Introduction to Software-Defined Radio :
Physical layer implementation using and USRP

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For the course :
Protocol for connected objects

1 Introduction

These three lab classes are focused on the processing of real communication signals like audio broadcasting or aeronautical communication. A single receiver can record almost all these signals and then process them numerically using GNURadio in order to recover all the information. We will first present the theoretical aspects of this acquisition device and then developp the receptio of frequency modulation broadcasting and will end on the reception of VOLMET messages in AM-SSB.

2 First part : presentation of the acquisition device

In this part, we will use a National Instruments USRP-2900 software defined radio transceiver. At the end of the reception chain, the computer stores the flow of samples in a file that we use during the following exercises under GNURadio.

2.1 Question 1

We assume that the the received signal is similar to the transmitted one ($r_{RF}(t) = S_{RF}(t)$) and we know the expression of $S_{RF}(t)$:

$$r_{RF}(t) = S_{RF}(t) = s_R(t)cos(2\pi f_0 t) - s_I(t)sin(2\pi f_0 t) \tag{1}$$

We also know that $\tilde{r}_R(t)$ and $\tilde{r}_I(t)$ come from multiplier outputs. So, we have :

$$\tilde{r}_R(t) = S_{RF}(t) * cos(2\pi f_c t) \tag{2}$$

$$\tilde{r}_I(t) = S_{RF}(t) * sin(2\pi f_c t) \tag{3}$$

First, we try to find $\tilde{r}_R(t)$:

$$\begin{aligned}
\tilde{r}_R(t) &= (s_R(t)cos(2\pi f_0 t) - s_I(t)sin(2\pi f_0 t)) * cos(2\pi f_c t) \\
&= s_R(t)cos(2\pi f_0 t)cos(2\pi f_c t) - s_I(t)sin(2\pi f_0 t)cos(2\pi f_c t) \\
&= s_R(t)(\frac{cos(2\pi f_0 t - 2\pi f_c t) + cos(2\pi f_0 t + 2\pi f_c t)}{2}) - s_I(t)(\frac{sin(2\pi f_0 t + 2\pi f_c t) + sin(2\pi f_0 t - 2\pi f_c t)}{2}) \\
&= \frac{s_R(t)}{2}(cos(2\pi t(f_0 - f_c)) + cos(2\pi t(f_0 + f_c))) - \frac{s_I(t)}{2}(sin(2\pi t(f_0 + f_c)) + sin(2\pi t(f_0 - f_c)))
\end{aligned}$$

And we do the same to find $\tilde{r}_I(t)$:

$$\begin{aligned}
\tilde{r}_I(t) &= (s_R(t)cos(2\pi f_0 t) - s_I(t)sin(2\pi f_0 t)) * sin(2\pi f_c t) \\
&= s_R(t)cos(2\pi f_0 t)sin(2\pi f_c t) - s_I(t)sin(2\pi f_0 t)sin(2\pi f_c t) \\
&= s_I(t)(\frac{cos(2\pi f_0 t - 2\pi f_c t) - cos(2\pi f_0 t + 2\pi f_c t)}{2}) - s_R(t)(\frac{sin(2\pi f_0 t + 2\pi f_c t) - sin(2\pi f_0 t - 2\pi f_c t)}{2}) \\
&= \frac{s_I(t)}{2}(cos(2\pi t(f_0 - f_c)) - cos(2\pi t(f_0 + f_c))) - \frac{s_R(t)}{2}(sin(2\pi t(f_0 + f_c)) - sin(2\pi t(f_0 - f_c)))
\end{aligned}$$

2.2 Question 2

We search the characteristics of the h filter to get $r_R(t) = s_R(t)$ and $r_I(t) = s_I(t)$ with $f_c = f_0$. With the previous question, we have the expression of $\tilde{r}_R(t)$ and $\tilde{r}_I(t)$:

$$\begin{aligned}
\tilde{r}_R(t) &= \frac{s_R(t)}{2}(cos(2\pi t(f_0 - f_0)) + cos(2\pi t(f_0 + f_0))) - \frac{s_I(t)}{2}(sin(2\pi t(f_0 + f_0)) + sin(2\pi t(f_0 - f_0))) \\
&= \frac{s_R(t)}{2}(cos(2\pi t(0)) + cos(2\pi t2f_0)) - \frac{s_I(t)}{2}(sin(2\pi t2f_0) + sin(2\pi t(0))) \\
&= \frac{s_R(t)}{2}(1 + cos(4\pi tf_0)) - \frac{s_I(t)}{2}(sin(4\pi tf_0) + 0) \\
&= \frac{s_R(t)}{2}(1 + cos(4\pi tf_0)) - \frac{s_I(t)}{2}sin(4\pi tf_0) \\
\tilde{r}_I(t) &= \frac{s_I(t)}{2}(cos(2\pi t(f_0 - f_0)) - cos(2\pi t(f_0 + f_0))) - \frac{s_R(t)}{2}(sin(2\pi t(f_0 + f_0)) - sin(2\pi t(f_0 - f_0))) \\
&= \frac{s_I(t)}{2}(cos(2\pi t(0)) - cos(2\pi t2f_0)) - \frac{s_R(t)}{2}(sin(2\pi t2f_0) - sin(2\pi t(0))) \\
&= \frac{s_I(t)}{2}(1 - cos(4\pi tf_0)) - \frac{s_R(t)}{2}(sin(4\pi tf_0) - 0)) \\
&= \frac{s_I(t)}{2}(1 - cos(4\pi tf_0)) - \frac{s_R(t)}{2}sin(4\pi tf_0)
\end{aligned}$$

The low-pass filter h cut the pic at a certain frequency without modifying the $s_R(t)$ and $s_I(t)$ signals. As said previously, the h filter must have $r_R(t) = s_R(t)$ and $r_I(t) = s_I(t)$. We also have $r_R(t) = \tilde{r}_R(t) * h$. In the frequency domain, we can use the Fourier transform :

$$\begin{aligned}
\tilde{R}_R(f) &= F\{\tilde{r}_R(t)\} \\
&= s_R(f) * \frac{1}{2}\delta(f) * [\delta(f) + \frac{\delta(f + 2f_0) - \delta(f - 2f_0)}{2}] - s_I(f) * \frac{1}{2}\delta(f) * [\frac{j}{2}\delta(f + 2f_0) - \delta(f - 2f_0)] \\
&= \frac{s_R(f)}{4}[2\delta(f) + \delta(f + 2f_0) + \delta(f - 2f_0)] - \frac{s_I(f)}{4j}[\delta(f - 2f_0) - \delta(f + 2f_0)] \\
&= \frac{1}{4}[2s_R(f) + s_R(f + 2f_0) - s_R(f - 2f_0) + js_I(f - 2f_0) - js_I(f + 2f_0)]
\end{aligned}$$

By analogy with the previous expression, we have :

$$\begin{aligned}
\tilde{R}_I(f) &= F\{\tilde{r}_I(t)\} \\
&= s_I(f) * \frac{1}{2}\delta(f) * [\delta(f) - \frac{\delta(f + 2f_0) - \delta(f - 2f_0)}{2}] - s_R(f) * \frac{1}{2}\delta(f) * [\frac{j}{2}\delta(f + 2f_0) - \delta(f - 2f_0)] \\
&= \frac{s_I(f)}{4}[2\delta(f) - \delta(f + 2f_0) - \delta(f - 2f_0)] - \frac{s_R(f)}{4j}[\delta(f - 2f_0) - \delta(f + 2f_0)] \\
&= \frac{1}{4}[2s_I(f) - s_I(f + 2f_0) - s_I(f - 2f_0) + js_R(f - 2f_0) - js_R(f + 2f_0)]
\end{aligned}$$

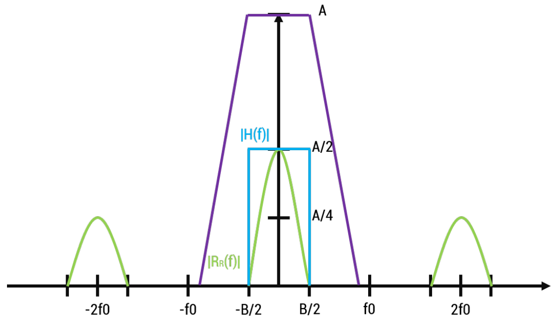

Figure 1 – Signal and filter representation in the frequency domain

The modules of real and imaginary parts are identical. The difference is in a phase shift oh π between these two.

Thus, we can deduce the characteristics of the h filter :

— low-pass with $\frac{B}{2} < f_c < 2f_0 - \frac{B}{2}$ as we only want to keep the central band of the signals $\tilde{R}_R(f)$ and $\tilde{R}_I(f)$

— narrowband with $f_0 > \frac{B}{2}$ to avoid spectral overlap

— $|H(f)| = \begin{cases} 2 \text{ if } \frac{B}{2} < f < 2f_0 - \frac{B}{2} \\ 0 \text{ if } f > 2f_0 - \frac{B}{2} \end{cases}$

2.3 Question 3

The IQ receiver cannot work properly with wide-band signals ($f_0 < \frac{B}{2}$). Indeed, a $\tilde{R}_R(f)$ (or $\tilde{R}_I(f)$) wide-band signal induces spectral overlap. So, it is almost impossible to isolate the baseband signal and demodulate it. A solution would be to use a Hilbert filter which is a non-causal filter. Thus, an analog implementation is not possible and a digital one could provide a good result but would induce delays. Finally, the introduced IQ receiver can only work correctly with narrowband signals.

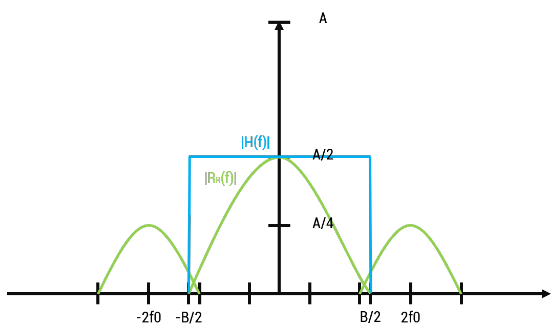

Figure 2 – Large band signal representation in the frequency domain

As you can see in the previous figure, the filter cannot keep the central part of the signal because of the overlap.

2.4 Question 4

In order to recover $r_R(t)$ ..., we have to choose the sampling period T_e checking the Nyquist-Shannon criterion, as $f_e > 2f_{max}$, with $f_e = \frac{1}{T_e}$ the sampling frequency and f_{max} the maximal frequency of the useful signal. In our case, $f_{max} = \frac{B}{2}$, thus $f_e > 2\frac{B}{2} = B$. Finally, the sampling period must check $T_e < \frac{1}{B}$. The respect of the Nyquist-Shannon criterion certificate the total conservation of the information transmitted by the analog signal. The obtained sampled signal allows the perfect recovery of the initial analog signal.

2.5 Question 5

Although it is theorically possible, interchanging frequency transposition and analog to digital conversion stages of the IQ demodulator does not seem relevant. Indeed, to respect the Nyquist-Shannon criterion, the constraint on the sampling becomes very hard, because the maximal frequency of the signal is largely higher ($f_{max} = f_0\frac{B}{2}$). Thus, $f_e > 2f_0 + B$ and finally, $T_e < \frac{1}{2f_0+B}$. The ADC must be very swift, but today, it is difcult to find one, or it is very expensive (ADC12J4000, 12 bits, 3 GHz, >2000€). Moreover, "hyperfrequency" components have many issues, particurlarly regarding their behaviour against the noise.

2.6 Question 6

We want to express , in the frequential domain then in the temporal domain, the analytic signal and the complex envelop dependinf of f_0 of the following signal :

$$\begin{aligned} s_{RF}(f) &= A(t)cos(2\pi f_0 t + \phi(t)) \\ &= s_R(t)cos(2\pi f_0 t) - s_I(t)sin(2\pi f_0 t) \end{aligned}$$

Using the Fourier transform, we can have :

$$\begin{aligned} s_{RF}(f) &= \frac{s_R(f)}{2}[\delta(f-f_0)+\delta(f+f_0)] - \frac{s_I(f)}{2j}[\delta(f-f_0)-\delta(f+f_0)] \\ &= \frac{1}{2}[s_R(f-f_0)+s_R(f+f_0)+js_I(f-f_0)-js_I(f+f_0)] \end{aligned}$$

Thus, the analytic signal can be modelized by :

$$\begin{aligned} s_a(f) &= s_{RF}(f) + j(-j * sgn(f)s_{RF}(f)) \\ &= s_{RF}(f) + sgn(f)s_{RF}(f) \\ &= 2s_{RF}(f) \\ &= s_{RF}(ff_0) + js_I(f-f_0) \\ &= [s_R(f) + js_I(f)] * \delta(f-f_0) \end{aligned}$$

In the temporal domain, using the inverse Fourier transform, the signal is :

$$s_a(t) = [s_R(t) + js_I(t)]\exp\{j2\pi f_0 t\}$$

The complex envelop in the frequential domain is :

$$s(f) = s_a(f+f_0) = s_R(f) + js_I(f)$$

In the temporal domain, the complex envelop is :

$$\begin{aligned} s(t) &= s_a(t)\exp\{-j2\pi f_0 t\} \\ &= s_R(t) + js_I(t) \end{aligned}$$

3 Second part : reception of frequency modulation broadcasting

In this part, we use the USRP to listen to FM radio signals. The Very High Frequency (VHF) is composed of all wavelengths from 1 to 10 meters and frenquencies between 30 and 300 MHz. We will focus on the bandwith from 87.5 MHz and 108 MHz dedicated to radiodiffusion Our goal was to study a specific recording and restore the content of this recording. To do this, we used GNURadio. The center frequency of the recording is $f_c = 99.5$ MHz and the sample frequency is $F_e = 1.5$ MHz.

3.1 Frequency analysis of the recording

3.1.1 Question 7

The following list presents the role of each block used in the processing chain :

— **Options** : gives info/options of the project (type of graphic library, author etc)

— **Variables** : assigns value and name to a unique variable (here sample frequency and cutoff frequency)

— **File source** : reads raw data values in binary format from the specified file (on repeat because the recording length is of 10s of seconds, so if it is not repeated the signal will stop)

— **Throttle** : flow of samples such that the average rate does not exceed the specified rate. It gives the sample frequency and if it is not implemented, the recording will not be read in real time

— **QT GUI Frequency Sink** : frequential graphic representation, fft to see power in terms of the other powers

— **Range** : dynamic modification of the variables

3.1.2 Question 8

The missing values are :

— $F_e = 1.5MHz$

— $f_c = 99.5MHz$

— sample rate = F_e

— center frequency = f_c

— bandwidth = F_e

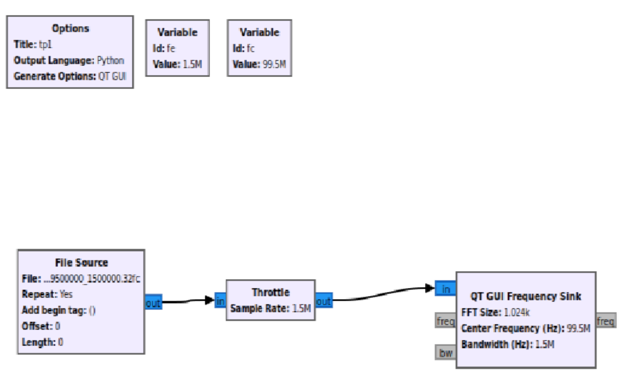

FIGURE 3 – Workspace with the missing values to visualize the signal

3.1.3 Question 9

We observe 3 frequency channels :

- **RFM**, centered on the frequency 99.1 MHz
- **Nostalgie**, centered on the frequency 99.5 MHz (not visible if max hold is not on, because there may be interferences on the INSA campus)
- **Skyrock**, centered on the frequency 100.0 MHz

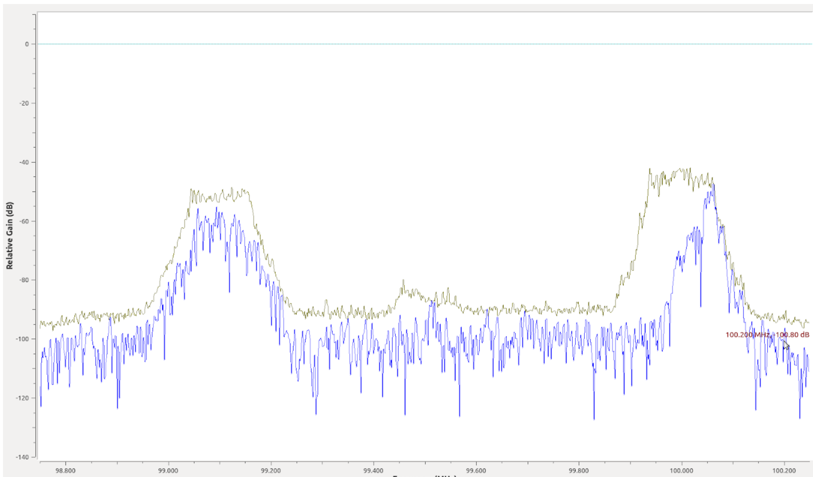

FIGURE 4 – Radio channels of the 3 different stations

3.1.4 Question 10

The signal-to-noise ratio is the ratio in Watt between the power of the signal and the power of the noise. In logarithm, it is the difference in Decibels between the power of the signal and the power of the noise.

To calculate this ratio, we use the maximum signal power and the maximum noise power at the foot of the radio channel.

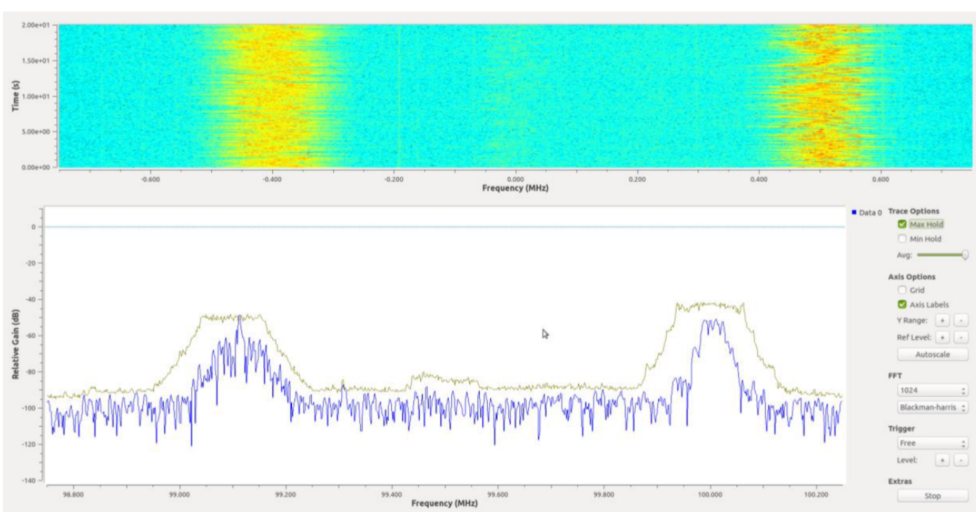

FIGURE 5 – SNR of the radio channels

The signal-to-noise ratios in decibel are :

- **RFM**, $SNR = -49.05 + 89.60 = 40.55dB$ (around 10000 more signal power than noise power)
- **Nostalgie**, $SNR = -81.12 + 87.91 = 6.79dB$ or between 3 and 6 dB (between 2 to 4 times more signal power than noise power)
- **Skyrock**, $SNR = -42.26 + 88.25 = 45.99dB$ (around 10000 more signal power than noise power)

The ratios are enough to demodulate the signals of RFM and Skyrock because they have around 10000 more signal power than noise power. For Nostalgie, the demodulation could be more complicated because the noise has a more important effect (we only have between 2 to 4 times more signal power than noise power).

3.1.5 Question 11

We calculate the bandwidth, using the bottom of the channel, at the threshold of the maximum noise.

- **RFM**, $bandwidth = 99.280 - 98.952 = 328kHz$
- **Nostalgie**, $bandwidth = 99.577 - 99.428 = 149kHz$
- **Skyrock**, $bandwidth = 100.148 - 99.808 = 340kHz$

3.2 Channel extraction by frequency transposition and low-pass filtering

The role of each new block is listed below :

- **Signal source** : to generate the complex exponential
- **QT GUI Range** : to define dynamically f_l during the software execution
- **Multiply** : to make the product

3.2.1 Question 12

The QI demodulator brings the signal back to base band in quadrature which means we bring 99,5 MHz to 0 Hz. Then, we sampled the signal.

The frequency offsets needed to center each channel are the following :

- **RFM**, $f = 99.1 - 100$
- **Nostalgie**, $f = 0$ already centered on this station
- **Skyrock**, $f = 99.5 - 100$

The offset is done by multiplying the signal by a cosinus with the right frequency.

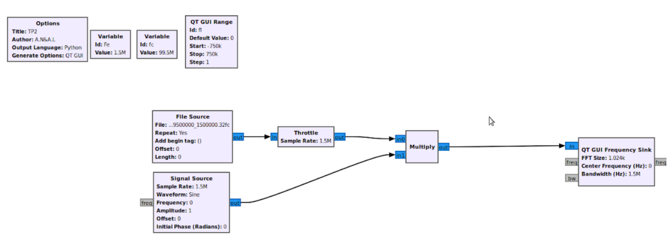

FIGURE 6 – Workspace updated with the multiplication of the signal by a cosinus

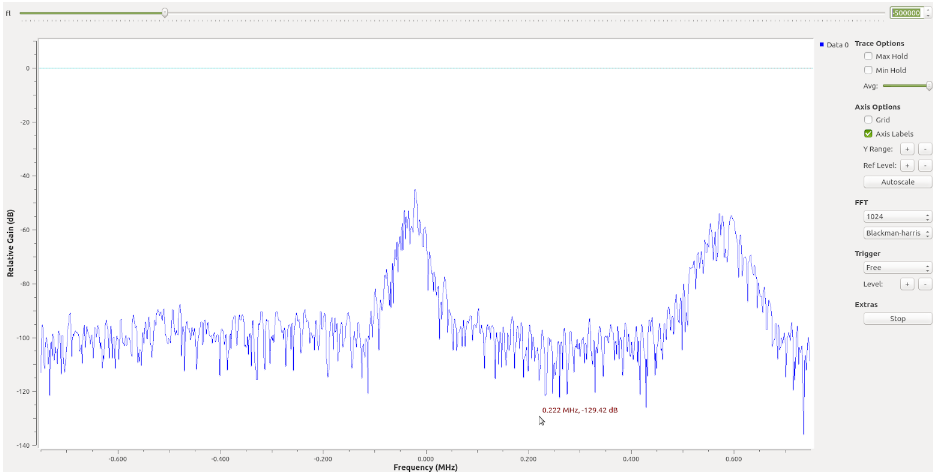

FIGURE 7 – Skyrock Channel with the associated offset

3.2.2 Question 13

When we use an offset of $\frac{f_e}{2}$ or another multiple of f_e, we have a complete rotation. There will be no difference because of the periodicity of the signal. When the offset varies, the 0 will be shifted around the center frequency.

The role of each new block is listed below :

- **Low Pass Filter block** : to implement a low-pass filter. This filter passes signals with a frequency lower than a selected cutoff frequency and attenuates signals with frequencies higher.
- **Decimation** : to delete samples which means we lose information. This modifies the maximum frequency we can retrieve. For example, when we decimate by 6 the sample frequency is also devided by 6 : $\frac{1500}{6} = 250kHz$.

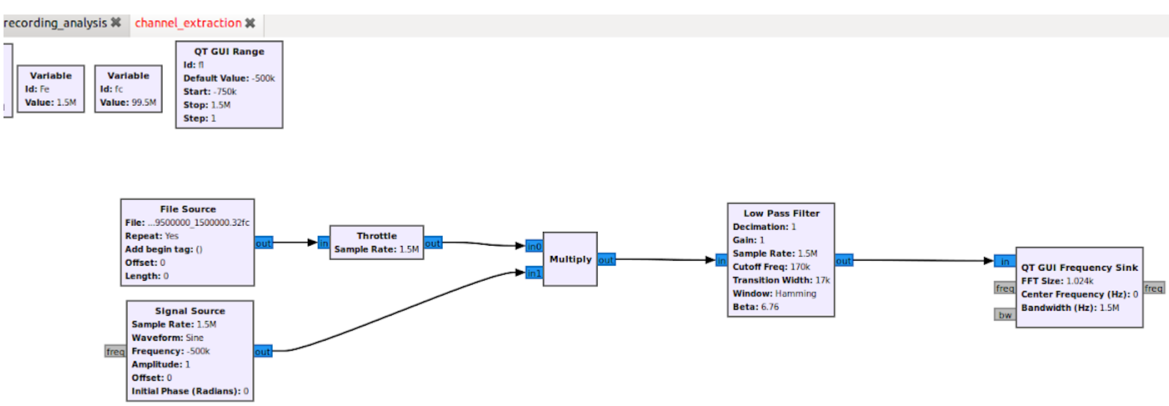

FIGURE 8 – Workspace updated with a low-pass filter

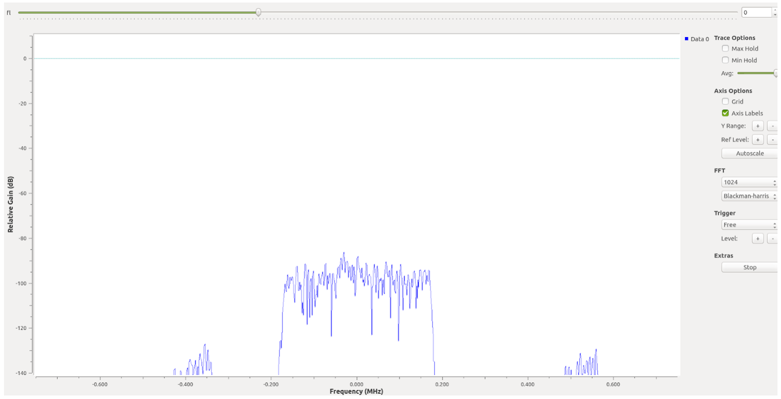

FIGURE 9 – Skyrock channel centered with the low-pass filter

3.2.3 Question 14

The parameters of the low-pass filter are the following :

- Transition frequency : 10 %
- Decimation : 1
- Cutoff frequency : 125 kHz
- Sample frequency : 250 kHz

The sample frequency is equal to the bandwidth in the output sink.

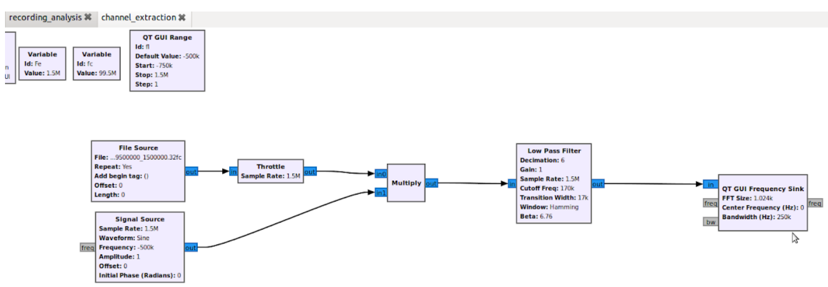

FIGURE 10 – Workspace updated with a decimation = 6 in the low-pass filter

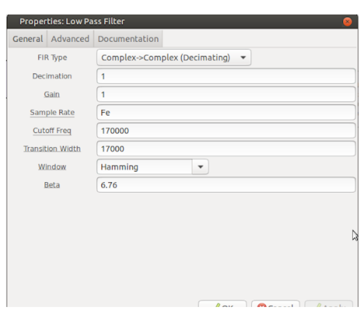

FIGURE 11 – Parameters of the low-pass filter with a decimation = 1

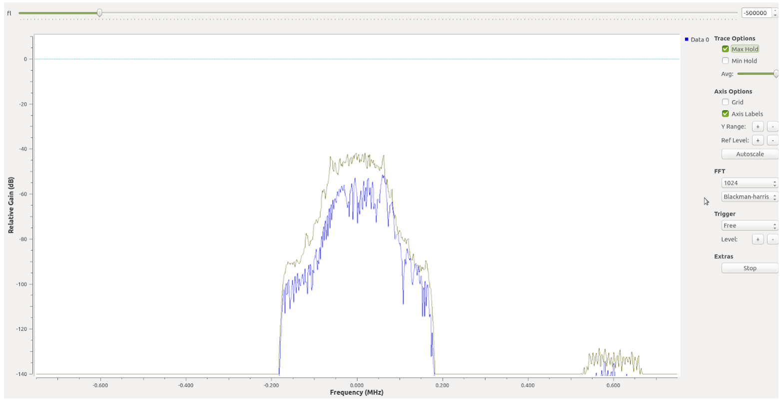

FIGURE 12 – Previous signal with a $f_l = 500kHz$

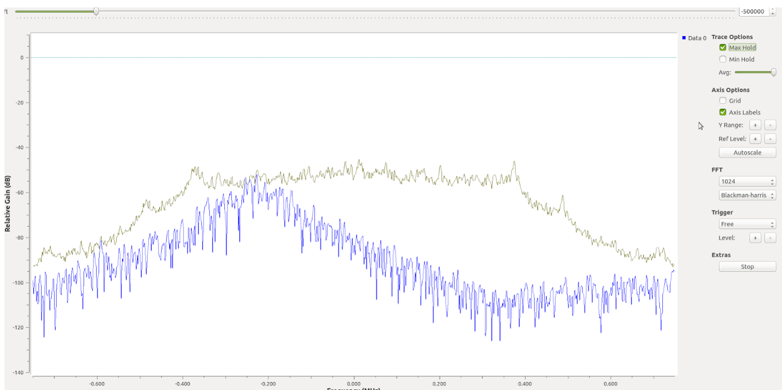

FIGURE 13 – Previous signal with a $f_l = 500kHz$ and a decimation of 6

3.3 Frequency demodulation and restitution

3.3.1 Question 15

With the Carson rule, we have :

$$\begin{aligned} B_{FM} &\approx 2(\Delta f + f_M) \\ &\approx 2(75000 + 57000) \\ &\approx 264000 \end{aligned}$$

with $\Delta f = \frac{100+75}{2} = 75$ kHz.

We know that FM channels have a bandwidth of 256 kHz. So, the results have the same order of magnitude.

3.3.2 Question 16

We have :

$$\begin{aligned} s_{RF}(t) &= A(t)cos(2\pi f_0 t + \phi(t)) \\ &= s_R(t)cos(2\pi f_0 t) - s_I(t)sin(2\pi f_0 t) \\ &= Acos(2\pi f_0 t + \frac{\Delta f}{max(|min(t)|)} \int_{-\infty}^{t} m(u)du) \end{aligned}$$

Thus, we can pose :

$$\phi(t) = \frac{\Delta f}{max(|min(t)|)} \int_{-\infty}^{t} m(u)du$$

Moreover, we have :

$$\begin{aligned} s_R(t) &= A(t)cos(\phi(t)) \\ s_I(t) &= A(t)sin(\phi(t)) \end{aligned}$$

We then obtain :

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

We also know, from question 6 :

$$s(t) = s_R(t) + js_I(t)$$

Thus, using the complex envelop, we have :

$$\begin{aligned} s(t) &= A(t)cos(\phi(t)) + jA(t)sin(\phi(t)) \\ &= A(t)\exp\{j\phi t\} \\ &= A(t)\exp\left\{j\frac{\Delta f}{max(|m(t)|)} \int_{-\infty}^{t} m(u)du\right\} \end{aligned}$$

We can discretise this complex envelop at the frequency F_e :

$$s[k] = A\left[\frac{k}{F_e}\right]\exp\left\{j\frac{\Delta f}{max(|min(t)|)} \sum_{i=0}^{k} m(i)\right\} + b[k]$$

As $A\left[\frac{k}{F_e}\right]$ is constant :

$$y_l[k] = A\exp\left\{jk_f \sum_{i=0}^{k} m(i)\right\} + b[k]$$

With $k_f = \frac{\Delta f}{max(|min(t)|)}$ and $b[k]$ a complex noise term introduced by the propagation channel as well as by the transceiver itself. It follows a normal distribution with a variance of σB and with a null mean. The frequency demodulation can be numerically realised with the following equation :

$$\tilde{m}_l[k] = arg(y_l[k]y_l * [k-1])$$

If we inject the $y_l[k]$ equation in the previous equation, we find a term corresponding to the transmitted message :

$$\begin{aligned}
\tilde{m}_l[k] &= arg(\exp\left\{j(\sum_{i=0}^{k} m(i) - \sum_{i=0}^{k-1} m(i)\right\})) \\
&= arg(A\exp\{jm[k]\}) \\
&= m[k]
\end{aligned}$$

We know that the real signal belongs to $\mathbb{R}$. To find it, we select the signal's real part.

$$\begin{aligned}
s_{RF}(t) &= Re\{A(t)\exp\{j(2\pi f_0 t)\}\} \\
&= Re\{A(t)[cos(2\pi f_0 t) + jsin(2\pi f_0 t)]\} \\
&= A(t)cos(2\pi f_0 t)
\end{aligned}$$

3.3.3 Question 17

We added three new blocks :

- **WBFM receive** : to demodulate a broadcast FM signal. The input is the downconverted complex baseband signal. The output is the demodulated audio which is a composite signal.
- **Low pass filter** : We want to retrieve the monophonic part which is between 0 and 15 kHz with a cutoff frequency of 15 KHz.
- **Audio Sink** : output to display the signal.

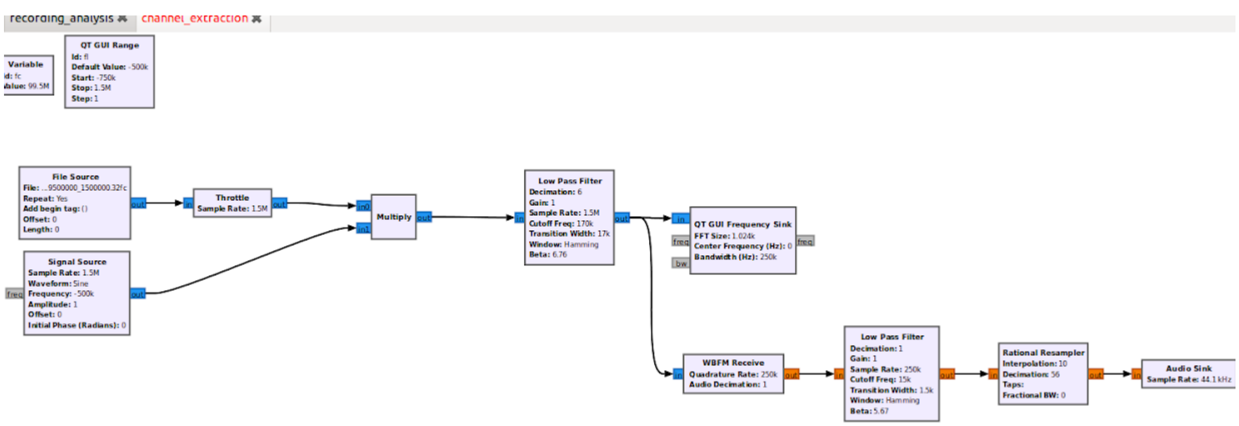

FIGURE 14 – Workspace updated with the three new blocks

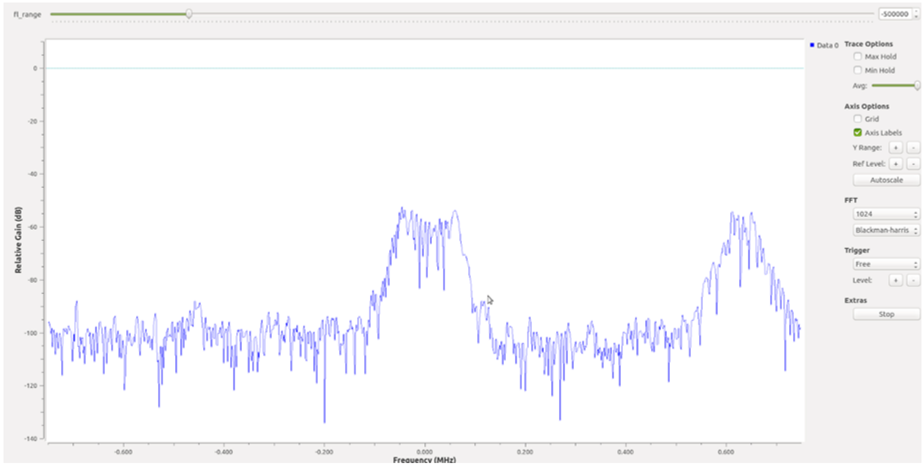

FIGURE 15 – Channel signal output

We also put a rational resampler with a interpolation parameter. Interpolation is the opposite of decimation : it allows to add samples between the existing samples.

3.3.4 Question 18

We demodulated the channels and can now hear what is recorded :

— **Skyrock**, the winner of the Sam Smith album is Jordie

— **RFM**, we are listening to One Republic

— **Nostalgie**, we are listening to YMCA by Village People

3.4 Real-time implementation with an USRP receiver

Here, we use a directive antenna with linear polarization in the PCB axis. Since the antenna works between 850 and 6500 MHz and the fm radio works between 80 and 100 MHz, we can deduct that it is the cable that acts as the real antenna.

We use the same workspace as the one in question 17 but we replace the data source. It is replaced by a UHD USRP Source block to get the data directly from the USRP. The USRP and our signal processing can be used to receive and demodulate Radio FM signals.

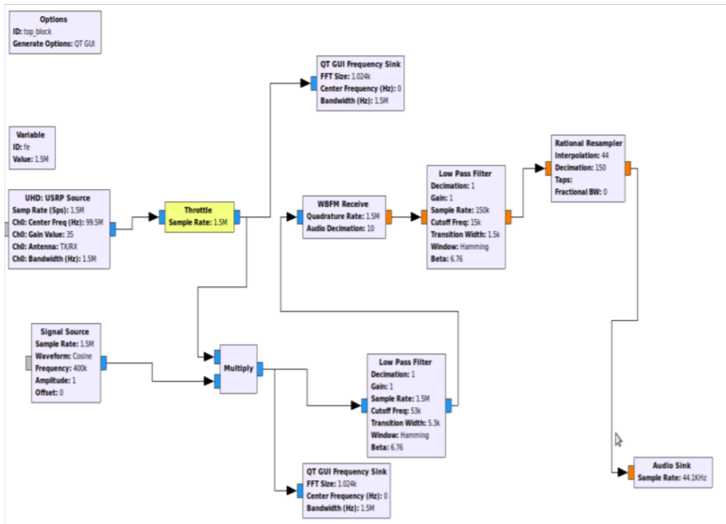

FIGURE 16 – New workspace with the UHD USRP source

The figure below represents the signal received by the USRP card. We can differenciate the radio stations : Skyrock, RFM and Nostalgie. We added a gain of 30 dB and we centered on the frequecny 99.6 MHz.

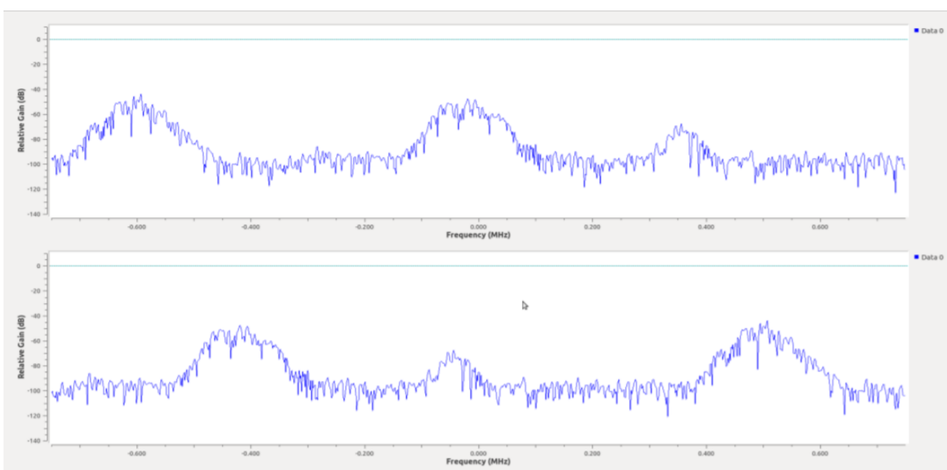

FIGURE 17 – Received signal

4 Third part : reception of VOLMET messages in AM-SSB

The frequency band named High Frequency (HF) ranges from 3 MHz to 30 MHz. Now, we are interested by the frequency sub-band between 11.175 MHz and 11.4 MHz, which is now reserved to the international aeronautic communications and in particular to the VOL METEO service (VOLMET). This is a periodic broadcasting of meteorological information, using a single sideband amplitude modulation.

The recording was obtained with a center frequency of $f_0 = 11.2965$ MHz and a sampling frequency of $F_e = 250$ kHz.

4.1 Frequency analysis of the recording

4.1.1 Question 19

Here, we plotted the modulus of the discrete Fourier transform in decibels, between $f_0 - \frac{F_e}{2}$ and $f_0 + \frac{F_e}{2}$ with the QT GUI Frequency Sink block.

By reading the maximum hold value of out signal output, we observe a maximum of -12,56 dB at a frequency of 11253,70 kHz. The http ://www.dxinfocentre.com/volmet.htm site indicates that the VOLMET station located in the English Royal Air Force has a frequency of 11,253 MHz which corresponds to the value we retrieved.

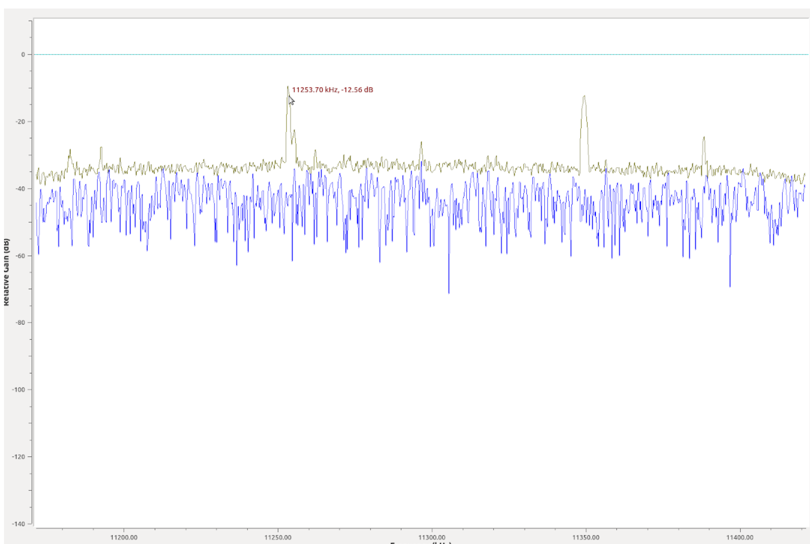

Figure 18 – Channel signal of the VOLMET file

11.3 MHz								
11.247	, 35	MTS	FLK	VIPER		-51 50 14	-58 28 14	IRREGULAR HOURS
11.253	Cont	MKL	ENG	MILITARY ONE		50 28 58	-5 00 00	
11.297	25 , 55	RLAP	RUS	ROSTOV	RR	47 15 12	39 49 02	DAY
11.318	00 , 30	UBB-2	RUS	SIVKAR	RR	61 38 17	50 31 49	DAY
	10 , 40	UNNN	SEO	NOVOSIBIRSK	RR	55 00 16	82 33 44	DAY
	15 , 45	RQCI	RUS	SAMARA	RR	53 11 00	49 46 00	DAY
11.369	01 ,	LWB	ARG	EZEIZA	SS	-34 49 59	-58 31 55	
11.387	00 , 30	VKA-931	AUS	AUSTRALIAN		-23 47 47	133 52 28	
	05 , 35	AWC	IND	KOLKATA		22 38 00	88 27 00	0305 - 1240 Z
	10 , 40	HSD	THA	BANGKOK		13 44 00	100 30 00	2310 - 1145 Z
	15 , 45	ARA	PAK	KARACHI		25 54 00	67 09 00	OUT OF SERVICE
	20 , 50	9VA-43	SNG	SINGAPORE		1 20 11	103 41 10	2250 - 1225 Z
	25 , 55	AWB	IND	MUMBAI		19 05 15	72 51 09	0325 - 1300 Z

Figure 19 – Table of VOLMET stations' frequencies