

Ovation

CINEMA SOUND PROCESSOR



Reference User's manual

Rev: February 2018

IMPORTANT SAFETY INSTRUCTIONS:

- Read these instructions. Keep these instructions.
- Heed all warnings. Follow all instructions.
- Unauthorised modification of this equipment is expressly forbidden by Trinnov Audio.
- Using the unit in the following locations can result in a malfunction:
 - ✓ In direct sunlight,
 - ✓ Locations of extreme temperature or humidity,
 - ✓ Locations of extreme temperature or humidity,
 - ✓ Locations of excessive vibration,
 - ✓ Locations of excessive vibration.
- Condensation can form on the inside of an amplifier if it is moved from a cold environment to a warmer location. Before switching the unit on, it is recommended that the unit be allowed to reach room temperature.
- Clean only with a dry cloth. Do not use liquid cleaners such as benzene or thinner, cleaning compounds or flammable polishes.
- Do not block any ventilation openings. Be careful not to let metal objects get into the equipment.
- Install in accordance with the manufacturer's instructions.
- Operating temperature: 0°C to 40°C. Relative operating humidity (without condensation): 20% to 80%.
- Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
- Always replace damaged fuses with the correct rating and type.
- Unplug this apparatus during lightning storms or when unused for long periods of time.
- Refer all servicing to qualified service personnel. Do not open the equipment case.
- Use only the supplied power cord which is compatible with the mains voltage supply in your area. Never break off the earth (ground) pin on the power supply cord.

TO COMPLETELY DISCONNECT THIS APPARATUS FROM THE AC MAINS, DISCONNECT THE POWER SUPPLY CORD PLUG FROM THE AC RECEPTACLE.



This symbol is intended to alert the user to the presence of uninsulated “dangerous voltage” within the product’s enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons



This symbol is intended to alert the user to the presence of important operation and maintenance (servicing) instructions in the literature accompanying the appliance

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1 Introduction

Thank you for choosing a Trinnov digital audio processor.

This User Guide is organized in the following chapters:

- **Introduction:** describes the main features of the Optimizer,
- **Getting Started:** provides a step-by-step guide to set up your system and configure the metering solution,
- **Hardware Guide:** describes the audio interfaces used in the Ovation, the 3D measurement microphone, the different remote options and update/support procedure,
- **Startup Options,**
- **System Software Guide:** describes all the pages of the Processor module, the core software and the loudspeaker/room optimization module,
- **Appendix.**

1.1 Main Features of the OVATION

The Ovation provides all the functionalities expected from a high quality and advanced digital audio processor:

- routing and level settings of inputs and outputs
- word clock input and output
- remote control options via Ethernet network
- Presets/Profiles save, backup and restore
- peak level metering of inputs and outputs
- calibrated, global gain/volume adjustments
- monitoring controller

This allows to integrate the Ovation into any commercial cinema environment and to meet the highest standards for sound quality and system flexibility and reliability.

1.1.1 Equalization

The Ovation is very useful including the Trinnov advanced technology **Optimizer** as it offers a comprehensive set of tools to fine-tune the results of the automated correction. This is usually done by using a separate acoustic measurements system to measure the results of the automatic optimization and improve them.

Two different tools are offered:

- State-of-the-art FIR EQ: based on Finite Impulse Response filters, the Optimizer's FIR EQ allows for accurate equalization without introducing additional phase problems.
- 1/3-octave EQ: 31 band Graphic Equalizers are also provided in order to support established methodologies and standards.

1.1.2 Active Crossovers

The Ovation features 2-ways, 3-ways and 4-ways active crossovers. Depending on the chosen audio interface, these crossovers may be used on up to 24 output channels. This makes Ovation a comprehensive equalization and affordable crossover solution for cinema sound systems.

1.2 Main Features of the Optimizer module

The Optimizer Runtime and the Optimizer Toolbox are complementary. As a whole, we simply call it the Optimizer. Its main features are as follows.

1.2.1 Level and Time Alignment

Based on its own acoustic measurements, the Optimizer automatically aligns the relative levels of each loudspeaker, and applies delays to time-align the system. This can be disabled in the *Optimizer Settings/Runtime* page, and may be configured separately for the front and surround loudspeakers in the *Optimizer Settings/Settings/Advanced Settings* page.

The Optimizer also levels and delays of multi-amps speakers allowing to have the best alignment of speaker ways.

The Optimizer also calibrates the absolute level.

1.2.2 Automatic Optimization

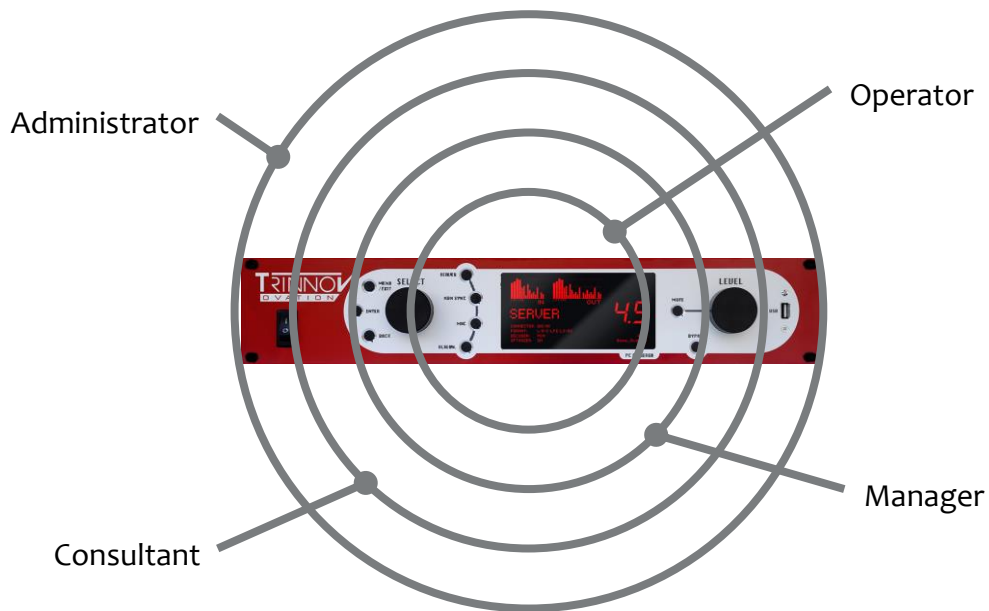
The Optimizer uses state-of-the art time-frequency algorithms to analyze calibration measurements and uses specific methods to compensate for direct sound, first reflections, late reflections/reverberation and room modes. All the subtlety of the Optimizer resides in knowing which defects may be corrected without creating additional problems.

- **Improved Phase Response:** the Optimizer corrects the frequency response of the speakers, both in amplitude and phase. This means that the Optimizer corrects both the tonal balance to obtain a neutral timbre for every speaker and it also works in the time domain to achieve a high resolution stereophonic image with well-focused phantom sources.
- **Target Curves:** the Optimizer automatically defines the filters that will achieve the required frequency response defined by a target curve. This is particularly useful in order to comply with the X-Curve SMPTE and ISO standards. Phase and Group Delay targets may also be defined, making the Optimizer a unique tool for sound system designers.
- **Fine-tuning options:** the Optimizer provides over 12 different parameters, such as maximum boost/cut, to customize the behaviour of the room correction algorithms. This opens many possibilities for fine-tuning the sound according to loudspeaker capabilities and listening tests.

1.3 Access levels

There is 4 access levels in the Ovation:

- **Operator:** default level when you power-on the machine. Normal use level,
- **Manager:** give access to Profiles configuration, need Login and password.
- **Consultant:** give access to global settings of the processor and Eqs on input and output channels. Need Login and password.
- **Administrator:** highest access level, give access to all parameters of the machine and calibration procedure. Level securized with Login and password.



Default Login and password:

- Manager level: login = advanced1, password = advanced1,
- Consultant level: login = advanced2, password = advanced2,
- Administrator level: login = fulladmin, password =fulladmin.



Be careful with processor levels access and passwords. We recommend to change passwords for new processors and keep the new ones confidential. You cannot modify Login.

NOTE: you always must type Login in lowercase. If you have passwords with upper and lowercase, you must respect the character's case.

2 Getting Started

2.1 Power on and Shutdown

2.1.1 Power on

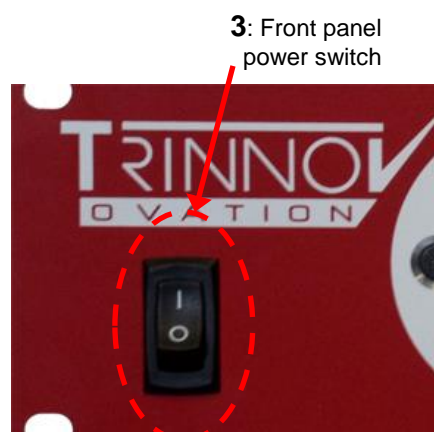
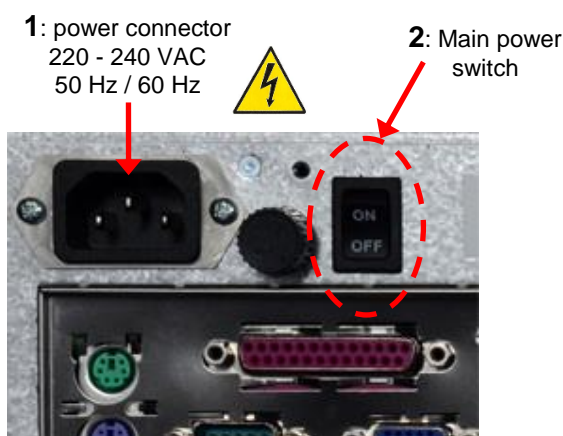


Important Note: don't forget that the speakers/amplifiers should always be powered up last.

Before connecting the power cord (1) to the electric installation, make sure the rear panel Main Power switch is « **OFF** » position and the front panel PC power switch is « **O** » position.

The rear panel Main power switch (2) always needs to be pressed **first** to supply the apparatus with electricity.

The front panel power switch (3) shall then be used to start the apparatus. It should illuminate the PC powered led **after a few seconds**. The back panel power switch always needs to be pressed first to supply the apparatus with electricity.



During the boot sequence of the Ovation, the following page appears on user's screen:



The boot sequence ended, the operating page appears on user's screen:



NOTE: It is possible to set the screen's contrast by turning the 'Level' knob while pushing simultaneously on the 'Back' button.

2.1.2 Shutdown

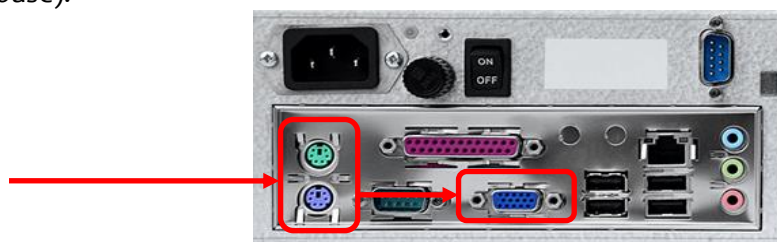
Ovation requires pushing the Front panel power switch once to initiate normal shut down. It is recommended to wait 15 to 30s between firstly switching off the Front panel power switch and then the Main power switch on the rear panel.

2.2 User Interface

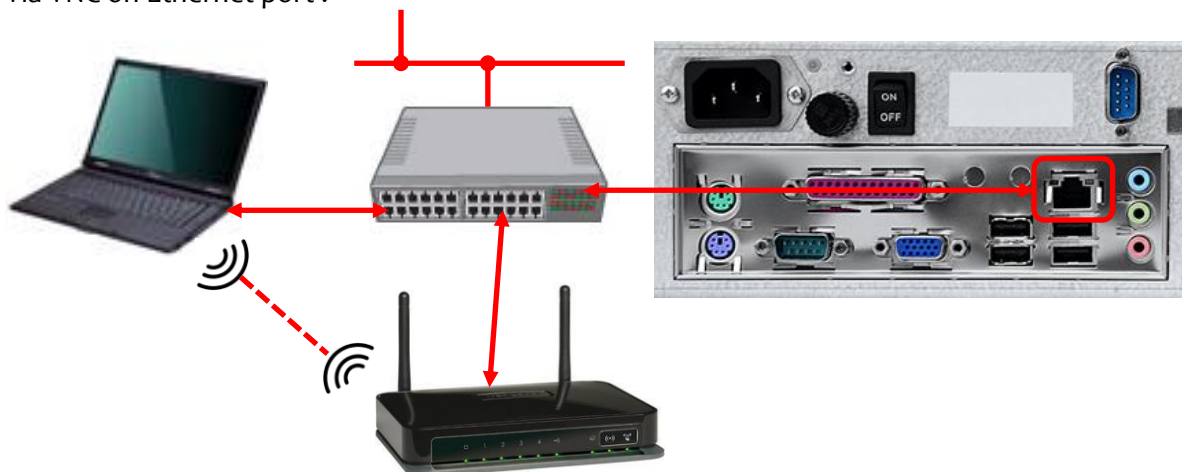
2.2.1 Connection to the Ovation

There are two ways to interface Ovation processor with users:

- KVM (Keyboard/Vidéo/Mouse):



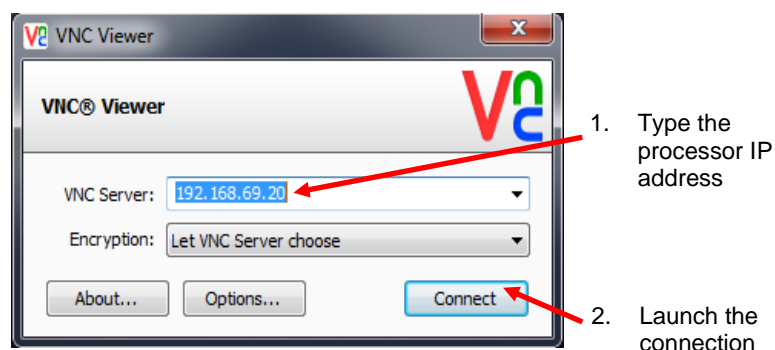
- Via VNC on Ethernet port :



These 2 possibilities can be combined and show the same views.

VNC procedure:

After installing on your laptop, launch VNC viewer (for example VNC-5.0.3-Windows.exe in the screenshots below):

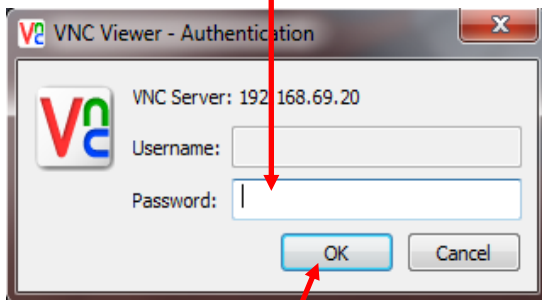


Confirm the connection by clicking on “Continue”:



Type the processor serial number on 6 digits: 000325 for the Ovation 325 for example. Serial number is indicated on a sticker of the rear panel of the processor.

3. Type the password = processor serial number on 6 digits



4. Click OK

Then you will arrive to the Ovation GUI (General User Interface).

Ovation has a built-in VNC Server that allows you to fully control the processor from any VNC client host device over the network.

In other words, VNC provides full control of the processor from a laptop (PC, Mac or Linux), smartphone or tablet (iOS, Android, Blackberry, Nokia...).

2.2.2 Screenshots

Screenshots of the graphical interface can be stored during operation by pressing the “print screen” key on a keyboard connected to the unit. These files then can be downloaded to a USB stick or through Ethernet.

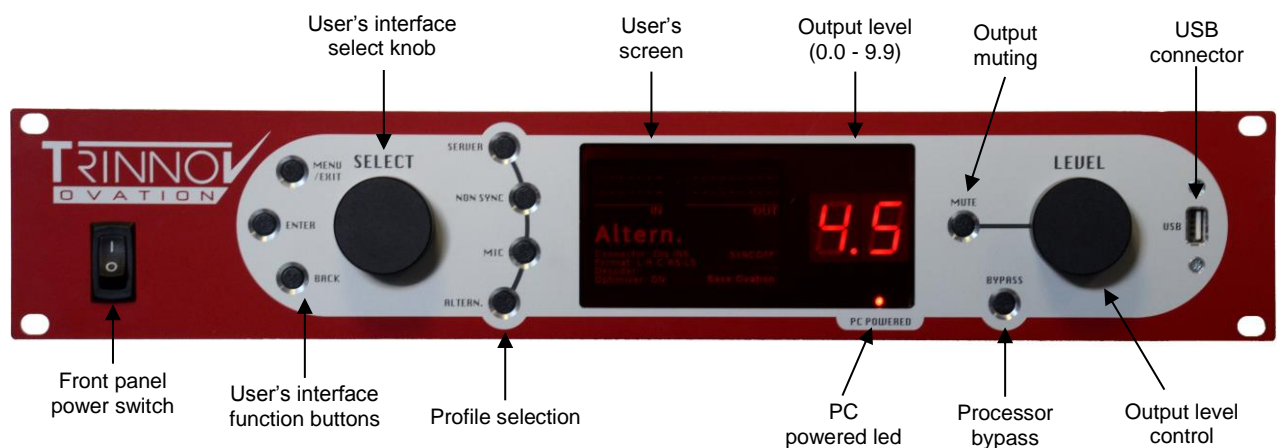
3 Hardware Guide

A wide range of hardware options makes the Ovation easy to integrate in your cinema's environment. The Hardware guide provides a description and technical specifications of the processor.

Ovation is equipped with the high performance Trinnov audio boards:

- The **Trinnov Audio Core (TAC)** is the central audio component inside the processor. It provides routing of the audio between the physical inputs/outputs and the software, activation of the relays when instructed by the user allowing safe shut-down, hardware source selection and clocking up to 192 kHz when available.
- The **Trinnov ADA4** is connected to the TAC, executes AD/DA conversions and offers 4 I/Os analog channels.
- The **Trinnov DAC8** is connected to the TAC, executes D/A conversions and offers 8 analog outputs.

3.1 Front panel



Profile selection: direct access to the 4 first Profiles (Server, Non Sync, Mic, Alternative) with 4 dedicated push buttons. All predefined Profiles are accessible by turning the 'SELECT' knob and confirm with 'ENTER' button.

User's interface buttons: access to the interface pages or back to the operating page with 'MENU/EXIT' button, value or choice confirmation with 'ENTER', back to the previous page with 'BACK' button.

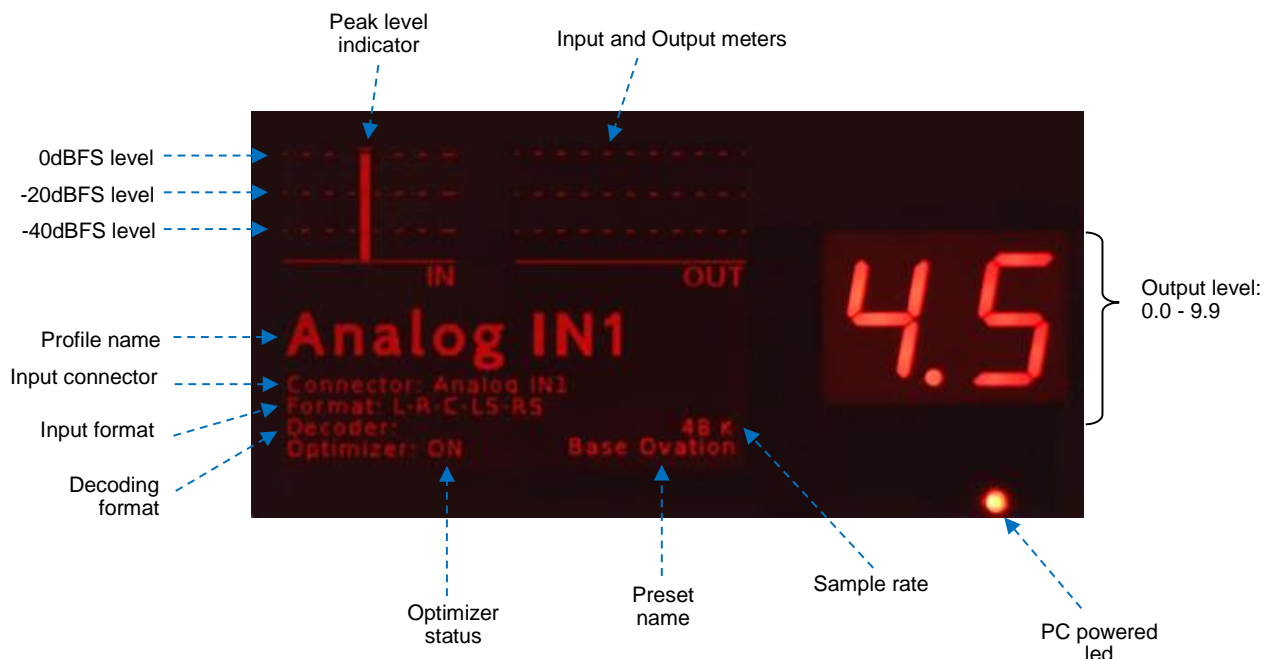
Output level controls: to adjust the global output level of the Ovation, please use the 'LEVEL' knob. To mute all output channels use the 'MUTE' button.

Bypass button can be used to put the Processor in Bypass mode: desactivation of the Optimisation.

USB port can be used to save or load Processor config, presets and Profiles. Can be used in parallel of the USB ports of the rear panel.

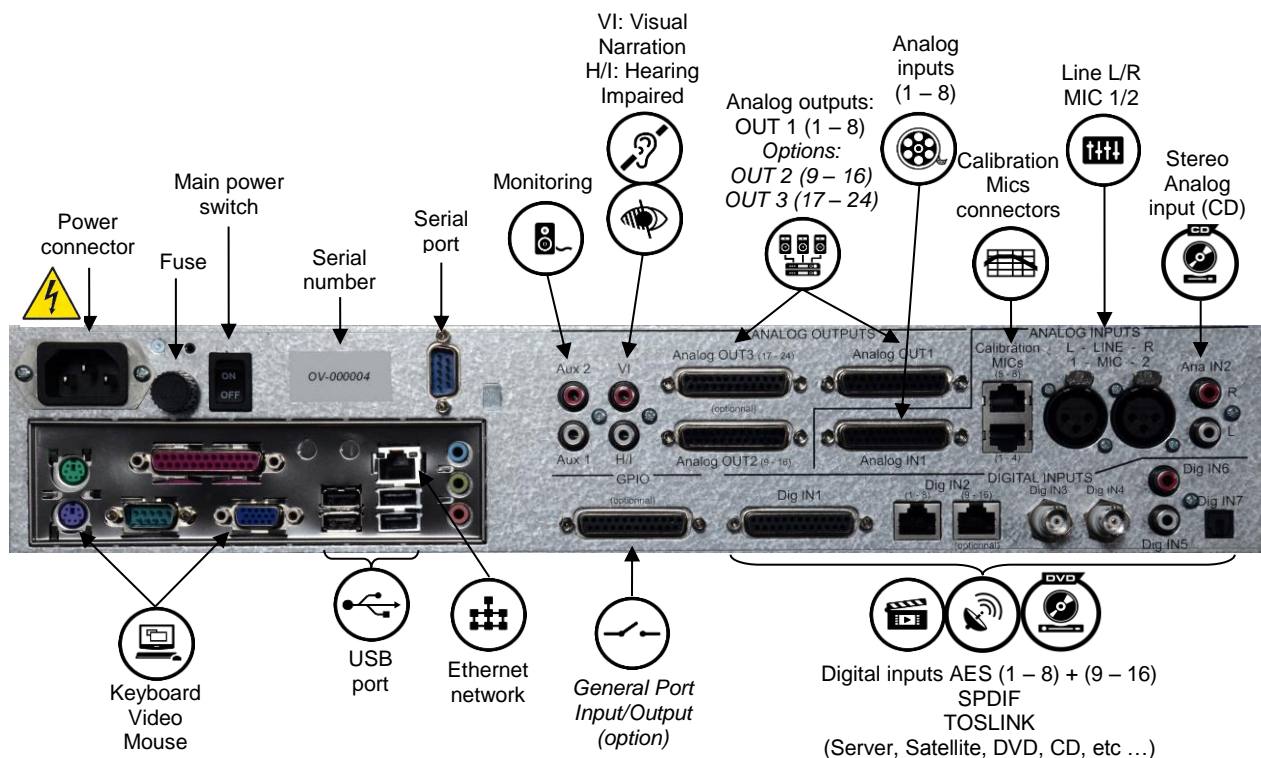
NOTE: it is recommended to install the processor in the 19" rack with the 4 screws in the 4 holes of the front panel.

3.2 Front Panel screen



NOTE: It is possible to set the screen's contrast by turning the 'Level' knob while pushing simultaneously on the 'Back' button.

3.3 Rear panel.



Digital inputs (sample rate: 44.1, 48 and 96 kHz):

- 8 AES DCI channels extensible to 16 (option) via DB25 connector (Dig IN1),
- 8 AES DCI channels extensible to 16 (option) via 2 x RJ45 connectors (Dig IN2),
- 2 x AES via BNC connector (Dig IN 3, Dig IN 4),
- 2 x SPDIF via RCA connector (Dig IN 5, Dig IN 6),
- 1 x SPDIF TOSLINK (Dig IN 7).

Analog inputs:

- 8 balanced inputs (ANALOG IN 1) via DB-25,
- 8 balanced inputs (Calibration MICs) via 2 x RJ45,
- 2 line level balanced inputs (LINE L-R) via 2 x XLR,
- 2 unbalanced inputs (Ana IN2) via 2 x RCA,
- 2 balanced microphone inputs via 2 x XLR (MIC 1-2) with phantom power.

Analog outputs:

- 1 to 8 balanced channels (Analog OUT1, 1-8) via DB-25,
- Options:
 - 9 to 16 balanced channels (Analog OUT2, 9-16) via DB-25,
 - 17 to 24 balanced channels (Analog OUT3, 17-24) via DB-25,
- Hearing-impaired: 1 unbalanced output via RCA (H/I),
- Visual Narration: 1 unbalanced output via RCA (VI),
- Auxiliary: 2 unbalanced output via 2 x RCA (Aux1, Aux2).

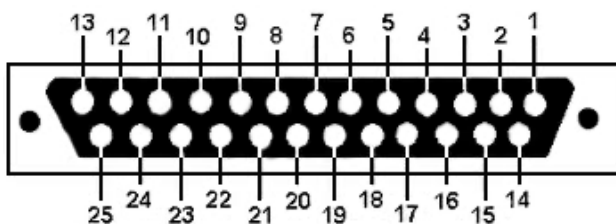
Other devices:

- GPIO port via DB25 (option),
- Ethernet (RJ45),
- USB port,
- KVM (keyboard/Video/Mouse),
- Serial port (DB9),
- Parallel port (DB25).

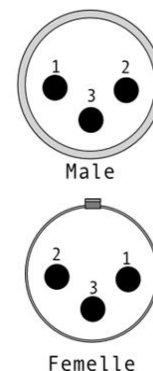
Ovation offer the following audio performance:

- A/D signal-to-noise ratio: 119 dB (A-Weighted)
- D/A signal-to-noise ratio: 118 dB (A-Weighted)
- 24 bits/96k support.
- Clock Recovery: jitter attenuation better than 50dB above 100Hz

3.4 Connectivity.



XLR wiring:
1: Ground
2: Hot (+)
3: Cold (-)



Analog DB25 connectivity for Analog In1, Analog Out1, Out2 and Out3 (TASCAM scheme):

Pin	Signal Description	Pin	Signal Description
1	Channel 8 hot (+)	14	Channel 8 cold (-)
2	Channel 8 ground	15	Channel 7 hot (+)
3	Channel 7 cold (-)	16	Channel 7 ground
4	Channel 6 hot (+)	17	Channel 6 cold (-)
5	Channel 6 ground	18	Channel 5 hot (+)
6	Channel 5 cold (-)	19	Channel 5 ground
7	Channel 4 hot (+)	20	Channel 4 cold (-)
8	Channel 4 ground	21	Channel 3 hot (+)
9	Channel 3 cold (-)	22	Channel 3 ground
10	Channel 2 hot (+)	23	Channel 2 cold (-)
11	Channel 2 ground	24	Channel 1 hot (+)
12	Channel 1 cold (-)	25	Channel 1 ground
13	not connected		

Mono-amplification and 35mm channels mapping:

Channel	Speaker	Connector
1	L	Analog out 1 / Analog in 1
2	R	Analog out 1 / Analog in 1
3	C	Analog out 1 / Analog in 1
4	LFE	Analog out 1 / Analog in 1
5	LS	Analog out 1 / Analog in 1
6	RS	Analog out 1 / Analog in 1
7	LB	Analog out 1 / Analog in 1
8	RB	Analog out 1 / Analog in 1

NOTE:

L = Left
R = Right
C = Center
LFE = Low Frequency Effect
LS = Left Surround
LR = Right Surround
LB = Left Back
RB = Right Back

Bi and tri-amplification channels mapping:

Channel	Speaker	Connector	Channel	Speaker	Connector
1	L low	Analog out 1	1	L high	Analog out 2
2	R low	Analog out 1	2	R high	Analog out 2
3	C low	Analog out 1	3	C high	Analog out 2
4	LFE	Analog out 1	4	HI	Analog out 2
5	LS	Analog out 1	5	L mid	Analog out 2
6	RS	Analog out 1	6	R mid	Analog out 2
7	LB	Analog out 1	7	C mid	Analog out 2
8	RB	Analog out 1	8	VI	Analog out 2

NOTE:

L low = Left low Freq.
R low = Right low Freq.
C low = Center low Freq.
LFE = Low Frequency Effect
LS = Left Surround
LR = Right Surround
LB = Left Back
RB = Right Back

L high = Left high Freq.
R high = Right high Freq.
C high = Center high Freq.
H/I = Hearing Impaired
L mid = Left medium Freq.
R mid = Right medium Freq.
C mid = Center medium Freq.
VI = Visual narration

HI/VI RCA connectors and Analog OUT2 channels 4 and 8 are parallels. DB25 outputs are balanced and RCA are unbalanced.

Digital DB25 connectivity (Dig IN1):

Pin	Signal Description	Pin	Signal Description
1	Channels 15 & 16 hot (+)	14	Channels 15 & 16 cold (-)
2	Channels 15 & 16 ground	15	Channels 13 & 14 hot (+)
3	Channels 13 & 14 cold (-)	16	Channels 13 & 14 ground
4	Channels 11 & 12 hot (+)	17	Channels 11 & 12 cold (-)
5	Channels 11 & 12 ground	18	Channels 9 & 10 hot (+)
6	Channels 9 & 10 cold (-)	19	Channels 9 & 10 ground
7	Channels 7 & 8 hot (+)	20	Channels 7 & 8 cold (-)
8	Channels 7 & 8 ground	21	Channels 5 & 6 hot (+)
9	Channels 5 & 6 cold (-)	22	Channels 5 & 6 ground
10	Channels 3 & 4 hot (+)	23	Channels 3 & 4 cold (-)
11	Channels 3 & 4 ground	24	Channels 1 & 2 hot (+)
12	Channels 1 & 2 cold (-)	25	Channels 1 & 2 ground
13	not connected		

NOTE: the 9 to 16 input channels are optionnals. It needs complementary cards and license to be operationnal.

DB25 GPIO connectivity:

GPIO - DB25 FEMELLE			
PIN	SIGNAL DESCRIPTION	PIN	SIGNAL DESCRIPTION
1	no connection	14	GPI 1 +
2	GPI 1 -	15	GPI 2 +
3	GPI 2 -	16	GPI 3 +
4	GPI 3 -	17	GPI 4 +
5	GPI 4 -	18	GPI 5 +
6	GPI 5 -	19	GPI 6 +
7	GPI 6 -	20	GPI 7 +
8	GPI 7 -	21	GPI 8 +
9	GPI 8 -	22	GPO 1
10	GPO 1	23	GPO 2
11	GPO 2	24	GPO 3
12	GPO 3	25	GPO 4
13	GPO 4		

Each GPI is photo coupled and admit 5 to 24 volts. Each GPO is a relay contact non-polarized.

GPI 1: load profile 1

GPI 2: load profile 2

GPI 3: load profile 3

GPI 4: load profile 6

GPI 5: load profile 7

GPI 6: load profile 8

GPI 7: switching mute (activate or disable alternatively)

GPI 8: switching dim (activate or disable alternatively)

GPO 1: indicates bypass mode

GPO 2: unaffected

GPO 3: unaffected

GPO 4: unaffected

4 Startup Options

4.1 Startup Modes

Momentarily displayed before the access to the GUI
Some options can be activated in the following screen.



- **Double speed mode:** this option does not concerne Ovation units and should not be used.
- **No default preset:** forces the processor to use the **built-in** factory preset at startup, overriding the default preset selected in the presets page.

After selecting these options, either "Audio Mode" or "Demo Mode" must be pushed to validate your choice.

- **Audio mode** is the normal mode
- **Demo mode** shows the functionalities, monitoring and pages without having to connect the unit to an audio source. This mode can simulate a calibration, even without microphone or loudspeakers connected.

4.2 Network Operation

Software updates and support can be performed remotely by Trinnov provided that the processor is connected to the internet and that outgoing connections to port 22 are open. Please note that any update requires advance approval and manual intervention from an engineer at Trinnov Audio's office in Paris.

When the processor is connected to Trinnov's server, the **Network Status** in the *Setup/Network* page will change to "**Connected to Trinnov Audio Server**". For more information on network configuration, please refer to 5.4.8 Setup/Network chapter.

If the processor is connected to a network, but the Network Status is "Local Network OK", it means that the Trinnov Server cannot be reached from your network. Please check the IP, DNS, Gateway and Mask addresses in the processor and what's needed in your network with your administrator.

5 System Software Guide

5.1 OPERATOR level

OPERATOR level is the default level when the processor is powered on. This is the first and basic access level of the Ovation. No Login, no password needed.

After boot sequence of the processor, user's interface shows *Home/Profile* and *presets* page.

5.1.1 Home/Profile and presets



Selection zone of Profile. 18 Profiles are available and configurable. Active Profile is in blue. **Confirmation of Profile choice will be asked.**

Selection zone of one preset in the ten first. Active preset is in blue.



Loading preset doesn't need confirmation. Be sure of your choice before changing preset.

All available Profiles are shown in this page. The active Profile is in blue and indicated in the front panel screen of the processor.

Only the 10 first presets are shown in this page. The Ovation can have a maximum of 29 presets. If the active preset is in the 10 first, it will be indicated in blue in this page. Preset name and number are indicated in the front panel screen.

PROFILE:

Profile includes parameters and all needed options for the room use. For example: "Server 5.1", "Conference", Mics ... Profiles are configurable in *Home/Profile Config* page from MANAGER level and above.

A Profile is always associated to a physical connector of the rear panel. The Profiles can be the same for all rooms and copied from one room to another.

PRESET:

Preset is the result of a room calibration including acoustics measurement and computing with parameters. Once the room is measured, it is possible to change parameters, compute new filtering and save new settings in another preset. This allows to switch from a preset to another, listen to different filtering and make the best choice

In another way, a preset is unique, depends on the room measurement, acoustics and parameters choice and can never be copied from one room to another.

A maximum of 29 presets are available in Ovation.

A preset is always associated to a Profile.

5.1.2 Home/Meters

Inputs and outputs Vu-meters page, dBFS scale.



In case of channel Overload (Peak), the channel label will be in red until clicking on 'Clear' button.

5.1.3 Help/About

The *Help/About* page provides useful information about software and hardware configuration of your processor:



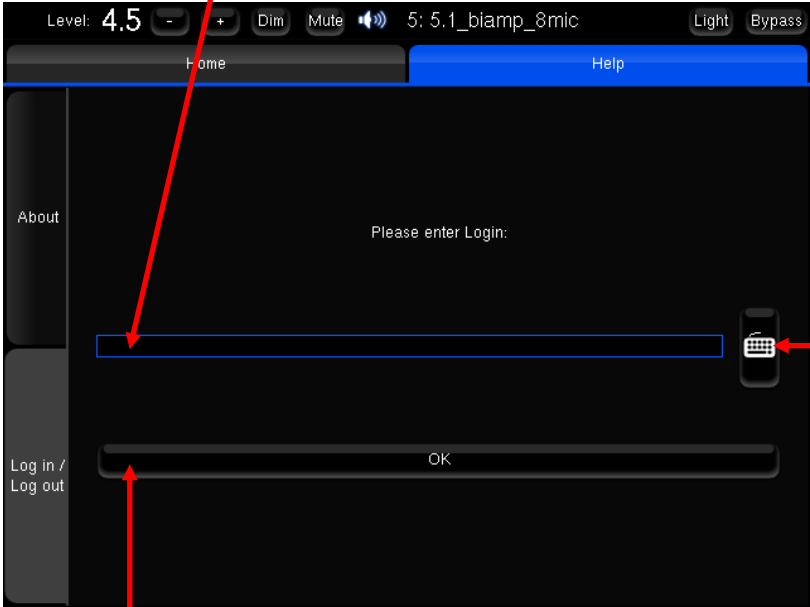
- **Version:** the exact software version installed on your processor,
- **Built:** the date this software version was built,
- **Product ID** (and serial number) of your processor, required as a password when connecting via VNC,
- **Microphone** ID that your processor is configured to be used with.
- **Soundcard:** displays Trinnov Audio Core.
- **Runtime mode:** the current runtime mode of the Optimizer:
 - “read & write” is the normal runtime mode,
 - “read only” is displayed when the Optimizer was started in read only mode to avoid any changes to the presets.
- **License:** the number of channels that can be processed in parallel (this depends on the Ovation config that has been purchased: 8, 16 or 24),
- **IP address (ethernet):** IP address on the network.

5.1.4 Help/Log in / Log out

Login page to access to any securised level. Can only be done at the OPERATOR level.

Logout from any level always bring you back to OPERATOR level.

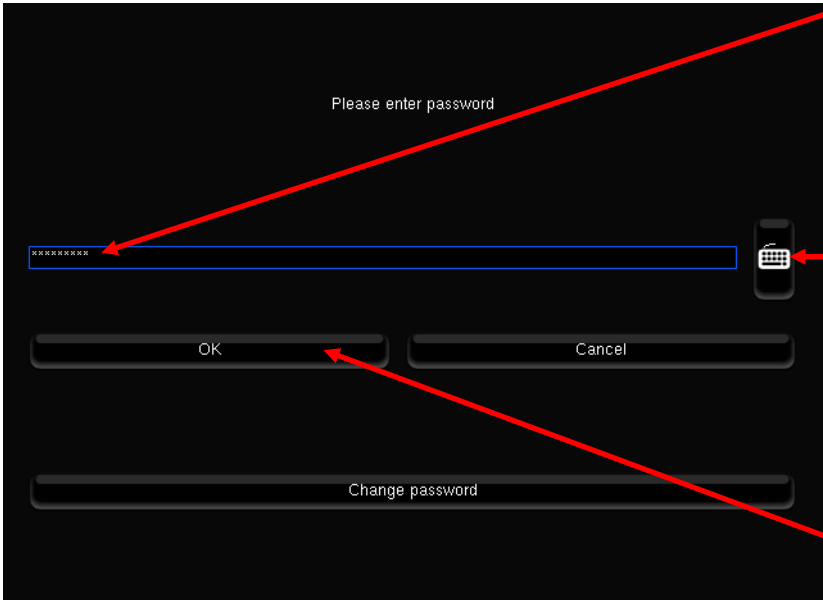
1. Type the level Login: advanced1, advanced2 or fulladmin



If you don't have keyboard, use this button to access to a virtual keyboard to type the Login.

2. Click 'OK'

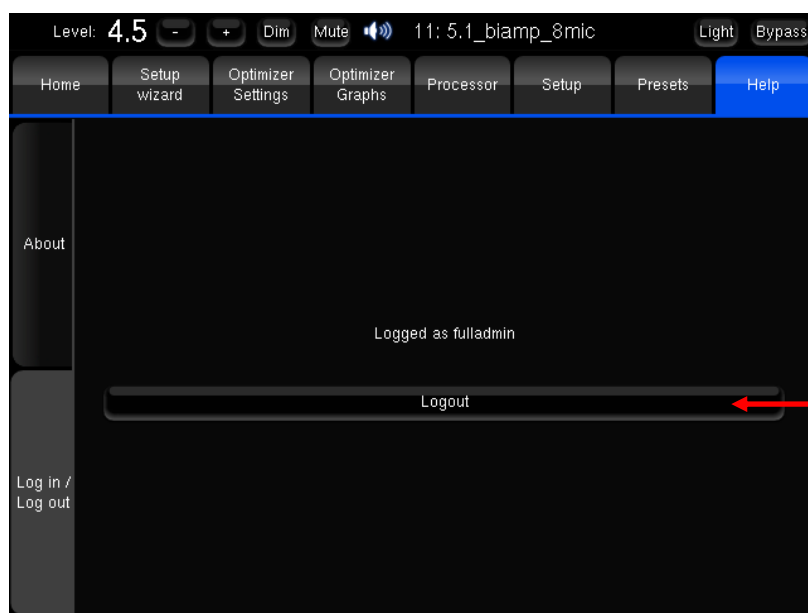
3. Type the default password or the one you have choose.



If you don't have keyboard, use this button to access to a virtual keyboard to type the Login.

4. Click 'OK'

Once logged at the desired level, a confirmation page is shown (ex. Fulladmin level below):

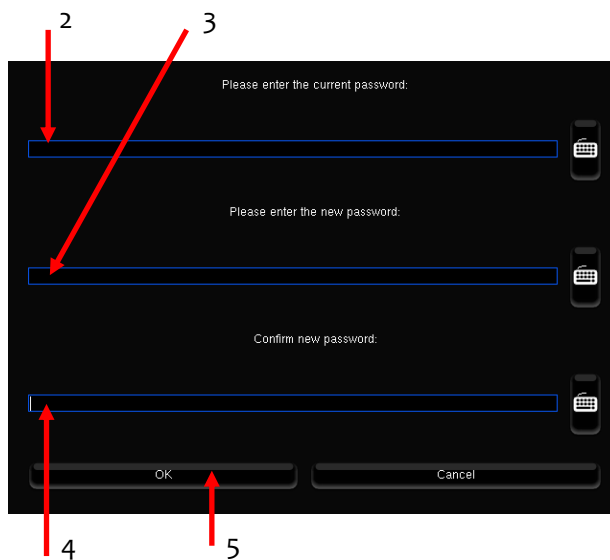
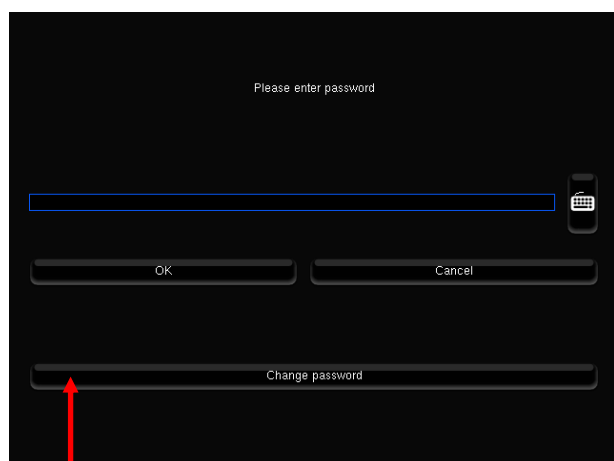


At any moment, you can go back to the OPERATOR level by clicking on « Logout » button.

Changing level password:

Once you've typed the desired level Login (you cannot change Login), don't type the level password, but:

1. click on 'Change password',
2. type the existing password of the level you want to change,
3. type the new password,
4. type again the new password to confirm,
5. click 'OK'.



Be careful with levels passwords. Give the password to someone means that this person could modify all the accessible parameters of the level. Keep the passwords in a secured place.

5.2 MANAGER level

Access Login: advanced1. Default password: advanced1. *Help* page is the same as in Operator level.

5.2.1 Home/Profiles Config

Home page has a new tab: *Profile Config*.

The *Home/Profile Config* tab allows to configure all Profiles with 3 main zones:



Mutual zone

For all Profiles:

- **Power on Volume:** setting of the global outputs volume when processor is powered on. Set the appropriate volume with '-' and '+' buttons. Default value is 4.5 on 0.0 – 9.9 scale.
If the setting is changed, you must save the modification with 'Save' button. If not, the new value won't be memorized.
- **Use last loaded Profile as power-on default Profile:** must be selected if you want the Ovation to start with the last used Profile before the powering off. This is the default setting.
Selecting 'Power-on Default' in a Profile config will unselect this option.
- **Save:** You need to press this button to apply and save the changes.





Be sure to save your new settings before loading another Profile. If not all settings will be lost.

- **Reload** : this button has two functions:

- As indicated: *'If profiles has been uploaded on the machine via the network (FTP), you have to reload them on the processor with button right here'.*
If you use USB stick to load Profiles, the list is automatically reloaded and you don't have to click on 'Reload' button.
- Reload the last Profile. All non saved modifications will be lost.

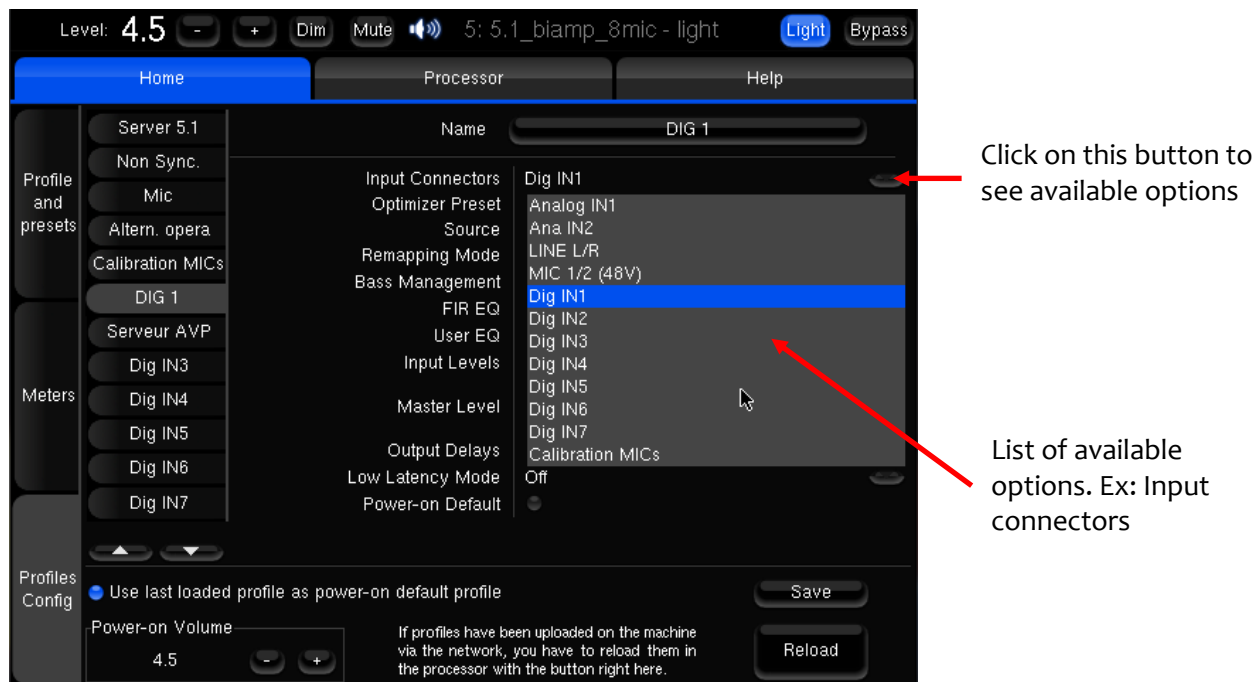
Select zone

Allows to select the Profile you want to edit.

You can use the  (UP) and  (DOWN) buttons to find the Profile you want to configure in the vertical tab on the left side. The Profile button you are editing is grey.

Edit zone

For each parameter, an options list appears by clicking on the button on the right side on the screen:



- **Name:** clicking on the “Name...” button will display a virtual keyboard allowing you to change the name of the profile.
- **Input connector:** used to associate a combination of physical inputs to the profile. As you recall the profile, the audio core will initiate a hardware source selection that will also affect the source routing matrix (**Setup/Sources Routing**).
- **Optimizer preset:** you can link a preset to a profile, thereby associating a hardware I/O routing to the relevant correction filter.



If an empty preset is associated to a Profile, this Profile can be saved but when it will be loaded the following message will appear: « Unable to load config file /var/config/[n°de preset] ».

- **Source:** you can select one of the software source of the preset associated to the profile. If no preset is associated, the only available option will be "As in preset".
- **Remapping Mode:** you can associate a remapping mode to a profile, force it over the current settings or use the remapping mode of the linked preset. This functionality need a calibration with the 3D Trinnov microphone. Not used in the cinema for now.
Default value : 'Off'.
- **Bass Management:** you can switch off the Bass Management or use the config editor over the current settings. You can also select one of the Bass Management option from a specific preset when linked to the profile.
Default value : 'Off'.
- **FIR EQ:** you can select a FIR EQ file or use the FIR EQ of a specific preset when associated to the profile.
- **User EQ:** you can select one of the available User EQ in the processor to be recalled when you hit the profile button.
- **Input Levels:** you can select an input control file or use the input levels of a specific preset when associated to the profile.
- **Master Level:** allows to apply a specific global output level (0.0 – 9.9 scale) when Profile's loaded.
Default value: 'Unchanged'. Typically when Profile changes come via GPIO card.
- **Output Delays:** you can choose whether you want to use the Output Delays of the associated Optimizer preset or not by selecting "As in preset" or "Inhibit".
Default value: 'As in preset'.
- **Low Latency Mode:** allows to reduce processor latency (phase and polarity correction bypassed).
Default value: 'Off'.
- **Power-on Default:** as for presets, you can choose to start the processor with the profile of your choice by selecting this option.

Important Note: for several settings, the following options are available:

- **No change:** if a specific setting is set to No Change, switching to the profile will have no effect on the current parameter.
- **As in preset:** the "As in preset" setting has two behavior:
 - It acts like the no change setting if no preset is associated to the profile: it has no effect on the current parameter.
 - If a preset is associated to the profile, any setting set to "As in preset" will use the preset parameter.

5.3 CONSULTANT level

Login: advanced2. Default password: advanced2.

Help and *Home* pages don't change. A new page *Processor* appears with five new tabs: *Meters*, *Master*, *Inputs*, *Outputs* and *Output Delays*.

As a complement to the Optimizer automatic equalization or as a stand-alone system, the *Processor* page includes a FIR Equalizer and Graphic Equalizers, levels and delays adjustments on every channel.

5.3.1 Processor/Meters

Same as *Home/Meters* page.

5.3.2 Processor/Master

Level and delays adjustments are usually performed as the last step in the calibration process.



- **Master Level:** is the reference level used by the processor for all presets. It affects both displayed and effective Level but it is not stored with presets. *Not stored in preset.*
- **Relative Level:** may be used to match different presets subjective levels since it can be saved. Therefore you could for instance carry out proper A/B comparisons between different settings such as Optimization On/Off. It does not affect the displayed level (on the top left corner of the screen). *Stored in preset.*
- **Level disp. offset:** its main utility is to display the level in dB SPL you would measure in the room with a -20 dBFS pink noise feeding the outputs of the Optimizer with a 0dB master level (102,6 dB in our

example). To do so, the system must have been calibrated and the dBC button selected. The Legacy option displays the level according to cinema standard 0.0 – 9.9 scale.

Level display offset affects displayed level only and has no impact on the level itself. It cannot be stored in presets. Without the dBC button pressed, it has the same features as Output SPL adjust except it cannot be saved. Once the dBC option activated, level adjustment is disabled.

- **Output SPL adjust:** affects both displayed and effective Level. Used along with Level display offset, it allows you for instance to set a reference level corresponding to a measured 85 dB SPL and to store it into a preset. It is a typical cinema use.
- **Level offset when correction is on:** this level offset will be applied when the Acoustic Correction is activated.
- **Level offset when remapping is on:** this level offset will be applied when 2D or 3D Remapping is activated *Optimizer Settings/Settings/Main Settings* page, in the ‘*Speaker Position Remapping*’ option. Not used in cinéma for now.
- **Master delay:** can be used to apply an additional delay to all channels and all presets. *Not stored in preset.*
- **Relative delay:** can be used to modify the delay of a preset. *Stored in preset.*

Processing latency informations:

Latency				
Processing latency:	32.5 ms	11.15 m	0.78 frames	
Master + relative delay (user-defined):	0.0 ms	0.00 m	0.00 frames	
In-out delay (processing + user-defined):	32.5 ms	11.15 m	0.78 frames	
Acoustic delay (loudspeaker distance):	40.0 ms	13.71 m	0.96 frames	
Total delay at measurement point:	72.5 ms	24.86 m	1.74 frames	
with frame rate:	<div> <div>24 fps</div> <div>25 fps</div> <div>30 fps</div> </div>			

- **Processing Latency:** corresponds to the latency of the processor algorithms. It can be modified by changing the Optimizer mode from “Normal mode” to “Low Latency Mode” in the *Optimizer Settings/Runtime* page, or Optimize setting (Amplitude + Phase has higher latency than Amplitude only) in the *Optimizer Settings/SettingMain Settings* page, or the Audio Buffer Size (in *Setup/Clock* page). This last possibility will only be active after a processor reboot.
- **Master + Relative Delay** is the sum of the master and relative delays of the *Processor/Master* page.
- **In-out Delay** is the sum of Processing Latency and User-defined delays. For the furthest speaker, it corresponds to the system’s delay from input to output.
- **Acoustic Delay:** corresponds to the distance of the furthest speaker to the reference measurement point (#1). When Time Alignment is activated, all the other speakers will be time-aligned to the furthest speaker.
This delay cannot be modified, it depends on the room size and architecture.
- **Total delay at measurement point** is the delay from one input until the sound reaches the measurement point.

NOTE: these delays (in ms) are converted in distance (m) and frames considering the speed of sound at 343m/s at 20°C.

- **Frame rate:** you can chose the frame rate in 'fps' (frames per second) to convert delays in frames.

5.3.3 Processor/Inputs

Processor/Inputs page can be considered as a kind of mixing console.
Most of the settings are stored in preset.



Channel order will be the same as defined in Setup/Sources page with the option 'Next order' and stored in the preset. LFE input will always be the last one.

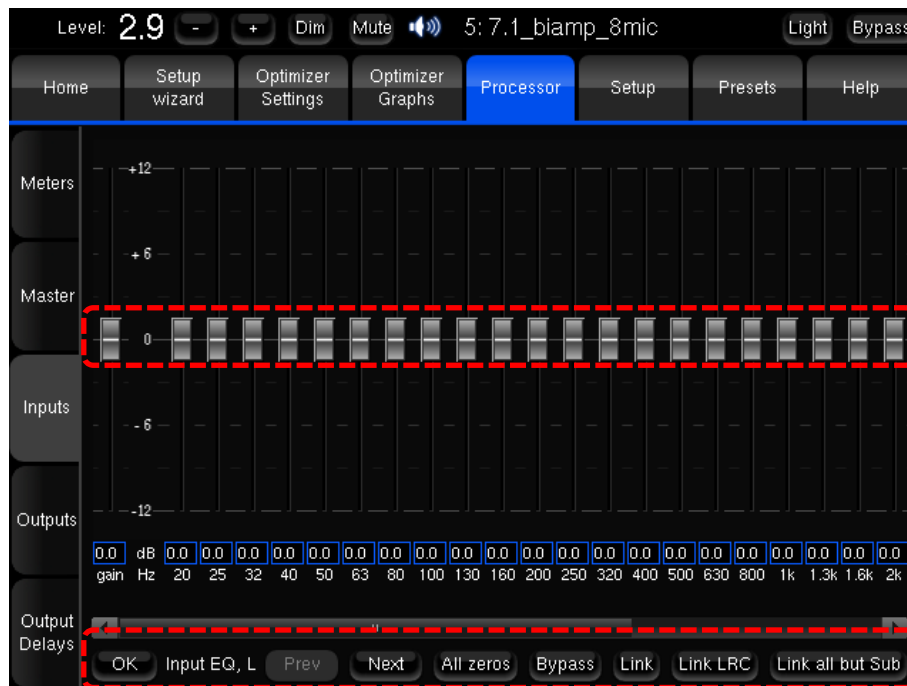


- **+1 / +0.1 / -1 / -0.1:** input level settings (in dB). Stored in Input Control file.
- **Solo:** allows to select the channel in an exhaustive way. Several solo can be combined. Stored in Input Control file.
- **Mute:** allows to mute input signal. Several mute can be combined. Stored in Input Control file.
- **Sine:** generates 1 kHz signal. Several sine can be combined. Not stored in Input Control file.
- **Pink noise:** generates Pink noise signal. Several can be combined. Not stored in Input Control file.
- **Input EQ:** opens the 1/3 octave « Input EQ » page of the channel. Stored in Input Control file.

Input EQ Page:

In addition to manually setting levels, the Processor/Inputs page includes 31-band, 1/3 octave Graphic Equalizers that allow for manual equalization of each input:

- Each Freq. is ajustable from - 12 to +12 dB.
- A global gain concerning all freqs is ajustable from - 12 to +12 dB.



- **OK:** allows to go back to the global Processor/Input page,
- **Prev:** allows to go to the Input EQ of previous channel,
- **Next:** allows to go to the Input EQ of next channel,
- **All zeros:** set all freqs to 0. To avoid bad manipulation, a confirmation will be asked.
- **Bypass:** allows to bypass the EQ to make a quick listening with/without.
- **Link:** allows to link several channels to make same modifications in one operation.
- **Link LRC:** allows to link the screen speakers L, R and C to be modified simultaneously.
- **Link all but Sub:** allows to link all channels but Sub.

- **Link:** allows to link several channels before going in the Input EQ page. *Stored in Input Control file.*
- **BM trim:** Bass Management (BM) ajustement gain in the Subwoofer for the selected channel.
Only possible when Bass Management functionality has been activated in Setup/Speakers page.
Stored in Input Control file.

Down the page, the active « Input Control » file name is indicated. If empty, no Input Control file is used.

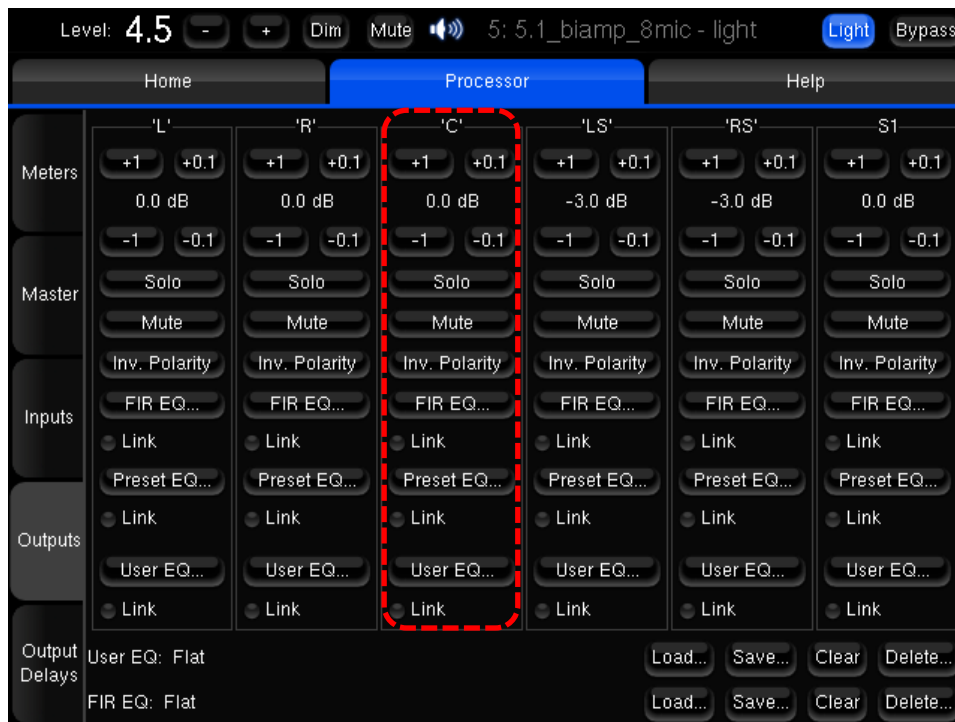
- **Load:** allows to load a Input Control file in the proposed list,
- **Save:** allows to save the current settings in a new or existing Input Control file,
- **Clear:** reset current settings,
- **Delete:** definitively delete the active Input Control file. Confirmation will be asked.

5.3.4 Processor/Outputs

Processor/Outputs page can be considered as a kind of mixing console.
Most of the settings are stored in preset.



Channel order will be the same as defined in Setup/Sources page with the option 'Next order' and stored in the preset. S1 will always be the last one.



- +1 / +0.1 / -1 / -0.1: settings of the input level (in dB). *Stored in Input Control file.*
- Solo: allows to select the channel in an exhaustive way. Several solo can be combined. *Stored in preset.*
- Mute: allows to mute input signal. Several mute can be combined. *Stored in preset.*
- Inv. Polarity: allows to reverse polarity of an output channel. *Stored in preset.*
- FIR EQ : For every speaker in the room, a FIR EQ can be defined in terms of:
 - Amplitude only
 - Amplitude and Phase
 - Amplitude and Group Delay

Please note: by default, since the length of the FIR filter is 20ms (as defined in *Optimizer Settings/Settings/Advanced Settings* page, “FIR & IIR settings” parameter), the FIR EQ has a constant resolution of 50Hz. This implies that it has very low resolution at low frequencies; it should therefore not be used to work on specific frequencies in the low range (under 200Hz). Instead, it should be used to change the tonal balance as a whole.
Stored in FIR EQ file.

- Link: allows to link several channels before going in the FIR EQ, preset EQ or User EQ pages. *Stored in preset.*

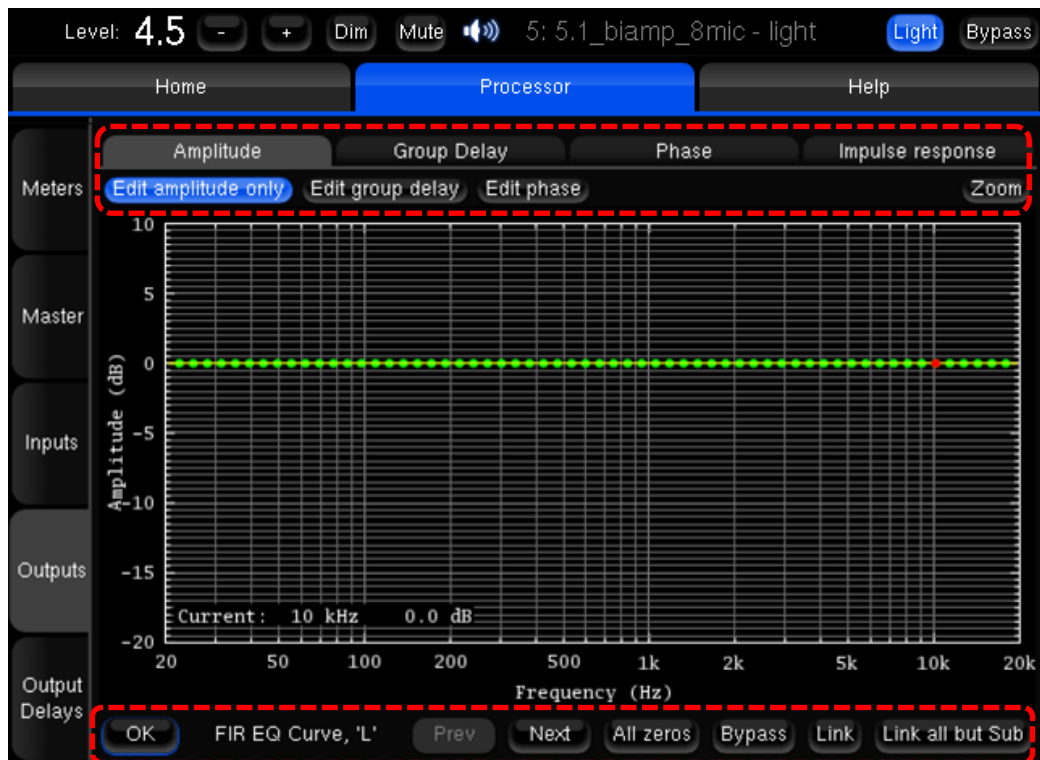
FIR EQ page:

The required curve can easily be edited with the arrows of the keyboard, or mouse:

- The **green dots** correspond to the required values for each frequency,
- The **yellow line** displays the expected result, taking into account the behaviour of the filter.

The FIR EQ will be saved and reloaded with the whole preset, through the usual presets page. Press the Close button to apply the changes and get back to the *Processor/Outputs* tab.

IMPORTANT: Please keep in mind that the changes will not be applied until you press the OK button.



4 horizontal tabs are available: *Amplitude*, *Group delay*, *Phase* and *Impulse response*. You can edit:

- Just Amplitude (« Edit amplitude only »),
or
- Amplitude and Group Delay (« Edit group delay »),
or
- Amplitude and Phase (« Edit phase »).

The choice of the curves to be edited can only be done in *Amplitude* tab.

Available Freqs (Hz): 20, 22, 25, 28, 31, 35, 39, 44, 49, 55, 62, 70, 78, 87, 98, 110, 120, 140, 150, 170, 190, 220, 240, 270, 300, 340, 380, 430, 480, 530, 600, 670, 750, 840, 940, 1.1k, 1.2k, 1.3k, 1.5k, 1.7k, 1.9k, 2.1k, 2.3k, 2.6k, 2.9k, 3.3k, 3.7k, 4.1k, 4.6k, 5.1k, 5.8k, 6.4k, 7.2k, 8.1k, 9.1k, 10k, 11k, 13k, 14k, 16k, 18k and 20kHz.

- Preset EQ: opens the “Preset EQ” of the channel. Settings are identical to “Input EQ”. Stored in preset.
- User EQ: opens the “User EQ” of the channel. Settings are identical to “Input EQ”. Stored in User EQ file.

Down the page, active “User EQ” and “FIR EQ” file names are indicated. If empty, no files used.

Load, Save, Clear, Delete: same functionality as in “Input EQ” page.

5.3.5 Processor/Output Delays

Processor/Output Delays page allows to set delays on output channels. Channels order's the same as Processor/Output page.



- +1 / +0.1 / -1 / -0.1: fine tuning of delay value. Stored in preset.

5.4 FULLADMIN level

5.4.1 Setup/Sources



This page is used to declare the number of sources and their respective format. For each source, the channel order can also be specified if necessary. Channel routing can then be done for each source in the input matrix (*Setup/Sources routing* page).

The input format information could notably be used for remapping as it gives the reference positions of the loudspeakers. The channel order is used for metering display and Input/Output Processor pages. Remapping isn't used in cinema for now.

- **Input format:** sets the input format: mono, stereo, 2 x stereo, 3 x stereo, 2/2, 3/1, 5.1 ITU, 5.1 SMPTE, 6.1 ITU, 6.1 SMPTE, 7.1, 3/4 SMPTE, 5/2 SDDS, 8 channels, 3/6 SMPTE, 12 channels or 16 channels.
- **Number of LFE:** sets the number of LFE signals feeding the Optimizer.

Please note: the difference between the “ITU” and “SMPTE” formats is the reference positions for the loudspeakers. “ITU” refers to the ITU R-775-1 specification for broadcast control rooms, while “SMPTE” refers to the SMPTE 202M standard for film theatres and mixing stages.

- **Channel order:** Sets the internal sort order of the multichannel signal. It is typically the sort order of the connections between your source and the Optimizer (if sources routing is straight). As an example, the default channel order for a 3/4 SMPTE DCI compliant source is:
L, R, C, LS, RS, LFE
If LFE channel is declared, the LFE signal is always placed in the *last position* after the other channels and is referred to as “LFE” in the *Setup/Sources Routing* page.
- **Listen:** used to select the source that is monitored.
- **Hide:** unused.
- **Name:** clicking on the “Name...” button will display a virtual keyboard allowing to change the source's name.
- **Remove:** used to remove a source. Removal cannot be cancelled.
- **Add:** used to add a new source.

5.4.2 Setup/Speaker



- **Loudspeaker Number:** sets the number of non LFE calibrated channels of your sound system.
NOTE: changing this setting requires re-calibration.
- **Subwoofer number:** sets the number of calibrated channels which will play the LFE signal and optional bass signal from other channels if bass management is activated. Subwoofers are labeled **S1, S2...** in the Routing and Meters views and are placed after the other loudspeakers. If you use several subwoofers, the system will create copies of the first input LFE signal.

NOTE:

- The maximum number of channels which can be processed simultaneously is determined by your license (“**loudspeaker number**” + “**Subwoofer number**”).
- If “Bass management” is activated, Subwoofers receive low frequencies from the main channels below the chosen Crossover Frequency.

5.4.2.1 Bass Management

The Optimizer is designed to support established Bass Management settings used in the Broadcast, Film and Music industries, as defined in standards such as EBU Tech 3276-E, AES TD 1001.1.01-10, ITU R-775-1, SMPTE 202M and ISO 2969 (curve X) and SMPTE 222M.

Important notes:



- The Optimizer always aligns the levels of all the loudspeakers, including the subwoofers.
- Regarding the option **+10dB on LFE input (S1)**:
 - In a professional environment, this option should be used as required in respect with the recommended calibration level of the subwoofer and in order to achieve the best gain structure.

- The LFE channel is recorded with a level offset of -10 dB. This offset has to be compensated for in the reproduction system. This option should therefore be used only when no other equipment within this chain applies this gain.
- This setting is independent of Bass Management on/off.

NOTE:

- The +10dB on LFE input (S1) is not affected by the bypass mode.
- Implication: if a Bypass mode is required, make a "Bypass" preset with appropriate LFE setting/level, rather than using the "Bypass" switch located in upper right corner.
- Recommendation: for optimized gain structure, set LFE amp at +10dB SPL relative to other channels before calibration. Activate "+10dB on LFE Input" in the Optimized preset and deactivate it in the "bypass" preset.

The bass management modes are as follows:

- **Off:** this means that no bass management is performed. Main speaker will reproduce the low frequency components of their respective channels, and the Subwoofer(s) will only reproduce the LFE channel.
- **On:** on each of the main channels the low frequencies are filtered at the cross-over frequency, summed with the LFE and sent to the Subwoofer(s). Please note: as defined in the industry standards, the LFE channel is not filtered. It is therefore sent full range to the Subwoofers.
- **Mono:** this is the standard bass management mode, the same signal is sent to all the subwoofers. This is the only possible mode when Subwoofer number is set to "1".
- **Stereo:** this bass management feature maintains stereo bass: the low frequencies from the Left channels (L and Ls) are sent to the first Subwoofer (S1) and the low frequencies from the Right channels (R and RS) are sent to the second Subwoofer (S2). The low frequencies from the Center channel are distributed equally between both subwoofers. This mode needs to declare a minimum of 2 Subwoofers.
- **Send LFE to L+R:** this bass management feature is useful when no subwoofer is available to monitor the LFE channel. The processor will equally distribute the LFE channel between the L and R loudspeakers. Particular care should be taken to make sure that the monitors can handle the additional power required to reproduce the LFE.
Important Note: particularly on ported bass drivers, it is imperative to set an appropriate hi pass filter (Target Curve or in Advanced Settings), so as to not damage the woofer from over excursion.

NOTE:

- Bass management can be set up and activated before or after calibration. It does not require computation and its effect is audible instantly. Bass management filtering is not represented in the Optimizer Graphs.
- Bass management uses 4th order Butterworth filters.
- The button **Use config editor** is used for backward compatibility with previous versions of the software, in which the bass management settings are defined in the XML file. If you don't know XML perfectly, please don't use the config editor.

5.4.2.2 Delay Lines

Delay lines are provided in order to delay additional audio monitoring systems with the outputs of the Ovation without being affected by the Optimization.

Delay lines typically are used in HI/VI treatment to align audio and video.

Once created in the Setup/Speakers page, delay lines can be set up in Setup/Sources Routing and Setup/Speakers Routing grids.

For example, if you create 2 delay lines to manage HI and VI input signals, 2 additional lines are consequently created in the Sources Routing grid and named HI and VI. First line is HI, second one is VI and you can affect these lines to HI and VI inputs on the grid.

In the Speakers Routing page, 2 additional lines are also created and can be affected to any physical output channel. **Note** that outputs 4 and 8 of the Analog OUT2 dB25 connector are parallels to HI and VI RCAs.

5.4.3 Setup/Active Xovers



5.4.3.1 Functionality

The active crossovers included in the Optimizer provide the following functionality:

- 2 ways, 3 ways and 4 ways crossovers,
- Constant-directivity horn correction (3 and/or 6dB/oct from 3kHz),
- Types of filters: Bessel order 2, 3 or 4; Linkwitz-Riley order 2 or 4; Butterworth order 2, 3 or 4,
- Level, Polarity and Delay adjustment on each way (driver),
- Graphic display of the crossover response

5.4.3.2 Procedure

The active crossovers can be implemented as follows:


1. The number and type of filters for each loudspeaker are set manually as well as crossover frequencies. If the screen speakers are identical, it's possible to **Link LRC** channels to set them in one operation.
2. Levels, Delays and polarity of each driver are set manually or determined automatically with a calibration if the Optimizer module is installed. **Link LRC musn't be used for these settings.** Automatic procedure measure the reality of the speakers in the room and will always give the best result.

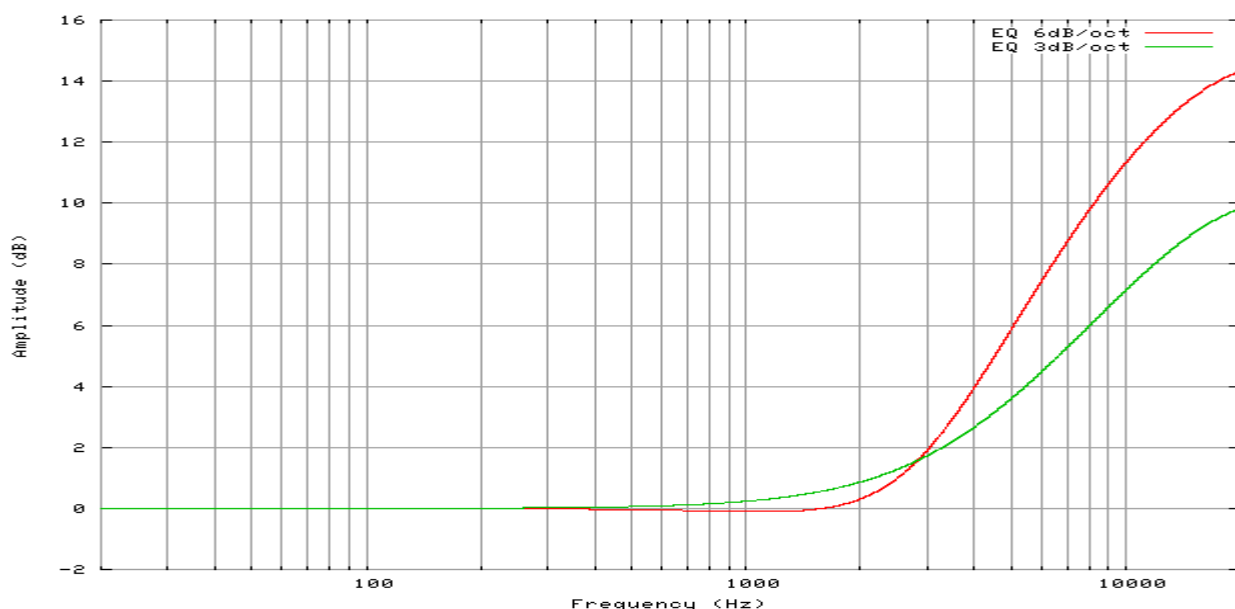
5.4.3.3 Manual settings


The Active Crossovers page displays one tab for each loudspeaker. The number of loudspeakers displayed depends on the number of loudspeakers declared in the *Setup/Speakers Settings* page.

Note: setup of the crossover filters can be done simultaneously for any speakers thanks to their respective **Link** button. Linking has to be done before changing parameters!

The following settings are available for each loudspeaker:

- The **number of ways** can be modified by pressing the **+1** and **-1** buttons,
- The **type** of high-pass or/and low-pass filters is chosen by pressing the  buttons,
- The **cut-off frequency** of each filter is set by clicking in the blue window and type the value.
- Two additional IIR filters are available under the name of “Constant-directivity horn” EQs. The purpose of such filters is to boost the high frequencies for speaker that use constant-directivity horns. These speakers tend to let the high frequency fall down at the sweet spot. Depending on the horn you are using, you may want to boost the high frequencies by a 3 and/or 6 dB/oct, from about 3kHz.



- The **Apply** button is highlighted as soon as a parameter is modified and is used to compute and load the new settings into the processor. Once compute is finished (gears icon disappears from the notification bar), filters are applied to the outputs.
- Physical outputs are affected to new ways. It is therefore necessary to check the speaker routing before being able to listen to the resulting filters.
- If change is unwanted, press the **Cancel** button to cancel your modifications
- If change is accepted, save the preset of your choice in the presets page. Otherwise changes will be lost.
- **Level** and **delay** can be adjusted on each way (driver). Use  buttons: 0,1 dB and 0,01ms on each click.
- **Mute** and **Invert Polarity** buttons are also available.

5.4.3.4 Automatic settings

With the Optimizer option, it is possible, for each speaker, to set up the level, delay and polarity of each driver automatically by a simple procedure that calibrates the drivers separately:

1. Set up the number of ways for the speaker, and the corresponding speakers routing (in the *Setup/Speakers Routing* tab)
2. Set up the low-pass and high-pass filters for each way: filter type, crossover frequency.
3. Press the Calibrate button and proceed as for a global calibration.

This procedure will automatically determine the levels, delays and polarities that should be applied on the drivers of the loudspeaker.

Once the calibration is finished, you can visualize the results in two forms:

- measured **impulse response** of each way: you can see whether the drivers are correctly synchronized.
Ex: 2 ways speaker impulse response on Center channel (Low Freqs in green, High Freqs in red).



- recombined **amplitude** response of the speaker: you can see whether the combination of the drivers is constructive, and you can observe the effects of level/delay/polarity modification on the combined amplitude of the speaker. Two curves are displayed, one showing the global power of the speaker including the room (cyan), and one showing the amplitude of the direct front and early reflections (magenta). Comparing both curves around the crossover Freq will indicate you whether the crossover conserves the directivity of the speaker: the more alike the two curves look, the more directive the speaker is towards the listening spot.



Please note:

- As they are going to be automatically tuned, the previously manual set up levels, delays and polarities will be ignored during the calibration. In other words, tuning these parameters before launching the calibration will have no effect on the result.
- Under some circumstances, the automatic crossover algorithm may suggest inverted polarities for a driver from one speaker to another. This can be explained by various factors:
 - o The physical polarity of one speaker is, indeed, inverted (from a cabling issue, as an example). In this case, the correction suggested by the Optimizer should be applied to improve audio quality
 - o Two adjacent drivers (for instance Mid and High) are phase-shifted by an amount of about 90°. In this case, the Optimizer provides more uncertain results, as it gets harder to see whether the drivers are in or out of phase. If you are not comfortable with the results provided by the Optimizer, you can correct them manually afterwards (via the button “Invert pola.”).

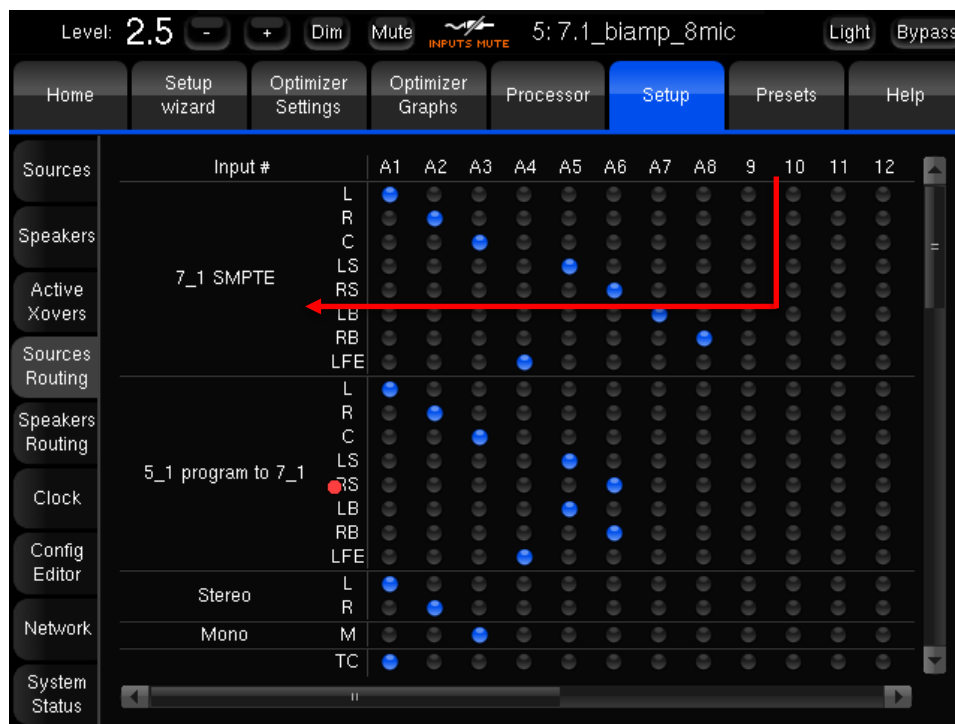
5.4.4 Setup/Clock



Page divided in 5 sub windows:

- **Status information :**
 - Current sample rate: shows the current clock frequency of the processor
 - Detected sample rate: shows the frequencies of detected external clock sources (AES, SPDIF, Word Clock...).
 - Detected sync: shows the types of detected clock sources.
 - Using sync: shows the selected clock.
- **Clock Mode:**
 - Slave or Master (at different rates). Turn loudspeakers OFF before changing: changing this setting may result in loud clicks, depending on the analog converters used.
 - Normal use in commercial cinema is Slave when Ovation receive contents from server.
- **Clock Source:** indicate on what physical input the clock comes in the processor.
- **Audio Buffer Size:**
 - 512 is the default value
 - Smaller values can be selected to decrease latency, but you should check that there are no sync losses. Note: changes are ignored until the next reboot.
- **Stored in preset:** if this button is on when you save a preset, clock settings will be saved and recalled with that preset.
- **CPU Load:** indicates if there is a CPU overload or not.

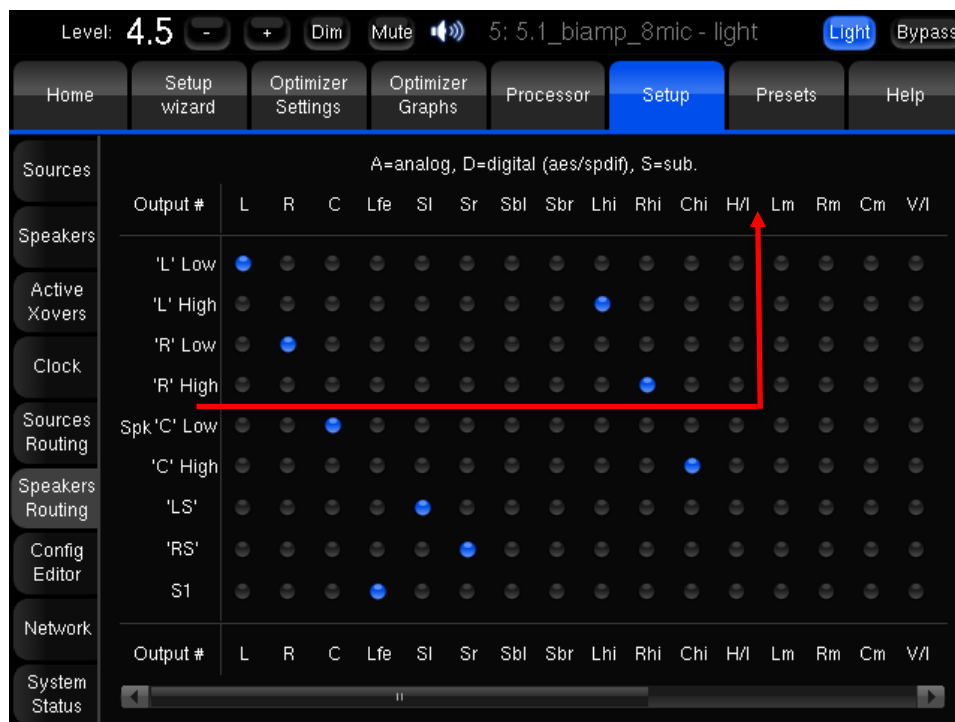
5.4.5 Setup/Sources Routing



This page allows you to affect physical inputs (columns) to the different channels of the software sources (rows).

- Channels (rows) names and order of display depends on the input format and channel order declared for each software source in **Setup/Sources** settings. You can assign physical inputs to any channel. **Note:** the system automatically applies the relevant attenuation if several channels of a same source are routed to the same physical input.
- The 8 signals of the calibration microphone, like any physical inputs, must also be routed. For convenience, 8 specific channels are dedicated to the microphone and regrouped in a special software source named "**Micro**". If additional "signal inputs" are needed, inputs can be routed to be shared for the microphone and a normal signal. (Note: in this case, both signals are routed simultaneously!).
- Sources routing may be changed after calibration.
- Note that LFE is physicaly in 4th position to be DCI compliant but in last position because of the *Setup/Sources* page settings.

5.4.6 Setup/Speakers Routing



- This page allows you to affect software output channels (rows) to the physical outputs of the Ovation (columns).
- The number of outputs channels displayed in the grid corresponds to the number of loudspeakers and subwoofers declared in the *Setup/Speakers* settings + number of ways declared in *Setup/Active Xovers* page.
- The sort order of output channels depends on the input format and channel order of the active software source defined in *Setup/Sources* page.
- If the number of speakers is bigger than the number of channels of the input format of the active source, additional channels will be displayed as numbers. They also correspond to the calibration order.

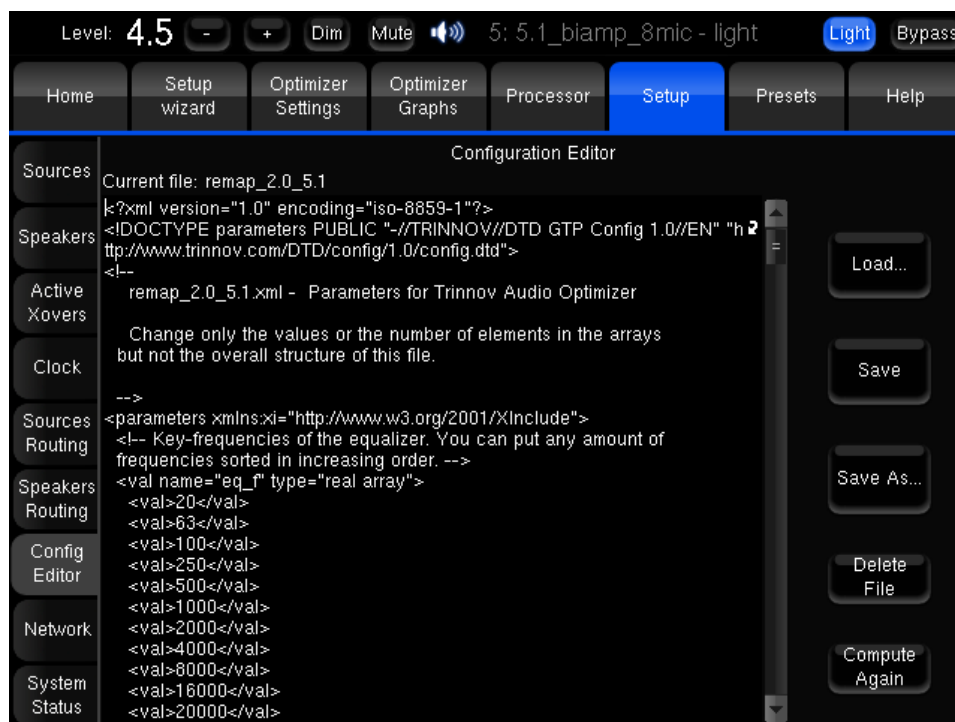
Notes:

- **Do not** change the Speakers Routing once calibration has been performed. Correction filters have a direct association between routing outputs and speakers. Changing Speakers Routing after Calibration will result in a mismatch of compensation by re-directing the corrected signal to the wrong Loudspeaker.
- Last software channels will always be the Subs: S1, S2 But, to be DCI compliant, the main LFE physical output will be the 4th.

5.4.7 Setup/Config Editor

Certain advanced settings have not yet been implemented in the user interface. Instead, they are stored in a text file, referred to as the “Config file”. Config files are based on XML, a computing standard that facilitates the sharing of data among computer programs.

Each preset is linked to *one and only one* XML file. Conversely, one XML file can be used by one or more presets. Each XML file includes a set of parameters that specify the behaviour of the Optimizer’s calibration, analysis and optimization algorithms.



Custom remapping matrix:

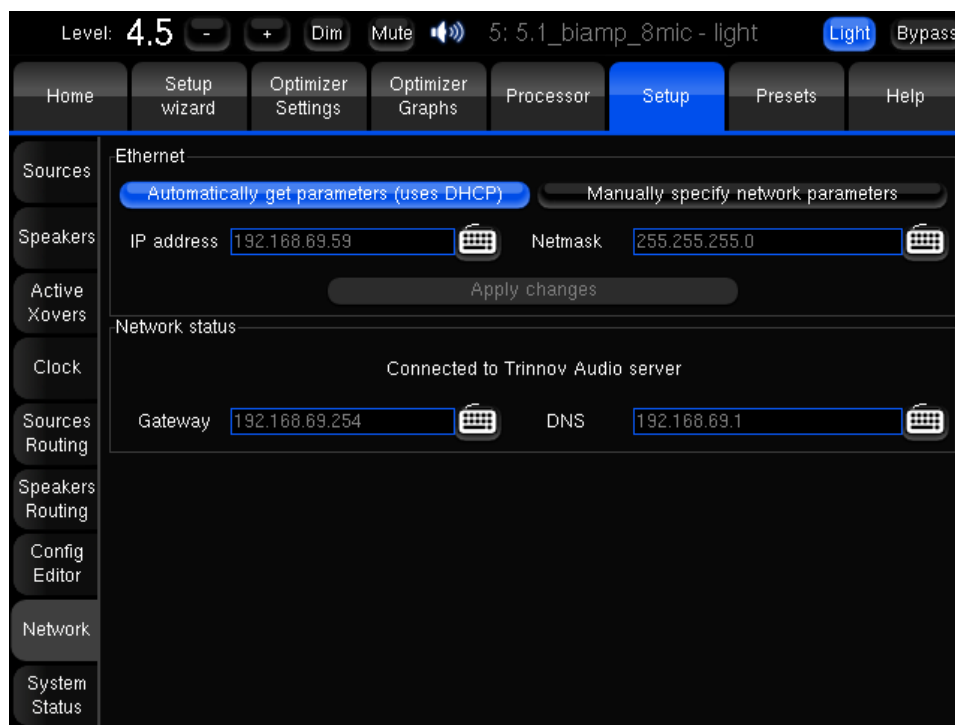
For specific requirements, a custom remapping matrix can be specified, for example to route one input channel to several output channels. For more information on this feature, please contact your Trinnov distributor.

Parametric filters:

Additional parametric filters can be defined on each channel, for more information on this feature, please contact your Trinnov distributor.

NOTE: don't change anything in Config Editor if don't have a perfect knowing of XML language and Ovation.

5.4.8 Setup/Network



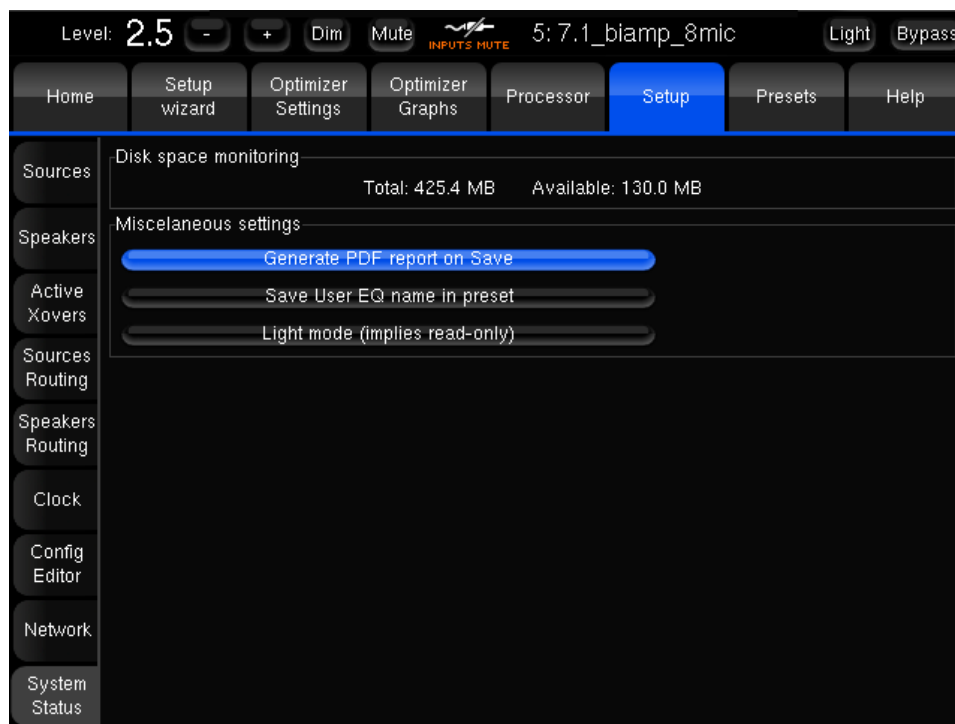
Two modes: DHCP on and DHCP off.

- DHCP on when the button “Automatically get parameters (uses DHCP)” is activated. IP address, Netmask, Gateway and DNS will be automatically defined by the network devices.
- DHCP off when the button “Manually specify network parameters” is selected. IP address, Netmask, Gateway and DNS can be defined in the corresponding blue windows. To validate any change, press the “Apply changes” button.

Notes:

- It is also possible to select DHCP on or off and set addresses on the front panel screen of the Ovation:
 - Turn Select knob to have Setup menu on screen, validate with Enter button,
 - First line indicate if the Ovation is connected to any network,
 - Use Select knob to go on the second line: DHCP, presse Enter haxe access to DHCP mode,
 - You can select On or Off and validate with Enter, press Back button to return to the previous page,
 - If DHCP is Off, turn Select to choose the address you want to set, press Enter,
 - With Select, choose the number you want to set, press Enter to access to the modification, turn Select to choose the right value, press Enter to validate,
 - When all numbers are setted, turn Select to go on “OK” and press Enter, then you come back to the previous menu.
- Trinnov software updates and remote support require Internet access, indicated in the Network Status display “Connected to Trinnov Audio Server.”

5.4.9 Setup/System Status



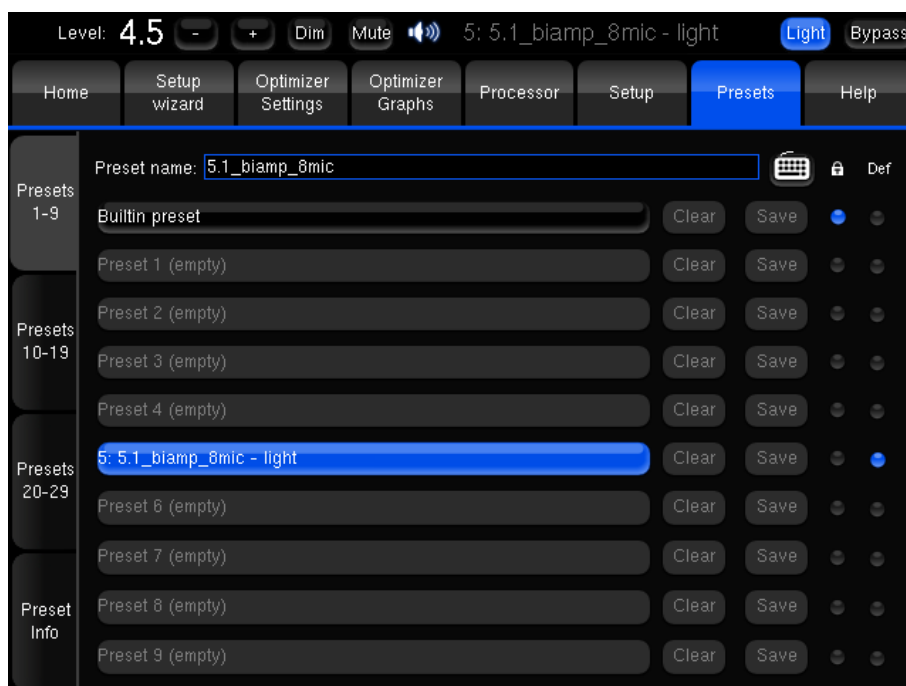
Disk space monitoring allows to check if the flash memory is full or nearly full.

Miscellaneous settings:

- **Generate PDF report on save:** when activated, a PDF report corresponding to the preset is generated when the preset is saved. Please note: when deactivated, the previous PDF file will be erased next time the preset is saved. This avoids having PDFs that do not correspond to the current preset. PDFs can then be copied onto USB sticks or transferred from FTP.
- **Light mode (implies read-only):**
 - When activated, the user will not be able to save or clear presets. Having a backup copy of all your presets and switching your system to light mode keeps you from sensitive data loss.
 - This button is exactly the same as the one in the upper banner of screen. The upper banner button is accessible from all pages instead of the one in this page.
 - This is the normal mode to assure the quickest loading of preset and Profile.
 - More than a preventive measure, the light mode also allows faster preset changes because it uses lighter preset versions. Light presets do not recall measurement data such as impulse responses and do not load graphs.

5.4.10 Presets/Presets 1-29

Ovation can store up to 29 memory presets. presets can be backed-up and restored to/from a USB key.



Except for the *Master Level* and the *Synchronization Mode* (unless you have selected the corresponding “Stored in preset” option) all setup data is saved in the preset: optimization settings, routing, levels, delays, FIR EQ and Graphic EQ settings, display choices...

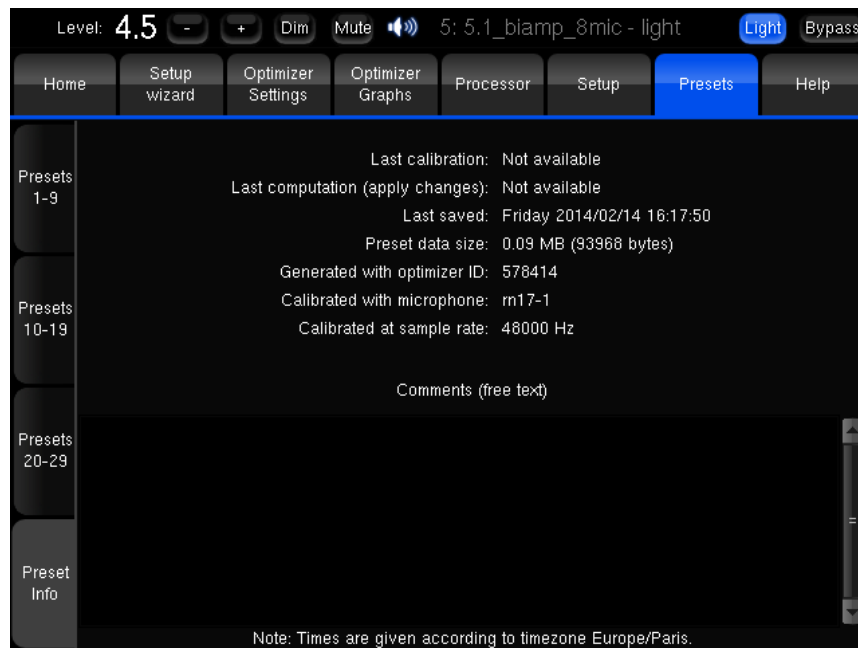
- Each preset can be locked (small lock icon) or deleted (Clear button).
- Once saved, a preset can be loaded with its button.
- One preset can be chosen as the default preset to be automatically loaded at each start-up if its Def button is checked. The “no default config” of the Startup Screen can disable this automatic load if desired.

If standard PS2 or USB keyboard is connected to Ovation, presets can be renamed by just typing a new name in the “Preset name” text blue window. If not, a virtual keyboard is also available to label your presets. Type a label in the “Preset name” window, then press the Save button.

5.4.11 Preset/Preset Info

The preset Infos tab provides useful information about the active preset:

- **Last calibration:** the date and time of the last calibration.
- **Last computation:** the last time the user pressed the Apply Changes button and the preset has been recomputed.
- **Last saved:** the last time the preset was saved.
- **Preset data size:** the size of the preset’s data on the flash memory.
- **Generated with optimizer:** the ID of the optimizer where the preset was calibrated.
- **Calibration microphone:** the ID of the microphone kit used for the calibration.
- **Calibrated at sample rate:** the sample rate used for the calibration.
- **Notes:** free space to type infos. This can be used to store a version history of the preset.

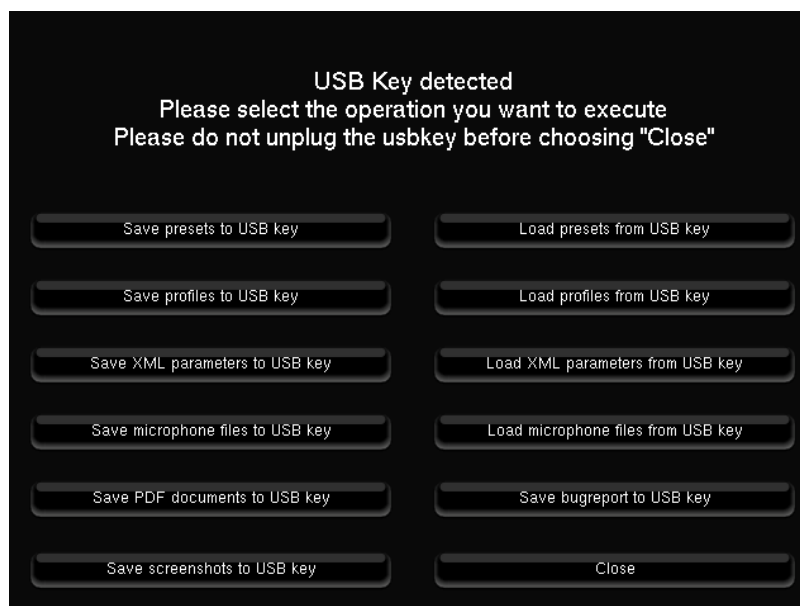


5.4.12 Backup/Restore presets

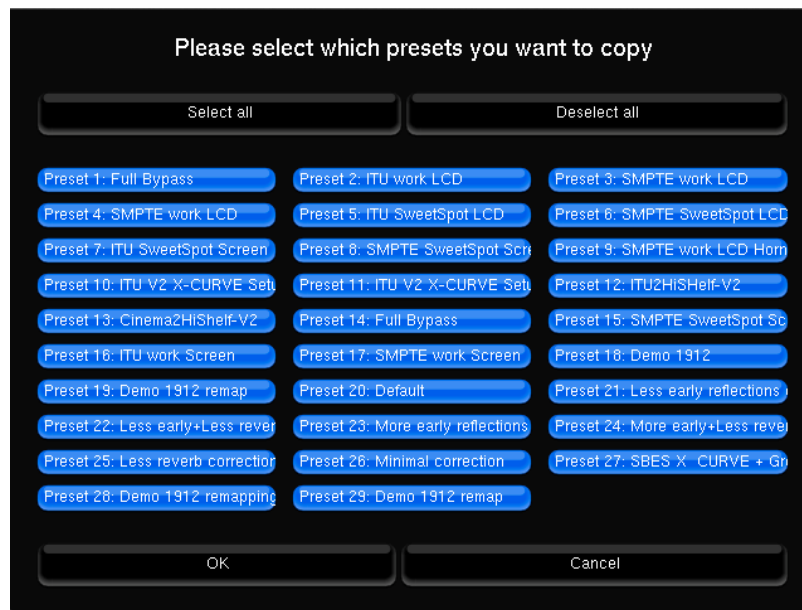
5.4.12.1 Backup / Restore with a USB Key

The purpose of this feature is to make a global or partial backup of the system and restore it. While the Trinnov unit is running, plug a USB memory stick in one of the USB ports of the Ovation (front or back panels). The menu below will appear. It allows one to restore settings from the USB key into the processor, or backup them from the processor into the USB stick:

- **Save or Load presets** include all the information related to a preset.
- **Save or Load Profiles** include all the information related to a Profile.
- **Save or load XML Parameters** correspond to the XML files used in the Config Editor.
- **Save or Load microphone compensation files.**
- **Save PDF Documents** are the measurement reports generated by the Optimizer.
- **Save bugreports and screenshots.**



Presets can be saved and restored individually.



Caution: the elements previously stored on the processor will be replaced. It is of course possible to remove files on the memory stick in order to restore only some specific settings.

5.4.12.2 Backup / Restore through the network (via FTP)

FTP functionality allows accessing the preset files, **report** files (.pdf) and the Screenshots of your Optimizer for backup and consultation. Use any FTP client from any computer of the LAN (web browsers like Internet Explorer, Firefox... have FTP capability) using the current IP address of your Optimizer (see *Setup/System Status* page for connection settings).

Example: ftp://192.198.0.5

The built-in FTP server will ask you to log in:

Login = srp

Password =
- the 7 digit **product ID** shown on the help page
- or the 6 digit serial number labeled on the rear panel

5.4.13 Optimizer Graphs

The Optimizer provides a complete and flexible tool for displaying the responses of the speakers before and after optimization, as well as the correction filters. It is possible to observe this information through various angles: amplitude, phase, group delay, impulse response.

These graphs are available in the *Optimizer Graphs* page. This page includes two main parts:



5.4.13.1 Commands and parameters

The commands and parameters zone includes two tabs: « *Display* » and « *Settings* ».

The « *Display* » part is made of 3 subparts:

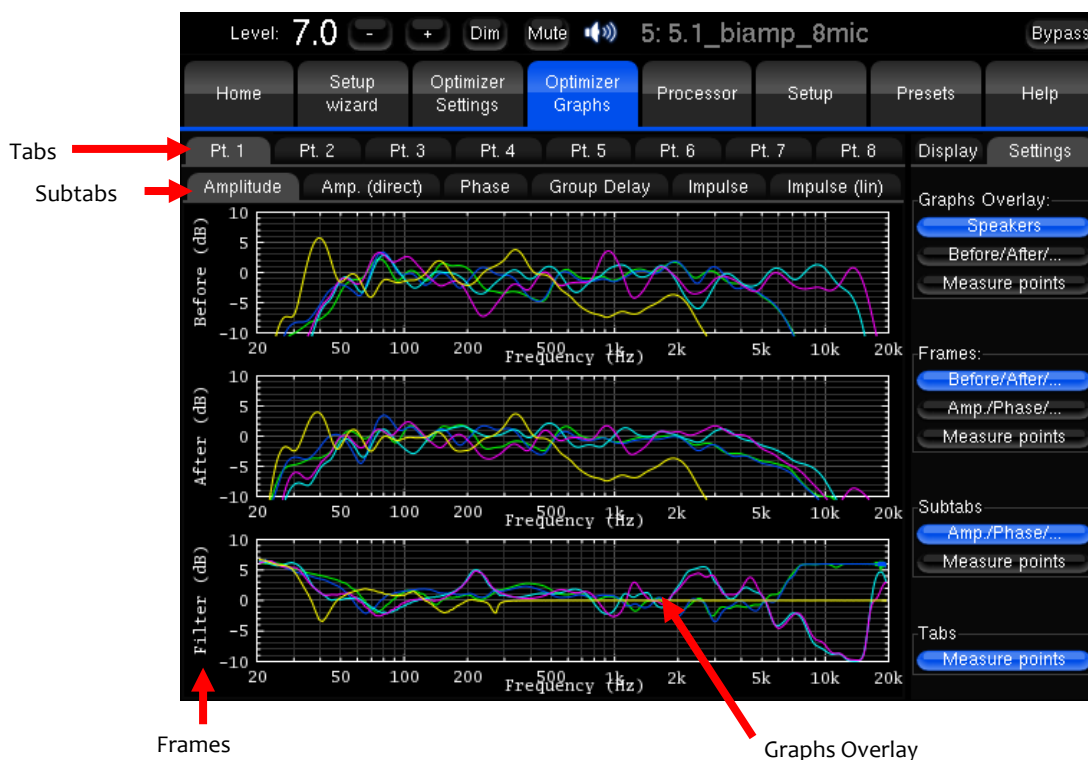
- **Graphs:** allows to select which graphs you want to visualize. For example: the amplitude response of L, R, LS, RS and S1 channels on measurement point 1.
- **Frames:** allows to select which frames you want to visualize: 'Before' (the filters application) and/or 'After' (filters applied) and/or 'Filter' (resulting of the computing phase)
- **Zoom:** allows modifying amplitude, frequency and time scales according to the type of representation. Modifications possible on axis X and/or Y.

The « *Settings* » part allows you to organize **Tabs**, **Subtabs**, **Frames** and **Graphs Overlay**.

In the example below:

- The **Graphs Overlay** is 'Speakers': you will visualize the output channels graphs associated to the speakers. These speakers are defined in « Setup/Speakers » page (Loudspeaker number, Subwoofer number).
In the « *Display* » part, it will be possible to choose which speakers you want to view in the available ones. For example: L, R, C, LS, RS, S1 (Loudspeaker number = 5 and Subwoofer number = 1).
- The **Frames** to display are 'Before/After/Filter': you will view or mask each one of these three frames in the field 'Frames' of the « *Display* » part. From the top to the bottom, the order of these frames is always: Before – After – Filter.

- The **Subtabs** option is 'Amp/Phase/Delay/Impul': here are defined the available pages in the horizontal subtabs. In the present case, it will be: Amplitude, Amplitude (direct), Phase, Groupe Delay, Impulse and Impulse (lin).
- The **Tabs** option is 'Measure points': here are defined the available pages in the horizontal tabs. In the present case, it will be the measurement points 'Pt.1' to 'Pt.8' for a Multi-microphones calibration mode.



The following combinations are possible:

Graphs overlay	Frames	Subtabs	Tabs
speakers	before/after/filter	amp/phase/delay/impul	measure points
speakers	before/after/filter	measure points	amp/phase/delay/impul
speakers	amp/phase/delay/impul	before/after/filter	measure points
speakers	amp/phase/delay/impul	measure points	before/after/filter
speakers	measure points	before/after/filter	amp/phase/delay/impul
speakers	measure points	amp/phase/delay/impul	before/after/filter
before/after/filter	speakers	amp/phase/delay/impul	measure points
before/after/filter	speakers	measure points	amp/phase/delay/impul
before/after/filter	amp/phase/delay/impul	speakers	measure points
before/after/filter	amp/phase/delay/impul	measure points	speakers
before/after/filter	measure points	speakers	amp/phase/delay/impul
before/after/filter	measure points	amp/phase/delay/impul	speakers
measure points	speakers	before/after/filter	amp/phase/delay/impul
measure points	speakers	amp/phase/delay/impul	before/after/filter
measure points	before/after/filter	speakers	amp/phase/delay/impul
measure points	before/after/filter	amp/phase/delay/impul	speakers
measure points	amp/phase/delay/impul	speakers	before/after/filter
measure points	amp/phase/delay/impul	before/after/filter	speakers

In all these combinaisons, you will use some of them more than the others. Try to configure your Optimizer Graphs page as convenient as possible.

NOTE: the viewing settings are saved in presets. This display only takes under consideration the automatic filters provided by the Optimizer. All the complementary manual settings (Input EQ, FIR EQ, preset EQ, User EQ, ...) are not included in this visualisation.

5.4.13.2 Visualisation

There is no possible action in the visualisation zone. You can just use Tabs and Subtabs to select the graphs you want to see regarding the configuration you made in the « Settings » part.

Please note:

- The Graphs colors are copied on the selecting buttons in the 'Graphs' field of the « Display » part,
- Pay attention to the horizontal and vertical unities (dB, ms, Hz, degree) in the way to use as best the graphs and the zoom options,
- Try to configure the viewing combinaisons in a way to avoid too many information displayed to ease graphs analysing.

5.4.14 Optimizer Settings

The Optimizer Settings page provides settings concerning the calibration and the optimization.

5.4.14.1 Optimizer Settings/Runtime



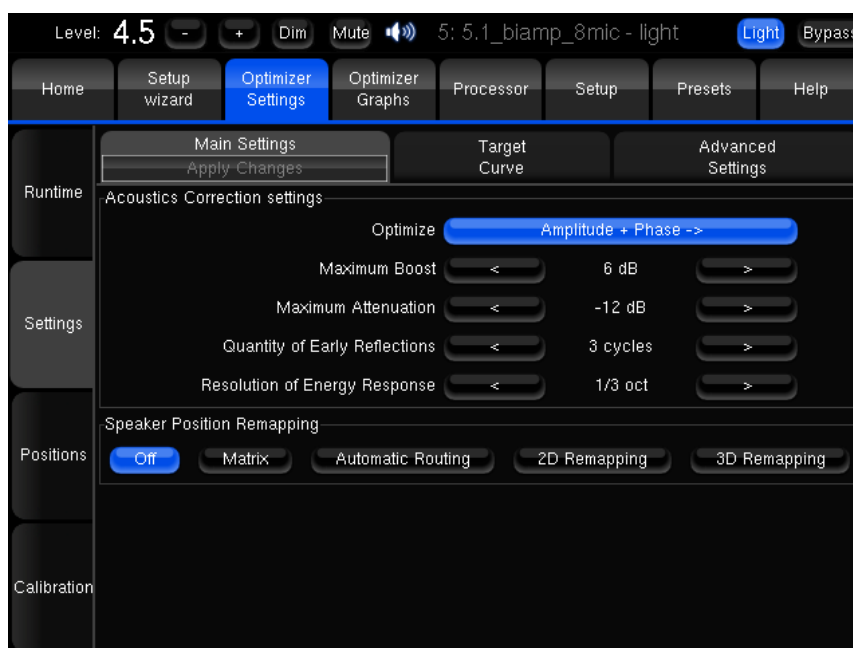
- **Optimization ON/OFF:** allows the user to bypass all the processing related to the Settings page: the acoustic correction, the automatic delay and level alignment, as well as the remapping options. When Optimization is OFF, only the processing defined in the other pages is applied:
 - routing, as defined in the Setup Sources and Speakers routing tabs,
 - levels, as defined in the Processor tabs,

- graphic EQs, as defined in the Processor tabs,
- bass management (not used in cinema).
- **Acoustic Correction** ON/OFF: when turned OFF, both the automatic equalization (defined by the target curve) and the FIR EQ are bypassed.
- **Level Alignment** ON/OFF: the automatic alignment of speaker levels can be disabled, meaning that no automatic gain changes will be applied to the outputs.
- **Delay Alignment** ON/OFF: the automatic alignment of speaker distances can be disabled, meaning that no automatic delays will be applied to the outputs.

Please Note: for more information about the global bypass mode, please refer to the appendix of this document.

5.4.14.2 Optimizer Settings/Settings

5.4.14.2.1 Optimizer Settings/Settings/Main settings



Main Optimization Settings

- **Optimize:**
 - Amplitude + Phase (default):** with this setting, the Optimizer will improve both the loudspeaker amplitude and the phase response of the loudspeaker. This greatly reduces the group delay of the speakers starting from about 200Hz,
 - Amplitude only:** this mode tells the Optimizer to work only on the amplitude of the loudspeaker's response. The phase behavior is not modified,
 - Low range only:** with this setting the automatic equalization will only use IIR filters up to frequency defined in the advanced settings. The automatic FIR filter is disabled, but the FIR EQ can still be applied,
 - According to L&R speakers:** this is a special mode that will optimize the center and surround speakers in order to achieve the same response as the Left and Right speakers. It is mostly useful in home cinema installations.
- **Maximum boost:**
 - It defines, in dB, the maximum amount of boost that will be performed by the algorithms. This parameter is used to avoid distortion.
 - This parameter has an important impact on the behaviour of the automatic equalization, and is applied to both the time-based and the energetic approach. Default value is 6dB.

- **Maximum attenuation:**

It defines, in dB, the maximum amount of attenuation that will be performed by the algorithms. This parameter also has an important impact on the behaviour of the automatic equalization, and is applied to both the time-based and the energetic approach. Default value is -12dB.

NOTES: about the parameters “Quantity of Early Reflections” & “Resolution of Energy Response”

The Optimizer uses two different approaches for the Optimization of Loudspeaker/Room Acoustics:

- 1) A time-based approach for the correction of Early Reflections (ER). The main parameter that defines the behaviour of this algorithm is the width of the time-frequency window “Quantity of Early Reflections”.
- 2) An energetic approach for the correction of Late Reverberation (LR), whose main parameter, Resolution of Energy Response is the smoothing applied to the energy response.

- **Quantity of Early Reflections:**

A simple parameter that characterizes the quantity of early reflections that the Optimizer tries to compensate is the width of the time-frequency window. The size of this window is defined by a number of cycles, hence the naming “ α/f ”, where α (alpha) is the number of cycles and f the frequency ($1/f$ being one cycle). Default value is 3 cycles.

The meaning of the time-frequency window is as follows: for each frequency a different duration (or width of the time window) is taken into account. At low frequencies the window typically starts at 150ms for 20Hz, and decreases constantly to be as low as 0.3ms at 10kHz.

- **Resolution of Energy Response:**

It defines, in number of octaves, how the room’s energy response is smoothed, and modifies the behaviour of the equalization performed by the Optimizer on the *Late Reverberation*. The behaviour of the energy optimization algorithm varies according to the smoothing applied to the room’s response. If the response is less smoothed, sharp peaks in the response will be more taken into account for the correction, while with more smoothing only the overall tonal balance of the room will be corrected. Default value is 1/3 octave.

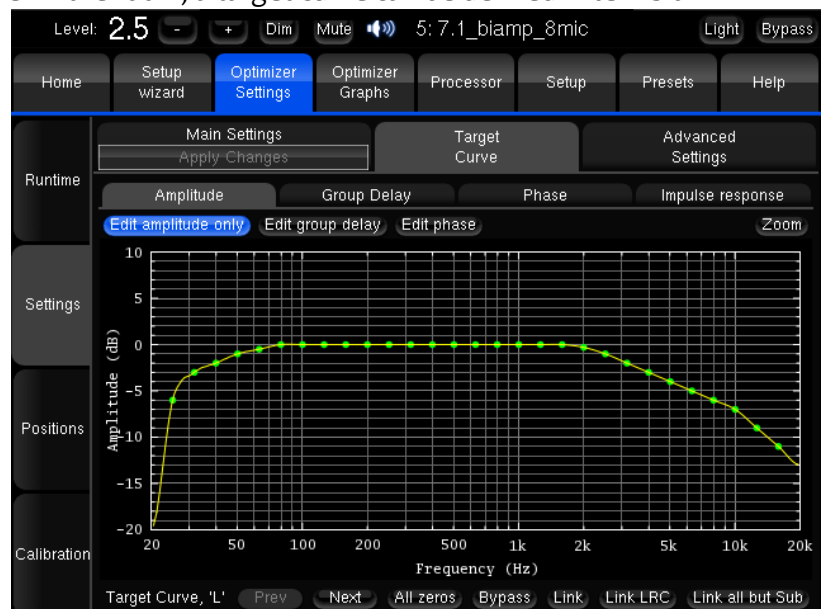
- **Speaker position remapping:** not used in cinema.

5.4.14.3 Target curves

The Optimizer implements Target Curves as a powerful tool for achieving the required frequency response from your sound system. For every speaker in the room, a target curve can be defined in terms of:

- Amplitude only
- Amplitude and Phase
- Amplitude and Group Delay

One or more channels can be linked in order to share the same target curve for several speakers. The most usual case is to link screen speakers. A specific button was implemented for this configuration: *Link LRC* button.



The curve can easily be edited with the arrows of the keyboard:

- the **green dots** correspond to the required values for each frequency,
- the **yellow line** displays the expected result, taking into account the behaviour of the filter,
- click on green dot, it will appear in red and current freq and level values will be indicated.



Please note:

- by default, the Optimizer will use both IIR and FIR filters to achieve the target curve. This can be changed to IIR only or FIR only in the “Use Filters” parameter of the *Optimizer Settings/Advanced Settings*,
- in addition to target curves, several other parameters define the behaviour of the automatic equalization.

5.4.15 Optimizer Settings/Runtime

- **Optimization ON/OFF:** allows the user to bypass all the processing related to the Optimizer Settings page: the acoustic correction, the automatic delay and level alignment, as well as the remapping options. When Optimization is OFF, only the processing defined in the other pages is applied:
 - Routing, as defined in the Setup/Sources Routing and Setup/Speakers Routing pages,
 - Levels, as defined in the Processor Master, Inputs and Outputs pages,
 - Graphic EQs, as defined in the Processor Inputs and Outputs pages,
 - Bass management.
- **Acoustic Correction ON/OFF:** when turned OFF, both the automatic equalization (defined by the target curve) and the FIR EQ are bypassed,
- **Level Alignment ON/OFF:** the automatic alignment of speaker levels can be disabled, meaning that no automatic gain changes will be applied to the outputs,



- **Delay Alignment ON/OFF:** the automatic alignment of speaker distances can be disabled, meaning that no automatic delays will be applied to the outputs.

Please Note: for more information about the global bypass mode, please refer to the appendix 7.3 of this document.

5.4.16 Optimizer Settings/Settings

5.4.16.1 Optimizer Settings/Settings/Main Settings



- **Optimize:**
 - **Amplitude + Phase (default):** with this setting, the Optimizer will improve both the loudspeaker amplitude and the phase response of the loudspeaker. This greatly reduces the group delay of the speakers starting from about 150Hz,
 - **Amplitude only:** this mode tells the Optimizer to work only on the amplitude of the loudspeaker's response. The phase behavior is not modified,
 - **Low range only:** with this setting the automatic equalization will only use IIR filters up to frequency defined in the *Optimizer Settings/Settings/Advanced Settings* tab(Default: 150Hz). The automatic FIR filter is disabled, but the FIR EQ can still be applied,
 - **According to L&R speakers:** this is a special mode that will optimize the center and surround speakers in order to achieve the same response as the Left and Right speakers. It is mostly useful in home cinema installations. The different options available for this mode are explained in the *Optimizer Settings/Settings/Advanced Setting* tab.
- **Maximum boost:**
It defines, in dB, the maximum amount of boost that will be performed by the algorithms. This parameter is used to avoid distortion. Its default value is 6dB.
This parameter has an important impact on the behaviour of the automatic equalization, and is applied to both the time-based and the energetic approach.
- **Maximum attenuation:**
It defines, in dB, the maximum amount of attenuation that will be performed by the algorithms. Its default value is -12dB.
This parameter also has an important impact on the behaviour of the automatic equalization, and is applied to both the time-based and the energetic approach.

About the parameters “Quantity of Early Reflections” & “Resolution of Energy Response”:

The Optimizer uses two different approaches for the Optimization of Loudspeaker/Room Acoustics:

- A time-based approach for the correction of Early Reflections (ER). The main parameter that defines the behaviour of this algorithm is the width of the time-frequency window “Quantity of Early Reflections”,
- An energetic approach for the correction of Late Reverberation (LR), whose main parameter, Resolution of Energy Response is the smoothing applied to the energy response.

- Quantity of Early Reflections:

A simple parameter that characterizes the quantity of early reflections that the Optimizer tries to compensate is the width of the time-frequency window. The size of this window is defined by a number of cycles, hence the naming “ α/f ”, where α (alpha) is the number of cycles and f the frequency ($1/f$ being one cycle). Default setting is 3 cycles.

The meaning of the time-frequency window is as follows: for each frequency a different duration (or width of the time window) is taken into account. At low frequencies the window typically starts at 150ms for 20 Hz, and decreases constantly to be as low as 0.3ms at 10 kHz.

- Resolution of Energy Response:

It defines, in number of octaves, how the room's energy response is smoothed, and modifies the behaviour of the equalization performed by the Optimizer on the *Late Reverberation*. Its default value is 1/3 (third octave).

The behaviour of the energy optimization algorithm varies according to the *smoothing* applied to the room's response. If the response is less smoothed, sharp peaks in the response will be more taken into account for the correction, while with more smoothing only the overall tonal balance of the room will be corrected.

- **Speaker position remapping:**

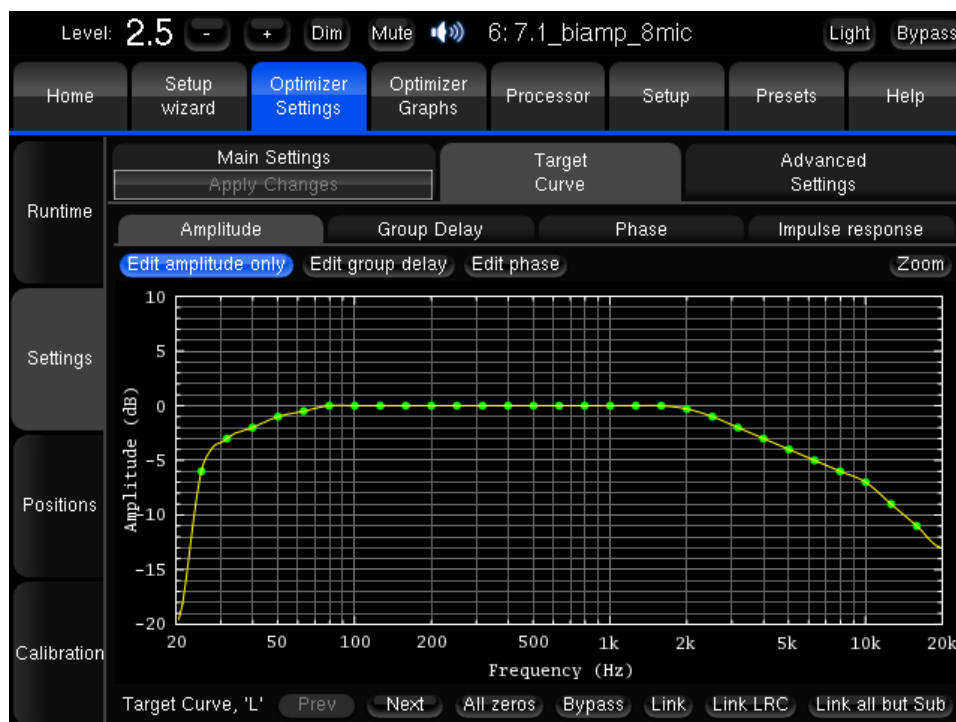
- **Off (default):** remapping is disabled. Normal use in commercial cinema.
- **Matrix:** specifies that the I/O routing to be taken into account is in the *Setup/Config Editor* page. To be used by advanced users.
- **Automatic routing:** This option uses the Optimizer's 3D speaker position information (from the calibration with the Trinnov 3D microphone) to automatically route each channel to the speaker closest to the reference position.
- **2D Remapping:** this feature performs a remapping of the loudspeakers positions, but only *in the horizontal plane*. This allows for :
 - Compensating for incorrect loudspeakers position, but only with respect to horizontal angles (azimuth).
 - Rendering a signal format (stereo, 5.1, 7.1...) on any number of loudspeakers. This includes up-mixes and down-mixes.
- **3D Remapping:** This mode enables the full spatial optimization in 3D. This allows for:
 - Compensating for incorrect loudspeaker position in azimuth and elevation.
 - Rendering a signal format (stereo, 5.1, 7.1...) on any number of loudspeakers. This includes up-mixes and down-mixes.

Important Note: Speakers Remapping mode can be controlled with Profiles.

5.4.16.2 Optimizer Settings/Settings/Target Curve

The Optimizer implements Target Curves as a powerful tool for achieving the required frequency response from your sound system. For every speaker in the room, a target curve can be defined in terms of:

- Amplitude only
- Amplitude and Phase
- Amplitude and Group Delay



One or more channels can be linked in order to share the same target curve for several speakers. The most usual case is to link LRC channels before making modifications. A specific button was implemented for this configuration.

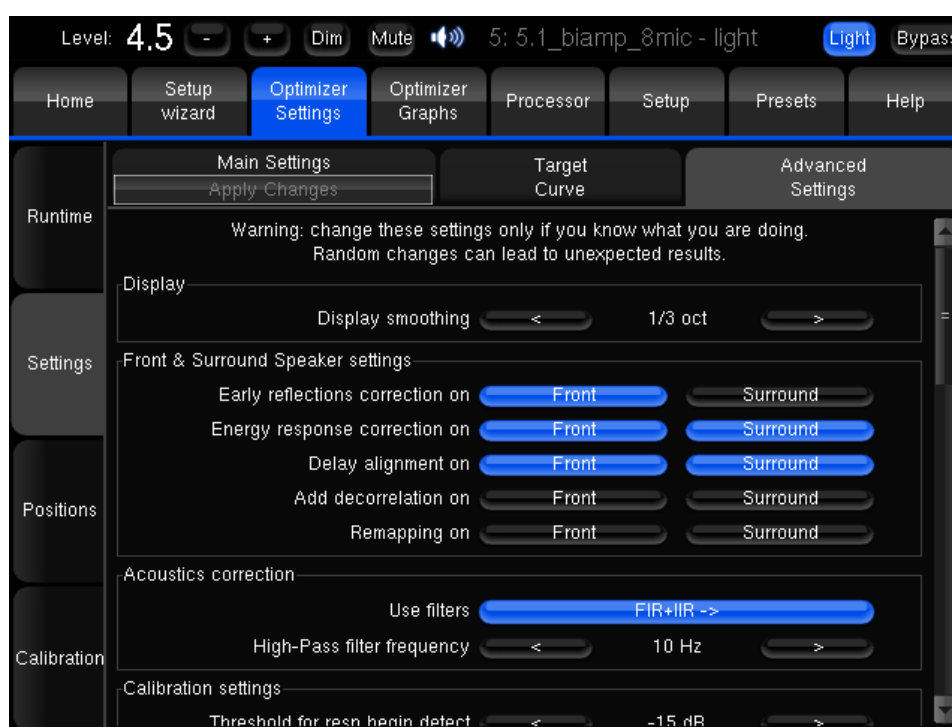
The curve can easily be edited with the arrows of the keyboard:

- The **green dots** correspond to the required values for each frequency,
- The **yellow line** displays the expected result, taking into account the behaviour of the filter.

Please note:

- By default, the Optimizer will use both IIR and FIR filters to achieve the target curve. This can be changed to IIR only or FIR only in the *Optimizer Settings/Settings/Advanced Setting/Acoustics correction* window with the “Use Filters” parameter.

5.4.16.3 Optimizer Settings/Settings/Advanced settings



All the following settings are accessible using the right scroll bar on the page.

- **Display smoothing:** defines the smoothing value used for displaying the frequency response curves, **in the pdf document** generated while saving, in amplitude and phase. A smaller value can be used to display more details in the frequency response.
Available settings: 1/24, 1/12, 1/6, 1/3, 1/2, 1, 2 and 3 oct.
Please note that this parameter has no effects on the curves displayed in the Optimizer Graphs page.
Default setting is 1/3 octave.
- **Front & Surround Speaker settings:**
The following features can be separately configured on for the **Front** speakers and on the **Surround** speakers.
Please note that Front speakers are defined as those whose azimuth is below 90 degrees (LRC). Conversely, Surround speakers are defined as those whose azimuth is above 90 degrees.
 - *early reflections correction* (default setting is Front),
 - *energy response correction* (default setting is Front and Surround),

- *delay alignment*, e.g. surround delay alignment may or may not be required depending on the application and recommendations (default setting is Front and Surround),
 - *add decorrelation*: in the case of sound editing and sound feature mixing rooms for cinema stages, the Optimizer can apply a decorrelation algorithm to the surround channels in order to simulate the diffuse field created by a belt of surround speakers (default setting is no Front and no Surround),
 - *remapping*: in certain monitoring situations, such as in dubbing theatres, it may be preferable not to apply remapping to the front speakers (default setting is no Front and no Surround).
- **Acoustics Correction:**
 - Use Filters:
 - **FIR + IIR**: default setting used by the Optimizer: both IIR and FIR filters are used to work on the speaker responses full range,
 - **FIR only**: no IIR filters are used, which means that the Optimizer will only work on the mid and high range of the speaker responses. The low range won't be optimized,
 - **IIR only**: with this setting, both the automatic FIR and the FIR EQ are disabled. The Optimizer will only use IIR filters to work on the low range. This setting could be used in specific cases where the user wants to be sure that no FIR filters at all are applied.
 - **High-pass filter frequency**: defines the cutoff frequency for the high-pass filter applied to all channels. Can be disabled. Default setting at 10 Hz.
 - **Calibration settings:**

Threshold for resp begin detect: is the threshold below the peak level of the impulse that is taken into consideration to determine the onset of the response. However, a lower threshold is recommended in a room with a huge amount of early reflections where the peak might not be detected correctly. Default setting at -15dB.
 - **Optimize according to L&R speakers settings:**

Processing on L&R speakers (default setting is "IIR only"): if the optimize mode is set to "Optimize according to L&R speakers" in the *Optimizer Settings/Settings/Main Settings* tab, the Optimizer will only use IIR filters on the L&R speakers. This can be switched to "None" to make sure the L&R speakers are not optimized at all.
 - **FIR and IIR settings:**
 - FIR filter length (default setting is 20ms): defines the length or number of taps of the FIR filter. The default setting of 20ms corresponds to 1024 taps at 48kHz and 2048 taps at 96kHz,
 - Number of IIR filters (default setting is 15): the number of IIR filters that will be used on every channel,
 - IIR filters minimal/maximal frequency (default settings are 20Hz/300Hz): IIR filters will be positioned from the min frequency up to the max frequency.
 - **Level alignment settings:**
 - Weighting used for levels (default setting is dBA): sets the type of weighting used by the optimizer for level alignment,
 - Width of level window (default setting is 16/f): this time-frequency window is used to compute the perceptual level of every speaker. Its width can be modified to improve the automatic level alignment,
 - Maximum/minimum gain on speakers (default settings are 10dB/-20dB): defines the maximum/minimum gain that will be applied for the automatic level alignment,
 - Minimal/maximal bandwidth frequency (default settings are 10Hz/Unlimited): defines the high end/low end of the bandwidth that is used as the starting point for level computations.

- **Subwoofer low-pass filter settings:**

- Cutoff frequency (default setting is “disabled”): defines the cutoff frequency for the low-pass filter that can be applied to the subwoofer.
- Filter type (default setting is Butterworth): defines the type of filter used at the cutoff frequency. The elliptic filter is sharper than the Butterworth but shows ripples in the whole bandwidth.
- Filter order (default setting is 4): defines the steepness of the filter. The slope gets stiffer as you increase the filter order. Please be aware that such changes may result in significant phase shifts
- Rp value (for elliptic filter) (default setting is 0.1 dB): sets a maximum amplitude ripple above the cutoff frequency.
- Rs value (for elliptic filter) (default setting is 80 dB): defines the attenuation below the cutoff frequency.
- Subwoofer delay alignment (default setting is Enable): allows to include the Subwoofer in the speakers delay alignment.

- **Decimation settings:**

It is strongly recommended not to change these settings unless you have been requested to do so by Trinnov Support.

- **Advanced FIR settings:**

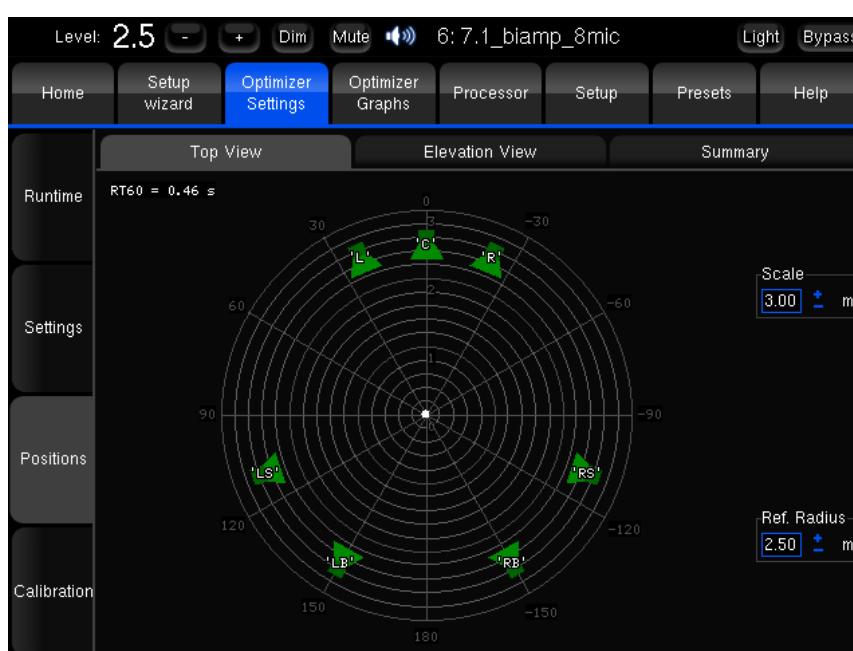
It is strongly recommended not to change these settings unless you have been requested to do so by Trinnov Support.

5.4.17 Optimizer Settings/Positions

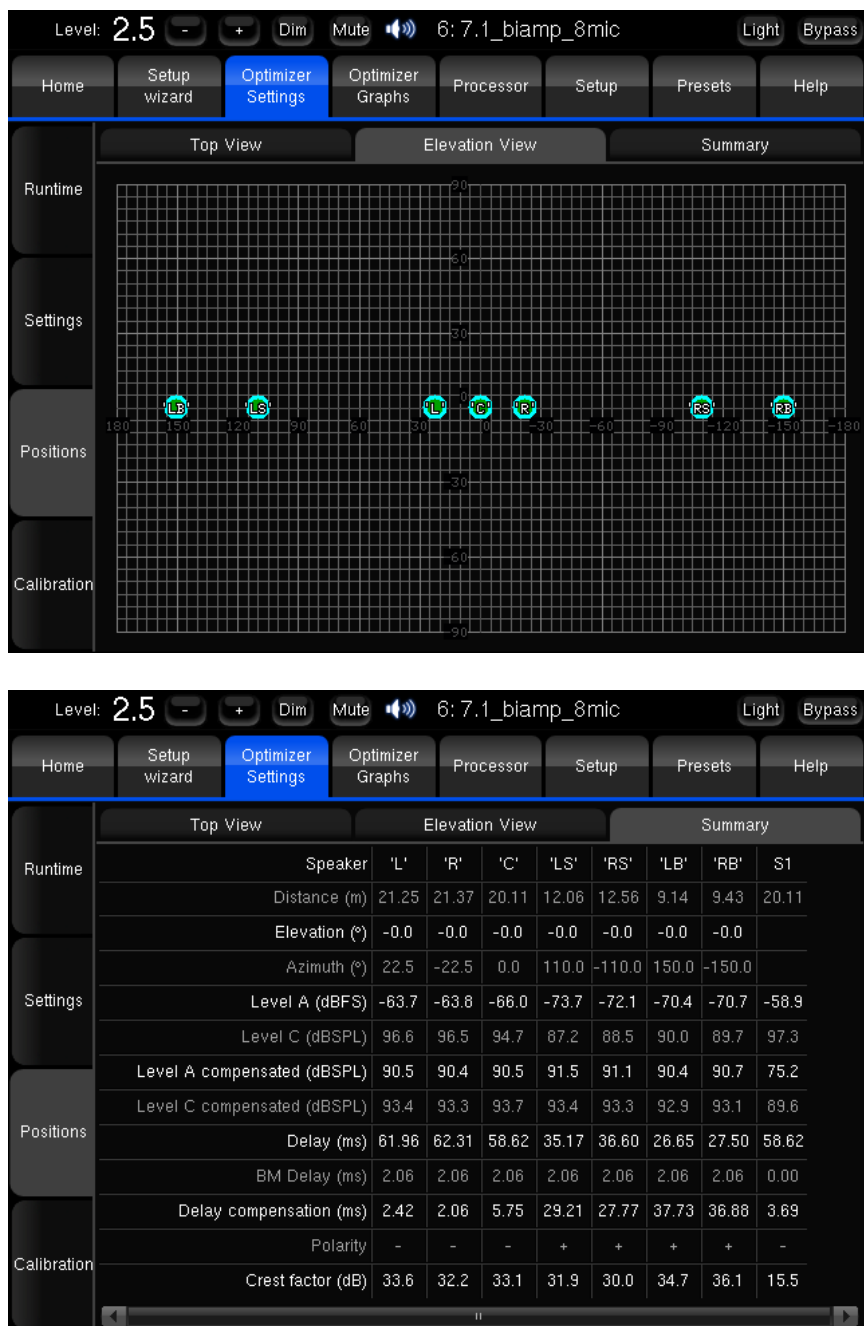
Three views are provided about the loudspeaker positions and additional details:

- The **Top view** tab: loudspeakers are seen from above,
- The **Elevation view** tab: Loudspeakers are seen from the listening point. Relative degrees of elevation and azimuth are displayed,
- **Summary** tab: a table listing the measured distance, elevation, azimuth, level, delay etc...

By default, the *Optimizer Settings/Positions/Top View* tab always displays the reference loudspeaker positions in green color. These positions are determined by the input format of the active source (Stereo, 5.1 ITU, 5.1 SMPTE or other). **Note that RT60 measurement is indicated in this page.**



The Top View (above) can be used in 2D measurement. Elevation View (below) needs the 3D microphone to be used in cinema theaters.



Measurement Summary:

- **Speaker** is the list of the speakers labels,
- **Distance** is the distance of the speaker (in meters) to the measurement point,
- **Elevation** is the elevation of the speaker (in degrees) to the measurement point (only for 3D measurement),
- **Azimuth** is the azimuth of the speaker (in degrees) to the measurement point (only for 3D measurement),
- **Level A** corresponds to the A-weighted level of the speaker in dBFS,
- **Level C** corresponds to the C-weighted level of the speaker in dBSPL,
- **Level A compensated** corresponds to the A-weighted level of the speaker in dBSPL after applying gain compensation and filtering,
- **Level C compensated** corresponds to the C-weighted level of the speaker in dBSPL after applying gain compensation and filtering,
- **Delay** corresponds to the distance of the speaker in time unit (ms) with a speed of sound = 343 m/s,

- **BM Delay** may be added to the subwoofer to improve the crossover with the satellites. More rarely, if the subwoofer is in advance, all the satellites are delayed,
- **Delay Compensation** is the delay that is added to every speaker to time align the system,
- **Polarity** specifies whether the Optimization will invert the polarity of the speaker. *Please note this is not the “measured polarity” of the speaker,*
- **Crest Factor** helps evaluating the quality of the measurement. **It should be higher than 30dB for front and surround speakers.** Low values could indicate background noise problems. Sub should be more than 15dB.

5.4.18 Optimizer Settings/Calibration

The Ovation allows to calibrate the B chain of the room with an easy and quick calibration procedure in 5 steps called **Setup Wizard**. For more details, please refer to §5.4.15.

You can enlarge your listening area by measuring your system in different points or series of points, a special algorithm finding automatically afterward the best compromise between the ideal corrections in these points.

In the *Optimizer Settings/Calibration* tab, you will find the list of all the measurements of the preset (one per line). A measurement contains the responses recorded during the test sound sequence by one microphone or several microphones placed at different points in the room. You can calibrate the selected measure by using the “**Calibrate**” button at the top of the page. During Calibration, the name of every speaker/channel is displayed while playing the MLS sequence.

Before the measurement, the calibration status is always: **No!**

Once every measure has been calibrated, the calibration status can be: **Yes** or **Partial**.

If the status is **Yes**, that means that all channels have been well measured for all microphones and then you can compute the acoustic correction filter that will best fit every point, accordingly to their weight.

The **Partial** status means that measurement went wrong on one (or few) channels and/or one (or few) microphones. Then you must find the cause of the issue and then calibrate again till you will obtain a good measurement. **Computing a bad measurement won’t give you the highest quality of calibration you could expect to have with the Optimizer.**

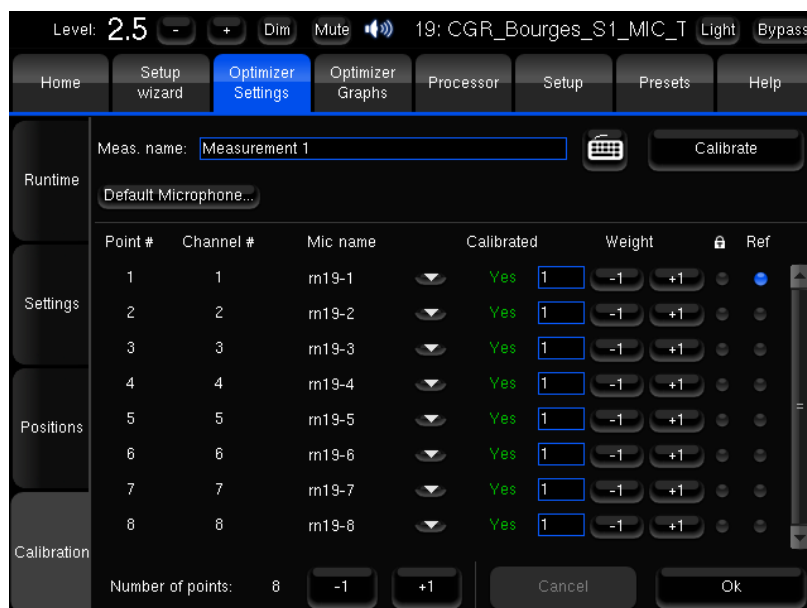
Each measurement can be renamed by typing a new name in the blue window or using the virtual keyboard.



List of measurement points:

Using the “**Configure**” button, you open the list of points that compose the measurement. You can add or remove points by pressing the “+1” or “-1” button at the bottom of the page.

This will increase or decrease the number of points indicated.



Please note that among the list of available microphones should only appear the microphones you have purchased and which are therefore linked to your processor. The Default microphone can be chosen by pressing the Default Microphone button.

The next column, called “**Calibrated**”, indicates the calibration status of each point. Three labels are possible: **Yes**, **Partial** or **No!**:

- **Yes** means that all channels have been correctly measured for this point,
- **Partial** means that some channels haven’t been correctly measured for this point. Fix the issue and recalibrate.
- **No!** means that no channel has been correctly measured for this point. Fix the issue and recalibrate.

The “**Weight**” column allows you to set a weight to each point, from 0 to 100, to emphasis for example some central points. You can bypass some points by setting their weight to 0, what can be useful in case one of the points have failed the calibration but you don’t want to perform a calibration again.

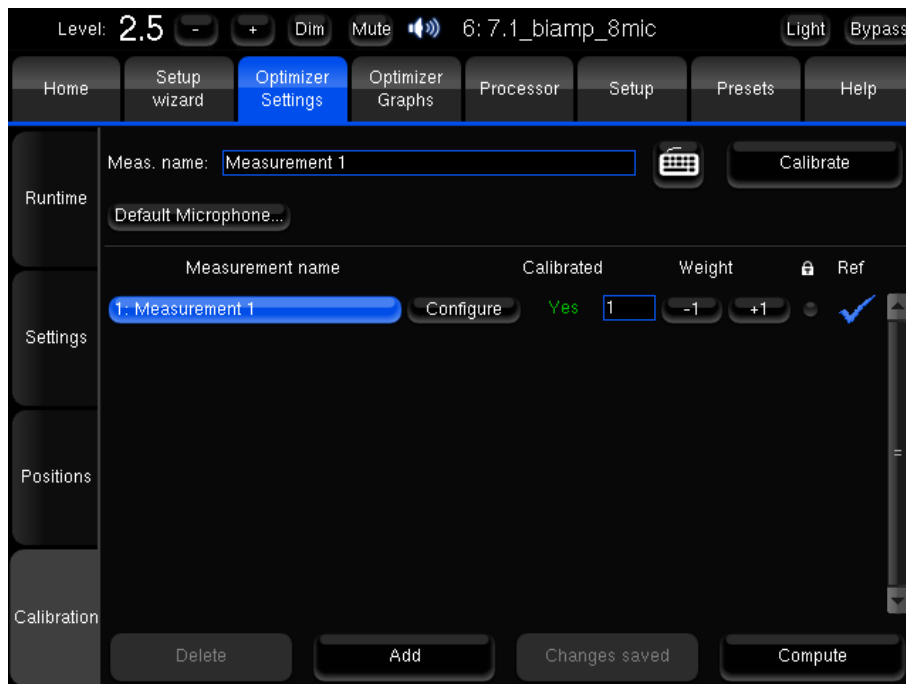
The weight can be adjusted after a calibration; you don’t have to run a complete calibration to take weight changes into account, just to press once the “**Compute**” button. If using your processor with an external keyboard, you can adjust the weight with the up (+1) and down (-1) arrows, or the page up (+10) and page down (-10) keys.

The **lock** will allow you to calibrate some point without losing the data you already recorded for this point (not functional yet in this version)

Ref is the unique reference point for all points of all measurements. That’s the point the delay and level correction are computed from. Default setting is Point #1.

“**Ok**” allows to keep your changes and go back to the measurement list. “**Cancel**” allows to go back to the measurements list without keeping the changes.

Measurements list:



You can add a new measurement with “**Add**” button at the bottom of the page, or delete the selected measurement by pressing “**Delete**”. When you add a new measurement, the points configuration is copied from the selected measurement, assuming you may want to run a new measurement of the same system with the same equipment (of course you can change these parameters). The removal of a measurement cannot be cancelled. You can also rename a measurement by editing the blue text box “Meas. name”.

For each measurement, the “**Calibrated**” column indicates “**Yes**” if all the points containing the measurement have been correctly calibrated, “**No!**” if none have been, and “**Partial**” if some points have successfully passed the calibration while some other did not, or if you added some more points after the calibration.

As for each point, you can set a **Weight** to each measurement; the final weight of each point will be the product of its own weight and the weight of its measurement. Setting a measurement’s weight to 0 will bypass it. The weight can be adjusted after a calibration but requires to be recomputed.

The **lock** disables measurement settings edition.

The “**Ref**” column indicates in which measurement the reference point is. This can be changed by pressing the “**Configure**” button of the measurement that contains the point you want choose as new reference or just by choosing the reference measurement point of your choice in case it contains only one point.

All the measurements with non-null weight should be fully calibrated before to compute the acoustic correction filters by pressing the “**Compute**” button. If some are partially or not calibrated at all, you will have fix the problem, remove the bad points or their entire measurements, for example by setting their weight to 0, before to be able to compute the acoustic filters.

Before saving your preset in the “**Presets**” page, you have to save the multipoint parameters by pressing the “**Save changes**” button. These parameters are automatically saved (and the button disabled) when you leave the point list, and when you run a calibration or a computation.

Please note that if computation is not performed after a partial or full calibration, the notification bar will show a red “NOT READY” message indicating that the outputs of the processor are muted.

5.4.19 Setup Wizard

The Setup Wizard page includes 5 tabs (Step 1 to 5) that are the 5 easy and quick steps to calibrate the sound of the B chain.

Select the **Step 1** tab, if another step was already selected.



5.4.19.1 Step 1:

Please follow the instruction and firstly mute the inputs by clicking on 'Inputs Mute' button. The input mute is active when the button is blue.

It won't be possible to launch the Pink Noise during the step 3 if the 'Inputs Mute' hasn't been selected in the step 1.

The following logo  appears in the upper banner as long as the 'Inputs Mute' is active.

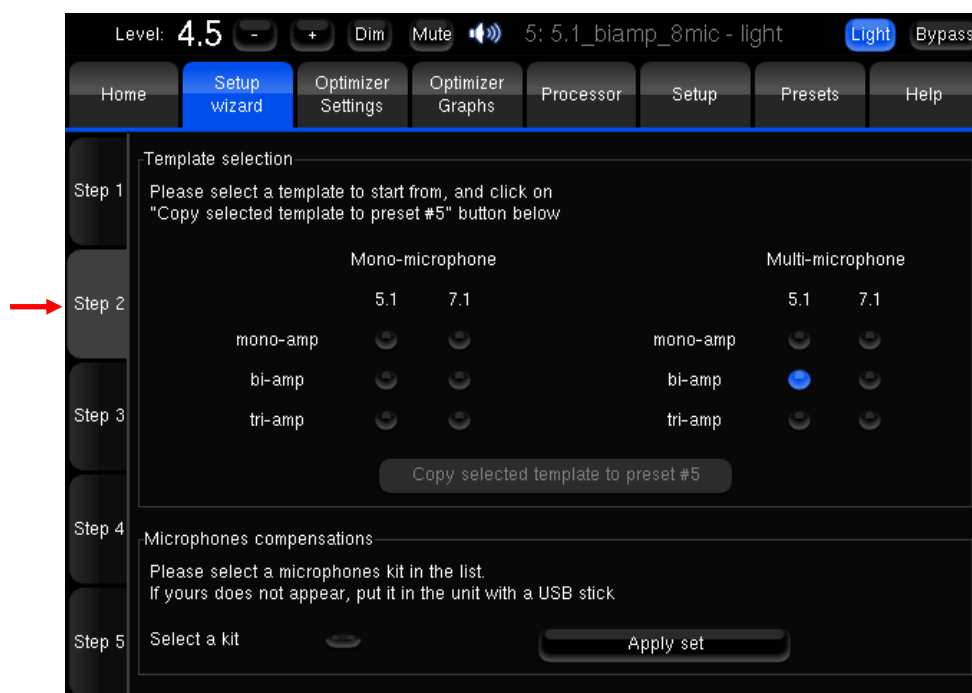
Disable the 'Light mode'. 'Light mode' is disable when the button is black.

NOTE: in the processor normal mode, the Ovation will compute all graphs when you'll load presets or Profiles. This operation can take a non negligible time depending on the amount of measurement points, input and output channels and can slow down the switch from a preset to another one, introducing sound cuts. 'Light Mode' allows to load quickly a preset without computing the graphs. It will be very useful when you want to quickly compare two presets before choosing the most adapted to your cinema theater.

Place the microphones in the room according to the recommendation in *Microphone placing* window.

Go to **Step 2**.

5.4.19.2 Step 2:



First of all, you must select the calibration Template, that means the configuration of your audio system and the microphone configuration for the calibration:

- calibration with 1 (Mono) or 8 (Multi) microphones,
- 5.1 ou 7.1 input audio format,
- mono, bi or tri-amplification audio system for the Left, Right and Center output channels,

then click on ‘Copy selected template to preset #5’ button to save these settings to default preset #5.

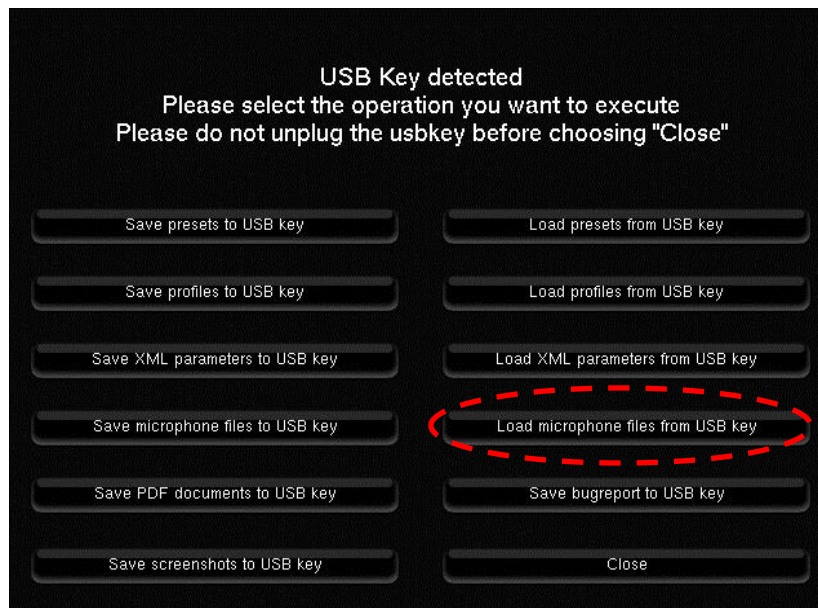
Selecting Microphones compensations file:

The Mics compensations file applies a frequency correction on each of your 8 microphones of the kit to obtain the most flat and linear response. Before starting calibration, you have to choose the Mics compensations file corresponding to the used kit in the list appearing by clicking on the small black button on the left beside the ‘Apply set’ button.

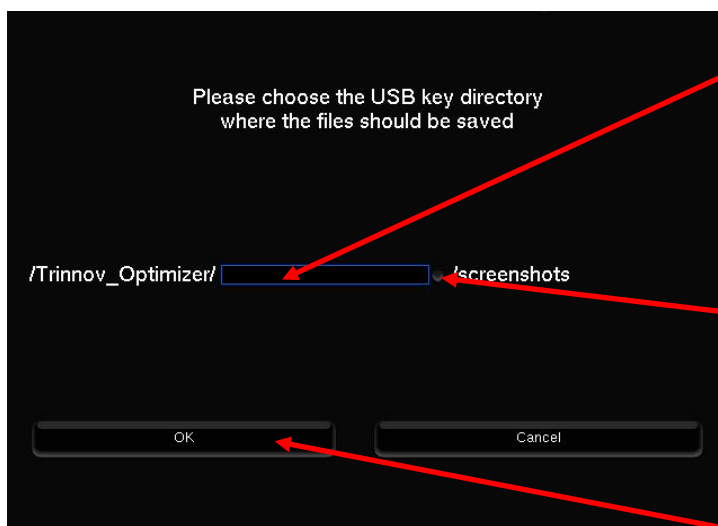
NOTE: the Ovation Mics compensation file is always named *rnXX*, with XX = kit number.

Once the appropriate file selected, click on ‘Apply set’ to confirm.

If the Mics compensations file you need isn’t already in the Ovation processor, you can load it from the USB stick included in the mics box. Plug the USB stick in the front panel USB port and select ‘Load microphone files from USB key’ in the following page that appears on screen:



After clicking on the appropriate button, the following page appears:



type the name of the folder
where the Mics compensations
file is stored on the USB stick

OR

choose the folder in the list
that will appear by clicking on
this button

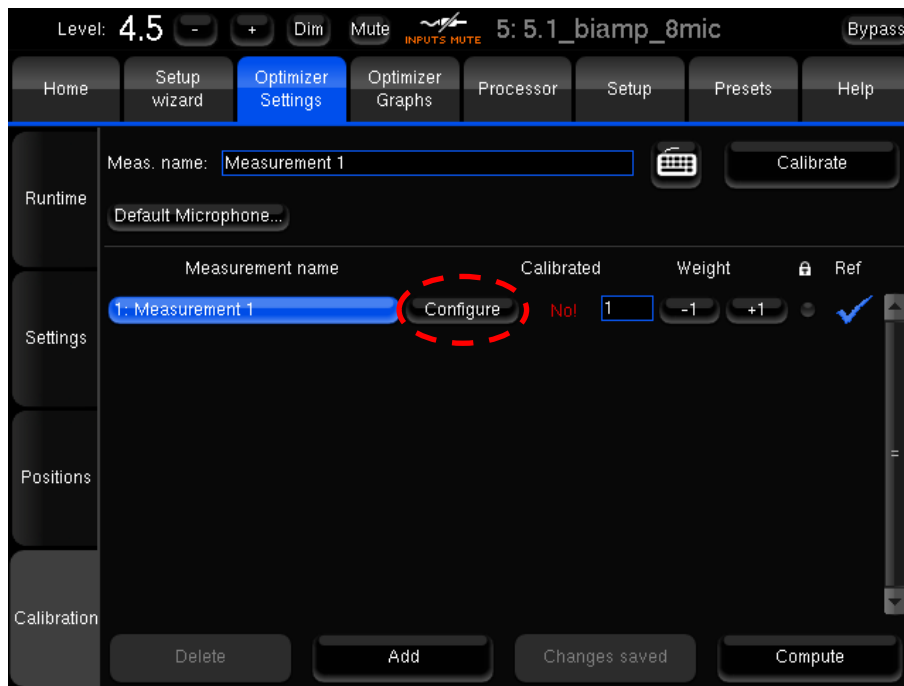
AND

click 'OK' to confirm

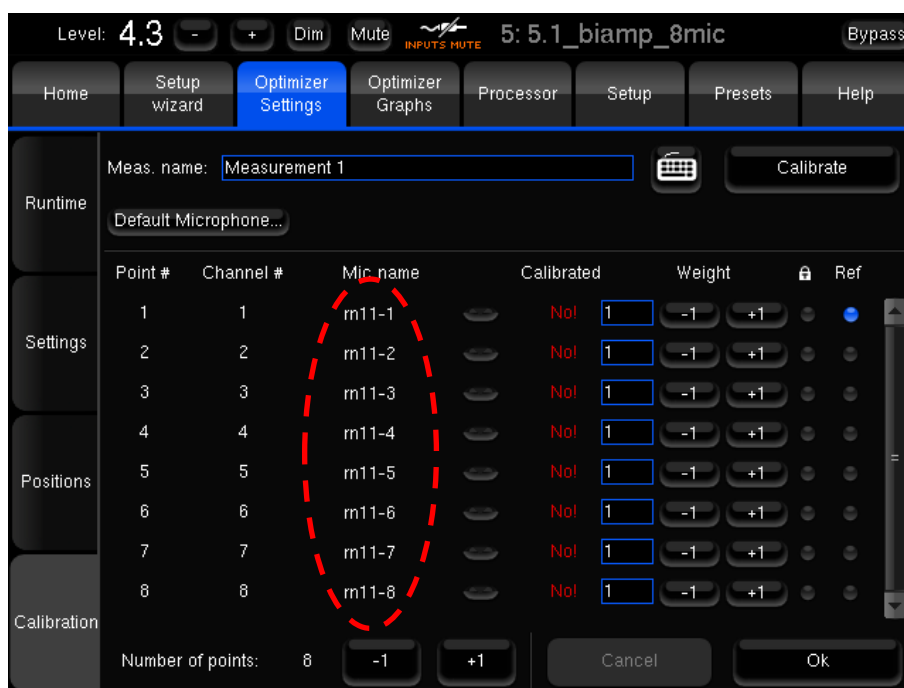
Back to the previous page, click on the 'Close' button to come back to step 2.

Click on 'Select a kit' on Step 2 page to choose the appropriate kit you currently use for the calibration. Confirm by clicking on 'Apply set' button.

Not necessary but, if you want, you can control that the 8 Mics compensations files of your kit have been rightly loaded in the processor in *Optimizer Settings/Calibration* page, by clicking on 'Configure' button:



Then you can verify than each Mic compensations file is applied to the corresponding microphone (example below: kit 11):



Go back to the *Setup Wizard* page and go to **Step 3**.

5.4.19.3 Step 3:



If the calibration devices are not installed at this step (Mics, Mic box, cables), please do it now.



Before launching the pink noise test, make sure that the output level isn't too high. Start with the output level set at 3.0 for example, then increase to 7.0.

You can launch the pink noise by clicking 'Start' button and then rise it slowly to reach the reference level (7.0). The default mode is Manual. You can switch from a channel to another with '<' or '>' buttons between which the speaker label is indicated: L, R, C, LS, RS and S1 for 5.1 format for example.

By clicking 'Auto' button, you activate the automatic pink noise (active when button is blue) which test successively all the output channels (approximately 5 sec. per channel).

In multi-amplification mode, **and for each concerned channel**, you can hear the whole enclosure by clicking 'All ways' button (active when button is blue) or each way of the selected channel with the '<' or '>' third line's buttons: High, Mid and Low.

With the Ovation output level at 7.0, set the levels of all amplifiers to obtain:

- around 65 dBC on each output channel (or each enclosure way),
- around 80 dBC on the subwoofer output channel.

A Microphones RTA analyser shows the frequency response of the calibration mics. You can select each microphone by clicking 'M1' to 'M8' buttons to control the consistency of the whole measurement points.

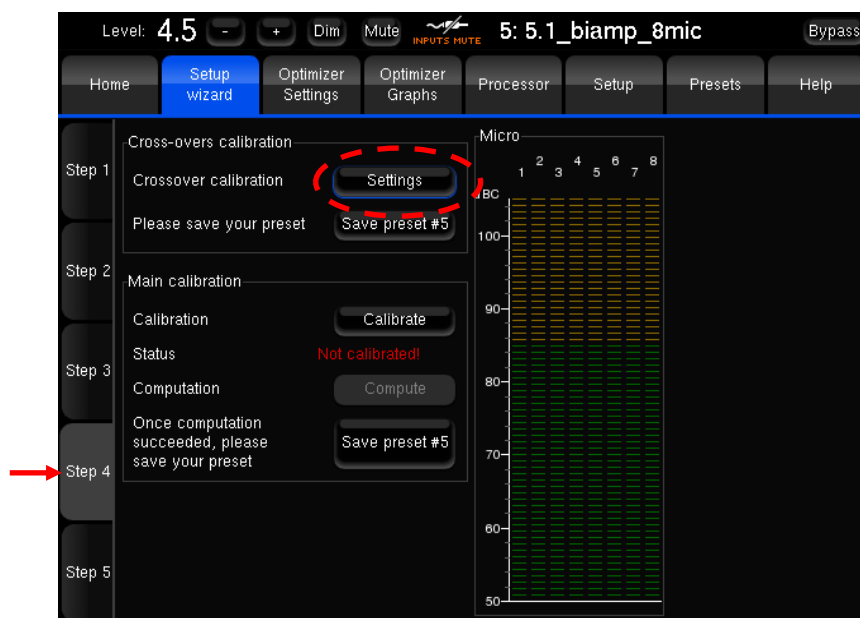
In any case, you can stop the pink noise by clicking the 'Stop' button.

Go on **Step 4** next page for multi-amps mode and **Step 4** further for mono-amp mode.

5.4.19.4 Step 4:

5.4.19.4.1 Multi-amplification

In multi-amplification mode, it's **firstly** necessary to perfectly calibrate the cross-overs filters of each screen speaker before calibrating a global output channel. You access to the Crossover calibration page by clicking 'Settings' button:



Then you reach the crossovers setting page of each output channel (Left on the following ex.):



The frequencies crossovers and filters types are set with default security values. Generally, the multi-amplified channels are the screen channels: Left, Right and Center. Multi-amps is also possible on surround and back channels.

Before launching crossovers calibration, you must type the appropriate crossover frequency values and choose the correct filters types respectively to the speaker manufacturer specifications.

NOTE: for identical enclosures, a modification of one's settings can be duplicated on the others. Before modifying the characteristics of the identical enclosures, you must link the corresponding channels. Screen speakers can be linked only by clicking on 'Link LRC' button.

Click the « Link » button on the output channels you want to link each others and validate with the «Apply » button which became highlighted.

When several channels are linked, the « Link » button is highlighted on each one.

The modifications done on the identicals channels, you must unlink them.

Click the « Link » button on the output channels you want to unlink and validate with the «Apply » button which became highlighted.

You can now proceed to the selected channel calibration. This procedure must be reconduct on each multi-amplified channel.

NOTE: for all multi-amplified speakers (Left, Right and Center generally), it is recommended to use identical amplifiers and to set them at the same level.



To avoid any risk of enclosures damage due to high output levels of the amplifiers or audio processor, set the Ovation output to low value and adjust it slowly to the appropriate level. Start at 2.0 and rise progressively the level between 5.0 and 6.0. A calibration at 7.0 is possible but could be very loud.

If the MLS sequence level is too low for the measurement, the following message will appear on Ovation and computer interface's screen: 'Crest factor too low for capsule X' (X is the microphone number).

This message will disappear if you click on it or when the measurement level will be enough for the processor after you've rised the output level. This message could appear again until a correct measurement level.

Once the crossover calibration procedure has finished on the active channel, a confirmation page is shown and you can see the results on the following graphs and bring some modifications if necessary.

The « **Crossovers filters** » graph shows the result of the electrical frequency bands summation considering the parameters introduced.

This graph can be modified by the level adjustment, the crossover frequency value and the filter type: Bsl (Bessel), L-R (Lindquist Riley) and Bwth (Butterworth) with the appropriate order.



The « **Impulse response** » graph shows how each speaker way is responding to an impulse signal. Then you can see whether the drivers are correctly synchronized and their respective levels. You can adjust the parameters to synchronize the graphs the best as possible.

This graph can be modified by Delay time and Level value.



The « **Amplitude (power & direct)** » graph shows the recombined amplitude response of the speaker: you can see whether the combination of the drivers is constructive, and you can observe the effects of level/delay/polarity modification on the combined amplitude of the speaker.

Two curves are displayed, one (cyan) showing the global power of the speaker (including the room), and one showing the amplitude of the direct front and early reflections (magenta). Comparing both curves will indicate you whether the crossover conserves the directivity of the speaker: the more alike the two curves look, the more directive the speaker is towards the listening spot.

This graph can be modified by Level value, Delay time and polarity.



When you have found the best compromise of your settings for the current channel, then you can calibrate the next one.

Once you have finished the calibration of all the multi-amplified channels then click on the '*Back to Wizard*' button to come back to page « Setup Wizard/Step 4 ».



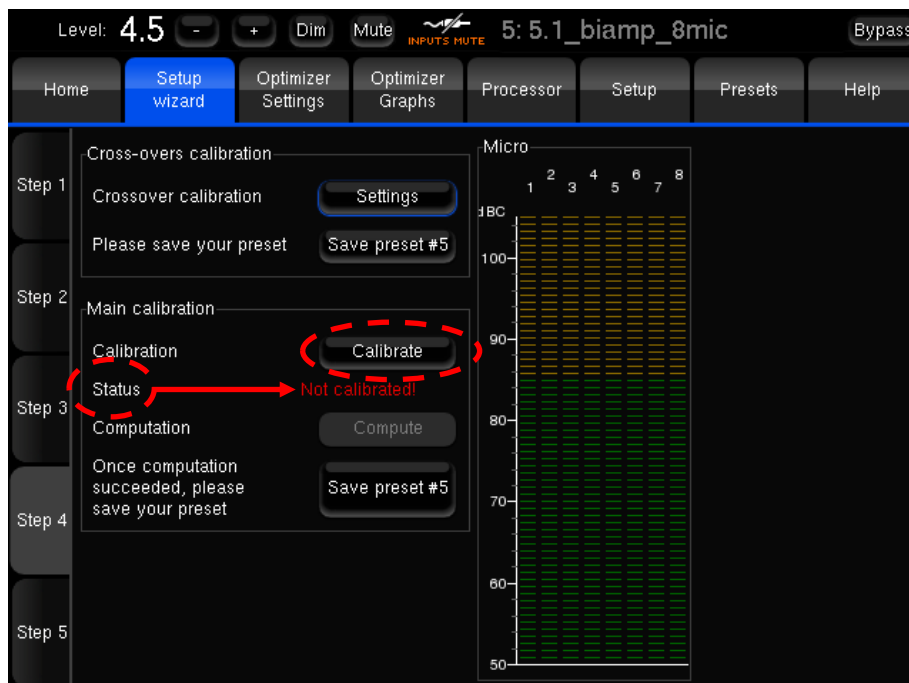
Save the crossover calibration by clicking on the '*Save preset #5*' button.

Preset #5 is always the default preset to save your settings.

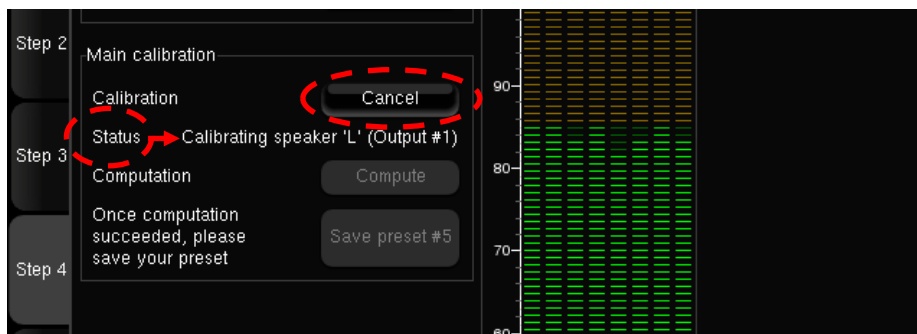
The calibration of each output channel will now begin as indicated in the following pages whatever you are on mono or multi amplification.

5.4.19.4.2 Mono-amplification

In mono-amplification, or once the crossover calibration's achieved, launch the MLS calibration sequence by clicking on 'Calibrate' button (Calibration status must be **Not calibrated !**):



At any moment you can stop the calibration sequence by clicking on the 'Cancel' button that appears in place of the 'Calibrate' button. 'Status' will change and indicate the channel that is currently calibrated.



During the calibration, the following logo appears in the upper banner:



If the MLS sequence level is too low for the measurement, the following message will appear on Ovation and computer interface's screen: 'Crest factor too low for capsule X' (X is the microphone number):

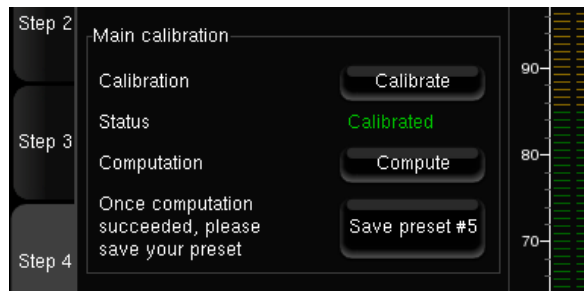


This message will disappear if you click on it or when the measurement level will be enough for the processor after you've rise the output level. This message could appear again until a correct measurement level.

NOTE: at the end of the calibration sequence, if the status field indicate '**Partially calibrated**', that means one or few measurement points has not be rightly calibrated.

It can be a cable damage, dropped microphone So you must fix the issue and launch a new calibration sequence.

When the calibration sequence achieve without problem for all the outputs channels, the status field will indicate '**Calibrated**':



By clicking on 'Compute' button, the processor will now calculate all the optimisation filters.

During the filters computing time, the following logo appears in the upper banner:



During the graphs's compute, the following logo appears in the upper banner:

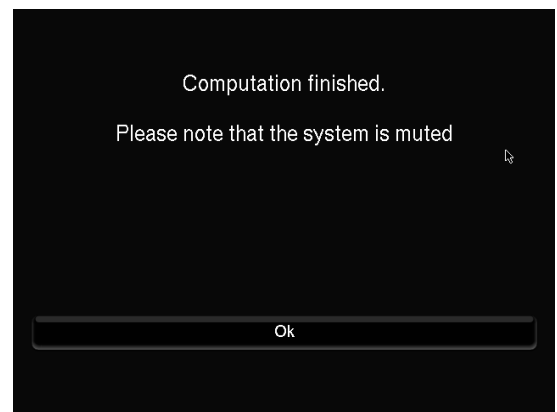


Depending on the input format (5.1, 7.1, ...), the amount of measurement points and the amplification mode (mono, bi or tri), the computing time differs.



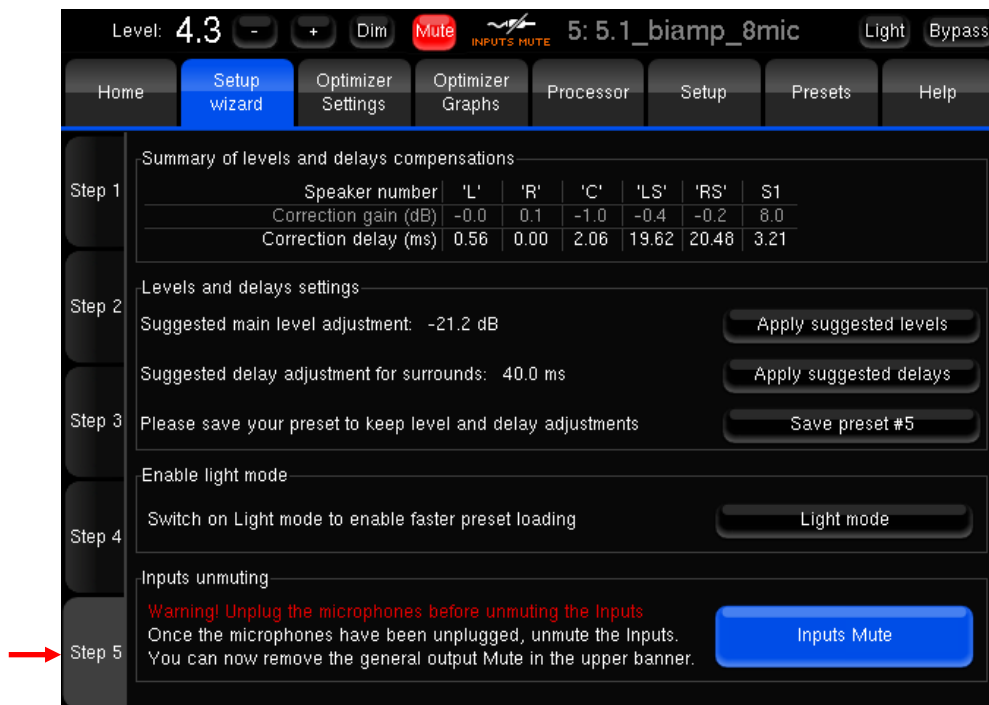
Once the computation has finished, you must click on 'OK' button to validate, and save it by clicking on 'Save preset #5' button on the « Setup Wizard/Step 4 » page.

Go on **Step 5**.



5.4.19.5 Step 5:

First of all, step 5 shows a summary of levels and delays compensations for all speakers. In the following example (5.1 format), the compute suggest a +8.0dB gain correction and a 3,21ms delay on the subwoofer output channel:



This array allows you to verify the consistency of the audio installation, or in the opposite, identify anomalies due to the cinema theater or to the measurement points.

In this case, it is necessary to analyse these anomalies and try to reduce them the most as possible or to cancel their causes. Then you can calibrate once again your cinema theater and note the improvements of your actions on the calibration result.

If everything's OK, click on the 'Apply suggested levels' and 'Apply suggested delays' buttons to apply the suggested adjustments and **finally save these settings by clicking on the 'Save preset #5' button.**

NOTE: If you want to compare several presets with different parameters and settings, copy the preset #5 on another emplacement with an appropriate name before starting a new calibration procedure.

Then you can switch from a preset to another by simply clicking on the appropriate preset button in the « preset » page and listen to the differences and make your choice. Being in 'Light mode' allows to switch quicker from a preset to another like it's explained in the §2.4.1 NOTE.

NOTE: At step 5 the microphones phantom power is off, so you can let the microphones plugged and unmute the inputs without danger for speakers.

If you want to listen some audio contents, you must firstly unmute the inputs by clicking on the 'Inputs Mute' button and unmute the outputs by clicking the 'Mute' button in the upper banner.

5.5 Reducing latency

For some applications, latency higher than a few milliseconds can be problematic. Latency highly depends on two parameters: the buffer size and the sampling frequency.

With standard optimizer settings, the following values can be used as reference:

Sampling Frequency	Buffer Size	Latency
44,1 kHz	256	28,2 ms
44,1 kHz	512	44,4 ms
48 kHz	256	23,2 ms
48 kHz	512	25,4 ms
88,2 kHz	256	20,1 ms
88,2 kHz	512	28,2 ms
96 kHz	256	17,4 ms
96 kHz	512	23,2 ms

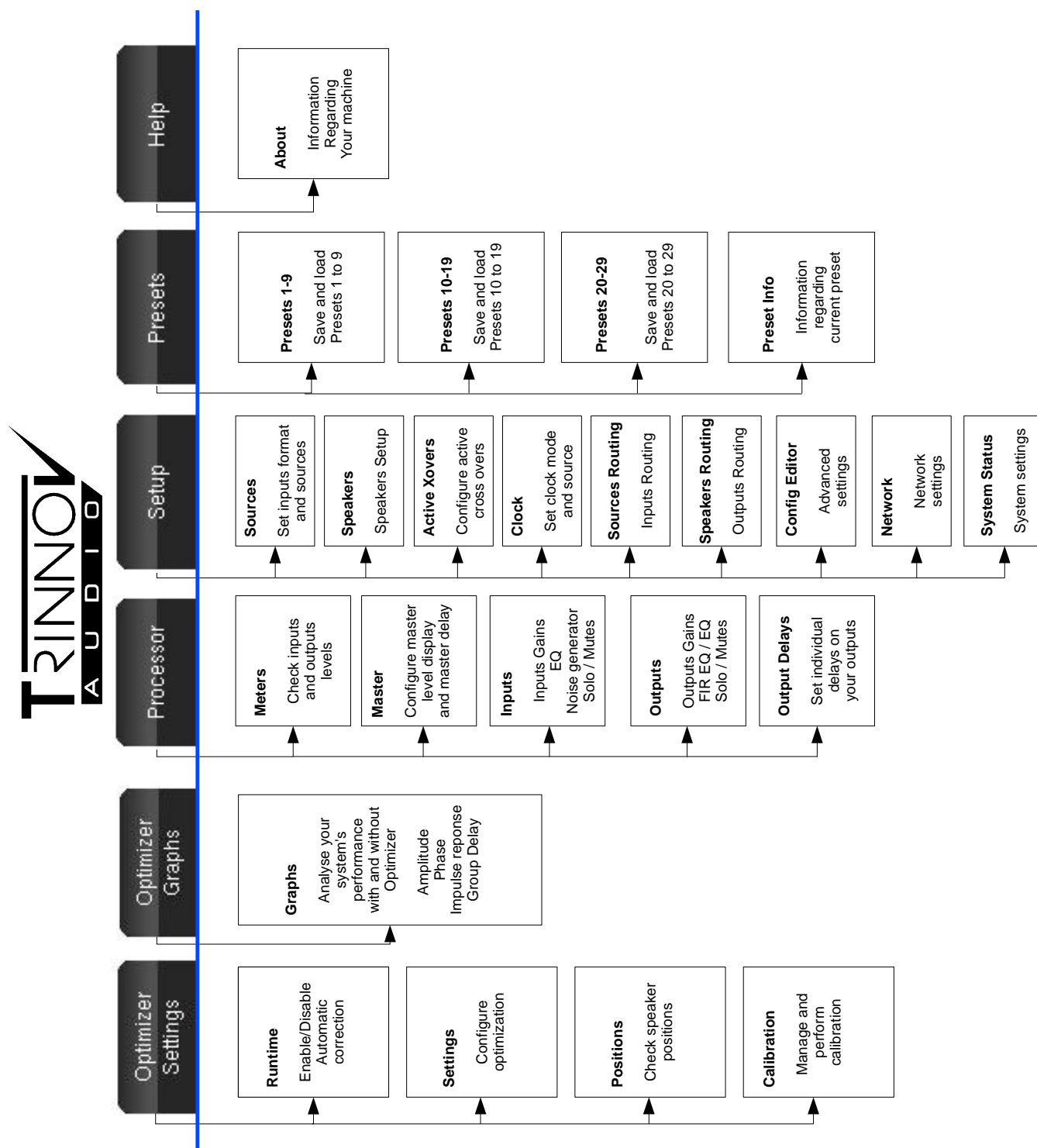
To reduce latency (**only if it's necessary !**):

- You can change the sampling frequency and the buffer size of your unit according to the table above. This can be done in the *Setup/Clock* page. Changing the buffer size requires rebooting the processor. Switching to a smaller buffer size can result in clicks and sync loss depending on the amount of CPU resources of your unit.
- In the *Optimizer Settings/Settings/Main settings* tab you can also try to change the Optimize setting to “Amplitude” or “Low range only”. Don’t forget to hit “Apply changes”.
- If no latency is allowed, you can just **Bypass** the Optimizer using the bypass button in the upper bar at the top of the screen. The resulting latency is the AD/DA conversion latency only as the processing section is bypassed.
On the other hand, turning the Optimization Off in the *Optimizer Settings/Runtime* page does not reduce latency since the audio still pass through the PC.

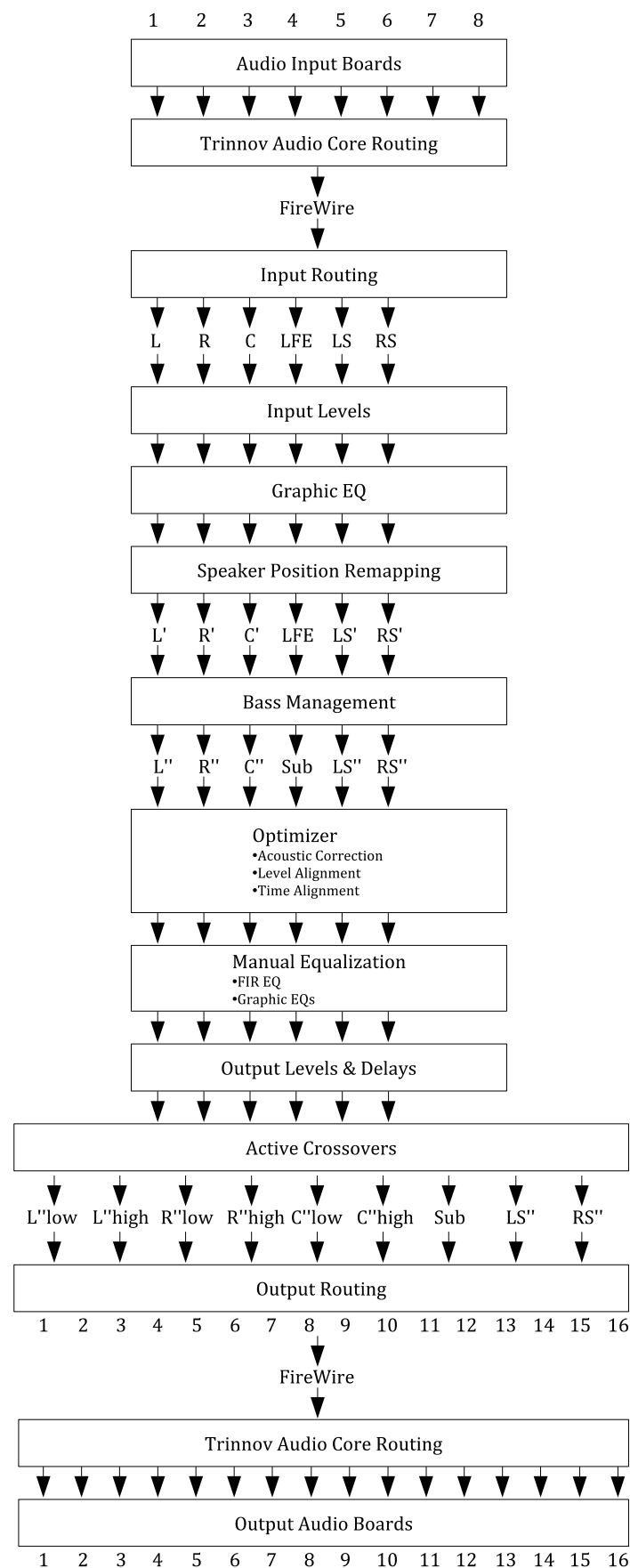
Note: Latency can be monitored in details in the *Processor/Master* page.

6 Appendix

6.1 Arborescence of the menus



6.2 Signal Flow of the Ovation



6.3 Bypass

The following table describes which parameters are affected by the different bypass modes:

	Optimizer Settings/Runtime				Notification bar
	Acoustics Correction Off	Level Alignment Off	Delay Alignment Off	Optimization Off	Bypass
Optimization (Automatic FIR + IIR)	Off	On	On	Off	Off
Level Alignment	On	Off	On	Off	Off
Delay Alignment	On	On	Off	Off	Off
Solo/Mute	On	On	On	On	On
FIR EQ	On	On	On	On	Off
Graphic EQ	On	On	On	On	Off
Inputs Gain	On	On	On	On	On
Outputs Gain	On	On	On	On	On
Outputs Delay	On	On	On	On	Off
Bass Management	On	On	On	On	Off
BM Trim	On	On	On	On	Off
Input Routing	On	On	On	On	On
Output Routing	On	On	On	On	On
Active X-Overs	On	On	On	On	Mutes the outputs
In-Out Latency	Unchanged	Unchanged	Unchanged	Unchanged	2 frames