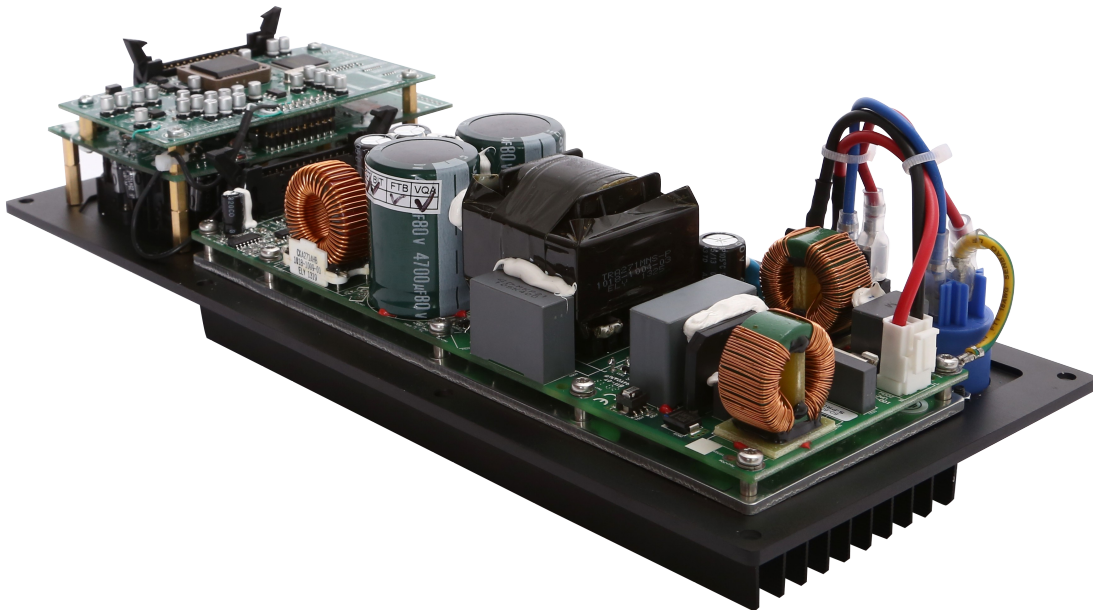


PDA500PF

PDA500PF is a complete solution dedicated to 1-way or 2-way self-powered loudspeakers. Designed to meet different applications, it provides 2 channels with output power of 2x500W@4 Ohm or 2x250W@8 Ohm. In addition it offers a full set of added value features such as on board DSP and RS485 connection for monitoring and control via dedicated PC software. To guarantee maximum reliability, the PDA500PF includes a highly efficient universal switch mode power supply with PFC (Power Factor Correction) which provides a total 1kW power to the 2 output channels. The 2 output stages use the well-known Pascal Class D SPRO2 module. The PDA500PF includes a set of sophisticated processes for loudspeaker, implemented by the powerful M704 DSP running 48kHz/24bit [96 bits precision for the internal intermediate processes] and high performance 24bit AD/DA Converters. Processes as Noise Gate, up to 512 taps FIR filters OR up to 48dB/Oct IIR Hp/Lp filters for X-over, parametric EQs per input and output sections, RMS compressor, alignment delay and All-Pass filters are available, all in all everything needed to optimize a self-powered loudspeaker. Moreover the Clip/Limiter function per channel provides output monitoring to prevent speaker damage with gentle gain reduction at clip threshold, in addition to the efficient heat dissipation system and Over-Heat protection which themselves ensure a reliability without compromises. Furthermore the PDA500PF is also equipped with a Dynamic Loudness function and an useful Pink/White noise generator. All setup parameters for input mixing, DSP features and the limiter setting are accessible by using the remote PC software.

A Powerful 512taps FIR is available for the System Phase Correction, based on System Response measurement.



Features

Outstanding Performance

High power output: 2 x 500W @ 4Ω or 2 x 250W @ 8Ω

Switched-Mode Power Supply with PFC and auto voltage sensing

Pascal Class D Amp Module - full bandwidth PWM modulator with ultra low distortion

Full protection circuitry including Over-Current, Over/Under-Voltage, Output DC and Over-Temperature

Excellent sonic performance with 24bit high end converters coupled with 48kHz sample rate

Top-grade DSP Engine

10 band parametric equalization per input channel

8 band parametric equalization per output channels

FIR OR IIR Filters for X-OVER:

The X-Over can be implemented both by FIR filters OR IIR Hp/Lp, as desired by the user which can select the FIR or IIR use by Sw.

FIR: Crossover filters can be created by the user selecting from 256 up to 512 taps, the FIR type and the Out Band attenuation or importing from external application the FIR coefficients as '.txt' file.

IIR: Crossover filters with slopes from 6dB/Octave up to 48dB/ Octave including Butterworth, Bessel and Linkwitz-Riley

Each output features a precision RMS Limiter with selectable Attack/Release time and Threshold.

On the Low Output, a Bell type Dynamic Filter for Driver protection is available.

Adjustable Delay time up to 161 ms for input and 81ms for output channels

Input channel includes a Noise Gate function, Pink/White noise generator, sophisticated Dynamic Loudness function and a High-Pass filter with slopes from 6dB/Octave up to -48dB/Octave including Butterworth, Bessel, Linkwitz-Riley and a RMS Compressor.

Direct PC/Network Connection

Rs485 connection for system setup, monitoring and control via fully manageable remote PC software

Control

Simultaneous control up to 32 units via PC software

8 Preset Selection by using rotary encoder switch

Security Lockout

Power & Amplifier Sections

Number of Channels	2
Max Output Power @ 8 ohms	2 x 250W
Max Output Power @ 4 ohms	2 x 500W
Output Circuitry	Class D - full bandwidth PWM modulator with ultra low distortion
Output Voltage	+/-70 V (SE Mode unloaded) / Bridged +/-140V (BTL Mode unloaded)
THD @ Rated power 4Ω (1kHz)	<0.005% (20 Hz - 20 kHz, 8Ω load, 3dB below rated power)
Signal To Noise Ratio	>120 dB (A-weighted, 20 Hz - 20 kHz, 8Ω load)
Frequency Response	20Hz - 20kHz ±0.5 dB
Power Supply	Switch mode power supply with PFC (Power Factor Correction) and integral standby converter
Operating Range	Universal Mains, 85-265V
Consumption / Current draw and Thermal dissipation @ 230 V	11.2W / -A / 38.3 BTU/h (Idle) 173W / -A / 143.4 BTU/h (1/8 max. power@4Ω)
Protections	Over-Current, Over/Under Voltage, Output DC and Over-Temperature

Audio

Analog Input	1 x XLR electronically balanced, +12dB
Analog Output	1 x XLR electronically balanced (Link)
AD&DA Converters	CS42528 24bit
Frequency Response (DSP)	20-20KHz; -0.5dBu at 20Hz and 20kHz

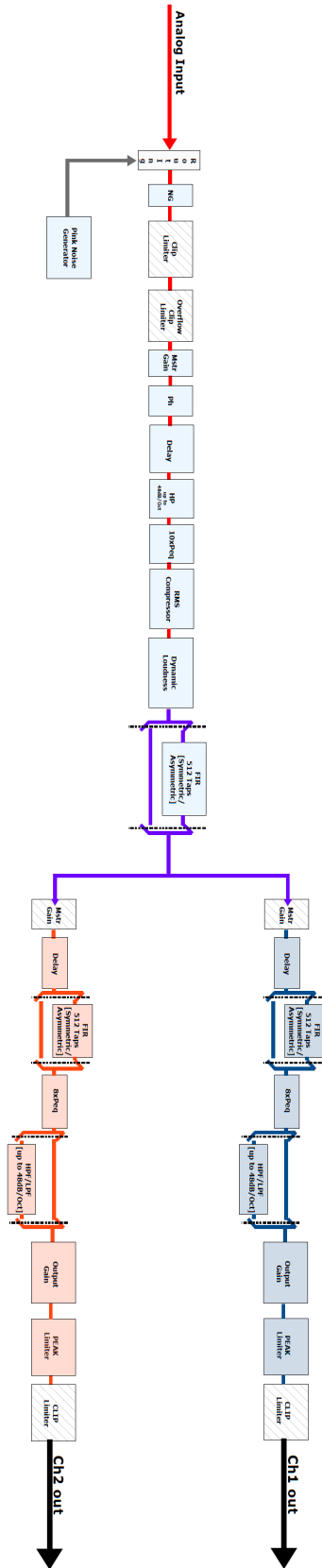
DSP & Processing

DSP Engine	MARANI M704
DSP Resolution	24bit (data) x 24 bit (coeff.), 54 bit accumulation registers, 96 bit precision on intermediate processing data
Parametric Equalization	10 filters input, 8 filters output channels
Filter Type	Bell, Low/High Shelving variable Q, Low/High Pass Qvar and Notch
Filter Gain	From -15dBu up to +15dBu by 0.5dBu resolution steps
Center Frequency	Adjustable with a 1Hz resolution step from 20Hz up to 20kHz
Bell Filter Q/BW	Q from 0.5 up to 10 by 0.1 resolution steps
System Phase Correction	512 taps FIR
Crossover section FIR	from 256 up to 512 taps coefficients, FIR type selection and out of band attenuation, operating from 250Hz, up as X-over point; accepted coefficients generated by external applications
Crossover section IIR	HPF/LPF Butterworth 6/12/18/24/48dB Octave Linkwitz-Riley 12/24/36/48dB Octave. Bessel 12/24dB Octave. Filter resolution 1Hz
Input Special Filters	FIR up to 512 Taps for System Phase Correction working with Internal Wizard Tool or importing Coefficients from External Application High-Pass filter up to 48dB/Oct Sophisticated Dynamic Loudness
Input RMS Compressor	Drive from -12 to 6dBu; Threshold from -18dBu up to -12dBu; Attack time from 5ms up to 100ms; Knee 0~100%; Ratio from 2:1 to 100:1; Release time from 40ms up to 1000ms (10ms resolution).
Output Peak Limiter	Threshold from -18dBu up to +12dBu Release time from 40ms up to 1000ms (10ms resolution up to 200ms,100ms res. up to 1000ms) Attack time from 1ms up to 100ms (1ms resolution up to 20ms, 10ms resolution up to 100ms)
Delay	161 ms, 20.8us increment / decrement steps on Input Channel and 81ms, 20.8us increment / decrement steps on each output Channel

General

Front	GAIN pot. -30dBu ~ 0dBu PRESET EQ 8 positions Rotary encoder switch Red LED (Power); Yellow LED (Link); Green LED (Signal presence) 1 x XLR female connector (Input) 1 x XLR male connector (Link Output) 2 x RJ45 connector (M-LAN Rs485) IEC C13 16A connector; Power on/off switch
Dimensions	355x70x155mm
Weight, Net / Shipping	6.61 lbs (3 Kg) / 8.82 lbs (4 Kg)

Processes Block Scheme



PDA500PF PC Sw Quick User's Guide

Before to install the 'setup.exe', please make sure that the Net Framework 4.0 is already installed on the PC used.

After install the 'setup.exe' and run the app “PDA500PF” then the system show the following window

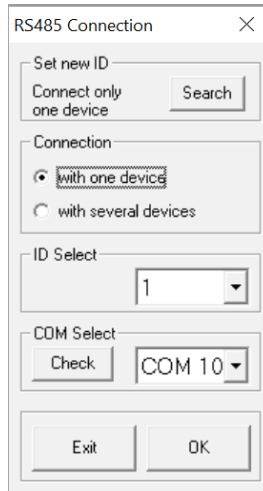


and after few seconds the main window



The user must click on the menu “**Connect**” in order to connect one or several devices.

The User has the possibility to connect the device via **RS485**



RS485 Connection

Set new ID
Connect only one device Search

Connection
 with one device
 with several devices

ID Select
1

COM Select
Check COM 10

Exit OK

If the Sw is used for the first time with a new Hw Plate, will be necessary assign to the plate the specific ID number, so to be able to put more plates in the same Network.

The ID assignment can be done asking the Sw to “Search” for the connected device (for the ID assignment only ONE device has to be connected to the Pc Sw) and once found, the ID can be assigned and 'stored” within the connected unit.

Once done this for more units, assigning them different Ids, then the all units can be connected in Network and to be controlled by the same Pc sw.

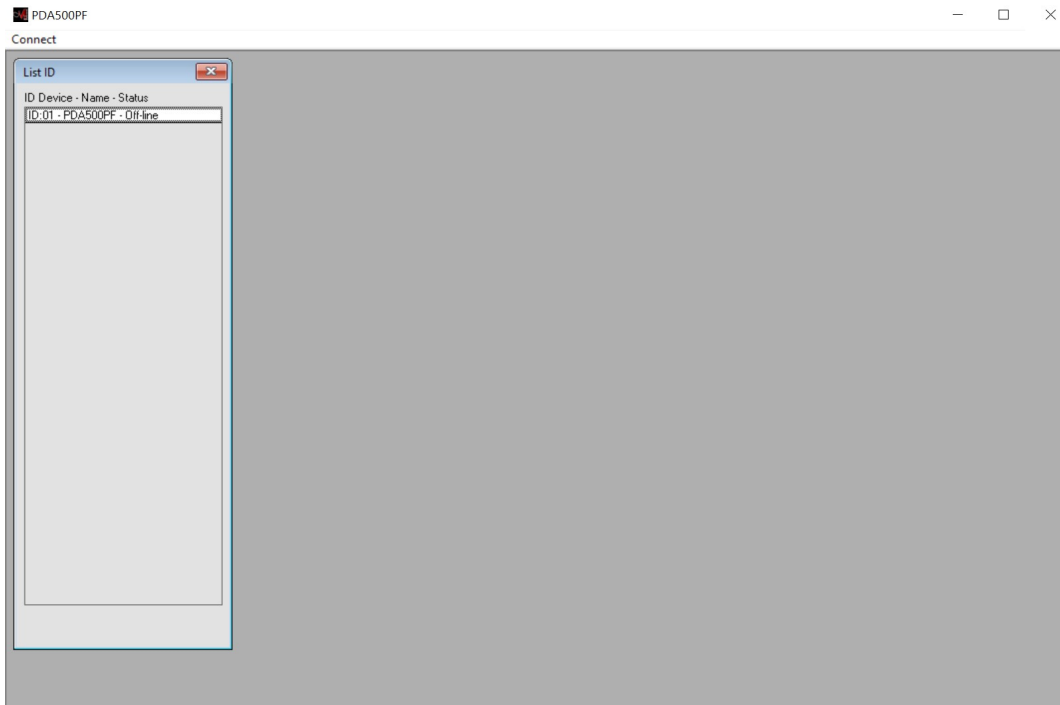
If only one unit with a defined Id is connected to the remote control, in order to control it will be enough to select as connection “with one device” and set as “ID Select” the one of the connected unit. If more units with their different ID are connected in network , then just need to select as Connection the “with several devices”.

The User must also set the number of the COM used to connect the device(s). When the RS485 is used and the User want connect several devices then is important that all device have an unique ID in order to avoid devices “overlap”.

If user not know the available COMs, then can use the button “Check” to see all COM connected on the PC.

Once set the Connection parameters, press the button “OK” and the system will start to check for the connected device(s).

When found the connected device(s) the Sw will show a list of the devices found (if no device are found then is possible to continue in demo mode “Off-line”) on the window “List ID”.



If no devices found, the Sw will ask if the user want to continue in demo mode for evaluating the available process of the **PDA500PF**.

To edit the device, double click on the name of the device present on the “List ID”

EDITING Windows

on the Edit window is possible edit all parameters of the Input and output Output channel, save and load configuration on the sw PC and see the current version of the firmware and software.

On the ALL Editing Windows, the Top Bar is fixed and shows ever the several available "tools" useful for the User to manage the Editing Job so as to save it on Pc or within the unit.

TOOL BAR



NOTE ON PRESET STRUCTURE ON PDA500PF: Before to proceed with the description of the functions related to the several icons and buttons on the tool bar, need to describe which is the presets' structure adopted for the PDA500PF. We did start from the idea that the presets have to involve ONLY the Input path parameters, and not to involve the Output Sections. This basically because we considered that the Plate, once set the proper configuration for the specific cabinet, has to have the output parameters, as Xover and delay, or Phase set, fixed, being those parameters strictly related to the cabinet itself and its set up, and not instead used for creating different "sounding styles", which are left to the user in terms of Input parameters set up. On the base of this principle, we did manage the preset structure so to give the possibility to Save on / load from Pc or store into / recall from the PDA500PF up to 8 Input configurations, and set the single Output configuration in a dedicated area of the PDA500PF internal memory.

To be more clear, let define the following as the **INPUT PARAMETERS**, which are the ONLY ONES saved on / Loaded from Pc and Stored into / recalled from the internal memory of the PDA500PF when a preset is created:

- Input Gain
- Input Mute
- Input Phase
- Input NoiseGate (Bypass,Thr,Rel,Atk)
- Input Delay
- Input RMS Compressor (Bypass, Thr, Ratio, Knee, Rel, Atk, Makeup)
- Input Dynamic Filter
- Input HP Filter
- Input 10 PEQ (Type, Freq,Q,Gain,Bypass)
- Enable/Disable FIR Phase correction
- Name Preset

Particularly, the one above are the ONLY parameters saved on Pc and loaded from Pc.

All other parameters, included the ones related to the Outputs configurations, are **stored ONLY within the PDA500PF memory.**

The internal memory of the PDA500PF is organized as follow:

Input preset 1

Input preset 2

Input preset 3

Input preset 4

Input preset 5

Input preset 6

Input preset 7

Input preset 8

Current input buffer

Fixed output parameters

System variables

When the PDA500PF is turned on, the MCU is taking some action

1. Read the Encoder position on the Plate, so to load as current preset, the one selected by the encoder itself, which will be actually the combination of the Input Preset selected and set it on the Current input buffer (temporary), then load the Output configuration resident within the Fixed output parameters location.
2. Set the all System variables.

After that, the PDA500PF is working without Pc connected, performing the selected preset.

If the Pc will be connected for an Editing Session, the Pc, connection time, will read the Current input buffer, the Fixed output parameters area and the System variables are, and show the current Input/Output configuration on the Editing windows. If a parameter not belonging to the previously defined INPUT PARAMETERS list, is modified, which most likely then is related to the Inputs configuration or to System variables, then the new value of the parameter is saved into the "Fixed" memory area. If a parameter belonging to the INPUT PARAMETERS list, is modified, then the new value of the parameter is saved into the "Buffer" memory area.

When finally the user is deciding to save on Pc or to Store into a location of the {DA500PF the new preset, the content of the "Buffer" area is saved on Pc or Stored into the selected location of the 8 available on the PDA500PF. The "Fixed" area is untouched by the Saving and storing actions.

On the base of the note above, Load/Save. Store and Recall icons allow the following...

Load: Load a previously saved Input Configuration from a defined Pc directory. The configuration loaded will be the one edited with its parameters set on the input editing windows. The configuration loaded is containing only the Input parameters

Save: Save the current Input Configuration (the all parameters set within the Input Window) into a defined Pc directory. The saved configuration is containing the Input parameters.

Store: Save the current Input Configuration (the all parameters set within the Input Editing Window) into one of 8 available locations within the **PDA500PF**. The saved configuration is containing the ALL parameters set in the Input Editing Window and can be recalled also from the unit's front panel.

Recall: Load a previously saved Input Configuration from one of 8 available locations within the **PDA500PF**. The configuration loaded will be the one edited with its parameters set on the Input editing window.

The stored Preset within the unit are re-callable on the PDA500PF, by the external available 8 positions absolute encoder.

Encoder Position: this frame is indicating the current position of the Absolute Encoder located on the PDA500PF Plate.

Overflow Process: this Led turn ON if an Overflow is occurring within the DSP processes and the Overflow Clip Limiter become therefore Active. [see IMPRTANT NOTE later]

Temperature: in this frame is displayed the current Temperature of the Power Stages. If the temperature is exceeding the 80Deg Celsius, the the DSP set Outputs (feeding the power Stages) are MUTED.

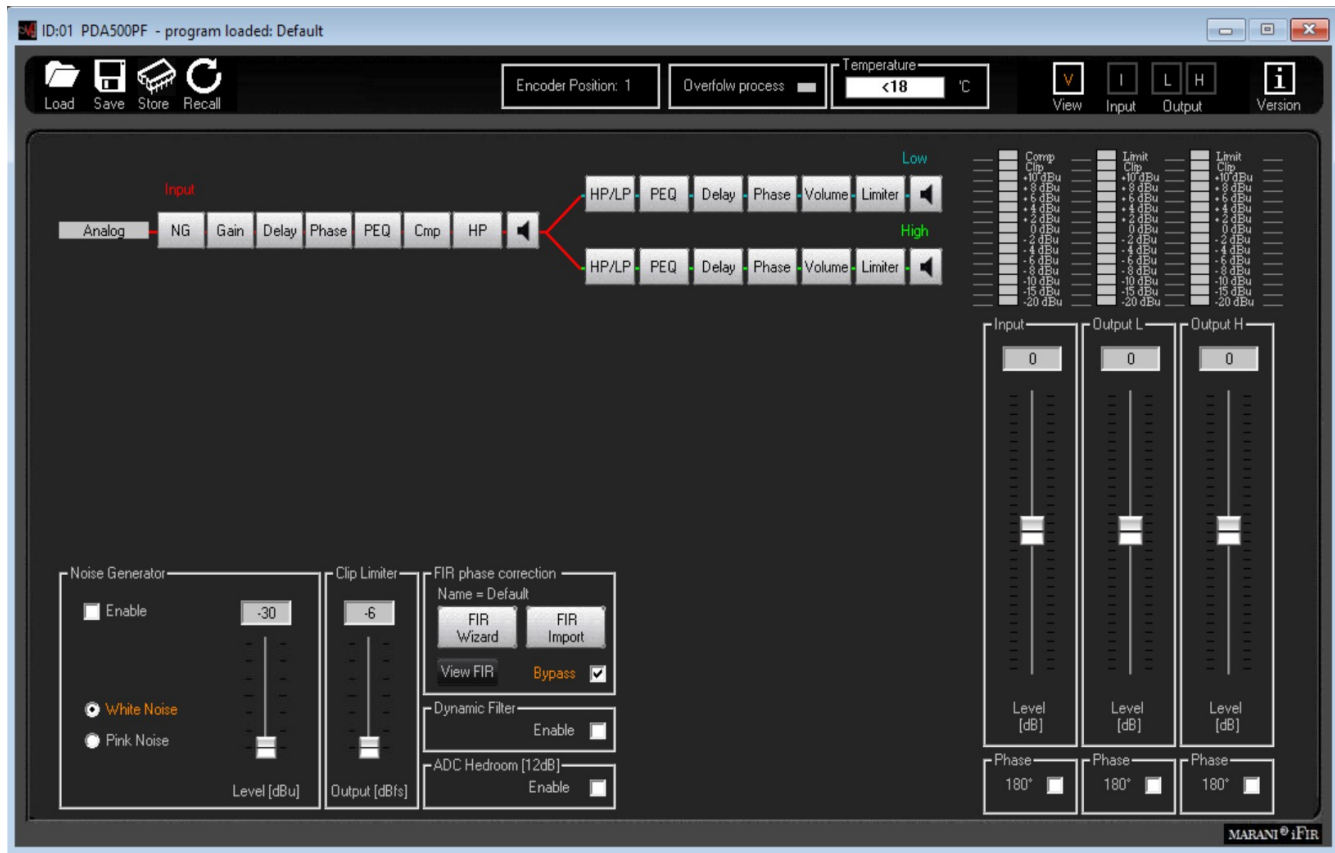
V: button allowing to access the GAINS and SYNOPTIC image Main Editing Page

I: buttons allowing to access the INPUT CHANNEL Editing Pages

L/H: buttons allowing to access the OUTPUT CHANNELS Editing Pages

i: info button, giving information about the Firmware and Pc Sw version.

GAIN and SYNOPTIC WINDOW



This Window is presenting a synoptic image of the several processes available on the Input and Output paths of the **PDA500PF**, so as a series of “frames” for the selection/setting of particular processes.

The Main Input of the PDA500PF, is a **Balanced Analog** one, coming from the the Front Panel XLR connector.

Noise Generator frame: In alternative to the Analog Input, a Noise Generator can be sent to the DSP processes and to the PDA500PF outputs.

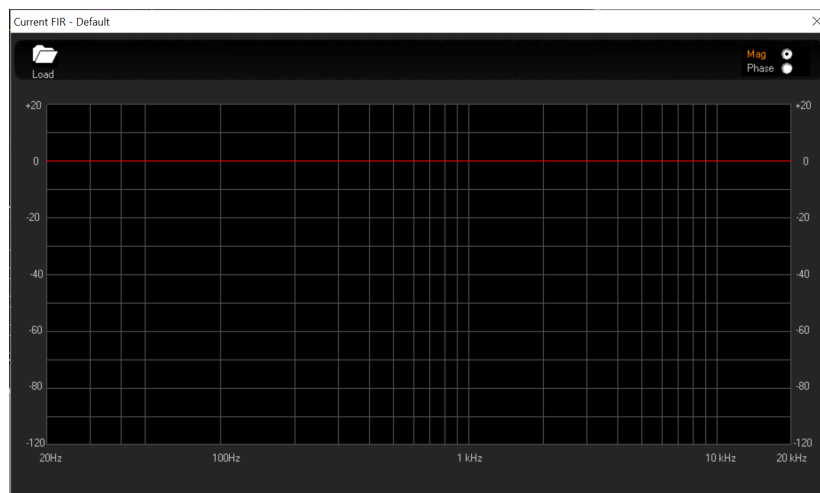
- **Noise Generator**: when enabled, the Analog Input is replaced by the output of the internal noise Generator. The internal Noise Generator can generate White or Pink Noise and its Level can be controlled by the related Slider ranging from -40dBu up to 0dBu.

Clip limiter frame: Here is possible to set a Clip limiter on outputs able to limit the output itself up to -6dBu from its maximum level. This Clip limiter can be useful, after the Peak Limiter on the output paths, to dramatically cut with infinite attack time, the output signal, there where APEC is used for processing signals going to external Class D amplifiers, in example, without internal Limiting circuitry.

FIR phase correction frame: here are available the controls for the use of a **512 Taps FIR** located on the Input path, which has been added mainly for being used for **PHASE CORRECTION** purposes, there where used together with the “Wizard” process embedded in the **PDA500PF Sw** (see NOTE A), able to control a measurement process bringing to the calculation of the necessary coefficients for the phase linearization of a 2 way system.

Nevertheless, the FIR on the input paths can also work with coefficients loaded by external files and generated by Third Party Sw able to generate FIR coefficients. In this case, the FIR turn to be a up to 512 Taps standard FIR, which can be loaded with any sort of coefficients for getting any sort of result (PEQ, Min Phase, Lin Phase, etc.). For the use of the external coefficients, need to generate them as text files and import them. See the details on the NOTE B.

- **Wizard**: this control is launching the “Wizard” Sw for the related FIR (see NOTE A for the Wizard Sw)
- **FIR Import**: here external coefficients can be imported into the FIR. Pressing the “FIR Import” button, the following window open and from the clicking on the LOAD icon, a .txt file (see NOTE B) with the Third Party calculated coefficients can be imported.



The Amplitude and Phase responses can be seen on the graphic which will show the characteristics of the imported filter.

- **View FIR**: once imported, or even internally generated, the FIR coefficients, the response of the FIR can be seen recalling it on graphic just pressing the 'View FIR" button.
- **Byp**: this check box is useful for making the loaded Fir Active or not. When not active, checked box, the FIR is simply transparent on the related input path. The Byp can therefore be used as powerful tool for A/B comparison between the system responses using or not the FIR.

Dynamic Filter frame: the Dynamic filter is in fact a Dynamic Loudness which is a Dynamic Filtering boosting automatically the Input level on Low and high Frequencies on the base of the Input Level RMS value. Particularly the boost is applied with a couple of Bell filters with different small Qs [pretty wide BW] centered at 50Hz and 12kHz. At these 2 frequencies the boosts have different amount: the boost on Lows at his maximum reaches +7dBV when the input signal is -28dBV from its max and the boost on Highs reaches its maximum boost with +3dBV. The amount of boost on the Lows and highs can anyway be tuned with sliders controlling in percentage the boosts amounts independently. The following table gives the details about the Boost amounts per Low and High frequencies in relation with the Input Signal level.

DFL Input [dBu]	50Hz / DFL Out [dBu]	50Hz filter gain [db]	DFL Input [dBu]	12KHz / DFL Out [dBu]	12KHz filter gain [db]
-16	-6.7	9.3	-16	-11	5
-14	-4.7	9.3	-14	-9	5
-12	-2.9	9.1	-12	-7	5
-11	-2.1	8.9	-11	-6.1	4.9
-10	-1.5	8.5	-10	-5.3	4.7
-9	-0.8	8.2	-9	-4.5	4.5
-8	-0.4	7.6	-8	-3.9	4.1
-7	0.1	7.1	-7	-3.1	3.9
-6	0.4	6.4	-6	-2.6	3.4
-5	0.8	5.8	-5	-1.9	3.1
-4	0.9	4.9	-4	-1.5	2.5
-3	0.7	3.7	-3	-1.1	1.9
-2	0.5	2.5	-2	-0.8	1.2
-1	0.8	1.8	-1	-0.2	0.8
0	1.3	1.3	0	0.6	0.6
1	2	1	1	1.5	0.5
2	2.5	0.5	2	2.3	0.3

The table has to be read as follow...

The 2dBu, 0dBV, represent the Input Level at the one the boost on both Lows and Highs stops, maintaining a constant value up to the Max input level of 12dBV. The level of -16dBu, which is -18dBV, is the Input Level at the one the Boost for both lows and highs reaches its maximum and remain constant for the all lower input signals.

ADC Headroom [12dB] frame: checking the ADC Headroom box, an internal process is made active, which is allowing to have onto the **PDA500PF** A/D converters a +12dB headroom. This means that the MAX output level of the Amplifier is reached yet when the input signal is at -12dBu from the Max accepted level accepted by the A/D converters. Each increase of the Input level in front of the A/D from the -12dBu, is contained by an internal Clip Limiting process. This feature is allowing to get then the Max power delivered by the Amp when still 12dB of Headroom are available on the Inputs, so allowing to have already the Max Power performances and avoid Clips there where the input signal is Exceeding the -12dBu from the max acceptance.

IMPORTANT NOTE: the **PDA500PF** has an internal Algorithm controlling the level coming from the Processes including gain variations as, Filtering Gains or Simply Level Gains, there where the Gain coming from the processes themselves can be a positive one. In this case, it is possible, given an input level close to the max available, to have internal processes overflow. The **PDA500PF** inner Clip limiting Algorithm is detecting the potential or already occurring overflows and is containing them up to a 12dB Headroom, with a process “distortion free” based on level reduction table coefficients. *[Note that this algorithm is different from the one used for the 12dB Headroom on the A/D converters, being the one here described “containing” digital potential overflows coming from the internal DSP processes].* When overflow over 12dB occurs on the internal processes, the Algorithm is containing them with a no distortion free process, which progressively is increasing the distortion level as more as the overflow containment is exceeding the +12dBu. The result of this “transparent to the user process”, is a NOT CLIPPED output signal even with large excess of gain in internal processes and Input levels close to the max accepted by the unit. The process is active ONLY in case of Overflow occurring; when the signal is not exceeding with the sum of the Input level and the processes gain, the unit's dynamic, then the process is not active.

Input Meters: the input meters show the Input levels from -36dBu up to +12dBu. The Limit Led is indicating if Active the RMS Compressors. The Clip Led is indicating if the Clip limiter Algorithm described here above is active on the Input path.

Output Meters: the output meters show the output levels from -36dBu up to +12dBu. The Limit Led is indicating if the Peak Limiter. The Clip Led is indicating if the Clip limiter Algorithm described here above is active on the Output path.

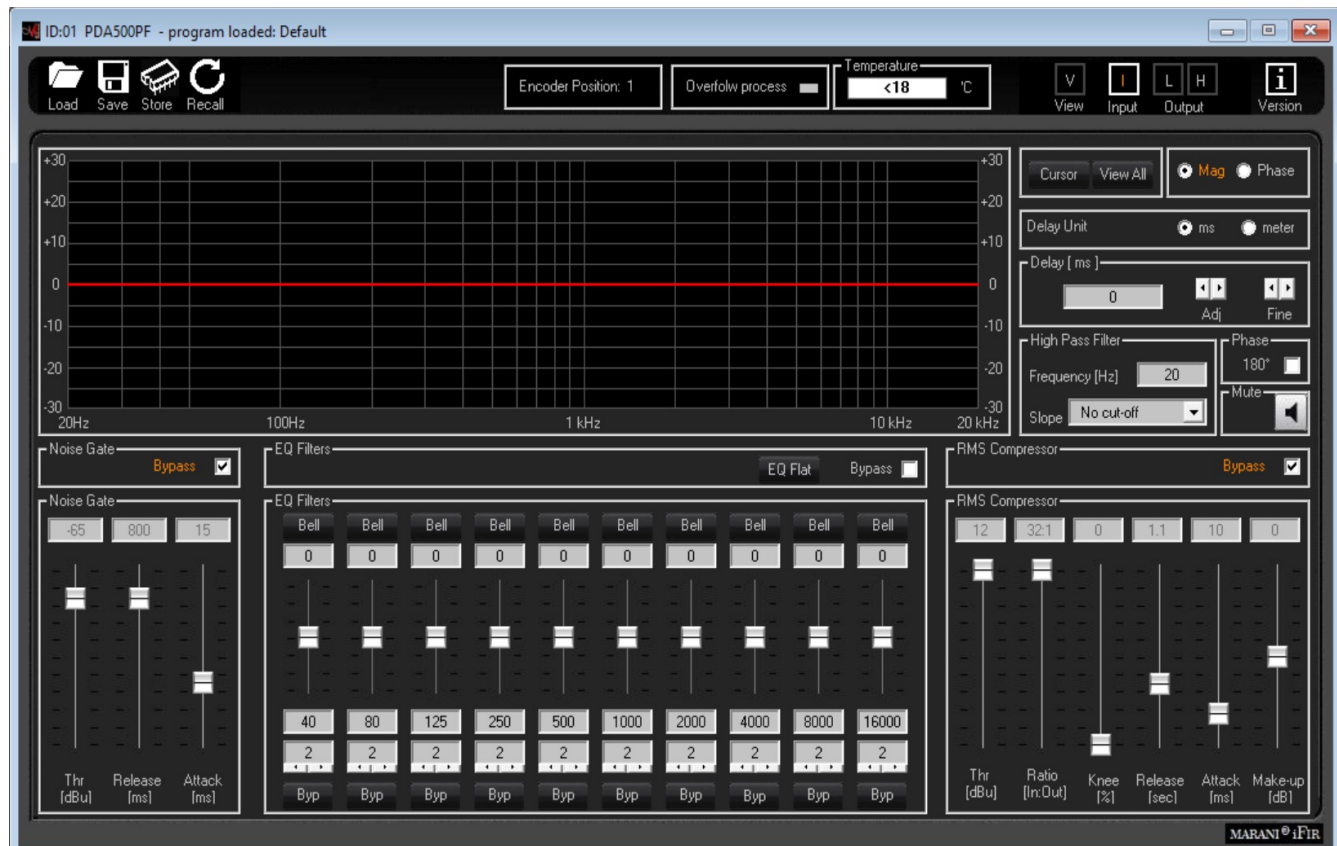
Sliders: the sliders control the Input/Output levels ranging from -12dBu up to +12dBu.

Phase (Polarity): control for inverting the signal polarity (180Deg.)

Overflow Process: this led is turning ON when an internal overflow is occurring at digital processes level.

Synoptic Image: The Synoptic Image is showing in clear way the available processes on Input and Output paths, so as gives the possibility to directly MUTE/UNMUTE input and any output, just clicking on the related loudspeaker icons. Clicking on any block of the Input path or Output path, the user is immediately brought to the editing window of the related Input or Output channel.

INPUT WINDOW



All the processes available on the Input Paths can be edited within the Input Windows.

Eq Flat and Bypass Controls: the EQ Flat control is allowing to flat in one shot the 10 bands Parametric Eq. The Bypass control, instead, is allowing to temporarily Bypass the Eq, without resetting the current setting.

EQ Filters: within this frame, the all 10 filters of the PEQ can be defined in terms of type, gain, Freq, Q

- **Filters type:** on top of the Slider and the box showing the gain level, there is a button that once pressed is opening a wide list of filters type which can be selected, for each filter of the parametric Eq. The available filters are **Bell** (constant Q), **Hi/Low Shelving Q** (Symmetrical with variable Q), **Lp/Hp Q** (12dB/Oct second order, variable Q) and **Notch**.

- **Gain:** the filter types which can gain, have the gain controlled by a slider ranging from -15dBu up to +15dBu.

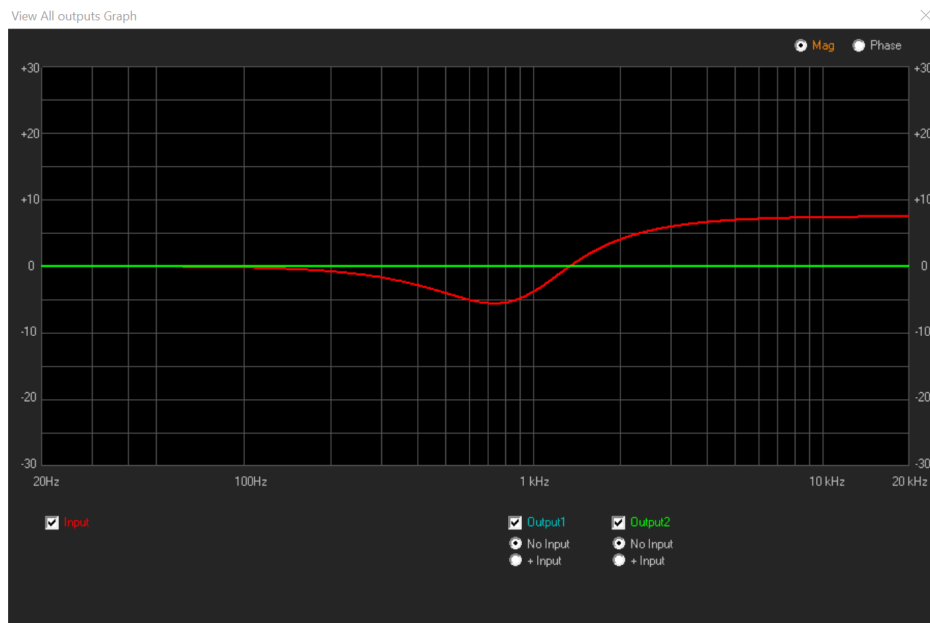
- **Freq:** here can be inserted the Center, Cutting Frequency of the selected filter, ranging from 20Hz up to 20kHz.

- **Q:** there where usable, the Q can be set with a range depending from the selected filter.

- **Byp:** any filter of the 12 of the Parametric Eq, can be singularly bypassed, so to have the possibility to a quick A/B comparison between the signal when active or not the filter currently in editing phase.

Cursor: clicking on this control, cursors will appear on the graphic showing the amplitude response of the parametric Eq, for the setting of the filters' gain by mouse action.

View All: this control is opening a window showing the full Eq setting on each Input or Output or combination of them.



Mag and Phase: it is possible to decide if showing on the Window's graphic the Magnitude response of the Parametric Eq setting (Mag selected) or the Phase response (Phase selected).

Delay: the Input delay can be set here. The delay (in ms or meter) can be set using the button up/down "Adj" with step 1ms or "Fine" with step 10.4us (max input delay = 480ms =163.5meter)

High Pass Filter: a Hp filter up to 48dB/Oct of Classic types is available there where the APEC sw is used for a cabinet working with a separate SUB.

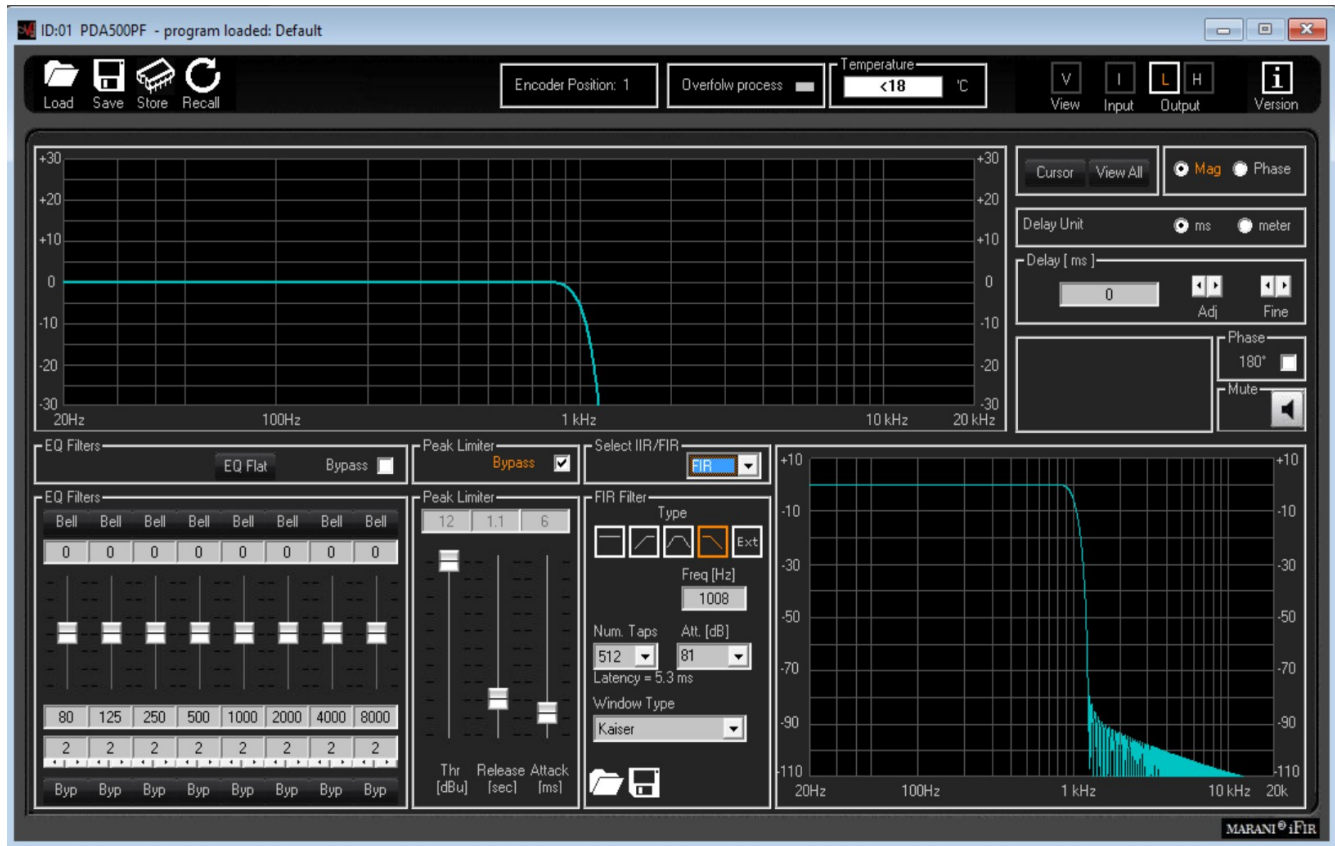
RMS Compressor: the **PDA500PF** is implementing a powerful RMS compressor on Inputs and Outputs, which compression coefficient is calculated on the base of tabled values giving a compression step of 0.1dB.

- **Bypass:** Compressor Active or not (Note that the all parameters here below are not accessible when the Compressor is set in Bypass)
- **Threshold:** from -10dBu to +20dBu step 0.2dBu
- **Ratio:** [input:Output] from 2:1 to 32:1 step 1
- **Knee:** from 0% to 100% (0% hard knee, 100% soft knee)
- **Release-time:** from 0.1sec to 3sec step 0.1 sec
- **Attack-time:** from 5ms to 200ms step variable
- **Make-Up:** from -12dB to +12dB step 0.1dB

Noise Gate: a noise gate is available

- **Bypass:** Noise Gate Active or not
- **Threshold:** from -80dBu to -50dBu step 5dBu
- **Release-time:** from 10ms to 1sec step variable
- **Attack-time:** from 1ms to 1sec step variable

OUTPUT WINDOW



All the processes available on the Output Paths can be edited within the Output Windows.

Eq Flat and Bypass Controls: the EQ Flat control is allowing to flat in one shot the 8 bands Parametric Eq. The Bypass control, instead, is allowing to temporarily Bypass the Eq, without resetting the current setting.

EQ Filters: within this frame, the all 8 filters of the PEQ can be defined in terms of type, gain, Freq, Q

- **Filters type:** on top of the Slider and the box showing the gain level, there is a button that once pressed is opening a wide list of filters type which can be selected, for each filter of the parametric Eq. The available filters are **Bell** (constant Q), **Hi/Low Shelving Q** (Symmetrical with variable Q), **Lp/Hp Q** (12dB/Oct second order, variable Q) and **Notch**.

- **Gain:** the filter types which can gain, have the gain controlled by a slider ranging from -15dBu up to +15dBu.

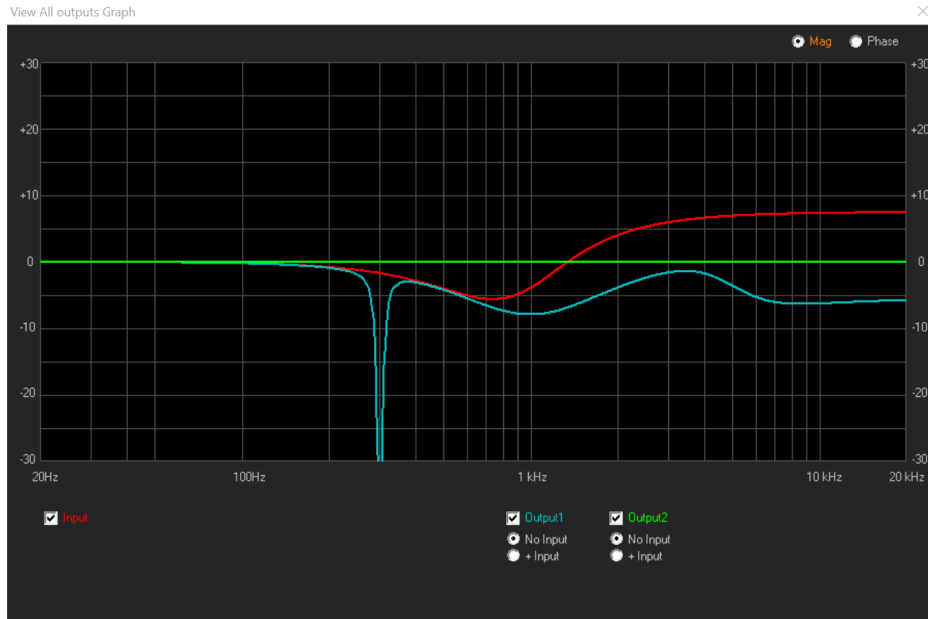
- **Freq:** here can be inserted the Center, Cutting Frequency of the selected filter, ranging from 20Hz up to 20kHz.

- **Q:** there where usable, the Q can be set with a range depending from the selected filter.

- **Byp:** any filter of the 8 of the Parametric Eq, can be singularly bypassed, so to have the possibility to a quick A/B comparison between the signal when active or not the filter currently in editing phase.

Cursor: clicking on this control, cursors will appear on the graphic showing the amplitude response of the parametric Eq, for the setting of the filters' gain by mouse action.

View All: this control is opening a window showing the full Eq setting on each Input or Output or combination of them.



Mag and Phase: it is possible to decide if showing on the Window's graphic the Magnitude response of the Parametric Eq setting (Mag selected) or the Phase response (Phase selected).

Delay: the Input delay can be set here. The delay (in ms or meter) can be set using the button up/down "Adj" with step 1ms or "Fine" with step 20.8us (max input delay = 680ms = 231meter)

RMS Compressor: the **PDA500PF** is implementing a powerful RMS compressor on Inputs and Outputs, which compression coefficient is calculated on the base of tabled values giving a compression step of 0.1dB.

- **Bypass:** Compressor Active or not (Note that the all parameters here below are not accessible when the Compressor is set in Bypass)
- **Threshold:** from -10dBu to +20dBu step 0.2dBu
- **Ratio:** [input:Output] from 2:1 to 32:1 step 1
- **Knee:** from 0% to 100% (0% hard knee, 100% soft knee)
- **Release-time:** from 0.1sec to 3sec step 0.1 sec
- **Attack-time:** from 5ms to 200ms step variable
- **Make-Up:** from -12dB to +12dB step 0.1dB

Peak Limiter: a Peak Limiter for Loudspeaker protection is available

- **Bypass:** Peak limiter Active or not
- **Threshold:** from --10dBu to 20dBu step 0.2dBu
- **Release-time:** from 100ms to 5sec step variable

X-over Section

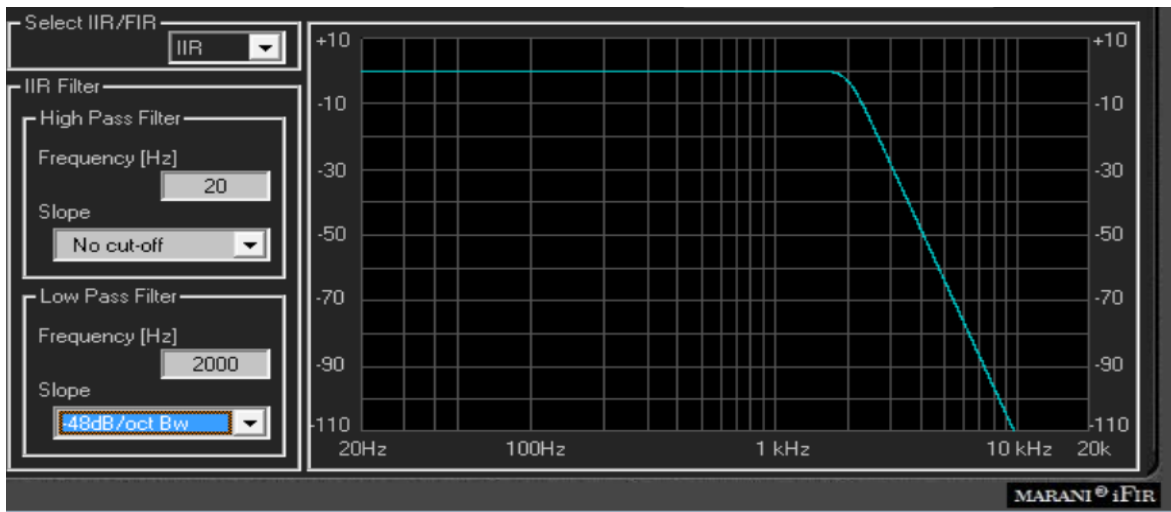
The X-over Section of the PDA500PF is a particular and pretty powerful one, which can even become “tricky” if used in combination with the Parametric Eq.

Basically the PURE X-overing Section can be in standard operation, operated by IIR filters or FIR filters. The FIR filters used for the X-over Section are 512 Taps ones and therefore cannot be used for cutting frequencies lowest than 250Hz.

Back the the X-over, for cutting frequencies lower than 250Hz, or for any other reason coming from the user, IIR classic Lp/Hp filters can be used.

IIR X-over

If selecting IIR in the “Select IIR/FIR” frame, the X-over section is implementing IIR Hp/Lp, and the interface for the IIR filters setting is the following

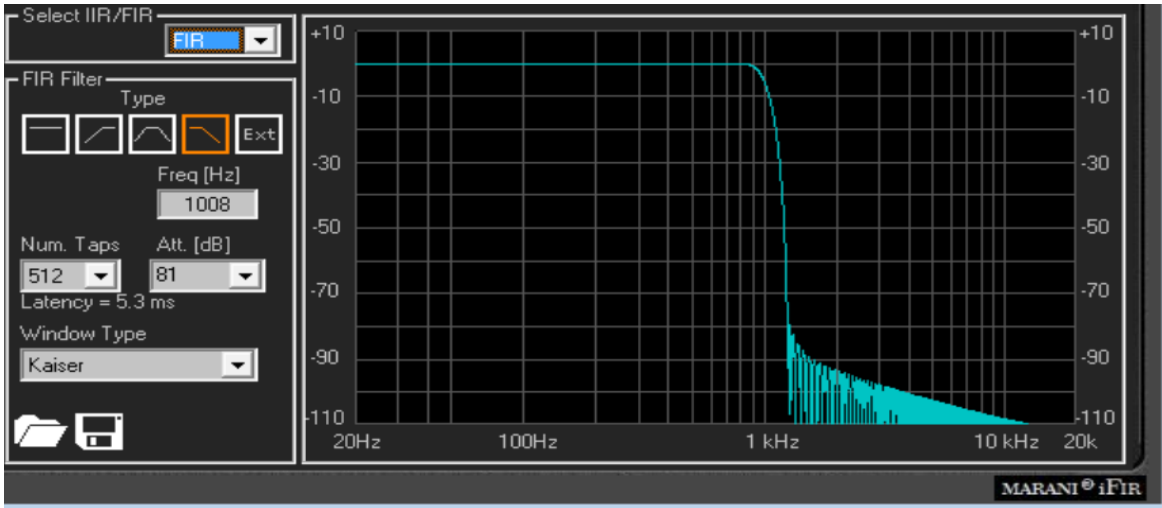


To edit the Hp or LP filter cutting frequency, input the value on the text box and press enter (Frequency from 20Hz to 20kHz step 1Hz)

To edit the Hp or LP filter type and slope, click on the relative Box and select the type (bypass, Butterworth, Linkwitz Riley, Bessel)

FIR X-over

If selecting FIR in the “Select IIR/FIR” frame, the X-over section is implementing FIR Hp/Lp, and the interface for the IIR filters setting is the following



Here is possible to select the FIR type, and the all parameters for its configuration. Particularly,

- click on the icon to select the Filter **Type** (High Pass, Band Pass, Low Pass or Bypass)
- insert the Cut-Off **Freq[Hz]**, input the value on the text box and press enter (Frequency from 250Hz to 20kHz step 1Hz)
- select the number of the **Num. Taps** of the FIR (from 256 to 512). The FIR latency is dependent from the number of the taps and, being the FIR running at 48kHz, the latency can be calculated as follow:

$$\text{FIR Latency} = \text{Ntaps}/2 \times \text{Ts} = \text{Ntaps} \times \text{Ts}/2 = \text{Ntaps} \times 10.4\mu\text{s}$$

- select the max attenuation **Att.[dB]** for the out band signal
- select the **Window Type** of the filter

Once the filter is set, it will be displayed in the window aside the parameters set. In order to make it active and performing on the Signal path, need to load it within the **PDA500PF** clicking on the blinking icon



which is appearing as soon as a modification ton the currently loaded FIR is done.

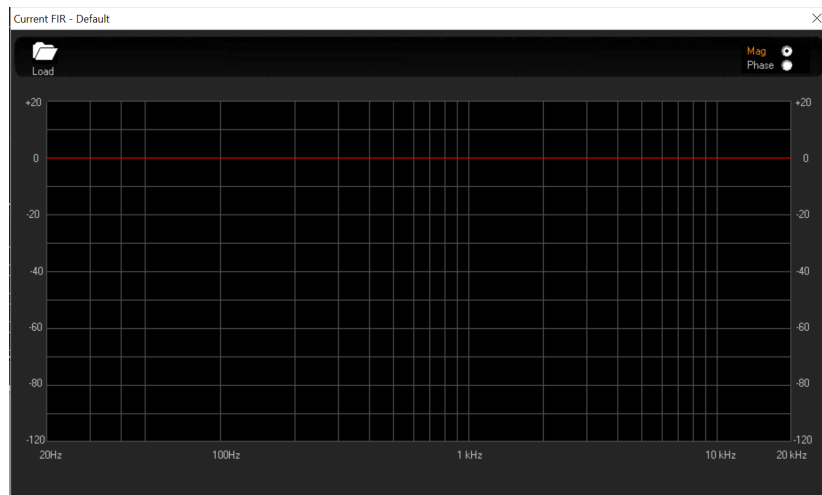
Once the set FIR is uploaded into the **PDA500PF**, the following Icons will become available



Allowing to save into a Pc directory the current edited filter for future use, or to upload from a Pc directory, previously saved FIR filters.

Also for the FIR used in the Output X-over Section, it is possible to upload coefficients imported from text files coming from Third Party Sw.

- **EXT** Icon: here external coefficients can be imported into the FIR. Clicking on the “EXT” Icon button, the following window open and from the clicking on the LOAD icon, a .txt file (see NOTE B) with the Third Party calculated coefficients can be imported.



The Amplitude and Phase responses can be seen on the graphic which will show the characteristics of the imported filter.

- **View** : once imported, or even internally generated, the FIR coefficients, the response of the FIR can be seen recalling it on graphic just pressing the 'View ’’ button.

FIR on Output not as X-over filter, using imported coefficients.

If the user would like to use an IIR filter as X-over, being necessary in example a cutting frequency below 250Hz, but still having available the FIR in the X-over section, a trick can be used for getting the result.

Particularly, we could say that we have three possible way of using the FILTERS of the output channels.

1. 4 bands parametric Eq + IIR X-over (no FIR used): the X-over cutting frequencies can be set everywhere by the user, without limitation
2. 4 bands parametric Eq + FIR X-over (no IIR used): the X-over cutting frequencies can be set everywhere above 250Hz/300Hz by the user, no reasonable slopes can be achieved with the FIR for lower cutting frequencies.
3. If necessary to set a cutting Frequency for a Lp or Hp below 250Hz, but still we'd like to have the IFR available, eventually for working with imported coefficients, a trick for getting the result can be used. If we'd like to combine the IIR filters for Hp/Lp for low frequencies and still have FIR filters available, let say for further filtering, as PEQ one or any sort of phase configuration, then the user can operate in the following way, "sacrificing" some of the filters of the PEQ: the user can create the Low Freq Hp or Lp, using the variable Q Hp/Lp filters available in the Peq. They are in cascade, therefore, for a Hp/Lp 48dB/Oct, you can use the 4 Filters of the PEQ..

Anyway, in this way, the user can have the IIR Hp or Lp and an available FIR 512Taps which can be used as the user desires if uploading the coefficients from an external source, using the FIR itself also for eventual Peq setting together with other features.

The user can see each one of the PEQ Filters as a IIOrd Cell, which if set as Hp or Lp with variable Q, can give a 12dB/Oct filter.

In example, setting the Filter N1. of the PEQ, set as Lp-Q at 1kHz and with $Q=0.7$, then can be gotten a 12dB/Oct Butterworth Lp Pass with 1kHz Cutting Frequency.

Selecting now the second filters also as Lp-Q, and also with 1kHz cutting Frequency, and setting the Q of the first filter to 0.55 and the Q of the second to 1.3, then a 4th order Butterworth can be set up.

Basically, to get Lp, filters from 12dB/Oct, up to 48dB/Oct, need you to set as follow the Q, given the same cutting freq. to all of them:

Butterworth

12dB/Oct -----> One Cell / Q1=0.70
24dB/Oct -----> Two Cells / Q1=0.55, Q2=1.30
36dB/Oct -----> Three Cells / Q1=1.95, Q2=0.70, Q3=0.50
48dB/Oct -----> Four Cells / Q1=2.55, Q2=0.90, Q3=0.60, Q4=0.50

Linkwitz-Riley

12dB/Oct -----> One Cell / Q1=0.70
24dB/Oct -----> Two Cells / Q1=0.70, Q2=0.70
36dB/Oct -----> Three Cells / Q1=0.55, Q2=1.30, Q3=0.70
48dB/Oct -----> Four Cells / Q1=0.55, Q2=1.30, Q3=0.55, Q4=1.30

Bessel

12dB/Oct -----> One Cell / Q1=0.50
24dB/Oct -----> Two Cells / Q1=0.55, Q2=0.80

NOTE A

FIR WIZARD - User Guide v1.0

The FIR Wizard is a smart and easy-to-use software tool designed for enhancing the audio quality of a loudspeaker by compensating the linear distortions, both in magnitude and phase, through FIR filtering.

The tool guides the user through the loudspeaker IR measurement and allows generating state of the art equalization FIR filters, with few essential synthesis parameters, that “invert” unwanted phase shift, with the aim to linearize the phase response of the loudspeaker.

A magnitude equalization option is also available, allowing to flatten the amplitude response of the loudspeaker, or to set up a target magnitude curve.

Moreover, the Wizard allows to reduce the nominal latency of the generated FIR filter, in such way to fulfill the latency constraint of live sound systems, while preserving, almost unchanged, the correction capability of the filter itself.

Preliminary

The FIR Wizard is a tool to aid the user in “optimizing” the response of a loudspeaker, but driver choice as well as good cabinet and Xover design still remain the primary steps to achieve the best sound quality.

In general, before using the FIR Wizard, the user should do all the needed steps to correctly setup its sound system and to start from a reasonable loudspeaker response.

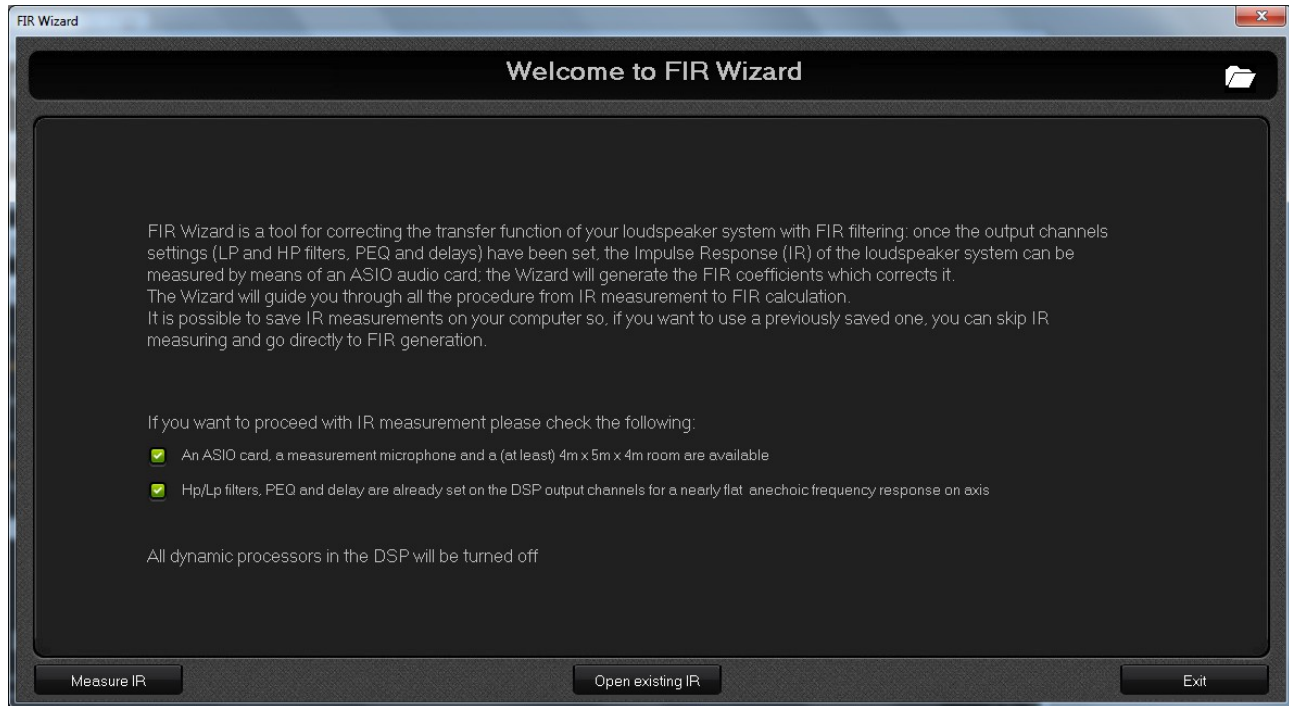
For example, in a multi-way loudspeaker, the response of each driver should be aligned in time to avoid dips in the Xover point, that can not be filled by FIR filtering without introducing audible artifacts.

Moreover, a preliminary coarse loudspeaker equalization is required to start from an almost flat overall frequency response, especially when FIR filtering is used to compensate both magnitude and phase. This allows to avoid strong boost of some frequencies, that may introduce undesired audible ringing effects.

If a non-flat target response is desired, the preferred procedure should be anyway to start from an nearly flat response and apply a post equalization after computing the compensation FIR filter, or, even better, to apply a target curve in the FIR filter itself.

Starting Up

The image below shows the home page of the FIR Wizard.

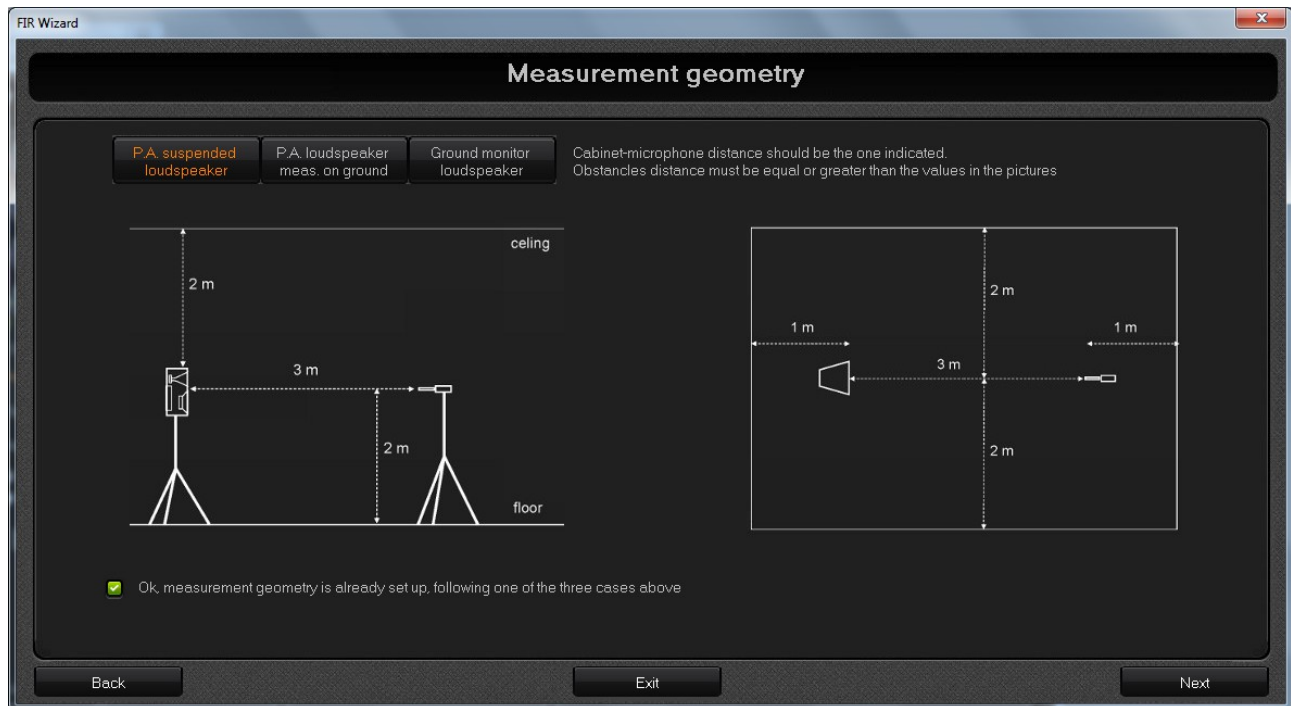


From the home page, the user can choose to:

- 1) Start from an existing Wizard project: click the open button at top right of the screen and a file browse window appears allowing to select a Wizard project file (.zwx)
- 2) Start from an existing IR, that has been already measured and saved from the FIR Wizard: click the "Open existing IR" button at the bottom center of the screen and open an IR file (.oir) from the file browse window.
- 3) Start with a new IR measurement: to do this, be sure you have an ASIO sound card, a measurement microphone and, at least, a 4m x 5m x 4m (L x W x H) obstacle free measuring room (if the available space is a little bit smaller check for possible tricks described in the following section). Moreover, as already described in the "Preliminary" section, it is suggested to set up the Xover, the PEQ and the output delays to have a nearly flat frequency response on axis. Then, click the two reminder checkboxes to enable the "Measure IR" button and click it to proceed with IR measurement steps. At this time, all the dynamic processes in the DSP will be automatically turned off, as well as the equalization FIR filter that could be already active.

Measurement Geometry

After clicking the “Measure IR” button, the “Measurement Geometry” page is shown and three different geometry set up is suggested.



Clicking the buttons at top left of the screen, the three measurement scenarios are shown (vertical section on the left, planimetry on the right). This is just a visualization and reminder for the user that should check every time the correct geometry and choose one of the three options. The choice doesn't affect any following measurement or calculation parameters, which are actually compatible with all the three scenarios.

For each measurement geometry, the minimum distances between loudspeaker, microphone and obstacles has been computed in order to maintain a minimum distance between:

- 1) the direct sound wave path, from the loudspeaker to the microphone
- 2) the reflected sound wave path, from an obstacle (e.g. the floor or the wall) to the microphone

This will allow the Wizard algorithm to cancel out reflected components in the IR without corrupting the direct component information, at least in a useful, well defined frequency band.

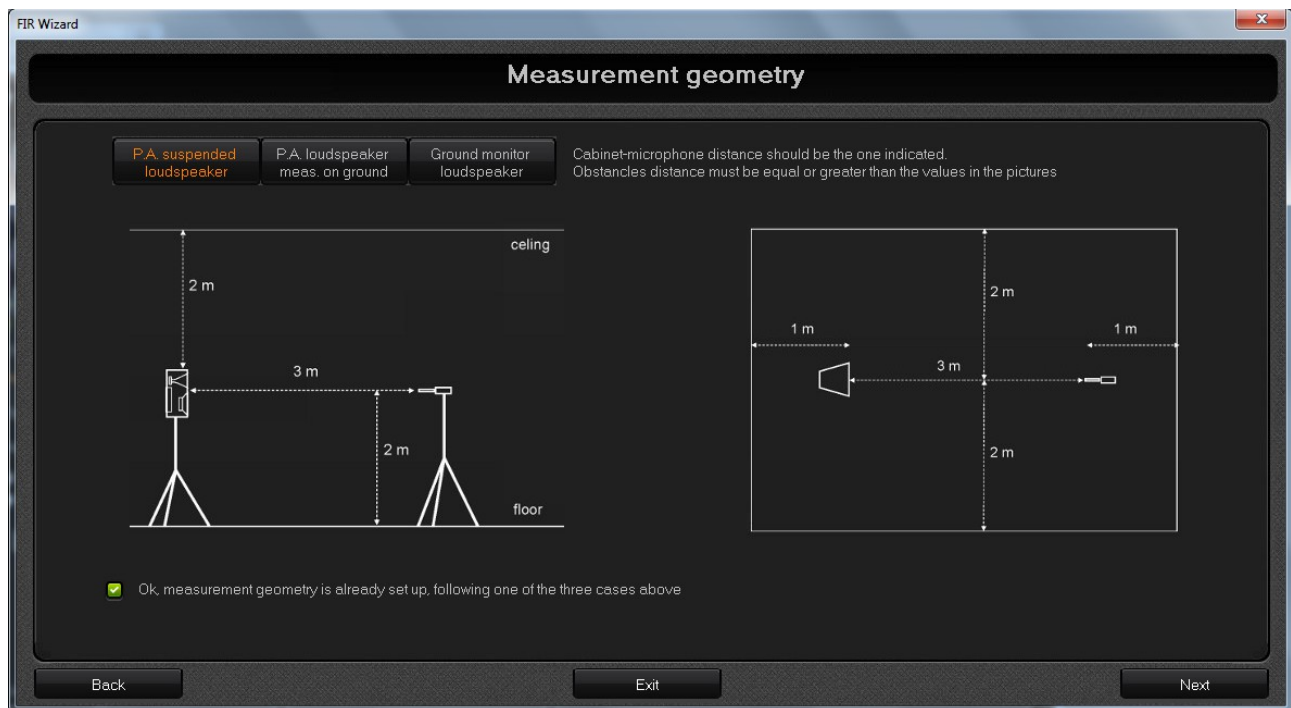
This technique creates a sort of virtual anechoic room, which is fundamental to make the FIR equalization strictly related to the sound system and independent from the measurement space: in this way, the FIR will work fine also in other environments.

Here below, the three measurement scenarios are briefly described.

1) Loudspeaker system and microphone suspended at half height

In this case, the measurement space should be at least 4m x 5m x 4m (L x W x H).

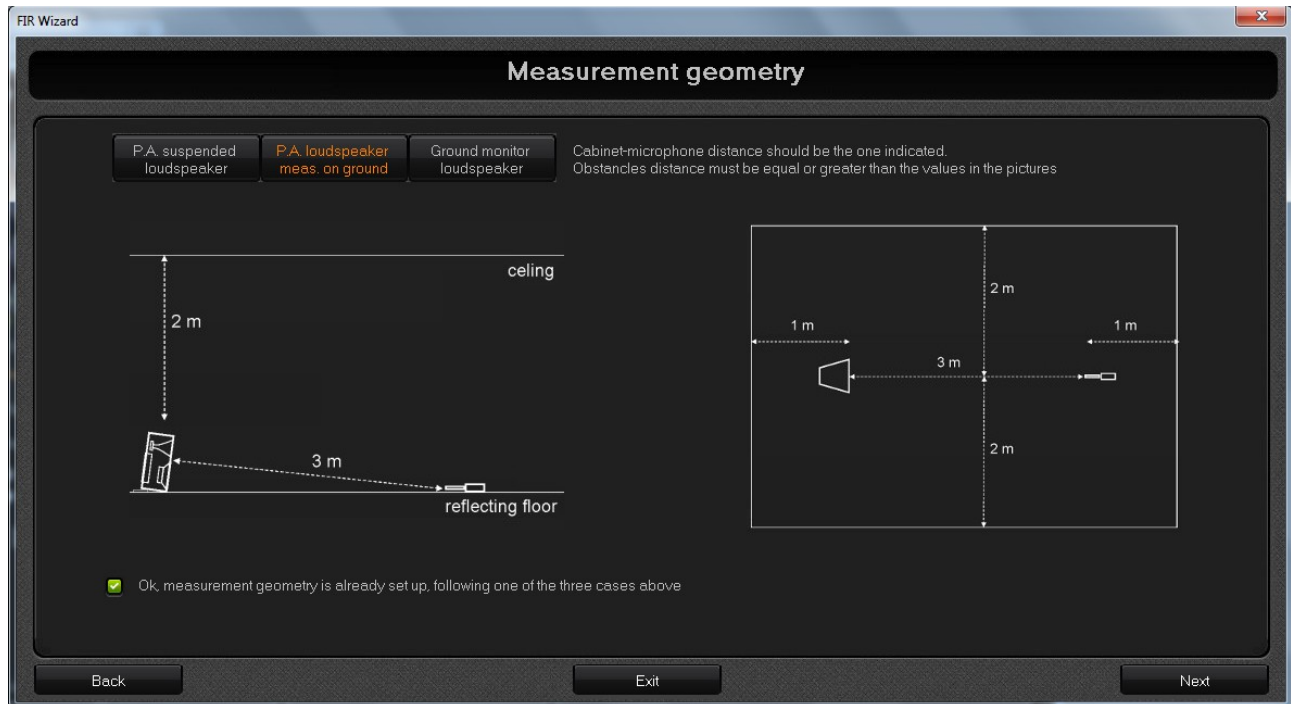
The microphone should be in axis with the loudspeaker; obstacles should be at least 2m far from the microphone.



2) Loudspeaker system and microphone lying on the ground

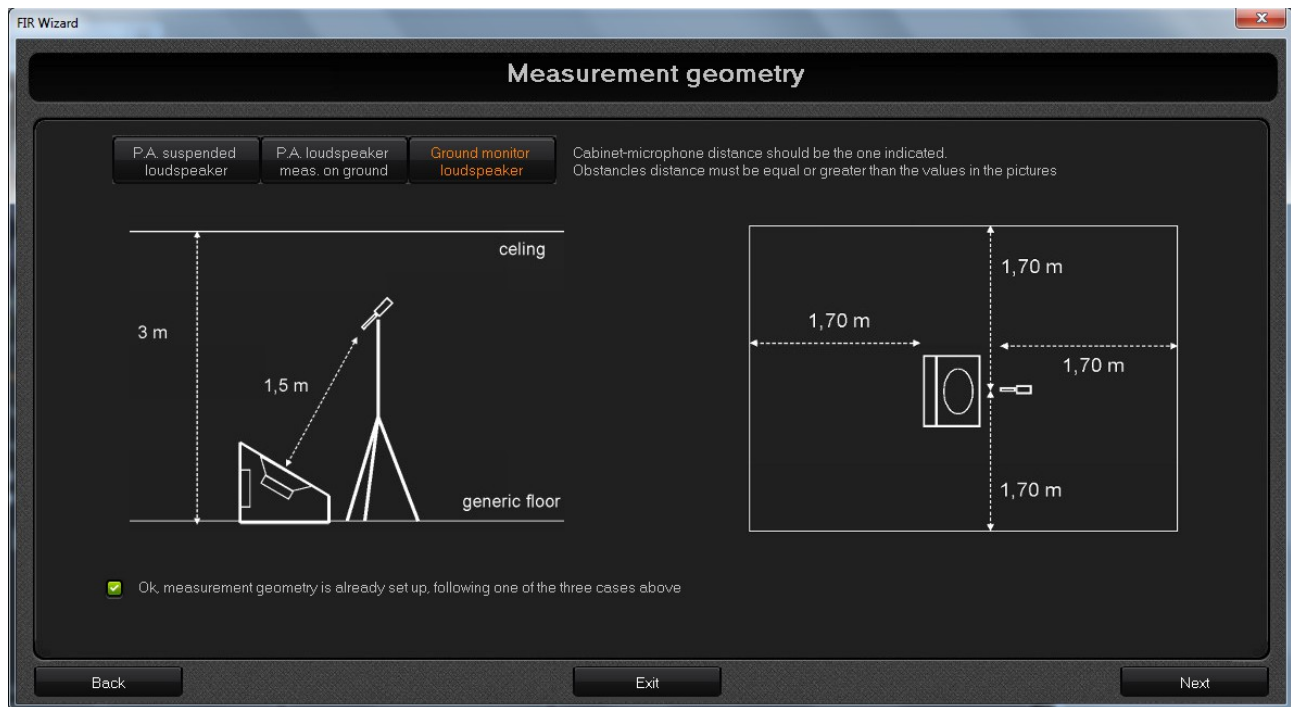
The loudspeaker must be tilted to point the microphone, that should be strictly in contact with the floor. The floor should be very reflecting (smooth), especially beneath the mic: a fine metal sheet under the mic can be useful for this purpose.

As the picture below shows, in this measurement setup a lower room height is required with respect to the preceding scenario, so it can be used when the room height is less than 4m.



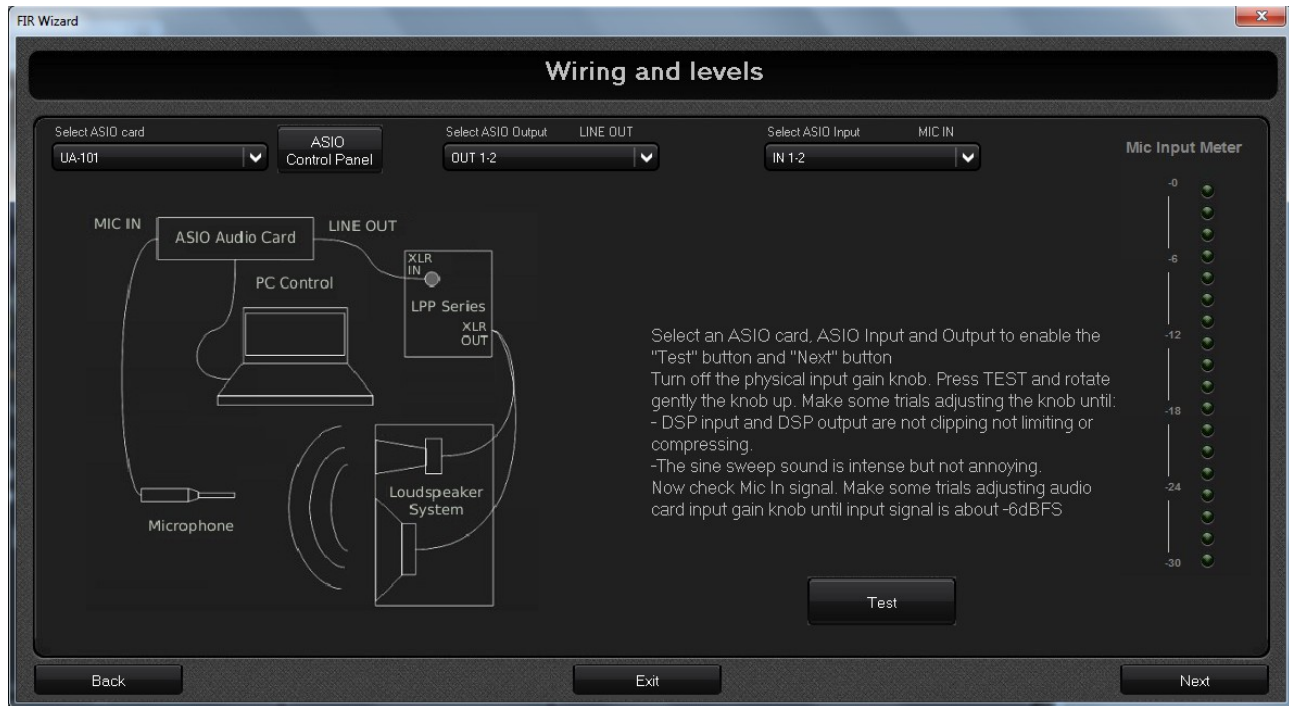
3) Ground monitor in operating position

The microphone should be in axis with the loudspeaker at an height of about 1.7m, where the listener's head will stay.



So, arrange the measurement choosing one of the three geometry setup and click "Next" button to proceed.

Wiring



Connect an ASIO audio card to the PC where the FIR wizard is running, with its proper connection (USB, Firewire, etc.). Connect the (normally balanced) ASIO card output (XLR or TRS) to the balanced input (XLR) of the DSP device (e.g the LPP Controller).

Connect the microphone output to the mic input (normally XLR) of the audio card and turn the phantom power (if needed). If the mic has its own preamplifier, connect it to a line input of the audio card. Finally, connect the DSP device to the loudspeaker system.

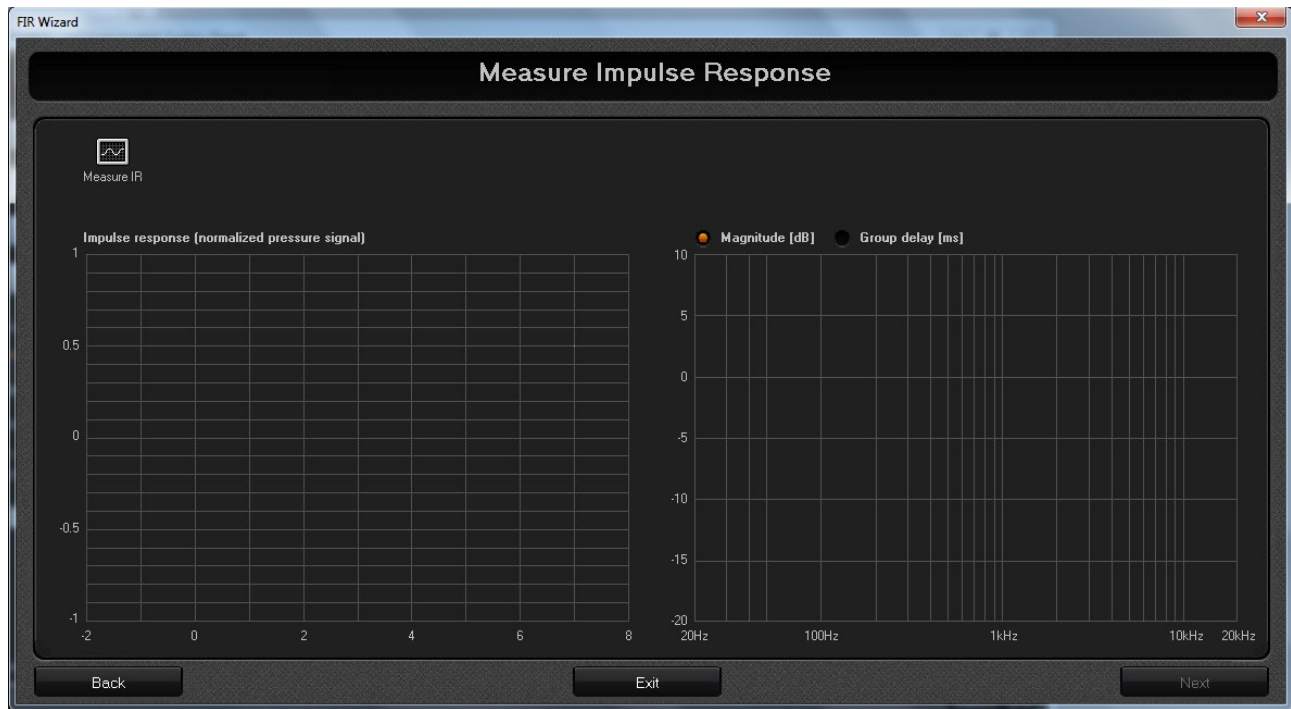
Therefore, in the FIR Wizard "Wiring and levels" page, select the ASIO audio card and the right input/output channels. using the list box at the top of the screen,

Now, click the "Test" button to start the microphone test, making sure the loudspeaker amplifier volume is low. Then, start rising the amplifier volume until a strong but not annoying "sine sweep" is heard. Click "Test" button again and adjust the microphone preamp gain until the measured signal peak reach roughly the -6 dBFS value.

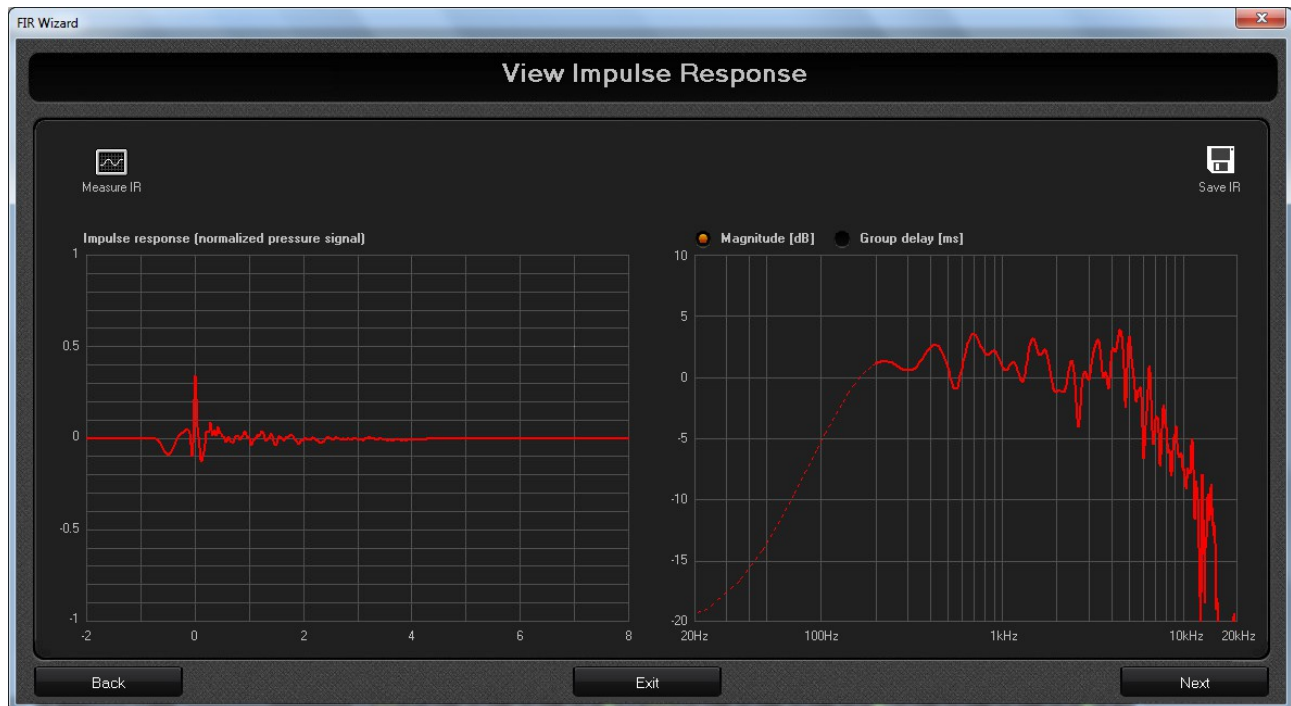
If during the microphone test some glitches occur, it is probably due to a latency issue of the sound card. To solve the problem, open the sound card control panel clicking the "ASIO Control Panel" button and try to choose an higher latency value (or, equivalently, a large buffering).

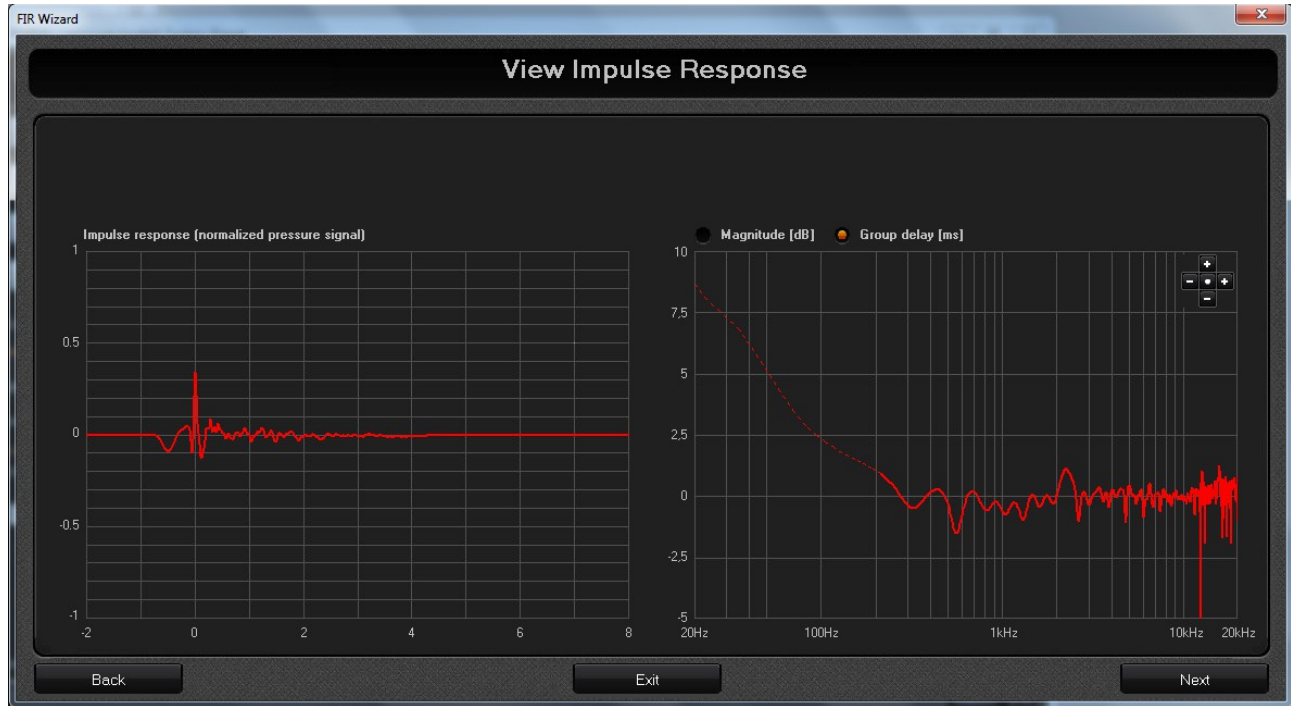
Click "Next" button and you are ready to measure the IR of the loudspeaker.

IR Measurement



Clicking the “Measure IR” button at the top left of the screen, the measurement process starts and an activity progress bar shows the activity completion status. Wait in silence for the process completion. After the sine sweep, the Wizard starts the IR computation and the measured IR is plotted on the left graph with its peak centered on zero, while the magnitude and group delay frequency response is shown on the right graph.





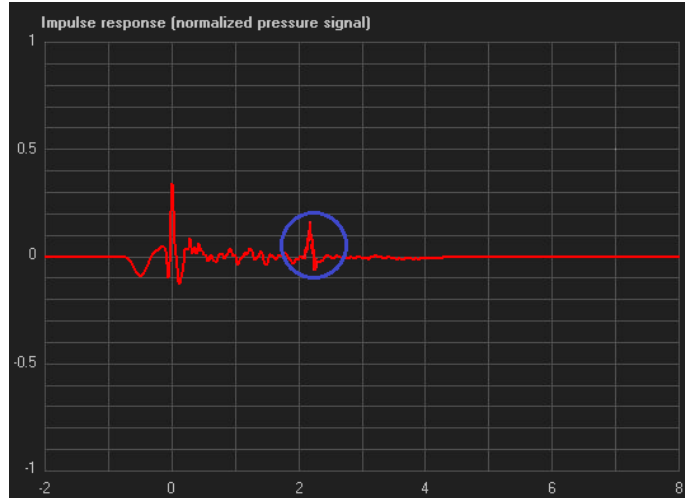
Discontinuities in the group delay view may appear (as the picture above shows near 12kHz), due to the poor phase computation accuracy in the frequency points where the magnitude response has a very deep dip.

Now, you can save the IR by clicking the “Save IR” button at the top right of the screen. Choosing to start from an existing IR at the Wizard home page would have lead you directly here.

The IR computed by the Wizard (and shown in the left plot) is a cut of the real measured IR at the microphone position. The calculation engine applies a windowing process to the first 10ms of the real IR (centered on its peak value) to remove unwanted obstacle reflections, in such a way to provide an environment independent IR.

This remarks the need to arrange the measurement environment as described in the “Measurement Geometry” section.

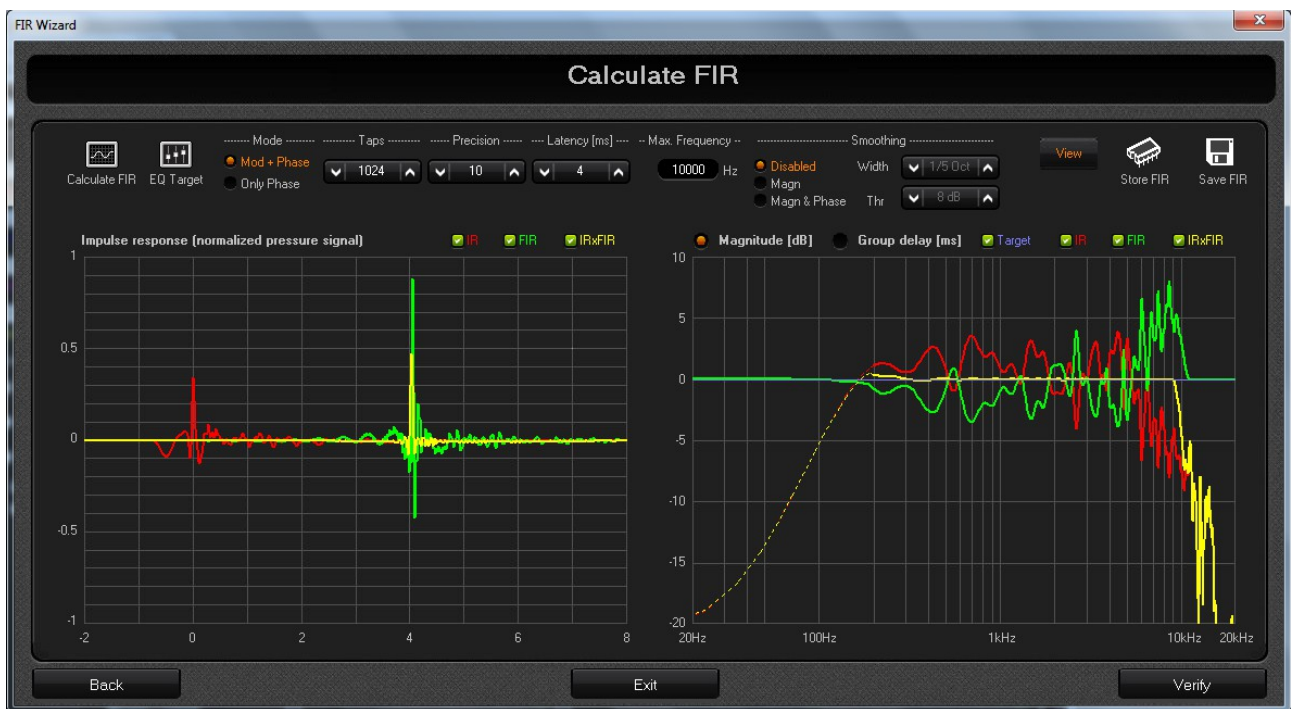
The picture below shows an example of how the IR plot would appear when the geometry constraint is not respected: reflections closer to the IR peak (see the blue circle) are not removed by the windowing process. This implies that the compensating filter will try to “correct” also the room response, so to violate the assumption of the “environment independent equalization”.



As a consequence of the windowing process, the computed IR is not accurate in the low frequency range: it is not possible to estimate the frequency response below the reciprocal of the analysis time interval, due to the time-frequency duality. So, to avoid misleading view, the frequency response plot shows a dotted line below 200Hz.

To proceed with the FIR filter computation, click "Next" button and the "Calculate FIR" page will be shown.

FIR Calculation

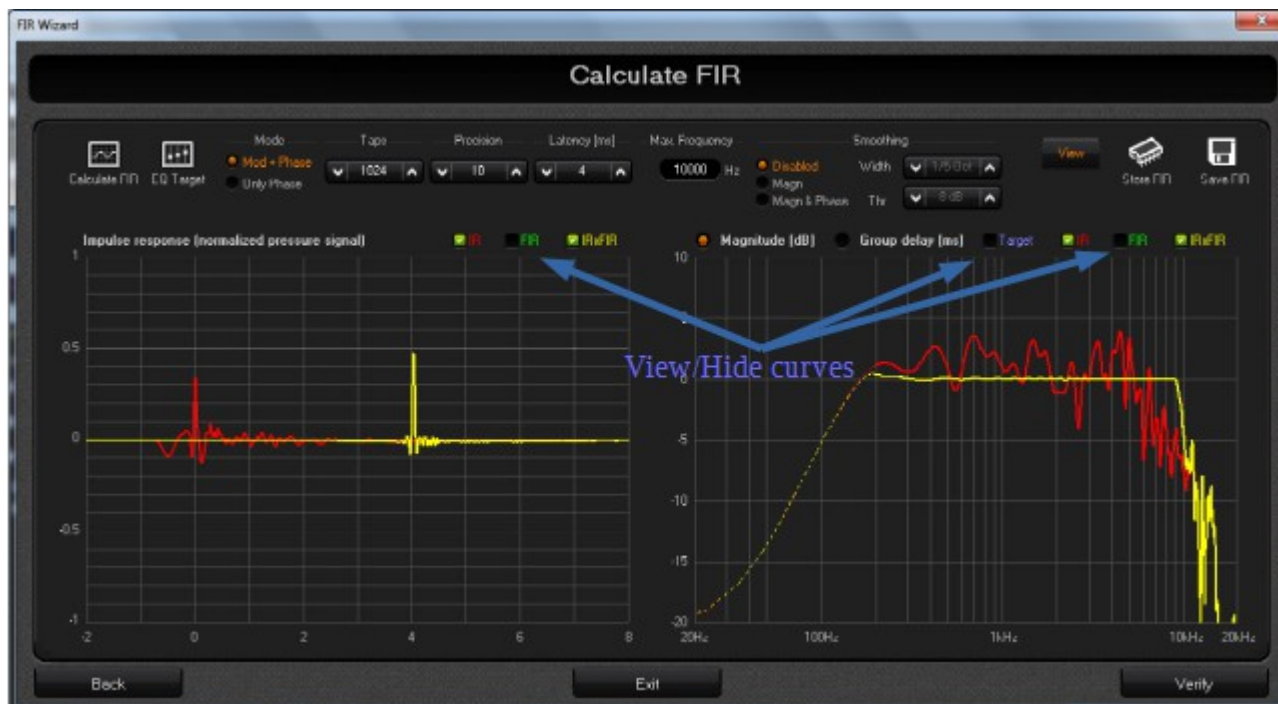


In this page the FIR filter coefficients are computed starting from default settings (or from the parameters set of the FIR Wizard project, when a project file is recalled at the Wizard home page).

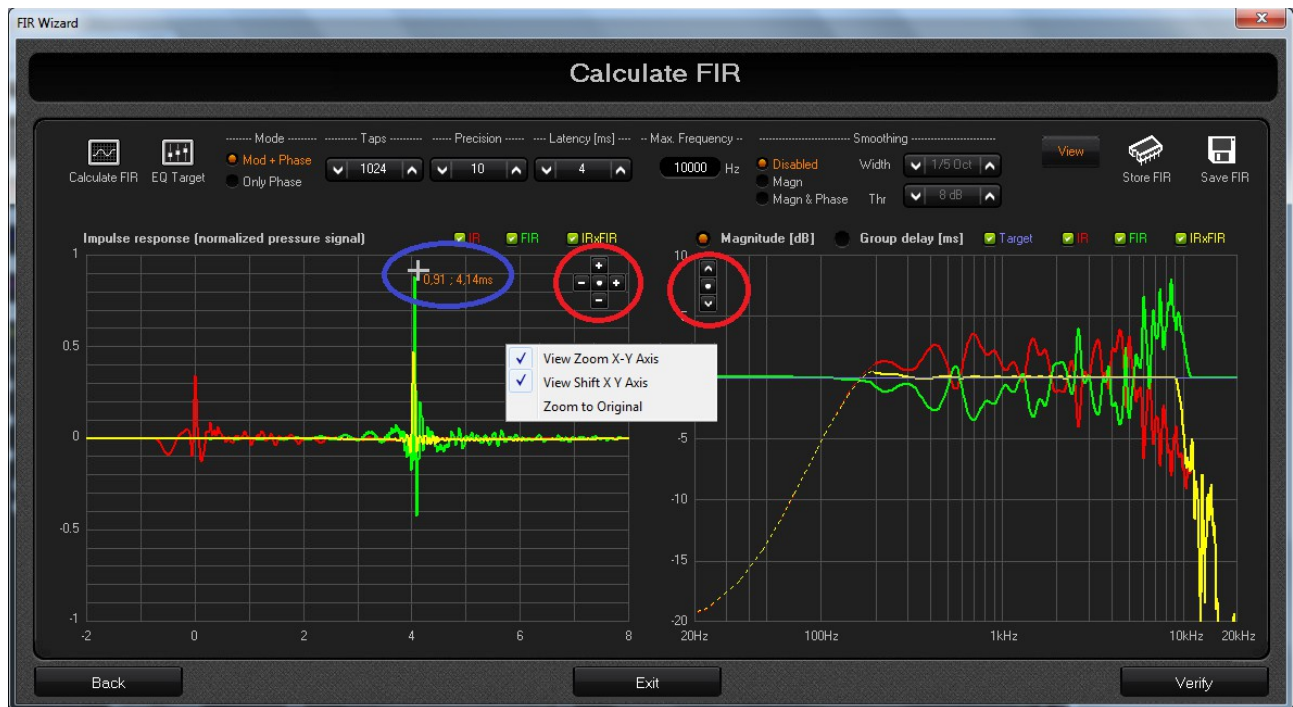
The two graphs in the screen show the time and frequency response of:

- 1) the loudspeaker IR (red line), as it appears in the preceeding page.
- 2) the compensating FIR filter (green line), which is normalized so that when the FIR filtering is enabled or disabled, the sound loudness remains roughly unchanged. This is very important for performing listening tests and it is a good feature in general.
- 3) the convolution between the IR and the FIR filter (IRxFIR, yellow line). It represents the result of the IR compensation process, that corresponds to what would be the IR if it is measured with FIR processing enabled, or in other words, the new loudspeaker IR when the computed FIR filter is applied.
- 4) The target magnitude curve, that is shown only in the frequency domain when the magnitude view is selected. By default the target magnitude is flat, but it can be changed by the user as described afterwards.

Curves in the graphs are overlaid to allow a better comparison between the IR, the FIR filter and the resulting convolution. Anyway, if needed, each curve can be hide acting on the checkboxes at the top right of the plotting area as shown in the picture below, where FIR filter and target curve are hidden.



Left clicking a point in the plotting area allows to show the cursor coordinates, while right clicking shows the plot zoom menu.



Let's now analyze in more details the linearization capability of the computed FIR by comparing the time and frequency domain plots of the IR and the IRxFIR curves, for a flat target magnitude.

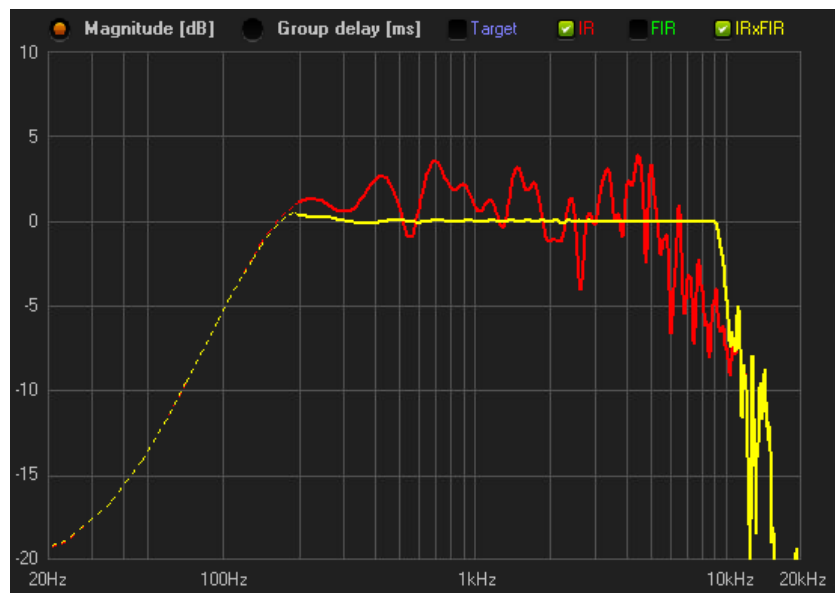
In the time domain what we should ideally get is a unitary pulse centered at 0ms (Dirac delta), that is the TD equivalent to a flat frequency response.



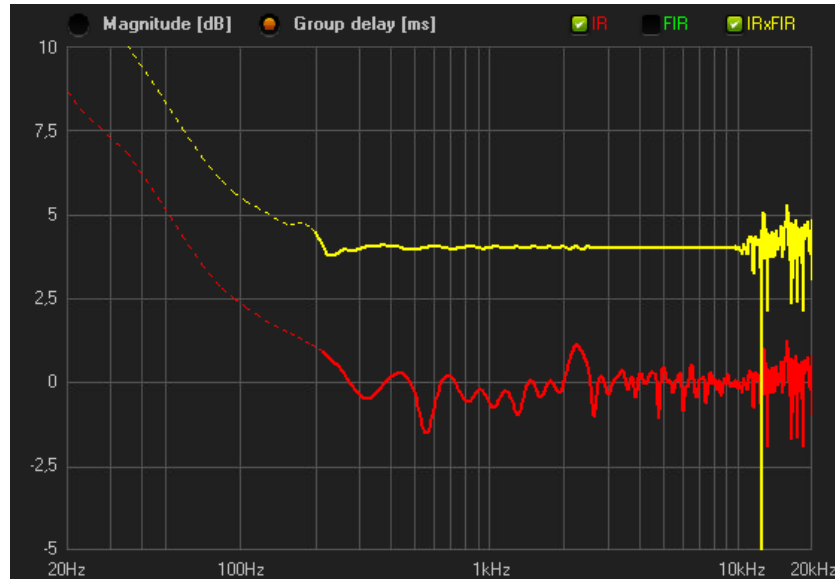
Looking the chart above, we can immediately note that the IRxFIR (the yellow curve) is much closer to the ideal Dirac delta than the measured IR (the red curve). This is the improvement added by the FIR filter.

The slightly difference between the IRxFIR and the Dirac delta is due to the fact that the FIR filter is not working on the whole frequency response, but it is limited to a frequency range from about 200Hz to a MaxFreq (10kHz in this example). The lower limit depends on the IR measurement technique (as described in the “IR Measurement” section), and on the poor resolution of the FIR filter at low frequencies. The upper limit (MaxFreq) is a user defined parameter that will be described afterwards. Moreover, the time domain graph shows the latency added by the FIR filter, that is defined as the time distance between the peak of the IR and the peak of the IRxFIR curve.

The magnitude plot here below shows the flattening effect of the FIR filter in the yellow curve, as well as the frequency range where the filter is acting.



The graph below shows the group delay flattening effect (see the yellow curve), that is the main achievement of the FIR filter: a flat group delay means a linear phase of the sound system (the group delay is the derivative of the phase with respect to the frequency).



Let's now describe the user parameters related to the FIR filter generation. Note that, every time a user parameter is changed, the “Calculate FIR” button at the top left of the screen start blinking, meaning the FIR filter needs to be recalculated. Once the button is clicked the synthesis algorithm computes the FIR filter with the new parameters and the corresponding plots are updated.

- **Mode**

Two different working mode is available:

- 1) “Magn+Phase”. When this option is selected the FIR filter will compensate both magnitude and phase of the measured IR. For the magnitude response a target curve can be applied.
- 2) “Phase only”. In this mode the FIR filter works on the phase response only, leaving the IR magnitude unchanged.

- **Max Frequency**

Before calculating the FIR, the first thing the user should do is to look the frequency response of the IR, mainly to check the upper cut frequency. This is useful to set the maximum operating frequency above which the FIR filter will be neuter leaving the original IR unchanged. Moreover, it should be remarked that:

- 1) The frequency response at very high frequencies can strongly depend on microphone position, since the most of audio transducers are very directive in this frequency range. So it is worth to try different microphone positions moving it of few centimeters, and choosing the one which gives the best result.

2) The frequency response at very high frequencies can be the real one or just what the microphone captures. If you are using a shoddy microphone, results in the high frequency range can be not affordable. This doesn't mean that a satisfactory FIR equalization can't be done on a slightly reduced band: just choose a max frequency value and try listening the result.

- Precision

The precision parameter, ranging from 1 to 10, is a measure of how accurate the magnitude correction should be, with respect to the target curve. In particular, if the IR shows a deep dips in the frequency response, using high values of the precision parameter implies that the FIR filter will totally compensate them, with corresponding peaks in the FIR frequency response, which can result in undesired audible ringing effects (tonal components will become visible in the time domain FIR plot). On the contrary, using low values of the precision parameter limits the FIR to accurately compensate only little lacks of the original IR, and to be less accurate on severe dips. This parameter should be tried empirically and results can be checked basing on plotted curves and listening tests.

The following picture compare two magnitude plots, where the precision parameter is set to 10 (max accuracy) in the right side graph and to 1 (min accuracy) in the left side graph. Group delay plots are not shown since the precision parameter doesn't affect the phase compensation accuracy.



- Smoothing

The smoothing parameters allows limiting the FIR filter action in such a way to “correct” the IR in its average behaviour. In other words, when the smoothing is enabled, the FIR filter doesn't apply a punctual IR compensation, but it is limited to correct only an averaged IR response.

Smoothing may slightly degrades the compensation capability of the FIR filter in axis (i.e. in the measured point), but it generally allows to enlarge the listening area where the IR correction is effective.

The smoothing can be applied to the magnitude response only or to both magnitude and phase.

For the magnitude smoothing, the algorithm scans the frequency response of the FIR filter searching for frequency points where the magnitude is above a given threshold Thr [dB], (or below $-Thr$ [dB]). In those points, a moving average window is applied, with the aim to smooth the magnitude peaks. This way, choosing a proper threshold value, a selective smoothing can be applied to the magnitude response.

The averaging window width is a control parameter, by which the user can increase or decrease the averaging effect.

The picture below shows how the magnitude smoothing works, by comparing the magnitude response before (left side plot) and after (right side plot) enabling the smoothing.

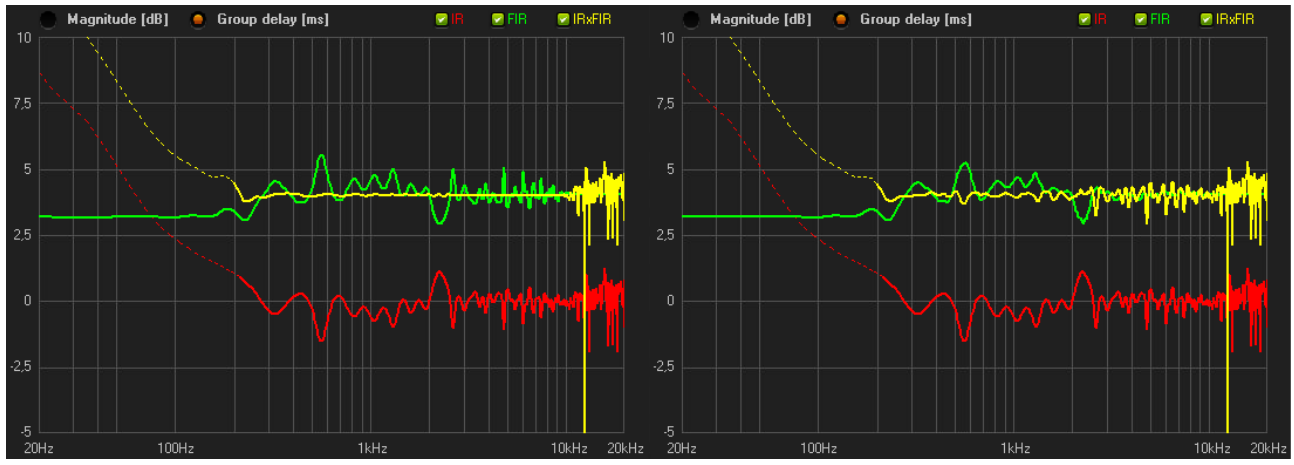


Setting up the threshold to 0dB, the smoothing is applied to the whole magnitude response, as shown in the following picture (left side plot, smoothing disabled; right side plot, smoothing enabled)



When the “Magn+Phase” mode is selected, the phase smoothing is always applied to the whole frequency range, even if the threshold is not equal to 0dB.

The following picture shows the phase smoothing effect, by comparing the group delay response before (left side plot) and after (right side plot) enabling the smoothing.

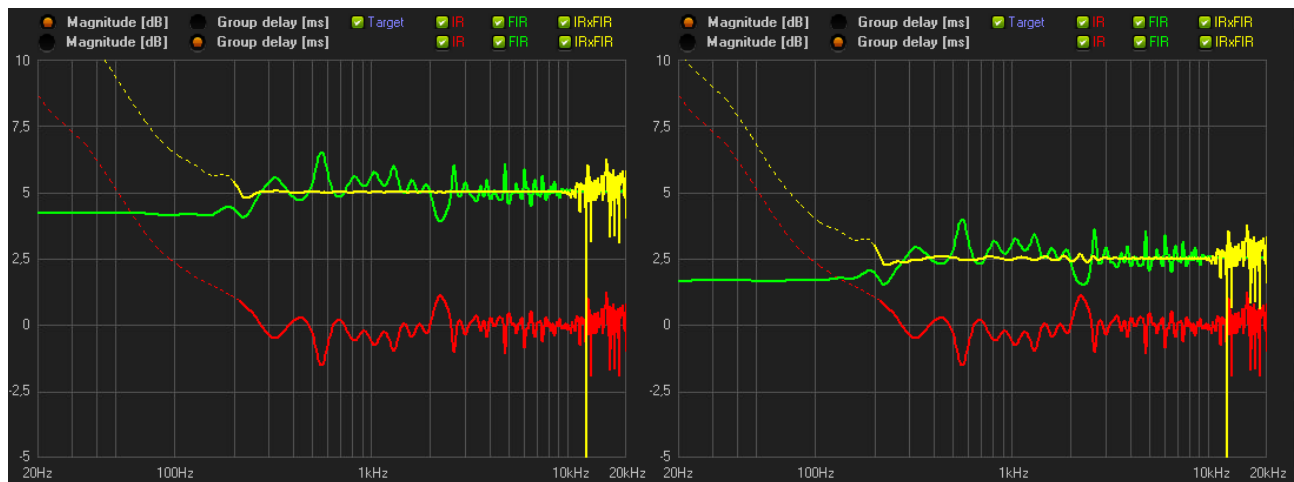


- Latency

The latency parameter allows to control the I/O delay added by the FIR filtering. As a general rule, higher latencies should be preferred where the compensation accuracy is the main target, while lower latency values is often used in live sound system, where the I/O delay is a critical application parameter.

The effectiveness in reducing the latency, leaving unchanged the compensation capability of the FIR filter, depends on the original loudspeaker IR, but, in the most cases, latency values greater or equal to 2.5ms allow to preserve almost unchanged the filter correction effects.

The following pictures compare the frequency responses (both in magnitude and group delay) of two filters with a latency value of 5ms (in the left side graph) and of 2.5ms (in the right side graph).



Reducing the latency below 2.5ms is even possible, with a lower compensation accuracy.

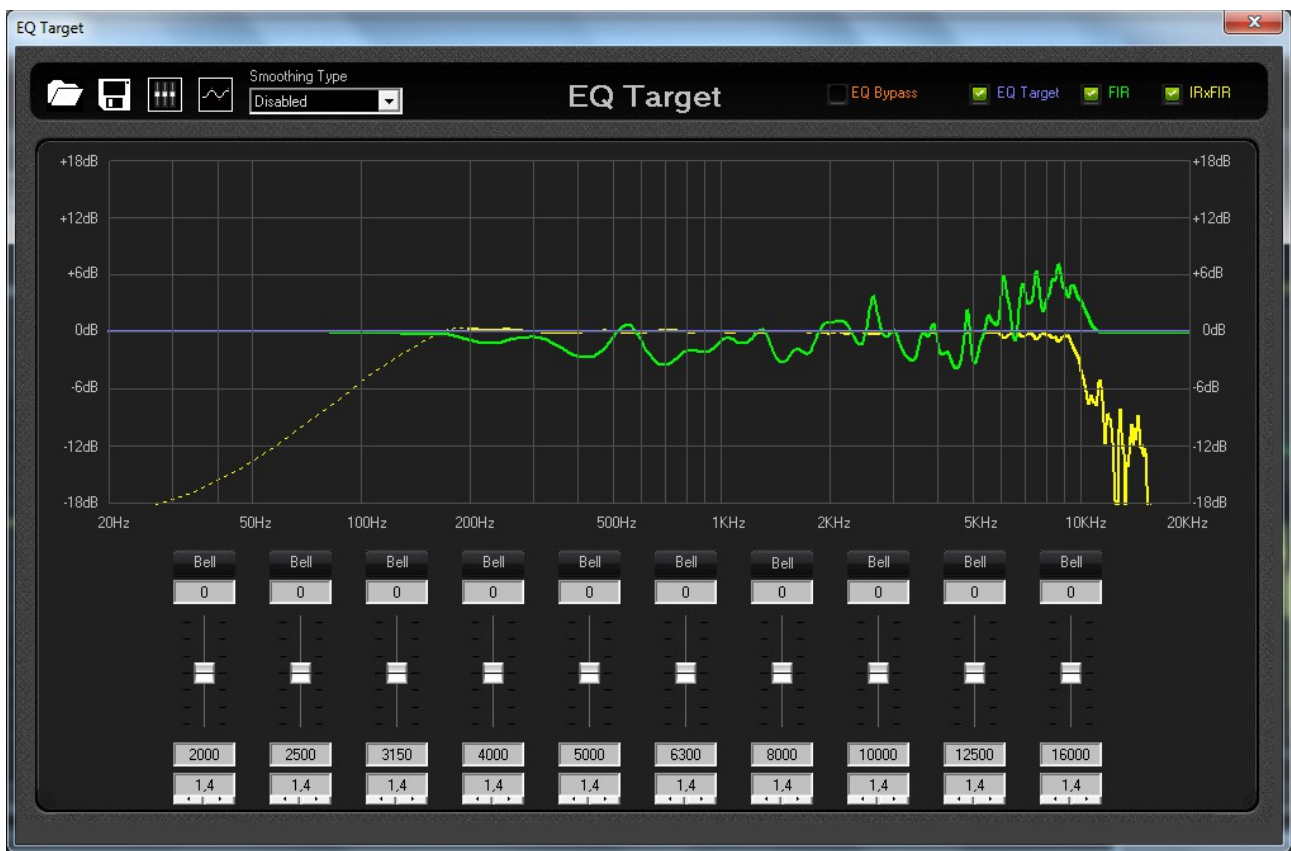
The picture below shows the magnitude and group delay plots for a latency value equals to 1.5ms: the frequency response starts to be significantly different from the ideal one, especially for the magnitude response (see the yellow curves)



As a general rule, it is suggested to try different values for the latency parameter and choose the lower one that still gives satisfying equalization results for the particular application, basing on plotted curves and listening tests.

- Target Curve

To set up a target magnitude curve, click the “EQ Target” button at the top left of the “Calculate FIR” page, and the “EQ Target” window is shown.



In the “EQ Target” page, the magnitude response of the target curve (blue line), of the FIR filter (green line) and of the IRx FIR (yellow curve) are overlaid. Curves in the plot can be hidden through the corresponding checkboxes at the top right of the screen.

To change the target curve, up to ten parametric IIR filter is available. The IIR filter prototype is just a drawing aid, since the target magnitude is included in the compensating FIR filter generation.

The parametric filter can be set by the controls at the bottom of the screen or by dragging the cursors in the plotting area.

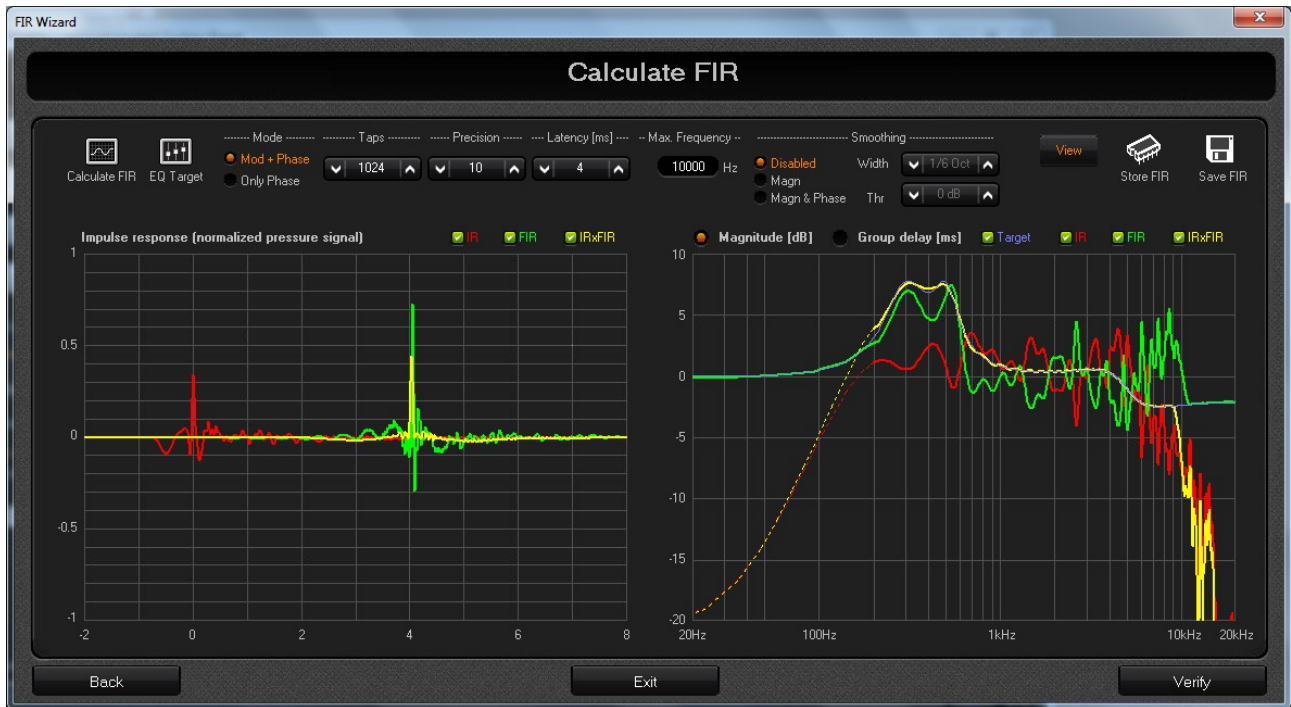


The EQ target can be bypassed (or enabled) by clicking the “EQ Bypass” checkbox at the top center of the screen. Clicking the “Flat EQ Target” button, the gain of each parametric filter is set to 0dB to flatten the target curve.

Enabling the target curve smoothing allows to reduce unwanted ripples in the magnitude response when low frequency filters are set with high gain and/or high Q.

The target curve can be also saved in a file to be easily recalled from a different Wizard project.

To apply the target curve to the compensating FIR filter the “EQ Target” window should be closed and the “OK” button must be clicked in the confirmation popup.



In “Phase Only” mode the target curve can't be set.

Once the FIR filter is computed with the user defined parameters, the corresponding coefficients can be saved in a file (and recalled from outside the Wizard) or stored in the DSP to start listening tests.

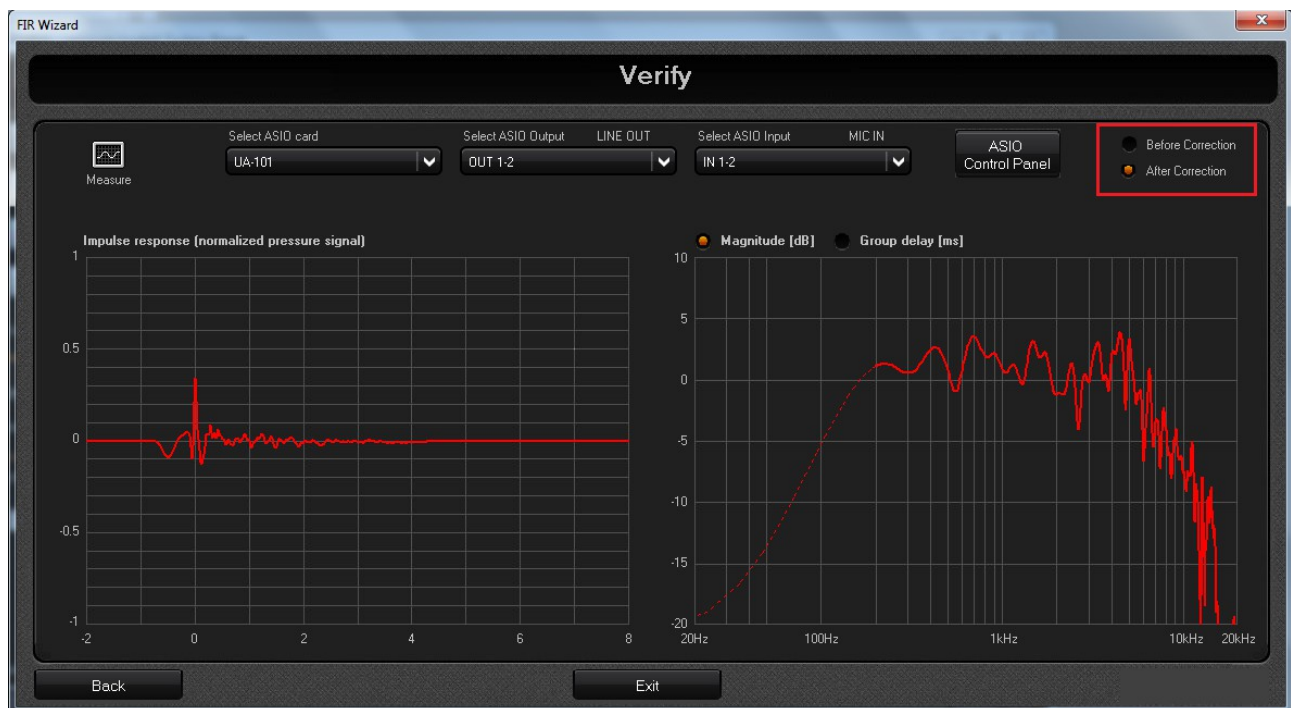
To save the filter file, click the “Save FIR” button at the top right of the “Calculate FIR” page: a file browse window appears allowing to select a folder in the PC and to enter a file name.

To store the FIR filter in the device, click the “Store FIR” button at the top right of the screen.

Verify

Once the FIR filter is stored in the DSP, the user has the possibility to verify the equalization result by measuring the new loudspeaker IR with the FIR filtering enabled. To do this, click the “Verify” button at the bottom right of the “Calculate FIR” page and the “Verify” page is shown.

Clicking the “Verify” button when the computed FIR filter is not yet stored in the DSP, even allows to proceed with the verify process, but care should be taken, since the IR measurement will be performed using the last stored FIR filter, that may not have been generated with the actual loudspeaker response, resulting in a misleading behaviour.



To start the verify process, click the “Measure” button at the top left of the screen and wait in silence. After the sine sweep completion, the new IR is computed and shown in the time/frequency domain plots. It is possible to compare the original and the compensated IR by acting on the “Before / After Correction” option button in the top right of the screen.

Now, it's the time to listen the sound improvement of the FIR filtering. So, close the Wizard window or click the “Exit” button and a message is shown on the screen asking to save the Wizard project. After saving (or skipping) the project file, the Wizard window is closed and you can enable the FIR filtering on the DSP to listen the IR equalization result.

NOTE B

For importing External FIR Calculated coefficients into the MARANI **PDA500PF**, the file format has to be the following:

FIR FILE FORMAT

- Standard floating point txt file
- One coefficient for each row
- Decimal separator: "."
- Max coefficient value: 4.0
- Max filter length (i.e. max file rows): Up to 512 taps (shorter filters are allowed)

Just to be make sure, here below an example of "FIRFormat.txt" which can be imported by the **PDA500PF**:

FIRFormat.txt

```
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-2.1021137868951E-06  
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8.92767141156887E-06  
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```


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