NOVA SCHOOL OF SCIENCE & TECHNOLOGY

MEEC/MIEEC

Analog Integrated Circuits

Analysis and design of a reference voltage buffer for an ADC

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3 Digital Communication

For this section two Quadrature Amplitude Modulation (QAM) techniques were used, **Quadrature Phase-Shift Keying (QPSK)** and **16-QAM**. This process involved generating a random stream of bits, modulating them into **QPSK** and **16-QAM** symbols and simulating their transmission through GNU Radio. In the simulation the effects of non-linear Power Amplifier (PA) and Low Noise Amplifier (LNA) were simulated as well as the noise of a Additive White Gaussian Noise (AWGN) channel.

3.1 Digital Modulation

QPSK places four equally spaced points on the unit circle:

$$s_k = e^{j\frac{\pi}{2}\left(k+\frac{1}{2}\right)}, \qquad k \in \{0,1,2,3\}.$$

Figure 1a, shows the mapping in the cartesian plane.

The mapper groups the encoded bit stream into two-bit tuples (b_1, b_0), converts each tuple to an integer index $(k = 2b_1 + b_0)$ and outputs s_k.

The theoretical bit-error probability for QPSK in an AWGN channel is given by Equation 1.

$$P_b^{\text{QPSK}} = Q\left(\sqrt{2\frac{E_b}{N_0}}\right) \quad [?] \tag{1}$$

For **QPSK**, demodulation is performed by simply de-mapping the bit values.

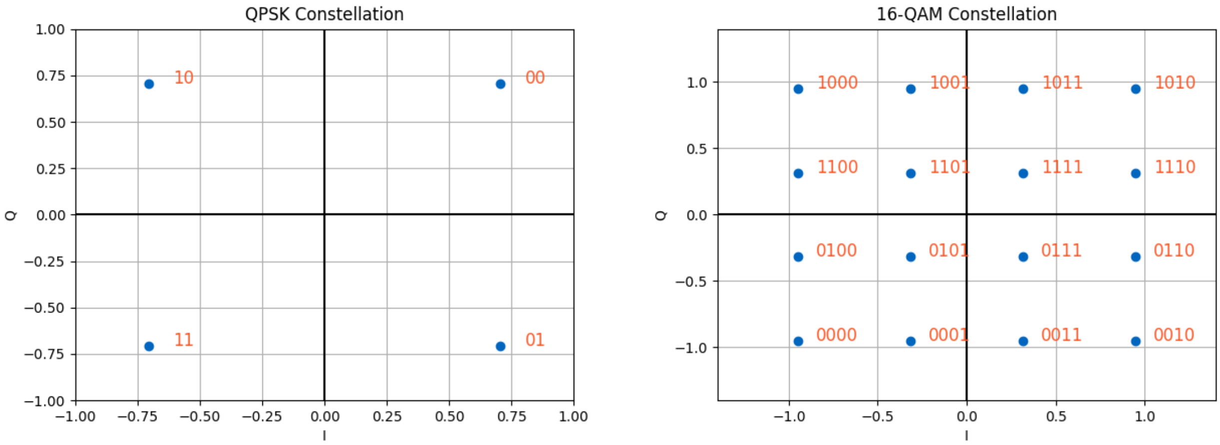
With **16-QAM** a 4×4 square constellation was used. What changes comparing to the previous mapping approach is the fact that the amplitude also changes and for this specific mapping the phase and amplitude will not change consistently. The symbol position in the cartesian frame will be:

$$I, R \in \{\pm 3,\ \pm 1\}$$

For **16-QAM** the theoretical **BER** for an AWGN channel with gray mapping is given by Equation 2.

$$P_b^{\text{16QAM}} \approx \frac{3}{4} \cdot Q\left(\sqrt{\frac{4}{5}\frac{E_b}{N_0}}\right) \quad [?] \tag{2}$$

The constellation points are labelled with *Gray coding*, thus every nearest neighbour differs in *exactly one* bit, this will minimize **BER**, since the most likely symbol error produces only one wrong bit. Figure 1b, shows how the codes are mapped.



(a) QPSK Constellation.

(b) 16-QAM Constellation.

Figure 1: Digital Modulation Constellations.

3.2 GNU Radio Implementation

The GNU Radio design aims to simulate an RF (radio frequency) QAM communication between a transmitter and a receiver through an AWGN channel. This simulation also includes non linear elements from the power amplifiers used in these circuits [Fig.7], and channel imperfections, such as signal attenuatin that occurs in the channel [Fig.6].

3.2.1 IQ Modulation

The transmitter modulates two different signals effectivly transfering the original signals from the original baseband to the channel frequency (Fc). These signals are modulated with a 90° angle phase shift between them [Fig.2 and 3]. This means the modulated signals are in quadrature with each other, thus allowing the transmitter to transmit both signals at the same time without them interfering with each other.

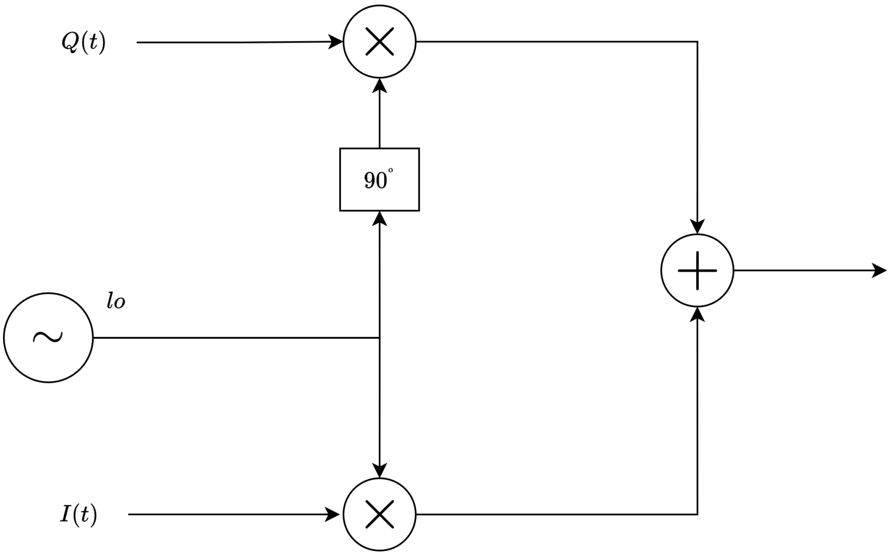


Figure 2: IQ Modulator Block Diagram

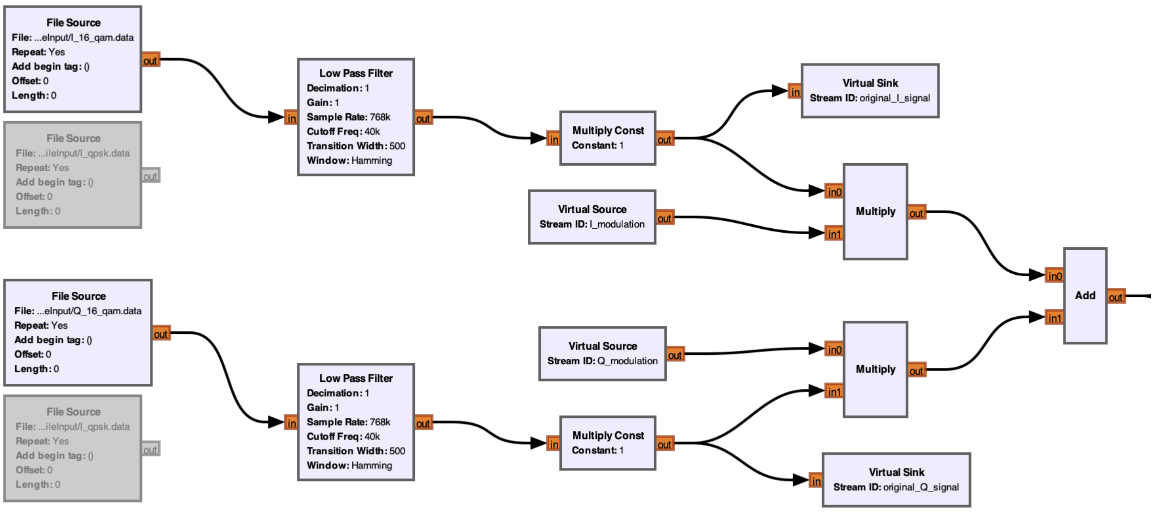


Figure 3: GNU IQ Modulator

On the receiver's side, the same method is applied to demodulate the received signal and recover both the original signals, [Fig.4 and 5].

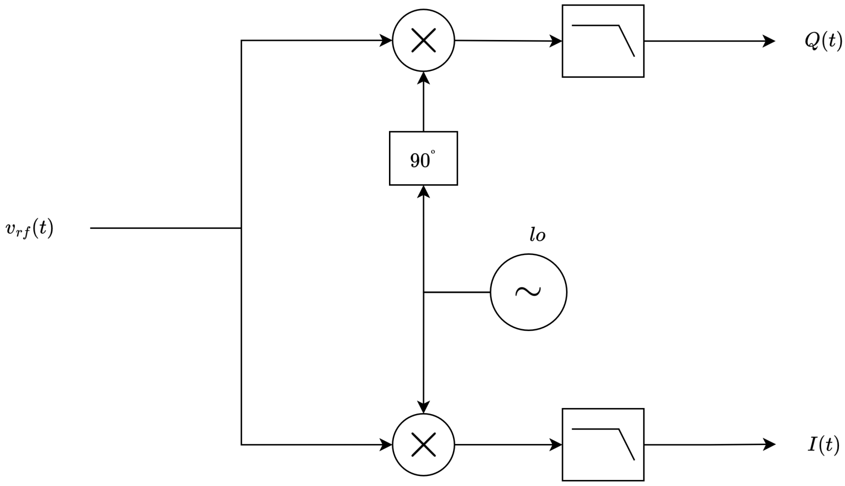


Figure 4: IQ De-Modulator Block Diagram

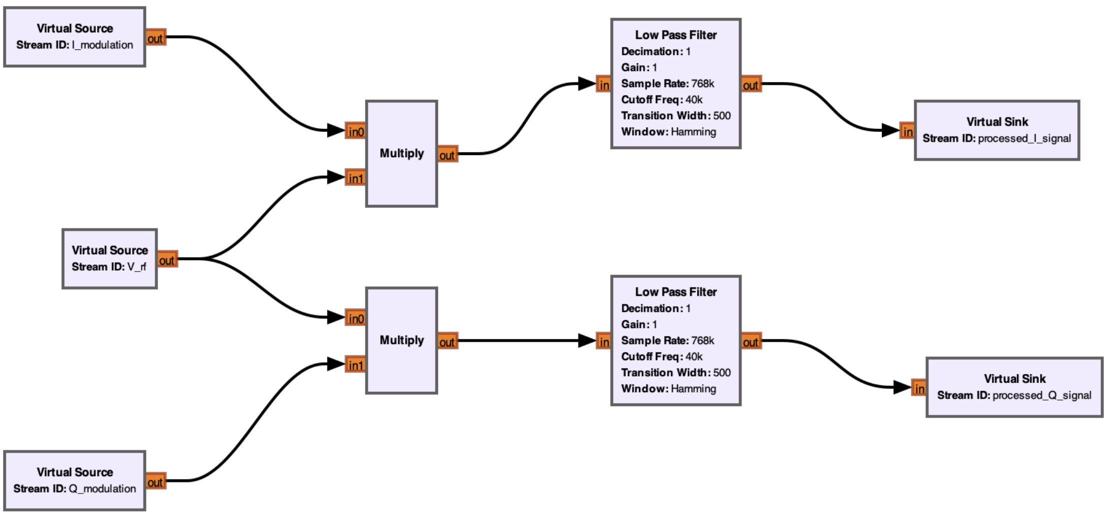


Figure 5: GNU IQ De-Modulator

Since the received signals are in quadrature with each other demodulating them with the same signal used on the transmitter, (a signal with the same frequency and pahse as used in the transmitter to modulate), the original signal is recovered.

3.2.2 Nonideal simulation elements

Arranjar outro titulo e texto a explicar 3 ordem de nao lin, limitacoes que vao apararecer e dizer que pro lNA é mais do mesmo

The cahnnel used for RF communications, in the real world, attenuates the transmitted signal and adds some white noise as well.

These effects are replicated in the simulation using a constant value multiplier in the channel with a constant value smaller than 1, and a adding a the transmitted signal to a signal produced by the signal generated by a white noise source, [Fig.6].

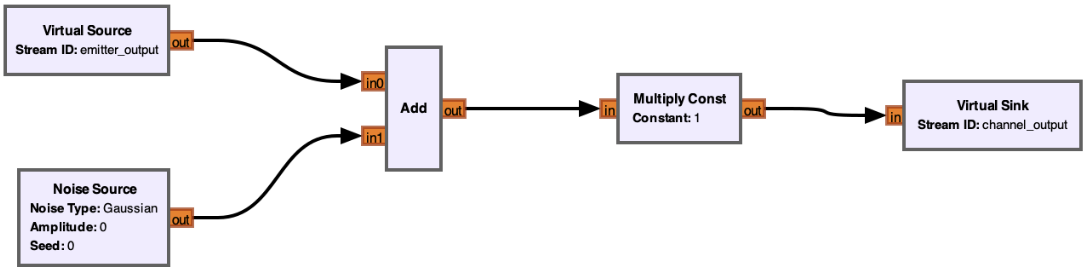


Figure 6: GNU Radio Channel

Dizer que o noise é adicionado antes da atenuaçao para manter o SNR mais facil de quantificar

To counteract the undesired effects of the channel an amplifier is introduced, both in the transmitter and the receiver.

Idealy these amplifiers would provide a linear gain to the signal, however the components used to create these amplifiers are ideal, in which case, the real amplifiers add a nonlinear component the signal gain.

Assuming the gain of these amplifiers can be expressed as a function of the input signal, then the output of these amplifiers can be described as $y(t) = Amp(x(t))$, then using taylor series' the output can be approximated to the result fo Equation 3.

$$y(t) = \sum_{n=1}^{\infty} \frac{\partial^n Amp(x_0)}{\partial x^n \cdot n} \cdot (x - x_0)^n \leftrightarrow y(t) \approx a_1 x(t) + a_2 x(t)^2 + a_3 x(t)^3 \tag{3}$$

Where x_0 is the dc operating point voltage of the input, was set to 0 to simplify calculations, a_n is the value of the nth derivative in respect to $x(t)$ evaluated at $x(t) = 0$ divided by the number of derivates taken.

To simplify the GNU Radio schematic only the 3 higher order Taylor series' components were used, these are also the most influencial components in the real circuit. The GNU Radio circuit is shown in Figure 7.

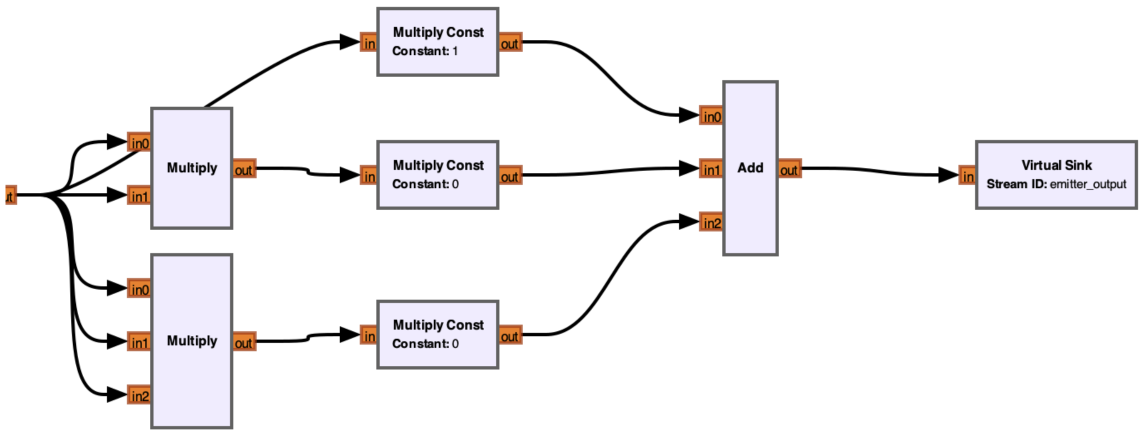


Figure 7: GNU Radio PA non Linear

3.3 Performance Analysis

The primary goal was to evaluate the system's performance by measuring the Bit Error Rate (BER) as a function of the Signal-to-Noise Ratio (SNR) in an AWGN channel.

A random bitstream of 3×10^6 bits was generated using `Gen_symbs.py` Meter cite aos anexos?. This stream was modulated and fed into the GNU Radio simulation. At the receiver, the `Read_Output.py` script was used to read the output files and calculate BER.

Finally with BER values were plotted against the theoretical performance curves, Equations 1 and 2, for both modulation schemes. As shown in Figure 8, in this figure there were no non-linear effects simulated.

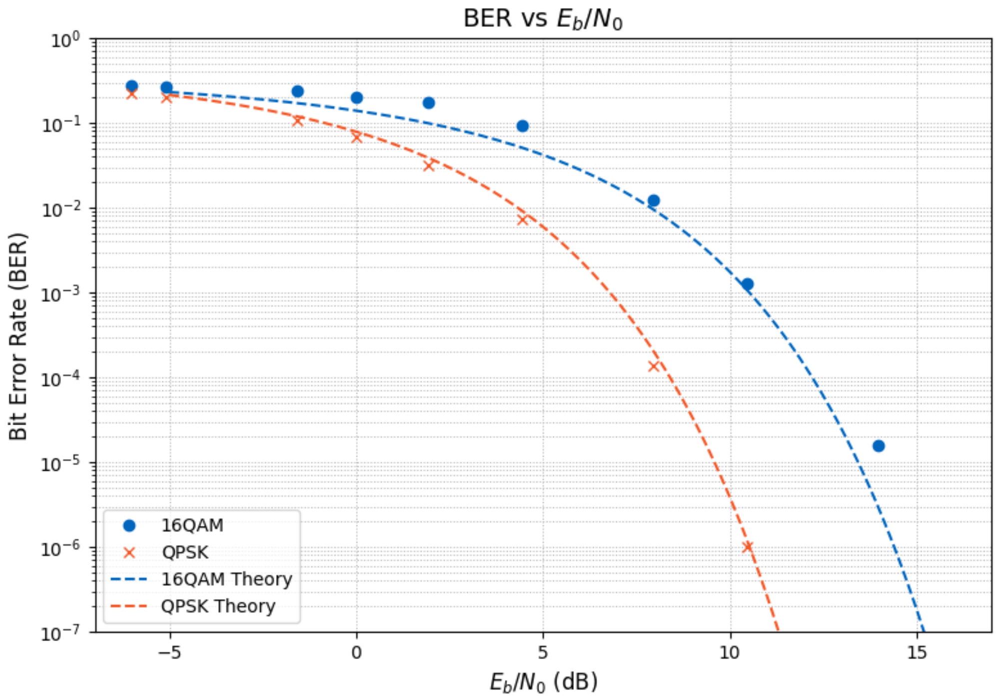


Figure 8: BER vs E_b/N_0

The results in Figure 8 clearly validate our simulation.

QPSK BER points (shown in orange) align almost perfectly with the theoretical QPSK performance curve. This confirms that the simulation chain, including the noise model and demodulator, is functioning correctly.

16-QAM similarly, the simulated 16-QAM data (in Blue) closely follows its theoretical curve.

4 SPICE simulation results and analysis

In this section, the circuit simulation of the AM Communication method demonstrated in Section **??** referenciar a secção do gnuradiowill be presented using Qucs-s.

4.1 Circuit analysis

In this simulation, the complete circuit consists on an AM Transmitter Block, a Channel Block, and finally the Receiver block as well. The simulated circuit can be found in Figure 9.

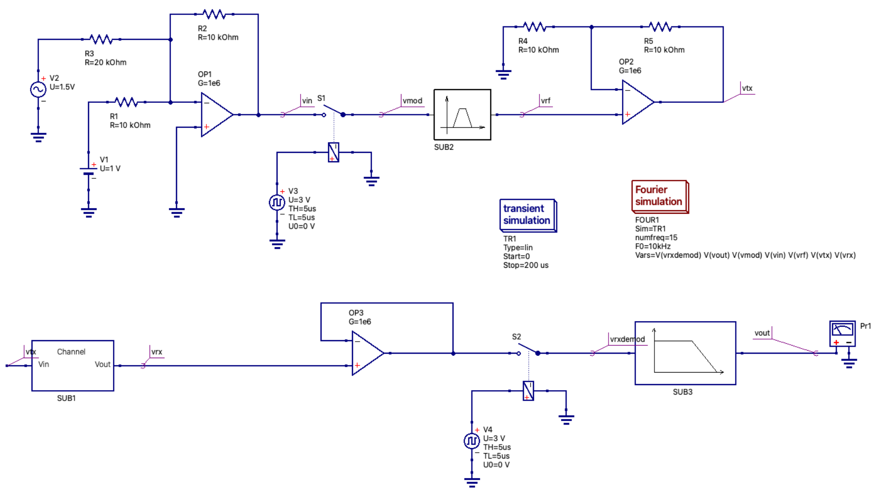


Figure 9: AM Communication circuit

The Transmitter block is implemented using, first a Summing OpAmp configuration, followed by one ideal switch, representing one Ideal Mixer, commuted by the Local Oscillator, with a frequency of $100kHz$, resulting in the modulation of the input signal.

Then to ensure that only the Base Band frequencies and the fundamental frequency of the carrier signal remain in the transmitted signal, an active Band Pass filter is used, with a central frequency corresponding to the carrier fundamental frequency, in this case $100kHz$. This filter was implemented using the Multiple-Feedback topology, resulting in the 4$^{\text{th}}$ order Bessel filter presented in Figure 10.

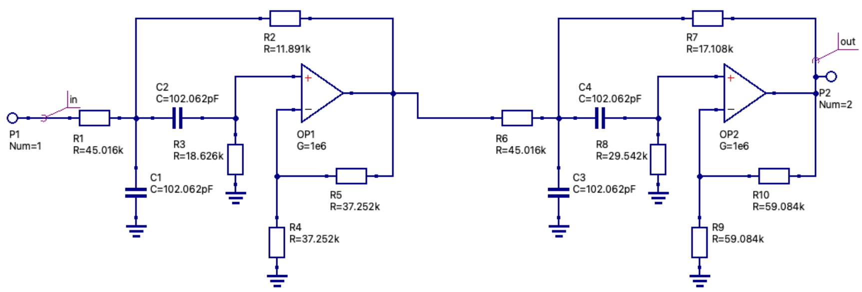


Figure 10: Band Pass filter circuit

The Bessel approximation was adopted to guarantee a constant group delay around the carrier frequency, avoiding signal distortion. Finally, the transmitted signal is amplified using a Power Amplifier, implemented using a Non-Inverter OpAmp circuit.

Following the Transmitter Block, the channel is simulated by attenuating the transmitted signal and adding noise to it.

Lastly, the signal is received by the Receiver Block, beginning with the Low Noise Amplifier, in this case, a Non-Inverter OpAmp with unitary gain, resulting in a Buffer configuration, followed by the step-down mixer, another Ideal Switch with the same frequency as the one present in the Transmitter Block.

At last, the signal passes through an active Low Pass Filter, isolating the baseband signal form the remaining images of the modulated signal created by the Mixer in the demodulation process. The Bessel active Low Pass Multiple-Feedback Filter can be seen in Figure 11.

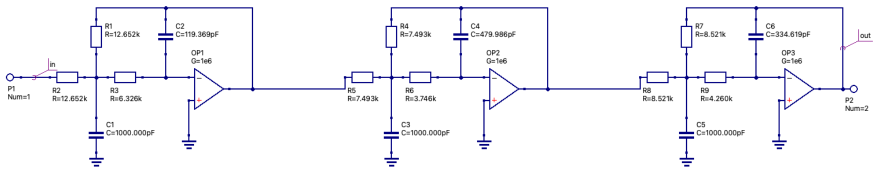


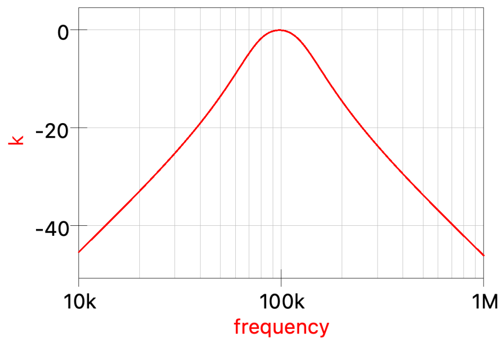
Figure 11: Low Pass filter circuit

The higher filter order, is a result of the proximity between the baseband and the carrier frequency, originating a small transition region between the passband and the stopband.

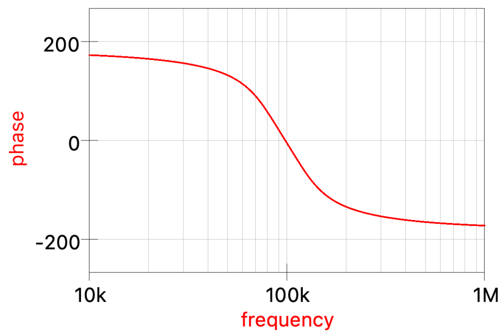
4.2 Results

4.2.1 Low Pass and Band Pass Filters

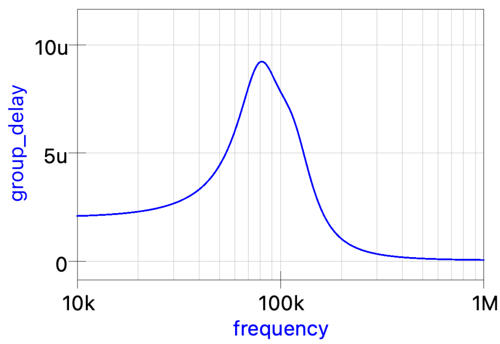
To measure the performance of the implemented filters, the frequency response was simulated, resulting in the Bode gain, phase and group delay graphics of both filters. First, the frequency response of the Band Pass Filter is presented in Figure 12



(a) Gain Bode Diagram.



(b) Phase Bode Diagram.

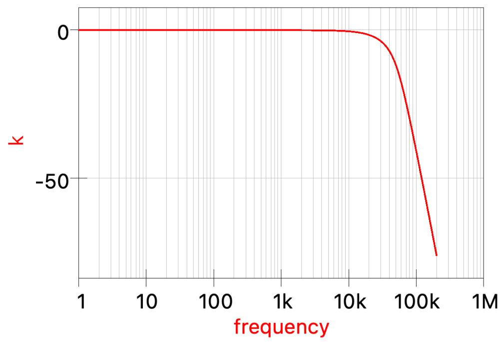


(c) Group Delay Bode Diagram.

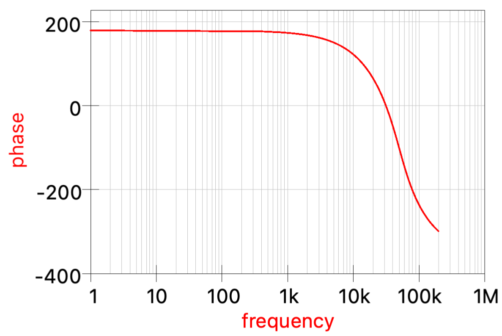
Figure 12: Band Pass Filter Bode Diagrams

Analyzing the Bode Diagrams of the Band Pass Filter, it is possible to conclude that the baseband signal around the carrier frequency will not be affected by the filter, but all frequencies of the Local Oscillator signal excepted the fundamental will be eliminated, as pretended.

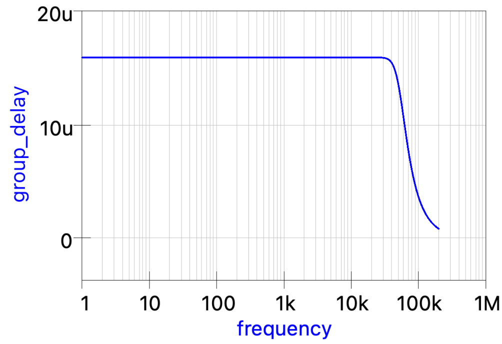
Passing to the Low Pass Filter, the frequency response of this filter is shown in Figure 13



(a) Gain Bode Diagram.



(b) Phase Bode Diagram.



(c) Group Delay Bode Diagram.

Figure 13: Low Pass Filter Bode Diagrams

Looking at the Bode Diagrams of the Low Pass Filter, the gain diagram shows that for the baseband signal, there is no attenuation, but the same cannot be said for the carrier frequency, so, the output of the filter will only be the desired signal.

4.2.2 Transmitter Block

Since this circuit involves signals with different frequencies and amplitudes, the best way to analyze the results of each component is through the use of the Fourier Transformation, allowing the analysis of the effect of all parts of the circuit for each frequency.

Starting with the mixer in the Transmitter, the Fourier Transform of the signal after the Mixer is shown in Figure 14

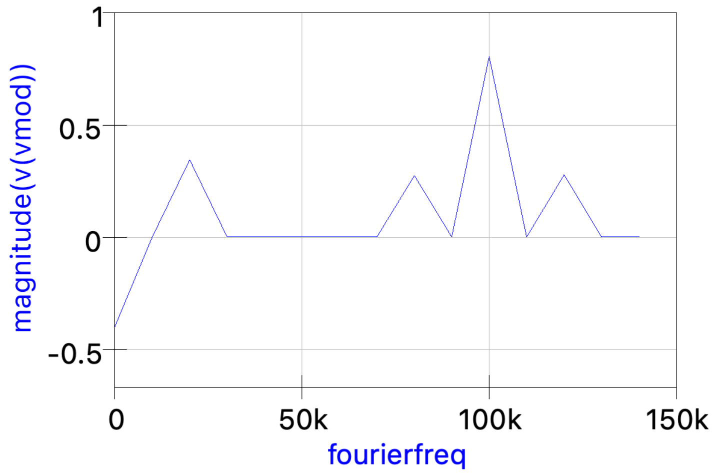


Figure 14: Fourier Transform of the Mixer output

In graphic above, is clear the presence of the carrier frequency and the baseband signal around and in its original location, so after the Band Pass Filter, the modulated signal is present in Figure 15.

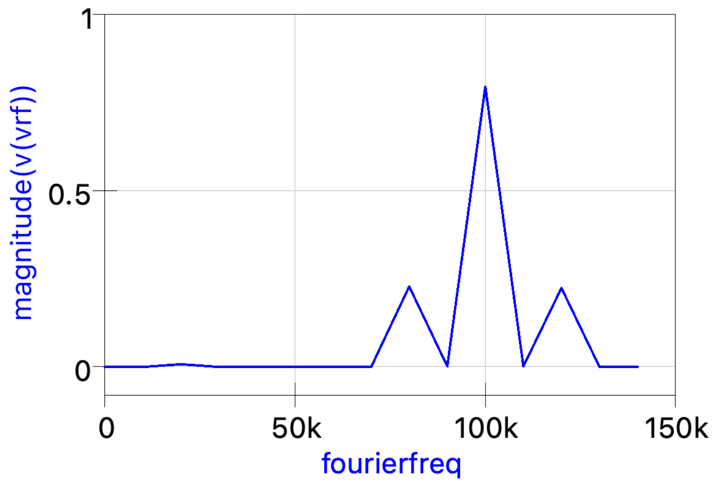


Figure 15: Fourier Transform of the Band Pass Filter output

Now only the desired band of frequencies is present in the signal. To confirm this fact, the signal in the time domain is displayed in Figure 16

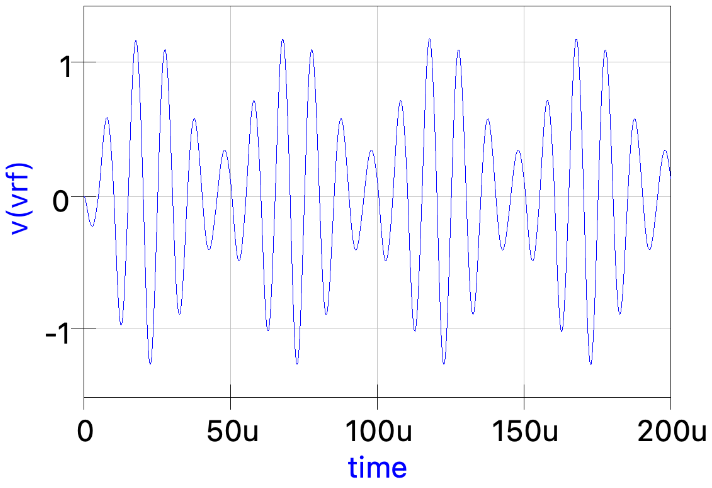
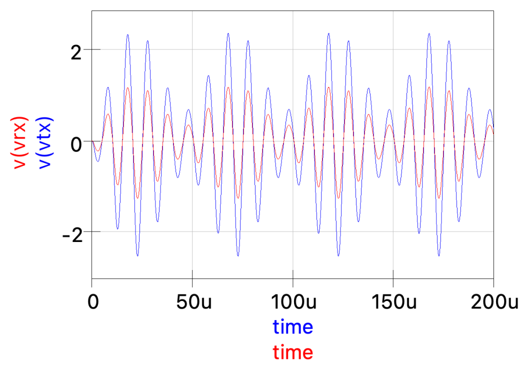


Figure 16: Band Pass Filter output

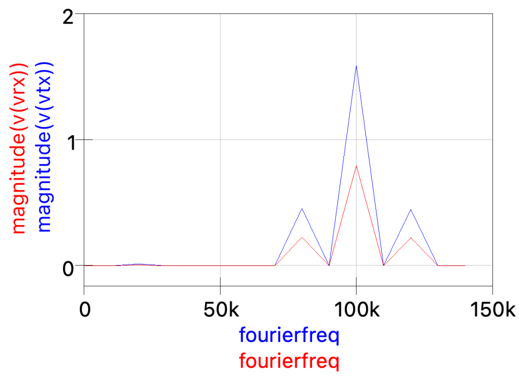
Analyzing the signal in the time domain, it is possible to see the envelope and the carrier signal outlined by it.

4.2.3 Channel Block

Now, to analyze the effect of the Channel Block, the signal after the Power Amplifier and at the Receiver can be compared, so both signal are presented in Figure 17 in both time and frequency domains.



(a) Time domain.



(b) Fourier Transform

Figure 17: Transmitted and received signals.

Figure 17 shows a difference in the amplitudes of both signals, validating the attenuation effect of the channel.

4.2.4 Receiver Block

Passing then to the Receiver Block, the signal needs to be demodulated, so, after passing through the Mixer, the resulting signal's Fourier Transform is displayed in Figure 18.

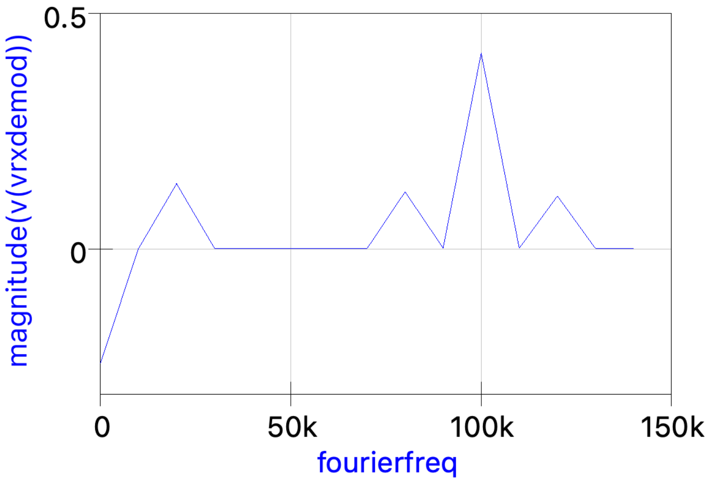


Figure 18: Fourier Transform of demodulated signal

After the mixer, the original signal is now in the original position, but the carrier frequency and the surrounding signal are still present, so, after the Low Pass Filter, the Fourier Transform of the filtered signal is shown in Figure 19.

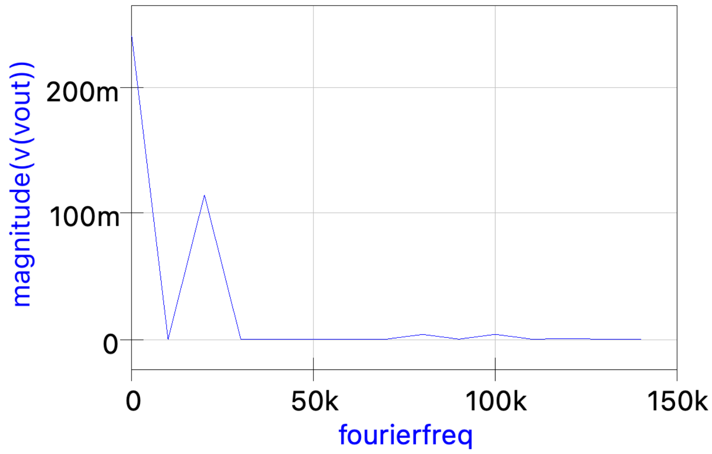


Figure 19: Fourier Transform of filtered received signal

Now, only the original baseband wave remains in the output, confirming the success of the AM Communication, to reinforce this observation, the comparison between the original transmitted signal and the one received can be seen in Figure 20.

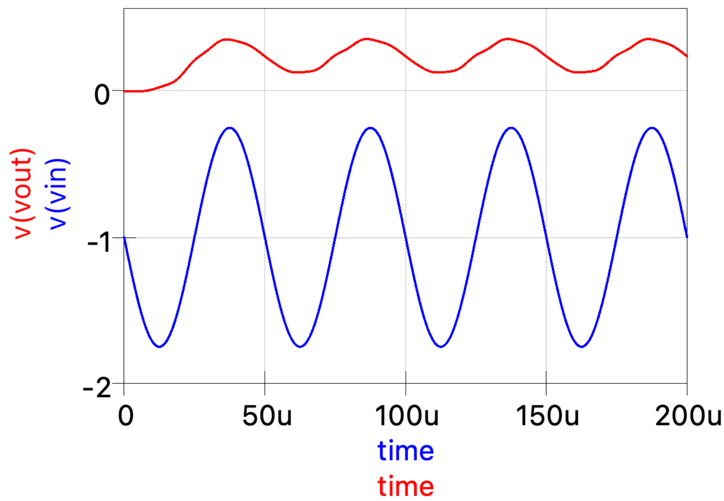


Figure 20: Transmitted and received signals

Comparing both waves, it is clear that the signal integrity remains unchanged, only the amplitude suffered a reduction in its value. This reduction is the result of both mixer stages, the channel attenuation and also a slight reduction caused by the filters, since this were not ideal filters and with small order.

5 VNA measurements and impedance transformation discussion

6 RFFE experiment setup, results, and analysis

7 Conclusions