D-MON | SERIES

USER MANUAL

D-Mon

User Guide

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To obtain the latest documentation and software downloads, please visit www.trinnov.com

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1 WELCOME

Welcome to the **D-Mon** from **Trinnov Audio**.

1.1 ABOUT THIS MANUAL

This document describes how to install, set up and operate the **D-Mon**. We recommend that you read this manual carefully before installing or operating a D-Mon system. The specification is valid for versions starting from 4.2.230.

The operation and setup of the different models are identical, but some functionalities such as the number of Optimizer® channels will vary. These differences are noted in the relevant topics.

For a short introduction, we recommend reading Chapter 5*: Quick Start*.

Also, look out for the following text boxes which indicate:

Notes - points of clarification.

Tips - useful tips and shortcuts.

ATTENTION!
Alerts you when an action should *always* be observed.

1.2 SOFTWARE UPDATES

Trinnov Audio employs an ongoing development programme and offers software updates for all **D-Mon** products. Releases may be downloaded from the website after registration (see below).
The following utility software applications are freely available from the Trinnov website (https://www.trinnov.com):

- **Trinnov App** – this is a separate application which must be installed if you wish to simply take control of your D-Mon, or control it from an Avid EUCON control surface remotely. See Chapter 8*: External Control* for details.
  NB: The app is currently only available on Mac computers.

1.3 USER REGISTRATION

For access to regular product updates and downloads, please register at https://register.trinnov.com

2 IMPORTANT SAFETY INSTRUCTIONS

ATTENTION!

To ensure optimal performance, please pay attention to the instructions in this Quick Start Guide:

- Read these instructions.
- Keep these instructions.
- Follow all instructions.
- Install the apparatus on a solid, flat, level surface that is dry, well ventilated and out of direct sunlight. Be sure that all four feet are supported.
- Do not use this apparatus near water.
- Clean only with a dry cloth. Do not use liquid solvent-based cleaners.
- Protect the detachable power cord from being walked on or pinched particularly at plugs, convenience receptacles, and the point where they exit from the apparatus. If the ac cord becomes damaged, do not use it. Immediately replace it with a new one of the same or better rating.
- Unplug this apparatus during lightning storms or when unused for long periods.
- Do not open the equipment case or remove any of the cover panels. There are no user serviceable parts in this equipment. Refer all servicing to qualified service personnel.
- To prevent fire or shock hazard, do not allow liquids to spill or objects to fall into any openings of the product.
- Use only attachments/accessories specified by the manufacturer.
- This unit is supplied with two 3-pin grounded AC plugs. Always insert the AC plug into a grounded outlet. Do not remove the ground pin or disable the ground for any purpose. The main AC must be protected by a 20 Ampere circuit breaker.
- Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding-type plug has two blades and a third grounding prong. The wide blade or the third prong is provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
- Before connecting the equipment, check that the main power supply voltage rating corresponds to the local main power supply. The rating of the main power supply voltage is printed on the equipment.
- If replacement of the ac line fuse and any internal fuse becomes necessary, replace only with the same value and type of fuse (110V: T1A Schurter FST 5x20; 220V: T800mA Schurter FST 5x20). Never bypass the fuse.
- It is imperative that these apparatus be operated in a well-ventilated environment and the immediate external temperature be maintained as specified. Do not expose this apparatus to humidity, steam, smoke or excessive dampness or dust. Maximum permissible operating conditions: 0°C to 40°C, 20-65% relative humidity. External cooling fans may be required in some cases.
- Do not stack any equipment directly above or below this apparatus to protect it from overheating, as well as the continued functionality of any equipment near and around it.
- To completely disconnect the apparatus from the AC, completely remove the power cable from the main outlet.
- Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as when the power-supply cord or plug-in damaged, liquid has been spilled, or object has fallen into the apparatus or the apparatus has been exposed to rain or moisture, does not operate normally or has been dropped.
- Do not expose this apparatus to dripping or splashing and ensure that no objects filled with liquids, such as vases, are placed on the apparatus.
- The main plug of the power-supply cord shall remain readily operable.

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 OF SUFFICIENT MAGNITUDE TO CONSTITUTE A RISK OF ELECTRIC SHOCK.
⚠	THE 220/110V SELECTION IS NOT AUTOMATIC. It can only be made by an authorized person. Check with your reseller for further information.

3 INTRODUCING THE D-MON

3.1 OVERVIEW

The **D-Mon** provides comprehensive monitoring solutions for professional recording, mixing and mastering studios. The processor is designed to meet varying needs ranging from simple stereo loudspeaker setups to more demanding immersive installations. In each case, the processor provides two main functions: an advanced monitoring controller along with a loudspeaker/room tuner, Trinnov Optimizer®.

Operations can also be controlled via the D-Mon Control Panel (a web-based user interface).
The **Trinnov La Remote** is the perfect companion for **D-Mon**, providing an extensive and fully configurable surface.

La Remote and D-Mon Processor

D-Mon Control Panel interface (common to all models)

Also, a wide range of control protocols is available for fine grained integration (via Avid EUCON protocol, Digidesign Icon protocol, or MIDI). The Colin Broad TMC1 also features a specific D-Mon compatible firmware.

Advanced Monitoring Control

- **Monitoring**: day-to-day control of source & speaker set selection, volume, mute, dim, talkback, listen-back, headphone and outputs.
- **Routing Matrix**: any physical input or mix bus can be assigned to any physical output.
- **Internal Mixer**: inputs can be mixed to mono/stereo/multichannel mix buses and thus routed anywhere.
- **Studio settings**: base setup of loudspeakers and studio configuration.
- ***La Remote* settings**: configuration of the Trinnov control unit.
- **Audio over IP settings**: advanced configuration of AoIP Ravenna streams.
- **Studio Presets & Session Snapshots**: customize the processor for your installation, and store and recall different monitoring arrangements and DAW (Digital Audio Workstation) session setups.

Loudspeakers to Room Tuning (Optimizer®)

The **Optimizer®** provides digital acoustic correction. The algorithms are included with every processor; the number of Optimizer channels starts at 6 as a standard, and can be increased with additional channels, 2 by 2, up to a maximum of 18 channels.

Each Optimizer® channel can be re-assigned via presets to support multiple configurations.

- Multiple "sweet spot" presets can be saved for the same room – e.g., mixing seat, producer desk, musicians couch, etc.
- Advanced parameters meet the needs of different sound engineers - target curve, frequency, time & level alignment, etc.
- Other functions include down-mixing, graphic EQs, DRC emulation (Dynamic Range Compression), etc.

3.2 COMPARISON OF MODELS

The **D-Mon** model describes the number of available **Optimizer®** channels. D-Mon|6 up to D-Mon|18, the machine can be upgraded to fit any requirement. All models share a common set of inputs and outputs:

	D-MON|6, D-MON|8, D-MON|10, D-MON6|12, D-MON|14, D-MON|16, D-MON|18
DIGITAL IN (SUBD-25)	16 (2×4 AES3)
DIGITAL OUT (SUBD-25)	16 (2×4 AES3)
ANALOG IN (SUBD-25)	8 Line Level
ANALOG OUT (SubD-25)* *8 first channels are mirrored on direct XLR)	16 Line Level
Audio over IP Ravenna	16 channels IN + 16 channels OUT
OPTIMIZER® **(DIGITAL ACOUSTIC TUNING)**	**From 6 channels up to 18 Channels by increment of 2**
ASSIGNABLE INTERCOM	2 x Talkback + 2 x Listen-back lines. Fully assignable. 2 x 36V-power analog inputs on MPIO.
WORD CLOCK	BNC Input & Output
CONTROL PROTOCOLS	Avid EUCON (MC5, S6, Dock…). Icon D-Command & D-Control (X-Mon 15p cable), general MIDI
GPIOS	user assignable 2 in / 1 out (footswitch, remote commands).
REMOTE & LOCAL PARAMETERS *(Specifications may vary with some controllers)*	
LEVELS	Level / Mute / Dim / Dim Level
SOURCE SELECTION	Fully Configurable
MONITOR SELECTION	Uses programmable presets & profiles. (Limited by what's physically available.)
ROUTING MATRIX	Any physical input, mix bus or Optimizer® output to any physical output.
SUMMING MATRIX	16 mono busses *could be aggregated for multichannel busses.*
OPTIMIZER SETTINGS	Full control over Ethernet and from the Processor itself. (VGA/HDMI screen & USB mouse/keyboard.)

Each model has been designed with the following applications in mind:

D-Mon|6: for studios (recording, mixing, mastering) where up to three stereo speaker-sets would be the Main and Alternate selections. On top of those three, non-optimized speaker-set could be defined for special purposes like TV or very small validation speakers.
For post-production studios where one surround (5.1) speaker set is mostly in use. Additional stereo setups or any arrangement of six speakers can be easily stored and recalled.

D-Mon|8, D-Mon|10, D-Mon|12: for studios producing 5.1 or 7.1 programs and working simultaneously with one 5.1 and one stereo speaker set. It could also fit for 3D mixing rooms producing programs in 7.4.1 formats such as Dolby Atmos, Auro-3D and DTS-X.

D-Mon|14, D-Mon|16, D-Mon|18: for mixing rooms producing the full-fledged Dolby Atmos contents, up to 9.3.6 with advanced bass-management.

Machines built before December 2021 have a model specific configuration.
The former configurations are listed here for reference:

Pre-2022 Products	D-MON|4	D-MON|6	D-MON|8 & D-MON|12
DIGITAL IN (SUBD-25)	8 (4x AES3)	8 (4x AES3)	16 (2×4 AES3)
DIGITAL OUT (SUBD-25)	8 (4x AES3)	8 (4x AES3)	16 (2×4 AES3)
ANALOG IN (SUBD-25)	4 Line Level	8 Line Level	8 Line Level
ANALOG OUT (SubD-25)* *8 first channels are mirrored on direct XLR)	12 Line Level	12 Line Level	16 Line Level
OPTIMIZER® **(DIGITAL ACOUSTIC TUNING)**	**4 Channels**	**6 Channels**	**8 Channels**
	e.g. 2 x stereo pairs or LCR+Lfe	e.g. 5.1 speaker set or 3 x stereo pairs or LCRS+1xSt	e.g. 5.1 + 1 stereo pair or 7.1 system or LCRS+2xSt

An upgrade kit is available for earlier hardware versions.

3.3 SYSTEM COMPONENTS

To configure and operate the system, you will need three hardware components:

- **D-Mon Processor** (always supplied): handles all audio processing, routing, mixing and I/O.
- **Trinnov 3D-Microphone** (optional): required during setup to perform the speaker/room calibration.
- **Trinnov La Remote** (optional): for easy day-to-day control of your monitoring.
- **Controlling Device** (not supplied): this can be a networked device like a computer or a tablet with the Trinnov App utility or a standard web browser (chrome preferred). Alternatively, a physical screen, mouse and keyboard may be connected directly to the unit.

D-Mon Processor

3D-Microphone (optional)

La Remote (optional)

3.4 METHODS OF CONTROL

The D-Mon processor can be remotely controlled using **the Trinnov App utility** or a regular **web-browser** or an **external controller**.

3.4.1 THE D-MON CONTROL PANEL

The **D-Mon Control Panel** is the main User Interface for the system.

A freely downloadable utility application, **Trinnov App**, is provided. It allows an automatic discovery of the D-MON device on the local network and the display of the user interface. It is the simplest way to take control of your device. It also includes a meters window (pre or post optimizer).

Also, each D-Mon processor includes a dedicated web server and, therefore, the interface can be accessed from a networked device with a web browser interface such as a desktop computer, laptop, tablet, etc. and no additional software is required. Multiple connections to the device are supported; the last user's changes always take effect and synchronize with other open connections. A range of pages provides fast access to everyday tasks as well as setup functions:

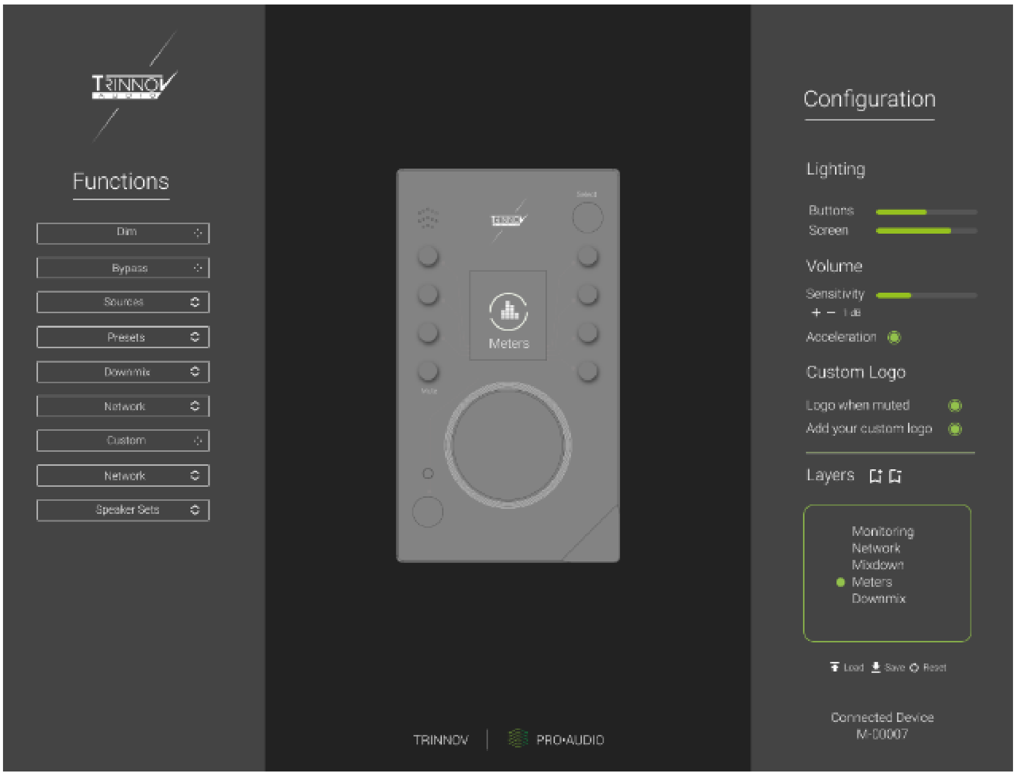
3.4.2 EXTERNAL CONTROLLERS

The D-Mon processor can also be remotely controlled. The level of integration depends on the protocol used. The options are:

Protocol or Hardware	Connection Method	Level of Integration
Trinnov La Remote	USB *either directly to the device or to the computer running the Trinnov App*	Fully customizable, 8 assignable buttons in unlimited layers + Volume + Mute
Avid EUCON	IP/Ethernet *requires the Trinnov App running on the main EUCON workstation*	EUCON specific monitoring, custom EUCON command keys
Avid ICON	"ICON" MPIO	D-Command & D-Control specific monitoring
Generic MIDI	"MIDI" MPIO	Main monitoring control functions
Griffin Volume Controller (discontinued)	USB *either directly to the device or to the computer running the Trinnov App*	Speaker Volume + Mute (or Talkback)
Trinnov network Protocols (proprietary)	Dependent on the interface	

Trinnov La Remote

Designed on purpose**, La Remote** offers a fully customizable experience. 8 user assignable buttons come along a dedicated mute button and a volume knob. The 8 user buttons are organized in layers and a dedicated select button allows the user to change the current layer or to change some specific functions like the headphone volume. A simple drag'n drop setup interface permits any user layout. Also, the screen display can be personalized with a custom logo, or other monitoring views. Multiple La remote could be connected on a D-MON device if needed.



Avid EUCON

All Trinnov products can integrate with the Avid EUCON ecosystem, with control surfaces like the Avid S6 or Avid Dock. The D-Mon is the only one that can handle all advanced monitoring functions like talkbacks, sources summation, cues mix busses… The Trinnov App is required and must be installed on the DAW (Digital Audio Workstation) which is linked to the control surface, along with the EUCON software package. The option "Enable EUCON gateway" should be check in the Trinnov App connexion windows, and then in the EuControl software, the Trinnov App should be locked as "monitoring" to ensure it will keep the control of this part while the main audio software (Pro Tools, Logic, …) is running.

Avid ICON Series / Generic MIDI Devices

Alternatively, an Avid ICON Series control surface such as the D-Command or D-Control, or any MIDI programmable device can be connected via the MPIO (Multi-Purpose In/Out) connector. In this instance, no additional software is required and the monitoring features remain active even with the DAW shutdown.

Two different MPIO breakout cables flavour are available:

- An **"ICON" MPIO** which has a DB15 ICON connector to connect to a D-Command/D-Control.
- A **"MIDI" MPIO** which is generic.

For more details, please see Chapter 8*: External Control*.

Griffin USB Volume Controller (discontinued)

This option provides physical control of speaker volume and mute (or talkback) functions. The hardware must connect either directly to the USB port of the D-Mon processor (front or rear), or a DAW running the Trinnov App. Note that connection via the web GUI in a browser is not supported (the Trinnov App is required). Multiple controllers can be installed – in this instance; all controllers will be assigned the same functionality.

Trinnov Protocols

Other interfaces such as the Colin Broad TMC-1 support Trinnov proprietary protocols. For more details, please contact your local Trinnov representative.

Functionality

The D-Mon Control Panel GUI assigns the processor’s available functions as follows:

- *SESSION SETTINGS -> Remote Controllers* – maps functions such as speaker sets, monitor sources, and cues to ICON control surfaces; and speaker sets and cues to Avid EUCON. For EUCON, the D-Mon sources are automatically seen with no need for manual declaration.
- *STUDIO SETUP -> Remotes* – maps Talkback, Mute, Dim, and AES Insert switching to the GPIOs. Options for the Griffin USB Volume Controller are also configured here.

4 THE D-MON PROCESSOR

This chapter provides an overview of the D-Mon processor **hardware**.

For more details on installing the processor, or wiring to and from the unit, please see *Chapter* 6*: Installation & Setup*.

4.1 FRONT PANEL

	Function		D-Mon Model
#A	**ON/OFF switch**	Used to boot-up and shut down the CPU. **ATTENTION!** This switch should *ALWAYS* be used to shut down the processor. Use the rear panel ON/OFF switch to cut the power only once the front green led is OFF. Brutal power loss or direct use of the rear panel ON/OFF may damage the internal components.	ALL
#B	**1 x USB 2.0 port**	Used to attach a remote controller like *La Remote*, a mouse, or a Griffin volume knob, or for inserting a USB key to save your presets and setup.	ALL

4.2 REAR PANEL

	Function	Connector
#1	Power plug: 220/110V (specified on order), with main fuse	CEE Main
#2	ON/OFF mains switch: applies power to the whole unit.	n/a
#3	VGA Output (depending on model)	SubD 15p
	HDMI Output	HDMI
#4	USB, for La Remote, Mouse, keyboard, or memory-stick inputs	4 x USB 2.0
#5	Network, gigabit ethernet	RJ-45
#6	AES Inputs & Outputs 1-8 (4 x AES3)	SubD-25*
#7	AES Insert 1-8 (4 x AES3)	SubD-25*
#8	Analog Inputs 1-8	SubD-25*
#9	Analog Outputs 1-8 (mirrored on both connector)	SubD-25*
		XLR-3
#10	AES Inputs & Outputs 9-16 (4 x AES3)	SubD-25*
#11	Analog Outputs 9-16	SubD-25*
#12	Multi-Purpose Inputs & Outputs	SubD-25
#13	75Ω Word clock input & output	2 x BNC

*SubD-25 audio connectors are wired according to the TASCAM standard – see *10.1.1 Audio SubD-25 Connectors*.

5 QUICK START

This chapter provides a **quick start** to get you up and running as quickly as possible.

We will assume that the D-Mon processor is already installed and configured – i.e., all connections made; all speaker sets defined and calibrated; the correct clock signal selected; etc. If not, please see *Chapter* 6*: Installation & Setup*.

Note that all the functions of the D-Mon Control Panel GUI are covered in more detail later in *Chapter* 7.

5.1 POWERING ON / BOOTING UP

Providing the processor has been fully installed and configured, it can be booted from the front panel.

- Press the front panel **ON/OFF** button to switch **ON** the processor:

Turn ON here

You will hear a click from the internal relay, and the **ON** button illuminates as shown above. The processor will take a few seconds to boot, so please wait a little while before opening the GUI.

At the end of the boot-up procedure, the processor may recall a specific preset (see *7.6.5: Setting the Default Preset*.) This should reset the processor's configuration so that it meets the needs of your studio.

ATTENTION!

To turn off the unit, *ALWAYS* use the front panel **ON/OFF** button to shut down the processor.

The rear ON/OFF button must only be used to completely remove the power, only once the front green led (which monitor the internal computer) is off. Premature removal of the power may damage the unit.

5.2 OPENING THE USER INTERFACE

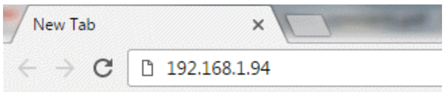
The **D-Mon Control Panel** can be accessed by opening a browser connection to the D-Mon processor from a networked device such as a desktop computer, laptop, smartphone or tablet.

System Requirements

- The device must be connected to the same IP network as used by the D-Mon processor.
- The D-Mon processor's network connection and IP settings must be properly configured (see *6.2.2: Setting up the CPU*).
- The device must run the Trinnov App, or alternatively support a suitable web browser: Chrome (recommended), Safari, Internet Explorer, Firefox, etc.

Opening the GUI

- ➢ Start the Trinnov App. The network discovery window opens. Select a D-Mon machine in the list and hit the connect button.
- ➢ Alternatively, start your browser software and type the IP address of the D-Mon processor into the search bar and press Enter – the GUI's **CONTROLLER** page should appear:



> The IP address of the D-Mon processor can be found in the "About" and "Setup -> Network" tabs of the Optimizer & Processor Control Panel (OPCP) during the setup of the processor. See *6.2.2: Setting up the CPU.*
>
> If a Trinnov La Remote is connected to the D-Mon, select the "network" layer to display its IP address.

Troubleshooting

If the machine is not listed in the Trinnov App, then check:

- Is the D-Mon processor booted? (the front panel **ON** button should be lit)
- Is the D-Mon in the same physical network as the computer (ie. in case of a dedicated control surface network), or logical (VLAN)?
- Is the Bonjour/Zero-conf protocol allowed in your network?

If the GUI does not appear, then check:

- Is the D-Mon processor booted? (the front panel **ON** button should be lit)
- Have you entered the correct IP address?
- Is there any computer firewall or browser limitation that prevents accessing the device ?
- Are both the D-Mon processor and your networked device connected to the same IP network?

5.3 RESIZING THE GUI

Before using the GUI, please check the screen resolution. In our example below, some functions are “missing” from the bottom of the **CONTROLLER** page as the GUI is being viewed on a 16:9 computer screen.

CONTROLLER Page (viewed at 16:9)

The GUI is optimized for 4:3. Therefore, please adjust your browser’s window size and screen resolution accordingly.

You should now be able to see all parameters - in our example, the headphones **ON**/OFF button, source select and volume.

CONTROLLER Page (resized for 4:3)

5.4 PAGES & NAVIGATION

The **D-Mon Control Panel** supports six pages – use the icons at the bottom of the GUI to change page:

> If you hover the cursor over an icon, then the page name is displayed (e.g., **CONTROLLER**).

The six pages are:

- **CONTROLLER** – this is the main operational page (shown above). From here, you can perform everyday tasks such as select a monitor source; switch between speaker sets; talk to a destination, and adjust speaker and headphone volume.
- **SESSION ROUTING** – controls the routing matrix. Here you can see the results of any automated switching, and make manual assignments for the session such as routing a mix bus to its output destination.
- **MONITORING MIXER** – controls the summing of the session's sources onto mix buses.
- **SESSION SETTINGS** – defines parameters which apply to the session such as the sources, mix buses, and outputs.
- **STUDIO SETUP** – defines parameters which apply to the studio such as the speaker sets, an audio clock signal, etc. If a red dot is lit, this denotes a major issue, either in the speaker calibration or in the audio clocking.
- **LE REMOTE SETUP** – defines parameters which apply to the attached *La Remote* control surface, opened in a new window.
- **AoIP SETUP** – defines parameters which apply to Audio over IP – Ravenna, opened in a new window.
- **CONFIGURATION PRESETS** – opens a pop-up menu where you can recall either a snapshot (to reset session parameters) or a preset (to reset the complete unit). The **MANAGE** button opens a full-size page with further file management options.

> The processor's model number can also be selected – in our example, **D6**. This opens the **Processor Control** GUI in a new browser window. It is used for setup functions such as the speaker/room calibration and is described later in this manual (see *6.4.1: Working with the D-Mon and Optimizer* Control Panel).

5.5 CONVENTIONS

In general, a color (usually green) indicates when something is enabled. So, in our example above, the **CONTROLLER** page is selected; the source named **Main** (on the left) is assigned to **Spk A** (on the right); the speaker listening level is open (indicated by the central rotary slider); the HEADPHONES are turned **ON,** and the headphone level is open (indicated by the horizontal slider).

5.6 RECALLING A PRESET

At this stage, it is a good idea to recall a preset to be sure that the processor's speaker sets, and other studio-related parameters, are configured correctly. Presets will be described in more detail later. For now, use the following steps to reset your system:

- Select **CONFIGURATION PRESETS** from the page icons at the bottom of the GUI – a pop-up menu appears.
- Check that the **STUDIOS PRESETS** tab is selected and scroll up/down to view the available presets.
- Click on a preset name to load its settings: e.g., **Dubbing session**.

It will take a second or two to load the new configuration; the preset highlights in green once the recall is complete.

CONFIGURATION PRESETS -> STUDIO PRESETS

Preset Recall Complete

Click here to open pop-up menu

- Click on the **X** (at the top right of the menu) to close the **CONFIGURATION PRESETS** pop-up.

For more details on snapshots and presets, please see *7.6: CONFIGURATION PRESETS.*

5.7 DAY-TO-DAY OPERATION

Once your processor is configured, all everyday tasks can be operated from the **CONTROLLER** page:

	Area	Primary Function
#1	SOURCES	Click to monitor an incoming source (e.g., **Main**) on the selected speaker set (#3). Multiple ones can be selected simultaneously in the “sum” mode.
#2	Speaker Level	Click and drag to adjust the speaker level.
#3	SPEAKERS	Click to select a speaker set (e.g., **Spk A**) – the individual speaker buttons update accordingly.
#4	OUTPUTS & TALKS	Click to monitor a mix or output on the selected speaker set. You can also talk to a mix bus if talkback is configured.
#5	HEADPHONES	Click to enable the headphone output (**ON**/OFF), select a source (e.g., **Main**) and adjust the level.

> The contents of the SOURCES (#1), OUTPUTS & TALKS (#4) areas will vary depending on the configuration done in the session settings. The content of the SPEAKERS (#3) area depends on the configuration done in the studio settings.
> The screenshot above is taken from a D-Mon|6 processor after recalling the **Built-in** preset.

	Area	Operations
#1	SOURCES (IN)	**Manages all incoming sources** defined in the SESSION SETTINGS. ➢ Click to monitor a source on the selected speaker set. ➢ Use the **TOGGLE** and **SUM** buttons to determine whether selections are exclusive or additive. You can listen to any combination of sources (#1) and outputs (#4). You may also see an AES INSERT **ON**/OFF button and an **INS** indicator beside some sources (if the "AES 1-8 Insert" option in the STUDIO SETUP -> Inputs & Outputs is enabled): ➢ Use the **ON**/**OFF** button to switch the AES insert return in and out of the circuit. See Switching the AES 1-8 Insert: Switching the AES 1-8 Insert for more details.
#2	Speaker Level	**Adjusts the listening level for the speakers** (applies to all speakers within all speaker sets): ➢ Click and drag on the dB value to increase or decrease the speaker level. ➢ Click on **DIM** to reduce the listening level – the button lights in orange when enabled. ➢ Click on **MUTE** to cut all speakers within the speaker set – the button lights in red when enabled. ➢ Click on **Ref** to reset the listening level to the reference value. The speaker level can be displayed as an unreferenced dB value (as shown above), referenced to the calibrated speaker measurement, or displayed as a Commercial Cinema way (from 0 to 9.9). This, plus the dim, reference and maximum speaker level are defined under STUDIO SETUP -> Options/Levels.
#3	SPEAKERS (OUT)	**Manages all speaker sets** defined in the STUDIO SETUP -> Speaker Sets. ➢ Click to switch to a different speaker set (if more than one is configured). A green "V" (from the Trinnov logo) appears beside each optimized speaker set. ➢ The buttons at the bottom of the SPEAKERS area can be used to solo, deliver pink noise to, or mute an individual speaker. First, choose the mode (e.g., **SOLO**), and then select a speaker (e.g. **C**). Each of the functions is color-coded to indicate when a speaker is muted (red), in solo (yellow) or sending out pink noise (turquoise). ➢ Click on **DownMix** or **DRC** (Dynamic Range Compression) to enable these functions. The drop-down menu chooses the downmix format (e.g., **Stereo**). A green checkmark refers to a speaker set fully handled by the Trinnov Optimizer, the lack of such mark (like in the above screenshot for **Spk B**) refers to a non-optimized speaker-set without acoustical correction (Cf. non-optimized speaker-set under STUDIO SETUP -> Speaker sets).
#4	OUTPUTS & TALKS	**Manages all mix buses and outputs** defined in the SESSION SETTINGS. ➢ Click on the button name (e.g., **Mix – Output #1**) to listen to the mix on the selected speaker set. ➢ Use the **TOGGLE** and **SUM** buttons (under #1 SOURCES) to determine whether selections are exclusive or additive. You can listen to any combination of sources (#1) and outputs (#4). If talkback to a mix bus is enabled in the SESSION SETTINGS page then you will see the relevant master **SLATE** button(s) plus a talkback symbol (on the right of the button name): ➢ Click on the talkback symbol to open a communication line between the talkback input and the selected destination. Click again to close the talkback line. Note that several talkback lines can be open at the same time. ➢ Use the **SLATE 1** and **SLATE 2** buttons to talk to multiple destinations: **SLATE 1** talks to ALL destinations assigned to talkback input 1, and **SLATE 2** to ALL destinations assigned to talkback input 2. The **SLATE** buttons override any individual talkback selections. If the listen-back inputs are enabled in the STUDIO SETUP -> Inputs & Outputs, then the relevant **LISTEN** buttons will appear: ➢ Click on **LISTEN 1** (or **LISTEN 2**) to monitor the listen-back input on the selected speaker set. Click again to cancel the listen-back. Listen-back can be used to return talkback from a particular studio location – for example, to "listen in" to the conductor's microphone during a recording session.

	Area	Operations
#5	HEADPHONES	**Controls the stereo headphone output**. ➢ Click on the **ON**/**OFF** button to activate/deactivate the headphones. ➢ Use the drop-down menu to select the monitor source – you can choose any of the sources, mixes or outputs defined in the *SESSION SETTINGS* page, or a listen-back input if they are enabled. If **Phones source follow monitor** is selected, under *SESSION SETTINGS -> Options*, 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. A maximum headphone level can be defined under *STUDIO SETUP -> Options/Levels*.

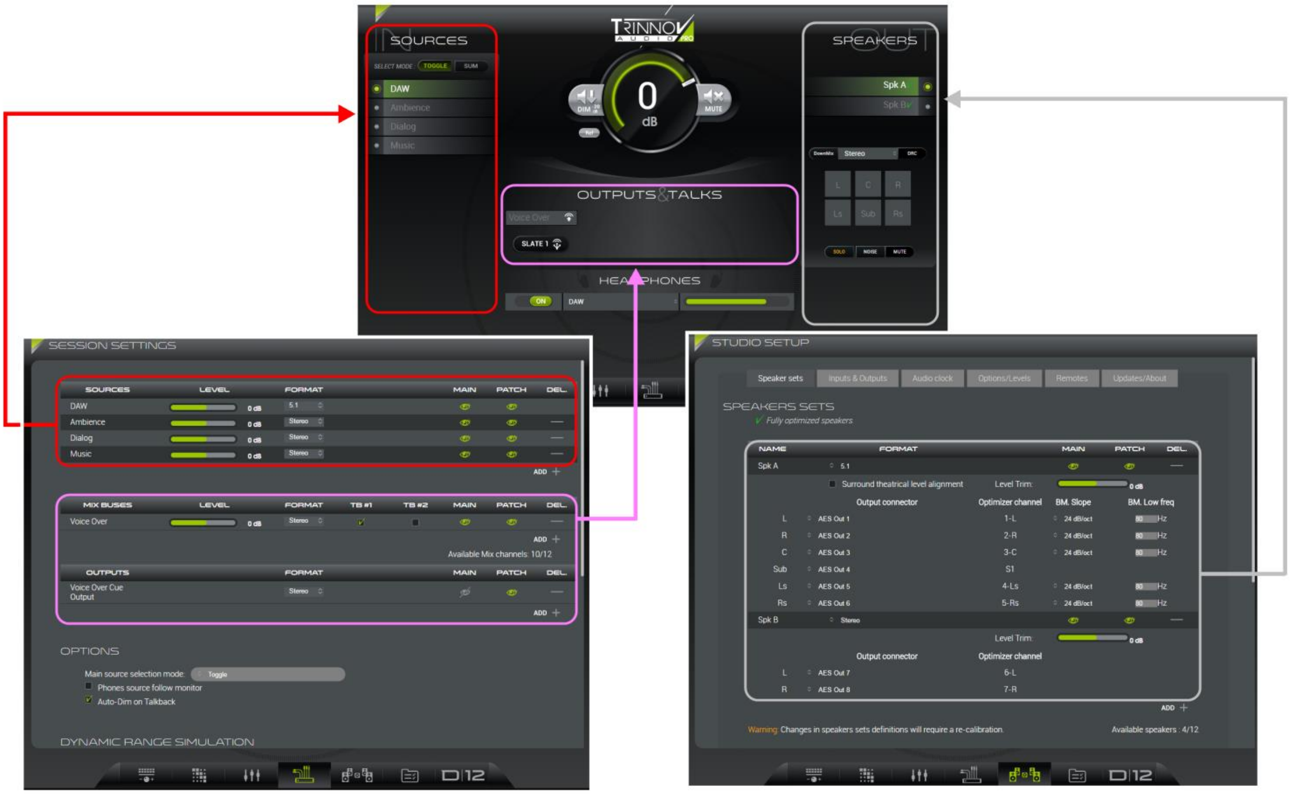
For more details on any of the above operations, please see *7.1: The CONTROLLER Page*.

5.8 MODIFYING THE CONFIGURATION

If the contents of the **CONTROLLER** page do not meet the needs of the session, then you will need to modify the configuration. First, it is useful to understand how the GUI pages interact, and how settings are saved using snapshots and presets.

5.8.1 USING THE GUI PAGES

The diagram below shows how the **CONTROLLER** page is affected by the **SESSION SETTINGS** and **STUDIO SETUP** parameters:



The system has been designed with the following in mind:

- All studio-specific parameters can be found in the **STUDIO SETUP**. They include the definition of the speaker sets (shown above) and other parameters such as the bass-management configuration, talkback and listen-back input assignments, audio clock options, etc.
- At a level above the **STUDIO, SETUP** are all session-related parameters. These are defined in the **SESSION SETTINGS, SESSION ROUTING,** and **MONITORING MIXER** pages. The **SESSION ROUTING** and **MONITORING MIXER** will be described later in chapter 7*: The D-Mon Control Panel*. For now, we will concentrate on the **SESSION SETTINGS** page which defines the sources, mix buses and outputs.

5.8.2 ABOUT SNAPSHOTS AND PRESETS

To save and load settings, the system uses two different file types:

- A studio **preset** stores everything required to reset the D-Mon processor – i.e., all the D-Mon Control Panel GUI parameters plus other lower-level settings defined during installation (such as the Optimizer channels DSP filters[1]).
- A session **snapshot** stores only session-related parameters – i.e., the current state of the **CONTROLLER** page, plus the **SESSION SETTINGS**, **SESSION ROUTING** and **MONITORING MIXER**.

You should use presets to store settings for the studio – for example, different monitoring arrangements. Then use snapshots to store settings for each DAW session – for example, the sends and returns to the DAW and studio floor.

This allows you to reset the unit completely using only a preset. OR, recall a preset followed by a snapshot to apply different session-related parameters to different monitoring arrangements.

For more details on managing snapshots and presets, see Chapter *7.6: CONFIGURATION PRESETS*.

5.8.3 RECOMMENDED WORKFLOW

During Installation

The **STUDIO SETUP** parameters and, in particular, the speaker sets[1] and audio clock, should be configured during installation. As a result, you should already have some user presets which will customize the D-Mon processor for your studio and its monitoring arrangements. See *5.6 Recalling a Preset*.

Note that, if you make changes to the **STUDIO SETUP** parameters, then you must remember to create a new preset, or overwrite an existing file, to store the changes.

Before a New Session

Before each new DAW session, you can then configure the necessary **SESSION SETTINGS**, **SESSION ROUTING,** and **MONITORING MIXER** parameters. These should be saved in snapshots for a recall after a reboot or at a later date.

It is strongly recommended that you save regularly so that you can recall a snapshot if you make a mistake while editing the session configuration.

Everyday Operation

Once both the studio and session parameters have been defined, the system is ready for operation from the **CONTROLLER** page.

[1] **About the Speaker/Room Calibration**

Each speaker set requires calibration for your listening environment. This is handled by the D-Mon processor's Optimizer®; the number of available Optimizer channels varies according to your D-Mon model.

The definition of each speaker set is handled by the **D-Mon Control Panel** (DMCP) GUI. Select the *STUDIO SETUP -> Speaker Sets* tab to view the current configuration. The calibration procedure is then performed at a lower level using the **Optimizer & Processor Control Panel** (OPCP) GUI. This can be opened in a new browser window by clicking on your D-Mon model number (e.g., **D|12**).

Both the definition of the speaker-sets and their calibration should be handled during the installation of the D-Mon processor, and stored in presets. You can then recall a preset to reset the system. Note that several presets can be created to support multiple monitoring arrangements and "sweet spots" within your studio (e.g., mixing seat, producer desk, etc.).

See *6.4: Configuring the System* for more details.

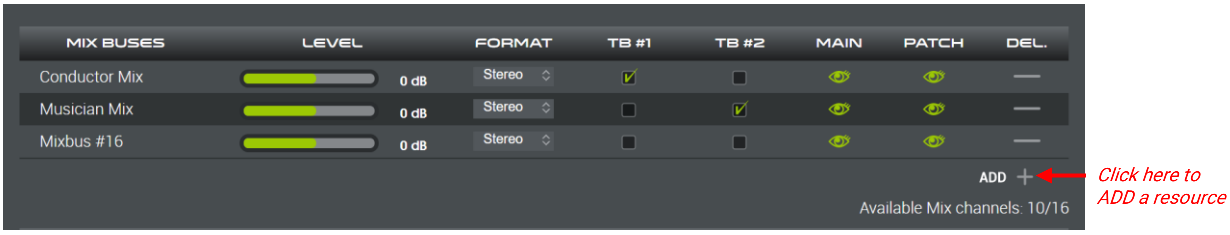
5.9 EDITING THE SESSION SETTINGS

At the top of the **SESSION SETTINGS** page, you will see the available SOURCES, MIX BUSES and OUTPUTS for the session. These determine what resources are available in the main **CONTROLLER**, **SESSION ROUTING** and **MONITORING MIXER** pages as described earlier (see *Using the GUI Pages*).

Possible Operations

For each type of resource:

- Click on the **ADD +** button (at the bottom of each section) to configure a new entry - the resource is given a default name (e.g. **Mixbus #16**) and a generic set of parameters (**Level** = **0dB**, **Format** = **Stereo**, **TB** = off, **MAIN** = on, **PATCH** = on):



The maximum number of SOURCES and OUTPUTS is limited only by the number of physical inputs and outputs. An input can be used in several sources, whereas an output can only be used once in the output and speakers definitions.

A green ethernet logo in the AoIP column denotes a source or an output patched to an AoIP channel. AoIP I/O channels cannot be associated with regular I/O (analog / AES3) within the same SOURCE or OUTPUT: all connectors must match the same kind for a given entry. AoIP SOURCES and OUTPUT may be created automatically upon stream declaration (see Chapter 9: AoIP operations).

The maximum number of MIX BUSES is 16 mono channels. The available number of mix channels is stated on-screen. In our example, **Available Mix channels: 10/16** means that we have ten mono mix buses remaining from the 16 available in our D-Mon.

- Click on a parameter to edit its value – all parameters are available for all resources, except for talkback which can only be enabled for mix buses, and levels which can only be adjusted for sources and mix buses. The available parameters are covered on the next page.
- Click on the DELETE symbol (in the right-hand column) to remove a resource.

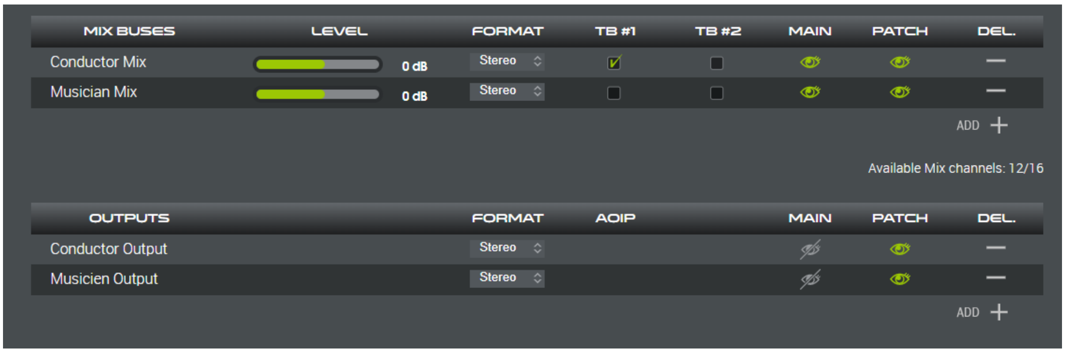
> There is no undo for the delete operation, so take care when selecting this symbol! Deleting a SOURCE, MIX BUS or OUTPUT will remove it from the **CONTROLLER**, **SESSION ROUTING** and **MONITORING MIXER** pages.

Available Parameters

- **Name** – click to edit the name. You can click once to position the cursor, or double-click to select the text you wish to overwrite. Type in a name; the usual copy, cut and paste functions are available. Each name can be up to 16 characters, and all alphabetical and numerical signs are accepted.
- **LEVEL** (SOURCES and MIX BUSES only) – click and drag on the slider to adjust the source or mix bus level. Alternatively, double-click on the white text field, type in a value (e.g., 6dB) and press Enter. (Does not accept decimal numbers)
- **FORMAT**– click to choose a format from the drop-down menu. If you choose a surround format, then the component channels are always allocated in the same order, up to the relevant number of channels: L, R, C, Sub, Ls, Rs, Lrs, Rrs, HL, HR, HLs, HRs.
- **TB #1, TB #2** (MIX BUSES only) – click to enable talkback to the mix bus from either or both of the talkback inputs. Once enabled, the relevant master **SLATE** button(s) plus a talkback symbol appear on the CONTROLLER page, see *7.1.8: Talking to an Output*.
- **MAIN** & **PATCH** – the "eye" icons determine whether a resource is visible in the **MAIN** (CONTROLLER) and **PATCH** (SESSION ROUTING) pages. When enabled, the icon lights in green. Typically, this feature is used to hide a resource from the CONTROLLER page - for example, to hide a tone source or cue output (see the tip below).

Configuring Cue Feeds

If you wish to use the D-Mon processor to handle your session's cue feeds, then you should add both a mix bus and an output for each cue – for example:



Once added to the **SESSION SETTINGS**, each cue mix can be adjusted from the **MONITORING MIXER** page, and each cue mix routed to its output from the **SESSION ROUTING** matrix. To make the connections, you will need to leave all the **PATCH** icons enabled (so that both the cue mixes and their outputs are visible in the **SESSION ROUTING**).

TB 1 is enabled for the Conductor Mix and TB 2 for the Musician Mix – this will allow independent talkback to each cue bus from the **CONTROLLER** page. To make sure you can access the talkback switching and monitor the cue bus, you should leave the **MAIN** icons enabled for the cue mixes (so that they are visible in the **CONTROLLER** page).

However, it is recommended to disable the **MAIN** icons for the cue outputs. This will simplify the **CONTROLLER** page, where it is not necessary to view both the cue mix and its output.

Note that the cue mix **LEVEL** can be set as you wish. This is the same as the cue mix fader level in the **MONITORING MIXER**.

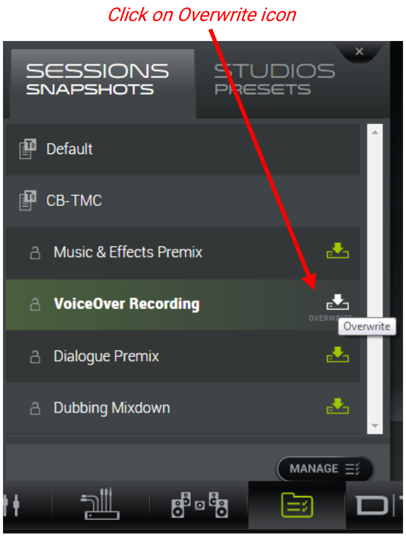
For more details on the remaining **SESSION SETTINGS**, please see *7.4: SESSION SETTINGS.*

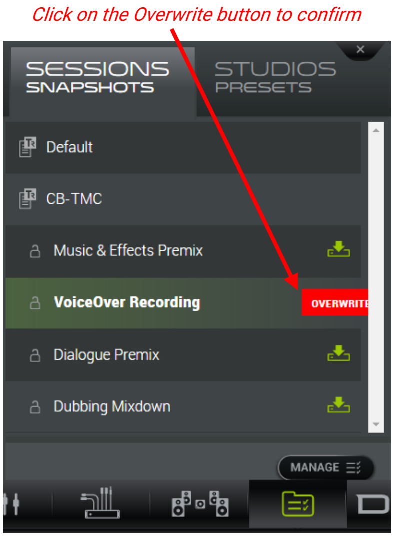
5.10 SAVING A SNAPSHOT

To save any changes to the **SESSION SETTINGS**, you should save a snapshot. There are two possibilities: overwrite an existing snapshot or create a new file.

To overwrite an existing snapshot:

- First, select **CONFIGURATION PRESETS** from the page icons at the bottom of the GUI – a pop-up menu appears.
- Check that the **SESSION SNAPSHOTS** tab is selected and scroll up/down to view the available snapshots.
- Click on the **Overwrite** icon (to the right of the file name) - a red **OVERWRITE** button appears.
- Click on the button to confirm – the current settings are written into the existing file:





- Click on the **X** (at the top right of the pop-up menu) to close the **CONFIGURATION PRESETS** pop-up menu.

For more details on managing snapshots and presets, please see *7.6: CONFIGURATION PRESETS*.

5.11 NEXT STEPS

This completes the quick start.

For more details on any of the GUI pages, please see Chapter 7*: The D-Mon Control Panel*.

6 INSTALLATION & SETUP

This chapter covers the **installation and setup** of the system.

Here we will focus on a stand-alone installation to be controlled via the D-Mon Control Panel GUI. If you wish to connect an external control surface, then please see Chapter 8*: External Control* for additional information.

> **ATTENTION!**
> You should read and observe *ALL* of the Important Safety Instructions *BEFORE* installing or operating your system.

Installation Checklist

To install and set up the device, you will need to:

- Unpack and check the contents of the shipping box.
- Start up the processor and configure its network connection.
- Connect the audio devices & other wirings (Wordclock, GPIOs, etc.)
- Configure the system – create and calibrate the speaker sets, define the audio clock, etc.

Further Information

Additional installation and wiring diagrams can be found in the Appendices, see *10.1: Wiring Diagrams*.

6.1 WHAT’S IN THE BOX?

Your shipping box includes the following items. Note that some items are optional.

Please check the contents, and in the event of any transport damage, contact the Trinnov service department at support@trinnov.com.

Item

Processor (19”, 2U Rack)

Mandatory.

Power cord

Mandatory.

3D-Microphone (optional)

Required for speaker/room calibration.

3D-Mic breakout cable (optional)

For quick connection of the 3D-Microphone to the analog inputs available on Sub-D25.

MPIO breakout cable “ICON” (optional)

To connect a D-Mon processor to an Avid ICON control surface such as D-Command or D-Control. Discrete inputs & outputs: GPIOs, Listen-back, LTC.

MPIO breakout cable “MIDI” (optional)

To connect a D-Mon processor to a MIDI controller. Discrete inputs & outputs: MIDI I/O, Talkback, Listen-back, Headphones, GPIOs, LTC.

La Remote USB Controller (optional)

For physical control of the monitoring functions,

including speaker volume and mute.

6.2 STARTING UP THE D-MON PROCESSOR

Before passing any audio signals, you should make sure that the D-Mon processor is booted and correctly set up. We recommend you do this before installing the unit into its final location (rack or housing), so that it is easier to access the connectors and switches.

Please follow each of the steps carefully and take note of all comments.

The location of all switches and connectors (#) can be found by referring back to topics *4.1: Front Panel* and *4.2: Rear Panel*.

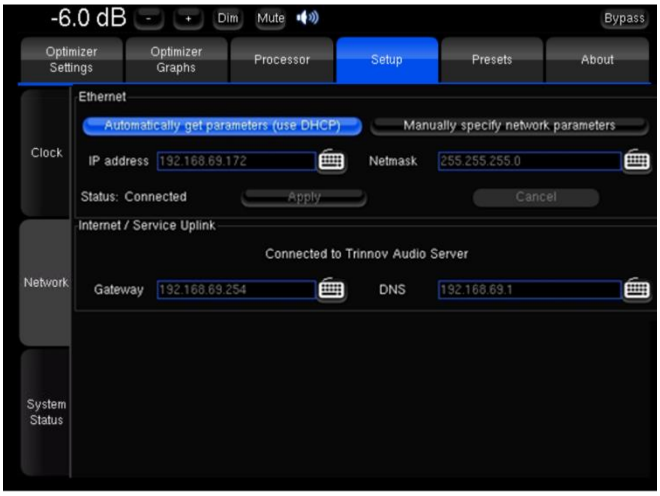
6.2.1 BOOTING UP THE CPU

Step	Instructions	Comments
1	Connect the unit to an appropriate power socket.	**THE 220/110V SELECTION IS NOT AUTOMATIC.** It can only be made by an authorized person. Check with your reseller for further information.
2	Switch **ON** the mains using the rear panel switch (#2) to apply power to the unit.	ATTENTION! NEVER use this switch to shut down the processor. Instead, use the front panel ON/OFF switch (#A) to shut down *BEFORE* switching off the mains power.
3	Connect a VGA or HDMI screen to the connectors (#3). Power up this device.	The screen and mouse can be removed once the unit's network connection is established. However, they are **highly recommended** when starting up for the first time as they provide easy access to the initial parameters.
4	Connect a USB mouse to one of the rear ports (#4) or the front panel (port #B).	
5	Connect an Ethernet cable (RJ45-CAT5/6) to the network port (#5).	The other side of the cable should connect to a DHCP router; any switch is allowed. The router will require internet access to allow the download of software updates and control setups at a later stage.
6	Switch **ON** the front panel button (#A) to boot up the processor. ATTENTION! *ALWAYS* use this switch to shut down the processor (*BEFORE* eventually powering off from the rear panel when the front red led is OFF).	The following should happen: ✓ Fans start to turn and then become silent. ✓ You will hear one click from an internal relay. ✓ The front panel **ON** switch illuminates. ✓ The screen displays the boot-up sequence.
7	Check the statuses displayed on the screen.	The D-Mon firmware should display the following on your VGA or HDMI screen: ✓ Computer motherboard logo ✓ Trinnov logo ✓ Trinnov boot page ✓ Optimizer & Processor Control Panel as shown on the next page.

Once successful, the D-Mon processor is up and running!

The next steps are to configure the CPU's network access so that the processor can be remotely controlled from a networked device.

6.2.2 SETTING UP THE CPU

Step	Instructions	Comments
1	From the Optimizer & Processor Control Panel , select the “About” page:	
2	Here you will see the following important information which should be noted down for future reference:	✓ Product ID: required for remote control via VNC. ✓ Serial Number: required for future connection via Ethernet. ✓ Microphone Number: required during the calibration procedure. ✓ License: the number of licensed Optimizer channels. ✓ IP address: the IP address of the system. If an address is present, then network access is already configured.
3	Select the “Setup” page followed by the “Network” sidebar tab:	
4	By default, the system is set to receive its IP address via DHCP. Therefore, if the unit has been connected to a DHCP network, you should see the following:	✓ Automatically get parameters (use DHCP) button is on. ✓ The “IP address” and “Netmask” fields are properly completed. ✓ The sentence “Connected to Trinnov Audio Server” is displayed. If any of the above conditions are missing, check your Ethernet cabling and IP settings. If necessary, click the **Manually specify network parameters** option to enter the processor IP settings manually. Please contact your network administrator for details.

At this stage, it is a good idea to test the network communication by opening a browser connection to the processor, see *5.2: Opening the* .

The optimizer control panel (shown above) is used for other setup functions such as calibration and audio clock selection. These will be described later in this chapter.

6.3 STUDIO INTEGRATION & WIRING

This section describes each of the rear panel connectors and their intended signal assignments.

D-Mon Processor Rear Panel

6.3.1 AUDIO CONNECTORS

All SubD-25 audio connectors are wired according to the Tascam standard – see *10.1.1: Audio SubD-25 Connectors*.

	Connector	Function
#5	**Ethernet** Gigabit	Used to send and receive AoIP signals for the Ravenna software option. Cf. chapter 9: AoIP operations **ATTENTION!** The D-MON cannot be elected as clock leader on the network, another AoIP device on the network will take this ownership. It is **MANDATORY** that this PTP leader device has its clock in sync with the D-MON internal one, either via word-clock or AES (the D-Mon can be either clock leader or clock follower).
#6	**AES I/O 1-8** SubD-25	Connects signals to and from your DAW session in the digital domain: **AES In 1-8 (4xAES3)** = the main signals coming from your DAW session, such as the main mixes, aux sends discrete tracks, etc. The routing of inputs is fully flexible, and any combination of formats is possible - for example 4 x stereo; 1 x 5.1 + 1 x stereo; 1 x 7.1; and so on. **AES Out 1-8 (4xAES3)** = signals returning to your DAW session, such as the analog inputs OR AES insert returns (if the INSERT is active).

	Connector	Function
#7	**AES I/O INSERT** SubD-25	Designed to offer an instant comparison between say a dry main mix (up to 7.1) and the wet mix returning from an external digital effects processor. The 8 AES in/out should be wired to and from the insert sends and returns of the external processor. **ATTENTION!** The insert sends *ALWAYS* follow the signals assigned to AES In 1-8 on connector #6. When the INSERT is active, all eight returns are switched to AES Out 1-8 on connector#6. This means: **INSERT OFF** = AES 1-8 signals come from AES In 1-8 on connector #6. **INSERT ON** = AES 1-8 signals come from AES In 1-8 on connector #7. Insert switching is *ALWAYS* made in one block; it is not possible to insert less than eight signals. Note that inserts are direct outputs extracted at the AES input stage. This means that no processing delay is added. Once the INSERT is active, the AES input is taken from the AES I/O INSERT connector, including the clock. If an external AES clocking is used and no reference signal is detected, the INSERT activation should be checked.
#8	**ANA IN 1-8** SubD-25	Designed to connect incoming signals from analog studio equipment like recorders, players, and external signal processors. Inputs 1 & 2 are reserved for Talkback and Listen-back and include phantom power (see note below). When performing an acoustic calibration via the Optimizer®, you will need to connect the 3D-Microphone (4 channels) to the first four analog inputs. Therefore, we recommend wiring these inputs to a patch bay which is easily accessible. If you are swapping out an X-Mon and want to remain in the analog domain, these inputs can carry the main mix or/and auxiliary signals. **ATTENTION!** Talkback and Listen-back can be moved to other inputs (analog or digital), but they will lose their 36V phantom power, and cease to appear on the MPIO (see connector #12).
#9	**ANA OUT 1-8** SubD-25/XLR-3	These outputs should connect first to your loudspeakers and then to other analog devices such as live headphones, recorders, and external signal processors. **ATTENTION!** Both connectors (XLR and SubD) carry the same signals.
#10	**AES I/O 9-16** SubD-25	Designed to connect signals to and from your digital studio equipment like external processors, recorders, etc. **ATTENTION!** When using multiple digital devices, take care that the whole studio is correctly synchronized by installing an appropriate word clock generator and distributing the clock signal via a “star” wiring network. Note that digital gear is always sensitive to proper earthing of all digital devices in the loop.

	Connector	Function
#11	**ANA OUT 9-16** SubD-25	Outputs 9 & 10 are reserved for headphones (see note below). You should then connect any other analog devices not covered by connector #9 – for example, non-optimized loudspeakers, live headphones, recorders, external signal processors, etc. Some suggested arrangements are: **D-Mon|4 & 6**: headphones on 9-10; miscellaneous signals on 11-12. **D-Mon|8**: headphones on 9-10; miscellaneous signals on 11-16. **D-Mon|12**: headphones on 9-10; the remaining four optimized speaker channels on 11-14; miscellaneous signals on 15-16. **ATTENTION!** The headphones can be moved to other outputs (analog or digital) but will require an external headphone amplifier, and cease to appear on the MPIO (see connector #12).

6.3.2 MPIO & BNC CONNECTORS

	Connector	Model	Function
#12	**Multi-Purpose I/O** SubD-25	ALL	This multi-purpose connector carries the following signals: • MIDI input & output: compliant with the General MIDI 2 standard • Talkback input: balanced input (with phantom power) • Listen-back input: balanced input (with phantom power) • Headphone output: stereo unbalanced output • 2 x GPI and 1 x GPO **ATTENTION!** The MPIO connector handles both audio and control signals. Specific breakout cables are available for different devices: either MIDI (for generic devices) OR Avid ICON (for D-Command & D-Control). The Talkback and Listen-back inputs are hard-wired to analog inputs 1 & 2 (on connector #8). The Headphone output is hardwired to analog outputs 9 & 10 (on connector #11). If any of these signals are re-assigned, then they will NOT appear on the MPIO connector. See Appendix *10.1.2: Multi-Purpose In/Out (MPIO) SubD-25 Connector* for wiring information.
#13	**WORDCLOCK IN & OUT** 2 x BNC (75Ω)	ALL	When using multiple digital devices, you should take care that the whole studio is correctly synchronized by installing an appropriate word clock generator and distributing the clock signal via a “star” wiring network. • **WORDCLOCK IN** – accepts Wordclock from the master generator. • **WORDCLOCK OUT** – can be used to send Wordclock to another digital device.

6.4 CONFIGURING THE SYSTEM

There are some configuration tasks which should be performed when you install a system. These are:

- Define and calibrate the speakers (using the Optimizer®).
- Define the audio clock signal.
- Define other studio-related parameters such as the talkback and listen-back inputs.
- Save all settings in a preset (or several presets).

> It is strongly recommended that you create and update a preset throughout the configuration process. This will ensure that you do not lose any settings should the processor reboot.

6.4.1 WORKING WITH THE D-MON AND OPTIMIZER CONTROL PANELS

Some of the configuration tasks can be performed from the **D-Mon Control Panel** (DMCP), while others can *only* be accessed from the **Optimizer & Processor Control Panel** (OPCP). Therefore, the remainder of this chapter will refer to both GUIs.

Both GUIs can be accessed by opening a browser connection to the processor from a suitable networked device.

See *5.2: Opening the* for details on opening the **D-Mon Control Panel** (DMCP).

From the DMCP, click on the D-Mon processor model number (e.g., **D|12**) to open the **Optimizer & Processor Control Panel** (OPCP) in a new browser tab:

D-Mon Control Panel (DMCP)

Optimizer & Processor Control Panel (OPCP)

The **Optimizer & Processor Control Panel** (OPCP) is a lower-level interface to the D-Mon processor than the DMCP GUI. Use the page buttons running across the top of the GUI to access its pages (e.g., **Optimizer Settings**, **Optimizer Graphs**, etc.).

> Many parameters can be accessed from both interfaces in parallel and, in this case, the DMCP GUI is recommended. Therefore, only the necessary OPCP functions are described in this manual. If an OPCP page/tab/parameter is not described, then please assume that the same function can be controlled from the DMCP GUI and is covered in *Chapter* 7.

6.4.2 USING THE OPTIMIZER & 3D-MICROPHONE

Before calibrating the system, it is useful to understand more about the Optimizer and 3D-Microphone.

6.4.2.1 "OPTIMIZER": THE TRINNOV DIGITAL ACOUSTIC CORRECTION

The "Optimizer" is a digital processing module included in every D-Mon processor. It provides studio monitoring loudspeakers with the best possible response **according to the particular room** where they are installed. This means that no "pre-calibration" can be made before installation and that after a change to the room or speakers, a new calibration should be made.

Thanks to **advanced, patented and unique algorithms,** the Optimizer module can fix acoustic problems such as changes in energy response, frequency response, time alignment, level alignment, early reflections interferences, room modes, phase coherence, and so on. The system works by assigning a dedicated "filter" channel to each loudspeaker within an Optimized Speaker Set. Thus, depending on the D-Mon model number, multiple speaker configurations can be handled at the same time.

Thanks also to **presets**, it is possible to manage the same loudspeakers in different arrangements – for example, create an optimized stereo pair from the LR of a 5.1 setup. Or, handle lots of different loudspeakers – for example, a **D-Mon| 6** could handle 12 different loudspeakers by using presets for 3 x optimized stereo pairs OR 1 x 5.1.

Making it happen:

The D-Mon processor must be installed at the final stage in the audio chain before the loudspeakers. This allows the Optimizer to process all the signals to be played out from the "tuned" loudspeakers (as with a standard loudspeaker EQ):

Your studio's mixing gear

Analog console,

DAW Audio Interfaces,

Mastering, Externals

Your Speaker Sets

5.1,

Stereo A,

Stereo B

Then, to compute the "correction filter" applied by each Optimizer channel, you must perform a calibration run:

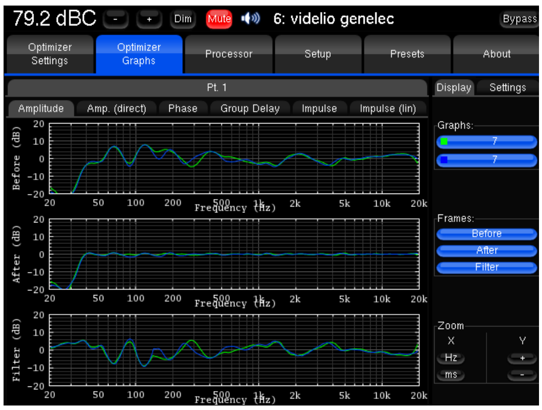
i. This process requires the Trinnov 3D-Microphone to be placed at one or more listening points to sample how the room reacts – for example, at the mixing position, the producer's couch, etc.

ii. At each listening point, the D-Mon processor generates a special noise (MLS bursts) and outputs this to the loudspeakers. It then samples how the noise bursts are reproduced.

iii. By comparing the generated noise bursts with the speakers' reproduction, the processor can calculate the most appropriate correction filters to apply. Lots of graphs are also generated at this stage to help with user-defined fine tuning.

iv. Following this initial acoustic tuning, many fine adjustments can be made to the frequency response, phase coherence, latency, impulse response, etc. via the OPCP GUI (shown the opposite). Depending on the actual acoustic situation, the Optimizer makes the best compromise between all user-defined parameters.

v. The final tunings can be stored in presets. This allows you to store and compare alternate fine tunings as required.



From here on, you will have the best possible sound from your room AND your loudspeakers.

6.4.2.2 THE TRINNOV 3D-MICROPHONE

The Trinnov 3D-Microphone has been designed specifically for Optimizer room correction. It is equipped with measurement transducers and is not designed for any other application (such as sound recording).

The design has been conceived alongside the Optimizer algorithms to sample the characteristics of the monitoring loudspeakers and their layout in the control room.

For each loudspeaker, the Optimizer gathers the following information:

- Full 3D position
- Amplitude response
- Phase response

Checking the Power Supply

The Trinnov 3D-Microphone is equipped with an internal battery that any user can change. The 9V PP3 LR61 battery is located within the body of the microphone and is accessible from the bottom of the unit.

To check and replace the battery:

- ✓ Power on the microphone, using the button on the bottom of the unit, and check the status of the On/Off LED. If the LED does not illuminate, then you will need to change the battery.
- ✓ Using an appropriate driver, remove the screw on the bottom of the unit, and slowly pull down the lower section of the housing to reveal the battery block.
- ✓ Unplug the battery from its connector and replace with a brand new 9V PP3 LR61.
- ✓ Power on and check the status of the On/Off LED.
- ✓ Carefully replace the housing by sliding it back onto the battery block, and tighten the screw to fix in place.

Checking the 3D-Microphone Compensation File

All Trinnov 3D-Microphones are verified and calibrated before leaving the factory. At this stage, a specific compensation file (for the microphone) is created. This allows any 3D-Microphone to be used with any Trinnov processor which supports the Optimizer as, by installing the correct compensation file, the overall measurement path can be flattened.

To check that your processor is using the correct microphone compensation file:

- ✓ Take note of the serial number of the 3D-Microphone (printed along the bottom of the microphone body). The serial number is usually written as "V9 N 268", where 9 is the lot number and 268 is the serial number.
- ✓ From the OPCP GUI, select "Optimizer Settings -> Calibration," and click on the **Configure** button beside the measurement name – from here you can select any pre-installed microphone compensation file (see below).
- ✓ Then select **Default Microphone** to select the file applied to every new preset created from the "built-in" preset.

In both cases, the compensation filename must match the 3D-Microphone serial number you are using. If not, then any Calibration made will provide absolutely the wrong room correction!

If the file for your 3D-Microphone is not available, then you must either upload or install a new file (as described below). Alternatively, you can select the filename "Vx-mean" where x = the lot number.

Only an appointed Trinnov reseller or the Trinnov Support Department can upload or provide a new compensation file:

- ✓ If possible, connect your processor to the internet and contact the Trinnov Support Department with the serial number of both the 3D-Microphone (printed on the microphone body) and D-Mon processor (displayed in the OPCP GUI "About" page). The support team will then upload the required file at the earliest opportunity.
- ✓ If an internet connection to the processor is not possible, then a USB Key can be provided. Plug the Key into one of the processor's USB ports – the Key File Manager automatically opens. Select the **Load microphone files from USB Key** option and follow the on-screen instructions.
- ✓ To refresh the microphone compensation file list, the unit needs to be rebooted.
- ✓ Once the upload/install is complete, the compensation file can be selected as described above.

6.4.3 PREPARING FOR CALIBRATION

6.4.3.1 DEFINING THE SPEAKER SETS

Before running the first calibration, you will need to define the speaker sets. This is handled from the D-Mon Control Panel GUI, as described later in *7.5.2: STUDIO SETUP -> Speaker Sets*. Note that:

- Speaker sets are divided into two categories, optimized and non-optimized. Both categories require calibration and, therefore, should be treated similarly during the calibration process.
- To create multiple arrangements using the same loudspeakers, you should add a separate Speaker Set for each arrangement and then assign the same physical outputs to the shared speakers. For example, to create a stereo pair from the LR of a 5.1 setup, define two Speaker Sets: Stereo and 5.1; then assign the same physical output to the Stereo front Left, and 5.1 front Left, and repeat for the Right front speaker.
- You can create multiple presets for the same Speaker Sets (e.g., to deal with different acoustic arrangements). In this case, each preset will require its calibration process.
- You cannot perform calibration without first defining a Speaker Set.

6.4.3.2 RECOMMENDATIONS FOR SPEAKER PLACEMENT

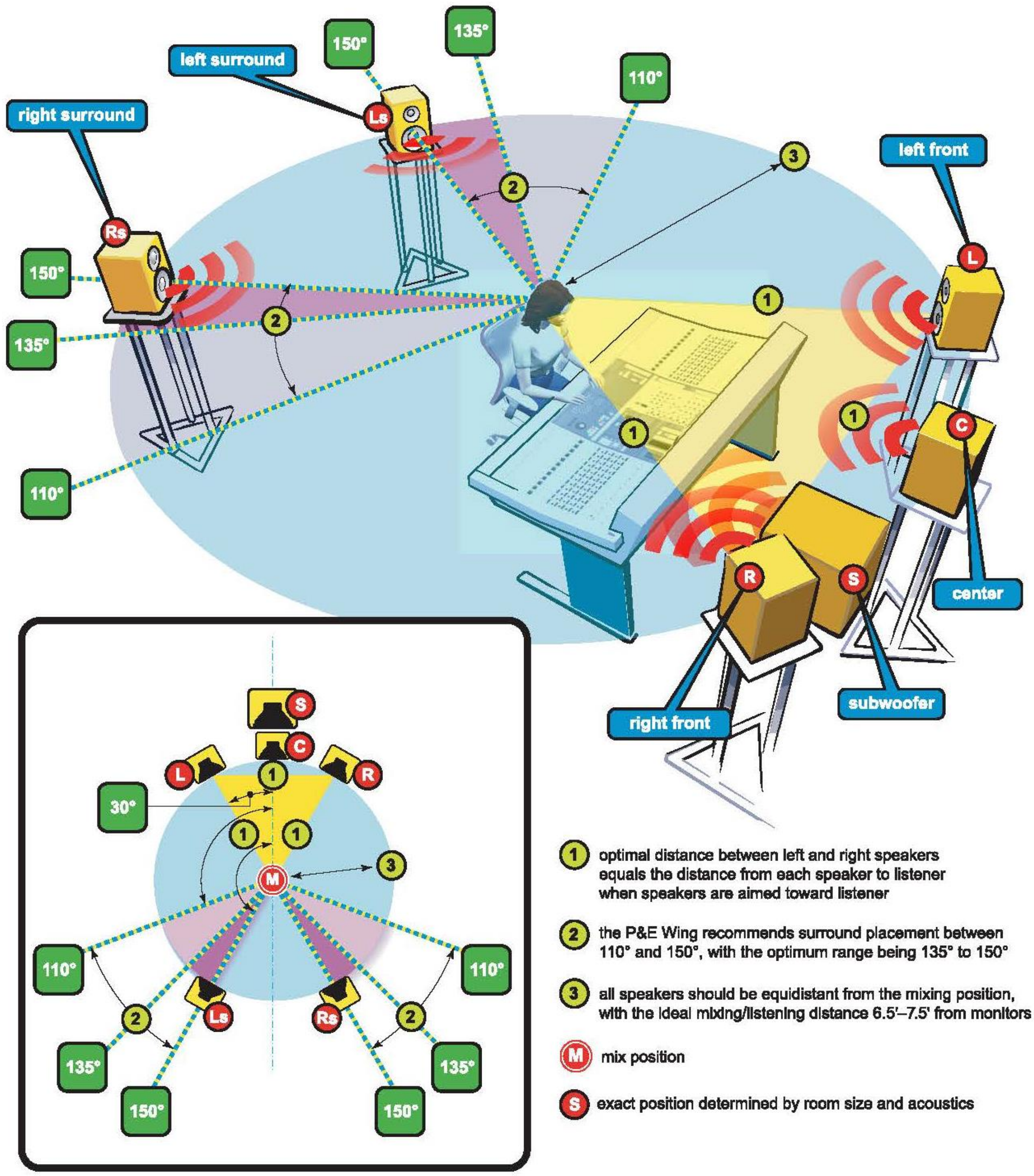
To get the best out of your control room when it comes to acoustics, you should consider the following advice and, if necessary, apply the changes before continuing:

- Be sure to follow the professional recommendations for stereo and surround loudspeaker installation. Standards such as the ITU-R-775-1, SMPTE 202M and so on, are what the Optimizer algorithms refer to regarding spatial positioning.
- The better the acoustic environment, the finer and more effective the Optimizer tuning. As a minimum, you should consider the room's form factor, its construction, and materials, adding acoustic panels or modular elements, and the placement of furniture and equipment. The help of a knowledgeable and experienced acoustician is highly recommended.
- You should avoid placing obstacles, such as computer screens, glass surfaces, large empty desks, etc. between the loudspeakers and the listening position.
- However, the listening position should be arranged according to the actual mixing conditions you will be working in. So, if DAW screens and other devices are required around you, they should be positioned before calibration.

The purpose of the Optimizer is to achieve the best response from your loudspeakers whatever the listening conditions are. Therefore, the more acceptable they are before calibration, the better the "post calibration" result.

The diagram on the next page shows the recommended positions and angles to apply in a professional recording/mixing room, according to the various standards used in the Optimizer.

Extract from the Recording Academy Recommendations



6.4.3.3 RECOMMENDATIONS FOR 3D-MICROPHONE PLACEMENT

The Optimizer algorithms support both single-point and multi-point calibrations.

In most cases, a single-point calibration will achieve the best possible result from the loudspeakers.

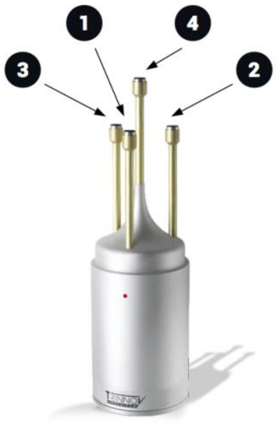
Multi-point calibration should be used if a larger sweet-spot is required (e.g., in front of a large-scale mixing desk); if there are simultaneous mixing positions; or when a room presents particularly complex acoustic issues. In this case, the help of a Trinnov-certified technician and an experienced acoustician can dramatically improve the methodology and the end result.

In both scenarios, a **reference measurement position** should be chosen carefully since it will be used by the Optimizer to calculate the loudspeakers' relative delay and level alignment as well as the Master delay and level. The "M" position is the best choice for presets to be used for mixing.

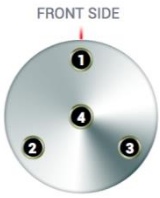
The calibration microphone should be placed at the listening position, using the listener's ears and capsule #4 as a reference for height. The **red LED** indicates the front of the 3D-Microphone which must be **pointed to the front and towards the center** of the sound stage. You may use the virtual line passing over capsules #1 and #4 to align it with the axis symmetry.

The 3D-Microphone must be perfectly stable and well settled, horizontally, on its mic stand.

Side View



View from Above



6.4.3.4 CONNECTING THE 3D-MICROPHONE

Once positioned, the 3D-Microphone must be connected to the D-Mon processor.

> Note that the connection is required during calibration only.
> It is NOT necessary to keep the 3D-Microphone connected to the processor after calibration.

The 3D-Microphone comes with a 4 x male XLR multi-cable (below left). This must connect to Analog Inputs 1-4, available via SubD-25 on the rear of the D-Mon processor. You can use the optional 4 x female XLR to SubD-25 breakout cable (below right) to access the inputs or refer to the SubD-25 wiring diagram in Appendix 10.1.1.

3D-Microphone with XLR multi-cable

Optional 3D-Mic Breakout Cable (XLR to SubD-25)

> Each microphone output (1-4) MUST connect to the same processor INPUT (1-4) since the Optimizer relies on the expected capsule positions to perform the 3D measurement. Therefore, using the optional 3D-Mic Breakout cable is recommended.

6.4.4 RUNNING THE FIRST CALIBRATION (STEP-BY-STEP)

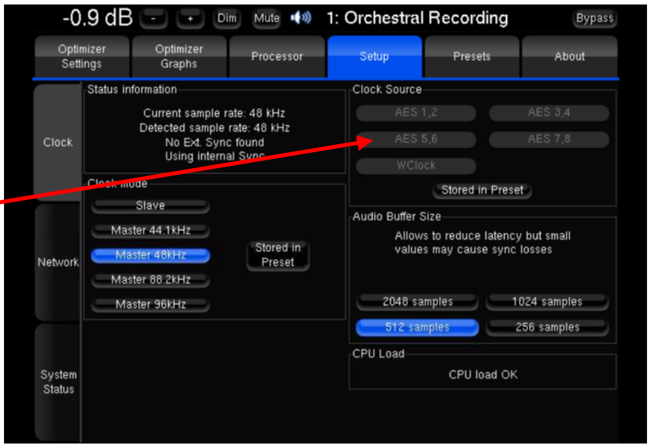
	SEVERE NOISE INJURY MAY OCCUR DURING THE CALIBRATION PROCESS IF THE FOLLOWING INSTRUCTIONS ARE NOT STRICTLY APPLIED. ANY PERSON PRESENT IN THE ROOM DURING CALIBRATION SHOULD BE WARNED ABOUT THE RISK OF NOISE OVERLOAD AND THE LARSEN EFFECT.

The Optimizer requires a minimum Sound Pressure Level for proper calibration. Until the minimum SPL is reached, the test signal will play through the same speaker. For safety reasons, we recommend running the first calibration as if it were a test. Start with a Master Level of -40 dB and increase the level until the test signal passes on to the second speaker. This confirms that the minimum SPL has been reached.

<table>
<tr><th>Step</th><th>Instructions</th><th>Comments</th></tr>
<tr><td>1</td><td>Open the OPCP GUI by clicking on the D-Mon processor model number from the DMCP.

See 6.4.1: Working with the D-Mon and Optimizer Control Panel.</td><td></td></tr>
<tr><td>2</td><td>Select the “Setup” page followed by the “Clock” sidebar tab:

✓ Set the sample rate to 48kHz, whether as the Master or Slave to an external clock.

✓ In the case of an external clock, select a valid Clock Source and make sure that a 48kHz sample rate is displayed on-screen to show that it has been accepted.</td><td>
</td></tr>
<tr><td>3</td><td>Select the “Processor” page:

✓ In the “Meters,” “Master” and “Outputs” sidebar tabs, set the output parameters to convenient values.

✓ Any parameter can be changed later and saved in a different preset.

✓ The default values ensure an overall gain of 0dB. Therefore, you should only change parameter values if necessary.</td><td>
</td></tr>
<tr><td>4</td><td>Select the “Optimizer Settings” page followed by the “Calibration” sidebar tab. From here:

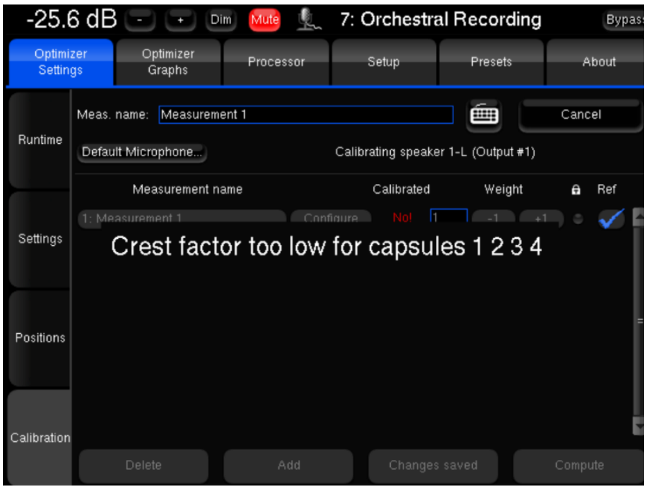
✓ Change the Measurement name if you wish (e.g., Measurement 1).

✓ Click on Configure to check and, if necessary, change the 3D-Microphone compensation file. See 6.4.2.2: The Trinnov 3D-Microphone.

✓ Adjust the calibration parameters.</td><td>
</td></tr>
</table>

5 Click on **Calibrate** to start the calibration procedure, and follow the on-screen instructions:

- ✓ First, Mute the system to avoid any trouble. It does not affect the calibration procedure.
- ✓ Power on the 3D-Microphone (its LED should be on) and press are **OK**.
- ✓ **The generator immediately sends out MLS noise bursts (three shots) to each of the loudspeakers.**
- ✓ Following a successful calibration, the “Calibrated” field updates to “Yes,” and you can proceed to step 7.
- ✓ If the calibration is unsuccessful, then a warning message will appear (an example is shown the opposite). Click on **Cancel** to stop the calibration, and see *Warning Messages & Troubleshooting* for more details.



6 To add more measurement points, for a multi-point calibration, click on the **Add** button and repeat steps 5 and 6:

- ✓ Use the value to the right of the "Calibrated" field to set the importance factor. A value of 0 ignores the point. (Default value is 1)
- ✓ Check the **Ref** field to set the reference. (The default reference measurement point is the #1)
- ✓ Click on **Delete** to delete a measurement point.

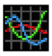
ATTENTION!

Switch off the 3D-Microphone after every calibration attempt!

7 Once all measurement points have been successfully calibrated, click on **Compute**. The Optimizer now calculates the correction filtering required.

Various icons appear in the top info bar to show the progress:

 = calculating filters.

 = calculating graphs.

 = calculations complete.

Note that it may take some time for the processor to reach the final step.

Optimization is now complete and you are ready to enjoy your new listening conditions!

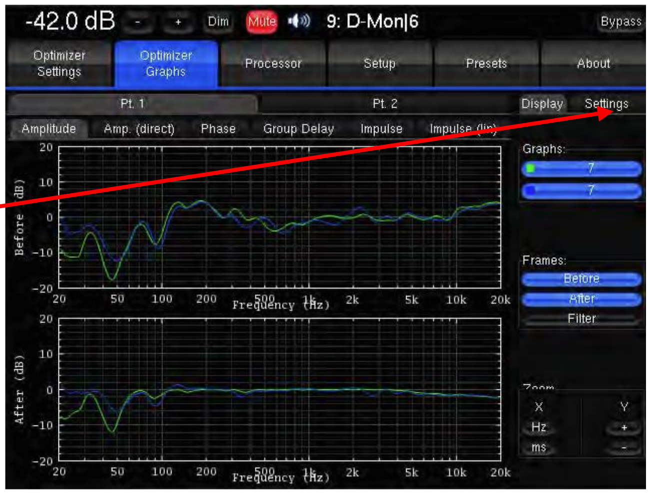
Calibration level may be loud. Do not forget to reduce the volume before using the unit. The processor is always muted after a calibration, so remember to unmute first!

We strongly recommend using the **Bypass** button (top right) to listen with and without the Optimizer. When making an A/B comparison, we recommend using professional sound checking material, if possible, rather than a normal mix.

The “Optimizer Graphs” page can be used to **view** the graphs while listening to the result at the same time.

- ✓ Use the tabs (**Amplitude**, **Phase**, etc.) to view the different responses.
- ✓ Click on **Settings** to rearrange how the graphs are displayed.

The “Optimizer Settings” page provides various options for fine-tuning the Optimizer parameters. See *6.4.5: Optimizer Settings* for details.



6.4.4.1 WARNING MESSAGES & TROUBLESHOOTING

During calibration, you may see a warning message indicating that there is a problem. The list below describes the most likely causes of each message and the steps which should be taken to provide a solution.

> **IMPORTANT**: if several digital “acoustic tuning” processors (external or integrated) are added to the loudspeaker audio chain, then it is highly likely you will achieve a bad overall result and see some warning messages from the D-Mon processor. Therefore, it is strongly recommended that only one processor be active at a time, and that you compare the results.

a) “Crest factor too low…”

This message appears if the processor cannot distinguish between the level of the background noise and the MLS noise coming from the loudspeakers.

Possible Causes	Solutions
The main level is too low. The direct sound is too weak, compared to the reverberation. There is too much background noise.	Turn up the volume, and avoid as much background noise as possible. Use the “Processor -> Meters” in the OPCP GUI to check the noise levels arriving at each microphone capsule (1-4).
There is an obstacle between the speaker and the microphone.	Remove the obstacle or reposition the speaker.
The system includes dipole and bipole speakers.	Reposition the 3D-Microphone so that it is closer to a single emitting axis.
The speaker, or something in the audio chain before the speaker, has some processing which is creating pre-ringing effects.	Select the “Optimizer Settings -> Settings -> Advanced Settings” in the OPCP GUI, and under Calibration Settings, raise the “Threshold for response begin to detect” value.

b) “Unstable Position for Speaker…”/“Unable to Localize Speaker…”

This message appears when the sound around the 3D-Microphone is too diffuse, and the system cannot find the emitting source.

Possible Causes	Solutions
Someone or something has moved during the calibration.	Do not move while performing a calibration. Do not let anything else move (e.g., doors, curtains, etc.).
Too many strong reflections around the 3D-Microphone are disturbing the impulse response measurement.	Remove the 3D-Microphone from these disturbances or attenuate the reflections.

There is an obstacle between the speaker and the microphone which is disturbing the measurement.	Remove the obstacle or reposition the speaker.
The bandwidth of one of the speakers is too narrow.	Make sure that any built-in filters (e.g., a subwoofer filter) are disabled.

c) “NOT READY.”

This message can appear in the top info bar if a problem has occurred during computation, or if the calibration has been corrupted for some reason.

In this instance, you should try running another calibration from scratch.

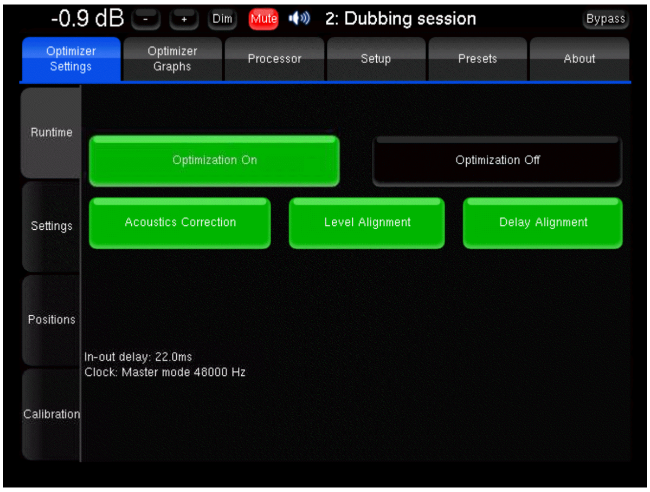
“Not ready” also means that the calibration has not been computed. In this case, please make sure you clicked on “Compute” after having calibrated the system.

6.4.5 OPTIMIZER SETTINGS

Having completed the first calibration, the “Optimizer Settings” page can be used for fine-tuning. The page has four sidebar tabs:

- “Runtime” – allows you to manage the overall Optimizer process.
- “Settings” – provides access to all normal and advanced parameters.
- “Positions” - displays how the speakers have been measured.
- “Calibration” – provides access to the calibration procedure (described earlier).

Optimizer Settings -> Runtime



From here you can enable or disable the complete Optimizer program or any of its parts:

- **Optimization ON/OFF** - allows the user to run or bypass all of the processing related to the “Optimizer Settings” page. This includes the acoustic correction, the automatic delay, and level alignment, as well as remapping options.

  Note that when **Optimization is OFF**, only processing defined in other pages is applied:
  - Routing – as defined in the DMCP.
  - Levels – as defined in the “Processor” page.
  - Graphic EQs – as defined in the “Processor” page.
  - Bass Management.
- **Acoustic Correction ON/OFF**: when turned OFF, both the automatic equalization (defined by the target curve) and the FIR EQ are bypassed.
- **Level Alignment ON/OFF**: the automatic alignment of speaker levels can be disabled, meaning that no automatic gain changes will be applied to the outputs.
- **Delay Alignment ON/OFF**: the automatic alignment of speaker distances can be disabled, meaning that no automatic delays will be applied to the outputs.

At the bottom, you will see the following information (for display purposes only):

- **In-out delay**: the latency applied by the algorithms. This varies according to the preset currently in use and how complex and effective the algorithms are.
- **Clock**: displays the current clock setting. (To adjust the clock setting, use the “Setup” page or DMCP).

Optimizer Settings -> Settings

This tab provides access to all Optimizer parameters. It is divided as follows:

- **Main Settings**: includes all the usual parameters you may wish to adjust (see next page).
- **Advanced Settings** includes more advanced parameters.
- **Target Curve**: allows the definition of target curves that the algorithms will respect.
- **Excursion Curve**: allows the definition of boost and cut values based upon frequency bandwidth.

The **Main Settings** are being described on the next page.

Advanced Settings are usually adjusted by a Trinnov-certified technician. Therefore, these are not described in this User Guide.

Optimizer Settings -> Settings (Continued)

The **Main Settings** are:

- **Optimize**:
  - **Amplitude + Phase** (default): with this setting, the Optimizer will improve both the amplitude and phase response of the loudspeaker. This greatly reduces the group delay of the speakers (starting from about 150Hz).
  - **Amplitude only**: in this mode, the Optimizer will work only on the amplitude response; the phase behavior is not modified.
  - **Low range only**: with this setting, the automatic equalization only uses IIR filters up to the frequency defined in the Advanced Settings (Default: 150Hz). The automatic FIR filter is disabled, but the FIR EQ can still be applied.
  - **According to L&R speakers**: this is a special mode that will optimize the center and surround speakers to achieve the same response as the Left and Right speakers. It is most useful in home cinema installations.
- **Maximum Boost/Attenuation** defines, in dB, the maximum amount of boost or attenuation applied by the algorithms. These parameters are used to protect against overloads. The default values are 6dB/10dB respectively. These parameters will dramatically impact upon the behavior of the automatic equalization, as they are applied to both the time-based and the energy processing. Accurate values can be defined by using the “Exclusion Curve” across the full frequency bandwidth.

Optimizer Settings -> Positions

This tab displays how the speakers have been measured. Click on **Summary** to view a comprehensive measurement table.

6.4.6 DEFINING THE AUDIO CLOCK

The audio clock source must be defined in the OPCP GUI using the “Setup” page and “Clock” tab.

Note that the clock settings and status can also be checked (but not adjusted) from the **D-Mon Control Panel** GUI, see *7.5.5 STUDIO SETUP -> Audio Clock*.

To change the audio clock source:

<table>
<tr><th>Step</th><th>Instructions</th><th>Comments</th></tr>
<tr><td>1</td><td>Open the OPCP GUI by clicking on the D-Mon processor model number from the DMCP.</td><td>See 6.4.1: Working with the D-Mon and Optimizer Control Panel.</td></tr>
<tr><td>2</td><td>Select the “Setup” page followed by the “Clock” sidebar tab.</td><td rowspan="2">

</td></tr>
<tr><td>3</td><td>To define an external clock source:
✓ Select Slave as the “Clock mode.”
✓ Then choose one of the “Clock Source” options – either an AES input or Wordclock.
The “Status information” updates and should show that a valid clock source is being received. If not, check both the clock source and its connections.
Alternatively, you can use the processor’s internal clock by selecting one of the master “Clock Mode” options (e.g., Master 48kHz).</td></tr>
<tr><td>4</td><td>✓ Select Store in Preset if you wish the “Clock mode” and “Clock source” to be saved and recalled by presets.
✓ Use the “Audio Buffer Size” options to adjust the latency of the processor. Note that too small a latency can result in syncing issues.</td><td></td></tr>
</table>

6.4.7 OTHER STUDIO-RELATED PARAMETERS

At this stage, it is a good idea to check the rest of the **STUDIO SETUP** parameters in the **D-Mon Control Panel** GUI so that when you save a preset, it will include all the settings required to reset the D-Mon processor for the studio. In particular, check the following tabs:

- **Inputs & Outputs** – defines the talkback and listen-back inputs, plus the AES insert.
- **Options/Levels** – sets automatic dimming on talkback, plus the dim and reference levels for the speaker volume.
- **Remotes** – GPIO mappings for external controllers and the optional USB Volume controller.

See *7.5: STUDIO SETUP* for details.

6.4.8 SAVING A PRESET (OR PRESETS)

To complete the installation, you will need to save a preset so that all settings can be recalled later.

Note that a preset store's everything required to reset the D-Mon processor – i.e., all **D-Mon Control Panel** pages (including the **STUDIO SETUP**), plus the **Optimizer & Processor Control Panel** settings (including the calibration and clock parameters).

For full details on how to store and recall presets, see *7.6: CONFIGURATION PRESETS*.

Note that if you wish to perform several calibrations, to re-configure the Optimizer for different monitoring arrangements, then you will need to save a preset for each different setup.

> It is also strongly recommended that you create and update a preset throughout the configuration process. This will ensure that you do not lose any settings should the processor reboot.
>
> Note that there are a few parameters which are NOT saved in either presets or snapshots. These settings are always absolute and remain at their last values. They are the:
>
> - **Speaker listening level**, **MUTE,** and **DIM** set in *The CONTROLLER* page of the DMCP GUI.
> - **Network** parameters defined in the OPCP GUI, see *6.2.2: Setting up the CPU*.
> - **Audio clock** parameters (if “Store in preset” is set to off in the OPCP GUI, see *6.4.6: Defining the Audio Clock*).

7 THE D-MON CONTROL PANEL

This chapter covers all the functions available from the **D-Mon Control Panel**.

Please read *Chapter* 5*: Quick Start* for a short introduction to the GUI and its operating principles.

7.1 THE CONTROLLER PAGE

The **CONTROLLER** page is the main operational page for everyday tasks. It is divided into five main areas:

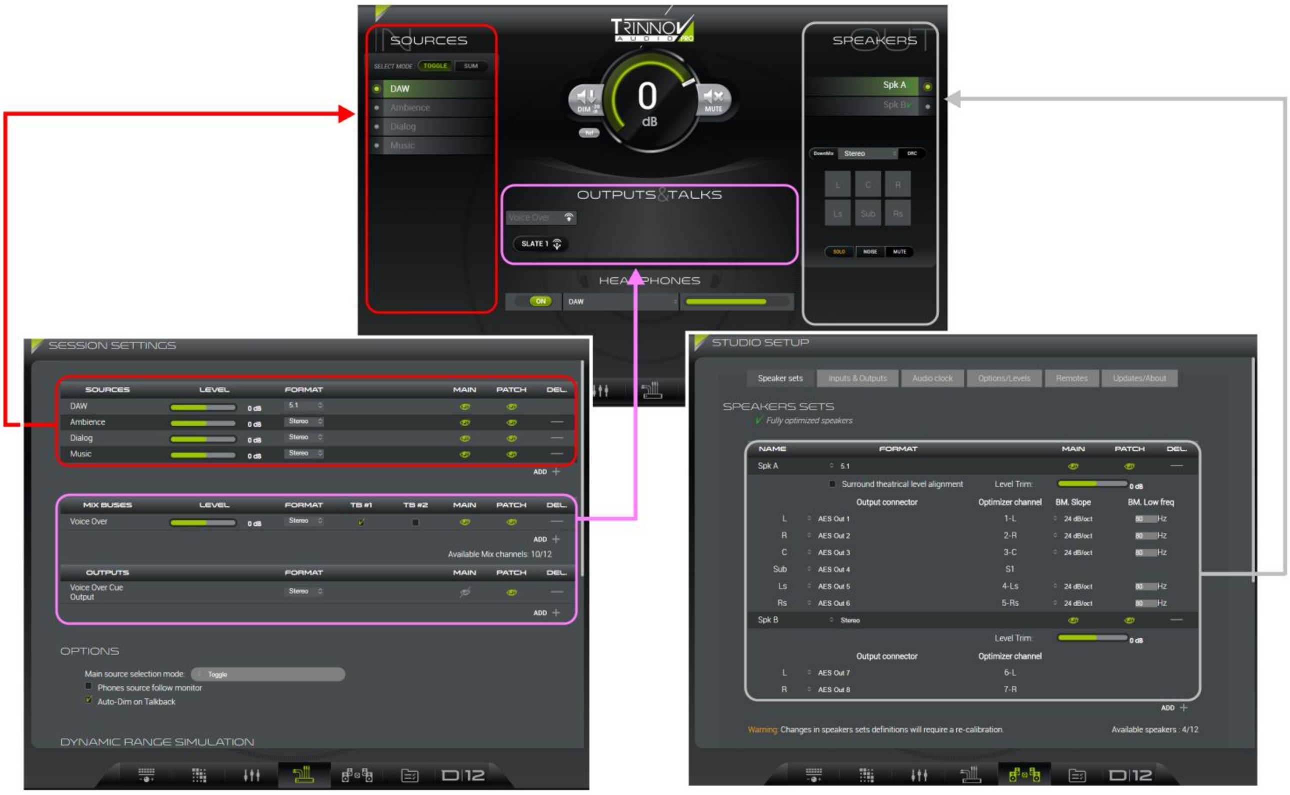
	Area	Function
#1	**SOURCES**	Manages all incoming sources defined in the *SESSION SETTINGS*.
#2	**Speaker Level**	Adjusts the listening level for the speakers (applies to all speakers within all speaker sets).
#3	**SPEAKERS**	Manages all speaker sets defined in the *STUDIO SETUP -> Speaker Sets*.
#4	**OUTPUTS & TALKS**	Manages all mix buses and outputs defined in the *SESSION SETTINGS*.
#5	**HEADPHONES**	Controls the stereo headphone output.

Our example above shows a D-Mon processor configured for an orchestral recording session. On the left, the sources (#1) have been given session-specific names, and on the right, there are two optimized speaker sets (#3) defined for the installation: **7.1.4** and **Stereo**. In the middle, two mix outputs (#4) are also defined, for the Conductor and Musicians; the Conductor's mix has talkback available, and both listen-back inputs are also enabled. The operator is currently listening to the **Musician Mix** on the **7.1.4** speaker set and the incoming **DAW** mix on the Headphones. Both the speaker and headphone levels (#2, #5) are open.

> If you have recalled a different preset, or are using a different D-Mon processor, then you will see different options within the SOURCES (#1), SPEAKERS (#3) and OUTPUTS & TALKS (#4) areas.

7.1.1 PREPARING THE CONTROLLER PAGE

The diagram below shows how the **CONTROLLER** page is affected by the **SESSION SETTINGS** and **STUDIO SETUP** parameters:



The available SOURCES and OUTPUTS & TALKS are determined by the *SESSION SETTINGS,* and the available SPEAKERS by the *STUDIO SETUP -> Speaker Sets*. Therefore, you will need to configure both of these pages before operating the system.

Note that:

- The **MAIN** “eye” icon (in the **STUDIO SETUP -> Speaker Sets**) must be enabled to view a speaker set in the SPEAKERS list.
- The **MAIN** “eye” icon (in the **SESSION SETTINGS**) must be enabled to view a source in the SOURCES list or mix bus or output in the OUTPUTS & TALKS area.

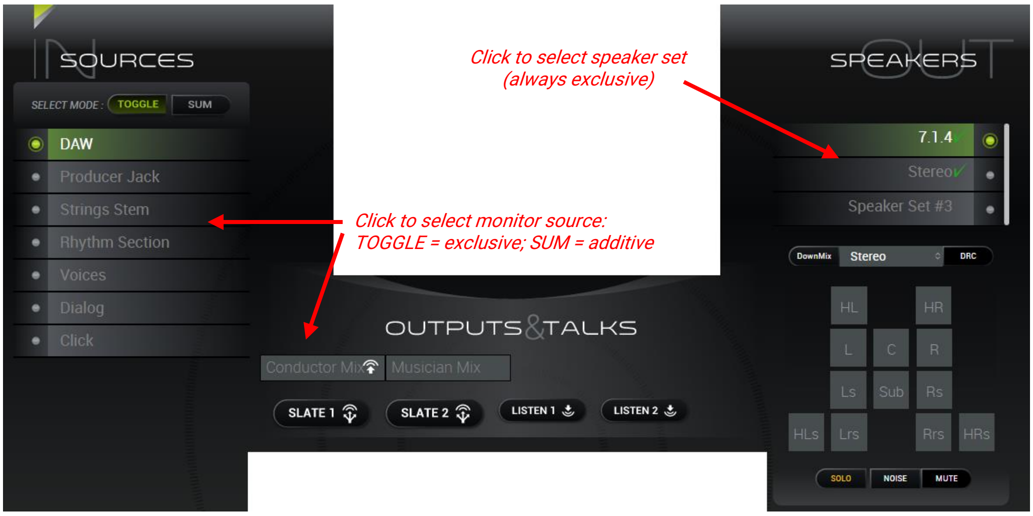
Once both the studio and session parameters have been defined, the system is ready for operation from the **CONTROLLER** page.

7.1.2 SAVING SETTINGS (IN SNAPSHOTS & PRESETS)

Nearly all selections made within the **CONTROLLER** page are saved in both snapshots and presets - for example, the current monitor source, selected speaker set, and headphone on/off, source and level.

Exceptions are the speaker listening level, **MUTE,** and **DIM** which are absolute and remain at their last value.

7.1.3 MONITORING AUDIO (ON THE SPEAKERS)



- From either the SOURCES or OUTPUTS & TALKS areas, click to choose a monitor source - the selected source or output (e.g., **DAW**) feeds the selected speaker set (e.g., **7.1.4**).

The available sources and outputs are defined in the *SESSION SETTINGS* page and are stored and recalled by snapshots. You can listen to any combination of SOURCES and OUTPUTS – use the **TOGGLE** and **SUM** modes to choose whether selections are exclusive or additive. When selecting an output, take care to click on the button name and NOT the talkback symbol!

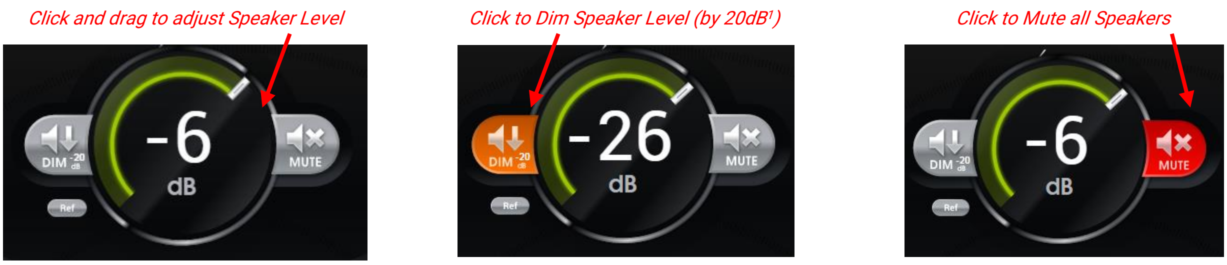
Note that the **LISTEN** buttons (see later) and **SPEAKERS** buttons in the *MONITORING MIXER* also change the monitor source.

- From the SPEAKERS area, click to change the speaker set.

The available speaker sets are defined under *STUDIO SETUP -> Speaker Sets*, and are stored and recalled by presets. If more than one speaker set is defined, then the switching is exclusive. A green "V" (from the Trinnov logo) appears beside each optimized speaker set; if there is no green "V," then the speaker set is non-optimized. As you switch between speaker sets, the individual loudspeaker buttons (**HL**, **HR**, **L**, **C**, **R**, etc.) update to reflect the current format.

Note that a monitor source and speaker set are always selected. Therefore, if you wish to stop audio feeding the speakers, use the **MUTE** button (described below).

7.1.4 ADJUSTING THE LISTENING LEVEL (SPEAKER VOLUME)



- Click and drag on either the dB value or slider handle to increase or decrease the listening level.

If using a mouse, it is easier to drag left/right or up/down (rather than in a circular motion). If a maximum speaker level[1] has been defined, then you cannot exceed this level. The speaker level can be displayed as an unreferenced dB value (as shown above) or using one of the referenced level options defined under *STUDIO SETUP -> Options/Levels*.

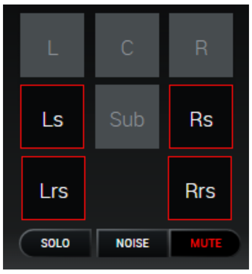
- Click on **DIM** to reduce the listening level[1] – the button lights in orange when enabled. Click again to cancel.
- Click on **MUTE** to cut all speakers within the speaker set – the button lights in red when enabled. Click again to cancel.
- Click on **Ref** to reset the listening level to the reference value[1] – this is a one-shot operation.

[1] The amount of dim, plus the reference and maximum speaker levels are defined under *STUDIO SETUP -> Options/Levels*.

Note that different level trims can be applied to individual speaker sets under *STUDIO SETUP -> Speaker Sets*.

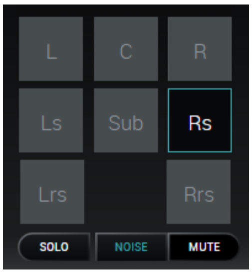
7.1.5 CHECKING INDIVIDUAL LOUDSPEAKERS

Speakers Muted



Centre Speaker in Solo

Pink Noise to Right Surround



Using the buttons at the bottom of the SPEAKERS area, you can solo, deliver pink noise to, or mute an individual speaker:

- First, choose the function – e.g., **MUTE**.
- Then, select a speaker or speakers (e.g., **Ls**, **Rs**, **Lrs**, **Rrs**).

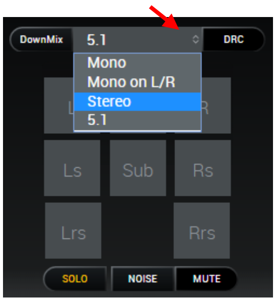
Each of the functions is color-coded to indicate when a speaker is muted (red), in solo (yellow) or sending out pink noise (turquoise).

- Cancel the operation either by de-selecting the speaker(s) or switching to a different mode – e.g., from **MUTE** to **SOLO**.

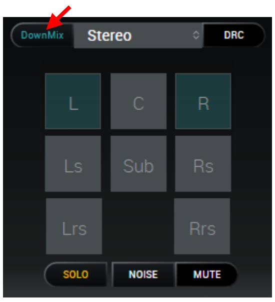
> It is possible to set up if the solo/mute/noise functions are applied at the input stage or the output stage of the optimizer process.
> This parameter is defined in the **SESSION SETTINGS** page. See 7.4.3 SESSION SETTINGS -> Options

7.1.6 LISTENING TO A DOWNMIX

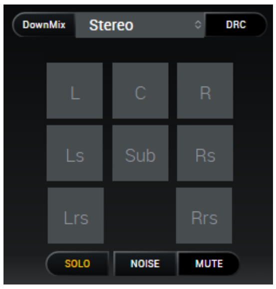
Click to select format



Click to enable DownMix



DownMix Off (normal monitoring)



You can use the **DownMix** button in the SPEAKERS area to check the listening compatibility for different speaker arrangements. For example, to check how a 7.1 mix will sound in 5.1, Stereo or Mono.

- First, select the source you wish to check from the SOURCES or OUTPUTS areas; and select the right speaker set (a 7.1 in our case)
- Then, choose a downmix format from the drop-down menu (e.g., **Stereo**).
- Enable the **DownMix** button – the active speakers are highlighted while the downmix is enabled (e.g., **L** and **R**).
- Click again on the **DownMix** button to cancel the function and return to "normal" monitoring (e.g., **7.1**).

Note that the available downmixes are factory-configured and determined by the selected speaker set. So, for example, if you are working with a stereo speaker set, there will be only one downmix available: **Mono on L/R**. If a surround speaker set is selected, there will be multiple downmix options.

When choosing to listen to a mono downmix on a surround speaker set, you can choose either:

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

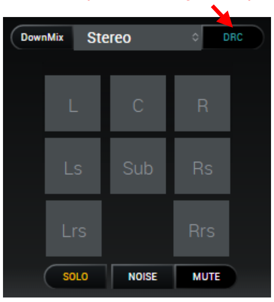
While listening to a Downmix, you can use the **SOLO** or **MUTE** functions to interrogate individual speakers.

Note that downmixing occurs automatically whenever there is a mismatch between a source and a destination – for example if you listen to a 5.1 source on a stereo speaker set. Also, you can force a downmix to occur by changing the routing assignments in the SESSION ROUTING page, see *7.2.5: Downmixing.*

In all cases, the downmix rules (coefficients) and the list of the available targets to be presented to the user in the DMCP are defined in the OPCP GUI, see *6.4.1: Working with the D-Mon and Optimizer* Control Panel.

7.1.7 DYNAMIC RANGE COMPRESSION (DRC)

Click to enable Dynamic Range Compression



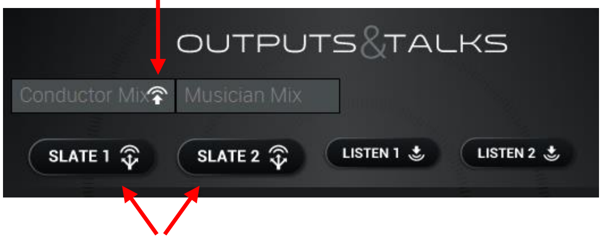
The **DRC** button in the SPEAKERS area can be used to simulate how your mix will sound when played through a consumer system (where dialogue normalization is applied).

The simulation complies with the SMPTE 85 standard used in Dolby encoding, and its parameters are defined in the **SESSION SETTINGS** page. See *7.4.4 SESSION SETTINGS -> Dynamic Range Simulation*.

- Click on the **DRC** button to enable or disable the simulation.

7.1.8 TALKING TO AN OUTPUT

Talk to Conductor



Talk to ALL

If talkback to a mix bus is enabled in the *SESSION SETTINGS* page then, in the OUTPUTS & TALKS area you will see the relevant master **SLATE** button(s) plus a talkback symbol (on the right of the button name):

- Click on the talkback symbol to open a communication line between the talkback input and the selected destination.

Take care to click on the talkback symbol and not the button name; selecting the button name will change the monitor source!

- Click again to close the talkback line. Note that several talkback lines can be open at the same time.
- Click on the **SLATE 1** or **SLATE 2** buttons to talk to multiple destinations.

SLATE 1 talks to ALL destinations assigned to talkback input 1, and **SLATE 2** to ALL destinations assigned to talkback input 2. The **SLATE** buttons override any individual talkback selections. Note that individual talkback is available for talkback input 1 only.

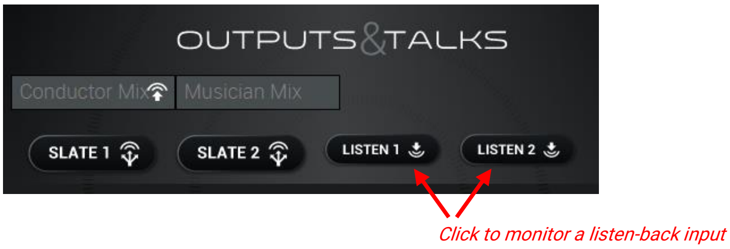
If the **Auto-Dim on Talkback** option is enabled in the *SESSION SETTINGS -> Options*, then the speaker level is dimmed whenever talkback is active.

7.1.9 LISTENING "IN" TO THE STUDIO

About the Talkback Inputs

The system supports two talkback inputs which can be used to talk to an individual mix bus or all buses (via the **SLATE** button). For each mix, you can determine whether talkback is enabled and which input will be used (TB1 or TB2), see *5.9 Editing the Session Settings*. These assignments are stored and recalled by snapshots, allowing you to store and recall different talkback configurations for each session.

The physical inputs and their level trims are specified under STUDIO SETUP -> Inputs & Outputs, and are stored and recalled by presets. They can use any analog or AES input, connected via the MPIO or SubD-25 connector. Analog input 1 is reserved for talkback and is named accordingly within the software. If TB2 is assigned to a different physical input, then you will be able to talk from different positions (e.g., from a sound engineer or producer).



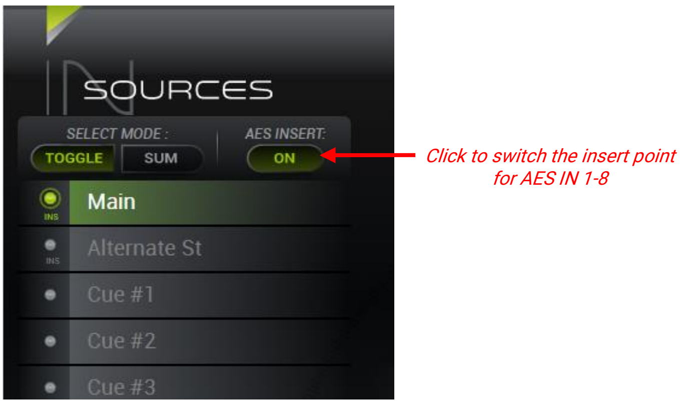
Listen-back can be used to return talkback from a particular studio location – for example, to "listen back" to the conductor's microphone during a recording session. If the listen-back inputs are enabled in STUDIO SETUP -> Inputs & Outputs, then the relevant **LISTEN** buttons appear in the OUTPUTS & TALKS area:

- Click on **LISTEN 1** or **LISTEN 2** to monitor the listen-back input on the selected speaker set.
- Click again to cancel the listen-back.

About the Listen-back Inputs

The system supports two listen-back inputs which you can listen to on either the selected speaker set (via the **LISTEN** buttons in the OUTPUTS & TALKS area) or on the Headphones (via the Headphones monitor source selector).
The physical inputs and their level trims are specified under STUDIO SETUP -> Inputs & Outputs, and are stored and recalled by presets. They can use any analog or AES input, connected via the MPIO or SubD-25 connector. Analog input 2 is reserved for listen-back and is named accordingly within the software.

7.1.10 SWITCHING THE AES 1-8 INSERT



If the "AES 1-8 Insert" option is enabled in the STUDIO SETUP -> Inputs & Outputs, then you will see an AES INSERT **ON**/OFF button and an **INS** indicator beside some sources – in our example, beside **Main** and **Alternate St**.

- Click on the AES INSERT **ON**/OFF button to switch the insert return in and out of the circuit. For example, to compare your “dry” mix (INSERT OFF) with a “wet” mix (INSERT ON) returning from a digital effects processor.

Note that:

- The insert switching affects only digital sources connected from AES In 1-8. In our example, this is why we see the **INS** indicator beside the source named **Main** (connected from AES In 1-6) and the source named **Alternate St** (connected from AES in 7-8). There is no insert point available for any other sources.
- When the INSERT is switched **ON**, all eight returns are switched together (i.e., the insert is switched on for both our **Main** and **Alternate St** sources). It is not possible to switch less than eight signals.

For more details on how to connect the AES signals, please see *6.3.1: Audio Connectors*.

> If the D-MON clock source is external and referenced to AES, when engaged, the INSERT also switches the clock source to the insert path. If no AES signal is present at the AES connector, the D-MON is no longer clocked.

7.1.11 MONITORING AUDIO (ON THE HEADPHONES)

Analog outputs 9 & 10 are reserved for the stereo headphones (see note below).

- Click on the **ON**/OFF button to activate or deactivate the headphone output.
- Use the drop-down menu to select a monitor source – e.g., **Cue #4**.

You can choose any of the sources, mixes or outputs defined in the *SESSION SETTINGS page,* or a listen-back input if listen-backs are enabled.

Note that the **PHONES** buttons in the *MONITORING MIXER* also change the headphone monitor source.

If the **Phones source follow monitor** option is enabled in the *SESSION SETTINGS -> Options*, then the headphones will automatically follow the speaker monitor source. In this instance, you can still make an independent selection, but as soon as the next speaker source selection occurs, the headphone monitor source will reset.

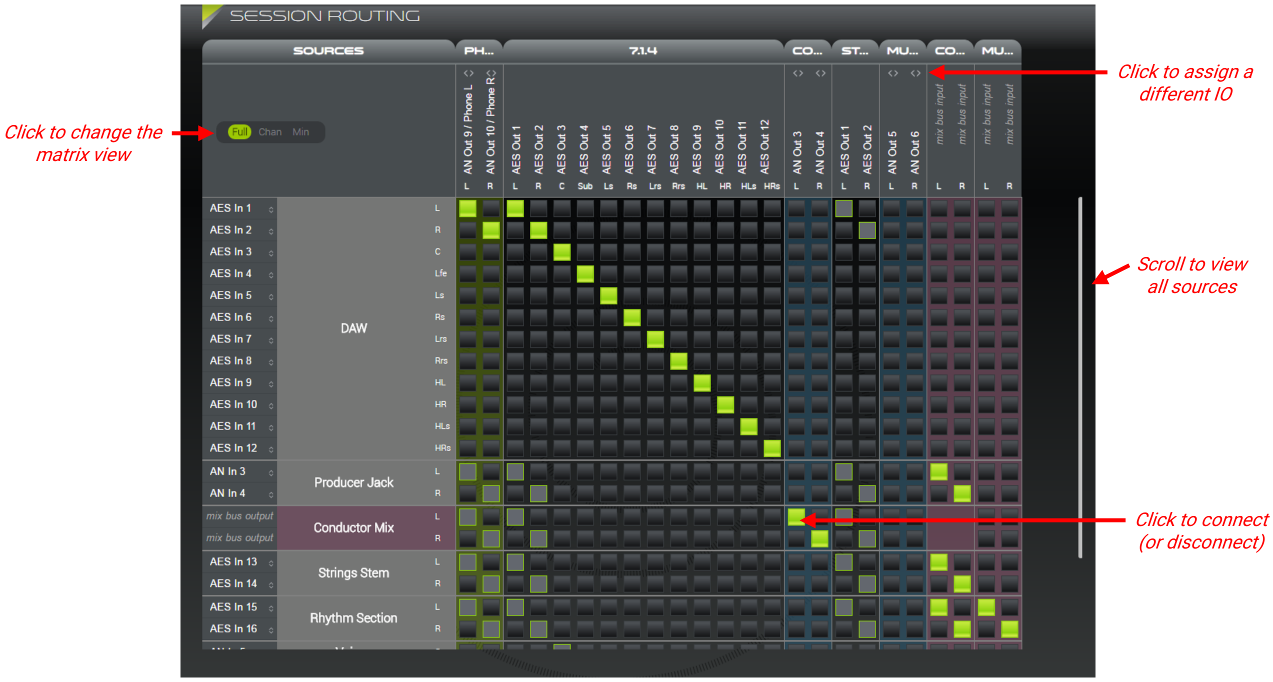
- Click and drag on the horizontal slider to increase or decrease the headphone level. If a maximum headphone level has been defined (under *STUDIO SETUP -> Options/Levels*), then you cannot exceed this level.

> If you hover over the slider, then a text read-out of the level is displayed.

> **About the Headphone Output**
>
> Analog outputs 9 & 10 are reserved for the stereo headphones and are named accordingly within the software. They are also hard-wired to the MPIO (and ICON console) headphone connector.
> Different outputs can be used by changing the "Phones" routing assignments in the *SESSION ROUTING* matrix. However, if the headphone output is moved away from analog outputs 9 & 10, then an external headphone amplifier will be required, and the headphones will cease to appear on the MPIO (or ICON) connector.

7.2 SESSION ROUTING

This page controls the routing matrix. It can be used to view all sources and destinations to the system; make manual routing assignments for the session – for example, to route a mix bus to its output destination; and change IO assignments.



Interrogating the Matrix

All available sources are listed on the left of the matrix and destinations across the top.

In **Full** view (shown above), you will see an X/Y crosspoint for every source and destination channel. The physical inputs and outputs assigned are also shown. If you click on the arrow beside/above the current assignment, then a drop-down menu appears where you can select a different IO.

Scrollbars will appear at the bottom and on the right of the matrix if needed - scroll left/right or up/down to see all available sources and destinations. Alternatively, you can enlarge the browser window or change the matrix view (see *Matrix Views*).

The sources and destinations are color-coded to help quickly identify the different types of signal:

- Inputs = light grey (e.g. **DAW**).
- Mix buses = purple (e.g. **Conductor Mix**).
- Outputs = blue (e.g. **Conductor Cue Output**).
- Speaker sets = black (e.g. **7.1.4**)
- Headphones = green (e.g. **Phones**)

Each crosspoint can be:

- Fully lit (GREEN) = crosspoint is active.
- Unlit (GREY) with GREEN OUTLINE = crosspoint is inactive but could be enabled by either the speaker or headphone monitor source selectors. This indicates that manual patching is not permitted to these destinations.
- Unlit (DARK GREY) with NO OUTLINE = crosspoint is inactive and available for manual patching.

Making Connections

➢ Click on a crosspoint to connect (or disconnect) the source and destination channel.

Note that, if you attempt to override an automated crosspoint (controlled by the monitor source selectors), then the connection will fail and be instantly reset to its automated status.

> In **Full** and **Chan** view, you can drag and drop to make consecutive assignments (see *7.2.4: Making Multiple Connections*).

7.2.1 PREPARING THE MATRIX

The diagram below shows how the matrix sources and destinations are affected by the **SESSION SETTINGS** and **STUDIO SETUP**:

The available sources are determined by the SOURCES in the *SESSION SETTINGS* page. The available destinations are determined by both the *SESSION SETTINGS* page (MIX BUSES and OUTPUTS) and *STUDIO SETUP -> Speaker Sets* (SPEAKERS). Therefore, you will need to configure both of these pages before setting up the matrix.

Note that:

- The headphones (**PHONES**) always appear as a destination.
- The **PATCH** "eye" icon (in the **STUDIO SETUP -> Speaker Sets**) must be enabled to view a speaker set in the matrix.
- The **PATCH** "eye" icon (in the **SESSION SETTINGS**) must be enabled to view a source, mix bus or output in the matrix.
- A mix bus will appear both as a source and a destination. This allows you to assign sources to a mix bus, and the same mix bus to its physical destination.
- If you assign a source to a mix bus in the **SESSION ROUTING** matrix, then this is reflected in the **MONITORING MIXER** page, and vice versa. See *7.3: MONITORING MIXER.*

7.2.2 SAVING SETTINGS (IN SNAPSHOTS & PRESETS)

All assignments made within the **SESSION ROUTING** matrix are saved in both snapshots and presets.

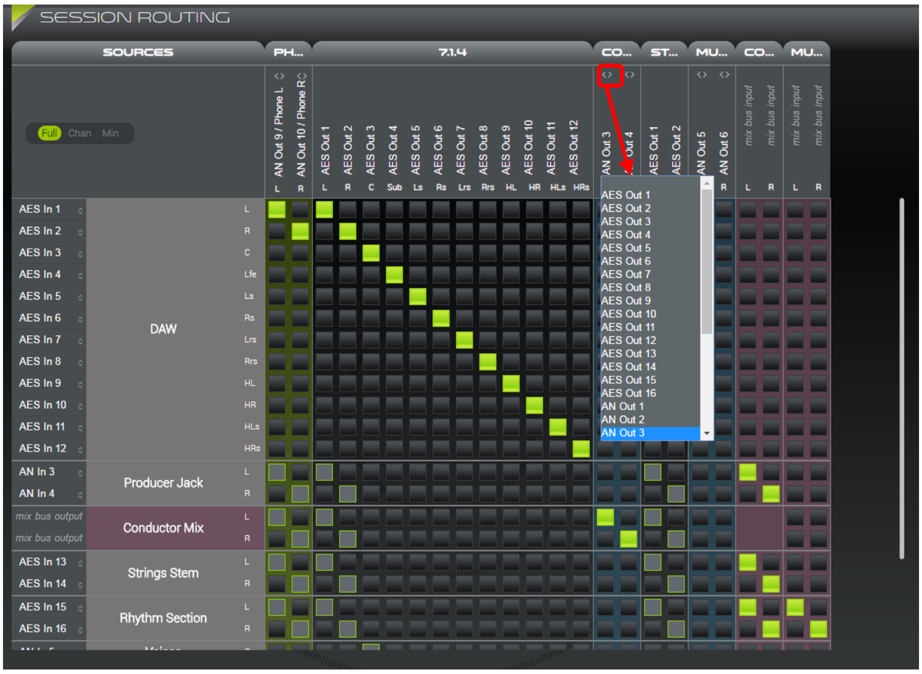
7.2.3 MATRIX VIEWS

The matrix can be displayed in one of three “views” – use the buttons at the top left to change between **Full**, **Chan** (channel) and **Min** (minimized) view. Once the matrix is configured, you can use the **Chan** and **Min** views to simplify its operation.

Full View

In **Full** view, you will see an X/Y crosspoint for every source and destination channel and its assigned input or output.

In addition to making connections, you can click on the arrow beside or above the I/O name to choose a different input or output. Any changes made are reflected in the **SESSION SETTINGS** page, and vice versa. In our example, we have assigned the left channel of the Conductor’s Cue Output to analog output 3:



> The I/O assignments of all sources and destinations can be adjusted except for the speaker sets (as their I/O assignments are part of the **STUDIO SETUP** rather than the **SESSION SETTINGS**). See *STUDIO SETUP -> Speaker Sets*.
> Analog outputs 9 & 10 are being reserved for the stereo headphones (**PHONES**). Different outputs can be used, but you will need an external headphone amplifier, and the headphones will cease to appear on the MPIO connector (as they are hard-wired to analog out 9 & 10).
> You can disable the **PHONES** output by selecting the first (blank) entry in the drop-down IO selector.

Channel View

In **Channel** view, the physical I/O assignments are hidden so that you see only the source and destination channels, but not their physical connections. This can be used to simplify the matrix:

Minimized View

In **Minimized** view, two things happen:

- Any destinations controlled by automated switching are hidden – i.e., all speaker sets and the headphones.
- For the remaining sources and destinations, the individual channels (Left, Right, etc.) are consolidated. This means that you will see a single row and column for each source and destination, regardless of their mono, stereo or surround format.

The result is a very simple view, where it is easy to see exactly which sources and destinations are connected:

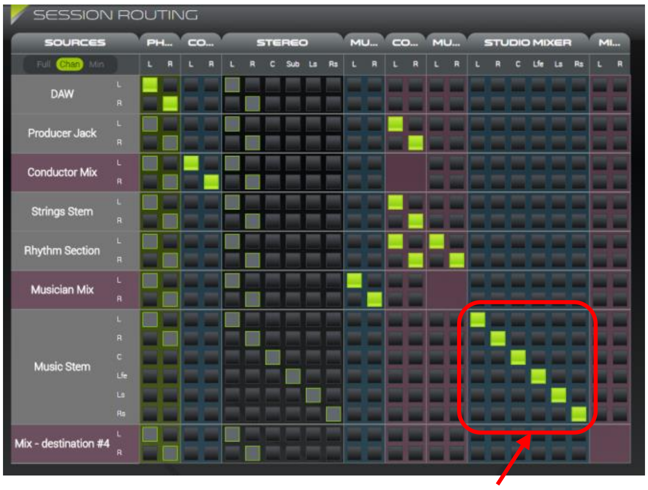
➢ Click on a crosspoint to connect (or disconnect) a source and destination.

If the source or destination is multi-channel, then the system connects the “hidden” channels consecutively, so source channel 1 to destination channel 1, source channel 2 to destination channel 2, and so on. This is ideal if the formats of the source and destination match as Left is assigned to Left, Right to Right, and so on.

However, if you wish to make a non-standard assignment (e.g., **Conductor Mix Left** to both **Conductor Cue Output Left** and **Right**), then this cannot be done in **Minimized** view. Instead, switch back to either **Channel** or **Full** view to assign the individual channels.

Note that if there is a mismatch between the number of sources and destination channels, then the connections wrap around. So, if you assign say a stereo source to a 4-channel destination, then source channel 1 is assigned to destination channels 1 & 3, and source channel 2 to destination channels 2 & 4.

7.2.4 MAKING MULTIPLE CONNECTIONS



Click here and drag diagonally to make selections

Release to connect (or disconnect)

In addition to making connections in the usual manner, you can “drag and drop” to make consecutive channel assignments. This method only works in **Full** or **Channel** view and is achieved as follows:

- First, switch to either **Full** or **Channel** view.
- Click at the top left corner of the first crosspoint and, while holding down your mouse button, drag diagonally across the matrix – as you click and drag, the crosspoints turn dark green to show that they are selected but not yet connected.
- Keep dragging until all the channels you wish to connect are selected – in our example, all six source channels.
- Then release your mouse button – the selected connections (or disconnections) are made.

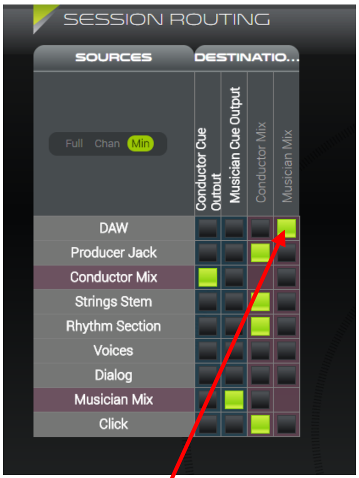
In our example, we have used this method to connect all six channels of our 5.1 music stem to the 5.1 studio mixer.

Note that:

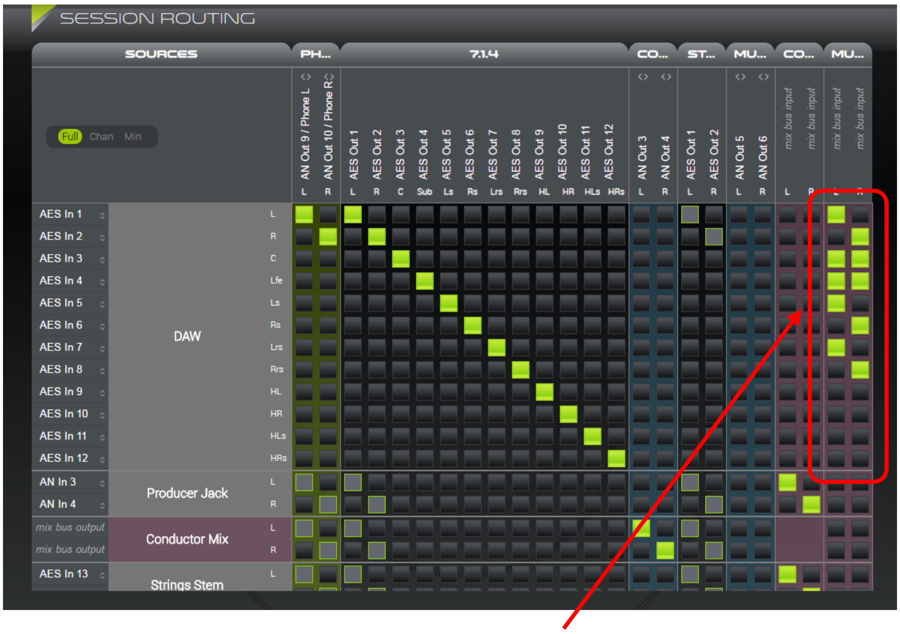
- You can use the same method to make multiple connections or disconnections.
- You can only select matching channel crosspoints – i.e., Left to Left, Right to Right, etc. (as shown above). If you wish to make a mismatched connection (e.g., source channel R to destination channel L), then you will need to click individually on each crosspoint.

7.2.5 DOWNMIXING

Downmixing occurs automatically whenever there is a mismatch between a source and a destination – for example if you assign a surround source to a stereo destination in **Min** view, and then switch back to **Full** view, you will see that an automatic routing has been applied. Wherever there is a summation (e.g., Centre channel to channels Left + Right), downmixing will occur.



12-channel source (DAW) assigned to stereo destination (Musician Mix)



Automatic routing & downmix coefficients applied

The downmix coefficients are identical to those used when downmixing from the SPEAKERS area on the CONTROLLER page (see *7.1.6: Listening to a DownMix*).

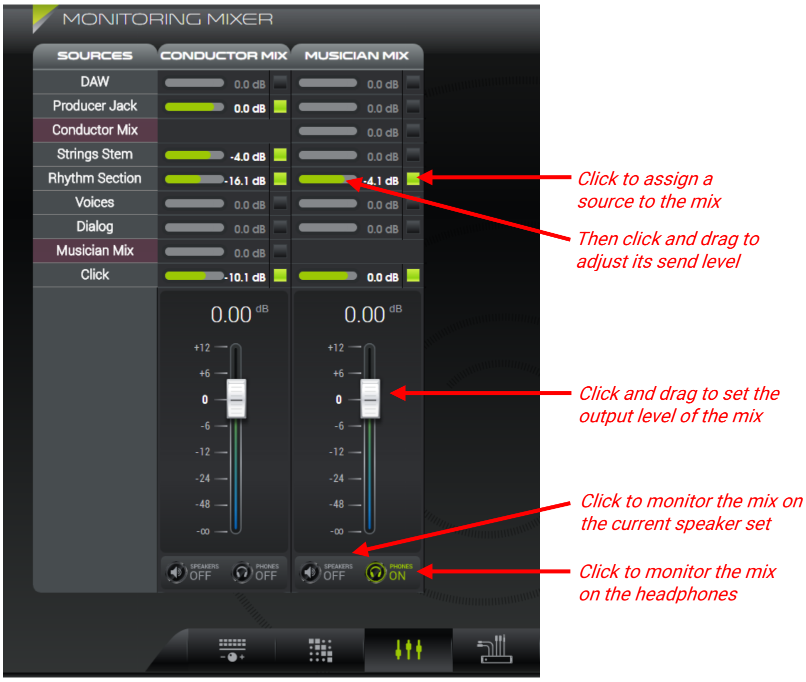
Downmixing will also be applied to summations forced manually in **Full** view –for example, if you click on individual crosspoints to create a manual fold down.

In all cases, the downmix coefficients are defined in the OPCP GUI, see *6.4.1: Working with the D-Mon and Optimizer* Control Panel.

7.3 MONITORING MIXER

This page controls the summing of the session's sources onto mix buses. It can be used, for example, to adjust a cue mix.

For each mix, you can set the individual summing points and levels, and the overall mix level. You can also choose to monitor a mix on the speakers or headphones using the buttons below each fader:



Using the Mixer Controls

In our example above, we have two cue mixes (**CONDUCTOR MIX** and **MUSICIAN MIX**) and a range of SOURCES. Note that each mix bus is also available as a source to allow mixes to be assigned to other mixes (mixes appear in purple in the SOURCES list).

Scroll bars will appear at the bottom and on the right of the mixer if needed - scroll left/right or up/down to see all available sources and mixes, or enlarge the browser window to fit the resources.

- Click on the square summing points to assign sources to each mix – the summing points light in green when active.

Any number of sources can be assigned. In our example, the **CONDUCTOR MIX** is receiving Producer Jack + Strings Stem + Rhythm Section + Click, while the **MUSICIAN MIX** is receiving Rhythm Section + Click.

- Click and drag on a horizontal slider to adjust the send levels to the mix – the sliders light in green once the level is open; the text readout displays the current level. To set a specific level, double-click on the text field, type in a value and press Enter. Note that you can only adjust a send level once the summing point is active.

You can use the send levels to control how the same source feeds different mixes. For example, the Click is feeding the CONDUCTOR MIX at -**10.1 dB**, while the MUSICIAN MIX is receiving Click at a much higher level (**0.0 dB**).

- Click and drag on a fader to adjust the output level of the mix – the large white text readout displays the current level. As above, you can double-click on the text field to enter a specific value.

In our example, the mixes are being used as cue feeds, and so you would use the faders to adjust the overall listening levels.

Monitoring a Mix

For convenience, the buttons below the faders allow you to quickly monitor any mix:

- Click on a **SPEAKERS** button to listen to the mix on the current speaker set.
- Click on a **PHONES** button to listen to the mix on the headphones.

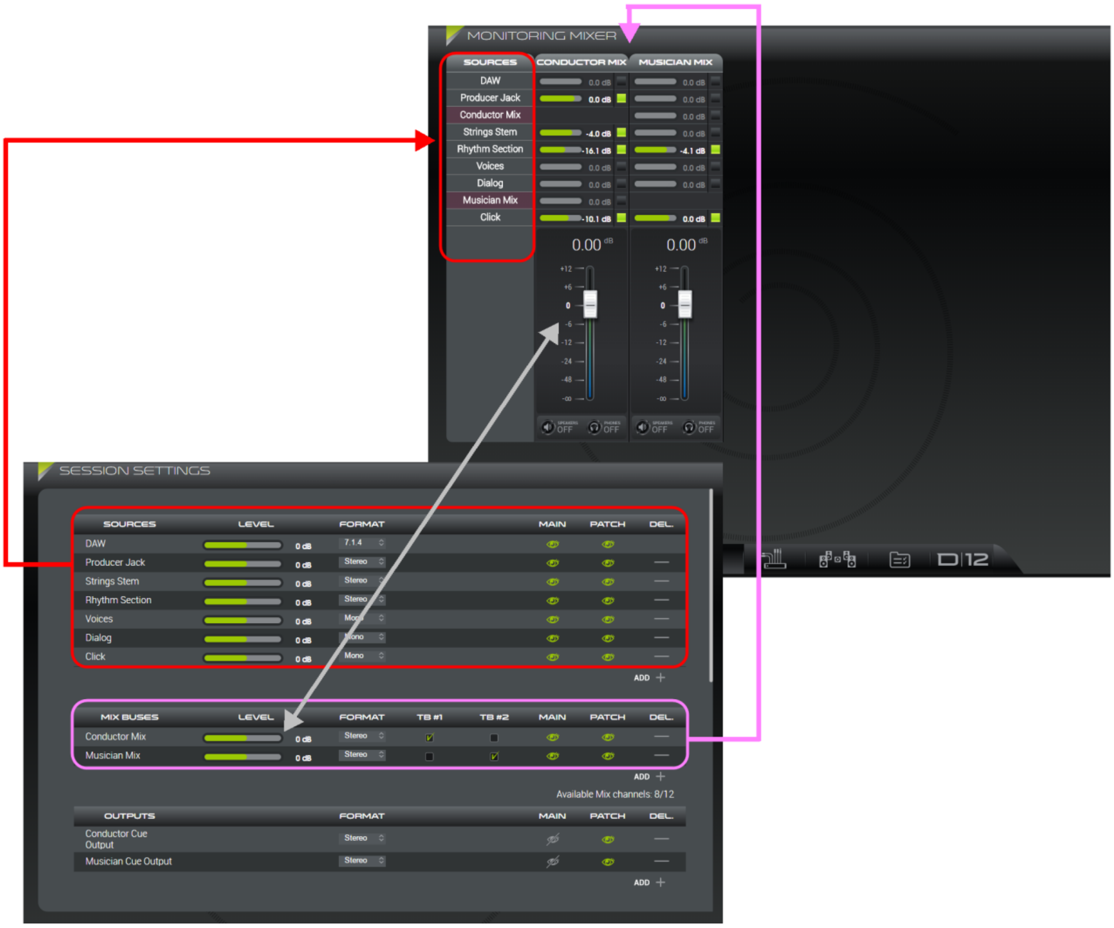
If the **Phones source follows the monitor** option is enabled in the *SESSION SETTINGS*, then the **PHONES** will automatically follow the **SPEAKERS** buttons. If this option is disabled, then you can choose to monitor a different mix on the speakers and headphones.

Turn off all the **SPEAKERS** and **PHONES** buttons to return to the previous monitor source (as selected in *The CONTROLLER Page*).

7.3.1 PREPARING THE MIXER

The diagram below shows how the available SOURCES and MIXES are determined by the *SESSION SETTINGS*. Therefore, you will need to configure this page before setting up the mixer.

Note that the LEVEL for MIX BUSES in the **SESSION SETTINGS** is the same as the fader level in the **MONITORING MIXER**.



Note also:

- If you assign a source to a mix using the summing points in the **MONITORING MIXER**, then this is reflected by the corresponding crosspoints in the **SESSION ROUTING** matrix. See *7.2: SESSION ROUTING*.
- The **SPEAKERS** and **PHONES** buttons in the **MONITORING MIXER** interact with the monitor source selectors in the **CONTROLLER** page. See *7.1.3: Monitoring Audio (on the Speakers)* and *7.1.10: Monitoring Audio (on the Headphones)*.

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7.3.2 SAVING SETTINGS (IN SNAPSHOTS & PRESETS)

All parameters in the **MONITORING MIXER** are saved in both snapshots and presets.

7.4 SESSION SETTINGS

This page defines parameters which apply to the session. It is divided into four separate areas which can be accessed by scrolling up and down the page:

- **Sources, Mix Buses & Outputs** (shown below) – define the resources available for the session.
- **Options** – for the monitoring such as “phones source follow monitor” and “auto-dim on talkback.”
- **Dynamic Range Simulation** – defines parameters for the Dynamic Range simulation (**DRC** button).
- **Remote Controllers** – defines the mapping functions to an external controller.

7.4.1 SAVING SETTINGS (IN SNAPSHOTS & PRESETS)

All parameters in the **SESSION SETTINGS** page are saved in both snapshots and presets.

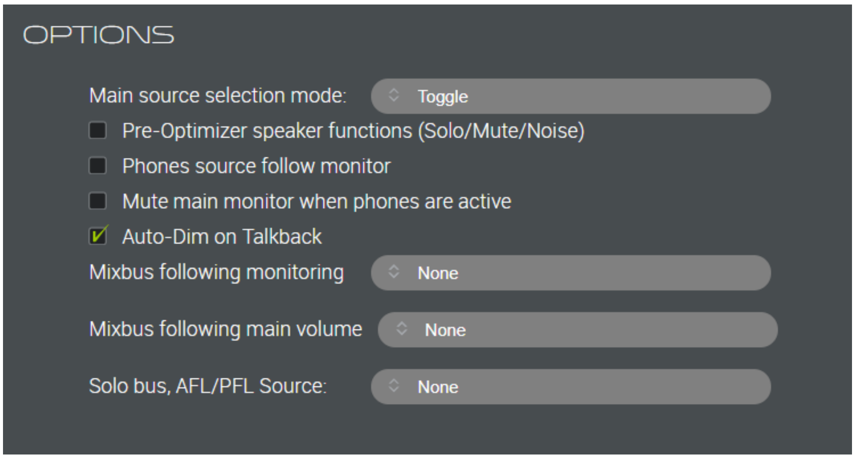
7.4.2 SESSION SETTINGS -> SOURCES, MIX BUSES & OUTPUTS

At the top of the page, you will see all the SOURCES, MIX BUSES and OUTPUTS which are available for the session. These affect the resources on the **CONTROLLER** page, **SESSION ROUTING** matrix and **MONITORING MIXER**.

The operation of this area is covered in the Quick Start chapter, so please see *5.9: Editing the Session Settings* for details.

7.4.3 SESSION SETTINGS -> OPTIONS

Scroll down the page to access the following options for the speaker and headphone monitoring:



- **Main source selection mode** – sets the default mode used at startup of the monitor source selector: either **Toggle** or **Sum**.
- **Pre-Optimizer speaker functions** – Defines if the per speaker SOLO / MUTE / NOISE speaker functions are done before or after the optimizer processing. This allows to include or not the bass management process: if the option is not checked, the function will be applied on the speaker output signal, else the function will be applied on the input signal.
- **Phones source follow monitor** – this option affects the headphone monitor source:
  - ON (ticked) = the headphones automatically follow the speaker monitor source.
  - OFF (not ticked) = the headphones monitor source can be selected independently.

  See *7.1.10: Monitoring Audio (on the Headphones)* for more details.
- **Mute main monitors when phones are active** – when engaged, this option automatically mute the main monitors upon headphone activation, and un-mute the main monitor when deactivating the headphones..
- **Auto-Dim on Talkback** – this option affects the speaker listening level:
  - ON (ticked) = the speaker level is dimmed whenever talkback is active.
  - OFF (not ticked) = the speaker level remains unchanged during talkback.

  When active, the dimming is triggered by both the individual talkback and **SLATE** buttons. See *7.1.8: Talking to an Output*. The amount of dim is the same as the normal dim level (defined under *STUDIO SETUP -> Options/Levels*.)

  You can use this feature to reduce the “noise” levels in the communication line and avoid feedback, via your talkback mic, in the speakers.
- **Mixbus following monitoring** – If a mixbus is selected there, its sources will be automatically updated to mirror the currently selected monitor sources. This can be used to have a mirror of the sources under listening without processing nor latency (i.e. For an external metering device).
- **Mixbus following main volume** – If a mixbus is selected there, its master output level will automatically follow the main volume level. This can be used to have a zero-latency monitoring (i.e. when tracking an artist) and control the output with the standard volume knob.
- **Solo bus, AFL/PFL source** – When a source is selected there, it can be activated with a GPI or with the griffin controller. This is likely to be linked with a GPO from the main DAW triggered upon a SOLO on any track, with a dedicated SOLO output bus.

7.4.4 SESSION SETTINGS -> DYNAMIC RANGE SIMULATION

Scroll down again to set the parameters for the Dynamic Range Simulation:

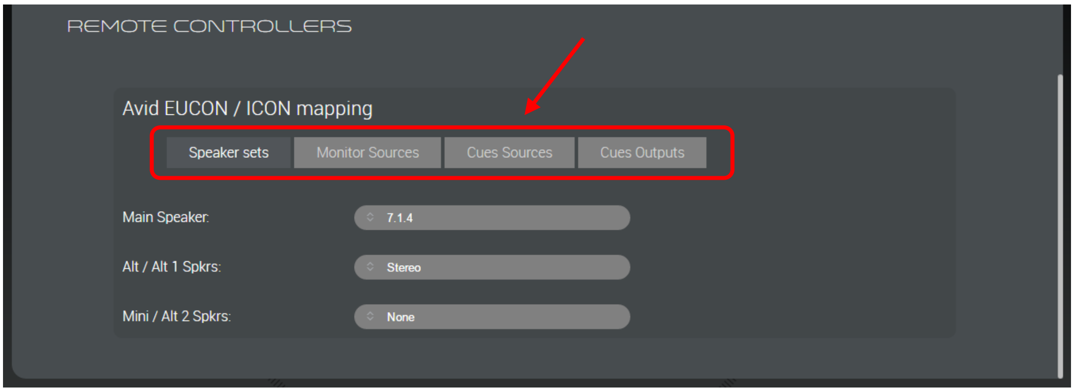
These parameters are applied whenever the **DRC** button on the **CONTROLLER** page is selected, see *7.1.7: Dynamic Range Compression (DRC)*. The simulation complies with the SMPTE 85 standard used in Dolby encoding:

- **Dialnorm ref. Level**: sets the “dialnorm” reference level in dB Full Scale.
- **Profile**: click to select a profile from the drop-down menu (shown above). This defines the compression curve applied (according to the SMPTE 85 standard).
- **RF Mode**: when enabled, both gain and the selected “profile” are applied. When disabled, the simulation runs in **Line Mode** (no additional gain).
- **Unity Gain**: tick this option to bypass the output gain applied by **RF Mode**. (Compression is still applied but without gain compensation.)
- **Delay Compensation**: tick this option to have the same latency whether **DRC** is on or off. (The **DRC** processing latency = 30ms. Therefore, when this option is enabled, a delay of 30ms is added when **DRC** is off and removed when **DRC** is on.)

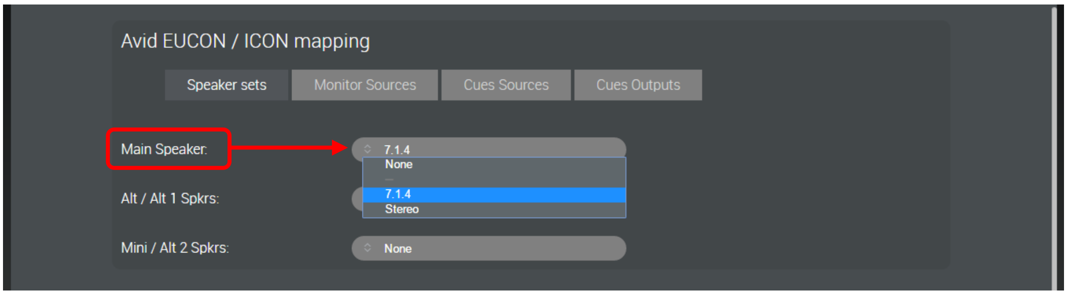
7.4.5 SESSION SETTINGS -> REMOTE CONTROLLERS

Scroll down to the bottom of the page to define the mapping of functions to an Avid EUCON or ICON controller.

There are four different sets of mappings which are accessed using the tabs at the top of the area: **Speaker sets**, **Monitor Sources**, **Cue Sources,** and **Cue Outputs**:



In each case, click to assign a D-Mon processor resource (e.g., a speaker set) to the controller function (e.g., **Main Speaker**):



The external controller functions are pre-determined by the connection protocol (either Avid EUCON or ICON). The available resources depend on the D-Mon processor configuration; in our example above, two speaker sets have been defined (in the *STUDIO SETUP -> Speaker Sets*) and, therefore, either of these can be assigned to the function **Main Speaker**.

> The EUCON protocol only requires speaker-sets and Cue Outputs mapping.
> The main and cues sources are automatically transposed.

Remember that all the SESSION SETTINGS parameters are stored and recalled in snapshots. This allows you to create different mappings for different users or controllers.

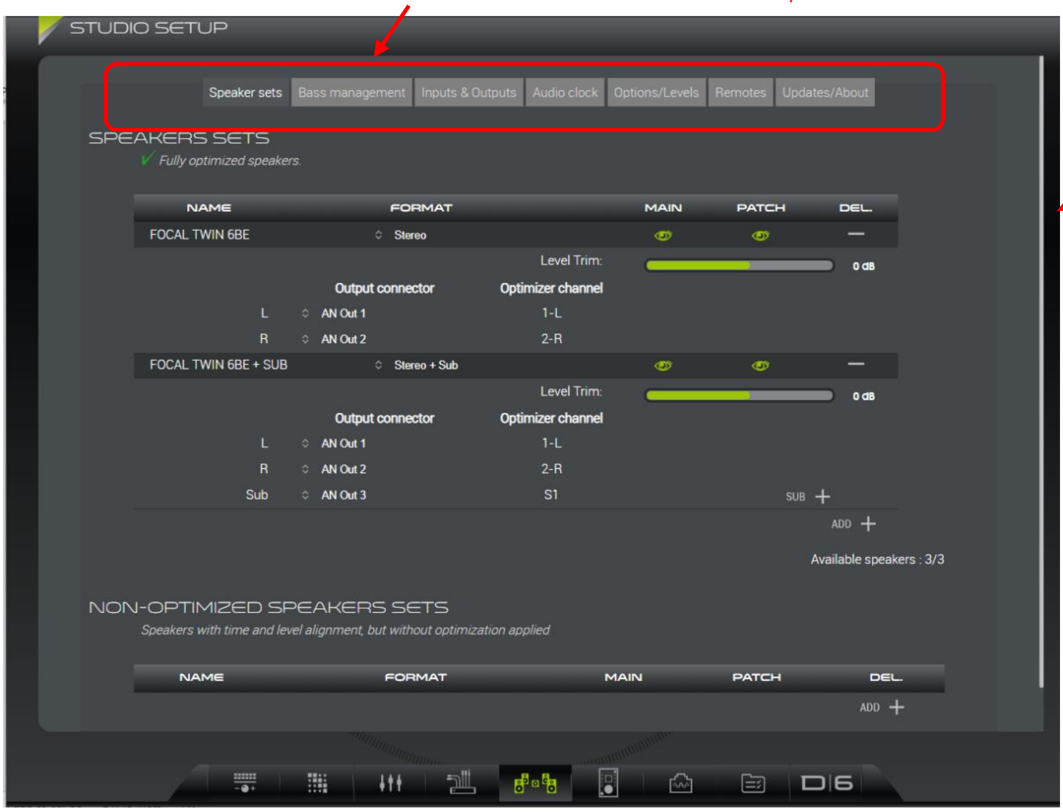
For more details on connecting the external control surface, see Chapter 8*: External Control.*

7.5 STUDIO SETUP

This page defines parameters which apply to the studio. It is divided into six different sets of parameters:

- **Speaker sets** – defines the studio's speaker sets (optimized and non-optimized).
- **Inputs & Outputs** – defines the talkback and listen-back inputs, plus the AES insert.
- **Audio clock** – options for the audio clock signal.
- **Options/Levels** – sets automatic dimming on talkback, plus the dim and reference levels for the speaker volume.
- **Remotes** – GPIO mappings for external controllers and the optional USB Volume controller.
- **Updates/About** – check for software updates and information about the product.

Use the tabs at the top of the page to access the parameter sets. In each case, you may need to scroll up and down:



7.5.1 SAVING SETTINGS (IN PRESETS)

All parameters defined in the **STUDIO SETUP** page are saved in presets, but not snapshots.

Most parameters are usually configured during the installation of the D-Mon processor, and a set of suitable presets for the studio prepared. This will allow you to reset the system for your studio by recalling a preset.

If you need to modify anything in the **STUDIO SETUP** page, then remember to save the change by overwriting an existing preset or creating a new preset file. See *7.6: CONFIGURATION PRESETS*.

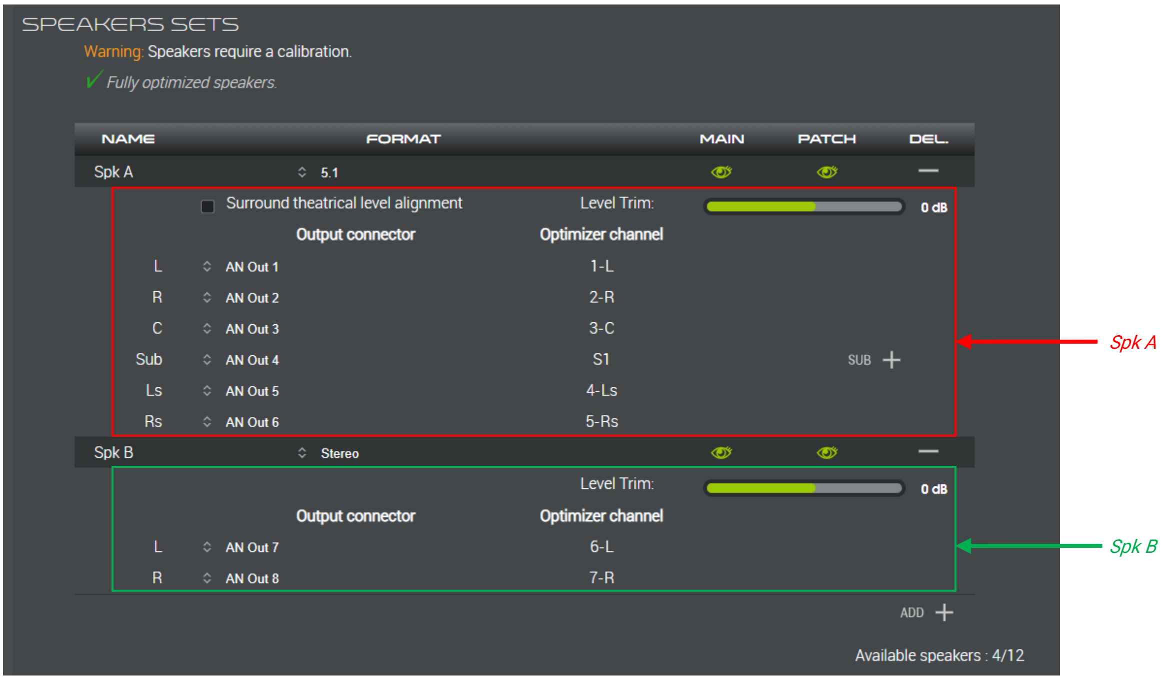
7.5.2 STUDIO SETUP -> SPEAKER SETS

This tab defines the name, format and physical connections for each speaker set, plus the bass management options. Once configured, each speaker set appears in the main **CONTROLLER** page and **SESSION ROUTING** matrix.

Note that speaker sets are divided into two categories, optimized and non-optimized. Optimized speaker sets appear first in the DMCP interface STUDIO SETUP -> Speaker Sets page. Scroll down to view the non-optimized speaker sets.

Optimized Speaker Sets	Non-optimized Speaker Sets
Fully calibrated (including acoustic correction)	Calibrated for time and level alignment only (no acoustic correction)
Designed for primary loudspeakers	Designed for secondary "reference" loudspeakers (e.g., Yamaha NS10s, Auratones, etc…)
Each loudspeaker uses an Optimizer® resource. Therefore, the maximum number is limited by the number of Optimizer® licences.	No Optimizer® resources are required. Therefore, the maximum number is limited only by the physical outputs available.

7.5.2.1 INTERROGATING THE OPTIMIZED SPEAKER SETS



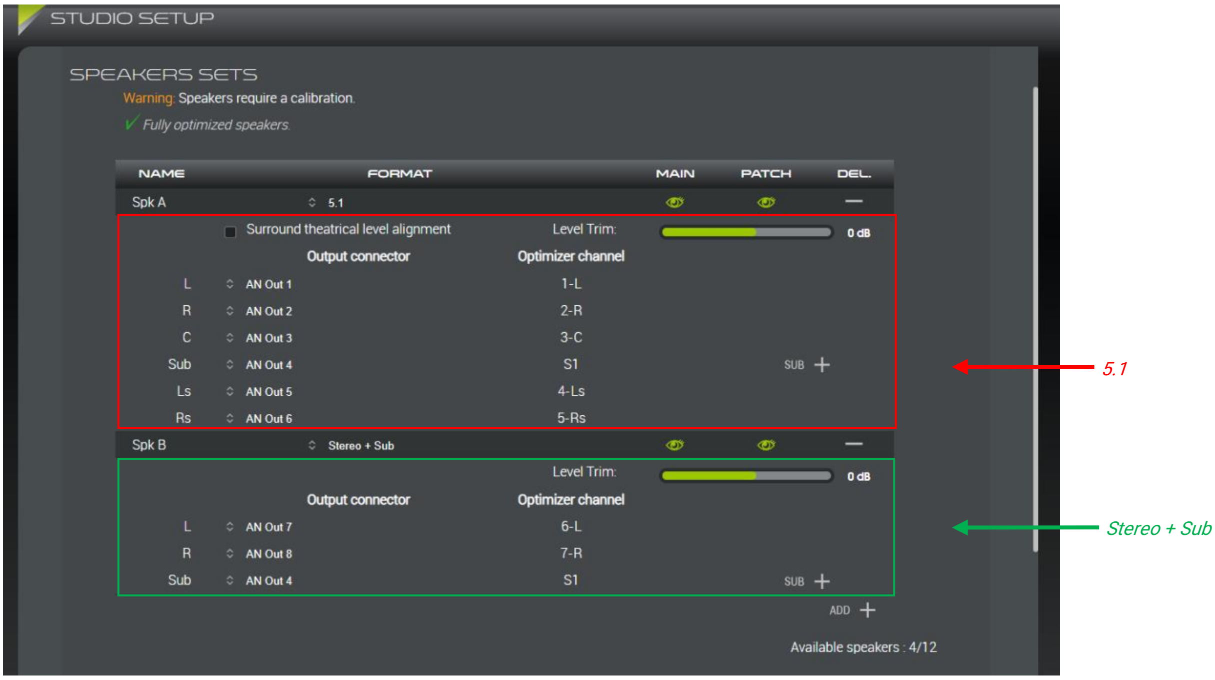
In the example above, two completely independent speaker sets have been defined:

- **Spk A** – is a 5.1 surround speaker set, connected to Analog outputs 1 to 6.
- **Spk B** – is a stereo + sub speaker set, connected to Analog outputs 7 & 8.

In this studio, **Spk A** will feed the 5.1 main surround monitoring, while **Spk B** will feed a pair of alternate stereo speakers. By choosing either **Spk A** or **Spk B** on the main CONTROLLER page, the user can switch between "Main" and "Alt" speakers.

Since each loudspeaker is fully optimized, it requires a dedicated Optimizer resource. So, in our **D-Mon|12** processors, we have used eight resources for our 5.1 + stereo speakers and have four remaining. This is indicated in the text readout at the bottom of the speaker sets: **Available speakers: 4/12**.

Configuration Example Two



In this example, there are still two speaker sets, but this time the subwoofers channel connects to the same physical output and, therefore, can share the same Optimizer channels (S1). We will see later that the bass management will allow custom property for the shared subwoofer depending on its referenced speaker-set.

This time when the user chooses a speaker set from the CONTROLLER page, they are listening either in **5.1** or **Stereo+Sub** but using the same subwoofer loudspeakers. Therefore, in this example, the **Available speaker's** count is still **4/12**.

Note that if the **Available speakers** show **0/x**, then all of the processor's Optimizer resources have been used. In this instance, it is still possible to define a non-optimized speaker set if you wish. See *Working with Non-Optimized Speaker Sets*.

Using Presets
Both the speaker set definitions and Optimizer settings are stored and recalled by presets. This means that you can create a range of presets to support different monitoring arrangements and "sweet spots" within your studio.
See *6.4: Configuring the System* for more details.

7.5.2.2 EDITING THE OPTIMIZED SPEAKER SETS

ATTENTION!

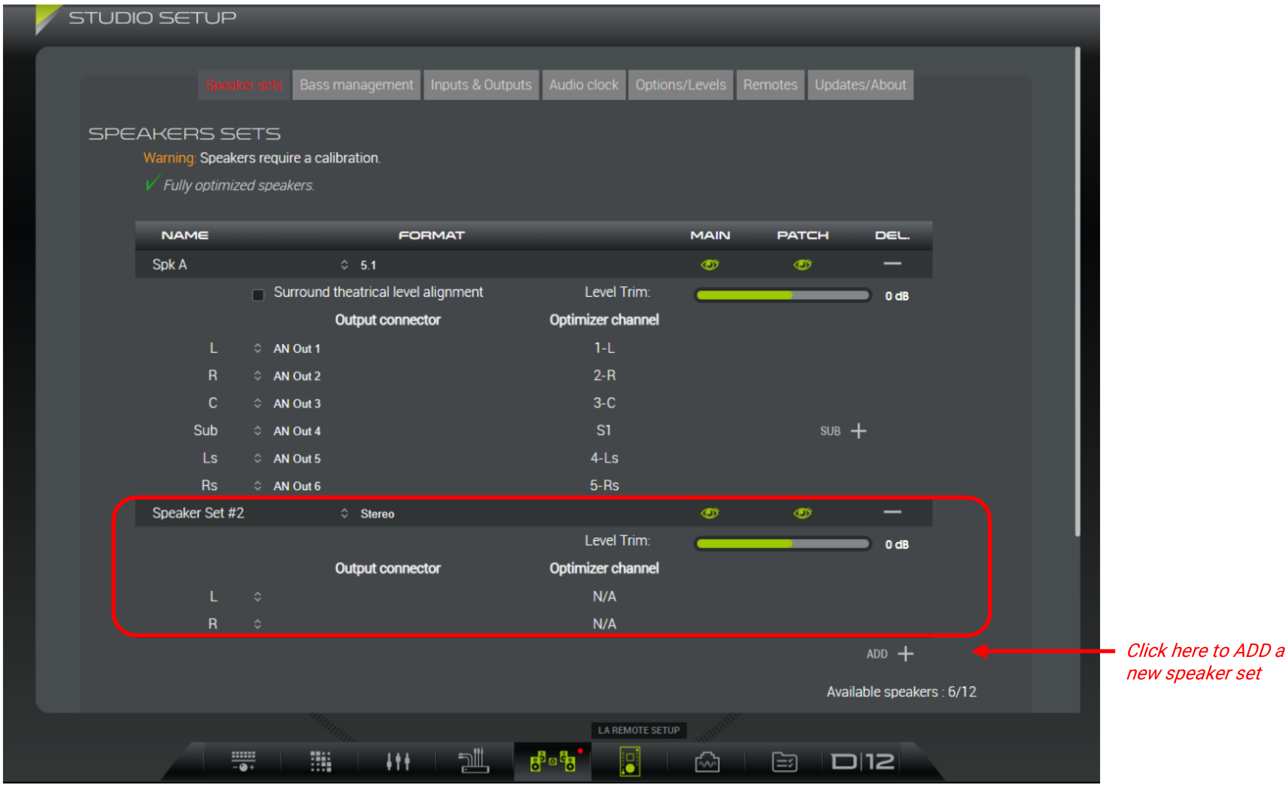
If you make changes to any speaker set (optimized or non-optimized), you will need to re-calibrate the speakers.

This means that if you change one output connector of a speaker, you MUST re-calibrate your whole system.

See *6.4: Configuring the System*.

Possible Operations

- Click on the **ADD +** button to create a new speaker set - the entry is given a default name (e.g. **Speaker Set #2**) and a generic set of parameters (**Format** = **Stereo, MAIN** = on, **PATCH** = on, **Level Trim** = **0dB**):



The number of optimized speaker sets is limited by the number of Optimizer® channels supported by your D-Mon processor. The number of channels remaining is indicated below the **ADD +** button. In our example, **Available speakers: 4/12** means that there are 4 mono Optimizer channels remaining from the 12 available in our D-Mon|12.

- For all speaker sets, you can define the following:

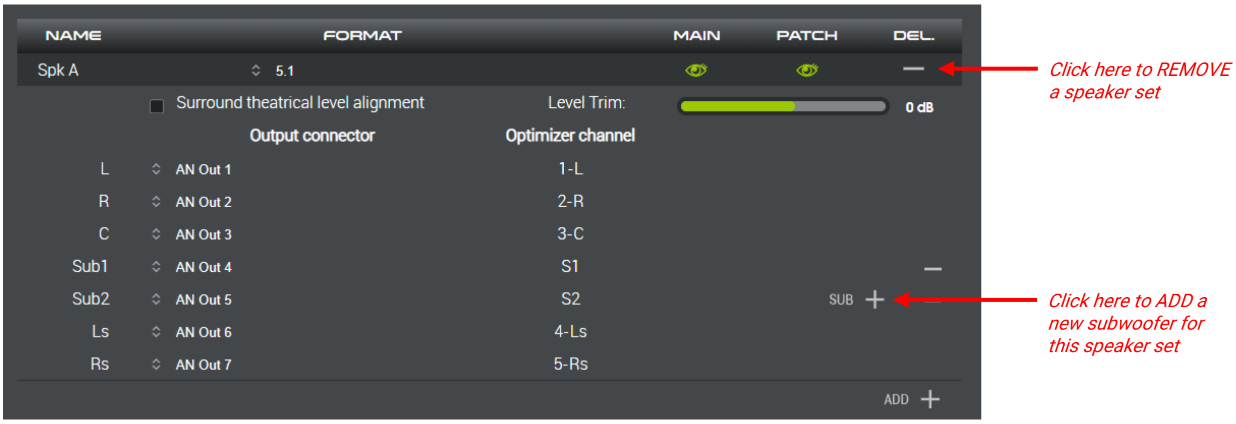
- **Name** – click to edit the name of the speaker set. You can click once to position the cursor, or double-click to select the text you wish to overwrite. Type in a name; the usual copy, cut and paste functions are available. Each name can be up to 16 characters, and all alphabetical and numerical signs are accepted.
- **FORMAT**– click to choose a format from the drop-down menu. The options will vary depending on your D-Mon model. If you choose a surround format, then the component channels are always allocated in the same order, up to the relevant number of channels: L, R, C, Sub, Ls, Rs, Lrs, Rrs, HL, HR, HLs, HRs.
- **MAIN** & **PATCH** – the “eye” icons determine whether a speaker set is visible in the **MAIN** (CONTROLLER) and **PATCH** (SESSION ROUTING) pages. When enabled, the icon lights in green.
- **Surround theatrical level alignment** – For speaker sets with surround channels, you can choose their alignment level: If this checkbox is ticked, the combined level of all surround speakers matches the level of one front speaker and conforms to the Dolby standard for cinema listening environments. If not ticked (default), each surround loudspeaker is level aligned like any other, each loudspeaker has an equal alignment level.
- **Level Trim** – click and drag on the slider to adjust the output level for all channels of the speaker set. Alternatively, double-click on the white text field, type in a value (e.g., 0dB) and press Enter.

  This level is done on top of the Optimizer calibration, which normalizes the levels across all speakers. Note that it is also possible to set a per speaker level trim in the OPCP interface Processor -> Outputs, which is therefore combined with this speaker set level at processing.

- **Output connector** – click to assign a physical output to each channel of the speaker set – you can choose any AES or analog output. Once assigned, the **Optimizer channel column** will update itself.

> **Re-using some speakers in different speakers sets**
> Is it possible to re-use an optimizer channel in another speaker set by selecting the same physical output for the same channel label (i.e. In the previous screen capture, re-use the analogue output 4 for the subwoofer channel of the Speaker-set #2).
>
> This tip allows the re-use of a subwoofer across several speaker sets, or to mix different speaker models for the same speaker base.
>
> It also makes possible a fine usage of the downmixes that are related to the current speaker set format.

- For speaker sets with a subwoofer (.1) channel, you can add several subwoofers to it.
- Click on the DELETE symbol (in the right-hand column) to remove a speaker set.



> **ATTENTION!**
> There is no undo for the delete operation, so take care when selecting this symbol!
> Deleting a speaker set will remove it from the **CONTROLLER** and **SESSION ROUTING** pages.

7.5.2.3 WORKING WITH NON-OPTIMIZED SPEAKER SETS

Non-optimized speaker sets can be edited identically to optimized speaker sets. However, the number which can be defined is not limited by the D-Mon processor's Optimizer® channel resource. In our example below, a stereo non-optimized speaker set (**Spk C**) has been added to an existing configuration (with one **Spk A** 5.1 optimized speaker set):

Note that:

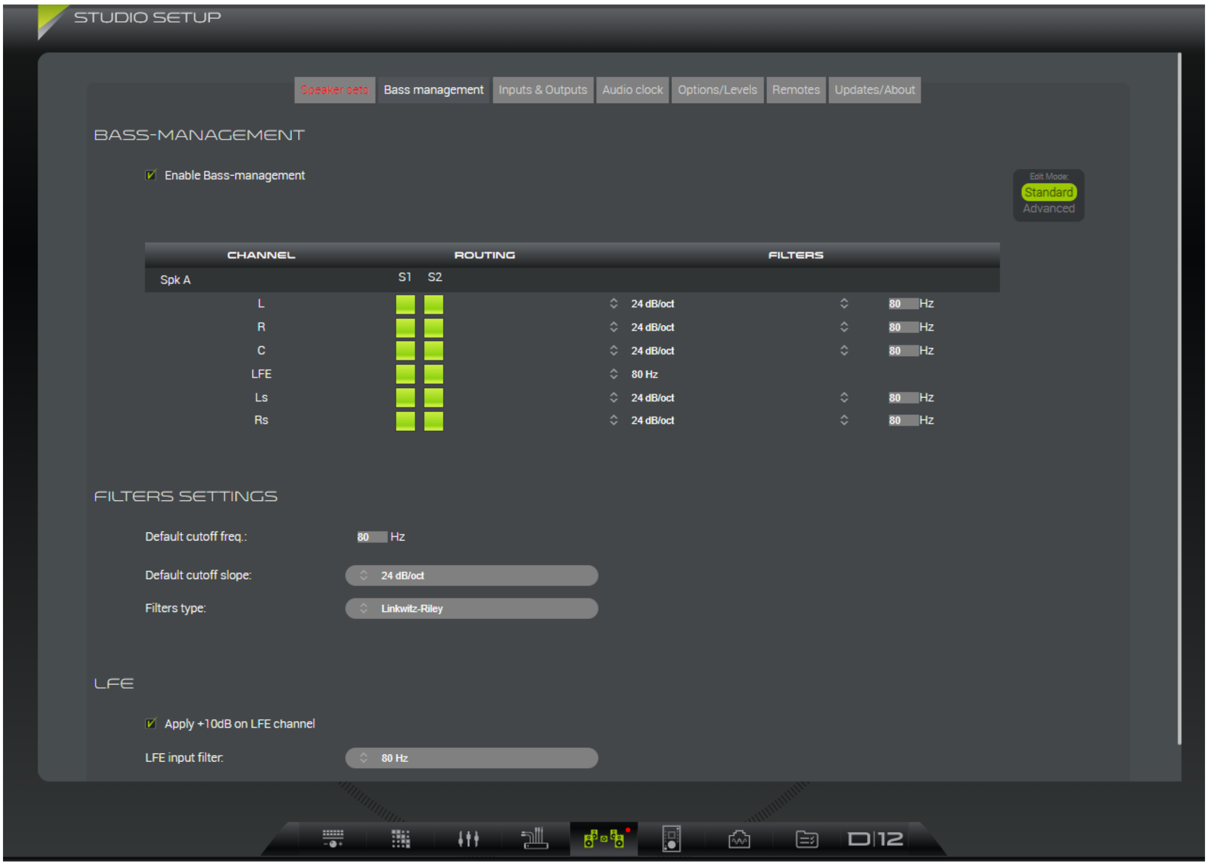
- Under OPTIMIZED SPEAKER SETS, the number of **Available speakers** remains unchanged at **6/12**.
- Under NON-OPTIMIZED SPEAKER SETS, an **Optimizer channel** is still assigned (in our example, **6-L,** and **7-R**). This is because the channel is still processed (to apply the time and level alignment).

ATTENTION!

If you make changes to any speaker set (optimized or non-optimized), you will need to re-calibrate the speakers.
See *6.4: Configuring the System*.

7.5.3 STUDIO SETUP -> BASS-MANAGEMENT

- Select the second top tab to access the Bass-Management section:



These options apply to speaker sets with a subwoofer (.1) channel only:

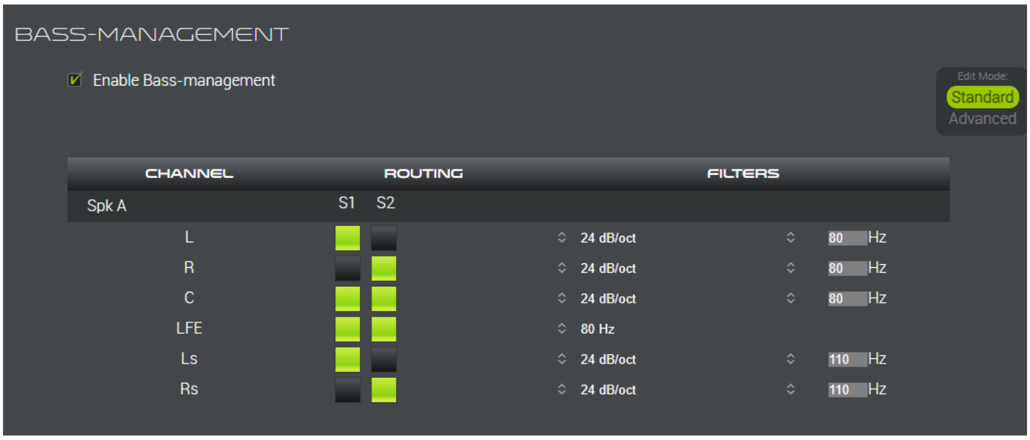
- **Enable Bass-management** – tick to enable the Bass-Management globally on your system.
  When enabled, the D-Mon processor will sum the low-frequency ends of all signals within the speaker set, and route the summed audio to the Sub speaker channel. You can select precisely which channel should be added to this summation, and the destination subwoofer (in case several subwoofers are declared). This could allow you to create a subwoofer channel from a mix without LFE.

Note that, if the **Enable Bass-management** option is off, then the subchannel is fed from the **LFE** channel of your monitor source in the usual manner. In this case, the LFE input filter is still applied if one frequency is selected.

Edit Modes

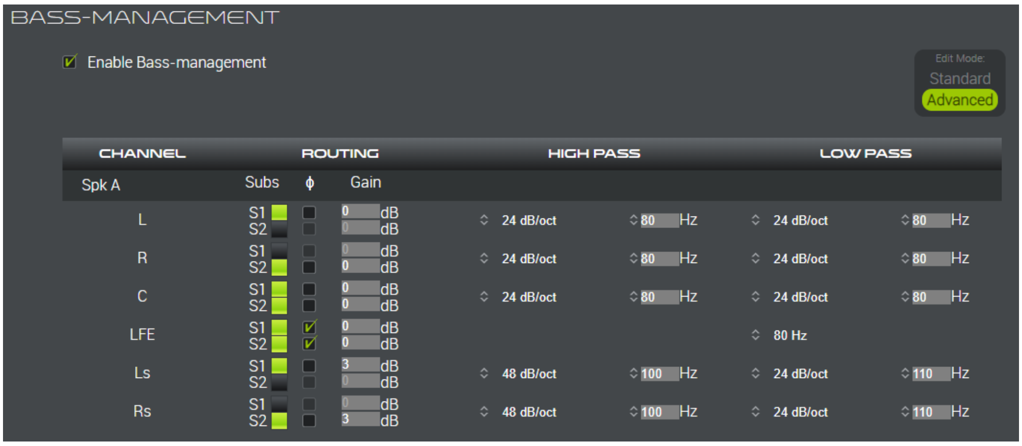
Two editing modes are available: the standard one (default), and the advanced one if you need some fine tweaking.

- **Standard Edit mode**



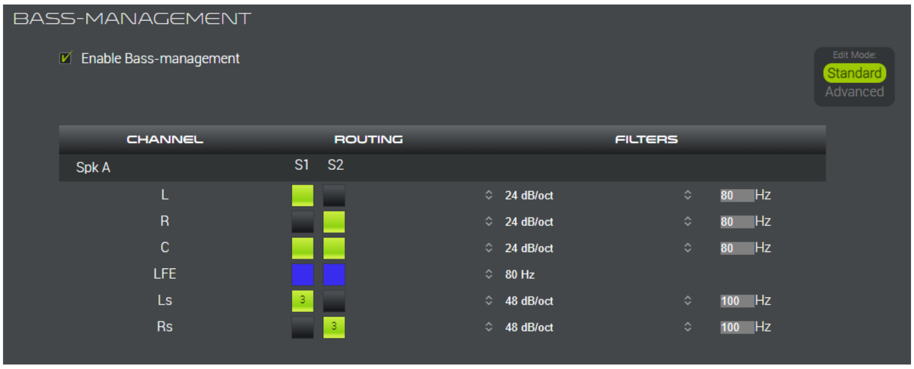
Here, a 5.1 speaker set has a bass-management balanced over two subwoofers, one for the left side and one for the right side. The center and LFE channels will be sent to both subwoofers, with a global resulting level coherent with this double send. Additionally, it is possible to specify the slope and frequency of the filters used for both the high-pass and low-pass. In the above screenshot, the surround speakers have a higher cut-off frequency.

- **Advanced Edit mode**



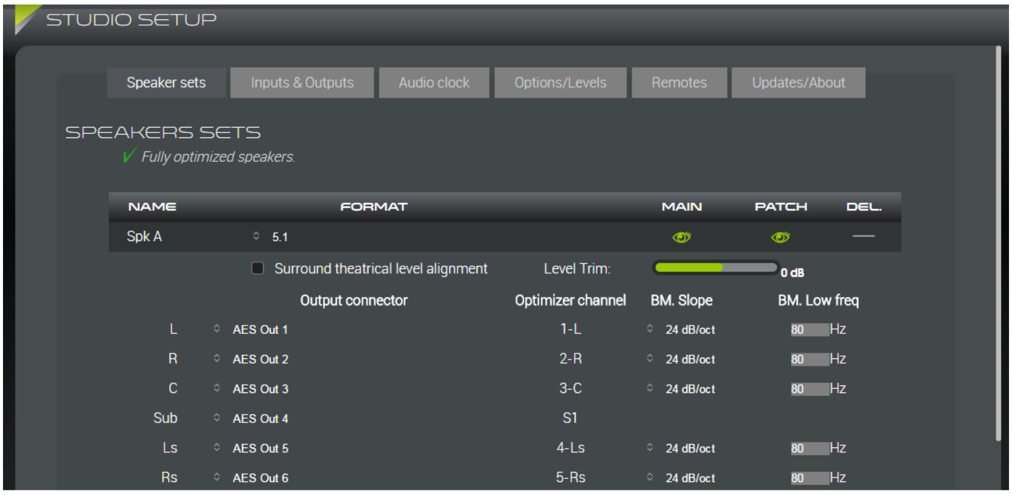
In this mode, it is possible to adjust a send gain and phase reverse from each speaker to each target subwoofer. Also it is possible to have different settings for the high-pass filter (channel content kept in its speaker) and low-pass filter (channel content sent to the subwoofer).

> Swapping the edit mode back to Standard after some fine tuning still keeps and applies the advanced parameters, therefore any custom send gain is displayed in the routing point and polarity reversal is denoted with a blue patch color.



Filters settings:

- **Default cut-off frequency** and **Default cut-off slope** – reset all filter values in all speakers to a common one. You can then edit the individual settings for each speaker channel.



- The **Filters type** defines the filtering type applied to all filters.

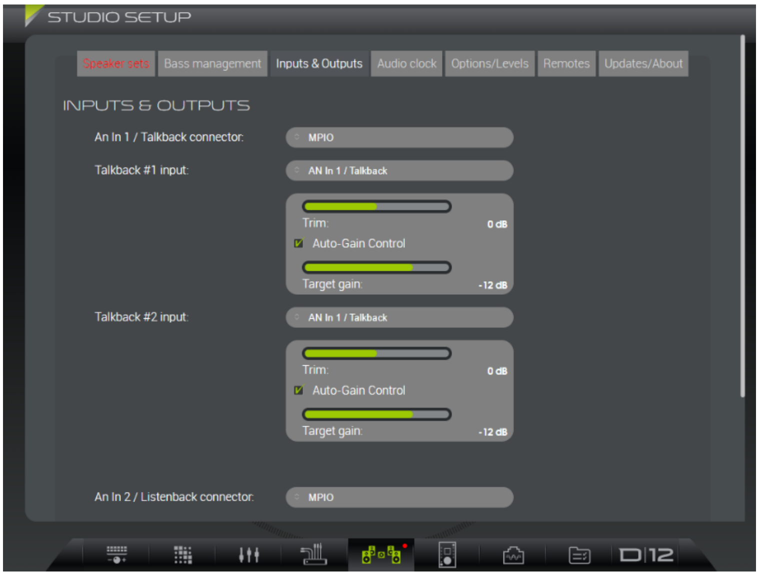
LFE:

Those parameters are applied even with the Bass-management disengaged.

- **Apply +10 dB on LFE channel** – tick to enable the usual additional gain in the LFE to subwoofer transition.
- **LFE input filter**– It is possible to select a cutoff frequency to only send the low end part of the LFE signal to the subwoofer.

7.5.4 STUDIO SETUP -> INPUTS & OUTPUTS

This tab defines the talkback and listen-back inputs, plus the AES insert.



Talkback

The system supports two talkback inputs which can be used to talk to the session's mix buses, see *7.1.8 Talking to an Output*. The physical input connectors, level trims, and limiter/auto-gain are specified here in the STUDIO SETUP:

- **An 1 / Talkback connector** – click on this field to choose the connector used for both inputs. You can choose either the **MPIO**, **SubD** connector or **Follow Talkback** (please read the note below).
- **Talkback #1 Input** - click on the I/O selector to assign an input. You can choose any analog or AES in (but please read the note below). Then click and drag on the **Trim** slider to adjust the input level, followed by the **Limiter** option and its **Target** slider to set an automatic target level (e.g., **12dB**). Note that, although this option is called "**Limiter**," the processor applies either limiting OR automatic gain to reach the desired target level. The limiter/auto-gain occurs after the input **Trim** level.
- **Talkback #2 Input** – repeat for talkback input 2 as required.

If the studio has a single talkback microphone, then you should define the same physical input for **Talkback input #1** and **Talkback input #2** as shown above. Alternatively, you can assign different physical inputs to support two talkback positions (e.g., to use **Talkback input #1** for a sound engineer and **Talkback input #2** for a producer).

Having defined the physical inputs, remember to use the SESSION SETTINGS page to determine how each talkback input feeds the session's mix buses. See *5.9 Editing the Session Settings*.

> Analog inputs 1 & 2 are the only inputs suitable for direct connection to a microphone (as they provide the necessary phantom power). Analog input 1 is reserved for talkback and analog input 2 for listen-back. Both are named accordingly within the I/O selector menus. Note that the SubD analog inputs (1 & 2) are hard-wired to the MPIO connector (TB & LB). This means that if you reassign talkback and listen-back away from analog inputs 1 & 2, you will lose phantom power and also the talkback & listen-back signals from the MPIO connector.
>
> It is possible to save 2 extra inputs by choosing the "Follow talkback" mode, which will select the subD connector by default, and automatically switch to the MPIO connector once a talkback is engaged.

Listen-back

Scroll down the page to access similar options for the listen-back inputs:

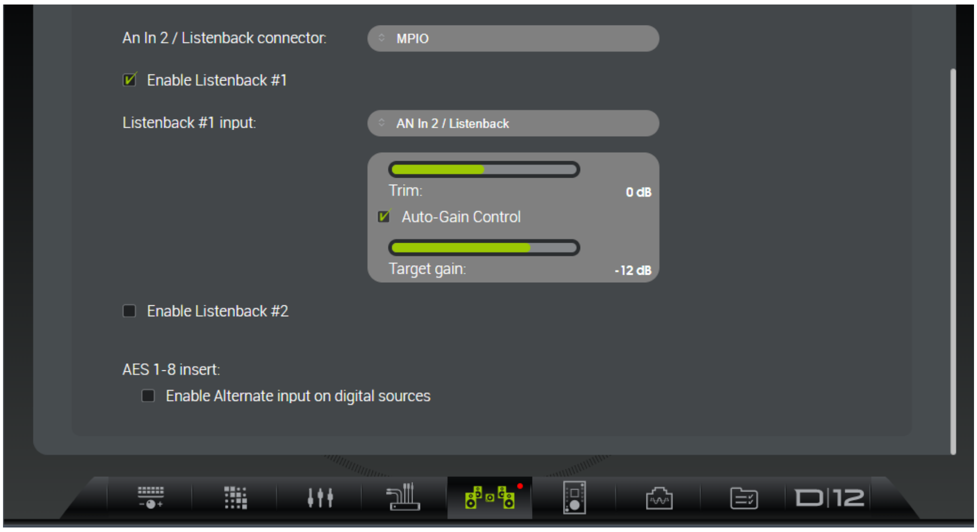
The system supports two listen-back inputs which can be used to return talkback from a particular studio location, see *7.1.9: Listening "In" to the Studio*.

- **An 2 / Listenback connector** - click on this field to choose the connector used for both inputs. You can choose either the **MPIO**, **SubD** connector or **Follow Talkback** (but please read the note above).

> The default value of the **An 1 / Talkback connector** and **An 2 / Listenback connector** is MPIO:
> if no signal comes to the first two channels, double-check this option.

- **Enable Listenback #1** – tick this checkbox to enable the first listen-back input. You can then define its input, level trim and limiter/auto-gain (in the same manner as for the talkback inputs). You can choose any analog or AES in (but please read the note above).
- **Enable Listenback #2** – repeat for listen-back input 2 as required.

In our example, only **Listenback #1** is enabled. Therefore, only **LISTEN 1** will appear in the various monitor source selectors (on the CONTROLLER page).

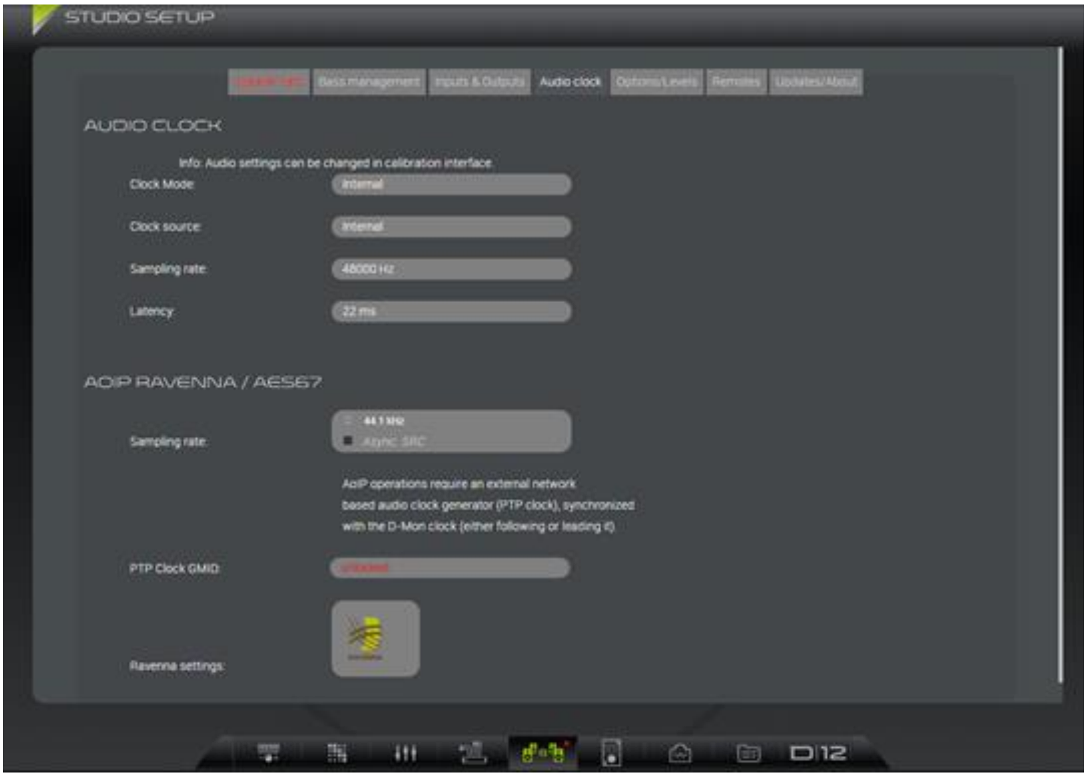


AES Insert

Scroll down to the bottom of the page to enable (or disable) the AES 1-8 Insert, see *7.1.10: Switching the AES 1-8 Insert*.

7.5.5 STUDIO SETUP -> AUDIO CLOCK

This tab shows the status of the audio clock sampling frequency and source:

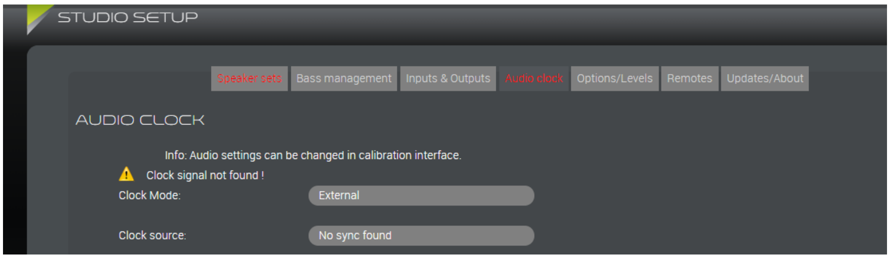


These parameters are “read-only” and are included for information only. To change the audio clock configuration, you will need to use the **Optimizer & Processor Control Panel** (OPCP) GUI. See *6.4.6: Defining the Audio Clock* for details.

When using Audio over IP, you can adjust here the sampling rate of the sent and received stream. In order to be able to send or receive audio into the D-Mon, this sampling rate must match the D-Mon main audio clock.

When a valid PTP (precise time protocol v2) leader clock is found locked on the network and the D-Mon AoIP stack locked to it, the **PTP Clock GMID** shows the MAC address of this leader clock. Please refer to chapter 9.2: AoIP clocks.

If there is a problem with the clock signal, then you will see a red “invalid clock” warning.



7.5.6 STUDIO SETUP -> OPTIONS/LEVELS/POWER-ON STATE

This tab sets various options for the speaker listening level:

Options

- **Auto-Dim on Talkback** – this option is identical to the one defined under *SESSION SETTINGS -> Options*. It is repeated here so that it can be saved and loaded by Presets.

Levels

These options affect how the speaker level (on the CONTROLLER page) is displayed, and the **Reference** and **DIM** functions, see *7.1.4: Adjusting the Listening Level (Speaker Volume)*.

- **Display unit** - defines how the speaker level is displayed:
  - **Unreferenced** = the level value is unreferenced and expressed in dB, from -60 to +20 in 1 dB steps.
  - **dB C** = 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 in 1 dB steps.
  - **Cinema** = the level value is relative to the current calibration level and is expressed as a range, from 0 to 10 in 0.1 steps.
- **Reference Volume recall level** – defines the speaker reference level value (recalled by the **Ref** button on the CONTROLLER page).
- **Maximum Volume level** – defines the maximum speaker level (available on the CONTROLLER page).
- **Dim offset level** – defines the amount of dim applied to the speaker level (when you press the **DIM** button or auto-dim on talkback is active).
- **Maximum Headphone level** – defines the maximum headphone level (available on the CONTROLLER page).

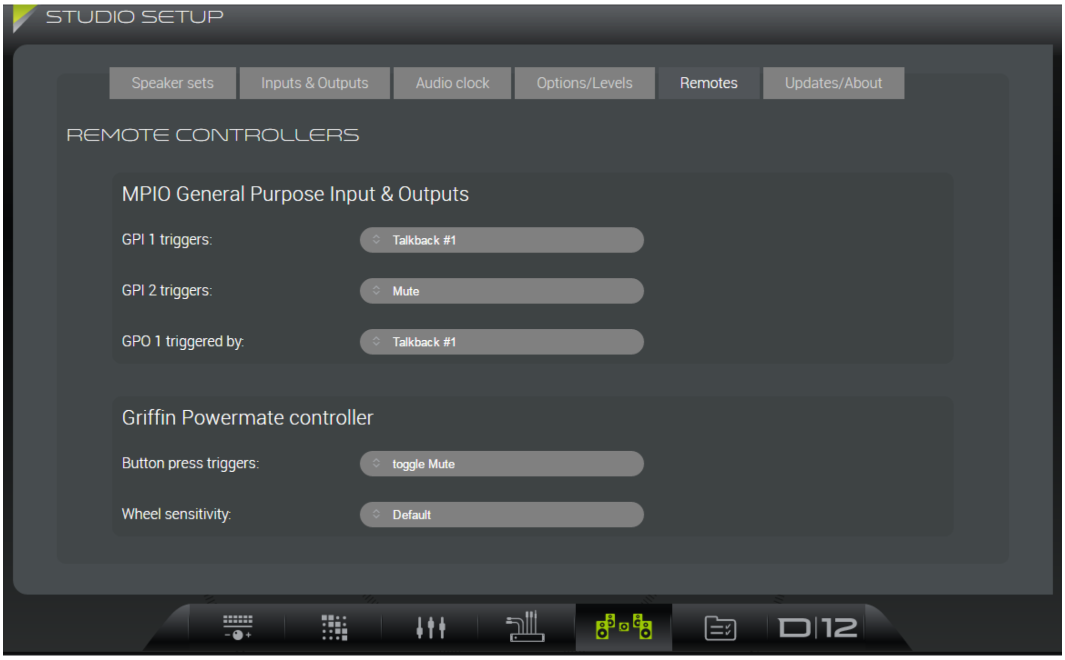
Power-On State

This option is not related to a preset and saved globally.

- **Auto-mute on power-on** – automatically engage the mute at machine startup. The user will need to unmute explicitly to send any signal to a speaker set.
- **Recall fixed level on power-on** – By default, at machine startup, the volume is set to the last level, at machine shutdown. This option allows starting the machine with a default level, independently of the last volume set at the previous shutdown.

7.5.7 STUDIO SETUP -> REMOTES

This tab sets the GPIO mappings for external controllers and the optional USB Volume controller.



MPIO General Purpose Inputs & Outputs

These fields will map a D-Mon processor function to a General Purpose Input or Output (GPI or GPO). There are two GPIs and one GPO available. The physical connections are made via the MPIO connector. See *10.1.2: Multi-Purpose In/Out (MPIO) SubD-25 Connector* for wiring information.

In each case, click on a field and choose a function from the drop-down menu.

You can choose from:

- **Talkback #1** or **Talkback #2**– the GPI will trigger the **TB1** or **TB2** button. See *7.1.8 Talking to an Output.*
- **Mute** or **Dim** – the GPI will mute or dim the speakers. See *7.1.4: Adjusting the Listening Level (Speaker Volume).*
- **AES Insert** – the GPI will switch the AES Insert in and out of the circuit. See *7.1.10: Switching the AES 1-8 Insert*.

For the GPO, you have the same options as above, but this time the output will be triggered by the function - for example, to illuminate a talk button when talkback is active.

Griffin Powermate controller (discontinued)

These fields configure the button and wheel on the USB Volume controller.

- Click on the first field to define the button - you can choose either :
  - to **Mute** the speakers
  - to trigger a talkback button (**TB #1** or **TB #2**), as Toggle or Momentary mode
  - to cycle thru speaker sets
  - to toggle the AFL/PFL source
  - to toggle the AES insert
- Click on the second field to set the sensitivity of the wheel.
  This is always mapped to the speaker level (volume).

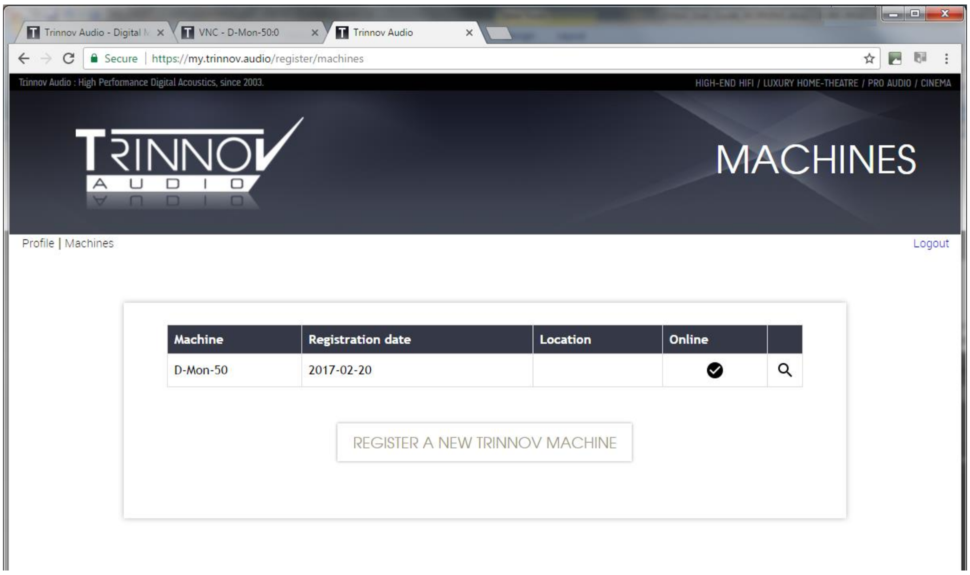
If more than one USB Volume controller is connected, then all devices are assigned the same functionality.

7.5.8 STUDIO SETUP -> UPDATES/ABOUT

This tab provides information about the product and its software status:

The unit can be updated (via the Trinnov server) once registered. Therefore, for a first-time update, please follow these steps:

Step	Instructions	Comments
1	Connect your unit to the internet and go to register.trinnov.com	
2	Follow the on-screen instructions to log in (or create a new user profile), and register your unit. Once registration is complete, you will see a list of your machines including their **Online** status:	



Step	Instructions	Comments
3	Click on the Search icon to reveal further information about the device. From here you can load microphone compensation files to the unit, and enable (or disable) software auto-updates:	

Step	Instructions	Comments
4	Set the **Software Auto-Updates** option to **Yes** to allow updates to be launched directly from the D-Mon Control Panel interface. (Note that you can set this option to **No** to prevent updates being launched from the user interface).	
5	Open the D-Mon Control Panel GUI and select the **STUDIO SETUP -> Updates/About** tab.	

6 Click on **Check for updates**. (The software will also be checked automatically at the next reboot).

In the STATUS field, you will see:

- "An update is available" along with the available release number and any release notes = proceed to step 7.
- "Unable to reach update server" = check your internet connection and registration details.
- "Machine is running the latest release" = your unit is up to date, and no further action is required.

7 Follow the on-screen instructions to **Switch OFF your amplifiers**, and then click on **Proceed to install** to start the update. The update server will now run the install. Once complete, the unit will reboot.

——————————

The D-Mon processor should now be up and running! You can check the software version from the STUDIO SETUP -> Updates/About tab.

7.6 CONFIGURATION PRESETS

This icon opens a pop-up menu where you can recall either a snapshot (to reset session parameters) or a preset (to reset the complete unit). The **MANAGE** button opens a full-size page with further file management options: create, lock, export, delete, etc.

7.6.1 ABOUT SNAPSHOTS AND PRESETS

To save and load settings, the system uses two different file types:

- A studio **preset** stores everything required to reset the D-Mon processor – i.e., all the GUI page parameters (including the **STUDIO SETUP**), plus other lower-level settings defined during installation (such as the Optimizer channels).
- A session **snapshot** stores only session-related parameters – i.e., the current state of the **CONTROLLER** page, plus the **SESSION SETTINGS**, **SESSION ROUTING** and **MONITORING MIXER**.

You should use presets to store settings for the studio – for example, different monitoring arrangements. Then use snapshots to store settings for each DAW session – for example, the sends and returns to the DAW and studio floor.

This allows you to reset the unit completely using only a preset. OR, recall a preset followed by a snapshot to apply different session-related parameters to different monitoring arrangements.

> Note that there are a few parameters which are NOT saved in either presets or snapshots. These settings are always absolute and remain at their last values. They are the:
>
> - **Speaker listening level**, **MUTE,** and **DIM** set in *The CONTROLLER* page.
> - **Network** parameters defined in the OPCP GUI during installation, see *6.2.2: Setting up the CPU*.
> - **Audio clock** parameters defined under *STUDIO SETUP -> Audio Clock* (if “Store in preset” is set to off in the OPCP GUI, see *6.4.6: Defining the Audio Clock*).
>
> Each unit can store up to 30 **presets** (29 user presets + the **Built-in** preset which cannot be deleted).
> The number of **snapshots** is unlimited.

7.6.2 FACTORY-CONFIGURED SNAPSHOTS AND PRESETS

All D-Mon processors include some factory-configured snapshots and presets.

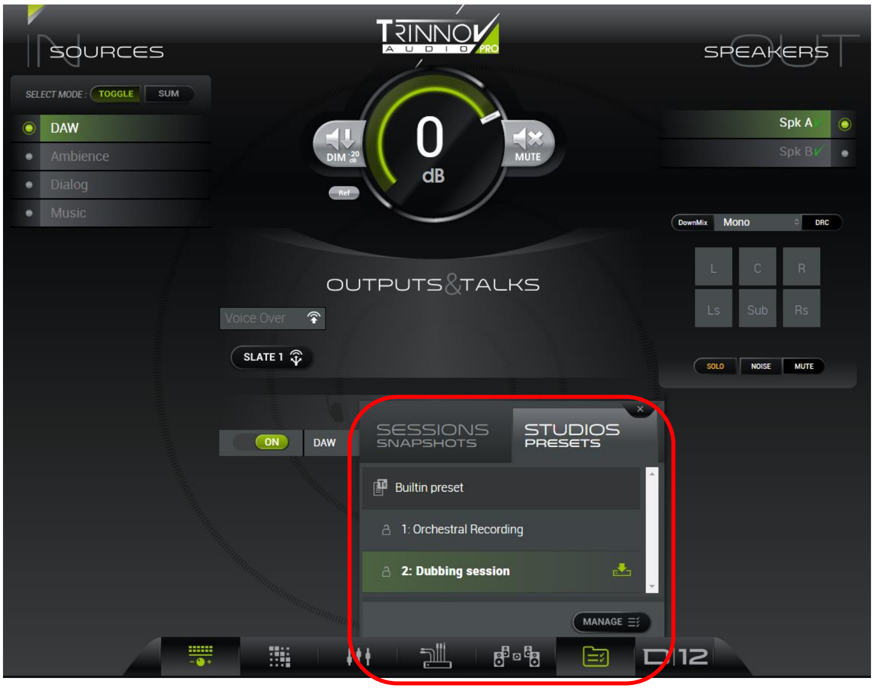
These are indicated by the **Tr** (Trinnov) symbol and cannot be renamed, overwritten or deleted. They will reset the relevant parts of the system to a known set of parameters, and provide a good starting point for defining customer-specific snapshots and presets. Note that the exact contents vary according to the D-Mon processor model.

The **Built-in** preset and **Default** snapshot appear on all D-Mon processor models. Note that the Default snapshot is the same as the Built-in state, but recalls only the session settings (rather than studio-specific parameters like the Optimizer channels).

Other factory-configured snapshots can appear depending on the unit’s options. For example, the **CB-TMC** snapshot is a snapshot that supports the Colin Broad TMC-1 remote controller.

7.6.3 USING THE POP-UP MENU

When you first select **CONFIGURATION PRESETS**, a pop-up menu appears over the top of the current page. This allows you to quickly access settings without having to change your view:



Possible Operations

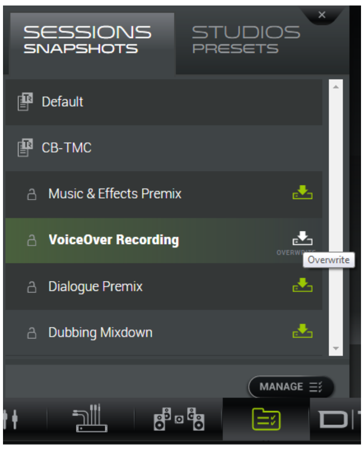
- ➢ Use the tabs at the top of the menu to view either **SESSION SNAPSHOTS** or **STUDIO PRESETS**.
- ➢ Scroll up and down to view all available files. Before each file name you will see one of the following icons:

- **Tr** – indicates a Trinnov (factory-configured) snapshot or preset. These files cannot be modified.
- **Padlock Open/Closed** – appears beside each user-defined snapshot or preset; if the padlock icon is closed, then the file is locked and cannot be overwritten, renamed or deleted.

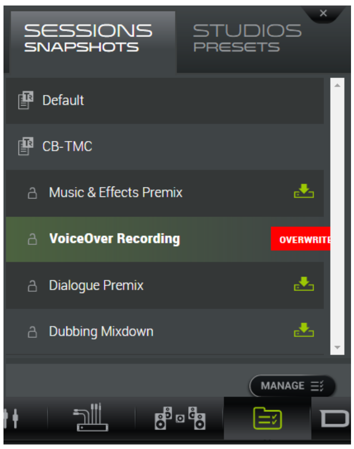
The current file is highlighted in green (e.g., **Dubbing session**); this is the last snapshot or preset to be recalled or overwritten.

- ➢ Click on the snapshot/preset name to load its settings; the file name highlights in green once the recall is complete.
- ➢ Click on the Overwrite icon (to the right of the file name) to store the current settings into a snapshot or preset. A red **OVERWRITE** button appears – click on the button to confirm, or click somewhere else to cancel.

Overwrite (step 1)



Overwrite (step 2)



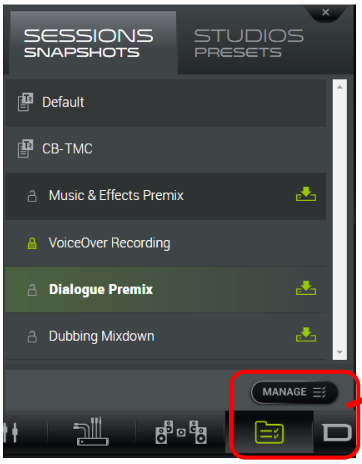
- ➢ Click on the **MANAGE** button to open a full-size page with further file management options.
- ➢ Click on the **X** (at the top right of the pop-up menu) to close the **CONFIGURATION PRESETS** pop-up menu.

7.6.4 USING THE FULL PAGE VIEW

To access other file management options such as Lock, Delete, etc., you will need to open the full page view.

➢ Select **CONFIGURATION PRESETS** followed by the **MANAGE** button – the pop-up menu is replaced by the full pageview:

Pop-up Menu



Full Page View

Possible Operations

➢ Use the tabs at the top of the menu to view either **SESSION SNAPSHOTS** or **STUDIO PRESETS**.

➢ Scroll up and down to view all available files. The last snapshot/preset to be recalled or overwritten is marked as CURRENT (e.g., **Dialogue Premix**).

➢ Click and type in the **Search** field to filter the list – as you type, only file names which include the text string are displayed.

➢ Select an option from the **Sort** menu to sort the list:

- **Last modified first** = the last file to be overwritten or created is displayed at the top of the list.
- **Alphabetical order** = files are sorted by name from A to Z.
- **Slot order** = files appear in the order in which they were created.

➢ Click on the green **RECALL** symbol to load a file – the relevant settings are recalled and the CURRENT indicator updates.

➢ Click in the name field to edit the file name. You can click once to position the cursor, or double-click to select the text you wish to overwrite. Type in a name; the usual copy, cut and paste functions are available. Each name can be up to 16 characters, and all alphabetical and numerical signs are accepted.

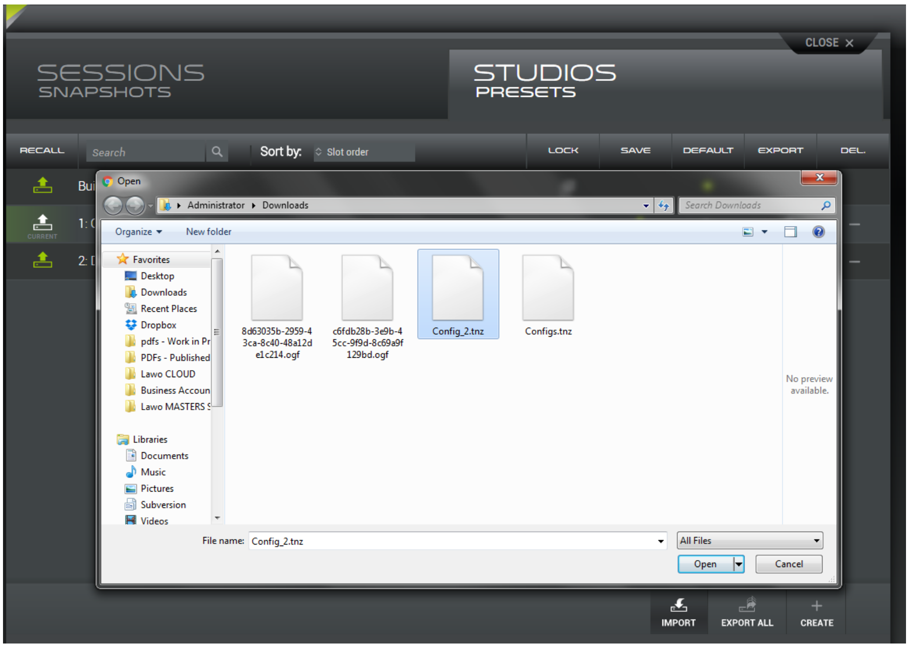
➢ Click on the **LOCK** symbol to lock (or unlock) a user-defined file — the **LOCK** symbol lights in green when active. A locked file cannot be renamed, overwritten or deleted. Factory-configured files are always locked and are indicated by the **Tr** icon.

➢ Click on the green **SAVE** symbol to overwrite the current settings into an existing file. A red **OVERWRITE** button appears – click on the button to confirm, or click somewhere else to cancel. You cannot overwrite a factory-configured or locked file.

➢ Click on the **EXPORT** symbol to export the contents of the snapshot or preset. This will allow you to import the file to another D-Mon processor, as described on the next page. The exported file is automatically named and placed in your browser's "Downloads" folder. Snapshots have a file suffix of **.ogf** and preset of **.tnz**. It is important that you do not change the suffix in Windows, as this is how the system recognizes the file types.

➢ Click on the **DELETE** symbol to delete the file. A red **DELETE** button appears – click on the button to confirm, or click somewhere else to cancel. You cannot delete a factory-configured or locked file. Deleting a snapshot or preset will remove it completely from the D-Mon processor's memory.

- Click on **EXPORT ALL** (at the bottom of the page) to export all user-defined presets in the list. The presets are packaged together in a single file. This is automatically named and placed in your browser's "Downloads" folder as before. You can now use the IMPORT function, as described below, to import all presets into the presets list.
- Click on **CREATE NEW** (at the bottom of the page) to save the current settings into a new snapshot or preset. The file appears immediately in the list and is named a date and time stamp. You can now edit the name and other parameters as described on the previous page.
- Click on **IMPORT** (at the bottom of the page) to import a snapshot or preset – a file explorer window appears asking you to choose a file:



In our example, we have a choice of two snapshots (**.ogf**) and one single preset (**Config_2.tnz**). There is also a packaged file (**Configs.tnz**) created from using the **EXPORT ALL** button (described on the next page).

Note that you cannot import a snapshot (**.ogf**) to the presets list OR a preset (**.tnz**) to the snapshots list, so take care to choose the correct file type.

- Click on **Open** to complete the operation – after a successful import you will see the file in the presets (or snapshots) list:

If the file uses the same name as an existing preset (or snapshot), as in our example, then it is a good idea to rename it now!

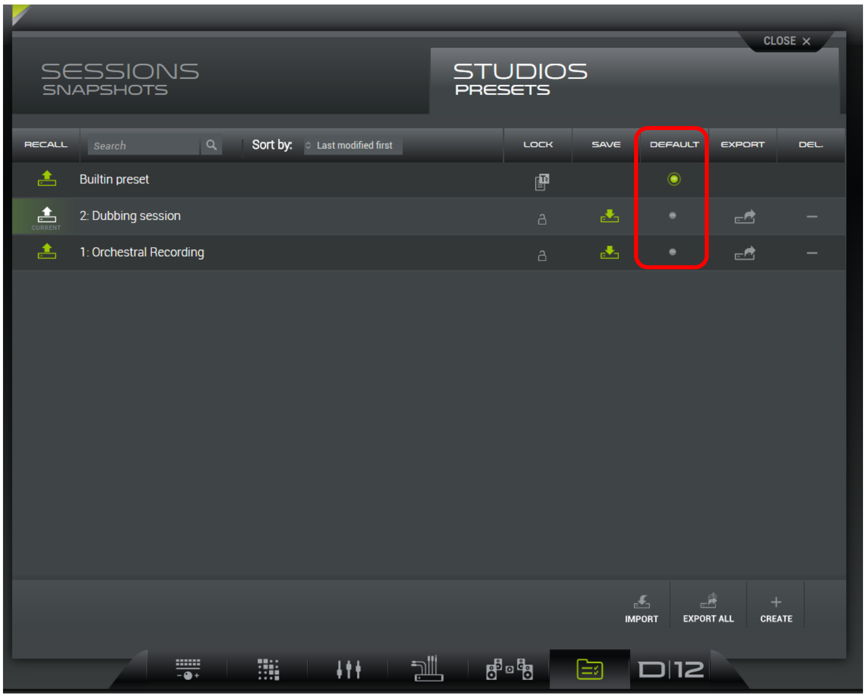
7.6.5 SETTING THE DEFAULT PRESET

There is one extra field in the Studio Presets tab which defines the **DEFAULT** preset.

This is the preset being recalled during boot-up, and it should be used to determine how the processor is configured following a reboot or power off/on.

- Select one of the **DEFAULT** radio buttons to set the default preset – only one preset can be selected at a time.

In our example below, we have chosen the **Built-in preset**:



8 EXTERNAL CONTROL

This chapter covers the additional steps required to control a D-Mon device using either **Avid EUCON** or **ICON** protocols.

Please see *3.4.2: External Controllers* for an overview of all remote control options.

8.1 AVID EUCON

To use an ethernet based control surface, like the Avid S6, an additional software application is required – the "Trinnov" App. The App must be installed on the DAW (Digital Audio Workstation) which is linked to the EUCON control surface. It can be downloaded, for free, from the Trinnov website at:

- **MacOS:** https://updates.trinnov.com/apps/trinnov/macos/current

The App detects which control surface(s) and D-Mon processor(s) are available in the network; and acts as a gateway between the D-Mon side and the control surface side. It also automatically opens the D-Mon Control Panel (DMCP).

Once you have downloaded the App to your DAW, please refer to the instructions and "Read Me" files included in the package.

EUCON libraries must be installed on the computer running the Trinnov app. Best practice is to have the DAW, EUCON and Trinnov App running on the same computer. Ensure that your control surface is up-to-date and runs the latest available firmware, and that your workstation is also running the latest EUCON package available.

The D-Mon is capable of handling all monitoring functions presented by EUCON :

- Main volume with main volume absolute level display
- Mute
- Dim and dim level
- Speaker-set selection
- Talkback activation and Talkback microphone level
- Downmixes and DRC activation
- Per-speaker solo/mute
- Main sources selection*
- Cues (MonA/B/C) master level and its sources

D-Mon processor functions are automatically mapped to the EUCON control surface. A few parameters require a pre-configuration using the D-Mon Control Panel GUI, see *7.4.5: SESSION SETTINGS -> Remote Controllers*:

- Speaker-set, as main or alternate
- Mixbusses, as Cues for MonA, MonB and MonC.

MonD is pre-allocated to control the headphones volume and source, and cannot be changed.

Sources can be created on the fly on D-Mon, and will appear automatically on the EUCON side.

ATTENTION!

With Avid S6 control surface, a specific "intercancel" or "sum" mode exists for source selection. The "intercancel" mode is not compatible with D-Mon, and the S6 monitoring mode must always stay on "sum".

If you wish to listen to only one source at a time, please select the "toggle" mode in the Trinnov interface. Swapping from a mode to another MUST be done at Trinnov interface level.

It is also possible to define custom keys that handle D-Mon specific features, like :

- Preset and snapshot recall
- Bass management
- Bypass
- AES insert
- Group Solo (front/surround/height)
- Talkback 2nd bus

Depending on the control surface, ensure to "lock" the monitoring part to the "Trinnov App", so when you switch to your main audio software the D-Mon is still notified of EUCON events.

8.2 AVID ICON SERIES

Alternatively, an Avid ICON Series control surface, such as the D-Command or D-Control, can be connected. In this instance, no additional software is required.

The D-Mon processor supports the monitoring section of the ICON Series control surface. Therefore, it can be considered a digital replacement for the X-Mon, which is usually provided with these consoles.

8.2.1 CONNECTING THE CONTROL SURFACE

The ICON Series control surface sends out all monitoring data through the RS422 protocol which carries the MIDI commands. The SubD-15 cable is, therefore, not directly compatible with a MIDI connector, and requires adaptive circuitry to meet the electronic MIDI standard which is integrated within the D-Mon processor.

Trinnov Audio manufactures a specific MPIO breakout cable which can be used for connection to the Avid ICON Series 15-pin cable. As soon as the two are connected, you will have immediate control of the D-Mon processor from the ICON console. Also, dedicated connectors are provided for the GPIOs, listen-back input, headphones and talkback signals (so that they may be re-routed to the console, just as in the X-Mon).

ATTENTION!

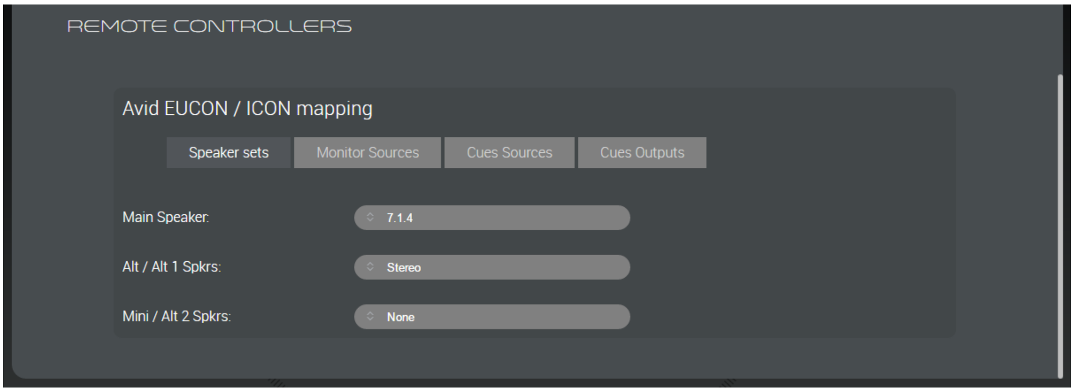
The ICON Series protocol does not accept remote commands via the 15-pin connector. This means that any actions made from the DMCP GUI are NOT reflected on the control surface.

Therefore, best practice is to use only the control surface and close any opened D-Mon interface to prevent direct change via the DMCP GUI.

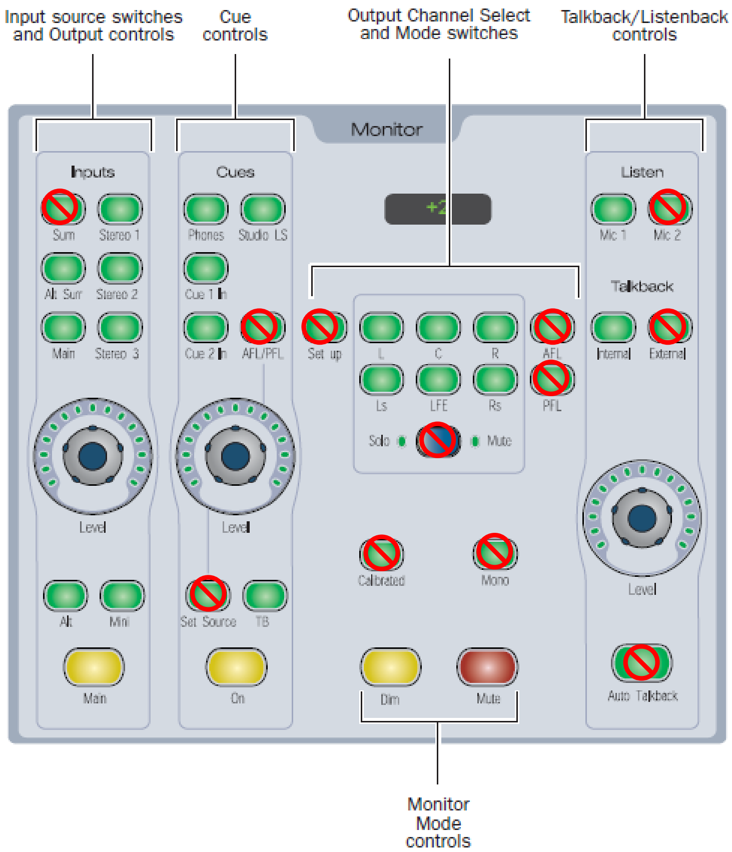
8.2.2 MAPPING THE FUNCTIONS

The mapping of the console's buttons and knobs to the D-Mon processor functions is handled using the D-Mon Control Panel (DMCP) GUI.

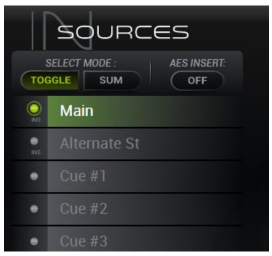
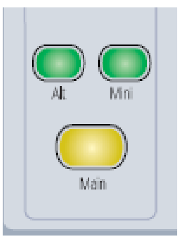
As soon as the D-Mon processor detects the presence of ICON signals on the MPIO connector, it creates a set of parameters in the *SESSION SETTINGS -> Remote Controllers* tab. Thus, you can define the functionality according to the session's needs.

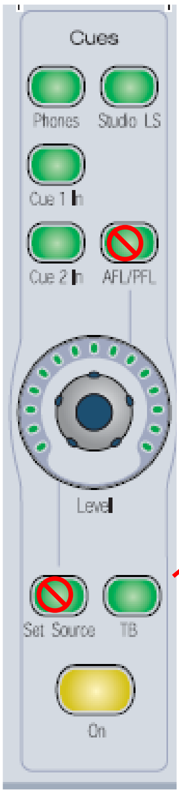


The ICON functions not *directly* supported are marked below with red checkboxes. Most are internal to the ICON and cannot be seen by the D-Mon processor. Some can be achieved indirectly using the DMCP GUI: e.g., **Sum** from the *MONITORING MIXER* tab; **Set Source** using the *SESSION ROUTING* matrix; individual speaker **Solo** and **Mute** from *The CONTROLLER Page*.



The supported functions are as follows:

Icon Control Panel	D-Mon Control Panel	Comments
	 	The **Main**, **Alt Sur**, **Stereo 1**, **Stereo 2** and **Stereo 3** input source select switches can be assigned to any incoming signal from the *SESSION SETTINGS -> Remote Controllers*. Output level is automatically assigned.
		The **Main**, **Alt** and **Mini** monitor output select switches can be assigned to any Speaker Set from the *SESSION SETTINGS -> Remote Controllers*.

<table>
<tr><th>Icon Control Panel</th><th>D-Mon Control Panel</th><th>Comments</th></tr>
<tr><td>


</td><td>

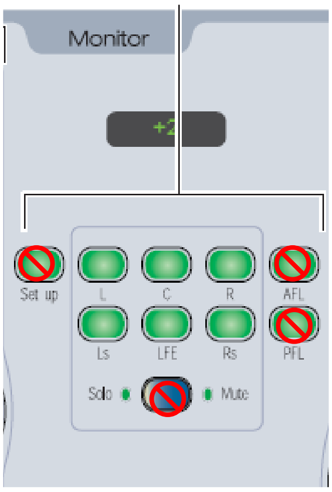

</td><td>

The Phones output is automatically assigned to the headphones.

The Studio LS, Cue 1 and Cue 2 outputs can be assigned from any mix bus using the SESSION SETTINGS -> Remote Controllers tab.

Once an output is selected, level and on/off are automatically assigned to the Level knob and On switch.

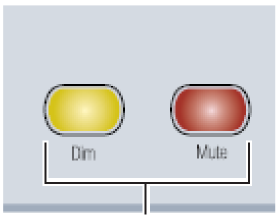
Once an output is selected, the TB switch can talk to the selected output, if TB #1 is enabled for the Mix Bus in the SESSION SETTINGS.

</td></tr>
<tr><td>


</td><td>


</td><td>

The loudspeaker select switches are mapped automatically according to the format of the selected Speaker Set.

</td></tr>
<tr><td>


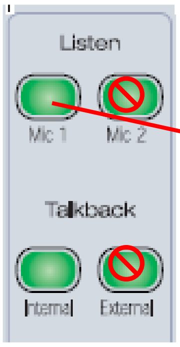
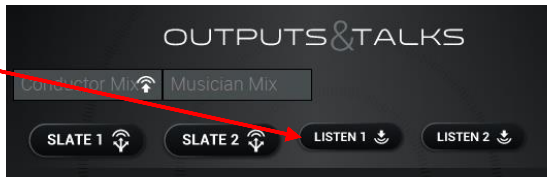
</td><td>


</td><td>

The Mute switch is automatically assigned.

Note that the DIM switch is not transmitted by the console, but the level is adjusted according to the console setup.

</td></tr>
</table>

Icon Control Panel	D-Mon Control Panel	Comments
		The **Mic 1** switch activates Listen-back #1. The Internal/External selection for talkback is not supported.

9 AOIP OPERATIONS

9.1 ABOUT RAVENNA

RAVENNA is a solution for real-time distribution of audio and other media content in IP-based network environments. Utilizing standardized network protocols and technologies, RAVENNA can operate in existing network infrastructures and is inherently fully AES67 and **SMPTE ST 2110**-compliant.

D-Mon integrates a Ravenna software solution built by Merging Technologies, and is fully compatible with ANEMAN to control the stream subscription.

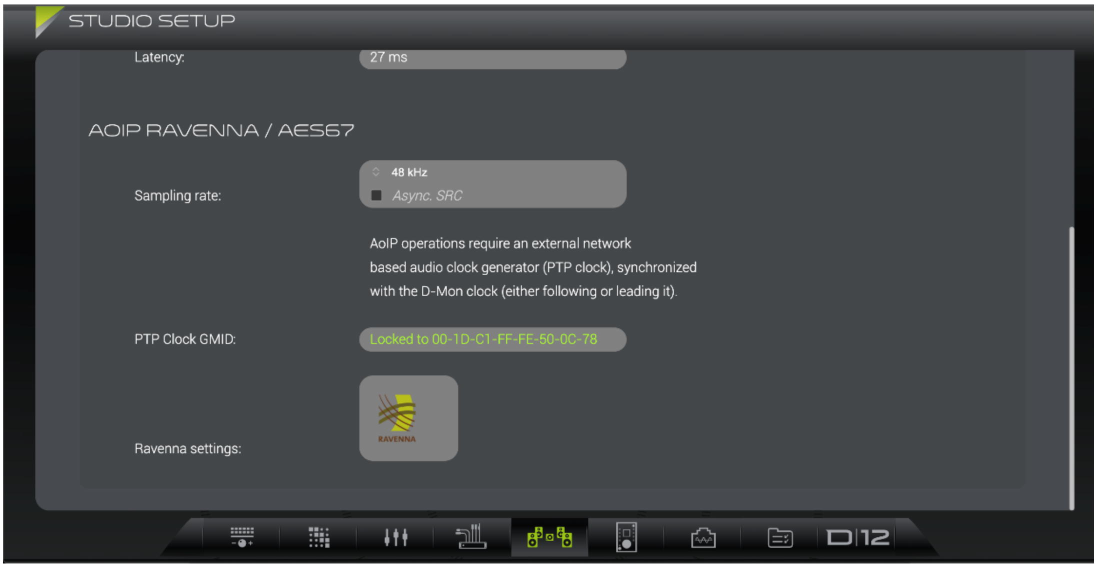
9.2 AOIP CLOCKS

Audio-over-IP is a combination of several software stacks to achieve the global audio exchange over a network. Two elements are at the core: the media data transport, upon RTP (real time protocol); and the clock used to synchronize all the devices, with PTPv2 (version 2 of the precision time protocol, capable of a typical precision around 100ns aka. 1/200 sample at 48kHz). This PTPv2 layer is established from an election process at bootup, where one of the most capable device wins this election and becomes the clock reference, also called "clock leader". It can be identified with the MAC ethernet address of the device ("GMID" for "grand master ID").

> **ATTENTION!**
> A PTP leader clock device is required on your local network in order to be able to use D-Mon audio-over-IP (AoIP).

D-Mon audio-over-IP software solution implements a PTP client software, but it is not able to take ownership of PTP and will always become PTP follower of the leader clock on the network. Therefore, you will need an AoIP device on your local network that can lead the PTP clock (i.e., it could be a digital to analog converter which has AES67 compatibility built-in).

Once a PTP leader clock has been detected and locked to it, the D-Mon interface displays its network MAC address in the STUDIO SETUP -> Audio Clock tab under "PTP clock GDMID" (grand master ID).

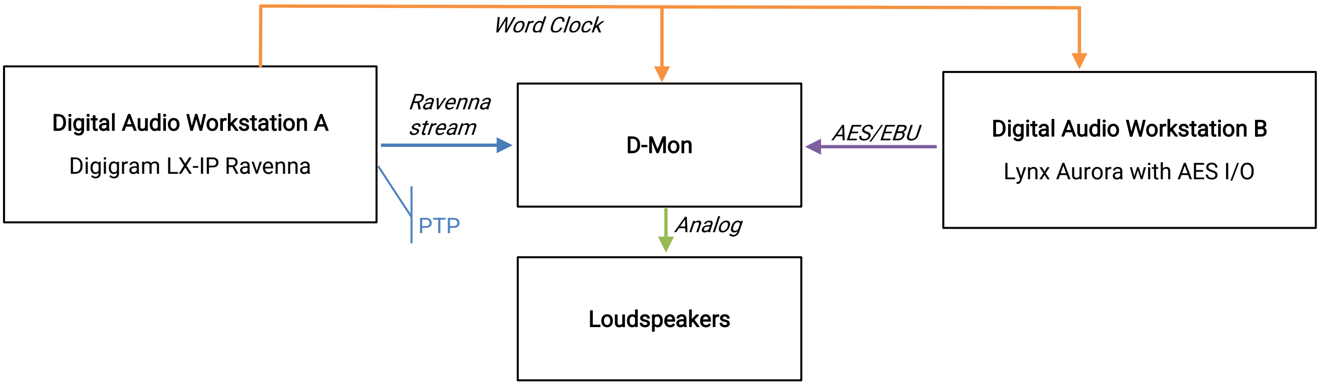


As the D-Mon will also have its own local audio (analog and AES3) inputs and outputs running, **it is required to keep in sync the D-Mon internal clock and the PTP master on the network** in order to avoid audio drops.

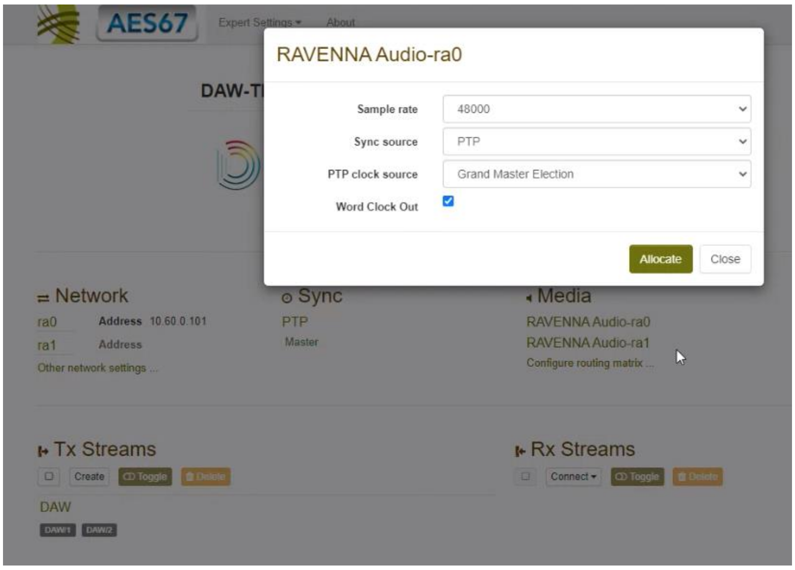
> **ATTENTION!**
> The PTP leader clock device must be physically synchronized with the D-MON, either via AES3 or word clock.

You will need a network PTP leader device that has a physical port that can be used for such a synchronization. D-Mon could be either clock leading or following this network device, via an AES3 (or DARS) link or word clock.

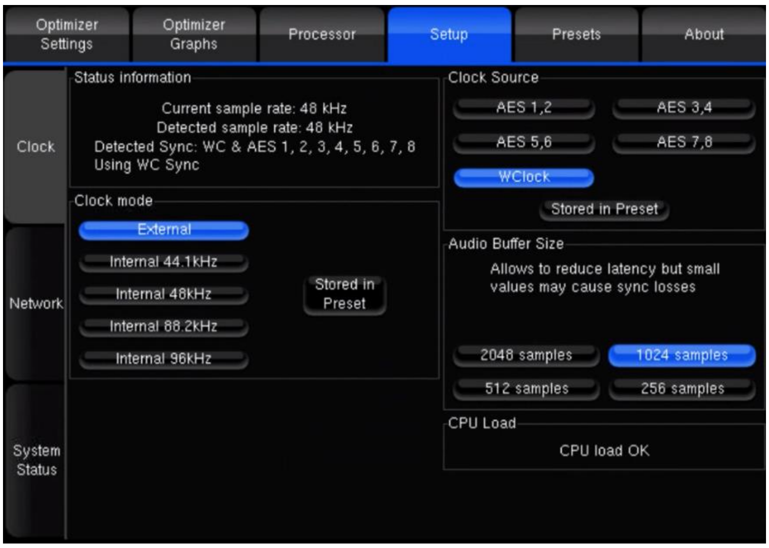
As an example, let's present a working setup, with two DAWs sharing the same D-Mon monitoring and speaker setup.



The DAW A is part of the PTP clock and generates a word clock signal for both the DAW B and the D-Mon.

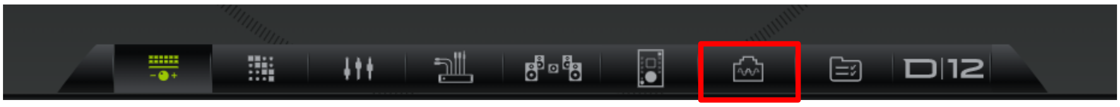


The D-Mon is following the external clock given by the Word Clock, and receive the signal from the DAW B also clocked from this same Word clock, and therefore receive the AES signal in sync.

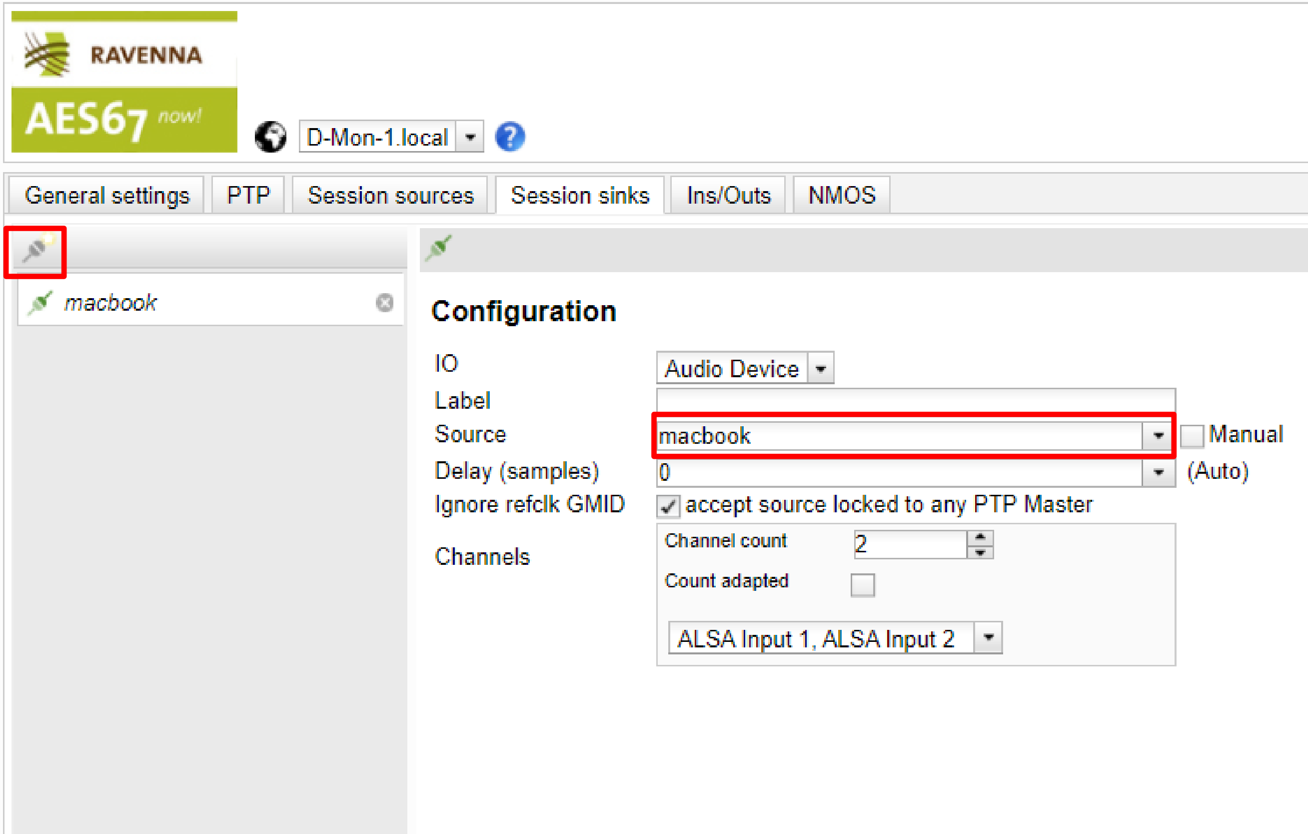


9.3 RECEIVING STREAMS

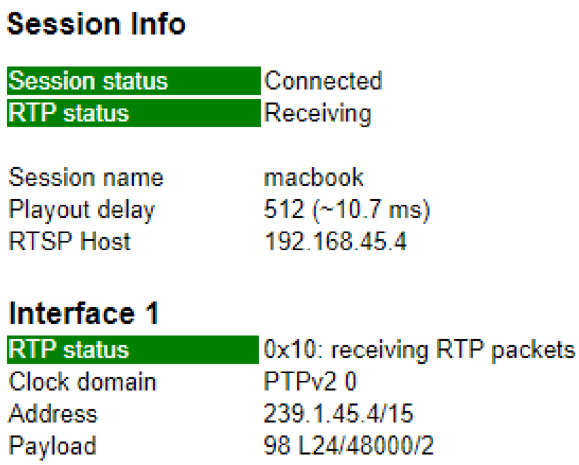
- In the D-Mon interface, under STUDIO SETUP -> Audio Clock, check that the sampling rate matches your current sampling rate and that the PTPv2 clock has been detected and locked.
- open the Ravenna windows, available from the bottom link



- In the Session sinks tab, create a new receiver sink:
  - click on the top left button to add a new stream
  - in the right configuration pane, under "Source", select the source you want to subscribe to. Here, the stream named "*macbook*" has been selected, and the stream details have been automatically retrieved and applied.



➢ Once the stream is established, the sessions info describes the current state

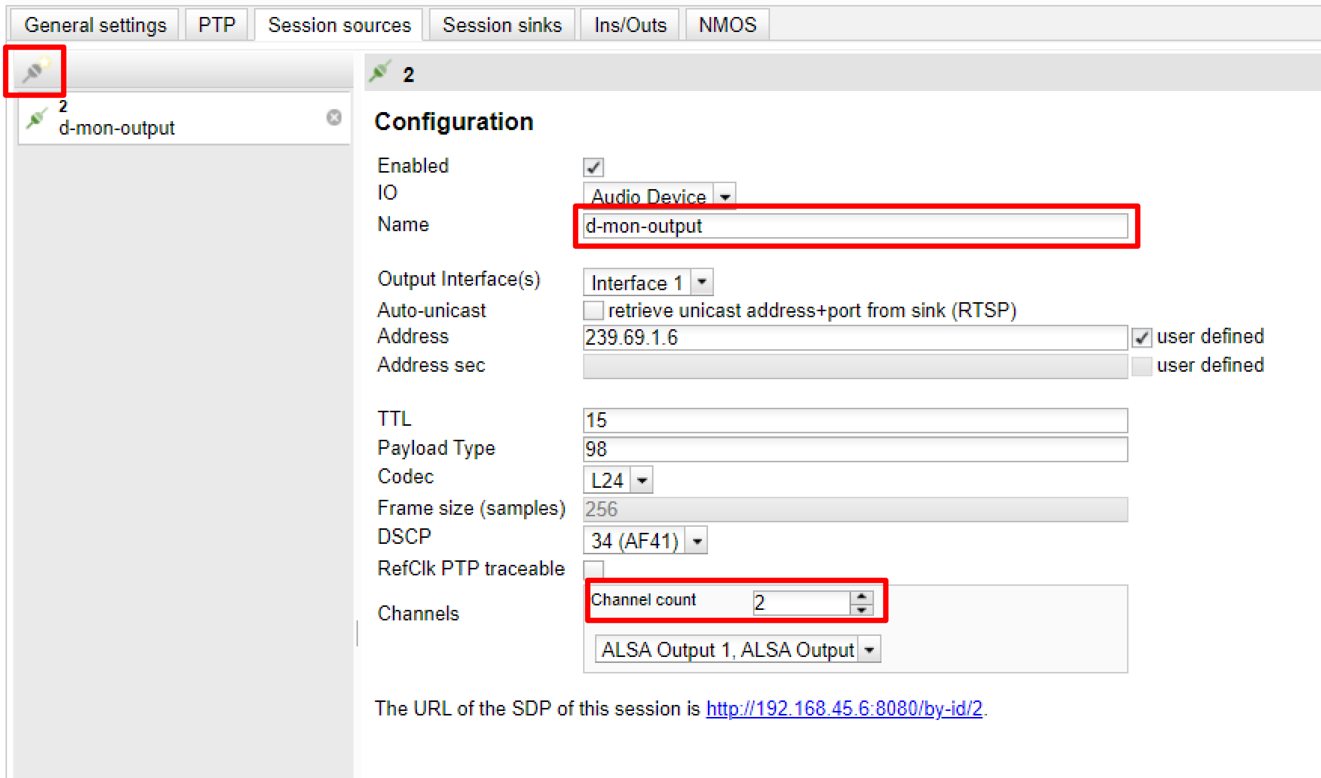


9.4 SENDING STREAMS

➢ Check the same clocking and sampling rate prerequisites as in the above chapter “receiving streams”.

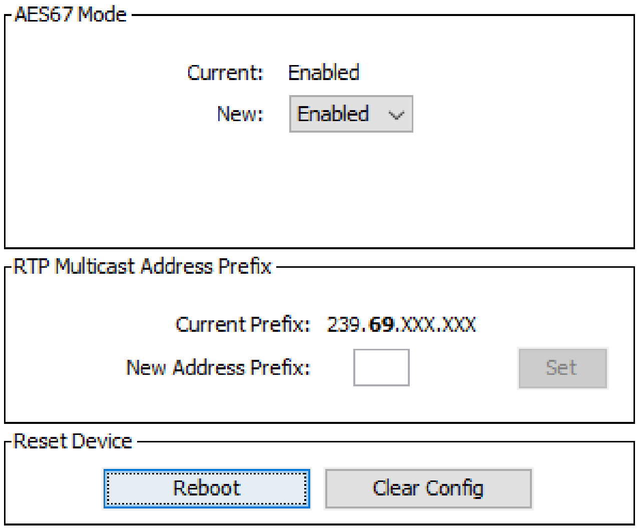
➢ In the “Session sources” tab, create a new emitter source:

   - click on the top left button to add a new stream
   - in the right configuration pane, set a custom “Name” that describes this stream (validate this name by pressing “enter”), and adjust the desired number of channels.

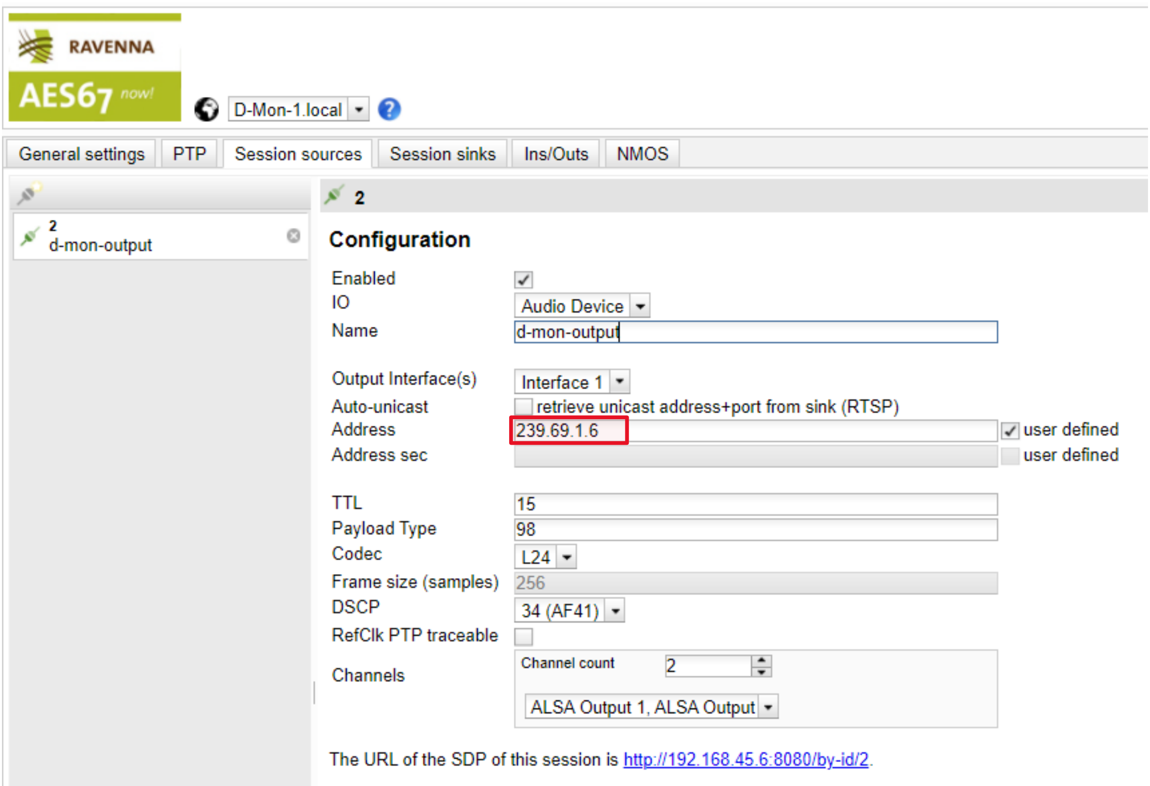


9.5 SENDING TO DANTE DEVICE

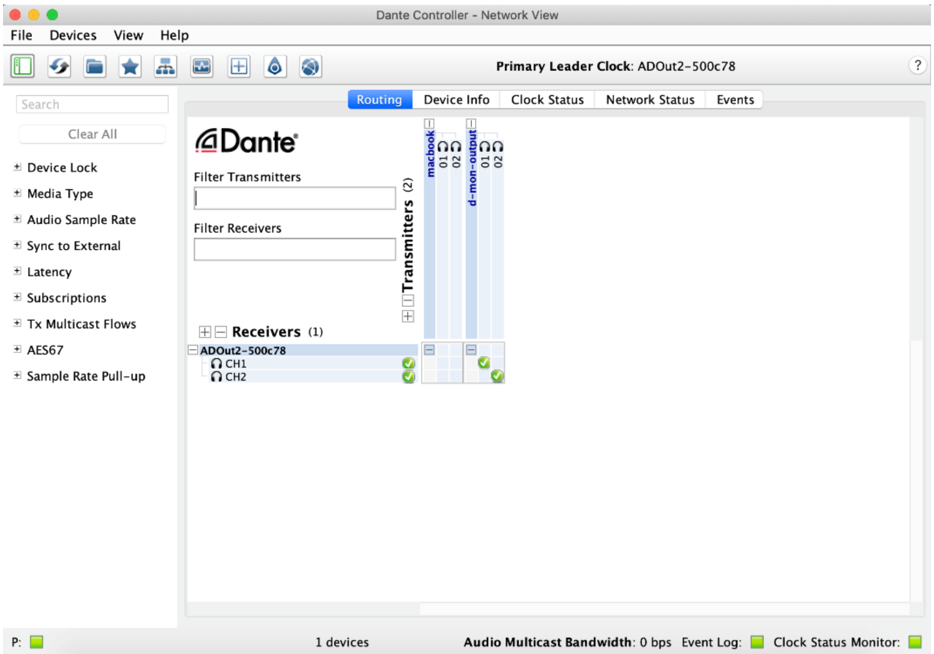
- In the Dante Controller, select your emitting device properties and make sure it has AES67 compatibility mode enabled and take note of the RTP multicast address prefix



- Create an output stream on the D-Mon Ravenna interface, like in the above chapter, set a custom name, adjust the number of channels,
- Adjust the first two numbers of the “Address” fields so that it starts with the same previous multicast address.



➢ In the Dante Controller, you can now see the D-Mon output, you can patch-it to your desired target(s).

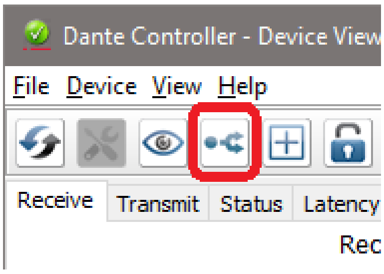


➢ In the D-Mon interface, you can now patch a source to this newly created output.

9.6 RECEIVING FROM DANTE DEVICE

➢ In the Dante Controller, select your emitting device properties and enable the AES67 compatibility.

➢ In the Dante Controller, select your emitting device properties and create a multicast flow on the desired channels



For more details, refer to the Dante user guide.

➢ You can now subscribe to those channels from the Ravenna interface, the stream is announced as SAP (Cf. above chapter “receiving streams”)

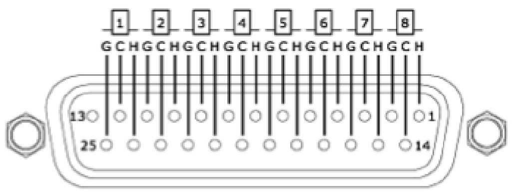
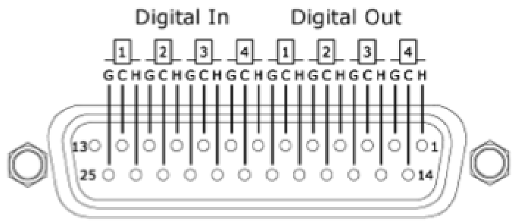
➢ In the D-Mon interface, a new source is automatically created and you can now select it in the monitoring controller

10 APPENDICES

10.1 WIRING DIAGRAMS

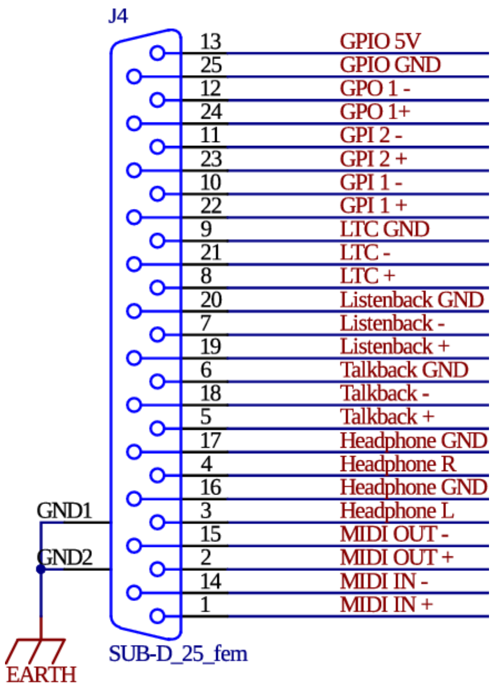
10.1.1 AUDIO SUBD-25 CONNECTORS

SubD-25 audio connectors are compliant with the **TASCAM** standard:

Analog connections	AES3 Digital Connections
	
G = Ground; C = Cold (–); H = Hot (+)	

10.1.2 MULTI-PURPOSE IN/OUT (MPIO) SUBD-25 CONNECTOR

This connector is the same for all D-Mon models. It handles the following signals:



2 x GPI (General Purpose Inputs)

Used to trigger events in the D-Mon processor from an external console switch or footswitch. See *STUDIO SETUP -> Remotes* for functionality.

1 x GPO (General Purpose Output)

Used to trigger an on/off event from the D-Mon processor.

LTC (Linear Time Code)

Compliant with SMPTE standard (balanced).

Talkback & Listen-back Inputs

Balanced inputs with 36V phantom power. These two inputs are wired, via a relay, to the same analog inputs as #1 and #2 on the related SubD-25.

Stereo Headphones

Unbalanced output. This output is wired, via a relay, to the same analog output as #9 and #10 on the related SubD-25.

Midi Input and Output

See the *MIDI Implementation* for details.

10.2 MIDI IMPLEMENTATION

Standard MIDI Control

The D-Mon processor can be remotely controlled using standard MIDI commands sent to the MIDI input port of the MPIO connector. Simple commands such as volume control changes and note on/off are used.

This allows the basic monitoring functions to be controlled by dedicated hardware on nearly any MIDI-equipped device.

The available commands are:

Function	Midi Channel	Message type	Parameters 1	Parameters 2
Main speaker set volume	1	Control change (0xB0)	Volume (0x07)	requested level
Headphones volume	2	Control Change (0xB1)	Volume (0x07)	requested level
Talkback volume	3	Control Change (0xB1)	Volume (0x07)	requested level
Listenback volume	4	Control Change (0xB1)	Volume (0x07)	requested level
Speaker Mute	1	Note On (0x80) Note Off (0x90)	key B6 (0x5F)	velocity ignored (must be > 0)
Speaker Dim	1	Note On (0x80) Note Off (0x90)	key A6 (0x5D)	velocity ignored (must be > 0)
Talkback	1	Note On (0x80) Note Off (0x90)	key #A6 (0x5E)	velocity ignored (must be > 0)
Talkback 2	1	Note On (0x80) Note Off (0x90)	key #A7 (0x6A)	velocity ignored (must be > 0)
Mono	1	Note On (0x80) Note Off (0x90)	key #G6 (0x5C)	velocity ignored (must be > 0)
Diff Mono	1	Note On (0x80) Note Off (0x90	key #B7 (0x6B)	velocity ignored (must be > 0)
Source selection	1	Note On (0x80) Note Off (0x90	key #A1 (0x21) to key #G2 (0x2C)	velocity ignored (must be > 0)
Speaker selection	1	Note On (0x80) Note Off (0x90	key #A2 (0x2D) to key #G3 (0x38)	velocity ignored (must be > 0)
Source selection mode	1	Note On (0x80) Note Off (0x90	key #C8 (0x6C)	velocity ignored (must be > 0)
Headphone mute	1	Note On (0x80) Note Off (0x90	key #C8 (0x6D)	velocity ignored (must be > 0)

The levels controlled by MIDI volume control changes are described by their second parameter according to the following table:

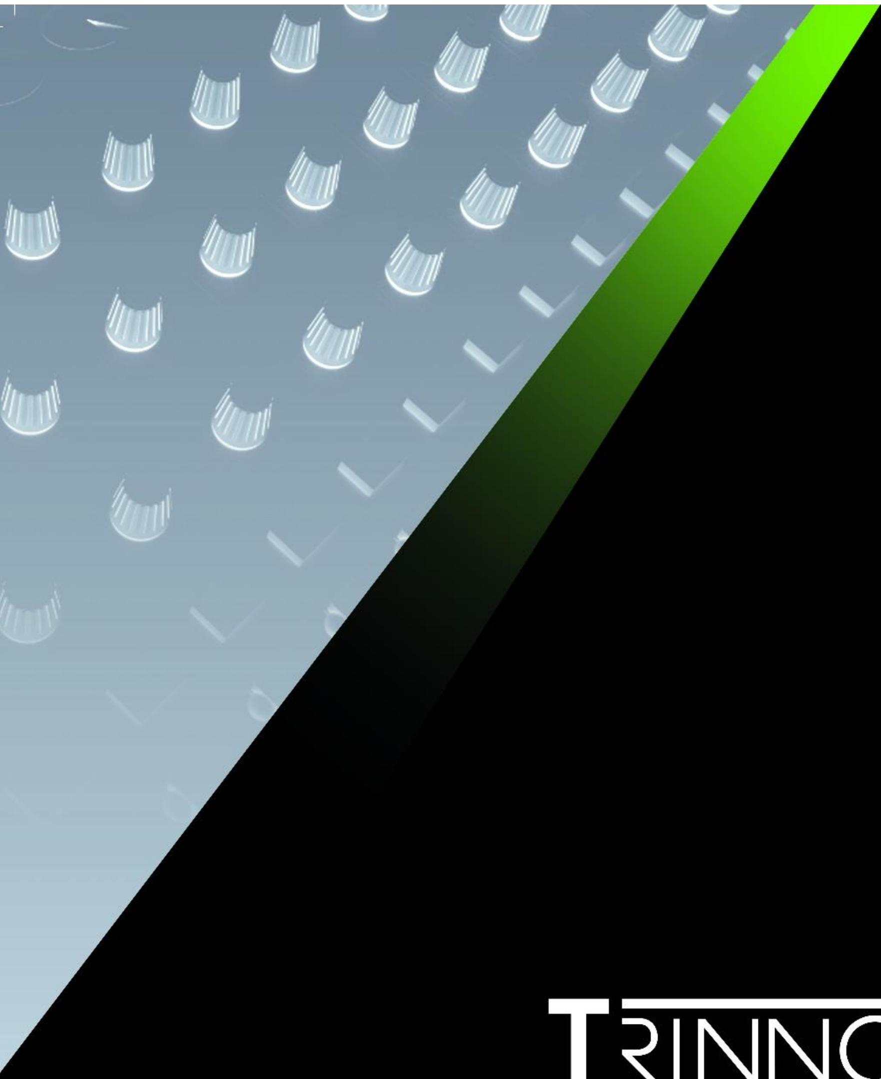
Meaning	Parameter 2 Value (decimal / hexadecimal)
OFF	0 / 0x00
-95 dB	1 / 0x01
+ 0. dB	96 / 0x60
+ 30 dB	126 / 0x7E
+ 31 dB	127 / 0x7F

In order to avoid midi messages saturation, a loopback detection feature is present. It sends at a regular time interval a heartbeat as NOTE OFF on channel 1 / key #C4. If such a loop is detected, all incoming midi commands are discarded by the D-Mon and a warning icon with a specific message is displayed in the settings page, under the remote controller section.

10.3 TECHNICAL DATA

10.3.1 MECHANICAL

- **Weight** = 13 kg / 28,60 lbs
- **Height** = 87,5 mm / 3,48 inches (compliant with 2U space rack)
- **Width** = 427 mm / 16,81 inches (compliant with 19’’ space rack)
- **Depth** = 410 mm / 16,14 inches (not including any additional connector)



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