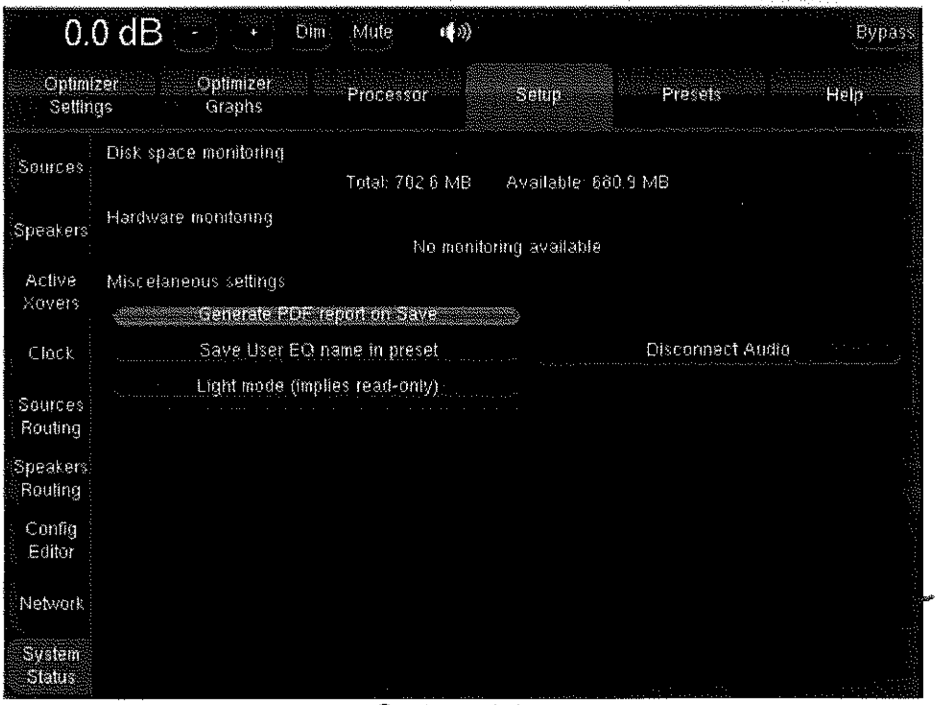
Custom remapping matrix:
For specific requirements, a custom remapping matrix can be specified, for example to route one input channel to several output channels. For more information on this feature, please contact your Trinnov distributor.

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

4.1.8 System Status



System status

Network setup
This concerns Ethernet and Internet operations: control over LAN with VNC, FTP file transfer and software updates.

Network Status
Indicates the connection status of the processor:

- No network detected: no cable or no DHCP
- Local network OK: connected to the local network
- Internet OK: ping from trinnov audio ok
- Connected to Trinnov Audio Server: SSH connection allowed and established

For Backup/Restore of presets or remote control of the unit, the Network Status "Local network OK" is sufficient. Trinnov software updates and remote support require Internet access, indicated in the Network Status display "Connected to Trinnov Audio Server." If this does not work, please read the Troubleshooting chapter about Network Connections.

Remote support and/or updates though Internet is not automatic and needs an operator in France. Please contact your Trinnov Audio distributor to schedule updates.

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.

4.2 Processor

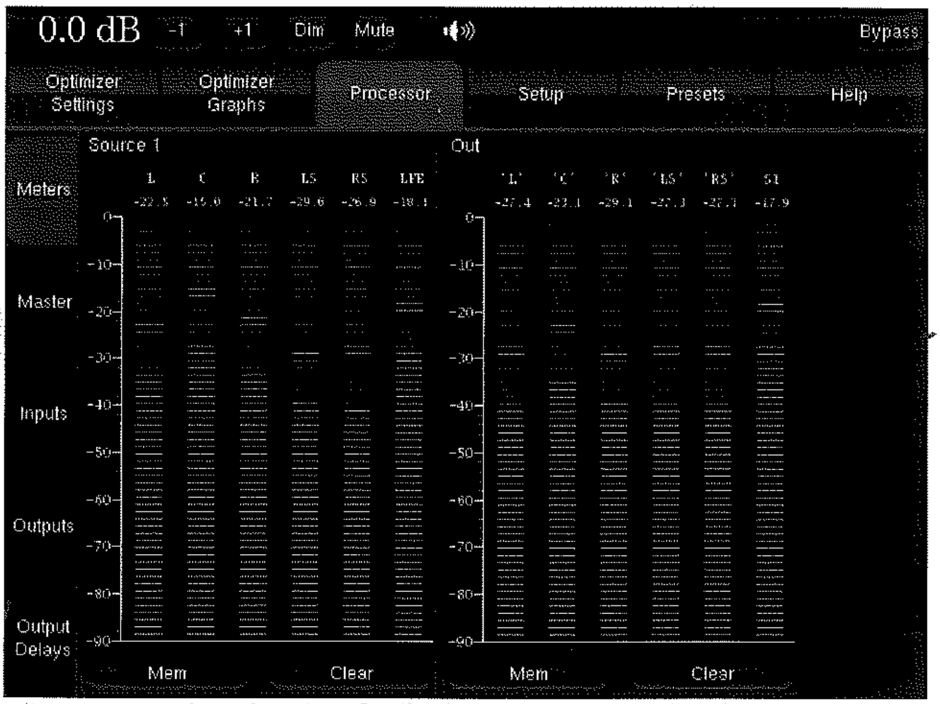
As a complement to the Optimizer automatic equalization or as a stand-alone system, the Magnitude32 includes a FIR Equalizer and Graphic Equalizers, levels and delays adjustments on every channel.

4.2.1 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.

Please note: on the input meters the LFE channel is always displayed as LFE as the last channel to the right. On the outputs meters the subwoofers are always displayed as S1, S2 etc...



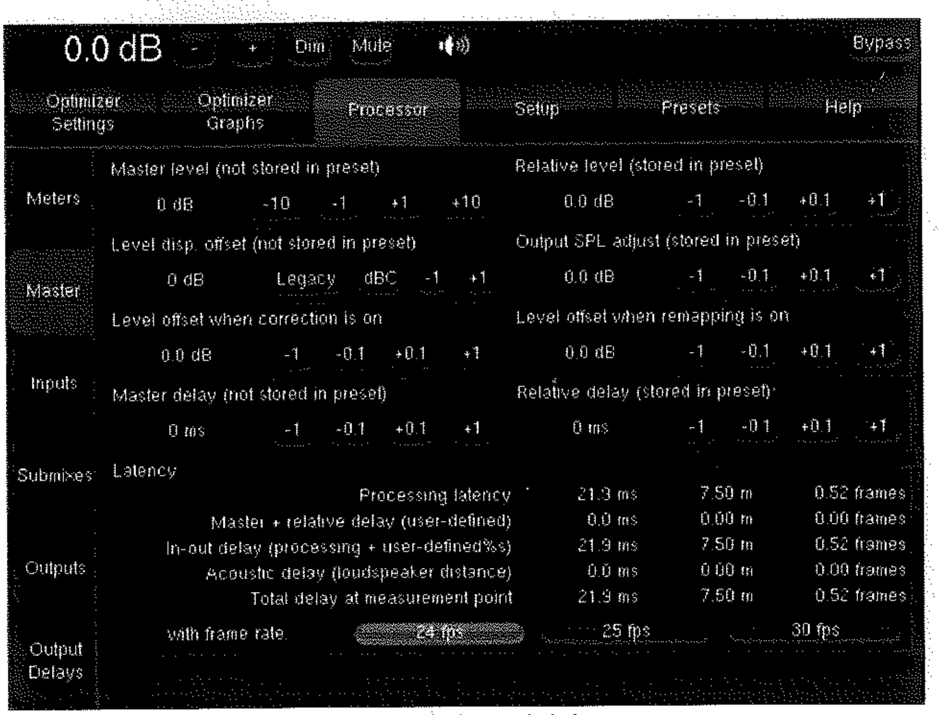
Level meters for the input and output signals

Important note: a digital signal cannot technically exceed 0 dBFS. Therefore the red tag on top of each input channel only let you know when the maximum level of 0dBFS is reached and does not necessarily indicate distortion or clipping.

4.2.2 Levels and Delays adjustments

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

4.2.2.1 Master Levels and Delays



Master levels and delays

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

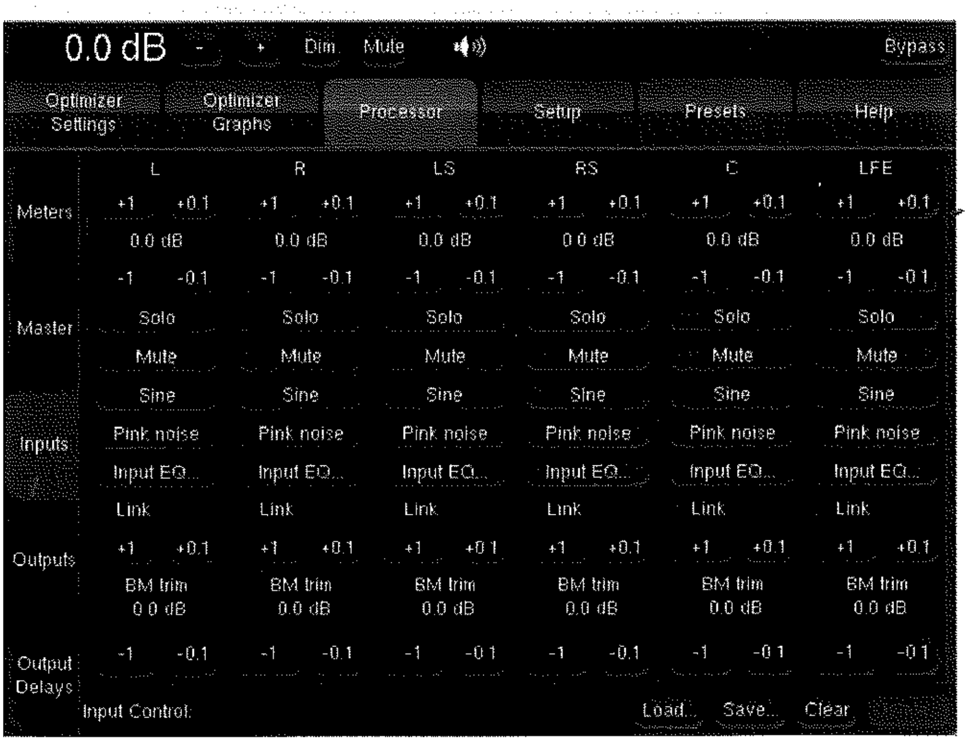
- **Master Level** is the reference level used by the processor for all presets. It affects both displayed and effective Level but it is not stored with presets.
- **Relative Level** may be used to match different presets subjective levels since it can be saved. Therefore you could for instance carry 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 (102,6 dB in our example). To do so, the system must have been calibrated and the dBC button selected. The Legacy option displays the level according to 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 activated, level adjustment is disabled.
- **Output SPL adjust** affects both displayed and effective Level. Used along with Level display offset, it allows you for instance to set a reference level corresponding to a measured 85 dB SPL and to store it into a preset. It is a typical cinema use.
- **Level offset when correction is On:** this level offset will be applied when the Acoustic Correction is activated.
- **Level offset when remapping is On:** this level offset will be applied when 2D or 3D Remapping is activated.
- **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.

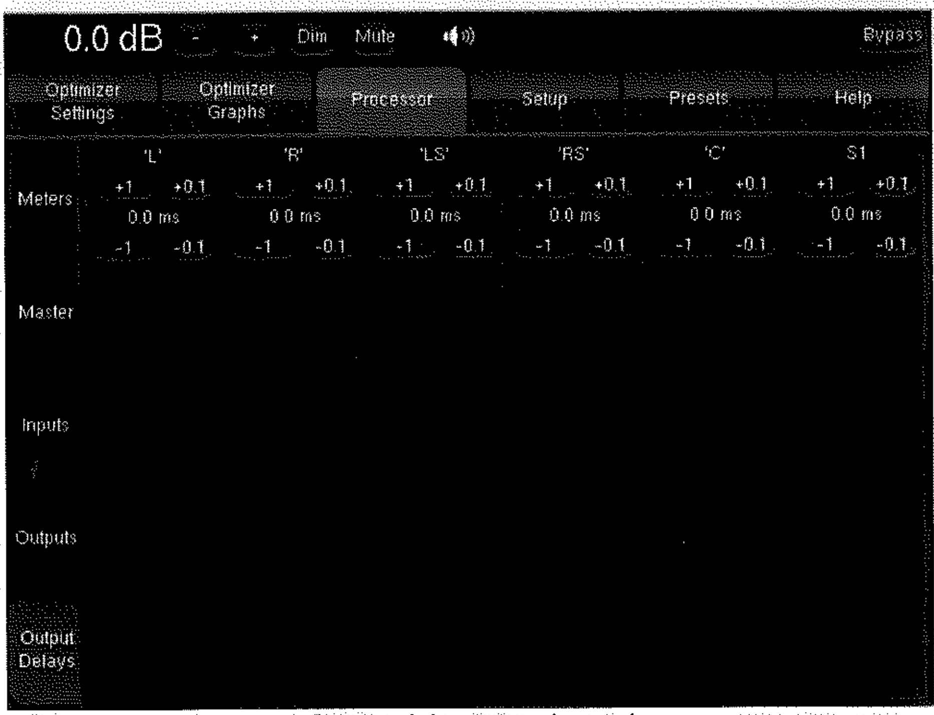
4.2.2.2 Channel-specific Levels and Delays

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



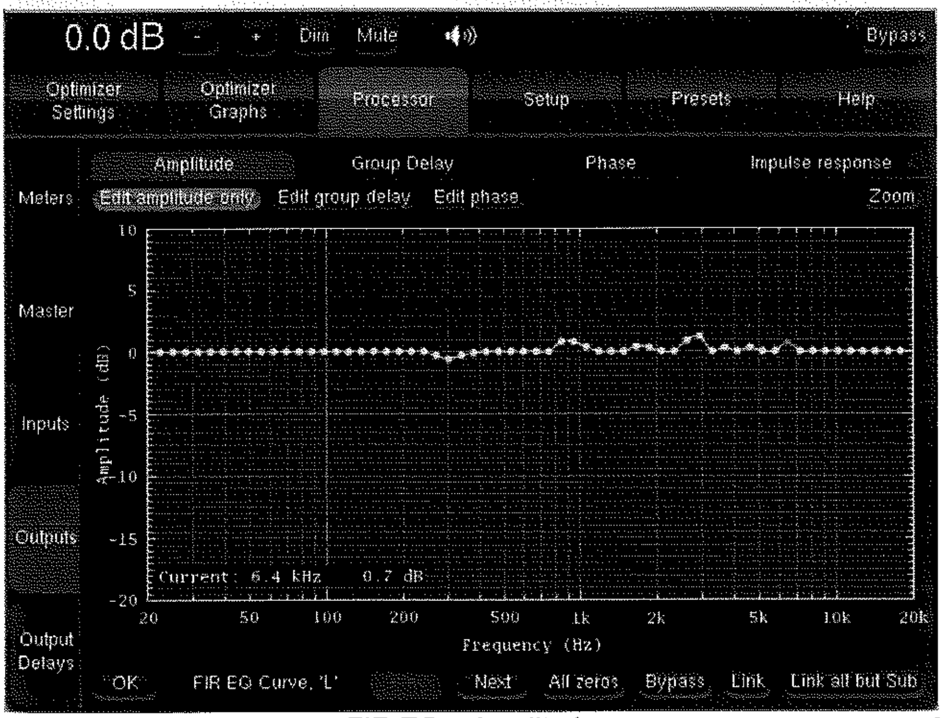
Output delays per channel

4.2.3 FIR EQ

For every speaker in the room, a FIR EQ can be defined in terms of:

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

One or more channels can be linked in order to apply the same FIR EQ to several speakers.



FIR EQ – Amplitude

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 Close button to apply the changes and get back to the Processor/Outputs tab.

Please note: by default, since the length of the FIR filter is 20ms (as defined in the Advanced Settings), the FIR Equalizer has a constant resolution of 50Hz. This implies that it has very low resolution at low frequencies; it should therefore not be used to work on specific frequencies in the low range. Instead, it should be used to change the tonal balance as a whole.

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

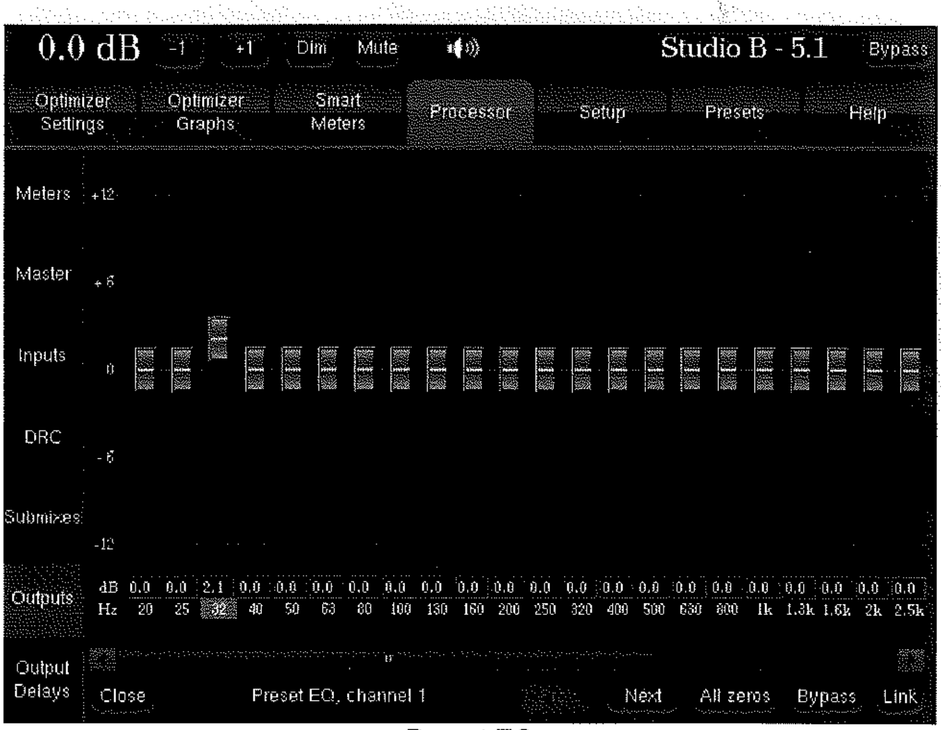
4.2.4 31 band Graphic Eqs

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.

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.

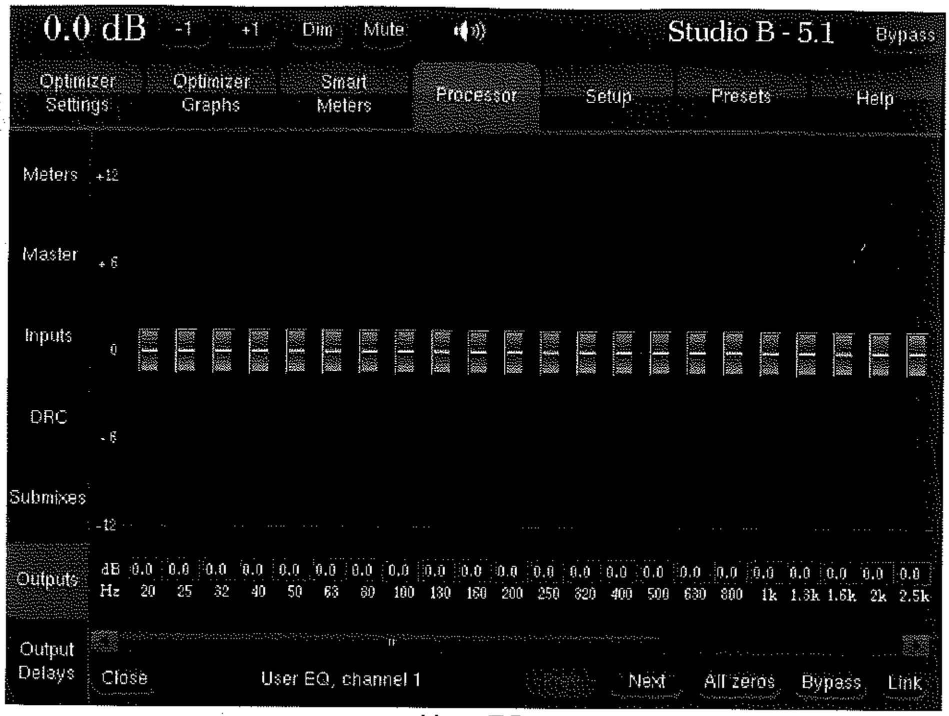


Preset EQ

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.



User EQ

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.

Notes:
You can use Pink Noise for setting up your equalization, as follows:

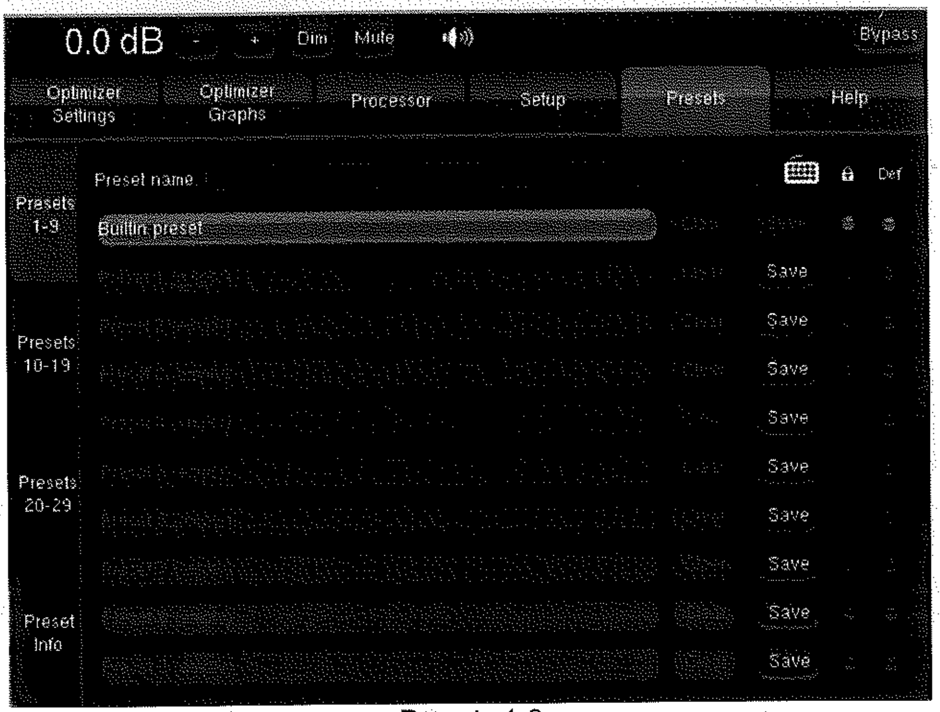
- In the Processor/Inputs page, press on the Pink Noise button for the channel you want to equalize,
- Switch to the Processor/Ouputs page and open the Preset EQ or User EQ for that *same* channel,
- Make your EQ adjustments,
- Use the Previous/Next buttons: the *pink noise will follow* to the same channel.

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.

4.3 Presets

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

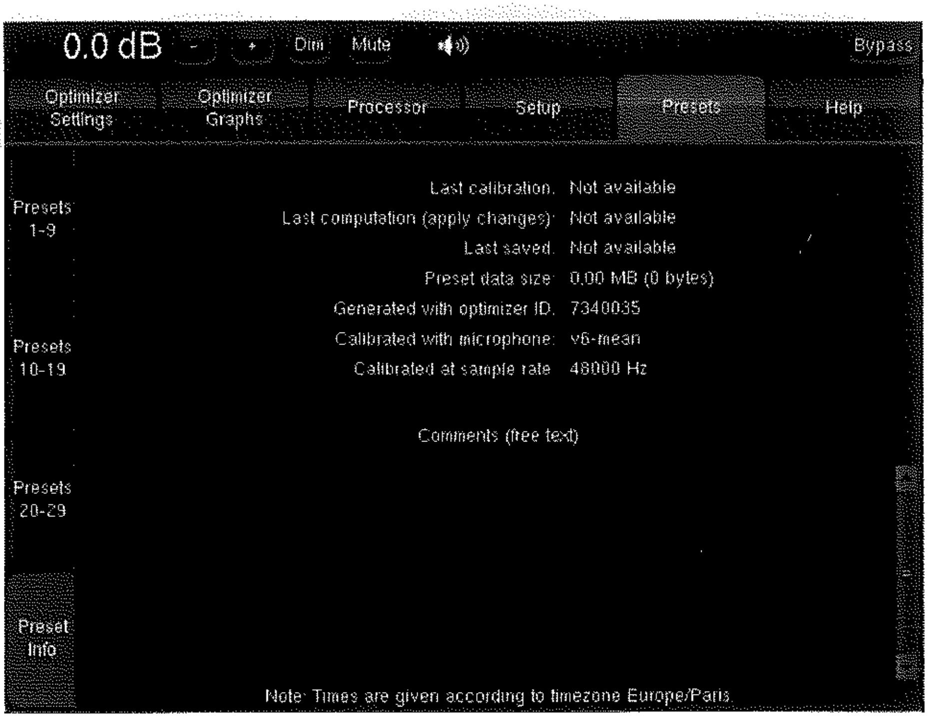
- Each preset can be locked (small lock icon) or deleted (Clear button).
- Once saved, a preset can be loaded with its button or with the corresponding number on the IR remote control (up to preset n°9).
- One preset can be chosen as the default preset to be automatically loaded at each start-up if its Def button is checked. The "no default config" of the Startup Screen can disable this automatic load if desired.

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 Optimizer.

4.3.2 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 where the preset was calibrated.
- **Calibration microphone**: the ID of the microphone used for the calibration
- **Calibrated at sample rate**: the sample rate used for the calibration.
- **Notes** about the preset provided by the user. This can be used to store a version history of the preset.



4.3.3 Backup/Restore Presets

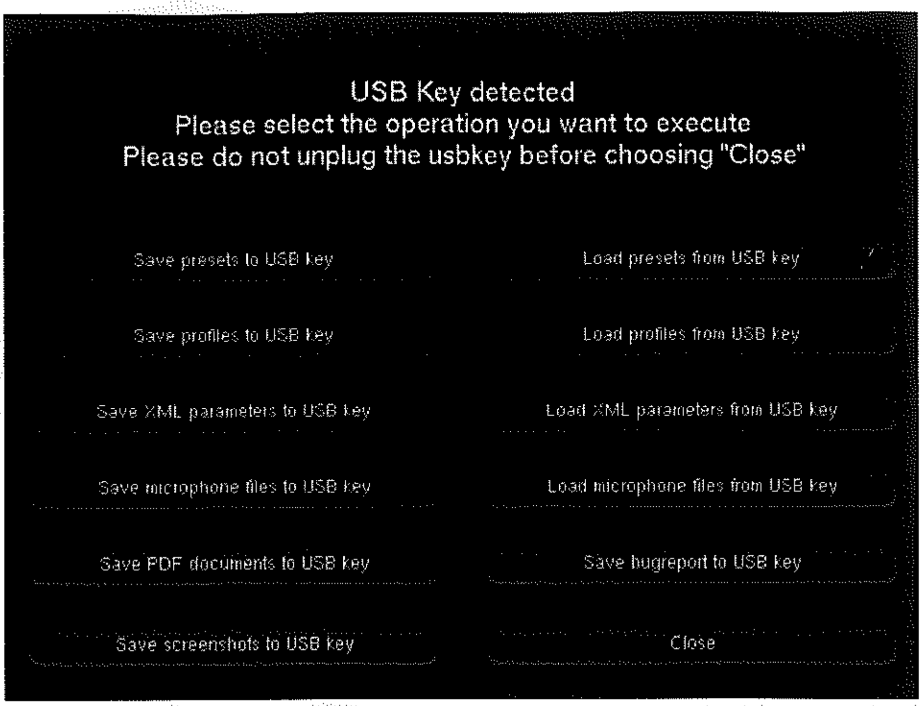
4.3.3.1 Screenshots

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

4.3.3.2 Backup / Restore with a USB Key

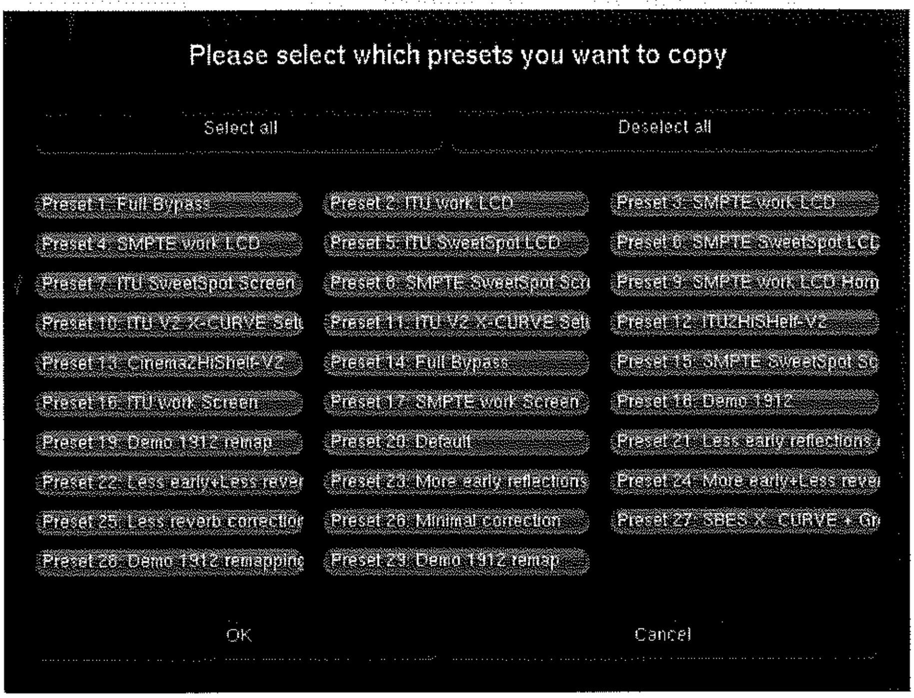
The purpose of this feature is to make a global or partial backup of the system and restore it. While the Trinnov unit is running, plug a USB memory stick in one of the USB ports of the Processor. A menu will appear. It allows one to restore settings from the USB key into the Processor, or backup them 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**



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. It is of course possible to remove files on the memory stick in order to restore only some specific settings.

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.

4.3.3.3 Backup / Restore through the network (via FTP)

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

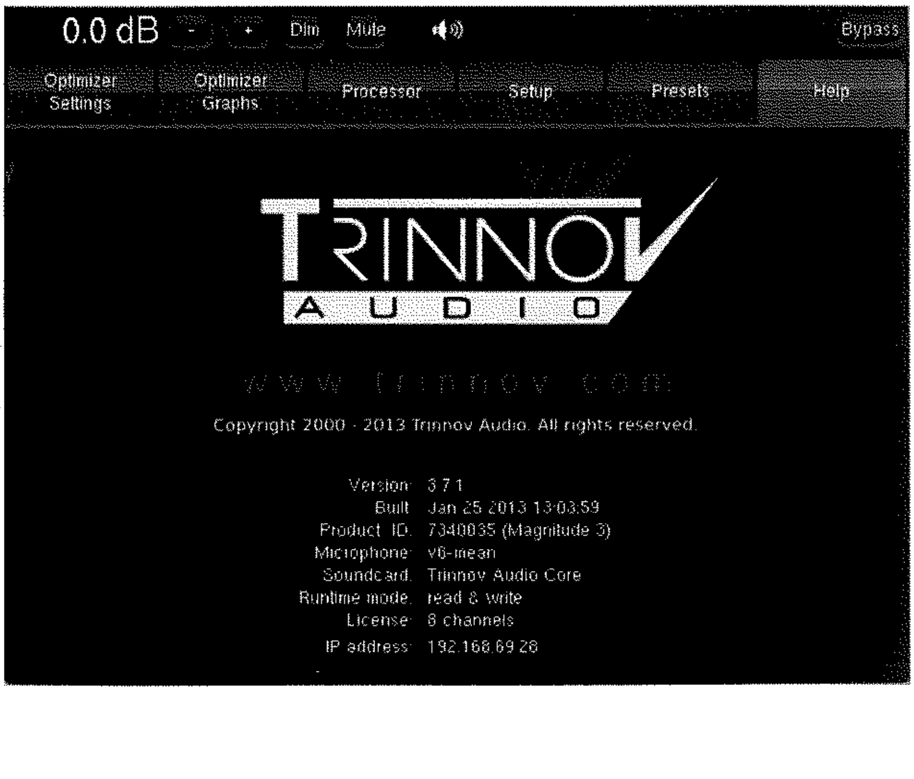
Example: ftp://192.198.0.5
The built-in FTP server will ask you to log in:
Login = srp
Password = *- the 7 digit **product ID** shown on the help page*
- or the 6 digit serial number labeled on the rear panel

4.4 Help

4.4.1 About

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:
  - o "Read & Write" is the normal runtime mode,
  - o "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),

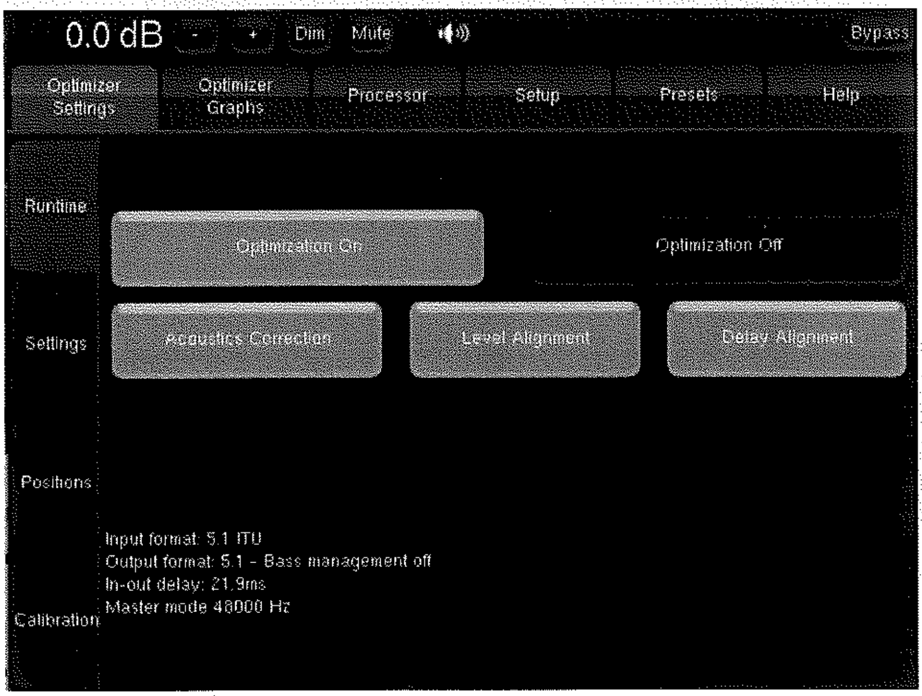


5 Optimizer Guide

5.1 Optimizer Settings

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

5.1.1 Runtime



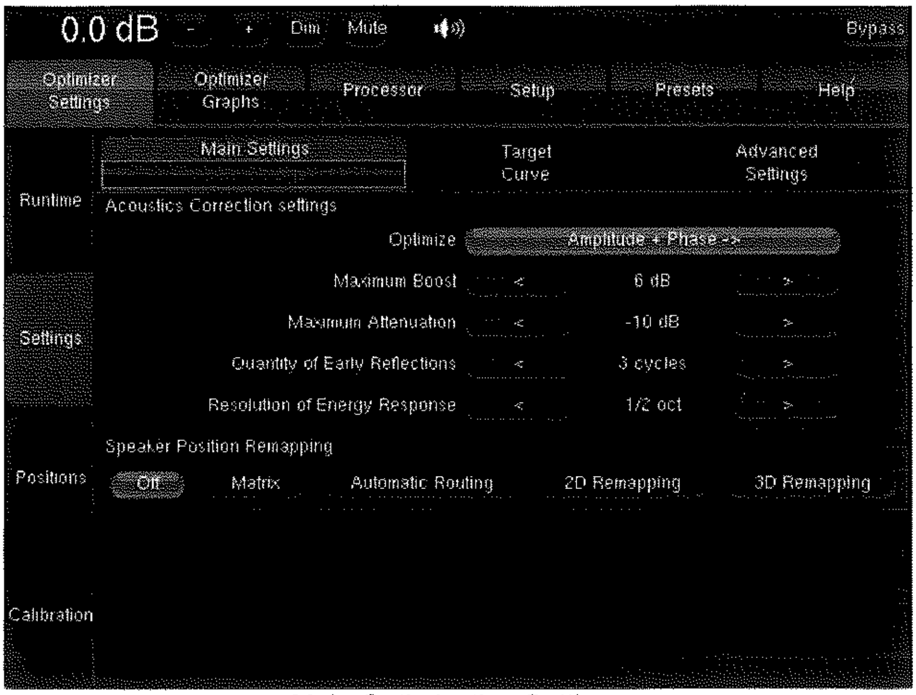
Runtime Settings

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

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

5.1.2 Settings

5.1.2.1 Main settings



Main Optimization Settings

- **Optimize**:

Amplitude + Phase (default): with this setting, the Optimizer will improve both the loudspeaker amplitude and the phase response of the loudspeaker. This greatly reduces the group delay of the speakers starting from about 200Hz;
Amplitude only: this mode tells the Optimizer to work only on the amplitude of the loudspeaker's response. The phase behavior is not modified;
Low range only: with this setting the automatic equalization will only use IIR filters up to frequency defined in the advanced settings. The automatic FIR filter is disabled, but the FIR EQ can still be applied;
According to L&R speakers: this is a special mode that will optimize the center and surround speakers in order to achieve the same response as the Left and Right speakers. It is mostly useful in home cinema installations.

- **Maximum boost**:
  It defines, in dB, the maximum amount of boost that will be performed by the algorithms. This parameter is used to avoid distortion. It's default value is 6dB.
  This parameter has an important impact on the behaviour of the automatic equalization, and is applied to both the time-based and the energetic approach.

- **Maximum attenuation**:
  It defines, in dB, the maximum amount of attenuation that will be performed by the algorithms. It's default value is -10dB.
  This parameter also has an important impact on the behaviour of the automatic equalization, and is applied to both the time-based and the energetic approach.

About the parameters "Quantity of Early Reflections" & "Resolution of Energy Response":
The Optimizer uses two different approaches for the Optimization of Loudspeaker/Room Acoustics:

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

- **Quantity of Early Reflections** (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 10kHz.

- **Resolution of Energy Response** (default is ½ octave):
  It defines, in number of octaves, how the room's energy response is smoothed, and modifies the behaviour of the equalization performed by the Optimizer on the *Late Reverberation*. It's default value is 0.5 (half an octave). Note that this is smoother than the 1/3 octave smoothing more commonly used.
  The behaviour of the energy optimization algorithm varies according to the *smoothing* applied to the room's response. If the response is less smoothed, sharp peaks in the response will be more taken into account for the correction, while with more smoothing only the overall tonal balance of the room will be corrected.

- **Speaker position remapping:**

  - **Off (default)**: remapping is disabled.

  - **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:** T–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:

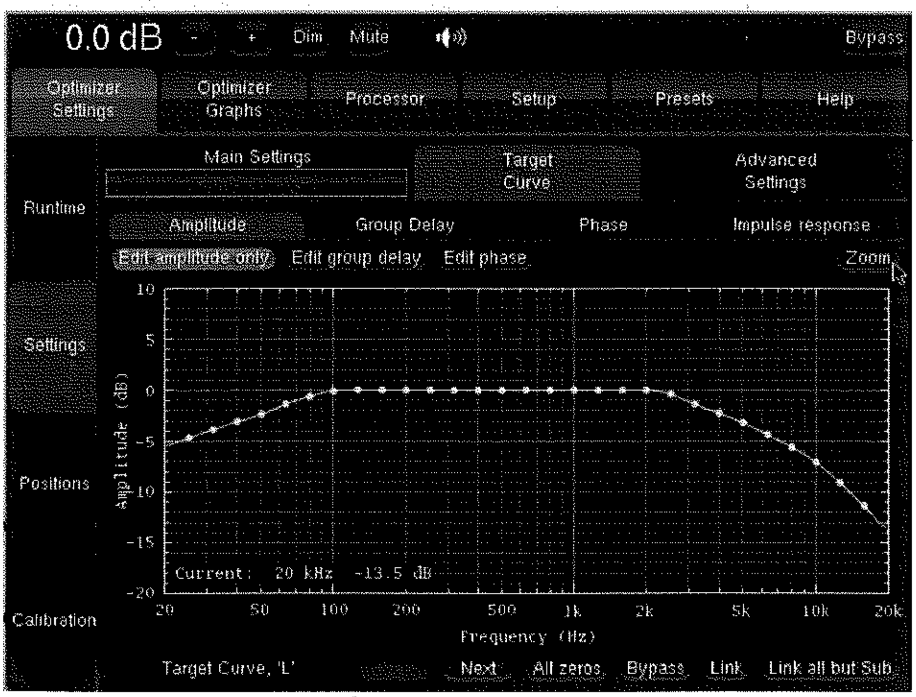
- compensating for incorrect loudspeaker position in azimuth and elevation.
- rendering a signal format (stereo, 5.1, 7.1...) on any number of loudspeakers. This includes up-mixes and down-mixes.

5.1.2.2 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.



Target Curve – X-curve example

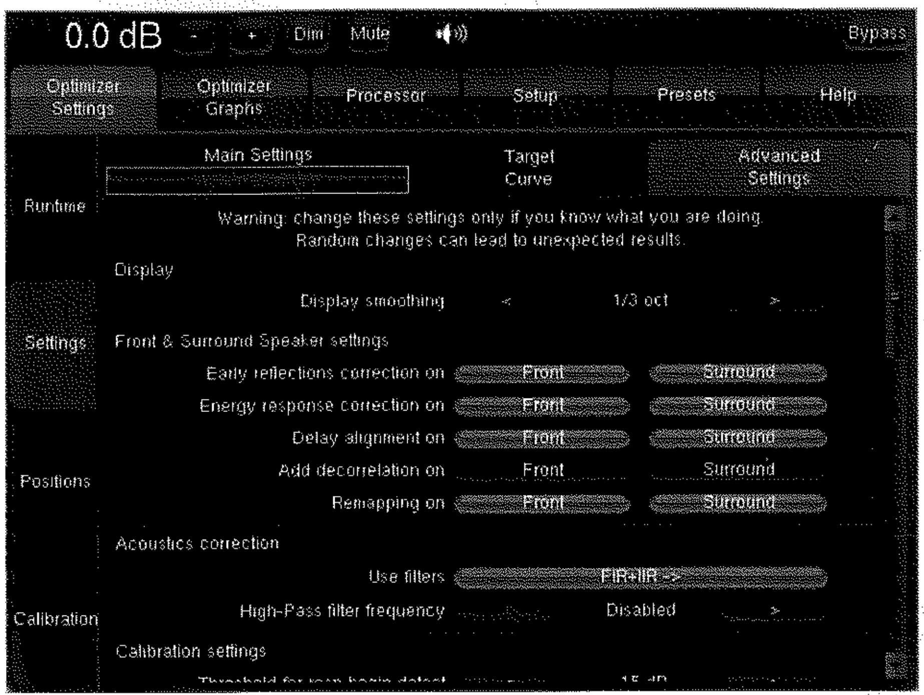
The curve can easily be edited with the arrows of the keyboard:

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

Please note:

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

5.1.2.3 Advanced settings



Advanced Settings

- **Display smoothing:** defines the smoothing value used for displaying the frequency response curves, **in the pdf document** generated while saving, in amplitude and phase. Its default value is 0.33 (1/3 octave). A smaller value, such as 0.1, can be used to display more details in the frequency response.
  Please note that this parameter has no effects on the curves displayed in the Optimizer Graphs page.

- **Front & Surround Speaker settings:**

The following features can be separately configured on for the front speakers and on the surround speakers.
Please note that Front speakers are defined as those whose azimuth is below 90 degrees. Conversely, Surround speakers are defined as those whose azimuth is above 90 degrees.

  - 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: in the case of sound editing and sound feature mixing rooms for cinema stages, the Optimizer can apply a decorrelation algorithm to the surround channels in order to simulate the diffuse field created by a belt of surround speakers;
  - remapping: in certain monitoring situations, such as in dubbing theatres, it may be preferable not to apply remapping to the front speakers.

- **Acoustics Correction:**
  - Use Filters:
    - **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:**
   - Processing on L&R speakers (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.

- **FIR and IIR settings:**
   - FIR filter length (default is 20ms): defines the length or number of taps of the FIR filter. The default setting of 20ms corresponds to 1024 taps at 48kHz and 2048 taps at 96kHz.
   - Number of IIR filters (default is 15): the number of IIR filters that will be used on every channel.
   - IIR filters minimal/maximal frequency (default is 20Hz/300Hz): IIR filters will be positioned from the min frequency up to the max frequency.

- **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.

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

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

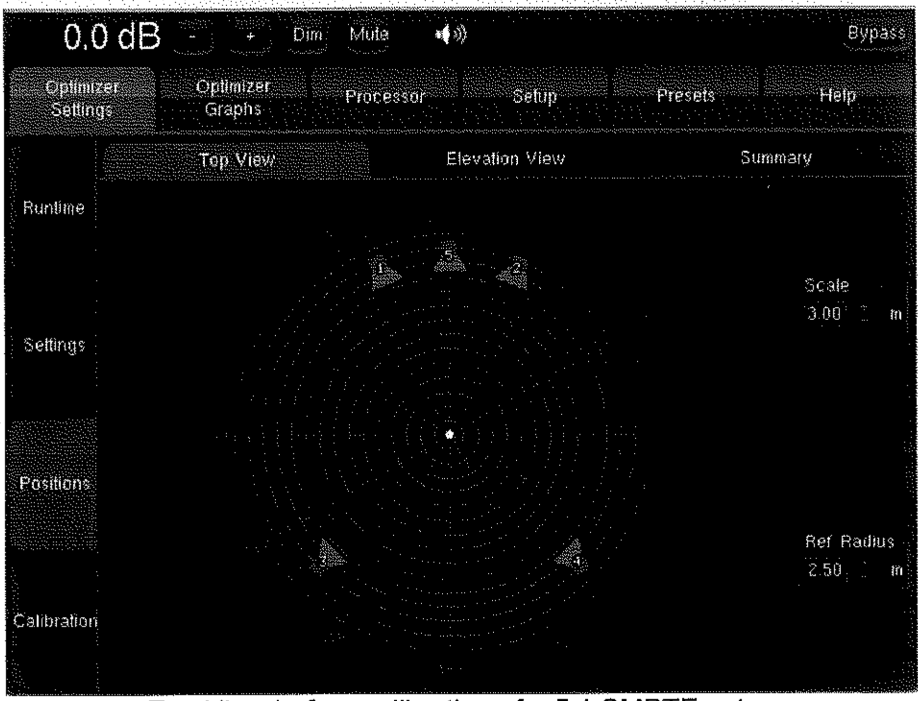
- **Advanced FIR settings:**
  It is strongly recommended not to change these settings unless you have been requested to do so by your Trinnov Distributor.

5.1.3 Positions

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

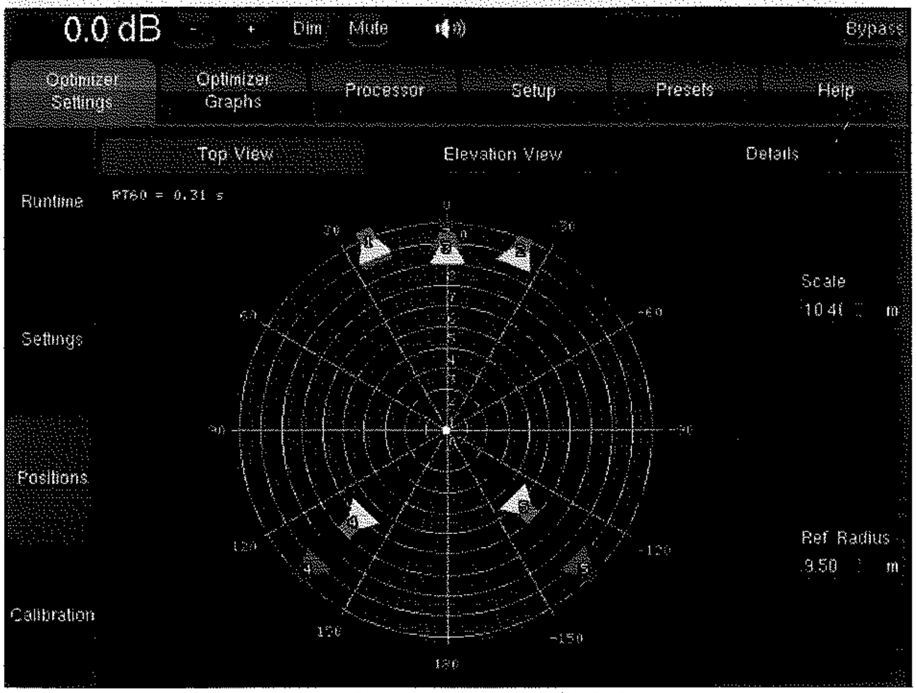
- The **Top view:** loudspeakers are seen from above;
- The **Elevation view:** Loudspeakers are seen from the listening point. Relative degrees of elevation and azimuth are displayed;
- **Details view:** a table listing the measured distance, elevation, azimuth, level, delay etc...

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

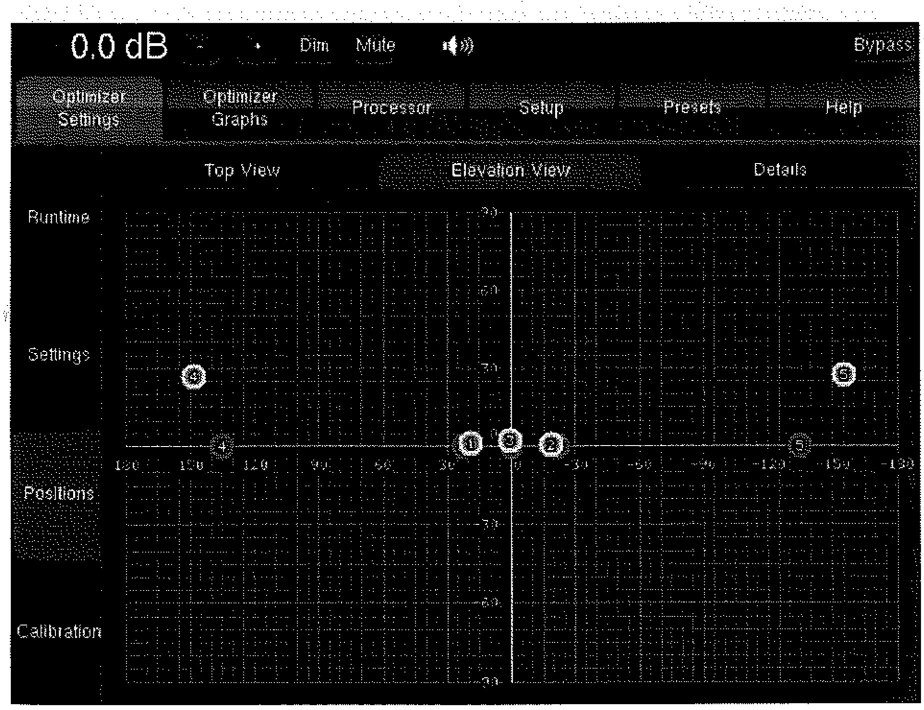


Top View before calibration of a 5.1 SMPTE setup

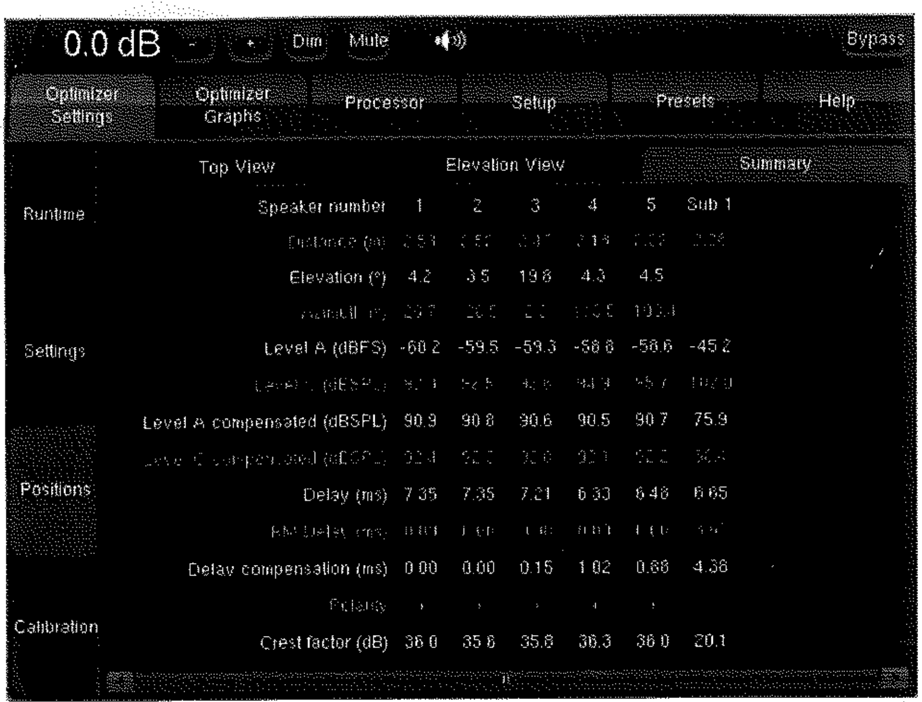
Once a calibration has been successfully performed, the Optimizer page will also display the actual loudspeaker positions of your system. The color of the loudspeakers depends on the Remapping settings.



Top View after calibration of a 5.1 setup



Elevation View



Loudspeaker Details view

Measurement details:

- **R** is the distance of the speaker (in meters) to the measurement point;
- **Elev** is the elevation of the speaker (in degrees) to the measurement point;
- **Azim** 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 Comp** 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.

5.1.4 Calibration

Once the system has been defined in the Setup page, the calibration page is used to perform a measurement of the full impulse response of every speaker in the room. For a description of the calibration process, please refer to the quickstart guide.

Important Note: Master 44,1 kHz and 48 kHz clock modes should be used for calibration. A 96 kHz measurement does not provide an impulse response sampled at 96 kHz but an impulse response sampled at 48 kHz that contains information within an extended frequency range (up to 96 kHz).

5.1.4.1 Overview

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

In the "Calibration" tab, you will find the list of all the measurements of the preset (one per line). A measurement contains the responses recorded during the test sound sequence by one microphone or several microphones placed at different points in the room. You can calibrate the selected measure by pressing the "**Calibrate**" button at the top of the page. During Calibration, the name of every speaker/channel is displayed while playing the MLS sequence. Once every measure has been calibrated, you can compute the acoustic correction filter that will best fit every point, accordingly to their weight.

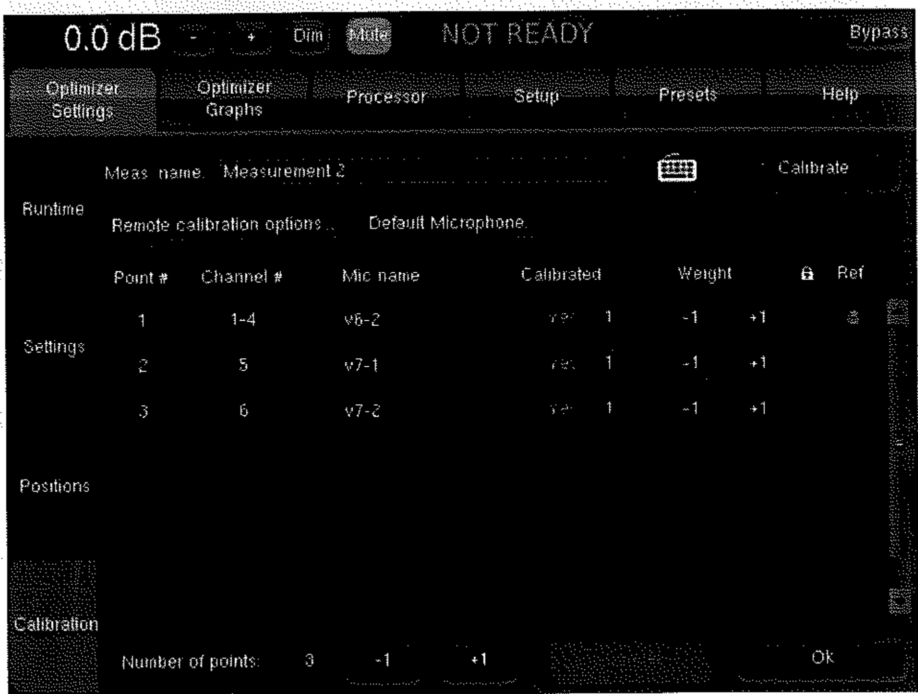
Measures list view

5.1.4.2 List of measurement points

When you press the **configure** button, you open the list of points that compose the measurement. You can add or remove points by pressing the "+1" or "-1" button at the bottom of the page.

You have to select for each point the **microphone(s)** you are using, by opening the list of available microphones. The system will find information about this microphone and will tell you on the 2nd column which input you have to connect it to.

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



Points list view

The next column, called "**Calibrated**", indicates the calibration status of each point. If the label "No!" remains after you ran a calibration, consider verifying your plugging and routing.

The "**Weight**" column allows you to set a weight to each point, from 0 to 100, to emphasis for example some central points. You can bypass some points by setting their weight to 0, what can be useful in case one of the points have failed the calibration but you don't want to perform a calibration again. The weight can be adjusted after a calibration; you don't have to run a complete calibration to take weight changes into account, just to press once the "Compute" button. If using your processor with an external keyboard, you can adjust the weight with the up (+1) and down (-1) arrows, or the page up (+10) and page down (-10) keys.

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

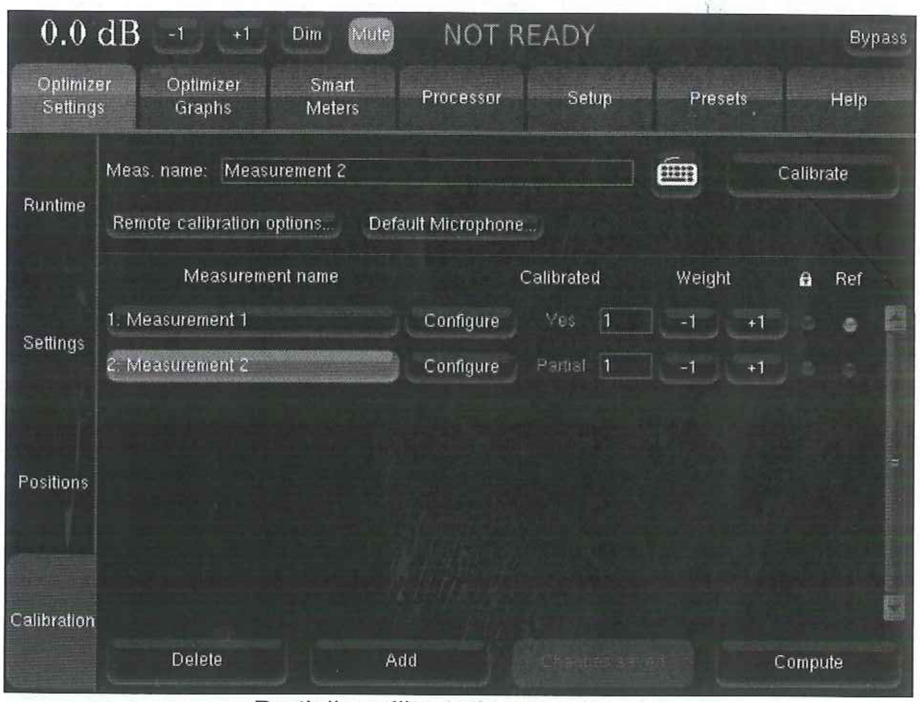
The **reference point** is unique for all points of all measurements. That's the point the delay and level correction are computed from.

You can exit this window by pressing "Ok" to keep your changes or "Cancel" to avoid them. You then come back to the measurements list.

5.1.4.3 Measurements list

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**". When you add a new measurement, the points configuration is copied from the selected measurement, assuming you may want to run a new measurement of the same system with the same equipment (of course you can change these parameters). The removal of a measurement cannot be cancelled. You can also rename a measurement by editing the text box "Meas. name".

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



Partially calibrated measurement

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

The **lock** disables measurement settings edition.

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

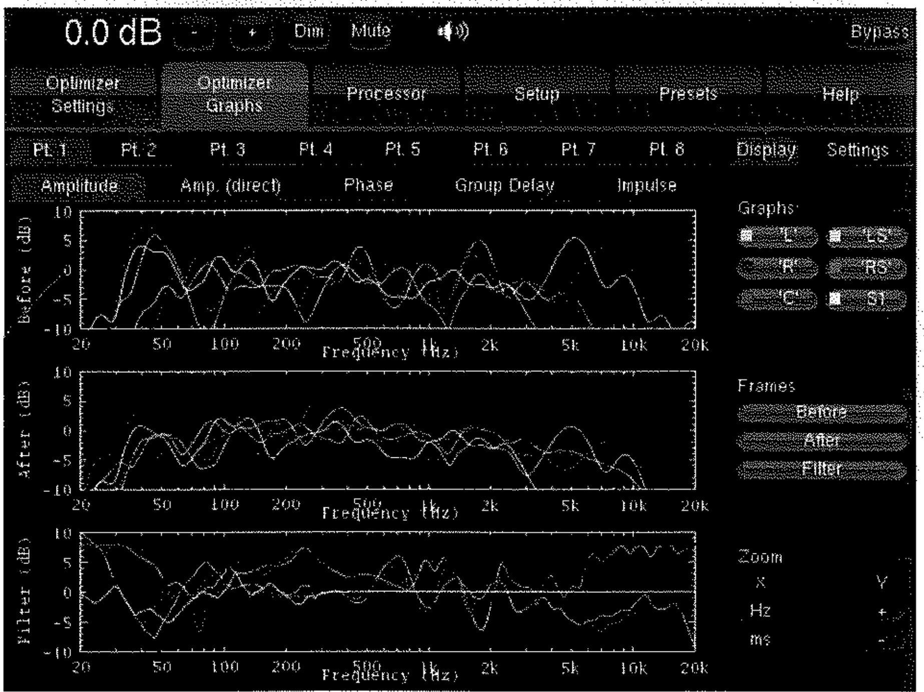
All the measurements with non-null weight should be fully calibrated before to compute the acoustic correction filters by pressing the "**Compute**" button. If some are partially or not calibrated at all, you will have to remove the bad points or their entire measurements, for example by setting their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

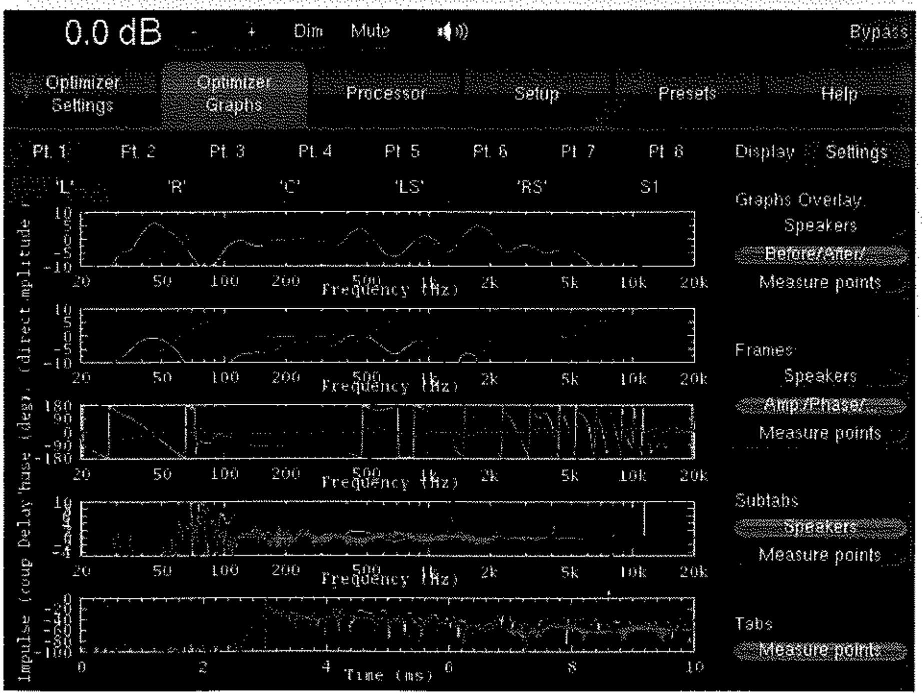
Before saving your preset in the "Presets" page, you have to save the multipoint parameters by pressing the "**Save changes**" button. These parameters are automatically saved (and the button disabled) when you leave the point list, and when you run a calibration or a computation.

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

5.2 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.



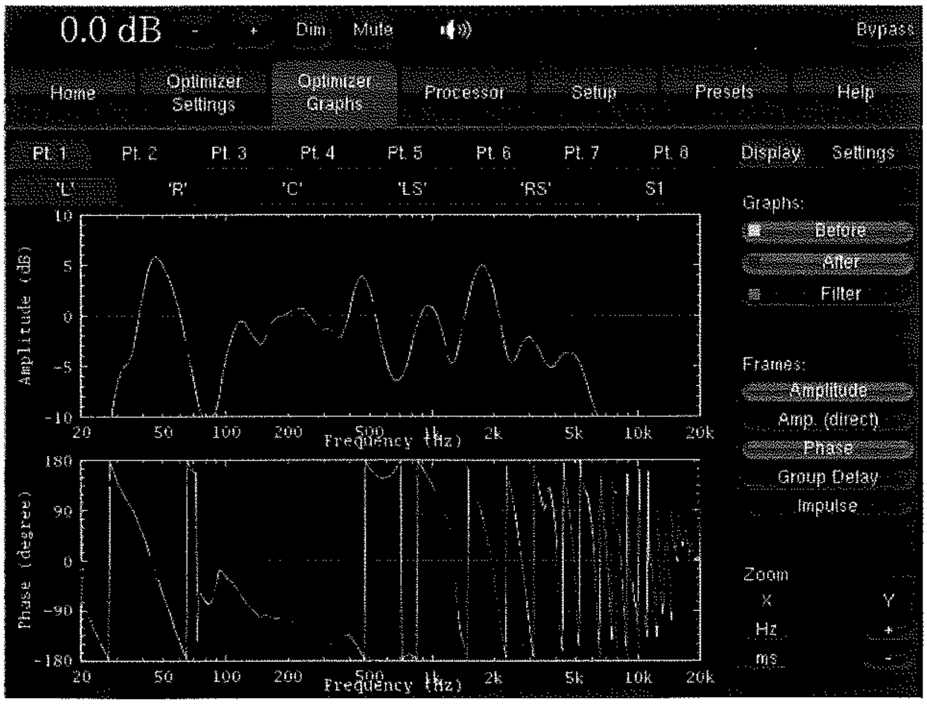


The flexibility of this feature allows you to visualize on a same screen various type of data, which you can organize as you wish by a few parameters:

- 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.



Zoom options allows modifying amplitude, frequency and time scales according to the type of representation.

Please note:

- the display settings will be saved in preset
- this display only takes under consideration the automatic filters provided by the Optimizer, not the manual additional adjustments.

6 Known Issues and Troubleshooting

6.1 Known Issues

6.1.1 Using the option "Send LFE to L+R"

Although the "Send LFE to L+R" option is designed for installations where no subwoofer is present, please note that in the current software version, the bass management option "Send LFE to L+R" can only be activated if **one** Subwoofer is defined in the *Subwoofer number* setting.

6.1.2 Calibration with wide bandwidth Subwoofers

In certain cases, with wide bandwidth subwoofers – going beyond 300Hz – the level calculation of the Optimizer may not work properly.

We strongly recommend to check the frequency response of the subwoofer After Compensation to make sure that it has been calibrated correctly.

6.1.3 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".

6.2 Troubleshooting

6.2.1 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.

6.2.2 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.

7 Useful Tips

7.1 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**.

7.2 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.

7.3 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.

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

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

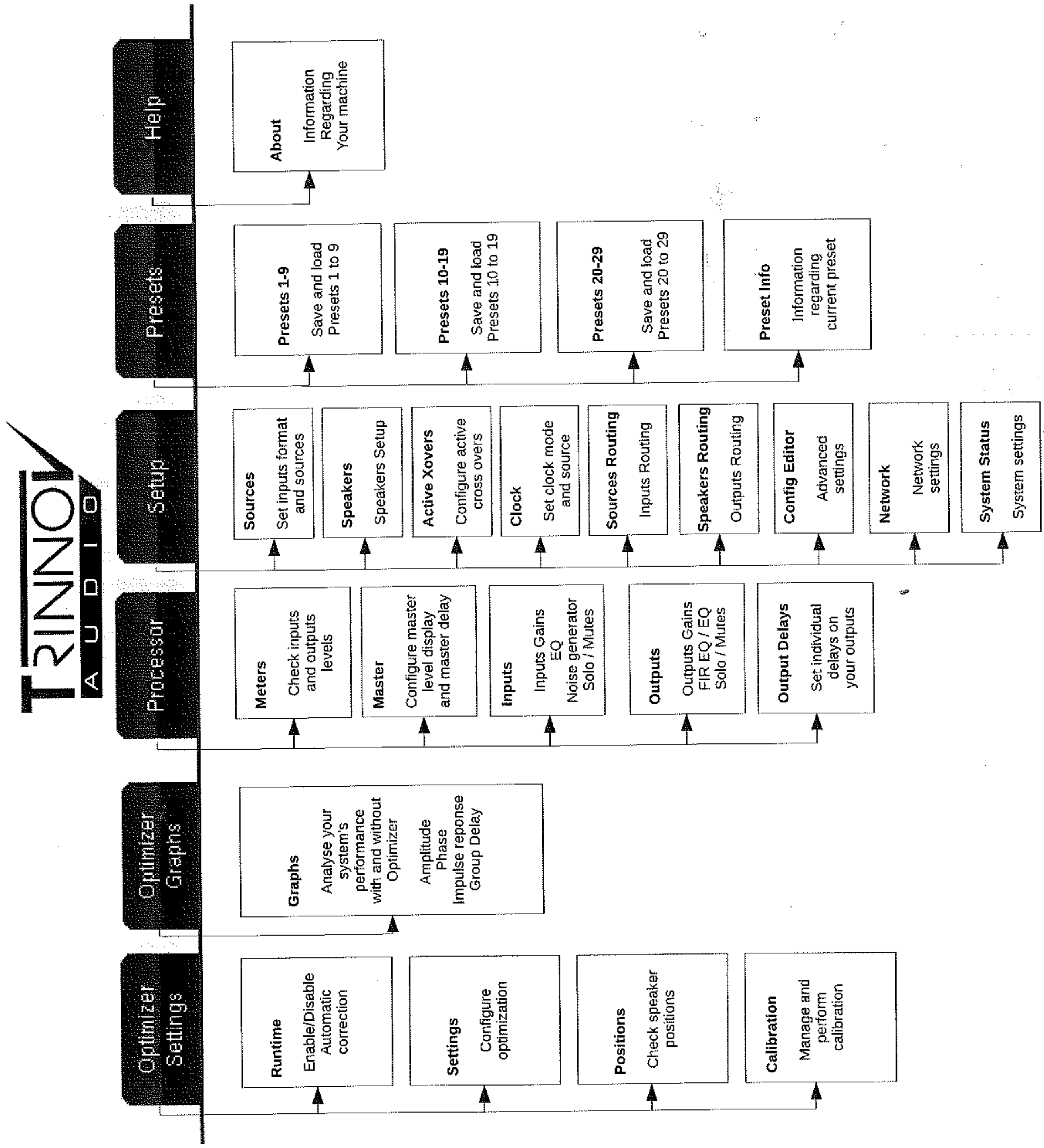
To reduce latency:

- You can change the sampling frequency and the buffer size of your unit according to the table above. This can be done in the **Setup/Clock** 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 hand, turning the Optimization Off in the **Optimizer Settings/Runtime** tab does not reduce latency since the audio still pass through the PC.

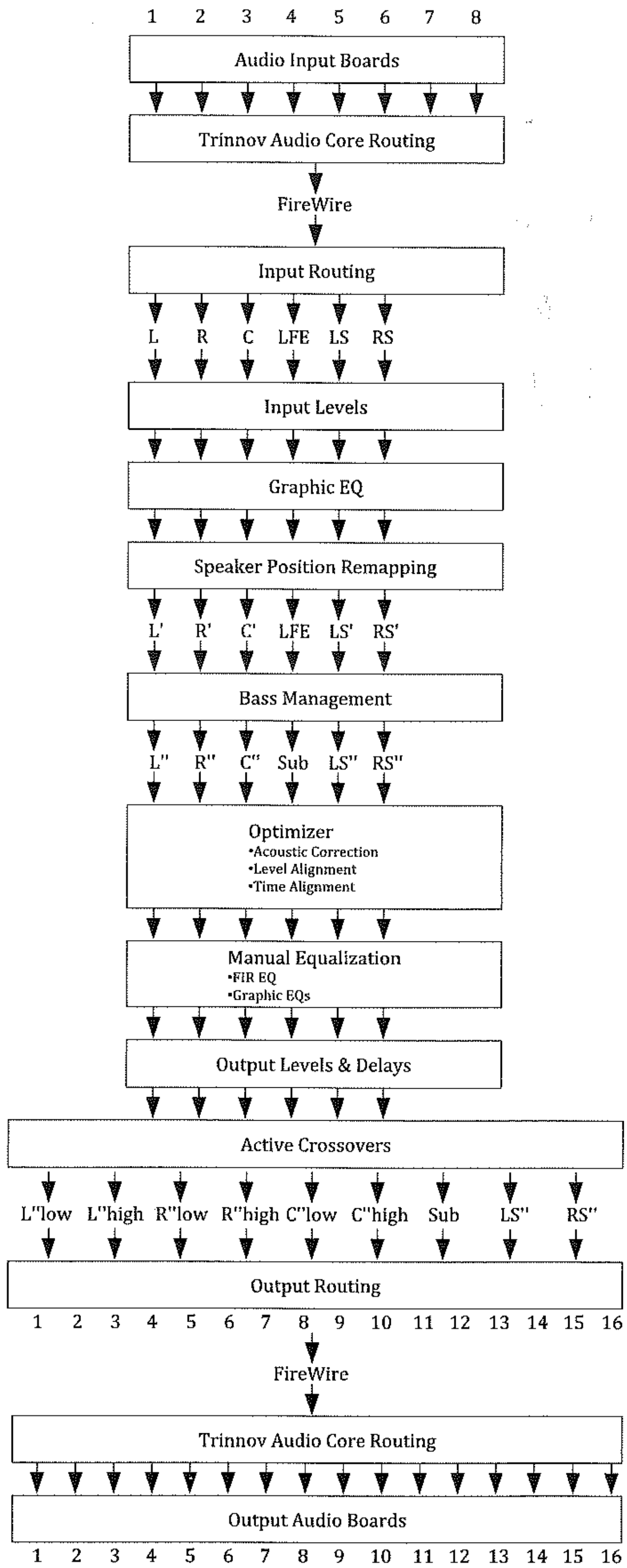
Please note: Latency can be monitored in details in the **Processor/Master** page.

8 Appendix

8.1 Arborescence of the menus



8.2 Signal Flow of the Magnitude32



8.3 Bypass

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

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