Software Defined Radio

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5 ISS - B1

First part: Presentation of the acquisition device: In-phase/Quadrature Software-Defined Radio transceiver

q1)

$$r_R(t) = \frac{s_R(t)}{2}\left[\cos(2\pi f_0 t + 2\pi f_c t) + \cos(2\pi f_0 t - 2\pi f_c t)\right]$$

$$- \frac{s_I(t)}{2}\left[\sin(2\pi f_0 t + 2\pi f_c t) + \sin(2\pi f_0 t - 2\pi f_c t)\right]$$

$$= \frac{s_R(t)}{2}\left[\cos(2\pi(f_0+f_c)t) + \cos(2\pi(f_0-f_c)t)\right]$$

$$- \frac{s_I(t)}{2}\left[\sin(2\pi(f_0+f_c)t) + \sin(2\pi(f_0-f_c)t)\right]$$

q2)

$Q_2 \rightarrow f_c = f_0$

$$r_R(t) = \frac{s_R(t)}{2}\left[\cos(2\pi(2f_0)t) + \cos(0)\right]$$

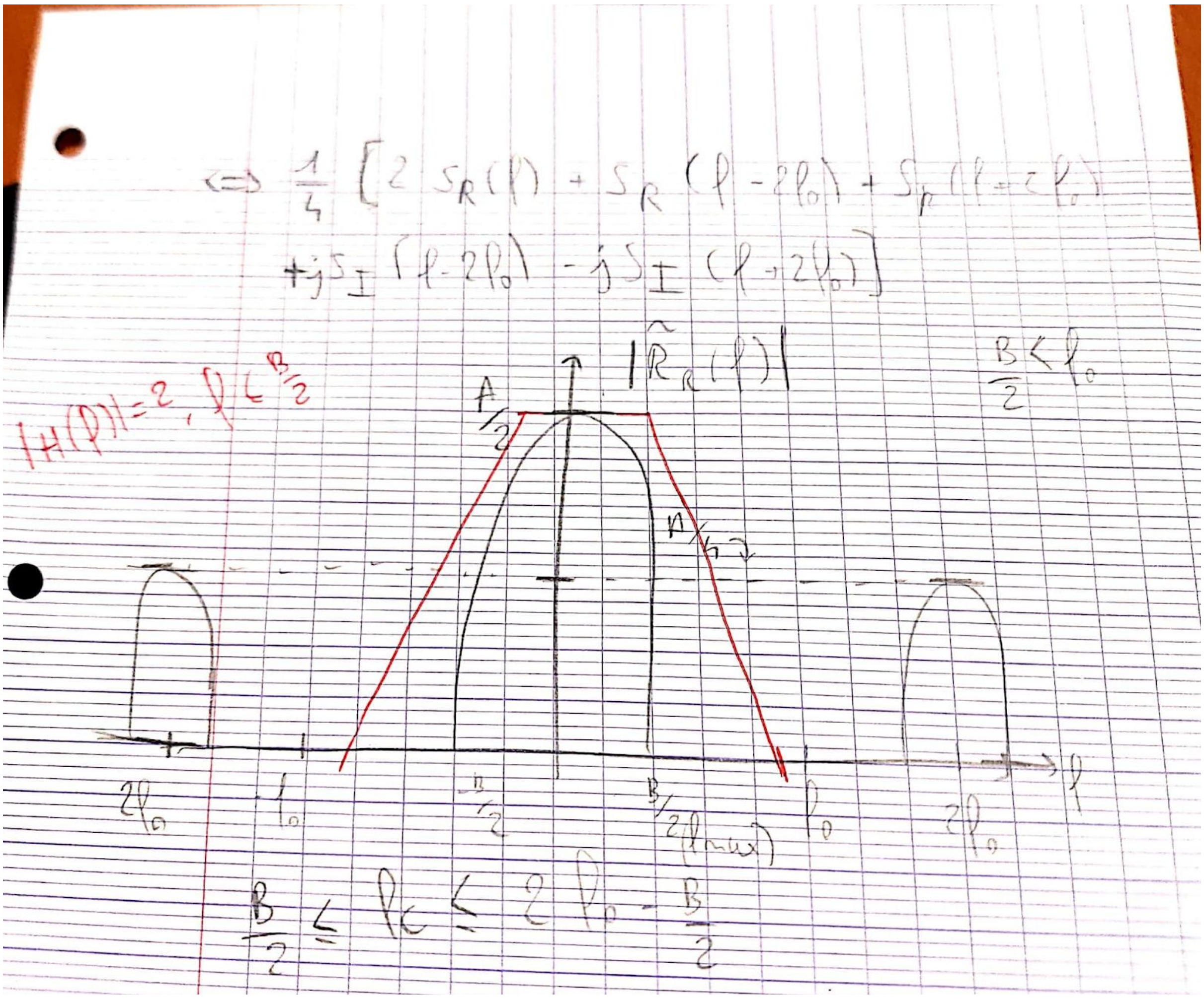
$$- \frac{s_I(t)}{2}\left[\sin(2\pi(2f_0)t) + \sin(0)\right]$$

$$= \frac{s_R(t)}{2} \cdot \left[\cos(2\pi(2f_0)t) + 1\right]$$

$$- \frac{s_I(t)}{2} \cdot \sin(2\pi(2f_0)t)$$

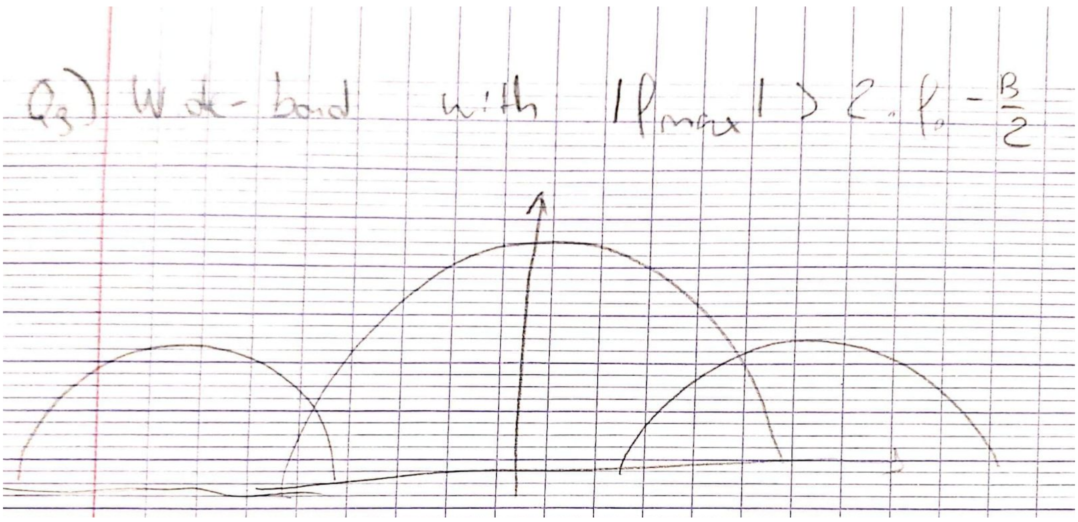
$$\Leftrightarrow R_R(f) = \frac{S_R(f)}{2} * \left[\frac{1}{2}\left(\delta(f+2f_0) + \delta(f-2f_0)\right)\right.$$

$$\left. + \delta(f)\right] - \frac{S_I(f)}{2} * \left[\frac{1}{2}\left(\delta(f-2f_0) + \delta(f+2f_0)\right)\right]$$



On veut récupérer le signal original = le signal du centre.
On elimine le 'bruit' sur le côté, car l'information est au centre. On utilisera donc un seul et même filtre.

q3)



On ne pourra pas recuperer le signal d'information du milieu.

q4)
Avec le theoreme de shannon, Fe > 2Fmax.
On prendra donc Fe = 1 / Te

q5)
Thériquement possible, cela couterais trop d'énergie et de temps, introduisant une latence innutile. On peut aussi dire que l'équipement physique necessaire pour cela couterais trop cher.

q6)

Q6) $S_{RF}(f) = \frac{S_R(f)}{2} * [\delta(f-f_0) + \delta(f+f_0)]$

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

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

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

$= S_R(f-f_0) + j (S_I(f-f_0))$

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

$S(f) = S_a(f-f_0) = S_R((f+f_0)-f_0) +$

$j S_I((f+f_0)-f_0)$

$= S_R(f) + j S_I(f)$

$\Rightarrow s(t) = s_R(t) + j s_I(t)$

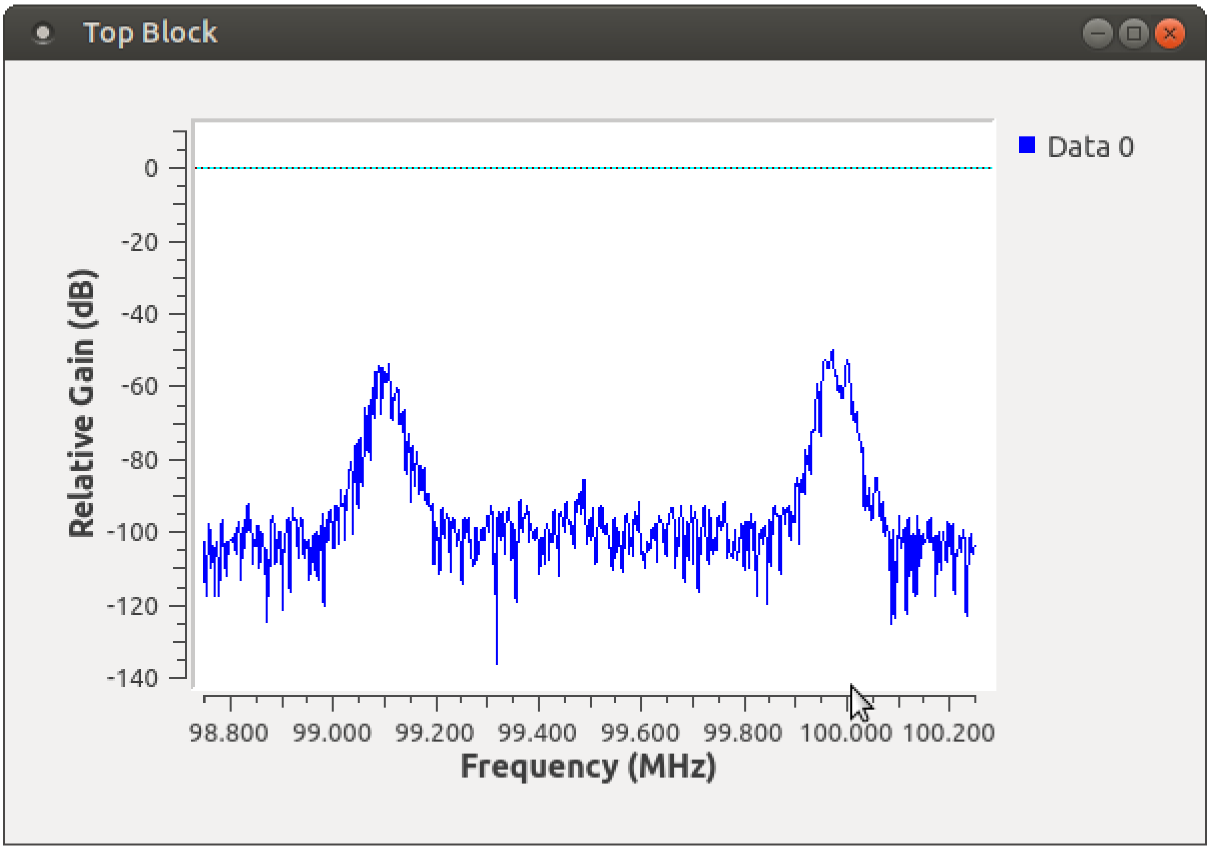
Second part : Reception of frequency modulation (FM) broadcasting

1. Frequency analysis of the recording

q7) The role of blocks are :
- **file source** : input for a file containing points of source signal
- **throttle** : to setup source signal at the right frequency
- **QT GUI freq sink** : realize real time FFT

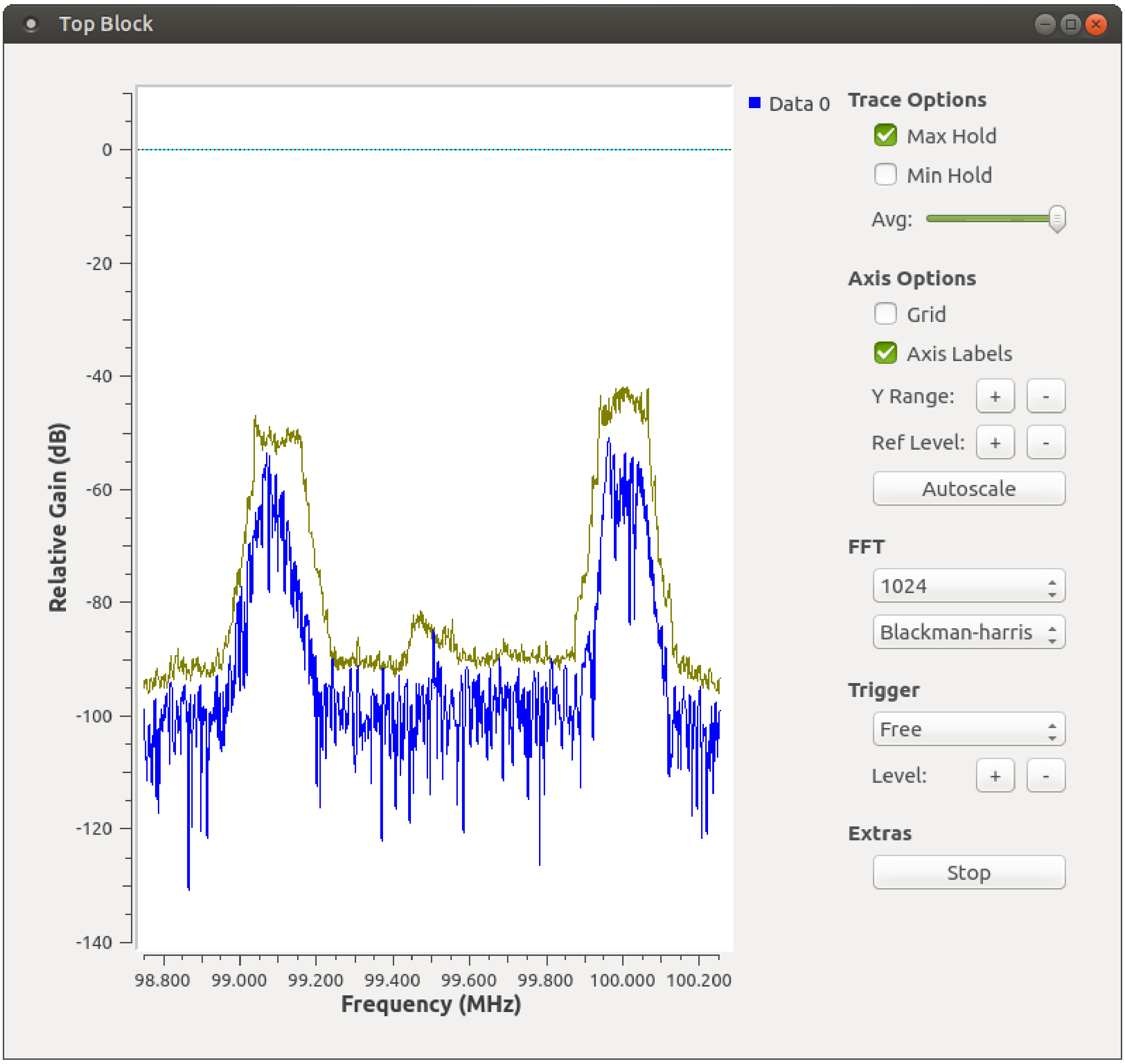
q8)
With Fe = 1.5 MHz and Fc = 99.5 MHz
Throttle - samp rate : Fe
Freq sink - center freq : Fc
Freq sink - bandwidth : Fe

q9)



We can distinguish 2 channels.
One at 99.1 MHz, RFM Toulouse, and one at 100MHz, Skyrock.
We can also distinguish a little peak at 99.5 MHz, for Nostalgie Toulouse.

q10)



Signal ~ -45dB
Noise ~ -87dB

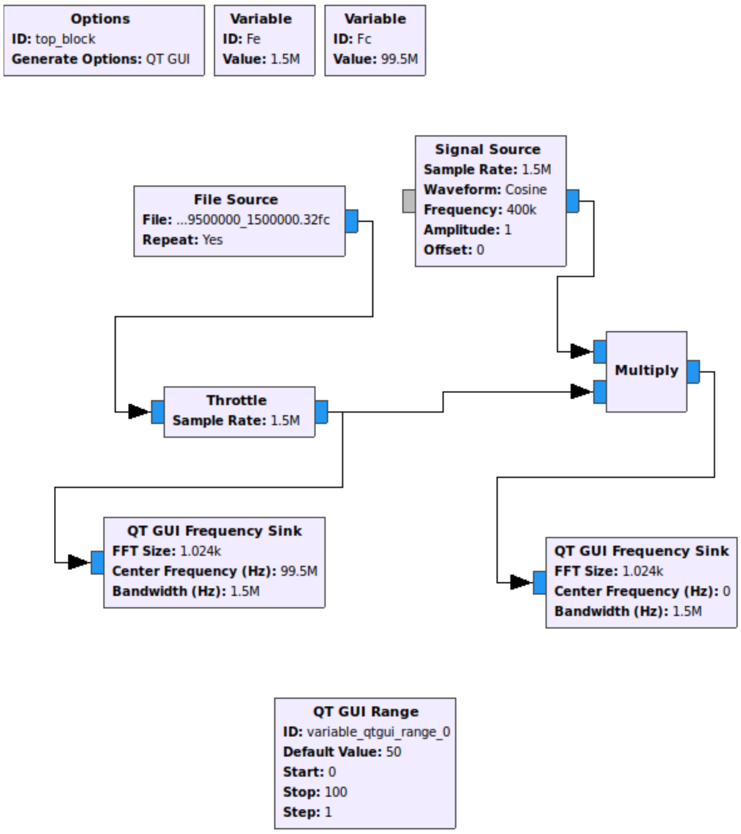
signal-to-noise ratio = 42 dB

signal-to-noise radio > 41 dB = EXCELLENT
(src : https://resources.pcb.cadence.com/blog/2020-what-is-signal-to-noise-ratio-and-how-to-calculate-it)

It will be enough to demodulate.

q11)
Bandwidth is ~250/300Khz

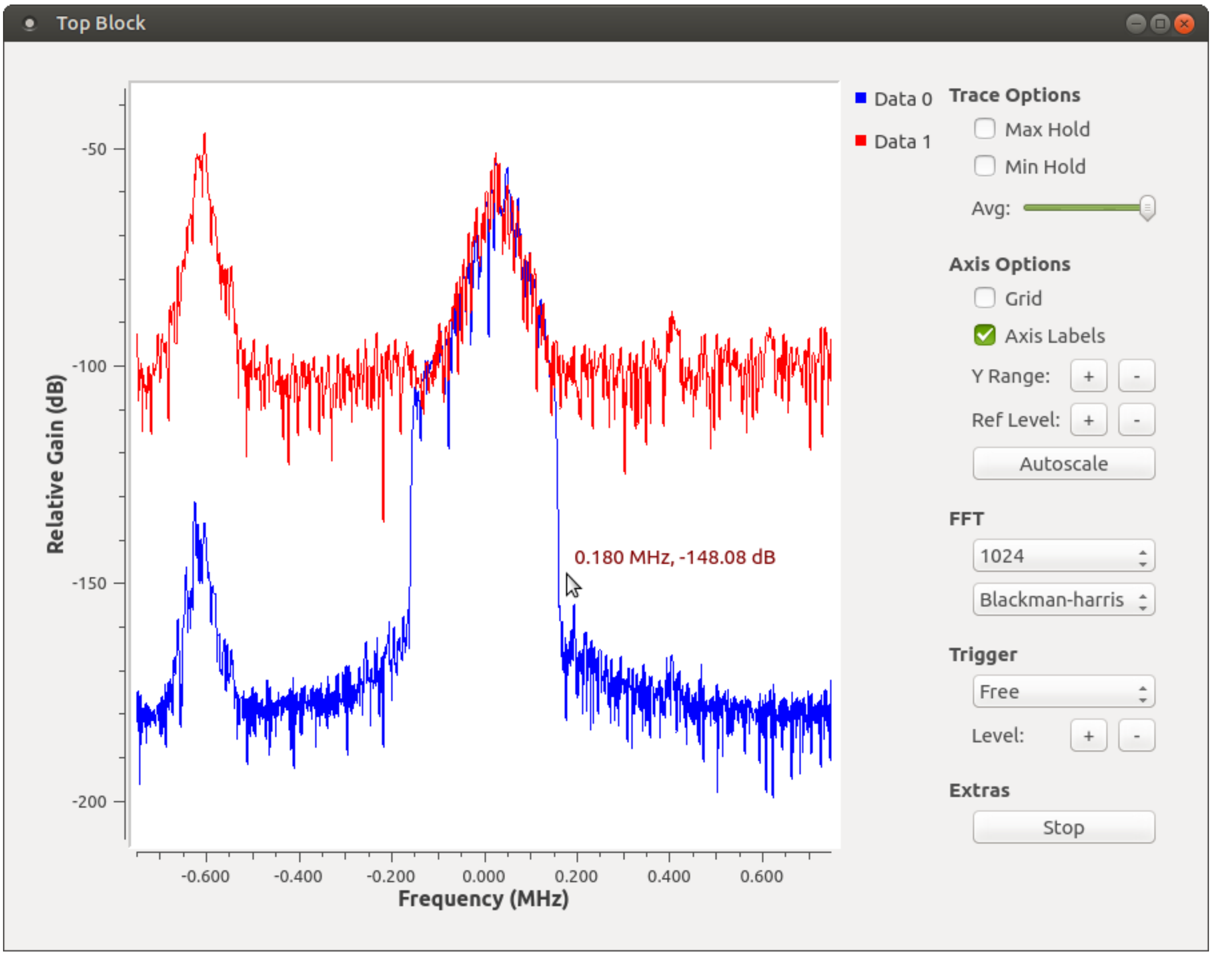
2. Channel extraction by frequency transposition and low-pass filtering



q12)
Offset are 99.5 MHz - (Channel frequency).
So we have 400KHz of offset for the 99.1 MHz channel and -500 KHz for the 100 MHz channel.
To get the offset, we need to multiply the signal by a cosine.

q13)
Using a frequency greater than Fe will ‘reset’ the offset. Using 1.5 MHz cosine will result in a null shift. Using 1.9 MHz cosine will result the same as 400 KHz one.

q14)
Low pass filter --
Sample rate : 1.5 MHz (Fe)
Cutoff freq : 150KHz (bandwidth of channel / 2)
Transition : 10% of cutoff = 15 KHz



Blue signal = after filtering
red signal = before filtering

Channel at 99.1 got reduced power, as we wanted to.

3. Frequency demodulation and restitution

q15)

Carson rule : $B_{FM} \approx 2 \cdot (\Delta f + f_m)$

Δf = 75 KHz

f_m = 53 KHz

donc $B_{FM} \approx 2 \cdot (\Delta f + f_m) \approx 256$ KHz, which correspond to previous measurement.

q16)

$$s_{RF}(t) = A(t) \cos\left(2\pi f_0 t + \frac{\Delta f}{\max(|m(t)|)} \int_{-\infty}^{t} m(u)\,du\right)$$

On a aussi

$$s_R(t) = A(t) \cos(\varphi(t)) \text{ et } s_I(t) = A(t) \sin(\varphi(t))$$

On peut déduire

$$s_{RF}(t) = A(t) \cos(\varphi(t)) \cos(2\pi f_0 t) - A(t) \sin(2\pi f_0 t)$$

On sait que $s(t) = s_R(t) + j\, s_I(t)$

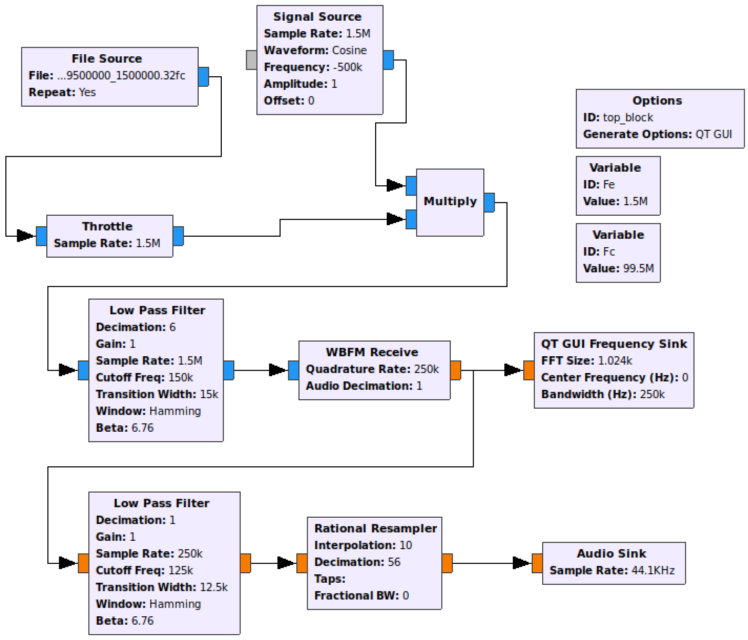
D'où : $s(t) = A(t) \cos(\varphi(t)) + j\, A(t) \sin(\varphi(t))$

$= A(t)\, e^{j\varphi(t)}$

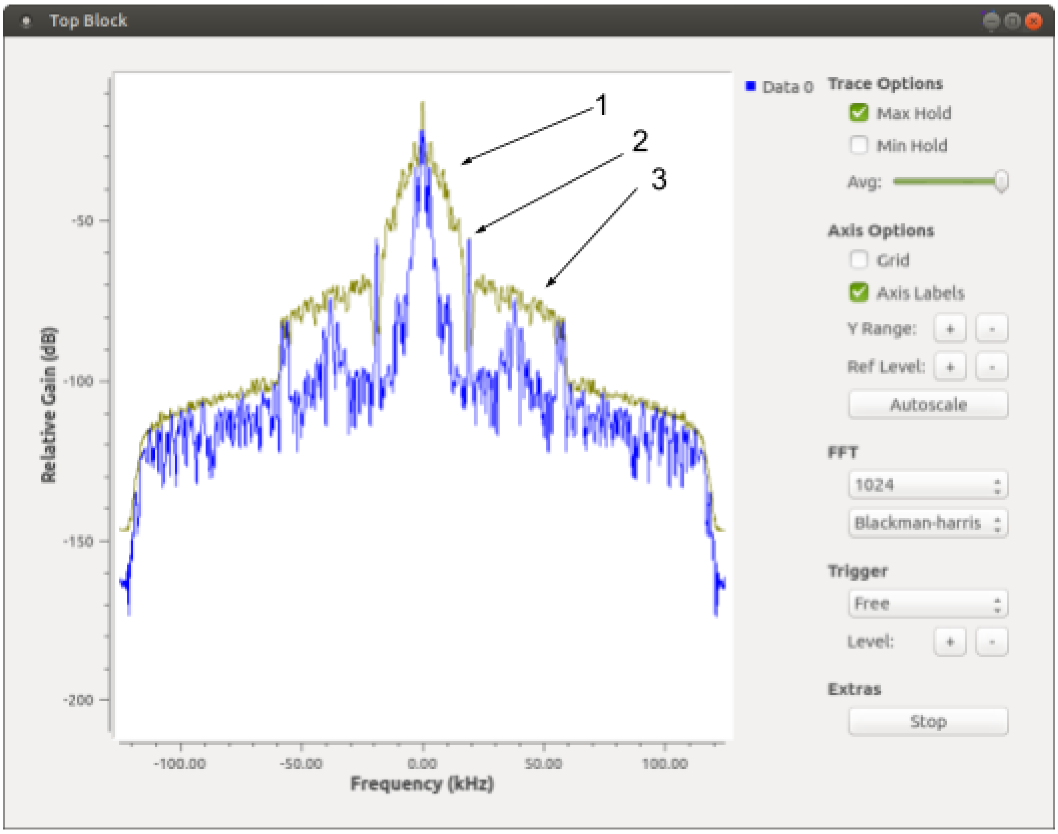
$$\Rightarrow s[k] = A\left[\frac{k}{F_e}\right] e^{j\varphi\left[k/F_e\right]} + b[k]$$

$$\Rightarrow y_f[k] = A\, e^{jk_f \sum_{i=0}^{k} m(i)} + b[k], \quad k_f = \frac{\Delta f}{\max(|m(t)|)}$$

q17)



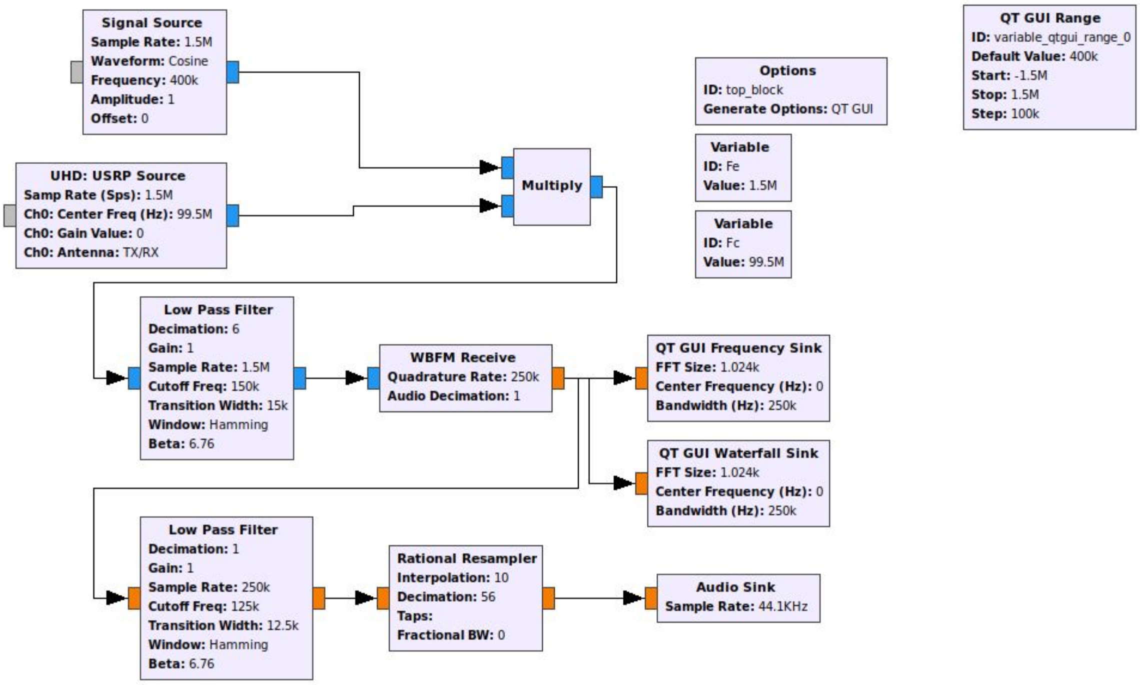
Decimation of 6 will allow a quadrature rate/sample rate of 250 KHz.

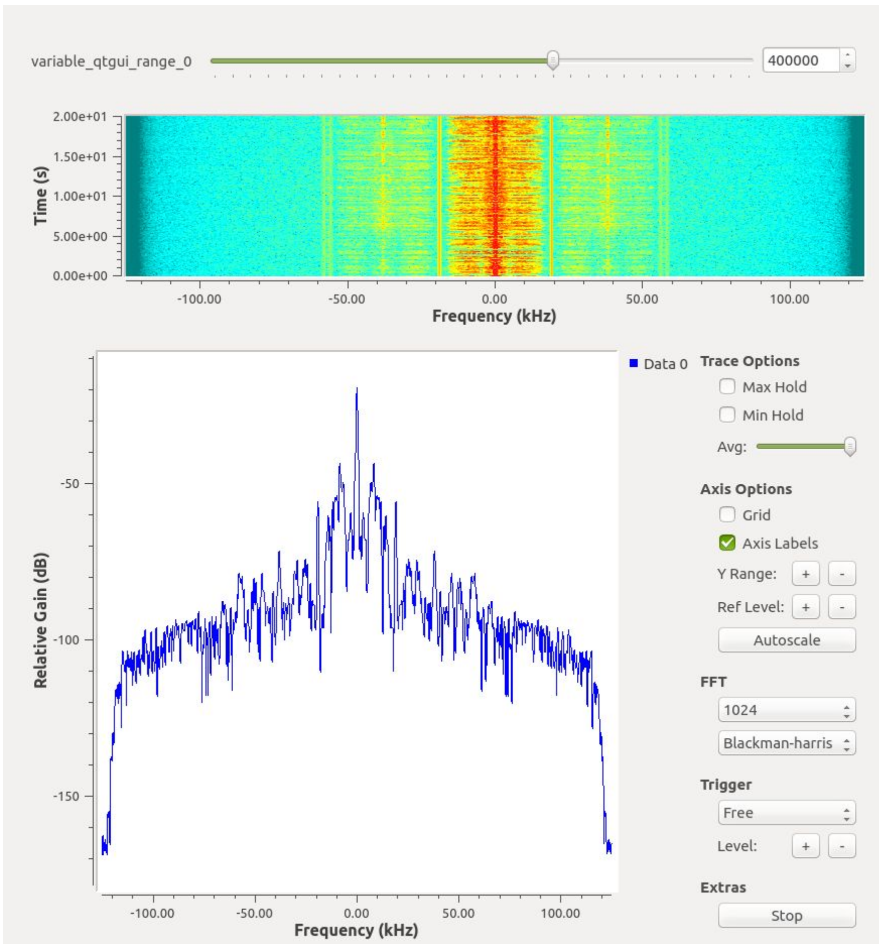


We can see the mono part at the center (1), then the carrier (2), and next, the stereo part (3).

q18)
On skyrock, it's "Jordi" who won the Sam Smith album.
On RFM, counting stars is played.

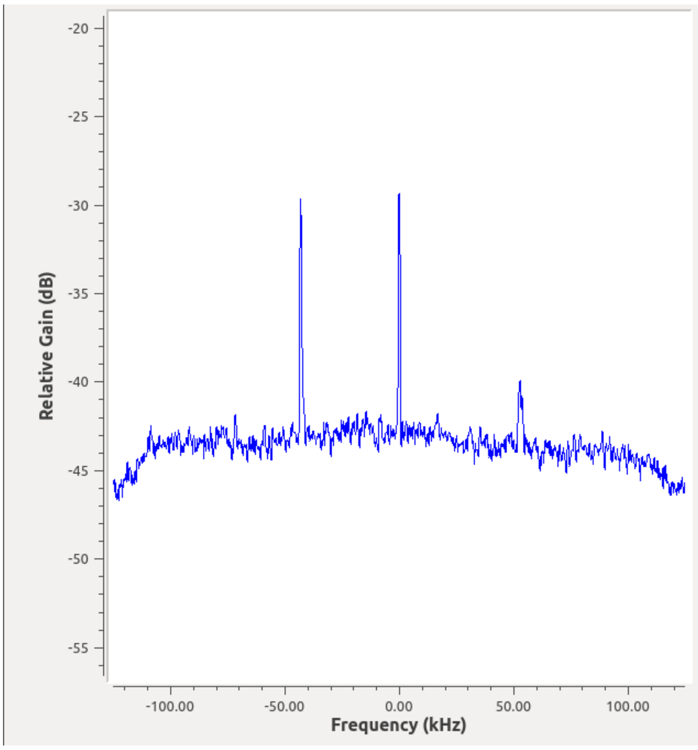
URSP


Signal Source
Sample Rate: 1.5M
Waveform: Cosine
Frequency: 400k
Amplitude: 1
Offset: 0
UHD: USRP Source
Samp Rate (Sps): 1.5M
Ch0: Center Freq (Hz): 99.5M
Ch0: Gain Value: 0
Ch0: Antenna: TX/RX
Multiply
Options
ID: top_block
Generate Options: QT GUI
QT GUI Range
ID: variable_qtgui_range_0
Default Value: 400k
Start: -1.5M
Stop: 1.5M
Step: 100k
Variable
ID: Fe
Value: 1.5M
Variable
ID: Fc
Value: 99.5M
Low Pass Filter
Decimation: 6
Gain: 1
Sample Rate: 1.5M
Cutoff Freq: 150k
Transition Width: 15k
Window: Hamming
Beta: 6.76
WBFM Receive
Quadrature Rate: 250k
Audio Decimation: 1
QT GUI Frequency Sink
FFT Size: 1.024k
Center Frequency (Hz): 0
Bandwidth (Hz): 250k
QT GUI Waterfall Sink
FFT Size: 1.024k
Center Frequency (Hz): 0
Bandwidth (Hz): 250k
Low Pass Filter
Decimation: 1
Gain: 1
Sample Rate: 250k
Cutoff Freq: 125k
Transition Width: 12.5k
Window: Hamming
Beta: 6.76
Rational Resampler
Interpolation: 10
Decimation: 56
Taps:
Fractional BW: 0
Audio Sink
Sample Rate: 44.1KHz


variable_qtgui_range_0
400000
Time (s)
2.00e+01
1.50e+01
1.00e+01
5.00e+00
0.00e+00
-100.00
-50.00
0.00
50.00
100.00
Frequency (kHz)
Relative Gain (dB)
-50
-100
-150
Data 0
Trace Options
Max Hold
Min Hold
Avg:
Axis Options
Grid
Axis Labels
Y Range:
Ref Level:
Autoscale
FFT
1024
Blackman-harris
Trigger
Free
Level:
Extras
Stop

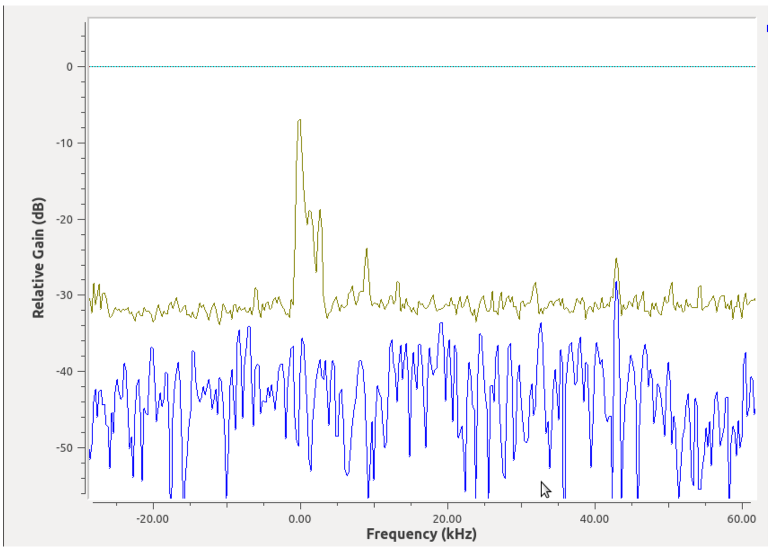
Third part : Reception of VOLMET messages in AM-SSB

q19)



q19)
Volmet is around 11.253 MHz, center peak is at 11.2965 MHz (Fc).
Fc - 43 KHz ~ 11.253 MHz, so the 'left' peak is volmet, with -43 KHz offset.

q20)
We need 43 KHz offset to center around volmet frequency.



q21)

$$s_{RF}(t) = A(t)\cos\left(2\pi f_0 t + \frac{\Delta f}{\max(|m(t)|)} \int_{-\infty}^{t} m(u)\,du\right)$$

...

On a aussi

$$s_R(t) = A(t)\cos(\varphi(t)) \text{ et } s_I(t) = A(t)\sin(\varphi(t))$$

On peut déduire

$$s_{RF}(t) = A(t)\cos(\varphi(t))\cos(2\pi f_0 t) - A(t)\sin(2\pi f_0 t)$$

On sait que $s(t) = s_R(t) + j\, s_I(t)$

D'où : $s(t) = A(t)\cos(\varphi(t)) + j\, A(t)\sin(\varphi(t))$

$$= A(t)\, e^{j\varphi(t)}$$

$$\Rightarrow s[k] = A\left[\frac{k}{F_e}\right] e^{j\varphi\left[k/F_e\right]} + b[k]$$

$$\Rightarrow y_f[k] = A\, e^{j k_f \sum_{i=0}^{k} m(i)} + b[k], \quad k_f = \frac{\Delta f}{\max(|m(t)|)}$$

- $s_{RF}(t) = R\{s(t)\, e^{-j2\pi f_0 t}\}$

$$= R\{m(t) + H\{m(t)\}\, e^{j2\pi f_0 t}$$

- $S_a(f) = S_{RF}(f) + jH\{S_{RF}(f)\}$

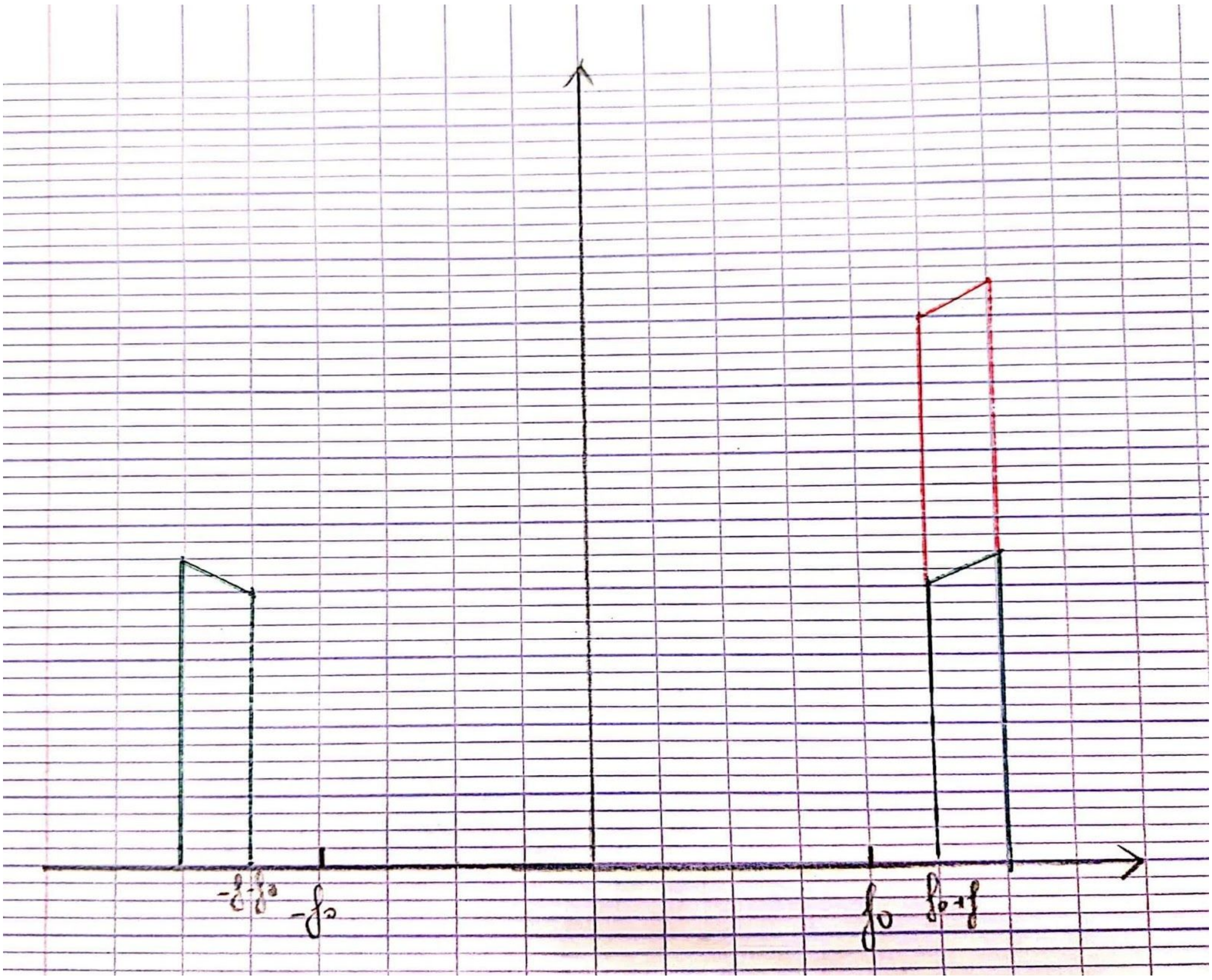
$$= S_{RF}(f) + \operatorname{sgn}(f)\, S_{RF}(f)$$

- $S_a(f) = S(f-f_0) = m(f-f_0) + \operatorname{sgn}(f-f_0)\cdot m(f-f_0)$

Donc

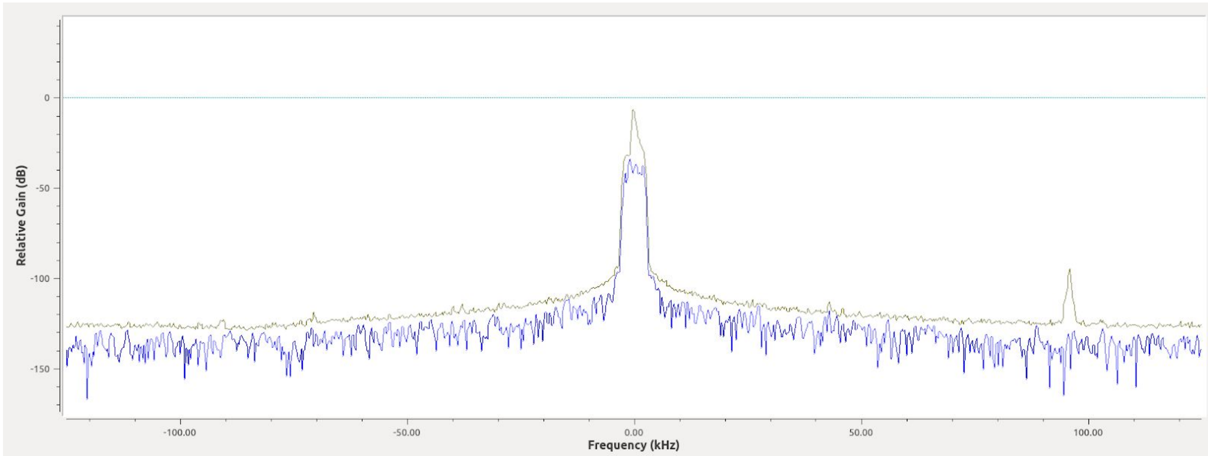
$$S_{RF} = \frac{1}{2}\left(S_a(f) + S_a^*(f)\right)$$

$$= \frac{1}{2}\left[m(f-f_0) + \operatorname{sgn}(f-f_0)\, m(f-f_0) + m(f-f_0) + \operatorname{sgn}(-f-f_0)\, m(-f-f_0)\right.$$



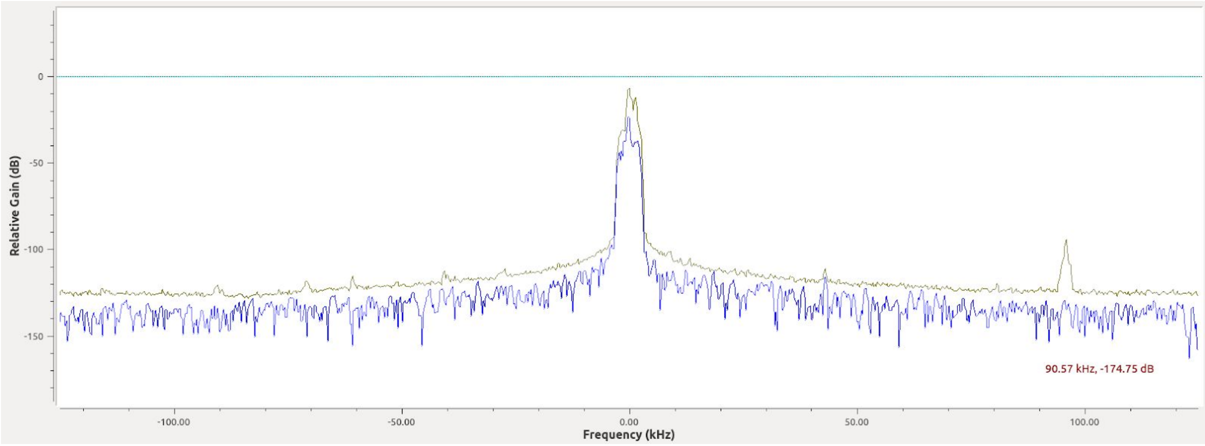
q22)
There is a sideband at +43khz, around 11.2965 MHz.
The upperband is conserved.

q23)



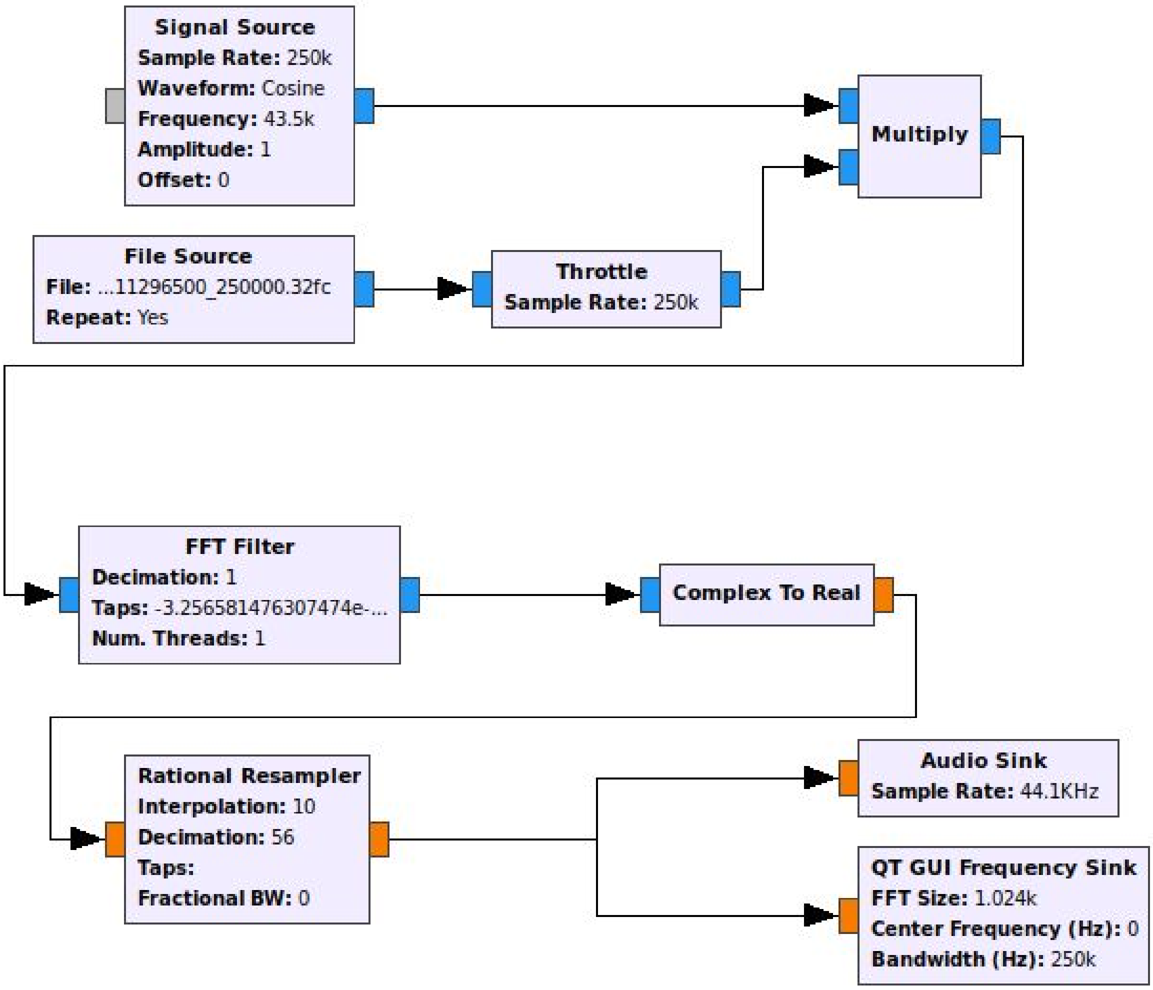
After filtering

q24)



With FFT Fitler.

q25)



With these blocks, we get the desired audio output.