User Manual

LPP-480F

USER MANUAL
Remote Control sw PC
LPP-480F

Hardware and Software requirement for installing
– O.S: XP (32-bit) and higher   (also 64-bit O.S)
– RAM: min. 1GB
– ROM: min. 10MB
– NET Framework 4.0 (installed before to installing the LPP-480F setup)
– USB Driver: "STM32_VCP_Driver_V1.3.1.exe"

Preliminary
– The remote control sw Pc can connect with the device(s) via USB, RS485, TCP/IP so the user must set the connection mode according with the device's setting.

– Each device can be connected by up to 4 connection (PC) at the same time. Is not possible to connect the same device at the same time via RS485 and TCP/IP. Is possible to use at the same time the USB and TCP/IP   or USB and RS485 connection (from different PC).

– When two remote control sw PC are connected to the same device then each modification done by one sw PC GUI, shall be reported also on the other one sw PC GUI automatically

– Each device have inside installed an "Announce" function in order to inform the sw PC when it is
connected via TCP/IP. So at the first start-up of the sw PC connected via TCP/IP with the device, the firewall of the O.S try to block the program (see the picture below)

Click the button "Unblock" if the User want that the sw Pc recognize automatically all devices connected via LAN.

– the default IP address of the device is 192.168.0.100, so if the User want to connect it with the sw PC via LAN is recommended to configure the PC with a static address on the range 192.168.0.x

– The parameters used on the device are divided in two group, system and program parameter.
System Parameter:
Device Name, Input Source Type, Auto switch source enable, Input/Output Link, Input/Output Keep linked channels level offset, Noise Generator Enable/Type/Level, ADC Headroom, Mode.
Admin/User Password, Admin Lock Mode, Lock GUI with and without password, Program recall option.
This parameters are used to configure the device and not will be change when the User change a

program. Many of this parameters are set under the "Settings" page and "Routing" page.

Program Parameter:
are all parameter used to configure the signal process (filters, noise gate, gains, compressor, limiter, delay,....)   this parameters can be load/saved on PC and Recall/Store to Device.
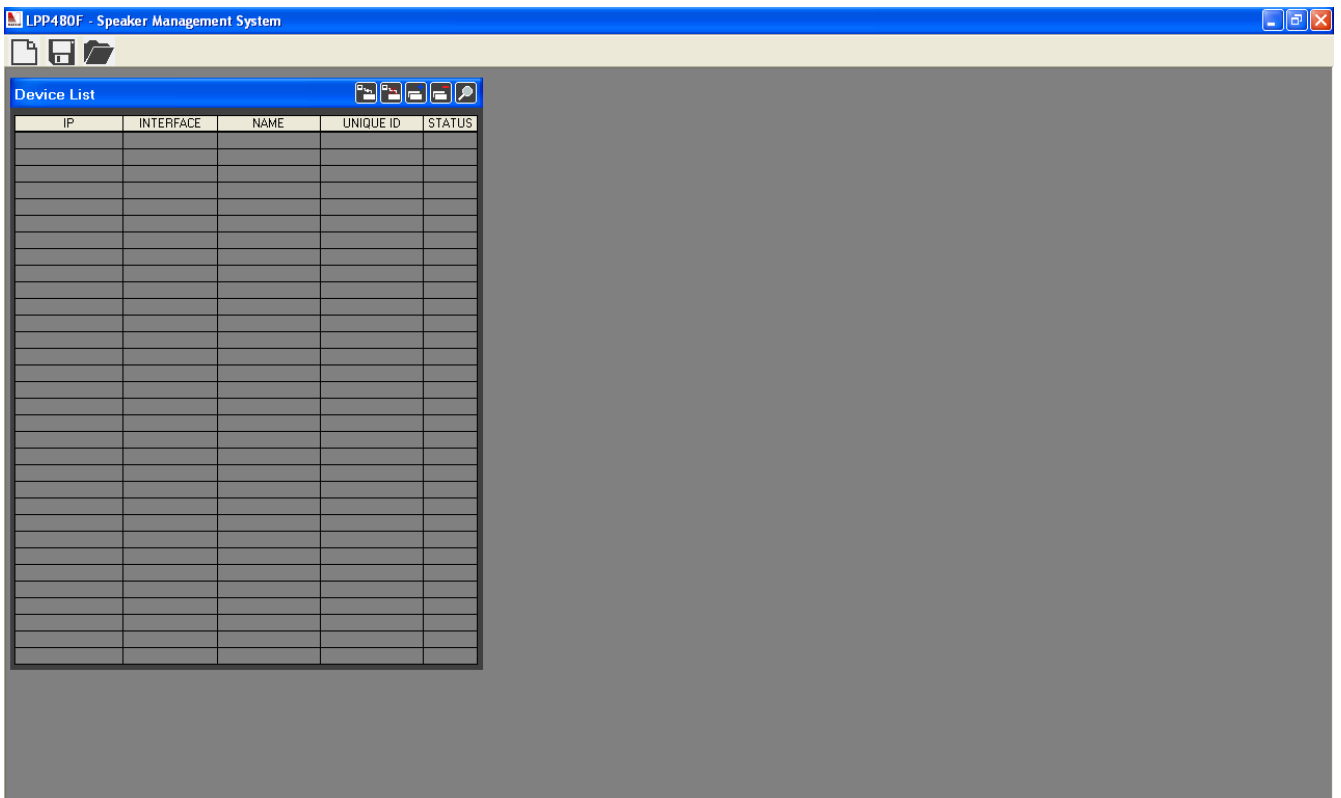
– Project,
the sw PC start every time with last project configuration, on the project file there are the information of each device (system parameter and current program parameter) added to a "Device-List" (see later)
the User can Save the current project to PC or load a saved project from PC.

– The User can also work in off-line Mode and if he want,
he can upload the setting on the device when try to connect (on-Line status) the sw PC with the device or he can download the setting from the device to the GUI when try to connect (on-Line status) the sw PC with the device.

USER GUIDE

after run the remote control sw PC, the first window shown is the splash-screen

after some seconds this window will be closed automatically and start the "Main" window:

where is possible to see the "Device-List" window and the three icon to manage the project set by User "New", "Save" and "Load" project. To close the App the User must click on the red button "X"

Each time that the User close the App, the system save automatically last configuration of the project in order to recall it at the next start-up of the App.

Device-List window
on the Device-List window are shown all devices found automatically by sw PC (if the device are connect via TCP/IP) and the devices added manually by the User.

If a device is connect via TCP/IP then the app can recognize automatically (if the permit on the **Firewall** is enabled) the device and ask to the USER to connect it or not and insert in the List the device found if it is not already inserted.
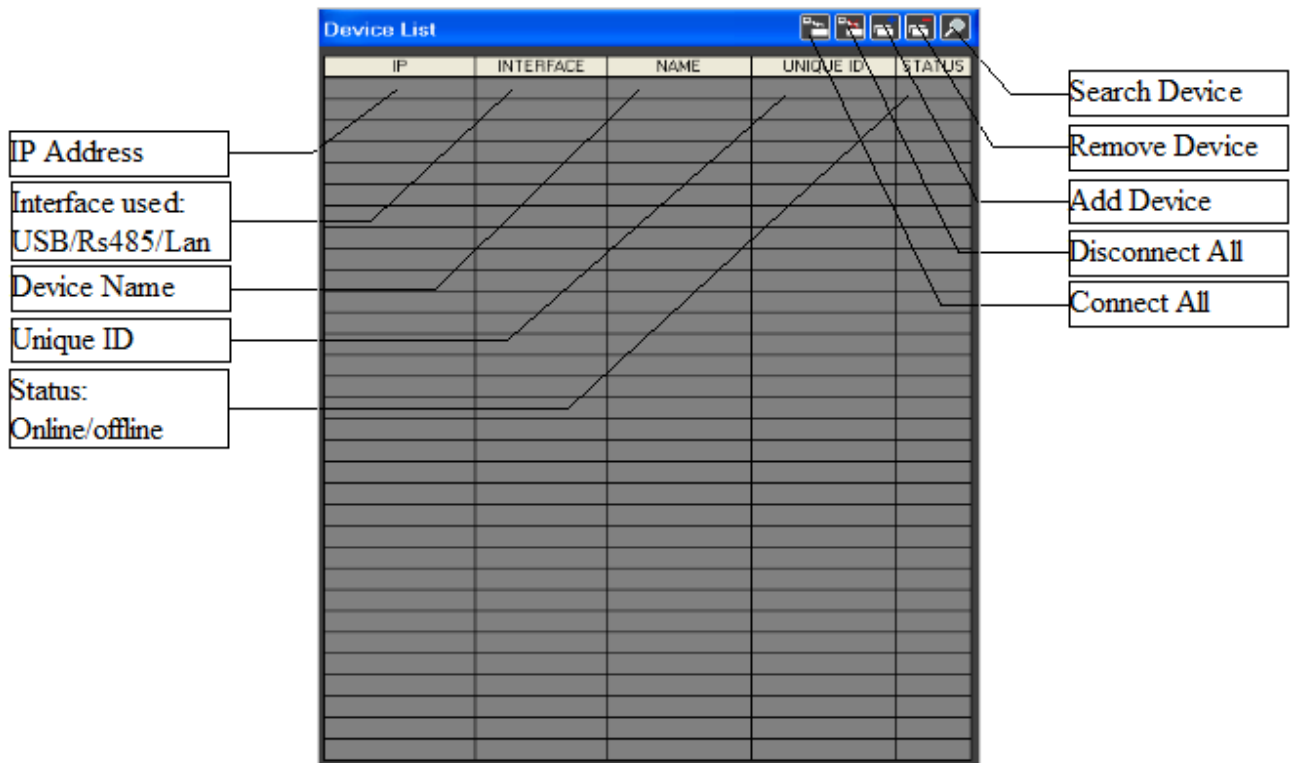
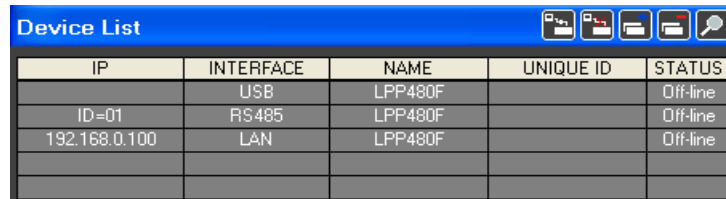**Add Device:** with the button "Add Device" is possible to insert manually a device
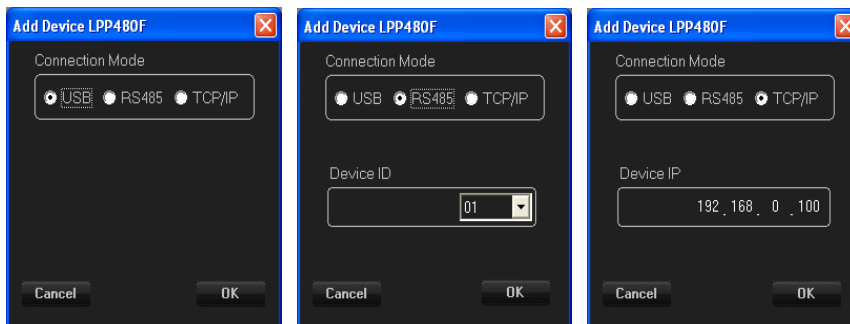
The Connection Mode (Interface in the Device List) are USB, RS485, TCP/IP.
If the User select:

- USB Mode, only one device can be selected in USB mode. If on the Device-List is already present a device set in USB mode then the system inform the User after pressed the button OK, that is not possible to insert other device in USB Mode
- RS485 Mode, must set an ID and this must be the same used on the device that the user want to connect. The ID range is from 1 to 32 (max 32 device can be added). If on the Device-List is already present a device set in RS485 mode with same ID then the system inform the User after pressed the button OK, that is not possible to insert other device with the same ID
- TCP/IP (LAN) Mode, must set an IP address and this must be the same used on the device that the user want to connect. The IP range is from 1.0.0.1 to 223.255.255.254. If on the Device-List is already present a device set in LAN mode with same IP then the system inform the User after pressed the button OK, that is not possible to insert other device with the same IP

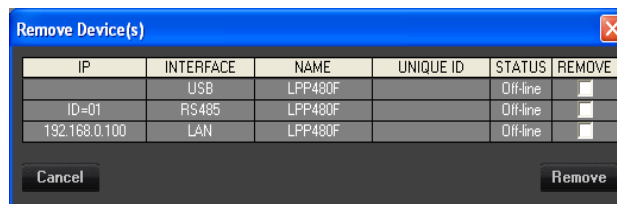After pressed the button OK if there are not duplicate device the system insert the device selected on the "Device-List"

the Unique ID field will be refilled when the sw is connected with the device (this info will be read from the device)
The Unique ID is used by sw PC to identify the device on the network.

Remove Device: with the button "Remove Device" is possible to remove one or several device from the Device-List.
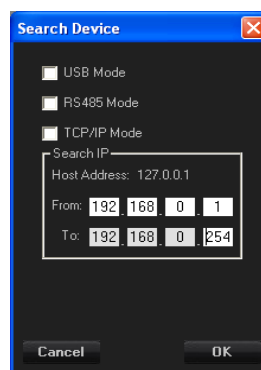

if a device on the Device-List is On-line or in editing then it will be not shown on the Remove-List.
To remove a device the User must set the check-box present on the column REMOVE and the row of the device that the User want remove.
When a device is removed from the User then the device will be not recognize automatically, he must insert it again manually if wanted. If the sw Pc will be closed and re-open then the sw PC can recognize the device removed.
After select the check-box press the button "Remove" to remove the device(s) in the "Device List".

Search Device: with the button "Search Device" is possible to check if there are devices connect to PC and still not inserted on the Device-List.

The User can select what to search on the network, the device connect in USB Mode, and or Device connect in RS485 Mode and or Device connect in TCP/IP mode.
If the USB check-box is selected then the system search only one device connect in USB Mode. The system check if there is a USB port connected to device (check all COM on the range from 1 to 64)
the system work with one COM for time, so if there is a device already On-line and connect via **USB or RS485** then the system can not found new device connect via USB

If the RS485 check-box is selected then the system search all device connect in RS485 mode with ID from 1 to 32 (is **important** that each device connect on the network via RS485 have an ID different from the other). The system work with one COM for time, so if there is a device already On-line and connect via **USB** then the system can not found new device connect via RS485

If the TCP/IP check-box is selected then the system search all device connect via LAN mode with IP on the range selected by User, the system can search devices from 1 to max 255 for time, in that case the first 3 byte of the IP address of the "From" and "To" are the same.
If the Host Address is not found (PC have the LAN not available) then the sw PC show the IP address :127.0.0.1 and in that case is not possible search device in TCP/IP mode.

After selected the search mode check-box press the button "Ok" and the system start to interrogate the network and when finish show the new device found on the Device-List or show a pop-up message with "No new device(s) found".

Connect/Disconnect ALL: with the button "Connect/Disconnect All"
is possible to connect (set in On-line mode) all device present on the Device-List in Off-line Mode
is possible to disconnect (set in Off-line mode) all device present on the Device-List in On-line Mode

Single Connection
if the device 's STATUS is On-line then if the User click on the colum "STATUS" of the device's row   then the device switch in Off-line mode after a confirmation, vice versa if the device is Off-line then switch in On-line mode.

To switch from Off-line to On-line mode the system ask to User if:
a. the GUI (Graphic User Interface) must be update with the data present on the device or
b. the Device must be update with the data already edit on the GUI (data of the device selected and already present on the project).
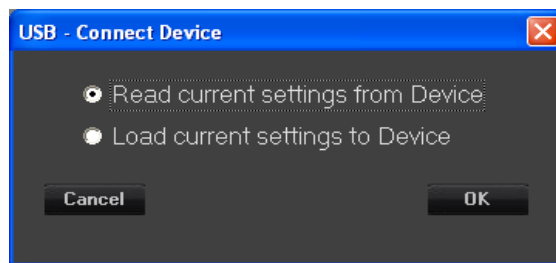
This window will be show also when the User use the button "Connect All"

After press the Button "OK" the system run the selected option set by User.
If the device selected is on the network then on the Device-List will be show the device status "On-Line".
EDIT DEVICE

The user can edit the configuration of the device in Off-line and On-line mode. If the Device is in On-line mode then the changed parameters will be update on the device in real-time mode.

To Edit a device the user must a double-click with mouse   in the device-List on the column IP (or Interface, or Name or Unique ID) of the device's row.
After the double-click the system show the following Edit window
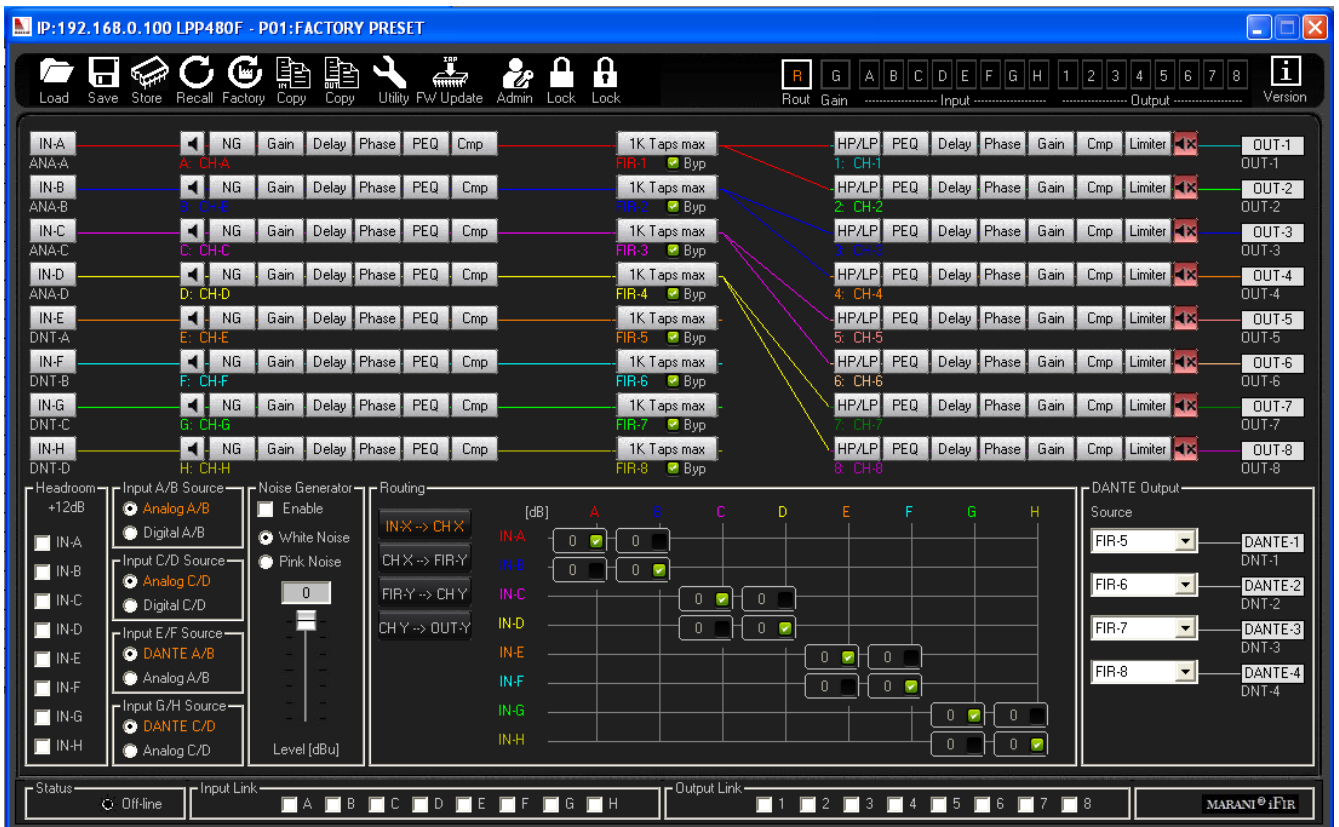
- – on the Title-bar of the window are present the device IP address (if the device is connect via TCP/IP) or device ID (if the device is connect via RS485), the device name and the current program loaded.

- – on the above tool-bar of the window there are:

1. Load Program saved to PC
2. Save Program to PC
3. Store Program to Device
4. Recall Program from Device
5. Reload the current Program with the Factory parameter
6. Copy one input channel to another input channel
7. Copy one output channel to another output channel
8. Utility
9. Firmware Update via USB
10. Enter a password to log-in as Admin
11. Lock interface without password
12. Lock inteface using a User password (default = "111111")
13. Pages selector, to select a page between Routing, Gains and Meter, input channel (A,...,H), output channel(1,...,8)
14. Read the current version of the Fw

see later how working the button 1,..,14 present on this tool-bar

– on the below tool-bar of the window there are:

Status	Input Link								Output Link								MARANI® iFIR
○ Off-line	☐ A	☐ B	☐ C	☐ D	☐ E	☐ F	☐ G	☐ H	☐ 1	☐ 2	☐ 3	☐ 4	☐ 5	☐ 6	☐ 7	☐ 8	

      15   16   17

15. Status of the connection (off-line / On-line)
16. Link input channel (when the check-box is enabled on the channel of the control modified then the value will be copied to the other channel with the link enabled)
17. Link output channel (same at the Link input channel)

see later how working the button 15,..,17 present on the tool-bar

– on the middle of the window there are the edit pages (depending from the button selected on the "Page Selector")

Routing Page:

on the routing page is possible select:

–   Headroom: when the check-box is set, on the relative input will be add +12dB

–   Input source: the User can change the input source in pairs,
    a. source A/B can be set as ANALOG A/B or DIGIATAL A/B
    b. source C/D can be set as ANALOG C/D or DIGIATAL C/D
    c. source E/F can be set as DANTE A/B or the selection on the Source A/B
    d. source G/H can be set as DANTE C/D or the selection on the Source C/D
    On the utility menu "Src Settings" (open it using the button Utility) is possible to set the automatic source priority "**Autoswitch**" on channel E/F and G/H.
    When the "Enable Autoswitch on Channel E/F" is set then, independently of the set on the Source E/F, if the signal is present on the DANTE input A/B, the source E/F will be set automatically to DANTE A/B.
    When the "Enable Autoswitch on Channel G/H" is set then, independently of the set on the Source G/H, if the signal is present on the DANTE input C/D, the source G/H will be set automatically to DANTE C/D

–   Noise Generator: on this frame is possible to set the Noise generator type (White, Pink) and the level of the noise. When the Noise Generator is enabled then in all input there is the Noise signal independently of the set on the Source channel

- Input-Routing (IN-X --> CH.X)

Enable/Disabel routing IN-A to CH.A

Fixed Amount of the IN-A to CH.A

- Ch Input to FIR - Routing (CH.X --> FIR-Y)

Enable/Disabel routing CH.A to FIR-1

Amount of the CH.A to FIR1 routing

- FIR to Ch Output - Routing (FIR-Y --> CH.Y)
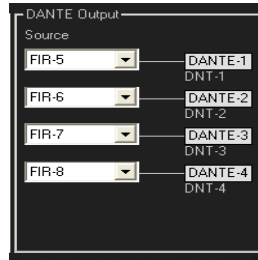

FIR Byp On/Off

Enable/Disabel routing FIR-1 to CH-1

Amount of the FIR-1 to CH-1

- Ch Output to Output - Routing (CH.Y --> OUT-Y)


Enable/Disabel routing CH-1 to OUT-1

Fixed Amount of the CH-1 to OUT-1

‒ DANTE Output Routing

- the output DANTE-1can take the signal from the output of the FIR-1 or FIR-5 block, or from OUT-1
or from OUT-5
- the output DANTE-2 can take the signal from the output of the FIR-2 or FIR-6 block, or from OUT-2   or from OUT-6
- the output DANTE-3 can take the signal from the output of the FIR-3 or FIR-7 block, or from OUT-3   or from OUT-7
- the output DANTE-4 can take the signal from the output of the FIR-4 or FIR-8 block, or from OUT-4   or from OUT-8

‒ FIR BLOCK
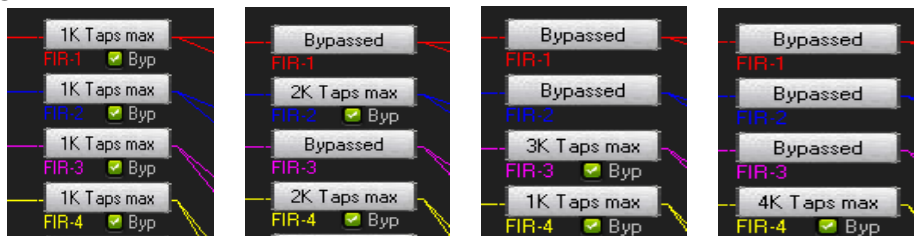
the FIR-1,FIR-2,....FIR-8 can be configured under utility menu "FIR Config." (open it using the button Utility)
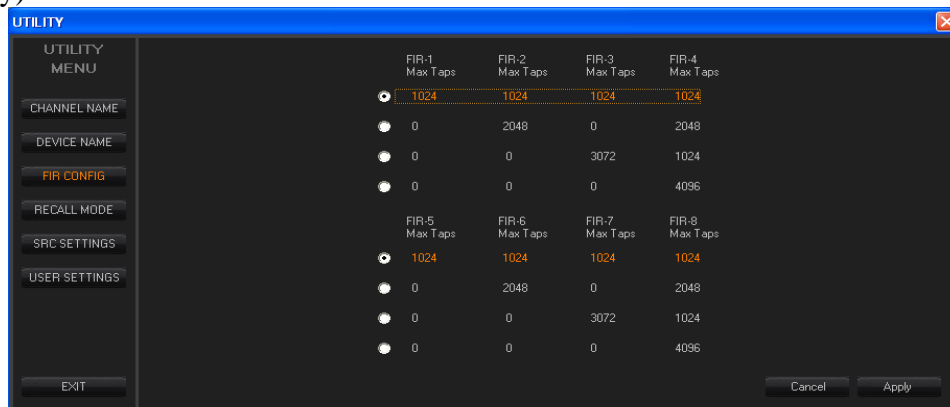
the System have split out the FIR block in two section FIR-1 to FIR-4 and FIR-5 to FIR-8 in order to have more flexibility on the number of the Taps of the FIR.
With the configuration present on the 1st row, the system have 4 FIR with 1024 Taps.
With the configuration present on the 2nd row, the system have one FIR on channel 2 and one FIR on channel 4 with 2048 Taps, the channel 1 and 3 have not FIR
With the configuration present on the 3rd row, the system have one FIR on channel 3 with 3072 Taps and one FIR on channel 4 with 1024 Taps, the channel 1 and 2 have not FIR
With the configuration present on the 4th row, the system have one FIR on channel 4 with 4096 Taps, the channel 1,2 and 3 have not FIR.
This is true also for the section FIR-5 to FIR-8

is a FIR Block is not "Bypassed" when the User, click with the mouse (left button) on the label "xx

Taps max" the system show a following pop-up menu
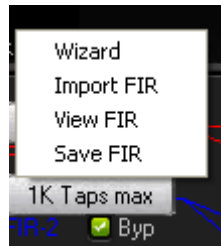
using this menu the User can:
1. use a Wizard in order to use a tool to create a FIR filter for the phase correction (the User need an external Asio audio card to recorder the Output signal using a microphone. See the appendix "Wizard" to understand how it is work)
2. Import an exernal FIR (see the appendix "Import FIR Filter")
3. View the current FIR loaded on the device
4. Save the current FIR loaded on the device to a file txt

To see the name of the FIR loaded the User must put the cursor of the mouse over the button and in this case the system show a pop-up with the name.

GAINS Page:

on the gains page is possible change the value of the input and output Gain (from -18dB to +12dB), set the Polarity, the Mute and see the input (before Gain) and output (after Limiter) Meter.
When the Mute button is red then the relative channel is in muted.
When the Polarity check-box is set then the polarity of the relative channel is inverted of 180°

On this page, when more then one link on input and/or on output are set,

then the User can also enable the option "Keep linked channels level offset"
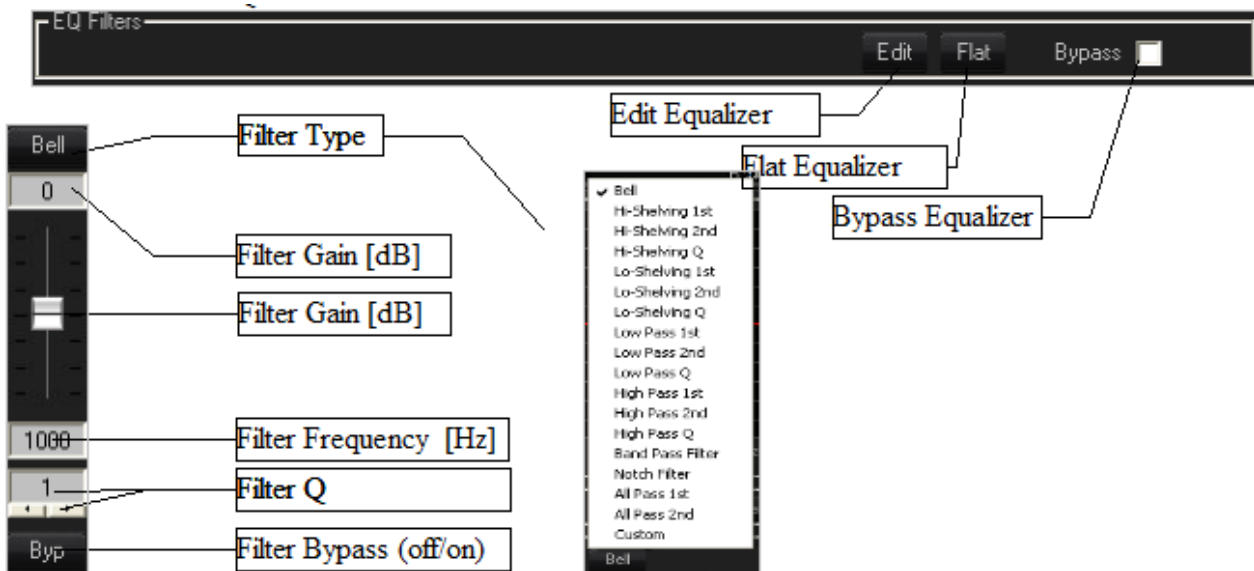

this mean that when move a gain on a channel linked the other channel linked change the value with the same quantity but maintaining the offset.
Examples:
– Input Link A and B are set.
– Keep linked channels level offset (on input section) is enabled
– Level of the CH.A is 0dB and the Level of the CH.B is +3dB
  if the User set the Level on CH.A to +5dB then the Level of the CH.B change to +8dB
  on this example the max level of the CH.A is +9dB because when the CH.A is +9dB the Level of the CH.B is +12dB (Max Level) and to maintain the offset of 3dB the CH.A max value will be fixed to +9dB.

INPUT A,B,...,H Page:

on each input channel is possible change the:
- Gain
- Mute
- Polarity
- Delay
- Noise Gate
- RMS Compressor
- 13 PEQ Filter
- Graphic filter (Magnitude or Phase)
- Meter of the Channel and the Compressor activity (Gain reduction)

On all control where is a text-box [1.5], the User can change the value of the control in the following mode:
- select the text-box
- input the value using the keyboard
- press Enter to validate the new value

On all control where is a Fader, the User can change the value of the control in the following mode:

select the fader with mouse (when the fader is selected the color change [ ] => [ ])
a1. press and hold the button left of the mouse and drag it to change the value
a2. after selected use the key "up"/"down" of the keyboard to increase/decrease the value
with a double click (button left) of the mouse the value will be changed to 0dB (this is true only for the gains/eqs fader control)

On all control where is a Chech-Box, when it is set [Bypass ✓] then the parameter is actived. Examples in the picture the bypass is enabled
Parametric Filter EQ

– when the equalizier is bypassed the user can working on the single filter but the graphic of the filter is shown as flat

– when the User press on the button "Flat", all filter will be set to 0dB (but if the filter type of the filter is Low Pass, High Pass, Notch, All Pass, then the single filter will be bypassed)

– to change the value of the Q, use the button Left/Right below the text-box or input the value on the text box and press "Enter". The Q value is depends from the filter Type.

– to change the value of the Frequency, input the value on the text box and press "Enter". The value must be in Hz without "," or "."

– to change the value of the Gain, the User must after select the fader with mouse   (when the fader is selected the color change   =>   ) press and hold the button left of the mouse and drag it   to up or down or after selected use the key "up"/"down" of the keyboard to increase/decrease the value or
input the value on text-box above the fader and press enter. (the range is from -15dB to +15dB step 0.5dB)

– with a double click (button left) of the mouse on the fader the value of the gain will be changed to 0dB

– to change the filter Type, the User must click with the mouse (left button) on the filter type button.
After press, the system show a pop-up menu with a list of the filter type.
If the Type is Hi-Shelving/Lo-Shelving 1st, 2nd then the Q is fixed so the control is disabled
If the Type is High/Low Pass, Notch filter then the Gain is fixed so the control is disabled
If the Type is All Pass 1st then the Gain,Q are fixed so the controls are disabled
If the Type is All Pass 2st then the Gain is fixed so the control is disabled
If the Type is Custom (IIR Filter imported by User) the system loads the last external IIR filter stored on the device and shows a frame with an "Edit" button (see the picture above). To working with Custom filter, please see the appendix "Custom IIR Filter"

Graphic's Parameter

- **Cursor Button**

  The button "Cursor" (toggle button) is used to draw the filter on the graphic using the mouse. After pressed, the button change state (the color of the label change from Gray-Orange and vice versa) when active (orange) the system show on the graphic some label (if the Equalizer is not in bypass) with the number of the filter (for input channel from 1 to 13, for output channel from 1   to 7)
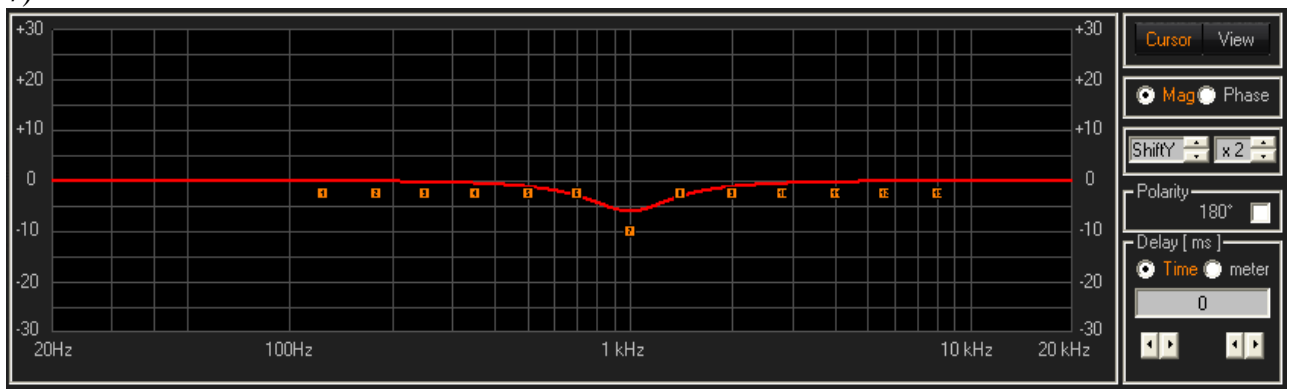
  The User can press and hold the left button of the mouse over the orange label and drag the it to modify the frequency and the Gain of the Filter selected (dipending of the filter Type, i.e for HP filter is not possible change the gain) after release the button then the filter will be loaded on the device if it is connected. To change the Q of the filter the User must use the control as described above (using the text-box or the button left/right).

- **Option Mag/Phase**

  Use the options "Mag" or "Phase" to see on the graphic the Magnitude or the Phase of the filters

- **Shift and Zoom control**

  the Shift control is used to shift the axis Y (dB) up or down,
  the Zoom control is used to increase or decrease the resolution on the axis Y (dB)
  with Zoom = x1 then the step from a label to another is 5dB
  with Zoom = x4 then the step from a label to another is 20dB
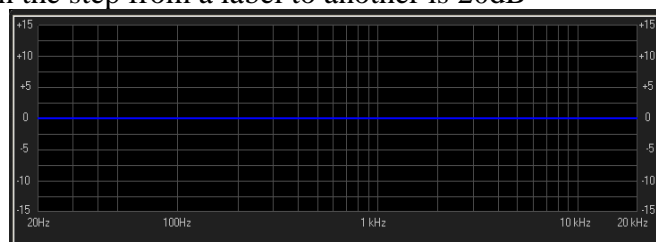
  x1
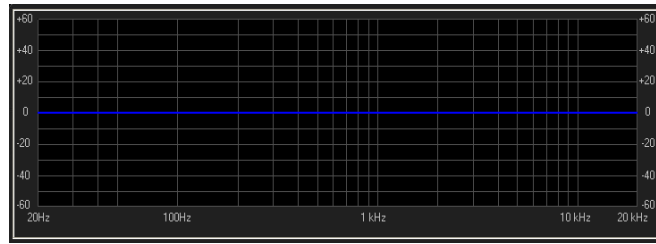
x4

– **View Button**
when the user press the button, the system open a new window where is possible only view all plot of all input and output channel on the graphic
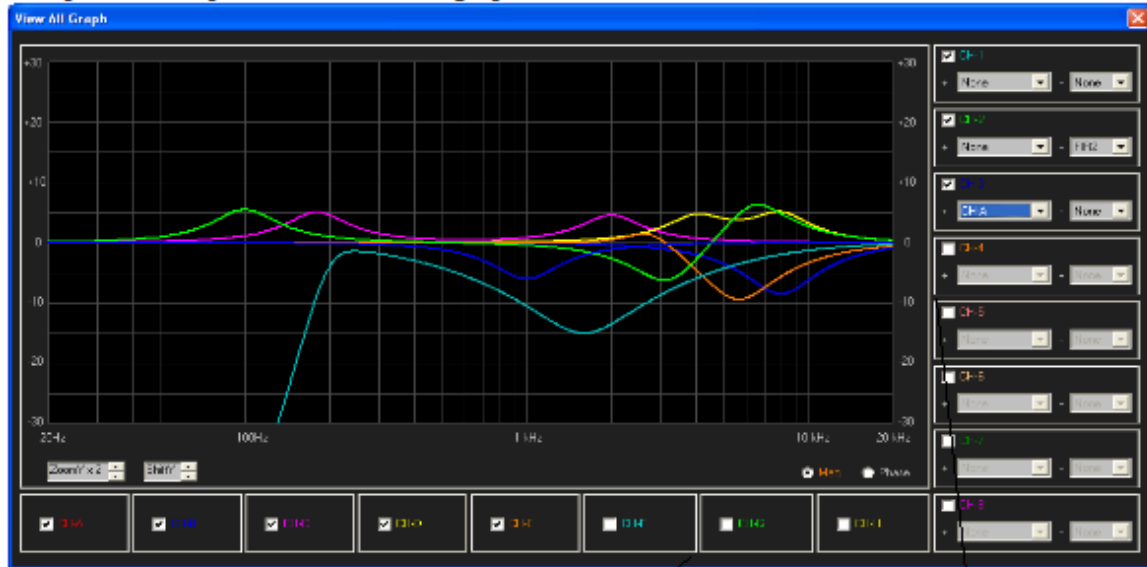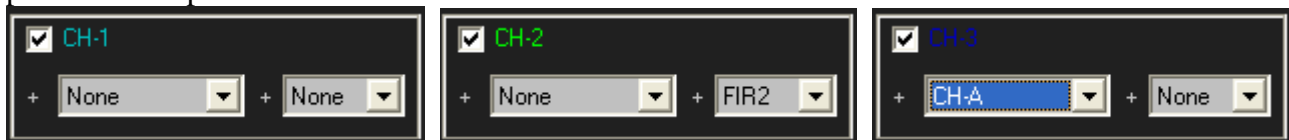

Input channel, when the relative check-box is enabled then the User can see on the graphic the plot of the input channel enabled
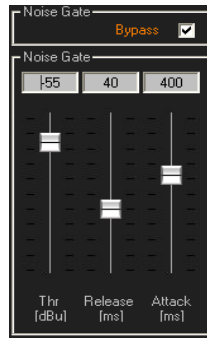

Output channel, when the relative check-box is enabled then the User can see on the graphic the plot of the output channel enabled.
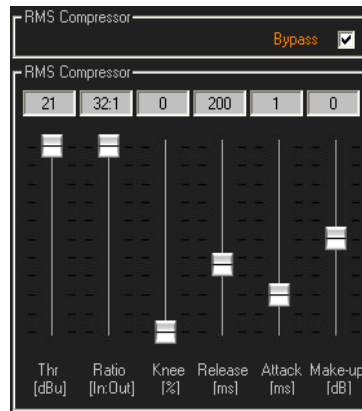

The User can see
1. the plot of the only Output.Y (i.e CH.1)
2. the plot of the Output.Y + FIR-Y (i.e CH.2)
3. the plot of the Output.Y + Input CH.X (i.e CH.3)
4. the plot of the Output.Y + Input CH.X   + FIR-Y

Noise Gate

In the Noise Gate control is possible edit the Bypass, Threshold, Release-Time, Attack-Time.
To change the value use the same management used for EQ

RMS Compressor

In the RMS Compressor control is possible edit the Bypass, Threshold, Ratio, Knee, Release-Time, Attack-Time, Make-up (same on the output channel). To change the value use the same management used for EQ

DELAY

Delay Adjust, step 1 ms

Delay Fine, step 10.4 us
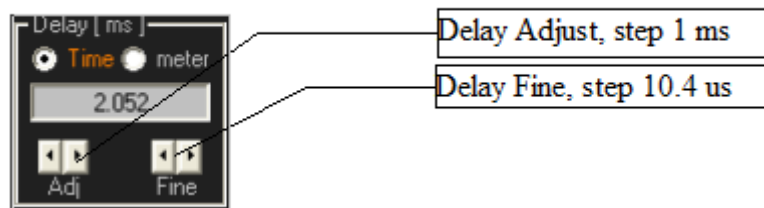
on the Delay control is possible select the unit of the delay (time [ms], distance [m]), change the value using the text-box or the buttons left/right below the text-box

Vu-Meter Channel
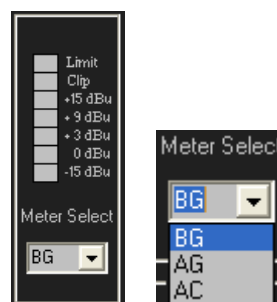
the user can select what see on the Meter: BG=Before Gain, AG=After Gain, AC=After Compressor
(on output page there are: AG=AfterGain, BL=Before Limiter, AL=After Limiter)

Compressor Activity
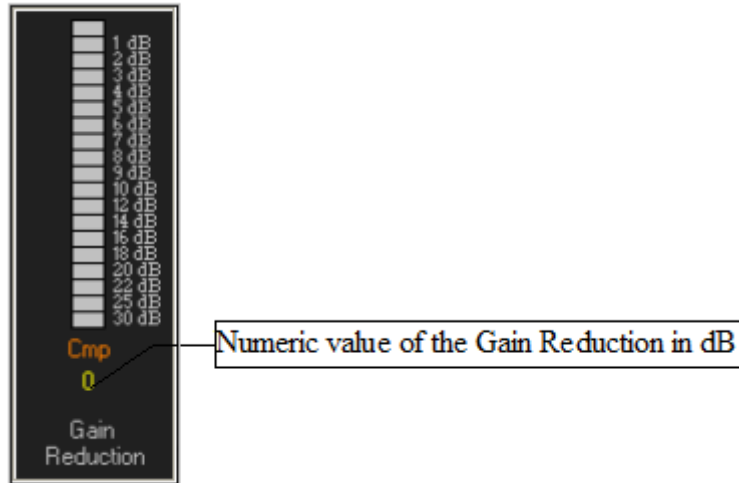
on this Meter the User can see the Gain Reduction when the compressor is active

MUTE/POLARITY/GAIN
they are the same control present on the GAINS page

OUTPUT 1,2,...,8 Page:
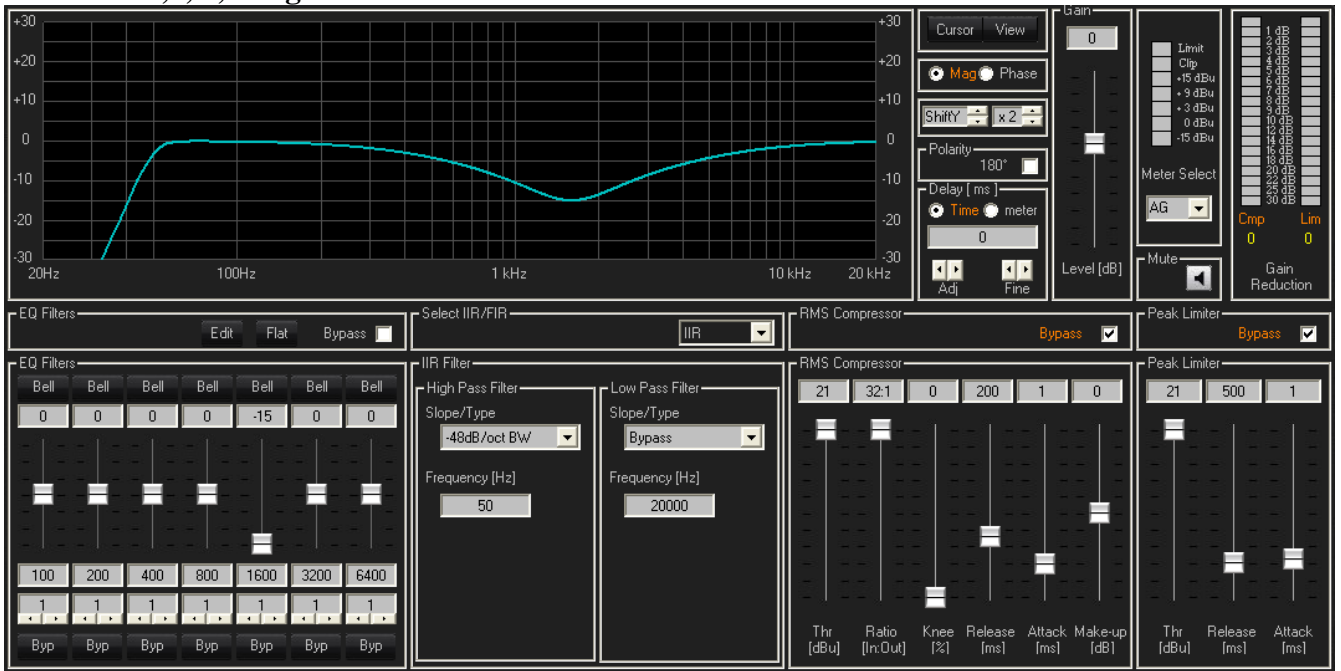
on each output channel is possible change the:

– Gain
– Mute
– Polarity
– Delay
– Peak Limiter
– RMS Compressor
– High/Low Pass IIR filter or FIR Filter (dependent of the selector IIR/FIR)
– 7 PEQ Filter
– Graphic filter (Magnitude or Phase)
– Meter of the Channel and the Compressor/Limiter activity (Gain reduction)

About the PEQ Filter, RMS Compressor, Delay, Polarity, Mute, Gain, Graphic filter, Meter and Gain Reduction please see the section "**INPUT A,B,...,H Page"** (above)

Peak Limiter
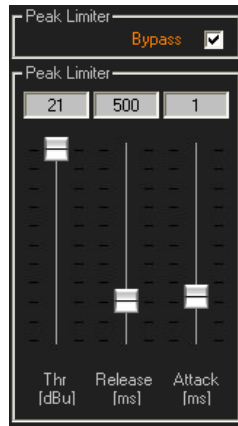
on the Peak Limiter control is possible edit the Bypass, Threshold, Release-Time, Attack-Time. To change the value use the same management used for EQ

Selector IIR/FIR

the device can work with the standard High and Low Pass IIR filter or use a FIR filter instead of the IIR Hp/Lp Filter.

The selection is possible using this control

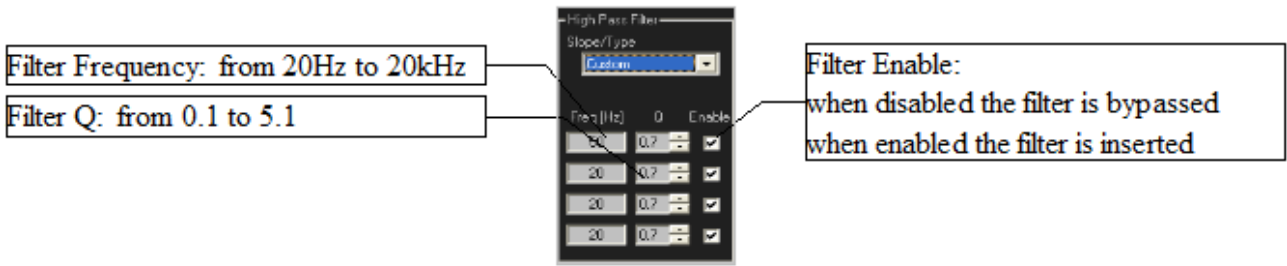– if the **IIR** is selected then, on the interface the system show the following control
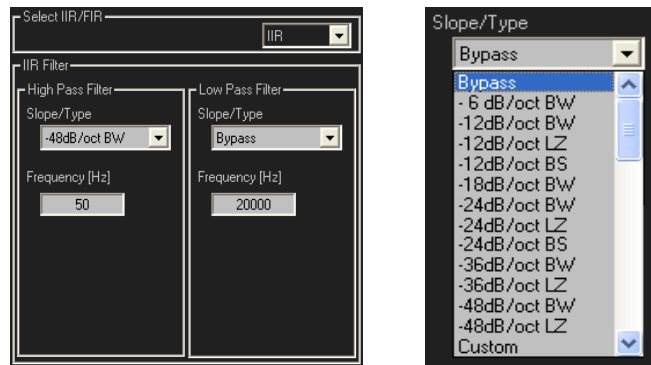
where is possible to see the standard Hp/Lp IIR filter, the frequency value is in the range from 20Hz to 20kHz and the filter Slope/Type are show on the picture above. To open the list of the type the user must click with the mouse over the control (Combo-box) where is present the string of the current filter loaded.
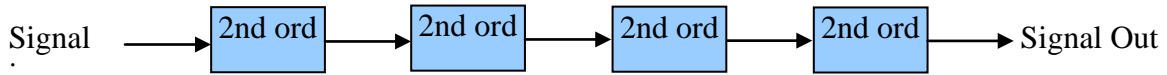
When Bypass is selected then the filter is not loaded (flat).

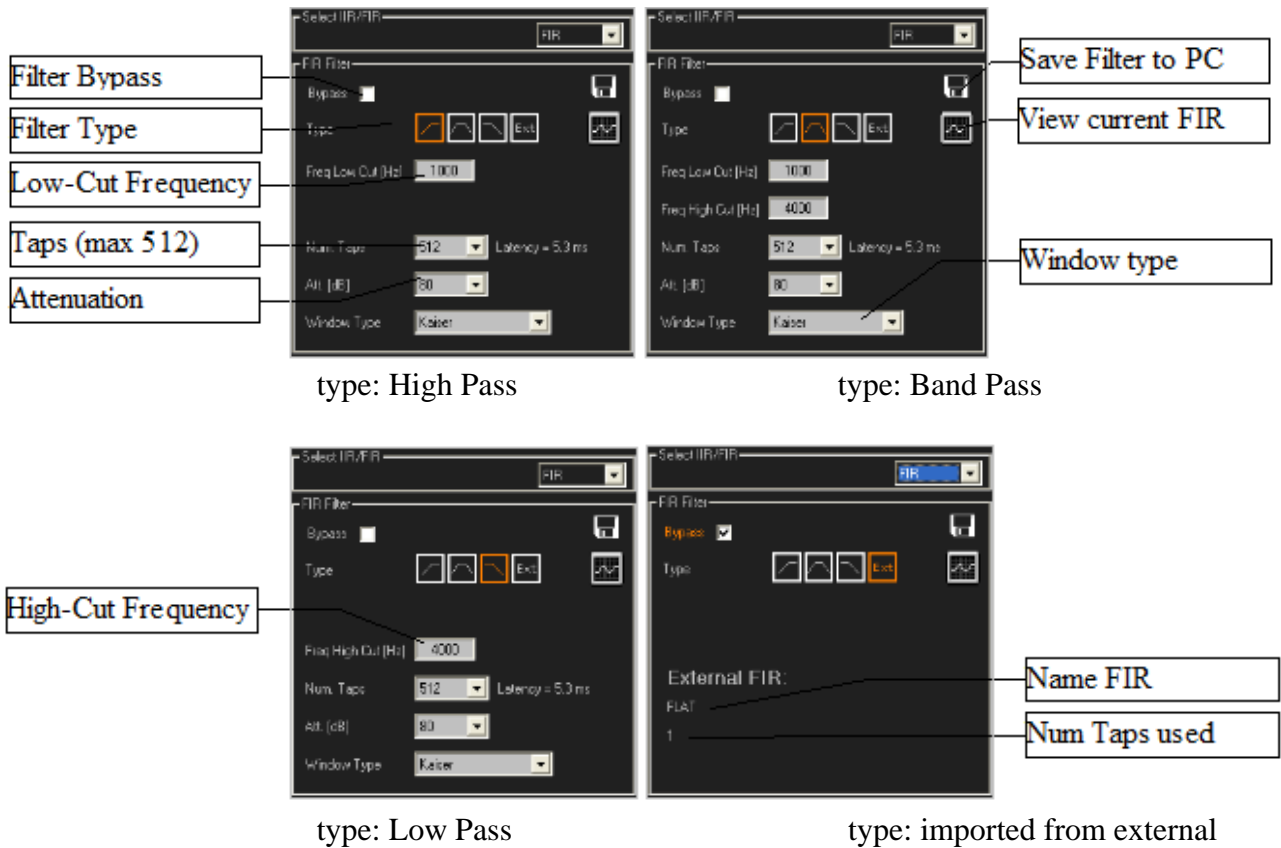BW is Butterworth type, LZ is Linkwitz type, BS is Bessel type.

When the Custom type is selected

Filter Frequency: from 20Hz to 20kHz

Filter Q: from 0.1 to 5.1

Filter Enable:
when disabled the filter is bypassed
when enabled the filter is inserted

then the user can create a filter using 4 standard independent IIR filter Cell (with each IIR cell is possible create a 2nd order filter, is possible change the Q, Frequency and the filter enable) in cascade mode

if the **FIR** is selected then, the system show the following control

Filter Bypass

Filter Type

Low-Cut Frequency

Taps (max 512)

Attenuation

Save Filter to PC

View current FIR

Window type

type: High Pass   type: Band Pass

High-Cut Frequency

Name FIR

Num Taps used

type: Low Pass   type: imported from external

Is possible to bypass the FIR using the relative check-box "Bypass", is possible to save the filter to a file.txt in order to reload it (as External filter) or use it for other applications, is possible to see the Amplitude and Phase of the FIR loaded (the page to "view only" the plot is the same page used to import an external FIR).

The FIR Filter type are HP, LP, BP (are internally calculate and dependent from the option taps, attenuation, window type and cut-off frequency) and External Filter (the user can import a file with the coefficient of the FIR already calculate with other tools, please see the appendix "Import FIR Filter" to understand how it is work)

If the FIR filter loaded is external and the User change type from "Ext" to another type then the External filter will be overwrite with the new internal FIR, to reload the preview Ext FIR the User must reload it (see the appendix "Import FIR Filter")

If the FIR Filter type are HP, LP, BP (the filter will be loaded on the device in real-time if it is connected) then the user must also set other parameter
– nTaps: number of the taps of the Filter   (min 256, max 512, step 32 taps)
– Frequency: only low-cut freq if HP is selcted, only high-cut freq if LP is selected and low/high-cut freq if BP is selcted (20Hz to 20kHz step 1Hz)
– Attenuation: is the value that the user must set to have an attenuation of the signal out the Band selected (the attenuation is dependent of the Window Type, max range -21dB to -120dB)
– Window Type: there are different type of window to calculate the FIR
    (Rect, Kaise, Sinc, Hanning, Hamming, Blackman, Blackman-Harris, Blackman-Nuttal, Nuttal, Kaiser-Bessel, Sine.

TOOL-BAR buttons

LOAD PROGRAM:

To Load a program from PC, the user must press the button .
the system show the standard window to open a file, select a file ".Fhe" and open it.
The file .Fhe are program saved with the "Save Program", inside the file there are all program parameter modified with this GUI.
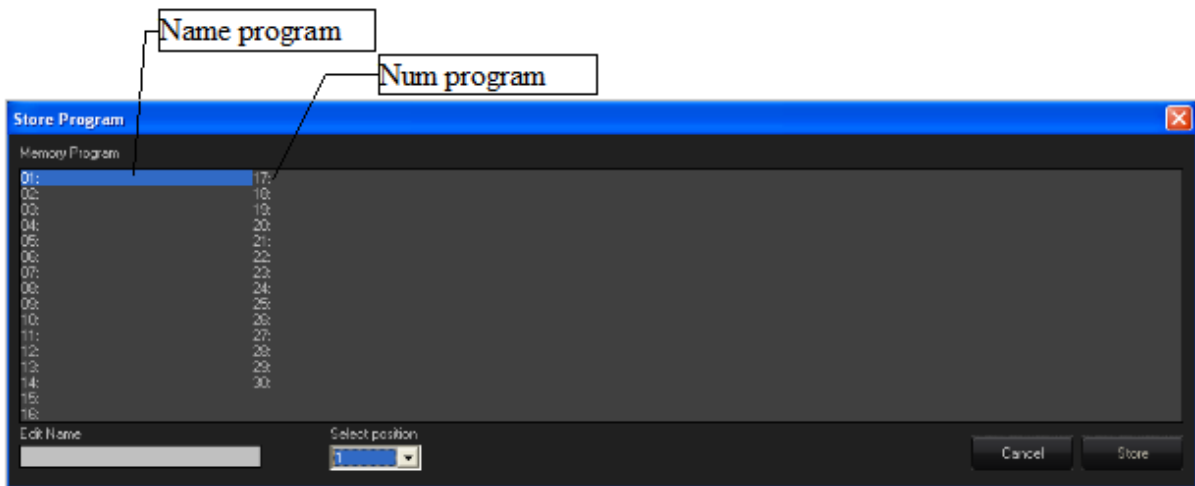
SAVE PROGRAM:

To Save a program to PC, the user must press the button .
the system show the standard window to save a file, input the wanted name of the file ".Fhe" and save it.
The file .Fhe saved have inside all program parameter modified with this GUI. The User can recall this file using the "Load Program"

STORE PROGRAM:

after press the button the system open the following window to store a program to device.
The device can memorize up to 30 program.

Name program

Num program

To store the current parameter set on the "Edit" window, the User must
1. select the position (from1 to 30)
2. input a name for the program on the "Edit Name" text-box (the max length of the name is 16 characters).
3. Press the button "Store"

after this three steps, the current configuration is saved to device and it can be recall from sw PC or recall from the User Interface of the device.

If the System working in ADMIN mode then is possible also see a "Lock Program"check-box
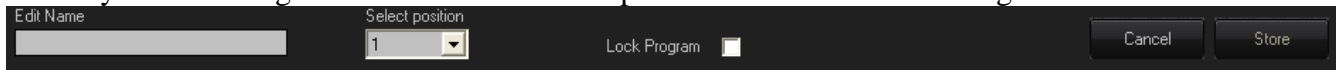
If the Admin set this check box then the program stored will be locked and in that case the User can't overwite this program. If the "Program Store Lock" is enabled then the User can't use the function Store Program (the button is disabled)

RECALL PROGRAM

after press the button Recall the system open the following window to recall a program (stored by PC or device) from the device.

The User can recall up to 30 program.

Name program

Num program
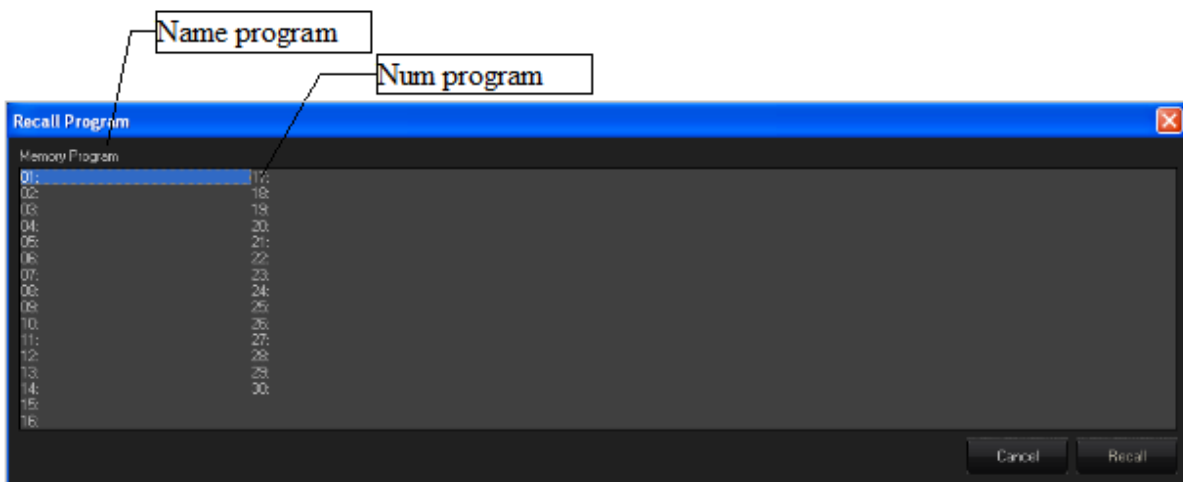
to recall a program the user must select the program and press the button "Recall"

when the system recall the program, if the option "Keep Actual Mutes" present on the Utility Page is set then the system replace the status of the mute present on the program recalled with the status of the current Mute.

RECALL FACTORY

the button is used to replace the current configuration (program parameters) with the factory configuration parameters.

If the "Factory Recall Program Lock" is enabled then the User can't use the Recall factory (the button is disabled)

Copy Input Channel

after press the button the system open the following window to copy a source input channel to a destination input channel
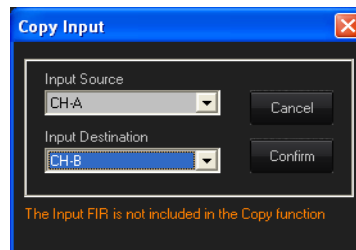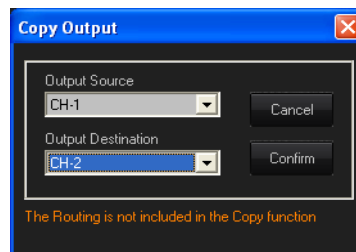

The Input channel source selected will be copied to the Input channel destination selected after pressed the button "Confirm". The Input FIR is not included in the Copy function.

Copy Output Channel

after press the button the system open the following window to copy a source output channel to a destination output channel

The Output channel source selected will be copied to the Output channel destination selected after pressed the button "Confirm". The Routing is not included in the Copy function.
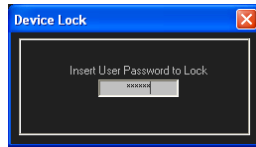
LOCK

To lock the GUI without using a password, the User must press the button . All parameters shall be locked (only the Admin can change the parameter after the log-in). To unlock, the User must press again the same button.

LOCK with Password

To lock the GUI using a password, the User must press the button , the system show a window to insert the "User Password".

If the password inserted is correct then all parameters will be locked (only the Admin can change the parameter after the log-in). To unlock, the User must press again the same button and insert the "User Password".

To insert the password, the user must input (on the text-box) 6 characters and press Enter.

The characters allowed are: " ","0",.....,"9","A",.....,"Z","a",.......,"z"
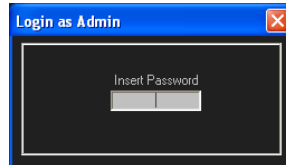
the default User Password is: "111111"

ADIM mode

there are some parameters that can be set only when the system is logged as ADMIN.

To login as Admin the User must press the button , the system show a window where is possible insert the "Admin Password".

To insert the password, the user must input (on the text-box) 6 characters and press Enter.

The characters allowed are: " ","0",.....,"9","A",.....,"Z","a",.......,"z"

the default Adim Password is: "000000"

If the password inserted is correct then the ADMIN is logged, in ADMIN mode the User's Lock are bypassed.

**UTILITY Page**

to open the utility page the User must press the button .

on each page of the Utility menu there is:

– a button .
Press the button "Apply" to save the modification of the page

– a button .
Press the button "Cancel" to remove the modification not applied of the page

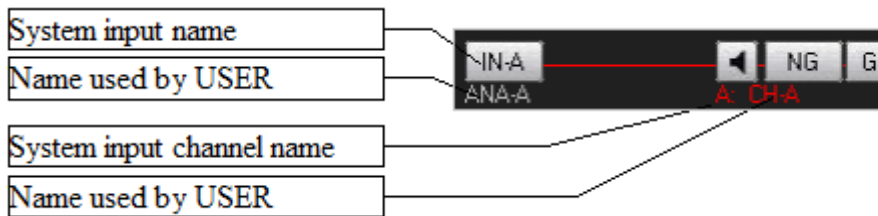**UTILITY MENU**

– CHANNEL NAME
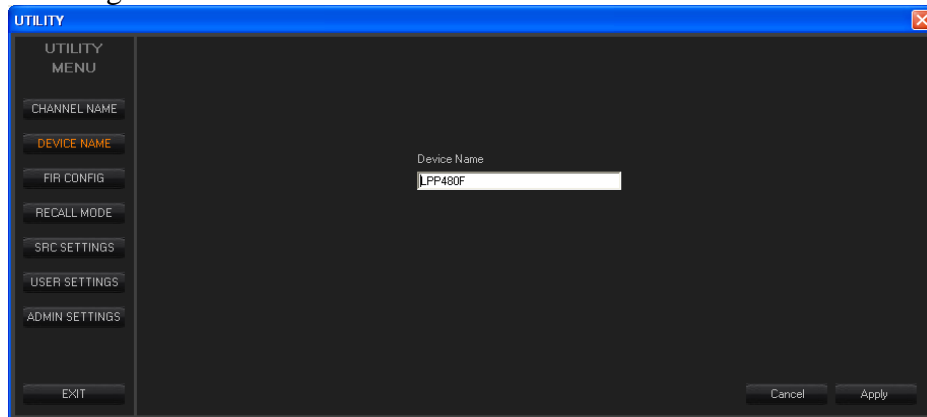the User can associate a name to an input channel/output channel/source/Output.....
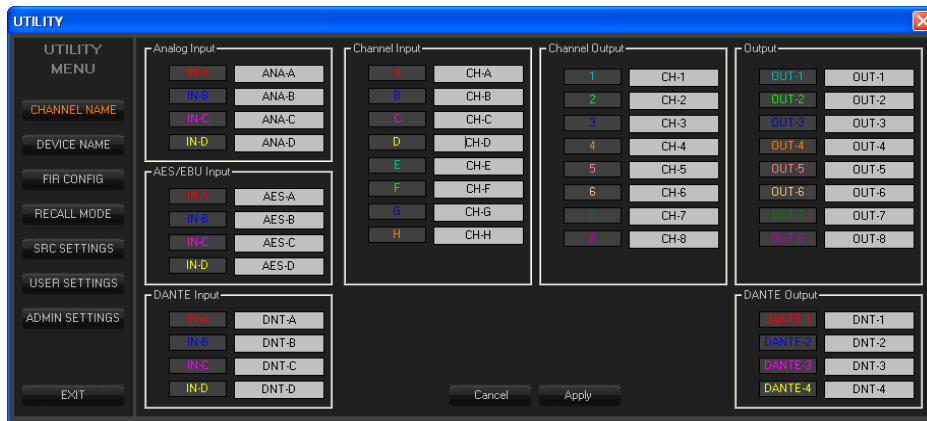
The User can input the wanted name on the relative text-box (max 6 characters).

Examples:

– DEVICE NAME
  the User can change the device name.

Input the wanted name on the relative text-box (max 16 characters).

– FIR CONFIG
  used to configure the max taps on the FIR block.
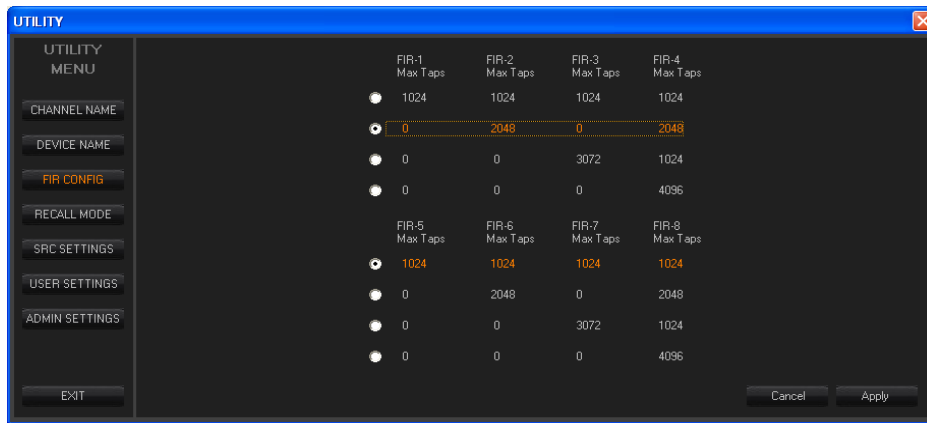
please see the "Routing Page – FIR BLOCK"

- RECALL MODE
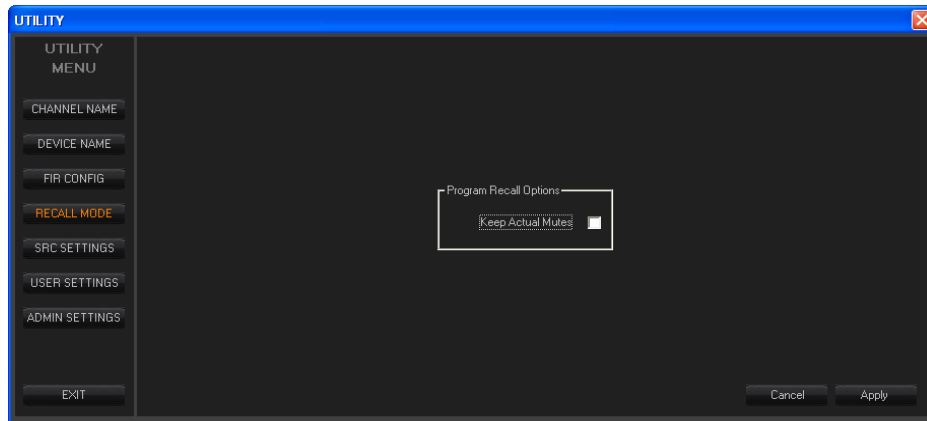
the option "Keep Actual Mutes" is used by device when a program is recalled.
If the option is set, the device replace the status of the mute channels saved to program recalled
with the current status of the mute.

- SRC SETTINGS
this setting will be used by device to understand if it must switch automatically the source input
E/F and G-H when the signal is present on the DANTE Input or not.
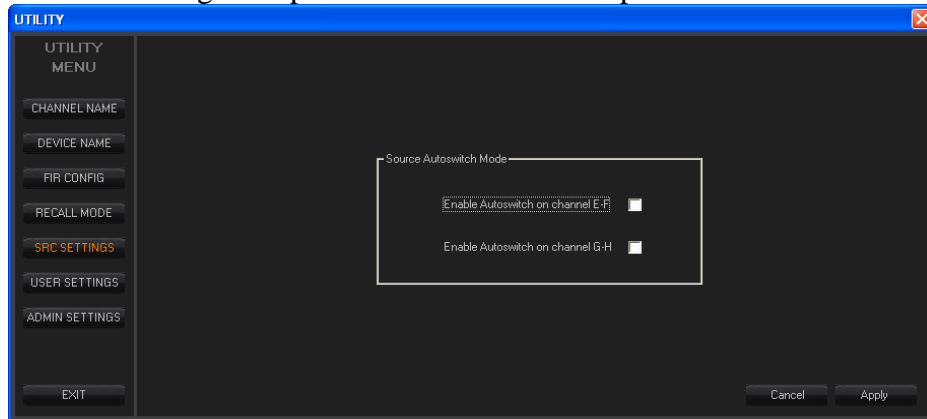
Please see the "Routing Page – Input source"

- USER SETTINGS
if the system is working in USER Mode then the User have not access to the "USER MODE
LOCKED FUNCTIONS".

the User can only change the USER Password. To change the password the User must:

1. Input the current USER Password on the text-box below the label "User Password" (6 characters)

2. Input the wanted USER Password on the text-box below the label "New User Password" (6 characters)

3. use the apply button to confirm it.

the "USER MODE LOCKED FUNCTIONS" parameters can be modify only in ADMIN Mode
- when set the "Program Store Lock" then the User can't store program to device

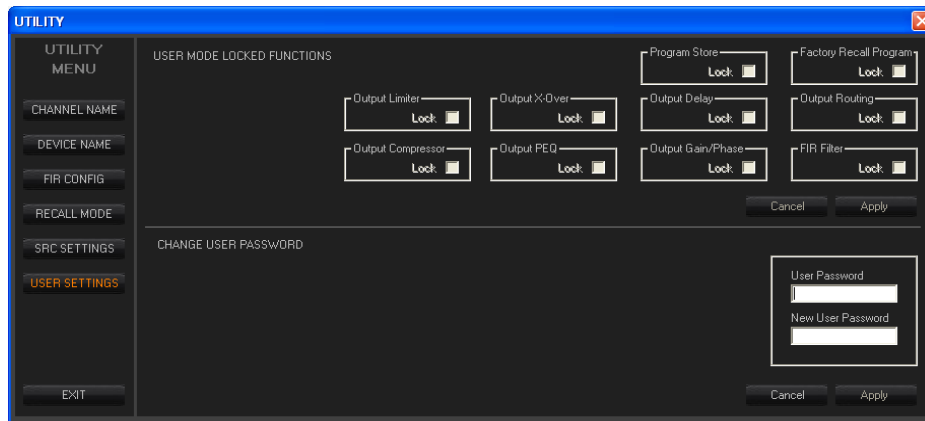- when set the "Factory Recall Program Lock" then the User can't use the button
- when set the "Output Limiter Lock" then the User can't edit all output Peak Limiter parameters
- when set the "Output Compressor Lock" then the User can't edit all output RMS Compressor parameters
- when set the "Output X-Over Lock" then the User can't edit all output HP/LP IIR/FIR filter parameters
- when set the "Output Delay Lock" then the User can't edit all output Delay parameters
- when set the "Output Gain/Phase Lock" then the User can't edit all output Gain and Phase
- when set the "Output Routing Lock" then the User can't edit all output Routing (CH.Y -> OUT-Y)
- when set the "FIR Lock" then the User can't edit all FIR (Block)

− ADMIN SETTINGS
this pages is visible only in ADMIN Mode.
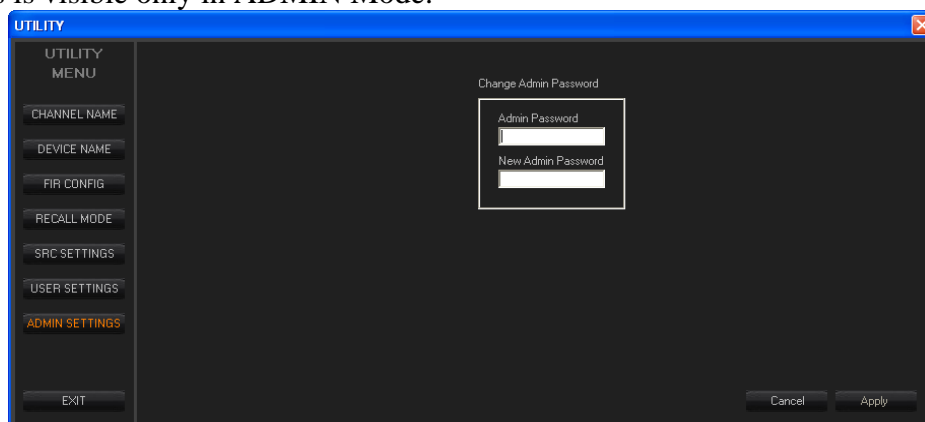
The Admin can change the "Adim password". To change the password the Admin must:

1. Input the current ADMIN Password on the text-box below the label "Admin Password" (6 characters)

2. Input the wanted ADMIN Password on the text-box below the label "New Admin Password" (6 characters)

3. use the apply button to confirm it.

– EXIT

Press the button to close the Utility page and comeback to the "Edit" window. All modification not applied will be lost.

APPENDIX
CUSTOM IIR FILTER

Editing external IIR PEQ Filter

To load an external PEQ filter the user has to click the "Edit PEQ Filter" button in the single external filter frame. The Edit external PEQ window will appear showing the actual external filter coefficients.
When no external PEQ filter is selected, the user can also open the Edit window by clicking the "Edit EQ" button: in this case the first PEQ filter will be shown.

Edit external PEQ window

After clicking the Edit button of the single filter or Edit EQ button, the system shows the following modal window (with the selection of 13 PEQ filters for the input channel and 7 PEQ filter for the output channels)
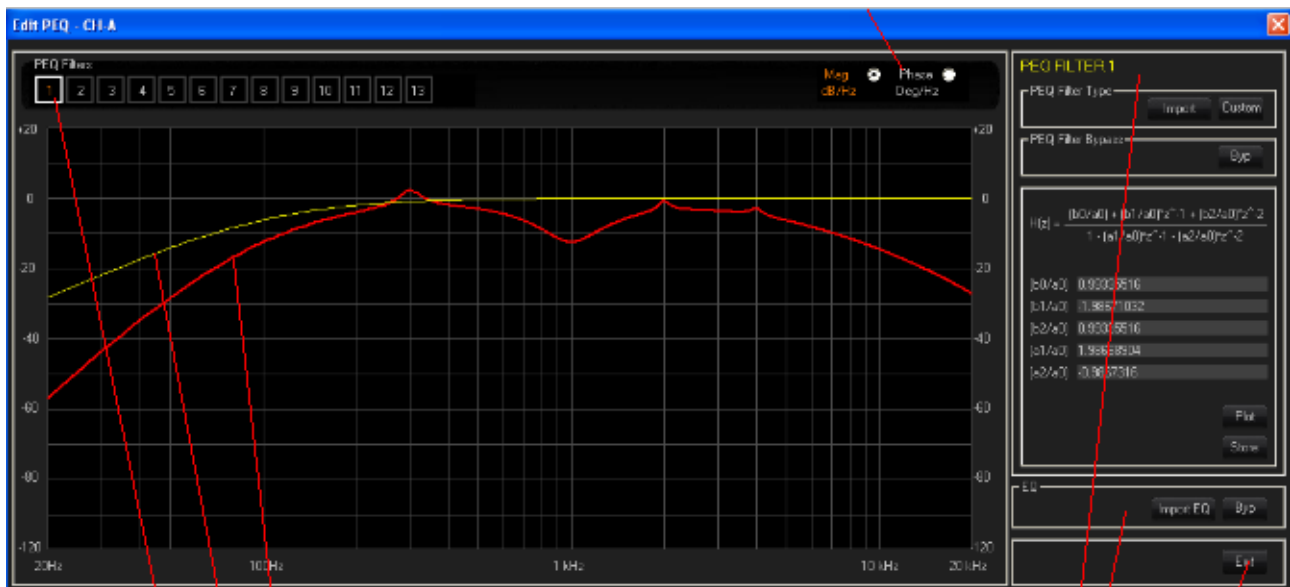
FDT of the overall equalizer (red curve)

FDT of the current selected Filter (yellow curve)

Current selected Filter

Frame with the selected filter settings

Bypass and import an equalizer (more then one filter)

Close window

The Edit window allows to:
- import a single external PEQ filter
- import more then one external PEQ filter
- manually input/edit the external PEQ filter coefficients
- Select a filter and edit it (type, freq, gain, Q, single PEQ bypass and Eq Bypass)
- View the FDT (amplitude or phase) of the single selected PEQ filter and the FDT of the overall

equalizer (amplitude or phase)

If the selected filter type is not "external" then the edit frame will appear as the following:

Select Type

Bypass PEQ

Edit:
Gain, Freq, Q

Current Filter

Overall EQ

Current Filter

Complete EQ
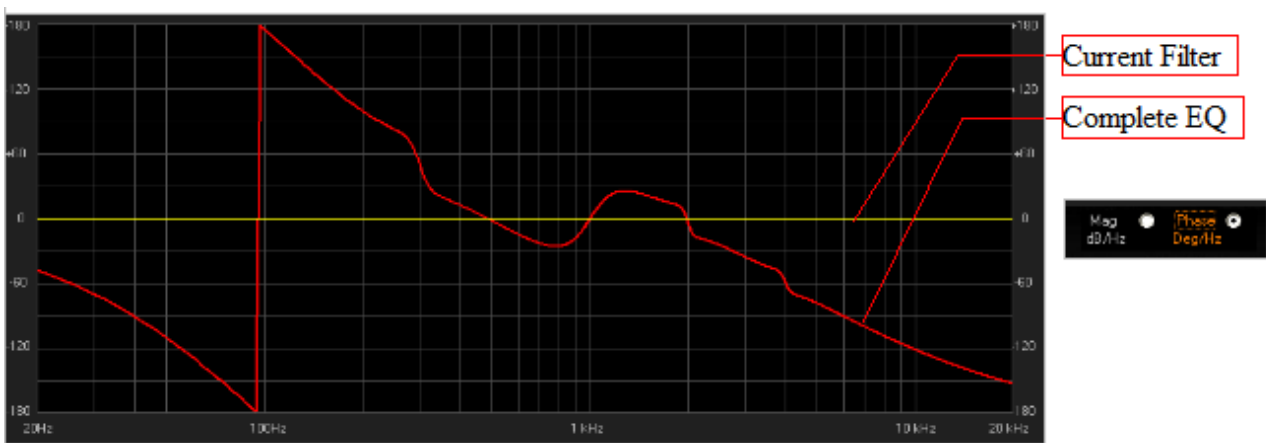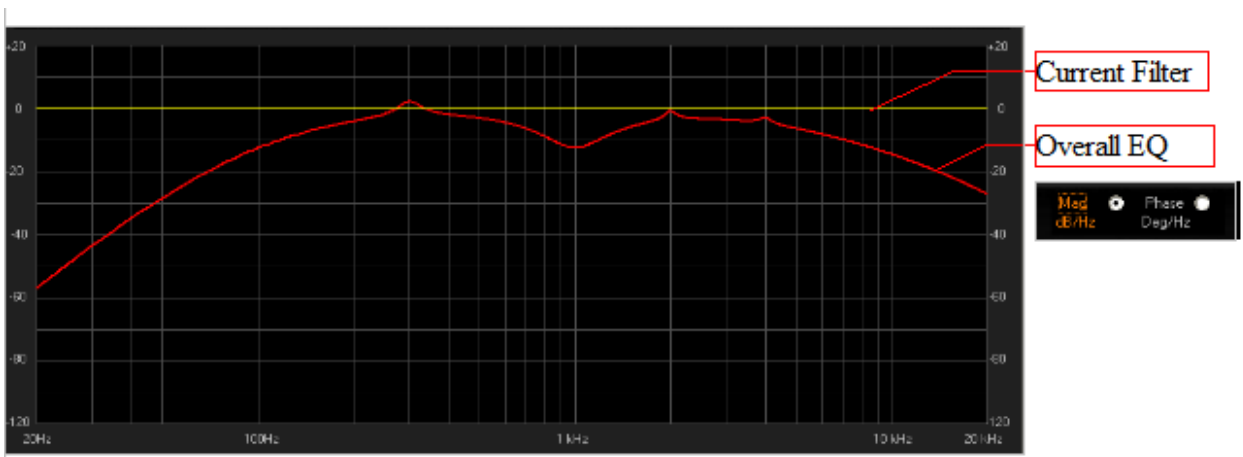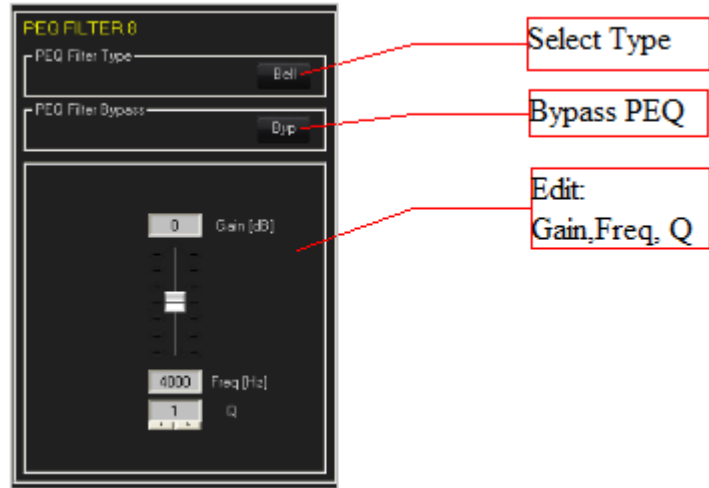
For "external" filters the edit frame will appear as the following:

Current Filter

Complete EQ

Mag dB/Hz   Phase
            Deg/Hz

Current Filter

Overall EQ

Mag dB/Hz   Phase Deg/Hz
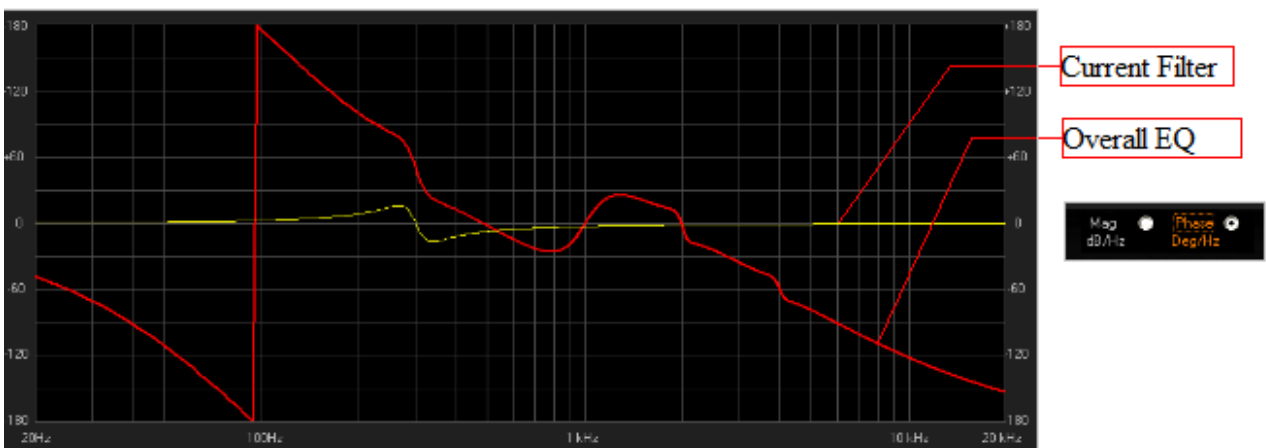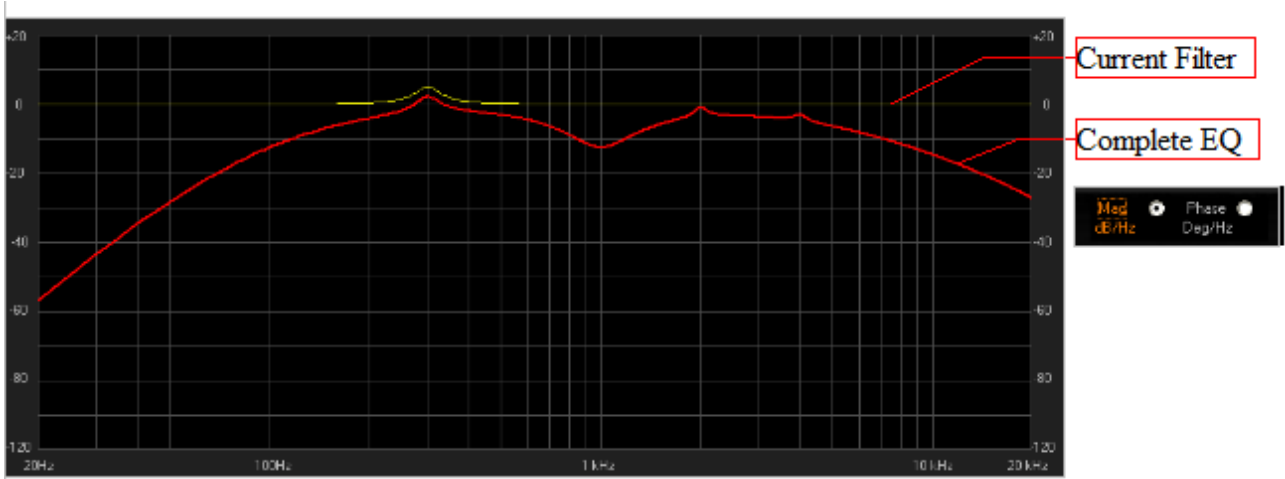
Import single PEQ filter from external file

PEQ FILTER 3

PEQ Filter Type

Import   Custom

PEQ Filter Bypass

Byp

Select Type

Bypass PEQ

Edit

$$H(z) = \frac{(b0/a0) + (b1/a0) \cdot z^{-1} + (b2/a0) \cdot z^{-2}}{1 - (a1/a0) \cdot z^{-1} - (a2/a0) \cdot z^{-2}}$$

(b0/a0) 1.00136094
(b1/a0) -1.99611796
(b2/a0) 0.99514177
(a1/a0) 1.99611796
(a2/a0) -0.99650271

Plot

Store

Typing coefficients:

1. input the right coefficients (floating point, Max< 2   Min >-2, "." as decimal separator) on the (b0/a0),....,(a2/a0) fields
2. the "Plot" and "Store" buttons start blinking to alert the User that something is changed on the filter.
3. Click the "Plot" button to see the FDT of the new filter coefficients

Current loaded Filter (yellow)

Current EQ (red)

Current filter FDT after clicking "Plot" but still not loaded on the device (orange)
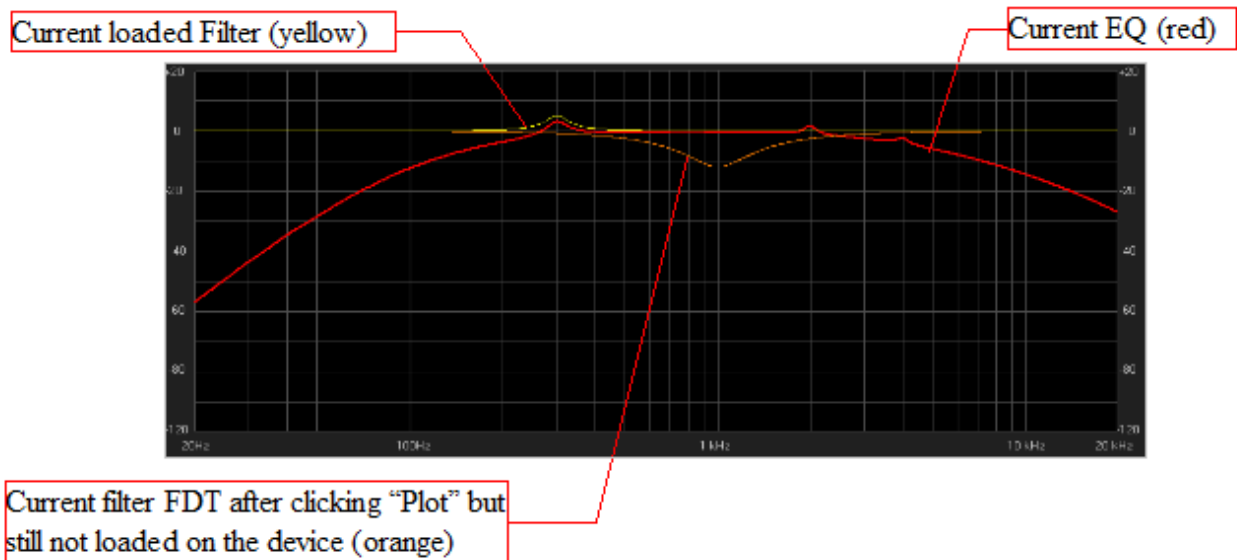
4. Click the "Store" button to load the new filter coefficients in the device. After clicking the graphic will be updated as the following
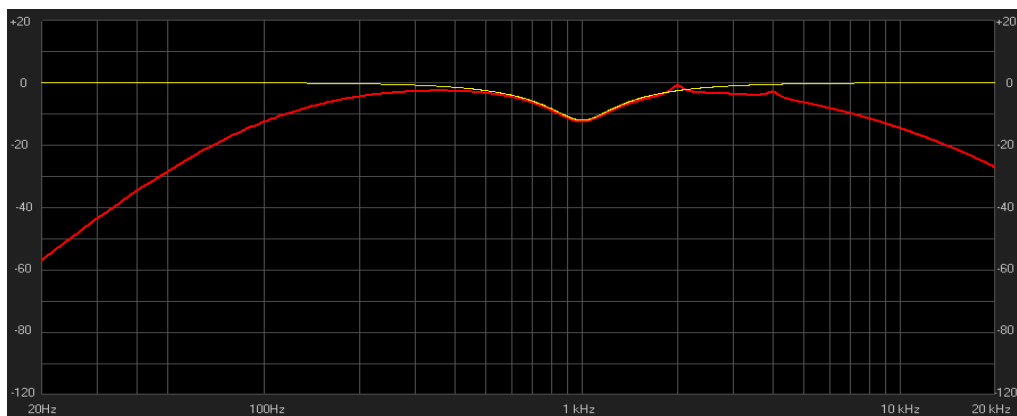
Importing single PEQ filter

To import a single external PEQ filter, the user has to click the "Import" button
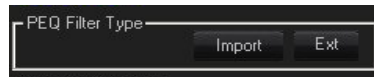
The system shows a dialog window to select a ".txt" or ".csv" filter file (see later for the file format).

After selecting the file and clicking the "open" button, the system shows the current filter imported and ask for a User confirmation to proceed. The system reads and loads the first 5 coefficients in the file, discarding any other further coefficient.
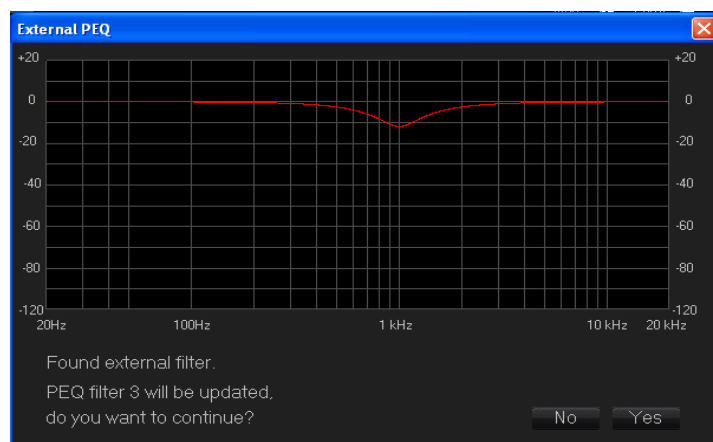
If the User clicks the "Yes" button, the selected external filter will be updated with the new one.

Import EQ filter

To import more then one external PEQ filter, the user has to click on the "Import EQ"button
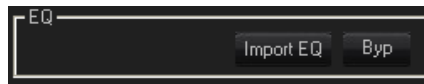
The system shows a dialog window to select a ".txt" or ".csv" filter file (see later for the file format).
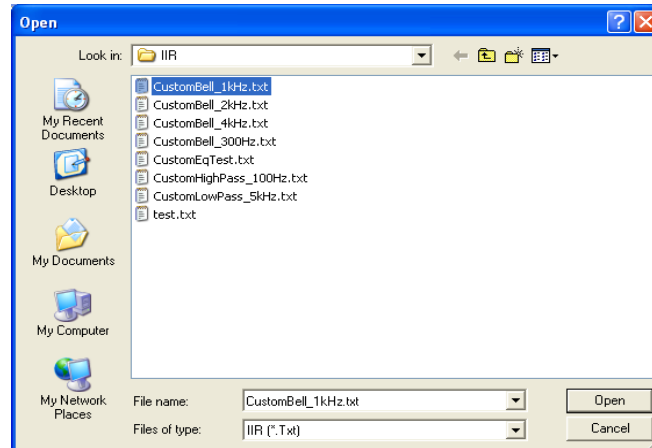
After selecting the file and clicking the "open" button, the system shows the current filter imported and ask for a User confirmation to proceed.

The file should contains 5 coefficients for each filter the user wants to load.

The system parses the file and build a filter every 5 coefficients.

So, if the file contains:

– 5 coefficients, then the system loads the first filter (PEQ1)
– 10 coefficients, then the system loads the first two filters (PEQ1-2)
– ...
– 40 coefficients, then the system loads the first 8 filters (PEQ1-2-...-8)
– ...
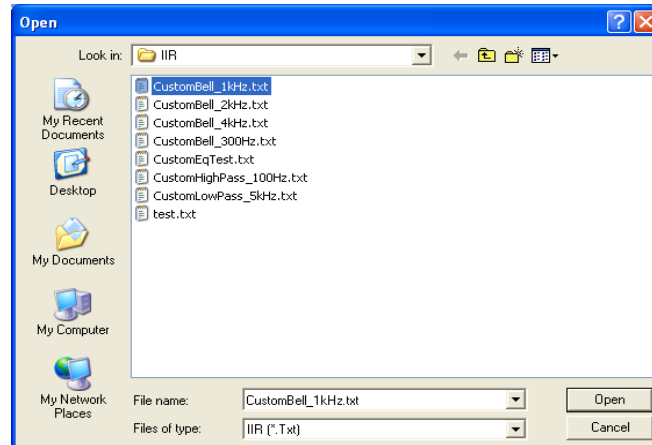– 50 coefficients, then the system loads the first 10 filters (PEQ1-2-...-10)
– more than 50 coefficients, then the system loads anyway the first 10 filters (or 8 filter for the input channel), discarding the additional coefficients.
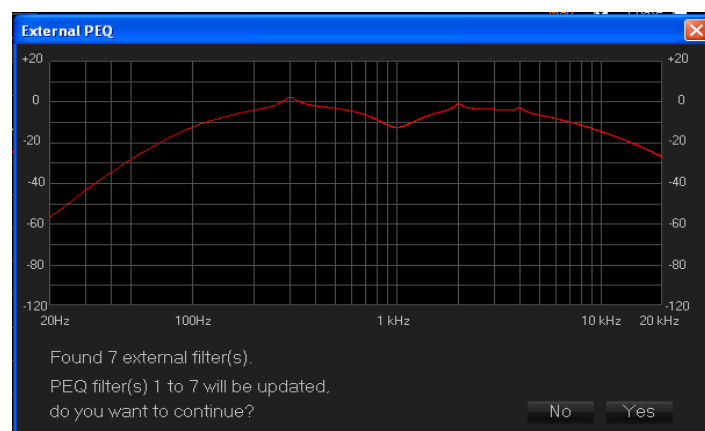
Now, clicking the "Yes" button, the external filter(s) will be updated with the new ones.

File Format

The system can import both ".txt" and ".csv" file. The file coefficients should match the following

structure:

$$H(z) = ((b0/a0) + [(b1/a0) * (z^{-1})] + [(b2/a0) * (z^{-2})]) / (1 - [(a1/a0) * (z^{-1})] - [(a2/a0) * (z^{-2})])$$

Single filter
Format .txt (with "." decimal separator)

b0/a0	0.96673063
b1/a0	-1.90704896
b2/a0	0.94441023
a1/a0	1.90704896
a2/a0	-0.91114086

Format .txt (with "," decimal separator)

b0/a0	0,96673063
b1/a0	-1,90704896
b2/a0	0,94441023
a1/a0	1,90704896
a2/a0	-0,91114086

Format .CSV (vertical)

b0/a0	0.96673063;
b1/a0	-1.90704896;
b2/a0	0.94441023;
a1/a0	1.90704896;
a2/a0	-0.91114086;

Format .CSV (vertical)

b0/a0	0.96673063,
b1/a0	-1.90704896,
b2/a0	0.94441023,
a1/a0	1.90704896,
a2/a0	-0.91114086,

Format .CSV (vertical)

b0/a0	0.96673063
b1/a0	-1.90704896
b2/a0	0.94441023
a1/a0	1.90704896
a2/a0	-0.91114086

Format .CSV (Horizontal)

b0/a0; b1/a0; b2/a0; a1/a0; a2/a0	0,96673063;-1,90704896;0.94441023;1,90704896;-0,91114086;

Format .CSV (Horizontal)

b0/a0, b1/a0, b2/a0, a1/a0, a2/a0	0.96673063,-1.90704896,0.94441023,1.90704896,-0.91114086,

//////////////////////////// **EQ filter (i.e. 7xPEQ txt format)** ////////////////////////////
0.99335516
-1.98671032
0.99335516
1.98668904
-0.98673160
0.01998113
0.03996227
0.01998113
1.42610018
-0.50602472
1.00136094
-1.99611786
0.99514177
1.99611786
-0.99650271
0.96673063
-1.90704896
0.94441023
1.90704896
-0.91114086
1.00281348
-1.96936654
0.98354664
1.96936654
-0.98636012
1.00347818
-1.90590082
0.96965554
1.90590082
-0.97313372
0.99335516
-1.98671032
0.99335516
1.98668904
-0.98673160

///

APPENDIX
IMPORT FIR FILTER

After clicked on "Import FIR" (FIR Block) or

button "Ext" (output section if FIR is selected)
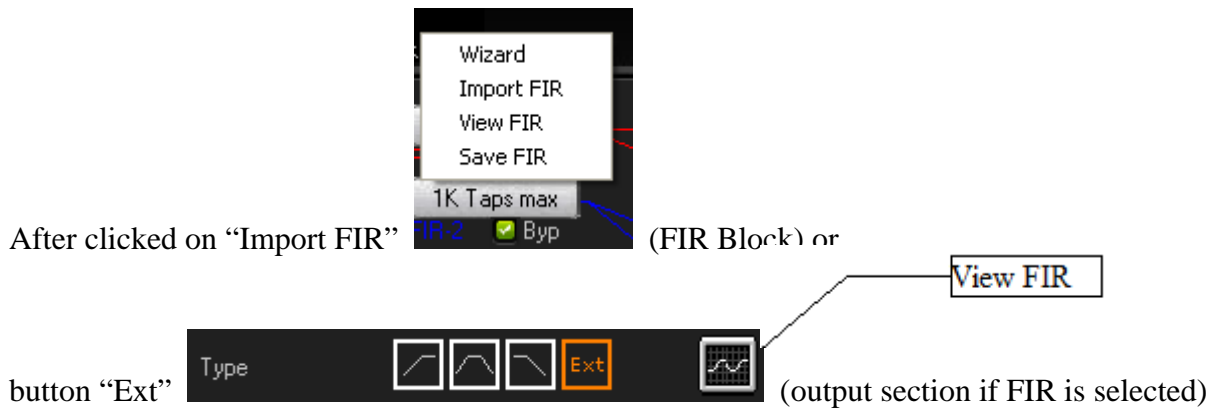
the system show the following modal window:
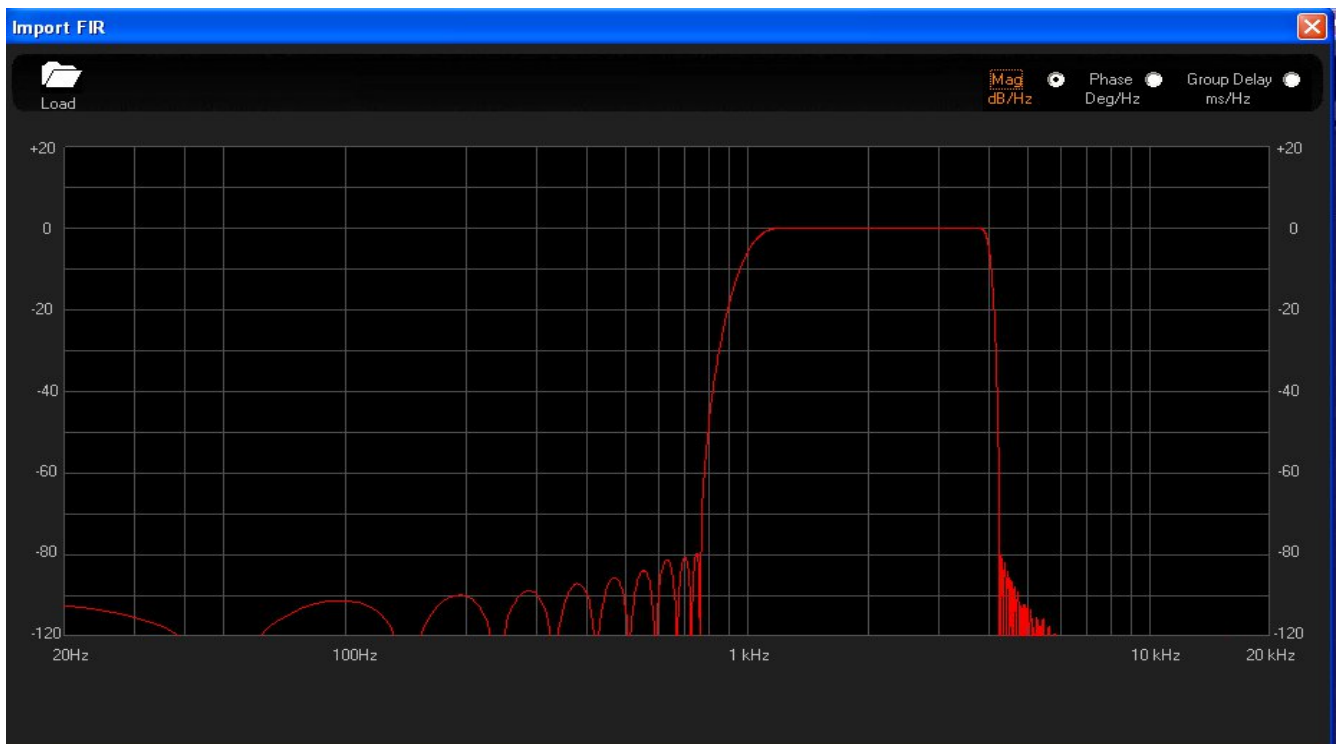
where is possible:
- Import an external FIR with max 512 taps (for output) or up to 4096 taps (for Input FIR)
- see the graphic (Magnitude, Phase, Group Delay) of the FIR Imported

if the User click on "View FIR" then the button "Load" is not visible.

After clicked on the button "Load", the system show a dialog window to select a file
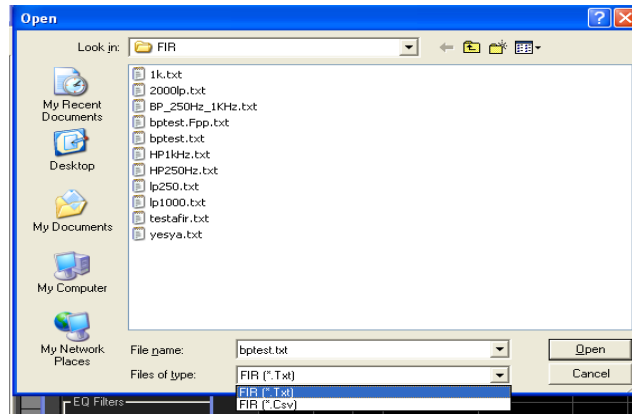
the file of Type can be ".txt" or ".Csv" (see later for the format of the file). After selected the file and clicked on the "open" button then system show the current filter FIR imported and a button "Store" will be show (blinking) to indicate the User that the current FIR is not stored on the device.

Before to store the external FIR loaded the User can see the Magnitude, Phase and Group Delay, if all is ok then click on the button Store and the Filter FIR will be updated with the new one, the Import FIR Window will be close automatically.
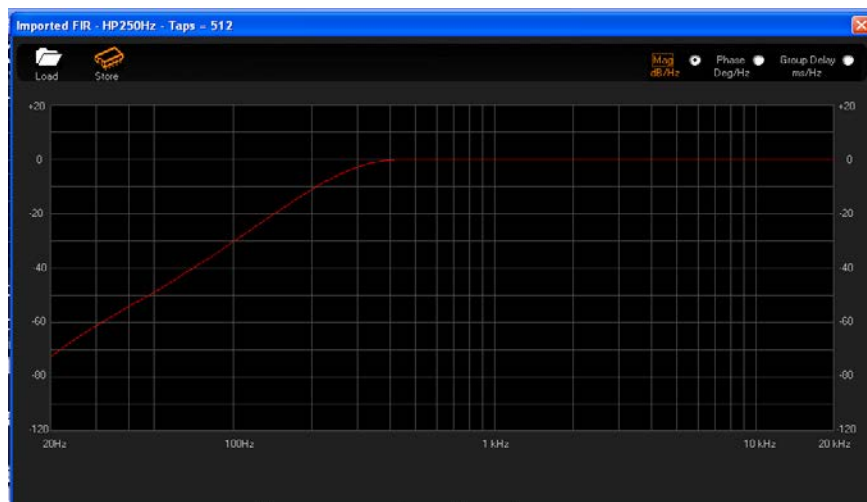
File Format
the system can import file .txt and .Csv, with coefficient in floating point.
The max number of the Taps is 512 (for Output), so the User can load filter FIR with number of the taps from 1 to 512. (if the file have taps > 512 the the system show a message "file wrong")

the range of each coefficient must be >-2.0 and < 2.0

Format .txt   (the separator of the decimal can be ".” or ",")   see the examples 2 type of txt

Coe1	4.76837158203125E-07	4,76837158203125E-07
Coe2	-4.76837158203125E-07	-4,76837158203125E-07
Coe3	-2.38418579101563E-06	-2,38418579101563E-06
Coe4	-5.24520874023438E-06	-5,24520874023438E-06
..	...	...
CoeN	...	...

Where N is the number of the taps

Format .CSV (vertical, the separator of the decimal can be ".” or ",” see 3 examples of coef

Coe1	4,76837158203125E-07;	4.76837158203125E-07,	4.76837158203125E-07
Coe2	-4,76837158203125E-07;	-4.76837158203125E-07,	-4.76837158203125E-07
Coe3	-2,38418579101563E-06;	-2.38418579101563E-06,	-2.38418579101563E-06
Coe4	-5,24520874023438E-06;	-5.24520874023438E-06,	-5.24520874023438E-06
..	...	...	...
CoeN	...	...	...

Where N is the number of the taps

Format .CSV (Horizzontal)   see 2 examples

Coe1;......,coeX;......,coeN	4,76837158203125E-07;......;-5,24520874023438E-06;.....

Coe1;......,coeX;......,coeN	4.76837158203125E-07,......,-5.24520874023438E-06,.....

APPENDIX
WIZARD

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 distorsions, 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.
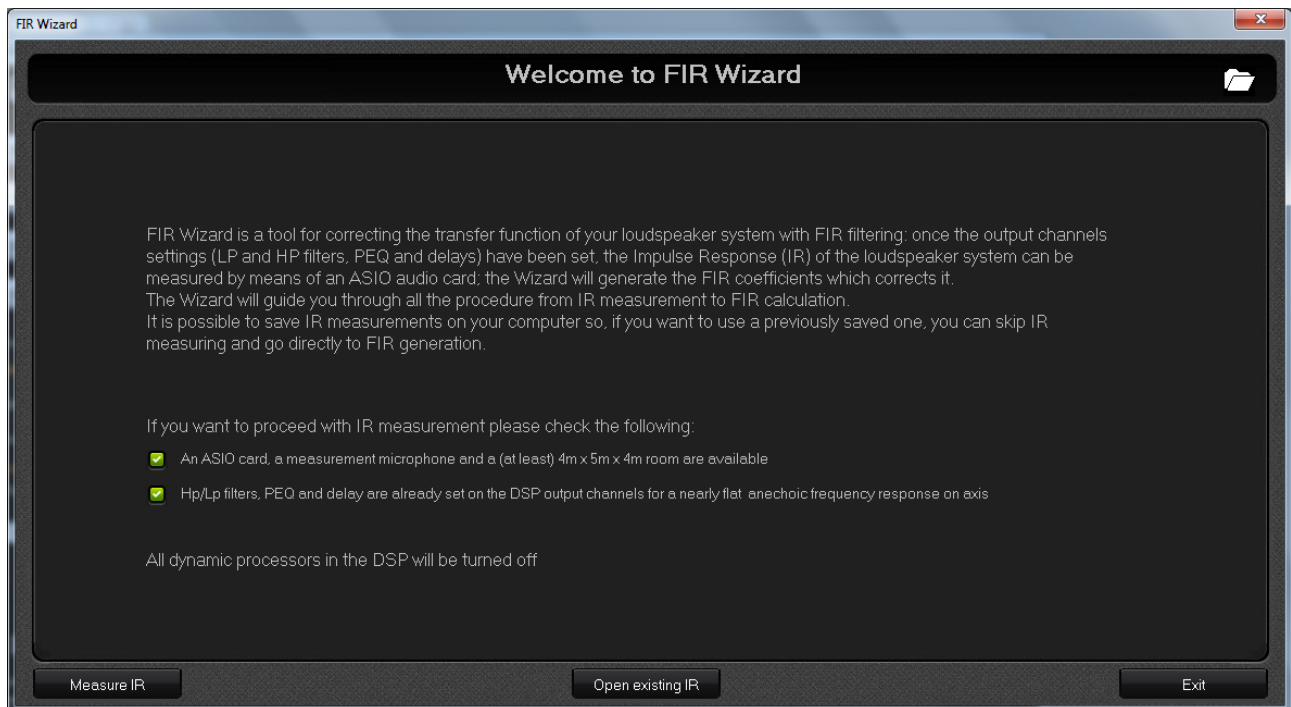
From the home page, the user can choose to:

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 (.zwz)

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 botton 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 dinamic 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.
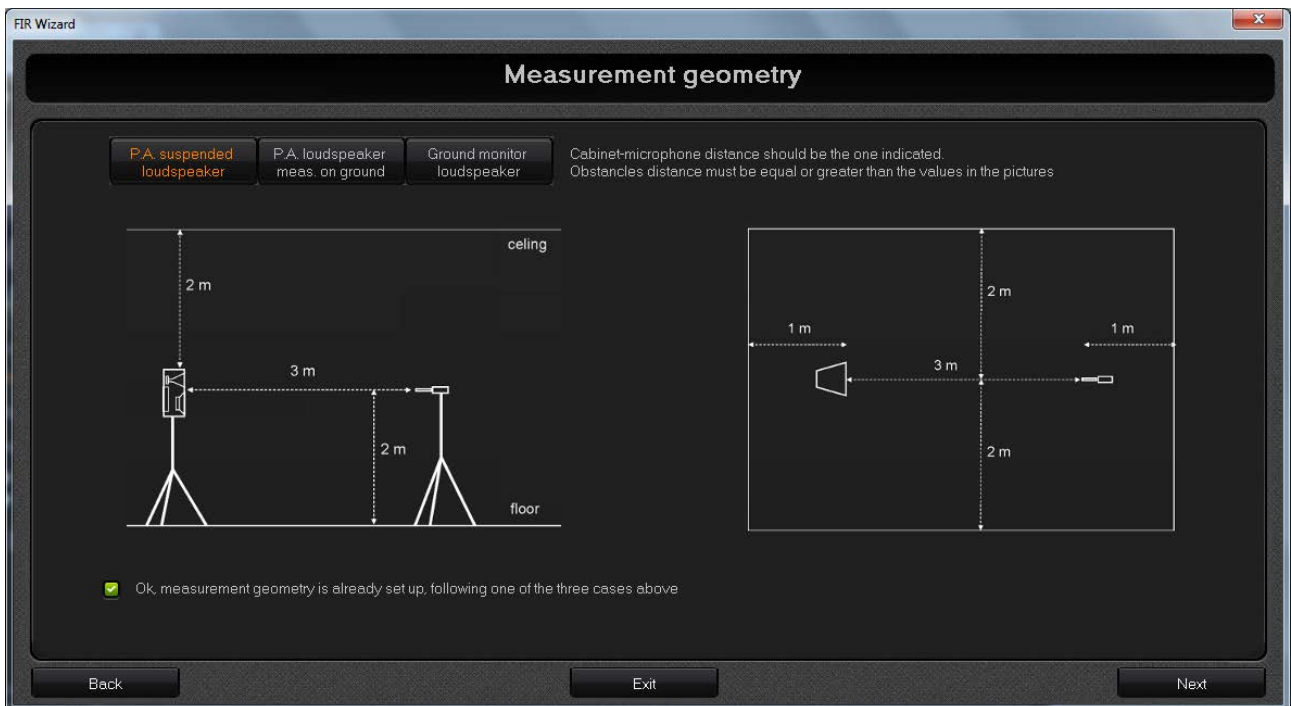
Clicking the buttons at top left of the screen. the three measurement scenarios are shown (vertical section on the left, planimetry on the right). This is just a visualization and reminder for the user that should check every time the correct geometry and choose one of the three options. The choice doesn't 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.
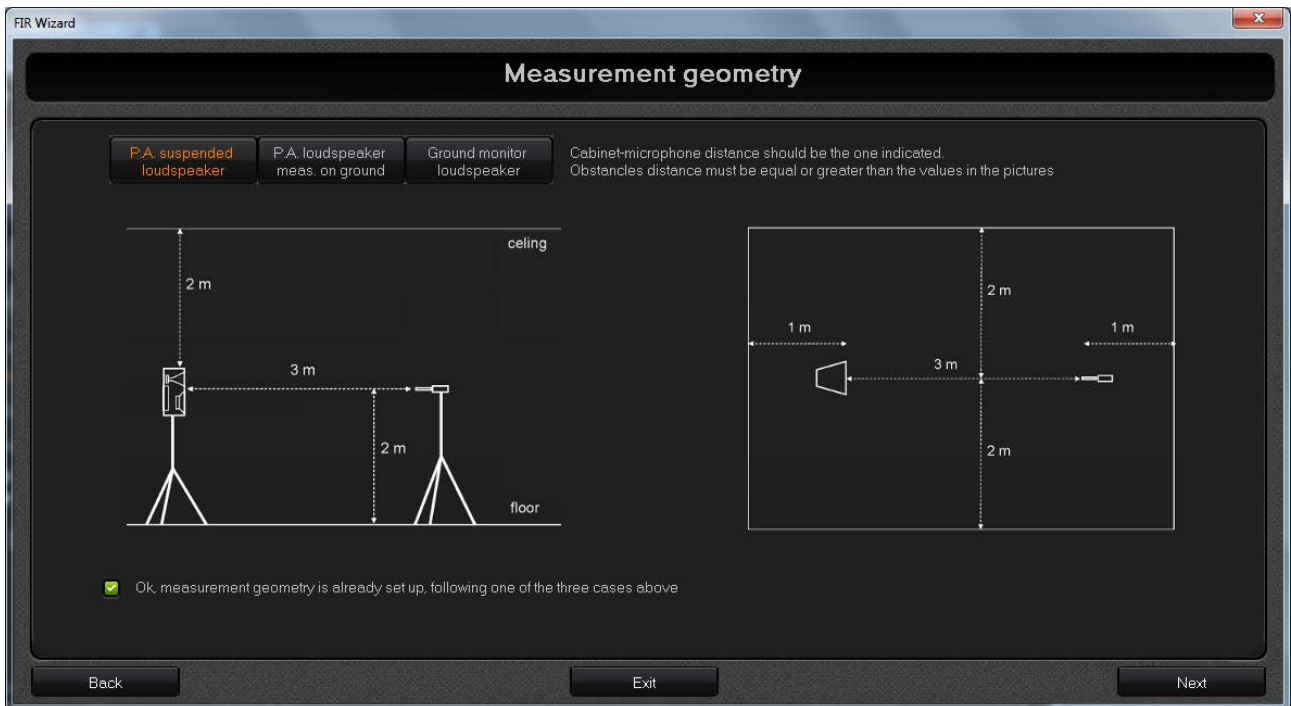
## 2) Loudspeaker system and microphone lying on the ground

The loudspeaker must be tilted to point the microphone, that should be stricly 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 preceeding scenario, so it can be used when the room height is less than 4m.

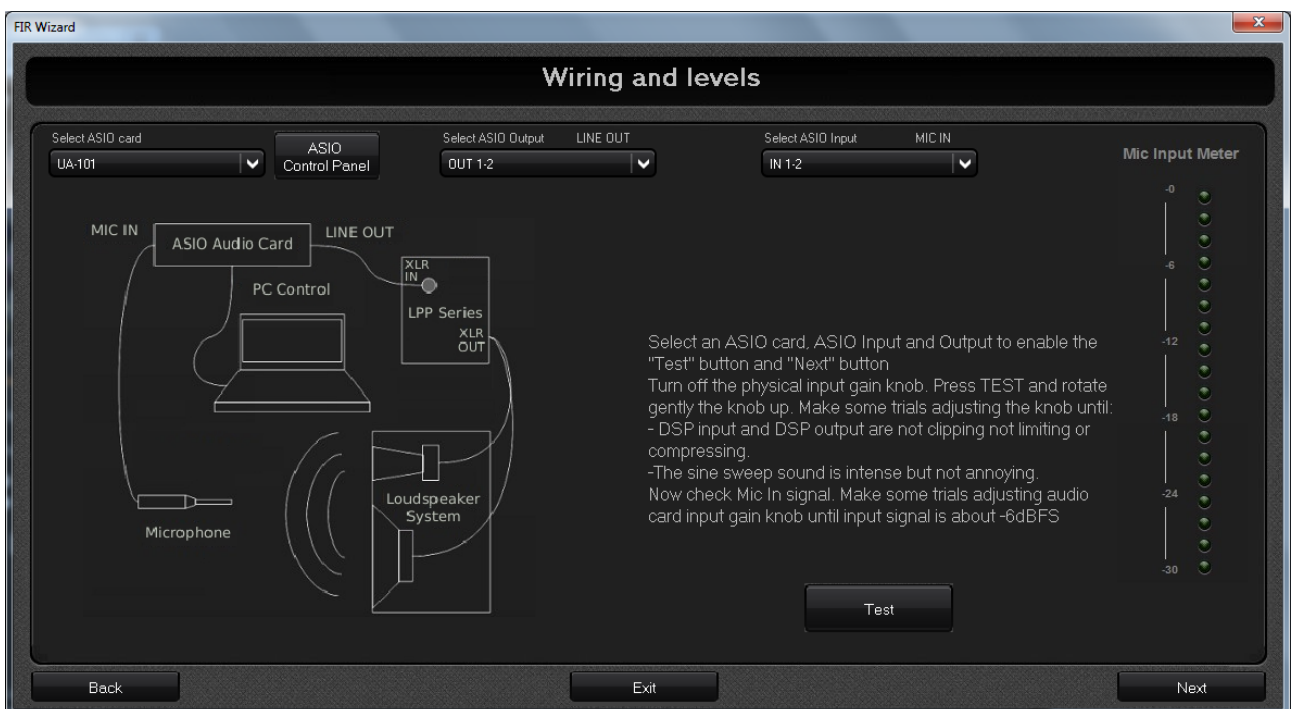## 3) Ground monitor in operating position

The microphone should be in axis with the loudspeaker at an height of about 1.7m, where the
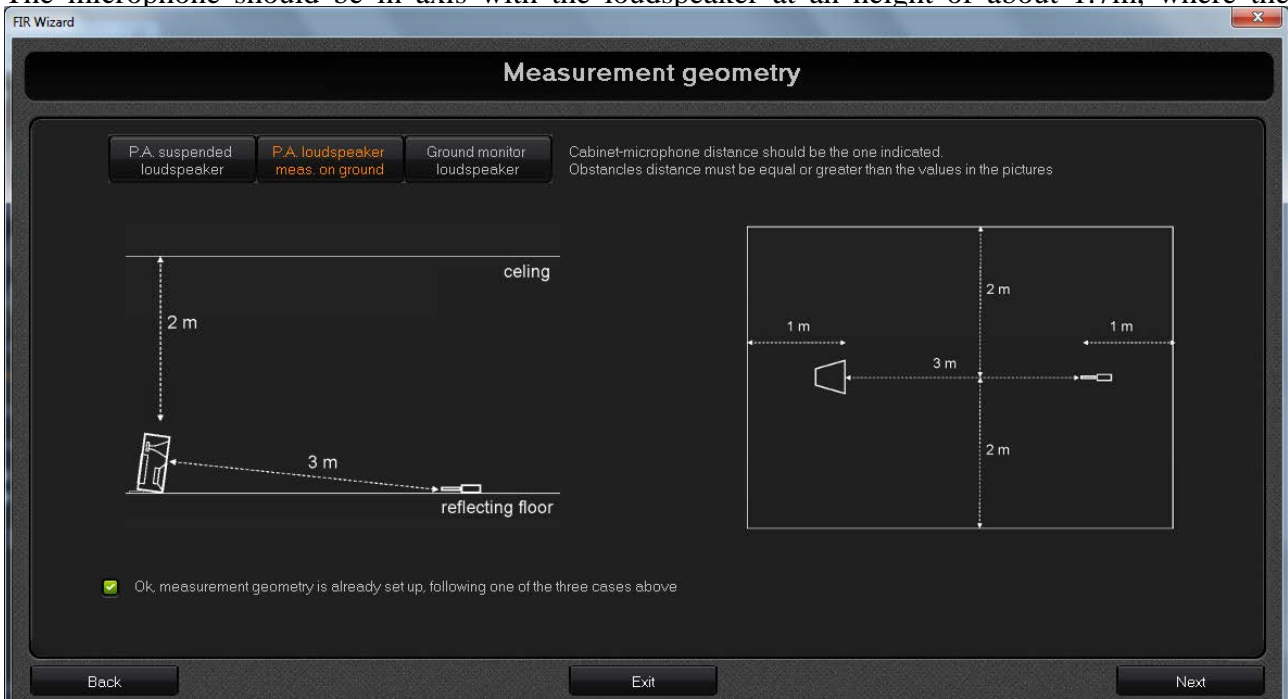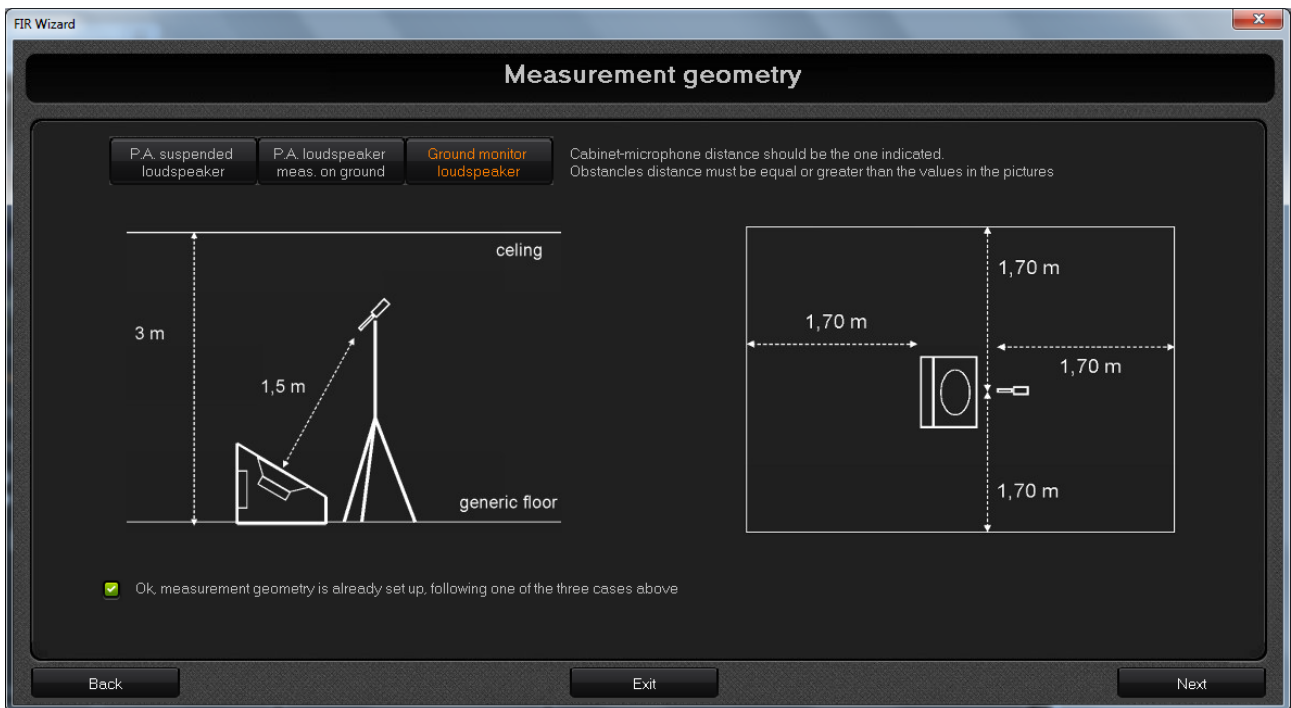
So, arrange the measurement choosing one of the three geometry setup and click "Next" button to proceed.

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 untill 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
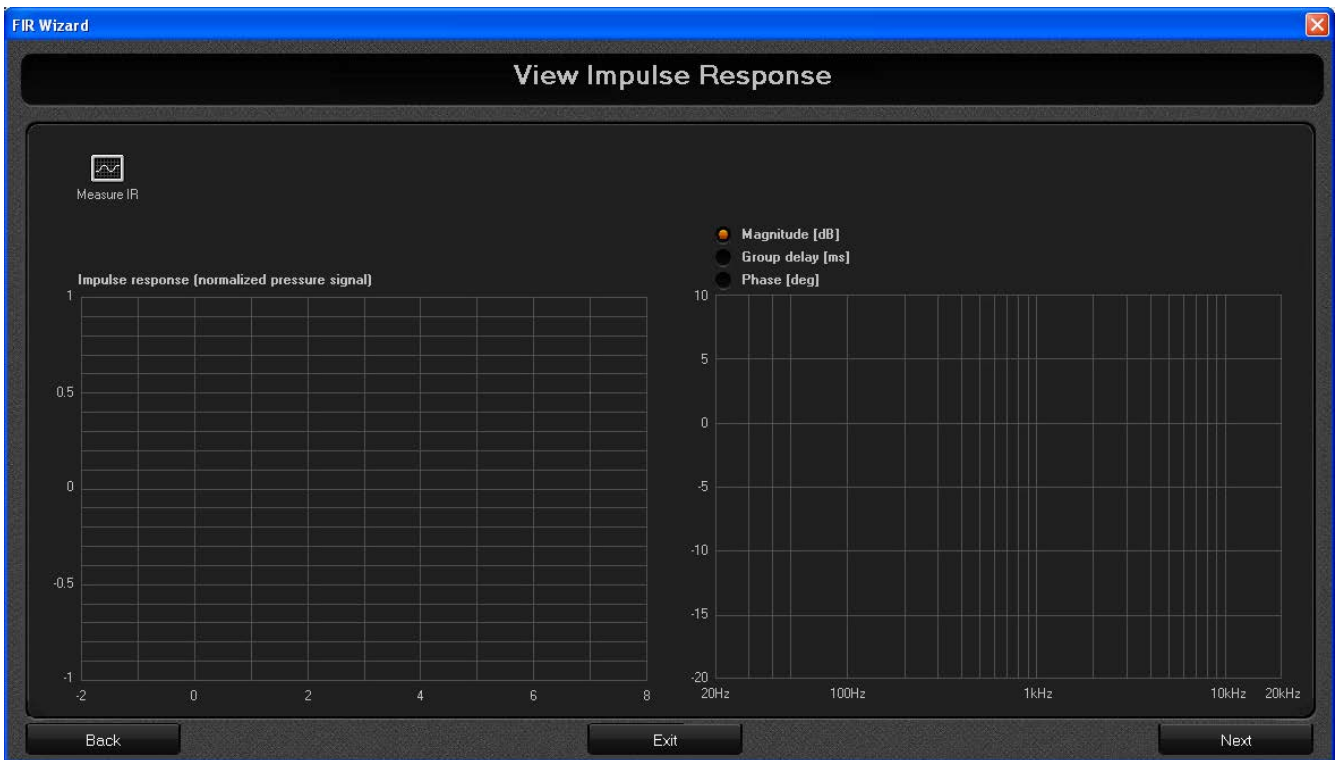
Clicking the "Measure IR" button at the top left of the screen, the measurement process starts and an
activity progress bar shows the activity completion status. Wait in silence for the process completion. After the sine sweep, the Wizard starts the IR computation and the measured IR is plotted on the left graph with its peak centered on zero, while the magnitude, group delay and wrapped phase frequency response is shown on the right graph..
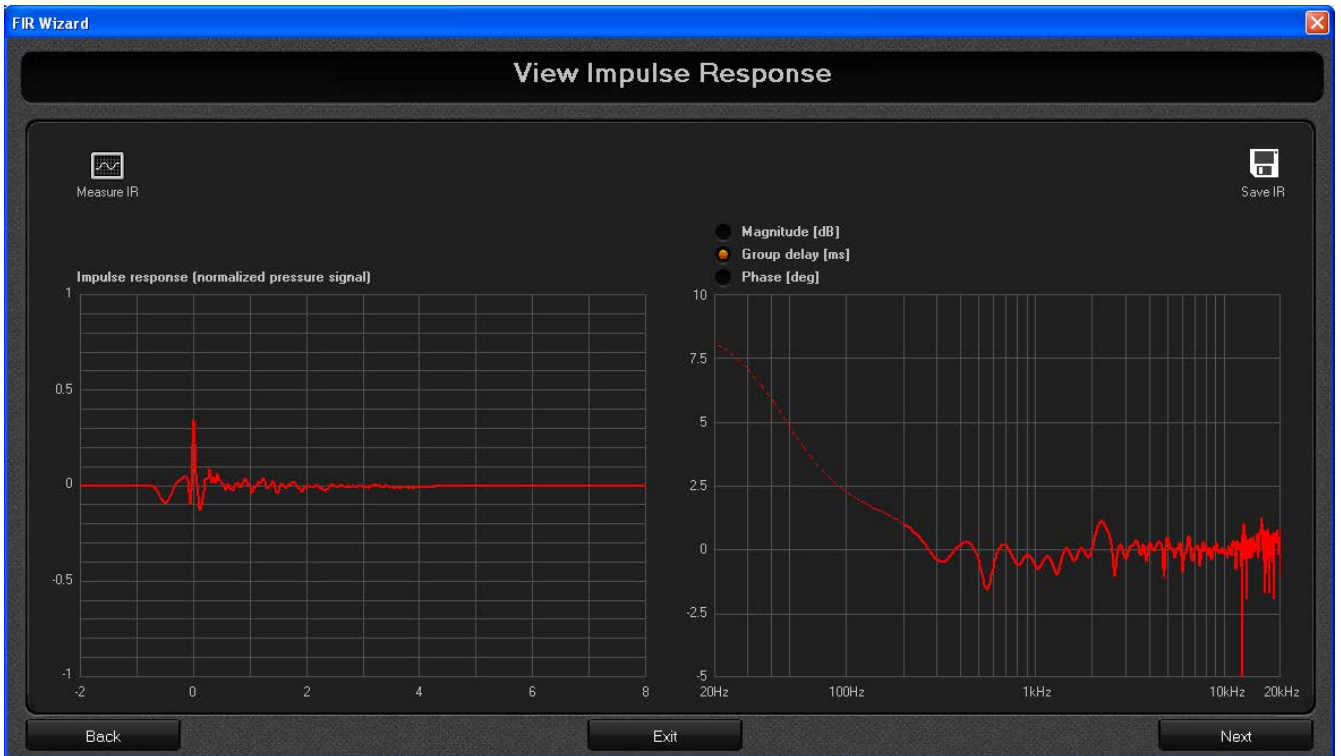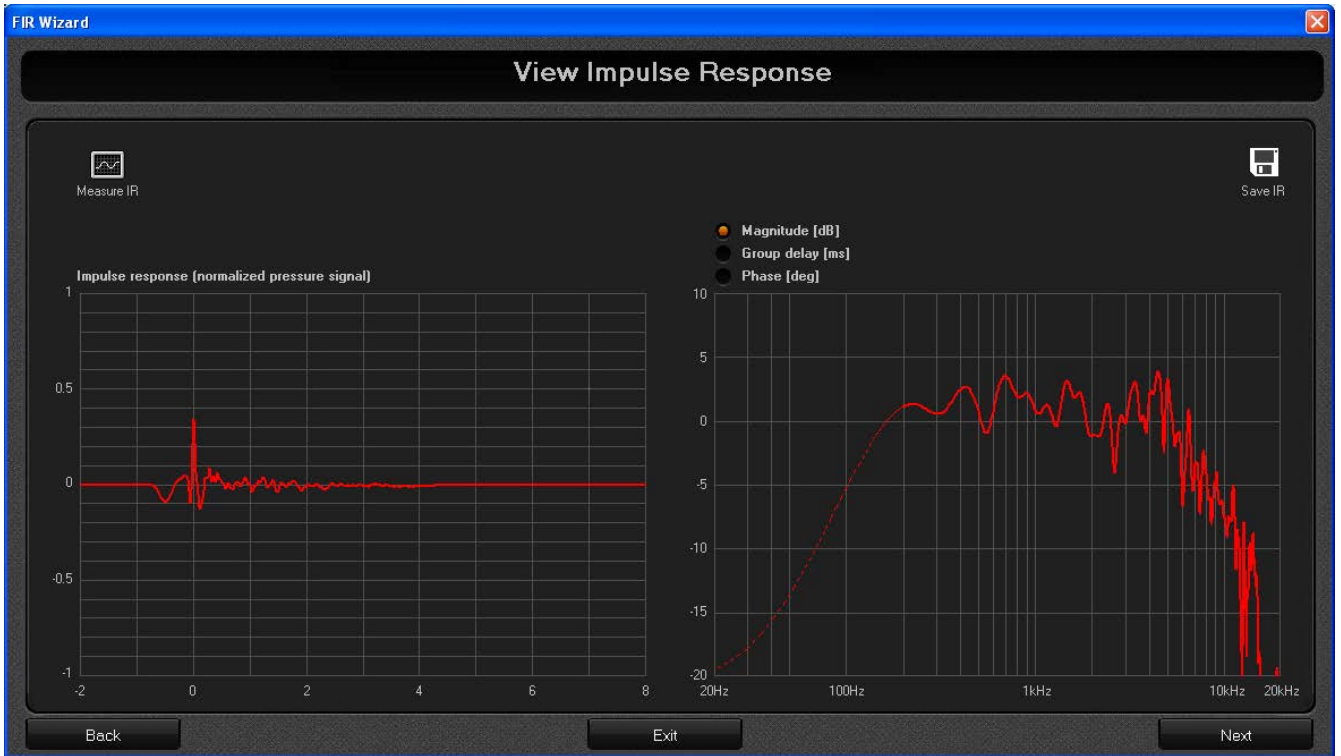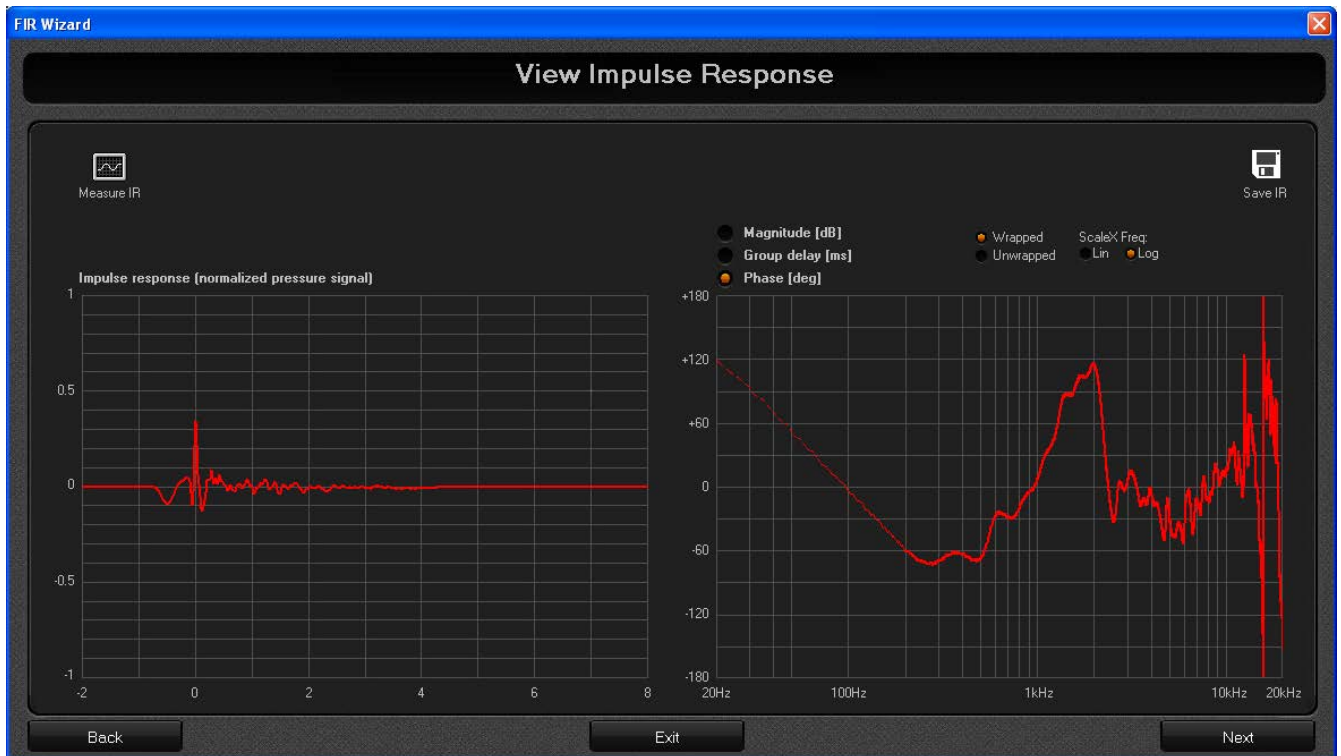
Discontinuities in the group delay view may appear (as the picture above shows near 12kHz), due to the poor phase computation accuracy in the frequency points where the magnitude response has a very deep dip.

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".

Impulse response (normalized pressure signal)
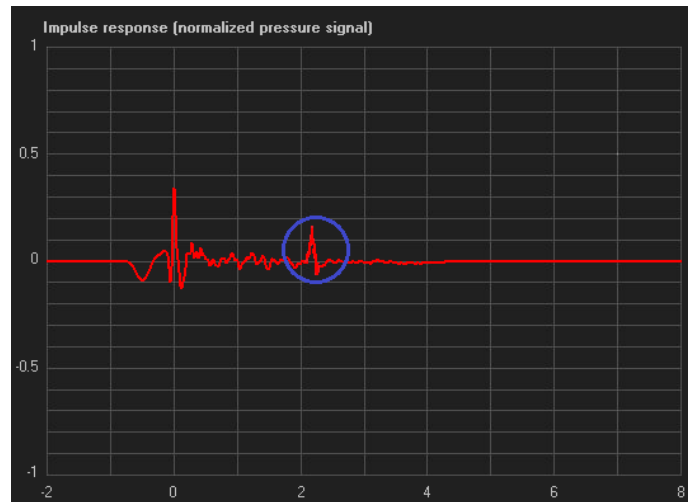
As a consequence of the windowing process, the computed IR is not accurate in the low frequency range: it is not possible to estimate the frequency response below the reciprocal of the analisys time interval, due to the time-frequency duality. So, to avoid misleading view, the frequency response plot 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
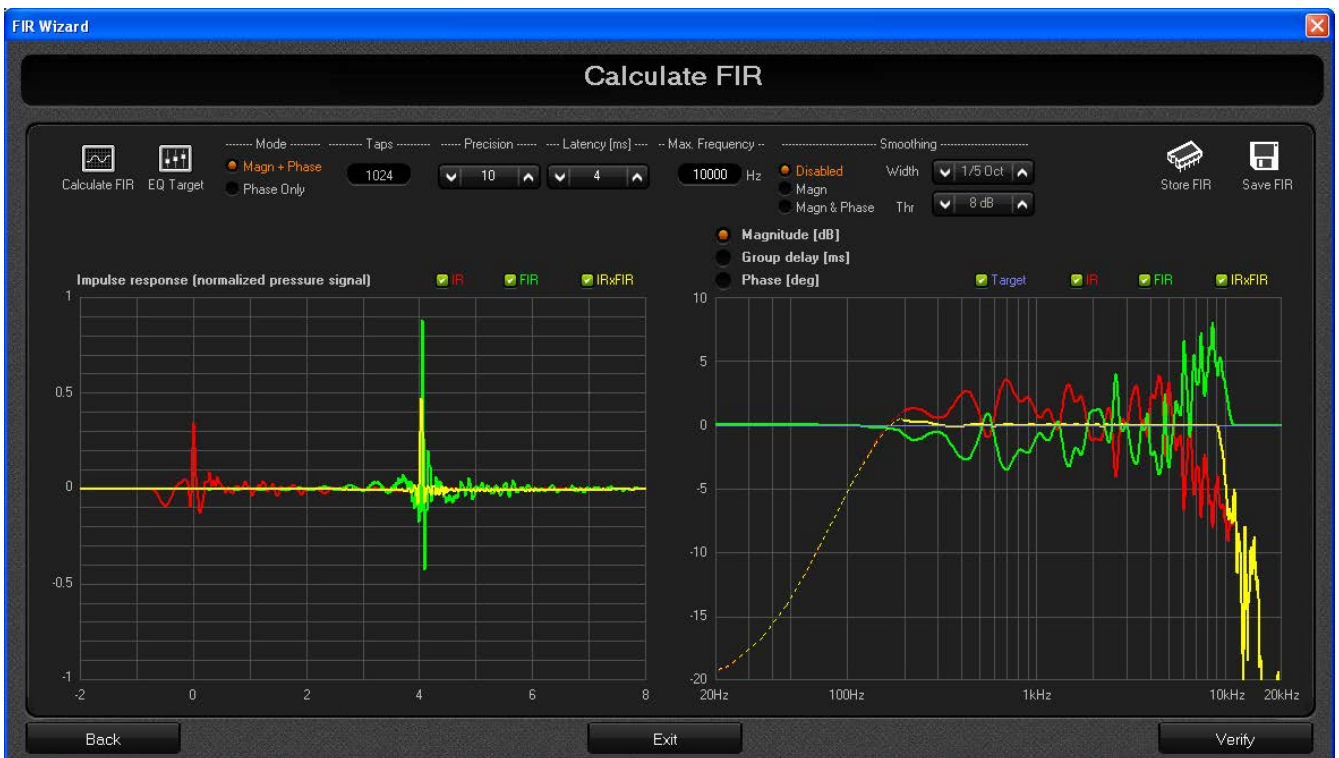
In this page the FIR filter coefficients are computed starting from default settings (or from the parameters set of the FIR Wizard project, when a project file is recalled at the Wizard home page).

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.
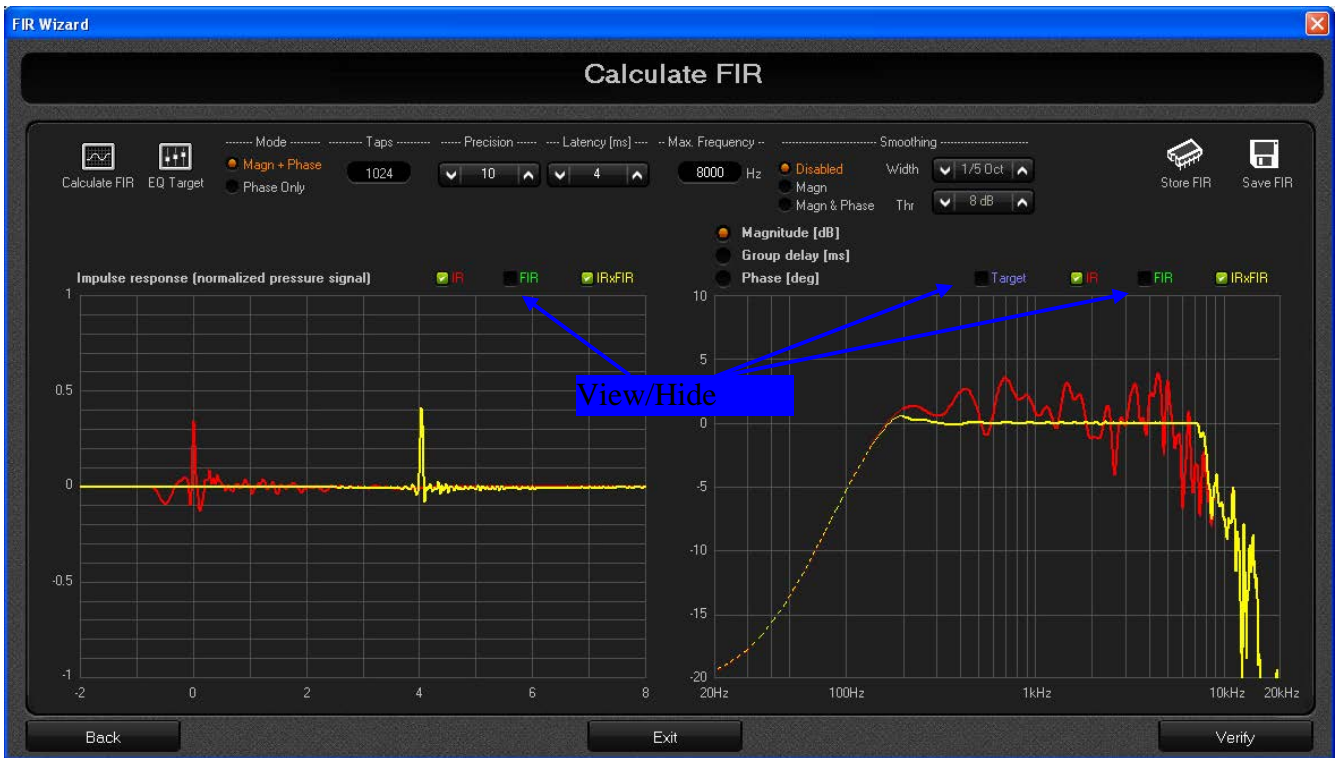
Left clicking a point in the plotting area allows to show the cursor coordinates, while right cliking shows the plot zoom menu.
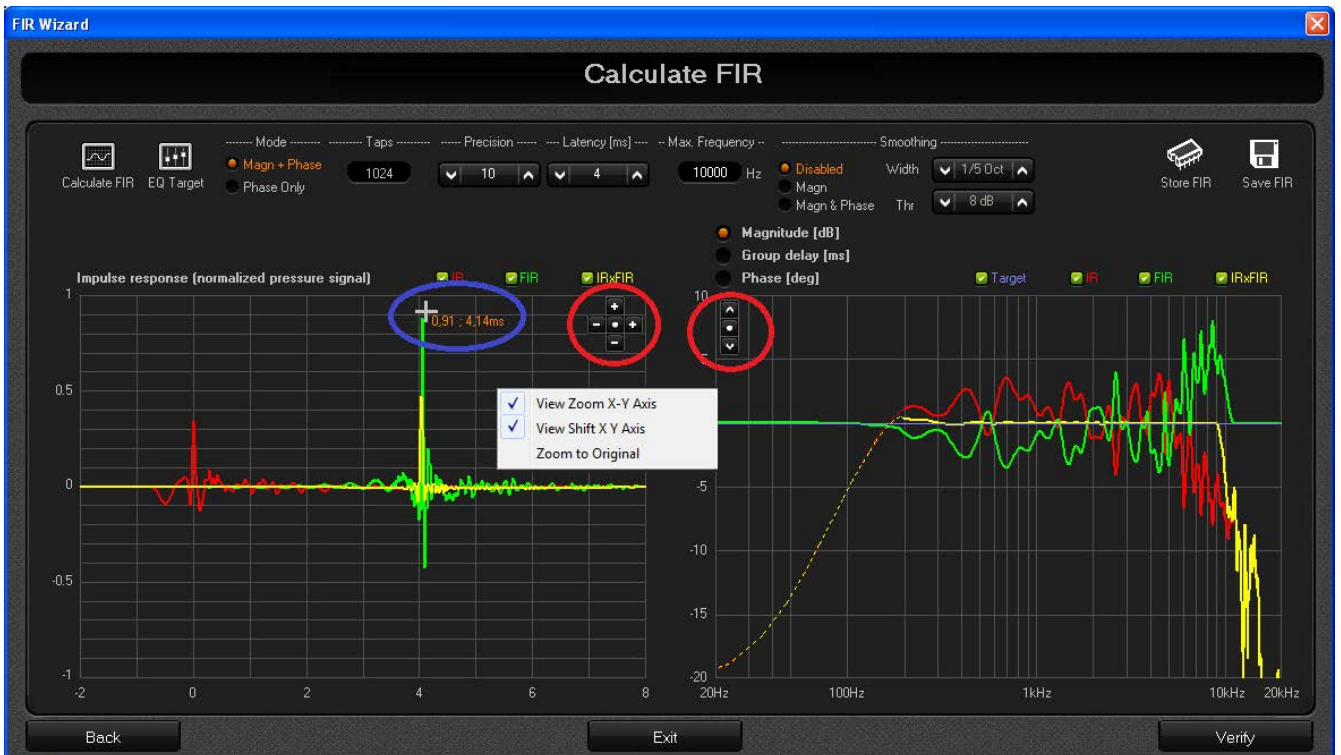
Let's now analize in more details the linearization capability of the computed FIR by comparing the time and frequency domain plots of the IR and the IRxFIR curves, for a flat target magnitude.
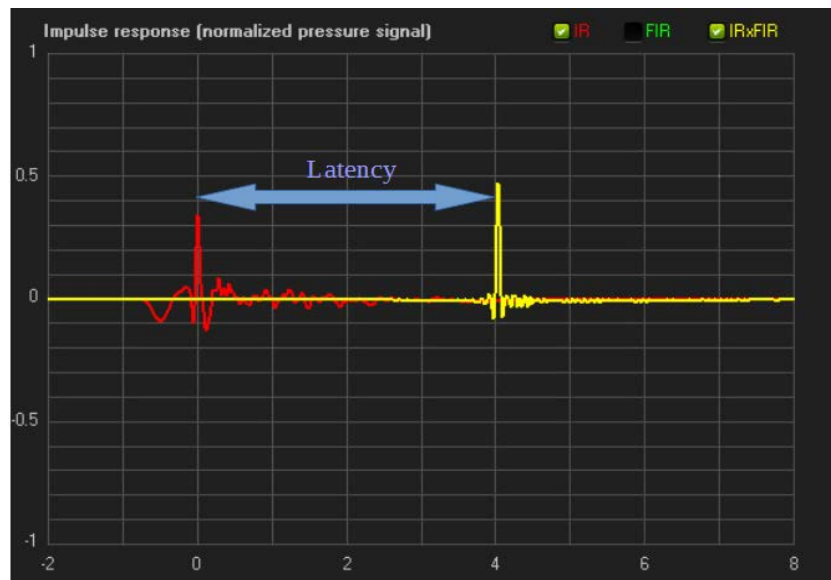
In the time domain what we should ideally get is a unitary pulse centered at 0ms (Dirac delta), that is the TD equivalent to a flat frequency response.

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).
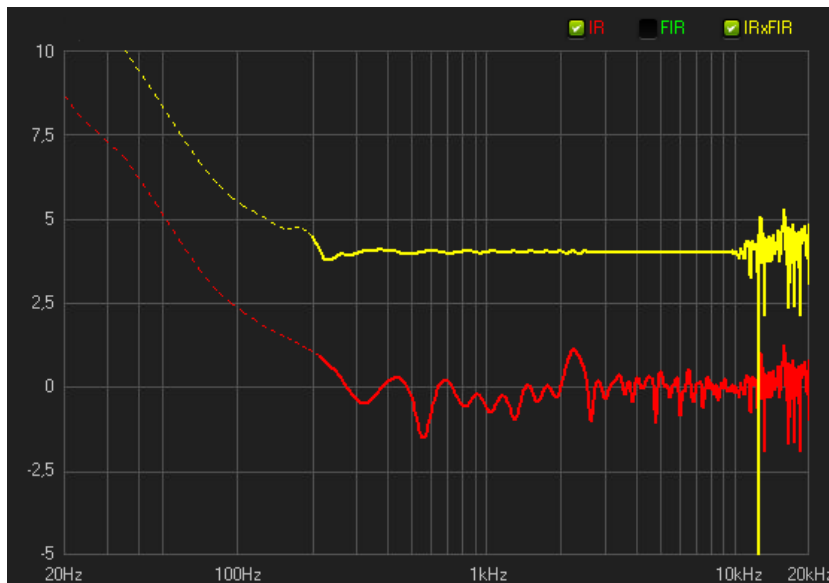
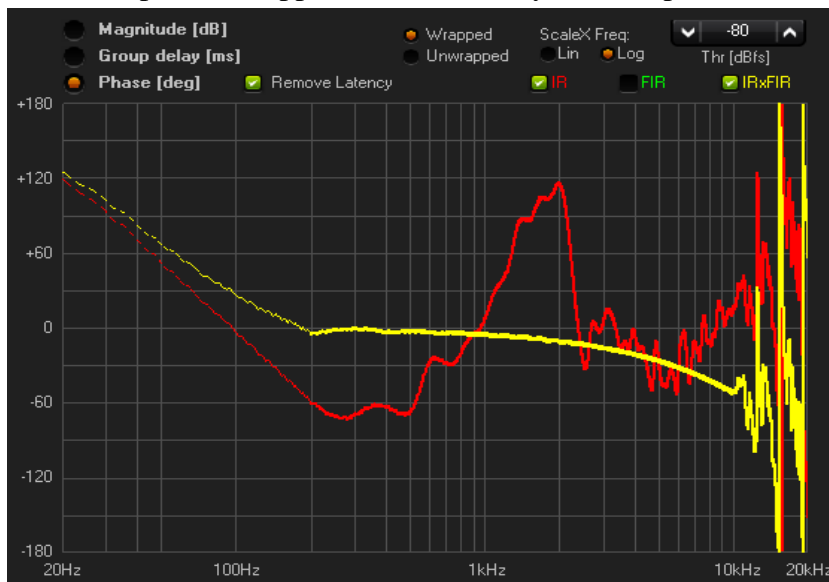The graph below shows the phase (wrapped without latency) linear up 10kHz (see the yellow curve)

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 capabilty of the FIR filter in axis (i.e. in the meausured 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)
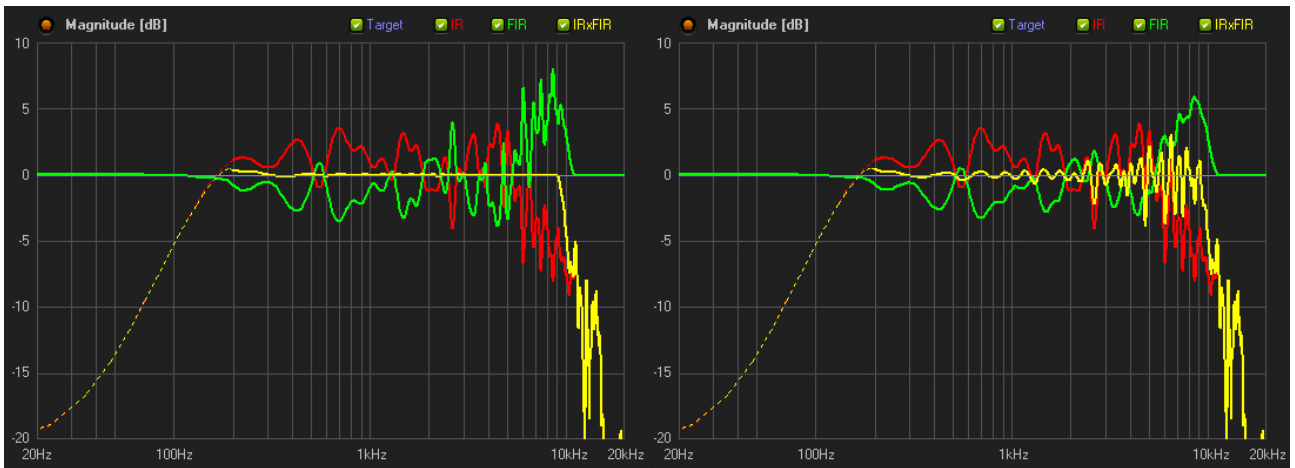
When the "Magn+Phase" mode is selected, the phase smoothing is always applied to the whole frequency range, even if the threshold is not equal to 0dB.

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.
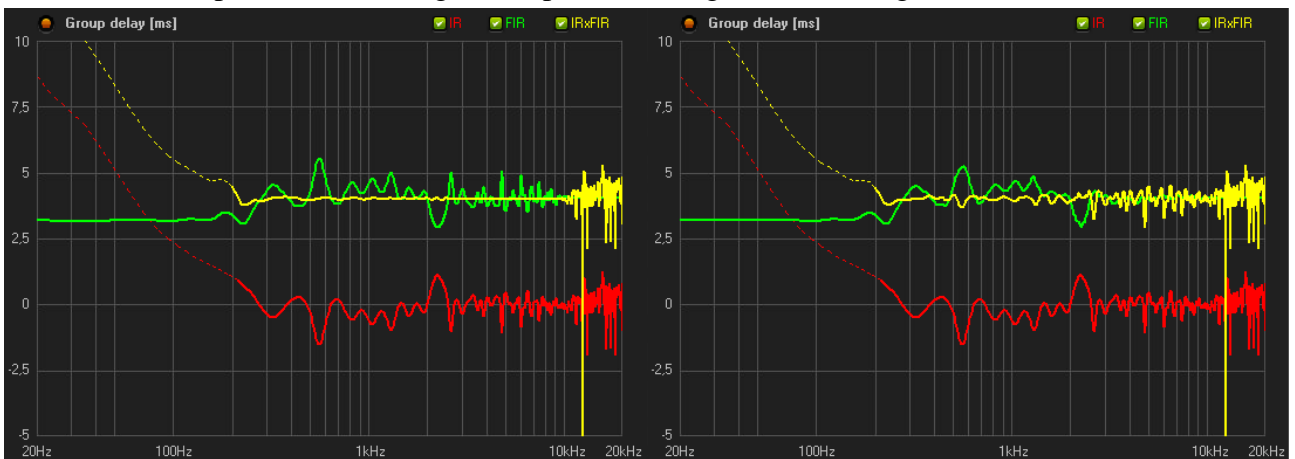
- *Latency*

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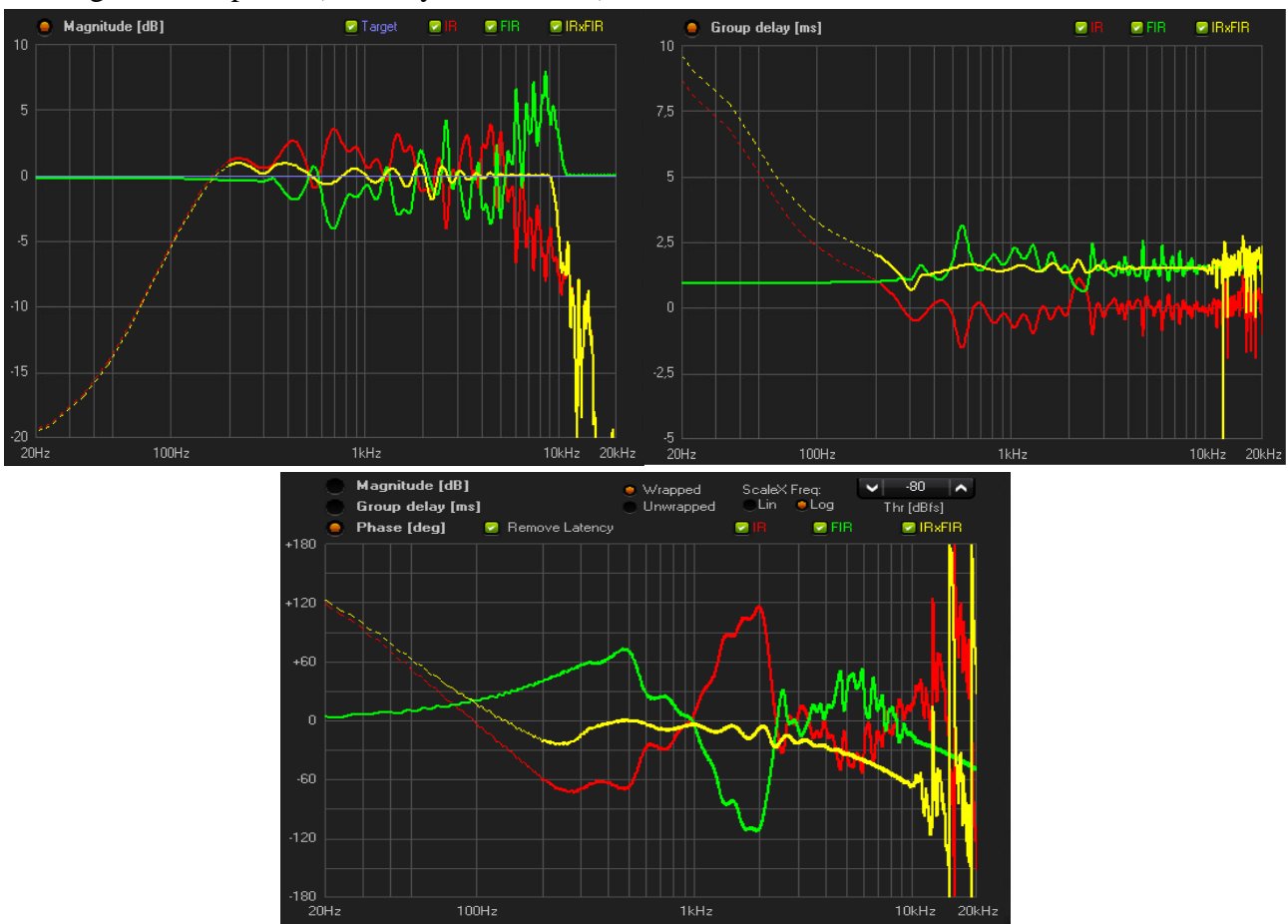
As a general rule, it is suggested to try different values for the latency parameter and choose the lower one that still gives satisfying equalization results for the particular application, basing on plotted curves and listening tests.

- *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.
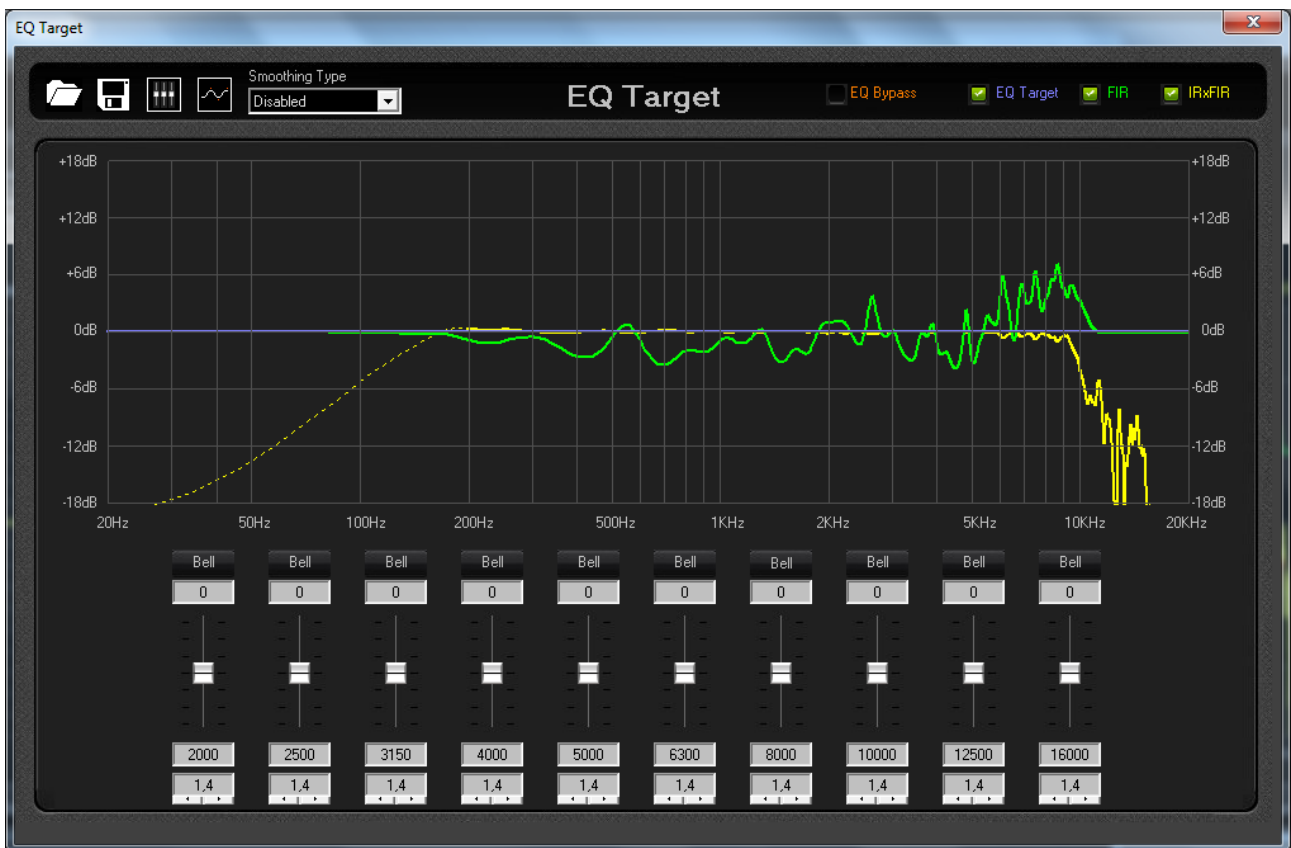
In the "EQ Target" page, the magnitude response of the target curve (blue line), of the FIR filter (green line) and of the IRxFIR (yellow curve) are overlaid. Curves in the plot can be hidden through the corresponding check-boxes at the top right of the screen.

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 pop-up.
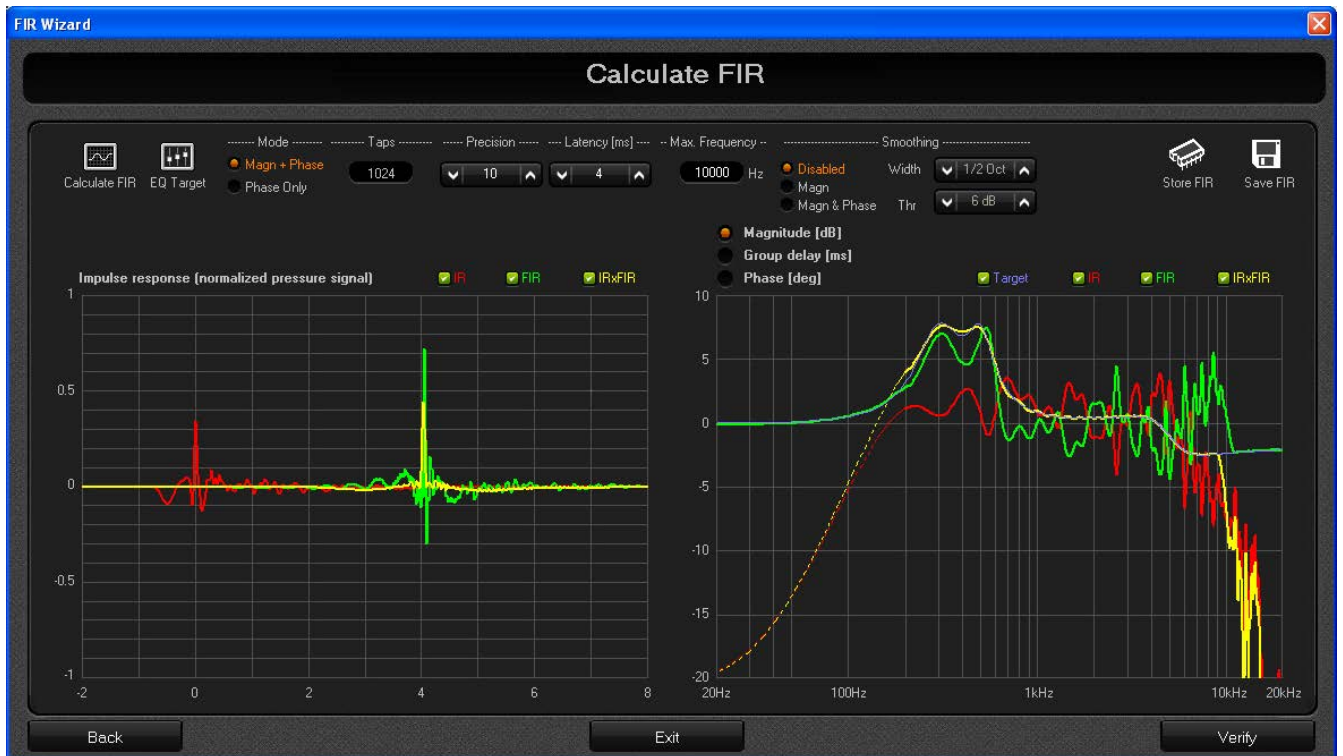
In "Phase Only" mode the target curve can't be set.

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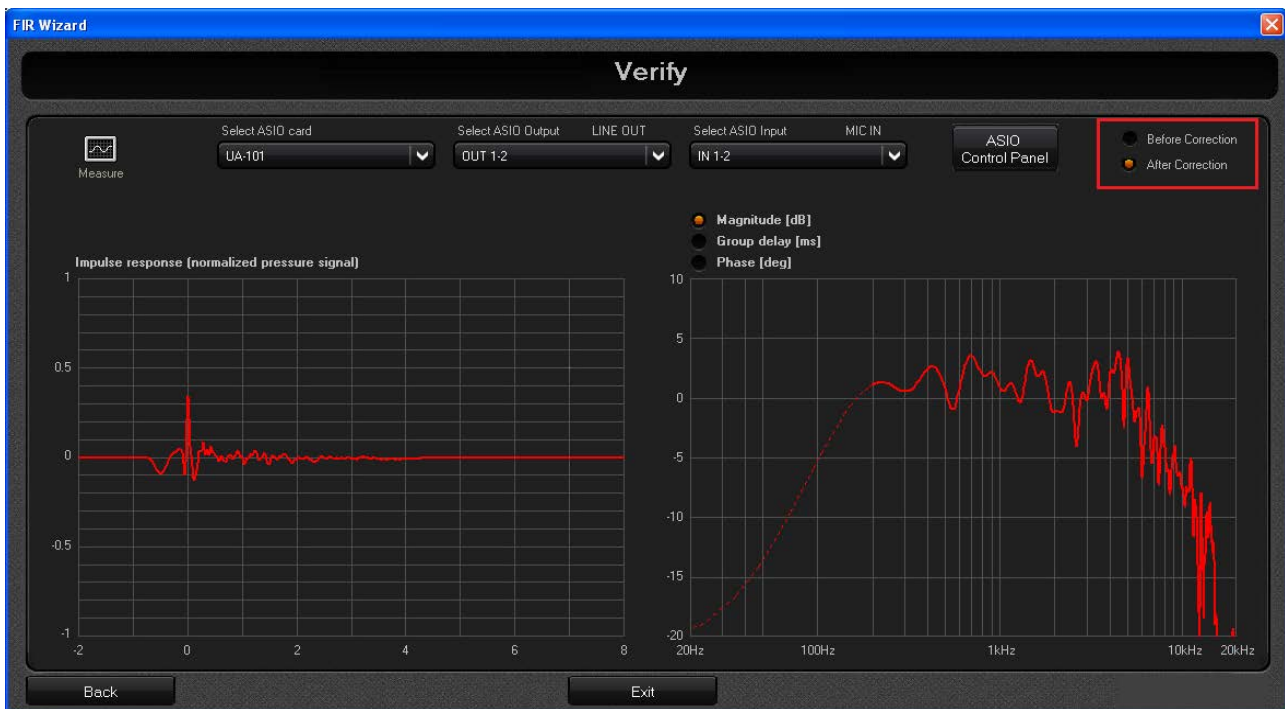
To start the verify process, click the "Measure" button at the top left of the screen and wait in silence. After the sine sweep completion, the new IR is computed and shown in the time/frequency domain plots. It is possible to compare the original and the compensated IR by acting on the "Before / After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

Phase's plot on graphic
there are different mode to see the plot of the phase.
– option to see the wrapped/Unwrapped phase
– option to see the frequency axis (scaleX) in Linear/Logarithm mode

- view the plot with/without the latency added by FIR.
- Threshold, to remove the phase value on the frequencies where the module of the response is under the threshold (this is because when the module of the response is very low, is not possible to calculate with precision the value of the phase)
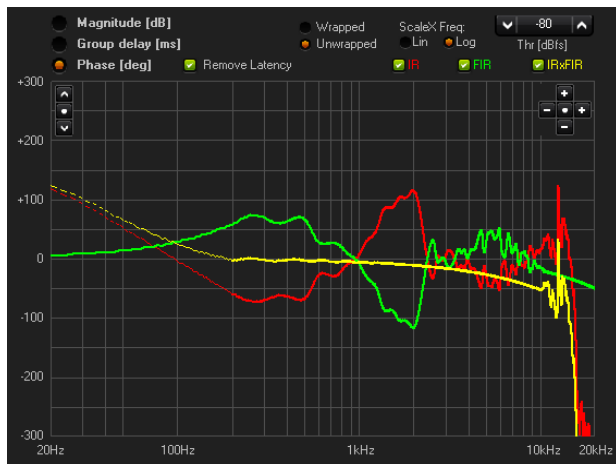
below the plot with wrapped phase, with Log and Lin scale frequencies and without latency:
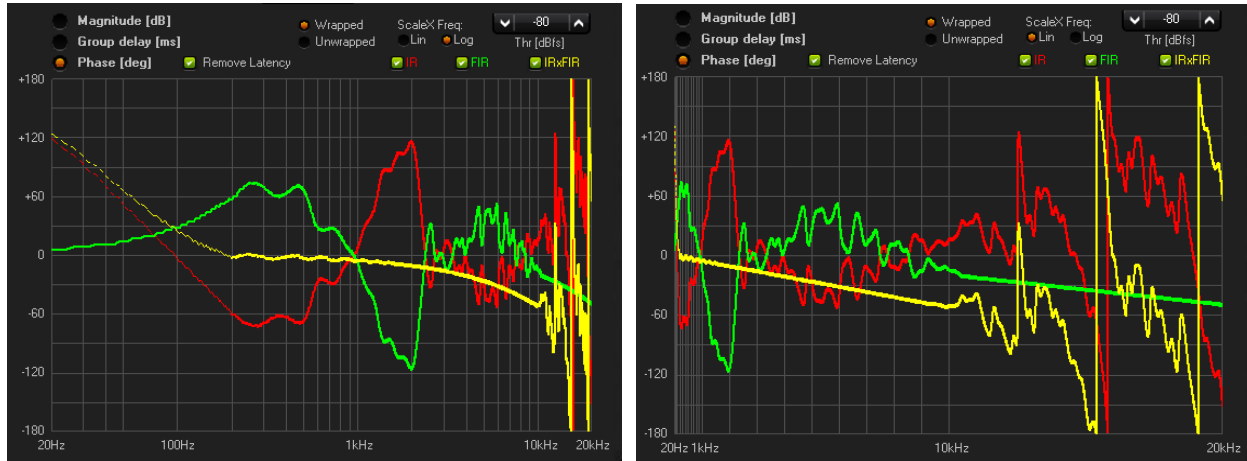
below the plot with wrapped phase, with Log scale frequencies and with latency:
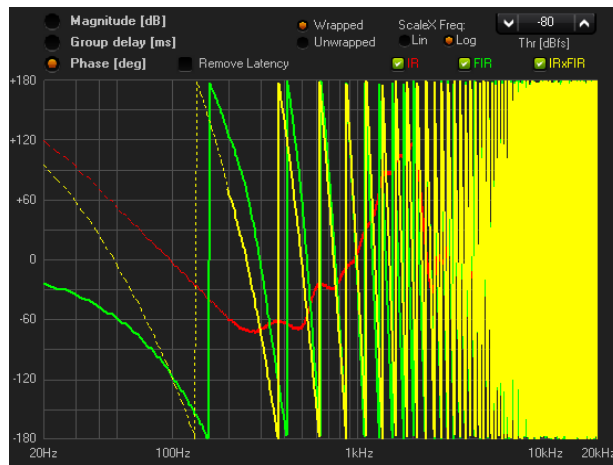
below the plot with unwrapped phase, with Log and Lin scale frequencies and without latency (scaleY: from -300 Deg to 300 Deg, used the Zoom and Shift)

below the plot with unwrapped phase, with Linear scale frequencies and without latency
(scaleY: from -2000 Deg to 1000 Deg, scaleX: from 1kHz to 2kHz, used the Zoom and Shift)

below the plot with unwrapped phase, Log scale frequencies and without latency
(scaleY: from -2000 Deg to 1000 Deg, scaleX: from 100Hz to 2kHz, used the Zoom and Shift)