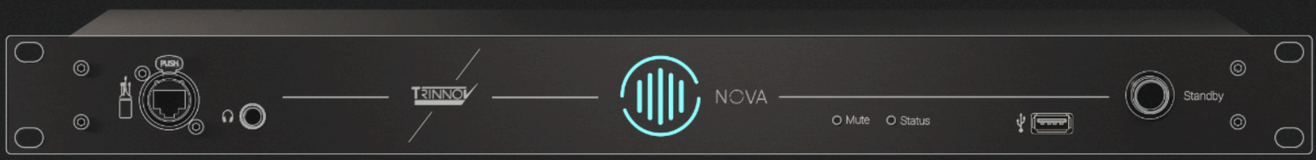
NOVA



USER MANUAL

WELCOME

Thank you for choosing Trinnov.

Our goal is to create products that meet, and even exceed, the monitoring requirements of the most demanding audio professionals. Our solutions enable them to produce the highest quality sound possible for music, broadcast, film, or any type of audio content.

This manual includes the essential information you need to start using your device in your studio setup.

We will help you through every step of the setup process, using NOVA as a room optimizer, a monitor controller, an audio interface, or all of the above.

Safety Instructions

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

2. Follow all warnings 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, block 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 cord from being walked on or pinched, particularly at plugs, convenience receptacles, and where they exit the apparatus. The AC outlet of the power supplier must remain accessible at all times

11. Always replace damaged fuses with the correct rating and type, **T500mA.**

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 voltage outlet other than what your unit is intended for. The power cord must be connected to an earthed mains socket.

15. This product must not be disposed of or landfilled with your other household waste. You are responsible for disposing of all your waste electronic or electrical equipment by taking it to the collection point specified for the recycling of this hazardous waste.

Table of Contents

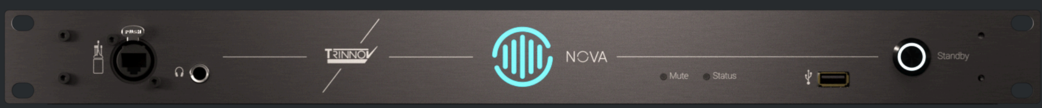
INTRODUCTION

YOUR NOVA

Package content

The package of the NOVA contains the following items:

- NOVA
- Protection bag
- One power cord
- One cat5e network cable
- Printed quick start guide.



Inputs & Outputs

On the front panel are located, from left to right, the etherCON microphone input, Headphones input, and La Remote input.

On the back panel are located from left to right:

- Power input with fuse case
- USB A service input
- Dante and Ethernet ports (machine control)
- Digital SPDIF inputs (Coaxial and Optical)
- Digital AES/EBU output
- 6 XLR balanced analog output
- 4 TRS Jack analog input
- 2 XLR analog input

Software Licenses

NOVA is shipped with **an Optimizer license for two channels** by default.

To optimize more than two speaker outputs simultaneously, additional software licenses must be purchased.

If you have already purchased additional licenses and that these licenses have already been assigned to your unit, please verify that:

- You have downloaded and installed the Trinnov Application on your macOS or Windows computer
- Your computer and NOVA are connected to the same network and the internet.

As you start the application for the first time and NOVA is detected, you will be prompted to update your unit, which will install your additional licenses.

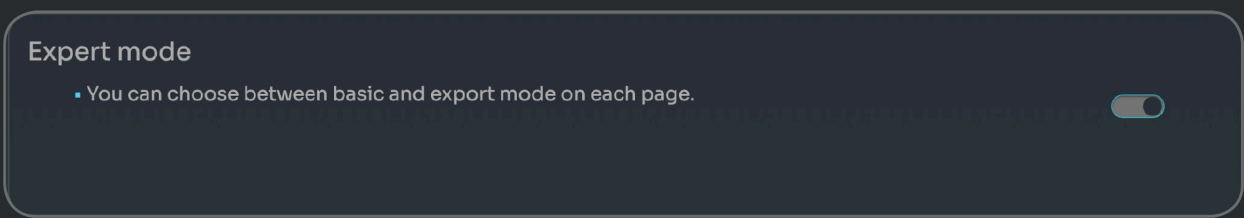
More in System updates and License upgrades.

User Interface

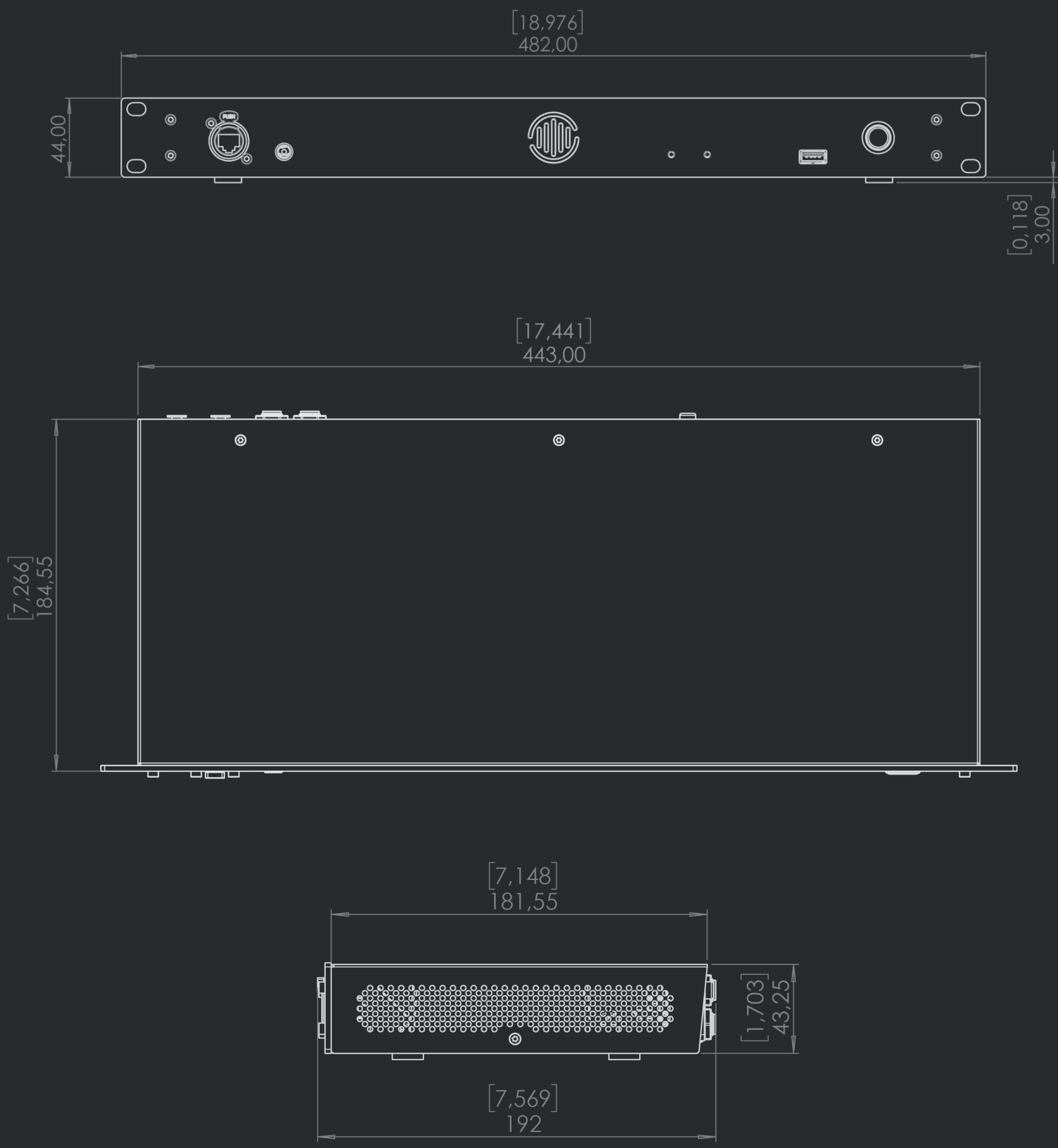
As part of NOVA's development, we designed a new User Interface to deliver a better and more intuitive user experience, including step-by-step instructions to help you configure NOVA and calibrate your monitoring system in the easiest way possible.

The User Interface uses color cues. Any menu item displayed in RED or ORANGE indicates something is wrong or misconfigured in your preset or in hardware bypass mode.

By default, the user interface shows only essential functions, but more advanced options can be displayed on most pages using the Expert Mode toggle button on the top left of the user interface.



Dimension and weight:



NOVA weighs 2.6 kilograms or 5.7 pounds.

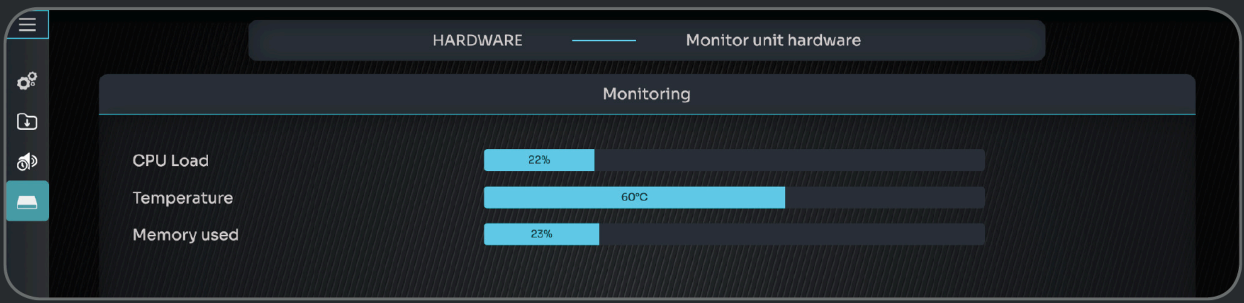
Important information

The hardware research team has designed a fanless unit to keep your environment as quiet as possible. They successfully achieved this by using the top lid of NOVA for heat dissipation.

Don't be worried if NOVA is getting hot, that's normal behavior.

All heat dissipation systems are linked to the lid; be careful not to put **heavy weight** or **hard pressure on it** . Improper handling of the lid can result in a broken motherboard.

You can also monitor the current temperature of NOVA.



About this User Manual

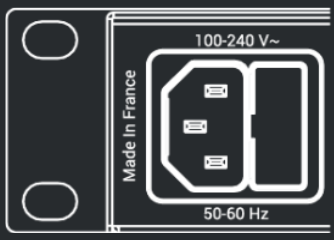
This manual is split into different sections:

- **Configuration and Start-Up** will get you through the essential things you need to know to get up and running
- **Notions** cover generic concepts and explain more specific features
- **Help** is dedicated to troubleshooting

CONFIGURATION AND START-UP

Connect NOVA

Power connection



NOVA's internal power supply (PSU) is connected via an IEC C14 inlet at the back of the device.

This high-performance switch mode power supply accepts 100V to 240V AC.

NOVA's PSU has a built-in inline filter, is fully regulated against voltage fluctuations, and suppresses mains interference.

Network connection

For the initial configuration, NOVA needs to be connected to your local network. This is how the Trinnov Application and the processor communicate to provide control over the unit.

An internet connection is also required to download new software versions and additional software licenses. We therefore strongly recommend connecting NOVA to the Internet before its first use.

A network connection is also required in most cases for daily operation.
Except for NOVA owners also using La Remote, which only requires a USB connection to NOVA (or to the computer running the Trinnov application (MAC OS app only, but a network connection is then needed too).

Product description

The rear panel of NOVA features two RJ45 connectors labeled primary and secondary, which provide standard Ethernet connectivity. The supported link speeds are 100 Mbps and 1000 Mbps, and cable lengths of up to 100 m are supported when using Cat 6e.

The default configuration makes NOVA act as a network switch. This means that you can control NOVA or use the Dante audio connectivity from either port.

Network connection

In case NOVA is connected to the local router or internet service provider box with default parameters, no specific configuration should be required.

No connection should be required if you connect NOVA directly to the computer running the Trinnov Application unless network settings are set manually in either device.

For more advanced network integration, please contact a professional administrator and read this manual's Network section.

Important notes:

- If you plan to use Dante audio over IP, ensure the network link is 1 GB/s.
- An etherCON port is present on the front panel. This connector is dedicated to the Trinnov measurement microphone and is incompatible with computer network hardware.
- If you connect NOVA to your local network via a switch or router, make sure you **do not connect both Ethernet ports at the same time** to the same network device. This will lead to a network infinite loop and loss of connectivity.

Power on/off

Power On

The NOVA has a power AC inlet at the rear panel and a standby switch at the front.

Perform the following steps to power on NOVA:

1. Ensure that the power inlets are correctly connected to a power source.
2. Press the front panel Standby button to boot the device.

Important note: if the front panel LEDs do not light up after the second step, please check the NOVA fuse next to the AC socket.

Power Off

Perform the following steps to power off NOVA:

1. Press and hold the front panel Standby button for 2 seconds. You can release it as soon as the button color changes to red.
2. The Standby button changes to yellow. Wait for all front LEDs to switch off.
3. You can then safely unplug the rear power inlet if needed.

Trinnov Application

Download the free Trinnov App from our website, https://www.trinnov.com/en/support, to discover and take control of your NOVA device.

This software **must be installed on a computer connected to the network** NOVA has been connected to.

NOVA is controlled by the Trinnov App.

macOS APP

WINDOWS APP

The app is also available on IOS to control your NOVA on an iPhone (volume, mute, preset selection) and an iPad for configuration.
You can download the application using this QR code.

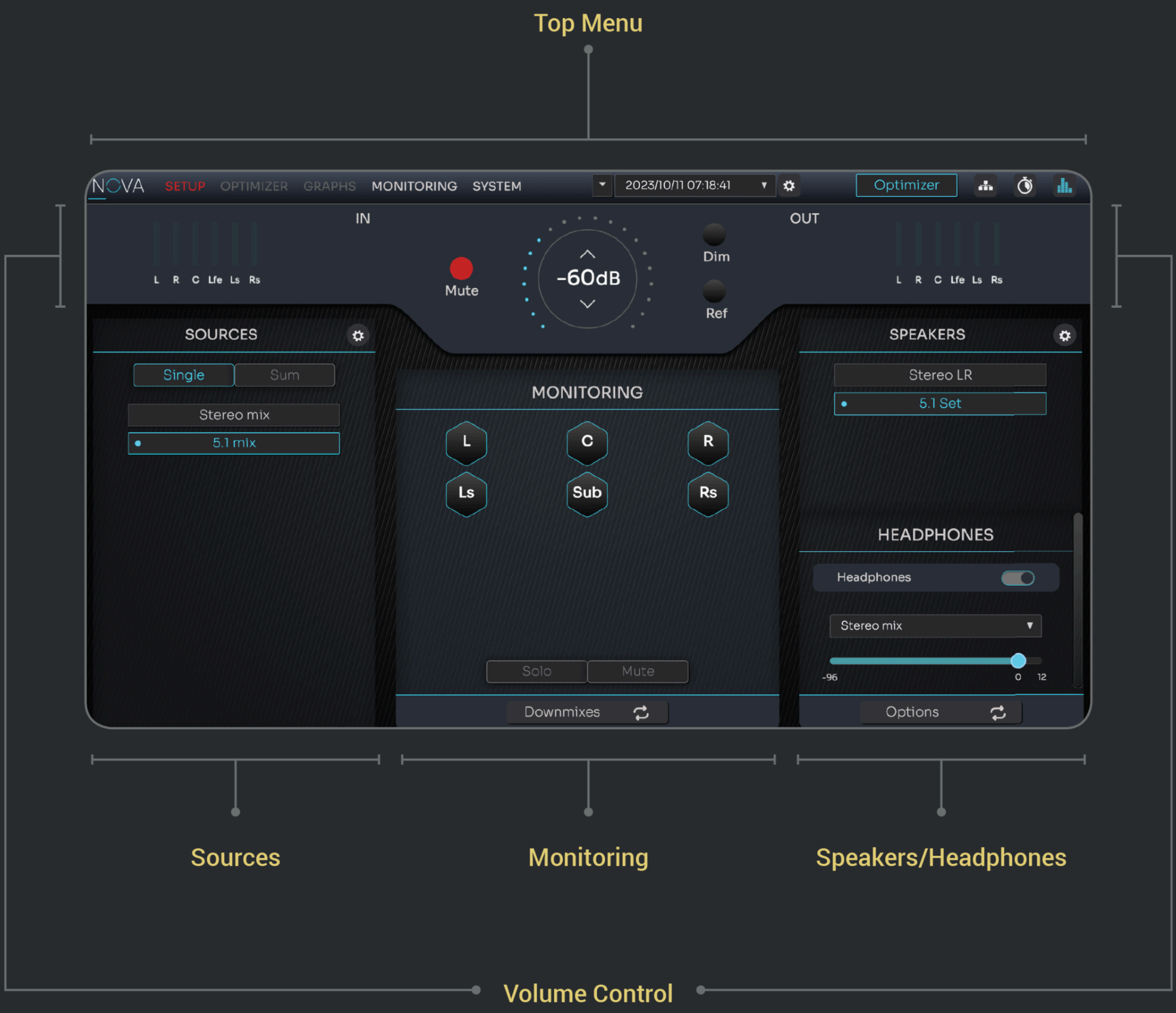
IOS APP

You can customize the name of your device in the Trinnov App to suit local configuration and simplify day-to-day identification.

Note that if you rename the unit in the Dante controller, this name will also be used in the Trinnov App.

Home Page

This new, completely redesigned homepage shows everything at one glance.



Top Menu

The top menu is always visible and includes:

NOVA **This is the home page shortcut.**
Clicking it will always bring you back to the home page.

Main menu items

SETUP OPTIMIZER GRAPHS MONITORING SYSTEM

Preset selection and management

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Optimizer Status.

Optimizer Blue means Activated

Optimizer Orange means Bypassed

Quick access to the configuration of :

Network

- Click this icon to access network settings.
- Note: You will lose the interface momentarily after a network change.

Clock

- If Nova does not play any sound, click this icon to check your audio clock settings.
- The audio clock settings are global and affect all presets.
- Calibration only works at 48kHz.
- You may change your clock after calibration is complete.

Volume

- Click on this icon to expand/collapse the volume control panel.
- This volume control panel also includes input and output meters.
- Click on the meters to display them in full screen.

Volume Control

Below the top menu and in the middle horizontal section, you will find the input and output meters, as well as the volume control panel.
Click on the input or output meters for an enlarged and more detailed meter representation.

The Volume control panel includes the main volume knob, mute, dim, and reference call-back buttons. Note that the Dim attenuation and reference volume level can be set in the system settings.

Sources

All the sources you have configured via the configuration wizard explained further down in this manual will be listed and available on the right-hand side of NOVA's Home Page.

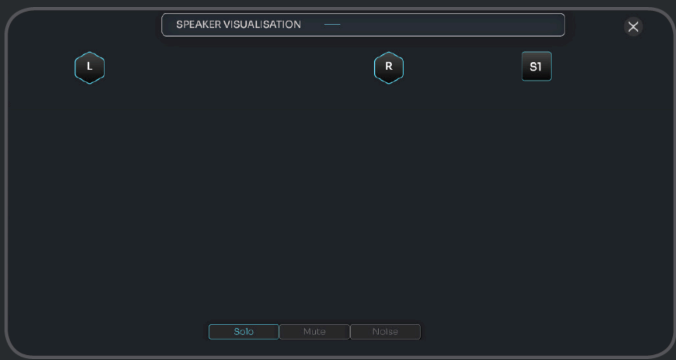
You can always reconfigure them after you have run the wizard by using the gear icon located in the top-right corner of the Sources selection panel.

Monitoring

The middle section of the home pages includes basic monitoring controls such as individual speaker mute and solo.

But with the flip button located at the bottom of this section, you can also access downmixes functions.

This item is called Speaker Visualisation. From this page, you can play pink noise, solo or mute .



Speakers

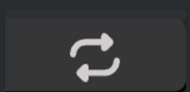
The right-hand side of the home page hosts the output control panel.

This panel includes speaker set selection, headphones level, and source control.

As for the source selection panel, you can find a gear icon in the top-right corner that will bring you straight to the output configuration. Please go to the Output Selection section of this guide for more detailed information

Important note: reconfiguring speakers will require a new acoustic calibration.

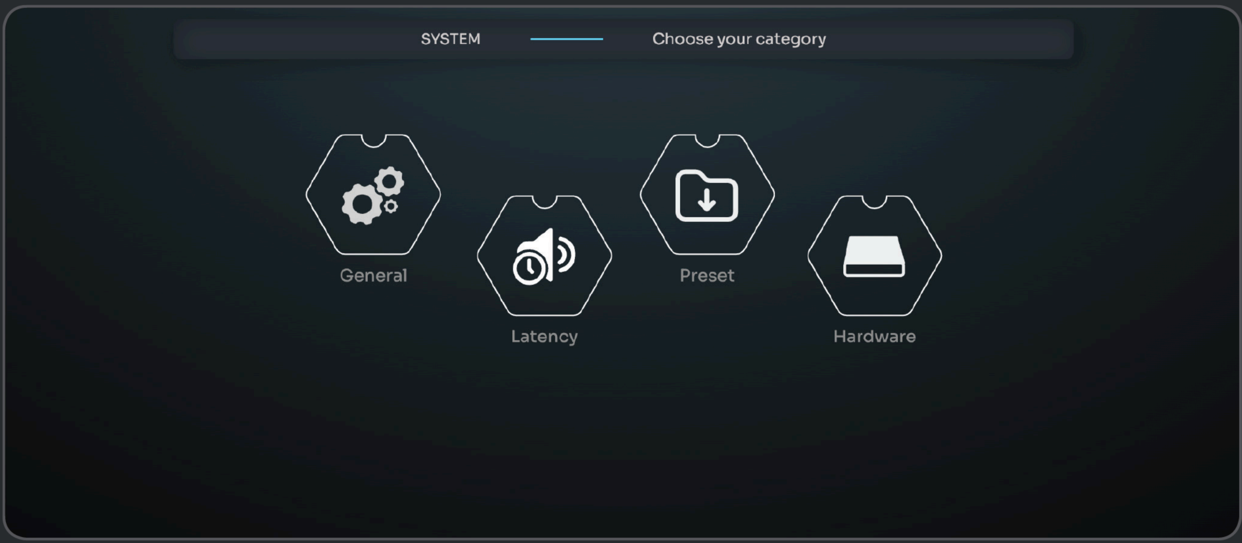
Flip buttons



You can find two flip buttons at the bottom of the monitoring and output control panels. These flip buttons give you access to useful yet less essential features related to each section.

System Settings

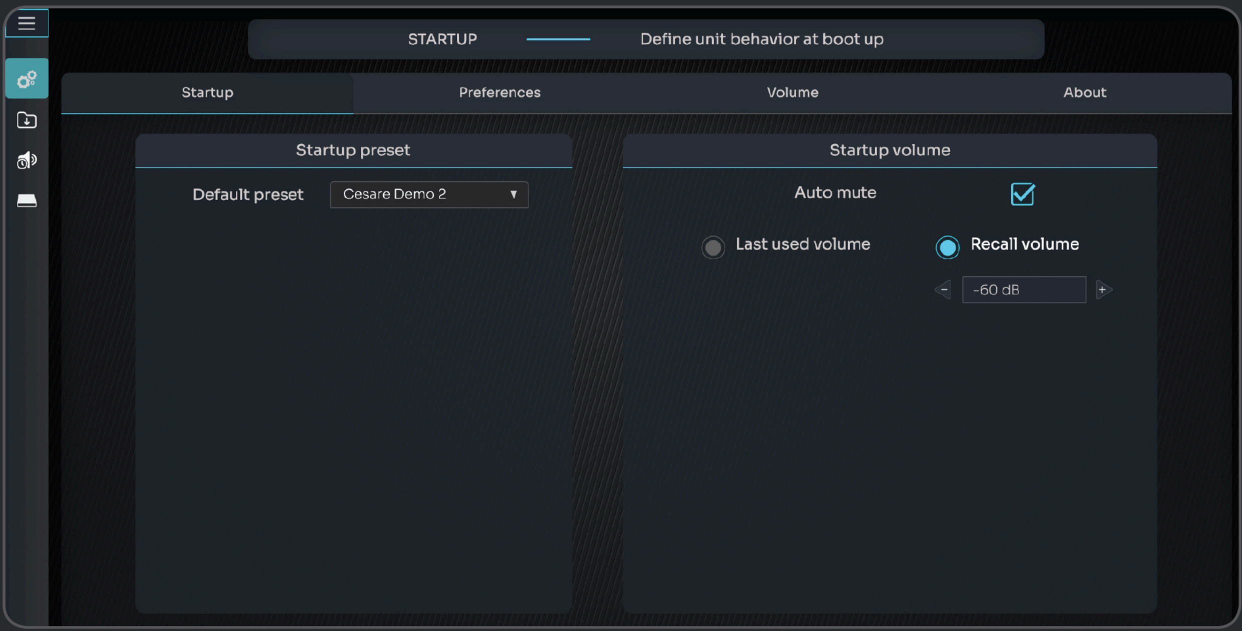
By clicking on SYSTEM on the toolbar, you will access the configuration of your settings.



General

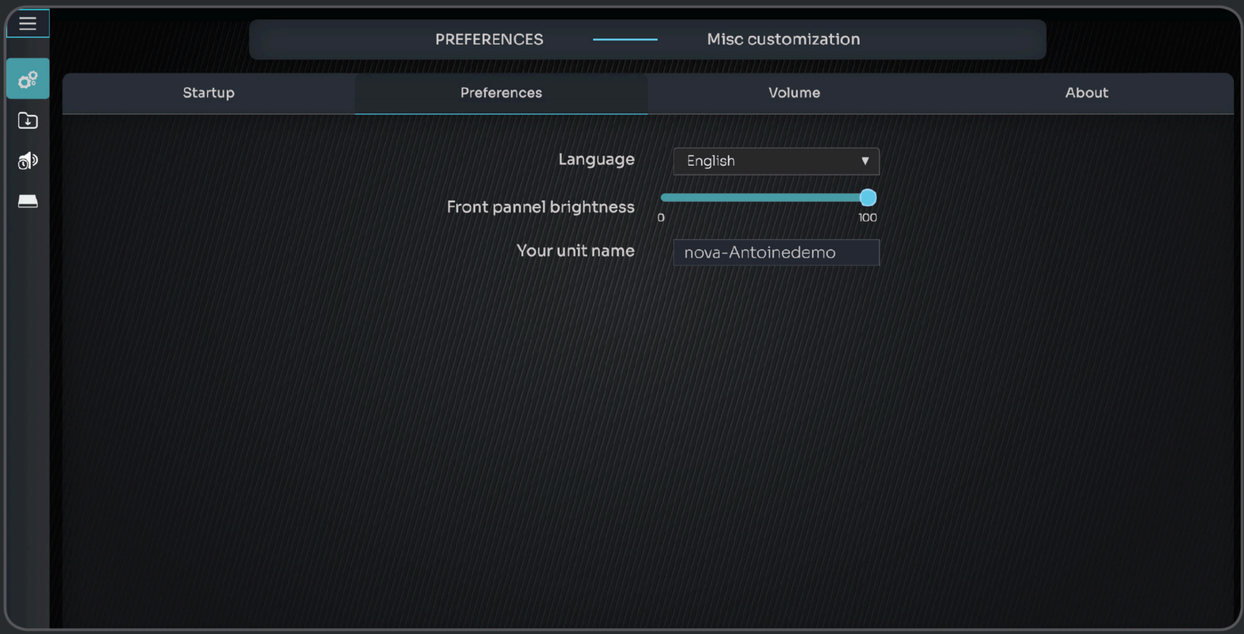
In the general section, you can adjust several parameters.

Startup



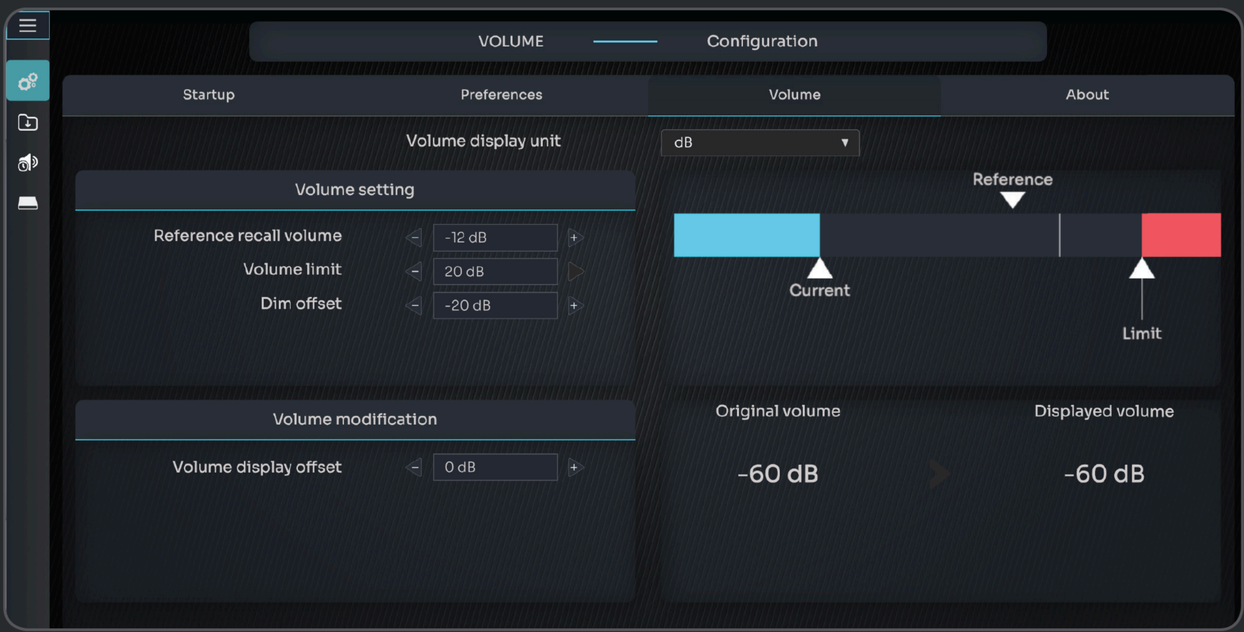
Default preset : the current preset loaded at startup.
Auto mute : The unit will automatically mute at startup.
Volume : You can either boot at a fixed volume or resume the level used during the last session

Preferences



Language: Select the main language of NOVA (only English available currently).
Front panel brightness: Select the preferred LED luminosity.
Your unit name: Apply a specific name for NOVA.

Volume



In this section, you can configure all the volume parameters.

Volume display unit: Define how the speaker level is displayed

- **dB:** Unreferenced, you can adjust the volume between -60 dB up to +20 dB.
- **dBC:** the level value is relative to the current calibration level (displayed on-screen once you have calibrated the speaker set). It is expressed in dB SPL, from 0 to +115.

Volume setting:

- **Reference recall volume**: defines the speaker reference level value (recalled by the Ref button on the home page).
- **Volume limit:** defines the maximum speaker level.
- **Dim offset:** defines the amount of dim applied to the speaker level.

Volume modification: Level display offset affects the displayed level only and does not impact the level itself. It cannot be stored in presets.

About

Critical information about the software version, serial number of NOVA, and the number of licenses installed.

Latency

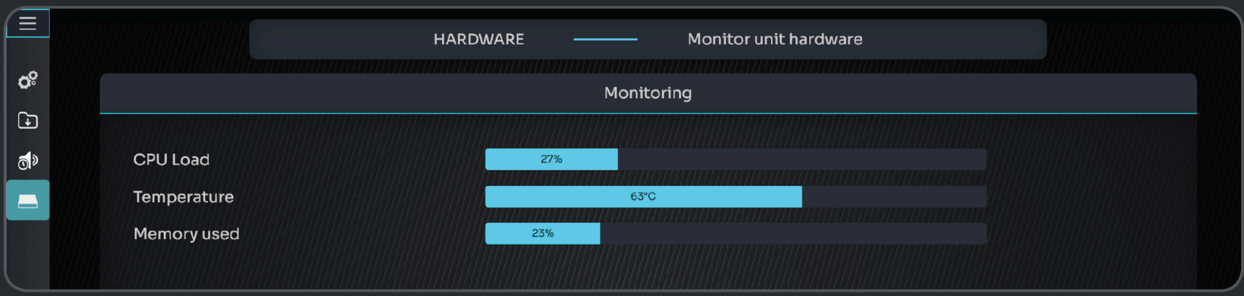
Master delay : can be used to apply an additional delay to all channels and all presets. It is not stored in the preset.

The following latency information is available:

- Processing Latency corresponds to the latency of the processor algorithms. It can be modified by changing the Optimizer settings (Amplitude + Phase has higher latency than Amplitude only).
- Master + Relative Delay is the sum of the master and relative delays of the Processor/Master page.
- In-out Delay is the sum of the Processing Latency and the User-defined delays. For the furthest speaker, it corresponds to the system's delay from input to output.
- Acoustic Delay corresponds to the distance of the furthest speaker to the measurement point. When Time Alignment is activated, all the other speakers are time-aligned to the furthest speaker.
- Total delay at the measurement point is the delay from one input until the sound reaches the measurement point.

Hardware

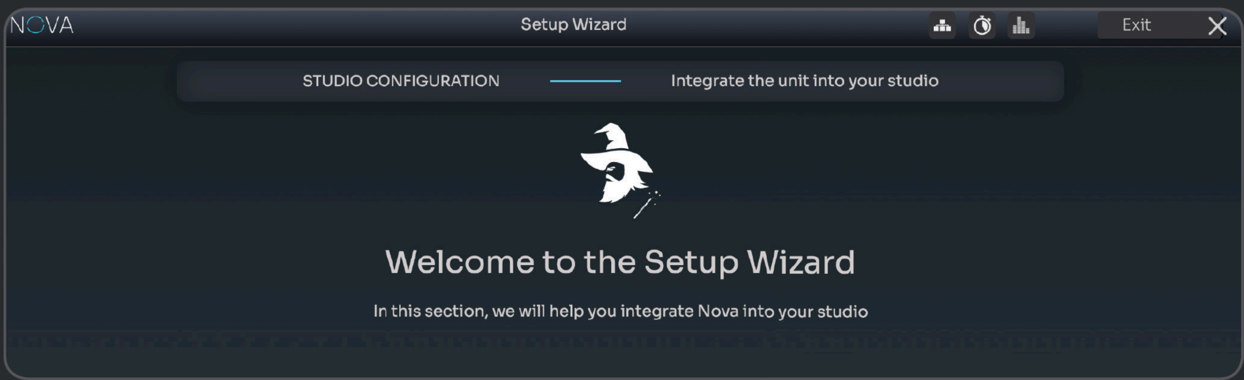
This is real-time information about NOVA.



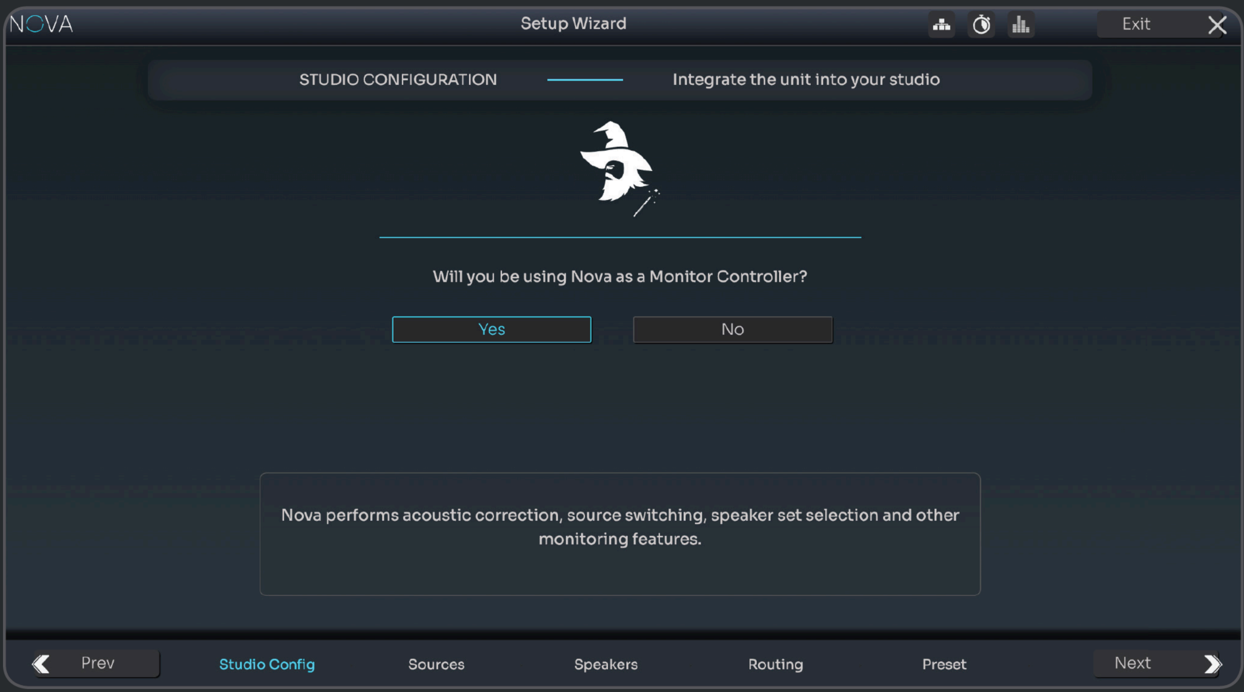
CPU load: Percentage of CPU usage in real-time. An overload of CPU can result in audio drops.
Temperature: Current temperature inside NOVA. if the temperature reaches more than 85 °C (185 °F), this can result in audio drops or the powering off NOVA.
Memory used: Percentage of the current memory used .

Quick Setups

Configuration Wizard



The Setup Wizard will start automatically the first time you use your NOVA or anytime you create a new preset.



The configuration Wizard takes you through different steps to guide you along the configuration of NOVA.

The different successive steps are indicated at the bottom of the screen.

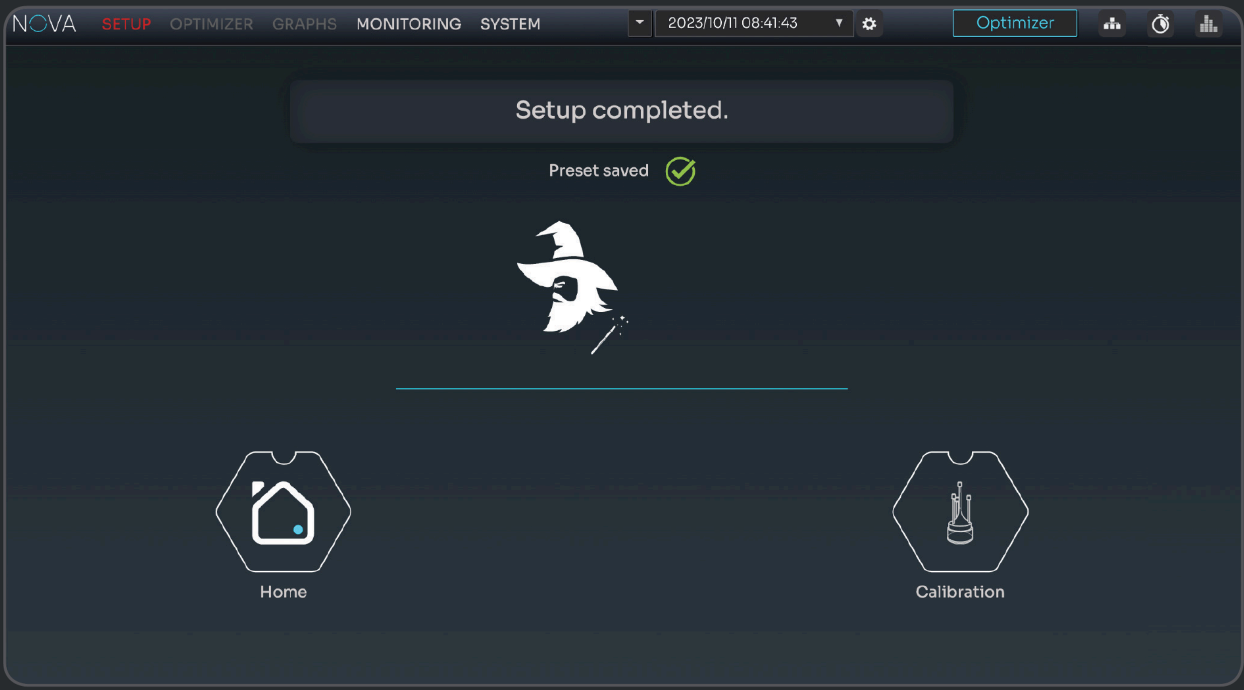
You can navigate through each step using the Next and Previous buttons unless missing information prevents NOVA from letting you through the next step.

The Configuration Wizard includes contextual information and descriptions to help you understand each step.

No Monitoring mode

No Monitoring mode allows routing inputs to outputs within NOVA and using the optimizer on these channels, making the concept of speaker sets irrelevant. This is useful if you're using an external device, such as a console or monitoring controller, to switch between speaker sets.

The no-monitoring mode is available in the configuration wizard, allowing direct connection of input to output connectors without format or channel settings. Names displayed will be connector names, not channel names. Note that some features, like direct outs and downmixes, are unavailable in this mode.



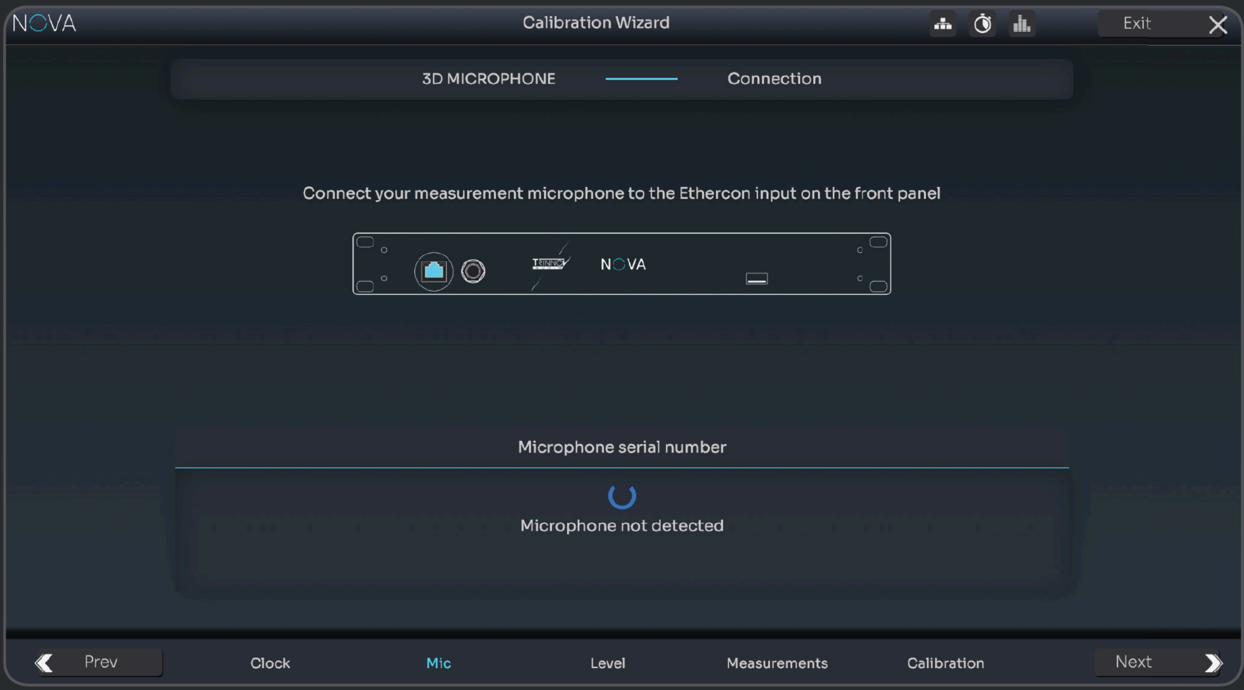
After completing the last step of the configuration, you can go back to the home page and start using NOVA. This allows you to play audio from any of the configured sources through any of the configured outputs.

However, if you cannot wait to enjoy the benefit of our Optimizer technology, you can jump straight to the calibration Wizard.

We highly recommend using the Calibration Wizard to get the full Trinnov experience.

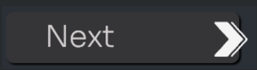
Calibration Wizard

The calibration Wizard follows the same principles as the configuration Wizard and helps you achieve the first acoustic optimization of your monitoring system.



As an example, you can see the second step of the calibration Wizard where you are prompted to connect your measurement microphone to the etherCON connector located on the front panel of NOVA. Once connected, NOVA automatically detects the microphone and imports its compensation profile, ensuring the most accurate acoustic measurement.

As for the Configuration Wizard, use the Prev and Next buttons to navigate through each step.



Once calibration is complete, a preset will be automatically created and saved.

NOTIONS

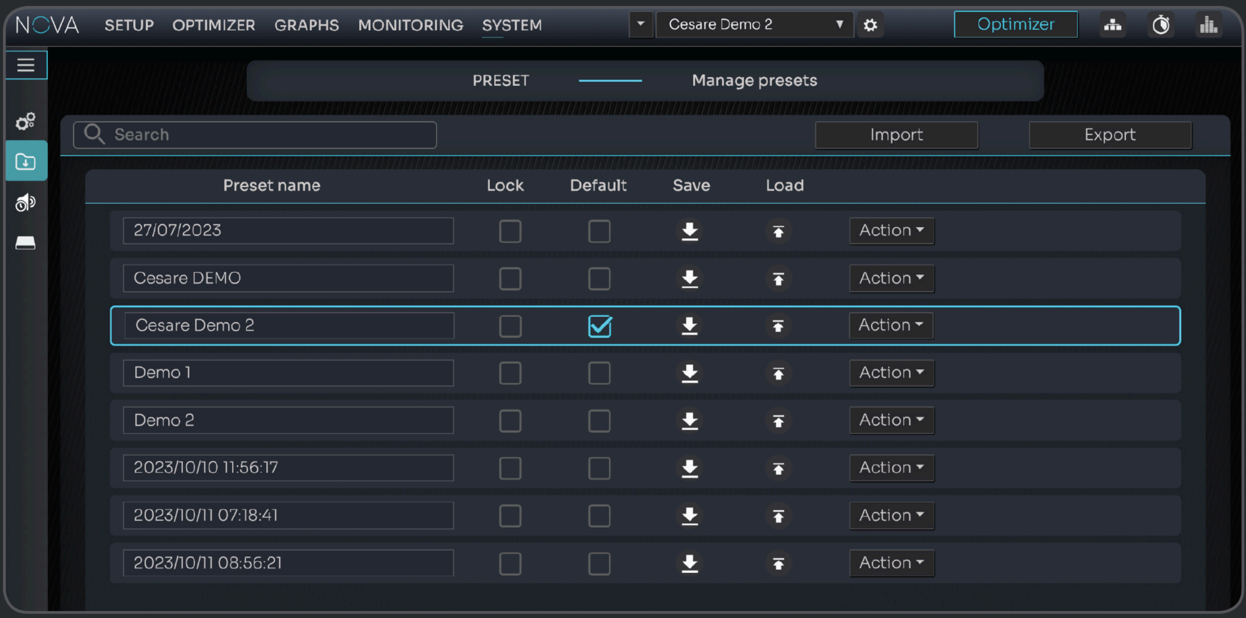
Read this section if, despite running NOVA's initial configuration and calibration, you still need to understand general concepts and more specific features in detail.

Presets

NOVA configurations are saved in presets, which include:

- Source and Speaker configuration
- Input & Output Routing
- Acoustic measurements and filters
- Acoustic optimization settings

You can save and reload presets from the top menu, but a dedicated PRESET section is also available in the SYSTEM menu.

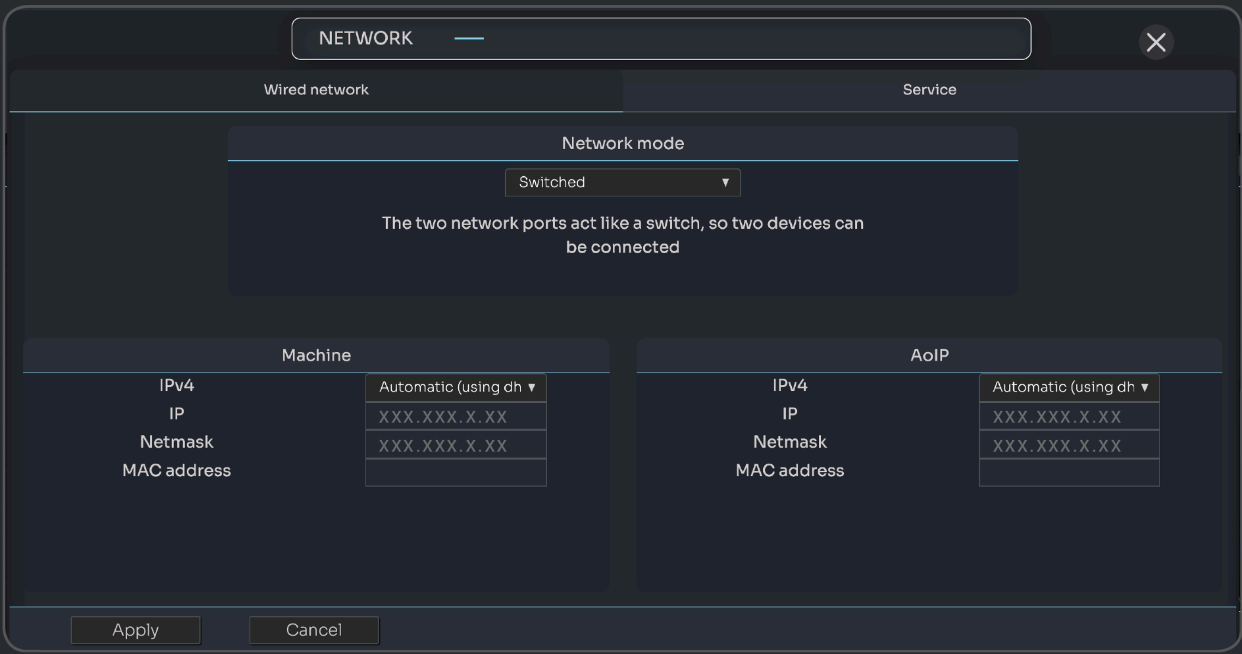


On this page, you can manage your presets with more flexibility and options:

- Load & Save: also available from the Top menu
- Duplicate and rename: useful to create an alternative preset without starting from scratch
- Import or export

You can also Lock presets from being accidentally deleted, as well as set the default preset you want loaded as the start-up default.

Network



A checkbox is now available in Network/Service to disable the connection between the unit and Trinnov servers. If no preset exists, this checkbox will also appear in the setup wizard's first step.

Mode

- **Switched:** The two network ports are equivalent, and are linked with the internal switch, so several devices can be daisy-chained.
- **Redundant:** Two parallel network connections run independently on two subnets to provide multiple paths
- **Isolated:** The first port is dedicated for Dante AoIP and the secondary is for machine control

Information

- **Status:** indicates the connection status of the Ethernet
- **MAC address:** indicates the MAC address of the network interface of your NOVA
- **IP address:** indicates the IP address of the Ethernet interface of your NOVA.
- **Netmask:** indicates the netmask of the local area network joined as a DHCP client via Ethernet.
- **Gateway:** indicates the address of the Gateway for network interconnection.

Automatic (using DHCP)

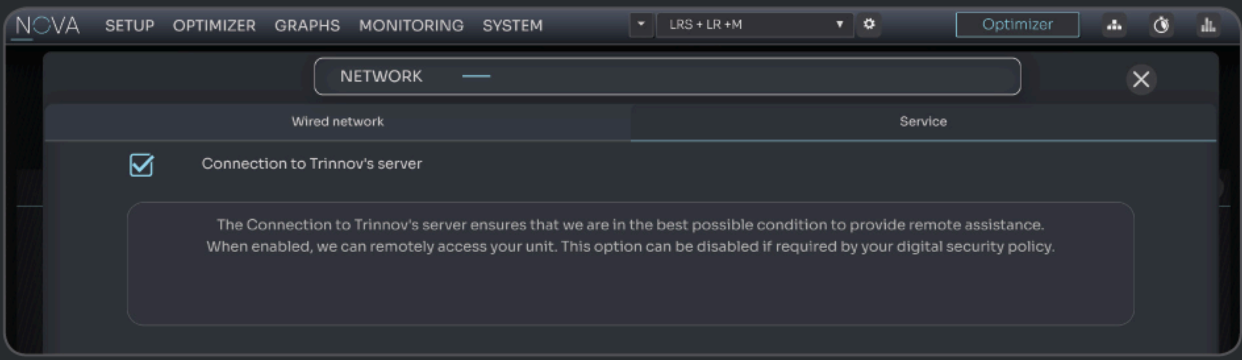
Dynamic Host Configuration Protocol (DHCP) is a network protocol used to automate the process of configuring devices on IP networks. With this mode selected, if no DHCP server (likely your FAI router box) is answering, NOVA will fall-back to a link-local connection to allow direct control with your computer.

Fixed IP

The IPv4 address, Netmask, Gateway, and DNS option can then be edited manually. Applying new settings with the Apply button is required for every change.

Trinnov Audio Server

This is not an obligation, but as Trinnov Audio can perform remote assistance, having a machine connected to the internet is strongly recommended. To perform maintenance, your unit must be connected to the Trinnov Audio Server.

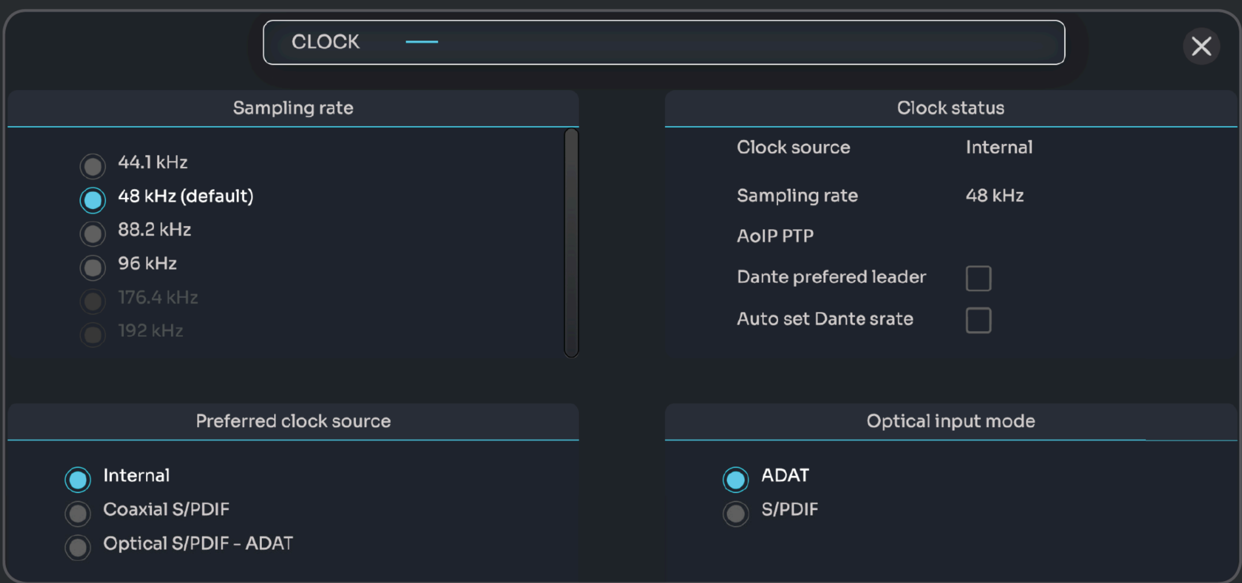


Port 22 is the port used by your unit to access the Trinnov Audio Server. If you want Trinnov Audio to assist you remotely, please enable the outgoing port 22.
You can check the port in your network configuration or ask your network provider.
Generally, port 22 is opened, but if the Connection to the Trinnov audio server is not working, it means that port 22 could be blocked.

A checkbox is now available in Network/Service to disable the connection between the unit and Trinnov servers. If no preset exists, this checkbox will also appear in the setup wizard's first step.

Clocking

The audio clock source **must be defined** in the clock configuration page, which is always available via the clock icon located on the top right-hand side of the top menu.



The normal behavior of NOVA is to switch automatically to the right clocking mode based on the connected sources:

- If no digital source is connected, NOVA will automatically be set to internal and use its own internal clock as a reference.
- In other cases, NOVA will follow the external clocking signal it detects from one of the digital source sources.

If several sources are connected to NOVA and can be used as an external clock reference, you may want to choose your **preferred clock source**.

If NOVA locks to an external clock source, it will automatically follow the external clock source's sampling rate.

If NOVA is using its internal clock, you can then select which **sampling rate** you want to use, from 44,1kHz to 192kHz. 48kHz is the default sampling rate.

The **Optical Input mode** panel lets you decide whether you wish to use the Optical input as S/SPDIF or as an ADAT input. Note that in ADAT, the number of available channels is directly dependent on the sampling rate you have selected in the source (the higher the sampling rate, the fewer channels available).

ADAT SMUX 96kHz operation now supported. In ADAT mode, the SMUX ratio is now displayed alongside the sampling rate.

The **Clock Status** panel in the upper-right corner of this page indicates what source mode (internal or external) is currently selected and at which sampling rate NOVA is running. This section also includes information about Dante. For more information about Dante and Audio over IP, please go to the corresponding chapter of this guide.

Important notes:

- **To run a calibration, it is mandatory to set the sample rate to 48kHz**, whether NOVA is using its internal clock or if it is synced to an external clock.
- The higher the sampling rate, the shorter the latency, but the higher the CPU consumption and heating. This is why we normally recommend working at 48kHz, which is the most used sampling rate.

Output Selection

Speaker Sets

Speaker sets are divided into two categories:

- Optimized Speaker Sets: acoustically optimized with Trinnov Optimizer technology
- Non-optimized Speaker Sets: non-optimized but aligned in level and delay with the Optimized speaker sets to ensure a smooth transition as you switch between optimized and non-optimized speaker sets.

Both require calibration and should be treated similarly during the calibration process.

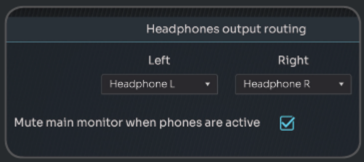
To create multiple speaker configurations using the same loudspeakers (ex: 2.0 and 2.1 with the same left and right speakers), you should add a separate Speaker Set for each configuration and then assign the same physical outputs to the shared speakers. Declaring the same speakers in multiple speaker sets does not consume additional Optimizer licenses.

You can also use presets to increase the number of Speaker Sets available and overcome the limitation of licenses. (Ex: preset 1: speaker set 1 & preset 2: speaker set 2) but bear in mind that each preset will require its own calibration. In addition, note that it takes longer to switch between presets than it does to switch between speaker sets in the same preset.

Headphones

NOVA has a dedicated Headphone output on the front panel of the unit. This output uses specific DACs which have been optimized for headphone listening.

Click on the ON/OFF button to activate/deactivate the headphones.



You can now automatically mute the main monitor when headphones are in use.

Use the drop-down menu to select the monitor source – you can choose any of the sources. If the headphones source "follows monitor" is selected, then the headphones will automatically follow the speaker source selection.

Click and drag on the horizontal slider to increase or decrease the headphone level. If you hover the cursor over the slider, then a text read-out of the level is displayed.

You can double-click on the slider to reset the headphone level.

You can also configure any output for headphone in monitoring > headphones menu.
There you can select auto-mute when headphones are in use.

Monitoring

The following page can be found in the MONITORING item from the top menu.

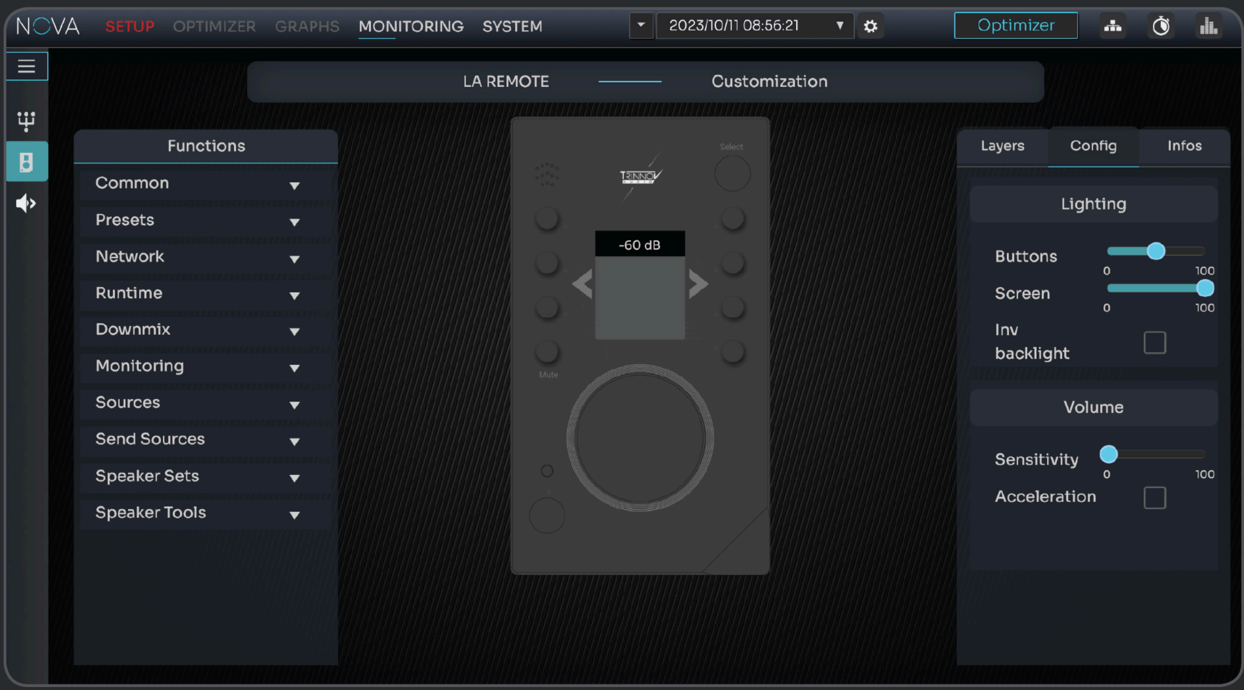
La Remote configuration

Designed for Trinnov processors exclusively, La Remote is fully customizable.

La Remote features eight user-assignable buttons, a fixed mute button, a main volume knob, and an additional selection knob.

Multiple layers can be set up to change the function of each assignable button.
Navigating from one layer to another is done via the selection knob located on the top right of La Remote.

The selection knob can also be pushed for additional functions.
As an example, you can push and turn the selection knob to change the headphone level.



La Remote configuration can be done from the NOVA interface in the MONITORING menu. From this page, you can assign any functions of NOVA to any assignable buttons with a simple drag-and-drop interface. This is also where layers are managed.

Furthermore, you can customize what the screen displays and other settings, such as the volume knob sensitivity.

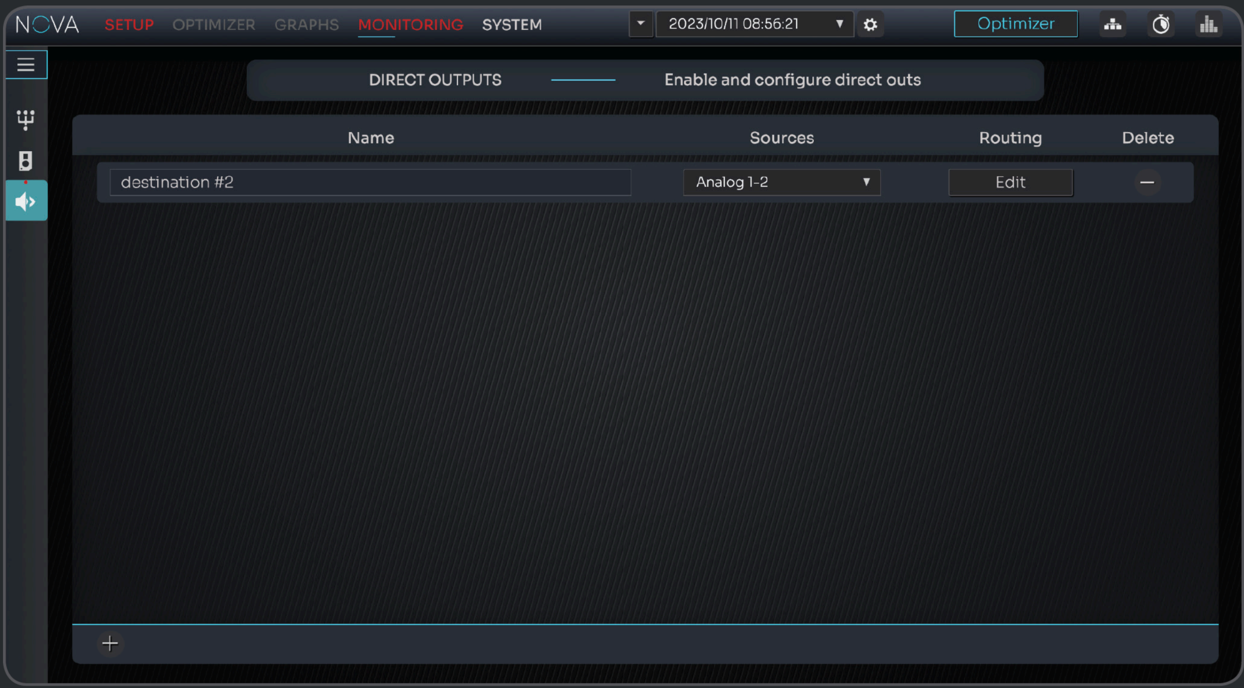
Please refer to the La Remote user manual for more details.

Direct Outputs

Direct outputs are generally used to feed external devices such as a metering system or a recorder.

More specifically, here is what you also need to know about Direct Outputs:

- A Direct Output has no processing delay. It is commonly referred to as "zero-latency".
- A Direct Output is not affected by volume changes. The selected source is redirected to the Direct Output with unity gain.
- A Direct Output feeds unprocessed signals



To configure a Direct Output, all you have to do is to:

1. Type a name for your Direct Output
2. Select the sources for your Direct Output
3. Edit the routing of your Direct Output.

Note that only available outputs will be visible for a Direct Output.

Downmixes

Downmixes are available in the monitoring section of the home page. Downmixes are typically used to check the compatibility of your mix across different formats. As an example, you would use Downmixes to check how a 5.1 mix would sound in Stereo or Mono.

To configure Downmixes, first activate any of the available downmixes listed on the right-hand side of the menu.

Each of the activated downmixes will be contextually displayed in the downmixes section of the Home Page according to the format of the source and speaker set in use.

As an example, if you wanted to listen to a mono mix of a 5.1 mix through a 5.1 speaker set and that you enabled both Mono downmixes, you could choose either:

- Mono = downmix is routed to the Center speaker only (discrete Center).
- Mono on L/R = downmix is routed to both Left and Right speakers at equal power ("phantom" Center).

While listening to a Downmix, you can use the SOLO or MUTE functions to listen to individual speakers.

You can then adjust the fold-down rules from the left section of this page. Fold-down rules are basically the amount of attenuation you apply on the original signals before downmixing them to a lesser number of speakers.

Center/LR" foldown rule affect both "Center " and "Mono on L/R" downmixes

Optimization

The Optimizer technology is probably the main reason why you purchased a NOVA.
Loudspeaker/Room Optimization is Trinnov Audio's main area of expertise, and mostly how the company has built its reputation since 2005.

The Optimizer is a digital correction technology complementary to acoustic treatment.
Using both active and passive acoustic treatment is the best way to achieve the highest possible performance in sound reproduction.

The Optimizer

The better the acoustic environment, the more effective the Optimizer processing.

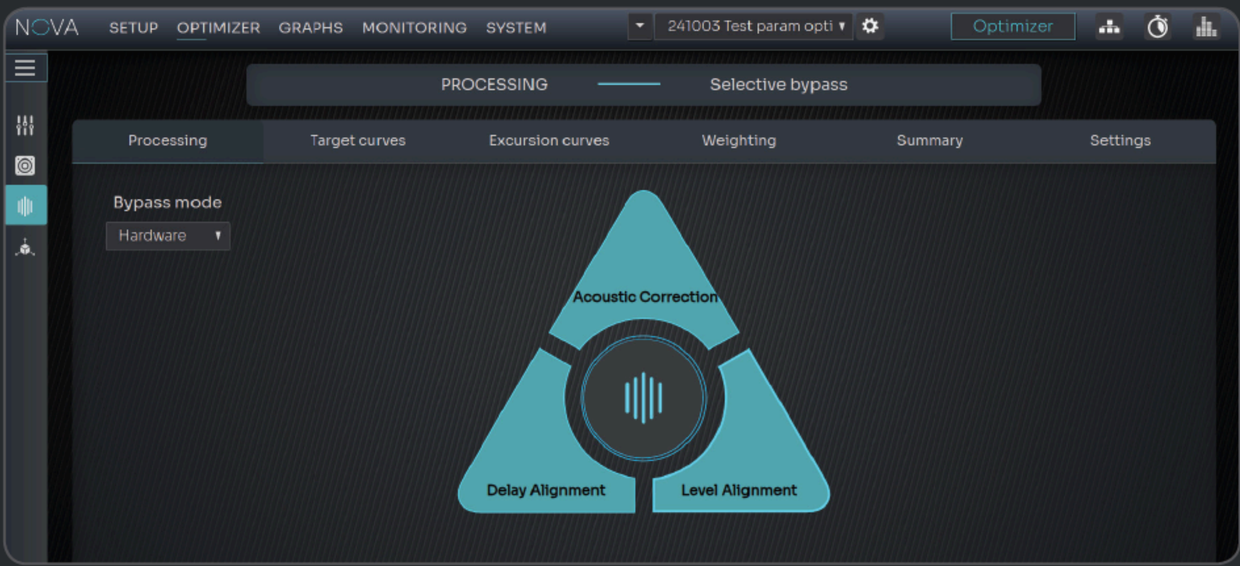
As a minimum, you should consider the room's dimensions, its construction and materials, and the mixing position. Adding acoustic panels or modular elements, or even just furniture and equipment can help although we strongly recommend consulting an experienced acoustician to help design your room.

You should avoid placing obstacles between the loudspeakers and the listening position, such as computer screens, glass surfaces, large empty desks, etc...

However, the listening position should be arranged according to the mixing conditions you will work in. So, if DAW screens and other devices are required around you, they should be positioned before calibration.

The Optimizer aims to achieve the best response from your loudspeakers, whatever the listening conditions.

- The Optimizer precisely measures the acoustic response of your speakers at several positions in the room. Variations in the acoustic response are often due to acoustic problems, which is why it is important to perform in-room measurements.
  The Optimizer extracts and uses a huge amount of information from each measurement, from speaker position to amplitude and timing information.
- The Optimizer classifies each acoustic issue (loudspeaker, early reflections, reverberation, room modes...) and addresses each acoustic concern using the most effective method.
- The Optimizer does not try to correct what can't be corrected electronically.



Hardware /Software Bypass

Enabling bypass disables the Optimizer for uncorrected listening. Switching to hardware bypass minimizes latency, making it ideal for recording. Please note that some features will be unavailable in this mode.

Unavailable :

- Setup : Source and speaker configuration
- Optimizer : Bass management is disabled and all optimizer functions
- Graphs : disabled
- Monitoring : Downmix, headphones, la remote configuration and direct output configuration
- System : Latency unavailable

In the main OPTIMIZER menu, the processing tabs offer you to selectively enable or disable different types of corrections applied by the Optimizer:

- Acoustic Correction: disables the Optimizer's acoustic correction, including both the amplitude and phase correction
- Delay alignment: disables all the delays applied to time align every speaker to the main listening position
- Level alignment: disables all the gains applied to level match every speaker to the main listening position

Finally, if you hit the Optimizer icon at the center of the triangle, all three corrections will be disabled at once. This is similar to the bypass button available from the top menu.

Graphs

The graphs generated after the calibration are designed to help you better understand your room and the way your speakers interact within it.

Basic Mode

By default, the layout of the Optimizer graphs is fixed, with preconfigured tabs used to show you the essential information about your system.

The right panel features basic filters and visualization options.

The visualization options are divided into two parts.

M1 stands for the "Measurement point", if several points have been done. You will be able to choose the corresponding point.
You can also select which graphs you want to visualize, by selecting one of those options in the dropdown menu :

Amplitude : The Amplitude graph displays the frequency response of the speaker and takes the whole Impulse's response time window into account : Direct Sound, Early Reflections and Late Reverberation.

Amplitude Direct : The Amplitude (Direct) graph displays the frequency response of the direct sound and excludes the end of the response.

Phase : The Phase graph displays Phase shift VS Frequency across the bandwidth.

Impulse Response : The Impulse Response graph is an amplitude VS time representation.

Impulse Response (lin): The Impulse Response graph (lin) is a linear and logarithmic amplitude scale.

Group Delay : The Group Delay graph displays Time arrival VS frequency across the bandwidth.

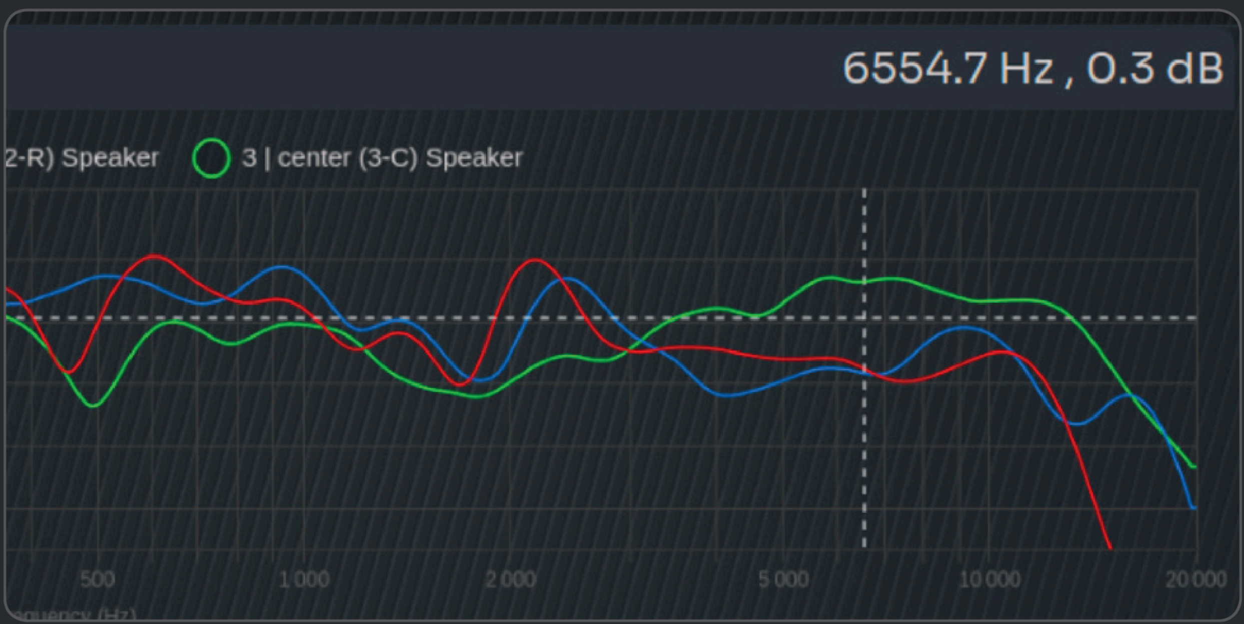
The **Overview** tab shows an overlay of every measured speaker before and after correction to understand the impact of the Optimizer processing.

The **Speakers** tab shows an overlay of the before and after correction for each speaker.

The **Measurement** tab shows an overlay of the different measurement positions for each loudspeaker to understand the variation of the acoustic response across the different measurement points.

The **Phase** tab shows the phase response of every speaker before and after correction to highlight the effect of the Optimizer in the time domain.

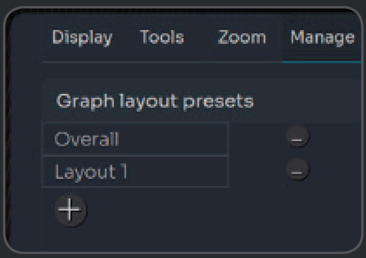
The basic mode allows you to use the cursor to show the exact information for both axis. In the example below, the exact frequency (x-axis) and amplitude (y-axis) values are displayed for the cursor location.



Expert Mode

Engaging the [Expert mode] in the Optimizer Graphs menu provides a lot of additional acoustic information and much greater flexibility in terms of display.
In Expert mode, the graphs truly turn NOVA into a comprehensive stand-alone measurement system.
As an example, you can create your own graph layouts and save them as presets.

Layouts



The different layouts are created by adjusting options in the Display menu.

You can determine what information you want to overlay, what information you would like to display in different frames, and what information to visualize in each of these frames.

The number of options is huge, and more than most people actually need.

Graph display

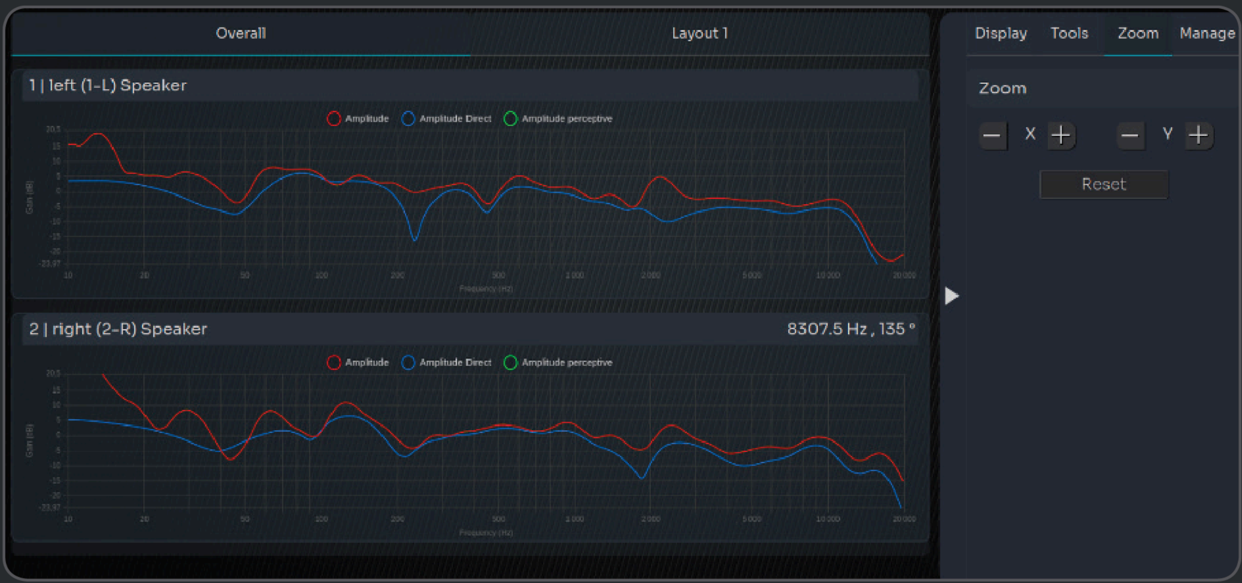
The available options to choose from to create your own graphs and layouts are:

- Speakers: display acoustic response by speakers (declared and calibrated speakers)
- Acoustic responses with frequency scale as the x-axis:
  - Amplitude
  - Amplitude Direct
  - Phase
  - Phase Direct
- Acoustic responses with timescale as the x-axis:
  - Impulse response
  - Linear Impulse response
  - Group delay
  - Group delay direct

- Before/After:
  - Speaker response Before correction
  - Speaker response After correction
  - Response of the correction Filter

- Measurement points: shows the responses of each measurement point

Zoom



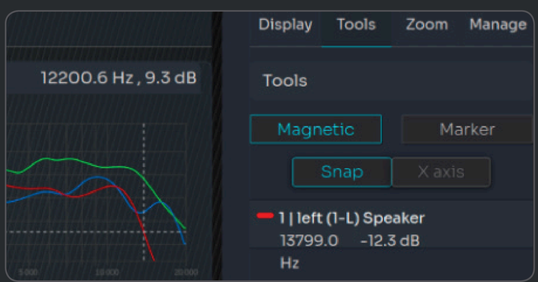
Zoom can be used in 3 different ways.

- using your pointer on the axis of the graph (x or y).
- using your pointer on the curve.
- using the -/+ under the zoom tab.

A reset button lets you switch back to the default position.

Graph analysis

In expert mode, you can access different analysis tools by clicking on the **Tools** tab.



Magnetic snap will automatically link to the nearest curve.

Magnetic X axis will draw a vertical line and display each curve's intersection.

You have the possibility to add points on a curve and compare the delta between points.

You can also add markers. Those lines are fixed and won't follow the zoom.

Bass Management

Bass management is used to redirect the low frequency portion of a speaker signal to a more capable subwoofer.

The purpose of bass management is to get greater bass extension and make sure every driver works at ease within its natural bandwidth, rather than trying too hard to reproduce frequencies it inherently cannot.
It is important to note that subwoofer integration is essential to make sure speakers and subwoofers blend perfectly.

To deliver a seamless result, the Optimizer makes sure that the system achieves a smooth response at the crossover region and that both speakers and subwoofers are perfectly time-aligned.

In Basic mode, every speaker is bass managed by every subwoofer with default 80Hz Linkwitz-Riley 24dB/octave crossover filters.

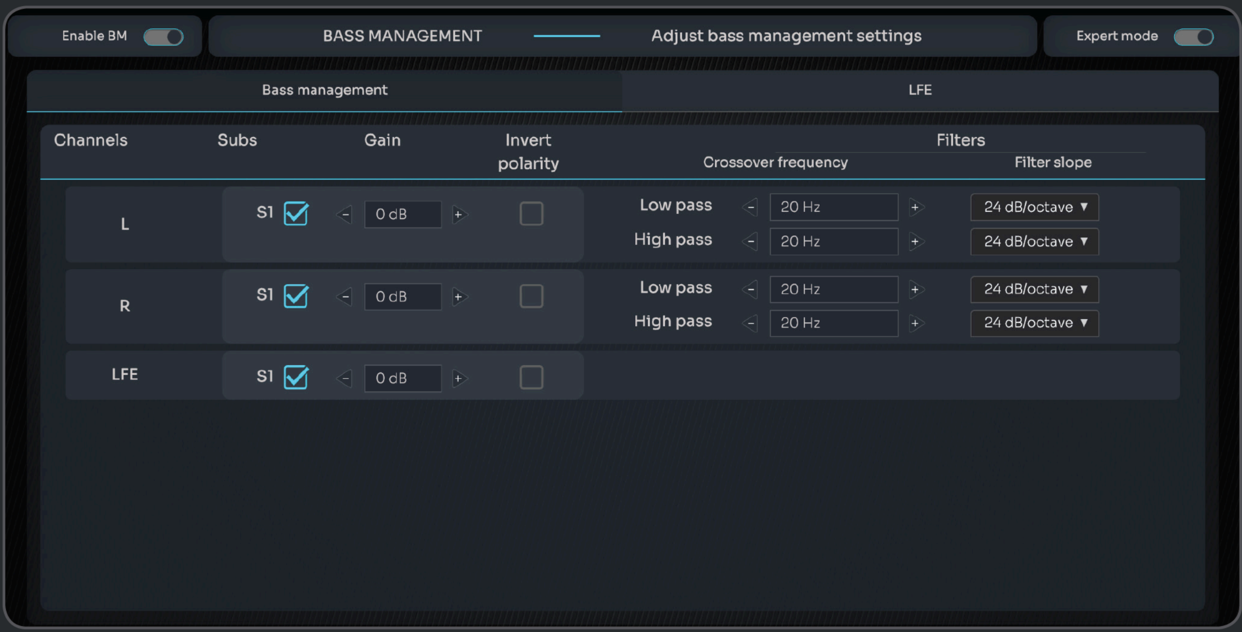
Bass management:

- **Crossover frequency**: the frequency that will be applied for the cut-off between the speakers and subwoofer. Subwoofers receive low frequencies from the main channels below the chosen Crossover Frequency.
- **Filter slope**: the amount of attenuation the filter applies per octave. As an example, a 24dB LR filter applies 24 decibels of attenuation per octave starting at the crossover frequency.

LFE : LFE stands for "Low Frequency Effects," and is an optional channel that may or may not be present in the source material, and which is the "x.1" channel. LFE signals are usually reproduced by one or more subwoofer(s).

- **Cutoff frequency:** This setting determines the cutoff frequency of the LFE channel.
- **Apply +10 dB on LFE:** In a professional environment, this option should be used as required with respect to the recommended calibration level of the subwoofer to achieve the best gain structure. The LFE channel is recorded with a level offset of -10 dB. This offset has to be compensated for in the system. This option should therefore be used only when no other equipment within this chain applies this gain.

In Expert mode, Bass Management is very flexible.



Once engaged, the expert modes give you multiple bass management options for every channel:

- Use any combination of subwoofers to bass manage each channel
- Add gain to increase the gain of the bass-managed portion of the signal sent to the subwoofer(s). Note, this is different from increasing the gain on the LFE input channel
- Invert polarity (used for troubleshooting purposes only)
- For every channel, determine the low pass and high pass filters, including:

• Low Pass – Sub(s): These settings will determine the cutoff frequency below which the signal from the channel is redirected to the subwoofer(s). These settings are only applied to the signal redirected to the subwoofer(s).

• Sub settings: These additional settings give the ability to invert the polarity and adjust the gain of the signal redirected to the subwoofer(s). These settings are only applied to the signal redirected to the subwoofer(s).

• High Pass – Satellites: This section determines the low-end cutoff frequency of the signal sent to the satellites. These settings are applied to the signal directed to the Satellites.

Target Curves

To understand what the behavior of a Target Curve is, let's compare it to an equalizer:

- With an equalizer, you directly apply corrections to achieve a certain acoustic response of the system
- With a Target Curve, you draw the acoustic response you want to achieve and let the Optimizer apply the correction for you

The Target Curve is possibly the most powerful and useful tool you can use to fine-tune the default result and shape the sound to your taste.

A different target curve can be applied for each individual speaker, but the recommendation is to apply the same target curve to the entire speaker set.

A **"Link all" button** has been added to Target curves, excursion curves, EQs, and FIR EQs.

The **Edition mode** drop-down menu lets you decide whether you want to edit the target curve freely or by section (using the segment tool).
The tick boxes under the **Manage** column allow you to link speakers to edit the target curves of multiple speakers at once.

As soon as you start editing the Target Curve, the **Apply** button will be highlighted.

Applying changes once the edition is completed is required for your changes to be considered. This will trigger a new computation of the filters, where the Optimizer will try to achieve the new target curve.

Tips:

- Don't forget to save your new settings to a different preset. Saving different configurations is useful to perform A/B comparisons between different settings and make sure you are moving in the right direction.
- You can also display in the background your speaker curves (before or after). This will help you determine how to shape the target curve based on your room results.

Excursion curves

The Excursion Curves function sets boundaries for the Optimizer filters.

In other words, the Excursion Curves limit the boost or attenuation the Optimizer can apply to the original acoustic response to achieve the target curve.

The Excursion Curves can be edited independently for each loudspeaker or subwoofer and allows for setting different maximum boost and attenuation at all frequencies.

The Excursion Curves offer the same **Edition mode** as the Target Curves: free or by segments.

The Excursion curves and Target Curves are interdependent and can result in NO correction at all. Let's take an example and assume we are talking about a single speaker:

- The maximum boost limit is set at + 6dB at 30Hz in the Excursion Curves
- The Target Curves is set to achieve +5dB at 30Hz
- The measured amplitude response of the speaker before correction is - 7dB at 30Hz

The result after correction cannot be more than -7db (original response) + 6dB (maximum boost) = -1dB, which is 4dB less than the target curve.

We recommend extreme caution when setting the excursion curve and especially when increasing the maximum authorized boost in the low frequency. You may damage your speakers or increase the level of distortion if you push them beyond what they can do.

The Excursion Curves also include a special feature called **Adaptive Limiter**.

When the Adapt limiter is on, the excursion curve will be automatically set to limit the correction in frequency regions where too much noise was measured. Indeed, too much noise in the original measurement may result in somewhat irrelevant processing, as we would rather not correct noise but correct actual signals.

The adaptive limiter should not drastically limit the amount of correction unless the measurement was performed at a low volume in a noisy environment.

Note that before the first speaker test signal is played during the calibration sequence, the Optimizer measures the noise floor and increases the test signal where too much noise is detected. This "pre-emphasized" measurement signal better rejects noise from the acoustic measurement.

Settings

The advanced settings of the Optimizer are divided in two sections:

- Algorithm parameters
- Filter parameters

The algorithm parameters determine the behavior of the Optimizer algorithms.
The filter parameters determine what filters the Optimizer can implement to achieve the best results.

This whole section should be used by experienced calibration or with a minimum understanding of digital signal processing.

Algorithm

Optimization mode

A global parameter that determines the algorithm's scope of action. Amplitude + Phase is strongly recommended.

Quantity of early reflections

Defines the size of the time/frequency window used by the algorithm to mitigate early reflections. More cycles mean a larger window. Not recommended in small to medium size rooms unless strong reflections occur in the immediate surrounding of the speakers.

Resolution of energy response

Determines the resolution of the correction applied to the energy response of the room, not the resolution of the filters. More resolution results in more analytical and accurate reproduction. Let your ears decide which setting suits you best.

Filters

Filters combination used by the algorithms. FIR filters operate in the time and frequency domains over the entire frequency range. IIR filters operate in the frequency domain only and are implemented to increase resolution in the low frequencies.

High-pass frequency

Protect the system from undesirable low frequencies and/or DC offset.

Algorithm limits

Maximum boost: defines, in dB, the maximum amount of boost that will be performed by the algorithms. This parameter is used to avoid distortion. Its default value is 6dB. This parameter has an important impact on the behavior of the automatic equalization and is applied to both the time- and amplitude-based approaches.

Maximum attenuation: defines, in dB, the maximum amount of attenuation that will be performed by the algorithms. Its default value is -10dB. This parameter also has a significant impact on the behavior of the automatic equalization and is applied to both the time-based and the energetic approach.

Filters

The optimizer uses two types of filters : IIR and FIR

IIR (Infinite Impulse Response)

Digital equivalent of what is typically used in analog electronic filters composed of resistors, capacitors, inductors (or even linear amplifiers) ; also found in early and low-cost digital processing.

The infinite response can be explained by using capacitors with a "memory" and their internal state never completely relaxes following an impulse (assuming the classical model of capacitors and inductors, where quantum effects are ignored).

The IIR filters are characterized by a response based on the values of the input and the previous values of the same response.

FIR (Finite Impulse Response)

FIR describes the impulse response for each sample: all the "echoes" that will follow in time from an initial stimulus. This description is done in a defined time range (the length of the filter)

Technically, FIR filters consume much more CPU than IIR filters. Its latency depends on the implementation, but is always a few dozen samples at minimum. The heavy calculus to be done is known as a "convolution product".

In the interface, you will find all the useful information about applying such settings.
It is strongly recommended not to change these settings unless you know what you are doing.

IIR Settings

- **Number of IIR filters** (default is 30): the number of IIR filters that will be used on every channel.
- **IIR filters minimal/maximal frequency** (default is 20 Hz/300 Hz): IIR filters will be positioned from the minimal frequency up to the maximal frequency.

  Note: In Automatic, the lower frequency of your speaker's bandwidth will be set as the minimal frequency used for the IIR filter.
- **Resolution of energy response for IIR:** Increasing this parameter can help achieve a more linear low frequency response
- **Low-freq auto-transition bandwidth**: This setting determines the number of octaves over which the max limiter is progressively inhibited. The special value disabled allows you to completely disable the automatic behavior of the limiters.

FIR Settings

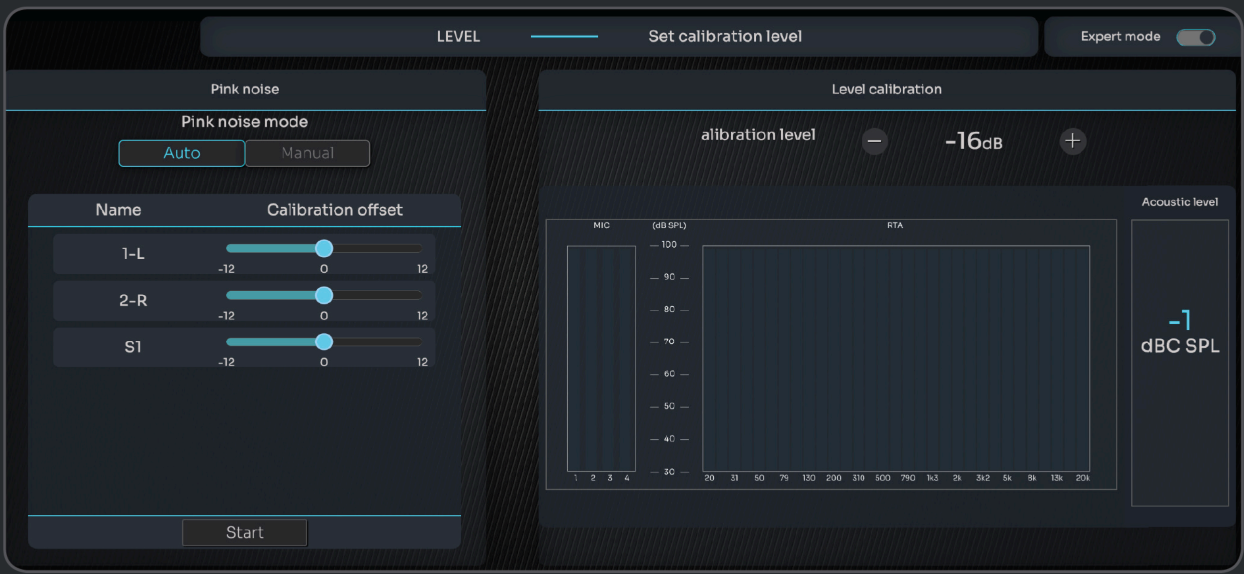
- **FIR filter length** (default is 100ms): The longer the FIR length, the higher the filter resolution across the spectrum and CPU load. The default setting already provides more than enough resolution in most situations.
- **FIR reference** : Increasing the FIR reference enables the Optimizer to correct phase and group delay in the low frequencies more effectively. It can considerably improve transient response, bass control, and stereo imaging. This may produce artifacts beyond a certain limit.
- **Room smoothing method** : Acoustic calculation can be done in Modulus and Square Modulus. Modulus is a calculation based on Amplitude. Square Modulus is a calculation based on Power.
- **Preringing reduction**: Exclusive technology from Trinnov to reduce artifacts when applying advanced correction in the time domain.

Calibration

Measurement

The Calibration process is essentially the process whereby the acoustic response of every loudspeaker is captured, and then corrected, in the room.

The Calibration wizard guides you through the process and gives you multiple options, based on the nature of your environment (single measurement position, multiple measurement positions…).



The calibration process requires placing a Trinnov 3D Microphone at one or more listening positions to sample how the room reacts – for example, at the mixing position, the producer's couch, etc.

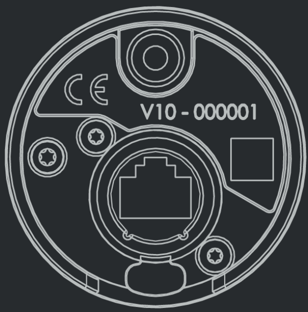
For each measurement position, NOVA follows the same calibration sequence: it first samples the noise floor and then generates a measurement test signal, which is played 3 times through every loudspeaker declared during the configuration. A MLS (Maximum Length Sequence) burst sounds like Pink Noise, but it is not. Unlike pink noise, an MLS sequence is used for impulse response measurement. By measuring the impulse measurement of each loudspeaker, the Optimizer not only collects frequency information, but frequency information over time.

The Optimizer can then compare the original test signal to the same signal measured in the room through each loudspeaker and compute the most appropriate filters to achieve the user-defined target curve.

The Optimizer then generates a set of acoustic graphs to help the user better understand the behavior of the system and make informed fine-tuning decisions.

Microphone

The microphone input of NOVA is located on the front panel and uses an etherCON connector. (Note that any CAT6 cable up to 50 meters will work).



NOVA uses the latest version of the Trinnov 3D measurement microphone, which brings two major upgrades and benefits:

- Once connected, the microphone automatically powers on and off, based on NOVA's instructions
- Once connected, the microphone automatically transfers its unique compensation file to NOVA to deliver a seamless experience to the user, whilst ensuring the most accurate measurement possible

Main Listening Position

The calibration should be performed in conditions that are as close as possible to the normal usage situation, although there might be some exceptions if your normal conditions are not ideal.

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

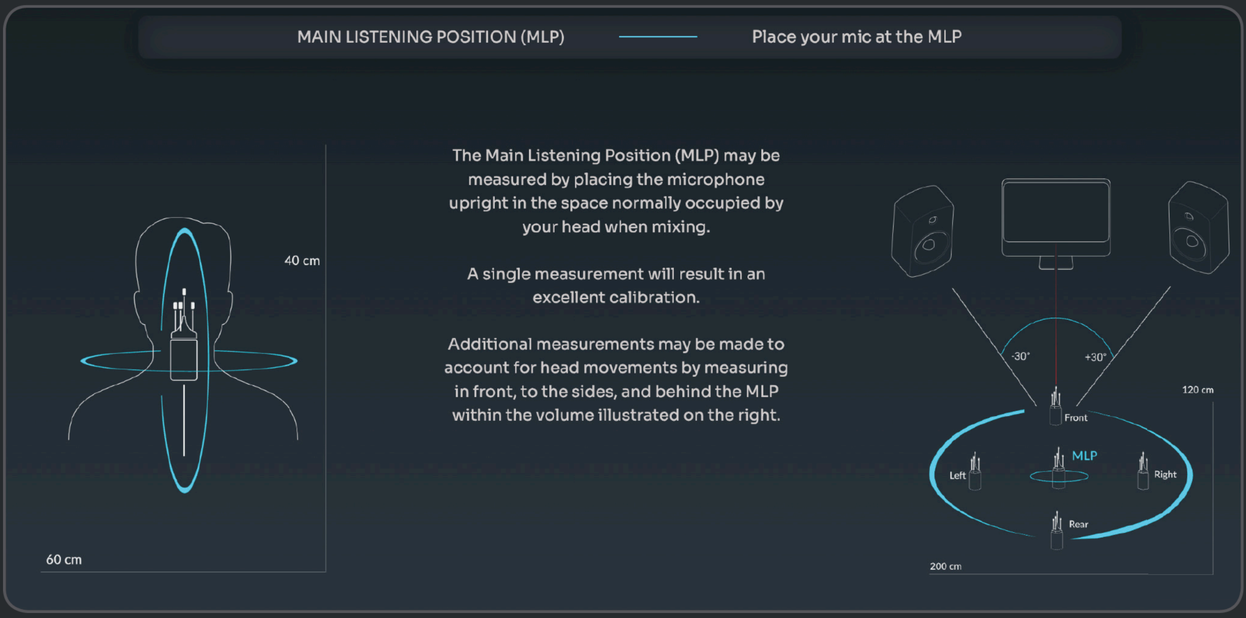
- No obstacle between the speaker and the microphone
- No highly reflecting surface (leather sofa, glass table...) close to the microphone
- No background noise during the measurements (air con, opening doors, windows...)
- No movement during the calibration (disturbs speaker localization)

The Optimizer supports single measurement and multiple measurement calibrations, but only a single main listening position, or reference position, can be set.

The Main Listening Position is used specifically as a reference for:

- Loudspeaker 3D localization
- Loudspeaker relative delay/level alignment
- Active crossover calibration (not available yet)
- Master delay/level calculation

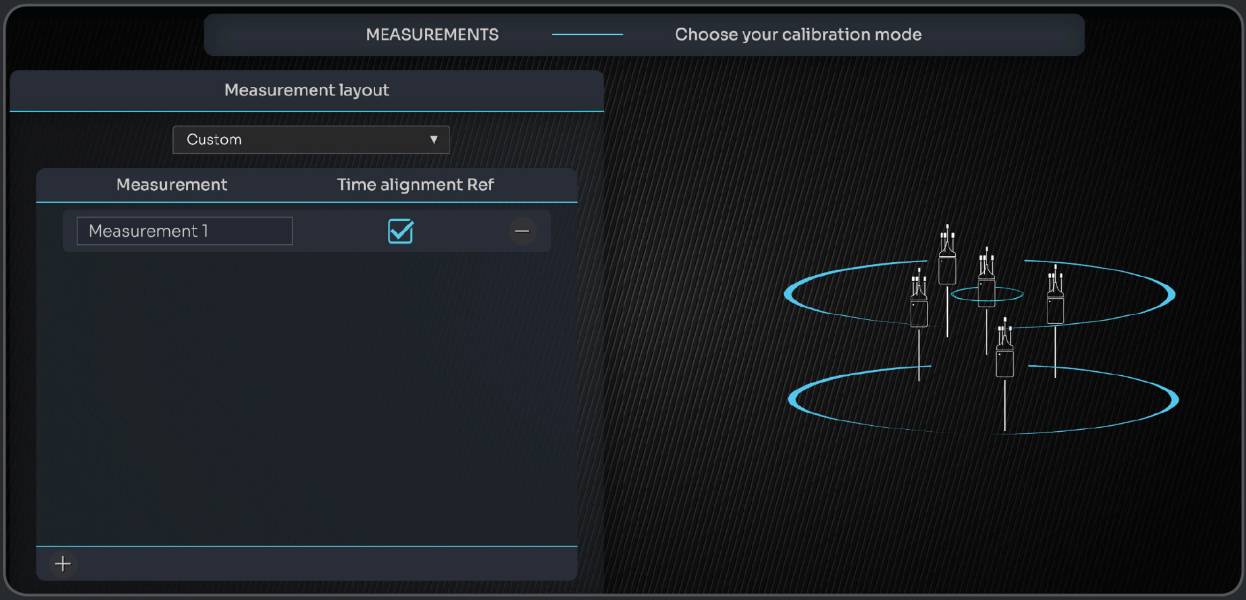
The main listening position is named appropriately: it should be as close as possible to your main mixing position.



Multipoint measurement

One of the most advanced features of the Optimizer is its unique multipoint measurement technology.

Instead of using an averaged response out of multiple measurement points, the Trinnov multipoint algorithm takes every single measurement point into account and tries to achieve the best result in both frequency and phase for each position.



The purpose of multi-measurement is not only to optimize a wider listening area but also to get additional information from measurements and increase reliability by considering important variations in the listening area.

After calibration, the Optimizer even lets you set a different weight for each measurement position to focus the optimization on the area of your choice.

Dante

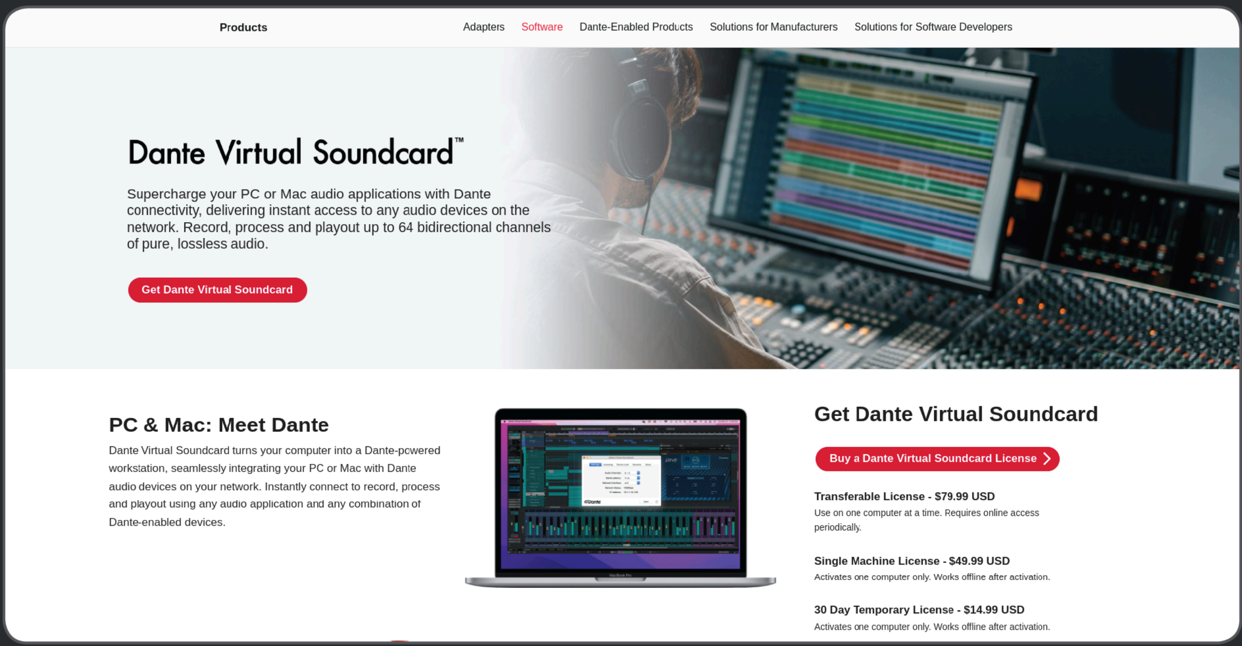
Certification

Although NOVA supports Dante, the purpose of this user guide is not to educate people about Audio over IP, which is of course a vast and complex subject.

Instead, we strongly recommend everyone sign up for the Dante certification, which is free of charge and available online, free of charge at the following address:
https://www.audinate.com/learning/training-certification/dante-certification-program

Dante Virtual Sound card

DVS is a software sold by Audinate, the Australian company behind Dante.



Considering what DVS does, it is fair to say that DVS is relatively inexpensive, and therefore it is difficult not to recommend it.

Consider DVS as a virtual audio interface on your workstation, which connects to a network audio hub (Dante Controller) and provides you with access to every Dante device on the network.

In other words, NOVA (or any other Dante device) connects to DVS via the Dante controller, which is another essential free software developed by Audinate. From there, DVS provides you with access to NOVA I/Os from your DAW as a virtual sound interface.

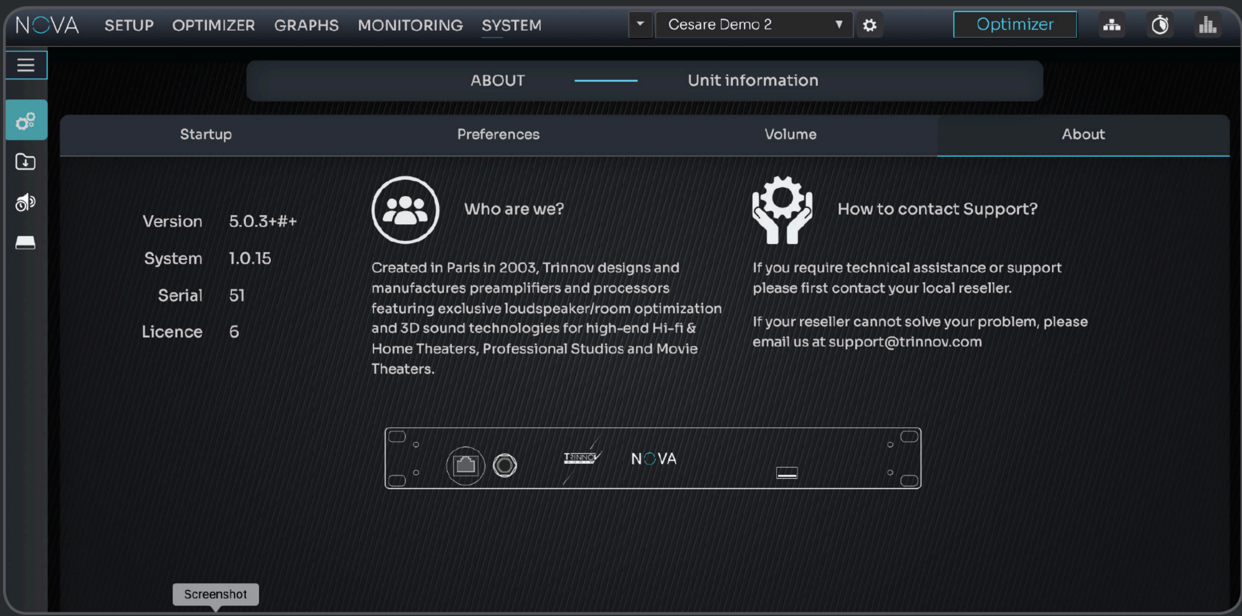
Warning about USB to network adapters

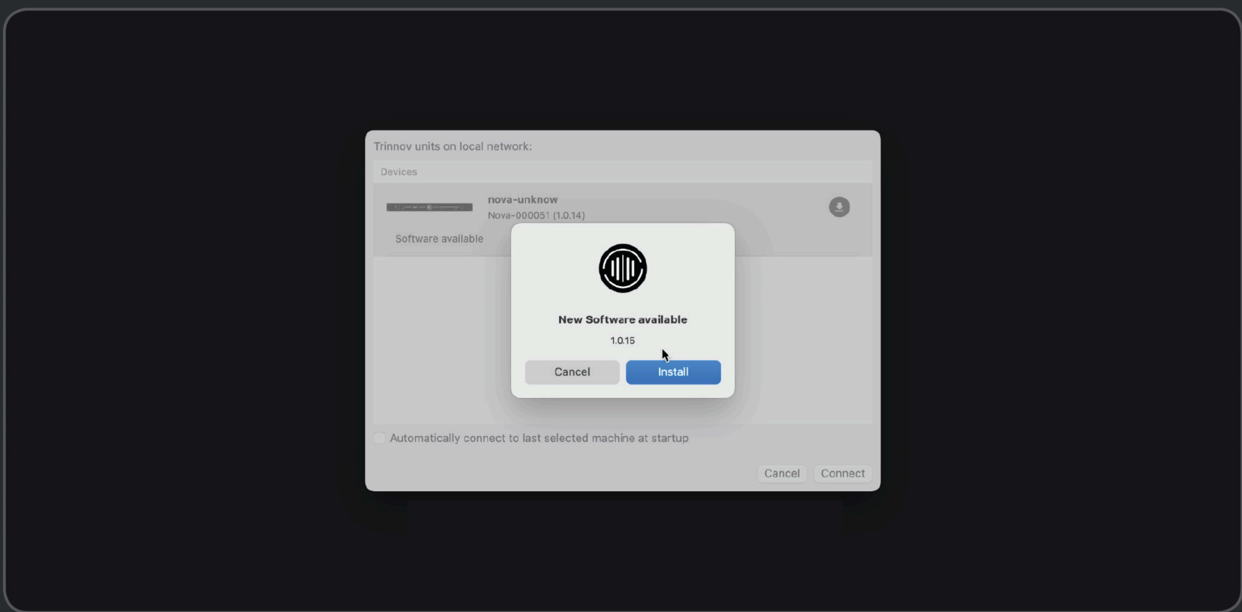
USB to network adapters used with laptops could potentially be problematic.
If Dante and audio over IP prove to be unreliable, please first check your hardware and the quality of the internal chipset used for USB to Ethernet translation.

HELP

System updates and License upgrades

Using the Trinnov application (macOS or Windows), you will be able to check the version on NOVA and see if a new version is currently available.





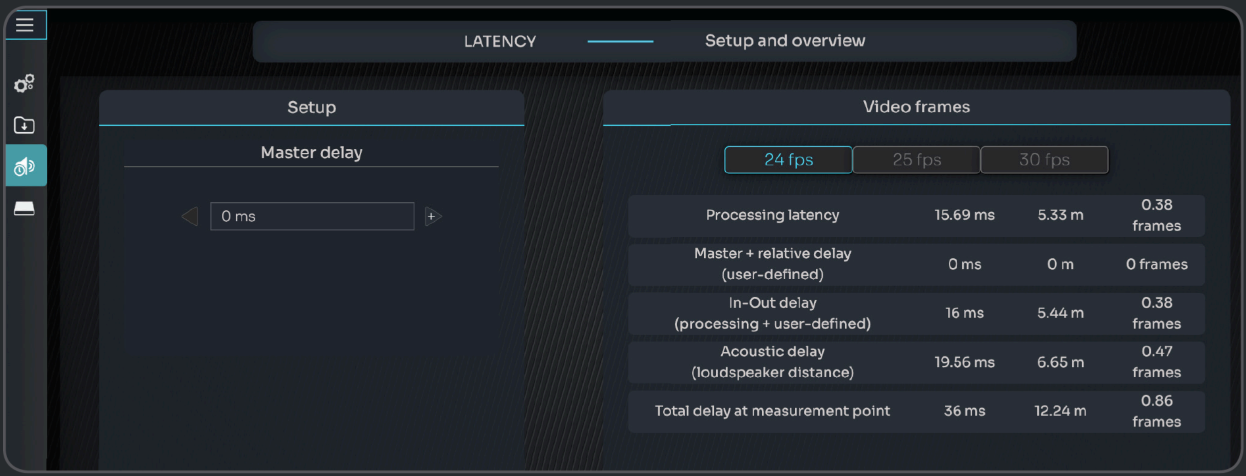
If a new version is available, a pop-up will appear.

The same is true for extra purchased licenses; they will be available through the Trinnov application.

Latency

Unlike more basic acoustic correction solutions available on the market, we developed expertise in time/frequency analysis. Sound is essentially a phenomenon that you can explain if you look at the time domain, and operating in the time domain necessarily implies latency to correct phase issues.

As such, you can expect an average latency of 20ms with the Optimizer engaged, although this may vary based on the buffer size, sampling rate, and some optimization parameters.



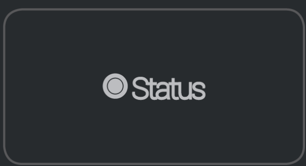
LEDs Status

Standby Button :

Blank: unit is off
Blue: unit is powered on
Yellow: unit is turning on or off
Red: a power failure occurred

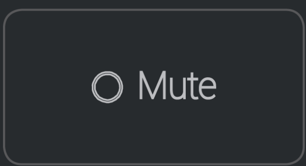
<u>If the button is pressed:</u>
Red: the unit will shutdown if the button is depressed
Pink: the unit will be forced shutdown if the button is depressed
Yellow: a network reset will be requested if the button is depressed
Blue: no action will be taken

Status LED:



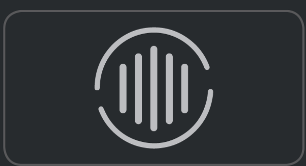
Blinking : booting/ loading a preset configuration
Green : ready
Red : system error occurred
Blue (blinking) : software update in progress
Green/Blue (blinking) : audio clock error
white: network reset in progress

Mute LED :



Red: all the speaker outputs of the NOVA unit are muted
Yellow : DIM is engaged, the speaker output is lowered

Optimizer LED :



Blue : optimizer is calibrated and running
Yellow : optimizer is NOT calibrated
Yellow (blinking) : Optimizer is not running. Either bypassed or computing filters.

APPENDIX

Version history

Version 1.0

November 2023