

**theremino**  
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System Theremino

# Theremino DAA

version 4.2

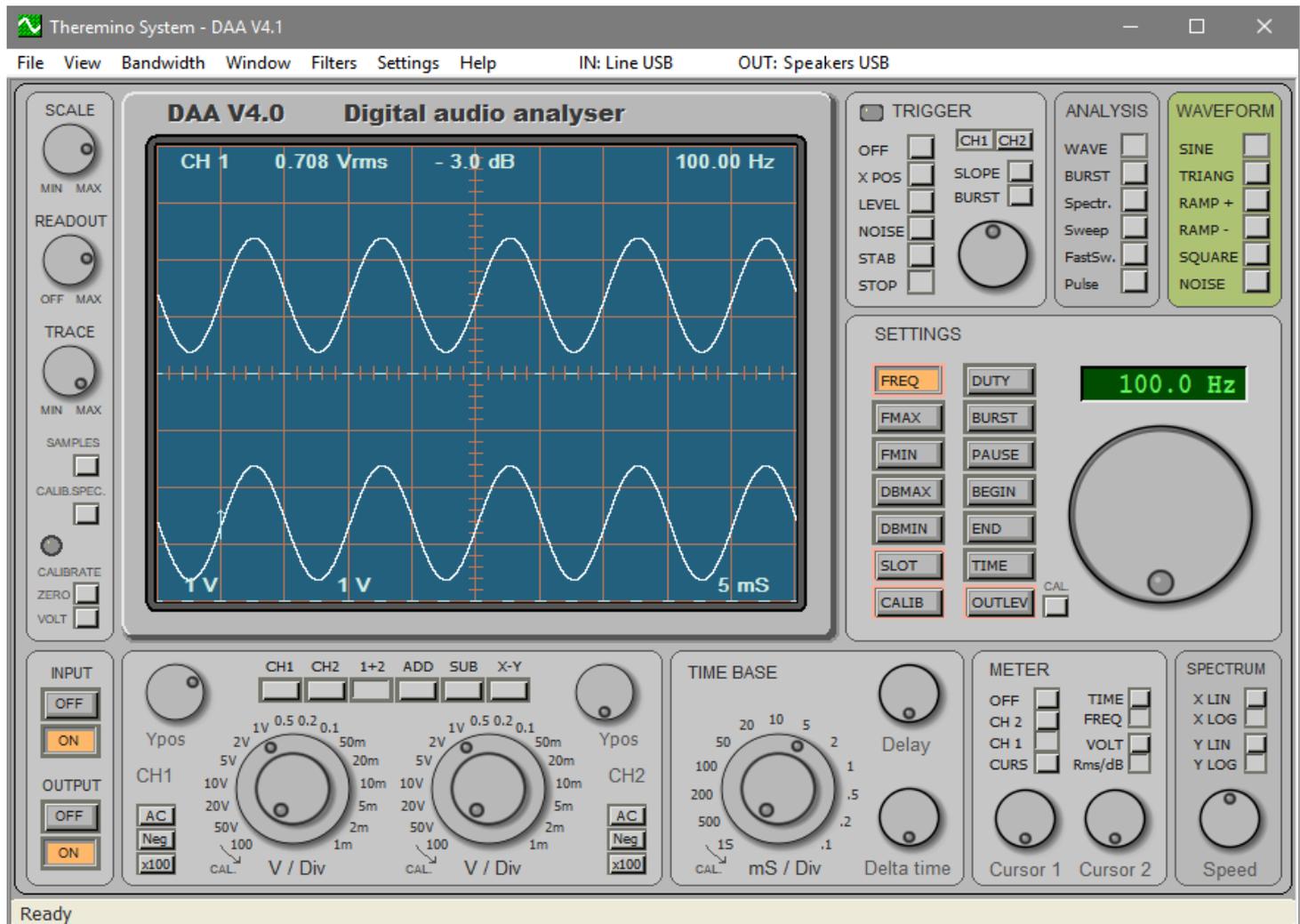
## Instructions

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<https://www.theremino.com>

<https://www.theremino.com/downloads/uncategorized#daa>

# The application DAA



*The DAA is a measuring instrument and test equipment for audio, also usable as a two-channel oscilloscope for frequencies up to a few tens of Khz.*

Like all applications of Theremino system, even the DAA is an application "portable". The "portable" applications do not require installation and does not change anything outside of the folder where they are. E 'can then copy them from one folder to another or from one computer to another.

With the "portable" applications, the operating system is not changed and the installation and uninstallation operations are simplified.

## ◆ Installation

Copy "Daa.exe" and the folder "Docs" in any folder.  
Then start the "Daa.exe" file.

## ◆ uninstalling

Delete all the files of the DAA.

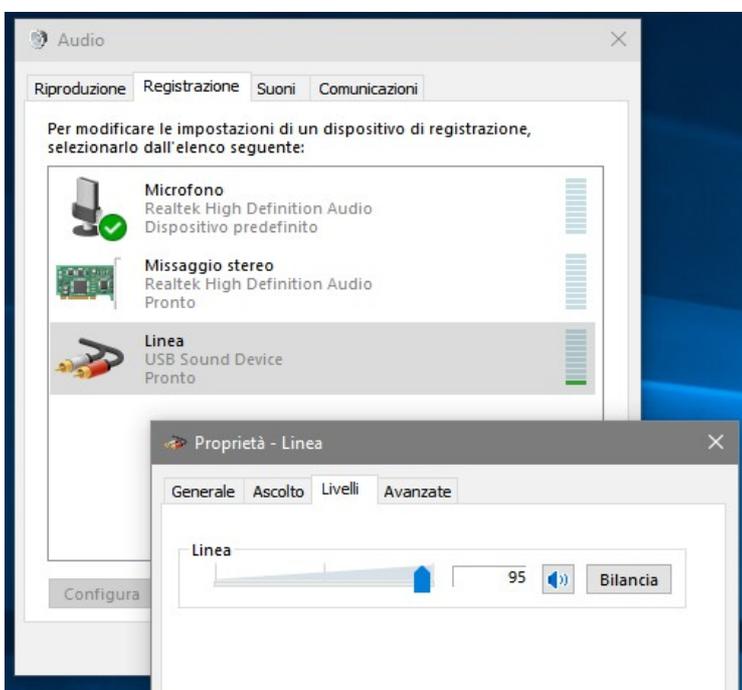
# Accuracy and minimum and maximum voltages

Using a USB sound card that costs less than ten Euros and with a simple change, which is done in two minutes (see the file "[DAA\\_V4\\_InputDevices](#)"), You get two input channels and two output, all coupled in continuous.

Features:

- ◆ precise adjustment of the tensions that allows measuring as a tester.
- ◆ Precise measurements of DC voltages, the mean value and the effective value.
- ◆ fast and accurate calibration, repeatable without requiring instrumentation.
- ◆ Precision and accuracy within a millivolt, and even better than a tenth of a decibel.
- ◆ Input impedance of 1 mega ohms (which is the standard for oscilloscopes)
- ◆ Possibility to measure voltages up to +/- 70 V (with respect to ground).
- ◆ With the addition of a 10: 1 probe (nine resistors 1 mega in series), you can measure voltages up to +/- 700 V
- ◆ With the addition of a high voltage probe, you can measure voltages up to several thousand volts.

*The minimum and maximum voltages measured depend on the input level adjustment "LINE IN" and are listed in the table on page 8, the file "[DAA\\_V4\\_InputDevices](#)".*



The level adjustments are made in panels that you see here on the left. **After changing the level you have to recalibrate.**

The calibrations are stored in the "DaaMainRegulations.ini" file and are automatically reset every time you throw the DAA program.

The calibration changes slightly with the change of temperature. In the first minutes of zero operation moves a few millivolts.

After half an hour of operation the temperature stabilizes, and from then on the calibration remains accurate within a fraction of a millivolt.

***If you change the input device, or recording level, calibration is no longer valid and you have to repeat it, as explained in the following pages.***

# The commands of the display panel



<b>SCALE</b>	Adjust the brightness of the grid (grid of the display)
<b>READOUT</b>	Adjust the brightness of the display written
<b>TRACE</b>	Adjust the brightness and color of the trace on the display
<b>SAMPLES</b>	View samples of all spectrum analysis
<b>CALIB.SPEC.</b>	Calibrate the baseline of all spectrum analysis
<b>CALIBRATE ZERO / VOLT</b>	Calibration of the zero and voltages

## **SAMPLES button**

This button only works with Analysis "Spectrum". "Sweep", "FastSweep" and "Pulse".

Pressing viewing the samples on which is based the analysis of the spectrum.

When making "Spectrum" type analysis. "Sweep", "FastSweep" and "Pulse" you can press this button and control the amplitude of the signals. The signal to be displayed must be large as much as possible (to act on the amplifier volume of Outlevel and the system mixer), but its tips must not be truncated.

If the amplitude of the signals is low there will be too much noise that will disturb the measurements, if it exceeds the maximum level tolerable by the ADC, the signals will be truncated and the measurement will be distorted.

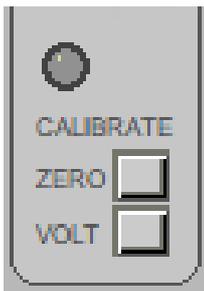
## **CALIB.SPEC button.**

This button only works with Analysis "Spectrum". "Sweep", "FastSweep" and "Pulse".

Pressing makes flat the response curve and then creates a reference on which to perform comparison measurements. The CALIBRATE LED turns green, indicating that there is the reference calibration. Pressing this button again, the LED goes off and the reference calibration is disabled.

If you change FMAX, FMIN, or TIME OF ANALYZES the reference is no longer valid, then it is automatically disabled and the LED turns off.

# Calibrations and ZERO VOLT



To obtain accurate measurements must be calibrated before the zero and then the scale. These two operations must be performed every time you change input device, input level or measuring probes.

*If you choose CH1 or CH2 will not be calibrated both channels, but only one chosen. Calibrate only one channel could be used to calibrate a high-voltage probe, but normally you should be careful to choose a "1 + 2".*

## Calibrate the reference (ZERO)

First you prepare a special jack, welding together the three poles and is preserved into the "MIC", ready for the zero calibration.



To calibrate the zero threading the special hall Jack "LINE IN", you press the "ZERO" button and wait a few seconds until the LED stops flashing.



## Calibrate the scale (VOLT)

Before you press the button VOLT you must have calibrated zero, and the calibration voltage must also have set in [panel SETTINGS](#).

Connect the inputs to an external calibration voltage, or remove the jack from the input "LINE IN" (and leave it open, without ferrules, so as to minimize interference). Then press the button VOLT and wait until the LED stops flashing.

## Use an external calibration voltage

The calibration voltage can be supplied from the outside with batteries or a power supply.

It measures the external voltage with a good digital multimeter, and is set in the SETTINGS panel. The external calibration voltage must be within the range of measurable voltages (see page 8 of the document "[DAA\\_V4\\_InputDevices](#)") And it must be connected to the inputs used (normally both the channels CH1 and CH2), before pressing the button VOLT.

## Use the internal calibration voltage

Using the internal calibration voltage is very convenient. It shoves the special Jack shorted and press "ZERO", then takes off the Jack and press "VOLT". **Within seconds, the calibration is done.**

The boards recommended by us have an internal voltage source. Before you press the button VOLT, you must set this calibration value in the SETTINGS panel.

*To accurately measure the internal calibration voltage, read the next page.*

# Measure the internal calibration voltage

The internal calibration voltage changes from one card to another, but once measured always remains valid. So just remember it or even write on a label behind the card.

The internal calibration voltage that is usually 1.35 to 1.45 volts. According to the data-sheet of the CM6206 chip that tension could also be 2.2 volts, but of all the boards we tested was less than 1.45 volts. If you want great accuracy you can then set up a voltage of 1.4 volts and is accurate to within a +/- 5% should get.

It is advisable, however, to accurately measure this voltage, using a good digital multimeter, with high input impedance (usually 10 mega ohm) and adjusted on a low scale (probably 2 volts DC).

While this measure is good not generate signals. Close the application or DAA press "OUTPUT OFF". Then connect the tester to the jack "FRONT OUT" (positive signal to ground and negative).

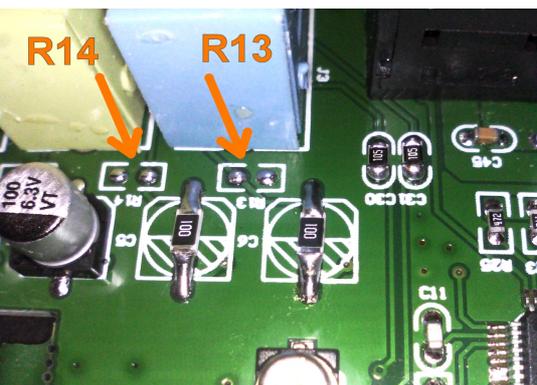


Even the LINE IN inputs are polarized with the same voltage, but not use them. Their high impedance (1 mega ohm) and the tester impedance (usually 10 mega ohm) would cause a measurement error of ten per cent, then about 140 millivolts less.

By measuring the voltage on the "FRONT OUT" outputs you get to an accuracy of a few millivolts.

The residual errors are dependent on two factors:

- ◆ Small differences in voltage between the two output channels (a few millivolts)
- ◆ Voltage drop (about 3 mllivolt) on the 100 ohm output resistors, caused by the resistors R13 and R14 47k, which have not been removed from the card.



To minimize the first error, it is advisable to join together the two output channels. Or to measure them separately and take the average.

To eliminate the second error is best to remove the resistors R13 and R14, as explained in the document "[DAA\\_V4\\_InputDevices](#)" On page 5.

## Refine the calibration voltage

To locate the calibration voltage with greater accuracy, measure a known voltage (possibly close to the maximum of the current measurement scale). Then it remodels the calibration value and repeats the calibration. Repeat several times until it fit properly tightened.

The calibration voltage so retouched remain valid for the future and there will be no need to measure it. You may want to write it back to the board.

# Calibrate the high voltage probes



The high voltage probes are described in the file "[DAA\\_V4\\_InputDevices](#)", From page 10 onwards.

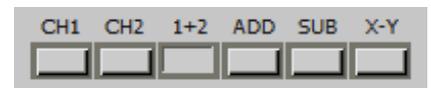
The calibration of high voltage probes is not different from the normal calibration, but there

are some caveats to consider.

## Choose the channel for probe

Usually you use a single high-voltage probe, for which we will calibrate a single channel, and not both as you normally do.

So before you start the calibration, instead of "1 + 2" We choose "CH1" or "CH2", depending on the channel which connects the probe.



## Calibration "ZERO"

To calibrate the probes **You do not use the special Jack** (Shown in page [Calibrations and ZERO VOLT](#)), But connecting the mass to the selected channel, **through the probe**.



## Calibration "VOLT"

The second phase of calibration is entirely identical to that explained in the page [Calibrations and ZERO VOLT](#). You can calibrate both the internal calibration voltage, which with a voltage supplied from the outside.

However, to calibrate the high voltage probes the internal voltage is too low and you would get an inaccurate calibration. The internal voltage is only one or two volts, i.e. from one hundredth to one-thousandth of full scale that are normally used with the probes.

**So to calibrate the high voltage probes is always better to use an external voltage high enough.**

The external voltage must be a DC voltage, stable enough, and should be towards the full scale flow that is being calibrated. But, in the absence of anything better, we'll do a few tens of volts.

For high flow rates you could use the 400 .. 500 volts, provided by a Geiger Adapter or from a feeder for the rooms to ions. These power supplies are simple and easy to build. You could prepare one to be used to calibrate the probes.

**To calibrate the external voltage is measured with the tester. It keeps connected to the tester so as not to vary by disconnecting the voltage. It sets the calibration voltage in [panel SETTINGS](#), And press "VOLT".**

# Calibrations default

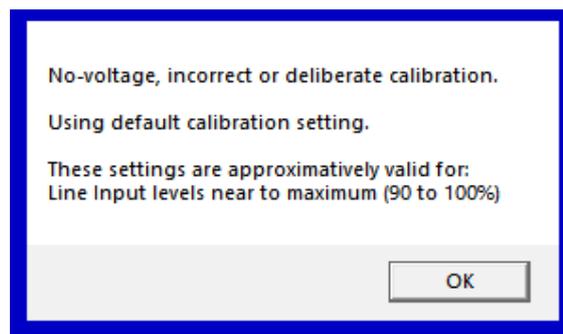
In previous pages we explained how to calibrate the sound card with the changes recommended by us.

Other devices (such as the microphone or the stereo Mixing) can not be calibrated because you can not connect their input to ground, and you can not even provide a reference voltage.

For these devices we recommend:

- ◆ Resetting the input signal (to be silent in the case of the microphone and off each signal generator in the case of Mixing Stereo or other software inputs)
- ◆ Press the "ZERO" button and wait a few seconds until the LED stops flashing.
- ◆ Press the "VOLT" button and wait a few seconds.

At this point it should open a message that informs that a default calibration is carried out.



This calibration is valid to make voltage measurements, but can be used for measures related to spectrum analysis and to see the waveforms.

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## CAUTION

Each new calibration of an input device deletes the previous calibrations.

So this calibration lose the primary calibration, the input line. When you re-use the line input you will have to redo its calibration.

# Calibration of the output signal

When using the signal generator, in order to obtain accurate measurements, it must also calibrate the output signal.

## preventive controls

First of all you should check the "Settings / Audio devices" menu that the output levels of "Altoparlanti - USB Sound Device" are at maximum. Also check in to "balance" that all levels are at 100.

Then go down to zero the sliders of "Line" and "Microphone". **Warning're talking about the speakers panel, the level of "Line" panel is not touched, it must remain what it is used for calibration and ZERO VOLT.**

You could also set the quality to 48000 Hz. It is also advisable to check that they are not set "Sound Space" or other "improvements."

**Before performing the calibration of the output signal is the calibration Zero Volt and must have carried out, as explained in [This Page](#).**

## Carry out the calibration of the output signal

- ◆ Connect a stereo cable Jack - Jack in the exit "FRONT OUT" and "LINE IN input."
- ◆ Make sure it is selected IN: Line (USB) and OUT: Speakers (USB)
- ◆ Check that INPUT and OUTPUT are ON
- ◆ Set WAVE ANALYSIS =, = SINE WAVEFORM and SETTINGS = OutLev.
- ◆ Press the "Cal" button located in the PANEL SETTINGS, to the right of OutLev button.
- ◆ Wait a few seconds and the calibration is done. It may then check that the level in decibels measured in WAVE analysis corresponds to the one set in OutLev. If everything in place it should match exactly, within a tenth of a decibel, both CH1 to CH2.



## The cable to be used for calibration

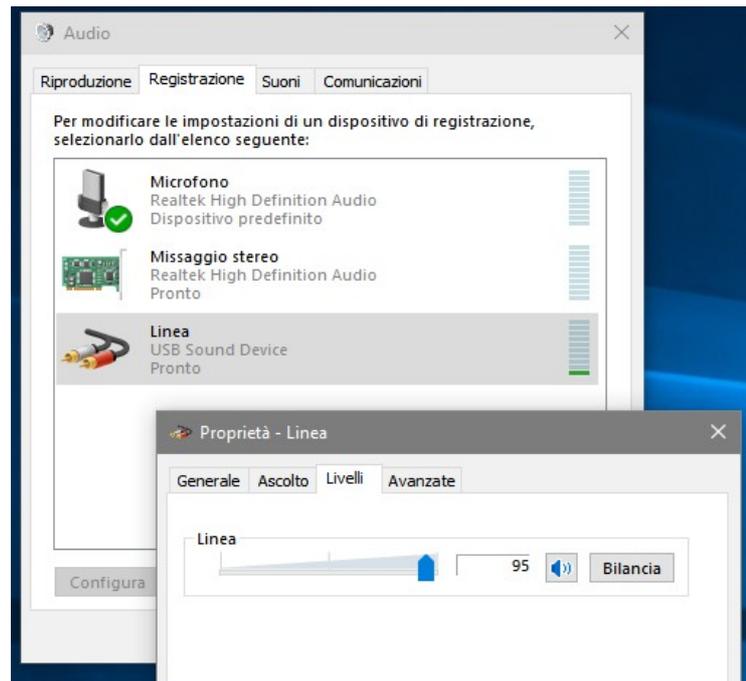
The output signal is stereo and the calibration adjusts individually the two outputs, so that the signal of each corresponds exactly to those set in decibels "OutLev" of the "SETTINGS" panel.

**So it's important to use a cable to connect the CH1 with CH1 and CH2 with CH2. In practice, the Jack tip connected to "FRONT OUT" must be connected with the tip of Jack connected to "LINE IN." Usually the cables are like that, but for safety, and especially in the case of manually soldered cables, you should check it with a meter.**

## Adjust the input level Line



- ◆ Use the "Settings" menu of the AAD
- ◆ Choose "Audio devices"
- ◆ Select "Registration"
- ◆ Double-click on "Line - USB Sound Device"
- ◆ Choose "Layers"
- ◆ Adjust the level from 0 to 100.
- ◆ Choose "Advanced"
- ◆ Set the quality to 48000 Hz.



**input levels to be set depending on the voltage values to be measured**

LEVEL	Input Voltage	
	minimum	maxim
96	+0.0 V	+2.5 V
93	-1.0 V	+3.5 V
88	-2.5 V	+5.0 V
86	-3.5 V	+6.0 V
80	-7 V	+9 V
75	-13 V	+15 V
65	-29 V	+30 V
58	-50 V	+50 V
30	-70 V	+70 V

This table is valid for cards with chip CM6206, modified as described in the file [DAA\\_V4\\_InputDevices](#)

In this short list we have collected the values to be set for the flow of most common use.

*In the file "[DAA\\_V4\\_InputDevices](#)" On page 8, there is a complete list of the minimum and maximum voltages measured, annotated and advice.*

**Tips**

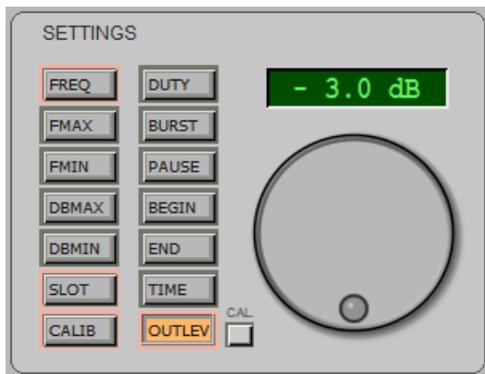
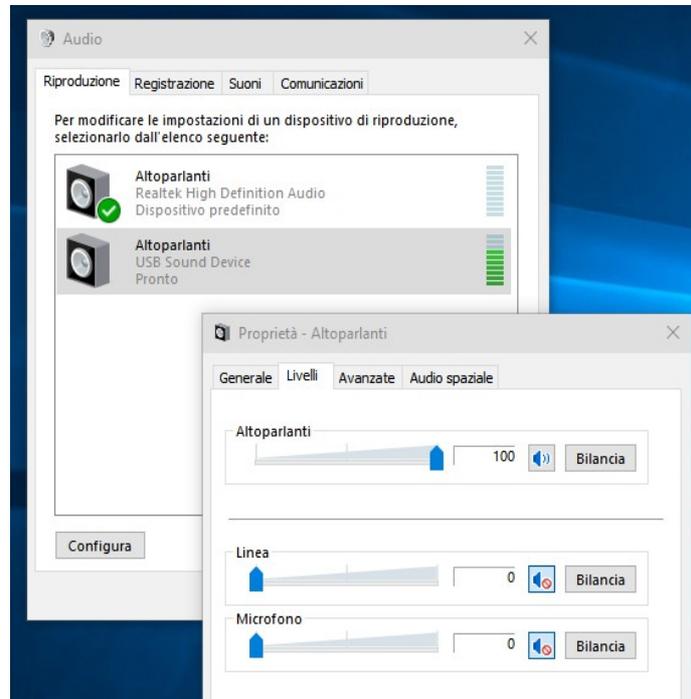
*To make more convenient the selection of inputs and outputs, it is good to disable the inputs and outputs that are not used (eg SPDF), and then remove the tick from "Show Disabled Devices."*

*Always pay attention, both for inputs and for outputs that are not enabled "Enhancements", "Spatial Audio", "Auto Levels" or other options like that.*

# Adjust the output levels



- ◆ Use the "Settings" menu of the application DAA.
- ◆ Choose "Audio devices"
- ◆ Select "Playback"
- ◆ Double-click on "Speakers - USB Sound Device"
- ◆ Choose "Layers"
- ◆ Adjust the level to 100
- ◆ Open "Balance" and check that both channels are 100
- ◆ Adjust the levels of line and microphone to zero
- ◆ Choose "Advanced"
- ◆ Set the quality to 48000 Hz.



The output level should always be kept at 100. The calibration of the signal generator, and the output level, are adjusted in the DAA application.

## Tips

*To make more convenient the selection of inputs and outputs, it is good to disable the inputs and outputs that are not used (eg SPDF), and then remove the tick from "Show Disabled Devices."*

*Always pay attention, both for inputs and for outputs that are not enabled "Enhancements", "Spatial Audio", "Auto Levels" or other options like that.*

# The panel INPUT OUTPUT



This panel enables and disables the reading of the input signals and the generation of the output signals.

## ◆ INPUT OFF

Disables sampling (the signal reading).

*This freezes command what has been sampled and allows to analyze it for a long time (in this condition the displayed signals originate from an internal buffer that contains the last 32 seconds sampled).*

*By moving the mouse cursor over the INPUT OFF and INPUT ON buttons, the status bar (located at the bottom), displays the minimum and maximum voltages measurable with the current calibration.*

## ◆ INPUT ON

Enables independent sampling on two channels (CH1 = Left / Right = CH2).

## ◆ OUTPUT OFF

Disable the signal generator.

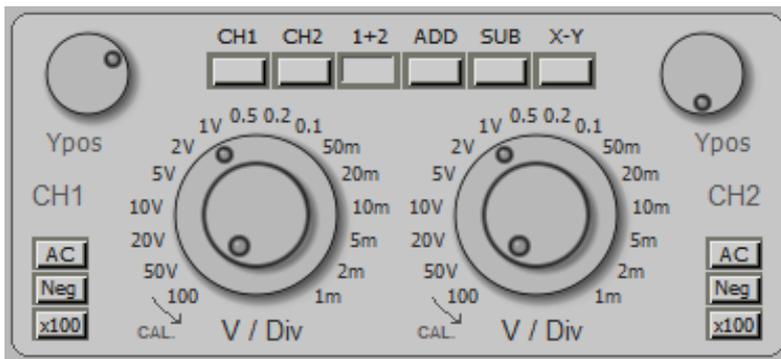
## ◆ OUTPUT ON

Enable the signal generator.

### **Notes to the signal generator**

*The generated signal is adjusted in amplitude by choosing SETTINGS - Outlevel and adjusting it with the large dial. With the WAVEFORM panel you choose the waveform.*

# The panel CH1 and CH2



This panel is part of the Oscilloscope section and you only use with WAVE BURST and analysis.

The panel is composed of a channel selection keyboard, by two knobs mV / Div, YPos by two knobs, and buttons AC - Neg - x100

The knobs and keyboards are doubled (left and right) and act on channels 1 and 2.

## CHANNEL SELECTION KEYBOARD

- ◆ **CH1**      The channel 1.
- ◆ **CH2**      The channel 2.
- ◆ **1 + 2**      View channels 1 and 2 simultaneously.
- ◆ **ADD**      Show only one track which is the sum of channels 1 and 2.
- ◆ **SUB**      Show only one track which is the difference of channels 1 and 2.
- ◆ **XY**      The channel 1 in X and channel 2 in Y.

## OTHER CONTROLS

- ◆ **yPos**      vertical track position.
- ◆ **V / Div**      Select the scale in millivolts per division of the display grid.  
The outer ring of twelve places select from 100 V to 1 mV.  
The potentiometer allows fine adjustment (turning hard left the actual value is indicated by the outer ring).  
The volts per division are also displayed on the display, at the bottom left.
- ◆ **B.C**      The DC component of the signal is eliminated.
- ◆ **Neg**      The signal is reversed (some sound cards the reverse).
- ◆ **x100**      The volts per square of all the flow rates are multiplied by 100.  
It uses this option to measure with high voltage probes, when the 100 volts per square are not enough.

# The panel TIME BASE



This panel is part of the OSCILLOSCOPE section and acts only with the WAVE BURST and analysis.

## mS / Div

This knob determines the scanning speed. The outer ring of twelve places select from S 1 to 0.1 mS. The potentiometer allows fine adjustment (turning hard left value effective is that indicated by the outer ring). The milliseconds per division are also displayed on the display, at the bottom right.

## Delay

Moves the display window along the entire 32-second buffer signal champions. If the DELAY knob is not zero, the "Delay" is lit in red and time value is displayed on the display (bottom) as: Dly xxx mS

## Delta time

It works only with CHANNEL SELECTION in position 1 + 2, ADD, SUB and XY. Move over time the channel 2 while maintaining the channel 1 fixed. The zero corresponds to a total rotation in a clockwise direction. If the DELTA TIME knob is not at zero, the "Delta Time" lights color red and the time value is displayed on the display (bottom) as: dT xxx mS

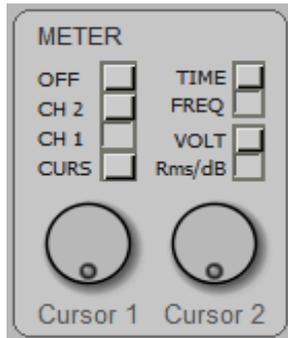


*Please note the words "Delay" and "Delta time". If they are lit in red color then the "Delay" knob and "Delta Time" are not zero and you may get strange display behavior.*

*For example, the timing (trigger) apparently could malfunction and the signal changes may appear late.*

*So (unless you want to use specially), these knobs should be always kept to zero.*

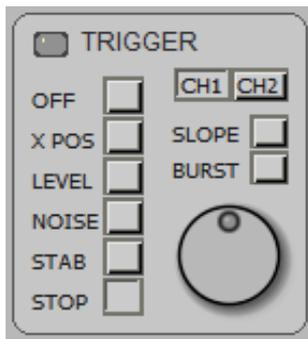
# The panel METER



This panel is part of the Oscilloscope section is only used with the WAVE BURST and analysis.

- OFF** All measurements are disabled.
- CH1** continuous measurement of the channel 1.  
On the top line of the display shows CH1, the average voltage and the voltage peak-to-peak, or the RMS voltage and decibels. On the right appears the time to cycle, or frequency.  
The measurement is independent of any adjustment, just that INPUT is ON.
- CH2** continuous measurement of the channel 2.  
On the top line of the display reads CH2, the average voltage and the voltage peak-to-peak, or the RMS voltage and decibels. On the right appears the time to cycle, or frequency.  
The measurement is independent of any adjustment, just that INPUT is ON.
- CURS** Enable the sliders to measure the voltage and the time between any two points of form the displayed waveform.
- TIME** The time measurements are displayed as time.
- FREQ** The time measurements are displayed as frequency.  
In this condition (using CH1 or CH2 METER always on the panel) is obtained a great frequency, accurate to the hundredth of a hertz and able to work with minimum voltages of a few millivolts.
- VOLT** The voltage is displayed as the average voltage and peak-to-peak voltage.
- Rms / dB** The veins tension displayed as RMS voltage and decibel (referenced to zero decibels which is conventionally considered to be 2.82 volts peak-to-peak)
- CURSOR 1** The CURSOR1 1 knob moves the cursor along the displayed waveform.  
The distance between the two cursors, in voltage and time, is displayed in the upper part of the display. When the cursor is to the far left is positioned to zero (which is also indicated with a dashed line).
- CURSOR 2** The cursor2 knob 2 moves the cursor along the displayed waveform. There distance between the two sliders, in voltage and time, is displayed in the upper part of the display. When the cursor is to the far left is positioned to zero (which is also indicated with a dashed line).

# The TRIGGER panel



This panel is part of the Oscilloscope section is only used with the WAVE BURST and analysis.

The trigger point (synchronizer), is represented by an arrow on the display pointing up or down depending on the condition of SLOPE, which moves to the right and to the left with XPOS, and in upward or downward with LEVEL.

When the INPUT is OFF in the arrow moves down and indicates the zero point of the stored signals of time.

- LED** It lights up if the TRIGGER acts (**Note 1**)
- OFF** The TRIGGER is disabled, and the waveform is presented continuously.
- XPOS** The knob adjusts the horizontal position of the TRIGGER point.
- LEVEL** The knob adjusts the vertical position of the TRIGGER point.
- NOISE** The knob adjusts the amount of noise tolerated by the TRIGGER (adjust for achieve maximum stability of the waveform).
- STAB** Adjust to obtain maximum stability of the waveform). This adjustment is normally held to a minimum and is only used for the type "Burst analysis" or enhance the timing of signals "difficult."
- STOP** The STOP voltage level is adjusted with the knob. (**Note 2**)
- CH1** The trigger and the spectrum analyzer using the signals from channel 1 (**Note 3**)
- CH2** The trigger and the spectrum analyzer using the signals from channel 2 (**Note 3**)
- SLOPE** Select the trigger on the rising edge or on the front of the signal down.
- BURST** Special operation to synchronize signals consist of pulse packages (**Note 4**)

**(Note 1)** Normally you get a good synchronism with Level = 50%, Noise = 25% and Stab to a minimum.

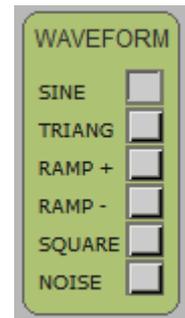
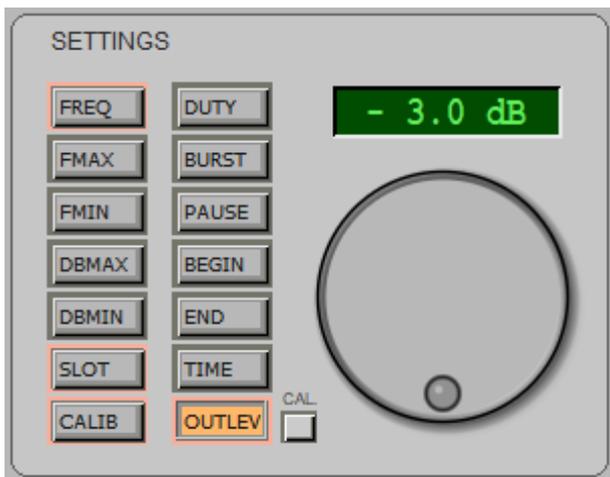
**(Note 2)** When the signal passes this level (both in positive and negative direction), the acquisition stops. For fine adjustments use CTRL or ALT and the mouse wheel.

**(Note 3)** The CH1 / CH2 switch, in addition to select the sync channel for the oscilloscope, also selects the input channel for all of spectrum analysis.

**(Note 4)** For BURST type waveforms you start adjusting Level = 50%, Noise = 50%, then rule Stab to the maximum and it is lowered until the LED lights up.

# Special measures

The ANALYSIS panels, WAVEFORM, SETTINGS and SPECTRUM, all located on the right side of the instrument, enable the special measurement functions not found in a simple oscilloscope.



## Signal generator (WAVEFORM)

It is available a generator of audio signals from 0.1 Hz to 22 KHz with six types of waveform including white noise and with possibility of burst (pulse packets).

## Spectrum analysis (Spectr.)

It allows to analyze a signal by dividing it into various components with respect to the frequency. This function can also be displayed in three dimensions (amplitude, frequency and time) by selecting Spectrum3D from the VIEW menu. The spectrum analysis used with the noise generator NOISE can measure the frequency response by canceling the effects of resonance but suffers very randomness of the noise and does not provide accurate measurements.

## type SWEEP Analysis

It allows to measure the frequency response with the SWEEP method (scanning frequencies over time with sinusoidal waves). The SWEEP is a rather slow but very precise method, however, is affected by the resonance effects of the devices under test and the environment in which the measurement is made. With reference calibration you can get to an accuracy of a tenth of a decibel.

# FAST-SWEEP and PULSE Analysis

These methods measure the frequency response quickly and accurately. The analysis is carried out with very short pulses that contain in equal measure all frequencies of the audio spectrum. The pulses from the filtered in analysis components are divided in frequency and is achieved in one fell swoop the entire spectrum. Also, if you start sampling immediately prior to the pulse arrival and ceases to be sampled before they arrive unwanted signals due to reflections in the environment can be excluded resonance and reflection phenomena.

The measurement must be carried out in a quiet environment and preferably with a microphone and a low-noise input pre-amplifier, the method is sensitive to periodic disturbances (eg mains interference).

All these factors mean that noise measurement dynamic is quite limited, and that the noise tends to distort the measurements especially in the lower part of the spectrum (below one hundred hertz).

When making measurements with FastSweep Pulse and check that the noise is enough below the curve displayed by putting OUTPUT OFF position, otherwise raise, if possible, the output level to overcome the background noise. To perform measurements on speakers is good to use a rather powerful external amplifier.

To partially overcome the abovementioned problems the DAA analyzer provides two FastSweep and Pulse optimized methods respectively for the upper and lower parts of the spectrum.

FastSweep uses a pulse that contains equal energy all frequencies and is affected by the effects of noise especially in the lower part of the spectrum.

Pulse uses a pulse sawtooth that starts steeply up to the maximum value and then with linear trend tends to zero, a pulse of this type contains much more energy at low frequencies and this disparity is compensated for during the analysis of the spectrum, Pulse affected by noise in the high part of the spectrum.

The method used to perform the spectrum analysis is the FHT Fast Hartley Transform that is faster and more accurate than FFT Fast Fourier Transform. All of fast processing methods are inherently linear nature and to get a logarithmic scale are converted mathematically.

Depending on the minimum and maximum frequency analysis of the Hartley transform is performed on 1024, 2048, 4096, 8192 or 18384 samples. If you select a minimum frequency below 50 Hz, the logarithmic scale X or X scale expanded very slightly slows down the speed of analysis.

For all types of spectrum analysis you can use the cursor Cursor 1 METER panel, to measure the value at a specific point of the curve.

## Mixer Settings

For all the measures it is necessary that the controls BASS and TREBLE Windows MIXER, are centrally located and 3Dstereo that the command is enabled. If you want to use these commands (for example, to evaluate their effectiveness), you have to keep the output volume to about twenty decibels below the maximum to prevent the cuts sound card the signal peaks.

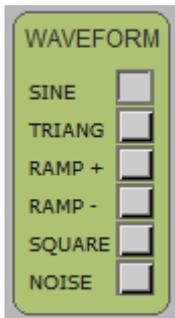
# The panel ANALYSIS



ANALYSIS With the panel determines the basic configuration of the analyzer.

- WAVE** Oscilloscope and signal generator.
- BURST** Showing oscilloscope type and signal generator to BURST (package pulses).  
The number of cycles that make up the package and the pause between packets is adjusted with SETTING - BURST and PAUSE.  
This type of analysis is used to highlight the resonance due to queues insufficient damping of speakers.
- Spectr** Spectrum analysis.  
The measuring scale is adjustable with SETTINGS / FMAX, FMIN, near Dbmax, DBMIN.  
The type of scale is selected by SPECTRUM, XLIN / xlog and Ylin / YLOG.  
The response speed is adjusted with SPECTRUM / Speed.  
The spectrum can also be displayed in three dimensions (amplitude, frequency and time) by selecting Spectrum3D from the VIEW menu.
- Sweep** Spectrum analysis in frequency sweep with a sinusoidal generator.  
Remember to place INPUT OUTPUT ON and select SINE WAVEFORM in position.  
The scanning time is adjustable with SETTINGS / TIME.  
The measuring scale is adjustable with SETTINGS / FMAX, FMIN, near Dbmax, DBMIN.  
The type of scale is selected by SPECTRUM / XLIN / xlog and Ylin / YLOG.
- FastSweep** impulsive Analysis of the frequency response optimized for maximum dynamic (Minimum noise) on the upper part of the spectrum.  
The measuring scale is adjustable with SETTINGS - FMAX, FMIN, near Dbmax, DBMIN.  
The type of scale is selected by SPECTRUM / XLIN / xlog and Ylin / YLOG.  
The response speed is adjusted with SPECTRUM / Speed.
- Pulse** impulsive Analysis of the frequency response optimized for maximum dynamic (Minimum noise) on the lower part of the spectrum.  
The measuring scale is adjustable with SETTINGS / FMAX, FMIN, near Dbmax, DBMIN.  
The type of scale is selected by SPECTRUM XLIN / xlog and Ylin / YLOG.  
The response speed is adjusted with SPECTRUM / Speed.

# The panel WAVEFORM



With this panel will select the waveform of the signal generator.

**SINE**      Sine wave.

**TRIANG**      Triangular wave.

**RAMP +**      Wave positive ramp.

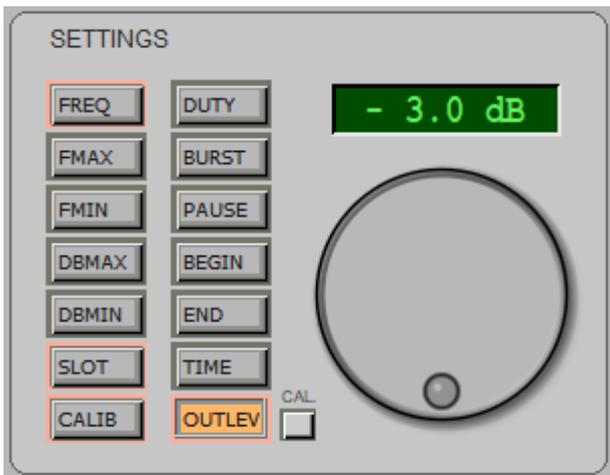
**ramp-** negative ramp wave.

**SQUARE**      Square wave.

**NOISE**      White noise.

These waveforms are obtained only if OUTPUT is ON and if WAVE ANALYSIS is in position, or BURST Spectr.

# The SETTINGS panel



This panel is composed of a selection keyboard and a knob and a display to adjust and display the selected value.

Depending on the type of analysis, active buttons have the edge illuminated in orange color.

**Display** Show the selected value. Click on the display with the left mouse button to edit the value, then press the ENTER key or click on the knob to confirm the value.

**Knob** This knob acts with an effect proportional to the speed rotation allows rapid changes but also precise adjustments.

---

**FREQ** Frequency of the signals generated from 1 Hz to 22 KHz.

**FMAX** minimum viewing frequency from 1 Hz to 22 KHz (SWEEP, SPECTRUM and PULSE). FMAX can not fall below FMIN.

**FMIN** maximum viewing frequency from 1 Hz to 22 KHz (SWEEP, SPECTRUM and PULSE) FMIN not exceed FMAX.

**near Dbmax** top of the scale limit for SPECTRUM, SWEEP and PULSE.

**DBMIN** scale low limit for SPECTRUM, SWEEP and PULSE.

**SLOT** If you set a number from 1 to 990, the three values visible in the upper part of the display They are continuously sent to three consecutive slots, starting from the set.

**CALIB** of the calibration voltage value (external or internal). To see [these pages](#).

---

**DUTY** pulse / space ratio for the square wave (SQUARE).

**BURST** Number of cycles that make up the pulse packet (BURST).

**PAUSE** Number of cycles that make the break between the pulse packets (BURST).

**BEGIN** Beginning of sampling (only for FastSweep and Pulse) (see Notes Begin - End) If you do not know how to adjust it is good to keep it to a minimum, ie zero.

**END** End of sampling (pears only Pulse) (see Notes to Begin - End) If you do not know how to adjust it to keep it at the maximum value, that is 200mS.

**TIME** Duration of scan in seconds (SWEEP). Use a long time to make the lower part of the spectrum more accurate frequencies.

**OutLev** Output level for the signal generator and for the Sweep pulses, and FastSweep Pulse. You can make small precise changes, the tenth of a decibel, which It would not be possible with the Windows Mixer.

# The output slot

If as SLOT (see previous page), you set a number from 1 to 990, the three values visible in the upper part of the display are sent continuously (50 times per second), in three consecutive slots, starting from that set.

The magnitudes which are measured and sent to the slot, and depend on the settings of ANALYSIS METER panels.

If, for example, you set SLOT = 1, here are some examples of the many possible:

- ◆ = Vms Slot1, Slot2 = dB, Slot3 Hz = (CH1 or CH2 depending on the settings)
- ◆ Slot1 = Vdc, Slot2 = Vpp, Slot3 Hz = (CH1 or CH2 depending on the settings)
- ◆ Slot1 = Vdc, Slot2 = Vpp, Slot3 = Sec (CH1 or CH2 depending on the settings)
- ◆ Slot1 Vpp = CH1, CH2 = Slot2 Vpp, Slot3 Hz = (Measured with the sliders)
- ◆ Slot1 = Vpp, dB = Slot2, Slot3 Hz = (With the cursor of the spectrum analysis)

*The units of measurement of the values sent to the slots are not the ones displayed (such as millivolts or milliseconds), but always those fundamental: Volt, Seconds, Hertz and Decibel.*

## Using the measured values

Once the measured values are in the slot, all other Theremino system applications can read them and use them in various ways.

For example, you may read the values and write them in a file at regular time intervals (with Theremino Datalogger or other similar applications).

Or you could develop values with Theremino Automation and generate events or error messages if they leave each values from an allowable range.

Or you could filter the data with Theremino Filters, send them to a Master Output Pin to control the servo mechanisms, etc ...

## Make more different measures, including using more than one simultaneously DAA

You can also open multiple DAA simultaneously and send different measures at different slots. In this case you must prepare more folders, each with a DAA, so as to maintain separate settings for each (see [these pages](#)).

If the application is minimized DAA its load on the CPU it is very low (less than 0.5%) and if the closes minimized, at the next start will reopen minimized.

# The SPECTRUM panel



Scale Settings for: Sweep, Spectr, PulseLo and PulseHi.

**XLIN** Frequency scale (linear axis X).

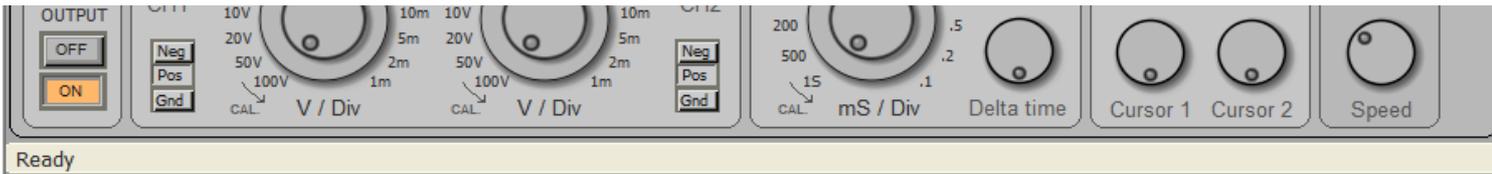
**xlog** Scale frequency (X-axis) logarithmically.

**Ylin** Scale of the amplitudes (Y linear axis).  
The linear scale is calibrated in volts.

**YLOG** Scale of the amplitudes (Y axis) logarithmically.  
The logarithmic scale is calibrated in decibels referenced to 1 volt effective (2.82 volts peak to peak)

**Speed** Rapid response of the display.  
Speed adjusting to a low value (counterclockwise), the average is carried out on a long time, improving the measuring accuracy and decreasing the noise.

# The status bar



The bottom row of the application is the status bar.

By scrolling the cursor commands appear on specific instructions for each command.

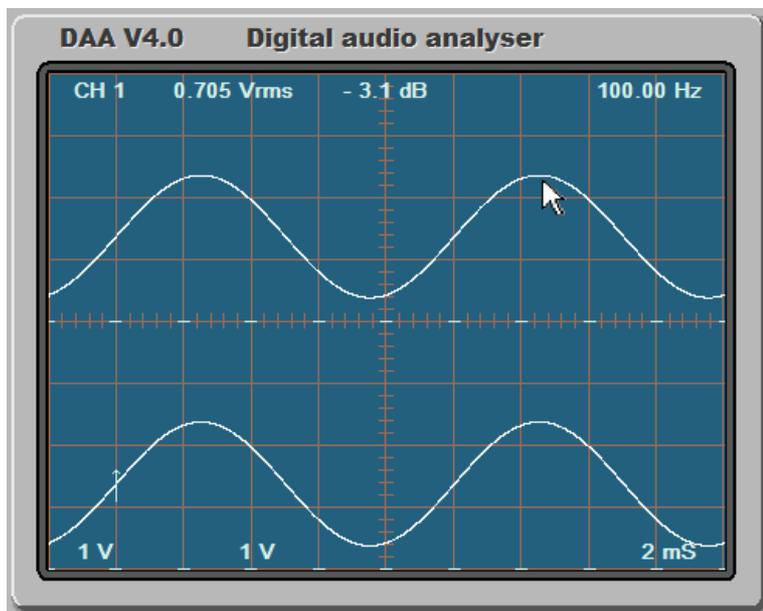
Pulse generator with Fourier spectrum analysis

Fast sweep generator with Fourier spectrum analysis

Start sampling correction: 3.15 mS

Sampling time: 200 mS - Lower bandpass: 5 Hz - Sound travels: 68.2 m

End sampling ( each mS is about 3 meters of distance )



By scrolling the cursor to appear on the display the values of voltage and time relative to the point indicated by the cursor.

If you are viewing a spectrum analysis then the decibels and frequency are displayed.

CH1 = 6.33 V    CH2 = 2.343 V    TIME = 14.437 mS

By moving the mouse cursor over the INPUT OFF and INPUT ON buttons, the status bar displays the minimum and maximum voltages measurable with the current calibration (see this page).

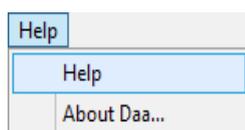
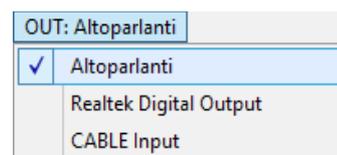
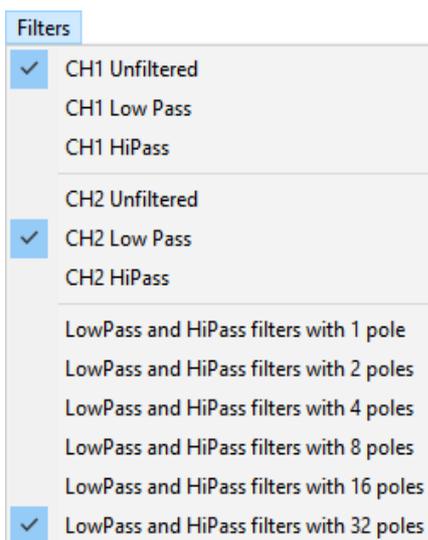
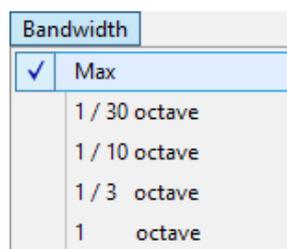
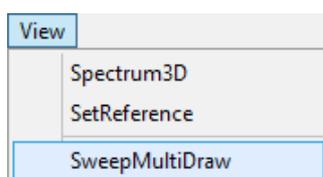
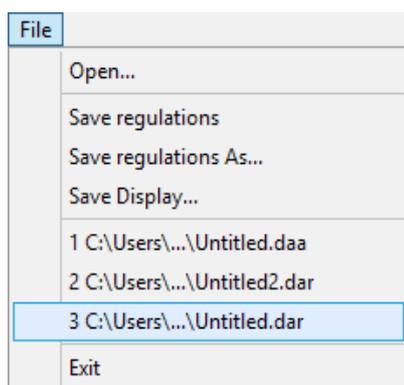
--- CH1 voltage range = -4.3 to 6.5 volt    --- CH2 voltage range = -4.3 to 6.5 volt

# The bar Menu



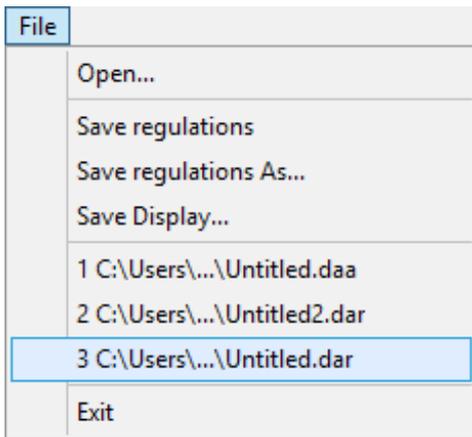
The menu bar provides access to files and to choose global functions and configurations, related to the entire application.

The last two menu items, relating to audio inputs and outputs, are variable depending on the sound card and the language of the operating system.



On the following pages menus are explained one by one.

# The File menu



**Open** Reads DAA file.

All analyzer settings are therefore present in the DAA file quickly restore a prepared configuration and saved in can previously.

**Save Regulations** Save the analyzer status on DAA currently selected file.

The current file name is visible in the title bar (top left).

If no file is selected, it opens a dialog box to choose it.

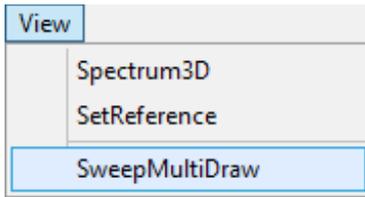
**Save Reg. As ...** Save the analyzer status of a selected file or a new file.

**Save Display** Save the image on the display in BITMAP format.

**Files List** The recently used files.

**Exit** It closes the DAA application.

# The View menu



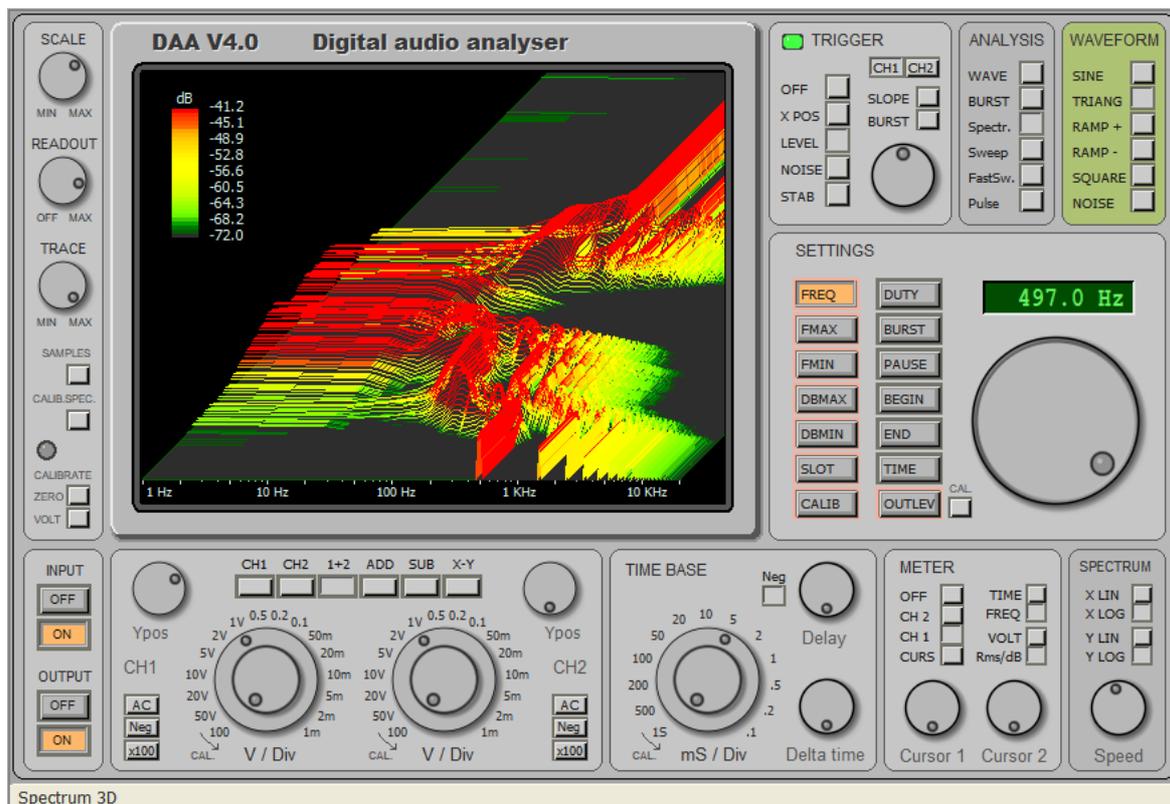
**Spectrum3D** and Displays the spectrum analysis in three dimensions (amplitude, frequency and time).

**SetReference** Freeze the shape of this waveform on the screen to use as a comparison and reference.

**SweepMultiDraw** Only it used with the SWEEP analyzes.

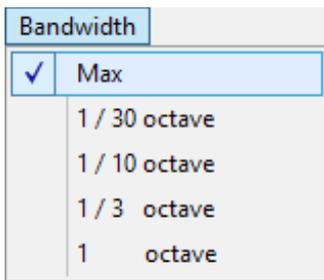
Press the right mouse button to start a new sweep.

The previous sweep are not deleted.



Display example that you get by selecting "3D Spectrum"

# The Bandwidth menu



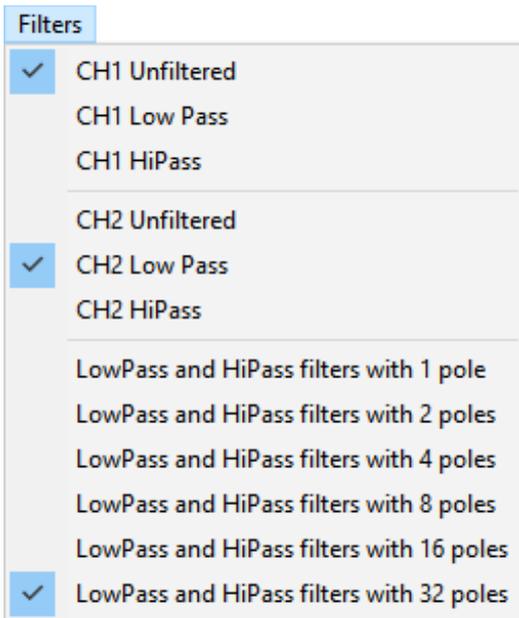
Selection of bandwidth, for SPECTRUM and PULSE analysis.

# The Window menu



Selection of the sampling window, for SPECTRUM analysis.

# The menu Filters

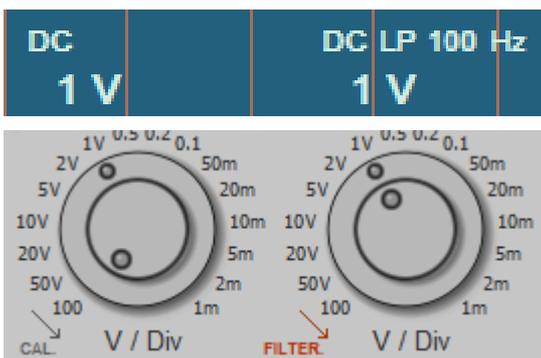


With these controls are assigned to the low-pass filters and high pass to the two input channels and choose the steepness of the filters.

The filters are digital but have been designed to behave exactly as simple RC filter.

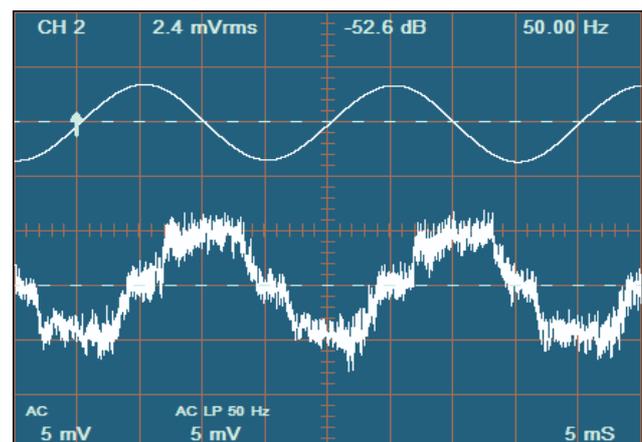
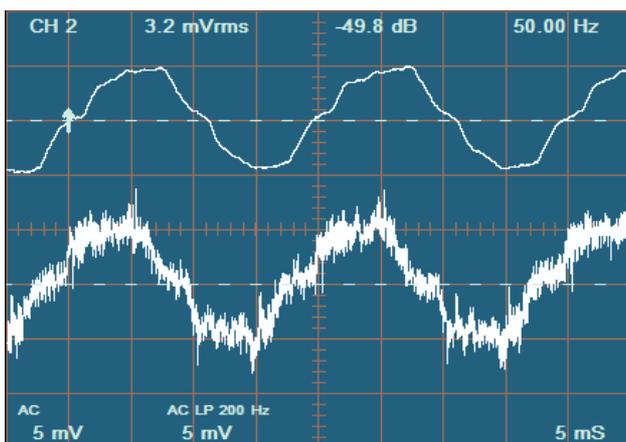
It increases the slope of the filter by increasing the number of RC cells (1 to 32).

When the cells are more than one software recalculates the values of each cell, so as to match the cutoff frequency of the overall filter to the value that is set manually. Furthermore, the digital implementation allows to obtain an ideal filter, in which the following cells do not load the previous ones, as would be the case with capacitor filters and real resistance.



The type of filter (LP or HP), and the cutoff frequency, are indicated in the display.

When a channel is filtered shows "FILTER" in red. The cutoff frequency is set with the central channel of the knob. For adjustments you can also use the mouse wheel and arrow keys, in combination with the CTRL and ALT keys that are used to obtain fine variations in frequency.

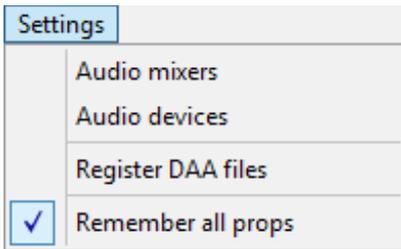


Here we see the 50 Hz of the electrical system, picked up with a piece of wire 10cm long and in the upper trace the same filtered signal.

- In the image on the left you show the filtered signal with a low pass from one pole to 200 Hz.
- In the image on the right you can see the filtered signal with a low pass from 32 poles to 50 Hz.

Using the Trigger for the filtrate channel is obtained a greater stability.

# The Settings menu



**Audio mixers** - Opens the mixer listening audio (out) and record (in).

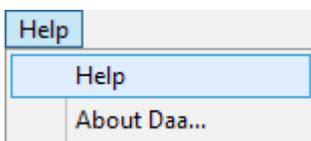
**Audio devices** - Open the control panel for configuration of audio devices.

**Register DAA files** - With this command you record DAA files to make them open with a double click with this application.

If you move the DAA application, or change name to its directory, the operating system is no longer able to associate the DAA file to the application. Instead, this command will always succeeds.

**Remember all props**- By enabling this option all settings are also remember closing the application. you deselect this option and then you close and re-launches the application DAA To restore basic adjustments.

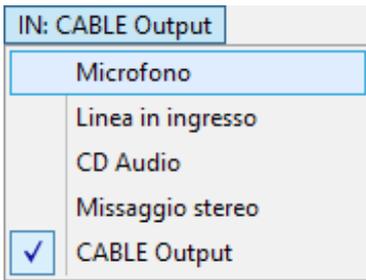
# The Help menu



**Support** - Open the folder that contains the documentation files.

**About** - Provides information on the DAA program.

# The menus of the Input Devices



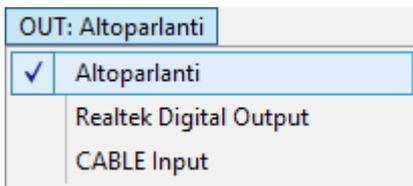
With this menu you select the input device.

The number of devices depends on the connected sound cards.

Some devices may not appear because disabled.

With AudioDevices menu you can view the hidden devices and enable them.

# The Output Device menu



With this menu, choose the output device.

The number of devices depends on the connected sound cards.

Some devices may not appear because disabled.

With AudioDevices menu you can view the hidden devices and enable them.

# Adjustments Using the Mouse and Keyboard

## Adjust the knobs with Wheel Mouse

It's important to note that it is not necessary to click on the knobs. Just bring the cursor on them and then rotate the wheel of the mouse.

For rotate the inner knob of the double knobs, holding down the CTRL key while rotating the wheel of the Mouse. Or you could use the ALT key to get smaller movements. See also other key combinations explained below.

## Adjust the knobs with the keyboard

You move the mouse cursor to a knob and it increases or decreases its value with the right arrow and left buttons (or up and down if you prefer). Using the PageUp and PageDown keys are obtained variations ten times more. When using the modifier keys ALT does not work. To make small changes it is best to use the mouse wheel in combination with CTRL, SHIFT + CTRL, ALT and CTRL + ALT.

## The keys that change the cruising speed

With the keys SHIFT - CTRL and ALT changes the speed of adjustment for both the mouse wheel for the Up Arrow and Page Up / Down.

- ◆ SHIFT multiplies the speed for 10
- ◆ CTRL It divides the speed of 10 times
- ◆ SHIFT + CTRL It divides the speed of 100 times
- ◆ ALT It divides the speed of 1000 times
- ◆ ALT + CTRL It divides the speed of 10000 times

## The functions that can be adjusted by clicking on the display

- ◆ When the cursor is the mouse wheel adjusts the display the time base.
- ◆ Pressing the left button on the screen and moving the mouse you can adjust the "Delay".
- ◆ If the button is pressed in the vicinity of the trigger or arrow cursors, you can move them.

## Bring back the controls to their default values

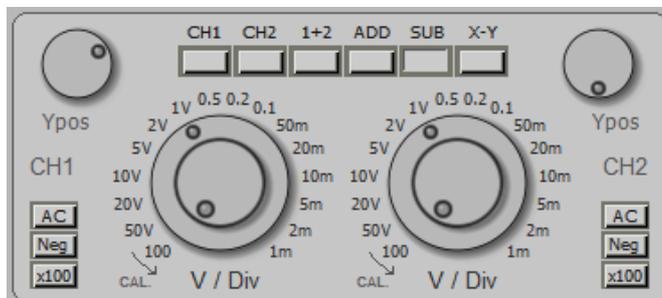
All knobs back to their default values with a double click of the left mouse button.

With a double click on the Trigger Settings and keyboard keys, in addition to select their function you also get the default of their value. The default the same is also obtained by selecting the button and then double-clicking on the knob.

# Measure the current

When the current to be measured is referred to ground we can measure interposing a resistor between the user and the mass. In these cases, with the two oscilloscope channels, we could also measure both simultaneously.

But if the current to be measured is located at a point under tension, to measure we have to use both channels connected to the two ends of a resistor. We'll link this resistor in series to the user, and then cutting the wire which carries current and interposing the resistor. Then we will connect the two test probes across the resistor and choose "SUB" of the input channels in the panel.



To do this measure the two knobs "V / Div" must always be adjusted equal to each other. The current picture will be equal to "V / Div" set of these two knobs, divided by the resistance in ohms. The following table shows some examples of the most common.

Resistor	0.5 V / Div	0.2 V / Div	0.1 V / Div	50 mV / Div	20 mV / Div	10 mV / Div	5 mV / Div
1 ohm	0.5 A / Div	200 mA / Div	100 mA / Div	50 mA / Div	20 mA / Div	10 mA / Div	5 mA / Div
0.5 ohms	A 1 / Div	400 mA / Div	200 mA / Div	100 mA / Div	40 mA / Div	20 mA / Div	10 mA / Div
0.2 ohms		A 1 / Div	500 mA / Div	250 mA / Div	100 mA / Div	50 mA / Div	25 mA / Div
0.1 ohms		2 A / Div	A 1 / Div	500 mA / Div	200 mA / Div	100 mA / Div	50 mA / Div
00:05 ohms			2 A / Div	A 1 / Div	400 mA / Div	200 mA / Div	100 mA / Div
00:02 ohms			5 A / Div	2.5 A / Div	A 1 / Div	500 mA / Div	250 mA / Div
00:01 ohms				5 A / Div	2 A / Div	A 1 / Div	500 mA / Div

Some boxes are empty because the dissipation on the resistor exceed 10 watts. Those that exceed the watts are highlighted in red. Those that exceed the quarter-watt orange and those that exceed the tenth of a watt dark yellow. In the arealight yellow you can rest easy even using very small resistors. The watts of calculation refers to a current corresponding to half the oscilloscope scale, namely four vertical divisions.

The resistance must be of low value, so that the current (amperes), multiplied by the resistance (ohms), produces a low enough voltage as not to disturb the circuit under test, and also to maintain a low dissipation in the resistor.

Usually this tension should be at most a few hundred millivolts. But measure low voltages makes the most inaccurate measures, then we must evaluate case by case basis.

It is not easy to account for everything and it takes experience to identify the best compromise.

There is a bit 'aid in the fact that the oscilloscope directly measure the voltage drop on the resistor, and tells us the value of this voltage, instant by instant.

# The Begin-End controls

By choosing a sampling window that begins when the impulse is to reach the microphone and ends before they reach the reflections on the walls of the environment you get a higher precision for FastSweep and Pulse analysis. These values, however, must be adjusted with care otherwise it can completely distort the measurement.

## BEGIN

BEGIN usually is held to zero, the automatic synchronism places the beginning of the sampling just before the 'beginning of the pulse to be analyzed. Sometimes, however, 'the synchronism does not act well and it is necessary to make a small manual corrections by BEGIN.

If BEGIN is exceeded, even slightly, the pulse start this is truncated, and the frequency response changes so completely an adjustment mode can be arriving until the response changes and then go back a little.

The most accurate method to adjust BEGIN however, is the following:

- ◆ Press the button SAMPLES
- ◆ Turn the knob TIME BASE 1 mS / Div, raise mV / Div channel one up to the good frame pulse start
- ◆ Adjust BEGIN to bring the surge began about a division of the TRIGGER on the right arrow.
- ◆ Also check that the pulse width is less than one volt (see VIEW Menu - ViewSamples).

## END

The END adjustment only with PULSE type analysis,

Normally the END value is left at maximum (200 mS), but it is possible to decrease it to truncate the sampling so as to exclude reflections on the walls of the room.

Adjust END to the distance between the speaker and the nearest opposite wall or side of the chest, plus the distance between this wall and the microphone.

If the environment is small and therefore the distance is low you lose precision on the low part of the spectrum and it is better to adjust END on a higher value even if doing so includes a part of the reflections.

The END value is measured in milliseconds but while it is possible to see in the rule at the bottom (status bar) the converted value in "meters traveled by the sound."

On the status bar you can also read the minimum valid frequency which is continuously recalculated and is visible until the mouse cursor remains positioned on the knob.

# Frequency Response Measurements

To perform frequency response measurements of acoustic systems (loudspeakers) is necessary to use a microphone. The microphone must have a sufficiently flat frequency response are therefore excluded from the dynamic microphones.

A great microphone for these measures is the type Electret which is easily commercially available.

The best Electret microphones ensure a frequency response within a decibel from 20 Hz to 20 KHz (do not use models with three-wire or with an outer diameter of 10..12 mm. Using the model to two terminals with a diameter of 6 mm).

The Electret microphones they need a source of direct current (5..10 volts in series with a resistor 4..10 K) that is usually already arranged on the incoming MIC sound cards (only if the MIC input is not stereo).

To connect Electret microphones to a MIC input of a sound card, use a stereo jack with two signal heads (right and left) joined (one of the heads provides power and the other carries the signal), and a cable shielded, not longer than three meters, with stocking and only one signal wire (check on the arrivals feeding microphone, from 1 to about 3 volts)

If the power supply for the electret microphone is not available, he wanted to use a connection wire is longer or requires greater sensitivity you will have to use an external preamplifier with power stack that greatly facilitate the measurements.

The external preamplifier advantages are:

- ◆ Adjustable sensitivity and greater,
- ◆ Low noise and elimination of disturbances on the cable
- ◆ Very low output impedance which allows to use a long cable up to 10..50 meters.

## What kind of pre-amp to use

You can build preamp for electret microphones in minutes. It takes some components that are commonly found and a matrix board.

For the construction see the file "[DAA\\_V4\\_InputDevices](#)"

# Tips on microphone position

To measure the response curve of the crates without having inaccuracies due to reflections exist various methods.

## First method

Put the outdoor case, lying with cones facing upwards, and the microphone hanging from a meter in height. (Inexpensive but also very uncomfortable)



## second method

Use an anechoic chamber.

## Third method ( "Begin" and "End")

Begin and End can be used only if you are working in a great environment (nightclub) where the nearest wall is over ten meters.

The microphone provides a few meters from the till. Lights only one case at a time otherwise interfere with each other and ruin the measures. You set "Begin" so as not to cut off the first part of the pulse and set "End" at twice the distance to the nearest wall. This "double" must be at least 20..25 meters measure otherwise it hurts the bass. In this way the sampling window excludes the reflections because they arrive too late, and when it comes to size and made.

## Fourth method

Keep the microphone very close to the speakers, it is the most convenient and, in some ways, even the most accurate provided you do some 'of evidence in order to understand what is being measured and not take blunders.

With the microphone to 5..50 centimeters from the cones (depending on the size of the speakers and the room) the sound pressure difference between the direct sound and the reflected and so high that the reflections do not affect most of the measures.

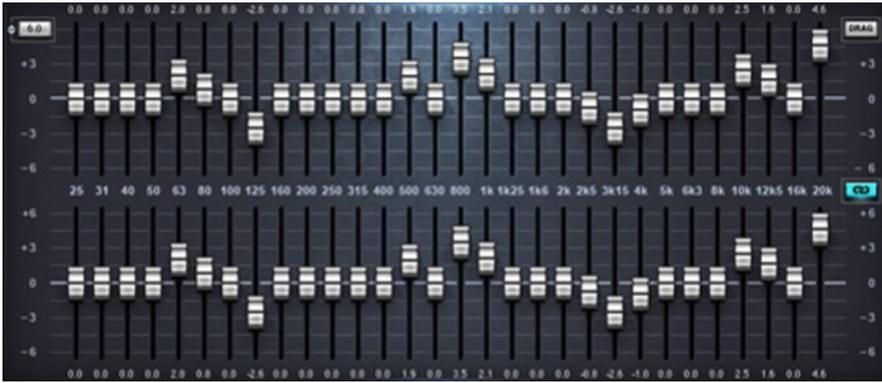
According to 5 cm errors are certainly less of a decibel. However, so 'doing cones are to be measured one at a time by placing the microphone exactly in front of the cone axis (at the same distance for all the cones).

It ends so 'to have three separate curves (or two or four depending on the number of routes) curves to be evaluated separately in order to understand the flatness and the tips and the holes of each band.

Then you move the microphone to a meter (on axis - a cash center) and, this time ignoring the spikes and holes that surely will be increased, the focus is on the middle level of the bands to be as similar as possible.

The fourth method may seem imprecise and insecure, but when you take your hand, is the best, even more accurate sound chamber. The next page explains further details of this method.

# Operation of the fourth method



If you place the microphone on the same axis, at one meter from the body (classical measuring in an anechoic chamber) are peaks and valleys (especially in the areas of crossover frequencies) that are due to the microphone-tweeter distances, microphone-midrange and microphone-woofer .

They are formed of the interference characteristics for that particular position of the microphone that completely change if you move the microphone (even 5 or 10 dB for a displacement of 50 cm to the right or left, back or top etc ..)

These spikes, caused by interference, do not have nothing to do with the environment, reverberation or less, and all cases with more than a cone, also the most expensive, even in an anechoic chamber, creating peaks and valleys at certain frequencies that change depending on the distance and the position of the microphone.

And 'an inevitable physical phenomenon, in some particular positions and frequencies pressures add up or cancel each other out. There is no remedy but have all sources in one place, even cones with coaxial tweeter are exempt (because of those two centimeters away), you should just use a single cone ... and you can not .

Ultimately if you try to measure the whole case you will always find the tips and valleys (even ten dB) that confuse the measures and that are missing anything with the characteristics of the cones, the speakers and crossover.

The speaker manufacturers know this and always make sure to minimize the inaccuracies that are measured at one meter, straight. Too bad that usually the cases are heard at a greater distance and almost always off-axis.

This is not to say that three meters crates will get worse, but that you should not fixate to measure and correct the gaps and spikes due to interference (much every point will listen to them different). Also careful not to lose hours with equalizers for perfect equalization, which will be valid only in the exact spot where the microphone is placed.

You could measure the response in, say, ten or twenty different positions spaced randomly throughout the listening area and then the average ... But the best thing would be to have a tool that interference will not see their own and, in fact, this is exactly what you get if you measure the cones one by one and in a short axis.

Then you move the microphone, you pretend that peaks and valleys do not exist, and the tie of the cones emissions by adjusting the gain of each band.

# Technical features

The instrument consists of the following functional blocks:

- ◆ Viewer signals
- ◆ Signal generator
- ◆ Sampler of signals
- ◆ Frequency
- ◆ Analyzer

## Features of the viewer (as an oscilloscope)

- ◆ amplitude adjustment from 100 V / div at 1 mV / div
- ◆ Time base from 1 sec / div to 100 uS / div
- ◆ Trigger positive or negative
- ◆ Triggers for pulsed signals
- ◆ Level control for manual trigger
- ◆ Two tracks, addition, subtraction and display X / Y.
- ◆ Tracks width and position.
- ◆ Visualization with delay and DeltaTime.
- ◆ Storage of the past thirty seconds with the maximum resolution.
- ◆ Zero calibration and voltage, to an accuracy of about one millivolt and one tenth of a decibel.
- ◆ Measurement of the average value, the effective value (rms), the peak to peak value and the decibel.
- ◆ Measurement times and frequencies.

## Features of the viewer (as a spectrum analyzer)

- ◆ X-axis linear or logarithmic with minimum and maximum frequency adjustment.
- ◆ Axis Y linear adjustment with maximum amplitude from 100 Vrms to 1 mVeff
- ◆ Y logarithmic with amplitude adjustments from +60 dB to -120 dB

## Features of the viewer (such as three-dimensional spectrum analyzer)

- ◆ X axis and Y as spectrum analysis
- ◆ Z axis with time adjustment.

## Features of the signal generator

- ◆ Frequency range: 1 Hz to 95 KHz (**Note 1**) in steps of 0.1 Hz.
- ◆ Forms of wave: Sine / Square / Triangular / Ramp positive / negative ramp / White Noise / Burst / Special pulses for frequency response measurements.
- ◆ Adjusting the duty cycle: 1 to 99% in steps of 0.001%
- ◆ distortion less than 0.002%
- ◆ Noise less than -96 dB
- ◆ Stability and accuracy of better than +/- 0.0001 Hz and one part in 200,000, equal to 0.0005%

## sampler Features

- ◆ Frequency range: 0.1 Hz to 95 KHz (**Note 1**)
- ◆ Noise less than -96 dB

## the frequency meter Features

- ◆ Measurement of frequencies from 0.1 Hz to 95 KHz (**Note 1**) accurately than 0.01 Hz.
- ◆ Measures of time from 1 second to 100 uS with an accuracy of 50 uS

## Features of 'analyzer

- ◆ Measurements of amplitude in dB and in volts.
- ◆ Measurements of timing and frequency offsets.
- ◆ Sliders measurement.

**(Note 1)** The software samples to 192 KHz, but so far we have not found sound cards with bandwidth above 20 kHz, and having two input channels.

# Precision Frequency Characteristics

With a good sound card the accuracy of time and frequency measurements is better than one part in a million.

Some sound cards, *in particular conditions*, they can be very inaccurate in the sampling frequency. It's so good to check your card with a known precise source (not the 50Hz mains).

Note also that if the sound card is used simultaneously by the DAA and one or more other precision frequency audio applications may, *in some cases*, deteriorate considerably.

In the event that two applications use different sample rates some sound card privilege the first application, and perform subsequent applications with an approximate sampling rate (*in some cases even errors of one percent*).

## Other characteristics

Some sound cards have an artificial system enlargement of the distance between the speakers (3d stereo enhancement), make sure it is disabled otherwise the frequency response is disturbed.

Also make sure that the tone controls are inactive.

Some notebooks have an ever-present equalizer to enhance the sound of their small speakers, often, but not always, you get a flat response when you insert a jack on the audio output. At times it may be necessary to turn off and on the audio output because the jack is detected.

Depending on the used sound card, as it is physically installed in the computer and of the shielding characteristics of the computer disorders may be present noise components outside the audio range (50 KHz - 200 Mhz) also of considerable amplitude.

Such disorders, which normally do not create problems in audio measurements, may be deleted using connection cables that incorporate a low-pass filter.

## How to see the generated signal

Most sound cards have the choice between inputs "Stereo Mixer", or "What U Hear". But there are cards that allow you to set as input only the line input or just the microphone. In these cases, to view the generated signal you have to put a wire (stereo jack) between the "LINE OUT" output and the "LINE IN input" or "MIC IN".

On Notebook it is always good to put a jack in "LINE OUT" so that the sound card excludes the interior and the internal equalizer speakers (it needs to sound a little 'best small speakers). Without equalizer you are obtained a flat frequency response that is important for the measurements.

# Links

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