ST2-HIFI

USER GUIDE V. 3.8

TRINNOV AUDIO

High Performance Digital Acoustics,
since 2003.

Light version

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THE ST2 HIFI

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.

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

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

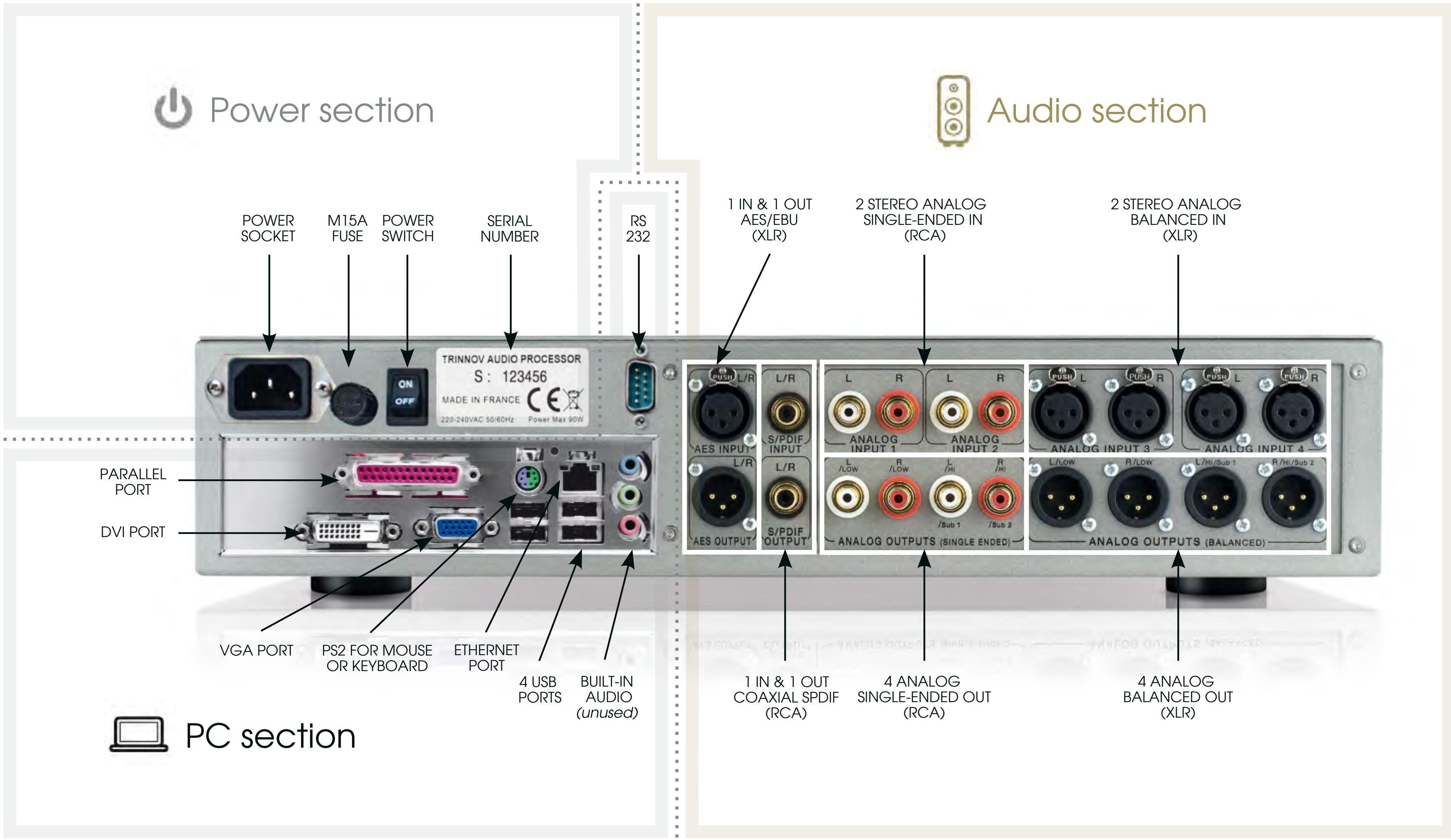
1.3 CONCEPT

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



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 +/-2° 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.

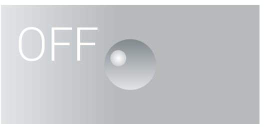
Flat response (within +/- 0,1 dB across the 20Hz-24kHz frequency range) is guaranteed by individual compensation filters.

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

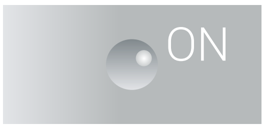
The second purpose of the frond led is to indicate the battery level.

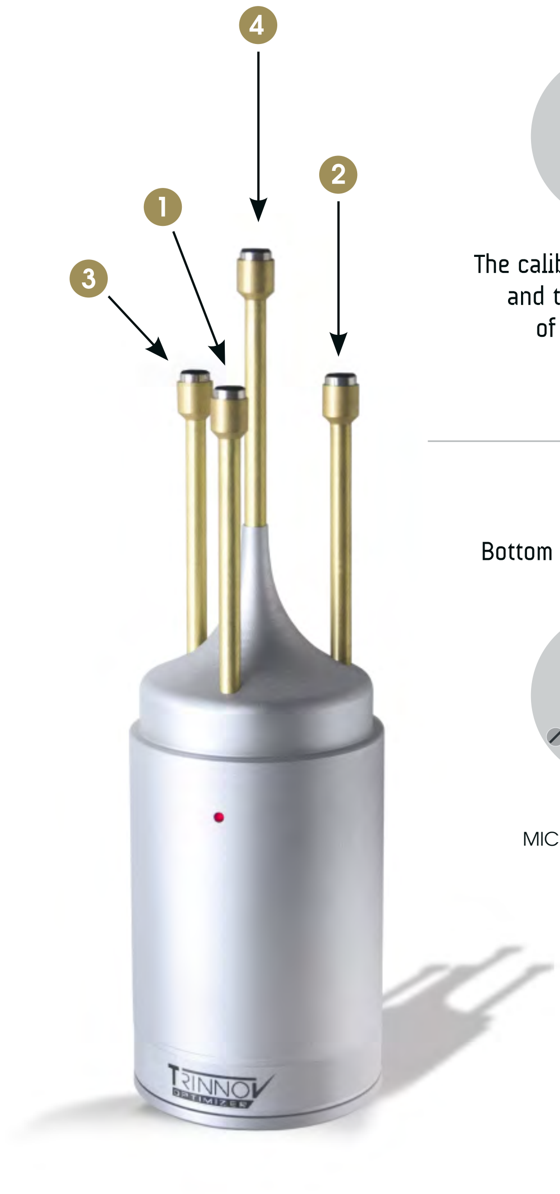
Back side of the microphone

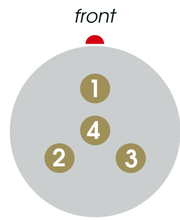
Switching off



Powering on

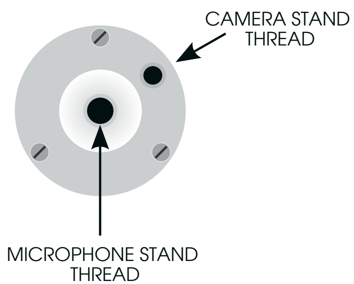






The calibration microphone and the arrangement of the capsules

Bottom of the microphone



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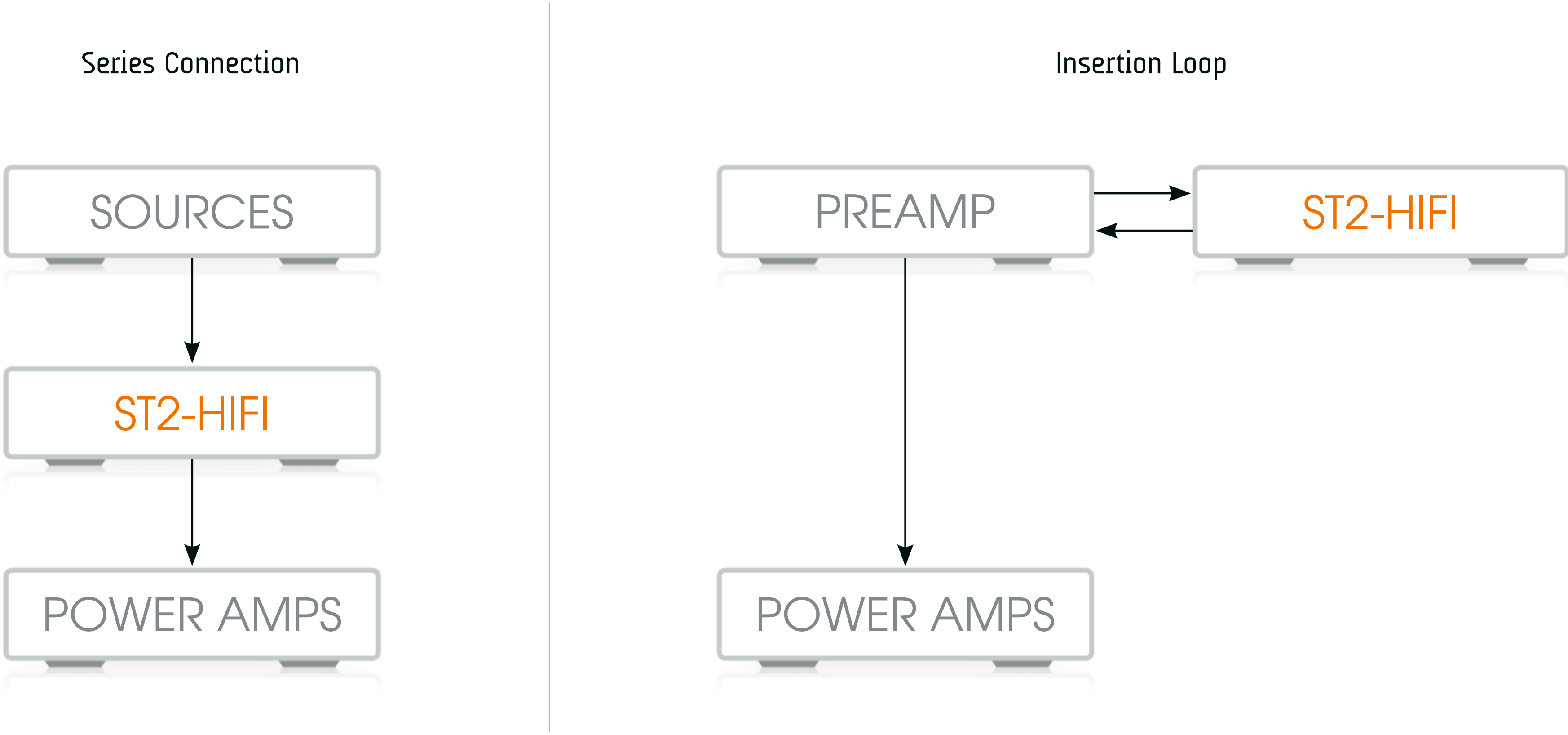
INSTALLATION

2.1 TYPICAL CONNECTIONS

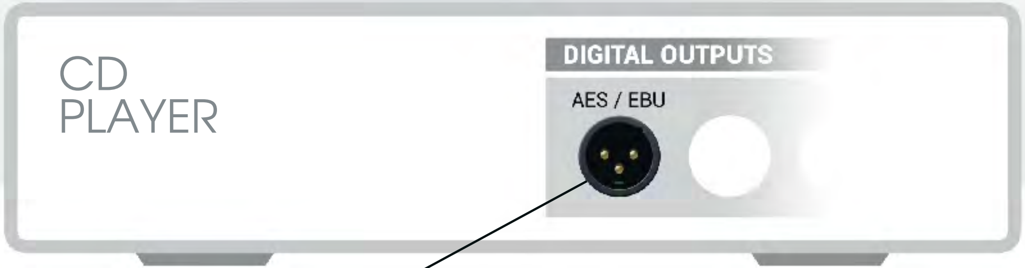


The ST2 HiFi seamlessly integrates your hifi installation.

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

2.2 CONNECTING A SOURCE

Connecting a CD Player
to the AES Input 1

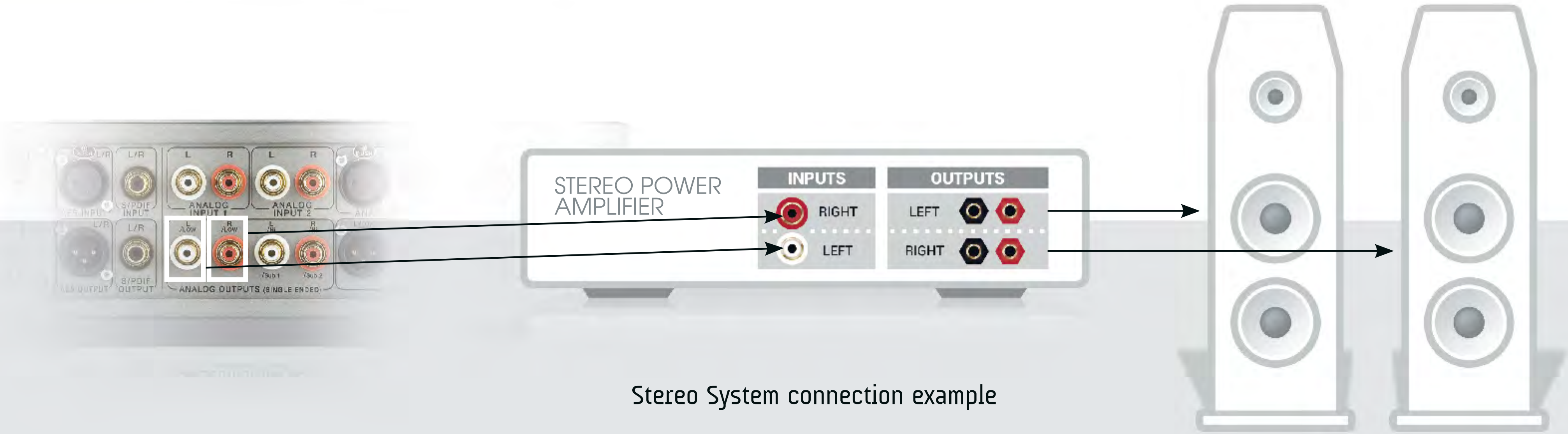


The ST2 HiFi allows you to connect and play any stereo source without any configuration.
This example shows how to connect the AES output from a CD player to the ST2 HiFi.

2.3 CONNECTING SPEAKERS

Default output connectors for a stereo system

This image shows the outputs that should be used as default stereo system output.
Please remind that these outputs play simultaneously.



Stereo System connection example

2.4 CONNECTING THE MICROPHONE

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

- 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

3

GETTING STARTED

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

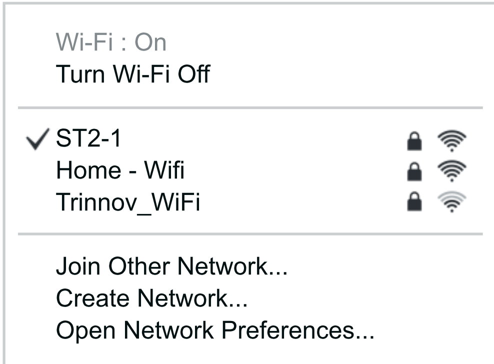
The ST2 HiFi has built-in Ethernet and optional WiFi.
It can simultaneously be part of a network as:

- 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.

3.3 USING THE ST2 HIFI AS A WIFI ACCESS POINT

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.

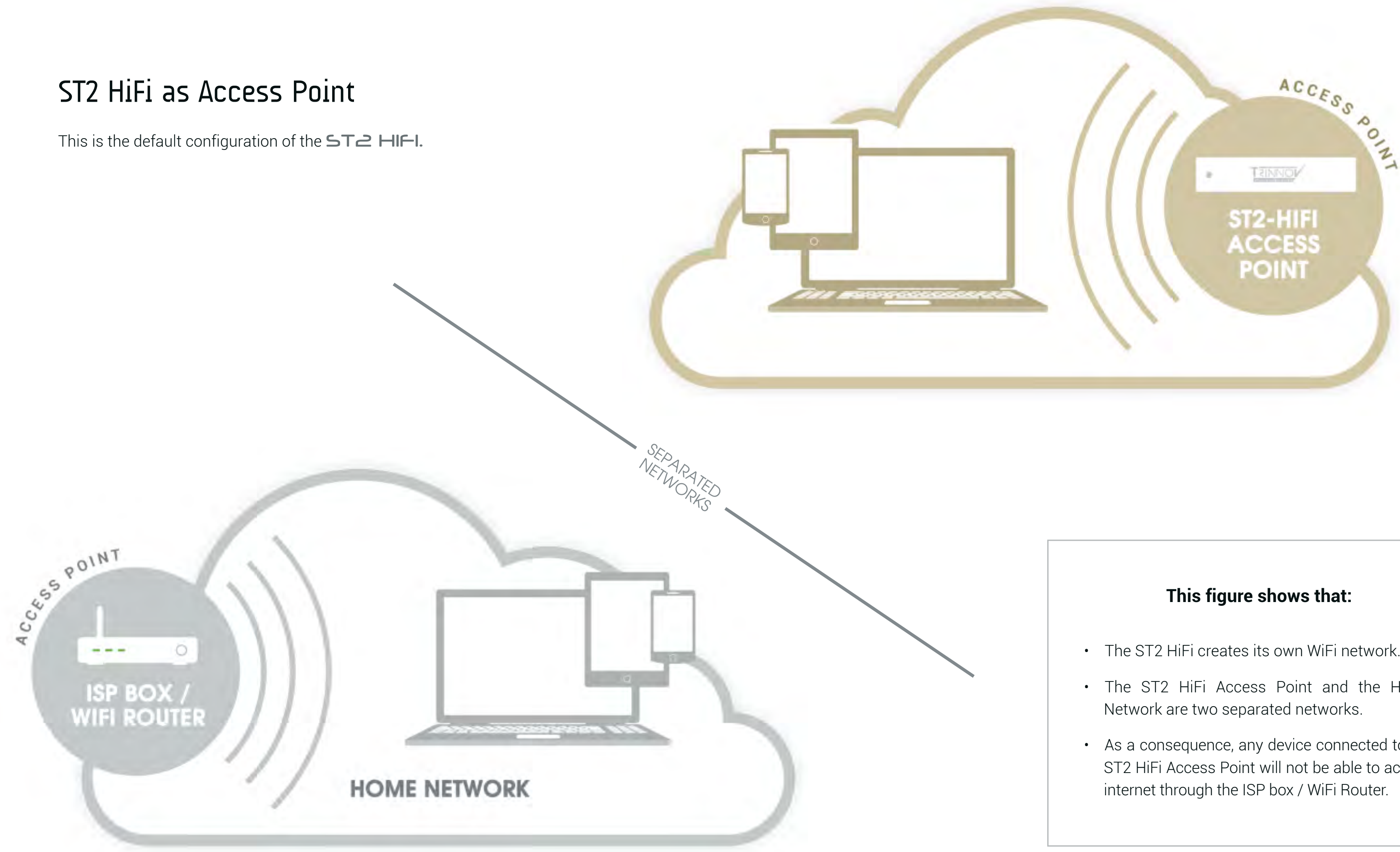


WiFi Networks List - Mac OS X

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 is the default configuration of the ST2 HIFI.



This figure shows that:

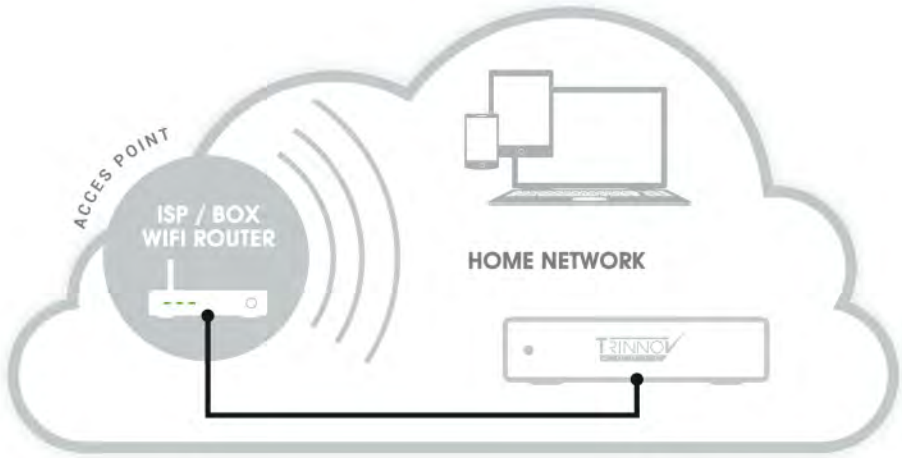
- The ST2 HiFi creates its own WiFi network.
- The ST2 HiFi Access Point and the Home Network are two separated networks.
- As a consequence, any device connected to the ST2 HiFi Access Point will not be able to access internet through the ISP box / WiFi Router.

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 though 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 enables full remote control from any VNC client host device over the network.

3.5.1 About VNC

VNC is a graphical desktop sharing system that transmit 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:

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

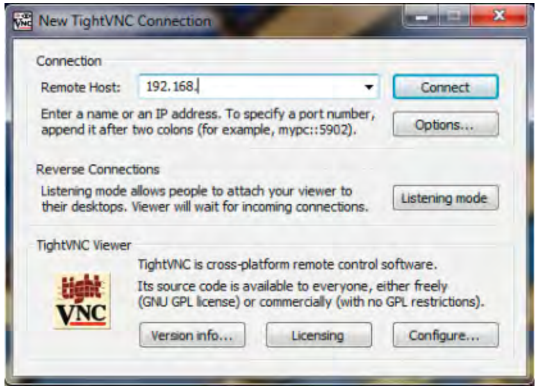
VNC control from Windows® 7

VNC control from Mac OS X

3.5.2 VNC Connection

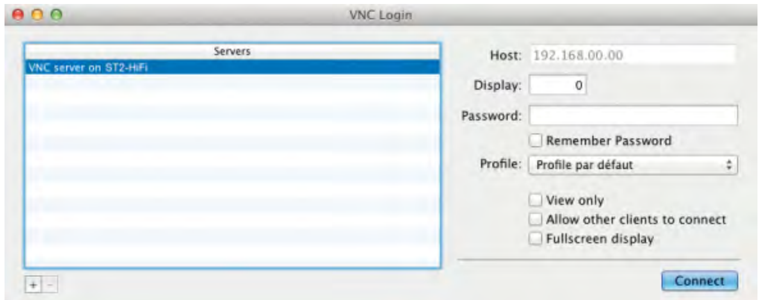
First install the VNC client application on the device you want to control the ST2 HiFi from. Make sure this device is connected to the same network (either wired or wireless network).
The VNC Client will ask you: the server address (or host depending on the application) + a display port + a password.

"TightVNC" Login Panel (Windows 7®)



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

Some clients automatically displays the available server's list:



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.

Mocha VNC Lite available server's list

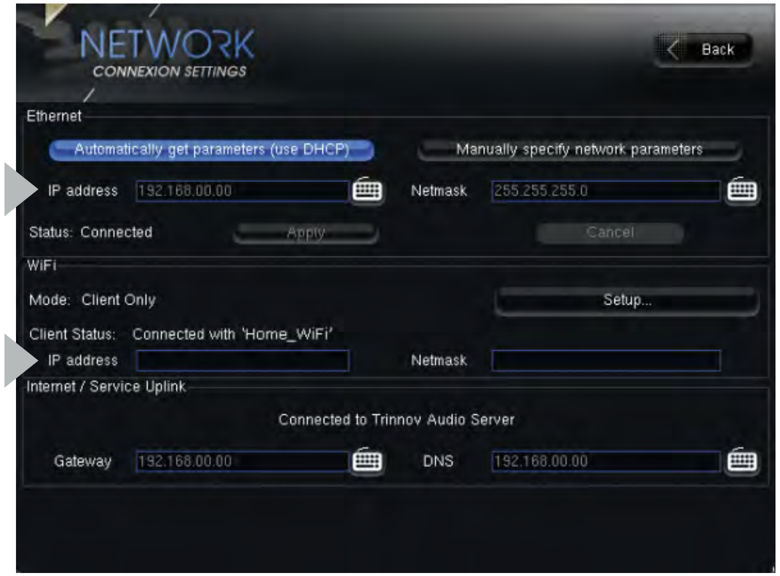
Likewise, the blue arrow displays the available servers list on the VNC client for IPhone/IPad "Mocha VNC Lite".

Server Adress

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 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 confortable network configuration since it allows both controlling the ST2 HIFI and browsing internet from a smartphone or tablet, without any cables.

This figure shows that:

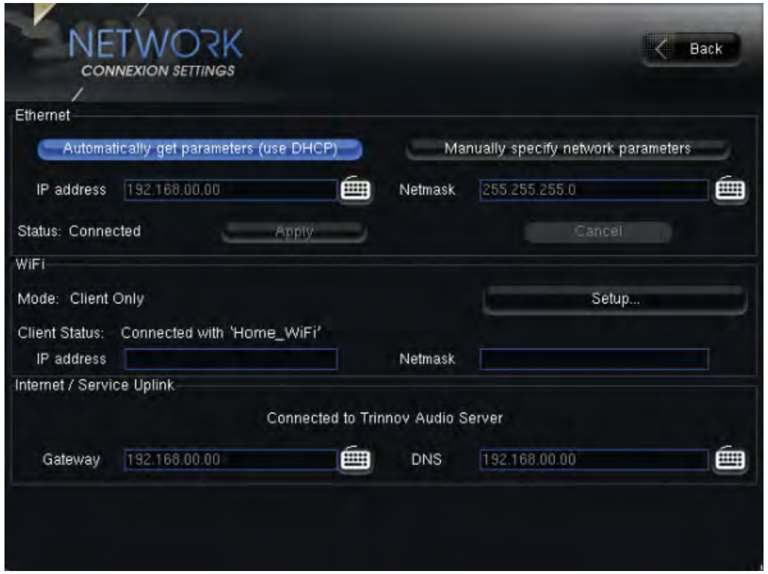
- The ST2 HIFI is part of the Home Network, like any other laptop, smartphone or tablet connected to the Home WiFi via Wireless connection.
- The WiFi AP of the Home Network is also the ISP Box; any devices controlling the ST2 HIFI can also access Internet.

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

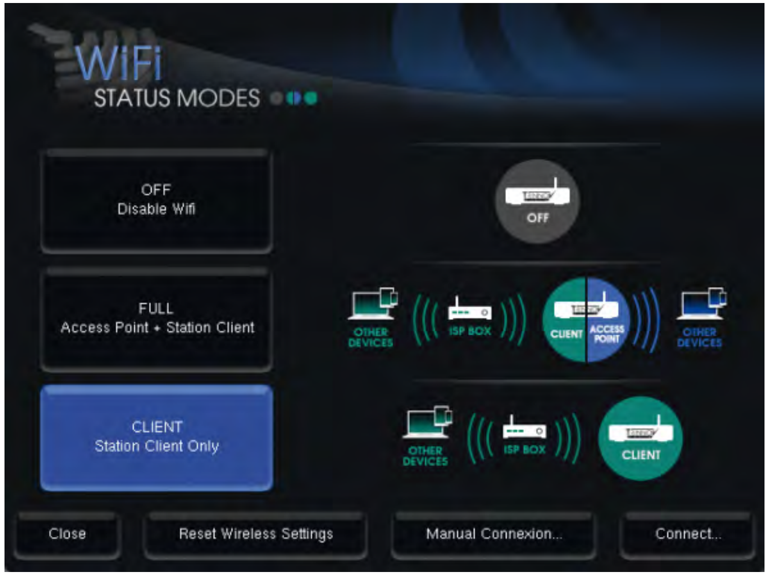
- **OFF** the WiFi module of the ST2 HIFI is disabled.
- **FULL** (default) the ST2 HIFI is both used as an Access Point and a Client.
- **CLIENT** the Access Point is disabled but the ST2 HIFI still operates as a WiFi client as long as the configuration is valid.

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 but only works as an Access Point until the WiFi client is configured).
3. Whether you selected the FULL or CLIENT mode, the Connect button display the available WiFi Access Points.
4. The encryption columns displays a locker icon when the network is protected by a passphrase and/or encrypted.
5. The signal columns displays a WiFi signal icon, indicating the strength of the WiFi signal.
6. If you select an encrypted Access Point, you will be asked a passphrase.



ST2 HIFI - Network Page



ST2 HIFI - WiFi Settings Page

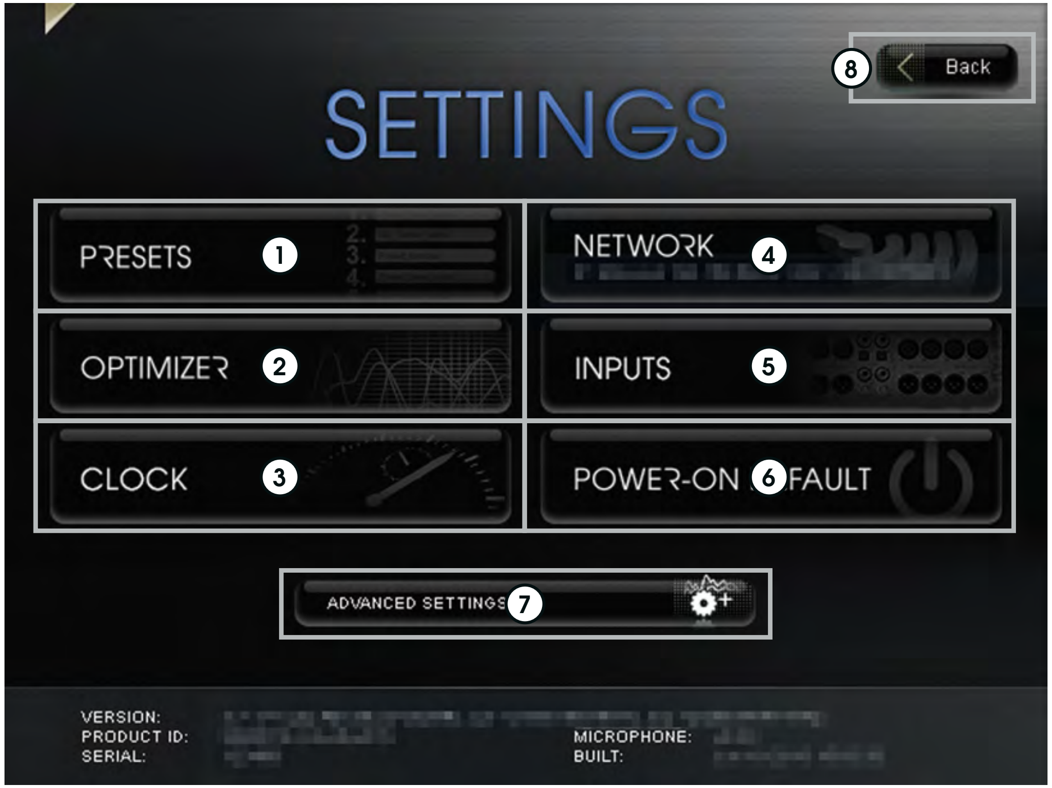
ST2 HIFI - Access Point list

4

CONFIGURATION

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 the Optimizer step-by-step calibration wizard
3. Setup the synchronization mode of the ST2 HIFI
4. Manage the ST2 HIFI network connectivity
5. Name and organize inputs
6. Configure the power-on default parameters
7. Unleashed the full power of the Loudspeaker/Room Optimization 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**.

- **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 master word clock. The choice of the clock signal is done in the ***Clock Source*** frame.
- **MASTER** mode: the ST2 HIFI uses its internal clock for synchronization. This is the default mode for analog inputs.

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.

In the ***Audio Buffer Size*** frame you can select the audio buffer size. A bigger buffer prevents loss of synchronization due to unregular incoming audio flow, but as a counterpart the processing time delay is longer. To set a different buffer size, select the required value and restart the ST2 HIFI.

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 INPUTS CONFIGURATION

The INPUTS configuration menu allows you to manage up to 6 sources which will be available on the ST2 HIFI's ***Home*** page.

A One vertical tab per input on the left.

B Input-specific parameters on the right.

Home > Settings > Inputs

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.

For each input, you can define:

- Its name whith the ***Edit name*** button,
- Its icon on the ***Home*** screen: by clicking on the button under the back button, a new page opens where you just have to click on the desired icon and confirm your choice by pressing the ***Close*** button,
- The input connector used amongst analog (Balanced and Single Ended) and digital (S/PDIF or AES). The selection is done via the drop-down menu labelled ***Input Connectors***,
- The clock setup,
- The preset to use,
- The FIR EQ (available in the ***Advanced Settings***. Please refer to a Trinnov certified installer) used by this input:
    - 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***.
- The User EQ used by this input :
    - 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 input, use ***As in preset***,
    - For more information on User EQ, please refer to *chapter 7.6*.

  User EQs and FIR EQs are optional manual EQs.

- The Input Levels Sensitivity can be used to adjust input levels and achieve a consistant playback level for every input.

NOTE
It is possible to directly recall your prefered settings upon input selection. To do so, assign the same input connector to different inputs and modify the other parameters.

The clock defined in the Inputs menu will prevail on the clock defined in the preset.

4.4 PRESETS MANAGEMENT

Except the inputs 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 NOTE

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

Settings Panel

Presets 1-9 page

The *Presets* page consists of four vertical tabs:

- **Presets 1-9**: includes the Built-in preset and presets 1 to 9
- **Presets 10-19**: includes presets 10 to 19
- **Presets 20-29**: includes presets 20 to 29
- **Preset Info.**: shows the current preset information

Each Preset tab consist of the following items:

- **PRESET NAME** used to enter or modify a preset's name.
- **10 MEMORY SLOTS** including from left to right:
  - **PRESET NAME/NUMBER** greyed when the slot is empty; blue when the preset is selected.
  - **CLEAR** to clear the memory slot. This function irreversibly deletes the preset's parameters.
  - **SAVE** to save current parameters as a preset in the memory slot.
  - **LOCK** 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.

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

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. Please refer to chapter 4.3.
Presets can be backed-up and restored to/from a USB key. Please refer to chapter 4.6.

4.5 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 straightfoward :

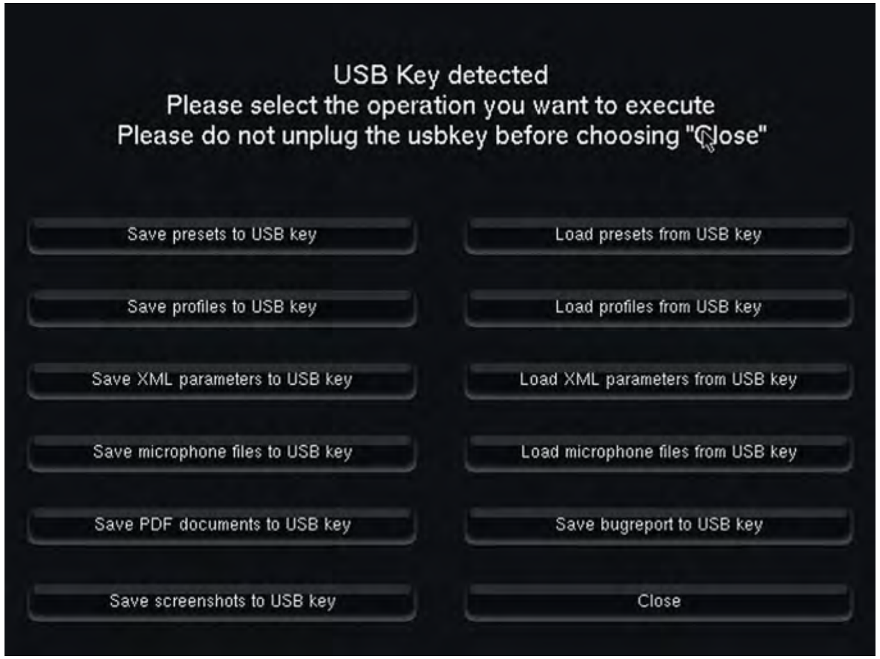
1. Change the preset in the *Power-on Optimizer Preset* frame with the < > arrows button
2. Select the required Input in the *Power-on* Input frame. It will then be higligthed in blue.
   The *Use Last Selected* button allows to use the input used when the unit was last powered off at start-up input.
3. Once your setting is over change click on the *Back* button to apply 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 occured 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 Interace by pressing the "print screen" key. These screenshots can be copied through the USB Key interface afterwards.



If you plug a USB key to one of the rear USB port of the ST2 HIFI a menu will pop-up automatically.

IMPORTANT NOTES

To ensure the data integrity of your USB Key please press the Close button and wait until the home page is displayed before unplugging it.

Be careful when reloading presets from a USB key: a preset stored on the ST2 HIFI on the same memory slot as the backup preset will be overwritten without warning! Make sure the presets you want to keep are locked to prevent any mishandling.

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

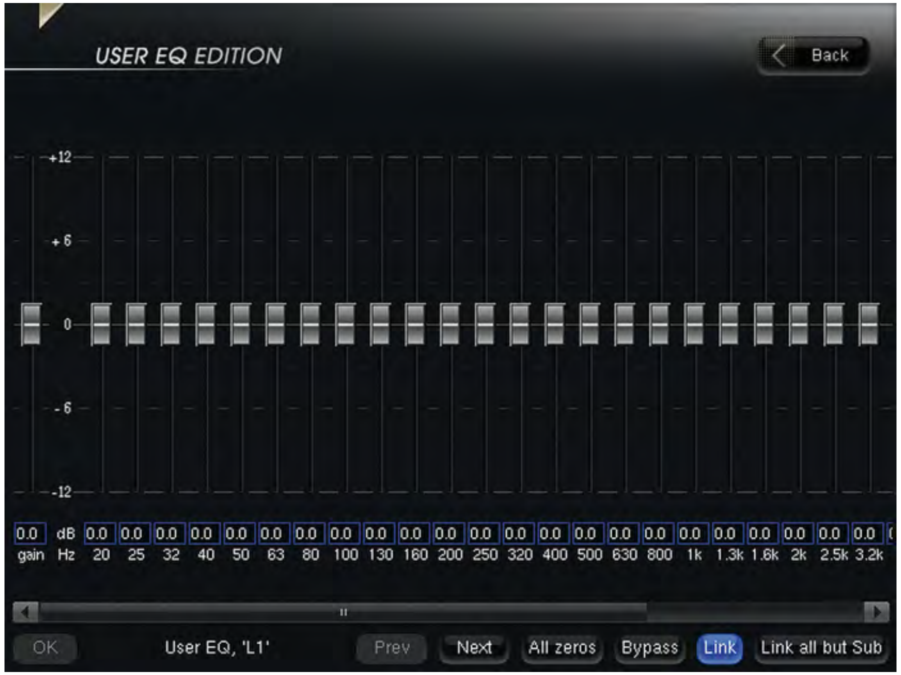
You can launch a test noise for each configured channel, for example check if your connections are correctly done and assigned.

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. The frequency you are modifying will be higlighted 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 Frequency sliders

4 Direct Input Gain values

5 Frequency sliders values

1 Scrollbar

2 EQ and Channel name

3 Change Channels button

4 Reset all values to 0

5 Bypass EQ

6 Link Channels with or without Sub

Once in the ***Channels*** page you can save your current EQ by presssing ***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 overwritable EQ to save the new one.
- 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:

1 **MUTE** it with the ***Mute*** button

2 **INVERSE THE POLARITY** with the polarity trigger (the polarity is inversed when the trigger is lighted up)

3 **ADJUST THE LEVEL** by +/-0.5dB step

4 **ADJUST THE DELAY** by +/-0.5ms. You can not input a negative delay.

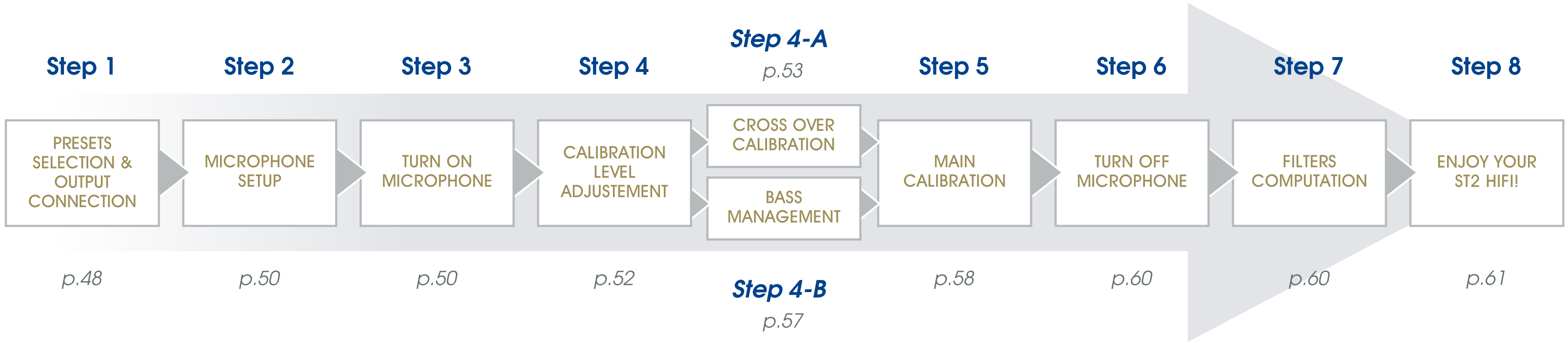
5

BASIC OPTIMIZER SETUP

5.1 OVERVIEW

The ST2 HIFI includes a step-by-step calibration assistant to helps 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.

Using the Wizard templates implies respecting the following speaker connection order:

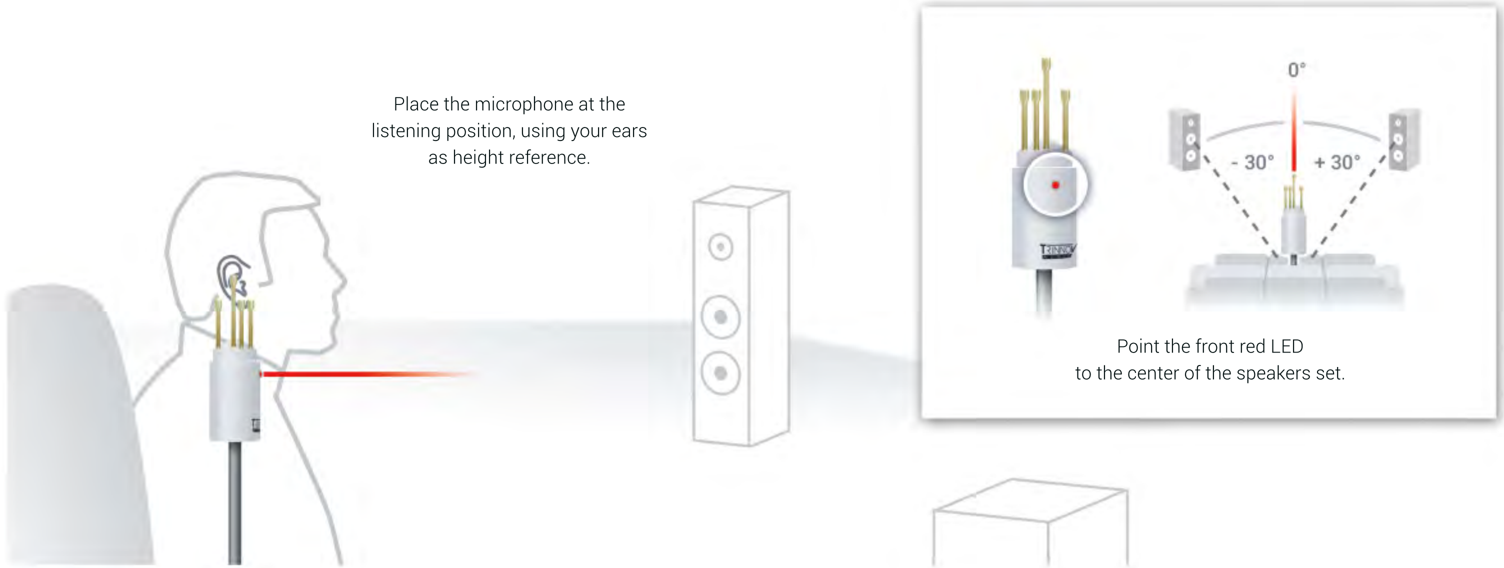
TEMPLATES	BALANCED ANALOG OUTPUTS				SINGLE-ENDED ANALOG OUTPUTS			
	L/LOW	R/LOW	L/HI/Sub 1	R/HI/Sub 2	L /LOW	R /LOW	L /HI /Sub 1	R /HI /Sub 2
2.0 MONO-AMP	LEFT	RIGHT			LEFT	RIGHT		
2.0 BI-AMP	LEFT Low	RIGHT Low	LEFT High	RIGHT High	LEFT Low	RIGHT Low	LEFT High	RIGHT High
2.1 BASS MGMT	LEFT	RIGHT		SUB	LEFT	RIGHT		SUB
2.2 MONO B-MGMT 2.2 STEREO B-MGMT	LEFT	RIGHT	SUB Left	SUB Right	LEFT	RIGHT	SUB Left	SUB Right

5.3 MICROPHONE SETUP

Step 2 - *Microphone Setup*

The first and 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 reflecting surface** (leather sofa, glass table, back wall...) close to the microphone, or cover it with a thick blanket.

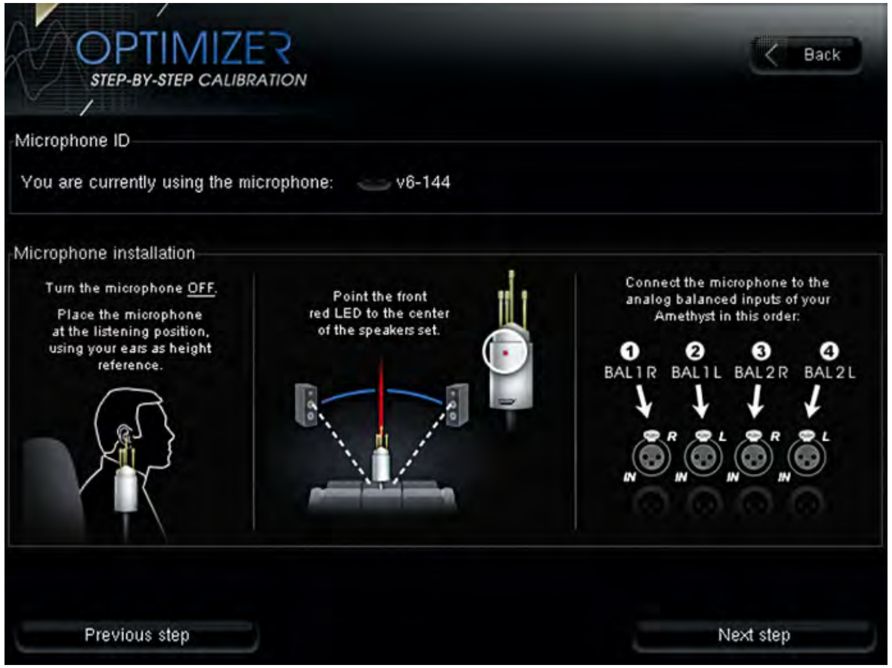


In this very straightfoward step, you just have to choose the model number (ie ID) of your microphone from a dropdown menu. That ID can be found directly at the back of your microphone. If by error your microphone is not referenced in the list, please contact your vendor for an update.

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 ST2 HIFI'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.

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 adjustement 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.

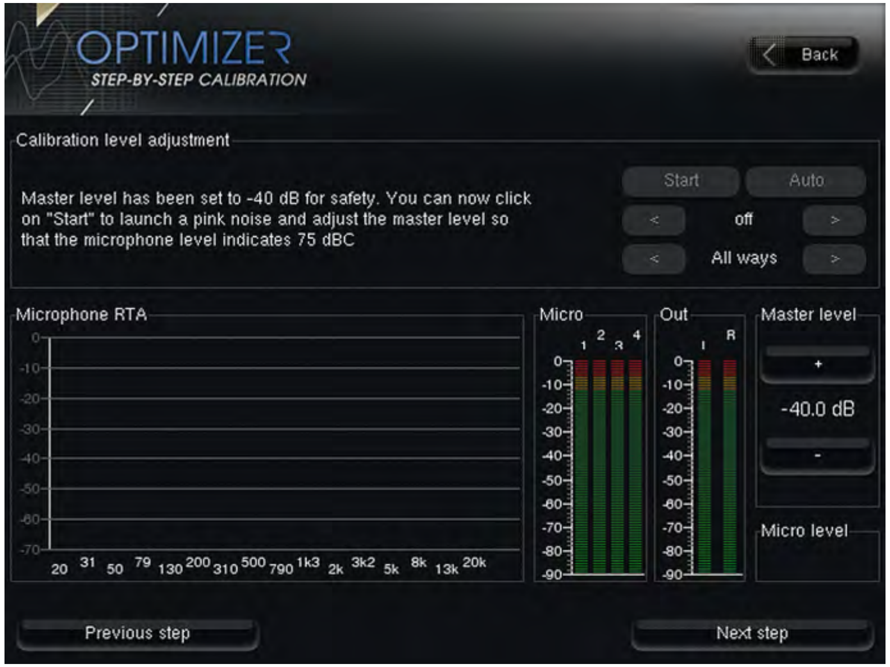
The *Microphone RTA* frame, where RTA stands for Real Time Analysis, gives you a real time visualisation of the frequency response of your system seen by the microphone.

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 adjustement* 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 adjustement:

1. Press *Start* to launch the pink noise
2. Check if a signal is played in the Out frame. If not, make sure you have pressed *Start*.
3. Check if a signal is received by the microphone in the *Micro* frame. If not, chack that your microphone is correctly plugged in and that it is switch 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 adjustement* 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.

Step 4-A - *Semi-automatic crossovers calibration*

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

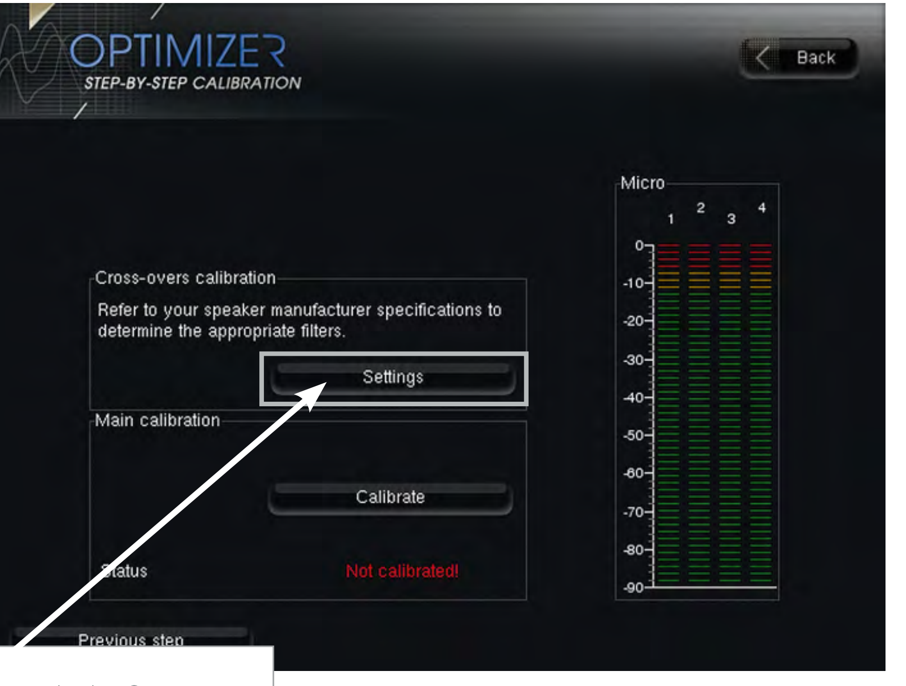
- The Cross-overs calibration is only available with the 2.0 Bi-amp template.
- The Bass management crossover frequency is only available with the 2.1 and 2.2 templates.

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 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 (default) and right speakers cross-overs settings as horizontal tabs.

The left and right speakers are linked (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:

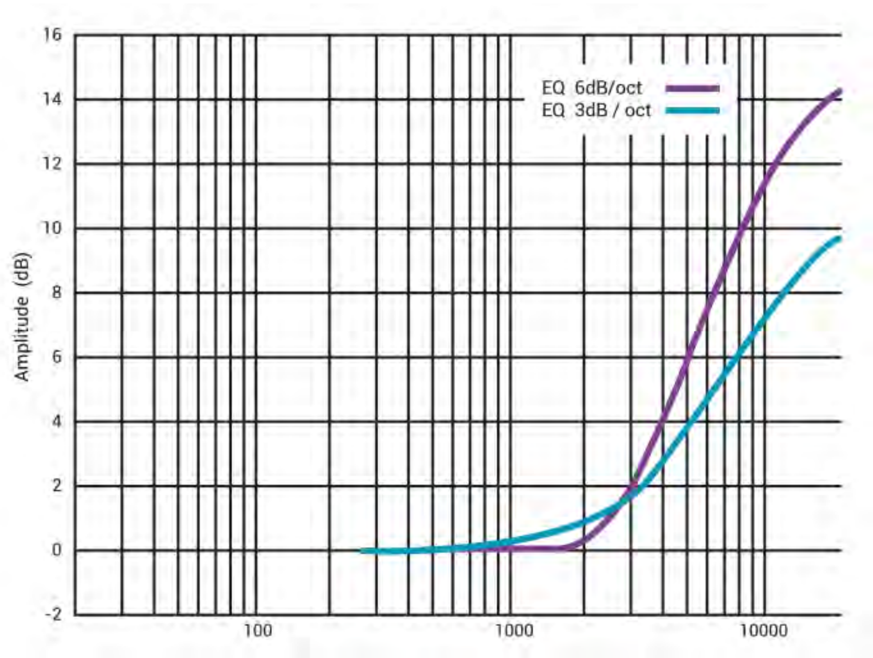
- Low / High cross-over frequencies (default 2000 Hz)
- Filter type (default 4th order Linkwitz-Riley). The following filters are available:
  - 2^{nd} and 4^{th} order Linkwitz-Riley
  - 2^{nd}, 3rd and 4^{th} order Bessel
  - 2^{nd}, 3^{rd} and 4^{th} order Butterworth
- Select the type of high-pass 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.

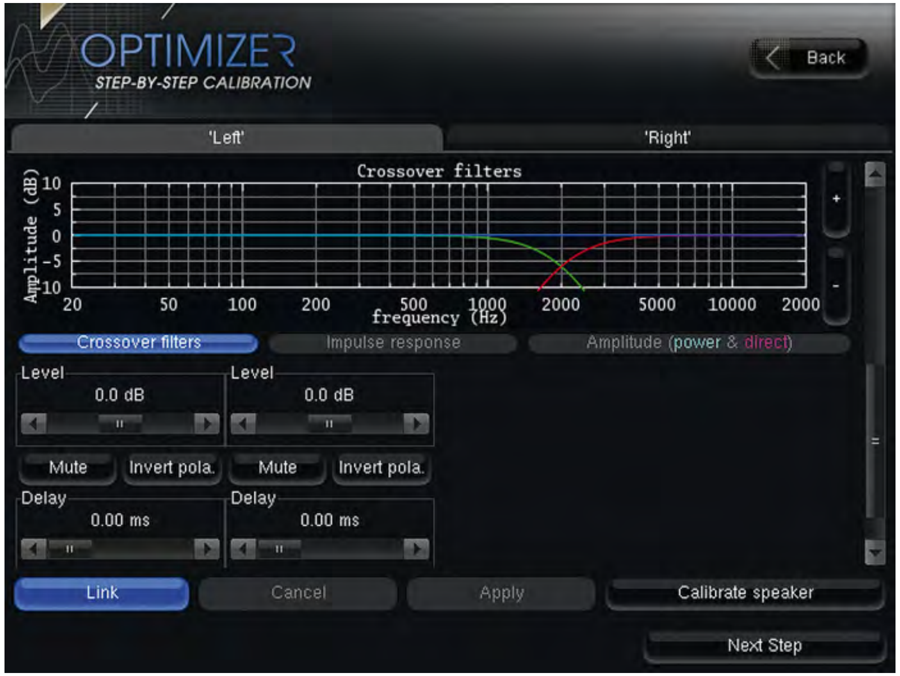
If change is unwanted, press the ***Cancel*** button.

"Constant-directivity horn" EQ filters



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

Crossover filters representation



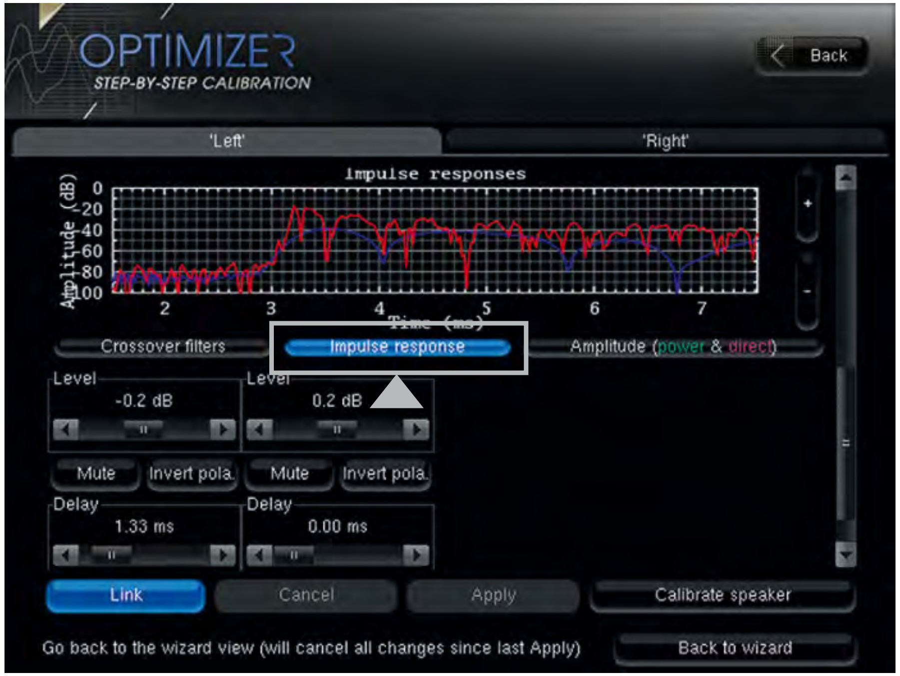
Scroll down to display the crossover filter representation.

- 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 Left speaker drivers and follow onscreen instructions. A test signal is sent to each driver.

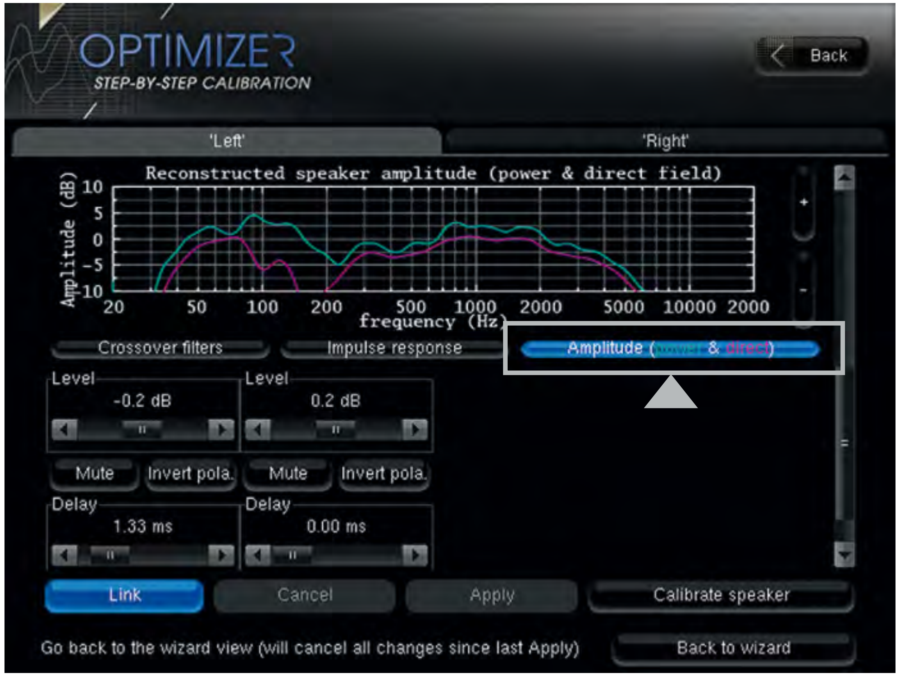
The following graphical representations are available after calibration.

Crossovers Impulse response after calibration



The ***Impulse response*** graph shows the measured impulse response of each driver and indicates 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 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.

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.

Step 4-B - *Bass Management setup*

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

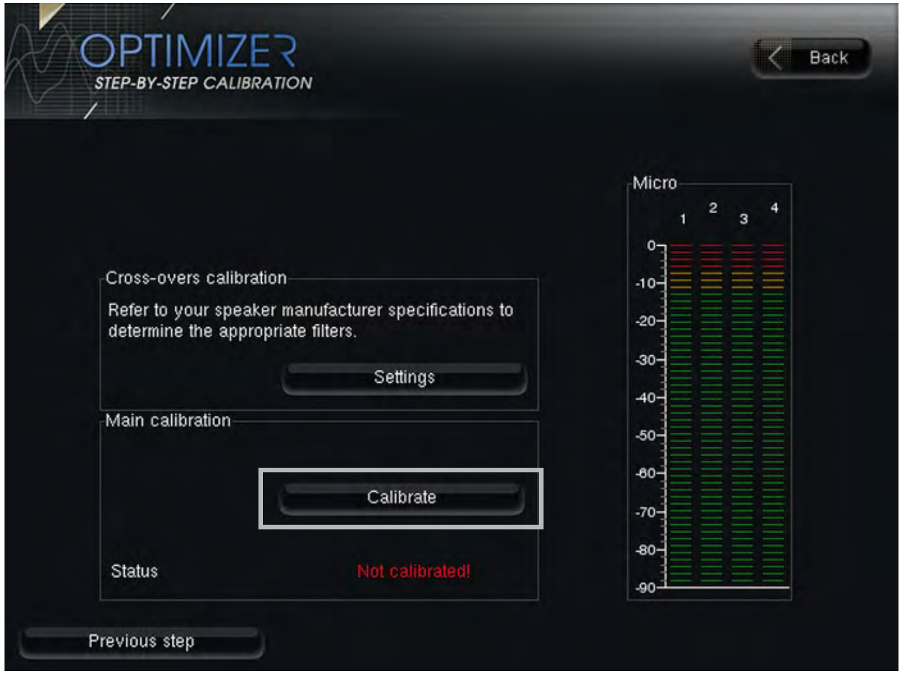
Low frequencies below the bass management crossover frequency will be filtered and sent to the subwoofer(s).

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

IMPORTANT NOTE
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 serie 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 indicated the microphone input levels during calibration.
If a problem occures, a warning message will be displayed in orange under the ***Status*** label.

Warning messages can be:

Crest factor too low for capsules...:
Check that your microphone is switch on and that your level meets the calibration requirements.

Unable to determine position... or *Unstable position...*:
Check that nothing stands in the path between the speaker and the microphone. If it is not enough 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 loose 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 irratable 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 ***Presets*** button.

You can also fine-tune automatic settings via the ***Channels*** page (see chapter 4.7) or dive into the ***Advanced parameters*** (via the ***Settings*** page).

6

ADVANCED OPTIMIZER SETUP

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 9.2).

6.2 ARBORESCENCE OF THE ADVANCED SETTINGS MENU

Optimizer Settings

RUNTIME
Enable/Disable automatic correction

SETTINGS
Configure optimization

POSITIONS
Check speakers position

CALIBRATION
Manage & perform calibration

Optimizer Graphs

GRAPHS
Analyze your system's performance with and without Optimizer

- Amplitude
- Phase
- Impulse response
- Group delay

Processor

Manual settings

MANUAL

METERS
Check Inputs & Outputs levels

MASTER
Configure Master level display and Master delay

INPUTS
- Inputs gain
- EQ
- Noise generator
- Solo / Mutes

OUTPUTS
- Outputs gain
- Fir EQ / EQ
- Solo / Mutes

OUTPUT DELAY
Set individual delays on your outputs

Setup

Automatically configured with Setup Wizard Templates

AUTO

SOURCES
Set Inputs format

SPEAKERS
Set number of speakers

ACTIVE XOVER
Configure active crossovers

SOURCES ROUTING
Route your Inputs

SPEAKERS ROUTING
Route your Outputs

NETWORK
Network settings

SYSTEM STATUS
System settings

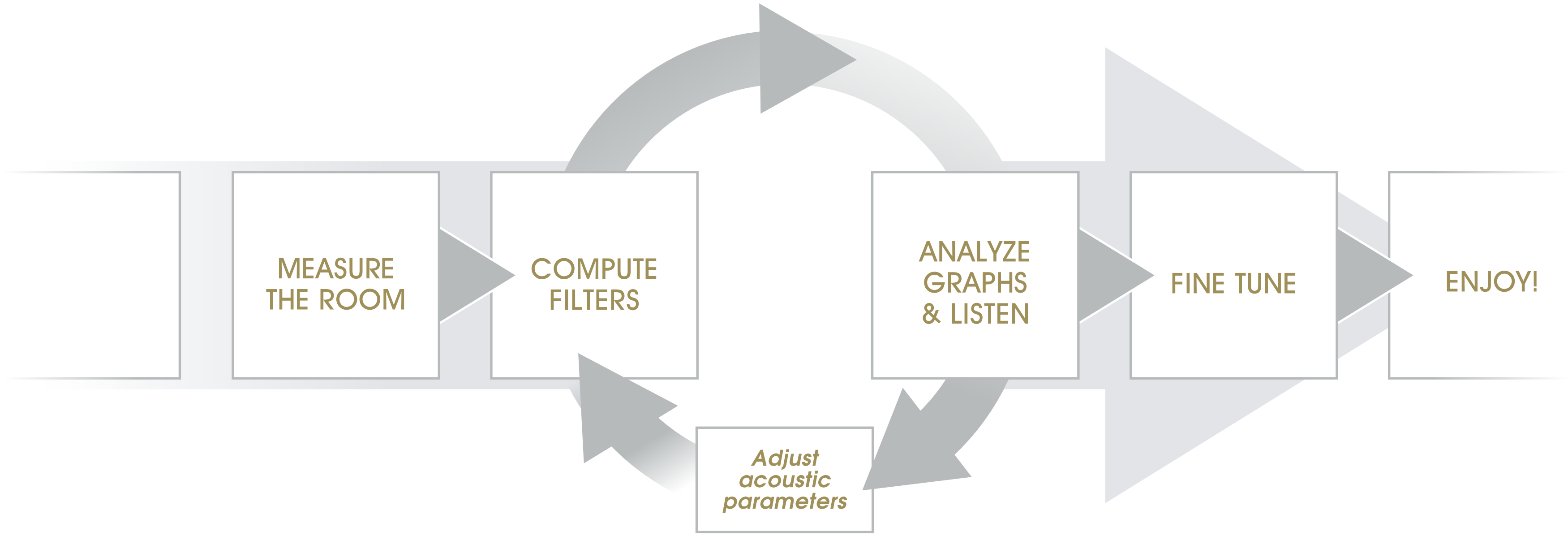
6.3 ITERATIVE PROCEDURE

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

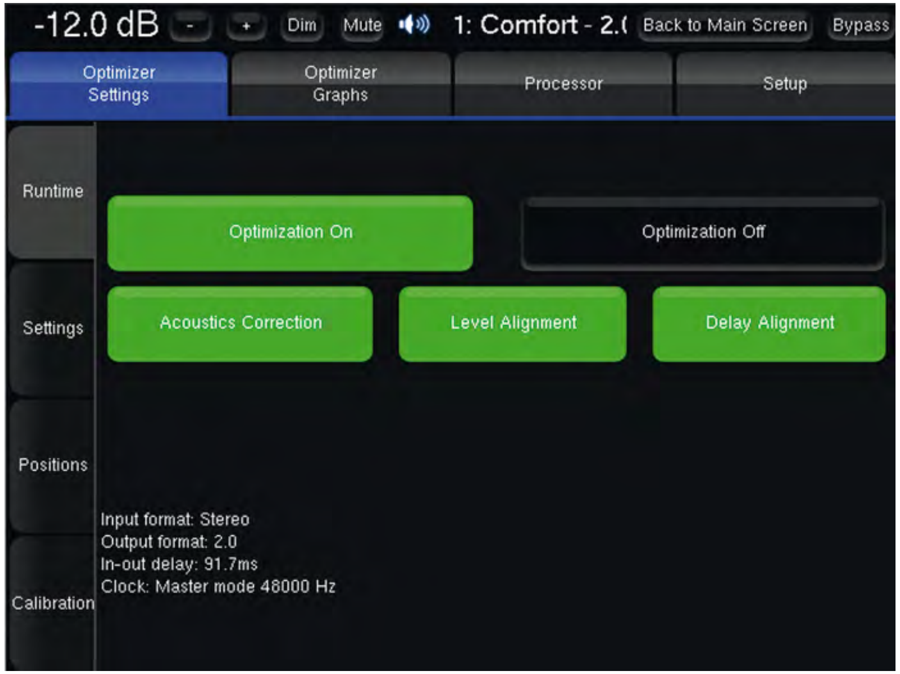


The iterative procedure

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***:

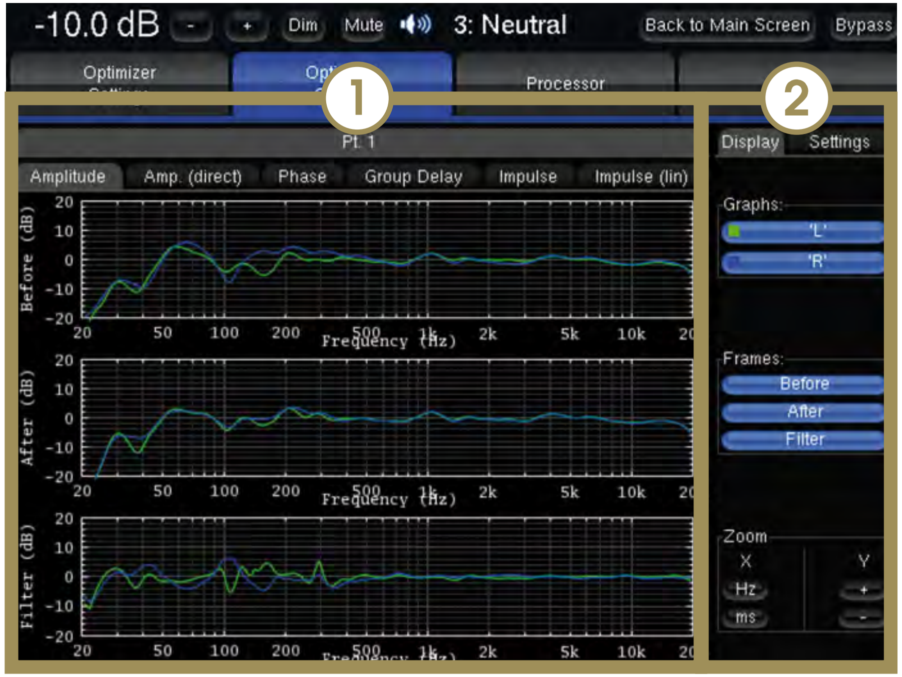
- ***Acoustics correction***: amplitude and phase response corrections
- ***Level Alignment***: level alignment of all speakers
- ***Delay Alignment***: time alignment of all speakers

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 accross 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 phases rotations and relative responses offset between speakers indicate a lack of definition and an unstable stereo image.

Group Delay : the graph delay is also a time-related representation and displays relative time arrival of all freqeuencies at the listening point.

Impulse response : 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.

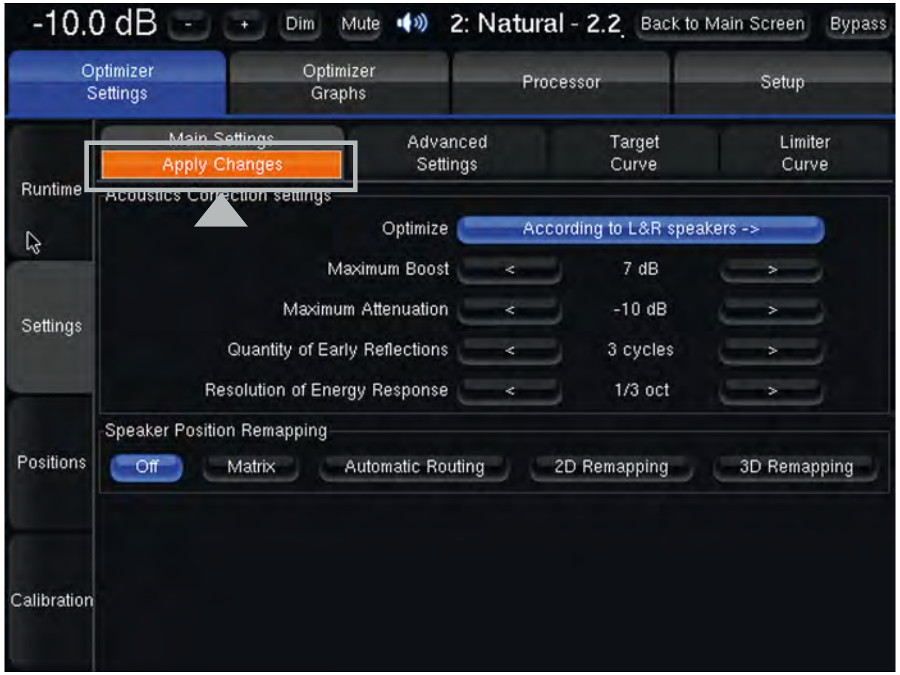
Computation & Graphs update

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 initates 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 incremently 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.



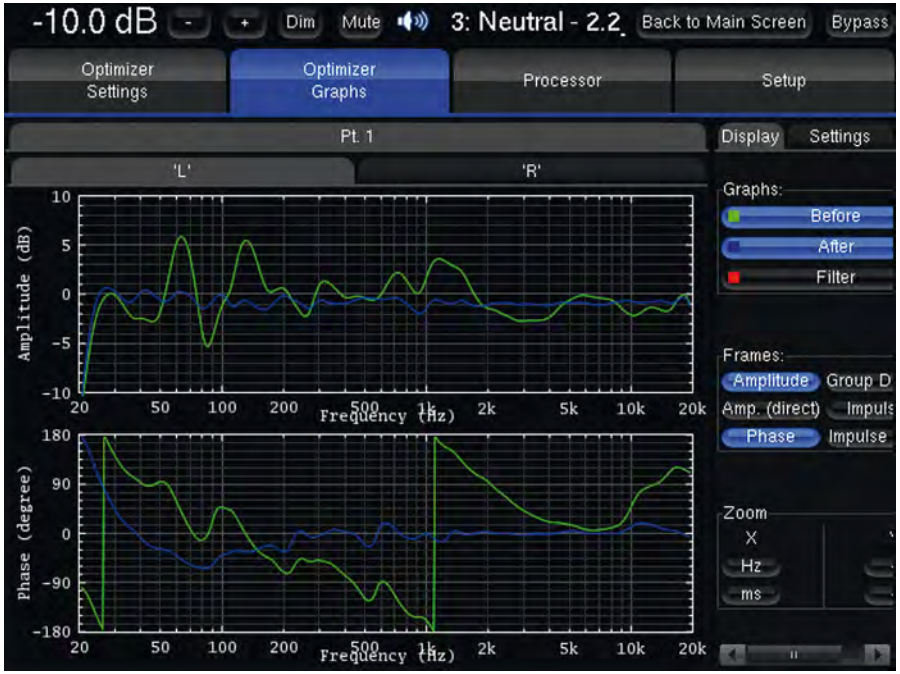
Optimize Modes

The *Maximum Boost* and *Maximum Attenuation* parameters are used to avoid over-correction.

Excessive boost can cause distorsion and damage speakers and should be avoided unless the amplification + speakers system has sufficient headroom.

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

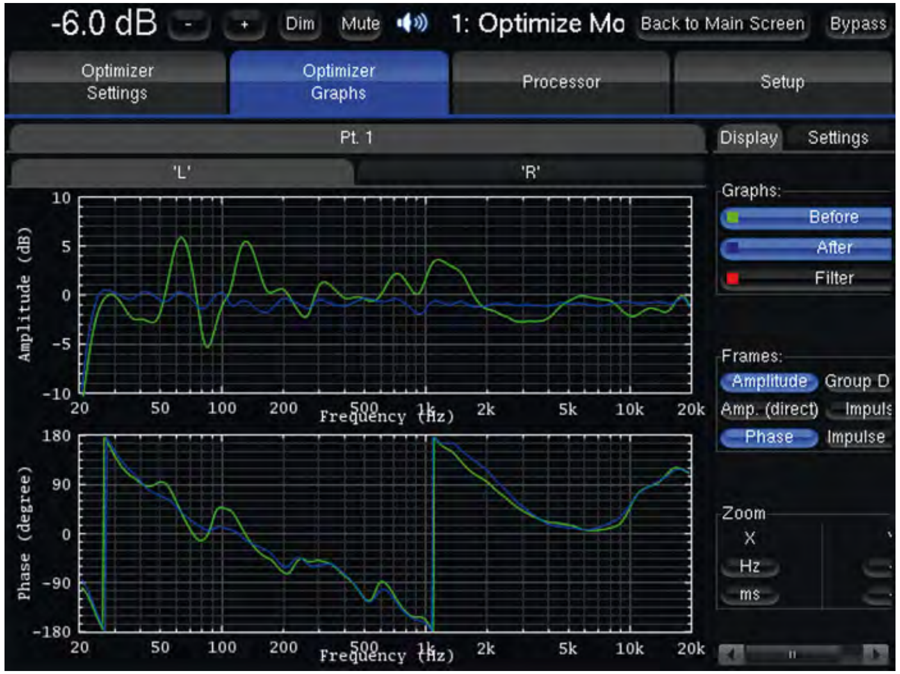
Amplitude + Phase



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

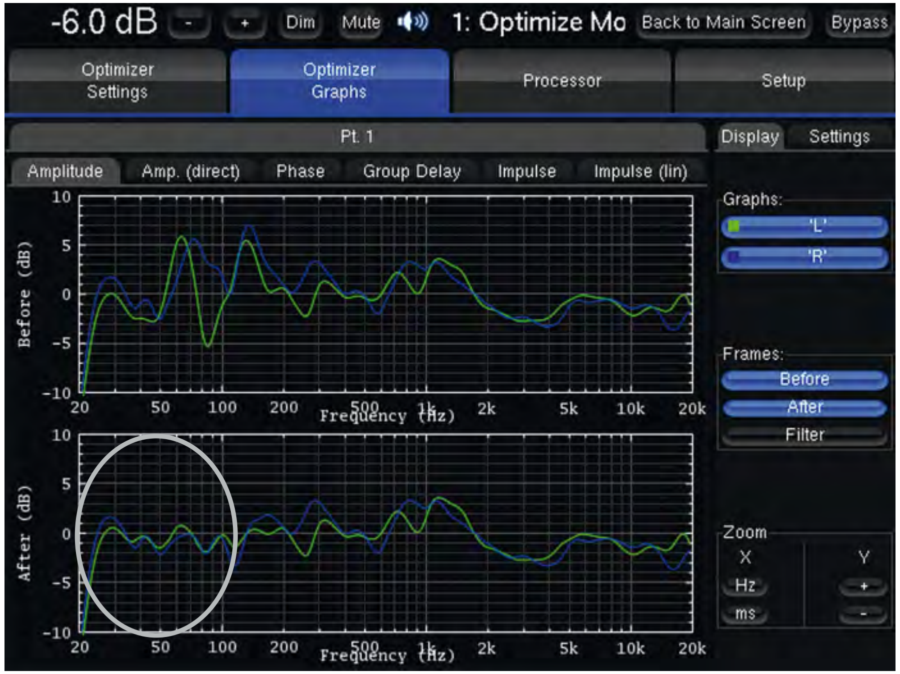
It is the default behaviour of the Optimizer and implies amplitude AND phase response corrections to achieve better transient responses and improved sound staging.

Amplitude only



The ***Amplitude mode only*** provides amplitude response correction and result 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)..

The range of action of this mode can be defined in the ***Advanced settings*** page.

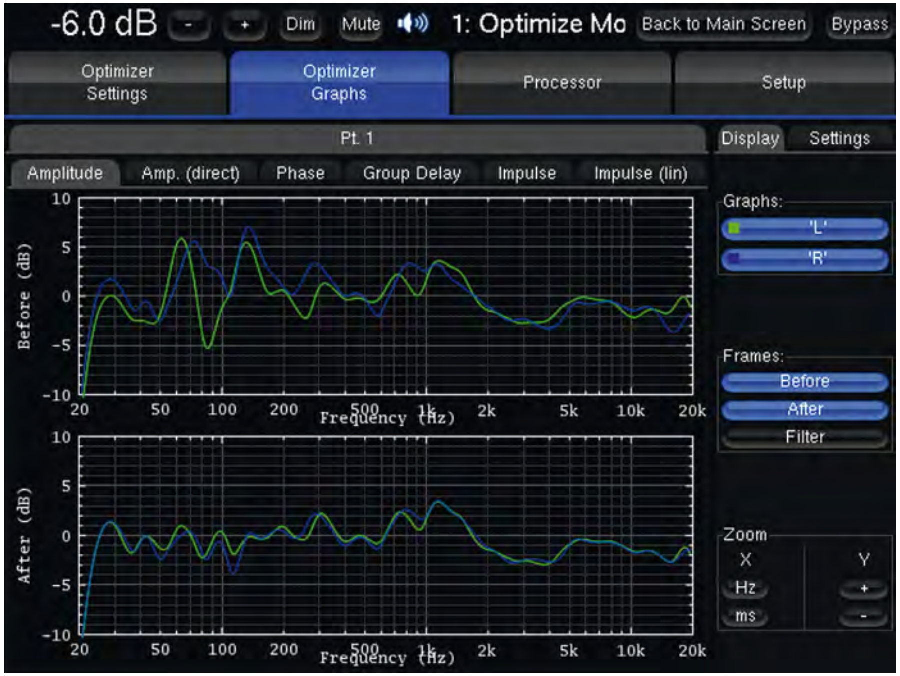
IIR Filters maximal frequency

IIR filters are dedicated to low range amplitude accidents.

The ***IIR filters maximum frequency*** can be modifed 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

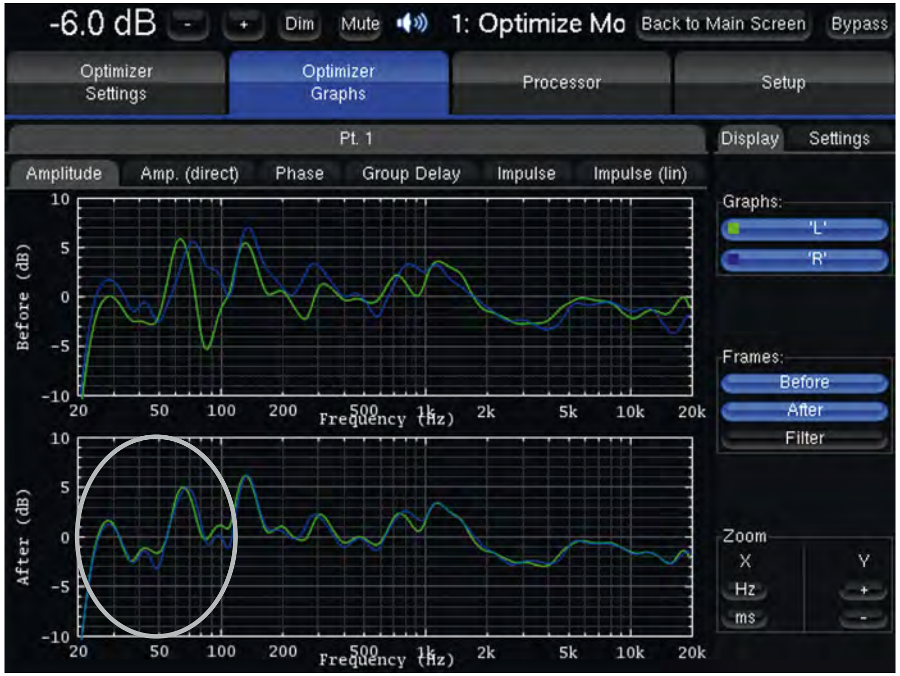


The ***According to L&R speakers*** mode takes the average response of the left and right loudspeakers as reference and automatically uses it as target curve.

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 :

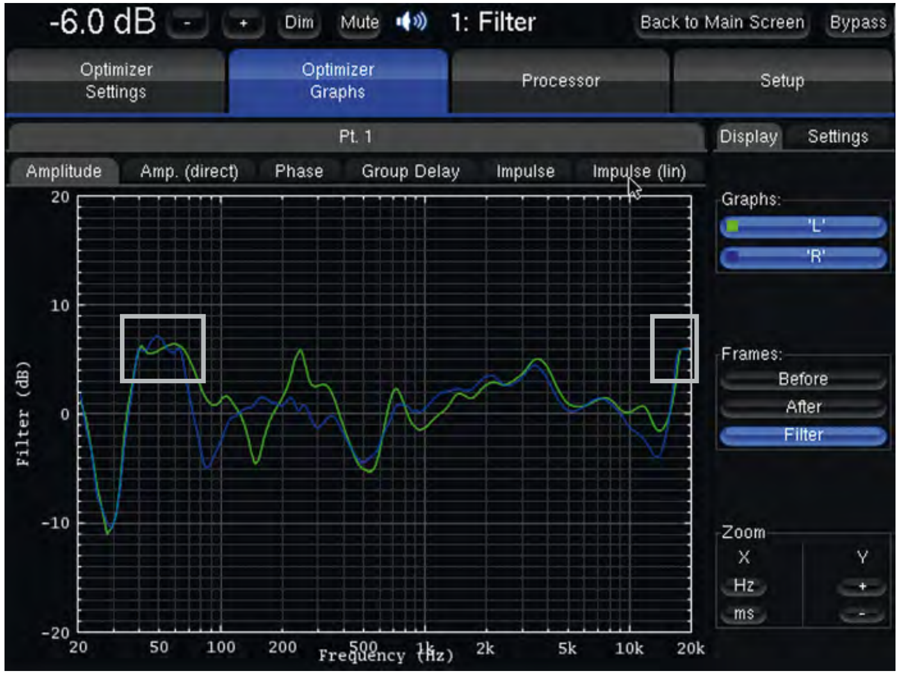
Switching the ***Processing on LR target*** from ***IIR Only*** to ***None*** will disable the bass correction.

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



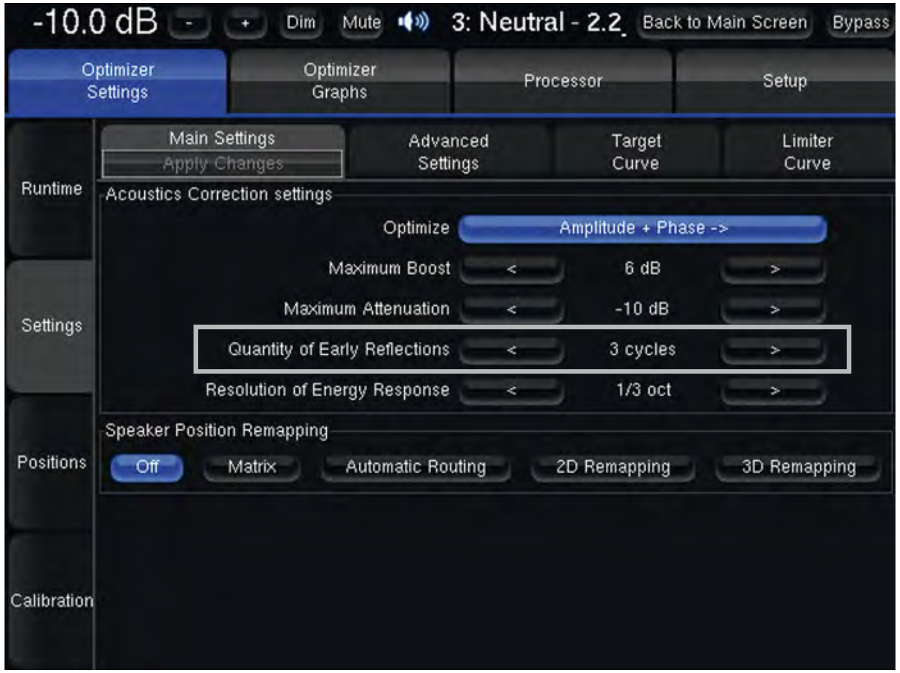
The *Maximum Boost* and *Maximum Attenuation* parameters are used to avoid over-correction.

Excessive boost can cause distorsion and damage speakers and should be avoided unless the amplification + speakers system has sufficient headroom.

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

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

The ***Resolution of the Energy response*** is the resolution of the room correction.

1/3 oct is the default settings.

The minium resolution is 1/3 oct and the maximum resolution is 1/24 oct.

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 suitable 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 most efficient way to edit the ***Target Curve*** is to use a keyboard, be it a physical keyboard or a virtual keyboard within a VNC client.

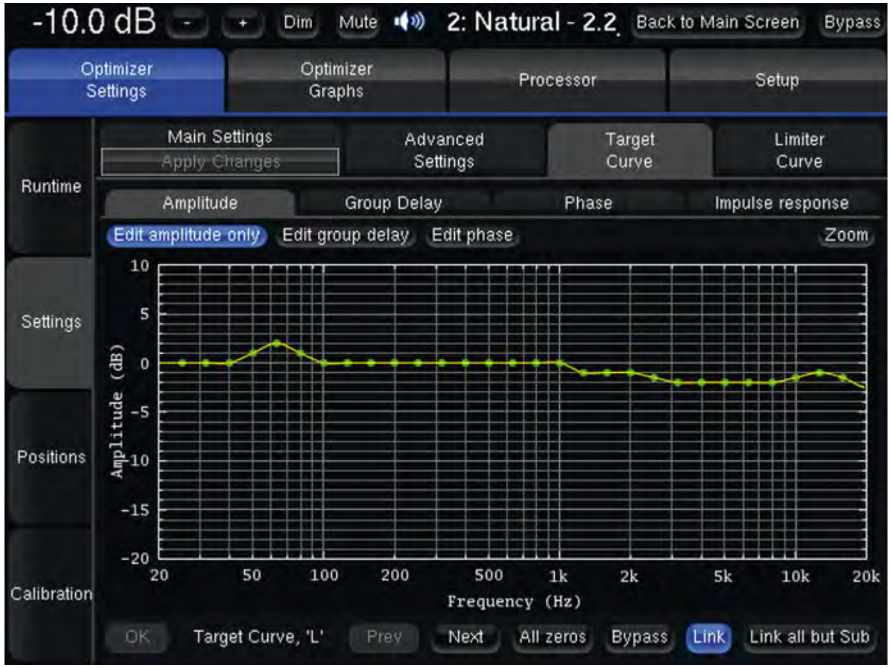
- The ***left and right arrows*** shall be used to change editing points.
- The ***up and down arrows*** shall be used to modify the amplitude of the target curve for the selected frequency with 0.1dB steps.
- ***PgUp and PgDown*** shall be used to modify the amplitude with 0.5dB steps.

The ***Target Curve*** can also be edited with a mouse or a tablet touch screen.

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

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.



Optimizer Settings > Settings > Target 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.

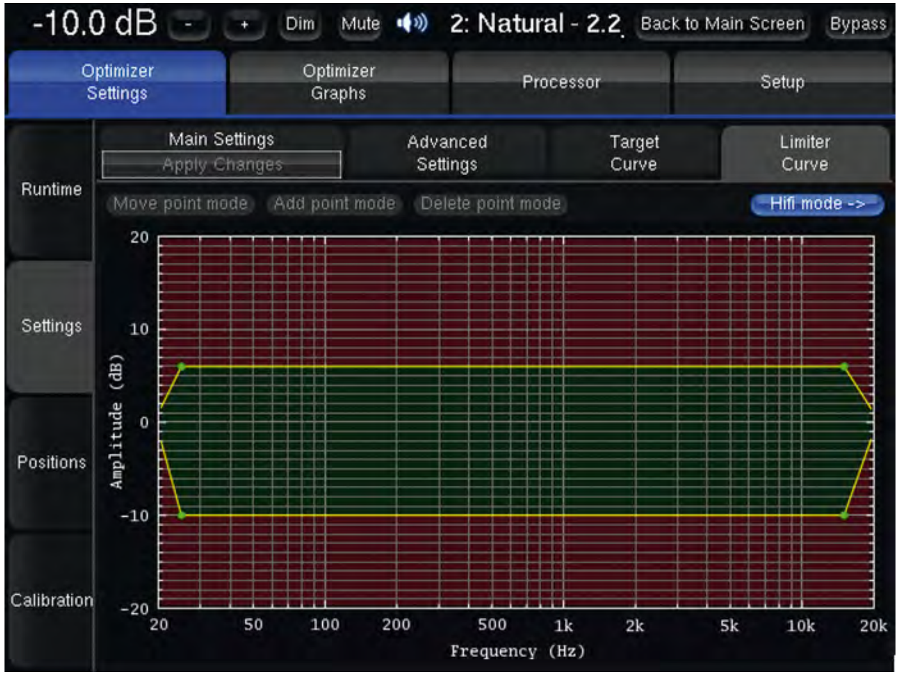
The Target Curve can only be achieved if the required correction comply within the maximum boost and attenuation values.

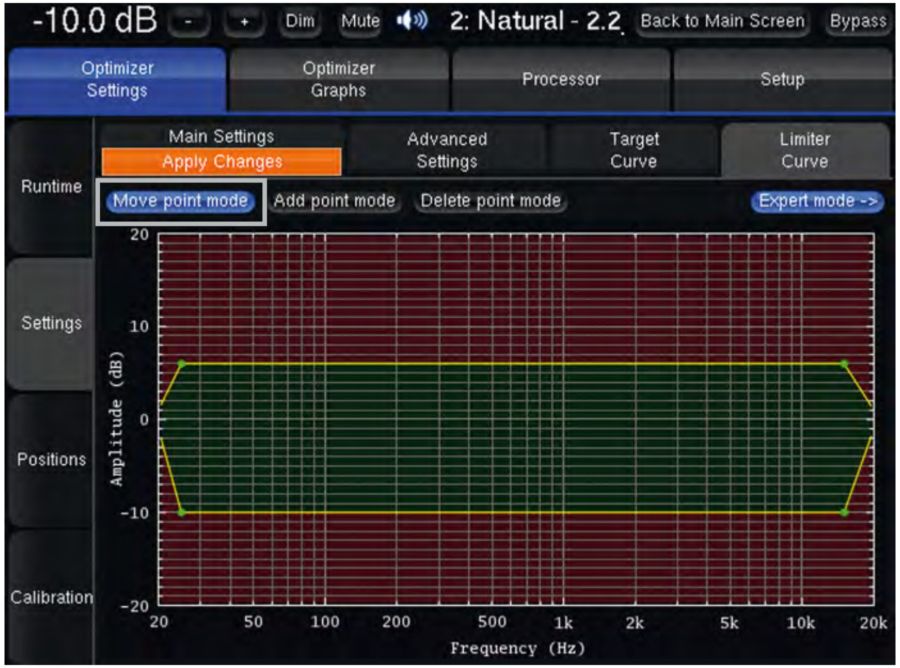
Limiter Curve

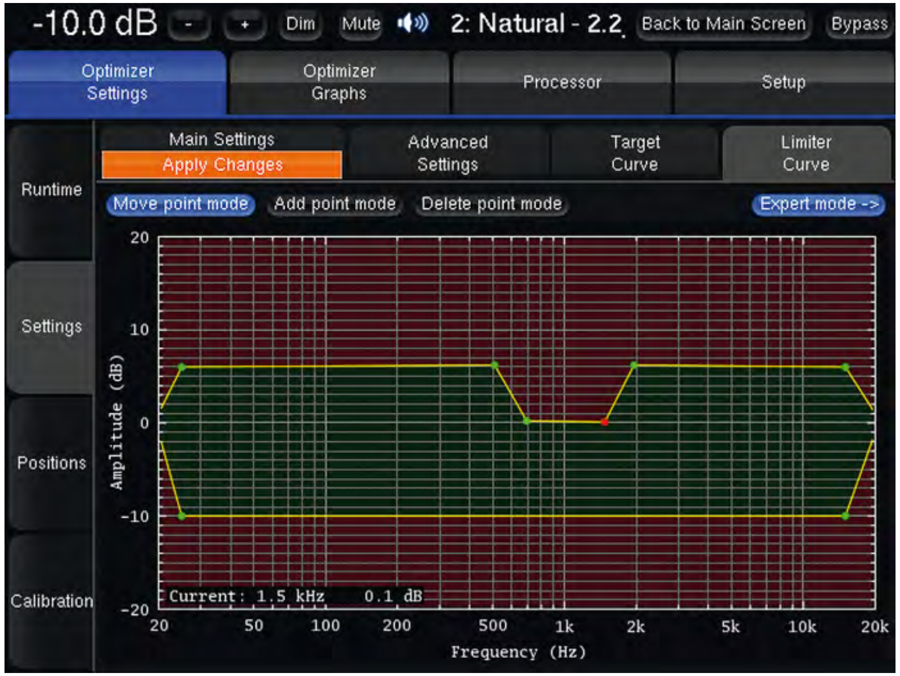
The ***Limiter Curve*** is a frequency-dependant version of the Maximum boost/attenuation parameters.

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 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.

6.8 ABOUT LATENCY

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

- Sampling Rate
- Buffer size
- Advanced audio processing parameters

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

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.

7

MULTIPOINT CALIBRATION

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 occuring 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 large area would work but result 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.

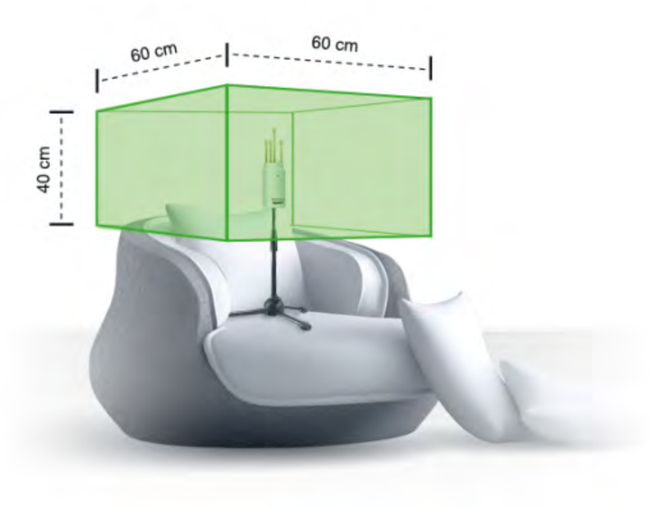
7.2 MEASUREMENT POSITIONS

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.

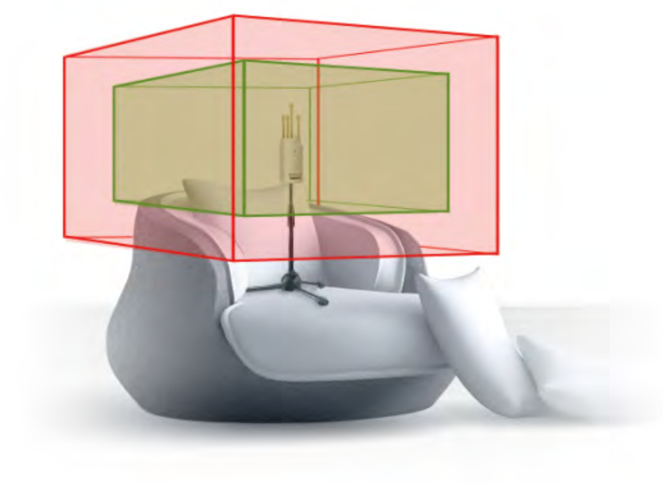
The instructions for the microphone placement concern the main measurement position, called Reference point.

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.

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 ***Calibration wizard*** regarding microphone connection, placement, calibration level and make sure the ***Analog 1 Balanced*** source is selected.

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

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 measurement point 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 mesurement 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).

8

MULTICHANNEL SETUPS

8.1 SOURCES CONFIGURATION

NOTE
Even though the ST2 HIFI has mainly been designed to reproduce stereo content on any 4-speakers layouts, 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***.

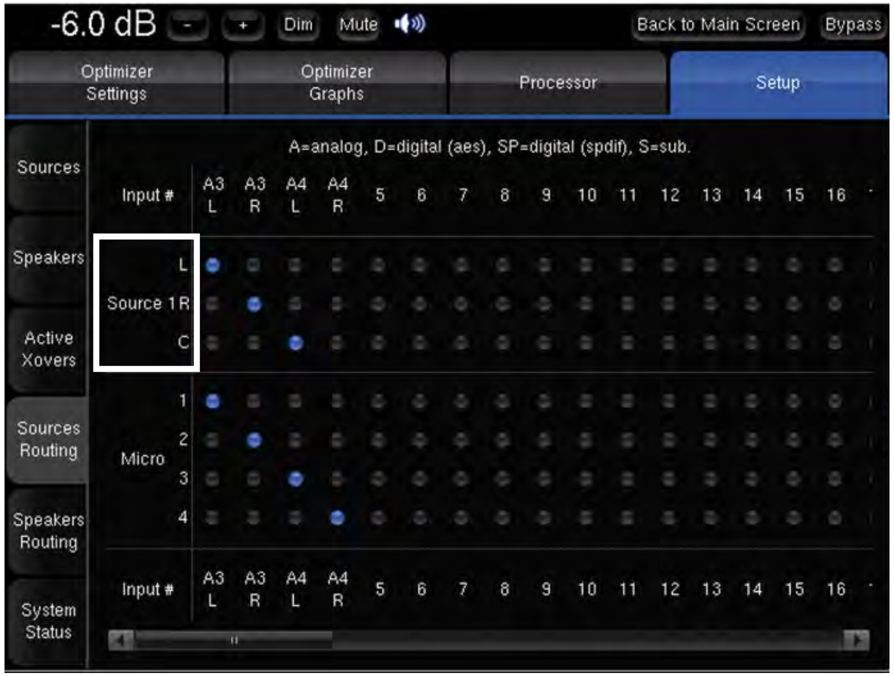
Select the required source format from the list using the ***Next*** and ***Prev***. buttons.

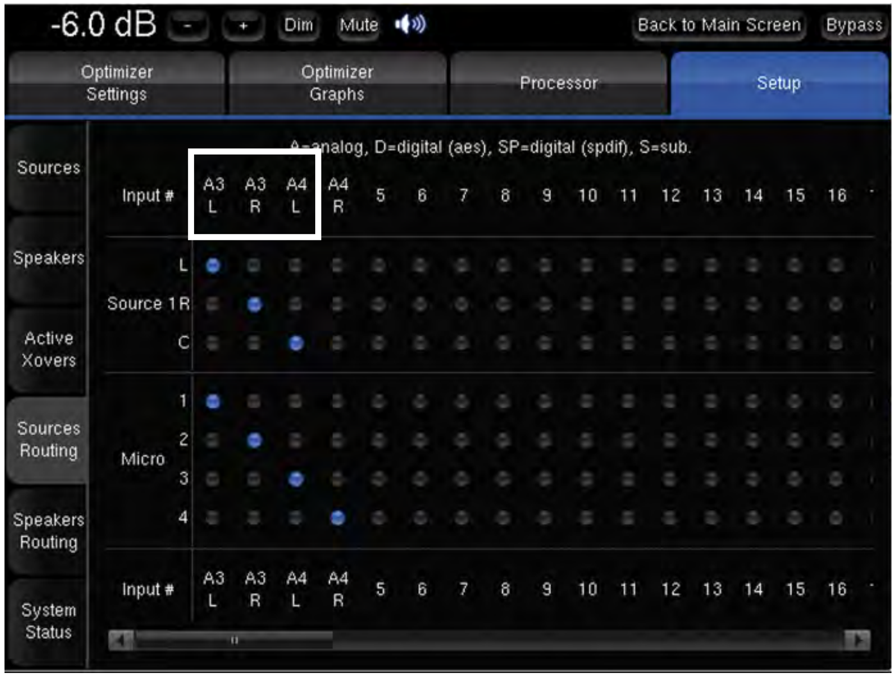
The number of LFE can also be selected using ***-1*** and ***+1*** buttons.

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.

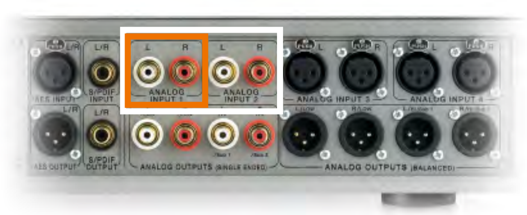
The caption above the routing matrix reminds the different types of inputs available on the ST2 HIFI:

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

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:

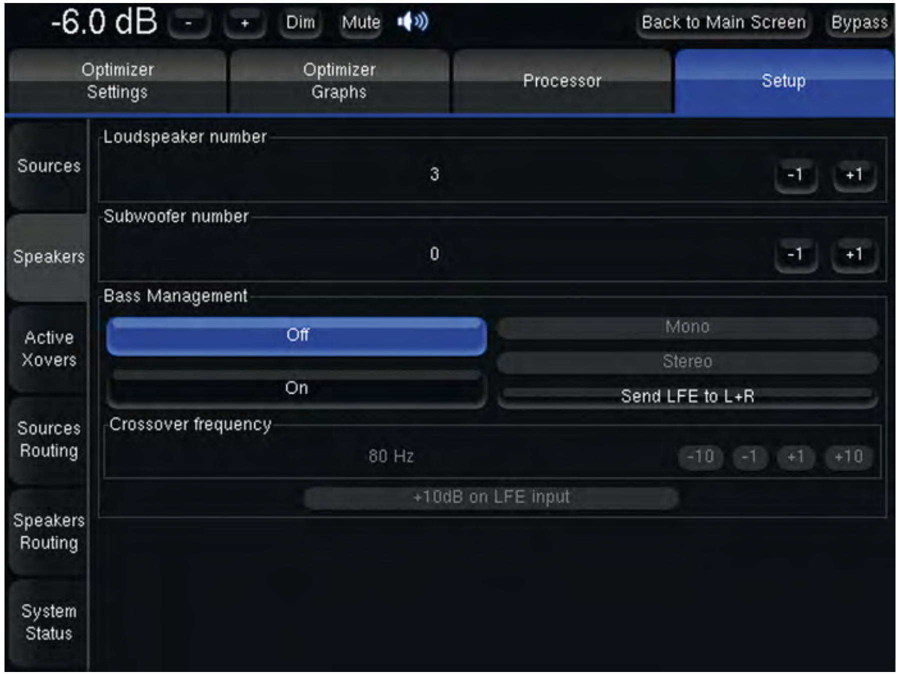
8.3 SPEAKERS CONFIGURATION

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.

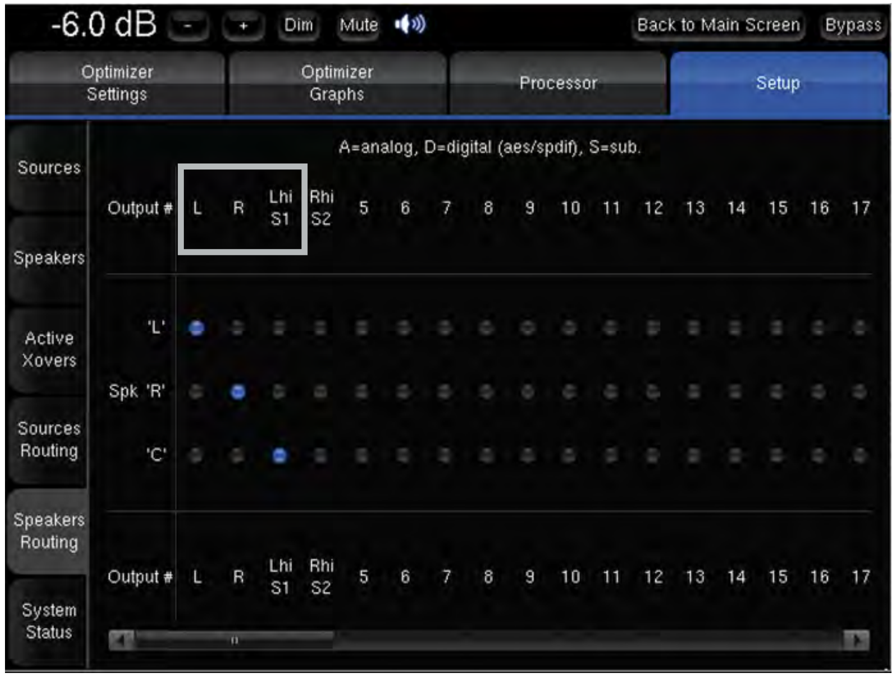
The ***Bass management*** settings can be modified after the calibration.



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

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