# **GNU Radio Tutorials**

Labs 1 – 5

Balint Seeber Ettus Research

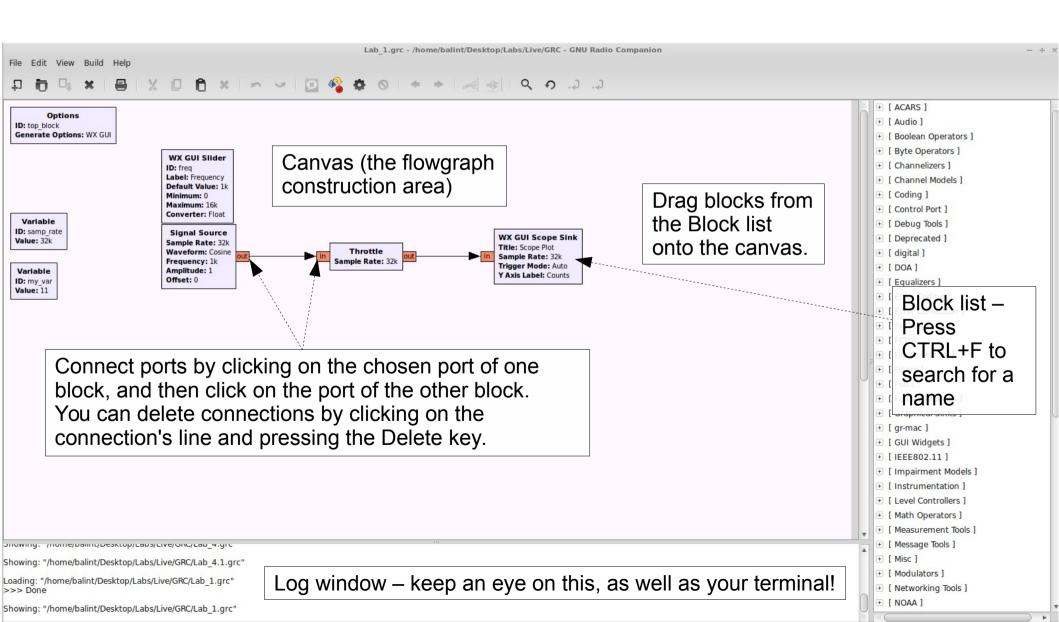
Version 1.0 (18<sup>th</sup> April 2014)

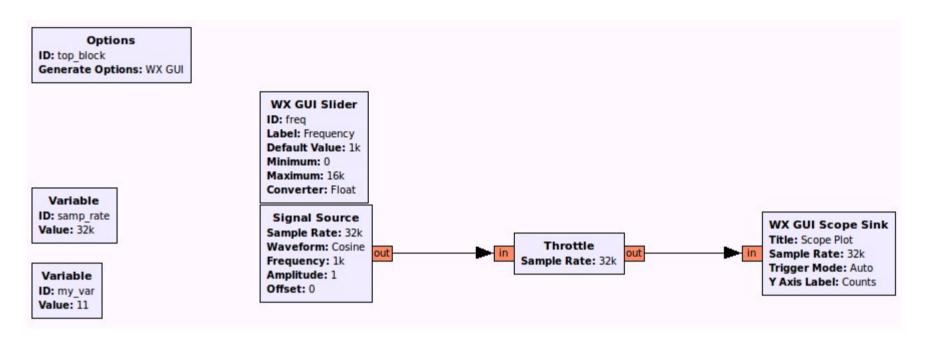
Comments & suggetions welcome:

balint@ettus.com

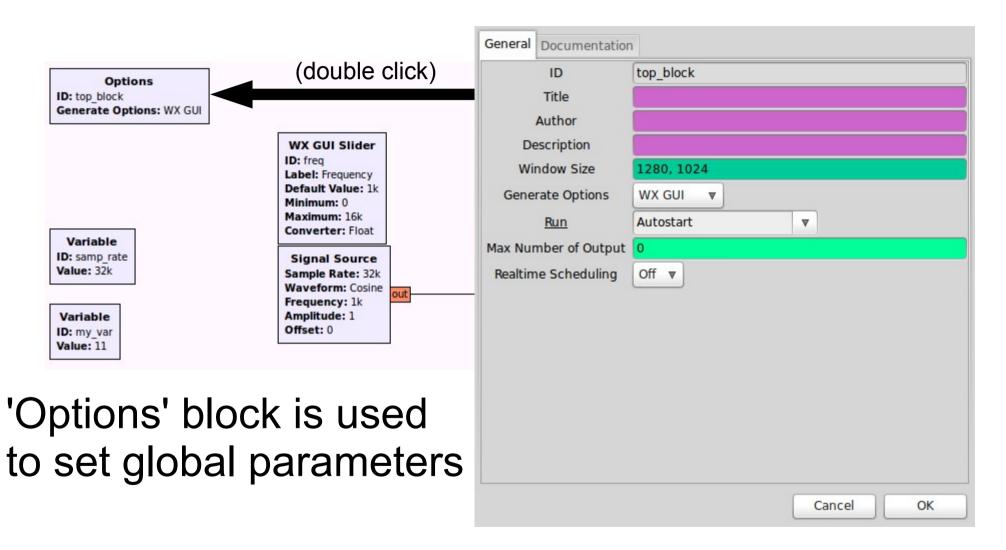
@spenchdotnet

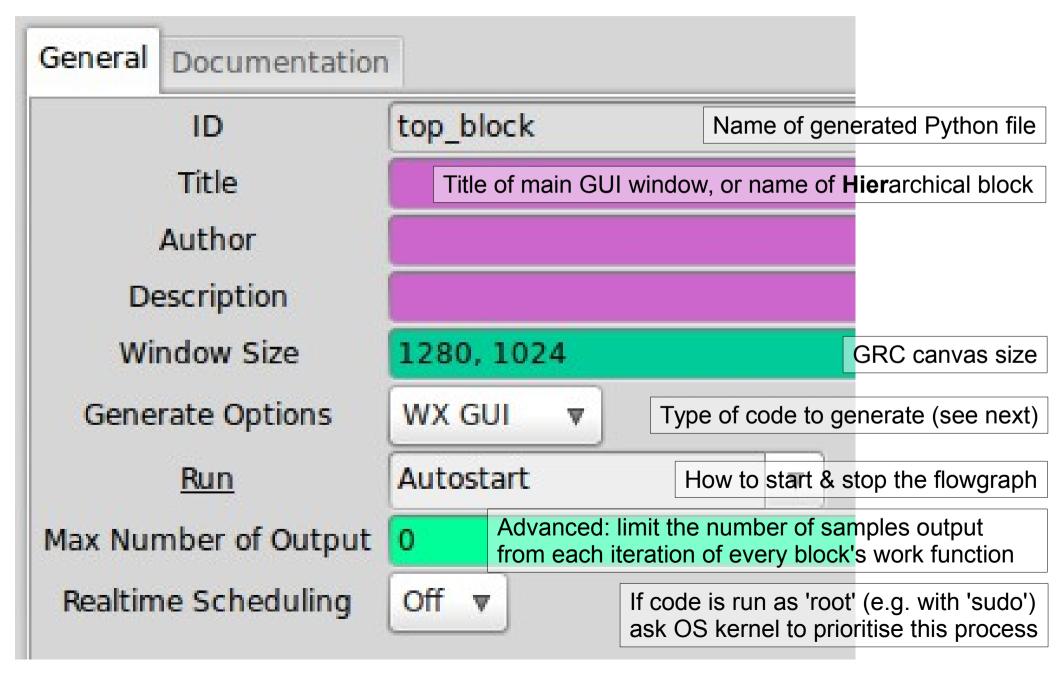
- Open GNU Radio Companion:
  - Open a Terminal/Console/Command Prompt
  - Run 'gnuradio-companion'

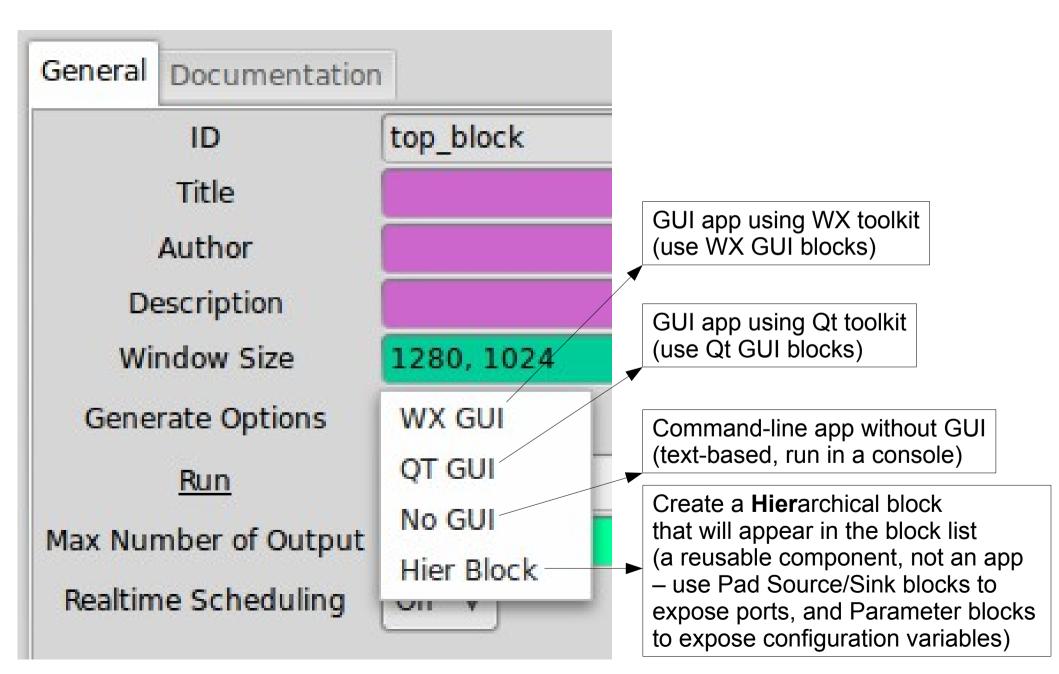


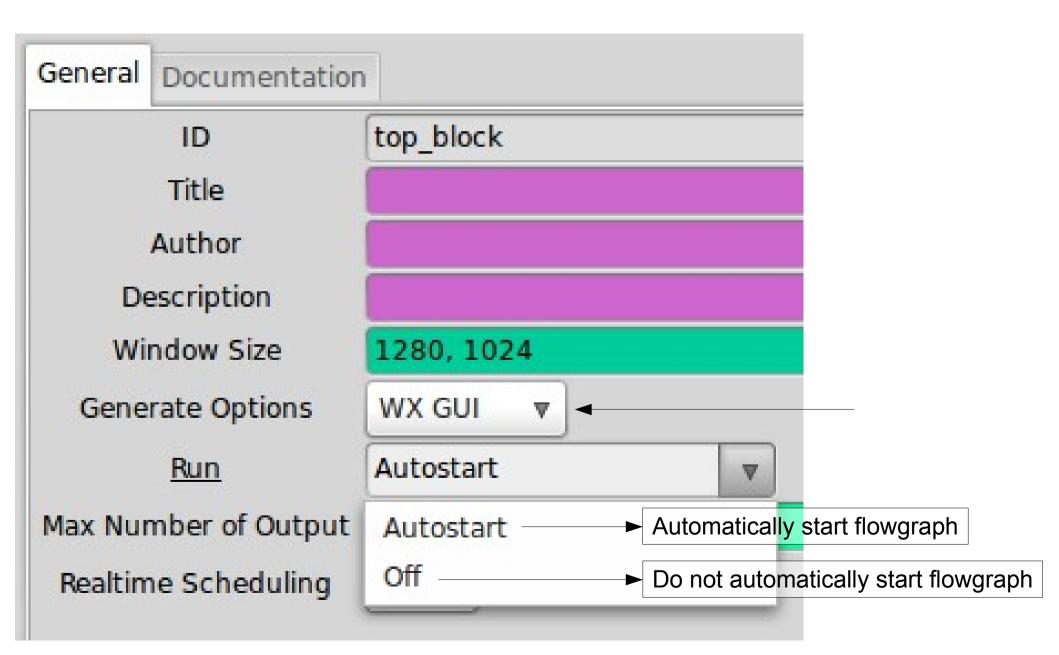


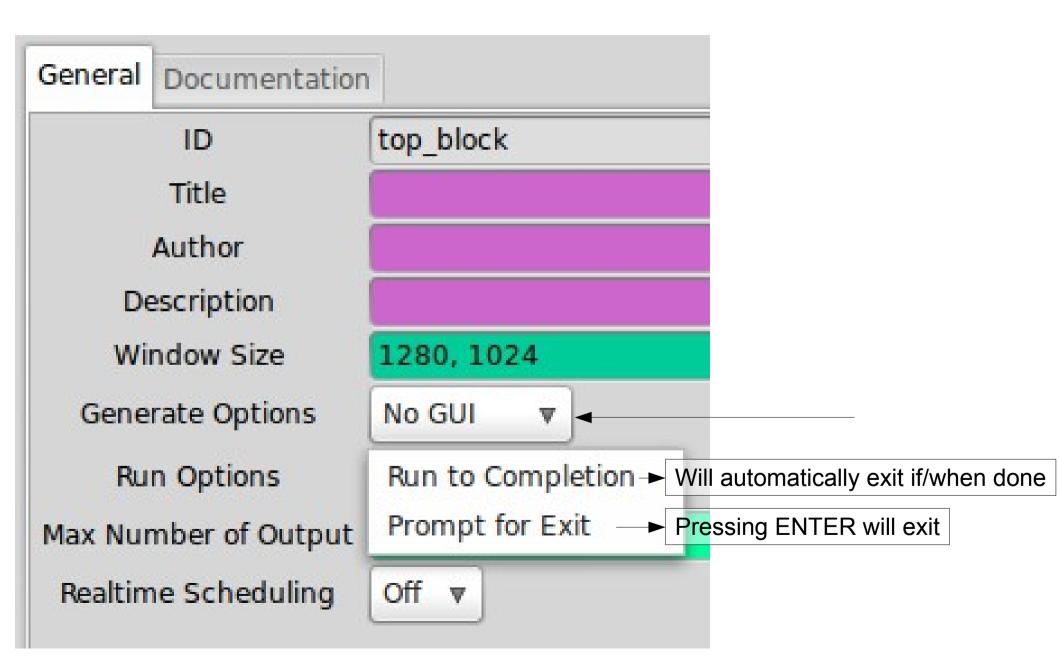
Create a sine wave & inspect the generated samples with a (time-domain) Scope Sink

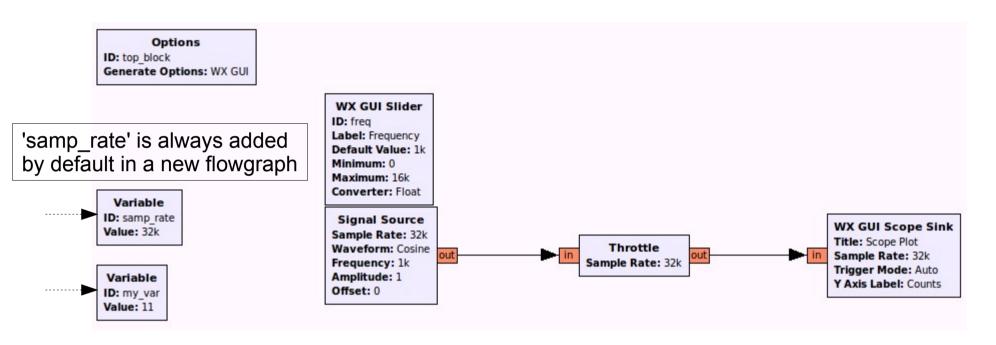






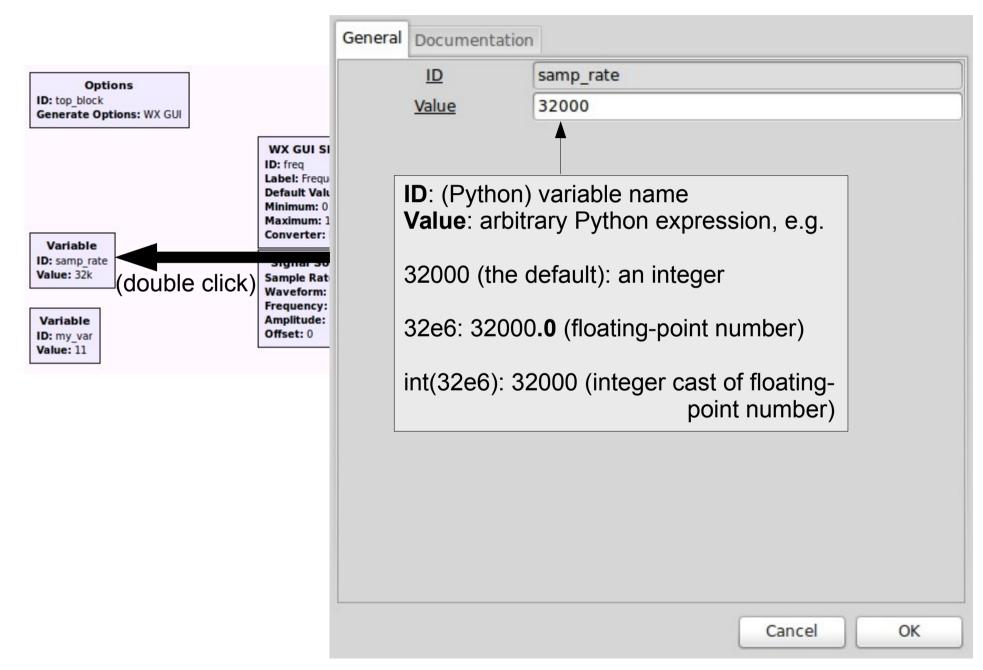


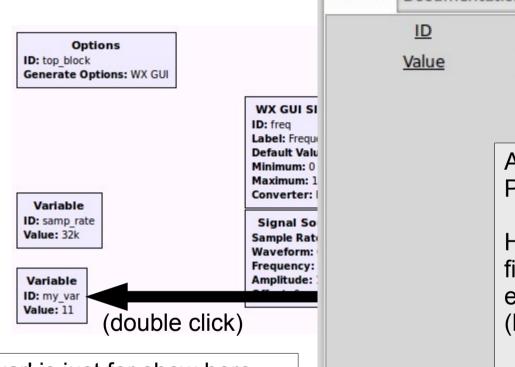




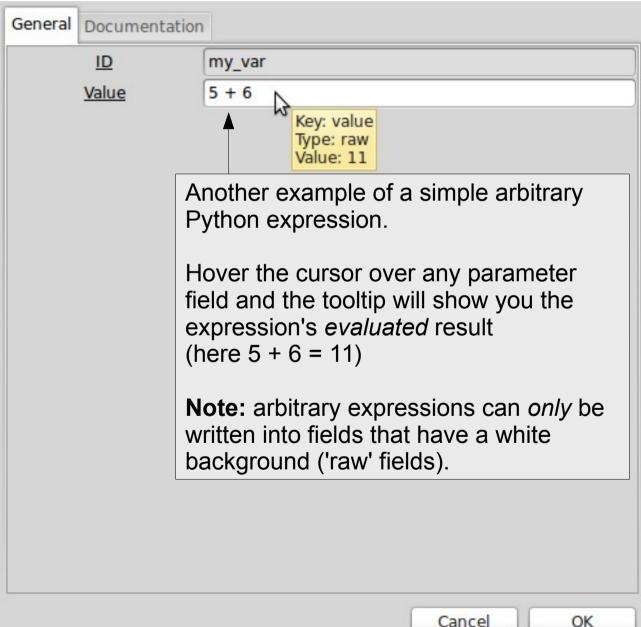
Variable: a block that contains an arbitrary Python expression.

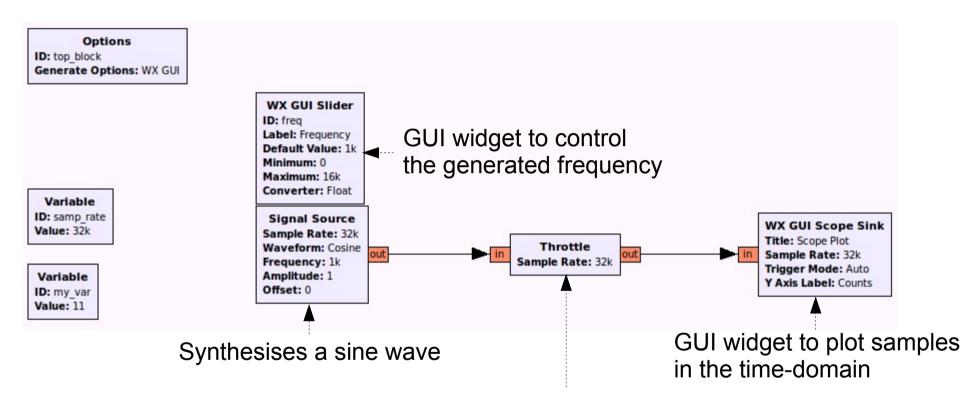
You can refer to it in another block by its **ID**.



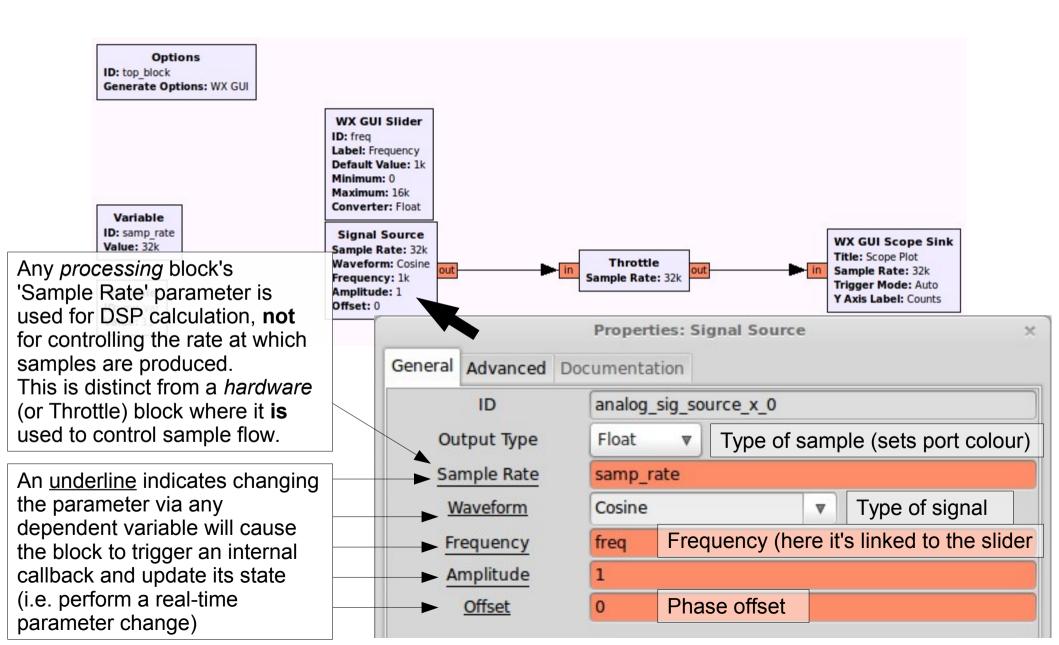


'my\_var' is just for show here (it doesn't actually do anything useful in this flowgraph).





Will throttle the rate at which samples pass through this block (thus setting the rate at which samples pass through the whole flow graph).



# Sample Rate (DSP)

- If calculating a sine wave where a given frequency in Hertz is desired, you actually need to know the sample rate too. This is because the mathematical representation requires both values to calculate the individual sample amplitude at any specific point in time.
- The actual sample rate value used can be anything. It just so happens you'll usually use the same value as in the rest of your flowgraph so that everything will be consistent (operate in the same sample rate domain).

# Sample Rate (DSP)

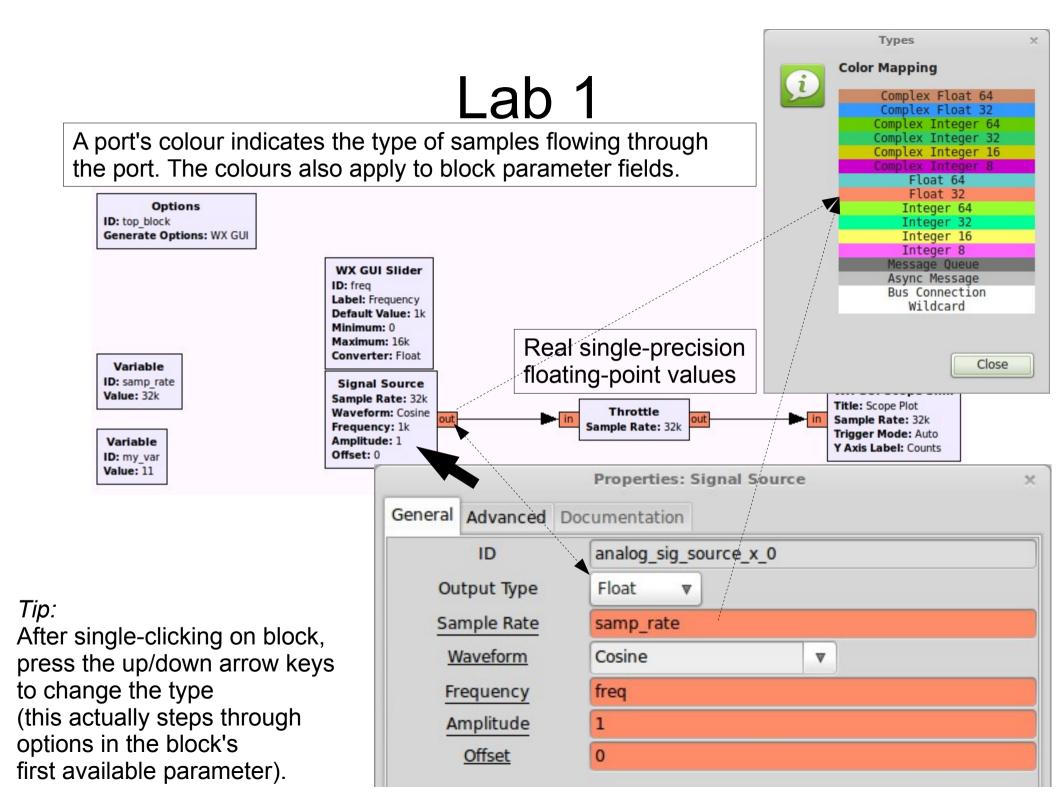
 Think of it as being used to calculate the discrete step size from one sample to the next within a DSP operation (e.g. the time step when calculating the amplitude of the next sample in the sine wave generator)

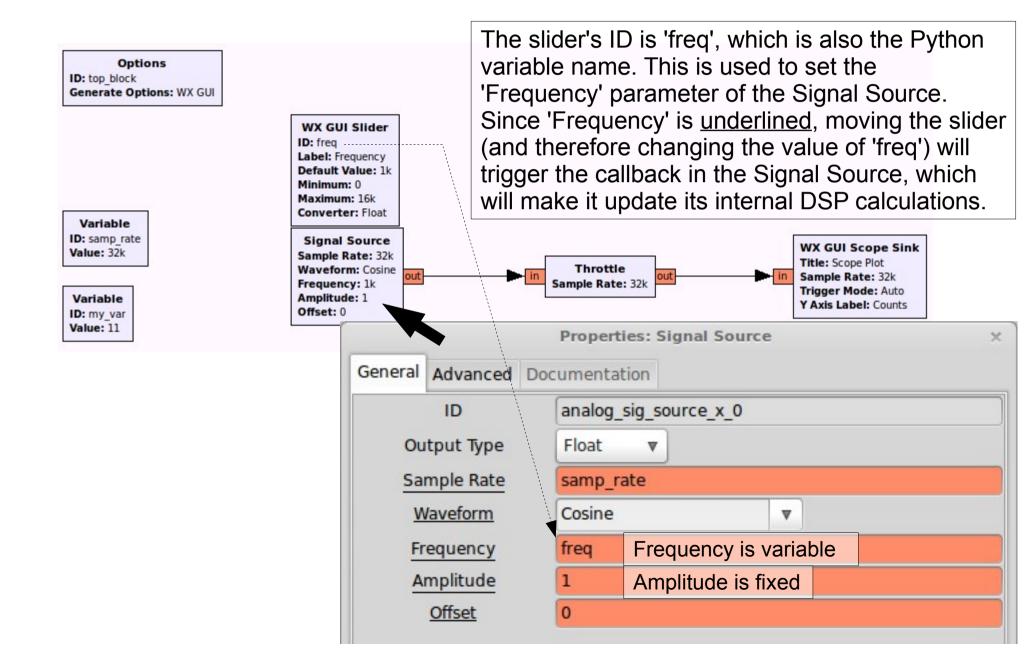
- Distinct from mathematical (DSP) calculation, sample rate also refers to the rate at which samples pass through the flowgraph.
- If there is no rate control, hardware clock or throttling mechanism, the samples will be generated, pass through the flowgraph and be consumed as fast as possible (i.e. the flowgraph will be CPU bound).
- This is desirable if you want to perform some fixed DSP on stored data as quickly as possible (e.g. read from a file, resample and write it back).

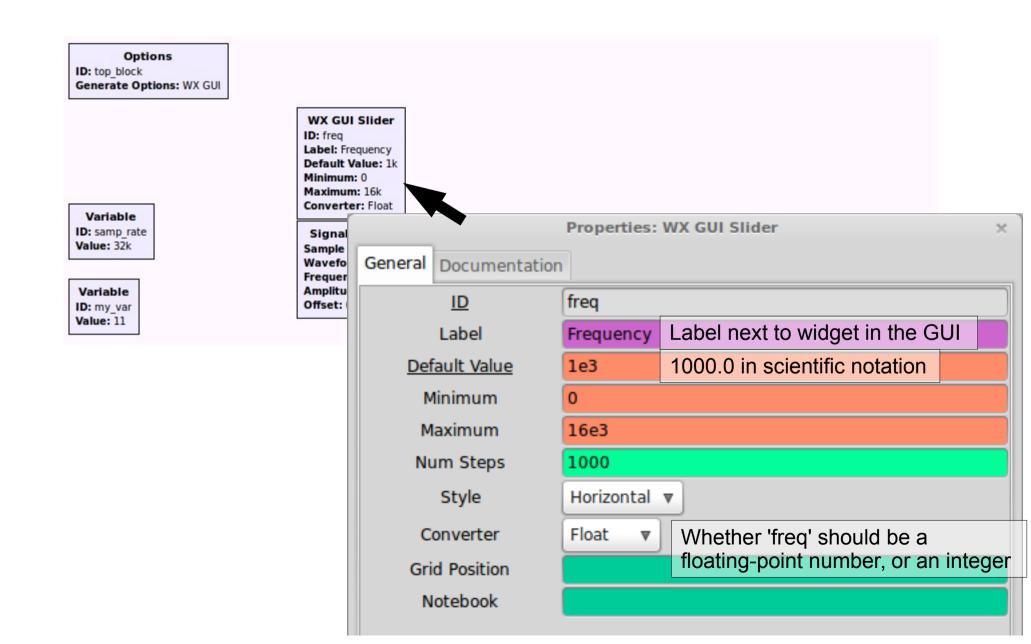
- Only a block that represents some underlying hardware with its own clock (e.g. USRP, sound card), or the Throttle Block, will use 'Sample Rate' to set that hardware clock, and therefore have the effect of applying rate control to the samples in the flowgraph.
- A Throttle Block will simply apply host-based timing (against the 'wall clock') to control the rate of the samples it produces (i.e. samples that it makes available on its outputs to downstream blocks).

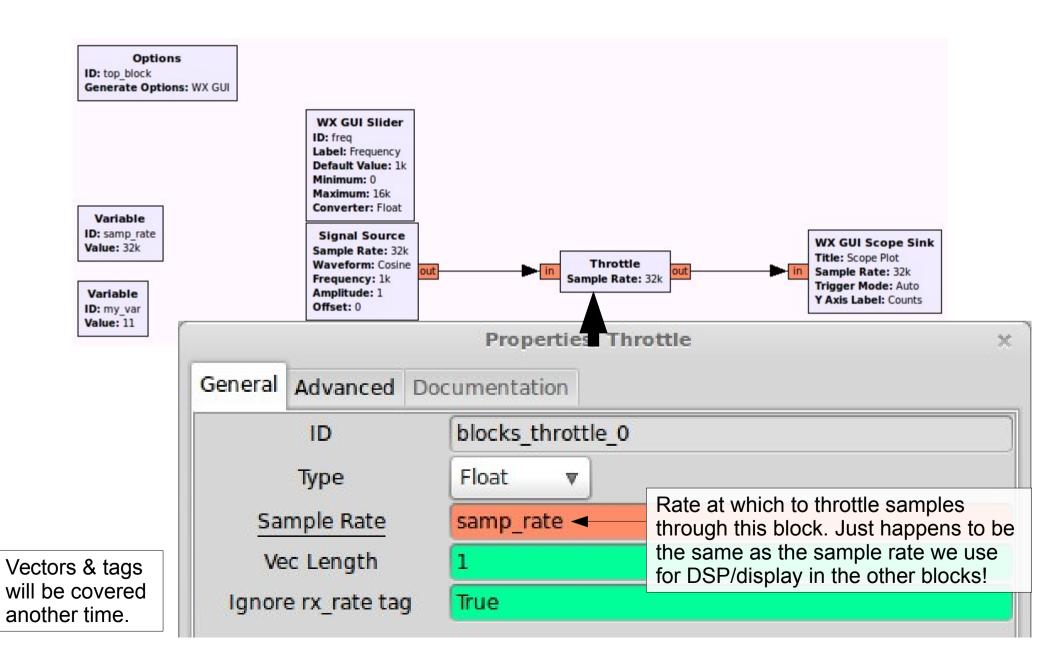
- A hardware Sink block will consume samples at a fixed rate (relative to the wall clock)
- The Throttle Block, or a hardware Sink block, will apply 'back pressure' to the upstream blocks (the rate of work of the upstream blocks will be limited by the throttling effect of this ratecontrolling block)
- A hardware Source block will produce samples at a fixed rate (relative to the wall clock)

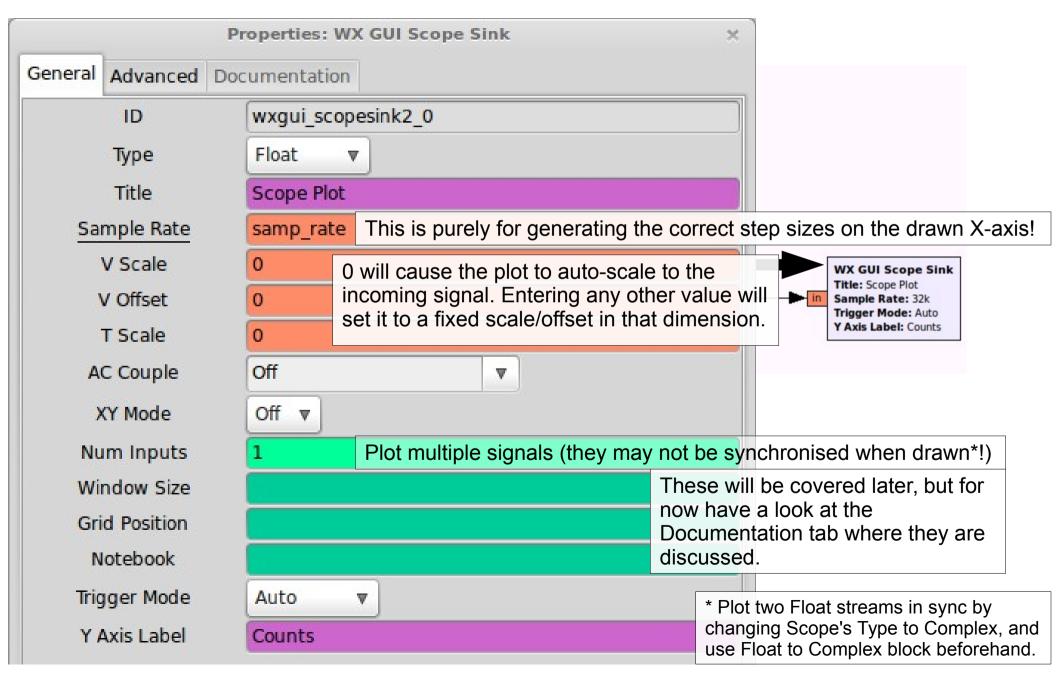
- In general, there should only ever be one block in a flowgraph that has the ability to throttle sample flow.
- Otherwise you need to be very careful with multiple, unsynchronised clock sources: they will eventually go out of sync and cause overflows/underruns as their production/consumption rates will differ.
  - This is the 'two clock' problem (discussed later)
  - Work arounds: allow non-blocking I/O, and/or tweak resampling rates to account for the clock offsets

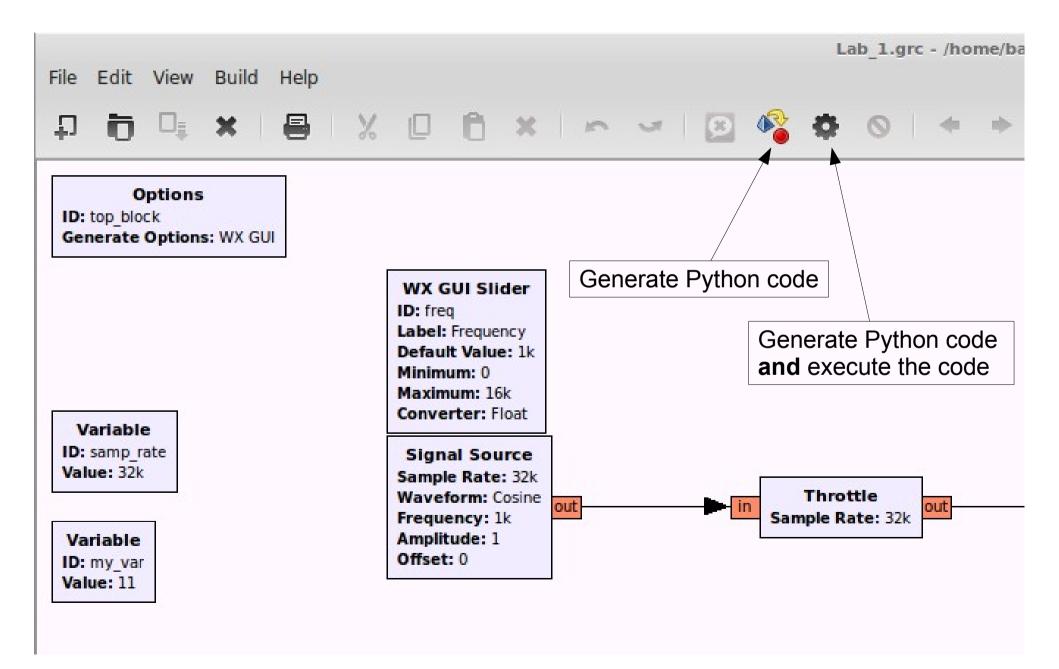












#### Python code generated by GRC

```
#!/usr/bin/env pvthon

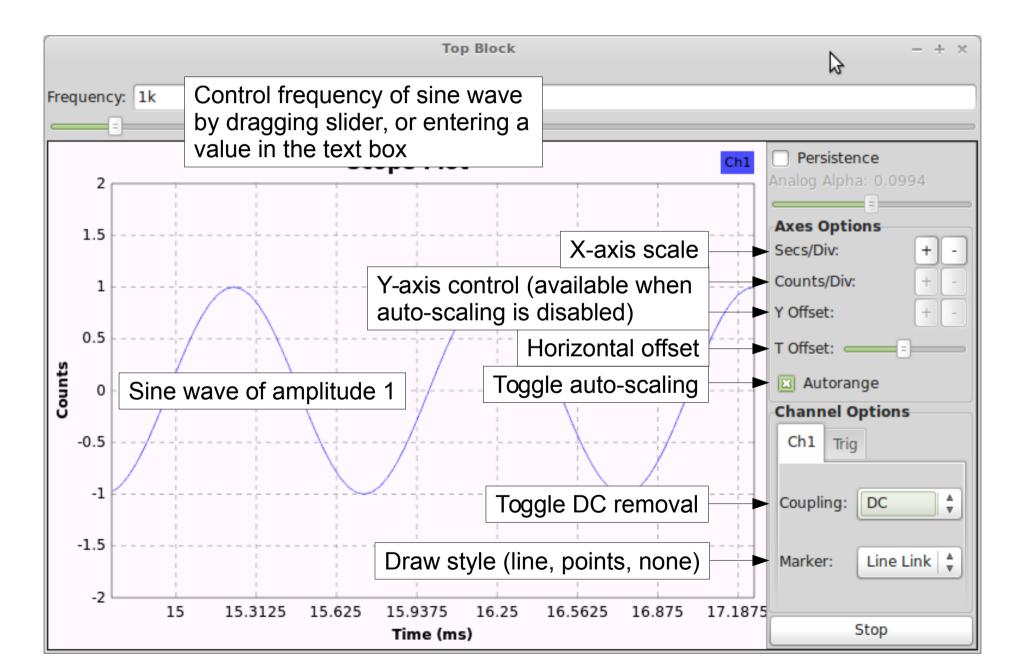
→ trig mode=wxgui.TRIG MODE AUTO,

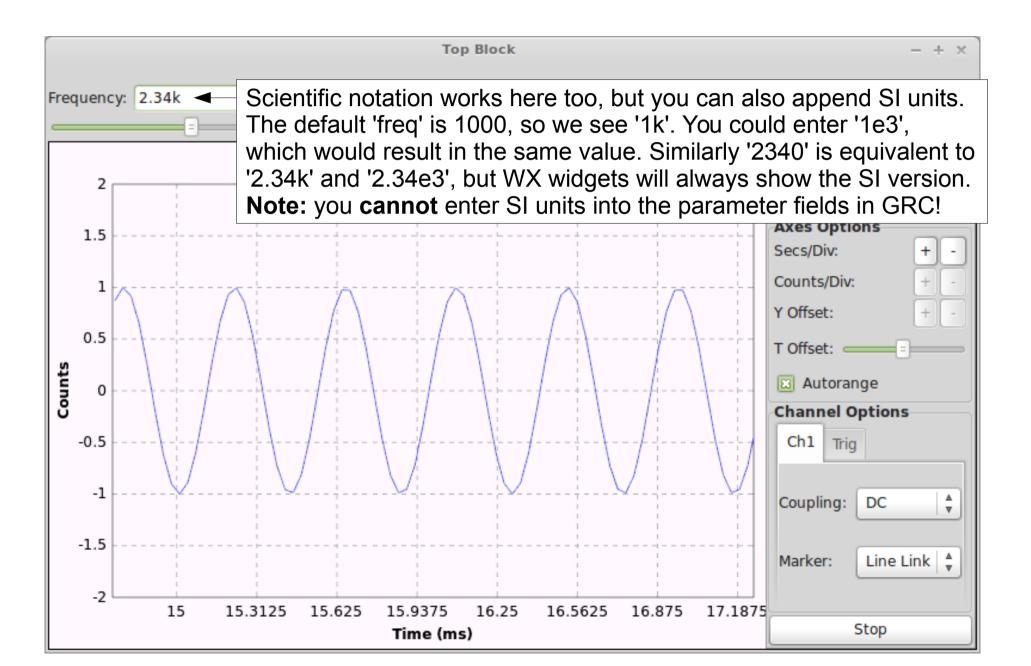
*************************************
                                                      ·····y axis label="Counts",
# Gnuradio Python Flow Graph
# Title: Top Block
                                                            self.Add(self.wxqui scopesink2 0.win)
# Generated: Wed Apr 16 14:11:52 2014
                                                            self.blocks throttle 0 = blocks.throttle(gr.sizeof float*1, samp rate, True)
*************************************
                                                            self.analog sig source x \theta = analog.sig source f(samp rate, analog.GR COS WAVE, freg, 1, \theta)
from gnuradio import analog
                                                            from gnuradio import blocks
                                                            # Connections
from gnuradio import eng notation
                                                            from gnuradio import gr
                                                            self.connect((self.analog sig source x 0, 0), (self.blocks throttle 0, 0))
from gnuradio import wxgui
                                                            self.connect((self.blocks throttle 0, 0), (self.wxgui scopesink2 0, 0))
from gnuradio.eng option import eng option
from gnuradio.filter import firdes
from gnuradio.wxgui import forms
                                                     # QT sink close method reimplementation
from gnuradio.wxgui import scopesink2
from grc gnuradio import wxgui as grc wxgui
                                                        def get samp rate(self):
from optparse import OptionParser
                                                     return self.samp rate
import wx
                                                      ---def set samp rate(self, samp rate):
class top block(grc wxgui.top block gui):
                                                      self.samp rate = samp rate
                                                     self.analog sig source x 0.set sampling freg(self.samp rate)
   def init (self):
                                                      self.wxgui scopesink2 0.set sample rate(self.samp rate)
      grc wxgui.top block_gui.__init__(self, title="Top Bloc self.blocks throttle 0.set sample rate(self.samp rate)
      *************************************
                                                      ---def get my var(self):
      # Variables
                                                      ······return self.my var
      self.samp rate = samp rate = 32000
                                                      ---def set my var(self, my var):
      self.mv var = mv var = 5 + 6
                                                     ····self.my var = my var
 self.freq = freq = 1e3
                                                      def get freq(self):
   -----
                                                     ·····return self.freq
      def set freq(self, freq):
     freq sizer = wx.BoxSizer(wx.VERTICAL)
                                                     ····self.freq = freq
      self. freq text box = forms.text box(
                                                     self.analog sig source x 0.set frequency(self.freq)
      → parent=self.GetWin(),
                                                     self. freq slider.set value(self.freq)

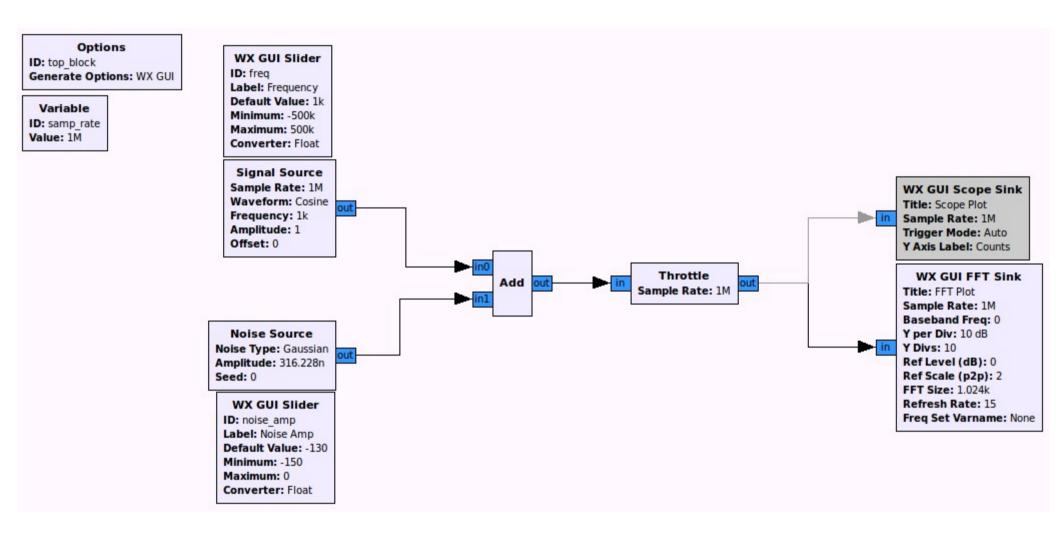
→ sizer= freq sizer,
                                                     self. freq text_box.set_value(self.freq)
     ··· -----callback=self.set freq,
                                                     if name == ' main ':
       → label="Frequency",
                                                      import ctypes
          converter=forms.float converter(),
                                                      · · · import sys
         →proportion=0,
                                                        if sys.platform.startswith('linux'):
      self. freq slider = forms.slider(
                                                          ----x11 = ctypes.cdll.LoadLibrary('libX11.so')
       →parent=self.GetWin(),
                                                      x11.XInitThreads()

→ sizer= freq sizer,
                                                      · · · · · · · except:
     ··---value=self.freq,
                                                      .....print "Warning: failed to XInitThreads()"

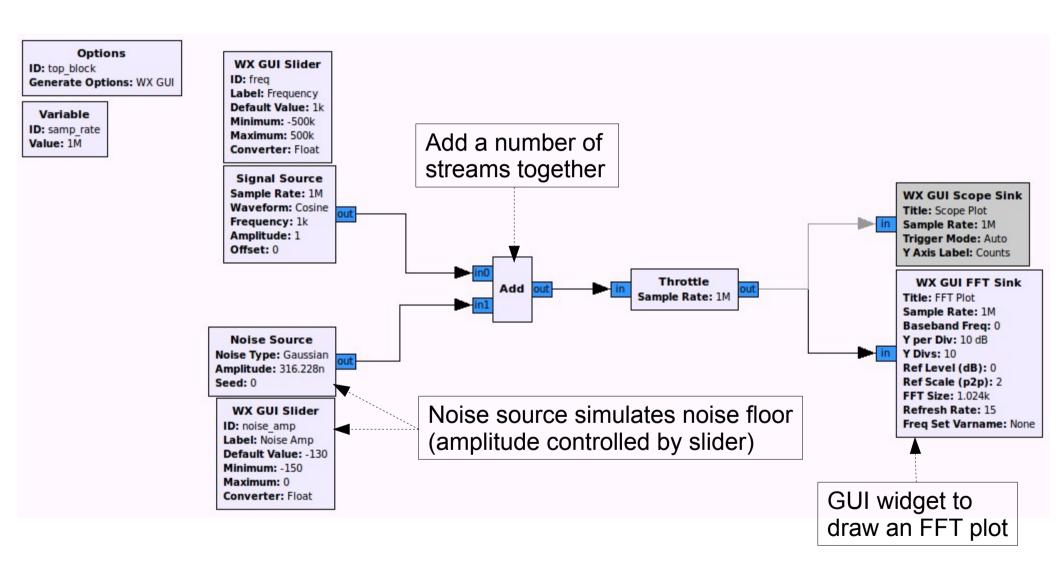
→ callback=self.set_freq,
                                                     parser = OptionParser(option_class=eng_option, usage="%prog: [options]")
         ∍minimum=0,
                                                     (options, args) = parser.parse args()
       →maximum=16e3,
                                                     ····tb = ·top block()
     ···—→num steps=1000,
                                                     · · · · tb.Start(True)
    ----style=wx.SL_HORIZONTAL,
                                                     ····tb.Wait()
   ····—→cast=float,
```

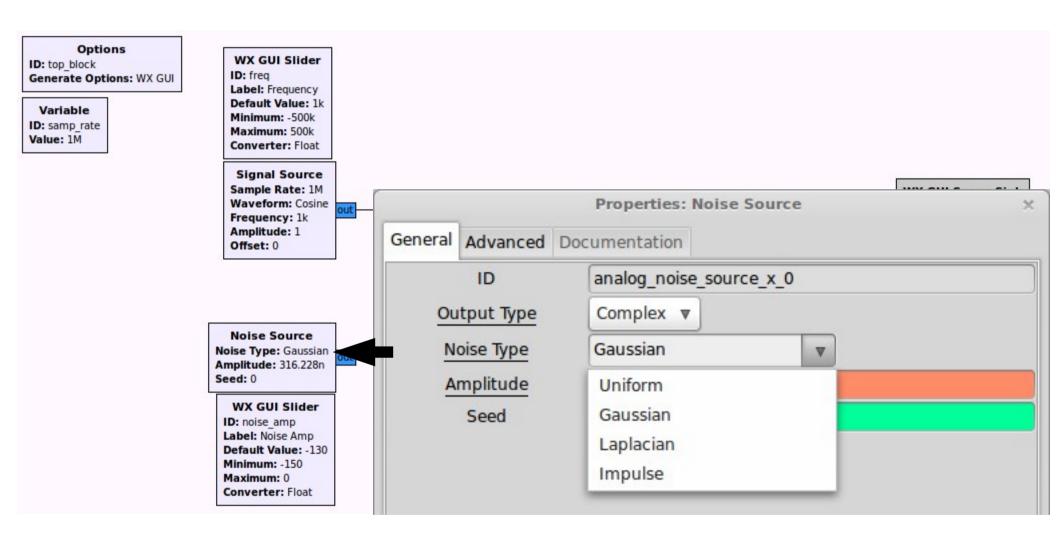


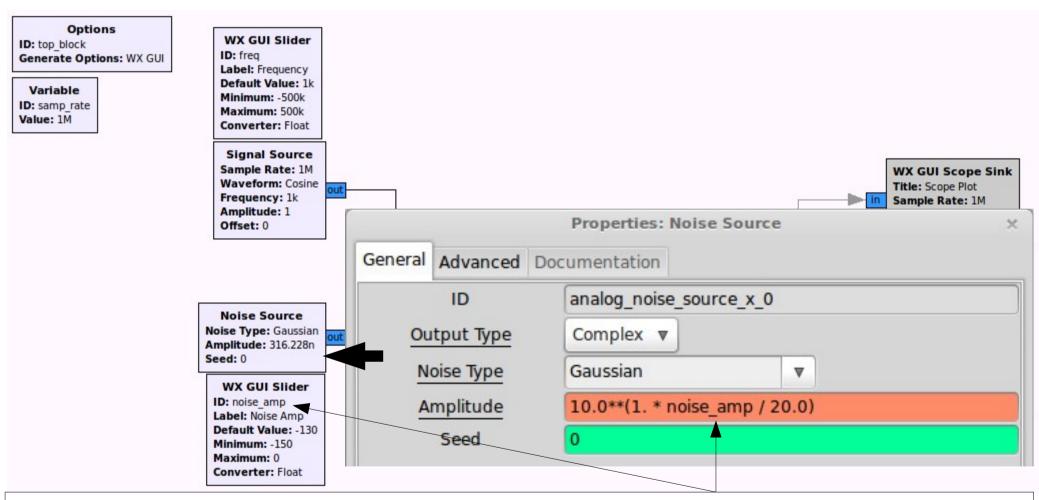




Generate a sine wave & some noise, add both, and plot the resulting signal in the frequency domain.

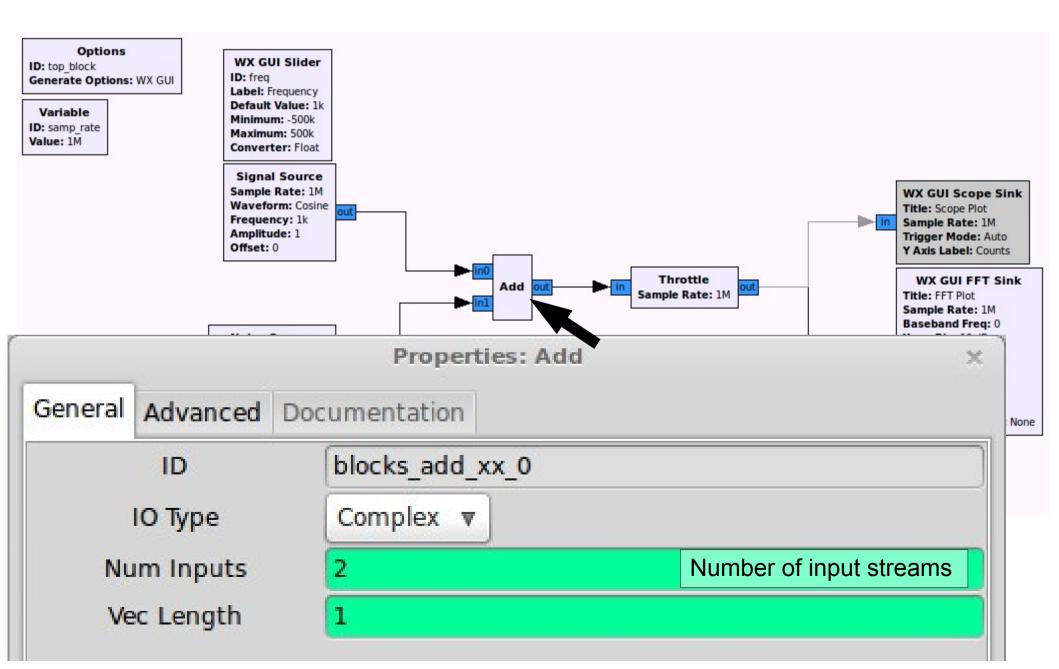


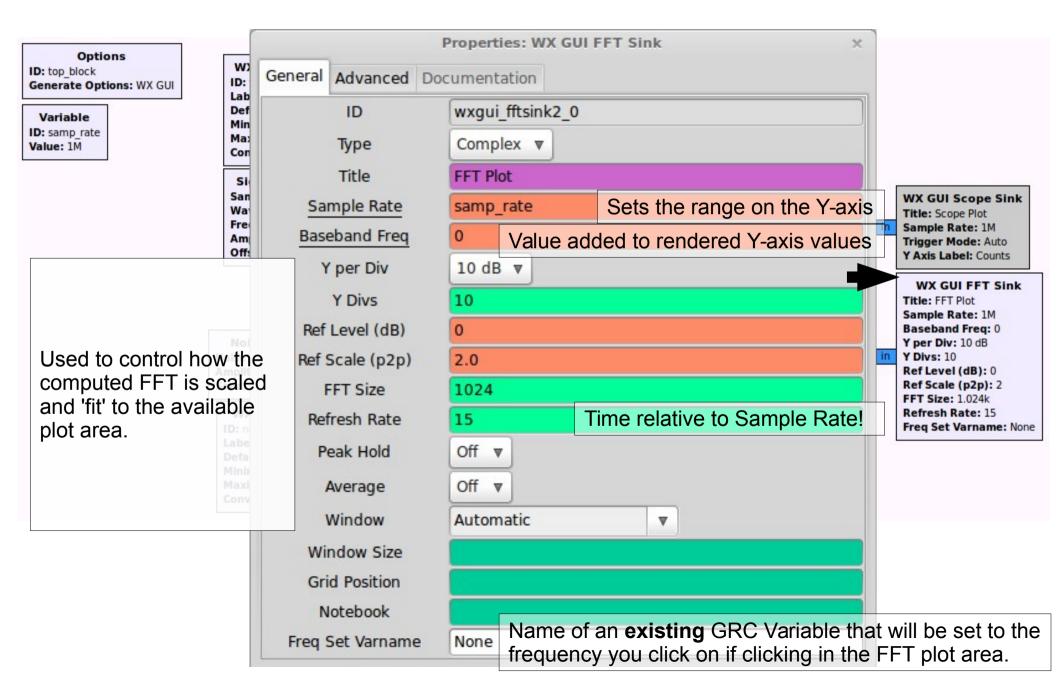


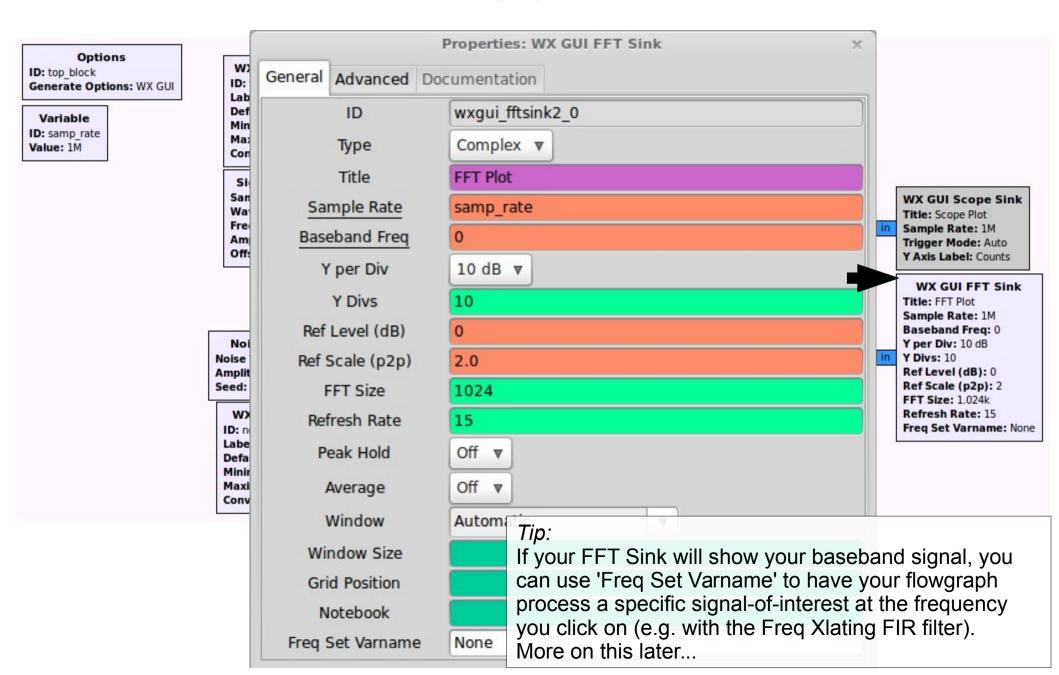


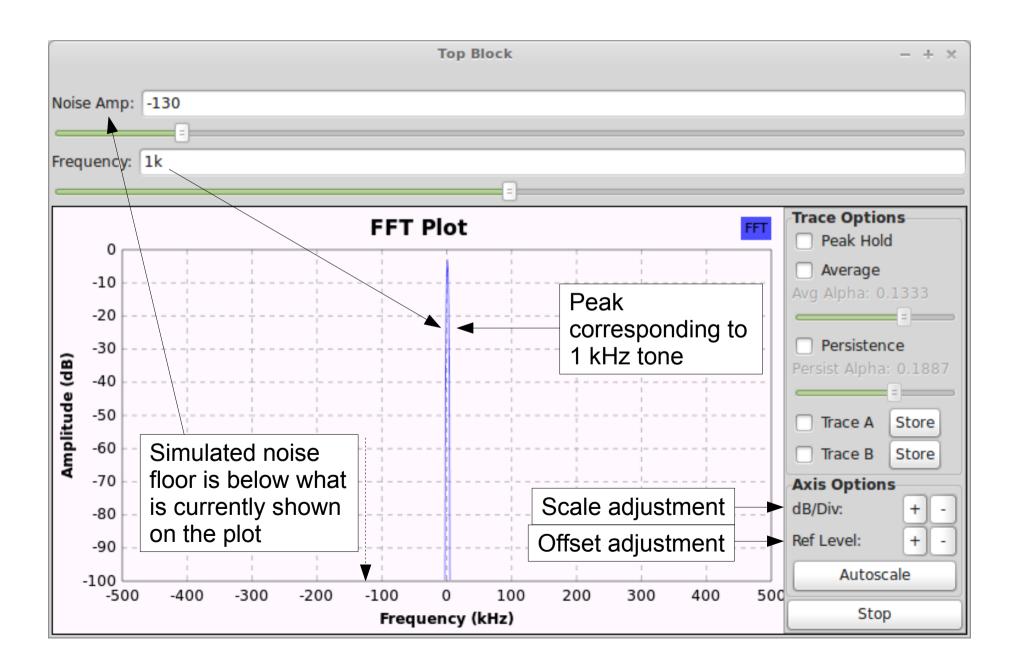
'noise\_amp' is the slider value, which (here) we interpret in dB, as opposed to a linear sample ampltitude value (e.g. '1.0').

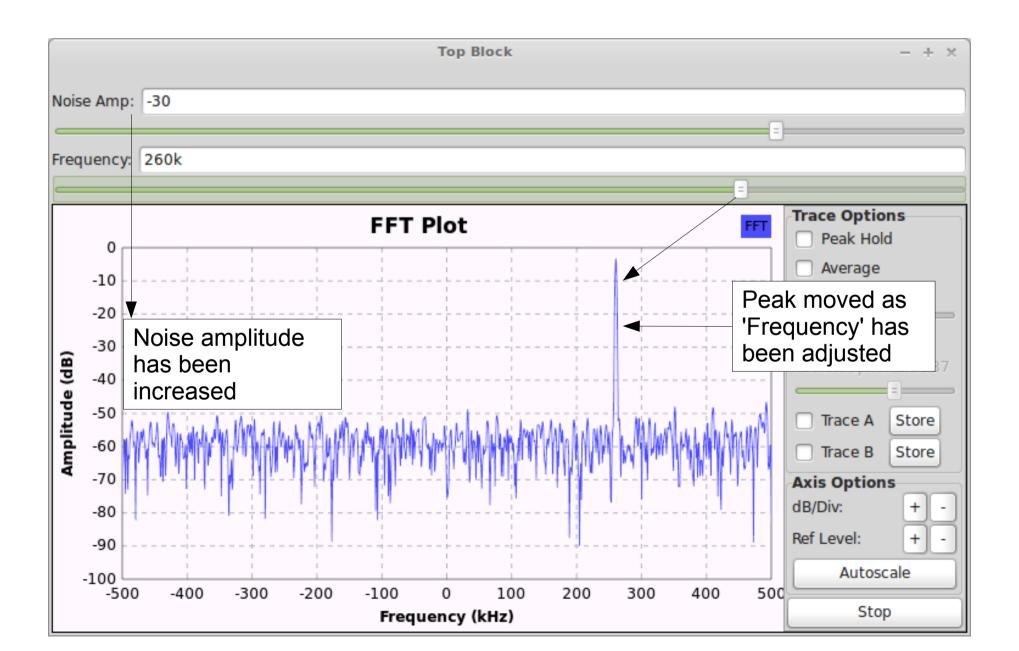
Therefore we need to convert the value in dB to an actual linear ampltiude value ('volts') for use by the block (i.e. reverse the 'log10' function). The decimal points are added to force Python to compute with floating-point values (otherwise it would round and produce integers).

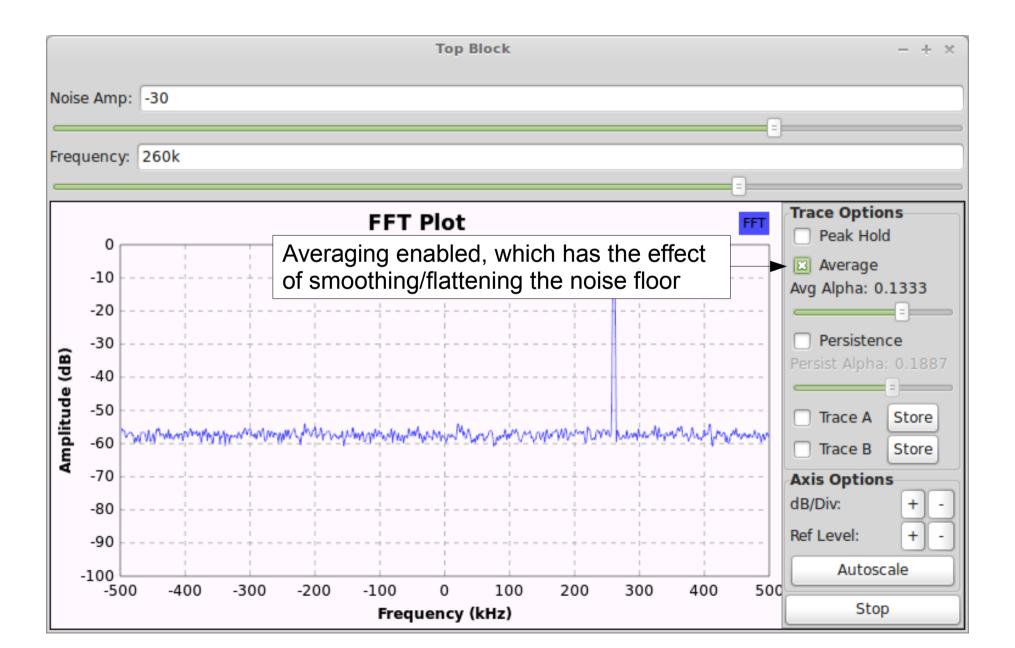


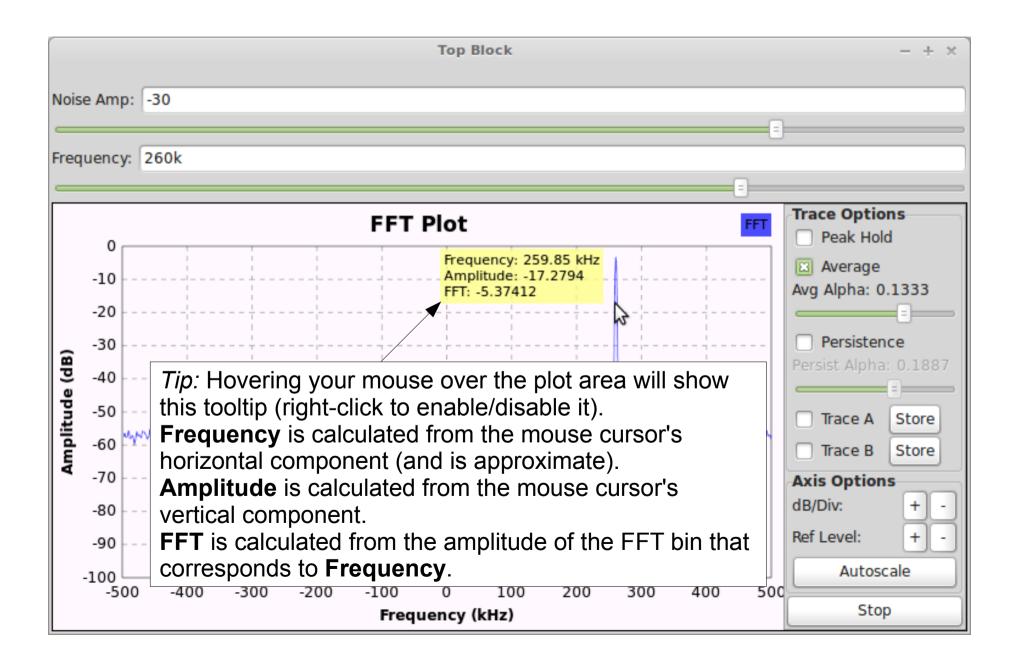


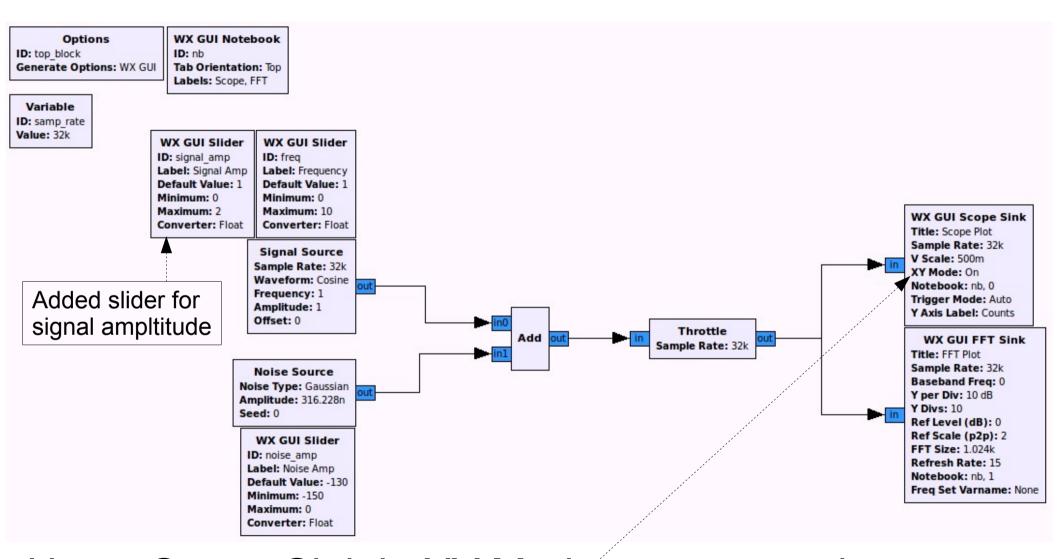




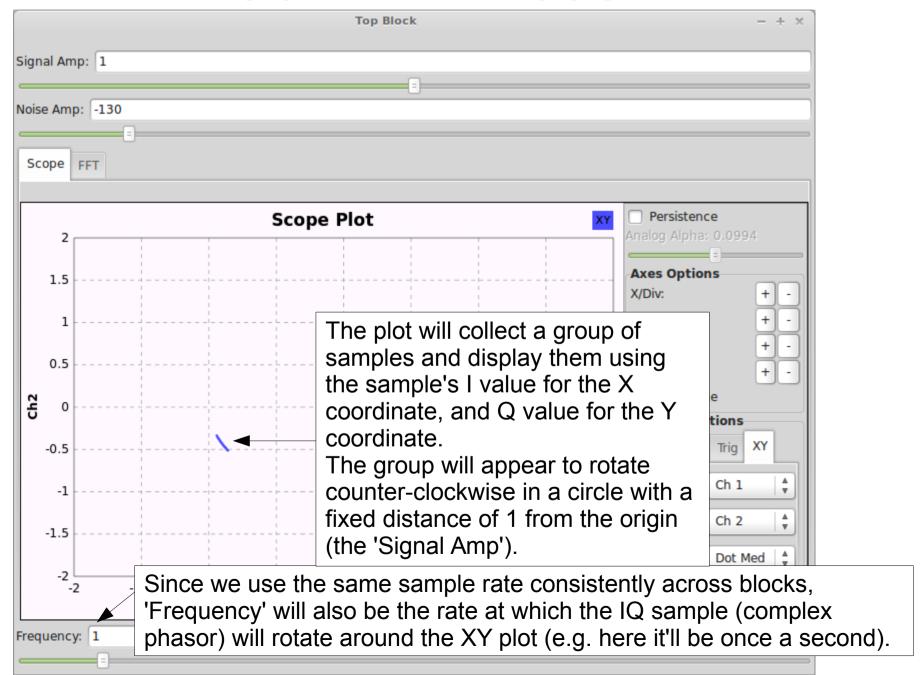


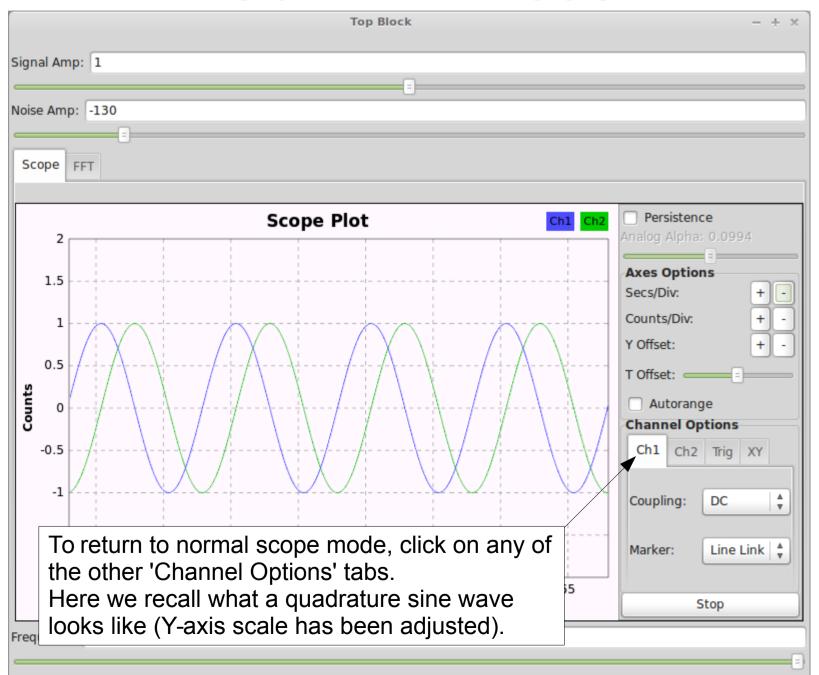


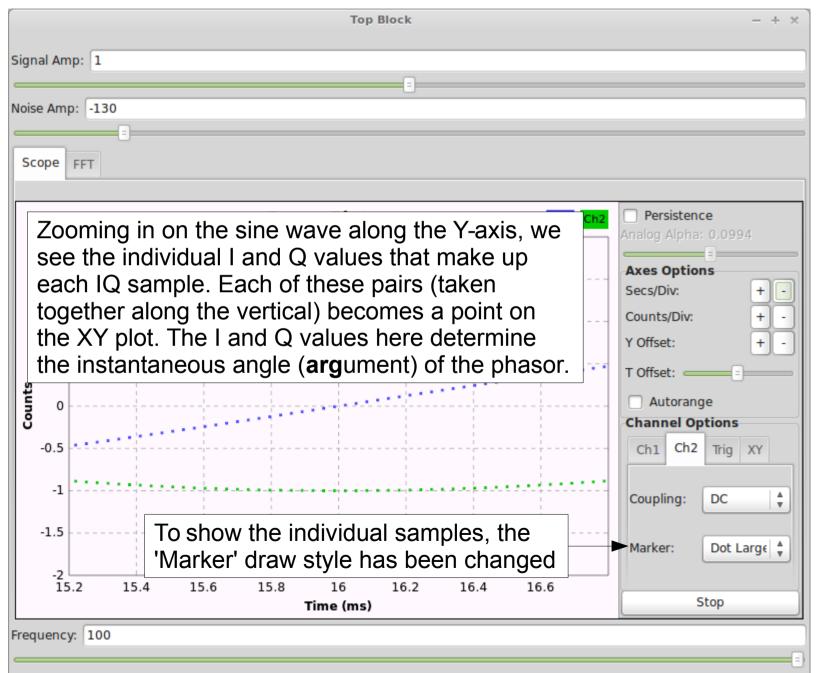


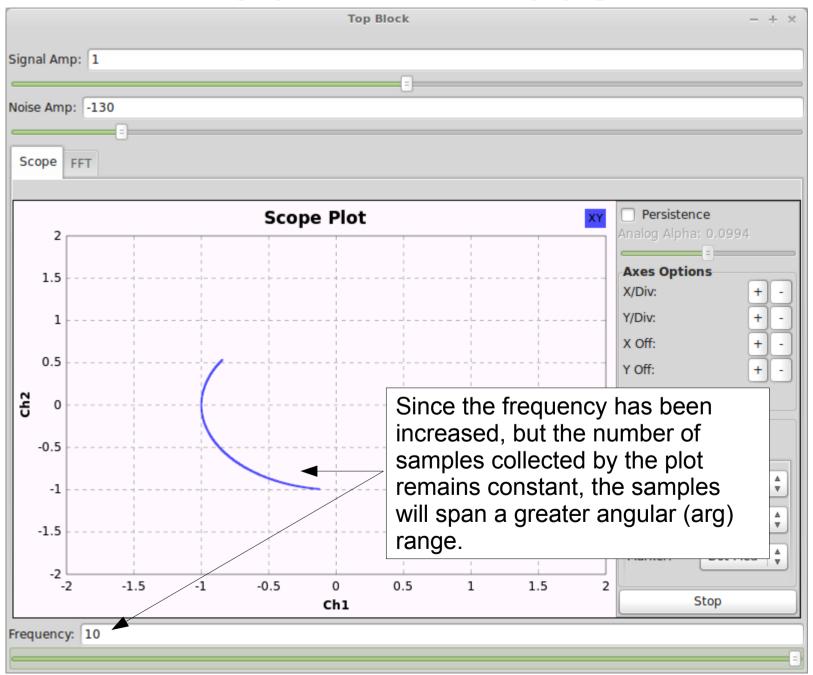


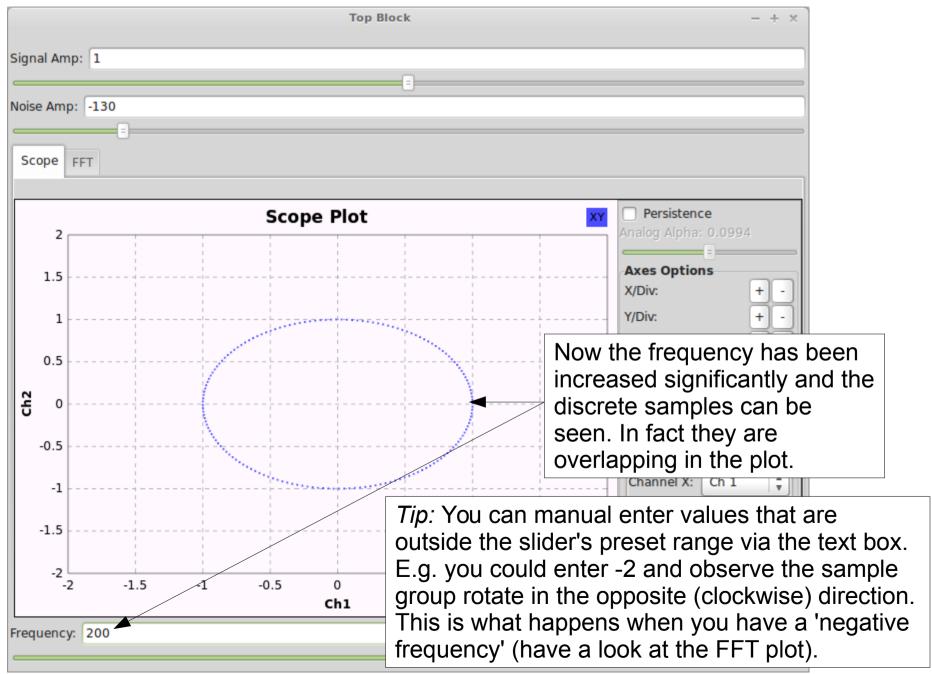
Use a Scope Sink in XY Mode so we can observe the characteristics of an IQ (quadrature) signal

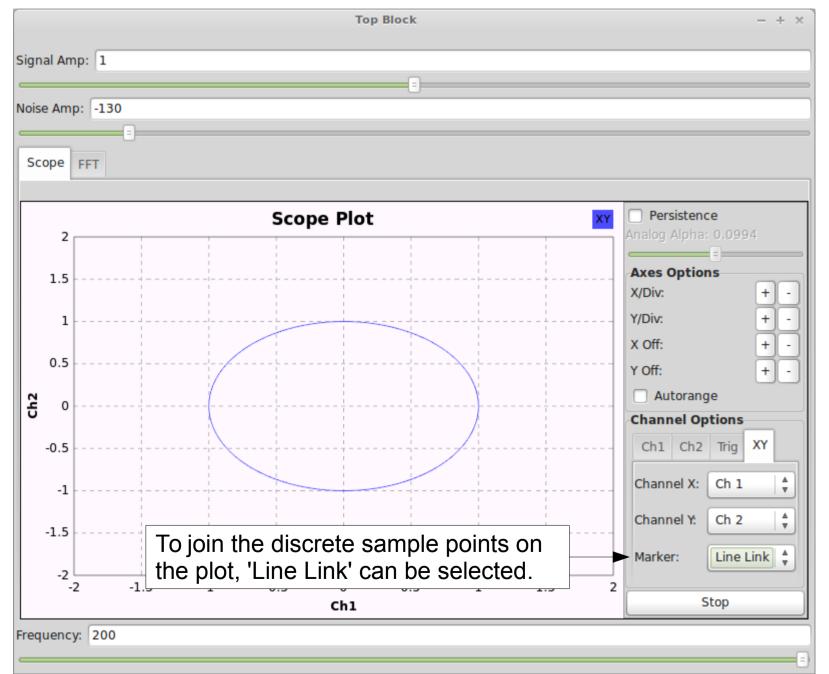


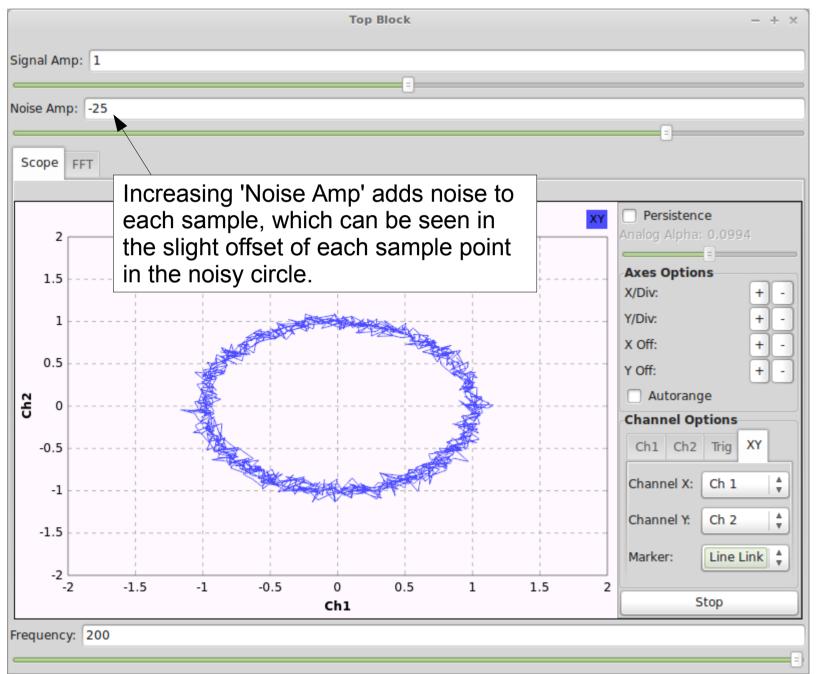


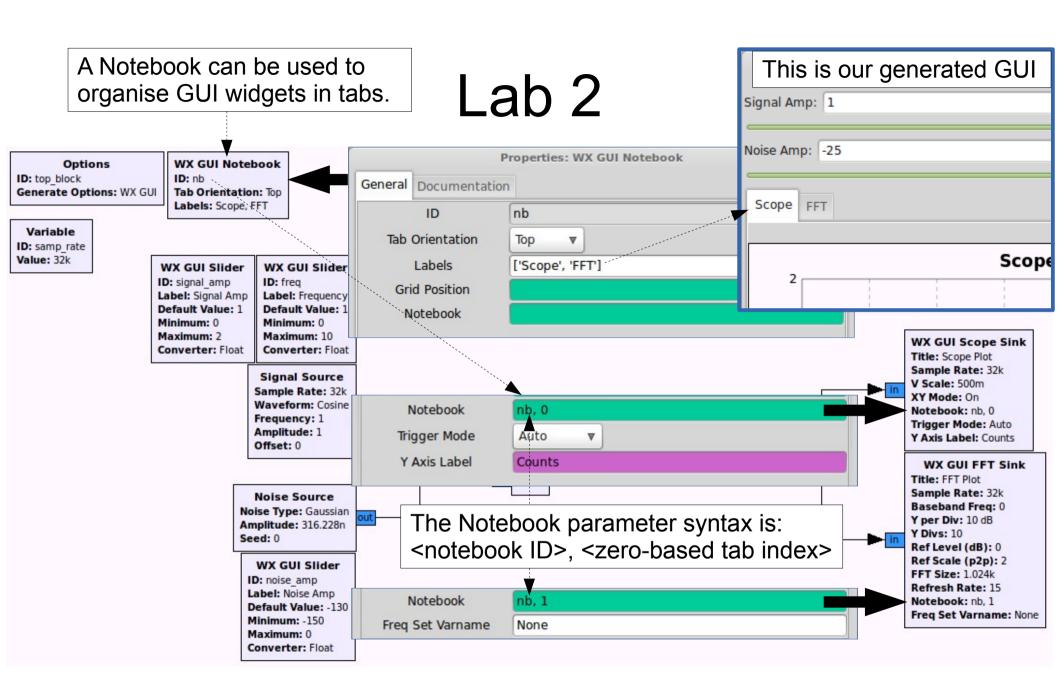


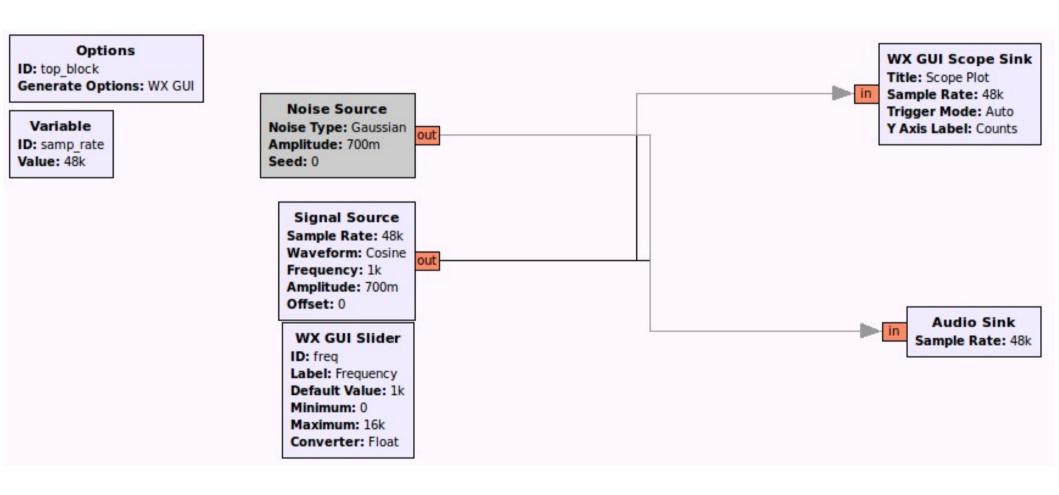




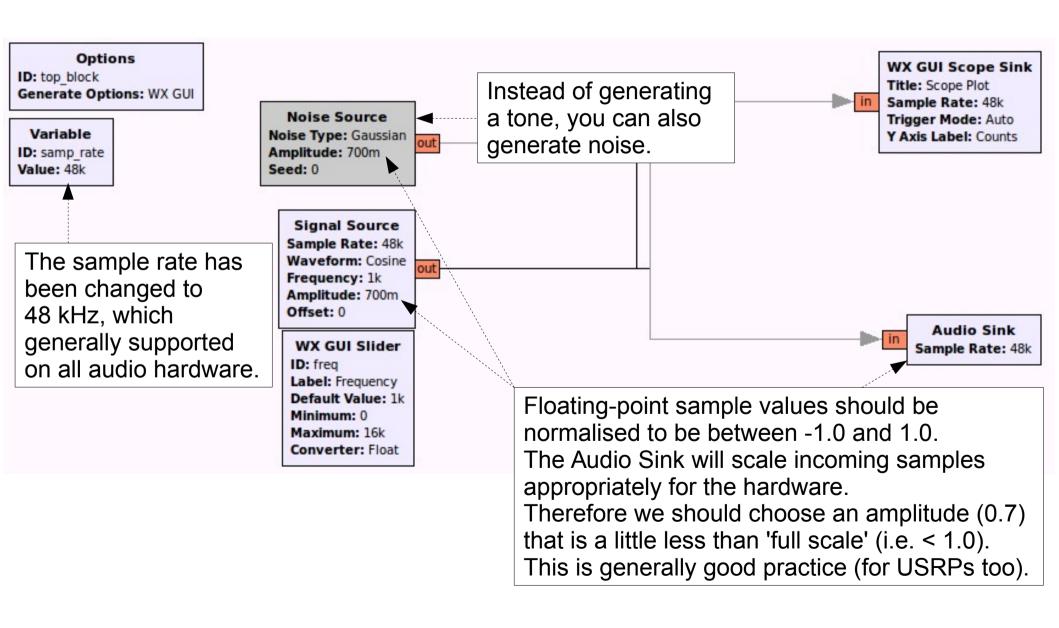


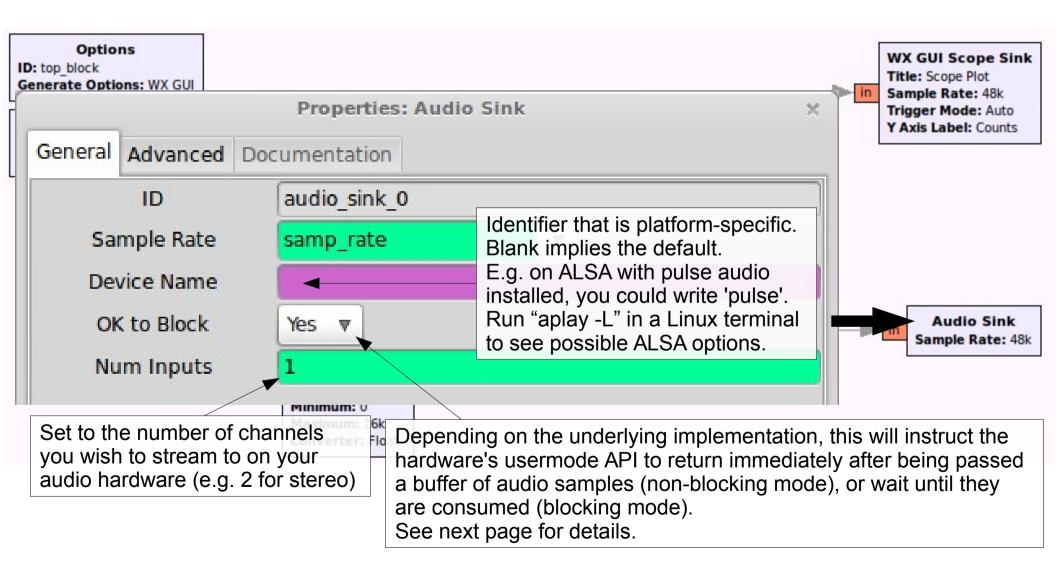






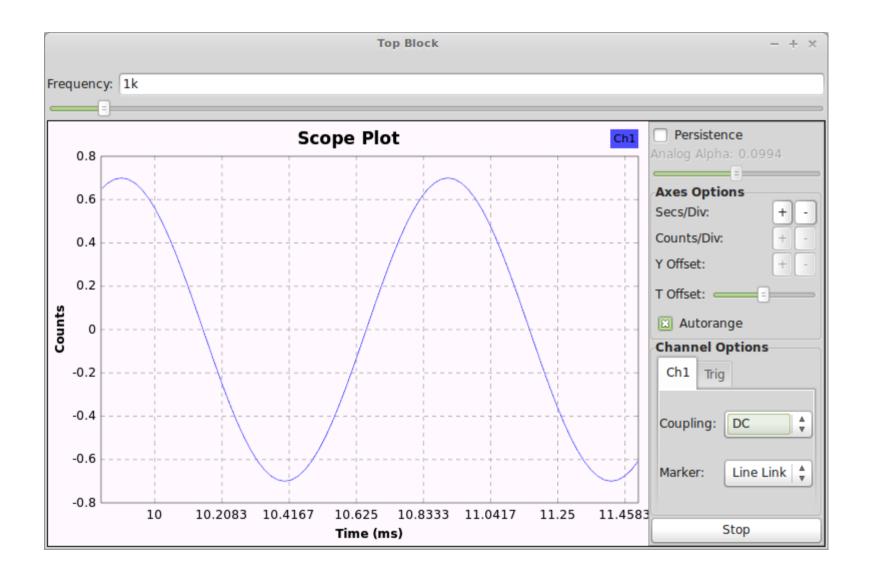
Output a single tone from the computer's soundcard



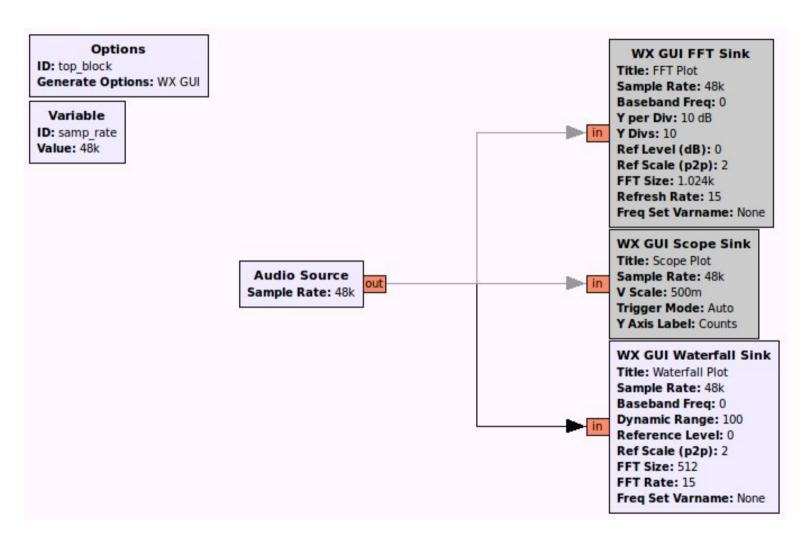


- Blocking mode ('OK to Block') will apply upstream backpressure, which is good when the Audio Sink is the only hardware device in the flowgraph.
- This can be problematic if the flowgraph source is, for example, a USRP. The source is then also hardware that has its own internal clock and will be throttling the sample production rate while the Audio Sink is throttling consumption with its own unsynchronised clock. This is called the 'two clock' problem.

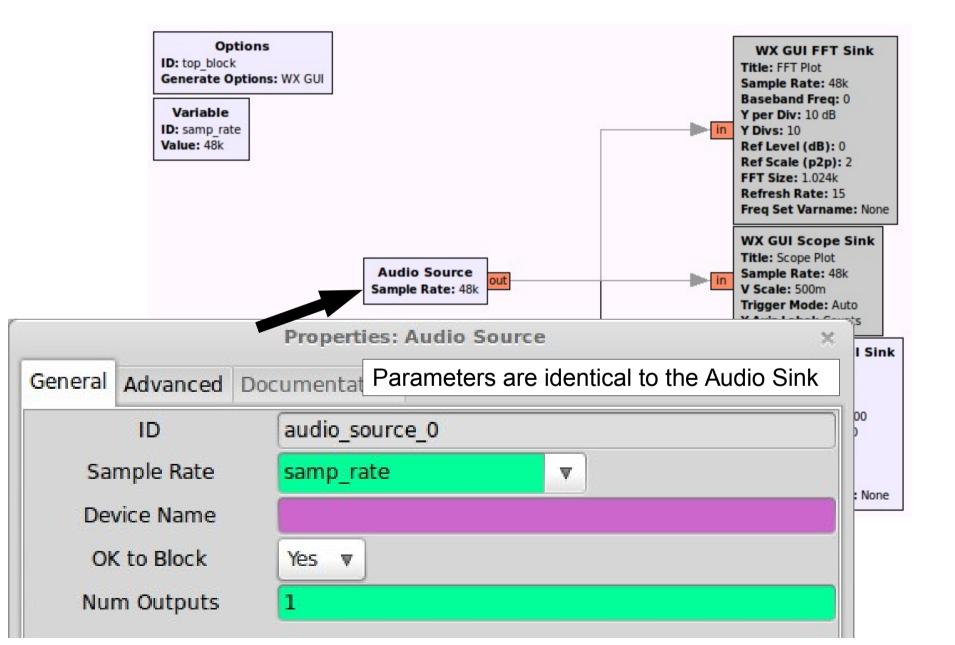
- To workaround this two clock problem, set the Audio Sink to non-blocking mode (not 'OK to Block') so that it will never hold up the flowgraph (i.e. not apply backpressure). It will consume samples as normal, but if there is ever an excess (e.g. the USRP is producing samples a little faster than the Audio Sink can consume) it will drop the samples (might cause audio glitches).
- This does not solve the case where samples are being produced *slower* than the Audio Sink's consumption rate (this will produce an underrun: audio will sound choppy and 'aU' will be printed).

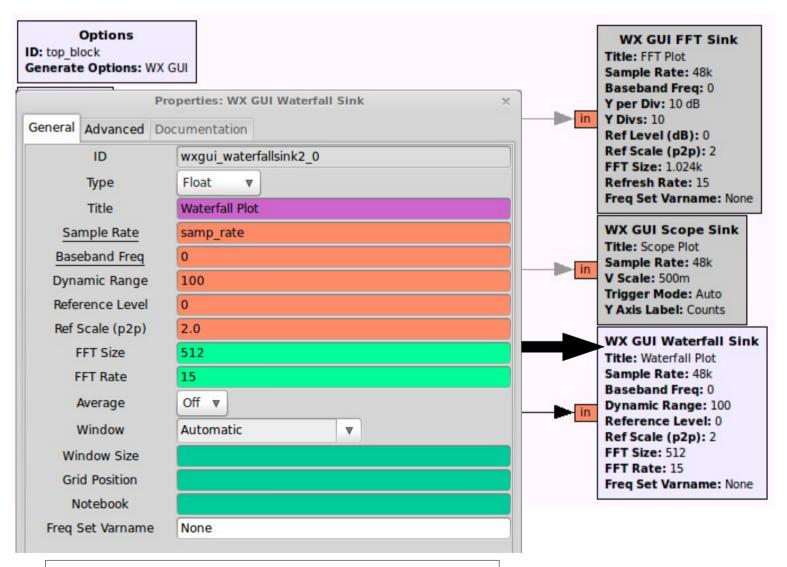


Same sine wave as before, but now we hear it emanating from the computer's speakers.

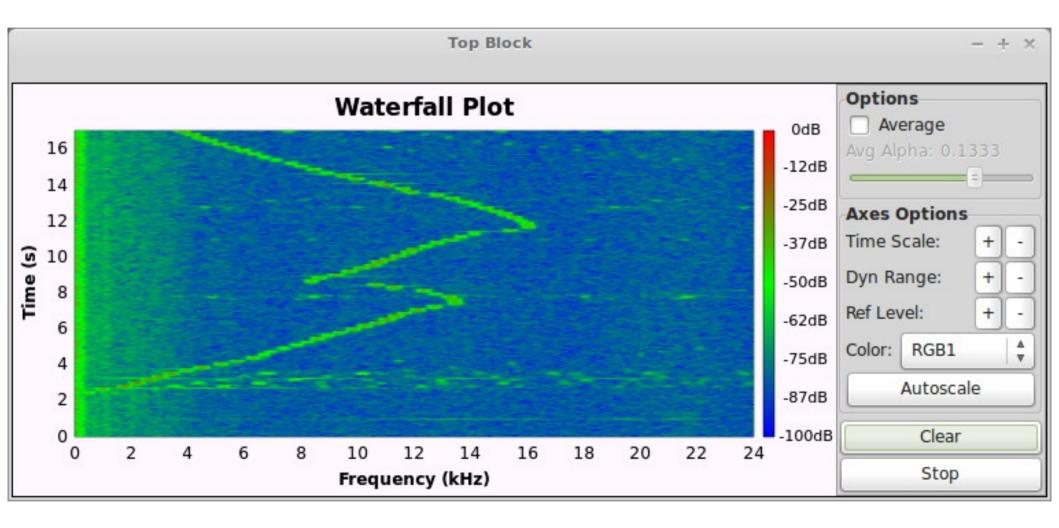


Visualise the audio sampled by a soundcard on a time-based scrolling FFT (waterfall/spectrogram).

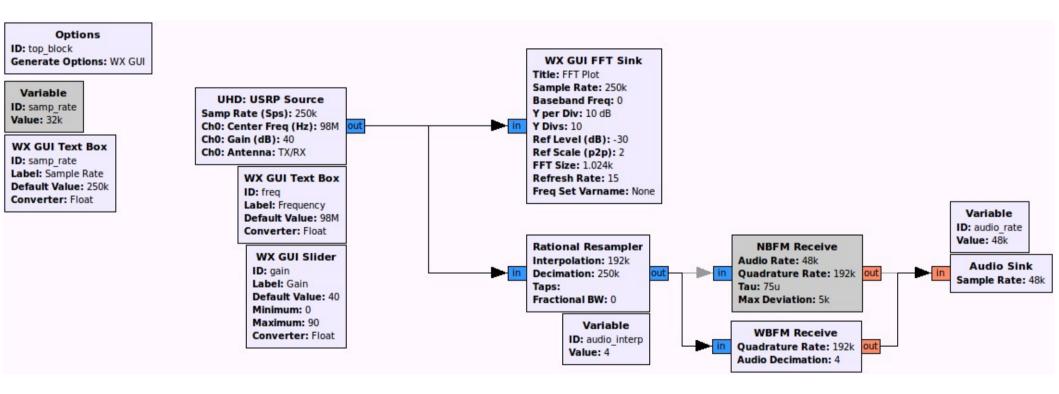




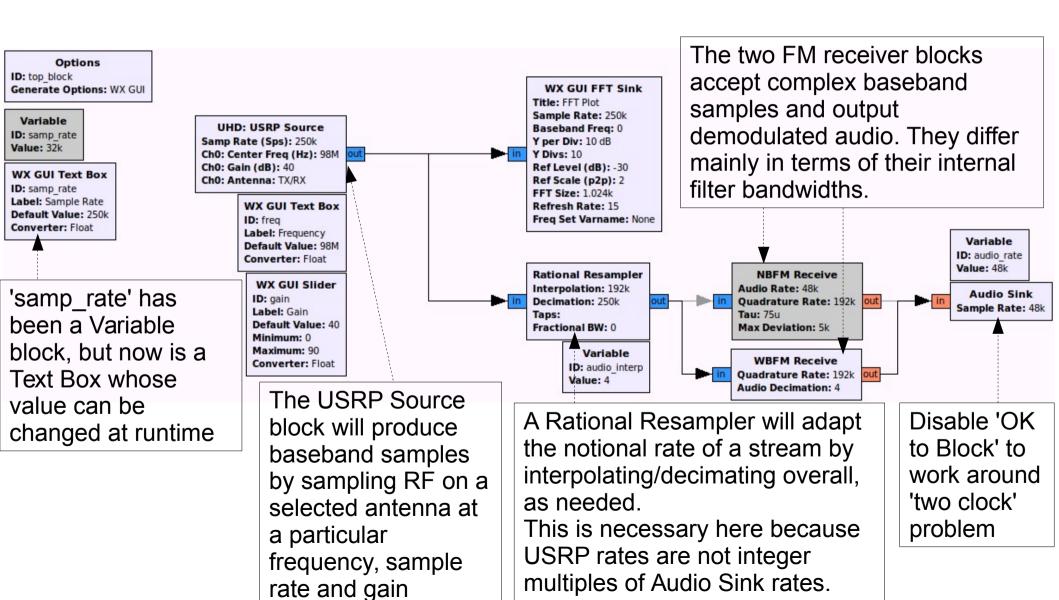
Parameters are identical to the FFT Sink



Running the sine wave generator program at the same time, and changing the frequency. This is a rough 'loopback' test where the computer's microphone listens to its speakers.

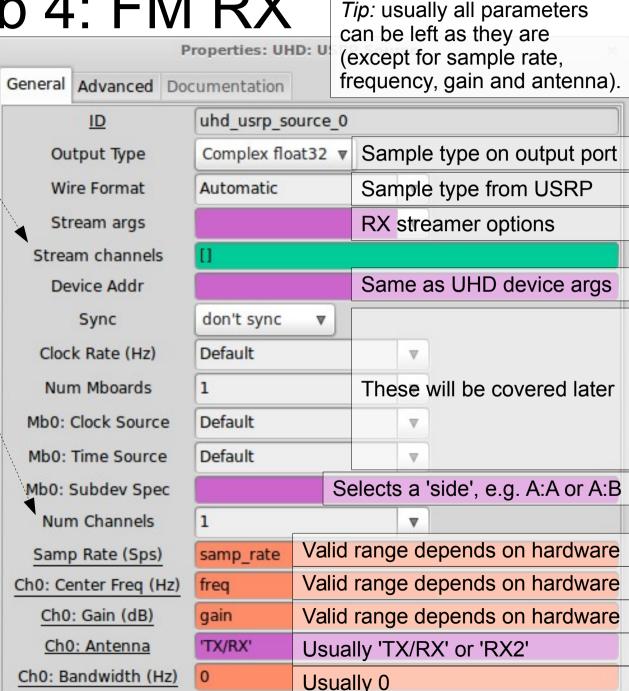


Receive a baseband signal using a USRP and listen to it using a narrow- or wide-band FM demodulator

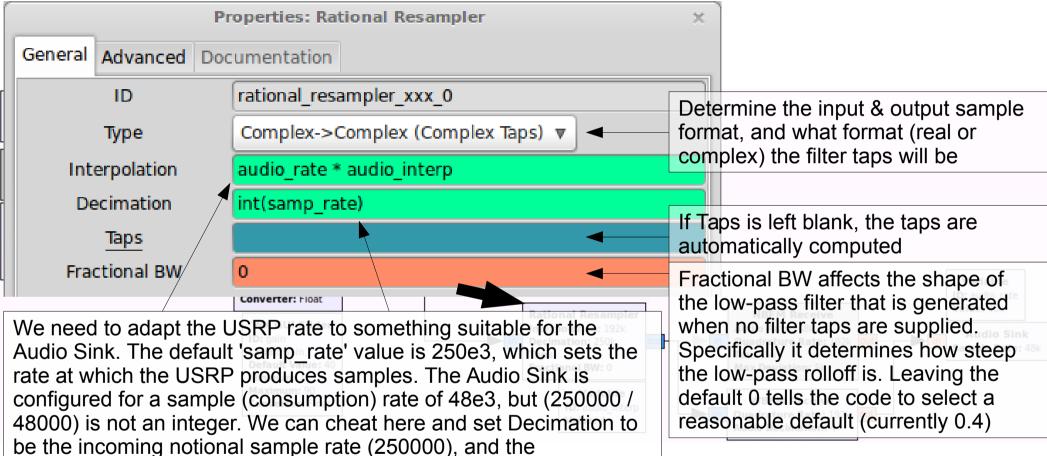


Mapping from physical (USRP) channel index to logical (GRC ID: top b port) channel index (zero-based). Leave as the empty list '[]' for the Variat default linear mapping. ID: samp Value: 32k Ch0: Center Freq (Hz): 98M Ch0: Gain (dB): 40 WX GUI Text Box ChO: Antenna: TX/RX ID: samp rate Label: Sample Rate **WX GUI Text Box** Default Value: 250k ID: freq Converter: Float Label: Frequency Default Value: 98M Sets the number of output ports and duplicates the channelspecific parameters accordingly. Maximum: 90 *Tip:* To be certain about any of the possble parameter values, consult

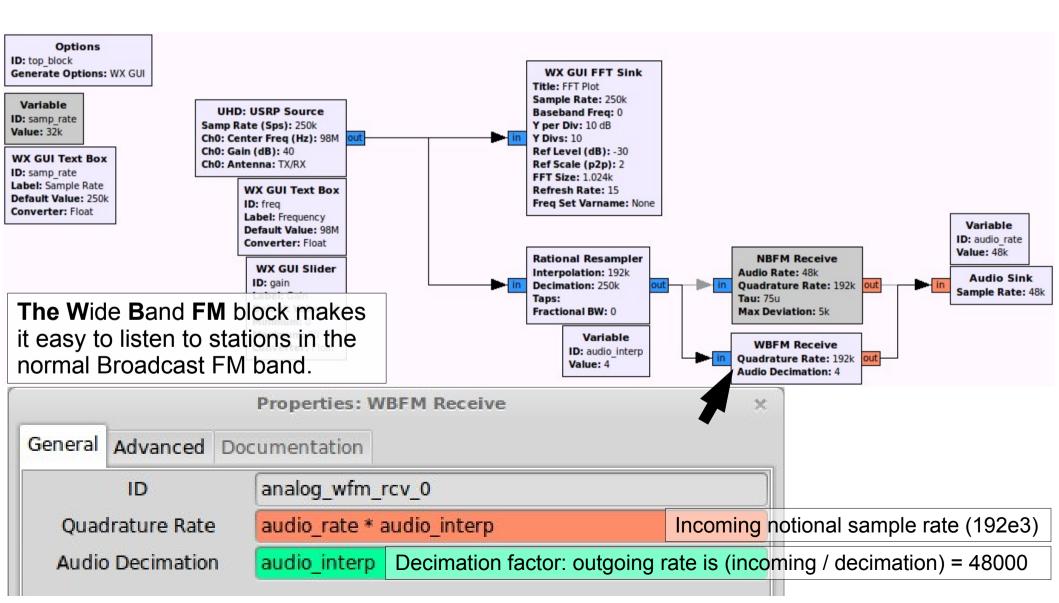
the online documentation for your device and/or daughterboard. You can also run 'uhd\_usrp\_probe' in a terminal for hardware specs. Watch your console during runtime for any warning messages from UHD regarding invalid settings!

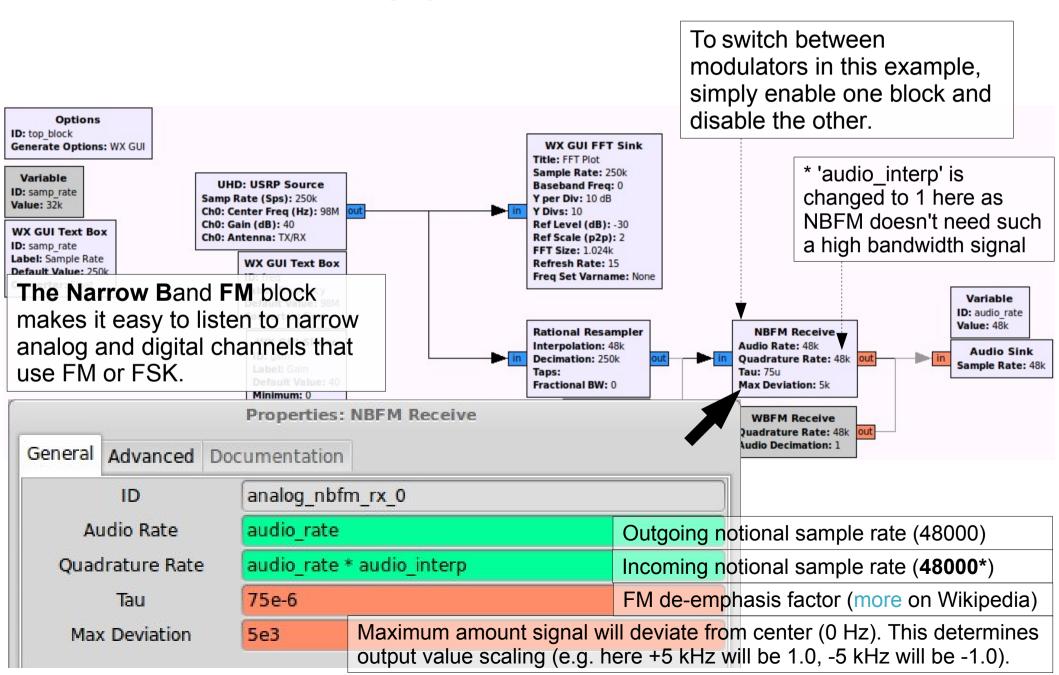


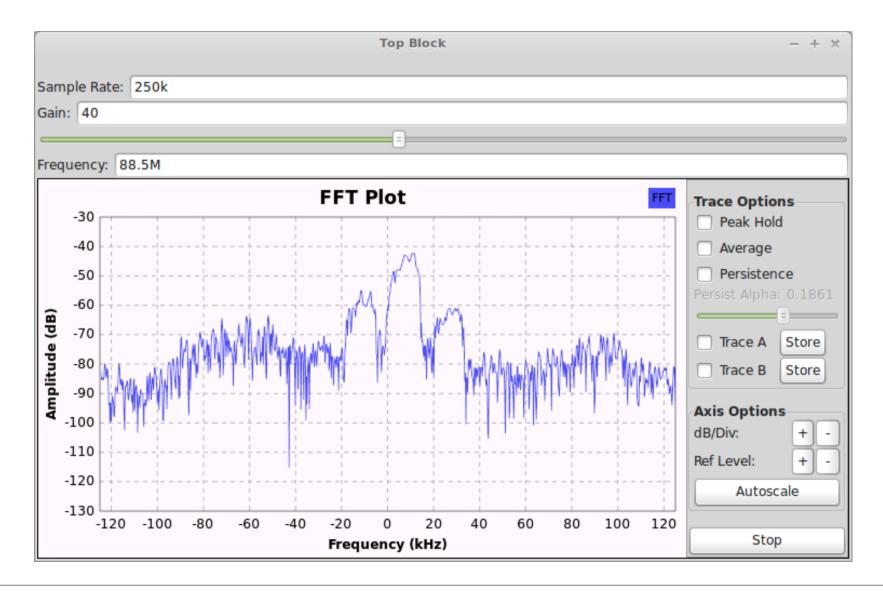
- This example uses the USRP B200
- Valid ranges:
  - Antenna: TX/RX, RX2
  - Frequency: 70 MHz 6 GHz
  - RX Gain: 0 − 73 (default of ~25 is a good starting point)
  - Sample Rate: 62.5 ksps 56 Msps (62.5e3 56e6)
    - Default **M**aster **C**lock **R**ate = 32e6 (max: 61.44e6)
    - (MCR / sample rate) must be an integer, and should be divisible by 4 for the best RF performance (flat spectrum)
    - MCR can be changed with "master\_clock\_rate=X" in Device Addr, where X is new MCR in Hz (e.g. 40e6)
- A 'O' on the console indicates an overrun, and occurs when the host is not able to consume samples quickly enough.



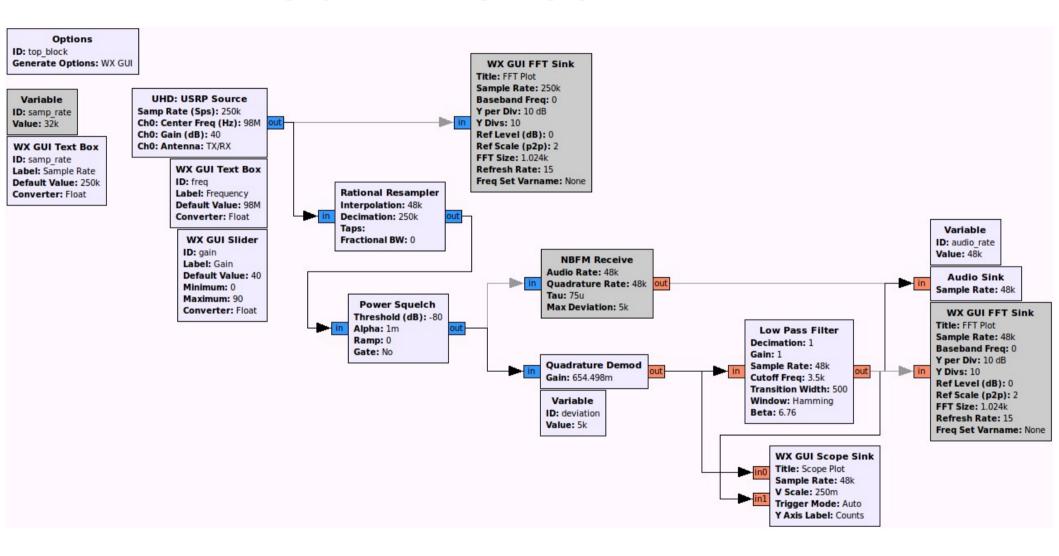
be the incoming notional sample rate (250000), and the Interpolation to be a different (non-divisible) outgoing notional sample rate (192000\*). The code will calculate the GCD. Since the parameters must be integers, and 'samp\_rate' is a floating-point number, we use the Python function 'int' to convert it to an integer. \* 'audio\_rate' is multiplied by 4 ('audio\_interp') because the demodulator blocks will perform additional decimation (by 4). Specifically the WBFM block should be given a high rate (i.e. high bandwidth signal since FM broadcast channels are 200 kHz wide).



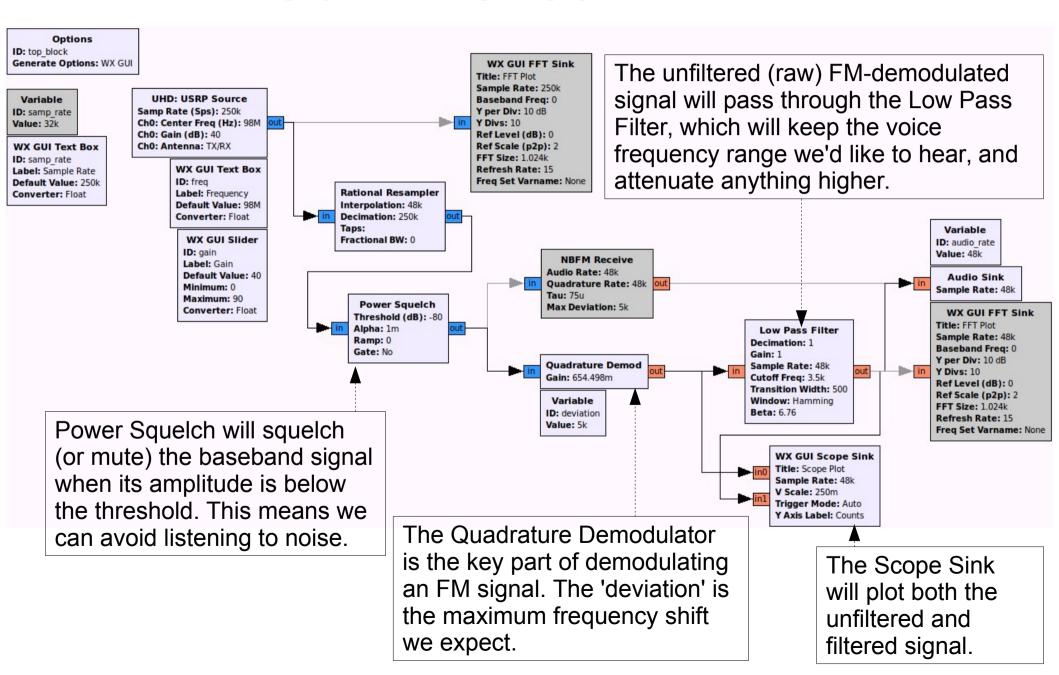


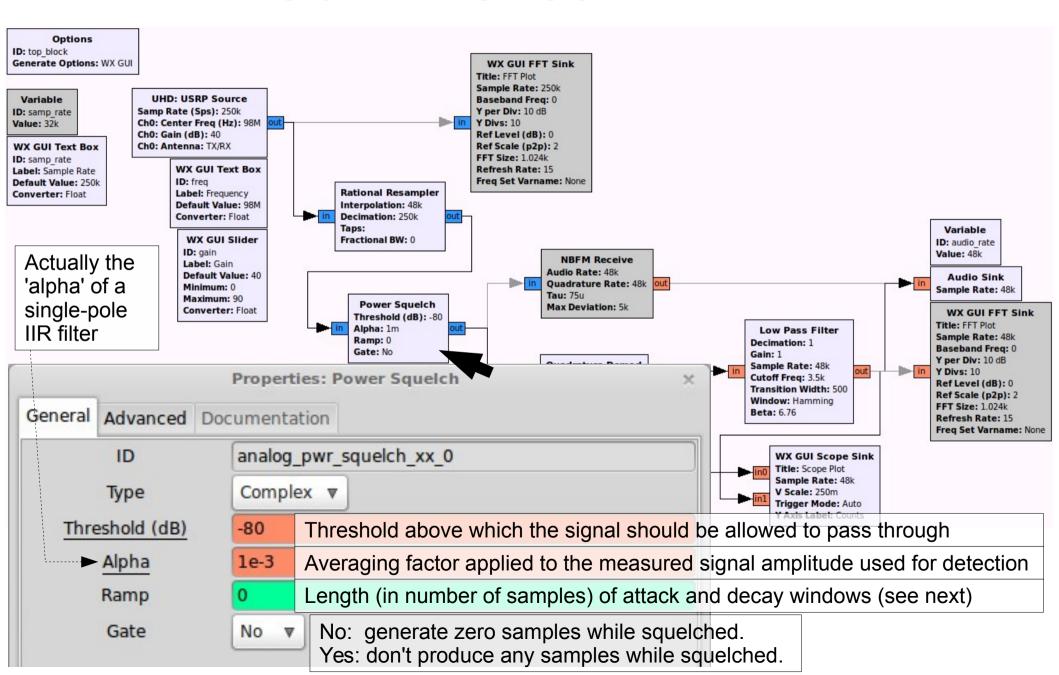


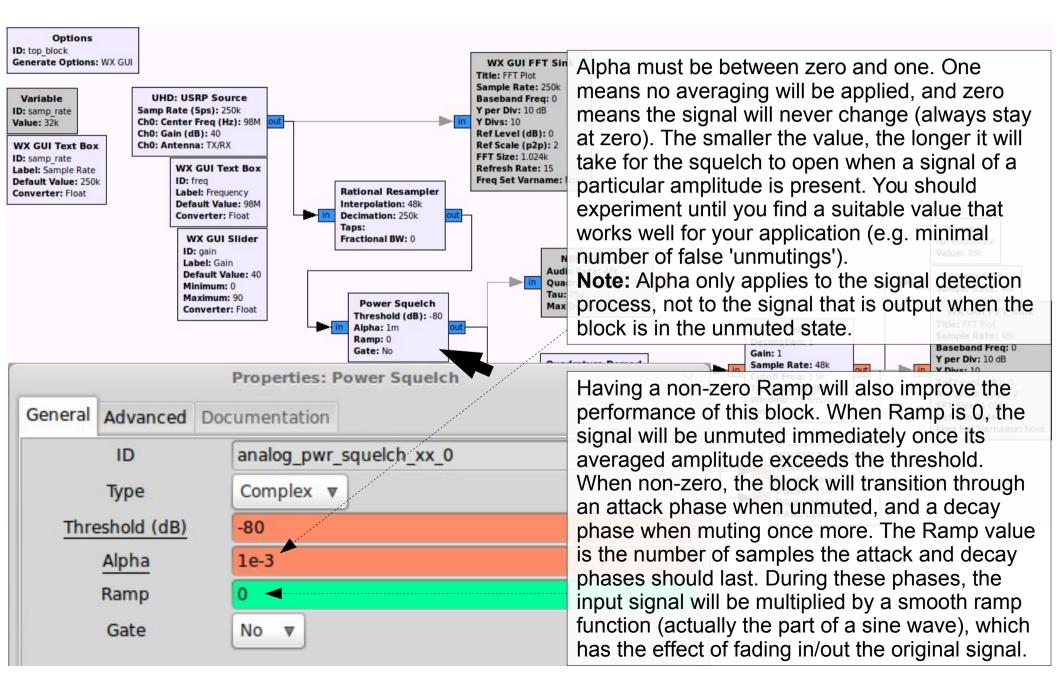
The baseband spectrum (a local radio station) in shown on the FFT plot, and the signal at the center of the spectrum is demodulated producing audio coming out of the host's soundcard.

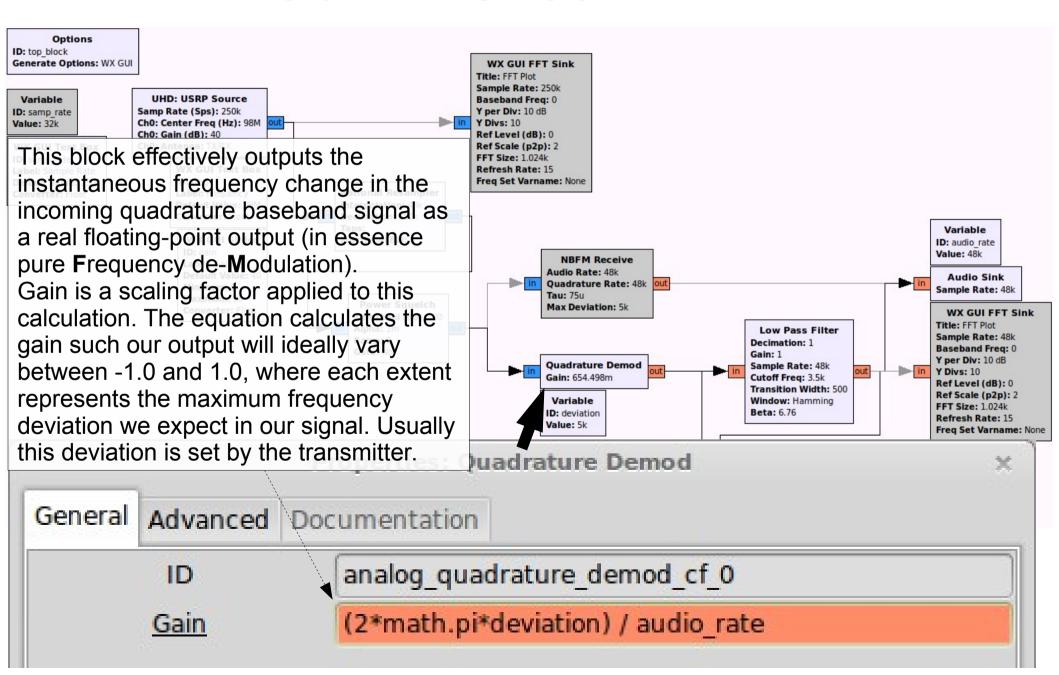


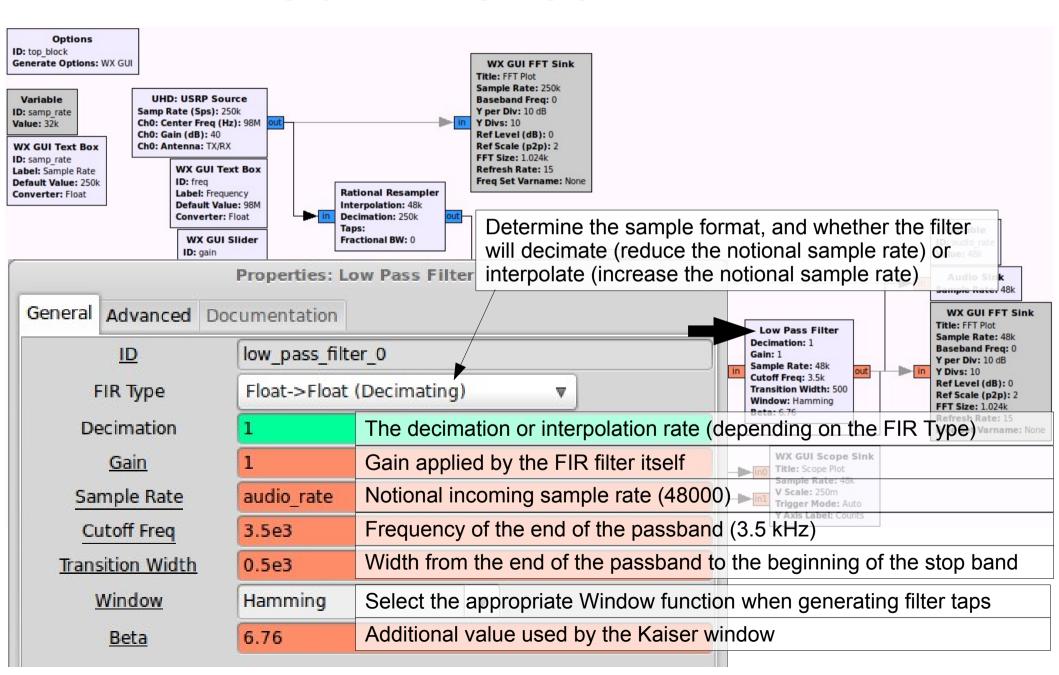
Repeat the Narrow Band FM reception example, but perform the individual demodulation steps.

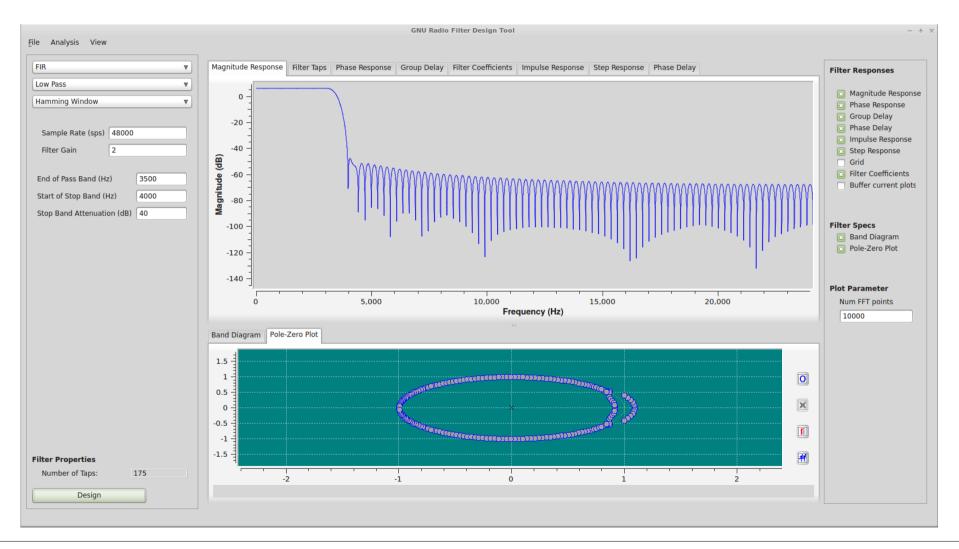












The Filter Design Tool (run 'gr\_filter\_design') is a GUI that allows you to interactively design different types of filters. Once you're happy with your design, you can place an Interpolating/Decimating FIR Filter block into your GRC flowgraph and set its taps using the filter coefficients output by the designer.



The baseband signal will be demodulated as before, however audio will only be heard if there is a strong-enough signal present at the center of the spectrum to open the squelch. The Scope Sink shows the raw demodulated signal (a whistle) in blue, and the low-pass filtered (and therefore slightly delayed) signal in green, which is output to the Audio Sink.

**GNU Radio:** 

http://gnuradio.org/

**CGRAN**:

http://cgran.org/

Ettus Research:

http://ettus.com/

**UHD Docs:** 

http://files.ettus.com/uhd\_docs/doxymanual/html/