V. 3.8

magnitude 32

USER GUIDE

magnitude 32

TRINNOV
AUDIO

Last Update: 25 March 2014

IMPORTANT SAFETY INSTRUCTIONS

1. Read the following instructions carefully. Save all instructions for future reference.

2. Follow all warning and instructions.

3. TRINNOV Audio expressly forbids unauthorized modification of this equipment.

4. Using the unit in the following locations can result in a malfunction:

   . In direct sunlight
   · Locations of extreme temperature or humidity
   · Excessively dusty or dirty locations
   · Locations of excessive vibration
   · Close to magnetic fields

5. Condensation can form on the inside of the apparatus if it is suddenly 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.

6. Clean only with a dry cloth. Do not use liquid solvent-based cleaners.

7. Do not cover of bloc ventilation slots or openings. Never push objects of any kind into ventilation slots on the equipment casing.

8. Install in conformance with the manufacturer's instructions.

9. Maximum permissible operating conditions: 0°C to 40°C, 20-65% relative humidity.

10. Protect the power chord from being walked on or pinched particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.

11. Always replace damaged fuses with the correct rating and type.

12. Unplug this apparatus during lightning storms or when unused for long periods of time.

13. Do not open the equipment case. There are no user serviceable parts in this equipment. Refer all servicing to qualified service personnel.

14. Please connect the designated AC/AC power supply to an AC outlet of the correct voltage. Do not connect it to an AC outlet of voltage other than that for which your unit is intended.

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 sufficient magnitude to constitute a risk of electric shock to persons.

Contents

1 INTRODUCTION

Thank you for choosing a magnitude32.

As a company, our goal is to offer a range of products that meet the requirements of the most demanding audio professionals and High-End enthousiasts.

The magnitude32 has been designed to achieve multichannel audio reproduction at the highest level of performance with speaker configurations ranging from 5.1 up to the most complex 3D audio systems. The magnitude32 can integrate any existing home cinema audio system but it may also be considered in the early stages of a home cinema design, as a fully dedicated loudspeaker/room optimization system.

The magnitude32 features the company's world-acclaimed Optimizer technology and Trinnov's unique 3D loudspeaker remapping.

1.1 THE OPTIMIZER IN A NUTSHELL

The magnitude32 features Trinnov's Loudspeaker/Room integrated correction technology and can deliver from 8 up to 32 simultaneous channels of optimized sound to your home cinema.

Based on acoustic meaurements performed in a single or multiple position across the room, the system provides different types of compensation and information at once:

Level and Time Alignment

The Optimizer considers a user-defined reference measurement position to automatically align 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 measures the absolute playback level.

Automatic Optimization

The Optimizer uses state-of-the art time-frequency algorithms to analyze acoustic 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 to comply with industry standards such as 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 to fine-tune the sound according to loudspeaker capabilities and listening tests.

Intelligent Active Crossovers

The Optimizer provides a semi-automatic calibration engine of 2-way, 3-way and 4-way active crossovers. These crossovers may be used on up to 32 output channels. This makes Magnitude32 a comprehensive equalization and crossover solution for high-end sound systems.

Bass Management

The magnitude32 supports an unlimited number of subwoofers, mono and stereo bass management. The bass management crossovers filters and the subwoofers delays are automatically computed by the Optimizer after calibration based on user-defined crossover frequency.

Loudspeaker Positions Remapping

Thanks to its measurement probe with 4 capsules, the Optimizer determines *in 3D* the relative position of each loudspeaker. From the Input Format setting, the Optimizer knows the reference loudspeaker positions defined by the standard for the target system, for example ITU 775. The Optimizer computes the *remapping matrix* that must be applied to the input signal to create the same acoustic field that would be obtained if the speakers were positioned correctly.

For an in-depth description of Trinnov's remapping technology, please read the AES Paper 6375, *Reproducing Multichannel Sound on Any Speaker Layout,* available in the download section of our website at **www.trinnov.com**.

1.2 ADDITIONNAL FEATURES OF THE MAGNITUDE32

The Magnitude32 also provides all the basic functionalities expected from a surround sound processor:
 • routing and level adjustment of inputs and outputs (Audio Matrix functionality)
 • remote control options via Ethernet, RS232 or through automation system modules.
 • presets save, backup and restore
 • basic peak level metering of inputs and outputs
 • calibrated, global gain/volume adjustments

Manual Equalization

The magnitude32 provides a set of manual equalization tools in addition to the Optimizer, to allow manual fine-tuning of the automatic correction results. This is usually done by using a separate acoustic measurements system.

Two different tools are offered:
 • 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: multiple layers of 31 band Graphic Equalizers are also provided in order to support established methodologies and standards.

2 GETTING STARTED

2.1 POWER ON AND SHUTDOWN

Power on

To power on the Magnitude32, first press the back panel power switch to supply the processor with electricity. The front power button shall then be used to start the processor. It should illuminate **after a few seconds.**
The system takes approximately 45 seconds to initialize.

IMPORTANT NOTE
Don't forget that the speakers/amplifiers should always be powered up last.

Shutdown

Push the front panel power switch to initiate normal shut down.
The system will need 10 seconds until complete stop.

NOTE
It is not recommended to cut the AC power (back panel), as the system saves several "last used" settings.

2.2 REMOTE ACCESS TO THE GRAPHICAL USER INTERFACE

The Magnitude32's open architecture offers a variety of solutions to access the user interface:
- from a VNC Client running on remote device (laptop, smartphone or tablet) connected through the network to the Magnitude32
- from an external monitor with a keyboard and a mouse.

About the VNC protocol

The magnitude32 has a built-in VNC Server that allow you to fully control the processor from any VNC client host device over the network.

VNC is a graphical desktop sharing system that transmits the keyboard and mouse events from one computer (server) to another (client), relaying the graphical screen updates back in the other direction, over a 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…), provided that both the server (magnitude32) and client (remote control device) are part of the same network.

Network configurations

The magnitude32 has built-in Ethernet and optional WiFi to perform various actions over the network.

The magnitude32 can simultaneously be part of a network as:
- a hard-wired client to join an existing network through an Internet Service Provider (ISP) Box / router.
- a WiFi Access Point (AP) to create its own WiFi network.
- a WiFi client to join an existing wireless network.

IMPORTANT NOTE
The USB WiFi dongle must be pluged to the magnitude32 before switching on the unit.

Using the magnitude32 as a DHCP client

The default Ethernet mode of the magnitude32 is set to DHCP client, meaning that it will automatically be detected and configured on most domestic networks as the magnitude32 is connected to the Ethernet.
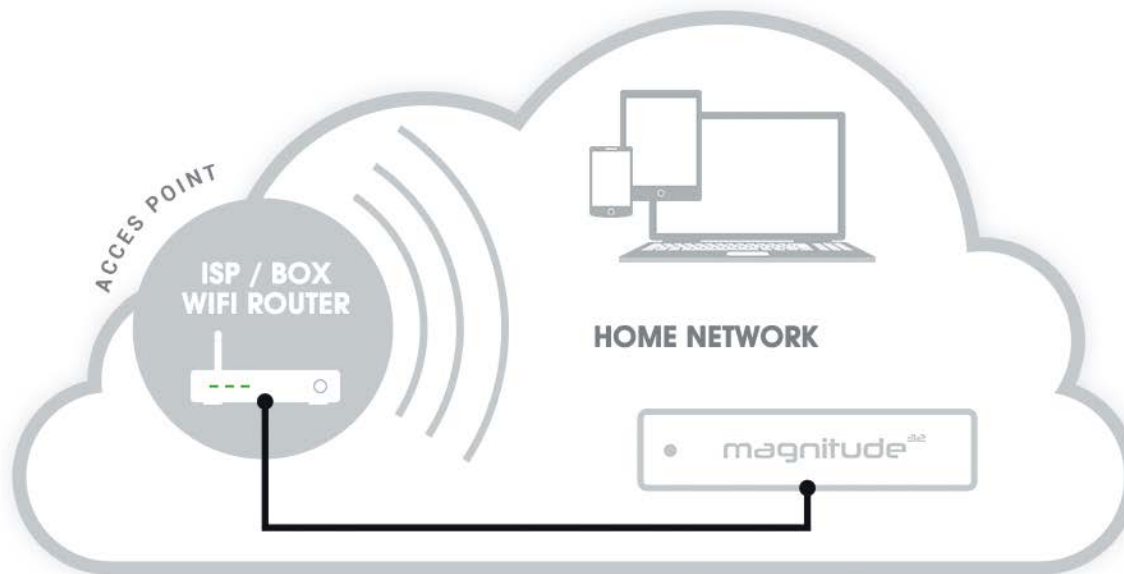
The figure above shows that the magnitude32 is hard-wired to the Internet Service Provider Box / WiFi Router of the local home network.

The magnitude32 can then be remotely accessed wirelessly from wifi devices connected to the local WiFi access point.

Using the magnitude32 as a WiFi Access Point
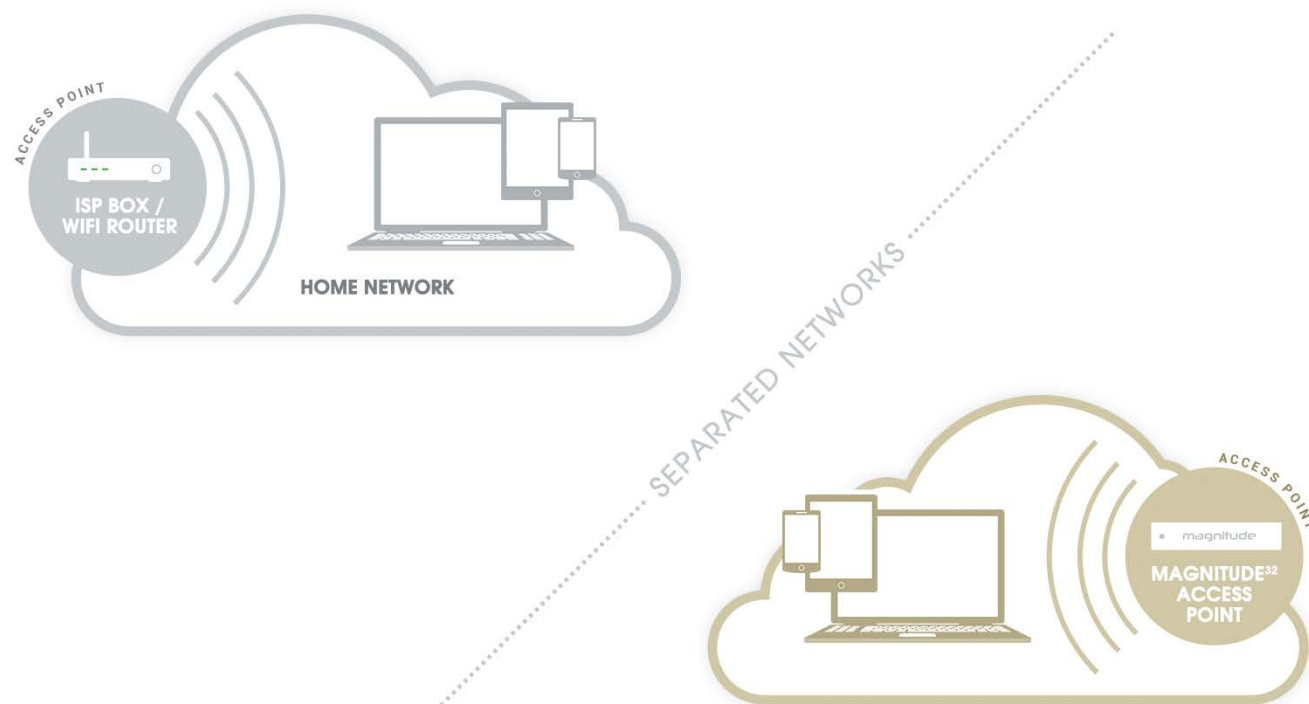
The default WiFi mode of the magnitude32 is set to Access Point.
Provided that the optional USB WiFi dongle is connected before starting the magnitude32, the magnitude32 will automatically create iits own WiFi Access Point. Once connected to the magnitude32 Access Point, it is possible to remotely access the magnitude32's graphical user interface but access to Internet will be lost, as indicated in the figure above,.

IMPORTANT NOTE
*The Access Point password is **calibration**.*

Using a VNC Client through the local network
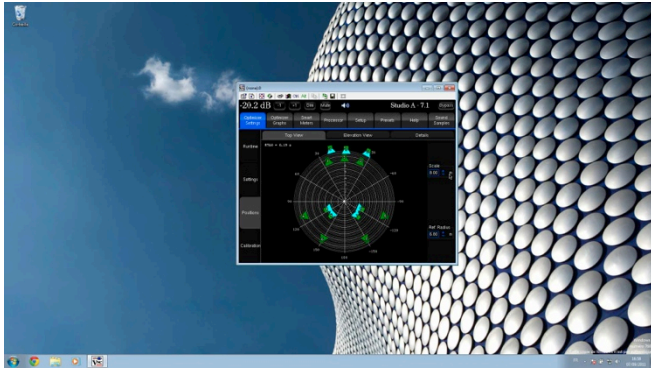
VNC control from Windows XP   *VNC control from Mac OS X*
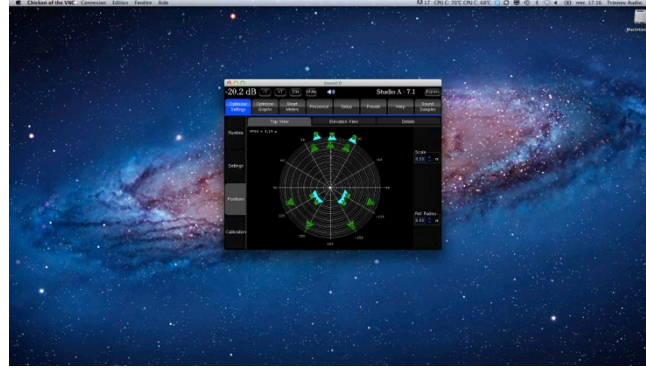
You can find different VNC Clients/Viewer on the Internet. As examples, we've chosen "TightVNC" for Windows 7, "Chicken of the VNC" for Mac OS X, and "Mocha VNC Lite" for iPad / Android.
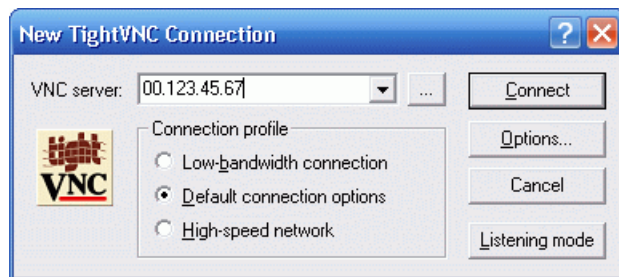
"TightVNC" Login panel (Windows)   *"Chicken of the VNC" Login panel (Mac OS X)*
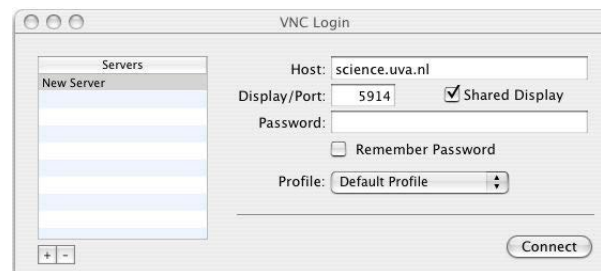
"Mocha VNC Lite" panel (iPad)

As you launch the VNC client, you will attempt to establish a connection to the server. The VNC Client requires the server address and a password:

- The server address is IP address of the Magnitude32.
- The password consists of:
  - o   the 6 digits serial number labelled on the back panel.
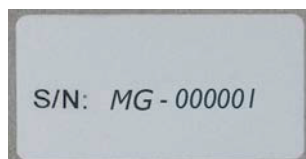
  - o   or the 7 digits "**Product ID**" number displayed in the **Help/About** page.

Depending on the VNC client, the magnitude32 is automatically recognized as an available server.
The IP address can therefore be retrieved from the VNC client using:
- The blue arrow, located at the right of the VNC Server address field in *"Mocha VNC Lite"*.
- As the magnitude32 is selected from the server list in *"Chicken of the VNC"*.

If the IP address of the magnitude32 is not automatically displayed or available, it can be retrieved from the router administration page, where the IP address of the DHCP clients are listed, or in the **Setup/Network** page of the graphical user interface, displayed to a monitor via the VGA or DVI port.
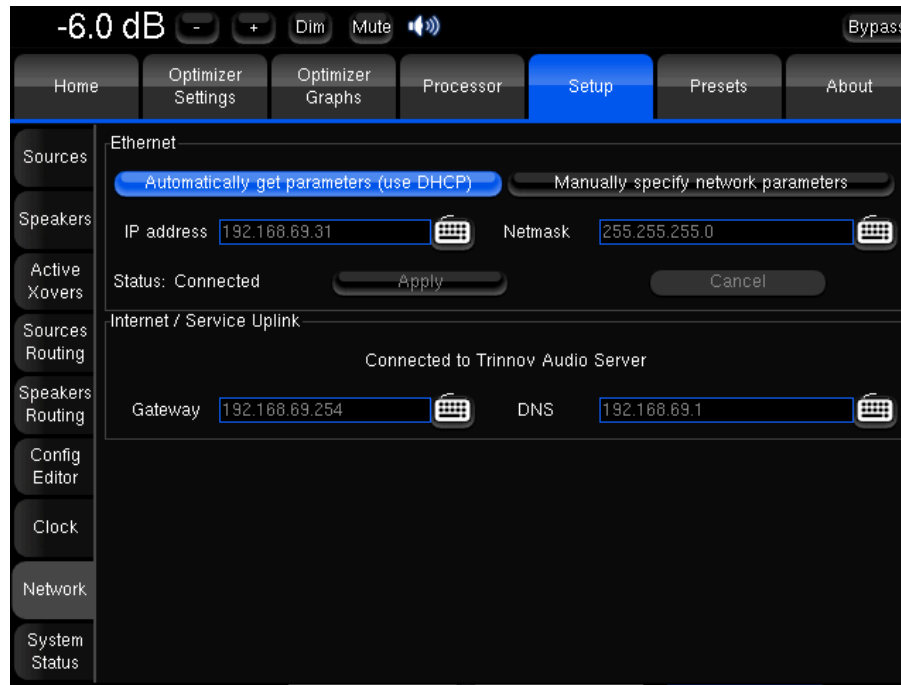
Setup/Network

Using the magnitude32 as a WiFi client to join an existing network

Once connected to the graphical user interface, the **Setup/Network** page will also display WiFi options, provided that the optional USB WiFi dongle was connected before starting up the unit.

Setup/Network page with WiFi Options

Press the WiFi **Setup** button to display the WiFi Status Modes page.
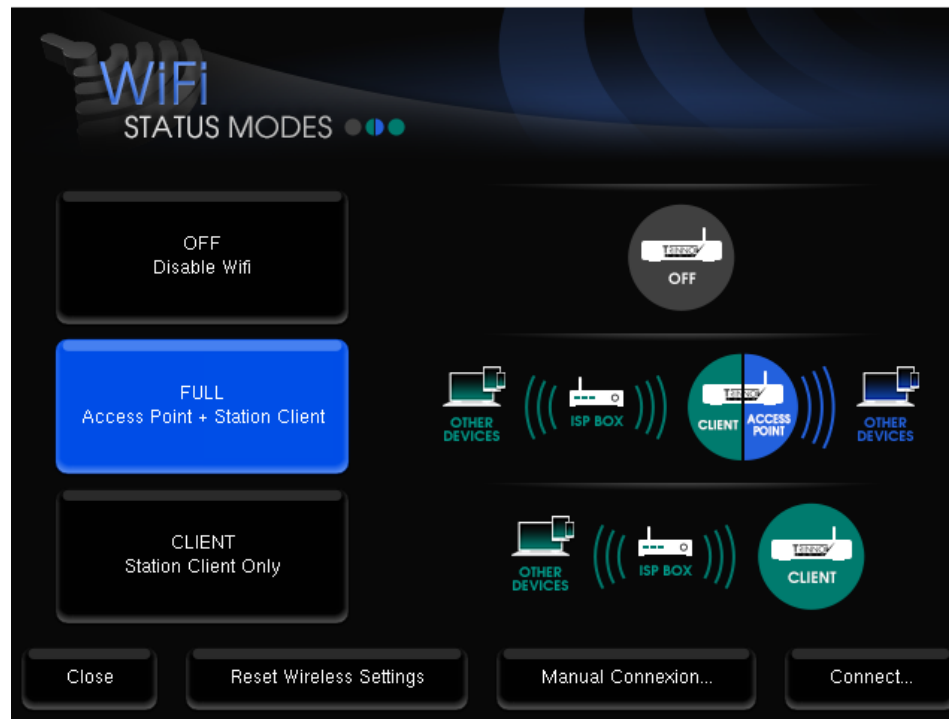
WiFi Status modes

There are 3 WiFi modes that can be selected:
- **OFF:** the WiFi module of the magnitude32 is disabled.
- **FULL (default):** the magnitude32 is both used as an Access Point and a Client
- **CLIENT:** the Access Point is disabled but the magnitude32 still operates as a WiFi client as long as the configuration is valid.

WiFi client configuratino procedure:

1. In the Settings/Network page, hit the Setup... button to display the WiFi configuration page.
2. Select the required WiFi mode (FULL is the default mode but only works as an Access Point until the WiFi client is configured).
3. Whether you selected the FULL or CLIENT mode, the Connect button display the available WiFi Access Points.
4. The encryption columns displays a locker icon when the network is protected by a passphrase and/or encrypted.
5. The signal columns displays a WiFi signal icon, indicating the strength of the WiFi signal.
6. If you select an encrypted Access Point, you will be asked a passphrase.
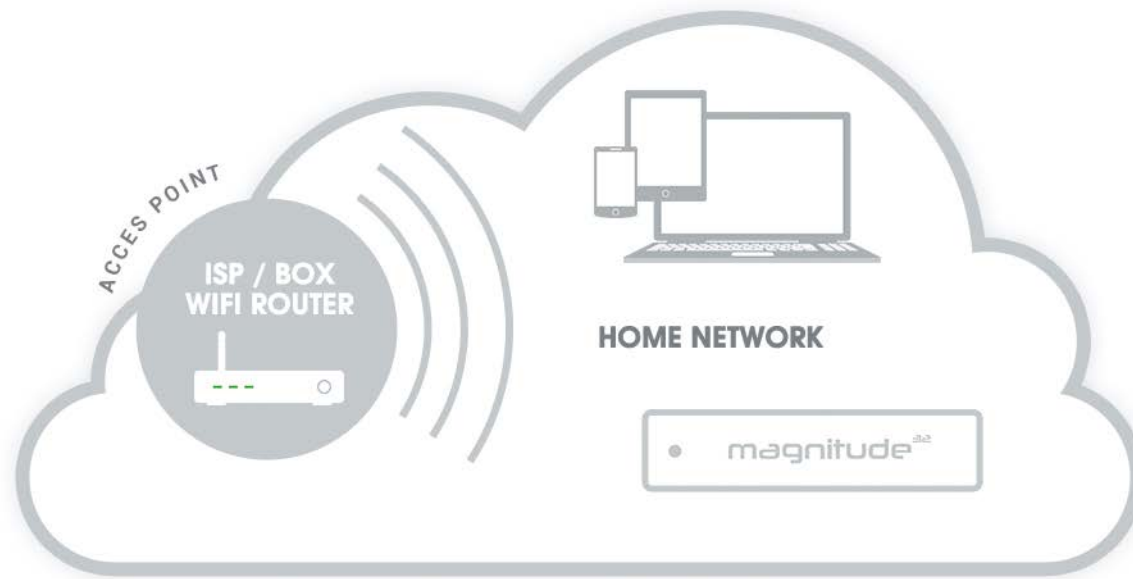
Using an external display, mouse and keyboard

Magnitude32 have VGA/DVI and PS2/USB ports at the back for standard PC monitor and mouse/keyboard use and may be used with KVM (Keyboard Video Mouse) extenders (available as an option). In this way, one has comprehensive, remote access to all the functionalities of the Processor.

2.3 CONNECTIONS

Magnitude32

The magnitude32 front panel is made of a brushed aluminum plate and integrates the power switch. Rack ears are provided to install the magniture32 in a 19-inch rack.

The magnitude32 product ranges consist of 5 models:

mg^{32} **88d** |mg^{32} **1212d** |mg^{32} **1616d** |mg^{32} **1624d** |mg^{32} **1632d**

Hardware upgrades are possible to increase the number of I/Os and processing channels.
All configuration share the same rear panel.

The magnitude32 enables extensive channels routing. Inputs and outputs are not labelled since they can be freely assigned.
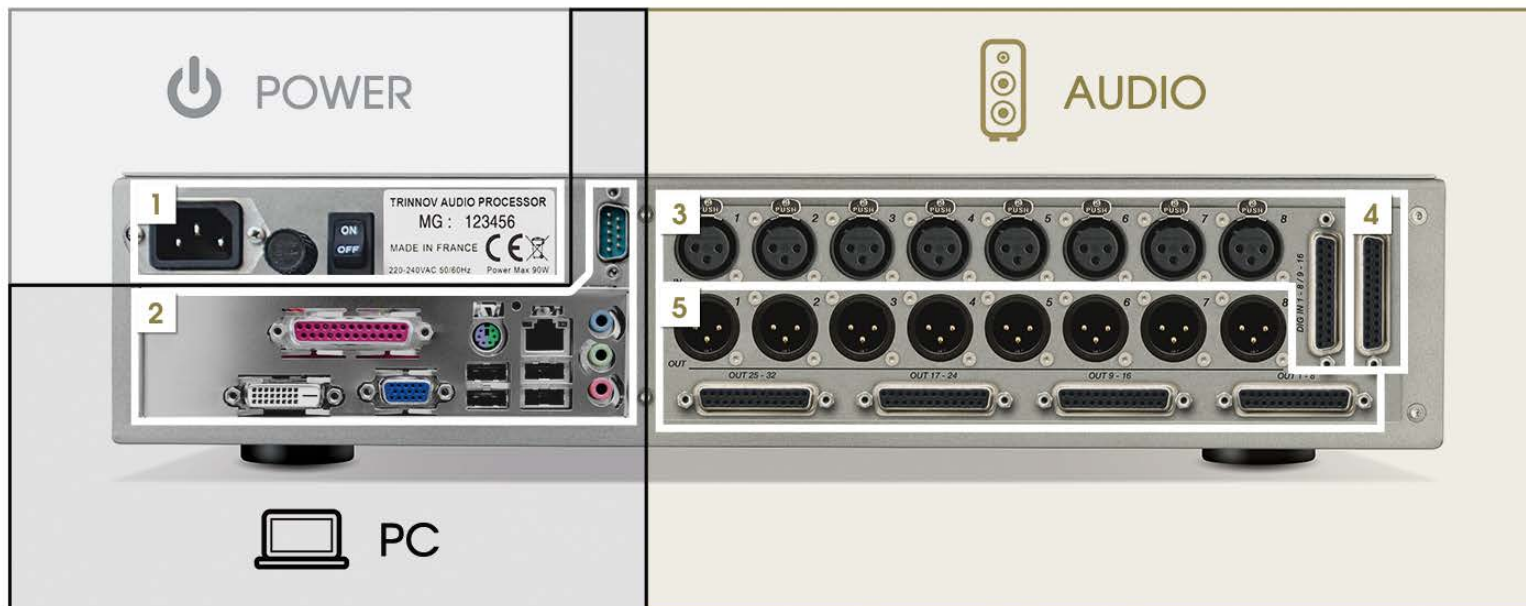
- **PC section:**
  - o 1 PS/2 port for mouse and keyboard
  - o 1 VGA connector
  - o 1 DVI connector
  - o 4 USB ports

  - o 1 Ethernet socket
  - o 1 parallel port
  - o 1 RS232 port
  - o on-board audio (inactive)

- **Audio section:**
  - o 1 DB25 for Digital Inputs 1-8 / 9-16
  - o 1 DB25 for Analog inputs 9-16

  - o 1 DB25 for Analog outputs 1-32
  - o 8 XLR for the inputs 1 to 8.
  - o 8 XLR for the outputs 1 to 8.

    AES-EBU inputs 9-16 are available as an option.

DB25 connectivity

The 25 pin D-sub connectors provide both analog and AES inputs and outputs using the widely spread **Tascam** scheme, which is also used by Avid.

Analog & AES/EBU DB-25 Connectors pinout

G: Ground   C: Cold / -   H: Hot / +
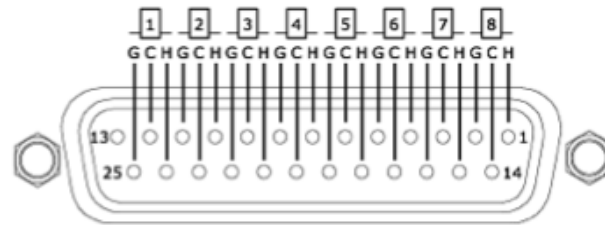
NOTES
Another quite popular pinout is the one from Yamaha. You should check which pinout your equipment is using before connecting it to the magnitude32.

Unbalanced connections must connect signal to SubD-Hot and shield to SubD-Cold.

Connection example

The example bellow shows a typical connection example.
The magnitude32 is inserted between an AV preamp and power amplifiers.
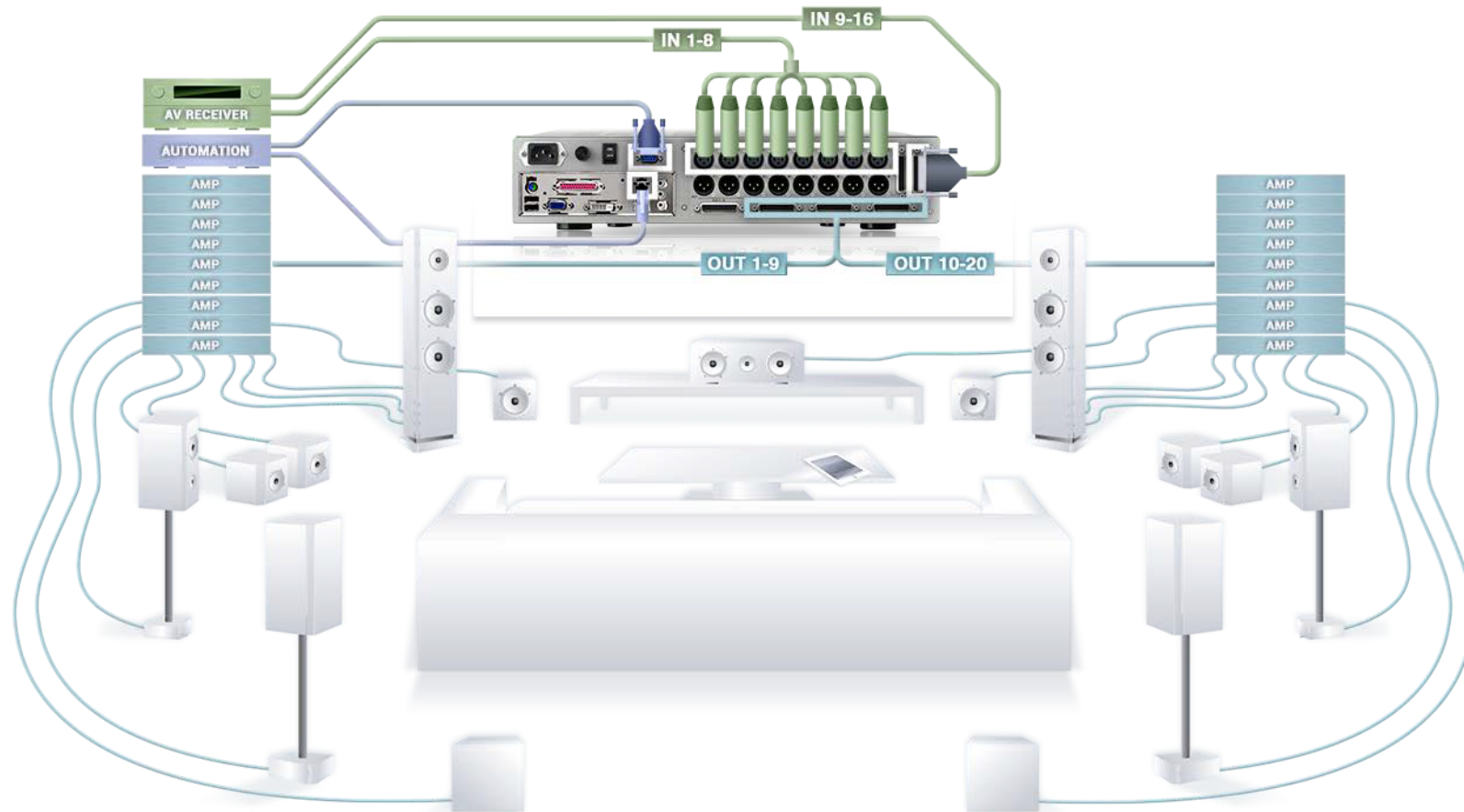
3 BASIC OPERATIONS

3.1 INPUT SELECTION

The Home/Select page provides input selection buttons to choose between analog and digital inputs.
The analog inputs should be selected before calibration as the microphone should be connected to the analog XLR input 1-4.
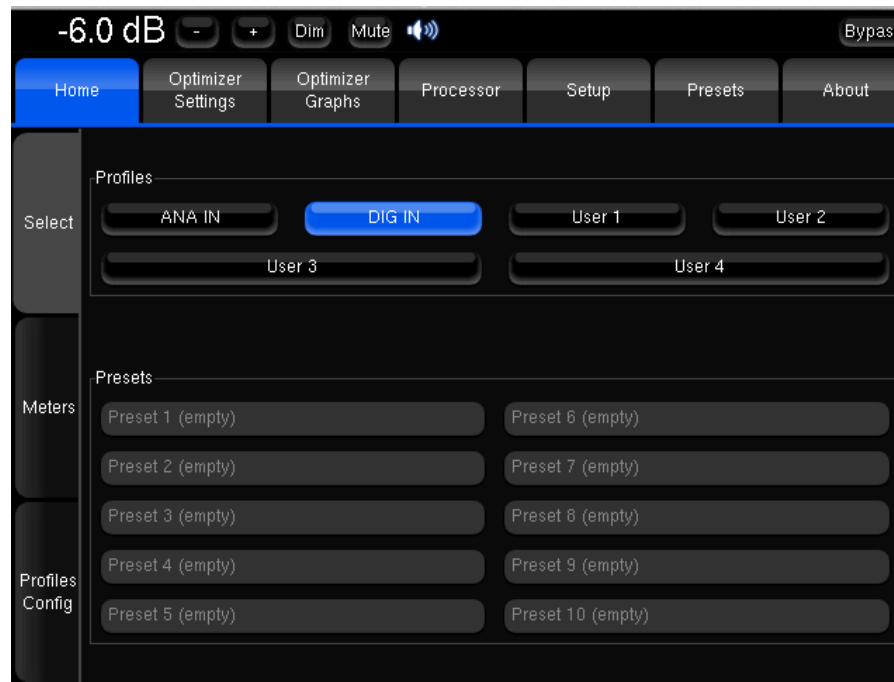
NOTE
The synchronisation mode of the magnitude32 depends on the selected inputs.

• *The magnitude32 works as a slave and synchronizes itself on the AES audio signal clock when the digital inputs are selected*
• *The magnitude32 works as master clock and uses its internal clock for synchronization when the analog inputs are selected*

3.2 VOLUME CONTROL

The graphical user interface allows to control the volume of the magnitude32.

The Master level defines the boost/attenuation applied to the outputs and can be adjusted by 1dB steps.

The **Dim** button performs a 20 dB attenuation.
The **Mute** button cuts the outputs.

0dB is unity gain and may overload the output stage if the compensation filters apply too much boost over the bandwidth.

As a safety measure, the master level should at least be attenuated by a value equal to the maximum boost settings of the Optimizer.

Volume control can also be performed via IP or RS232 commands or an automated system.

3.3 PRESET MANAGEMENT

Built-in Preset

At first start-up, the magnitude32 loads the factory preset, called built-in preset. The built-in preset cannot be modified.

This preset is set for a 5.1 source and speaker layout.
- Inputs 1-6 (analog or digital) are respectively considered as channels L, R, LS, RS, C, LFE.
- Outputs 1-6 should likewise feed speakers L, R, LS, RS, C, LFE.

Presets 1-29

Magnitude32 can store up to 29 memory presets. Presets can be backed-up and restored to/from a USB key.

Presets 1-9

Except the Master Level and the input selection, presets include and store almost every settings of the magnitude32.
* Each memory slot can be used to store configuration (Save button), deleted (Clear button) or locked (small lock icon).
* Push any of the saved memory slot button to recall a preset. The preset currently in use is highlighted in blue. An empty memory slot is greyed.
* One preset can be chosen as the default preset to be automatically loaded at start-up if its Def (default) button is checked.

A virtual keyboard (icon next to the "preset name" text window) can be used to label your presets. Type a label in the "Preset name" window, then press the save button. A standard PS2 or USB keyboard can also be used if connected to the magnitude32.

IMPORTANT NOTE
Any parameter change will be lost if not saved in a preset. Please manipulate presets with care since saving, overwriting and deleting preset do not require confirmation.

Preset Info

The Preset Info tab provides useful information about the preset:
- **Last calibration**: the date and time of the last calibration.
- **Last computation**: corresponds to the last time the user pressed the Apply Changes button.
- **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 used for.
- **Calibration microphone**: the ID of the microphone used for the calibration
- **Calibrated at sample rate**: the sample rate used for the calibration.
- **Comments** about the preset provided by the user. This can be used to store a version history of the preset.
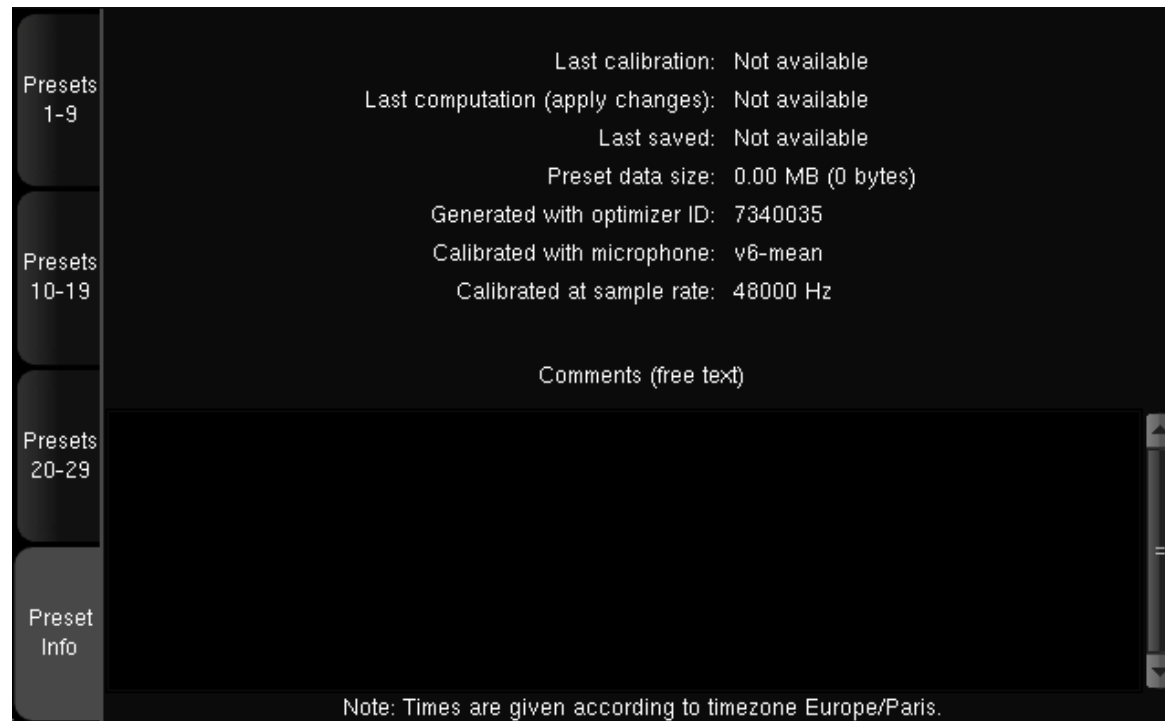
Backup/Restore Presets

Screenshots

Screenshots of the graphical interface can be stored during operation by pressing the "print screen" key on a keyboard connected to the unit.

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 Processor. A menu will appear. It allows one to restore settings from the USB key into the Processor, or backup them from the Processor into the USB key:

- **the Presets** include all the information related to a preset.
- **the XML Parameters** correspond to the XML files used in the Config Editor.
- **the microphone files** include all the compensation files of the microphones installed on the system.
- **the PDF Documents** are the measurement reports generated by the Optimizer.
- **the bugreports**
- **the screenshots**

USB Key detected
Please select the operation you want to execute
Please do not unplug the usbkey before choosing "Close"

Save presets to USB key   Load presets from USB key

Save profiles to USB key   Load profiles from USB key

Save XML parameters to USB key   Load XML parameters from USB key

Save microphone files to USB key Load microphone files from USB key

Save PDF documents to USB key   Save bugreport to USB key

Save screenshots to USB key   Close

The **Save** function will copy elements currently stored in the system to the directory of your choice in the USB Key.
The **Load** function will copy to the Processor the elements from the USB Key directory that you specify.

Presets can be saved and restored individually.

Caution: the elements previously stored on the system will be replaced.
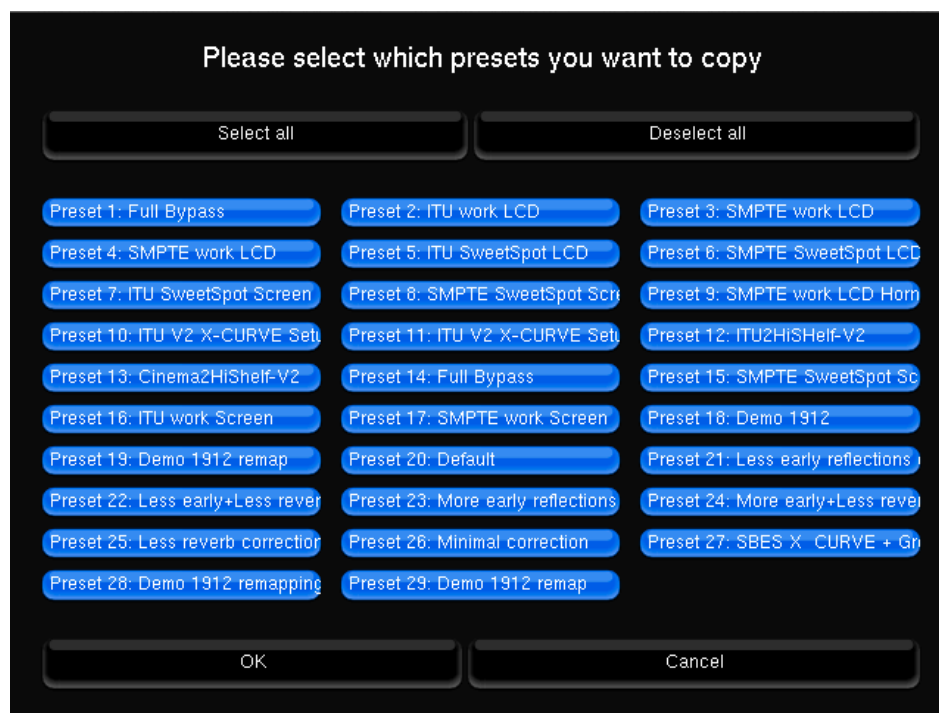
IMPORTANT NOTE
The USB port should only be used for this function with no critical audio, as it may result in momentary clicks or pops.

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 magnitude32 (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

3.4 INPUT CONFIGURATION

The magnitude32 allows to link several parameters to an input. The purpose of such input go beyond input selection. Therefore, inputs are alternatively named profiles in this menu. Profiles settings can be configured in the **Home/Profiles Config** page.

-6.0 dB   -   +   Dim   Mute

Bypass

| Home | Optimizer Settings | Optimizer Graphs | Processor | Setup | Presets | About |

Name — DIG IN

ANA IN		

	Input Connectors	Digital
DIG IN	Clock	Default
	Optimizer Preset	No Change
	Source	As in preset
User 1	Remapping Mode	As in preset
	Bass Management	As in preset
	FIR EQ	No Change
User 2	User EQ	No Change
	Submixes	As in preset
	Input Levels	No Change
User 3		
	Master Level	Unchanged Unchanged
User 4	Output Delays	No Change

Profiles Config

○ Use last loaded profile as power-on default profile   Save

If profiles have been uploaded on the machine via the network, you have to
reload them in the processor with the button right here.   Reload

The analog and digital inputs are represented as vertical tabs on the left of the page.

For each input, several parameters can be configured using a drop-down menu. The "save" button shall be used to store and apply new parameters.

- **Name:** clicking on the "Name…" button will display a virtual keyboard allowing you to change the name of the input.
- **Input Connectors:** is used to select the physical input affected to the profile. This option may be used to modify the input's purpose. The physical inputs displayed in the source routing matrix depend on the input connector affected to the current profile.
- **Clock:** you can choose a clock mode and sample rate from the list to control the synchronization mode of the magntude32 via profile selection.
- **Optimizer Preset:** you can link an Optimizer preset to a profile, thereby associating a loudspeaker/room compensation to an input.
- **User EQ:** you can recall of the User EQ available on the magnitude32 on input selection.
- **Master Level:** you can recall a master level with a profile to align the output level between analog and digital inputs. To set a master level, click on the unchanged button to display and modify the master level with the ++/-- (+/- 1 dB) and +/- (+/- 0,1 dB) buttons.

IMPORTANT NOTES
- the following settings interact with the Optimizer preset associated to the profile.
- the list of available options for the "Source", "Remapping" and "Bass Management" settings is updated according to the associated profile.
- **Source:** you can configure different input format for each preset and affect a specific format 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 or use the remapping mode of the current preset.
- **Bass Management:** profiles allow you to recall a specific bass management option on input selection.
- **FIR EQ:** you can recall a specific FIR EQ file on input selection.
- **Input Levels:** you can select an input control file or use the input levels of a specific preset when associated to the profile.
- **Output Delays:** you can use the current preset or disable the output delays on input selection by using the "Inhibit" option.

The "Power-on Default" option shall be ticked to use the input as default input on start-up.
The other option is "Use last loaded profile as power-on default", located on the left of the "Save button" at the bottom of the screen.

IMPORTANT NOTE
• You need to press the "Save" button to apply changes.

• The "Reload" button cancel changes made before saving.

NOTE

It is possible to define a default input and a default preset. However, the default input is recalled after the default preset at start-up. Therefore, the default preset won't be recalled if another preset is linked to the default input.

4 SETTING UP THE SYSTEM

4.1 SOURCES SETTINGS / ROUTING

Sources Settings

The main purpose of the Sources Settings menu is to define the input format

NOTE

This page also allows to configure the source with the following parameters:

- *Input format*
- *Number of LFE*
- *Channel order*

The list of available input format depends on the model of your magnitude32 and include every industry professional standards up to 9.1 SMPTE as well as 12/16 channels to comply with third parties proprietary formats and DCP.

Mono	LCRS	SDDS
Stereo	QUAD	9.1 SMPTE
2.1	5.1 ITU / SMPTE	12 channels
LCR	6.1 ITU / SMPTE	16 channels
LCR+LFE	7.1 ITU / SMPTE	

The input format also defines the speaker layout used as target for 2D/3D loudspeaker remapping.

Channel order:

The channel order can be modified using the **Next Order** button and defines:
- The sorting order of the connections between your source and the magnitude32.
- The sorting order of input channels in the routing matrix.
- The calibration order.

IMPORTANT NOTE
The channel order should not be changed after calibration. The filters would then be applied to the wrong physical outputs.

Number of LFE:

The number of LFE defines the number of LFE feeds for the magnitude32.

IMPORTANT NOTE
In case of multiple subwoofers, declaring several LFE and feeding them with the same signal is required to automatically apply the required attenuation to subwoofers output channels and keep the same sound pressure level out of the subwoofers.

LFE duplication
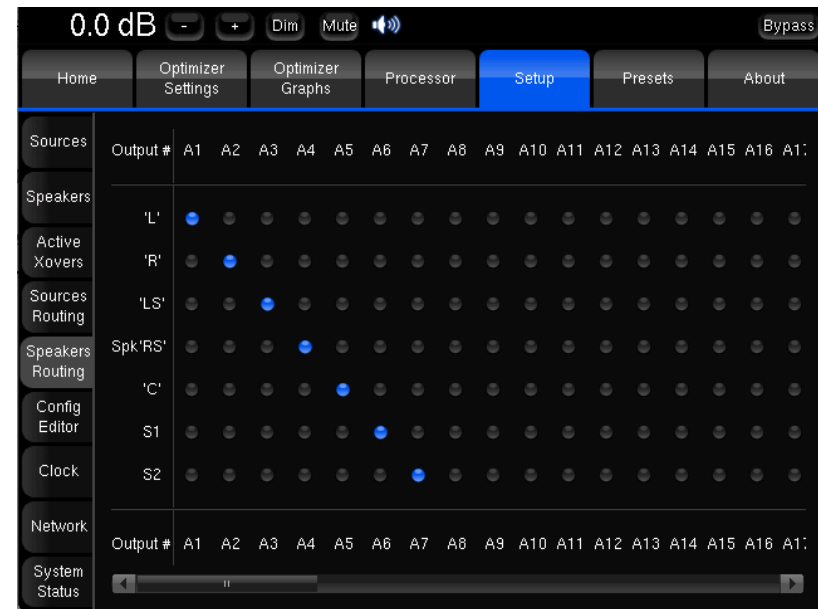
Routing of the attenuated signals to multiple subwoofers

Likewise, declaring a 7.1 input format and routing duplicated signals to different speakers will apply the required attenuation to keep the sound pressure level constant.
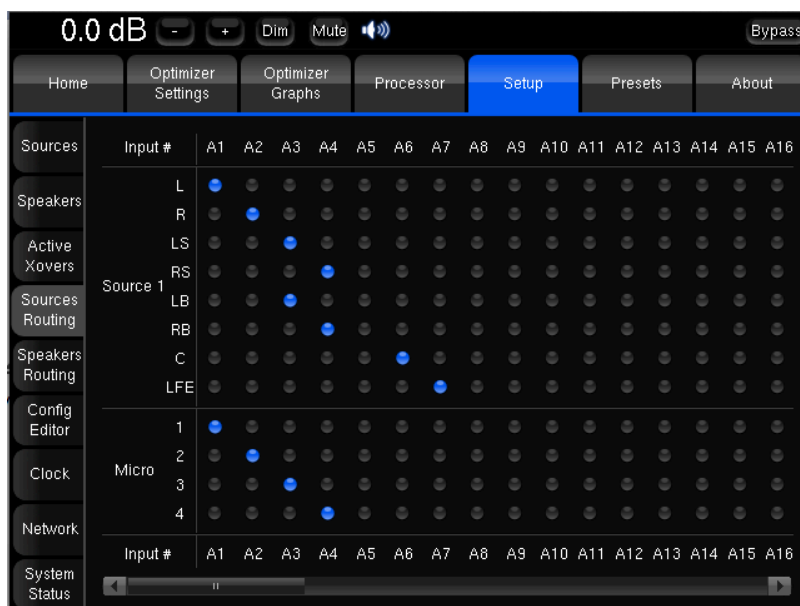
Ls/Rs dupplication via 7.1 source
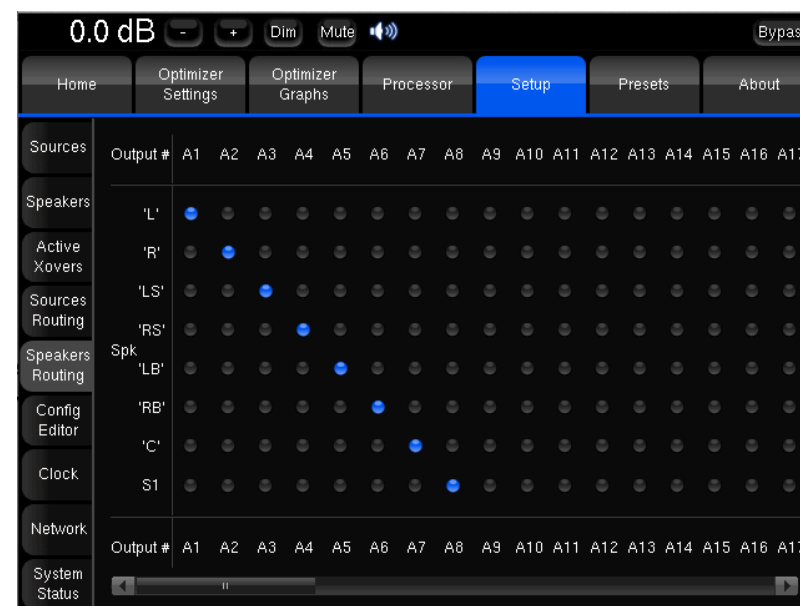
Routing of the attenuated 5.1 Ls/Rs signals to the 7.1 layout satellites

Sources Routing

This page allows you to affect physical inputs (columns) to the different channels defined by the input format/channel order (rows).

Physical inputs are prefixed with an A (Analog) and a D (Digital) depending on the selected input.

NOTE
It is possible to modify the routing of the 4 microphone channels if required

4.2 SPEAKER SETTINGS / ROUTING

Speaker Settings

The Speaker settings page is used to define the speaker layout of the multichannel audio system.

The available parameters are:
- Number of speakers: sets the number of loudspeakers of your sound system.
- **Please note**: changing this setting requires re-calibration.
- Number of subwoofers: sets the number of loudspeakers which will play the LFE channel 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.

Please refer to the dedicated chapter for bass management information.

Speakers Routing
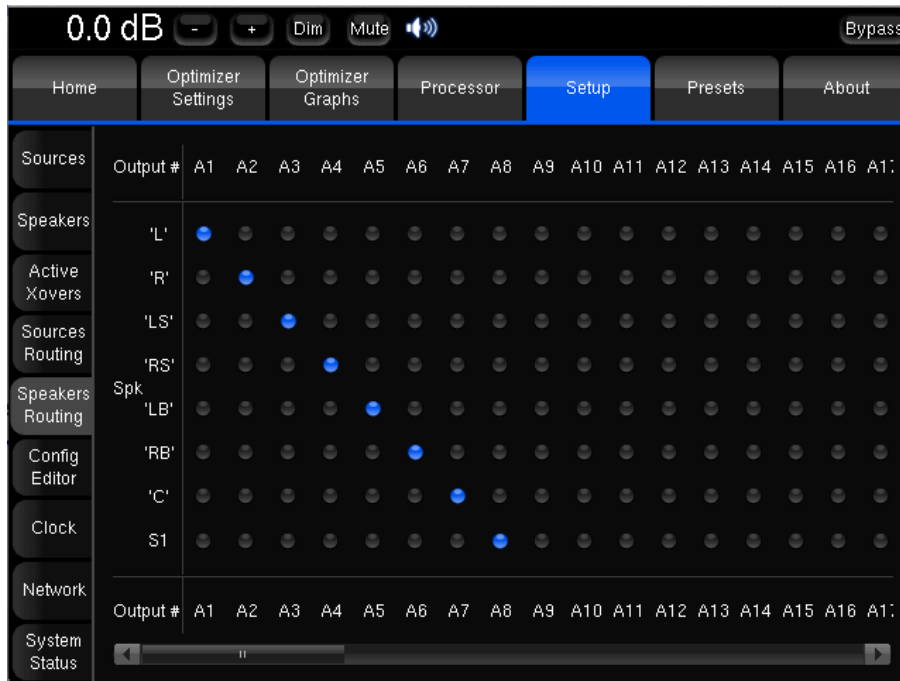
This page allows you to affect output channels (rows) to the physical outputs of the magnitude32 (columns).

The number of output channels displayed in the grid corresponds to the number of loudspeakers, number of ways and subwoofers declared in the Setup/Speaker Settings and Active Xovers 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.

If **Remapping (Spatial Optimization)** is used, each input channel may be mapped across two or more output channels AND the number of loudspeakers does not have to match the number of input channels. Spatial Optimization is a unique feature of the Optimizer.

4.3 CLOCK SETTINGS

The clock settings page is used to select the synchronization mode of the magnitude32.

The synchronization mode depends on the audio input type:

- **Analog Input:** the magnitude32 uses its internal clock for audio synchronisation. The sampling rate 48kHz is recommended.
- **Digital Input:** the magnitude32 uses the incoming AES/EBU signal as external clock. The sampling rate used by the magnitude32 therefore depends on the sampling rate of the incoming audio.

On selection of the input, the magnitude32 will automatically switch to the required synchronization mode.

Status information:

- **Current sample rate:** shows the current clock frequency of the magnitude32
- **Detected sample rate:** shows the detected clock frequency.

4.4 AUDIO UTILITIES

The processor tab of the graphical user interface includes an extensive set of audio utilities to perform various tests and manual operations with the magnitude32.

Meters

Input and Output levels can be monitored through the Meters tab. Both the peak level and the RMS level are displayed.

You can visualize the peak level of each channel. The highest level registered on each channel can be displayed by pressing the Mem button. When the level gets close enough to the saturation level, the name of the channel appears highlighted in red. The Clear button resets the memory and the saturation indicators for every channel of the group.

IMPORTANT NOTE
a digital signal cannot technically exceed 0 dBFS. Therefore the red tag on top of each digital input channel only let you know when the maximum level of 0dBFS is reached and does not necessarily indicate distortion or clipping.
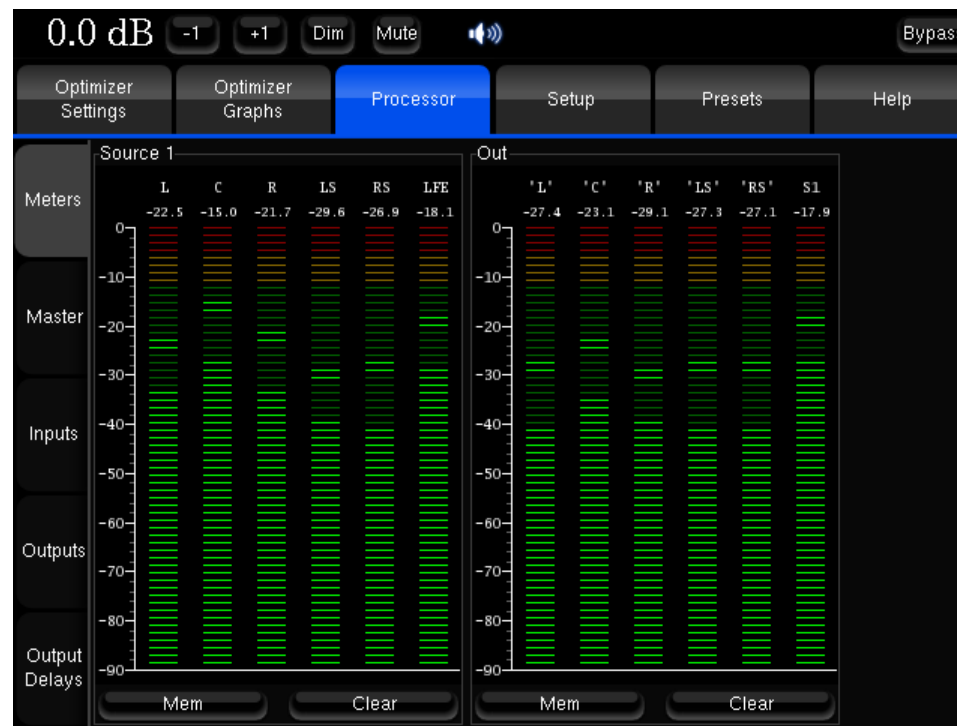
Level meters for the input and output signals

Test Signals

The Inputs page includes 1000Hz sinusoidal signals and pink noises for every channel.

Be careful about the output level before playing a test signal, the sinusoidal signal and pink noise are both generated at -20dBFS RMS.

Input channels

For instance, you can use Pink Noise for setting up your equalization, as follows:
- In the Processor/Inputs page, press on the Pink Noise button for the channel you want to equalize,
- Switch to the Processor/Outputs page and open the Preset EQ or User EQ for that *same* channel,
- Make your EQ adjustments,
- Use the Previous/Next buttons: the *pink noise will follow* to the same channel.

The pink noise will only follow if you start the EQ from the same channel where the pink noise has been activated. A typical use of this graphic EQ is ISO X curve compliance verification by a consultant or voluntary small change of the tonal balance.

Solo/Mutes

Output channels

Both inputs and outputs channel can be soloed and muted individually

5 CALIBRATION

5.1 FUNDAMENTAL CONCEPTS

Procedure

The magnitude32 is inserted before loudspeakers to correct imperfections of the loudspeakers and the room.

During calibration, speakers are calibrated with a 3D acoustic probe.

Signal flow of the magnitude32
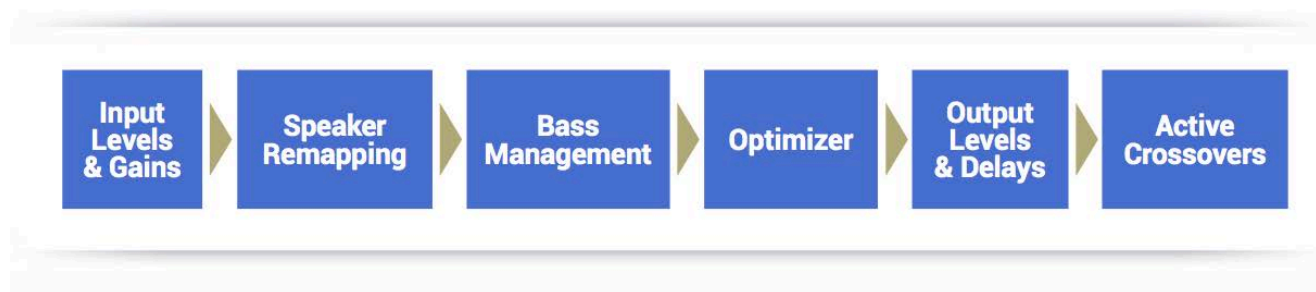
Not mentioned:
Inputs & outputs routing / Physical inputs & outputs
Manual input & output Eqs

Procedural order & relation between filters
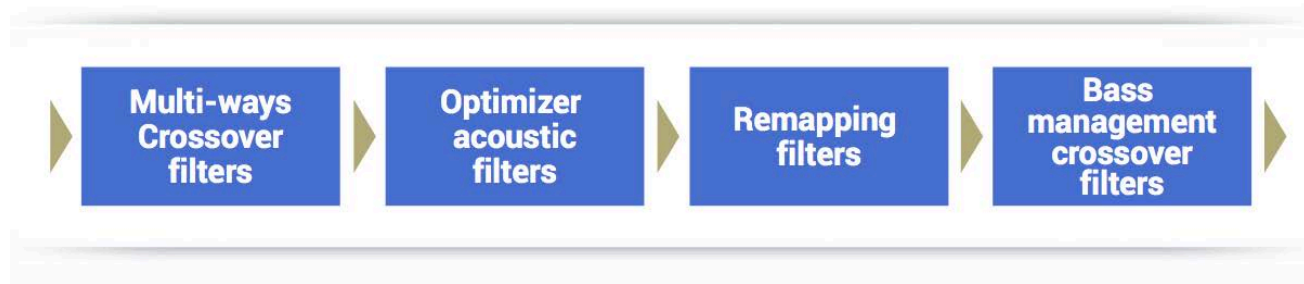
1) Active-Crossovers are set up first following a semi-automatic calibration procedure

2) The Optimizer's filters calculation is made from the speaker's measurement

3) Remapping filters are applied based on the source's input format and loudspeaker's position measured during the Optimizer Calibration, as soon as one of the remapping mode is enabled manually

4) Bass management crossover filters are applied after and regardless of the loudspeakers calibration.
   The crossover frequency and bass management mode can therefore be modified after calibration

5.2 3D MEASUREMENT MICROPHONE

The purpose of the calibration microphone is to measure the characteristics of the loudspeaker and their layout in the room.

For each loudspeaker, the Optimizer gathers the following information:

- full 3D position
- amplitude response
- phase response

Microphone specifications

- 4 capsules tetrahedron figure.
- Individual compensation filters
- Frequency response within +/- 0.1dB across the 20Hz-24kHz frequency range
- Spatial resolution below +/-2°
- Built-in preamp
- A 9V PP3 LR61 battery powers capsules and electronic
- Camera / Microphone thread.

Bottom of the microphone
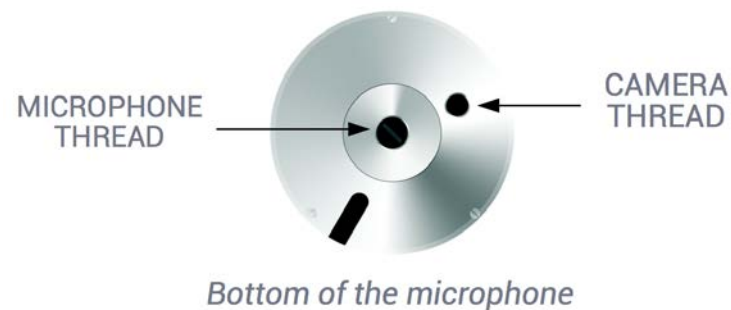
Power supply

The power supply located in the base of the microphone uses a standard 9V PP3 LR61 battery. The red led indicates the battery Level. Replacing the battery requires to take off the 3 screws at the bottom of the microphone.

Individual compensation file

The individual compensation file of the microphone used by the magnitude32 can be modified in the **Optimizer Settings/Calibration** page.
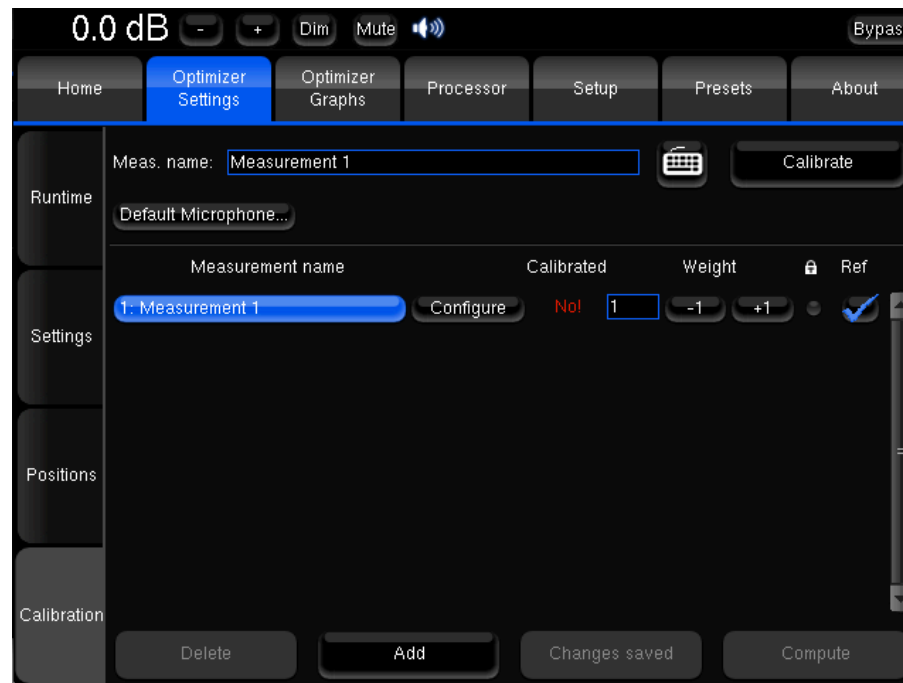
Calibration

Always make sure the microphone's serial number indicated in the **About** page matches the label at the bottom of your microphone.

If not, use the **Configure** button of the required Measurement to select the correct compensation file.

For future measurements, use the **Default Microphone** button to make the required compensation file automatically selected in the Factory preset.

The 3D localisation of the speaker can only be performed if the connection order of the capsules is respected.

Connection Order
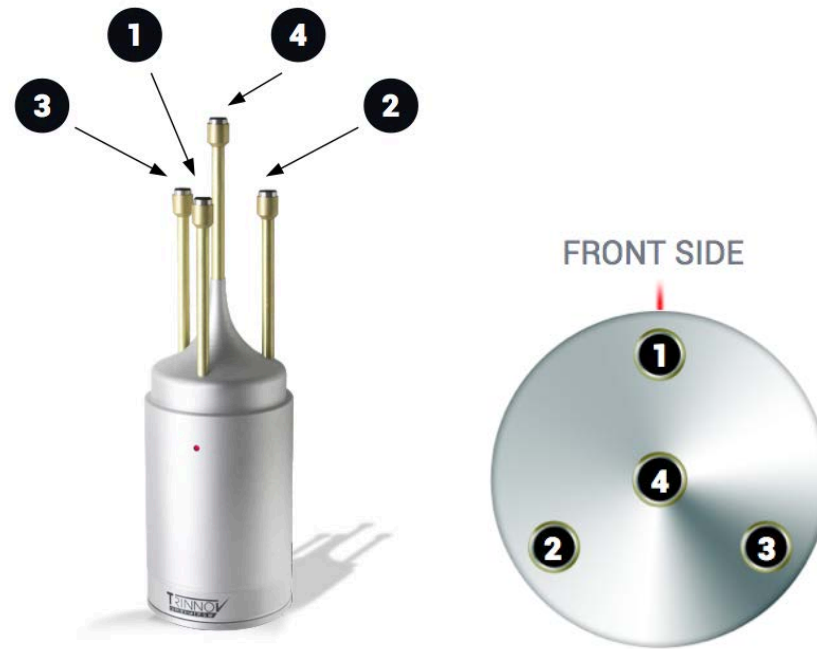
5.3 MICROPHONE REFERENCE POSITION

Measurement environment requirements

At least a few basic rules should be respected to ensure a robust measurement:

- No obstacle between the speaker and the microphone.
- No highly reflecting surface (leather sofa, glass table...) close to the microphone.
- No background noise during the measurements (Air conditionning, open doors, windows etc...)
- No movement during the calibration (disturbs speaker localisation)

Reference measurement point

The magnitude32 allows for multi-point calibration but whether the calibration is performed with a single or multiple position, the reference measurement position should be chosen carefully since it will be used by the Optimizer for:

- Cross-over drivers alignment
- Loudspeaker 3D localisation
- Loudspeaker 2D/3D remapping
- Loudspeaker relative delay/level alignment
- Master delay/level calculation

Position and orientation of the microphone for the reference point

The Calibration microphone should be placed at the listening position, using ears as height reference. A red led shows the front of the microphone. This end must be pointed to the front of the sound stage, where the center channel should be (or is, if the loudspeaker setting is respecting the ITU recommendation).

Below is the reference position:
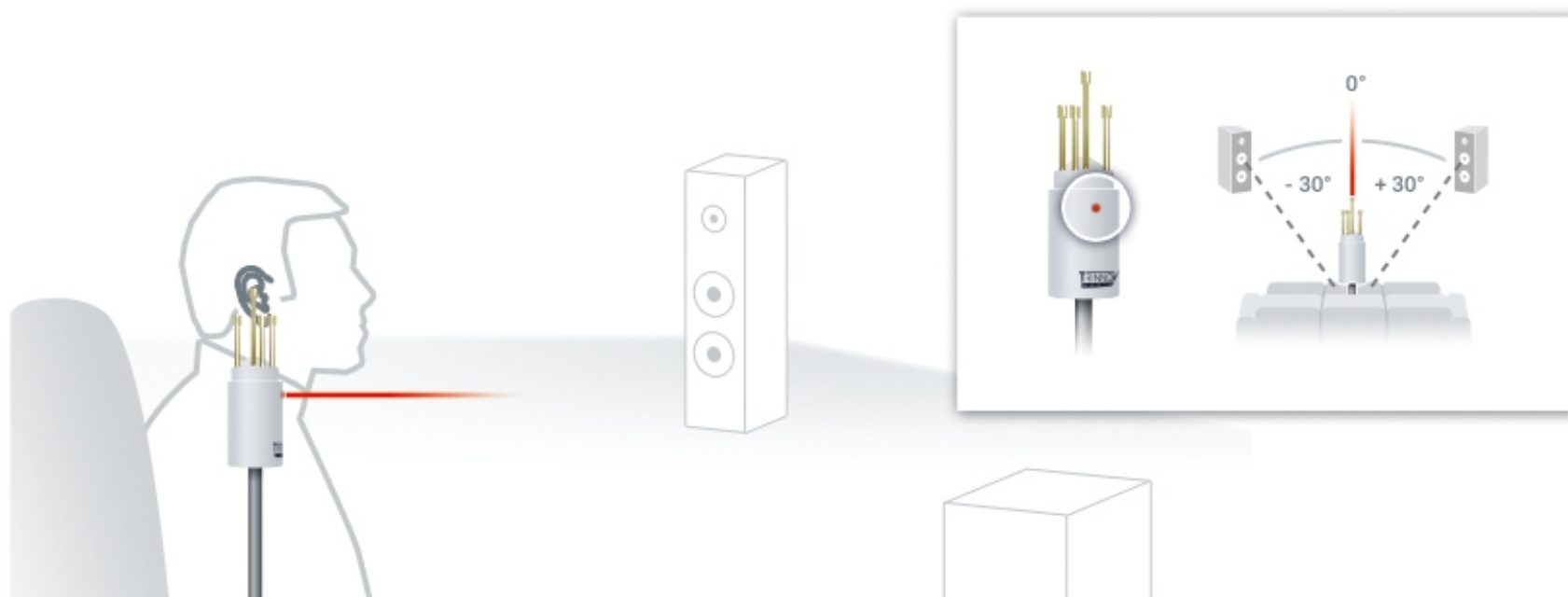
During the calibration process the microphone should be screwed on a microphone or camera stand.

5.4 ACTIVE XOVERS CALIBRATION
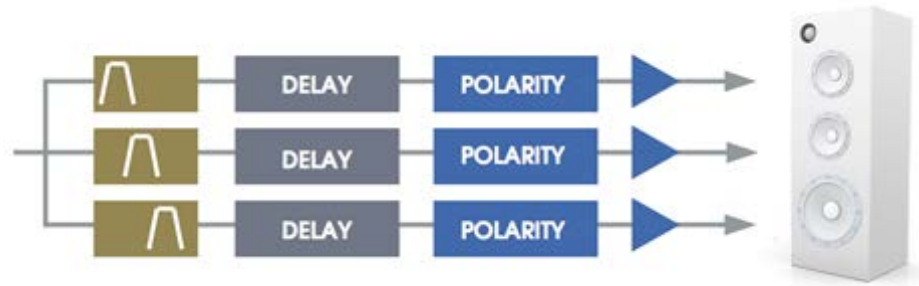
The Active Crossover Calibration is accessible via the **Setup/Active Xovers** page.

Functionality

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

- 2 way, 3 way and 4 way crossovers
- up to 32 output channels, e.g. managing up to 8 four-way loudspeakers with a 32 channels magnitude32
- Types of filters: Bessel order 2, 3 and 4; Linkwitz-Riley order 2 and 4; Butterworth order 2, 3 and 4.
- Level, Polarity and Delay adjustment on each way (driver)
- Graphic display of the crossover response

Procedure

The active crossovers calibration is a semi-automatic procedure:

1. The number and type of filters for each loudspeaker are set manually as well as crossover frequencies.
2. Levels, Delays and polarity of each driver are set manually or determined automatically with a calibration.

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.
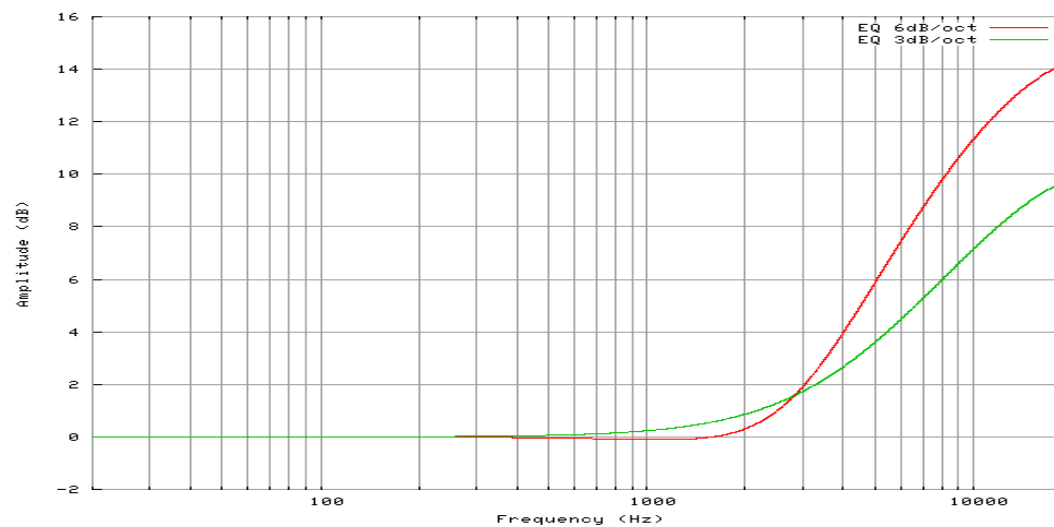
- The **Apply** button is highlighted as soon as a parameter is modified and is used to compute and load the new settings into the processor. Once compute is finished (gears icon disappears from the notification bar), filters are applied to the outputs.
- If change is unwanted, press the **Cancel Changes** button to cancel your modifications
- If change is accepted, press the **Save** button of the preset of your choice in the Presets page to *save the changes*. Otherwise changes will be lost.

Upon configuration of multi-way speakers, the output routing matrix shows new output channels.
Make sure every output channels is routed to the required physical output.

Routing Xovers

The theoreticalmodel of the specified filters is represented as follows:
- The blue curve represents the theoretical low-pass filter response
- The red curve represents the theoretical high-pass filter response
- The green curve represents the theoretical resulting response of the speakers
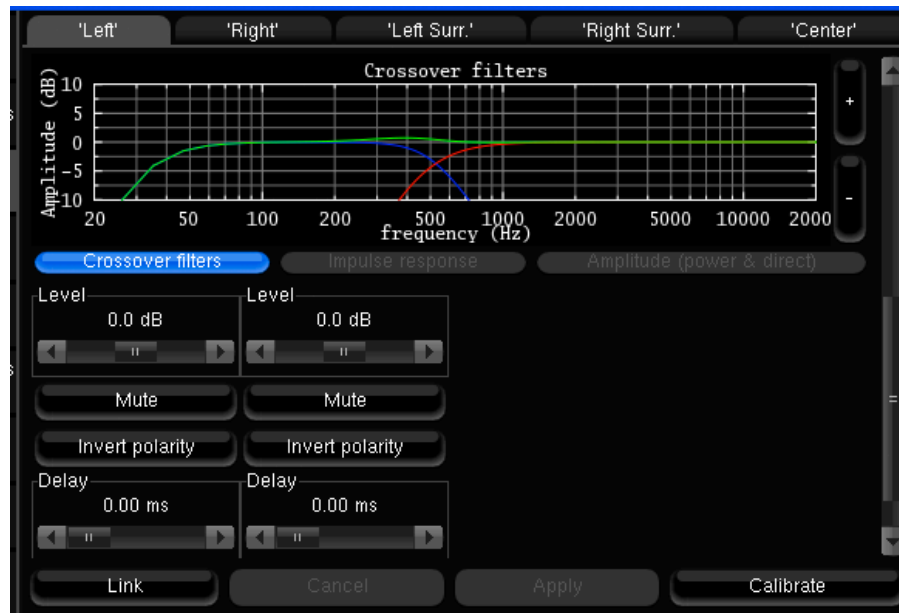
- **Level** and **delay** can be adjusted on each way (driver),
- **Mute** and **Invert Polarity** buttons are also available.

Automatic calibration

The Optimizer automatically set up the level, delay and polarity of each driver automatically by a simple procedure that calibrates the drivers separately.

Before calibration, please remind that:
- Each speaker is calibrated **individually**.
- It is required to **unlink** speakers before calibration
- A MLS signal is sent to each driver of the selected speaker when the **Calibrate** button is hit.
- A warning message states that a global calibration is required.
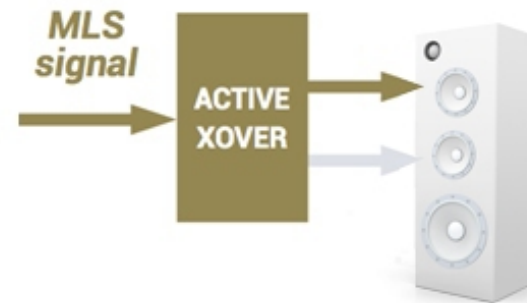
Calibration instructions:
- Respect the reference microphone placement instructions.
- Make sure the measurement environment is ok.
- Refer to the calibration chapter regarding output level during calibration.

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

1 measured **impulse response** of each way: you can see whether the drivers are correctly synchronized.
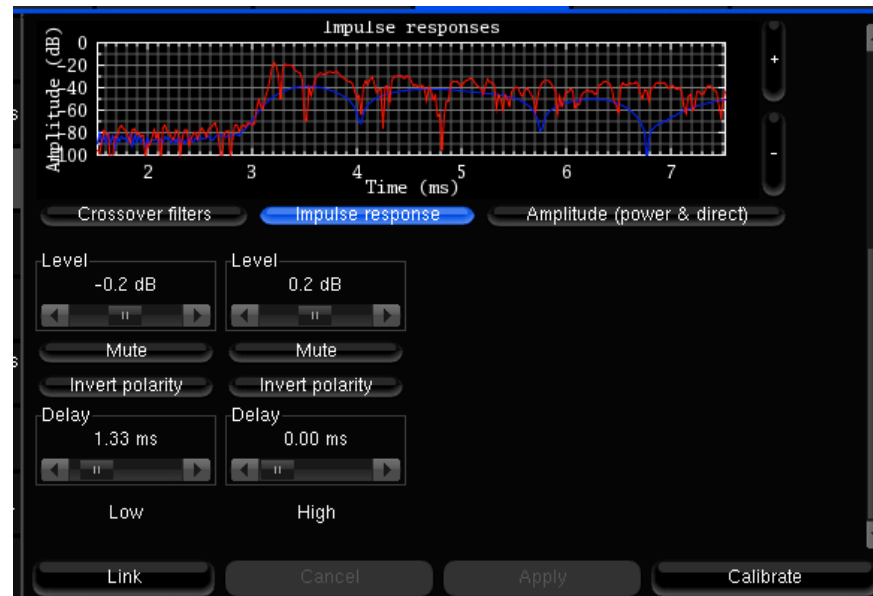
2 the recombined **amplitude** response of the speaker: you can see whether the combination of the drivers is constructive, and you can observe the effects of level/delay/polarity modification on the combined amplitude of the speaker. Two curves are displayed, one showing the global power of the speaker (including the room), and one showing the amplitude of the direct front and early reflections. 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.
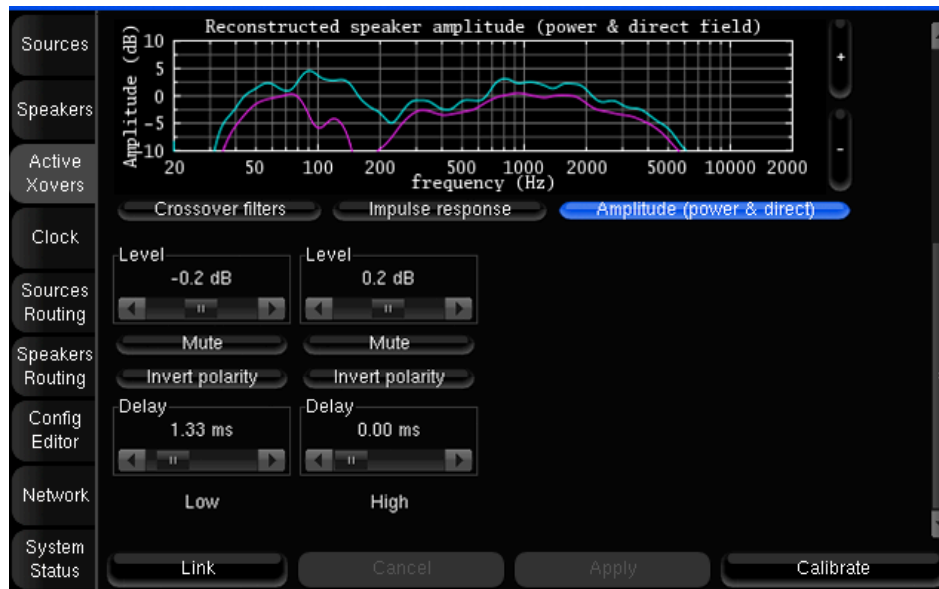
Please note:
- As they are automatically tuned, previously set up levels, delays and polarities are ignored during the calibration. In other words, tuning these parameters before launching the calibration will have no effect on the result.
- Under some circumstances, the automatic crossover algorithm 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 polarity").

5.5 CALIBRATION

The loudspeaker calibration is made from the Optimizer Settings/Calibration page.
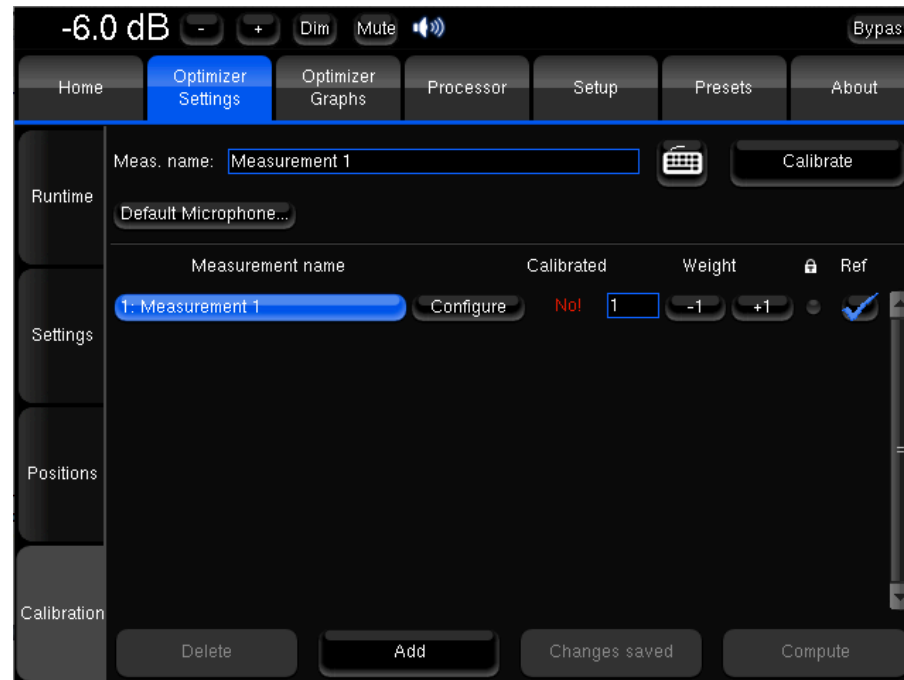
Useful tips

Avoiding feedback loops:
- Always MUTE the processor as the first step of the calibration procedure. Muting the outputs does not affect the calibration signal.
- Be careful with feedback issues when you change the routing you could make loops between the microphone and the loudspeakers.

Avoiding warning messages during calibration:
- Disable subwoofers built-in filters: a narrow bandwidth can disturb the measurement.
- Adjust levels between amplifier/active speakers before calibration to avoid dangerous level jumps.

Level Adjustment:

The Optimizer requires a minimum Sound Pressure Level for proper calibration. Until that minimum SPL is reached, the test signal will keep playing through the same speaker. For safety matters, we recommend to perform that first calibration as if it were a test.

1. Set a safe output level:
   a. Press the Mute button to cut the outputs
   b. Decrease Master Level to -40 dB
   c. Press the **Calibrate** button
   d. Follow on-screen instructions regarding the microphone.
2. Adjust output level:
   a. Increase the master level until the test signal pass on the second speaker
   b. If possible, you can increase that level to get a better quality measurement and ensure the minimum SPL is reached for every loudspeakers.
   c. Use the **Cancel** button to stop the calibration.

Calibration procedure
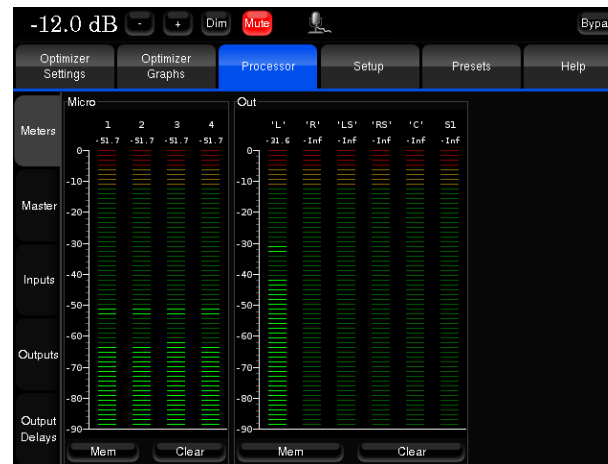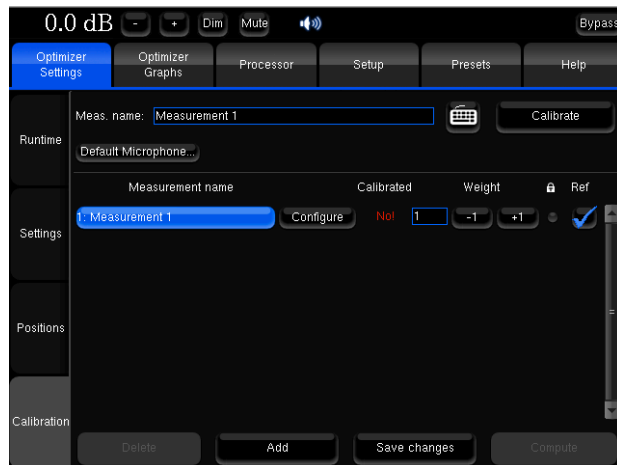
1) Press the **Calibrate** button to launch the Calibration sequence.
2) Follow on-screen instructions.

Normal calibration sequence

- The calibration initiates three MLS bursts for every speaker and subwoofer.
- The calibration page indicates which speaker is being calibrated

Problem during calibration

- The calibration initiates more than three MLS bursts for a speaker and a warning message is displayed:
  - **"Crest factor too low for capsules":**
    - Check the Input Meters in the **Processor/Meters** page.
    - If the Input level is null on at least one of the 4 channels, go back to the **Optimizer Settings/Calibration** page and stop the calibration.
    - Check that the selected analog physical input is selected in the Home/Select page.
    - Make sure the microphone routing is correct.
    - It can also be the PP3 battery inside the microphone that is out of energy, the red led at the front of the microphone indicates the battery level.
    - The microphone or cables could also be damaged.
  - **"Unstable position for speaker":**
    - Someone may have moved during the calibration.
    - Too many strong reflections around the microphone environment disturb the impulse response measurement.
    - Obstacle between the microphone and the speakers disturb the measurement.
    - The bandwidth of one of the speaker is too narrow.

Calibration status

For each measurement, the calibration status indicates Yes if no problem occurred during the measurement. After calibration, the Top Bar indicates NOT READY, which means no sound can go through the processor before computation of the compensation.

If the calibration had to be stopped during the measurement, the status will indicate Partial.

The status displays NO! when the measurement has not been performed.
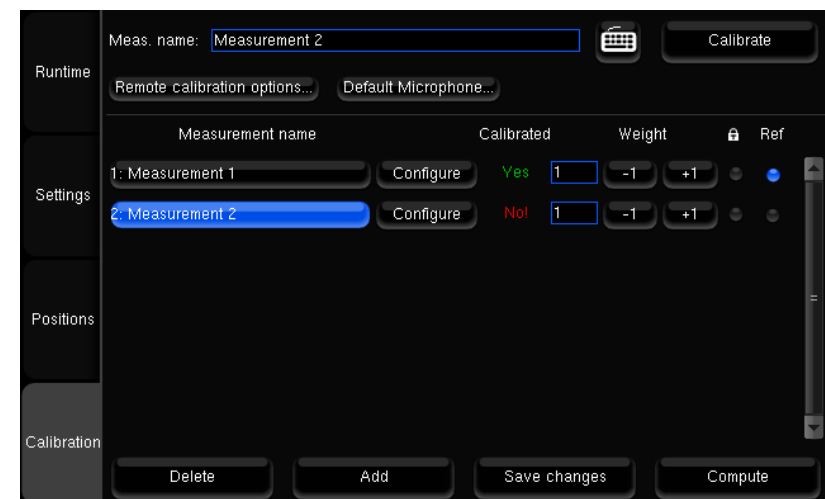
Checking loudspeakers position during calibration

The Optimizer Settings/Positions page displays loudspeaker localisation in real-time during the calibration.
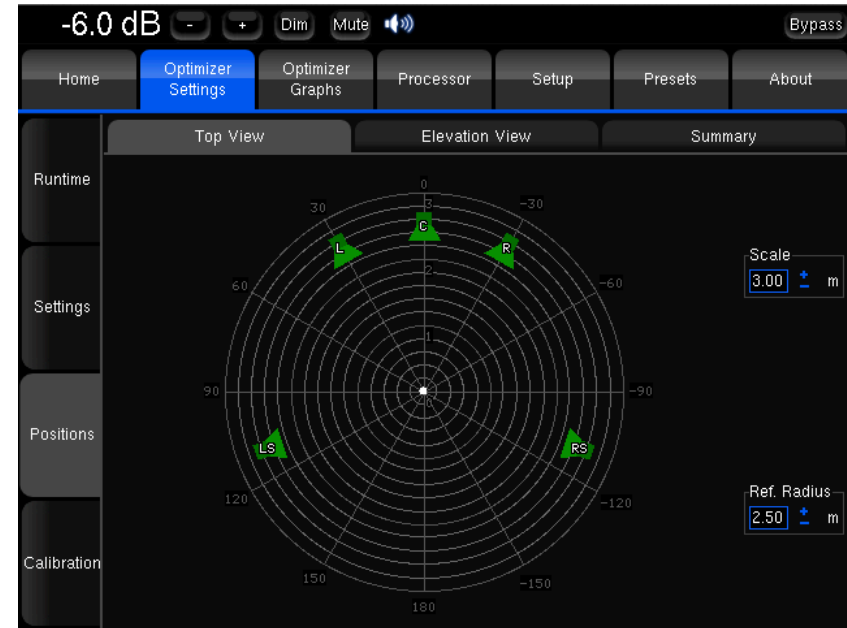
Optimizer Settings/Positions

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

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

By default, the Optimizer page always displays the reference loudspeaker positions in green color. These positions are determined by the input format of the source (Stereo, 5.1 ITU, 5.1 SMPTE or other).
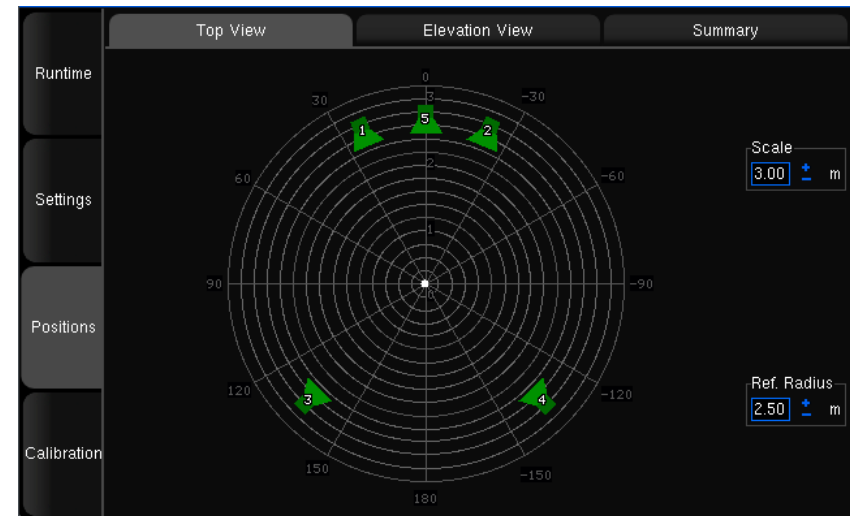
Top View before calibration of a 5.1 SMPTE setup

As the calibration is performed, the real position of each speaker is displayed after the 3rd MLS burst.

• The Top View allows to identify a wrong speaker placement. It might be necessary to recalibrate if the microphone is not perfectly pointed towards the center of the front stage, or activating the remapping will result in a slightly off-axis rotated sound stage.

• The Top view also provides the RT60.

• The Scale and Ref. Radius options allow to overlay both theoretical and actual representations and verify compliance with the placement recommendation.

• The number displayed on each speaker is related to the channel order defined in the sources settings page.

• The color of the loudspeakers depends on the Remapping settings.
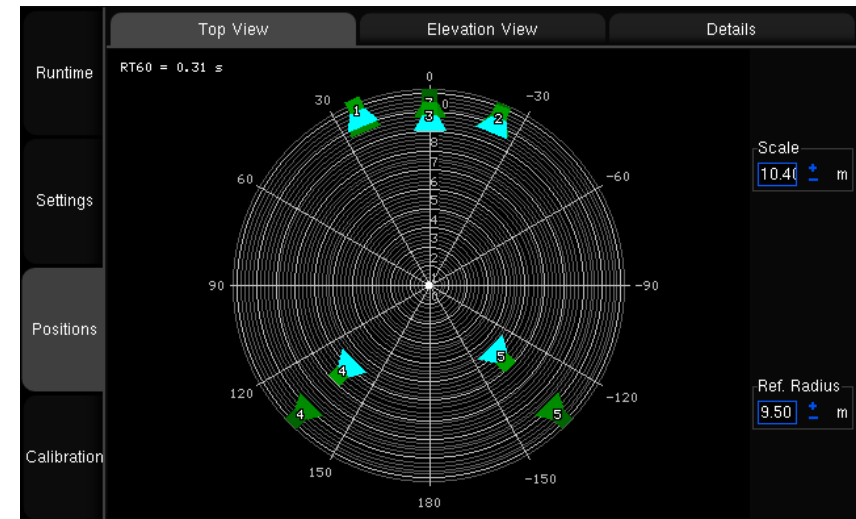
Top View after calibration of a 5.1 setup

• The Elevation View uses the same color codes to display the real speakers position.
• It indicates whether the microphone was or was not correctly leveled during the calibration.

If one speaker cannot be localized during calibration, it will be conventionally displayed at the center position.

An unlocalized speaker cannot be used for loudspeaker remapping.

The Summary page provides localization information for each speaker as numerical values.
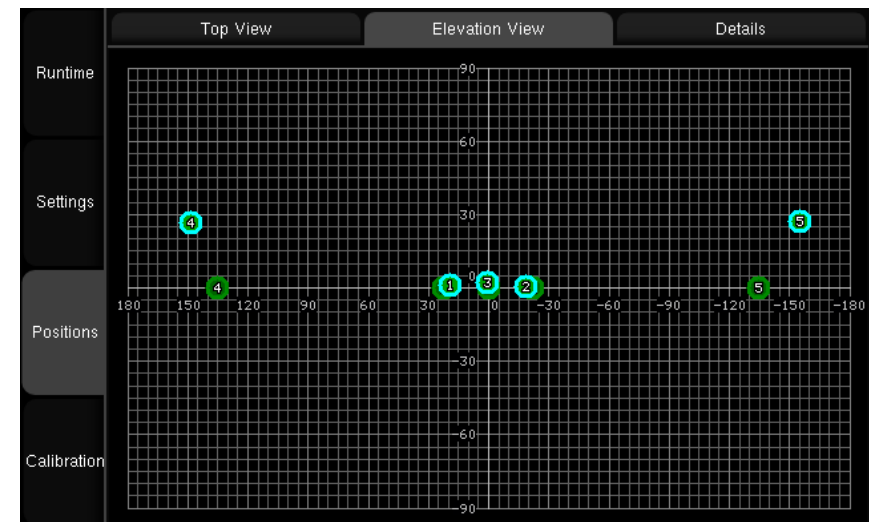
Elevatioon view

Speaker number	1	2	3	4	5	Sub 1
Distance (m)	2.53	2.52	2.47	2.18	2.22	2.28
Elevation (°)	4.2	3.5	19.8	4.3	4.5	
Azimuth (°)	28.7	-28.5	2.3	113.5	-108.4	
Level A (dBFS)	-60.2	-59.5	-59.3	-58.8	-58.6	-45.2
Level C (dBSPL)	92.3	92.5	93.8	94.9	95.7	102.0
Level A compensated (dBSPL)	90.9	90.8	90.6	90.5	90.7	75.9
Level C compensated (dBSPL)	92.4	92.3	92.0	92.1	92.2	88.4
Delay (ms)	7.35	7.35	7.21	6.33	6.48	6.65
BM Delay (ms)	0.00	0.00	0.00	0.00	0.00	3.67
Delay compensation (ms)	0.00	0.00	0.15	1.02	0.88	4.38
Polarity	+	+	+	+	+	-
Crest factor (dB)	36.0	35.6	35.8	36.3	36.0	20.1

Summary Details view

Measurement details:
- **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;
- **Azimuth** is the azimuth of the speaker (in degrees) to the measurement point;
- **Level A** corresponds to the A-weighted level of the speaker;
- **Level C** corresponds to the C-weighted level of the speaker;
- **Delay** corresponds to the distance of the speaker;
- Correction details:
- **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. Low values could indicate background noise problems.

5.6 COMPUTE

After calibration, the Top Bar indicates **NOT READY**, which means no sound can go through the processor before computation of the compensation.


Press the **Compute** button to initiate calculation of the compensation filters and generation of the Optimizer graphs.

The top bar will display a gear wheel and a graph icon during computation.

An onscreen message notifies the end of the computation.

The Speaker icon indicates that the Optimizer is ready.

Computation time varies depending on the number of speakers of the system and Optimizer parameters.

Once the speaker system has been calibrated, every change can be applied through recalculation of the filters. Performing new measurements is not required unless:
- Speakers/Furniture/listening positions change in the room
- Active Xovers filters are modified
- The number of speakers is modified
- The channel order is modified

5.7 MULTI-POINT MEASUREMENT

The Optimizer integrates a unique multi-point measurement algorithm, able to take every single measurement point into account rather than using an averaged multi point response.

The purpose of multi-point measurement is not only to optimize a wider area but also to get more information from measurements and increase reliability by taking differences occuring in the listening area into account.

Performing acoustic measurement in multiple position in the room ensures a more reliable calibration but choosing relevant positions is highly critical.

Multi-point measurement positions

There is no accurate rules in regards to the choice of multi-point measurement positions.

Making measurement outside of the area covered by every loudspeakers can be considered as irrelevant.

Relevant multi-point measurement positions

As an example, multi point measurement can be performed for a single listening position:
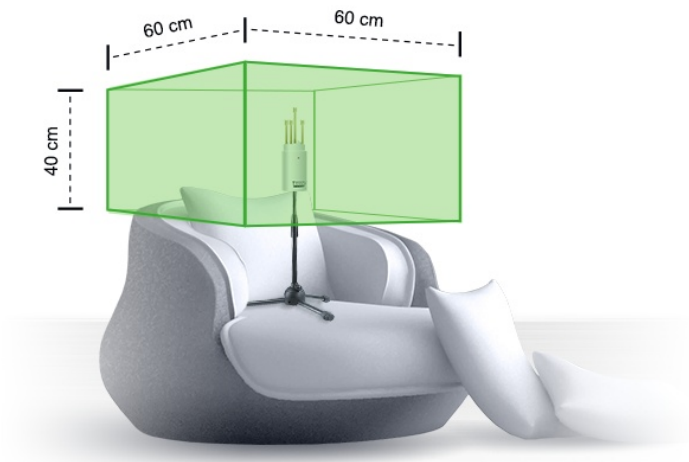
Relevant multi-point measurement area for single listening position
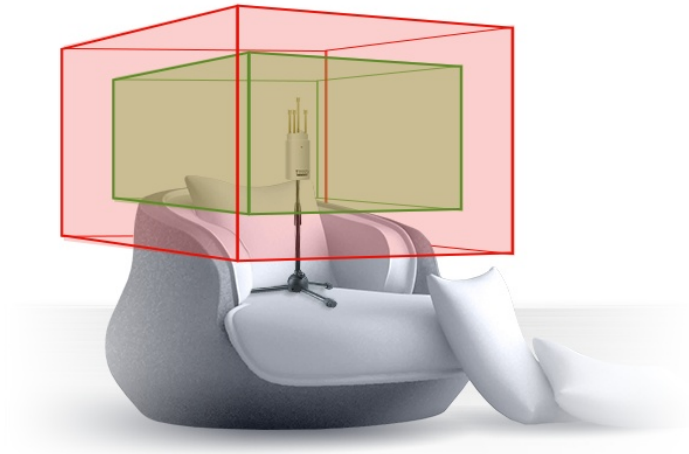
Irrelevant multi-point measurement area for single listening position

Multi-measurement calibration

Multi point measurement can be performed sequentially using the same microphone.

Measures list view

You can add a new measurement by pressing the "**Add**" button at the bottom of the page, or delete the selected measurement by pressing "**Delete**". You can also rename a measurement by editing the text box "Meas. name".

For each measurement, the "**Calibrated**" column indicates "Yes" if the measurement has been correctly performed, "No" if it has not, and "Partial" if the calibration was interrupted before completion.

Each measurement contains the responses recorded during the test sound sequence.

Select the required measurement and press the "**Calibrate**" button to launch the calibration sequence.

Once every measurement has been calibrated, you can compute the acoustic correction filter that will best fit every point, accordingly to their weight.

Multi-measurement weighting

The "**Weight**" column allows you to set a weight to each measurement, 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.

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 to 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.

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** disables measurement settings edition.

The "**ref**" column indicates which measurement is the reference point. The reference point is unique, it can be modified after the calibration. Taking a new measurement into account as reference point requires to recompute the compensation 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.

6 OPTIMIZER GUIDE

6.1 OPTIMIZER GRAPHS

The main Optimizer graphs displays multiple acoustic responses for every speakers as graphic representations:
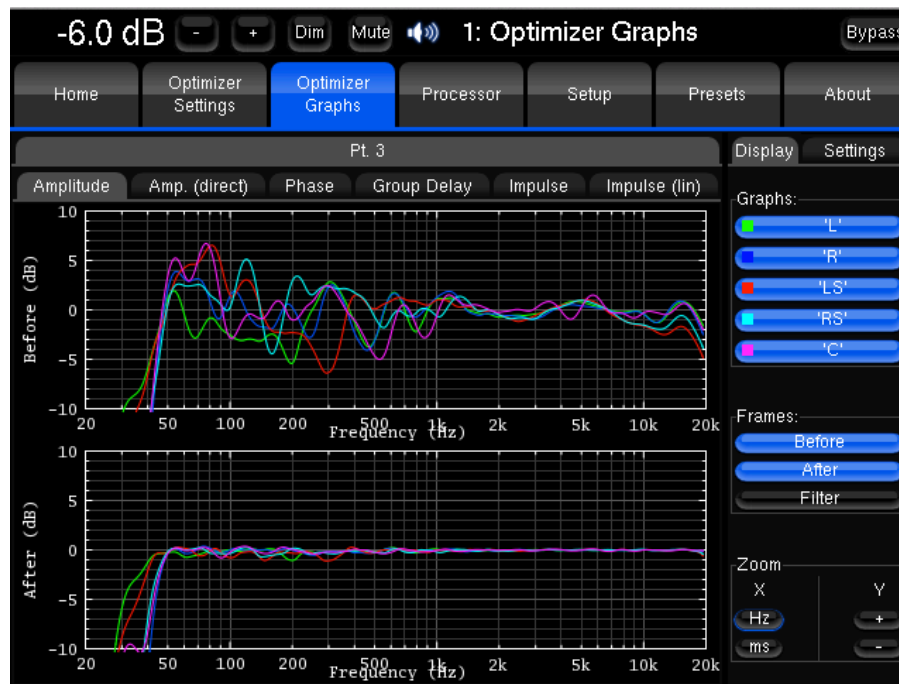
Optimizer Graphs

About the Optimizer Graphs

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

The after optimization graphs can be provided as many times as required to represent the effects of new optimization parameters. These graphs are therefore based on calculations only, not on measurement. Measurements with external kits have always proved the accuracy of these calculations.

The Optimizer Graphs only take under consideration the automatic filters provided by the Optimizer, not the manual additional adjustments.

Overview

The Optimizer graphs consist of a Display area, on the left, and a Config area, on the right.
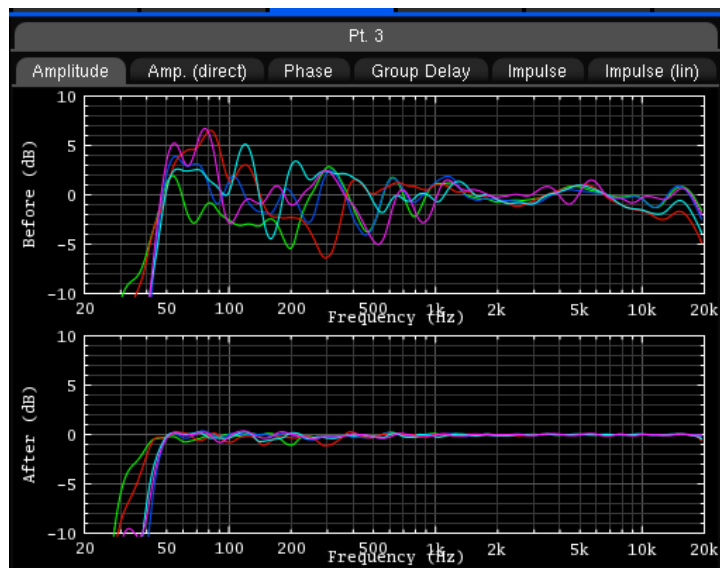
The Display area is organized in **Tabs & Subtabs**, **Graphs** and **Frames**

The Configuration area is organized in **Display filters** and **zoom Settings.** Zoom options allows modifying amplitude, frequency and time scales according to the type of representation.

Graphs Settings
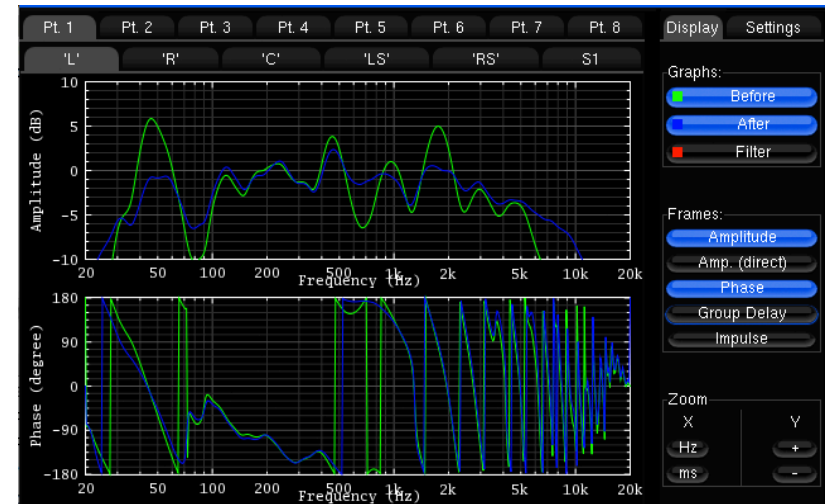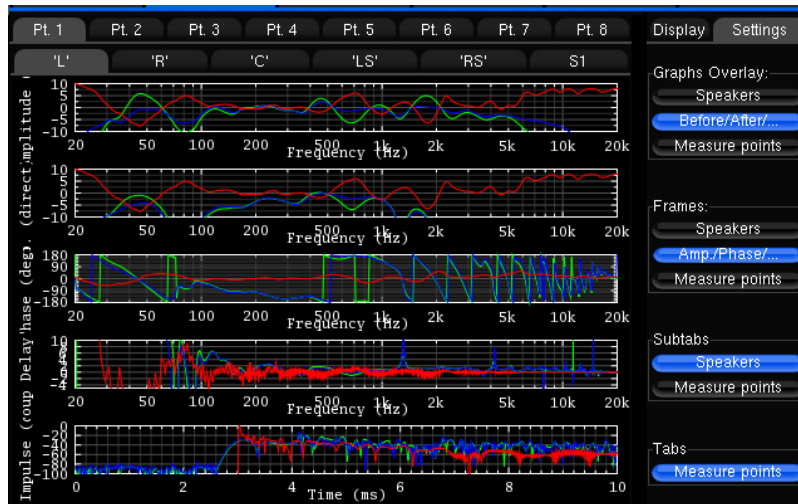
The flexibility of this feature allows you to visualize on a same screen various type of data, which you can organize as you wish by a few parameters:

- the **Graphs Overlay**: type of data you want to overlay on a same graph (or **Frame**)
- the **Frames**: type of data disposed vertically on a same tab
- the type of data you want to visualize on each **tab** and **subtab**.

  For each level, you have the possibility to order the data through various criteria:
- the **speaker** (L, R, C, Sub, …)
- the **point of measurement**: in the case of a multipoint configuration, you can visualize the response measured on each position, and the effect the optimization has on it
- the **type of response**: speaker response before optimization, speaker response after optimization, or filter response
- the **type of visualization**: amplitude, amplitude of the direct front and early reflections only, phase, group delay, or impulse response.

NOTES
The display settings will be saved in preset.

6.2 RUNTIME / BYPASS

While the Optimizer Graphs display both the system's response before and after optimization, the bypass and runtime modes can be used to listen the system with and without correction and enable a proper correlation between the listening experience and graphs analysis.

Bypass

The bypass button located at the top right corner of the graphical user interface is constantly accessible and has the same function as the Optimization Off button described below.

Optimizer Runtime

The Optimizer runtime modes allows the user to selectively enable/disable the different corrections applied by the Optimizer.

- **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:
  - o   routing, as defined in the Setup page
  - o   levels, as defined in the Processor page
  - o   graphic EQs, as defined in the Processor page
  - o   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.
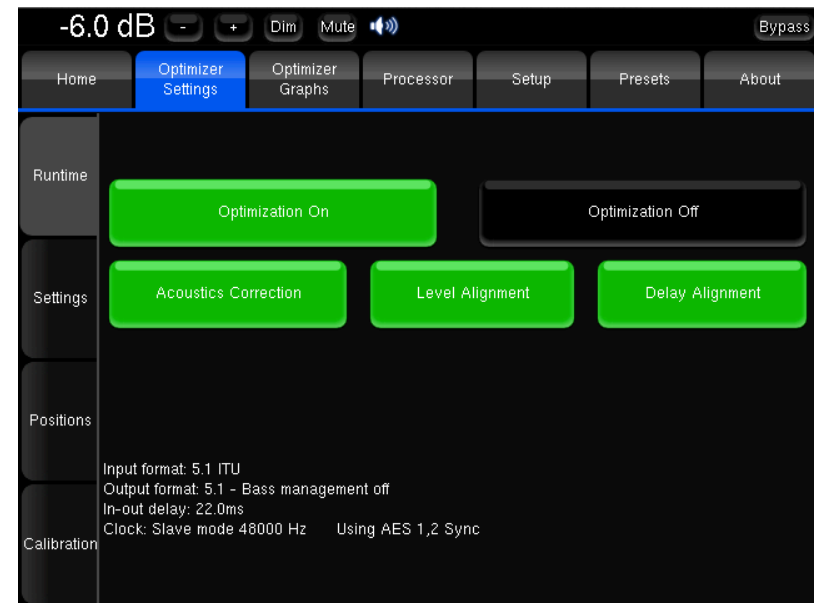
Runtime Settings

6.3 MAIN SETTINGS

Reminder: the Optimizer Settings do not require new acoustic measurements. New parameters are recomputed on the fly using the **Apply changes** button.

The Optimizer Main Settings are located in the **Optimizer Settings/Settings/Main Settings** page.

Optimization Modes

The Optimizer offers 4 different optimization modes with very specific behaviour:

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 the frequency defined in the advanced settings (Default: 150Hz). The automatic FIR filter is disabled, but manual FIR EQ can still be applied;

According to L&R speakers: this is a special mode that will optimize the every speakers in order to achieve the same response as the averaged Left+Right speakers. Different options are available for this mode, please refer to the Advanced settings chapter.

Optimizer Main Settings

77

Maximum Boost/Attenuation

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. It's default value is 6dB.
This parameter has an important impact on the behavior 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 -10dB.
This parameter also has an important impact on the behavior of the automatic equalization, and is applied to both the time-based and the energetic approach.

Acoustic Parameters

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 (default is 3 cycles):
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)*.

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 10 kHz.

Resolution of Energy Response (default is 1/3 octave):
It defines, in number of octaves, how the room's energy response is smoothed, and modifies the behavior 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.

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 every speaker without the sub. A specific button was implemented for this configuration and is used by default.
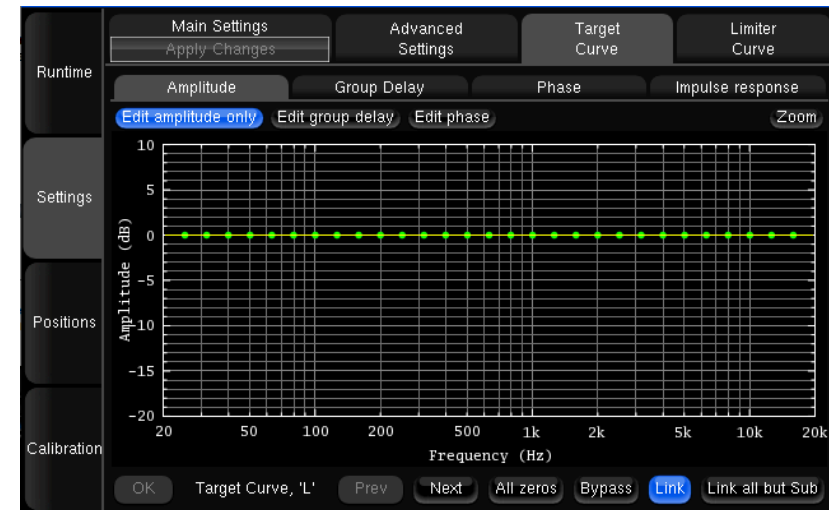
Target Curve

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.

NOTES
• by default, the Optimizer will use both IIR and FIR filters to achieve the target curve.

• in addition to target curves, several other parameters define the behaviour of the automatic equalization.

Limiter Curve

Unlike the Maximum Boost / Attenuation parameters, the limiter curve allows to set up frequency-dependant maximum boosts&attenuation. This curve is not engaged by default.

Limiter curve

Limiter curve example

Use the Add point mode to create editing points around the required frequency range.
Use the Move point mode to edit the limiter curve for the required frequency range.
Use the Delete point mode to delete editing points.

Steep limitation should be avoided.
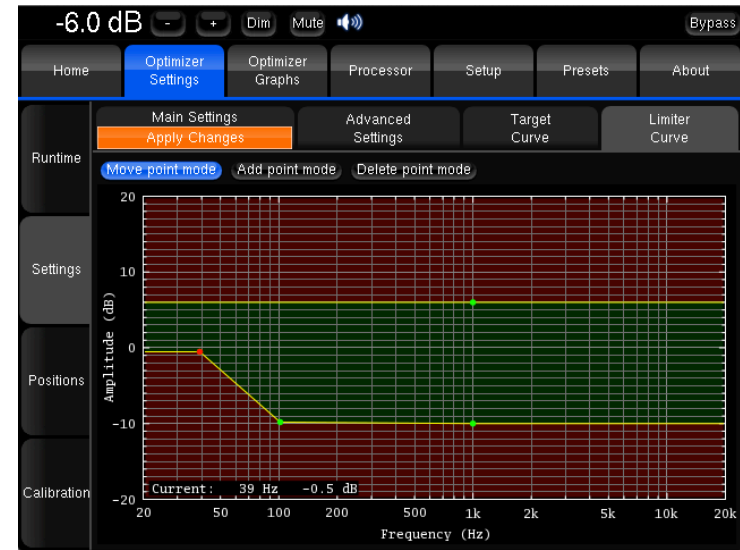
As an example, the limiter curve above will keep the Optimizer from attenuating too much below 100 Hz.

Please note that editing the maximum boost and maximum attenuation limiter curve disables the global Maximum Boost/Attenuation settings in the Optimizer Main Settings page.

6.4 ADVANCED SETTINGS

The Optimizer provides a large set of advanced settings in the **Optimizer settings/Settings/Advanced Settings** page.

Advanced Settings

Display

The **Display smooting** defines the smoothing value used to display the Amplitude Response in the Optimizer Graphs as well as **in the pdf document** generated while saving preset. Its default value is (1/3 octave). A smaller value, such as 1/12 octave, can be used to display more details in the frequency response.

Front & Surround Speaker settings

The following features can be separately enabled/disabled for the front speakers and on the surround speakers:
- Early reflections correction;

- Energy response correction;
- Delay alignment, e.g. surround delay alignment may or may not be required depending on the application and recommendations;
- Add decorrelation: 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;
- Remapping: in certain situations, such as in dubbing theatres, it may be preferable not to apply remapping to the front speakers.

Please note that Front speakers are defined as those whose azimuth is below 90 degrees. Conversely, Surround speakers are defined as those whose azimuth is above 90 degrees.

Acoustics Correction

The user can determine which filters the Optimizer should use to perform the automatic correction.

- **FIR + IIR**: this is the 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 will not 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.

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. -15 dB is the default value and works in most cases. However, a lower threshold is recommended in a room with a huge amount of early reflections where the peak might not be detected correctly.

Optimize according to L&R speakers settings

The Optimizer mode "According to L&R speakers" can be configured with specific settings:

- **Processing on L&R target** (default is IIR only): if the optimize mode is set to "Optimize according to L&R speakers" in the 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.
- **Align L&R on target** (default is On): if this setting is on, the Optimizer will optimizer the Left & Right speaker using their average response as target. If this setting is off, only the center and surround channels will be optimized.
- **Optimizer Phase** (Default is On): determines whether the Optimizer should optimize the phase response of the speakers or not when the According to L&R speakers mode is on.

FIR and IIR settings

FIR stands for Finite Impulse Response.
FIR filters are used to compensate for amplitude and/or phase

IIR stands for Infinite Impulse Response.
IIR filters are minimal phase filters used by the automatic compensation in addition to FIR for the low frequencies.

- **FIR filter length** (default is 100ms): defines the length or number of taps of the FIR filter. The longer the FIR filter, the higher the resolution of the correction is.
- **Number of IIR filters** (default is 10): the number of IIR filters that will be used on every channel.
- **IIR filters minimal/maximal frequency** (default is Automatic/150Hz): IIR filters will be positioned from the min frequency up to the max frequency. The Automatic parameter uses the minimum frequency of the speaker bandwidth and the Low-freq auto transition bandwidth parameter to calculate the IIR filter minimal frequency.
- **Low-freq auto transition bandwidth** (default is 1 octave): determines the bandwidth of the auto transition. The auto transition is the frequency range that separates the Optimized frequency range from the unfiltered frequency range in the low end. The upper boundary of this auto transition bandwidth is the minimal frequency of the bandwidth of the speaker.
- **Room smoothing method** (default is Modulus): for advanced users only.
- **Norm used for level_hp (!= align)** (default is Room): for advanced users only.

Level alignment settings:

- **Weighting used for levels** (default is dBA): sets the type of weighting used by the optimizer for level alignment.
- **Width of level window** (default 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 is 10dB/-20dB): defines the maximum/minimum gain that will be applied for the automatic level alignment.
- **Minimal/maximal bandwidth frequency** (default is 10Hz/Unlimited): defines the high end/ low end of the bandwidth that is used as the starting point for level computations.

Advanced Energy Response smoothing settings

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

Subwoofer low-pass filter settings

The Optimizer includes a subwoofer low-pass filter with the following options:

- **Cutoff frequency** (default is "disabled"): defines the cutoff frequency for the low-pass filter that can be applied to the subwoofer.
- **Filter type** (default 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 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 is 0.1 dB): sets a maximum amplitude ripple above the cutoff frequency.
- **Rs value** (for elliptic filter) (default is 80 dB): defines the attenuation below the cutoff frequency.
- **Subwoofer delay alignment:** the automatic delay alignment can be disabled to time align the subwoofer manually.

Decimation settings & Advanced FIR settings

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

Advanced FIR settings:

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

6.5 2D/3D LOUDSPEAKER POSITION REMAPPING

About the remapping

The Loudspeaker remapping is one of Trinnov's patented technology, based on the ability to calculate the acoustic field that is produced by a set of loudspeakers.

Its purpose is to reduce spatial distorsion caused by a wrong loudspeakers positioning or to increase spatial resolution by using more speakers.

A typical application for the Loudspeaker reamapping is illustrated below.

Another typical application is to use a 7.1 speaker layout to reproduce a 5.1 soundtrack with an increased spatial accuracy.

Principle

The source system layout is the reference speaker positions, defined by the user-defined input format. The target speaker layout is the real positions of the speakers, localized during calibration with Trinnov's 3D measurement probe.

The remapping therefore relies on an accurate localisation of the speakers during calibration.

Remapping mode can be enabled instantaneously within the Optimizer Settings/Settings/Main Settings page.
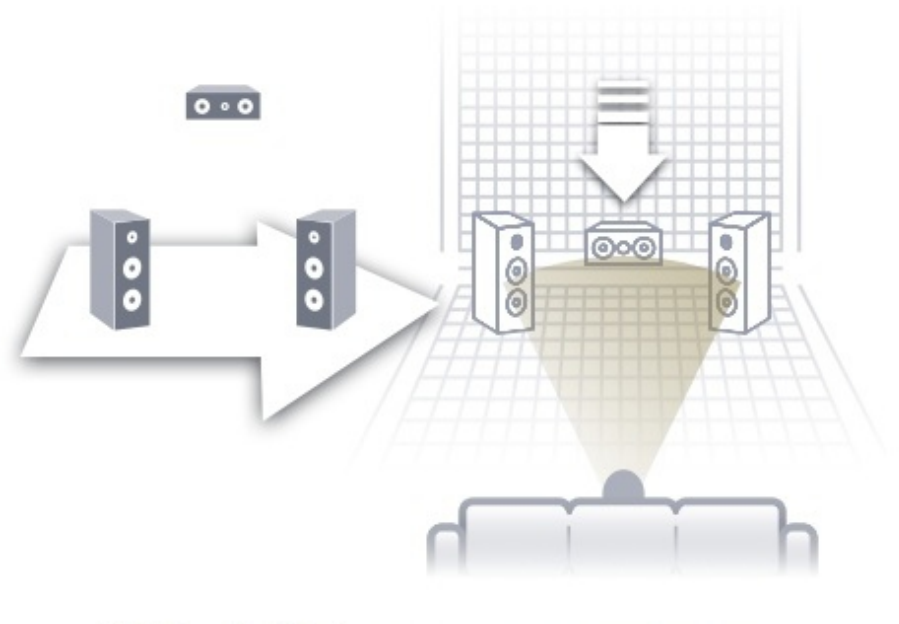
Remapping modes

Off (default):
Remapping is disabled.

Matrix:
Specifies that the i/o routing to be taken into account is in the Config Editor. To be used by advanced users.

Automatic routing:
This option uses the Optimizer's 3D speaker position information (from the calibration) to automatically route each channel to the speaker closest to the reference position. Auto-route works with all input formats (defined in Setup -> Sources settings). For example, to find out which loudspeaker corresponds to the "Left channel" (reference position = 30°), the Optimizer will look which of the measured loudspeakers is closest to 30°. If the nearest one is loudspeaker number 4 at 25°, then the Optimizer will "auto-route" the Left channel to Loudspeaker 4.
- This feature frees the user from having to manually verify that the speakers routing is correct. When activated, the Optimizer automatically sends each channel to the loudspeaker that is nearest to its intended position.
- This option can be more convenient when no remapping is required but you don't want to manually verify that each loudspeaker is cabled on the right output.

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.

When the 2D and 3D remapping are enabled, speakers will respectively be displayed in orange and red.

Optimizer Main Settings

Use cases
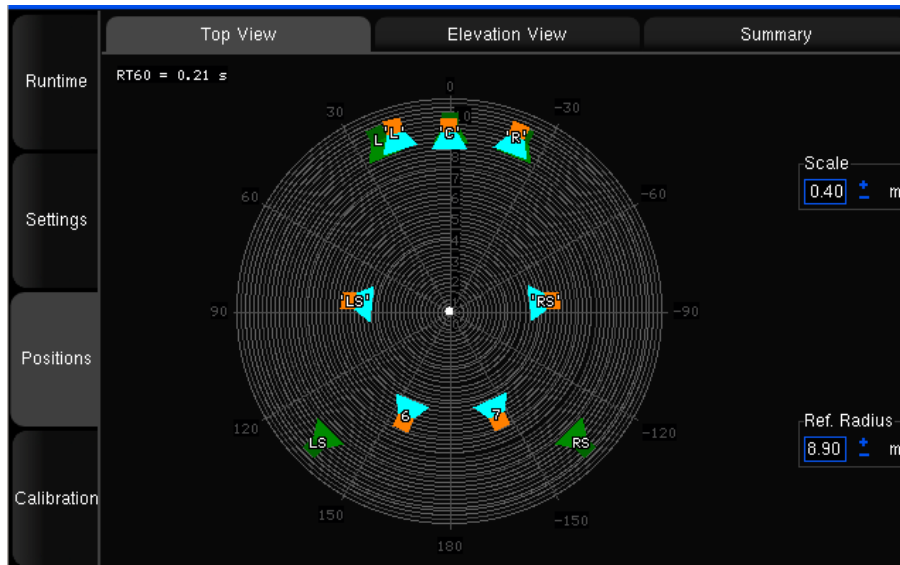
2D Remapping enabled

3D Remapping enabled

2D Remapping:

In this case, the remapping will not only add delay to virtually push sides and back speakers backward but will also compensate for a wrong azimuth.

The user-defined input format is 5.1 SMPTE but the real speaker layout is 7.1.
The 2D remapping will perform a spatial upmix and use both sides and back speakers to reproduce the Left and Right surround channels of the 5.1 incoming soundtrack.

3D Remapping:

The remapping uses adjacent speakers to virtually correct wrong speaker placement. In this example, the wrong center elevation will be compensated by 3D remapping, but surround speakers height positions won't be compensated for.

6.6 BASS MANAGEMENT

About 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.

Bass Management modes

The Bass management mode can be configured after the calibration in the Setup/Speakers page.

Speakers

IMPORTANT NOTE

• 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.

The bass management modes are as follows:

Off

This means that no bass management is performed. In other words, the 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.

Stereo

This bass management feature maintains stereo bass: the low frequencies from the Left channels (for example L and Ls) are sent to the first Subwoofer (S1) and the low frequencies from the Right channels are sent to the second Subwoofer (S2). The low frequencies from the Center channel are distributed equally between both 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.

NOTES

• 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.

• 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.*

7 MANUAL ADJUSTMENTS

As a complement to the Optimizer automatic equalization or as a stand-alone system, the Magnitude32 includes a FIR Equalizer and Graphic Equalizers, levels and delays adjustments on every channel.

7.1 LEVELS AND DELAYS ADJUSTMENTS

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

Master levels and delays

The following options are offered in order to adjust levels and delays for all channels at the same time:

- **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.
- **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).
- **Level display 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. To do so, the system must have been calibrated and the dBC button selected. The Legacy option displays the level according to another cinema standard scale with values from 0 to 10.

  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 is 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. The preset needs to be saved and reload to effectively apply the offset.
- **Level offset when remapping is On:** this level offset will be applied when 2D or 3D Remapping is activated. The preset needs to be saved and reload to effectively apply the offset.
- **Master Delay** can be used to apply an additional delay to all channels and all presets. It is not stored in the preset.
- **Relative Delay** can be used to modify the delay of a preset.

  The following latency information is available:
- **Processing Latency** corresponds to the latency of the processor algorithms. It can be modified by changing the **Optimize** setting (Amplitude + Phase has higher latency than Amplitude only) or the **Audio Buffer Size** (in Setup/Clock Settings).
- **Master + Relative Delay** is the sum of the master and relative delays of the Processor/Master page.
- **In-out Delay** is the sum of the Processing Latency and the 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 measurement point. When Time Alignment is activated, all the other speakers will be time-aligned to the furthest speaker.
- **Total delay at measurement point** is the delay from one input until the sound reaches the measurement point.

Reducing Latency

The following parameters affect the processing latency:

- **Audio Buffer Size** (Setup/Clock page):
  - 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
- **Sampling Rate** (Setup/Clock page):
  - The higher the sampling rate, the smaller the latency
- **Optimize Mode** (Optimizer Settings/Settings/Main Settings):
  - The amplitude only mode reduces latency.

7.2 CHANNEL-SPECIFIC LEVELS AND DELAYS

Inputs and outputs levels and additional delays can be manually and individually adjusted. Solo and Mute functions are found in the Outputs but also Inputs pages because the result can be different if the Remapping is activated (in that case one input signal can feed several loudspeakers). Another typical use is to set +10dB to the input for LFE channel.
These settings are to be saved with a preset in the Setup/Presets page.

Input Levels per channel

Output Levels per channel
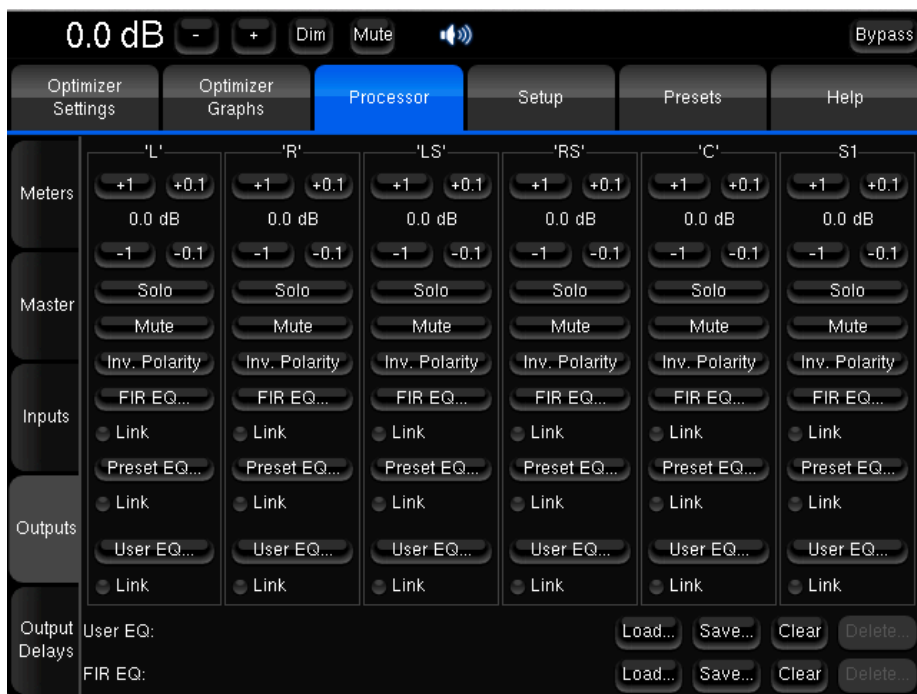

Output delays per channel

7.3 FIR EQ

Input Levels per channel

FIR EQ – Amplitude

Inputs and outputs levels and additional delays can be manually and individually adjusted. Solo and Mute functions are found in the Outputs but also Inputs pages because the result can be different if the Remapping is activated (in that case one input signal can feed several loudspeakers). Another typical use is to set +10dB to the input for LFE channel.
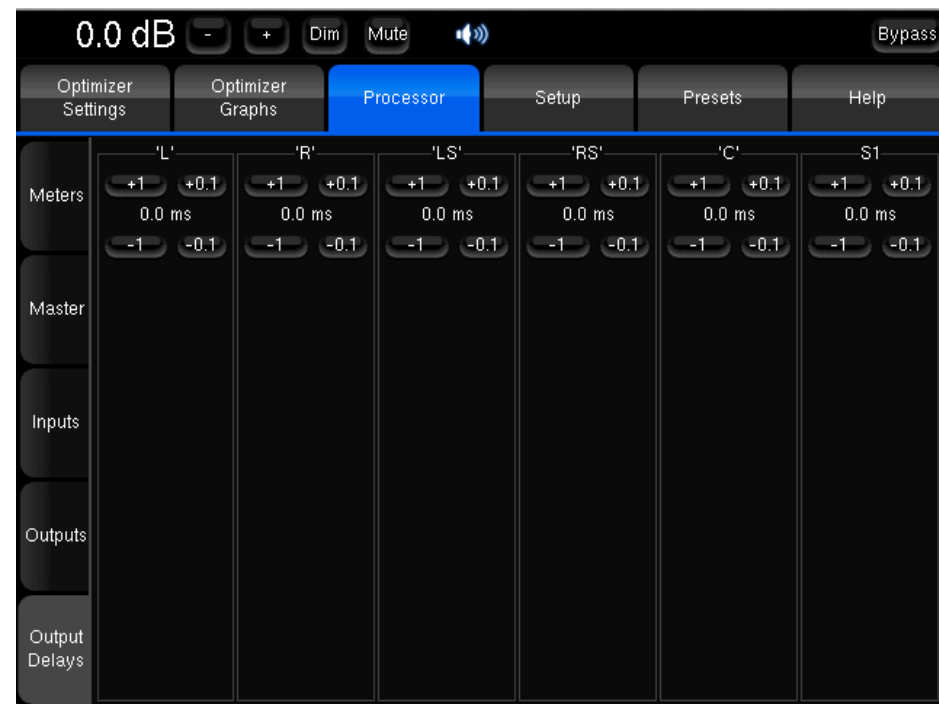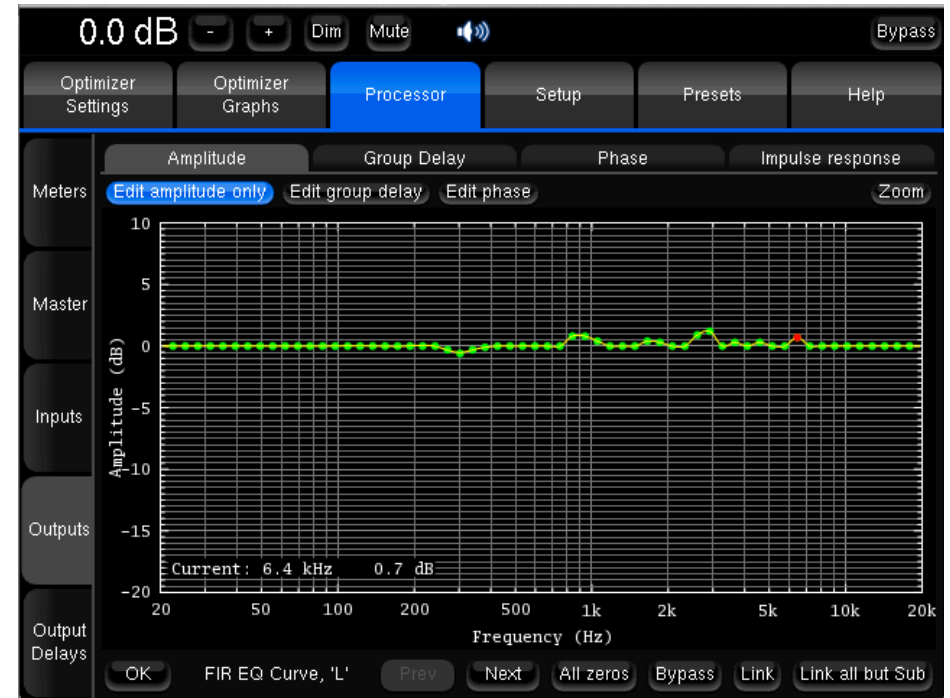These settings are to be saved with a preset in the Setup/Presets page.

For every speaker in the room, a FIR EQ 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 apply the same FIR EQ to several speakers.

The required curve can easily be edited via the touch screen or 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.

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

7.4 31 BAND GRAPHIC EQS

In addition to manually setting levels, the Processor/Inputs and the Processor/Outputs pages include 31-band, 1/3 octave Graphic Equalizers that allow for manual equalization of each input or output channel:
- Input EQ is available on the input channels,
- Preset EQ and User EQ are available on the output channels

The Input EQ behaves exactly in the same way as the Preset EQ described below. The only difference is that it is applied to the inputs instead of the outputs.
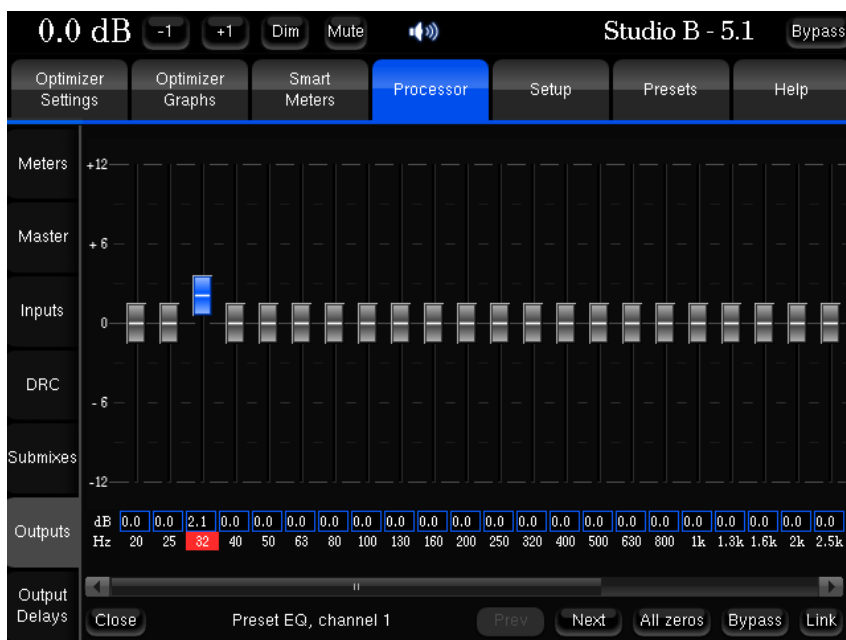
Preset EQ

User EQ

The "Preset EQ" and "User EQ" buttons in the Processor/Output page give access to two independent Graphic EQs. Both EQs have the same interface (only the current EQ name at the bottom left of the window is added for the User EQ) but are saved in 2 different places.

The Preset EQ will be saved and reloaded with the whole Preset, through the usual Presets page:
- Press Preset EQ, this shows the Graphic EQ for the selected channel,
- move the faders to setup your equalization,
- use the Previous/Next buttons to move to other channels,
- press Close to come back to the Processor/Output page
- go to Presets page to save these changes in a preset.

The modifications on these filters are applied in real time so you can hear them, but they are not visible on the frequency response curves in the Optimizer Graph page.
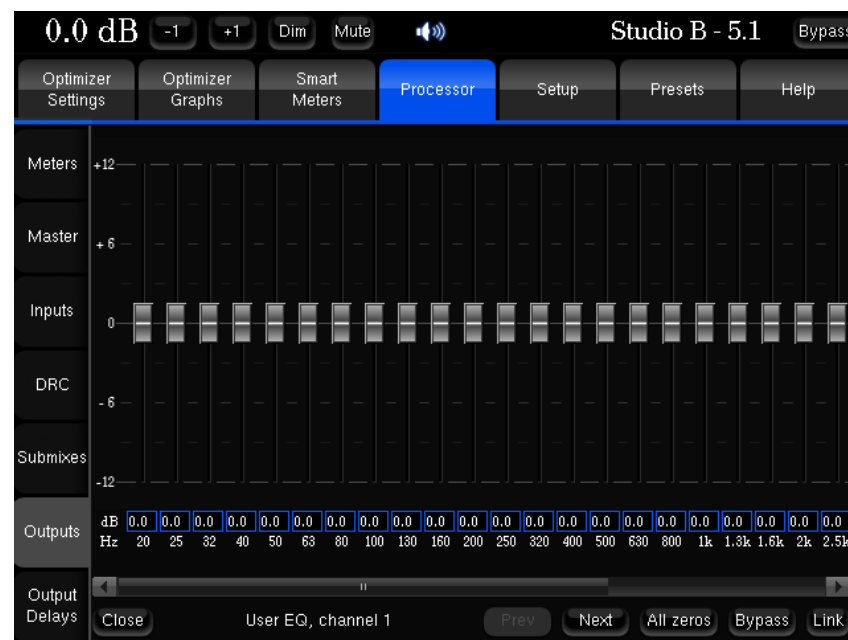
Multiple User EQs can be saved and reloaded, independently from the current preset:
- Press User EQ, this shows the EQ for the selected channel,
- move the faders to setup your equalization,
- use the Previous/Next buttons to move to other channels,
- press Close to come back to the Processor/Output page,
- press the Save button at the bottom-right to save this User EQ and give it a name, using the virtual keyboard or a connected keyboard,
- you can now see the name of your new current User EQ at the bottom of the Processor/Output page.

This gives you the flexibility of having one "reference" EQ (User EQ) that can be used on top of Optimizer presets. As an example, User EQ could be recalled for a specific room. This also could be used as "User EQ" according to the users preference.

7.5 SUBMIXES

A submix is a specific matrix used to inject any input signal to any output signals with the required attenuation/polarity.
The submix can be edited in the Processor/Submixes/Matrix Editor page.
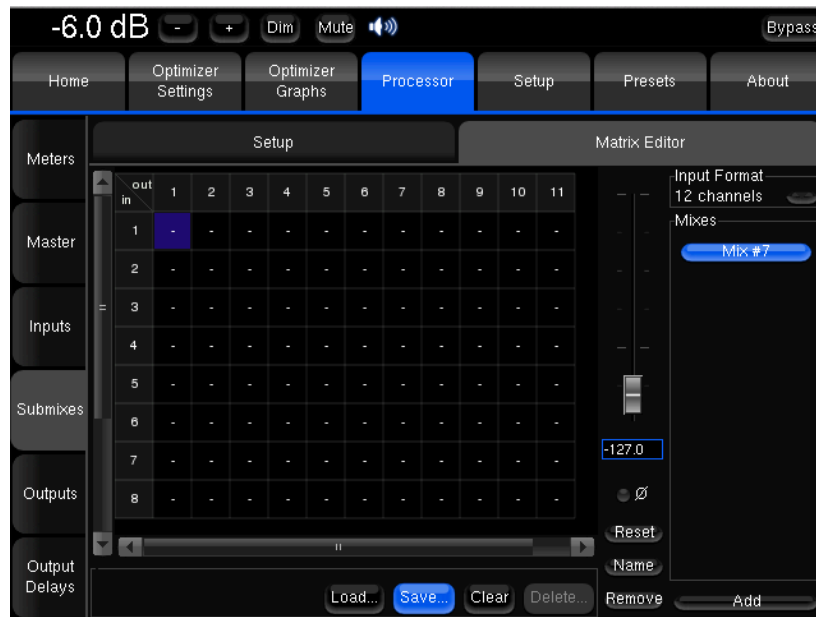
Submixes

Submixes activation

Submixes editing

Choose the input format of the matrix and use the Add button to create a submix matrix.
The submix matrix can then be renamed using the Name button.
The Matrix lines are input channels and the matrix columns are output channels.
Click on a matrix field to edit it.
Right click on another matrix field to add it to the editing group.
Use the fader, the mouse's scroll wheel or the numerical field to edit level.
0,0 dB means that the input signal Is sent to the required output without attenuation.

Use the polarity button to invert polarity. The field will then be displayed with a red underline.

Submixes activation

Once the submix has been edited, go to the Processor/Submixes/Setup page.

Provided that the source format correspond to the subxmix matrix format, the custom matrix will be displayed and will be available as Active Downmixes & Matrixes.

Select it and tick the speaker box to apply the matrix instantaneously.

8 NETWORK OPERATIONS

The magnitude32 can perform various actions over the network:

Remote control / Automation   Software updates & remote assistance

Remote Control / Automation

Trinnov Control Protocol

Trinnov's telnet protocol makes it easy to program any automation system for remote control of the magnitude32 through the network.

Most functions can be controlled with an automation system:

Volume control	Optimization bypass	Acoustic Correction
Preset change	Delay Alignment	Remapping
Input selection (Profiles)	Level Alignment	Bass Management

Savant

Trinnov is partner of Savant's partner in excellence program. The module can be retrieved directly from Savant's website.

Crestron

The magnitude32 can be remotely controlled by a Crestron automation system. A Crestron module is available and can be downloaded from the Support / Download section of our website www.trinnov.com

Please note that the magnitude32 cannot be remotely controlled via a Crestron system using RS232, unless implementing the module from Trinnov's RS232 protocol.

Software Updates & Remote Support

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 offices 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"***.

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 read the Troubleshooting chapter about Network Connection.

9 KNOWN ISSUES AND TROUBLESHOOTING

9.1 KNOWN ISSUES

Clicks and Sync losses

Every Magnitude32 was designed to run at sampling rate up to 96 kHz. However, depending on the amount of processing power that is used by the Optimizer, running at the highest sampling rate can result in CPU overload, the audio outputs will click constantly and Optimizer will lose the Sync.

This problem can easily be avoided by increasing the buffer size in the **Setup/Clock Settings** page. You can also try to change the Optimize setting to "Amplitude only" or "Low range only".

9.2 TROUBLESHOOTING

Calibration

Here is a list of settings to check if your calibration does not succeed.

Synchronisation :
> Page: Setup → Clock
> Check: Is "Current sample rate" information correct and stable?

Loudspeaker number:
> Page: Setup → Speakers settings → Speaker number
> Check: Does the number of connected loudspeakers match this number?

Input channel order
Page: Setup → Sources → Input format, and
> Page: Setup → Sources routing
Check: is this the right order of the connections between the source and the Optimizer (also regarding possible modifications of the sources routing)?

Output Channel order
> Page: Setup → Speakers routing
Check: does the speakers routing (lines) correspond to the wiring of the loudspeakers (columns)?

Microphone signals
Once the calibration is started and if you can hear the test signals, you should see the 4 input signals on the level meters corresponding to what the microphone is recording. If not, check the microphone routing:
Setup → Sources routing → Micro
Check also the power switch and the battery of the microphone.

Warning message at start-up
You may have chosen a default **preset** at startup, which could cause problem after changing your installation (synchronization for example). It is then possible to choose not to load this **preset** by pressing "no default config" on the startup screen, without forgetting to press "audio mode just after to validate your choice.

Network Connection for Software Updates & Remote Support

In order to be able to perform software updates, and connect the processor for remote support, the Network Status in the Setup/Network page must display "Connected to Trinnov Audio Server".

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. The Trinnov Server, located in Trinnov's offices near Paris, is listening on port 22 on the internet address "bry.trinnov.com". To check whether your firewall is blocking outgoing connections to port 22, you can perform the following command from your computer's terminal:

telnet srpserver.trinnov.com 22

The result should be:

Trying 217.128.95.110...
Connected to bry.trinnov.com.
Escape character is '^]'.
SSH-2.0-OpenSSH_4.6

If you cannot connect from your own computer to the Trinnov Server, then the processor will not be able to connect either. You should contact your network administrator and ask him to **open outgoing connection to port 22**. Please note that it is not necessary to open *incoming* connections, only *outgoing* connections to port 22.

10   USEFUL TIPS

10.1   AVOIDING FEEDBACK LOOPS

- Remember: always **MUTE** the processor as the first step of the calibration procedure. Muting the outputs does not affect the calibration signal: even if the processor is muted, it will play the calibration signal through the outputs defined in the Speakers Routing;
- If you change the routing, be careful with feedback issues, you could make loops between the microphone and the loudspeakers. See **Setup/Sources Routing**.

10.2   POSITIONING AND ORIENTATING THE MICROPHONE

The following tips only apply when the processor is used on a layout where the Left and Right loudspeakers are positioned *symmetrically* with respect to the listening position. In this case the microphone's placement and orientation are very sensitive.

- First of all, it is very important to position the microphone *exactly on the axis of symmetry*. Otherwise, if the microphone is slightly off axis, it will measure a different distance for the L & R speakers and try to compensate for it, which will shift the stereo image sideways.

- Furthermore, when using the Remapping functionality, the microphone's orientation is also very important: it should point exactly in the middle between the L & R speakers

Here are a few simple rules to position and orientate the microphone correctly and achieve consistent results with the Optimizer:

1) During the calibration, use the Positions --> Details tab to check the *distance* (R column) and azimuth (Phi column) of the L & R loudspeakers;

2)   Move the microphone and repeat the calibration until the distance and azimuth of L & R speakers are *almost identical* (within 1 or 2 centimetres for the distance, and within 1 or 3 degrees for the azimuth)

Please note: when a Center loudspeaker is used, and positioned exactly in the middle between the L & R speakers, you should also check that the measured azimuth of the Center speaker is zero degrees.

10.3   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.

To reduce latency:
- You can change the sampling frequency and the buffer size of your unit. This can be done in the **Setup/Clock** tab. 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 and that you don't have an insert S&R on your mixing desk that can be made inactive, you can just **bypass** the Optimizer using the bypass button 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 end, turning the Optimization Off in the **Optimizer Settings/Runtime** tab does not reduce latency since the audio still pass through the PC.
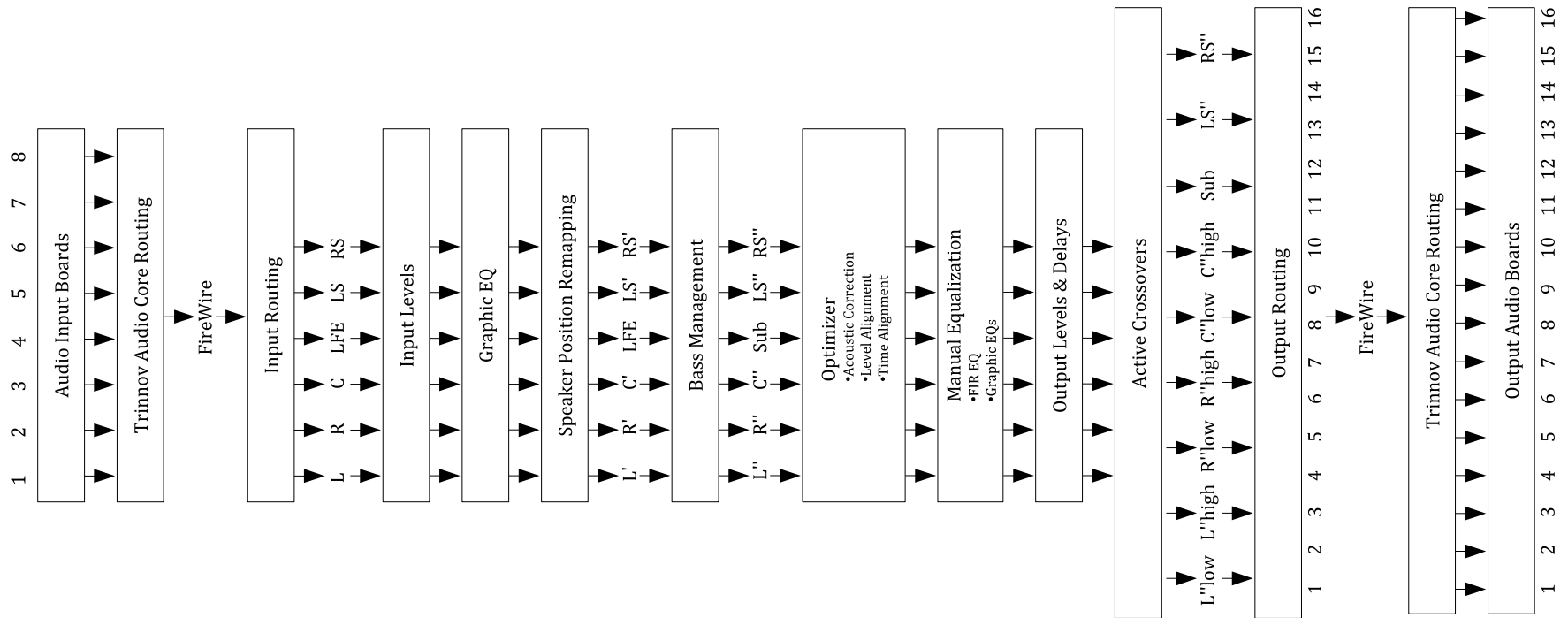
Please note: Latency can be monitored in details in the **Processor/Master** page.

11   APPENDIX

11.1   ARBORESCENCE OF THE MENUS
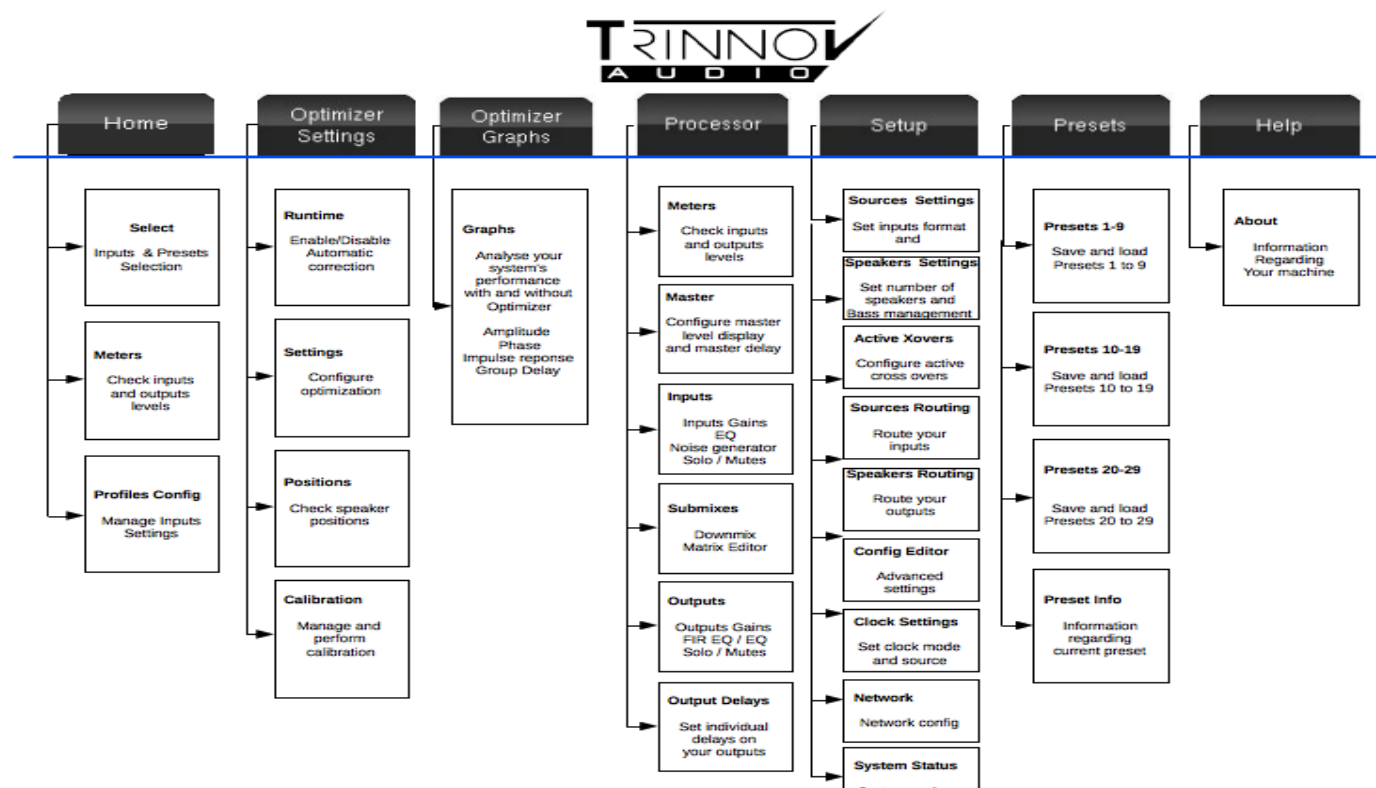
11.3   ABOUT YOUR PROCESSOR

The 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,
- the Product ID of your processor, required as a password when connecting via VNC,
- the 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 magnitude32 model that was purchased),

11.4   STARTUP OPTIONS

• **Double speed mode**: this option does not concerne magnitude32 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.

o **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.

11.5   AUDIO INTERFACES & CHASSIS

Magnitude32 are equipped with the new, high performance, Trinnov audio boards:

Trinnov Audio Core (TAC)

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.

Trinnov ADA4

The Trinnov ADA4 is connected to the TAC, executes AD/DA conversions and offers 4I/Os analog channels.

Trinnov DA8

The Trinnov DA8 is connected to the TAC, executes D/A conversions and offers 8 analog outputs.

Physical characteristics

Chassis: 2U
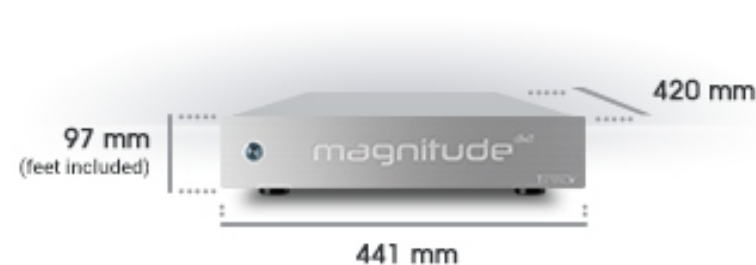**Power supply:** 240V AC / 50-60 Hz. Option: 130V AC
Consumption: 90 W max.
Weight: ~10,2kg
Environnemental conditions:
- Operating: 0°C – 40°C (32°F – 104°F)
- Humidity: 20% – 80% relative humidity (without condensation)

97 mm (feet included)   420 mm   441 mm

Audio interfaces

The following table specifies the audio interface used in each Magnitude32:

Magnitude32	Audio Interface	Description
mg32-88d	Trinnov Audio Core + 2 ADA4 + 1 AES	8 analog inputs/outputs Upgradable to 1212, 1616, 1624 ou 1632.
mg32-1212d	Trinnov Audio Core + 3 ADA4 + 1 AES	12 analog inputs/outputs Upgradable to 1616, 1624 ou 1632.
mg32-1616d	Trinnov Audio Core + 4 ADA4 + 1 AES	16 analog inputs/outputs Upgradable to 1624 ou 1632.
mg32-1624d	2 Trinnov Audio Core + 4 ADA4 + 1 DA8 + 1 AES	16 analog inputs, 24 analog outputs Upgradable to 1632.
mg32-1632d	2 Trinnov Audio Core + 4 ADA4 + 2 DA8 + 1 AES	16 analog inputs, 32 analog outputs

Technical specifications

Magnitude32 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

Magnitude32 processors are powered by a quad-core Intel i5 to process a maximum of 32 simultaneous audio channels at 96 kHz. Despite this amount of power, the unit remain silent thanks to custom heat sinks and additional slow fans. Magnitude32 also use flash disk technology removing the need for a mechanical hard drive, and ensuring mechanical robustness.

11.6   MISCELLANEOUS

Delay Lines

Delay lines are provided in order to delay additional audio systems with the outputs of the Magnitude32 without being affected by the Optimization. Delay lines typically are used in broadcast environment to align audio monitoring between audio and video control rooms.
Once created in the Speaker Settings page, delay lines can be set up in sources and speakers routing grids such as in the following example.

In the example below, we created 5 delay lines to align a 5.0 monitoring system with the Trinnov Optimized speakers. The first screenshot shows that five additional lines are consequently created in the Sources Routing grid. Inputs 1 to 5 have been affected to the active software source while inputs 6 to 10 as alternative sources to be routed to the Delay Lines outputs. To set a mirrored output of the main source, we would just have selected Inputs 1 to 5 instead.

In the Speakers Routing Page, selecting Outputs for the Delay Lines will define the Outputs towards which the previously selected Delay Lines Inputs will be sent. In the example, Inputs 6 to 10 are routed to the Outputs 6 to 10.
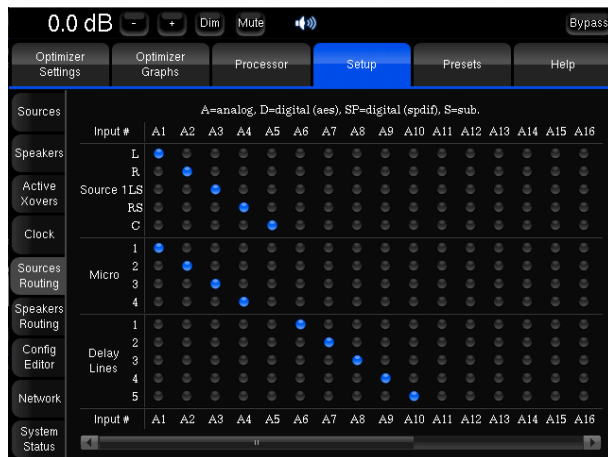
Physical inputs 6 to 10 are affected to the delay lines
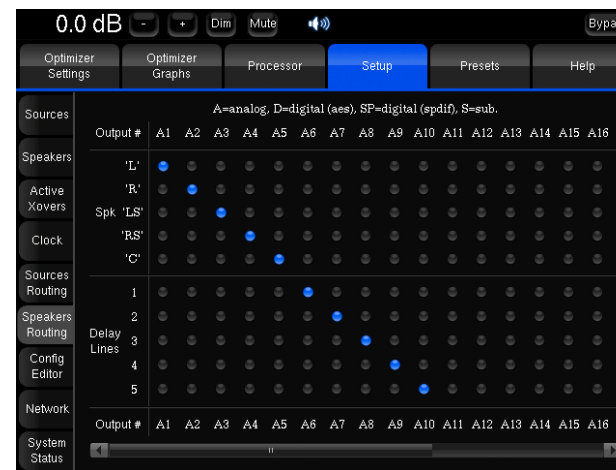
and routed to the physical outputs 6 to 10.

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 Time-frequency window:
For specific requirements, the "alpha/f" time-frequency window can be replaced by a custom time-frequency window specified in the XML file.
Two lists of values are used to specify this window:
- tab_ft: provides the list of key frequencies of the time-frequency window.
- tab_t: provides the list of window widths for each of the key frequencies specified above.

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.
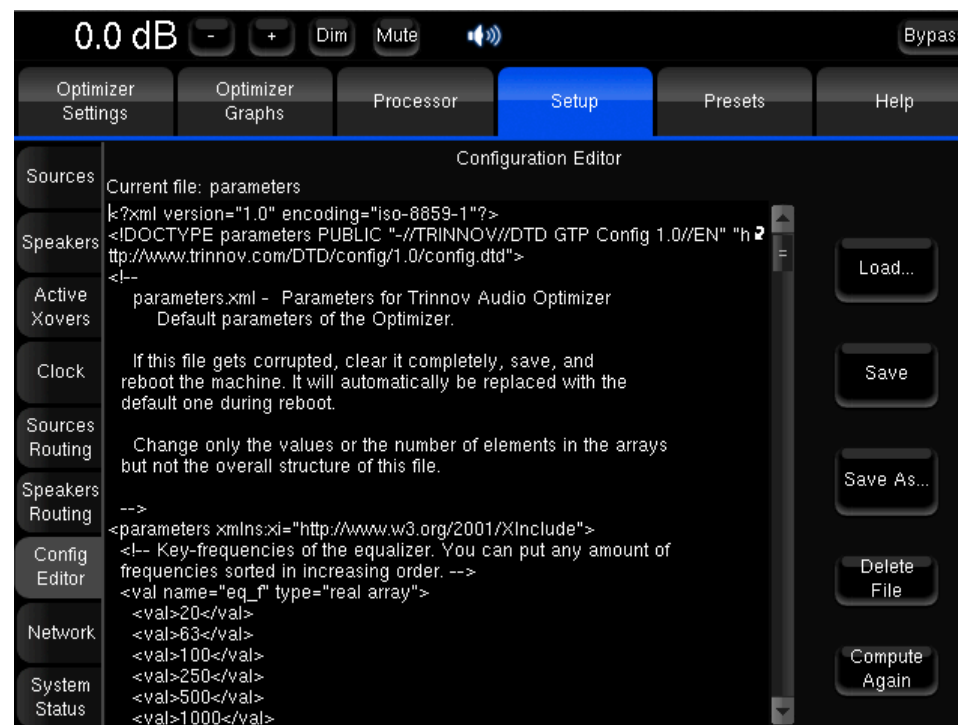
Config Editor

System Status

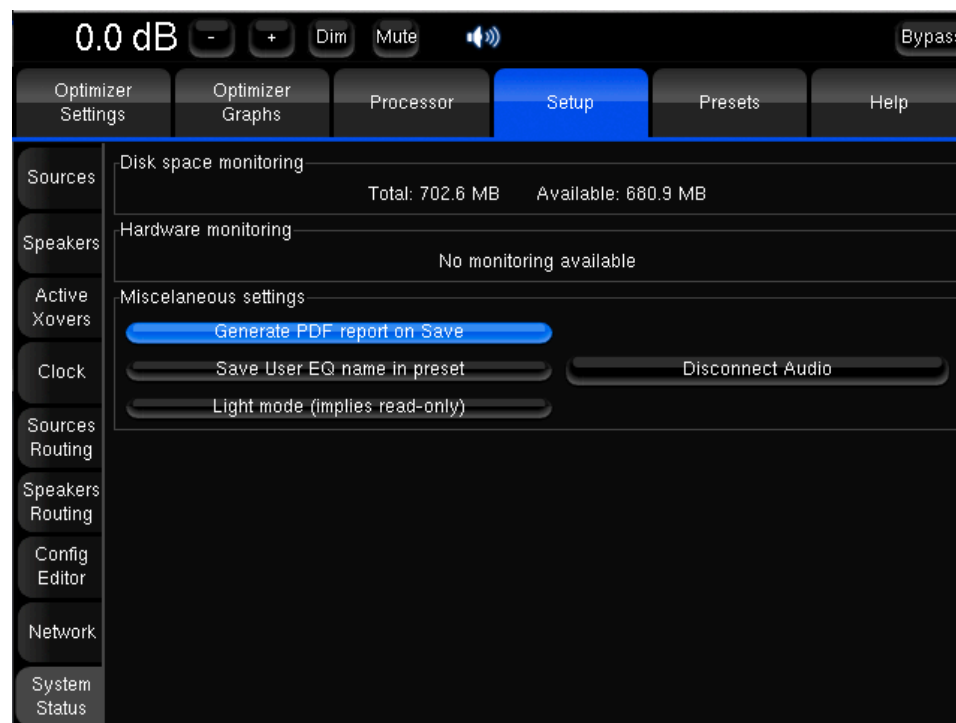**Disk space monitoring** allows to check if the flash memory is full or nearly full.

Hardware monitoring displays information related to the cooling of the system. It should be monitored when no air conditioning is available in the studio, under warm ambient conditions.

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.
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.

System Status