RESISTIVITY VARIATION MEASUREMENT

Objectives:

- We intend to measure variations in resistivity in a medium with a bacteria culture. This is achieved using an electronic system.
- Develop an electronic circuit capable of detecting small resistivity variations in a medium using a resistive array in a Wheatstone bridge configuration.
- Develop a Digital-Analogical capture card with an USB communication interface that allows analogical data acquisition and its transfer to a computer on a binary format.
- Develop acquisition and analysis information software.

Materials:

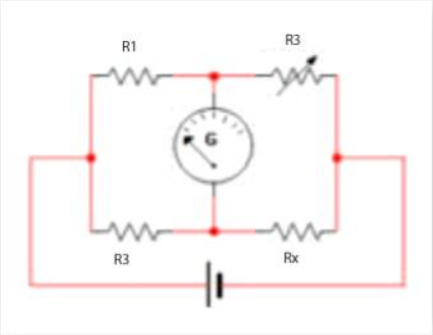
These are the electronic components used to develop the electronic system and their theoretical justification:

Wheatstone bridge:

A Wheatstone bridge is an electronic circuit designed to measure the resistivity (impedance, in a general way) of a component in the circuit where the resistivity of another three components is known.

On picture 1 you may observe that R_x is the resistance to be estimated where R_1, R_2 and R_3 have known resistance values. Also, R_2 resistance is adjustable. If the resistances ratio in the known arm R_1/R_2 equals that in the unknown arm R_x/R_3, there will be no voltage V_D between the upper and lower red points and there will not be any electric current between these points.

During the measuring phase, R_2 resistance is varied until it reaches equilibrium.



Picture 1: Wheatstone bridge diagram.

The direction of the electric current, during disequilibrium, reveals if R_2 is too high or too low (The value of the power source (E) of the generator is indifferent and does not affect the measurements).

When the bridge is constructed in a way that R_1 equals R_3, R_x will equal R_2 in an equilibrium condition (no electric current through galvanometer).

Likewise, during equilibrium the following always holds true:

If the values for R_1, R_2 and R_3 are accurately known, the value for R_x can be determined with much precision. Small variations in R_x disrupt equilibrium, thus being accurately detected by a galvanometer (Or, as in our case, by an amplifying system).

Instrumentation Amplifier:

An instrumentation amplifier is a device made up by simple operational amplifiers. It’s designed aiming for high entry impedance and an efficient common mode rejection. (CMRR, relation of amplification of signals which are totally equal in both of their entries. This property allows discarding induced noise).

It can be built based on discrete components or it can be found encapsulated (e.g. INA122P, AD60).

This array performs a voltage signal subtraction on each of its two entries:

$V_{div} - (V_2 - V_1)$ (1)

Differential voltage obtained V_{div} is multiplied by a gain or amplification factor determined by the next equation:

$Vdiv= 1 + 2R_1/R_x$ (2)

By joining equations (1) and (2) the instrumentation amplifier output equation can be obtained:

$$V_{sal} = (V_2 - V_1)\left(1 + \frac{2R_1}{R_x}\right) \qquad (3)$$

Active Filter:

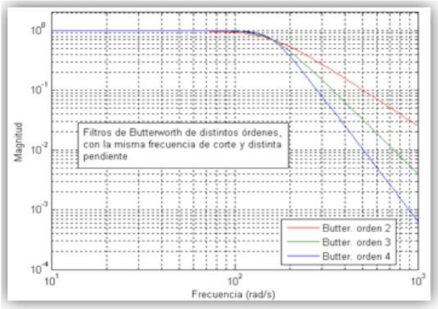
An active filter is an analogical electronic filter that uses one or more active components (that proportionate a specific way for amplifying energy), that what makes it different from passive filters, which normally use only passive components. Commonly, this active element can be a vacuum tube, a transistor or an operational amplifier.

An active filter can have a partial or complete increase on its output signal with respect to the entry signal. Its implementation combines active and passive elements, where operational amplifiers are frequently used to obtain resonance and a high Q factor with no coil usage.

Among others, low pass, high pass and band pass filters with Sallen-Key or State variable configuration can be implemented.

Butterworth Filter:

A Butterworth filter is one of the most basic electronic filters designed to generate an output as flat as possible until cutoff frequency. In other words, output remains constant until cutoff frequency, then, it lowers at a ratio of 20n dB per decade (or ~6n dB per eigth), where n is the number of poles of the filter.



Picture 3: Graphic representation of frequency answer for a Butterworth filter

Design:

Let H be the frequency response, the first 2N-1 derivatives from $|H(\Omega)|^2$ should be equal to zero when $\Omega = 0$ and $\Omega = \infty$. It only has poles and its transfer function is:

$|H(\Omega)|^2 = 1/(1 + (\Omega/ \Omega_c)^{2N})$

Where N represents the filter's order, Ω is the cutoff frequency (where the output reaches 3 dB under the passant band) and Ω is a complex analogical frequency (Ω - jω).

Design is independent of implementation, which can be through Sallen-Kay or Rauch cells, discrete components etc.

Active electric current source:

For integrated circuits where instrumentation amplifiers are included, it is necessary to polarize its component transistors by adjusting polarization resistances which display variations on its resistive value as a consequence of temperature variation, in this way modifying the polarization point in transistors and affecting confidentiality of the circuit under construction.

A way to avoid variation in polarization levels due to temperature or resistance changes in the sensor is the implementation of an electric current source for

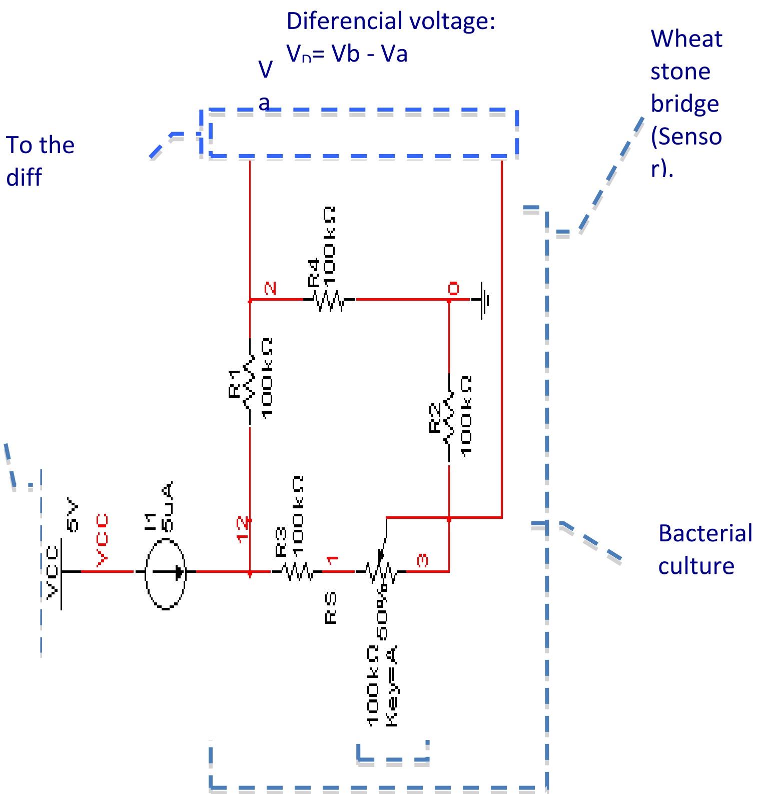
Wheatstone bridge. This way we are provided with a constant reference for determining resistance in the sensing medium.

Developing:

Sensor Resistivity:

Detection of conductivity variations in the bacterial culture is achieved by introducing two platinum-covered chrome electrodes into the medium. The sensing methodology consists in using a Wheatstone bridge resistive array. Since high medium resistivity is expected, we use four resistances of 100 kΩ, one of these is variable (R_3) and intends to simulate electric resistance variation in the medium where the resistance terminals work as electrodes (Because of issues regarding the simulation).

The array is fed by a direct electric current source that injects a current equivalent to I_T=5 uA.



Picture 2: Sensor diagram

We intend to find an equation that relates input current I_T, differential voltage obtained from the Wheatstone bridge and resistances R_1, R_2, R_3 y R_4, which can be simplified to R_1= R_2= R_3 = R_4,= R following Ohm law:

$V= RI_T$ (6)

It is noticeable that node potentials vary in a proportional way with respect to resistance variation in each of the arms of the Wheatstone bridge, where the node potential Va is:

$Va = I_T((2R + Rs)/(4R + Rs))$

And node Vb potencial is:

$Vb = I_T(2R/(4R +Rs))$

From the concept of differential voltage V_D expressed as:

$V_D=Vb – Va$ (9)

And solving for Rs, an equation that relates electric current I_T and differential voltage V_D with the medium resistance Rs was obtained.

$Rs = 4RV_D/(RI_T –V_D)$

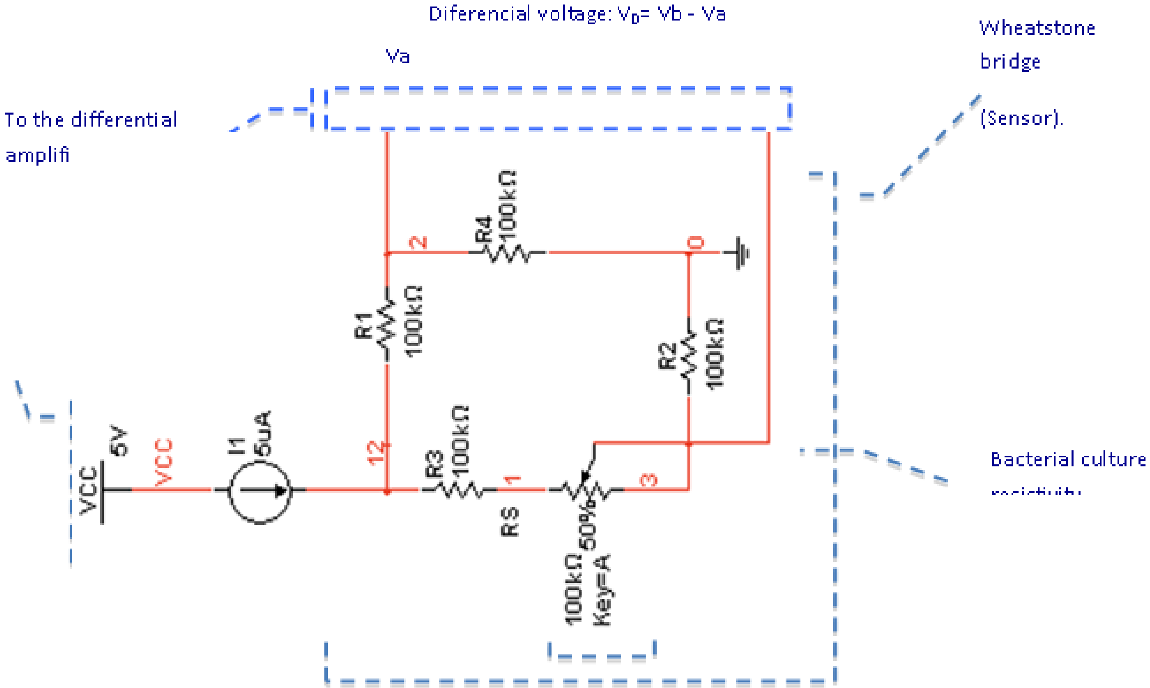
Amplifcation:

Given that the magnitude of the signal detected by the sensor is too small, it is necessary to implement the instrumentation amplifier described previously. The amplification phase is made by an AD620 integrate.

Analog Filtration:

Once the signal has been amplified at intervals interpretable by the low sensibility electronic stages it is filtrated for discarding noise and rendering it adequate for digital processing, making a digital re-filtering for a better interpretation of obtained information possible.

The described filter has a passing frequency equal to 20 Hz and the cutoff frequency is equal to 30 Hz where the gain equals -1 dB. The filter is of a sixth order Butterworth type, as a consequence, it has a slope of 120 dB/dec between the cutoff and passing frequency.



Picture 1: Sensor diagram

Analog to Digital conversion of acquired signal.

Analogical to digital conversion transcribes an analogical signal to a digital signal, facilitating its processing and reducing noise susceptibility.

The conversion consists in a periodical measuring of signal amplitude and its successive translation it into a numerical language. The algorithm consists of three steps:

1. Sampling: Periodical sampling of wave amplitude. Where sampling frequency (velocity) is defined as the number of samples taken every second.
2. Quantification: Quantifies voltage in each sample and relates every signal interval value to discrete output values.
3. Codification: Traduces quantification values to binary code.

Once the filtering stage is completed and the signal is interference free, consecutive amplification stages becomes a possibility for signal manipulation, according to analogical-digital conversion stage requirements for a posterior processing and visualization.

Digitalization stage implements a microcontroller Microchip© PIC18F4550® which has a 10 bits ADC resolution frequently used with an analogical reference voltage from 0V to 0.752 V, allowing a system sensibility equal to 0.752 V/ 2^{10} bits = 0.734 mV/ bit.

From equation (10) and fixing V_D at 0.743 mV, we find that system sensibility given in ohms/bit (resistivity variation per bit) equals:

Sensibility=4(100kΩ)(0.734mV/bit)/((100kΩ)(5uA) – (0.734mV/bit)) - 588.06Ω/bit

Another important aspect to take into account is sampling frequency F_S, which in this system is of 1000 Hz. According to Nyquist theorem the maximum detectable signal is defined as 500 Hz, (The velocity of the change of resistivity in the medium is expected to be lower than 500 Hz).

Sensor - PC Communication interface.

The designed sensor has a 2.0 USB (Universal Serial Bus) module for communicating with the PC. Through it, the continue vector containing sampled data previously treated by a rectangular window filter with a 10 Hz cutoff frequency that allows elimination of abrupt voltage V_D variations due to electromagnetic noise. Host (PC) employs a driver that identifies the sensor and assigns a communication channel and its respective memory and time processing resources. Once the sensor has been identified by the operating system it is possible to establish communication and to obtain data that is being continuously sent by the operating system through a user application that recognizes such device.

The application that uses data transmitted by the sensor was designed in Microsoft© Visual C. This application establishes communication with the sensor and defines time lapses for capturing analogical signal. Once the application in the PC has captured whole data, it displays a tabulated list containing indicative columns for sample number, digital value obtained, differential voltage V_D correspondence and the equivalent resistance offered by the medium in the sampling instant. At the same time, the application plots resistance data obtained versus time.

When previously stated recording time ends, it is possible to store the obtained data in an ascii file with tabular format for its posterior processing.

Complete sensing circuit:

Below, the complete amplification and filtration circuit developed in a (Multisim 10) electronic simulation environment is shown.

Simulation and Results:

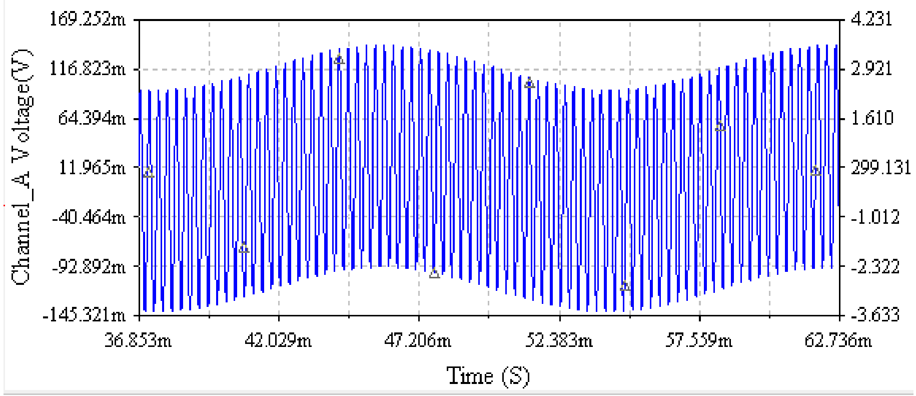
The system here described was tested in an electronic graphic simulation environment (Multisim 10), where fallowing results were obtained.

Amplified signal with noise in transmission line:

If the output signal from the first amplification stage is analyzed, it is noticeable that the noise, as well as the signal of interest, from the experimentation environment has been amplified.

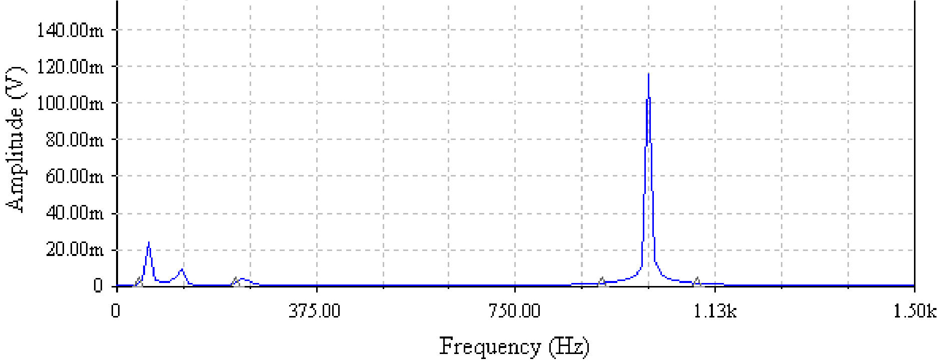
The graph below clearly shows the 1 kHz signal (interest signal) modulated by the interference signal found in the 60 Hz and its most significant harmonics from the transmission line.

Here it is evident that variations in voltage caused by interference, have a 30% contribution in the signal of interest. As a consequence, wrong readings concerning medium resistivity are registered.



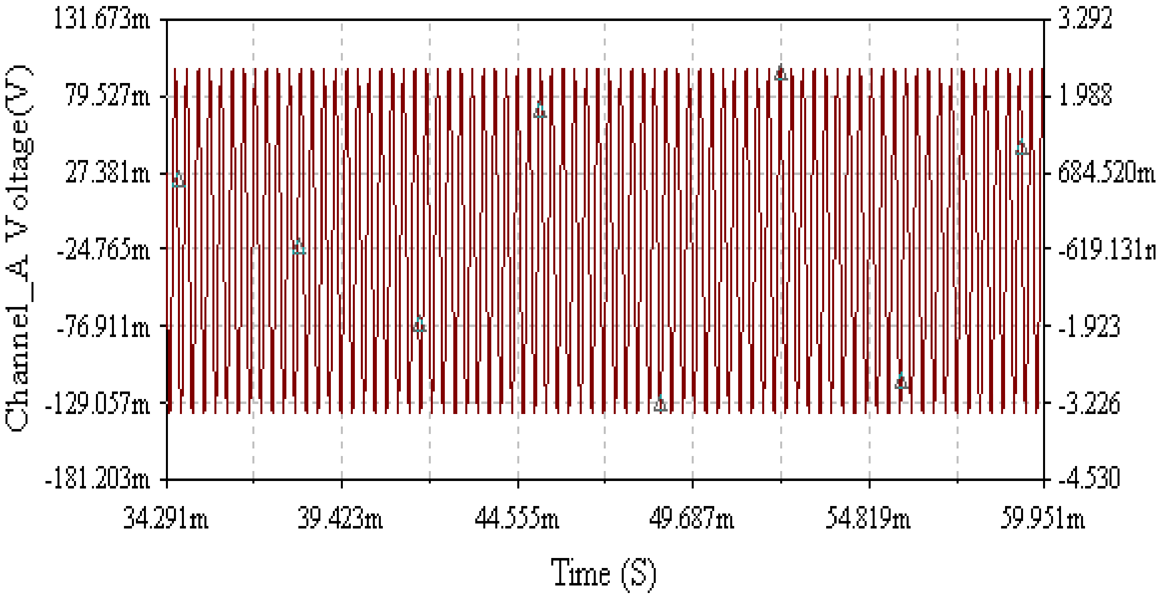
Picture 9: Signal contaminated with low frequencies

If a Fourier analysis under FFT operation is performed, a frequency spectrum that contains the signal of interest (1 kHz) and those that are interference, is obtained.



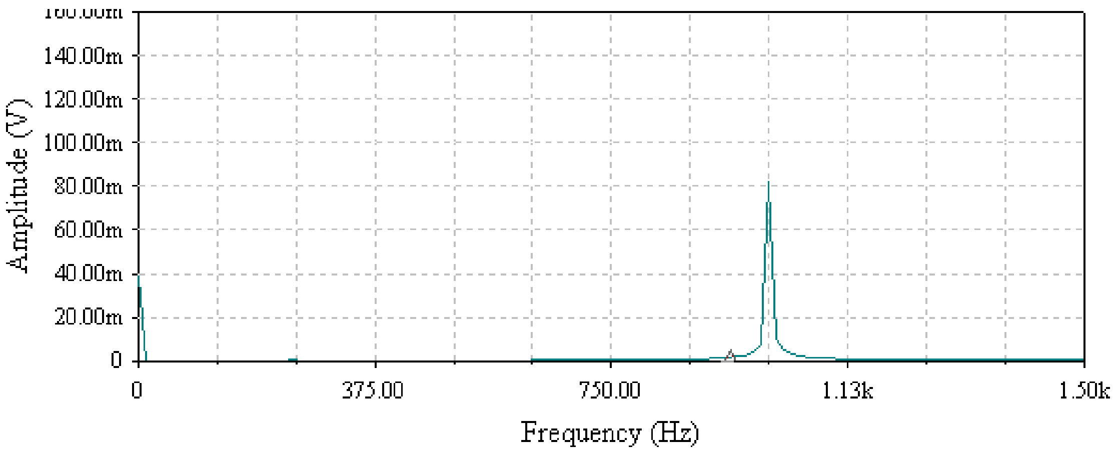
Picture 10: Frequency spectrum

After applying the analogical filter, the signal displays the following behavior:



Picture 11: Filtered signal

And the associated frequency spectrum is:



Picture 12: Frequencies spectrum for filtered signal

Therefore, it is demonstrated that the interference frequencies have been eliminated.

Analysis of the Measurements

Design of a digital filter for eliminating measurements noise

A digital filter is a system that applying mathematical processing on the signal, generally consisting on a discrete time Fourier analysis, discards unwanted components based on its frequencies. The interval of frequencies that goes through the filter are called *passing band* while discarded frequencies are known as *suppression bands*.

According with the spectrum is allowed to pass or the one that is attenuated filters are classified as:

- *High pass filter:* Attenuates low frequency components allowing high frequency components to pass through, with the possibility of being amplified by active components.
- *Low pass filter:* Attenuates higher frequencies allowing lower frequencies.

And according with the type of response to the unitary entry:

- *FIR (Finite Impulse Response):* If the entry signal is an impulse, the output vector is composed by a finite number of not null elements.
- *IIR (Infinite Impulse Response):* If the entry signal is an impulse, output vector is composed by an infinite number of not null terms. This implies that it never returns to the basal state.

As filters of finite impulse response have a linear response phase to impulses (symmetry) and are faster and easier to handle than infinite response filters, we decided to use one of those filters designed by the window method. The window method consists on the unification of the filter response sign by multiplying the window by each of the elements on the signal vector. In this way, noising data that obstacles information analysis can be discarded.

For filtering the signals generated by our system we used a digital filter (low pass) of finite impulse response. A window, of 150 Taps and a sampling frequency of 1 kHz, was implemented. This window was chosen based on previous knowledge about the noise frequencies found in the environment as well as the oscillation frequency of the signal of interest.

MathLab was used to obtain the coefficients composing the filter.

Signal Filtering:

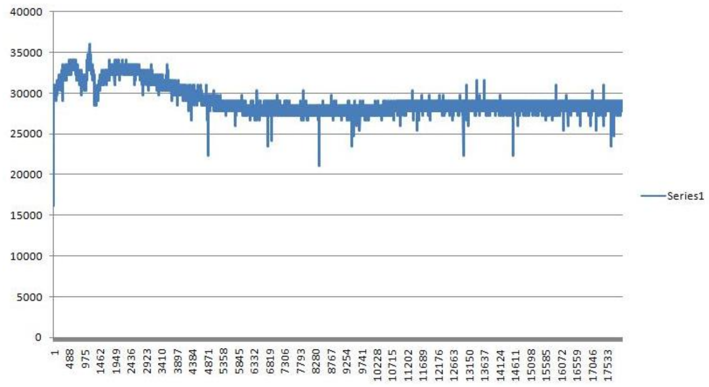
Once the coefficients that define the filter are known, the filter is applied. Mechanistically, filtration consists on:

Let V_1 be the vector that describes the filter and V_2 the one that represents the signal to be filtered.

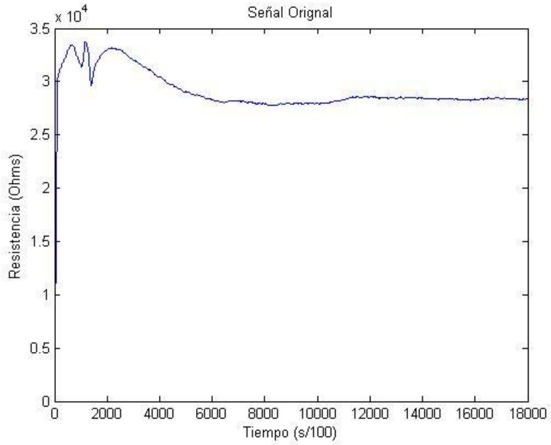
1. Multiply each of the samples in the actual V_2 by defined coefficients in V_1.
2. The sum of the successive multiplications results composes a third vector that represents the output of the actual instant.

As a result of the previous described convolution, a vector is obtained. It unifies digital signal with the filter vector, discarding this way frequencies over the cutoff frequency. Data obtained this way are saved on a new file that contains the elements of the filtered vector. For visual analysis simplification, the option for generating a graphic distribution of filtered data was implemented in the same program.

Applying the described algorithm to each of the measurements, noise from the environment and electronic interference of the electronic components is filtrated. This is easily visualized in the graphs presented below:



Picture 13: Unfiltered signal describing behavior of cells transformed with YohM⁻+ pBB + RcnA; after addition of 500µM of nickel sulfate.



Picture 14: Filtered signal describing behavior of cells transformed with YohM⁻+ pBB + RcnA; after addition of 500µM of nickel sulfate.