

V.3.8

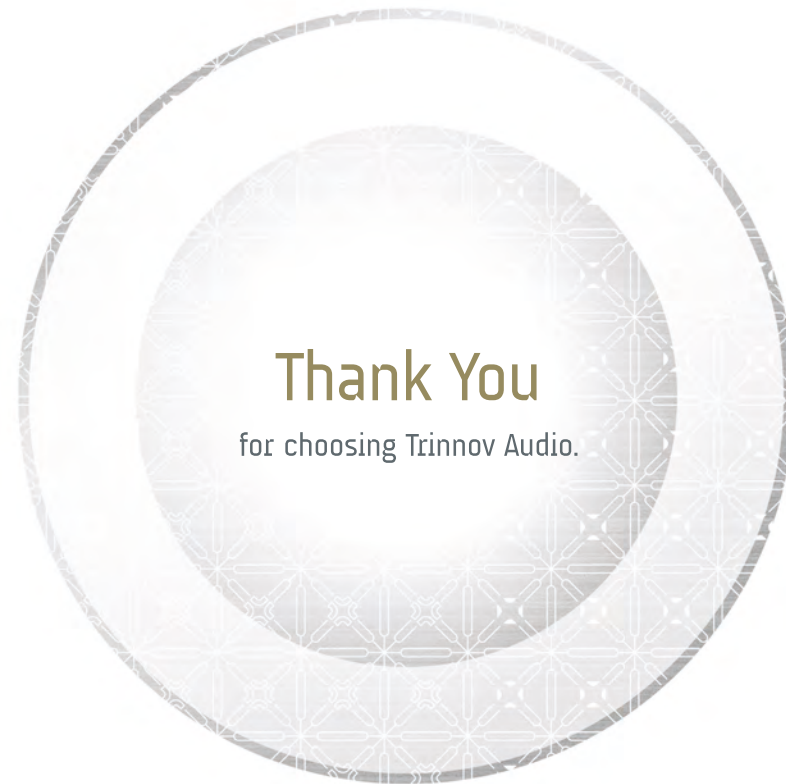


USER GUIDE

# ST2-HiFi

TRINNOV  
AUDIO





Your ST2 HiFi:    Serial Number:

Product ID:

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## 1.1 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: 3,15 AT.
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.
- 15. 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.



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

1.2

## WHAT'S IN YOUR BOX

1



2



3



- 1 THE ST2-HIFI
- 2 1 TRINNOV 3D MICROPHONE *(optional)*
- 3 1 POWER CABLE

# ACOUSTIC FIDELITY

The weakest element of any high-fidelity system is the room. Typical rooms introduce up to 10dB of distortion in the frequency response. Furthermore, loudspeakers with a perfect impulse response don't exist.

Trinnov's ST2 HiFi solves the acoustic equation. It takes your high-end system to a whole new level of accuracy, from High-Fidelity to Acoustic-Fidelity.

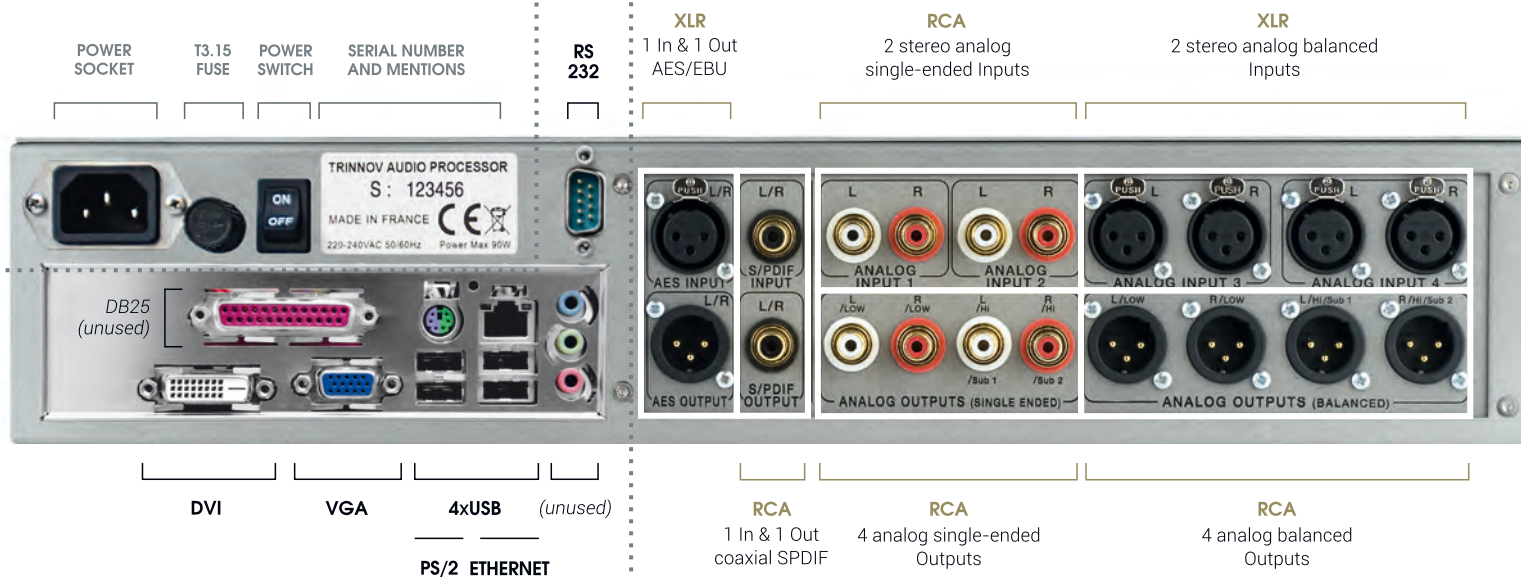
## 1.4 THE REAR PANEL



Power section



Audio section



PC section

## 1.5 SPECIFICATIONS

### HIGH PERFORMANCE AUDIO

RESOLUTION / SAMPLING RATE	24 bits/96 kHz – 192kHz ready
A/D SIGNAL-TO-NOISE RATIO	119 dB (A-Weighted)
THD+N ADC	-103 dB
D/A SIGNAL-TO-NOISE RATIO	118 dB (A-Weighted)
THD+N DAC	-98 dB
CLOCK / JITTER	Variations over 25ps are recovered, jitter attenuation superior to 50 dB is achieved above 100Hz
POWER SUPPLY	Independant for audio and processing sections
SAFETY COMPONENTS	AntiPop relays on each analog output
DESIGN AND ASSEMBLY	All Audio boards designed and manufactured in France by Trinnov Audio

### PHYSICAL CHARACTERISTICS

**CHASSIS:** 2U

**POWER SUPPLY:** 240V AC / 50-60 Hz. *Option: 130V AC*

**CONSUMPTION:** 90 W max.

**WEIGHT:** above 9,4kg

**ENVIRONNEMENTAL CONDITIONS:**

- Temperature: 0°C – 40°C (32°F – 104°F)
- Humidity: 20% – 80% relative humidity (without condensation)





## CONNECTIVITY

### ANALOG INPUTS/OUTPUTS

- 4 single-ended channels inputs via 4 x cinch (10k Ohms)
- 4 single-ended channels outputs via 4 x cinch (10k Ohms)
- 4 balanced channels inputs via 4 x XLR (20k Ohms)
- 4 balanced channels outputs via 4 x XLR (20k Ohms)

### AES INPUTS/OUTPUTS

- 2 channels input via 1 x XLR (110 ohms)
- 2 channels output via 1 x XLR (110 ohms)

### SPDIF INPUTS/OUTPUTS

- 2 channels inputs via 1 x cinch (75 ohms)
- 2 channels outputs via 1 x cinch (75 ohms)

**IMPORTANT NOTE** SPDIF, AES, Single-Ended and Balanced Analog outputs are mirrored.

## PROCESSING SECTION

PROCESSOR Intel Dual-Core 1,8 GHz

DATA WIDTH 64 bits, floating point

RAM 1 Go DDR3

STORAGE Flash Drive 1 GB

COOLING SYSTEM Custom heat sinks + additionnal slow fans

MAX. NUMBER OF PROCESSING CHANNELS 4 channels at 96 kHz

## 1.5 THE TRINNOV 3D MEASUREMENT MICROPHONE

The measurement microphone is one of the most critical component in the Trinnov calibration procedure.

The Optimizer's sophisticated algorithms not only rely on very accurate acoustic measurements but also on the ability to localize speakers positions and to detect early reflection origin.

The microphone consists of 4 capsules mounted at the top of thin brass tubes to avoid diffraction.

The capsules form a tetrahedron figure, ideal to identify distance, azimuth and elevation altogether with a spatial resolution below  $\pm 2^\circ$  in every direction.

Consequently, capsules are identified from 1 to 4 and the microphone cables are labelled accordingly.

A led incorporated in the body of the microphone indicates the front of the microphone that should be pointed at the center of the soundstage before proceeding with a measurement.

Flat response (within  $\pm 0,1$  dB across the 20Hz-24kHz frequency range) is guaranteed by individual compensation filters.

The microphone uses a standard 9V PP3 LR61 battery to power capsules and electronic.

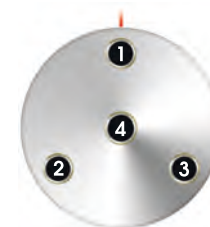
The second purpose of the frond led is to indicate the battery level.

*Back side of the microphone*



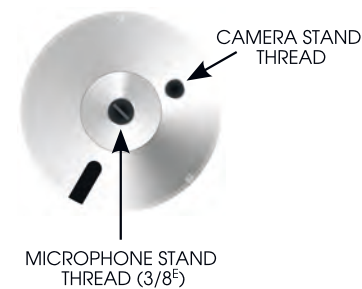
Top view

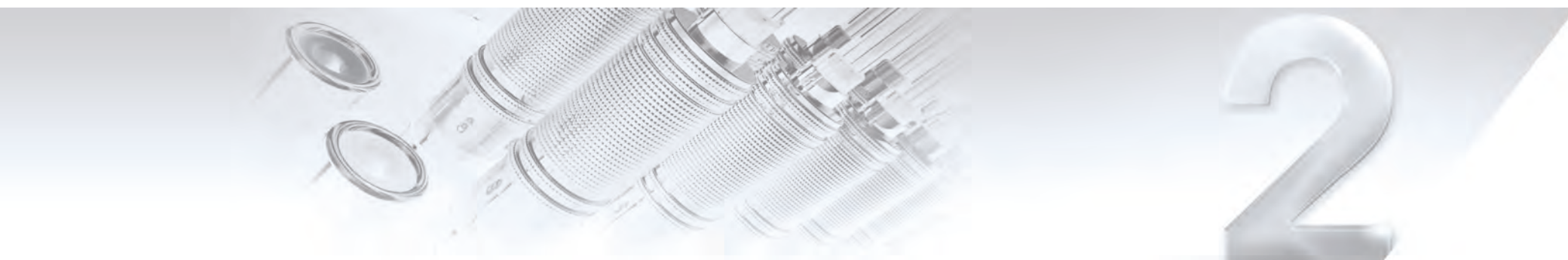
*Front of the microphone*



CAPSULES NUMBERING SYSTEM

Bottom view





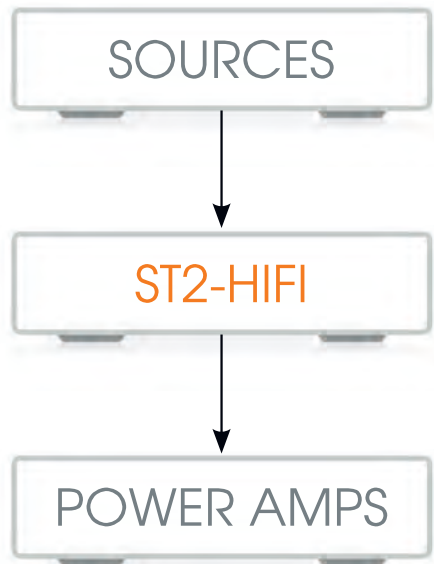
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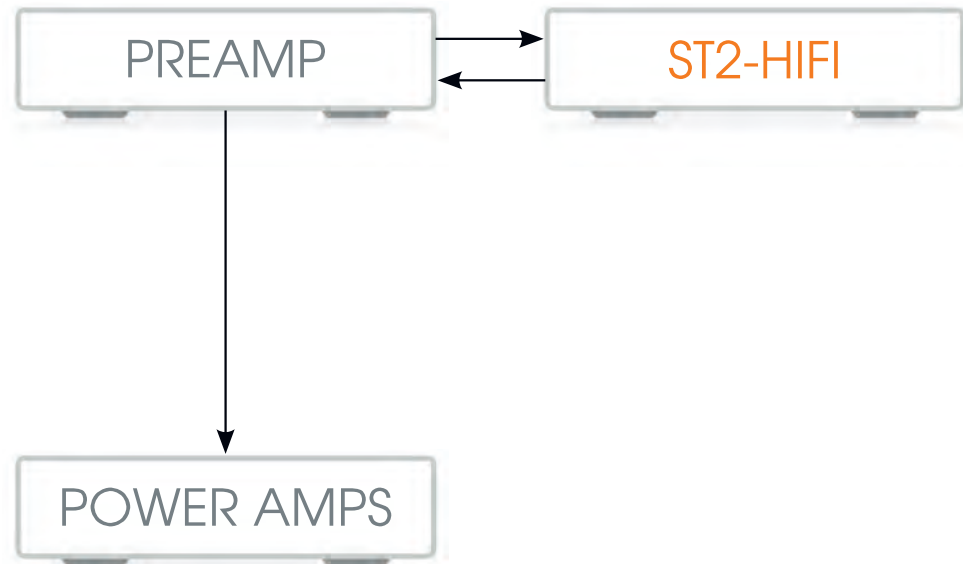
## 2.1 TYPICAL CONNECTIONS

The ST2 HiFi seamlessly integrates your hifi installation. Whether your system allows for series connection and/or insertion loop, both analog and digital inputs/outputs of the ST2 HiFi can be used.

Series Connection



Insertion Loop



## 2.2 CONNECTING A SOURCE

Connecting a CD Player  
to the AES Input 1

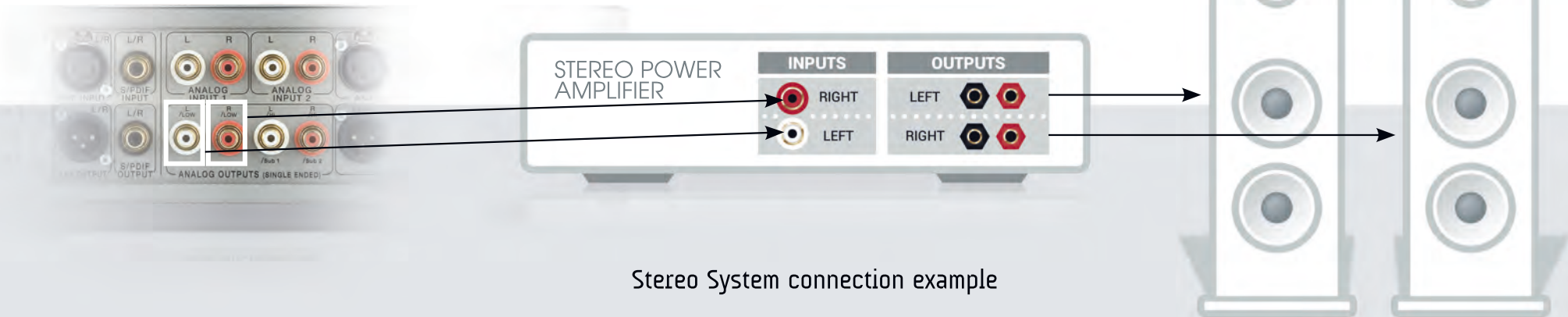


The ST2 HiFi allows you to connect and play any stereo source without any configuration.  
This example shows how to connect the AES output from a CD player to the ST2 HiFi.

## 2.3 CONNECTING SPEAKERS

### Default output connectors for a stereo system

This image shows that any first pair of outputs should be used as default stereo system output. Please remind that these outputs play simultaneously.



## 2.4 CONNECTING THE MICROPHONE

The Trinnov 3D measurement microphone must be connected to the balanced analog inputs of the ST2 HiFi.



**IMPORTANT** Always make sure the microphone is turned OFF before you connect it to the ST2 HiFi.



The microphone cables are labelled and numbered from 1 to 4. Please respect the connection order of the microphone, such as indicated in this figure:



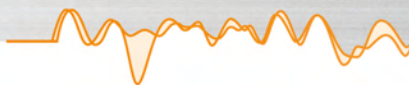
Connection of the Microphone





Stereo

HIGH END







# GETTING STARTED

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## 3.1 POWERING ON THE ST2 HiFi

### STARTING UP



**IMPORTANT NOTE** Don't forget that the amplifier should always be powered up last.

Turn on the main power switch on the rear panel to supply the ST2 HiFi with electricity, then press the front panel power button to start the ST2 HiFi. The system takes **45 seconds to initialize**.



### SHUTTING DOWN



**NOTE** It is not recommended to cut the AC Power, as the system saves several "last-used" settings while shutting-down.

To shut the ST2 HiFi down, press the front panel power button once. The system will need approximately 10 seconds until complete stop.



## 3.2 REMOTE ACCESS TO THE GRAPHICAL USER INTERFACE (GUI)

The ST2 HiFi allows several options for remote control:

- Remote control from a tablet, laptop or smartphone over the network, using VNC.
- Telnet command or automation system (Crestron, Savant...)
- Screen, mouse and keyboard directly connected to the PC



The ST2 HiFi has built-in Ethernet and optional WiFi. It can simultaneously be part of a network as:

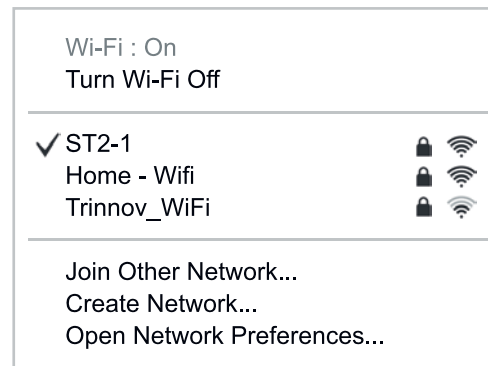
- a WiFi Access Point (AP) to create its own WiFi network;
- a hard-wired client to join an existing network through an Internet Service Provider (ISP) Box/Router;
- a WiFi client to join an existing wireless network.



### IMPORTANT NOTE

The optional WiFi USB dongle must be connected to a USB 2.0 port (black) of the ST2 HiFi before ignition.

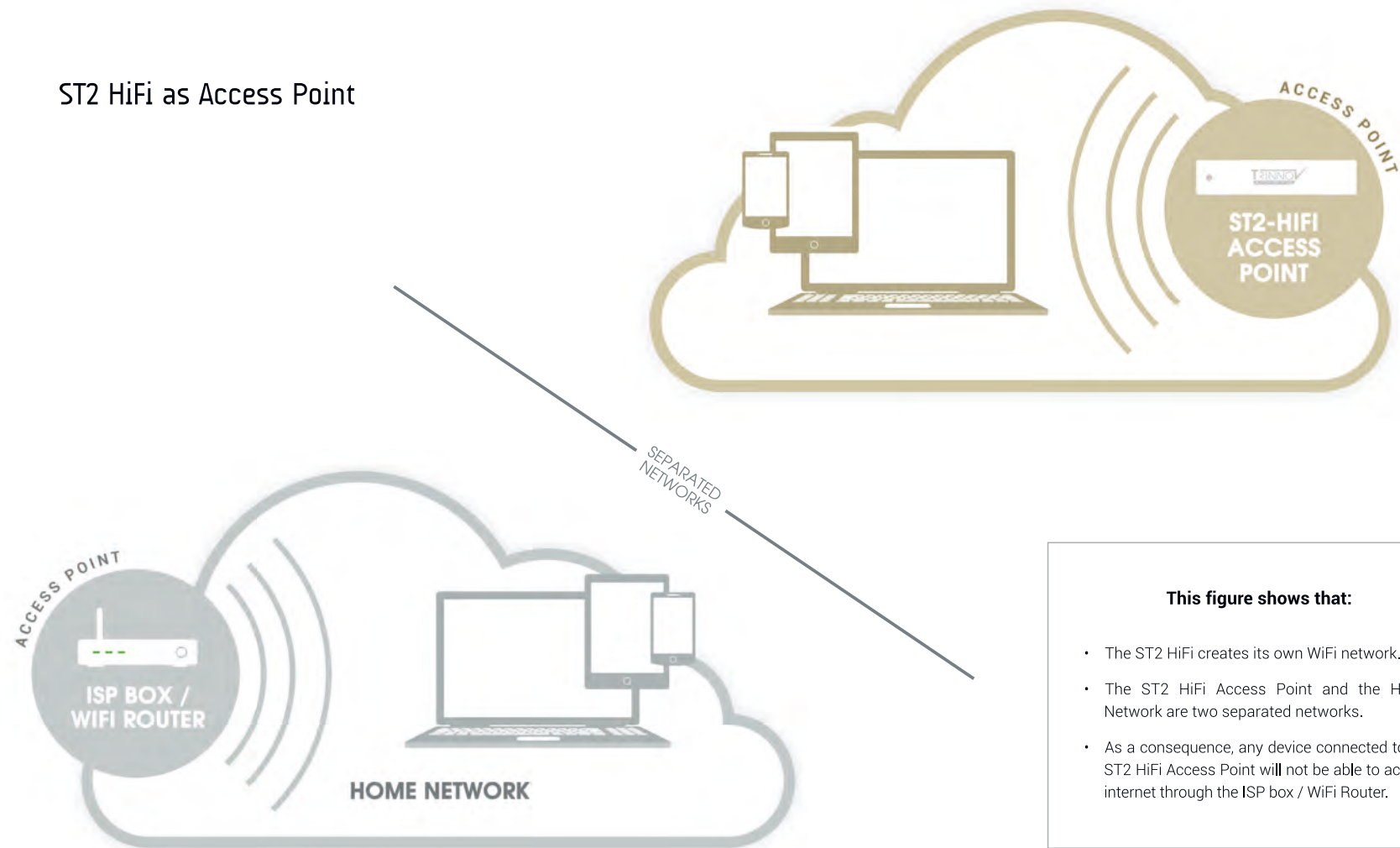
Provided that the optional WiFi dongle has been inserted into one of the USB port of the ST2 HiFi before ignition, a dedicated WiFi Access point should appear in the WiFi Networks list of your tablet/smartphone/computer under the name "ST2-xx" where xx is the serial number of your ST2 HiFi.



WiFi Networks List - Mac OS X

Once you've selected the ST2 HiFi Access Point, you must enter the default password "**calibration**" to establish the connection.

## ST2 HiFi as Access Point



### This figure shows that:

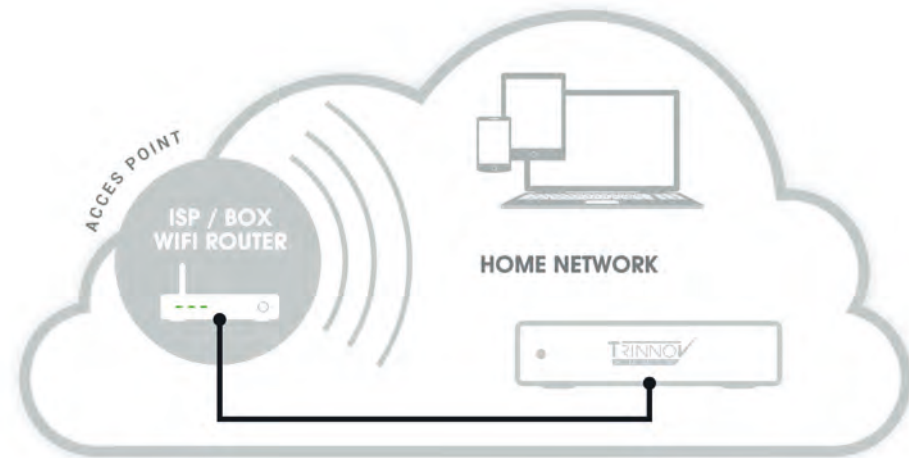
- The ST2 HiFi creates its own WiFi network.
- The ST2 HiFi Access Point and the Home Network are two separated networks.
- As a consequence, any device connected to the ST2 HiFi Access Point will not be able to access internet through the ISP box / WiFi Router.

## 3.4 USING THE ST2 HiFi AS A DHCP CLIENT

The default Ethernet mode of the ST2 HiFi is set to DHCP client, meaning that it will automatically be detected on most domestic networks.

**This figure shows that:**

- The ST2 HiFi is hard-wired to the ISP Box / WiFi Router.
- This same ISP Box / WiFi Router is the Access Point of the Home Network.
- Potential control devices (laptops, tablets and smartphones) are connected to the ISP Box through the WiFi.
- As a consequence, control devices are able to communicate with the ST2 HiFi over the network and access Internet.



### NOTES

- This configuration only works if the WiFi router includes an active DHCP server.
- With ISP Box standard firewall settings, this configuration should allow the ST2 HiFi to reach Trinnov Audio's Server for software updates through Internet.
- The network parameters of the ST2 HiFi can also be set manually in the Settings/Network page of the Graphical User Interface

## 3.5

## REMOTE ACCESS FROM A TABLET, LAPTOP OR SMARTPHONE

The ST2 HiFi has a built-in VNC server that allows you to fully control the Optimizer from any VNC client host device over the network.

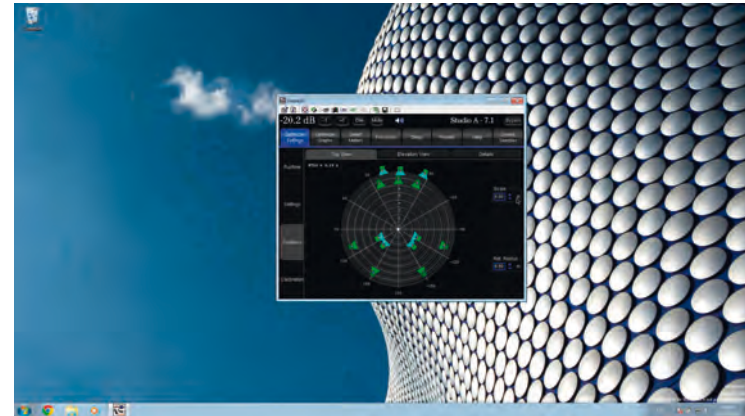
### 5.4.1 About VNC

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.

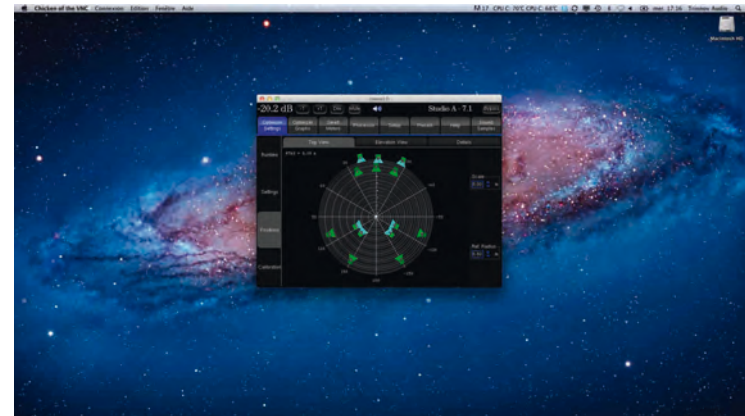
In other words, VNC provides full control of the ST2 HiFi from a tablet, smartphone (iOS, Android, Blackberry, Nokia...) or laptop (PC, Mac or Linux).

You can find different VNC Clients/Viewers on Internet/Application stores:

<b>Windows®:</b>	TightVNC
<b>MAC OS:</b>	Chicken of the VNC
<b>iOS/Android:</b>	Mocha VNC Lite



VNC control from Windows® 7



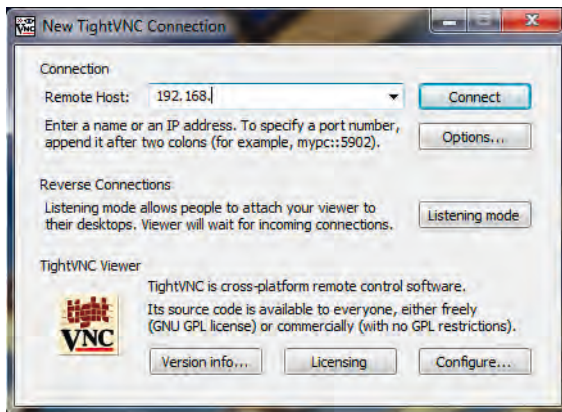
VNC control from Mac OS X

## 5.4.2 VNC Connection

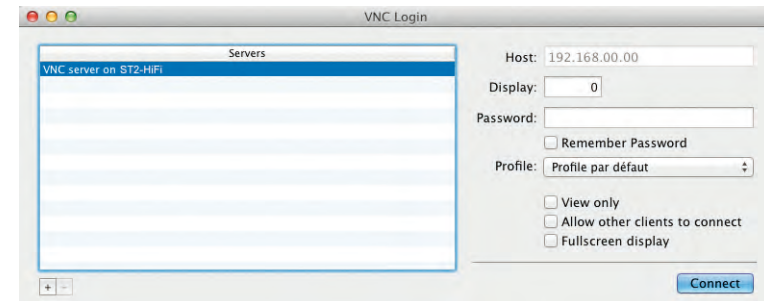
First install the VNC client application on the device you want to control the ST2 HiFi from. Make sure this device is connected to the same network (either wired or wireless network).

The VNC Client will ask you to enter the following informations:  
the server address (or host depending on the application) + a display port + a password.

Some client automatically displays the available server's list:

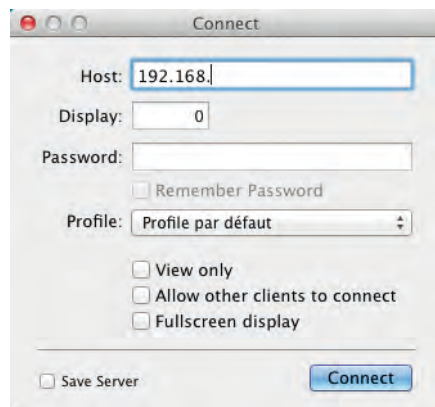


"TightVNC" Login Panel (Windows 7®)

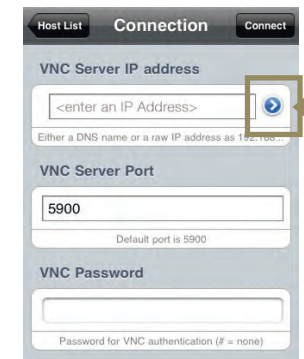


Chicken of the VNC available server's list

The example above shows the connection screen of the VNC client for Mac "Chicken of the VNC".  
In that case, the server address is automatically filled out.



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



Mocha VNC Lite available server's list

Likewise, the blue arrow displays the available servers list on the VNC client for iPhone/iPad "Mocha VNC Lite".



## Server Address

The server address is the IP address of the ST2 HIFI:

- The Ethernet IP address if the ST2 HIFI is hardwired to the local network.
- The WiFi IP address if the VNC client is connected to the ST2 HIFI Access Point.

IP addresses can also be retrieved from the graphical interface in the **Setup/Network** page:



## Display Port

Depending on your VNC client, the default display port must either be set to 5900 or 0.

## Password

The password is the last 6-digit serial number of the ST2 HIFI.

## 3.6

## CONNECTING THE ST2 HIFI TO AN EXISTING WIFI NETWORK

### 3.6.1 Introduction

Unlike using the ST2 HiFi as an Access Point or DHCP client, connecting the device to a home network as a wifi client requires some configuration. This is the most flexible and comfortable network configuration since it allows both controlling and streaming to the ST2 HiFi from a smartphone or tablet while keeping the access to internet, without any cables.



**This figure shows that:**

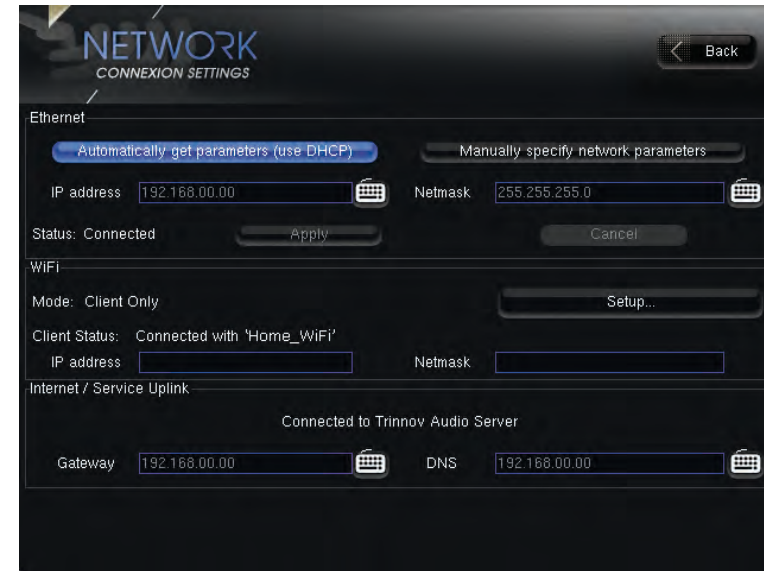
- The ST2 HiFi is part of the Home Network, like any other laptop, smartphone or tablet connected to the Home WiFi via Wireless connection.
- The WiFi AP of the Home Network is also the ISP Box; any devices controlling the ST2 HiFi can also access Internet.

**There are 3 WiFi modes** that can be selected in the *Settings > Network > WiFi Setup* page:

- **OFF** the WiFi module of the ST2 HiFi is disabled.
- **FULL** (default) the ST2 HiFi is both used as an Access Point and a Client.
- **CLIENT** the Access Point is disabled but the ST2 HiFi still operates as a WiFi client as long as the configuration is valid.

## 3.6.2 Configuration of the client

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 (Access Point + client) 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 displays the available WiFi Access Points.
4. The **Encryption** column displays a locker icon when the network is protected by a passphrase and/or encrypted.
5. The **Signal** column 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.



ST2 HIFI - Network Page



ST2 HIFI - WiFi Settings Page



ST2 HIFI - Access Point list

Presets

BACKUP / RESTORATION

Clock

AUDIO BUFFER SIZE

Sources

POWER-ON DEFAULT

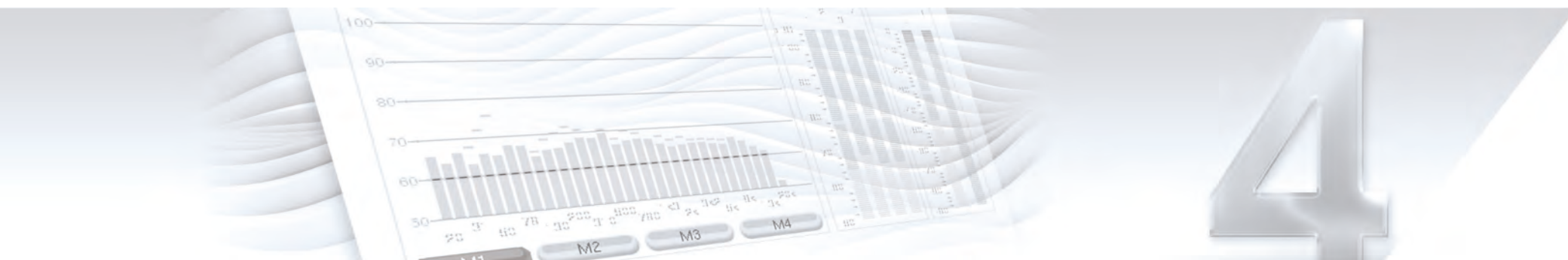
INPUTS/OUTPUTS CONTROL

USB backup

EQ EDITION

Channels

PRESETS MANAGEMENT



# CONFIGURATION

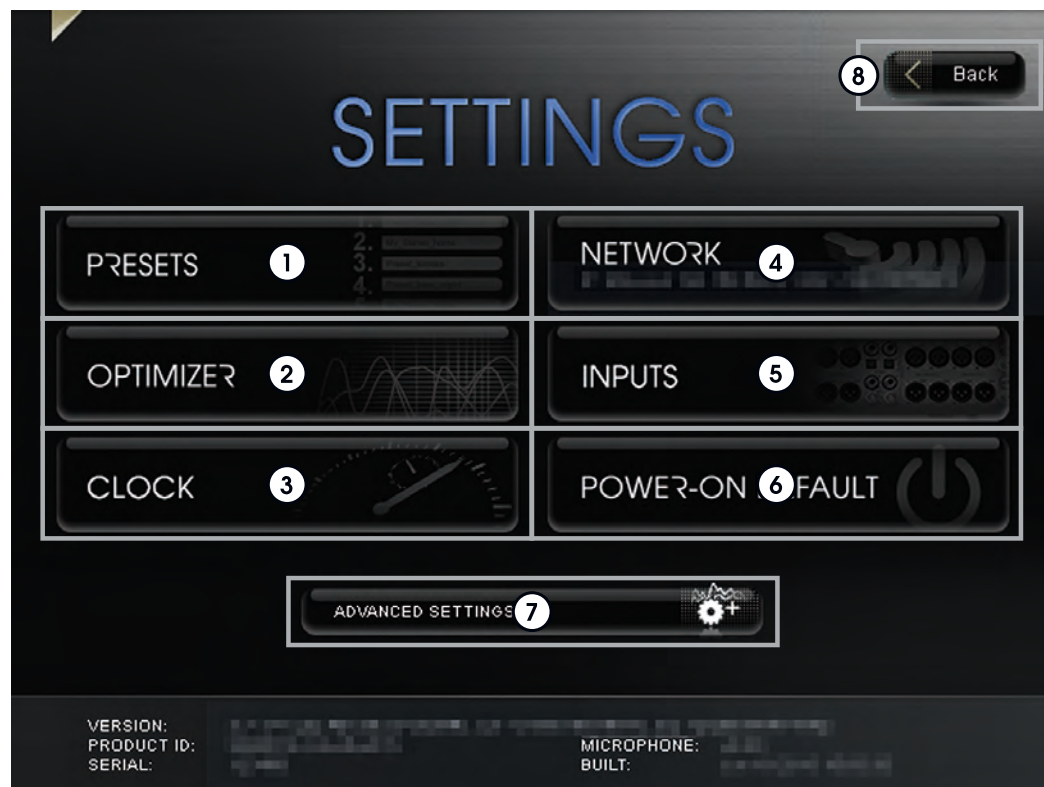
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## 4.1 OVERVIEW

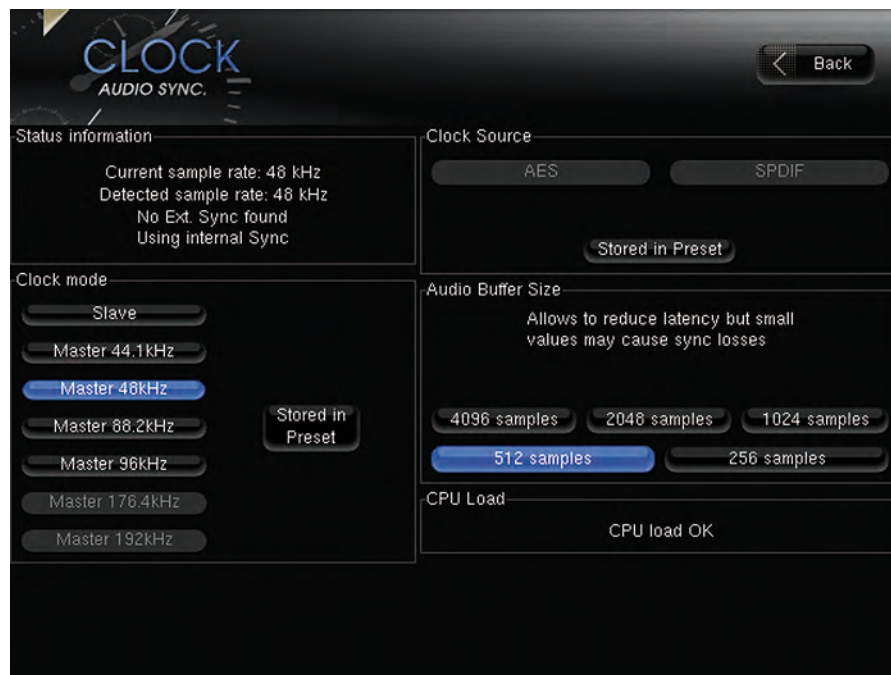


You can access to the Advanced preamplifier settings of the ST2 HiFi on the home page by pressing the **Setting** button.



- ① Rename, save, copy or delete presets.
- ② Access to the wizard, The first step to your Optimized world, via the *Optimizer* menu.
- ③ Internal or external Audio Clock are ajustables in the *Clock* menu.
- ④ Manage the ST2 HiFi connectivity with *Networks* settings.
- ⑤ Name and Organize audio inputs with the *Sources* menu.
- ⑥ Choose your preferred parameters to load on start-up.
- ⑦ Unleashed the full power of your Optimizer with the *Advanced settings* menu.
- ⑧ Back to the Home page.

## 4.2 SETTING UP AUDIO CLOCK



Home > Settings > Clock

The **Status information** frame displays the current and detected sample rate. Its purpose is to indicate whether or not the ST2 HiFi is correctly synchronised.

The **Clock mode** frame proposes two modes: **Slave** & **Master**.

- In **Slave** mode, the ST2 HiFi synchronizes itself on an external clock signal emitted by an SPDIF or AES audio signal from a digital transport, or by an external dedicated world clock. The choice of the clock signal is done in the **Clock Source** frame.
- In **Master** mode, the ST2 HiFi uses its own internal clock for synchronization. This is the default mode for the Network music playing. As a consequence, the **Clock Source** parameters become unavailable since the Slave mode synchronization does not apply.

**Clock mode** and **Clock Source** frames have a **Stored in Preset** button to store the current parameters in the current preset. You still have to save the preset to permanently set that configuration on a preset change or after an ST2 HiFi shutdown/startup sequence.

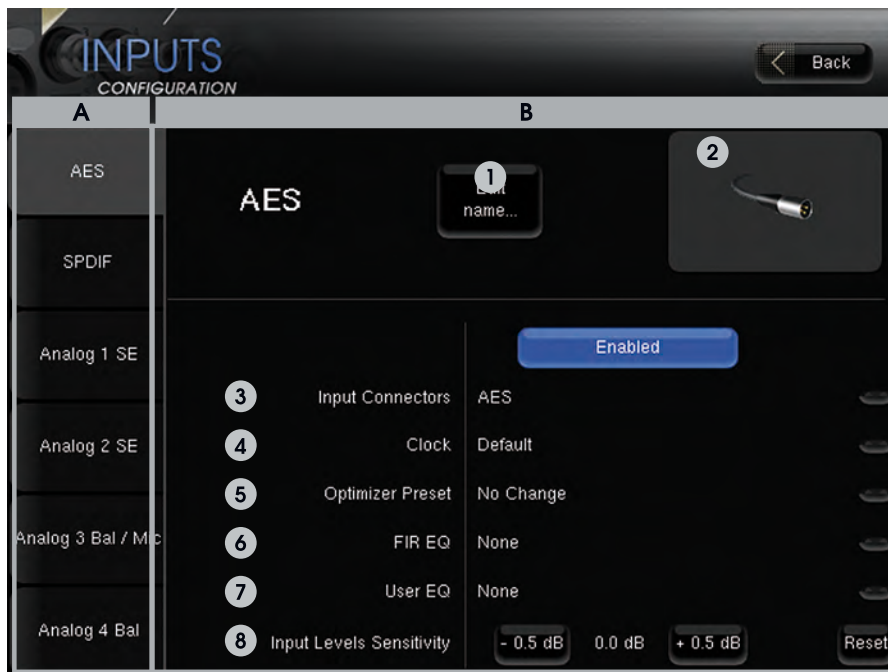
In the **Audio Buffer Size** frame you can select the audio buffer size. A bigger buffer allows to prevent loss of synchronization due to unregular incoming audio flow, but as a counterpart the processing time delay is longer.

The **CPU Load** frame provides an estimation of processor's usage. If the message isn't **CPU load OK** there is a risk of synchronization loss. That phenomenon can be attenuated by selecting a bigger audio buffer in the **Audio Buffer Size** frame (see explanation above).



## 4.3 SOURCES CONFIGURATION

Home > Settings > Sources



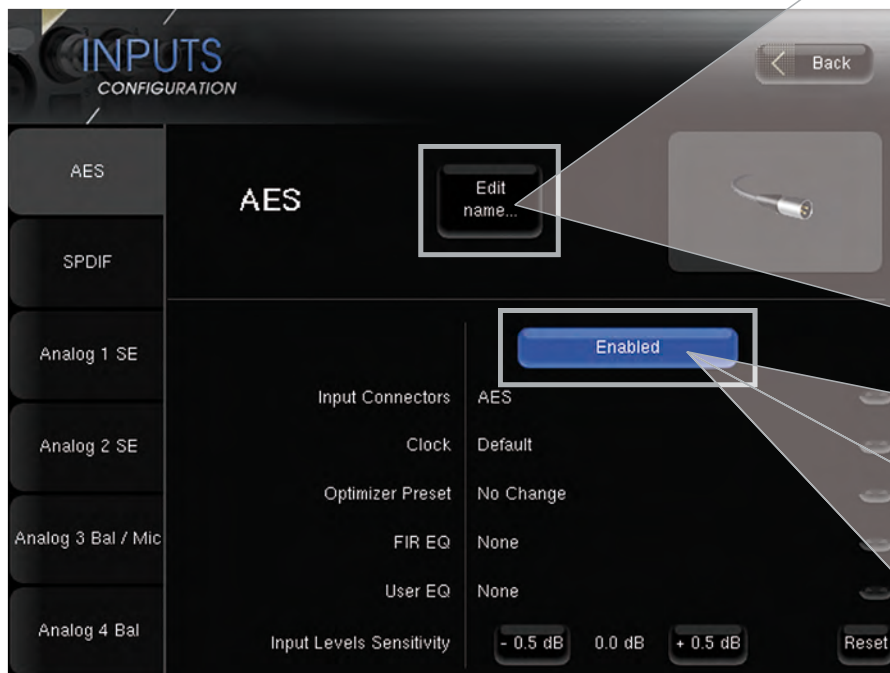
**A** One vertical tab per source (profile) on the left. **B** Source-specific parameters on the right.

The *Sources Configuration* screen allow you to manage up to 12 sources, wich will be displayed on the start screen.

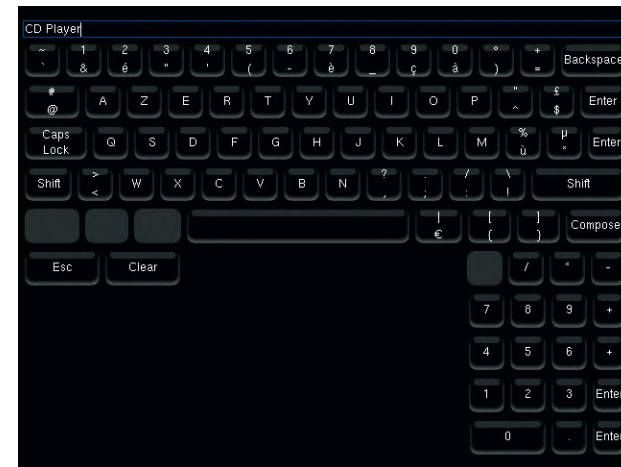
For each source, you can:

- 1 **Edit its name** with the *Edit name* button;
- 2 **See the icon** that will be displayed on the homepage;
- 3 **Choose the input connector** used among the analog (Balanced and Single Ended), digital (S/PDIF, optical or AES) and network. The selection is done via the listbox labelled *Input Connectors*;
- 4 **Set up the clock.** Note that the clock defined in the *Source* menu will prevail on the clock defined in the preset;
- 5 **Choose the preset to use;**
- 6 **Setting up the FIR EQ** (available in the *Advanced Settings*. Please refer to a Trinnov certified installer) used by this source:
  - Select *No Change* to keep the current FIR EQ
  - Select *None* not to use FIR EQ
  - To use the FIR EQ stored in the preset assigned to the source, use *As in preset*.
- 7 **Choose the User EQ** used by this source:
  - Select *No Change* to keep the current User EQ,
  - Select *None* not to use User EQ,
  - To use the User EQ stored in the preset assigned to the source, use *As in preset*,
  - For more information on User EQ, please refer to *chapter 7.6*.
- 8 **Choose the Input Levels Sensitivity.** This parameter can be used to adjust input levels and achieve a consistant playback level for every source.

FIR EQs and User EQs are optional manual EQs.



### Virtual keyboard



Unused sources can be hidden from the *Home* with the **Enabled** button:

- Black button : the source won't be displayed on the *Home* screen.
- Blue lighted button : the source will be displayed on the *Home* screen.

## 4.4 PRESETS MANAGEMENT

Except the sources configuration and power-on default settings, all parameters are saved into presets. The ST2 HiFi can store up to 29 user presets.

Home Page - Presets list



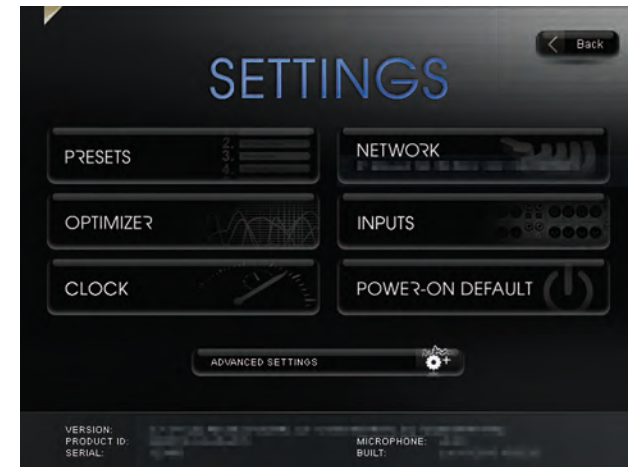
The presets list is available from the **Home** page. After first start-up, the only preset available in the unit is the built-in preset. This **Presets** menu is only used to select and load presets. Presets management is handled from a dedicated **Presets** menu available through the **Settings** panel.



### IMPORTANT

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.

Settings Panel



Presets 1-9 page



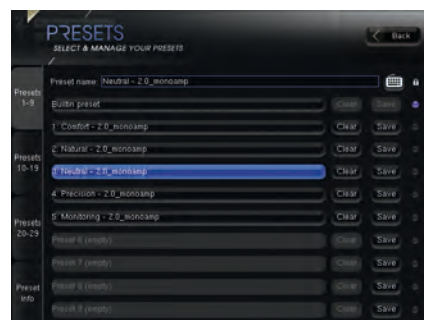
The Presets page consists of four vertical tabs, on the left side of the screen:

- **Presets 1-9:** includes the Built-in preset and presets 1 to 9;

#### NOTE

The Built-in preset is locked and cannot be overwritten.

- **Presets 10-19:** presets 10 to 19;
- **Presets 20-29:** presets 20 to 29;
- **Preset Info.:** shows the current preset informations.



Each Preset tab consist of the following items:

- The **Preset name** field: used to enter or modify a preset's name.
- 10 memory slots including from left to right:
  - The **Preset name/number**, greyed when the slot is empty, highlighted in blue when the preset is selected.
  - A **Clear** button to clear the memory slot. This function irreversibly deletes the preset's parameters.
  - A **Save** button to save current parameters as a preset in the memory slot.
  - A **Lock** checkbox to protect the preset: disables the **Clear** & **Save** buttons.

As an example, the recommended procedure to duplicate a preset and copy it in a different memory slot would be to:

1. Reload the preset to recall the exact configuration you want to duplicate.
2. Use the **Save** button of an empty preset to recopy it.

## ABOUT THE "SETUP WIZARD"

The setup Wizard always uses a bank of 5 presets. You may therefore have to move your presets before starting a new calibration via the wizard.



#### NOTE

A specific preset can be linked to each source.  
Presets can be backed-up and restored to/from a USB key. Please refer to chapter 4.6.

## 7.5 POWER-ON DEFAULT SETTINGS



In the **Power-on Default** menu you can setup which source and which preset to be loaded at startup.

The setup is straightforward :

1. **Choosing the default preset :** Change the preset in the **Power-on Optimizer Preset** frame with the < > arrows button
2. **Choosing the default source :** Select the Source in the **Power-on Source** frame by clicking on its name. It will then be highlighted in blue. The **Use Last Selected Source** allows to restart on the same source the ST2 HiFi was on at the last power off.
3. Once your setting is over change click on the **Back** button to apply the changes.

## 4.6 USB BACKUP/RESTORE



### How to use a USB key to save/load presets and to save acoustic reports.

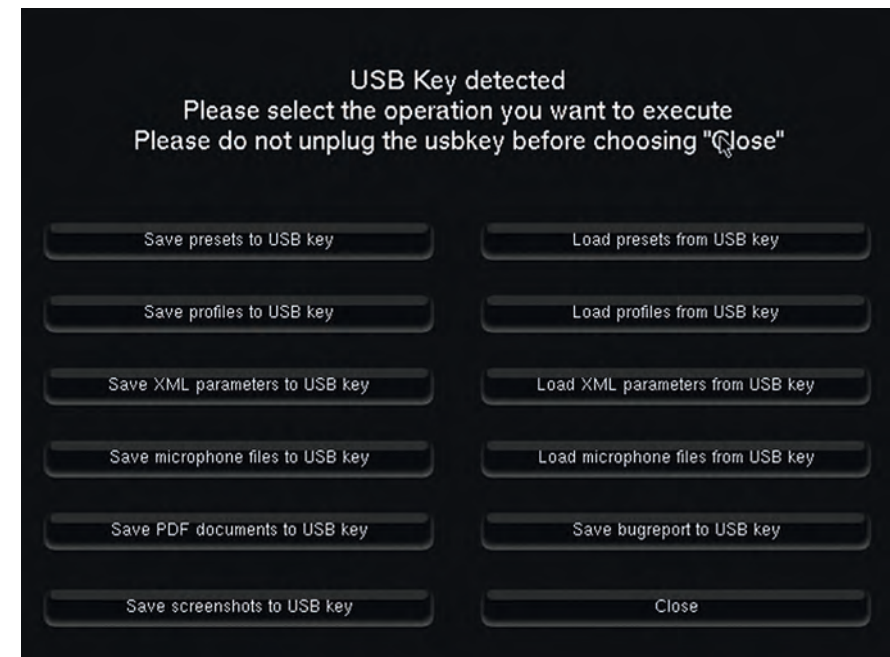
You can:

- Load or save presets to a USB key.
- Save or load profiles (= sources configuration) to a USB key.
- Save or load XML parameters to a USB key.  
**These parameters are meant to be modified by installers only.**
- Save or load the microphone calibration files to a USB key.
- Generate and save a PDF report for each of your presets on a USB key. This report includes the summary of your system measurements and the correction applied.
- Save a bug report to a USB key: when an unexpected error occurred the system is halted to prevent damage to your system and a bug report is generated. After the restart of the ST2 HiFi you can save that bug report on the USB key and then send it to Trinnov for analysis.
- Save screenshots to a USB key: using a keyboard you can take screenshots of the ST2 HiFi Graphical User Interface by pressing the "print screen" key. These screenshots can be copied through the USB Key interface afterwards.



#### Modal dialog window displayed when a USB dongle is detected

When you plug a USB dongle in one of the ST2 HiFi USB connector, a modal window will automatically be opened, offering you all related functionalities:



#### IMPORTANT NOTES

When pulling out your USB dongle from the ST2 HiFi while ensuring its data integrity, please first close the modal dialog window (Close button) and wait until the home page is displayed before unplugging it.

Please be careful when you load your presets from USB dongle : if a slot of the ST2 HiFi is already filled with another preset on the USB dongle, this USB preset will overwrite the processor preset. In order to prevent all incidental overwriting, you can lock the presets you want to protect.



## 4.7 CHANNELS SETTINGS

Channels menu, available through the **Channels** button on the Home page give you access to fine-tuning tools. Meters are also available for every Inputs and Outputs.

### Inputs

You can launch a test noise for each configured channel, for example check if your connections are correctly done and assigned.

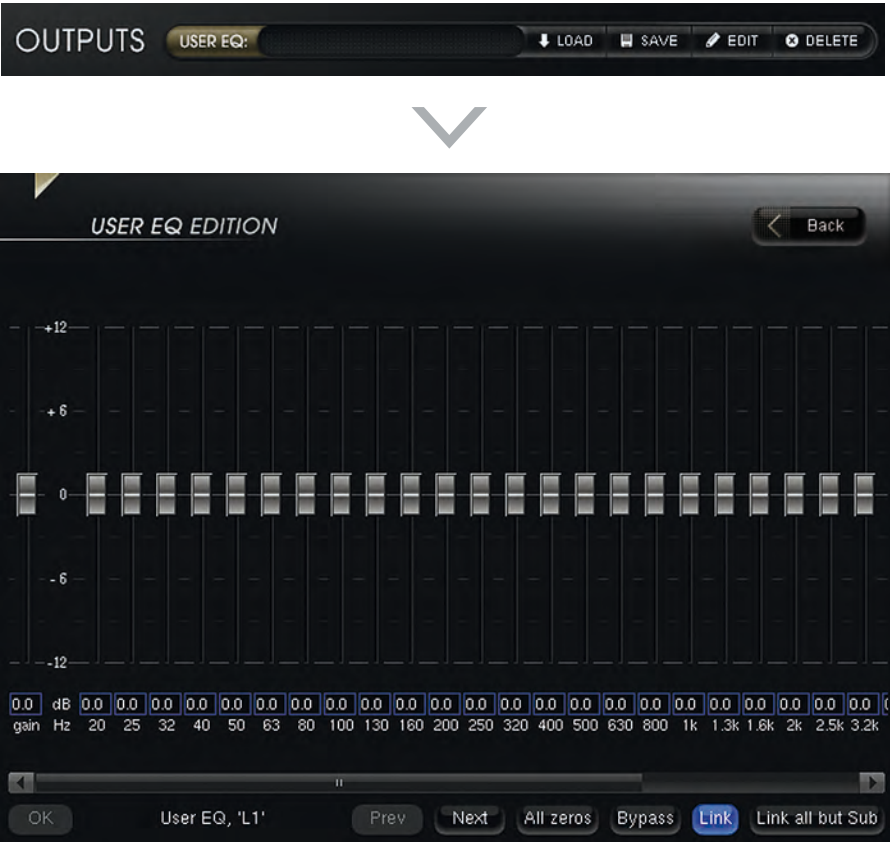
### Outputs

For each channel, you can apply a graphical EQ (Equalizer) which can easily be saved, loaded edited or deleted.



If your preset includes more than two channels, you can browse through channels by pressing the right (>) and left (<) arrows.

To create an EQ, press the *Edit* button on the *User EQ* bar, a new window *User EQ Edition* will open:



This is a standard 31 bands EQ. Press the *Link* button on each channel on which you want your settings to be applied. You can raise and lower gains with sliders within a +12/-12 dB scale. The gain will be reported in the small frame with a blue border under the corresponding slider. If you modify a frequency gain value, it will be highlighted in red. You can also click in the blue bordered gain's frame and directly input the gain's value with a keyboard. A master level gain for the current channel is also available on the left.



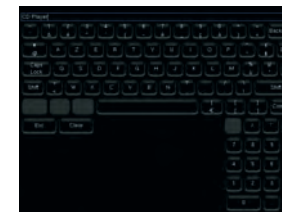
- 1 Master Gain slider
- 2 dB Scale
- 3 Gain per frequency sliders
- 4 Direct Input Gain values for each frequency
- 5 Frequencies



- 1 Scrollbar
- 2 EQ and Channel name
- 3 Change Channels button
- 4 Reset all values to 0
- 5 Bypass EQ
- 6 Link Channels or link all Channels except subwoofers

Once in the **Channels** page you can save your current EQ by pressing **Save**; from this point two choices are offered to you:

- You already have a user EQ saved and want to overwrite it with your new EQ: just click on the name of the EQ to overwrite.
- You want to create a new entry for your EQ: click **Create New**, the virtual keyboard will be displayed in order to let you name your EQ. When finish press **Enter** to confirm and save your EQ, press **Esc** otherwise.



### ① LOAD AN EQ

To load an EQ, click on **Load** button and then on the name of your desired EQ. If you want to disable the User EQ select **(NONE)**. When no user EQ is set or when it is set to **(NONE)** the field where the name is normally displayed is empty.

### ② SAVE button.

### ③ EDIT AN EQ

To edit an existing EQ, you first have to load it (see previous chapter) and then click on the **Edit** button. If you click on **Edit** without having selected an EQ first you will create a new EQ.

### ④ DELETE AN EQ

To delete an EQ, you have to first load it with the **Load** button, then you have to click **Delete**. A confirmation will be asked before effectively deleting the EQ as it is not possible to undelete an EQ.

### ⑤ CREATE A NEW USER EQ entry.

### ⑥ OVERWRITE an existing EQ.



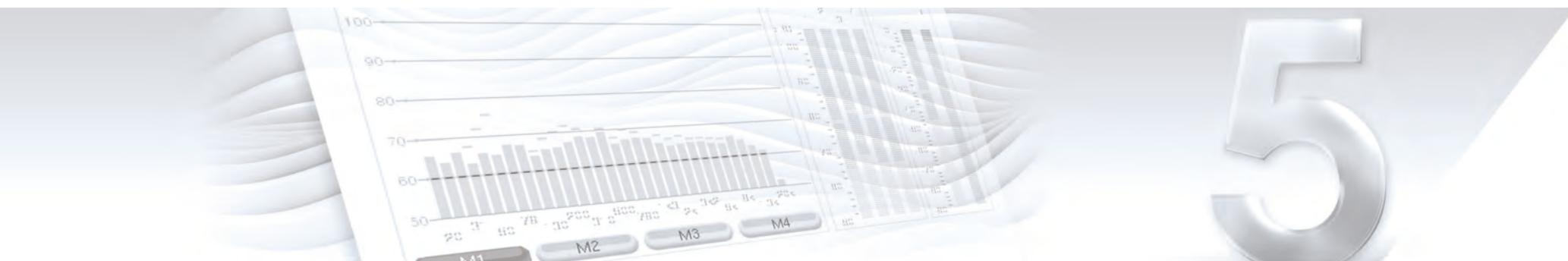


For each Output channel you can:

- ⑦ **MUTE** it with the *Mute* button
- ⑧ **INVERSE THE POLARITY** with the polarity trigger (the polarity is inversed when the trigger is lighted up)
- ⑨ **ADJUST THE LEVEL** by +/-0.5dB step
- ⑩ **ADJUST THE DELAY** by +/-0.5ms. You can not input a negative delay.







# BASIC OPTIMIZER SETUP

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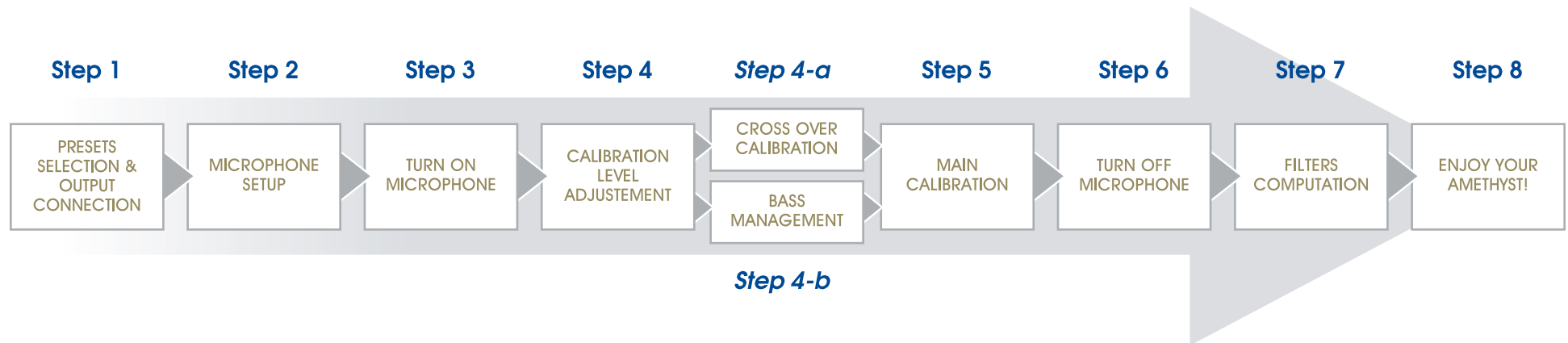
This chapter will help you setting up your system by:

1. Using the "Optimizer" wizard, the step-by-step guide to the unique Trinnov acoustic correction.
2. If you want to go further, you can follow an iterative procedure and modify the main acoustic parameters to efficiently converge to the best results.
3. Manage your presets.

## 5.1 OVERVIEW

The ST2 HiFi includes a step-by-step calibration assistant to help you throughout the calibration process and benefit from the Room/Speaker correction technology Trinnov Optimizer with just a few clicks.

Wizard's steps quick reference :



## 5.2 PRESETS SELECTION & OUTPUT CONNECTIONS

### Step 1 - Presets selection & Output connections

You have 2 choices to make in the wizard's first step:

1. **Select a preset range** where the filters issued from calibration will be saved.  
The ST2 HiFi automatically computes 5 presets at the end of the calibration process, so a slot of 5 presets is required to achieve the wizard.

**IMPORTANT NOTE** If you select already saved memory slots, their content will be erased as soon as you click **Next step**. There is no possible fallback, so please ensure you have saved the presets you want to keep on other slots or on a usb key before proceeding further with the wizard.



2. **Select your speaker layout:**

- **2.0 Mono-amp:** standard stereophonic configuration. You have two main speakers to reproduce all the audible spectrum. Each speaker receive one audio amplification channel.
- **2.0 Active Bi-amp:** specific configuration where the cross-over between channels (intended to separate high frequencies from low frequencies) of your loudspeaker is done directly by the ST2 HiFi and not by the internal speaker crossover. There are 4 channels on total, each requiring a dedicated amplifier.

Active Bi-amp is different from passive Bi-amp or Bi-wiring. Please contact your speakers manufacturer if you have a doubt on your speaker's capacity to handle active bi-amping.

- **2.1 Bass Management:** in this mode, the lowest frequencies from both main speakers are sent to a subwoofer.
- **2.2 Mono Bass Management:** same as 2.1, but two independant subwoofers receives the bass management signal.
- **2.2 Stereo Bass Management:** unlike 2.2 (Mono Bass Management), the signals sent to the subwoofers are different since the left subwoofer receives the low frequencies from the left channel and the right subwoofer the low frequencies from the right channel.

For each configuration, a connexion diagram will be displayed in the Output connections frame. Follow these instructions to connect the ST2 HiFi to your amplifier(s) via balanced or single ended output.

**Choose how to prefix preset names using the Preset base name field.**

Once your configuration is over, click **Next Step**. Note that this action will overwrite the presets you have selected. If you need to check the availability of these memory slots, you can quit the wizard at this step by pressing the **Back** button.

# CONNECTIONS TEMPLATES TO FOLLOW AT THE BEGINNING OF THE CALIBRATION PROCESS

**NOTE** Balanced outputs and single-ended outputs work simultaneously.

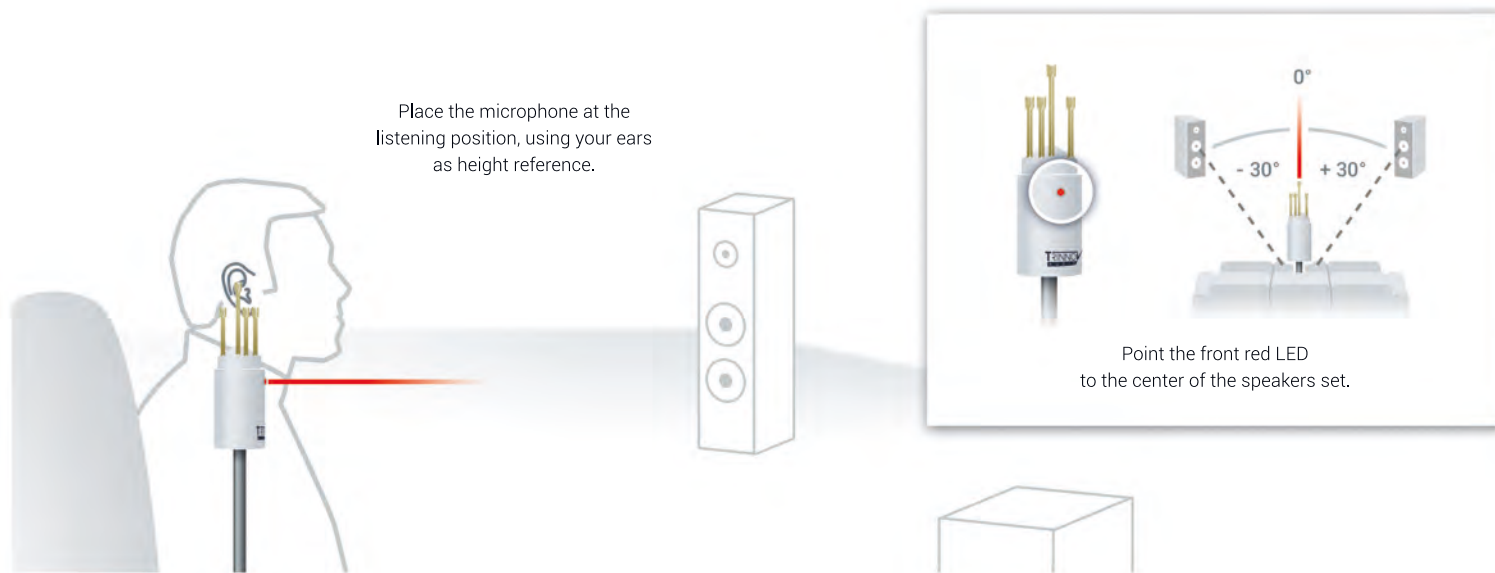
TEMPLATES	BALANCED ANALOG OUTPUTS	SINGLE-ENDED ANALOG OUTPUTS
2.0 MONO-AMP		
2.0 BI-AMP		
2.1 BASS MGMT		
2.2 MONO B-MGMT 2.2 STEREO B-MGMT		



## Step 2 - Microphone Setup

The most critical step of the Optimization procedure is the measurement (or calibration), which first relies on a correct microphone placement.

1. During the calibration procedure, the microphone should be screwed on a microphone or camera stand.
2. The **red LED**/battery level shows the front of the microphone. This end **must be pointed to the front** of the sound stage, typically to the center of the speakers.
3. Place the **microphone at the listening position** using your ears as height reference for the top capsule of the microphone, as shown in the following figure.
4. Make sure there is **no obstacle** between the speaker and the microphone.
5. Make sure there is **no highly acoustically reflecting surface** (leather sofa, glass table, back wall...) close to the microphone, or cover it with a thick blanket.





In this very straightforward step, you just have to choose the model number (ie ID) of your microphone from a dropdown menu. This microphone ID can be found directly at the back of your microphone. If by error your microphone is not referenced in the list, please contact your vendor for an update.

The frame **Microphone installation** is describing precisely how to determine your microphone position from your listening and speakers position. Follow that guide to ensure optimal measurements and results.

The last diagram shows you how to connect the microphone to the ST2 HiFi. As it uses all XLR inputs, you may have to disconnect your sources to do the measurement. Once the Wizard is over and your microphone switched off as instructed, you can safely plug back your sources.

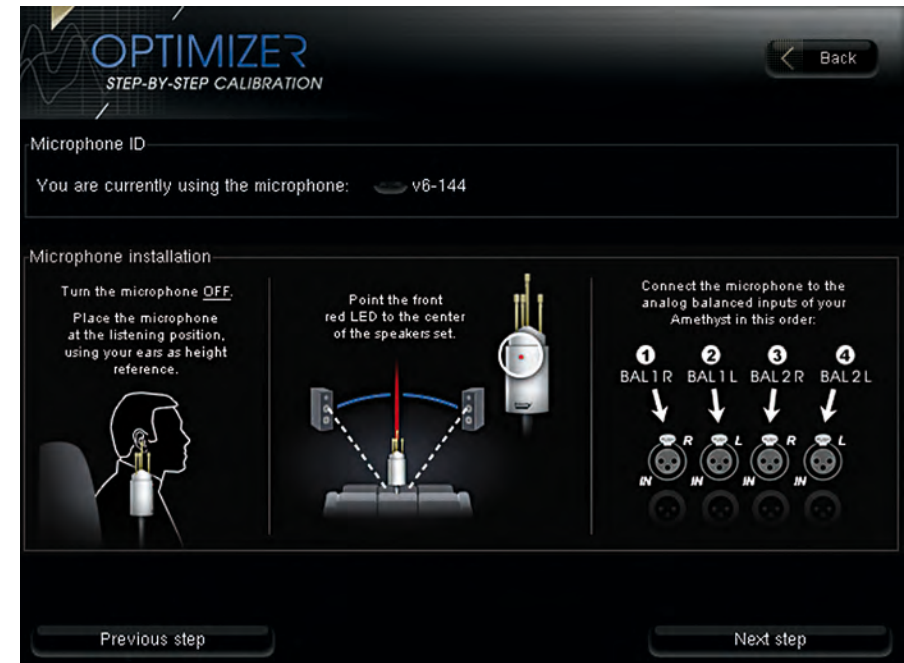
When your microphone setup is done, press **Next step**. Pressing **Previous Step** will lead you to the first wizard step, and **Back** to the Amethyst's setting panel.



**NOTE** Pressing **Back** at this point, and in the following steps, won't restore your presets as they were before entering the wizard: they have been overwritten in step 1.

## Step 3 - Turn on the Microphone

Depending on the model of your microphone, a red light on the front should light up when it's turned on. From this moment, you can press **Next Step** to continue. If you decide to go to the previous step or quit the wizard via the **Back** button, don't forget to switch the microphone off.



## 5.4 CALIBRATION LEVEL ADJUSTMENT

### Step 4 - Calibration Level adjustment

In this step, you will adjust the output level of your speakers to the minimum required level for calibration. To avoid excessive level output, the output is automatically set to -40dB .

Press **Start** in the Calibration level adjustment frame to launch a pink noise. If everything is set up properly, you should see the meters (bargraph) moving in the **Micro** and **Out** frames.

The **Micro** frame represents the level received by the four capsules, and the **Out** frame, the level on the output of the ST2 HiFi.

The **Microphone RTA** frame, where RTA stands for Real Time Analysis, gives you a real time visualisation of the frequency response of your system seen by the microphone.

At last, the **Micro level** frame indicates the current level of the microphone as a Sound Level Meter would. The volume is given in dBC.

In the **Calibration level adjustment** frame, the Start button is used to start and stop the pink noise. The **Auto** button will automatically switch to the next speaker every 5 seconds.

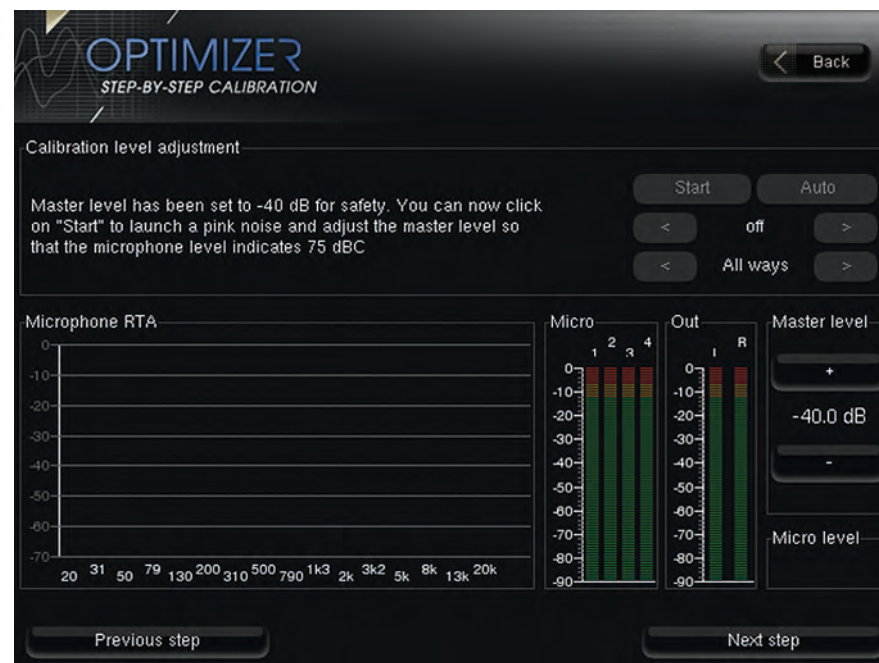
You can use the second row's arrows (< >) to manually switch speakers.

The last row of buttons (< >) is only used with the active biamplification template and allows to check independently the level of the low/high frequencies, or both at the same time.

Here is a little guideline for the level adjustment:

1. Press **Start** to launch the pink noise
2. Check if a signal is played in the **Out** frame and check that a pink noise is played by the first speaker. If not, verify the speakers connections and make sure you have pressed **Start**.
3. Check if a signal is received by the microphone in the **Micro** frame. If not, check that your microphone is correctly plugged in and that it is switched on (see previous chapter on step 2).
4. Adjust the **Master Level** with the plus (+) and minus (-) buttons until the **Micro level** is displayed in green. If the **Micro level** is displayed in blue, the level is too low; press the plus (+) button to raise it. If the level is displayed in red, the level is too high; press the minus (-) button to lower the level until the level is displayed in green.
5. Repeat the operation for every speaker by pressing the > button in the **Calibration level adjustment** frame.

When all the lights are green, press **Next Step** to continue...



#### NOTE

For the subwoofers, you may have to use their own level settings to align them to your main speaker level.

## 5.5 CROSSOVER CALIBRATION

### Step 4-a - Semi-automatic crossovers calibration

The Step 4 displays different settings depending on the selected template:

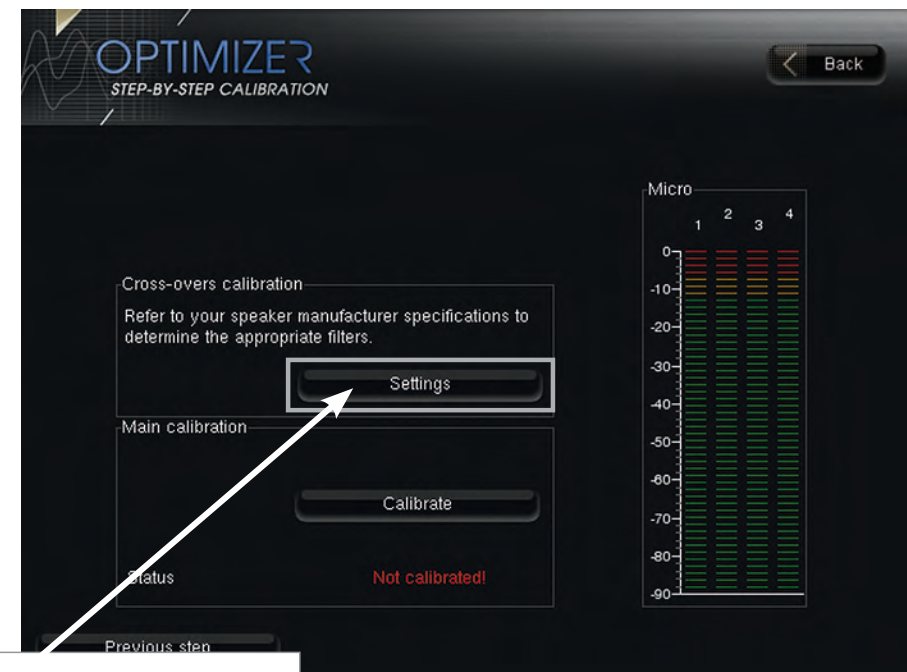
- The Cross-overs calibration is only available with the 2.0 Bi-amp template.
- The Bass management crossover frequency is only available with the 2.1 and 2.2 templates.

**The calibration of bi-amplified speakers is a two-steps procedure:**

- Semi-automatic crossovers calibration.
- Main calibration

**The active crossovers are implemented as follows:**

1. The type of filters and crossover frequencies are set manually.
2. Levels, Delays and polarity of each driver are determined automatically by calibrating separately each speaker.



Click the **Settings** button in the **Cross-overs calibration** section to display the cross-overs settings and calibration page.

## Cross-overs settings page



This page displays the left (shown by default) and right speakers cross-overs settings as horizontal tabs.

The left and right speakers are linked by default (the link button is blue highlighted in both tabs). Every manual change is therefore applied on both speakers.

Please refer to your speaker's manufacturer specifications to determine the appropriate filters:

- Low / High cross-over frequencies (default 2000 Hz)
- Filter type (default 4<sup>th</sup> order Linkwitz-Riley). The following filters are available:
  - 2<sup>nd</sup> and 4<sup>th</sup> order Linkwitz-Riley
  - 2<sup>nd</sup>, 3<sup>rd</sup> and 4<sup>th</sup> order Bessel
  - 2<sup>nd</sup>, 3<sup>rd</sup> and 4<sup>th</sup> order Butterworth
- Select the type of high-pass or/and low-pass filters with the "<" and ">" buttons.
- The cut-off frequency of each filter is set by sliding the scrollbar or by using the arrows.
- Two additional filters are available under the name of **Constant-directivity horn EQ**.

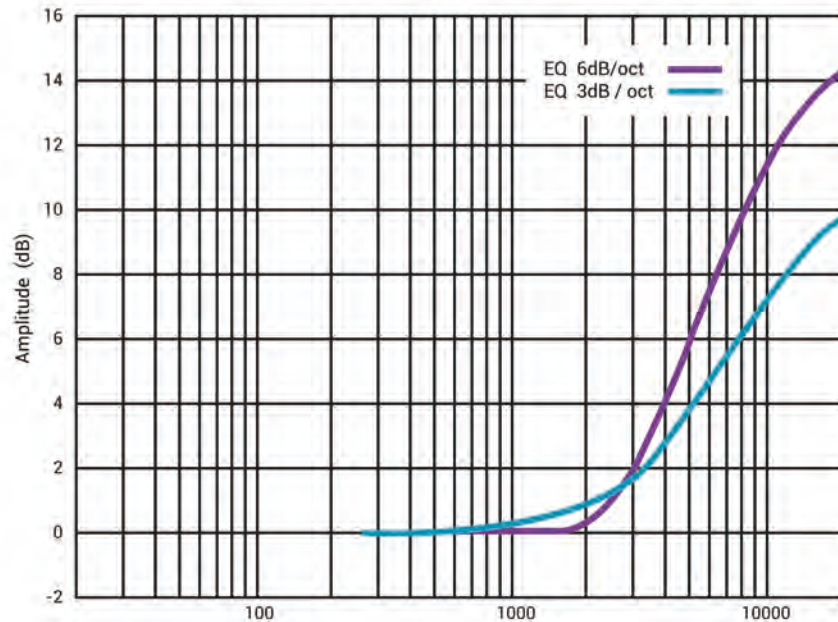
The purpose of such filters is to boost the high frequencies for speaker that use constant-directivity horns.

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 3 or 6db/oct, from about 3 kHz.

The **Apply** button is highlighted as soon as a parameter is modified and is used to compute and load the new settings. Once compute is finished, filters are applied to the outputs.

If change is unwanted, press the **Cancel** button.

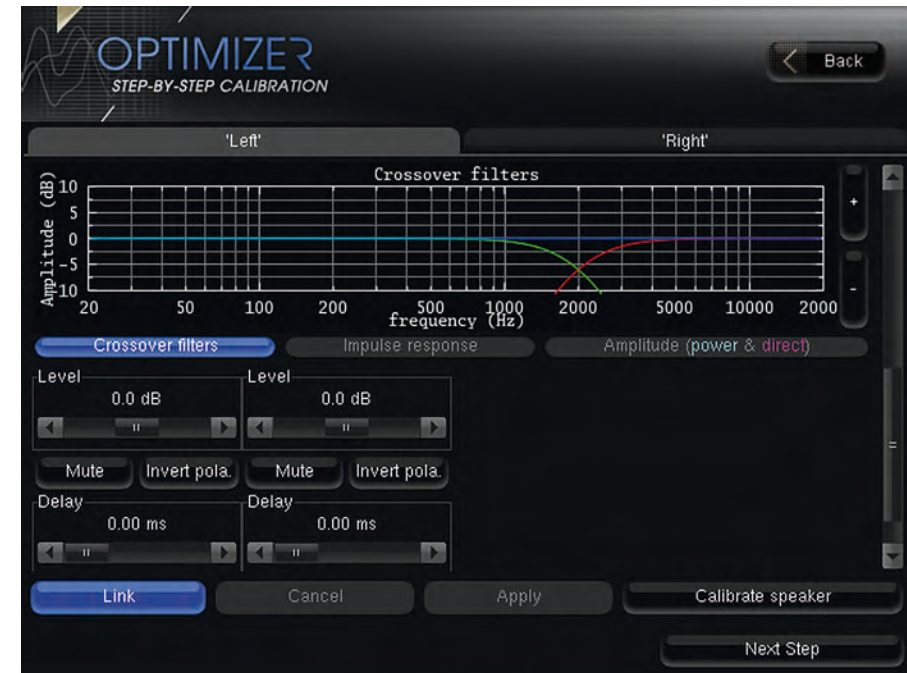
"Constant-directivity horn" EQ filters



**NOTE**

The calibration of multi-way speakers usually requires high output level.

Crossover filters representation



Scroll down to display the crossover filter representation.

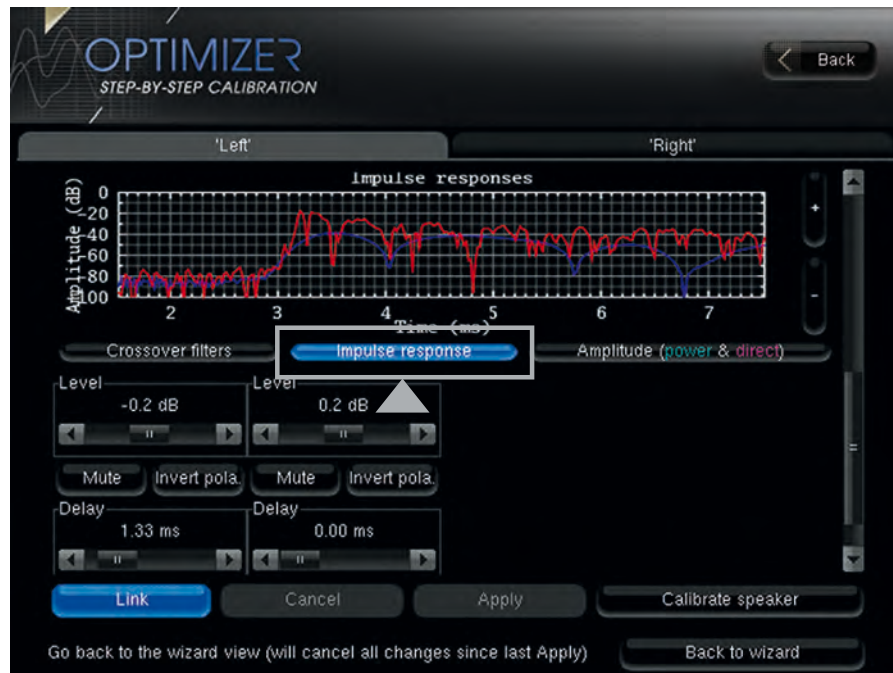
- The green curve represents the theoretical low-pass filter response.
- The red curve represents the theoretical high-pass filter response.
- The blue curve represents the theoretical resulting response of the speaker.

Press the **Calibrate speaker** button to automatically determine levels, polarities and delays of the selected speaker and follow onscreen instructions. A test signal (MLS) is sent to each driver.

The following graphical representations are available after calibration.

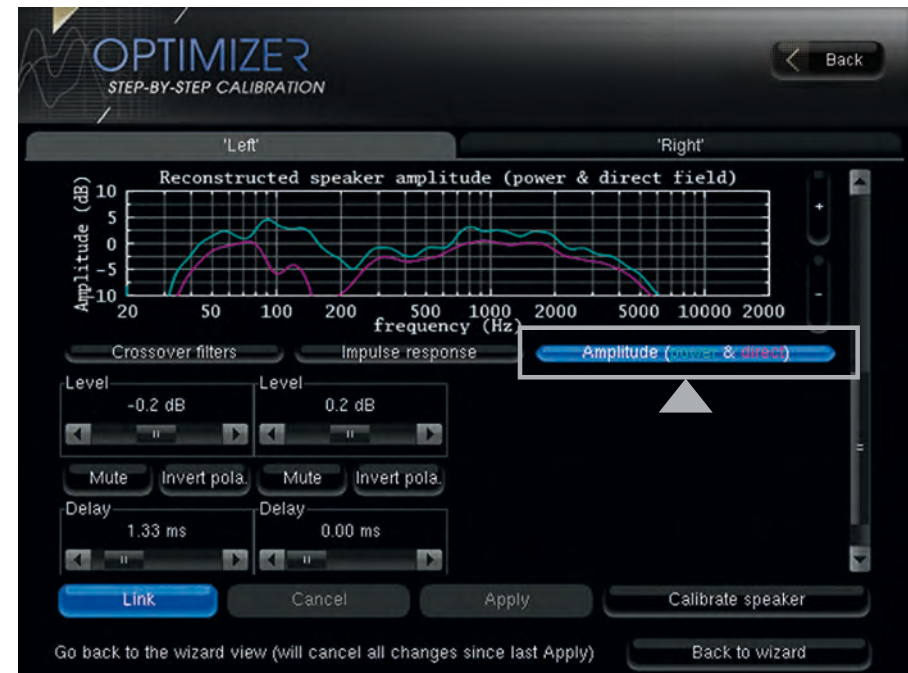


Crossovers Impulse response after calibration



The **Impulse response** graph shows the measured impulse response of each driver and allow to check whether they are correctly synchronized or not.

Amplitude (power & direct) response after calibration



The **Amplitude (power & direct)** graph shows the recombined amplitude response of the speaker. You can see whether the combination of the drivers is consistent, 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.

You can then select the **Right** speaker tab to calibrate the other speaker. Then press the **Back to wizard** button to go back to the main calibration page.

## 5.6 SPEAKER CALIBRATION

### Step 4-b - Bass Management setup

The Bass management crossover frequency window is displayed with 2.1 and 2.2 Bass management templates.

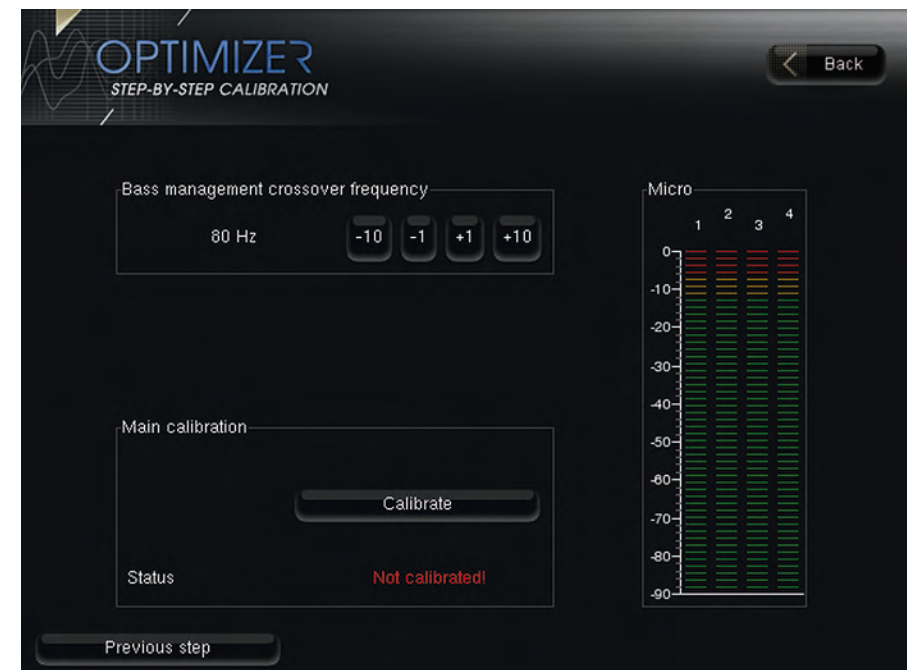
Low frequencies below the bass management crossover frequency will be filtered and sent to the subwoofer(s).

Press the **-10/+10** and **-1/+1** buttons to respectively decrease or increase the crossover frequency by 10 and 1dB.



#### IMPORTANT

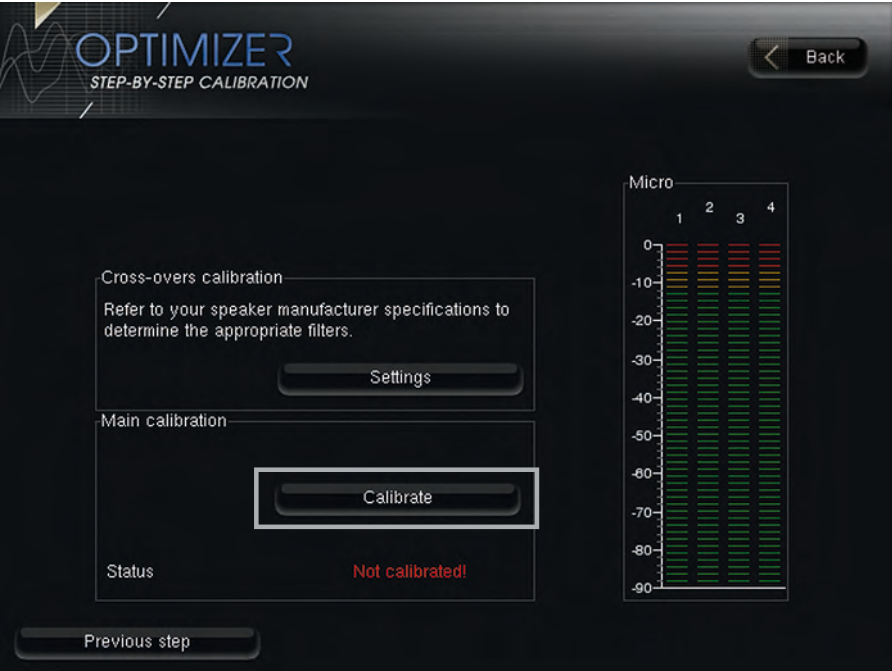
The Bass management is not taken into account during calibration. It can therefore be changed at any time.



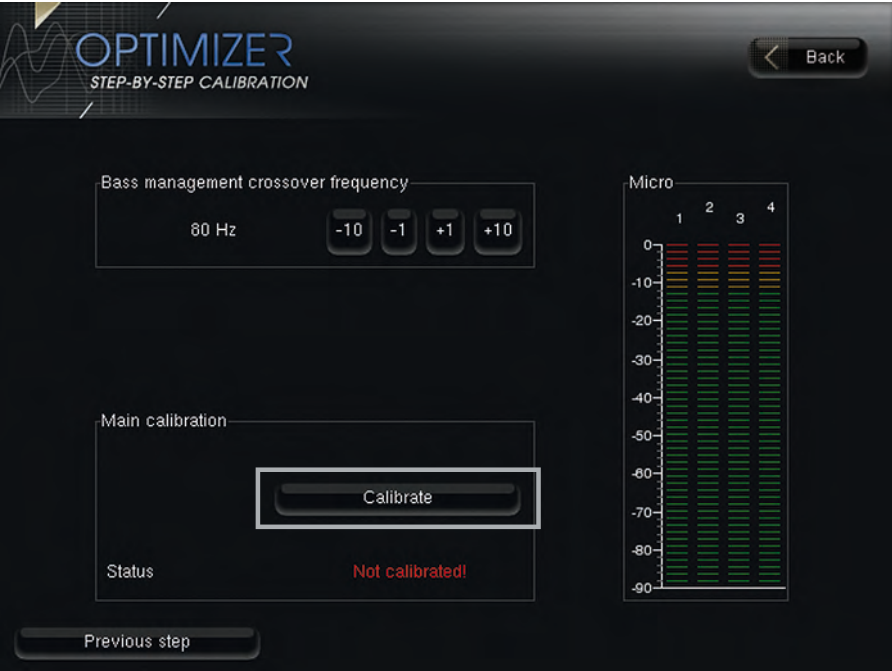


# Step 5 - Main Calibration

If you are coming from step 4-A after the cross-over calibration, press the *Calibrate* button in the *Main Calibration* frame to launch the speaker calibration sequence.



If you are coming from step 4-B after having setup your bass management cross-over frequency press the *Calibrate* button in the *Main Calibration* frame to launch the speaker calibration sequence.



Pressing the **Calibrate** button initiates a series of 3 MLS (Maximum Length Sequence, Calibration signal) minimum, that will be played and measured on each speaker (subwoofers included). The **Status** label will display information on the current speaker being calibrated. The **Micro** frame indicates the microphone input levels during calibration. If a problem occurs, a warning message will be displayed in orange under the **Status** label.

Warning messages can occurs:

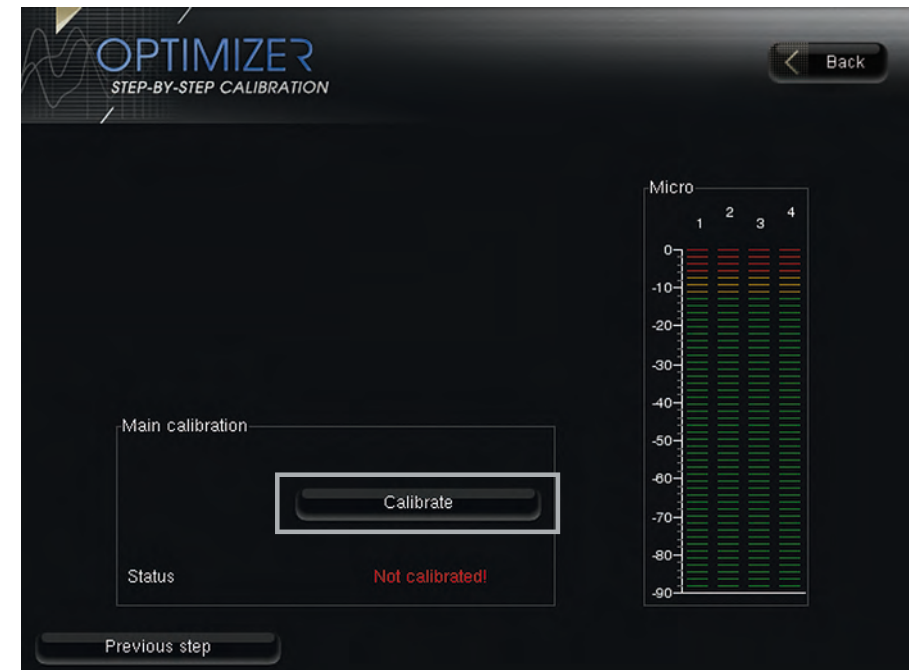
***Crest factor too low for capsules...:***

Check that your microphone is switched on and that your level meets the calibration requirements.

***Unable to determine position... or Unstable position...:***

Check that nothing stands in the acoustic path between the speaker and the microphone. If it is not sufficient, try to calibrate at a higher level or change the microphone's position by a few centimeters.

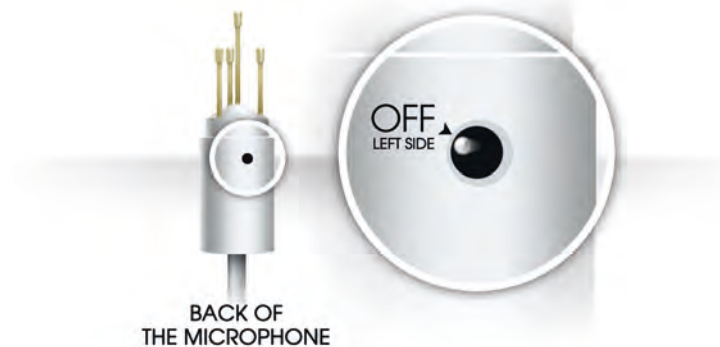
If you are coming from step 4 press directly the **Calibrate** button in the **Main Calibration** frame to launch the speaker calibration sequence.



## 5.7 AUTOMATIC PRESETS

### Step 6 - Turn off Microphone

Turn off the microphone, the front light of the microphone will fade off. Press the **Next Step** button to continue.



### Step 7 - Filters computation

The ST2 HIFI is now computing 5 filters/presets. The computation can take 2 to 5 minutes. A progress bar indicates remaining computation time.

The 5 presets tend to cover a large spectrum of listening types; here is a short description for each of them :

#### COMFORT

You like the idea of having a system which sticks to the truth, but sometimes a little more roundness on harsh CD, a more tolerant listening on those badly mixed but lovely materials is needed.

#### NATURAL

You like the sound of your speakers, you want to correct the phase, the reflections etc. but you don't want to lose the sound signature of your speakers.

#### NEUTRAL

Could be considered as the default preset, it is designed to offer a flat response from bottom to high: listen to the music, not your system.

#### PRECISION

Is it the second or the third violin who missed that note ?  
If you are eager to know, switch to the Precision preset.

#### MONITORING

Audio, sound and music are your work on a daily basis. You want and need to hear everything, including that irritable little defect in the recording that every one else would try to mask. The monitoring preset is a magnifying glass for your records.

When the computation is over, the system will automatically go to the final step.

## Step 8 - Enjoy your ST2 HiFi!

---

Congratulations the Wizard is over! You can now press the **Enjoy your ST2 HiFi** button and compare the five presets described in the previous page.

Note that your system is muted for a security matter, so it is the good time to unplug your microphone and plug your favorite source.



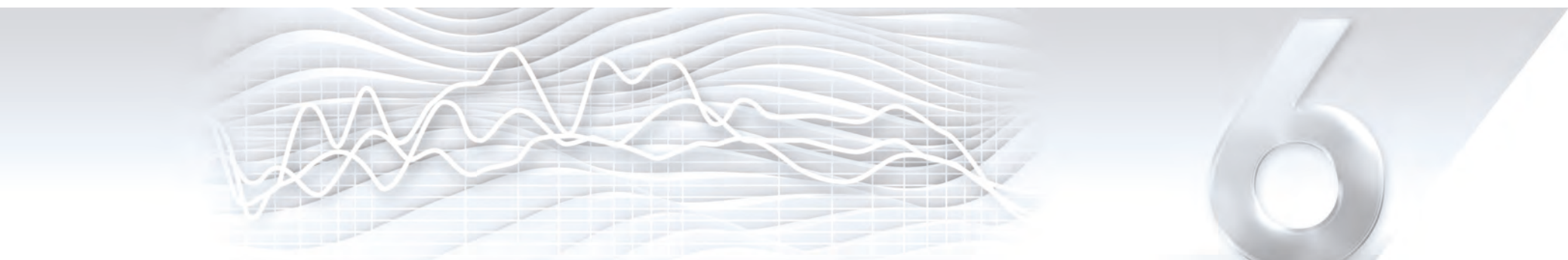
The presets are directly available in the home page via the **Preset** button.

You can also fine-tune automatic settings via the **Channels** page or dive into the **Advanced parameters** (via the **Settings** page).

The background features a large, semi-transparent gear centered on the page. From the top of the gear, several bright, white, three-dimensional rectangular beams radiate upwards, resembling rays of light or data. The overall color palette is light blue and white, with a dark grey horizontal band across the middle.

# Optimizer

ADVANCED



# ADVANCED OPTIMIZER SETUP

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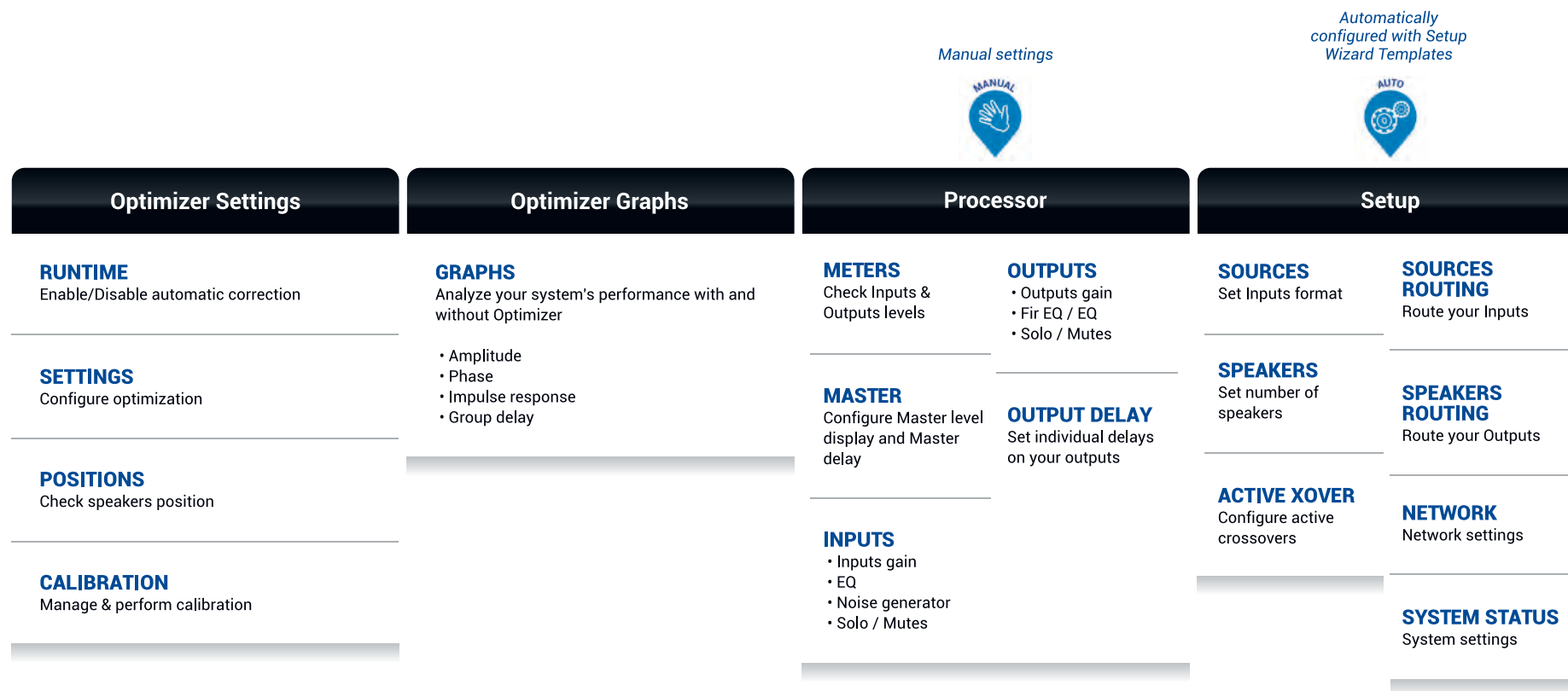
## 6.1 TRINNOV CERTIFIED INSTALLERS

Depending on how far you want to dig into its features, the Optimizer is either a straightforward and easy-to-use automatic compensation system or an incredibly flexible and powerful tool, including numerous advanced filter parameters, target curves and manual EQ's, all being recomputable on the fly.

By following an iterative installation procedure, the Optimizer achieves results beyond expectations in record time but requires a good understanding of the audio and acoustic basics and more especially experience to clearly identify a problem and therefore use a relevant method to solve it. That's why we recommend at least to consider certified installers services.

The ST2 HiFi provides multiple tools to adjust the presets automatically generated by the Wizard. For a fresh start, load the preset you want to adjust and follow the iterative procedure (see chapter 6.3).

## 6.2 ARBORESCENCE OF THE ADVANCED SETTINGS MENU





## 6.3 ITERATIVE PROCEDURE

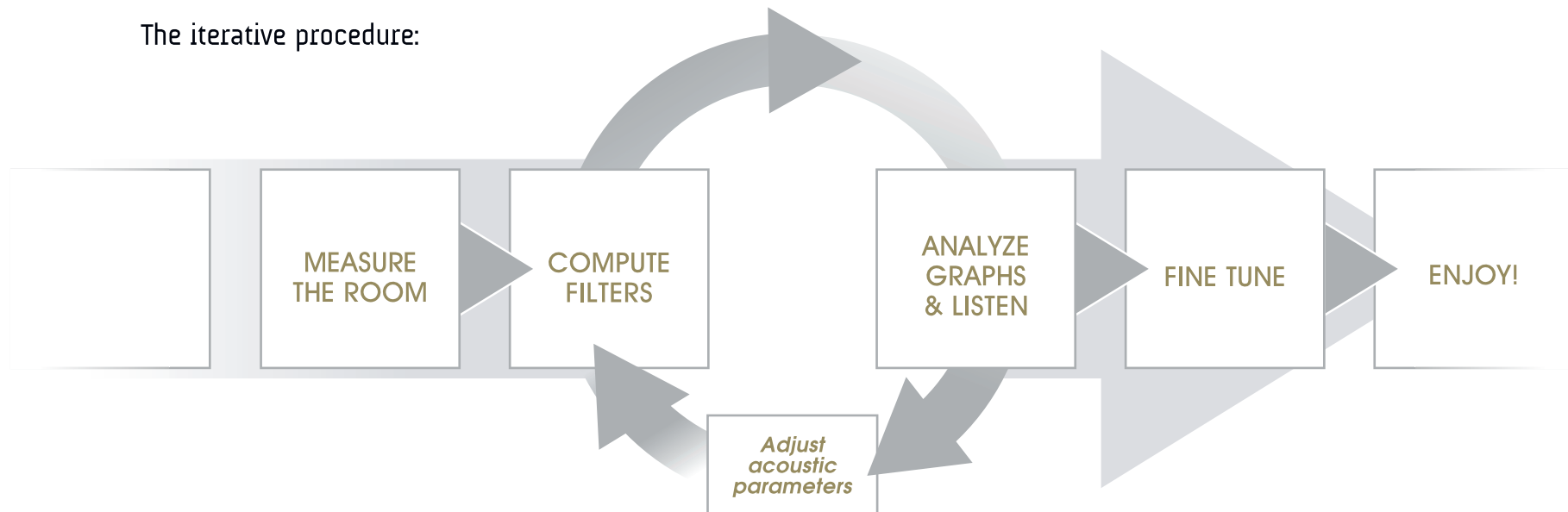
Whether it is using the Optimizer Settings of the Setup Wizard, the Target Curve or other parameters described in this chapter, the method to converge to the best results efficiently remains the same.

1. Establish a correlation between the Optimizer graphs and the listening experience.
2. Adjust acoustic parameters.
3. Compute the filters.

The main and most efficient parameters are available in the *Advanced settings*:

- Optimize Modes
- Maximum Boost/Attenuation
- Target Curve
- Limiter curve

The iterative procedure:

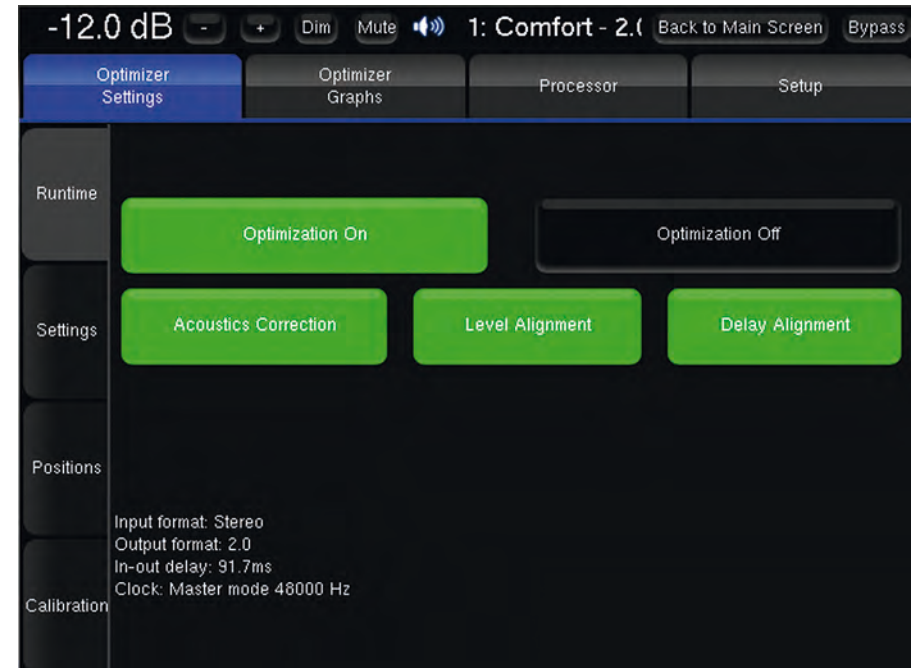


Settings screen



The **Advanced settings** menu can be accessed from the **Settings** screen.

Advanced Settings screen



The page automatically displayed is the **Runtime** menu, used to switch the different corrections of the **Optimizer On** and **Off**:

- **Acoustics correction**: amplitude and phase response corrections
- **Level Alignment**: level alignment of all speakers
- **Delay Alignment**: time alignment of all speakers

The **Optimization Off** button allows you to disable all corrections to understand the effect of the Optimizer. The **Bypass** button located on the top-right corner of the interface also allows you to switch all corrections off and remains visible regardless of the selected menu.

## 6.4

## OPTIMIZER GRAPHS

Reading and interpreting the Optimizer Graphs require some acoustic knowledge and is therefore dedicated to advanced users or certified installers.

The Optimizer Graphs include a **display** (1) and a **config** (2) area.

The **display** area is organized in:

- Tabs & Subtabs
- Graphs
- Frames

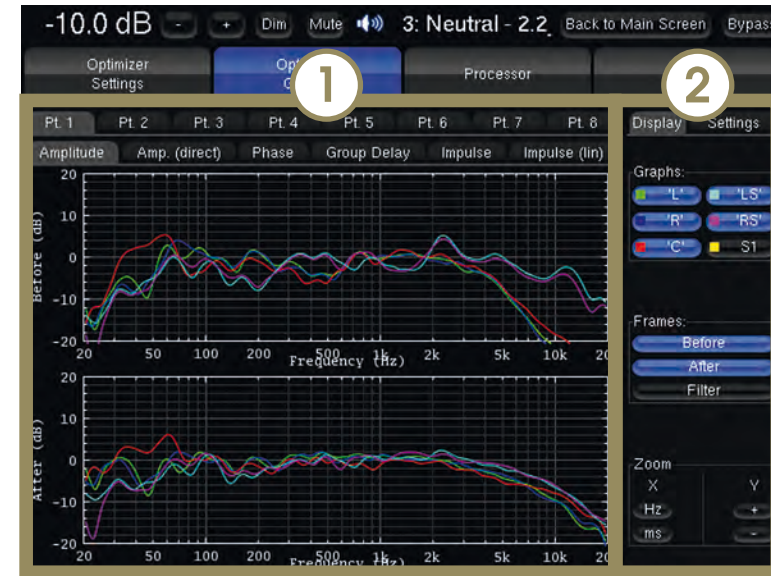
The **config** area is split in two tabs:

- **Display:** includes filters to decide which available informations should be displayed as Graphs / Frames and allows to adjust different zoom options.
- **Settings:** select what information should be displayed as graphs/frame/tabs/subtabs.

The **Zoom options** allow to display specific frequency range, to adjust the amplitude and time scales. The available zoom options depend on the displayed graphs.

The default layout is as follows :

- Tab: measurement points
- Subtab: responses
- Graphs: speakers
- Frames Before/After/Filter way graphs are organized



For every speaker and every measurement points, the following responses can be displayed:

**Amplitude** the amplitude response is the most commonly used and the most understandable graph available in the Optimizer. It represents the amplitude versus frequency response across the 20Hz-20kHz frequency range. It clearly shows dips and boosts which, with a trained ear, can easily be correlated with the listening experience and related to the global perceptive tonal balance of the system.

**Amp. (Direct)** this representation shows the amplitude response of the direct sound and early reflections. This representation better reveals acoustical problems (cross-over filters problems, room modes, reflections...) as well as the direct sound response.

**Phase** the phase response of a speaker shows the phase rotation versus frequency. Many phase rotations and relative responses offset between speakers indicate a lack of definition and an unstable stereo image.

**Group Delay** the group delay is also a time-related representation and displays relative time arrival of all frequencies at the listening point.

**Impulse response** it shows the main impulse of the speaker versus time and reveals the behaviour of the speaker depending on the room. The most sensible information is the amplitude of the main impulse versus the amplitude of the first reflections.

## Computation & Graphs update

The parameters described in the following chapters require a computation to be applied.

As soon as a parameter is modified, the **Apply changes** button will highlight.

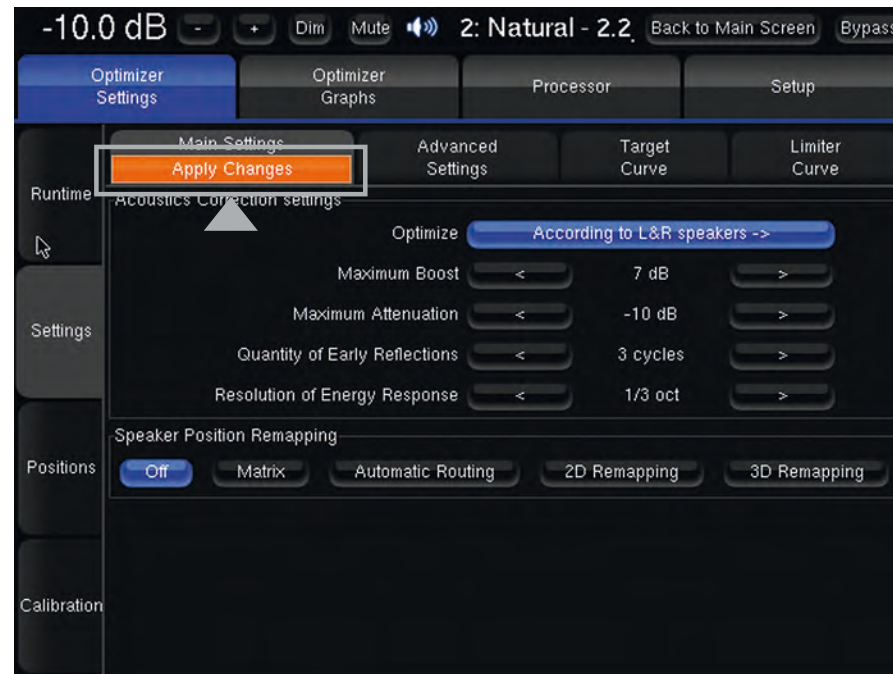
Pushing this button initiates a new calculation.

On modification of an Optimizer parameter and computation of the new compensation filters, the **Optimizer graphs response (After)** will be updated to show the new response and allow further correlation with the new listening experience.

After each parameter change, you can make new iterations and incrementally save new parameters as different presets to be able to recall each change with just a click.

### NOTE

Despite the fact that the after correction responses are calculated instead of measured responses, re-measuring the Optimized system has always proved the accuracy of this calculation.



## 6.5 OPTIMIZER MODES

### Optimize Modes



The **Optimize** setting allows to select the acoustic correction mode.

### Amplitude + Phase

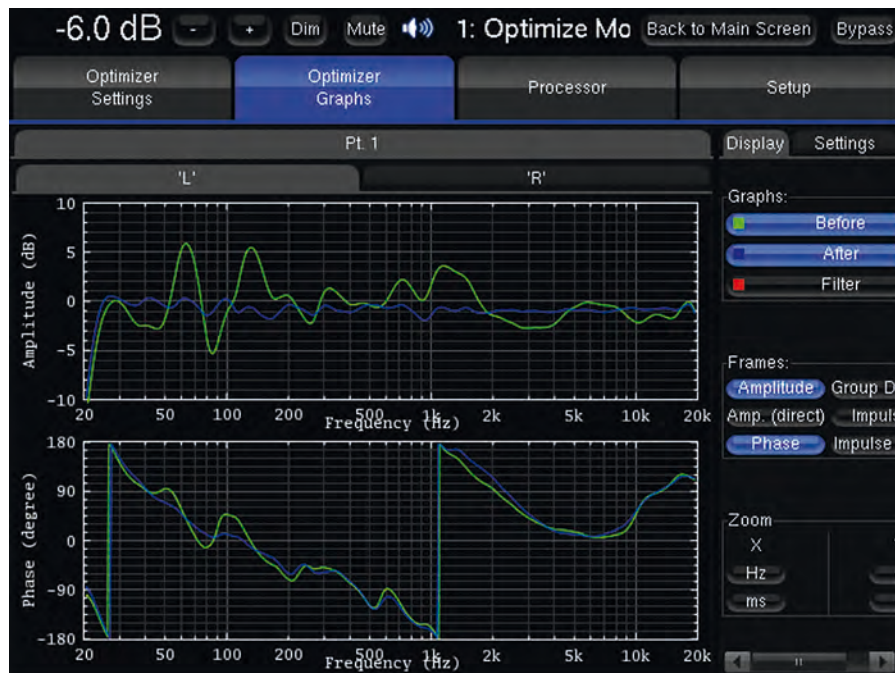


The **Amplitude + Phase** mode is used in every factory presets except the natural preset.

It is the default behaviour of the Optimizer and implies amplitude AND phase corrections to achieve better transients and to obtain the best sound staging.

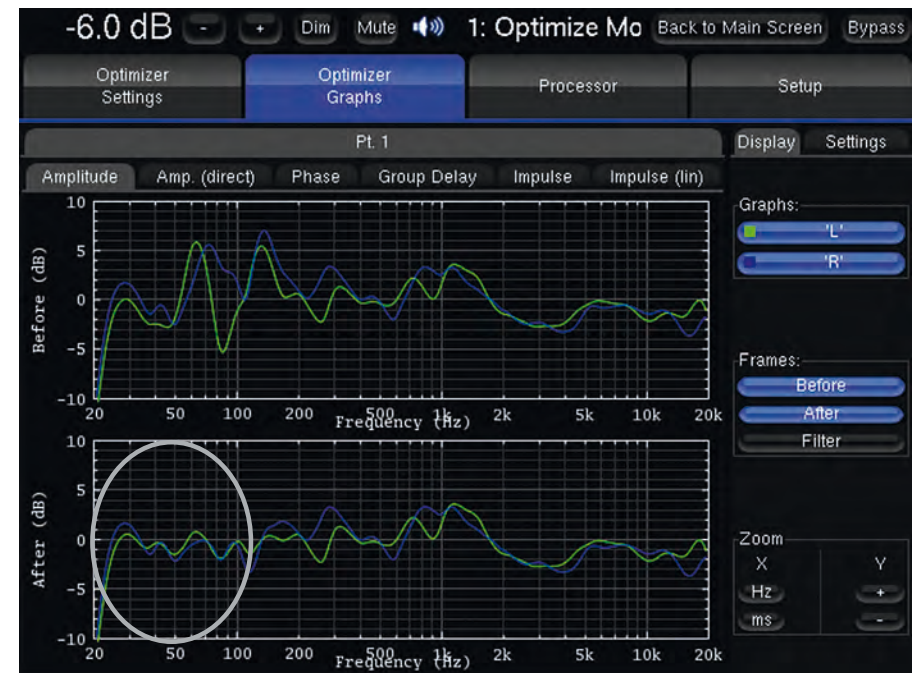


## Amplitude only



The **Amplitude mode only** provides amplitude responses corrections only (thus not the phase corrections) and results in less spatial accuracy and stability.

## Low Range only



The **Low range only** mode does not provide phase response correction and only compensate the amplitude response up to 150Hz (Default)..

This action limit of this mode can be defined in the **Advanced settings** page.

## IIR Filters maximal frequency

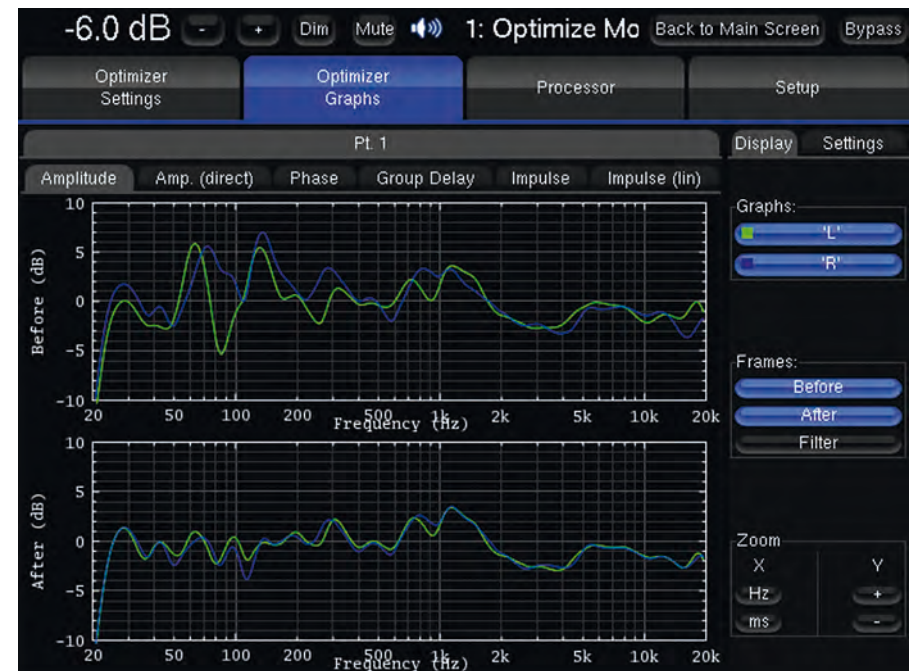


IIR filters are dedicated to low range amplitude accidents.

The **IIR filters maximum frequency** can be modified and requires a computation.

The parameter **Number of IIR filters** should be increased with the **IIR filter maximal frequency** to maintain sufficient resolution and achieve efficient correction.

## According to L&R speakers mode



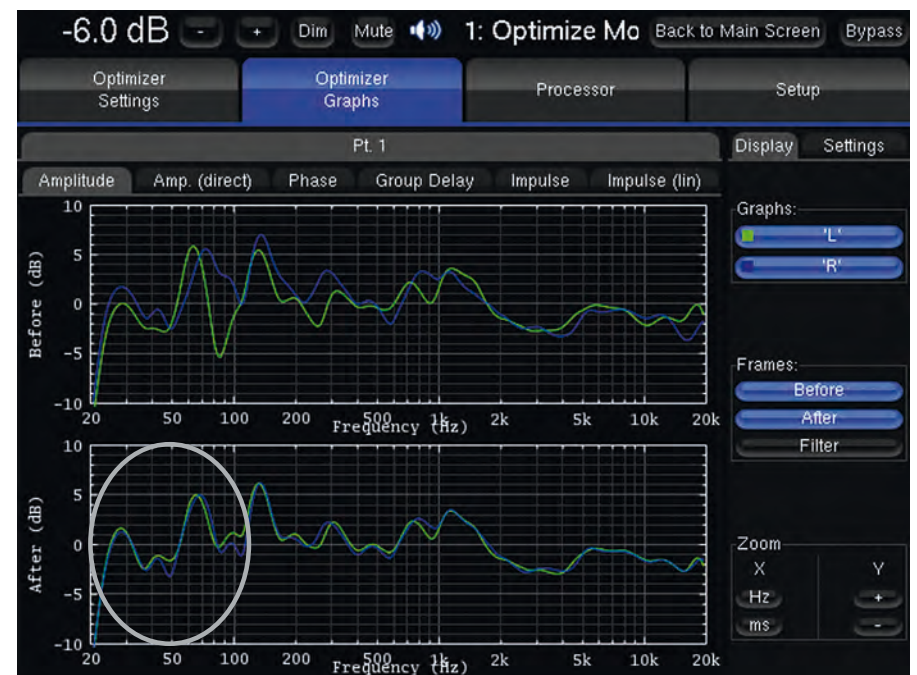
The **According to L&R speakers** mode takes the average response of the left and right loudspeakers as reference and automatically uses it as target curve.

This mode compensates the phase response of the speakers and the amplitude response below 150 Hz.

The range of action of the bass correction can be modified the same way as for the **Low Range only** mode.



## According to L&R speakers



The **According to L&R speakers** has its own specific parameters :

Switching the **Processing on LR target** from **IIR Only** to **None** will disable the bass correction.

Disabling the **Align on L&R target** option will result in a flat target curve for mid/high frequencies.

**Optimizer phase Off/On** determines whether or not the phase correction is applied in **According to L&R Optimizer** mode.

## 6.6 OPTIMIZER SETTINGS

### Maximum Boost & Attenuation



The **Maximum Boost** and **Maximum Attenuation** parameters are used to avoid over-correction, unless the amplification and the speakers have been sized consequently.

Displaying the **Filter response** should highlight frequency regions where the maximum boost/attenuation limiters are in action.



#### IMPORTANT

Excessive boost can cause distortion and damage speakers and should be avoided unless the audio diffusion system (amplification and speakers) has sufficient headroom.



These parameters affect the bandwidth.

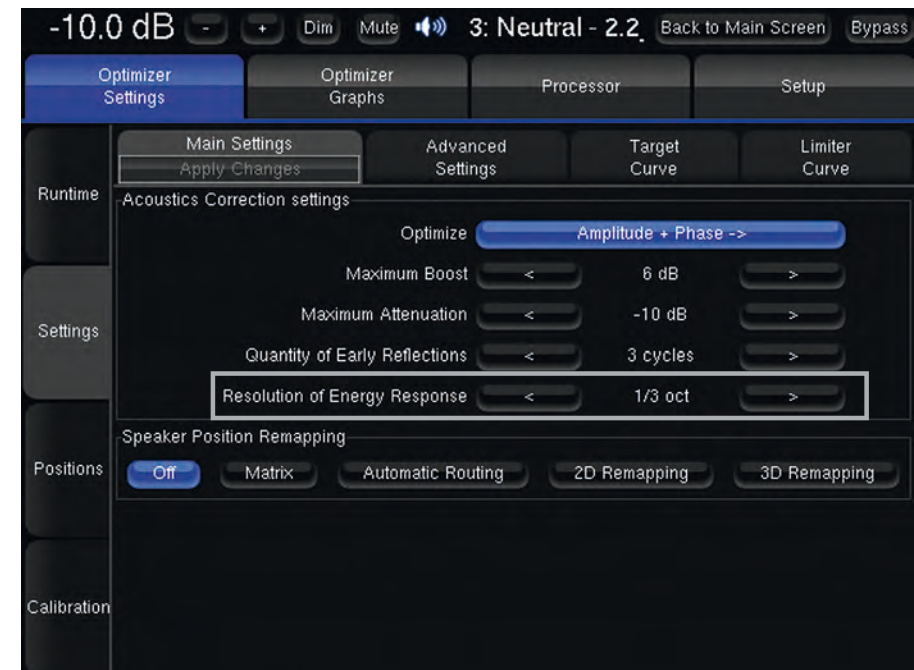
## Quantity of Early Reflections



The parameter **Quantity of Early Reflections** determines the amount of early reflections taken into account for acoustic compensation.

3 cycles is the default settings.

## Resolution of the Energy response



**Resolution of the Energy response** allows to set the degree of accuracy with which the automatic correction algorithm will take the energy response of the room into account.

1/3 oct is the default settings.

The minimum resolution is 1/3<sup>rd</sup> octave and the maximum resolution is 1/24<sup>th</sup> octave.

## 6.7 TARGET CURVE / LIMITER CURVE

### Target Curve

The **Target Curve** is certainly the most powerful and the easiest fine-tuning tool within the advanced settings to adjust the automatic calibration and achieve a personalized tonal balance.

The correlation between graphical representation and listening experience can be made by comparing the before and after amplitude responses graphs while bypassing the correction using the **Bypass** button located in the upper part of the GUI.

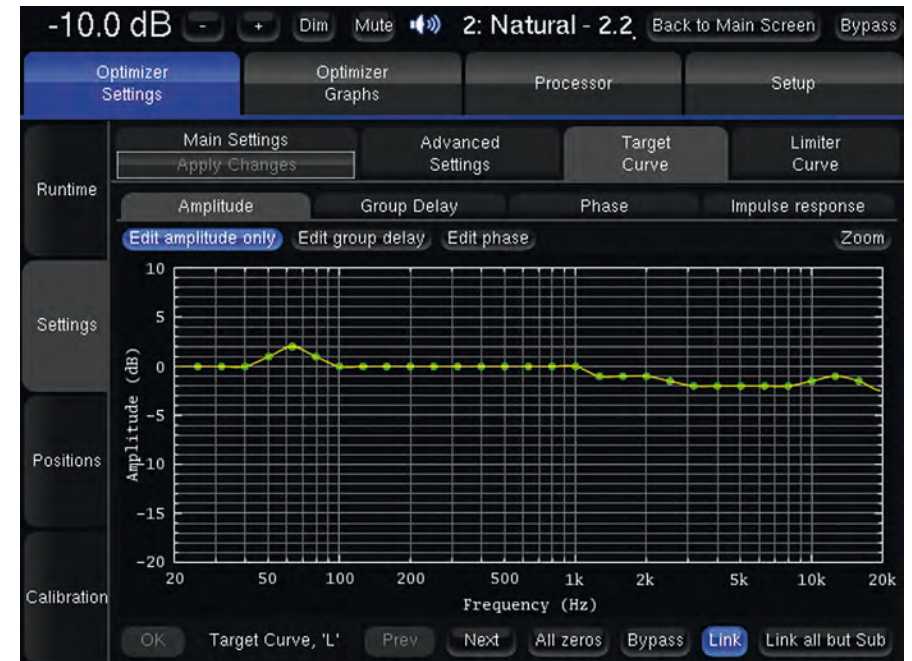
The **Target Curve** tool is located in the **Optimizer Settings > Settings > Target Curve > Amplitude** page (see below).

The required curve can easily be edited:

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

The target curve can be edited with mouse or gesture on touch screen. However, the most efficient way to edit the **Target Curve** is to use a keyboard, be it a physical or a virtual keyboard within a VNC client.

- Right and left keys allow to switch between editable points.
- Up and down keys shall be used to modify the amplitude of the target curve for the selected frequency with 0.1dB steps.
- PageUp and PageDown keys shall be used to modify the amplitude with 0.5dB steps.



The **All zeros** button resets the curve.

The **Link** highlighted button indicates that the editing is performed for a group of speakers.

You can use the **Next** and **Previous** buttons and the **Link** button to exclude speakers from this editing group and apply speaker-specific correction.

The Target Curve can only be achieved if the required correction comply within the maximum boost and attenuation values.

#### Please respect the following procedure:

1. Pay attention to the amplitude scale while editing the target curve.
2. Preferably apply smooth corrections.
3. Press the **Apply Changes** button to recompute the filters.

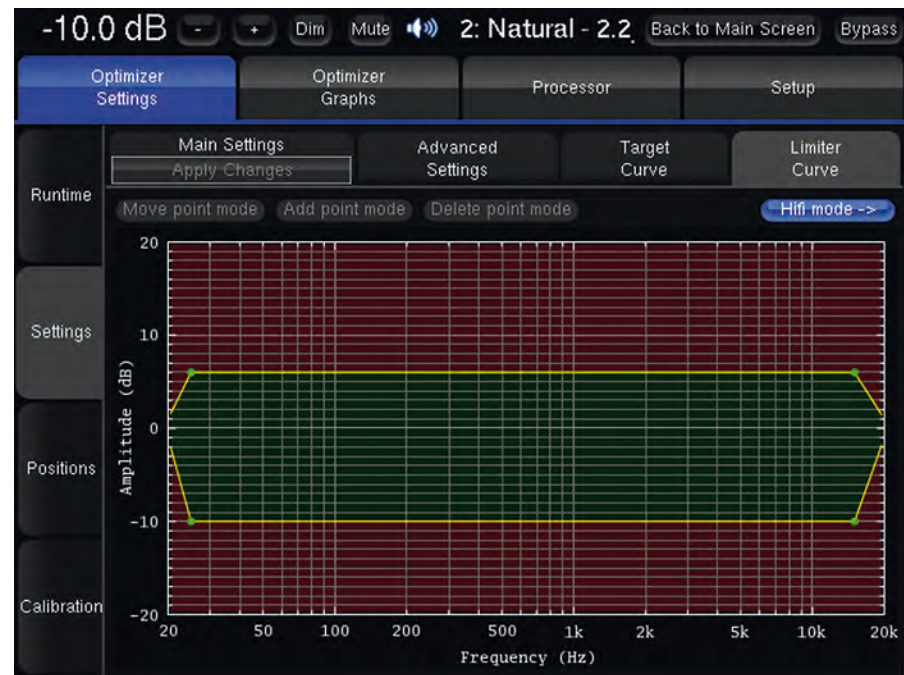


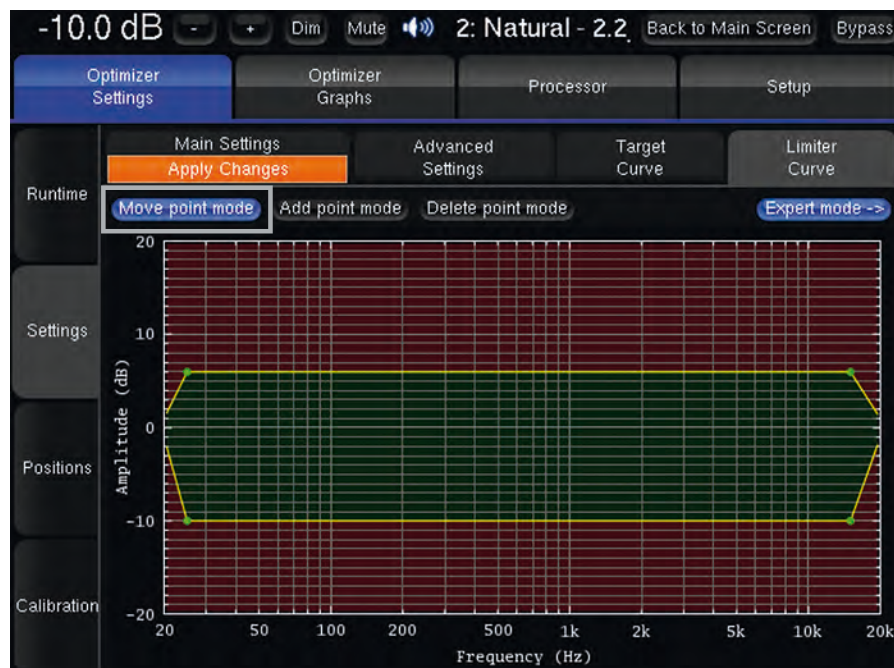
## Limiter Curve

The **Limiter Curve** is a frequency-dependant version of the Maximum boost/attenuation parameters.

The **default Limiter curve** limits boost and attenuation in the extreme frequencies.

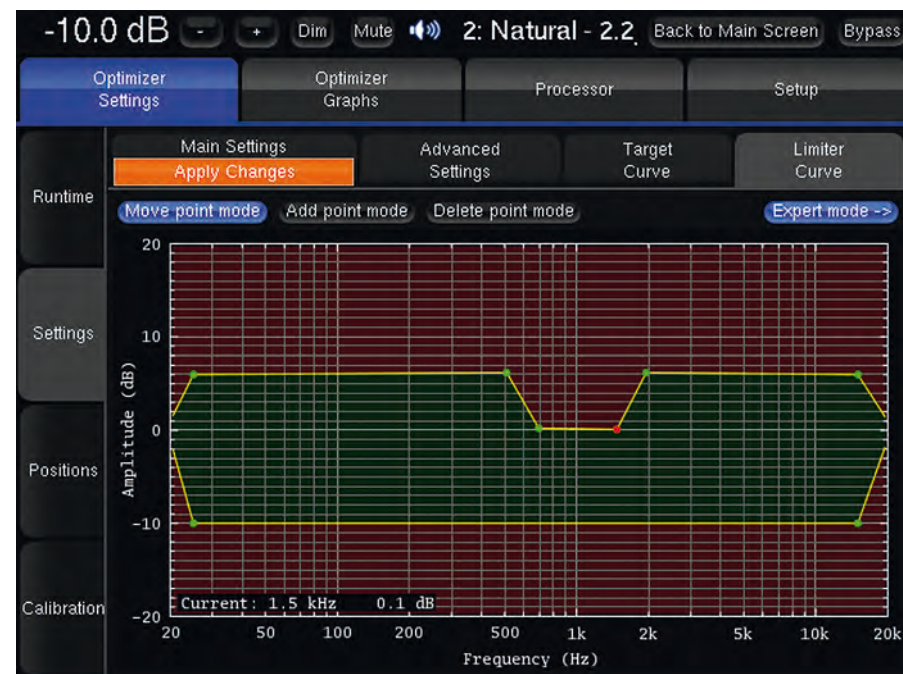
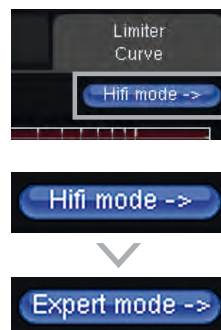
The **Limiter curve** shows both the maximum boost and maximum attenuation allowed on a graphs with the amplitude in dB as ordinate and the frequency in Hertz as abscissa.





Pushing the HiFi mode switch the Limiter curve to **Expert mode** and enables three edit functions:

- **Move point** mode: click and shift an edit point in the vertical/horizontal direction to respectively adjust the amplitude/frequency allowed for the filter.
- **Add point** mode: click on the boost or attenuation curve at the required frequency to add an edit point.
- **Delete point** mode: click on an edit point to delete it.



In this example, the compensation filter will not be allowed to boost the amplitude response by more than 0.1 dB between approximately 700 and 1500 Hz.

It is recommended to apply smooth amplitude corrections with soft slopes.

The processing implies a latency, which depends on the following parameters :

- Sampling Rate
- Buffer size
- Advanced audio processing parameters

MAIN PRINCIPLE: The latency increases with the buffer size whereas it decreases with the sampling rate.

**IMPORTANT**

Depending on the requirements of the application, you can try to reduce the buffer size to lower the latency but please be aware that it might result in CPU overloads, sync loss and eventually loud digital clicks, especially when processing 192kHz audio on multiple channels.





# MULTIPOINT CALIBRATION

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## 7.1 PRINCIPLES

The purpose of multipoint calibration is not only to optimize a wider area but also to gather more information from measurements and increase reliability by taking differences occurring in the listening area into consideration. However, taking different measurements into account is irrelevant if the listening area considered is too much disparate acoustically. In other words, trying to generate compensation filters that works for a too large area would work but results in a lower improvement compared to the result provided at the listening position with a single measurement.

Choosing relevant positions for multipoint calibration is highly critical and will determine whether the target, which is achieving the best possible improvement for the widest possible area is reached or not.

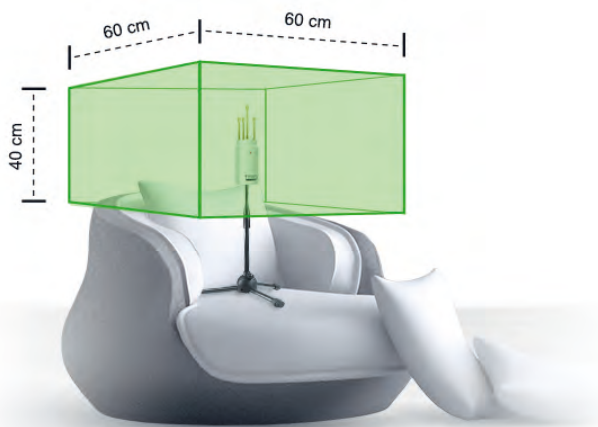
## 7.2 MEASUREMENT POSITIONS

The following illustrations show a relevant and an irrelevant measurement areas for a single listening position. The idea is to perform measurements inside a volume that include head movements.

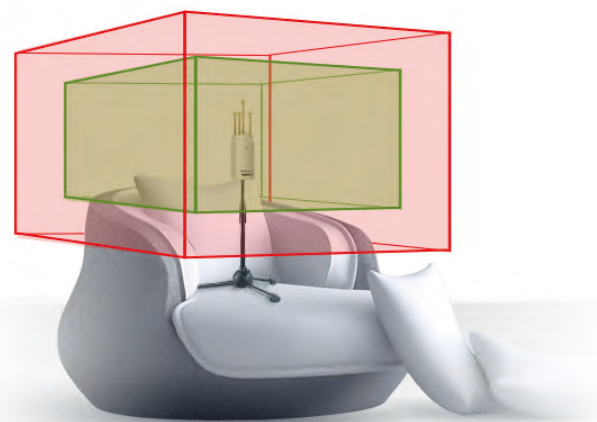
The instructions for the microphone placement concern the main measurement position, called Reference point.

Additional measurements can be made ahead, aside and above the reference point within the volume described above.

For multi seats measurement, it is recommended to choose measurement locations included in the center area and avoid the positions furthest away from the reference point.



Good position multipoint measurement for single position



Wrong multipoint measurement for single position

## 7.3 MULTIPOINT ENGINE

A mutipoint calibration can be performed in the *Optimizer settings/Calibration* page, available in the *Advanced settings*.

Unless the preset loaded in the ST2 HiFi contains multi-measurement, this page should only display one measurement.

Every measurement features a calibration status, a weight and a *Lock* option.

The *Calibration status* indicates whether a successful calibration has been performed for the measurement point or not.

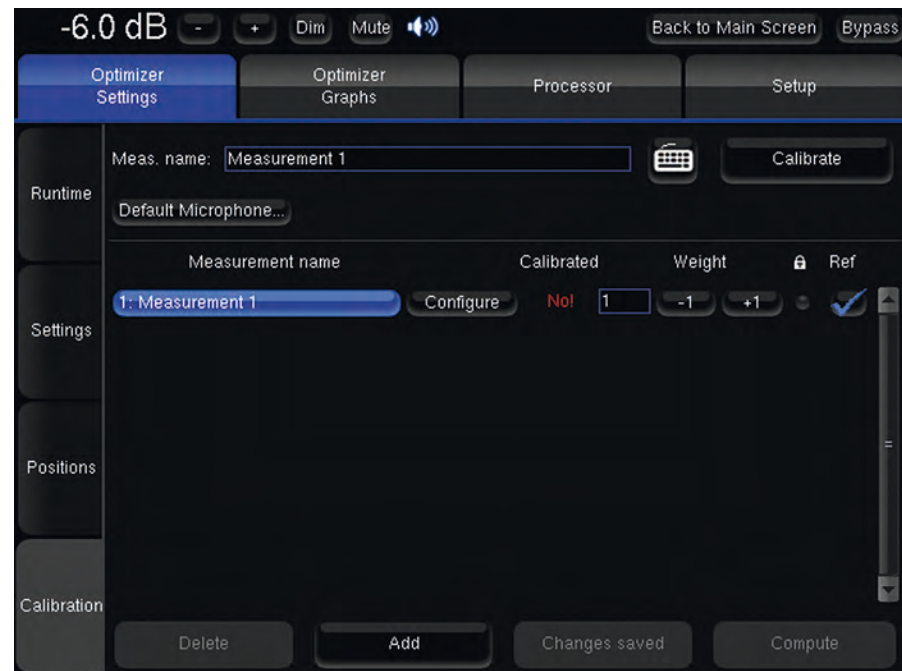
The weight can be used to emphasis a measurement point or another, or even to exclude a measurement point from the compensation calculation (indicate zero in this line).

The *Lock* option makes the calibration impossible (the calibrate button becomes greyed and not clickable) for the measurement point.

The *Ref* lockbox is used to change the reference point.

Only the Reference point is used for:

- Cross-over drivers alignment
- Loudspeaker 3D localization
- Loudspeaker 2D/3D Remapping
- Loudspeaker relative delay/level alignment
- Master delay/level calculation



To perform multipoint measurement, please first refer to the **Optimizer** wizard regarding microphone connection, placement, calibration level.

Unlike the **Calibration wizard**, please note that every step needs to be made manually.

#### IMPORTANT

As first step, push the **Mute** button and decrease the **Master level** to avoid feedback loop.



Then for each measurement point, follow the instructions below:

- Use the **Add** button to create a new measurement.
- Select the new measurement.
- Use the **Meas. name** field to rename the measurement.
- Use the **Calibrate** button to go through calibration of every speakers.

You can select and delete a measure using the **Delete** button.

However, it is rather recommended to use a null weight in order to exclude the measurement point from calculation.

To apply changes, whether it's to take different weighting into account or to use a different measurement as reference point, you need to recompute the compensation filters.

The Optimizer graphs will be updated accordingly.

In the example above, the Reference point has the same weight as the other points altogether.

In total, the measurement information of the reference point will be considered for 50% of the computation (5/10).





# MULTICHANNEL SETUPS

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**NOTE**

The ST2 HiFi has mainly been designed to reproduce stereo content on any 4-speakers layouts, but the unit can also handle 4-channels sources, including 3.0, 3.1, LCRS and Quad.

The Wizard does not include templates for such configurations. It is therefore necessary to proceed manually to configure both the input format, and the speaker layout.

The source format can be chosen in the **Setup > Sources** page of the **Advanced settings**.

Select the required source format from the list using the **Next** and **Prev.** buttons.

The number of LFE can also be selected using **-1** and **+1** buttons.

The channel order displayed on the bottom line is also the calibration order.

Please keep in mind that the source format combined with the number of LFE cannot exceed 4 channels.



## 8.2 SOURCES ROUTING



The next step is to configure the Sources routing to make sure the channel affectation complies with the input connection.

Each row of the sources routing matrix represents a source channel and depends on the sources format configured during the previous step.



Each column of the sources routing matrix corresponds to a physical Input.

The caption above the routing matrix reminds the different types of inputs available on the ST2 HiFi:

- **A** : Analog (both balanced and single-ended)
- **D** : AES
- **SP** : SPDIF (both coaxial and optical)

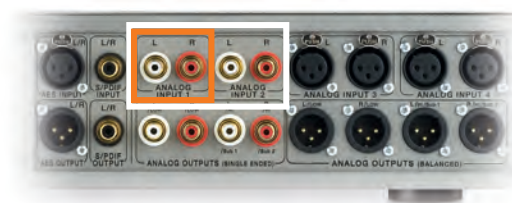


## 4-Channels sources

---

The physical Inputs available in the routing matrix depends on the selected source:

Selecting «Analog 1 SE» activates Analog 1 SE + Analog 2 SE Inputs:



Selecting «Analog 3 Bal» activates Analog 3 Bal + Analog 4 Bal Inputs:



## 8.3

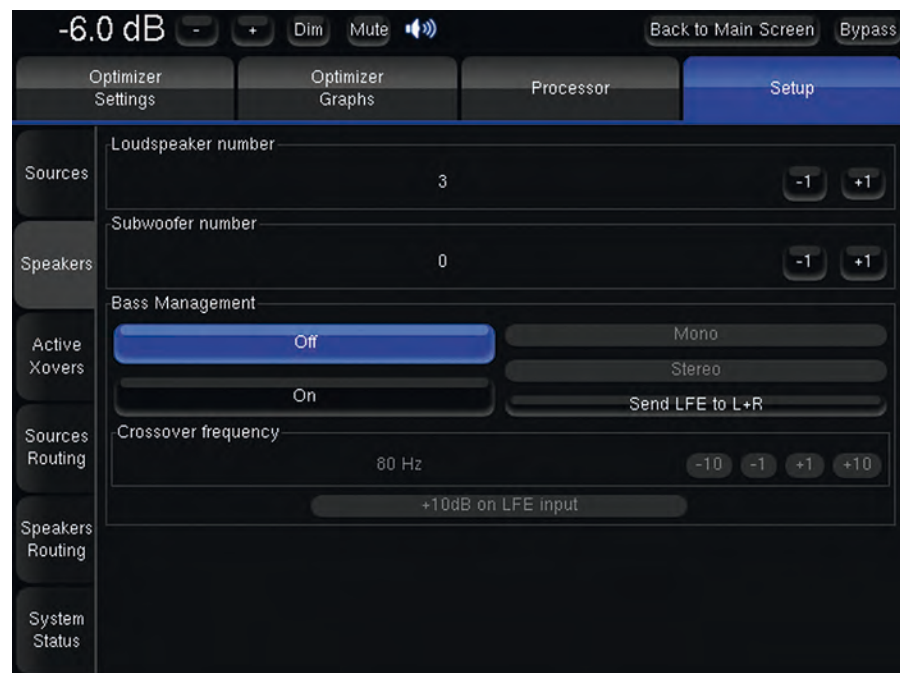
## SPEAKERS CONFIGURATION

The Speaker layout can be configured in the **Setup > Speakers** page of the **Advanced settings**.

The number of speakers and subwoofer can be defined using the respective **-1** and **+1** buttons and cannot exceed 4 in total.

If the speaker layout includes subwoofer(s), the subwoofer will be calibrated full-range.

The **Bass management** settings can be modified after the calibration.



## 8.4 SPEAKERS ROUTING

The Speakers routing matrix is similar to the sources routing matrix.



Each row of the speakers routing matrix represents a source channel and depends on both the sources format and speaker layout respectively configured during the previous steps.



Each column of the speakers routing matrix corresponds to a physical Output.

# Outputs

Unlike for sources, all physical Outputs are used simultaneously.

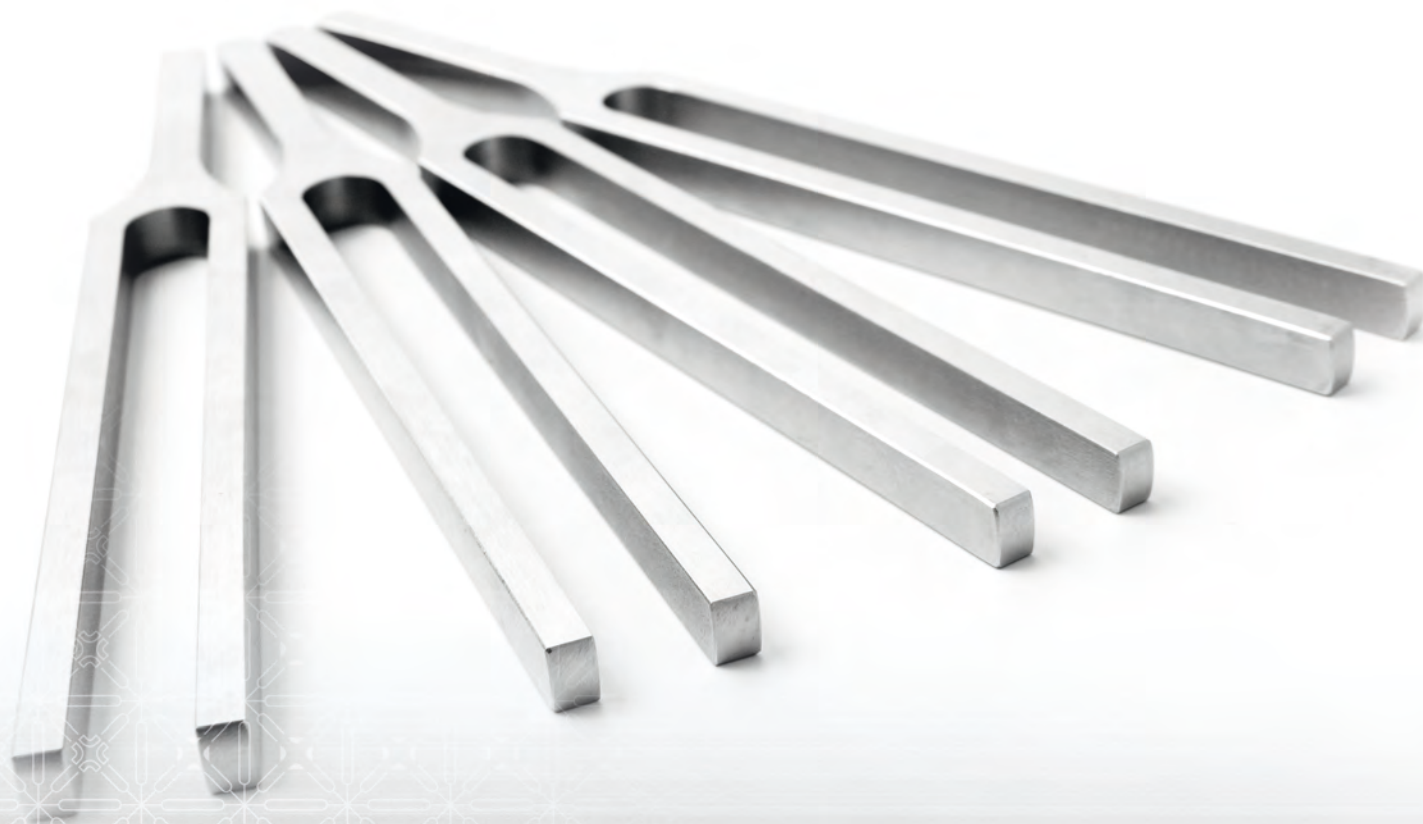


The signals affected to **the first two columns** of the speakers routing matrix are simultaneously sent to the following Outputs:

- AES
- SPDIF
- Ana Bal 1
- Ana SE 3

The signals affected to **the last two columns** of the speakers routing matrix are simultaneously sent to the following Outputs:

- Ana Bal 2
- Ana SE 4



# Musically YOURS





ST2-HiFi

ACOUSTIC FIDELITY



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