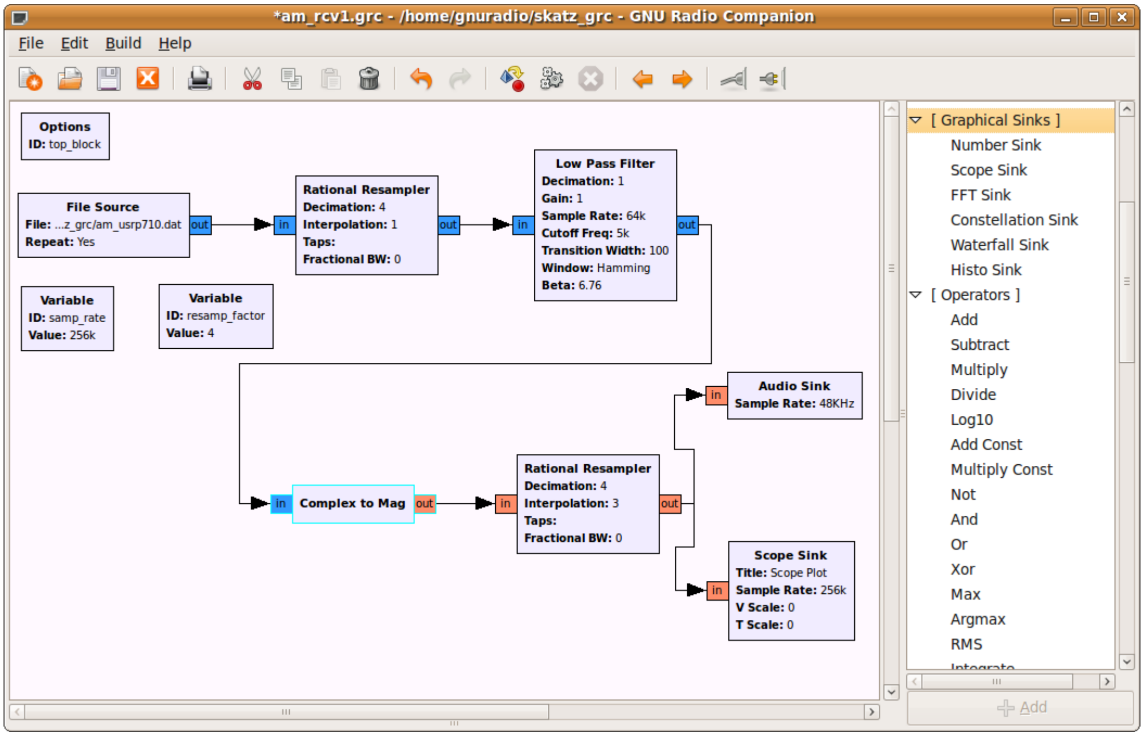
ID: low_pass_filter_0
FIR Type: Complex->Complex (Decimating)
Decimation: 1
Gain: 1
Sample Rate: samp_rate/resamp_factor
Cutoff Freq: 5000
Transition Width: 100
Window: Hamming
Beta: 6.76

Documentation:

This filter is a convenience wrapper for an fir filter and a firdes taps generating function.

Sample rate, cutoff frequency, and transition width are in Hertz.

10. Execute the flow graph. You should see that only the station between +/- 5KHz remains.
11. The next step is to demodulate the signal. In the case of AM, the baseband signal is the envelope or the magnitude of the modulated waveform. GNU Radio contains a Complex to Mag block (in the Type Conversions menu) that can be used for this purpose. Again, the use of complex signal representation will be dealt with in depth in the future. Insert this block between the Low Pass Filter and the Throttle.
12. Note that the titles of some of the blocks are now red and the Execute Flow Graph icon is dimmed. This indicates an error. Prior to adding this block, all of the block inputs and outputs were complex values. However, the output of the Complex to Mag block is Float (a real number). Thus, any blocks following this block should be Type: Float. Modify the Throttle and FFT Sink accordingly.
13. Execute the flow graph. You should now observe the spectrum of the baseband signal in the FFT Sink. Note that since the input data type to the FFT Sink is Float, only the positive frequency spectrum is displayed.
14. The next step is to listen to this demodulated waveform to confirm that it is in fact receiving the baseband signal. Remove the Throttle and the FFT Blocks. Add an Audio Sink block to the output of the Complex to Mag block. Note that the default sample rate for the Audio Sink is 32KHz. Most current audio cards require a sample rate of at least 44.1KHz. Change the sample rate to 48 KHz.
15. Note that the sample rate out of the Complex to Mag block is 64K and the input to the Audio Sink is 48K. In order to convert 64K to 48K we need to divide (decimate) by 4 and multiply (interpolate) by 3. Insert a Rational Resampler between the Complex to Mag and Audio Sink blocks and set the decimation and interpolation as noted above. Also set its Type to Float→Float.
16. Place a Scope Sink at the output of the Rational Resampler (in addition to the Audio Sink). Change its Type to Float. The flow graph should be similar to the one shown below.



Options
ID: top_block

File Source
File: ...z_grc/am_usrp710.dat
Repeat: Yes

Rational Resampler
Decimation: 4
Interpolation: 1
Taps:
Fractional BW: 0

Low Pass Filter
Decimation: 1
Gain: 1
Sample Rate: 64k
Cutoff Freq: 5k
Transition Width: 100
Window: Hamming
Beta: 6.76

Variable
ID: samp_rate
Value: 256k

Variable
ID: resamp_factor
Value: 4

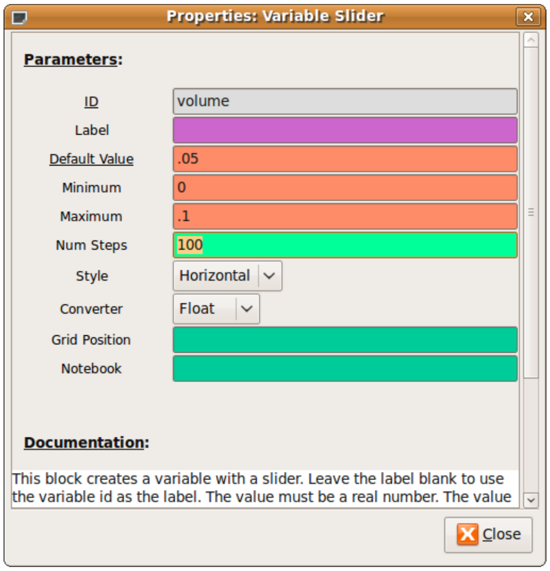
Audio Sink
Sample Rate: 48KHz

Complex to Mag

Rational Resampler
Decimation: 4
Interpolation: 3
Taps:
Fractional BW: 0

Scope Sink
Title: Scope Plot
Sample Rate: 256k
V Scale: 0
T Scale: 0

17. Execute the flow graph. The Scope Sink should open and display the output waveform. However, you may not yet hear the audio from your speaker. This is due to the fact that the values of the samples going in to the Audio Sink block are too large for the speaker. We need to apply an attenuator to reduce the size of the samples.
18. Insert a Multiply Const block from the Operators category between the Complex to Mag block and the Rational Resampler. Set the IO Type of the block to Float.
19. Add a Variable Slider block. Set the parameters as shown below.



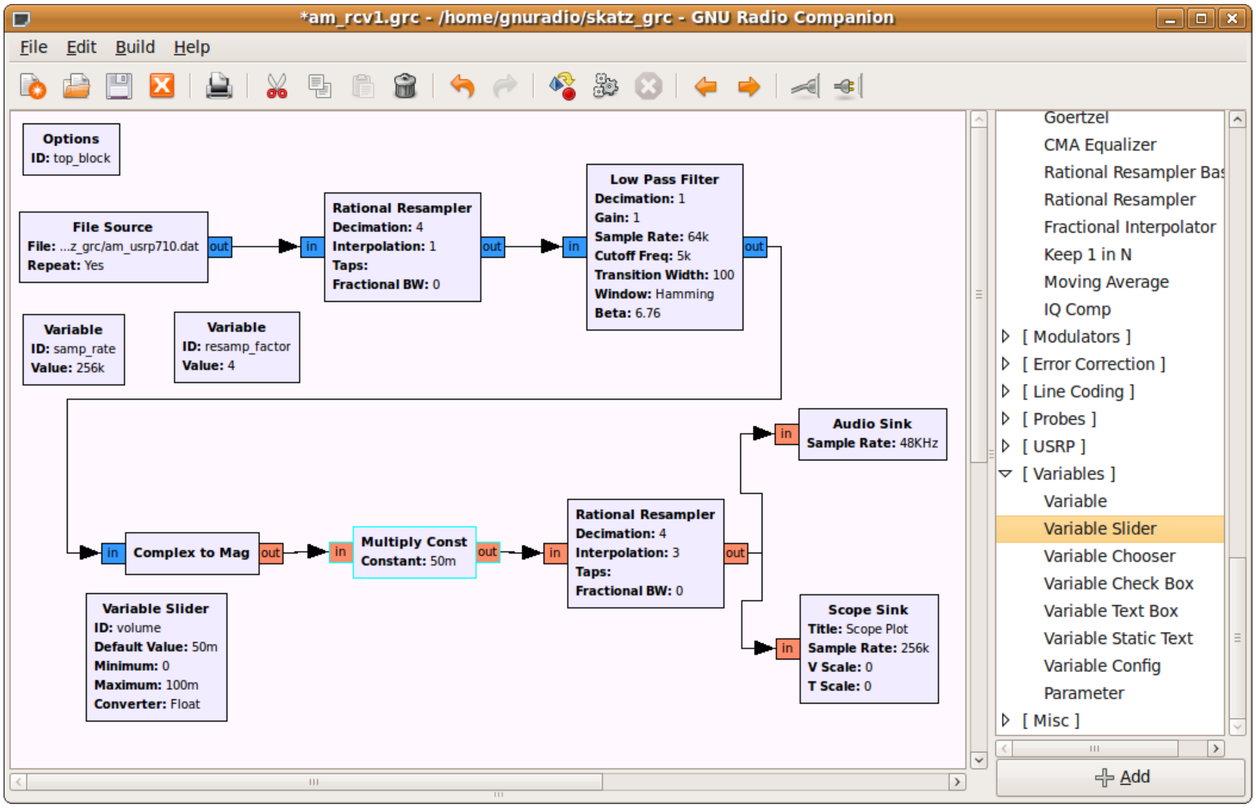
Parameters:

ID	volume
Label	
Default Value	.05
Minimum	0
Maximum	.1
Num Steps	100
Style	Horizontal
Converter	Float
Grid Position	
Notebook	

Documentation:

This block creates a variable with a slider. Leave the label blank to use the variable id as the label. The value must be a real number. The value

20. Set the Constant in the Multiply Const block to “volume” so that the slider controls it. The final flow graph is shown below.



Options
ID: top_block

File Source
File: ...z_grc/am_usrp710.dat
Repeat: Yes

Rational Resampler
Decimation: 4
Interpolation: 1
Taps:
Fractional BW: 0

Low Pass Filter
Decimation: 1
Gain: 1
Sample Rate: 64k
Cutoff Freq: 5k
Transition Width: 100
Window: Hamming
Beta: 6.76

Variable
ID: samp_rate
Value: 256k

Variable
ID: resamp_factor
Value: 4

Audio Sink
Sample Rate: 48KHz

Complex to Mag

Multiply Const
Constant: 50m

Rational Resampler
Decimation: 4
Interpolation: 3
Taps:
Fractional BW: 0

Variable Slider
ID: volume
Default Value: 50m
Minimum: 0
Maximum: 100m
Converter: Float

Scope Sink
Title: Scope Plot
Sample Rate: 256k
V Scale: 0
T Scale: 0

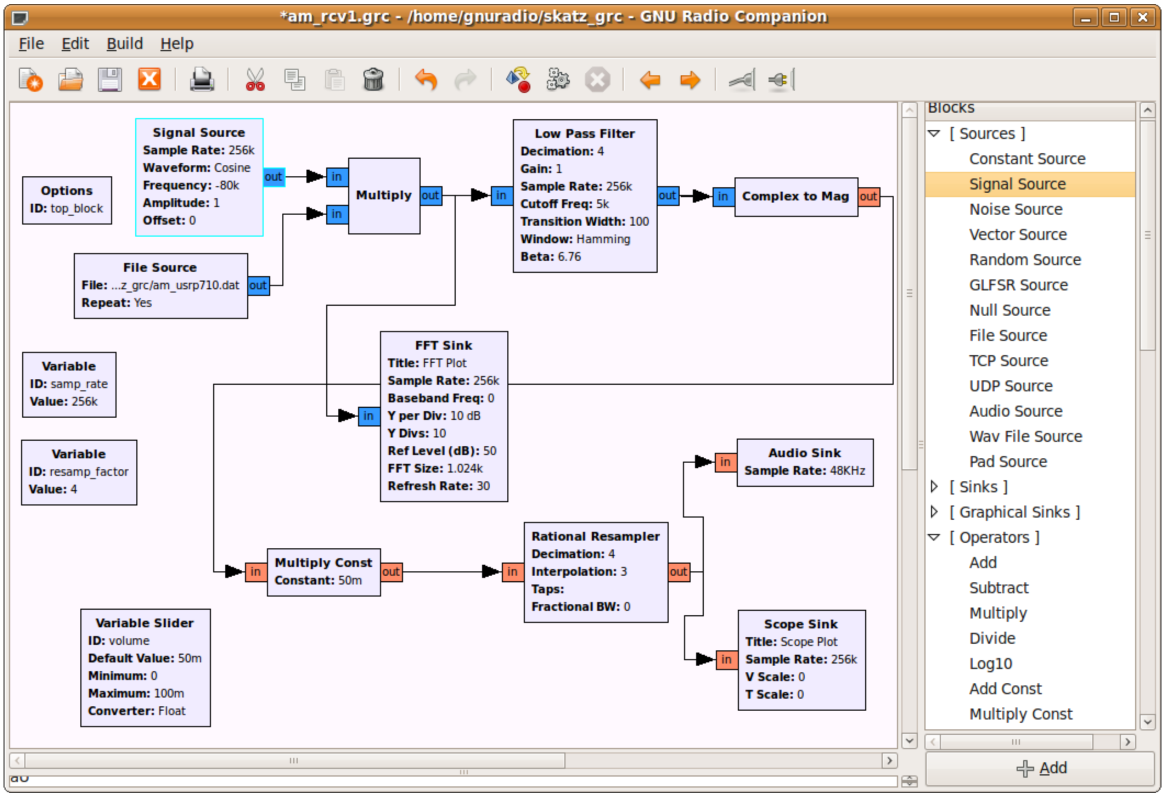
21. Execute the flow graph. You will probably hear a distorted signal coming out of the speaker. Reduce the slider until the audio is easy to understand. This will probably be around 3m (.003). You now have a functioning AM receiver.

22. Double click on the Low Pass Filter. Note that it can also decimate. Change the decimation in the filter block to resamp_factor and remove the Rational Resampler. The filter now handles both operations. Test the receiver again.

23. Place an FFT sink at the output of the file source, leaving the rest of the flow graph unchanged. Execute the flow graph and observe the location of the other stations in the spectrum. Note that there is a fairly strong signal at 80 KHz (really 710 + 80 = 790 KHz).

24. In order to receive this signal we need to shift it down to zero frequency so that it will pass through the low pass filter. One way to accomplish this is to multiply it by a sinusoid. Modify the flow graph as shown below. Add a Signal Source and set its parameters to output a Cosine at a frequency of -80000. This negative frequency will shift the entire spectrum to

the left by 80KHz. Use a Multiply block and move the FFT sink to observe its output. Test this receiver.



Options
ID: top_block

Signal Source
Sample Rate: 256k
Waveform: Cosine
Frequency: -80k
Amplitude: 1
Offset: 0

Multiply

Low Pass Filter
Decimation: 4
Gain: 1
Sample Rate: 256k
Cutoff Freq: 5k
Transition Width: 100
Window: Hamming
Beta: 6.76

Complex to Mag

File Source
File: ...z_grc/am_usrp710.dat
Repeat: Yes

Variable
ID: samp_rate
Value: 256k

Variable
ID: resamp_factor
Value: 4

FFT Sink
Title: FFT Plot
Sample Rate: 256k
Baseband Freq: 0
Y per Div: 10 dB
Y Divs: 10
Ref Level (dB): 50
FFT Size: 1.024k
Refresh Rate: 30

Audio Sink
Sample Rate: 48KHz

Multiply Const
Constant: 50m

Rational Resampler
Decimation: 4
Interpolation: 3
Taps:
Fractional BW: 0

Variable Slider
ID: volume
Default Value: 50m
Minimum: 0
Maximum: 100m
Converter: Float

Scope Sink
Title: Scope Plot
Sample Rate: 256k
V Scale: 0
T Scale: 0

25. Add another Variable slider so that you can adjust the frequency with a slider. Test your flow graph and demonstrate that it works. You may need to adjust your volume slider for each station. This is because the stations are at varying distances away from the receiver and have different transmitted power. (Remember the link equation?)

26. The volume adjustment can be automated with an Automatic Gain Control (AGC) block. This block works by sampling its own output and adjusting its gain to keep the average output at a particular level. Insert the AGC2 block (found under Level Controls) between the Low Pass Filter and the Complex to Mag block. The audio card can only accept values between -1.0 and 1.0. We want the AGC to adjust the gain so that the sample values are always in this range. The Threshold parameter controls this. Set Threshold to 0.2. This means the AGC block will adjust its gain so that the average output value is 0.2. We will discuss the other parameters later. Set the Attack to 6.25E-4 and the Decay to 1E-5. Leave the other parameters at their default values. Set the range of the Volume variable from 0 to 1. Test the receiver again. Adjust the volume to a comfortable level on the first station you

hear and tune up and down the band. Notice you no longer need to adjust the volume. The radio functions the same as a hardware radio.