

Infrasound Microphone

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Abstract

My project was about building a microphone to be used to see infrasound noise present in LIGO's apparatus and this article describes the work I did to build it.

Chapter 1

Basic Ideas

The basic principles on which the microphone is based are easy to understand: first I thought to build my microphone using the same principles as common loudspeakers and microphones are built: a membrane fixed to a coil with a magnet inside, so that any air pressure variation on the membrane induces a movement of the membrane itself and of the coil, and so a current on the coil itself.

However I should abandon this idea soon because I realized that my infrasound microphone had to work at frequencies below 20 Hz, or at wavelengths above 17 m: too much compared with the displacement of the membrane, so that, at the first order, my apparatus couldn't see any pressure variation, because too weak. So I looked for another method, not based on any pressure variation on loudspeaker, but, better, based on the effective pressure on it. This method, that I eventually found, consists of this idea: there's a coil fixed to a membrane, as in the loudspeaker, but instead of a magnet inside the coil there's another coil above the first one and fixed to a retaining structure (in my case to the metal framework of the membrane), so that one coil is movable and the other one is fixed. Then the movable one carries an alternate (sinusoidal) current. In the other coil an induced alternate current appears, with the same frequency as the first one, but with an amplitude which depends (in first approximation) only on the distance between the 2 coils.

To be clearer, call A the fixed coil and B the movable one, also call z the mutual distance between the 2 coils, Φ_A the flux of the magnetic field in the fixed coil and i_A and i_B the currents that flow in each coil. Then it's easy to see that

$$\Phi_A = M(z)i_B = M(z)i_{0B} \sin(\omega t + \phi)$$

where $M(z)$ is a factor that depends only on the relative distance of the coils and on the geometry of the system. From the Faraday-Neumann law you have:

$$f.e.m._A = -\frac{d\Phi_A}{dt} = -\frac{dM(z)}{dt}i_B - M(z)\frac{di_B}{dt}.$$

The first term in the approximation of small variation of z is negligible compared to the second one, so that the current i_A in the movable coil as an amplitude that depends on the relative distance of the coils:

$$i_A = \frac{f.e.m._A}{R_A} = -\frac{M(z)}{R_A}\frac{di_B}{dt} = -\frac{M(z)}{R_A}\omega i_{0B} \cos(\omega t + \phi)$$

where R_A is the resistance of the movable coil and i_A is still a sinusoidal function with the same frequency as i_B .

With this in mind it's easy to understand that if there's any pressure variation on the membrane, then there will be an amplitude variation of current (and so of tension) in the fixed coil, and this variation can be detected by an electronic circuit, as the one described below. All this system had been fixed inside the lid of a metal barrel, isolating so doing a side of the membrane and improving the response of the apparatus to any external air pressure variation.

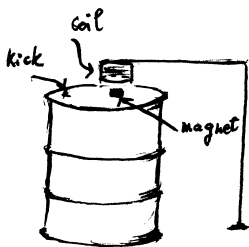


Figure 1.1: measurement of barrel resonances, as explained below

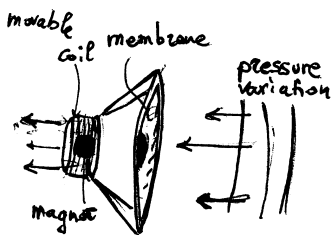


Figure 1.2: picture of the principle on which common loudspeakers are built

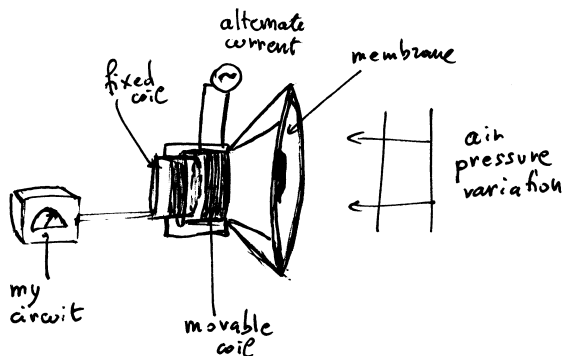


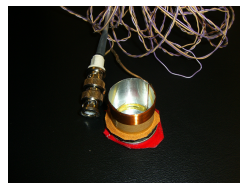
Figure 1.3: picture of the principle on which my loudspeaker was built

Chapter 2

Construction

To build my microphone I used two loudspeakers with the diameter of 10 cm and a metal barrel of size 60cm(diameter)·80cm(height), at the top of which, inside the lid, I fixed the membrane of one of the two speakers.

First I roughly measured the low-frequency resonances of the barrel to understand if I had to strengthen its walls, cutting so off all these resonances that could decrease the response of the membrane to the air pressure variation because of the additional response of the barrel's walls to the pressure waves. I did it using a little magnet fixed to the top of the barrel and a coil above it fixed through a rigid support independent from the barrel and linked to the ground. Then I hit the barrel on different points.



I measured the electric signal, induced by the vibration of the magnet in the coil, on an oscilloscope and I realized that there were some resonances at about 10 – 20 Hz. Below there's a graphic representing one of these measurements, after having isolated the resonance vibrations.

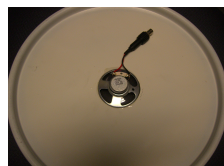


Figure 2.1: coil fixed in the lid of the barrel



Figure 2.2: barrel

I decided to cover the top and the bottom of the barrel with an aluminium beehive-shape material 1 cm thick and two more lids (like a sandwich), to make all more rigid. I then built my microphone based on the LVDT's method: I removed in one speaker, composed of a membrane and a coil fixed to it, its magnet and I pasted a second coil, taken from the other speaker, above the first coil, fixing it to the metal framework of the first speaker itself (LVDT method). Then I did a hole on the top lid of the barrel and I pasted the whole structure in this hole, with one side of the membrane inside the barrel.

I also did a small hole on the lid to blow in, and so to measure the characteristic relaxation time of the membrane, understand if there could be some air losses inside the barrel, and, if so, understand if these losses could decrease the response of my apparatus to noise under 20 Hz. In fact I found out that (with the small hole closed) there were some losses that could be relevant at the noise frequency of the order of 10 mHz. Below it's possible to see the graphics of the membrane relaxations with open and closed hole.

To set my microphone I built a wood box bigger than the barrel, I put the whole system inside this box and I fixed a subwoofer at the top of it. Then I sealed the wood box to isolate my system and used the woofer to produce pressure waves, in order to test the response of my microphone to any pressure variations. All this analysis was done through an electric circuit used to detect any change of the current's amplitude on the fixed coil.

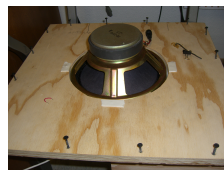


Figure 2.4: woofer



Figure 2.5: wood box

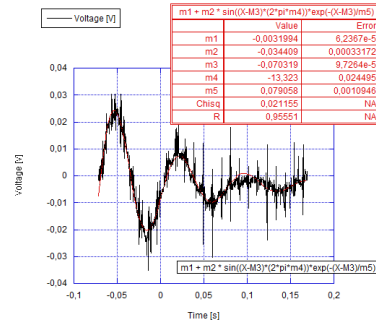


Figure 2.3: the blue and black line represent the relaxation curve of the membrane with the hole opened and closed

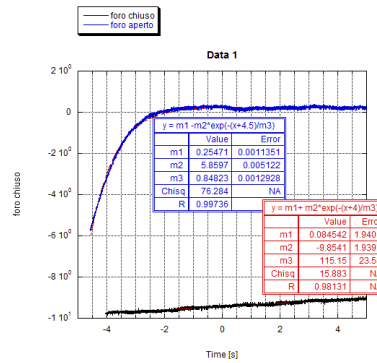


Figure 2.6: the blue and black lines represent the decay laws respectively for the opened and closed hole

Chapter 3

Electric Circuit and Measurement

With the LVDT method in mind I built a circuit with the aim of detecting pressure variations on the membrane of my microphone. After having built it I soldered it in order to reduce some additional noise coming for example from the board

Each coil of my microphone had a resistance of 8 ohm and a maximum power dissipation of 2 W. I made a fixed current of frequency at about 10 khz and amplitude at about 100 mV flow in the movable coil. As already said, in the fixed coil an induced current appeared, with an amplitude depending on the mutual distance of the 2 coils. My circuit had in input both the signals coming from the coils, and was divided into 3 parts:

1. the two signals are amplified up to tensions of the order of Volts
2. the two signals are put in opposition of phase (phasators) and then added (that is they are subtracted). So doing I get a signal whose amplitude variation is proportional to a pressure variation on the membrane. Then a time average of the modulus of this signal is taken through a diode and a bass-pass filter (with a cutting frequency at about).
3. the final signal is amplified once more, with respect to a tension reference, chosen so that without pressure on the membrane the resulting signal at the output of my circuit is 0 Volts

So doing I got at the end of my circuit a signal proportional to the air pressure variation on the membrane. The sketches below represent all that:

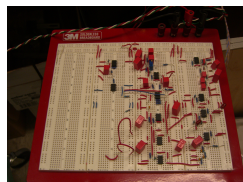


Figure 3.1: electric circuit

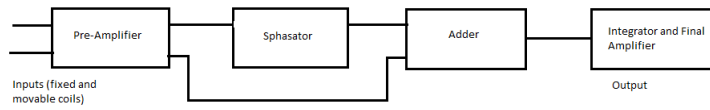


Figure 3.2: general sketch

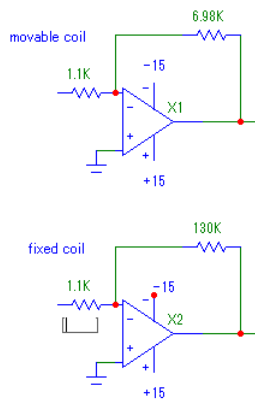


Figure 3.3: Pre-amplifier

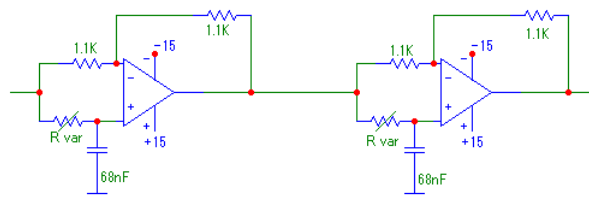


Figure 3.4: Sphasator

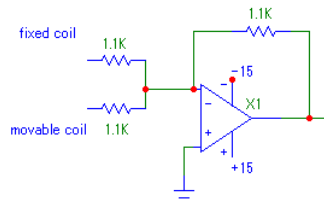


Figure 3.5: Adder

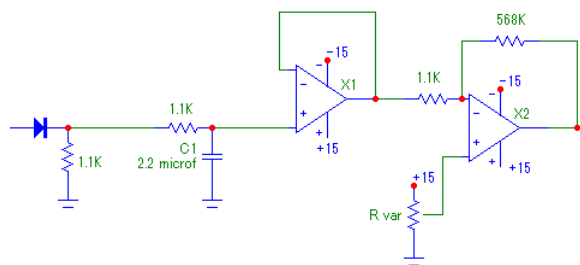


Figure 3.6: Integrator and final amplifier

Then I used the subwoofer, as said before, to test my microphone: I measured the response of my circuit to pressure waves of different frequencies produced by the woofer and at the end I plotted all that in a graphic getting the transfer function of my microphone. Below it's possible to see the signal sent in the subwoofer (constant in amplitude for all values of the frequencies used) and the response signals detected. For each of them I reported the real signal measured, its average over a period (trend) and its sinusoidal part, calculated by the real signal after having subtracted the trend part.

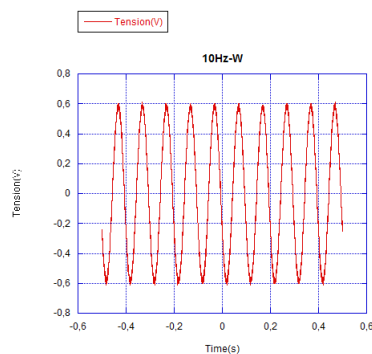
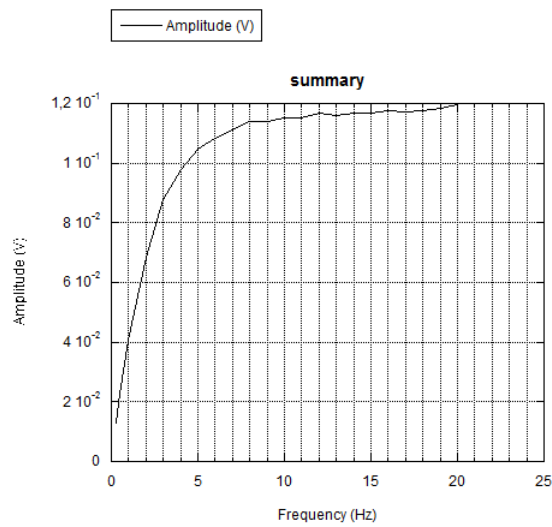


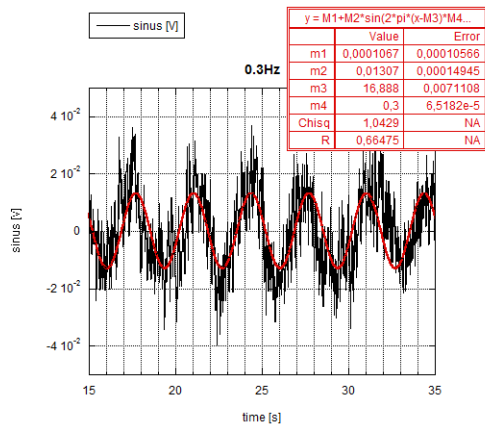
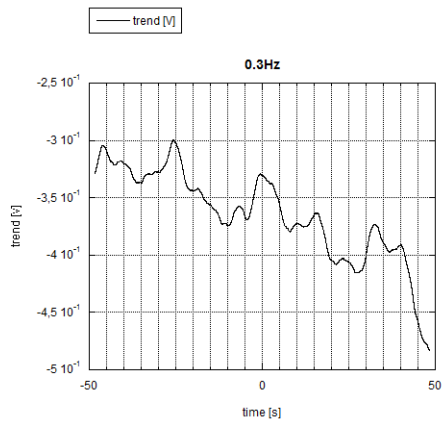
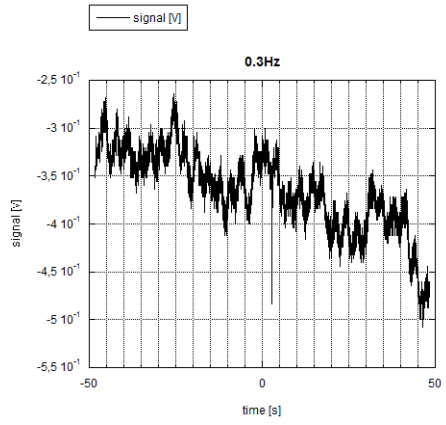
Figure 3.7: sample of woofer signal of 10 Hz frequency

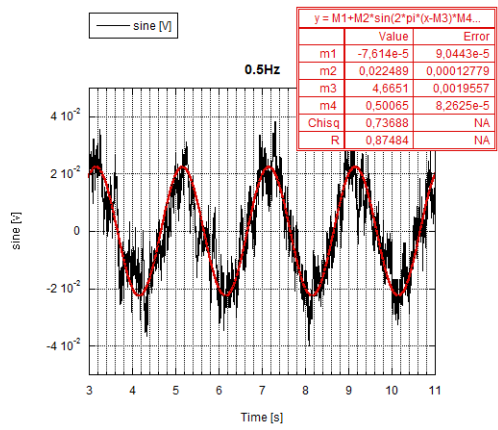
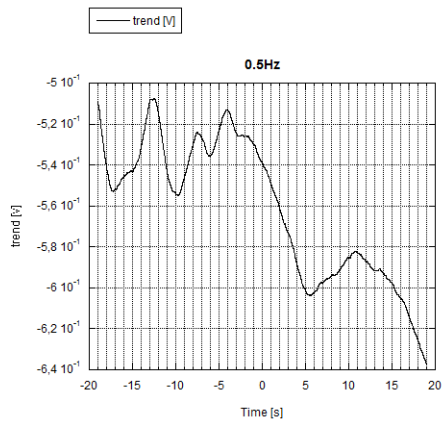
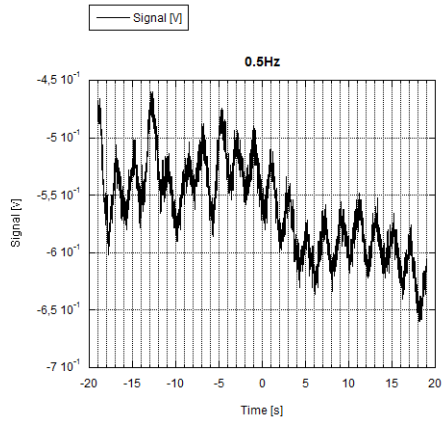
Chapter 4

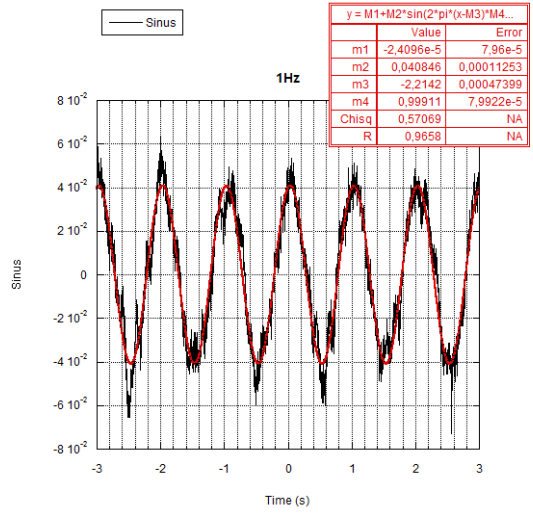
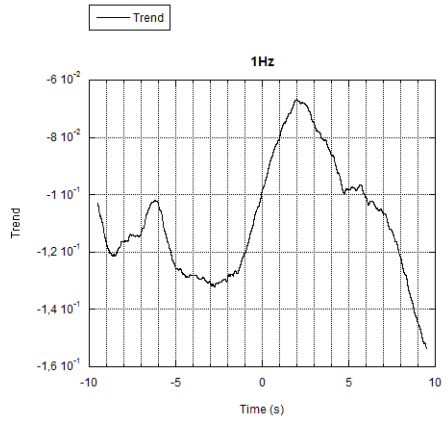
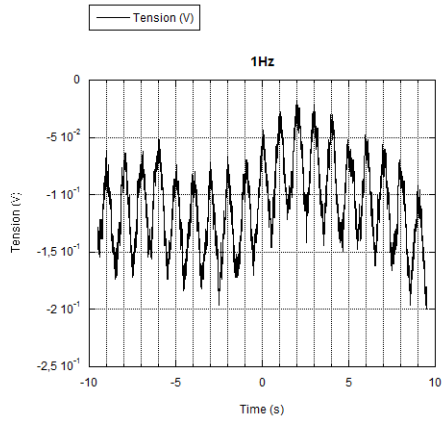
Conclusion

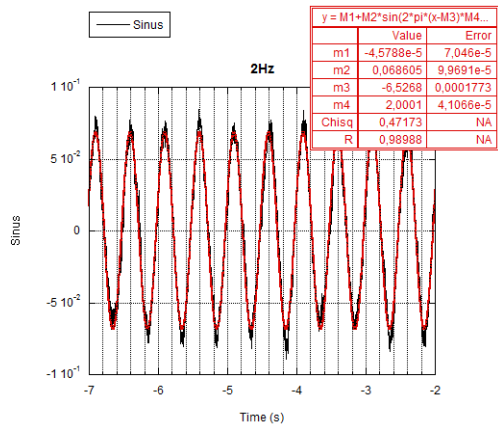
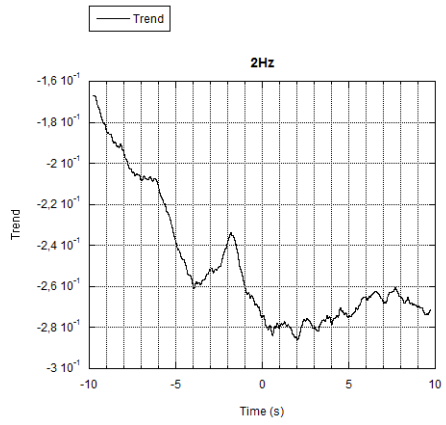
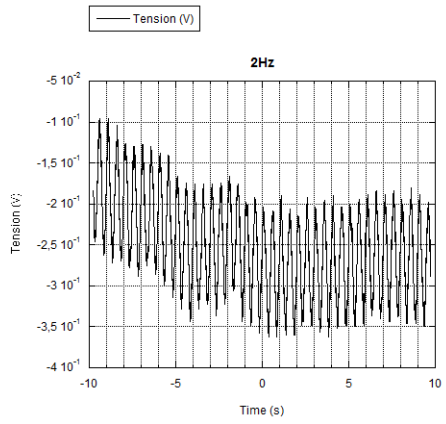
Below it's possible to see the plot representing the response of my microphone at different frequencies. In particular it's evident that there is a region, delimited by a cut-off frequency of about 2 Hz (the frequency at which the response is about one half the maximum response value), in which my microphone doesn't work well.

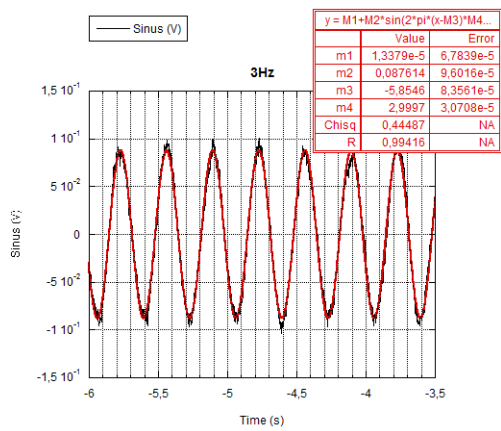
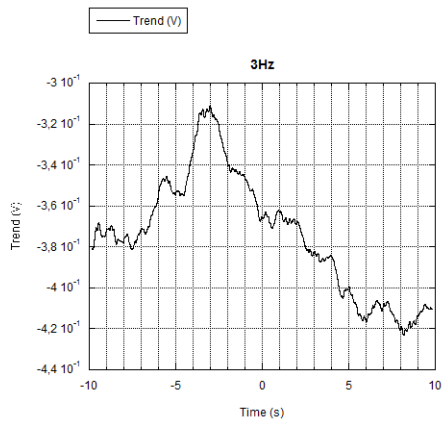
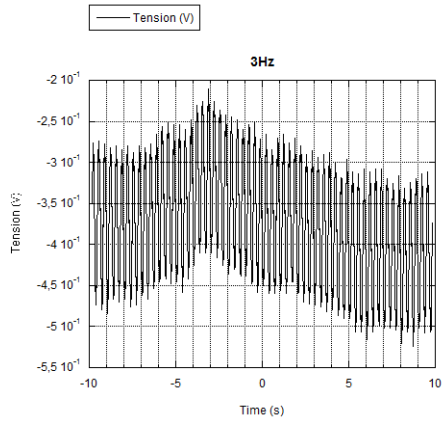


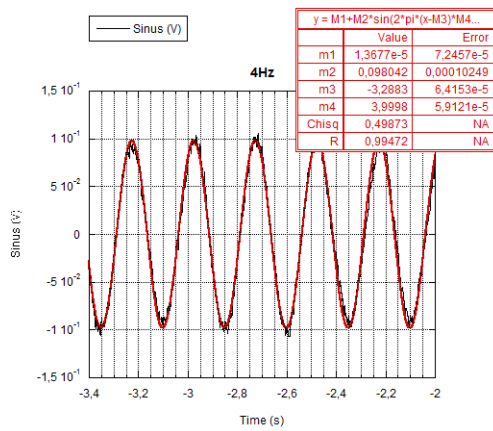
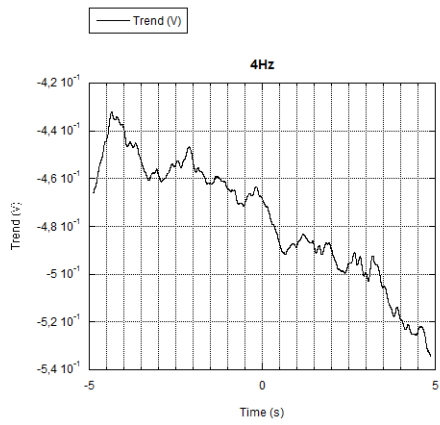
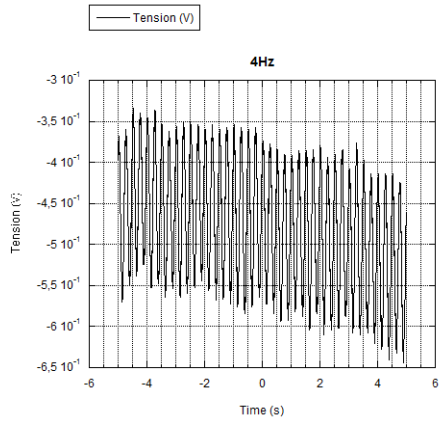


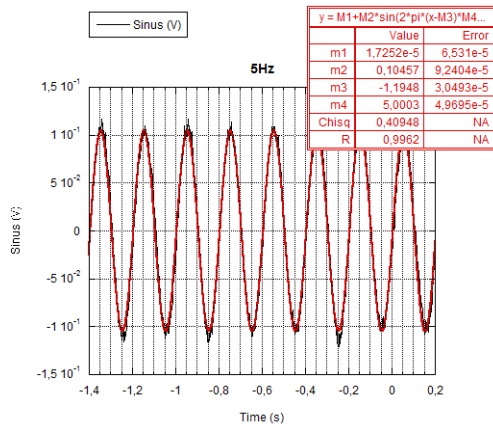
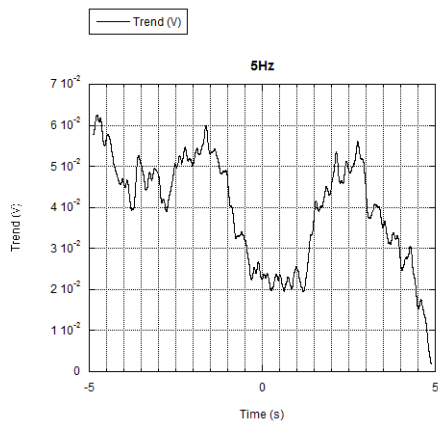
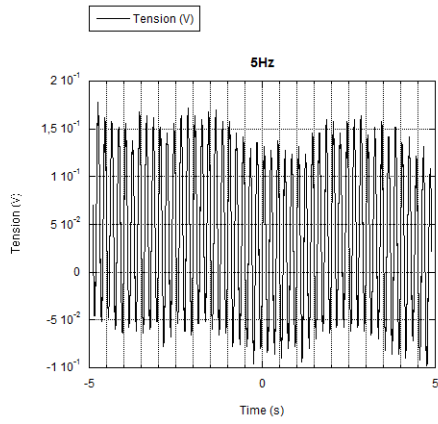


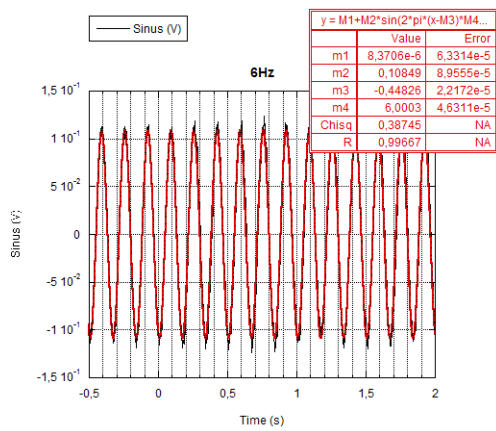
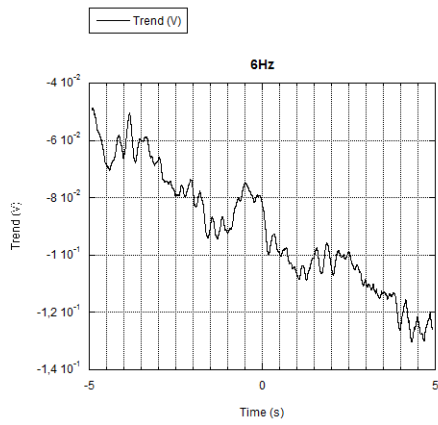
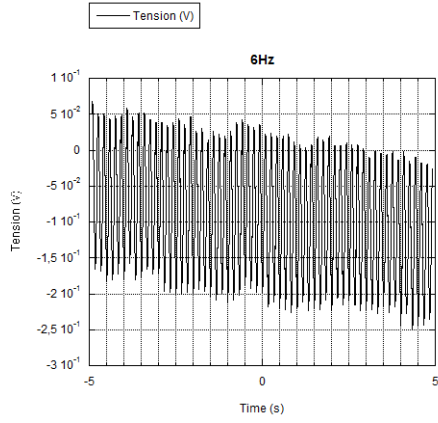


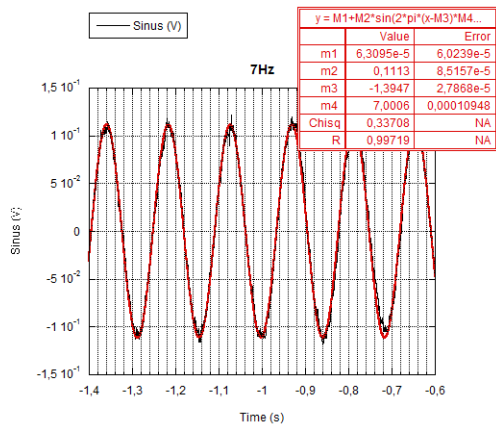
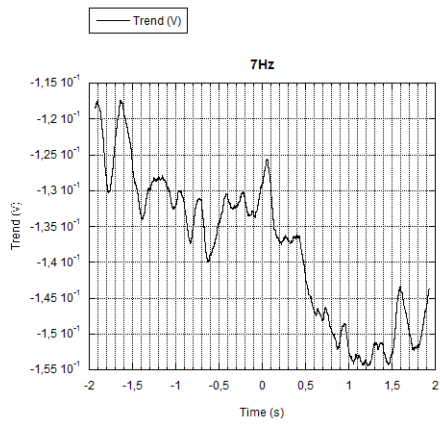
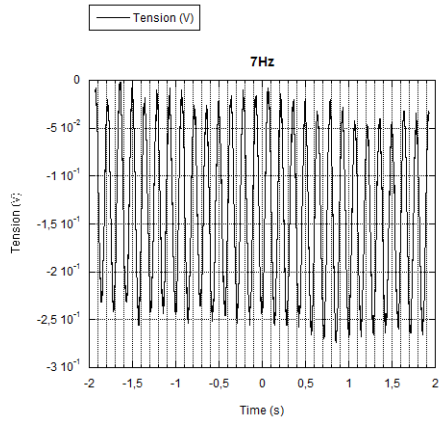


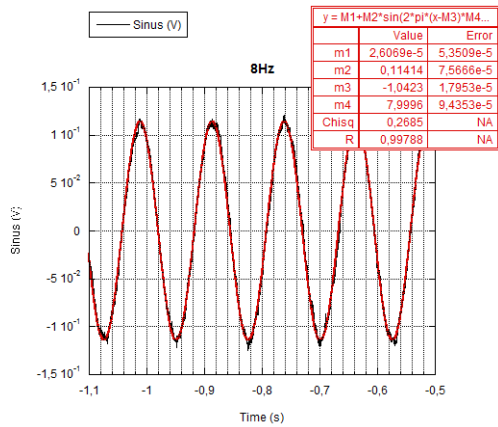
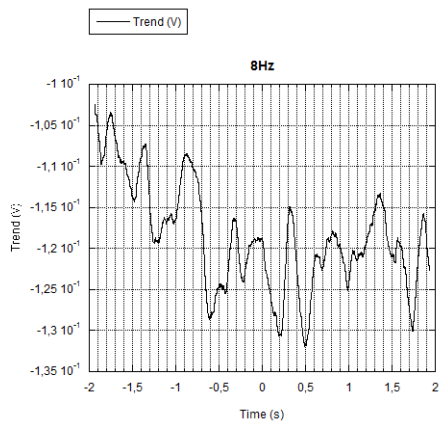
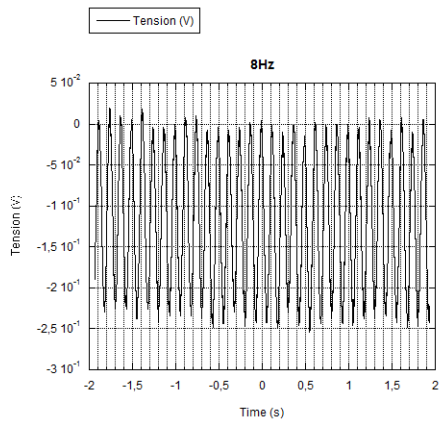


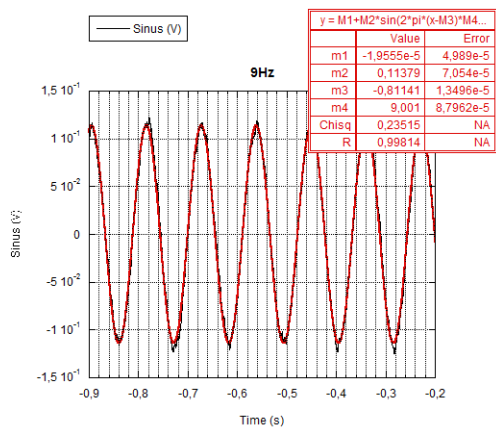
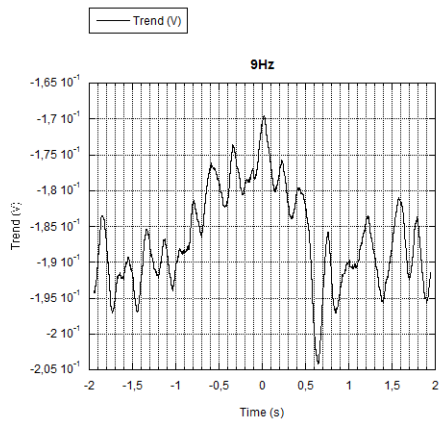
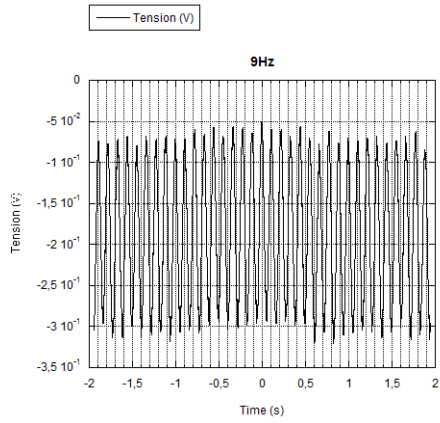


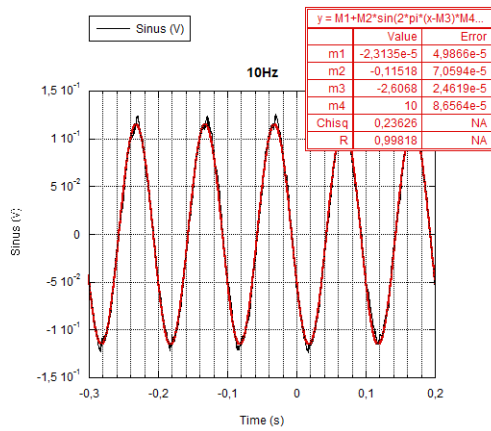
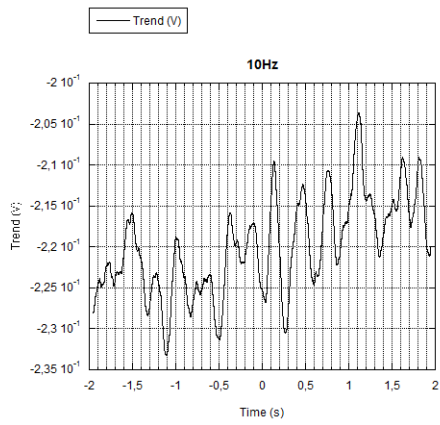
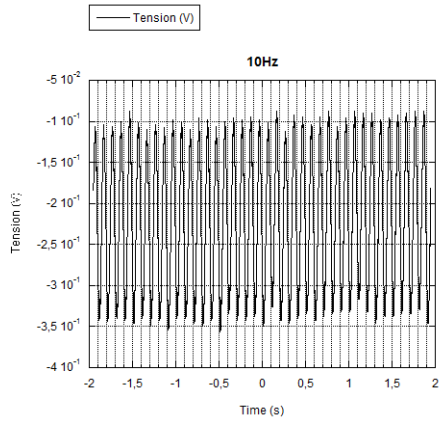


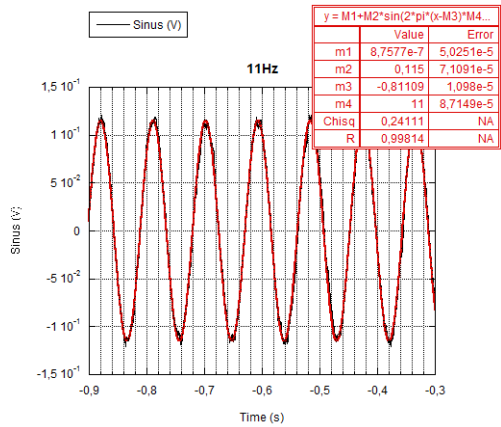
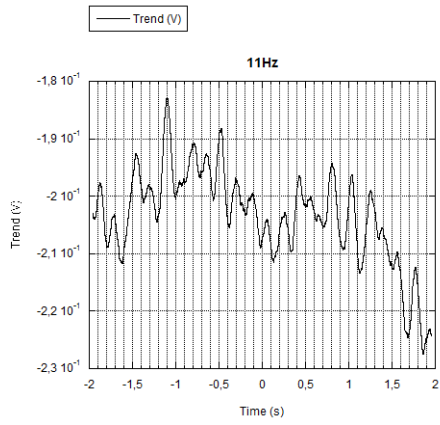
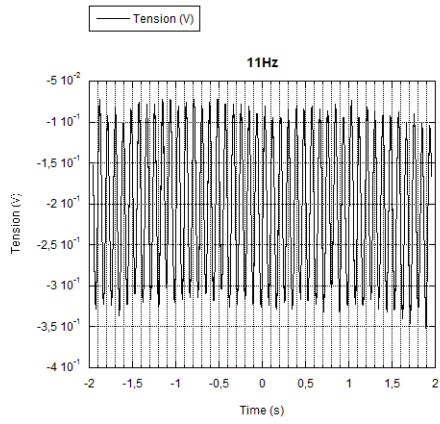


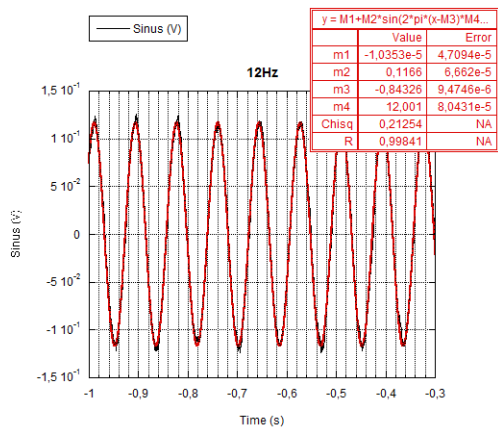
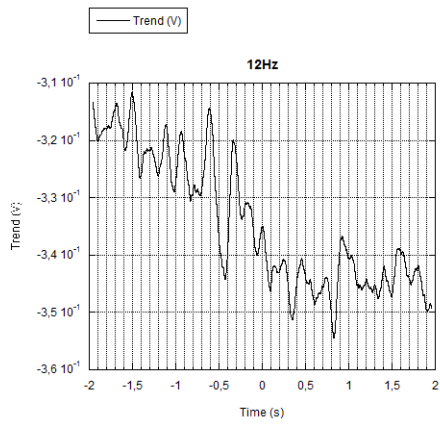
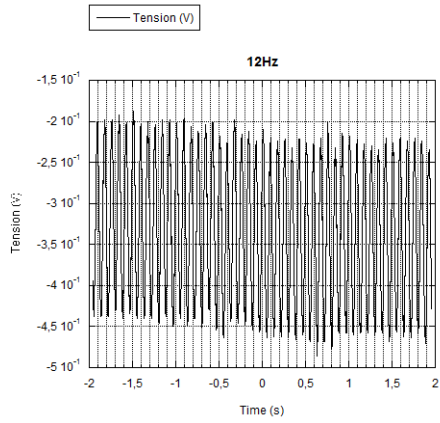


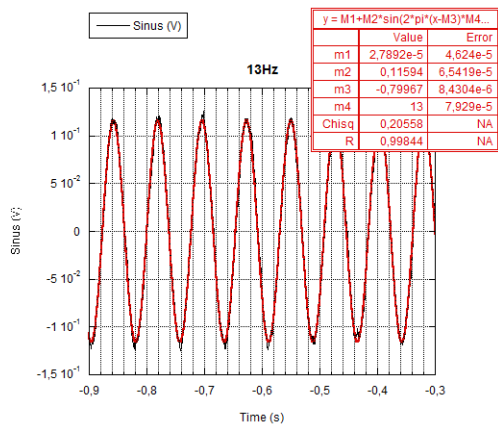
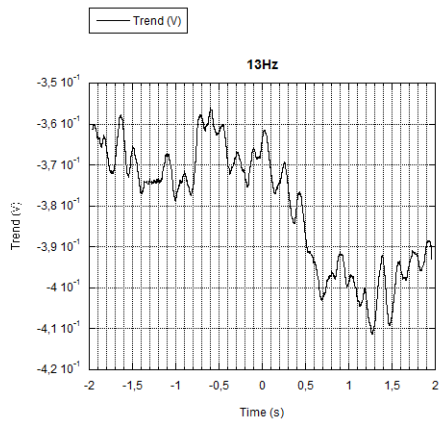
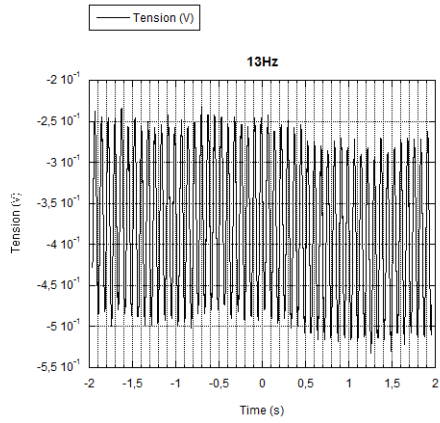


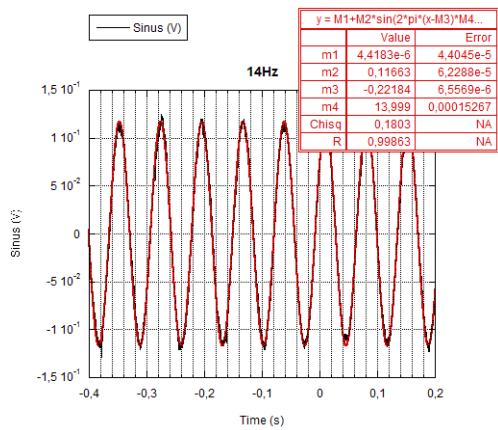
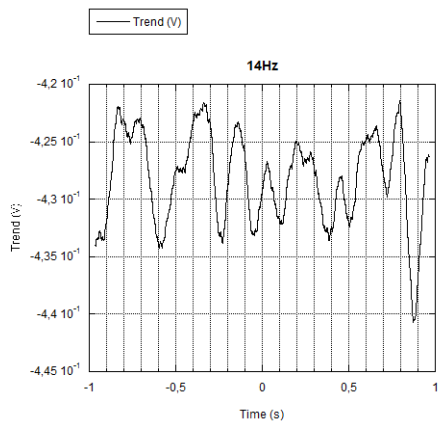
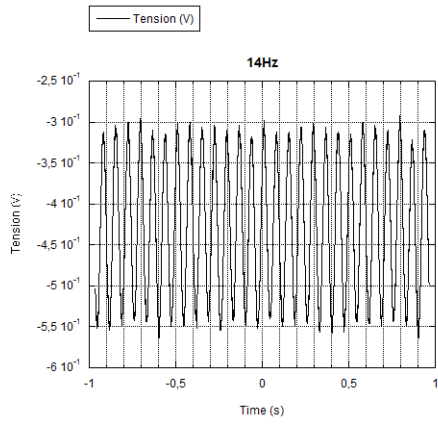


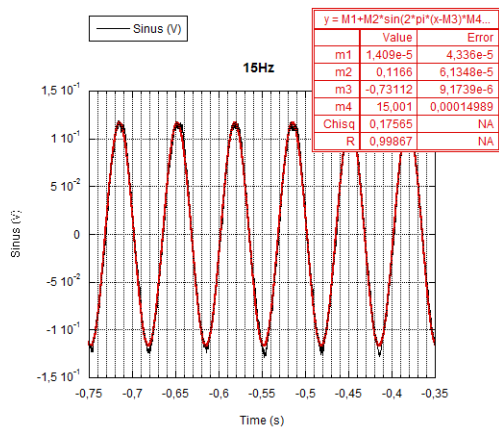
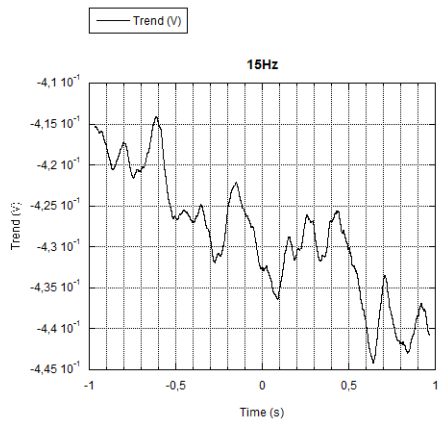
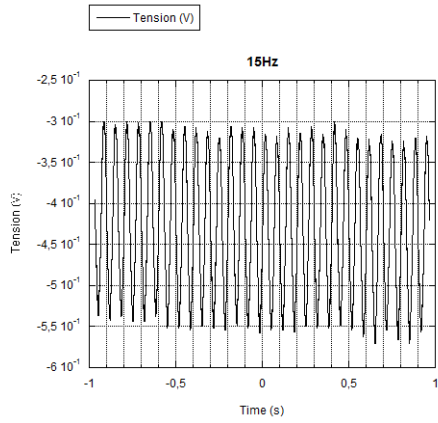


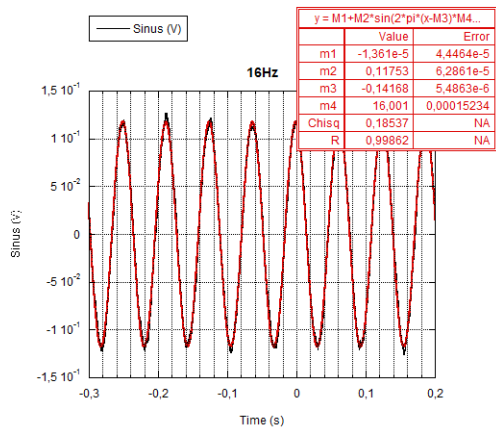
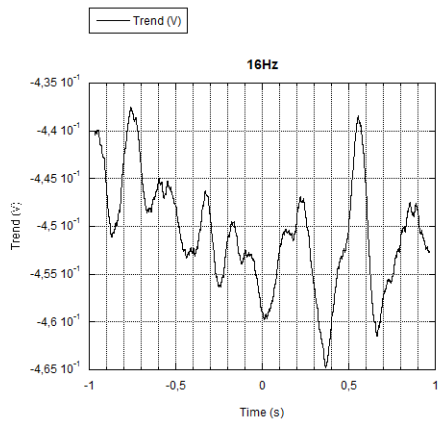
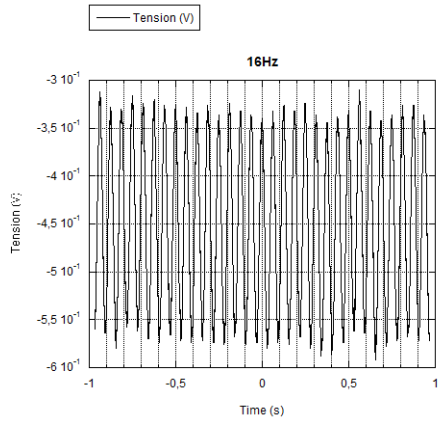


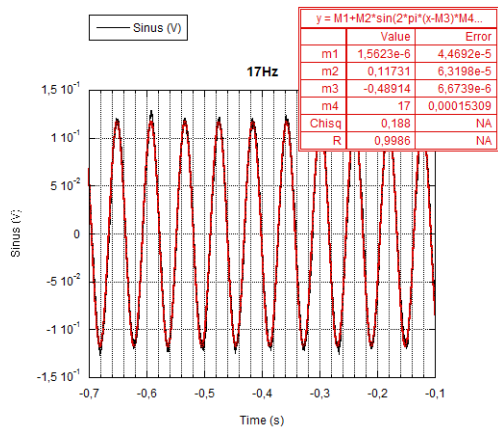
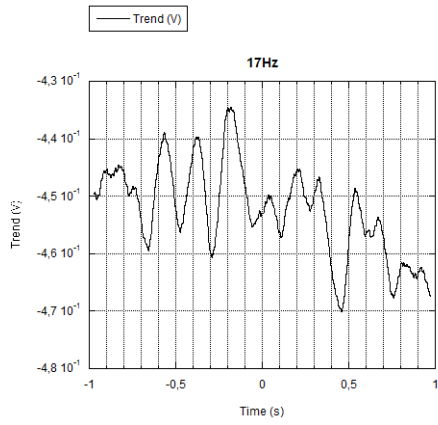
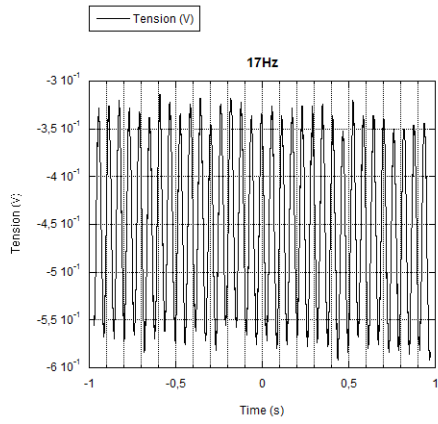


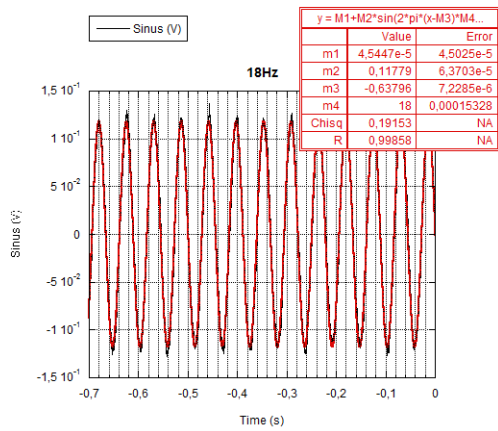
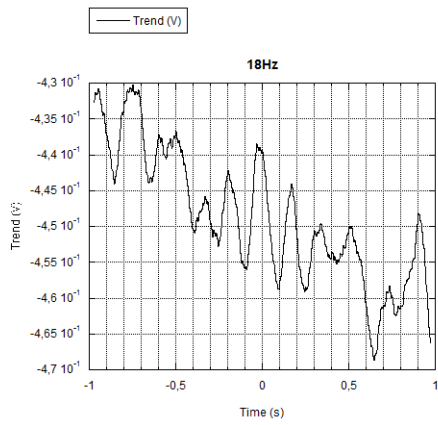
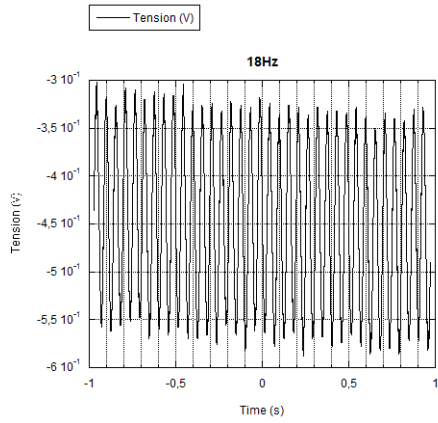


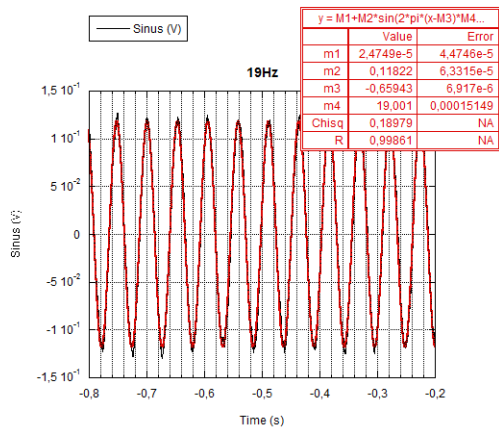
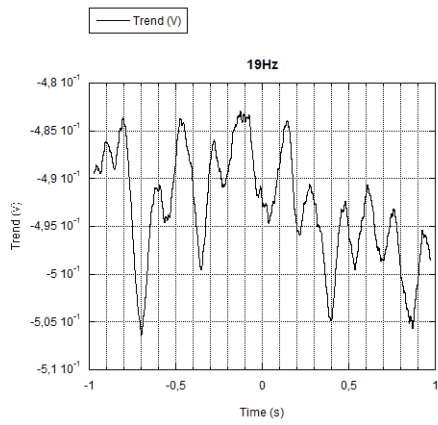
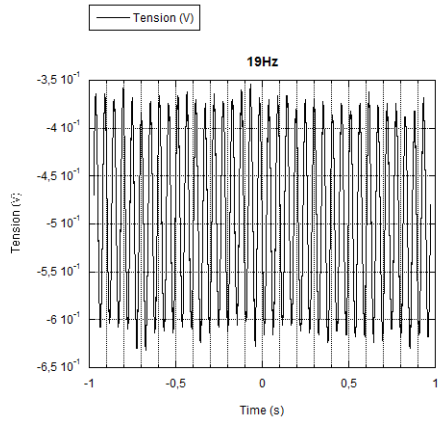


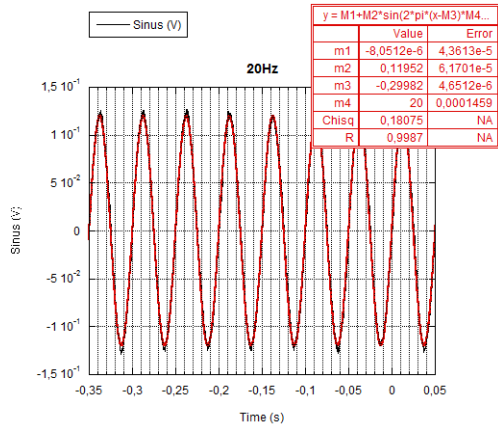
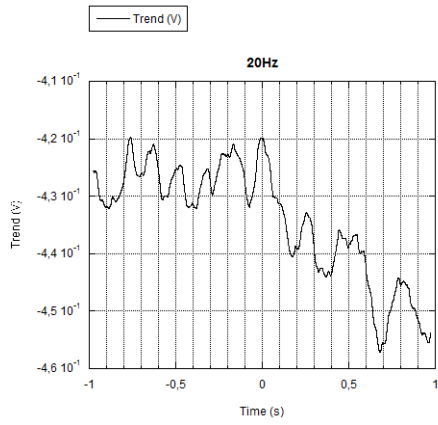
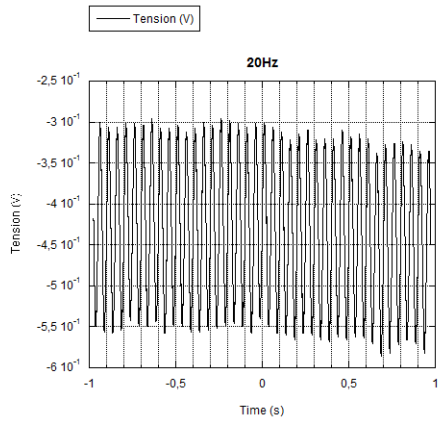












Chapter 5

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