Software Defined Radio Report

Xu Andy, Zennaro Thomas, ISS-Groupe B1

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Contents

Introduction

The objective of the labs is to focus on the reception of real communication signals. First of all, we have studied theoretically the behavior of the USRP, by using signal processing formulas, about how demodulation, signal amplitude and data transmission are made. Then, we made an analysis of the receiver. We studied a FM broadcasting record with the GNURadio development environment. Finally, we focused on the reception of VOLMET messages in AM.

1 Analysis of the SDR transceiver

1.1 Question 1

The objective was to express the $\widetilde{r}_R(t)$ and $\widetilde{r}_I(t)$ signals in function of $s_R(t)$, $s_I(t)$, f_0 and f_c:

$$r_{RF}(t) = s_{RF}(t) = s_R(t) \times \cos(2\pi f_0 t) - s_I(t) \times \sin(2\pi f_0 t)$$

$$\widetilde{r}_R(t) = r_{RF}(t) \times \cos(2\pi f_c t)$$
$$\widetilde{r}_I(t) = r_{RF}(t) \times (-\sin(2\pi f_c t))$$

$$\widetilde{r}_R(t) = [s_R(t) \times \cos(2\pi f_0 t) - s_I(t) \times \sin(2\pi f_0 t)] \times \cos(2\pi f_c t)$$

$$\widetilde{r}_R(t) = s_R(t) \times \cos(2\pi f_0 t) \times \cos(2\pi f_c t) - s_I(t) \times \sin(2\pi f_0 t) \times \cos(2\pi f_c t)$$

$$\widetilde{r}_R(t) = \tfrac{s_R(t)}{2}[\cos(2\pi(f_0 - f_c)t) + \cos(2\pi(f_0 + f_c)t)] - \tfrac{s_I(t)}{2}[\sin(2\pi(f_0 + f_c)t) + \sin(2\pi(f_0 - f_c)t)]$$

With the same way, we con obtain the expression of $\widetilde{r}_I(t)$:

$$\widetilde{r}_I(t) = \tfrac{-s_R(t)}{2}[\sin(2\pi(f_0 + f_c)t) - \sin(2\pi(f_0 - f_c)t)] + \tfrac{s_I(t)}{2}[\cos(2\pi(f_0 - f_c)t) - \cos(2\pi(f_0 + f_c)t)]$$

1.2 Question 2

fo is the carrier frequency and fc the USRP oscillator frequency. The carrier is a frequency that modulates signal. In order to recover the signal, the oscillator frequency is set to the frequency of the carrier: fc = f0. this is to bring the signal back to the base band. Using the Fourier transform, we will be able to determine the characteristics of the required filter (gain...).
If we suppose $f_0 = f_c$, we can rewrite the previous equations:

$$\widetilde{r}_R(t) = \tfrac{s_R(t)}{2}[1 + \cos(4\pi f_0 t)] - \tfrac{s_I(t)}{2}[\sin(4\pi f_0 t)]$$

$$\widetilde{r}_I(t) = \tfrac{-s_R(t)}{2}[\sin(4\pi f_0 t)] + \tfrac{s_I(t)}{2}[1 - \cos(4\pi f_0 t)]$$

The Fourier transformations are (we the variable change $F = 2f_0$):

$$\widetilde{R}_R(t) = \tfrac{S_R(f)}{2} * [\tfrac{1}{2}(\delta(f + 2f_0) + \delta(f - 2f_0)) + 1] - \tfrac{S_I(f)}{2} * [\tfrac{j}{2}(\delta(f + 2f_0) - \delta(f - 2f_0))]$$

$$\widetilde{R}_R(t) = \tfrac{1}{4}x[2S_R(f) + S_R(f + 2f_0) + S_R(f - 2f_0) - j.S_I(f + 2f_0) + j.S_I(f - 2f_0)]$$

and with the same way:

$$\widetilde{R}_I(t) = \tfrac{1}{4}x[2S_I(f) - S_I(f + 2f_0) - S_I(f - 2f_0) - j.S_R(f + 2f_0) + j.S_R(f - 2f_0)]$$

We can now deduce the filter characteristics:

- we need a gain of 2
- the cutoff frequency will be taken between $\frac{B}{2}$ and $2f_0 - \frac{B}{2}$

Furthermore, as we can see in Figure 1, we need a low-pass filter.

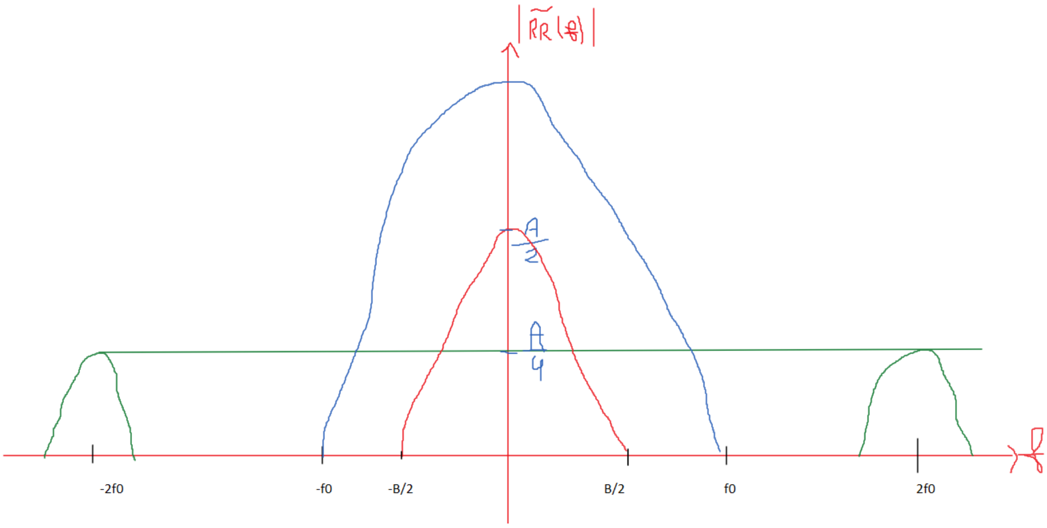


Figure 1: Frequency spectrum of the real part of the receiver signal (before filtering)

1.3 Question 3

In order to avoid a superposition between the middle band and the next band centering in $2f_0$, we need to satisfy the following inequality: $f_c < 2f_0$. Furthermore, in order to have the whole signal, we need to respect $f_c > B$. Nonetheless, on wide-band case, $f_0 < B/2$, so we have necessarily a superposition.

1.4 Question 4

If we applied the Shannon theorem:

$$F_e \geq 2f_{max} \longrightarrow F_e \geq B$$

1.5 Question 5

We do not interchange the stages of frequency transposition and analog to digital conversion because else we have to convert high frequencies (because it is before filtering) so error spreading can be increase during the conversion. Furthermore, we need more digits to convert high frequencies.

1.6 Question 6

$S_{RF}(f) = \frac{S_R(f)}{2} * [\delta(f-f_0) + \delta(f+f_0)] - \frac{S_I(f)}{2j} * [\delta(f-f_0) - \delta(f+f_0)]$

$S_{RF}(f) = \frac{1}{2} * [S_R(f-f_0) + S_R(f+f_0) + j.S_I(f)(f-f_0) - j.S_I(f+f_0)]$

Then we have the analytic signal:

$S_a(f) = S_{RF}(f) + sgn(f).S_{RF}(f) = 2.S_{RF}(f)$

$S_a(f) = [S_R(f) + j.S_I(f)] * \delta(f-f_0)$

So:

$s_a(t) = (s_R(t) + j.s_I(t)).\exp(j2\pi f_0 t)$

Then: $S(f) = S_a(f+f_0) = S_R((f+f_0)-f_0) + j.S_I((f+f_0)-f_0) = S_R(f) + j.S_I(f)$

$\longrightarrow s(t) = s_R(t) + j.s_I(t)$

2 Reception of frequency modulation

2.1 Frequency analysis of the recording

2.1.1 Question 7

The description of each block is given below:

- Variable: used to define the variable(s) used inside the processing chain
- File source: used to read row data values (of a record file) in binary format; the repeat mode implies a loop process
- Throttle: used to define the sampling rate; this blocks prevents that the limit sampling rate is not exceeded
- QT GUI Frequency Sink: used to return the analytic signal (complex envelope) of given signal, the FFT of the signal

2.1.2 Question 8

Then we specified the values of the missing parameters. You can found them below in Figure 2.

- Sample Rate: Fe
- Center Frequency: fc
- Bandwith: Fe

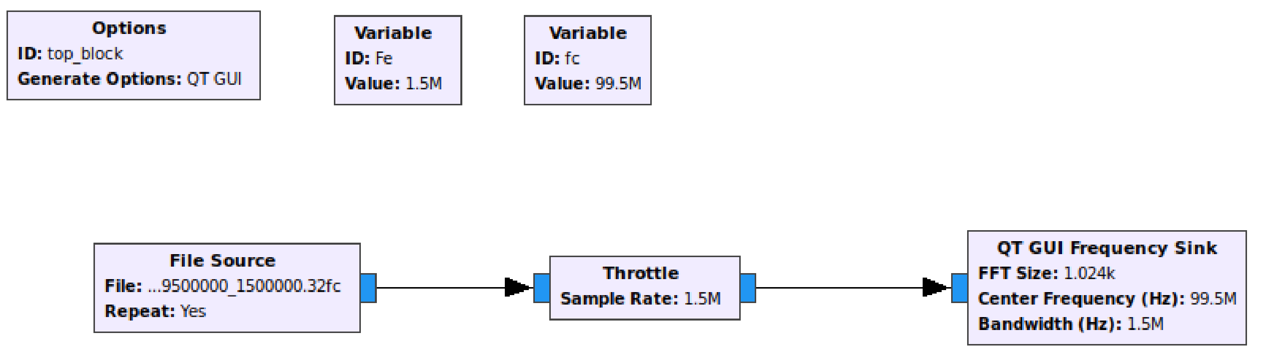


Figure 2: Block schematic of the processing chain

2.1.3 Question 9

As we can see on Figure 3, we noticed 3 channels. After checking *annuradio*, we linked each FM band with their associated radio as follow:

- 99.1MHz: RFM Toulouse
- 99.5MHz: Nostalgie
- 100MHz: Skyrock

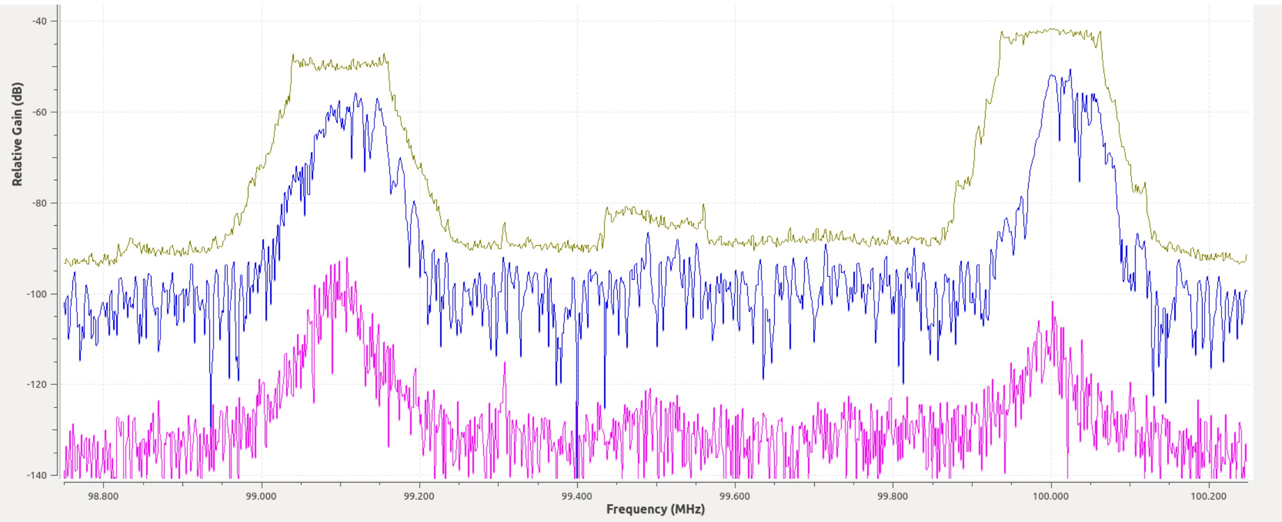


Figure 3: FFT of the output signal of the processing chain

Furthermore, we can notice that Nostalgie is near the noise threshold, so it is not easy at the beginning to identify this radio unless we do not correctly configure the control panel.

2.1.4 Question 10

We use the max hold function and calculate for each signal the signal-to-noise ratio. We choose each time the max of each signal (given by the green curve). As the values are directly in decibel, we simply have to do the difference between signal and noise values:

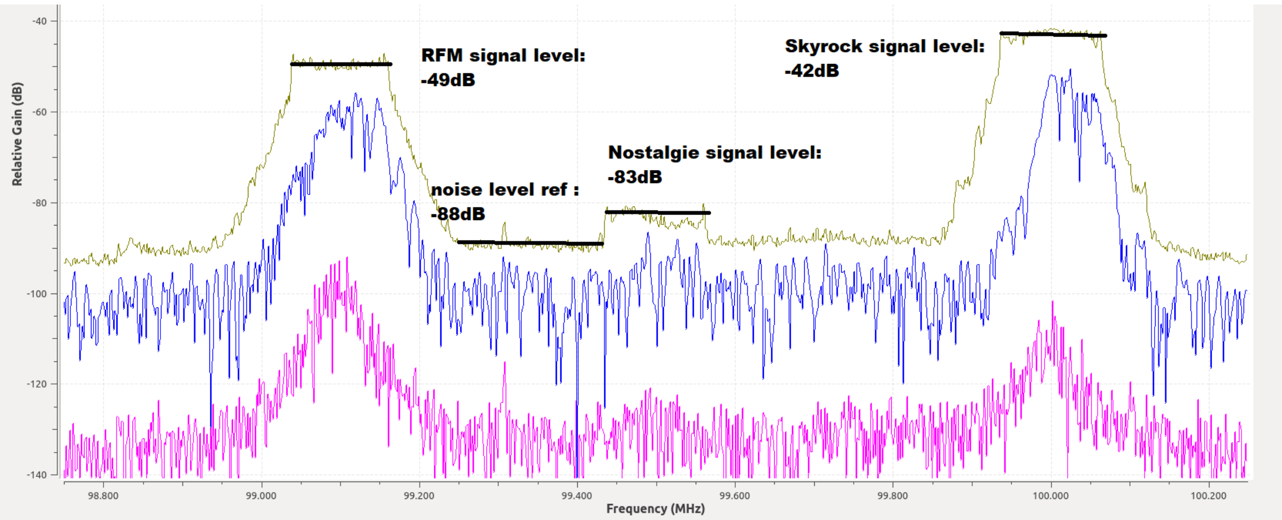


Figure 4: Thresholds for each signal and for the noise

RFM Toulouse: $|-49db-(-88db)| = 39dB$
Nostalgie: $|-83db-(-88db)| = 5dB$
Skyrock: $|-42db-(-88db)| = 46dB$

We can remark that as the ratio is important for RFM and Skyrock, there will not have any problem to demodulate. Nonetheless, it could be more difficult for Nostalgie because the ratio is low.

2.1.5 Question 11

To determine the bandwidth for each channel, we have measure the frequency bandwidth by starting at the maximum signal value until -3dB oh this maximum value.

At -3db, we have approximately 130kHz of band for each channel.

Indeed, thanks to both following figures, we have: $100.065MHz - 99.935MHz = 130kHz$.

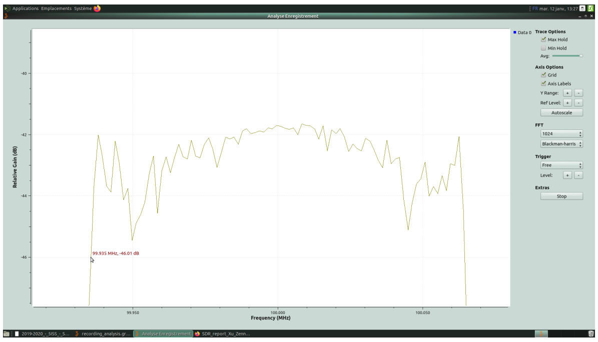


Figure 5: Determination of f_1 for Skyrock bandwidth

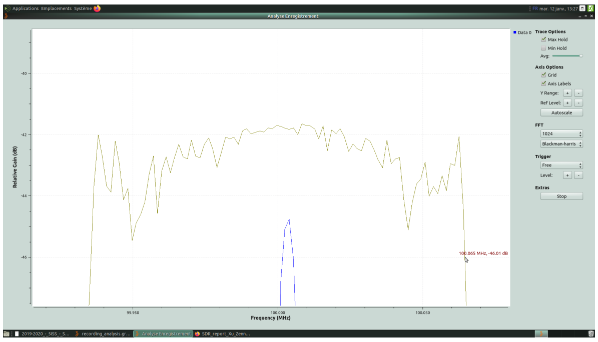


Figure 6: Determination of f_2 for Skyrock bandwidth

2.2 Channel extraction by frequency transposition and low-pass filtering

2.2.1 Question 12

The description of each block used for this new chain is given below:

- Signal source: used to the type of wanted signal; we need to define the sample rate, the waveform, the frequency and the amplitude for our case
- Qt GUI range: used to define a variable (defined between min and max terminals) that can be used in other block as input parameter
- Multiply: used to time two signals
- QT GUI Frequency Sink: returns the analytic signal (complex envelope) of given signal, the FFT of the signal

So if we take the frequency of Nostalgie as the center ($99.5MHz$) in order to sweep the spectrum to reach the other stations, we need to shift the following frequency values:

- RFM Toulouse: 400kHz
- Nostalgie: 0Hz
- Skyrock: 500kHz

2.2.2 Question 13

If the frequency offset is higher than the sampling frequency, we could observe a signal aliasing. Indeed, Shannon criteria is not respected anymore.

2.2.3 Question 14

The new block used for the filtering chain is Low pass filter. It is used to filter the unwanted high frequency defined with a frequency threshold. We need to define some parameters. Indeed, we have to configure the decimation (example: we choose a decimation of 6 which means that during the sampling of the signal we are going to take only one value over 6), the sampling frequency ($1.5MHz$), the cut-off frequency, the transition frequency and the window type for the FFT.
Taking into account the information of the subject, we define the parameters of our low-pass filter as follow:

- Gain: 1
- Cut-off frequency: $\frac{BW}{2}$ = $\frac{130}{2}$ = $65kHz$
- Transition frequency: $10\% * Gain * f_{cLP}$ = $6.5kHz$

A decimation of 6 means that $F_e \longrightarrow \frac{F_e}{6} = \frac{1.5MHz}{6} = 250kHz$
The filtering chain is given in Figure 7 and the result of filtering for Skyrock station is given in Figure 8.

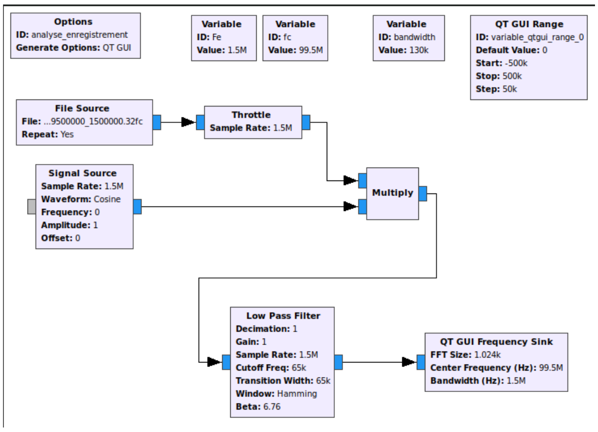


Figure 7: Filtering chain for channel extraction

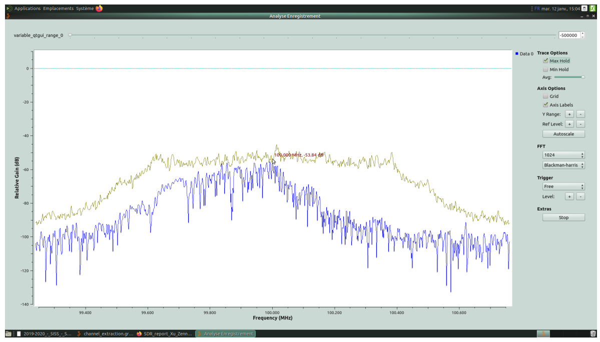


Figure 8: FFT of Skyrock signal after low-pass filtering

2.3 Frequency demodulation and restitution

2.3.1 Question 15

By applying the Carson rule, we have:

$$B_{FM} = 2 * (\Delta f + f_m) = 2 * (75 + 53) = 256kHz \tag{1}$$

So it confirms the theory.

2.3.2 Question 16

In this part we need to frequency demodulate and play audio stream by modifying the previous processing chain. For that we added some new blocks that are depicted below:

- FM demod: used to demodulate a broadcast FM signal. We need to confgure the decimation (in order to fit with previous block), the deviation (here we put $75kHz$ as suggested) and the channel rate (for audio sound card sample rate: $44.1kHz$).
- Rational resampler: changing the sample rate; the FM demodulated signal as a sample rate of $250kHz$ and the sound card sample rate is $44.1kHz$, so we need a ratio of $250/44.1 = 5.67$ which means that we need an interpolation of 10 and a decimation of 56.
- Audio sink: used to play the resulted audio.

The mathematical demonstration was given by **Philippe Herail**.

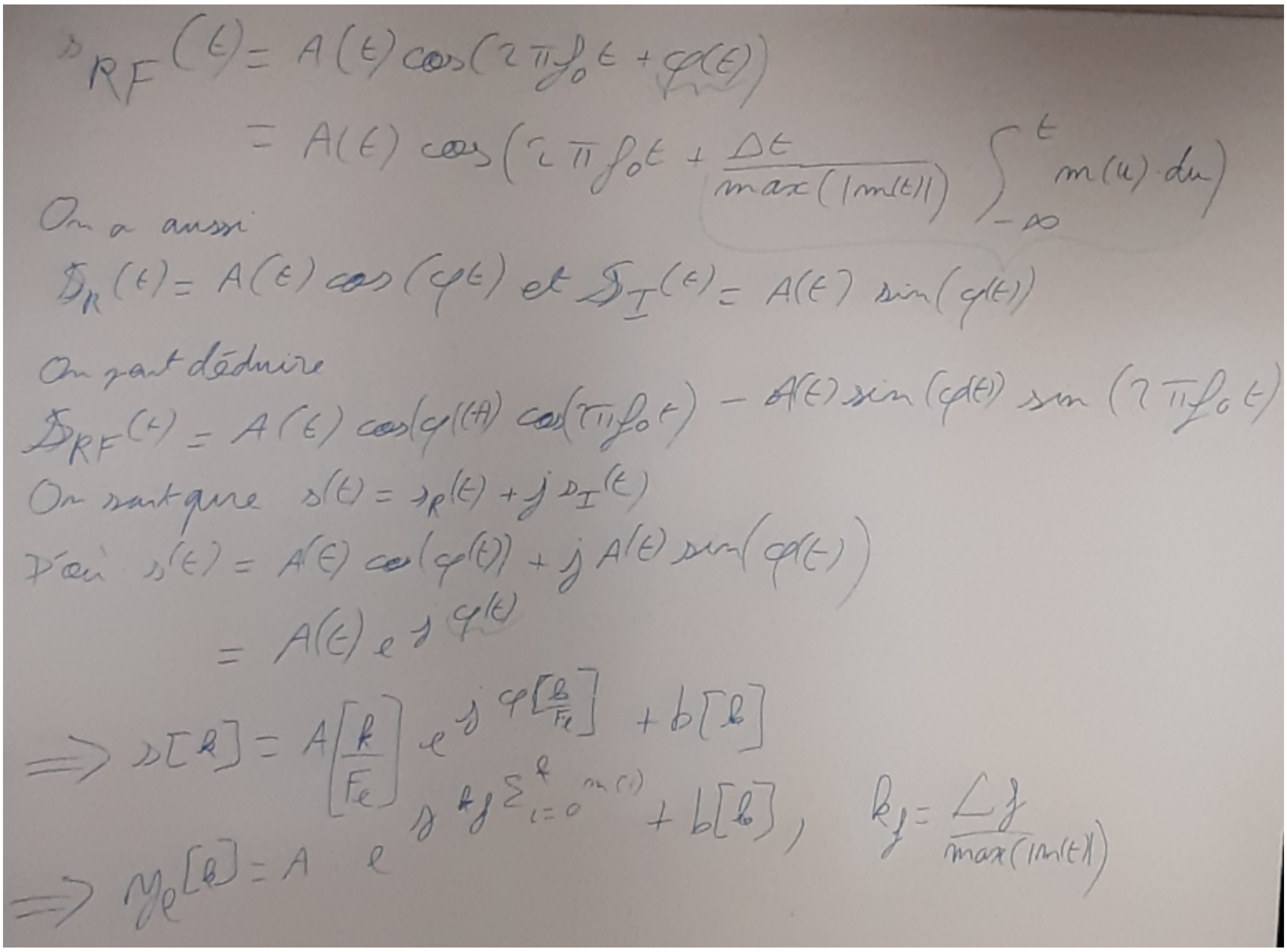


Figure 9: Demonstration

The chain process is given below:

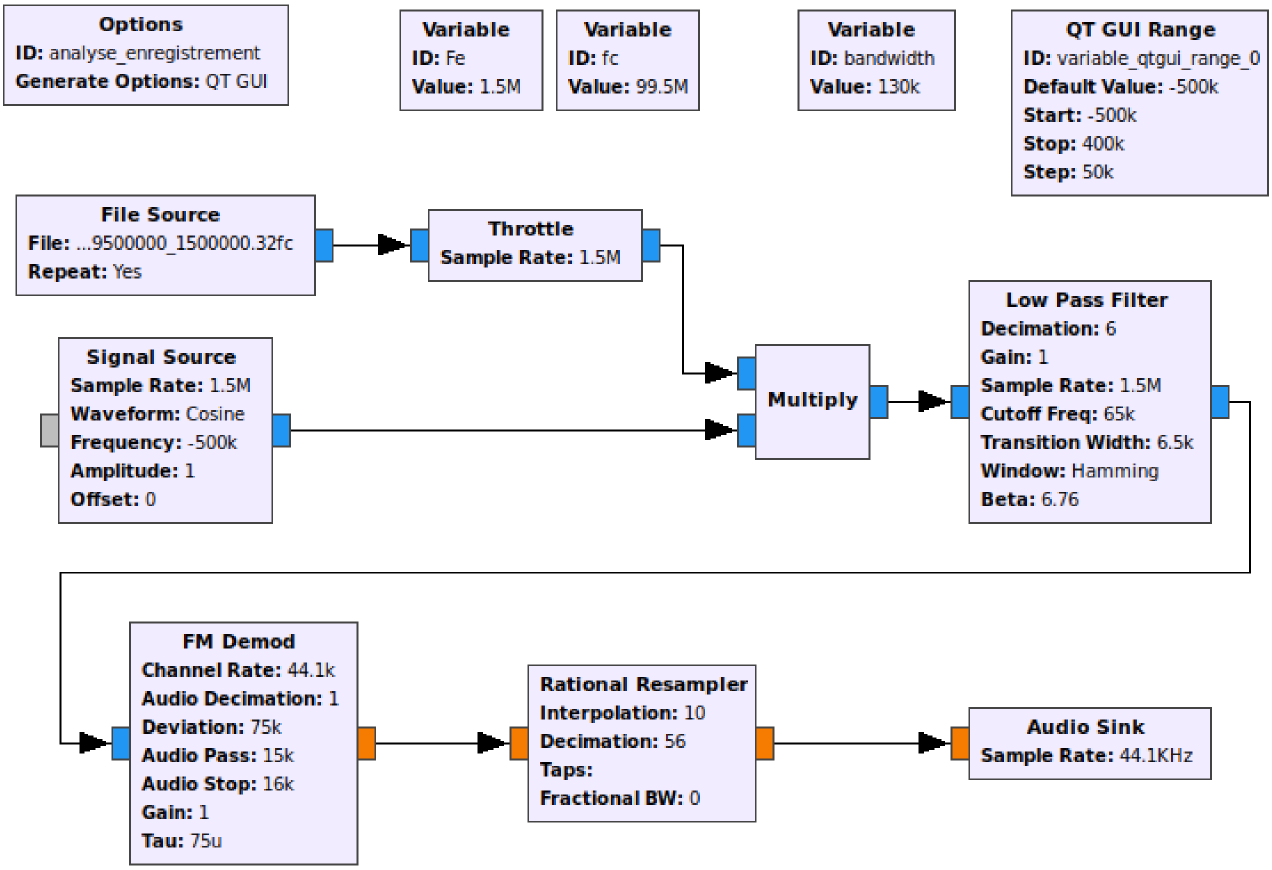


Figure 10: FM demodulation of the signal to recover the audio stream

2.3.3 Question 17

Here we find only the mono sound part which corresponds to the first block of fig. 6 on the left side.

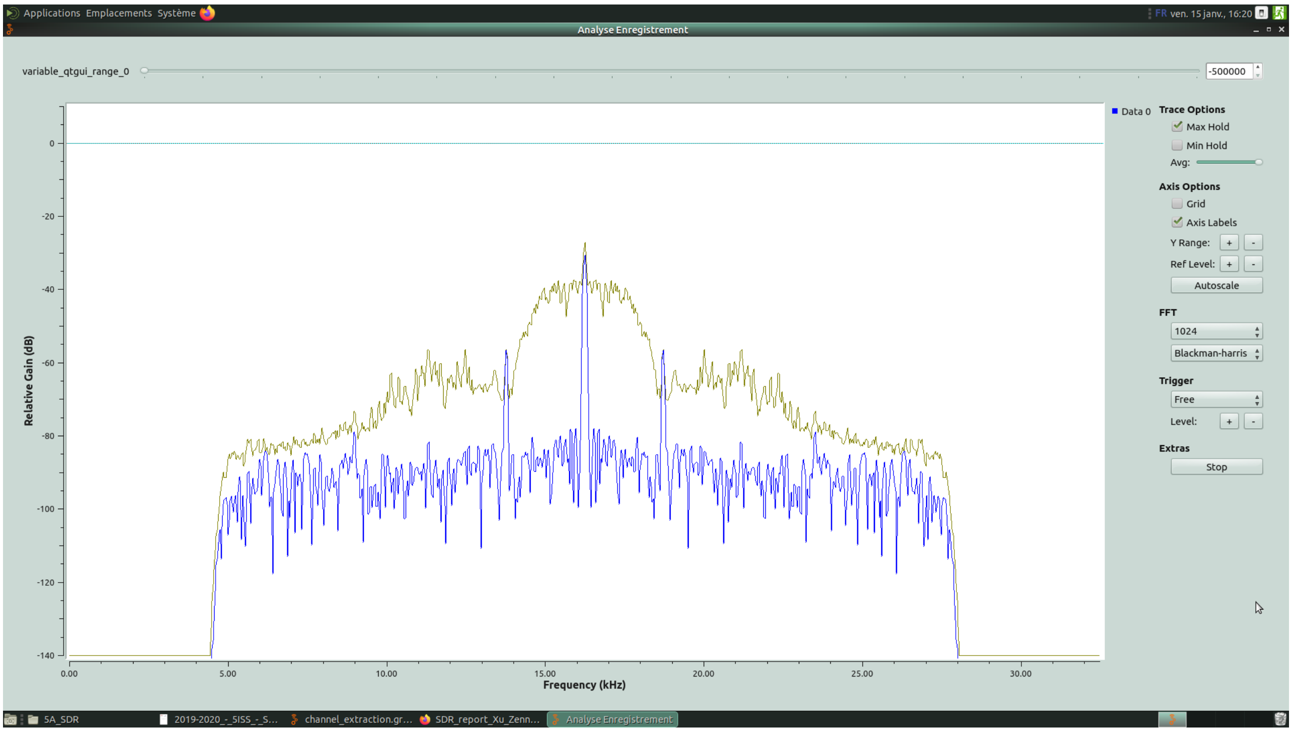


Figure 11: Mono sound demodulation

2.3.4 Question 18

Sam Smith? To listen to other stations, we do like previously and multiply our sound signal with the generated signal with the correctly chosen frequency to shift the center frequency.

2.4 Real time implementation with an USRP receiver

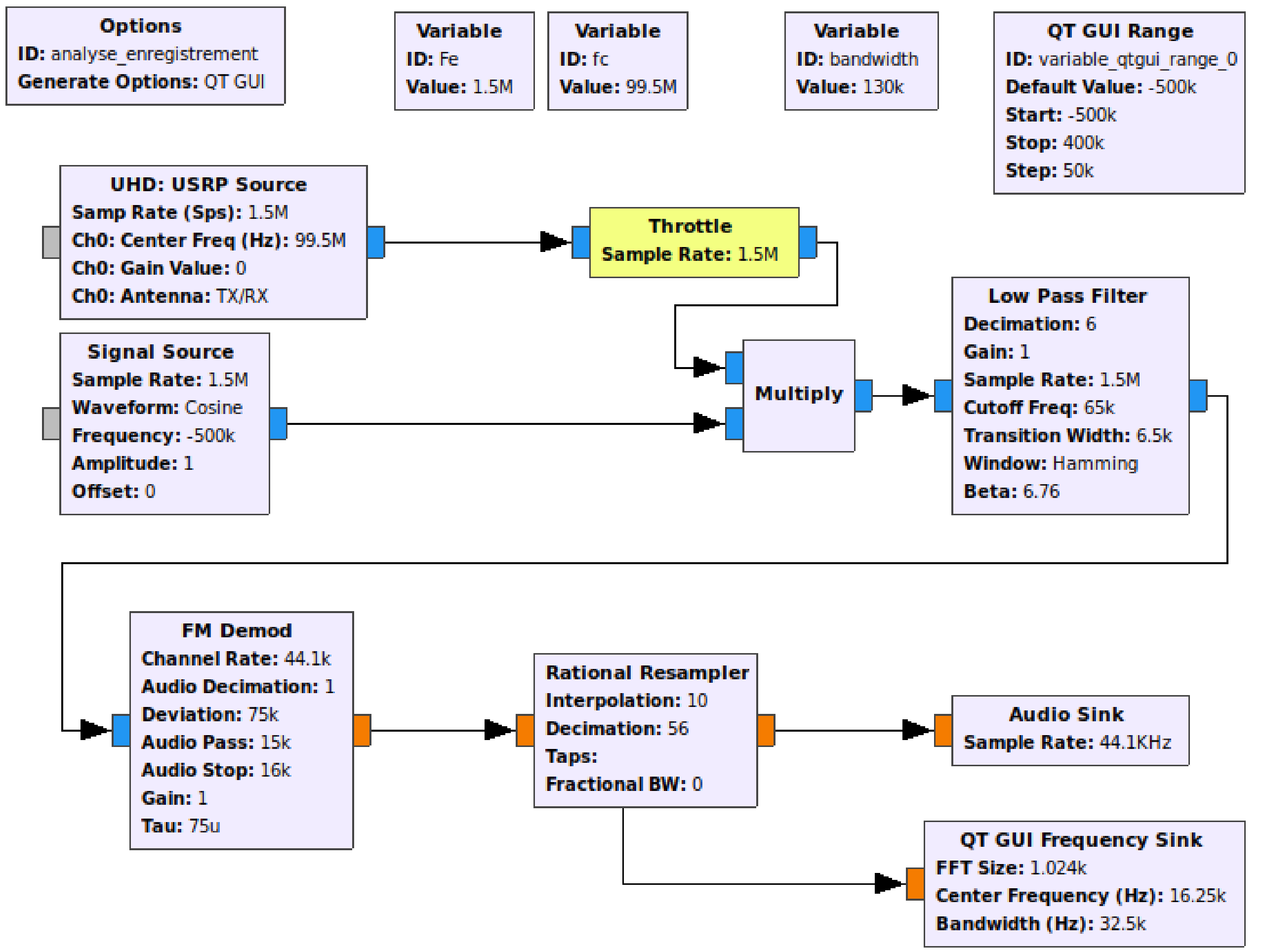


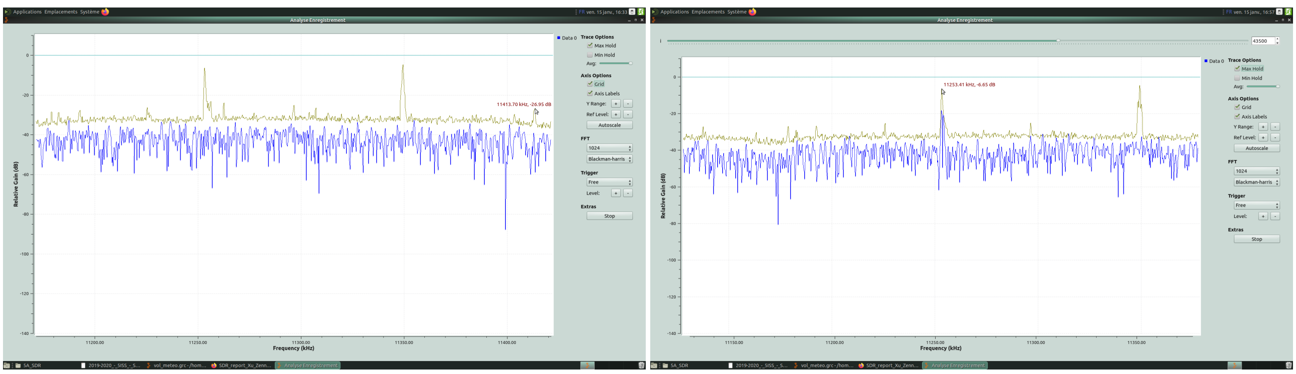
Figure 12: Processing chain using an USRP instead of a file source

3 Reception of VOLMET messages in AM-SSB

3.1 Frequency analysis of the recording

3.1.1 Question 19

We can see 2 spikes in Figure 13a and 11.253MHz is known to be RAF Volmet while 11.3xxMHz is maybe another radio emitting at that time.



(a) FFT of VOLMET signal

(b) Recentered FFT at wanted frequency

3.2 Frequency transposition

3.2.1 Question 20

We need to shift our signal by 43.5kHz to recenter our signal as it is showed in Figure 13b.

3.3 Single sideband amplitude demodulation

3.3.1 Question 21

As the spectre of a signal is symmetrical in terms of frequencies, we can use only one side of it when sending the signal over the radio to limit the bandwidth usage.

The mathematical demonstration was given by **Philippe Herail**.

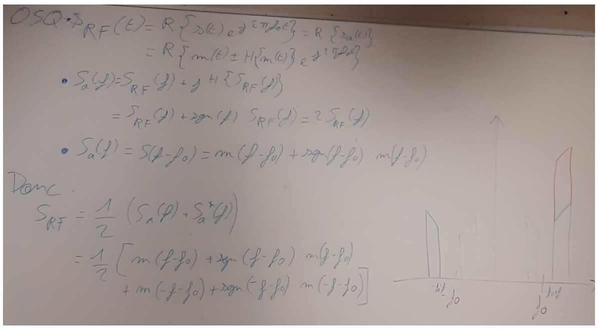


Figure 14: Demonstration

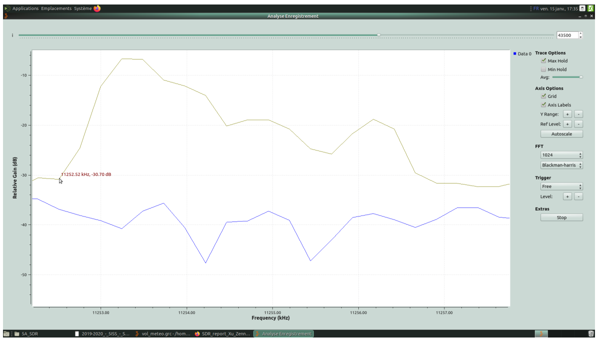


Figure 15: FFT of the signal

3.3.2 Question 22

In our case, only the upper sideband is conserved that can be deduced by the asymmetrical signal in Figure 15.

3.3.3 Question 23

Figure 16 is the design following the given criteria.

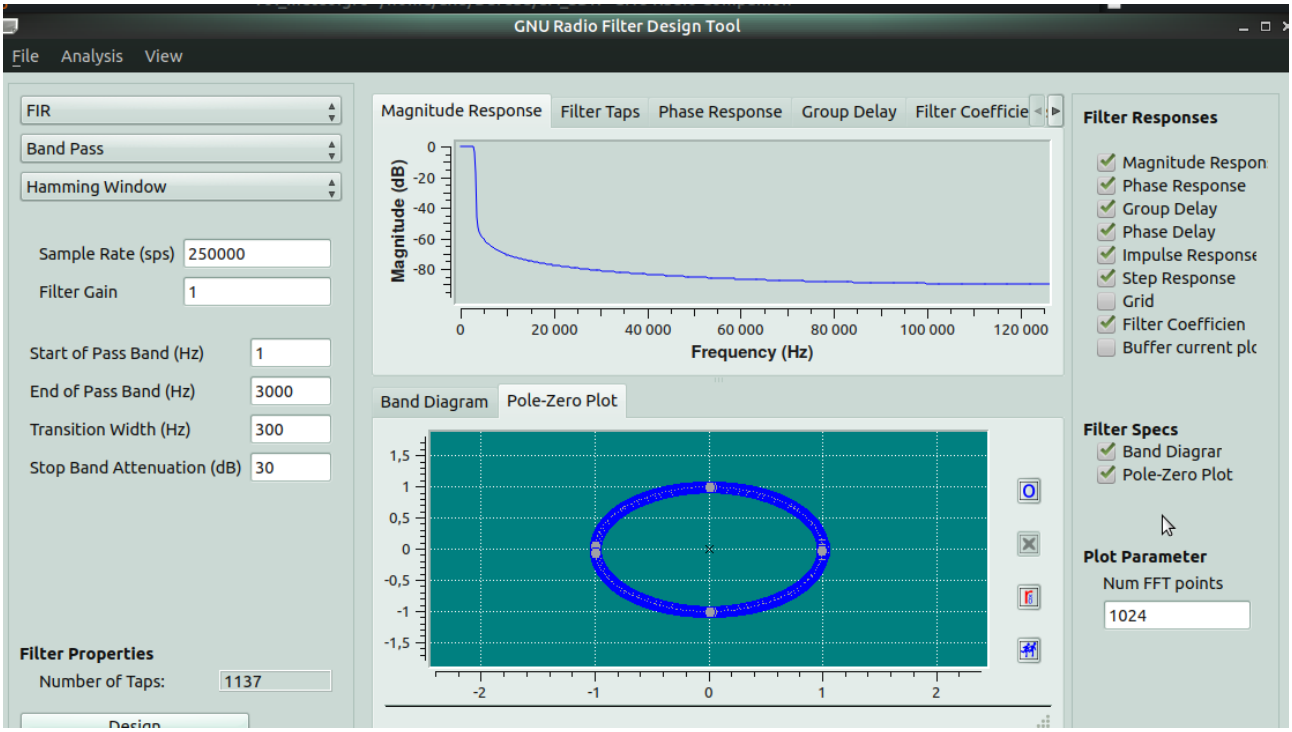
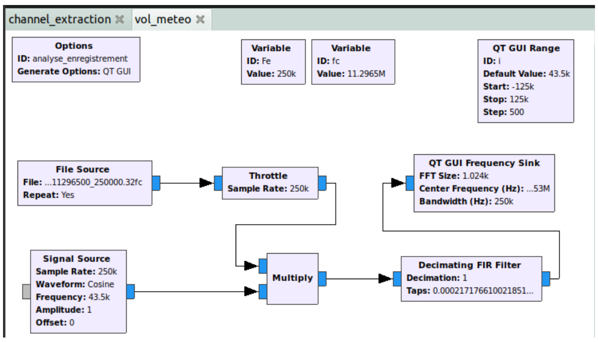


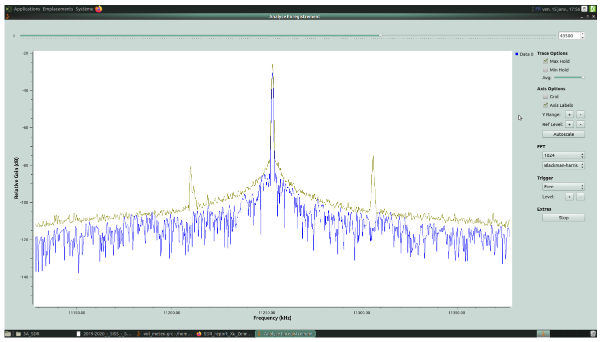
Figure 16: FIR filter design

3.3.4 Question 24

Next we apply our filter to the signal and observe that the design was correctly chosen.



(a) Filtering chain with the designed FIR filter



(b) FFT of the filtered signal

3.3.5 Question 25

Finally we listen to the demodulated signal to appreciate our work.

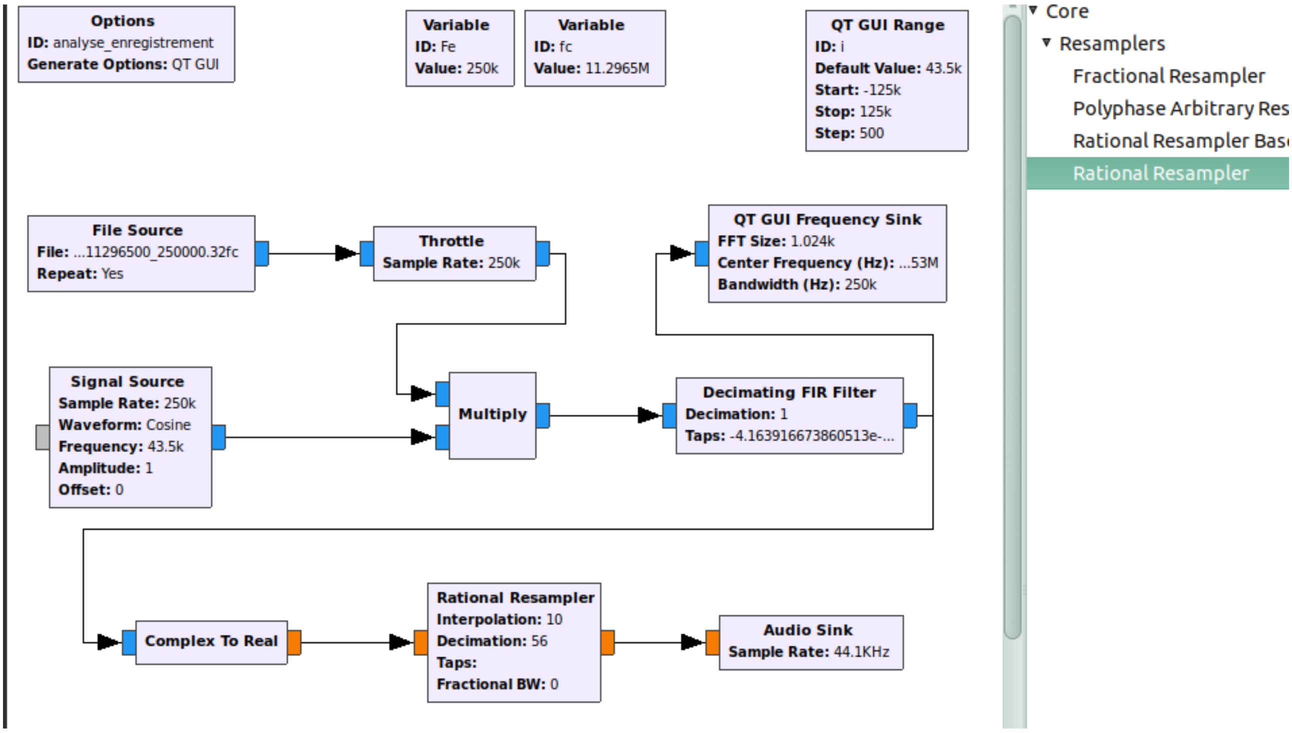


Figure 18: Volmet demodulation to recover the audio stream

Conclusion

In this lab we worked on one major application of signal treatment that is radio communication. For us that come from the Automatique et Électronique field, this lab is actually quite similar to one we did in our fourth year where we designed a phase-locked loop (PLL) as it is the principle behind FM modulation and demodulation for analog. The difference is that this lab was more focused on the mathematical principle behind and the numerical approach using computer science, thus the name software-defined radio.

List of Figures