Filter lab: Analysis, design and application

Objective

The objective of this exercise is the design of LC filters and their application to a basic communications system. The student must show the following skills:

- Definition of the specifications required for each filter.
- Selection of the type of filter and approximation in order to fulfill the specifications.
- Implementation of the doubly-loaded filter with inductors and capacitors.
- Time-domain characterization of the filter with the circuit simulator.
- Integration of the filter in a system, and understanding its effects.

Simulator

The simulator that is proposed is an open source development by Paul Falstad and Iain Sharp.

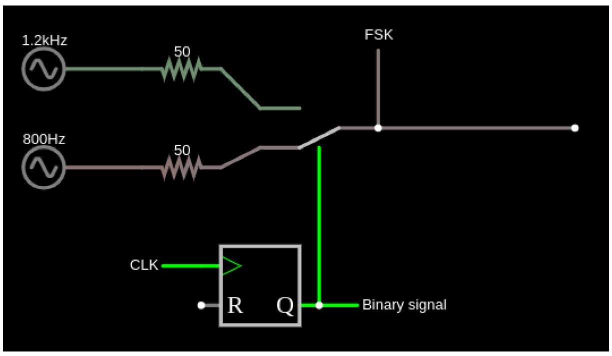
If the student has experience with other software for circuit simulation (for example, PSpice/OrCAD), he/she may use it. Notice that time-domain simulations are required to validate the designs, comments and conclusions. Of course, other types of simulations may be included if deemed necessary to complement the time-domain ones.

Lab instructions, evaluation criteria and submission

Same as in the first lab.

Approach

The circuit shown in the following figure is a model of a binary digital signal transmission system using FSK modulation, in which pulses of equal amplitude but different frequency are used to transmit the symbols “0” and “1”.



At the node `Binary signal` you can measure the binary digital signal to be modulated, and at the node `FSK` you can measure the modulated signal (link).

This modulation allows transmitting the signal through a channel occupying a frequency interval centered on:

$$f_0 = \frac{f_1 + f_2}{2}$$

and with approximate bandwidth

$$\mathrm{BW} = f_2 - f_1 + \frac{2}{T_s}$$

where f_1 y f_2 are the frequencies used for each symbol, and T_s the duration of each symbol (i.e, $\frac{1}{T_s}$ is the transmission bit rate, selected by the frequency of the clock signal `CLK`).

When receiving this signal, it is necessary to demodulate the FSK signal to recover the binary data. To do this, we can use a low pass filter that *discriminates* the frequency f_1 and the frequency f_2. The circuit of the following figure (link) is an example of this technique:

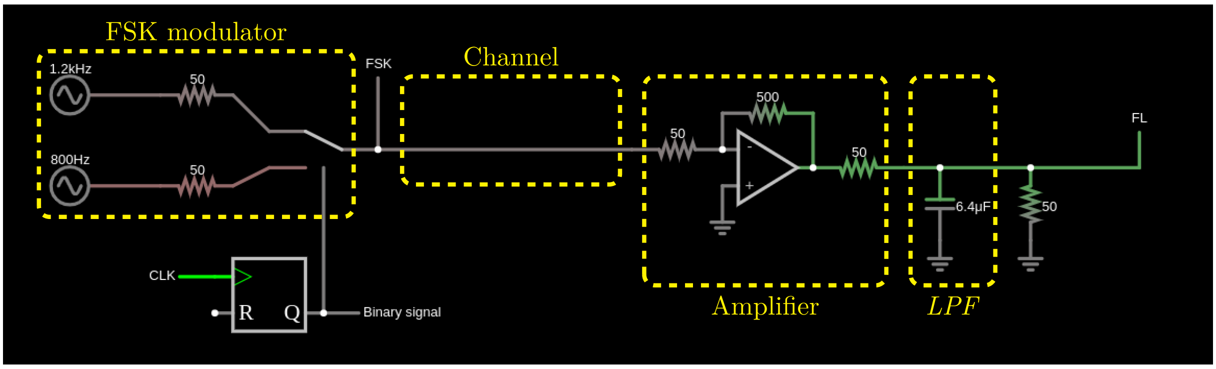


Figure 1: FSK with a low pass filter to discriminate frequencies.

The following blocks can be seen in the circuit above:

FSK modulator. Está implementado mediante un conmutador controlado por la señal digital binaria.

Channel. It is a direct connection.

Amplifier. It is an inverting amplifier based on an ideal operational amplifier, with an output impedance of $50\,\Omega$.

LPF. Low-pass filter that should attenuate the symbols transmitted with the frequency $f_2 = 1200\,\text{Hz}$ more than those transmitted with the frequency $f_1 = 800\,\text{Hz}$.

At the output of the low pass filter (at node `FL`) it can be seen that the pulses transmitted at different frequencies have different amplitudes (i.e. the FSK modulation has been converted into an ASK modulation). Therefore, an envelope detector could recover the baseband binary signal.

Alternatively, a high-pass filter could have been used to discriminate frequencies (link):

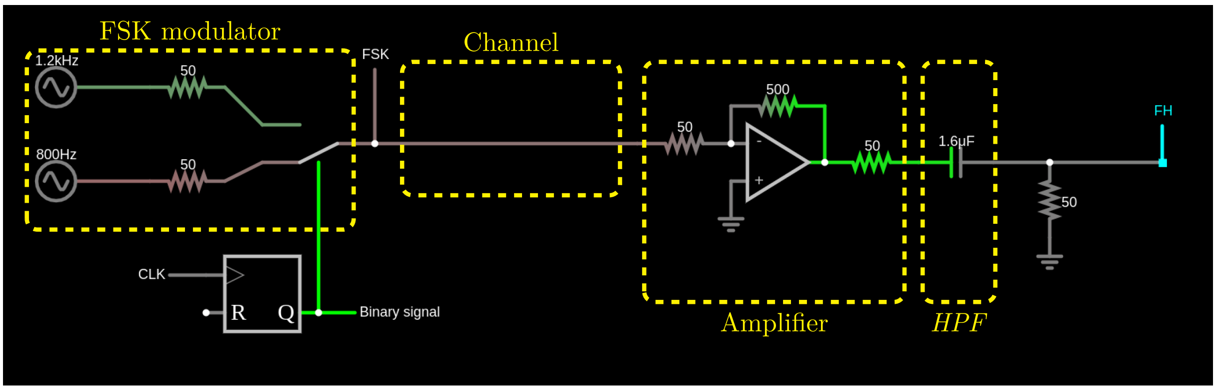


Figure 2: FSK with a high pass filter to discriminate frequencies.

The only change is to replace the block *LPF* by the block *HPF*:

HPF. High-pass filter that should attenuate the symbols transmitted with the frequency $f_2 = 800\,\text{Hz}$ more than those transmitted with the frequency $f_1 = 1200\,\text{Hz}$.

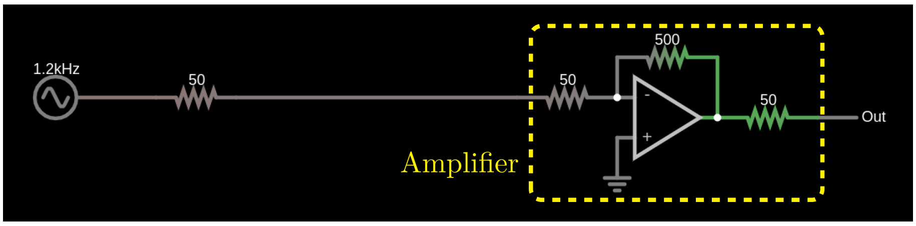
The result is again a signal with different amplitude levels for each symbol, but with the high and low levels interchanged.

Session 1

Response of the original low pass filter and high pass filter

1.1 Preparatory homework: Characterisation of the initial filters. [10 %]

1. Show that the impedance *seen* on the left by the low pass filter in Figure **??** is $50\,\Omega$. That is, verify that the impedance of the following circuit is $50\,\Omega$:

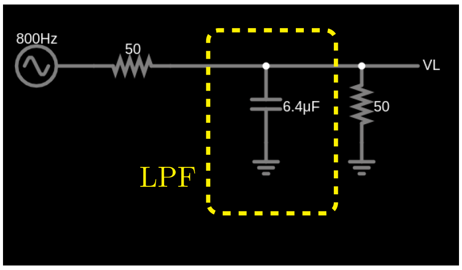


   What is the insertion *gain* (the inverse of the insertion loss) of the block marked *Amplifier* when we connect a $50\,\Omega$ resistor at its output (node `Out`)? Write your answer in dB.

2. Obtain, by means of circuit analysis, the following properties of the low pass filter used in the circuit shown in the Figure 1:

   - Filter order.
   - Transfer function.
   - Frequency at which the filter attenuates 3 dB.
   - Atenuación del filtro a frecuencias $f_1 = 800\,\text{Hz}$ and $f_2 = 1200\,\text{Hz}$.

   NOTE: Be aware that the filter is fed on the left side by a source with an internal impedance of $50\,\Omega$ (the output impedance of the amplifier), and is loaded on its right by another impedance of $50\,\Omega$. That is, the analysis is that of the following circuit:



3. Repeat the previous item for the high pass filter in the circuit of Figure 2.

4. It is desired to replace the previous filter with a more selective filter. For the new filters, the attenuation at one the frequencies must be less than 0.5 dB, and the attenuation at the other frequency must be more then 10 dB. Plot the specification mask of the new filters and then overlay on it the response of the initial filters (the ones obtained in the previous items). Use a computer tool that you can use during the laboratory session to create this graph. (Matlab, Python, Desmos, Geogebra,...)

   During the laboratory session, you will design a low-pass filter and a high-pass filter that meets these specifications, characterise it experimentally, and verify that these specifications are indeed met.

1.2 Lab work: Design and characterization of a new low-pass filter [40 %]

If the measurements taken in the lab do not correspond to the theoretical result (obtained in the previous work or during the lab session), you must either find the error in the theoretical approach, or find the error in the measurement, or try to explain the discrepancy.

1. (a) Check, by means of a Falstad simulation, the value of the frequency at which the low pass filter characterized in the previous work attenuates 3 dB. Take a screenshot of your measurement and explain how you have verified the result.

   (b) Same as in the previous section, but with the high pass filter.

   [5 %]

2. Design the low pass filter whose specification mask you have obtained in the preparatory work. [10 %]

   (a) Determine the type and order of the approximation to be used.

   (b) Obtain the filter implementation (take screenshots and explain the mesurements at each step):

   - Select a prototype of the tables. Check the prototype response at $\bar{\omega} = 1\,\mathrm{rad/s}$. In order to display such low frequency signals, you will need to increase the value of the `Time step size` in `Other options`. (In general this parameter should be between 100 and 10 000 times smaller than the period of the signal).
   - Denormalise impedances and check that the transfer function does not change. Do the voltages change? And the currents?
   - Apply the denormalisation on the frequency axis to the circuit, and check that the measurement does not change if it is now measured at the corresponding scaled frequency. (Do not forget to reset the `Time step size`.) Use a normalization frequency that allows you to meet specifications loosely in both the passband and the attenuated band.

3. Use low-pass to high-pass transformation $s = \frac{\omega_0^2}{p}$ to transform the low pass filter designed in the previous section into a high pass filter whose attenuation at 800 Hz and 1200 Hz match the attenuation of the low pass filter at 1200 Hz and 800 Hz respectively. Obtain the high pass filter and verify the requirements. [5 %]

4. Calculate and plot on the specification mask the amplitude response of the original low-pass filter and high-pass filters, and the new designed low-pass filter and high-pass filters(to make this plot, use the computer procedure you had to prepare in the preliminary work). [5 %]

5. Characterise the designed filters (Low-pass and high-pass) using the circuit simulator, taking measurements of the filter response (**only in modulus**) at significant points on the frequency axis. Add the measurements to the plot made in the previous paragraph and check the correspondence. [10 %]

   Note that since you are using a simulator, you can measure several frequencies simultaneously. Simply copy and paste the circuit. Do not forget to include the screenshots.

6. Use the filters that you have just designed in the circuit shown here, where the symbols transmitted in an FSK signal are detected by comparing the output of the high pass filter and the low pass filter.

   - Compare the result obtained at nodes `FL` and `FH` with the original filters and the new filters. Explain the result.
   - Compare the result obtained in the `RX signal` node. Explain the result

   [5 %]

Session 2

Band-pass filter at the channel.

2.1 Preparatory homework: Pass-band filter design. [15 %]

In order to avoid interference and, most importantly, noise, it was decided to add a bandpass filter between the *Channel* block and the *Amplifier* block in the circuit shown in Figure 2 or in Figure 2.

Design a filter that selects the frequency band of the FSK signal and filtesr the frequencies below and above. The attenuation in the pass-band should be no more than 0.5 dB. Leaving a transition band of between 400-600Hz approximately (justify your decision), the attenuation in the attenuated band must be greater than 20 dB for high frequencies, and greater than 30 dB for low frequencies.

In the design process, the following steps must be followed:

1. Draw the specification mask.
2. Apply frequency transformation to the specifications.
3. Choose order and type of approximation.
4. See the corresponding low-pass prototype in the design tables.
5. Apply frequency transformation to the low-pass prototype. (NOTE: may be the same transformation as the one used in step 2, or a different one if you decide to have a response that is not “touching” adjusted to the specification of the passband).
6. Taking into account the output impedance of the modulator, and the input impedance to the amplifier, scale and/or match the impedances of the design obtained.