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PTP Innovative Smart Systems - Protocols for the Connected Objects

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Introduction to Software-Defined Radio

INNOVATIVE SMART SYSTEMS

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Contents

Introduction

This report is the result of the lab session of Signal Defined Radio. During this practical session, we have focused on the reception of real communication signals in order to connect the course, the theory and the applications. We have first demonstrated that we can receive narrow-band signals with our technology without loss of any data in a theoretical way. Then, we have worked on FM radio, with a record first and with real-time signal then. Finally, we have worked on VOLMET AM but we only managed to make it work on the recording cause our USPR couldn't work at VOLMET frequency. During this lab, we implemented our SDR using GNU Radio to process our signals and and USRP-2900 from National Instruments to receive our signals.

Chapter 1

Presentation of the acquisition device

In this part, we are going to study the way of running of the USRP transceiver.
Let's first have a look to the block diagram of the USRP:

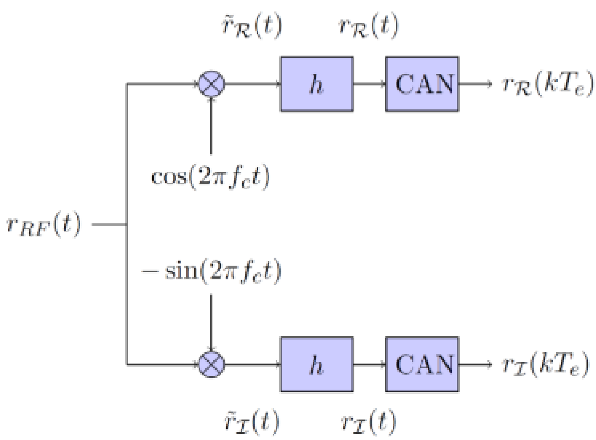

Figure 1.1: Receiver block diagram

In the figure above, we can extract several information about the USRP. First, it receives a temporal signal ($r_{RF}(t)$ on the left) which has a real and an imaginary part. These two parts are treated independently.
On one side, a cos is applied to the real part of the signal, to obtain the $\tilde{r}_R(t)$ signal. On the other side, a sin is applied to the imaginary part of the signal, to obtain the $\tilde{r}_I(t)$ signal.

Now, we are going to express the value of both $\tilde{r}_R(t)$ and $\tilde{r}_I(t)$ in function of $s_{RF}(t)$, f_0 and f_c, considering that the received signal is equal to the transmitted one ($r_{RF}(t) = s_{RF}(t)$):

$$\begin{cases} \widetilde{r_R}(t) &= \quad r_{RF}(t)\cos(2\pi f_c t) \\ \widetilde{r_I}(t) &= -r_{RF}(t)\sin(2\pi f_c t) \end{cases} \tag{1.1}$$

Since, we assume the following hypothesis: $r_{RF} = s_{RF}$, we have,

$$\Rightarrow \begin{cases} \widetilde{r_R}(t) &= \quad s_R(t)\cos(2\pi f_0 t)\cos(2\pi f_c t) - s_I(t)\sin(2\pi f_0 t)\cos(2\pi f_c t) \\ \widetilde{r_I}(t) &= -s_R(t)\cos(2\pi f_0 t)\sin(2\pi f_c t) + s_I(t)\sin(2\pi f_0 t)\sin(2\pi f_c t) \end{cases} \tag{1.2}$$

And therefore,

$$\Rightarrow \left\{ \begin{array}{rcl} \widetilde{r_R}(t) & = & \frac{s_R(t)}{2}[\cos(2\pi(f_0 - f_c)t) + \cos(2\pi(f_0 + f_c)t)] \\ & & -\frac{s_I(t)}{2}[\sin(2\pi(f_0 + f_c)t) + \sin(2\pi(f_0 - f_c)t)] \\ \widetilde{r_I}(t) & = & -\frac{s_R(t)}{2}[\sin(2\pi(f_0 + f_c)t) - \sin(2\pi(f_0 - f_c)t)] \\ & & +\frac{s_I(t)}{2}[\cos(2\pi(f_0 - f_c)t) - \cos(2\pi(f_0 + f_c)t)] \end{array} \right. \tag{1.3}$$

Our objective is to demodulate the signal to obtain the same information than the one which was sent initially ($r_R(t) = s_R(t)$ and $r_I(t) = s_I(t)$). To do that, we are going to make a representation of $\tilde{R_R}(f)$ and $\tilde{R_I}(f)$, after using the Fourier transforms.

We got the following hypothesis $f_c = f_0$, and $2f_0 = F$ thus,

$$\Rightarrow \left\{ \begin{array}{rcl} \widetilde{r_R}(t) & = & \frac{1}{2}s_R(t)[\cos(2\pi Ft) + 1] - \frac{1}{2}s_I(t)[\sin(2\pi Ft)] \\ \widetilde{r_I}(t) & = & -\frac{1}{2}s_R(t)[\sin(2\pi Ft)] + \frac{1}{2}s_I(t)[-\cos(2\pi Ft) + 1] \end{array} \right. \tag{1.4}$$

$$H(\omega) = \left\{ \begin{array}{l} 2 \text{ for } |f| \leq f_c \\ 0 \text{ for } |f| > f_c \end{array} \right. \text{with } f_c \geq \frac{B}{2} \tag{1.5}$$

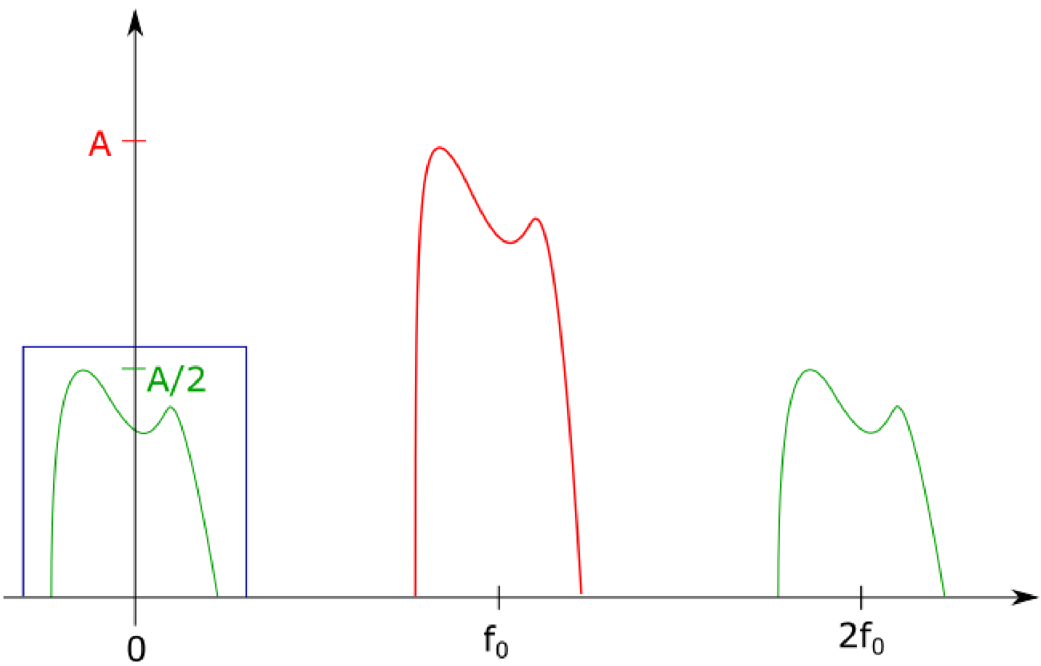

Figure 1.2: Signal demodulation

Using the receiver presented in Fig. 2, we have to be careful about the bandwidth of signals. In fact, let B the bandwidth of the signal, if $\frac{B}{2} \geq f_0$, then we are going to face spectral problems as you can see in the picture below:

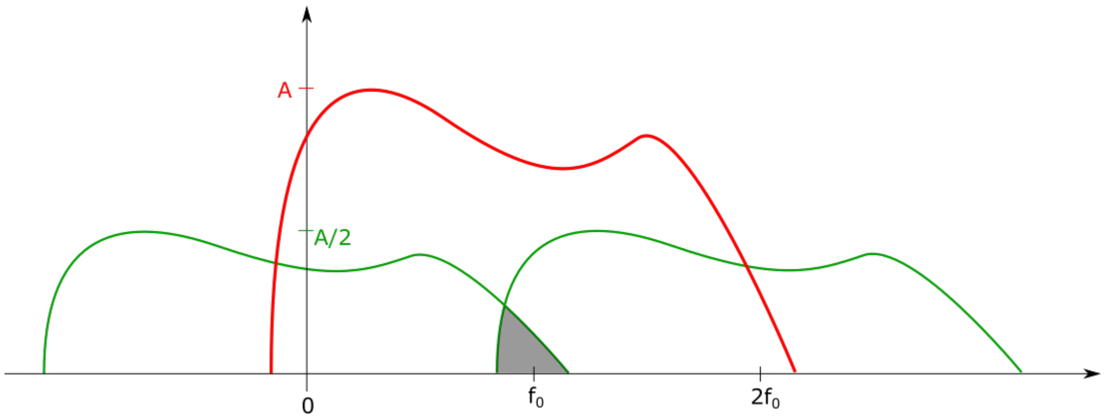

Figure 1.3: Spectral recovering when $\frac{B}{2} \geq f_0$

The grey part of the spectrum corresponds to a spectral recovering due to too large bandwidth compared to the central frequency f_0 of the signal.

To choose the sampling method T_e in order to recover $r_R(t), t \in \mathbb{R}$ from $r_R(k.T_e), k \in \mathbb{Z}$, we use the Shannon theorem which says that $f_e \geq 2 * f_{max}$, and we obtain:

$$f_e \geq 2 \times \frac{B}{2} = B \Rightarrow T_e \leq \frac{1}{B} \text{ if } B \neq 0 \tag{1.6}$$

It is important to note that we work with frequency transposition to use lower frequencies and consequently cheaper hardware. In fact, hardware able to work with high frequencies is really expensive because it needs high precision.

Supposing a real narrow-band signal $s_{RF}(t) = A(t).cos(2.\pi.f_0.t + \varphi(t)), t \in \mathbb{R}$, its analytic signal and its complex envelop in function of f_0 (knowing that $S_{RF}(f) = S^*_{RF}(-f)$) are:

$$s_{RF}(t) = s_R(t)\cos(2\pi f_0 t) - s_I(t)\sin(2\pi f_0 t) \tag{1.7}$$

$$\begin{aligned}
S_{RF} &= \mathcal{F}\{s_{RF}\}(f) && (1.8)\\
&= S_R(f) * \frac{1}{2}|\delta(f - f_0) + \delta(f + f_0)| + S_I(f) * \frac{j}{2}|\delta(f - f_0) - \delta(f + f_0)| && (1.9)\\
&= \frac{1}{2}[S_R(f - f_0) + S_R(f + f_0)] + \frac{j}{2}[S_I(f - f_0) - S_I(f + f_0)] && (1.10)
\end{aligned}$$

$$\begin{aligned}
S_a(f) &= S_{RF}(f) + j(-j \times sign(f)S_{RF}(f)) && (1.11)\\
&= S_R(f - f0) + S_R(f + f_0) + jS_I(f - f_0) - jS_I(f + f_0) && (1.12)\\
&= S_R(f - f_0) + jS_I(f - f0) \text{ because } S_R(f + f_0) = S_I(f + f_0) = 0 && (1.13)
\end{aligned}$$

Thus,

$$S_a(f) = \begin{cases} 2S_{RF}(f) \text{ for } f \geq 0 \\ 0 \text{ for } f < 0 \end{cases} \tag{1.14}$$

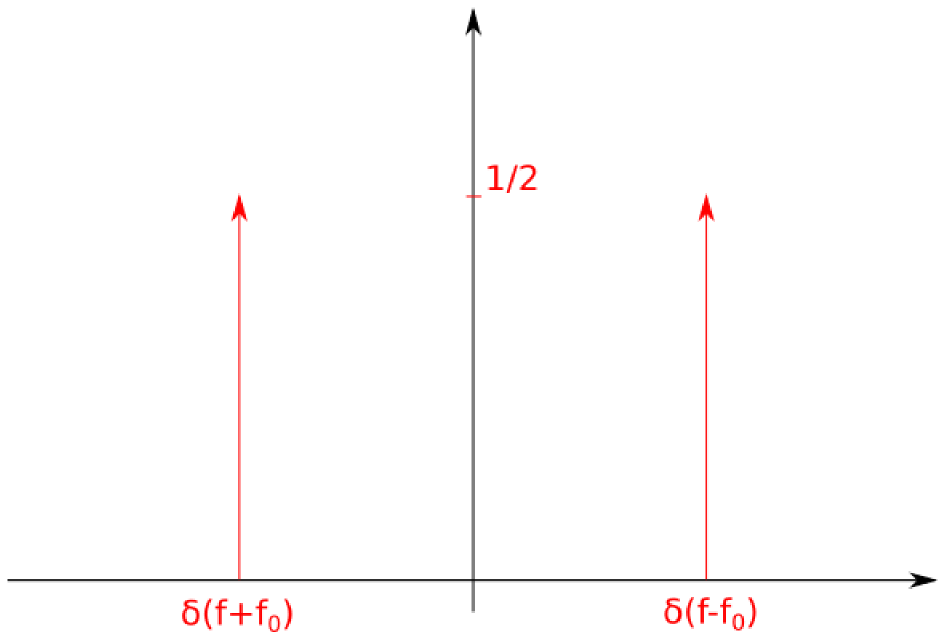

Figure 1.4: Fourier transform of the signal

Chapter 2

Reception of frequency modulation (FM) broadcasting

2.1 Frequency analysis of the recording

We are now going to receive a FM signal, and analyze it with the GNURadio Companion software. Here is a screenshot of the interface of the software:

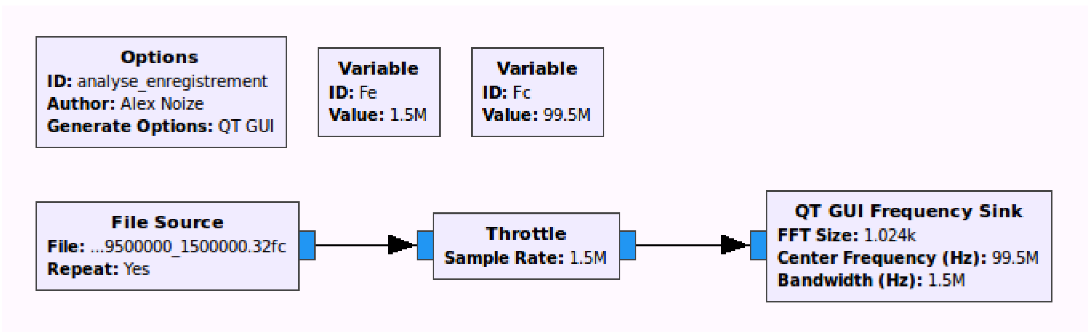

Figure 2.1: GNURadio Companion processing chain for recording analysis

We are going to briefly analyze the main blocks used in the processing chain above.
First, the *Options* block is mandatory, and sets the properties of the flowgraph.
Then, each *Variable* block is used to create a variable with a name and a value.
After that, the *Throttle* block throttles the flow of samples red from a data file with an average rate which does not exceed sample_per_sec. Finally, the *QT GUI Frequency Sink* block displays a graphical interface with the analyzed signals.

The recording red with the *File Source* block was sampled with a sample frequency F_e of 1.5 MHz. Moreover, the central frequency F_c of the signal to analyze is 99.5 MHz. That's why the variables F_e and F_c are set with these values.
Also, the *Throttle* block has a sample rate of 1.5 MHz, which is more exactly the value of F_e that we set as parameter.
Finally, the *QT GUI Frequency Sink* block is set with the center frequency F_c, and a bandwidth of 1.5 MHz. The bandwidth was chosen using the Shannon theorem: the sampling frequency F_e has to be

greater or equal to the double of the maximum frequency ($F_e \geq 2$ * F_{max}), since we transpose the signal around the /! JE NE SAIS PLUS POURQUOI, IL FAUT L'ECRIRE... /!

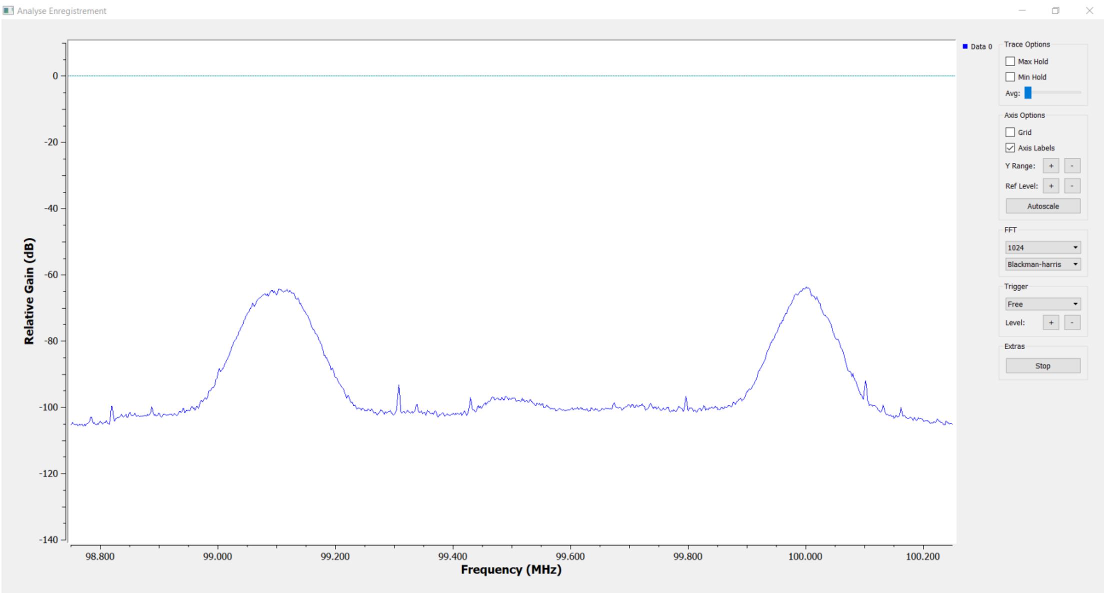

Figure 2.2: Signal obtained by recording analysis

On the frequency analysis above, we can see three frequency channels:

- L_1 = 99,1 MHz, which corresponds to RFM Toulouse
- L_2 = 100,0 MHz, which corresponds to Skyrock
- L_3 = 99,5 MHz, which corresponds to Nostalgie

The Signal-Noise Ratio (SNR) of a signal is the ration between the maximum amplitude of the signal and the noise measured. It characterizes the quality of the data transmission.
For each one of these channels, we measured the SNR:

- L_1: SNR_1 = -65 - (-102) = 37 dB
- L_2: SNR_2 = -64 - (-102) = 38 dB
- L_3: SNR_3 = -97 - (-102) = 5 dB

To determine the value of the SNR for each frequency, we chose to take the maximum amplitude of the signal as explained before. But we could also take the effective power of the signal by taking only the half of the amplitude, and so estimate the SNR differently. This measure is less optimistic than our.
Taking the half of the maximum amplitude amounts to subtract 3 dB to the maximum amplitude (-3 dB $\Leftrightarrow \div 2$).

We made the choice to continue working with L_1 and L_2 only because the SNR of the L_3 channel was too bad to obtain good results.
Thanks to the signal analysis, we determined the bandwidth of the two channels, which are:

- L_1: 100 kHz
- L_2: 131 kHz

These measures were took at half the maximum amplitude of the signal at channel's central frequency.

2.2 Channel extraction by frequency transposition and low-pas filtering

The next objective is to demodulate the signal. To do that, we have to transpose the signal around the 0 Hz frequency. Then, we apply a low-pass filter to mute the noise and only listen to the relevant part of the signal.

To center each channel, we have to apply an offset which depends on the central frequency of the channel and on the central frequency of the signal. Consequently, the offset is calculated as following: $offset = F_c - F_{c_{channel}}$. These offsets are:

- 99,5 MHz - 99,1 MHz = +400 kHz for L_1
- 99,5 MHz - 100,0 MHz = -500 kHz for L_2

If the frequency offset is higher than the sampling frequency F_e, we trying to retrieve sample that we do not have and ,thus, it will not work.

The low-pass filter parameters are the following :

- Decimation = 6
- Gain = 1
- Sample rate = F_e
- Cut-off frequency = $\frac{F_{chan}}{2}$
- Transition width = $0.1 \times \frac{F_{chan}}{2}$

2.3 Frequency demodulation and restitution

In the previous part, we measured a bandwidth of 100 kHz.
The Carson rule says that the bandwidth is equal to twice the addition of the maximum frequency excursion of the modulation and the maximum frequency of the composite signal $m(t)$ ($B_{FM} = 2 * (\delta f + f_m)$.
Let's check if the measured bandwidth is relevant:

$$B_{FM} = 2 * (\delta f + f_m) \tag{2.1}$$

$$B_{FM} = 2 * (75.10^3 + 53.10^3) \tag{2.2}$$

$$B_{FM} = 256kHz \tag{2.3}$$

The result obtained shows that the measured bandwidth was not really relevant, and that we have to change f_{canal} to $f_{canal} = 250kHz$.

From the expression (10) of the transmitted signal and the affected processes until now (frequency transposition and low-pass filtering), we want to show that the signal $y_l[k]$ can be noted $y_l[k] = A.e^{j.k_f \cdot \sum_{i=0}^{k} m[i]} + b[k]$:

$$s_{RF}(t) = A.\cos(2\pi f_0 t + \varphi(t)) \text{ with } \varphi(t) = \frac{\Delta f}{max(|m(t)|} \int_{-\infty}^{t} m(u)\,\mathrm{d}u \tag{2.4}$$

We got,

$$\begin{cases} s_R(t) &= A(t)\cos(\varphi(t)) \\ s_I(t) &= A(t)\sin(\varphi(t)) \end{cases} \tag{2.5}$$

Since,

$$\begin{aligned} s_{RF}(t) &= s_R(t)\cos(2\pi f_0 t) - s_I(t)\sin(2\pi f_0 t) && (2.6)\\ &= A(t)\cos(\varphi(t))\cos(2\pi f_0 t) - A(t)\sin(\varphi(t))\sin(2\pi f_0 t) && (2.7)\end{aligned}$$

We have,

$$\begin{aligned} s(t) &= s_R(t) + j s_I(t) && (2.8)\\ &= A(t)\cos(\varphi(t)) + jA(t)\sin(\varphi(t)) && (2.9)\\ &= A\,\mathrm{e}^{j\varphi(t)} && (2.10)\\ &= A\,\mathrm{e}^{j\frac{\Delta f}{max(|m(t)|}\int_{-\infty}^{t} m(u)\,\mathrm{d}u} && (2.11)\end{aligned}$$

We discretize our signal,

$$s[k] = A[k]\,\mathrm{e}^{j\frac{\Delta f}{max(|m[k]|}\sum_{i=0}^{k} m[i]} + b[k] \tag{2.12}$$

By identifying with $y_l[k] = A(t)\,\mathrm{e}^{jk_f \sum_{i=0}^{k} m[i]}$, we got

$$k_f = \frac{\Delta f}{max(|m[k]|} \tag{2.13}$$

Let's now have a look to the demodulated Skyrock channel (L_2):

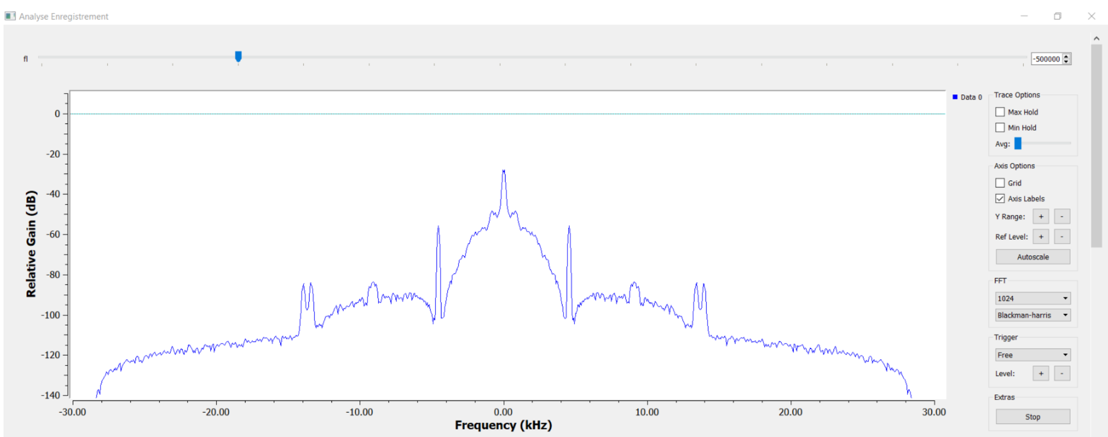

Figure 2.3: Skyrock channel demodulated

Listening to the Skyrock frequency (L_2), we can hear the radio host announcing to Jordi that he won the Sam Smith album and 50€.

2.4 Real time implementation with an USRP receiver

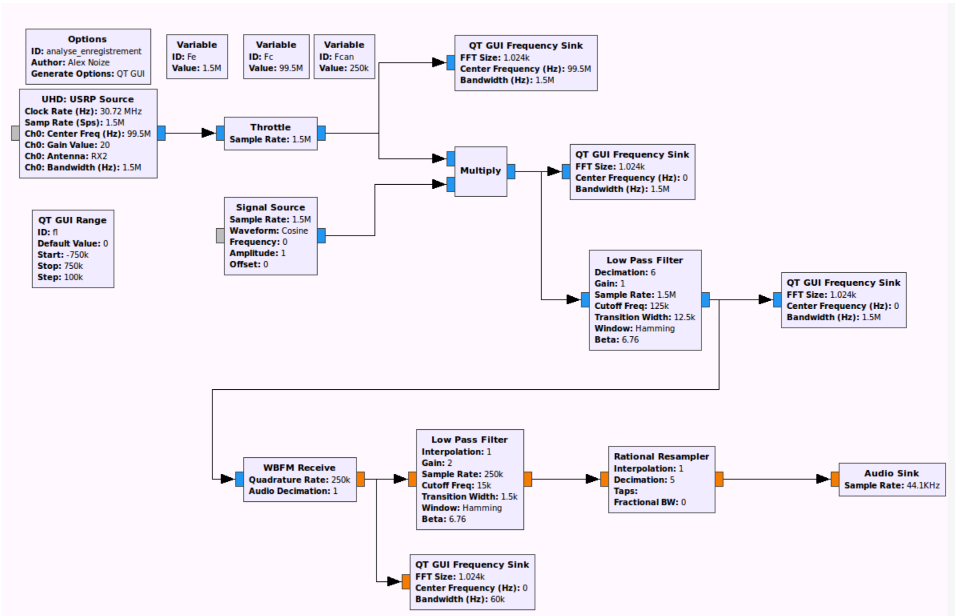

Figure 2.4: Real time implementation

We implemented the previous architecture and wioth use of an USRP we haven been able to demodulate radio signals in real-time. We were able to listen to the radio channel previously listed.

Chapter 3

Reception of VOLMET messages in AM-SSB

3.1 Frequency analysis of the recording

We analyzed a VOLMET recording, and this is the result we obtained:

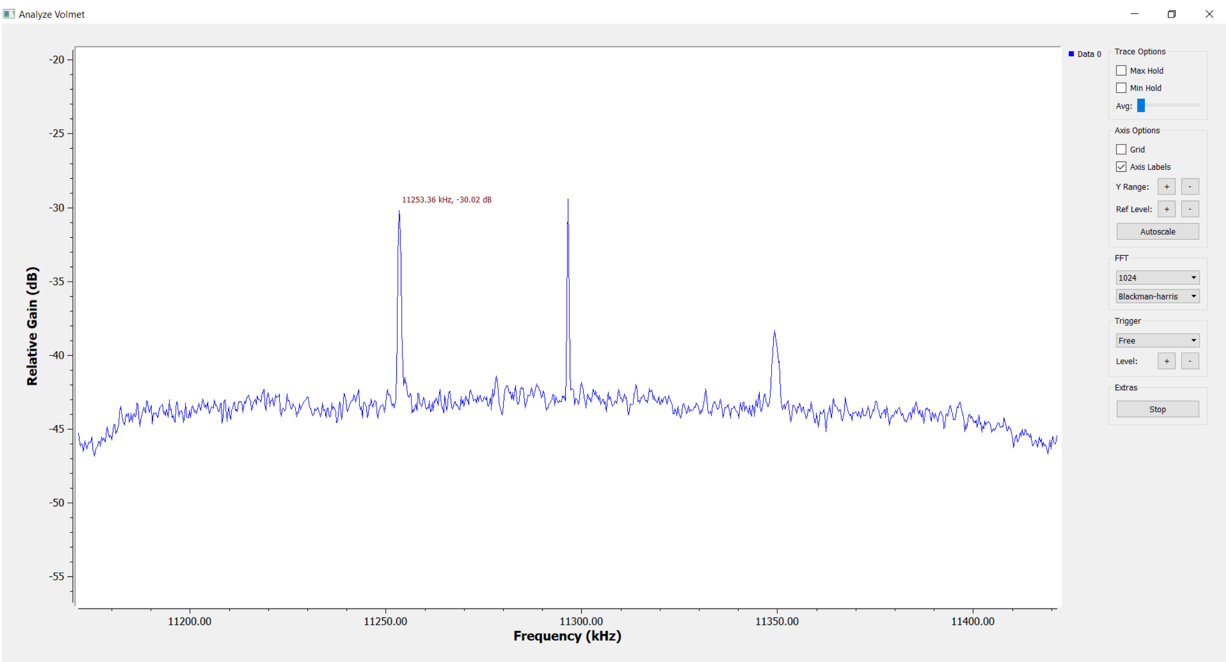

Figure 3.1: Frequency analysis of the VOLMET signal

On the spectrum obtained, we can see two peaks. The left one corresponds to the Royal Air Force VOLMET channel, which is at 11.253 MHz frequency.

11.253	*Cont*	MKL	GBR	ST EVAL (MILITARY ONE)	4000	50 28 58	-5 00 00

Figure 3.2: VOLMET frequency of the Royal Air Force

3.2 Frequency transposition

Now, we have to transpose the signal to demodulate the required frequency. To do that, we computed the offset to apply to the signal to center the relevant frequency: $offset = F_c - F_{RAF} = 11.2965MHz - 11.253MHz = 43.5kHz$

Here is the plotted modulus of the discrete Fourier transform in dB between $f_0 - \frac{f_e}{2}$ and $f_0 + \frac{f_e}{2}$:

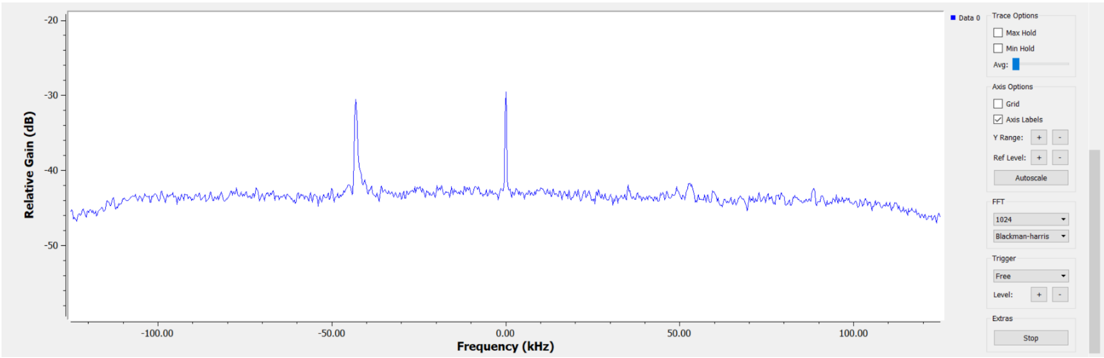

Figure 3.3: Discrete Fourier transform of the Volmet signal transposed at 43.5 kHz

3.3 Single sideband amplitude demodulation

The transmission process used by the VOLMET service is a single sideband amplitude modulation. First, we are going to study the VOLMET signal to apply the relevant filter.

At the beginning, we compute the module of the VOLMET signal to identify which part we want to filter.

$$s_{RF}(t) = Re(s(t).e^{j2\pi f_0 t}) = Re(m(t) \pm jH(m(t))e^{j2\pi f_0 t}) \tag{3.1}$$

with,

$$S_A(f) = S_{RF}(f) + sign(f)S_{RF}(f) \tag{3.2}$$

$$\begin{aligned} S_{RF}(f) &= \mathcal{F}[s_{RF}](f) && (3.3)\\ &= M(f) * \delta(f - f_0) && (3.4)\end{aligned}$$

$$\begin{aligned} S_A(f) &= M(f) * \delta(f - f_0) + sign(f)(M(f) * \delta(f - f_0) && (3.5)\\ &= \delta(f - f_0) * (M(f) + sign(f)M(f) && (3.6)\\ &= M(f - f_0) + sign(f - f_0)M(f - f_0) && (3.7)\end{aligned}$$

$$\begin{aligned} S_{RF}(f) &= \frac{1}{2}(S_A(f) + S_A^*(f) && (3.8)\\ &= \frac{1}{2}(M(f - f_0) + sign(f - f_0)M(f - f_0) + M(-f - f_0) + sign(-f + f_0)M(-f - f_0)) && (3.9)\end{aligned}$$

If $f - f_0 > 0$,

$$S_{RF}(f) = M(f - f_0) \tag{3.10}$$

If $f - f_0 < 0$

$$S_{RF}(f) = M(-f - f_0) \tag{3.11}$$

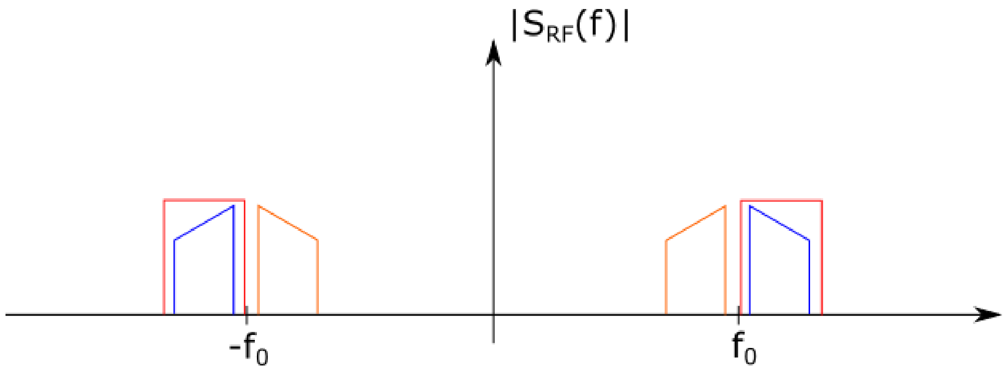

Figure 3.4: VOLMET signal received

We see that our bandwidth is 3 kHz (11.256 MHz - 11.253 MHz), so we only have half the initial bandwidth ($\frac{B}{2}$), and we only have the positive part of the spectrum.

We designed this filter, and this are its modulus and its phase:

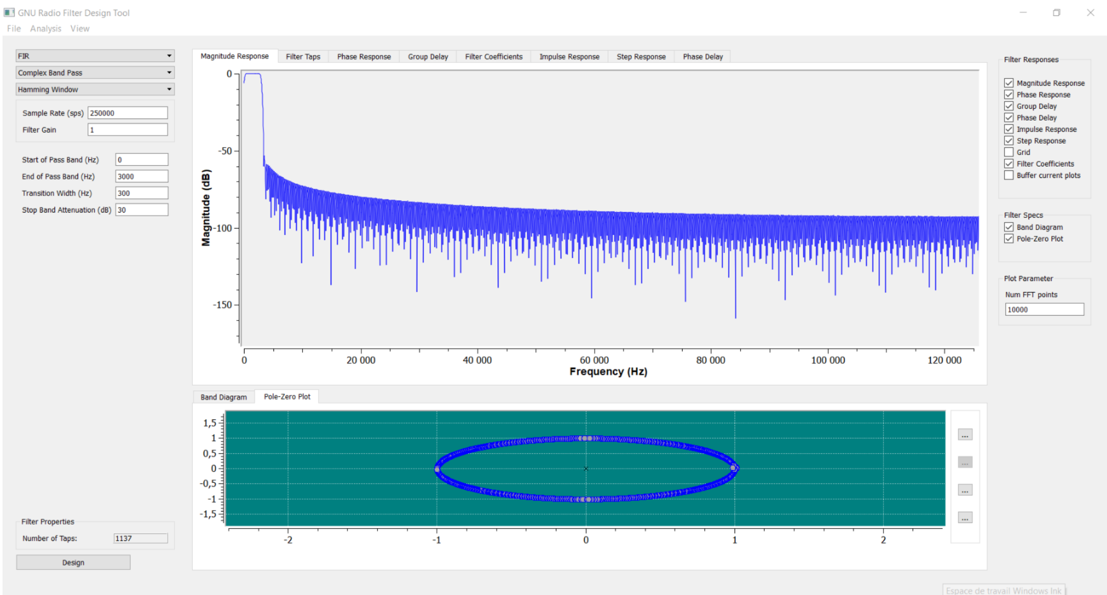

Figure 3.5: Modulus and phase of the designed pass-band filter

We have now to filter the $r_1[k]$ signal with a pass-band filter to obtain as output $y[k]$ which corresponds to the single upper sideband of the amplitude modulated signal (blue signal on the right on Figure 3.4).

Figure 3.6: Pass-band filter applied to $r_1[k]$ to obtain $y[k]$

The filter applied to the signal corresponds to the red filter on the right of Figure 3.4, and is a pass-band filter acting between f_0 and $f_0 + 3kHz$.

We finally tried to recover the real signal to the form $\tilde{m}[k] = m[k] + \tilde{b}[k]$:

$$r_1[k] = m[k] + jH(M[k]) + b[k] \tag{3.12}$$

$$\begin{aligned} \tilde{m}[k] &= Re(r1[k]) &(3.13)\\ &= m[k] + \tilde{b}[k] \text{ with } \tilde{b}[k] = Re(b[k]) &(3.14) \end{aligned}$$

We implemented the following block chain:

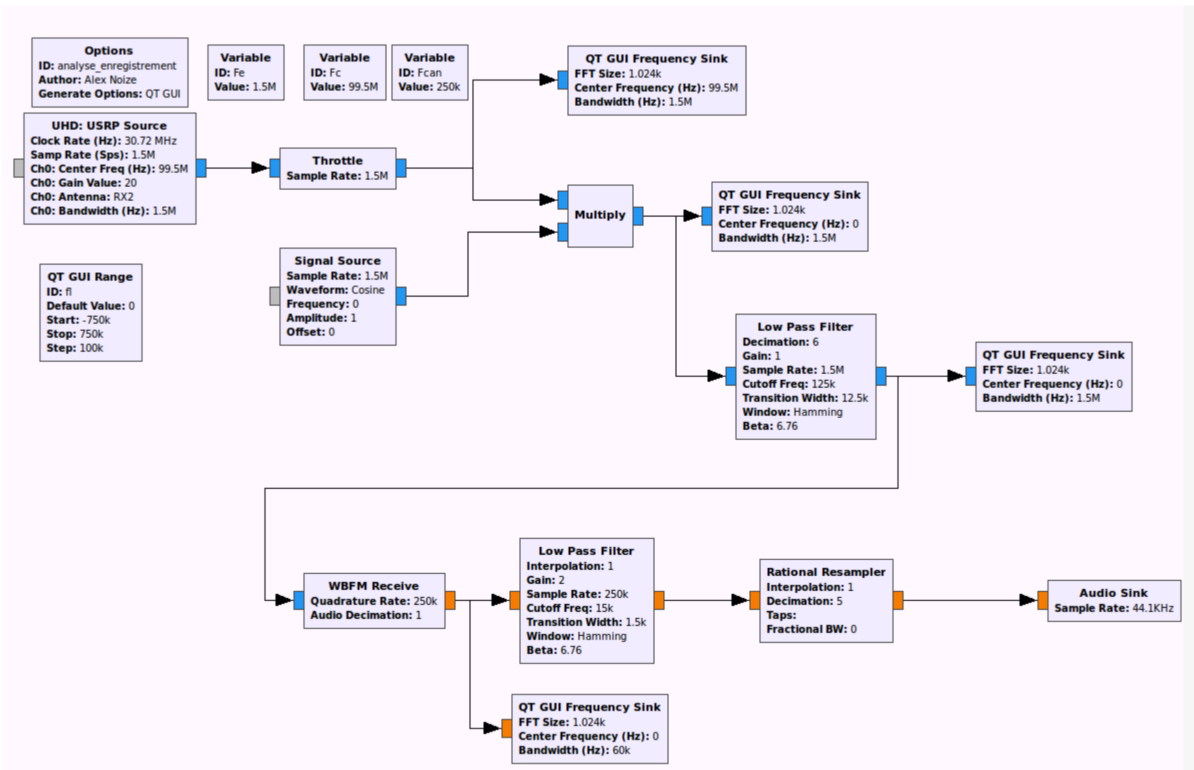

Figure 3.7: USRP received VOLMET demodulation block chain

But we faced an issue: the maximum receiving frequency of the USRP is 70 MHz, and real-time VOLMET signals are sent between 3 MHz and 17 MHz. Consequently, we were unable to receive a VOLMET signal and test our block chain in real conditions.

Chapter 4

Conclusion

In this lab, we have been introduced to the concept of software defined radio. We have first mathematically proved that we can demodulate our signals thanks to an IQ transceiver. Then, we have work on frequency modulation (FM) in order to demodulate radio signals. We could perform real time demodulation and, thus, listen to radio channels. Finally, we worked on amplitude demodulation and VOLMET messages but we couldn't perform real time decoding because our USRP did not cover this band.

Moreover, we could appreciate the strengths and the power of both USRP and software defined radio. A wide variety of applications can be implemented using the same hardware parts and only change the software part. This key advantage is particularly interesting for IoT technologies , indeed, the technologies are evolving really fast in this domain and new standards are set everyday. Thanks to SDR and USRP, you can update the your communication part without changing any part of the hardware and, thus, save a lot of money, time and energy.