Trinnov Audio

High End Digital Acoustics

magnitude32

Reference Manual

v3.6 Software Release

www.trinnov.com

TRINNOV AUDIO

Integrated Monitoring Solutions

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Contents

1 Introduction

Thank you for choosing a Trinnov digital audio processor. As a company, our goal is to offer a range of products that meet the audio monitoring of the most demanding audio professionals and High-End enthousiasts.

This User Guide is organized in the following chapters:

- **Introduction**: describes the main features of the Optimizer.
- **Getting Started**: provides a step-by-step guide to set up your system.
- **Hardware Guide**: describes the audio interfaces used in the magnitude32, the 3D measurement microphone, the different remote options and update/support procedure;
- **Software Guide - System Software**: describes all the pages of the Processor module, the core software of all Trinnov products.
- **Software Guide - Optimizer**: describes all the pages of the loudspeaker/room optimization module.
- **Know Issues**: describes some limitations that have been identified.
- **Troubleshooting**: provides answers to the most common problems that may be encountered during installation or operation.
- **Useful tips:** provides useful tips regarding the system
- **Appendix**

1.1 Main Features of the Magnitude32

The Magnitude32 provides all the basic functionalities expected from a digital audio processor:

- routing and level settings of inputs and outputs (Audio Matrix functionality)
- word clock input and output
- remote control options via an Ethernet network
- presets save, backup and restore
- basic peak level metering of inputs and outputs
- calibrated, global gain/volume adjustments
- monitoring controller

1.1.1 Manual Equalization

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

Two different tools are offered:

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

1.1.2 Active Crossovers

The Processor features 2-way, 3-way and 4-way active crossovers. Depending on the chosen audio interface, 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.

1.2 Main Features of the Optimizer module

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

1.2.1 Level and Time Alignment

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

The Optimizer also calibrates the absolute level.

1.2.2 Automatic Optimization

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

- Improved Phase Response: the Optimizer corrects the frequency response of the speakers, both in amplitude and phase. This means that the Optimizer corrects both the tonal balance to obtain a neutral timbre for every speaker and it also works in the time domain to achieve a high resolution stereophonic image with well-focused phantom sources.
- Target Curves: the Optimizer automatically defines the filters that will achieve the required frequency response defined by a target curve. This is particularly useful in Film Studios in order to comply with the X-Curve SMPTE and ISO standards. Phase and Group Delay targets may also be defined, making the Optimizer a unique tool for sound system designers.

- Fine-tuning options: the Optimizer provides over 12 different parameters, such as maximum boost/cut, to customize the behaviour of the room correction algorithms. This opens many possibilities for fine-tuning the sound according to loudspeaker capabilities and listening tests.

1.2.3 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*.

2 Getting Started

2.1 Power on and Shutdown

2.1.1 Power on

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

Every Magnitude32 have power switches on both back panel and front panel.

The back panel power switch always needs to be pressed first to supply the apparatus with electricity.

The front power button shall then be used to start the apparatus. It should illuminate **after a few seconds.**

2.1.2 Shutdown

Magnitude32 requires pushing the power button only once to initiate normal shut down.

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

2.2 User Interface

Magnitude32 open architecture offers a variety of solutions to access the user interface:

- from an external monitor with a keyboard and a mouse,
- from a VNC Client running on remote device (laptop, smartphone or tablet) connected through the network to the Magnitude32

2.2.1 Using an external display, mouse and keyboard

Magnitude32 have VGA (and/or 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.

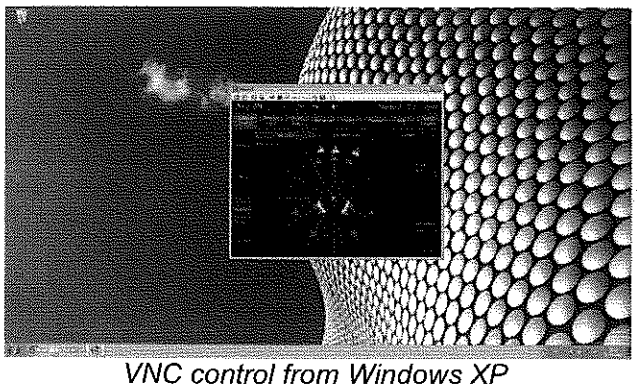
Using an external display is also the easiest way to find the IP address of the processor that is needed to use VNC (**Setup/System Status**).

2.2.2 Using a VNC Client through the local network

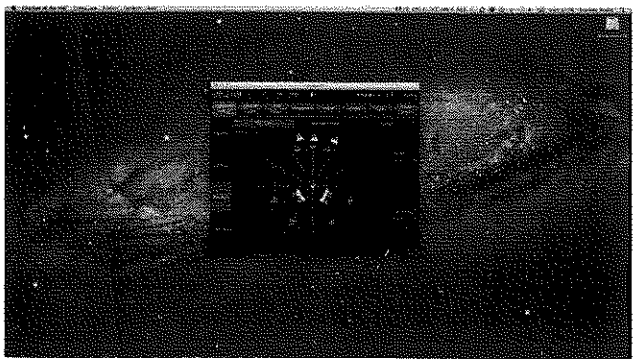
Magnitude32 have a built-in VNC Server that allow you to fully control the processor from any VNC client host device over the network.

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...)

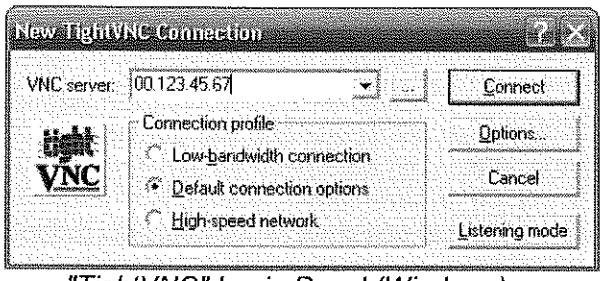


VNC control from Windows XP

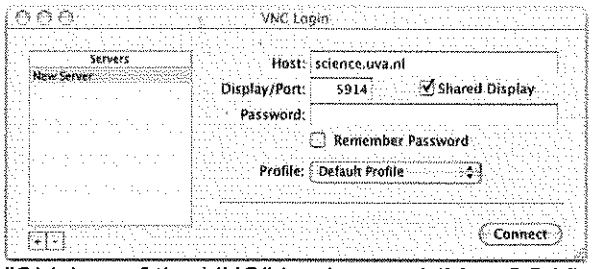


VNC control from Mac OS X

You can find different VNC Clients/Viewer on the Internet. As examples, we've chosen "TightVNC" for Windows 7 and "Chicken of the VNC" for Mac OS X.



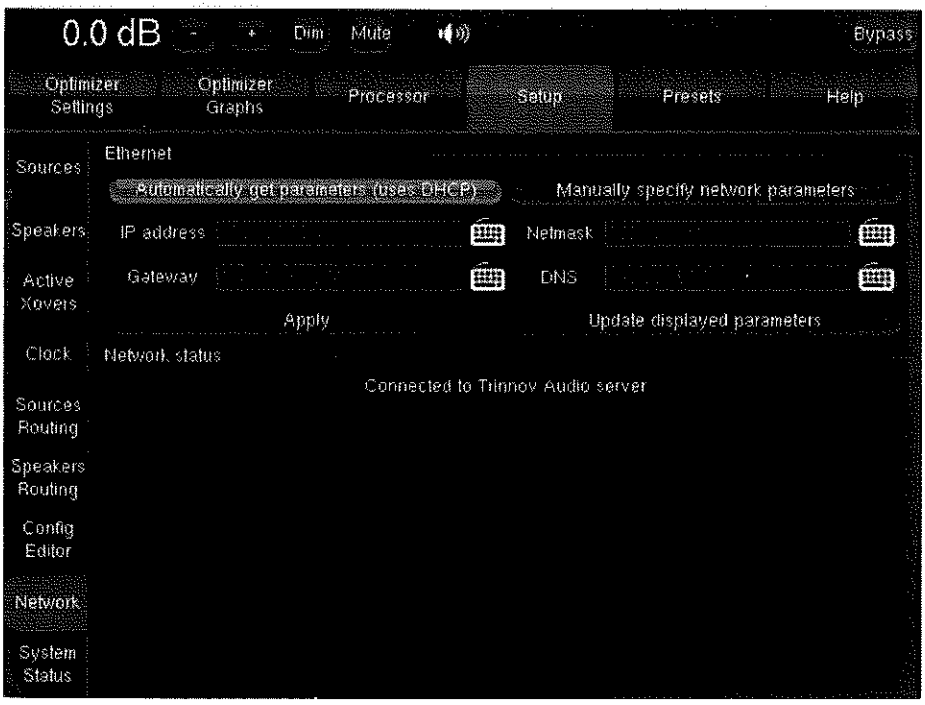
"TightVNC" Login Panel (Windows)



"Chicken of the VNC" Login panel (Mac OS X)

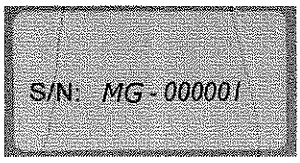
As you launch the VNC client, you will attempt to establish a connection to the server. The VNC Client will ask you the server address and a password:

- The server address is IP address of the Magnitude32. You will find it in the **Setup/Network** page:



Note (advanced user): You can also find the IP address in the DHCP Client list of your router/set-top box administration page

- a password which consists of:
  - the 6 digits **serial number** labeled on the back panel.



  - the 7 digits "**Product ID**" number displayed in the **Help/About** page.

2.3 Optimizer - Quickstart

2.3.1 Calibration settings

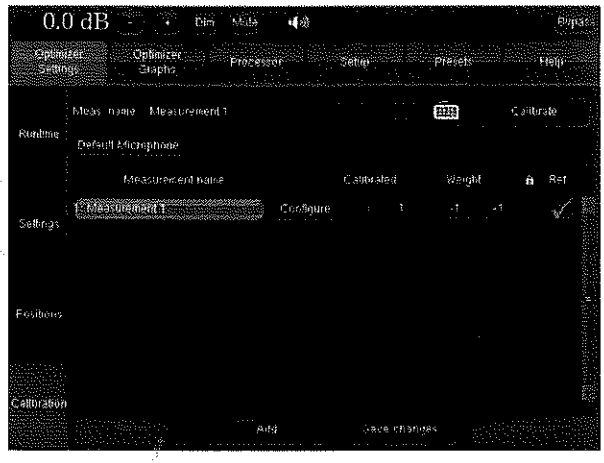
The following parameters must be configured correctly before calibration:

- Software source routing (**Setup/Sources Routing**) based on the connection of the microphone;
- Number of Speakers/Subwoofers of the reproduction system (**Setup/Speakers Settings**);
- Speaker routing according to the output connections (**Setup/Speakers Routing**);
- Clock mode set to Master 48 kHz (**Setup/Clock Settings**);

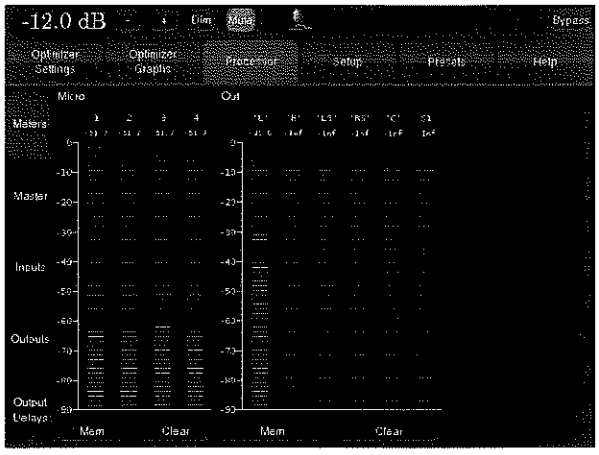
2.3.2 Calibration

Important Note: Active Crossovers need to be calibrated prior to the main calibration. Please refer to chapter 4.2.3.

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. Start with a Master Level of -40 dB and increase it until the test signal pass on the second speaker, meaning you reached that minimum SPL. If not too loud for your system you can even increase that level to get a better quality measurement.



Calibration Page



Meters during Calibration

1) Go to the **Optimizer Settings/Calibration** page.
2) Press the **Calibrate** button to launch the Calibration sequence.
3) Power **ON** the microphone when you are invited to do so, and press **OK**.
4) Listen to the calibration sequence (MLS) on the left loudspeaker. Calibration will initiate a minimum of three MLS bursts.
    a. If the test signal pass on to the right speaker:
        i. The Sound Pressure Level is equal or above the minimum required.
        ii. This indicates a good, robust calibration.
    b. If the first loudspeaker keeps repeating the calibration signal:
        i. 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, stop the calibration and refer to the **Sources Routing** chapter of the User Guide to verify that the microphone routing is correct. It can also be the PP3 battery inside the microphone that is out of energy, and/or the microphone or cables that are damaged.
        ii. If the 4 Input Meters are modulating, keep increasing the master Level.

iii. As soon as you reach the minimum required SPL, cancel calibration and start the procedure over.

5) Power **OFF** the microphone when you are invited to do so, and press **OK**.
6) In the **Optimizer Settings/Calibration** page, the calibrated column should indicate **Yes!**. Press **Compute** so that the Optimizer processes the data and calculate appropriate filters.
7) **Save** this new calibration/correction in a free/blank **Preset** (one of the **Save** button In a **Presets** page).

3 Hardware Guide

A wide range of hardware options makes the Optimizer easy to integrate in your studio's environment. The Hardware guide provides a description and technical specifications of each Processor.

3.1 Audio Interfaces & Chassis

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

- The **Trinnov Audio Core** (TAC) is the central audio component inside the processor. It provides routing of the audio between the physical inputs/outputs and the software, activation of the relays when instructed by the user allowing safe shut-down, hardware source selection and clocking up to 192 kHz when available.
- The **Trinnov ADA4** is connected to the TAC, executes AD/DA conversions and offers 4I/Os analog channels.
- The **Trinnov DA8** is connected to the TAC, executes D/A conversions and offers 8 analog outputs.

Magnitude32 are mounted in a Trinnov 2U 19" Chassis.

Height: 100 mm
Width: 445 mm
Depth: 410 mm
Weight: 8 kg

3.1.1 Audio interfaces

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

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

3.2 Magnitude32

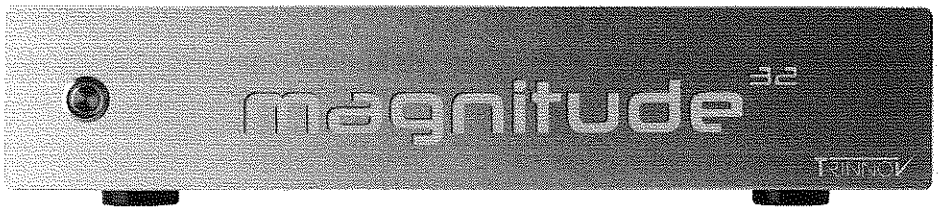
3.2.1 Technical Description

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

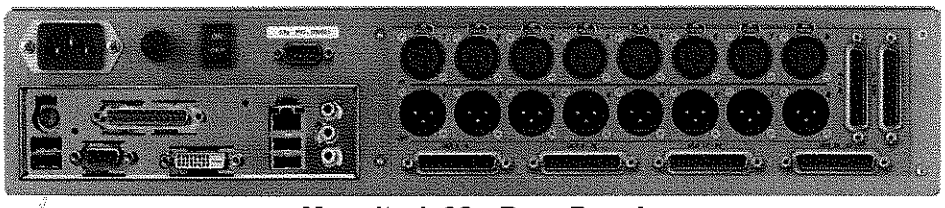
Magnitude32 processors are powered by a quad-core Intel i5 to process a maximum of 32 simultaneous audio channels at 96 kHz. Despite this amount of power, the unit remain silent thanks to custom heat sinks and additional slow fans. Magnitude32 also use flash disk technology removing the need for a mechanical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

3.2.2 Front/Rear panels



Magnitude32 - Front Panel

Magnitude32 front panel are made of a brushed aluminium plate with a power switch. Rack ears provided.



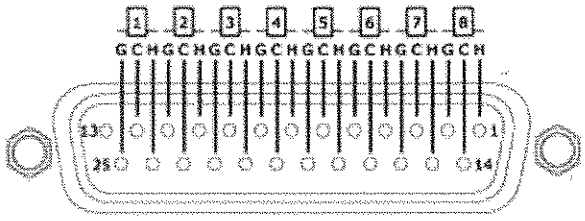
Magnitude32 - Rear Panel

The rear panel connectivity is the same for every magnitude32 processor.

- **PC section:**
  - 1 PS/2 port for mouse and keyboard
  - 1 VGA connector
  - 1 DVI connector
  - 4 USB ports
  - 1 Ethernet socket
  - 1 parallel port
  - 1 serial port
  - on-board audio (inactive)
- **Audio section:**
  - 8 XLR for the inputs 1 to 8.
  - 8 XLR for the outputs 1 to 8.
  - 2 DB25 for the inputs 1 to 16
  - 4 DB25 for the outputs 1 to 32

3.2.1 DB25 connectivity

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



Analog DB-25 Connectors

G: Ground C: Cold / - H: Hot / +

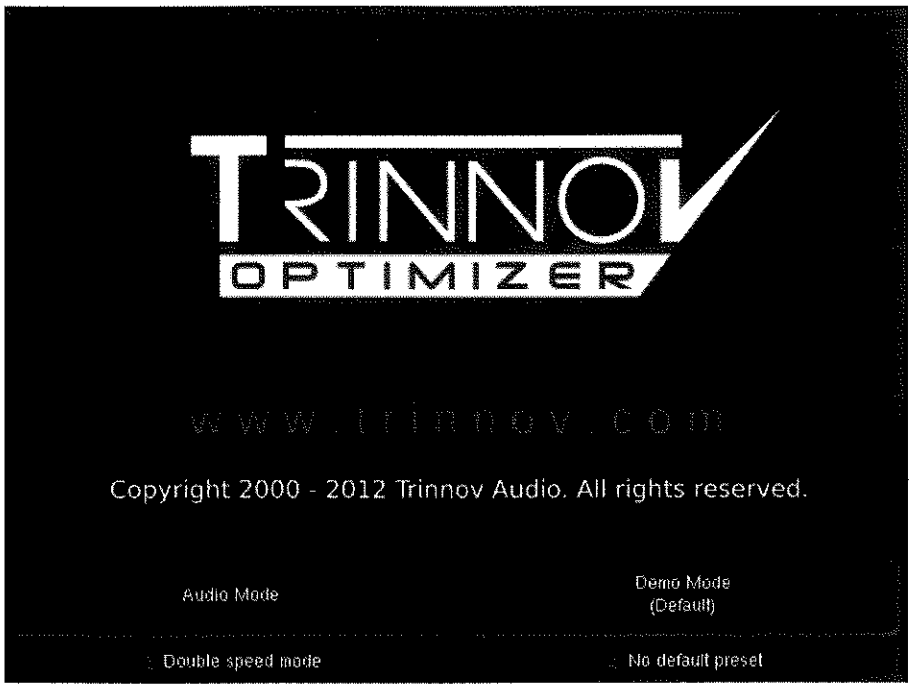
Please note:
- Another quite popular pinout is the one from *Yamaha*. You should check which pinout your equipment is using before connecting it to the magnitude32.

3.3 Startup Options

3.3.1 Startup Modes

Momentarily displayed before the access to the GUI

Some options can be activated in the following screen.



- **Double speed mode**: this option does not concerne 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.

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.

3.4 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

3.4.1 Position and orientation of the microphone

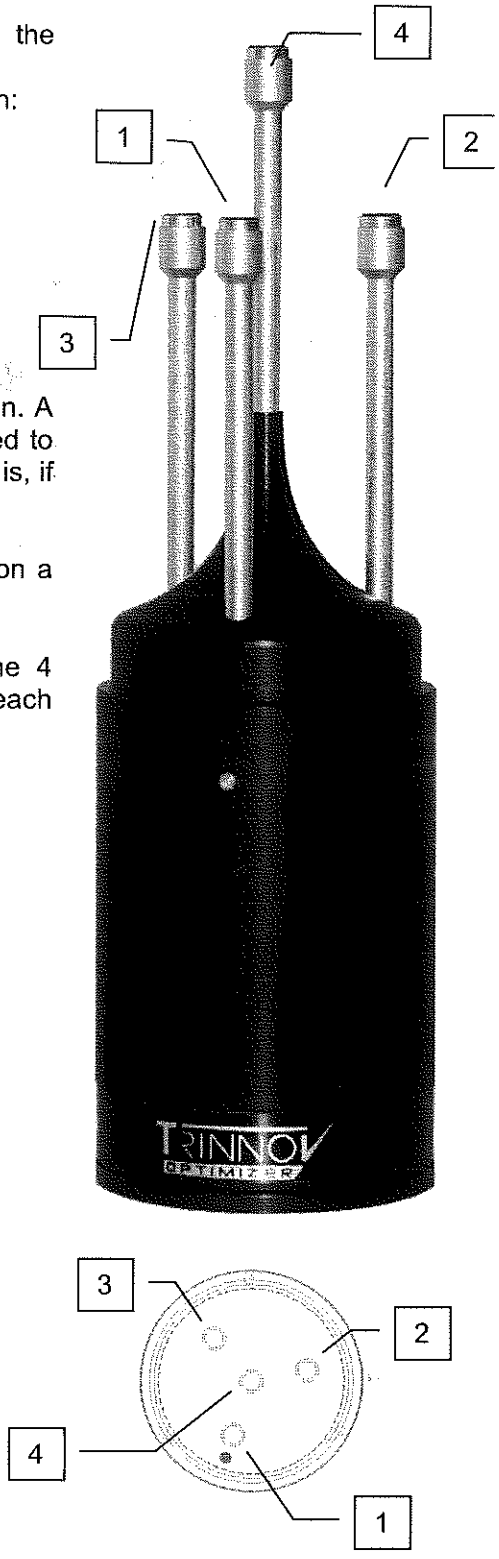
The Calibration microphone should be placed at the listening position. A red led shows the front of the microphone. This end must be pointed to the front of the sound stage, where the center channel should be (or is, if the loudspeaker setting is respecting the ITU recommendation).

During the calibration process the microphone should be screwed on a microphone or camera stand.

The numbers on the photograph indicate the routing order of the 4 capsules. The correct order can be checked by smoothly scratching each capsule.

3.4.2 Power supply

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



The calibration microphone and the arrangement of the capsules

3.5 Network Operation

3.5.1 Crestron module

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.

3.5.2 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.

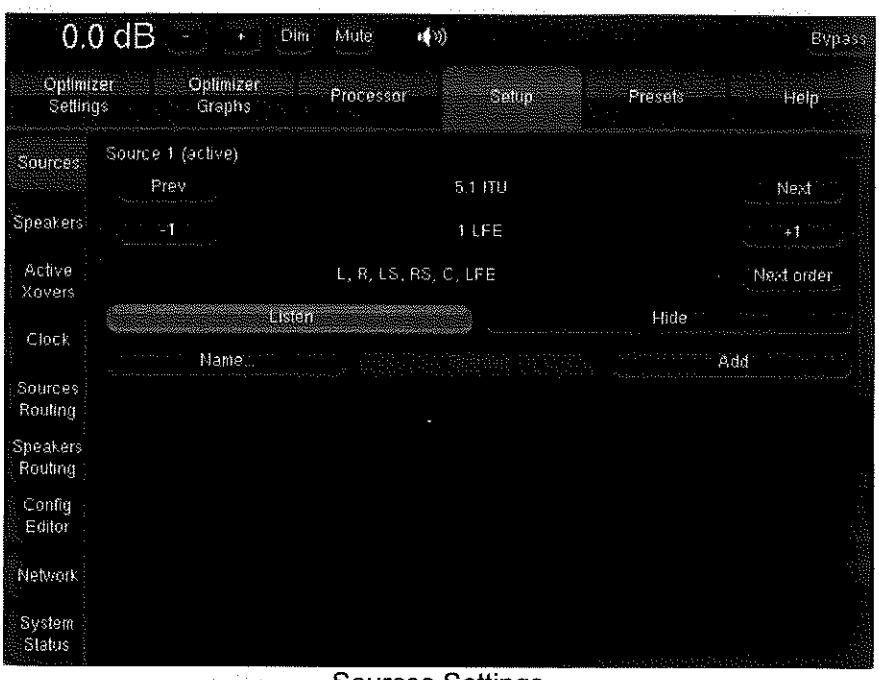
When the processor is connected to Trinnov's server, the **Network Status** in the Setup/Network page will change to ***"Connected to Trinnov Audio Server"***. For more information on network configuration, please refer to chapter 4.2.8.

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.

4 System Software Guide

4.1 Setup

4.1.1 Sources Settings



Sources Settings

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

The input format information is notably used for remapping as it gives the reference positions of the loudspeakers. The channel order is used for metering display.

Standard Settings:

- **Input format**: sets the input format: mono, stereo, 2 x stereo, 2/2, 3/1, 3/0.5.1, 6.1, 7.1, 8 channels, 12 channels or 16 channels.

- **Number of LFE:** sets the number of LFE signals feeding the Optimizer.

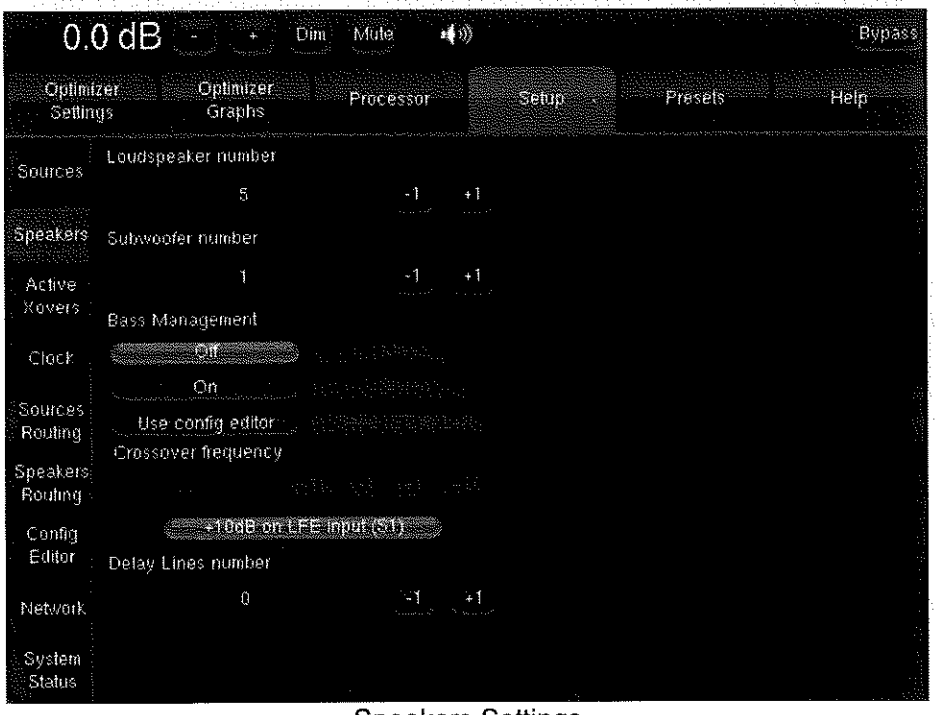
Please note: the difference between the "ITU" and "SMPTE" formats is the reference positions for the loudspeakers. "ITU" refers to the ITU R-775-1 specification for broadcast control rooms, while "SMPTE" refers to the SMPTE 202M standard for film theatres and mixing stages.

- **Channel order**: Sets the internal sort order of the multichannel signal. It is typically the sort order of the connections between your source and the Optimizer (if sources routing is straight). As an example, the default channel order for a 3/1 source is:
  Left - Right – Surround - Center
  If there is a LFE channel declared in the **Sources settings**, the *LFE* signal is always placed in the *last position* after the other channels and is referred to as "LFE" in the Sources Routing.

- **Listen:** used to select the source that is monitored.

- **Hide:** unused.
- **Name:** clicking on the "Name..." button will display a virtual keyboard allowing to change the source's name.
- **Remove**: used to remove a source. Removal cannot be cancelled.
- **Add:** used to add a source.

4.1.2 Speaker Settings



Speakers Settings

Standard Settings:

- **Loudspeaker Number**: sets the number of loudspeakers of your sound system.
  Please note: changing this setting requires re-calibration.
- **Subwoofer number**: 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 note:
  - The maximum number of channels which can be processed simultaneously is determined by your license ("**loudspeaker number**" + "**Subwoofer number**").
  - If "Bass management" is activated, Subwoofers receive low frequencies from the main channels below the chosen Crossover Frequency.

4.1.2.1 Bass Management

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

Important notes:

- The Optimizer always aligns the levels of all the loudspeakers, including the subwoofers.
- Regarding the option **+10dB on LFE input (S1):**
  - In a professional environment, this option should be used as required in respect with the recommended calibration level of the subwoofer and in order to achieve the best gain structure.
  - The LFE channel is recorded with a level offset of -10 dB. This offset has to be compensated for in the reproduction system. This option should therefore be used only when no other equipment within this chain applies this gain.
- This setting is *independent* of Bass Management on/off.

Please note:

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

The bass management modes are as follows:

- **Off**: this means that no bass management is performed. 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.

Please note:

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

4.1.2.2 Delay Lines

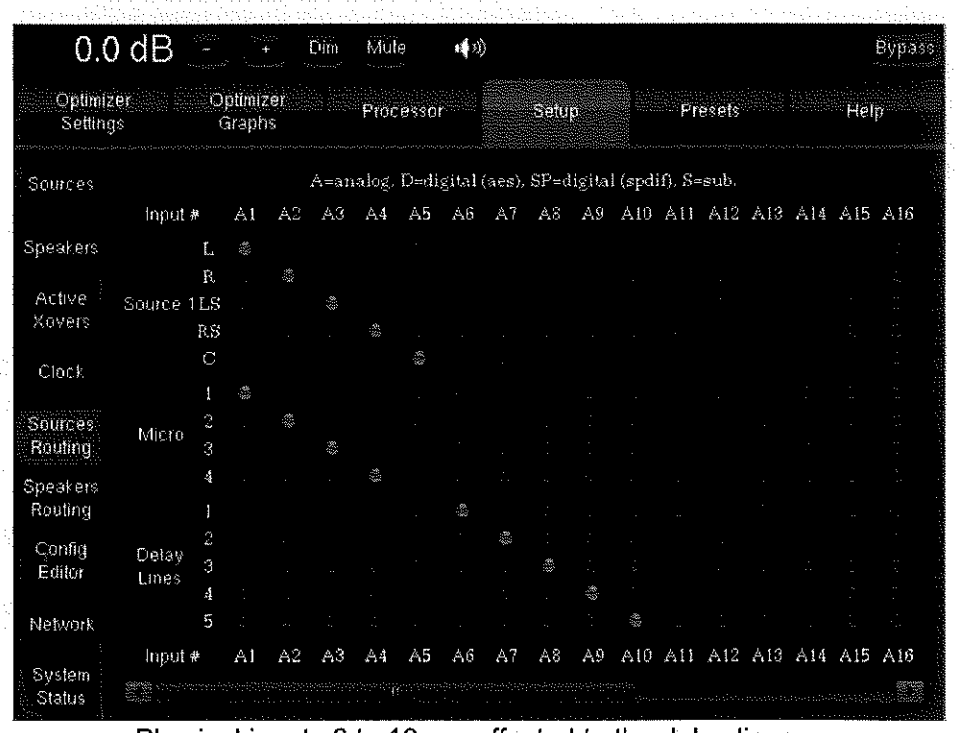
Delay lines are provided in order to delay additional audio monitoring systems with the outputs of the Magnitude32 without being affected by the Optimization.
Delay lines typically are used in broadcast environment to align audio monitoring between audio and video control rooms.

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.

Setting up Delay Lines:



Physical inputs 6 to 10 are affected to the delay lines...

...and routed to the physical outputs 6 to 10.

4.1.3 Active Xovers

4.1.3.1 Functionality

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

- 2 way, 3 way and 4 way crossovers
- up to 24 output channels, e.g. managing up to 6 four-way loudspeakers
- 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

4.1.3.2 Procedure

The active crossovers can be implemented as follows:

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

4.1.3.3 Manual settings

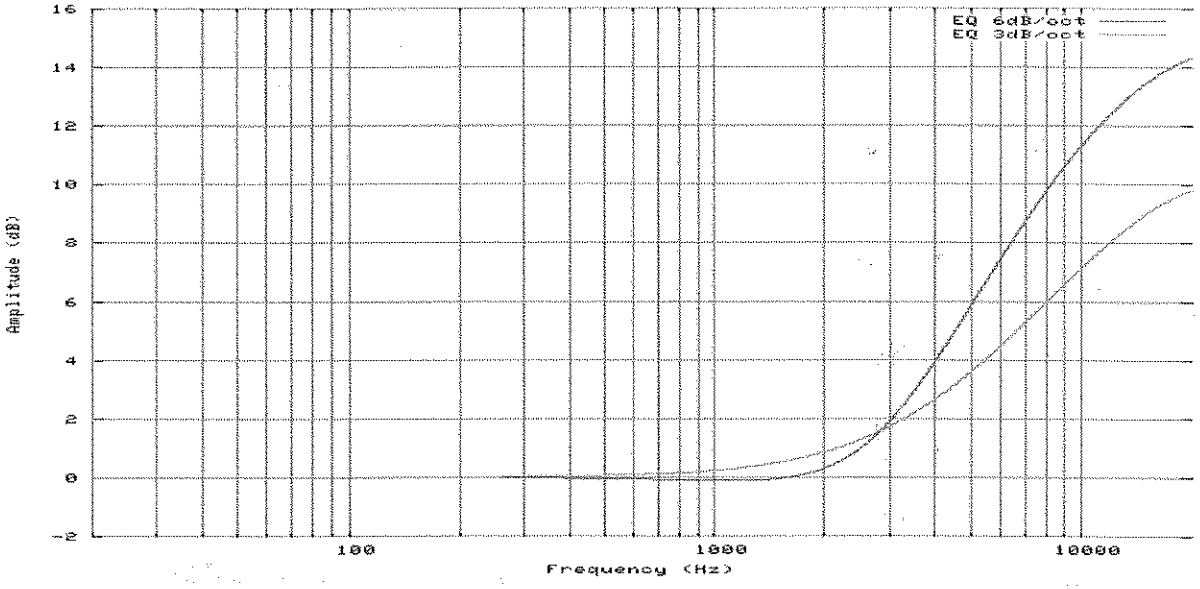
The Active Crossovers page displays one tab for each loudspeaker. The number of loudspeakers displayed depends on the number of loudspeakers declared in the **Setup/Speakers Settings** page.

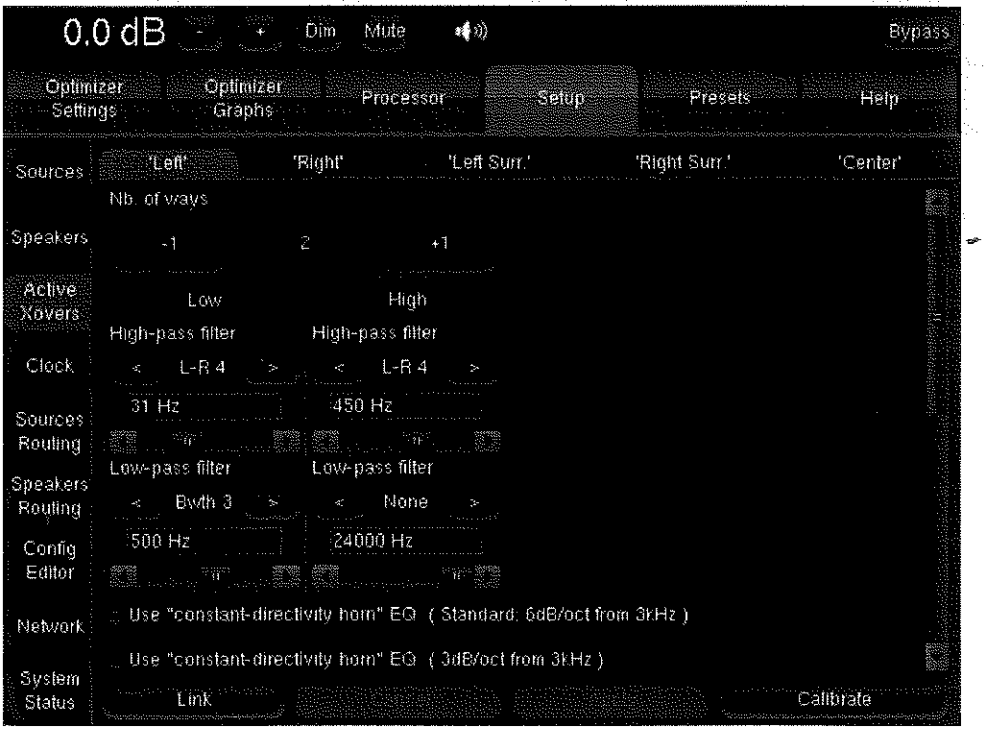
Note: setup of the crossover filters can be done simultaneously for any speakers thanks to their respective **Link** button. Linking has to be done before changing parameters!

The following settings are available for each loudspeaker:

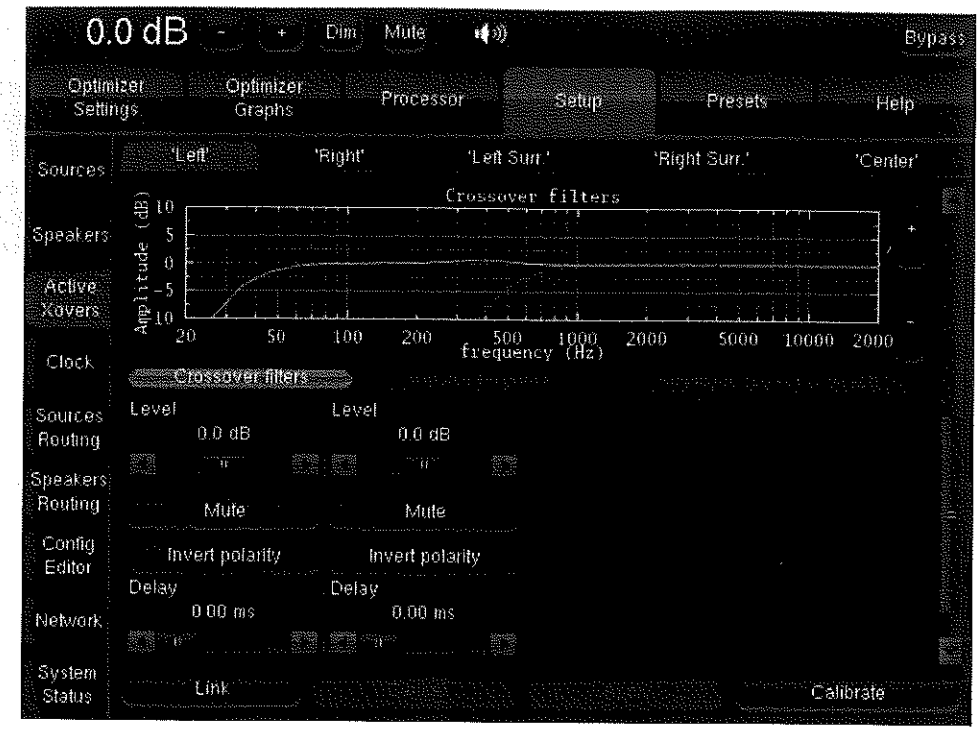
- The **number of ways** can be modified by pressing the **+1** and **-1** buttons,
- The **type** of high-pass or/and low-pass filters is chosen by pressing the "<" and ">" buttons,
- The **cut-off frequency** of each filter is set by sliding the scrollbar or by using the arrows.
- Two additional IIR filters are available under the name of "Constant-directivity horn" EQs. The purpose of such filters is to boost the high frequencies for speaker that use constant-directivity horns. These speakers tend to let the high frequency fall down at the sweet spot. Depending

on the horn you are using, you may want to boost the high frequencies by a 3 or 6 dB/oct, from about 3kHhz.





- The **Apply** button is highlighted as soon as a parameter is modified and is used to compute and load the new settings into the processor. Once compute is finished (gears icon disappears from the notification bar), filters are applied to the outputs.
- Physical outputs are affected to new ways. It is therefore necessary to check the speaker routing before being able to listen to the resulting filters.
- If change is unwanted, press the **Cancel 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.



- **Level** and **delay** can be adjusted on each way (driver),
- **Mute** and **Invert Polarity** buttons are also available.

4.1.3.4 Automatic settings

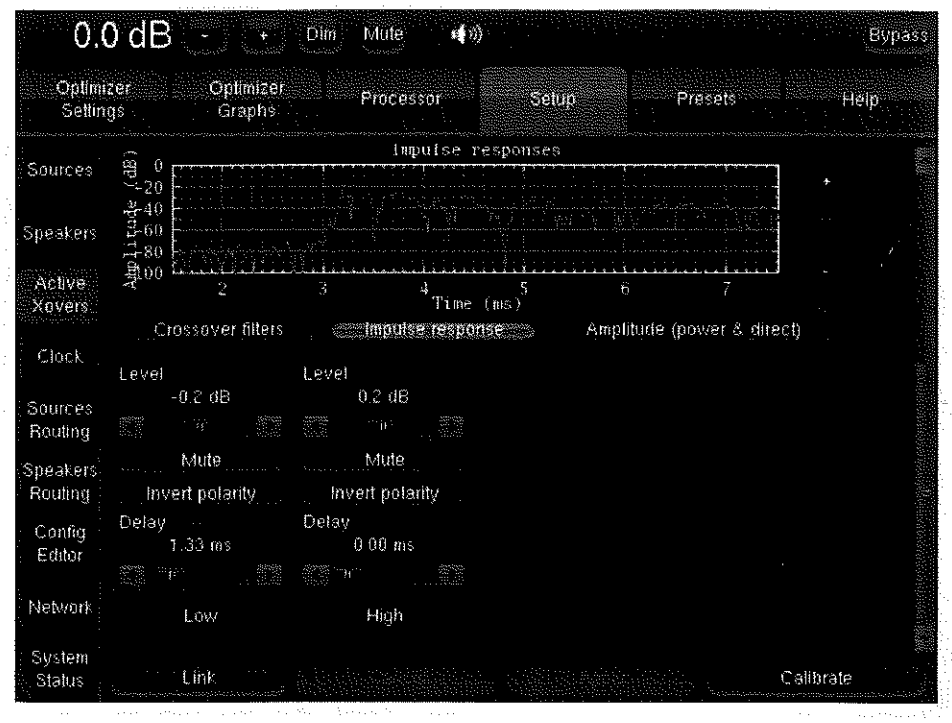
With the Optimizer option, it is possible, for each speaker, to set up the level, delay and polarity of each driver automatically by a simple procedure that calibrates the drivers separately:

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

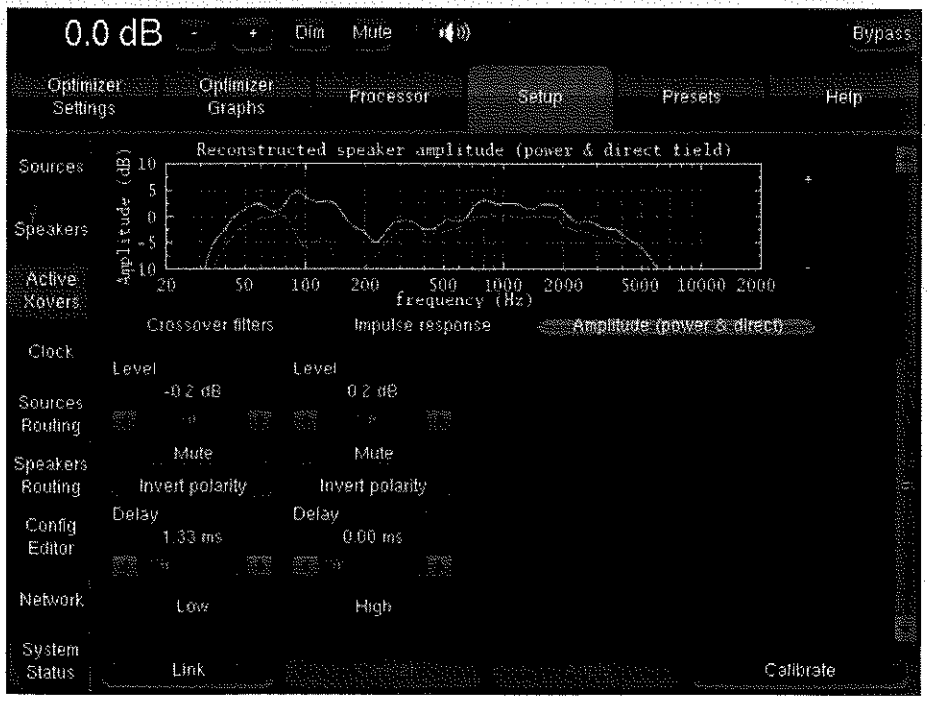
This procedure will automatically determine the levels, delays and polarities that should be applied on the drivers of the loudspeaker.

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

4. measured **impulse response** of each way: you can see whether the drivers are correctly synchronized.



5. 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:

6. As they are going to be automatically tuned, the previously set up levels, delays and polarities will be ignored during the calibration. In other words, tuning these parameters before launching the calibration will have no effect on the result.
7. On the Automatic Crossover main page, you also have the possibility to use linear regression for delay optimization, via the button that suggests so. This simply selects another algorithm for automatic crossover (the button must be toggled or untoggled **before launching the calibration**) Depending on the room and on the speakers, one algorithm can provide more accurate results than the other. Y
8. 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:
   - 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
   - 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").

4.1.4 Clock Settings

Clock Settings page

Standard Settings:

- **Status information** :
  - **Current sample rate**: shows the current clock frequency of the processor
  - **Detected sample rate**: shows the detected clock frequency.

- **Clock Mode**:
  - Master. Turn loudspeakers OFF before changing: changing this setting may result in loud clicks, depending on the analog converters used.

- **Audio Buffer Size:**
  - 512 is the default value

- Smaller values can be selected to decrease latency, but you should check that there are no sync losses. Note: changes are ignored until the next reboot.

- **Stored in Preset:** if this button is on when you save a Preset, clock settings will be saved and recalled with that preset.

4.1.5 Sources Routing

Sources Routing page

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

- Channels (rows) names and order of display depends on the input format and channel order declared for each software source in **Setup/Sources settings**. You can assign physical inputs to any channel. **Note:** the system automatically applies the relevant attenuation if several channels of a same source are routed to the same physical input.

  The Builtin preset source on a Magnitude32 is a 5.1 source with the following channel order and therefore routing:

Input 1	→ Ch 1 (L)	: Left
Input 2	→ Ch 2 (R)	: Right
Input 3	→ Ch 3 (LS)	: Left Surround
Input 4	→ Ch 4 (RS)	: Right Surround
Input 5	→ Ch 5 (C)	: Center
Input 6	→ Ch 6 (LFE)	: Subwoofer

- The 4 signals of the calibration micrphone, like any physical inputs, must also be routed. For convenience, 4 specific channels are dedicated to the microphone and regrouped in a special software source named "**Micro**". If additional "signal inputs" are needed, inputs can be routed to be shared for the microphone and a normal signal. (Note: in this case, both signals are routed simultaneously!).
- Sources routing may be changed after calibration.

Notes:

- Thanks to the **Auto Route** function of the remapping, the Optimizer will localize every loudspeaker and set correct channel or polarity feeds even if they are incorrectly connected. (The only requirement is that the loudspeaker plays its calibration signal).

- The **Spatial Optimization** also allows the use of more (or less) speakers than input channels. If this Spatial Optimization is not selected the first channel will be sent as per routing.

- Do not change the Speakers Routing once calibration has been performed. Correction filters have a direct association between routing outputs and speakers. Changing Speakers Routing after Calibration will result in a mismatch of compensation by re-directing the corrected signal to the wrong Loudspeaker.

4.1.7 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.

Config Editor

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.