M1-IRELE

**ELEC-H401 Modulation and coding**

# DVB-C project

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**Academic year :**

2024 - 2025

# Contents

# Introduction

This report aims to complete the code that simulates a DVB-C transmission chain in matlab. It provides additionnal information from the theoretical part of the project.
The first part aims to simulate each block of the transmission chain and to link them together such that the received signal is the same as the transmitted signal (in a noiseless case).

## Second part to be explained later

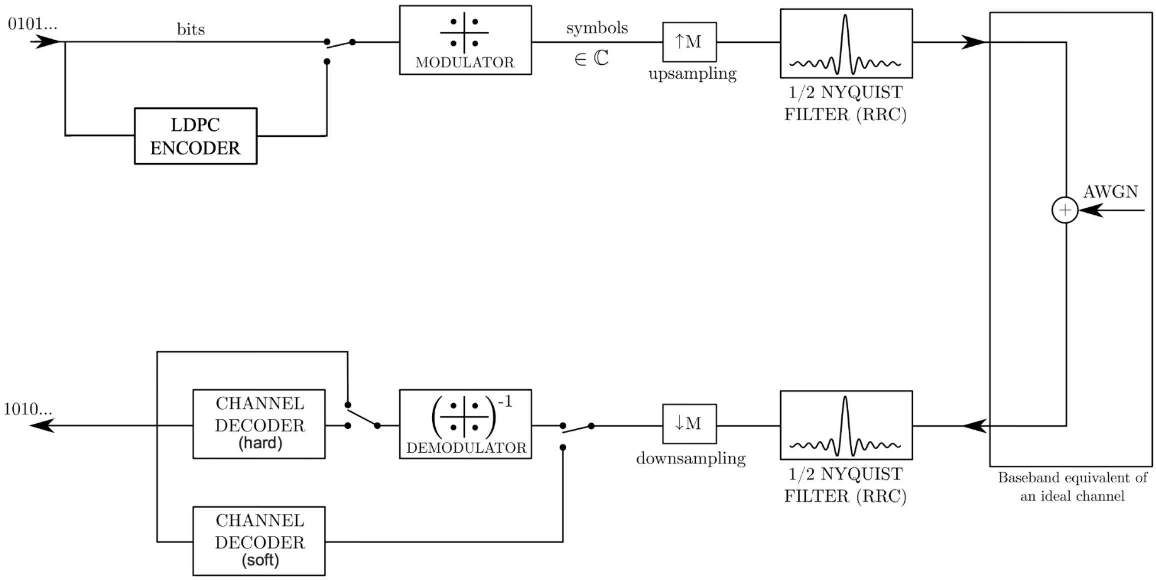



Figure 1: DVB-C transmission chain

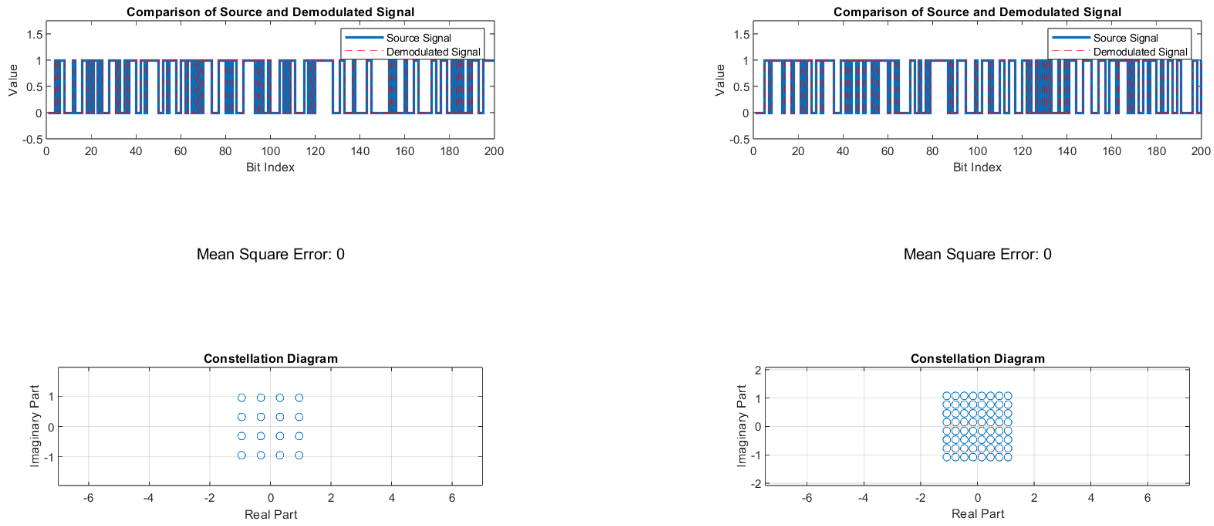
# Transmission chain blocks

## Baseband representation

By looking at the block diagram of the transmission chain 1, one can see we never move the baseband signal to the carrier frequency. As the simulation runs on a computer, using the bandpass representation of the signal would require much more samples as the sampling frequency would need to be at least twice the carrier frequency. By simulating the chain in baseband, the minimal sampling frequency is reduced to the symbol rate in order to have at least one sample per symbol.
Because the signal is oversampled, the sampling frequency is then equal to the symbol rate multiplied by the oversampling factor.

## Modulation and Demodulation

After generating N random bits, they are modulated. This allows to send fewer symbols than the number of bits. We chose QAM modulation as it combines ASK and PSK. Depending on the number of bits per symbol ($N_{\text{bps}}$), the number of bits sent ($N$) had to be chosen such that $N/N_{\text{bps}} \in \mathbb{N}$.
Figure 2 compares the modulation the constellation diagrams obtained for QAM-16 and QAM-64. As the constellations points are more spaced on the left, QAM-16 is less prone to a wrong demodulation (when noise will be added). This comes at the cost of a lower bitrate: for the same symbol rate, QAM-64 will send 6 bits while QAM-16 only send 4. It clearly shows a compromise between reliability and capacity.




(a) QAM-16 modulation

(b) QAM-64 modulation

Figure 2: Comparison of QAM modulations, where the mean square error is computed between the transmitted and received bitstream

## Pulse shaping

With modulation only, the bandwidth of the transmitted signal is infinite. This is problematic as it could interfere with neighboring channels. A filtering is applied to resolve this but the chosen filter must respect two other constraints: it must cancel inter-symbol interference (ISI) and must maximize the SNR.

The raised cosine filter is chosen as it limits the bandwidth and cancels ISI. To maximize the SNR, it is applied as a matched filter by using the square root of it at the transmitter and at the receiver.

The time domain and frequency domain representation of the raised cosine filter is shown in Figure 3. Figure 4 shows how the signal is shaped in the time domain and how there is indeed no ISI. Finally, the spectrum of the transmitted signal is plotted in figure 5 where the frequency band is limited to $[-3,3]$MHz.

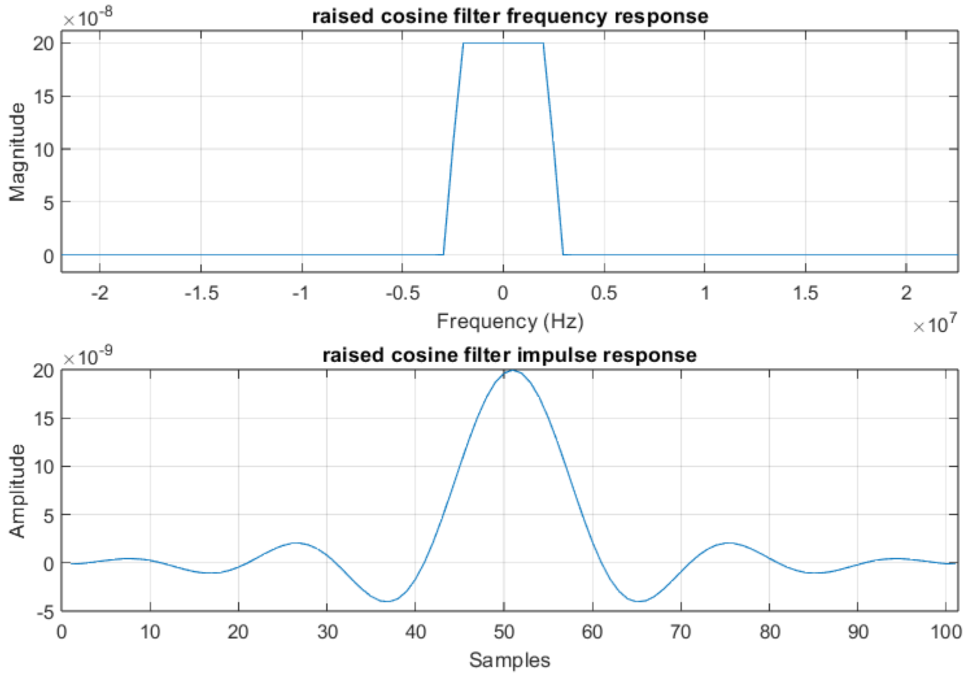



Figure 3: Time and frequency domain representation of the raised cosine filter

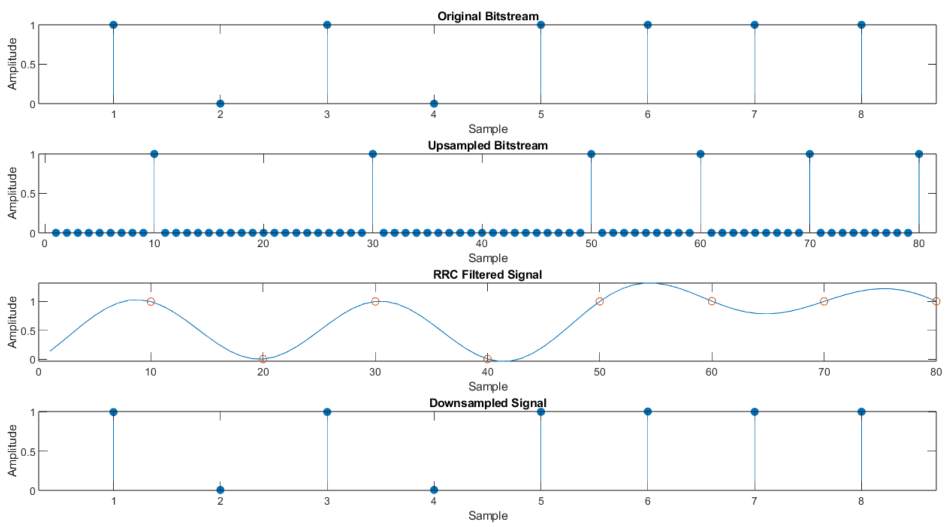



Figure 4: Pulse shaping with a raised cosine filter

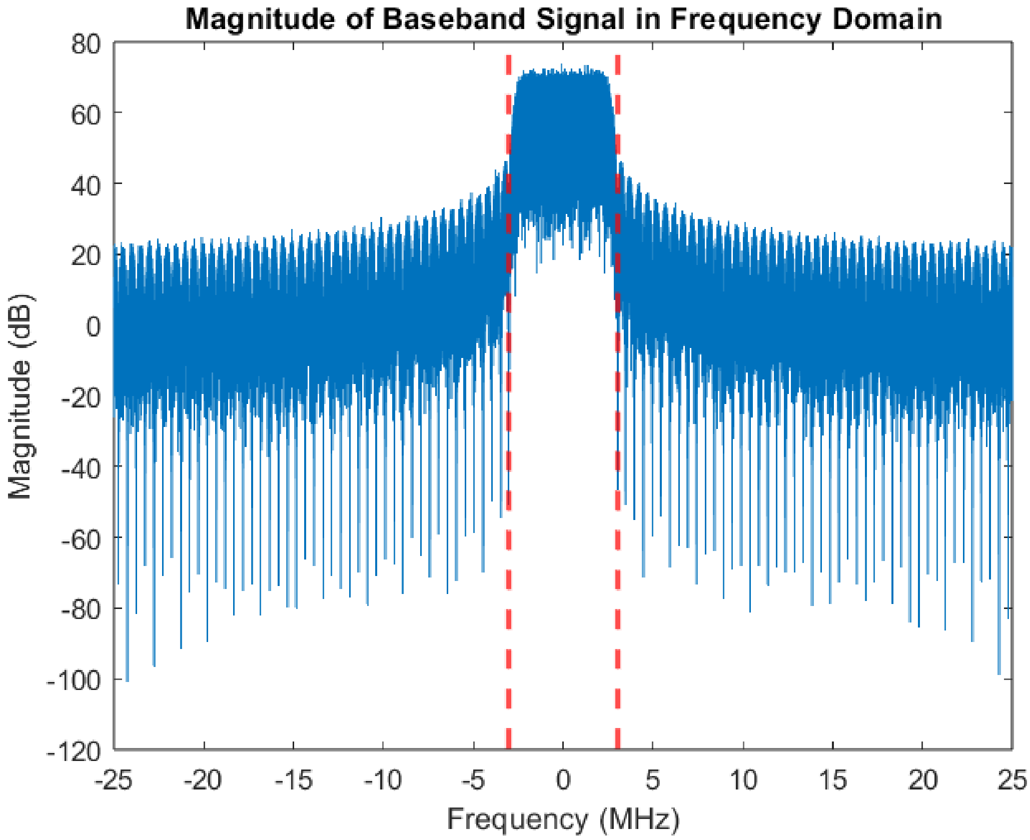



Figure 5: Spectrum of the transmitted signal after pulse shaping

## Noise addition

## About the simulation

Because the simulation runs on a computer, it has to store signals on discrete time intervals. Because of the Shannon theorem, the sampling frequency (or the inverse of the time interval) has to be at least twice larger than the highest frequency of the signal. This is why the whole transmission is simulated in baseband. Working in bandpass would require much more samples to store the same signal, leading to a longer simulation time and a larger memory usage.

The simulation sampling frequency has been chosen such that each transmitted symbol is sampled at least once. As a real communication channel is not always transmitting, each symbol is separated by some null symbols (*oversampling*). $f_s$ is then equal to the symbol rate multiplied by the oversampling factor.

Amaury : (*How do you make sure you simulate the desired $E_b/N_0$ ratio?*)

The number of transmitted packet and their length is chosen such that the simulation has enough samples to generate a reliable BER curve. If not correctly chosen, the BER would be less smooth.

## About the communication system