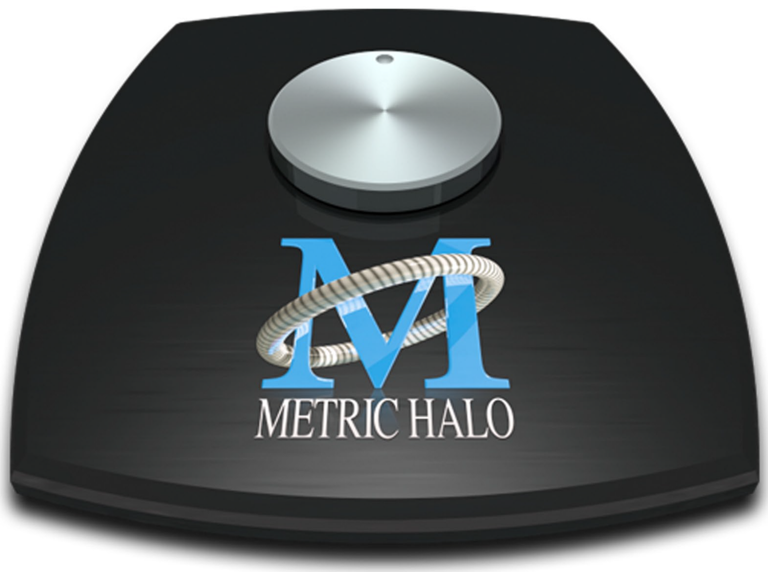

Metric Halo 3d
Users Guide

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Metric Halo

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Table of Contents

List of Figures

Welcome to Metric Halo 3d!

On behalf of everyone at Metric Halo, thank you and congratulations on your decision to bring Metric Halo 3d into your creative process. Whether you are upgrading a "2d-style" Metric Halo box or starting fresh with us, we remain resolved to pushing the boundaries of musical transparency and technological innovation while maintaining the highest levels of customer service.

As audio professionals, we have been assaulted with both diminishing budgets and diminishing deadlines. 3d is here to help on both fronts. 3d has been developed in direct response to the fundamental real-world problems that audio professionals are faced with every day - not as a band-aid, not as an alternative work-around, but as an overall solution to those problems.

As an upgrade to your prior Metric Halo investment, 3d provides a ridiculous boost in functionality to every box we've ever made while at the same time being much easier to use. For both prior and new owners, the higher level of musical transparency and sheer audio quality 3d offers, combined with its ease of use, dynamic expandability and no-limits routing and processing power will make all aspects of your creative process not just better-sounding, but more efficient as well.

That's the design philosophy behind 3d as a whole. Metric Halo's 3d developmental stomping grounds include professional installations working at the highest levels of:

- multi-room professional recording facilities
- analog and digital tape master archival / preservation transfers
- immersive-format orchestral recording
- acoustical analysis and correction
- on-location performance recording
- high-resolution music mastering and restoration
- virtual reality soundscape design and capture
- production audio, music and effects for film and broadcast, and
- FOH and monitor mix from local events to large arenas, with performance recording

...to name just a few.

The brutal vetting endured in these disparate workflows and the (often merciless) scrutiny of the veteran engineers and musicians submitting their reports from the field has resulted the audio production platform you now have in your hands. Quick setup, less time spent "nailing the sound", otherworldly spatial representation (that "you are inside the music" feeling), a remarkable reduction of ear fatigue over long sessions... these are the kinds of comments our customers bring back to us that tell us we're on to something special.

Welcome to the Metric Halo family. Let us know how it goes!

Introduction

The following is a "Start Here" readme for those of us who prefer to just connect a new gadget and start exploring right away. Unintentional cliché aside, 3d really does add a new dimension beyond yesterday's traditional audio interfaces and audio networking solutions. The difference with 3d is, it's as easy as you want it to be but as powerful as you need it to be.

So, here's this whole MHLink thing in a nutshell:

When you connect one 3d box to another 3d box with MHLink, you get a bigger box. That's what happens. That's it. If you stop reading here, you've got the basic concept.

Connect via MHLink and start listening. Live, while running audio through the first box, connect a second box to the first, power it up and get double the DSP processing and immediate access to all the added audio I/O with no added latency and *with no interruption of your currently-playing audio*.

The new box automatically shows up in the MIOConsole3d System Status, and all new ports instantly appear in the routing windows. No glitches, no manual registration, literally plug and play.

Your computer and DAW still see them as "one box", even if you daisy-chain ten of them. Plus, since they're running on Gigabit Ethernet, your "one box" can include analog and digital I/O in another room or on a stage 100 meters away. Since all Gigabit Ethernet jacks are transformer-coupled, all connections are galvanically isolated - which means no ground loops, wherever you want to plug them in.
Try *that* with USB3 or Thunderbolt.

The way MHLink works is very simple: The computer is connected to one 3d device - the first box in the MHLink chain. We call this the **Root** box. All other 3d boxes are connected, in turn, from the Root box.

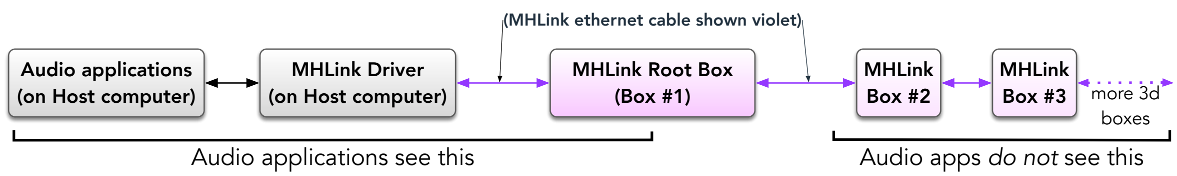


Figure 1: MHLink connections

The Root box is the only thing the computer can see, and it is the Root box that manages all other 3d boxes in the MHLink chain. So, as far as the computer (and your DAW) is concerned, there literally is only ever that one physical audio device. Everything that happens on *the other side* of the Root box is managed entirely by the Root box itself, and remains hidden from the computer operating system and your DAW. Since the routing management system in every 3d device is smart enough to manage its own domain, any 3d device can serve as a Root box at the full 128 channel bandwidth and 192kHz max sample rate.

New boxes synchronize their DSP and I/O, and lock to the system clock of the Root box automatically upon connection. Adding boxes does not add system latency - ten boxes act and respond the same as one. MHLink provides more channels at higher sample rates with less latency (and *way* less setup) than audio-over-IP solutions. ...And of course, truly superior sound quality all the way through.

There is a lot more to love than just ease of use, and the next few pages will help you get a handle on how it all comes together. So even if you are a single-box user, please take the time to read through this brief summary of terms to take best advantage of your 3d system right from the start.

It is a bit of an info-dump and we apologize for that, but keeping these concepts in mind when connecting the boxes and clicking around MIOConsole3d will definitely help get you into your comfort zone much faster.

Many thanks to the hyper-talented and immensely diverse Metric Halo beta team for helping us to compile this pre-emptive FAQ.

Decoding 3d: Important Terms and Concepts

Part 1: Hooking things up

- **MHLink** - "MHLink" is the proprietary state-of-the-art Metric Halo technology which uses Gigabit Ethernet as a communications backplane to automatically synchronize and unify multiple 3d boxes and present them to the computer as a single audio device. Unlike AVB and AES-67, MHLink is not an IP-based network protocol, so your computer will not see MHLink boxes as typical network devices.
- **Daisy-chain** - Each 3d box has two MHLink ports on the rear panel. The correct method for connecting multiple boxes is in series, or "daisy-chained". This is essential for the proper inter-communication and synchronization of all the connected boxes. Adding boxes to the chain does *not* add latency to the signal path.

  MHLink is a fully dynamic protocol, specifically designed for 3d boxes to be added and removed at the end of the chain while the system is live and in use, with no interruption to existing audio paths.
- **MHLinkDriver** - This software driver is required for your computer to recognize, synchronize and communicate with MHLink boxes connected to your Gigabit Ethernet port. This is a standard driver installation requiring a reboot of the computer to complete the installation. From the MIOConsole3d application "I/O" menu, the MHLinkDriver can be set to allocate 2, 8, 16, 32, 64 or 128 channels into and out of the computer at all sample rates up to 192kHz. The default setting is 32 channels of I/O.
- **USB** - A single USB-C port is provided on all 3d devices as a secondary, cross-platform compatible alternative computer connection port. This port supports the UAC2 class audio driver built into all USB2-compliant devices.

  3d USB connections can provide up to 48 channels of I/O at 44.1 and 48kHz, 24 channels at 88.2 and 96kHz and 12 channels at 176.4 and 192kHz, user-configurable within the MIOConsole3d software. Most modern macOS, Windows 10, Linux, iOS and Android devices are capable of hosting a 3d USB connection. The current 3d USB interface is limited to single-box operation only.

  Note: Connecting to USB does not require the installation of a software driver. See MIOConsole3d Preferences > Discovery before connecting USB to a computer which is also connected to an MHLink ethernet interface.
- **MHLink and shared IP networks** - MHLink requires a minimum of Gigabit Ethernet over CAT5e or higher spec ethernet cable. 10/100Base-T Ethernet is not supported. The preferred method of connection to a computer is to dedicate a Gigabit Ethernet port exclusively for MHLink (ie. not on a shared network). While not every Gigabit Ethernet adapter on the market has been tested, generally the current crop of USB3 and Thunderbolt Gigabit Ethernet adapters work well.

  That said, MHLink should work just fine on a basic high-quality Gigabit Ethernet switch alongside common IP (Internet Protocol)-based network traffic. As you might expect, this configuration introduces a lot of extra variables, and is discouraged for critical audio workflows demanding high channel counts and/or high sample rates.

  MHLink will not pass through Ethernet routers or WiFi.

  > When connecting MHLink through a shared network, any computer on that network which has the MHLinkDriver installed will be able to see those boxes as available audio devices. This should be avoided as it creates the possibility of two computers trying to control the same boxes at the same time. Connecting MHLink to a dedicated Gigabit Ethernet port is preferred.

- **MHLink Domain** - An "MHLink Domain" is any Metric Halo 3d box or daisy-chained group of boxes connected to your computer; the boxes in an MHLink Domain have one single audio clock shared between the units. Essentially, *a Domain IS the audio device...* any boxes you add to the MHLink chain simply make that audio device bigger.

  Domains are displayed in the System Status Pane at the left side of the MIOConsole3d UI. MIOConsole3d controls all Domains attached to the Host computer as separate audio devices, each operating with its own system clock and driver channel count, with fully independent routing, mixer desk, cue and monitoring functions.

  The graphic below illustrates how domains can connect to a Host computer. A single box connected via USB = a Domain. Four boxes daisy-chained together on a dedicated Gigabit Ethernet port = a Domain.

  Note that there are two domains hanging off the Gigabit Ethernet switch (outlined in red). When you attach a box (or boxes) to separate ports on a Gig-E switch they register as different ethernet hardware port addresses, so each port will register as its own domain.

  Future 3d updates may include enhancements to support specialized scenarios involving multiple domains. Much of the information provided here is so you will understand what is going on as you explore 3d and MHLink.

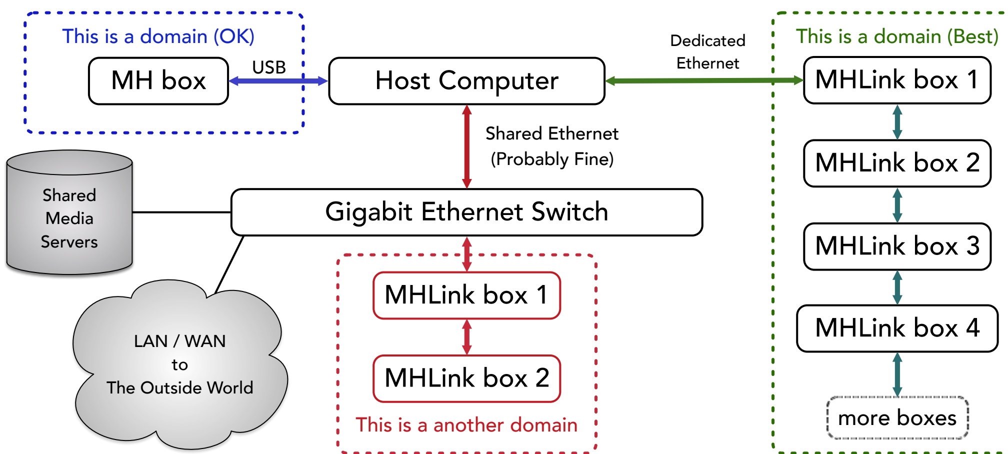


Figure 2: MIOConsole3d Domain configurations

- **Best - Dedicated Gigabit Ethernet port:** (Shown in Green)
  Full support of multibox MHLink domains with the highest channel counts and greatest flexibility.
- **OK - USB:** (Shown in Blue)
  Single-box operation with limited channel count. Cross-platform (Mac, Windows 10, Linux, iOS & Android) connectivity. Performance subject to variables in USB bus activity, host computer OS, software and hardware support, and adapter interconnects.
- **Probably Fine - MHLink and IP network connection shared through a Gigabit Ethernet Switch:** (Shown in Red)
  Performance subject to network traffic and configuration/capability of the Gigabit Ethernet switch. May require adjustment of I/O channel count when streaming audio or video across the network.

Please Note: *Only one instance of MIOConsole3d is allowed to be running under any circumstance where more than one computer can see the same boxes, whether connected by Gigabit Ethernet or USB.* More than one instance of MIOConsole3d would create a conflict where two computers were trying to control the same box at the same time. While this will not damage anything, it would require quitting the offending copy of MIOConsole3d, and might also require a power cycle of the Root box to clear the conflict.

Part 2: Operations

- **MIOConsole3d** - The MIOConsole3d application is your command and control interface to all the 3d boxes visible to your computer through network and/or USB connections. When you launch MIOConsole3d, the program scans for all MH 3d boxes and identifies and synchronizes each box into its Domain. The MIOConsole3d application does no audio processing; it is the controller for the audio routing and processing which is all executed in the DSP and FPGA processors in the hardware boxes. Any processor inserts you run from within the MIOConsole3d are running on the boxes, not on the computer. All MIOConsole3d processors are zero-latency.

- **the Host** - The computer which is directly connected to the MHLink Domain and runs MIOConsole3d is called the "Host". The Host computer sees each MHLink Domain (via AudioMIDI Setup or the Sound Preference Pane) as a single audio device.

  Future 3d updates may explore supporting specialized scenarios involving shared or split Hosting between MHLinked computers, but for the time being there can be only one.

- **Root box** - The Metric Halo 3d box which is directly connected to your Host computer is called the **Root** box for that Domain. The **Root** box is always Box #1 in the Domain list and is responsible for managing communications between the MIOConsole3d application on the Host computer and all the boxes in the Domain (see the "MIOConsole3d Domain configurations" graphic on the previous page). *All audio to and from the Host computer for all boxes in that Domain is routed through the Root box.*

  With that in mind, as long the Root box stays connected to the Host, the MHLink architecture allows the addition and removal of daisy-chained boxes on a live MHLink domain even while audio is running. New boxes added to the end of the chain will automatically join the domain and appear in the MIOConsole3d mixer with new input strips ready to be routed. Boxes you remove will show as "Offline" in the MIOConsole3d Domain Status window and can be removed by right-clicking and selecting "Delete Offline Unit".

- **Routing** - The Routing window is arguably the most important single interface to be comfortable with in the Console. MHLink Domains combine many individual boxes transparently into a single unit for the Host. Each box has an array of physical analog and digital inputs and outputs, 64 bus channels plus the 128 channel routes from and to the Host computer, easily making hundreds of audio routes available at any given time. The Routing window interface includes 'sorting' filters to bring this power under control. All routes in the Routing windows are listed by box number in the daisy-chain, the box model (2882, ULN8, etc.), the type of interface (Analog, ADAT, AES, Host etc.), and the channel number of that interface on that box. In the Routing window you can click *"All Units"* to show the Routing options for all of the boxes in the Domain, or just one box. You can filter your options further using the selection categories to the left. Note that output (or return) routes can only be assigned from one source. All available routes will be presented as black and selectable in the routing window. Unavailable routes are shown as "[Unrouted]".

  Cascade is a feature common to some DAWs and digital mixing consoles which applies sequential assignments across many channel strips at once to speed mixer setup. *Cascade* always starts at the leftmost selected strip and increments from left to right, the same as you may have seen in other audio environments. In MIOConsole3d, *Cascade* is further refined by applying the selection filters in the Routing window. *Cascade* will take the filtered list on the right pane of the Routing window and apply that list sequentially to either selected range or a set number of strips. If *Cascade* is set to 24 strips, the routes shown in the Routing list will be assigned to the 24 strips to the right of the first selected strip. Note that Adhoc groups are fully supported by *Cascade*.

  Cascade Hints: When using *Cascade*, it is highly recommended that you always scroll down and verify your routing targets *before* hitting *"Apply"* to commit the new routes. To remove routes from multiple strips at once, select the target strips, open the Routing window, scroll to the top and select *"None"* at the top of the selection list, then hit *"Apply"*.

Part I. Quick Start Guides

Table of Contents

1. ULN-8 Quick Start Guide

Figure 1.1: ULN-8

Prepare the unit for use

Unpack the ULN-8 and make sure all the parts are there:

- One Mobile I/O ULN-8 unit
- One IEC Power Cord appropriate for your area
- One 18-volt 60-watt world-ready external power supply
- One 12” CAT5e Ethernet Cable
- One 14' CAT5e Ethernet Cable
- One USB-A to USB-C Cable
- Two Rack Ears w/ fasteners
- Rubber feet
- Warranty/Registration Card

Welcome to your new Metric Halo ULN-8 3d! Once you’re finished checking the box physically and installing the rack ears (or rubber feet), connect the power supply. Now connect your input and output cables along with your monitors and signal sources. Your monitors should be connected to Analog Out 1/2, either via a DB25 cable or the 1/4" TRS jacks. Turn the unit on using the front panel switch.

> ***Super QuickStart!***
> Plug your computer into the USB-C port on the back of the ULN-8.
>
> Select “ULN-8” as your computers’ audio device.
>
> Done.
>
> The ULN-8 will receive and send audio from/to channels 1 through 12 of your USB host device at all resolutions up to 192kHz/24 bit. Audio channels 1 and 2 of your host device will play audio out both the headphone jack and the analog channels 1 and 2 jacks and DB25 analog out on the rear panel.
>
> *Note:* While this is indeed the fastest way to get audio going, it is also the most boring. Connecting your Host computer through MHLink is by far the more powerful and expansive interface (and certainly the preferred method), but rest assured the USB connection will take on a much more interesting role later in the Advanced section. Please, do read on.

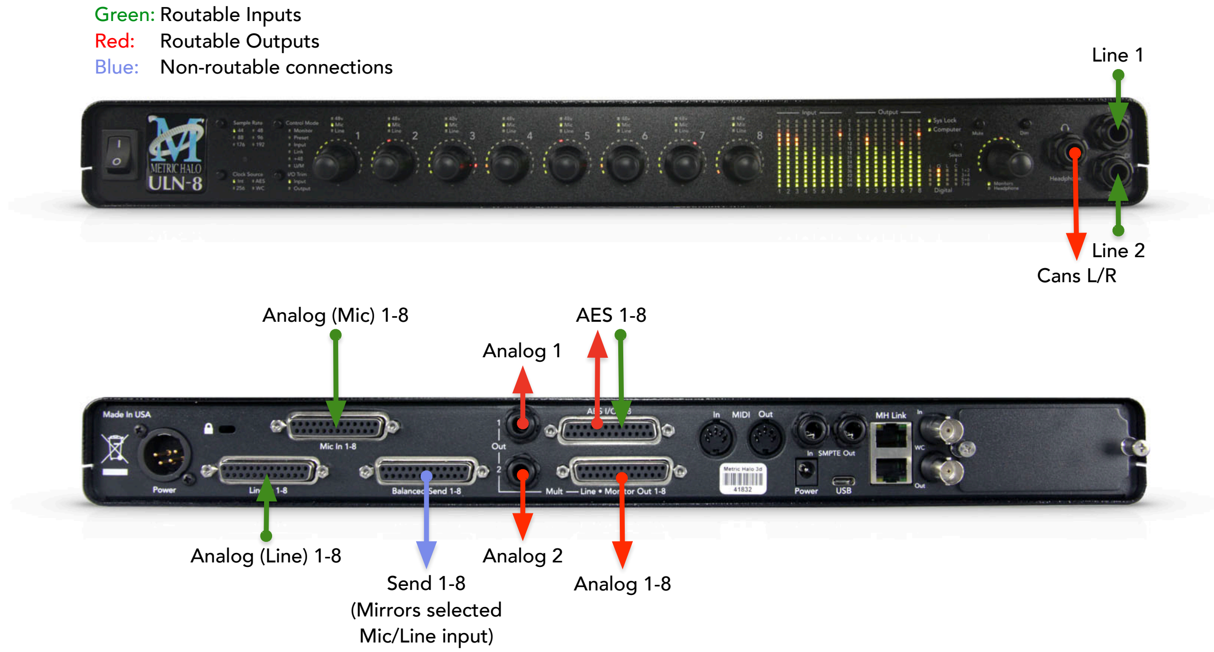


Figure 1.2: ULN-8 Routing

Connect the ULN-8

Install the latest MIOConsole3d installer package from: macOS MIOConsole3d installers (this will require a restart). Please see the section on software installation if you need clarification.

> **Driver Installation on macOS 10.13 and newer**
> With High Sierra (macOS 10.13) Apple introduced another layer of driver (system extension) software security.
>
> In order for third-party drivers (system extensions) to be activated on new installs of High Sierra, the user of the computer (you) must explicitly approve the activation, after the installation has occurred.
>
> You must do this before you will be able to use your hardware via MHLink.
>
> The authorization process is the same for all third-party drivers. If you are unfamiliar with the process, please read our FAQ answer here: macOS Driver User Authentication.
>
> Additionally, with macOS 10.14 and 10.15, you must select “Yes” when asked to give MIOConsole3d permission to access the “Microphone”. If in doubt, double-check this setting: go to the Apple menu: System Preferences.../Security & Privacy pane, select “Microphone” in the column to the left, and make sure that MIOConsole3d is checked.

Connect an ethernet cable between the ULN-8 and an available ethernet port on your computer, then go to the System Preferences and select “MHLink Audio” as the system’s sound input and output.

Get familiar with the front panel

Look at the LED under the Monitor Control encoder; the “Monitor Control” LED should be illuminated. If “Cans” is lit, push the encoder to switch it (all encoders have shaft buttons). The LED should be green; if it is yellow, push the “Dim” button. If it is red push the “Mute” button. Now that you’re sure that you’re looking at the Monitor Controller and that is not dimmed and unmuted, turn it down. The meters are now showing the gain value of the Monitor Controller. This will happen any time you adjust an encoder.

Take a listen

With MHLink, you need to launch MIOConsole3d to establish the routing between MHLink and the Analog Outputs on the box. The routing is established automatically as soon as you launch MIOConsole3d.

If you connect with USB, you don't need to launch MIOConsole3d to establish the output routing; it is setup automatically by the box itself.

Note: The ULN-8 default configuration is set to receive audio from channels 1 and 2 from your computer, and play audio out both the headphone jack and analog out channels 1 and 2 from the db25 and 1/4" TRS jacks on the rear panel.

Using Legacy Interfaces

If you wish to compare your new 3d interface with an older “2d” Firewire MIO interface, you can leave it connected as always and run them both at the same time.

All FireWire-based Metric Halo interfaces will operate as fully independent audio devices alongside your Metric Halo 3d Ethernet-based interfaces.

MIOConsole3d

Figure 1.3: MIOConsole3d ULN-8 default view

MIOConsole3d Default Window Layout

At first launch of MIOConsole3d with your ULN-8 you will be presented with the default layout shown above.

There are a few important things to look at here so let's take a few minutes to familiarize ourselves with the interface/GUI.

First of all, if the Mixer window takes up too much room or appears too small on your screen, select Mixer/Mixer UI Scale from the menu bar and adjust the visual scale of the mixer surface. Note that scaling does not change the size of all of the Mixer strip elements - the fader/meters section remains elastic and will shrink or stretch as other Mixer elements are resized, added or removed. Try selecting the 85% scale setting so you can see what it does.

Now resize the Mixer window by grabbing the window corner (or any of the edges) so you can see how the mixer elements react.

You can also change the width of mixer strips (wider strips are helpful to better see 8-channel surround meters) and show or hide any mixer strip UI element. We will explore these features in more detail below.

As you have probably noticed as you check things out in the MIOConsole3d, most elements in the Mixer, Monitor Controller and Cue Controls windows have mouse-over pop-up hints which display expanded details for the button beneath.

The regular mixer surface button labels will often be abbreviated to save space, but the mouse-overs will display all the details of each control setting. This is an especially important feature as you add more boxes to your system domain.

1. In the upper left side of the main mixer window you see the System Status Pane.

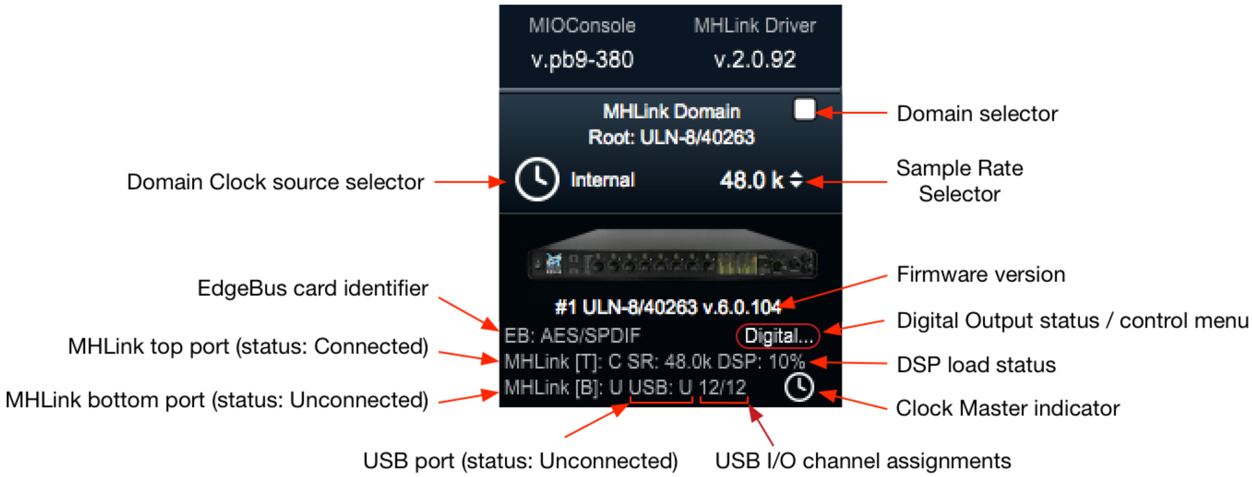


Figure 1.4: ULN-8 System and Unit Status display

The System Status Pane displays critical information for every 3d MH device visible to your computer.

The Status Pane header lists the current MIOConsole version and MHLink Driver version. When the version number turns orange, there is a new version available: click to download and install.

The MHLink Domain "Root" box (your 2882 connected directly to the host computer) is listed next and also occupies the #1 spot in the **Unit Status Display** directly below. Click the sample rate indicator to the right select your system sample rate, and the clock icon to the left to set your clock source (currently defaulted to "Internal").

Below the 2882 icon is the box ID label and current firmware version. As with the Console and Driver displays in the Status Pane header, when the firmware version turns orange, click to download and install the new version.

Click on the **"Digital..."** label (circled in red below the firmware version) to open the **Digital I/O Status** window for this box.

You'll see a pop-up with that device's type and serial number, with Clock Lock status for each digital audio port and a selector to choose "ADAT" or "TOSLINK" as the output format for the 2882's built-in optical port. Note that the optical input port automatically senses ADAT/SMUX or TOSLINK formats and is independent of the output format.

This window will also identify any digital audio ports present on the EdgeCard™ (abbreviated "EB") installed in the unit.

Other status info shown the example above - the MHLink Top port is Connected, MHLink Bottom port is Unconnected and the USB port is Unconnected. We can also see that the sample rate on the DSP card is 48K, the current DSP load is 10%, and the small clock icon in the lower right indicates that this box is the Clock Master for the domain.

Each 3d box in your domain will have a similar Unit Status Display.

2. Directly below the Unit Status Displays is a currently empty field for Mute Groups controls, which we will get more into later.

3. The Link Groups section, including "Selected Strips", is at the bottom. With "Selected Strips" checked, shift- or command-click to select a few random faders in the mixer. You now have an "Selected Strips" link group of just those selected faders.

   With the "Rel" (as in "Relative") setting selected, it behaves just as one would expect - move one fader and the rest follow.

   Now click "Rel" and select "Inv" (for "Inverse"). In this mode moving one fader causes the other faders in the group to move in the opposite direction.

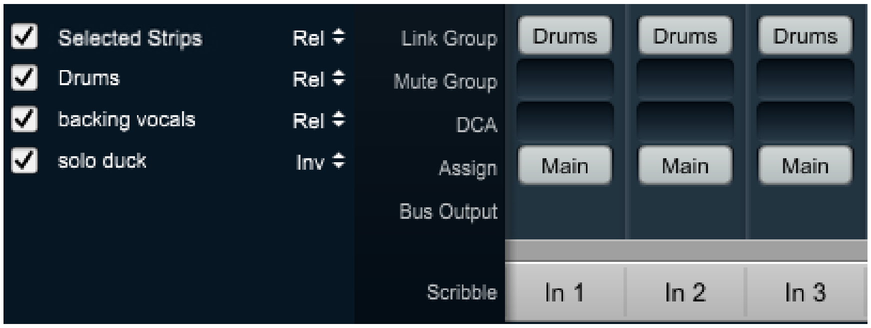


Figure 1.5: Link Groups example

   Any Link Groups you create will be listed below "Selected Strips" in this section and can be assigned using the Link Groups selectors in each mixer strip (as illustrated in the example above). Saved Link Groups have additional operational modes, which will be detailed in the dedicated Link Groups section of the manual.

   The visibility of the System Status and Link Groups section of the 3d Console interface can be toggled with the " [" key (or the "Mixer > Show System Status Pane" menu command).

4. To the right of the System Status Pane is the primary default Mixer window.

   There are three "Mix Panes" available to help organize your mixer. Each Mix Pane can show any combination of Input strips, Bus strips (Main, Aux and Group) or DCA strips. Mixer strip layouts (i.e.: which mix desk strip controls are shown) are configured independently for each Mixer Pane, and are controlled from the "hamburger menu" icon in the upper left corner of each Mix Pane. This lets you set up the strips in one Mix Pane for tracking and the same channel strips in another pane for mixing. Additionally, each Mix Pane has it's own independent scroll bar.

   In the default view, only Mix Pane 1 is set up and visible - Mix Panes 2 and 3 are currently unconfigured and are accessible by the two "Hamburger Menus" at the far top right of the Mixer window.

   Click on the Mix Pane 1 hamburger menu and uncheck "Show Input Strips" - that hides the Input strips in Mix Pane 1. Now bring them back, and select "Configure Channel Strip Elements...". This sub-window let's you show or hide any of the control elements for the strips within that Mix Pane.

   Click on the Show/Hide box for each of the Strip elements so you can see what happens. This feature allows you optimize your Mixer layout for whatever workflow you may require.

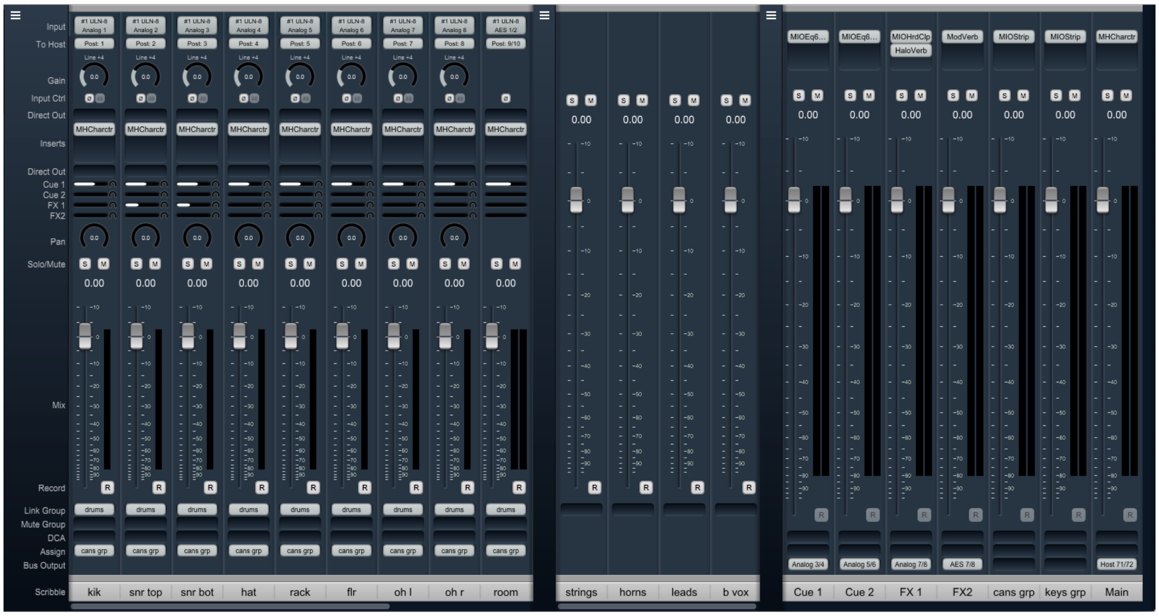


Figure 1.6: ULN-8 Mixer Panes example

In the example above we have Input strips in Mix Pane 1, DCA strips in Mix Pane 2 and Cue/Effects Aux buses, submix Groups and the Main master bus in Mix Pane 3. As you add Input strips, Aux buses, Group buses and DCA strips to your mixer, tweaking the strip layout across the three Mixer Panes helps you maximize the efficiency of your control surface while conserving valuable screen space.

5. In the upper right of the default layout is the Monitor Controller. This is a separate resizable window from the main Mixer window which can also be set to "float" so it is always visible on your display. Across the top of the Monitor Controller is the Monitor Source selector, showing the Main master bus and the four Aux buses from your default Mixer layout.

   All Monitor Controller Sources and Outputs are completely user-configurable - these are just the default settings.

Figure 1.7: Monitor Controller

Below the Source selection buttons is the speaker and volume level control/display.

The new 3d Monitor Controller is a vast upgrade to previous versions with its own dedicated DSP processing path and supports all channel configurations from Mono through Dolby Atmos™ 7.1.4.

There is a wealth of knowledge in the Monitor Controller section on how to set up and use the Monitor Controller.

The Monitor Controller window can be hidden by toggling the " m " key. Hold the Option key to engage scroll wheel/two finger scroll control of Monitor Level control. Option-Click returns controls to the default setting.

6. Below the Monitor Controller is the Cue Controls window. This window will expand downwards as you add more Cue Sends.

   You can think of Cues as named sends to physical outputs; they provide advanced routing controls to allow you feed them with the MC source, dedicated busses (for cue mixes) and with a talkback or listen back signal.

   The Cues give you top level control of the physical output level for each Cue. The controllable routing logic allows you to easily control the source and level of the signals routed to the cues which allows you to use them as a monitoring console for live monitoring and/or as part of a send system for bus output routing.

   By default, there will be one Cue that mults the selected input in the MC and routes to the Headphones of the root box. This mults the selected MC source to both the Monitor output and headphones. You can think of this Cue as your Control Room headphones.

   The Talkback Source at the top of this window can be set to any mono audio input. "Listenback" sends the currently selected Monitor Controller source to all Cue Sends when selected. Multiple independent Cues Sends are fully supported.

   A logical place to start would be to name the first two Aux mix buses "Cue 1" and "Cue 2", then add more Cue and effects mix buses as you build up your session; this allows you to provide independent monitor mixes to performers during tracking.

   Probably the most straightforward application of Cue Sends in an MHLink domain would be have an MHLink box at each musicians' station with one Cue mix output assigned to the headphone jack of each box. That way each person has direct monitoring and their own independent volume control knob with plenty of inputs for their mics and instruments, and it all still appears to the DAW as a single audio device.

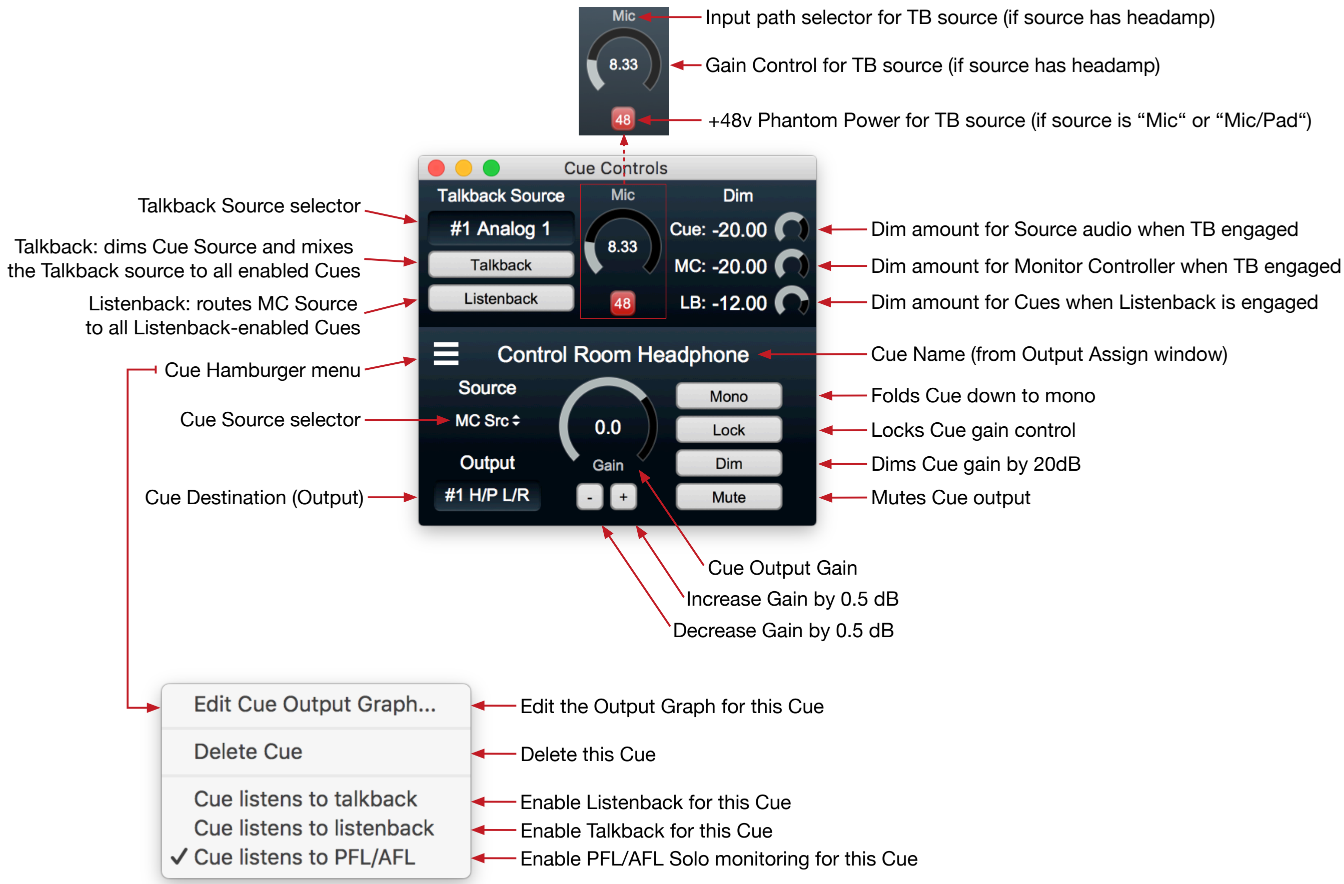


Figure 1.8: Cue Controller

To add a Cue, hit the “p” key, or go to the Monitor menu and select “Add Cue Controller”. You can delete a Cue from its hamburger menu. The Cue Controls can be hidden by toggling the “c” key. Hold the Option key to engage scroll wheel control of Talkback and Cue Level controls. Option-Click returns controls to the default setting.

Check out the Cue Controller section for more details.

The Mixer window

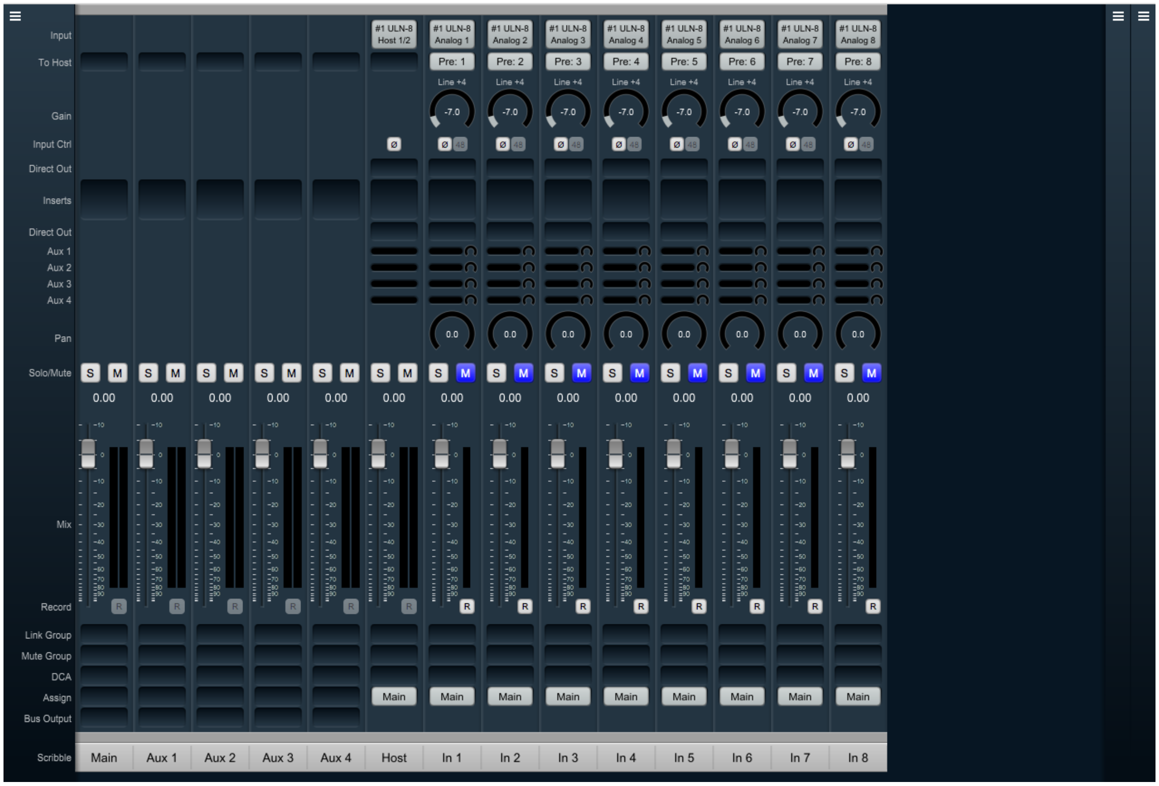


Figure 1.9: Mixer Window

The Mixer window presents you with the primary mixer interface to the ULN-8.

Moving from left to right, the default Mixer surface has a stereo “Main” bus, stereo Aux buses 1 through 4, and nine Input channel strips. The first input strip is the “Host”, which is a stereo mixer input routed from channels 1 and 2 of your host computer (ie.: channels 1 and 2 from your DAW). The remaining eight input strips are all mono, and are routed from the hardware analog inputs from your ULN-8.

Let’s look at the input strips first.

Input Strips

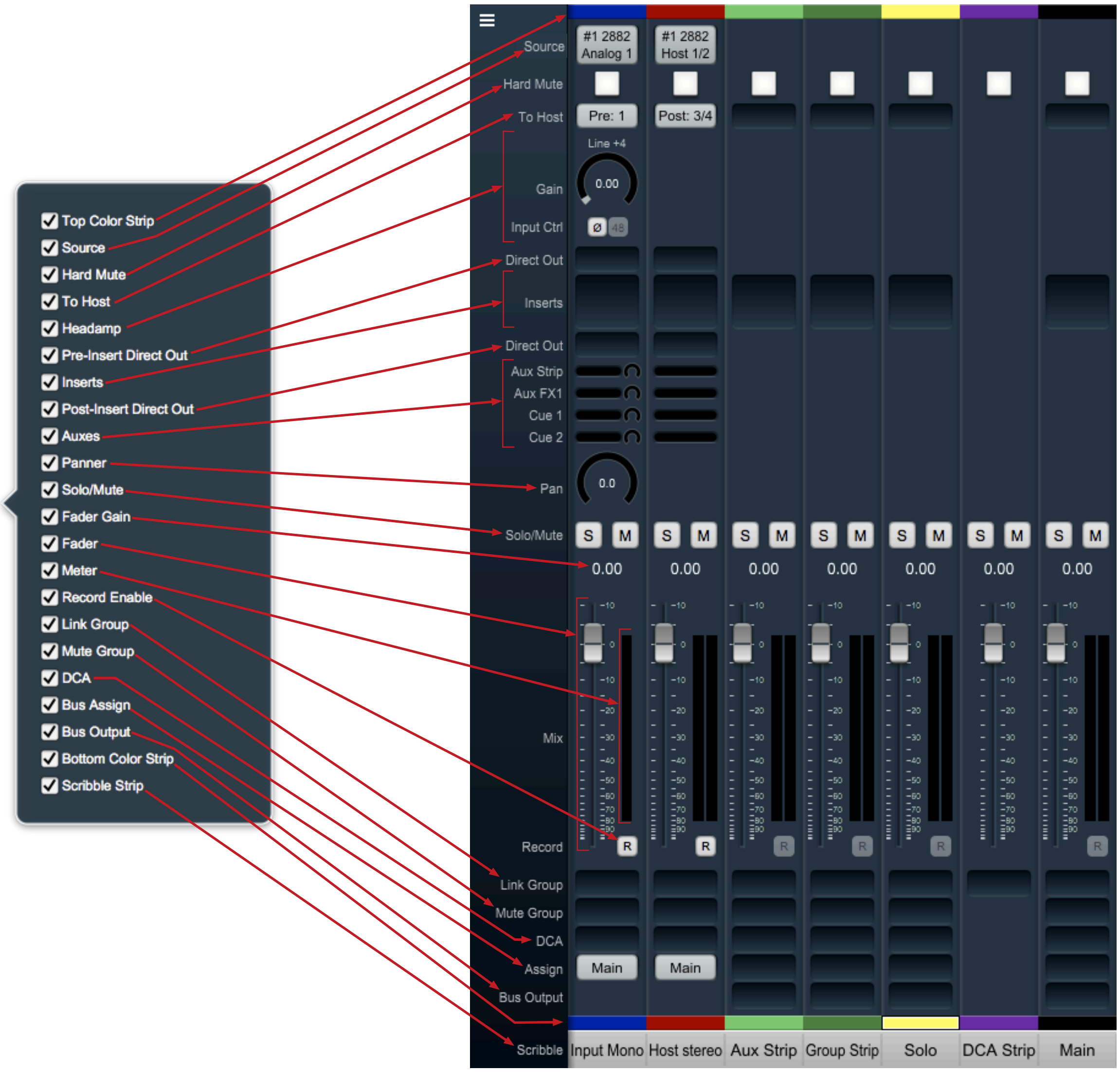


Figure 1.10: Mixer Strip elements exposed

At the top of every input strip is the Input Source selector. Click on it to see the Input Source Selector window. Within the Input Source Selector window you can choose the type of inputs you want to select from, show all the inputs from all boxes in the domain or only one box, how many channels you want your new input strip(s) to include, and how many strips you want to create (or modify) with Cascade.

"Hard Mute" is a kill switch at the source of the desk strip designed for live sound applications. "Hard Mute" is hidden and disabled by default.

Analog inputs get a head amp where you can select the input path type, invert signal phase, toggle phantom power and set input gain.

Below the input selector is the "To Host" menu. This lets you set pre- and/or post-insert return paths directly to your Host computer. Since the 3d MIOConsole3d mixer is designed to support hundreds of hardware and Host computer i/o channels, "To Host" is optimized for fast setup of large mixer layouts, and will automatically assign the *next available Host return channel*.

Below the head amp is the pre-insert Direct Out. The Direct Outs can send signal to any physical analog or digital output, as well as provide additional returns back to the Host computer.[1]

Below the Pre-insert Direct Out are the Inserts slots. Here you can insert plugins, route to sends (additional buses), add hardware I/O inserts, call up macros (previously-saved DSP processors) and instantiate custom DSP graphs. The Inserts section expands as you add more processors up to a maximum of ten per strip. To delete an insert, open the Insert selection window and select “None”. *Note:* For users upgrading from earlier MIO versions, “Character” has been moved to the Inserts section.

Below the Inserts section is a Post-insert Direct Out. Any signal routed from this Direct Out includes all of your insert processing.

Aux send controls appear below the post-Direct Out. Each Aux has a send level control and (where appropriate) a panner to the bus on each strip.

Aux buses can be named in the “Configure Mixer” window or by double-clicking the scribble field at the bottom of the Aux return strip.

For more precise control of the send levels to an Aux bus, click the name of the Aux bus in the strip legend to the left of the mixer pane. A dot will mark the selected bus name, the gray fader knobs in the main mixer surface will become yellow, and the Aux mix will move onto the main surface.

So, yellow faders in your mixer means you are looking at and controlling an Aux mix, not your main mix. Click the dotted Aux name in the legend again to return to the main mix desk. *Hint:* Aux buses are also used for Cue mix sends.

Panners are next. The panner type configures automatically to match the channel width of the strip to the bus that it is assigned to:

- There are no pan controls on a channel assigned to a mono bus.
- Mono input channels will have a pan knob when assigned to stereo or LCR buses.
- Mono input channels assigned to LCRS through 7.1 buses will have joysticks. Right-clicking on the joystick will allow you to hard assign the input channel to a specific output channel, i.e. Center, Left, etc.
- Multichannel inputs (stereo and above) have no pan control, and are direct routed to the matching channels in the bus.

The fader/meters section dynamically expands and contracts based on the size of the Mixer window and the number of control elements currently visible in the strip. The number of meters will always reflect the number of channels handled in the strip. Meters display “pre-fader” by default, and can be set to read “post-fader” signal in the MIOConsole3d Mixer menu.

All input and bus strips have Solo, Mute and Record Enable buttons.

Please note that ‘Record Enable’ is active only if there is audio routed from that strip back to the Host computer either via “To Host” or a Direct Out (the audio you want to record must be routed back to the Host or the Host computer has no way to record it to the hard drive).

Below the fader/meters section are controls for managing assignments for Link Groups, Mute Groups and DCA groups, with Bus Assign selectors (all currently defaulted to the Main bus) as the bottom controls on the input strip. The Assign control is for routing that strip the Main bus and/or any available Group bus.

[1] *Hint regarding Direct Outs:* Use Direct Outs whenever you need to route to a specific-numbered return channel to the Host. “To Host” is designed to automate the addition of many channestrips in a single step and is sometimes just not the right tool for the job. Using Direct Outs (often coupled with ‘Selected Strips’ groups and Cascade) to route your Host returns allows you to precisely target specific channels or groups of channels from the MIOConsole3d mixer to your DAW.

At the very top and bottom of each strip is a color band (default color: gray) for differentiating strips at a glance. Click on a color band to open the color selector. Setting the color of a strip will apply to multiple strips at once based on either a 'Selected Strips' group or the assigned link group. (Note: Strip colors can be applied to the entire strip as a preference.)

See the Input Strip Details section of the MIOConsole3d overview for more information.

Bus Strips

In the default mixer configuration you have the one stereo Main "master" bus and four Aux bus return strips. These bus strips all have To Host (post-fader only), Inserts, Solo, Mute, Record Enable, Link Group, Mute Group, DCA, and Assign control elements which all work just like the Input strips.

Aux, Group and Main buses have an additional routing selector below "Assign" on the strip called "Bus Output". Use Bus Output to route bus strip outputs to any hardware port or as additional post-fader return routes to the Host computer.

There are two other types of buses which are not shown in the default mixer, but are shown in the "Mixer Strip elements exposed" graphic: Group buses and the Solo bus. The Solo bus appears when a PFL or AFL Solo mode is chosen in "Configure Mixer" (default is Solo-In-Place). Group buses are also created in "Configure Mixer". We will get into more detail on MIOConsole3d busing architecture later but the general flow is:

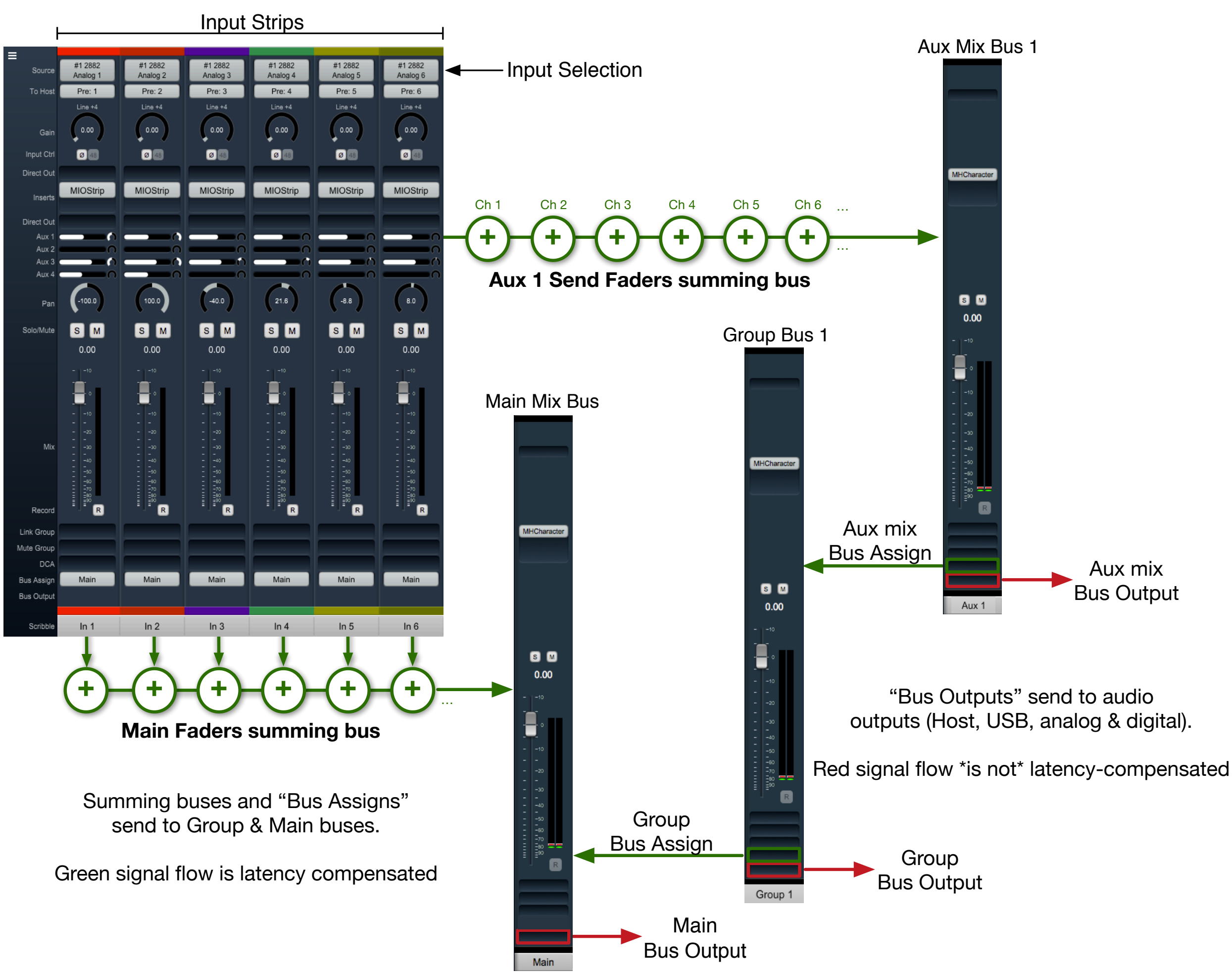


Figure 1.11: Basic 3d Mixer signal flow

- External audio sources enter the 3d mixer through Input strips. Input strips can send to any output, bus or both. Input strips are internally latency-compensated from their input through their assigned output bus.
- Aux buses include their own fader, pan and mute for each Input strip independently of the Main mix bus. Aux buses are for building a completely independent mix and can be used for effect sends (to be returned to a Group or Main mix bus), or for independent cue mixes (to be routed out to the world). Aux buses are internally latency-compensated when assigned to Group buses or the Main mix bus.
- Group buses are sub-mix buses. All the buses assigned to an input strip (Main + any groups) all share the same fader, mute, and pan settings. Group buses can be routed to hardware ports out to the world, to other Group buses, or the Main mix. Routing through Group buses within the mixer will be internally latency compensated.

The fact that the buses are latency compensated means that you can sum channels to any of the buses, and if you assign one bus to another bus the shared signals will be time-aligned and will not introduce any phasing. You can utilize this to do parallel compression on submix buses or sum in a reverb bus that includes a dry component.

Routing

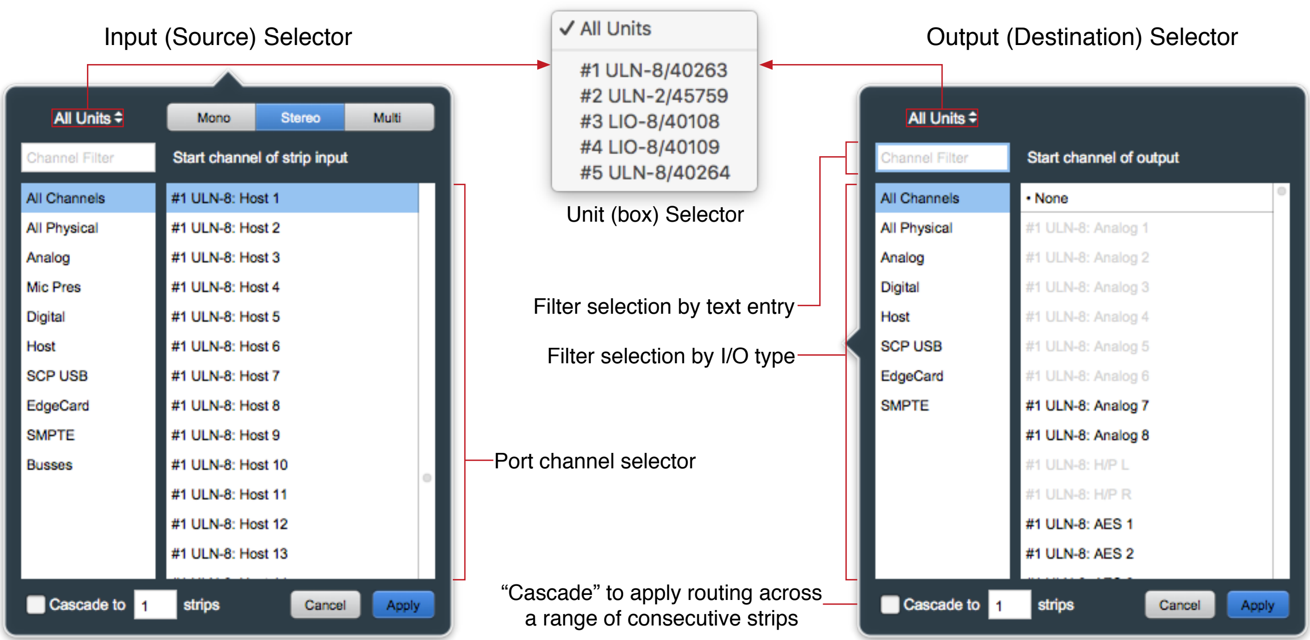


Figure 1.12: Input source and Output destination routing selectors

Since 3d/MHLink domains are completely dynamic (in that you can add or remove boxes live, and up to 100m between...) the Input and Output routing selector UIs both offer filter-by-box, filter-by-port-type and text-entry search tools to help manage the madness. Use the "Unit Selector" when you know which box you need when they are in different locations, select by the port type when the boxes are clustered in a rack, or combine search filters at will. Text-entry searches are dynamic - type "tos" and you are presented with every TOSLINK port visible in the domain.

In the above graphic, note that some of the output ports in the Output selector on the right are greyed-out and unavailable. In the case shown, those ports are occupied as a 5.1 Monitor Controller output and a stereo Cue send, both to the ULN-8 root box #1. Automatic direct mults to outputs are not currently supported in MIOConsole3d, but you can always use a Group or Aux bus to accomplish the same thing.

Two things to be aware of when routing in the 3d mixer:

- Routing from a bus output to an Input strip is allowed, but as mentioned in the previous section, will not be automagically latency-compensated.
- Since routing search filters can be combined, typing "tos" to find a TOSLINK port while the "Analog" port type filter is selected will return zero results (because, after all, TOSLINK is a digital port). Best practice is to select "All Channels" or "All Physical" (which includes all regular analog and digital hardware ports, but not Host computer and SCP USB ports) when searching by text.

For more information regarding configuration and routing of ADAT/TOSLINK, MADI and USB ports see the Digital I/O Status / Control Menu and USB Port Status and Configuration sections of the main manual.

Using the Mixer

You should now be in a good position to integrate the ULN-8 with your audio workstation. First, make sure "Main" is selected as the Monitor Controller source. To play it safe (if you haven't already done so), reduce the Monitor Controller level from it's default "0.0" and bring it up after the music starts. Select "MHLink Audio" (or, if you are connecting via USB, "ULN-8") as your audio interface in your DAW's hardware configuration. To play from your DAW, send your DAW's signal to outputs 1/2 and it will also come into the ULN-8 on the Host 1/2 strip. Hit "Play" in your DAW and enjoy.

Now that you have some tunes going, you might feel like messing around with some insert processes and see what this thing can really do. To get started, go to your Host input strip and click in an Insert slot. The dialog that appears list categories for "Plugin", "Macro", "Graph" and "I/O".

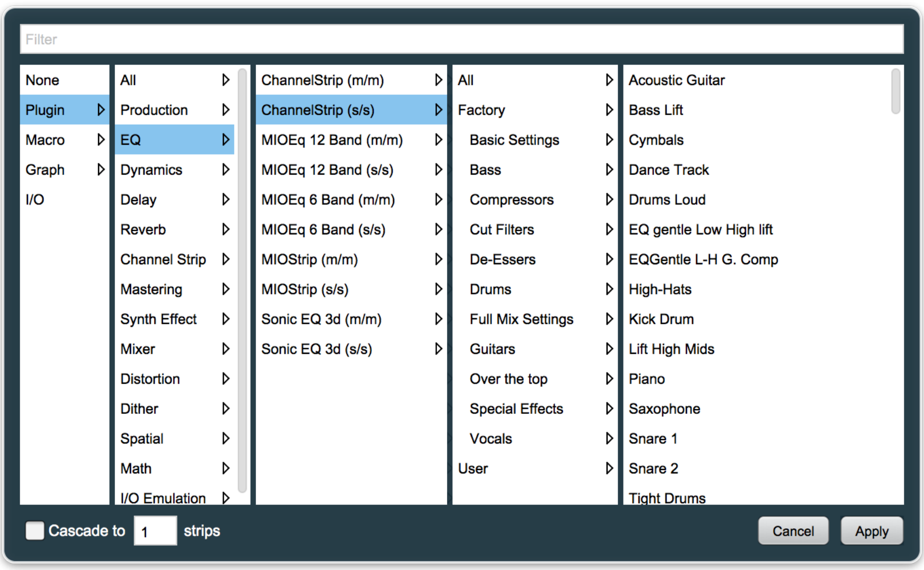


Figure 1.13: Mixer Strip Insert: Plug-in selector

- Plugin: Includes the 3d-native MH Production Bundle , Sonic EQ 3d and many very useful processor "building blocks". Click the *MH Production Bundle* link above to learn about sharing presets between your Metric Halo AU/VST plug-ins and the 3d DSP versions.
- Macro: Most are essentially presets of more complex process chains in a DSP Graph. Feel free to use these as a starting point to create your own personal custom signal processors. The Reverb macros can not be represented as modules in a normal graph, and can not be modified as-is.

- Graph: A graph is a “DSP playground” where you can build your own signal processing chains and save them for later recall. The MIOConsole3d Graph is arguably the most powerful audio hardware DSP environment available to the public. Creative users have built custom stereo to 5.1 spatial up-mix processors, analog tape noise reduction, and multi-layered FOH speaker array time delay and acoustic correction tools, to name just a very few. Graphs can be very simple - just build a quick little tool to do some one-off task not gracefully handled by a packaged plug-in… Graphs are especially spectacular for creating new types of parallel processors because no matter how complex the processing it is all fully internally latency-compensated.
- I/O: A basic signal path insert-within-an-insert, if you will. From within any strip, send to any hardware output or the Host computer and return from any hardware input or the Host. I/O is very useful for an extra Direct Out, as a sidechain source, or for mults to secondary systems (external backup recorder, video truck feed...), etc.

  Note: Since an I/O send/return insert sends audio to external systems outside the mixer DSP and back, I/O loops can not be automatically latency-compensated.

As a reminder, all of the mixing, gain control and processing which is shown and accessed through the MIOConsole3d is operating solely in the box(es) and not on the computer, even though the user interface for the 3d Mixer is controlled from the computer display. This is an important distinction to keep in mind so you can properly manage your sessions and take best advantage of the strengths of both the DAW and the 3d hardware.

Now let's walk through adding the ULN-8's eight AES channels into the MIOConsole3d mixer and your DAW. This procedure works with any kind of input from any box in the domain.

First hit command-shift-A to create a new input strip. In the routing window that pops up, select "Digital" from the filter list on the left (to show only the digital inputs), select "AES 1", type "8" in the Cascade box at the bottom of the routing window and hit Apply (or Return or Enter).

With these few clicks, you have instructed the Mixer to create eight new input strips assigned starting at AES Input 1 (i.e.: AES inputs 1-8).

The Mixer defaults to automatically assign routes to the Host computer for each strip, starting with the next available To Host channel... so, since the eight analog input strips are already using To Host routes 1-8, these new strips will use To Host channels 9-16.

Now, go back to your DAW, create eight new input strips in your DAW mixer and assign them to input 9-16. You have built a virtual patchbay routing audio between your DAW and the ULN-8. You can route pre- and post-insert Direct Outs, Aux buses, Group buses and the Main bus back to the Host DAW in the same way.

To disable any route, just open the routing window and select "None" at the very top of the strip inputs list, or mouse over the strip inputs button and click the "X" to delete that route and free it up for another task.

Mixer Strip Color Bars

Clicking on the Color Bar at the top or bottom brings up a macOS color selector, from which you may choose a color for that Mixer strip. In the Appearance pane of the MIOConsole3d Preferences you may select whether to color the entire mixer strip or just the top and bottom color bars.

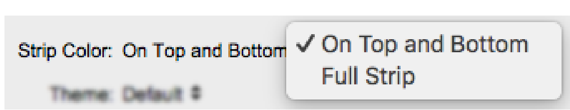


Figure 1.14: Console3d Prefs: Appearance: Strip Color configurator

The Bottom and Top Color Bars mirror each other, so what applies to one applies to the other. You may show or hide the opt or bottom color bars independently with “Configure Mixer Strip Controls...”.

Scribble Strip

Double-click the text field at the bottom of any strip to enter the strips name. The name you enter in the Scribble Strip will be propogated throughout the 3d user interface wherever a Mixer Strip name is found, including plug-in headers, Link Group, Mute Group, DCA and Bus Assign selection windows.

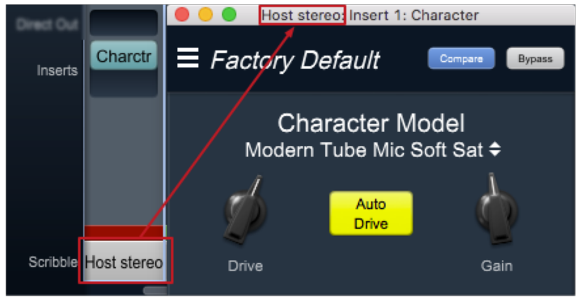


Figure 1.15: Scribble Strip name to Mixer strip Insert header

Scribble Strip names are also used to name sound files recorded in the Record Panel.

Mixer strip Insert controls

Once you have selected a plug-in, it will be listed in the assigned insert slot:

Figure 1.16: Inserted MIOEQ6 Plug-in (as shown in Mixer strip)

Plug-in names will generally appear abbreviated in order to save space ("MIOEQ6" is shown above).

When you move the mouse over an inserted plug-in, the Insert label will change to show three control icons. The tooltips for each of these controls have been exposed in the example graphic below.

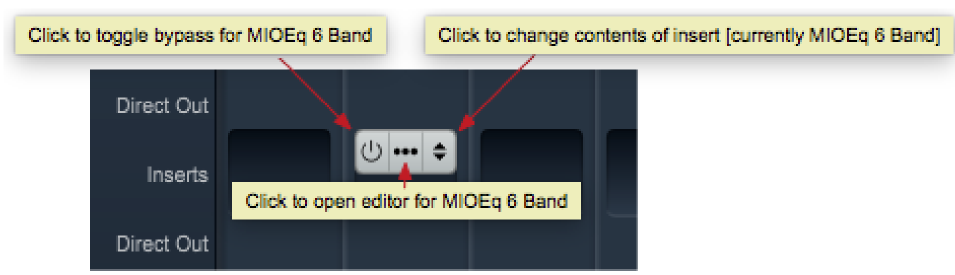


Figure 1.17: Inserted plug-in controls

- The "On / Off" switch icon on the left is the plug-in Bypass. When Bypassed, the Insert button will turn yellow.
- Clicking the "..." icon in the middle opens the inserted plug-in editor UI. When the plug-in editor is open/visible, the Insert button will turn blue.
- Clicking the "up/down" arrows icon at the right opens the Insert selector window, where you may select a replacement plug-in, or navigate to directly open a different saved preset without having to open the plug-in editor UI.

Insert Control modifier key shortcuts

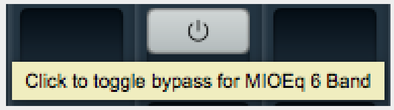


Figure 1.18: "⌘-click" / <Command>-click to Bypass Insert

<Command>-click the Insert button to Bypass the Insert.

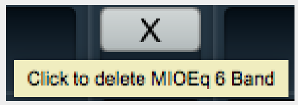


Figure 1.19: <Control-Option-Command>-click to Delete Insert

Use "^⌥⌘-click" / <Control-Option-Command>-click to delete an Insert directly from the Mixer desk.

Quick Copy/Paste Plug-ins

Option-click-drag any plug-in instance from one Insert slot to another anywhere in the Mixer to clone that Plug-in to the new location. Plug-in instances will automatically adapt to the channel width of the target Insert as necessary

"Sweeping" controls

Toggle buttons on consecutive strips in the 3d Mixer desk can be switched in a single move by a click-hold-sweep gesture.

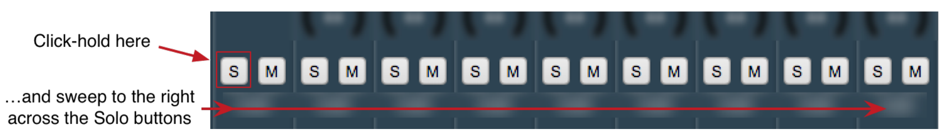


Figure 1.20: "Sweep" to toggle multiple buttons in one gesture

To try it, click on a Solo button, and while holding the mouse button down, drag the cursor to the right or left across the adjacent Solo buttons.

The move works with Polarity Invert, Solo, Mute and Record Enable buttons.

The ULN-8 and surround

If you are working in surround, it’s easy to configure the ULN-8 for surround processing and monitoring. Let's take a minute to walk through the process:

First, go to the Mixer menu and select “Configure Mixer” (cmd-shift-C). This window is the main setup page for your Mixer layout. The Main Mix configuration is set at the top - it is currently set to “Stereo”. If you click on the Main Mix selector, you can change it from “Stereo” to whatever width you need.

Let’s say you’re working in 5.1. Select “5.1 SMPTE/ITU” and hit OK… Voila! Your Main bus is six channels wide and all the mono input channels now have joysticks instead of pan knobs.

Your Host input strip is still bringing your DAW in on just two channels, so let's fix that too. Go to the top of the “Host” input strip and click on the input assignment pulldown. At the top of this window you have the choice of a mono, stereo or a multi-channel strip. Select “Multi” and click “Apply”.

Your Host strip is automatically set to match the Main bus configuration of your mixer: in this case 5.1 SMPTE/ITU. Now your audio will come into the ULN-8 in SMPTE surround format as positioned by your host on channels 1 through 6.

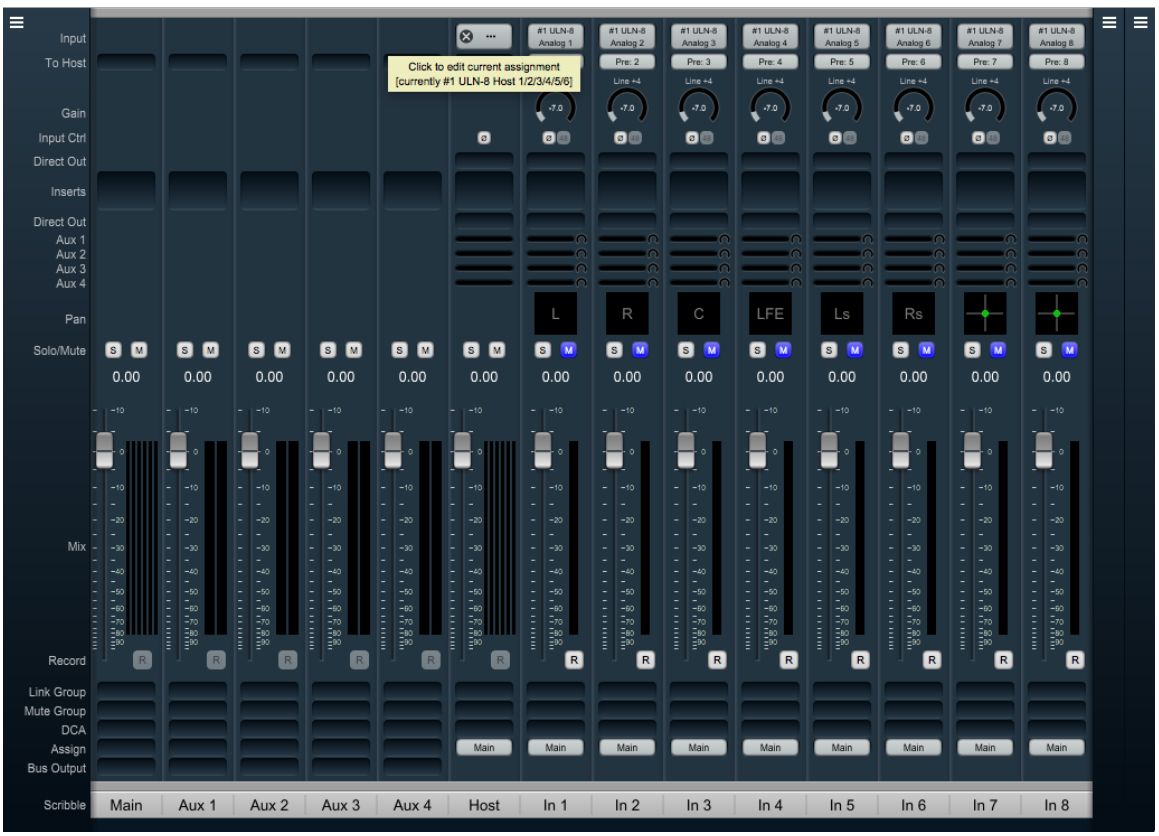


Figure 1.21: Surround Mixer

Your surround mixer should look pretty similar to this. You can right-click (or ctl-click) the panner on the first six mono input strips to hard-assign each strip to a particular speaker channel as is shown on the example above.

Note especially the mouse-over pop-up at the top of the Host desk strip. Ordinarily the text “#1 ULN-8 Host 1/2/3/4/5/6” would be too small to read, so in the interest of saving screen real estate these mouse-overs come in very handy.

If you can afford the space, use the "Mixer - Set Mixer Strips Width" menu to adjust your mixer desk strips' width to taste.

Surround Monitoring Setup

In order to monitor in surround, you need to create a 5.1 output in the Monitor Controller. Go to the "Monitor" menu and select "Add Monitor Output" to get the setup box. You'll see that the "Name" field is already selected, so go ahead and type "5.1" to name the monitor output.

The "Type" selector to the right of this window is where you choose your new Monitor Controller output configuration: Select "5.1 SMPTE/ITU".

To set up 5.1 Monitor Controller outputs, click the Left channel selector (currently set to "None") in the "Destinations" field. The routing box should appear as in the figure below.

If your speaker channels are all in ITU order, you can just double-click "ULN-8 Analog 1" to automatically cascade L, R, C, LFE, Ls, Rs to channels 1, 2, 3, 4, 5 and 6 respectively. Otherwise de-select "Cascade" at the bottom of this window and assign each speaker output individually.

Click "Apply" to route the speaker assignments. Note that you can apply individual level trim and delay for each speaker from this menu, should you desire.

Click "Add" to confirm and establish the new Monitor Controller Output parameters and finish up. You can return to the Monitor Controller Output configuration screen any time by right-clicking the MC output selector, or going to the Monitor menu and selecting "Edit Current Monitor Output".

Figure 1.22: Monitor Controller speaker assignment

Unleash the DSP

The ULN-8 is based on the 3d processing engine and includes a very powerful arsenal of high-precision audio processing tools. All plug-ins and processors work in any channel configuration and include full parameter save and recall with lots of factory presets to get you started.

There are two ways to access the DSP:

- You can insert processes directly in the mixer strip inserts; this works well for standalone processes like eq, compressors, etc.
- Insert a graph. This lets you chain plugins together, use them in parallel, and create custom processors with routing configurations that would be very difficult (or impossible) with other platforms.

  Graphs are also available in the Monitor Controller and Cue Controller signal path. High-precision room EQ/acoustic correction, custom crossovers and bass management with none of the expense, inconvenience or limitations imposed by running such processes in outboard gear or a DAW monitor bus.

You should definitely check out the following processes:

- MIOStrip: Gating, EQ and compression powerhouse. Very clean, very transparent - put it up against any precision digital mastering hardware and be just a little stunned. Mix it with some Character for a tracking console vibe.
- Character and MHCharacter: The sound of different analog circuits and devices available on any input, output or bus. Both have manual and automatic drive and gain. Character is something of a late 80’s/90’s animal with an extra lo-fi bite. MHCharacter is geared more towards higher precision, warmth and detail enhancement.
- Haloverb: Great sounding reverb, doesn't use any CPU from your host. What’s not to love?
- Transient Control: Super fast, super clean initial strike control.

You should read the DSP Implementation section to learn more about how to work with DSP, and the DSP documentation details the over 100 plug-ins available in the 3d DSP engine.

This should get you started with the ULN-8!

Additional Resources

The ULN-8 is an exceptionally deep product, and there are many features, applications and workflows to discover. We have published a series of technical notes and tutorial movies that go in depth about the Mobile I/Oplatform. Please take a look at them to learn more about the ULN-8: https://mhsecure.com/metric_halo/support/tutorials.html

2. LIO-8 Quick Start Guide

Figure 2.1: LIO-8

Prepare the unit for use

Unpack the LIO-8 and make sure all the parts are there:

- One Mobile I/O LIO-8 unit
- One IEC Power Cord appropriate for your area
- One 18-volt 60-watt world-ready external power supply
- One 12” CAT5e Ethernet Cable
- One 14' CAT5e Ethernet Cable
- One USB-A to USB-C Cable
- Two Rack Ears w/ fasteners
- Rubber feet
- Warranty/Registration Card

Welcome to your new Metric Halo LIO-8 3d! Once you’re finished checking the box physically and installing the rack ears (or rubber feet), connect the power supply. Now connect your input and output cables along with your monitors and signal sources. Your monitors should be connected to Analog Out 1/2, either via a DB25 cable or the 1/4" TRS jacks. Turn the unit on using the front panel switch.

> ***Super QuickStart!***
> Plug your computer into the USB-C port on the back of the LIO-8.
>
> Select “LIO-8” as your computers’ audio device.
>
> Done.
>
> The LIO-8 will receive and send audio from/to channels 1 through 12 of your USB host device at all resolutions up to 192kHz/24 bit. Audio channels 1 and 2 of your host device will play audio out both the headphone jack and the analog channels 1 and 2 jacks and DB25 analog out on the rear panel.
>
> *Note:* While this is indeed the fastest way to get audio going, it is also the most boring. Connecting your Host computer through MHLink is by far the more powerful and expansive interface (and certainly the preferred method), but rest assured the USB connection will take on a much more interesting role later in the Advanced section. Please, do read on.

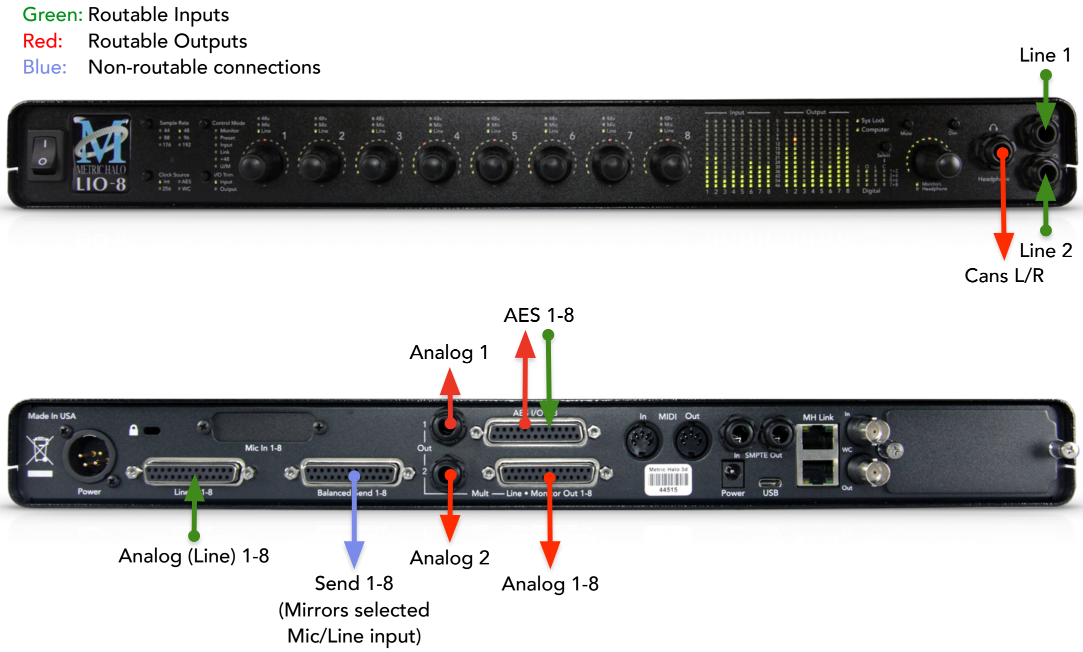


Figure 2.2: LIO-8 Routing

Connect the LIO-8

Install the latest MIOConsole3d installer package from: macOS MIOConsole3d installers (this will require a restart). Please see the section on software installation if you need clarification.

> **Driver Installation on macOS 10.13 and newer**
> With High Sierra (macOS 10.13) Apple introduced another layer of driver (system extension) software security.
>
> In order for third-party drivers (system extensions) to be activated on new installs of High Sierra, the user of the computer (you) must explicitly approve the activation, after the installation has occurred.
>
> You must do this before you will be able to use your hardware via MHLink.
>
> The authorization process is the same for all third-party drivers. If you are unfamiliar with the process, please read our FAQ answer here: macOS Driver User Authentication.
>
> Additionally, with macOS 10.14 and 10.15, you must select "Yes" when asked to give MIOConsole3d permission to access the "Microphone". If in doubt, double-check this setting: go to the Apple menu: System Preferences.../Security & Privacy pane, select "Microphone" in the column to the left, and make sure that MIOConsole3d is checked.

Connect an ethernet cable between the LIO-8 and an available ethernet port on your computer, then go to the System Preferences and select "MHLink Audio" as the system's sound input and output.

Get familiar with the front panel

Look at the LED under the Monitor Control encoder; the "Monitor Control" LED should be illuminated. If "Cans" is lit, push the encoder to switch it (all encoders have shaft buttons). The LED should be green; if it is yellow, push the “Dim” button. If it is red push the “Mute” button. Now that you’re sure that you’re looking at the Monitor Controller and that is not dimmed and unmuted, turn it down. The meters are now showing the gain value of the Monitor Controller. This will happen any time you adjust an encoder.

Take a listen

With MHLink, you need to launch MIOConsole3d to establish the routing between MHLink and the Analog Outputs on the box. The routing is established automatically as soon as you launch MIOConsole3d.

If you connect with USB, you don't need to launch MIOConsole3d to establish the output routing; it is setup automatically by the box itself.

Note: The LIO-8 default configuration is set to receive audio from channels 1 and 2 from your computer, and play audio out both the headphone jack and analog out channels 1 and 2 from the db25 and 1/4" TRS jacks on the rear panel.

Using Legacy Interfaces

If you wish to compare your new 3d interface with an older “2d” Firewire MIO interface, you can leave it connected as always and run them both at the same time.

All FireWire-based Metric Halo interfaces will operate as fully independent audio devices alongside your Metric Halo 3d Ethernet-based interfaces.

MIOConsole3d

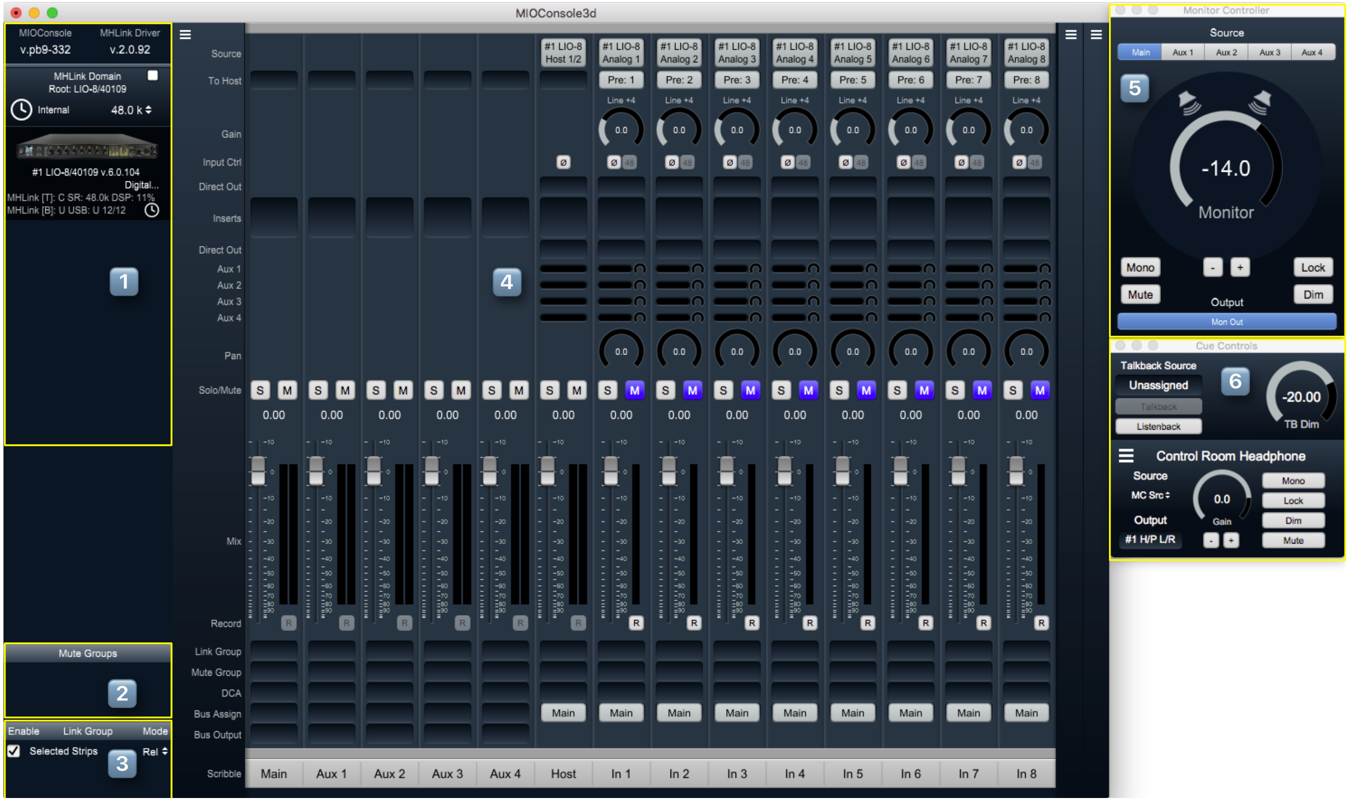


Figure 2.3: MIOConsole3d LIO-8 default view

MIOConsole3d Default Window Layout

At first launch of MIOConsole3d with your LIO-8 you will be presented with the default layout shown above.

There are a few important things to look at here so let's take a few minutes to familiarize ourselves with the interface/GUI.

First of all, if the Mixer window takes up too much room or appears too small on your screen, select Mixer/Mixer UI Scale from the menu bar and adjust the visual scale of the mixer surface. Note that scaling does not change the size of all of the Mixer strip elements - the fader/meters section remains elastic and will shrink or stretch as other Mixer elements are resized, added or removed. Try selecting the 85% scale setting so you can see what it does.

Now resize the Mixer window by grabbing the window corner (or any of the edges) so you can see how the mixer elements react.

You can also change the width of mixer strips (wider strips are helpful to better see 8-channel surround meters) and show or hide any mixer strip UI element. We will explore these features in more detail below.

As you have probably noticed as you check things out in the MIOConsole3d, most elements in the Mixer, Monitor Controller and Cue Controls windows have mouse-over pop-up hints which display expanded details for the button beneath.

The regular mixer surface button labels will often be abbreviated to save space, but the mouse-overs will display all the details of each control setting. This is an especially important feature as you add more boxes to your system domain.

1. In the upper left side of the main mixer window you see the System Status Pane.

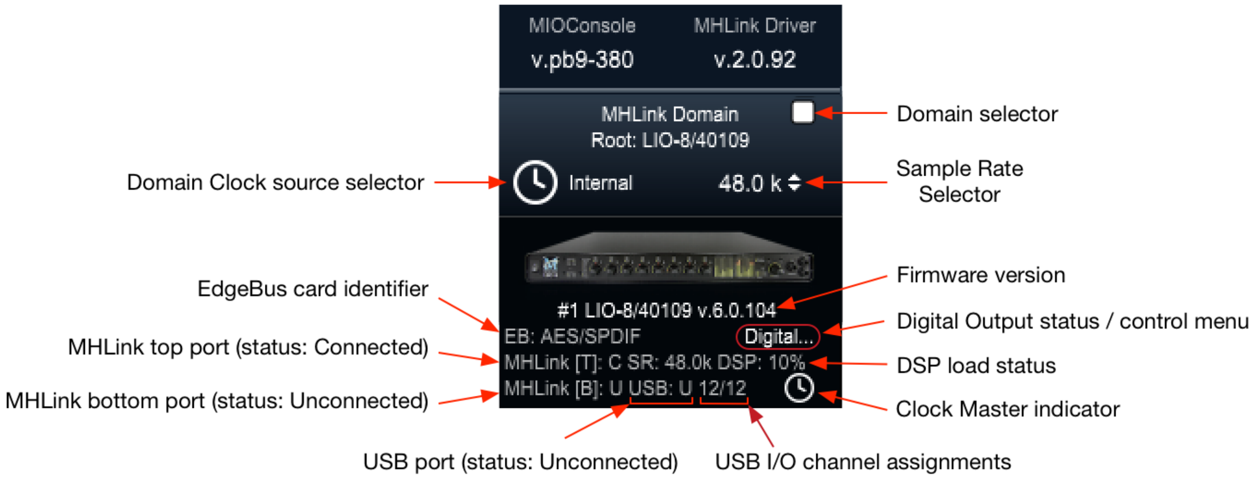


Figure 2.4: LIO-8 System and Unit Status display

The System Status Pane displays critical information for every 3d MH device visible to your computer.

The Status Pane header lists the current MIOConsole version and MHLink Driver version. When the version number turns orange, there is a new version available: click to download and install.

The MHLink Domain "Root" box (your LIO-8 connected directly to the host computer) is listed next and also occupies the #1 spot in the **Unit Status Display** directly below. Click the sample rate indicator to the right select your system sample rate, and the clock icon to the left to set your clock source (currently defaulted to "Internal").

Below the LIO-8 icon is the box ID label and current firmware version. As with the Console and Driver displays in the Status Pane header, when the firmware version turns orange, click to download and install the new version.

Click on the **"Digital..."** label (circled in red below the firmware version) to open the **Digital I/O Status** window for this box.

You'll see a pop-up with that device's type and serial number, with Clock Lock status for each digital audio port and a selector to choose "ADAT" or "TOSLINK" as the output format for the LIO-8's built-in optical port. Note that the optical input port automatically senses ADAT/SMUX or TOSLINK formats and is independent of the output format.

This window will also identify any digital audio ports present on the EdgeCard™ (abbreviated "EB") installed in the unit.

Other status info shown the example above - the MHLink Top port is Connected, MHLink Bottom port is Unconnected and the USB port is Unconnected. We can also see that the sample rate on the DSP card is 48K, the current DSP load is 10%, and the small clock icon in the lower right indicates that this box is the Clock Master for the domain.

Each 3d box in your domain will have a similar Unit Status Display.

2. Directly below the Unit Status Displays is a currently empty field for Mute Groups controls, which we will get more into later.

3. The Link Groups section, including "Selected Strips", is at the bottom. With "Selected Strips" checked, shift- or command-click to select a few random faders in the mixer. You now have an "Selected Strips" link group of just those selected faders.

With the “Rel” (as in “Relative”) setting selected, it behaves just as one would expect - move one fader and the rest follow.

Now click “Rel” and select “Inv” (for “Inverse”). In this mode moving one fader causes the other faders in the group to move in the opposite direction.

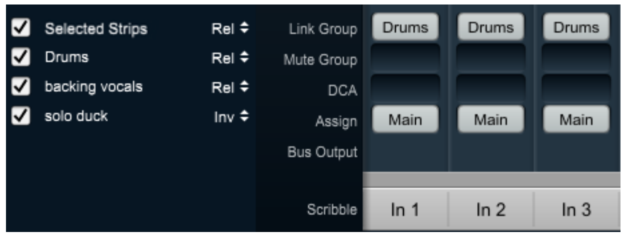


Figure 2.5: Link Groups example

Any Link Groups you create will be listed below “Selected Strips” in this section and can be assigned using the Link Groups selectors in each mixer strip (as illustrated in the example above). Saved Link Groups have additional operational modes, which will be detailed in the dedicated Link Groups section of the manual.

The visibility of the System Status and Link Groups section of the 3d Console interface can be toggled with the “ [” key (or the “Mixer > Show System Status Pane” menu command).

4. To the right of the System Status Pane is the primary default Mixer window.

   There are three “Mix Panes” available to help organize your mixer. Each Mix Pane can show any combination of Input strips, Bus strips (Main, Aux and Group) or DCA strips. Mixer strip layouts (i.e.: which mix desk strip controls are shown) are configured independently for each Mixer Pane, and are controlled from the “hamburger menu” icon in the upper left corner of each Mix Pane. This lets you set up the strips in one Mix Pane for tracking and the same channel strips in another pane for mixing. Additionally, each Mix Pane has it's own independent scroll bar.

   In the default view, only Mix Pane 1 is set up and visible - Mix Panes 2 and 3 are currently unconfigured and are accessible by the two “Hamburger Menus” at the far top right of the Mixer window.

   Click on the Mix Pane 1 hamburger menu and uncheck “Show Input Strips” - that hides the Input strips in Mix Pane 1. Now bring them back, and select “Configure Channel Strip Elements...”. This sub-window let’s you show or hide any of the control elements for the strips within that Mix Pane.

   Click on the Show/Hide box for each of the Strip elements so you can see what happens. This feature allows you optimize your Mixer layout for whatever workflow you may require.

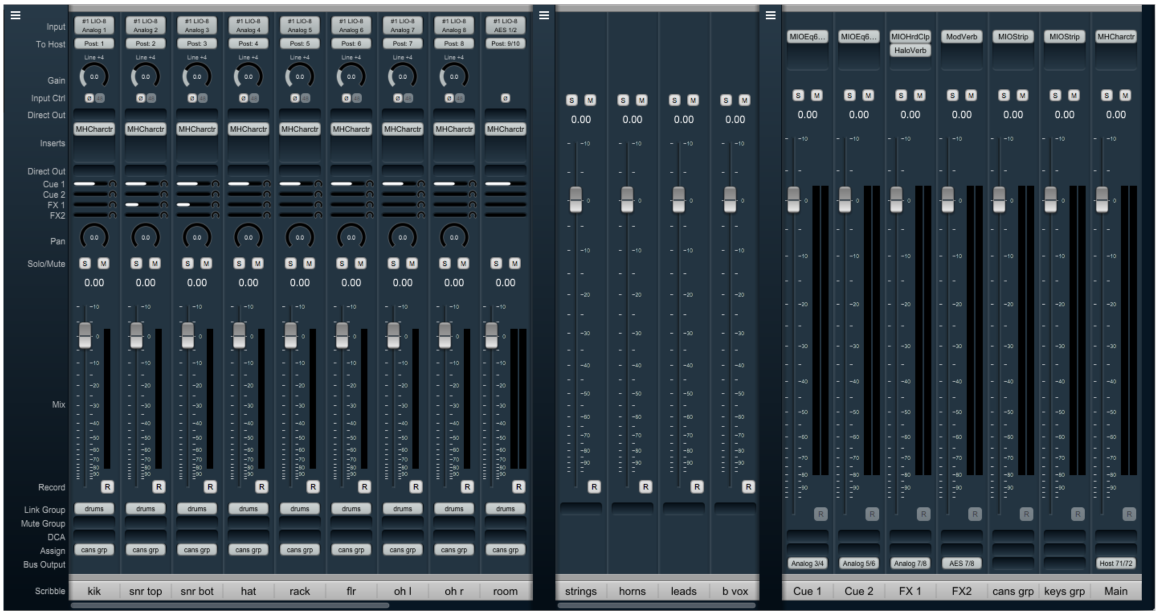


Figure 2.6: LIO-8 Mixer Panes example

In the example above we have Input strips in Mix Pane 1, DCA strips in Mix Pane 2 and Cue/Effects Aux buses, submix Groups and the Main master bus in Mix Pane 3. As you add Input strips, Aux buses, Group buses and DCA strips to your mixer, tweaking the strip layout across the three Mixer Panes helps you maximize the efficiency of your control surface while conserving valuable screen space.

5. In the upper right of the default layout is the Monitor Controller. This is a separate resizable window from the main Mixer window which can also be set to "float" so it is always visible on your display. Across the top of the Monitor Controller is the Monitor Source selector, showing the Main master bus and the four Aux buses from your default Mixer layout.

   All Monitor Controller Sources and Outputs are completely user-configurable - these are just the default settings.

Figure 2.7: Monitor Controller

Below the Source selection buttons is the speaker and volume level control/display.

The new 3d Monitor Controller is a vast upgrade to previous versions with its own dedicated DSP processing path and supports all channel configurations from Mono through Dolby Atmos™ 7.1.4.

There is a wealth of knowledge in the Monitor Controller section on how to set up and use the Monitor Controller.

The Monitor Controller window can be hidden by toggling the " m " key. Hold the Option key to engage scroll wheel/two finger scroll control of Monitor Level control. Option-Click returns controls to the default setting.

6. Below the Monitor Controller is the Cue Controls window. This window will expand downwards as you add more Cue Sends.

   You can think of Cues as named sends to physical outputs; they provide advanced routing controls to allow you feed them with the MC source, dedicated busses (for cue mixes) and with a talkback or listen back signal.

   The Cues give you top level control of the physical output level for each Cue. The controllable routing logic allows you to easily control the source and level of the signals routed to the cues which allows you to use them as a monitoring console for live monitoring and/or as part of a send system for bus output routing.

   By default, there will be one Cue that mults the selected input in the MC and routes to the Headphones of the root box. This mults the selected MC source to both the Monitor output and headphones. You can think of this Cue as your Control Room headphones.

   The Talkback Source at the top of this window can be set to any mono audio input. "Listenback" sends the currently selected Monitor Controller source to all Cue Sends when selected. Multiple independent Cues Sends are fully supported.

   A logical place to start would be to name the first two Aux mix buses "Cue 1" and "Cue 2", then add more Cue and effects mix buses as you build up your session; this allows you to provide independent monitor mixes to performers during tracking.

   Probably the most straightforward application of Cue Sends in an MHLink domain would be have an MHLink box at each musicians' station with one Cue mix output assigned to the headphone jack of each box. That way each person has direct monitoring and their own independent volume control knob with plenty of inputs for their mics and instruments, and it all still appears to the DAW as a single audio device.

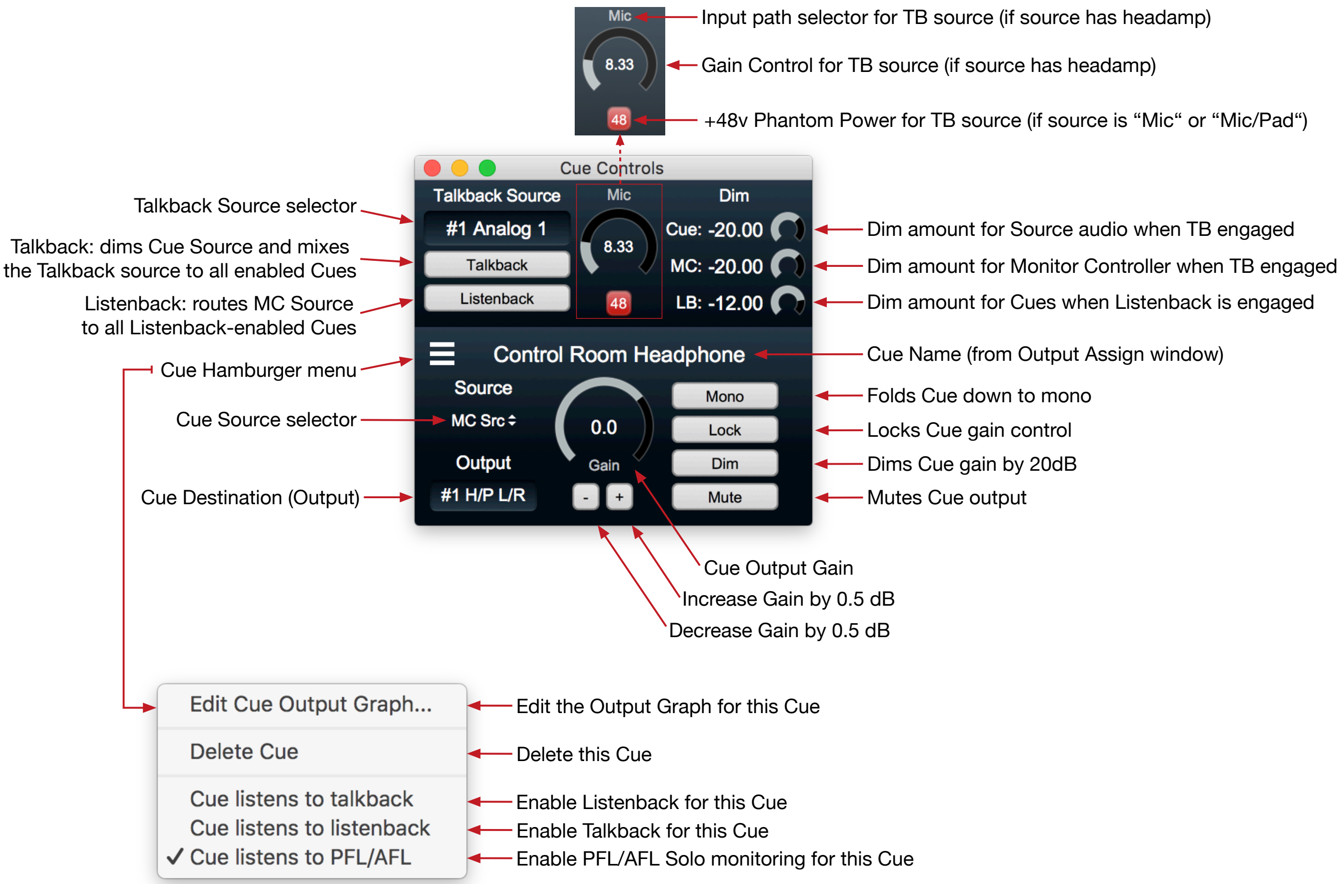


Figure 2.8: Cue Controller

To add a Cue, hit the "p" key, or go to the Monitor menu and select "Add Cue Controller". You can delete a Cue from its hamburger menu. The Cue Controls can be hidden by toggling the "c" key. Hold the Option key to engage scroll wheel control of Talkback and Cue Level controls. Option-Click returns controls to the default setting.

Check out the Cue Controller section for more details.

The Mixer window

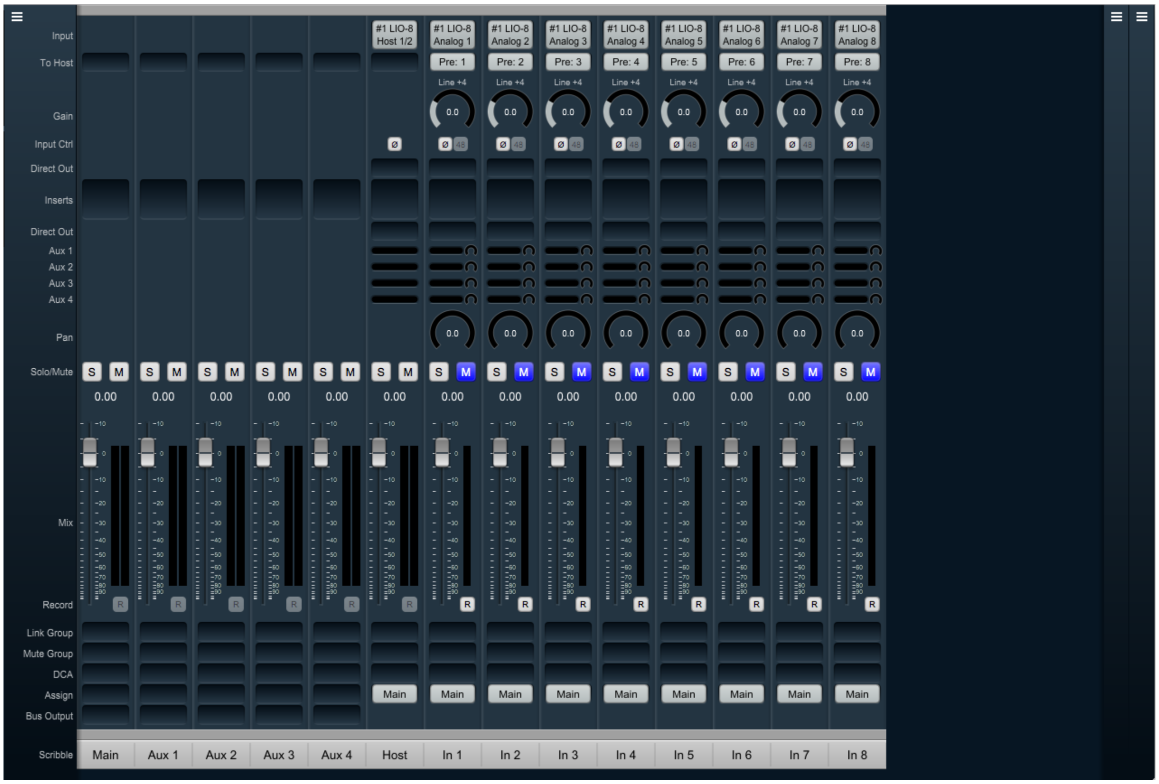


Figure 2.9: Mixer Window

The Mixer window presents you with the primary mixer interface to the LIO-8.

Moving from left to right, the default Mixer surface has a stereo “Main” bus, stereo Aux buses 1 through 4, and nine Input channel strips. The first input strip is the “Host”, which is a stereo mixer input routed from channels 1 and 2 of your host computer (ie.: channels 1 and 2 from your DAW). The remaining eight input strips are all mono, and are routed from the hardware analog inputs from your LIO-8.

Let’s look at these input strips first.

Input Strips

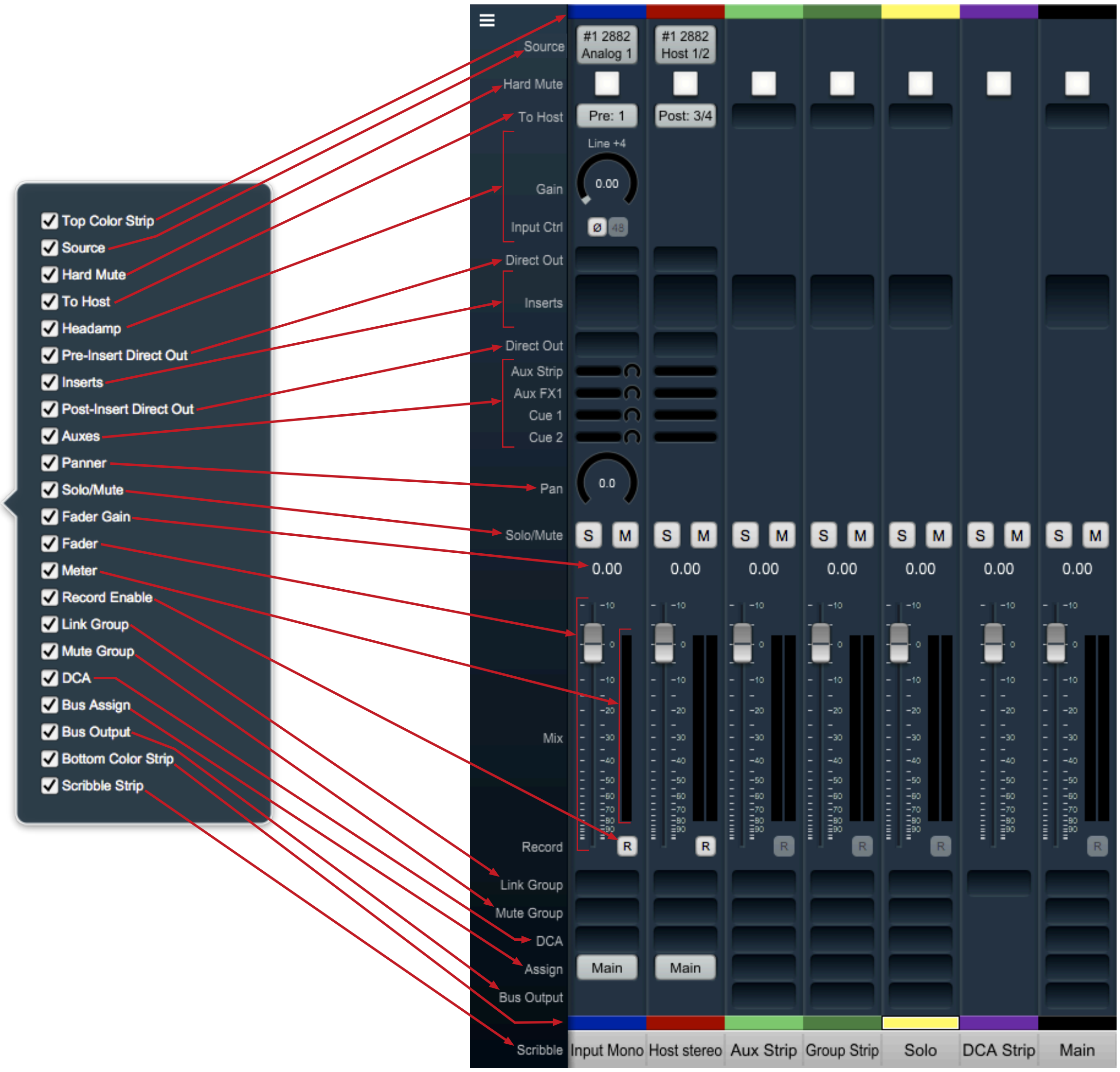


Figure 2.10: Mixer Strip elements exposed

At the top of every input strip is the Input Source selector. Click on it to see the Input Source Selector window. Within the Input Source Selector window you can choose the type of inputs you want to select from, show all the inputs from all boxes in the domain or only one box, how many channels you want your new input strip(s) to include, and how many strips you want to create (or modify) with Cascade.

"Hard Mute" is a kill switch at the source of the desk strip designed for live sound applications. "Hard Mute" is hidden and disabled by default.

Analog inputs get a head amp where you can select the input path type, invert signal phase, toggle phantom power and set input gain.

Below the input selector is the "To Host" menu. This lets you set pre- and/or post-insert return paths directly to your Host computer. Since the 3d MIOConsole3d mixer is designed to support hundreds of hardware and Host computer i/o channels, "To Host" is optimized for fast setup of large mixer layouts, and will automatically assign the *next available Host return channel*.

Below the head amp is the pre-insert Direct Out. The Direct Outs can send signal to any physical analog or digital output, as well as provide additional returns back to the Host computer.[1]

Below the Pre-insert Direct Out are the Inserts slots. Here you can insert plugins, route to sends (additional buses), add hardware I/O inserts, call up macros (previously-saved DSP processors) and instantiate custom DSP graphs. The Inserts section expands as you add more processors up to a maximum of ten per strip. To delete an insert, open the Insert selection window and select "None". *Note:* For users upgrading from earlier MIO versions, "Character" has been moved to the Inserts section.

Below the Inserts section is a Post-insert Direct Out. Any signal routed from this Direct Out includes all of your insert processing.

Aux send controls appear below the post-Direct Out. Each Aux has a send level control and (where appropriate) a panner to the bus on each strip.

Aux buses can be named in the Configure Mixer window or by double-clicking the scribble field at the bottom of the Aux return strip.

For more precise control of the send levels to an Aux bus, click the name of the Aux bus in the strip legend to the left of the mixer pane. A dot will mark the selected bus name, the gray fader knobs in the main mixer surface will become yellow, and the Aux mix will move onto the main surface.

So, yellow faders in your mixer means you are looking at and controlling an Aux mix, not your main mix. Click the dotted Aux name in the legend again to return to the main mix desk. *Hint:* Aux buses are also used for Cue mix sends.

Panners are next. The panner type configures automatically to match the channel width of the strip to the bus that it is assigned to:

- There are no pan controls on a channel assigned to a mono bus.
- Mono input channels will have a pan knob when assigned to stereo or LCR buses.
- Mono input channels assigned to LCRS through 7.1 buses will have joysticks. Right-clicking on the joystick will allow you to hard assign the input channel to a specific output channel, i.e. Center, Left, etc.
- Multichannel inputs (stereo and above) have no pan control, and are direct routed to the matching channels in the bus.

The fader/meters section dynamically expands and contracts based on the size of the Mixer window and the number of control elements currently visible in the strip. The number of meters will always reflect the number of channels handled in the strip. Meters display "pre-fader" by default, and can be set to read "post-fader" signal in the MIOConsole3d Mixer menu.

All input and bus strips have Solo, Mute and Record Enable buttons.

Please note that 'Record Enable' is active only if there is audio routed from that strip back to the Host computer either via "To Host" or a Direct Out (the audio you want to record must be routed back to the Host or the Host computer has no way to record it to the hard drive).

Below the fader/meters section are controls for managing assignments for Link Groups, Mute Groups and DCA groups, with Bus Assign selectors (all currently defaulted to the Main bus) as the bottom controls on the input strip. The Assign control is for routing that strip the Main bus and/or any available Group bus.

[1] *Hint regarding Direct Outs:* Use Direct Outs whenever you need to route to a specific-numbered return channel to the Host. "To Host" is designed to automate the addition of many channestrips in a single step and is sometimes just not the right tool for the job. Using Direct Outs (often coupled with 'Selected Strips' groups and Cascade) to route your Host returns allows you to precisely target specific channels or groups of channels from the MIOConsole3d mixer to your DAW.

At the very top and bottom of each strip is a color band (default color: gray) for differentiating strips at a glance. Click on a color band to open the color selector. Setting the color of a strip will apply to multiple strips at once based on either a 'Selected Strips' group or the assigned link group. (Note: Strip colors can be applied to the entire strip as a preference.)

See the Input Strip Details section of the MIOConsole3d overview for more information.

Bus Strips

In the default mixer configuration you have the one stereo Main "master" bus and four Aux bus return strips. These bus strips all have To Host (post-fader only), Inserts, Solo, Mute, Record Enable, Link Group, Mute Group, DCA, and Assign control elements which all work just like the Input strips.

Aux, Group and Main buses have an additional routing selector below "Assign" on the strip called "Bus Output". Use Bus Output to route bus strip outputs to any hardware port or as additional post-fader return routes to the Host computer.

There are two other types of buses which are not shown in the default mixer, but are shown in the "Mixer Strip elements exposed" graphic: Group buses and the Solo bus. The Solo bus appears when a PFL or AFL Solo mode is chosen in Configure Mixer (default is Solo-In-Place). Group buses are also created in Configure Mixer. We will get into more detail on MIOConsole3d busing architecture later but the general flow is:

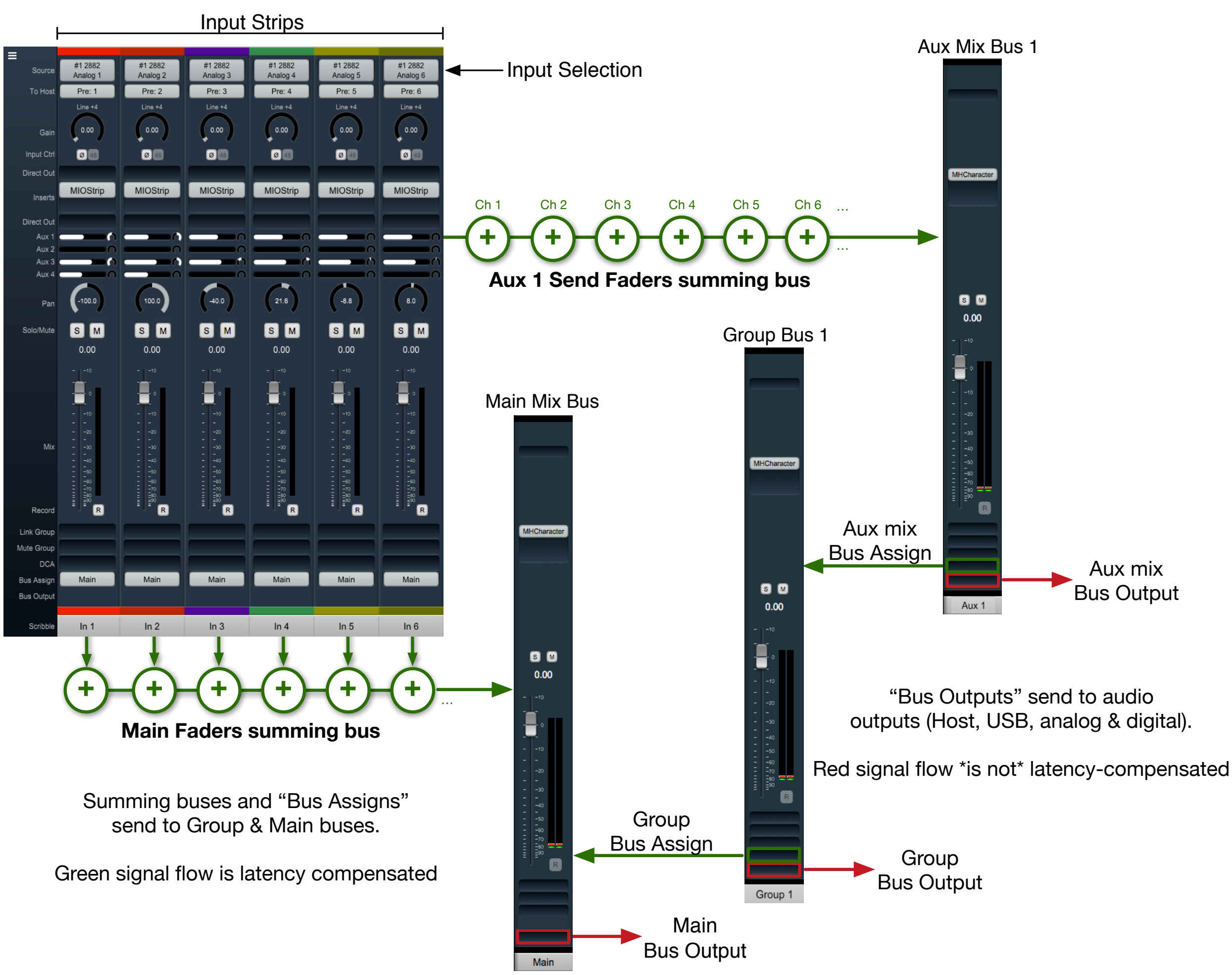


Figure 2.11: Basic 3d Mixer signal flow

- External audio sources enter the 3d mixer through Input strips. Input strips can send to any output, bus or both. Input strips are internally latency-compensated from their input through their assigned output bus.
- Aux buses include their own fader, pan and mute for each Input strip independently of the Main mix bus. Aux buses are for building a completely independent mix and can be used for effect sends (to be returned to a Group or Main mix bus), or for independent cue mixes (to be routed out to the world). Aux buses are internally latency-compensated when assigned to Group buses or the Main mix bus.
- Group buses are sub-mix buses. All the buses assigned to an input strip (Main + any groups) all share the same fader, mute, and pan settings. Group buses can be routed to hardware ports out to the world, to other Group buses, or the Main mix. Routing through Group buses within the mixer will be internally latency compensated.

The fact that the buses are latency compensated means that you can sum channels to any of the buses, and if you assign one bus to another bus the shared signals will be time-aligned and will not introduce any phasing. You can utilize this to do parallel compression on submix buses or sum in a reverb bus that includes a dry component.

Routing

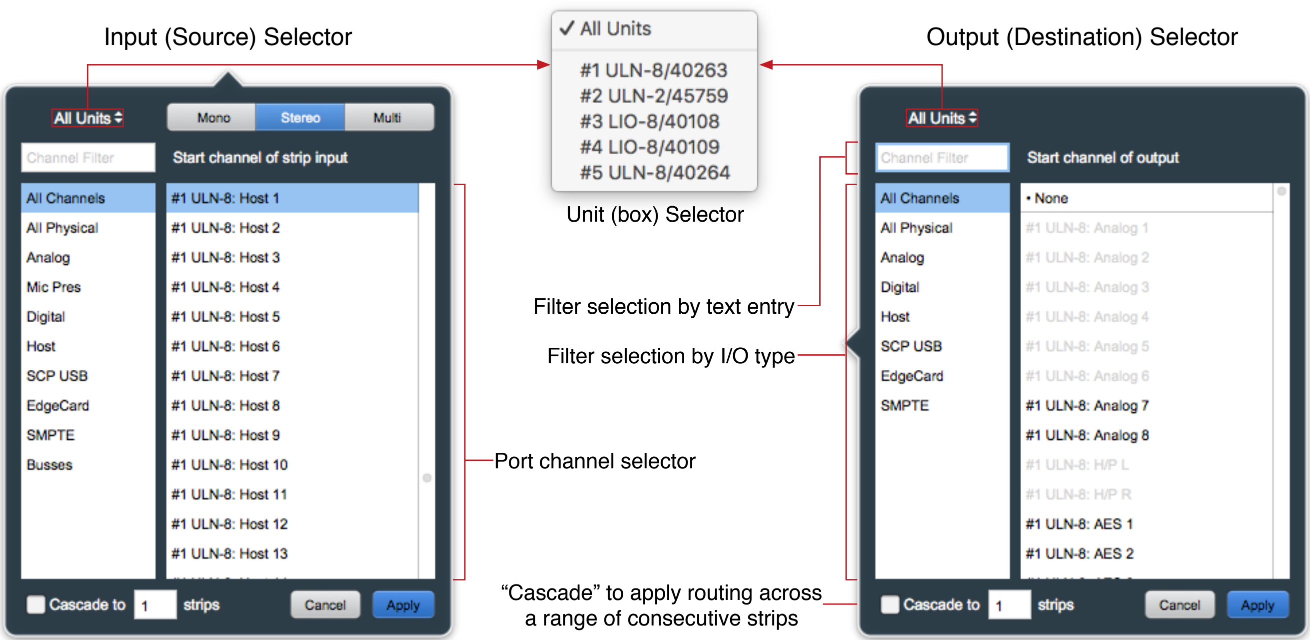


Figure 2.12: Input source and Output destination routing selectors

Since 3d/MHLink domains are completely dynamic (in that you can add or remove boxes live, and up to 100m between...) the Input and Output routing selector UIs both offer filter-by-box, filter-by-port-type and text-entry search tools to help manage the madness. Use the "Unit Selector" when you know which box you need when they are in different locations, select by the port type when the boxes are clustered in a rack, or combine search filters at will. Text-entry searches are dynamic - type "tos" and you are presented with every TOSLINK port visible in the domain.

In the above graphic, note that some of the output ports in the Output selector on the right are greyed-out and unavailable. In the case shown, those ports are occupied as a 5.1 Monitor Controller output and a stereo Cue send, both to the ULN-8 root box #1. Automatic direct mults to outputs are not currently supported in MIOConsole3d, but you can always use a Group or Aux bus to accomplish the same thing.

Two things to be aware of when routing in the 3d mixer:

- Routing from a bus output to an Input strip is allowed, but as mentioned in the previous section, will not be automagically latency-compensated.
- Since routing search filters can be combined, typing “tos” to find a TOSLINK port while the “Analog” port type filter is selected will return zero results (because, after all, TOSLINK is a digital port). Best practice is to select “All Channels” or “All Physical” (which includes all regular analog and digital hardware ports, but not Host computer and SCP USB ports) when searching by text.

> For more information regarding configuration and routing of ADAT/TOSLINK, MADI and USB ports see the Digital I/O Status / Control Menu and USB Port Status and Configuration sections of the main manual.

Using the Mixer

You should now be in a good position to integrate the LIO-8 with your audio workstation. First, make sure "Main" is selected as the Monitor Controller source. To play it safe (if you haven't already done so), reduce the Monitor Controller level from it's default "0.0" and bring it up after the music starts. Select “MHLink Audio” (or, if you are connecting via USB, "LIO-8") as your audio interface in your DAW's hardware configuration. To play from your DAW, send your DAW’s signal to outputs 1/2 and it will also come into the LIO-8 on the Host 1/2 strip. Hit “Play” in your DAW and enjoy.

Now that you have some tunes going, you might feel like messing around with some insert processes and see what this thing can really do. To get started, go to your Host input strip and click in an Insert slot. The dialog that appears list categories for "Plugin", "Macro", "Graph" and "I/O".

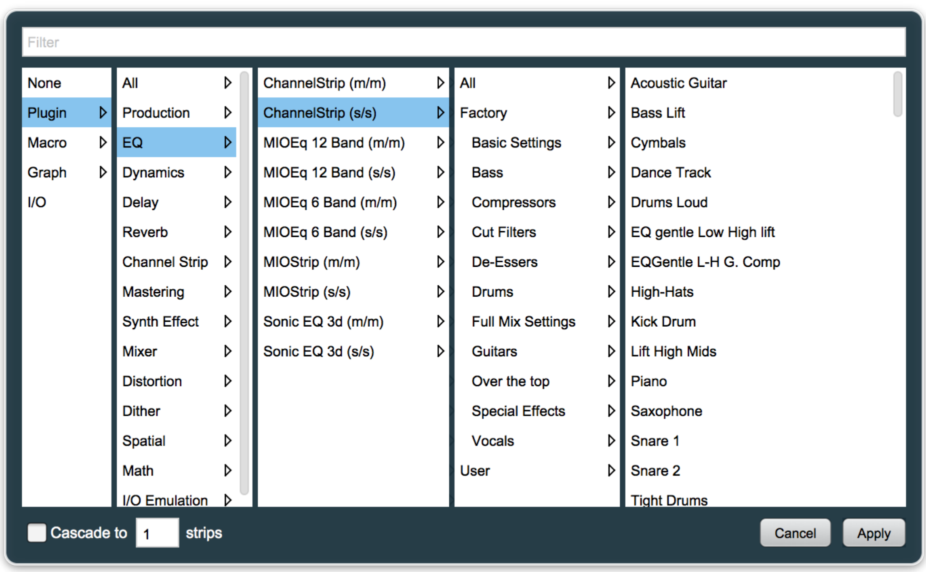


Figure 2.13: Mixer Strip Insert: Plug-in selector

- Plugin: Includes the 3d-native MH Production Bundle, Sonic EQ 3d and many very useful processor “building blocks”. Click the *MH Production Bundle* link above to learn about sharing presets between your Metric Halo AU/VST plug-ins and the 3d DSP versions.
- Macro: Most are essentially presets of more complex process chains in a DSP Graph. Feel free to use these as a starting point to create your own personal custom signal processors. The Reverb macros can not be represented as modules in a normal graph, and can not be modified as-is.

- Graph: A graph is a "DSP playground" where you can build your own signal processing chains and save them for later recall. The MIOConsole3d Graph is arguably the most powerful audio hardware DSP environment available to the public. Creative users have built custom stereo to 5.1 spatial up-mix processors, analog tape noise reduction, and multi-layered FOH speaker array time delay and acoustic correction tools, to name just a very few. Graphs can be very simple - just build a quick little tool to do some one-off task not gracefully handled by a packaged plug-in… Graphs are especially spectacular for creating new types of parallel processors because no matter how complex the processing it is all fully internally latency-compensated.
- I/O: A basic signal path insert-within-an-insert, if you will. From within any strip, send to any hardware output or the Host computer and return from any hardware input or the Host. I/O is very useful for an extra Direct Out, as a sidechain source, or for mults to secondary systems (external backup recorder, video truck feed...), etc.

  Note: Since an I/O send/return insert sends audio to external systems outside the mixer DSP and back, I/O loops can not be automatically latency-compensated.

As a reminder, all of the mixing, gain control and processing which is shown and accessed through the MIOConsole3d is operating solely in the box(es) and not on the computer, even though the user interface for the 3d Mixer is controlled from the computer display. This is an important distinction to keep in mind so you can properly manage your sessions and take best advantage of the strengths of both the DAW and the 3d hardware.

Now let's walk through adding the LIO-8's eight AES channels into the MIOConsole3d mixer and your DAW. This procedure works with any kind of input from any box in the domain.

First hit command-shift-A to create a new input strip. In the routing window that pops up, select "Digital" from the filter list on the left (to show only the digital inputs), select "AES 1", type "8" in the Cascade box at the bottom of the routing window and hit Apply (or Return or Enter).

With these few clicks, you have instructed the Mixer to create eight new input strips assigned starting at AES Input 1 (i.e.: AES inputs 1-8).

The Mixer defaults to automatically assign routes to the Host computer for each strip, starting with the next available To Host channel... so, since the eight analog input strips are already using To Host routes 1-8, these new strips will use To Host channels 9-16.

Now, go back to your DAW, create eight new input strips in your DAW mixer and assign them to input 9-16. You have built a virtual patchbay routing audio between your DAW and the LIO-8. You can route pre- and post-insert Direct Outs, Aux buses, Group buses and the Main bus back to the Host DAW in the same way.

To disable any route, just open the routing window and select "None" at the very top of the strip inputs list, or mouse over the strip inputs button and click the "X" to delete that route and free it up for another task.

Mixer Strip Color Bars

Clicking on the Color Bar at the top or bottom brings up a macOS color selector, from which you may choose a color for that Mixer strip. In the Appearance pane of the MIOConsole3d Preferences you may select whether to color the entire mixer strip or just the top and bottom color bars.

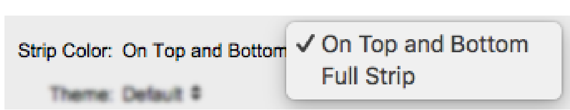


Figure 2.14: Console3d Prefs: Appearance: Strip Color configurator

The Bottom and Top Color Bars mirror each other, so what applies to one applies to the other. You may show or hide the opt or bottom color bars independently with "Configure Mixer Strip Controls...".

Scribble Strip

Double-click the text field at the bottom of any strip to enter the strips name. The name you enter in the Scribble Strip will be propogated throughout the 3d user interface wherever a Mixer Strip name is found, including plug-in headers, Link Group, Mute Group, DCA and Bus Assign selection windows.

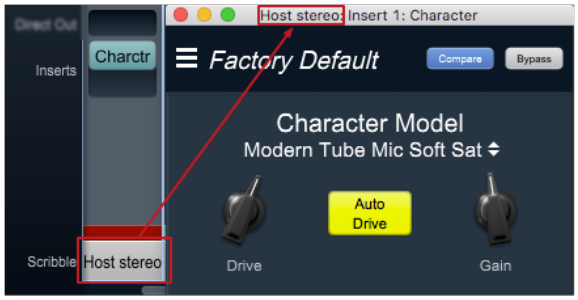


Figure 2.15: Scribble Strip name to Mixer strip Insert header

Scribble Strip names are also used to name sound files recorded in the Record Panel.

Mixer strip Insert controls

Once you have selected a plug-in, it will be listed in the assigned insert slot:

Figure 2.16: Inserted MIOEQ6 Plug-in (as shown in Mixer strip)

Plug-in names will generally appear abbreviated in order to save space ("MIOEQ6" is shown above).

When you move the mouse over an inserted plug-in, the Insert label will change to show three control icons. The tooltips for each of these controls have been exposed in the example graphic below.

Figure 2.17: Inserted plug-in controls

- The "On / Off" switch icon on the left is the plug-in Bypass. When Bypassed, the Insert button will turn yellow.
- Clicking the "..." icon in the middle opens the inserted plug-in editor UI. When the plug-in editor is open/visible, the Insert button will turn blue.
- Clicking the "up/down" arrows icon at the right opens the Insert selector window, where you may select a replacement plug-in, or navigate to directly open a different saved preset without having to open the plug-in editor UI.

Insert Control modifier key shortcuts

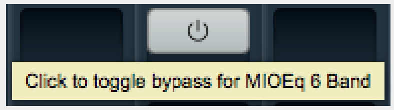


Figure 2.18: "⌘-click" / <Command>-click to Bypass Insert

<Command>-click the Insert button to Bypass the Insert.

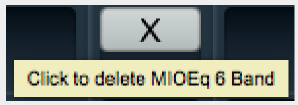


Figure 2.19: <Control-Option-Command>-click to Delete Insert

Use "^⌥⌘-click" / <Control-Option-Command>-click to delete an Insert directly from the Mixer desk.

Quick Copy/Paste Plug-ins

Option-click-drag any plug-in instance from one Insert slot to another anywhere in the Mixer to clone that Plug-in to the new location. Plug-in instances will automatically adapt to the channel width of the target Insert as necessary

"Sweeping" controls

Toggle buttons on consecutive strips in the 3d Mixer desk can be switched in a single move by a click-hold-sweep gesture.

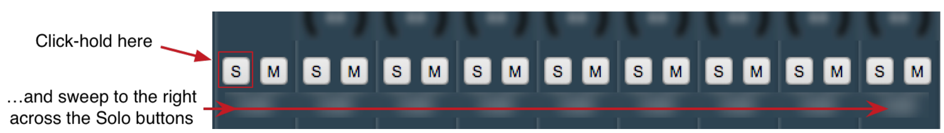


Figure 2.20: "Sweep" to toggle multiple buttons in one gesture

To try it, click on a Solo button, and while holding the mouse button down, drag the cursor to the right or left across the adjacent Solo buttons.

The move works with Polarity Invert, Solo, Mute and Record Enable buttons.

The LIO-8 and surround

If you are working in surround, it's easy to configure the LIO-8 for surround processing and monitoring. Let's take a minute to walk through the process:

First, go to the Mixer menu and select "Configure Mixer" (cmd-shift-C). This window is the main setup page for your Mixer layout. The Main Mix configuration is set at the top - it is currently set to "Stereo". If you click on the Main Mix selector, you can change it from "Stereo" to whatever width you need.

Let's say you're working in 5.1. Select "5.1 SMPTE/ITU" and hit OK… Voila! Your Main bus is six channels wide and all the mono input channels now have joysticks instead of pan knobs.

Your Host input strip is still bringing your DAW in on just two channels, so let's fix that too. Go to the top of the "Host" input strip and click on the input assignment pulldown. At the top of this window you have the choice of a mono, stereo or a multi-channel strip. Select "Multi" and click "Apply".

Your Host strip is automatically set to match the Main bus configuration of your mixer: in this case 5.1 SMPTE/ITU. Now your audio will come into the LIO-8 in SMPTE surround format as positioned by your host on channels 1 through 6.

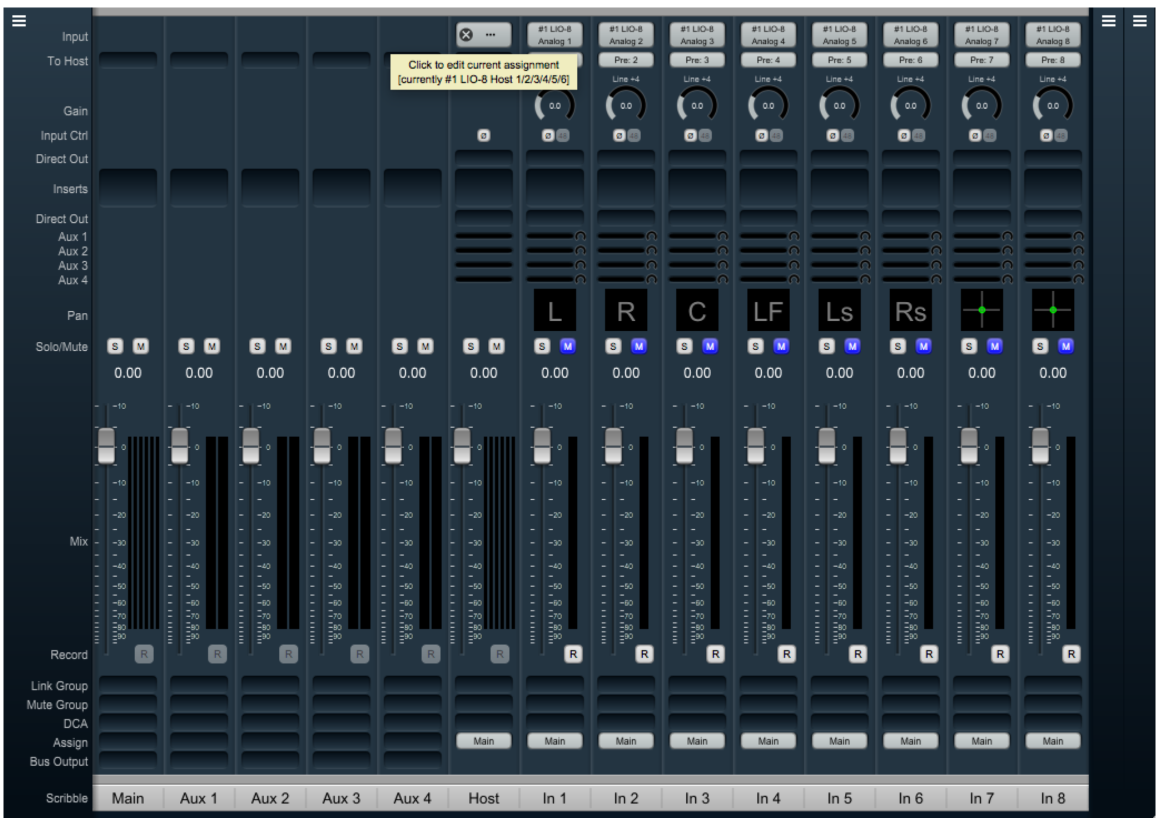


Figure 2.21: Surround Mixer

Your surround mixer should look pretty similar to this. You can right-click (or ctl-click) the panner on the first six mono input strips to hard-assign each strip to a particular speaker channel as is shown on the example above.

Note especially the mouse-over pop-up at the top of the Host desk strip. Ordinarily the text "#1 LIO-8 Host 1/2/3/4/5/6" would be too small to read, so in the interest of saving screen real estate these mouse-overs come in very handy.

If you can afford the space, use the "Mixer - Set Mixer Strips Width" menu to adjust your mixer desk strips' width to taste.

Surround Monitoring Setup

In order to monitor in surround, you need to create a 5.1 output in the Monitor Controller. Go to the "Monitor" menu and select "Add Monitor Output" to get the setup box. You'll see that the "Name" field is already selected, so go ahead and type "5.1" to name the monitor output.

The "Type" selector to the right of this window is where you choose your new Monitor Controller output configuration: Select "5.1 SMPTE/ITU".

To set up 5.1 Monitor Controller outputs, click the Left channel selector (currently set to "None") in the "Destinations" field. The routing box should appear as in the figure below.

If your speaker channels are all in ITU order, you can just double-click "LIO-8 Analog 1" to automatically cascade L, R, C, LFE, Ls, Rs to channels 1, 2, 3, 4, 5 and 6 respectively. Otherwise de-select "Cascade" at the bottom of this window and assign each speaker output individually.

Click "Apply" to route the speaker assignments. Note that you can apply individual level trim and delay for each speaker from this menu, should you desire.

Click "Add" to confirm and establish the new Monitor Controller Output parameters and finish up. You can return to the Monitor Controller Output configuration screen any time by right-clicking the MC output selector, or going to the Monitor menu and selecting "Edit Current Monitor Output".

Figure 2.22: Monitor Controller speaker assignment

Unleash the DSP

The LIO-8 is based on the 3d processing engine and includes a very powerful arsenal of high-precision audio processing tools. All plug-ins and processors work in any channel configuration and include full parameter save and recall with lots of factory presets to get you started.

There are two ways to access the DSP:

- You can insert processes directly in the mixer strip inserts; this works well for standalone processes like eq, compressors, etc.
- Insert a graph. This lets you chain plugins together, use them in parallel, and create custom processors with routing configurations that would be very difficult (or impossible) with other platforms.

  Graphs are also available in the Monitor Controller and Cue Controller signal path. High-precision room EQ/acoustic correction, custom crossovers and bass management with none of the expense, inconvenience or limitations imposed by running such processes in outboard gear or a DAW monitor bus.

You should definitely check out the following processes:

- MIOStrip: Gating, EQ and compression powerhouse. Very clean, very transparent - put it up against any precision digital mastering hardware and be just a little stunned. Mix it with some Character for a tracking console vibe.
- Character and MHCharacter: The sound of different analog circuits and devices available on any input, output or bus. Both have manual and automatic drive and gain. Character is something of a late 80’s/90’s animal with an extra lo-fi bite. MHCharacter is geared more towards higher precision, warmth and detail enhancement.
- Haloverb: Great sounding reverb, doesn't use any CPU from your host. What’s not to love?
- Transient Control: Super fast, super clean initial strike control.

You should read the DSP Implementation section to learn more about how to work with DSP, and the DSP documentation details the over 100 plug-ins available in the 3d DSP engine.

This should get you started with the LIO-8!

3. ULN-2 Quick Start Guide

Figure 3.1: Mobile I/O ULN-2

Prepare the unit for use

Unpack the ULN-2 and make sure all the parts are there:

- One Mobile I/O ULN-2 unit
- One IEC Power Cord appropriate for your area
- One 18-volt 60-watt world-ready external power supply
- One 12” CAT5e Ethernet Cable
- One 14' CAT5e Ethernet Cable
- One USB-A to USB-C Cable
- Two Rack Ears w/ fasteners
- Rubber feet
- Warranty/Registration Card

Welcome to your new Metric Halo ULN-2 3d! Once you’re finished checking the box physically and installing the rack ears (or rubber feet), connect the power supply, and connect your monitors to Monitor Out L/R.

> ***Super QuickStart!***
> Plug your computer into the USB-C port on the back of the ULN-2.
>
> Select “ULN-2” as your computers’ audio device.
>
> Done.
>
> The ULN-2 will receive and send audio from/to channels 1 through 12 of your USB host device at all resolutions up to 192kHz/24 bit. Audio channels 1 and 2 of your host device will play audio out both the front panel headphone jack and the Monitor out and analog channels 1 and 2 jacks on the rear panel.
>
> *Note:* While this is indeed the fastest way to get audio going, it is also the most boring. Connecting your Host computer through MHLink is by far the more powerful and expansive interface (and certainly the preferred method), but rest assured the USB connection will take on a much more interesting role later in the Advanced section. Please, do read on.

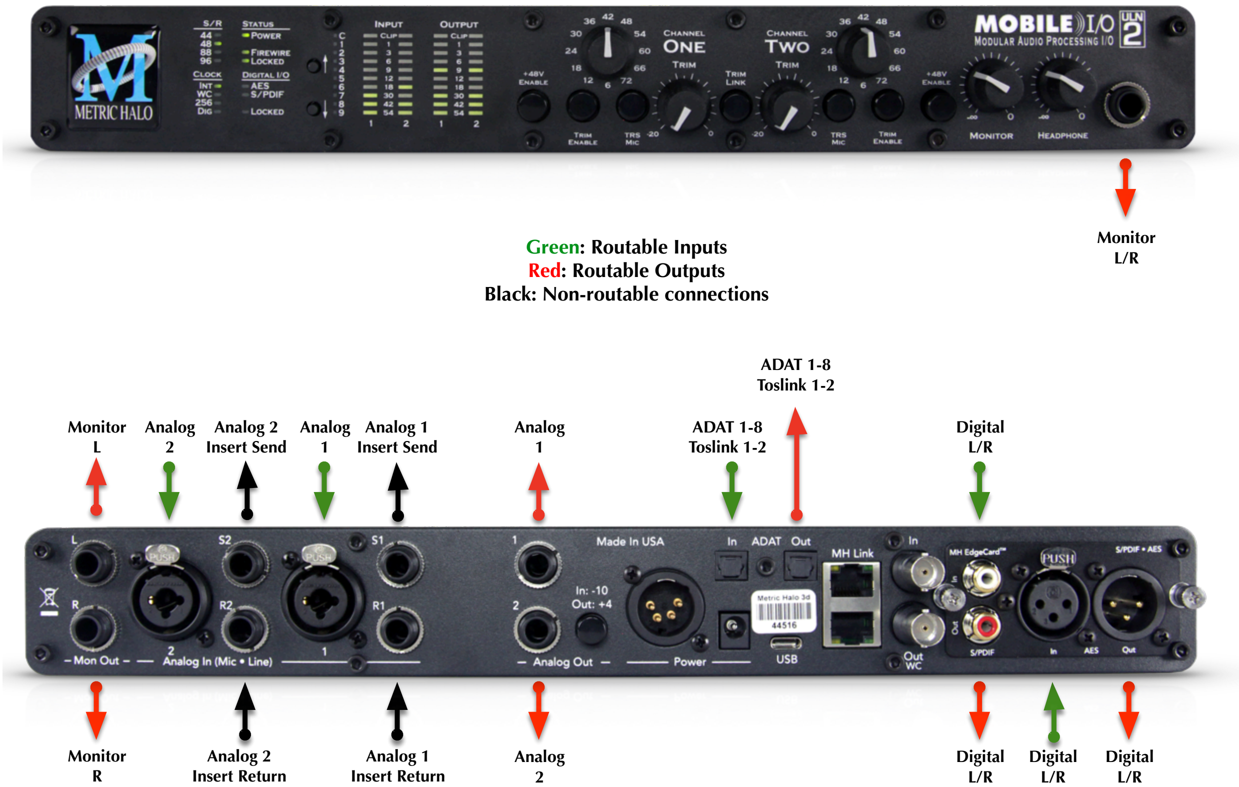


Figure 3.2: ULN-2 Routing

Connect the ULN-2

Install the latest MIOConsole3d installer package from: macOS MIOConsole3d installers (this will require a restart). Please see the section on software installation if you need clarification.

> **Driver Installation on macOS 10.13 and newer**
> With High Sierra (macOS 10.13) Apple introduced another layer of driver (system extension) software security.
>
> In order for third-party drivers (system extensions) to be activated on new installs of High Sierra, the user of the computer (you) must explicitly approve the activation, after the installation has occurred.
>
> You must do this before you will be able to use your hardware via MHLink.
>
> The authorization process is the same for all third-party drivers. If you are unfamiliar with the process, please read our FAQ answer here: macOS Driver User Authentication.
>
> Additionally, with macOS 10.14 and 10.15, you must select "Yes" when asked to give MIOConsole3d permission to access the "Microphone". If in doubt, double-check this setting: go to the Apple menu: System Preferences.../Security & Privacy pane, select "Microphone" in the column to the left, and make sure that MIOConsole3d is checked.

Connect an ethernet cable between the ULN-2 and an available ethernet port on your computer, then go to the System Preferences and select “MHLink Audio” as the system’s sound input and output.

Take a listen

With MHLink, you need to launch MIOConsole3d to establish the routing between MHLink and the Analog Outputs on the box. The routing is established automatically as soon as you launch MIOConsole3d.

If you connect with USB, you don't need to launch MIOConsole3d to establish the output routing; it is setup automatically by the box itself.

Note: The ULN-2 default configuration is set to receive audio from channels 1 and 2 from your computer, and play audio from the headphone jack and the Monitor outs and analog channel 1 and 2 outputs on the rear panel.

Using Legacy Interfaces
If you wish to compare your new 3d interface with an older “2d” Firewire MIO interface, you can leave it connected as always and run them both at the same time.

All FireWire-based Metric Halo interfaces will operate as fully independent audio devices alongside your Metric Halo 3d Ethernet-based interfaces.

MIOConsole3d

Figure 3.3: MIOConsole3d ULN-2 default view

MIOConsole3d Default Window Layout

At first launch of MIOConsole3d with your ULN-2 you will be presented with the default layout shown above.

There are a few important things to look at here so let's take a few minutes to familiarize ourselves with the interface/GUI.

First of all, if the Mixer window takes up too much room or appears too small on your screen, select Mixer/Mixer UI Scale from the menu bar and adjust the visual scale of the mixer surface. Note that scaling does not change the size of all of the Mixer strip elements - the fader/meters section remains elastic and will shrink or stretch as other Mixer elements are resized, added or removed. Try selecting the 85% scale setting so you can see what it does.

Now resize the Mixer window by grabbing the window corner (or any of the edges) so you can see how the mixer elements react.

You can also change the width of mixer strips and show or hide any mixer strip UI element. We will explore these features in more detail below.

As you have probably noticed as you check things out in the MIOConsole3d, most elements in the Mixer, Monitor Controller and Cue Controls windows have mouse-over pop-up hints which display expanded details for the button beneath.

The regular mixer surface button labels will often be abbreviated to save space, but the mouse-overs will display all the details of each control setting. This is an especially important feature as you add more boxes to your system domain.

1. In the upper left side of the main mixer window you see the System Status Pane.

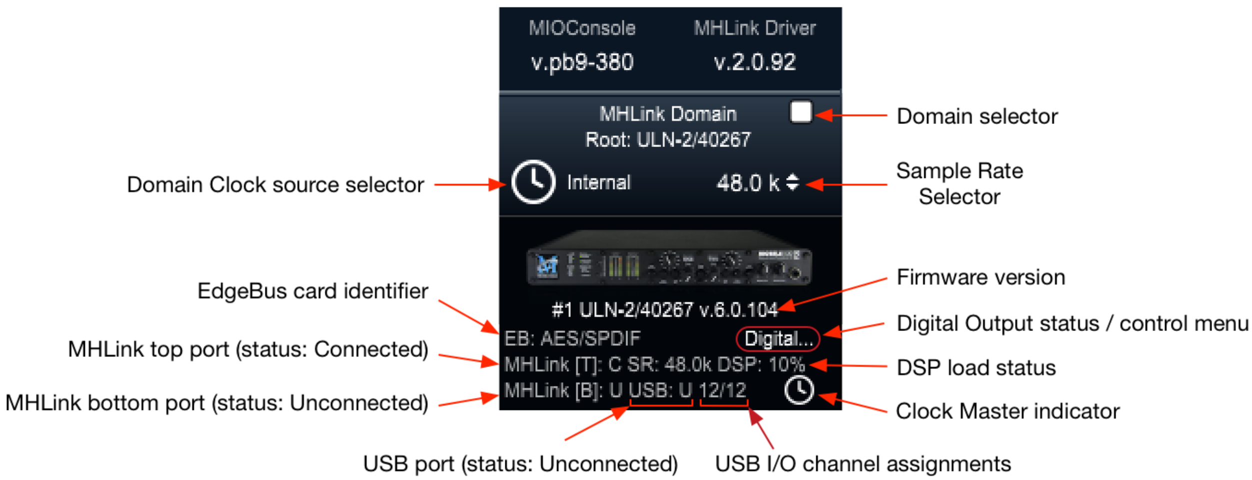


Figure 3.4: ULN-2 System and Unit Status display

The System Status Pane displays critical information for every 3d MH device visible to your computer.

The Status Pane header lists the current MIOConsole version and MHLink Driver version. When the version number turns orange, there is a new version available: click to download and install.

The MHLink Domain "Root" box (your ULN-2 connected directly to the host computer) is listed next and also occupies the #1 spot in the **Unit Status Display** directly below. Click the sample rate indicator to the right select your system sample rate, and the clock icon to the left to set your clock source (currently defaulted to "Internal").

Below the ULN-2 icon is the box ID label and current firmware version. As with the Console and Driver displays in the Status Pane header, when the firmware version turns orange, click to download and install the new version.

Click on the **"Digital..."** label (circled in red below the firmware version) to open the **Digital I/O Status** window for this box.

You'll see a pop-up with that device's type and serial number, with Clock Lock status for each digital audio port and a selector to choose "ADAT" or "TOSLINK" as the output format for the ULN-2's built-in optical port. Note that the optical input port automatically senses ADAT/SMUX or TOSLINK formats and is independent of the output format.

This window will also identify any digital audio ports present on the EdgeCard™ (abbreviated "EB") installed in the unit.

Other status info shown the example above - the MHLink Top port is Connected, MHLink Bottom port is Unconnected and the USB port is Unconnected. We can also see that the sample rate on the DSP card is 48K, the current DSP load is 10%, and the small clock icon in the lower right indicates that this box is the Clock Master for the domain.

Each 3d box in your domain will have a similar Unit Status Display.

2. Directly below the Unit Status Displays is a currently empty field for Mute Groups controls, which we will get more into later.

3. The Link Groups section, including "Selected Strips", is at the bottom. With "Selected Strips" checked, shift- or command-click to select a few random faders in the mixer. You now have an "Selected Strips" link group of just those selected faders.

   With the "Rel" (as in "Relative") setting selected, it behaves just as one would expect - move one fader and the rest follow.

   Now click "Rel" and select "Inv" (for "Inverse"). In this mode moving one fader causes the other faders in the group to move in the opposite direction.

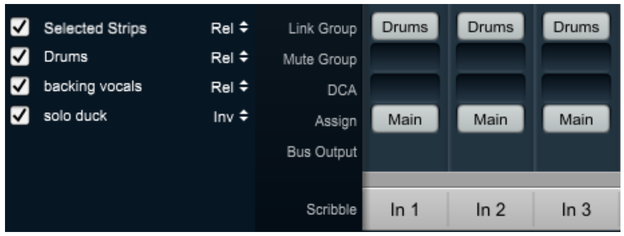
   

   Figure 3.5: Link Groups example

   Any Link Groups you create will be listed below "Selected Strips" in this section and can be assigned using the Link Groups selectors in each mixer strip (as illustrated in the example above). Saved Link Groups have additional operational modes, which will be detailed in the dedicated Link Groups section of the manual.

   The visibility of the System Status and Link Groups section of the 3d Console interface can be toggled with the " [" key (or the "Mixer > Show System Status Pane" menu command).

4. To the right of the System Status Pane is the primary default Mixer window.

   There are three "Mix Panes" available to help organize your mixer. Each Mix Pane can show any combination of Input strips, Bus strips (Main, Aux and Group) or DCA strips. Mixer strip layouts (i.e.: which mix desk strip controls are shown) are configured independently for each Mixer Pane, and are controlled from the "hamburger menu" icon in the upper left corner of each Mix Pane. This lets you set up the strips in one Mix Pane for tracking and the same channel strips in another pane for mixing. Additionally, each Mix Pane has it's own independent scroll bar.

   In the default view, only Mix Pane 1 is set up and visible - Mix Panes 2 and 3 are currently unconfigured and are accessible by the two "Hamburger Menus" at the far top right of the Mixer window.

   Click on the Mix Pane 1 hamburger menu and uncheck "Show Input Strips" - that hides the Input strips in Mix Pane 1. Now bring them back, and select "Configure Channel Strip Elements...". This sub-window let's you show or hide any of the control elements for the strips within that Mix Pane.

   Click on the Show/Hide box for each of the Strip elements so you can see what happens. This feature allows you optimize your Mixer layout for whatever workflow you may require.

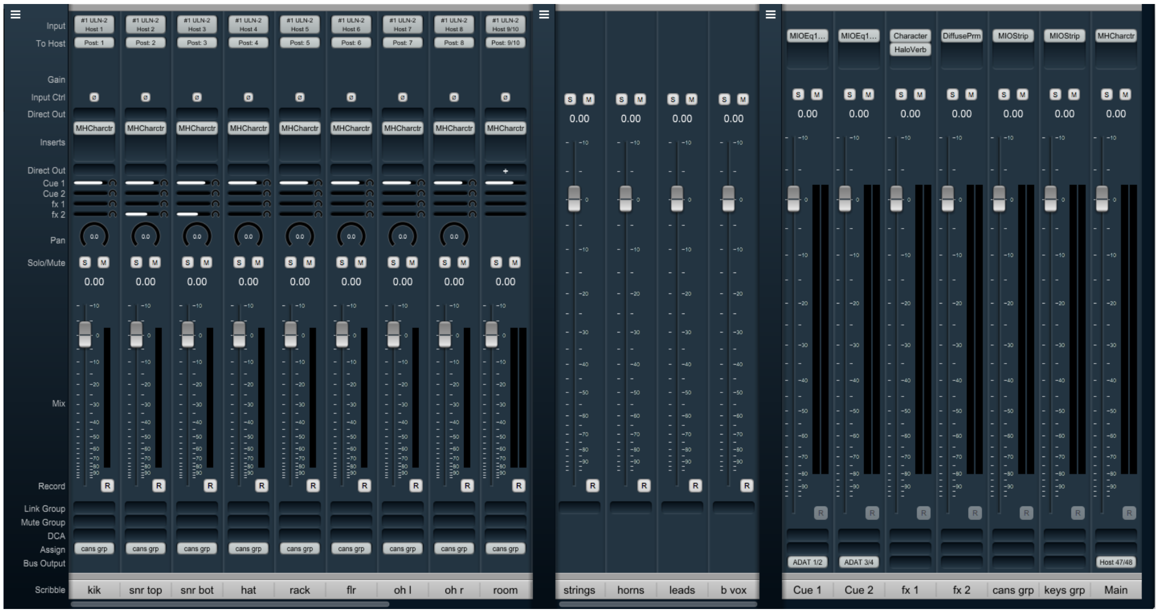


Figure 3.6: ULN-2 Mixer Panes example

In the example above we have Input strips in Mix Pane 1, DCA strips in Mix Pane 2 and Cue/Effects Aux buses, submix Groups and the Main master bus in Mix Pane 3. As you add Input strips, Aux buses, Group buses and DCA strips to your mixer, tweaking the strip layout across the three Mixer Panes helps you maximize the efficiency of your control surface while conserving valuable screen space.

5. In the upper right of the default layout is the Monitor Controller. This is a separate resizable window from the main Mixer window which can also be set to “float” so it is always visible on your display. Across the top of the Monitor Controller is the Monitor Source selector, showing the Main master bus and the four Aux buses from your default Mixer layout.

   If you are primarily only interested in using the ULN-2 stereo monitor outs, you will probably just leave the Monitor controller set to “0”, close the MC window and use the front panel control... but just in case, here's a run-down of how it works.

   All Monitor Controller Sources and Outputs are completely user-configurable - these are just the default settings.

Figure 3.7: Monitor Controller

Below the Source selection buttons is the speaker and volume level control/display.

The new 3d Monitor Controller is a vast upgrade to previous versions with its own dedicated DSP processing path and supports all channel configurations from Mono through Dolby Atmos™ 7.1.4, and although the ULN-2 is primarily a stereo device in the analog domain, it is possible to work in multichannel by using the optical I/O or EdgeCard interfaces.

There is a wealth of knowledge in the Monitor Controller section on how to set up and use the Monitor Controller.

The Monitor Controller window can be hidden by toggling the " m " key. Hold the Option key to engage scroll wheel/two finger scroll control of Monitor Level control. Option-Click returns controls to the default setting.

6. Below the Monitor Controller is the Cue Controls window. This window will expand downwards as you add more Cue Sends.

   You can think of Cues as named sends to physical outputs; they provide advanced routing controls to allow you feed them with the MC source, dedicated busses (for cue mixes) and with a talkback or listen back signal.

   The Cues give you top level control of the physical output level for each Cue. The controllable routing logic allows you to easily control the source and level of the signals routed to the cues which allows you to use them as a monitoring console for live monitoring and/or as part of a send system for bus output routing.

   By default, there will be one Cue that mults the selected input in the MC and routes to the Headphones of the root box. This mults the selected MC source to both the Monitor output and headphones. You can think of this Cue as your Control Room headphones.

The Talkback Source at the top of this window can be set to any mono audio input. "Listenback" sends the currently selected Monitor Controller source to all Cue Sends when selected. Multiple independent Cues Sends are fully supported.

A logical place to start would be to name the first two Aux mix buses "Cue 1" and "Cue 2", then add more Cue and effects mix buses as you build up your session; this allows you to provide independent monitor mixes to performers during tracking.

Probably the most straightforward application of Cue Sends in an MHLink domain would be have an MHLink box at each musicians' station with one Cue mix output assigned to the headphone jack of each box. That way each person has direct monitoring and their own independent volume control knob with plenty of inputs for their mics and instruments, and it all still appears to the DAW as a single audio device.

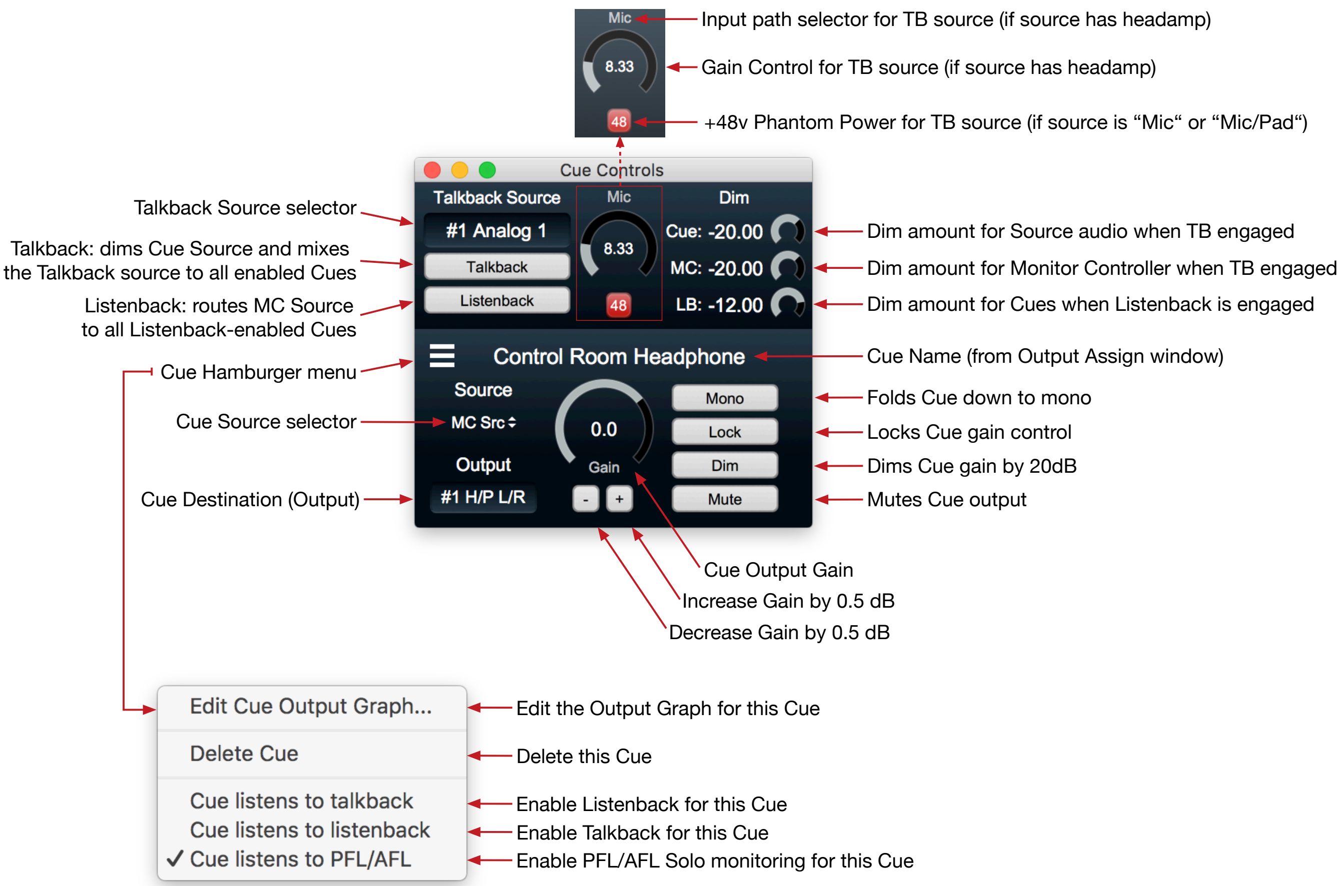


Figure 3.8: Cue Controller

To add a Cue, hit the "p" key, or go to the Monitor menu and select "Add Cue Controller". You can delete a Cue from its hamburger menu. The Cue Controls can be hidden by toggling the "c" key. Hold the Option key to engage scroll wheel control of Talkback and Cue Level controls. Option-Click returns controls to the default setting.

Check out the Cue Controller section for more details.

The Mixer window

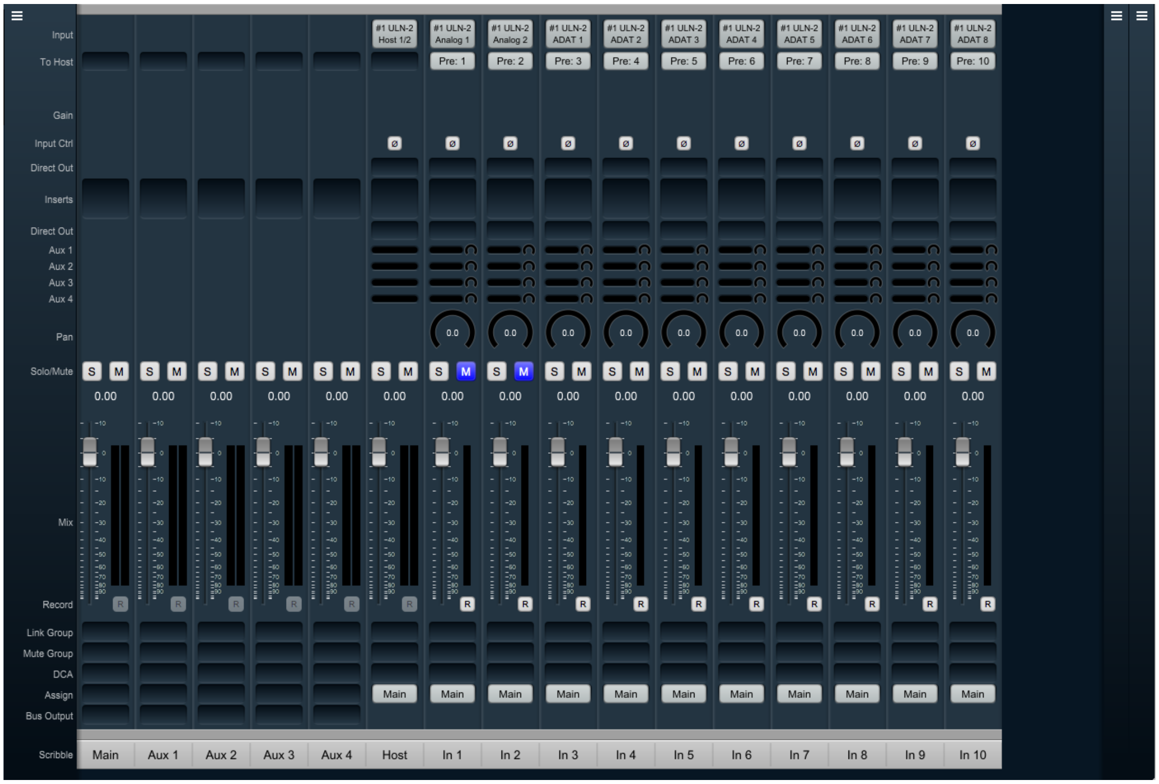


Figure 3.9: Mixer Window

The Mixer window presents you with the primary mixer interface to the ULN-2.

Moving from left to right, the default Mixer surface has a stereo "Main" bus, stereo Aux buses 1 through 4, and nine Input channel strips. The first input strip is the "Host", which is a stereo mixer input routed from channels 1 and 2 of your host computer (ie.: channels 1 and 2 from your DAW). Input strips In 1 and In 2 are routed from the hardware analog inputs from your ULN-2. For illustration purposes we have included the ADAT optical inputs on strips 3 through 10.

Let's look at these input strips first.

Input Strips

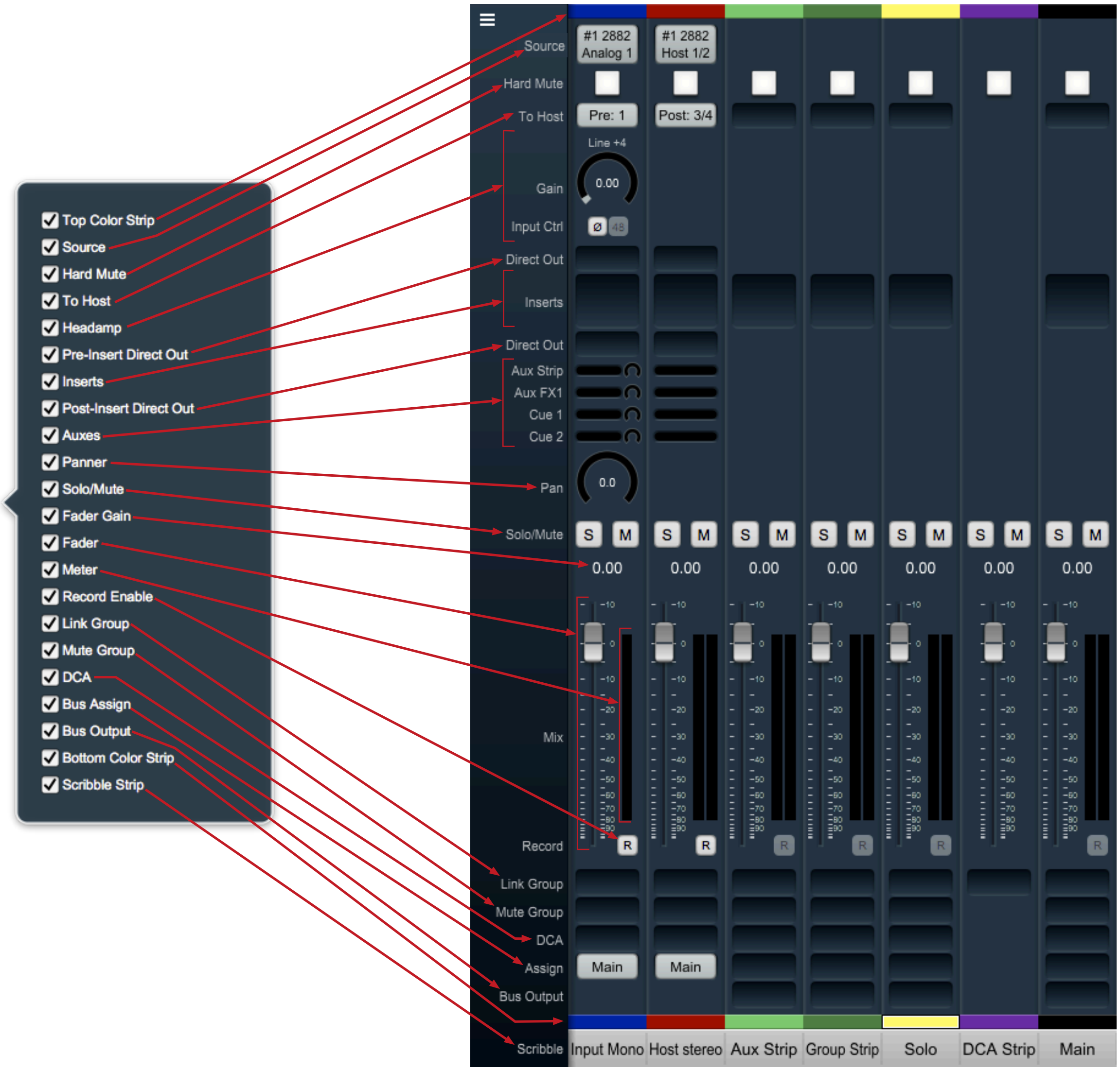


Figure 3.10: Mixer Strip elements exposed

At the top of every input strip is the Input Source selector. Click on it to see the Input Source Selector window. Within the Input Source Selector window you can choose the type of inputs you want to select from, show all the inputs from all boxes in the domain or only one box, how many channels you want your new input strip(s) to include, and how many strips you want to create (or modify) with Cascade.

“Hard Mute” is a kill switch at the source of the desk strip designed for live sound applications. “Hard Mute” is hidden and disabled by default.

Analog inputs get a head amp where you can select the input path type, invert signal phase, toggle phantom power and set input gain.

Below the input selector is the “To Host” menu. This lets you set pre- and/or post-insert return paths directly to your Host computer. Since the 3d MIOConsole3d mixer is designed to support hundreds of hardware and Host computer i/o channels, “To Host” is optimized for fast setup of large mixer layouts, and will automatically assign the *next available Host return channel*.

Below the head amp is the pre-insert Direct Out. The Direct Outs can send signal to any physical analog or digital output, as well as provide additional returns back to the Host computer.[1]

Below the Pre-insert Direct Out are the Inserts slots. Here you can insert plugins, route to sends (additional buses), add hardware I/O inserts, call up macros (previously-saved DSP processors) and instantiate custom DSP graphs. The Inserts section expands as you add more processors up to a maximum of ten per strip. To delete an insert, open the Insert selection window and select "None". *Note:* For users upgrading from earlier MIO versions, "Character" has been moved to the Inserts section.

Below the Inserts section is a Post-insert Direct Out. Any signal routed from this Direct Out includes all of your insert processing.

Aux send controls appear below the post-Direct Out. Each Aux has a send level control and (where appropriate) a panner to the bus on each strip.

Aux buses can be named in the Configure Mixer window or by double-clicking the scribble field at the bottom of the Aux return strip.

For more precise control of the send levels to an Aux bus, click the name of the Aux bus in the strip legend to the left of the mixer pane. A dot will mark the selected bus name, the gray fader knobs in the main mixer surface will become yellow, and the Aux mix will move onto the main surface.

So, yellow faders in your mixer means you are looking at and controlling an Aux mix, not your main mix. Click the dotted Aux name in the legend again to return to the main mix desk. *Hint:* Aux buses are also used for Cue mix sends.

Panners are next. The panner type configures automatically to match the channel width of the strip to the bus that it is assigned to:

- There are no pan controls on a channel assigned to a mono bus.
- Mono input channels will have a pan knob when assigned to stereo or LCR buses.
- Mono input channels assigned to LCRS through 7.1 buses will have joysticks. Right-clicking on the joystick will allow you to hard assign the input channel to a specific output channel, i.e. Center, Left, etc.
- Multichannel inputs (stereo and above) have no pan control, and are direct routed to the matching channels in the bus.

Keep in mind that the while ULN-2 is primarily a stereo device in the analog domain, it is possible to work in multichannel by using the optical I/O or EdgeCard interfaces.

The fader/meters section dynamically expands and contracts based on the size of the Mixer window and the number of control elements currently visible in the strip. The number of meters will always reflect the number of channels handled in the strip. Meters display "pre-fader" by default, and can be set to read "post-fader" signal in the MIOConsole3d Mixer menu.

All input and bus strips have Solo, Mute and Record Enable buttons.

Please note that 'Record Enable' is active only if there is audio routed from that strip back to the Host computer either via "To Host" or a Direct Out (the audio you want to record must be routed back to the Host or the Host computer has no way to record it to the hard drive).

[1] *Hint regarding Direct Outs:* Use Direct Outs whenever you need to route to a specific-numbered return channel to the Host. "To Host" is designed to automate the addition of many channestrips in a single step and is sometimes just not the right tool for the job. Using Direct Outs (often coupled with 'Selected Strips' groups and Cascade) to route your Host returns allows you to precisely target specific channels or groups of channels from the MIOConsole3d mixer to your DAW.

Below the fader/meters section are controls for managing assignments for Link Groups, Mute Groups and DCA groups, with Bus Assign selectors (all currently defaulted to the Main bus) as the bottom controls on the input strip. The Assign control is for routing that strip the Main bus and/or any available Group bus.

At the very top and bottom of each strip is a color band (default color: gray) for differentiating strips at a glance. Click on a color band to open the color selector. Setting the color of a strip will apply to multiple strips at once based on either a 'Selected Strips' group or the assigned link group. (Note: Strip colors can be applied to the entire strip as a preference.)

See the Input Strip Details section of the MIOConsole3d overview for more information.

Bus Strips

In the default mixer configuration you have the one stereo Main "master" bus and four Aux bus return strips. These bus strips all have To Host (post-fader only), Inserts, Solo, Mute, Record Enable, Link Group, Mute Group, DCA, and Assign control elements which all work just like the Input strips.

Aux, Group and Main buses have an additional routing selector below "Assign" on the strip called "Bus Output". Use Bus Output to route bus strip outputs to any hardware port or as additional post-fader return routes to the Host computer.

There are two other types of buses which are not shown in the default mixer, but are shown in the "Mixer Strip elements exposed" graphic: Group buses and the Solo bus. The Solo bus appears when a PFL or AFL Solo mode is chosen in Configure Mixer (default is Solo-In-Place). Group buses are also created in Configure Mixer. We will get into more detail on MIOConsole3d busing architecture later but the general flow is:

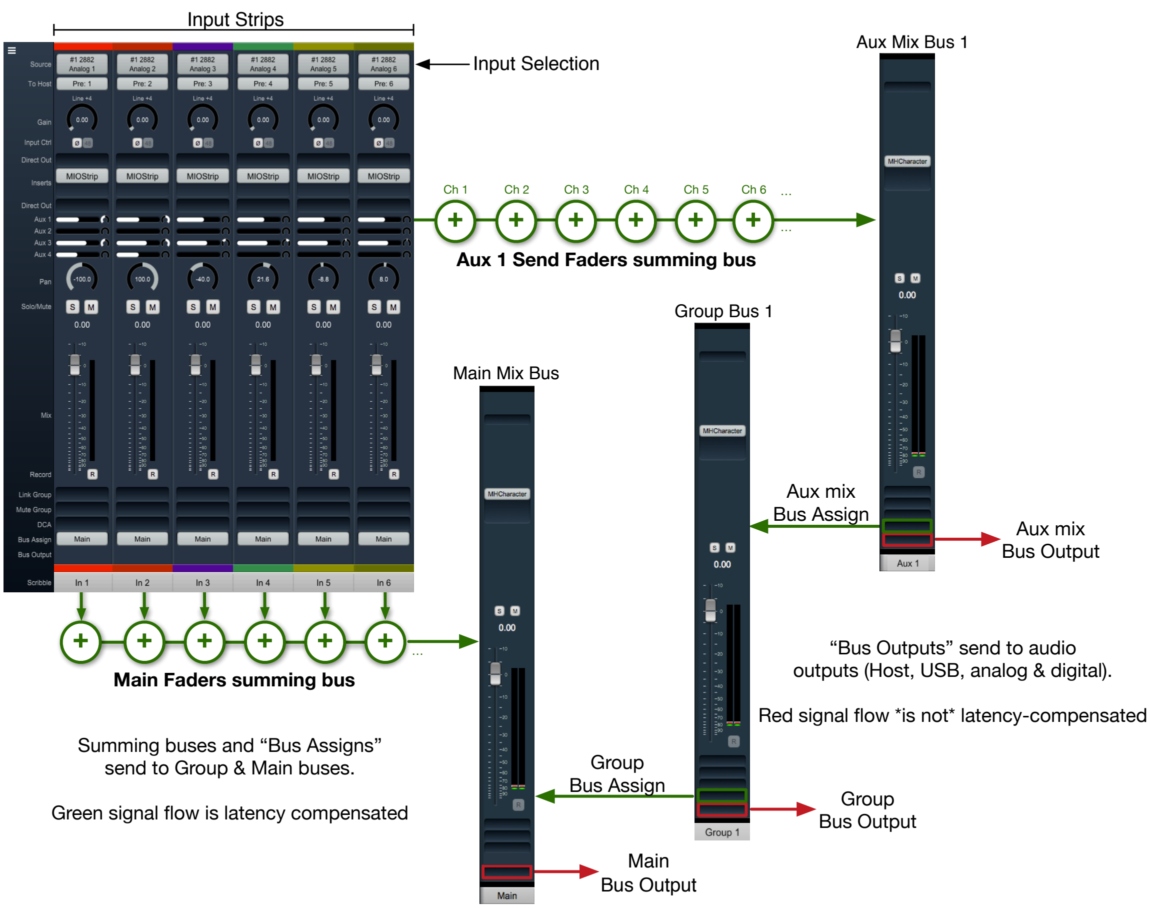


Figure 3.11: Basic 3d Mixer signal flow

- External audio sources enter the 3d mixer through Input strips. Input strips can send to any output, bus or both. Input strips are internally latency-compensated from their input through their assigned output bus.
- Aux buses include their own fader, pan and mute for each Input strip independently of the Main mix bus. Aux buses are for building a completely independent mix and can be used for effect sends (to be returned to a Group or Main mix bus), or for independent cue mixes (to be routed out to the world). Aux buses are internally latency-compensated when assigned to Group buses or the Main mix bus.
- Group buses are sub-mix buses. All the buses assigned to an input strip (Main + any groups) all share the same fader, mute, and pan settings. Group buses can be routed to hardware ports out to the world, to other Group buses, or the Main mix. Routing through Group buses within the mixer will be internally latency compensated.

The fact that the buses are latency compensated means that you can sum channels to any of the buses, and if you assign one bus to another bus the shared signals will be time-aligned and will not introduce any phasing. You can utilize this to do parallel compression on submix buses or sum in a reverb bus that includes a dry component.

Routing

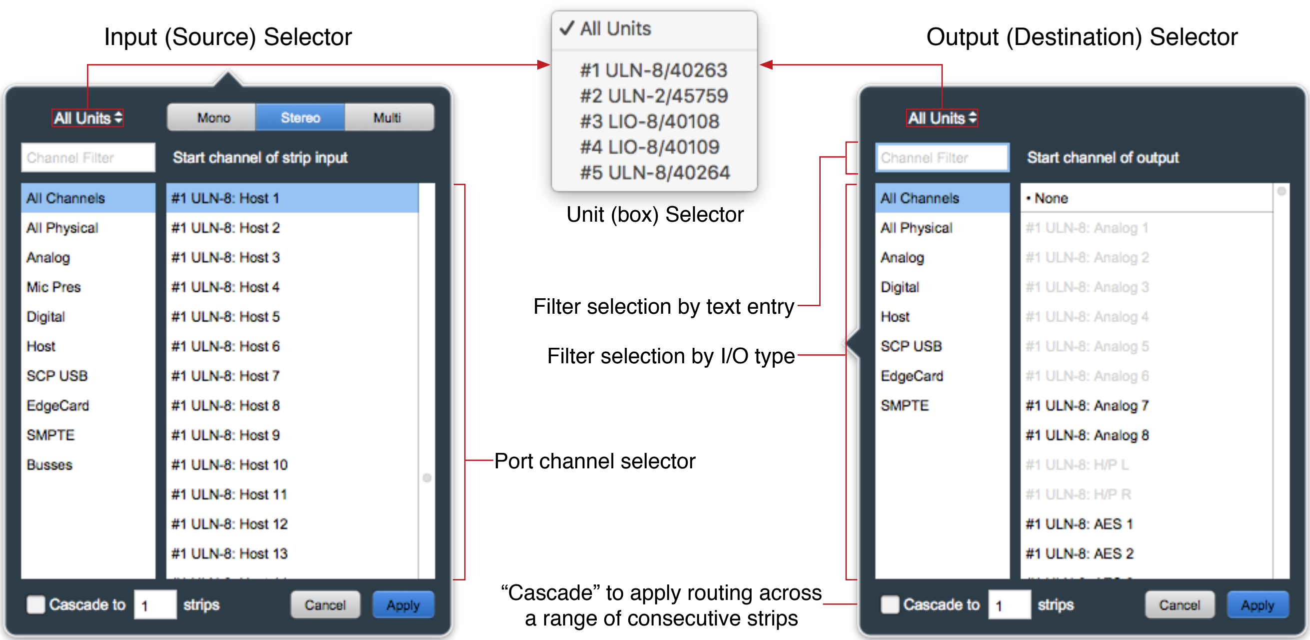


Figure 3.12: Input source and Output destination routing selectors

Since 3d/MHLink domains are completely dynamic (in that you can add or remove boxes live, and up to 100m between...) the Input and Output routing selector UIs both offer filter-by-box, filter-by-port-type and text-entry search tools to help manage the madness. Use the "Unit Selector" when you know which box you need when they are in different locations, select by the port type when the boxes are clustered in a rack, or combine search filters at will. Text-entry searches are dynamic - type "tos" and you are presented with every TOSLINK port visible in the domain.

In the above graphic, note that some of the output ports in the Output selector on the right are greyed-out and unavailable. In the case shown, those ports are occupied as a 5.1 Monitor Controller output and a stereo Cue send, both to the ULN-8 root box #1. Automatic direct mults to outputs are not currently supported in MIOConsole3d, but you can always use a Group or Aux bus to accomplish the same thing.

Two things to be aware of when routing in the 3d mixer:

- Routing from a bus output to an Input strip is allowed, but as mentioned in the previous section, will not be automagically latency-compensated.
- Since routing search filters can be combined, typing “tos” to find a TOSLINK port while the “Analog” port type filter is selected will return zero results (because, after all, TOSLINK is a digital port). Best practice is to select “All Channels” or “All Physical” (which includes all regular analog and digital hardware ports, but not Host computer and SCP USB ports) when searching by text.

> For more information regarding configuration and routing of ADAT/TOSLINK, MADI and USB ports see the Digital I/O Status / Control Menu and USB Port Status and Configuration sections of the main manual.

Using the Mixer

You should now be in a good position to integrate the ULN-2 with your audio workstation. Select “MHLink Audio” (or, if you are connecting via USB, "ULN-2") as your audio interface in your DAW's hardware configuration. To play from your DAW, send your DAW’s signal to outputs 1/2 and it will also come into the ULN-2 on the Host 1/2 strip. Hit “Play” in your DAW and enjoy.

Now that you have some tunes going, you might feel like messing around with some insert processes and see what this thing can really do. To get started, go to your Host input strip and click in an Insert slot. The dialog that appears list categories for "Plugin", "Macro", "Graph" and "I/O".

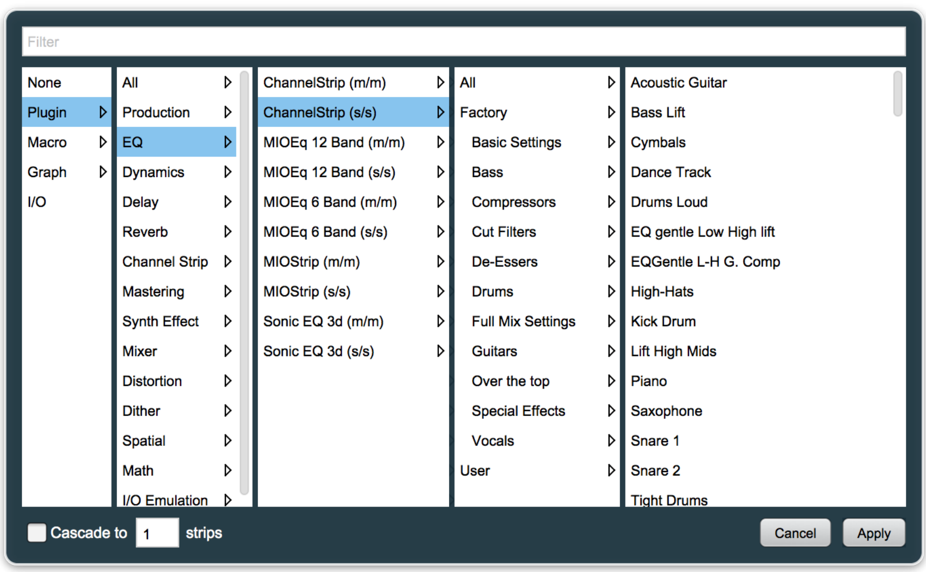


Figure 3.13: Mixer Strip Insert: Plug-in selector

- Plugin: Includes the 3d-native MH Production Bundle, Sonic EQ 3d and many very useful processor “building blocks”. Click the *MH Production Bundle* link above to learn about sharing presets between your Metric Halo AU/VST plug-ins and the 3d DSP versions.
- Macro: Most are essentially presets of more complex process chains in a DSP Graph. Feel free to use these as a starting point to create your own personal custom signal processors. The Reverb macros can not be represented as modules in a normal graph, and can not be modified as-is.
- Graph: A graph is a “DSP playground” where you can build your own signal processing chains and save them for later recall. The MIOConsole3d Graph is arguably the most powerful audio hardware

DSP environment available to the public. Creative users have built custom stereo to 5.1 spatial upmix processors, analog tape noise reduction, and multi-layered FOH speaker array time delay and acoustic correction tools, to name just a very few. Graphs can be very simple - just build a quick little tool to do some one-off task not gracefully handled by a packaged plug-in… Graphs are especially spectacular for creating new types of parallel processors because no matter how complex the processing it is all fully internally latency-compensated.

- I/O: A basic signal path insert-within-an-insert, if you will. From within any strip, send to any hardware output or the Host computer and return from any hardware input or the Host. I/O is very useful for an extra Direct Out, as a sidechain source, or for mults to secondary systems (external backup recorder, video truck feed...), etc.

  Note: Since an I/O send/return insert sends audio to external systems outside the mixer DSP and back, I/O loops can not be automatically latency-compensated.

As a reminder, all of the mixing, gain control and processing which is shown and accessed through the MIOConsole3d is operating solely in the box(es) and not on the computer, even though the user interface for the 3d Mixer is controlled from the computer display. This is an important distinction to keep in mind so you can properly manage your sessions and take best advantage of the strengths of both the DAW and the 3d hardware.

Now let's walk through adding the ULN-2's eight ADAT channels into the MIOConsole3d mixer and your DAW. This procedure works with any kind of input from any box in the domain.

First hit command-shift-A to create a new input strip. In the routing window that pops up, select "Digital" from the filter list on the left (to show only the digital inputs), select "ADAT 1", type "8" in the Cascade box at the bottom of the routing window and hit Apply (or Return or Enter).

With these few clicks, you have instructed the Mixer to create eight new input strips assigned starting at ADAT Input 1 (i.e.: ADAT inputs 1-8), and your mixer should now look like the one in the example above.

The Mixer defaults to automatically assign routes to the Host computer for each strip, starting with the next available To Host channel... so, since the two analog input strips are already using To Host routes 1 and 2, these new strips will use To Host channels 3-10.

Now, go back to your DAW, create eight new input strips in your DAW mixer and assign them to input 3-10. You have built a virtual patchbay routing audio between your DAW and the ULN-2. You can route pre- and post-insert Direct Outs, Aux buses, Group buses and the Main bus back to the Host DAW in the same way.

To disable any route, just open the routing window and select "None" at the very top of the strip inputs list, or mouse over the strip inputs button and click the "X" to delete that route and free it up for another task.

Mixer Strip Color Bars

Clicking on the Color Bar at the top or bottom brings up a macOS color selector, from which you may choose a color for that Mixer strip. In the Appearance pane of the MIOConsole3d Preferences you may select whether to color the entire mixer strip or just the top and bottom color bars.

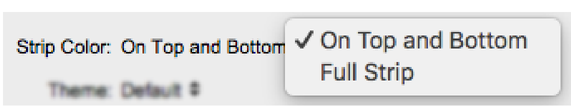


Figure 3.14: Console3d Prefs: Appearance: Strip Color configurator

The Bottom and Top Color Bars mirror each other, so what applies to one applies to the other. You may show or hide the opt or bottom color bars independently with "Configure Mixer Strip Controls...".

Scribble Strip

Double-click the text field at the bottom of any strip to enter the strips name. The name you enter in the Scribble Strip will be propogated throughout the 3d user interface wherever a Mixer Strip name is found, including plug-in headers, Link Group, Mute Group, DCA and Bus Assign selection windows.

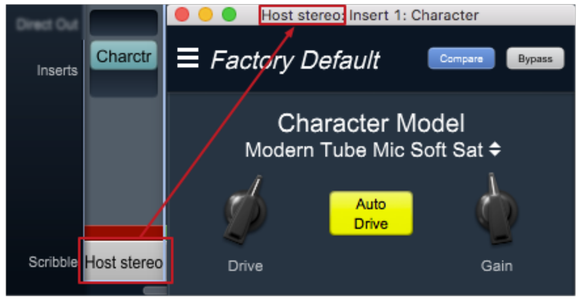


Figure 3.15: Scribble Strip name to Mixer strip Insert header

Scribble Strip names are also used to name sound files recorded in the Record Panel.

Mixer strip Insert controls

Once you have selected a plug-in, it will be listed in the assigned insert slot:

Figure 3.16: Inserted MIOEQ6 Plug-in (as shown in Mixer strip)

Plug-in names will generally appear abbreviated in order to save space ("MIOEQ6" is shown above).

When you move the mouse over an inserted plug-in, the Insert label will change to show three control icons. The tooltips for each of these controls have been exposed in the example graphic below.

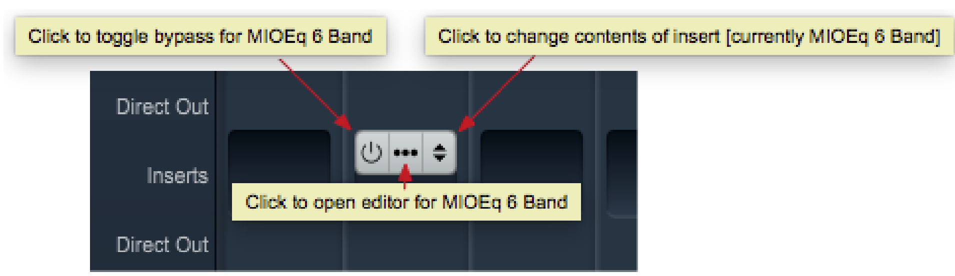


Figure 3.17: Inserted plug-in controls

- The "On / Off" switch icon on the left is the plug-in Bypass. When Bypassed, the Insert button will turn yellow.
- Clicking the "..." icon in the middle opens the inserted plug-in editor UI. When the plug-in editor is open/visible, the Insert button will turn blue.
- Clicking the "up/down" arrows icon at the right opens the Insert selector window, where you may select a replacement plug-in, or navigate to directly open a different saved preset without having to open the plug-in editor UI.

Insert Control modifier key shortcuts

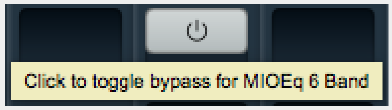


Figure 3.18: "⌘-click" / <Command>-click to Bypass Insert

<Command>-click the Insert button to Bypass the Insert.

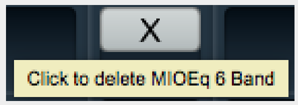


Figure 3.19: <Control-Option-Command>-click to Delete Insert

Use "^⌥⌘-click" / <Control-Option-Command>-click to delete an Insert directly from the Mixer desk.

Quick Copy/Paste Plug-ins

Option-click-drag any plug-in instance from one Insert slot to another anywhere in the Mixer to clone that Plug-in to the new location. Plug-in instances will automatically adapt to the channel width of the target Insert as necessary

"Sweeping" controls

Toggle buttons on consecutive strips in the 3d Mixer desk can be switched in a single move by a click-hold-sweep gesture.

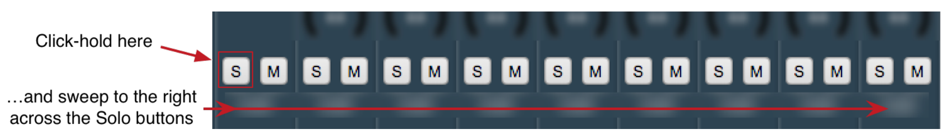


Figure 3.20: "Sweep" to toggle multiple buttons in one gesture

To try it, click on a Solo button, and while holding the mouse button down, drag the cursor to the right or left across the adjacent Solo buttons.

The move works with Polarity Invert, Solo, Mute and Record Enable buttons.

The ULN-2 and surround

If you are working in surround, it's easy to configure the ULN-2 for surround processing and monitoring. Let's take a minute to walk through the process:

First, go to the Mixer menu and select "Configure Mixer" (cmd-shift-C). This window is the main setup page for your Mixer layout. The Main Mix configuration is set at the top - it is currently set to "Stereo". If you click on the Main Mix selector, you can change it from "Stereo" to whatever width you need.

Let's say you're working in 5.1. Select "5.1 SMPTE/ITU" and hit OK… Voila! Your Main bus is six channels wide and all the mono input channels now have joysticks instead of pan knobs.

Your Host input strip is still bringing your DAW in on just two channels, so let's fix that too. Go to the top of the "Host" input strip and click on the input assignment pulldown. At the top of this window you have the choice of a mono, stereo or a multi-channel strip. Select "Multi" and click "Apply".

Your Host strip is automatically set to match the Main bus configuration of your mixer: in this case 5.1 SMPTE/ITU. Now your audio will come into the ULN-2 in SMPTE surround format as positioned by your host on channels 1 through 6.

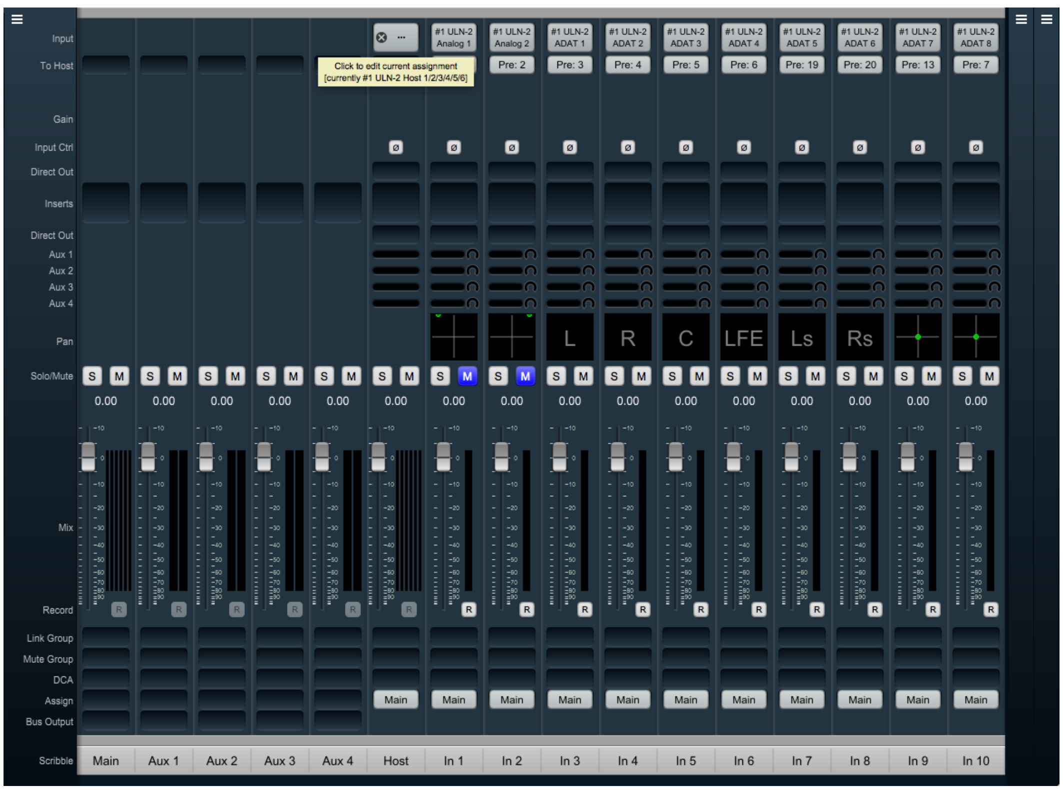


Figure 3.21: Surround Mixer

Your surround mixer should look pretty similar to this. You can right-click (or ctl-click) the panner on the first six mono input strips to hard-assign each strip to a particular speaker channel as is shown on the example above.

Note especially the mouse-over pop-up at the top of the Host desk strip. Ordinarily the text "#1 ULN-2 Host 1/2/3/4/5/6" would be too small to read, so in the interest of saving screen real estate these mouse-overs come in very handy.

If you can afford the space, use the "Mixer - Set Mixer Strips Width" menu to adjust your mixer desk strips' width to taste.

Surround Monitoring Setup

In order to monitor in surround, you need to create a 5.1 output in the Monitor Controller. Go to the "Monitor" menu and select "Add Monitor Output" to get the setup box. You'll see that the "Name" field is already selected, so go ahead and type "5.1" to name the monitor output.

The "Type" selector to the right of this window is where you choose your new Monitor Controller output configuration: Select "5.1 SMPTE/ITU".

To set up 5.1 Monitor Controller outputs, click the Left channel selector (currently set to "None") in the "Destinations" field. The routing box should appear as in the figure below.

If your speaker channels are all in ITU order, you can just double-click "ULN-2 ADAT 1" to automatically cascade L, R, C, LFE, Ls, Rs to channels 1, 2, 3, 4, 5 and 6 respectively. Otherwise de-select "Cascade" at the bottom of this window and assign each speaker output individually.

Click "Apply" to route the speaker assignments. Note that you can apply individual level trim and delay for each speaker from this menu, should you desire.

Click "Add" to confirm and establish the new Monitor Controller Output parameters and finish up. You can return to the Monitor Controller Output configuration screen any time by right-clicking the MC output selector, or going to the Monitor menu and selecting "Edit Current Monitor Output".

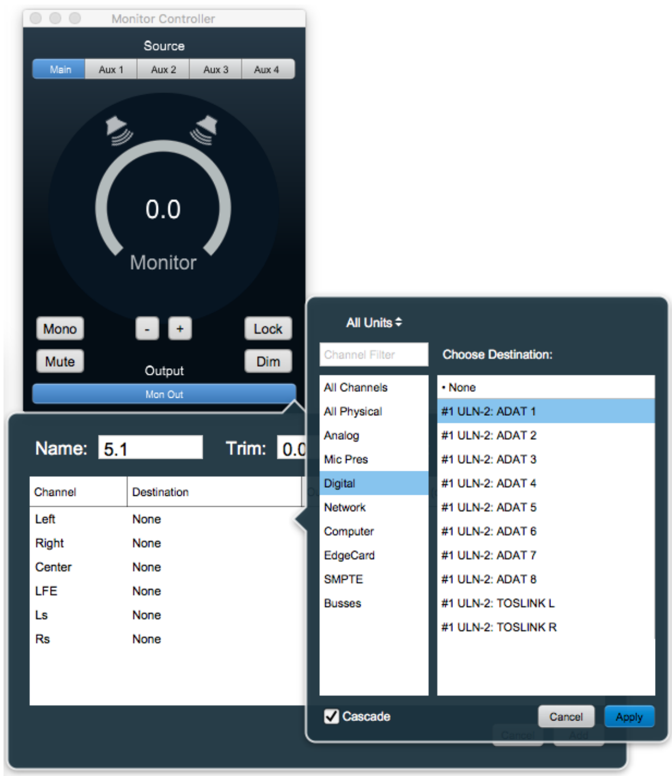


Figure 3.22: Monitor Controller speaker assignment

Unleash the DSP

The ULN-2 is based on the 3d processing engine and includes a very powerful arsenal of high-precision audio processing tools. All plug-ins and processors work in any channel configuration and include full parameter save and recall with lots of factory presets to get you started.

There are two ways to access the DSP:

- You can insert processes directly in the mixer strip inserts; this works well for standalone processes like eq, compressors, etc.
- Insert a graph. This lets you chain plugins together, use them in parallel, and create custom processors with routing configurations that would be very difficult (or impossible) with other platforms.

  Graphs are also available in the Monitor Controller and Cue Controller signal path. High-precision room EQ/acoustic correction, custom crossovers and bass management with none of the expense, inconvenience or limitations imposed by running such processes in outboard gear or a DAW monitor bus.

You should definitely check out the following processes:

- MIOStrip: Gating, EQ and compression powerhouse. Very clean, very transparent - put it up against any precision digital mastering hardware and be just a little stunned. Mix it with some Character for a tracking console vibe.
- Character and MHCharacter: The sound of different analog circuits and devices available on any input, output or bus. Both have manual and automatic drive and gain. Character is something of a late 80's/90's animal with an extra lo-fi bite. MHCharacter is geared more towards higher precision, warmth and detail enhancement.
- Haloverb: Great sounding reverb, doesn't use any CPU from your host. What's not to love?
- Transient Control: Super fast, super clean initial strike control.

You should read the DSP Implementation section to learn more about how to work with DSP, and the DSP documentation details the over 100 plug-ins available in the 3d DSP engine.

This should get you started with the ULN-2!

4. 2882 Quick Start Guide

Figure 4.1: Mobile I/O 2882

Prepare the unit for use

Unpack the 2882 and make sure all the parts are there:

- One Mobile I/O 2882 unit
- One IEC Power Cord appropriate for your area
- One 18-volt 60-watt world-ready external power supply
- One 12” CAT5e Ethernet Cable
- One 14' CAT5e Ethernet Cable
- One USB-A to USB-C Cable
- Two Rack Ears w/ fasteners
- Rubber feet
- Warranty/Registration Card

Welcome to your new Metric Halo 2882 3d! Once you’re finished checking the box physically and installing the rack ears (or rubber feet), connect the power supply, and connect your monitors to Analog Out 1/2. Many people find it useful to put a pad between their amps or powered monitors and the 2882’s outputs; this allows the 2882 to work at full digital resolution without overdriving the amplifiers.

> ***Super QuickStart!***
> Plug your computer into the USB-C port on the back of the 2882.
>
> Select “2882” as your computers’ audio device.
>
> Done.
>
> The 2882 will receive and send audio from/to channels 1 through 12 of your USB host device at all resolutions up to 192kHz/24 bit. Audio channels 1 and 2 of your host device will play audio out both the headphone jack and the analog channels 1 and 2 jacks on the rear panel.
>
> *Note:* While this is indeed the fastest way to get audio going, it is also the most boring. Connecting your Host computer through MHLink is by far the more powerful and expansive interface (and certainly the preferred method), but rest assured the USB connection will take on a much more interesting role later in the Advanced section. Please, do read on.

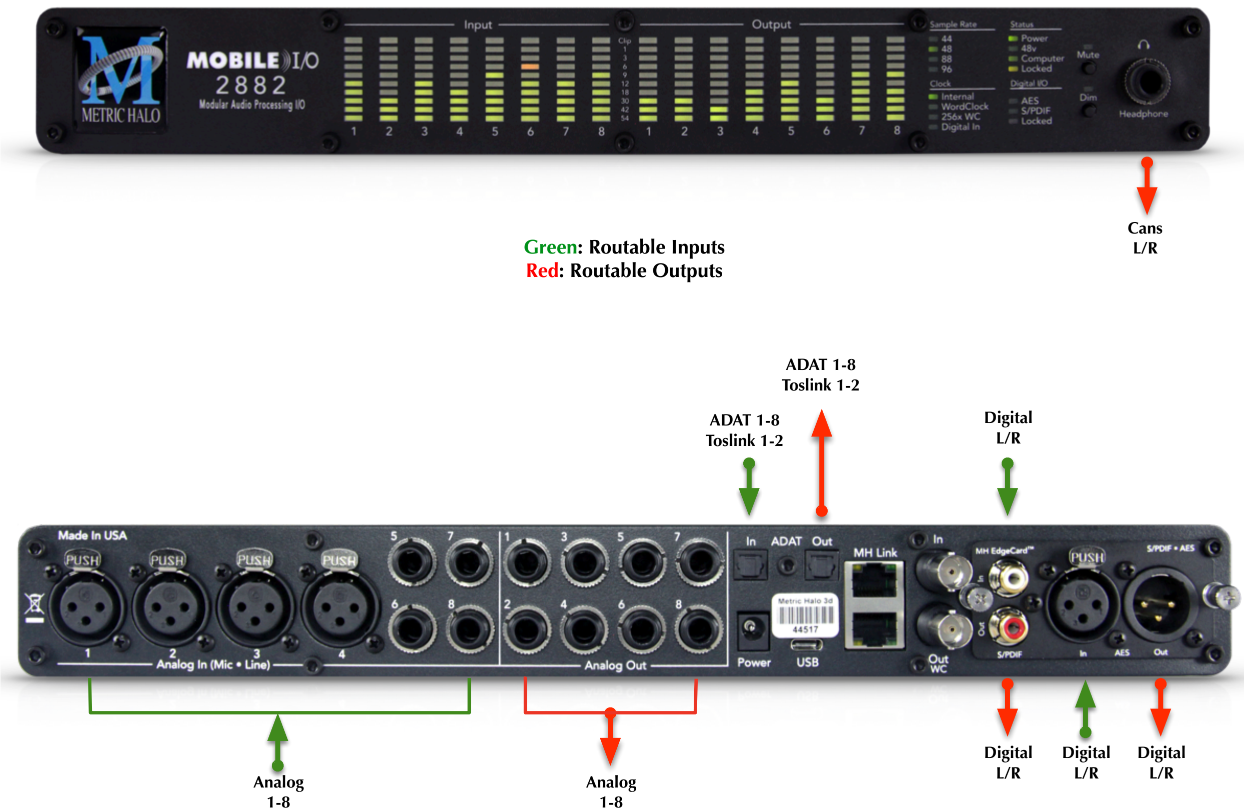


Figure 4.2: 2882 Routing

Connect the 2882

Install the latest MIOConsole3d installer package from: macOS MIOConsole3d installers (this will require a restart). Please see the section on software installation if you need clarification.

> **Driver Installation on macOS 10.13 and newer**
> With High Sierra (macOS 10.13) Apple introduced another layer of driver (system extension) software security.
>
> In order for third-party drivers (system extensions) to be activated on new installs of High Sierra, the user of the computer (you) must explicitly approve the activation, after the installation has occurred.
>
> You must do this before you will be able to use your hardware via MHLink.
>
> The authorization process is the same for all third-party drivers. If you are unfamiliar with the process, please read our FAQ answer here: macOS Driver User Authentication.
>
> Additionally, with macOS 10.14 and 10.15, you must select "Yes" when asked to give MIOConsole3d permission to access the "Microphone". If in doubt, double-check this setting: go to the Apple menu: System Preferences.../Security & Privacy pane, select "Microphone" in the column to the left, and make sure that MIOConsole3d is checked.

Connect an ethernet cable between the 2882 and an available ethernet port on your computer, then go to the System Preferences and select "MHLink Audio" as the system's sound input and output.

Take a listen

Now we're ready to listen.

With MHLink, you need to launch MIOConsole3d to establish the routing between MHLink and the Analog Outputs on the box. The routing is established automatically as soon as you launch MIOConsole3d.

If you connect with USB, you don't need to launch MIOConsole3d to establish the output routing; it is setup automatically by the box itself.

Note: The 2882 default configuration is set to receive audio from channels 1 and 2 from your computer, and play audio out both the headphone jack and the analog channels 1 and 2 jacks on the rear panel.

Open iTunes, turn its volume slider down and play some music. Bring the volume up to a comfortable level. Enjoy simply listening to music.

Using Legacy Interfaces

If you wish to compare your new 3d interface with an older "2d" Firewire MIO interface, you can leave it connected as always and run them both at the same time.

All FireWire-based Metric Halo interfaces will operate as fully independent audio devices alongside your Metric Halo 3d Ethernet-based interfaces.

MIOConsole3d

Figure 4.3: MIOConsole3d 2882 default view

MIOConsole3d Default Window Layout

At first launch of MIOConsole3d with your 2882 you will be presented with the default layout shown above.

There are a few important things to look at here so let's take a few minutes to familiarize ourselves with the interface/GUI.

First of all, if the Mixer window takes up too much room or appears too small on your screen, select Mixer/Mixer UI Scale from the menu bar and adjust the visual scale of the mixer surface. Note that scaling does not change the size of all of the Mixer strip elements - the fader/meters section remains elastic and will shrink or stretch as other Mixer elements are resized, added or removed. Try selecting the 85% scale setting so you can see what it does.

Now resize the Mixer window by grabbing the window corner (or any of the edges) so you can see how the mixer elements react.

You can also change the width of mixer strips (wider strips are helpful to better see 8-channel surround meters) and show or hide any mixer strip UI element. We will explore these features in more detail below.

As you have probably noticed as you check things out in the MIOConsole3d, most elements in the Mixer, Monitor Controller and Cue Controls windows have mouse-over pop-up hints which display expanded details for the button beneath.

The regular mixer surface button labels will often be abbreviated to save space, but the mouse-overs will display all the details of each control setting. This is an especially important feature as you add more boxes to your system domain.

1. In the upper left side of the main mixer window you see the System Status Pane.

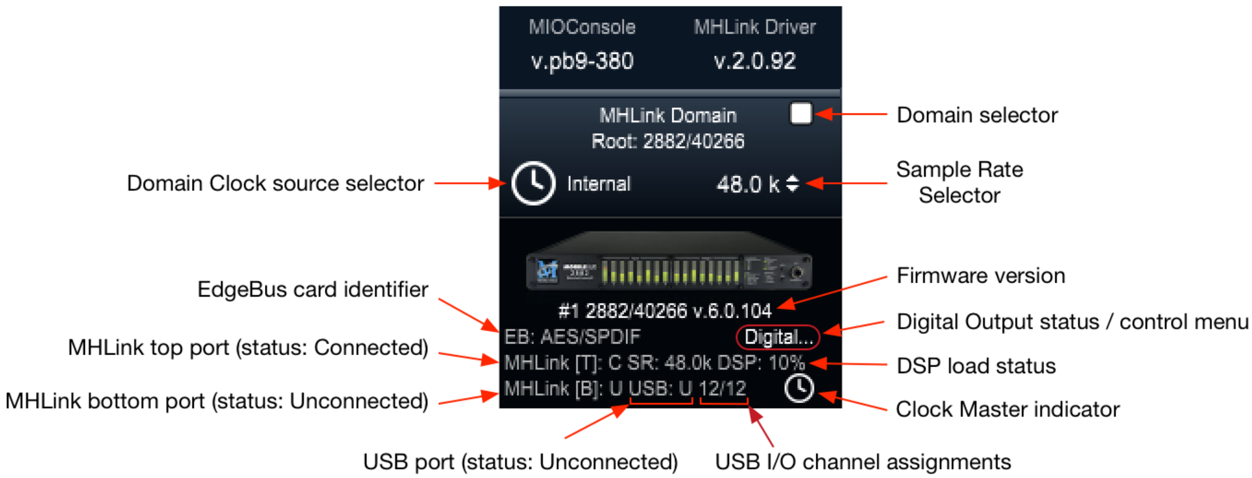


Figure 4.4: 2882 System and Unit Status display

The System Status Pane displays critical information for every 3d MH device visible to your computer.

The Status Pane header lists the current MIOConsole version and MHLink Driver version. When the version number turns orange, there is a new version available: click to download and install.

The MHLink Domain "Root" box (your 2882 connected directly to the host computer) is listed next and also occupies the #1 spot in the **Unit Status Display** directly below. Click the sample rate indicator to the right select your system sample rate, and the clock icon to the left to set your clock source (currently defaulted to "Internal").

Below the 2882 icon is the box ID label and current firmware version. As with the Console and Driver displays in the Status Pane header, when the firmware version turns orange, click to download and install the new version.

Click on the **"Digital..."** label (circled in red below the firmware version) to open the **Digital I/O Status** window for this box.

You'll see a pop-up with that device's type and serial number, with Clock Lock status for each digital audio port and a selector to choose "ADAT" or "TOSLINK" as the output format for the 2882's built-in optical port. Note that the optical input port automatically senses ADAT/SMUX or TOSLINK formats and is independent of the output format.

This window will also identify any digital audio ports present on the EdgeCard™ (abbreviated "EB") installed in the unit.

Other status info shown the example above - the MHLink Top port is Connected, MHLink Bottom port is Unconnected and the USB port is Unconnected. We can also see that the sample rate on the DSP card is 48K, the current DSP load is 10%, and the small clock icon in the lower right indicates that this box is the Clock Master for the domain.

Each 3d box in your domain will have a similar Unit Status Display.

2. Directly below the Unit Status Displays is a currently empty field for Mute Groups controls, which we will get more into later.

3. The Link Groups section, including "Selected Strips", is at the bottom. With "Selected Strips" checked, shift- or command-click to select a few random faders in the mixer. You now have an "Selected Strips" link group of just those selected faders.

With the “Rel” (as in “Relative”) setting selected, it behaves just as one would expect - move one fader and the rest follow.

Now click “Rel” and select “Inv” (for “Inverse”). In this mode moving one fader causes the other faders in the group to move in the opposite direction.

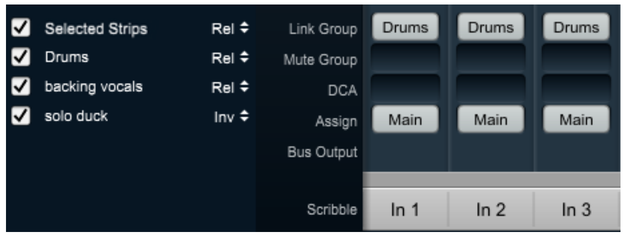


Figure 4.5: Link Groups example

Any Link Groups you create will be listed below “Selected Strips” in this section and can be assigned using the Link Groups selectors in each mixer strip (as illustrated in the example above). Saved Link Groups have additional operational modes, which will be detailed in the dedicated Link Groups section of the manual.

The visibility of the System Status and Link Groups section of the 3d Console interface can be toggled with the “ [” key (or the “Mixer > Show System Status Pane” menu command).

4. To the right of the System Status Pane is the primary default Mixer window.

   There are three “Mix Panes” available to help organize your mixer. Each Mix Pane can show any combination of Input strips, Bus strips (Main, Aux and Group) or DCA strips. Mixer strip layouts (i.e.: which mix desk strip controls are shown) are configured independently for each Mixer Pane, and are controlled from the “hamburger menu” icon in the upper left corner of each Mix Pane. This lets you set up the strips in one Mix Pane for tracking and the same channel strips in another pane for mixing. Additionally, each Mix Pane has it's own independent scroll bar.

   In the default view, only Mix Pane 1 is set up and visible - Mix Panes 2 and 3 are currently unconfigured and are accessible by the two “Hamburger Menus” at the far top right of the Mixer window.

   Click on the Mix Pane 1 hamburger menu and uncheck “Show Input Strips” - that hides the Input strips in Mix Pane 1. Now bring them back, and select “Configure Channel Strip Elements...”. This sub-window let’s you show or hide any of the control elements for the strips within that Mix Pane.

   Click on the Show/Hide box for each of the Strip elements so you can see what happens. This feature allows you optimize your Mixer layout for whatever workflow you may require.

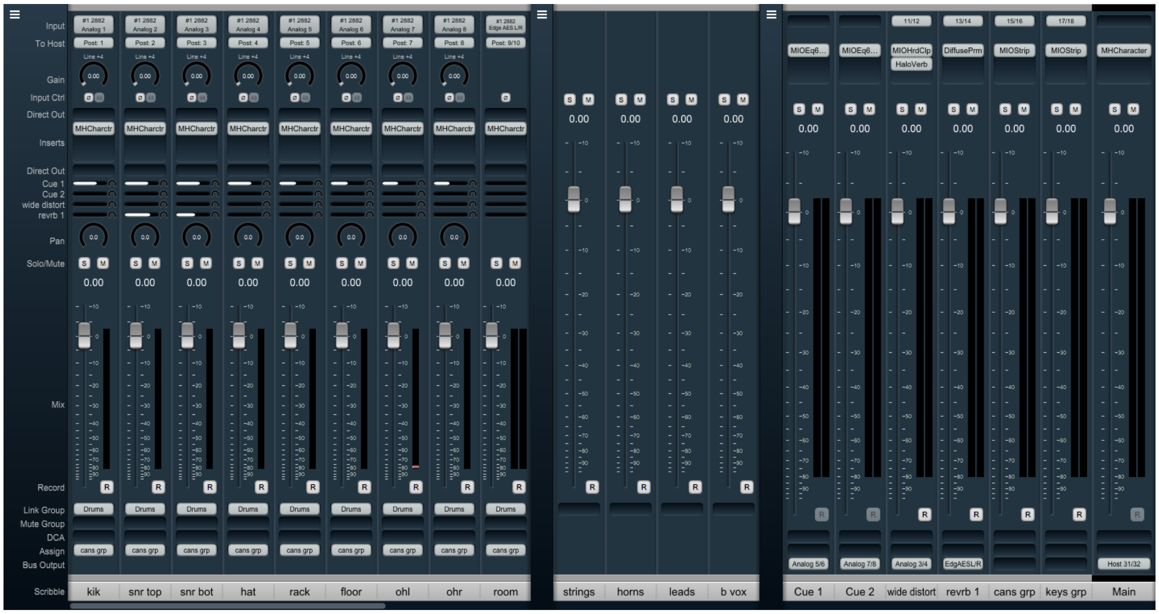


Figure 4.6: 2882 Mixer Panes example

In the example above we have Input strips in Mix Pane 1, DCA strips in Mix Pane 2 and Cue/Effects Aux buses, submix Groups and the Main master bus in Mix Pane 3. As you add Input strips, Aux buses, Group buses and DCA strips to your mixer, tweaking the strip layout across the three Mixer Panes helps you maximize the efficiency of your control surface while conserving valuable screen space.

5. In the upper right of the default layout is the Monitor Controller. This is a separate resizable window from the main Mixer window which can also be set to “float” so it is always visible on your display. Across the top of the Monitor Controller is the Monitor Source selector, showing the Main master bus and the four Aux buses from your default Mixer layout.

   All Monitor Controller Sources and Outputs are completely user-configurable - these are just the default settings.

Figure 4.7: Monitor Controller

Below the Source selection buttons is the speaker and volume level control/display. Since the 2882 does not have digitally-controlled analog gain stages like the LIO8 and ULN8, the default setting for the 2882 is line level out (i.e.: no signal attenuation).

The new 3d Monitor Controller is a vast upgrade to previous versions with its own dedicated DSP processing path and supports all channel configurations from Mono through Dolby Atmos™ 7.1.4.

There is a wealth of knowledge in the Monitor Controller section on how to set up and use the Monitor Controller.

The Monitor Controller window can be hidden by toggling the " m " key. Hold the Option key to engage scroll wheel/two finger scroll control of Monitor Level control. Option-Click returns controls to the default setting.

6. Below the Monitor Controller is the Cue Controls window. This window will expand downwards as you add more Cue Sends.

   You can think of Cues as named sends to physical outputs; they provide advanced routing controls to allow you feed them with the MC source, dedicated busses (for cue mixes) and with a talkback or listen back signal.

   The Cues give you top level control of the physical output level for each Cue. The controllable routing logic allows you to easily control the source and level of the signals routed to the cues which allows you to use them as a monitoring console for live monitoring and/or as part of a send system for bus output routing.

   By default, there will be one Cue that mults the selected input in the MC and routes to the Headphones of the root box. This mults the selected MC source to both the Monitor output and headphones. You can think of this Cue as your Control Room headphones.

The Talkback Source at the top of this window can be set to any mono audio input. "Listenback" sends the currently selected Monitor Controller source to all Cue Sends when selected. Multiple independent Cues Sends are fully supported.

A logical place to start would be to name the first two Aux mix buses "Cue 1" and "Cue 2", then add more Cue and effects mix buses as you build up your session; this allows you to provide independent monitor mixes to performers during tracking.

Probably the most straightforward application of Cue Sends in an MHLink domain would be have an MHLink box at each musicians' station with one Cue mix output assigned to the headphone jack of each box. That way each person has direct monitoring and their own independent volume control knob with plenty of inputs for their mics and instruments, and it all still appears to the DAW as a single audio device.

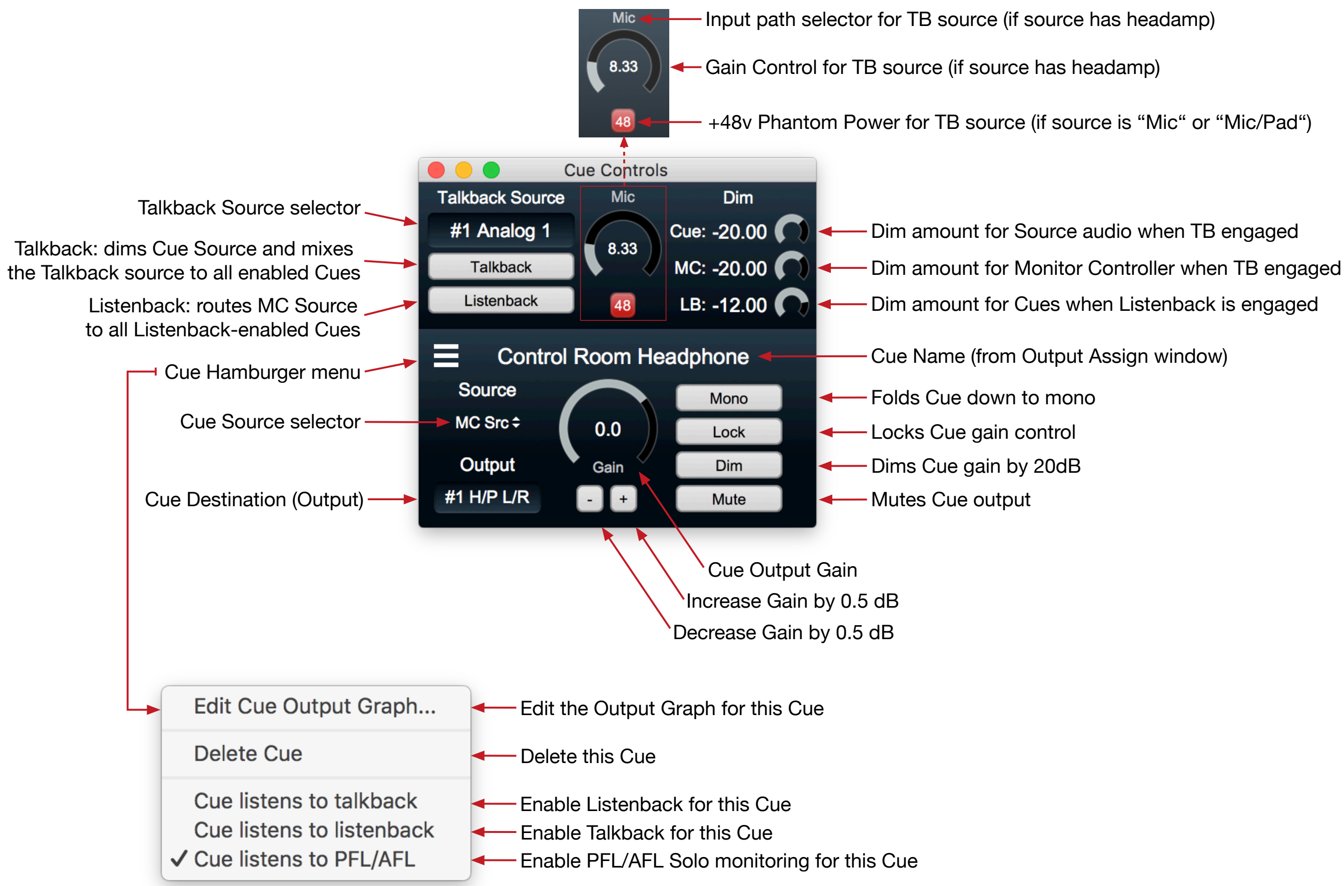


Figure 4.8: Cue Controller

To add a Cue, hit the "p" key, or go to the Monitor menu and select "Add Cue Controller". You can delete a Cue from its hamburger menu. The Cue Controls can be hidden by toggling the "c" key. Hold the Option key to engage scroll wheel control of Talkback and Cue Level controls. Option-Click returns controls to the default setting.

Check out the Cue Controller section for more details.

The Mixer window

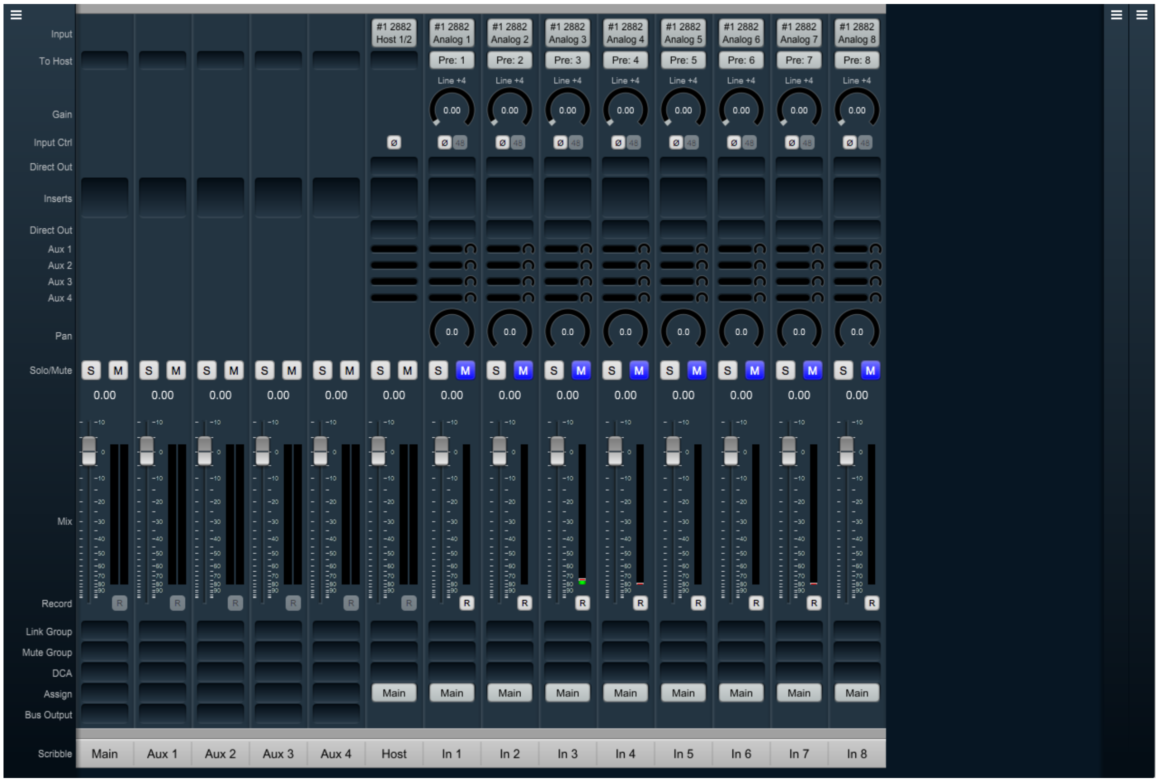


Figure 4.9: Mixer Window

The Mixer window presents you with the primary mixer interface to the 2882.

Moving from left to right, the default Mixer surface has a stereo "Main" bus, stereo Aux buses 1 through 4, and nine Input channel strips. The first input strip is the "Host", which is a stereo mixer input routed from channels 1 and 2 of your host computer (ie.: channels 1 and 2 from your DAW). The remaining eight input strips are all mono, and are routed from the hardware analog inputs from your 2882.

Let's look at these input strips first.

Input Strips

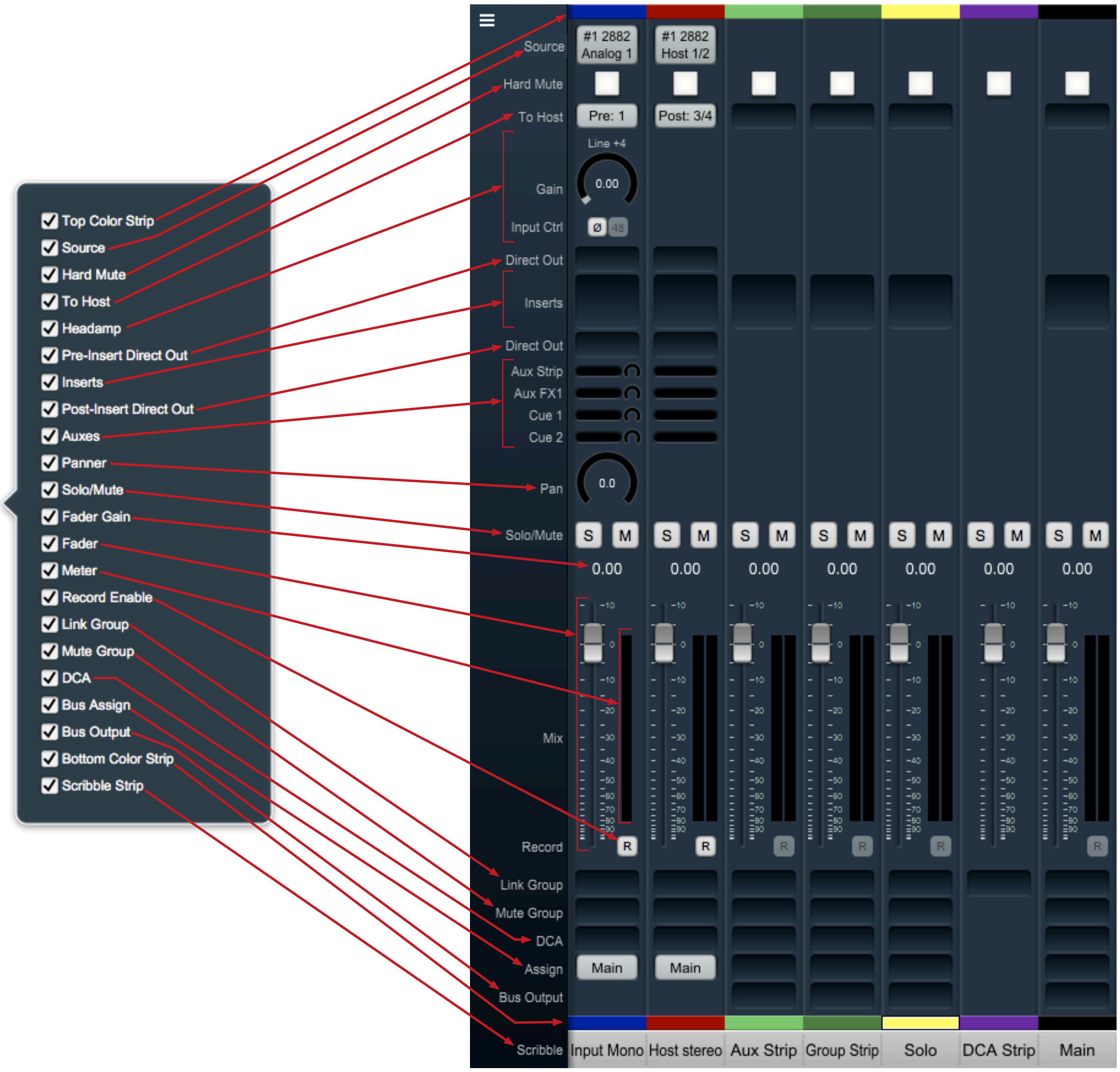


Figure 4.10: Mixer Strip elements exposed

At the top of every input strip is the Input Source selector. Click on it to see the Input Source Selector window. Within the Input Source Selector window you can choose the type of inputs you want to select from, show all the inputs from all boxes in the domain or only one box, how many channels you want your new input strip(s) to include, and how many strips you want to create (or modify) with Cascade.

"Hard Mute" is a kill switch at the source of the desk strip designed for live sound applications. "Hard Mute" is hidden and disabled by default.

Analog inputs get a head amp where you can select the input path type, invert signal phase, toggle phantom power and set input gain.

Below the input selector is the "To Host" menu. This lets you set pre- and/or post-insert return paths directly to your Host computer. Since the 3d MIOConsole3d mixer is designed to support hundreds of hardware and Host computer i/o channels, "To Host" is optimized for fast setup of large mixer layouts, and will automatically assign the *next available Host return channel*.

Below the head amp is the pre-insert Direct Out. The Direct Outs can send signal to any physical analog or digital output, as well as provide additional returns back to the Host computer.[1]

Below the Pre-insert Direct Out are the Inserts slots. Here you can insert plugins, route to sends (additional buses), add hardware I/O inserts, call up macros (previously-saved DSP processors) and instantiate custom DSP graphs. The Inserts section expands as you add more processors up to a maximum of ten per strip. To delete an insert, open the Insert selection window and select "None". *Note:* For users upgrading from earlier MIO versions, "Character" has been moved to the Inserts section.

Below the Inserts section is a Post-insert Direct Out. Any signal routed from this Direct Out includes all of your insert processing.

Aux send controls appear below the post-Direct Out. Each Aux has a send level control and (where appropriate) a panner to the bus on each strip.

Aux buses can be named in the Configure Mixer window or by double-clicking the scribble field at the bottom of the Aux return strip.

For more precise control of the send levels to an Aux bus, click the name of the Aux bus in the strip legend to the left of the mixer pane. A dot will mark the selected bus name, the gray fader knobs in the main mixer surface will become yellow, and the Aux mix will move onto the main surface.

So, yellow faders in your mixer means you are looking at and controlling an Aux mix, not your main mix. Click the dotted Aux name in the legend again to return to the main mix desk. *Hint:* Aux buses are also used for Cue mix sends.

Panners are next. The panner type configures automatically to match the channel width of the strip to the bus that it is assigned to:

- There are no pan controls on a channel assigned to a mono bus.
- Mono input channels will have a pan knob when assigned to stereo or LCR buses.
- Mono input channels assigned to LCRS through 7.1 buses will have joysticks. Right-clicking on the joystick will allow you to hard assign the input channel to a specific output channel, i.e. Center, Left, etc.
- Multichannel inputs (stereo and above) have no pan control, and are direct routed to the matching channels in the bus.

The fader/meters section dynamically expands and contracts based on the size of the Mixer window and the number of control elements currently visible in the strip. The number of meters will always reflect the number of channels handled in the strip. Meters display "pre-fader" by default, and can be set to read "post-fader" signal in the MIOConsole3d Mixer menu.

All input and bus strips have Solo, Mute and Record Enable buttons.

Please note that 'Record Enable' is active only if there is audio routed from that strip back to the Host computer either via "To Host" or a Direct Out (the audio you want to record must be routed back to the Host or the Host computer has no way to record it to the hard drive).

Below the fader/meters section are controls for managing assignments for Link Groups, Mute Groups and DCA groups, with Bus Assign selectors (all currently defaulted to the Main bus) as the bottom controls on the input strip. The Assign control is for routing that strip the Main bus and/or any available Group bus.

[1] *Hint regarding Direct Outs:* Use Direct Outs whenever you need to route to a specific-numbered return channel to the Host. "To Host" is designed to automate the addition of many channestrips in a single step and is sometimes just not the right tool for the job. Using Direct Outs (often coupled with 'Selected Strips' groups and Cascade) to route your Host returns allows you to precisely target specific channels or groups of channels from the MIOConsole3d mixer to your DAW.

At the very top and bottom of each strip is a color band (default color: gray) for differentiating strips at a glance. Click on a color band to open the color selector. Setting the color of a strip will apply to multiple strips at once based on either a 'Selected Strips' group or the assigned link group. (Note: Strip colors can be applied to the entire strip as a preference.)

See the Input Strip Details section of the MIOConsole3d overview for more information.

Bus Strips

In the default mixer configuration you have the one stereo Main "master" bus and four Aux bus return strips. These bus strips all have To Host (post-fader only), Inserts, Solo, Mute, Record Enable, Link Group, Mute Group, DCA, and Assign control elements which all work just like the Input strips.

Aux, Group and Main buses have an additional routing selector below "Assign" on the strip called "Bus Output". Use Bus Output to route bus strip outputs to any hardware port or as additional post-fader return routes to the Host computer.

There are two other types of buses which are not shown in the default mixer, but are shown in the "Mixer Strip elements exposed" graphic: Group buses and the Solo bus. The Solo bus appears when a PFL or AFL Solo mode is chosen in Configure Mixer (default is Solo-In-Place). Group buses are also created in Configure Mixer. We will get into more detail on MIOConsole3d busing architecture later but the general flow is:

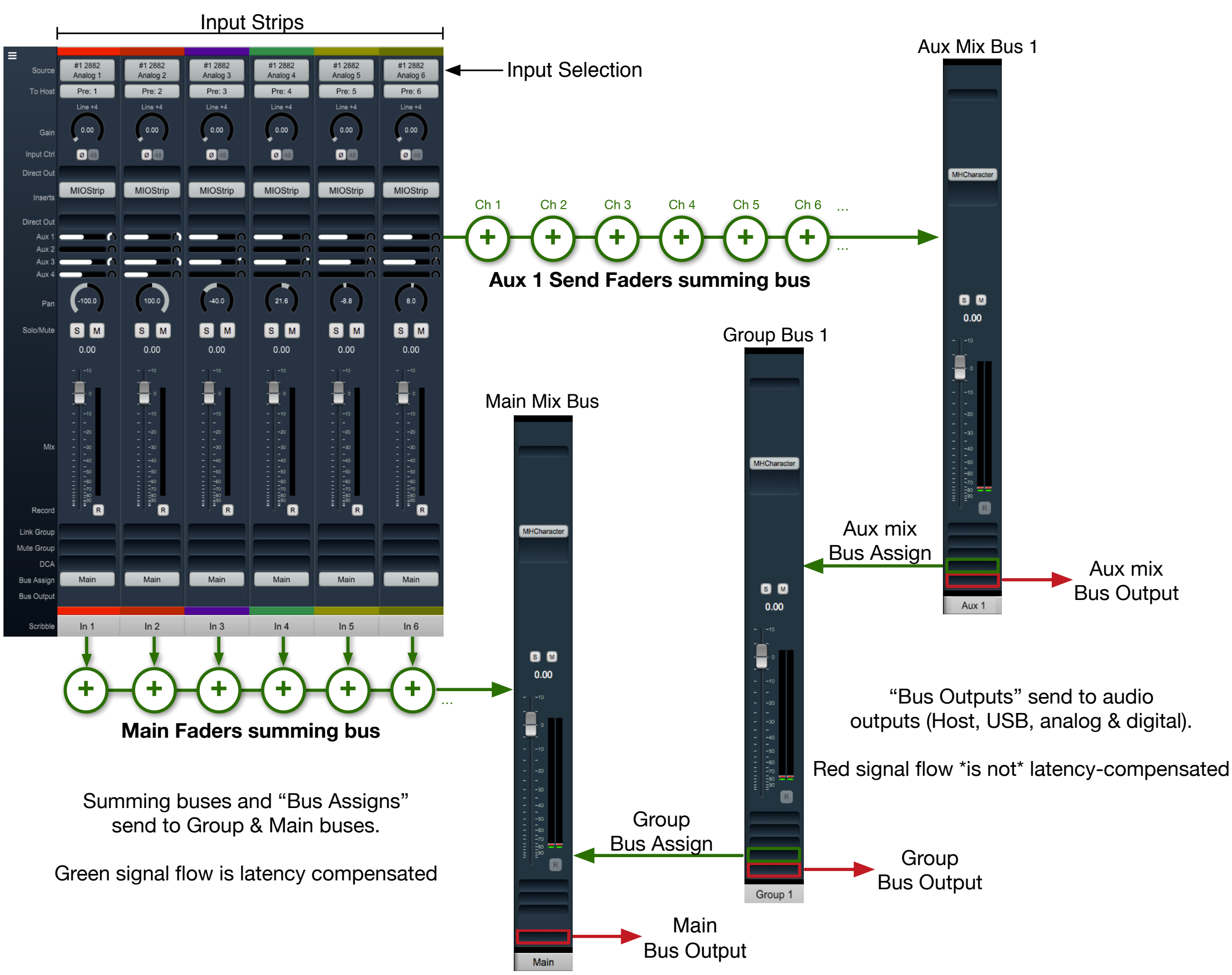


Figure 4.11: Basic 3d Mixer signal flow

- External audio sources enter the 3d mixer through Input strips. Input strips can send to any output, bus or both. Input strips are internally latency-compensated from their input through their assigned output bus.
- Aux buses include their own fader, pan and mute for each Input strip independently of the Main mix bus. Aux buses are for building a completely independent mix and can be used for effect sends (to be returned to a Group or Main mix bus), or for independent cue mixes (to be routed out to the world). Aux buses are internally latency-compensated when assigned to Group buses or the Main mix bus.
- Group buses are sub-mix buses. All the buses assigned to an input strip (Main + any groups) all share the same fader, mute, and pan settings. Group buses can be routed to hardware ports out to the world, to other Group buses, or the Main mix. Routing through Group buses within the mixer will be internally latency compensated.

The fact that the buses are latency compensated means that you can sum channels to any of the buses, and if you assign one bus to another bus the shared signals will be time-aligned and will not introduce any phasing. You can utilize this to do parallel compression on submix buses or sum in a reverb bus that includes a dry component.

Routing

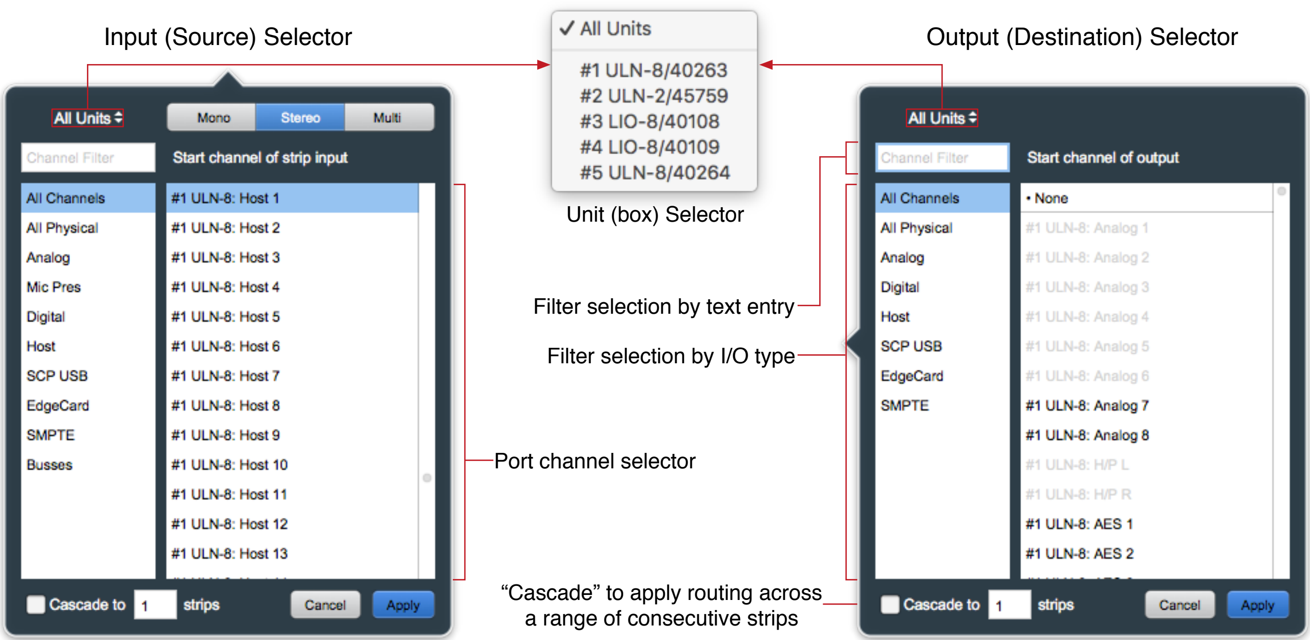


Figure 4.12: Input source and Output destination routing selectors

Since 3d/MHLink domains are completely dynamic (in that you can add or remove boxes live, and up to 100m between...) the Input and Output routing selector UIs both offer filter-by-box, filter-by-port-type and text-entry search tools to help manage the madness. Use the "Unit Selector" when you know which box you need when they are in different locations, select by the port type when the boxes are clustered in a rack, or combine search filters at will. Text-entry searches are dynamic - type "tos" and you are presented with every TOSLINK port visible in the domain.

In the above graphic, note that some of the output ports in the Output selector on the right are greyed-out and unavailable. In the case shown, those ports are occupied as a 5.1 Monitor Controller output and a stereo Cue send, both to the ULN-8 root box #1. Automatic direct mults to outputs are not currently supported in MIOConsole3d, but you can always use a Group or Aux bus to accomplish the same thing.

Two things to be aware of when routing in the 3d mixer:

- Routing from a bus output to an Input strip is allowed, but as mentioned in the previous section, will not be automagically latency-compensated.
- Since routing search filters can be combined, typing “tos” to find a TOSLINK port while the “Analog” port type filter is selected will return zero results (because, after all, TOSLINK is a digital port). Best practice is to select “All Channels” or “All Physical” (which includes all regular analog and digital hardware ports, but not Host computer and SCP USB ports) when searching by text.

> For more information regarding configuration and routing of ADAT/TOSLINK, MADI and USB ports see the Digital I/O Status / Control Menu and USB Port Status and Configuration sections of the main manual.

Using the Mixer

You should now be in a good position to integrate the 2882 with your audio workstation. First, make sure "Main" is selected as the Monitor Controller source. To play it safe (if you haven't already done so), reduce the Monitor Controller level from it's default "0.0" and bring it up after the music starts. Select “MHLink Audio” (or, if you are connecting via USB, "2882") as your audio interface in your DAW's hardware configuration. To play from your DAW, send your DAW’s signal to outputs 1/2 and it will also come into the 2882 on the Host 1/2 strip. Hit “Play” in your DAW and enjoy.

Now that you have some tunes going, you might feel like messing around with some insert processes and see what this thing can really do. To get started, go to your Host input strip and click in an Insert slot. The dialog that appears list categories for "Plugin", "Macro", "Graph" and "I/O".

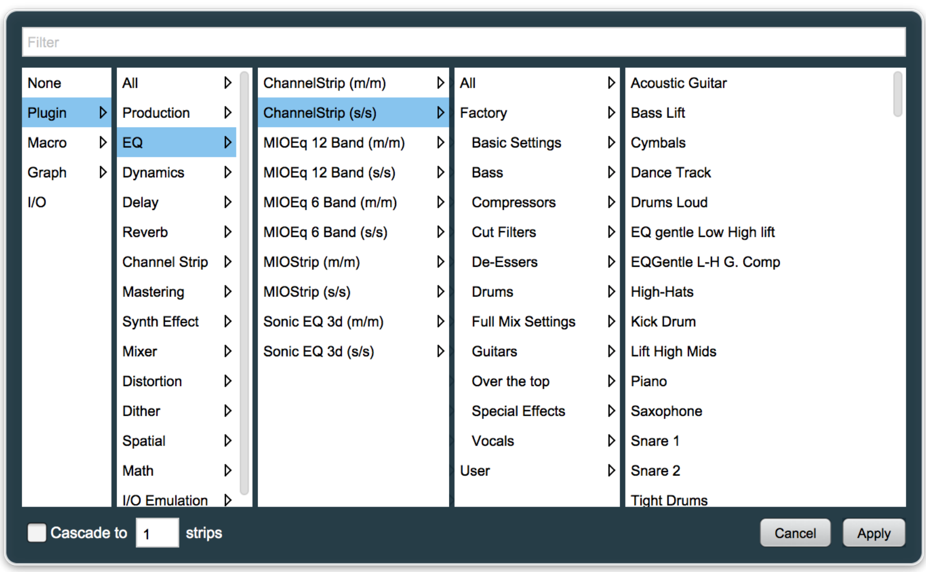


Figure 4.13: Mixer Strip Insert: Plug-in selector

- Plugin: Includes the 3d-native MH Production Bundle, Sonic EQ 3d and many very useful processor “building blocks”. Click the *MH Production Bundle* link above to learn about sharing presets between your Metric Halo AU/VST plug-ins and the 3d DSP versions.
- Macro: Most are essentially presets of more complex process chains in a DSP Graph. Feel free to use these as a starting point to create your own personal custom signal processors. The Reverb macros can not be represented as modules in a normal graph, and can not be modified as-is.

- Graph: A graph is a “DSP playground” where you can build your own signal processing chains and save them for later recall. The MIOConsole3d Graph is arguably the most powerful audio hardware DSP environment available to the public. Creative users have built custom stereo to 5.1 spatial up-mix processors, analog tape noise reduction, and multi-layered FOH speaker array time delay and acoustic correction tools, to name just a very few. Graphs can be very simple - just build a quick little tool to do some one-off task not gracefully handled by a packaged plug-in… Graphs are especially spectacular for creating new types of parallel processors because no matter how complex the processing it is all fully internally latency-compensated.
- I/O: A basic signal path insert-within-an-insert, if you will. From within any strip, send to any hardware output or the Host computer and return from any hardware input or the Host. I/O is very useful for an extra Direct Out, as a sidechain source, or for mults to secondary systems (external backup recorder, video truck feed...), etc.

  Note: Since an I/O send/return insert sends audio to external systems outside the mixer DSP and back, I/O loops can not be automatically latency-compensated.

As a reminder, all of the mixing, gain control and processing which is shown and accessed through the MIOConsole3d is operating solely in the box(es) and not on the computer, even though the user interface for the 3d Mixer is controlled from the computer display. This is an important distinction to keep in mind so you can properly manage your sessions and take best advantage of the strengths of both the DAW and the 3d hardware.

Now let's walk through adding the 2882's eight ADAT channels into the MIOConsole3d mixer and your DAW. This procedure works with any kind of input from any box in the domain.

First hit command-shift-A to create a new input strip. In the routing window that pops up, select "Digital" from the filter list on the left (to show only the digital inputs), select "ADAT 1", type "8" in the Cascade box at the bottom of the routing window and hit Apply (or Return or Enter).

With these few clicks, you have instructed the Mixer to create eight new input strips assigned starting at ADAT Input 1 (i.e.: ADAT inputs 1-8).

The Mixer defaults to automatically assign routes to the Host computer for each strip, starting with the next available To Host channel... so, since the eight analog input strips are already using To Host routes 1-8, these new strips will use To Host channels 9-16.

Now, go back to your DAW, create eight new input strips in your DAW mixer and assign them to input 9-16. You have built a virtual patchbay routing audio between your DAW and the 2882. You can route pre- and post-insert Direct Outs, Aux buses, Group buses and the Main bus back to the Host DAW in the same way.

To disable any route, just open the routing window and select "None" at the very top of the strip inputs list, or mouse over the strip inputs button and click the "X" to delete that route and free it up for another task.

Mixer Strip Color Bars

Clicking on the Color Bar at the top or bottom brings up a macOS color selector, from which you may choose a color for that Mixer strip. In the Appearance pane of the MIOConsole3d Preferences you may select whether to color the entire mixer strip or just the top and bottom color bars.

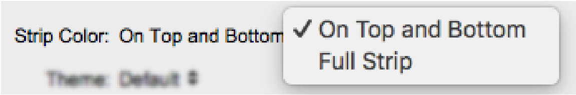


Figure 4.14: Console3d Prefs: Appearance: Strip Color configurator

The Bottom and Top Color Bars mirror each other, so what applies to one applies to the other. You may show or hide the opt or bottom color bars independently with “Configure Mixer Strip Controls...”.

Scribble Strip

Double-click the text field at the bottom of any strip to enter the strips name. The name you enter in the Scribble Strip will be propogated throughout the 3d user interface wherever a Mixer Strip name is found, including plug-in headers, Link Group, Mute Group, DCA and Bus Assign selection windows.

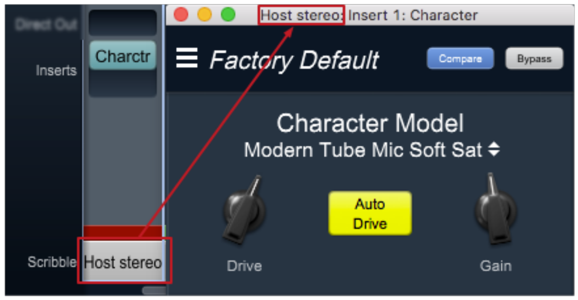


Figure 4.15: Scribble Strip name to Mixer strip Insert header

Scribble Strip names are also used to name sound files recorded in the Record Panel.

Mixer strip Insert controls

Once you have selected a plug-in, it will be listed in the assigned insert slot:

Figure 4.16: Inserted MIOEQ6 Plug-in (as shown in Mixer strip)

Plug-in names will generally appear abbreviated in order to save space ("MIOEQ6" is shown above).

When you move the mouse over an inserted plug-in, the Insert label will change to show three control icons. The tooltips for each of these controls have been exposed in the example graphic below.

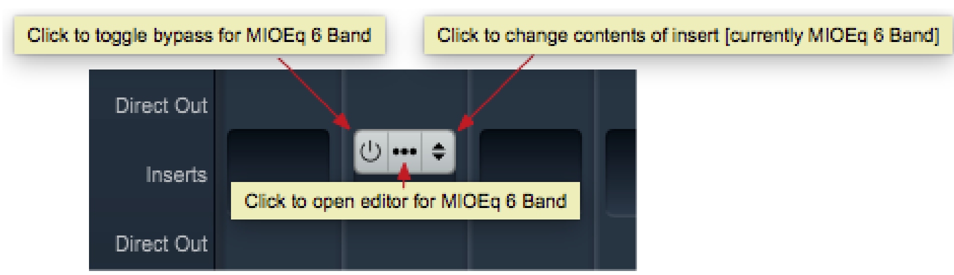


Figure 4.17: Inserted plug-in controls

- The "On / Off" switch icon on the left is the plug-in Bypass. When Bypassed, the Insert button will turn yellow.
- Clicking the "..." icon in the middle opens the inserted plug-in editor UI. When the plug-in editor is open/visible, the Insert button will turn blue.
- Clicking the "up/down" arrows icon at the right opens the Insert selector window, where you may select a replacement plug-in, or navigate to directly open a different saved preset without having to open the plug-in editor UI.

Insert Control modifier key shortcuts

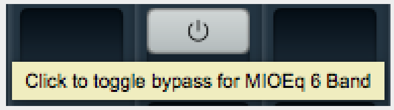


Figure 4.18: "⌘-click" / <Command>-click to Bypass Insert

<Command>-click the Insert button to Bypass the Insert.

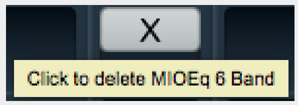


Figure 4.19: <Control-Option-Command>-click to Delete Insert

Use "^⌥⌘-click" / <Control-Option-Command>-click to delete an Insert directly from the Mixer desk.

Quick Copy/Paste Plug-ins

Option-click-drag any plug-in instance from one Insert slot to another anywhere in the Mixer to clone that Plug-in to the new location. Plug-in instances will automatically adapt to the channel width of the target Insert as necessary

"Sweeping" controls

Toggle buttons on consecutive strips in the 3d Mixer desk can be switched in a single move by a click-hold-sweep gesture.

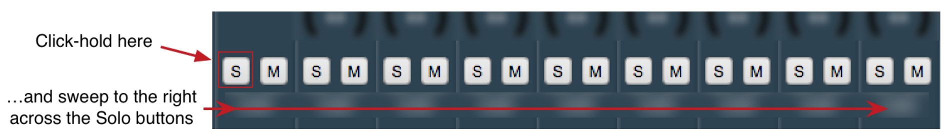


Figure 4.20: "Sweep" to toggle multiple buttons in one gesture

To try it, click on a Solo button, and while holding the mouse button down, drag the cursor to the right or left across the adjacent Solo buttons.

The move works with Polarity Invert, Solo, Mute and Record Enable buttons.

The 2882 and surround

If you are working in surround, it’s easy to configure the 2882 for surround processing and monitoring. Let's take a minute to walk through the process:

First, go to the Mixer menu and select “Configure Mixer” (cmd-shift-C). This window is the main setup page for your Mixer layout. The Main Mix configuration is set at the top - it is currently set to “Stereo”. If you click on the Main Mix selector, you can change it from “Stereo” to whatever width you need.

Let’s say you’re working in 5.1. Select “5.1 SMPTE/ITU” and hit OK… Voila! Your Main bus is six channels wide and all the mono input channels now have joysticks instead of pan knobs.

Your Host input strip is still bringing your DAW in on just two channels, so let's fix that too. Go to the top of the “Host” input strip and click on the input assignment pulldown. At the top of this window you have the choice of a mono, stereo or a multi-channel strip. Select “Multi” and click “Apply”.

Your Host strip is automatically set to match the Main bus configuration of your mixer: in this case 5.1 SMPTE/ITU. Now your audio will come into the 2882 in SMPTE surround format as positioned by your host on channels 1 through 6.

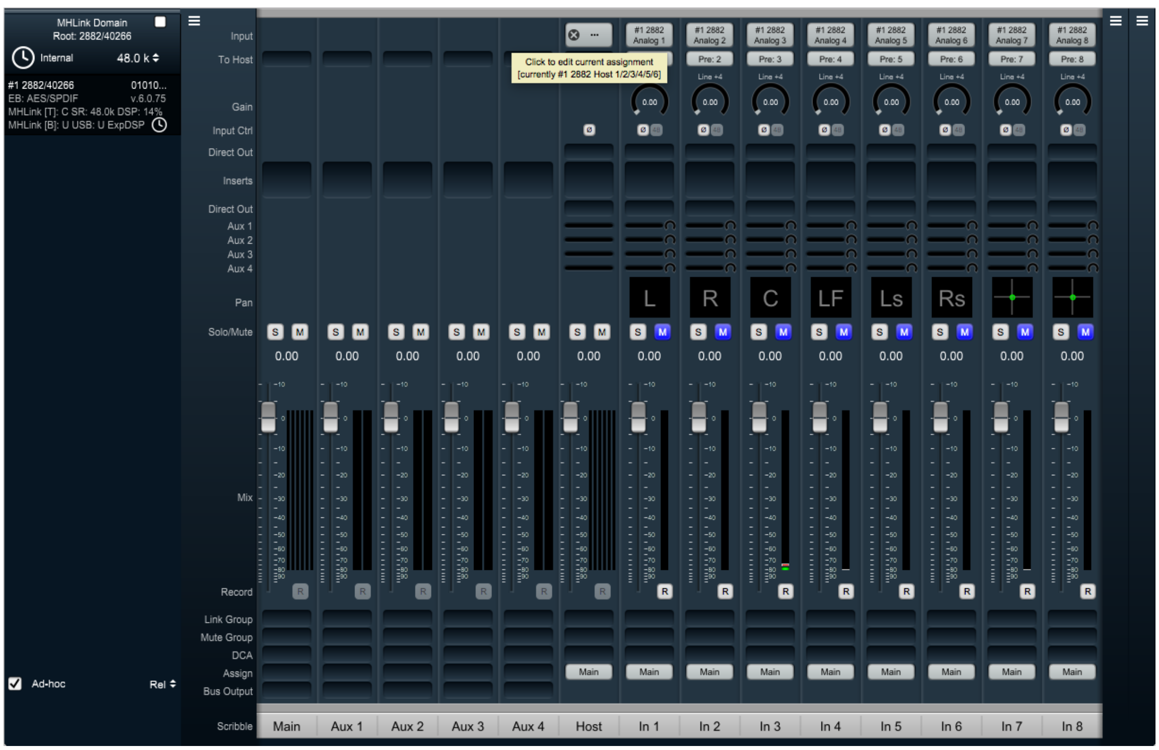


Figure 4.21: Surround Mixer

Your surround mixer should look pretty similar to this. You can right-click (or ctl-click) the panner on the first six mono input strips to hard-assign each strip to a particular speaker channel as is shown on the example above.

Note especially the mouse-over pop-up at the top of the Host desk strip. Ordinarily the text “#1 2882 Host 1/2/3/4/5/6” would be too small to read, so in the interest of saving screen real estate these mouse-overs come in very handy.

If you can afford the space, use the “Mixer - Set Mixer Strips Width” menu to adjust your mixer desk strips’ width to taste.

Surround Monitoring Setup

In order to monitor in surround, you need to create a 5.1 output in the Monitor Controller. Go to the "Monitor" menu and select "Add Monitor Output" to get the setup box. You'll see that the "Name" field is already selected, so go ahead and type "5.1" to name the monitor output.

The "Type" selector to the right of this window is where you choose your new Monitor Controller output configuration: Select "5.1 SMPTE/ITU".

To set up 5.1 Monitor Controller outputs, click the Left channel selector (currently set to "None") in the "Destinations" field. The routing box should appear as in the figure below.

If your speaker channels are all in ITU order, you can just double-click "2882 Analog 1" to automatically cascade L, R, C, LFE, Ls, Rs to channels 1, 2, 3, 4, 5 and 6 respectively. Otherwise de-select "Cascade" at the bottom of this window and assign each speaker output individually.

Click "Apply" to route the speaker assignments. Note that you can apply individual level trim and delay for each speaker from this menu, should you desire.

Click "Add" to confirm and establish the new Monitor Controller Output parameters and finish up. You can return to the Monitor Controller Output configuration screen any time by right-clicking the MC output selector, or going to the Monitor menu and selecting "Edit Current Monitor Output".

Figure 4.22: Monitor Controller speaker assignment

Unleash the DSP

The 2882 is based on the 3d processing engine and includes a very powerful arsenal of high-precision audio processing tools. All plug-ins and processors work in any channel configuration and include full parameter save and recall with lots of factory presets to get you started.

There are two ways to access the DSP:

- You can insert processes directly in the mixer strip inserts; this works well for standalone processes like eq, compressors, etc.
- Insert a graph. This lets you chain plugins together, use them in parallel, and create custom processors with routing configurations that would be very difficult (or impossible) with other platforms.

  Graphs are also available in the Monitor Controller and Cue Controller signal path. High-precision room EQ/acoustic correction, custom crossovers and bass management with none of the expense, inconvenience or limitations imposed by running such processes in outboard gear or a DAW monitor bus.

You should definitely check out the following processes:

- MIOStrip: Gating, EQ and compression powerhouse. Very clean, very transparent - put it up against any precision digital mastering hardware and be just a little stunned. Mix it with some Character for a tracking console vibe.
- Character and MHCharacter: The sound of different analog circuits and devices available on any input, output or bus. Both have manual and automatic drive and gain. Character is something of a late 80's/90's animal with an extra lo-fi bite. MHCharacter is geared more towards higher precision, warmth and detail enhancement.
- Haloverb: Great sounding reverb, doesn't use any CPU from your host. What's not to love?
- Transient Control: Super fast, super clean initial strike control.

You should read the DSP Implementation section to learn more about how to work with DSP, and the DSP documentation details the over 100 plug-ins available in the 3d DSP engine.

This should get you started with the 2882!

Part II. Interfaces

Table of Contents

5. ULN-8 Users Guide

ULN-8 Overview

Figure 5.1: ULN-8

What it is

The ULN-8 is a portable, archival–quality, modular multi-format audio converter, interface, and processor for professional audio applications. The ULN-8 is equipped with eight balanced mic inputs, eight balanced line inputs, eight channels of AES digital I/O, eight balanced sends and eight balanced analog outputs, as well as wordclock in/out, MIDI in/out, SMPTE in/out.

Host computer connection and box daisy-chaining are provided by two MHLink Gigabit Ethernet ports, each of which support 128 channels of bidirectional audio at 192khz/32 bit data rate. A UAC2 class-compliant USB-C connection is available as an alternative computer interface.

All inputs and outputs are capable of 24-bit/192kHz operation.

What it has

- 18 simultaneous input channels and 20 simultaneous output channels
- Full 24 bit/192kHz audio
- 44.1, 48, 88.2, 96, 176.4, 192kHz Sampling Rates
- 8 channels of 24 bit A/D converters
- 10 channels of 24 bit D/A converters
- 8 Balanced Microphone Inputs with switchable phantom power – DB25
- 8 Balanced Line Inputs – DB25
- 8 Balanced line level Sends – DB25
- 8 Balanced Analog Outputs – DB25, channels 1 and 2 multed on 1/4" TRS
- 8 channels of AES I/O, single wire for 44.1-192KHz operation - DB25
- 2 channels of DI available on the front panel - 1/4" TRS
- Front panel cans output with discrete high quality D/A and amplification - 1/4" TRS
- Word Clock 1x on 75ohm terminated BNC
- SMPTE I/O on 1/4" TRS
- MIDI I/O on 5-pin DIN

- MH EdgeBus programmable and pluggable audio expansion slot.
- Enhanced processing power and massive hardware memory. All 3d hardware includes over a hundred DSP plug-ins and the unique Metric Halo Graph environment within the 3d mixer.
- 128 input x 64 bus (at all sample rates) multi-box unified zero-latency mixer
- 1024 x 1024 internal audio routing matrix (per box)
- Front Panel Metering for Analog Inputs and Outputs on 15 segment multicolor LEDs
- Front Panel signal present, lock and clock selection multicolor LEDs for AES I/O
- 9 detented Front Panel encoders
- Full console metering of every channel and mix bus
- Total recall of every console parameter
- Portable Capabilities – Battery Powerable
- Rack Mount Kit

Options (can be installed before or after purchase)

- Metric Halo Edge Card expansion modules
  Click here for more information on the available Edge Card configurations

What you need to use it

- Computer:
  - Any Mac that supports Mac OS X 10.8.5 with a Gigabit Ethernet or UAC2 class-compliant USB connection
  - Cat 5e or better cable (e.g. with all 8 conductors)
  - Mac OS X 10.8.5 or newer required
  - Mac OS X 10.11.5 or newer recommended
- Peripheral Gigabit Ethernet Adaptors supported:
  - Apple Thunderbolt Gigabit Ethernet Adapter
  - PCIe Gigabit Ethernet Adapter
  - Third-party Thunderbolt Gigabit Ethernet Adapter (e.g. as part of a dock)
  - USB3 Gigabit Ethernet Adapter
- Software: All CoreAudio compliant software is compatible with Mobile I/O, including:
  - Cubase
  - Digital Performer
  - GarageBand
  - iTunes
  - Live
  - Logic
  - Nuendo

- Pro Tools
- Studio One
- Reaper
- MixBus / Ardour
- and hundreds more…

What comes with it

Your ULN-8 package contains the following items:

- One ULN-8 unit:

Figure 5.2: ULN-8 Unit

- One IEC Power Cord appropriate for your area
- One 18-volt 48-watt world-ready external power supply
- One 12” CAT-5e Gigabit Ethernet Cable
- One 14' CAT-5e Gigabit Ethernet Cable
- Two Rack Ears w/ fasteners
- Jumpers for internal configuration
- Warranty/Registration Card

If any of these items are missing from your package when you open it, please contact Metric Halo or your dealer immediately for assistance.

Using the ULN-8 Hardware

ULN-8 Front Panel

Figure 5.3: ULN-8 Front Panel

The front panel provides ULN-8 system control and status at a glance:

- Monitor - Select the Monitor Controller input or output
- Preset - Recall one of the eight stored system configurations
- Input - Step through the available input sources
- Link - Link multiple input and output encoders for stereo or multichannel use
- +48 - Enable phantom power, per channel
- U/M - User mode (for future expansion)

- Power switch for rear 4 pin XLR connector (jumper defeatable)
- Sample Rate (nominal 44.1, 48, 88.2, 96, 176.4 or 192KHz)
  - The Sample Rate is determined from the currently selected clock source so it will accurately indicate the current sample rate, even when the clock source is being provided by an external device.
- Clock source:
  - Internal indicates that the system is internally clocked
  - Wordclock indicates that system is being clocked from the wordclock input
  - Digital In indicates that the system is being clocked from the selected digital input
- Control Mode — Indicates what parameters the encoders are modifying:
- I/O Trim — Indicates whether you are modifying input or output channel gain.
- Input Status - Indicates mic input, line input or mic s/r by channel.
- Encoders - Eight detented encoders for multipurpose control, with push-switches.
- 15-segment metering for the 8 analog inputs and outputs using multicolor LEDs, which also display gain values during encoder use. The meters are fast PPM peak reading meters with auto-resetting peak holds.
- SysLock — Indicates that the system clock recovery circuit is properly locked to the selected clock source. If this light is not illuminated, the ULN-8 will not be locked to a clock and will revert to its failsafe internal clock source. Even if the Locked light is not illuminated, the actual sample rate will still be indicated on the front panel display.

- FireWire — Indicates that the ULN-8 has been successfully connected to either a USB or MHLink Gigabit Ethernet connection to the Host computer.
- Signal present indicators for AES input and output, as well as digital lock and cable select button and indicators.
- Monitor Control Section:
  - The Mute and Dim buttons provide instant access to simple level control for the selected Monitor path or headphone output. The Mute button provides a quick, tactile “panic switch” which mutes the monitors or front panel headphone output in case of accidental feedback loops and other audio unpleasantries. The Dim button attenuates the selected path by 20 dB.
  - The Monitor Control encoder provides front panel adjustment for your audio. By pressing the encoder, it can toggle between affecting the Monitor Control section or Headphones. There are two multicolor LEDs below the encoder that indicate which path the encoder is modifying as well as mute and dim status.

The ULN-8 front panel also provides access to the Headphone output. The headphone output jack is a TRS 1/4” jack that provides the Left Channel on the tip, the Right Channel on the ring and the ground return for the two channels on the sleeve. These signals are all ground referred, so they may also be split and fed single-ended (unbalanced) to an external audio device.

The ULN-8 headphone output is suited to a wide range of impedances. Headphones with lower impedance would be expected to get louder than those with higher impedance.

There are two DI inputs on the front panel, which are paralleled with Line inputs 1 and 2. The DI inputs can provide 0, 10, or 20db of gain selectable via internal jumpers.

More information regarding the front panel can be found in the ULN/LIO-8 Front Panel Guide.

ULN-8 Rear Panel

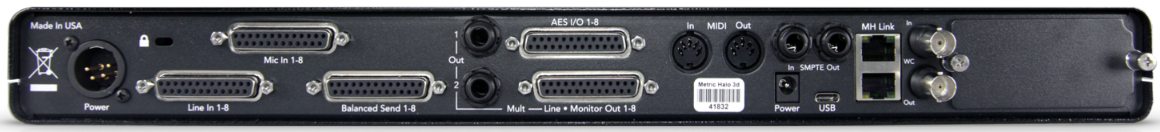


Figure 5.4: ULN-8 Rear Panel

The ULN-8 rear panel features:

- 4 pin XLR DC power jack (14v - 28v, 32 Watts)
- 8 channels balanced mic inputs on DB25. Each input has:
  - -22 dB to +91.5 dB of gain range
  - Dynamic Range (-60 dB, flat 0-22.05 kHz, typ): 115 dB
  - +0/-1.0dB @ fs = 192KHz: 2.9 Hz - 64.7 kHz
  - Noise Floor (flat 0-22.05 kHz, typ): 115 dB
  - remote switchable 48v Phantom power, with 10mA current limit
- 8 channels balanced line inputs on DB25. Each input has:
  - -12 dB to +31.5 dB of gain range
  - Dynamic Range (-60 dB, flat 0-22.05 kHz, typ): 115 dB
  - +0/-1.0dB @ fs = 192KHz: 1.8 Hz - 64.7 kHz
  - Noise Floor (flat 0-22.05 kHz, typ): 115 dB
- 8 channels balanced line level sends on DB25 that mirror the selected input
- 8 channels balanced line/monitor outputs on DB25. Each output has:
  - 24-bit 192kHz D/A converters (120dB SNR)
  - Gain range from -96 dB to +30 dB
  - Outputs 1 and 2 also multed to 1/4" TRS connectors
- 8 channels AES digital I/O on DB25
  - Single wire mode for full 8 channels at 44.1-192KHz operation
- Wordclock input/output on BNC connectors
- MIDI I/O to connect a control surface to MIOConsole3d
- SMPTE input and output on 1/4" TRS
- 2 MHLink Gigabit Ethernet ports
- 1 USB-C port
- Kensington security slot
- 1 4-pin XLR power port, switched, compatible with any 4-pin XLR power system with the following characteristics: 9v - 30v DC, Pin 4 Hot, 15 Watts

ULN-8 3d Signal Flow

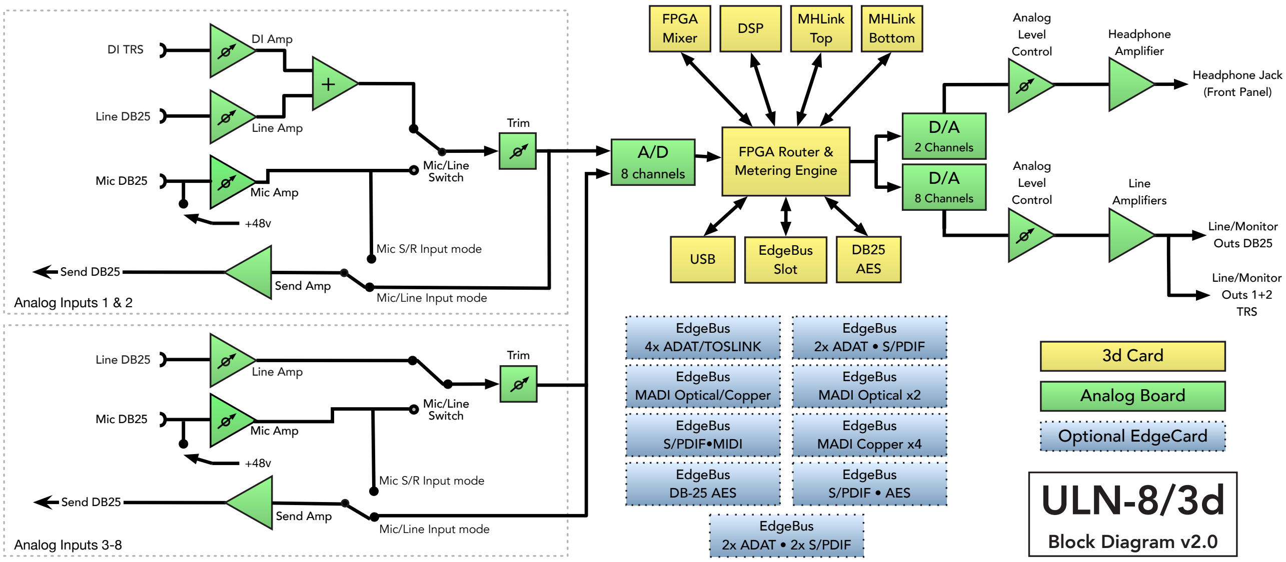


Figure 5.5: ULN-8 3d Signal Flow

Click here for a larger version

Making connections to the ULN-8

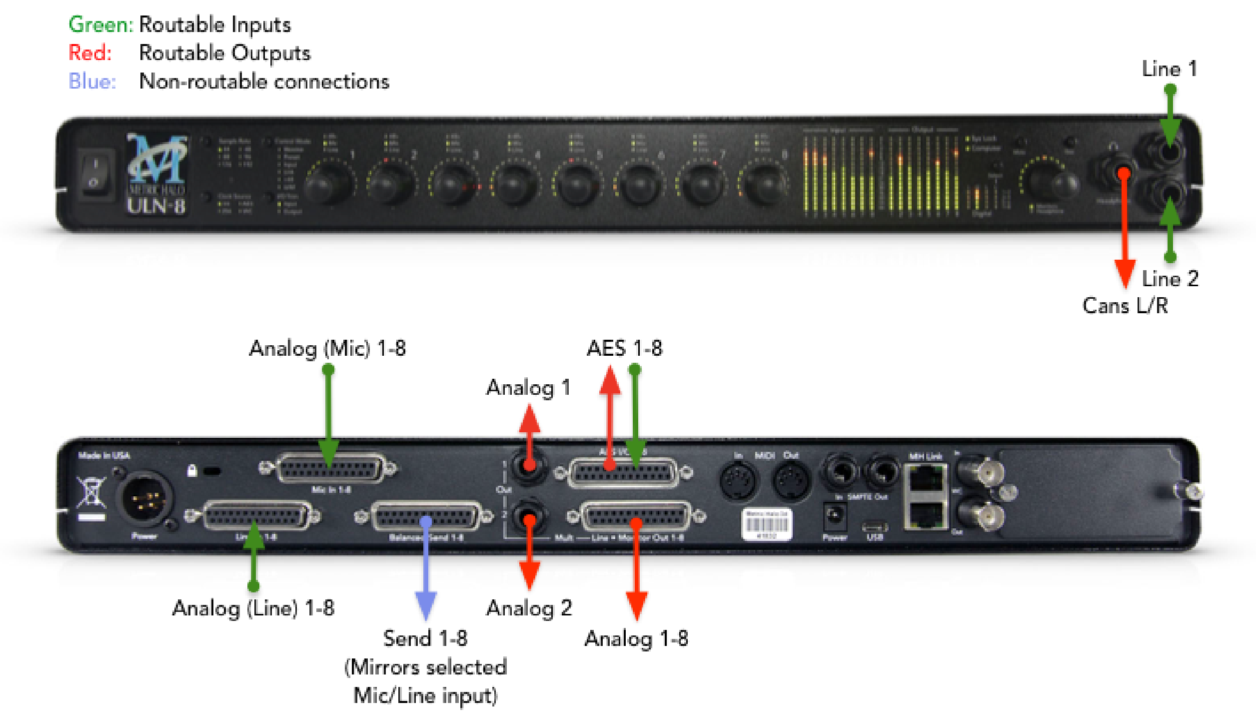


Figure 5.6: ULN-8 3d Routing

Click here for more information on the available Edge Card configurations

There are eight classes of connections you can make to the ULN-8 3d hardware:

1. Gigabit Ethernet (MHLink)
2. USB
3. Analog Audio
4. AES Digital Audio
5. Clock Sync
6. MIDI
7. SMPTE
8. Power

Computer Connections

- **MHLink (Gigabit Ethernet)** - The preferred method of connecting your computer to the ULN-8 (or any Mobile I/O 3d audio device) is with a Gigabit Ethernet cable to one of the MHLink ports. This method enables the full capabilities of the 3d/MHLink environment with 128 channels at 32bit integer resolution input and output at all sample rates up to 192kHz. Daisy-chaining additional MHLink boxes dynamically and transparently integrates the new devices into the currently-running 3dConsole mixer, making all the added I/O ports and DSP engine power immediately available for use.

  MHLink requires the installation of the MHLinkDriver software. The latest MIOConsole3d installer package is available from: macOS MIOConsole3d installers (this will require a restart). Please see the section on software installation if you need clarification.

  > **Driver Installation on macOS 10.13 and newer**
  > With High Sierra (macOS 10.13) Apple introduced another layer of driver (system extension) software security.
  >
  > In order for third-party drivers (system extensions) to be activated on new installs of High Sierra, the user of the computer (you) must explicitly approve the activation, after the installation has occurred.
  >
  > You must do this before you will be able to use your hardware via MHLink.
  >
  > The authorization process is the same for all third-party drivers. If you are unfamiliar with the process, please read our FAQ answer here: macOS Driver User Authentication.

  Note: With MHLink, you need to launch MIOConsole3d to establish the routing between MHLink and the attached box(es). The routing is established automatically as soon as you launch MIOConsole3d.

- **USB** - The secondary method of connecting to your ULN-8 is with USB. USB connections for audio transport will be limited compared to MHLink. In a pinch, however, your ULN-8 can be seen as an audio device by any computer with a UAC2 class-compliant USB port.

  Note: Connecting to USB does not require the installation of a software driver. See MIOConsole3d Preferences > Discovery before connecting USB to a computer which is also connected to an MHLink ethernet interface.

Analog Audio Connections

The analog I/O connections on the ULN-8 have been engineered for maximum flexibility in that they support both balanced and unbalanced connections with a wide range of input and output levels and a wide range of matching impedances. The microphone inputs have an impedance of 3.3k Ω, and the line inputs are 10k Ω. With that in mind, there are a number of aspects of the design that you should take into account when interfacing with the ULN-8. All multichannel audio connections are made using industry standard DB25 connectors, which are pin- compatible with cables using the Tascam/Digidesign standard.

Wiring details for this connector can be found in the DB25 Pinouts appendix.

Whenever possible, use balanced connections with the ULN-8. The performance of balanced interconnects is much higher and much more resistant to noise interference and electrical (power) wiring problems. The expense of balanced interconnects is not substantially higher than unbalanced connections, so if the gear that you are interfacing with supports balanced connection — use it. If you cannot utilize balanced interconnects, there are connection schemes that you can use that will maximize performance.

On input, at line level, it is sufficient to simply use standard unbalanced (TS) connections. If you are interfacing with the Mobile I/O XLR inputs, you will need to ensure that pin 3 is grounded in the unbalanced adapter cable. More information about adjusting the input level can be found in the MIOConsole3d software chapter.

ULN-8 DB25 cables should be wired pin 2 or Tip hot. The front and rear 1/4" connectors are wired Tip hot. The headphone connector is wired Tip/Left, Ring/Right, Sleeve/Ground.

Connecting to the DIs

The DI inputs provide an excellent high impedance input for directly connecting instruments to the ULN-8. These inputs are unity-gain summed with the corresponding line input from the back panel. This means that if you have signal connected to both the DI and line input for a given channel the ADC will see a mix of those two inputs. If you do not wish to have signal from one of the connectors you need to physically disconnect that input or otherwise ensure that no signal is present at the unused input.

You may notice that the MIOConsole3d has an input selection for "Inst". This can be used as a visual reminder that you are plugged into the DI, but has no operational effect on the input. Selecting "Inst" on any channel of the ULN-8 sets the input to Line +4; the only DI input is available from the front panel jacks for inputs 1 and 2.

Each DI input supports balanced connections via TRS cable. You can connect a source to the DI via a balanced interconnect (if the source is actually a balanced output), an unbalanced interconnect (if the source is unbalanced like most instruments), or a telescoping shielded cable (which is an alternate unbalanced connection that provides additional shielding).

Since the DI inputs provide an ultra-high input impedance, if you connect an unbalanced source to the DI via a balanced cable you will float the negative input and will effectively inject a large DC offset into the output of the DI which will cause the associated ADC to mute. As a result you must use either a TS cable or a telescoping shield cable with unbalanced sources.

When you remove the connector from the DI jack, the DI connects its inputs to ground internally to remove any possible residual DC offset or pickup noise. If you disconnect the cable from the source but leave it connected to the DI input, the cable will function as an antenna and will inject noise into your inputs and may cause enough of a DC offset to mute the ADC. As a result you should always disconnect the cable from the DI rather than the instrument.

Balanced Sends and Mic S/R Mode

The ULN-8 has eight balanced sends; these mirror the analog input you have selected to feed that channel's A/D converter. For example, if you select the Mic input on analog channel 1, the output of that mic preamp will be available at send 1. If you select the Line input on analog channel 1, the Line in signal will be available at send 1. This allows you to use the sends as a splitter to feed a mixing console, recorder or other equipment.

Using analog inserts on the ULN-8
The "Mic S/R" input mode creates pre-converter inserts on a per-channel basis by utilizing the Mic input as the channel input and Line input as the insert return. For example, to insert a compressor on Analog 1:

1. Set Analog 1's input type to Mic S/R
2. Connect your signal (mic or line level) to Mic input 1
3. Connect Send 1 to your compressor's input
4. Connect your compressor's output to Line input 1

You have now inserted your compressor between the output of the mic preamp and the A/D converter.

You cannot route signal from a DAW channel to the balanced sends; to route signals from your computer to the D/A converters, you must use the Line/Monitor outputs.

AES Digital Audio

The ULN-8 supports 8 channels of digital audio over copper-based connections. The native format of the ULN-8 is AES, but can be converted to SPDIF or optical using third-party adapters. The ULN-8 operates in single-wire mode, providing 8 channels of digital audio at all sample rates. The ULN-8's digital I/O connections are made using industry standard Tascam/Digidesign pinout DB25 cables.

Wiring details for this connector can be found in the DB25 Pinouts appendix.

Digital audio interface options can be greatly expanded through the addition of a Metric Halo EdgeCard™.

Clock Sync

Clock sync is a serious consideration in any digital audio system.

If you are recording analog sources with ULN-8, you can simply use the unit's high-quality internal clock source to drive the converters. This is the easiest case to deal with.

If you need to interface with other devices digitally or ensure sample accurate sync with video sources, the extensive clock synchronization capabilities of ULN-8 will prove to be more reliable (and better sounding) than much higher priced alternatives.

There are three different ways to get external clock information into the unit:

1. Sending a word clock signal into the WC Input BNC.
2. Sending an AES signal into the Digital input.

The BNC word clock input port is a 75 Ohm terminated coaxial input. It should be driven by a 75 Ohm source driver and interconnected with 75 Ohm coaxial cable. If you do not use proper cabling and source drive, you will introduce reflections on the word clock cable which will propagate jitter into the recovered word clock.

The AES recommended procedure for distributing clock is to use an AES clock signal. The AES clock signal is an AES digital audio signal with no audio activity. The LIO-8 only uses the AES preambles for clock recovery, so it is immune to data dependent jitter effects. This means you can reliably use the Digital Input as a clock source with or without audio data.

MIDI

The ULN-8 offers MIDI input and output ports for direct connection of a control surface. These ports are only active while MIOConsole3d is running; the ULN-8 cannot utilize a control surface in standalone operation. MIOConsole3d makes these ports available to other applications while it is running, but the MIDI implementation is currently optimized for control use only - it is not recommended to use these ports for connecting keyboards or other devices that require accurate timing.

SMPTE

The ULN-8 has SMPTE input and output ports dedicated for timecode use. The SMPTE input and output are routed from the "SMPTE" category in the MIOConsole3d Source and Destination routing menus. Please note that MIOConsole3d does not natively decode or encode MTC at this time; if you need to synchronise to MTC, you will need third party software.

Power

One of ULN-8's great strengths is the flexibility of its power system. The ULN-8 can be powered from any DC source in the range of 14V to 28V as long as it provides 32 Watts of power. The DC inputs on the ULN-8 are a 2.1mm coaxial power connector, center positive and a 4-pin XLR connector Pin 4 Hot. So if you are powering the unit with a third party power source and it supplies 14V, the power source will have to provide 2.3 amps of current. If you are powering the unit with 24V, the power source will have to provide 1.3 amps of current, and so on.

The ULN-8 ships with a world-ready 18 volt, 2 amp power supply. You can plug this supply into any AC power source from 90V to 240V, 50Hz - 60Hz, using an appropriate IEC power cord, and it will supply the proper power to the ULN-8 on the 4 pin XLR power connector. Unfortunately, the power capacity supplied through a USB bus is unsufficient to power a ULN-8.

As with all electronic devices, when connecting an external power source to the ULN-8, you should first connect the power source to the ULN-8 while it is in an unenergized state (e.g. not connected to the mains or switched off). After the connection to the ULN-8 has been made, you should energize the power source.

If you connect an energized power source to the ULN-8's 2.1mm power connector you may see a small spark when you make the connection. This is due to surge current and is normal if you connect a power source in this way. While this will not damage the ULN-8 in any way, to avoid the spark just connect the power connector to ULN-8 before connecting the power source to the wall.

ULN-8 Specifications

Rails	
Preamp Rails	±15.8 Volts
Analog Rails (low power)	±9.9 Volts
Analog Rails (high power)	±12.6 Volts
Phantom Power	48 ± 0.1 Volts
Max Phantom Current Per Mic	10 ma

Table 5.1. ULN-8 Voltage Rails

Maximum I/O Levels (Balanced)	
Peak Line Output @ 0 dBFS (no jumper/low power)	+18.5 dBu
Peak Line Output @ 0 dBFS (output jumper/low power)	+22.0 dBu
Peak Line Output @ 0 dBFS (output jumper/high power)	+24.5 dBu

Maximum I/O Levels (Balanced)	
Analog Send Max Output	+21.5 dBu
Mic Pre Max Input	+20 dBu
Line In Max Input	+24.5 dBu
Output Impedance	5 Ω

Table 5.2. ULN-8 Maximum I/O Levels (Balanced)

Monitor Controller	
Nominal FS output (Balanced) Output Jumper Off	-19.0 dBu
Nominal FS output (Balanced) Output Jumper On	-12.0 dBu
Maximum Output	Same as Line
Gain Range	-96 dB to +30 dB
Gain Precision	±0.05 dB
Gain Step	0.5 dB

Table 5.3. ULN-8 Monitor Controller

Mic Pre Input + ADC	
Input Impedance	3.3k Ω
Dynamic Range (-60 dB, flat 0-22.05 kHz, typ)	115 dB
Dynamic Range (-60 dB, A-weighted, typ)	118 dB
Noise Floor (flat 0-22.05 kHz, typ)	115 dB
Noise Floor (A-weighted, typ)	117.9 dB
Gain Range	-22 dB to +91.5 dB
Gain Precision	±0.05 dB
Gain Step	0.5 dB
THD D/A/A/D loop @ -12 dBFS	0.0007 %
THD D/A/A/D loop @ -12 dBFS +12 dB Gain	0.0015 %
THD D/A/A/D loop @ -0 dBFS	0.0023 %
Crosstalk @ 1kHz	-110dB
IMD 1k component (19 kHz/20kHz @ +8dBu)	-104 dBu
EIN @ 60 dB Gain (150 Ω Source Impedance)	-130.50 dBu
EIN @ 60 dB Gain (0 Ω Source Impedance)	-133.25 dBu
Analog Send Calibration (ADC = 0 dBFS)	+21.5 dBu
Phantom Power (Switchable Per Channel)	+48 Volts

Table 5.4. ULN-8 Mic Pre Input + ADC

Mic Pre + ADC Frequency Response	
+0/-0.1dB @ fs = 44100 Hz	8.9 Hz - 20.5 kHz
+0/-1.0dB @ fs = 44100 Hz	2.9 Hz - 21.0 kHz
+0/-0.1dB @ fs = 96000 Hz	8.9 Hz - 43.9 kHz
+0/-1.0dB @ fs = 96000 Hz	2.9 Hz - 45.4 kHz

Mic Pre + ADC Frequency Response	
+0/-0.1dB @ fs = 192000 Hz	8.9 Hz - 42.1 kHz
+0/-1.0dB @ fs = 192000 Hz	2.9 Hz - 64.7 kHz
5° low-end in-channel phase deviation point	20.0 Hz
Interchannel phase 0 Hz - 20 kHz	< ±0.05°
Crosstalk from SMPTE Input	< -142 dB

Table 5.5. ULN-8 Mic Pre + ADC Frequency Response

Line Input + ADC	
Input Impedance	10k Ω
Dynamic Range (-60 dB, flat 0-22.05 kHz, typ)	115 dB
Dynamic Range (-60 dB, A-weighted, typ)	118 dB
Noise Floor (flat 0-22.05 kHz, typ)	115 dB
Noise Floor (A-weighted, typ)	117.9 dB
THD D/A/A/D loop @ -12 dBFS	0.0005 %
THD D/A/A/D loop @ -0 dBFS	0.0015 %
Crosstalk @ 1kHz	-127 dB
IMD 1k component (19 kHz/20kHz @ +8dBu)	-104 dBu
Gain Range	-12 dB to +31.5 dB
Gain Precision	±0.05 dB
Gain Step	0.5 dB

Table 5.6. ULN-8 Line Input + ADC

Line Input + ADC Frequency Response	
+0/-0.1dB @ fs = 44100 Hz	5.7 Hz - 20.5 kHz
+0/-1.0dB @ fs = 44100 Hz	1.8 Hz - 21.0 kHz
+0/-0.1dB @ fs = 96000 Hz	5.7 Hz - 43.9 kHz
+0/-1.0dB @ fs = 96000 Hz	1.8 Hz - 45.4 kHz
+0/-0.1dB @ fs = 192000 Hz	5.7 Hz - 42.1 kHz
+0/-1.0dB @ fs = 192000 Hz	1.8 Hz - 64.7 kHz
5° low-end in-channel phase deviation point	10.7 Hz
Interchannel phase 0 Hz - 20 kHz	< ±0.05°
Crosstalk from SMPTE Input	< -142 dB

Table 5.7. ULN-8 Line Input + ADC Frequency Response

Latency	
A/D	63 samples
D/A	44 samples

Table 5.8. ULN-8 Converter Latency

Input Processing	
M/S Decode	Instantiable

Input Processing	
parametric EQ	Instantiable
Dynamics	Instantiable
Limiting	Instantiable
Character	Instantiable
Transient Control	Instantiable
Reverb	Instantiable
Delay	Instantiable
Nearly Infinite Combinations via +DSP	Instantiable

Table 5.9. ULN-8 Input Processing

Output Processing	
M/S Encode	Instantiable
parametric EQ	Instantiable
Dynamics	Instantiable
Limiting	Instantiable
Character	Instantiable
Transient Control	Instantiable
Reverb	Instantiable
Dither	Instantiable
Mix Foldown	Instantiable
Signal Generation	Instantiable
Nearly Infinite Combinations via +DSP	Instantiable

Table 5.10. ULN-8 Output Processing

Front Panel	
Indicator LEDs (Bicolor)	495
Meters	16
Segments per Meter	15
Channel Encoder Knobs	8
Encoders Have Shaft Push Buttoms	Yes
Monitor Controller Encoder Knobs	1
Monitor Controller Mute	Yes
Monitor Controller Dim	Yes
Indicators Per Encoder	16
Sample Rate Indicators	6
Clock Source Indicators	4
AES Clock Source Indicators	4
AES Lock Indicators	4
AES Input Signal Indicators	4
AES Output Signal Indicators	4

Front Panel	
System Lock Indicator	1
Firewire Indicator	1
Front Panel Modes	14
Headphone Output (Dedicated DAC)	TRS Stereo
DI Inputs (2)	TRS Balanced
DI Input Impedance	10M Ω
DI Input Fixed Gain (Jumper Selectable)	0/10/20 dB
DI Input Variable Gain	-12 dB to 31.5 dB
Power Switch (Jumper Defeatable)	Toggle

Table 5.11. ULN-8 Front Panel

Back Panel	
MIDI Connectors (In and Out)	5-Pin DIN
SMPTE Connectors (In and Out)	Balanced TRS
Word Clock Connectors (In and Out)	75 Ω BNC
AES Connector (8 Channels In and Out)	DB-25
AES Connector Pinout	Tascam/Digidesign Digital
Mic Input Connector (8 Channels)	DB-25
Mic Input Pinout	Tascam/Digidesign Analog
Line Input Connector (8 Channels)	DB-25
Line Input Pinout	Tascam/Digidesign Analog
Send Output Connector (8 Channels)	DB-25
Send Output Pinout	Tascam/Digidesign Analog
Line/Monitor Output Connector (8 Channels)	DB-25
Line/Monitor Output Pinout	Tascam/Digidesign Analog
Line/Monitor Mult Output Connectors (Analog 1/2)	TRS Balanced
Gigabit Ethernet MHLink Connectors (2)	RJ-45 8-pin
USB-C	
Metric Halo EdgeCard™ expansion slot	
Power (Unswitched)	2.1mm Coaxial
Power (Switched)	4-Pin XLR
Security Slot	Kensington

Table 5.12. ULN-8 Back Panel

Software	
Driver	CoreAudio Mac OS X
Mac OS X	10.8.5 or newer
Architecture	Intel
MIOConsole3d	Included
Record Panel	Included

Software	
LTC Decoder	Included
Mixer	Included
DSP Processing	Included

Table 5.13. ULN-8 Software

Power	
Voltage	14v - 28v
Power	32 Watts
External Supply (Input)	100-240VAC
External Supply (Max Input Current)	1.2 A
External Supply (Output)	18 VDC
External Supply (Max Output Current)	2.0 A
External Supply Connector	4-Pin XLR

Table 5.14. ULN-8 Power

Case	
Material	Powder Coated Aluminum
Fasteners	3mm Phillips Head
Weight	6 lbs
Weight	2.7 kg
Dimensions	17" x 13" x 1.72"
Dimensions	432 x 330 x 44 mm
Rack Ears (included)	Powder Coated Steel
Rear Rack Mounts	Available

Table 5.15. ULN-8 Case

6. LIO-8 Users Guide

LIO-8 Overview

Figure 6.1: LIO-8

What it is

The LIO-8 is a portable, archival–quality, modular multi-format audio converter, interface, and processor for professional audio applications. The LIO-8 is equipped with eight balanced line inputs, eight channels of AES digital I/O, eight balanced sends and eight balanced analog outputs, as well as wordclock in/out, MIDI in/out and SMPTE in/out.

Host computer connection and box daisy-chaining are provided by two MHLink Gigabit Ethernet ports, each of which support 128 channels of bidirectional audio at 192khz/32 bit data rate. A UAC2 class-compliant USB-C connection is available as an alternative computer interface.

All inputs and outputs are capable of 24-bit/192kHz operation.

What it has

- 18 simultaneous input channels and 20 simultaneous output channels
- Full 24 bit/192kHz audio
- 44.1, 48, 88.2, 96, 176.4, 192kHz Sampling Rates
- 8 channels of 24 bit A/D converters
- 10 channels of 24 bit D/A converters
- 8 Balanced Line Inputs – DB25
- 8 Balanced line level Sends – DB25
- 8 Balanced Analog Outputs – DB25, channels 1 and 2 multed on 1/4" TRS
- 8 channels of AES I/O, single wire for 44.1-192KHz operation - DB25
- 2 channels of DI available on the front panel - 1/4" TRS
- Front panel cans output with discrete high quality D/A and amplification - 1/4" TRS
- Word Clock 1x on 75ohm terminated BNC
- SMPTE I/O on 1/4" TRS
- MIDI I/O on 5-pin DIN
- MH EdgeBus programmable and pluggable audio expansion slot.

- Enhanced processing power and massive hardware memory. All 3d hardware includes over a hundred DSP plug-ins and the unique Metric Halo Graph environment within the 3d mixer.
- 128 input x 64 bus (at all sample rates) multi-box unified zero-latency mixer
- 1024 x 1024 internal audio routing matrix (per box)
- Front Panel Metering for Analog Inputs and Outputs on 15 segment multicolor LEDs
- Front Panel signal present, lock and clock selection multicolor LEDs for AES I/O
- 9 detented Front Panel encoders
- Full console metering of every channel and mix bus
- Total recall of every console parameter
- Portable Capabilities – Battery Powerable
- Rack Mount Kit

Options (can be installed before or after purchase)

- 4 or 8 channels of ULN-R Microphone preamps
- Metric Halo Edge Card expansion modules
  Click here for more information on the available Edge Card configurations

What you need to use it

- Computer:
  - Any Mac that supports Mac OS X 10.8.5 with a Gigabit Ethernet or UAC2 class-compliant USB connection
  - Cat 5e or better cable (e.g. with all 8 conductors)
  - Mac OS X 10.8.5 or newer required
  - Mac OS X 10.11.5 or newer recommended
- Peripheral Gigabit Ethernet Adaptors supported:
  - Apple Thunderbolt Gigabit Ethernet Adapter
  - PCIe Gigabit Ethernet Adapter
  - Third-party Thunderbolt Gigabit Ethernet Adapter (e.g. as part of a dock)
  - USB3 Gigabit Ethernet Adapter
- Software: All CoreAudio compliant software is compatible with Mobile I/O, including:
  - Cubase
  - Digital Performer
  - GarageBand
  - iTunes
  - Live
  - Logic
  - Nuendo

- Pro Tools
- Studio One
- Reaper
- MixBus / Ardour
- and hundreds more…

What comes with it

Your LIO-8 3d package contains the following items:

- One LIO-8 3d unit:

Figure 6.2: LIO-8 Unit

- One IEC Power Cord appropriate for your area
- One 18-volt 48-watt world-ready external power supply
- One 12” CAT-5e Gigabit Ethernet Cable
- One 14' CAT-5e Gigabit Ethernet Cable
- Two Rack Ears w/ fasteners
- Jumpers for internal configuration
- Warranty/Registration Card

If any of these items are missing from your package when you open it, please contact Metric Halo or your dealer immediately for assistance.

Using the LIO-8 Hardware

LIO-8 Front Panel

Figure 6.3: LIO-8 Front Panel

The front panel provides LIO-8 system control and status at a glance (The Front Panel includes features that are only enabled if you install one or two of available add-on Mic Pre boards. When no Mic Pre boards are installed, these features are disabled and are present for future expansion. The Mic Pre related features are noted below.):

- Monitor - Select the Monitor Controller input or output
- Preset - Recall one of the eight stored system configurations
- Input - Step through the available input sources (only active if fitted with optional mic pres)
- Link - Link multiple input and output encoders for stereo or multichannel use
- +48 - Enable phantom power, per channel (only active if fitted with optional mic pres)
- U/M - User mode (for future expansion)

- Power switch for rear 4 pin XLR connector (jumper defeatable)
- Sample Rate (nominal 44.1, 48, 88.2, 96, 176.4 or 192KHz)
  - The Sample Rate is determined from the currently selected clock source so it will accurately indicate the current sample rate, even when the clock source is being provided by an external device.
- Clock source:
  - Internal indicates that the system is internally clocked
  - Wordclock indicates that system is being clocked from the wordclock input
  - Digital In indicates that the system is being clocked from the selected digital input
- Control Mode — Indicates what parameters the encoders are modifying:
- I/O Trim — Indicates whether you are modifying input or output channel gain.
- Input Status - Indicates mic input, line input or mic s/r by channel (fixed on “Line” if no Mic Pre is fitted).
- Encoders - Eight detented encoders for multipurpose control, with push-switches.
- 15-segment metering for the 8 analog inputs and outputs using multicolor LEDs, which also display gain values during encoder use. The meters are fast PPM peak reading meters with auto-resetting peak holds.
- SysLock — Indicates that the system clock recovery circuit is properly locked to the selected clock source. If this light is not illuminated, the LIO-8 will not be locked to a clock and will revert to its

failsafe internal clock source. Even if the Locked light is not illuminated, the actual sample rate will still be indicated on the front panel display.

- FireWire — Indicates that the LIO-8 has been successfully connected to either a USB or MHLink Gigabit Ethernet connection to the Host computer.
- Signal present indicators for AES input and output, as well as digital lock and cable select button and indicators.
- Monitor Control Section:
  - The Mute and Dim buttons provide instant access to simple level control for the selected Monitor path or headphone output. The Mute button provides a quick, tactile "panic switch" which mutes the monitors or front panel headphone output in case of accidental feedback loops and other audio unpleasantries. The Dim button attenuates the selected path by 20 dB.
  - The Monitor Control encoder provides front panel adjustment for your audio. By pressing the encoder, it can toggle between affecting the Monitor Control section or Headphones. There are two multicolor LEDs below the encoder that indicate which path the encoder is modifying as well as mute and dim status.

The LIO-8 front panel also provides access to the Headphone output. The headphone output jack is a TRS 1/4" jack that provides the Left Channel on the tip, the Right Channel on the ring and the ground return for the two channels on the sleeve. These signals are all ground referred, so they may also be split and fed single-ended (unbalanced) to an external audio device.

The LIO-8 headphone output is suited to a wide range of impedances. Headphones with lower impedance would be expected to get louder than those with higher impedance.

There are two DI inputs on the front panel, which are paralleled with Line inputs 1 and 2. The DI inputs can provide 0, 10, or 20db of gain selectable via internal jumpers.

More information regarding the front panel can be found in the ULN/LIO-8 Front Panel Guide.

LIO-8 Rear Panel

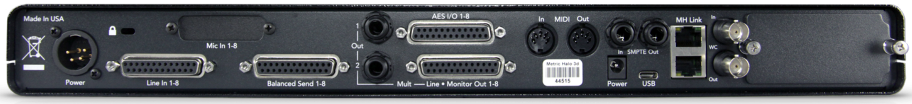


Figure 6.4: LIO-8 Rear Panel

The LIO-8 rear panel features:

- 4 pin XLR DC power jack (14v - 28v, 32 Watts)
- 8 channels balanced line inputs on DB25. Each input has:
  - -12 dB to +31.5 dB of gain range
  - Dynamic Range (-60 dB, flat 0-22.05 kHz, typ): 115 dB
  - +0/-1.0dB @ fs = 192KHz: 1.8 Hz - 64.7 kHz
  - Noise Floor (flat 0-22.05 kHz, typ): 115 dB
- 8 channels balanced line level sends on DB25 that mirror the analog inputs
- 8 channels balanced line/monitor outputs on DB25. Each output has:
  - 24-bit 192kHz D/A converters (120dB SNR)
  - Gain range from -96 dB to +30 dB
  - Outputs 1 and 2 are also multed to 1/4" TRS connectors
- 8 channels AES digital I/O on DB25
  - Single wire mode for full 8 channels at 44.1-192KHz operation
- Expansion Port for DB-25 for 4 or 8 channels of Mic Pre (field installable)
- Wordclock input/output on BNC connectors
- MIDI I/O to connect a control surface to MIOConsole3d
- SMPTE input and output on 1/4" TRS
- 2 MHLink Gigabit Ethernet ports
- 1 USB-C port
- Kensington security slot
- 1 4-pin XLR power port, switched, compatible with any 4-pin XLR power system with the following characteristics: 9v - 30v DC, Pin 4 Hot, 15 Watts

LIO-8 3d Signal Flow

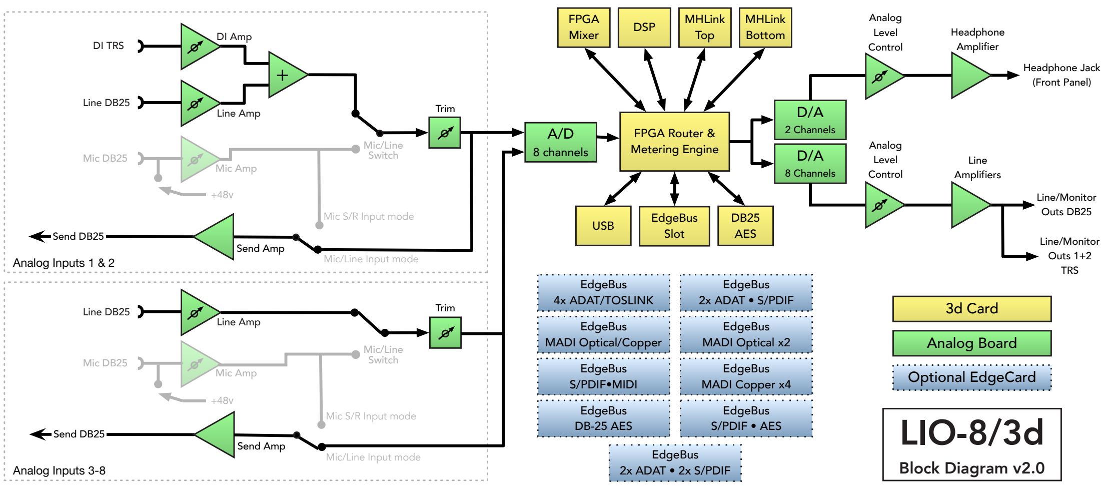


Figure 6.5: LIO-8 3d Signal Flow

Note: Signal flow of the optional ULN-R Mic preamp module is shown ghosted in the graphic above.

Click here for a larger version

Making connections to the LIO-8 3d

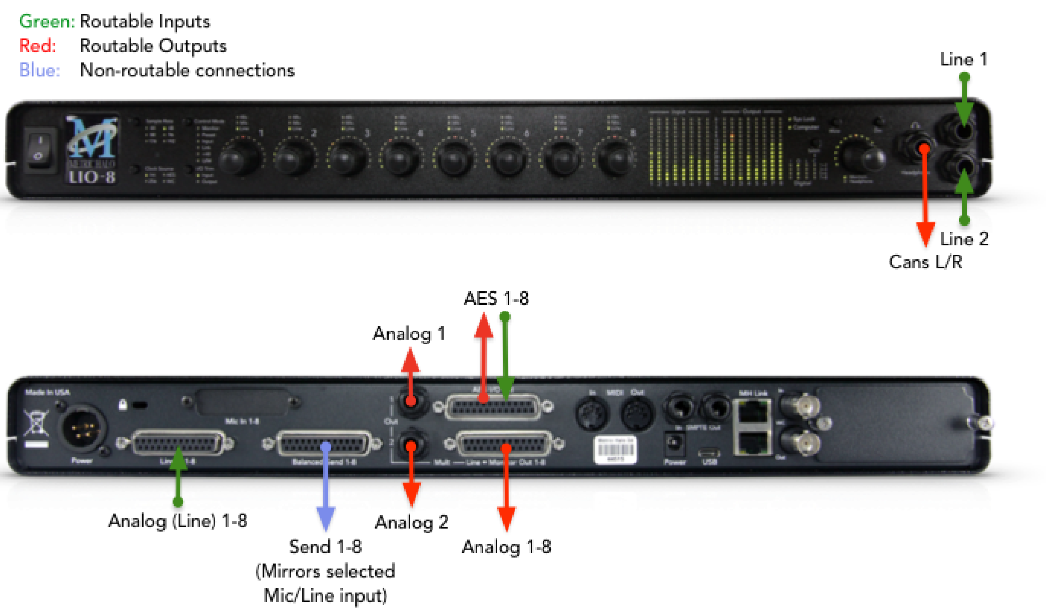


Figure 6.6: LIO-8 3d Routing

Click here for more information on the available Edge Card configurations

There are eight classes of connections you can make to the LIO-8 3d hardware:

1. Gigabit Ethernet (MHLink)
2. USB
3. Analog Audio
4. AES Digital Audio
5. Clock Sync
6. MIDI
7. SMPTE
8. Power

Computer Connections

- **MHLink (Gigabit Ethernet)** - The preferred method of connecting your computer to the LIO-8 (or any Mobile I/O 3d audio device) is with a Gigabit Ethernet cable to one of the MHLink ports. This method enables the full capabilities of the 3d/MHLink environment with 128 channels at 32bit integer resolution input and output at all sample rates up to 192kHz. Daisy-chaining additional MHLink boxes dynamically and transparently integrates the new devices into the currently-running 3dConsole mixer, making all the added I/O ports and DSP engine power immediately available for use.

  MHLink requires the installation of the MHLinkDriver software. The latest MIOConsole3d installer package is available from: macOS MIOConsole3d installers (this will require a restart). Please see the section on software installation if you need clarification.

  > **Driver Installation on macOS 10.13 and newer**
  > With High Sierra (macOS 10.13) Apple introduced another layer of driver (system extension) software security.
  >
  > In order for third-party drivers (system extensions) to be activated on new installs of High Sierra, the user of the computer (you) must explicitly approve the activation, after the installation has occurred.
  >
  > You must do this before you will be able to use your hardware via MHLink.
  >
  > The authorization process is the same for all third-party drivers. If you are unfamiliar with the process, please read our FAQ answer here: macOS Driver User Authentication.

  Note: With MHLink, you need to launch MIOConsole3d to establish the routing between MHLink and the attached box(es). The routing is established automatically as soon as you launch MIOConsole3d.

- **USB** - The secondary method of connecting to your LIO-8 is with USB. USB connections for audio transport will be limited compared to MHLink. In a pinch, however, your LIO-8 can be seen as an audio device by any computer with a UAC2 class-compliant USB port.

  Note: Connecting to USB does not require the installation of a software driver. See MIOConsole3d Preferences > Discovery before connecting USB to a computer which is also connected to an MHLink ethernet interface.

Analog Audio Connections

The analog I/O connections on the LIO-8 have been engineered for maximum flexibility in that they support both balanced and unbalanced connections with a wide range of input and output levels and a wide range of matching impedances. The line inputs have an impedance of 10k Ω. With that in mind, there are a number of aspects of the design that you should take into account when interfacing with the LIO-8. All multichannel audio connections are made using industry standard DB25 connectors, which are pin-compatible with cables using the Tascam/Digidesign standard.

Wiring details for this connector can be found in the DB25 Pinouts appendix.

Whenever possible, use balanced connections with the LIO-8. The performance of balanced interconnects is much higher and much more resistant to noise interference and electrical (power) wiring problems. The expense of balanced interconnects is not substantially higher than unbalanced connections, so if the gear that you are interfacing with supports balanced connection — use it. If you cannot utilize balanced interconnects, there are connection schemes that you can use that will maximize performance.

On input, at line level, it is sufficient to simply use standard unbalanced connections. If you are interfacing with the LIO-8 line inputs, you will need to ensure that the cold pin is grounded in the unbalanced adapter cable (Pin 3 if connecting to an XLR loom, Ring if connecting to a TRS loom). More information about adjusting the input level can be found in the MIOConsole3d software chapter.

On output, at line level, it is sufficient to simply use standard unbalanced connections. If you are interfacing with the LIO-8 line inputs, for best performance you should ensure that the cold pin is left floating in the unbalanced adapter cable (Pin 3 if connecting to an XLR loom, Ring if connecting to a TRS loom). More information about adjusting the output level can be found in the MIOConsole3d software chapter.

LIO-8 DB25 cables should be wired pin 2 or Tip hot. The front and rear 1/4" connectors are wired Tip hot. The headphone connector is wired Tip/Left, Ring/Right, Sleeve/Ground.

Connecting to the DIs

The DI inputs provide an excellent high impedance input for directly connecting instruments to the LIO-8. These inputs are unity-gain summed with the corresponding line input from the back panel. This means that if you have signal connected to both the DI and line input for a given channel the ADC will see a mix of those two inputs. If you do not wish to have signal from one of the connectors you need to physically disconnect that input or otherwise ensure that no signal is present at the unused input.

Each DI input supports balanced connections via TRS cable. You can connect a source to the DI via a balanced interconnect (if the source is actually a balanced output), an unbalanced interconnect (if the source is unbalanced like most instruments), or a telescoping shielded cable (which is an alternate unbalanced connection that provides additional shielding).

Since the DI inputs provide an ultra-high input impedance, if you connect an unbalanced source to the DI via a balanced cable you will float the negative input and will effectively inject a large DC offset into the output of the DI which will cause the associated ADC to mute. As a result you must use either a TS cable or a telescoping shield cable with unbalanced sources.

When you remove the connector from the DI jack, the DI connects its inputs to ground internally to remove any possible residual DC offset or pickup noise. If you disconnect the cable from the source but leave it connected to the DI input, the cable will function as an antenna and will inject noise into your inputs and may cause enough of a DC offset to mute the ADC. As a result you should always disconnect the cable from the DI rather than the instrument.

Balanced Sends

The LIO-8 has eight balanced sends; these mirror the analog signal present at the Line input. This allows you to use the sends as a splitter to feed a mixing console, recorder or other equipment.

You cannot route signal from a DAW channel to the balanced sends; to route signals from your computer to the D/A converters, you must use the Line/Monitor outputs.

AES Digital Audio

The LIO-8 supports 8 channels of digital audio over copper-based connections. The LIO-8 operates in single-wire mode, providing 8 channels of digital audio at all sample rates. The LIO-8's digital I/O connections are made using industry standard Tascam/Digidesign pinout DB25 cables.

Wiring details for this connector can be found in the DB25 Pinouts appendix.

Digital audio interface options can be greatly expanded through the addition of a Metric Halo EdgeCard™.

Clock Sync

Clock sync is a serious consideration in any digital audio system.

If you are recording analog sources with LIO-8, you can simply use the unit's high-quality internal clock source to drive the converters. This is the easiest case to deal with.

If you need to interface with other devices digitally or ensure sample accurate sync with video sources, the extensive clock synchronization capabilities of LIO-8 will prove to be more reliable (and better sounding) than much higher priced alternatives.

There are three different ways to get external clock information into the unit:

1. Sending a word clock signal into the WC Input BNC.
2. Sending an AES signal into the Digital input.

The BNC word clock input port is a 75 Ohm terminated coaxial input. It should be driven by a 75 Ohm source driver and interconnected with 75 Ohm coaxial cable. If you do not use proper cabling and source drive, you will introduce reflections on the word clock cable which will propagate jitter into the recovered word clock.

The AES recommended procedure for distributing clock is to use an AES clock signal. The AES clock signal is an AES digital audio signal with no audio activity. The LIO-8 only uses the AES preambles for clock recovery, so it is immune to data dependent jitter effects. This means you can reliably use the Digital Input as a clock source with or without audio data.

MIDI

The LIO-8 offers MIDI input and output ports for direct connection of a control surface. These ports are only active while MIOConsole3d is running; the LIO-8 cannot utilize a control surface in standalone operation. MIOConsole3d makes these ports available to other applications while it is running, but the MIDI implementation is currently optimized for control use only- it is not recommended to use these ports for connecting keyboards or other devices that require accurate timing.

SMPTE

The LIO-8 has SMPTE input and output ports dedicated for timecode use. The SMPTE input and output are routed from the "SMPTE" category in the MIOConsole3d Source and Destination routing menus. Please note that MIOConsole3d does not natively decode or encode MTC at this time; if you need to synchronise to MTC, you will need third party software.

Power

One of LIO-8's great strengths is the flexibility of its power system. The LIO-8 can be powered from any DC source in the range of 14V to 28V as long as it provides 32 Watts of power. The DC inputs on the LIO-8 are a 2.1mm coaxial power connector, center positive and a 4-pin XLR connector Pin 4 Hot. So if you are powering the unit with a third party power source and it supplies 14V, the power source will have to provide 2.3 amps of current. If you are powering the unit with 24V, the power source will have to provide 1.3 amps of current, and so on.

The LIO-8 ships with a world-ready 18 volt, 2 amp power supply. You can plug this supply into any AC power source from 90V to 240V, 50Hz - 60Hz, using an appropriate IEC power cord, and it will supply the proper power to the LIO-8 on the 4 pin XLR power connector. Unfortunately, the power capacity supplied through a USB bus is unsufficient to power a LIO-8.

As with all electronic devices, when connecting an external power source to the LIO-8, you should first connect the power source to the LIO-8 while it is in an unenergized state (e.g. not connected to the mains or switched off). After the connection to the LIO-8 has been made, you should energize the power source.

If you connect an energized power source to the LIO-8’s 2.1mm power connector you may see a small spark when you make the connection. This is due to surge current and is normal if you connect a power source in this way. While this will not damage the LIO-8 in any way, to avoid the spark just connect the power connector to LIO-8 before connecting the power source to the wall.

LIO-8 Specifications

Rails	
Analog Rails (low power)	±9.9 Volts
Analog Rails (high power)	±12.6 Volts

Table 6.1. LIO-8 Voltage Rails

Maximum I/O Levels (Balanced)	
Peak Line Output @ 0 dBFS (no jumper/low power)	+18.5 dBu
Peak Line Output @ 0 dBFS (output jumper/low power)	+22.0 dBu
Peak Line Output @ 0 dBFS (output jumper/high power)	+24.5 dBu
Analog Send Max Output	+21.5 dBu
Line In Max Input	+24.5 dBu
Output Impedance	5 Ω

Table 6.2. LIO-8 Maximum I/O Levels (Balanced)

Monitor Controller	
Nominal FS output (Balanced) Output Jumper Off	-19.0 dBu
Nominal FS output (Balanced) Output Jumper On	-12.0 dBu
Maximum Output	Same as Line
Gain Range	-96 dB to +30 dB
Gain Precision	±0.05 dB
Gain Step	0.5 dB

Table 6.3. LIO-8 Monitor Controller

Line Input + ADC	
Input Impedance	10k Ω
Dynamic Range (-60 dB, flat 0-22.05 kHz, typ)	115 dB
Dynamic Range (-60 dB, A-weighted, typ)	118 dB

Line Input + ADC	
Noise Floor (flat 0-22.05 kHz, typ)	115 dB
Noise Floor (A-weighted, typ)	117.9 dB
THD D/A/A/D loop @ -12 dBFS	0.0005 %
THD D/A/A/D loop @ -0 dBFS	0.0015 %
Crosstalk @ 1kHz	-127 dB
IMD 1k component (19 kHz/20kHz @ +8dBu)	-104 dBu
Gain Range	-12 dB to +31.5 dB
Gain Precision	±0.05 dB
Gain Step	0.5 dB

Table 6.4. LIO-8 Line Input + ADC

Line Input + ADC Frequency Response	
+0/-0.1dB @ fs = 44100 Hz	5.7 Hz - 20.5 kHz
+0/-1.0dB @ fs = 44100 Hz	1.8 Hz - 21.0 kHz
+0/-0.1dB @ fs = 96000 Hz	5.7 Hz - 43.9 kHz
+0/-1.0dB @ fs = 96000 Hz	1.8 Hz - 45.4 kHz
+0/-0.1dB @ fs = 192000 Hz	5.7 Hz - 42.1 kHz
+0/-1.0dB @ fs = 192000 Hz	1.8 Hz - 64.7 kHz
5° low-end in-channel phase deviation point	10.7 Hz
Interchannel phase 0 Hz - 20 kHz	< ±0.05°
Crosstalk from SMPTE Input	< -142 dB

Table 6.5. LIO-8 Line Input + ADC Frequency Response

Latency	
A/D	63 samples
D/A	44 samples

Table 6.6. LIO-8 Converter Latency

Input Processing	
M/S Decode	Instantiable
parametric EQ	Instantiable
Dynamics	Instantiable
Character	Instantiable
Reverb	Instantiable
Delay	Instantiable
Nearly Infinite Combinations via DSP Graph	Instantiable

Table 6.7. LIO-8 Input Processing

Output Processing	
M/S Encode	Instantiable
parametric EQ	Instantiable

Output Processing	
Dynamics	Instantiable
Character	Instantiable
Reverb	Instantiable
Dither	Instantiable
Mix Foldown	Instantiable
Nearly Infinite Combinations via DSP Graph	Instantiable

Table 6.8. LIO-8 Output Processing

Front Panel	
Indicator LEDs (Bicolor)	495
Meters	16
Segments per Meter	15
Channel Encoder Knobs	8
Encoders Have Shaft Push Buttoms	Yes
Monitor Controller Encoder Knobs	1
Monitor Controller Mute	Yes
Monitor Controller Dim	Yes
Indicators Per Encoder	16
Sample Rate Indicators	6
Clock Source Indicators	4
AES Clock Source Indicators	4
AES Lock Indicators	4
AES Input Signal Indicators	4
AES Output Signal Indicators	4
System Lock Indicator	1
Firewire Indicator	1
Front Panel Modes	14
Headphone Output (Dedicated DAC)	TRS Stereo
DI Inputs (2)	TRS Balanced
DI Input Impedance	10M Ω
DI Input Fixed Gain (Jumper Selectable)	0/10/20 dB
DI Input Variable Gain	-12 dB to 31.5 dB
Power Switch (Jumper Defeatable)	Toggle

Table 6.9. LIO-8 Front Panel

Back Panel	
MIDI Connectors (In and Out)	5-Pin DIN
SMPTE Connectors (In and Out)	Balanced TRS
Word Clock Connectors (In and Out)	75 Ω BNC
AES Connector (8 Channels In and Out)	DB-25

Back Panel	
AES Connector Pinout	Tascam/Digidesign Digital
Line Input Connector (8 Channels)	DB-25
Line Input Pinout	Tascam/Digidesign Analog
Send Output Connector (8 Channels)	DB-25
Send Output Pinout	Tascam/Digidesign Analog
Line/Monitor Output Connector (8 Channels)	DB-25
Line/Monitor Output Pinout	Tascam/Digidesign Analog
Line/Monitor Mult Output Connectors (Analog 1/2)	TRS Balanced
Gigabit Ethernet MHLink Connectors (2)	RJ-45 8-pin
USB-C	
Metric Halo EdgeCard™ expansion slot	
Power (Unswitched)	2.1mm Coaxial
Power (Switched)	4-Pin XLR
Security Slot	Kensignton

Table 6.10. LIO-8 Back Panel

Software	
Driver	CoreAudio Mac OS X
Mac OS X	10.8.5 or newer
Architecture	Intel
MIOConsole3d	Included
Record Panel	Included
LTC Decoder	Included
Mixer	Included
DSP Processing	Included

Table 6.11. LIO-8 Software

Power	
Voltage	14v - 28v
Power	24 Watts
External Supply (Input)	100-240VAC
External Supply (Max Input Current)	1.2 A
External Supply (Output)	18 VDC
External Supply (Max Output Current)	2.0 A
External Supply Connector	4-Pin XLR

Table 6.12. LIO-8 Power

Case	
Material	Powder Coated Aluminum
Fasteners	3mm Phillips Head

Case	
Weight	6 lbs
Weight	2.7 kg
Dimensions	17" x 13" x 1.72"
Dimensions	432 x 330 x 44 mm
Rack Ears (included)	Powder Coated Steel
Rear Rack Mounts	Available

Table 6.13. LIO-8 Case

7. ULN-2 Users Guide

ULN-2 Overview

Figure 7.1: Mobile I/O ULN-2

What it is

ULN-2 is the result of a dream to create a piece of audio gear that provides unbelievable audio quality while at the same time offering a degree of mobility and convenience that until very recently was simply not possible. The successful integration of world-class analog stages, excellent A/D/A conversion, its amazing digital mixing, routing, DSP and digital audio formats conversion with the dynamically expandable MHLink platform makes the ULN-2 a uniquely flexible and powerful computer audio interface.

The ULN-2 is equipped with two balanced analog inputs on Neutrik™ combo connectors, two balanced analog outputs (1/4" TRS)and two balanced monitor outputs for connecting directly to power amps and self powered monitors. The digital side includes fully configurable ADAT® / TOSLINK optical I/O (with full auto-SMUX support), BNC wordclock in/out plus independent stereo AES/EBU and S/PDIF Digital I/O ports (provided on the base-configuration S/PDIF•AES EdgeCard™). Host computer connection and box daisy-chaining are provided by two MHLink Gigabit Ethernet ports, each of which support 128 channels of bidirectional audio at 192khz/32 bit data rate. A UAC2 class-compliant USB-C connection is available as an alternative computer interface.

Analog inputs are capable of 24-bit/ 96kHz operation, although the analog outputs, digital i/o and internal routing and processing paths support sample rates up to 192kHz.

What it has

- 12 simultaneous input channels and 14 simultaneous output channels
- A/D and D/A conversion at 44.1, 48, 88.2, 96kHz Sampling Rates (D/A conversion operates at 176.4 and 192kHz. A/D conversion is not supported at 4x sample rates.)
- 2 independent channels of high gain, low-noise mic-pre with switchable phantom power
- 24 bit 110 dB Dynamic Range A/D converters
- 24 bit 120 dB Dynamic Range D/A converters
- Digital I/O, DSP processing, routing and formats conversion at 44.1, 48, 88.2, 96, 176.4 and 192kHz Sampling Rates
- 8 channels of ADAT® Optical Input and Output (or 2 TOSLINK). Optical inputs auto-sense ADAT® vs. TOSLINK format. Optical output format is user-selectable. SMUX is automatically implemented to support high sample rates.
- MH EdgeBus programmable and pluggable audio expansion slot.
  ULN-2 base configuration includes the S/PDIF•AES Stereo Digital I/O EdgeCard pre-installed

- Enhanced processing power and massive hardware memory. All 3d hardware includes over a hundred DSP plug-ins and the unique Metric Halo Graph environment within the 3d mixer.
- 128 input x 64 bus (at all sample rates) multi-box unified zero-latency mixer
- 1024 x 1024 internal audio routing matrix (per box)
- Front Panel Metering for Analog Inputs and Outputs
- Full console metering of every channel and mix bus
- Total recall of every console parameter
- Portable Capabilities – Battery Powerable
- Rack Mount Kit

Optional (can be installed before or after purchase)

- 1 or 2 channels of Jensen input transformers

What you need to use it

- Computer:
  - Any Mac that supports Mac OS X 10.8.5 with a Gigabit Ethernet or UAC2 class-compliant USB connection
  - Mac OS X 10.8.5 or newer required
  - Mac OS X 10.11.5 or newer recommended
- Peripheral Gigabit Ethernet Adaptors supported:
  - Apple Thunderbolt Gigabit Ethernet Adapter
  - PCIe Gigabit Ethernet Adapter
  - Third-party Thunderbolt Gigabit Ethernet Adapter (e.g. as part of a dock)
  - USB3 Gigabit Ethernet Adapter
- Software: All CoreAudio compliant software is compatible with Mobile I/O, including:
  - Cubase
  - Digital Performer
  - GarageBand
  - iTunes
  - Live
  - Logic
  - Nuendo
  - Pro Tools
  - Studio One
  - Reaper
  - MixBus / Ardour

- and hundreds more…

What comes with it

Your ULN-2 3d package contains the following items:

- One ULN-2 3d unit:

Figure 7.2: Mobile I/O Unit

- One IEC Power Cord appropriate for your area
- One 18-volt 48-watt world-ready external power supply
- base configuration includes one S/PDIF•AES Stereo Digital I/O EdgeCard (pre-installed) Note: alternate EdgeCard configurations are available at time of order.
- One 12" CAT-5e Gigabit Ethernet Cable
- One 14' CAT-5e Gigabit Ethernet Cable
- Two Rack Ears w/ fasteners
- Warranty/Registration Card

If any of these items are missing from your package when you open it, please contact Metric Halo or your dealer immediately for assistance.

Using the ULN-2 Hardware

ULN-2 Front Panel

Figure 7.3: ULN-2 Front Panel

The ULN-2 front panel provides ten-segment metering for the 2 analog inputs and the main outputs as well as knobs and switches to control the input, monitor, and headphones sections. The meters are fast VU meters with auto-resetting peak holds.

Each input channel has the following controls:

- Input gain knob — This is a 12 position gold-contact rotary switch which allows you to control the gain of the selected input.
- Phantom Power enable switch — This is a push-button switch which enables/disables Phantom power. Push the switch IN to enable phantom power.
- Trim Enable switch — This is a push-button switch which allows you to control whether the attenuator trim pot is in the signal path or not. Push the switch IN to enable the trim pot. The attenuation range of the trim pot is 0dB to -20dB.
- Mic/TRS switch — This is a push-button switch which selects the input stage. The ULN-2 has two distinct input stages: The Mic Amp and the DI Amp.
  - The Mic Amp is optimized for high gain and very low noise with low impedance sources like microphones. This input is connected to the XLR portion of the Neutrik combo connector. Maximum gain is 72 dB. Push the Mic/TRS switch IN to select the MIC input.
  - The DI amp is optimized for high impedance sources like magnetic pick-ups. This input is connected to the TRS portion of the Neutrik combo connector. Maximum gain is 63 dB. The Mic/TRS switch should be in the OUT position to use this input.
- Trim Pot — The trim pot controls a passive attenuator. The attenuator is buffered between the return receiver and the A/D converter so its operation is transparent with regard to sound quality. Push the the trim enable switch IN to enable the trim pot. The attenuation range of the trim pot is 0dB to -20dB.

The front panel also provides ULN-2 system status at a glance:

- Sample Rate (nominal 44.1, 48, 88.2, 96, 176.2 or 192kHz)
  - The Sample Rate is determined from the currently selected clock source so it will accurately indicate the current sample rate, even when the clock source is being provided by an external device.
  - 176.4kHz sample rate is indicated when both 44.1 and 48 are lit.
    192kHz sample rate is indicated when both 88.2 and 96 are lit.

    Note: A/D conversion is not available at 4x sample rates. Interestingly, the D/A converters operate fine at 4x rates, even though they are 96kHz-specified chips.
- Clock source:
  - Internal - indicates that the system is internally clocked

  - Wordclock - indicates that system is being clocked from the wordclock input
  - Digital In - indicates that the system is being clocked from the selected digital input (Optical, AES or S/PDIF)
- Power — Indicates that the ULN-2 is receiving power.
- FireWire — Indicates that the ULN-2 has been successfully connected to either a USB or MHLink Gigabit Ethernet connection to the Host computer.
- Locked — Indicates that the system clock recovery circuit is properly locked to the selected clock source. If this light is not illuminated, the ULN-2 will not be locked to a clock and will revert to its failsafe internal clock source. Even if the Locked light is not illuminated, the actual sample rate will still be indicated on the front panel display.
- Digital I/O Section:
  - The AES and S/PDIF lights indicate when a valid incoming digital audio stream is present. The Locked light indicates when the digital receiver is locked to the incoming digital audio signal.

The ULN-2 front panel provides access to the level control knobs for headphones and for the monitor outs. The headphone output jack is on the front panel and the monitor output jacks are located on the back of the unit.

The headphone output jack is a TRS 1/4" jack that provides the Left Channel on the tip, the Right Channel on the ring and the ground return for the two channels on the sleeve. The ULN-8 headphone output is suited to a wide range of impedances. Headphones with lower impedance would be expected to get louder than those with higher impedance.

The Monitor output jacks are balanced TRS connectors.

ULN-2 Rear Panel

Figure 7.4: ULN-2 Rear Panel

The ULN-2 rear panel features:

- 2 channels balanced MIC/LINE/INSTRUMENT inputs on Neutrik™ Combo connectors. Each input has:
  - 24-bit 96kHz A/D converters (110dB SNR)
  - high gain, low noise Mic amps with up to 72 dB of gain (fed by the XLR connector)
  - high gain, low noise DI amps with up to 63 dB of gain (fed by the TRS connector)
  - switchable input impedance characteristics (Mic input 3.3k Ohms, DI input 200k Ohms)
  - switchable 48V Phantom power (on XLR connector)
  - balanced analog inserts (S1, S2, R1, R2 jacks) which are post preamp but pre A/D. You can use the inserts to patch in analog processing between the preamp and the A/D converter. The send jack can also be used to send a mult of the input signal to another device while still using the A/D section of the ULN-2. This allows the ULN-2 to be used as an active mic splitter.
- 2 channels balanced TRS main outputs. Each output has:
  - 24-bit 96kHz D/A converters (120dB SNR) (functional at 4x SR)
  - switchable +4/-10 level
- 2 channels balanced Monitor output with front panel level control
  - Connect these outputs directly to power amps or self powered monitors
- 4-pin XLR power port for use with broadcast batteries
  - Compatible with any 4-pin XLR power system with the following characteristics: 9v - 30v DC, Pin 4 Hot, 15 Watts
- TOSLINK connectors for ADAT Optical or Optical S/PDIF I/O
  - 8 channels of ADAT® Lightpipe input (auto-senses and switches between ADAT® and Optical S/PDIF)
  - 8 channels of ADAT® lightpipe output (user-selectable between ADAT® and Optical S/PDIF)
- Wordclock input/output on BNC connectors
- Stereo S/PDIF input/output on RCA connectors (with base configuration S/PDIF•AES Edge Card)
- Stereo AES/EBU input/output on XLR connectors (with base configuration S/PDIF•AES Edge Card)
- 2 MHLink Gigabit Ethernet ports
- 1 USB-C port
- 1 2.1mm DC power jack (9v - 30v, center positive, 15 Watts)

Signal Flow

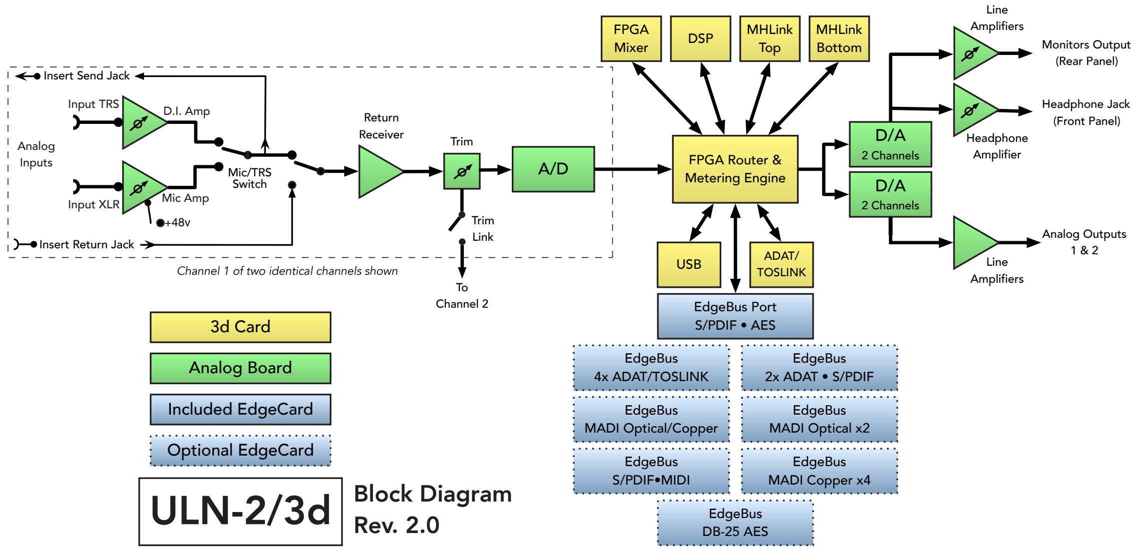


Figure 7.5: ULN-2 Signal Flow

Click here for a larger version

Making connections to the ULN-2

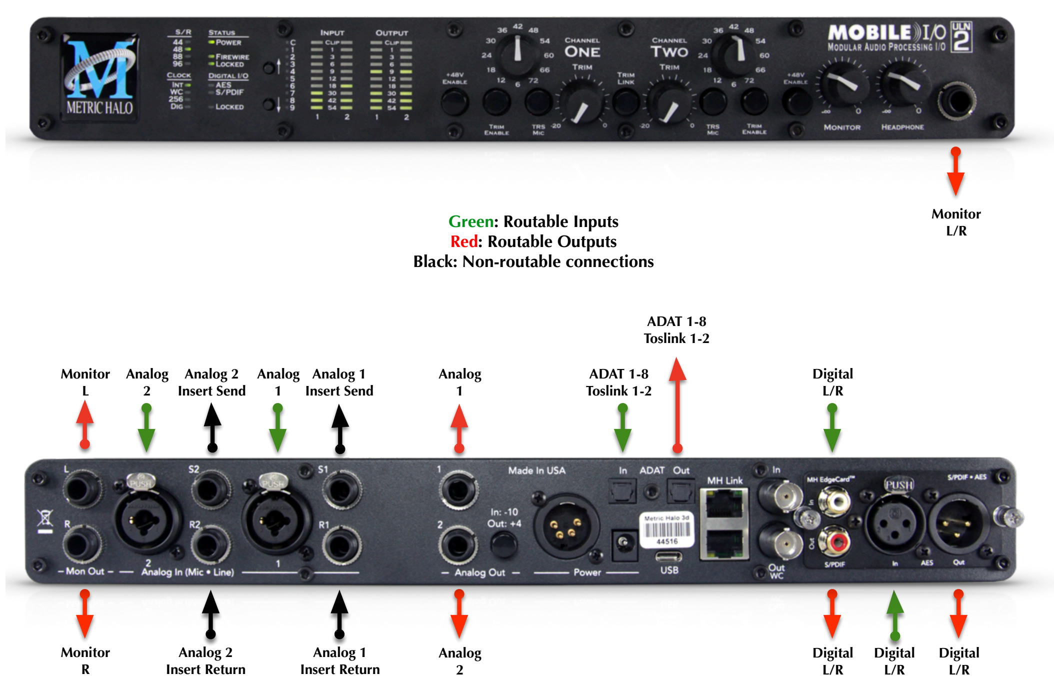


Figure 7.6: ULN-2 Routing (shown with standard S/PDIF•AES Edge Card installed)

Click here for more information on the available Edge Card configurations

What's the difference between the Analog outs and Monitor outs, and how do I send to the headphones?

- The Analog outputs get their signal from "Analog 1/2" in MIOConsole3d, and their level is controlled via software
- The Monitor outputs and headphones get their signal from "Monitor L/R" in MIOConsole3d, and their output levels are controlled via the front panel knobs

There are seven classes of connections you can make to the ULN-2 3d hardware:

1. Gigabit Ethernet (MHLink)
2. USB
3. Analog Audio
4. Copper-based Digital Audio
5. Optical-based Digital Audio
6. Clock Sync
7. Power

Computer Connections

- **MHLink (Gigabit Ethernet)** - The preferred method of connecting your computer to the ULN-2 (or any Metric Halo 3d audio device) is with a Gigabit Ethernet cable to one of the MHLink ports. This method enables the full capabilities of the 3d/MHLink environment with 128 channels at 32bit integer resolution input and output at all sample rates up to 192kHz. Daisy-chaining additional MHLink boxes dynamically and transparently integrates the new devices into the currently-running Console3d mixer, making all the added I/O ports and DSP engine power immediately available for use.

  MHLink requires the installation of the MHLinkDriver software. The latest MIOConsole3d installer package is available from: macOS MIOConsole3d installers (this will require a restart). Please see the section on software installation if you need clarification.

  Driver Installation on macOS 10.13 and newer
  With High Sierra (macOS 10.13) Apple introduced another layer of driver (system extension) software security.

  In order for third-party drivers (system extensions) to be activated on new installs of High Sierra, the user of the computer (you) must explicitly approve the activation, after the installation has occurred.

  You must do this before you will be able to use your hardware via MHLink.

  The authorization process is the same for all third-party drivers. If you are unfamiliar with the process, please read our FAQ answer here: macOS Driver User Authentication.

  Note: With MHLink, you need to launch MIOConsole3d to establish the routing between MHLink and the attached box(es). The routing is established automatically as soon as you launch MIOConsole3d.

- **USB** - The secondary method of connecting to your ULN-2 is with USB. USB connections for audio transport will be limited compared to MHLink. In a pinch, however, your ULN-2 can be seen as an audio device by any computer with a UAC2 class-compliant USB port.

  Note: Connecting to USB does not require the installation of a software driver. See MIOConsole3d Preferences > Discovery before connecting USB to a computer which is also connected to an MHLink ethernet interface.

Analog Audio Connections

The analog I/O connections on the ULN-2 have been engineered for maximum flexibility in that they support both balanced and unbalanced connections with a wide range of input and output levels and a wide range of matching impedances. This means that ULN-2 handles sources from mic level to line level and from mic impedance to guitar impedance. With that in mind, there are a number of aspects of the design that you should take into account when interfacing with ULN-2.

There are really three distinct analog input stages available in a ULN-2 input:

1. The Mic amp, which is fed by the XLR portion of the Combo connector.
2. The DI amp which is fed by the TRS portion of the combo connector.
3. The TRS return jack. This is a line level input which is the shortest path to the A/D converter.

Each input path is optimized for specific sources, but each is capable of handling a wide variety of sources. For example, both the Mic amp and the DI amp are capable of receiving Line level inputs. Additionally the DI input is capable of 63 dB of gain and can be used with dynamic microphones (phantom power is only available with the Mic Amp).

Feel free to experiment with the different input paths and choose the one which works best for a given application.

Whenever possible, use balanced interconnects with ULN-2. The performance of balanced interconnects is much higher and much more resistant to noise interference and electrical (power) wiring problems. The expense of balanced interconnects is not substantially higher than unbalanced connections, so if the gear that you are interfacing with supports balanced connection — use it.

If you cannot utilize balanced interconnects, there are connection schemes that you can use that will maximize performance.

On input, at line level, it is sufficient to simply use standard unbalanced (TS) connections. If you are interfacing with the ULN-2 XLR inputs, you will need to ensure that pin 3 is grounded in the unbalanced adapter cable. The ULN-2 XLR inputs are all wired pin 2 hot and the 1/4” inputs are wired Tip hot.

> **TIP:**
> To use the ULN-2 TRS input with guitar or bass, you can simply use a standard TS guitar cable (patch cord) and it will work fine. However, you can take advantage of the balanced input design of the ULN-2 to get more noise rejection than you thought possible on a guitar input.
>
> In order to do this, you will need to make a pseudo-balanced telescoping shield guitar cable. This can be constructed with a TRS connector, a TS connector and balanced microphone cable. This cable will treat the guitar as a floating balanced source and provide a telescoping shield from the ULN-2 ground.

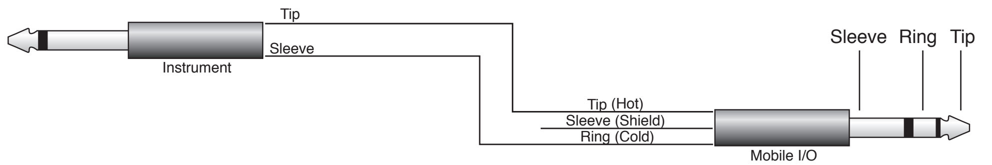


Figure 7.7: Telescoping Shield Cable for Instruments

If you want to use the TRS inputs with balanced microphones, you will need an XLR female to 1/4” TRS balanced plug adapter cable. These are available commercially, or you can construct one easily. The connections are Tip to Pin 2, Ring to Pin 3 and Sleeve to Pin 1:

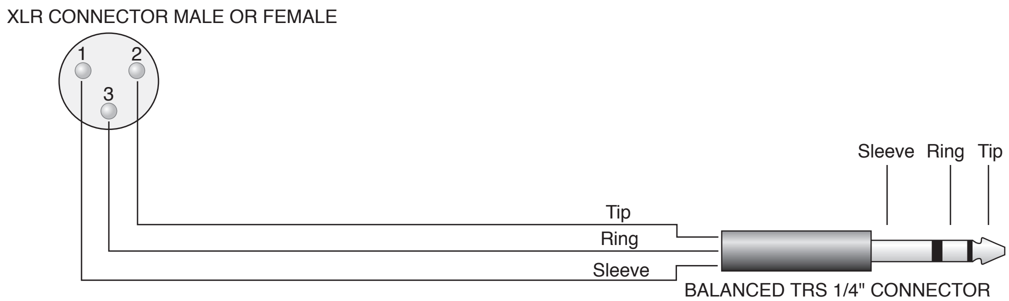


Figure 7.8: XLR to Balanced TRS Cable

On output, the situation is a bit more complex. If you are driving an unbalanced load, you will get the best performance by not connecting the ring of the TRS jack to ground. In order to do this, you can simply use a balanced TRS/TRS connector with the unbalanced gear. You can also construct a special cable with a TRS connector and a TS connector. In this cable, you just let the ring of the TRS connector float:

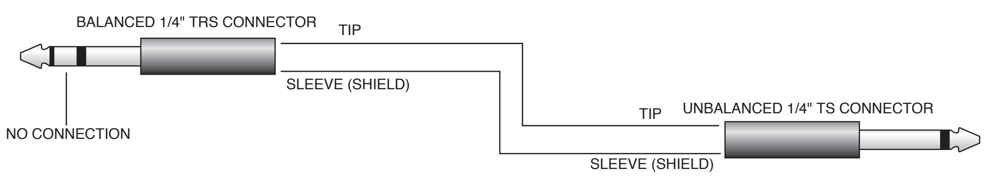


Figure 7.9: TRS to TS Unbalanced Cable

Alternatively, the TS connector can be replaced with an RCA connector for interfacing with gear that has RCA unbalanced interconnects.

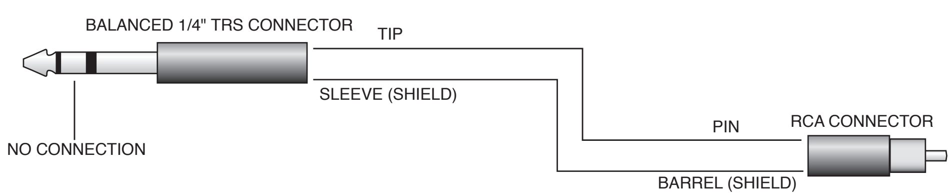


Figure 7.10: TRS to RCA Unbalanced Cable

Making a 1/4” connection
When you connect a 1/4” plug to a ULN-2 jack, insert it straight and firmly, ensuring that the plug is fully inserted into the jack. If the plug is not fully inserted you will get level shifts, phase flips, distortion, or no sound.

To disconnect a 1/4" plug, firmly pull the plug straight out from the connector body. The connectors on ULN-2 are stiff, so you may have to exert some force to remove the plug.

Making an XLR connection
When you connect a Male XLR plug to a ULN-2 jack, ensure that you have aligned the pins with the connector body and insert firmly until the retention tab clicks.

To disconnect the plug, press the metal retention tab flush against the box, and pull the plug from the ULN-2.

Copper-Based Digital Audio

The ULN-2 supports 4 channels of digital audio over copper-based connections using the S/PDIF interconnects with the RCA connectors and the AES interconnects using the XLR connectors. Even though only one of the AES or S/PDIF inputs can be a clock master source at any given time, you can have different digital sources connected to each of the input connectors at the same time – you use the MIOConsole3d application to select the active input. Audio routed to the digital outputs will be mirrored by both S/PDIF and AES outputs. This allows you to send the same stereo pair to two devices at once.

We recommend that you use the AES interconnect mechanism to establish the digital communication between the ULN-2 and other digital devices. The jitter and electrical noise tolerance on AES interconnects is substantially better than with S/PDIF interconnects. The AES interconnect standard is equivalent to balanced audio interconnections. If you need to use S/PDIF interconnects, try to use the shortest cables you can and, if possible, use special purpose 75 ohm S/PDIF or video cables.

The RCA connectors used for S/PDIF are friction fit coaxial connectors. When you connect them, ensure that they are fully inserted and tight.

The XLR connectors used for AES are fully locking. When connecting to them, make sure that you align the pins and insert firmly. When you remove the connector, make sure that you release the lock by pressing the lock release button before you pull the connector out of the ULN-2.

Wild digital input streams are still wild

Normally, when working with digital audio transport, you must take care to ensure that all devices communicating with one another are synchronized to the same audio clock. While you may find that the 3d digital audio inputs can often recover clean-sounding audio from unlocked audio devices running at the same sample rate, this is honestly just an accidental quirk of the over-built quality of the 3d hardware and *not a supported scenario*.

In production or critical listening situations, please always follow digital audio clocking best practices and ensure that your outboard devices are properly clock-slaved to the Metric Halo box, or the domain is properly slaved and locked to your outboard gear.

That said, if you find this quirk useful for a quick audition or casual listening, you are in good company.

Optical-Based Digital Audio

ULN-2 provides two TOSLINK™ connectors on the back panel. One is a transmit connector and the other is a receive connector. These connectors are used with Plastic Optical Fiber (TOSLINK) cables to communicate with other devices. The TOSLINK connectors can be used to communicate with either the ADAT® Optical communication protocol or the Optical S/PDIF communication protocol. Each port can be independently switched between the two protocols via the 3d Console.

The S/PDIF Optical communication protocol allows a device to transmit 2 channels of 24 bit audio at 44.1k through 192k, along with digital audio clock information.

The ADAT Optical standard allows a device to transmit 8 channels of 24-bit audio at up to 50kHz along with digital audio clock information.

After the original 8 channel ADAT specification was finalized, Alesis and other third parties extended the standard to support 4 channels of 24-bit audio at 2x sample rates and 2 channels at 4x sample rates. This extension of the standard is commonly referred to as SMUX. The ULN-2 automatically supports SMUX transport over the ADAT Optical connections when the box is in 2x mode (either 88.2k or 96k sampling frequency) or 4x mode (176.4k or 192k). While operating in SMUX mode, the audio data for each of the first four “ADAT” channels is placed on pairs of optical channels. ADAT 1 is placed on optical channels

1 and 2. Each optical channel continues to transmit data at 1x speeds, and both devices are required to multiplex and demultiplex the audio into and out of the optical channel pairs.

Clock Sync

Clock sync is a serious consideration in any digital audio system. The ULN-2 and any of it's digital audio interfaces can be used as the master clock source either for the ULN-2 by itself or for a full daisy-chained MHLink domain. In this regard, all 3d-equipped Metric Halo boxes are equal - only the physical I/O complement differentiates the box models.

If you are recording analog sources with ULN-2 by itself or as the root box of an MHLink domain, you can simply use the next-generation 3d internal clock source to drive the converters. This is the easiest case to deal with.

If you need to interface with other devices digitally or ensure sample accurate sync with video sources, the extensive clock synchronization capabilities of the 3d ULN-2 will prove to be more reliable (and better sounding) than much higher priced alternatives.

There are four different ways to get external clock information into the unit:

1. Sending a word clock signal into the WC Input BNC.
2. Sending an AES signal into the XLR Digital input.
3. Sending an S/PDIF signal into the RCA Digital input.
4. Sending an ADAT or S/PDIF signal into the Optical Digital input.

The BNC word clock input port is a 75 Ohm terminated coaxial input. It should be driven by a 75 Ohm source driver and interconnected with 75 Ohm coaxial cable. Even though the clock recovery qualities of the 3d hardware are second-to-none, incorrect cable impedance will introduce reflections on the word clock cable (jitter) which may adversely affect clock stability and audio quality.

The AES recommended procedure for distributing clock is to use an AES clock signal. The AES clock signal is an AES digital audio signal with no audio activity. The ULN-2 only uses the AES preambles for clock recovery, so it is immune to data dependent jitter effects. This rule applies to both the XLR AES and RCA digital inputs, which means you can reliably use the Digital Inputs as a clock sources with or without audio data.

The Optical input port will automatically adjust to any ADAT or TOSLINK signal it recieves at whatever sample rate. For example, incoming ADAT data SMUXed at 96kHz will be decoded automatically as 4 channels of audio clocked at 96kHz requiring no adjustments from the user. That said, ADAT/TOSLINK optical connections are generally more susceptible to jitter artifacts than copper AES, S/PDIF or word clock sources.

Power

One of the ULN-2’s great strengths is the flexibility of its power system. The ULN-2 can be powered from any DC source in the range of 9V to 30V as long as it provides 12 Watts of power. The ULN-2 provides two power supply inputs: a 2.1mm coaxial power connector (center positive), and a broadcast battery-compatible 4-pin XLR male (pin 4 hot). So if you are powering the unit with a third party power source and it supplies 9V, the power source will have to provide 1.4 amps of current. If you are powering the unit with 12V, the power source will have to provide 1 amp of current, and so on.

The ULN-2 ships with a world-ready 18 volt, 2 amp power supply. You can plug this supply into any AC power source from 90V to 240V, 50Hz - 60Hz, using an appropriate IEC power cord, and it will supply the proper power to the ULN-2 3d on the 4-pin XLR power socket.

Unfortunately, the power capacity supplied through a USB bus is unsufficient to power a Mobile I/O.

As with all electronic devices, when connecting an external power source to the ULN-2, you should first connect the power source to ULN-2 while it is in an unenergized state (e.g. not connected to the mains or switched off). After the connection to ULN-2 has been made, you should energize the power source.

If you connect an energized power source to the ULN-2's 2.1mm power connector you may see a small spark when you make the connection. This is due to surge current and is normal if you connect a power source in this way. While this will not damage the ULN-2 in any way, to avoid the spark just connect the power connector to ULN-2 before connecting the power source to the wall.

ULN-2 Specifications

Mic Inputs	
Stepped Gain Range	+6 dB – +72 dB (6 dB Steps)
Gain Range with Trim	-14dB – +70 dB
Input Impedance	3.3k Ω
• Harmonic Distortion @1kHz • (+9dBu in @ 6 dB Gain)	0.0005%
• Intermodulation Distortion 1k component • (19 kHz/20kHz @ +8dBu)	-96 dBu

Table 7.1. Mic Inputs

Equivalent Input Noise (E.I.N.)	
150 Ω Source	-130.5 dBu
50 Ω Source	-132.0 dBu
0 Ω Source	-134.0 dBu

Table 7.2. ULN-2 Equivalent Input Noise (E.I.N.) 20 Hz – 20 kHz Flat @ 72 dB Gain

Frequency Response	
18 Hz – > 20 kHz	± 0.1 dB
8 Hz – > 50 kHz	-± 1.0 dB
3 Hz – >100 kHz	± 3.0 dB

Table 7.3. ULN-2 Frequency Response

Crosstalk @1kHz	
trim link engaged	-107 dB
trim link disengaged	-132 dB

Table 7.4. ULN-2 Crosstalk @1kHz

Maximums	
Max Gain	72 dB
Preamp Headroom	20 dB above Digital Clip
Phantom Power	+48v Regulated, high current, individually switch-able, P48 test compliant, short circuit/ hot-swap protected

Table 7.5. ULN-2 Maximums

Latency	
A/D	39 samples
D/A	28 samples

Table 7.6. ULN-2 Converter Latency

Input Processing	
M/S Decode	Instantiable
parametric EQ	Instantiable
Dynamics	Instantiable
Character	Instantiable
Reverb	Instantiable
Delay	Instantiable
Nearly Infinite Combinations via DSP Graph	Instantiable

Table 7.7. ULN-2 Input Processing

Output Processing	
M/S Encode	Instantiable
parametric EQ	Instantiable
Dynamics	Instantiable
Character	Instantiable
Reverb	Instantiable
Dither	Instantiable
Mix Foldown	Instantiable
Nearly Infinite Combinations via DSP Graph	Instantiable

Table 7.8. ULN-2 Output Processing

Front Panel	
Indicator LEDs	64
Meters	4
Segments per Meter	10
Controls	• Stepped Gain (12 steps 6dB – 72 dB) • Continuous Attenuator on return • Attenuation link • Mic/Line Select switch • Independent Monitor Volume Controls • Independent Phantom Power Controls • Preset Recall Buttons
Sample Rate Indicators	4
Clock Source Indicators	4
Digital I/O Source Indicators	2

Front Panel	
System Lock Indicator	1
Firewire Indicator	1
Power Indicator	1
Preset Recall Indicators	10
Headphone Output (Dedicated DAC)	TRS Stereo

Table 7.9. ULN-2 Front Panel

Back Panel	
Word Clock Connectors (In and Out)	75 Ω BNC
AES Connectors (2 Channels In and Out) • w/S/PDIF•AES Edge Card	XLR
SPDIF Connectors (2 Channels In and Out) • w/S/PDIF•AES Edge Card	RCA
Optical Connectors (8 or 2 Channels In and Out)	TOSLINK (automatic format sensing and SMUX support)
Mic/Line Input Connectors (2 Channels)	Neutrik™ Combo XLR/TRS
Insert Send Connectors (2 Channels)	TRS
Insert Return Connectors (2 Channels)	TRS
Analog Output Connectors (2 Channels, switchable +4/-10)	TRS
Monitor Output Connectors (2 Channels)	TRS
Gigabit Ethernet MHLink Connectors (2)	RJ-45 8-pin
USB-C	
Metric Halo EdgeCard™ expansion slot	
Power (Unswitched)	2.1mm Coaxial
Power (Unswitched)	4-Pin XLR
Security Slot	Kensington

Table 7.10. ULN-2 Back Panel

Software	
Driver	CoreAudio Mac OS X
Mac OS X	10.8.5 or newer
Architectures	Intel
MIOConsole3d	Included
Record Panel	Included
LTC Decoder	Included
Mixer	Included
DSP Processing	Included

Table 7.11. ULN-2 Software

Power	
Voltage	9v - 30v

Power	
Power	8 Watts
External Supply (Input)	100-240VAC
External Supply (Max Input Current)	1.2 A
External Supply (Output)	18 VDC
External Supply (Max Output Current)	2.0 A
External Supply Connector	2.1mm coaxial, 4-Pin XLR

Table 7.12. ULN-2 Power

Case	
Material	Powder Coated Aluminum
Fasteners	7/64" Hex Socket Head
Weight	4 lbs
Weight	1.8 kg
Dimensions	13.5" x 8.25" x 1.73"
Dimensions	34.3 cm x 21 cm x 4.4 cm
Rack Ears (included)	Powder Coated Steel

Table 7.13. ULN-2 Case

8. 2882 Users Guide

2882 Overview

Figure 8.1: Mobile I/O 2882

What it is

Mobile I/O is a portable, high–quality, modular multi-format audio converter, interface, and processor for professional audio applications. The 2882 model line is equipped with eight balanced analog inputs (4 on XLR and 4 on 1/4" TRS), eight balanced analog outputs (1/4" TRS), fully configurable ADAT® / TOSLINK optical I/O (with full auto-SMUX support), BNC wordclock in/out plus independent stereo AES/EBU and S/PDIF Digital I/O ports (provided on the base-configuration EdgeCard™).

Host computer connection and box daisy-chaining are provided by two MHLink Gigabit Ethernet ports, each of which support 128 channels of bidirectional audio at 192khz/32 bit data rate. A UAC2 class-compliant USB-C connection is available as an alternative computer interface.

Analog inputs are capable of 24-bit/ 96kHz operation, although the analog outputs, digital i/o and internal routing and processing paths support sample rates up to 192kHz.

What it has

- 20 simultaneous input channels and 20 simultaneous output channels (with the base S/PDIF•AES EdgeCard™)
- 8 independent channels of mic-pre with switchable phantom power
- A/D and D/A conversion at 44.1, 48, 88.2, 96kHz Sampling Rates (D/A conversion operates at 176.4 and 192kHz. A/D conversion is not supported at 4x sample rates.)
- 24 bit 110 dB Dynamic Range A/D converters
- 24 bit 120 dB Dynamic Range D/A converters
- 8 Balanced Analog Inputs – 4 XLR, 4 TRS 1/4"
- 8 Balanced Analog Outputs – 8 TRS 1/4"
- Digital I/O, DSP processing, routing and formats conversion at 44.1, 48, 88.2, 96, 176.4 and 192kHz Sampling Rates
- 8 channels of ADAT® Optical Input and Output (or 2 TOSLINK). Optical inputs auto-sense ADAT® vs. TOSLINK format. Optical output format is user-selectable. SMUX is automatically implemented to support high sample rates.
- MH EdgeBus programmable and pluggable audio expansion slot.
  2882 base configuration includes the S/PDIF•AES Stereo Digital I/O EdgeCard™ pre-installed

- Enhanced processing power and massive hardware memory. All 3d hardware includes over a hundred DSP plug-ins and the unique Metric Halo Graph environment within the 3d mixer.
- 128 input x 64 bus (at all sample rates) multi-box unified zero-latency mixer
- 1024 x 1024 internal audio routing matrix (per box)
- Front Panel Metering for Analog Inputs and Outputs
- Full console metering of every channel and mix bus
- Total recall of every console parameter
- Portable Capabilities – Battery Powerable
- Rack Mount Kit

What you need to use it

- Computer:
  - Any Mac that supports Mac OS X 10.8.5 with a Gigabit Ethernet or UAC2 class-compliant USB connection
  - Mac OS X 10.8.5 or newer required
  - Mac OS X 10.11.5 or newer recommended
- Peripheral Gigabit Ethernet Adaptors supported:
  - Apple Thunderbolt Gigabit Ethernet Adapter
  - PCIe Gigabit Ethernet Adapter
  - Third-party Thunderbolt Gigabit Ethernet Adapter (e.g. as part of a dock)
  - USB3 Gigabit Ethernet Adapter
- Software: All CoreAudio compliant software is compatible with Mobile I/O, including:
  - Cubase
  - Digital Performer
  - GarageBand
  - iTunes
  - Live
  - Logic
  - Nuendo
  - Pro Tools
  - Studio One
  - Reaper
  - MixBus / Ardour
  - and hundreds more…

What comes with it

Your Mobile I/O 2882 3d package contains the following items:

- One Mobile I/O 2882 3d unit:

Figure 8.2: Mobile I/O Unit

- One IEC Power Cord appropriate for your area
- base configuration includes one S/PDIF•AES Stereo Digital I/O EdgeCard (pre-installed) Note: alternate EdgeCard configurations are available at time of order.
- One 18-volt 48-watt world-ready external power supply
- One 12” CAT-5e Gigabit Ethernet Cable
- One 14' CAT-5e Gigabit Ethernet Cable
- Two Rack Ears w/ fasteners
- Warranty/Registration Card

If any of these items are missing from your package when you open it, please contact Metric Halo or your dealer immediately for assistance.

Using the 2882 Hardware

2882 Front Panel

Figure 8.3: 2882 Front Panel

The 2882 front panel provides ten-segment metering for the 8 analog inputs and outputs. The meters are fast VU meters with auto-resetting peak holds.

The front panel also provides 2882 system status at a glance:

- Sample Rate (nominal 44.1, 48, 88.2, 96, 176.2 or 192kHz)
  - The Sample Rate is determined from the currently selected clock source so it will accurately indicate the current sample rate, even when the clock source is being provided by an external device.
  - 176.4kHz sample rate is indicated when both 44.1 and 48 are lit.
    192kHz sample rate is indicated when both 88.2 and 96 are lit.

    Note: A/D conversion is not available at 4x sample rates. Interestingly, the D/A converters operate fine at 4x rates, even though they are 96kHz-specified chips.
- Clock source:
  - Internal - indicates that the system is internally clocked
  - Wordclock - indicates that system is being clocked from the wordclock input
  - Digital In - indicates that the system is being clocked from the selected digital input (Optical, AES or S/PDIF)
- Power — Indicates that the 2882 is receiving power.
- Phantom power — Indicates that at least one of the preamps has the phantom power enabled.
- FireWire — Indicates that the 2882 has been successfully connected to either a USB or MHLink Gigabit Ethernet connection to the Host computer.
- Locked — Indicates that the system clock recovery circuit is properly locked to the selected clock source. If this light is not illuminated, the 2882 will not be locked to a clock and will revert to its failsafe internal clock source. Even if the Locked light is not illuminated, the actual sample rate will still be indicated on the front panel display.
- Digital I/O Section:
  - The AES and S/PDIF lights indicate when a valid incoming digital audio stream is present. The Locked light indicates when the digital receiver is locked to the incoming digital audio signal.

The Mobile I/O front panel also provides access to the Headphone output and some associated controls. The headphone output jack is a TRS 1/4” jack that provides the Left Channel on the tip, the Right Channel on the ring and the ground return for the two channels on the sleeve. These signals are all ground referred, so they may also be split and fed single-ended (unbalanced) to an external audio device.

The 2882 headphone output is suited to a wide range of impedances. Headphones with lower impedance would be expected to get louder than those with higher impedance.

The Mute and Dim buttons provide instant access to simple level control for the headphone output. The Mute button provides a quick, tactile "panic switch" which mutes the front panel headphone outputs in case of accidental feedback loops and other audio unpleasantries. The Dim button attenuates the front panel headphone output by 18 dB.

2882 Rear Panel

Figure 8.4: 2882 Rear Panel (shown with standard S/PDIF•AES Edge Card installed)

The Mobile I/O rear panel features:

- 4 channels balanced XLR inputs. Each input has:
  - 24-bit 96kHz A/D converters (110dB SNR)
  - remote controllable pre-amps with 40 dB of gain
  - remote switchable input impedance characteristics
  - remote switchable 20dB pad
  - remote switchable 48v Phantom power, with 10mA current limit
- 4 channels balanced TRS inputs. Each input has:
  - 24-bit 96kHz A/D converters (110dB SNR)
  - remote controllable pre-amps with 40 dB of gain
  - remote switchable input impedance characteristics
  - remote switchable 20dB pad
  - remote switchable 48v Phantom power, with 10mA current limit
- 8 channels balanced TRS outputs. Each output has:
  - 24-bit 96kHz D/A converters (120dB SNR) (functional at 4x SR)
  - remote controllable output gain (from -12dBV up)
- TOSLINK connectors for ADAT Optical or Optical S/PDIF I/O
  - 8 channels of ADAT® Lightpipe input (auto-senses and switches between ADAT® and Optical S/PDIF)
  - 8 channels of ADAT® lightpipe output (user-selectable between ADAT® and Optical S/PDIF)
- Wordclock input/output on BNC connectors
- Stereo S/PDIF input/output on RCA connectors (with base configuration S/PDIF•AES Edge Card)
- Stereo AES/EBU input/output on XLR connectors (with base configuration S/PDIF•AES Edge Card)
- 2 MHLink Gigabit Ethernet ports
- 1 USB-C port
- 1 2.1mm DC power jack (9v - 30v, center positive, 15 Watts)

2882 Signal Flow

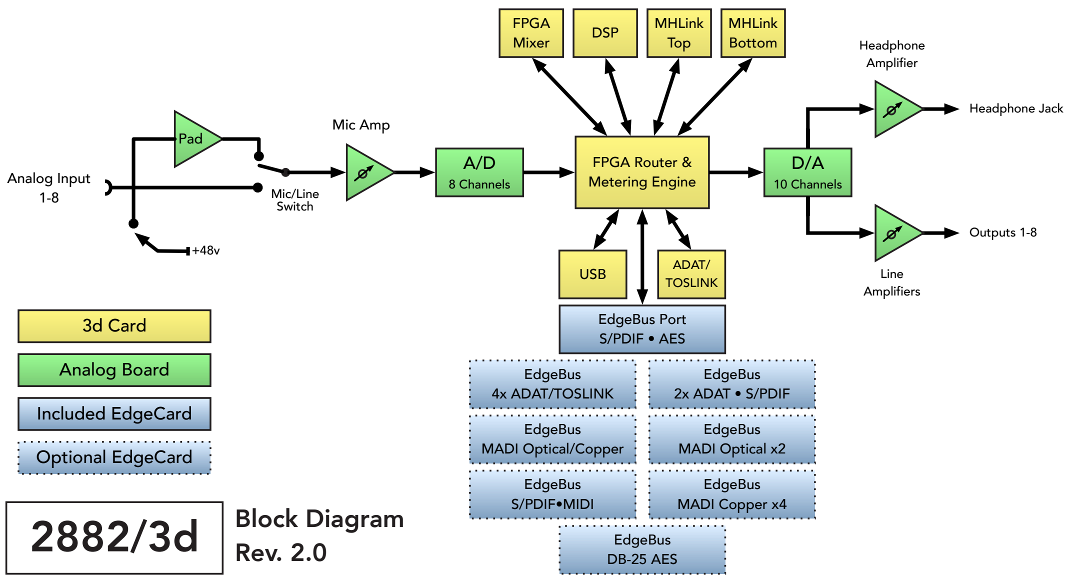


Figure 8.5: 2882 Signal Flow

Click here for a larger version

Making connections to the 2882

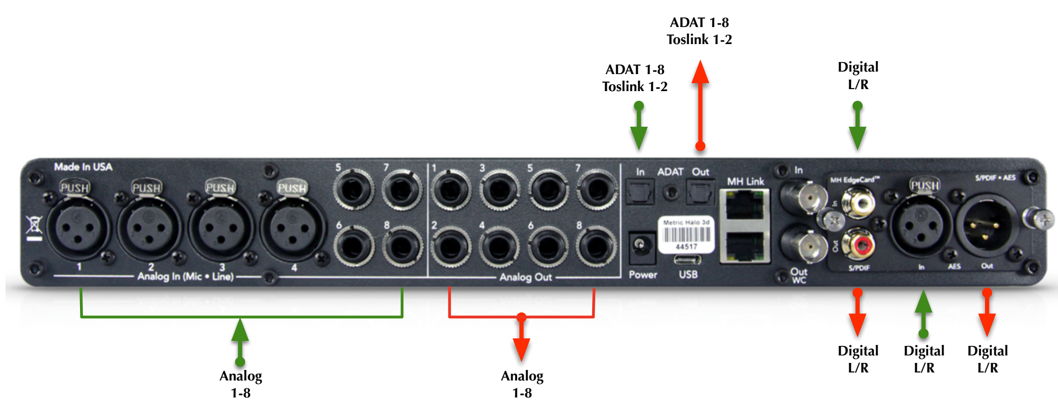


Figure 8.6: 2882 3d Routing (shown with standard S/PDIF•AES Edge Card installed)

Click here for more information on the available Edge Card configurations

There are seven classes of connections you can make to the 2882 hardware:

1. Gigabit Ethernet (MHLink)
2. USB
3. Analog Audio
4. Copper-based Digital Audio
5. Optical-based Digital Audio
6. Clock Sync
7. Power

Computer Connections

- **MHLink (Gigabit Ethernet)** - The preferred method of connecting your computer to the 2882 (or any Mobile I/O 3d audio device) is with a Gigabit Ethernet cable to one of the MHLink ports. This method enables the full capabilities of the 3d/MHLink environment with 128 channels at 32bit integer resolution input and output at all sample rates up to 192kHz. Daisy-chaining additional MHLink boxes dynamically and transparently integrates the new devices into the currently-running 3dConsole mixer, making all the added I/O ports and DSP engine power immediately available for use.

  MHLink requires the installation of the MHLinkDriver software. The latest MIOConsole3d installer package is available from: macOS MIOConsole3d installers (this will require a restart). Please see the section on software installation if you need clarification.

  > **Driver Installation on macOS 10.13 and newer**
  > With High Sierra (macOS 10.13) Apple introduced another layer of driver (system extension) software security.
  >
  > In order for third-party drivers (system extensions) to be activated on new installs of High Sierra, the user of the computer (you) must explicitly approve the activation, after the installation has occurred.
  >
  > You must do this before you will be able to use your hardware via MHLink.
  >
  > The authorization process is the same for all third-party drivers. If you are unfamiliar with the process, please read our FAQ answer here: macOS Driver User Authentication.

  Note: With MHLink, you need to launch MIOConsole3d to establish the routing between MHLink and the attached box(es). The routing is established automatically as soon as you launch MIOConsole3d.

- **USB** - The secondary method of connecting to your 2882 is with USB. USB connections for audio transport will be limited compared to MHLink. In a pinch, however, your 2882 can be seen as an audio device by any computer with a UAC2 class-compliant USB port.

  Note: Connecting to USB does not require the installation of a software driver. See MIOConsole3d Preferences > Discovery before connecting USB to a computer which is also connected to an MHLink ethernet interface.

Analog Audio Connections

The analog I/O connections on the Mobile I/O have been engineered for maximum flexibility in that they support both balanced and unbalanced connections with a wide range of input and output levels and a wide range of matching impedances. This means that Mobile I/O handles sources from mic level to line level and from mic impedance to guitar impedance. With that in mind, there are a number of aspects of the design that you should take into account when interfacing with Mobile I/O.

Whenever possible, use balanced interconnects with Mobile I/O. The performance of balanced interconnects is much higher and much more resistant to noise interference and electrical (power) wiring problems. The expense of balanced interconnects is not substantially higher than unbalanced connections, so if the gear that you are interfacing with supports balanced connection — use it. If you cannot utilize balanced interconnects, there are connection schemes that you can use that will maximize performance.

On input, at line level, it is sufficient to simply use standard unbalanced (TS) connections. If you are interfacing with the Mobile I/O XLR inputs, you will need to ensure that pin 3 is grounded in the unbalanced adapter cable. More information about adjusting the input level can be found in the MIOConsole3d software chapter.

The Mobile I/O XLR inputs are all wired pin 2 hot and the 1/4" inputs are wired Tip hot.

> **TIP:**
> To use the 2882 TRS input with guitar or bass, you can simply use a standard TS guitar cable (patch cord) and it will work fine. However, you can take advantage of the balanced input design of the 2882 to get more noise rejection than you thought possible on a guitar input.
>
> In order to do this, you will need to make a pseudo-balanced telescoping shield guitar cable. This can be constructed with a TRS connector, a TS connector and balanced microphone cable. This cable will treat the guitar as a floating balanced source and provide a telescoping shield from the 2882 ground.

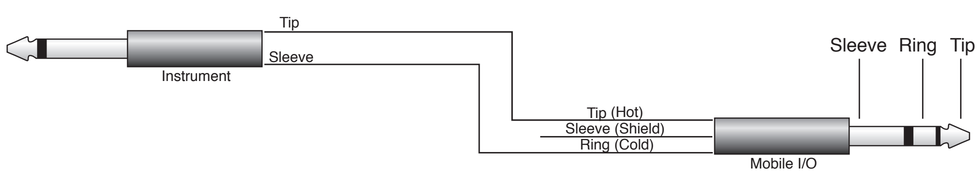


Figure 8.7: Telescoping Shield Cable for Instruments

If you want to use the TRS inputs with balanced microphones, you will need an XLR female to 1/4" TRS balanced plug adapter cable. These are available commercially, or you can construct one easily. The connections are Tip to Pin 2, Ring to Pin 3 and Sleeve to Pin 1:

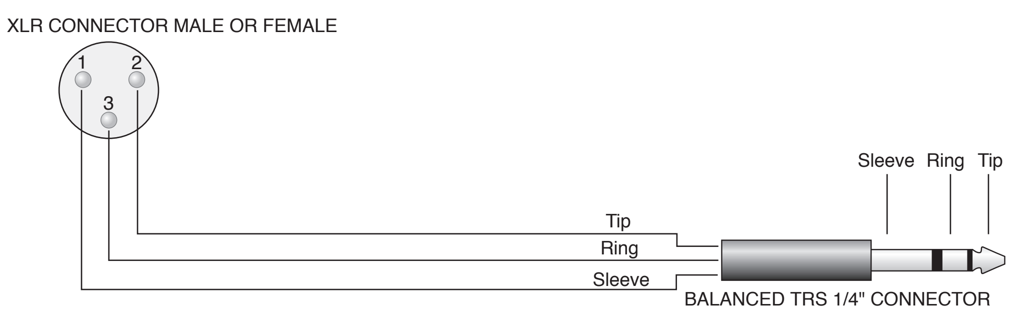


Figure 8.8: XLR to Balanced TRS Cable

On output, the situation is a bit more complex. If you are driving an unbalanced load, you will get the best performance by not connecting the ring of the TRS jack to ground. In order to do this, you can simply use a balanced TRS/TRS connector with the unbalanced gear. You can also construct a special cable with a TRS connector and a TS connector. In this cable, you just let the ring of the TRS connector float:

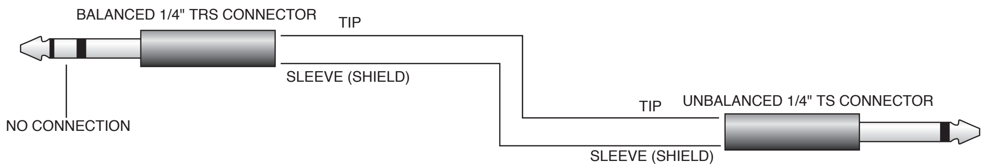


Figure 8.9: TRS to TS Unbalanced Cable

Alternatively, the TS connector can be replaced with an RCA connector for interfacing with gear that has RCA unbalanced interconnects.

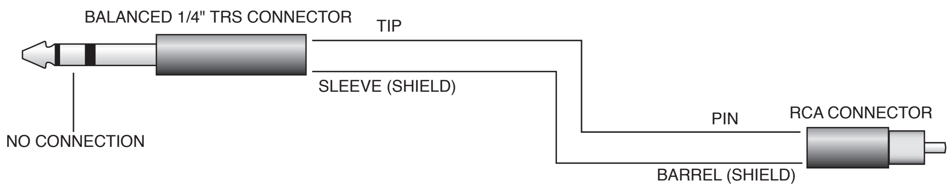


Figure 8.10: TRS to RCA Unbalanced Cable

Making a 1/4” connection

When you connect a 1/4” plug to a 2882 jack, insert it straight and firmly, ensuring that the plug is fully inserted into the jack. If the plug is not fully inserted you will get level shifts, phase flips, distortion, or no sound.

To disconnect a 1/4” plug, firmly pull the plug straight out from the connector body. The connectors on 2882 are stiff, so you may have to exert some force to remove the plug.

Making an XLR connection

When you connect a Male XLR plug to a 2882 jack, ensure that you have aligned the pins with the connector body and insert firmly until the retention tab clicks.

To disconnect the plug, press the metal retention tab flush against the box, and pull the plug from the 2882.

Output Attenuation

You may find that the 2882's analog outputs are too "hot" for your amps or powered speakers; if you find yourself:

- Turning your amps or speakers down further than normal (barely on)
- Bringing the master fader in MIOConsole3d or the Monitor Control level down more than a few dB

Then you are overdriving the inputs of your monitors. In this case, you should use a pad (also called an inline attenuator) on the outputs of the 2882. You should use a pad with 20-40dB of attenuation. These devices are available through music gear stores or online.

Copper-Based Digital Audio

The 2882 supports 4 channels of digital audio over copper-based connections using the S/PDIF interconnects with the RCA connectors and the AES interconnects using the XLR connectors. Even though only one of the AES or S/PDIF inputs can be a clock master source at any given time, you can have different digital sources connected to each of the input connectors at the same time – you use the MIOConsole3d application to select the active input. Audio routed to the digital outputs will be mirrored by both S/PDIF and AES outputs. This allows you to send the same stereo pair to two devices at once.

We recommend that you use the AES interconnect mechanism to establish the digital communication between the 2882 and other digital devices. The jitter and electrical noise tolerance on AES interconnects is substantially better than with S/PDIF interconnects. The AES interconnect standard is equivalent to balanced audio interconnections. If you need to use S/PDIF interconnects, try to use the shortest cables you can and, if possible, use special purpose 75 ohm S/PDIF or video cables.

The RCA connectors used for S/PDIF are friction fit coaxial connectors. When you connect them, ensure that they are fully inserted and tight.

The XLR connectors used for AES are fully locking. When connecting to them, make sure that you align the pins and insert firmly. When you remove the connector, make sure that you release the lock by pressing the lock release button before you pull the connector out of the 2882.

> **Wild digital input streams are still wild**
> Normally, when working with digital audio transport, you must take care to ensure that all devices communicating with one another are synchronized to the same audio clock. While you may find that the 3d digital audio inputs can often recover clean-sounding audio from unlocked audio devices running at the same sample rate, this is honestly just an accidental quirk of the over-built quality of the 3d hardware and *not a supported scenario*.
>
> In production or critical listening situations, please always follow digital audio clocking best practices and ensure that your outboard devices are properly clock-slaved to the Metric Halo box, or the domain is properly slaved and locked to your outboard gear.
>
> That said, if you find this quirk useful for a quick audition or casual listening, you are in good company.

Optical-Based Digital Audio

The 2882 provides two TOSLINK™ connectors on the back panel. One is a transmit connector and the other is a receive connector. These connectors are used with Plastic Optical Fiber (TOSLINK) cables to communicate with other devices. The TOSLINK connectors can be used to communicate with either the ADAT® Optical communication protocol or the Optical S/PDIF communication protocol. Optical inputs will auto-sense the incoming signal format. Optical output ports can be independently switched between the two protocols via the 3d Console.

The S/PDIF Optical communication protocol allows a device to transmit 2 channels of 24 bit audio at 44.1k through 192k, along with digital audio clock information.

The ADAT Optical standard allows a device to transmit 8 channels of 24-bit audio at up to 50kHz along with digital audio clock information.

After the original 8 channel ADAT specification was finalized, Alesis and other third parties extended the standard to support 4 channels of 24-bit audio at 2x sample rates and 2 channels at 4x sample rates. This extension of the standard is commonly referred to as SMUX. The 2882 automatically supports SMUX transport over the ADAT Optical connections when the box is in 2x mode (either 88.2k or 96k sampling frequency) or 4x mode (176.4k or 192k). While operating in SMUX mode, the audio data for each of the first four “ADAT” channels is placed on pairs of optical channels. ADAT 1 is placed on optical channels 1 and 2. Each optical channel continues to transmit data at 1x speeds, and both devices are required to multiplex and demultiplex the audio into and out of the optical channel pairs.

Since the 2882 provides direct routing within the box, you can easily configure the unit to work as an ADAT based 8 channel A/D/A. Refer to the chapter on MIOConsole3d for information about configuring the routing.

Clock Sync

Clock sync is a serious consideration in any digital audio system. The 2882 and any of it's digital audio interfaces can be used as the master clock source either for the 2882 by itself or for a full daisy-chained MHLink domain. In this regard, all 3d-equipped Metric Halo boxes are equal - only the physical I/O complement differentiates the box models.

If you are recording analog sources with 2882 by itself or as the root box of an MHLink domain, you can simply use the next-generation 3d internal clock source to drive the converters. This is the easiest case to deal with.

If you need to interface with other devices digitally or ensure sample accurate sync with video sources, the extensive clock synchronization capabilities of the 3d 2882 will prove to be more reliable (and better sounding) than much higher priced alternatives.

There are four different ways to get external clock information into the unit:

1. Sending a word clock signal into the WC Input BNC.
2. Sending an AES signal into the XLR Digital input.
3. Sending an S/PDIF signal into the RCA Digital input.
4. Sending an ADAT or S/PDIF signal into the Optical Digital input.

The BNC word clock input port is a 75 Ohm terminated coaxial input. It should be driven by a 75 Ohm source driver and interconnected with 75 Ohm coaxial cable. Even though the clock recovery qualities of the 3d hardware are second-to-none, incorrect cable impedance will introduce reflections on the word clock cable (jitter) which may adversely affect clock stability and audio quality.

The AES recommended procedure for distributing clock is to use an AES clock signal. The AES clock signal is an AES digital audio signal with no audio activity. 2882 only uses the AES preambles for clock recovery, so it is immune to data dependent jitter effects. This rule applies to both the XLR AES and RCA digital inputs, which means you can reliably use the Digital Inputs as a clock sources with or without audio data.

The Optical input port will automatically adjust to any ADAT or TOSLINK signal it recieves at whatever sample rate. For example, incoming ADAT data SMUXed at 96kHz will be decoded automatically as 4 channels of audio clocked at 96kHz requiring no adjustments from the user. That said, ADAT/TOSLINK optical connections are generally more susceptible to jitter artifacts than copper AES, S/PDIF or word clock sources.

Power

One of 2882’s great strengths is the flexibility of its power system. 2882 can be powered from any DC source in the range of 9V to 30V as long as it provides 12 Watts of power. The DC input on the 2882 is a 2.1mm coaxial power connector, center positive. So if you are powering the unit with a third party power source and it supplies 9V, the power source will have to provide 1.4 amps of current. If you are powering the unit with 12V, the power source will have to provide 1 amp of current, and so on.

The 2882 ships with a world-ready 18 volt, 2 amp power supply. You can plug this supply into any AC power source from 90V to 240V, 50Hz - 60Hz, using an appropriate IEC power cord, and it will supply the proper power to the 2882 on the 2.1mm coaxial power connector.

Unfortunately, the power capacity supplied through a USB bus is unsufficient to power a Mobile I/O.

As with all electronic devices, when connecting an external power source to the 2882, you should first connect the power source to 2882 while it is in an unenergized state (e.g. not connected to the mains or switched off). After the connection to 2882 has been made, you should energize the power source.

If you connect an energized power source to the 2882's 2.1mm power connector you may see a small spark when you make the connection. This is due to surge current and is normal if you connect a power source in this way. While this will not damage the 2882 in any way, to avoid the spark just connect the power connector to 2882 before connecting the power source to the wall.

2882 Specifications

Mic/Line Inputs	
Line +4 Gain Range	-2 dB – +40.5 dB
Line -10 Gain Range	-13.8 dB – +28.7 dB
Inst Gain Range	0 dB – +42.5 dB
Mic Gain Range	0 dB – +42.5 dB
Mic Pad Gain Range	-20 dB – +22.5 dB
Line Input Impedance	10k Ω
Instrument Input Impedance	200k Ω
Mic Input Impedance	200k Ω (12k Ω with phantom)
Mic Pad Input Impedance	10k Ω (6k Ω with phantom)

Table 8.1. Mic/Line Inputs

Maximums	
Max Gain	42.5 dB
Preamp Headroom	20 dB above Digital Clip
Phantom Power	+48v Regulated, high current, individually switchable, P48 test compliant, short circuit/ hot-swap protected
Output	+26 dBu

Table 8.2. 2882 Maximums

Latency	
A/D	39 samples
D/A	28 samples

Table 8.3. 2882 Converter Latency

Input Processing	
M/S Decode	Instantiable
parametric EQ	Instantiable
Dynamics	Instantiable
Character	Instantiable
Reverb	Instantiable
Delay	Instantiable
Nearly Infinite Combinations via DSP Graph	Instantiable

Table 8.4. 2882 Input Processing

Output Processing	
M/S Encode	Instantiable
parametric EQ	Instantiable
Dynamics	Instantiable
Character	Instantiable
Reverb	Instantiable
Dither	Instantiable
Mix Foldown	Instantiable
Nearly Infinite Combinations via DSP Graph	Instantiable

Table 8.5. 2882 Output Processing

Front Panel	
Indicator LEDs	177
Meters	16
Segments per Meter	10
Controls	• Mute • Dim
Sample Rate Indicators	4
Clock Source Indicators	4
Power Indicator	1
Phantom Power Indicator	1
Computer Connection Indicator (Firewire)	1
System Lock Indicator	1
Digital I/O Source Indicators	2
Digital Input Lock Indicator	1
Mute Indicator	1
Dim Indicator	1
Headphone Output (Dedicated DAC)	TRS Stereo

Table 8.6. 2882 Front Panel

Back Panel	
Word Clock Connectors (In and Out)	75 Ω BNC

Back Panel	
AES Connectors (2 Channels In and Out) • w/S/PDIF•AES Edge Card	XLR
SPDIF Connectors (2 Channels In and Out) • w/S/PDIF•AES Edge Card	RCA
Optical Connectors (8 or 2 Channels In and Out)	TOSLINK (automatic format sensing and SMUX support)
Mic/Line/Inst Input Connectors (8 Channels)	Ch. 1-4: Neutrik™ Combo XLR/TRS Ch. 5-8: TRS
Analog Output Connectors (8 Channels)	TRS
Gigabit Ethernet MHLink Connectors (2)	RJ-45 8-pin
USB-C	
Metric Halo EdgeCard™ expansion slot	
Power (Unswitched)	2.1mm Coaxial
Security Slot	Kensington

Table 8.7. 2882 Back Panel

Software	
Driver	CoreAudio Mac OS X
Mac OS X	10.8.5 or newer
Architectures	Intel
MIOConsole3d	Included
Record Panel	Included
LTC Decoder	Included
Mixer	Included
DSP Processing	Included

Table 8.8. 2882 Software

Power	
Voltage	9v - 30v
Power	8 Watts
External Supply (Input)	100-240VAC
External Supply (Max Input Current)	1.2 A
External Supply (Output)	18 VDC
External Supply (Max Output Current)	2.0 A
External Supply Connector	2.1mm coaxial

Table 8.9. 2882 Power

Case	
Material	Powder Coated Aluminum
Fasteners	7/64" Hex Socket Head
Weight	4.5 lbs
Weight	2 kg
Dimensions	13.5" x 11" x 1.73" (34.3 cm x 27.9 cm x 4.4 cm)
Rack Ears (included)	Powder Coated Steel

Table 8.10. 2882 Case

Part III. The MIOConsole3d Application

Table of Contents

9. Installation and Registration

Installing MIOConsole3d

Software Installation

The **MIOConsole3d** application manages all software, driver and firmware installation and updates.

To get started, launch the MIOConsole3d application.

Software Updates

When you launch MIOConsole3d, it automatically checks to see if any updates are available from Metric Halo. This update check is performed every time you launch the Console.

If an update to the Console application is found, the Software Version header in the very upper left of the MIOConsole3d main window will show the Console3d application version number as **orange text**.

Click on the orange text to download the update.

Note: Especially on the maiden installation, definitely pull the new Console build before proceeding. It's just the application package and should only take a few seconds - no restart will be required until you install the MHLinkDriver.

Once you have downloaded the newest MIOConsole version, place it in your Applications folder and launch as usual.

Driver Installation

The **MHLink Driver** is required for connecting via Gigabit Ethernet, where the full power of the 3d platform is unleashed. If no MHLink Driver is found on your computer, the "Software Version" header in the very upper left of the MIOConsole3d main window will show the current version number of the Console3d application, and the term " <Not Loaded> " in place of the MHLink Driver version number (as shown below).

Figure 9.1: MIOConsole3d - First launch status (prior to MHLink Driver installation)

Clicking on the orange MHLink Driver indicator will quit MIOConsole3d and launch the driver installation process. Your computer will need to be rebooted to complete the MHLink Driver installation. (See below for further details regarding handling of operating system security and audio driver installation.)

> Whenever you see this orange-colored text, the Console is telling you that a new version of the Console application, driver or 3d device firmware has become available.
>
> Click on an orange version indicator to initiate the update procedure.

Firmware Installation

Firmware updates will be indicated (again, as orange text) in the Unit Status display of each connected 3d box which requires the update.

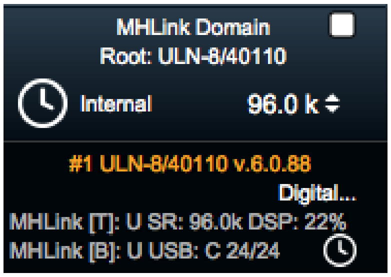


Figure 9.2: MIOConsole3d - a new firmware update is available!

The firmware update will be applied to all boxes in the selected domain to ensure parity. You will not need to quit MIOConsole3d, but all boxes must be fully power cycled for the firmware installation to be complete. The boxes will automatically re-appear in the System Status Pane when they come back online.

macOS Installation Concerns

Driver Installation on High Sierra (10.13)

With High Sierra (macOS 10.13) Apple has introduced another layer of device driver system extension security. In order for third-party drivers (system extensions) to be activated on new installs of High Sierra, the user (you) of the computer must explicitly approve the activation, after the installation has occurred.

This activation is only required for new installs of High Sierra. If you update to High Sierra, and you already have the 3d Drivers installed, the activation should not be required. After the driver has been activated, updates of the driver or installations of other new drivers provided by Metric Halo should not require activation. But if you do a fresh install of High Sierra, and then install the 3d Driver, you will need to *Allow* it to be activated before you can use the MIO with CoreAudio.

The activation event will occur the first time your box is connected after the driver has been installed. If activation is required, macOS will show an alert similar to the following:

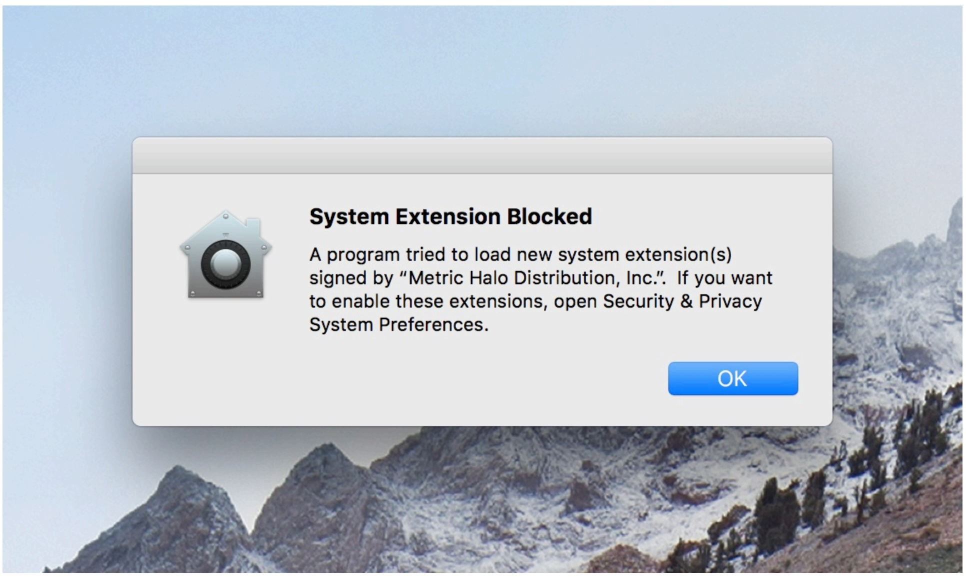


Figure 9.3: 3d Driver installation blocked by High Sierra

We strongly recommend activating the driver immediately, as the Apple-provided UI for activation has a timeout and will disappear after 30 minutes.

To activate the driver, follow these steps:

1. Open "System Preferences..." from the Apple Menu:

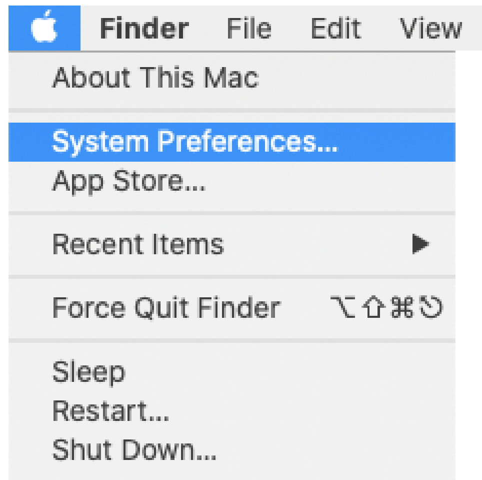


Figure 9.4: Apple Menu: "System Preferences..."

2. Select "Security & Privacy" (highlighted in red):

Figure 9.5: System Preferences: Security & Privacy

3. If the lock at the bottom of the window is locked, click it and enter your password when prompted. Then click the "Allow" button (highlighted in red):

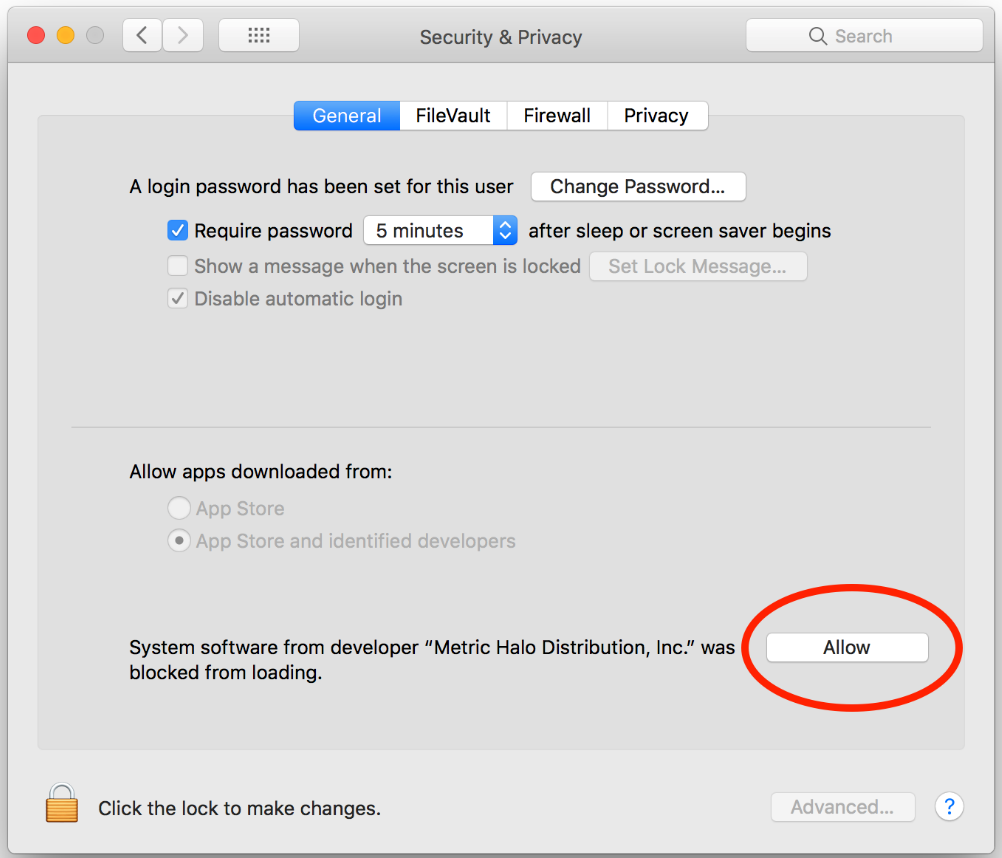


Figure 9.6: Security & Privacy: "Allow MHLink Driver to load"

4. That's it. At this point the driver should load and your MIO will appear in CoreAudio. This approval is persistent, so you should not have to do this again, unless you do a clean install of macOS.

Why can‘t I get audio input in my audio app on my new macOS 10.14 or 10.15 machine???

When you first start running audio to any audio program, Mojave puts up a dialog asking you if you want to let the App access the “Microphone”. This was a new security measure added in Mojave (macOS 10.14), is also present in Catalina (10.15) and Big Sur (macOS 11).

You **must** grant permission. This is a one-time thing.

You might think, “But I dont wan’t to use my computer’s microphone”. And you would be correct. But: Mojave treats all audio inputs on the system (including third-party interfaces) as “Microphones”. If you don’t *grant permission*, the app will be able to connect to your audio device, but CoreAudio will zero out all the samples before sending them to the app.

Oops...

You can fix this by going to System Preferences > Security + Privacy > Privacy > Microphone. Check the appropriate checkboxes to allow the appropriate Apps (specifically, in this case: MIOConsole3d) to access audio from audio devices. You may need to do this for other audio applications if you accidently denied access:

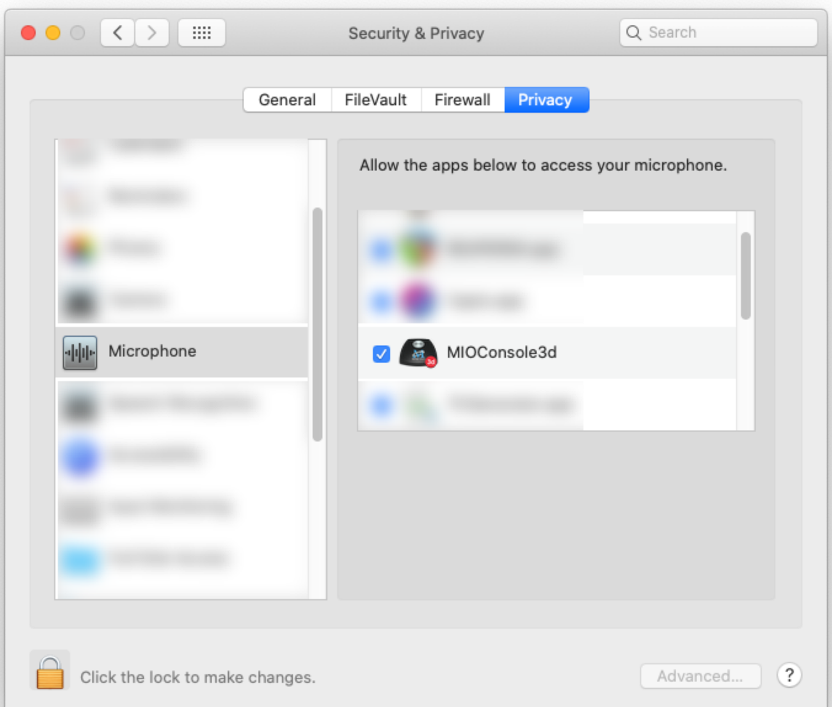


Figure 9.7: Security + Privacy / Privacy tab: “Allow MIOConsole3d access to Microphone”

Support for Apple Silicon

All Metric Halo MIOConsole3d and the MHLink Driver code has been ported, tested and fully supports Apple Silicon Macs. MIOConsole3d will automatically install the correct driver for your hardware. (See sidebar at the end of this section regarding EuCon support)

However, due to increased security features implemented in macOS 11 Big Sur, you need to ensure that your Secure Boot settings are configured to allow third-party drivers. Big Sur will warn you of this if your Secure Boot settings do not allow third-party drivers, but the instructions may be confusing.

To modify your Secure Boot settings, you must first reboot your Mac in Recovery Mode and open the Startup Security Utility from the menu bar 'Utilities' menu (same linked page, just scroll down a bit).

Apple's instructions for this are:

1. Reboot your Mac with Apple silicon into Recovery mode.
2. Set the security level to Reduced security.
3. Allow the loading of third-party kexts.
4. Reboot back to macOS.

After the Secure Boot settings have been changed, you can install the driver, and follow these steps to authorize it being loaded:

1. Open the Security & Privacy System Preferences.
2. Authenticate to make changes.
3. Allow the system to load your kext.
4. Wait for the system to load the kext and rebuild the auxiliary kext collection.
5. Reboot to load the new auxiliary kext collection.

> Please note: Recovery Mode is actually something every modern Mac user needs to be familiar with, so if you are not already aware, please take a moment to scan afore-linked page while you have it up… it's not long and the knowledge could save you a *lot* of hassle later on.

> The MIOConsole3d software is at full parity between Intel and Apple Silicon with the exception of one feature: EuCon support. EuCon support relies on Avid providing Apple Silicon EuCon support, which is not available yet. When running MIOConsole3d natively on Apple Silicon, EuCon will not be available. If you run it in Rosetta (using the Intel slice) then EuCon is available.
>
> Check here for EuCon and other Avid products compatibility with Big Sur and Apple silicon.

Choosing your connections

Each Metric Halo 3d device provides two ways for you to physically connect your computer to the audio I/Os.

- USB
- MHLink (Gigabit Ethernet)

Please note that both the USB and MHLink ports are configured and controlled via the MIOConsole3d application on your host computer. Care should be taken that only one instance of the MIOConsole3d application is visible to your 3d boxes at a time, regardless of how you are connected. This will guard against any conflicts that may occur due to two MIOConsole instances attempting to control the boxes at the same time.

The USB Class Audio connection

USB is everywhere... Pretty much any gadget smart enough to have an updatable operating system has the potential to host a USB audio connection. So for compatibility, USB connectivity is massive.

Compatibility is the raison d'être of the USB port on your 3d boxes.

That said, USB is not an optimal port for audio transport. Many USB chipsets are built with tiny I/O memory buffers - a signature of a system designed for small asynchronous information transmission, such as a keyboard, mouse or secondary storage media. Anyone with a DAW knows that tiny buffers are not the friend of stable audio transfer - they are built for speed, and speed alone. This ‘tiny buffer’ issue also evidently applies to USB 3.x and later silicon as well.

Additionally, USB as a data transport protocol has no concept of real-time clock (something essential for audio), limited support for time-sensitive sustained data throughput (such as audio streams), and is a low-priority peripheral port in the resource allocation hierarchy of modern motherboards and operating systems. Even among systems that are verified to have implemented the USB UAC2 driver, performance is wildly variable between OS dot-revisions, BIOSes and hardware.

This is why most products which use USB for audio transport write custom software drivers, even if they are only passing stereo audio. These software drivers rely on a combination of pre-loading audio and larger buffers, providing protection to the audio stream at the cost of of extra transfer latency, the inevitable need to code versions for each operating system, and requiring a driver update every time the operating systems make internal changes.

Metric Halo has incorporated a custom firmware implementation of the UAC2 class audio transport within the 3d hardware as an alternative to software drivers. Providing an optimized ‘tiny buffer’ transfer mode in the 3d USB hardware has the advantage not only of lower latency, but far greater stability and cross-platform compatibility than a software driver solution.

3d USB port Compatibility

macOS version 10.10.x and later, and iOS devices version 11 and later perform consistently well over USB connections. iOS devices require USB-C or Lightning Camera adapters that support iOS USB Host mode.

Microsoft Windows 10 (starting with release 1703) includes the UAC2 class audio driver as a WASAPI audio device.

UAC2 driver code for the Linux kernel has been available since around 2015, and versions of Linux which incorporate this driver should register 3d boxes as ALSA audio devices.

USB Host mode for Android has been available since Android 3.1, via USB OTG (On-The-Go) adapter cables.

> Over the course of testing we have found that some (but not all) of the Windows audio APIs do not properly play back to USB devices with more than 8 outputs. As a result, things like system sounds and ASIO4All worked fine with the default 12/12 I/O count but other applications (like Edge and Chrome) did not - and in fact would fail to play audio at all and then crash.
>
> This appears to be a bug in Windows (or these apps), but, in principal, simply changing the USB I/O count to 8 or less via MIOConsole3d should resolve this issue. When you change the I/O count the new count is stored persistently in the 3d hardware, so this only needs to occur once.
>
> Taking all of the above into consideration, YMMV is the keyword when incorporating USB into your audio workflow.

The maximum S/R available via the USB port varies with the maximum channel count set for the USB connection. The maximum channel count can be changed with MIOConsole3d and is stored persistently in the box.

- 4-12 channels works up to 4x rates (44.1k - 192k)
- 4-24 channels works up to 2x rates (44.1k - 96k)
- 4-48 channels works only for 1x sample rates (44.1k - 48k)

This has the side effect that a USB port channel setting of the root box (box #1, connected directly to your computer) will affect the available sample rates for the domain. If you find yourself unable to set the domain sample rate higher than your current setting, check the USB I/O channel count of the root box (see sidebar below).

If your target sample rate is 4x, make sure all USB ports are set to 12 I/O channels or less.

If your target sample rate is 2x, make sure all USB ports are set to 24 I/O channels or less.

The 3d device will appear in your operating system audio preferences as a standard USB class audio device and will default with direct routes set up between USB and the Analog I/O ports on the box, with the headphones and analog outputs monitoring channels 1 and 2 from the USB host device.

> **Configuring 3d USB ports**
> Configuring the USB port channel allocation on each 3d device is managed from within the MIOConsole3d application. You can configure the number of channels if you are connected either via USB or MHLink.
>
> When you make the change, the box USB connection must reset. It will go offline for a moment then automatically re-connect.
>
> For USB status and configuration details, refer to the USB Port Status and USB I/O Configuration sections of the manual.

The MHLink (Gigabit Ethernet) connection

MHLink supports up to 128ch I/O (128 channels in and 128 channels out) at up to 192kHz/32bits (integer) resolution. The driver defaults to 32 channels of I/O.

MIOConsole3d provides a menu bar pull-down (appropriately called "I/O") to control the number of channels provided by the driver. Currently you must have MIOConsole3d running in order for your I/O channel preference to be re-asserted if you disconnect and reconnect your hardware.

When using MHLink as the connection to the computer, you can chain multiple boxes using any Gigabit Ethernet-qualified (Cat 5e or better) connection cables up to 100 meters in length. The MHLink driver will automatically recognize the chained boxes and aggregate their resources to appear as a single audio device to the computer, and as a domain in MIOConsole3d.

MHLink / Gigabit Ethernet considerations

The MHLink Driver will discover 3d devices connected to the computer over Gigabit Ethernet. Currently the driver, console and firmware do not provide any support for arbitrating or sharing the 3d hardware between multiple computers. *Only run one copy of MIOConsole3d at a time if multiple computers can see the same 3d hardware via the network or a USB connection.*

MHLink cannot heal the audio clock across the packet buffering stages in an ethernet switch. This means there is no way to synchronize multiple 3d units directly connected to an IP network switch. Each box would appear as its own MHLink clock domain, and as separate devices to the Host operating system (as depicted in red in the graphic below).

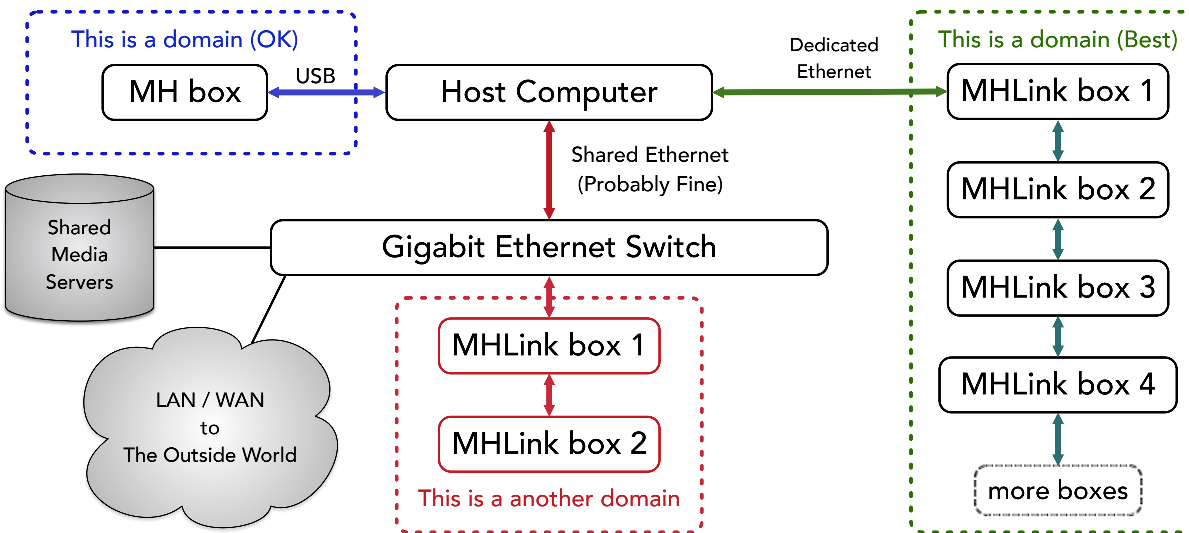


Figure 9.8: MIOConsole3d Domain configurations

To have an unified, aggregated system, 3d boxes must be daisy-chained via the MHLink ports, with one end of the chain connected from the 3d device directly to the computer (preferable) or the IP network. **Do not create Ethernet loops.**

Optimizing Host Computers for MHLink

Most of the usual things you can do to improve your computers' performance (SSD storage, adding RAM, keeping the OS up to date and well-matched to your computer) are of course recommended as a matter of course.

While your MHLink domain is designed to run at the max 128 channels at 192kHz with your default computer networking configuration, your computer is not a dedicated hardware audio environment. Most of the time your computer is thinking about something other than delivering audio samples in a timely fashion.

Luckily, there are a few things you can do to make your computers' workload a bit easier.

Consider the following optional for casual and light workload users, but best practice and highly recommended for professional installations.

Dedicate an internal, PCIe or Thunderbolt Gigabit Ethernet connection

The most significant stability/performance enhancement you can make is to dedicate an internal or PCIe/Thunderbolt Gigabit Ethernet NIC solely to MHLink audio transport.

A dedicated port will allow your computer to communicate with your MHLink domain to the full 128 channels in and out at all sample rates with the highest responsiveness and efficiency.

Keeping MHLink isolated from your IP network simplifies managing network performance and ensures maximum available bandwidth for network data transfers. This is especially the preferred configuration for server-based production houses, for all the obvious reasons.

Important Note! USB-C Gigabit Ethernet adapters marketed as "Thunderbolt 3 port compatible" are most likely *NOT* using Thunderbolt. Unless the adapter is physically labeled with the Thunderbolt logo (shown below), it is almost certainly some flavor of USB 3, and you will not experience nearly the stability or performance provided by a motherboard, PCIe or Thunderbolt NIC.

Figure 9.9: Thunderbolt icons

I/O Channel Count

The quickest and easiest way to reduce the load on your computer is to reduce the number of MHLink audio I/O channels. Both the computer and your DAW have to reserve memory and resources per the number of hardware I/O channels they have to keep track of. Fewer I/O channels means less work for the computer and therefore more cpu headroom.

To change MHLink I/O channel allocation, go to the MIOConsole3d menu bar I/O menu, and drop the channel count as low as you can manage without sacrificing functionality.

Keep in mind, most professional DAWs really hate it when you change the channel count while live, so it is always best to save and quit any running audio applications that are talking to MHLink before making changes to the I/O.

Disabling IPv4

For those able to dedicate a Gig-Ethernet port to audio, disabling Internet Protocol to the MHLink port is one of those tweaks that just makes sense.

Basically, since MHLink does not use internet protocol services, turning off IP services will stop the computer from peppering the MHLink ethernet port looking for IP-based network devices to talk to.

It's a preventative measure to be sure, and one less potentially complicating factor.

To disable IP for your MHLink port:

- In the network control panel, select the ethernet port that you are using for MHLink:
- select the Network connection that the MIO is attached to
- set IPv4 to "Off":

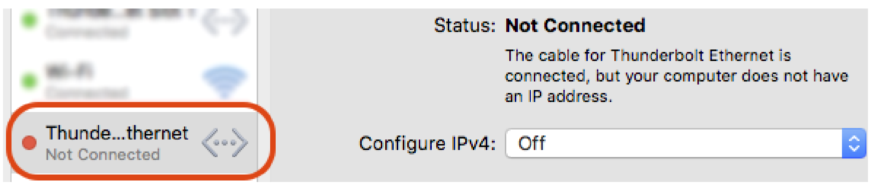


Figure 9.10: Network preference pane: IPv4 = "Off"

Note the "Status" entry in the above screenshot. The MHLink port will always say that same thing whether or not IPv4 is enabled. This is a good way to confirm that this port is dedicated to MHLink.

The 'cable being connected' part is telling you that you have a good physical connection and that there is legitimate Ethernet data being passed back and forth across that connection.

The "your computer does not have an IP address" phrase is telling you that said legitimate Ethernet connection is not using Internet Protocol. That right there pretty well confirms that this is your MHLink port, and that turning off IPv4 services will not affect your internet and file-sharing network connection.

Jumbo Gigabit Ethernet packets

When dedicating a Gigabit Ethernet port to MHLink, setting the MHLink ethernet port to use *Jumbo* packets will improve your host computers' CPU load significantly. This can become quite important any time you are pushing the CPU load in your DAW, and most especially when running 4x sample rates and/or high channel counts.

Rather than suggest that you blindly tinker with your computers' settings, here's a little background…

Ethernet transports information in packets of data called "Transfer Units". These packets can be of variable size, up to a maximum size which must be supported and agreed upon by all devices attached to the Ethernet network.

The standard maximum Gigabit Ethernet packet size (i.e. Maximum Transfer Unit or MTU) is 1500 bytes. Unfortunately, when streaming many channels of highly dense audio data, that 1500 byte packet size limits the number of sample frames we can place in a given MHLink packet. So when we have a lot of audio data to transport, this causes the packet rate between the computer and the box to be quite high. In your host computer, all that packet handling is managed by your main CPU.

The high packet rate does cause the low-level Ethernet NIC driver in the computer to do a lot of work, and it uses a lot of CPU as a result.

By utilizing Jumbo packets (e.g. 9000 byte packets rather than 1500 bytes), we can decrease the rate that packets are sent to and from the computer and substantially reduce the overhead that is generated by the low-level Ethernet driver. In short, the computer only needs to process one-sixth the number of packets to deliver the same amount of raw data within the same period of time.

The MHLinkDriver will automatically utilize packets up to the maximum specified for the underlying NIC driver. Since every device on an Ethernet network needs to agree on the MTU, this is only useful if you will dedicate an Ethernet port to MHLink, or if your Ethernet network is already running Jumbo packets throughout.

In order to use this feature, you need to manually configure the MTU for the Ethernet interface that you use for MHLink:

1. In the network control panel, for the ethernet port that you are using for MHLink, you need to increase the MTU:
   - select the Network connection that the MIO is attached to
   - click the "Advanced" button:

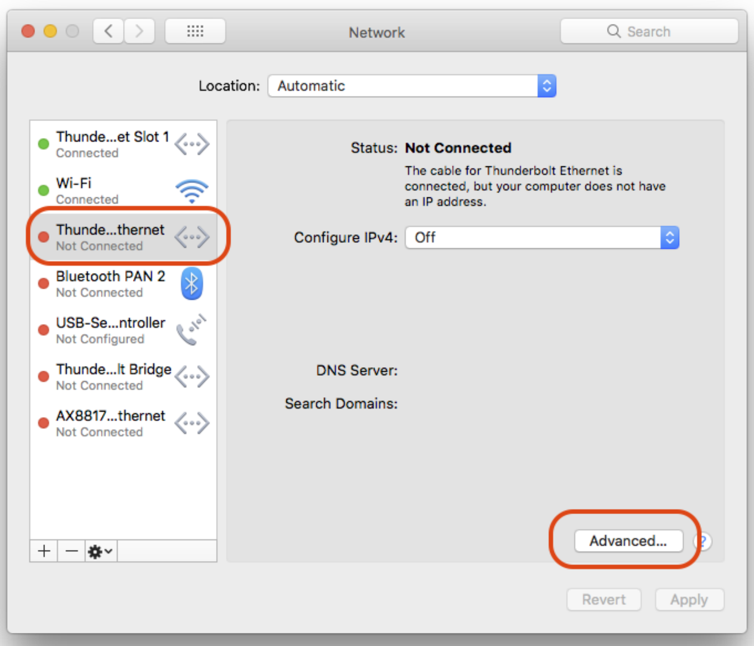


Figure 9.11: macOS Network preference pane

2. This shows the Advanced sheet:
   - First select the Hardware pane
   - Then set the controls as follows:
     - Configure: Manually
     - Speed: 1000BaseT
     - Duplex: full-duplex
     - MTU: Jumbo (9000)

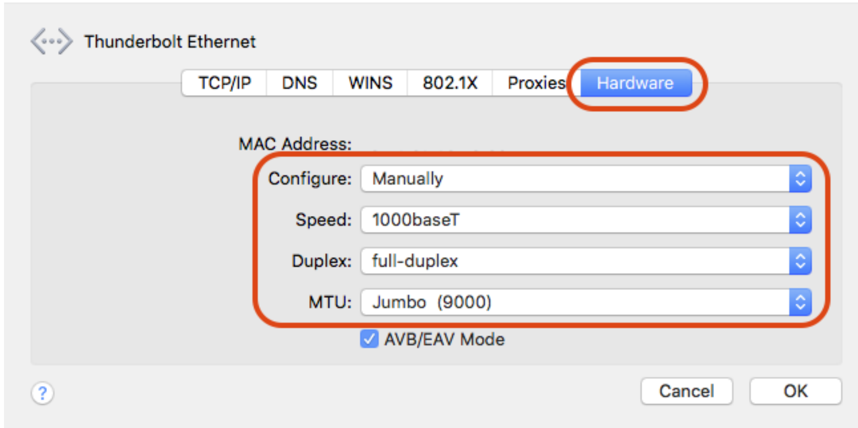


Figure 9.12: macOS Network / Hardware / Advanced preference pane

3. Click "OK"
4. Then in the main Network preference pane, hit "Apply"

Again, this is an optional optimization you can choose to do with a dedicated Ethernet connection for MHLink. It is not required.

Registration & Licensing

Registering your interface

> **Internet required!**
> Your computer must be online to register interfaces and manage licenses.

When you launch MIOConsole3d with an unregistered interface attached, you'll see the following dialog:

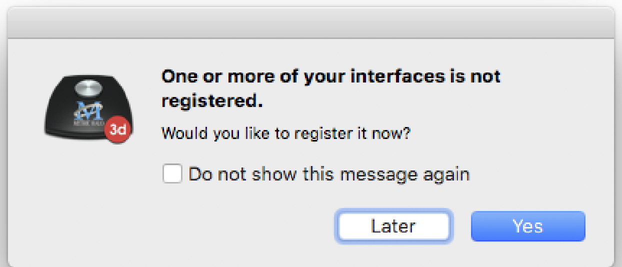


Figure 9.13: Registration Message

Your options are:

- *Yes:*Opens the registration window.
- *Later:*Closes the message. The message will appear again the next time you launch MIOConsole3d.
- *Do not show this message again:*This will set MIOConsole3d to no longer warn you about unregistered units after this message is closed. You can set MIOConsole3d to check for unregistered units again by going to MIOConsole3d's Preferences/Discovery panel and setting the "Check for unregistered units" option to “Always”.

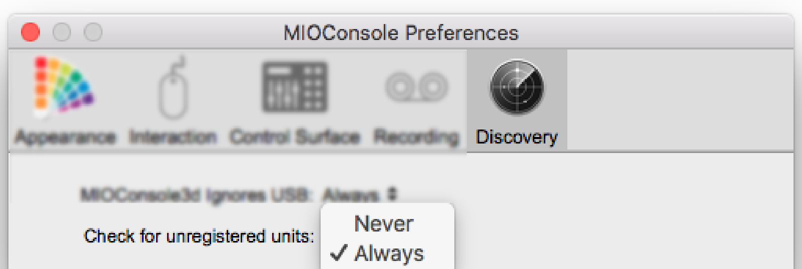


Figure 9.14: Preferences: Discovery: "Check for registered units" selector

Click OK, and the registration window will open:

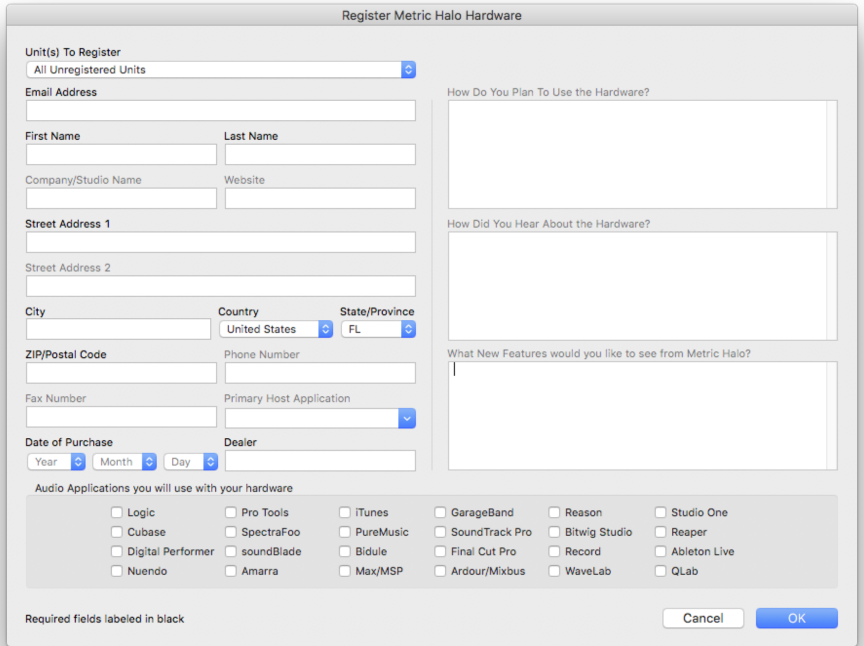


Figure 9.15: Registration Window

In the upper left corner is a menu to select which interface to register (if more than one unregistered unit is connected):

Figure 9.16: Registration Selection

You can choose to register all attached units or select a specific interface.

After selecting the unit(s) you wish to register, fill in the registration form; required fields are labeled in black, optional information is labeled in gray. When you're finished, hit OK. If there are any required fields empty, you'll receive a warning and the missing information will be labeled in red.

Once your information is correct and accepted, you will see this message:

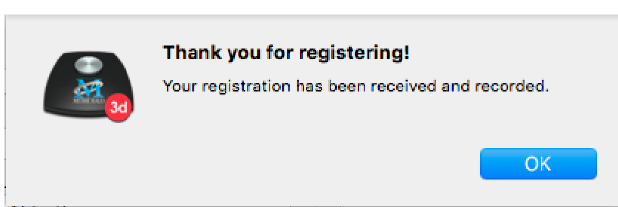


Figure 9.17: Registration Successful

You will receive an email from confirming your registration. This email will also include information on support, help resources and other useful links.

10. MIOConsole3d Operations

Overview

The MIOConsole3d application is your command and control center for every Metric Halo 3d device connected to your computer. When launched, the application scans, registers and synchronizes all available 3d devices and presents you with total control of the combined I/O and DSP processing power of every box in a clean, intuitive interface. 3d boxes connected to different physical ports on your computer will register as separate audio devices called Domains, which can be managed from the Domain Status Pane to the left of the main Console window (see example graphic on the next page).

With MIOConsole3d running, 3d boxes connected to the MHLink daisy-chain are automatically and transparently added to the existing 3d Domain. I/O and DSP resources of the newly-attached boxes register and become available in the background with no interference to audio already playing through the existing boxes. Removal of boxes from the Domain is a simple matter of manually unmounting them, and is equally transparent to audio running on the remaining active boxes.

The MIOConsole3d software allows you to easily verify and control clock source and sample rate, routing of all Analog and Digital I/O port and Host computer routes, control Mic Pre and Line Input and Output channel parameters, route, mix and and process your audio in any way you desire.

All routing and processing paths within the MIOConsole3d mixer environment are internally latency-compensated from Input strips through Aux and Group buses to the Main bus. MHLink system latency (e.g. from the Host computer through the last box in the daisy-chain) remains constant regardless of how many 3d boxes are added to the chain. All summing, plug-in and Graph processing within MIOConsole3d takes place on the MH 3d hardware and takes up none of your computers' cpu.

The I/O Routing interface includes a search engine and lets you narrow the selection view by box and port type, making it easy to target just the ports you want, even in large installations with hundreds of available computer, analog and digital I/O ports. Cascade and Group assignment features are included to simplify and speed up creation and modification of high channel-count systems.

All MIOConsole3d GUI windows are visually scalable and include mouse-over hints providing informational details for the majority of buttons and controls. The Appearance pane in MIOConsole3d preferences lets the user modify the UI font and color scheme and save/recall these modifications as "Themes".

The Mixer features complete show/hide control over all mixer strip elements. Three scrollable Mixer desk panes are available to help organize sessions. Comprehensive Solo modes include Solo-In-Place, PFL, AFL, PFL(Mono) and AFL(Mono). Auxiliary Mix buses are available to create Cue and Effects mixes which can route to the outside world and/or through to the Group submix and Main mix buses. All input and bus strip signal paths support channel widths from mono to 7.1. Extensive routing options and control management tools including fully-customizable Link Groups, Mute Groups and DCAs, are available on all Mixer Input and Bus strips.

MIOConsole3d employs a comprehensive preset and template management system at both a global level (one file with all I/O routing, gain structure, mix surface, monitor, cue and window layout parameters) and micro level, with saved presets recallable for all individual plug-ins and DSP Graphs. Like the routing selectors, templates and presets are organized by user-defined categories and can be searched by keyword as your library grows. This system can be used for pre-configuring your entire signal path offline so you will be ready to plug-in once at the gig, for managing separate location setups, or for recalling complex routing, gain and cue maps for quick changeover between sessions and sets.

An intuitive, interactive "smart-remapping" feature is provided for sharing routing maps, mixer setups and/or Monitor Controller templates between studios and venues with differing 3d hardware setups.

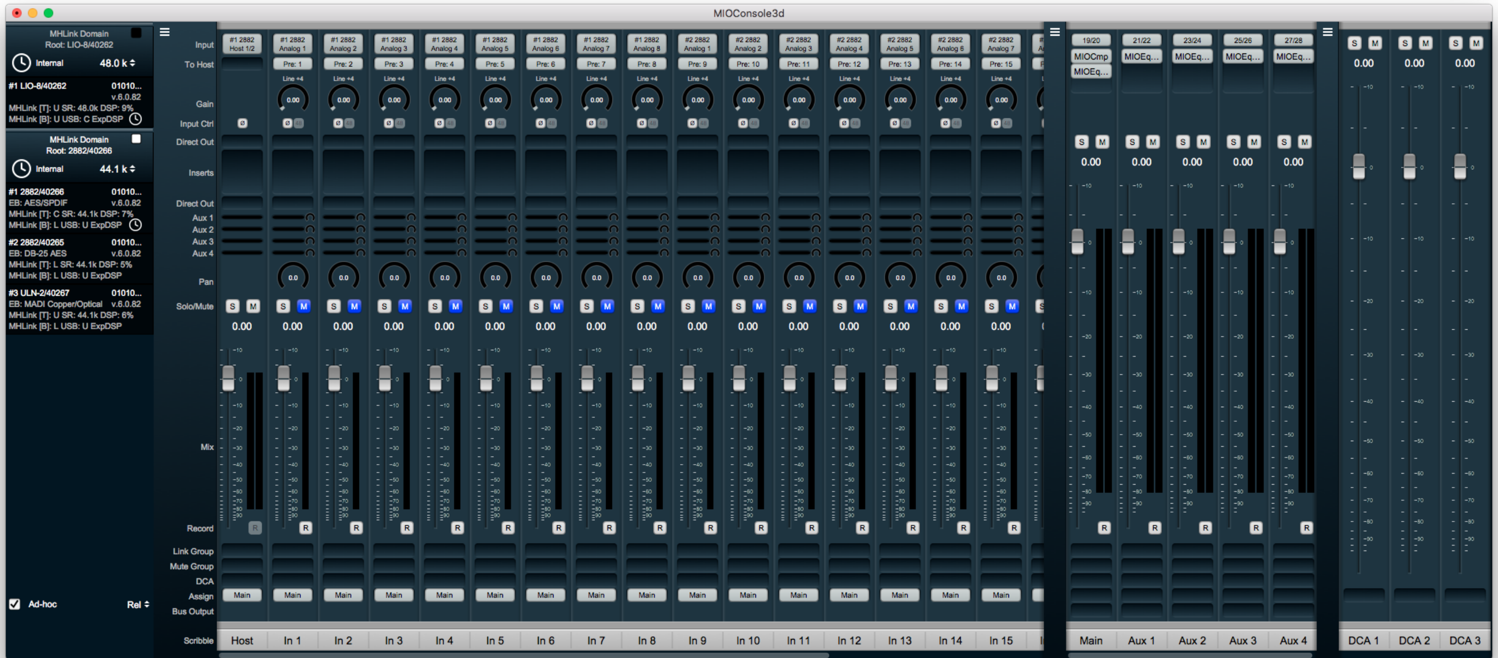


Figure 10.1: MIOConsole3d Mixer window

Powerful Monitoring and Cue Control sub-systems are provided as separate windows. The Monitor Controller supports multiple assignable sources and destinations in all channel formats from mono through Atmos 7.1.4. Cue Controllers are added as needed, with Talkback, Listenback and individual source and destination, mono fold-down, mute, dim, lock and volume controls per Cue. All Monitor and Cue outputs include a dedicated DSP processing section for every output channel, perfect for room acoustics correction, arrayed speaker delays, noise rejection filters, etc.

Figure 10.2: MIOConsole3d Monitor Controller and Cue Controls windows

The MIOConsole3d Menu Bar

The MIOConsole3d menu bar hosts many important functions you will use often when working in the 3d environment. All menu bar items can be assigned keystroke command controls from the **Edit > Edit Key Commands** window (default key command: ⌥K [<option>-K]). Default key commands are visible in the menu screenshots below, next to their respective commands.

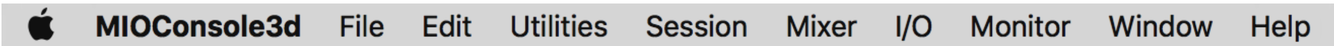


Figure 10.3: MIOConsole3d Menu Bar

MIOConsole3d Application Menu

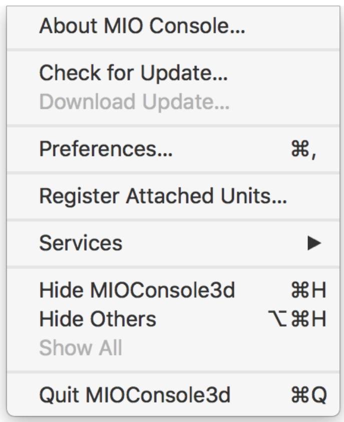


Figure 10.4: MIOConsole3d Menu

- Selecting **About MIOConsole3d...** will open an information box displaying the software distribution package designation and the currently loaded MHLinkDriver version. Please take note of the "Copy Version Info" button located at the bottom center of the window.

Figure 10.5: About MIOConsole3d

Clicking this button will load the Mac copy buffer with a status summary of the MIOConsole3d application, driver and each box listed in the currently running Console, ready for you to paste into session records and/or email correspondence with Metric Halo. This information is very helpful when requesting any kind of technical support.

Note: This system status information is automatically included when using the MIOConsole3d "Help" menu service forms.

- Although MIOConsole3d automatically checks the Metric Halo servers for new versions of the MIOConsole3d application when it launches, **Check For Update...** lets you initiate the process manually. When a new version is found, an information box is presented and the version number in the System Status Pane will turn orange.
- **Download Update...** becomes active when an update is available and will download the new build to your "Users/Downloads" folder.
- **Preferences...** (default key command: ⌘,) opens the MIOConsole3d Preferences window. Please see the MIOConsole3d Preferences section for the full overview.
- **Register Attached Units...** scans your system for connected 3d units and opens the hardware registration form. Ordinarily the MIOConsole3d Preferences are set to auto-launch the Registration form when it finds an unregistered unit, but if you routinely borrow, rent or otherwise move 3d units between systems, you may disable this feature in the "Discovery" preference pane.

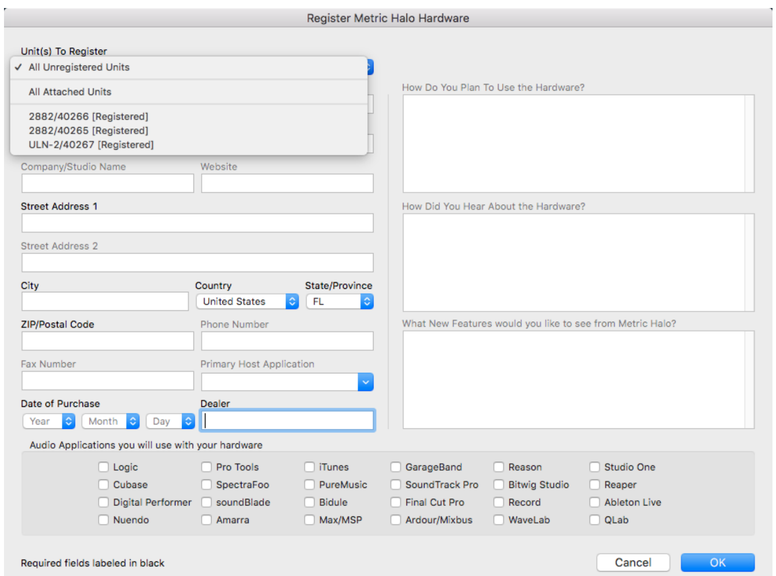


Figure 10.6: MIOConsole3d Registration Form

The registration form lets you register "All Unregistered Units", "All Attached Units", or to select a unit from a list of attached units. You can use the registration form to update your address or other registration information at any time.

- The standard macOS **Services** do not currently apply to MIOConsole3d, so the submenu only has an entry to the mac OS "Preferences > Services" pane.
- Standard macOS Finder commands to **Hide MIOConsole3d** (default key command: **⌘H**), **Hide Others** (default: **⌥⌘H**), **Show All** and **Quit MIOConsole3d** (default key command: **⌘Q**) fill out the rest of the MIOConsole3d menu bar menu.

File Menu

The “File” menu houses the standard macOS file system commands for: **Open…**, **Open Recent…**, **Close**, **Save** (default: **⌘S**) and **Save As…** (default: **⇧⌘S**) MIOConsole3d mixer files.

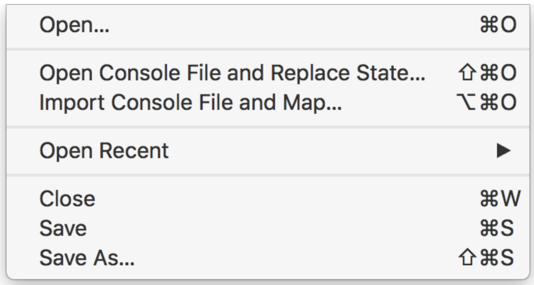


Figure 10.7: File Menu

Opening MIOConsole3d files…

As you know, the whole idea behind 3d and MHLink is the ability to freely combine multiple 3d boxes together to form a single insanely flexible and powerful audio interface. By the same token, you can freely *remove* 3d devices as well, making it easy to say, grab three boxes from your 5-box studio rig to go do a last-minute remote recording gig. The point is, you can treat MIOs like Legos™ to a certain extent, taking them apart and re-connecting them at will.

So, following the above scenario, let’s say you worked up a pretty good rough mix in the 3d Mix desk while at the aforementioned remote gig, and now you want to bring it all back and finish out the mix in your home studio. But, the studio rig is a different setup from the remote - different boxes, different connections to different gear, plus you have your studio monitors tuned with a graph in the LIO-8 serving as your studio Monitor Controller.

This is where **Box Mapping** comes in. In a nutshell, Box Mapping opens a saved MIOConsole3d file in a window which presents the entire Domain from the saved file - every 3d box with every I/O route and all Mix desk settings and Insert plug-ins - in relation to the boxes in your current domain. This allows you to translate or “map” the entire signal flow from the original session directly into your own studios’ routing configuration.

In this way, MIOConsole3d files can be easily shared between studios, making collaborative workflows a relative breeze.

> Note: In *all* cases, MIOConsole3d Box Mapping prioritizes keeping your current studio Monitor Controller Output settings exactly as they are. Importing ‘.cnsl3d’ files from a different studio using the “Open…” command will not contaminate your existing monitoring configuration. Importing foreign Monitor Controller output configurations (.mc3d files) can be accomplished with the menu bar > Monitor > “Load Domain Monitor Controller…” command.

With that in mind, let’s cut back to the menu bar “File” menu.

There are three commands available in the “File” menu to open a saved Console file:

- **“Open…”** (default: **⌘O**) is the standard “Open” command that is used when you double-click a saved ‘.cnsl3d’ file in the Finder, drag a file onto the application icon, or open a file from the “File > Open Recent” submenu. It may be used to open any of the file types that MIOConsole3d supports, including ‘.cnsl3d’ files (mixer/routing configuration), ‘.3dTheme’ files (Appearance Theme and font), and ‘.mc3d’ (Monitor Controller Output configuration) files.

"Open…" is a "smart" function in that it automatically analyzes the file that you have selected, compares it to the current MHLink domain and determines how best to open it. When you are working with a single 3d box or a stable multi-box rig, you'll just use "Open…" as usual, and it will bring up your mixer and monitor configurations immediately as usual.

If you have changed your MHLink domain by adding or removing 3d boxes since you saved the file, that tells the Console that some I/O routes are different between the file being loaded and the current system setup, and opens the Box Mapping window so you can decide how you would like to load the file.

- **"Open Console File and Replace State..."** (default: **⇧⌘O**)

  In special cases, it may be desired to simply load all contents of a saved '.cnsl3d' file exactly as it was saved, with no "re-mapping" from the saved Domain to the currently running system Domain.

  The **"Open Console File and Replace State..."** command completely replaces the current system state in MIOConsole3d with the contents of the file, asserts the updated configuration on any attached boxes, and loads the default state for any current boxes that were not in the file.

  All domains saved within the file will be loaded as 'offline domains' in the Console System Status Pane. Selecting each domain will reveal the entire signal path of every 3d box that domain.

> **Note:** This command is frankly overkill for normal production scenarios. However, some users have found it handy for examining all the stored information in a saved file verbatim (including Monitor Controller configurations and room tuning graphs), or to perform a full system reset to a specific configuration as part of a calibration or troubleshooting procedure.
>
> That said, it won't hurt anything to play with it. Even if it doesn't behave how you expected for whatever reason, you can always just open a known-working MIOConsole3d file and get right back to work.

- **"Import Console File and Map..."** (default: **⌥⌘O**) will display the mapping window, even if MIOConsole otherwise thinks that it can map or replace the data without user intervention. This is generally used for importing '.cnsl3d' files from a different multi-box MHLink system, or if you are opening an older '.cnsl3d' file and have made changes to the I/O routing of your system since you last opened it.

  Since "Import Console File and Map…" directly opens any MIOConsole3d file into the Box Mapping interface, it is especially handy for checking the contents of a saved file.

  As always, once you have examined the contents of an imported file, you can 'Cancel' the import without committing any changes.

Box Mapping: **How it works**

The Box Mapping interface walks you through the process of translating all box settings, routing assignments and mixer parameters from a saved console mixer file to your local 3d system. There are three stages to the process: first, selecting the Domain to import, then matching which source boxes will map to each box in your domain, and finally you can fine-tune individual routings as necessary.

This sounds like real work, but in practice the Console3d software makes importing even the most complex systems easy. To begin with, most MHLink systems consist of a single domain, which makes deciding which domain to map to kind of a no-brainer.

So the **"Open…"** command follows a set of rules to determine how it should handle the opening of each file. In the first two cases listed below, the file just immediately loads as usual with no extra steps.

1. If the domain(s) in the file and the system exactly match, then it will directly load all mixer routing and parameters from the file to the boxes.
2. If the both the file and the system contain only a single box, the mixer routing and parameters just map straight into your box, even if the box I/O is different (as with a ULN-2 to a ULN-8).
3. In all other cases, the system determines that the user needs to be involved, and will begin the Box Mapping process:

   Importing Domains>: When you open a file that contains a domain configuration that does not match the actual configuration of your system, the "Map Domain from File to System" window will appear:

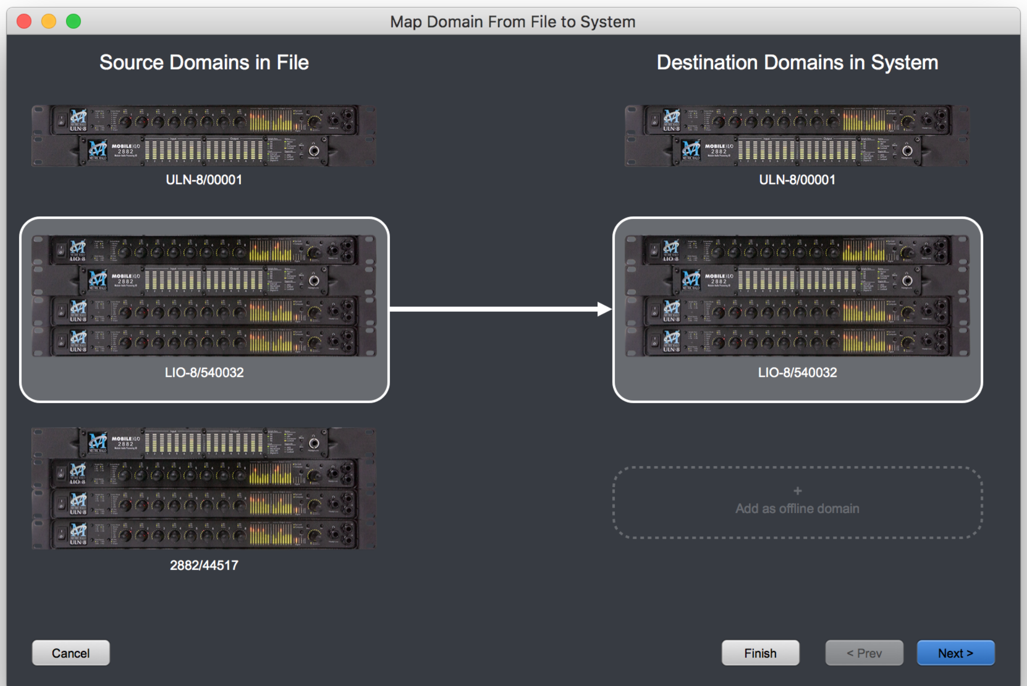


Figure 10.8: Map Domain from File to System

The left column lists the domains that are in the file. On the right are the domains that are in your running system. Click on one domain in the left column and one in the right. The domain on the left will provide the source data (mixer, gain controls, routing, MC, cues) that will be mapped onto the hardware selected on the right.

If the selected source domain has a root unit that does not appear as a root unit in one of your existing domains, the "Add as offline domain" target will be enabled, and you can select that as a destination. This provides a mechanism to examine the contents of the imported domain in the Console without over-writing your current mixer configuration. (It is, of course, not possible for the current root box to exist as both "online" and "offline" at the same time, so "Add as offline domain" will not be available as a target when the Source domain includes your current root box as its root (as in the example above)).

Once your selections have been made, you can proceed to the next step by clicking the "Next" button.

Generally, there will be one domain on the left and one on the right, making this step pretty straightforward. Since 3d does allow for advanced systems with multiple domains, this is how they are handled.

> Note: at any point in the process you may accept the defaults that will be applied in the subsequent mapping steps and commit the mapping by clicking the "Finish" button.
>
> If you decide you need to back up and change a setting in a previous step, click the "Prev" button to go back.
>
> If you want to stop the mapping process without making any changes to your existing configuration, click the "Cancel" button or the window's 'Close' button.

After you click the "Next" button, you will see the Box Mapping pane:

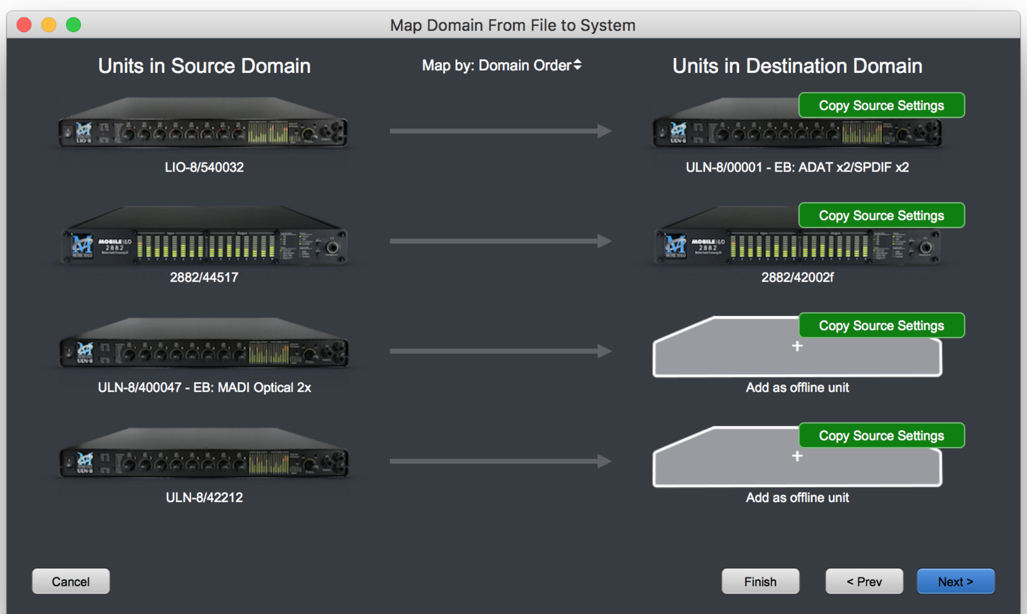


Figure 10.9: Map Domain: Source Boxes To System Boxes

The **Box Mapping** pane allows you to choose how the saved boxes in the file are mapped to the actual boxes you have in your system. You can click and drag the boxes in the right hand column (drag them up and down) to set the mapping from a given source box to a destination box.

The analog (including level trims) and digital I/O configurations (including SCP USB) will be applied from the source to the destination. Any channels that are routed in the mixer that connect to the physical I/O of the source box will be mapped to the closest corresponding physical I/O on the destination box.

While you move the boxes in the destination domain around for mapping purposes, the order of the boxes in the domain in your system (after the mapping is complete) will still be determined by the physical connection order.

By default, the boxes will be ordered by the order they appear in the domain. By dragging the boxes around you can set the mapping to whatever custom order meets your needs. That being said, there are a few automatic mappings you might want to use. You can select from them via the "Map By" popup menu:

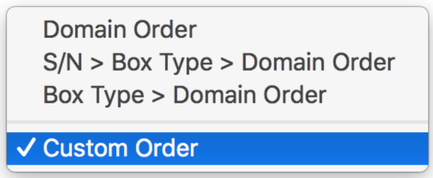


Figure 10.10: Map Boxes: Box Mapping order options

- **Domain Order** matches by connection order in the respective domains. It is the default, and is appropriate for mapping from one identically configured system to another (but with boxes with different serial numbers).
- **S/N > Box Type > Domain Order** matches first by Serial Number, then by Box Type and then any remaining unmatched boxes are matched by domain order. This is appropriate if you are mapping from one domain to another where the serial numbers are the same or mostly the same.
- **Box Type > Domain Order** matches first by Box Type and then any remaining unmatched boxes are matched by domain order. This is appropriate if you are mapping from one mixed domain to another where you are trying to get the best match of box models (ULN-2 to ULN-2, 2882 to 2882, etc.).
- **Custom Order** indicates that you have manually dragged the boxes to a custom order.

Mapping Mode: The green button attached to each box in the destination domain is the 'Mapping Mode' button. The default mapping mode is "Copy Source Settings", but there may be cases where you wish to keep the physical I/O routing on a particular box unaltered (such as mic input settings, loops to outboard processing gear, etc.). Click the button on a box to choose how you would like to map routes to each box:

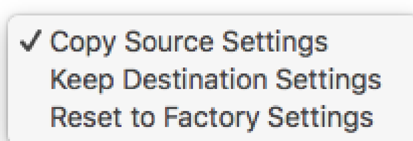


Figure 10.11: Map Boxes: Destination Box mapping mode

You will be able to review each box mapping before committing, so feel free to try things out. You can always hit 'Prev' to back up and make changes.

- **Copy Source Settings** copies the data from the source box as well as it can be matched (this will be exact if the source and destination box are of the same type).
- **Keep Destination Settings** leaves the configuration data on this unit alone (e.g. it remains with your current system settings). This does not have an impact on things like mixer routings, but will affect things like input and output gains.

- **Reset to Factory Settings** resets the configuration data on this unit back to the initial default 'power on' state. This doesn't have an impact on things like mixer routings, but will affect things like input and output gains.
- **Delete Offline Unit** appears as an option on the "Add as offline unit" boxes; this will simply drop that box from the mapped domain (as if you selected "Delete Offline unit" on the box after the mapping is completed). This will remove any mapped routings that refer to this unit from the mixer.

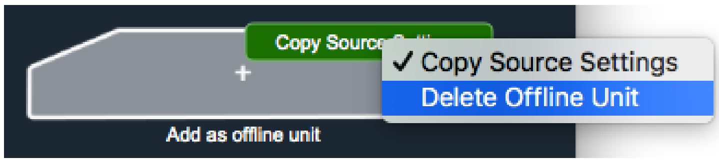
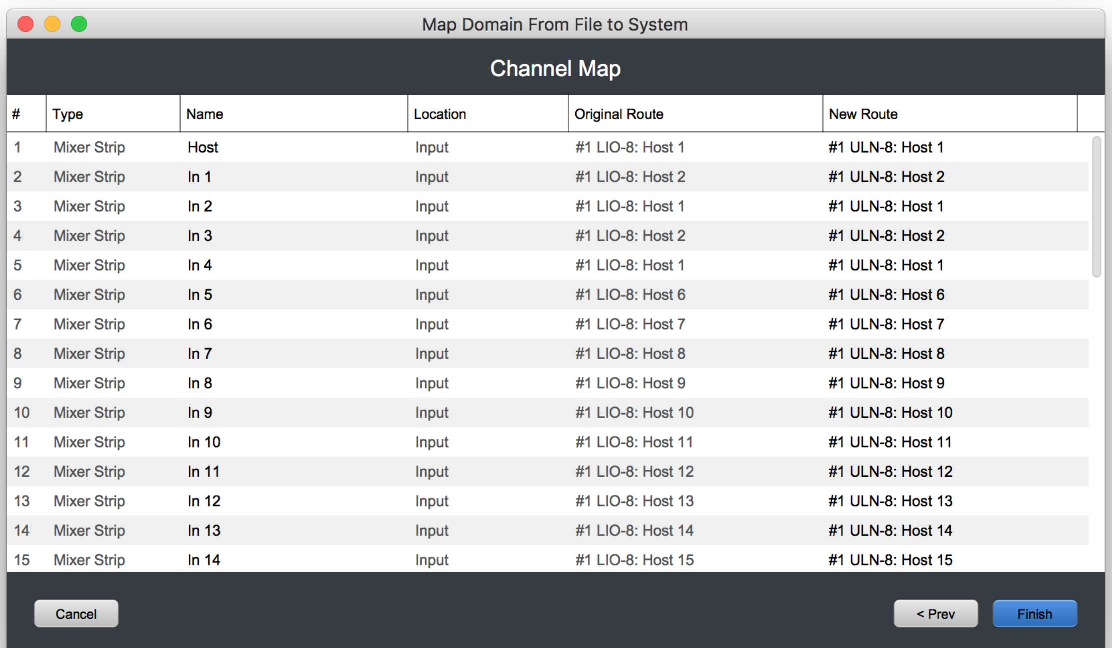


Figure 10.12: Map Boxes: Offline Unit options

Once you have made your mapping selections you can either click "Finish" to finalize the mapping with the I/O channels automatically set based on the box mapping you have established, or you can click "Next" to go to the **Channel Mapping Pane**:



#	Type	Name	Location	Original Route	New Route
1	Mixer Strip	Host	Input	#1 LIO-8: Host 1	#1 ULN-8: Host 1
2	Mixer Strip	In 1	Input	#1 LIO-8: Host 2	#1 ULN-8: Host 2
3	Mixer Strip	In 2	Input	#1 LIO-8: Host 1	#1 ULN-8: Host 1
4	Mixer Strip	In 3	Input	#1 LIO-8: Host 2	#1 ULN-8: Host 2
5	Mixer Strip	In 4	Input	#1 LIO-8: Host 1	#1 ULN-8: Host 1
6	Mixer Strip	In 5	Input	#1 LIO-8: Host 6	#1 ULN-8: Host 6
7	Mixer Strip	In 6	Input	#1 LIO-8: Host 7	#1 ULN-8: Host 7
8	Mixer Strip	In 7	Input	#1 LIO-8: Host 8	#1 ULN-8: Host 8
9	Mixer Strip	In 8	Input	#1 LIO-8: Host 9	#1 ULN-8: Host 9
10	Mixer Strip	In 9	Input	#1 LIO-8: Host 10	#1 ULN-8: Host 10
11	Mixer Strip	In 10	Input	#1 LIO-8: Host 11	#1 ULN-8: Host 11
12	Mixer Strip	In 11	Input	#1 LIO-8: Host 12	#1 ULN-8: Host 12
13	Mixer Strip	In 12	Input	#1 LIO-8: Host 13	#1 ULN-8: Host 13
14	Mixer Strip	In 13	Input	#1 LIO-8: Host 14	#1 ULN-8: Host 14
15	Mixer Strip	In 14	Input	#1 LIO-8: Host 15	#1 ULN-8: Host 15

Figure 10.13: Map Channels: Channel Mapping Pane

The Channel Mapping Pane lays out all Mixer Strip routings present in the Console file you are importing. Here, you can review the current channel routing map and make individual adjustments, or click the "Prev" button to return to the box mapping page.

The table in this pane includes the following columns:

- **Index** lists the number of the strip in the Mixer desk layout starting at the left (#1) and moving to the right.

- **Type** Indicates the channel map routing object type:
  - **Mixer Strip:** A channel route point on a mixer input strip
  - **Bus Strip:** A channel route point on a mixer bus master strip
  - **Talkback:** A channel route point in the talkback controller
  - **MC:** A channel route point in the monitor controller or cue controller
- **Name** lists the name of the routing object (e.g. the mixer strip or bus name). This is editable, and can be used to change the name when the mapping is applied.
- **Location** is the name of the route point in the routing object (e.g. input, output, etc.).
- **Original Route** shows where the signal is sourced from (if it's an input) or sent to (for an output) in the source domain configuration. This field specifies whether the route is 'pre-' or 'post-process' direct out, an I/O insert route, Monitor Controller, Cue, etc.
- **New Route** defines where the signal will be sourced from or sent to in the destination domain configuration. This routing is established by the box mapping you configured in the previous 'box-to-box' routing page. You can click on these entries to re-assign channels individually, or click the "Prev" button to return to the box mapping page and make box-level adjustments there.

Once you are done you can click "Finish" to finalize the mapping; the settings will be applied to your system state and the mapping window will close. The new console state will be named "(original file name) mapped". You can now save the mapped configuration if you so desire, or "Save As…" to rename the Console file before saving.

Edit Menu

Edit is also a standard macOS menu with the core text and item editing commands:

- **Undo** (⌘Z),
- **Redo** (⇧⌘Z),
- **Cut** (⌘X),
- **Copy** (⌘C),
- **Paste** (⌘V),
- **Clear** (no default key command),
- **Select All** (⌘A) and
- **Deselect All** (⌥⌘A).

Figure 10.14: Edit Menu

As in most applications, these core OS commands are contextual and will be applied according to the current item selection. In the Mixer window, they apply to all text and numeric data entry fields. In the Session editing space, they also function as expected on timeline and segment selections.

"Edit Key Commands" provides an interface to re-assign any existing default key command, including the core macOS commands listed above. The "Edit Key Commands" details section follows on the next page.

Key Commands

With **Edit Key Commands** (⌥K), you can define custom command keystrokes for every menu item available within the MIOConsole3d interface, plus special commands specifically for navigating and controlling 3d Mixer functions from control surfaces.

There are Key Command categories for each menu bar pull-down: Application (MIOConsole3d in the menu bar), File, Edit, Utilities, Session, Mixer, I/O, Monitor, Window and Help. Each category provides a list of each command from each menu, with controls that let you modify or assign keystrokes to each command.

There are additional Key Command categories for Monitor Controller, Parameter Controls, and Talkback Controller.

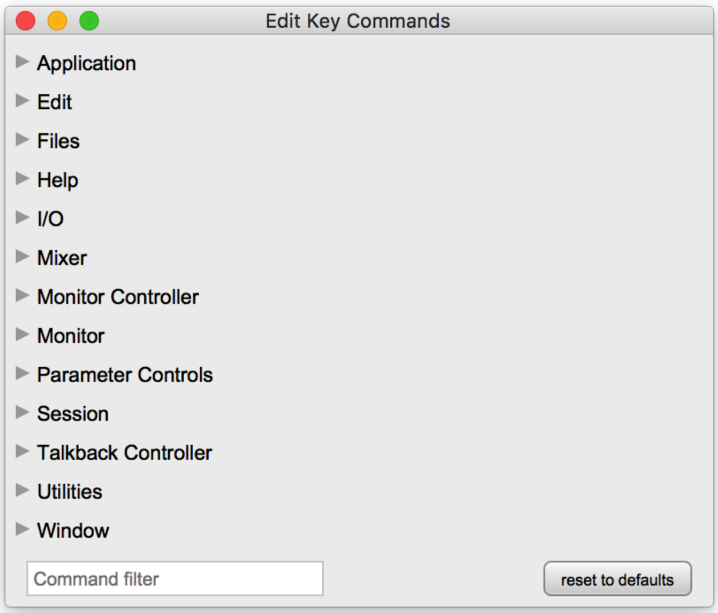


Figure 10.15: Key Commands categories

The screenshot above shows the Edit Key Commands window with all categories collapsed. You can option-click any closed menu expansion arrow to expand all categories, and option-click an open arrow to collapse them.

We will overview the Edit Key Commands interface here so you know how it all works, and reference the default command keystrokes as we progress into MIOConsole3d Mixer and Session operations.

Some key commands are specifically geared towards controlling Mixer and Monitor functions from external control surfaces, and do not have equivalent control buttons in the MIOConsole3d UI.

The very first thing to note in the Edit Key Commands window is the **"Command Filter"** text entry field at the bottom left. This is a dynamic search engine specifically for key commands. Type the command name or type of command you are looking for and the list will be instantly filtered accordingly.

For example, if you type in 'play', the key command list will show key commands related to Session 'play', 'playback' and 'playhead' functions.

At the bottom right-hand corner is the **"reset to defaults"** button, should you wish to reset all key commands to the original factory default settings.

Below is the default view when you open Edit Key Commands…

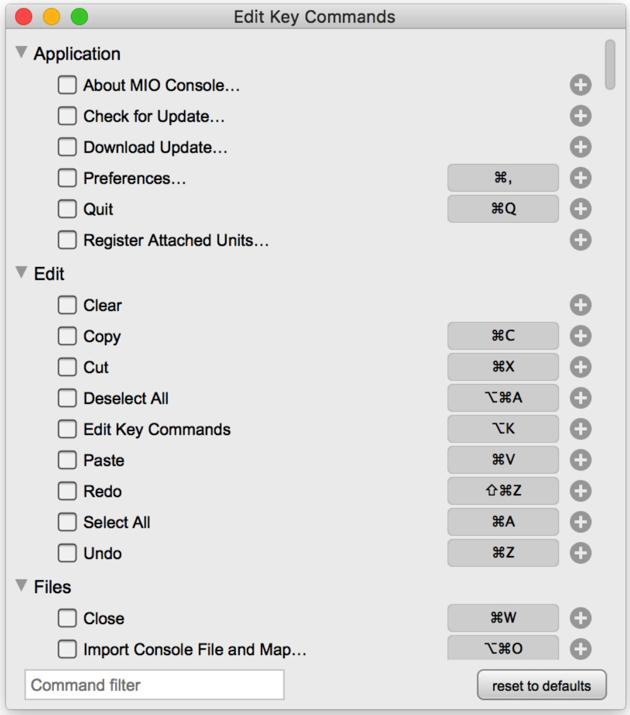


Figure 10.16: Default view: Edit Key Commands default window

…and here is the Key Commands controls layout.

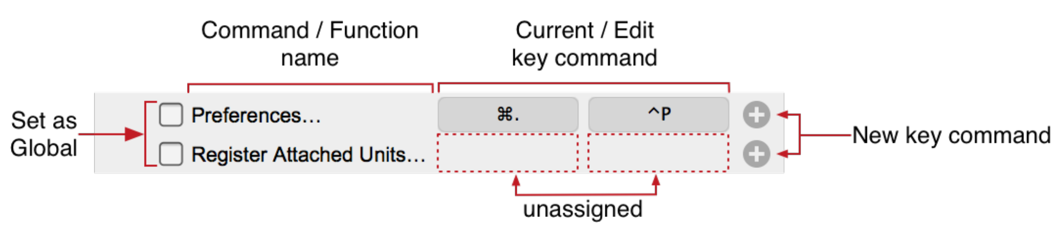


Figure 10.17: Edit Key Command UI

At the left are the Global Key Commands selection boxes, detailed later in this chapter. Global key commands will trigger MIOConsole3d functions even when the MIOConsole3d application is not in the foreground.

New key commands may be entered by clicking the plus sign “add” icon to the far right.

Each MIOConsole function listed in the command key library may have up to three keystrokes assigned.

Clicking the plus sign “add” icon of a function that already has a key command will let you add a second or third key command for that function. After adding a third keystroke, the “add” icon disappears.

To modify an existing keystroke, select the current command key button, and enter your new key command for that function.

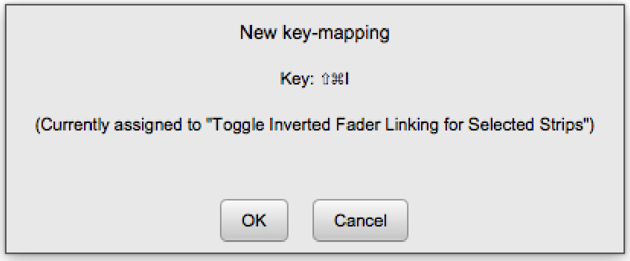


Figure 10.18: Edit Key Commands Conflict box

A dialog box will pop up to indicate any key map conflicts. This dialog box reveals the conflicted keystroke in question as well as the command that keystroke is currently assigned to.

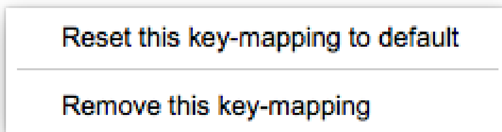


Figure 10.19: Reset / Remove Key-map Entry box

Right-clicking an existing entry will provide options to 'Reset' the current key-mapped command to its default setting, or 'Remove' the key command entirely.

There are three extra Key Commands categories; **Monitor Controller**, **Parameter Controls** and **Talkback Controller** provide additional keyboard controls not available through the menu bar items or their submenus.

The Monitor Controller section provides keyboard control for the Monitor Controller Dim, Mute and Mono on/off toggle switches, Volume Control nudge up and down commands, and selection of up to eight Monitor Sources and eight Monitor Outputs. See Monitor Controller keymaps for the default keystroke settings.

Parameter Controls provides Coarse and Fine 'increment up' and 'increment down' controls for any numeric parameter control under your mouse cursor. Coarse adjustments are generally a factor of ten larger than fine adjustments. For example, a single "Coarse Up" command on a mixer fader will increase the gain on that fader by 5.0dB, whereas a "Fine Up" will increase the gain by 0.5dB.

The Talkback Controller category lets you assign keyboard commands to toggle Talkback and Listenback sends to your Cue Controller sends. Note that the "Talkback" command that you assign here will be an on/off toggle, not a momentary switch like clicking the "Talkback" button in the Cue Controller window. The Talkback default key commands are used as the example in the "Global Key Commands" section on the next page.

A tooltip describing the function of each command will appear when hovering the mouse cursor over the command.

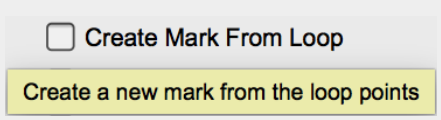


Figure 10.20: Edit Key Commands menu tooltip example

Shown here is the Session timeline command, "Create Mark from Loop".

Global Key Commands

Due to the nature of Console3d as the control interface for your MIO units, there are a number of Console3d-specific key commands that you might want to trigger while working in a different application. Primary examples include operating the Monitor Controller, Talkback and/or Session transport while editing in a DAW.

In the “Edit Key Commands” window, each command has a checkbox. If the box is checked, the key command will be established *globally* throughout your computer system - it will be active no matter which application is frontmost. To make a command *global*, check the box. To make the command *local* (e.g. operational only from within MIOConsole3d), uncheck the box.

IMPORTANT NOTE: Keep in mind, while MIOConsole3d is running, any *global* key-command will trigger whenever you press that keystroke *no matter what program you are working in*.

This will make it impossible to use that keystroke for any other purpose in any application (including MIOConsole3d). Therefore, if you plan on making the key-command global, you should make sure to use keystrokes that are not meaningful to any other applications (including the operating system). In particular, you should not use un-modified single character keystrokes for global commands as this would make typing that character into a text field (in any application) non-functional.

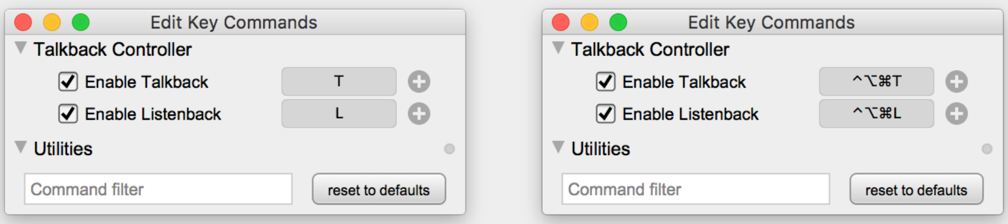


Figure 10.21: Global Key Commands example

For example, in the picture above, you can see we changed the keystroke for the Talkback command from the default ‘**t**’ to ‘**^⇧⌘T**’ (control-shift-command-T) before making it global. That allows you to use ‘**^⇧⌘T**’ to toggle talkback from your DAW or any other program.

Utilities Menu

The Utilities Menu currently only has three items.

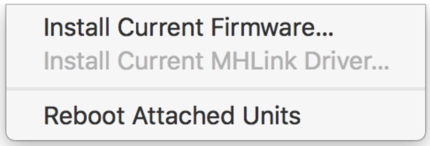


Figure 10.22: Utilities Menu

Install Current Firmware…

Selecting "Install Current Firmware…" will manually initiate the firmware update procedure for all boxes in the active Domain, using the firmware version included in the currently running version of MIOConsole3d. Selecting "Install Current Firmware…" will open the "Install Firmware Update" dialog box, which details the version of the firmware to be installed, with a reminder that all 3d boxes in the domain will need to be powered down and restarted to complete the update process. You may Cancel the update procedure here at this point without interrupting your currently running session.

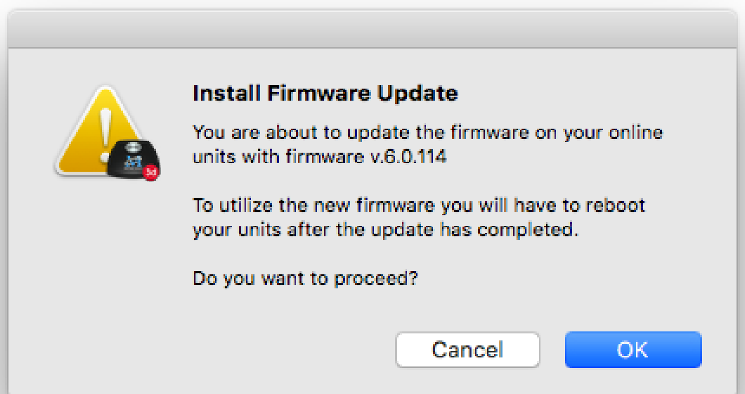
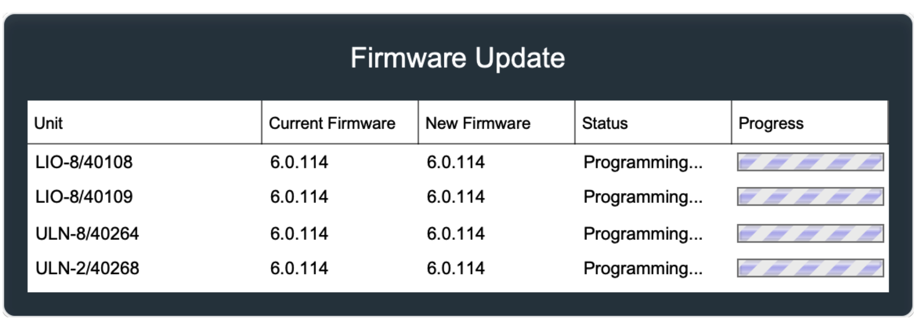


Figure 10.23: Firmware Update dialog

Clicking "OK" in the "Install Firmware Update" dialog box will immediately launch the update process window, shown below.



Unit	Current Firmware	New Firmware	Status	Progress
LIO-8/40108	6.0.114	6.0.114	Programming...	
LIO-8/40109	6.0.114	6.0.114	Programming...	
ULN-8/40264	6.0.114	6.0.114	Programming...	
ULN-2/40268	6.0.114	6.0.114	Programming...	

Figure 10.24: Firmware Update window

This window details the old firmware, the new firmware you are installing, and the progress of the installation process for each box in your current Domain. The update process will register success for each box sequentially, and will automatically close after the last box is done. When the window disappears, you must manually power cycle the boxes for the new firmware update to be complete.

Quitting the MIOConsole3d application is not necessary. When you power-on, the boxes will automatically sync with MIOConsole3d and the new firmware version will be visible for each box in the Domain Status Pane.

Install Current MHLink Driver…

The "Install Current MHLink Driver…" will become available only when a new version of the MHLInk Driver is available. Selecting "Install Current MHLink Driver…" will open the "Install MHLink Driver" dialog box, which details the version of the firmware to be installed, with a reminder that your host computer requires restarting to complete the installation process. You may Cancel the update procedure at this point without interrupting your currently running session.

Reboot Attached Units

"Reboot Attached Units" commands all 3d boxes in your Domain to restart. While this "soft reboot" is not quite the same as a full 'power-off - wait 20 seconds - power-on' cycle, it is a quick way to get boxes back online on the rare occasion a box becomes unresponsive (such as can happen when accidentally launching the MIOConsole3d application on two different computers attached to the same boxes via USB or MHLink). All boxes will re-synchronize and resume in the same state they were when the Reboot command was issued.

> *Note:* The soft reboot performed by this command will only work if the computer is in communication with the Root box. If the Root box is unable to receive commands from the computer, it will not reboot. In these rare cases, confirm your physical connections, and a power cycle of the boxes will bring them back on-line in a few seconds.

Session Menu

The Session panel has been designed with an eye towards hardware-based multitrack studio workflows, with the MIOConsole mix desk as a hardware mix desk and patchbay (which it actually is), and the Session panel as the multitrack tape deck.

In the case of a studio-based multitrack recording session, one would set up the mix desk and patchbay for the available outboard gear and the instruments to be recorded at the start of the session, and use that basic setup throughout the recording session, making relatively minor adjustments as new songs require.

That basic setup would include not just signal processing, but link groups, VCA/DCAs, aux send level processing and panning, sub-mix busing and cue routing.

This way of working was refined over the decades specifically to get the technology out of the way of the creative process, and let the artists and engineers concentrate on the reason they were there: capturing the musical performance, maintaining a creative flow to the recording session, and lending in-the-moment impact to the songs being recorded and mixed.

The MIOConsole3d menu bar Session menu provides commands for the Session window, including track/file import and export, Record and Play transport controls and track overviews navigation, even when the Session window is hidden or closed.

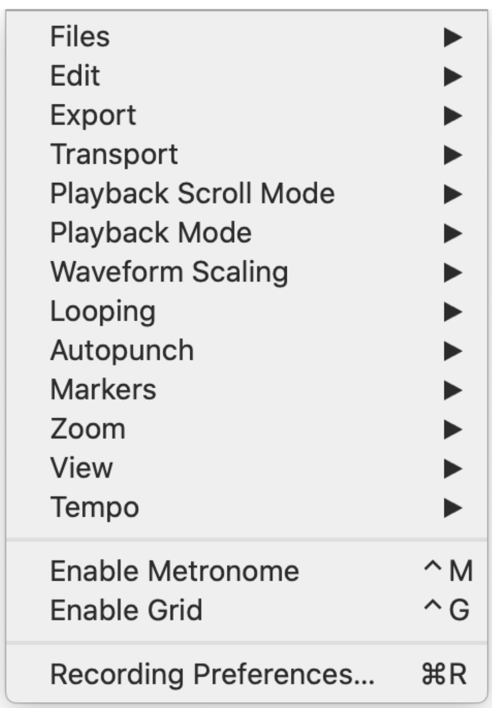


Figure 10.25: Session Menu (top level)

Menu commands for the Session panel are categorized to subfolders for ease of navigation and to allow for future feature expansion.

Many Session menu items include default key commands, and all Session key commands may be edited through the menu bar “Edit” menu: “Edit Key Commands”) interface.

Session: Files menu

The "Files" menu lets you select folders for recording, importing source audio for playback, and basic file-related session functions.

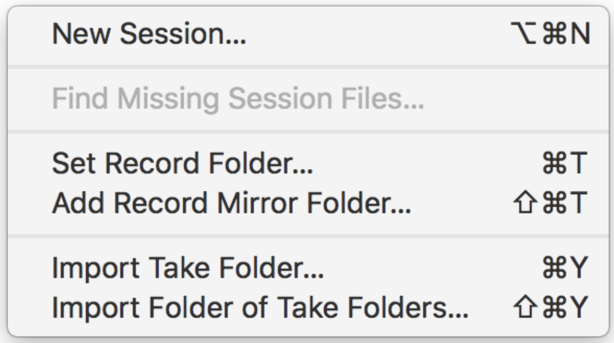


Figure 10.26: Session: Files menu

- New Session... (default key command: ⌥⌘N) will clear all existing Session audio tracks without disturbing the current mix desk. This allows easy re-use of an existing mixer layout with a completely new set of audio tracks.

  If you are working from a modified, but unsaved console, selecting "New Session..." will open a dialog asking if you wish to save your current console file before creating a new session.

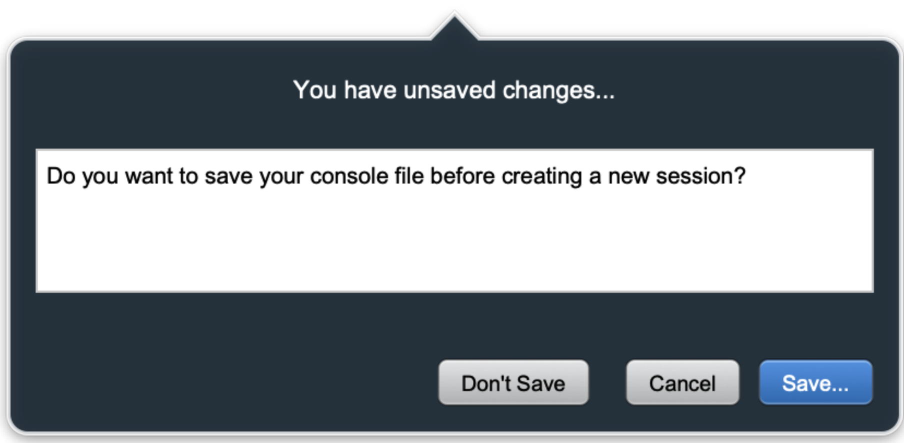


Figure 10.27: Session menu: Files: New Session "You have unsaved changes..." dialog

  You may choose to continue without saving your changes, "Cancel", or "Save...", which will prompt you to choose a location and a name for your saved console file.

  Upon naming and saving your console file, all existing audio files will be cleared from the Session tracks, ready for you to import new source audio.

- The **Find Missing Session Files...** menu item will be active when you open a saved Session file, but some or all of the audio files from that Session are unavailable. This may be because an external drive is not mounted, or if the Session is being moved from another computer.
  A 'Missing Files…' notification will also appear in the upper left corner of the Session panel.

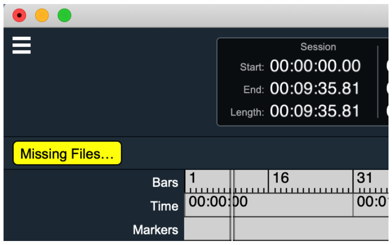


Figure 10.28: "Missing Files" notification

Both the "Find Missing Session Files…" menu item and the "Missing Files" warning button, when selected, open the "Locate Missing Sound Files" window.

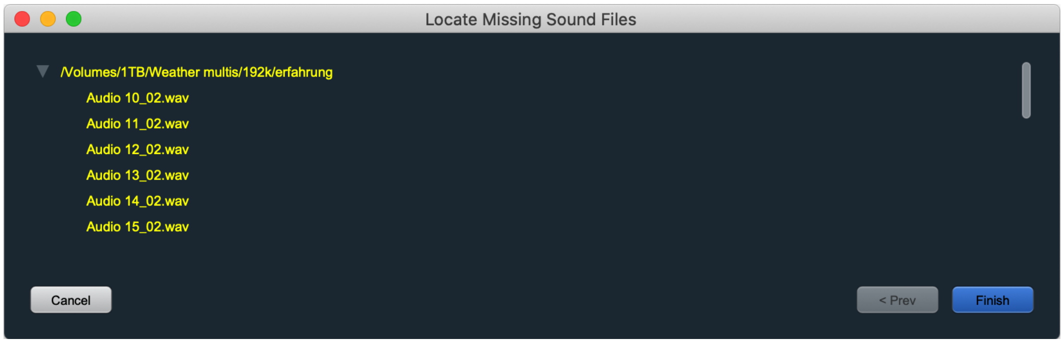


Figure 10.29: "Locate Missing Sound Files"

Missing files and/or their enclosing folder path will be listed in yellow. Selecting either the enclosing folder or a soundfile will automatically copy the name of the selected item and open a Finder navigation window. Select the Search field and paste the copied item name to do a Finder search, or navigate manually to locate the selected missing folder or file.

Once you find the target file or folder, double-click your selection, or hit **Return**, **Enter** or the "Open" button. The selected item name will turn white and show the old location redirected to the new location. Any other listed 'missing' files and folders in the new directory path will also turn white, ready to be loaded.

Hit **Return**, **Enter** or the "Finish" button to finish.

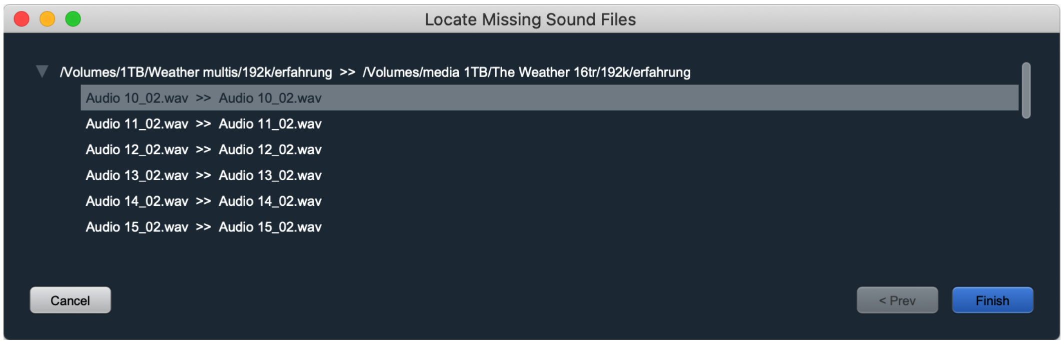


Figure 10.30: Missing Sound Files located

- **Set Record Folder...** (default: ⌘T) opens a Finder navigation window where you can either select an existing folder or create and name a new folder as the primary destination for audio files captured in the Record Panel.

- **Add Mirror Record Folder...** (default: ⇧⌘T) performs essentially the same function, but directs captured audio from the Record Panel to a secondary 'mirror' folder as a confidence backup against potential problems with the primary recording drive. It is also very handy just to have spare copies available at the end of the session.

  Between the extreme efficiency of the MIOConsole3d Record Panel and the speed of todays SSD drives on USB3 and Thunderbolt connections, it is possible to assign multiple record mirror safety destinations for every session.

  Keep in mind that solid state storage is not immune to directory or block errors, and different media may be more tuned to fast access than sustained data writes. With any critical recording process, please, always run a full rehearsal with the storage devices you wish to use in the configuration you wish to employ *before* allowing yourself to fully trust it.

- **Import Take Folder...** (default: ⌘Y) opens a Finder navigation window where you can select an existing folder from which to load audio files captured into your Session.

  Use this command to load a single folder consisting of one set of audio files into the Play Panel.

  All the files in the selected folder will be loaded in alpha-numeric order as new tracks, placed at 00:00:00 in the Session timeline, with a new mix desk strip for each track added to your current Console. This allows you to audition each track individually and decide whether to drag an imported audio file to an existing track (taking advantage of that mixer strips' EQ, dynamics and bus routing), or just use the freshly imported track as-is.

- **Import Folder of Takes Folders...** (default: ⇧⌘Y) opens a Finder navigation window to target a folder of Takes you wish to audition in the Session Panel. Takes will be loaded into the Session Panel as grouped segments in the timeline, in sequence with the first take at the head of the timeline at the left. A location marker for each Take will be placed in the timeline header with each Take Name for identification and quick navigation.

Session: Edit menu

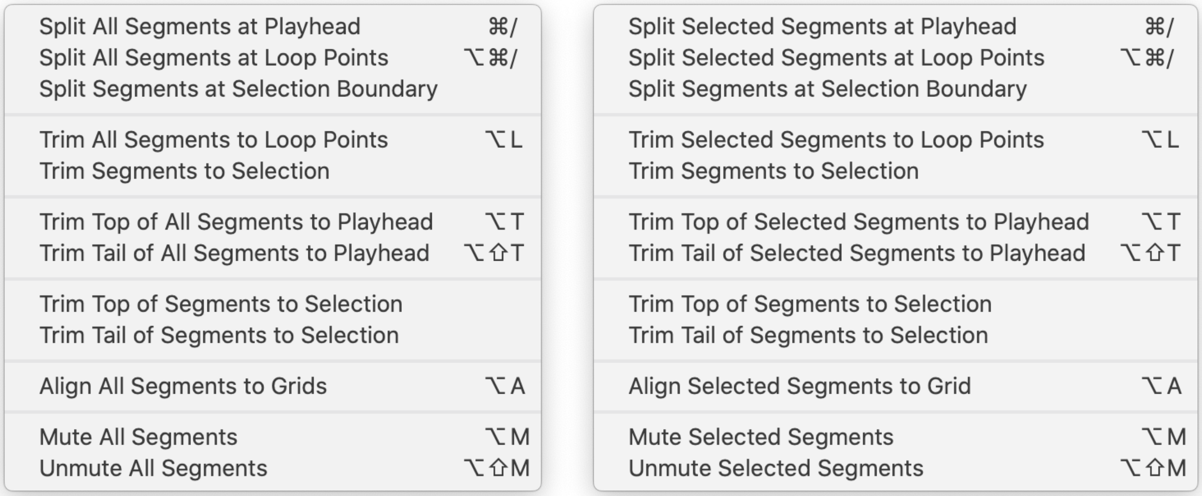


Figure 10.31: Session: Edit menu

Session: Edit menu commands are context-sensitive, as shown above.

When no segments are selected, Edit menu commands will apply to all tracks in the Session. For example, if your session is a live concert recording with 60 tracks, you would use the Edit menu commands to break out each song by editing all tracks simultaneously. This would be the same function as selecting *all* segments in the Session.

When there is a timeline range, segment or segments selected, the menu text changes to reflect that, and the Edit menu commands will apply only to the selected elements.

Note that "Align Segments to Grid" is the only command in the current Session Edit command set that actually moves audio position on the timeline. All "Split" and "Trim" commands merely create or move segment start and/or end points (along with their fades), leaving the audio within the segment in place.

See widget bar "Xfade" and "Time" for "Split" and "Trim" fade and crossfade details.

The definitions below describe the "Selected Segments" context for brevity.

Please Note: Session: Edit menu commands using a timeline selection will apply to all tracks within the timeline selection boundaries. Use "Segment Selection" to apply Session: Edit menu commands specifically to audio in selected tracks.

- **Split Selected Segments at Playhead** (default: ⌘/) places an edit splice in all selected segments at the current playhead location.
- **Split Selected Segments at Loop Points** (default: ⌥⌘/) uses the current Loop Start and End points as edit in/out points, and places edit splices at those points in all selected segments.
- **Split Segments at Selection Boundary** uses the current timeline selection boundaries as edit in/out points, and places edit splices at those points in all selected segments.
- **Trim Selected Segments to Loop Points** (default: ⌥L) uses the current Loop Start and End points as Edit In and Out points, and removes any audio before and after the Loop points in the selected segments.
- **Trim Segments to Selection** uses the current timeline selection boundaries as Edit In and Out points, and removes any audio before and after the highlighted area of the selected segments.

- **Trim Top of Selected Segments to Playhead** (default: ⌥T) moves the In point of all selected segments to the current Playhead position, thereby removing any unwanted audio between the previous In point and the Playhead. Any non-selected segments earlier in the timeline will remain untouched.
- **Trim Tail of Selected Segments to Playhead** (default: ⌥⇧T) moves the Out point of all selected segments to the current Playhead position, removing unwanted audio between the Playhead and the previous Out point of the selected segments. Any non-selected segments later in the timeline will remain untouched.
- **Align Selected Segments to Grids** (default: ⌥A) moves all selected segments to the nearest timeline Grid division. This command is disabled when the Grid is turned off or set to "None".
- **Mute Selected Segments** (default: ⌥M) mutes the selected segments in the timeline. to be clear, 'Mute Selected Segments' is applied specifically to segments in the timeline and is independent of the Mixer Strip Mute control.
- **Unmute Selected Segments** (default: ⌥⇧M) unmutes selected muted segments in the timeline.

  Note: Segment Mute/Unmute commands are not toggles, so 'unmuting' an non-muted segment will have no effect.

Session: Export menu

Session **Export** functions are high-speed file-level commands designed to support fast interchange of Session audio to third-party DAWs.

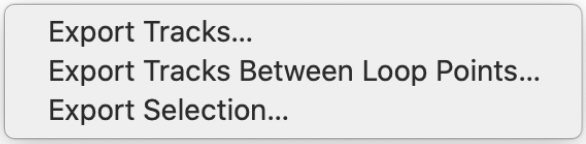


Figure 10.32: Session: Export menu

- **Export Tracks...** exports each track as a single soundfile, starting at timeline 00:00:00.00 through the end of the last segment in the Session, such that all exported soundfiles start at 00:00:00.00 and are of the same duration.

  Empty space in each track will be at digital zero. Edits, fades, segment gains and any summing of track lanes will be executed per-track and included in the exported file.

  Real-time mix desk processing is not applied to Exported soundfiles.

  Think of 'Export Tracks' as a track consolidation process, designed to make all files in a Session easily transportable to a third-party DAW or archive media.

- **Export Tracks Between Loop Points...** is similar to "Export Tracks", but exports the audio only from the Loop Start point to the Loop End point.

  Exported files will be named for their source track, but will contain audio only from within the Loop. It is best to export these files to a folder named with a description of the loop from which the file is derived.

- **Export Selection...** is similar to "Export Tracks", but exports the audio only from within the current Timeline Range selection. It is best to export these files to a folder named with a description of the selected audio, with a reference to the Session from which it was derived.

All Session Export functions follow the same rules as Export Tracks, where each exported Track file in the Export will be of uniform length (with digital zeros padding any empty spaces) and edits and segment-level processing executed per-track.

When selecting an "Export…" function, a dialog will open asking you to select a destination folder for the export. Upon creating or selecting the destination folder, the export will do its thing and open the folder containing the newly-exported files.

> **Note regarding Session exports:**
> All exported files are named for their originating Track. In the case of duplicate filenames in the export destination folder, the newer exported file names will be disambiguated as detailed here in the Session: "Take" recording section. For the most part you can replace the word *Take* with *Export* as you read to place it all in context.
>
> It is always advised that you save every export to a uniquely-named folder, preferably a folder name which describes the content of the exported files in detail.

Session: Transport menu

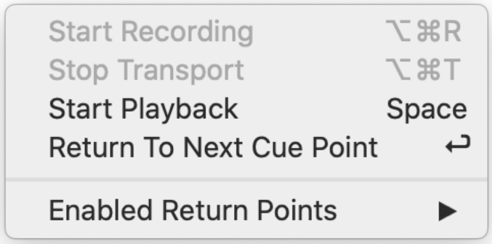


Figure 10.33: Session: Transport menu

- The **Start Recording** (default: **⌥⌘R**) command is available whenever one or more MIOConsole3d mixer strips are Record Enabled (i.e.: the Record Enable button is RED).

  If the Session Record Trigger mode is set to 'Manual', **Start Recording** will start recording immediately, including audio already captured in the preroll buffer.

  If the Session is in either "Level" or "TimeCode" trigger mode, **Start Recording** will arm the record engine, and Record will engage per the settings you've made in the Recording Preferences window.

  Hitting **Start Recording** while a take is already in progress will immediately start a new take, including any manual break overlap assigned in the Recording Preferences. (see the Preferences: Recording section for further details)

- **Stop Transport** (default: **⌥⌘T**) is like the Stop button on any tape deck or DAW: if Session is actively recording or playing back, **Stop Transport** stops it.

- **Start Playback** (default: **spacebar**) will start play, or stop play/record.

- **Return To Next Cue Point** (default: **Return key**) will toggle the playhead through the cue points selected in the "Enabled Return Points" submenu (immediately below). Each time you hit "Return" will move the playhead to the next cue point.

- The **Enabled Return Points** submenu lets you select the cue points to be included in the "Return To Next Cue Point" toggle rotation.

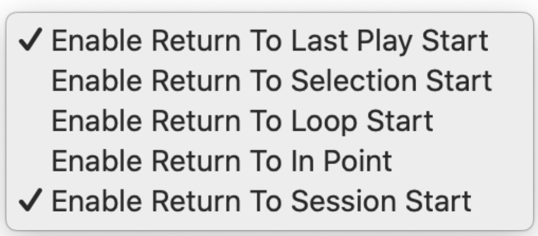


Figure 10.34: Session: Transport: Enabled Return Points selection menu

Session: Playback Scroll Mode menu

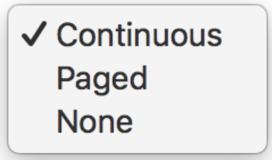


Figure 10.35: Session: Playback Scroll Mode menu

- The **Continuous** scrolling mode places the playhead in the middle of the timeline view, stationary, with the audio waveforms scrolling behind it from right to left in time with audio playback.
- In **Paged** scrolling mode, the playhead moves from left to right with the waveforms remaining still. Just before the playhead reaches the right edge of the panel, both the playhead and the waveforms snap back to the left edge of the screen. and the Playhead resumes from left to right again.
- **None** is a fully manual timeline view. The waveforms display is stationary, and the Playhead is free to roll offscreen while you zoom in and examine a specific waveform or edit.

Session: Playback Mode menu

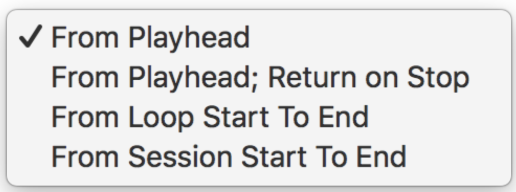


Figure 10.36: Session: Playback Mode menu

"Playback Mode" controls the behavior of the Session panel timeline playback.

- **From Playhead** plays from the current Playhead location, and after stopping, continues playing from the last point the playhead stopped.
- **From Playhead; Return on Stop** plays from the current Playhead position, and when stopping, returns the Playhead to its previous playback position.
- **From Loop Start to End** plays from the start of the Loop to the end of the Loop. If Loop mode is engaged, the loop will repeat.
- **From Session Start to End** plays from the start of the Session through the last segment in the Session.

Session: Waveform Scaling menu

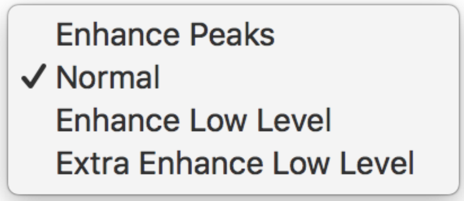


Figure 10.37: Session: Waveform Scaling menu

"Waveform Scaling" provides four relative waveform amplitude scales to better expose peak transients or to reveal low-level signal detail.

- **Enhance Peaks** exaggerates the amplitude scale to enhance transient peaks - useful for beat location and time-aligning tracks.
- **Normal** is the standard linear amplitude scale, good for general overview and scales well to small track heights.
- **Enhance Low Level** is scaled to show low level signals more clearly without completely obscuring high-amplitude peaks.
- **Extra Enhance Low Level** pushes up the low amplitude level view for tight editing and noise floor examination.

Session: Looping menu

Figure 10.38: Session: Looping menu

Session provides a number of looping modes for rehearsal, overdubbing and editing. The Session transport will loop even 192kHz multitracks with zero latency, so no time is lost at loop transitions.

- **Loop Playback** (default: **⌘L**) defaults as 'enabled' when opening a new session.

  The Session Transport "Loop On/Off" button and the currently looped section of the timeline turn yellow to indicate that "Loop Playback" mode is engaged.

  Looped audio will play continuously unit stopped.

- **Set Loop When Cueing to Mark** (default: **⌥⌘L**)

  When selecting a Loop Mark in the Marker List, "Set Loop When Cueing to Mark" will automatically turn on looped playback for that mark.

- **Set Loop Start to Playhead** (default: **⇧⌘<**) sets the Loop Start cursor to the current playhead position.
- **Set Loop End to Playhead** (default: **⇧⌘>**) sets the Loop End cursor to the current playhead position.
- **Set Loop End to Selection** sets the Loop Start and Loop End to the extents of the current selection range.

Session: Autopunch menu

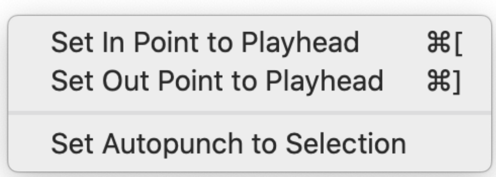


Figure 10.39: Session: Autopunch menu

The Autopunch "In/Out" timeline bar shows the start and end points of the current Autopunch selection.

- **Set In Point to Playhead** (default: **⌘[**) sets the Autopunch In point to the current playhead position.
- **Set Out Point to Playhead** (default: **⌘]**) sets the Autopunch Out point to the current playhead position.
- **Set Autopunch to Selection** sets the Autopunch In and Out points to the current timeline selection boundaries.

Session: Markers menu

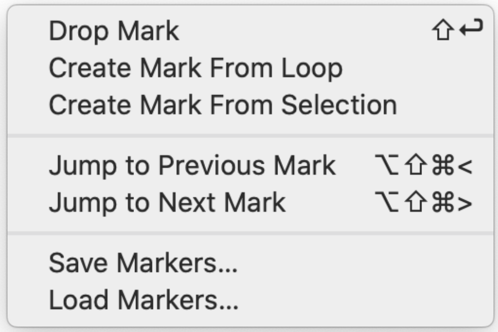


Figure 10.40: Session: Markers menu

"Markers" in the Session can be used as standard location pointers, for editing remarks, QC comments or playback Cues. In this case a Marker would have a duration of zero, simply indicating a point on the timeline.

MH Session Markers can also have a duration, used to define loops, song arrangement sections, production comments, or any other longer-running events on the timeline.

Markers are displayed in the order in which they occur in the timeline Marker List to the right of the Session Tracks waveform overviews. Use the Marker List to name and add comments and notes to each marker using the editable "Name", "Reason" and "Notes" fields.

Note: "Start", "End" and "duration" fields display location and duration information as defined from the timeline and are not editable in the Marker List.

- **Drop Mark** (default: **⇧ Return/Enter**) immediately drops a locator mark at the current position of the playhead. Dropped marks have no duration.
- **Create Mark from Loop** creates a Mark with start and end points copied from the current timeline Loop. The command works the same whether the Loop is active or inactive.
- **Jump to Previous Mark** (default: **⇧⌥⌘<**) immediately locates the Playhead to the previous mark in the timeline.

  Note: 'Jump to Previous Mark' is also triggered by EuCon and Mackie Control surface transport control " **<<** ".
- **Jump to Next Mark** (default: **⇧⌥⌘>**) immediately relocates the playhead to the next mark in the timeline.

  Note: 'Jump to Next Mark' is also triggered by EuCon and Mackie Control surface transport control " **>>** ".
- **Save Markers...** opens a Finder "Save…" dialog box where you can save current Marker List metadata as an independent 'json' format markup text file. The file extension for saved Markers files is '.markers3d'.
- **Load Markers...** lets you locate and load saved 3d Session Markers metadata to your current Session timeline and Marker List. Please note that loading a '.markers3d' file will overwrite your existing Marker List.

Session: Zoom menu

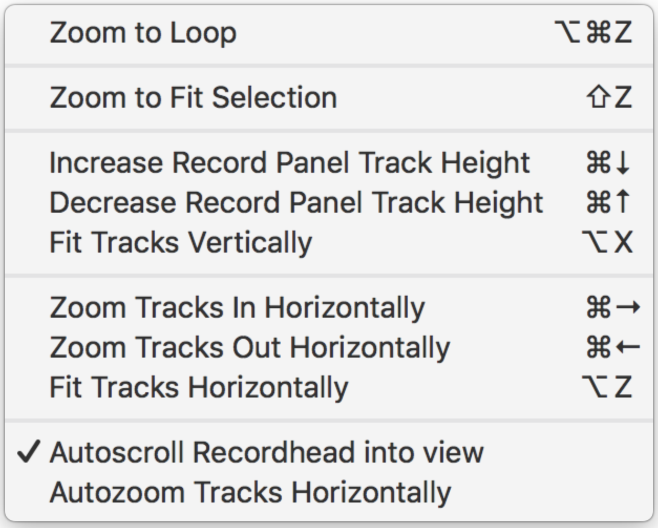


Figure 10.41: Session: Zoom menu

- **Zoom to Loop** (default: ⌥⌘Z) horizontally zooms your timeline view to the current Loop.
- **Zoom to Fit Selection** (default: ⇧Z) focuses your timeline view to the currently selected track segments.
- **Increase Session Tracks Height** (default: ⌘↓) incrementally increases the size of all tracks in the Session overview, centered on the current cursor position.
- **Decrease Session Tracks Height** (default: ⌘↑) incrementally decreases the size of all tracks in the Session overview, centered on the current cursor position.
- **Fit Tracks Vertically** (default: ⌥X) is a toggle control which will vertically squeeze all tracks in the Session overview into the available space of that window. If you've got a lot of tracks in there, they're going to be tiny, but they will only squeeze down to the point you can still read the channel name.

  Toggling 'Fit Tracks Vertically' off again will return to your previous vertical zoom view.

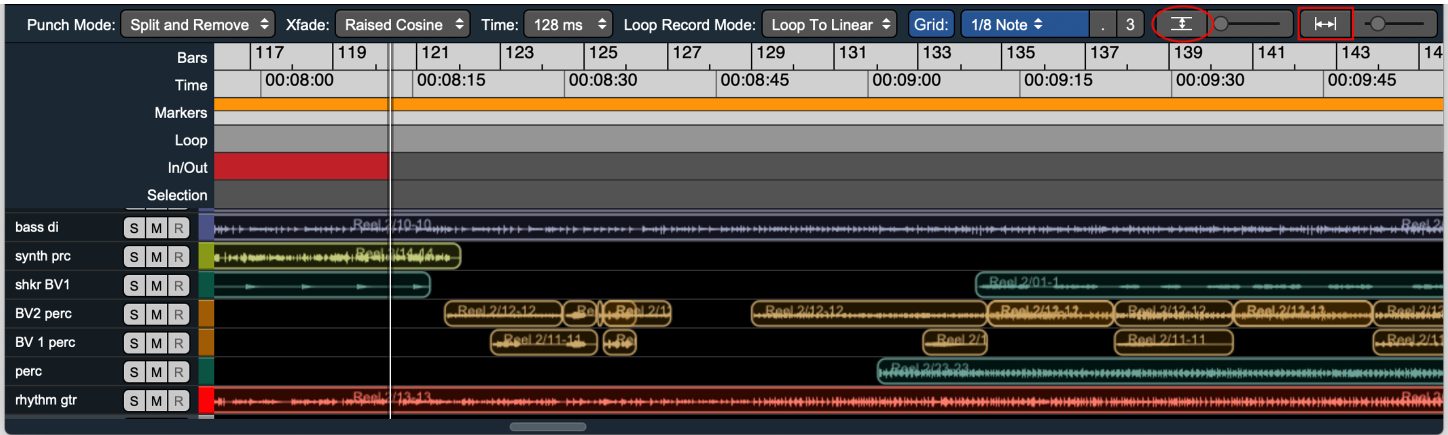


Figure 10.42: Session overview: '*Fit Tracks Vertically*' (minimum view)

The screenshot above shows tracks squished to their minimum height.

Note towards the upper right above the Track List, the widget bar 'Fit Tracks Vertically' button (circled in red) is colored blue, indicating it is enabled.

- **Zoom Tracks In Horizontally** (default: ⌘→) does pretty much just that. If the Overviews display is actively recording or playing with **Autoscroll Recorded Into View**, this command will zoom in towards the playhead at the right of the Overview. In all other conditions, **Zoom Tracks In Horizontally** will focus towards the beginning of the audio file at the left edge of the Overviews window.

  Note to the right of the 'Fit Tracks Vertically' button above the Track List, the 'Fit Tracks Horizontally' button (in the red rectangle) is colored gray, indicating it is disabled.

- **Zoom Tracks Out Horizontally** (default: ⌘←) zooms the tracks out as you would expect, centered on the current playhead position.

- **Fit Tracks Horizontally** is another toggle control which, when enabled, zooms out to show the entire length of all sound files in the current Session timeline.

  Like its vertical **Fit Tracks Vertically** counterpart above, **Fit Tracks Horizontally** has a widget bar control above the Tracks List which turns blue when engaged, and when disengaged will return you to your previous horizontal zoom view.

- **Autoscroll Recordhead into view** will draw the recorded waveforms all the way to the right edge of the current overview window, and then auto-scroll the waveforms back when it reaches the edge. This mode always shows the most recently recorded audio in the Overview.

- **Autozoom Tracks Horizontally** zooms out automatically as you record, so the right edge of the timeline stays at 00:00:00.00, and the Recordhead is at the far left of the Tracks Overview. In this mode, all takes in the Session are always visible within the Tracks Overview window.

Session: View menu

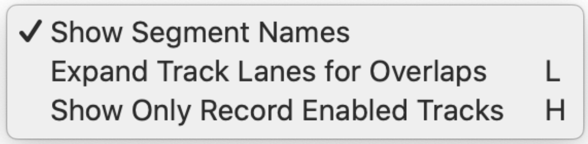


Figure 10.43: Session: View menu

- **Show Segment Names** shows the segments' audio file name and its enclosing folder name along the top of each segment waveform. The segment name is placed such that it is always visible regardless of zoom level, unless the segment itself is too small to contain the file name.
- **Expand Track Lanes for Overlap** (default: **L**) expands or collapses Track Lanes in the Session overview.

  Track *Lanes* are employed to visually manage overlapping Takes and/or segments within a track. New Track lanes are created to accomodate any new audio segments, whether a new recorded take or an audio segment copied or moved from another track.

  Collapsed track lanes take up less space, with the waveforms of overlapped audio superimposed over each other.

  Expanding the track lanes exposes all segments within the track to separate lanes to facilitate editing, fade adjustment and segment gain tweaking.
- **Show Only Record Enabled Tracks** (default: **H**)hides any tracks which are not currently record-enabled, such that only *Record-Enabled* tracks are visible.

Session: Tempo menu

The “Tempo” menu is provided to calculate the tempo of any selected musical phrase in the timeline, and, if desired, apply that tempo to the bars/beats timeline and drive the metronome.

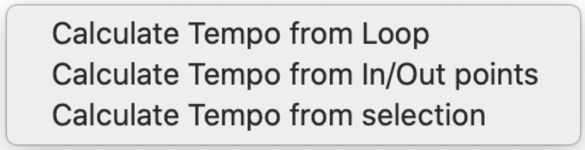
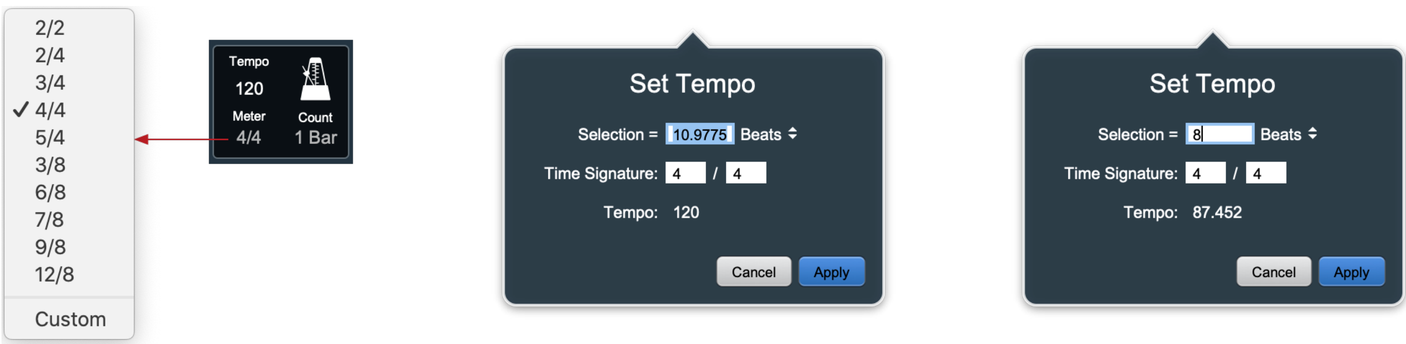


Figure 10.44: Session: Tempo menu

- Select a passage of music as a timeline loop, making sure the time signature for the loop is correct in the Session Meter Control block. Open **Calculate Tempo from Loop** to enter in the number of beats in your loop and calculate the loops tempo. This tempo is then applied to the Bars/Beats timeline and drives the metronome.



select time signature   enter the number of beats in the Loop   tempo is immediately calculated

Figure 10.45: Session: Set Tempo

Note: If you have an odd meter which is not present in the Meter pull-down menu, you can select “Bars” as opposed to “Beats" in the Set Tempo interface, enter the number of measures in the selected loop, and manually enter in any legit time signature (say, 15/16).

- **Calculate Tempo from In/Out Points** is the same function as “Calculate Tempo from Loop”, but uses the “In/Out” timeline range for the tempo calculation rather than the Loop start and end.
- Likewise, **Calculate Tempo from Selection** uses the current segment selection for tempo calculation.

Session menu: Enable Metronome

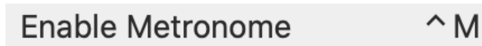


Figure 10.46: Session: Enable Metronome menu item

Enable Metronome (default: **^M**) enables/disables the Session transport metronome. When enabled, the Metronome icon in the Transport Header Tempo block will be highlighted white, and a “Click” Input Strip will appear at the far right of the Mix desk. The “Click” is assigned by default to the Main bus, and may be routed anywhere in the system like any other Input Strip.

Session menu: Enable Grid

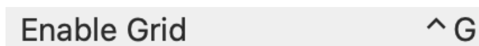


Figure 10.47: Session: ‘Enable Grid’ menu item

Enable Grid (default: **^G**) enables/disables the Session Track Overviews Grid. The Grid is like a quantization field for playhead and edit cursor placement within the Session track overview, making all cursor placement snap to a Bars/Beats boundary.

Turn on the Grid, and in the Widget Bar just below the Transport controls set the Grid to "Bar". Zoom in a bit and drag the playhead around. As you move the Playhead, note that it snaps to the top of each Bar.

When enabled, the Grid affects the placement all edit moves, Loop and Autopunch In/Out points, and Selection points.

Session menu: Recording Preferences

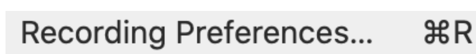


Figure 10.48: Session: ‘Recording Preferences’ menu item

Recording Preferences (default: **⌘R**) directly opens the MIOConsole3d Preferences: ‘Recording’ pane.

Mixer Menu

The Mixer menu items provide control over a wide variety of core mix desk functions, behaviors and display preferences, most of which you will use (either from the key commands or the menu itself) every time you fire up MIOConsole3d.

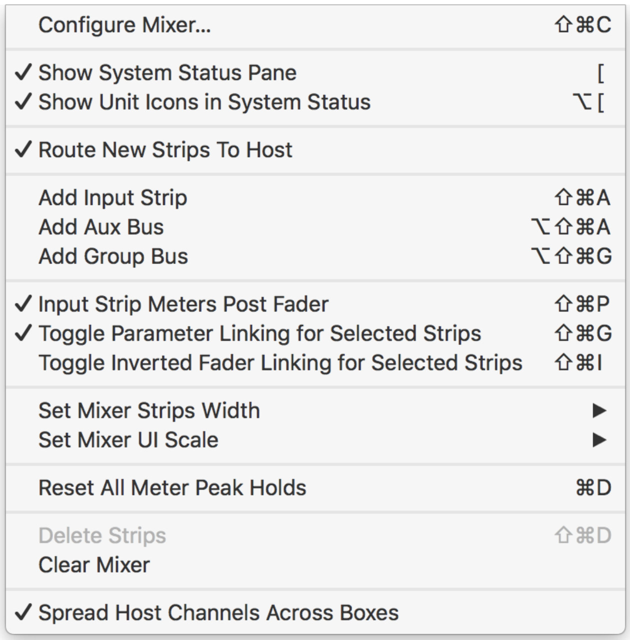


Figure 10.49: Mixer Menu

- **Configure Mixer** is where you design the busing layout of your Mixer desk. The 'Configure Mixer' window lets you configure your Main, Aux and Group buses, DCAs and Mute groups, assign your Solo Mode, and enable/disable the Hard Mutes feature. This is generally the first place to go when you are ready to configure a new mixer. The default key command to open 'Configure Mixer' is (command-shift-C).

  See the 'Configure Mixer' section for the full breakdown.

- **Show System Status Pane** shows/hides the System Status Pane on the far left side of the Mixer window. The System Status Pane is where you manage MHLink Domain selection, box status, system clock source, sample rate, digital audio and SCP I/O configuration. Mute Group and Link Group naming, behavior and enable/disable controls are also located here.

- **Show Unit Icons in System Status** is a toggle control to show or hide the Unit Icons within the Status Pane.

Figure 10.50: Status Pane Unit Icon

Unit Icons are enabled by default, as they help to visually break up the unit status sections and make them easier to read. Unit Icons can be hidden to accomodate more devices or extended Mute and Link Groups lists.

- **Route New Strips To Host**, when checked, will automatically create new routes back to your Host computer on every new Input or Bus strip you create. These new routes will be assigned to the next available return channels to the Host computer… so if you already have 12 mono input strips returning to the Host computer on channels 1-12 and you create a new stereo strip, that strip will be created with "To Host" returns assigned to channels 13 and 14.

> *Note:* "To Host" always automatically cascade-assigns the next available route back to the Host computer. This is by design and is intended for building or modifying large groups of Input-to-Host computer connections quickly and accurately.
>
> Whenever you need to use a *specific channel* to return to the Host, use Direct Outs, I/O Inserts or a Bus output.
>
> *Note Also:* When MIOConsole3d auto-assigns or cascade-assigns stereo routes automatically, the assigned stereo pairs will always use odd numbers for the left channel and even numbers for the right channel. So when auto-assigning, if the next available route is #12 and you create a stereo strip, the Console will skip channel #12 and assign the new stereo returns to channels 13/14. This logic also follows for multichannel surround groupings: left channels will be odd, rights will be even (except where the target configuration implicitly specifies otherwise). Additionally, multichannel sources and outputs will always be created of consecutive channel numbers.

Add Input Strip

…is for adding new Input strips to your mixer. If you have a strip selected in the mixer and invoke **Add Input Strip**, a new empty strip will be added to the right of your selected strip. with no strip selected, the new strip is placed after the the last strip of your mix desk, all the way to the right. The default key command for "Add Input Strip" is **⇧⌘A** (Shift + Command + A)

This new strip will be named "In __" at the bottom of the strip. The number assigned reflects the total number of Input strips already in your mixer, plus one.

Figure 10.51: New Input Strip source button

To assign an Input Source for the new strip, click on the 'Source' button at the top. This will open the input source selection menu (shown below). Here you choose to create Mono, Stereo or Multichannel strips, select an Input source for the strip(s), and if you wish, use 'Cascade' to create multiple strips of the same type in sequence.

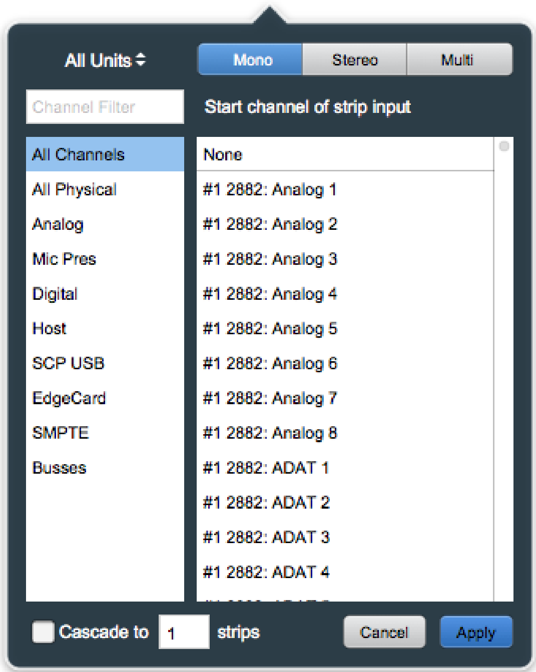


Figure 10.52: Source selection window

Input Source Routing

Since an MHLink domain presents hundreds of available input sources to choose from, all routing selection boxes allow you to filter the visible choices by both the individual box and by the type of source you wish to target.

Select "Mono" to create mono input strips, "Stereo" to create stereo strips, and "Multi" to create multichannel strips. Multichannel strips will always be configured to the same channel format as your Main mixer bus.

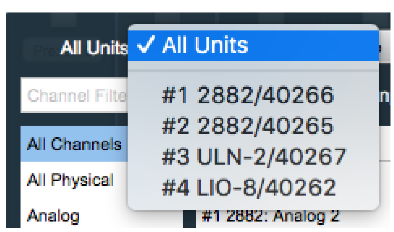


Figure 10.53: "View sources by box"

Clicking on "All Units" opens a drop menu where you can narrow your selection view to show only the routable sources from a particular box on your MHLink daisy-chain.

Beneath the "view sources by box" selector is a search field where you can enter the name of the type of input you are looking for. This is especially useful for searching particular types of digital inputs. Text entry search is dynamic, so type in 'TO' to show only TOSLINK, 'MA' for MADI, etc.

The categories in the list on the left let you refine your view as follows:

Category	
All Channels	shows all input sources
All Physical	filters out MHLink (ethernet), USB and SMPTE ports
Analog	shows only analog inputs

Category	
Mic Pres	shows only analog inputs with internal mic preamps (ie. no LIO analog ins)
Digital	shows only physical digital ports (AES, ADAT, TOSLINK, SPDIF and MADI)
Host	shows the routable sources from your Host computer (either USB or MHLink)
SCP USB	shows the routable sources from non-Host USB ports
EdgeCard	lists only inputs from EdgeCards
SMPTE	lists only the dedicated SMPTE inputs on ULN8 and LIO8 units
Busses	shows the currently available Aux and Group buses in the current Mixer

Table 10.1. 3d Routing interface filter categories

> *Note:* “Cascade”, when checked, will create multiple input strips of the same configuration, but with sequential input channel assignments (much like you have seen in other DAWs and digital consoles).
>
> For example, to create four stereo inputs from an ADAT input, hit [command-shift-A] (the keystroke for **Add Input Strip** shown in the Mixer menu).
>
> In the routing window that pops up, click “Stereo” to create stereo input strips, select "Digital" from the filter list on the left (to show only the digital inputs), select "ADAT 1” (as the first channel you want to route from), type "4" in the Cascade box at the bottom of the routing window (to create your 4 stereo strips), then hit ‘Apply’.
>
> You will have created four stereo ADAT input strips, assigned from ADAT 1 & 2, ADAT 3 & 4, ADAT 5 & 6 and ADAT 7 & 8, respectively.
>
> This procedure is the same whether creating 2 strips or 72, mono, stereo or multichannel.

- **Add Aux Bus** is for adding a new Auxiliary mix bus to your mixer. If you have a strip selected in the mixer and invoke **Add Aux Bus**, the new Aux will be added to the right next to your selected strip. If no strip is selected, the new Aux will appear after the the last strip of your mix desk, all the way to the right. The default key command for “Add Aux Bus” is **⇧⌥⌘A** (Shift + Option + Command + A).

  See “Configure Mixer: Aux Buses” to customize your Aux bus behaviors.

- **Add Group Bus** is for adding a new Group bus. As above, if you have a strip selected in the mixer and invoke **Add Group Bus**, the new Group will be added to the right next to your selected strip. If no strip is selected, the new Group will appear after the the last strip of your mix desk, all the way to the right. The default key command for “Add Group Bus” is **⇧⌥⌘G** (Shift + Option + Command + G).

  See “Configure Mixer: Group Buses” to customize your Group bus configuration.

  > **Note**: Multichannel Input strips, Aux mix buses and Group buses are always created with the same channel configuration as the Main master bus. In other words, if your Main is configured as ‘5.1 ITU’, a new Aux, Group or multichannel Input strip will also be configured ‘5.1 ITU’ to match the Main.

- **Input Strip Meters Post Fader** sets the Mixer strip meters to reflect signal level after the fader. This is a global setting for all Mixer fader meters.

- **Toggle Parameter Linking for Selected Strips** toggles the "Selected Strips" link group function (visible in the "Link Groups" section of the Status Pane to the left of the Mixer window).
- **Toggle Inverted Fader Linking for Selected Strips** toggles the 'Inverted' mode of the 'AdHoc link group'.

  Hint: Having easy-to-remember key commands for the two above toggles is especially handy for performance-oriented mixing

- **Set Mixer Strips Width** will change the width of any selected mixer strips. There are four widths to choose from: **Narrow**, **Normal**, **Wide** and **Extra Wide**. **Wide** or **Extra Wide** can make is easier to see multichannel meters in strips with greater than 6 channels.

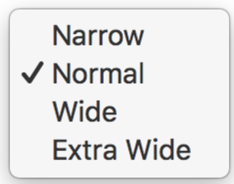


Figure 10.54: Mixer Strip Width selector

- **Set Mixer UI Scale** adjusts the vertical scale of Mixer desk control elements to better accommodate different display sizes. There are five relative scales to choose from. Combining 'Mixer UI Scale' with 'Mixer Strips Width' and **Configure Channel Strip Elements** (from the Mixer Pane hamburger menus) lets you build efficient Mixer desks that show only the controls you need and will fit comfortably on any computer display arrangement.

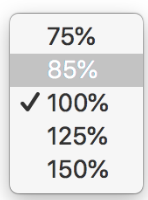


Figure 10.55: Mixer UI Scale selector

  Note that Mixer UI scaling does not change the size of all of the Mixer strip elements - the fader/meters section remains elastic and will shrink or stretch as other Mixer elements are resized, added or removed.

- **Reset All Meter Peak Holds** clears peak hold markers on all Mixer desk meters. You can also ⌥-Click (<option>-Click) any peak hold marker.
- **Delete Mixer Strips** will delete any selected Mixer Input or Bus strips and clears all I/O routes, processors and Link and Mute Group relationships assigned to the deleted strips.
- **Clear Mixer** removes all Mixer Input strips and clears all I/O routes, processors and Input strip Link and Mute group associations to the Input strips.

  Clear Mixer does *not* clear Bus strips, Link or Mute Groups, Monitor Controller or Cue Controller configurations. Bus strips and configurations managed from the **Configure Mixer** window are not affected by the **Clear Mixer** command.

- **Spread Host Channels Across Boxes** enables distribution of DSP processing load across all MHLinked boxes in a Domain… add a box, double the processing power, add another box, triple it, etc. Generally the root box bears the biggest load since it manages the traffic between the Host computer and all the other boxes, but you can help manage that load a bit by keeping the following in mind:

1. DSP processes assigned within a single Mixer strip will be assigned to the same box, so if you have a ton of insert processes running on one particular strip, that process load will not be shared across boxes.

2. Processing on signal paths routed to a physical output port rather than a bus - Monitor Controller and Cue Controller Graph, etc. - gets assigned to the 3d box connected to the physical output of that path. So, if you have a 6-channel surround MC output with room correction, delay and EQ in the DSP Graph going to the analog outputs on box #3, that DSP Graph processing will be executed on box #3.

I/O Menu

The **I/O Menu** provides two separate functions critical to the operation of MHLink. The first manages the bandwidth between the Root box and your Host computer, and the second optimizes MHLink data flow between the Root box and the daisy-chain for large-scale MHLink domains.

See USB I/O Configuration for 3d USB ports configuration.

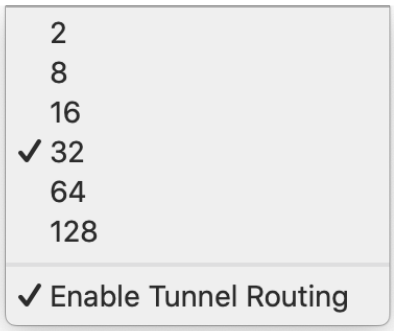


Figure 10.56: I/O Menu

- **MHLink Host channel allocation:** Selecting ‘2’ through ‘128’ determines the number of bi-directional channels presented to the Host computer over MHLink Gigabit Ethernet.

  Setting the I/O to to ‘64’ means your Host computer (and therefore your DAW) will see 64 channels in and out of the MHLink Core Audio device. The default is 32 channels in and 32 channels back out to the Host, which is a pretty easy load to bear, even at 192kHz sample rate.

  The '2 channel' I/O option is available for compatibility with mono/stereo-only internet audio/video conferencing applications.

  The thing to remember here is, higher channel counts and higher sample rates combine to make for a heavier load on your Host computer. Most computers suitable for DAW use can handle 128 channels of I/O at up to 96kHz without too much trouble, but if you don’t really need 128 channels, 64 is easier to manage anyway and any Host DAW can always use a little more cpu headroom.

> *Note:* If you can not dedicate a Gigabit Ethernet port exclusively to MHLink, and need to run MHLink over your regular IP network in a business or corporate environment, you might find 64 or 32 channels of I/O strikes a better balance between network response and audio bandwidth than the full load of 128 channels. The same might be true if you routinely run hi-res video from a local server alongside your audio streams. When performance issues on a shared network arise, try reducing your I/O channel count and see if that helps.

- **Tunnel Routing** is enabled by default; it is what allows MHLink to route audio between boxes without introducing additional latency. Due to the bandwidth reservation required to make Tunnel routing work, if you will be using more than 6 boxes in an MHLink chain @ 4x sample rates you will need to turn Tunnel routing off.

  In some future release of the software, MIOConsole3d will manage this automatically and this menu item will be removed.

Monitor Menu

The Monitor menu houses controls for various Monitor Controller functions including creation and editing of Monitor sources, outputs and Cue Controllers, editing Monitor Output Graph processors and Load/ Save of Monitor Output configurations files.

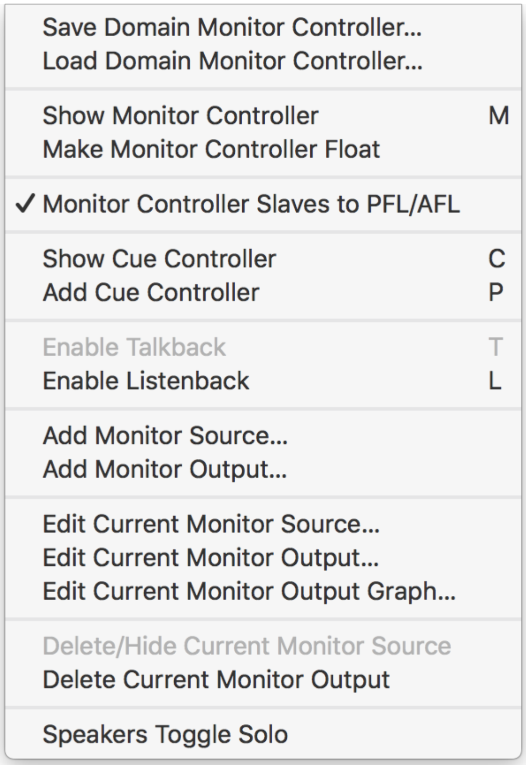


Figure 10.57: Monitor Menu

- **Save Domain Monitor Controller...** opens a 'save file' dialog box where you can choose the location to save the current Monitor Controller output configurations. Saved Domain Monitor Controller (.mc3d) files contain all source and output routes, trims, offsets, delays and graph processes associated with your current monitor configurations. This saved file can be later recalled after importing a foreign MIOConsole3d mixer which had a different Monitor Controller setup than your current system

  Please note: Speaker 'Solo' and 'Mute' states are not saved in Monitor Controller presets.

- **Load Domain Monitor Controller...** will manually load a previously saved Monitor Controller configuration to your current MIOConsole3d Mixer.

  See also Interaction Preferences for details regarding Monitor Controller loading options.

- **Show/Hide Monitor Controller** is a toggle to show or hide the Monitor Controller window.

- **Make Monitor Controller Float**, when selected, makes the Monitor Controller window float in front of other windows on your display, so it is visible even when MIOConsole3d is in the background.

- **Monitor Controller Slaves to PFL/AFL** - When the Mixer is in one of the PFL/AFL modes, having this command checked will automatically set the Solo bus as the MC Source whenever a "Solo" is engaged. De-selecting this control leaves the MC Source uninterrupted when Solo is engaged. This function is available for each of the Cue Controller sends, so you could have, say, a Cue send to headphones slaved to the PFL/AFL Solo bus, while not interrupting the control room mix for a producer.

- **Show/Hide Cue Controller** shows/hides the Cue Controls window.
- **Add Cue Controller** adds a new Cue Controller to the Cue Controls stack ready to be configured.
- **Add Monitor Source** opens a window where you can name, select input routes, and set trims, gain offsets and channel configurations for a new Monitor Controller source.

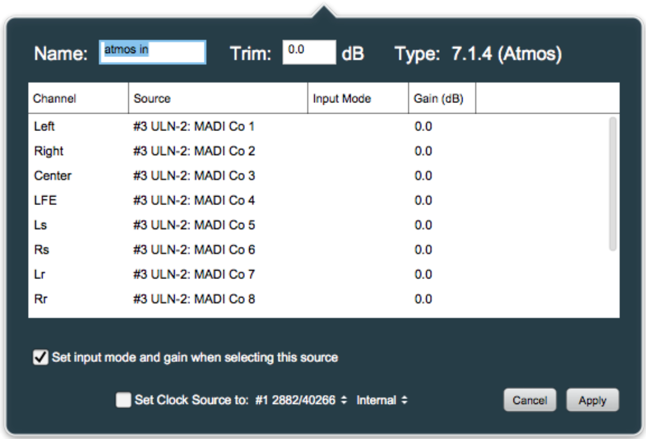


Figure 10.58: Monitor Source routing window

The checkbox "Set input mode and gain when selecting this source" allows you to configure the physical input when you select this input. For instance you could use this to, say, switch between the 'Line +4' and the 'Mic S/R' inputs on a ULN8 depending on your MC Source selection.

The "Set Clock Source to:" checkbox lets you switch the Domain clock source to the selected digital input, useful for connecting a wild digital input from a CD, DAT or other digital device which can not be clocked to the MHLink Domain. Switching to another Monitor Source resets the domain clock to it's original state.

- **Add Monitor Output** opens a window where you can name the new output, select output destinations, trims, output mode, gain offsets, delay offsets and channel configurations for a new Monitor Controller Output.

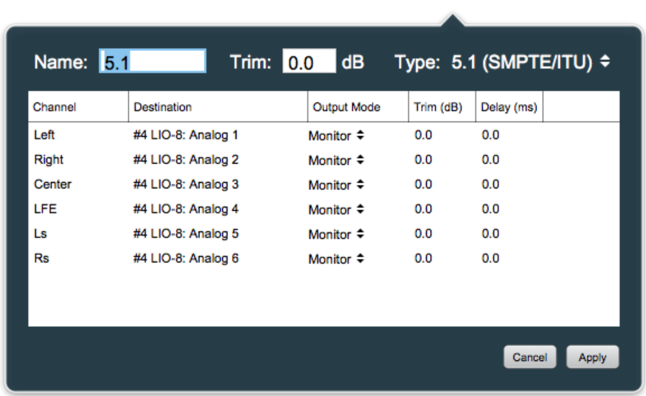


Figure 10.59: Add Monitor Output window

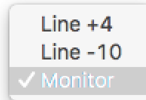


Figure 10.60: MC Output Mode selector

Output mode sets the analog gain stage for the selected output channel between +4, -10 and Monitor, which pads the output signal 30dB for directly feeding amplifiers and powered monitors.

- **Edit Current Monitor Source** opens a small control box where you can adjust the gain trim of the selected Monitor Source input.
- **Edit Current Monitor Output** opens a window for the currently selected Monitor Output where you can edit the Output parameters. This interface is the same as the **Add Monitor Output** window shown above.
- **Edit Monitor Output Graph** opens the DSP Graph window for the selected Monitor Output. The Graph can be used to build custom crossovers, apply gnats-hair-accurate acoustic correction, phase offset, bass management, theatrical monitoring filter curves or any other type of processing that one might imagine applying a monitor path. Monitor Controller Graphs are completely independent of the Mixer process path. Graphs can be saved, recalled and edited freely, as can each component processor used within the DSP Graph framework. Click the link for detailed functions and capabilities of the DSP Graph.
- **Delete Current Monitor Output** removes the currently selected Monitor Output, releases all routed output connections and frees up all associated DSP Graph resources.
- **Delete/Hide Current Monitor Source** will delete the currently selected Monitor Source from the MC window. If the selected Monitor Source is an Aux or Group bus referenced in the 'Configure Mixer' window, the Monitor Control status of that bus will be changed from "In MC" to "Not In MC". The Main Mix bus monitor source feed can not be deleted or hidden.
- **Speakers Toggle Solo** - By default, clicking on a speaker icon in the Monitor Controller window will turn that speaker icon red and mute that speaker feed. When **Speaker Toggle Solo** is checked, clicking on a speaker icon will turn the icon yellow, and will solo that speaker feed.

See the Monitor Controller and Cue Controllers sections for detailed operational commands for these features.

Window Menu

The Window menu features controls for MIOConsole3d window behaviors, views and navigation.

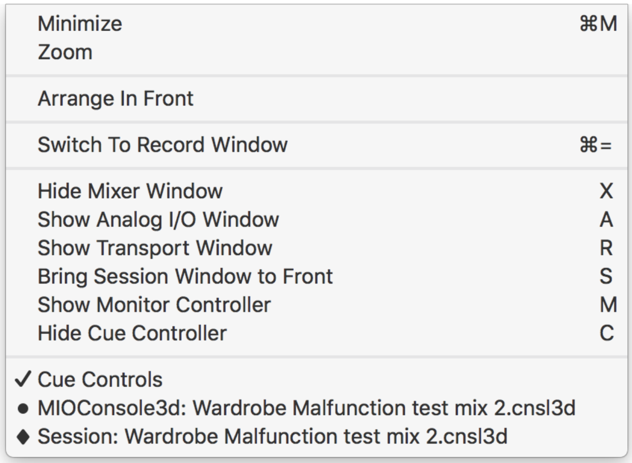


Figure 10.61: Window Menu

- **Minimize** sends the currently selected window to the Finder Dock.
- **Zoom** is a toggle command will expand the main Mixer window to fill the current display screen, then back to it's original size.
- **Arrange In Front** brings all MIOConsole3d windows to the front of other application windows that might be hiding them.
- **Switch to Record Window** toggles the current view between the Mixer window and the Record Panel window.

  When "Preferences: Interaction > Switch to Mixer/Record Window" is set to 'Hide switched-from window', toggling to the Mixer will hide the Record Panel.

  When "Preferences: Interaction > Switch to Mixer/Record Window" is set to 'Does not hide switched-from window', toggling switches focus between the Mixer and Record Panel windows normally.
- **Show/Hide Mixer Window**, **Show/Hide Analog Window**, **Show/Hide Monitor Controller** and **Show/Hide Cue Controller** toggle the visibility of each window. **Bring Session Window to Front** indicates that the Session window is open, but not currently in focus. 'Bring Window to Front' will also refer to any windows that are open but minimized.
- Below the 'Show/Hide' window items is the standard macOS 'Current Window' selector list, which shows all open windows for the current application. The 'MIOConsole3d' window is the main console interface with the Status Pane and mixer desk, and will always show the currently loaded MIOConsole3d file.

  Note in the Window menu screenshot at the top of this section, a check mark '✓' indicates the window currently in focus, in this case, the Cue Controls.

  When the MIOConsole3d mixer window is not focus, the dot '●' next to the MIOConsole3d name indicates that the currently loaded MIOConsole3d file has been modified but has not been saved.

  The diamond '♦' next to the Session window indicates that the window is open, but minimized to the macOS Dock.

Help Menu

The Help menu provides an interface to Metric Halo information resources, web links and submissions of feature requests and trouble reports.

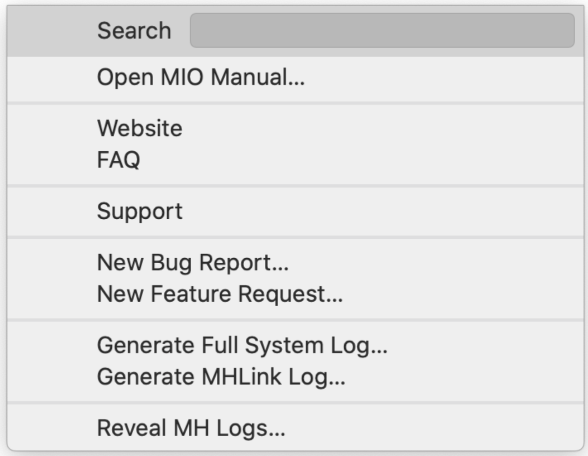


Figure 10.62: Help Menu

- **Search** searches the MIOconsole3d menu bar menu items first, and then the macOS Help menus.
- **Open MIO Manual...** opens the MobileIO Users Guide. If there is a new version of the manual available from the mothership, the new version will download automatically.
- **Website** opens your web browser and takes you to the main Metric Halo website landing page.
- **FAQ** opens your web browser and takes you to the Metric Halo 3d Frequently Asked Questions web page.
- **Support** opens the Metric Halo Support Help Desk web page, where you can submit requests for Sales Support or Technical Support.
- **New Bug Report...** opens the Metric Halo Bug Reporter. Use this form to submit reports of any problems or just unexpected behaviors you find while using MIOConsole3d.

  Please be as specific as possible regarding the sequence of events that revealed the problem, explain what you were trying to do when the problem occurred, and describe what happened that was unexpected (or just plain wrong). To address any technical issue, we must be able to re-create the issue in the lab, so the more information we have to do this, the better the chances we can resolve the issue and post a fix quickly.

  If you ever need to use the Bug Reporter, please make sure to include your return email address so we can follow up with you and make sure we get the issue resolved as quickly as possible.

  Note: The Bug Reporter will pop up automatically the next time you launch MIOConsole3d after an application crash.
- **New Feature Request...** opens the Metric Halo Bug Reporter form, but will be directed to the "Feature Request" queue at MH HQ rather than the Tech Support queue. If there is something you really want 3d to do, definitely do not be shy about asking for it! The more requests we get for any given feature the faster it becomes real.
- **Generate Full System Log...** scans the operating system log and saves a text file with the log from the last hour called `mhlink_system.log` in your current User Home folder. After the log has been

created, it will be revealed in the Finder. MH Support may ask you to run this command and send the generated log for support or troubleshooting purposes.

- **Generate MHLink Log...** scans the operating system log and saves a text file with the log items generated by the MHLinkDriver over the last hour called `mhlink_system.log` to your current User Home folder. After the log has been created, it will be revealed in the Finder. MH Support may ask you to run this command and send the generated log for support or troubleshooting purposes.
- **Reveal MH Logs...** opens the temporary folder which contains the realtime MIOConsole3d and individual MHLink 3d box activity logs.

The System Status Pane

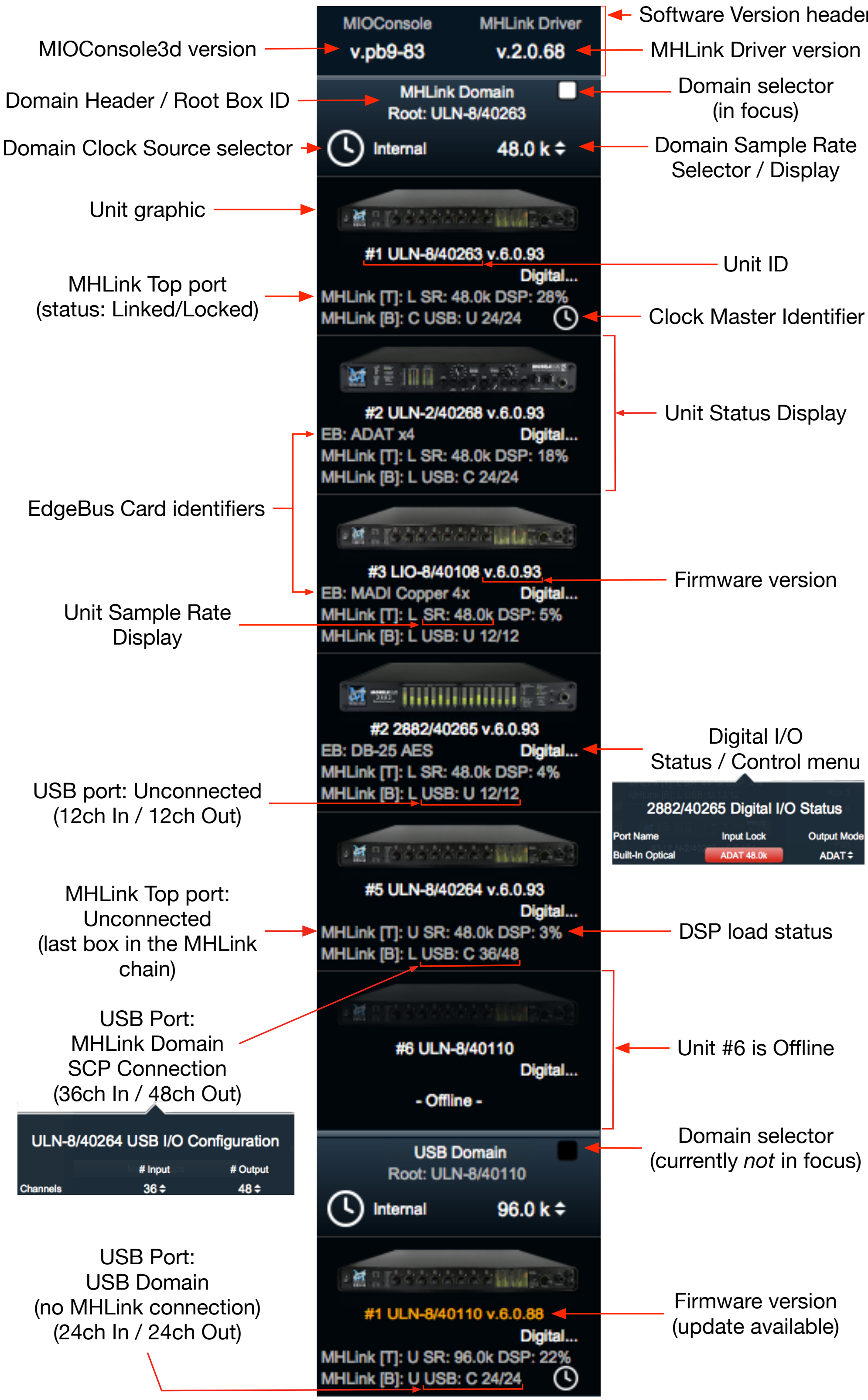


Figure 10.63: System Status Pane map

The **System Status Pane** resides at the far left side of the main MIOConsole3d Mixer window, and provides realtime status information for every 3d device visible to your computer. It is also where you monitor and configure all clocking parameters and physical digital input and output ports on each of those devices, including the USB ports.

The System Status Pane can be hidden entirely by clicking the "[" key, or toggling the menu bar Mixer menu: "Show System Status Pane" menu item.

The full map of the System Status Pane is available for reference on the previous page.

Let's break down how the Status Pane is laid out, then get into the details of each section.

System Status Pane organization

There are five sections graphically represented within the Status Pane, but only the first three concern us here. The other two sections, Mute Groups and Link Groups are covered in their own sections of the "Mixer Strip Controls Breakdown" chapter.

Software Version Header

The **"Software Version Header"** sits at the very top of the System Status Pane, displaying the currently-installed MIOConsole3d application and MHLink Driver versions.

MIOConsole3d automatically checks for new software updates upon launch. When an update is found, the Software Version header text will turn orange. Click the orange header text to download the new Console to your "User"/Downloads folder.

See the Software Installation section for full installation details, including important macOS security considerations when installing audio software in general. (Short version, for those who know the drill: launch the new Console and click the orange text…)

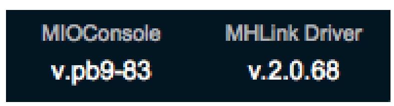


Figure 10.64: System Status Pane: Software Version Header

Note that updating the MHLink Driver will require a computer reboot, while updating the MIOConsole3d application does not.

Domain Header

Immediately below the 'Software Version Header' is the **Domain Header**. Each Domain is seen by your computer as an independent hardware audio interface, even though they are all controlled from the MIOConsole3d application. Domain controls such as sample rate and clock master operate just like any other audio interface, regardless of how may individual MHLinked 3d devices there are in the Domain.

Figure 10.65: Domain Header

There will be a Domain Header for every domain visible to your computer, whether the boxes are attached via Gigabit Ethernet or USB ports.

The 3d box directly attached to the computer is designated as the "Root Box" for that Domain, and is identified as such by model and serial number in the Domain Header, as shown above.

When you add boxes to a running MHLink daisy-chain, those boxes will sync up and appear within that Domain at the bottom of the list with all input and output routes immediately available in the routing dialogs and their analog input channels automatically added as input strips to your current mixer, ready to get to work.

You can add or remove boxes from the end of the daisy-chain while audio is running without interrupting the audio streams.

Note: There is no way to directly bridge between domains, aside from a regular audio connection. Workflows involving multiple-domain scenarios have not been explored to a great degree as of yet, but the framework is in place for you to experiment as you wish.

Avoid situations where two computers can communicate with the same 3d units either by USB or MHLink, as that introduces the likelihood of a communications conflict.

Regarding USB Domains: MHLink multibox aggregation is not currently supported with USB computer connections, so 3d boxes connected to your computer via USB will operate as single-box Domains. Any 3d boxes connected to the MHLink ports of a USB Root box will be shown in the Status Pane for identification purposes only, and will not be active as audio devices.

Unit Status Displays

Below the Domain Header will be one or more **Unit Status Displays**. The Unit Status Display includes a Unit Icon graphic with the model and serial number of the unit, EdgeCard™ details (if installed), current configuration details plus the digital I/O port controls for that 3d box. There is a Unit Status Display for every box in each Domain.

Figure 10.66: Unit Status Displays

The Unit Icon can be hidden if desired by clicking the (default) "**⇧[** (Shift-[ )" key command, or toggling the menu bar -> Mixer menu: "Show Unit Icons in System Status" menu item.

As mentioned previously, the box directly connected to the Host computer is designated as the 'Root' box, and is always at the top of the list. 3d boxes on the MHLink daisy-chain will always appear by their place in the chain: the Root is box #1, connected to that is box #2, then #3 and so on down the list.

Domain Status and Management

Each Domain appears to the computer and behaves as its own distinct Core Audio device, operating with its own system clock and driver channel count, with fully independent routing, mixer desk, cue and monitoring functions. All functions for all Domains are controlled from within this same Console3d interface.

Domain Header Controls

Information and controls located in the **"Domain Header"** are global for every 3d box within that domain. The "Domain Header" shows the box type and serial number of the "Root" box of that Domain. The "Root" box is always Box #1 in the Domain list, signifying its connection to the Host computer, and also its control of all other 3d Units that Domain.

Figure 10.67: MHLink Domain Header details

The **"Domain Selection"** status box at the right of the "Domain Header" turns 'white' to indicate the Domain currently controlled by the MIOConsole3d UI. In other words, the Mixer desk, Monitor Controller, Cues and Session timeline that are currently visible are all controlling the Domain that is in focus (status box = 'white').

When a Domain is not in focus, its status box will be dark. Note that when a Domain is not in focus, everything keeps running on those boxes just as you left it. When you change Domain focus you are just telling your Console3d interface to control a different set of boxes - rather like switching a MIDI keyboard controller from playing one synth on MIDI channel 1 to playing a different synth on MIDI channel 2. Same general idea.

Click on the "Domain Header" to switch your mixer desk, routing, Monitor Controller, Cues Controls and Record Panel focus to control that Domain.

Domain Master Clock Source

The **"Domain Master Clock Source"** selector opens a menu where you can select the master clock for your Domain. "Internal" will currently always clock to the Root box. You can, if you wish, clock the Domain from any other physical digital input on any box in the Domain, and this is where you do it.

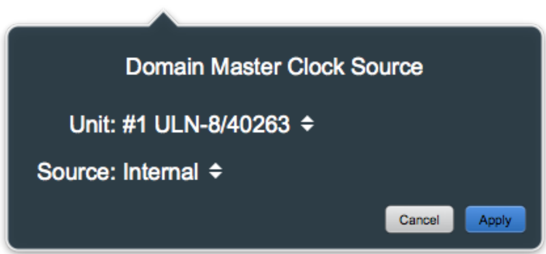


Figure 10.68: Domain Master Clock Source selector

There are two pull-down menus available in the Master Clock Source selector. With the **"Unit:"** pull-down, you select the physical Unit (or box) in your Domain which houses the digital input you wish clock from.

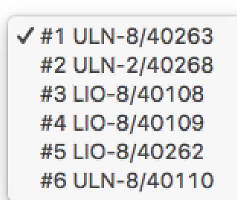


Figure 10.69: Master Clock "Unit" selector

Once you have selected the desired Unit, use the **"Source"** menu to choose a clock source from the digital ports available on that box.

Each box in your domain may have an different set of digital inputs from which to choose, so the **Source** menu will vary from box to box.

All 3d boxes have a Word Clock input. All ULN-8s and LIO-8s have 8-channels of AES I/O built-in. All 2882s and ULN2s have an ADAT/TOSLINK optical digital I/O built-in with an 'SPDIF • AES' EdgeCard™ included by default (unless you ordered a different card). The list of available clock sources in the **Unit Clock "Source" selector** will also include the digital ports of any EdgeCard™ you have installed. The example below is a stock LIO-8 with a 'SPDIF x2 • ADAT x2' EdgeCard™, so it lists word clock, the four built-in AES channel pairs, plus the two ADAT optical and SPDIF RCA digital ports on the EdgeCard™ as available clock sources.

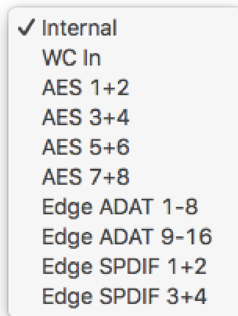


Figure 10.70: Unit Clock "Source" selector

When you have selected an external clock source, the **"Domain Master Clock Source"** selector will change from 'Internal' to display the digital audio clock source from which the Domain is now clocking. This will include the box model and serial number, and the port on that box receiving the master clock signal.

Figure 10.71: Domain Master Clock Source display: set to external

In the example above, your new Domain master clock source is from the EdgeCard™ SPDIF port 1 on LIO-8 sn40109, and it is running at 48kHz (per the "Sample Rate Selector / Display", detailed in the next entry).

Sample Rate Selector / Display

The **"Sample Rate Selector / Display"** (located to the immediate right of the "Domain Clock Source Selector") will always display the current Domain sample rate.

In the example below, the screenshot on the left shows the "Domain Clock Source" set to 'Internal', with the sample rate set to 96kHz. The screenshot to the right shows the system slaved to an external clock source, which is naturally controlling the sample rate for the Domain.

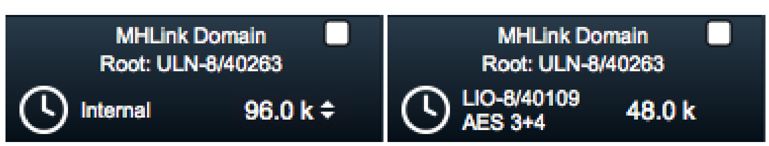


Figure 10.72: Domain Clock Source and Sample Rate Selector/Display examples

Note the up/down arrows next to the sample rate selector when set to 'Internal' - this indicates the selector is live and in control of the sample rate.

However, if you are synced to external clock (as shown on the right), this control becomes display-only, showing the sample rate provided by the external clock. The up/down arrows disappear as further confirmation that control of the sample rate is now external and you can no longer change the sample rate from here.

Clock Master Identifier

The domain master clock source can be set to any word clock or digital audio input port on any 3d box in the domain. The Clock Master Identifier icon in the lower right corner of the Unit Status Display indicates exactly *which* box in the Domain is the clock master. Sometimes you need to know the box type and serial number, sometimes you just need to know which one it is in the rack… in this case, it's box 4.

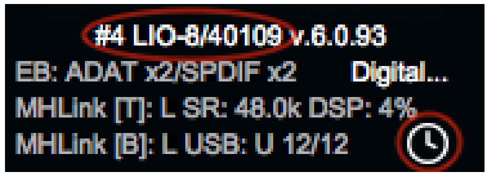


Figure 10.73: Clock Master Identifier icon

The "Clock Master Identifier" icon will always be attached to the Root box when the Domain is set to 'Internal' clock, even if you set the 'Unit' selector to a different box.

Offline Domains

You may at some point have a Domain show up with all its boxes listed as "Offline". This can occur if you have powered off or lost connection to the root box, removed that set of 3d units from the system, or have opened a MIOConsole3d file which contains references to 3d hardware not connected to your current Host computer.

This is a feature requested by touring and location engineers which allows them to examine and edit all MIOConsole3d settings and parameters without being connected to the boxes.

You can save renamed versions of the edited Console file or overwrite the original as desired, and when you reconnect the domain, your changes will be applied just as if you had made them with the boxes attached.

You can select an offline domain by clicking its domain header, and a snapshot of that entire console will be recalled with all the routing, gains and inserts in that mix desk visible. All hardware input and outputs will be shown italicized since the physical boxes are unavailable.

You can clear an offline Domain by "right-clicking" or "control-clicking" the offline Domain Header. This will open a pop-up menu which will remove the offline Domain from the System Status Pane.

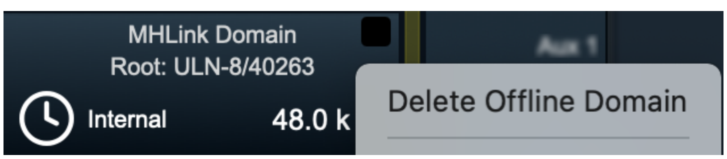


Figure 10.74: "Delete Offline Domain" pop-up

Boot States and Snapshots

MIOConsole3d and your interfaces keep their data in what are called *states*. A state is the collection of data that tells the system how the interface is configured, how your mixer is set up, etc.

This data is stored in three ways:

- *MIOConsole3d file*: This saves your state as a file on your computer with a ".cnsl3d" suffix and can be opened by MIOConsole3d.
- *Boot state*: This is a state stored in your interface that defines how the interface will be configured when it is powered up.
- *Snapshot*: Snapshots are stored in the interface and can be used when the box is used standalone (not connected to a computer).

Unlike Console files, Boot states and Snapshots are for using the hardware "offline" (without MIOConsole3d). They are appropriate for the following use cases:

- Pre-configuring routing and/or processing to use with a system that doesn't have MIOConsole3d available (e.g. Windows, Linux, iOS, Android)
- Pre-configured routing and processing for standalone fixed installation or conversion (e.g. as a side-car for a console or a front-end for a recorder)
- Pre-configuring fixed routing to allow the use of the HW with a Mac without having to launch MIOConsole3d (e.g. to configure the MIO as a "dumb" interface / converter)

The only difference between Boot States and Snapshots is that Boot States are automatically applied by the hardware when it is booting up; Snapshots must be recalled manually. The Boot State is just a Snapshot that is recalled automatically during boot. In the following, we refer to both Boot States and Snapshots as Snapshots.

When you save a snapshot to the hardware, everything required to restore the current running state of the hardware is gathered up and stored on the hardware. This includes:

- Routing
- Physical I/O and Mixer Gains
- Pans
- Mutes
- DSP plug-ins and processing parameters
- Interbox Routing
- Clock Source
- Sample Rate
- Computer Routing including Host and SCP

When the Snapshot is recalled, the state of the running system is restored to the state it was in when the snapshot was saved. Again, this reflects the state of the entire MHLink domain.

Sample Rate Considerations

The state of all the signal processing in the system is configured for the sample rate the system was running at when the Snapshot was saved. The Snapshot will recall the Sample Rate, if the boxes are on internal clock. Changing the sample rate (either from the front panel or via an external clock) will not update the coefficients for the DSP (unless MIOConsole3d is running and connected to the domain), so the sample rate cannot be changed with boot- states or Snapshots if there is any DSP processing that is sample rate dependent (e.g. EQ, Comp, Delay etc.); if the sample rate is changed, the incorrect coefficients from the snapshots' sample rate will continue to be used.

Multibox Considerations

Since 3d and MHLink aggregate multiple boxes into a single audio device, all MIOConsole3d file, Boot State and Snapshot operations operate at the domain level, across all boxes as a unit.

In the case of a multibox domain, this includes all the complex routing between the boxes. **For a multibox system to recall properly, everything must be connected in exactly the same way it was when the Snapshot was saved.**

This means that all boxes must be connected in the same order with the same ports used for the connections). If this rule is not followed, the recall will happen correctly, but the physical ports will not match the routing stored in the Snapshot and the signals will not route properly.

The model is a rack of 3d boxes in a road case, all connected within the case and traveling as a unit with only the external power, analog and digital audio breakouts to be attached at each site. As long as the all the connections within that road case remain intact, all Snapshots will work as programmed.

How Snapshots work

When the system is powered up (or rebooted), the hardware will check to see if there is a valid boot-state snapshot stored on the box. If there is, it will be asserted onto the hardware, and then the last state of the hardware gain controls (that is stored separately) will be restored; this means that if you change gains on the unit when it is running stand-alone with a boot state, and then you power cycle it, it will come back the way it was before you turned it off.

The system sets a non-volatile flag when asserting a snapshot; the flag is cleared after the snapshot is fully asserted. If the system detects the flag during boot, that indicates that the Hardware crashed while asserting the boot state snapshot and the HW will skip the boot state. The flag will get cleared if you save a new boot state or erase the boot state.

Bypassing The Boot State

If you run into some problem with the boot state (or simply want to temporarily avoid loading it during boot), you can hold down the front panel Mute button (or the up-arrow button on the ULN-2/3d) while booting the box. If the Mute button is depressed during boot the HW will skip loading the boot state.

> **Note:** While it is safe to launch MIOConsole3d or connect to a computer running MIOConsole3d, doing so will assert the Console state onto the boxes, replacing the currently running bootstate/snapshot configuration. This will not affect the snapshot states stored in non-volatile memory on the hardware.

Setting Boot States and Snapshots for offline operation

Each 3d device can support one Boot State and up to ten Snapshots.

The first storage slot is always the Boot State.

Configuring and storing snapshots in the box is done in MIOConsole3d:

1. First, attach the interface to the computer and start up MIOConsole3d.
2. Use MIOConsole3d to configure the boxes. Set up all aspects that you care about. Once you have the configuration as you like it, you are ready to save the snapshot.
3. Control-Click the Domain Header to open the Domain Header contextual menu:

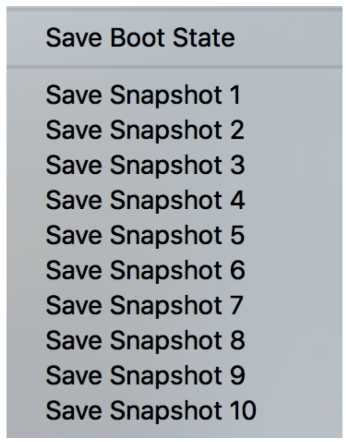


Figure 10.75: Domain Header Contextual Menu: “Save Boot State / Snapshot”

   - “Save Boot State” will save the current states of each connected box in the domain.
   - To save the snapshot to one of the other snapshot slots, choose the appropriate “Save Snapshot x” item (where x is the appropriate number).

4. Even if the snapshot is taken from a saved .cnsl3d file, it is good practice to save a copy of the current Console state to a file on your hard disk with an appropriate tag (like “Snapshot 1” for the 1st snapshot) added to the filename. Keeping an “offline operation” copy of the box configurations.

Pre-configuring Snapshots for USB

There are two ways to pre-configure a Snapshot to function properly as a USB audio device connected to a computer:

1. Connect the 3d box to your host Mac via USB and configure to taste, with the Mixer Host routes naturally addressing the USB port. This is obviously the most intuitive method, but possibly a bit inconvenient for some users.
2. Alternatively, an MHLink-attached box with SCP USB I/O assigned will function as-configured when removed from the domain and its USB port connected to another computers’ USB port.

Erasing Boot States and Snapshots

Below the “Save...” commands in the Domain Header contextual menu is a set of commands to *Erase* the saved Boot State or Snapshots.

Ordinarily it is fine to overwrite a Snapshot in a given slot with a new one, but the Erase commands are available as a troubleshooting tool or just to clear the state memory for a fresh start.

Recalling Snapshots

> **Note:** At the current time, it is not safe to load a snapshot while MIOConsole3d is running and attached to a domain.
>
> Doing so will replace the running state of the hardware without MIOConsole3d being able to synchronize with the hardware.
>
> This will eventually cause the hardware to crash, requiring a reboot of all the attached boxes.
>
> This will be addressed in an upcoming release of MIOConsole3d.

Due to the differences in front panel design, recalling Snapshots is a different process for each 3d box model.

As part of an offline MHLink domain, selecting any snapshot from the front panel of any box in the domain will immediately reflect that selection to every other box in the domain.

ULN-2 Snapshot Recall

The two tact-switches on the left-side of the front-panel (between the status indicators and the meters) may be used to select the snapshot that you want to use to configure the ULN-2. These buttons are labeled with up and down arrows. The currently selected snapshot is indicated by the column of LEDs labeled C, 1, 2, 3, 4, 5, 6, 7, 8, 9.

"C" is the Boot State, followed by Snapshots 1-9 as programmed in MIOConsole3d.

When the ULN-2 turns on, the "C" indicator will be illuminated, indicating that the unit has booted up from its pre-configured Boot State.

Figure 10.76: ULN-2 Front Panel Snapshot Controls

Pressing the up arrow will move to the next higher snapshot in the list (e.g. if you are currently on snapshot 3, you will move to snapshot 2). Conversely, pressing the down arrow will move to the next lower snapshot in the list (e.g. if you are currently on snapshot 3, you will move to snapshot 4). If you are at either beginning of the list and you press the up arrow, you will wrap around to the last item in the list.

When you select a new snapshot, the new snapshot is applied to the box immediately.

ULN/LIO-8 Snapshot Recall

The ULN-8 and LIO-8 provide for front panel recall of eight snapshots; they are stored in the same manner as the ULN-2, and are recalled by going to the "Preset" Control Mode and pushing the channel encoder that corresponds to the preset number you wish to recall.

Figure 10.77: ULN-8 Front Panel in Preset mode

On the ULN-8 and LIO-8, encoder 1 recalls the Boot State. User snapshot #1 is recalled on encoder 2 and so on through snapshot #7 on encoder 8. There is no snapshot recall on the Monitor Control encoder, since that button toggles between Monitor and Cans.

Following the above selector convention, ULN-8 and LIO-8 encoder LED rings show green to indicate that a Snapshot is programmed in that slot, and are dark if the slot is empty. The currently selected and active Snapshot slot is shown orange.

2882 Snapshot Recall

While the 2882 3d motherboard stores and recalls all Snapshots just like the other units, it has no physical controls to allow Snapshot selection from the 2882 front panel.

When connected as part of an offline MHLink domain, the 2882 will follow the snapshot selection commands of the other boxes in the domain.

Unit Status Display Overview

Unit Status Displays list each box in the Domain, starting with "Root" box #1 directly below the Domain header. These fields provide realtime status information for each 3d device as well as access points to the digital audio port controls for each box.

"Right-clicking" or "control-clicking" a Unit Status Display brings up a contextual menu with three important functions.

"Delete Offline Unit" is, appropriately, for deleting offline units, and is detailed in the sections below.

"Identify Box", when selected, triggers a front panel meters dance to visually identify the selected box.

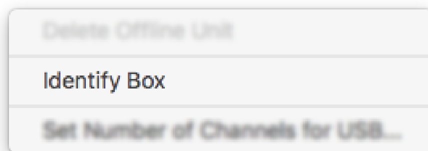


Figure 10.78: "Identify Box" contextual menu

"Set Number of Channels for USB…" is covered in the USB I/O Configuration section later in this chapter.

Unit ID and Firmware

The **Unit ID and Firmware** are listed immediately below the Unit Icon graphic.

Figure 10.79: Unit ID and Firmware version

The 'Unit ID and Firmware' text will turn orange when a new version of the firmware becomes available. Click on the orange firmware version number to initiate the firmware update process.

The firmware update process will be applied to all 3d boxes in the currently selected domain, and will require all boxes to be powered down and restarted to complete the installation process. It is not necessary to quit MIOConsole3d - the boxes will come back on-line automatically once restarted.

Regarding 'Offline' units…

If a box is not communicating with the rest of the domain, the **Unit Status Display** will show the box model and serial number (but no firmware version), and display "Offline" in the Status field section of the display. The "Digital I/O Status" menu remains visible as it includes the EdgeCard™ configuration for the offline 3d device.

If you know why the unit is showing up as "Offline" (or you just want it gone), you can clear it from the Domain Status list by "right-clicking" or "control-clicking" the offline "Unit Status Display" box. This will open a pop-up menu which will remove the offline unit from the Domain list.

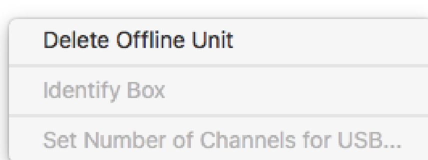


Figure 10.80: "Delete Offline Unit" pop-up

So, say the MHLink connection between boxes #2 to box #3 goes down. Because the boxes are chained serially from one to the next, box #3 *and all the boxes after #3* in the chain will go “Offline” as well. If you re-establish the connection going to box #3, they will all come back - you do not have to restart the Console3d app.

Try it - pull the box #2 ethernet link. It won't hurt anything. This is why we built MHLink, after all. Even ProTools won’t care (as long as it’s not the computer to Root box connection you yank).

Another behavior to note is, when you remove a box from a domain, the box you removed will show up at the end of the Domain list, registered as “Offline”. Active boxes in a Domain will always be listed consecutively, in order from the Host/Root box connection at the top, down to the end of the MHLink chain, with any offline boxes at the bottom.

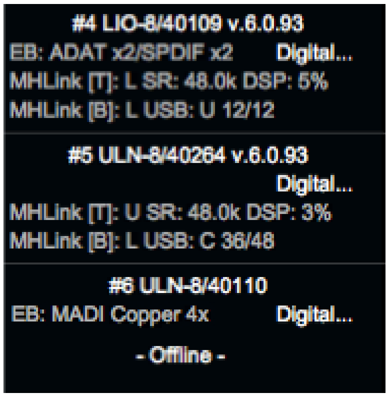


Figure 10.81: Offline Unit detail

Now, you may be thinking, “Hold on, how does the Console know there even *is* another box if it’s not connected?” Fair question…

When you build up a Domain and configure a Mixer with all the I/O routes, Monitor settings and Cue mix sends and save it, that saved MIOConsole3d file retains the state of every 3d box within that Domain, with all analog and digital I/O routes to each box referenced to that box’s serial number. So in the case above, ULN-8 #40110 with an EdgeBus 4x Copper MADI card *was* in the rig at some point, and the Console remembers.

Offline Units in the 3d Mixer Desk

Mixer strip input and output selector buttons assigned to offline boxes will turn yellow, and the source and destination text becomes italicized to indicate that those routes are not currently available.

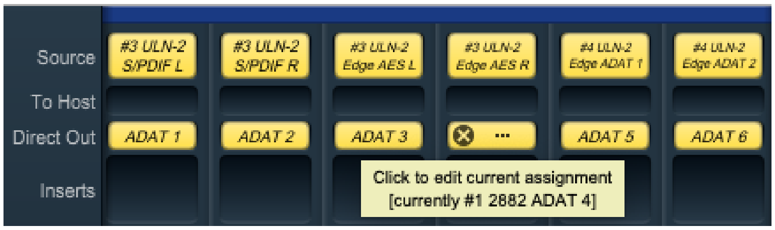


Figure 10.82: MHLink Port Status displays

In the figure above two ULN-2s (boxes #3 and #4 in the MHLink chain) are offline. You can see the inactive source and destination ports info as usual. Tooltip details also remain available for the offline ports.

Why do this? Three reasons:

1. When you re-attach the missing 3d box(es), all I/O routes and DSP processors are restored as if they were never removed.
2. You can use the Box Mapping feature of MIOConsole3d to import entire domains from one facility to another, with all routing, gain structure, mixer parameters, etc. fully intact.
3. You can create or modify mixer files for your standard 3d rig without the boxes being on-line (or even in the same country). In fact, you can modify the mixer files on any computer running the Console3d application. Just fire up the Console3d app, open your Console3d file, edit freely and save the new Console file (appropriately renamed, of course) back to the Host (or email it ahead). When you get to your destination the new file loads up fully functional, just as if it were built with the boxes all hooked up.

MHLink Port Status

The **"MHLink Port Status"** displays the connection status of every MHLink port in the Domain.

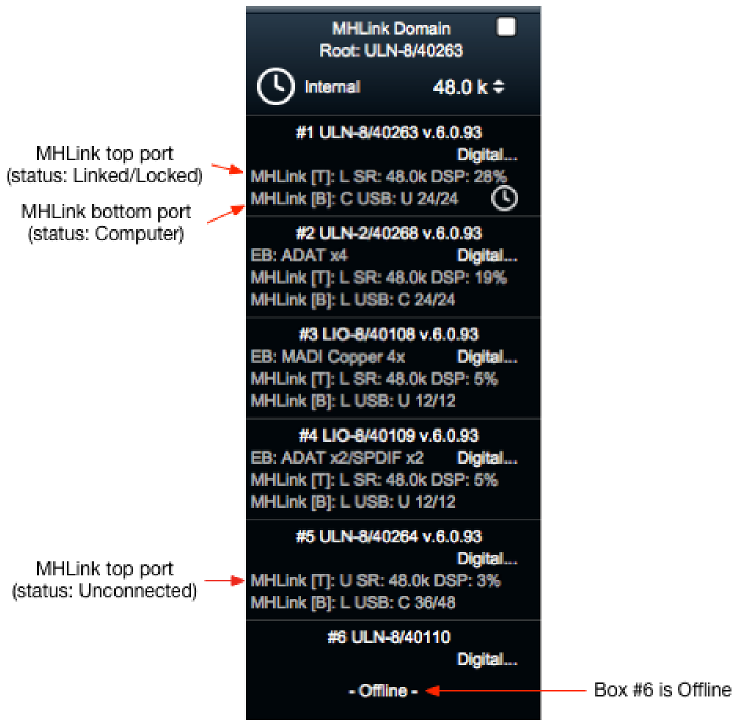


Figure 10.83: MHLink Port Status displays

"MHLink [T]" shows the status of the 'Top' MHLink port and "MHLink [B]" shows the 'Bottom' port of each 3d box. There are three possible states for any MHLink connection. Per the example above, starting at the top:

- "MHLink [T]: L" means the 'Top' MHLink port is **L***inked* and **L***ocked*, meaning it has a good physical connection and is communicating with another MHLink device in the daisy-chain.
- "MHLink [B]: C" indicates the Host **C***omputer* connection, in this case to the 'Bottom' MHLink port. Only box #1, the Root box, will have this designation. If by chance you ever see a "C" on any MHLink unit port other than the Root, you want to pull that plug. You can also think of it as the "**C**onsole" connection, because without this link Console3d cannot communicate with the boxes.
- "MHLink [T]: U" indicates the 'Top' port is **U***nconnected*, meaning one of three things: either there is no MHLink cable attached to that port, the cable that is attached is bad or disconnected at the other end, or the box to which it is attached is turned off.

So in the example above, we know that the Host computer is connected to the Bottom port of the Root box (box #1), we have good MHLink communication with boxes 2 through 5, and that box #6 is disconnected from the Top port of box #5 (MHLink [T]: U) for some reason… and maybe we should check it out.

EdgeCard™ Identifier

"EdgeCard™ identifier": Every 3d hardware unit has an "EdgeBus Card" expansion slot in the rear panel. The type of card you have installed (if any) will be identified in each Unit Status Display, immediately below the Unit ID.

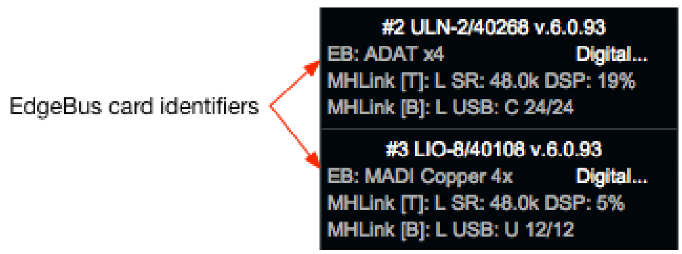


Figure 10.84: EdgeCard™ identifiers

In the example above, box #2 has a 'ADAT x4' EdgeCard™ installed, and box #3 has a 'MADI Copper 4x' card.

Digital I/O Status / Control Menu

At the upper right of the Unit Status, click "**Digital...**" to open the **Digital I/O Status** window for that 3d box.

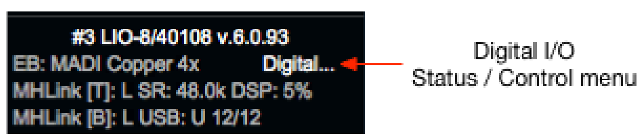


Figure 10.85: Digital I/O Status and output controls

This window provides details regarding every physical digital audio I/O port in the selected 3d unit.

Figure 10.86: Digital I/O Status window

The header at the top of the window includes the model and serial number of the box in focus.

On the left, **Port Name** lists all the digital audio ports available on that box, with 'built-in' ports at the top followed by the EdgeCard™.

Input Lock tells you *if* there is a digital audio signal present on that port, and (on ports which support multiple data formats) tells you what format it is. All 3d digital audio inputs auto-sense the sample rate and data format of the incoming signal and report it here. In the example above, the only signal coming into this box is on port #1 of the MADI Copper 4x EdgeCard. The incoming MADI stream protocol uses the '**Base|Legacy**' data format to identify the sample rate and channel frames, respectively. (MADI formats are detailed at the end of this section.)

'**Output Mode**' is where you set the digital stream format for that particular port. This feature applies to the MADI ports and Optical output ports. The cool thing here is, the 'Output' format you set here is completely independent of the 'Input' format, so you can perform formats conversions on the fly using the same I/O port.

Let's take a look at **Input Lock** states. In the previous example, most of the Input ports are colored dark grey and say "Invalid / No Signal" - meaning there is no usable signal coming in that port.

Here we see three states of functional digital audio streams coming into 3d:

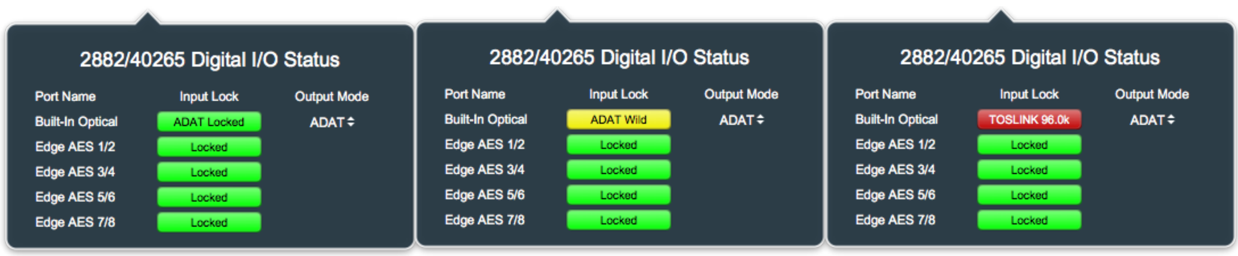


Figure 10.87: Digital I/O: Input States

'Green' and 'Locked' status displays indicate that the incoming signal is fully synced and locked to the Domain system clock.

'Yellow' and 'Wild' means that the incoming signal is the same sample rate as the Domain system clock, but is not fully synced and locked. Ok… here's the thing with 'Wild' digital signals and 3d: 3d has this quirk where incoming digital signals tend to resolve cleanly without clocking artifacts, even when they are technically not supposed to. So chances are pretty good if you are using reasonably modern gear, a 'Wild' digital input stream might sound just fine, and in a pinch be perfectly usable for auditioning unclocked sources (like CD players). It's a cheat, yes, but quite convenient.

For this reason incoming 'Wild' signals are not muted by the system, and it is up to you whether or not to use them as-is. What you need to keep in mind here is that, eventually, the difference in clock rates *absolutely will* drop samples.

Therefore, for any professional production, live and archival recording and critical listening, follow the rules and make sure all your clocking is locked up nice and tight.

A 'Red' status display indicates that a legitimate incoming digital stream is present, but it is too far off the Domain sample rate to resolve a lock. The display reports the foreign incoming sample rate as a troubleshooting tool. 'Red' input ports are muted at the receiver to ensure that no digital artifacts or clocking noises (a phenomenon more common with older digital gear) will get into the 3d Domain.

Note in the case of both MADI and ADAT/TOSLINK optical ports, the format of the incoming signal appears in the 'Input Lock' display along with the lock status. If it can be read (even if it can not be synced), the format will be displayed. For ADAT/TOSLINK optical, the format will either be ADAT or TOSLINK. SMUX is automatic for ADAT at 2x and 4x sample rates, with the usual reduction to 4 channels and 2 channels, respectively. Off-rate MADI signals will report the incoming signals 'Base/Native' and 'Legacy/Long' frame and channel formats.

The MADI I/O has been built as much as possible to be compatible with legacy format machines as well as modern MADI devices. To support this, both the optical and copper-based 3d MADI implementations can be configured on the fly to match up with whatever interface you might come across.

MADI sample rate formats

There are two ways 3d references sample rate for MADI:

'**Base**' reads the audio stream sample rate as the 1x 'Base' rate, where 44.1kHz is the 'Base' sample rate for streams carrying 88.2kHz(2x) and 176.4kHz(4x) audio data, and 48kHz is the 'Base' rate for 96kHz(2x) and 192kHz(4x) audio. The audio data frame doubles in size with each multiplier, which consequently halves the number of audio channels with each multiplier. This is the same as the 'Dual-Wire' and 'Quad-Wire' configurations used to carry AES audio data back in the '90's, when digital I/O chipsets could only handle 44.1k and 48k sample rates. **Base** is generally the place to start for 44.1kHz and 48kHz, and may be required for decoding "SMUX'd" high sample rate streams.

Native is (again, generally) used at high sample rates and frames the audio at the 'actual' or 'native' rate, so 'Native' 96kHz sample rate is actually read as 96kHz and so on.

At 1x rates, **Base** and **Native** are the same.

MADI frame/channel formats

'**Legacy**' is the older 56 channel frames format, where at 1x sample rates you get 56 channels, 2x you get 28 channels and 4x gets you 14 channels.

'**Long**' is the more recent 64 channel frames format, where at 1x sample rates you get 64 channels, at 2x you get 32 channels and 4x gets you 16 channels.

Some manufacturers may exclusively use the 'Legacy' 56-channel format, some only use 'Long'. With newer gear, try 'Long' first.

In practice, you always want to use Word Clock to sync MADI transfers. Word Clock is a requirement in the MADI spec. But even with great word clock, and especially when dealing with gear from the '90's and '00's, clean MADI lock can sometimes be a bit of trial and error. One piece of gear may not handle 'Native' frame rates at 4x, so you try 'Base', and hey, if it works it works, don't complain. Another prominent manufacturer may not support 'Base' frames or SMUX at all. The key here is compatibility, and between the 'Base/Native' and 'Legacy/Long' formats controls you should be able to communicate perfectly with any MADI gear you come across.

To that end, 3d MADI has been tested with some pretty cantankerous old beasts. Below is a first-generation ULN-2, updated to 3d with a copper/optical MADI EdgeCard™, slaved to the MADI stream of a stock Sony 3348HR at 44.1kHz. Solid as a rock. Of course, for the real transfers the 3348HR was slaved to the 3d word clock.

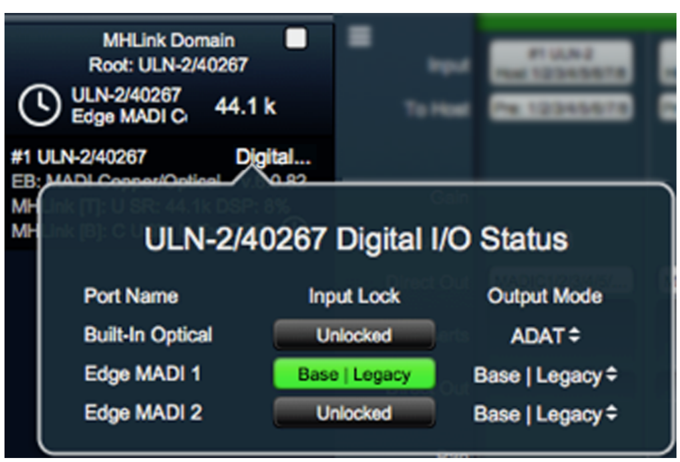


Figure 10.88: Test session: first-generation ULN-2 with 3d MADI clocked and locked to Sony 3348HR

Figure 10.89: Test session: 3d MADI and Sony 3348HR

Unit Sample Rate Display

The **"Unit Sample Rate Display"** shows the internal sample rate of the primary DSP processing engine within each box. This field is reserved for future 3d platform developments, and for the time being will always mirror the Domain master sample rate.

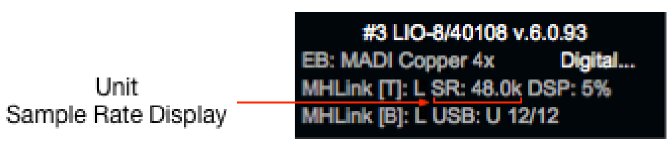


Figure 10.90: Unit Sample Rate Display

DSP Load Status

The **"DSP Load Status"** monitors how much DSP processing is happening in each box. 'Load Status' readings above 98% may result in some audible digital artifacts during playback. This can be immediately cleared by removing the last-inserted DSP plug-in. When using a multi-box Domain, you can enable **"Spread Host Channels across boxes"** under the menu bar "Mixer" menu to distribute the DSP load between the available boxes and maximize the available processing power.

Figure 10.91: DSP Load Status

USB Port Status

Every Metric Halo 3d unit includes a USB-C port on the rear panel, right next to the MHLink ports.

The USB port on every 3d device includes the UAC2 Class Audio driver in hardware, and serves three important functions:

1. An alternative to MHLink for connecting a 3d device to Mac, Windows 10 or Linux host computers and iOS/Android mobile devices. Click here for details.

2. Low-latency MIDI transport between the ULN-8, LIO-8 and SPDIF•MIDI EdgeCard MIDI ports and computer-based MIDI applications.

3. **"Satellite Computer Port"** audio connection between any 3d unit within an MHLink domain and a "satellite" computer or iOS/Android device.

The USB Port Status Display

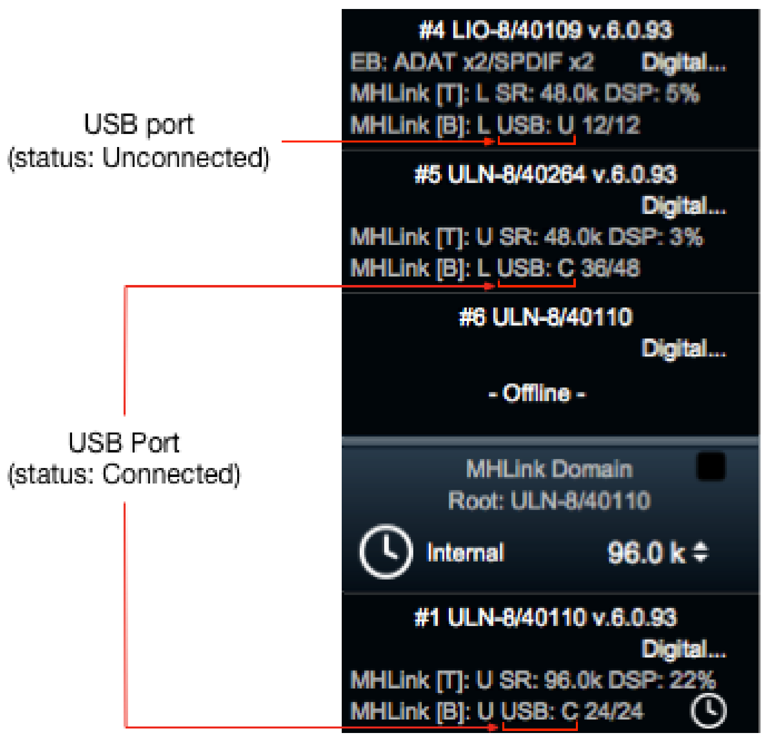


Figure 10.92: USB Port Status

- **"USB: U"** In the example above, the USB port on box #4 is **U***nconnected*. "U" on a USB port can mean one of three things: either there is no USB cable attached to that port, the computer, smartphone or tablet to which it is attached is turned off, or the USB cable that is attached is not compatible (see sidebar below).

- The **"USB: C"** status on box #5 means that USB is **C***onnected* and **C***ommunicating* as a "Satellite Computer Port". The Metric Halo 3d "Satellite Computer Port" feature allows an external USB device to send and receive a maximum of 48 audio channels directly with an MHLink domain.

- Note that the **"USB: C"** status on the solo box in the second Domain means that USB is **C**onnected and **C**ommunicating *not* as a "Satellite Computer Port", but as a "USB Host Computer" connection. We know this because this unit is identified as the Root box of its own Domain, and both MHLink ports are **Unconnected**.

USB I/O Configuration

You can configure the USB port by "right-clicking" or "control-clicking" the "Unit Status Display" box. Select "Set Number of Channels for USB…" as shown below.

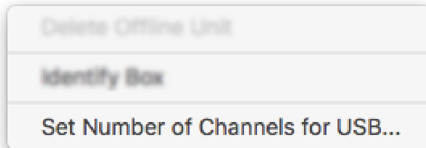


Figure 10.93: "Set Number of Channels for USB…" selector

This will open the "USB I/O Configuration" window. The box model and serial number are displayed in the header at the top. Below the header are pull-down menus from which to independently set the number of **Input** (From Computer) and **Output** (To Computer) channels available at that USB port. This allows you to trade off USB channel count for higher sample rates. "Maximum USB S/R:" will reflect the highest supported sample rate based on your channel count selections:

- 4-12 channels works up to 4x rates (44.1k - 192k)
- 4-24 channels works up to 2x rates (44.1k - 96k)
- 4-48 channels works only for 1x sample rates (44.1k - 48k)

Figure 10.94: "USB I/O Configuration" windows

USB I/O port channel configurations remain persistent to that unit even after it is powered down. This lets you pre-configure a 3d box for use on a PC or Linux box. Just remember, as noted above, the sample rates available to the new computer will be limited by the number of USB I/O port channels you set.

Important! *In all cases*, the MIOConsole3d application maintains control over all 3d units which appear in the Status Pane, and the 3d unit USB ports are slaved to the Domain sample rate and clock. Changing USB I/O channel allocations or Domain sample rate will trigger a reset of the USB port, and may (probably) require re-assigning your audio driver connection on the external "Satellite Computer Port" device.

Connecting an Apple tablet or smartphone to 3d USB requires a 'Lightning to USB Camera Adapter' or it's equivalent. Android users need a "USB On-The-Go"-compatible cable or adapter. Both of these solutions enable 'USB Embedded Host Mode' in the phone or tablet, which is necessary to communicate with the 3d USB port. These adapters are commonly used to connect digital cameras, USB flash drives, mice and keyboards to your Android or Apple portable device.

Please note that due to the immense and ever-expanding landscape of computers, smartphones and tablets which include some form of (often poorly documented) USB interface, the vast majority of configurations have not received substantial testing.

The MIOConsole3d Mixer Grand Tour

The MIOConsole3d Mixer is where it all comes together.

Something to keep in mind is that the 3d mixer model is based solely upon user-configuration. There are no assumptions made regarding signal flow, channel formats, monitoring or input devices. The advantage to this is that you can build the exact mixer you need for any specific task or set of circumstances.

Some will use the 3d mixer as nothing more than a routing interface. Others use 3d for multi-tiered FOH installations running everything from stage mics and multiple IEM cue mixes, feeds for acoustically-corrected and time-arrayed speaker systems and high-resolution performance capture, supplying word clock and secondary stems to audio to video and broadcast systems… simultaneously, through a combination of MHLink, analog and digital I/O interconnections. There are some crazy rigs out there.

In this section we begin where everyone begins: the default 'start-from-scratch' mixer state. From there we dive into customizing the Mixer desk layout, defining the types of mixer strips, mixer strip controls, routing and signal processing best practices, UI shortcuts and maybe a few surprises along the way.

First though, hit the "[" key to hide the System Status Pane, so we can concentrate on just the Mixer desk itself.

Mixer Default Configurations

Even though the 3d Mixer makes no assumptions and lets you configure as you please, there is a default setup when starting from scratch. Not only does this give you a little jump start to getting a mixer built, but it lets you know that the new box has been properly synced and incorporated into a 3d Domain, ready to get to work.

Default Setup for New Domains

When you connect a new 3d box to your computer, MIOConsole3d syncs and registers the new box by model and serial number, checks the installed firmware, and sets up a default configuration for that box in the Mixer. The default configuration includes Input routes from the Host computer and the analog inputs of all attached 3d boxes, with outputs to the Monitor Controller and/or Cue controller (depending on the Root 3d box model).

Default configuration settings are as follows:

At the Domain level:

- Sample rate: 48kHz
- System Status Pane: visible
- **Mixer Desk Configuration:**
  Settings in this section are found in **"Configure Mixer"** [cmd-shift-C] and the Menu bar: "Mixer" menu
  - Main mix bus: Stereo, unmuted, is the default Monitor Controller Source, is the default 'Control Room Headphone' Cue controller source
  - Solo Mode: Solo-In-Place
  - Hard Mutes: disabled (and hidden from the Mixer strips)
  - Auto-route new strips to Host: enabled
  - Input strip meters: show pre-fader levels
  - Four Aux mix buses: Stereo, unmuted, no inputs assigned, each is a Monitor Controller Source
    - Input strip assign: post-insert

    - Input strip Fader: pre-fader
    - Visibility: on-strip
    - Monitor Control: In MC
  - Group buses: 0
  - DCAs: 0
  - Mute Groups: 0
  - One stereo "Host" Input strip: unmuted, input routed from Host computer channels 1/2, assigned to Main mix bus
  - One mono Input strip per analog input: Input head amp set to 'Line +4' at unity gain, includes Pre-Insert "To Host" returns, is assigned to the Main mix bus, is panned to center, and is muted for safety at the Fader stage.
  - 'Selected Strips' Link Group: enabled, **Rel**ative behavior
  - 'Mixer Strips Width': Normal
  - 'Mixer UI Scale': 100%
- **Monitor Controller:**
  - Monitor Controller is visible, not floating
  - Monitor Controller Sources: Main (default) and stereo Aux mixes 1 through 4
  - Monitor Controller Graph: passthrough
  - Monitor Controller Output: stereo, assigned to box #1 analog output channels 1/2, Monitor mode (LIO and ULN-8 only), Trims and Delays at 0
  - Monitor Controller Slaves to PFL/AFL: ON

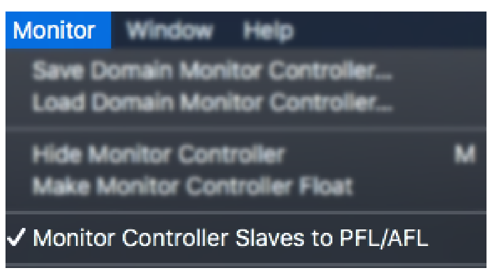


Figure 10.95: Menu bar "Monitor" menu: "Monitor Controller Slaves to PFL/AFL" control

People who spend most of their time in DAWs working in 'Solo-In-Place' mode should take note of this feature. This setting is found in the 'Monitor' menu bar pull-down.

When the Mixer is in one of the PFL/AFL modes, this setting makes the Monitor Controller play back the 'Solo' bus whenever Solo is engaged. This is the most common default mode for PFL/AFL use on a hardware console.

De-selecting this control leaves the MC Source uninterrupted when Solo is engaged.

This function is also available separately for each of the Cue Controller sends, so you could have, say, a Cue send to headphones slaved to the PFL/AFL Solo bus, while not interrupting the control room 'Main' bus mix for a producer.

- **Cue Controller:**
  - Cue Controller is visible
  - Cue Controller Source: stereo, 'MC Source' (Cue source follows Monitor Controller source selection)
  - Talkback Source: unassigned (which is why the "Talkback" trigger button is grayed out)
  - Cue Controller Graph: passthrough
  - Cue Controller Output: stereo, assigned to box #1 headphone output L/R

  Note: Since ULN-2s have dedicated a stereo Monitor route which goes to both the front panel headphone jack and stereo TRS monitor outs on the back panel, only a default Monitor Controller output is established. The default Cue Controller for a new ULN-2 root box is unconfigured.

Mixer Desk Default configuration: When you add a new box

If the new box is an addition to an existing Domain, a mono analog Input strip will be created for each analog input channel on that box (eight strips for a 2882, LIO-8 or ULN-8, two strips for a ULN-2). New strips created by adding a new box will always appear at the far right of the existing Mixer desk.

These new Input strips will include Pre-Insert “To Host” returns and will be routed to the current Main mix bus. Default Input strips are always mono, panned to center and are muted for safety at the Fader stage.

Console3d Mix Desk Organization

Mixer Panes and the Hamburger menus

At the top left corner of the Mixer window is a weird little icon called a Hamburger Menu.

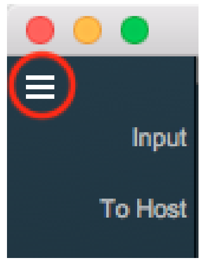


Figure 10.96: Mixer Panes and the Hamburger Menus

There are two more hamburger menu icons across the top of the mixer desk in the upper right corner. Each of these three Hamburger Menus are used to configure one of the three Mixer Panes provided to help organize your mixer. In the default view, only Mix Pane 1 is set up and visible - Mix Panes 2 and 3 are currently un-configured and are accessible by the two Hamburger Menus at the top right of the Mixer window.

Each Mix Pane is re-sizable by dragging the left edge of the Pane, and can show any combination of Input strips, Bus strips (Main, Aux and Group) or DCA strips. Mixer strip layouts (i.e.: your choice of which Input and Bus strip types are shown) are configured independently for each Mixer Pane.

Each Mix Pane has its own independent scroll bars, and will scroll automatically using standard scroll wheels and scrolling gestures when the mouse cursor is above the pane. Similarly, opt-scroll above a Mixer control element will operate that control, and opt-click on a control will return the control to its default setting.

You can have a different set of Mixer Strip controls visible per Pane as well. This lets you set up the strips in one Mix Pane for tracking, another pane for mixing with a third just for masters and DCAs.

Click on this hamburger menu to open the "Mixer Pane Configuration" menu.

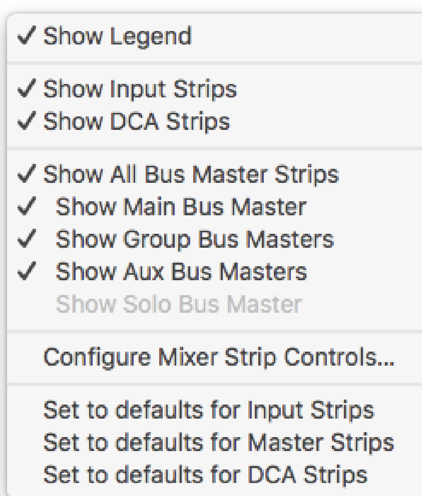


Figure 10.97: Mixer Pane Hamburger Menu

> Note that the Mix Pane 2 and 3 hamburger menus also have "Show Legends" enabled, but no legend is shown on those panes. If the Mixer Pane is set to show no Input, Bus or DCA strips, the Legend will be automatically hidden and the empty Mixer Pane will collapse to the right.

- The top entry in this menu, **"Show Legend"**, lets you show or hide the strip control labels running down the left side of the mix desk. Usually this is left enabled so you have an easy visual reference to the current controls, but if you are really pinched for desk space, turning the Legend off will buy you another strip width or so.
- **"Show Input Strips"** will show or hide the Input strips in the Mix Pane view. Input strips are easily distinguished from Master Bus and DCA strips by the presence of "Input Source" selectors at the top of the strip, Aux mix bus sends in the middle of the strip, and the lack of a "Bus Output" selector and the bottom of the strip.
- **"Show DCA Strips"** shows/hides DCA strips in the current Mixer Pane. DCAs in the 3d Mixer are essentially a specialized fader strip controlling Solo, Mute and output gain parameters across its assigned Input or Bus strips. DCA strips are the easiest to identify in a busy desk, as they have no audio routing, processing or meters, just Solo, Mute, Fader and Link Group controls.
- **"Show All Master Bus Strips"**, when selected, shows all types of bus strips in the current Mixer Pane (including the Solo bus if a PFL/AFL Solo mode is engaged). A common practice is to place the Master strips in a different Mixer pane so they can always be kept in view while you scroll across your vast array of Input strips in another pane.

  If you de-select "Show All Master Bus Strips", you can then show/hide each type of bus individually. When selecting "Show All Master Bus Strips", the Main, Group and Aux Bus Master selectors will all be checked, but your previous selections are remembered.

  - **"Show Main Bus Master"** shows/hides only the Main master bus strip.
  - **"Show Group Bus Masters"** shows or hides the Group sub-master bus strips.
  - **"Show Aux Bus Masters"** toggles visibility of the Aux mix bus strips.
  - **"Show Solo Bus Masters"** will be available when any of the PFL/AFL modes are enabled in "Configure Mixer" (as shown below). When the Solo bus is available, "Show Solo Bus Masters" will show or hide it.

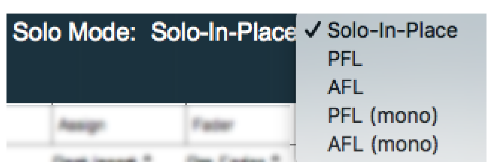


Figure 10.98: "Configure Mixer": Solo Mode selector

- **"Configure Mixer Strips Controls"** opens a window where you can choose what mixer strip control elements are shown on the Mixer strips and Legend. In the default configuration, all mixer controls are shown except for "Hard Mute", which is disabled by default in "Configure Mixer". "Configure Channel Strips Controls" are covered in depth in the next section.

- The following menu items switch the current Mixer controls to match the maximum available controls for each type of Mixer strip.
  - **"Set to defaults for Input Strips"**: Input Strips inherently have the most control elements, including source input routing, head-amp control of hardware inputs, routing to all three master bus types (Aux, Group and Main), pre- and post-Insert direct outs plus the fader block and parameter linking controls. The Input strip controls set is the default view for new Mixers.
  - **"Set to defaults for Master Strips"**: Master bus strips have no need for hardware Source controls, head amp controls, direct outs or Aux bus sends, so all these controls are hidden. Pan is included for mono Aux and Group buses to feed a stereo or multichannel Main mix bus.
  - **"Set to defaults for DCA Strips"**: DCA strips are just control elements themselves, so all audio routing and processing controls are hidden, leaving only Solo, Mute, Fader and Link Group, with Meters and Bus Assign controls left available for any Insert and Bus strips in the same Pane.

Mix Desk Command & Control

Here are some tips to help you move around and control the Console3d mixer environment.

- **Console3d UI: Parameter "Nudge"**

  Any numeric control element in the Console3d user interface (faders, panners, EQ frequency, etc.) can be "Nudged" to increment or decrement the parameter value. This command is available throughout the MIOConsole3d environment.

  The amount of "nudge" varies by the type of parameter. For example, Fader Gain 'coarse' nudges are 5dB steps per nudge and 'fine' nudges are 0.5dB.

  Pressing and holding the key command will repeat the nudge command per the Keyboard: "Key Repeat" settings in your computer operating system preferences.

  Default Nudge keystrokes are:
  - ^⌥Z (Control + Option +Z) to coarse nudge up
  - ^⌥X (Control + Option +X) to coarse nudge down
  - ⌥Z (Option + Z) to fine nudge up
  - ⌥X (Option + X) to fine nudge down

- **Scroll wheel increment / decrement**

  Mouse scroll wheel and trackpad scrolling gestures are supported as an alternative to "Nudging". Depending on your scroll wheel/trackpad sensitivity settings, scrolling can be a very helpful complement to nudging.

  By default, moving the scrollwheel while over the mixer desk surface will scroll the fader strips to the left or right. However, when holding the **Opt** (on Mac) or **Alt** (on PC) modifier key, the scrollwheel will move a fader or other control element.

  Generally the default settings strike a nice balance, but depending on your particular setup, you may wish to customize scrolling, Option/Alt-click and double-click behaviors.

  See MIOConsole3d Preferences: Interaction to modify these behaviors.

- **"Sweeping" controls**

  Toggle buttons on consecutive strips in the 3d Mixer desk can be switched in a single move by a click-hold-sweep gesture.

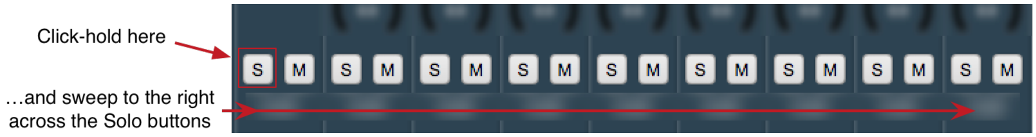


Figure 10.99: "Sweep" to toggle multiple buttons in one gesture

  To try it, click on a Solo button, and while holding the mouse button down, drag the cursor to the right or left across the strips.

  The move works with Polarity Invert, Solo, Mute and Record Enable buttons.

- Mixer strip Insert controls

Once you have selected a plug-in, it will be listed in the assigned insert slot:

Figure 10.100: Inserted MIOEQ6 Plug-in (as shown in Mixer strip)

Plug-in names will generally appear abbreviated in order to save space ("MIOEQ6" is shown above).

When you move the mouse over an inserted plug-in, the Insert label will change to show three control icons. The tooltips for each of these controls have been exposed in the example graphic below.

Figure 10.101: Inserted plug-in controls

- The "On / Off" switch icon on the left is the plug-in Bypass. When Bypassed, the Insert button will turn yellow.
- Clicking the "…" icon in the middle opens the inserted plug-in editor UI. When the plug-in editor is open/visible, the Insert button will turn blue.
- Clicking the "up/down" arrows icon at the right opens the Insert selector window, where you may select a replacement plug-in, or navigate to directly open a different saved preset without having to open the plug-in editor UI.

Insert Control modifier key shortcuts

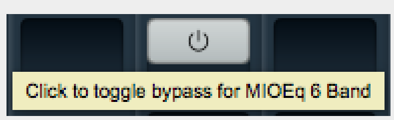


Figure 10.102: "⌘-click" / <Command>-click to Bypass Insert

<Command>-click the Insert button to Bypass the Insert.

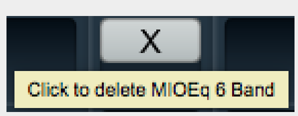


Figure 10.103: <Control-Option-Command>-click to Delete Insert

Use "^⌥⌘-click" / <Control-Option-Command>-click to delete an Insert directly from the Mixer desk.

- **Quick Copy/Paste Plug-ins**

  Option-click-drag any plug-in instance from one Insert slot to another anywhere in the Mixer to clone that Plug-in to the new location. Plug-in instances will automatically adapt to the channel width of the target Insert as necessary.

  NOTE: Due to the hardware-routing nature of I/O Inserts, copying routed I/O Inserts is not supported. To avoid potential routing conflicts, in all cases I/O Inserts are instantiated as unrouted.

Routing windows: [Unrouted] and [Offline]

One feature often requested by advanced users over the years has been the ability to create and/or modify Console configurations with the boxes offline, such as on a plane between shows or tweaking configuration templates on off hours.

To support this, the routing windows in MIOConsole3d always list the maximum possible routes supportable by the most recent hardware configuration or the hardware configuration saved in a MIOConsoe3d '.cnsl3d' state file, even when the boxes are not connected. All these routes may be routed and saved, and when the 3d hardware is attached they become active.

This also allows you to work offline without the full complement of boxes, maintaining the full system configuration for the next set or session.

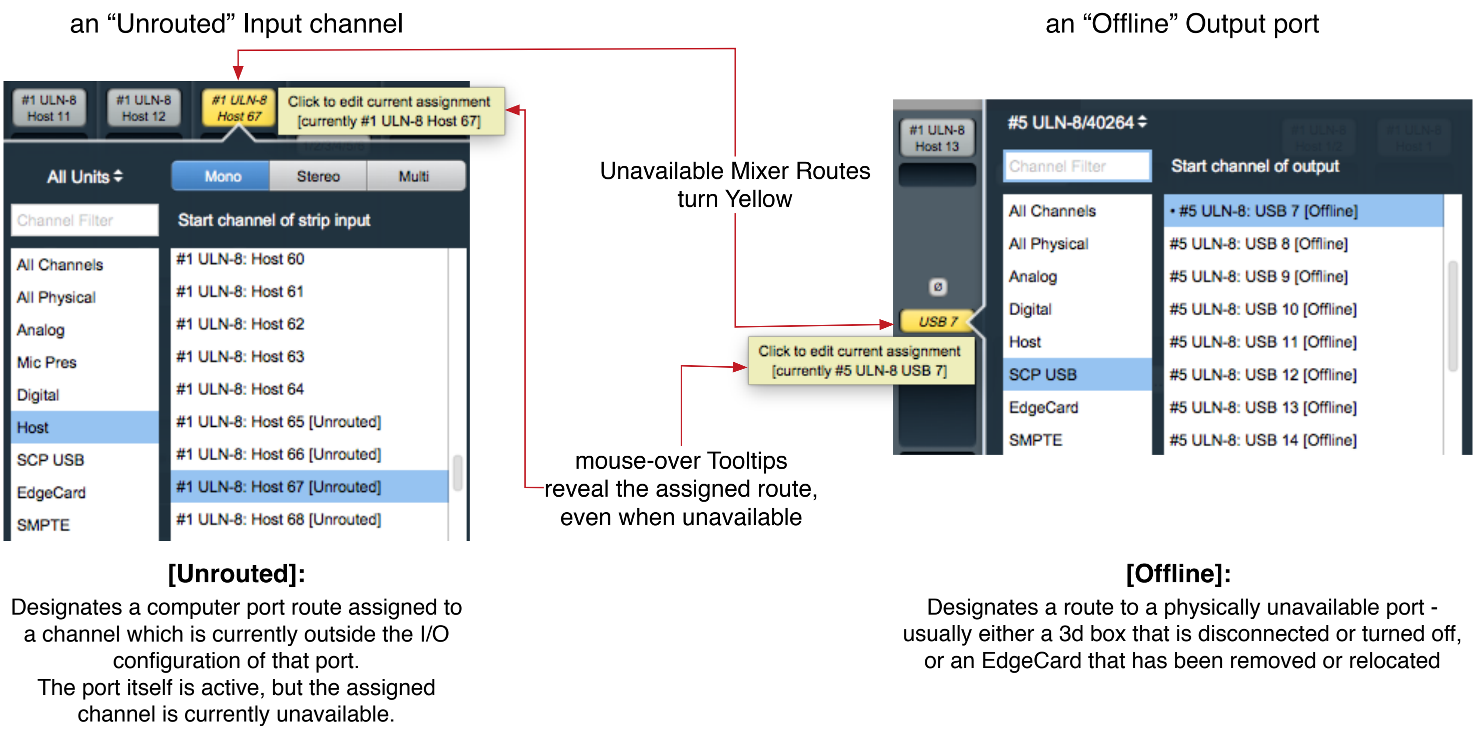


Figure 10.104: Unavailable Routes in the Routing windows

The graphic above illustrates the two types of "Unavailable" routes and their context within the MIOConsole3d UI.

"Unavailable" routes will show in Yellow routing button with italics text. In all cases, mouse-over Tooltips are available to show the entire routing path.

- Routing assignments listed as **[Unrouted]** reflect channels on an active port, but which fall outside the current I/O configuration. In the case above left, the MHLink I/O menu is set to 64 channels of I/O, so the route from Host Input channel #67 is out of range. Switching the I/O to 128 channels would immediately enable and connect the channel #67 route.

  Another example of [Unrouted] would be an Direct Out send to channel 18 of a USB port currently set to 24 channels In and 12 channels Out. In this case, Input channel 18 would be available, but the USB port Output setting would need to be changed to 20 channels or higher to enable USB output channel 18.

- **[Offline]** routes indicate a routing path that is physically not attached to the current Domain. In the case above right, we have a Direct Out again routed to a USB port, but the route is listed as [Offline] because that box is literally offline - either turned off or disconnected from the MHLink domain.

Another example of an [Offline] route would be a set of connections to a MADI EdgeCard which is no longer installed.

Launching MIOConsole3d with some hardware offline will generally initiate the Box Remapping page, where, if you wish, you can re-assign any existing routes to an appropriate I/O port in your currently-attached domain. This new Console configuration will have the string "[mapped]" added to the Console filename to avoid overwriting the original configuration.

Mixer Strip Controls Breakdown

The **"Configure Mixer Strips Controls"** menus for each pane let you show or hide any control element in the Mixer. The selected view is independent for each of the three Mixer Panes, allowing you to really dial in your Mixer setup to your particular workflow.

The "Configure Mixer Strips Controls" window can be opened either from the Mix Pane Hamburger Menu, or by right-clicking the Legend strip of the Mix Pane you want to tweak.

Note that hiding a control does not remove any settings you have applied, it just hides the control from the desk. Any outboard control surface mappings will still function as always.

The graphic below shows the "Configure Mixer Strips Controls" menu on the left, with all Mixer strip control elements exposed and mapped to their corresponding controls on the Legend and Mixer strips. The check boxes in "Configure Mixer Strips Controls" show or hide each control element from the Mixer strips. The remaining Mixer strip controls will dynamically adjust to fill empty spaces, or make room as necessary.

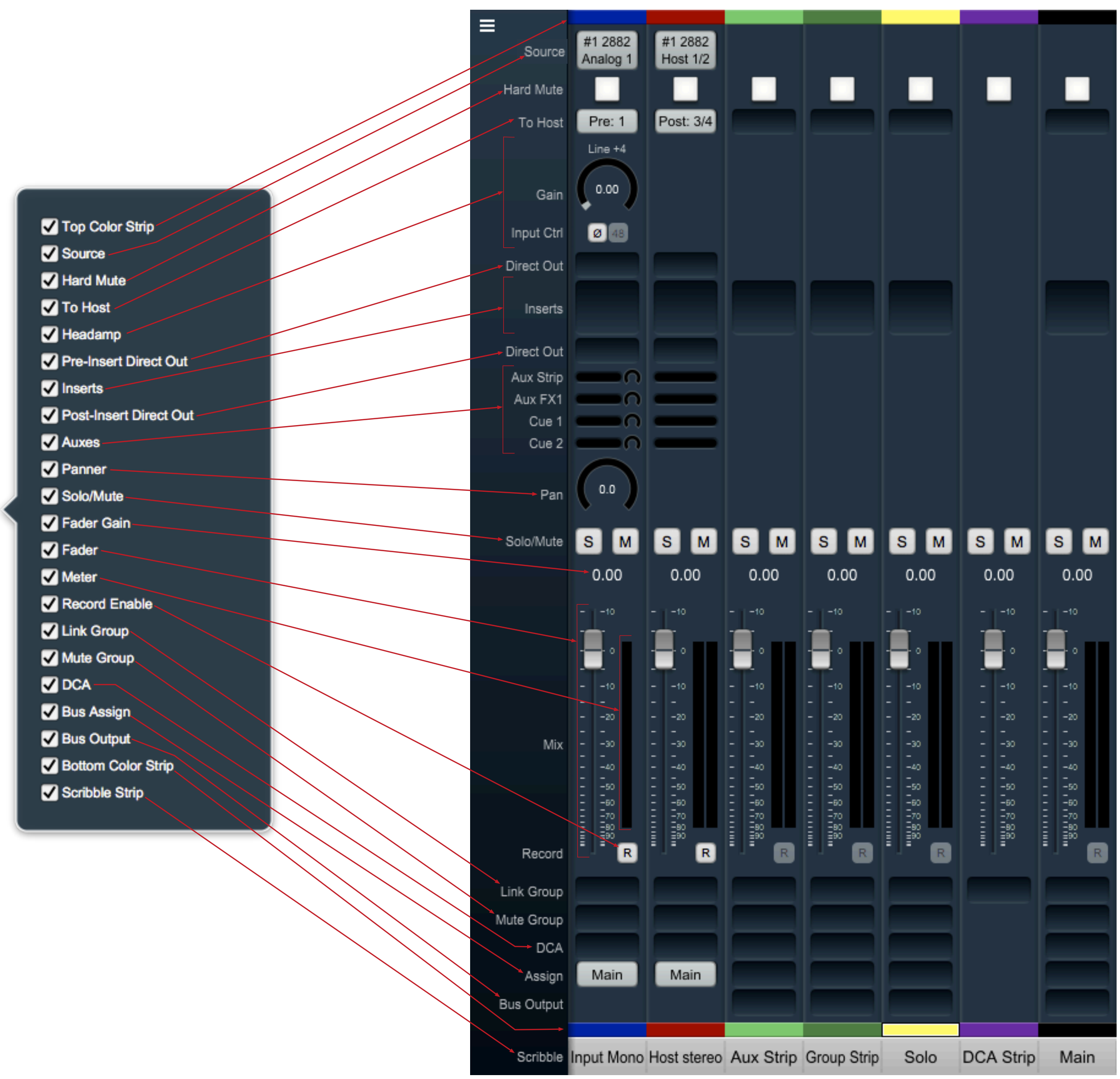


Figure 10.105: "Configure Mixer Strips Controls": Legend and Mixer Strips map

Mixer Strip: Color Bars, 'Source' selector, 'Hard Mute', 'To Host' returns, 'Headamp'

Now that we have a map, let's get down to it. From top to bottom, the Grand Tour begins…

Figure 10.106: Top Color Bar, Input Source, Hard Mute, 'To Host' Return and Headamp

Top Color Bar

Clicking on the **Top Color Bar** brings up a macOS color selector, from which you may choose a color for that Mixer strip. In the Appearance pane of the MIOConsole3d Preferences you may select whether to color the entire mixer strip or just the top and bottom color bars.

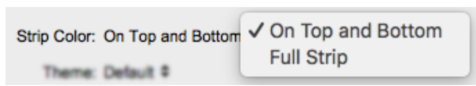


Figure 10.107: Console3d Prefs: Appearance: Strip Color configurator

Top and Bottom Color Bars mirror each other, so what applies to one applies to the other, except that you may show or hide each independently with "Configure Mixer Strip Controls…".

Source

This control is available only on Input strips. When you select **Source**, it opens the input source routing menu (shown below). Input strips can source audio streams from practically anywhere in the Mixer - all analog and digital hardware inputs, bus outputs, computer connections over MHLink ethernet and SCP USB, even the ULN-8 and LIO-8 SMPTE inputs are fair game.

While the primary function of this control is to choose a new source input, this window allows you to modify the current strip to Mono, Stereo or Multichannel. Additionally, you can re-purpose multiple existing strips simultaneously using Cascade. This last trick is most useful for re-assigning groups of inputs from one port in a domain (say an 8-channel AES in) to another (like an 8-channel ADAT in) without altering your existing mixer configuration.

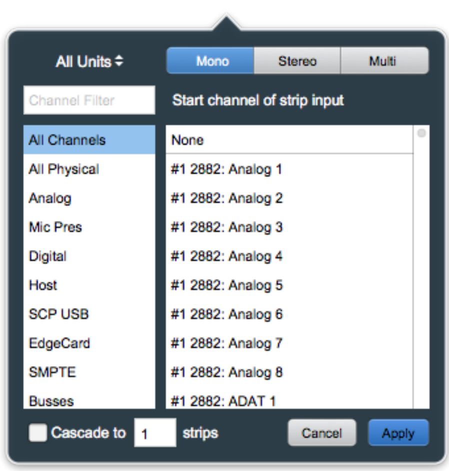


Figure 10.108: Input Strip Source selection window

Since any MHLink domain consists of hundreds of available input sources to choose from, all routing selection boxes allow you to filter the visible choices by both the individual box and by the type of source you wish to target.

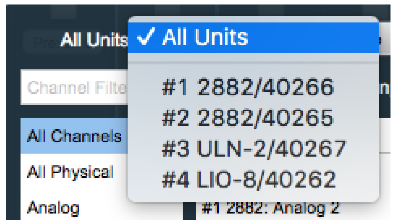


Figure 10.109: "View sources by box"

Clicking on "All Units" opens a drop menu where you can narrow your selection view to show only the routable sources from a particular box on your MHLink daisy-chain.

Beneath the "Unit"/box selector is a search field where you can enter the name of the type of input you are looking for. This is great for searching particular types of digital inputs. Text entry search is dynamic, so type in 'TO' to show only TOSLINK, 'MA' for MADI, etc. and the results pop up immediately as you type.

The categories in the list on the left let you refine your view as follows:

Category	
All Channels	shows all input sources
All Physical	shows all analog and digital audio ports (AES, ADAT, TOSLINK, SPDIF and MADI)
Analog	shows only analog inputs
Mic Pres	shows only analog inputs with internal mic preamps (ie. no LIO analog ins)
Digital	shows only physical digital ports (AES, ADAT, TOSLINK, SPDIF and MADI)
Host	shows the routable sources from your Host computer
SCP USB	shows the routable sources from non-Host USB ports
EdgeCard	lists only inputs from EdgeCards
SMPTE	lists only the dedicated SMPTE inputs on ULN8 and LIO8 units
Buses	shows the currently available Aux and Group buses in the current Mixer

Table 10.2. 3d Routing interface filter categories

Hard Mute

When engaged, **Hard Mute** kills a hardware input source just like you were pulling the input cable, only without the huge disconnection noise. It mutes 'To Host' sends, all Direct Outs, all Aux sends, Insert processes, Bus Outs… all signal through the strip is stopped.

Figure 10.110: Configure Mixer window: Hard Mutes enable

This control is disabled by default and appears grayed-out in the "Configure Mixer Strips Controls" window. It must be enabled manually in "Configure Mixer" [cmd-shift-C].

Enabling the "Hard Mutes" control places a big high-visibility square button at the top of every Input and Bus strip in the mixer, above "To Host". The button is white when passing signal, and turns bright blue hen engaged

To Host

To Host lets you route pre- and/or post-process returns directly to your Host computer. Input strip "To Host" selector options are "None", "Pre-Insert", "Post-Insert" and "Both" - where you are returning both an unprocessed signal and signal processed through the Insert processing path.

Note that Mixer Bus "To Host" returns are "Post-Fader" only.

"To Host" is specifically optimized for fast setup of large mixer layouts, and will automatically assign the next available Host return channel. So, if you already have 12 mono input strips returning to the Host computer on channels 1-12 and you add Host returns on a stereo strip, those Host returns will be assigned to channels 13 and 14.

Additionally, when MIOConsole3d auto-assigns or cascade-assigns routes, it will always maintain stereo pairs - in other words, stereo pairs will always use odd numbers for the left channel and even numbers for the right channel. This logic also applies to multichannel surround groupings: left channels will be odd, right channels will be even (unless of course the channel layout of the selected multichannel format dictates otherwise). Multichannel sources and outputs will always be assigned in groups of consecutive channel numbers.

For sending specific channel routes back to the Host computer, use a Direct Out from an Input Strip, or "Bus Output" from a Bus Strip

Important! - "To Host" and "Direct Out" returns to the Host are how you get audio back into your Host computer for use in your DAW and for recording in the Console3d Record Panel. You must have Host routes assigned either in "To Host", a "Direct Out" or a "Bus Output" to activate Record Enable on any Mixer strip.

Headamp

The **Headamp** section adapts to the type of audio signal selected in the Input strip "Source". For digital sources such as Host, USB and hardware digital audio streams, the headamp will only show the "Polarity Invert" button.

The analog Headamp has controls for signal Input Mode (level standard, impedance, etc.), gain/trim, phantom power toggling and polarity inversion.

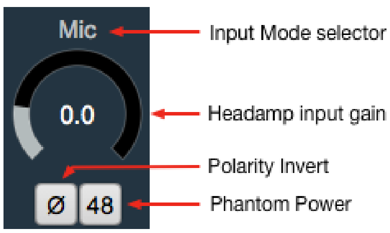


Figure 10.111: Analog Headamp controls

When an analog input source is selected, the headamp section expands to include controls specific to the particular MIO model routed by the Input strip.

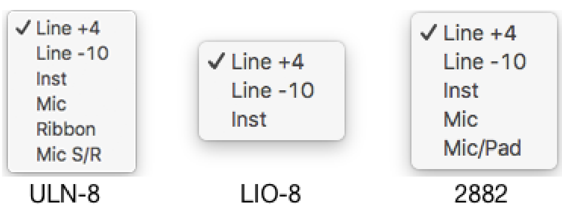


Figure 10.112: Analog Headamp Input Modes by model

- The ULN-8 offers settings for **Line +4** (10kΩ), **Line -10** (10kΩ), **Instrument** (200kΩ), **Mic** (3.3kΩ, 12kΩ with +48v phantom engaged), **Ribbon** and the **Mic S/R** send/return loop.

  'Mic Send/Return' physically routes the mic pre outputs to the 'Balanced Send 1-8' db25 connector on the back panel of the ULN-8 for external analog processing before the a/d converters.

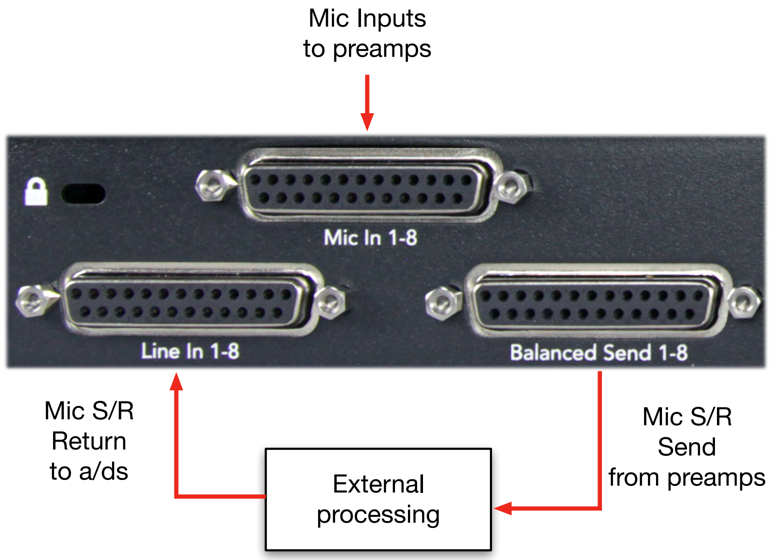


Figure 10.113: ULN-8 'Mic S/R' external process loop

  This graphic shows the signal path for the Mic S/R external processing loop. Note that the 'Balanced Send 1-8' mirrors whatever analog input you select in the Mixer Input strips, Mic, Line or any combination thereof. This allows you to use the sends as a splitter to feed a mixing console, recorder or other equipment.

- The LIO-8 offers **Line +4** (10kΩ), **Line -10** (10kΩ) and **Instrument** (200kΩ) settings. Any channels with ULN-R mic pre-amp modules include all ULN-8 input options as listed above, including Mic S/R.

- The 2882 includes **Line +4** (10kΩ), **Line -10** (10kΩ), **Instrument** (200kΩ), **Mic** (200kΩ, 12kΩ with phantom engaged) and **Mic/Pad** (10kΩ, 12kΩ with phantom engaged), for high-spl situations like condenser mics on kicks.

- The ULN-2 has no Headamp option here because it has a fully analog front end with all selection and input gain/trim controls on its front panel.

> **"Polarity Invert"** is available on all Input strips regardless of source type.
>
> Polarity Inversion is applied early in the signal path, before the Pre-Insert "To Host" routing stage. For cases where Polarity invert at such and early stage is not desired, a "Polarity Invert" plug-in is available for Input strip and Mixer Bus inversion insertion.

Mixer Strip: 'Pre- and Post-Insert Direct Outs', 'Insert' processor slots, 'Aux Sends'

This section is concerned primarily with routing options from the Mixer Strip and processing within the Mixer Strip.

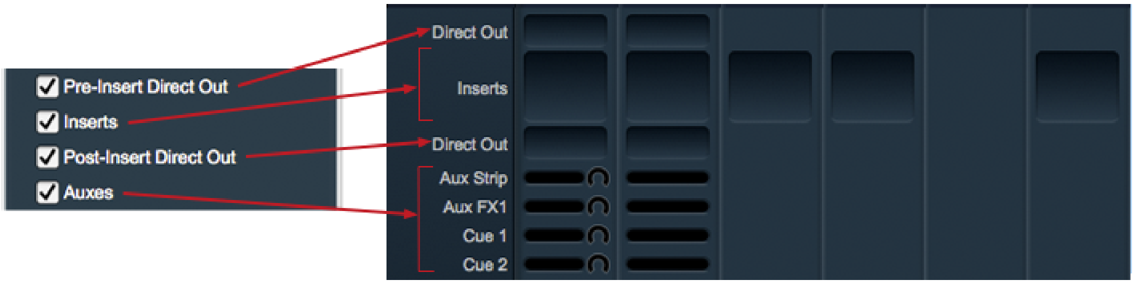


Figure 10.114: Pre-Insert Direct Out, Inserts, Post-Insert Direct Out, and Aux sends

Direct Outs

Both Pre-Insert and Post-Insert Direct Outs are available on all Input strips, placed above and below the "Inserts" section per their locations in the signal path. Direct Outs can send to any available output port in the Domain - hardware, Host, or SCP.

The **Pre-Insert Direct Out** sends signal from the mixer strip prior to Insert processing.

Inserts

DSP processing **Inserts** are available on all Input and Bus strips.

Clicking in the Inserts section opens a dialog box that lists categories for "Plugin", "Macro", "Graph" and "I/O", with "None" listed at the top. When you open an existing Insert plug-in, clicking "None" will delete it.

Selecting one of the listed categories will reveal the nested contents of that category. Selecting a nested item will in turn reveal a list of 'saved preset' categories pertaining to that item, whether it be a processor, macro or Graph. Selecting a preset category will in turn reveal all the saved presets within that category, as shown in the graphic below.
Shortcut! Double-clicking a plug-in or preset name will immediately instantiate that plug-in (at it's default setting) or the plug-in with the preset settings.

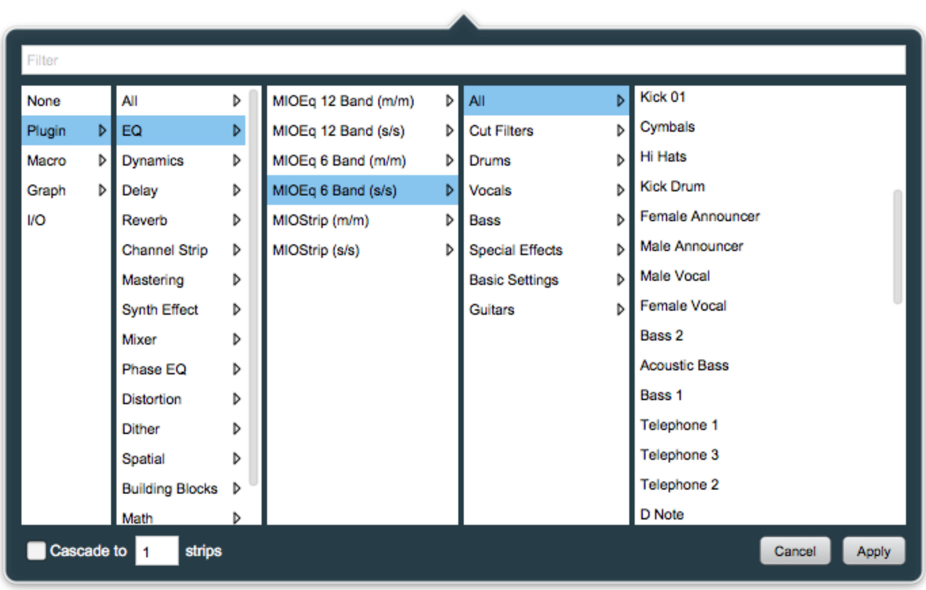


Figure 10.115: Insert selection categories with nested 'Saved Presets' hierarchy

- Plugin: All the basic stand-alone processors, including EQs, dynamics, reverb, distortion (like the 3d-native MH Production Bundle and Sonic EQ 3d) and some extremely useful processor “building blocks“ (M/S processors, summing matrices, delays, etc.).
- Macro: Most macros are essentially presets of more complex process chains in a DSP Graph. Feel free to use these as a starting point to create your own personal custom signal processors. Some macros (such as the Reverbs) can not be represented as modules in a normal graph, and can not be modified as-is.

> Note that mono and stereo versions of Plugins and Macros are available only on mono and stereo insert strips, respectively. The ”2d Amps”, ”2d Pedals”, and ”Cabinets” macro categories are only available in mono mixer strips, as these processors have only one input and one output. Likewise, the ”2d Reverbs” category is only available on stereo strips as they are designed specifically for stereo source content.
>
> Macros and stereo plugins are not currently available for multichannel strip insertion, but the mono plugins are available within the Graph context to create your own custom multichannel processors.

- Graph: A graph is a “DSP playground“ where you can build your own signal processing chains and save them for later recall. The MIOConsole3d Graph is arguably the most powerful audio hardware DSP environment available to the public. Creative users have built custom stereo to 5.1 spatial up-mix processors, analog tape noise reduction, and multi-layered FOH speaker array time delay and acoustic correction tools, to name just a very few.

  Graphs can be very simple - just build a quick little tool to do some one-off task not gracefully handled by a packaged plug-in…

  Graphs are especially spectacular for creating new types of parallel processors with signal paths possible only within the 3d DSP engine.

- I/O: A basic Input/Output routing patch point, with Bypass. I/O Inserts can be used as Send Only, Return-only (a switchable, alternate Input source) or to create outboard processing loops. From any point in the strip processing chain, send to any available hardware or computer output port and/or return from any hardware or computer input port.

  Note: Since an I/O send/return insert sends audio to external systems outside the 3d Mixer engine and back, I/O loops can not be automatically latency-compensated.

> As a reminder, all of the mixing, gain control and processing which is shown and accessed through the MIOConsole3d Mix desk is operating solely in the box(es) and not on the computer, even though the user interface for the 3d Mixer is controlled from the computer display. This is an important distinction to keep in mind so you can manage your sessions to take best advantage of the strengths of both the DAW and the 3d hardware.

Post-Insert Direct Out

The **Post-Insert Direct Out** sends processed signal from the mixer strip to any available output port in the Domain: analog, digital, Host, or SCP.

Aux Sends

Aux Sends appear immediately below the 'Post-Insert Direct Out'.

Aux mix buses are set up in the "Configure Mixer" window (opened with [cmd-shift-C] or the Menu bar: "Mixer" menu). In "Configure Mixer", Aux sends are set to be shown or hidden on Input strips, and configured to feed the Aux mix either pre- or post-Insert. Lastly an Aux mix bus may be set up Pre-Fader, or may have the source Input strip Fader and Mute settings applied to the Aux send signal.

For details regarding Aux bus configuration, see Configure Mixer: Aux Buses.

Aux mix buses can be named either in "Configure Mixer", or by double-clicking on their bus master scribble strip at the bottom of the Aux bus strip.

Each Aux send includes a small horizontal 'Aux Send Fader'. Aux Sends from mono Input strips feeding stereo or multichannel Aux buses will include a mini-panner alongside the mini-Fader. As with faders and panners throughout the 3d Mixer, option-clicking the Aux send mini-Fader will set it to 0.0 dB unity gain, and opt-clicking the mini-panner will set it to Center.

Click on the Aux name in the Mixer Pane Legend to bring that Aux Mixer layer into focus on the Mixer desk - essentially swapping the mini-fader and mini-panner for the main mix desk fader and panner. All Input strip Fader knobs will turn yellow, indicating that the primary Faders, Solos, Mutes and Pans are now controlling that Aux bus mixer rather than the Main bus mix. A dot will appear next to the name of the Aux that is currently in focus.

Mixer Strip: 'Panner', 'Solo/Mute', 'Fader Gain', 'Fader', 'Meter', 'Record Enable'

The Fader section is the same for all Input strips and Bus strips, and consists of a stereo or surround panner, Solo, Mute and Fader controls, an editable Fader Gain readout, pre- or post-Fader meters and the Record Enable button.

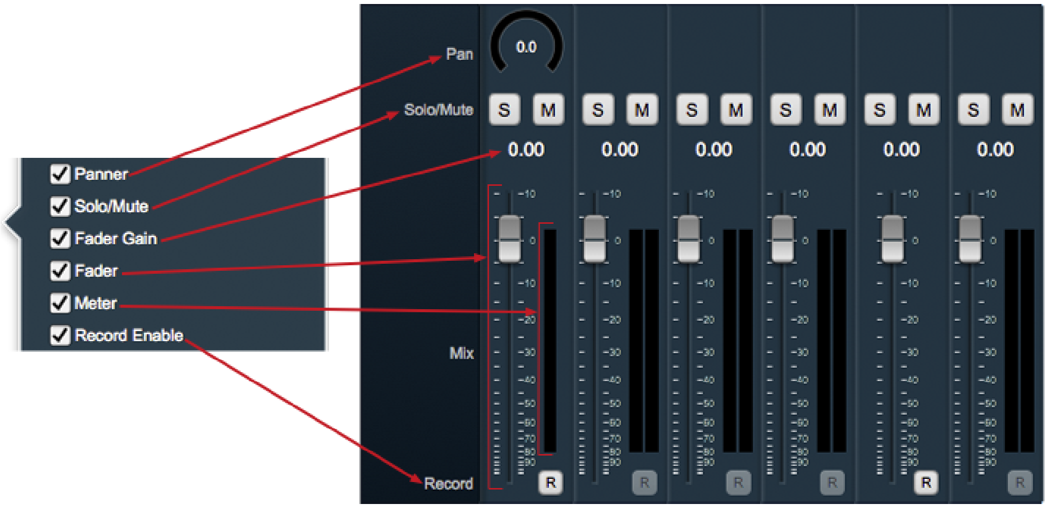


Figure 10.116: Panner, Solo/Mute, Fader Gain, Fader, Meter and Record Enable

Mixer Strip Panner

Panners appear on mono Input and Bus strips when the Main mix bus set to 2 or more channels wide.

When the Main mix bus is set to stereo or LCR, you will get a stereo panner:

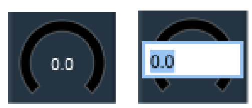


Figure 10.117: Stereo / LCR Panner

Click on the numeric readout to enter a position percentage: '0.0' is center, entering '-50' pans 50% to the left, '50' pans 50% to the right, and so on. Stereo and LCR panners follow the standard -3dBu center pan law.

If your Main mix bus is set to four channels or more, you will see a basic surround panner:

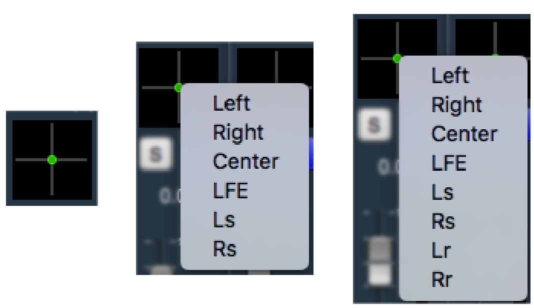


Figure 10.118: Surround Panner with direct assigns

The default position for the surround panner will be at the center of the sound field, with equal signal sent to the five main program channels. The LFE channel is not routed in the default configuration.

Right-clicking on the surround panner will open a pull-down menu where you can directly assign your mono input to any speaker feed. To remove a direct-assigned pan and return to the panner UI, click on the panner.

Solo

The Solo button turns yellow when engaged.

In Solo-In-Place mode, the 3d Mixer will only pass signal for channels that have the solo button engaged.

In PFL/AFL modes, the Solo'd signal is passed to the Solo bus, where it is routed to the Monitor Controller and Cue Controllers per the rules set by menu bar “Monitors”: ‘Monitor Controller Slaves to PFL/AFL’, and the hamburger menu settings of each individual Cue.

Solo'ing a bus passes the signal from all input and bus strips feeding the solo'd bus, with all panning and DSP processing from the source strips intact.

- ⌘ (Command)-click will “exclusively solo” your selected input strip. In other words, command-clicking any solo button will clear all solos in the Mixer except for the solo you click on. This is especially useful for quickly comparing individual mixer strips isolated from the full mix.
- ⌥ (Option)-click on any engaged solo button will toggle all solos in the Mixer. You can use this to clear all solos on the bus. Note that Option-clicking a bus solo will toggle that bus strip and all the Input strips solos.
- ⇧ (Shift)-click solo will add strips to the current selection, or remove them if already active.
- Click on a Solo button, hold, and drag left or right to sweep-toggle Solo buttons across your mixer, regardless of strip type.

Mute

When **Mute** is engaged signal in the associated strip is muted at the mixer. This does not affect the signal at the direct outs or pre-fader sends. The mute button turns blue when engaged.

Click-hold-and-drag left or right to sweep-toggle Mute buttons across your mixer, regardless of strip type.

Fader Gain display

The numeric **Fader Gain** displays the channel gain level in dB. Click in the readout display to manually set the desired Fader gain. Manual gain entries to 1/1000th of a dB are allowed and applied in the gain stage, but the display will be rounded to two decimal places.

Mix Desk Fader

The **Fader** adjusts the output level of the strip to its assigned bus. Fader maximum output gain is +10dBFS, 0.0dBFS is digital unity (no gain stage processing applied), and setting the Fader to minumum outputs digital zero from that strip (i.e.: no audio data - not even dither).

Fader Meter

The **Fader Meter** shows the Post-Insert signal level. Default meter readout is pre-Fader, and can be changed to show post-fader signal using the menu bar “Mixer” > “Input Strip Meters Post-Fader” command, or (Command-Shift-P).

Fader signal level meters are calibrated to the fader hashmarks, with major ticks every 10db and minor ticks each 5dB. Meters throughout the mixer display signal levels from -80dBFS to 0dBFS with a 2-second peak hold indicator.

Digital OVERs are displayed as sustained red peak-holds above 0dBFS. Click on the fader meter to clear the OVER flag, or Option-Click a meter to clear all OVER indications in that mixer pane.

Record Enable

Record Enable ("R") becomes available on a mixer strip *only* when "To Host" or a "Direct Out" is routed back to the host computer. Routes from the 3d mix environment back to the computer are required any time audio is fed from the 3d hardware back to host-based storage or DAW inputs.

Click on 'Record Enable' to arm the track for recording in Session, indicated by the button flashing red. Audio from this strip will now be captured when Record Start is triggered. While actively recording, the Record Enable button will become solid red.

Line Input Monitor

The **Input** ("I") control is attached to Record Enable on all mixer input strips.

When engaged (button is lit yellow), 'Input' manually locks the mixer strip source to its assigned hardware line input regardless of the Mixer Mode or Session playback state.

The Input button itself is shown/hidden in tandem with Record Enable.

When disengaged, mixer strip playback will follow the behaviors set in the Input Monitor Preferences pane.

> Note that in MIOConsole3d, many Fader section controls are also available in the Session Track headers, making it possible to control the key aspects of a capture-on-the-fly mix without having to swing back to the main Mixer desk:
>
> 
>
>
> **Figure 10.119: Session Track Header: duplicate Mixer Strip controls**

Mixer Strip: 'Link Group', 'Mute Group' and 'DCA' assigns, 'Bus Assign', 'Bus Output', 'Bottom Color Bar' and the 'Scribble Strip'

The bottom section of the Mixer Strips handles per-Strip parameter linking, Input Strip to sub-Group and Main Bus assignments, additional physical Bus outputs, Strip name and color identification.

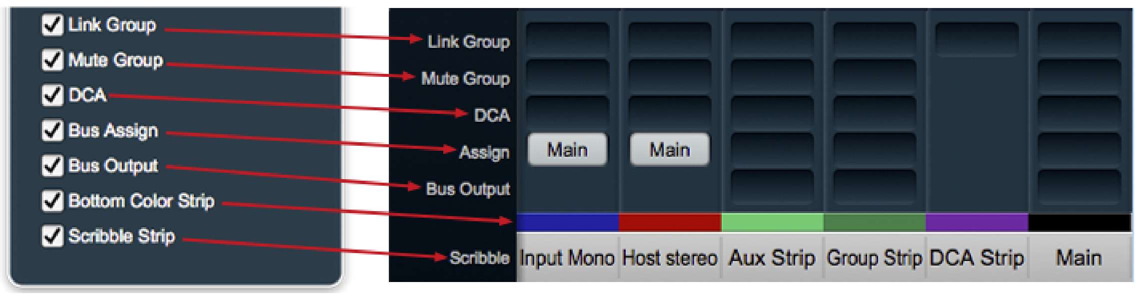


Figure 10.120: Link and Mute Group Assigns, DCA assigns, Bus Assign, Bus Output, Color bar and Scribble Strip

Link Group

Clicking the **Link Group** button will open the 'Link Group Assign' window. Here you can select one or multiple Link Groups, create a new Link Group, and apply your Link Group selection across multiple strips in one shot.

The screenshot below breaks out all the functions pertinent to managing Link Groups within the 3d Mixer. The 'Link Group Assign' window is at the top, the System Status Pane 'Link Groups List' on the lower left, and the Mixer Strip Link Groups buttons peeking out on the bottom right. These controls are all closely related, so let's run through how Link Groups work as a whole.

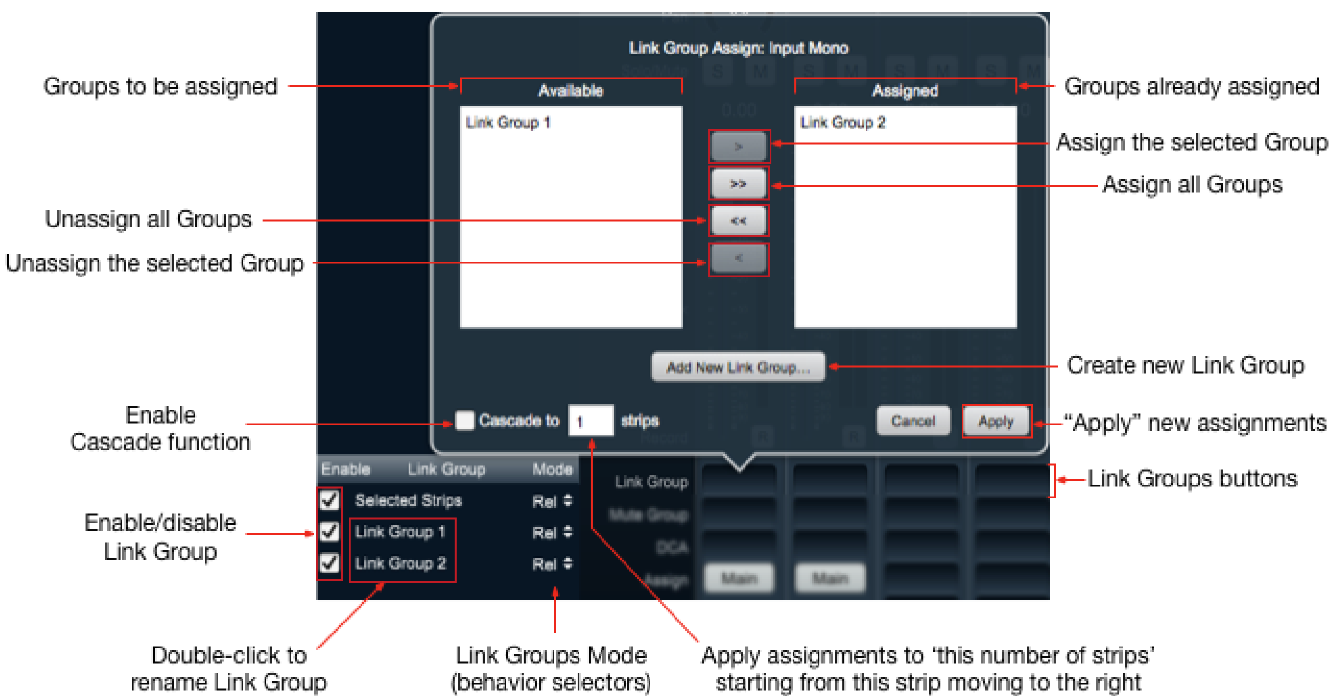


Figure 10.121: 'Link Group Assign' window with related Mixer Desk features overview

You start working with Link Groups by hitting the 'Link Groups' button at the bottom of any Input or Bus strip. This opens the 'Link Group Assign' window. The window is titled at the top with the window function and the name of the Mixer Strip from which you opened the window.

On the left, the 'Available' list presents you with all Link Groups not currently assigned to your target Strip. On the right, under 'Assigned', are listed Link Groups which are already assigned to your selected strip.

Between these two lists are four buttons with arrows. We will get to these in a moment, but first, click on the 'Add New Link Group…' button below to get things rolling…

- **'Add New Link Group...'** will immediately create a new Link Group in the 'Assigned' list. Note that this new Link Group also appears immediately in the Link Groups List in the System Status Pane.

  In the Link Groups List, double-click on a Link Group name to edit the name, or Control-click a Link Group to open a sub-menu where you can rename or delete the Link Group:

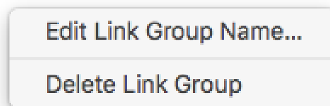
  

  Figure 10.122: Status Pane:Link Groups List: 'Edit Link Group Name…', 'Delete Link Group' submenu

  Please note: Although 'Add New Link Group…' immediately creates and assigns a new Link Group, all other assignment functions must be applied by hitting the 'Apply' button, '**Enter**' or '**Return**' keys.

  The 'Cancel' button, '**Esc**' key or '**cmd .**' key command will cancel new assignments, but will not undo newly created Link Groups.

- The **Single Arrow** buttons in the 'Link Groups Assign' window are for moving Link Groups between the 'Available' and 'Assigned' lists in the direction the arrow points. Make a selection in the 'Available' list and click the "**>**" button to move your selection to 'Assigned'. Use the standard macOS "shift-click" or "command-click" keystrokes to make multiple selections of the Link Groups in either list and move them all at once.

- The **Double Arrow** buttons will move all of the Link Groups from 'Available' to 'Assigned', and vice versa.

- A little shortcut: double-click a Link Group to move it immediately to the opposite list.

- By entering a number in the **'Cascade to __ strips'** box and enabling 'Cascade', you can apply your Link Group assignment across that number of strips, starting with your selected strip and moving to the right across the Mixer. So, in the example above, entering "4" in 'Cascade to __ strips' and checking the 'Cascade' enable box would assign your selected Link Groups to the four strips visible in the screenshot.

- By selecting multiple strips in the Mixer, then hitting one of the highlighted strips 'Link Group' buttons, the 'Link Groups Assign' window will replace 'Cascade to __ strips' with **'Cascade to selected strips'**:

Figure 10.123: 'Link Group Assign' window with multiple strip selection

'Cascade to selected strips' will be checked, and upon hitting 'Apply' your Link Group selection will be applied to all of your selected Mixer strips. In the above case, the Link Group selection would be applied to "Input Mono", "Host stereo", and "In 3" through "In 6", ignoring "Aux Strip" and "Group Strip" since they were not part of the original selection.

- When multiple Link Groups are assigned to a Mixer strip, the Link Group button for that strip will show as much of the first Link Group name as will fit in the width of the Strip, plus a **+** sign or an ellipsis **...** to indicate there's more than one. Hover the mouse cursor over the Link Group button and a tool-tip will pop up showing all Link Groups assigned to that strip:

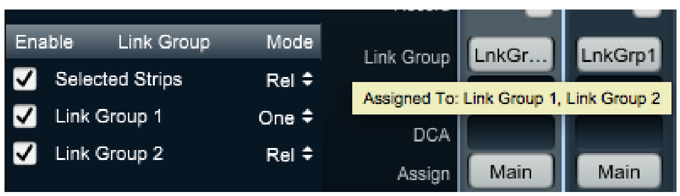


Figure 10.124: Tooltip revealing multiple Link Group assignments

- Link Groups can be Enabled or Disabled at will by clicking their checkbox in the Status Pane: Link Groups List. Disabling (unchecking) all Link Groups for a strip will make the Link Groups button for that strip turn yellow, indicating "All Link Groups are Bypassed".

Link Group Fader Behaviors

- **Abs**: (for "Absolute") will snap all faders in the group to the gain setting of the selected fader.
- **Rel**: "Rel" (as in "Relative") is the default linked fader behavior setting, in which linked faders follow the selected fader, always maintaining the same relative position between the fader you are moving and the faders following.
- **Inv**: (for "Inverse" Relative). In this mode moving one fader causes the other faders in the group to move in the *opposite* direction, while maintaining an inverse relative position between the fader you are moving and the faders following.
- **One** is a specialty behavior, especially suited to live shows where there are multiple soloists and you need to pick one out while ducking the others on the fly. Start with your linked faders all down, then move one up, as if that mic is the current featured soloist. Then when another solist comes up, move that fader up and the previous fader will duck out of the way. Repeat. This linking scheme generally does not play well with other Link Groups assigned to the same faders.

- **Link Groups Parameter Link map:** by strip position from top to bottom

Strip Control		
Source	Not Linked	
Hard Mute	Linked	
To Host	Not Linked	
Headamp	**Linked:** Input Type and Input Gain	**Not Linked:** Polarity Invert, Phantom Power
Direct Outs	Not Linked	
Inserts	Not Linked	
Aux	Linked	
Pan	Linked	
Strip Solo and Mute	Linked	
Fader Gain	Linked	
Record Enable	Linked	

Table 10.3. 'Link Groups' Parameter Link map

- **'Selected Strips' Parameter Link map**

Strip Control		
Source	Linked	
Hard Mute	Linked	
To Host	Linked	
Headamp	**Linked:** Input Type and Input Gain	**Not Linked:** Polarity Invert, Phantom Power
Direct Outs	Linked	
Inserts	**Linked:** Add/Remove processor	**Not Linked:** Processor parameters
Aux	Linked	
Pan	Linked	
Strip Solo and Mute	Linked	
Fader Gain	Linked	
Record Enable	Linked	

Table 10.4. 'Selected Strips' Parameter Link map:

- **Hold the "Shift" key to momentarily defeat Link Groups and operate a fader or control independently.**

  Once you release the "Shift" key, the fader will re-link with the group in its new position.

  Hold Shift+Option to reset a single parameter within a Link Group to its default setting.

Mute Group

Clicking the 'Mute Group' button will open the 'Mute Group Assign' window. Here you can select one or multiple Mute Groups, create a new Mute Group, and apply your Mute Group selection across multiple strips in one shot.

The screenshot below breaks out all the functions pertinent to creating and managing Mute Groups from the 3d Mixer desk. The 'Mute Group Assign' window is at the top, the System Status Pane 'Mute Groups List' on the lower left, and the Mixer Strip Mute Groups buttons shown on the bottom right.

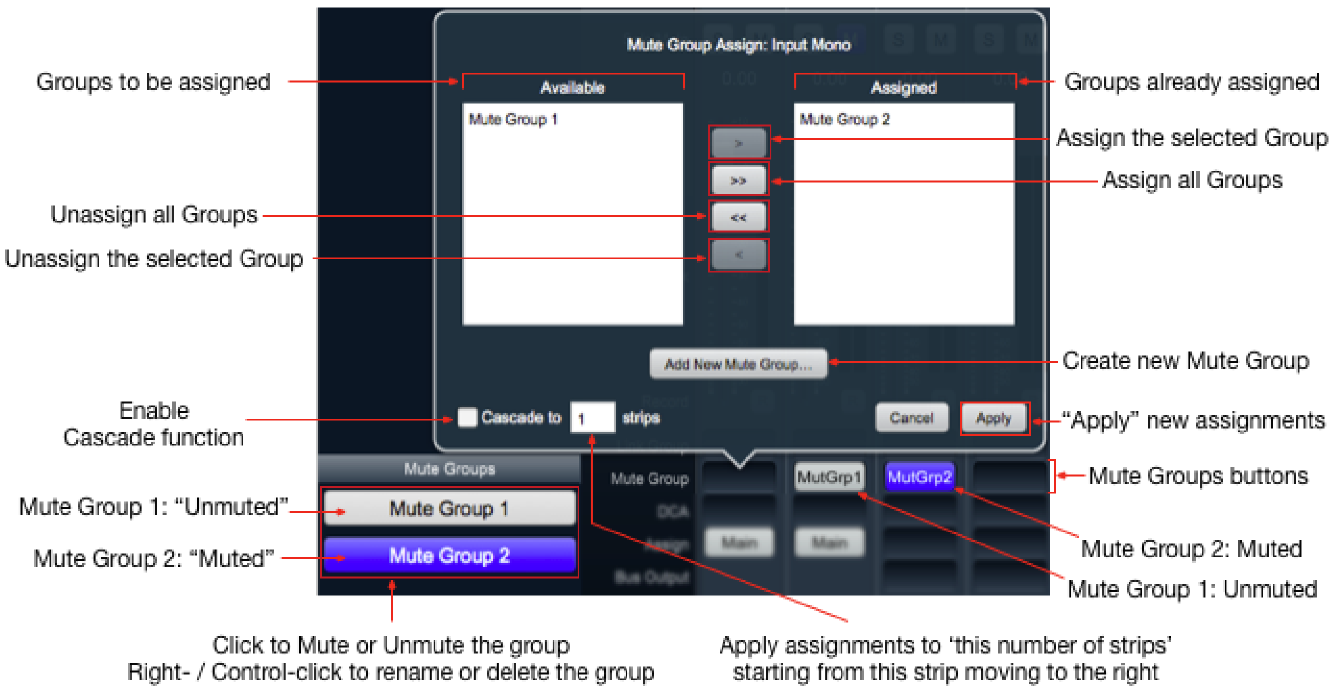


Figure 10.125: 'Mute Group Assign' window with related Mixer Desk features overview

As you can see, the Mute Group Assign window is the same interface as the Link Group Assign window, and the group creation and assignment operates the same exact way. Note that Mute Groups can also be added, deleted and renamed in the "Configure Mixer" window (opened with [cmd-shift-C] or from the Menu bar: "Mixer" menu):

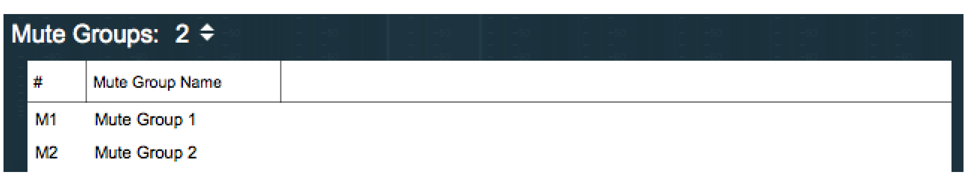


Figure 10.126: "Configure Mixer" Mute Groups Assign window

You can create and assign Mute Groups from the Mixer desk surface by hitting the 'Mute Group' button at the bottom of any Input or Bus strip. This opens the 'Mute Group Assign' window. The window is titled at the top with the window function and the name of the Mixer Strip from which you opened the window.

On the left, the 'Available' list presents you with all Mute Groups not currently assigned to your target Strip. On the right, under 'Assigned', are listed Mute Groups which are already assigned to your selected strip.

Click on the 'Add New Mute Group…' to begin

- **'Add New Mute Group...'** will immediately create a new Mute Group in the 'Assigned' list. Note that this new Mute Group also appears immediately in the Mute Groups lists in both the System Status Pane and the Mute Groups section of the "Configure Mixer" window.

  Please note: Although 'Add New Mute Group…' immediately creates and assigns a new Mute Group, all other assignment functions will only be applied by hitting the 'Apply' button, '**Enter**' or '**Return**' keys.

  The 'Cancel' button, '**Esc**' key or '**cmd .**' key command will cancel new assignments, but will not undo newly created Mute Groups.

- The **Single Arrow** buttons in the 'Mute Groups Assign' window are for moving Mute Groups between the 'Available' and 'Assigned' lists in the direction the arrow points. Make a selection in the 'Available' list and click the "**>**" button to move your selection to 'Assigned'. Use the standard macOS "shift-click" or "command-click" keystrokes to make multiple selections of the Mute Groups in either list and move them all at once.

- The **Double Arrow** buttons will move all of the Mute Groups from 'Available' to 'Assigned', and vice versa.

- Double-click a Mute Group in either list to move it immediately to the opposite list.

- By entering a number in the **'Cascade to __ strips'** box and enabling 'Cascade', you can assign your Mute Group across that number of strips, starting with your selected strip moving to the right across the Mixer. In the example above, entering "4" in 'Cascade to __ strips' and checking the 'Cascade' enable box would assign your selected Link Groups to the four strips visible in the screenshot.

- By selecting multiple strips in the Mixer, then hitting one of the highlighted strips' 'Mute Group' buttons, the 'Mute Groups Assign' window will replace 'Cascade to __ strips' with **'Cascade to selected strips'**.

  'Cascade to selected strips' will be checked automatically, and upon hitting 'Apply' your Mute Group selection will be applied to all of your selected Mixer strips.

- When multiple Mute Groups are assigned to a Mixer strip, the Mute Group button for that strip will show as much of the first Mute Group name as will fit in the width of the Strip, plus a (**+**) sign or an ellipsis (**...**) to indicate there's more than one. Hover the mouse cursor over the Mute Group button and a tool-tip will pop up showing all Mute Groups assigned to that strip:

Figure 10.127: Tooltip revealing multiple Mute Group assignments

Mute Group Behaviors

Mute Groups in MIOConsole3d are designed primarily for the needs of live sound, and behave differently than muting through Link Groups or hitting a Mixer strip mute button. For starters, Mute Groups do not use the Mixer Strip mute buttons.

Mute Groups are engaged by hitting the Mute Group button in the Mute Groups list in the Status pane to the left of the Mixer desk. The advantage here is pretty obvious - those buttons can always be visible and accessible regardless of your Mixer strip layout.

Further, when a strip is muted by a Mute Group, the Mute Group selector button at the bottom of the strip turns blue, and the Fader mute button blinks. This tells you at a glance that a Mute Group is muting that fader. In cases where a strip muted both by a Link Group and a Mute Group, both the Fader mute button and the Mute Group selector will be a solid, unblinking blue.

Another major difference from Link Group mutes is that strips muted by Mute Groups will *stay muted until you expressly un-mute that Mute Group* - there is no way to override a Mute Group on an individual strip. If a fader is assigned to multiple Mute Groups, you must un-mute *all* of those Mute Groups to un-mute that fader.

Fader Mute and Link Group mutes do not work this way. When you have multiple Link Groups muting a fader, releasing a mute on any one of those Link Groups will release a mute initiated by either that strip's Mute button or a mute command from another Link Group.

To rename or delete a Mute Group from the Mixer desk, right- or control-click the Mute Group to open a sub-menu, as shown below:

Figure 10.128: Status Pane: Mute Groups List: 'Edit Mute Group Name…', 'Delete Mute Group' submenu

DCAs

Digitally Controlled Amplifier strips in MIOConsole3d work the same as DCAs on a hardware digital console. No audio signal passes through a DCA strip, the DCA fader simply tells faders under its control to move up or down by the amount the DCA fader knob moves. This gain change is not reflected by the slaved faders visibly moving, like they would when moving a fader in a Link Group.

DCA strip Solo and Mute commands are reflected on slaved strips, however.

When a mixer strip is muted by a DCA, the DCA selector button at the bottom of the strip turns blue, rather than the Fader mute button. This tells you at a glance that a DCA is muting that fader. In cases where a strip muted by a Link Group, a Mute Group and a DCA, the Fader mute button, the Mute Group selector *and* the DCA selector will all show blue.

The same applies when a strip is solo'd by a DCA command - the DCA selector button will turn yellow to show it's solo'd by a DCA.

Like Mute Groups, DCA's can also be added, deleted and renamed in the "Configure Mixer" window (opened with [cmd-shift-C] or from the Menu bar: "Mixer" menu):

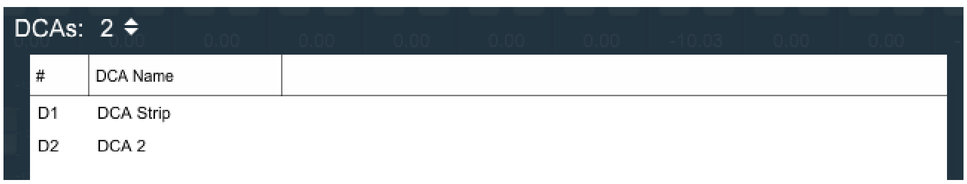


Figure 10.129: "Configure Mixer" DCAs Assign window

DCA strips can be deleted directly from the Mixer desk by selecting the DCA strip and hitting (cmd-shift-D) and from the Menu Bar: Mixer: "Delete Strips" command.

DCA strips can be shown or hidden per Mixer Pane by using the hamburger menus in the upper left corner of each Mixer Pane.

The screenshot below breaks out all the functions pertinent to creating and assigning DCAs from the 3d Mixer desk.

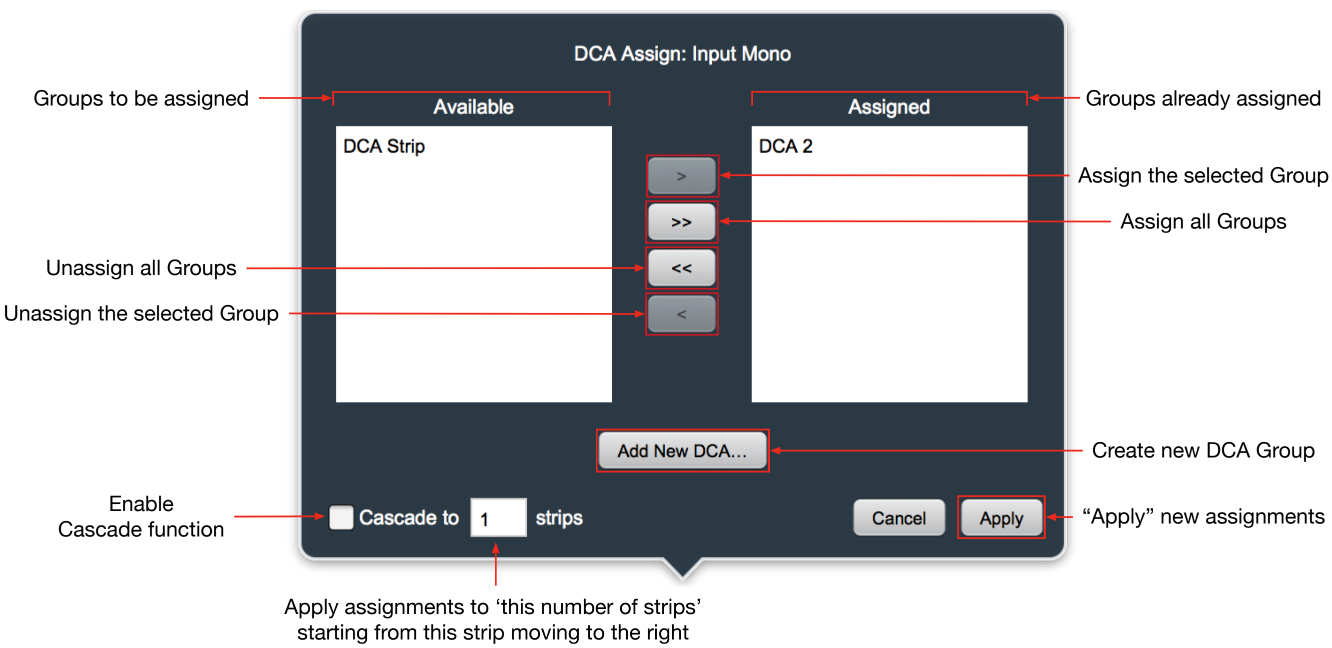


Figure 10.130: 'DCA Assign' window

You can create and assign DCAs from the Mixer desk surface by hitting the 'DCA' selector button at the bottom of any Input or Bus strip. This opens the 'DCA Assign' window. The window is titled at the top with the window function and the name of the Mixer Strip from which you opened the window.

On the left, the 'Available' list presents you with all DCAs not currently assigned to your target strip. On the right, under 'Assigned', are listed DCAs which are already assigned to your selected strip.

Click on the 'Add New DCA…' to begin.

- **'Add New DCA…'** will immediately create a new DCA in the 'Assigned' list. Note that this new DCA also appears immediately in the DCAs section of the "Configure Mixer" window.

  Please note: Although 'Add New DCA…' immediately creates and assigns a new DCA, all other assignment functions must be applied by hitting the 'Apply' button, '**Enter**' or '**Return**' keys.

  The 'Cancel' button, '**Esc**' key or '**cmd .**' key command will cancel new assignments, but will not undo newly created DCAs.

- The **Single Arrow** buttons in the 'DCAs Assign' window are for moving DCAs between the 'Available' and 'Assigned' lists in the direction the arrow points. Make a selection in the 'Available' list and click the "**>**" button to move your selection to 'Assigned'. Use the standard macOS "shift-click" or "command-click" keystrokes to make multiple selections of the DCAs in either list and move them all at once.

- The **Double Arrow** buttons will move all of the DCAs from 'Available' to 'Assigned', and vice versa.

- Double-click a DCA to move it immediately to the opposite list.

- By entering a number in the **'Cascade to __ strips'** box and enabling 'Cascade', you can assign your DCA across that number of strips, starting with your selected strip moving to the right across the Mixer. In the example above, entering "4" in 'Cascade to __ strips' and checking the 'Cascade' enable box would assign your selected DCAs to the four strips visible in the screenshot.

- By selecting multiple strips in the Mixer, then hitting one of the highlighted strips 'DCA' selector buttons, the 'DCA Assign' window will replace 'Cascade to __ strips' with **'Cascade to selected strips'**

  'Cascade to selected strips' will be checked, and upon hitting 'Apply' your DCA selection will be applied to all of your selected Mixer strips.

- When multiple DCAs are assigned to a Mixer strip, the DCA selector button for that strip will show as much of the first DCA name as will fit in the width of the strip, plus a (**+**) sign or an ellipsis (**…**) to indicate there's more than one. Hover the mouse cursor over the DCA selector button and a tool-tip will pop up showing all DCAs assigned to that strip:

Figure 10.131: Tooltip revealing multiple DCA assignments

- To delete a DCA entirely, select the DCA control strip and hit **cmd-shift-d**, or "Delete Strips" from the menu bar Mixer menu.

Bus Assign

Use the **Bus Assign** selector to route the output of any Input strip, Aux Bus strip, or Group Bus strip to Group buses and/or the Main mix bus. The screenshot below breaks out all the functions pertinent to creating buses and assigning strips to buses on the 3d Mixer desk.

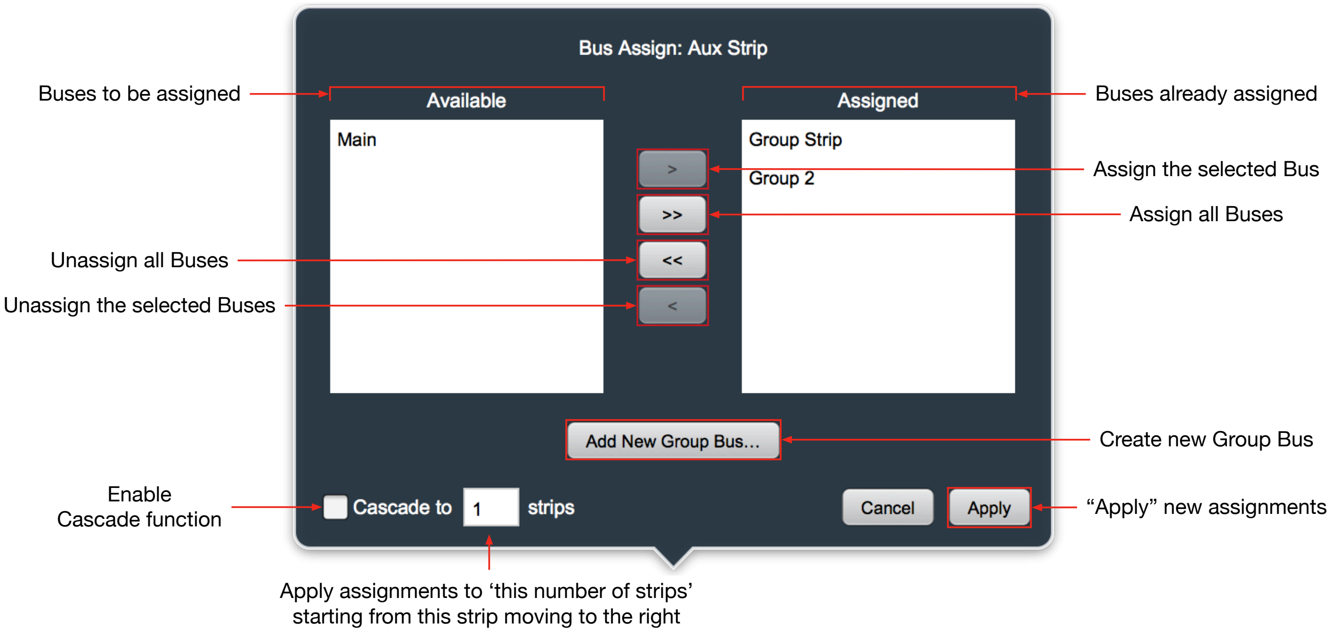


Figure 10.132: 'Bus Assign' window

A few notes regarding buses in the 3d Mixer:

New Input Strips added to an existing Mixer will be assigned by default to the 'Main' mix bus. Aux and Group buses added to an existing Mixer via "Configure Mixer" do not make that assumption, allowing you to assign them manually as you desire. The 'Bus Assign' window lets you create and assign buses in one interface.

Aux mix buses include their own mix engine, with fader, pan and mute controls for each Input strip independently of the Main mix bus. Aux buses are for building a completely independent mix and can be used for effect sends, or for independent cue mixes. Aux buses may be routed to Group buses, the Main bus, Host or USB computer ports, and/or to any hardware output.

Like most mixer desks, the 3d mixer supports one 'Main' mix bus as the final summing stage within the mix environment. Group buses are essentially sub-mix buses which share the same fader, mute, and panning settings as the Main mix bus. Group buses can be routed to hardware and computer ports out to the world, to other Group buses, or the Main mix.

You can create and assign Group Buses from the Mixer desk surface by hitting the 'Bus Assign' selector button at the bottom of any Input or Bus strip. This opens the 'Bus Assign' window. The window is titled at the top with the window function and the name of the Mixer Strip from which you opened the window - in the case above, we are assigning a new bus to the aptly-named Aux mix return strip: "Aux Strip".

On the left, the 'Available' list presents you with all Group buses (and possibly the Main mix bus) not currently assigned to your target Strip. On the right, under 'Assigned', are listed Group Buses which are already assigned to your selected strip.

Click on the 'Add New Group Bus…' to begin

- **'Add New Group Bus...'** will immediately create a new Group Bus in the 'Assigned' list. Note that this new Group Bus also appears immediately in the Group Buses section of the "Configure Mixer" window.

  Please note: Although 'Add New Group Bus…' immediately creates and assigns a new Group Bus, all other assignment functions must be applied by hitting the 'Apply' button, '**Enter**' or '**Return**' keys.

  The 'Cancel' button, '**Esc**' key or '**cmd .**' key command will cancel new assignments, but will not undo newly created Group Buses.

- The **Single Arrow** buttons in the 'Bus Assign' window are for moving buses between the 'Available' and 'Assigned' lists in the direction the arrow points. Make a selection in the 'Available' list and click the "**>**" button to move your selection to 'Assigned'. Use the standard macOS "shift-click" or "command-click" keystrokes to make multiple selections of the buses in either list and move them all at once.
- The **Double Arrow** buttons will move all of the buses from 'Available' to 'Assigned', and vice versa.
- Double-click a Bus to move it immediately to the opposite list.
- By entering a number in the **'Cascade to __ strips'** box and enabling 'Cascade', you can assign your DCA across that number of strips, starting with your selected strip moving to the right across the Mixer. In the example above, entering "4" in 'Cascade to __ strips' and checking the 'Cascade' enable box would assign your selected DCAs to the four strips visible in the screenshot.
- By selecting multiple strips in the Mixer, then hitting one of the highlighted strips 'Bus Assign' selector buttons, the 'Bus Assign' window will replace 'Cascade to __ strips' with **'Cascade to selected strips'**

  'Cascade to selected strips' will be checked, and upon hitting 'Apply' your Bus Assign selection will be applied to all of your selected Mixer strips.

- When multiple buses are routed from a Mixer strip, the Bus Assign selector button for that strip will show as much of the first bus name as will fit in the width of the strip, plus a (**+**) sign or an ellipsis (...) to indicate there's more than one. Hover the mouse cursor over the Bus Assign selector button and a tool-tip will pop up showing all buses assigned to that strip:

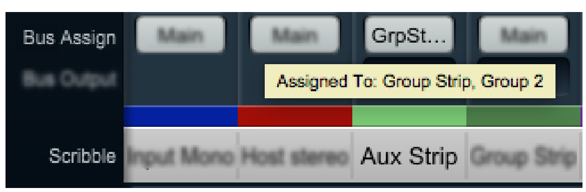


Figure 10.133: Tooltip revealing multiple Group Bus assignments

Bus Output

The **Bus Output** selector button is hidden from the Mixer UI by in the default Input Strips mixer view (Mixer Pane hamburger menu: "Set to Defaults for Input Strips"), but can be shown at any time using "Configure Mixer Strips Controls…" or selecting "Set to Defaults for Bus Strips".

Use the 'Bus Output' selector to send the post-fader signal from any bus strip to an external destination.

Note that all bus strips already include "To Host" routing at the top of each strip, plus dedicated routes behind the Mixer UI to the Monitor Controller and Cue Controllers. "In MC" is the default setting for all newly-created buses, unless specifically disabled within the "Configure Mixer" window, as shown below.

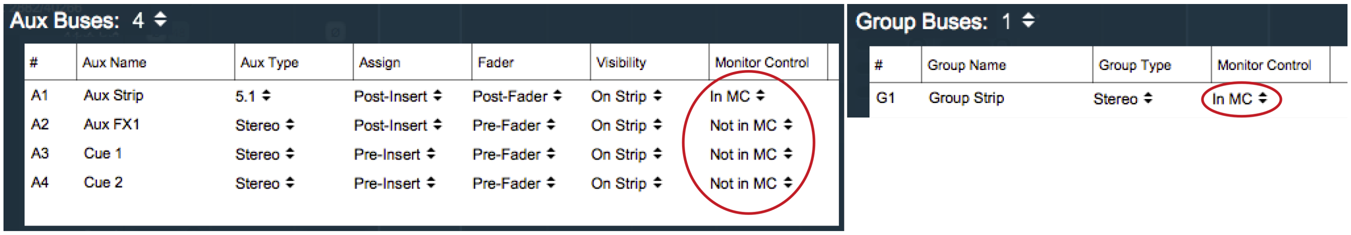


Figure 10.134: "Configure Mixer": Bus Strip routing to Monitor/Cue Controllers

So 'Bus Output' can be considered an extra post-fader bus send to any available analog or digital output, SCP USB out, or as an additional return route to your Host computer. Bus Outputs are often used to feed the 3d Record Panel, Spectrafoo (Metric Halo's host-based metering and audio analysis program), and outboard hardware encoding and metering systems.

Bottom Color Bar

Clicking on the Bottom Color Bar brings up a macOS color selector, from which you may choose a color for that Mixer strip. In the Appearance pane of the MIOConsole3d Preferences you may select whether to color the entire mixer strip or just the top and bottom color bars.

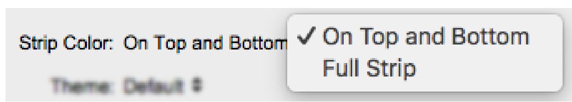


Figure 10.135: Console3d Prefs: Appearance: Strip Color configurator

The Bottom and Top Color Bars mirror each other, so what applies to one applies to the other, except that you may show or hide each independently with "Configure Mixer Strip Controls…".

Scribble Strip

Double-click to name any Mixer strip. The name you enter in the Scribble Strip will be propagated throughout the 3d user interface wherever a Mixer Strip name is found, including plug-in headers, Link Group, Mute Group, DCA and Bus Assign selection windows.

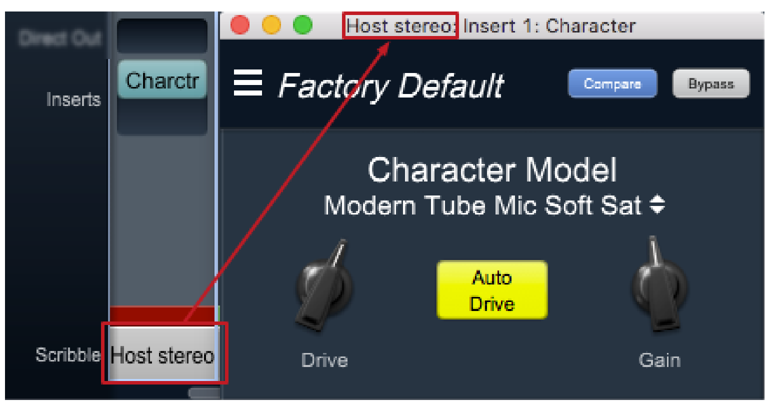


Figure 10.136: Scribble Strip name to Mixer strip Insert header

Scribble Strip names are also used to name sound files recorded in the Record Panel.

'Configure Mixer' Breakdown

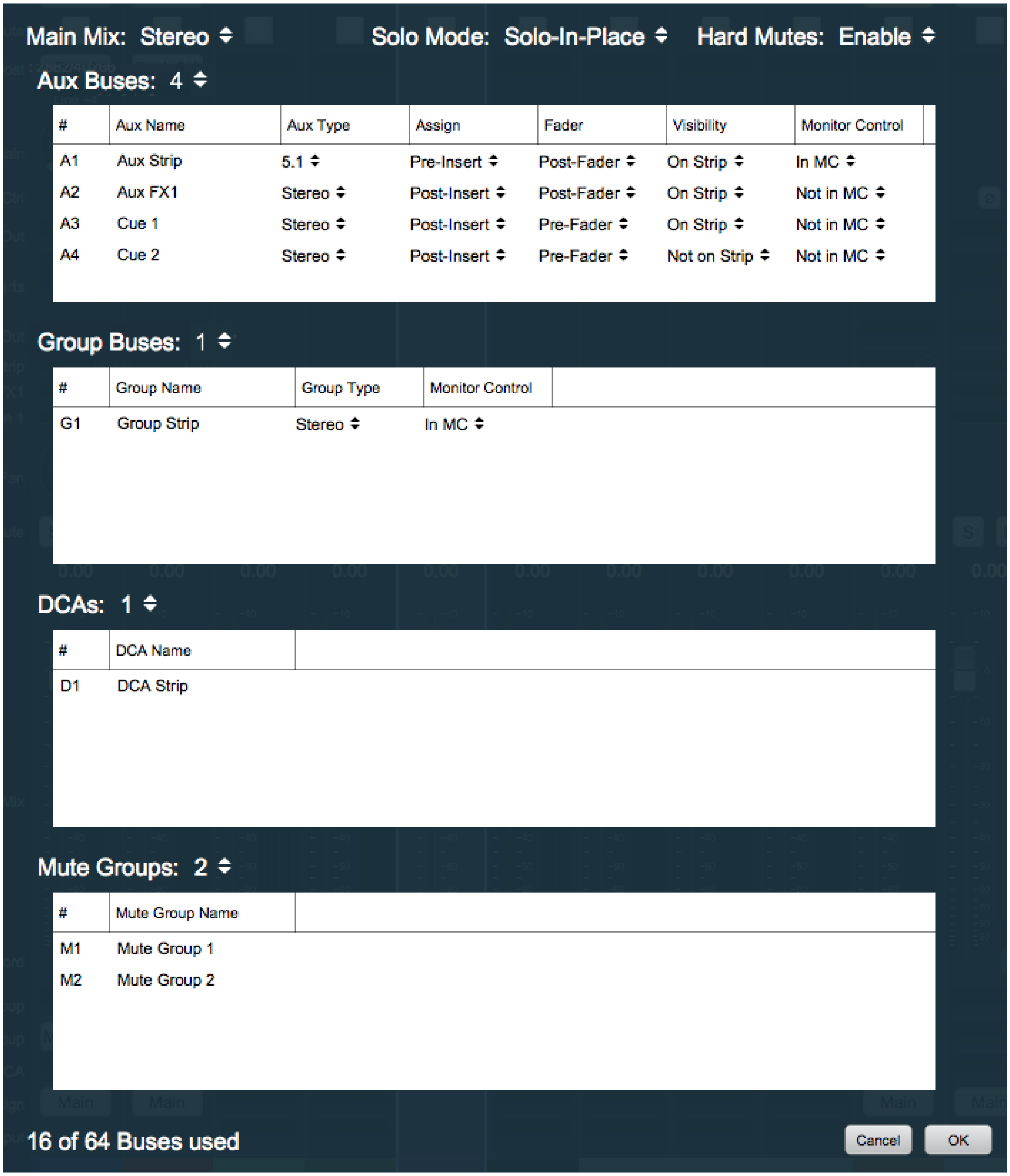


Figure 10.137: Configure Mixer Breakdown

From the top, the control elements comprising the 'Configure Mixer' interface are as follows:

- Main Mix:

  At the upper left corner of the window is the "**Main Mix:**" channel configuration menu. This menu determines the channel width and layout of your Main Mix bus.

  This setting determines the channel configuration for newly-created 'multichannel' Input Strips.

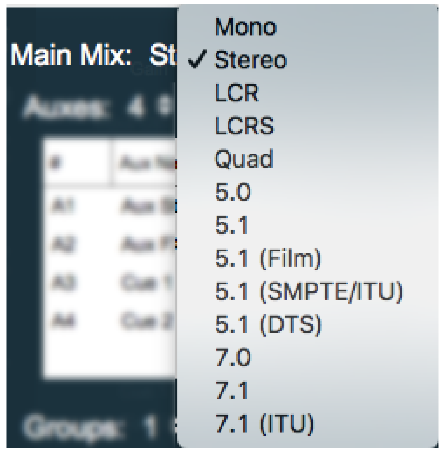


Figure 10.138: "Main Mix:" configuration menu

Channel layouts for the Main Mix bus are available from Mono to 7.1. Note that having a 5.1 SMPTE/ ITU Main Mix bus does not preclude you from having Input Strips, Auxes, Group buses or Monitor Controller outputs in mono, stereo or other surround channel configurations.

- **Solo Mode:**

  The **Solo Mode:** selector lets you choose between the default Solo-In-Place mode and your choice of PFL, AFL, PFL (mono) or AFL (mono) solo operation modes.

  The PFL/AFL modes instantiate a stereo Solo Bus strip at the left end of the Mixer desk. The PFL/ AFL Solo bus offers "To Host" return routes and Insert processor slots, with routing available for Main and Group buses and hardware outputs.

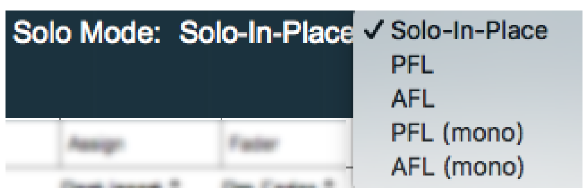


Figure 10.139: "Solo Mode:" configuration menu

  Solo modes 'Pre-Fader Listen' and 'After-Fader Listen' follow the usual self-descriptive rule: 'PFL' sends Pre-Fader/Panner signal to the Solo Bus, and 'AFL' sends Post-Fader/Panner. The PFL/AFL Mono modes send a summed mono signal to the Solo bus.

- **Hard Mutes:**

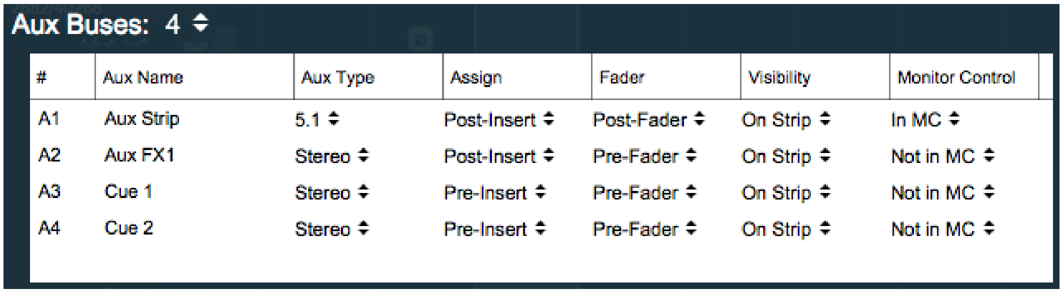
  'Hard Mute' controls are Disabled and hidden from Mixer strips by default. Enabling "Hard Mutes" places a big white square button at the top of every Input and Bus strip in the mixer, just below the Input routing selector. 'Hard Mutes', when enabled, will kill *all* audio routes through the strip, including signal to 'Pre-Insert To Host' return routes, Insert processes, Direct Outs and all assigned buses.

  Note that enabling 'Hard Mute' does not take the place of, or otherwise affect regular Fader Mute operation.

Aux Buses

This section of "Configure Mixer" is used for the creation of Auxiliary mix buses.

Use the **Aux Buses:** pull-down menu to select the number of Aux mix buses you wish to create. MIOConsole3d currently supports up to 32 Aux buses per mixer.



Aux Buses: 4

#	Aux Name	Aux Type	Assign	Fader	Visibility	Monitor Control
A1	Aux Strip	5.1	Post-Insert	Post-Fader	On Strip	In MC
A2	Aux FX1	Stereo	Post-Insert	Pre-Fader	On Strip	Not in MC
A3	Cue 1	Stereo	Pre-Insert	Pre-Fader	On Strip	Not in MC
A4	Cue 2	Stereo	Pre-Insert	Pre-Fader	On Strip	Not in MC

Figure 10.140: "Aux Buses:" configuration menu

The 'Aux Buses' configuration field consists of six columns with parameters for configuring your Auxiliary mix buses:

- Click in the **Aux Name** field to enter a name for the Aux bus.
- Use **Aux Type** to select the bus width and channel layout of the bus. The options are the same as the "Main Mix" configuration menu, ranging from mono to 7.1 ITU.

  Although Aux buses are commonly used for effects mixes returned to the Main mix bus, they are also often used for independent Cue mixes for headphone monitoring or complete alternate mixes for auditioning in mono, stereo or a different surround audio format. The channel width and layout can be set up independently of the Main Mix bus configuration to support this. Folddown of surround buses to mono and stereo cues is handled automatically.
- **Assign** determines if the Aux bus send from each Mixer Input strip originates before (pre) or after (post) the Insert slots. The menu selects between "Pre-Insert" and "Post-Insert", and applies to all Mixer sends to the selected Aux bus.
- **Fader** determines how the Input strip fader gain, mute and solo will be applied to the Aux send. The menu selects between "Pre-Mute","Pre-Fader", and "Post-Fader". The setting for the bus applies to all Mixer sends to the Aux bus. Since each strip's Aux send has its own panner (where appropriate), Input strip Pan control is not applied to Aux sends.
  - **Pre-Mute** sends to the Aux ignoring the Input strip 'Mute', 'Solo' and 'Fader' states.
  - **Pre-Fader** ignores the Input strip 'Fader' level, but follows the strip 'Mute' and 'Solo' states.
  - **Post-Fader** follows the Input strip 'Mute' and 'Solo' states, and applies the strip 'Fader' level as an offset to the Aux send feed.

  See the Mixer Aux Bus Fader Modes in Detail section for a more detailed description of how these modes function.
- **Visibility**: "On-Strip" shows this Aux bus send on the Mixer Input strips, "Not On Strip" hides the Aux from Mixer Input strips.
- **Monitor Control**: "In MC" adds the Aux bus output as a new Source in the Monitor Controller, "Not in MC" does not.

Mixer Aux Bus Fader Modes in Detail

Aux buses have three different modes available for linking the controls on the Mixer Strip to the controls for the send from that strip to the Aux bus.

These three different modes allow you to configure your auxes to meet your mixing needs. The **Pre-Mute** and **Post-Fader** correspond to the normal way that Pre-fader and Post-fader sends work in most DAWs. The **Pre-Fader** mode works the way Pre-fader works on most consoles.

Mode	Fader Gain	Mute	Solo (SIP)
Pre-Mute	ignore	ignore	ignore
Pre-Fader	ignore	logical ‘or’	logical ‘or’
Post-Fader	apply gain offset	logical ‘or’	logical ‘or’

Table 10.5. Aux Bus Send “Fader” controls table: Solo-In-Place (SIP) mode

In **Pre-Mute** mode now causes the Aux sends are unaffected by the mute and solo controls on the strip.

In **Pre-Fader** mode the Aux sends follow the Mute and Solo of the strip (strictly, this is a logical OR - e.g. if the send Mute is on OR the strip mute is On, the send will be muted; same for Solo).

In **Post-Fader** mode the Aux sends follow the Fader Mute and Solo of the strip, with the Fader gain for the strips being used to offset the send gain.

Newly created Auxes (either manual or automatic) will default to the new “Pre-Fader” mode.

Practical Implications:

If you have all of your Aux buses set to either “Pre Fader” or “Post Fader” the following things will be true:

If you Mute a strip with the normal mixer or control surface ‘Mute’ button, the follow things will be muted:

- The send to Main (if assigned)
- All sends to any assigned groups
- All sends to any Auxes that are either Pre-Fader or Post-Fader

…and the follow things will NOT be muted:

- Sends to the Computer
- Sends to Pre/Post direct outs
- Sends to to Assigned Send outputs in I/O inserts

If you want the I/O Insert sends (as opposed to the mixer sends) to be muted, you will need to use the Hard Mute facility, which is effectively a strip disconnect function.

If an aux is set to “Pre Mute” the solo and mute on the strip will have no effect on the Aux sends.

Similarly, with these modes selected, Solo (for SIP) on the strip will propagate to the Aux sends; as a result the aux buses will have the same solo state as the main mix; this has the effect of making it so the audio from the strip will be soloed to both the main bus and the aux(es). For things like sending to a reverb aux bus, this is cool because it will mute the sends from un-soloed strips without having to actually manually solo the aux sends individually, such that the soloed strip(s) are the only signals feeding the reverb.

Yes, this is one of those things that is much easier demonstrated than explained. Let’s walk through it a bit…

You have a bunch of strips and an Aux. The strips are assigned to Main. The Aux is also Assigned to Main. You are in SIP Solo mode.

The Aux bus “Fader” parameter is set to “Pre-Mute”:
When you solo a strip, the Main bus sees that a strip is soloed, and all the other strips assigned to the Main bus are muted (this would include the Aux master that is assigned to the Main bus, because you have not explicitly soloed it).

The Aux bus “Fader” parameter is set to Pre- or Post-Fader:
When you solo the strip, the Main bus sees that the strip is soloed. But the solo is also propagated to the Aux bus assign for that strip. So the Aux bus sees that the send is soloed. All the sends from other strips on the aux are muted. All the other (non-soloed) strips on the Main bus are muted. But the Aux master reports that it is soloed (because it has one or more sends that are soloed), so it will NOT be muted.

The net effect is that soloing the strip will cause the the strip + aux to be summed into the Main (with all the other non-soloed strips muted) and the only thing that will be feeding the aux will be the send from the soloed strip. So, effectively the Aux master is dynamically “solo safe” because it has a soloed send assigned to it.

Note that in SIP mode, if you are using the Auxes for Monitor or Cue mixes and you leave them as Pre-Fader (as opposed to Pre-Mute), soloing in the main mix will also solo on those Monitor/Cue mix buses. If you don’t want that, then you will need to set the Configure Mixer “Solo Mode” to PFL or AFL, so that soloing does not affect the mix/monitor buses and is instead routed through a solo bus.

So, if the “Solo Mode” is set to PFL or AFL (as you would have it set in a live mixing situation), the table becomes:

Mode	Fader Gain	Mute	Solo (PFL/AFL)
Pre-Mute	ignore	ignore	ignore
Pre-Fader	ignore	logical ‘or’	ignore
Post-Fader	apply gain offset	logical ‘or’	ignore

Table 10.6. Aux Bus Send “Fader” controls table: PFL and AFL modes

So in PFL/AFL Solo Mode, setting Aux Buses: ‘Fader’ to either Pre- or Post-Fader essentially follows the operational model of a live mixing console.

Important Aux Bus Fader Modes information for Early Adopters

The Fader Modes behavior described in the preceding pages reflects MIOConsole3d pb10-39 and newer. This differs from how MIOConsole3d behaved in previous versions.

Based on user feedback, the Mixer strip 'Mute' and 'Solo' button behaviors have been refined to be more in line with what people expect. The following describes what the changes are, and what you might need to do to make things work the way you want.

The behavioral settings of existing .cnsl3d files are unchanged. So, when you open an existing .cnsl3d file you will see any Aux buses that were set to **Pre Fader** will now show as being set to **Pre Mute**- e.g. no operational change in behavior from before.

If you want the new **Pre Fader** behavior for your Auxes on existing mixers, you will need to open "Configure Mixer", and set the "Fader" mode of the Auxes to "Pre-Fader".

Newly created Aux buses (either manual or automatic) will default to the new "Pre-Fader" mode.

Background:

Up until build pb10-39 there were two user selectable "Modes" of internal linking between the strip Mute, Solo and Fader: **Pre Fader** &**Post Fader**:

Mode	Fader Gain	Mute	Solo (SIP)
Pre-Fader	ignore	ignore	ignore
Post-Fader	apply gain offset	logical 'or'	logical 'or'

Table 10.7. previous Aux Bus Send "Fader" controls table

Pre-Fader as a result was effectively pre-fader, pre-mute and pre-solo.

Based on user feedback we have come to understand that this is unexpected for a lot of people, and especially unworkable for using MIOConsole in a live (FOH) context.

That said, it is the way things like Pro Tools work, so it is also important to not lose the ability to support pre-mute, pre-fader sends to aux buses.

How it has changed:

The way we are addressing this is as follows. We have added a new user selectable mode for the Auxes: **Pre Mute** and have redefined **Pre Fader** to be post-mute/post-solo.

Group Buses

Use the **Group Buses:** pull-down menu to select the number of Group buses you wish to create. MIOConsole3d currently supports up to 32 Group buses per mixer.

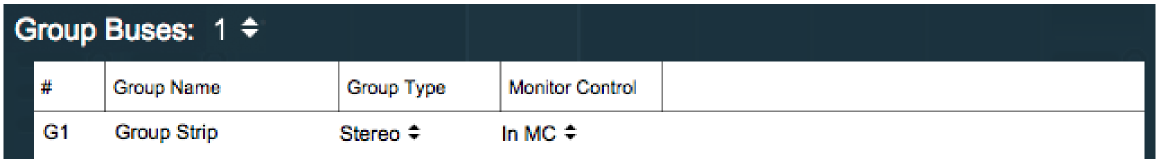
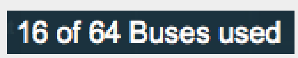


Figure 10.141: "Group Buses:" configuration menu

- Click in the **Group Name** field to enter a name for the Group bus.
- Use **Group Type** to select the bus width and channel layout of the Group bus. The options are the same as the "Main Mix" configuration menu, ranging from mono to 7.1 ITU.
- **Monitor Control**: "In MC" adds the Group bus output as a new Source in the Monitor Controller, "Not in MC" does not.

> Each Domain Mixer desk has 64 bus channels available. The 'Buses Used' display at the bottom of the 'Configure Mixer' window shows the number of buses currently assigned. In the example shown at the top of the 'Configure Mixer' section, the stereo Main Mix uses 2 buses, Aux Strip 1 is a 5.1 configuration so it uses 6 buses, Auxes 2, 3 and 4 are all stereo so that's another 6, plus the stereo Group strip for a total of 16 of 64 buses used.
>
> Note that switching the 'Solo Mode' to PFL/AFL will add the stereo PFL/AFL Solo bus strip to the mixer, using another two bus channels.
>
> 
>
>
> **Figure 10.142: 'Configure Mixer' window: "Buses Used" display**

DCAs

Use the **DCAs:** pull-down menu to select the number of DCAs you wish to create. MIOConsole3d currently supports up to 32 DCAs per mixer.

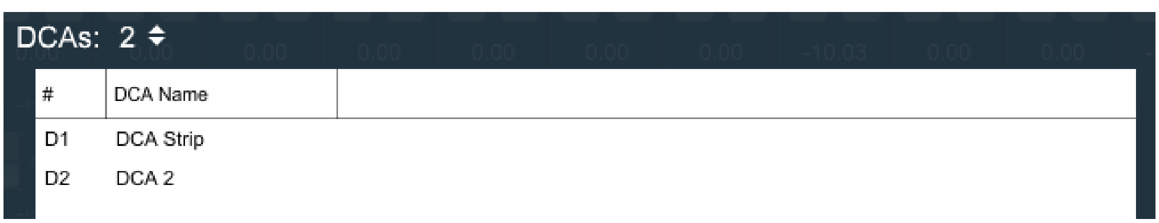


Figure 10.143: "DCAs:" configuration menu

Click in the **DCA Name** field to enter a name for the DCA.

For details on DCA operations, see the DCAs section of the Mixer Strip Controls breakdown.

Mute Groups

Mute Groups are just Link Groups dedicated to Muting, available for those situations where linking all the strip parameters is not desired.

Use the **Mute Groups:** pull-down menu to select the number of DCAs you wish to create. MIOConsole3d currently supports up to 32 Mute Groups per mixer.

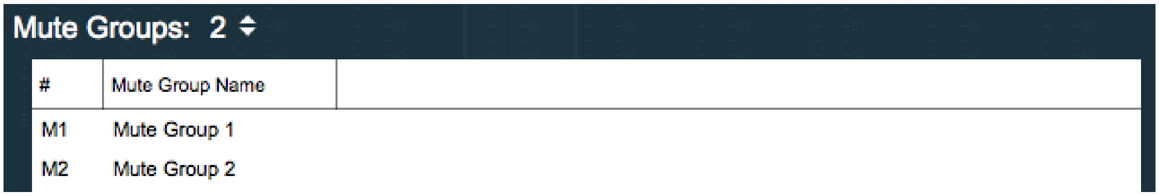


Figure 10.144: "Mute Groups:" configuration menu

Click in the **Mute Group Name** field to enter a name for the Mute Group.

For details on Mute Group operations, see the Mute Groups section of the Mixer Strip Controls breakdown.

The Analog I/O Pane

The Analog I/O Pane (default key command: "A") is provided as a central reference page for all analog inputs and outputs within a 3d domain.

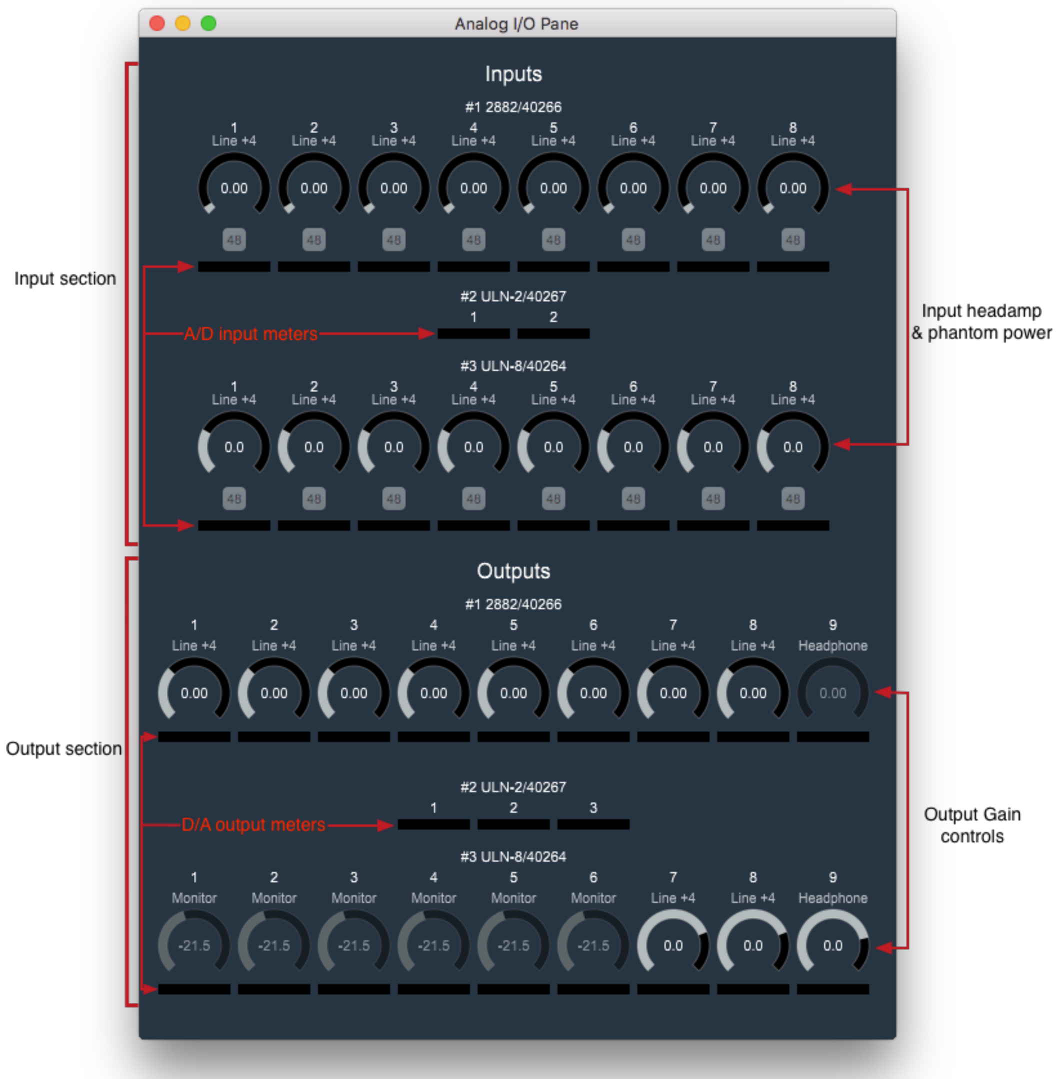


Figure 10.145: Analog I/O Pane

Analog Input headamps are shown for each box in the domain. These are the same as shown at the top of Mixer desk Input strips (aside from the 'Polarity Invert' control).

Input meters reflect the digital signal input level immediately from the A/D converters.

Output meters reflect the digital output level feeding the D/A converters.

Outputs assigned to Monitor or Cue Controller outputs (such as the box #1 Headphone output and box #3 outs 1 through 6 in the example above) are controlled specifically by their respective MC and Cue controls and appear greyed-out in the Analog I/O pane. Gain settings reflected in the UI are synced to the MC and Cue controls.

Since the ULN-2 has discrete analog controls on its front panel, there are no software gain controls.

USB: Satellite Computer Port (SCP) fundamentals

When your 3d box is connected and running as part of an MHLink domain, the USB port on the back of each 3d device becomes a **Satellite Computer Port** (heretofore known as "**SCP**").

Simply put, SCP turns each USB port in your MHLink daisy-chain into an extra digital I/O port, allowing a direct computer interface from each 3d box in your domain to a 'satellite' computer, iOS or Android device. (See The USB Class Audio connection for background on the mechanics behind this connection, if you haven't already.)

Here's how it works:

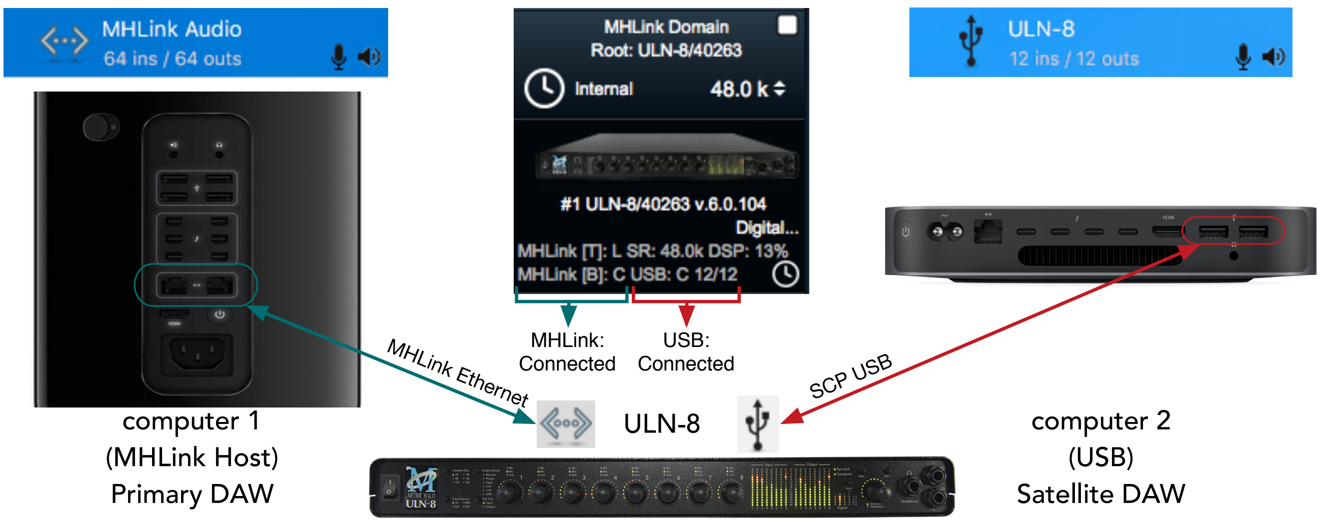


Figure 10.146: MHLink Host with Satellite computer

- On the left is the primary MHLink host computer (a Mac cylinder), in the middle is a ULN-8, on the right is a Mac Mini satellite computer.
- The primary host is connected to the ULN-8 via MHLink Gigabit-Ethernet, in this example with MH-Link domain I/O set to 64 channels (as shown in the blue macOS 'Audio MIDI Setup' graphic). The MIOConsole3d application is controlling the ULN-8 from this primary MHLink host computer.
- The satellite Mac Mini on the right is connected to the USB port of that same ULN-8. Note the USB connection in both the satellite Mac audio device graphic and the MIOConsole3d Unit Status Display shows "Connected" and set to 12 channels In and 12 channels Out.

So in this configuration, an audio app (such as a DAW) running on the primary Computer 1 will connect to MHLink with 64 channels of I/O, and those channels will show up in the MIOConsole mixer routing interfaces as "Host" routes 1-64 as usual.

A secondary audio application on Computer 2 connect to the USB "ULN-8" port with 12 channels of I/O.

Those channels are shown in the MIOConsole mixer routing interfaces as "#1 ULN-8/40263 SCP USB" routes 1-12.

The key term to keep in mind here is "Satellite". In all cases, the MIOConsole3d application on the MHLink Host computer maintains total control over every aspect of all 3d boxes in the Domain. Software running on the satellite computer will not be able to control the sample rate of the 3d connection. This is no different from connecting any outboard digital processor or DAW to a hardware digital mixing console: the Console controls everything.

The 3d Domain *is* after all a fully-functional hardware digital mixing/routing/processing environment - the only difference is you are controlling the hardware mixer with a computer UI (MIOConsole3d, with maybe a control surface or two).

The above connection model applies to as many boxes as you have in your MHLink domain. If you have four boxes on MHLink, you can do stuff like this:

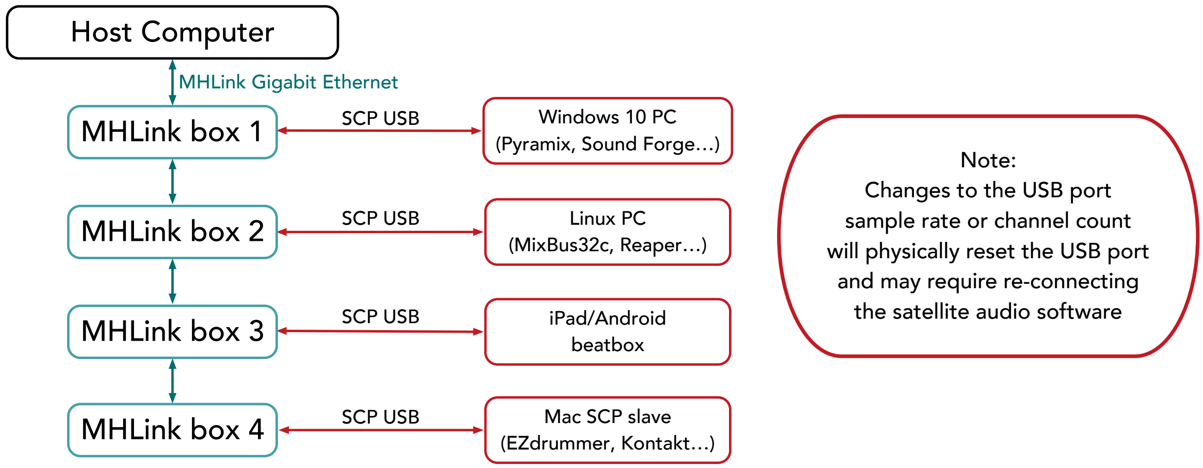


Figure 10.147: 4-box MHLink domain with four Satellite computers

Each satellite computer will show up as the SCP USB connections of the box to which it is attached - the Win 10 PC on box 1 SCP USB, Linux on box 2, iOS on box 3, etc.

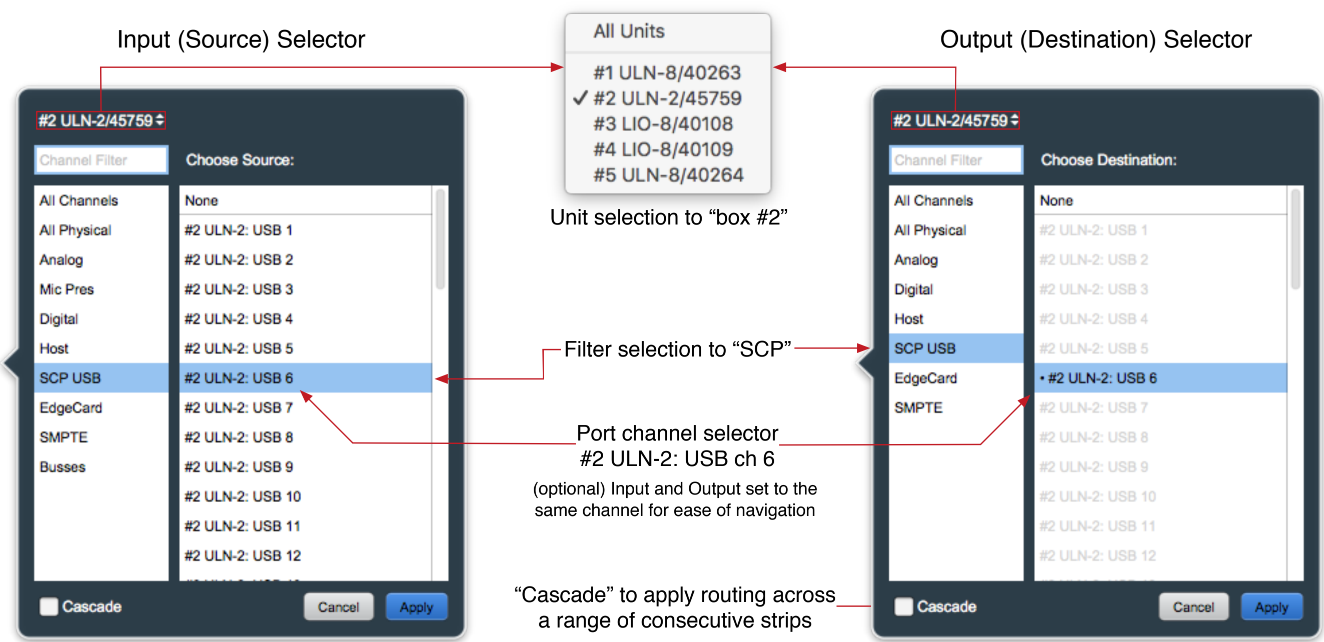


Figure 10.148: SCP USB Source and Destination routing

When assigning routes to and from USB ports, it is highly recommended to filter the routing UI to the desired 3d Unit in both the Source (input) and Destination (output) routing windows. It is easy (and frustrating) to be moving quickly and assign a Send to one box and the Return from a different box.

Further filtering by selecting “SCP USB” is helpful to keep scrolling through the lists at a minumum.

A few real-world things to keep in mind with SCP USB connections…

- Changing the host domain sample rate or USB port channel configuration will necessarily reset the 3d USB port. Audio software running on the satellite computer will lose connection to the USB port at these times. Most software will automatically reconnect on its own, but if not just manually reassign the software to the SCP USB port.

  To minimize the possibility of unstable audio connections, it is recommended that you quit or temporarily stop the audio connection of your satellite audio application, iOS or Android SCP clients when reconfiguring the USB ports from MIOConsole3d.

- As with a 3d USB Host computer connection, the maximum sample rate available via each USB port varies with the maximum channel count set for the USB connection. The number of input and output channels can be set independently. The larger channel count of the two will determine the maximum sample rate for the USB port.

  - 2-12 channels works at all sample rates (44.1k - 192k)
  - 2-24 channels works up to 2x rates (44.1k - 96k)
  - 2-48 channels works only for 1x sample rates (44.1k - 48k)

  Attempting to work beyond these maximums would overdrive the USB bus with inevitably undesirable results, often requiring a full reset of the USB bus. Due to the possibility of an external clock source being changed accidentally, the 3d hardware disallows SR settings that violate the USB I/O limits of any active SCP connections.

  This has the side effect that a connected USB I/O setting can lock out higher sample rates for the domain.

  So, if you find yourself unable to set the domain sample rate higher than your current setting, check the USB I/O channel count of each 3d box.

  If your target sample rate is 4x (176.4 or 192kHz), make sure all active 3d USB ports in the domain are set to 12 I/O channels or less.

  If your target sample rate is 2x (88.2 or 96kHz), make sure all active 3d USB ports in the domain are set to 24 I/O or less.

  Again, the above only applies to USB ports with an active connection to a satellite computer. Unconnected port settings are ignored by MIOConsole3d.

- Chances are, audio is not the only kind of data that will be bouncing around your computers’ USB buses. Best practice is always to keep critical audio connections isolated on their own USB bus. Use the macOS System Information **USB** section to display all USB ports and connected devices on your computer.

  Gaming mice and keyboards, USB control surfaces and especially USB storage and video peripherals can all have an impact on USB port audio stability. The degree of that impact will vary with each individual computer configuration and software applications.

  If you are getting tight on ports, a good-quality Thunderbolt dock can provide one or more isolated USB ports, as well as HDMI and (more importantly) a Gigabit Ethernet port.

  The good news is, it is very hard to actually damage a USB port, so feel free to try out and see what works and what doesn’t. At the worst, you can restore a stuck USB port by power cycling the 3d box.

  As always with this computer stuff, YMMV.

- **Special case: multiple MHLink domains**

  Unfortunately, there is currently no supported method to interconnect two MHLink domains across a computer connection. For the time being you will need to go "old school" to directly connect domains with analog, MADI, AES or optical audio I/O.

  If you have a need, you can do this, though:

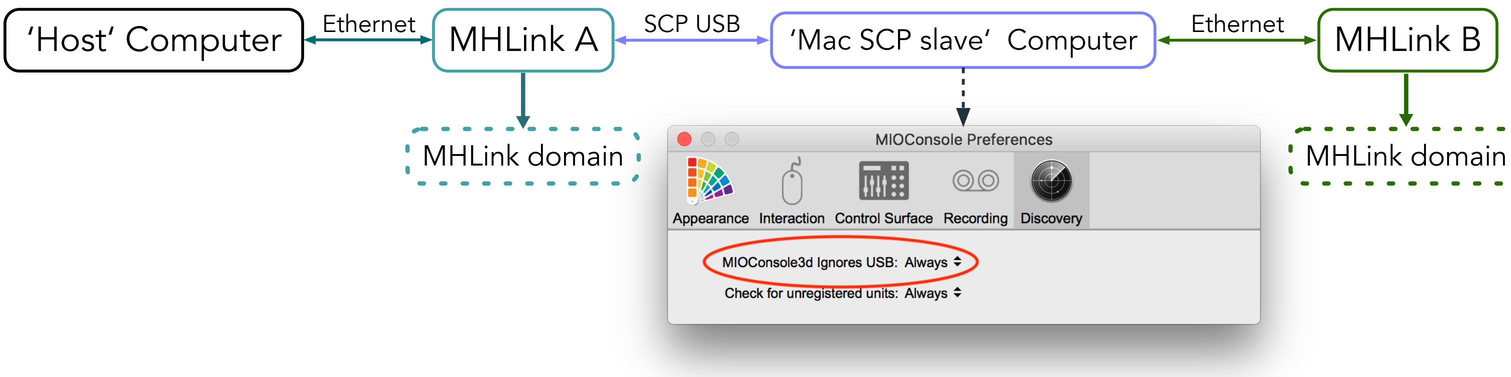


Figure 10.149: SCP connection between two MHLink host computers

In this scenario, we have the "MHLink A" 3d unit providing an SCP USB connection to second computer (called 'Mac SCP Slave') which is also host to its own MHLink domain. Both computers are running MIOConsole3d, but the 'Mac SCP Slave' Console3d prefs are set to ignore the USB port when booting up, meaning Console3d on that computer will *NOT* try to take control of the "MHLink A" box connected to the USB port. This setting (shown above) is found in the "MIOConsole Preferences: Discovery" pane:
'MIOConsole3d Ignores USB:' = "Always".

Note: This preference does not take effect until you quit and re-launch the MIOConsole3d application!

To be sure this is a fairly extreme case, but odd situations are common in studio life, and the possibility of needing an application or processor that resides on a computer in another room in the facility is not all that uncommon. For that one time you really need it, being able to take advantage of MHLinks' 100 meters between 3d units is worth adding one little preference.

> *Hint:* if you do not have the 'MIOConsole Ignores USB: Always' preference set, and you launch MIOConsole3d on the 'SCP Slave' computer while connected to the USB port, the slave computer will not function until you disconnect the USB plug.
>
> This is a temporary condition which does not damage the boxes in any way. In the worst case the 3d box may require a brief power cycle to reset the connection. Just set the 'MIOConsole Ignores USB: Always' preference, quit and re-launch MIOConsole on the 'SCP Slave' computer, re-connect the SCP USB cable. If necessary, power-cycle the 3d box and you will be up and running again.

11. Monitor and Cue Controllers

Overview

MIOConsole3d provides comprehensive monitor and cue facilities which take full advantage of the 3d mixer and MHLink multi-box environment.

The Monitor Controller is an independent entity which taps audio feeds from the mixing engine behind the scenes, simplifying routing. Likewise, Cues are a subsystem of the Monitor Controller which tap off the sources assigned in the Monitor Controller.

A Monitor Controller Source can be routed from any Aux bus, Group bus, the PFL/AFL Solo bus (in PFL/AFL mode), the Main mixer bus, or any hardware input. You can have as many Monitor Controller sources as you like.

The Monitor Controller Sources and Outputs can be any channel width from mono to 7.1.4. Multiple channel configurations for both Source and Output are fully supported.

Monitor Controller functions can be mapped and operated via key commands and EuCon or Mackie Control control surfaces.

Any feed routed to the Monitor Controller Source selector is available as a Cue Controller source. Cues can be slaved to follow the current Monitor Controller Source or can be specific to any Monitor Controller Source (such as a dedicated Cue mix bus).

Each Cue has independent source and output controls and their own talkback/listenback behaviors.

Each Cue can be any channel width from mono to 7.1.4.

Every Monitor and Cue output includes independent output gain modes (+4, -10, Monitor (or Headphone when appropriate)), level trim and Delay (in milliseconds) for each speaker channel. *Note*: Delay time is calculated to the nearest sample at every sample rate, so feel free to be brutally precise!

Every Monitor and Cue output path includes a dedicated in-line signal processing Graph, which can be used for anything from speaker crossovers and acoustics correction to spatial simulation or downmixes for IEMs. Monitor and Cue output Graph processors are placed in the signal path just prior to the final output gain stage.

Tooltips are available for all Monitor and Cue Controller selectors and commands - hover you mouse over a UI element for a second or two to reveal information about that button.

You can configure the Monitor Controller as a floating window so it always remains accessible, even when working in other applications.

With a LIO-8 or ULN-8 as your Monitor or Cue output, the front panel Mute, Dim and Monitor/Cans Volume controls automatically sync with the MIOConsole3d MC graphical interface. Same for the Mute and Dim buttons on the front panel of the 2882. (The ULN-2 has purely analog controls on its front panel.)

Monitor Controller output configurations can be saved and loaded as part of the overall MIOConsole3d mixer, or independently as a separate file. When importing a full MIOConsole3d mix session to your own tuned room, just open your saved Monitor file in the imported session and all your speaker assignments, trims, delay parameters and custom room tuning graphs will be applied to the session. The mix session with all routing and processing remains intact, but is now routing into your custom Monitor configuration. All Monitor Source inputs will be inherited from the imported session, just as if you had created the session in your own facility.

The Monitor Controller Interface

This section will walk through each control in the MC window. While the Monitor Controller window functions themselves are pretty straightforward, if you have not done so already please review the menu bar Monitor menu commands for important Monitor and Cue configuration commands and context. That section links back to here, so it is a painless and worthwhile little detour.

Figure 11.1: Monitor Controller controls

- **Source** Selector: Click on the Source you wish to monitor. MC Sources are created by default for each bus (or as assigned in Configure Mixer).
- **Speaker Mute/Solo** icons: Click on the Speaker icon you wish to mute. When muted, the speaker icon turns red.

  Solo Mode can be toggled by selecting the bottom entry of the menu bar "Monitor" menu:

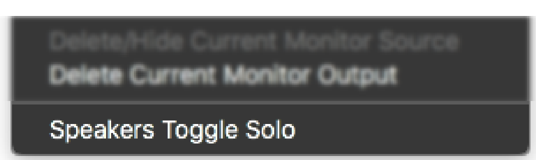


Figure 11.2: Monitor menu: MC Mute/Solo toggle

  In Solo mode, clicking the MC speaker icon will "solo" that speaker. When solo'd, the speaker icon will turn yellow.

- **Gain Control**: Yes, it's the volume control. Double-click on the numeric readout to enter a volume level. MC Gain is controlled by mouse scrollwheel/trackpad gestures, and when addressing a LIO or ULN-8, automatically syncs with the front panel monitor control of that unit. See Monitor Controller "0.0" reference for details about the MC volume control and output headroom.
- **"-"** and **"+"** (Decrement and Increment Gain): Clicking these buttons will subtract 0.5dB from, or add 0.5dB to the current volume level, respectively.

- **Lock** Gain: Click it, it turns purple, and it turns off the MC gain control (although the gain can still be controlled remotely).
- Folddown to **Mono**: All channels fold to Mono... stereo, surround, everything.
- **Dim**: Dims all speaker channels uniformly -20dB.
- **Mute**: Mutes the Monitor outs.
- **Output** selector: Right-click to open the output routing interface where you configure the speaker feeds for this Monitor Out. Name the Monitor Out in the text field titled “Name”, select a speaker configuration at “Type”, and assign the outputs, output trim and speaker time delay (if any) in the routing section.

The Cue Controller Interface

There are two important Cue Controller commands which appear in the menu bar Monitor pulldown menu:

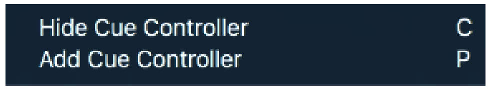


Figure 11.3: Monitor menu Cue commands

- **Hide Cue Controller** is a toggle to show/hide the Cue Controller interface, with the default key command "C".
- **Add Cue Controller** adds a Cue. Hit the default key command "P". As you add more Cues, they will stack progressively below the Talkback control header.

As always, the default key commands are editable under the menu bar "Edit / Key Commands" menu.

The Cue Controls interface itself consists of two sections.

The top section is for setup and control of the Talkback source and Listenback which will feed all Cue sends, plus Dim level settings (shown here at their default values).

Figure 11.4: Cue Controls interface with three Cues

The section below is the Cue sends stack (shown above with 3 cue sends set up). As you add more cues, new cues will add to the bottom of the cues interface.

Talkback controls

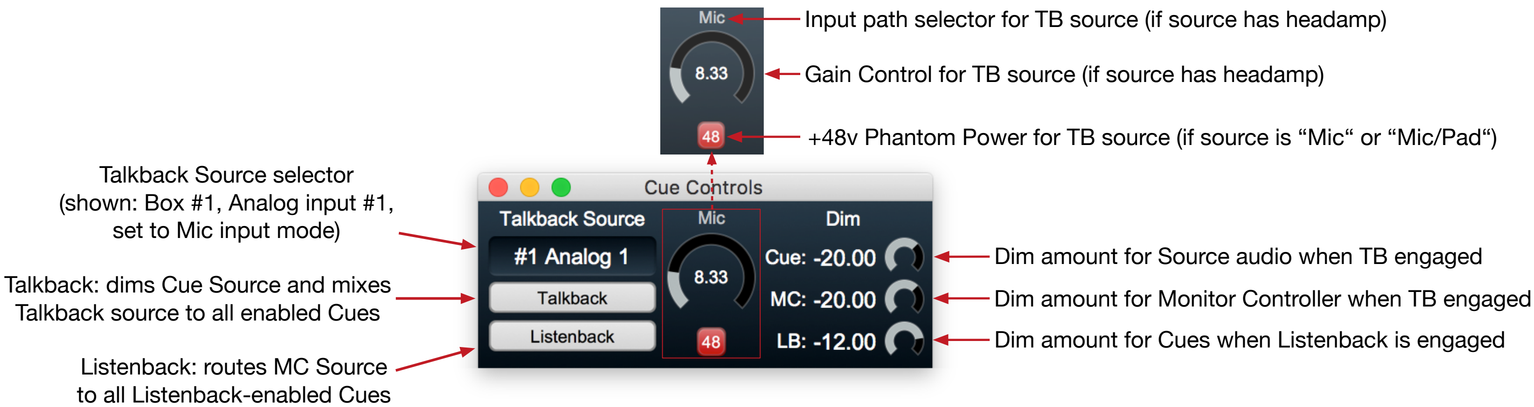


Figure 11.5: Cue Talkback controls

- The left side of the Talkback controls UI is where you select your talkback source, and control the Talkback and Listenback functions.
  - **Talkback Source** selector: Clicking this selector opens the usual MIOConsole3d Input Source window, from which you can choose any mono audio input. Traditionally, this would be a mic pre input, but there are no restrictions. The type of input selected here will determine what, if any, headamp controls appear to the immediate right.
  - **Talkback**: Clicking this button will dim the main audio stream and mix in the talkback signal to all enabled Cues. The Talkback button turns red when engaged.

    Key command and mouse-click activation of Talkback is latching. The default keyboard talkback trigger key is the letter "T".

    Click-and-hold the Talkback trigger will release Talkback when you release the mouse.
  - **Listenback**: This button routes the currently selected Monitor Controller Source audio to all Cues that have Listenback enabled. Note that this button latches whether engaged by mouse click or key command (the default key trigger being the letter "L").
- The middle of this controller is the Headamp section (shown labeled above the main graphic)
  - Talkback **Input path** selector: Most analog source selections will bring up a headamp in this space, and this control lets you select which analog path (Mic, line +4, line -10, Inst., etc.) to use for your Talkback source.
  - **Gain Control**: Sets your talkback source input gain.
  - **Phantom power** switch: If you have selected a Mic input, this button will enable phantom power. It turns red when phantom power is turned on.
- There are three controls on the right of the Talkback control section for customizing the talkback system to your particular workflow.
  - **TB Dim**: Sets the amount that the Cue source audio will dim when talkback is engaged. The default is -20db.
  - **MC Dim**: Sets the amount to dim the Monitor Controller output when talkback is engaged. The default is -20db. This setting is primarily used to avoid feedback from the Control Room monitors through the TB mic.

- **LB Dim**: Sets the amount that the Cue sends will dim when Listenback is engaged. The default is -12db. This is used to compensate for the level difference between the live mix and a more heavily-compressed bus mix.

Cue controls

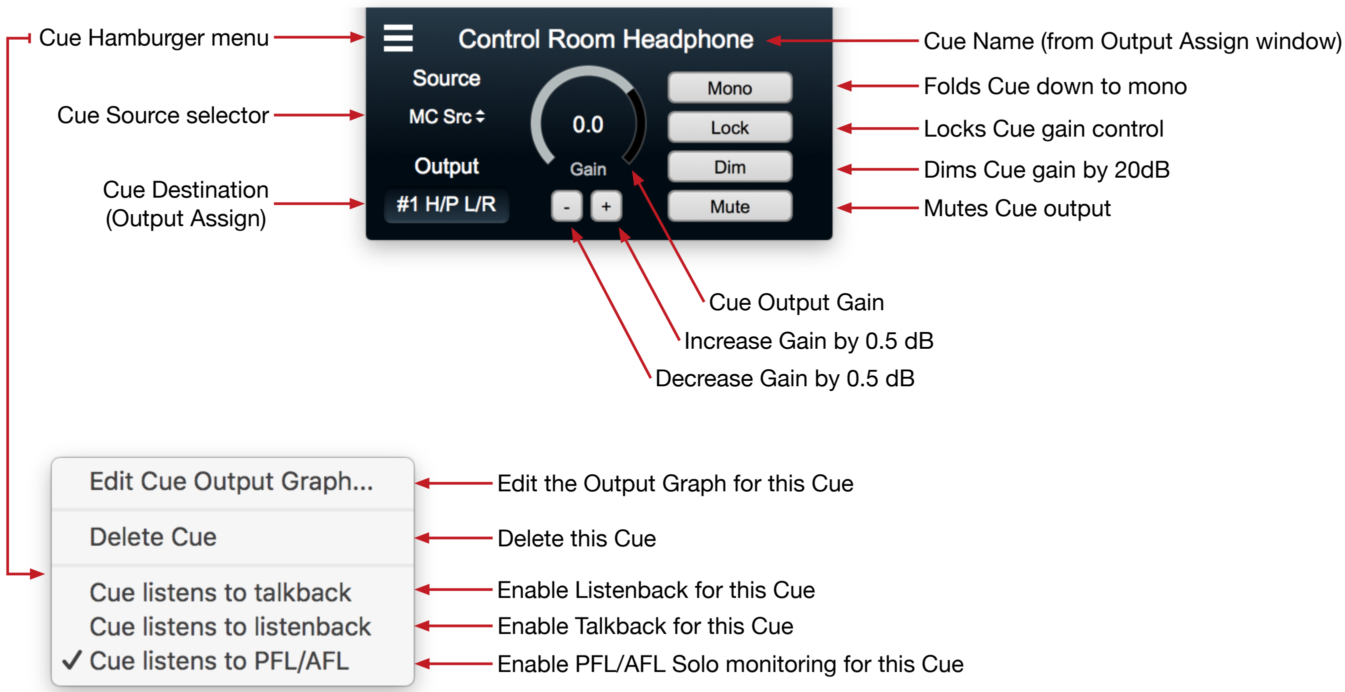


Figure 11.6: Cue Controls

Cue Hamburger menu

Every Cue includes a hamburger menu in its upper left corner (exposed at the lower left of the graphic above). The hamburger menu presents the following crucial functions:

- **Edit Cue Output Graph**: Opens the DSP processor Graph window for this Cue. The default window will open with no processors (as shown on the left). Right-click in the Graph window to open the processor selection window.

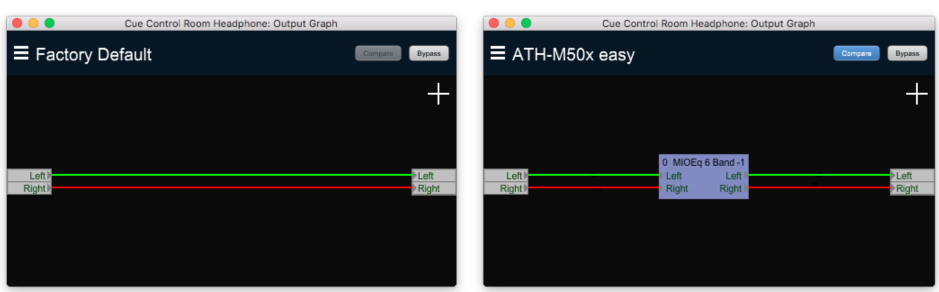


Figure 11.7: Basic Cue output graph with headphone EQ

In the example on the right, a basic EQ has been inserted to tweak the Cue mix for this particular musicians' taste.

See the DSP Graph Overview section of the manual for more information.

- **Delete Cue**: Deletes this Cue, releases any DSP Graph processes and frees up the assigned Cue Outputs for other uses.
- **Cue Listens to talkback**: routes the talkback to this Cue. When checked, activating the Talkback button will duck the main Cue audio and mix in the Talkback signal.
- **Cue listens to listenback**: routes the Listenback feed to this Cue. When checked, activating Listenback will send the currently selected Monitor Controller Source audio to this Cue.
- **Cue listens to PFL/AFL**: routes the output of the PFL/AFL Solo bus to this Cue.

  Note that this feature is inactive in the default "Solo-In-Place" solo mode. See Configure Mixer: Solo Modes for information on activating PFL and AFL solo modes.
- Cue **Source** selector: lets you choose any Monitor Controller source as the feed for this Cue.

Figure 11.8: Cue Source selector

  "MC Source" follows the current Monitor Controller Source selection, changing the feed to this Cue as the MC Source is changed.

  All other selections are static, eg. selecting "Main" will aways send the output of the "Main" bus to this Cue, regardless of the MC Source selection.
- Cue **Output** Assign: opens the output routing window for the Cue. Type in a name for the Cue in the text field titled "Name", and assign the outputs for the Cue in the routing section.

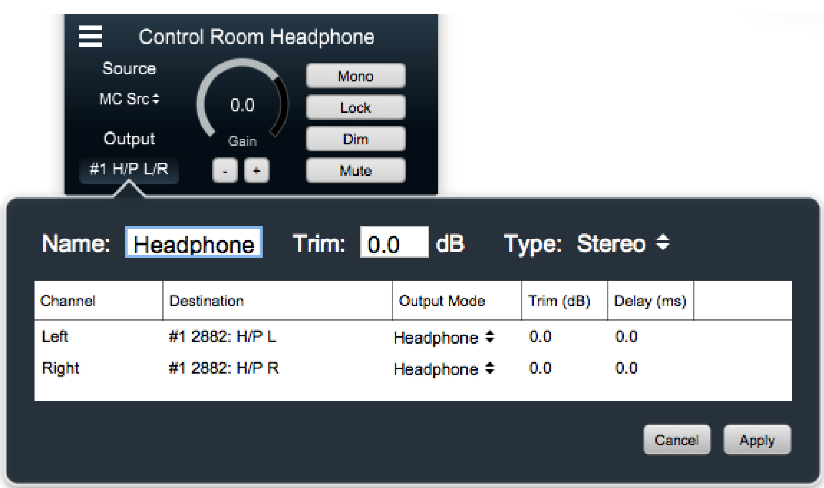


Figure 11.9: Cue Output assign

  This example shows the default settings for Cue #1: a stereo cue assigned to the headphone jack of box #1.
- **Cue Name**: Shows the Cue name you entered in the Cue Output assign window. As shown in the example, the name may be longer than is shown in the name entry field.

- **Mono**: Folds a stereo or multichannel Cue feed down to mono.
- **Lock** Gain: When selected, it turns purple and disables the Cue gain control.
- **Dim**: Dims the Cue output -20dB.
- **Mute**: Mutes the Cue.
- Cue output **Gain**: Volume control for this Cue.
- **"-"** and **"+"** (Decrement and Increment Gain): Clicking these buttons will subtract 0.5dB from, or add 0.5dB to the current volume level, respectively.

Monitor/Cue source management

The MC Source selection is also used as a dynamic input source by the Cue Controllers. It is always at the top of each Cue Source selection menu, listed as "MC Source". With "MC Source" selected, that Cue will play the currently selected Monitor Controller Source.

- Signal from the 3d Mixer Main Mix bus will always be the first selection in the 'MC Source' selector. This is a permanent connection from the 3d mixer.
- Group and Aux buses are assigned as Monitor Controller sources by default. Each bus may be Monitor Controller is managed from the menu bar Mixer: "Configure Mixer" window.

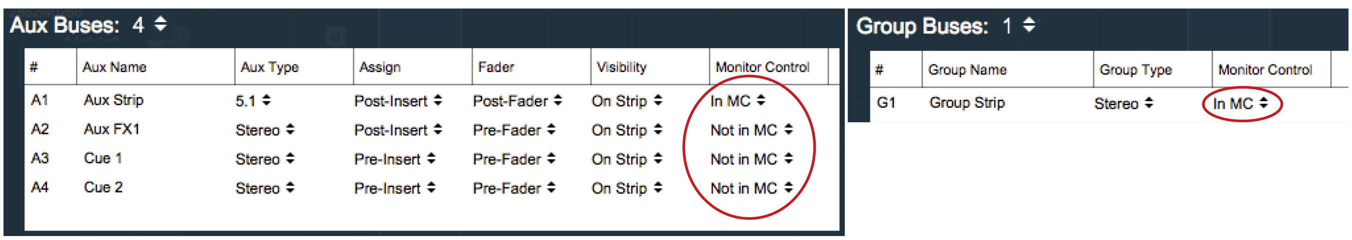


Figure 11.10: Assignment of bus outputs as Monitor Controller sources

The graphic above shows the Aux and Group 'Monitor Control' settings in the "Configure Mixer" window. In this example, Aux #1 and Group #1 are both selected as "In MC", meaning they are available as audio sources in the Monitor Controller source selector. Aux buses #2, 3 and 4 are "Not In MC" and therefore are not routed to the Monitor Controller source selector.

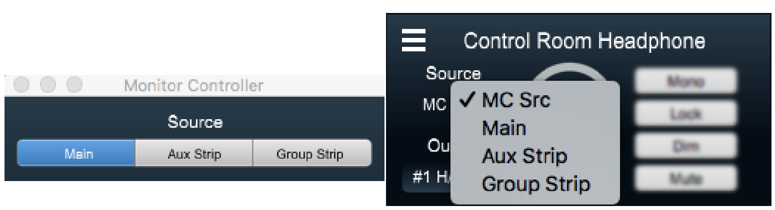


Figure 11.11: Monitor and Cue Controller sources as assigned

Please keep in mind: Since all Cue Controller inputs are provided by the Monitor Controller source selector, removing a Monitor Controller source will also remove it as a Cue Controller source.

- MC sources from Group and Aux buses and any hardware inputs, will be listed left-to-right in the order they are created.

Regarding large-scale multichannel surround monitoring...

With MHLink, all boxes in a domain regardless of physical distance from the host computer are completely time-aligned and synchronized to operate as a single physical unit. A practical upshot of this is, within the MIOConsole3d Monitor Controller you can assign the outputs of multiple 3d boxes to feed a multichannel immersive speaker array.

This is how 10 and 12-channel Atmos monitoring is supported in the current Monitor Controller. The physical returns from your Atmos decoder (whether via MADI, AES, analog or even SCP USB) are assigned as a Monitor Source input, and the analog outputs of two or more MHLinked boxes send to your various multichannel speaker feeds.

Default Monitor Controller Key Commands

⇧ = Shift	^ = Control	⌥ = Option	⌘ = Command

Command	**Key Sequence**
Show/Hide Monitor Controller	**M** (M)
Toggle Dim	⌘⌥^D
Toggle Mute	⌘⌥^M
Toggle Mono	⌘⌥^N
Volume Down	⌘⌥^↓
Volume Up	⌘⌥^↑
Show/Hide Cue Controller	**C** (C)
Add Cue Controller	**P** (P)
Enable Talkback (disabled until input assigned)	**T** (T)
Enable Listenback	**L** (L)
Select Monitor Source 1	⌘⌥^1
Select Monitor Source 2	⌘⌥^2
Select Monitor Source 3	⌘⌥^3
Select Monitor Source 4	⌘⌥^4
Select Monitor Source 5	⌘⌥^5
Select Monitor Source 6	⌘⌥^6
Select Monitor Source 7	⌘⌥^7
Select Monitor Source 8	⌘⌥^8
Select Monitor Output 1	⌘⌥1
Select Monitor Output 2	⌘⌥2
Select Monitor Output 3	⌘⌥3
Select Monitor Output 4	⌘⌥4
Select Monitor Output 5	⌘⌥5
Select Monitor Output 6	⌘⌥6
Select Monitor Output 7	⌘⌥7
Select Monitor Output 8	⌘⌥8

Table 11.1. MIOConsole3d default Monitor Controller key commands

Commands with no default key command have not been listed for brevity. *All* key commands are editable in the menu bar "Edit / Key Commands" window.

Monitor and Cue Controller default configuration

When you first connect a 3d box to your computer and launch MIOConsole3d, the Monitor Controller and Cue Controls windows will appear to the right of the main console window.

Figure 11.12: Default Monitor Controller and Cue Control windows

By default, MIOConsole3d will create a Monitor Controller for your domain. If you don't want to use the monitor controller, you can just delete the Monitor Output to return Analog outputs 1+2 to manual control.

> **Important note:**
>
> The Monitor Controller Source Selector is always in-line, routing audio to the various Cue Controller Listenback and MC Source feeds. So even if you re-assign your Monitor Controller outputs for other purposes, the MC Source path is still in the Cues loop. This lets you use keystroke commands and outboard EuCon or MCP control surfaces to select Cue sources with the Monitor Controller hidden.

Monitor Controller default configuration

At the top of the Monitor Controller window are the MC Sources. In the default configuration, each bus from the default mixer is available as a Source.

- Hold the mouse cursor over the blue "Mon Out" button at the bottom of the MC. A tooltip will appear, showing the current (default) setting for your Monitor Output: Box #1, Analog outputs 1 and 2.

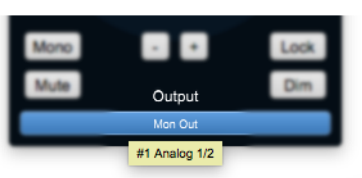


Figure 11.13: Monitor Controller Output tooltip

The default Monitor Controller output destination will always be to analog channels 1 and 2 of the 3d box connected directly to your host computer (i.e.: the "root" box, which is always designated as box #1). In the case of a ULN-2, which has a discrete stereo monitor output stage built-in, the default routing tooltip will show "#1 Monitor L/R".

- **Summary of Monitor Controller default settings**
  - Monitor Controller window is visible, but not "floating"
  - Monitor Controller is Muted (LIO and ULN models only)
  - Monitor Controller Sources: Main (selected) and stereo Aux mixes 1 through 4
  - Monitor Controller Output Graph: passthrough (no in-line processing)
  - Monitor Controller Output: stereo (like the Main bus), assigned to box #1 analog output channels 1/2, Monitor mode (LIO and ULN models only), with output Trims and Delays at 0
  - Solo Mode: Solo-In-Place

    Monitor Controller Slaves to PFL/AFL: ON (disabled in "Solo-in-Place" mode)

-

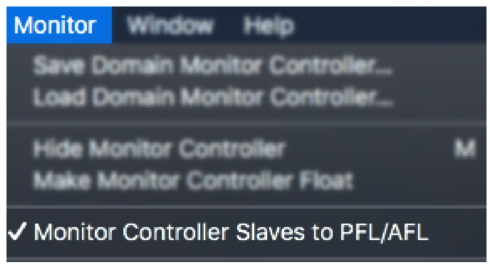


Figure 11.14: Menu bar "Monitor" menu: "Monitor Controller Slaves to PFL/AFL" control

Folks who spend most of their time in DAWs working in 'Solo-In-Place' mode should take note of this feature. This setting is found in the 'Monitor' menu bar pull-down.

When the Mixer is in one of the PFL/AFL modes, this setting makes the Monitor Controller play back the 'Solo' bus whenever Solo is engaged. This is the most common default mode for PFL/AFL use on a hardware console.

De-selecting this control leaves the MC Source uninterrupted when Solo is engaged.

This function is also available separately for each of the Cue Controller sends, so you could have, say, a Cue or two slaved to the PFL/AFL Solo bus, while not interrupting the control room 'Main' bus mix for a producer.

Cue Controller default configuration

The Cue Controller window defaults with the Talkback Source for Cues unassigned.

There is one Cue Controller (named "Control Room Headphone") set up to source audio from the currently selected Monitor Controller Source, with the output routed to the box #1 headphone jack.

Summary of Cue Controller default settings

- Cue Controller is visible
- Cue Controller Source: stereo, 'MC Source' (Cue source follows Monitor Controller source selection)
- Talkback Source: unassigned (which is why the "Talkback" trigger button is grayed out)
- Cue Controller Graph: passthrough
- Cue Controller Output: stereo, assigned to box #1 headphone output L/R

Note: Since ULN-2s have dedicated a stereo Monitor route which goes to both the front panel headphone jack and stereo TRS monitor outs on the back panel, only a default Monitor Controller output is established. The default Cue Controller for a new ULN-2 root box is unconfigured.

Freeing up Monitor Controller outputs

If your workflow does not require the best-sounding Monitor Controller on the planet, highlight the Monitor output you want to remove, then select "Delete Current Monitor Output" from the menu bar "Monitor" menu.

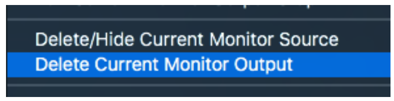


Figure 11.15: Disabling Monitor Controller Outputs

This will immediately remove the MC output and make the outputs available in the output routing dialogs.

Editorial note for LIO-8 and ULN-8 owners: Before re-purposing the Monitor outputs, you owe it to yourself to at least audition the MIOConsole3d Monitor Controller one time.

Just connect the analog 1 and 2 outputs to your speaker amplifiers and listen to a few familiar songs. If you use eq or DSP acoustic correction, first try monitoring with the room correction hard-bypassed (LIO-8/ULN-8 connected direct to the amps), then again with your processing in-circuit.

Monitor Controller "0.0" reference

The Monitor Controller determines the monitor gain headroom above "0.0" in the MC volume control by how much gain it can apply before a full scale digital signal would clip at the converter. The available headroom varies per the analog stages of each 3d device model.

On the ULN-8 and LIO-8, there is an analog domain gain control block after the converter that supports 30 dB of gain above monitor level.

When the MC determines that it has access to those gain blocks on all of the output channels of the output path, it knows that it can add up to 30 dB of gain in the analog domain without introducing any digital clipping. So that causes the zero point of the monitor knob to allow you to add 30 dB of gain. When you add the -10 trim, that moves the zero point down 10 dB and gives you 40 dB of gain above zero that can be done in the analog domain.

When one or more of the channels in the output path does *not* have the analog domain gain stage available, then the MC gain is done in the digital domain and the maximum gain that can be applied in the digital domain without running the danger of having a digital clip is 0 dB. So the maximum gain on the MC in that case is 0 dB and when you add the -10 trim, that moves the zero point 10 dB down and gives you 10 dB of gain before digital clip.

This will be the case when you have the output Destination set to "None", to a digital output or to a 2882 or ULN-2 analog out.

12. MIOConsole3d Session

Preliminary Documentation

Figure 12.1: The sections that follow are preliminary and in-progress. Please look for frequent manual updates!

Session Origins

Recording and playback capabilities have been a part of Metric Halo products since 1997 with the Capture Panel in the Spectrafoo Audio Analysis and Metering software package, shown here with gain control, fades, loop mode and Spectral History engaged.

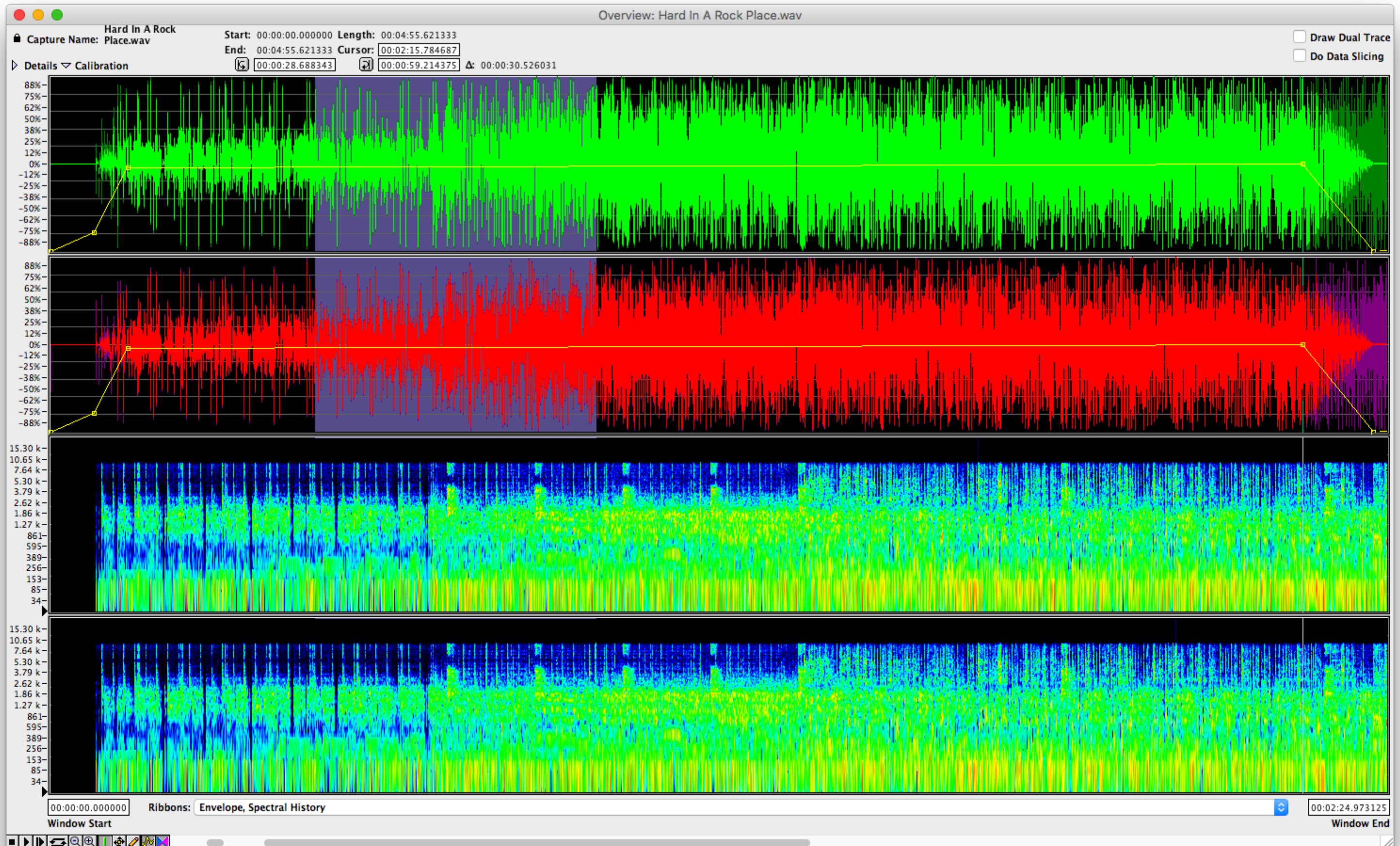


Figure 12.2: Spectrafoo Capture Panel

In 2002, elements of Spectrafoo's Capture Panel found a new home as the multitrack MIOConsole Record Panel, included with each Mobile I/O 2882, ULN-2, and later with the second-generation ULN-8 and LIO-8 devices.

Renowned for its superior reliability, sound quality and ease-of-use, the MIOConsole Record Panel remains a go-to recording tool for high-end mastering, audio preservation, live recording and technical analysis engineers today.

In fact it was the Record Panel that had accrued the most extensive feature request list from Metric Halo users.

OK, wish(es) granted.

With 3d, the enhancements to the recording system became so extensive that we have renamed the *Record Panel*; it is now called **Session**.

Session: Not Just Another DAW

Figure 12.3: MIOConsole3d Session

The new **Session** feature set originally was driven by the requirement to add back support for playback to the Record Panel. In addition to basic playback there were a number of longstanding feature requests for playback enhancements that we felt we needed to address.

- The ability to see multitrack waveform overviews for the audio playing back,
- The ability to cue playback within a given take, against the waveforms,
- The ability to cue between multiple takes without having to load the takes into some third party DAW, and
- The ability to generate marks during recording and/or playback that are correlated with the recording for denoting cue points, recording events or musically relevant events.

As we worked through the process of implementing playback with these requirements, we determined that in order to provide the required features, we needed to implement a full-featured internal playlist playback engine.

As soon as we implemented the internal playlist playback engine we realized that we had all the underlying technology required to support the other major recording related feature that has been at the top of the list of feature requests for the Record Panel for years: **Overdubbing**.

The same underlying Engine functionality also supports basic editing functionality including segment drag, drop and copy, timeline selections, cut copy and paste, all with integrated fades.

Notably, since **Session** is part of MIOConsole3d and has deep control of the hardware, overdubs are performed with dynamic monitor switching on record enabled channels in the hardware mixer with crossfades on punch in/out, and with the ability to monitor both recorded tracks and live inputs through zero-latency DSP processing.

While the Recording Engine has been enhanced to support the new recording features in the system, including a connection to the session timeline while recording, it still fundamentally supports the original *Take based* recording model that enables easy import of takes into other DAW systems and also enables take-based workflows simultaneously with the session based workflow added in this release of MIOConsole3d.

First and foremost, **Session** is not intended to be a full-featured DAW; rather it is intended to be the best tracking and capture tool on the planet.

The combination of zero latency monitoring with signal processing, automatic routing, pre-recording, overdubs and editing allows you to optimize your multichannel recording process and concentrate on the music, much like you would in a large hardware-based multitrack recording studio.

Whether you are doing long form live recordings, take-based recordings, or full-blown multipass recordings with overdubs, **Session** will allow you to capture quickly and easily with ultimate fidelity and stability.

Session's level-based trigger, loop-recording, and Autopunch functionality makes it just as easy to record by yourself as it is to record for others.

We have included track export functionality so that when you are ready to transition from tracking to production or mix down you can easily migrate your tracks to the DAW(s) of your choice.

Session: How it works

The Session Tracks Overview presents a waveform history view of every channel being recorded or edited, organized horizontally as Tracks, with each Track interfaced directly to their Mixer Strips in the MIOConsole3d Mixer. Here's the nutshell view of how the Mixer Strip to Session Track work together:

The Session Track layout is synced to your Mixer desk layout, so the left-most strip on your Mix desk will be the topmost Track in the Overview. If you move a strip (or strips) position in your Mixer, the corresponding track will relocate to reflect that new position(s) in the Session. Likewise, relocating tracks in the Session will apply to the Mixer strips layout. Simply select a strip or track and drag to reposition. Moving multiple strips/tracks at once using Shift- or Command-key selection is fully supported.

When you record-enable a mixer strip, the system automatically sends the *pre-insert* audio input to the Session record engine using the next available **To Host** route. The audio passes into the strip, through the inserts, panner and fader to your bus, and the bus goes to wherever you route it as usual.

The audio that is recorded is the signal that is the input to the strip **without the insert processing or the panner or fader gain**.

When you play back from session, the audio from the track is auto-routed back to the input point of the associated strip, and so passes through the inserts, panner and fader and then is summed to the bus, etc.

This has the effect of playing back the audio into the mix such that the mix you set up when recording is the same mix that is being used for playing back.

Since the recorded audio is dry (i.e. un-processed), you can tweak your mix or add or change processing without having to worry about double-printing effects.

Recording both pre-insert and post-insert simultaneously is supported, but the Session will only include the pre-insert audio. The post-insert signal will be recorded to the Session Record folder with the same filename as the pre-insert audio, but with the suffix *[POST]* added to the filename. If you wish to utilize the POST file in the session, you can drag it from the Finder into a free playback track for editing and insertion into the mix as desired.

While you can record buses into Session, bus tracks do not play back. Because bus master strips are fed by the mixer and not by a routable hardware input, there is no routing point for playback of audio on bus track. If you wish to play back the audio you have recorded onto a bus track, you will need to manually move the recorded segment to an open track associated with an input strip.

Assigning Host or hardware Input routes to the Mixer behaves as it always has. Changing the Input strip Source will not affect the ability of the strip to play back its Track audio, and an input strip does not need to have a physical input assigned in order to play back from its track. The “tape return” routing happens behind the scenes automatically and is independent of the Input strip Source route.

For overdubbing, you can record-arm the track to be recorded to, start playback (with a count-in, if desired), and the track will play back the recorded audio from the track until you drop into Record mode, at which point the input to the Mixer strip will automatically switch to the assigned Input Source (the live instrument being recorded). This is basically the traditional tape-style “Auto-Input” or “Auto-Record” punch-in behavior from days of yore.

The overdubbing toolset has been enhanced and optimized for everything from improv and double-tracking to compositional layering and beat-building in a band performance/studio jamming context. Tracking and dubbing fine-tuning controls can be found on the Session Widget Bar.

Channels to Strips to Tracks

As with all things in the audio world, everyone comes from a different background and may have different view of the audio recording and production process. One person's view of audio signal flow might be heavily influenced by their background in analog multitrack recording, another might have a more non-linear view. So that everyone is on the same page, we present a quick primer on the operational flow and terms used in the context of the 3d Session:

- **Channel:** Any monophonic audio stream. A mono mic feed carries one channel. A stereo keyboard input is two channels, a 5.1 submix bus is six channels, etc. The Tracks Overview shows the audio waveform of every channel being recorded. Audio source channels are organized, routed and processed in the Console3d Mix desk as "Mixer Strips".
- **Mixer Strip:** Strips are the vertically-oriented audio routing and processing interfaces which reside in the Mixer desk. Each strip can contain up to eight channels.

  Mixer Strips can be Input strips, Aux bus strips, Group bus strips, the Main bus strip (and the Solo bus, if you're in PFL/AFL Solo mode). All Mixer Strips are eligible to be recorded, but only audio recorded on an Input strip will have an automatic playback route. All mixer strips automatically have an associated track in the Session, but only Input mixer strips have an input routing point to play back from Session Tracks.

Strip Type	Record to Track?	Playback from Track?
Input strip	yes	yes
Aux Bus	yes	no
Group Bus	yes	no
DCA	no	no
Main Bus	yes	no
Solo Bus	yes	no

  Table 12.1. Mixer Strip Type: Session Integration

- **Track:** "Tracks" directly correspond to, and are named by, the Mixer Strip sending audio to and playing back from its Track. A Track can not exist without a corresponding Mixer Strip (it would have no way to communicate with the Mixer!). To create a Track, use the Mixer "Add Input Strip" (**⇧⌘A**) and a new Track will be created for that strip.

  As far as audio routing is concerned, the relationship between Mixer Strip and Session Track is analogous to a channel strip on a hardware recording console and the tracks on a 2" 24-track tape deck... the console mixer strips all do the processing and routing to and from the multitrack deck, and the deck records to tracks and plays them back.

  That said, there is a small difference in that Tracks in Session can be multichannel. When you record to a Track, the audio recorded inherits the channel width of the associated Mixer Strip (e.g. mono strips record mono files, stereo strips record stereo files, etc.).

  Each Track contains audio segments and waveforms representing the audio files containing the audio itself, which can be any channel width from mono to eight channels interleaved. You can mix channel widths within a single track by dragging in files of a different channel width, and the Session engine will do its best to accomodate. Stereo files dragged into a mono Track will be summed to mono, and stereo files dragged to a 5.1 track will play as channels L and R. Mono files into wider Tracks will send the audio to all the channels of the strip.

  For those of you coming from a "all recorded files are mono" background, please do not be freaked out that the file sizes for multichannel tracks are vastly larger than mono tracks of the same length when you look in the Take folder. Writing to interleaves is hugely helpful in keeping session recordings organized and making sure all multichannel tracks stay grouped together in the correct channel layout.

- **Track Lanes:** Each Track can contain multiple lanes. Track lanes allow multiple takes to be recorded into the same track, edited, crossfaded, comped, and mixed within the track itself, without taking up extra mixer strips.

  Lanes are created automatically whenever audio segments within the same track overlap on the timeline. All overlaps and crossfades are summed within the playback engine before being sent to the 3d Mixer. All segments have independent gain with full control of In- and Out-fades.

  There is no limit to the number of simlutaneous overlapping crossfades or segments.

  You can expand or collapse the Track Lanes view with the default key command: **L**. This command currently opens all track lanes in the Session overview, not just one or two at a time.

- **Selections:** There are two types of editing selections you can make in the current Session UI; time range selections, and segment selections. Both time range and segment selections may be applied across any number of Tracks.

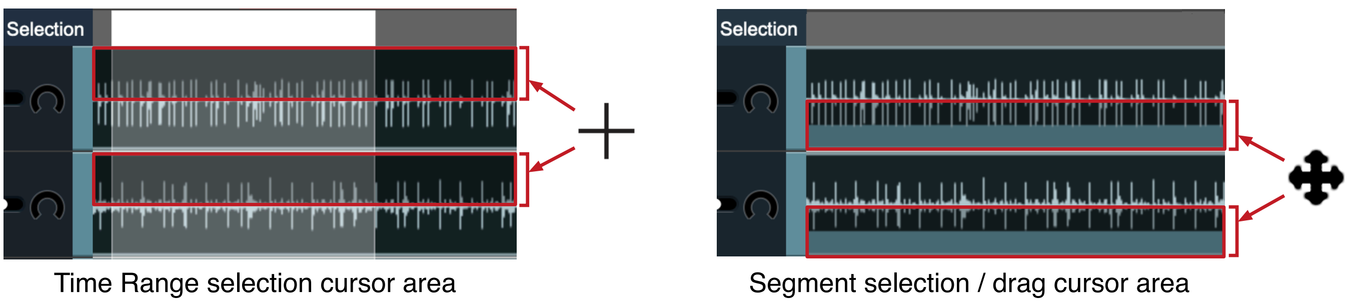


Figure 12.4: Time Range Selection (left) - Segment Selection (right)

Session Capabilities

For the merely curious, here is a brief overview of the current **Session** feature set.

- Each domain has its own session
- Time-based, multi-track, multi-region session for recordings, saved as part of console file
- Each track supports unlimited Audio Segments
- Each Audio Segment has an adjustable:
  - Fade-in & Fade-out
    - Multiple fade types
    - Adjustable Fade length
  - Gain
  - Mute
- Each track supports an unlimited number of Audio Segment overlaps
- Session supports time signature and tempo to provide a musical timeline
- Session can calculate tempo from Loop, Autopunch, and Selection regions
- Session provides the user-selectable timeline rulers:
  - Bars & Beats
  - SMPTE Time Code
  - Time
  - Sample
- Session has loop points for loop Playback and Recording
- Session has in and out points for Auto punch Recording
- Session supports musical grid
  - Bars down to 1/64 notes; support for triplet and dotted spacing
  - Related to Time signature and Tempo
  - Loop, In/Out, Playhead and segments are snapped to grid when enabled
- Session supports an unlimited number of marks
  - Each mark has:
    - Start Time
    - Duration
    - Name
    - Reason Field
    - Notes Field
  - Marks with duration can recall loop points
- Session includes a metronome locked to Time Signature and Tempo with a dedicated strip in the mixer
- Session Playback Engine
  - Plays each track to its own strip in the mixer
  - Automatically routes playback to input strip tape returns
  - Supports Loop playback
  - Supports Count off
  - Supports instant cueing to location
  - Supports instant cueing to mark
  - Supports delayed cueing to mark when currently playing a loop
  - Multichannel files are down-mixed when added to narrower tracks
  - Mono files are up-mixed when added to wider tracks
  - Responds instantly to Changes, even while in loop playback.
    - Cueing
    - Loop-point changes
    - Segment Gain, Mute, Fades postion
    - Adding and removing segments
- Session supports simplified editing model including
  - Auto-editing modes for drags and punches
    - Split and Add (splits existing segments on overlap, adds new segment)

    - Split and Remove (splits and removes existing segments on overlap, adds new segment)
    - Split and Mute (splits and mutes existing segments on overlap, adds new segment)
    - Add (adds new segment)
  - Auto-edits support user selectable cross fade times
  - Multiple Segments can be selected
  - Time regions can be selected
  - Segments can be split, topped, and tailed at the playhead or selection
  - Segments can be split or trimmed to the loop points or selection
  - Segments can be named
  - Segments can be moved or copied
  - Segments can be snapped to the grid
  - Segments can be removed
  - Selections can be cut or copied to the clipboard
  - The clipboard can be pasted at the playhead or selection point
  - Media file for Segment can be replaced
  - Audio files can be revealed in the Finder from the Segment
- Import/Export
  - Segments can be exported as new audio files (with trimming, gain and fades applied)
  - Individual tracks can be exported as new audio files
  - Individual tracks can be exported as new audio files, trimmed to loop points
  - All tracks can be exported as new audio files
  - All tracks can be exported as new audio files, trimmed to loop points
  - The Selected area of the Session can be exported as new audio files
  - Folders of existing audio files can be imported into new tracks
  - Folders of takes can be imported into new tracks, with takes loaded successively on new tracks
  - Audio files can be dragged directly into tracks in the **Session** window
  - Multiple Audio files can be dragged and dropped onto successive tracks
- File management
  - **Session** detects when media files are not found
    - All playlist information is maintained
  - Intelligent multi-file find tool to locate missing files
    - Supports finding files that have been renamed or moved
    - Found files are immediately reattached to playlist
- Recording
  - Record-enabled tracks record to record folders (with mirroring)
    - User specified file and take folder naming rules
    - Take number management
    - Tracks are recorded to files with natural width
      - Mono tracks record to mono files
      - Stereo tracks record to stereo files
      - 5.1 tracks record to 5.1 files
      - etc.
    - Session always records pre-insert, and plays back through inserts
  - It is possible to record onto bus tracks
    - Bus tracks do not play back
    - If you want to play back a bus track, you need to move the file to an input track
  - Mixer now supports “tape” returns and is transport-state aware
    - When transport is stopped, mixer monitors inputs
    - When transport is in Play
      - For tracks with Segments
        - Not record enabled
          - Mixer monitors tape returns
        - Record Enabled
          - Mixer monitors mix of input + tape returns

    - When transport is in Record
        - For tracks with Segments
            - Not record enabled
                - Mixer monitors tape returns
            - Record Enabled
                - Mixer monitors input
    - Monitoring follows transport state dynamically, providing cross-faded punch in support
- Recorded files continue to support pre-roll/pre-record
- Segments are added starting at trigger time, and ending at end of record
    - Segments are added using the specified Punch Mode (described above) and cross fade Time
    - Audio file may extend before and after segment; additional material can be revealed by dragging segment trim points
    - Segments take into account latencies when recording and are aligned with the playback timeline
- Recording supports multiple modes:
    - **Take**: Recording starts at current playhead time, but does not start playback. Playhead is advanced at each new take
    - **Count In: None** - Playback and Recording start at the current playhead position
    - **Count In: 1-4 Bars** - Playback starts specified number of bars before the current playhead position, and recording punches in when the playhead reaches its original position.
    - **Count In** mode uses the loop start or punch-in point as the starting point if one of those modes is active
- Loop Recording Modes
    - **Loop Mode**: Recording starts when triggered, and punches out at end of loop and is rearmed for next loop pass; this means all new material is recorded within the boundaries of the loop
    - **Loop to Linear Mode**: Playback follows the specified loop, and recording starts at the playhead position when triggered, but continues recording linearly while playback loops. This is great for recording non-looping material against a looped basic track.

Session UI: Quick Reference

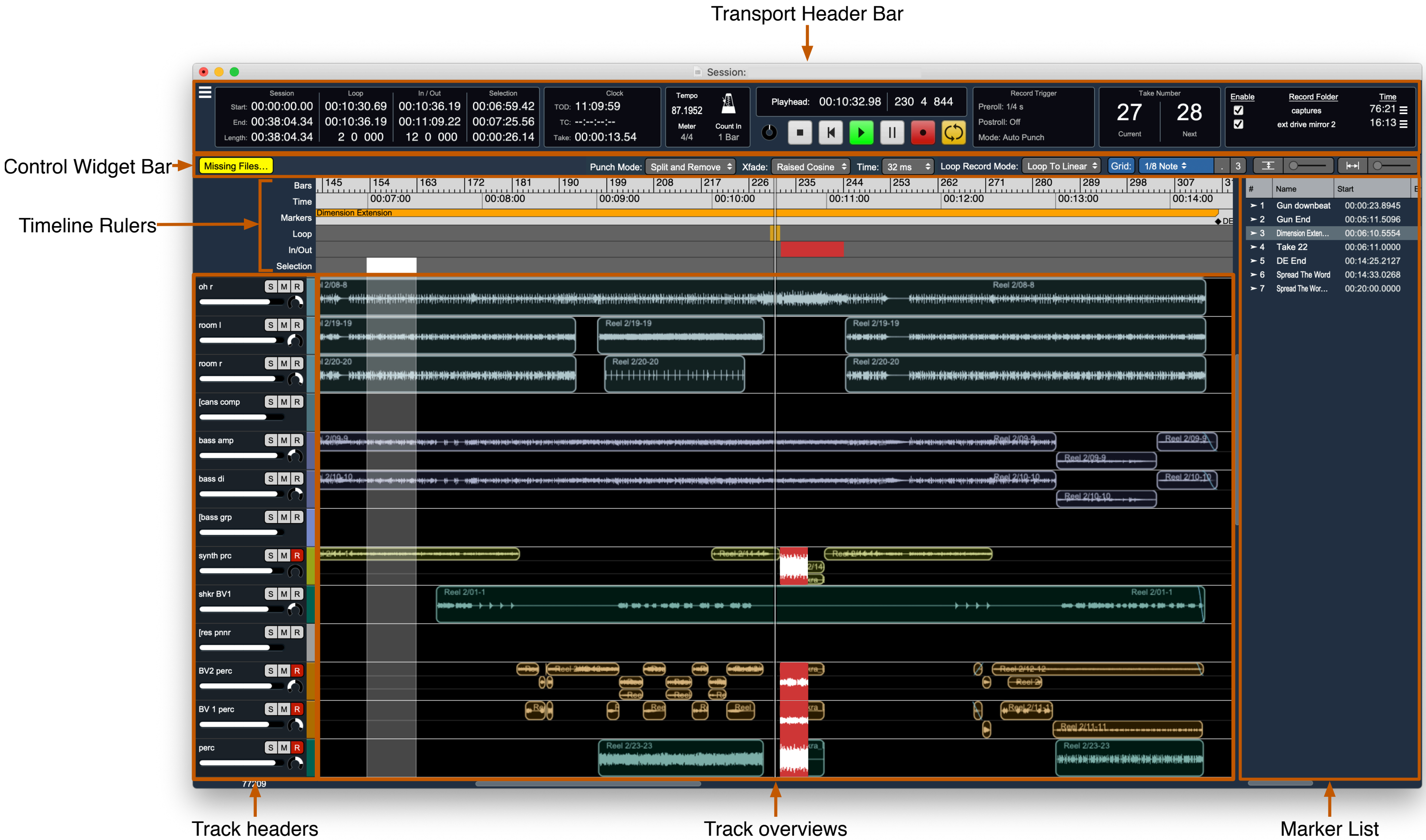


Figure 12.6: Session window area map

The **Session** window is comprised of 6 main areas, as shown above.

The Transport Header Bar contains the Session Hamburger menu (used to control which UI elements are shown in the Transport Header Bar), and a variety of transport related UI elements, including:

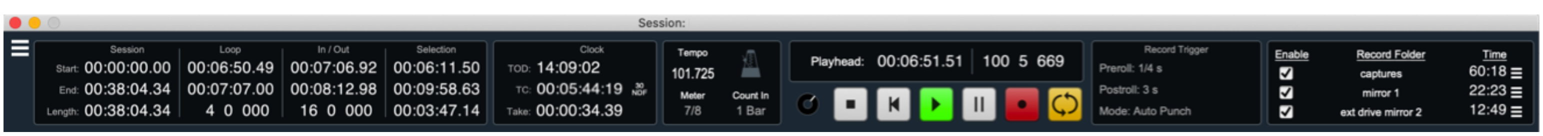


Figure 12.7: Session Transport Header Bar

- Time Readout block, with:
  - Session Start, End, and Duration
  - Loop Start, End, and Duration
  - Auto Punch In, Out, and Duration
  - Selection Start, End, and Duration
  - Each of these time readouts can be set to one of the following formats:
    - Time (HH:MM:SS.ss)
    - Time Fractional (HH:MM:SS.ssss)
    - Bars and Beats
    - Samples

- Clock Readout block, with:
  - Wall-clock Time (e.g. Time Of Day)
  - LTC TC reader Time Code
  - Duration of current (or last) take
- Tempo / Meter Control block
  - Tempo control
  - Time Signature control
  - Metronome Enable
  - Count off Mode
    - Take (Record does not start play)
    - None (Record does start play, but no count in)
    - 1, 2, 3 or 4 bars (Count-in the specified number of bars before punching into record)
- Playhead Position Readout
  - Left side shows postion in HH:MM:SS.ss
  - Right side shows position in Bars and Beats
- Transport Controls
  - Record Ring Buffer status
  - Stop (stops all play and record transport - ends any recording in progress)
  - Return to Zero
  - Play
  - Pause
  - Record (arms recording according to trigger mode)
  - Loop Playback Enable
- Record Trigger Mode block
  - Preroll sets the pre-record time
  - PostrollPostroll sets a delay time from an automated de-trigger command to the end of recording
  - Mode sets the record trigger Mode
    - Manual: Trigger/de-trigger on arm toggle
    - Level: Trigger/de-trigger based on input level
    - TC: Trigger/de-trigger based on LTC TC lock
    - Auto Punch: Trigger/de-trigger on playhead position
- Scene/Song Block
  - Enter a four-character Scene code per US/UK iXML film/video systems, or...
  - Enter a Song title for music workflows
- Take Block
  - Basic number count. The Take number auto-increments at each record stop
  - Take number resets to 1 when a new Scene/Song is entered
  - Left side shows current/last take number
  - Right side shows next take number (click to enter new take number)
- Slate Control Block
  - Basic number count. Slate auto-increments at each record stop but only resets manually. Used as an alternate differentiator from Take
  - Left side shows current/last slate number
  - Right side shows next slate number (click to enter new slate number)
- Session Metadata Block
  - Project field: enter Project name (mirrors Project field in Record Preferences pane)
  - Director/Producer/Client:
  - Engineer:
  - Soundroll:
- Record Folder block
  - Lists Enabled Record folders, along with available time

The **Control Widget Bar** contains controls for editing, overdubbing, grid and view modes, as well as dynamic status indicators. From left to right:

Figure 12.8: Control Widget Bar

- The **Missing Files** button will be visible if the session detects that any segments refer to files that cannot be found automatically. Clicking this button will launch the **Locate Missing Sound Files** window. If there are no missing files, this button will not be visible.
- The **Punch Mode** popup controls the automatic edit mode used when overlaps occur due to either dragging a segment or from overdubs. The available modes are:
  - Split and Add (splits existing segments on overlap, adds new segment)
  - Split and Remove (splits and removes existing segments on overlap, adds new segment)
  - Split and Mute (splits and mutes existing segments on overlap, adds new segment)
  - Add (adds new segment without affecting existing segments)
- The **Xfade** popup lets you set the type of fade or crossfade to be created automatically when segments are recorded or edited. Options include "None" (no crossfade - a butt-splice), "Linear", "Raised Cosine", "Cosine" and "Sqrt Cosine" (Square-Root Cosine).

  Please Note: When "None" is set as the fade type, you will still be able to grab the segment fade duration handle holding by the "f" key as usual, but since there is no fade slope to show, the fade duration will be drawn as a light blue line along the top of the segment, ending at the fade handle cursor.
- The **Xfade Time** popup lets you set the length of automatically created fades when segments are created during overdubs or edited after the fact. Options range from "None" (zero-duration fade - a butt-splice) to 512 milliseconds. Naturally fade durations may be adjusted manually as desired.
- The **Loop Record Mode** popup lets you choose between the different loop recording modes when you record with loop playback enabled:
  - **Loop Mode**: Recording starts when triggered, and punches out at end of loop and is rearmed for next loop pass; this means all new material is recorded within the boundaries of the loop
  - **Loop to Linear Mode**: Playback follows the specified loop, and recording starts at the playhead position when triggered, but continues recording linearly while playback loops. This is great for recording non-looping material against a looped basic track.
- The **Grid** button enables snap to grid
- The **Grid** popup sets the granularity of the grid in terms of musical units from bars down to 1/64 notes. The **.** and **3** buttons modify the selected grid time to turn it into dotted (3/2) or triplet (2/3) duration respectively.
- The **Vertical Scale** slider sets the height of the track overview lanes. The button associated with it toggles between the current view and zoomed all the way out to fit all the tracks into view (or set them to minimum size if there are too many to fit in the current window at the minimum track height).
- The **Horizontal Scale** slider sets the horizontal zoom of the track overviews The button associated with it toggles between the current view and zoomed all the way out to fit the entire session timeline into view.

The **Marker list** table lists all the markers you have added to the session. By default it shows these columns:

#	Name	Start	End	Duration	Reason	Notes
➢ 1	Gun downbeat	00:00:23.8945	00:00:23.8945	00:00:00.0000	bypass intro	
➢ 2	Gun End	00:05:11.5096	00:05:11.5096	00:00:00.0000		
➢ 3	Dimension Extension	00:06:10.5554	00:14:25.1970	00:08:14.6416	fx sidecar processing loop	brilliant
➢ 4	Take 22	00:06:11.0000	00:06:36.6449	00:00:25.6448	loop to linear o'd test	delete as necessary - just a test
➢ 5	DE End	00:14:25.2127	00:14:25.2127	00:00:00.0000		
➢ 6	Spread The Word	00:14:33.0268	00:14:33.0268	00:00:00.0000	fx intro	vibey w/ a bit o' snark
➢ 7	Spread The Word End	00:20:00.0000	00:20:00.0000	00:00:00.0000		

Figure 12.9: Session Marker List

- Marker Index

- Marker Name
- Start Time
- End Time
- Duration
- Reason
- notes

You can <control>-click in the header of the Marker list table to hide or show columns; your configuration is saved in the session.

Clicking on the marker index will cue the playhead to the mark, and select it in the list. If the Session is playing, and is currently in loop mode the cueing command will be deferred until the end of the current loop pass.

<control>-clicking the marker index will popup a context menu with the following commands:

- Zoom to Mark: fit the entire marked region horizontally in the overviews
- Zoom to Selected Marks: fit all the selected marks horizontally in the overviews
- Set Loop to Mark: Updates the loop points to the region defined by the mark
- Move Mark to Playhead: Moves the start time of the mark to the current playhead position
- Delete Mark: Removes the clicked on mark
- Delete Selected Marks: Removes all the selected marks

Clicking on the text fields in the list will allow you to edit them.

You can click and drag the strip in-between the **Marker list** and the **Track Overviews** to adjust how the window is split between the two areas. The split you set is saved and restored with the Session state.

The **Timeline rulers** provide rulers to help you navigate your session. The following rulers are available:

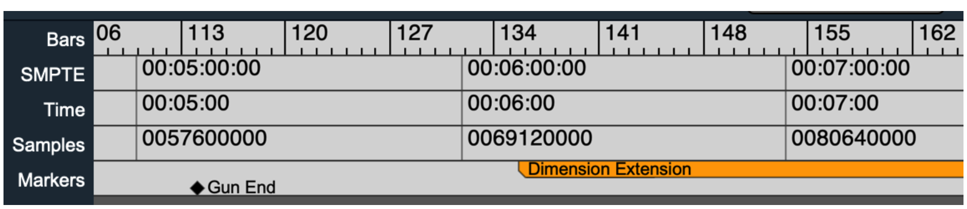


Figure 12.10: Timeline rulers

- Bars and Beats
- SMPTE
- Time
- Samples
- Markers

You can <control>-click in the Timeline rulers to popup a context menu that allows you to choose which rulers are visible. Clicking on one of the rulers will cue the playhead to that position. Double clicking will cue the Playhead and start playback.

You can click and drag markers to change their positions. <command>-click drag the edge of a marker to change its start or end time independently (thus changing the duration).

Below the rulers are three lanes that are always visible - the top one is for the current loop region, the middle one for the current Autopunch region and the bottom one is for the timeline selection region. You can click and drag a region to move it, or click and drag the edges to move the edges independently. Click and drag in the lane (outside of the existing region) to directly set the region. A simple click (with no drag) will cue the playhead to the click location.

The **Track headers** show the name of the track (taken from the associated mixer strip scribble strip), the track color, and if there is room the Solo, Mute, Record Enable, Fader and Pan from the associated mixer strip. When the track-height is small enough, these extra controls will be hidden.

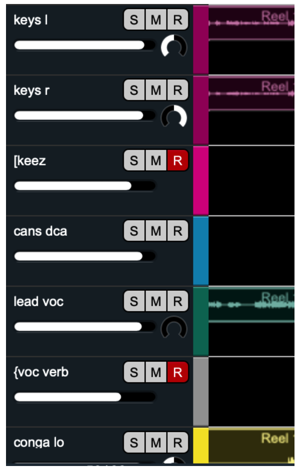


Figure 12.11: Track headers

The **Track headers** are directly connected with the **Track overviews** and scale and scroll along with the associated overview track. The **Track headers** reflect the selection state of the associated mixer strip and the tracks have the same order as the mixer strips. You can select tracks by clicking the background of the track header, and you can reorder the tracks by dragging the track headers up and down.

The **Track overviews** show the audio segments in the track; the segments are drawn based on the track color, and also show the in and out fades, as well as the mute status of the segment. The waveform overviews reflect the applied fades and the segment gain.

Figure 12.12: Session Track Overviews (track lanes exposed)

You can control the scaling of the waveform overviews with the Session menu: Waveform Scaling) menu command; this feature allows you to enhance low-level features of the waveform or emphasize transients in the waveform while still keeping the entirety of the dynamic range of the waveform in view.

You can choose whether or not to show the segment names in the track overviews.

Track Lanes: Unlike the usual implementation of **Track Lanes**, a unique feature of the Session playback engine is that you can have an unlimited number of segments overlapping at any given moment within the track. A new lane is created as-necessary whenever segments overlap, and since each segment has independent segment gain control, you can create full mini-submixes and/or multichannel comps within the track itself, without any extra routing and without ever touching the mix desk... All fades, segment gain and summing within the track is handled in the playback engine, and yes, segment gains and fades can be adjusted on the fly so you can tweak in context while listening. (See also: Session: Export Tracks)

You can toggle the track lanes view using the “L” key, or from the Session menu to expose each segment with its fades in relation to the other segments in their track lanes.

In addition to being able to scroll around the track overviews with the trackpad/scrollwheel, the track overviews support the following gestures:

- <option> scroll to zoom in and out vertically
- <option><command> scroll to zoom in and out horizontally around the playhead
- <option><command><control> scroll to zoom in and out horizontally around the mouse pointer
- trackpad pinch-to-zoom to zoom in and out horizontally around the playhead
- <shift>-click to re-cue the Playhead
- <shift>-double-click to re-cue the Playhead and start playback

And the following key/menu commands

- <shift>Z - zoom to fit the current selection
- <option><command>Z - zoom horizontally to fit current loop
- <command>Down Arrow - Increase Track Height
- <command>Up Arrow - Decrease Track Height
- <option>X - Fit Tracks Vertically
- <command>Right Arrow - Zoom In Horizontally
- <command>Left Arrow - Zoom Out Horizontally
- <option>X - Fit Session Horizontally

As you move the mouse over various hotspots in the Overviews, the cursor will dynamically change to indicate what will be affected if you click and drag.

From the bottom half of a track, click to select that segment, <command>-click or click-drag to select multiple segments.

From the top half of a track, click-drag for time range selections.

You can click and drag and drop segments within tracks and between tracks.

- Hold the option key to copy rather than move a segment when dragging it.
- Hold the shift key (after starting the drag) to constrain the segment to its original time on the timeline.
- Hold the command key down while dragging to defeat the current grid setting.

Click and drag the edges of segments to trim the start or end of the segment.

Click and drag fade line to adjust the length of the fade.

> Note:
> If you hold the "f" key down, clicking and dragging at the edge of the segment will prefer adjusting the fade length rather than the segment trim.
>
> Hold the "g" key and click on a segment to adjust the segment gain.

<control> click a segment to popup a context menu with segment commands:

- Reveal in Finder…
- Reveal Take folder in Finder…
- Rename Segment…
- Trim Top to Click
- Trim Tail to Click
- Trim Top to Playhead
- Trim Tail to Playhead
- Split Segment at Click
- Split Segment at Playhead
- Split Segment at Loop Points
- Trim Segment to Loop Points
- Align to grid
- Delete Segment
- Mute Segment
- Adjust Segment Gain…
- Export Segment to File…
- Bounce Track to File…

Zoom in to highlight and <control> clicking a fade will add the **Fade Types** to the segment commands list.

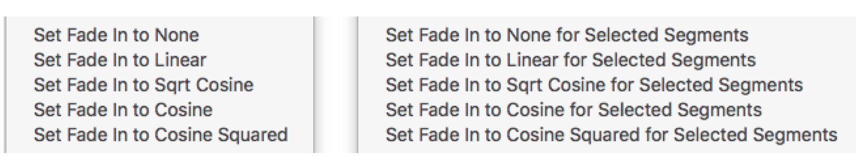


Figure 12.13: Session: Segment Fade Types contextual menu

The Fade Type items are contextual, and will show "for Selected Segments" when multiple segments are selected (above right).

The Session menu has replaced the old **Recording** menu, enhanced with additional commands to support the new **Session** implementation:

- Session
  - Files
    - New Session…
    - Find Missing Session Files…
    - Set Record Folder…
    - Add Mirror Record Folder…
    - Import Take Folder…
    - Import Folder of Take Folders…
  - Edit
    - Split [All/Selected] Segments at Playhead
    - Split [All/Selected] Segments at Loop Points
    - Split Segments at Selection Boundary
    - Trim [All/Selected] Segments to Loop Points
    - Trim Segments to Selection
    - Trim Top of [All/Selected] Segments to Playhead
    - Trim Tail of [All/Selected] Segments to Playhead
    - Trim Top of Segments to Selection
    - Trim Tail of Segments to Selection
    - Align [All/Selected] Segments to Grid
    - Mute [All/Selected] Segments
    - Unmute [All/Selected] Segments
  - Export
    - Export Tracks…
    - Export Tracks Between Loop Points…
    - Export Selection…
  - Transport
    - Start Recording
    - Stop Transport
    - Start Playback
    - Return To Next Cue Point
    - Enabled Cue Points:
      - Enable Return To Last Play Start
      - Enable Return To Selection Start
      - Enable Return To Loop Start
      - Enable Return To In Point
      - Enable Return To Session Start
  - Playback Scroll Mode
    - Continuous
    - Paged
    - None
  - Playback Mode
    - From Playhead
    - From Playhead; Return on Stop
    - From Loop Start to End
    - From Session Start to End
  - Waveform Scaling
    - Enhance Peaks
    - Normal
    - Enhance Low Level
    - Extra Enhance Low Level

- Looping
  - Loop Playback
  - Set Loop when Cueing to Mark
  - Set Loop Start to Playhead
  - Set Loop End to Playhead
  - Set Loop to Selection
- Autopunch
  - Set In Point to Playhead
  - Set Out Point to Playhead
  - Set Autopunch to Selection
- Markers
  - Drop Mark (at play or record head)
  - Create Mark from loop
  - Create Mark from Selection
  - Jump to Previous Mark
  - Jump to Next Mark
  - Save Markers…
  - Load Markers…
- Zoom
  - Zoom to Loop
  - Zoom to Fit Selection
  - Increase Record Panel Track Height
  - Decrease Record Panel Track Height
  - Fit Tracks Vertically
  - Zoom Tracks In Horizontally
  - Zoom Tracks Out Horizontally
  - Fit Tracks Horizontally
- View
  - Show Segment Names
  - Expand Track Lanes for Overlaps
  - Show Only Record Enabled Tracks
- Tempo
  - Calculate Tempo from Loop
  - Calculate Tempo from In/Out points
  - Calculate Tempo from Selection
- Enable Metronome
- Enable Grid
- Recording Preferences…

Session: Transport Header bar

The Session Transport Header bar is comprised of seven primary sections: Time Display block, Clock Display, Tempo Controls, Transport Controls, the "Take" Number Display, Record Trigger settings, and the Record Folders section.

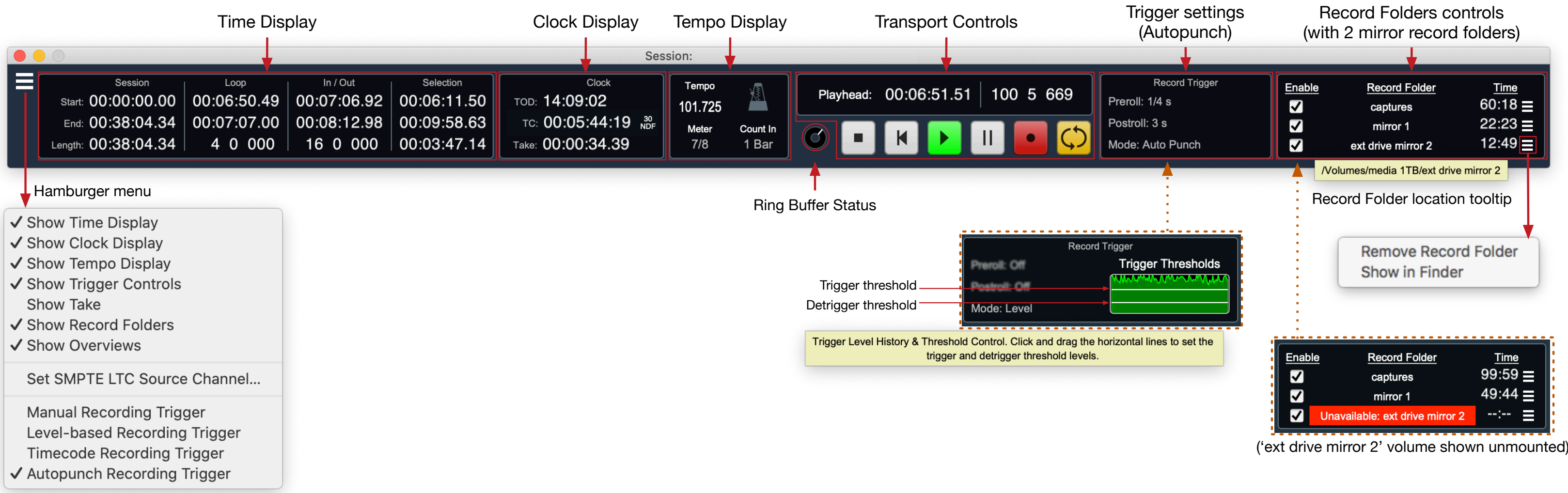


Figure 12.14: Session Transport panel (minus "Take" display)

Session Hamburger Menu

At the upper left corner of the Transport panel is a hamburger menu with controls to show or hide Transport display blocks, set the timecode source input, and set your record trigger mode. Refer to the graphic as we go through the hamburger menu functions, and we will break out all the operational details in the sections to follow.

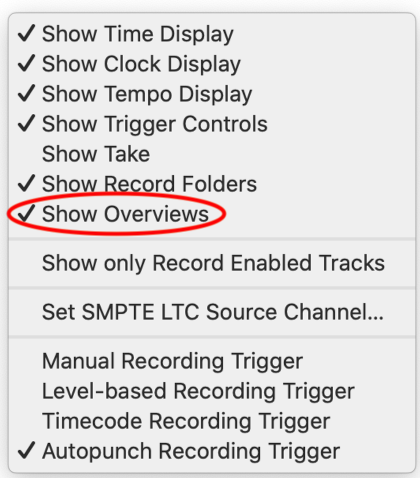


Figure 12.15: Transport Header bar hamburger menu

Hiding a section of the Transport Header lets you make the Transport window narrower without cutting off an info/control block, so it's a handy way to make desktop space available when things get tight.

The Session window can be configured all the way down to show only the show Transport Controls block, after which, well, you just minimize or close the window itself. The Transport Controls block can not be independently hidden.

Session Transport Burger menu options

- **Show Time Display** shows/hides the "Time" display box (with Session timeline, Autopunch In/Out and current Loop timings).
- **Show Clock Display** shows/hides the "Clock" display box (with Time of Day, LTC timecode and current Take duration).
- **Show Tempo** shows/hides the "Tempo" control block (tempo, meter, count-in mode and metronome).
- **Show Trigger Controls** shows/hides the Preroll/Postroll controls and Record Trigger display and mode settings.
- **Show Take** shows/hides the "Take" display.
- **Show Record Folders** shows or removes the Record Folder section from the Transport panel.
- **Show Overviews** hides the entire channel Overviews section, including the Session widget bar.
- **Show Only Record Enabled Tracks**, when selected, hides any tracks which are not currently record-enabled, such that only *Record-Enabled* tracks are visible.
- **Set SMPTE LTC Source Channel...** opens a routing window where you can select input source of the incoming SMPTE stream. Note that only ULN-8 and LIO-8 boxes have dedicated hardware SMPTE I/O ports.
- **Manual Recording Trigger** sets the Record Trigger mode to "Manual" operation. Enable "Record", hit "Play" to start, hit "Stop" to stop.
- **Level-based Recording Trigger** sets the Record Trigger mode to "Level-based": Record start triggers at a set input trigger level threshold, and de-triggers when signal level drops below a set de-triggers level threshold.
- **Timecode Recording Trigger** enables recording to automatically start on SMPTE LTC lock, and de-triggers on unlock.
- **Autopunch Recording Trigger** engages Autopunch recording mode. In this mode, the Transport will start play in Record Ready mode (respecting the 'Count-in' setting), and automatically punch in and out of Record based on the "In/Out" timeline selection range.

Time display block

The Time block shows the current Session timeline region selection status.

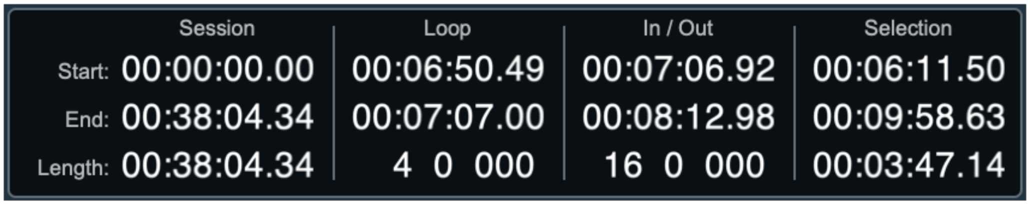


Figure 12.16: Transport Header bar: Session Time display

All fields in this display are read-only. Each field may be independently set to display in regular HH:MM:SS time, time with fractional seconds (HH:MM:SS.ssss), Bars & Beats or samples.

Control-click any Time display field to open a contextual menu and select a time format for that field.

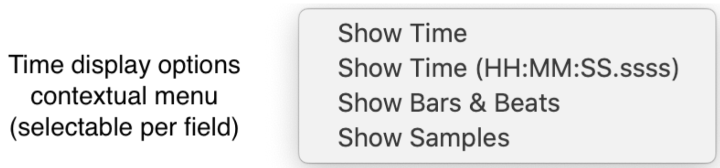


Figure 12.17: Transport Header bar: Time display format options

The Time block **Session** section displays the current 'Start' and 'End' times, and the 'Length' of the current Session timeline.

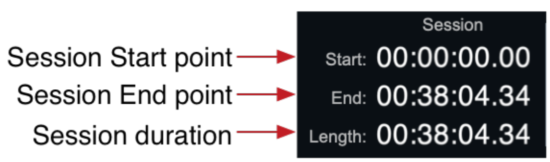


Figure 12.18: Transport Header bar: Session timeline display

Session 'Start' is the timeline start point (usually 00:00:00.00). Support for timeline Start point offset has not as yet been implemented.

Session 'End' shows the end mark location of the last audio segment in the timeline, plus 90 seconds. Ninety seconds of empty timeline space is always maintained to provide room at the end of the timeline UI to start a new recording set, drop in additional audio files, or as temporary holding space for audio segments (or groups of segments) to be moved out of the way and re-inserted later.

Session 'Length' is the total time between Session 'Start' and Session 'End'.

The Time block **Loop** section displays the Start, End and Length of the currently selected timeline Loop.

You create a loop by dragging a range in the "Loop" timeline. You can create a Mark with duration from the current loop; go to the Session "Markers" menu and choose "Create Mark from Loop". Markers with duration can be used to set the the loop by right clicking on the associated arrow in the Marker List to the right of the Session Overview.

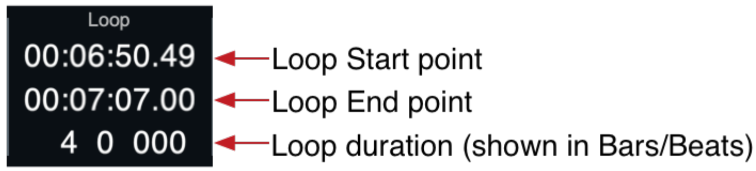


Figure 12.19: Transport Header bar: Loop time display

When Loop playback mode is enabled, the current Loop range will be highlighted *yellow* on the Loop timeline (as shown below left). The loop range will be shown gray otherwise (shown below right). (Please also refer to this graphic as an illustration of In/Out and Selection modes.)

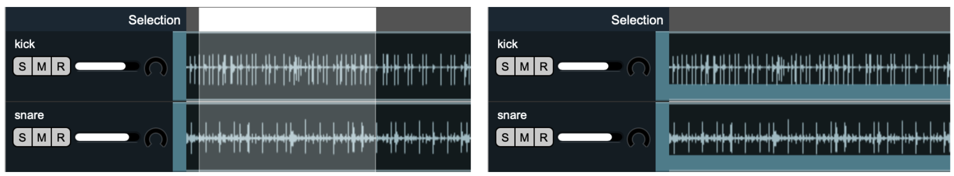


Figure 12.20: Transport Header bar: Timeline Loop, In/Out and Selection ruler and selection modes

Loop ‘Start’ reflects the start point of the currently selected loop.

Loop ‘End’ shows the end point of the selected loop.

Note that the Loop time display will also reflect loops defined by the selected Marker when in Loop playback mode.

The Time block **In/Out** section shows the Start, End and duration of the current Autopunch range.

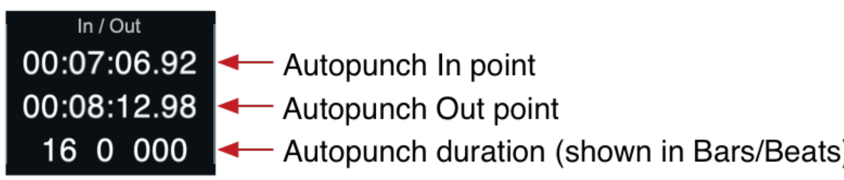


Figure 12.21: Transport Header bar: Autopunch In/Out time display

An Autopunch range is created by dragging and adjusting a range in the **In/Out** timeline, as shown in the timeline rulers graphic in the Loop section above.

When Autopunch record trigger mode is enabled, the Autopunch range will be highlighted *red* on the **In/Out** timeline (as shown above left). The punch range will be shown gray in all other record trigger modes (shown above right).

The Time block **Selection** fields show the Start, End and duration of the current timeline or segments selection.

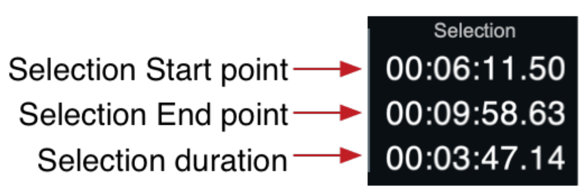


Figure 12.22: Transport Header bar: Selection time display

Timeline selections (shown above left) are displayed as a white range in the Selection timeline ruler. Timeline selections may be created by selecting a range in the timeline ruler which will include all audio within that range across all tracks in the session, or by clicking in the top half of any track and dragging a selection field to include only the audio within the field.

Segment selections are created by clicking in the bottom half of a segment, cmd-clicking the bottom half of multiple segments, or clicking in the bottom half or any track and draggin a selection field.

Note that only one selection range is possible at any time.

Clock display block

The Transport Control bar Clock is a read-only display, presenting the current time of day, incoming SMPTE time code and frame rate (if any) and the duration of the currently-recording take.

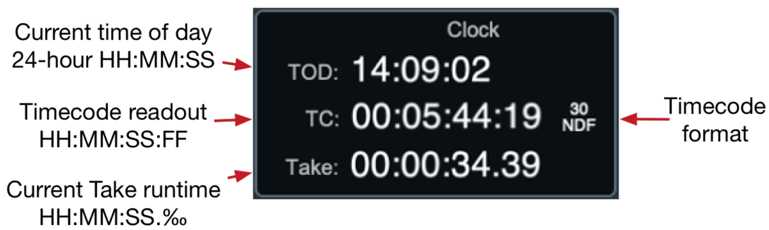


Figure 12.23: Transport Header bar: Clock display

Tempo Control block

The Tempo Control block houses controls for setting Tempo, time signature (meter), recording take Count-In mode, and metronome enable/disable.

Figure 12.24: Transport Header bar: Tempo controls

- **Tempo** can be set manually by clicking on the Tempo field and typing in the desired tempo. The Bars/Beats timeline and metronome will immediately re-adjust.

  Tempo can be also be calculated for existing audio by setting a loop, In/Out or timeline selection to a musical phrase, and applying the appropriate "Calculate Tempo from..." command from the menu bar "Session: Tempo" menu.

- Click the **Meter** selector to open a menu of commonly used time signatures. Other time signatures can be entered manually in "Calculate Tempo from...", selecting "Bars" as opposed to "Beats" in the Set Tempo interface, enter the number of measures in the selected loop, and manually enter your desired time signature.

- The **Metronome** can be toggled on/off by clicking its icon or with the default keystroke <control>-M. When engaged, the metronome icon will highlight white, and a mixer strip and Session track are created, appropriately named "Click".

  The Metronome sound itself is generated within the hardware, and follows the usual metronome convention of emphasizing the "1" as it marks the beats during transport playback.

  The "Click" strip/track is similar to a regular Input strip, allowing the click to be processed and routed to direct or bus outs. Because the click sound is generated internally, the "Click" strip has no input route, and so does not have the "Source" or "To Host" routing controls at the top of the strip and can not be used for playback.

  Note that the "Click" track will remain in your mix desk and session when the metronome is disabled. It can be deleted manually (and will be re-created the next time you enable the metronome), but any routing or processing applied in the "Click" strip would be lost upon deletion.

- **Count In** provides record mode controls for manual, loop and autopunch recording. In all cases, Preroll and Postroll settings and Record Trigger modes are respected.

  - **Take** mode starts recording from the playhead position the moment you hit "Record", but does *not* engage playback. Hitting "Record" while recording in "Take" mode will instantly advance the playhead to the new position in the timeline and start recording a new take.

    "Take" mode operates like the old Record Panel, and is generally the preferred mode for live performance capture or location recording where transport playback is not a factor.

    Hitting "Play" while recording in "Take" mode will start transport playback at the current playhead position or Loop Start mark (when Loop play mode is engaged).
    Hit "Play" once again to stop Transport playback and exit recording.

    This technique, combined with "Loop to Linear" record mode allows you to play loops of any section of audio within the timeline while recording a new linear track at a completely different location, improvising new lines against the loops.

    Use the keystroke **⇧ Return/Enter** to drops timeline markers on the fly while recording in "Take" mode.

    Be aware, the Metronome currently only engages during transport playback, and will not be heard in "Take" mode until play is engaged.

  - **Count In: None** will start both playback and recording at the current playhead position, Loop Start mark (when Loop play mode is engaged), or the Autopunch In point (when Autopunch trigger mode is engaged).

  - **Count In: 1, 2, 3 or 4 Bars** starts playback at the designated number of bars before the current playhead position, then punches into record at the original playhead point.

  Note that the above **Count In** settings will use the Loop Start or Autopunch In points as the record start point when one of those modes is active. When both Autopunch and Loop modes are engaged, recording will start at the Autopunch In point.

Transport Control block

The Transport block displays information crucial to the operation of Session recording and playback and houses the Transport operational controls.

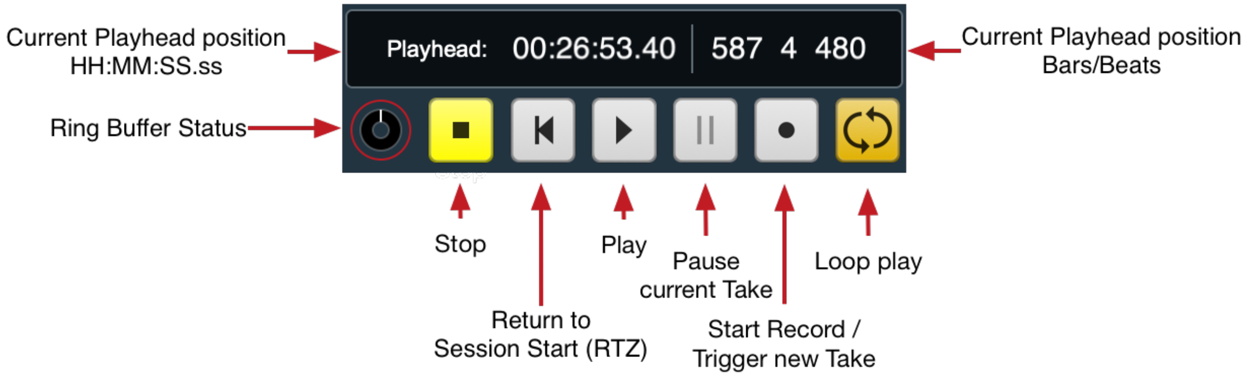


Figure 12.25: Transport Header bar: Transport stopped with Loop mode engaged

Transport Control is the only always-visible section the Session window.

- The **Current Playhead position** is shown (in both HH:MM:SS.ss and Bars/Beats). This is a read-only display.
- The **Ring Buffer Status** buffer status ring:

  Since the Ring Buffer is a crucial element of the recording engine, the Ring Buffer Status display is shown in the Transport section where it will always be visible.

  How you use the Ring Buffer is detailed by the Preroll entry in the "Record Trigger block" section which follows. In the meantime, here's what it is and what the display represents:

  Inherent to the 3d Session record engine is a 60-second FIFO ring buffer memory cache capable of handling 128 channels of 32-bit audio at 192kHz. This ring buffer rolls on unnoticed in the background whenever "Record Enable" on any Session track is engaged ("R" button = Red), continuously replacing old audio with new (hence "FIFO", meaning "First In, First Out").

  This 60-seconds of buffered audio is used as a data confidence measure when writing to slower storage media, and to provide from 1/4 second to 60 seconds of automatic **Preroll** to all Session audio files.

  So, if the Ring Buffer Status ring display is rotating, that means that some track somewhere, whether visible or not, is record-enabled, and the ring buffer is running in the background.

  The white "memory status" section of the ring buffer display grows larger as your Preroll setting increases. You will see it start "filling up" with audio upon record-enabling a track; so if you set Preroll to 10 seconds, it requires 10 seconds of pre-loading from the moment you record-enable your tracks to fill that preroll memory allocation (after all, it can't write audio to a file that it hasn't had time to load yet).

  When the "memory status" indicator stops growing your preroll time allocation has been reached.

  Again, the ring buffer itself is always FIFO-ing the full 60 seconds of audio - the white section only describes your current Preroll allocation.

  The spin rate of the display increases at higher sample rates, indicating data is moving faster through the ring buffer FIFO engine. As long as channels are record-enabled, even with 'Preroll = Off' the ring buffer continues to operate and the display will continue to rotate, as the ring buffer also acts as a safety measure against dropped samples when recording to slower storage media.

When no tracks are record-enabled, "memory status" bar dims, the display stops rotating and the ring buffer memory is cleared.

- The actual **Transport Controls** in general terms function as one would expect.
  - **Stop** is a full transport stop, regardless of Play or Record mode, and regardless of Trigger Mode. When either the Playhead or Recordhead (or both) are moving or the Record button is flashing red (indicating "Record Standby"), hitting "Stop" will stop play and record transport and disengage recordings-in-progress or Record Standby states.

    The Stop button will be colored yellow at all times the Transport is fully stopped (both the Playhead and Recordhead are idle).

Figure 12.26: Transport Full Stop

When Transport is running, the "Stop" button will be colored gray.

The Session Playback Mode can be set to automatically return the playhead to its prior 'play start' position, or leave the playhead where it was stopped, to continue play from there.

  - **Return To Zero**: From Transport Stop, relocates the Playhead to the Session Start location on the timeline (currently 00:00:00.00).

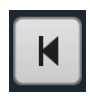


Figure 12.27: Transport "Return To Zero" (RTZ) button

Hitting RTZ once will return the playhead to the point on the timeline at which you last engaged "Play".

Hitting RTZ a second time will relocate the Playhead to 00:00:00.00 on the timeline.

RTZ will continue to toggle the Playhead between the last "Play start" location and 00:00:00.00 on the timeline.

  - The **Play** button engages "Play mode", in which the Play button turns green, the Stop button turns gray, the Playhead starts moving to the right and audio plays back.

Figure 12.28: Transport "Play" mode

Note that in Count-In: "Take" mode, the 'Play' button will start the Playhead moving independently of the Recordhead.

- **Record Pause:** "Pause" is a record-mode-only control which serves a specific purpose for live or field location recording, where the goal is to record contiguous soundfiles across extended periods of time, spanning hours or days with the possibility of extended breaks between recording events. It is generally not used in tracking or mixing.

  'Pause' is used to temporarily suspend recording while remaining in Record mode and within the current Take. When you click 'Pause' the Pause button turns blue, waveform drawing in Overview stops and audio stops being written to your drives. The Record button will remain red, confirming that you are still in record mode.

Figure 12.29: Record Pause (Count-In "Take" mode)

  During Pause, the Preroll ring buffer will continue operating in the background as usual.

  When you click 'Pause' again or 'Record' to resume recording, the preroll-buffer audio will be printed to the existing Take fileset and recording will continue as usual.

  Note that hitting 'Stop' while in Record Pause will disarm Record mode and immediately end the Take.

- **Record:** In Count-In: "Take" mode, hitting "Record" will immediately start recording your new Take. The 'Stop' button will turn gray, the "Record" button red, and the Preroll buffer will write its data to the beginning of the new audio files, capturing the audio which occurred just before you hit 'Record' into the current Take. count-In "Take" mode does not engage Play, so the transport "Play" button remains gray (inactive), there is no playback and the Playhead remains stationary.

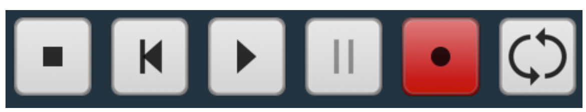


Figure 12.30: Recording in Count-In "Take" mode

  In the other Count-In modes, hitting the "Record" button will engage both Play and Record for mixing or overdubbing, following your Count-In selection (None, 1, 2, 3 or 4 bars).

Figure 12.31: Recording in Count-In overdubbing modes

  All Record Trigger Modes are available for all Count-In modes, adding flexibility and semi-automatic operation options to suit a wide scope of self-recording, studio and location production workflows.

> Please be aware that some combinations of Trigger and Count-In modes are designed for specific workflows and were not designed to work together. For example, “Take” record mode is designed for take-based recording-only functions (i.e., no Session Playback), such as live performance and location recording, whereas “Autopunch” trigger mode is built more for overdubbing, double-tracking and loop recording with or without Session playback.
>
> That said, there are no safeguards in place to restrict you from experimenting. Enjoy!

Record Trigger Control block

The Record Trigger block manages recording Preroll and Postroll settings, and the various recording trigger modes and controls.

Preroll, Postroll (and how that Ring Buffer thing fits in)

So have you ever been recording sound for an event and just missed hitting “Record” when the band or the announcer snuck in their intro earlier than scheduled? ...or maybe the crowd started singing “Louie, Louie” really loud but somehow in perfect pitch? Ladies and gentlemen, please say hello to Preroll in MH 3d Session.

In a nutshell, the Session **Preroll** lets you capture the audio immediately *before recording is triggered* in any record mode... up to a full minute (60 seconds) before you hit record.

When any new recording (i.e., a “Take”) is triggered, the contents of the Ring Buffer are immediately written to the record target drives at the head of the new Take files, thereby seamlessly capturing the audio just before you hit Record.

Selecting a Preroll duration merely tells the record engine *how much* ring buffered audio to write to the beginning of each new Take file.

During active recording, the ring buffer continues working in the background, so when you trigger new Takes manually by hitting the Record button without stopping in-progress recording, the Preroll option can kick in to automatically overlap the end of the previous Take at the beginning of the new Take.

Since the ring buffer is always operating at the full 60-second capacity, you can adjust the Preroll setting on the fly, lengthening or shortening the preroll duration to best suit the program being captured. The new preroll setting will be applied to the beginning of the next Take.

Remember, if you set Preroll to 10 seconds, it requires 10 seconds of pre-loading from the moment you record-enable your tracks to fill that preroll memory allocation. If you start recording a take befoer that buffer is full, it will work fine, but the amount of preroll will be a bit short of the full 10 seconds, is all.

Please be aware that in the Session Track overview of your recorded files, the Take fade-in is placed at the location that “Record” was triggered. Preroll audio will be hidden in the file *before* (to the left of) the Take fade-in.

Drag the Take fade-in to the left to expose the preroll audio.

> **Important Note:** The **Ring Buffer** will only cache audio from record-enabled tracks (because it’s silly to use a ton of memory to store data that will not be used). This means that *changing the record-enable state of any Track at any time will clear and re-start the ring buffer.*
>
> In practice this is rarely an issue, because in most cases it only takes a few seconds to re-fill the Preroll allocation. However, in cases where very long Preroll settings will be applied, this is something to keep in mind.

The Preroll setting is also an option for both the “Auto-Break Overlap” and “Manual Break Overlap” preferences, set in ‘MIOConsole3d Preferences: Recording’, and described in context below.

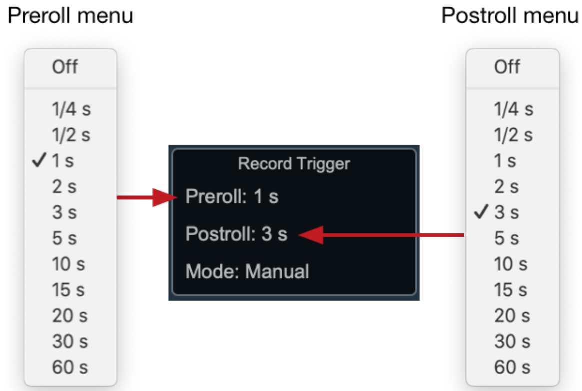


Figure 12.32: Session Transport: Preroll, Postroll and Ring Buffer

- Use the **Preroll** selection button as shown above to open the preroll menu. Record preroll can be set as short as 1/4 second up to 60 seconds. The preroll ring buffer is inherent to the record engine and operates whenever a channel is record-enabled. Setting Preroll to "Off" will naturally result in no preroll at the head of the recorded audio files.

- **Postroll** duration sets the amount of time a recording will continue after a "de-trigger" command is received by the record engine. Like Preroll, Postroll can be set to durations from 1/4 second to 60 seconds, or turned off entirely.

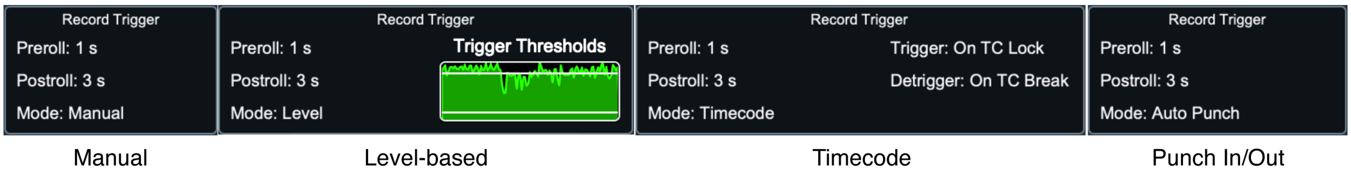
  "De-trigger" commands occur in Level-based record mode when the incoming audio level drops below de-trigger level threshold. Timecode record mode de-triggers when incoming timecode stops or otherwise becomes unlocked.

  Postroll is a just a timer, whose function is to delay the end of the Take beyond the de-trigger command. Postroll can ensure that you capture slow fades and long ringouts, or to continue recording a Take over a quiet passage in the music or between songs.

> Use the Transport 'Stop' button to disarm Record Standby, and to override a Postroll and immediately end the current Take.

Record Trigger modes

You can select your record trigger mode from the Record Trigger control block in the hamburger menu or by selecting "Manual", "Level", "Timecode" or "Auto Punch". The Record Trigger control UI morphs to reflect each recording trigger mode, as shown below.



Session Record Trigger modes

Figure 12.33: Session Transport Controls: Record Trigger modes

Manual Trigger Mode

This is the most basic recording trigger mode, In Manual mode, you hit the 'Record' button to start recording, and the 'Stop' button to stop recording.

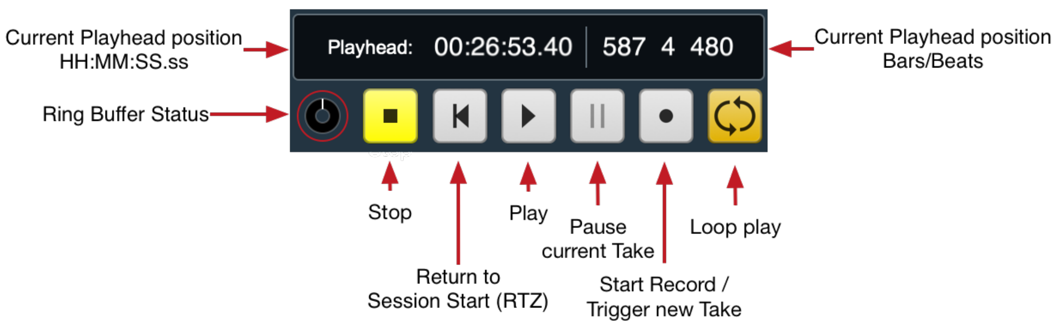


Figure 12.34: Session Transport: Manual trigger mode

The 'Stop' button will remain yellow (indicating 'Standby') until you hit the 'Record' button.

Manual Record Start: Hitting 'Record' immediately starts recording your new Take. The 'Stop' button will turn white, the "Record" button red, and the Preroll buffer will write its data to the beginning of the new audio files, capturing the audio which occurred just before you hit 'Record' into the current Take.

Record Pause: While recording, you can hit 'Pause' to temporarily suspend recording while remaining within the current Take. When you click 'Pause' the Pause button turns blue, waveform drawing in Overview stops and audio stops being written to your drives. The Record button will remain red, confirming that you are still in record mode.

During Pause, the Preroll ring buffer will continue operating in the background as usual.

When you click 'Pause' again or 'Record' to resume recording, the preroll-buffer audio will be printed to the existing Take fileset and recording will continue as usual.

Note that hitting 'Stop' while in Record Pause will disarm Record mode and immediately end the Take.

Manual Break: If you want to break up your recording on the fly and start capturing a new Take fileset in a new Take folder, hit 'Record' again while record is running. This is called a "Manual Break". The new audio files can include an audio overlap as defined by the 'MIOConsole3d Preferences: Recording' 'Manual Break Overlap'" setting.

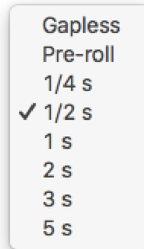


Figure 12.35: MIOConsole Preferences: Recording pane: 'Manual Break Overlap' selector

The **Manual Break Overlap** preference lets you set the amount of overlap between the two takes in case you want to edit them back together later. "Gapless" places zero-duration fades at Take file transitions, meaning the last sample of the previous Take is immediately followed by the first sample of the new Take.

Generally, Manual Break Overlaps will be pretty short, the idea being to give you a bit of extra room to drop in your own fade at a spot that works best for the recorded program and for the edit model of your particular audio workstation. In case you want a longer overlap, you can always use the Preroll setting, as shown above.

In any case, overlapping audio will be in the file created *after* the break; this means the currently recording Take will end when you break the Take, and the newly started Take will include the specified amount of overlap from *before* the break.

Level-based Trigger Mode

With *Level-based* record triggering, Session can be set up to operate unattended. Recording will start only when the incoming signal level reaches your set trigger threshold, and automatically stop when the level drops below your "de-trigger" threshold. Preroll and Postroll become especially powerful features in Level-based mode, allowing you to set the trigger thresholds high enough to avoid false starts even in live concert recordings, and still capture extended lead-ins and fade-outs at the head and tail of each set.

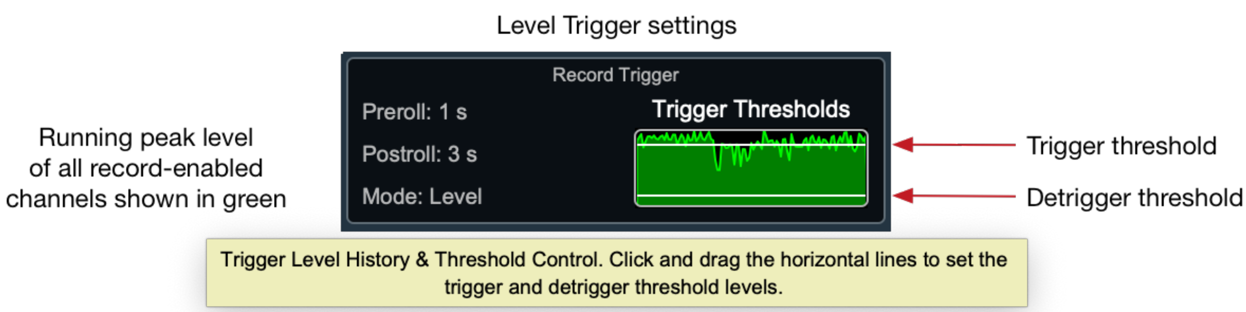


Figure 12.36: Session Transport: Level-Based Trigger Mode

When Level-based trigger mode is engaged, the 'Trigger Thresholds' display appears as shown above. When you have channels record-enabled and passing audio, the display will show a running peak level in green of all audio coming into the Session panel.

At this point, the transport "Stop" button will be yellow and the transport "Record" button will be white, indicating that the system is ready to record, but is not yet record-armed.

Before you arm the record engine, use the peak audio level display as a reference to set your Trigger and De-trigger levels.

The two horizontal lines running across the display can be dragged up or down to set your trigger and de-trigger thresholds (or un-trigger, if you prefer).

Trigger Threshold: The upper line is your Trigger threshold. When you click on it, the line will highlight as you move it and the Trigger Threshold reference level in dBFS will appear above the display. When the incoming signal level hits this threshold, recording will start, with preroll audio written at the start of the new Take.

The maximum Trigger Threshold setting is -3.0dBFS, and the minimum is -50dBFS.

Trigger and Preroll: Judicious setup of Preroll with respect to Trigger level allows you to set the threshold high enough to avoid false starts due to venue noises, but still capture quieter pre-downbeat events like intro music swells at the beginning of a live concert. The applications are endless, and the features really beg to be played with.

De-trigger Threshold: The lower line is the 'De-trigger' threshold. De-trigger will always be below the trigger threshold. Clicking on the de-trigger threshold will highlight the threshold line and display the De-trigger reference level.

The maximum De-trigger Threshold setting is -6.0dBFS, and the minimum is -55dBFS.

De-trigger and Postroll: Postroll plays a huge role in getting the most out of level-based recording. When the audio level drops to the De-trigger level, it cues the postroll timer, delaying the end of the Take. If, during postroll, the audio level rises back to the Trigger threshold, postroll is cancelled and the recording continues uninterrupted.

You can see how useful this feature becomes for any number of recording scenarios from live concert capture to spoken word to production sound to semi-automated preservation transfers.

> Mini-editorial
> Probably *the* best thing about level-based record triggering is that, as a working sound engineer with a ton of stuff to keep track of, the feature lets you relax just that bit more than usual, confident that the system is running and you don't have to be literally tied to the tech rig with your finger on the trigger for every second of a session. That five to twenty seconds of pre/postroll can feel like an eternity in-session, and knowing that the Session has your back, and can buy you the time to get back to the console without missing some unexpected event lets you concentrate more on enjoying the gig and less on the stress.
>
> The *other* best thing about level trigger mode is, in your home studio you can leave Record engaged all the time... Thus in the middle of the night (or whenever inspiration strikes) you can stagger into the studio and just start playing - Session will record your ideas automagically (including a lovely preroll, if so desired).

Timecode Trigger Mode

If you have access to incoming SMPTE timecode, Session can be set to automatically trigger Take recording immediately upon timecode lock, and stop recording on timecode unlock.

Routing SMPTE LTC in MIOConsole3d

Like any other incoming audio, timecode must first be routed in MIOConsole3d so the Session record transport can read it. Here is an example of Mixer Input Strip for SMPTE timecode:

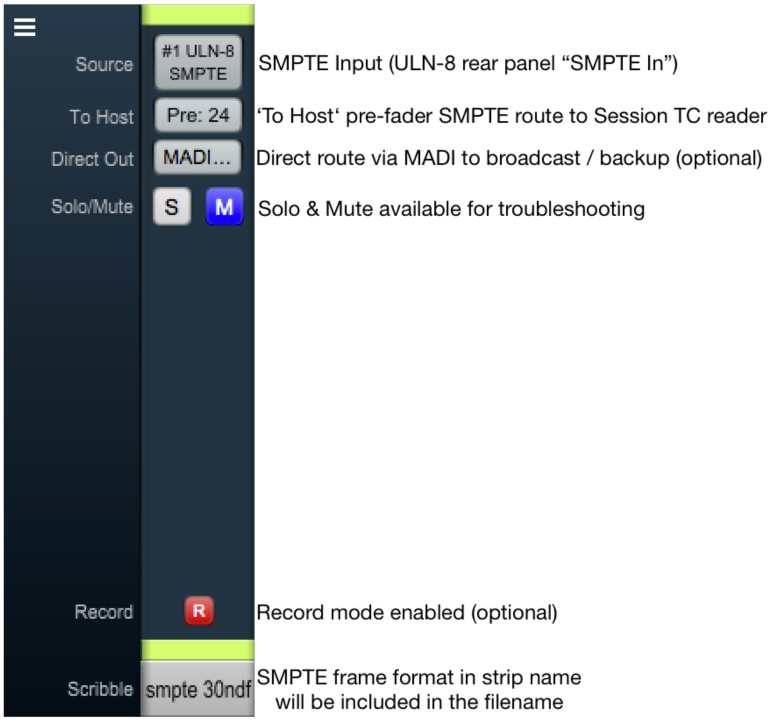


Figure 12.37: 3d Mixer: a SMPTE-specific Input Strip

Since this input strip is specifically for routing timecode to the Session, the only really crucial settings are "Source", "To Host" and the mixer strip name. You may find the other strip elements shown to be useful as well.

- Input **Source (required):** Set the Input strip "Source" to your incoming SMPTE port. ULN-8 and LIO-8 3d boxes have dedicated 1/4" TRS SMPTE timecode I/O ports on the back panel (shown routed above).

  Your SMPTE source could just as easily come from any input including from a file being played on a channel from your Host DAW. All are fair game depending on your particular workflow.

- **To Host (required):** Immediately below the "Source" selector, select a "To Host" route to send the SMPTE channel to the Host computer so the Session time code reader can see it, and the Session engine can record it to your drives.

  Generally, this would be "Pre Insert" selection, but should you need to add processing to clean up a poor SMPTE signal, "Post Insert" is always available.

- **Direct Out (optional):** The Input strip Direct Outs can be used to relay your SMPTE feed to a backup recorder (perhaps another computer via SCP USB), broadcast truck (say, copper or optical MADI), etc.

- **Solo and Mute (optional):** Solo and Mute can be handy for checking your SMPTE feed and verifying levels during setup.

- **Record Enable (optional):** Establishing the SMPTE "To Host" route and assigning it as the "SMPTE LTC source channel" in the Session (detailed below) are the only steps required for Timecode Trigger Mode operation. However, if you wish to record your incoming SMPTE timecode stream as a reference along with your audio program, click "Record Enable" to arm recording SMPTE ("Record Enable" = red).

  While technically optional, recording a SMPTE reference track alongside audio program is standard practice and always welcome.

- **Scribble Strip (required):** The mixer strip name is used in both the routing UIs and the recorded audio file name, so enter 'SMPTE' and the source frame rate here.

SMPTE Timecode Trigger assignment

Once you have your SMPTE input strip set up, go to the Session and click the Time Display "Timecode" readout to reveal the "SMPTE Source" selector.

You can also open this selector from the Session Hamburger Menu "Set SMPTE LTC Source Channel..." item.

Figure 12.38: Session: "SMPTE LTC source channel" selector

Clicking on the selection name (or "None", if unassigned) will open a list of all Mixer Input strips with "To Host" routes to the Session (below). The "SMPTE LTC Source" selection lists the strip names with the original source port for each strip shown in parentheses. Select your SMPTE Mixer strip from the list to assign it.

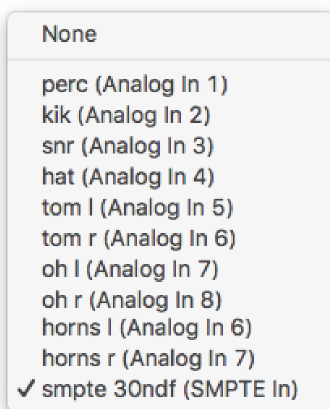


Figure 12.39: Session: "Set SMPTE LTC source channel..." example selection list

Once assigned, the currently selected SMPTE trigger source strip name will be displayed, with the timecode source port shown in parentheses.

Note that Aux, Group and Main bus "To Host" returns will not be included in the SMPTE LTC source assignment list.

Timecode Trigger Mode Settings

The **Timecode Reader** will become active in the Transport Header Clock block when SMPTE LTC is present and readable by the SMPTE timecode reader. The timecode is displayed as SMPTE standard HH:MM:SS:FF (hours: minutes: seconds: frames) including the frame rate/format read from the stream.

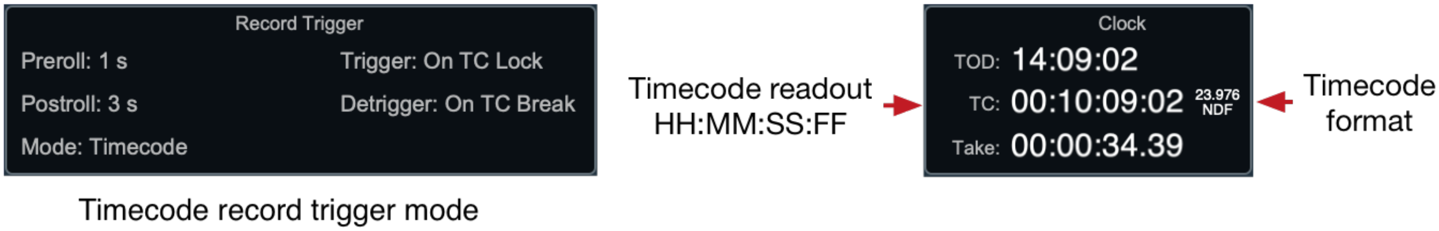


Figure 12.40: Session Transport: Timecode Trigger Mode

To the right of the Preroll and Postroll controls in the Record Trigger UI are the “Trigger” and "De-trigger" mode selectors. Both Preroll and Postroll are operational in Timecode Trigger mode and have a profound affect on timecode-based recording functionality.

Timecode Trigger can be set to either “Manual” or “On TC Lock”.

“On TC Lock” will trigger a new Take immediately upon timecode stream establishment at the timecode reader. As with Manual and Level-based triggering, audio in the Preroll buffer will be printed at the head of the new Take, ensuring that no audio is lost by a delay in the establishment of a good timecode stream.

Timecode De-trigger can be set either to “Manual” or “On TC Break”.

“On TC Break” triggers the Postroll timer when the timecode stream stops, ending the Take when Postroll times out.

If Timecode Trigger mode is set to “On TC Lock”, Postroll can act like a ‘freewheel’ over breaks and pauses in timecode - if timecode resumes before the Postroll runs out, the Postroll timer is cancelled and recording will continue until the next timecode break.

Regarding audio file timestamps:
When SMPTE is present, the SMPTE frame location at the moment record is engaged is used as the audio file start timestamp. This ensures that audio files recorded in Session can be directly imported by timestamp to a third party DAW timeline and recorded program will be located in sync with the original SMPTE.

Any preroll in the audio files is automatically compensated in the timestamp such that SMPTE and program remains in sync.

Autopunch Trigger Mode

AutoPunch trigger mode automatically enters recording at a user-specified time (the In point) and exits record mode at a later user-specified time (the Out point).

Figure 12.41: Session Transport: Auto Punch Trigger Mode

Autopunch In/Out points are specified in the In/Out timeline. The selected Autopunch range is highlighted red when active

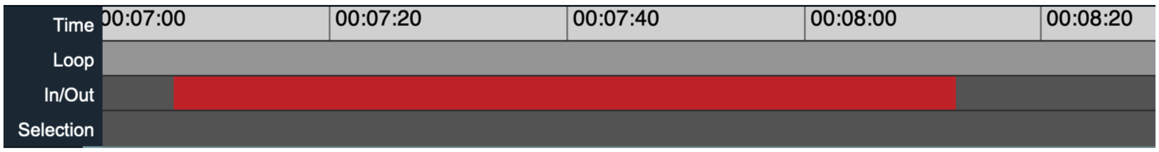


Figure 12.42: Session Transport Time block: Autopunch In/Out display

...and the Autopunch In point, Out point and Length (the duration of the punch) are reflected in the Transport header bar In/Out display:

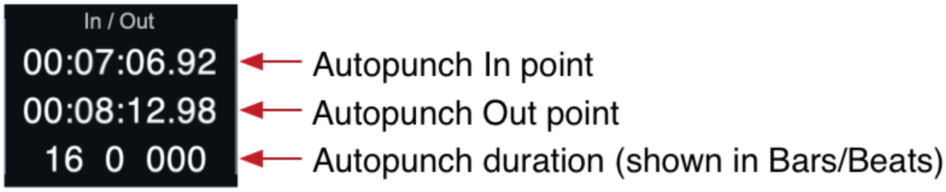


Figure 12.43: Session Transport Time block: Autopunch In/Out display

Again, as with all the time block displays, right (or control)-clicking any of the time fields will let you change the readout to hours:minutes:seconds, bars/beats or samples.

“Take” block

The **“Take”** block appears to the right of the Transport Control section.

- **Current** Take shows the number of the currently recording Take. This number will remain current after a recording is stopped, and will update only at the start of a new Take.
  “Current” Take Number is a read-only field.

Figure 12.44: Session: “Take” display

- The **Next Take** field shows the number of the next Take. Click on this field to edit the ‘Next Take’ number.

Figure 12.45: “Next Take” field: click to edit

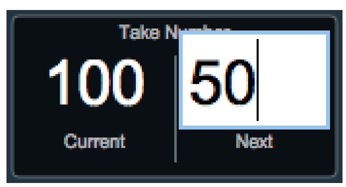


Figure 12.46: “Next Take” field: with new Next Take number

- Hitting the ‘Return’ or ‘Tab’ key commits the change:

Figure 12.47: “Next Take” field: new Next Take number committed

- When the new recording is triggered, the ‘Next Take’ number is transferred to ‘Current Take’ (and is used as the take number for the recorded filenames), and the Next Take number is automatically incremented:

Figure 12.48: “Next Take” field: auto-increments at the start of each new Take

Organizing and Identifying Takes

Safety First - Duplicate Filename Handling

The record engine doesn't really have a concept of takes once they hit the disk, so it keeps track of the take number, and, if you have "New folder per take" enabled, generates the directory name for the Take folder and the Track file names based upon the templates you have set up and the current take number (and other dynamic metadata).

It then creates the folder (if you have "New folder per take" checked) and the files for the take when the take is started.

If any of the files or folders that it attempts to create already exist, then it appends a disambiguating sequence number to the end of the name to ensure that nothing is overwritten.

If, for example, you have the following numbered Take files already recorded in your Record folder:
Take 0100
Take 0101
Take 0102

and you set the next take number to 99, and then hit 'Record' three times you'll get:
Take 0099 ---- (New take)
Take 0100 ---- (Old take)
Take 0100(2) - (New take with disambiguation number)
Take 0101 ---- (Old take)
Take 0101(2) - (New take with disambiguation number)
Take 0102 ---- (Old take)

If you happen to have the Take number reset to 1, and you wind up doing, say, three different recordings into the same record folder (when the next take is 1, e.g. after relaunching the console or manually resetting it), you'll wind up with:
Take 0001
Take 0001(2)
Take 0001(3)

The same thing happens for the files within a Take; generally, that's not relevant if you have "New folder per take" checked, but if you don't then multiple takes will go into the same folder, and if the file name doesn't disambiguate between the takes then this mechanism will.

Also, for the file names if you wind up somehow generating the same file name within a take (e.g. you just use the trackname for the file name, and you don't include the file number or track number, you can wind up with the same file name if you happen to have multiple strips with the same name), the console uses the same mechanism to ensure the names are unique (so that no data is lost).

This disambiguation is stateless within MIO Console, so it only is going by the name it wants to create and what already exists on the disk; the disambiguation number starts at 2 and it increments it in a loop until it gets to one that allows it to create a new filesystem object without overwriting one that already exists.

This means that the sequence number is determined per folder/object and if there are multiple duplicates on a given take, they may not all have the same sequence number — but that is a pretty degenerate case; you'd have to go out of your way to get that to happen.

This also means that the sequence numbers may be different for the same take in two different record folders.

Take Folder and Audio File naming tools

The Take management engine described above ensures that the Session record engine will not inadvertently over-write any previously recorded files, but the generic Take number is hardly sufficient to keep your recordings organized.

The 'MIOConsole3d Preferences: Recording' panel provides a scripting engine to help make Take management more livable.

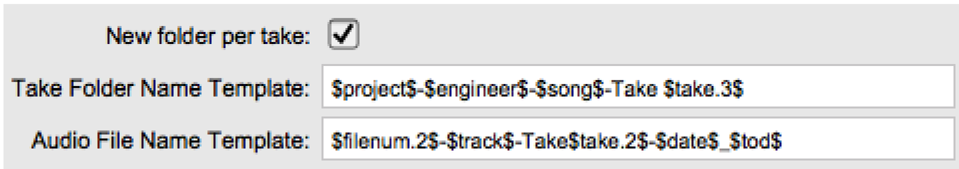


Figure 12.49: Recording Prefs: "New Folder Per Take", "Take Folder" and "Audio Filename" template fields

- **New Folder per Take** when checked, places all the files of each take in its own nested folder within the Record Folder and Mirror Record Folder(s). These Take folders are named per the parameters set below…
- **Take Folder Name Template**: This text field lets you auto-name new 'take' folders using the text entered in the Recording Preferences **'Project'**, **'Engineer'**, **'Song'** and **'Take Name'** fields, as well as a number of other variables. Understanding how to use this feature makes organizing crazed recording sessions *much* easier. If you use Session, this feature will be like a new best friend - you may hate it at first, but once you get to know it you will not want to live without it. Here is the **Take Folder Name Template** tooltip pop-up:

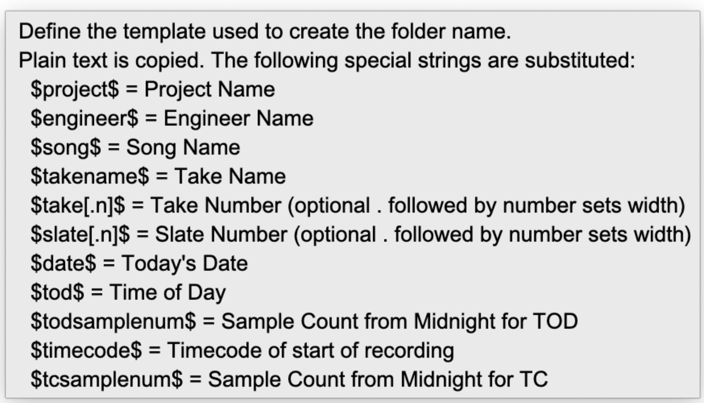


Figure 12.50: Take Folder Name Template **tooltip pop-up:**

…and here's the step-by-step breakdown:

The 'dollar sign's are used to define "tokens". These "tokens" command the folder-naming script where to look for information to copy into the folder name.

The **$project$** token says: look in the **"Project"** field, copy that text and paste it in the folder name.

The **$engineer$** token says: look in the **"Engineer"** field, copy that text and paste it in the folder name.

$song$ says: look in the **"Song"** field, copy that text and paste it in the folder name.

$takename$ says: look in the **"Take Name"** field, copy that text and paste it in the folder name.

$take$ says: copy the 'Next Take' number and paste in into the folder name. The [.n] sets the number of digits used to represent the take number. If you don't specify a number here, the take numbers will be 1, 2, 3, 4 etc. If you enter 2 (for 2 digits), you get a leading '0', so the take numbers go 01, 02, 03, 04 etc. The leading '0' makes listing files alphanumerically much cleaner. If you expect more than 100 takes, set it to 3 (for two leading zeros), set to 4 for a thousand takes, etc. The brackets don't get typed in, just the dot and the number. For example, if you are at take #4 and you had entered **$take.2$** in your template, the template script would insert **Take 04** in the folder name.

$date$ says: look up the current date and paste it in the folder name. The 'date' format is year-month-day, so July 22, 2019 would appear in the folder name as "2019-7-22".

tod says: paste the current Time Of Day into the folder name. The 'time' format is hours-minutes-seconds using a 24-hour clock, so 4:28 and 17 seconds in the afternoon would appear in the folder name as "16-28-17".

$todsamplenum$ will insert the number of audio samples counted from midnight to the start of the recording. This is essentially just a hyper-accurate measure of the 'Time Of Day' the take was initiated. This feature is mostly used when providing files to automated media ingestion systems.

$timecode$ will insert the SMPTE timestamp at the start of the recorded file.

$tcsamplenum$ will insert the number of audio samples calculated from a linear time code reference. (Note: if the LTC is locked to 'Time Of Day', this will be the same as **$todsamplenum$**). This is essentially just a hyper-accurate measure of the time of day the take was initiated. This feature is mostly used when providing files to automated media ingestion systems.

- **Audio File Name Template** is similar to **Take Folder Name Template**, but as one might expect, it is for audio file names rather than take folders. Here is the tooltip pop-up:

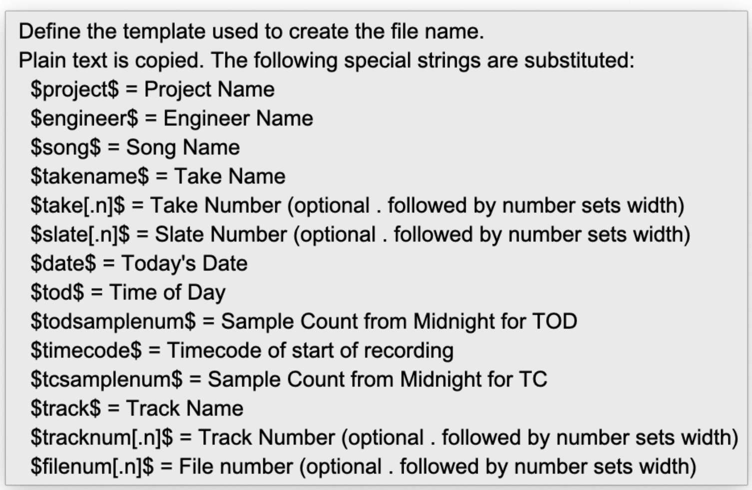

Define the template used to create the file name.
Plain text is copied. The following special strings are substituted:
$project$ = Project Name
$engineer$ = Engineer Name
$song$ = Song Name
$takename$ = Take Name
$take[.n]$ = Take Number (optional . followed by number sets width)
$slate[.n]$ = Slate Number (optional . followed by number sets width)
$date$ = Today's Date
tod = Time of Day
$todsamplenum$ = Sample Count from Midnight for TOD
$timecode$ = Timecode of start of recording
$tcsamplenum$ = Sample Count from Midnight for TC
$track$ = Track Name
$tracknum[.n]$ = Track Number (optional . followed by number sets width)
$filenum[.n]$ = File number (optional . followed by number sets width)

Figure 12.51: Take Folder Name Template **tooltip pop-up:**

The syntax for file naming is the same as for folder naming, although there are some extra "tokens" specifically for use in audio file names. Here's the step-by-step breakdown:

$project$ copies the text from the **"Project"** field and pastes it in the file name.

$engineer$ copies the text from the **"Engineer"** field and pastes it in the file name.

$takename$ copies the text from the **"Take Name"** field and pastes it into the file name.

$take$ copies the 'Next Take' number and pastes in into the file name. The [.n] sets the number of digits used to represent the take number. If you don't specify a number here, the take numbers will be 1, 2, 3, 4 etc. If you enter 2 (for 2 digits), you get a leading '0', so the take numbers go 01, 02, 03, 04 etc.

$date$ pastes the current date to the file name. The 'date' format is year-month-day, so July 22, 2019 would be entered as "2019-7-22"

tod pastes the Time Of Day at the start of the take into the file name. The 'time' format is hours-minutes-seconds and uses a 24-hour time scale, so 1PM is represented as '13' hours.

$todsamplenum$ will insert the number of audio samples counted from midnight to the start of the recording into the file name. Again, this feature is mostly used when providing files to automated media ingestion systems.

$timecode$ will insert the SMPTE timestamp at the start of the recorded file into the file name.

$tcsamplenum$ will insert the number of audio samples calculated from a linear time code reference. (Note: if the LTC is locked to 'Time Of Day', this will be the same as **$todsamplenum$**).

$track$: Each Track in the Session corresponds to a Mixer strip. The name of each strip being recorded is always visible in the track Overviews in the Session window. **$track$** pastes the name of the track into the file name. In cases where the audio routed to the track is post-process (either 'To Host: Post-Insert' or using the post-process Direct Out) the term **[POST]** will be added to the track name.

$tracknum[.n]$: The Session Tracks Overview window shows each Mixer strip which has return routes to the Host computer assigned. The position of these tracks corresponds to their position in the Mixer: so strips at the left of the Mixer are shown as tracks at the top of the Session Overviews. Using the Session Overviews as a map, the topmost track will be track 1, the one below it is track 2, and so on.

$tracknum[.n]$ numbers all tracks shown in the Session Overview, whether they are Record-Armed or not. In cases where not all tracks are armed for any given take, the recorded track numbers will have gaps in their sequence, but they will always match the same source audio and **$track$** name from the Mixer.

$filenum[.n]$ is similar to **$tracknum[.n]$**, but counts only the files actually being recorded.

Recording “Takes”: Manual and Automatic features

Taking a cue from film and video production, Session refers to each new recording as a ‘Take’. A Session Take can be defined as a set of files recorded simultaneously to the same media directory with a common naming convention and identical start and end timestamps.

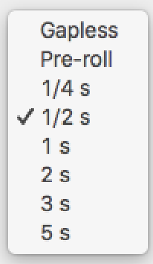
Takes can be triggered manually by hitting Record Start, or automatically via Session Record Trigger Modes.

The Take count is automatically incremented at each record start.

The Take count is reset to **1** when a new scene or song title is entered in the Scene/Song field.

Please note that the Slate count, while incrementing with each Take, is reset manually. This affords a method of grouping Takes by day, by geographical location, etc. across multiple scenes or songs.

New Takes can also be manually initiated during a running recording by hitting the ‘Record’ button. Manually triggering a new Take during recording is called a “Manual Break”. Triggering a Manual Break will immediately start writing a new set of files, with (or without) overlapping audio with the previous Take, and appropriately named per the Take Number display and your settings in ‘MIOConsole3d Preferences > Recording’ window.

> “Manual Break Overlap” is a preference in the ‘MIOConsole3d Preferences > Recording’ window, which lets you set the amount of overlap between the two takes in case you want to splice them back together.
>
> 
>
>
> **Figure 12.53: Manual Break Overlap selector**
>
> “Manual Break Overlap” can be set to ‘Gapless’ (ie. a butt-splice with no overlap), a set duration between 1/4 second and 5 seconds, or it can use the Session Record Trigger ‘Preroll’ setting.
>
> Overlapped audio will always be included in the New Take files. In other words, the currently recording Take will end immediately at the break, and the beginning of the new Take will include the overlapping audio from *before* the break.

In ‘Count In > Takes’ mode, each Take gets a marker dropped at its start point fade-in. If Preroll is enabled, the preroll audio will be present but hidden to the left of the start point fade-in. Postroll audio is included as part of the Take. Take markers will show the full duration of the Take (not including the preroll), and are automatically added to the Marker List the the left of the Session Track Overview.

> Note that Session automatically selects all segments of a Take the moment the Take is finished recording. This makes it easy to quickly delete the Take in case of false starts. This does not delete the soundfiles, just the Take segments within the Session Overview.

Slate block

The **Slate** block appears to the right of the Take section.

Slate is a numeric tally which increments automatically along with each Take, *but unlike Take*, Slate does not automatically reset its count when entering a new Scene/Song title. This lets you sequence slates by day, by geographical location, etc. across multiple scenes or songs while the Take counts respect each scene or song.

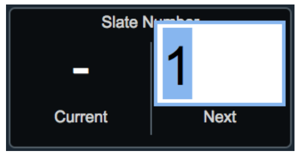


Figure 12.54: Slate field: click to edit

Functionally the Slate block works exactly the same as the Take block: click on "Next" to enter the next slate number (usually a reset to **1**) and the tally starts from there at the next recording start. See Take Block numbering for a more graphical presentation of the process.

The Slate number is included in the file and folder naming templates scripts.

Session Metadata block

The **Session Metadata** block appears to the right of the Slate section, serving as an accessible data entry point for the Recording Preferences **Project**, **Director**, **Engineer** and **Soundroll** metadata fields.

Figure 12.55: Slate field: click to edit

Click in the desired field to edit, and your changes will be applied to the folder and file naming scripts and metadata fields of the next recording.

Take Notes block

The **Take Notes** block appears to the left of the Record Folder section for free-form notes describing the next Take to be recorded.

Figure 12.56: Take Notes field: click to edit

Click in the desired field to edit. Take Notes may be entered live during a recording, and will be applied to the iXML metadata at the end of the recording.

To be clear, Take Notes is an iXML metadata field, and is unrelated to Session timeline Marker notes.

Record Folder block

The far right of the Session Transport bar houses the Record Folders control interface. Here you can assign destination folders for Session captures. If you have "New Folder per Take" selected in your Recording Preferences, every Take will be recorded in its own folder nested within this Record Folder.

- **The Record Folder**

  Upon first launch of the MIOConsole3d, a default Record Folder called "Metric Halo Session" will be created in the Music folder of your home directory.

  Holding the mouse cursor over the folder name will pop up a tooltip revealing the full file system path to that Record Folder:

Figure 12.57: Record Folder location path tooltip

  This locator tooltip is always available for any folder listed in the Record Folder interface, even if the folder is not currently visible to the Host computer.

  Clicking on the name of the current Record folder will open a folder navigation window focused on your Record Folder. From here you can designate a different folder for your recordings or create a new Record Folder in any location you choose. This function is also found in the menu bar "Session" menu:

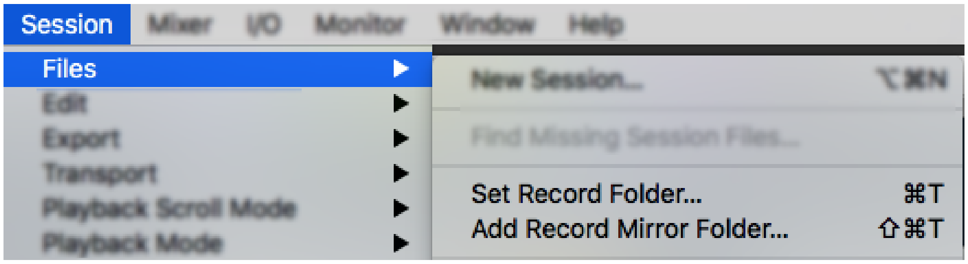


Figure 12.58: Session: Files: Set Record Folder...

- **Record Mirror Folders**

  Session also supports simultaneous recording to multiple additional record folders, called "Record Mirror Folders". Record Mirror Folders are also created from the menu bar "Session" menu, as shown above.

  Mirrors simultaneously capture the same data as the main Record Folder - every Take folder, every channel, every literal bit, just in a different location.

  Mirror folders can be located on any volume visible to your Host computer, generally secondary internal drives or external USB3 or Thunderbolt hard drives.

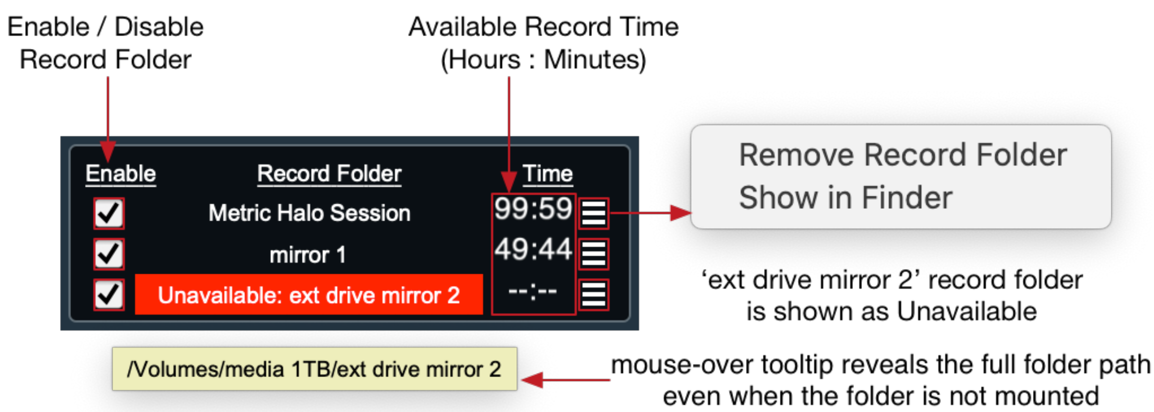


Figure 12.59: Transport Panel: Record Folder controls detail

- **Enable / Disable Record Folder**

  To the left of the Record Folder selector is a checkbox where you can "Enable" or disable any given Record folder temporarily. This feature could be used, say, to prepare multiple record volumes ahead of a long day of recording and easily switch to fresh record volumes as the first set of drives starts getting full.

- **Available Recording Time display**

  To the immediate right of the Record Folder controls is the available recording time display ("Time" for short). This readout shows the amount of recording time you have available on each record volume, based on your sample rate, your Preferences for recording bit depth and filetype, and the number of currently *Record Enabled* channels in your Mixer (ie. with the Record Enable button showing Red).

  The available record time is displayed in Hours and Minutes.

- **Record Folder hamburger menus**

  Next to the Time display is a small hamburger menu for the Record folder and each Mirror Folder. Clicking on these will open a menu where you can 'Remove' the selected folder, or 'Show' the folder in the Finder.

- **Unavailable Volumes**

  If a Record or Mirror folder is not visible to the Session, it will be highlighted bright red and labeled "Unavailable" in the Record Folders list. Use the folder path tooltip to reveal the location of the target folder.

  Re-mounting the missing volume will automatically restore it as available in the Record Folders list.

  If the folder was a backup on an external drive which has already gone home with the client, use the hamburger menu to remove that Record Folder from the list.

Record Drive considerations

Even though the massive pre-roll ring buffer is always active and working to compensate for slow record drive write underruns, faster drives are naturally always recommended for the primary Record and Record Mirror volumes. SSDs or 7200rpm (or better) enterprise-class hard drives should be considered minimum requirements for critical recording sessions.

> **Very important:** If you are using a USB3.x -> Gigabit Ethernet adapter for your MHLink connection, by all means isolate your record drives on a separate USB3 bus or on Thunderbolt. While on paper there should be plenty of bandwidth, both HD/SSD and Gigabit Ethernet adapters are asynchronous by nature and the drives tend to burst data in big chunks rather than stream it in a steady flow (like you need for audio). Having both on the same USB3 bus creates a situation where data bursts for the drive might interfere with the MHLink data streams.
>
> Due to the ever-expanding array of USB3 adapters on the market, we are unable to comprehensively test this scenario, but it is a very real possibility…

Recording to SDHC or USB3 flash memory modules is possible, but such devices are highly optimized for fast transfer of large blocks of data rather than the constant stream of time-sensitive chunks that you write when capturing audio. Additionally, these modules are typically set up to *Read* much faster than they *Write*, which is exactly wrong for confidently recording audio. Even worse, it is not uncommon to find that the same flash drive will write data twice as fast on one USB3 interface than it does on another.

Given all of the above, it is always recommended that you test any prospective record volumes (regardless of form factor) with a well-reviewed disk speed test application, and run a full real-world rehearsal of both record and playback performance before going into any session.

Recording logfiles

Every Take captured by Session is accompanied by a "Recording.log" file. The Recording.log is a generic text file which can be opened in any text reader, and includes timestamps, file path and file type for every file in the Take, Number of Tracks in the Take, Number of Channels in the Take, Number of Mirrors, Number of Drives, Sample Rate, average bandwidth per drive... lots of metadata which proves surprisingly useful when prepping recordings for submission to automated ingestion systems, and for monitoring the relative health of your recording system.

Session: Control Widget Bar

Immediately below the Transport Header Bar lives the **Control Widget Bar**. The Widget bar includes controls for overdubbing, editing, grid modes and window zoom controls.

The Widget Bar is shown and hidden along with the timelines and tracks when toggling *Show Overviews* in the Session hamburger menu.

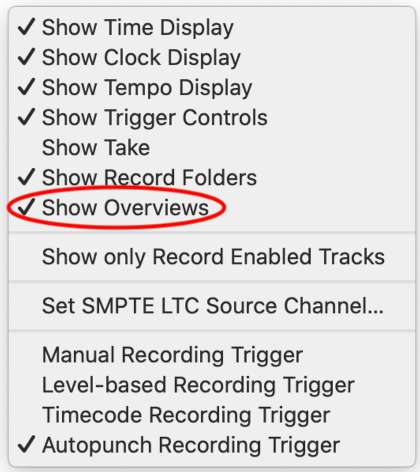


Figure 12.60: Transport Header bar hamburger menu: Show Overviews

"Missing Files…" flag

The space at the left edge of the Widget Bar may upon occasion display a yellow *Missing Files…* button. This is a notification that your Session needs help locating some of its audio files.

Figure 12.61: Widget Bar: Missing Files

Clicking *Missing Files…* will launch the "**Locate Missing Sound Files**" window, where you can locate the missing files, or remove references to those files from the session.

Locating missing files is a smart process, in that locating one file in a particular location will automatically register any other missing files found in that location.

In cases where files are organized within a common folder heirarchy which has been relocated intact, select the highest enclosing folder at its new location, and all of the audio folders and files within will be recognized and loaded.

Recognized folder and filenames will change from yellow text to white. Hit **Return**, **Enter** or the "Finish" button to finish, and remember to **⌘ S** (Command + S) to "Save" the Session with the new audio locations, or **⌘⌥** (Command + Option + S) to "Save As…" with a new name.

Tracking and Overdubbing controls

Session supports unlimited overdubs to be recorded to any track. Each new take will be added as an audio segment in its own track lane, with the original take in the top lane, and the overdub takes added below.

During active recording, waveforms of the audio being recorded will be always displayed using the entire track height, regardless of track lanes view mode. This is to ensure maximum visibility of the captured audio.

The screenshot below shows audio being punched-in over a previous take. While the background track waveform color will vary based on your mixer strip color choices, audio being recorded will always be shown as pure white waveforms on a red background.

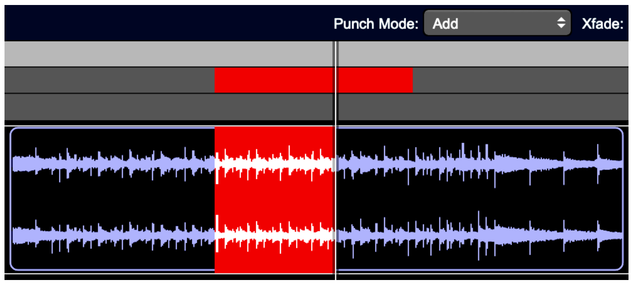


Figure 12.62: Recording in progress (autopunch mode shown)

The Widget Bar **Punch Mode**, **XFade** and **Time** controls define the behavior of new audio recordings to a given track.

Figure 12.63: Punch / new segment behavior controls

Punch Mode controls how new audio is inserted into a track when overlaps occur due to either dragging a segment or from overdubs.

All lanes within a track are summed in the playback engine before being routed back to the MIOConsole3d mixer, including individual segment gains and fades. The various punch modes are designed to streamline overdubbing such that multiple passes can be taken immediately without breaking the flow.

XFade sets the fade shape applied to new segments and edits.

Time sets the duration of new segment and edit fades.

Set the Xfade and Time fade parameters to ensure seamless crossfade transitions for your specific program material. Generally the default settings will work well, but if you are dubbing to an up-tempo rhythmic loop, a faster fade slope with a shorter duration can help keep take transitions in the pocket. Conversely, layering atmospheric pad loops might benefit from more linear fades and longer crossfade durations.

Punch Mode

Punch Mode determines what happens to existing audio in the track when you overdub into occupied space in the timeline.

There are four dubbing behaviors to choose from:

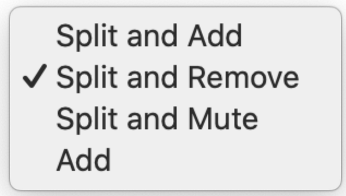


Figure 12.64: Punch Mode menu

The *Split* modes place edit crossfades in all prior audio segments at the punch in and out points. The automatic segment handling of the Split modes helps to speed selection, auditioning and editing of the original take and dubbed takes.

Split and Add

Split and Add is especially useful when layering multiple instrument passes over a loop to be mixed and comped in-place within the track.

Figure 12.65: Punch Mode: Split and Add

The overdubbed take is split to a new segment at the punch points, with the new take appearing as its own segment in a new lane below. The overdubbed audio segment remains unmuted, and when auditioned both the original and new take will be heard.

The advantage to this mode is that all takes including the original can immediately be soloed, muted or gain-shifted and mixed in place with transparent crossfades to the original audio.

Split and Add is perfect for quickly double-tracking a solo or harmony line without having to add extra mixer channels.

Loop recording in this mode greatly streamlines the creation of complex beats based on live recordings as well as improvising multiple passes over a given phrase.

Split and Remove

Split and Remove splits the overdubbed audio at the punch points, records the new take and removes all prior audio from the overlap.

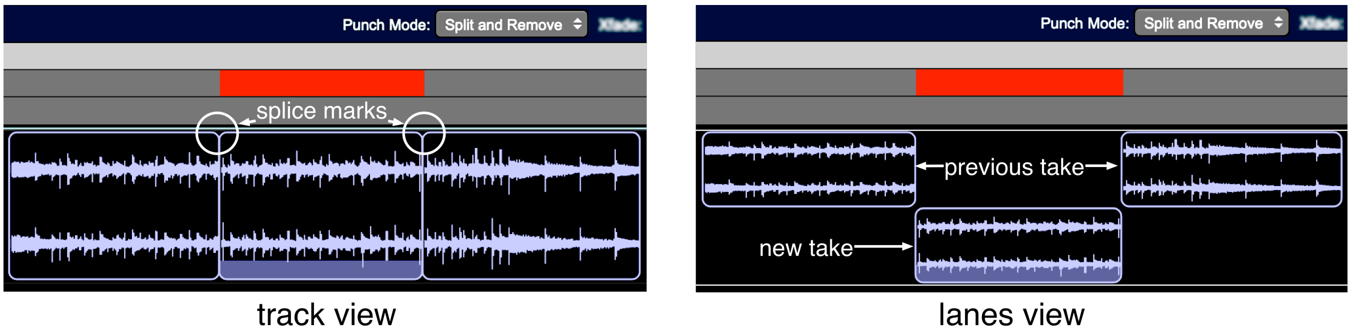


Figure 12.66: Punch Mode: Split and Remove

Use this mode to quickly replace a bad phrase or passage. Additional passes over the same punch points will remove the previously recorded segment.

Functionally, it works like punching over a track on a physical tape recorder. A Split and Remove punch will perform like it is overwriting the existing audio, when it's really just an insert edit, keeping the original audio file intact behind the edit.

Split and Mute

Split and Mute works the same as Split and Remove, but rather than removing previous takes, it mutes existing segments in the overlap, leaving only the newest take audible.

Figure 12.67: Punch Mode: Split and Mute

Split and Mute is for capturing multiple rehearsal takes while working out parts and adding variations to a base phrase. This would be the go-to mode for experimenting with new lines played over a looped phrase, as any previous passes would be muted allowing you to concentrate on the next idea.

Add

Add mode records new audio segments as additional layers in the track, without splitting the previous audio at the overlap.

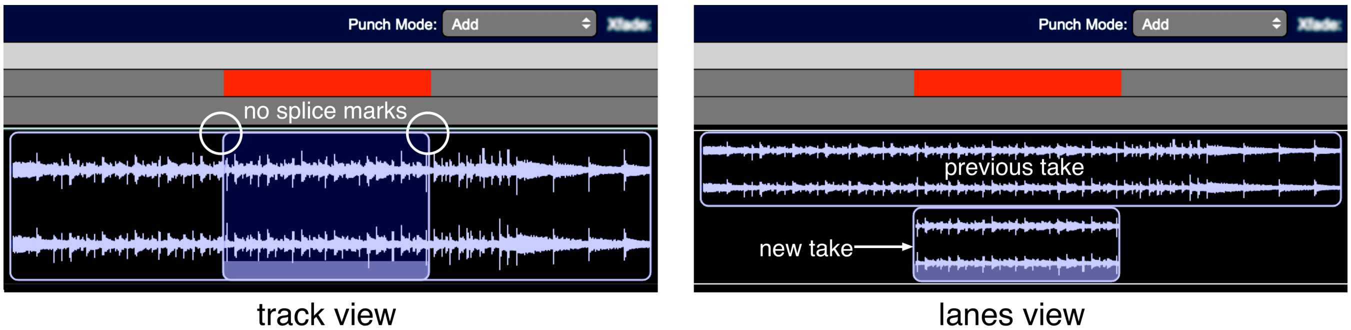


Figure 12.68: Punch Mode: Add

The sole difference between Add and Split and Add modes is the convenience of the perfectly-placed splices Split and Add inserts on the prior audio takes.

This mode would be advantageous for single-pass takes, free-form music recordings, or in any case where you know ahead of time you will be editing/mixing takes independently of the punch in and out points.

In practice, you will find that tailoring the punch mode to your workflow and managing takes within track lanes will provide more control over your gain structure, simplify your mix desk layout and even reduce the processing load in your mixer strips.

XFade: Crossfade / Fade slope type

The **Xfade** popup lets you set the slope type of fade or crossfade to be applied to new segments.

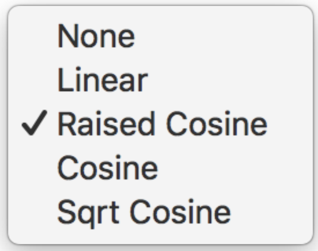


Figure 12.69: Xfade menu

All crossfades generated by punch-in or drag-n-drop file importing to a track are mathematically complementary, providing uniform gain across the duration of the crossfade. In the majority of cases these automatic edits will be inaudible, but there is no "one size fits all" in audio. Adjust the Xfade shape (and Time) to optimize fades to your program material and workflow.

Xfade: None

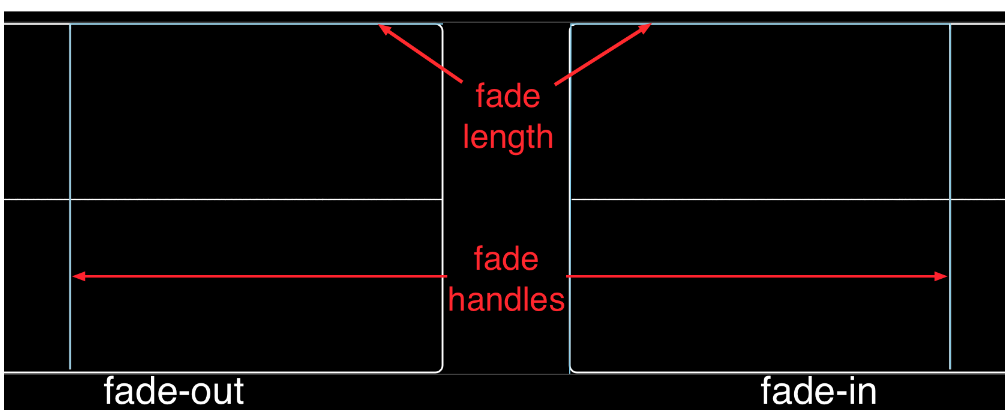


Figure 12.70: ***None*** **(zero-duration crossfade - a butt-splice)**

When *None* is set as the fade type, there is of course no fade slope to show. The fade tool itself is still present and you will still be able to grab the segment *fade duration* handle holding by the "F" key as usual, but in lieu of a fade slope, the crossfade duration will be drawn as a light blue line along the top of the segment, ending at the fade handle cursor opposite the segment end.

If your goal is to ensure that the recorded audio file ends precisely at the zero-duration fade, make sure pre-roll and post-roll are both set to "Off" in the Transport Bar Record Trigger control.

Xfade: Linear

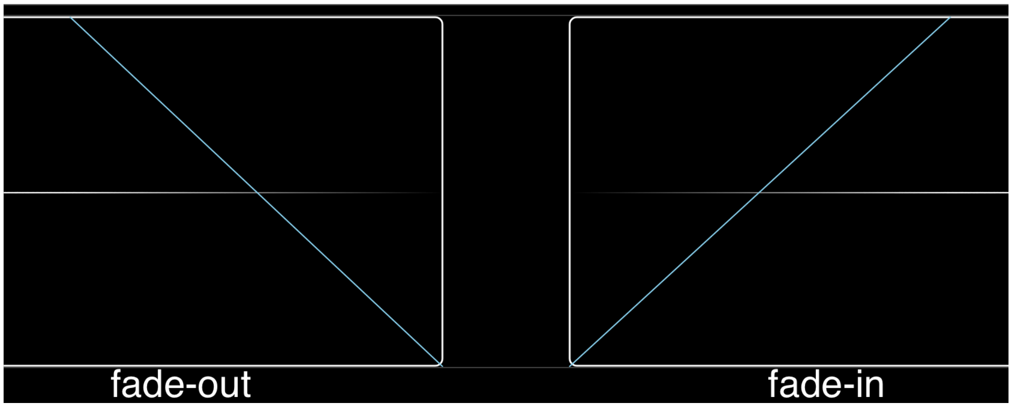


Figure 12.71: ***Linear***

The Linear fade shape provides a constant gain change across the entire duration of the fade. Linear is probably the most common fade type, and the default in many DAWs.

Xfade: Raised Cosine

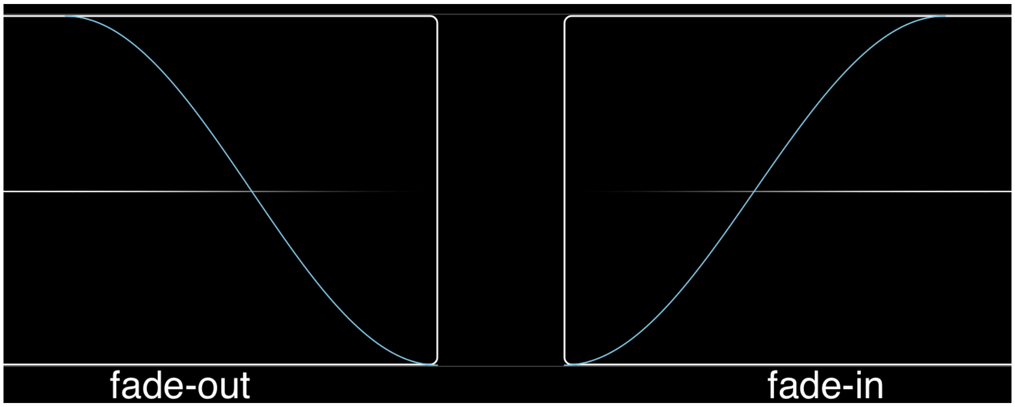


Figure 12.72: ***Cosine Squared* (Raised Cosine)**

Cosine Squared is the default Session fade shape. Cosine Squared features a slower gain change at the entry and exit of the fade, with a more rapid slope through the middle of the amplitude range. The more gradual entry and exit allows for tighter insertion tolerances where there may be amplitude offsets between the crossfaded segments or transient events close to the edit point.

Xfade: Cosine

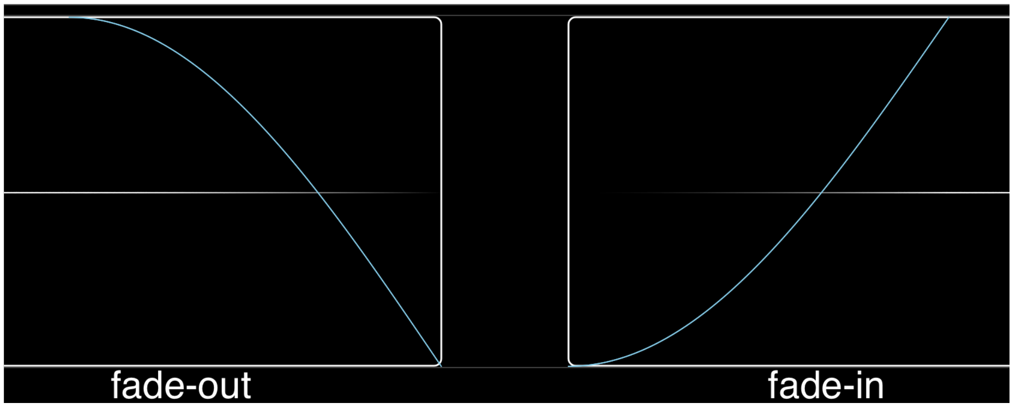


Figure 12.73: ***Cosine***

Cosine provides a slow gain change at the entry of the fade, with a near-linear slope at the exit. Cosine favors placement after a transient event, requiring a fast fade but still retaining a natural ring-out.

Xfade: Sqrt Cosine

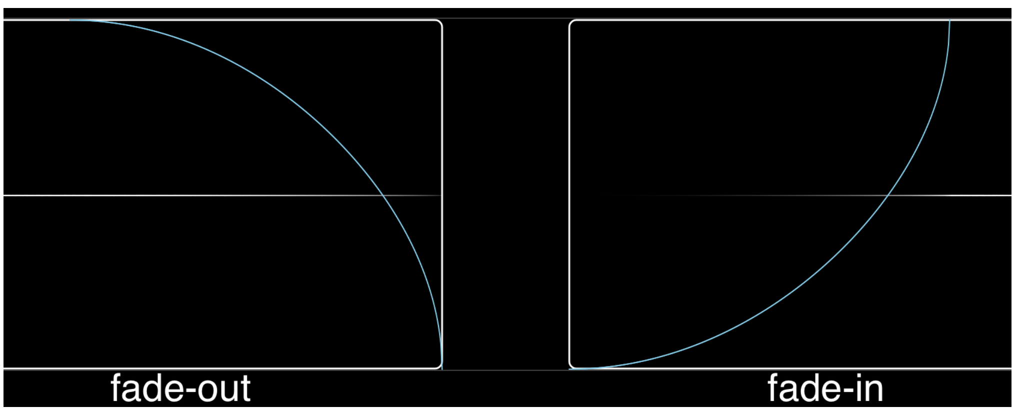


Figure 12.74: ***Sqrt Cosine*** **(Square-Root Cosine)**

Square-Root Cosine is similar in effect to Cosine but more exaggerated, with an even slower entry and a faster exit slope.

Time: Crossfade / Fade length

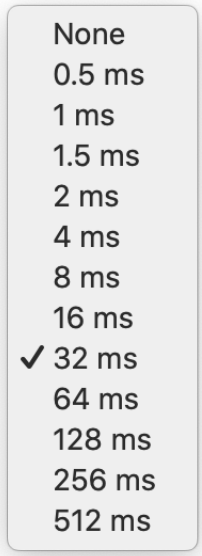


Figure 12.75: fade/crossfade length menu

The **Xfade Time** popup lets you set the length of automatically created fades when segments are created during overdubs or edited after the fact. Options range from “None” (zero duration fade - a butt-splice) to 512 milliseconds.

Naturally fade durations may be adjusted manually as desired.

Loop Record Mode

The **Loop Record Mode** popup lets you choose between the different loop recording modes when you record with loop playback enabled.

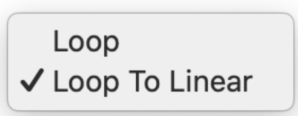


Figure 12.76: loop record mode menu

- **Loop Mode**: Recording starts when triggered, punches out at end of the loop and is rearmed for next loop pass; this means all new material is recorded within the boundaries of the loop. Each recorded pass drops to a new track lane with sample-accurate crossfades allowing seamless transitions.
- **Loop to Linear Mode**: Playback follows the specified loop, and recording starts at the playhead position when triggered, but continues recording linearly while playback loops. This mode is great for capturing long improvisations against looped phrases anywhere in the session.

  Note: Take recording mode allows you to record anywhere in the Session timeline while playing loops independently at completely different locations in the same timeline.

Grid

The **Grid** button enables snap to grid

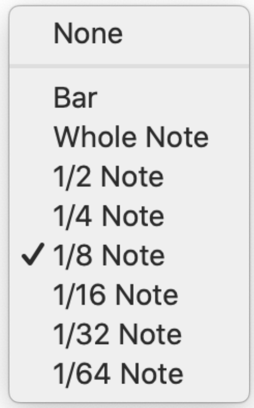


Figure 12.77: loop record mode menu

The **Grid** popup sets the granularity of the grid in terms of musical units from bars down to 1/64 notes. The **.** and **3** buttons modify the selected grid time to turn it into dotted (3/2) or triplet (2/3) duration respectively.

Vertical Scale

The **Vertical Scale** slider sets the height of the track overview lanes. The button associated with it toggles between the current view and zoomed all the way out to fit all the tracks into view (or set them to minimum size if there are too many to fit in the current window at the minimum track height).

Horizontal Scale

The **Horizontal Scale** slider sets the horizontal zoom of the track overviews The button associated with it toggles between the current view and zoomed all the way out to fit the entire session timeline into view.

Command	**Key Sequence**
Zoom to Fit Selection	⇧Z
Increase Session Track Height	⌘↓
Decrease Session Track Height	⌘↑
Scroll Up / Down	scrollwheel
Fit Tracks Vertically	⌥X
Zoom Tracks In Horizontally	⌘→
Zoom Tracks Out Horizontally	⌘←
Scroll Left / Right	⇧ scrollwheel
Fit Tracks Horizontally	⌥Z

Table 12.2. Session navigation: default key commands

13. DSP Implementation Guide

Signal Flow and Processing in the 3d Mixer

Every Input strip, Aux Bus, Group Bus and the Main Bus strip supports the insertion of latency-compensated plug-in processing.

All Input strips may be routed to any or all Aux mix buses, Group buses and the Main bus strip.

- All plug-in processing within the 3d Mixer occurs in the shared DSP/FPGA engines of the Metric Halo 3d hardware, completely independent of your Host computers' processor. The more 3d boxes you connect to your MHLink domain, the more processing power you have.
- All plug-in processing is fully latency compensated within the 3d Mixer domain, following the basic signal flow of: Insert strip -> Aux Bus strip -> Group strip -> Main Bus. Looping through multiple Group buses (with processing) on the way to the Main Bus is also latency compensated.

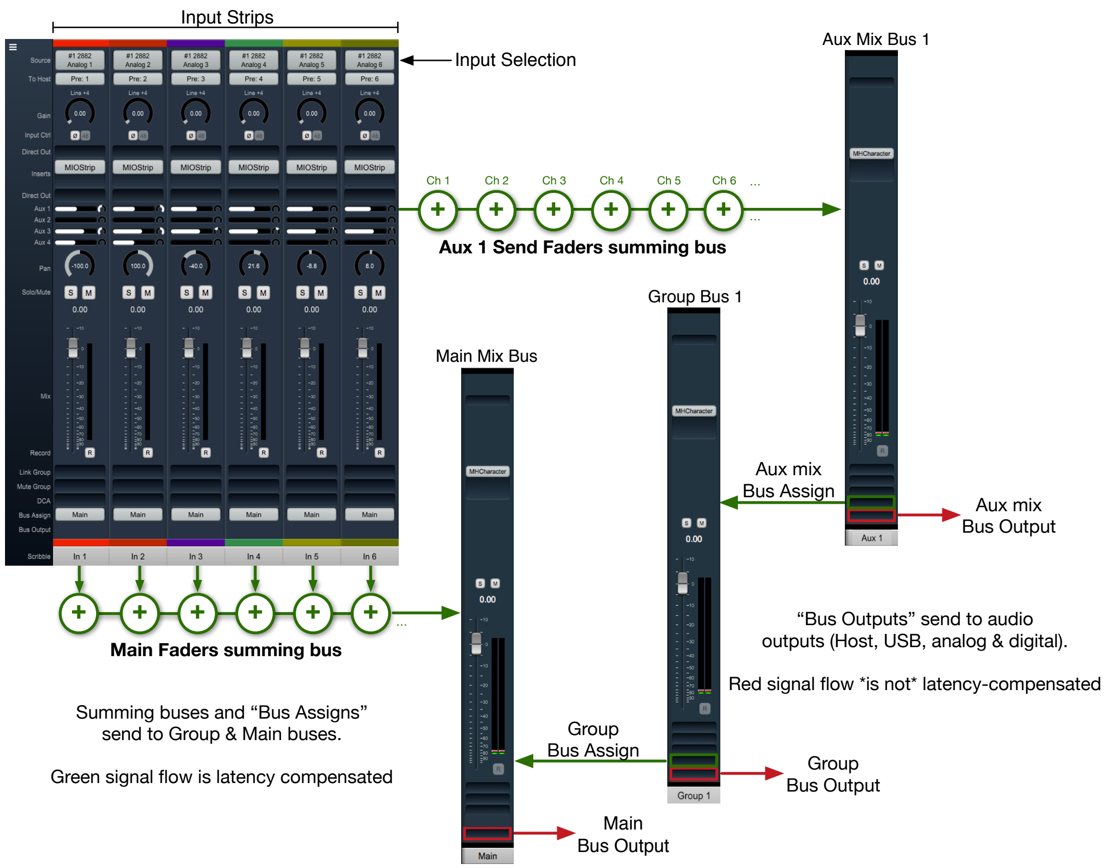


Figure 13.1: Basic 3d Mixer signal flow

In practical terms, this means a signal coming through any Input strip direct to the Main bus will be time-aligned with the same signal also routed through Aux bus mixer, then through Group buses, and then to the Main bus, regardless of how much extra processing has occurred at each bus stage (as shown in green, above).

Of course, any signal path that loops out to external devices via analog, digital or USB can not be automatically latency compensated. This includes "I/O" inserts, and any signal which leaves the mixer and returns via an Input strip (shown in red).

Inserts

Click on any empty stereo insert slot, and you will see the 3d Insert selector menu:

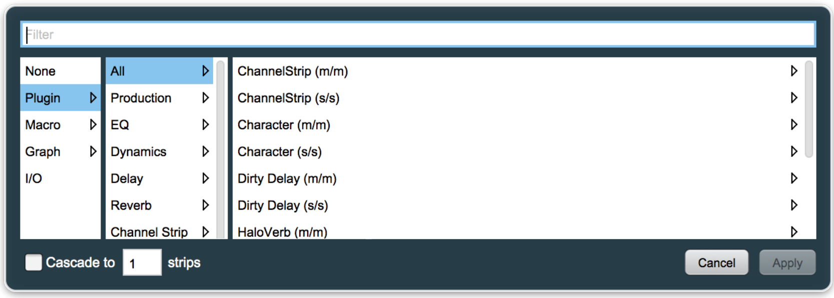


Figure 13.2: Insert Selection Menu

Across the top of the Insert selection menu is a search field. Type in any text string to search for processor type or saved presets within your system.

Below the search field are columns which break down the types of processes by broad terms (to the left) becoming more specific as you progress to the right. To see how this works, in the very left column select "Plugin", and note the subheaders that appear to its right.

Now select the "EQ" processor type. The next column lists MIOEQ12, MIOEQ6 and MIOStrip, mono and stereo versions.

As an example, select MIOStrip. Two more columns appear, one lists preset type categories (Cut Filters, De-Essers, Drums, etc.), and the next (to the right) contains saved presets within each of those categories.

Here you can select available plug-ins from the categories in the Plug-in menu, or select a I/O insert. You can also select Macros and Graphs, which are detailed in the Graphs section.

The plug-ins that may be inserted in any given slot depend on the number of channels of a given input channel strip. All mono plug-ins may always be inserted in any slot; if you insert a mono plug-in into a strip that has a multichannel input (for example a bus master strip or bus return strip, or a multichannel input strip), the mixer will automatically instantiate multiple copies of the plug-in (for example — two plug-ins into a stereo strip and 5 plug-ins into a 5.0 strip), and link the parameters of the instances so that when you control the inserted plug-in, it will control all instances. I/O inserts follow the same rule: an I/O inserted in a stereo mixer strip will route only stereo sends and stereo returns, etc.

If you are working with a multi-channel strip, only plug-ins that make sense for the number of channels of the strip will also be available. For example, with a stereo strip, you will see both the mono and stereo versions of the MIOComp and MIOLimit dynamics processors. You can select the version that works best for you. At the present time, there are few processors that have specifically been built for multi-channel strips with more than two channels. If you expect to use (or change to) a bus with multiple channels beyond stereo, you will probably want to use the mono version of the plug-in as they can be automatically instantiated as you adjust the number of channels of the strip. More info on using plug-ins via Inserts can be found on the DSP Implementation Guide.

Cascade: By checking the *Cascade* box at the bottom of the window, you can apply your selected plug-in to whatever number of strips you choose. The number in the *Cascade to ___ strips* box determines how many mixer strips will receive the new plug-in.

New plug-in inserts created in this way will all be placed in the same Insert slot across the desired number of channels. This method can also be used to replace existing Inserts.

Mixer strip Insert controls

Once you have selected a plug-in, it will be listed in the assigned insert slot:

Figure 13.3: Inserted MIOEQ6 Plug-in (as shown in Mixer strip)

Plug-in names will generally appear abbreviated in order to save space ("MIOEQ6" is shown above).

When you move the mouse over an inserted plug-in, the Insert label will change to show three control icons. The tooltips for each of these controls have been exposed in the example graphic below.

Figure 13.4: Inserted plug-in controls

- The "On / Off" switch icon on the left is the plug-in Bypass. When Bypassed, the Insert button will turn yellow.
- Clicking the "…" icon in the middle opens the inserted plug-in editor window. When the plug-in editor is open/visible, the Insert button will turn blue.

  Note: Normally opening a new plug-in window will replace any currently-open plug-in UI, such that only one is open at a time. <Shift>-click to open a new plug-in editor window while keeping existing plug-in UIs open.
- Clicking the "up/down" arrows icon at the right opens the Insert selector window, where you may select a replacement plug-in, or navigate to directly open a different saved preset without having to open the plug-in editor UI.

Insert modifier key shortcuts

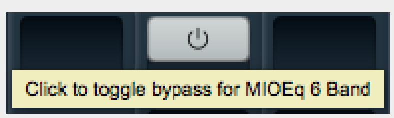


Figure 13.5: "⌘-click" / <Command>-click to Bypass Insert

<Command>-click the Insert button to Bypass the Insert.

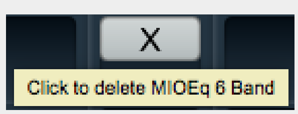


Figure 13.6: <Control-Option-Command>-click to Delete Insert

Use "^⌥⌘-click" / <Control-Option-Command>-click to delete an Insert directly from the Mixer desk.

Quick Copy/Paste Plug-ins

Option-click-drag any plug-in instance from one Insert slot to another anywhere in the Mixer to clone that Plug-in to the new location. Plug-in instances will automatically adapt to the channel width of the target Insert as necessary.

"Sweeping" controls

Toggle buttons on consecutive strips in the 3d Mixer desk can be switched in a single move by a click-hold-sweep gesture.

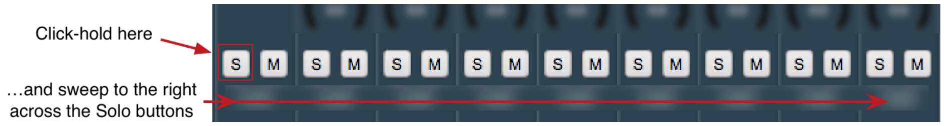


Figure 13.7: "Sweep" to toggle multiple buttons in one gesture

To try it, click-hold on a Solo button, and while holding the mouse button down, drag the cursor to the right or left across the mixer.

The move works with Polarity Invert, Solo, Mute and Record Enable buttons.

Insert types, Categories and UIs

So let's take a closer look at that Insert selector interface.

As you can probably tell by poking at it a bit, the columns are arranged with broad categories to the left, with more detailed subcategories popping up progressively to the right. Eventually the right-most column lists individual saved presets for each plug-in.

The hierarchy from left-to-right goes:
Insert Type - Processor Type - Plug-in Name - Plug-In Preset Category - Plug-in Preset Name

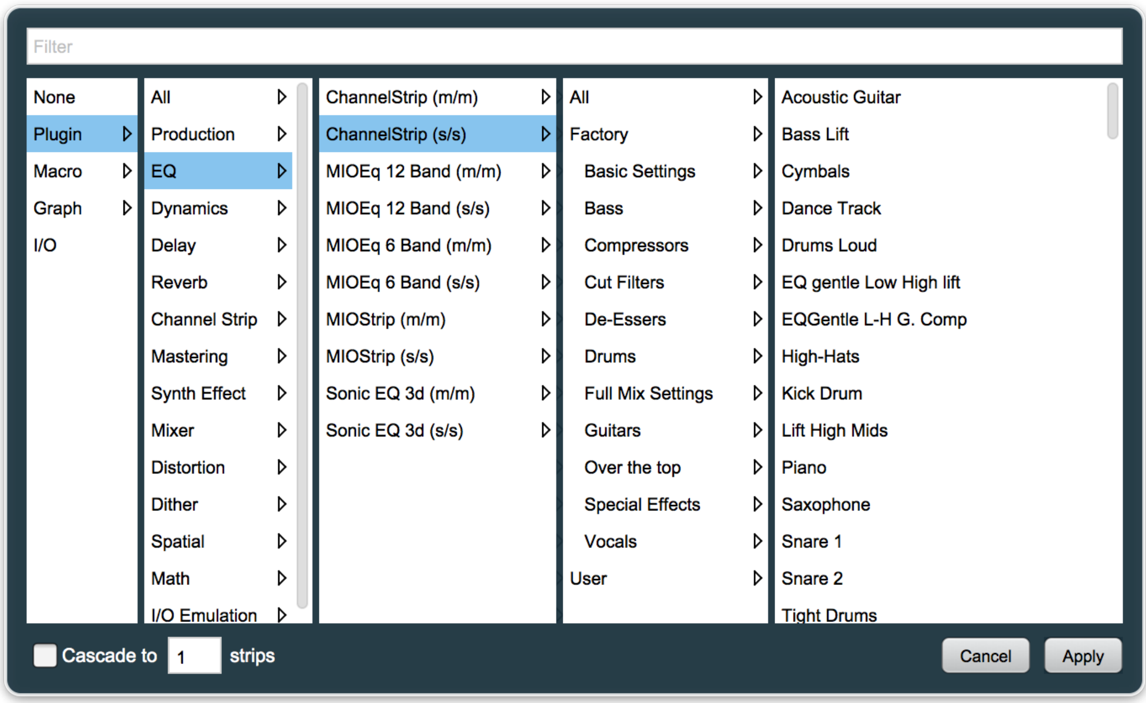


Figure 13.8: Insert Preset Categories and saved presets (Channelstrip stereo)

Note that the MH Production Bundle plug-ins (represented by ChannelStrip above) include sub-categories for *Factory* and *User* presets.

Across the top of the window is a text-entry field titled "Filter". This is a dynamic search engine which will filter the processors and saved presets shown in the selector. Type in a preset name, processor

Selecting a plug-in type, name or preset category will narrow the search to within your selection. In the example above, a search would show results only from ChannelStrip (stereo) Factory and User presets, whereas selecting "EQ" would return results from all of the EQ plug-in presets.

Shortcut! In the above selection window, double-clicking a plug-in name will immediately instantiate that plug-in at its default setting.

By the same token, double-clicking a preset name will immediately instantiate that plug-in with the preset settings.

Insert Types

The left-most column of the Insert selector lists one command followed by four types of inserts:

Removing Inserts: "None"

As noted earlier, you can use "^⌥⌘-click" / <Control-Option-Command>-click to delete an Insert directly from the Mixer desk.

Alternatively, "None" is always listed at the top of the left-most column of the Input selector as a method to remove an inserted plug-in. If you want to remove an Insert, open the Insert selector and select "None", then hit "Apply", the **Return** or **Enter** key.

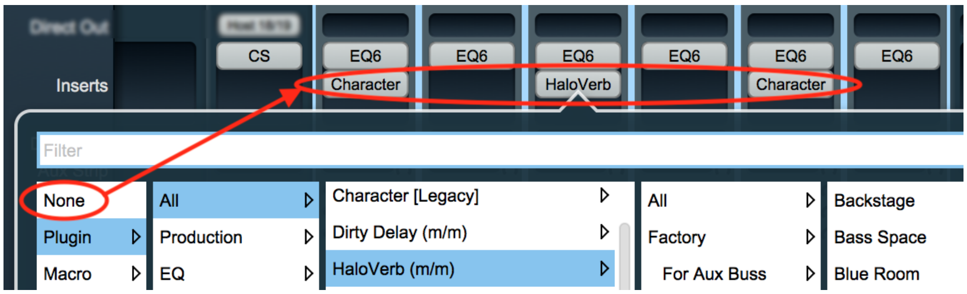


Figure 13.9: "Selected Strips": multiple Insert deletion

Note that selecting "None" in the selector UI respects "Selected Strips" link groups (when enabled), and allows deletion of multiple inserts in the same insert row simultaneously.

In the example above, the selection menu of the 'Haloverb' insert in slot 2 of the selected strips (highlighted in the lighter gray) is open. By selecting "None", and hitting "Apply", all of the slot 2 inserts in the 'selected strips' link group will be deleted.

- **Plugin:** Lists all of the mixer-insertable processors, from single-function building blocks (like time-corrected delays, M/S modules and dither blocks) to the 3d Production, "Exclusives" and "MIO Core" processor families.
- **Macro:** Currently there are actually two types of Macros.
  1. The first type are 'open' and can be thought of as Graph templates you can tinker with at will. These Graph-type macros can not be overwritten, but you can modify them as you like and save the results. Graph-type macros include all the plug-ins in the '2d Effects' and '2d Mastering' categories.
  2. The mono-specific '2d Amps', '2d Pedals' and 'Cabinets', and the stereo-only '2d Reverbs' macro categories are of the second type. These Macros have no user-tweakable parameters, and are designed for quick plug-and-play insertion.

     Need to cut a quick guitar line? Plug your guitar into the LIO/ULN8 front panel DI input, insert a 2d Amps/"British Mil Spec" macro and start playing.

     Need a short stereo drum room filler but don't have the time to mess around? Drop in "Early Diffuse Prime" and move on. It's amazing how removing the urge to tweak a plug can save time in-session.
- **Graph:** Ah, the Graph. You can make literally any audio signal process chain you can imagine in the 3d DSP graph, as exemplified by the Graph-type macros. The 3d Graph makes time-compensated signal path loops possible that can not be accomplished in any other digital or analog audio environment. Users of Max and Bidule will feel right at home.

  See the Graphs section for more details.

I/O

'I/O' inserts are not plug-ins per se, but use a special mechanism to provide a direct Input/Output loop at the selected Insert point within a mixer strip. I/O inserts are not DSP processors and are not currently implemented within a Graph context.

'I/O' inserts can address any hardware audio port whether analog or digital, including SCP USB and Host computer sends/returns.

'I/O' insert "Send" routes are active from the moment they are routed. "Bypassing" an 'I/O' insert defeats the "Return" path, but the "Send" is always sending. This lets you use 'I/O' inserts as extra Direct Outs, with an optional switchable alternate input path in the strip.

Reminder: Since 'I/O' inserts route to physical ports outside the 3d mixer, they can not be automagically latency-compensated like plug-in processor Inserts. All physical routes outside and back into the Mixer will incur some form of latency - whether that bit of delay is acceptable will depend on the individual use case.

Plug-in UIs

To open the UI for a plug-in, click on the "…" section of the Mixer insert.

Most of the small, single purpose plug-ins have basic, generic control sets - one that is automatically created from the parameters in the plug-in. The MIO Delay plug-ins, for example, use this generic interface:

Figure 13.10: MIO Delay (15k) Plug-in UI

The more complex 3d processors such as the 3d Production and Core processors, use custom interfaces generated by us with layout and special UI elements specific to the requirements of each processor.

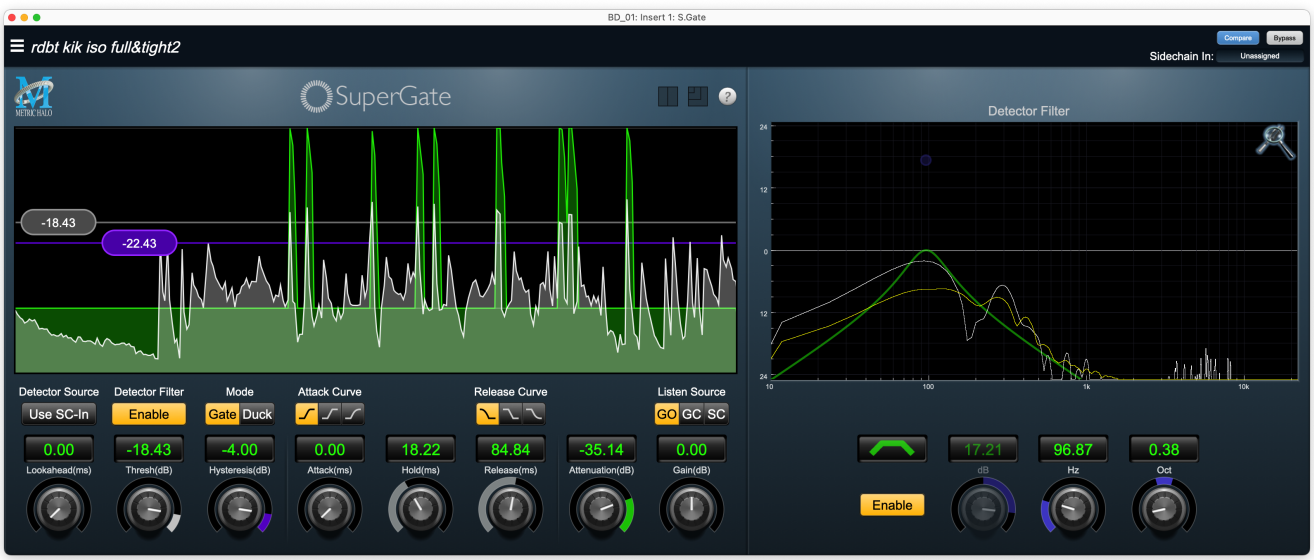


Figure 13.11: MH SuperGate UI

Plug-in Headers

All of the plug-in UI's share the plug-in header at the top of the window. This header provides generic services for managing the state of any plug-in.

The plug-in window header includes a hamburger menu, Preset name/selection, "Compare" and "Bypass".

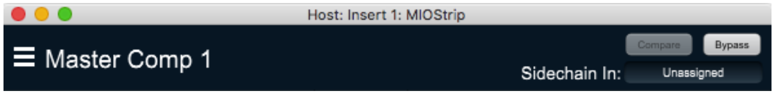


Figure 13.12: Plug-in Header

Plug-ins with integrated side-chain inputs also include a "Sidechain In:" selector. Sidechain inputs may be routed from any input or bus available on the system, except for the Main bus.

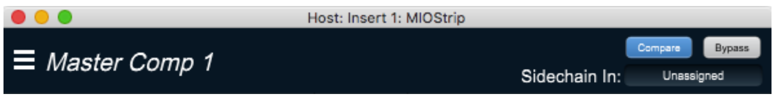


Figure 13.13: Plug-in Header (indicating edited parameters)

When you make changes to any plug-in parameters, the plug-in preset selector title will italicize, and the "Compare" button will activate. Toggling "Compare" switches between the saved preset settings (named in italics) and the changes you've made.

- The **Plug-in Hamburger menu** breaks down as follows:

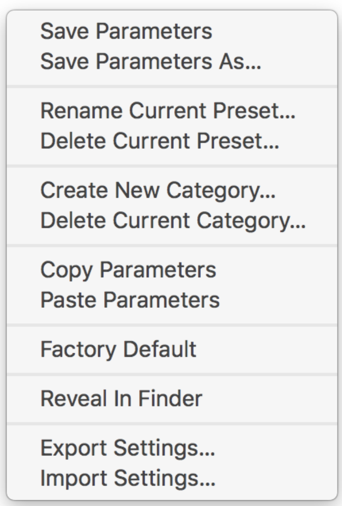


Figure 13.14: Plug-in Header: Hamburger menu

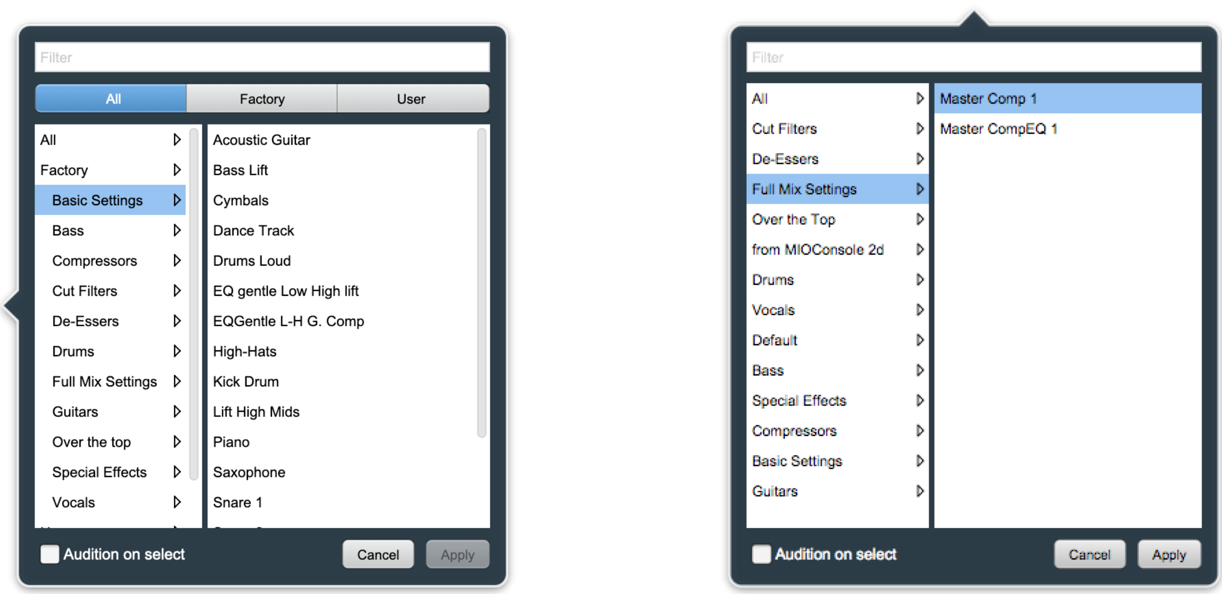
- **Save Parameters** writes the current plug-in parameters to the current preset.
- **Save Parameters As...** opens a dialog box where you can name and choose a category for your current plug-in settings.
- **Rename Current Preset...** lets you rename the current preset.
- **Delete Current Preset...** deletes the current preset.
- **Create New Category...** lets you create a new preset category for the current plug-in type.
- **Delete Current Category...** deletes the current preset category.

- **Copy Parameters** copies the current parameter set so you can paster them to another instance of the same type plug-in.
- **Paste Parameters** pastes the copied parameters. Note that pasting a parameter set over an existing named preset will change the preset name field to: **[No Preset]**.
- **Factory Default** loads the factory default settings for this plug-in.
- **Reveal In Finder** opens the folder in which the current preset is saved. Usually this will be your ~User/Library/Preferences/MIOPlugInPresets folder, but there are no limitations.
- **Export Settings...** is like "Save Parameters As...", but lets you save the preset to any location on any available storage.
- **Import Settings...** will import any Metric Halo 2d or 3d preset file of the same type as the current plug-in.

  Shortcut: Drag any Metric Halo preset file into the desired Insert slot in the 3d Mixer. Remember to "Save Parameters As...", to keep the import available in your preset library.

  The parameter library is automatically shared amongst all instances of a particular plug-in type, including preset categories. Actually, it is automatically shared amongst all instances of compatible plug-in types, so MIOStrip Mono and MIOStrip Stereo automatically share preset libraries.

- **Preset Name/selector menu:**



Channelstrip MIOStrip

Figure 13.15: Plug-in Header: Preset selector menus (ChannelStrip versus MIOStrip)

The Preset selector will open to show all the available preset categories, and the presets within those categories.

Note that the Production plug-ins which share presets with the AU/VST/AAX *MH Production Bundle* plugs also include "All", "Factory" and "User" preset category headers.

Figure 13.16: Preset selector menu: Audition on select

With "**Audition on select**" enabled at the bottom of the window, selecting a preset will immediately load those parameters so you can hear the effect on the audio you are playing, without actually committing the preset to the Mixer strip.

Click **Cancel** to revert to your previous settings and close the selector window.

Hit **Apply** to commit the new preset parameters and close the preset selector window.

- **Sidechain:** For plug-ins that have a sidechain, there is a pop-up that allows you to select any input or bus as a sidechain source.

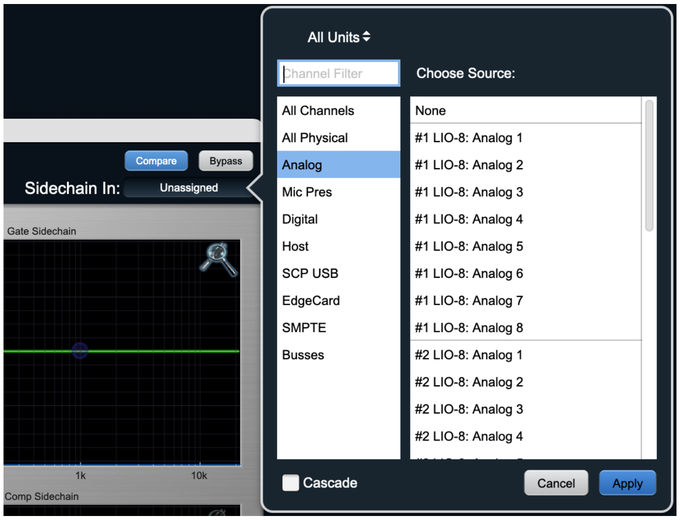


Figure 13.17: Plug-in Header: Sidechain Assign (ChannelStrip)

- Toggling **Compare** switches between the saved preset settings and the changes you've made since instantiating the plug-in.
- The **Bypass** button in the header bar is a master bypass for all processing in the plug-in.

Multichannel Plug-ins

Many plug-ins come in two versions; m/m (mono in – mono out) and s/s (stereo in – stereo out). MIOConsole3d's inserts are context aware; you will always have the m/m option, and will have the s/s option on stereo tracks. If you need to insert a plug-in on a multichannel (LCR to 7.1) input or bus, simply insert the m/m version and MIOConsole3d will instantiate the plug-in on every channel of the input or bus. The UIs will be linked; adjustments you make will be applied to all channels of the plug-in. If you wish to adjust the plug-in on a per-channel basis, you can:

- Use mono input channels instead of a multichannel source
- Insert a graph into the multichannel source or bus, and instantiate multiple plug-ins there

> **What's the difference between using a m/m vs. s/s plug-in on a stereo channel?**
> When you insert an m/m plug-in on a stereo channel, you are creating two separate mono signal paths. For example, inserting an m/m compressor on a stereo input will create two plug-ins with each detector fed from a single channel. The two channels will be processed independently. A stereo compressor would feed the detector from both channels and process them together.

Graphs

A graph is a freeform area in which you can create your own DSP processing chains.

When you insert a Graph in the mixer, a graph UI is automatically generated with input and output ports to match the number of channels of the strip that it is inserted into. The default state of an inserted graph is for the inputs to be connected directly to the outputs.

There is no routing or processing delay within a plug-in graph. So if you make mults with different plug-in paths on each side of the mult, the two paths will remain time-aligned. This allows you to configure parallel processing paths without the annoying (and virtually impossible) task of manually time-aligning the parallel paths.

When you open the graph UI for the insert, you can insert any set of plug-ins into the graph that is shown in the graph UI window. These plug-ins can be connected by virtual cables, their UIs opened, and parameters set. The graph I/O connections will automatically be routed to the appropriate points in the strip that hosts the graph. The graph will be saved and recalled with the rest of the mixer. You can also choose to save the graph independently as type of a "preset" that can be inserted again and again into the mixer.

To open an empty graph, double-click the Graph insert type from the insert menu:

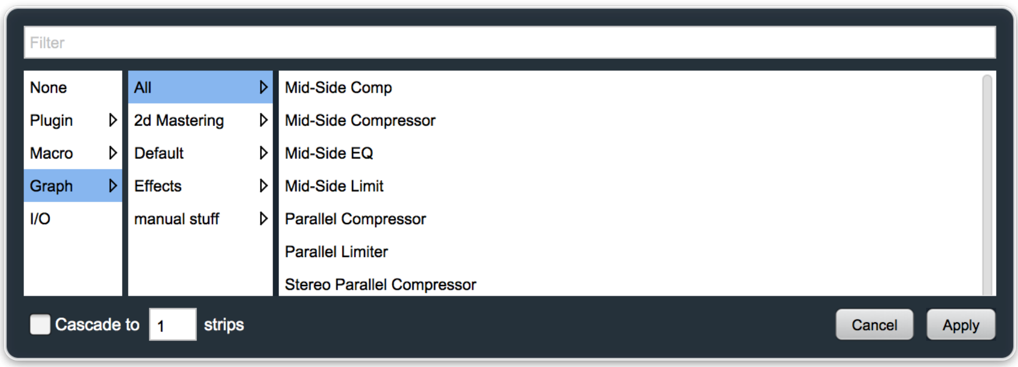


Figure 13.18: Selecting the Graph

and you will see this window:

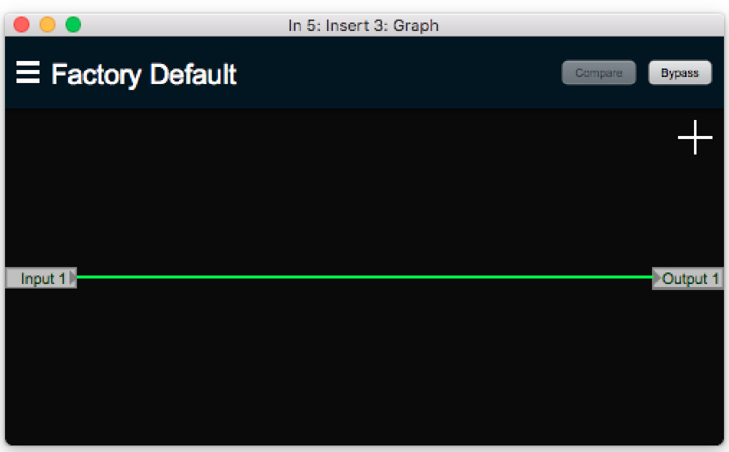


Figure 13.19: A default mono Graph window

To the left are the inputs for the graph, and to the right are the outputs. As noted above, this newly instantiated graph has the input and output connected, so that it will not interrupt the signal. The I/O will match the number of channels in the mixer strip the graph is inserted in; a mono strip will have one input on the left and one out on the right, a stereo strip will have two channels of I/O, a 5.1 strip will have six channels in and out, etc.

As you can see, the Graph has the same plug-in window header, hamburger menu and preset menu as every other plug-in. In this regard, Graph preset categories and save/recall is no different than any other plug-in.

Right-click any empty space, or hit the big "+" in the upper right corner. The Plug-in pop-up menu contains all of the available instantiable graph plug-ins. When you select a plug-in from this menu, a new instance is created in the Graph, and you may drag the instance to a convenient location in the Graph workspace.

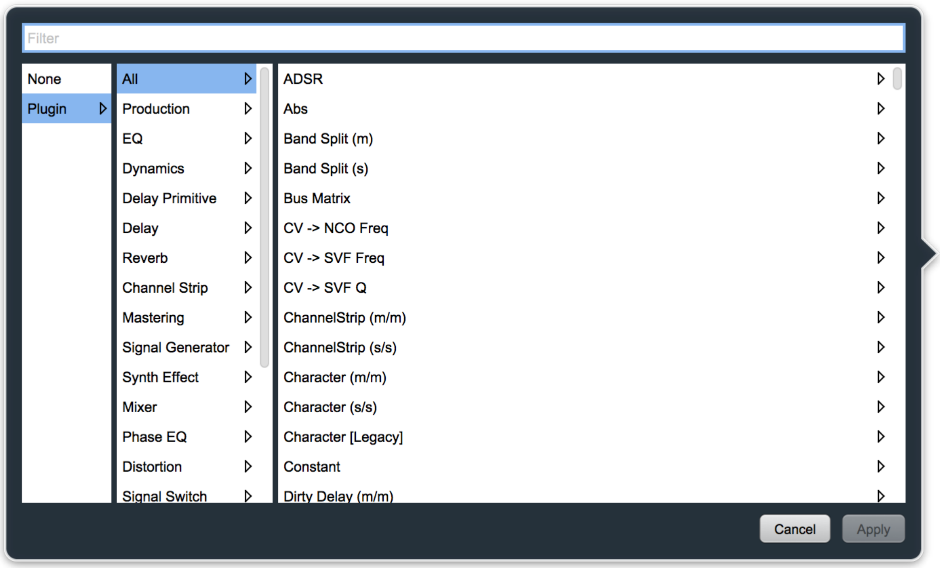


Figure 13.20: Selecting a new plug-in instance from the Plug-in Menu

Note that, yes indeed, you can insert full-blown processors like Channelstrip, TransientControl, Sonic EQ 3d, MIOStrip and Haloverb in the Graph without restriction, right alongside CV control and mathematical function instruments. It truly is a playground.

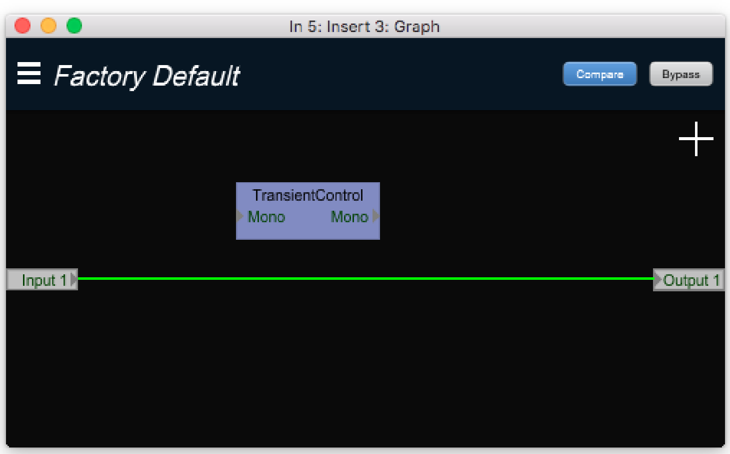


Figure 13.21: Positioning the new plug-in instance in the Graph

Once you have added the Plug-ins you want to use, you can wire them up. To make connections, click on a port (one of the small gray triangles next to the port name), and then drag the connection to the target. When you have made a valid connection, the connection line will switch from Gray to Green. You can make as many mults as you like of a signal source but only one connection can be made to a processor input or process bus port. If you make a new connection to an input that already has a connection, the old connection will be automatically disconnected. To remove a connection without establishing a new one, ^ (Control)-click on the input port to which the connection is made.

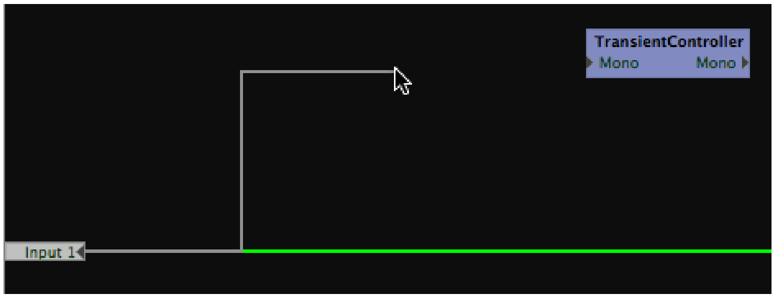


Figure 13.22: Starting a Connection

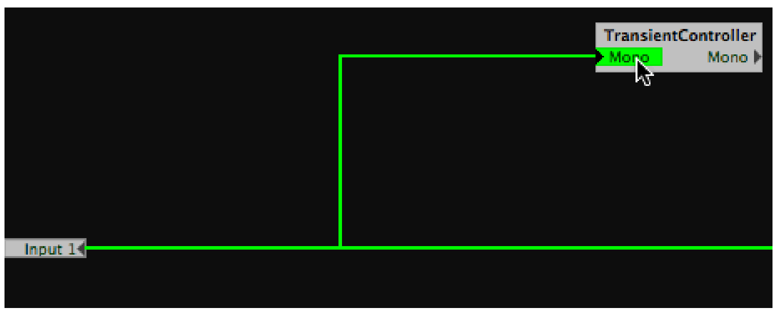


Figure 13.23: Completing the Connection

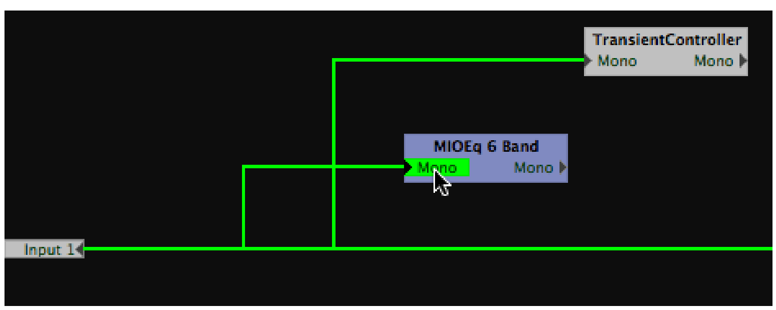


Figure 13.24: Making a Mult

After everything is placed and wired up, you will need to ensure that you have routed the output of your signal processors to the appropriate port on the output side of the graph. When you are done, you will have a complete graph. For example:

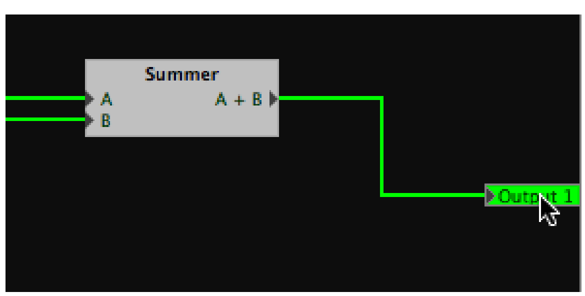


Figure 13.25: Connecting the Output

As you can see, once the output of the final plug-in was connected to the output port, the input-to-output connection was removed, and you are now listening to the processing chain.

Sidechain Inputs When sidechain-capable plug-ins are used in a Graph, the sidechain input is exposed as a routing point

Figure 13.26: Sidechain routing in the Graph

…and takes the place of the "Sidechain Assign" header element in the plug-in UI.

The graph is continuously modifiable with or without audio running. You can drag the plug-ins around as you like, and you can add new plug-ins, even while you are processing audio with the existing graph. You can make and break connections as you please.

Remember: **^-click** *(Control)-click on the input port to remove a connection* or hit Bypass to remove the graph from the signal path.

At this point, you will want to be able to control each plug-in's parameters. Double-click on any plug-in to view its UI as described previously.

See The DSP Toolchest for the full breakdown of every available Graph instrument.

Graph special commands

The Graph window supports a special set of key commands to help speed navigation within the Graph itself. These commands are separate from menu bar *Edit: Edit Key Commands* assignment, and apply only when the Graph UI window is active and in focus:

Command	Key
Delete selected blocks	delete
Move selected blocks up	up arrow
Move selected blocks down	down arrow
Move selected blocks left	left arrow
Move selected blocks right	right arrow
The following keys work for both upper and lower case:	
Select all blocks in graph window	A
Bypass/Unbypass selected blocks (all blocks will be the same after this command)	B
Toggle bypass on selected blocks (this toggles the bypass on each block individually)	T
Deselect all blocks in graph window	D
Reverse I/O direction of selected block	F
Redraw connections in straight lines (un-snag)	R

Metric Halo 3d Production Processors

Figure 13.27: The Metric Halo Production Bundle

3d DSP meets AU/VST/AAX (DAW software)

Naturally, all of Metric Halo's MIO 2d DSP processors have been updated to run natively in the 3d DSP engine, including MIOComp, MIOLimit, MIOEQ 6 & MIOEQ 12. All 2d Graph building blocks have also carried over to the new platform, ensuring even your most complex custom processing creations will smoothly transition over to 3d.

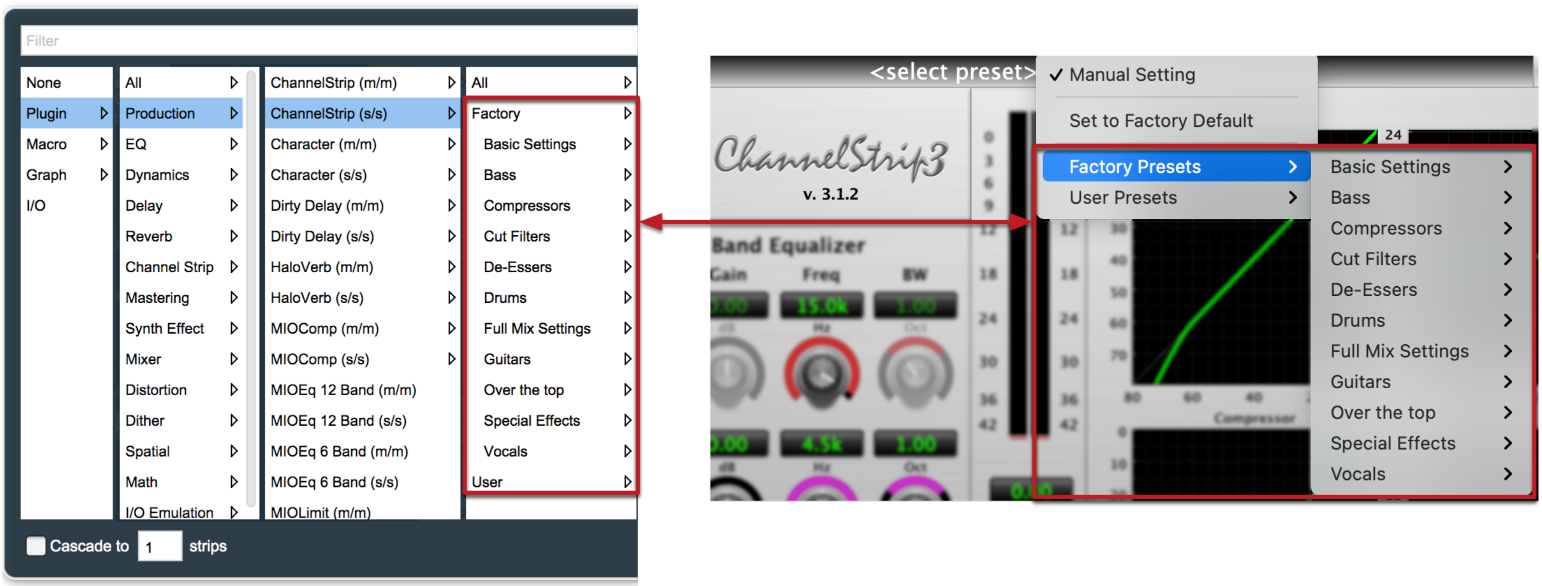
Equally naturally, Metric Halo's world-renowned AU/VST/AAX MH Production Bundle has found its way into the 3d hardware mixer environment as well.

The Metric Halo 3d Production processors run in the 3d hardware mix desk and are functionally identical to the AU/VST/AAX *MH Production Bundle* plug-ins running in your DAW host.

The preset libraries for each plug-in are shared between the 3d hardware and AU/VST/AAX software versions, so any parameter presets you save in either environment are accessible to the other.

Sharing presets between AU/VST/AAX and 3d DSP Production Bundle plug-ins is automatic as long as you use the *Plugin Presets Selector* popup menu (shown below).

Metric Halo AU/VST/AAX Plugin Presets Selector



ChannelStrip Factory presets
(MIOConsole3d Mixer Inserts selector)

ChannelStrip Factory presets
(AU/VST/AAX)

Figure 13.28: AU/VST/AAX and 3d DSP preset menus

Factory and User preset libraries are reflected in both plugin preset managers …and yes, any presets you have made over the years of using any of the MH Production Bundle plug-ins are immediately available to their 3d counterparts.

Please note that presets saved from within the DAW plugin headers are stored elsewhere in a proprietary format to each respective DAW, and will not be shared.

All 3d Production processors may also be used as part of larger processor chains within a 3d DSP Graph context.

Each 3d Production Bundle plug-in has been ported with the same familiar user interface, including the "Help" button toggle, re-sizable plug-in window, and (where applicable) show/hide control of the Spectrafoo™ graphics displays.

- **Plug-In Headers**

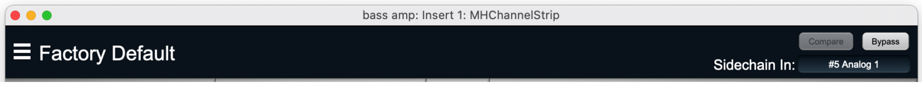


Figure 13.29: 3d plug-in header

All 3d insert and Graph processors share the 3d plug-in header, housing the hamburger menu controls, preset management, Compare and Bypass. The *ChannelStrip* header also includes sidechain input routing.

See the 3d plug-in header section for full details.

- **Help Button**

Figure 13.30: Help Button

This button toggles the tooltip display. When enabled, tooltips will be shown when the mouse hovers over a control. When the tooltip display is disabled, you may still see tooltips by holding down the **?** key and hovering over a control.

- **UI Size Selector**

Figure 13.31: UI Size Selector

This button switches the plugin user interface between small, medium and large sizes to accomodate different display resolutions.

- **Graph Visible Selector**

Figure 13.32: Graphs Closed

Figure 13.33: Graphs Open

This button toggles visibility of various response graphs. Not all Production Bundle plug-ins will have this button. This button allows you to maximize screen real-estate while still providing details on the processing when they are needed. Click on this control to toggle the visibility of the graphs. The plug-in window will automatically become smaller when you hide the graphs.

Character

Plug-in categories: **Production**, **Distortion** and **I/O Emulation**

Introduction

Character is an audio circuit modeling plug-in for MIOConsole3d. We measured a number of our customers' favorite analog signal processors to determine what they did to the audio signal being run through them. By measuring the harmonic distortion fingerprint of each device, we have been able to recreate their unique properties in Character.

Figure 13.34: Character's User Interface

By combining circuit modeling with variable gain, Character is capable of shaping your sound with subtle or over-the-top results.

There are several ways to use Character:

- Inserted on a single channel: This lets you add a little "flavor" to a channel by itself or as a pre-processor for another plug-in. By automating the Drive and Output Gain controls you can use Character as a creative effect.
- Inserted on all channels: Create a virtual mixing console, or use different models for different groups of inputs; mic pres for vocals, DIs for bass, tubes for drums…
- Inserted on the master bus: Put a Soft Sat model on the master bus for the analog "glue" to finish off your mix.
- All of the above!

Operation

The Character user interface uses a few different control elements to control its processing. These elements are:

Character Model Menu

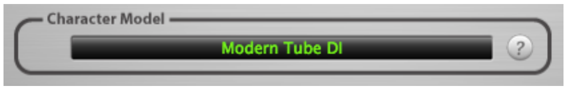


Figure 13.35: Character Model Menu

This pop-up menu selects Character's model. There are over 20 models available, which are detailed in the Processing chapter.

Control Knob

Figure 13.36: Control Knob

Control tips

You can change the value of each knob in a number of different ways.

- Click and drag the knob to change the value continuously.
- Dragging up or to the right will increase the value, while dragging down or to the left will decrease the value.

  Hint: If you hold down the ⌘ (Command) key when you click, you will be able to adjust the value with finer precision.
- ⌥ (Option)-click to reset any knob to its default value. You may also double-click a knob to reset it.
- **Rapid parameter entry**
  - Double-click the numeric readout of the knob to type in a number directly. The text-entry field will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, or **⌘.** (Command + .) keys.
  - Hit **return** or **enter** to confirm the value and dismiss the pop-up.
  - Hit the **tab** key to confirm the value and immediately activate the entry field for the next control. **⇧-tab** (Shift + tab) will display the entry field for the previous control).
  - Hit **⌘.** (Command + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.

Toggle Button

Figure 13.37: Toggle Button (On)

Toggle buttons are simple on/off switches. They light up when they are on and are dark when they are off. Toggle the state of the button by clicking on it. This button is used to enable Auto Drive within Character.

Output Meter

Figure 13.38: Output Meter

For the main output stage of Character we have provided meters driven with SpectraFoo metering technology. These meters show, in addition to the peak metering provided for the input stages, RMS level and VU level. The peak level is represented by the floating colored bar, the RMS level by the solid colored bar and the VU level by the overlaid gray bar. Both the Peak and RMS level are represented with fast PPM ballistics. The VU meter shows IEEE standard 300 ms RMS average level. When Character is on a mono insert there will be a single meter. When Character is running in stereo mode the top meter shows the left channel output level and the bottom meter shows the right channel output level. The output section clip lights activate if there is an over in the output stage or in any of the processing section input stages. It is reset by clicking on the meter; Mac ⌥ (Option)–click or Windows **Alt**-click to reset the clip lights on all the meters.

> **A Note About Clipping Indicators:**
> The clip lights do not mean that the plug-in is clipping; it means that the audio level in the DSP is currently over 0 dBFS. If you do not lower the signal level you run a chance of actually clipping the input of another processor or D/A convertor.

Processing

A Detailed Description

In this chapter we discuss what each parameter does and how the controls work. While Character has a fixed number of model types, there are an infinite number of combinations in conjunction with the Drive control.

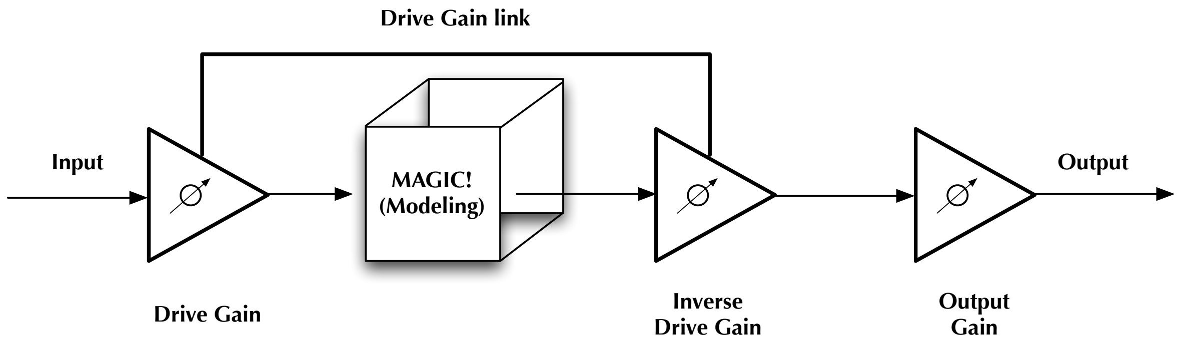


Figure 13.39: Character Block Diagram

The block diagram above illustrates the overall structure of the processing system provided by Character. This diagram does not indicate the metering blocks.

Now lets examine the various processing blocks indicated in the diagram.

Drive

The Drive control applies ±24 dB of gain to the signal before it goes to the modeling section. This allows you attenuate the signal for a subtle effect, or boost to get a more dramatic effect.

As gain is changed before the modeling stage, the inverse gain is applied after. For example, if you set the Drive control to +6 dB, the gain is boosted 6 dB before the model and cut by 6 dB after. This allows you to "push" the modeling section with no increase in output level. You may also hit the model with less signal by setting the Drive control to a negative value without loss of overall volume.

You may still experience in increase in signal level when using positive Drive gain with some models; you can use the Output Gain control to correct this.

Please note that the Drive parameter may have a greater range than you need for a given model, particularly when Auto Drive is engaged.

Auto Drive

When creating a new instance of Character, Auto Drive is turned off by default. When engaged, the Auto Drive button enables a detector that will automatically sense lower level input signals and apply more drive gain that varies with the signal level.

The effect of enabling Auto Drive is to have a more consistent amount of distortion applied for all input levels. As a result, the effect of the various circuit models becomes more pronounced with Auto Drive engaged. In some instances can lead to heavy distortion. Turn the drive knob to the left to compensate, or leave Auto Drive off for a more subtle effect.

Character Model

Each Model represents a digital "copy" of an analog device; some models are made from measuring a device with different combinations of settings. The most effective way to utilize Character is to listen to how each model affects your sound rather than relying on the name alone.

- None: No modeling is applied.
- Transformer: Applies the harmonic distortion signature of a transformer-coupled input.
- Valve: A tube-based EQ input stage.
- FET: Model of a solid state (transistor) front end.
- Soft Sat: Tube-based EQ with saturation.
- Boutique Tube: Hand-made tube mic pre.
- American Transformer 1: A variation of the "Transformer" model.
- American Transformer 2: Second variation of the "Transformer" model.
- California Tube Mic: American designed tube mic pre.
- California Tube Line: American designed tube line input.
- Modern Tube DI: Mastering quality tube DI.
- Modern Tube EQ: Mastering quality EQ.
- Modern Tube Soft Sat: Mastering quality EQ with saturation.
- Modern Tube LG: A tube mic pre with a low gain setting.
- Modern Tube MG: A tube mic pre with a medium gain setting.
- Modern Tube HG: A tube mic pre with a high gain setting.
- Modern Tube Sym: Mastering quality EQ
- Modern Tube Soft Sat: Mastering quality tube mic pre with saturation.
- Classic British Mic Pre: A favorite large console mic pre.
- American Solid State: FET mastering EQ.
- California Vocal Box: Transformer coupled tube vocal processor.
- California Vocal Box Drive: Transformer coupled tube vocal processor with increased gain.
- British Mic Pre Clone: A popular clone of a favorite British mic pre.

The "soft sat" variations are particularly good at providing the "analog glue" that many engineers want for their final mixes. The "sat" stands for "saturation", such as you would get with analog tape. The "soft" part of the name indicates that the level is *lower* after processing. You may want to use the Output Gain to make up the lost gain.

While most of the models can be applied to every channel, overuse of the saturation models can lead to undesired effects.

Output Gain

The Output Gain applies ±24 dB of gain to the signal after it has been through the modeling stage. This can be used to increase the final output level after using a soft saturation model or otherwise gain-stage the signal for the next processor.

ChannelStrip

Plug-in categories: **Production**, **EQ**, **Dynamics** and **Channel Strip**

Introduction

ChannelStrip is a MIOConsole3d plug-in which provides the essential basic channel processing found in the channel strip of a modern mixing console.

Processing functions include:

- Input level control
- Phase Invert
- Expander/Gate with filtered sidechain
- Compressor with filtered sidechain
- Fully interpolated 6 band Parametric EQ with scalable display
- Channel delay
- Limiter
- Advanced metering
- Sidechain input source from any input or bus
- Sidechain listen
- Scalable gain reduction meters
- Selectable *Smooth*, *Warm*, *Fast* and *MIO* compressor characters
- High precision SpectraFoo™ spectrum analysis
- Processes are automatically enabled when adjusted

Figure 13.40: ChannelStrip's User Interface

Operation

As with most channel strips, ChannelStrip provides many copies of controls that are all operated in a similar manner. The ChannelStrip user interface uses a few different control elements to control all of the processing. These elements are:

Control Knob

Control Knobs are used to control the value of various continuous parameters of a process. Examples of these types of parameters include: Attack time, Release Time, Threshold, etc. There are four styles of encoders:

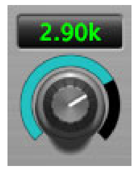


Figure 13.41: Swept Knob

The rings around these encoders sweep from a minimum to maximum value, normally from left to right. One exception is the compressor threshold, which sweeps from right to left.

Figure 13.42: Plus/Minus Knob

The rings around these encoders start at 12 o'clock and sweep to either side. These knobs are used for gain control, where straight up is no gain change, turning to the left cuts the signal and turning to the right boosts it.

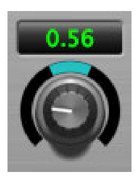


Figure 13.43: Spread Knob

The rings around these encoders start at 12 o'clock and spread to both sides equally as the control is increased. These knobs are used for bandwidth controls.

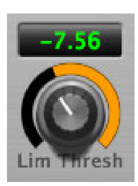


Figure 13.44: Limiter Knob

The ring around this encoder displays the amount of gain reduction from the limiter. The display sweeps around the encoder from right to left, with a fully left display indicating 12dB of gain reduction.

Control tips

You can change the value of each knob in a number of different ways.

- Click and drag the knob to change the value continuously.
- Dragging up or to the right will increase the value, while dragging down or to the left will decrease the value.

  Hint: If you hold down the ⌘ (Command) key when you click, you will be able to adjust the value with finer precision.
- ⌥ (Option)-click to reset any knob to its default value. You may also double-click a knob to reset it.
- **Rapid parameter entry**
  - Double-click the numeric readout of the knob to type in a number directly. The text-entry field will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, or ⌘. (Command + .) keys.
  - Hit **return** or **enter** to confirm the value and dismiss the pop-up.
  - Hit the **tab** key to confirm the value and immediately activate the entry field for the next control. **⇧-tab** (Shift + tab) will display the entry field for the previous control).
  - Hit ⌘. (Command + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.
  - When you enter a number into the pop-up entry, you can use a couple of abbreviations: "k" multiplies the number by 1000 and "m" divides the number by 1000. So if you want to enter 16,500 Hz you can just type 16.5k.

Toggle Button

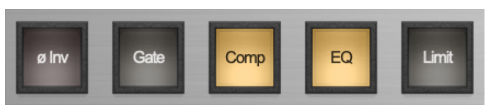


Figure 13.45: Toggle Buttons

Toggle buttons are simple on/off switches. They light up when they are on and are dark when they are off. You toggle the state of the button by clicking on it. These buttons are each labeled by the function they control. The ChannelStrip Master Enables are shown above with the Comp and EQ processors enabled.

Fader

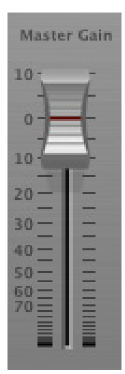


Figure 13.46: Master Fader

The fader is unique in that only one fader is used in the interface for ChannelStrip. It works in much the same fashion as the control knobs. Instead of dragging up/right or down/left to change the value, you directly drag the fader knob. The other "tricks" described for the knobs also work with the fader. The fader is used to control the master output gain of the plug-in before the limiter stage.

Filter Type

Each filter band in the strip (6 EQ bands and 2 Side-chain bands) has a filter type control that allows you to choose the shape of the filter applied by that band. Each band provides 6 different types of filter shapes:

- **Peaking/Parametric** – a second order bell-shaped parametric boost/cut filter. The Gain control has a boost/cut range of ± 24 dB. When the boost is greater than +15 dB the filter gains a resonant quality. The center frequency of the filter can be any frequency between 20 Hz and 20 kHz. The bandwidth of the filter is continuously variable between 0.1 octaves and 2.5 octaves.

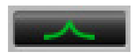


Figure 13.47: Peaking/Parametric

- **Low Shelf** – a shelving filter that applies boost/cut to low frequencies. Boost/cut has a range of ± 24 dB. The bandwidth controls the dip/peak that is added at the end of the transition band.

Figure 13.48: Low Shelf

- **High Shelf** – a shelving filter that applies boost/cut to high frequencies. Boost is limited to +12 dB (reflected in the EQ Transfer Function display) whereas cut goes to –24dB. The bandwidth controls the dip/peak that is added at the end of the transition band.

Figure 13.49: High Shelf

- **High Cut** – a 12 dB/octave high cut filter with a -3dB point that is continuously adjustable between 20 Hz and 20 kHz.

Figure 13.50: High Cut

- **Low Cut** – a 12 dB/octave low cut filter with a -3dB point that is continuously adjustable between 20 Hz and 20 kHz.

Figure 13.51: Low Cut

- **Bandpass** – a bandpass filter with 6dB per octave skirt on the high and low ends of the pass band. The width of the pass band can be adjusted between 0.1 octaves and 2.5 octaves and the center of the pass band is continuously adjustable between 20 Hz and 20 kHz.

Figure 13.52: Bandpass

You can select from these types via three different methods. Each time you click on the Filter Type control, the band will switch to the next type in the list (and wrap to the beginning when you hit the end of the list). If you click and hold the mouse button, a pop-up menu listing all of the types will appear after about 1/4 of a second. You can select the type directly from this pop-up menu. If you want to access the menu without having to wait, hold down the ^ (Control) key when you click or right-click.

Sidechain Input selector

Figure 13.53: "Sidechain In" selector window

ChannelStrip can use any input or bus to feed the gate and/or compressor detector sidechain. The "Sidechain In" selector is located at the upper right of the UI header. Click to open the input routing selection window.

Sidechain Routing Switch

Figure 13.54: Sidechain Routing Switch

Each sidechain routing switch allows you to control the signal sent to the sidechain input of its associated gate or compressor. By default, the level detectors in the dynamics processors key off of the signal that they are processing. Under some circumstances, you may want to use a different signal to open the gate or compress the signal. Most DAWs allow you to specify an input or bus as the source for ChannelStrip's sidechain input. The Sidechain routing switches allow you to choose the input to the level detector from the sidechain (key) input or the signal being processed. To toggle the state, click the Sidechain routing switch.

Sidechain Listen Button

Figure 13.55: Sidechain Listen Button (Disabled)

Figure 13.56: Sidechain Listen Button (Enabled)

The sidechain listen button allows you to listen to the signal being sent to the gate or compressor sidechain input. This allows you to monitor the audio being routed from the key input, and hear the effect of sidechain filtering.

Compressor Character

Figure 13.57: Compressor Character Switch

The compressor character controls the time constants of the compressor section. It functions identically to the Filter Type control, except there are only four choices: Smooth, Warm, Fast and MIO. See the section on the compressor for more information.

ChannelStrip also uses a number of standard visual representations to give you feedback about what is happening within the processor. These elements are:

Peak Meter

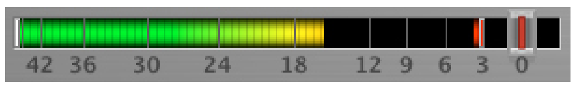


Figure 13.58: Peak Meter

ChannelStrip provides a peak-reading meter at the input stage of each processing block. The meter uses the fast PPM standard for decay time (0.9 seconds per 20 dB) and the digital PPM standard legend for calibration. On the dynamics sections (gate and compressor) a white bar is visible on top of the meter and indicates the current detector level. For the dynamics sections the processor threshold is indicated by the red slider over the input peak meter. This slider can be manipulated directly with the mouse. The top segment of the meter (above 0dB) is used as a clip indicator and is illuminated red if the input section of the processor detects an over. The clip light remains illuminated until you click on the meter. ⌥ (Option)–click any meter to reset the clip lights on all of the meters in ChannelStrip. When ChannelStrip is running in stereo mode, this meter shows the higher of the two input levels and will detect an over on either input channel.

Gain Reduction Meter

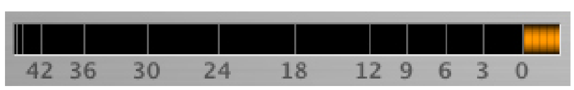


Figure 13.59: Gain Reduction Meter

The gain reduction meter, which has an orange bar and grows down from 0 dB, shows the amount of attenuation being applied by its associated dynamics processor at any given time. If you right-click or ^ (Control) click on the meter, you may set the scale of the gain reduction meter to any of the following values:

- 54 dB
- 24 db
- 12 db
- 6 db
- 3 db

Peak, RMS, VU Output Meter

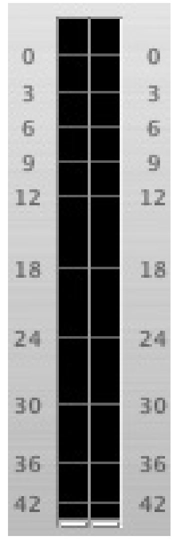


Figure 13.60: Output Meter

For the main output stage of ChannelStrip we have provided meters driven with SpectraFoo™ metering technology. These meters show, in addition to the peak metering provided for the input stages, RMS level and VU level. The peak level is represented by the floating colored bar, the RMS level by the solid colored bar and the VU level by the overlaid gray bar. Both the Peak and RMS level are represented with fast PPM ballistics. The VU meter shows IEEE standard 300 ms RMS average level. When ChannelStrip is on a mono insert there will be a single meter. When ChannelStrip is running in stereo mode the left meter shows the left channel output level and the right meter shows the right channel output level. The output section clip lights activate if there is an over in the output stage or in any of the processing section input stages. It is reset by clicking on the meter; Mac ⌥ (Option)–click to reset the clip lights on all the meters.

> **A Note About Clipping Indicators:**
> The clip lights do not mean that the plug-in is clipping; it means that the audio level in the DSP is currently over 0 dBFS. If you do not lower the signal level you run a chance of actually clipping the input of another processor or D/A convertor.

EQ Transfer Function

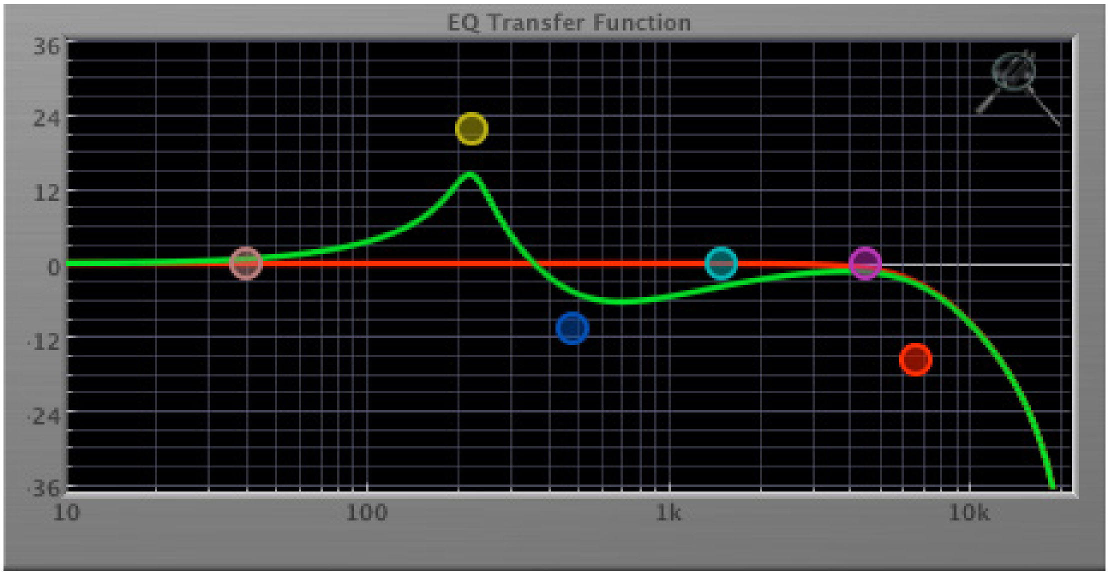


Figure 13.61: EQ Transfer Function

The following information applies to the EQ processing block as well as the sidechain filters of the gate and compressor.

The EQ transfer function is a combination of a visual representation of how the EQ is processing the signal and an intuitive controller for the associated filter bands. This display is sometimes called a "Cartesian Graph" by other EQ manufacturers.

The horizontal axis provides frequency calibration in Hertz (Hz). The vertical axis provides level calibration in decibels (dBr). The heavy green line indicates the relative change in level at each frequency that is created by the combined effects of all of the active bands in the equalizer. Each EQ band is represented by a colored dot in the transfer function. The color of the dot matches the color of the rings around the knobs for the corresponding EQ band.

The band that is currently being edited will be displayed along with the overall response curve. If the associated band is a parametric filter there will also be two smaller colored dots that can be used to control the bandwidth of the filter. Clicking on a large colored dot and dragging will allow you to adjust the frequency and gain of the associated band. ⌘ (Command)–click or double-click the dot to toggle the band enable. ⌥ (Option)–click the dot to adjust the bandwidth (dragging right increases the bandwidth, left decreases the bandwidth). ⌘⌥ (Command + Option)–click the dot to switch the band filter type. Click and drag the smaller dots associated with a larger dot to adjust the filter bandwidth.

To dismiss the filter curve, click anywhere in the black area of the transfer function. This will deselect the filter point, and the only trace displayed will be the green master curve.

If you right-click or ^ (Control) click on the transfer function, you will see a menu to set the vertical dB scale for the display. The values are:

- ±3 dB
- ±6 dB
- ±12 dB
- ±24 dB
- ±36 dB

This menu also allows you to specify whether adjusting a filter causes it to automatically be enabled. This preference is for all instances of ChannelStrip.

Spectragraph Analyzer

Clicking the SpectraFoo™ logo in the upper right hand corner of the transfer function will activate the spectragraph, showing the realtime frequency analysis of your signal:

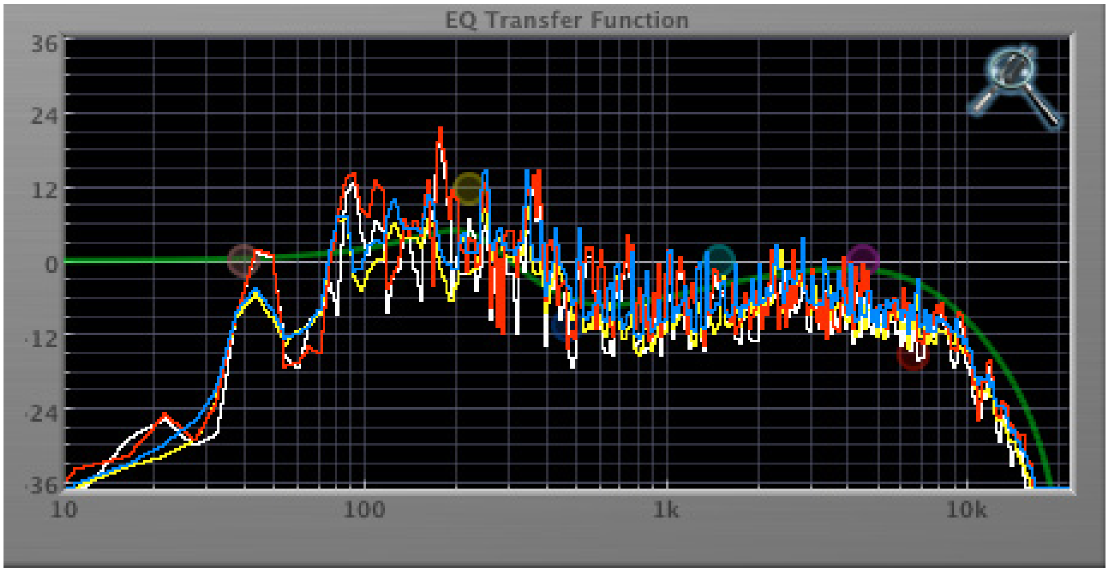


Figure 13.62: Spectragraph Display

The traces are:

- White: Left channel instantaneous display
- Red: Right channel instantaneous display
- Yellow: Left channel average display
- Blue: Right channel average display

The *instantaneous* trace updates in real-time, allowing you to see the immediate peak level of your audio. The *average* trace displays the level as averaged over a short period, giving you a more general view.

The spectragraph analyses the signal post-filter, allowing you to see the effect of your EQ filter(s); the EQ's spectragraph is also after the Master Gain and Limiter. The spectragraph may only be used in one window at a time and will toggle between sections. For example, if your are viewing the spectragraph in the EQ window and then click the the SpectraFoo™ icon in the compressor's sidechain window, the spectragraph will switch to that window. To disable the spectragraph entirely, click the active 'Foo icon.

If you right-click or ^ (Control) click on the transfer function, you will see a menu to set options for the spectragraph:

- Show Instantaneous Trace: Toggles whether the spectragraph shows the instant response of your audio.
- Show Average Trace: Toggles whether the spectragraph shows the averaged response of your audio.
- Show Left Channel: Toggles the left channel spectragraph display.
- Show Right Channel: Toggles the right channel spectragraph display.

These settings are stored for each transfer function window separately, and for each instance of ChannelStrip.

Dynamics Knee

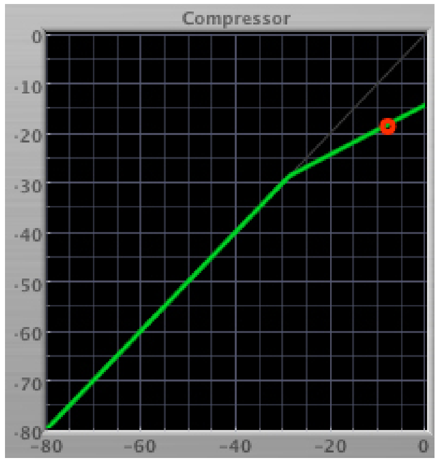


Figure 13.63: Dynamics Knee

ChannelStrip contains a Dynamics Knee diagram for the gate and compressor processing sections. The diagram provides feedback on the response of the associated dynamics processor. Both the horizontal and vertical axes are calibrated in dBFS. The horizontal axis corresponds to the input level and the vertical axis represents the output level. The heavy line shows the dynamic of the associated processing block. This means that if you sent in a sine wave at a given input level, the output level would be equal to the level shown on the graph. When the processor is working with real dynamic signals, the graph is a good approximation of the response when the attack is fast and the release is slow.

In most cases, however, the dynamic response of the processor will not match its static response. In order to represent this, we have included a “bouncing ball” meter for both the gate and the compressor. This metering is shown as a red circle that is overlaid on the knee diagram. The red circle is placed so its horizontal position is equal to the instantaneous input level and its vertical level is equal to the instantaneous output level. Examining this meter while you are adjusting the dynamics controls will provide you with a great deal of information about how the processor is operating and how the controls interact.

Auto Enables

The processing sections of ChannelStrip will automatically enable when one of the EQ Transfer Function parameters are adjusted. For example if the EQ master enable is off, adjusting any EQ parameter will turn it on. This way you will never make “phantom” adjustments where you make adjustments and hear no change. The same is true for the sidechain filters.

Note that the Auto Enable applies only to the EQ band enables, and does not affect the master plug-in Bypass state. Auto Enable does not apply to the 6-Band EQ control knobs.

Processing

A Detailed Description

In this chapter we'll discuss what each processing block does and how the controls work.

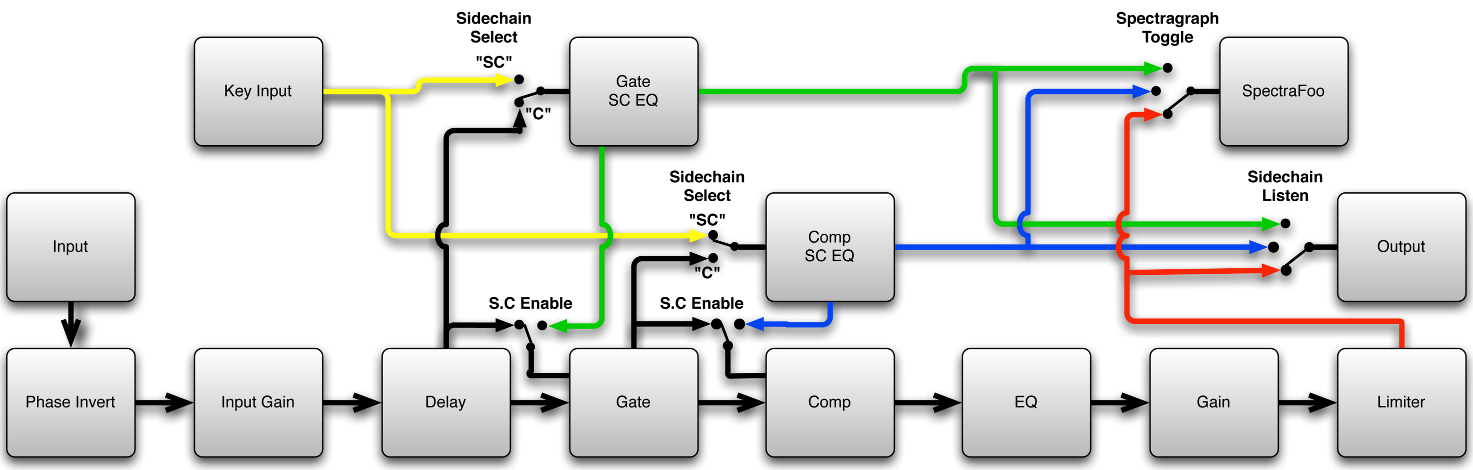


Figure 13.64: Signal Flow: Compressor is Pre-EQ

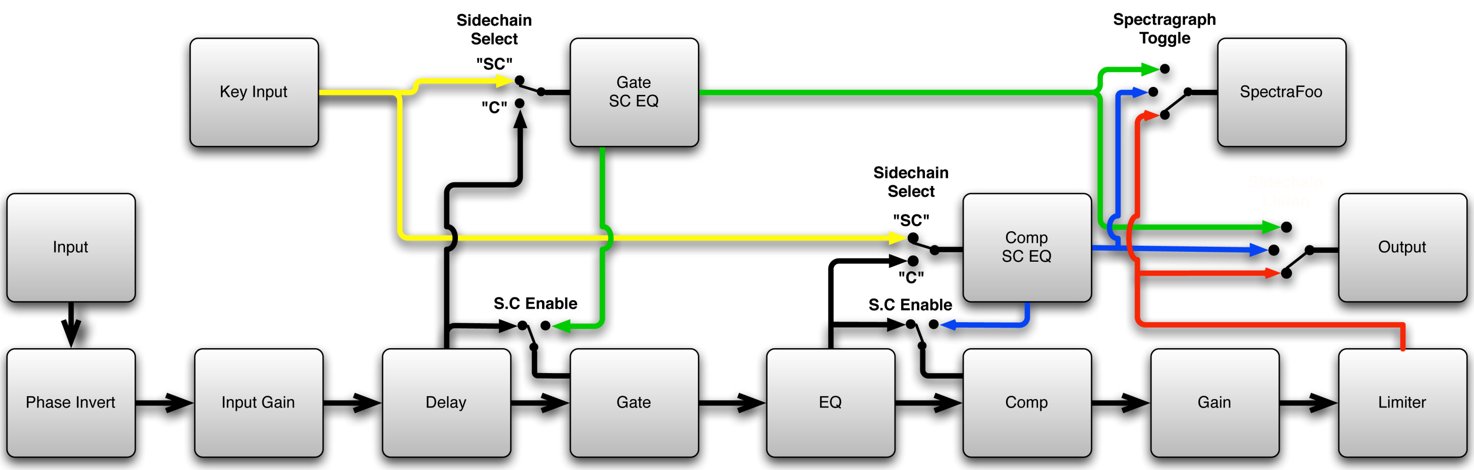


Figure 13.65: Signal Flow: Compressor is Post-EQ

The block diagrams above illustrate the overall structure of the processing system provided by ChannelStrip. The diagrams do not indicate the various metering blocks. The configuration is controlled by the "Post EQ" button in the compressor and it changes the path of the signal through the processor. The internal signal switching provided by ChannelStrip allows you to explore the various effects of re-ordering processing blocks without having to waste time stopping the transport, removing and reinserting plug-ins and then finally restarting the transport.

Now lets examine the various processing blocks indicated in the diagram.

Input Conditioning

After the signal is routed to ChannelStrip it runs through the phase invert block. This block is controlled by the "ø Inv" switch. When the phase invert is enabled the polarity of the signal will be flipped. The signal is cross-faded between the uninverted and inverted states so the signal level will drop briefly when you flip the state of the phase invert switch, but it will not introduce a glitch or click into your audio.

After the the phase invert block the audio is routed through an input gain block that provides input gain of up to +24 dB. You can use this gain to condition signals that are low in level.

This input gain may also be used to pad out signals by up to –24 dB. While you may find this attenuation useful to just bring down the level through the strip simply and quickly, you must realize that this gain is applied after the signal reaches ChannelStrip and will not pad out any clipping that occurs in the A/D converters or in a plug-in that is inserted before ChannelStrip.

The input gain is controlled by the "In Gain" knob.

Gate/Expander

The next processing block is the Gate. The gate is used to adjust the low level dynamics of the signal being processed. Through the use of the external side-chain the gate can be used to do acoustic triggers. In addition, the side chain filter may be used to make the control of the dynamics frequency sensitive. This can be useful when you are trying to gate out a noisy signal that has a specific, very strong signal in a limited frequency range when you want the gate to open.

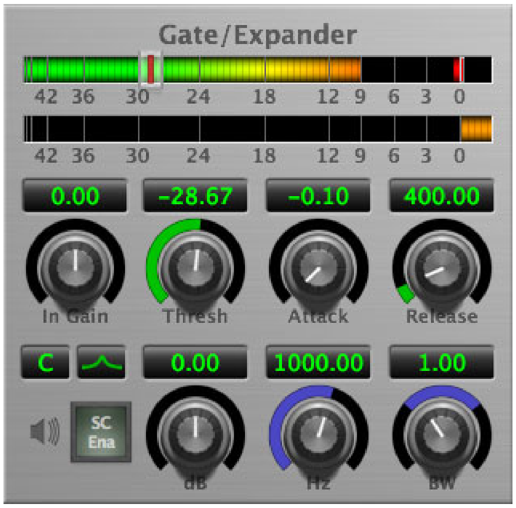


Figure 13.66: Gate/Expander

Theory Of Operation

Based upon your setting for the sidechain routing switch, either the sidechain input signal or the channel signal is fed to the sidechain filter. The sidechain filter provides one band of equalization that may be used to accentuate or cut certain frequencies (parametric or shelf filters) or limit the key to a certain range of frequencies (cut or bandpass filters). You control the filter type and the filter parameters with the filter type button and the "dB", "Hz" and "BW" knobs.

You can enable the side chain filter with the green "SC Ena" button.

After the sidechain signal has been processed by the sidechain filter it is measured by a level detector that determines the instantaneous level of the signal (in the case that ChannelStrip is running in stereo mode the detector is linked with the other channel in the stereo pair and the higher level of the two channels is used).

When the gate is enabled the signal will be attenuated based on how much the detected level is below the threshold you set with the "Thresh" knob.

The dynamic behavior of the opening and closing of the gate is controlled with the "Attack" and "Release" knobs.

Gate Enable

This button is in the Master Enables section. When this button is off, the gate will not change the signal.

Threshold Control

The “Thresh” knob controls the level at which the gate opens and closes. When the detector level is above the threshold level the gain through the gate is 0 dB. When the detector level is below the threshold level, the gain is reduced at a ratio of 1:2. This means that if the detector is 3dB below the threshold the signal output will be 6dB below the threshold or 3dB below the input level.

The gate threshold level is also indicated by the red bar above the gate input meter. You can adjust the threshold level using this indicator as well as by using the “Thresh” knob.

Attack Control

The “Attack” knob allows you to adjust how quickly the gain reduction is decreased to 0 dB when the detector level goes above the threshold level. When this control is set to Auto, the attack rate is controlled by how much the detector level is above the threshold. When you set the attack to another value other than “Auto” that value, measured in milliseconds, will control how quickly the gate opens. The maximum value is 100 milliseconds. Attack times other than auto are especially useful when using the gate as a trigger. If the key signal is a little early you can use the attack to delay the trigger slightly. It is also useful to remove the initial transients of impulsive sounds.

Release Control

The “Release” knob controls the release time of the gate. This parameter is measured in milliseconds and can range from 5 ms to 5 sec. The release time controls how quickly the gate closes after the detector drops below the threshold value. For settings below 90 ms or so the gate closes pretty abruptly and may introduce unwanted artifacts into your audio, depending on the signal.

Using The Sidechain Key Input w/Selectable Filter

The gate provides a sidechain that processes audio before the detector determines the current level. The sidechain can process either the channel signal or some external side chain input signal.

Sidechain Listen Button

This button (the speaker icon next to the Sidechain Enable) allows you to monitor the audio being sent to the gate's detector. This will allow you to listen to external audio that is being routed to the sidechain, and also hear the effect of the sidechain filter. When you are done listening to the sidechain, click this button again to hear ChannelStrip's normal output.

Sidechain Routing Button

This button (labeled “C” in the illustration) is used to control the routing of the input signal to the gate sidechain. When the button is in the “C” state, the signal used by the sidechain is the signal being processed by ChannelStrip. When the button is in the “SC” state, the signal used by the sidechain is the input or bus selected in the “side chain input” pop-up in your DAW's plug-in window header. If nothing is selected in that pop-up, the input to the sidechain will be silence and the gate will never open.

Sidechain Filter Enable

This button (labeled “SC Ena”) allows you to toggle the sidechain filter in and out of the audio path to the detector.

Filter Type Button

This button (indicating a peaking/parametric filter in the illustration) is used to select the filter type of the single band of side chain EQ. You may choose from the 6 different types of filters detailed in the filter type section in “Operating the Strip”.

Filter Band Boost/Cut Control

Use this knob (labeled "dB" in the illustration) to adjust the gain of the filter band for the peaking, high and low shelf filter types. This parameter is ignored for the other filter types. In the shelving filters the maximum boost is +12 dB and the maximum cut is -24 dB. In the peaking filters the maximum boost/cut is ± 24 dB. When you increase the boost for a filter band above 15 dB, the filter gets nicely aggressive and resonant.

Filter Band Frequency

Use this knob (labeled "Hz" in the illustration) to adjust the characteristic frequency of the filter. For the peaking and bandpass filter types this controls the center frequency of the filter. For the high and low cut filter types this control adjusts the 3 dB point of the filter. For the shelving filters this control adjusts the shelf transition point.

Filter Bandwidth

Use this knob (labeled "BW" in the illustration) to adjust the characteristic width of the filter. This control only has effect for peaking, shelving and bandpass filter types. Please note that this parameter controls the bandwidth (measured in octaves), not the quality factor (or "Q"). If you have been using Q controls, the numbers will be backwards from what you are used to. Small numbers mean narrow filters and large numbers mean wide filters. For peaking and bandpass filter types, this parameter controls the bandwidth of the filter in octaves. For the high and low shelving filter types this parameter adjusts the amount of dip/peak and the slope of the shelf. When this parameter is set to 0.1 you will get the largest dip/slope available and when the parameter is 2.5, you will get a classic first order shelf (which has a transition band that is about 1 decade wide; e.g. if it is a high shelf with a frequency of 10 kHz and a gain of 10 dB, the gain will be at 0 dB near 1kHz).

Compressor

Depending on the state of the "Post EQ" button (the default state is for the compressor to come first in the signal chain), the next block in the signal processing chain is the compressor. The compressor is used to adjust the high–level dynamics of a signal. As with the gate, the sidechain can be used to make the compressor frequency sensitive (so it can be used like a de-esser) or to reduce the gain of the signal in response to some external event (this allows the compressor to be used like a ducker or for other creative effects).

Figure 13.67: Compressor

Often, you will want to compress the signal before you equalize it. Sometimes you will need to equalize the signal before you compress it. ChannelStrip provides that flexibility with the “Post EQ” button. This is a very important part of ChannelStrip because it allows you to test different processing scenarios quickly and easily. It also allows you to compare the two different approaches without having to stop the transport so you can get a much more visceral comparison.

Theory Of Operation

The operation of the compressor is very similar to the gate. Based upon your setting for the sidechain routing switch, either the sidechain input signal or the channel signal is fed to the sidechain filter. The sidechain filter provides one band of equalization that may be used to accentuate or cut certain frequencies (parametric or shelf filters) or limit the key to a certain range of frequencies (cut or bandpass filters). You control the filter type and the filter parameters with the filter type button and the “dB”, “Hz” and “BW” knobs.

You can enable the sidechain filter with the enable button. After the sidechain signal has been processed by the sidechain filter it is measured by a level detector that determines the instantaneous level of the signal (in the case that ChannelStrip is running in stereo mode the detector is linked with the other channel in the stereo pair and the higher level of the two channels is used).

When the compressor is enabled the signal will be attenuated based on how much the detected level is above the threshold you set with the “Thresh” knob and what compression ratio is set with the “Ratio” knob.

The dynamic behavior of the opening and closing of the gate is controlled with the “Attack” and “Release” knobs and the compressor character switch.

Audio Dynamics

Compressors are important in controlling the dynamic range of the source material you are working with. While the instruments, your ears, the microphones and your digital audio workstation all have dynamic ranges that are greater than 100 dB, most reproduction and delivery media have significantly reduced dynamic ranges. Compression is used, in its simplest form, to help reduce the dynamic range of your project or elements of the project to a range that is reproducible. It does this by making the soft material louder and the loud material softer. This type of processing can also be used to change the character of the sound instead of just adjusting the dynamic range. The compressor in ChannelStrip excels at both types of processing.

Compressor Enable

This button is in the Master Enables section. When this button is off, the compressor section will not change the signal. The order of this button will change depending on whether the compressor is set to "Post EQ".

Post EQ

The "Post EQ" button places the compressor section after the equalizer in the signal chain. By providing the capability to switch the routing on the fly, ChannelStrip allows you to determine the most effective routing for your particular signal quickly and easily.

Auto Gain

When the "Auto Gain" button is enabled the compressor automatically adjusts the makeup gain in the compressor output stage so that if the manual "O Gain" knob is set to 0 dB the static gain reduction for a 0 dB input level will be about 7 dB. This number was chosen because it works well with the default settings of the "Attack" and "Release" knobs to provide enough pad to not clip fast transients. The "O Gain" knob will apply additional trim to the internal automatic gain. If the threshold is set very low (e.g. –60 dB) and auto gain is enabled, you will not be able to add very much manual gain (only about 1 – 2 dB) even though the readout on the knob will go up to + 30 dB. This is an internal limitation of the compressor.

Compressor Knee Control

The "Knee" knob allows you to adjust shape of the compressor transfer function when the Compressor Character is set to MIO (the Knee control has no effect for the other compressor character algorithms and will not be visible in those modes).

When the Knee control is set to 0, the transfer function of the compressor is a classic "hard-knee" in which the compressor applies no gain reduction when the detector is below the threshold, and the gain is reduced by the ratio when the detector level is above the threshold.

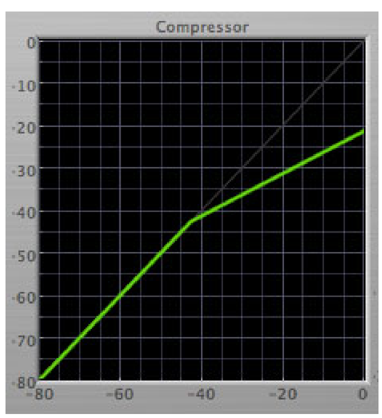


Figure 13.68: Knee Control at 0

When you increase the Knee parameter from 0 to 1 the knee of the transfer function gradually softens until the compressor functions as a soft-knee compressor when the Knee parameter is 1.

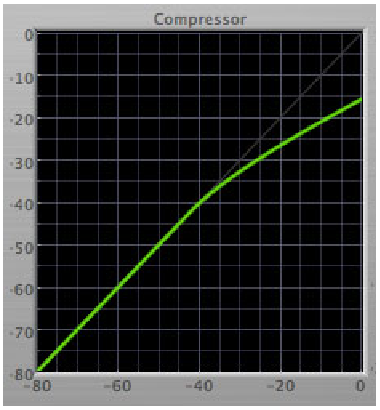


Figure 13.69: Knee Control at 1

You can also adjust the Knee parameter to negative values, which has the effect introducing a "kink" in the compressor transfer function at the threshold. This can yield useful results on percussive material.

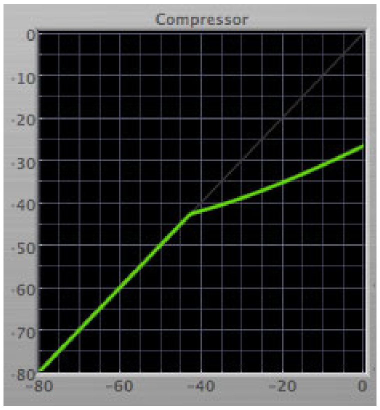


Figure 13.70: Knee Control at -0.5

Manual Make-Up Gain

The "O Gain" knob allows you to manually adjust the makeup gain applied to the signal after the gain reduction applied by the compressor. If the Auto Gain switch is off, this is the amount of makeup gain applied. If the Auto Gain switch is on, then this parameter is a trim added to the internally computed makeup gain. The makeup gain is enabled and disabled along with the rest of the compressor.

Compressor Character

Use the compressor character button to determine the overall dynamic characteristics of the compressor. There are three "classic" ChannelStrip 2 settings to choose from:

- Smooth – appropriate for full mixes or single instruments that do not have big transients. Provides very smooth compression with few artifacts, no distortion and limited transient control.
- Warm – the most versatile setting for the compressor. Balances transient control with audibility of the compression. Appropriate for a wide range of signals including harmonic instruments with large transients (e.g. Plucked bass).
- Fast – provides significant transient control at the expense of transparency and added distortion. Appropriate for impulsive signals with significant transients. Supports very fast (e.g. 1 sample) gain reduction attacks.

as well as the original MIOStrip/MIOComp hardware compressor algorithm:

- MIO – This compressor algorithm is very different from the other three. The MIO compressor generates its gain reduction directly from the detector level. The Attack and Release parameters directly control the measurement of the detector level. This allows the MIO compressor to function as a limiter as well as a compressor and a leveling amplifier. The MIO compressor algorithm also supports an adjustable Knee. In general, the MIO algorithm is more flexible and controllable than the other algorithms, and as a result, we have made it the factory default for ChannelStrip.

Threshold Control

The "Thresh" knob controls the level at which the compressor begins to reduce the gain applied to the signal. When the detector level is below the threshold level, no gain reduction is applied. As the detector level increases above the threshold level, the gain is reduced as indicated by the knee diagram associated with the compressor. The compressor knee is soft. The ratio increases as the difference between the detector level and the threshold increases.

The compressor threshold level is also indicated by the red bar above the gate input meter. You can adjust the threshold level using this indicator as well as by using the "Thresh" knob.

Ratio Control

The "Ratio" knob controls the 'terminal' ratio used to compute the gain reduction of the compressor. When the ratio associated with the soft knee hits the ratio specified by the ratio knob, the knee 'hardens' and remains at the same constant ratio. If you set the ratio to 1000:1 the compressor will have a soft knee for all input levels and thresholds. This makes the compressor work like a classic all tube limiter/compressor.

Attack Control

The "Attack" knob allows you to adjust how quickly the gain reduction is increases when the detector level goes above the threshold level. This control is calibrated in milliseconds and values range from 0 to 500 ms. The compressor has an 8 sample look-ahead buffer that allows it to have an "instant attack" when you set the attack time to 0. Fast attack times will control the transients of impulsive sounds. Use longer attack times to let the transients through but control the sustains.

Release Control

The "Release" knob controls the release time of the compressor. This knob is calibrated in milliseconds and can range from 5 ms to 5 sec. The release time controls how quickly the gain reduction returns to zero after the detector drops below the threshold value. For settings below 40 ms or so the compressor releases pretty abruptly and may introduce unwanted artifacts into your audio, depending on the signal. In addition, be careful making the release time faster than the attack time.

Using The Sidechain Key Input w/Selectable Filter

The compressor provides a sidechain that processes audio before the detector determines the current level. The sidechain can process either the channel signal or some external side chain input signal.

Sidechain Listen Button

This button (the speaker icon next to the Sidechain Enable) allows you to monitor the audio being sent to the compressor's detector. This will allow you to listen to external audio that is being routed to the sidechain, and also hear the effect of the sidechain filter. When you are done listening to the sidechain, click this button again to hear ChannelStrip's normal output.

Sidechain Routing Button

This button (labeled "C" in the illustration) is used to control the routing of the input signal to the compressor sidechain. When the button is in the "C" state, the signal used by the sidechain is the signal being processed by ChannelStrip. When the compressor is in the "SC" state, the signal used by the sidechain is the input or bus selected in the "side chain input" pop-up in your DAW's plug-in window header. If nothing is selected in that pop-up, the input to the sidechain will be silence and the compressor will never compress.

You can use the filter to achieve a de-essing effect by using the bandpass filter to only compress when the “ess” is present. You can also achieve a combined compression/de-essing effect by using a peaking filter to accentuate the “ess” frequencies and adjusting the threshold and ratio to perform compression when the “ess” is not present and limiting when the “ess” is present. Take a look a the presets for examples.

Sidechain Filter Enable

This button (labeled “SC Ena”) allows you to toggle the sidechain filter in and out of the audio path to the detector.

Filter Type Button

This button (indicating a peaking/parametric filter in the illustration) is used to select the filter type of the single band of side chain EQ. You may choose from 6 different types of filters detailed in the filter type section in “Operating the Strip”.

Filter Band Boost/Cut Control

Use this knob (labeled “dB” in the illustration) to adjust the gain of the filter band for the peaking, high and low shelf filter types. This parameter is ignored for the other filter types. In the shelving filters the maximum boost is +12 dB and the maximum cut is -24 dB. In the peaking filters the maximum boost/cut is ± 24 dB. When you increase the boost for a filter band above 15 dB, the filter gets very aggressive and resonant.

Filter Band Frequency

Use this knob (labeled “Hz” in the illustration) to adjust the characteristic frequency of the filter. For the peaking and bandpass filter types this controls the center frequency of the filter. For the high and low cut filter types this control adjusts the 3 dB point of the filter. For the shelving filters this control adjusts the shelf transition point.

Filter Bandwidth

Use this knob (labeled “BW” in the illustration) to adjust the characteristic width of the filter. This control only has effect for peaking, shelving and bandpass filter types. Please note that this parameter controls the bandwidth (measured in octaves), not the quality factor (or “Q”). If you have been using Q controls, the numbers will be backwards from what you are used to. Small numbers mean narrow filters and large numbers mean wide filters. For peaking and bandpass filter types, this parameter controls the bandwidth of the filter in octaves. For the high and low shelving filter types this parameter adjusts the amount of dip/peak and the slope of the shelf. When this parameter is set to 0.1 you will get the largest dip/slope available and when the parameter is 2.5, you will get a classic first order shelf (which has a transition band that is about 1 decade wide; e.g. if it is a high shelf with a frequency of 10 kHz and a gain of 10 dB, the gain will be at 0 dB near 1kHz).

Equalizer

The next processing section is the Equalizer. The equalizer may appear in the signal chain before the compressor section depending on the state of the “Post EQ’ button in the compressor. The equalizer in ChannelStrip is a very flexible, fully parametric 6 band equalizer. Each band in the equalizer can be configured as any of the six available filter types. Each parameter in the equalizer is continuously adjustable throughout its entire range, so you can set the exact EQ that you need.

Figure 13.71: EQ Section

Theory Of Operation

The equalizer in ChannelStrip work just like every other EQ under the sun with the exceptions that is more flexible, more efficient and sounds better. By adjusting the various parameters associated with each band in the EQ you can control the tonal and timbral balance of the signal. The resonance effect of the peaking filters provides a facility to recreate acoustic resonances that are lacking in the source material with which you are working. One of the nicest aspects of the filters in ChannelStrip is their time domain performance. These filters ring significantly less than comparable filters in other signal processors. This allows you equalize signals without the normal time smearing that you encounter with other equalizers.

Master Enable Button

This button is in the Master Enables section. When this button is off, the EQ section will not change the signal. The order of this button will change depending on whether the compressor is set to "Post EQ".

Filter Type Button

This button (indicating a peaking/parametric filter in the illustration) is used to select the filter type of the single band of side chain EQ. You may choose from 6 different types of filters:

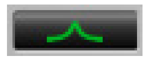


Figure 13.72: Peaking/Parametric

Peaking/Parametric – a second order bell-shaped parametric boost/cut filter. Boost/cut has a range of ± 24 dB. When the boost is greater than +15 dB the filter gains a resonant quality. The center frequency of the filter can be any frequency between 20 Hz and 20 kHz. The bandwidth of the filter is continuously variable between 0.1 octaves and 2.5 octaves.

Figure 13.73: Low Cut

Low Cut – a 12 dB/octave low cut filter with a -3dB point that is continuously adjustable between 20 Hz and 20 kHz.

Figure 13.74: High Cut

High Cut – a 12 dB/octave high cut filter with a -3dB point that is continuously adjustable between 20 Hz and 20 kHz.

Figure 13.75: Low Shelf

Low Shelf – a shelving filter that applies boost/cut to low frequencies. Boost/cut is limited to +12 dB/–24dB. The bandwidth controls the dip/peak that is added at the end of the transition band.

Figure 13.76: High Shelf

High Shelf – a shelving filter that applies boost/cut to high frequencies. Boost/cut is limited to +12 dB/–24dB. The bandwidth controls the dip/peak that is added at the end of the transition band.

Figure 13.77: Bandpass

Bandpass – a bandpass filter with 6dB per octave skirt on the high and low ends of the pass band. The width of the pass band can be adjusted between 0.1 octaves and 2.5 octaves and the center of the pass band is continuously adjustable between 20 Hz and 20 kHz.

Filter Enable Button

Use this toggle button to enable each filter band. When the filter band is turned off the signal will pass through the filter unchanged.

Filter Band Boost/Cut Control

Use this knob (labeled "dB" in the illustration) to adjust the gain of the filter band for the peaking, high and low shelf filter types. This parameter is ignored for the other filter types. In the shelving filters the maximum boost is +12 dB and the maximum cut is -24 dB. In the peaking filters the maximum boost/cut is ± 24 dB. When you increase the boost for a filter band above 15 dB, the filter gets very aggressive and resonant. You can use this feature to good effect when you need to reconstruct a resonance for a recorded instrument that lacks one. For example, you could place a narrow +24 dB peaking filter between 60 and 80 Hz on a kick drum track that lacked a "belly" for the drum.

Filter Band Frequency

Use this knob (labeled "Hz" in the illustration) to adjust the characteristic frequency of the filter. For the peaking and bandpass filter types this controls the center frequency of the filter. For the high and low cut filter types this control adjusts the 3 dB point of the filter. For the shelving filters this control adjusts the shelf transition point.

Filter Bandwidth

Use this knob (labeled "BW" in the illustration) to adjust the characteristic width of the filter. This control only has effect for peaking, shelving and bandpass filter types. Please note that this parameter controls the bandwidth (measured in octaves), not the quality factor (or "Q"). If you have been using Q controls, the numbers will be backwards from what you are used to. Small numbers mean narrow filters and large numbers mean wide filters. For peaking and bandpass filter types, this parameter controls the bandwidth of the filter in octaves. For the high and low shelving filter types this parameter adjusts the amount of dip/peak and the slope of the shelf. When this parameter is set to 0.1 you will get the largest dip/slope available and when the parameter is 2.5, you will get a classic first order shelf (which has a transition band that is about 1 decade wide; e.g. if it is a high shelf with a frequency of 10 kHz and a gain of 10 dB, the gain will be at 0 dB near 1kHz).

Controlling The EQ With The Transfer Function

As described in the operation guide earlier in this manual, you can control each band of the EQ directly from the EQ transfer function display associated with the 6 band equalizer.

Master Gain

The one fader in ChannelStrip's user interface controls the master gain of the plug-in. This fader is not shown in the illustration of the EQ section, but it is shown in the overall processor illustration at the beginning of this manual. The "Master Gain" fader allows you to add up to +10 dB of gain or up to -160 dB of attenuation to the output signal from the EQ processor or compressor block (depending on the processor order) before going to the limiter.

Delay Section

You can add up to 255 samples of delay to the output of ChannelStrip. This is useful for dynamically slipping tracks, doing acoustical time alignment or compensating for the delay of other plug-ins in your mix.

You can use automation on the delay to create interesting dynamic flanging effects. Simply duplicate a track and enable automation on the delay control for one of the copies. As you change the delay through one of the copies you will create a nice, controllable phasey flanging sound.

Limiter

Figure 13.78: Limiter

The final processing block in ChannelStrip 3 is a limiter. The limiter can be used to ensure that the final output of ChannelStrip doesn't distort due to extreme EQ boosts. This also makes it very easy to use ChannelStrip as a mastering plug-in on your final mix bus.

The limiter threshold may be set between 0 and -12 dB using the Limiter Threshold knob. The gain reduction is displayed via the ring around the encoder, sweeping from right to left.

The limiter applies autogain to the audio when engaged; as the threshold is lowered, the output gain is increased a complementary amount.

Dirty Delay

Plug-in categories: **Production** and **Delay**

Introduction

Dirty Delay is a DSP plug-in for MIOConsole3d which provides time delay effects for adding depth and ambience to your recordings. The plug-in includes support for setting the time delay in terms of tempo and note lengths. It also allows you to add filtering, distortion characteristics and feedback to the delay path for cool musical effects.

Features include:

- Independent Delay Path with Variable Time Controls
- Shelf and Cut Filters
- Character Processing on the Feedback

Figure 13.79: Dirty Delay's User Interface

WARNING!: Dirty Delay functions as a "feedback" delay. There are gain elements within the delay path that allow you to increase signal levels above unity gain. This can lead to signal accumulation at the output, which can lead to clipping and extremely loud sounds coming through your speakers! You must remain aware of the gain elements that could cause these undesired effects and be prepared to act accordingly if you encounter extreme feedback events.

The particular parameters to watch closely are the Filter (particularly when set above the 0 line), Character Gain, Feedback and Crossfeed. Turn these down to tame feedback. Also be aware that when you automate these parameters, playback through the complete automated sequence might provide the desired control over feedback levels for the song. However, stopping playback at particular points in the automated sequence where these parameters are high may lead to feedback. A quick solution is to mute your speakers, pull down the offending parameters, or jump to a silent section and start the transport.

Operation

The Dirty Delay user interface uses several control elements to operate the plugin. These elements are:

Link Control

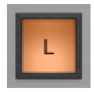


Figure 13.80: Link Control

This button toggles stereo linking of the left and right channel parameters. When engaged, any changes made to the left channel control knobs will also change the right channel's knobs, maintaining any pre-existing offset. Changing the right channel's knobs directly will change only for the right channel in order to adjust the desired offset with respect to the left channel knobs.

Control Knob

Control Knobs are used to control various parameters.

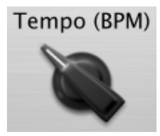


Figure 13.81: Control Knob

Some control knobs sweep from a minimum to maximum value, from left to right. Others are centered by default and can sweep left or right to subtract or add value respectively, or to set the parameter as a percentage.

You can change the value of each knob in a number of different ways. Click and drag the knob to change the value continuously. Dragging up or to the right will increase the value, while dragging down or to the left will decrease the value. If you hold down the ⌘ (Command) key when you click, you will be able to adjust the value with finer precision. If you hold the ⌥ (Option) key when you click, the knob will reset to its default value. You may also double-click a knob to reset it.

Click on the number (readout) of the knob to display a text entry field that allows you to type in a number directly. The pop-up will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, **⌘.** (Command + .) or **ESC** keys. Hit **return** or **enter** to confirm the value and dismiss the pop-up. Hit the **tab** key to confirm the value and display an entry field for the next control. **⇧-tab** (Shift + tab) will display the entry field for the previous control). Hit the **⌘.** (Command + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.

When you enter a number into the pop-up entry, you can use a couple of abbreviations: "k" multiplies the number by 1000 and "m" divides the number by 1000. So if you want to enter 16,500 Hz you can just type 16.5k.

Toggle Button

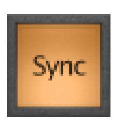


Figure 13.82: Toggle Button (On)

Toggle buttons are simple on/off switches. They light up when they are on and are dark when they are off. Toggle the state of the button by clicking on it. This button is used to enable Tempo Sync within Dirty Delay.

Radio Button

Radio buttons are like toggle buttons, except that they simultaneously disable another parameter when enabled. These buttons are used to modify the delay note values with triplet or dotted note time.

Tempo Tap

Figure 13.83: Tempo Tap Pad

The Tempo Tap Pad allows you to set the delay time in BPM with successive mouse clicks. Tap in quarter-note beats to set the Tempo.

Note Length

Figure 13.84: Note Length Popup Menu

This pop-up menu selects the note value of the delay with respect to the tempo.

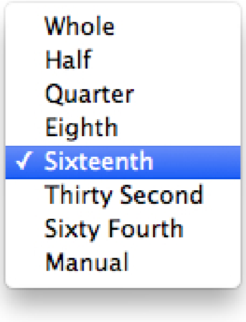


Figure 13.85: Note Length Selection

Standard musical note values are available, as well as Manual mode. When set to a musical note, the correct delay time is set automatically and the associated delay knob will be grayed out. When set to Manual, you can set the delay time using the delay knob.

Delay Filter

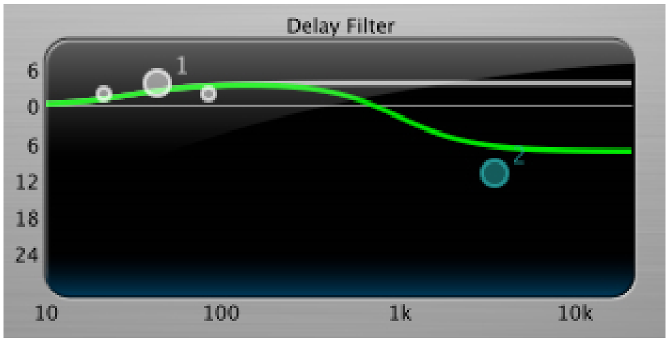


Figure 13.86: Delay Filter

This section allows detailed control over the 2-band EQ applied to the delayed signal. The horizontal axis provides frequency calibration in Hertz (Hz). The vertical axis provides level calibration in decibels (dBr). The heavy green line indicates the relative change in level at each frequency that is created by the combined effects of all the active bands in the equalizer. Each EQ band is represented by a numbered, colored dot in the transfer function.

The band that is currently being edited will be displayed along with the overall response curve. If the associated band is a parametric filter there will also be two smaller colored dots that can be used to control the bandwidth of the filter. Clicking on a large colored dot and dragging it will allow you to adjust the frequency and gain of the associated band. Mac Command-click, Windows Control-click or double click the dot to toggle the band enable. Mac Option-click or Windows Alt-click the dot to adjust the bandwidth (drag right to increase, and left to decrease bandwidth). Mac Command+Option-clic or Windows Control+Alt-click the dot to switch the band filter type. Click and drag the smaller dots associated with a larger dot to adjust the filter bandwidth.

To dismiss the filter curve, click anywhere in the black area of the transfer function. This will deselect the filter point, and the only trace displayed will be the green master curve.

Filter Type

Each filter band has a filter type control that allows you to choose the shape of the filter applied by that band. Each band provides 6 different types of filter shapes:

- **Peaking/Parametric** – a second order bell-shaped parametric boost/cut filter. Boost/cut has a range of ± 24 dB. When the boost is greater than +15 dB the filter gains a resonant quality. The center frequency of the filter can be any frequency between 20 Hz and 20 kHz. The bandwidth of the filter is continuously variable between 0.1 octaves and 2.5 octaves.

Figure 13.87: Peaking/Parametric

- **Low Shelf** – a shelving filter that applies boost/cut to low frequencies. Boost/cut has a range of ± 24 dB. The bandwidth controls the dip/peak that is added at the end of the transition band.

Figure 13.88: Low Shelf

- **High Shelf** – a shelving filter that applies boost/cut to high frequencies. Boost is limited to +12 dB (reflected in the Transfer Function display) whereas cut goes to –24dB. The bandwidth controls the dip/peak that is added at the end of the transition band.

Figure 13.89: High Shelf

High Cut – a 12 dB/octave high cut filter with a -3dB point that is continuously adjustable between 20 Hz and 20 kHz.

Figure 13.90: High Cut

- **Low Cut** – a 12 dB/octave low cut filter with a -3dB point that is continuously adjustable between 20 Hz and 20 kHz.

Figure 13.91: Low Cut

- **Bandpass** – a bandpass filter with 6dB per octave skirt on the high and low ends of the pass band. The width of the pass band can be adjusted between 0.1 octaves and 2.5 octaves and the center of the pass band is continuously adjustable between 20 Hz and 20 kHz.

Figure 13.92: Bandpass

You can select from these types via three different methods. Each time you click on the Filter Type control, the band will switch to the next type in the list (and wrap to the beginning when you hit the end of the list). If you click and hold the mouse button, a pop-up menu listing all of the types will appear after about 1/4 of a second. You can select the type directly from this pop-up menu. If you want to access the menu without having to wait, hold down the ^ (Control) key when you click or right-click.

Character Model Menu

Figure 13.93: Character Popup Menu

This pop-up menu selects the Character model applied to the delayed signal. There are 8 models available, which are detailed in the Processing chapter. The pop-up style menus can also be stepped through using the mouse scroll wheel.

Output Level Meter

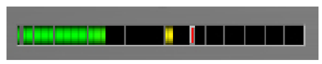


Figure 13.94: Dirty Delay Output Level Meter

Dirty Delay provides a meter displaying the output level of the plug-in. The right-most segment of the meter (above 0dB) is used as a clip indicator and is illuminated red if the output section of the processor detects an over. The clip light remains illuminated until you click on the meter. Mac ⌥ (Option)–click or Windows **Alt**-click the meter to reset the clip light.

> **A Note About Clipping Indicators:**
> The clip lights do not mean that the plug-in is clipping; it means that the audio level in the DSP is currently over 0 dBFS. If you do not lower the signal level you run a chance of actually clipping the input of another processor or D/A converter.

Processing

A Detailed Description

In this chapter we'll discuss what each processing block does and how the controls work.

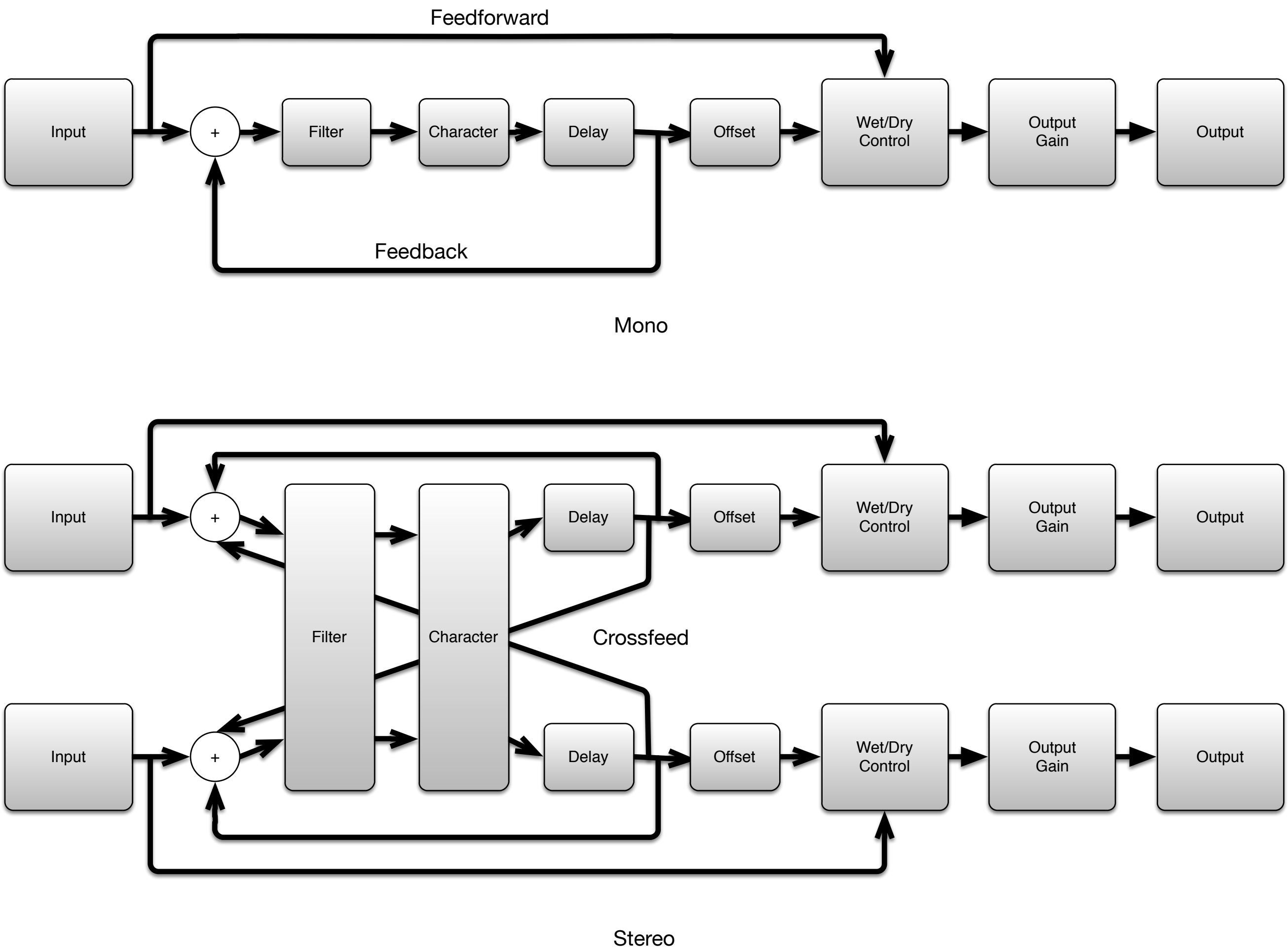


Figure 13.95: Signal Flow

The block diagram above illustrates the overall signal path through Dirty Delay for both the mono and stereo versions. The diagram does not indicate the metering blocks.

Now lets examine the various processing blocks indicated in the diagram and elaborate on additional control elements built around them.

Filter

The filter block provides frequency shaping to a copy of the input signal prior to being delayed. The available controls allow you to set the frequency and level of the variable pre-filters of the delay. Using these controls allows you to tailor desired frequencies with up to +6 dB of boost to -24 dB of cut. See the Delay Filter section in the Operation section for detailed information.

Character

The Character block follows the filter, and applies low order harmonics to the signal being delayed. Each Model represents a digital "copy" of an analog device; some models are made from measuring a device with different combinations of settings. The most effective way to utilize Character is to listen to how each model affects your sound rather than relying on the name alone.

- None: No modeling is applied.
- Valve: A tube-based EQ input stage.
- Soft Sat: Tube-based EQ with saturation.
- Transformer 1 Applies the harmonic distortion signature of a transformer-coupled input.
- Transformer 2: A variation of the “Transformer 1” model.
- Tube DI: Mastering quality tube DI.
- Tube EQ: Mastering quality EQ.
- Tube Soft Sat: Mastering quality EQ with saturation.
- Tube Symmetrical Clipper: Mastering quality EQ

While most of the models can be applied to all types of material, particularly after judicious filtering, overuse of the saturation models can lead to undesired effects.

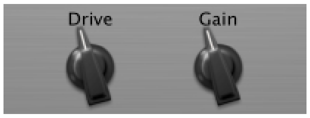


Figure 13.96: Character Drive and Gain Controls

The Drive control applies ±24 dB of gain to the signal before it goes to the modeling section. This allows you attenuate the signal for a subtle effect, or boost to get a more dramatic effect.

As gain is changed before the modeling stage, the inverse gain is applied after. For example, if you set the Drive control to +6 dB, the gain is boosted 6 dB before the model and cut by 6 dB after. This allows you to “push” the modeling section with no increase in output level. You may also hit the model with less signal by setting the Drive control to a negative value without loss of overall volume.

You may still experience in increase in signal level when using positive Drive gain with some models; you can use the Gain control to correct this.

Tempo Control

This block generates the reference tempo for metrical time delays. Depending on the setting of the controls in the UI, this may generate the tempo from data provided by the host, by the rate of taps on the tap button or directly from the Tempo knob if the tempo is set manually.

With Sync toggled on, the base tempo automatically locks to the tempo set in your host. Toggle Sync off to manually enter the delay in ms on a per channel basis (stereo), or to tap tempo.

Click the tap button successively at the desired rhythm to set the tempo. When Sync is engaged, the Tap button will be disabled.

Delay

The Note Length sets the resolution of the delay time. The values are defined in note duration (Whole, Half, Quarter, etc.) with respect to the tempo grid.

Figure 13.97: Triplets Button

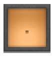


Figure 13.98: Dotted Notes Button

Any note value can be modified to repeat in the form of triplets or dotted notes by engaging these buttons.

Offset

The offset block adjusts when you hear the delayed signal relative to the total feedback delay. The time offset controlled by this block is adjusted with the Offset Knob in the UI. The Offset knob allows you to make fine adjustments to the groove of the delayed audio. Turning the knob to the left will make the delays arrive earlier, and turning to the right will make the delays arrive later. The range of the Offset knob is defined by the delay time. You can adjust the offset in each direction by up to half of the delay time.

Feedback Path

The Feedback path takes the output of the delay line and sums with the input. The Feedback knob sets how much of the delayed signal is fed back into the delay path. Higher settings will generate more repeats, and can eventually cause the output to clip over time as the repeated signals accumulate. The Feedback phase invert button causes the feedback to be inverted when it is engaged; every time the signal goes around the feedback loop the polarity will be inverted.

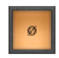


Figure 13.99: Phase Invert Button

> Dedicated Phase Invert Buttons are available for both the Feedback and Crossfeed controls.

Crossfeed

The crossfeed controls are only available when using Dirty Delay as a stereo plug-in. The left crossfeed control sets how much of the left channel delay feedback is fed into the right channel as a percentage, and vice versa for the right crossfeed control.

Wet/Dry Control

The Wet/Dry control sets the mix between the delay (wet) and original (dry) signals as a percentage. Setting this control to 0 (full dry) lets no delayed sound through. Setting this control to 100 (full wet) lets only delayed sound pass; you would want to use this setting when inserting Dirty Delay on an aux bus. If you are inserting Dirty Delay directly on an input or bus, use this control to balance the original audio and delay.

Output Gain

The Out Gain control allows you to vary the final output level of Dirty Delay by ±24dB.

HaloVerb

Plug-in categories: **Production and Reverb**

Introduction

HaloVerb is a DSP plug-in for MIOConsole3d which provides a flexible algorithm-based reverb to add depth and ambience to your mixes.

HaloVerb is not a vintage effects emulation, convolution or special effects reverb, rather it has been designed to create a simple neutral space in which to stage your mix elements in the soundfield.

The idea is: space doesn't have a "sound"… a sense of space doesn't diminish perception, it enhances it.

You will find HaloVerb fits in places you might not ordinarily think of using a reverb. For instance, just a touch of a short RT60 HaloVerb added to a headphone cue mix can work wonders for your talents' comfort factor and fighting off ear fatigue (especially with IEMs).

Features include:

- Comprehensive controls
- Graphic display of the Low and High Frequency Pre-Filters
- Graphic display of the Reverb Impulse

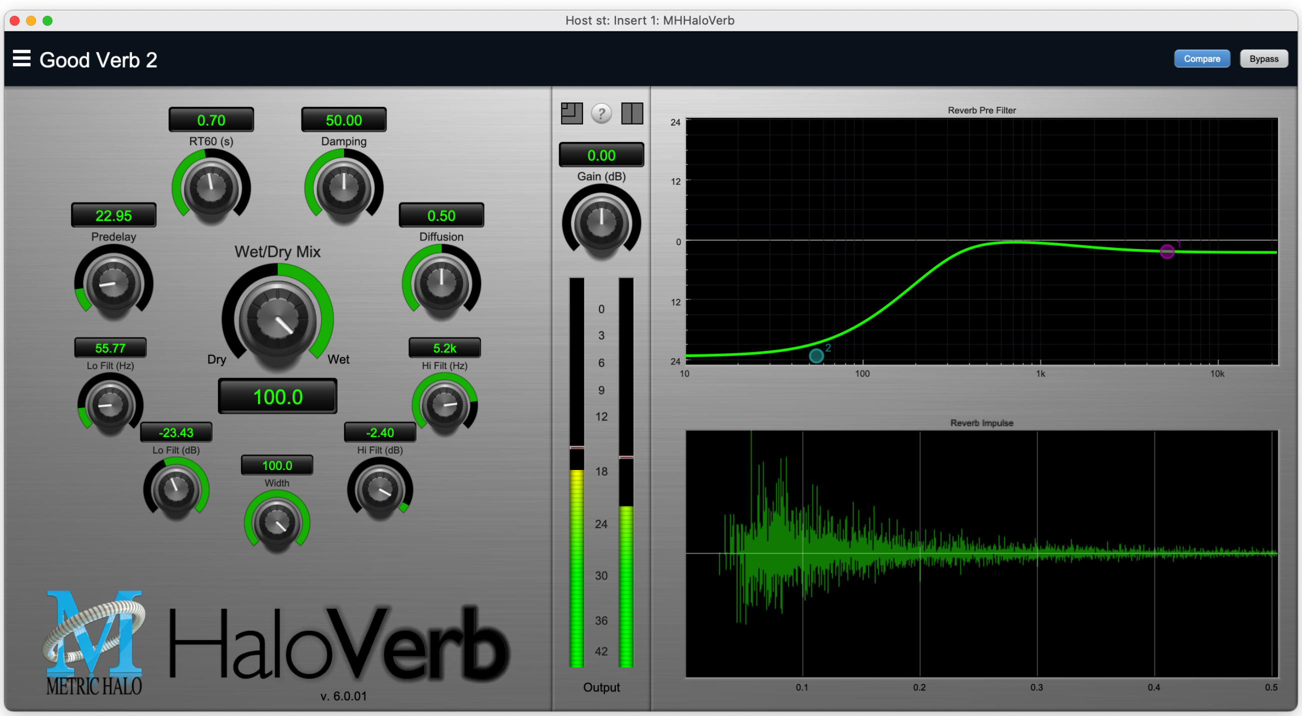


Figure 13.100: HaloVerb's User Interface

Operation

The HaloVerb user interface uses a few different control elements to control all of the processing. These elements are:

Control Knob

Control Knobs are used to control the value of various continuous parameters of a process. There are three styles of encoders:

Figure 13.101: Swept Knob

The rings around these encoders sweep from a minimum to maximum value, from left to right. The exception is the Cutoff controls, which sweep from right to left.

Figure 13.102: Spread Knob

The ring around this encoder starts at 12 o'clock and spreads to both sides equally as the Stereo Width is increased.

Figure 13.103: Plus/Minus Knob

The rings around these encoders start at 12 o'clock and sweep to either side. These knobs are used for Wet/Dry and Gain controls.

Control tips

You can change the value of each knob in a number of different ways.

- Click and drag the knob to change the value continuously.
- Dragging up or to the right will increase the value, while dragging down or to the left will decrease the value.

  Hint: If you hold down the ⌘ (Command) key when you click, you will be able to adjust the value with finer precision.
- ⌥ (Option)-click to reset any knob to its default value. You may also double-click a knob to reset it.
- **Rapid parameter entry**
  - Double-click the numeric readout of the knob to type in a number directly. The text-entry field will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, or **⌘.** (Command + .) keys.
  - Hit **return** or **enter** to confirm the value and dismiss the pop-up.
  - Hit the **tab** key to confirm the value and immediately activate the entry field for the next control. **⇧-tab** (Shift + tab) will display the entry field for the previous control).
  - Hit **⌘.** (Command + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.
  - When you enter a number into the pop-up entry, you can use a couple of abbreviations: “k” multiplies the number by 1000 and “m” divides the number by 1000. So if you want to enter 16,500 Hz you can just type 16.5k.

Output Level Meter

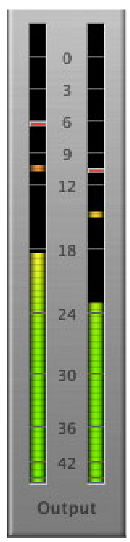


Figure 13.104: HaloVerb Output Level Meter

The HaloVerb provides a meter displaying the output level of the plug-in. The top segment of the meter (above 0dB) is used as a clip indicator and is illuminated red if the output section of the processor detects an over. The clip light remains illuminated until you click on the meter. Mac ⌥ (Option)–click or Windows **Alt**-click the meter to reset the clip light.

> **A Note About Clipping Indicators:**
> The clip lights do not mean that the plug-in is clipping; it means that the audio level in the DSP is currently over 0 dBFS. If you do not lower the signal level you run a chance of actually clipping the input of another processor or D/A convertor.

Low and High Frequency Pre-filter

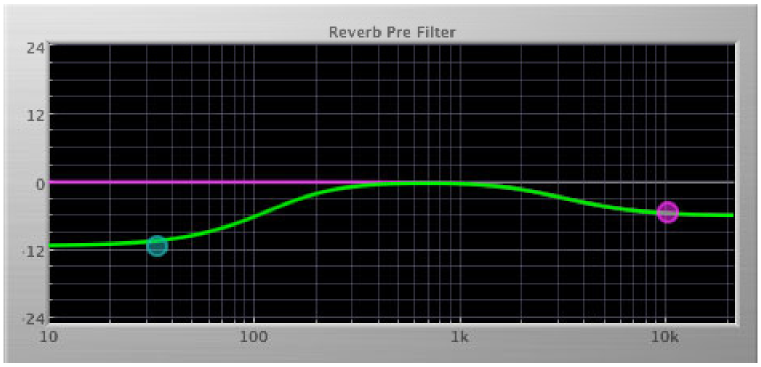


Figure 13.105: Low and High Frequency Pre-filter

The pre-filter graph is a visual representation of the filter shape as set by the Cutoff Hz and Cutoff dB controls for the Low and High filters. As these controls are changed, the dots shows the knee freqencies and levels. You may also click and drag the dots to change the filter placements.

Reverb Impulse Display

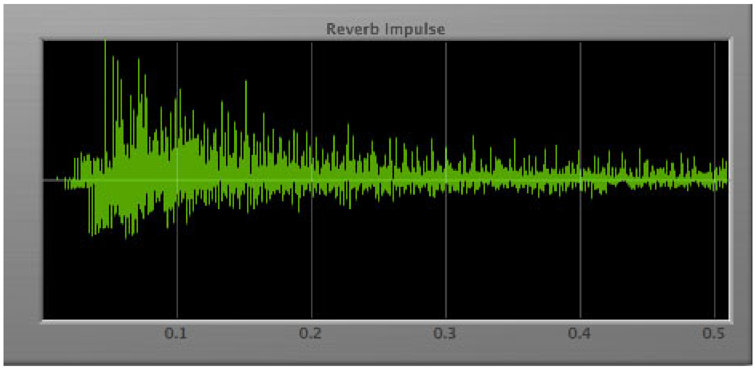


Figure 13.106: Reverb Impulse Display

The reverb impulse display gives you a visual indication of what the reverb parameters are doing. The horizontal axis represents time (up to .5 seconds) and the vertical axis represents unscaled level. Clicking on the display will toggle between the Reverb Impulse Display and Energy Time Curve.

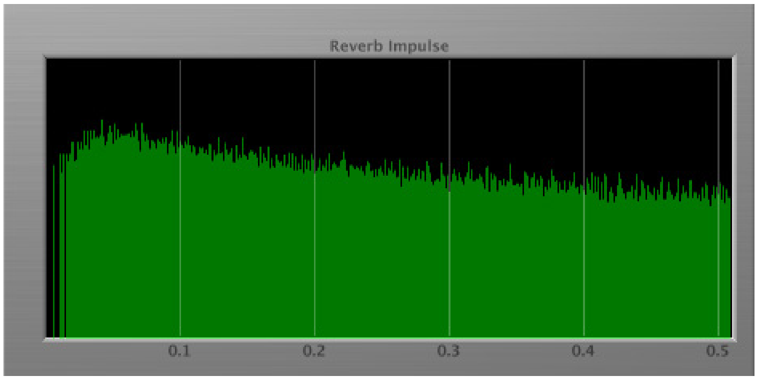


Figure 13.107: Energy Time Curve

Processing

A Detailed Description

In this chapter we'll discuss how the controls work.

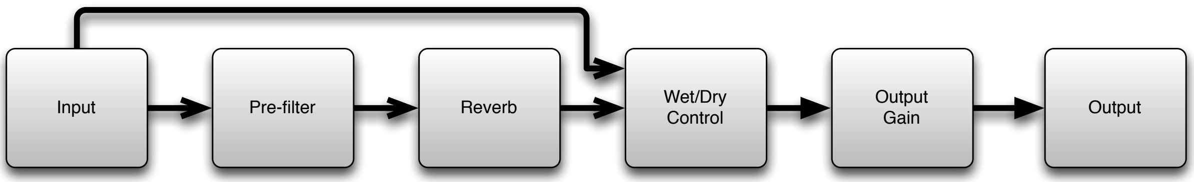


Figure 13.108: Signal Flow

The block diagram above illustrates the overall structure of the processing system provided by HaloVerb. The diagram does not indicate the metering blocks.

Now lets examine the various processing blocks indicated in the diagram.

Prefilter

These controls set the frequency and level of the shelving pre-filters. Using these controls allows you to tailor how much low and high frequency material is removed before the signal enters the reverb process.

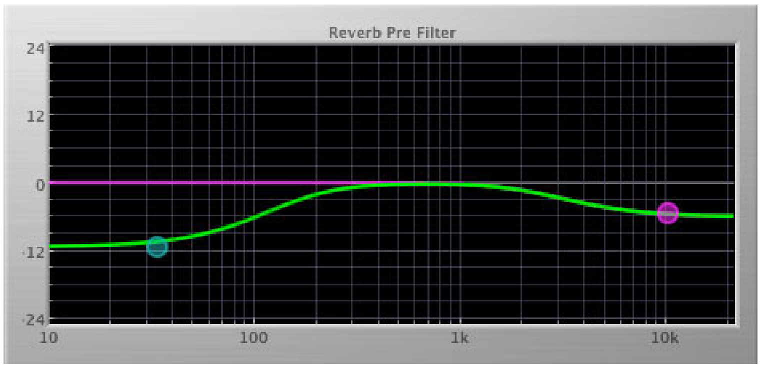


Figure 13.109: Pre-filters at 34 Hz and 9kHz

Reverb

Let's take a look at what HaloVerb's parameters do, using the Reverb Impulse display. All controls will be at the factory default except the control being discussed, so that you can see how that control changes the reverb characteristics.

Please note that some combinations of settings will look similar in the impulse display, but will sound very different.

Predelay

The Predelay control sets the time in milliseconds before the first reflection of the reverb is heard, up to a maximum of 170 ms. When looking at the Reverb impulse, the predelay is seen as the time between the left side of the window and the beginning of the impulse:

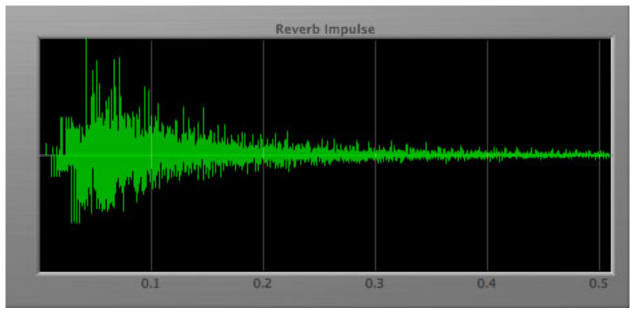


Figure 13.110: Predelay of 0 ms

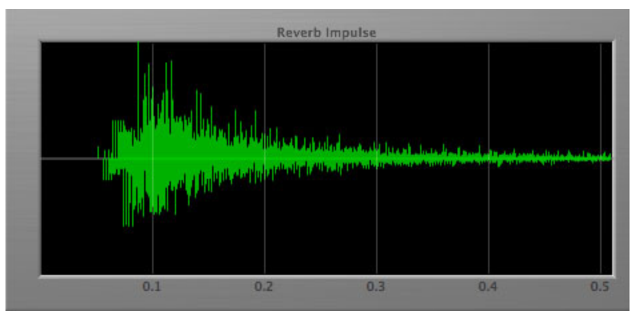


Figure 13.111: Predelay of 45 ms

RT60

The RT60 control sets the overall reverb time in seconds; RT60 is a standard measurement of how long it takes the acoustic energy in a room to fall below 60 dB. When looking at the Reverb impulse, changes in the RT60 will be seen as a "thickening" of the reverb tail:

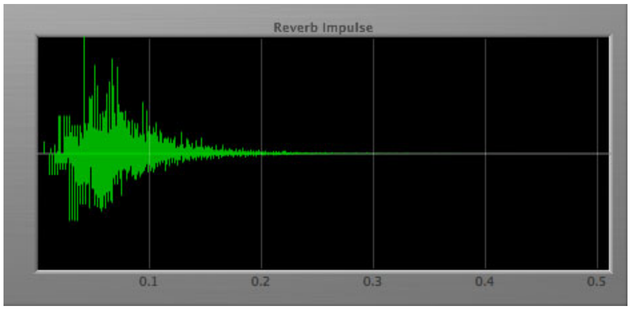


Figure 13.112: RT60 of .1 Seconds

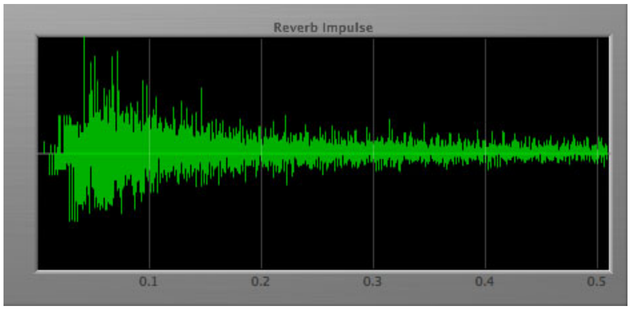


Figure 13.113: RT60 of 7 Seconds

Damping

The Damping control sets how much of the reverb signal is removed before being fed back into itself as a percentage. Low damping settings can lead to a ringing, metallic sound.

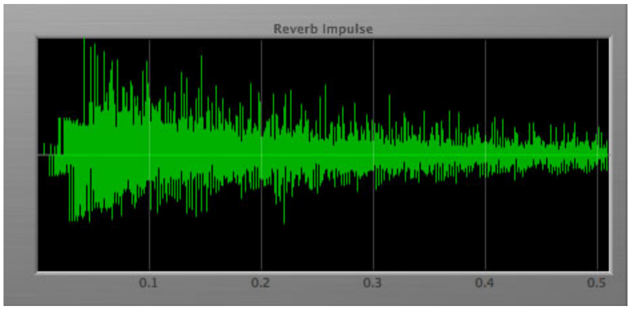


Figure 13.114: Damping of 0%

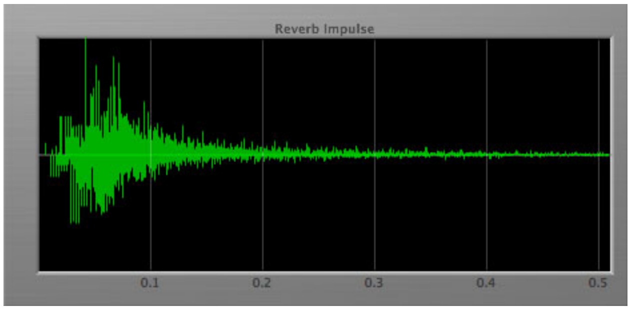


Figure 13.115: Damping of 100%

Diffusion

The Diffusion control determines how dense the reverb is. A low diffusion setting emphasizes individual echos, while a higher diffusion "pushes" the echos together and thickens the reverb. Some higher settings will cause the reflections to *decrease*due to phase cancellation.

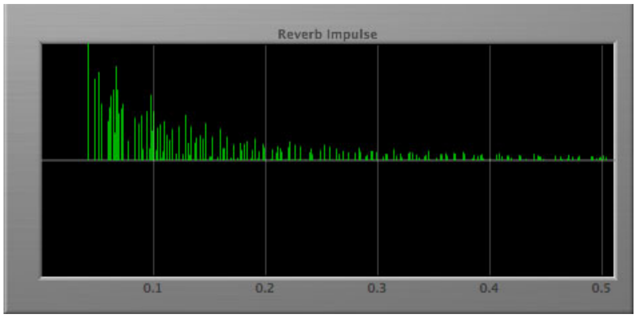


Figure 13.116: Diffusion of 0%

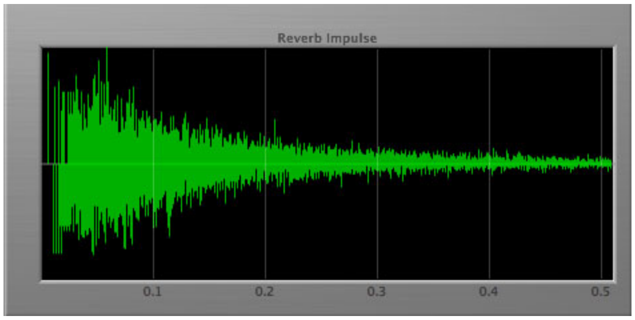


Figure 13.117: Diffusion of 75%

Width

The Width control is only available when using HaloVerb as a stereo plug-in. Width allows you to vary the "spread" of the reverb from mono (panned to the center of the image) to full stereo.

Wet/Dry Control

The Wet/Dry control sets the mix between the reverb (wet) and original (dry) signals as a percentage. Setting this control to 0 (full dry) lets no reverberated sound through. Setting this control to 100 (full wet) lets only reverb pass; you would want to use this setting when inserting HaloVerb on an aux bus. If you are inserting HaloVerb directly on an input or bus, use this control to balance the original audio and reverb.

Output Gain

The Gain control allows you to vary the final output level of HaloVerb by ±20dB

Multiband Dynamics

Plug-in categories: **Production**, **Dynamics** and **Mastering**

Introduction

Multiband Dynamics is a DSP plug-in for MIOConsole3d which provides frequency based dynamic processing.

Features include:

- Three-way crossover
- Three independent channels of compression
- Limiter
- Dynamic transfer function display
- SpectraFoo™ spectrum analysis

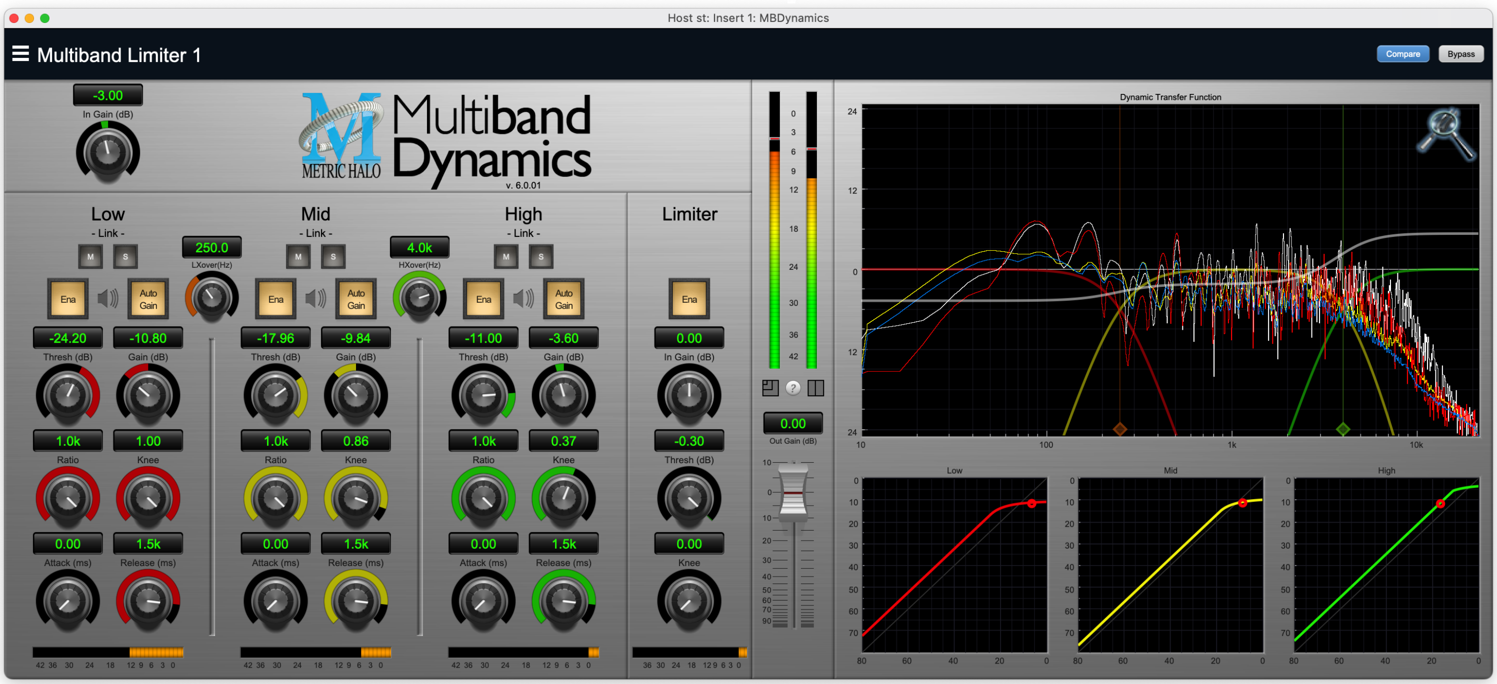


Figure 13.118: Multiband Dynamics's User Interface

Operation

As with most plug-ins, Multiband Dynamics provides many copies of controls that are all operated in a similar manner. The Multiband Dynamics user interface uses a few different control elements to control all of the processing. These elements are:

Control Knob

Control Knobs are used to control the value of various continuous parameters of a process. Examples of these types of parameters include: Attack Time, Release Time, Threshold, etc. There are two styles of encoders:

Figure 13.119: Swept Knob

The rings around these encoders sweep from a minimum to maximum value, normally from left to right. The one exception is the threshold controls, which sweep from right to left.

Figure 13.120: Plus/Minus Knob

The rings around these encoders start at 12 o'clock and sweep to either side. These knobs are used for gain control, where straight up is no gain change, turning to the left cuts the signal and turning to the right boosts it.

Control tips

You can change the value of each knob in a number of different ways.

- Click and drag the knob to change the value continuously.
- Dragging up or to the right will increase the value, while dragging down or to the left will decrease the value.

  Hint: If you hold down the ⌘ (Command) key when you click, you will be able to adjust the value with finer precision.
- ⌥ (Option)-click to reset any knob to its default value. You may also double-click a knob to reset it.
- **Rapid parameter entry**
  - Double-click the numeric readout of the knob to type in a number directly. The text-entry field will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, or **⌘.** (Command + .) keys.
  - Hit **return** or **enter** to confirm the value and dismiss the pop-up.
  - Hit the **tab** key to confirm the value and immediately activate the entry field for the next control. **⇧-tab** (Shift + tab) will display the entry field for the previous control).
  - Hit **⌘.** (Command + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.
  - When you enter a number into the pop-up entry, you can use a couple of abbreviations: "k" multiplies the number by 1000 and "m" divides the number by 1000. So if you want to enter 16,500 Hz you can just type 16.5k.

Toggle Button

Figure 13.121: Toggle Button (Off)

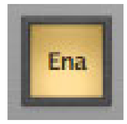


Figure 13.122: Toggle Button (On)

Toggle buttons are simple on/off switches. They light up when they are on and are dark when they are off. You toggle the state of the button by clicking on it. These buttons are used to enable compressor sections and the limiter, and to select auto gain.

Fader

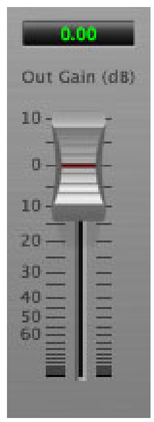


Figure 13.123: Output Gain

The fader is unique in that only one fader is used in the interface for Multiband Dynamics. It works in much the same fashion as the control knobs. Instead of dragging up/right or down/left to change the value, you directly drag the fader knob. The other "tricks" described for the knobs also work with the fader. The fader is used to control the master output gain of the plug-in.

Band Solo Button

Figure 13.124: Band Solo Button (Disabled)

Figure 13.125: Band Solo Button (Enabled)

The band solo button allows you to listen to an individual crossover band or combination of them. This allows you to hear the effect of each compressor section independantly.

Master / Slave Link Controls

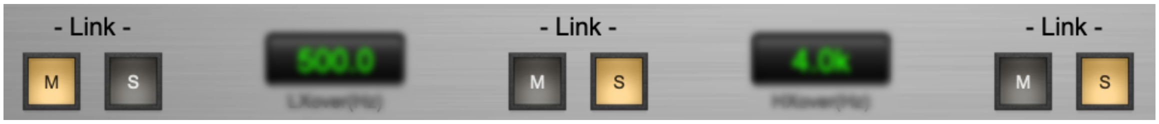


Figure 13.126: Master and Slave Link Buttons

This "M" button sets which band will be the link control master. Only one band can be set as the master. Click "M" to toggle the master link control for the band on or off. Clicking the master link control for a different band automatically toggles off the previously active master band. Changes made in the master link band to Threshold, Gain, Ratio, Knee, Attack and Release will change the same parameters in any slaved bands by the same amount.

The "S" button sets which band(s) will be slaved to the master. Click "S" to toggle the slave link controls on or off. Changes made in the slave linked bands to Threshold, Gain, Ratio, Knee, Attack and Release will change their relative offset with respect to the corresponding control in the master band.

Gain Reduction Meter

Figure 13.127: Gain Reduction Meter

The gain reduction meter, which has an orange bar and grows down from 0 dB, shows the amount of attenuation being applied by its associated dynamics processor at any given time. If you right-click (Mac/Win) or Mac ^ (Control) click on the meter, you may set the scale of the gain reduction meter to any of the following values:

- 54 dB
- 24 db
- 12 db
- 6 db
- 3 db

Output Meter

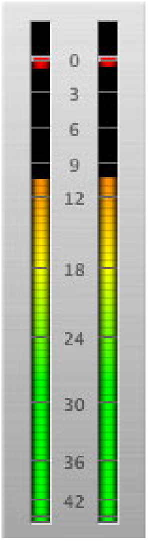


Figure 13.128: Output Meter

For the main output stage of Multiband Dynamics we have provided meters driven with SpectraFoo™ metering technology. These meters show, in addition to the peak metering provided for the input stages, RMS level and VU level. The peak level is represented by the floating colored bar, the RMS level by the solid colored bar and the VU level by the overlaid gray bar. Both the Peak and RMS level are represented with fast PPM ballistics. The VU meter shows IEEE standard 300 ms RMS average level. When Multiband Dynamics is on a mono insert there will be a single meter. When Multiband Dynamics is running in stereo mode the left meter shows the left channel output level and the right meter shows the right channel output level. The output section clip lights activate if there is an over in the output stage or in any of the processing section input stages. It is reset by clicking on the meter; ⌥ (Option)–click to reset the clip lights on all the meters.

Dynamic Transfer Function

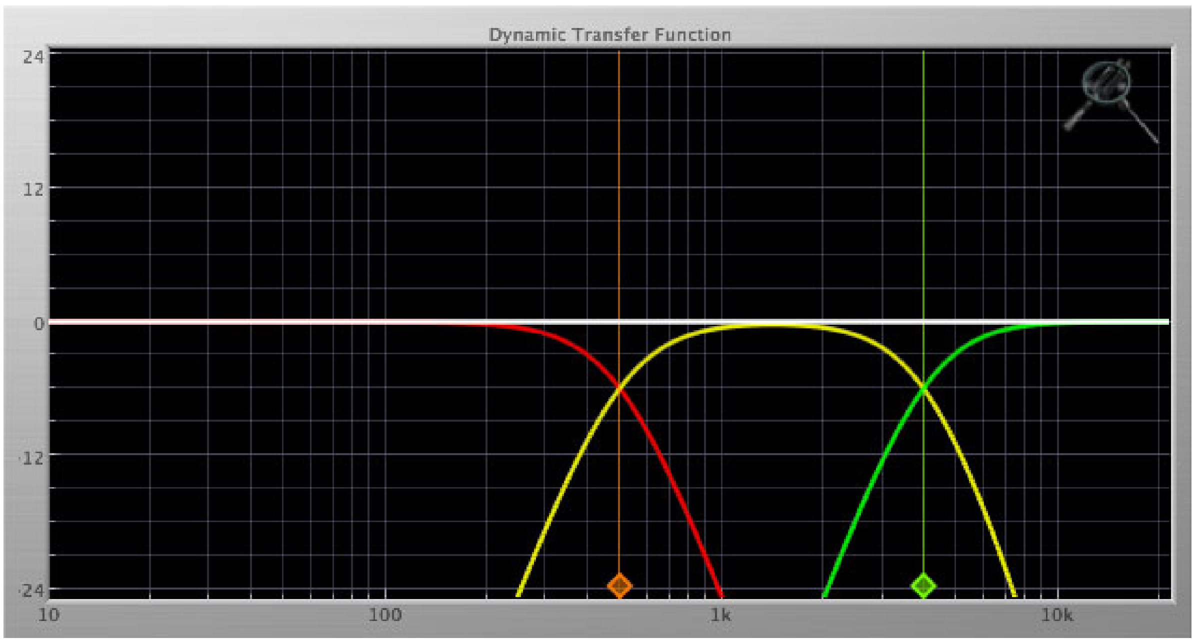


Figure 13.129: Dynamic Transfer Function

The dynamic transfer function is a combination of a visual representation of how the plug-in is processing the signal and an intuitive controller for the crossover filter bands.

The horizontal axis provides frequency calibration in Hertz (Hz), while the vertical axis provides level calibration in decibels (dBr). The heavy white line indicates the relative change in level at each frequency that is created by the combined effects of all of the active bands in the plug-in. Each crossover point (low to mid and mid to high) is represented by a colored dot in the transfer function. You may drag the dots to adjust the crossover points.

If you right-click or ^ (Control) click on the transfer function, you will see a menu to set the vertical dB scale for the display. The values are:

- ±3 dB
- ±6 dB
- ±12 dB
- ±24 dB
- ±36 dB

Spectragraph Analyzer

Clicking the SpectraFoo™ logo in the upper right hand corner of the transfer function will activate the spectragraph, showing the realtime frequency analysis of your signal:

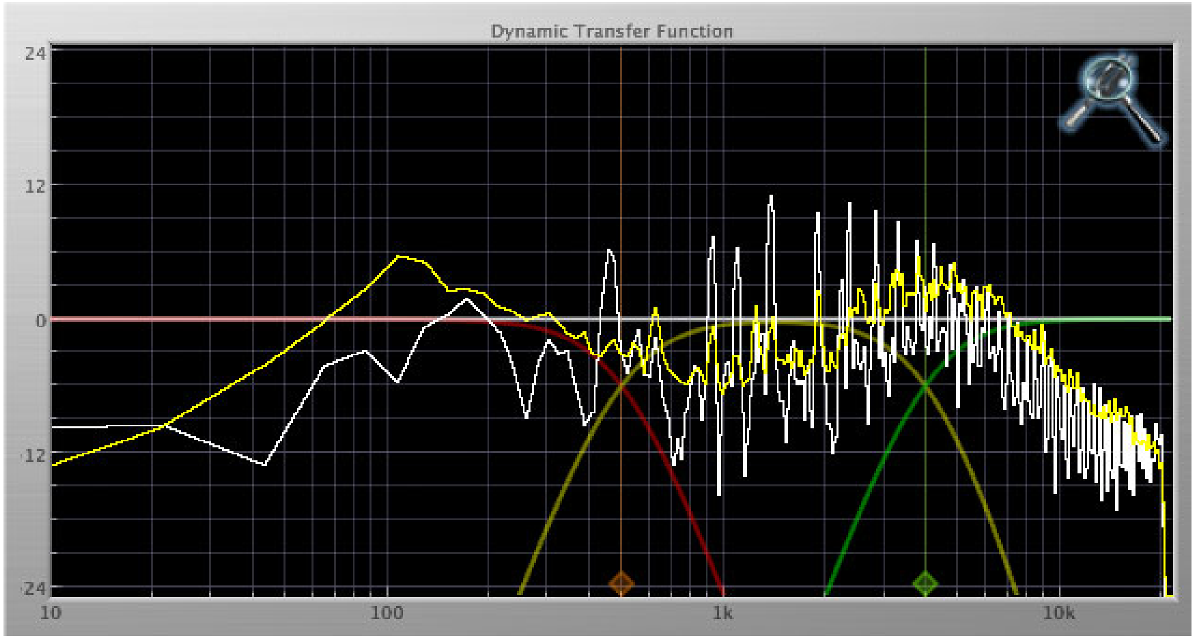


Figure 13.130: Spectragraph Display

The traces are:

- White: Left channel instantaneous display
- Red: Right channel instantaneous display
- Yellow: Left channel average display
- Blue: Right channel average display

The *instantaneous* trace updates in real-time, allowing you to see the immediate peak level of your audio. The *average* trace displays the level as averaged over a short period, giving you a more general view.

The spectragraph analyses the signal post-output gain, allowing you to see the effects of the processors. To disable the spectragraph entirely, click the active 'Foo icon.

If you right-click or ^ (Control) click on the transfer function, you will see a menu to set options for the spectragraph:

- Show Instantaneous Trace: Toggles whether the spectragraph shows the instant response of your audio.
- Show Average Trace: Toggles whether the spectragraph shows the averaged response of your audio.
- Show Left Channel: Toggles the left channel spectragraph display.
- Show Right Channel: Toggles the right channel spectragraph display.

These settings are stored for each transfer function window separately, and for each instance of Multiband Dynamics.

Dynamics Knee

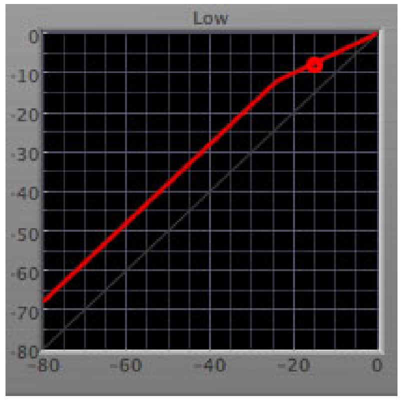


Figure 13.131: Dynamics Knee

Multiband Dynamics contains a Dynamics Knee diagram for each compressor processing section. The diagram provides feedback on the response of the associated dynamics processor. Both the horizontal and vertical axes are calibrated in dBFS. The horizontal axis corresponds to the input level and the vertical axis represents the output level. The heavy line shows the dynamic of the associated processing block. This means that if you sent in a sine wave at a given input level, the output level would be equal to the level shown on the graph. When the processor is working with real dynamic signals, the graph is a good approximation of the response when the attack is fast and the release is slow.

In most cases, however, the dynamic response of the processor will not match its static response. In order to represent this, we have included a “bouncing ball” meter for both the gate and the compressor. This metering is shown as a red circle that is overlaid on the knee diagram. The red circle is placed so its horizontal position is equal to the instantaneous input level and its vertical level is equal to the instantaneous output level. Examining this meter while you are adjusting the dynamics controls will provide you with a great deal of information about how the processor is operating and how the controls interact.

Processing

A Detailed Description

In this chapter we'll discuss what each processing block does and how the controls work.

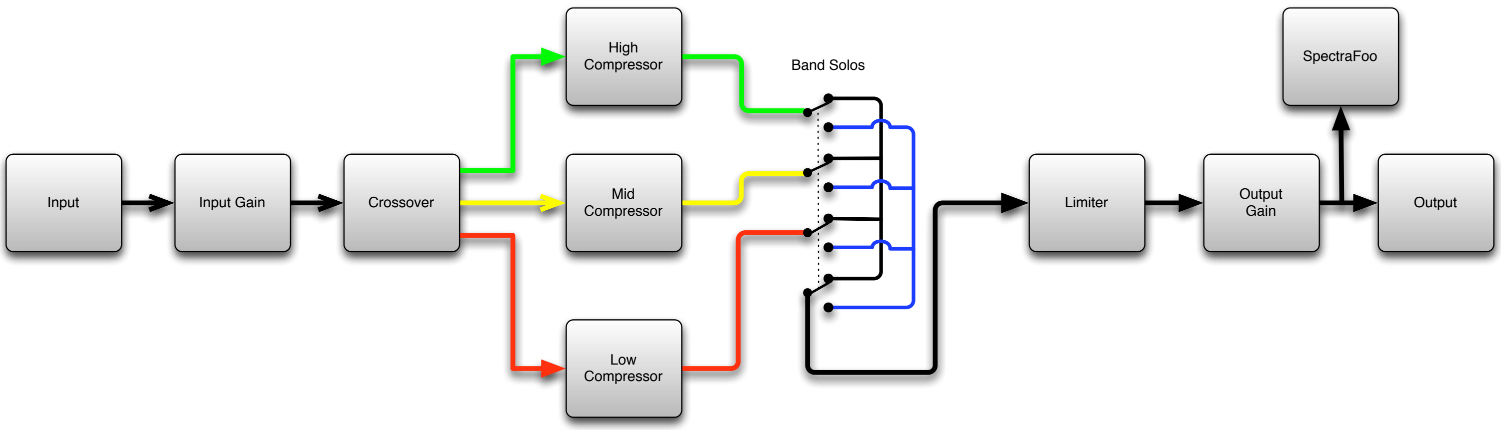


Figure 13.132: Multiband Dynamics Signal Flow

The block diagram above illustrates the overall structure of the processing system provided by Multiband Dynamics. The diagram does not indicate the various metering blocks.

Now lets examine the various processing blocks indicated in the diagram.

Input Conditioning

After the signal is routed to Multiband Dynamics it runs through an input gain block that provides input gain of up to +30 dB. You can use this gain to condition signals that are low in level.

This input gain may also be used to pad out signals by up to –30 dB. While you may find this attenuation useful to just bring down the level through the strip simply and quickly, you must realize that this gain is applied after the signal reaches Multiband Dynamics and will not pad out any clipping that occurs in the A/D converters or in a plug-in that is inserted before Multiband Dynamics.

The input gain is controlled by the "In Gain" knob.

Crossover

The crossover seperates the audio input into low, mid and high bands; each of these bands is then fed through a compressor section. There are two controls to set the crossover points:

Figure 13.133: Crossover Controls

The control marked "LXover" sets the frequency of the low to mid crossover, and the control marked "HXover" sets the frequency of the mid to high crossover.

You may also adjust the crossover points in the dynamic transfer function:

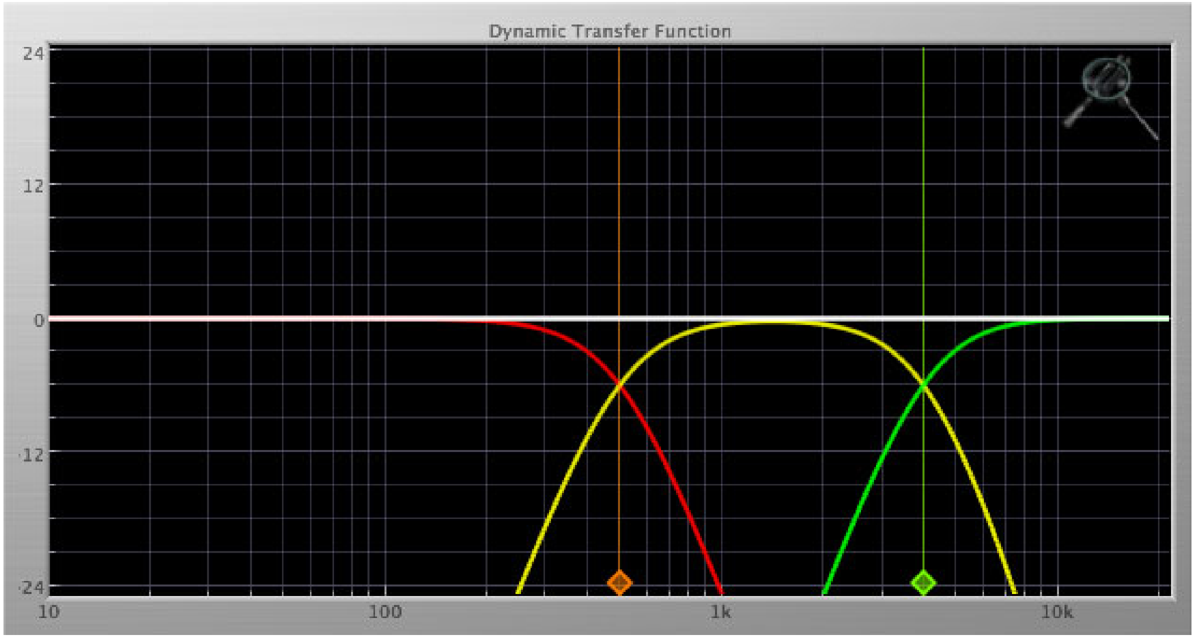


Figure 13.134: Dynamic Transfer Function

Simply grab one of the handles at the bottom of the display and drag to desired frequency. The orange handle adjusts the low/mid crossover point and the green handle adjusts the mid/high crossover point.

Compressor

Figure 13.135: Compressor band

Introduction

Multiband Dynamics contains three separate compressors, one for each crossover band. They are identical, so we will only describe their operation once.

Compressor Enable

When this button is off, the compressor section will not change the signal.

Band Solo Button

This button (the speaker icon next to the Enable button) allows you to monitor the selected compressor's output. This will allow you to listen one or more bands of compression in isolation. When you are done listening to the soloed band(s), click this button again to hear Multiband Dynamics' normal output.

Auto Gain

When the "Auto Gain" button is enabled the compressor automatically adjusts the makeup gain in the compressor output stage so that if the manual "O Gain" knob is set to 0 dB the static gain reduction for a 0 dB input level will be about 7 dB. This number was chosen because it works well with the default settings of the "Attack" and "Release" knobs to provide enough pad to not clip fast transients. The "O Gain" knob will apply additional trim to the internal automatic gain. If the threshold is set very low (e.g. –60 dB) and auto gain is enabled, you will not be able to add very much manual gain (only about 1 – 2 dB) even though the readout on the knob will go up to + 30 dB. This is an internal limitation of the compressor.

Threshold Control

The "Thresh" knob controls the level at which the compressor begins to reduce the gain applied to the signal. When the detector level is below the threshold level, no gain reduction is applied. As the detector level increases above the threshold level, the gain is reduced as indicated by the knee diagram associated with the compressor. The compressor knee is soft. The ratio increases as the difference between the detector level and the threshold increases.

The compressor threshold level is also indicated by the red bar above the gate input meter. You can adjust the threshold level using this indicator as well as by using the “Thresh” knob.

Manual Make-Up Gain

The “Gain” knob allows you to manually adjust the makeup gain applied to the signal after the gain reduction applied by the compressor by ±30 db. If the Auto Gain switch is off, this is the amount of makeup gain applied. If the Auto Gain switch is on, then this parameter is a trim added to the internally computed makeup gain. The makeup gain is enabled and disabled along with the rest of the compressor.

Ratio Control

The “Ratio” knob controls the ‘terminal’ ratio used to compute the gain reduction of the compressor. When the ratio associated with the soft knee hits the ratio specified by the ratio knob, the knee ‘hardens’ and remains at the same constant ratio. If you set the ratio to 1000:1 the compressor will have a soft knee for all input levels and thresholds. This makes the compressor work like a classic all tube limiter/compressor.

Compressor Knee Control

The “Knee” knob allows you to adjust shape of the compressor transfer function. When the Knee control is set to 0, the transfer function of the compressor is a classic “hard-knee” in which the compressor applies no gain reduction when the detector is below the threshold, and the gain is reduced by the ratio when the detector level is above the threshold.

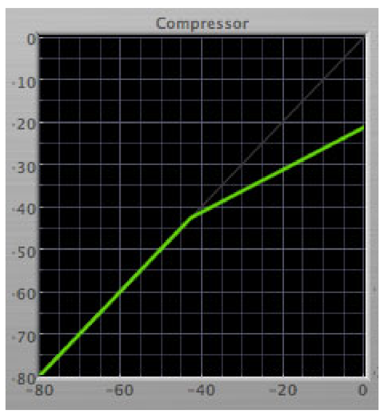


Figure 13.136: Knee Control at 0

When you increase the Knee parameter from 0 to 1 the knee of the transfer function gradually softens until the compressor functions as a soft-knee compressor when the Knee parameter is 1.

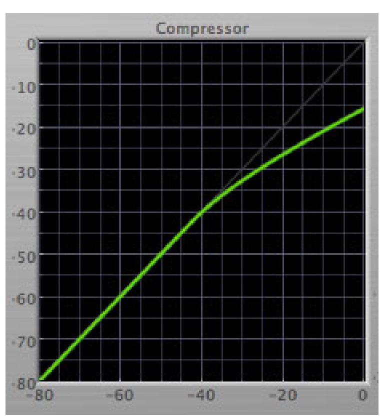


Figure 13.137: Knee Control at 1

You can also adjust the Knee parameter to negative values, which has the effect introducing a "kink" in the compressor transfer function at the threshold. This can yield useful results on percussive material.

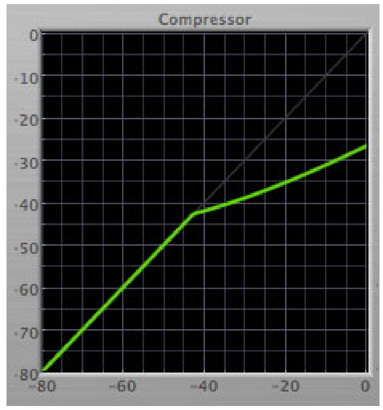


Figure 13.138: Knee Control at -0.5

Attack Control

The "Attack" knob allows you to adjust how quickly the gain reduction is increases when the detector level goes above the threshold level. This control is calibrated in milliseconds and values range from 0 to 500 ms. The compressor has an 8 sample look-ahead buffer that allows it to have an "instant attack" when you set the attack time to 0. Fast attack times will control the transients of impulsive sounds. Use longer attack times to let the transients through but control the sustains.

Release Control

The "Release" knob controls the release time of the compressor. This knob is calibrated in milliseconds and can range from 5 ms to 5 sec. The release time controls how quickly the gain reduction returns to zero after the detector drops below the threshold value. For settings below 40 ms or so the compressor releases pretty abruptly and may introduce unwanted artifacts into your audio, depending on the signal. In addition, be careful making the release time faster than the attack time.

Limiter

Figure 13.139: Limiter

The final processing block in the Multiband Dynamics is a limiter. The limiter can be used to ensure that the final output of ChannelStrip doesn't distort due to extreme EQ boosts. This also makes it very easy to use Multiband Dynamics as a mastering plug-in on your final mix bus.

Limiter Enable

When this button is off, the limiter section will not change the signal.

Input Gain

This control allows you to attenuate or boost the signal level going in to the limiter's input by ±30 dB.

Threshold

The “Thresh” knob controls the level at which the limiter begins to reduce the gain applied to the signal. When the detector level is below the threshold level, no gain reduction is applied. As the detector level increases toward the threshold level, the gain is reduced as determined by the knee control. The limiter knee is soft. The ratio increases as the difference between the detector level and the threshold increases.

Limiter Knee Control

The “Knee” knob allows you to adjust shape of the limiter gain reduction. When the Knee control is set to 0, the transfer function of the limiter is a classic “hard-knee” in which the compressor applies no gain reduction when the detector is below the threshold, and the gain is reduced by the ratio when the detector level is at the threshold.

Multiband Expander

Plug-in categories: **Production**, **Dynamics** and **Mastering**

Introduction

Where Multiband Dynamics concentrates on compression and limiting, the Multiband Expander specializes in frequency-based dynamic expansion.

Multiband Expander's features include:

- Input level control
- Three-way crossover
- Three independent channels of expansion
- Dynamic transfer function display
- SpectraFoo™ spectrum analysis

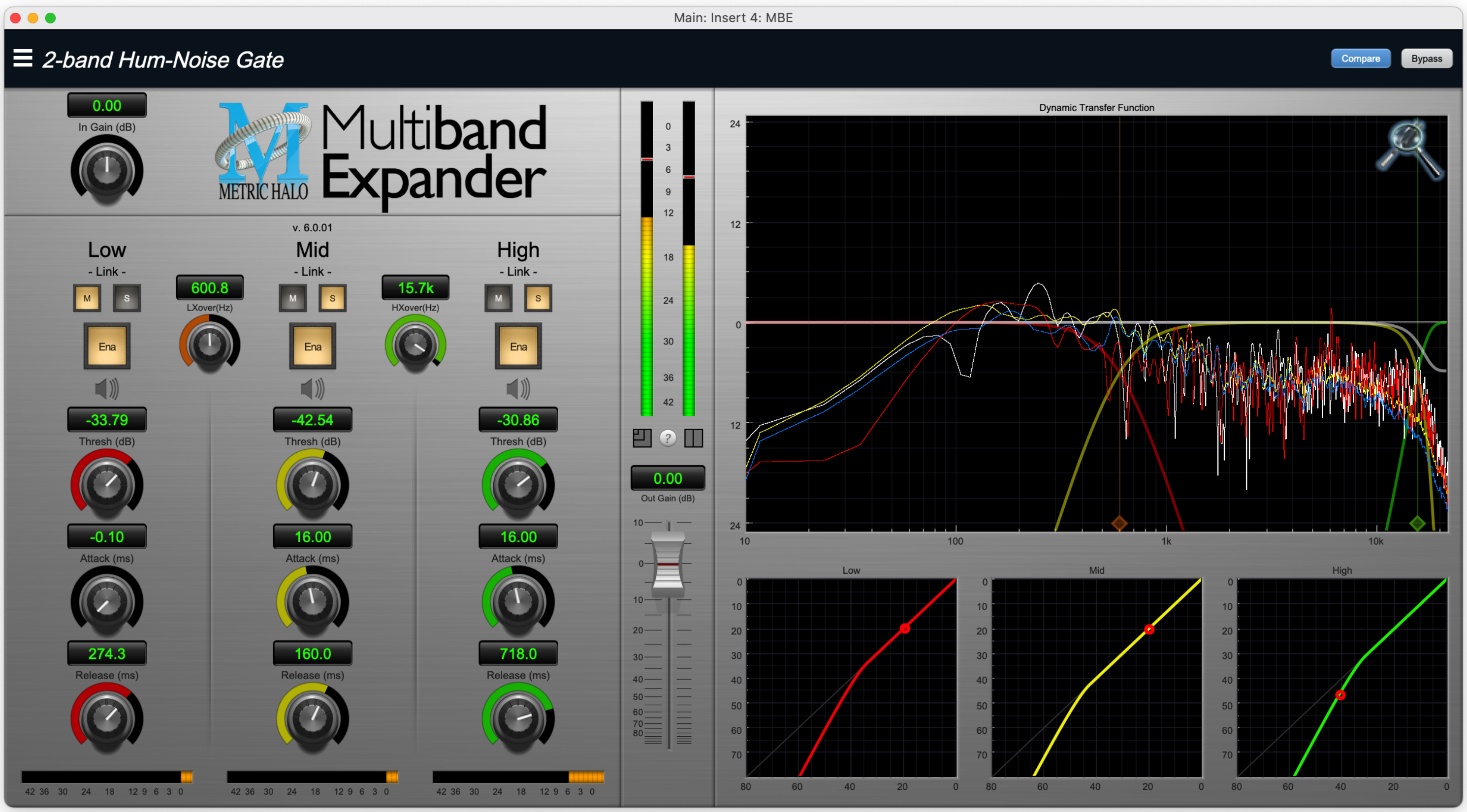


Figure 13.140: Multiband Expander's User Interface

Operation

As with most plug-ins, Multiband Expander provides many copies of controls that are all operated in a similar manner. The Multiband Expander user interface uses a few different control elements to control all of the processing. These elements are:

Control Knob

Control Knobs are used to control the value of various continuous parameters of a process. Examples of these types of parameters include: Attack Time, Release Time, Threshold, etc. There are two styles of encoders:

Figure 13.141: Swept Knob

The rings around these encoders sweep from a minimum to maximum value, normally from left to right. The exceptions are the crossover and threshold controls, which sweep from right to left.

Figure 13.142: Plus/Minus Knob

The rings around these encoders start at 12 o'clock and sweep to either side. These knobs are used for gain control, where straight up is no gain change, turning to the left cuts the signal and turning to the right boosts it.

Control tips

You can change the value of each knob in a number of different ways.

- Click and drag the knob to change the value continuously.
- Dragging up or to the right will increase the value, while dragging down or to the left will decrease the value.

  Hint: If you hold down the ⌘ (Command) key when you click, you will be able to adjust the value with finer precision.
- ⌥ (Option)-click to reset any knob to its default value. You may also double-click a knob to reset it.
- **Rapid parameter entry**
  - Double-click the numeric readout of the knob to type in a number directly. The text-entry field will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, or **⌘.** (Command + .) keys.
  - Hit **return** or **enter** to confirm the value and dismiss the pop-up.
  - Hit the **tab** key to confirm the value and immediately activate the entry field for the next control. **⇧-tab** (Shift + tab) will display the entry field for the previous control).
  - Hit **⌘.** (Command + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.
  - When you enter a number into the pop-up entry, you can use a couple of abbreviations: “k” multiplies the number by 1000 and “m” divides the number by 1000. So if you want to enter 16,500 Hz you can just type 16.5k.

Toggle Button

Figure 13.143: Toggle Button (Off)

Figure 13.144: Toggle Button (On)

Toggle buttons are simple on/off switches. They light up when they are on and are dark when they are off. You toggle the state of the button by clicking on it. These buttons are used to enable compressor sections and the limiter, and to select auto gain.

Fader

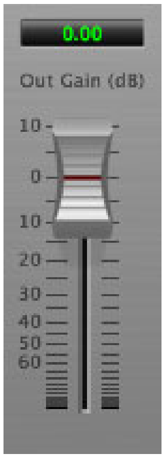


Figure 13.145: Output Gain

The fader is unique in that only one fader is used in the interface for Multiband Expander. It works in much the same fashion as the control knobs. Instead of dragging up/right or down/left to change the value, you directly drag the fader knob. The other "tricks" described for the knobs also work with the fader. The fader is used to control the master output gain of the plug-in.

Band Solo Button

Figure 13.146: Band Solo Button (Disabled)

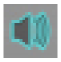


Figure 13.147: Band Solo Button (Enabled)

The band solo button allows you to listen to an individual crossover band or combination of them. This allows you to hear the effect of each compressor section independantly.

Master / Slave Link Controls

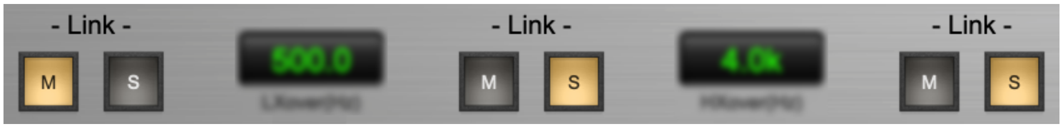


Figure 13.148: Master and Slave Link Buttons

This "M" button sets which band will be the link control master. Only one band can be set as the master. Click "M" to toggle the master link control for the band on or off. Clicking the master link control for a different band automatically toggles off the previously active master band. Changes made in the master link band to Threshold, Gain, Ratio, Knee, Attack and Release will change the same parameters in any slaved bands by the same amount.

The "S" button sets which band(s) will be slaved to the master. Click "S" to toggle the slave link controls on or off. Changes made in the slave linked bands to Threshold, Gain, Ratio, Knee, Attack and Release will change their relative offset with respect to the corresponding control in the master band.

Gain Reduction Meter

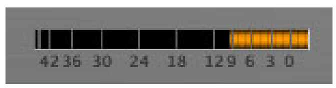


Figure 13.149: Gain Reduction Meter

The gain reduction meter, which has an orange bar and grows down from 0 dB, shows the amount of attenuation being applied by its associated dynamics processor at any given time. If you right-click or ^ (Control) click on the meter, you may set the scale of the gain reduction meter to any of the following values:

- 54 dB
- 24 db
- 12 db
- 6 db
- 3 db

Output Meter

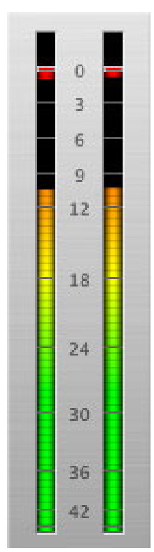


Figure 13.150: Output Meter

For the main output stage of Multiband Expander we have provided meters driven with SpectraFoo™ metering technology. These meters show, in addition to the peak metering provided for the input stages, RMS level and VU level. The peak level is represented by the floating colored bar, the RMS level by the solid colored bar and the VU level by the overlaid gray bar. Both the Peak and RMS level are represented with fast PPM ballistics. The VU meter shows IEEE standard 300 ms RMS average level. When Multiband Expander is on a mono insert there will be a single meter. When Multiband Expander is running in stereo mode the left meter shows the left channel output level and the right meter shows the right channel output level. The output section clip lights activate if there is an over in the output stage or in any of the processing section input stages. It is reset by clicking on the meter; ⌥ (Option)–click to reset the clip lights on all the meters.

> **A Note About Clipping Indicators:**
> The clip lights do not mean that the plug-in is clipping; it means that the audio level in the DSP is currently over 0 dBFS. If you do not lower the signal level you run a chance of actually clipping the input of another processor or D/A convertor.

Dynamic Transfer Function

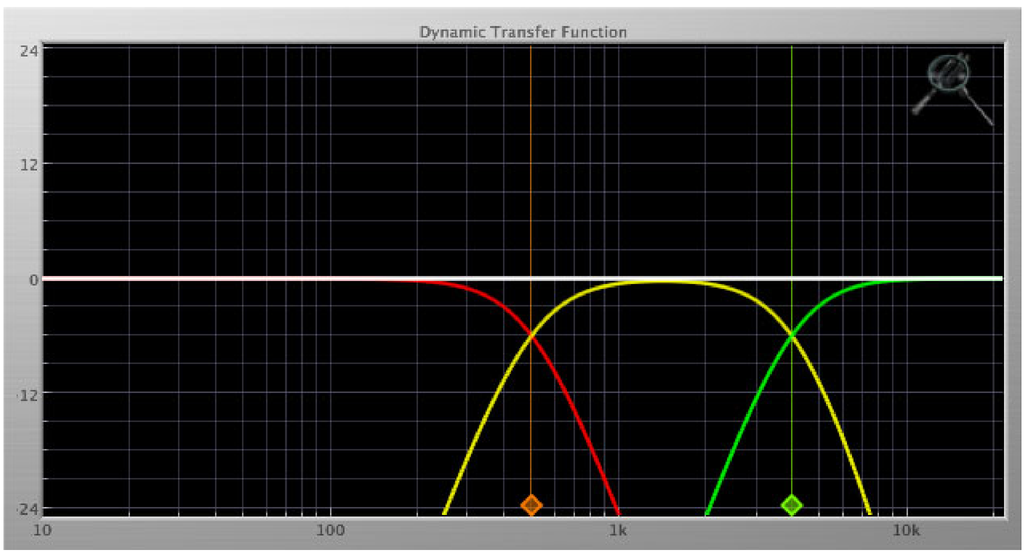


Figure 13.151: Dynamic Transfer Function

The dynamic transfer function is a combination of a visual representation of how the plug-in is processing the signal and an intuitive controller for the crossover filter bands.

The horizontal axis provides frequency calibration in Hertz (Hz), while the vertical axis provides level calibration in decibels (dBr). The heavy white line indicates the relative change in level at each frequency that is created by the combined effects of all of the active bands in the plug-in. Each crossover point (low to mid and mid to high) is represented by a colored dot in the transfer function. You may drag the dots to adjust the crossover points.

If you right-click or ^ (Control) click on the transfer function, you will see a menu to set the vertical dB scale for the display. The values are:

- ±3 dB
- ±6 dB
- ±12 dB

- ±24 dB
- ±36 dB

Spectragraph Analyzer

Clicking the SpectraFoo™ logo in the upper right hand corner of the transfer function will activate the spectragraph, showing the realtime frequency analysis of your signal:

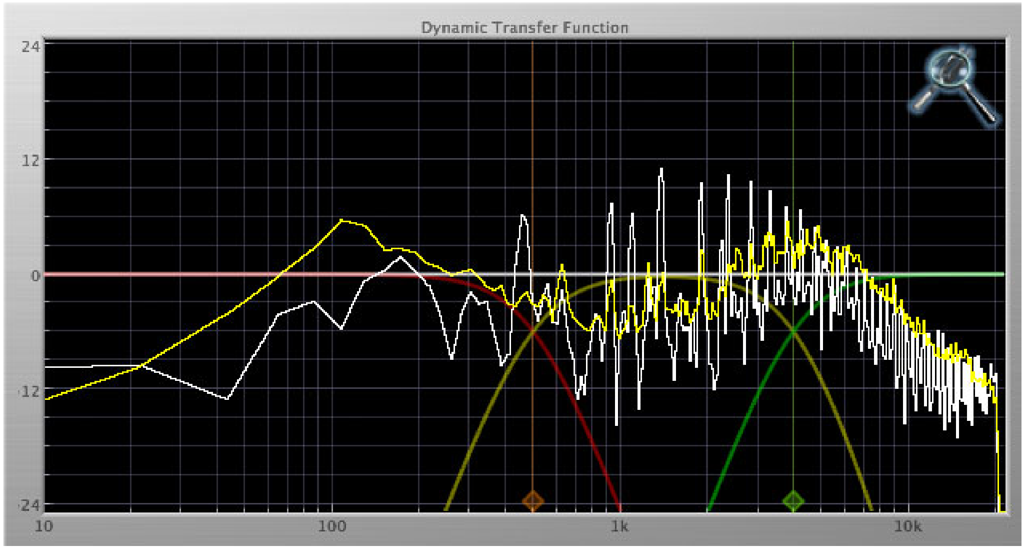


Figure 13.152: Spectragraph Display

The traces are:

- White: Left channel instantaneous display
- Red: Right channel instantaneous display
- Yellow: Left channel average display
- Blue: Right channel average display

The *instantaneous* trace updates in real-time, allowing you to see the immediate peak level of your audio. The *average* trace displays the level as averaged over a short period, giving you a more general view.

The spectragraph analyses the signal post-output gain, allowing you to see the effects of the processors. To disable the spectragraph entirely, click the active 'Foo icon.

If you right-click or ^ (Control) click on the transfer function, you will see a menu to set options for the spectragraph:

- Show Instantaneous Trace: Toggles whether the spectragraph shows the instant response of your audio.
- Show Average Trace: Toggles whether the spectragraph shows the averaged response of your audio.
- Show Left Channel: Toggles the left channel spectragraph display.
- Show Right Channel: Toggles the right channel spectragraph display.

These settings are stored for each transfer function window separately, and for each instance of Multiband Expander.

Dynamics Knee

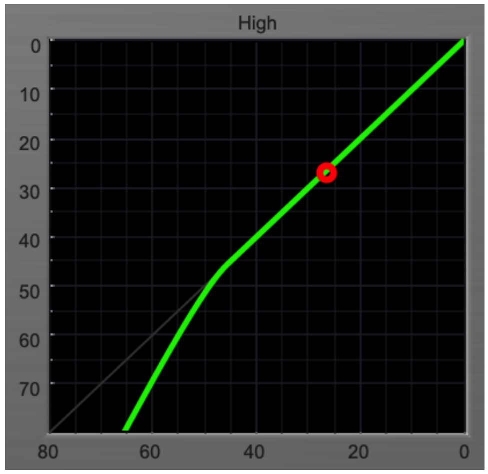


Figure 13.153: Dynamics Knee

Multiband Expander contains a Dynamics Knee diagram for each expander processing section. The diagram provides feedback on the response of the associated dynamics processor. Both the horizontal and vertical axes are calibrated in dBFS. The horizontal axis corresponds to the input level and the vertical axis represents the output level. The heavy line shows the dynamic of the associated processing block. This means that if you sent in a sine wave at a given input level, the output level would be equal to the level shown on the graph. When the processor is working with real dynamic signals, the graph is a good approximation of the response when the attack is fast and the release is slow.

In most cases, however, the dynamic response of the processor will not match its static response. In order to represent this, we have included a “bouncing ball” meter for each expansion band. This metering is shown as a red circle that is overlaid on the knee diagram. The red circle is placed so its horizontal position is equal to the instantaneous input level and its vertical level is equal to the instantaneous output level. Examining this meter while you are adjusting the dynamics controls will provide you with a great deal of information about how the processor is operating and how the controls interact.

Processing

A Detailed Description

In this chapter we'll discuss what each processing block does and how the controls work.

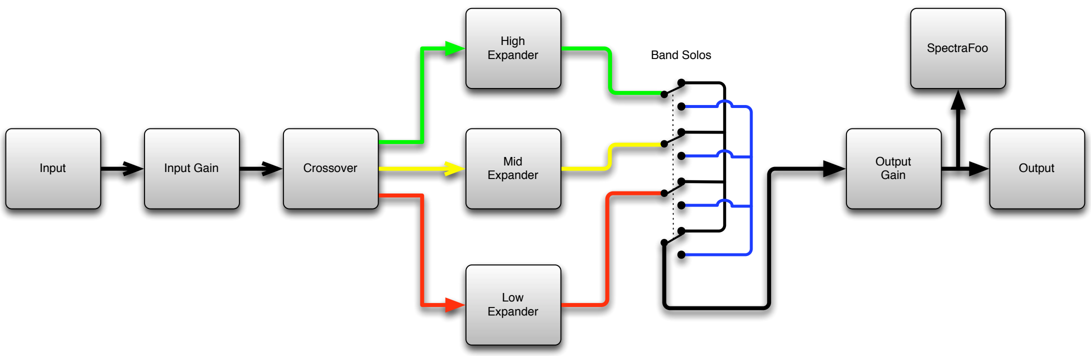


Figure 13.154: Multiband Expander Signal Flow

Input Conditioning

After the signal is routed to Multiband Expander it runs through an input gain block that provides input gain of up to +30 dB. You can use this gain to condition signals that are low in level.

This input gain may also be used to pad out signals by up to –30 dB. While you may find this attenuation useful to just bring down the level through the strip simply and quickly, you must realize that this gain is applied after the signal reaches Multiband Expander and will not pad out any clipping that occurs in the A/D converters or in a plug-in that is inserted before Multiband Expander.

The input gain is controlled by the “In Gain” knob.

Crossover

The crossover seperates the audio input into low, mid and high bands; each of these bands is then fed through a compressor section. There are two controls to set the crossover points:

Figure 13.155: Crossover Controls

The control marked “LXover” sets the frequency of the low to mid crossover, and the control marked “HXover” sets the frequency of the mid to high crossover.

You may also adjust the crossover points in the dynamic transfer function:

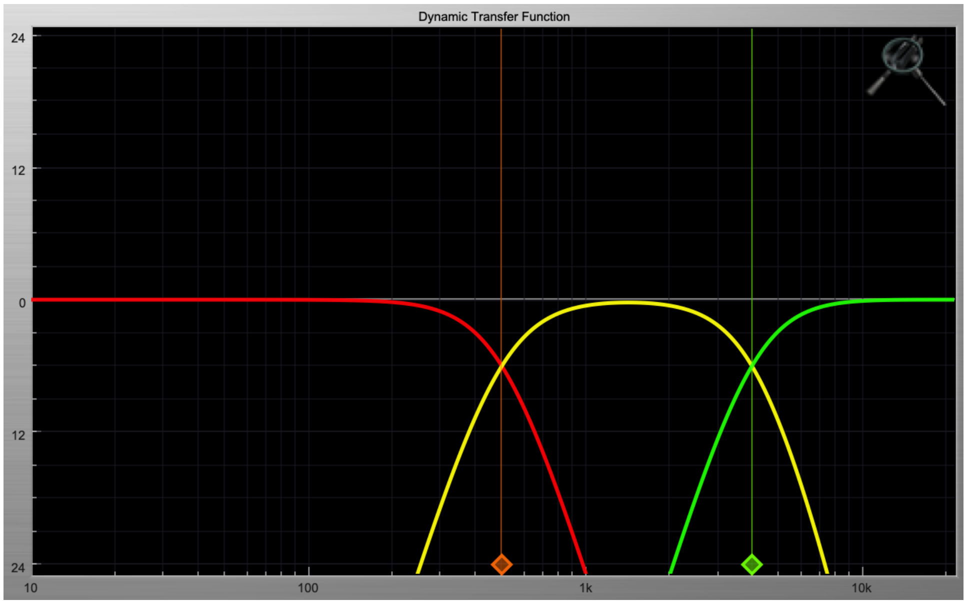


Figure 13.156: Dynamic Transfer Function

Simply grab one of the handles at the bottom of the display and drag to desired frequency. The orange handle adjusts the low/mid crossover point and the green handle adjusts the mid/high crossover point.

Expander

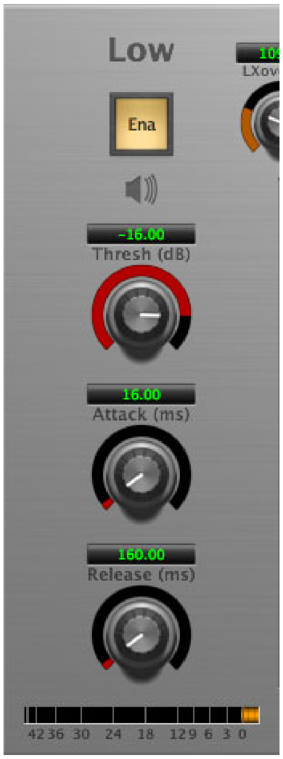


Figure 13.157: Expander (Low Frequency Shown)

Introduction

Multiband Expander contains three separate expanders, one for each crossover band. They are identical, so we will only describe their operation once.

Expander Enable

When this button is off, the expander section will not change the signal.

Band Solo Button

This button (the speaker icon next to the Enable button) allows you to monitor the selected expander's output. This will allow you to listen one or more bands of compression in isolation. When you are done listening to the soloed band(s), click this button again to hear Multiband Expander' normal output.

Threshold Control

The “Thresh” knob controls the level at which the expander begins to reduce the gain applied to the signal. When the detector level is below the threshold level, the gain reduction is applied. As the detector level decreases below the threshold level, the gain is reduced as indicated by the knee diagram associated with the expander.

The expander threshold level is also indicated by the red bar above the gate input meter. You can adjust the threshold level using this indicator as well as by using the “Thresh” knob.

Attack Control

The “Attack” knob allows you to adjust how quickly the gain reduction is reduced when the detector level goes above the threshold level. This control is calibrated in milliseconds and values range from 0 to 500 ms, and also includes an Auto mode. Increasing the attack time causes the gate to open more slowly, and can have the effect of cutting off the initial transients. This may be something that you want for creative effect, but if you are using the expander to simply cut down on bleed, you will probably want to choose the Auto setting, which adaptively removes the gain reduction when incoming transients are detected.

Release Control

The “Release” knob controls the release time of the expander. This knob is calibrated in milliseconds and can range from 5 ms to 5 sec. The release time controls how quickly the gain reduction is applied when the detector drops below the threshold value. For settings below 40 ms or so the expander releases pretty abruptly and may introduce unwanted artifacts into your audio, depending on the signal. In addition, be careful making the release time faster than the attack time.

Precision DeEsser

Plug-in categories: **Production**, **Dynamics** and **Mastering**

Introduction

The Precision DeEsser is a DSP plug-in for MIOConsole3d which provides a processor to tame excessive sibilance in vocal tracks.

Features include:

- Advanced process metering
- Scalable gain reduction meter
- Precision filter
- Flexible dynamics processor
- SpectraFoo™ spectrum analysis

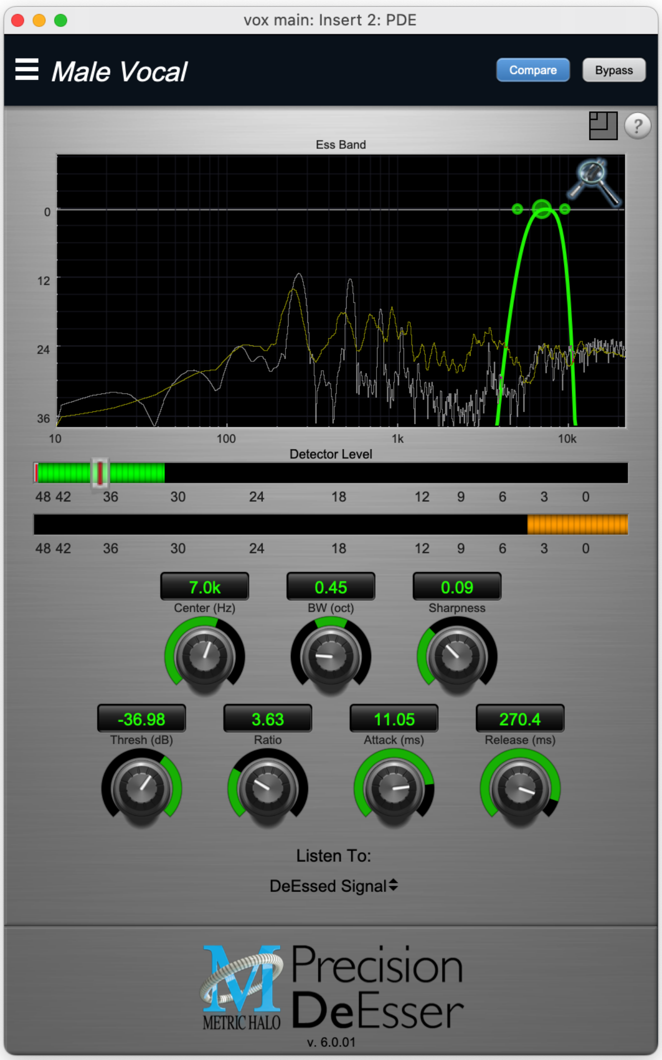


Figure 13.158: Precision DeEsser's User Interface

Operation

The Precision DeEsser user interface uses a few different control elements to control all of the processing. These elements are:

Control Knob

Control Knobs are used to control the value of various continuous parameters of a process. Examples of these types of parameters include: Attack time, Release Time, Threshold, etc. There are two styles of encoders:

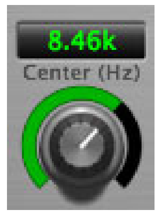


Figure 13.159: Swept Knob

The rings around these encoders sweep from a minimum to maximum value, from left to right. One exception is the threshold control, which sweeps from right to left.

Figure 13.160: Spread Knob

The ring around this encoder starts at 12 o'clock and spreads to both sides equally as the control is increased. This knob is used for the bandwidth control.

You can change the value of each knob in a number of different ways.

- Click and drag the knob to change the value continuously.
- Dragging up or to the right will increase the value, while dragging down or to the left will decrease the value.

  Hint: If you hold down the ⌘ (Command) key when you click, you will be able to adjust the value with finer precision.
- ⌥ (Option)-click to reset any knob to its default value. You may also double-click a knob to reset it.
- **Rapid parameter entry**
  - Double-click the numeric readout of the knob to type in a number directly. The text-entry field will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, or **⌘.** (Command + .) keys.
  - Hit **return** or **enter** to confirm the value and dismiss the pop-up.
  - Hit the **tab** key to confirm the value and immediately activate the entry field for the next control. **⇧-tab** (Shift + tab) will display the entry field for the previous control).
  - Hit **⌘.** (Command + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.
  - When you enter a number into the pop-up entry, you can use a couple of abbreviations: "k" multiplies the number by 1000 and "m" divides the number by 1000. So if you want to enter 16,500 Hz you can just type 16.5k.

Listen Control

Figure 13.161: Listen Control

This button selects what part of the audio signal you are monitoring:

- *DeEssed Signal*: The final output of the plug-in after processing
- *Esses*: The output of the compressor, without the non-processed audio
- *Detector Signal*: This is the post-filter, pre-compressor audio
- *Out-of-Band Signal*: The Detector signal subtracted from the input; this allows you to hear audio that is *not* being processed
- *Removed Material*: The Esses signal subtracted from the Detector signal; this allows you to hear only the audio that is being removed

DeEsser Detector Level Meter

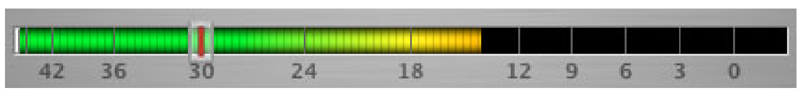


Figure 13.162: DeEsser Detector Level Meter

The Precision DeEsser provides a meter displaying the level of the detector (derived from the from the Ess-band) when the De-Esser is active. The processor threshold is indicated by the red threshold slider above the detector meter. This red bar can be manipulated directly with the mouse. While this meter should never clip (the signal would have to be exceedingly high), the top segment of the meter (above 0dB) is used as a clip indicator and is illuminated red if the input section of the processor detects an over. The clip light remains illuminated until you click on the meter.

⌥ (Option)–click the meter to reset the clip light.

Gain Reduction Meter

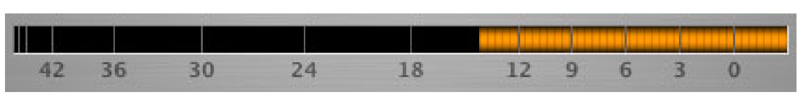


Figure 13.163: Gain Reduction Meter

The gain reduction meter, which has an orange bar and grows down from 0 dB, shows the amount of attenuation being applied by the dynamics processor at any given time. If you right-click or ^ (Control) click on the meter, you may set the scale of the gain reduction meter to any of the following values:

- 54 dB
- 24 db
- 12 db
- 6 db
- 3 db

Ess Band Transfer Function

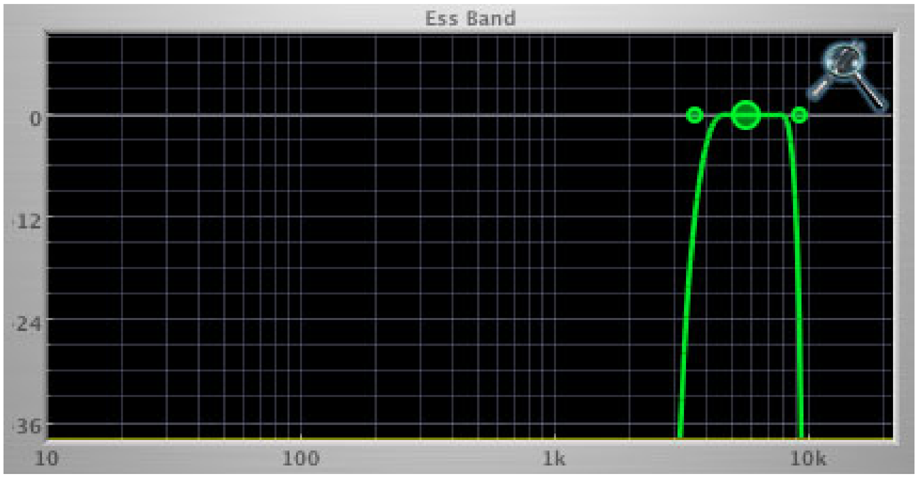


Figure 13.164: Ess Band Transfer Function

The ess band transfer function is a combination of a visual representation of how the de-esser is processing the signal and an intuitive controller for the associated filter. This display is sometimes called a "Cartesian Graph" by other manufacturers.

The horizontal axis provides frequency calibration in Hertz (Hz). The vertical axis provides level calibration in decibels (dBr). The heavy green line indicates the shape of the filter. The center frequency is represented by a large dot in the transfer function with two smaller dots flanking it.

Clicking on the large dot and dragging will allow you to adjust the center frequency. ⌥ (Option)–click the dot to adjust the bandwidth (dragging right increases the bandwidth, left decreases the bandwidth). Click and drag the smaller dots associated with a larger dot to adjust the filter bandwidth.

To dismiss the filter curve, click anywhere in the black area of the transfer function. This will deselect the filter point, and the only trace displayed will be the green master curve.

If you right-click or ^ (Control) click on the transfer function, you will see a menu to set the vertical dB scale for the display. The values are:

- ±3 dB
- ±6 dB
- ±12 dB
- ±24 dB
- ±36 dB

Spectragraph Analyzer

Clicking the SpectraFoo™ logo in the upper right hand corner of the transfer function will activate the spectragraph, showing the realtime frequency analysis of your signal:

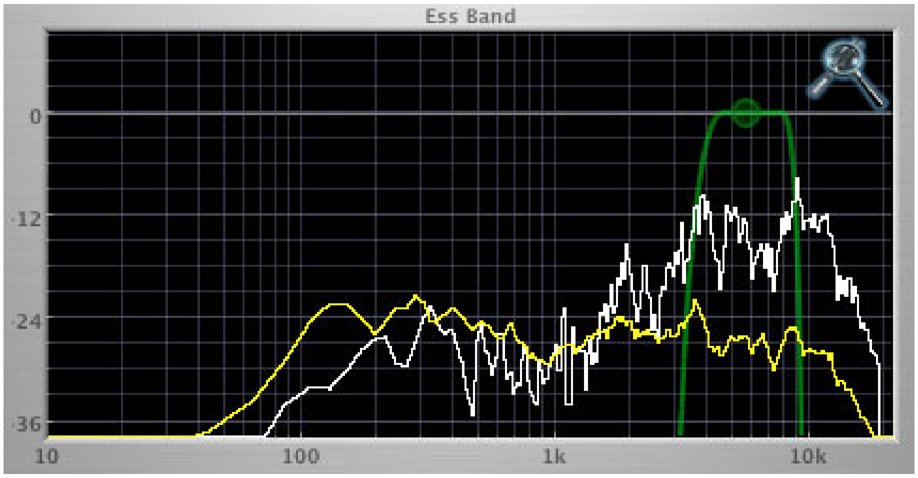


Figure 13.165: Spectragraph Display

The traces are:

- White: Instantaneous display
- Yellow: Average display

The *instantaneous* trace updates in real-time, allowing you to see the immediate peak level of your audio. The *average* trace displays the level as averaged over a short period, giving you a more general view.

The spectragraph analyses the signal currently being audition, as set by the "Listen To:" button. To disable the spectragraph entirely, click the active 'Foo icon.

If you right-click or ^ (Control) click on the transfer function, you will see a menu to set options for the spectragraph:

- Show Instantaneous Trace: Toggles whether the spectragraph shows the instant response of your audio.
- Show Average Trace: Toggles whether the spectragraph shows the averaged response of your audio.

These settings are stored for each instance of the Precision DeEsser.

Processing

A Detailed Description

In this chapter we'll discuss how the controls work.

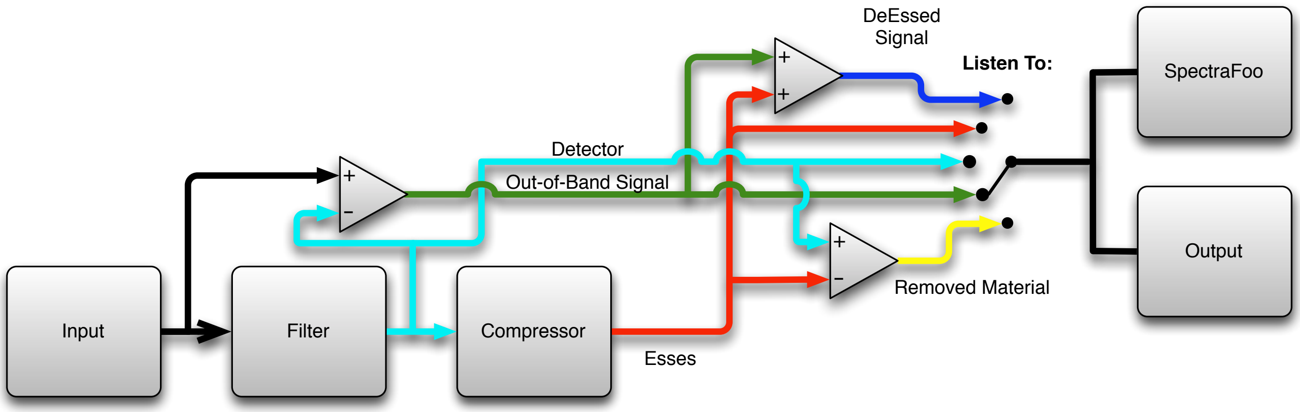


Figure 13.166: Signal Flow

The block diagram above illustrates the overall structure of the processing system provided by the Precision DeEsser. The diagram does not indicate the metering blocks.

Now lets examine the various processing blocks indicated in the diagram.

Ess Band Filter

When the signal is routed to the Precision DeEsser it runs through a filter block that allows you to define the frequency area to be processed.

Figure 13.167: Ess Band Filter Controls

The filter controls are:

- *Center*: Sets the center frequency of the ess band filter.
- *BW (Oct)*: Sets the bandwidth of the ess band filter. Higher values widen the filter.
- *Sharpness*: Sets the slope of the filter

To use the filter, set the Center parameter as close as possible to the undesired frequency. Use the Band-Width control to set the overall width of the filter; a narrow bandwidth will let less audio through to the compressor, while large bandwidths will process more overall. By using the Sharpness control you may shape the slope of the filter, to adjust the transition between the processed and non-processed audio.

Compressor

The next processing block is the compressor. The compressor is used to reduce the level of the audio within the filter boundary once it exceeds the threshold.

Figure 13.168: Compressor Controls

Threshold Control

The “Thresh” knob controls the level at which the de-esser begins to process audio. When the detector level is below the threshold level no processing takes place. When the detector level is above the threshold level, the gain is reduced at a ratio set by the Ratio control.

The threshold level is also indicated by the red bar above the input meter. You can adjust the threshold level using this indicator as well as by using the “Thresh” knob.

Ratio Control

The “Ratio” knob controls the ratio used to compute the gain reduction of the compressor.

Attack Control

The “Attack” knob allows you to adjust how quickly the gain reduction is decreased to 0 dB when the detector level goes below the threshold level. The maximum value is 100 milliseconds.

Release Control

The “Release” knob controls the release time of the compressor. This parameter is measured in milliseconds and can range from 10 ms to 1000 ms. The release time controls how quickly the compressor closes after the detector drops below the threshold value.

Thump

Plug-in categories: **Production** and **Synth Effect**

Introduction

Thump synthesizes bass notes from the input signal, allowing you to generate anything from simple drum reinforcement to synthetic bass lines.

The AU/VST/AAX version for your DAW is avaliable as a free download from the Metric Halo website.

Figure 13.169: Thump's User Interface

Operation

As with most plug-ins, Thump provides many copies of controls that are all operated in a similar manner. The Thump user interface uses a few different control elements to control all of the processing. These elements are:

Control Knob

Control Knobs are used to control the value of various continuous parameters of a process. Examples of these types of parameters include: Attack Frequency, Envelope Attack, Out Gain, etc.

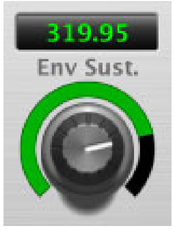


Figure 13.170: Swept Knob

The rings around these encoders sweep from a minimum to maximum value, from left to right.

You can change the value of each knob in a number of different ways.

- Click and drag the knob to change the value continuously.
- Dragging up or to the right will increase the value, while dragging down or to the left will decrease the value.

  Hint: If you hold down the ⌘ (Command) key when you click, you will be able to adjust the value with finer precision.
- ⌥ (Option)-click to reset any knob to its default value. You may also double-click a knob to reset it.
- **Rapid parameter entry**
  - Double-click the numeric readout of the knob to type in a number directly. The text-entry field will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, or **⌘.** (Command + .) keys.
  - Hit **return** or **enter** to confirm the value and dismiss the pop-up.
  - Hit the **tab** key to confirm the value and immediately activate the entry field for the next control. **⇧-tab** (Shift + tab) will display the entry field for the previous control).
  - Hit **⌘.** (Command + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.
  - When you enter a number into the pop-up entry, you can use a couple of abbreviations: "k" multiplies the number by 1000 and "m" divides the number by 1000. So if you want to enter 16,500 Hz you can just type 16.5k.

Toggle Button

Figure 13.171: Toggle Button (Off)

Figure 13.172: Toggle Button (On)

Toggle buttons are simple on/off switches. They light up when they are on and are dark when they are off. You toggle the state of the button by clicking on it. These buttons are used to enable the oscillators.

Fader

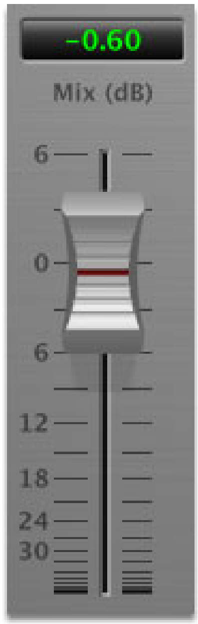


Figure 13.173: Mix Fader

The faders are used to control the output gain of the oscillators. This allows you to set the mix level of each tone.

Oscillator Pitch History

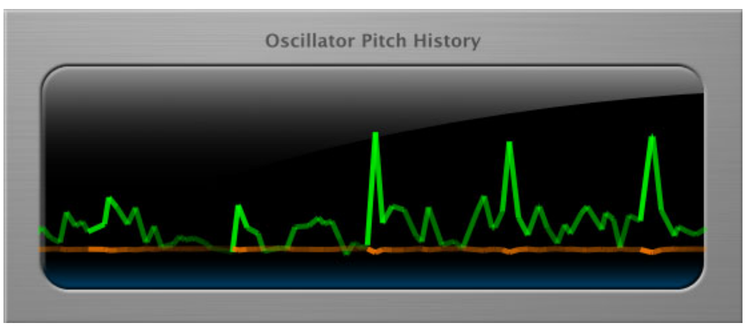


Figure 13.174: Pitch History Display

This display shows the activity of the two oscillators; Oscillator 1 is shown in green, Oscillator 2 in orange. As the display scrolls from right to left, each trace displays the oscillator output. As the oscillator pitch goes up in frequency, the trace goes higher in the window. As the oscillator output gets louder, the trace gets brighter.

Output Meter

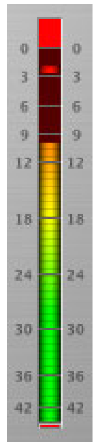


Figure 13.175: Output Meter

For the main output stage of Thump we have provided meters driven with SpectraFoo™ metering technology. These meters show, in addition to the peak metering provided for the input stages, RMS level and VU level. The peak level is represented by the floating colored bar, the RMS level by the solid colored bar and the VU level by the overlaid gray bar. Both the Peak and RMS level are represented with fast PPM ballistics. The VU meter shows IEEE standard 300 ms RMS average level. When Thump is on a mono insert there will be a single meter. When Thump is running in stereo mode the left meter shows the left channel output level and the right meter shows the right channel output level.

The output section clip lights activate if there is an over in the output stage or in any of the processing section input stages. It is reset by clicking on the meter; ⌥ (Option)–click to reset the clip lights on all the meters.

> **A Note About Clipping Indicators:**
> The clip lights do not mean that the plug-in is clipping; it means that the audio level in the DSP is currently over 0 dBFS. If you do not lower the signal level you run a chance of actually clipping the input of another processor or D/A convertor.

Click the Logo…

Clicking on the Thump logo will bring you to Metric Halo's web site where you can learn more about the Production Bundle and other MH audio products.

Processing

A Detailed Description

In this chapter we'll discuss what each processing block does and how the controls work.

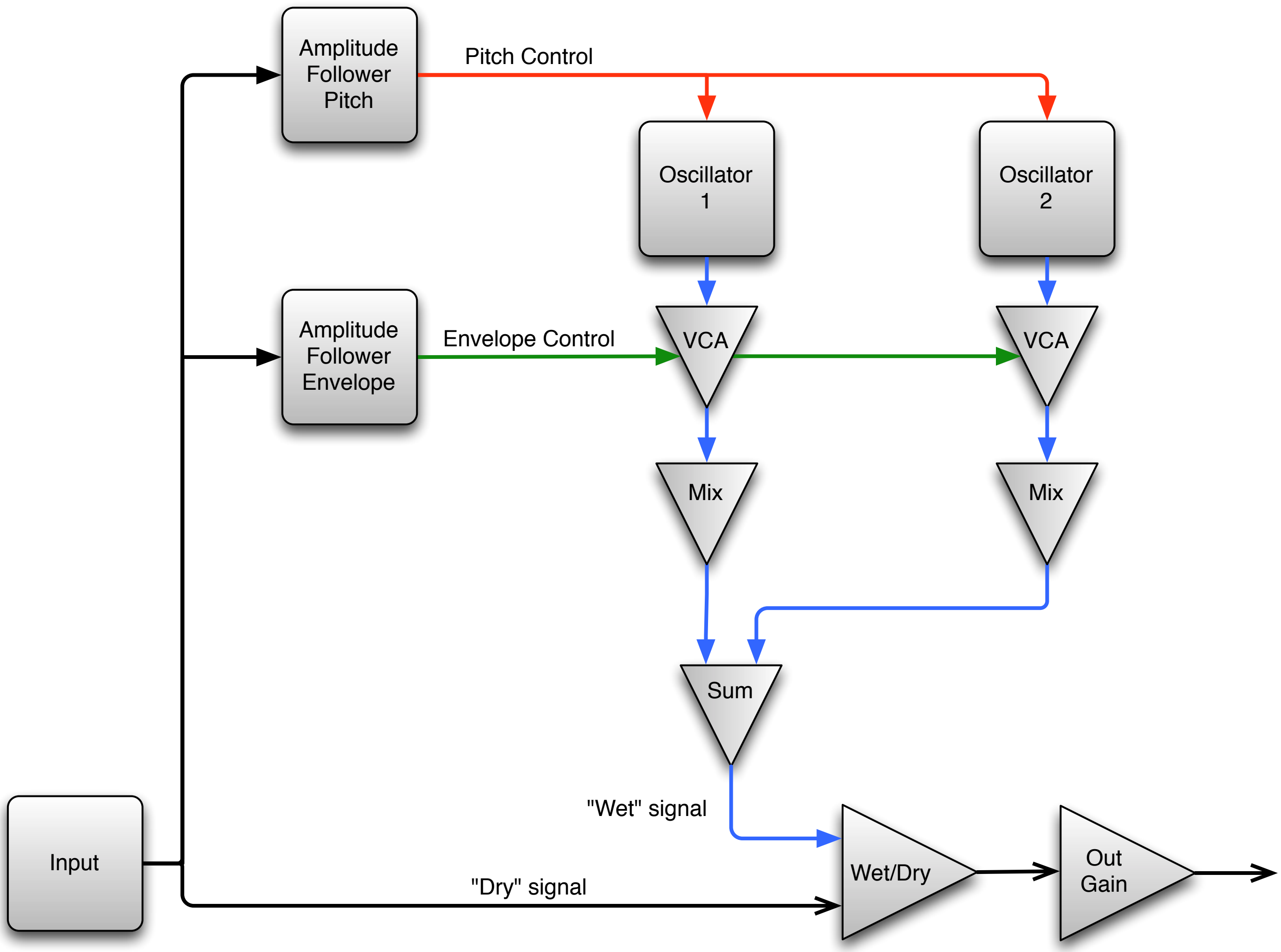


Figure 13.176: Thump Signal Flow

The block diagram above illustrates the overall structure of the processing system provided by Thump. The diagram does not indicate the various metering blocks.

Now let's examine the various processing blocks indicated in the diagram.

Amplitude Followers

Thump's Amplitude Followers convert the audio input into control signals for the oscillators. The Envelope Amplitude Follower extracts an envelope that is used to control the amplitude of the output of the oscillator. The Pitch Amplitude Follower extracts an envelope that is used to control the pitch of the oscillators.

Each Amplitude Follower functions as an envelope detector that is driven by the input audio signal. The Amplitude Follower extracts the amplitude envelope from the signal. Each Amplitude Follower provides the following controls that you can use to control its behavior:

- The *Envelope Atk.* and *Pitch Atk.* controls set the attack time constant in milliseconds (ms) of their respective Amplitude followers. The attack time constant controls how quickly the envelope output increases when the level of the input signal is higher than than the envelope output. Smaller values cause the envelope to follow the signal more closely, but values that are too small could lead to distortion, depending on the characteristics of the input signal.
- The *Envelope Sust.* and *Pitch Sust.* controls set the release time constant in milliseconds (ms) of their respective Amplitude followers. The release time constant controls how quickly the envelope output decreases when the level of the input signal is lower than than the envelope output. Smaller values cause the envelope to follow the signal more closely, but values that are too small could lead to distortion, depending on the characteristics of the input signal.

Oscillators

The envelope outputs from the Amplitude Followers are used to control the two oscillators that make up Thump's synthesis section. Each oscillator has two controls to allow you to adjust how the frequency of the oscillator changes with the control input from the Pitch Amplitude Follower:

- *Atk. Frequency*: This control sets the frequency of the oscillator when the Pitch Amplitude Follower output is at its maximum value (e.g. the input signal is at full-scale). This control ranges from 1 to 440 Hz.
- *Sust. Frequency*: This control sets the frequency of the oscillator when the Pitch Amplitude Follower output is at its minimum value (e.g. the input signal is silent). This control ranges from 1 to 440 Hz.

As the output of the Pitch Amplitude Follower swings from its minimum value to its maximum value, the oscillator's frequency will swing from the value set by the *Sust. Freq.* to the value set by the *Atk. Freq.* Each of the controls can take any value in the range, so that the oscillator frequency can be decreasing as the signal decays away, increasing as the signal decays away or even constant (if both controls are set to the same value).

The traditional decaying pitch drum sound can be made by setting the attack frequency higher than the sustain frequency. The frequency characteristics of each oscillator can be set independently.

The output of the Envelope Amplitude Follower is used to control the amplitude envelope of the output of the oscillators. The overall amplitude of each oscillator is independently controlled by its associated mix parameter. By adjusting the *Env Atk.* and *Env Sust.* parameters you can control how audible the attack frequency sweep is, and how long the decay of the sound will continue to be audible.

The oscillators have two master controls:

- *Enable*: This button turns each oscillator on and off.
- *Mix*: this fade sets the output level of each oscillator, from -∞ to +6 dB.

Output

Thump's output section has two controls:

- *Wet/Dry Mix*: This controls the balance between the original signal and synthesized audio. 0% is full dry (no effect), 50% is equal balance and 100% is full effect (no original audio). If you are using Thump on an aux bus, you would traditionally set the Wet/Dry Mix to 100.
- *Out Gain*: This sets the final output level of Thump, and ranges from -24 to +24 dB.

TransientControl

Plug-in categories: **Production**, **Dynamics** and **Mastering**

Introduction

TransientControl is a DSP plug-in for MIOConsole3d which provides a unique dynamic shaping tool.

Figure 13.177: TransientControl's User Interface

TransientControl looks at two components of audio:

- Transient: The audio's "attack". This is the pick of a guitar or bass, hit of a snare drum, etc.
- Sustain: This is the part of the audio around the transient.

TransientControl uses envelope detectors to separate what is transient and what is sustain. Once these elements are separated, TransientControl can manipulate them to boost or cut the desired section of audio.

Operation

The TransientControl user interface uses a few different control elements to control its processing. These elements are:

Control Knob

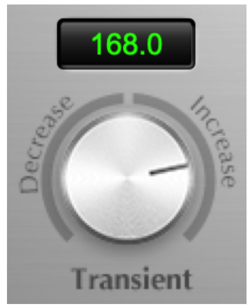


Figure 13.178: Control Knob

Control Knobs are used to control the value of various continuous parameters of a process. Examples of these types of parameters include: Transient, Sustain, Gain, etc.

You can change the value of each knob in a number of different ways.

- Click and drag the knob to change the value continuously.
- Dragging up or to the right will increase the value, while dragging down or to the left will decrease the value.

  Hint: If you hold down the ⌘ (Command) key when you click, you will be able to adjust the value with finer precision.
- ⌥ (Option)-click to reset any knob to its default value. You may also double-click a knob to reset it.
- **Rapid parameter entry**
  - Double-click the numeric readout of the knob to type in a number directly. The text-entry field will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, or ⌘. (Command + .) keys.
  - Hit **return** or **enter** to confirm the value and dismiss the pop-up.
  - Hit the **tab** key to confirm the value and immediately activate the entry field for the next control. **⇧-tab** (Shift + tab) will display the entry field for the previous control).
  - Hit ⌘. (Command + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.

Output Meter

Figure 13.179: Output Meter

For the main output stage of TransientControl we have provided meters driven with SpectraFoo metering technology. These meters show, in addition to the peak metering provided for the input stages, RMS level and VU level. The peak level is represented by the floating colored bar, the RMS level by the solid colored bar and the VU level by the overlaid gray bar. Both the Peak and RMS level are represented with fast PPM ballistics. The VU meter shows IEEE standard 300 ms RMS average level. When TransientControl is on a mono insert there will be a single meter. When TransientControl is running in stereo mode the top meter shows the left channel output level and the bottom meter shows the right channel output level.

The output section clip lights activate if there is an over in the output stage or in any of the processing section input stages. It is reset by clicking on the meter. ⌥ (Option)–click to reset the clip lights on all the meters.

UI Mode Button

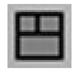


Figure 13.180: UI Mode Button

The three-segment button icon just below the header hamburger menu (which looks suspiciously like a miniature of the TransientControl interface itself) brings the three sections of TransientControl's user interface into view as you require them. Clicking this icon (shown magnified below) toggles through the three UI modes:

Figure 13.181: TransientControl UI Mode "zones"

- Basic: The primary controls for *Transient*, *Sustain* and *Gain* are always visible along with the *Output Meter*. This is the default UI mode, and the most convenient for quickly tightening or taming your target transients.
- Basic with Process Metering: From the Basic setting, clicking the UI Mode button once adds the *Process Meter* display for a visual reference of your processed signal.
- Advanced with Process Metering: One more click selects the third UI Mode, which adds access to the *Advanced Envelope Controls* drawer.

Clicking the UI Mode button while all interface segments are open will cycle you back to the Basic control mode.

Process Meter

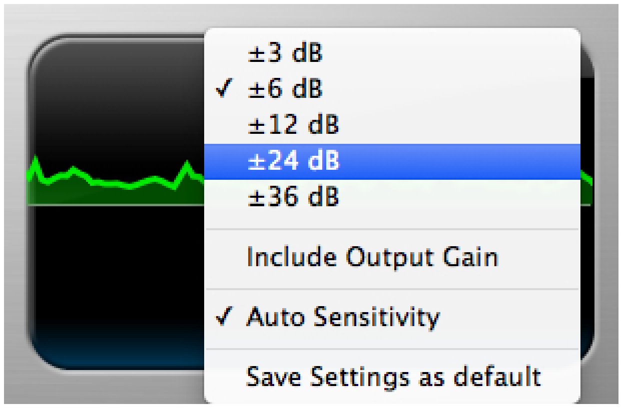


Figure 13.182: Process Meter with Settings Menu

The Process Meter shows how TransientControl is modifying the audio signal. Activity above the horizon of the meter shows the amount of gain increase from the *Transient* and *Sustain* adjustments, while activity below the horizon shows the amount of gain reduction.

Click on the Process Meter display to reveal the settings menu (shown above).

The default scale of the display is ±24dB, with five scale options from ±3dB to ±36dB.

Selecting *Include Output Gain* in this menu will let you configure whether the gain adjustment from the Gain control is factored into the meter. This provides a handy check towards maintaining headroom going into the next stage in your signal path.

The "Auto Sensitivity" parameter can be set individually on each instance of the plugin. It is enabled by default when you create a new instance of Transient Control, and enables a detector that will automatically sense lower level input signals and apply the plugin processing as if the input were at higher levels. This is especially valuable for program that varies a lot in level, as it will help smooth the overall response.

Lastly, "Save Settings as default" allows you to save your current meter display preferences as your default for new instances of the processor.

Basic Controls

The Basic UI mode gives you access to the most often used controls:

- **Transient**: Controls the gain applied to the transient portion of the signal. Adjust this parameter to boost or cut the transient “spike” of your signal. For example, Transient boost can bring out the pick attack in a bass line.
- **Sustain**: Controls the gain applied to the sustain portion of the signal. Adjust this parameter to boost or cut the audio material around the transient “spike” of your signal. For example, Sustain boost can round out the sound of an acoustic guitar.
- **Gain**: Master output gain in dB. Use this to set the output level after setting the Transient and Sustain controls.

Advanced Controls

The Advanced UI mode gives you access to the controls used to fine tune the transient and sustain envelope detectors.

You will not usually need to adjust these parameters, but may find it necessary on challenging material. You can also change these settings to create special effects.

Processing

A Detailed Description

In this chapter we discuss what each processing block does and how the controls work.

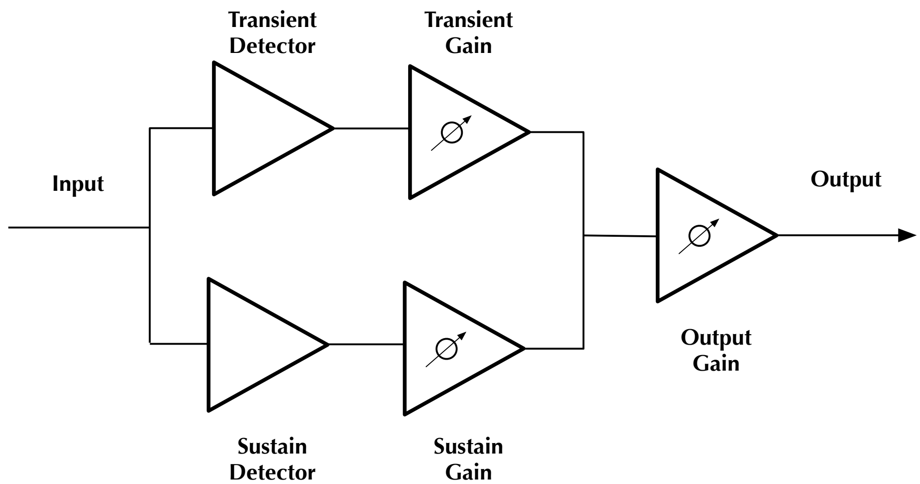


Figure 13.183: TransientControl Block Diagram

The block diagram above illustrates the overall structure of the processing system provided by Transient-Control. This diagram does not indicate the various metering blocks.

Now lets examine the various processing blocks indicated in the diagram.

Transient Detector

The transient detector separates the transient component of the audio from the surrounding sustain. The detector envelope is tuned using the following controls, which are available in the Advanced UI view:

- **Fast Attack**: Sets the fast attack time of the transient detector, from 0 to 100ms.
- **Slow Attack**: Sets the slow attack of the transient detector, from 0 to 100ms.
- **Release**: Sets the release of the transient detector, from 0 to 1000ms.

The Fast Attack and Slow Attack controls define the threshold that the audio must exceed to be considered "transient", while the Release control sets how long the envelope stays open.

You generally will not need to change these controls from the default values.

Transient Gain

This control allows you to boost or cut the level of the audio that falls within the transient detector's envelope. The gain range is ±300%

Sustain Detector

The sustain detector separates the sustain component of the audio from the transient "spike". The detector envelope is tuned using the following controls, which are available in the Advanced UI view:

- **Attack**: Sets the attack of the sustain detector, from 0 to 100ms.
- **Fast Release**: Sets the fast release of the sustain detector, from 0 to 1000ms.
- **Slow Release**: Sets the slow release of the sustain detector, from 0 to 1000ms.

The Fast Release and Slow Release controls define the thresholds that the audio must be within to be considered "sustain", while the Attack control sets how long the envelope stays open.

The transient detector algorithms and default settings have been painstakingly engineered to provide the most consistent and musical results across the widest possible range of audio program. Generally you will not need to change these controls from the default values. That said, these controls are included for those rare occasions where that little extra soupçon of precision is required.

Sustain Gain

This control allows you to boost or cut the level of the audio that falls within the sustain detector's envelope. The gain range is ±300%.

Output Gain

This control allows you to boost or cut the output level after all processing has been applied. The gain range is ±24 dB.

Auto Sensitivity

This control is set via the pop menu in the Process Meter. When Auto Sensitivity is turned on, the transient and sustain detectors become level independent and you will be able to apply the processing across a broader range of signals. Auto sensitivity is turned on by default.

Metric Halo 3d Exclusives

(…watch this space…)

The MH 3d "Exclusives" category consists of processors that can be found nowhere else but within the Metric Halo 3d shared-DSP environment.

Candidates for the "Exclusives" banner can be anything from unique vintage signal processors in need of resurrection away from the restrictions of host-based DAWs, to cutting-edge experiments expanding the capabilities of audio signal processing as we know it.

3d Exclusive processors can be found listed in the mixer Plug-in: *Production* menu category, and by each processors' functional subcategory (EQ, Dynamics, etc.).

Sonic Eq (Metric Halo 3d Edition)

Plug-in categories: **Production**, **EQ**, and **Mastering**

Introduction

Sonic EQ 3d has six independent minimum-phase filter bands. Each band offers your choice of 8 filter types and 5 function-specific equalization topologies. The plug-in can be used as a mono, stereo or multichannel instantiation.

Processing functions include:

- Original Sonic Solutions SonicSystem™ filter coefficients
- Per-band adjustable filter slope order
- Passband Ripple and Stopband Ripple resonant filter controls (2nd through 4th order)
- Updated and improved SpectraFoo™ spectrum analysis

Figure 13.184: Sonic EQ 3d

The Sonic EQ has gone through many iterations since Dr. James A. Moorer designed the original filter coefficient structures for Lucasfilms' SoundDroid project in the '80s, the most recent being the Sonic Studio Mastering EQ of the last decade.

In every generation of Sonic EQ, two primary characteristics have remained consistent:

1. the *SOUND* - a full, sweet yet powerful musicality which somehow brings a presence and clarity to literally anything you pass through it,

2. and it's *versatility*. Whether your need is forensic restoration, sound design, high-end mastering, post-production, recording, re-recording, multi-tracking or live sound, the power and simplicity of this EQ package will get you the sound you are looking for faster and sounding better than you may be willing to believe.

   You will find you can make EQ curves with Sonic EQ 3d that look completely broken, yet still sound perfectly clean and natural (refer to the screenshot on the preceding page).

It is no understatement to say the Sonic EQ has been and still is one of the most highly respected and coveted digital equalizer packages on the planet. Originally packaged exclusively with the Sonic Studio branded Model 302, 303 and 304 Firewire audio interfaces, it has been reborn in 3d with a cleaner and more responsive layout, Spectrafoo™ spectral and level metering and the higher order of musicality, precision and power inherent to the 3d DSP environment.

Operation

Sonic EQ 3d's control interface employs the simplified "virtual knob" encoder element seen throughout the MIOConsole3d environment.

Control Knobs

Control Knobs are used to control the value of various continuous parameters of a process. There are three styles of encoders used in SonicEQ3d:

Figure 13.185: Swept Knob

Swept Knob: The rings around these encoders sweep from a minimum to maximum value, normally from left to right. Swept knobs are used for Frequency (Hz), Passband Ripple (dB) and Stopband Ripple (dB).

Figure 13.186: Plus/Minus Knob

Plus/Minus Knob: The rings around these encoders start at 12 o'clock and sweep to either side. These knobs are used for Gain (dB) control, where straight up is no gain change, turning to the left cuts the signal and turning to the right boosts it.

Figure 13.187: Spread Knob

Spread Knob: The rings around these encoders start at 12 o'clock and spread to both sides equally as the control is increased. These knobs are used for the "Q" controls.

Control tips

You can change the value of each knob in a number of different ways.

- Click and drag the knob to change the value continuously.
- Dragging up or to the right will increase the value, while dragging down or to the left will decrease the value.

  Hint: If you hold down the ⌘ (Command) key when you click, you will be able to adjust the value with finer precision.
- ⌥ (Option)-click to reset any knob to its default value. You may also double-click a knob to reset it.
- **Rapid parameter entry**
  - Double-click the numeric readout of the knob to type in a number directly. The text-entry field will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, or ⌘. (Command + .) keys.
  - Hit **return** or **enter** to confirm the value and dismiss the pop-up.
  - Hit the **tab** key to confirm the value and immediately activate the entry field for the next control. **⇧-tab** (Shift + tab) will display the entry field for the previous control).
  - Hit ⌘. (Command + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.
  - When you enter a number into the pop-up entry, you can use a couple of abbreviations: "k" multiplies the number by 1000 and "m" divides the number by 1000. So if you want to enter 16,500 Hz you can just type 16.5k.

> **Note:** The variable filter types in Sonic EQ 3d use differing sets of control elements. For example, HiPass and LoPass filters employ active Frequency, Passband and Stopband parameters but no Gain control. First-order parametric filters have Gain, Frequency, and Q, whereas higher-order parametrics add the Passband Ripple control.
>
> Inactive control elements are shown with empty knob placeholders (i.e. no numeric parameter entry field).
>
> The De-emphasis, DC and RIAA filters are dedicated-purpose with no adjustable parameters.

Toggle Button

Figure 13.188: Filter Enable (On)

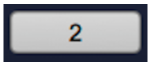


Figure 13.189: Filter Enable (Off)

Toggle buttons are simple on/off switches. They light up (colored) when they are on and are dark (uncolored) when they are off. You toggle the state of the button by clicking on it. These buttons are used to enable and disable filter bands within Sonic EQ 3d and engage "Compare" and "Bypass" in the plug-in header.

Filter Type selector

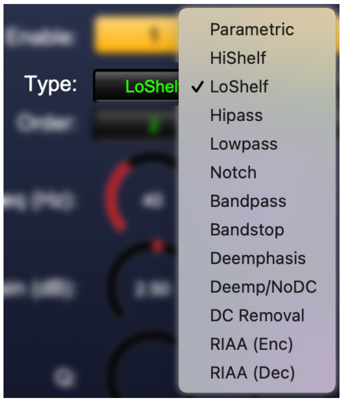


Figure 13.190: Filter Type selector (with submenu)

Below the band on/off toggle is the filter selector. Click this control to bring up the filter selection submenu.

Filter Order selector

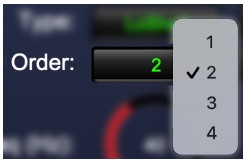


Figure 13.191: Filter Order selector (with submenu)

Click the Order selection menu to select your filter order (1 - 4). Note that 2nd order and higher filters will enable the *Pass Ripple* control.

Fader

Figure 13.192: Master Fader

The fader is used to control the master output gain of the plug-in. It works in much the same fashion as the control knobs, but instead of dragging up/right or down/left to change the value, you directly drag the fader knob. The other "tricks" described for the knobs (option-click to reset, etc.) also work with the fader.

EQ Transfer Function

Figure 13.193: EQ Transfer Function

The EQ transfer function is a combination of a visual representation of how the EQ is processing the signal and an intuitive controller for the associated filter bands. This display is sometimes called a “Cartesian Graph” by other EQ manufacturers.

The horizontal axis provides frequency calibration in Hertz (Hz). The vertical axis provides level calibration in decibels (dBr). The heavy green line indicates the relative change in level at each frequency that is created by the combined effects of all of the active filter bands in the equalizer. Each EQ band is represented by a colored dot in the transfer function. The color of the dot matches the color of the knob rings for the corresponding EQ band.

The band that is currently being edited will be displayed along with the overall response curve. If the associated band is a symmetrical filter (parametric) there will also be two smaller colored dots that can be used to control the Q of the filter. Clicking on a large colored dot and dragging will allow you to adjust the frequency and gain of the associated band.

- ⌘ (Command)–click or double-click the dot to toggle the band enable (single-click to disable a band when “Auto Enable Bands” is turned on).
- Click and drag the smaller dots associated with a larger dot to adjust the filter Q. You can also ⌥ (Option)–click the dot to adjust the Q (dragging left increases the Q, right decreases the Q).
- ⌘⌥ (Command + Option)–click the dot to switch the band filter type.

To dismiss the filter curve, click anywhere in the black area of the transfer function. This will deselect the filter point, and the only trace displayed will be the green master curve.

If you right-click or ^ (Control) click on the transfer function, you will see a menu to set the vertical dB scale for the display. The values are:

- ±3 dB
- ±6 dB
- ±12 dB
- ±24 dB
- ±36 dB

This menu also allows you to specify whether adjusting a filter causes it to automatically be enabled. This preference is for all instances of Sonic EQ 3d.

Spectragraph Analyzer

Clicking the SpectraFoo™ logo in the upper right hand corner of the transfer function will activate the spectragraph, showing the realtime frequency analysis of your signal:

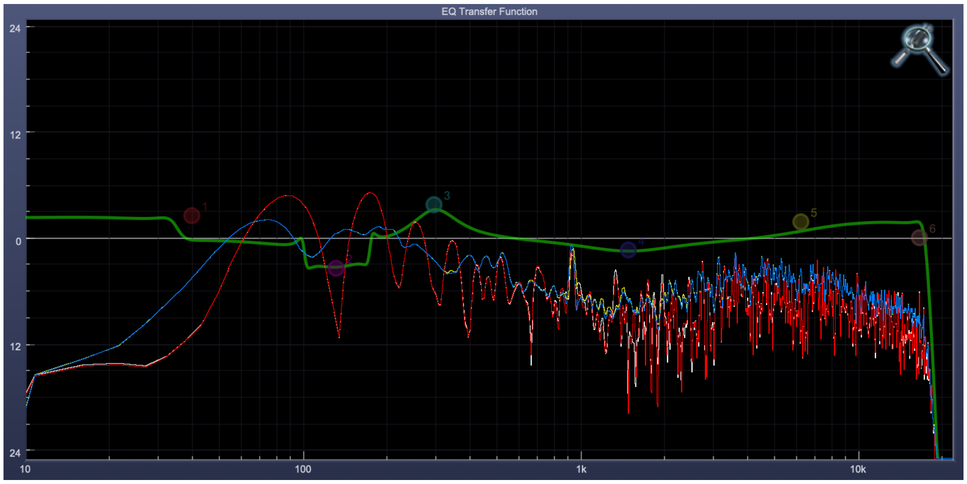


Figure 13.194: Spectragraph Display

The traces are:

- White: Left channel instantaneous display
- Red: Right channel instantaneous display
- Yellow: Left channel average display
- Blue: Right channel average display

The *instantaneous* trace updates in real-time, allowing you to see the immediate peak level of your audio. The *average* trace displays the level as averaged over a short period, giving you a more general view.

The spectragraph analyses the signal post-filter, allowing you to see the effect of your EQ filter(s).

To disable the spectragraph entirely, click the active 'Foo icon.

If you right-click or ^ (Control) click on the transfer function, you will see a menu to set options for the spectragraph:

- Show Instantaneous Trace: Toggles the instant response traces for both channels.
- Show Average Trace: Toggles the averaged response traces for both channels.
- Show Left Channel: Toggles the left channel spectragraph display.
- Show Right Channel: Toggles the right channel spectragraph display.
- Auto Enable Bands when band parameter changes: Automatically enables a filter band when one of its parameters is adjusted.

These settings are stored independently for each instance of Sonic EQ 3d.

Auto Enables

The processing sections of Sonic EQ 3d will automatically enable when one of the EQ Transfer Function parameters are adjusted. For example if the EQ master enable is off, adjusting any EQ parameter will turn it on. This way you will never make "phantom" adjustments where you make adjustments and hear no change.

Note that the Auto-Enable applies only to the filter band enables, and does not affect the master plug-in Bypass state. Auto Enable does not apply to the 6-Band EQ control knobs.

Processing

Sonic EQ 3d filter nomenclature

Sonic EQ filters are described and controlled in terms of Resonant Frequency, Gain, Q (for *Quality Factor*) and filter Order (filter slope). Secondary controls of Passband Ripple and Stopband Ripple are available where appropriate on the 2nd-order and higher filter settings.

Expressing the width of a filter as a 'Quality Factor', rather than bandwidth, provides a more intuitive sense of the filter's subjective "sound", since the same value of Q will produce different bandwidths at different frequencies. The higher the frequency, the wider the bandwidth will be for a given Q value, which roughly corresponds to our auditory mechanism's ability to perceive a filter's action. As an example, a parametric filter with a Q of 1 has a bandwidth of 100 Hz when its center frequency is set to 100 Hz but, it has a bandwidth of 1000 Hz when the center frequency is set to 1000 Hz.

- Filter **Order** menu items 1 - 4 are used to select a 6, 12, 18 or 24dB per octave slope, respectively, for each filter band. Filter orders 2 - 4 activate the Pass Ripple control parameter (detailed below).
- Resonant **Frequency (Hz)** is the 'knee' of a single-stage pass or shelf filter, or the center frequency of a symmetrical filter such as a notch, parametric "bell", band-pass or band-stop. Sonic EQ 3d adjustable Frequency range is 1Hz to 20.1kHz.
- **Gain** is the gain of the filter at the center frequency, expressed in dB. If the value is negative, the filter attenuates (cuts) frequencies within its band; if the value is positive, it boosts them. Sonic EQ 3d Gain range is ±24dB.
- **Q** is a numerical value that describes the filter's bandwidth as a function of its center frequency. Bandwidth is defined as the frequency span between the upper and lower -3dB points in the filter's pass band (i.e. the part of the frequency spectrum being processed by the filter and passed on to the output). The equation relating Q to bandwidth is:

$$BW = \frac{f}{Q}$$

  where BW is bandwidth and f is the center frequency. Sonic EQ 3d Q range is 0.10 (super wide) to 100 (super narrow).

- The **Pass Ripple** (short for Passband Ripple) controls the ripple amplitude inside the filter pass band.

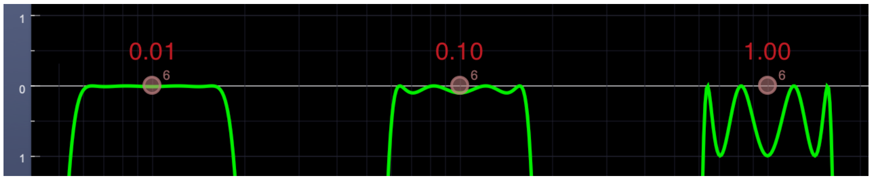


Figure 13.195: Pass Ripple examples

This figure shows the passband ripple of a Bandpass filter at the minimum 0.01dB setting, 0.10db (the default) and 1.00dB.

As you can see, the minimum setting results in barely perceptible ripple, the default setting of 0.10dB equates to a passband ripple amplitude of 0.1dB below unity gain, and 1.00dB of Pass Ripple gives you exactly 1dB of passband ripple. Note, however, that even zoomed in this closely you can still see differences in the filter slope introduced by these relatively small Pass Ripple adjustments. A tiny variation to the passband of the filter means a progressively increasing difference at -10, -30 and on downwards.

The maximum Pass Ripple is 5.98dB, which is pretty much sound effects design territory for audio filters.

- **Stop Ripple** (short for Stopband Ripple) describes the amount of amplitude variation or ripple in a filter's out of band response. Indirectly, it describes two more important parameters. One is out of band suppression or, how much "leakage" of unwanted signal you receive, and the other is phase shift and group delay.

  Stopband ripple is a negative parameter, instructing the filter how much signal is to be removed. The control provides a range of -40dB to -120dB. So if the Stop Ripple is 40dB, that would mean that the attenuation in the stop band will be -40dB *minimum*; it might be more attenuation at some frequencies. However, because the stopband ripple value is so low, the phase response and resultant temporal response of the filter will be less audibly apparent. At the other extreme of its range, stopband ripple will be 120 dB down from the (unity) passband gain but, the phase response will suffer, the group delay will be severe and the resulting temporal smearing may be unacceptable. As with any filter, careful listening will determine the tradeoff between stopband suppression and side effects.

  Please note that, since the EQ Transfer Function is focused to within ±36dB of unity gain, the majority of the stopband activity is outside (below) the range of the graph. Changes to the filter slope as you adjust the Stop Ripple provide a clue as to what is going on, but even more than Pass Ripple, this is very much a "feel" type of parameter.

Filter Types and Parameters

Parametric Filters

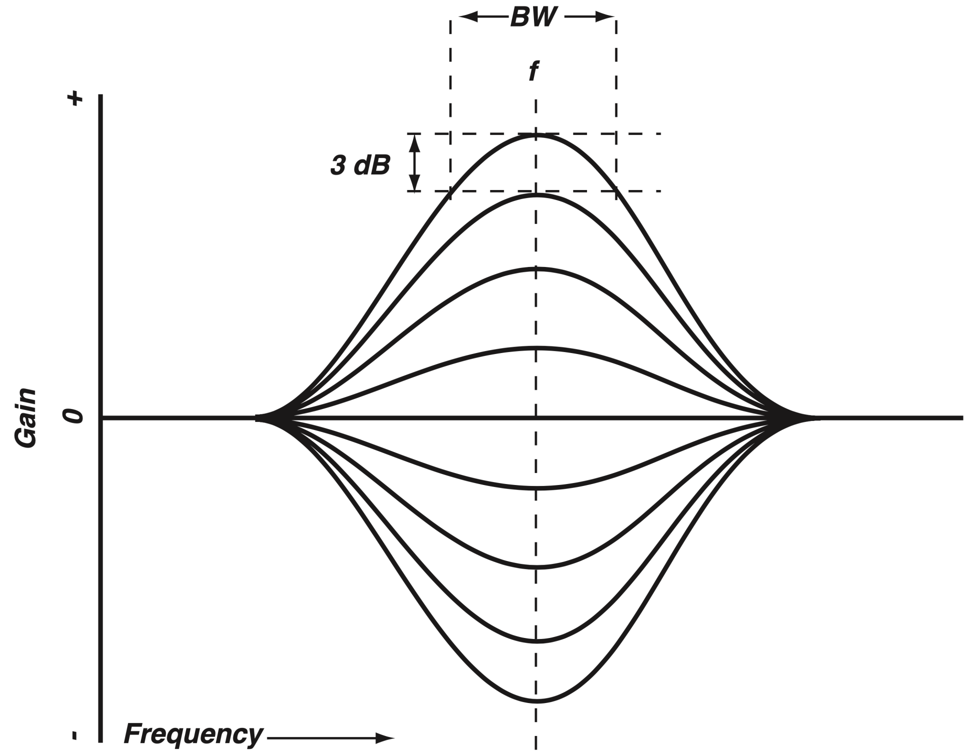


Figure 13.196: Parametric Filter Response Curves

A parametric filter affects a defined region of the audio spectrum.

- **Order** - Selects the filter order. Higher order equals a steeper filter slope:
  1. 6dB/oct (default)
  2. 12dB/oct
  3. 18dB/oct
  4. 24dB/oct
- **Freq (Hz)** - Sets the center frequency of the pass band.
- **Gain (dB)** - Adjusts the gain of the filter at the center frequency.
- **Q** - Adjusts the Q value and, inversely, the filter's bandwidth.
- **Pass Ripple** - Adjusts the ripple amplitude within the passband. Available to 2nd, 3rd & 4th order filters.

Shelving Filters

Shelving filters affect all frequencies above (high shelf) or below (low shelf) a given frequency (called the cutoff frequency).

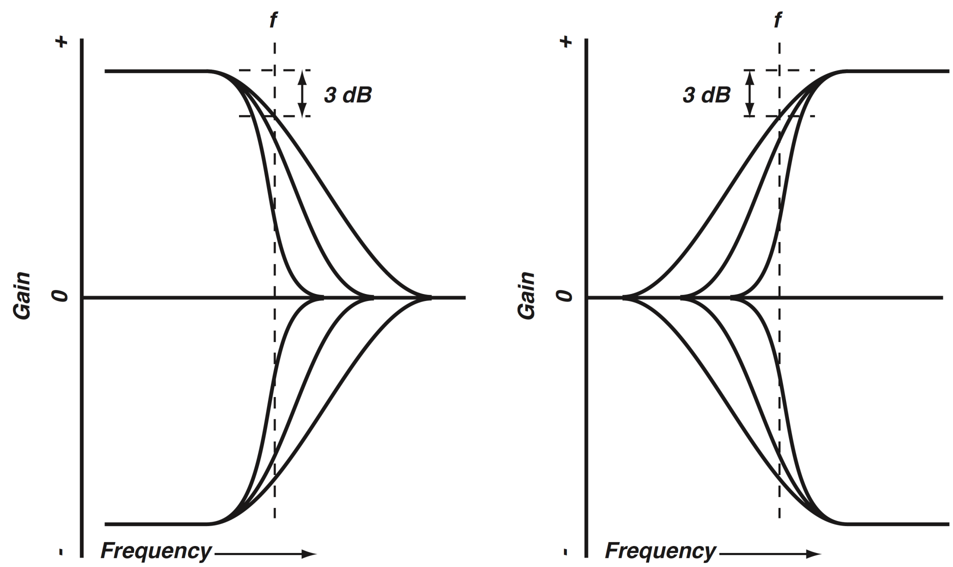


Figure 13.197: LoShelf (left) and HiShelf (right) Filter Response Curves

The above shows the response characteristics of low and high shelving filters at three different orders. The dotted lines indicate the cutoff frequency for the lowest-order (least steep) filter.

- **Order** - Selects the filter order. Higher order equals a steeper filter slope:
  1. 6dB/oct (default)
  2. 12dB/oct
  3. 18dB/oct
  4. 24dB/oct
- **Freq (Hz)** - Sets the cutoff frequency of the filter. Cutoff frequency is defined as the -3 dB point in the filter's response
- **Gain (dB)** - Adjusts is the gain of the filter above (high shelf) or below (low shelf) the cutoff frequency, expressed in dB. If the value is negative, the filter attenuates (cuts) frequencies within the range that it affects; if the value is positive, it boosts them.
- **Pass Ripple** - Adjusts the ripple amplitude within the passband. Available to 2nd, 3rd & 4th order filters.

Low Pass and High Pass Filters

Low and high pass filters affect all frequencies above (low pass) or below (high pass) a given frequency (called the cutoff frequency). Low and high pass filters are cut only: their purpose is to attenuate (or eliminate) spectral energy above or below a given frequency.

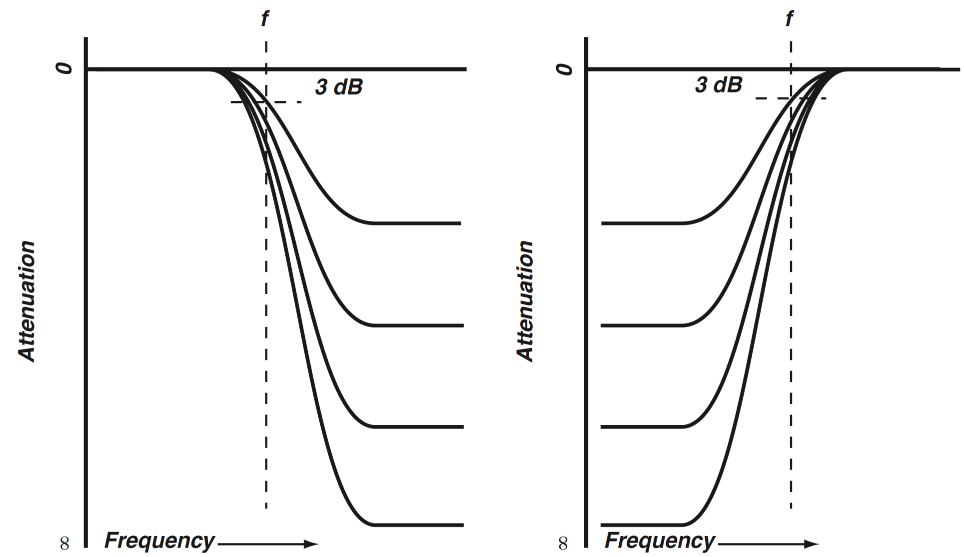


Figure 13.198: LowPass (left) and HiPass (right) Filter Response Curves

The graphic shows low and high pass filter responses for various settings of cut. The effect of the Order parameter on these filters is similar to that shown for shelving filters. Up to four parameters define the characteristic of a Lowpass or Hipass filter:

- **Order** - Selects the filter order. Higher order equals a steeper filter slope:
  1. 6dB/oct (default)
  2. 12dB/oct
  3. 18dB/oct
  4. 24dB/oct
- **Freq (Hz)** - Sets the cutoff frequency of the filter. Cutoff frequency is defined as the -3 dB point in the filter's response
- **Pass Ripple** - Adjusts the ripple amplitude within the passband. Available to 2nd, 3rd & 4th order filters.
- **Stop Ripple** - Adjusts the ripple amplitude in the stopband. Available to 2nd, 3rd & 4th order filters.

Bandpass Filters

A bandpass filter passes only frequencies within a defined region of the audio spectrum, and rejects frequencies outside of that range.

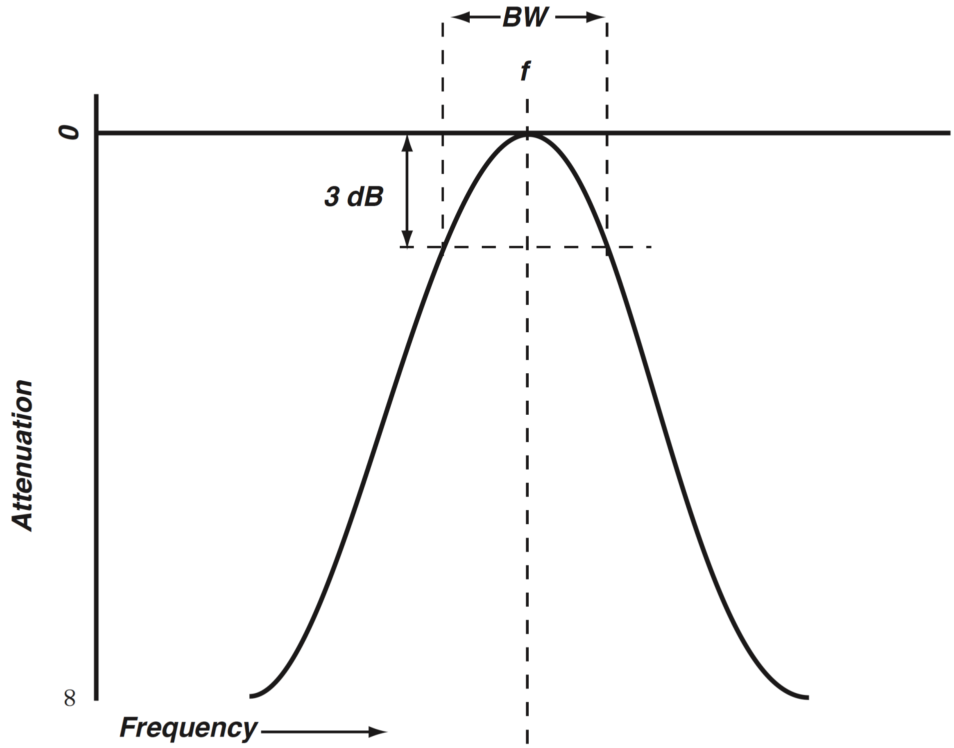


Figure 13.199: Bandpass Filter Response Curve

The graphic shows a typical bandpass filter response curve. The dotted lines indicate the -3 dB frequencies of the filter, which define its bandwidth. Up to five parameters define the characteristic of a Bandpass filter:

- **Order** - Selects the filter order. Higher order equals a steeper filter slope:
  1. 6dB/oct (default)
  2. 12dB/oct
  3. 18dB/oct
  4. 24dB/oct
- **Freq (Hz)** - Sets the the center frequency of the band affected by the filter, expressed in Hertz (Hz).
- **Q** - Adjusts the Q value and, inversely, the filter's bandwidth.
- **Pass Ripple** - Adjusts the ripple amplitude within the passband. Available to 2nd, 3rd & 4th order filters.
- **Stop Ripple** - Adjusts the ripple amplitude in the stopbands below and above the passband. Available to 2nd, 3rd & 4th order filters.

Bandstop Filter

A band stop filter is the inverse of a bandpass: it eliminates frequencies within a defined region of the audio spectrum, and passes frequencies outside of that range.

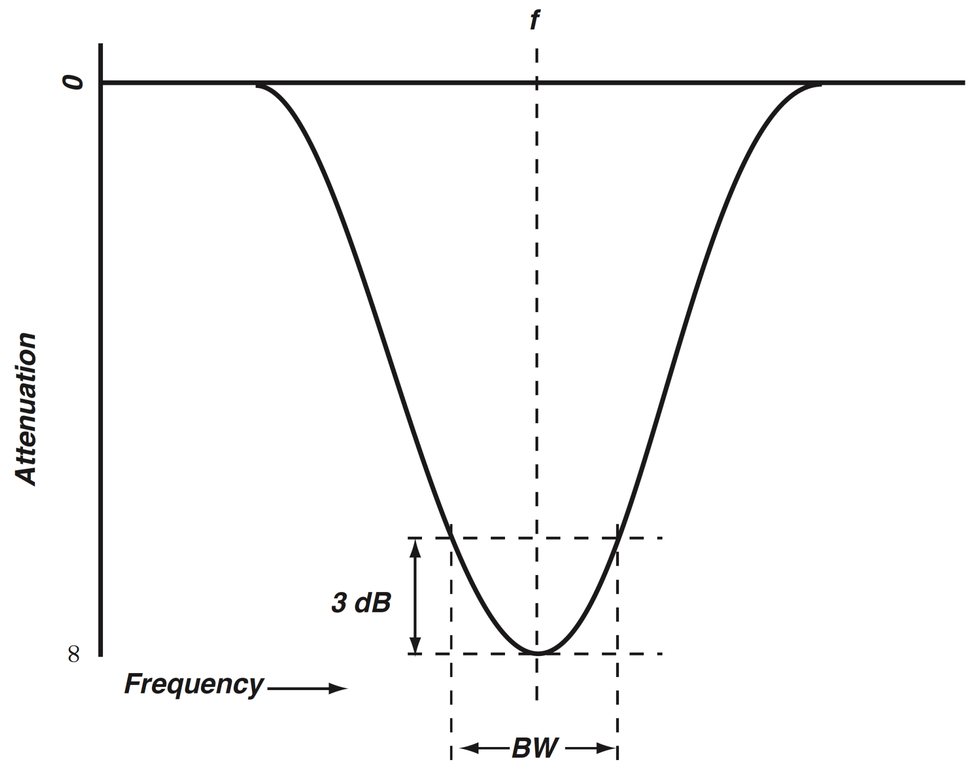


Figure 13.200: Bandstop Filter Response Curve

The graphic shows a typical band stop filter response curve. The dotted lines indicate the -3 dB frequencies of the filter, which define its bandwidth. Note that Bandstop is the only filter to employ Pass Ripple and Stop Ripple controls in the 1st-order filter slope setting. Five parameters define the characteristic of a Bandstop filter:

- **Order** - Selects the filter order. Higher order equals a steeper filter slope:
  1. 6dB/oct (default)
  2. 12dB/oct
  3. 18dB/oct
  4. 24dB/oct
- **Freq (Hz)** - Sets the the center frequency of the band affected by the filter, expressed in Hertz (Hz).
- **Q** - Adjusts the Q value and, inversely, the filter's bandwidth.
- **Pass Ripple** - Adjusts the ripple amplitude within the passbands below and above the stopband.
- **Stop Ripple** - Adjusts the ripple amplitude in the stopband.

Notch Filter

A notch filter eliminates frequencies within a very narrow region of the audio spectrum, and passes frequencies outside of that range. Notch filters are used to attenuate unwanted single-frequency signals such as AC line hum.

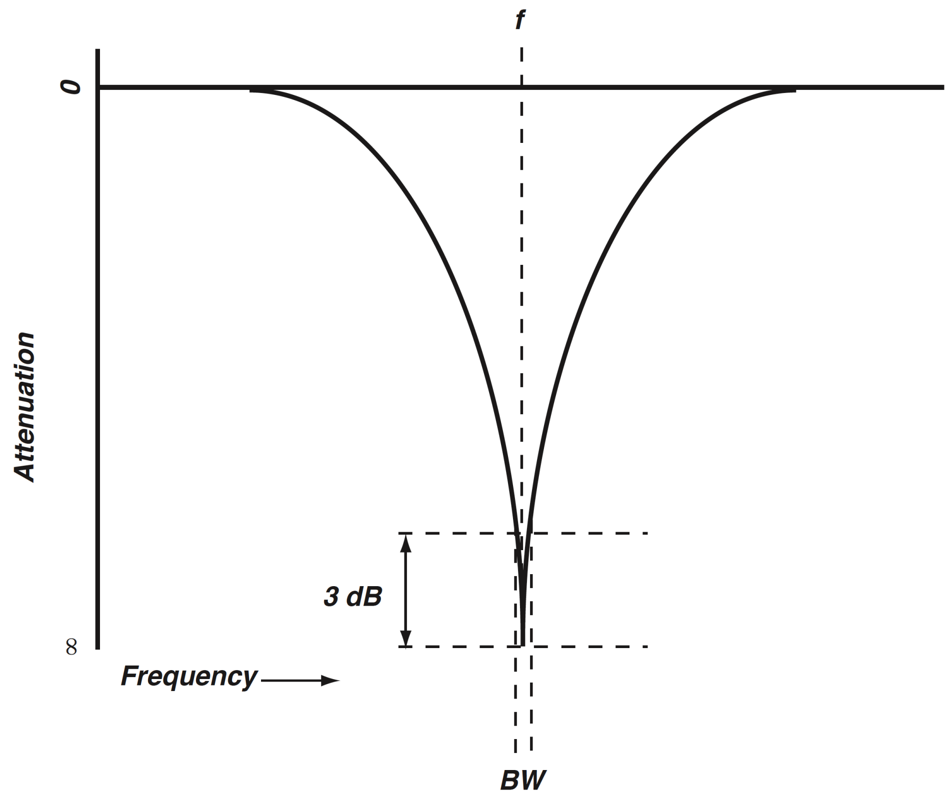


Figure 13.201: Notch Filter Response Curve

The graphic shows a typical notch filter response curve. The dotted lines indicate the -3 dB frequencies of the filter, which define its bandwidth. Two parameters define the characteristic of a Notch filter:

- **Freq (Hz)** - Sets the the center frequency of the band affected by the filter, expressed in Hertz (Hz).
- **Q** - Adjusts the Q value and, inversely, the filter's bandwidth.

Deemphasis and Deemp/NoDC Filters

The Deemphasis and Deemp/NoDC Filters are specifically for playing back legacy AES/EBU digital audio signals encoded with 75µs AES/EBU pre-emphasis.

The original intention of pre-emphasis/de-emphasis in digital recording was to enhance signal-to-noise and headroom at the highest frequencies. The quality of modern digital-to-analog conversion has long since evolved beyond the need for such trickery, so this filter will only be of interest in obsolete digital format preservation transfers.

Back in those days a/d conversion was often subject to DC offset contaminating the digital audio data. The Deemp/NoDC variation adds a DC Removal filter to address cases of emphasized digital audio containing a DC component.

DC Removal

DC Removal is a high-order, minimum phase high pass filter at 1Hz specifically designed to remove a DC component from a digital stream.

Due to the 1Hz knee and relative steepness of the filter slope, the DC Removal filter is barely visible in the EQ Transfer Function display when engaged.

RIAA Filters

The RIAA filter characteristic is a pre-emphasis/de-emphasis standard developed for stereophonic vinyl records. RIAA filters are used to add high-frequency emphasis when the record is mastered; an inverse RIAA filter in the phonograph cartridge preamplifier restores flat frequency response when the record is played on a turntable.

In Sonic EQ 3d, the emphasis (mastering) filter is named RIAA (Enc); the de-emphasis (playback) filter is named RIAA (Dec).

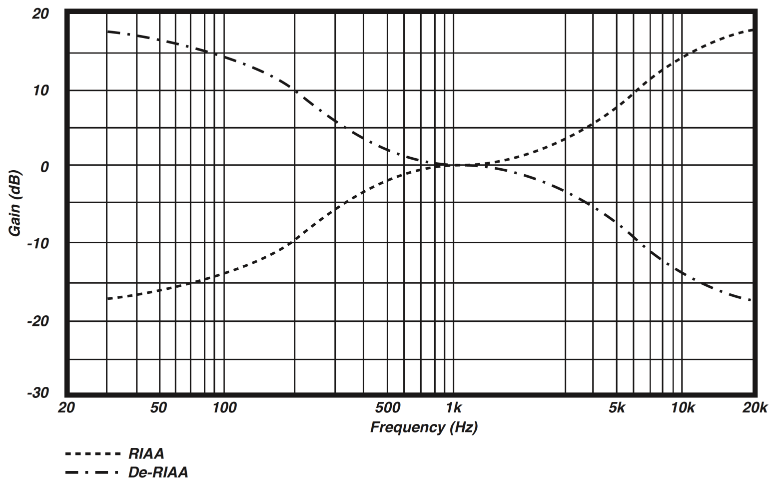


Figure 13.202: Notch Filter Response Curve

The graphic shows the response curves for the RIAA (Enc) and RIAA (Dec) filters (named RIAA and De-RIAA per the original Sonic HD manual from which this graphic was sourced). There are no adjustable parameters for these filters.

> WARNING: The RIAA (Dec) filter is +18dB at 20Hz and should never be engaged while monitoring audio.

SuperGate

Plug-in categories: **Production and Dynamics**

Introduction

SuperGate is a 3d-exclusive DSP plug-in specializing in isolating (or removing) instruments or audio events based on the program dynamic signature.

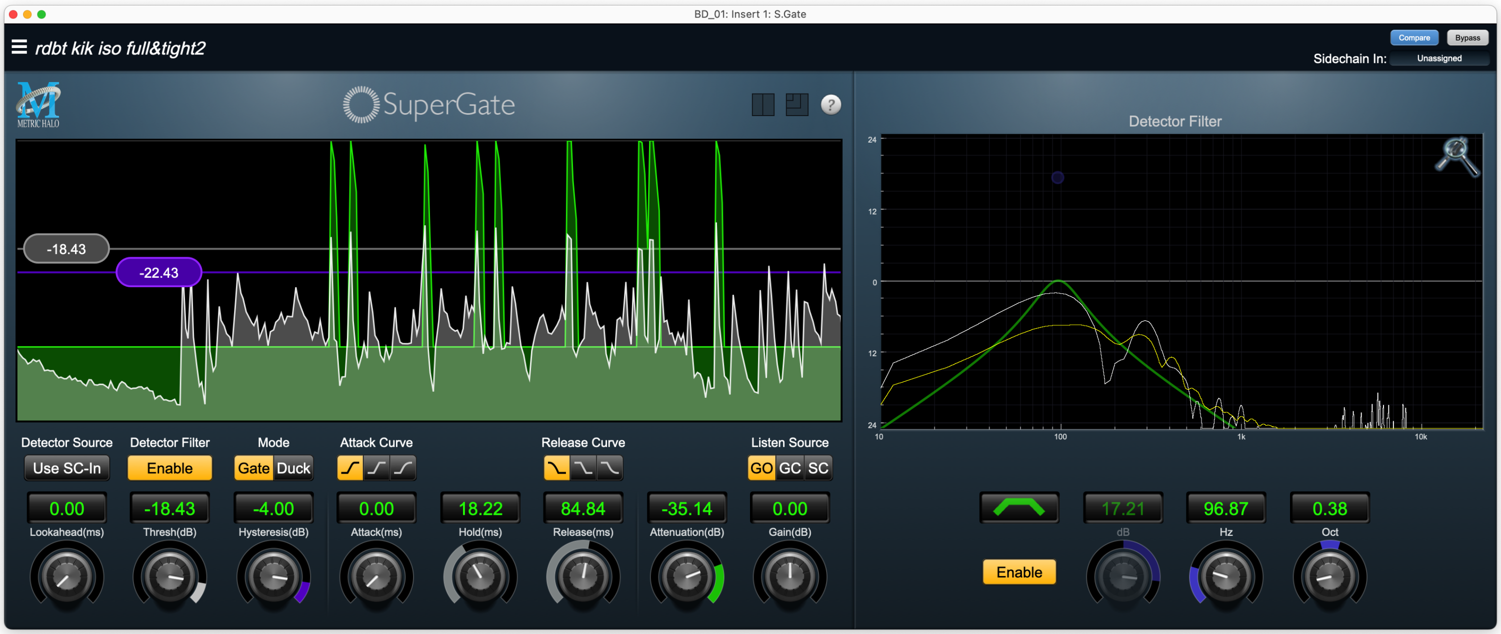


Figure 13.203: SuperGate's User Interface

SuperGate employs a number of controls not usually found in an audio gate, providing unprecedented control over the detector behavior and the attack/release shapes of the gate attenuation stages.

Features include:

- State machine "gate open" and "gate close" operation eliminates modulation between gate gain stages and the source audio
- Threshold (gate open) and Hysteresis (gate close) detector level controls
- Detector Lookahead aids detection of slower transient events
- Independent Gate Attack Time (up to 1000ms) and Attack Curve shape controls
- Gate Hold up to 2 seconds before Gate Release engages
- Independent Gate Release Time (up to 4000ms) and Release Curve shape controls
- Sidechain Detector Filter with six fully tune-able filter curves
- External Detector Source input routable from any MIOConsole hardware input or bus
- Listen Source selections monitor the *Gated* signal, *Gate Closed* (program the gate is rejecting) or the *SideChain* post-filter feed to the detector
- Attenuation control determines the gain reduction applied to audio rejected by the gate (from 0dB to -144dB)
- Duck mode: reverses the gate action, so the gate *closes* to attenuate audio when the detector is above Threshold, and *opens* when below Threshold

The SuperGate Detector/Gain History shown above to the left displays the filtered detector response to your source audio as a white line with grey fill, scrolling from right to left. The applied gate attenuation is overlaid as the green line with green fill.

Simply put: Green passes through the gate, grey does not.

Operation

SuperGate has a re-sizable plug-in window and full support for tooltip popups, along with the ability to show or hide the Detector Filter.

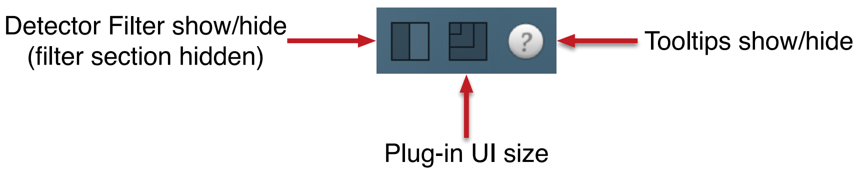


Figure 13.204: SuperGate plug-in view controls

These controls are located to the right of the SuperGate logo, above the Detector/Gain History display.

The SuperGate user interface uses a few different control elements to control its processing. These elements are:

Control Knob

Control Knobs are used to control the value of various continuous parameters of a process. There are three styles of encoders:

Figure 13.205: Swept Knob

The rings around these encoders sweep from a minimum value on the left to a maximum value to the right.

Note that *Threshold*, *Hysteresis* and *Attenuation* are negative-going parameters sweeping from 0.00dB on the right to their maximum negative value all the way left.

Figure 13.206: Spread Knob

The ring around the Detector Filter *Oct*ave encoder starts at 12 o'clock and spreads to both sides equally as the Detector Filter bandwidth is increased.

Figure 13.207: Plus/Minus Knob

The rings around these encoders start at 12 o'clock and sweep to either side. These knobs are used for the output *Gain* and Detector Filter passband boost/cut controls.

Control tips

You can change the value of each knob in a number of different ways.

- Click and drag the knob to change the value continuously.
- Dragging up or to the right will increase the value, while dragging down or to the left will decrease the value.

  Hint: If you hold down the ⌘ (Command) key when you click, you will be able to adjust the value with finer precision.
- ⌥ (Option)-click to reset any knob to its default value. You may also double-click a knob to reset it.
- **Rapid parameter entry**
  - Double-click the numeric readout of the knob to type in a number directly. The text-entry field will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, or **⌘.** (Command + .) keys.
  - Hit **return** or **enter** to confirm the value and dismiss the pop-up.
  - Hit the **tab** key to confirm the value and immediately activate the entry field for the next control. **⇧-tab** (Shift + tab) will display the entry field for the previous control).
  - Hit **⌘.** (Command + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.
  - When you enter a number into the pop-up entry, you can use a couple of abbreviations: "k" multiplies the number by 1000 and "m" divides the number by 1000. So if you want to enter 16,500 Hz you can just type 16.5k.

Toggle Buttons

Figure 13.208: *Detector Source* "Use SC-In" is Off, *Detector Filter* is On)

Toggle buttons are simple on/off switches. They light up (colored) when they are on and are dark (uncolored) when they are off. Click to toggle.

In SuperGate, these buttons are used by the *Detector Source* to select between the internal audio and an external sidechain key input, and *Detector Filter* to engage/disengage the audio input filter to the detector. Note that the *Detector Filter* button is duplicated in the main UI and the (sometimes hidden) Detector Filter pane.

Radio Buttons

Figure 13.209: Radio Buttons (to choose one of each set)

Radio buttons are like toggle buttons, except that they simultaneously disable another parameter when enabled. Selecting a button will light it up (colored) and make the other buttons in the set go dark.

Radio buttons are used to select between *Gate* or *Duck* modes, the three gate *Attack Curve* & *Release Curve* settings, and the three *Listen Source* audio feeds.

Detector/Gain History display

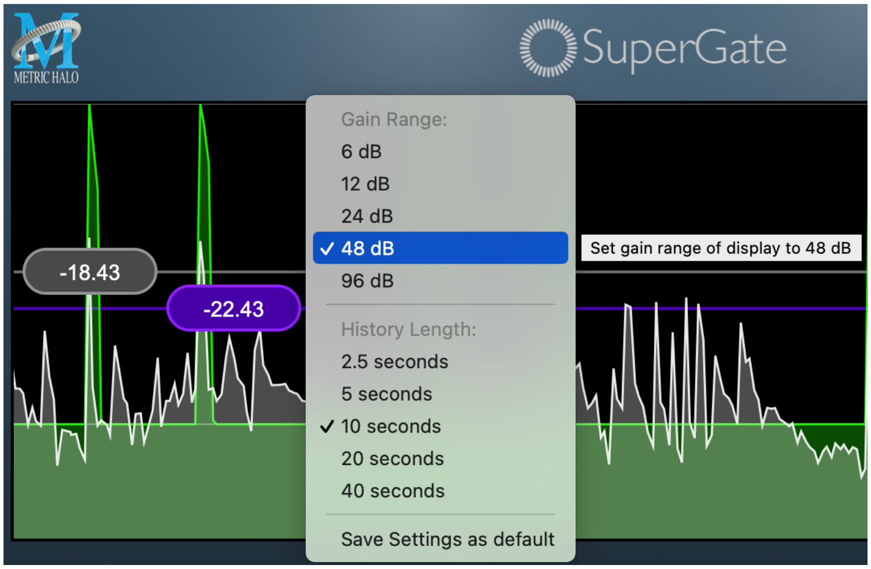


Figure 13.210: Detector/Gain History with Settings Menu

The SuperGate Detector/Gain History displays the filtered detector response to your source audio as a white line with grey fill scrolling from right to left. The applied gate attenuation is overlaid as the green line with green fill.

The top of the display is always 0dBFS. Solid green reaching the top of the display indicates that the gate is fully open. Since the display is showing only detector response and gain reduction, hitting the top of the display is not an indication of excessive audio level.

Click on the display to reveal the settings menu (shown above).

The default **Gain Range** of the display is 96dB, so it's 0dBFS at the top of the display and -96dBFS at the bottom. This widest display range is the default view since it will reveal any audio program activity down to noise floor levels. There are five options for focusing in on the detector action:

- 6 dB
- 12 dB
- 24 dB
- 48 dB
- 96 dB

History Length is the period of time shown scrolling across the window. Events happening "Now" take place at the right edge of the screen, with recent events scrolling down the path of history to the left. The default History Length shows 20 seconds of audio events in the display. There are five settings available:

- 2.5 seconds
- 5 seconds
- 10 seconds
- 20 seconds
- 40 seconds

The shortest 2.5 seconds setting provides the most detailed view of the detector and gate attenuation response, but it moves by very quickly. Longer History Length settings offer a longer time window at the cost of individual event details.

Lastly, **Save Settings as default** allows you to save your current meter display preferences as your default setting for new instances of the processor.

Supergate Theory of Operation

Yes, it's that much better. This is why…

> *State* (or *Discrete*) data is defined as information that can take only certain values, like the number of coins in a jar, a "true or false" statement, or digital audio samples.
>
> *Continuous* data can describe a value between any other two values, such as measures of heat, energy, attitude, amplitude, frequency and time (absolute and relative).
>
> Specifically, a *State* value corresponds to a digital quantity and a *Continuous* value corresponds to an analog quantity.

So, one of the biggest problems with many traditional triggered continuous gain stage topologies (no matter how smart they are) is the interaction between those gain stages and the audio program they are gating. "Continuous Gain" gates are analog-style volume controls, no different than a fader control, that open at a set "trigger" volume level (dBr) and close after either a set time period or at a "de-trigger" volume level (dBr), with variable attack and release times respectively.

The thing is, in the digital world, applying continuously variable mathematical changes to a moving stream of audio data has audible consequences in the form of aliasing artifacts (zipper noise, static, etc.). To avoid those artifacts, the volume change process must be interpolated to mitigate and smooth out those effects. That interpolation must take place over time (the duration of the level change), and that interpolated gain-shift-over-time by it's very nature has a modulating effect on the source audio.

Granted, that modulation is subjectively less offensive than fingernails across a blackboard, but it does have enough smearing effect to oftentimes make accurate gating an exercise in frustration. At that point, many just take the much longer road of editing the unwanted audio at the track waveform level.

Some more advanced "smart gates" apply AI-style learning algorithms and/or envelope-follower gate designs to trigger and track the source audio more intelligently. There really is some very cool tech out there, unfortunately many of these tools are also continuous-gain processors with these same fundamental process limitations.

SuperGate takes an entirely different approach.

In SuperGate, the shapes of the *Open Gain Curve* and the *Close Gain Curve* are calculated and mapped as a state function, and applied as a state operation at the 'gate open' trigger and 'gate close' de-trigger points you set up in the detector UI.

These states are literally digital snapshots of the gate gain curves (which you have tuned by ear), overlaid on the audio data starting with the trigger point sample value as its starting reference. From that first sample, each consecutive sample value is mapped from the source audio samples to follow the programmed gain curve, like applying a level curve in a photo editing program.

Applying the gain curves in this way completely avoids aliasing artifacts, the interpolation stages required to correct for them, and the modulation and smearing effects that degrade the gain transition.

The audible advantage of this approach is immediately apparent as you work with the processor. The *Open* and *Close* gate curves are applied with perfect repeatability as if they were copy/pasted onto the audio stream at the trigger point (which is in fact exactly the case), and all the minute variations of each audio event which provide feel and personality can come through the gate completely untouched.

After all, first and foremost, the job of an audio gate is to let the good stuff through unaffected and only attenuate the unwanted signal, right?

Processing

The block diagram below illustrates the overall structure of the SuperGate.

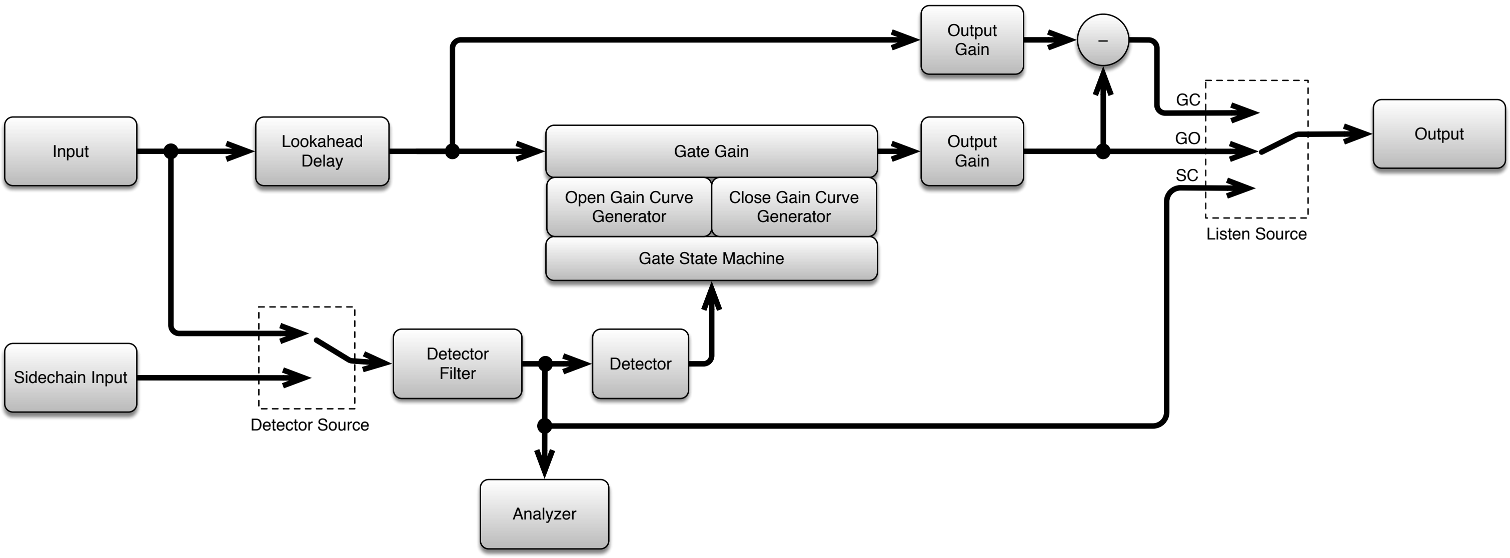


Figure 13.211: SuperGate Block Diagram

If you follow the horizontal line drawn from the Input to the Output, that is your primary audio signal path. There are actually only two controls in the SuperGate UI directly that directly affect the audio path - the *Lookahead* delay and the *Output Gain* control.

Right in the middle of the audio path is the *Gate Gain* stage. Both of SuperGates graphics displays and all of the remaining controls operate in the modules shown below the Gate Gain stage, in the detector sidechain path. It is here that the gate control settings you create by ear (using the displays as a visual guide) are calculated as state gain operations and imprinted on the source audio.

Here's the breakdown:

- The detector sidechain may be fed from either the Input path or an external key signal.
- The Input (or external key) feeds the *Detector Filter*, where you can fine-tune the signal so the Detector can trigger on a specific frequency range unique to the sound you are trying to gate.
- The Detector's job is twofold:
  1. …to relay the filtered sidechain transient information to the *Gate State Machine*, and
  2. …to feed back the *Attenuator*, *Threshold* and *Hysteresis* control data to the *Detector Gain/History* display. This provides instant visual feedback of your parameter settings in relation to the filtered sidechain audio as you listen (as in, "Oh! So that's what that knob does…").
- The *Gate State Machine*, combines the Detector sidechain output with your control input from the SuperGate UI to program the gain curve map data for the *Open Gain Curve Generator* and the *Close Gain Curve Generator*.
- The Gain Curve Generators translate that curve map data into a series of sample values, with the beginning of each gate curve as the first sample value at the 'Gate Open' and 'Gate Close' trigger points, respectively.
- The *Gate Gain* stage then applies this series of gain values to the audio stream, sample-by-sample, starting at the gate trigger point for the duration of each Gate Gain curve. This imprinting of the 'Gate Open' and 'Gate Closed' gain curves as new samples on the way to the Output stage is a state operation - no aliasing, no interpolation… instant gratification with zero artifacts.

The detector itself doesn't actually react to any of the parameter controls; it is the state machine and the gain curve generators that do.

Detector / Gate State Gain machine interface breakdown

Below is a map of the sidechain audio, detector and state gain machine process blocks, and how they are connected to SuperGates' UI controls and displays.

Detailing the detector sidechain first will provide context for the "Lookahead" and "Output Gain" analog path controls later on.

Black arrows are audio signal and red arrows are state control data.

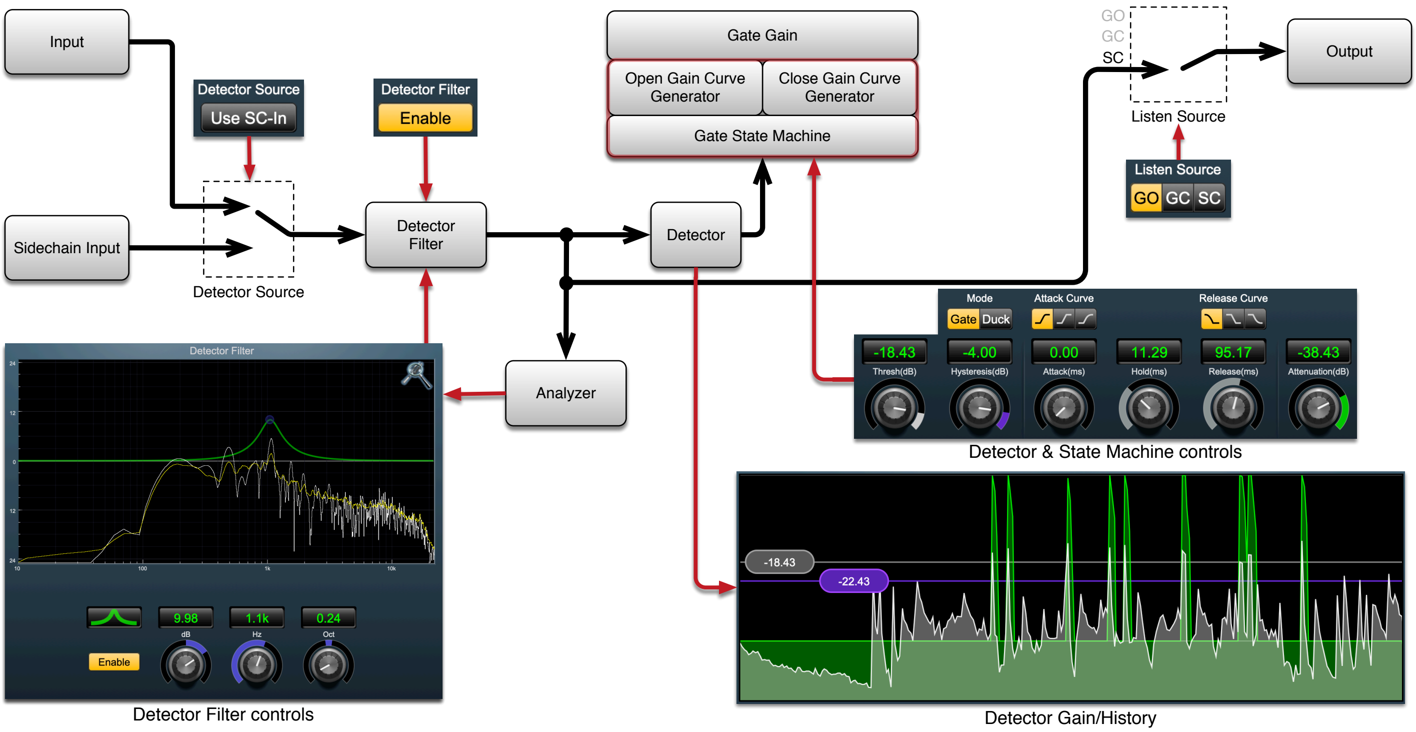


Figure 13.212: SuperGate Detector and Gate State Machine control map

In the following sections we will follow the detector sidechain and gate control signal path through the process blocks from Input to Output.

Editorial Note

Those of us who come from a more analog-style audio background tend to approach signal flow as a serial/linear path, traveling from the input through a series of processors and gain stages laid out one after the other, eventually reaching the final output bus.

While the signal processing of SuperGate is non-traditional, the controls and Gain History display have been set up with familiar realtime music workflows in mind. A few minutes auditioning a drum or guitar line and moving the controls while watching the Gain History display will reveal SuperGate's various functions quite clearly.

So, whatever your background, do not be put off by the science… this processor is an absolute blast to play with.

Keep in mind, the SuperGate state gain machine is **attenuation-only**, so unless you manually add too much Output Gain at the final stage, SuperGate is incapable of clipping your source audio. You can feed absolutely evil things through the detector sidechain without fear, often resulting in fascinating and bizarre effects on the audio dynamics.

Detector Source

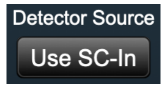


Figure 13.213: Detector Sidechain-In enable

Detector Source selects the Detector input feed between either the primary audio source (the "Input" block) or the externally-routed sidechain input (the "Sidechain Input" block). When dark, the audio Input is selected. When "Use SC-In" is illuminated, the external Sidechain Input is feeding the Detector.

Note that if "Use SC-In" is illuminated but there is no Sidechain Input routed, there will be no incoming signal to the Detector.

Sidechain Input selector

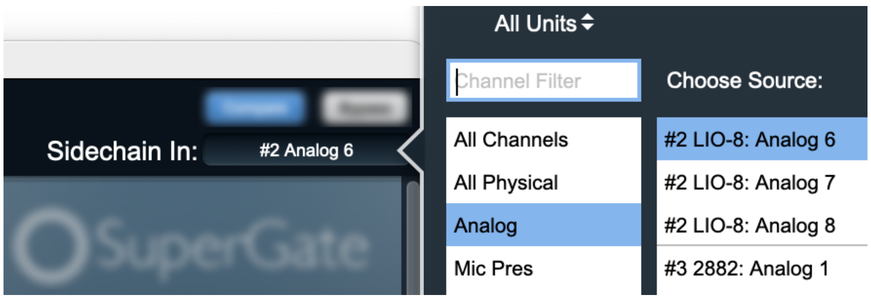


Figure 13.214: "Sidechain In" routing selector

The "Sidechain In" routing selector is located at the upper right of the UI header. SuperGate can use any hardware input or MIOConsole bus as a source to feed the detector sidechain. Click to open the input routing selection window.

Since "Sidechain In" routing is a MIOConsole header bar function common to a number of plug-ins, it is not represented graphically in the control map diagram on the previous page.

It corresponds the "Sidechain Input" block at the left edge of the controls map.

The Detector Filter

The *Detector Filter* is a single-stage audio EQ with six filter shapes for conditioning the frequency response of the sidechain audio prior to the detector.

The filter window

This circuit is depicted at the bottom left quadrant of the *Detector and Gate State Machine* control map.

The filter processing stage itself is represented in the controls map as the detector filter block, and it is the "Analyzer" block which feeds the filtered signal to the Detector Transfer Function and Detector Spectragraph displays.

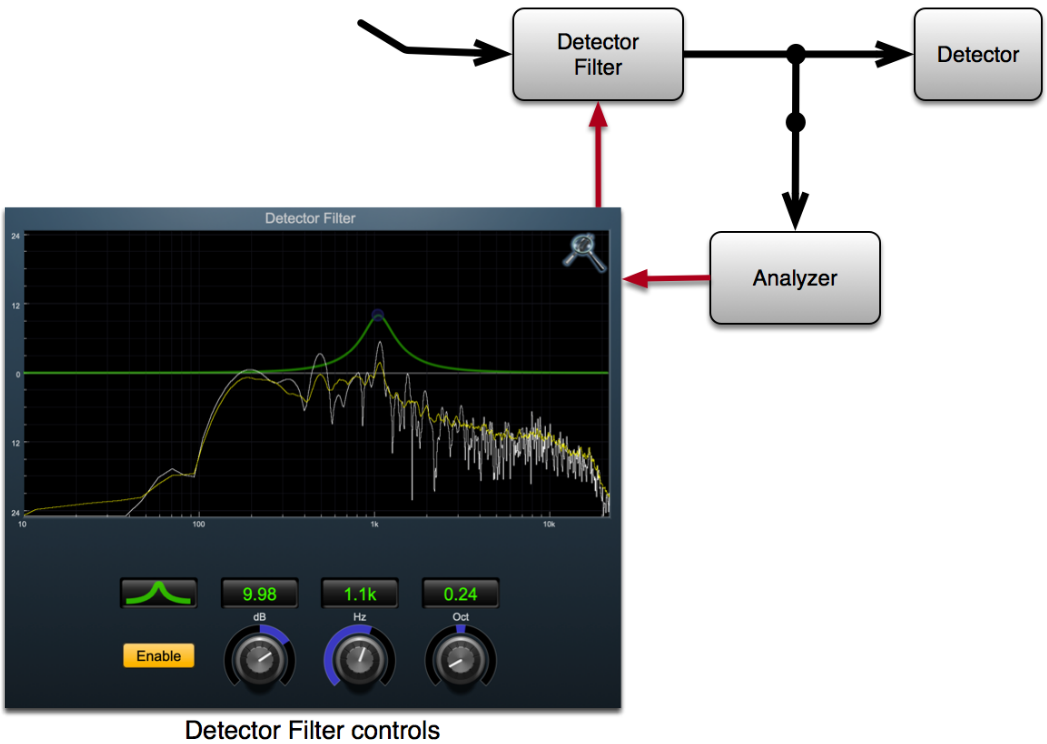


Figure 13.215: Detector Filter (flow chart focus)

Detector Filter Enable

The *Detector Filter Enable* switch is used to insert the "Detector Filter" into the sidechain feed to the Detector. When dark, the filter is bypassed. When "Enable" is lit, the filter is in-line and processing.

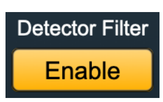


Figure 13.216: Detector Filter Enable

The "Detector Filter Enable" control is duplicated in both the primary control panel (under the Detector Gain/History display) and in the Detector Filter window.

Detector Filter Transfer Function

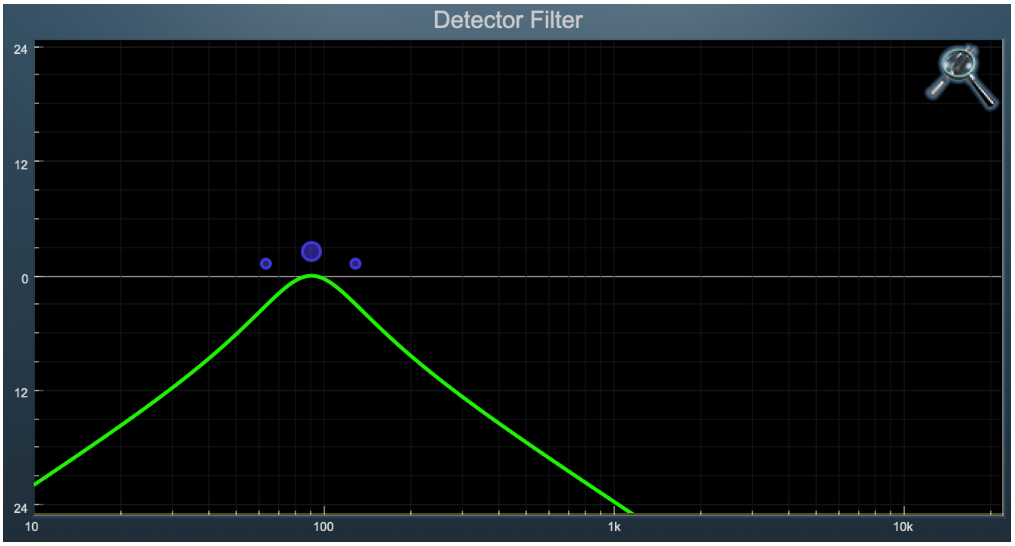


Figure 13.217: Detector Filter Transfer Function (bandpass filter shown)

The Detector Filter transfer function is both an intuitive control interface and a visual representation of how the filter is processing the signal.

The horizontal axis provides frequency calibration in Hertz (Hz). The vertical axis provides level calibration in decibels (dBr). The heavy green line indicates the relative change in level introduced by the filter.

Clicking the large colored dot and dragging horizontally will allow you to adjust the center (or knee) frequency. Parametric and shelf filters also allow you to drag vertically to boost or cut passband gain (±24dB).

When using a parametric, shelf or bandpass filter there will also be two smaller colored dots that can be used to control the bandwidth of the filter.

⌘ (Command)–click or double-click the larger 'frequency' dot to toggle the band enable. ⌥ (Option)–click the dot to adjust the bandwidth (dragging right increases the bandwidth, left decreases the bandwidth).

⌘⌥ (Command + Option)–click the dot to switch the filter type.

Click and drag the smaller dots associated with a larger dot to adjust the filter bandwidth.

Click anywhere in the black area of the transfer function to deselect the filter point and dim the filter curve trace when you need to concentrate more on the spectragraph analysis readout.

If you right-click or ^ (Control) click on the transfer function, you will see a menu to set the vertical dB scale for the display.

The values are:

- ±3 dB
- ±6 dB
- ±12 dB
- ±24 dB
- ±36 dB

This menu also allows you to specify whether touching the filter controller dots will automatically enable the filter. This preference applies across all instances of SuperGate in your mixer.

Spectragraph Analyzer

The spectragraph display is always live and available as a visual reference even when the Detector Filter itself is not enabled. Clicking the SpectraFoo™ logo in the upper right hand corner of the transfer function will activate the spectragraph, showing the realtime post-filter frequency analysis of your signal:

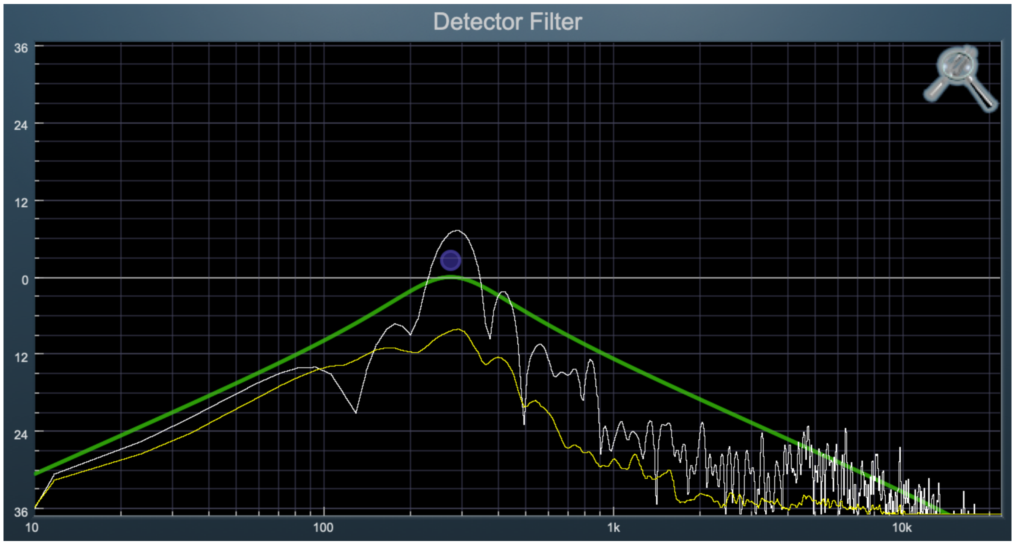


Figure 13.218: Detector Filter Spectragraph (±36dB view range shown)

The traces are:

- White: post-filter instantaneous display
- Yellow: post-filter average display

The *instantaneous* trace updates in real-time, allowing you to see the immediate peak level of your audio. The *average* trace displays the level as averaged over a short period, giving you a more general view.

The spectragraph analyses the signal post-filter, allowing you to see the effect of your filter before it hits the detector stage. To disable the spectragraph entirely, click the active 'Foo icon.

If you right-click or ^ (Control) click on the transfer function, you will see a menu to set options for the spectragraph:

- Show Instantaneous Trace: Toggles whether the spectragraph shows the instant response of your audio.
- Show Average Trace: Toggles whether the spectragraph shows the averaged response of your audio.

These settings are stored for each transfer function window separately, and for each instance of SuperGate.

> **Hint:** The Bandpass filter is especially efficient in SuperGate if you loop a section of your target audio, narrow the filter to a very tight bandwidth, and slowly move the center frequency while listening. The goal is to eliminate as much spurious signal as possible while retaining the maximum peak level possible from your target instrument.
>
> The more peak signal the detector has to work with, the more room you have to play with the Threshold trigger and Hysteresis de-trigger controls, and the more control you have for shaping the sound of the gated material. If you need more peak signal at a particular frequency for your target instrument, slightly widen the bandpass filter bell until you start to hear bleed from other instruments, then back off a touch.

Filter Type

The SuperGate detector filter stage provides 6 different types of filter shapes:

You can select from these types via three different methods. Each time you click on the Filter Type control, the band will switch to the next type in the list (and wrap to the beginning when you hit the end of the list). If you click and hold the mouse button, a pop-up menu listing all of the types will appear after about 1/4 of a second. You can select the type directly from this pop-up menu. If you want to access the menu without having to wait, hold down the ^ (Control) key when you click or right-click.

- **Peaking/Parametric** – a second order bell-shaped parametric boost/cut filter. The Gain control has a boost/cut range of ± 24 dB. When the boost is greater than +15 dB the filter gains a resonant quality which should not affect detector performance but may be audible when monitoring the detector filter sidechain directly. The center frequency of the filter can be any frequency between 20 Hz and 20 kHz. The bandwidth of the filter is continuously variable between 0.1 octaves and 2.5 octaves.

Figure 13.219: Peaking/Parametric

- **Low Shelf** – a shelving filter that applies boost/cut to low frequencies. The frequency knee (Hz) is continuously variable from 20 Hz to 20 kHz. The gain control (dB) has a boost/cut range of ± 24 dB. The bandwidth (Oct) controls the dip/peak that is added at the end of the transition band.

Figure 13.220: Low Shelf

- **High Shelf** – a shelving filter that applies boost/cut to high frequencies. The frequency knee (Hz) is continuously variable from 20 Hz to 20 kHz. The gain control (dB) has a boost/cut range of ± 24 dB. The bandwidth (Oct) controls the dip/peak that is added at the end of the transition band.

Figure 13.221: High Shelf

- **High Cut** – a 12 dB/octave high cut filter with a -3dB point that is continuously adjustable between 20 Hz and 20 kHz.

Figure 13.222: High Cut

- **Low Cut** – a 12 dB/octave low cut filter with a -3dB point that is continuously adjustable between 20 Hz and 20 kHz.

Figure 13.223: Low Cut

- **Bandpass** – a bandpass filter with 6dB per octave skirt on the high and low ends of the pass band. The width of the pass band can be adjusted between 0.1 octaves and 2.5 octaves and the center of the pass band is continuously adjustable between 20 Hz and 20 kHz. There is no boost/cut control for the bandpass filter.

Figure 13.224: Bandpass

Gate State Machine: Levels controls

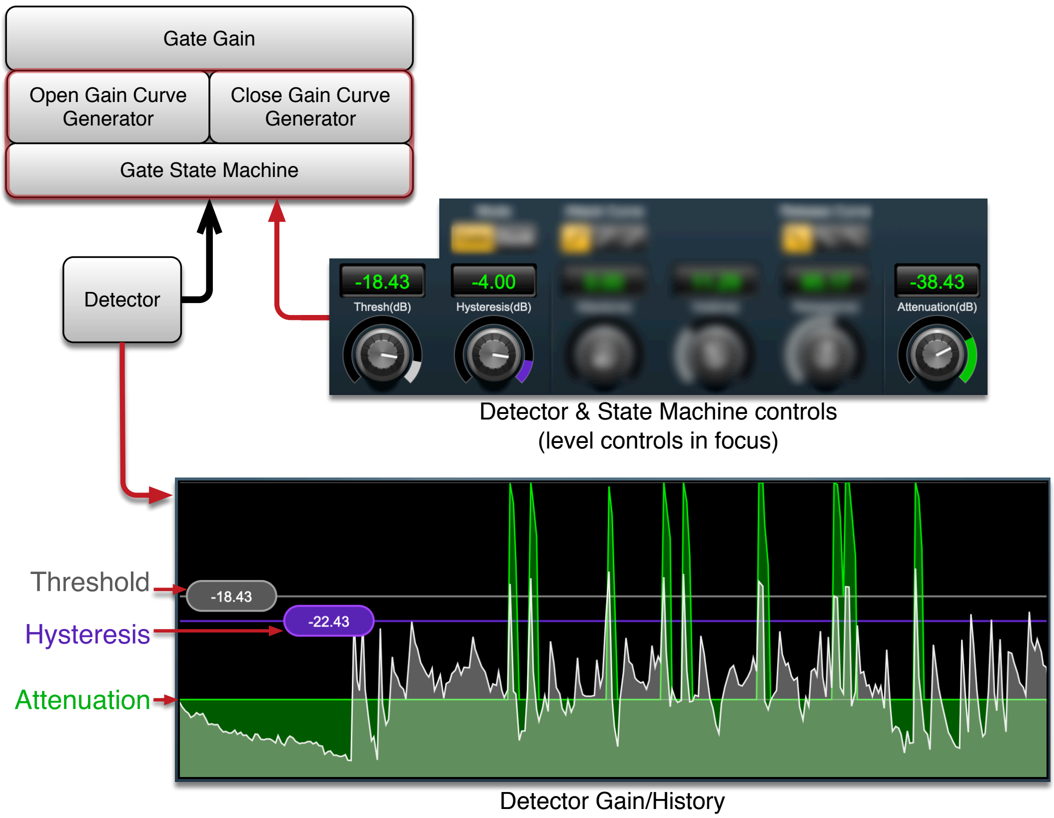


Figure 13.225: Gate State Machine: Level controls

Attenuation(dB): Attenuation level

Attenuation defines the maximum amount of gain reduction applied to the audio when the gate is closed, from the default -144dB (no audio) to 0dB (no attenuation). The Attenuation level serves as a base reference level for the Gate State Machine gain curve operation, and is shown in the Detector Gain/History as the horizontal green overlay upon which the Attack (gate open) and Release (gate close) Curves sit.

The green overlaid audio below the Attenuation line is passed through, the grey audio above the line is removed.

Attenuation is applied to all source audio upon entering the Gate Gain stage.

Note that even though Attenuation is technically applied as the first thing in the gain stage, it is usually the last thing to be set before closing SuperGate and moving on to the next phase of your mix.

In practice, Attenuation sets the amount of non-target instrument bleed allowed into the mix. Very often when trying to isolate an instrument or sound in a messy track, it is extremely helpful to back off the Attenuation and listen to the un-gated audio mixed with your gated audio settings to make sure you didn't miss any soft hits, flams, fluffs or grace notes, and to make sure your gate settings are providing the character of the instrument that best serves the performance.

This is why the Attenuation control is placed at the far right of the control cluster near the Output Gain and Listen Source controls: switching your Listen Source between the *Gate Open*, *Gate Closed* and isolated *Sidechain* feeds while adjusting Attenuation bleed-through not only helps you dial in your settings but often inspires more creative effects for your mix.

Thresh(dB): Threshold level

The *Threshold* control sets the 'gate open' trigger level (in dB) from the default 0dB to -144dB.

When the Threshold level is reached, the first sample of the Attack gain curve is printed on the audio data, proceeding to embed the gate curve until the gate is fully open. The gate will remain fully open until the Hysteresis level is reached and the gate Hold time expires.

Hysteresis(dB): Hysteresis level

The *Hysteresis* defines the level at which the gate Hold timer starts counting down to 'gate close' and the imprinting of the Release Curve.

Hysteresis maintains a set level window relative to the Threshold, ensuring consistent 'gate open' and 'gate close' response to even the most subtle variations in the source audio.

Detector/Gain History "Control Bubbles"

Figure 13.226: Threshold and Hysteresis controls shown within the display range (left) and below the display range (right)

The grey and violet "bubbles" that appear in the Detector/Gain History are handles for the detector *Threshold* (grey) and *Hysteresis* (violet) control knobs. The grey and violet lines extending from these handles across the display make it easy to set the precise detector level at which 'gate open' is triggered (Threshold) and when 'gate close' is triggered (Hysteresis).

Note that the violet *Hysteresis* value in the bubble shows the actual parameter level in dB, whereas the numeric Hysteresis value shows the difference between the Threshold and Hysteresis. In the graphical context of aligning the level with respect to audio events, one uses the physical level referenced to 0dBFS, but in the context of detector 'gate open' and 'gate close' commands, the size of the window between those levels becomes more important.

As you move the Threshold control, you will see the Hysteresis follow, maintaining a consistent relative difference between the two. This allows you to change the detector level threshold across a wide range without affecting the behavior of the gate gain stages. Follow the link for more details.
When the Threshold and Hysteresis control bubbles are set below the current Detector/Gain History Gain Range, they rest on the bottom of the display window with a downwards-pointing arrow indicating they are below the windows' view (as shown above right).

Gate State Machine: Gain Curve controls

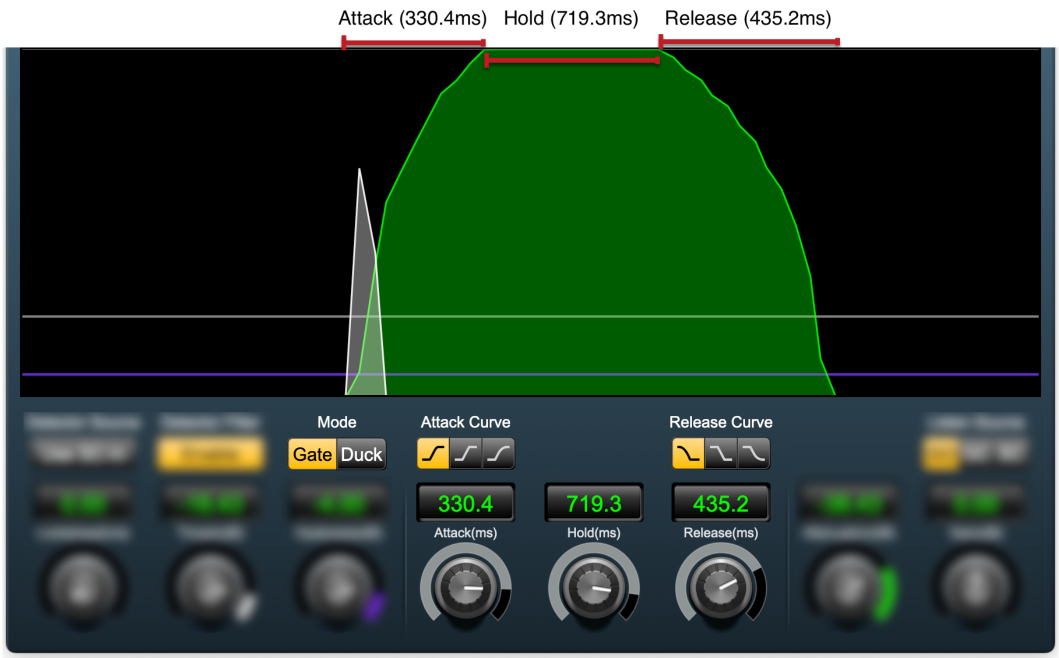


Figure 13.227: Gate State Machine: Gain Curve shape, Attack, Hold and Release controls

There are five controls in SuperGate dedicated to the shape of the 'gate open' and 'gate close' gain curves.

It is important to remember that these controls are not dynamic by nature, and have no effect on the detector input response.

In SuperGate, the Gain Curve shape controls bear much more resemblance to synthesizer key response programming, audio edit fade curves or graphics luminance level parameters than what you normally find in a dynamics processor.

Attack(ms): Gain Curve Attack time

Attack controls how quickly the gate opens from 0ms (the default) to a max of 1000ms. The Attack Gain Curve automatically adapts it's shape to match the duration entered.

The 'Attack' gain curve is applied at the moment the detector reaches the Threshold trigger level, opening the gate starting at the Attenuator level and progressing until the gate is fully open.

Hold(ms): Gain Curve Hold time

The *Hold* control determines how long the gate stays fully open at 0dB of attenuation.

Gate 'Hold' duration may be from 0ms to 2000ms, starting from the moment the detector hits the Hysteresis de-trigger level.

Release(ms): Gain Curve Release time

Release controls how quickly the gate closes from 0ms (the default) to a max of 4000ms.

The 'Release' gain curve is applied at the moment the Hold duration times out.

Attack Curve shape selector

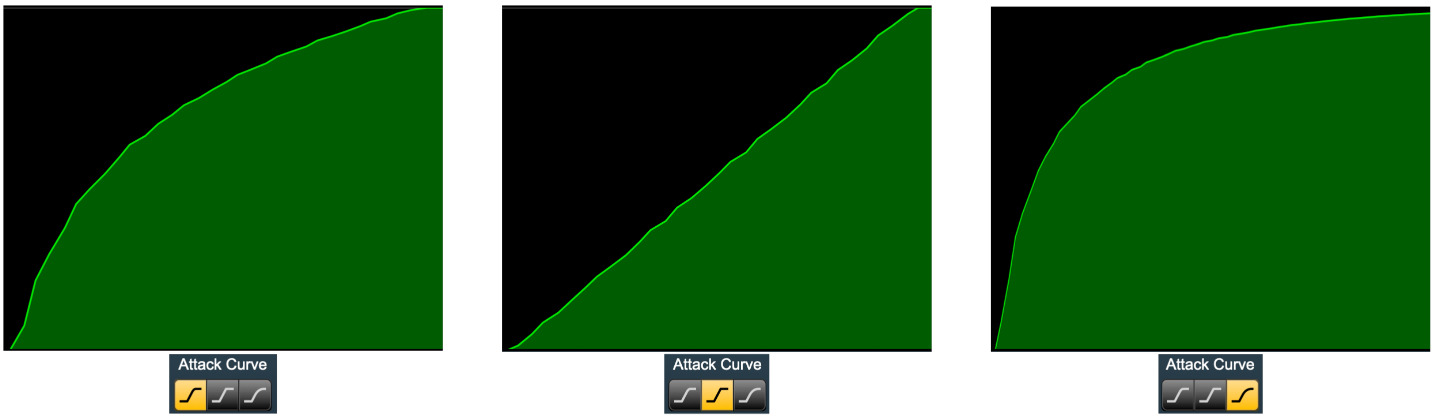


Figure 13.228: Attack Curve shapes

There are three Attack Curve shapes to choose from. As you would expect, the audible differences between them become more subtle as the Attack time decreases, but even at minimal attack times swtiching between the curves reveals a different feel.

- The leftmost (default) curve has a linear gain characteristic, suitable for most purposes and overall a good starting point.
- The mellow child in the middle has a slower ramp-up making it a good choice for slower-responding instruments, pushing sounds back in the soundfield by damping initial transient energy, or in some cases even click/crackle mitigation.
- The third curve has the quickest overall ramp-up, but damps the top of the transient somewhat at longer attack times by taking longer to fully open the gate over that last few dB. This curve is visually similar to the default (leftmost) attack curve but offers a full yet maybe less aggressive option on faster, louder sources.

Release Curve shape selector

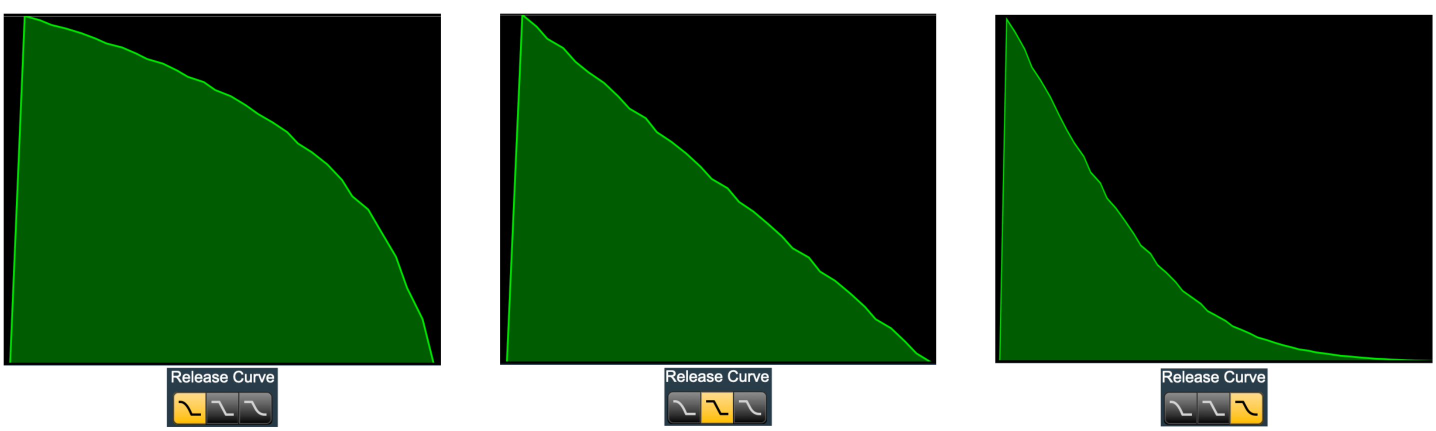


Figure 13.229: Release Curve shapes

The three Release Curves essentially mirror the Attack Curve shapes as far as energy curves go, but in practice there is a huge interplay with the gate Hold parameter which is not present with the Attack Curve.

- As with the Attack side, the leftmost (default) Release Curve is a good general purpose linear decay, appropriate for gating instruments, vocals or dialog. The rapid drop-off at the end of the curve provides good control for isolating instrument events on tracks with significant noise to reject.

- The slower drop-off of the middle Release Curve is especially handy for legato program transitions or emulating manual fader-riding in Duck mode.
- The third release curve has a rapid initial drop-off followed by a gradual transition to the Attenuation floor. This curve is well-suited to transitioning from events requiring a longer Hold setting to reproduce the body of an instrument (say, a kick drum ringout) but needs a fast controlled release to reject spurious instrument noise.

Gate Gain Stage Mode selector

Figure 13.230: Gate vs. Duck Mode

The SuperGate "Gate/Duck" control does not impact gain curve shapes per se, rather it takes the existing gain curves and flips everything upside down (or backwards...?).

Specifically, please note in the graphic above that Duck mode swaps the Attack gain curves for the Release gain curves without changing any of the level thresholds or time parameters. This creates a literal mirror image of the normal gating behavior.

Ducking with a mild Attenuation setting is particularly useful for broadcast or podcasting to turn down a music bed whenever voiceover program occurs.

Audio path controls

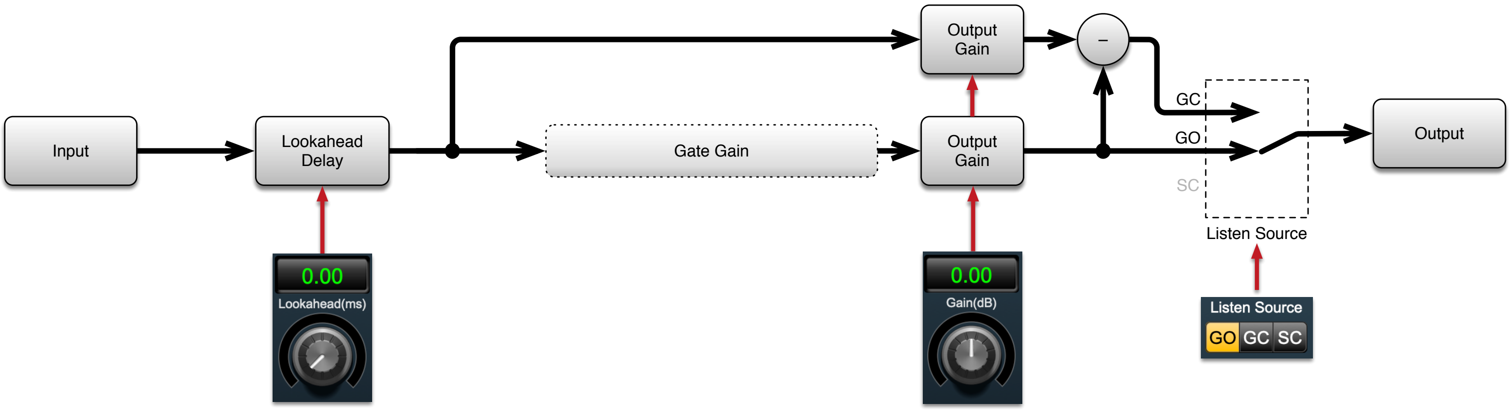


Figure 13.231: SuperGate primary audio path controls

Lookahead

Lookahead delays the primary audio input signal relative to the detector sidechain.

This is useful in cases where the 'Gate Open' or 'Gate Close' need to be triggered slightly ahead of the Threshold trigger point.

If you hear slight clicks at the gate transitions (sometimes an issue when monitoring the 'Gate Closed' signal), try adding 'Lookahead' until the click goes away.

The 'Lookahead' delay range is from 0ms to 10ms. Keep in mind adding this delay may be audible in some circumstances where relative phase issues may be exposed.

Output Gain

This control allows you to boost or cut the output level after all processing has been applied. The gain range is ±24 dB.

Since the only processing up to this point is creative attenuation, Output Gain is very often left alone at the default 0.0 dB.

On the other hand, since SuperGate is obviously much more than "just another gate", this manual gain stage is included for instances of picking out very low level signals which need a boost, or for dropping the output gain to match with the next processing stage.

Listen Source selector

The **Listen Source** radio button selects between the signal being passed through the gate, the signal being rejected by the gate, or the filtered detector sidechain input signal.

As you can see on the signal flowchart, there are actually two parallel output gain stages linked to the Output Gain control. One passes the original unprocessed audio and the other passes the gated audio from the Gate Gain engine.

After the output gain stages, the gated audio (on the lower path) continues to the *Gate Open* (GO) Listen Source feed.

This Gate Open audio also feeds into a phase-inverter and is summed with the unprocessed signal (on the upper path). Since both gated and unprocessed signals are time-aligned, the inverted gated audio cancels from the original unprocessed signal.

The result of removing the Gate Open audio from the unprocessed original leaves the signal rejected by the gate, and is available at the Listen Source *Gate Closed* (GC) selector.

The filtered detector sidechain audio can of course be monitored at the SideChain (SC) Listen source selector.

> It should be noted here that removing an instrument from audio is every bit as legitimate as isolating an instrument, and SuperGate is often surprisingly good at it. If you dial in an instrument as if to isolate it, then switch to monitoring the Gate Closed feed you may be surprised just how effectively that instrument has been removed. At that point a bit of tweaking the Attenuation and Lookahead a bit to smooth the gate transitions a bit you may find yourself with a quick and effective mix-minus.
>
> With a little imagination, the speed and accuracy of the state-based gating engine makes for a very flexible and powerful dynamics-based gain control element you can drop in anywhere in your mix.

MIO 3d Core processors

The MIO 3d "Core" processors are direct updates from the older 2d platform, skinned with the new 3d GUI.

The Core processors can be thought of as the basic components which formed the basis of the more fully-realized MH Production Bundle and 3d Production processors.

The Core plugs are extremely useful as standalone elements where you just need one job done quickly and efficiently. Along with basic EQ and dynamics, the Core plugs category includes M/S modules, sample-rate-aware delay lines and dithered word-length reduction blocks.

A-hem... It should probably be noted here that *MIOComp* and *MIOStrip* have both retained 100% of the qualities which made them indispensable in the creation of so many unique and powerful DSP Graph processors over the years.

3d Core processors can be found listed in the mixer Plug-in: Processor menu category, and by each processors' functional subcategory (EQ, Dynamics, etc.).

MIOComp

Plug-in categories: **Production**, **Dynamics** and **Mastering**

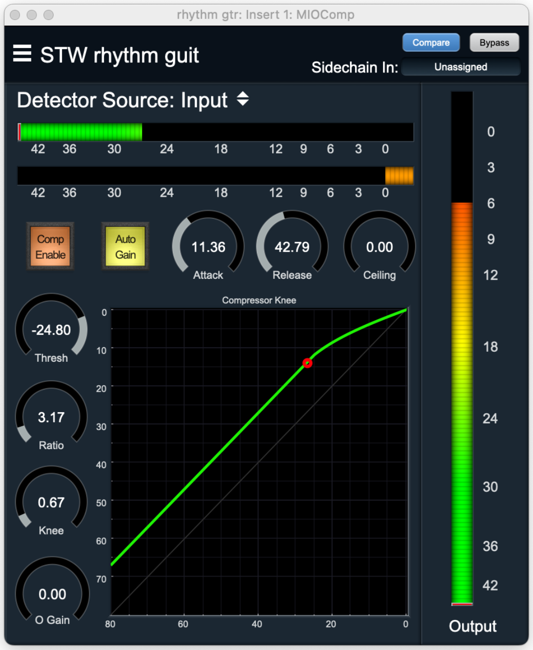


Figure 13.232: MIOComp

Description:

The original dynamics processor behind the MH ChannelStrip "MIO" compressor model.

MIOComp is a flexible, full featured dynamics processor that provides compression/limiting functionality.

The compressor provides a flexible detector that can be driven from the input signal or a sidechain input. The detector computes the input level of the source signal, using the user specified `Attack` and `Release` times to control the responsiveness of gain cell of the compressor.

When the detected level is below the `Threshold`, the gain cell passes the signal with no change. Conversely, when the detected level is above the `Threshold`, the signal is attenuated based upon the `Ratio`, `Knee` and the level of the signal above the `Threshold`.

When the `Knee` is set to 0, MIOComp acts a hard-threshold compressor. For every dB (x) the signal is above the `Threshold`, the output signal increase above `Threshold` by x ÷ `Ratio`.

When the `Knee` is set to a value above 0, the knee will be softened, and the effective `Ratio` will increase gradually from 1 to the specified `Ratio`. The transition is made more gradual as the `Knee` is increased from 0 to 1.

After the compressor gain-reduction has been applied to the signal, an additional make-up gain block applies a user specified `Makeup Gain` to the signal. If the `Auto Gain` has been enabled, the user specified `Makeup Gain` is applied in addition to the compressor computed make-up gain.

After the make-up gain has been applied, the output signal is limited to the level specified by the `Out Ceiling`. This output ceiling limiter captures the effects of fast-transient breakthrough and is especially useful when using MIOComp on highly impulsive signals with large compression ratios.

Parameters:

`Comp Enable` {Off, On}

Enables the compressor.

`Threshold` [-100, 0] dB

Sets the threshold at which the compressor begins to act. There will be no gain reduction applied by the compressor when the input signal is below this threshold.

`Ratio` [1, 100k]

The gain reduction ratio that is applied to the input signal when the signal is above threshold. The actual ratio applied also depends on the knee parameter which can be used to soften the knee and make the applied ratio progressive.

`Attack` [0, 1k] ms

The characteristic attack time of the application of the gain reduction. Longer attack times will cause MIOComp to be less responsive to short transients. Short attack times are required to use MIOComp as a signal limiter, but very short attack times can lead to distortion in the output signal. The attack time is applied in the detector stage of MIOComp.

`Release` [0, 5k] ms

The characteristic release time of the application of the gain reduction. Longer release times will cause MIOComp to be maintain the gain reduction for a longer period of time after transients. Longer release times are appropriate when using MIOComp as a leveling amp/AGC. Alternatively, shorter release times are appropriate when using MIOComp to increase the apparent level of impulsive material. Very short release times can lead to distortion in the output signal. The release time is applied in the detector stage of MIOComp.

`Knee` [-1, 20]

Adjusts the shape of the knee of the compressor as the detector goes above threshold. When knee is set to 0, the transition from no gain reduction to the specified ratio is immediate at the specified threshold. As this parameter is increased, the transition between the 1:1 ratio and the specified ratio is made more and more progressive, thus "softening" the action of the compressor.

`Out Ceiling` [-60, 0] dB

Sets the maximum output level of the compressor. The output signal is hard-clipped, post compressor, at this signal level.

`Makeup Gain` [-30, 30] dB

Sets the additional gain applied to the signal post-compressor, regardless of whether or not there is any gain reduction engaged. Without `Auto Gain`, it allows you to increase the the output level. With `Auto Gain`, it allows you to trim the output level to taste.

`Auto Gain` {Off, On}

When enabled, causes the compressor to automatically adjust the makeup gain to keep full-scale input approximately at full-scale on output. This is a heuristic rule based on extensive real-world testing across a wide variety of program material, intended for quick setup of basic automatic level control. The makeup gain that is auto-generated is only strictly correct when the attack is set to 0; so apply the manual `Makeup Gain` to trim the final output signal as necessary.

`Detector Source` {Input, SC-In}

Chooses the detector input source. Select between the input signal, and the sidechain input.

`Master Bypass` {On, Off}

When enabled, fully bypasses the plugin.

See also:

MIOLimit

MIOLimit

Plug-in categories: **Production**, **Dynamics** and **Mastering**

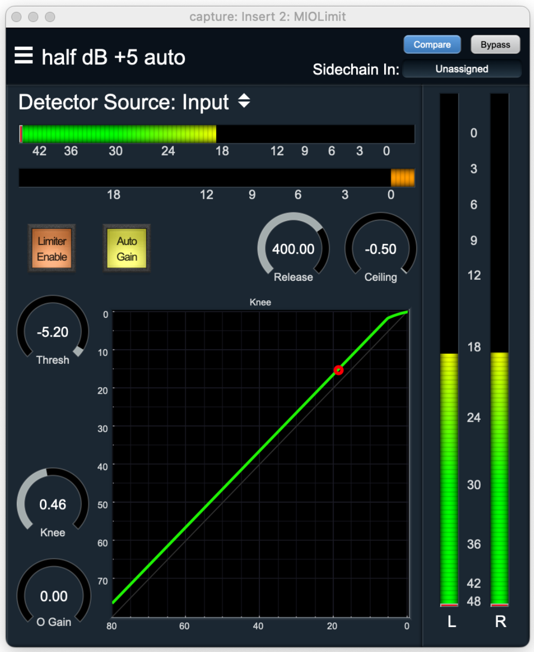


Figure 13.233: MIOLimit

Description:

Straight ahead, transparent limiting stage with independent sidechain input.

MIOLimit is a flexible, full featured dynamics processor that provides signal limiting functionality.

The limiter provides a flexible detector that can be driven from the input signal or a sidechain input. The detector computes the input level of the source signal, using the user specified `Release` time to control the responsiveness of gain cell of the limiter.

When the detected level is below the `Threshold`, the gain cell passes the signal with no change. Conversely, when the detected level is above the `Threshold`, the signal is attenuated based upon the `Knee` and the level of the signal above the `Threshold`.

When the `Knee` is set to 0, MIOLimit acts a hard-threshold brickwall limiter.

When the `Knee` is set to a value above 0, the knee will be softened, and the effective ratio will increase gradually from 1 to infinity. The transition is made more gradual as the `Knee` is increased from 0 to 1.

After the limiter gain-reduction has been applied to the signal, an additional make-up gain block applies a user specified `Makeup Gain` to the signal. If the `Auto Gain` has been enabled, the user specified `Makeup Gain` is applied in addition to the limiter computed make-up gain.

After the make-up gain has been applied, the output signal is limited to the level specified by the `Out Ceiling`. This output ceiling limiter captures the effects of fast-transient breakthrough and is especially useful when using MIOLimit on highly impulsive signals with large compression ratios.

Parameters:

`Enable` {Off, On}

Enables the limiter.

`Threshold` [-60, 0] dB

Sets the threshold at which the limiter begins to act. There will be no gain reduction applied by the limiter when the input signal is below this threshold.

`Release` [0, 5k] ms

The characteristic release time of the application of the gain reduction. Longer release times will cause MIOLimit to be maintain the gain reduction for a longer period of time after transients. Longer release times are appropriate when using MIOLimit as a leveling amp/AGC. Alternatively, shorter release times are appropriate when using MIOLimit to increase the apparent level of impulsive material. Very short release times can lead to distortion in the output signal. The release time is applied in the detector stage of MIOLimit.

`Knee` [0, 1]

Adjusts the shape of the knee of the limiter as the detector goes above threshold. When knee is set to 0, the transition from no gain reduction to the specified ratio is immediate at the specified threshold. As this parameter is increased, the transition between the 1:1 ratio and the specified ratio is made more and more progressive, thus "softening" the action of the limiter.

`Out Ceiling` [-120, 0] dB

Sets the maximum output level of the limiter. The output signal is hard-clipped, post limiter, at this signal level.

`Makeup Gain` [-30, 30] dB

Sets the additional gain applied to the signal post-compressor, regardless of whether or not there is any gain reduction engaged. Without `Auto Gain`, it allows you to increase the the output level. With `Auto Gain`, it allows you to trim the output level to taste.

`Auto Gain` {Off, On}

When enabled, causes the compressor to automatically adjust the makeup gain to keep full-scale input approximately at full-scale on output. This is a heuristic rule based on extensive real-world testing across a wide variety of program material, intended for quick setup of basic automatic level control. The makeup gain that is auto-generated is only strictly correct when the attack is set to 0; so apply the manual `Makeup Gain` to trim the final output signal as necessary.

`SC Source` {Off, On}

Chooses the detector input source. Select between the input signal, and the sidechain input.

`Master Bypass` {On, Off}

When enabled, fully bypasses the plugin.

See also:

MIOComp

MIOStrip

Plug-in categories: **EQ**, **Dynamics** and **Channel Strip**

Figure 13.234: MIOStrip

Description:

MIOStrip provides a complete channel strip processor in one self-contained signal processor. It includes dynamics (gate and compressor with side-chain EQ) and EQ in a flexible, internally routable package.

MIOStrip is a mono or stereo processor and provides a mono sidechain input channel.

The default signal flow through the processor is:

input → gate → compressor → EQ → output.

When `Post EQ` (next to the `Comp Enable`) is engaged, the EQ and compressor order is swapped, and the signal flow becomes:

input → gate → EQ → compressor → output.

The dynamics processor blocks normally detect the level from the signal that is at the input of the dynamics block, optionally pre-processed by the block's associated side-chain EQ. It is also possible to route the side-chain input channel to the input of each block's side-chain EQ and detector. This allows you to use an external key signal to control the dynamics processing in the channel strip. You can select the side-chain source of each dynamics block independently.

Note: Each band of EQ is capable of being configured as one of 6 types of EQ Filter. The types are: Peaking/Parametric, Low Shelf, Hi Shelf, Hi Cut, Low Cut, Band Pass, and RIAA. The first 5 types are parametric filters that allow you to use the Frequency, Gain and Bandwidth controls to adjust position and shape of the filter that is applied. The RIAA type is special. It is used to decode a signal that has been RIAA encoded (for example, the signal on a phonograph disc). This filter type would be selected if you need to decode the output of a phonograph player that is directly connected to the inputs of your interface, without the use of an external phono preamp. Only one band of EQ should be set to RIAA on a channel as using the filter twice will not provide useful results.

Parameters:

`Gate Enable` {Off, On}

Enables the gate. When disabled, the gate is hard-bypassed, and does not consume DSP power.

`Gate Threshold` [-100, 0] dB

Sets the threshold at which the gate begins to act. There will be no gain reduction applied by the gate when the input signal is above this threshold.

`Gate Attack` [0, 1k] ms

The characteristic attack time of the removal of the gain reduction. Longer attack times will cause the gate to cut off the initial transients that bring the gate out of gain reduction. When the attack time is negative, the Gate detector applies an auto-attack to the incoming signal to accurately track fast transients smoothly. The attack time is applied in the detector stage of the Gate block.

`Gate Release` [0, 5k] ms

The characteristic release time of the application of the gain reduction. Longer release times will cause the gate to be less responsive to short transient signal drops. Very short release times can lead to distortion in the output signal. The release time is applied in the detector stage of the Gate block.

`Gate Sidechain EQ SC Source` {Off, On}

Chooses the detector input source. Select between the processed signal and the sidechain input.

`Gate Sidechain EQ Enable` {Off, On}

Enables the EQ on the gate's detector input.

`Gate Sidechain EQ Gain` [-24, 24] dB

Sets the parametric gain on the EQ band. For peaking filters, this is the gain at the peak of the band. For shelving filters, this is the gain at the extreme limit of the filter. This parameter is ignored for the high and low pass filter types as well as the bandpass filter type.

`Gate Sidechain EQ Frequency` [20, 22k] Hz

Sets the characteristic frequency of the filter band.

`Gate Sidechain EQ Bandwidth` [0.01, 2.5] Oct

Sets the bandwidth of the filter band in octaves.

`Gate Sidechain EQ Type` {Peaking/Parametric, Low Shelf, Hi Shelf, Hi Cut, Low Cut, Band Pass, RIAA}

Chooses the type of filter that is applied for the band.

`Comp Enable` {Off, On}

Enables the compressor. When disabled, the compressor is hard-bypassed, and does not consume DSP power.

`Comp Threshold` [-100, 0] dB

Sets the threshold at which the compressor begins to act. There will be no gain reduction applied by the compressor when the input signal is below this threshold.

`Comp Ratio` [1, 1k] :1

The gain reduction ratio that is applied to the input signal when the signal is above threshold. The actual ratio applied also depends on the knee parameter which can be used to soften the knee and make the applied ratio progressive.

`Comp Attack` [0, 1k]

The characteristic attack time of the application of the gain reduction. Longer attack times will cause MIOStrip to be less responsive to short transients. Short attack times are required to use MIOStrip as a signal limiter, but very short attack times can lead to distortion in the output signal. The attack time is applied in the detector stage of MIOStrip.

`Comp Release` [0, 5k] ms

The characteristic release time of the application of the gain reduction. Longer release times will cause MIOStrip to be maintain the gain reduction for a longer period of time after transients. Longer release times are appropriate when using MIOStrip as a leveling amp/AGC. Alternatively, shorter release times are appropriate when using MIOStrip to increase the apparent level of impulsive material. Very short release times can lead to distortion in the output signal. The release time is applied in the detector stage of MIOStrip.

`Comp Knee` [-1, 20]

Adjusts the shape of the knee of the compressor as the detector goes above threshold. When knee is set to 0, the transition from no gain reduction to the specified ratio is immediate at the specified threshold. As this parameter is increased, the transition between the 1:1 ratio and the specified ratio is made more and more progressive, thus "softening" the action of the compressor.

`Comp Makeup Gain` [-30, 30] dB

Sets the additional gain applied to the signal post-compressor, regardless of whether or not there is any gain reduction engaged. Without `Comp Auto Gain`, it allows you to increase the the output level. With `Comp Auto Gain`, it allows you to trim the output level to taste.

`Comp Auto Gain` {Off, On}

When enabled, causes the compressor to automatically adjust the makeup gain to keep full-scale input approximately at full-scale on output. This is a heuristic rule based on extensive real-world testing across a wide variety of program material, intended for quick setup of basic automatic level control. The makeup gain that is auto-generated is only strictly correct when the attack is set to 0; so apply the manual `Comp Makeup Gain` to trim the final output signal as necessary.

`Comp First` {Off, On}

When enabled, the signal will flow through the compressor block first and the output of the compressor block will then flow into the EQ block. If this is not enabled, the signal will run through the EQ first and then through the compressor.

`Comp Sidechain EQ SC Source` {Off, On}

Chooses the detector input source. Select between the processed signal and the sidechain input.

`Comp Sidechain EQ Enable` {Off, On}

Enables the EQ on the compressor's detector input.

`Comp Sidechain EQ Gain` [-24, 24] dB

Sets the parametric gain on the EQ band. For peaking filters, this is the gain at the peak of the band. For shelving filters, this is the gain at the extreme limit of the filter. This parameter is ignored for the high and low pass filter types as well as the bandpass filter type.

`Comp Sidechain EQ Frequency` [20, 22k] Hz

Sets the characteristic frequency of the filter band.

`Comp Sidechain EQ Bandwidth` [0.01, 2.5] Oct

Sets the bandwidth of the filter band in octaves.

`Comp Sidechain EQ Type` {Peaking/Parametric, Low Shelf, Hi Shelf, Hi Cut, Low Cut, Band Pass, RIAA}

Chooses the type of filter that is applied for the band.

`EQ Enable` {Off, On}

Enables the EQ block. When disabled, the EQ is hard-bypassed, and does not consume DSP power.

`EQ Gain` [-80, 10] dB

Master gain applied to the signal at the output of the EQ block. Since the EQ is implemented as a high-resolution floating point process, you can bring the output signal of the EQ out of any degree of clipping using this parameter.

`EQ Band Enable` {Off, On}

Enables the band. When disabled, the band is hard-bypassed.

`EQ Band Gain` [-24, 24] dB

Sets the parametric gain on the EQ band. For peaking filters, this is the gain at the peak of the band. For shelving filters, this is the gain at the extreme limit of the filter. This parameter is ignored for the high and low pass filter types as well as the bandpass filter type.

`EQ Band Frequency` [20, 22k] Hz

Sets the characteristic frequency of the filter band.

`EQ Band Bandwidth` [0.01, 2.5] Oct

Sets the bandwidth of the filter band in octaves.

`EQ Band Type` {Peaking/Parametric, Low Shelf, Hi Shelf, Hi Cut, Low Cut, Band Pass, RIAA}

Chooses the type of filter that is applied for the band.

`Master Bypass` {On, Off}

When enabled, hard-bypasses the entire plug-in, and substantially reduces the amount of DSP power used.

MIOEq 6 Band

Plug-in categories: **Production**, **EQ**, and **Mastering**

Figure 13.235: MIOEQ6

Description:

MIOEq 6 Band implements a mono or stereo 6-band IIR EQ processor. Each band is fully parametric, and can be set to any of the filter types. All the parameters of all the bands can be set to any of valid settings (as detailed below). The composite EQ curve is applied to the input signal, and the master gain is then applied to the processed signal.

Note: Each band of EQ is capable of being configured as one of 6 types of EQ Filter. The types are: Peaking/Parametric, Low Shelf, Hi Shelf, Hi Cut, Low Cut, Band Pass, and RIAA. The first 5 types are parametric filters that allow you to use the Frequency, Gain and Bandwidth controls to adjust position and shape of the filter that is applied. The RIAA type is special. It is used to decode a signal that has been RIAA encoded (for example, the signal on a phonograph disc). This filter type would be selected if you need to decode the output of a phonograph player that is directly connected to the inputs of your interface, without the use of an external RIAA phono preamp. Only one band of EQ should be set to RIAA on a channel as using the filter twice will not provide useful results.

Master Parameters:

`Enable` {Off, On}

Enables the EQ block. When disabled, the EQ (including the master gain) is hard-bypassed, and does not consume DSP power.

`Gain` [-80, 10] dB

Master gain applied to the signal at the output of the EQ block. Since the EQ is implemented as a high-resolution floating point process, you can bring the output signal of the EQ out of any degree of clipping using this parameter.

`Master Bypass` {On, Off}

Disables the EQ block. When disabled, the EQ (including the master gain) is hard-bypassed, and does not consume DSP power.

Per-filter Parameters:

`Enable` {Off, On}

Enables the band. When disabled, the band is hard-bypassed.

`Gain` [-24, 24] dB

Sets the parametric gain on the EQ band. For peaking filters, this is the gain at the peak of the band. For shelving filters, this is the gain at the extreme limit of the filter. This parameter is ignored for the high and low pass filter types as well as the bandpass filter type.

`Frequency` [20, 22k] Hz

Sets the characteristic frequency of the filter band.

`Bandwidth` [0.01, 2.5] Oct

Sets the bandwidth of the filter band in octaves.

`Type` {Peaking/Parametric, Low Shelf, Hi Shelf, Hi Cut, Low Cut, Band Pass, RIAA}

Chooses the type of filter that is applied for the band.

See also:

MIOEq 12 Band

MIOEq 12 Band

Plug-in categories: **Production**, **EQ**, and **Mastering**

Figure 13.236: MIOEQ12

Description:

MIOEq 12 Band implements a mono or stereo 12-band IIR EQ processor. Each band is fully parametric, and can be set to any of the filter types. All the parameters of all the bands can be set to any of valid settings (as detailed below). The composite EQ curve is applied to the input signal, and the master gain is then applied to the processed signal.

Note: Each band of EQ is capable of being configured as one of 6 types of EQ Filter. The types are: Peaking/Parametric, Low Shelf, Hi Shelf, Hi Cut, Low Cut, Band Pass, and RIAA. The first 5 types are parametric filters that allow you to use the Frequency, Gain and Bandwidth controls to adjust position and shape of the filter that is applied. The RIAA type is special. It is used to decode a signal that has been RIAA encoded (for example, the signal on a phonograph disc). This filter type would be selected if you need to decode the output of a phonograph player that is directly connected to the inputs of your interface, without the use of an external RIAA phono preamp. Only one band of EQ should be set to RIAA on a channel as using the filter twice will not provide useful results.

Master Parameters:

`Enable` {Off, On}

Enables the EQ block. When disabled, the EQ (including the master gain) is hard-bypassed, and does not consume DSP power.

`Gain` [-80, 10] dB

Master gain applied to the signal at the output of the EQ block. Since the EQ is implemented as a high-resolution floating point process, you can bring the output signal of the EQ out of any degree of clipping using this parameter.

`Master Bypass` {On, Off}

Disables the EQ block. When disabled, the EQ (including the master gain) is hard-bypassed, and does not consume DSP power.

Per-filter Parameters:

`Enable` {Off, On}

Enables the band. When disabled, the band is hard-bypassed.

`Gain` [-24, 24] dB

Sets the parametric gain on the EQ band. For peaking filters, this is the gain at the peak of the band. For shelving filters, this is the gain at the extreme limit of the filter. This parameter is ignored for the high and low pass filter types as well as the bandpass filter type.

`Frequency` [20, 22k] Hz

Sets the characteristic frequency of the filter band.

`Bandwidth` [0.01, 2.5] Oct

Sets the bandwidth of the filter band in octaves.

`Type` {Peaking/Parametric, Low Shelf, Hi Shelf, Hi Cut, Low Cut, Band Pass, RIAA}

Chooses the type of filter that is applied for the band.

See also:

MIOEq 6 Band

MIO M/S Decoder

Plug-in category: **Spatial**

Figure 13.237: MIO M/S Decoder

Description:

M/S Decoder is a Mid/Side Decoder. It can be used to decode M/S microphone capsule inputs and/or recorded material that was previously encoded as Mid/Side.

Parameters:

`Mid Invert` {Off, On}

When enabled, inverts the 1st input channel before processing.

`Side Invert` {Off, On}

When enabled, inverts the 2nd input channel before processing.

`Width` [0, 1]

Adjusts the width of the transformed field. When this parameter is 0, the ouput will be mono. When this is 1, the full stereo width is represented.

`Master Bypass` {On, Off}

Bypasses all of the processing.

MIO M/S Processor

Plug-in category: **Spatial**

Figure 13.238: MIO M/S Processor

Description:

MIO M/S is a flexible Mid/Side channel processor. It can be used to decode material that was previously encoded as Mid/Side. It can also be used to encode a left/right stereo field into a mid/side signal. In addition to being used to encode or decode existing signals, if you also use two instances of MIO M/S to build a transcoding signal chain in the 3d DSP graph.

Since you have access to the individual Mid and Side signals after encoding, you can insert additional processing blocks between the encoder and the decoder, including EQ, delay, compression and limiting. This allows you to build signal chains that alter the overall size, nature and depth of the stereo field.

Parameters:

`Enable` {Off, On}

When engaged, enables all the processing of the module.

`Mid Invert` {Off, On}

When enabled, inverts the 1st input channel before processing.

`Side Invert` {Off, On}

When enabled, inverts the 2nd input channel before processing.

`Mode` {Decode (Constant Amplitude), Decode (Constant Power), Encode (Constant Amplitude), Encode (Constant Power), LR->MS->LR}

Selects the processing mode.

Decode (Constant Amplitude)

Makes input 1 be the Mid Channel and input 2 the Side Channel. In this mode, the processor decodes the M/S input to Left and Right outputs, keeping the total amplitude constant. The center point is output compensated to avoid clipping.

Decode (Constant Power)

Makes input 1 be the Mid Channel and input 2 the Side Channel. In this mode, the processor decodes the M/S input to Left and Right outputs, keeping the total power constant (the amplitude increases, and may go over 1). It is possible to clip audio using this setting.

Encode (Constant Amplitude)

Makes input 1 be the Left Channel and input 2 the Right Channel. In this mode, the processor encodes the Left and Right inputs to M/S outputs, keeping the total amplitude constant. The center point is output compensated to avoid clipping.

Encode (Constant Power)

Makes input 1 be the Left Channel and input 2 the Right Channel. In this mode, the processor encodes the Left and Right inputs to M/S outputs, keeping the total power constant (the amplitude increases, and may go over 1). It is possible to clip audio using this setting.

LR->MS->LR

Makes input 1 be the Left Channel and input 2 the Right Channel. In this mode, the processor encodes the Left and Right inputs to an internal M/S representation, applies the width and rotation transformations in the M/S representation and then decodes the transformed M/S channels back to Left and Right outputs.

`Width` [0, 1]

Adjusts the width of the transformed field. When this parameter is 0, the ouput will be mono. When this is 1, the full stereo width is represented.

`Rotation` [-180, 180] °

Rotates the stereo field by the specified angle.

`Master Bypass` {On, Off}

Bypasses all of the processing.

MIO Delays (15k and 96k)

Plug-in category: **Delay**

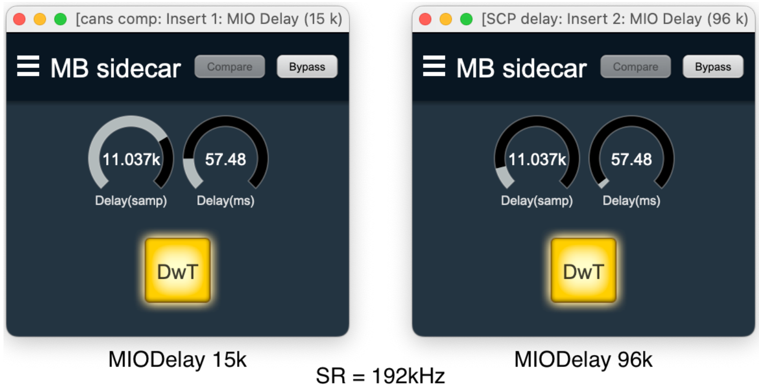


Figure 13.239: MIO Delay (15k) and (96k)

SR	MIO Delay (15k) *maximum delay*	MIO Delay (96k) *maximum delay*
44.1kHz	348.28 ms	2.18 sec
48 kHz	319.98 ms	2 sec
88.2 kHz	174.14 ms	1.09 sec
96 kHz	159.99 ms	999.99 ms
176.4 kHz	87.07 ms	544.21 ms
192 kHz	79.99 ms	500.00 ms

Table 13.1. MIODelay (15k) and (96k) maximum delay times by sample rate

Description:

Provides an integer sample delay from the input to the output. Changes in the delay time are interpolated, allowing you to change the delay time with no glitches. For large changes in the delay time, the interpolation has the effect of shifting the pitch of the delayed signal as the delay time is changing.

Parameters:

`Delay(samp)` [0, 15.359 k / 95.999 k] samples

Specifies the delay through the process block in samples.

`Delay(ms)` [0, 319.98 / 2.0 k] ms

Specifies the delay through the process block in milliseconds.

`DwT (Drive with Time)` {On, Off}

When engaged (button is lit/colored, as shown above), ensures that the `Delay(samp)` parameter will automatically track the sample rate such that the `Delay(ms)` time remains consistent regardless of the system sample rate.

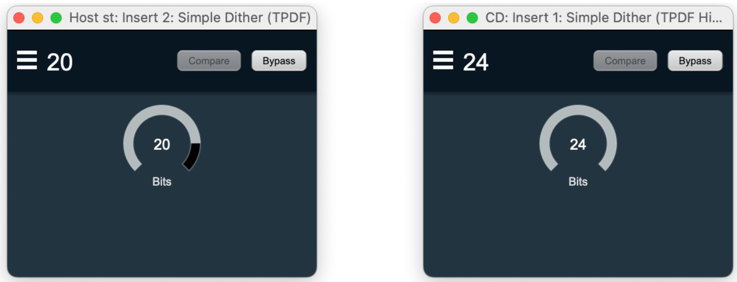
When `DwT` is disengaged (button is dark/uncolored) the `Delay(samp)` parameter will remain unchanged, and the `Delay(ms)` time will vary based on the system sample rate.

`Master Bypass` {On, Off}

When engaged sets the delay through the block to 0.

Simple Dither (TPDF and TPDF Hipass)

Plug-in category: **Dither**



Simple Dither (TPDF) @ 20 bits   Simple Dither (TPDF Hipass) @ 24 bits

Figure 13.240: Simple Dither (TPDF) and (TPDF Hipass)

Description:

Simple Dither (TPDF) applies white power spectrum dither to the input signal at the bit-depth specified by the `Bits` parameter, and truncates the dithered signal to the specified bit depth.

Simple Dither (TPDF Hipass) applies high-pass filtered dither noise to the input signal at the bit-depth specified by the `Bits` parameter, and truncates the dithered signal to the specified bit depth.

The initial dither noise for both modules is created with 2 LSB pk-pk triangular probability distribution function white noise, but only the (TPDF Hipass) is additionally filtered with a high-pass characteristic. Dither noise is phase-decorrelated per audio channel.

Note that Simple Dither (TPDF) will only generate dither noise when there is active modulated signal present. When there is no input signal, Simple Dither (TPDF) is silent.

Simple Dither (TPDF Hipass), on the other hand will output dither noise commensurate with the `Bits` parameter setting at all times unless `Master Bypass` is engaged.

Dither should be considered at any transition stage where word length would otherwise be truncated from a higher precision to 24 bits resolution or lower. Let your ears be your guide. While 32 bit float is still only 24 bits of actual data precision, the application of dither noise even at that low level can often perceptably smooth the transition.

Parameters:

`Bits` [1, 24] bits

Used to specify the final the bit depth of the dithered and truncated signal.

`Master Bypass` {On, Off}

Polarity Invert

Plug-in category: **Math**

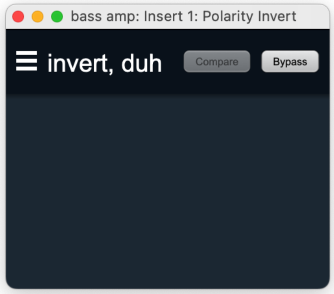


Figure 13.241: Polarity Invert

Description:

Polarity Invert mathematically reverses the signal polarity of the incoming signal. Polarity inversion can be inserted at any point in the Mixer strip Insert signal path or as a Graph process block.

Parameter:

`Master Bypass` {On, Off}

Deprecated [Legacy] plug-ins

The 3d processors listed with a [Legacy] tag in the plug-ins selection UI have been directly supplanted by the inclusion of their 3d Production Processor counterparts.

These plug-ins will remain available for the foreseeable future to provide backwards compatibility, but we recommend using the newer Production versions moving forward.

HaloVerb [Legacy]

Description:

HaloVerb is a mono to mono or stereo to stereo reverb that can be used as a send processor or an inline processor. This version of HaloVerb is optimized for 3d and will run at all sample rates.

Parameters:

`Room Size` [65, 99] M

Sets the effective size of the room; can also be thought of as Reverb Time. Larger numbers make the room bigger and the reverb time longer.

`Decay` [0, 100] %

Sets the damping of the regenerative filters of the room. Larger values roll the high–end off faster.

`Reverb dB` [-inf, 9.54243] dB

Sets the level of the reverb signal mixed into the output.

`Direct dB` [-inf, 9.54243] dB

Sets the level of the direct (dry) signal mixed into the output.

`Width` [0, 100] %

Controls how "Stereo" the output of the reverb is. Setting this to 0 will generate a mono reverb output. Setting this to 100 will provide maximally stereo reverb output.

`Diffusion` [0, 1] %

Controls how much diffusion is applied to reverberant field.

`PreDelay` [0, 45] ms

Sets how much delay (in milliseconds) is applied to the reverb signal before it is mixed in with the dry signal.

`Cutoff` [20, 20k] Hz

Sets the cutoff frequency (in Hz) of the high-end contouring filter.

`HiDamp` [-40, 0] dB

Sets the gain (in dB) of the the high-end contouring filter.

`Master Bypass` {On, Off}

When bypassed, the reverb passes the input through un-touched.

Character [Legacy]

Description:

Character is a mono inline processor which applies any of a selection of analog circuit-modeled non-linear distortion maps. The various character models offer unique, high quality harmonic and tonal coloration to your signal. Choose "None" to disable character modeling.

The "soft sat" variations are particularly good at providing the "analog glue" that many engineers want for their final mixes. The "sat" stands for "saturation", such as you would get with analog tape. The "soft" part of the name indicates that the level is *lower* after processing. You may want to use the Output Gain to make up the lost gain.

While most of the models can be applied to every channel, overuse of the saturation models can lead to undesired effects.

Parameters:

`Drive` [-20, 20]

The Drive control applies ±20 dB of gain to the signal before it goes to the modeling section. This allows you attenuate the signal for a subtle effect, or boost to get a more dramatic effect.

As gain is changed before the modeling stage, the inverse gain is applied after. For example, if you set the Drive control to +6 dB, the gain is boosted 6 dB before the model and cut by 6 dB after. This allows you to "push" the modeling section with no increase in output level. You may also hit the model with less signal by setting the Drive control to a negative value without loss of overall volume.

You may still experience in increase in signal level when using positive Drive gain with some models; you can use the Output Gain control to correct this.

Please note that the Drive parameter may have a greater range than you need for a given model, particularly when Auto Drive is engaged.

`Character Model` {preset list}

The following list describes the device from which each model was derived:

- None: No modeling is applied.
- Transformer: Applies the harmonic distortion signature of a transformer-coupled input.
- Valve: A tube-based EQ input stage.
- FET: Model of a solid state (FET transistor) front end.
- SoftSat: Tube-based EQ with saturation. This is the 'SoftSat' from the MH Production Bundle Character plug-in.
- SoftSat Gain: Tube-based EQ with saturation, gain adjusted so the saturation point lines up with 0dB (e.g. the signal will get louder). This 'SoftSat' is specific to the 3d DSP.
- Boutique Tube: Hand-made tube mic pre.
- American Transformer 1: A variation of the "Transformer" model.
- American Transformer 2: Second variation of the "Transformer" model.
- California Tube Mic: American designed tube mic pre.

- California Tube Line: American designed tube line input.
- Modern Tube DI: Mastering quality tube DI.
- Modern Tube EQ: Mastering quality EQ.
- Modern Tube Soft Sat: Mastering quality EQ with saturation.
- Modern Tube LG: A tube mic pre with a low gain setting.
- Modern Tube MG: A tube mic pre with a medium gain setting.
- Modern Tube HG: A tube mic pre with a high gain setting.
- Modern Tube Sym: Mastering quality EQ
- Modern Tube Soft Sat: Mastering quality tube mic pre with saturation.
- Classic British Mic Pre: A favorite large console mic pre.
- American Solid State: FET mastering EQ.
- California Vocal Box: Transformer coupled tube vocal processor.
- California Vocal Box Drive: Transformer coupled tube vocal processor with increased gain.
- British Mic Pre Clone: A popular clone of a favorite British mic pre.

`Auto Drive` {On, Off}

When creating a new instance of Character, Auto Drive is turned off by default. When engaged, the Auto Drive button enables a detector that will automatically sense lower level input signals and apply more drive gain that varies with the signal level.

The effect of enabling Auto Drive is to have a more consistent amount of distortion applied for all input levels. As a result, the effect of the various circuit models becomes more pronounced with Auto Drive engaged. In some instances can lead to heavy distortion. Turn the drive knob to the left to compensate, or leave Auto Drive off for a more subtle effect.

`Gain` [-20, 20] dB

The Output Gain applies ±20 dB of gain to the signal after it has been through the modeling stage. This can be used to increase the final output level after using a soft saturation model or otherwise gain-stage the signal for the next processor.

`Master Bypass` {On, Off}

When bypassed, the input passes through un-touched.

MHCharacter [Legacy]

Description:

MHCharacter has the the same internal processing as Character (listed above), but it wraps the processing in an oversampling block, which can make a difference for heavier distortion (especially with higher drive). MHCharacter does use more DSP power than Character (2x to 3x).

Like Character, MH Character is a mono inline processor which applies any of a selection of analog circuit-modeled non-linear distortion maps. The various character models offer unique, high quality harmonic and tonal coloration to your signal. Choose “None” to disable character modeling.

The “soft sat” variations are particularly good at providing the “analog glue” that many engineers want for their final mixes. The “sat” stands for “saturation”, such as you would get with analog tape. The “soft” part of the name indicates that the level is *lower* after processing. You may want to use the Output Gain to make up the lost gain.

While most of the models can be applied to every channel, overuse of the saturation models can lead to undesired effects.

Parameters:

`Drive` [-20, 20]

The Drive control applies ±20 dB of gain to the signal before it goes to the modeling section. This allows you attenuate the signal for a subtle effect, or boost to get a more dramatic effect.

As gain is changed before the modeling stage, the inverse gain is applied after. For example, if you set the Drive control to +6 dB, the gain is boosted 6 dB before the model and cut by 6 dB after. This allows you to “push” the modeling section with no increase in output level. You may also hit the model with less signal by setting the Drive control to a negative value without loss of overall volume.

You may still experience in increase in signal level when using positive Drive gain with some models; you can use the Output Gain control to correct this.

Please note that the Drive parameter may have a greater range than you need for a given model, particularly when Auto Drive is engaged.

`Character Model` {preset list}

The following list describes the device from which each model was derived:

- None: No modeling is applied.
- Transformer: Applies the harmonic distortion signature of a transformer-coupled input.
- Valve: A tube-based EQ input stage.
- FET: Model of a solid state (FET transistor) front end.
- SoftSat: Tube-based EQ with saturation. This is the ‘SoftSat’ from the MH Production Bundle Character plug-in.
- SoftSat Gain: Tube-based EQ with saturation, gain adjusted so the saturation point lines up with 0dB (e.g. the signal will get louder). This ‘SoftSat’ is specific to the 3d DSP.
- Boutique Tube: Hand-made tube mic pre.
- American Transformer 1: A variation of the “Transformer” model.

- American Transformer 2: Second variation of the “Transformer” model.
- California Tube Mic: American designed tube mic pre.
- California Tube Line: American designed tube line input.
- Modern Tube DI: Mastering quality tube DI.
- Modern Tube EQ: Mastering quality EQ.
- Modern Tube Soft Sat: Mastering quality EQ with saturation.
- Modern Tube LG: A tube mic pre with a low gain setting.
- Modern Tube MG: A tube mic pre with a medium gain setting.
- Modern Tube HG: A tube mic pre with a high gain setting.
- Modern Tube Sym: Mastering quality EQ
- Modern Tube Soft Sat: Mastering quality tube mic pre with saturation.
- Classic British Mic Pre: A favorite large console mic pre.
- American Solid State: FET mastering EQ.
- California Vocal Box: Transformer coupled tube vocal processor.
- California Vocal Box Drive: Transformer coupled tube vocal processor with increased gain.
- British Mic Pre Clone: A popular clone of a favorite British mic pre.

`Auto Drive` {On, Off}

When creating a new instance of Character, Auto Drive is turned off by default. When engaged, the Auto Drive button enables a detector that will automatically sense lower level input signals and apply more drive gain that varies with the signal level.

The effect of enabling Auto Drive is to have a more consistent amount of distortion applied for all input levels. As a result, the effect of the various circuit models becomes more pronounced with Auto Drive engaged. In some instances can lead to heavy distortion. Turn the drive knob to the left to compensate, or leave Auto Drive off for a more subtle effect.

`Gain` [-20, 20] dB

The Output Gain applies ±20 dB of gain to the signal after it has been through the modeling stage. This can be used to increase the final output level after using a soft saturation model or otherwise gain-stage the signal for the next processor.

`Master Bypass` {On, Off}

When bypassed, the input passes through un-touched.

TransientControl [Legacy]

Description:

TransientControl is a dynamics shaping processor. By using the `Transient` and `Sustain` controls, it is possible to accentuate the transient or sustained components of a signal; for example, by increasing the Transient and decreasing the Sustain of a snare drum, you can accentuate the hit and decrease the ring. Conversely, by decreasing the Transient of an electric bass, you can remove the attack of a pick. The `Gain` control can be used to apply makeup gain (or padding) after processing.

The top row consists of the main `Transient`, `Sustain` and `Gain` controls along with input level metering.

Figure 13.242: TransientControl: main controls, Input Level and Gain History meters (at default)

The `Gain History Meter` display to the right shows the amount of gain increase from the `Transient` and `Sustain` adjustments, while activity below the horizon shows the amount of gain reduction.
The scale of this display can be adjusted by control-clicking the meter UI.

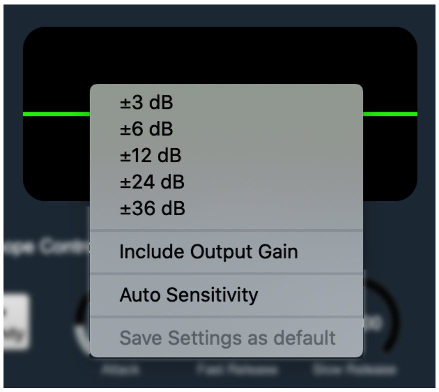


Figure 13.243: Gain History Meter contextual menu

This menu will let you:
- Set the meter to show a gain range from ±3dB to ±36dB
- Configure whether the gain adjustment from the `Gain` control is factored into the meter display
- Enable / disable `Auto Sensitivity`
- Save your current metering preferences as default

Transient detector

Along the bottom are the `Fast Attack`, `Slow Attack`, `Release`, `Attack`, `Fast Release` and `Slow Release` controls. These are used to fine tune the transient and sustain detectors. For most purposes the default values will work well, but these will allow you to create dramatic effects or fine tune the processor for your needs.

`Auto-Sensitivity` puts a compressor/limiter stage into the detector chain, helping to ensure that the signal going into the detector is stable and the transients are processed consistently.

The `Fast Attack` and `Slow Attack` define the upper and lower limits of the Transient detector envelope; *Fast* is the top and *Slow* is the bottom. If the Fast Attack is set lower than the Slow Attack the operation of the Transient detector is reversed; setting the Transient control to *decrease* will actually *increase* the level and vice versa. The audio that falls within the detector envelope is boosted or cut as set by the `Transient` control. The Fast and Slow Attack controls range from 0-100 ms.

For example, when the difference between the Fast and Slow parameters is small (in the example below they are 10 and 30), less material is processed:

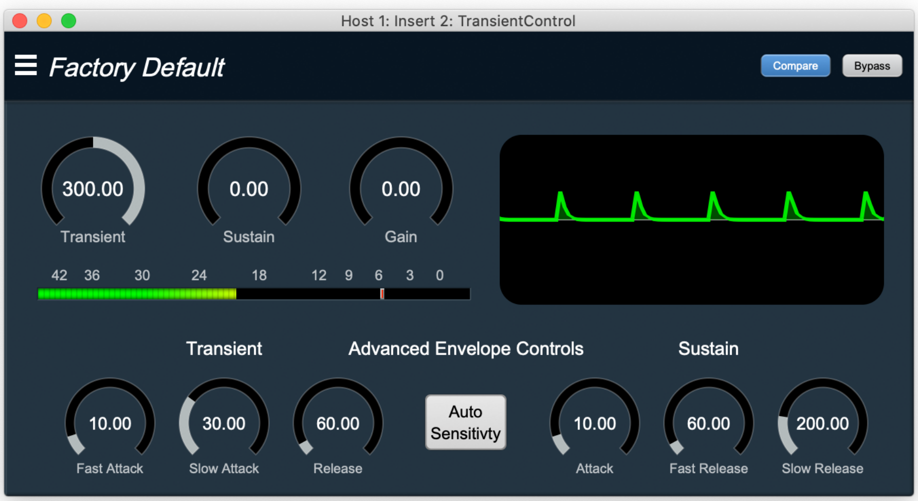


Figure 13.244: Small Detector Window

When the difference between the Fast and Slow Attacks are larger (1 and 60), more audio is processed:

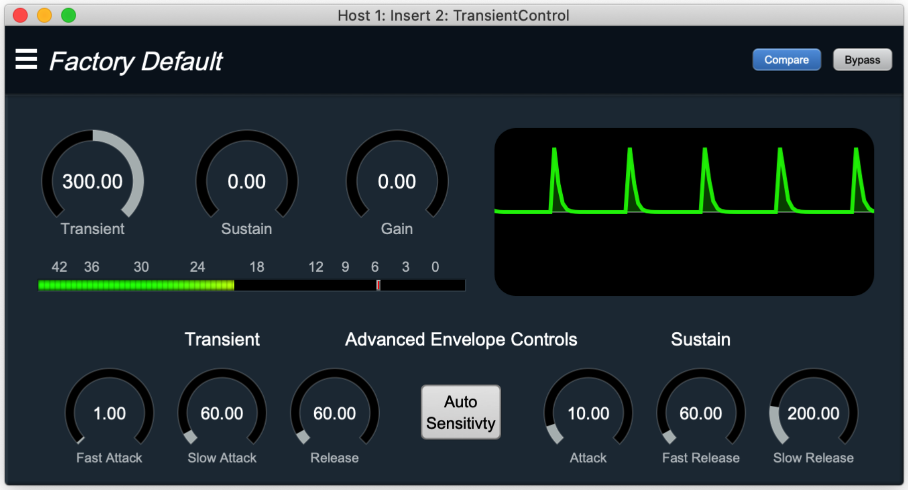


Figure 13.245: Large Detector Window

The Release control sets how long the envelope stays open; this allows you to adjust how quickly the transient ends after the signal transitions from transient to sustain:

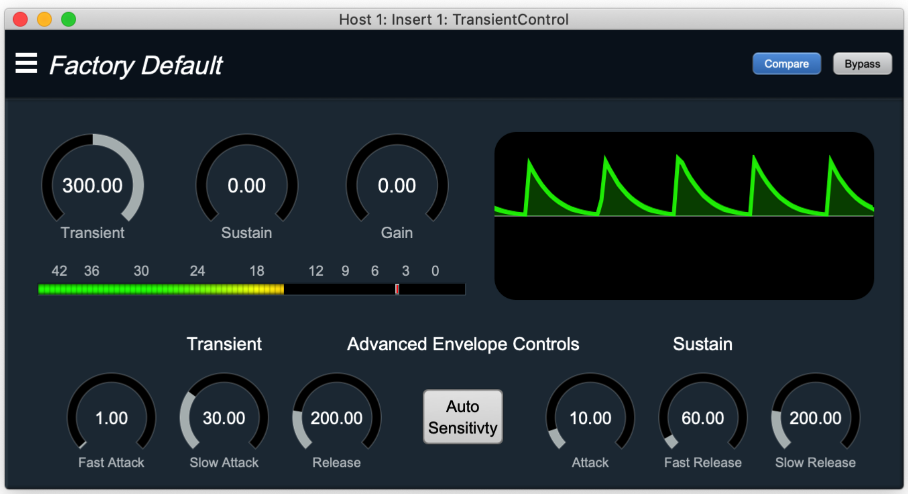


Figure 13.246: Large Release Value

The Release control ranges from 0-1000 ms.

In the figure above, the Transient Attack controls are the same as before but the Release has been made longer (from 60 to 200). This allows more audio to be processed.

If you make the Release control too short, you may find the applied processing sounds choppy.

Sustain detector

The Fast Release and Slow Release define the upper and lower limits of the Sustain detector envelope; Fast is the top and Slow is the bottom. If the Slow Release is set lower than the Fast Release the operation of the Sustain detector is reversed; setting the Sustain control to "Decrease" will actually *increase* the level and vice versa. The audio that falls within the detector envelope is boosted or cut as set by the Sustain control. The Fast and Slow Attack controls range from 0-1000 ms.

If you make either of the Release controls too short, you may find the applied processing sounds choppy.

For example, when the difference between the Fast and slow parameters is small (here they are 60 and 200), less material is processed:

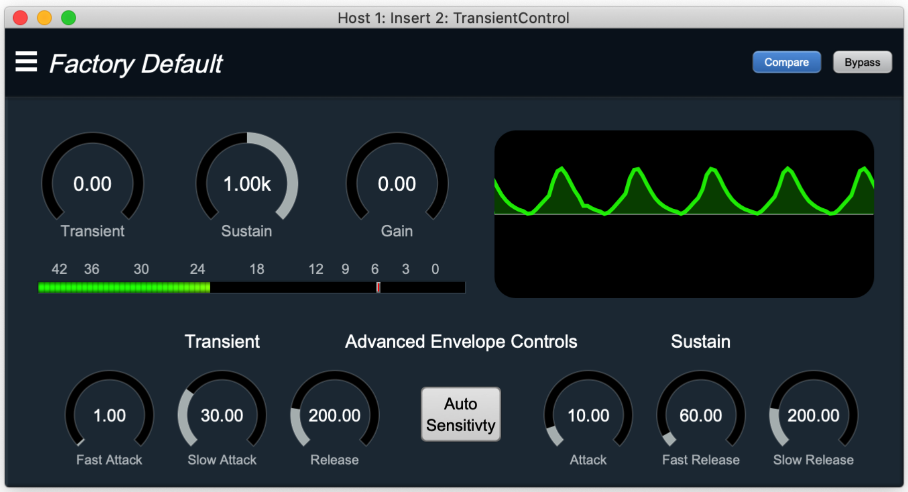


Figure 13.247: Small Detector Window

When the difference between the Fast and Slow Attacks are larger (1 and 300), more audio is processed:

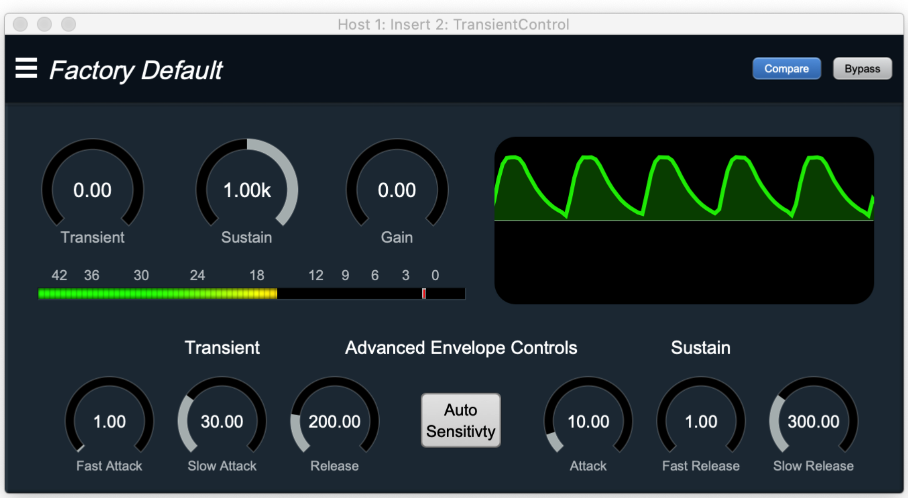


Figure 13.248: Large Detector Window

The Attack control sets how quickly the envelope opens; this allows you to adjust how long after the transient occurs that the sustain begins:

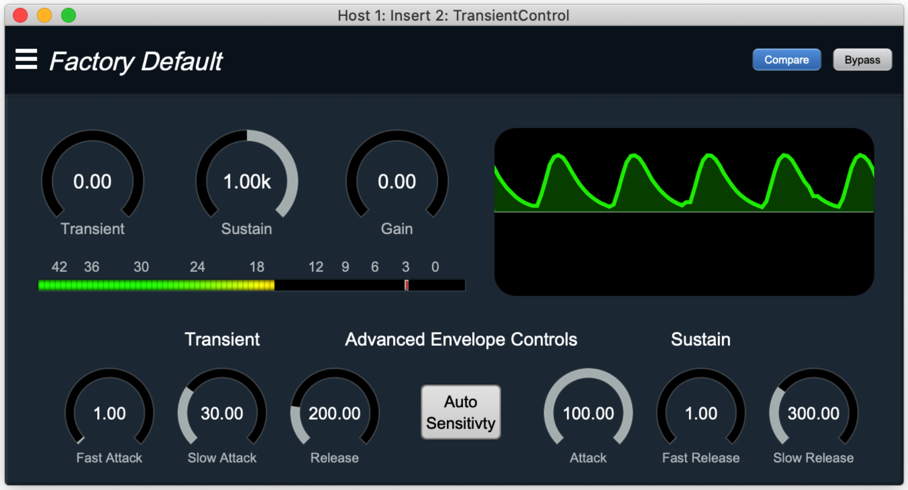


Figure 13.249: Large Attack Value

The Attack control ranges from 0-100 ms.

In the figure above, the Sustain Release controls are the same as before but the Attack has been made longer (from 10 to 100). This allows less audio to be processed.

Parameters:

`Transient` [-300, 300] %

Controls the gain applied to the transient portion of the signal.

`Sustain` [-1k, 1k] %

Controls the gain applied to the sustain portion of the signal.

`Gain` [-24, 24] dB

Master output gain in dB.

`Fast Attack` [0, 100] ms

Sets the fast attack time of the transient detector.

`Slow Attack` [0, 100] ms

Sets the slow attack of the transient detector.

`Release` [0, 1k] ms

Sets the release of the transient detector.

`Attack` [0, 100] ms

Sets the attack of the sustain detector.

`Fast Release` [0, 1k] ms

Sets the fast release of the sustain detector.

`Slow Release` [0, 1k] ms

Sets the slow release of the sustain detector.

`Bypass` {On, Off}

When enabled, fully bypasses the plugin.

`Display Gain Range` {±3, ±6, ±12, ±24, ±36}

Sets the gain range for the Gain History process metering display.

`Include Output Gain` {On, Off}

Sets whether the gain from the `Gain` control is figured into the process metering.

`Auto Sensitivity` {On, Off}

Enables a comp/limiter levelling stage in the detector circuit such that regardless of the source input level, the signal reaching the detector is stable and consistent.

14. MIOConsole3d Preferences

Accessing the preferences

MIOConsole3d has a number of preferences that you can set to control aspects of its behavior. These preferences are accessed via the *MIOConsole3d > Preferences…* command (or via ⌘, (Command + comma) key sequence). The Preferences command opens this window:

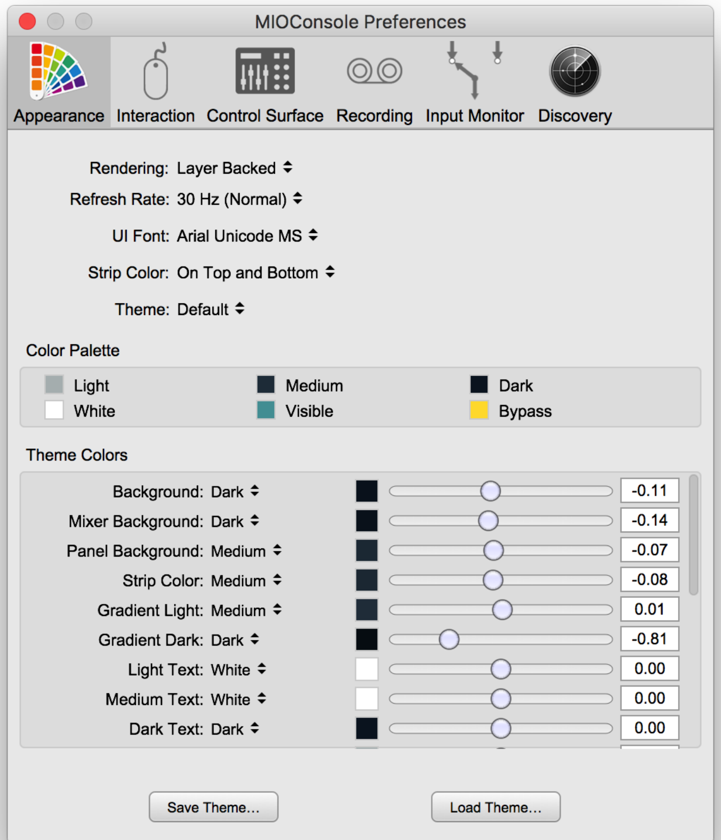


Figure 14.1: MIOConsole3d Preferences

Across the top of the window are five selector tabs dividing the various Preferences categories: Appearance, Interaction, Control Surface, Recording and Front Panel. These are primarily global preferences controlling behaviors within the MIOConsole3d UI and across all 3d devices attached to your Host computer. Front Panel preferences include some settings which are box-specific.

Appearance Preferences

- **Rendering**

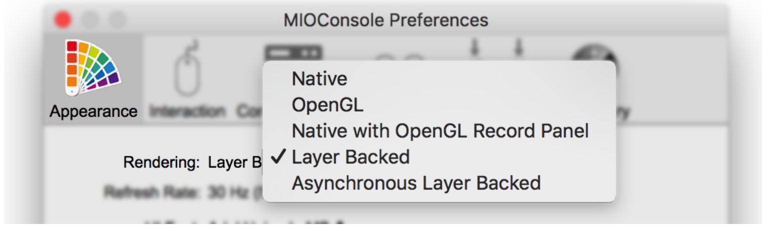


Figure 14.2: Rendering submenu

Most modern Macintoshes come equipped with "Native" CPU graphics rendering, plus "OpenGL", "Layer Backed" and "Asynchronous Layer Backed" GPU rendering engines. In an ideal world, the software would run optimally with just one of them, but in our testing we see variations in performance for the different engines between different versions of macOS and different macOS hardware and GPUs… enough of a difference to make it worth breaking out the controls to the user.

The best settings for most users fall as follows:
- Intel machines: *Layer Backed*
- Intel machines on older OS's: *OpenGL*
- Apple Silicon machines: *Layer Backed*

With Macs running OS's prior to macOS 10.14 (Mojave), "Native with OpenGL Record Panel" might also be worth a shot, traditionally offering a nice compromise between the smoother text and outline rendering of the Native engine, with the more complex scrolling waveforms in Session drawn more smoothly and efficiently by the OpenGL rendering engine.

Switching between *Native*, *Layer Backed* and *Asynchronous Layer Backed* should be immediate, however on some older machines and OS's you will need to quit and re-launch the MIOConsole3d application when switching to and from *OpenGL*.

Note: The *Rendering* and *Refresh Rate* settings are global preferences and are not saved as part of a Theme.

- **Refresh Rate**

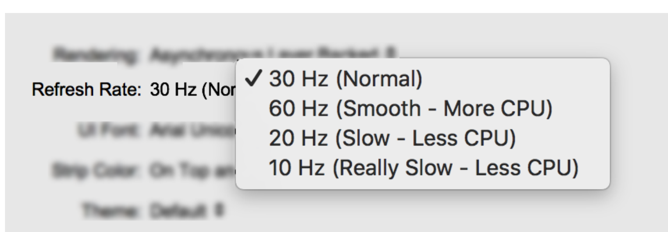


Figure 14.3: Refresh Rate submenu

Refresh Rate controls how quickly MIOConsole3d will attempt to update the meters and the Playhead. 30 FPS is the default setting. You can set it to 60 FPS for buttery smooth graphics at the expense of higher CPU load, or you can reduce it to 20 or 10 FPS for a lower CPU load. 20 FPS is pretty reasonable; 10 FPS is visibly choppier but still gets the job done.

Refresh Rate is applied immediately, so feel free to tinker at will.

- **Theme**

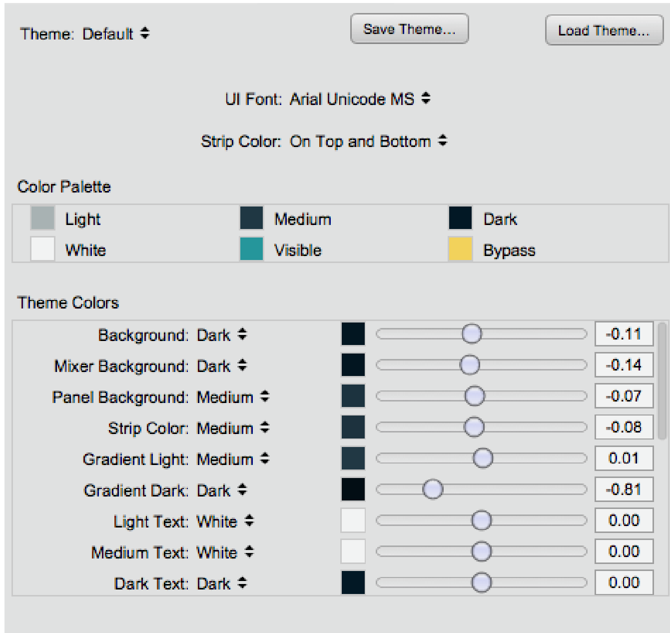


Figure 14.4: Themes: Preset Menu, Save and Load Themes, Customization

The *Theme* pop-up menu includes the Default theme plus 24 factory presets, and is where you recall your personalized themes. The "Default" theme is always located at the top of the menu, followed by factory presets below, and your saved custom themes at the bottom of the list.

Save Theme brings up a standard file naming and directory path dialog box where you can name and store your new themes. The default path is your User "Library/Preferences/MIOConsole3d/Themes" folder, which is where the *Theme* pop-up menu looks for your files.

The *Load Theme* button can be used to import any themes that are not already in your "Themes" Preferences folder (and are therefore not in the *Theme* pop-up menu).

The *Customized* entry in the *Theme* pop-up is just a holding slot for works-in-progress, and serves as an indicator in the Appearance: Theme menu button that the current mixer theme has been modified but has been not saved.

Consider the provided factory preset themes as starting points. Find one with a base color scheme you like and start tinkering.

- **UI Font**

This submenu allows you to change the primary text font throughout the Console user interface. The default font "Arial Unicode MS" looks pretty good, is nicely readable and most importantly, being Unicode it works with non-English character sets.

But, maybe you feel like injecting a little more personality into your mixer... Popular alternatives include "Futura", "Roboto" and the various "Stone Sans" flavors. "Comic Sans" is always good for a 90's vibe, and if you favor lengthy prose from Hobbiton you might like "Luminari". Fonts like these which are structured similarly to Arial Unicode (which was the model for the MIOConsole UI) will tend to work best, but may not support every character required in the UI.

Font switching takes place immediately in some parts of the UI, but not all. Quit and re-launch MIOConsole to load your new font choice throughout the entire UI.

- **Strip Color**

  Each mixer strip in the MIOConsole3d mix desk has a small color bar at the top (above the Input selector) and at the bottom (just above the Scribble Strip). Clicking on these color bars brings up Color Selector where you can choose a color for that strip. The "Strip Color" preference lets you choose whether the color is applied just to the color bars at the top and bottom of the mixer strip, or to the entire strip.

  Reminder: As with all of the UI elements in the Mixer strips, the top and bottom color bars can be hidden/shown in the Mixer Pane hamburger menu "Configure Channel Strip Elements..." window.

- **Color Palette: Controls and Customization**

  In addition to mixer strip colors and user-selectable fonts, the color scheme of MIOConsole3d is completely user-tweakable.

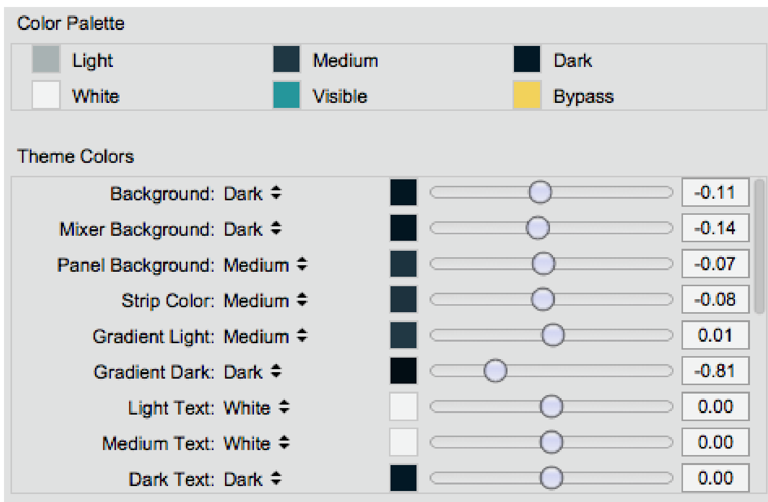


Figure 14.5: Color Palette controls

There are two control types for color palette manipulation. The Theme Colors affect the underlying color structure of the various UI elements across all MIOConsole3d windows directly. The six Color Palettes are like having six pre-set groups. Two are for specific cases: Bypass defines the color for when a plug-in is bypassed. Visible defines the color when a plug-in UI is open and visible. The other Color Palettes can be linked to Themes to allow for quick and easy manipulation of multiple Theme Colors at once.

Theme Colors controls can be set to bypass Color Palette groups by assigning them to "Custom" in the group selection pull-down. The 'fader' with each Theme Color is a basic 'Brightness' control.

At the same time, you can make individual adjustments to any of the Theme Colors independently of their Palette group, adding a little variety and differentiation between similar UI elements while still keeping them visually related.

Each square color button brings up the standard color selection dialog box with the color wheel, 'crayon box' and the other color selection options provided by the operating system. Opacity controls within the OS color selectors are respected, so you can create color blending effects for added depth.

- **Color customization hints/suggestions**

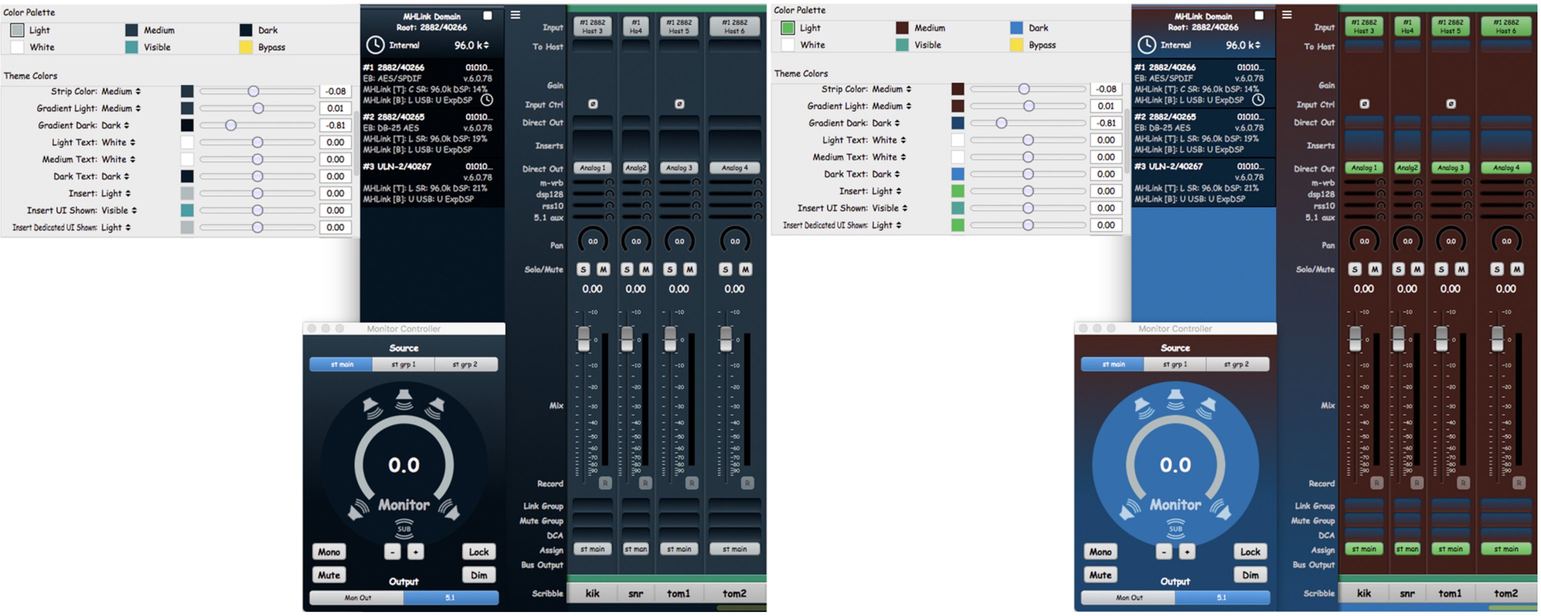


Figure 14.6: Just three little changes...

The above (apologetically unattractive) example illustrates just how much trouble you can get into with just thirty seconds of downtime between takes. Starting with the default color scheme, quick color wheel tweaks were made to the “Light”, “Medium” and “Dark” Color Palette group controls. You can see the changes reflected in the Theme and Color Palette controls, and their effect to the Mixer and Monitor Controller windows. The blue vs. ruddy red also nicely reveals the interaction between the “Gradient Light” and “Gradient Dark” Theme Color controls (which belong to the “Medium” and “Dark” Color Palette groups, respectively).

You can see what changes are being made to the MIOConsole3d UI as you make them, so really the best way to understand what each control element does is to try them out and see how they interact. The Theme Control labels make sense once you see them in action.

It should be noted that there is a real purpose for custom themes beyond just making cool-looking mixer desks. Working long hours in front of computer screens can be easily as fatiguing to the eyes as listening to high levels of audio for extended periods is to the ears. A dark-background, high-contrast desk surface with vivid (but not blaring) colors on important interface elements is much easier to look at over long periods than eight channel groups of brilliantly clashing colors at maximum brightness.

For example, the color palette for the ‘Night Vision’ preset was sampled from photos of the cockpit of a British Naval aircraft on night maneuvers, then modified to taste. Note that, depending on the display, white and light text and control elements can be too contrasty (if that’s a word) against deep dark backgrounds, and can be hard to look at. Lightening very dark backgrounds to let the controls "sit on a surface" so to speak, emulates a more natural context for your eyes to deal with, further lessening fatigue. Think of it as the visual equivalent of adding a mild room reverb to dry instrument tracks.

Interaction Preferences

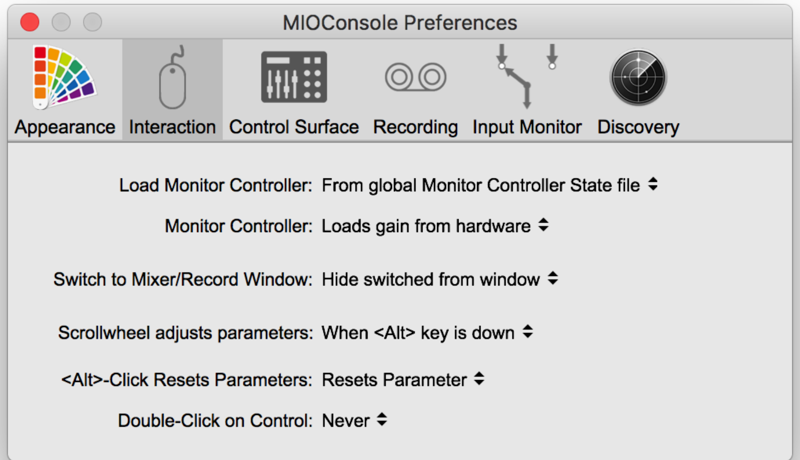


Figure 14.7: Interaction: MIOConsole3d Parameter Controls

- **Load Monitor Controller**
  MIOConsole3d maintains a "last unsaved state" file in your Preferences folder - essentially a snapshot taken of every Console parameter including Mix desk, Cues and Monitor Controller at the time you quit the MIOConsole3d application. This is so the next time you launch the application, everything comes up exactly as you left it, even if you did not explicitly hit "Save" or "Save As..." before quitting. In the case of Monitor Controller configurations, you may have a baseline reference configuration to which you need to return after having made temporary changes to accomodate a particular session.

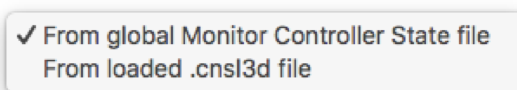


Figure 14.8: 'Load Monitor Controller configuration' behavior selector

"Load Monitor Controller" provides you with the option to always load "From global Monitor Controller State file" (the temporary 'last unsaved state'), or "From loaded .cnsl3d file" (the last *saved* state).

- **Monitor Controller**
  While "Load Monitor Controller" lets you choose the Monitor Controller configuration, the "Monitor Controller" preference determines whether or not to apply the saved gain settings from that MC configuration, or to keep using the current volume setting in your hardware. This is handy for avoiding unexpected monitor level jumps when loading a Console file from another facility with a different speaker/amp setup, or any time you simply do not know how loud they were monitoring when they saved the file. On the other hand, when sharing consoles between calibrated rooms, keeping the monitor levels consistent is important.

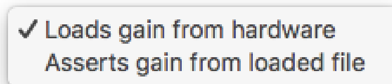


Figure 14.9: 'Load Monitor Controller: Apply Gain From...' behavior selector

  The default selection "Loads gain from hardware" maintains your current monitor levels even when loading a new monitor configuration from a saved console file.

  "Asserts gain from loaded file" applies the monitor levels from the loaded file to your monitoring chain. When using this option, it is best to double-check the levels it loads to your mains and Cue feeds to head off any unintended surprises.

- **Switch to Mixer/Record Window**
  The key sequence (⌘= (Command + equals sign)) can be used to quickly switch between the Session window and the main Mixer window. Usually this command merely switches the focus between the two windows, bringing one to the front while keeping them both open. This preference allows you to completely hide the 'other' window to better manage limited screen space and/or to conserve graphics cpu resources.

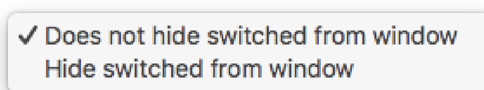


Figure 14.10: ...when switching between *Mixer* and *Session* windows...

- **Scrollwheel Adjusts Parameters**
  By default, moving the scrollwheel while over the mixer desk surface will scroll the fader strips to the left or right, but the scrollwheel will not move a fader or other control element without holding the **Opt** (on Mac) or **Alt** (on PC) modifier key. Setting "Scrollwheel adjusts parameters" to "Always" allows the scrollwheel to always affect whatever movable parameter is underneath it (potentially increasing the likelihood of unintentional parameter changes). The "Never" setting disables scroll-wheel parameter control completely.

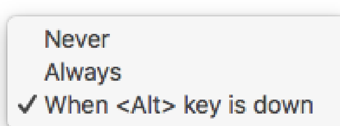


Figure 14.11: Scrollwheel behavior selector

- Alt-**Click Resets Parameters**
  By default, holding the **Opt** (on Mac) or **Alt** (on PC) and clicking on a parameter will return that parameter to its default position. Use this selector to change that behavior if you would prefer **Alt**-Click to open a a text-entry box, or "Do Nothing".

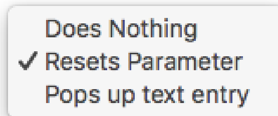


Figure 14.12: Alt-**Click behavior selector**

- **Double-Click On Control**
  Set to "Never" by default, set this parameter to "Always" if you like to double-click a control to reset to its default position.

Control Surface Preferences

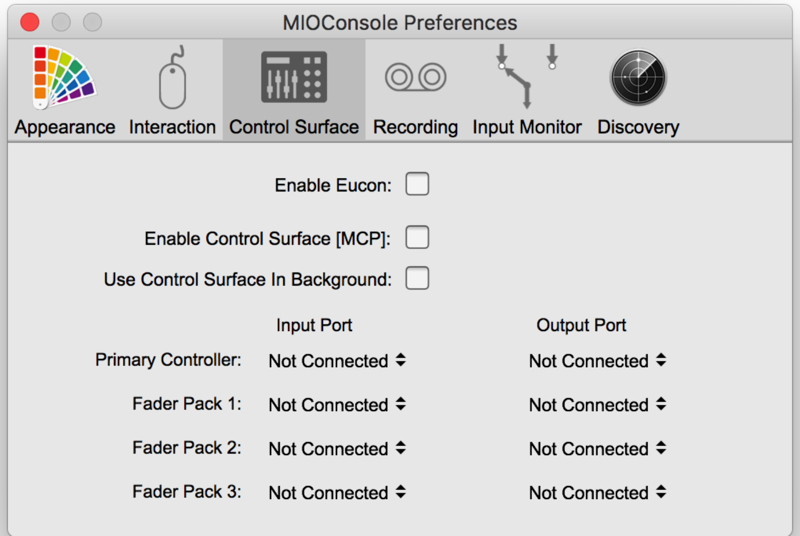


Figure 14.13: Control Surface: Parameter Controls

- **Enable EuCon Support**

  When checked, MIOConsole3d will automatically connect to EuControl services running on your host computer. This box is not checked by default. If you have a EuCon control surface, you must check this box in order to use it with MIOConsole3d. If you do not wish to use your EuCon control surface with MIOConsole3d, uncheck this box and MIOConsole3d will disconnect from EuCon services.

- **Enable Control Surface [MCP]**

  When checked, MIOConsole3d will listen to **Mackie Control Protocol** messages on the specified MIDI Ports. Use the controller pop-ups to select the MIDI I/O for your control surface. The default is for this to be unchecked. Toggle this box off and on to reset the MCP connection.

  - **Use Control Surface in Background** — When checked, MIOConsole3d will respond the Mackie Control Protocol control surfaces when MIOConsole3d is hidden or in the background. Uncheck this if you will use the same control surface units in another host program too. This preference does not affect EuCon control surfaces as EuCon manages application switching within its own UI.
  - **Primary Controller** — Select the MIDI ports for the primary (generally the master section) Mackie Control Protocol control surface unit. MIOConsole3d will treat this as the right-most controller in terms of fader layout.
  - **Fader Pack 1** — Select the MIDI ports for the Mackie Control Protocol first fader pack used in the system. MIOConsole3d will place this fader pack to the left of the Primary Controller in terms of fader layout.
  - **Fader Pack 2** — Select the MIDI ports for the Mackie Control Protocol first fader pack used in the system. MIOConsole3d will place this fader pack to the left of Fader Pack 1 in terms of fader layout.
  - **Fader Pack 3** — Select the MIDI ports for the Mackie Control Protocol first fader pack used in the system. MIOConsole3d will place this fader pack to the left of Fader Pack 2 in terms of fader layout.
  - Detailed information on Control Surface operations is available in the Control Surface Support chapter.

Recording Preferences

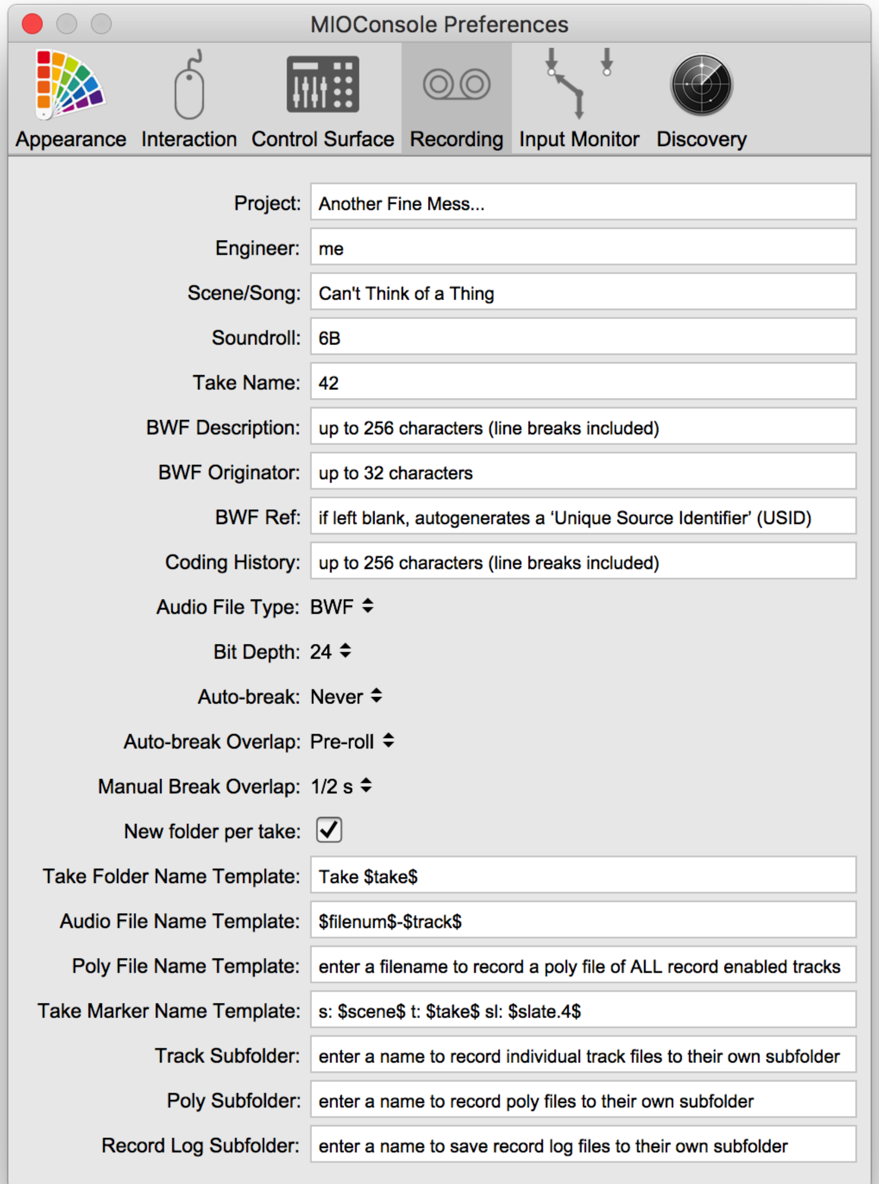


Figure 14.14: Recording: Filenaming, Metadata and Filetype Parameters

The top section of the "Recording" Preferences window presents you with text-entry fields for identifying and organizing your recorded audio files. Text entered in the **"Project"**, **"Engineer"**, **"Scene/Song"**, **"Soundroll"** and **"Take Name"** fields can be used as sources for auto-naming Take folders, audio files, poly files and take markers according to rules set up in the 'Name Template' fields at the bottom of this window (details to follow).

iXML

iXML is the de facto standard Field Recorder/Post Production media metadata interchange format, with broad hardware and software support across the industry. It is designed to facilitate consistent and unambiguous file and project metadata exchange across a variety of location recording, IP A/V streaming, film, video and post-production workflows. Click the link at the head of this paragraph to visit the specification website, or a more nicely human-readable overview can be read here.

Polyphonic soundfile recording is available alongside regular audio track recording. Enter a filename in the Poly File Name Template field to enable poly file recording.

MIOConsole3d Session propagates all project, Session Metadata and Take Notes information to iXML file header chunks for all BWF and poly soundfiles.

BWF (Broadcast Wave File) Metadata fields

If you record using the BWF file format, Session will include a BEXT chunk with recording meta-data in the header of every audio file. At this point BWF is pretty much the de facto standard professional audio recording format, and the metadata can be read by most professional audio and video workstations and media database systems. The BWF metadata fields allow you to set the following codes in the BEXT chunk:

- **BWF Description** (256 characters max) Sets the BWF Description field (supports the entry of line breaks).
- **BWF Originator** (32 characters max) Sets the BWF Originator field.
- **BWF Reference** (32 characters max) Sets the BWF Originator field. If this field is left blank, a reference 'Unique Source Identifier code' (USID) is automatically generated for each audio file.
- **Coding History** (256 characters max) Sets the BWF Coding History field (supports the entry of line breaks).

  More information about BEXT chunks is available from the EBU at "EBU BWF User Guide" documentation.
- **Audio File Type** selects the type of audio files to be recorded.

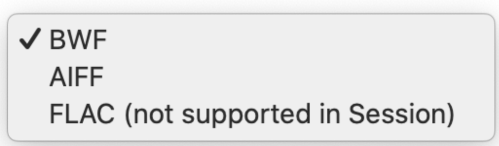


Figure 14.15: Audio File Type selector

- **BWF** is the EBU standard audio file based upon the WAV file format. It requires finalization at the end of recording. The Session record engine will automatically convert the file to a RF64 (64-bit Wave file) if the file size exceeds 4 GB. It supports timestamping and standardized metadata. BWF is the current de facto standard professional PCM audio file format.

> Most current audio software supports RF64 files, but some older products do not; if you will be working with older software it would be safest to use the Auto-break feature to keep files under 2GB.

   - **AIFF** file type is the standard Apple Audio file. It requires finalization at the end of recording and has a 2 GB filesize limit. It does not support timestamping.

     "32 (float)" audio bit depth is not available for the AIFF file format.

   - **FLAC (Free Lossless Audio Codec)** records lossless data-compacted audio files which can save up to 40-50% of the file size while maintaining perfect bit-accuracy of the original audio stream. Data reduction rates vary depending on content. With continuous music recordings, actual file size savings are generally closer to 70% compared to uncompressed BWF or AIFF files.

     Since FLAC is not a Linear PCM audio file forrmat, it is not currently supported for playback in Session. Waveforms will be shown during capture but will be removed at the end of each take, with the file location shown in the Missing Files UI in the Session widget bar.

     "32 (float)" audio bit depth is not available for the FLAC file format.

- **Bit Depth** sets the bit depth resolution of the recorded audio files to 16- or 24-bit (integer) or 32-bit (float). 24-bit integer is the default setting. If you select 16-bit, you should apply dither using a MIO SimpleDither plugin on the channels you will record; the Session record engine does not automatically dither down to 16-bit.

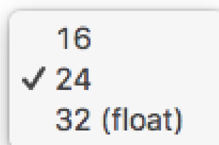


Figure 14.16: Audio Bit Depth selector

- **Auto-break:** The Auto-break function automatically breaks audio files into segments as you are recording.

  When the largest file in the take reaches the limit specified by this setting, all the files in the take are written with "completed" file headers and a new set of audio files begins. The new segments will continue to be recorded in the same location, continuing the current Take, and each file will have a 'segment number' filename suffix added for easy reconstruction.

  There are a number of reasons you might consider breaking up recorded audio files into smaller segments:

  1. Filesystem Compatibility - some older file systems do not support files larger than 2GB (FAT16) or 4GB (FAT32).
  2. Filetype Compatibility - most software will not support AIFF files larger than 2GB.
  3. Software Compatibility - some audio programs can not recognize audio files of larger than 2GB or 4GB, even if the file type supports larger files.
  4. Security - When recording with a file type that requires finalization, in the event of a power failure during recording, you don't need to worry about recovering a un-finalized file. While un-finalized files can be finalized after the fact, it is more work than just having files that get finalized periodically.

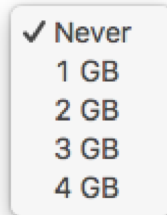


Figure 14.17: Auto-break file size selector

- **Auto-break Overlap** allows you to overlap file segment breaks, or leave them with no overlap. With "Gapless" files, the last sample of one segment is immediately followed by the first sample of the next segment. Many DAWs understand this and can treat segmented files as if they were one. Alternatively, you can concatenate gapless files together very quickly with file system commands.

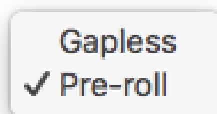


Figure 14.18: Auto-break Overlap selector

Setting an Auto-break Overlap is useful when you will be editing the files manually anyway and wish to have a measure of control over how the segment gaps are edited. The length of the overlap follows your Pre-Roll settings in the Session Record Trigger control block (settings from 1/4 seconds to 60 seconds).

The overlapped samples will be at the head of the file created after the break. To expose the preroll audio, drag the "In-Fade" at the start of the recorded take segment to the left.

- **Manual Break Overlap**: One of the most popular features of Session recording is the ability to trigger a new take without interrupting the recording in progress. To trigger a new take while in Record, simply tap the Record button. Manual Break Overlap lets you set the amount of overlap between the two takes in case you want to splice them back together.

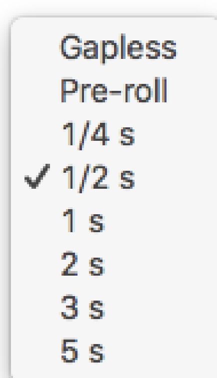


Figure 14.19: Manual Break Overlap selector

The overlapped samples will be at the beginning of the files in the new Take. In other words, the currently recording Take will end immediately at the break, and the beginning of the new Take will include overlapping audio from *before* the break. To expose the preroll audio, drag the "In-Fade" at the start of the recorded take segment to the left.

> Note: To be clear, the Pre-roll, Post-roll and Overlap fade-ins and fade-outs are never printed into the file itself. All Session fades and gains are executed in the playback engine before sending to the 3d mixer.

- **New Folder per Take** when checked, places all the files of each take in its own nested folder within the Record Folder and Mirror Record Folder(s). These Take folders are named per the parameters set below…
- **Take Folder Name Template**:

  This text field lets you auto-name new 'take' folders using the text entered in the Recording Preferences **'Project'**, **'Engineer'**, **'Song'** and **'Take Name'** fields, as well as a number of other variables. Understanding how to use this feature makes organizing crazed recording sessions *much* easier. If you use Session for "Take-based" recording workflows, this feature will be like a new best friend - you may hate it at first, but once you get to know it you won't want to live without it. Here is the **Take Folder Name Template** tooltip pop-up:

Define the template used to create the folder name.
Plain text is copied. The following special strings are substituted:
 $project$ = Project Name
 $engineer$ = Engineer Name
 $song$ = Song Name
 $takename$ = Take Name
 $take[.n]$ = Take Number (optional . followed by number sets width)
 $slate[.n]$ = Slate Number (optional . followed by number sets width)
 $date$ = Today's Date
 tod = Time of Day
 $todsamplenum$ = Sample Count from Midnight for TOD
 $timecode$ = Timecode of start of recording
 $tcsamplenum$ = Sample Count from Midnight for TC

Figure 14.20: Take Folder Name Template **tooltip**

…and here's the step-by-step breakdown:

The 'dollar sign's are used to define "tokens". These "tokens" command the folder-naming script where to look for information to copy into the folder name.

The **$project$** token says: look in the **"Project"** field, copy that text and paste it in the folder name.

The **$engineer$** token says: look in the **"Engineer"** field, copy that text and paste it in the folder name.

$song$ says: look in the **"Song"** field, copy that text and paste it in the folder name.

$takename$ says: look in the **"Take Name"** field, copy that text and paste it in the folder name.

$take$ says: copy the 'Next Take' number from the Session "Takes" counter and paste in into the folder name. The [.n] sets the number of digits used to represent the take number. If you don't specify a number here, the take numbers will be 1, 2, 3, 4 etc. If you enter 2 (for 2 digits), you get a leading '0', so the take numbers go 01, 02, 03, 04 etc. The leading '0' makes listing files alphanumerically much cleaner. If you expect more than 100 takes, set it to 3 (for two leading zeros), set to 4 for a thousand takes, etc. The brackets don't get typed in, just the dot and the number. For example, if you are at take #4 and you had entered **Take $take.2$** in your template, the template script would insert **Take 04** in the folder name.

$slate$ copies the 'Next Slate' number from the Session "Slate" counter and pastes in into the folder name. The [.n] sets the number of digits used to represent the slate number. If you don't specify a number here, the slate numbers will be 1, 2, 3, 4 etc. If you enter 2 (for 2 digits), you get a leading '0', so the slate numbers go 01, 02, 03, 04 etc. The script logic for Slate is identical to Take (detailed above).

$date$ says: look up the current date and paste it in the folder name. The 'date' format is year-month-day, so July 22, 2019 would appear in the folder name as "2019-7-22".

tod says: paste the current Time Of Day into the folder name. The 'time' format is hours-minutes-seconds using a 24-hour clock, so 4:28 and 17 seconds in the afternoon would appear in the folder name as "16-28-17".

$todsamplenum$ will insert the number of audio samples counted from midnight to the start of the recording. This is essentially just a hyper-accurate measure of the 'Time Of Day' the take was initiated. This feature is mostly used when providing files to automated media ingestion systems.

$timecode$ will insert the SMPTE timestamp at the start of the recorded file.

$tcsamplenum$ will insert the number of audio samples calculated from a linear time code reference. (Note: if the LTC is locked to 'Time Of Day', this will be the same as **$todsamplenum$**). This is essentially just a hyper-accurate measure of the time of day the take was initiated. This feature is mostly used when providing files to automated media ingestion systems.

- **Audio File Name Template** is similar to **Take Folder Name Template**, but as one might expect, it is for audio file names rather than take folders. Here is the tooltip pop-up:

Define the template used to create the file name.
Plain text is copied. The following special strings are substituted:
$project$ = Project Name
$engineer$ = Engineer Name
$song$ = Song Name
$takename$ = Take Name
$take[.n]$ = Take Number (optional . followed by number sets width)
$slate[.n]$ = Slate Number (optional . followed by number sets width)
$date$ = Today's Date
tod = Time of Day
$todsamplenum$ = Sample Count from Midnight for TOD
$timecode$ = Timecode of start of recording
$tcsamplenum$ = Sample Count from Midnight for TC
$track$ = Track Name
$tracknum[.n]$ = Track Number (optional . followed by number sets width)
$filenum[.n]$ = File number (optional . followed by number sets width)

Figure 14.21: Audio Folder Name Template **tooltip**

The syntax for file naming is the same as for folder naming, although there are some extra "tokens" specifically for use in audio file names. Here's the step-by-step breakdown:

$project$ copies the text from the **"Project"** field and pastes it in the file name.

$engineer$ copies the text from the **"Engineer"** field and pastes it in the file name.

$takename$ copies the text from the **"Take Name"** field and pastes it into the file name.

$take$ copies the 'Next Take' number from the Session "Takes" counter and pastes in into the file name. The [.n] sets the number of digits used to represent the take number. If you don't specify a number here, the take numbers will be 1, 2, 3, 4 etc. If you enter 2 (for 2 digits), you get a leading '0', so the take numbers go 01, 02, 03, 04 etc.

$slate$ copies the 'Next Slate' number from the Session "Slate" counter and pastes in into the file name. The [.n] sets the number of digits used to represent the slate number. If you don't specify a number here, the slate numbers will be 1, 2, 3, 4 etc. If you enter 2 (for 2 digits), you get a leading '0', so the slate numbers go 01, 02, 03, 04 etc.

$date$ pastes the current date to the file name. The 'date' format is year-month-day, so July 22, 2019 would be entered as "2019-7-22"

tod pastes the Time Of Day at the start of the take into the file name. The 'time' format is hours-minutes-seconds and uses a 24-hour time scale, so 1PM is represented as '13-00-00' hours.

$todsamplenum$ will insert the number of audio samples counted from midnight to the start of the recording into the file name. Again, this feature is mostly used when providing files to automated media ingestion systems.

$timecode$ will insert the SMPTE timestamp at the start of the recorded file into the file name.

$tcsamplenum$ will insert the number of audio samples calculated from a linear time code reference. (Note: if the LTC is locked to 'Time Of Day', this will be the same as **$todsamplenum$**).

$track$: Each Session track corresponds to a Mixer strip. The name of each strip being recorded is always visible in the Session Track Overviews window. **$track$** pastes the name of the track into the file name. In cases where the audio routed to the track is post-process (either 'To Host: Post-Insert' or using the post-process Direct Out) the term **[POST]** will be added to the track name.

$tracknum[.n]$: The Session Track Overview window shows each Mixer strip which has return routes to the Host computer assigned. The position of these tracks corresponds to their position in the Mixer: so strips at the left of the Mixer are shown as tracks at the top of the Session Track Overviews. Using the Session Track Overviews as a map, the topmost track will be track 1, the one below it is track 2, and so on.

$tracknum[.n]$ numbers all tracks shown in the Session Track Overview, whether they are Record-Armed or not. In cases where not all tracks are armed for any given take, the recorded track numbers will have gaps in their sequence, but they will always match the same source audio and **$track$** name from the Mixer.

$filenum[.n]$ is similar to **$tracknum[.n]$**, but counts only the files actually being recorded.

Note: Plain text inserted in the 'Template' fields (outside of the token variables) will be applied to the foldername or filename as entered.

Spaces are allowed in folder and filenames, but keep in mind that some internet transmission systems, media ingestion systems and computer file systems have different opinions of how to handle spaces and uncommon characters. It is a good habit to use underscores _ or hyphens - instead of spaces to head off any file-reading issues down the road.

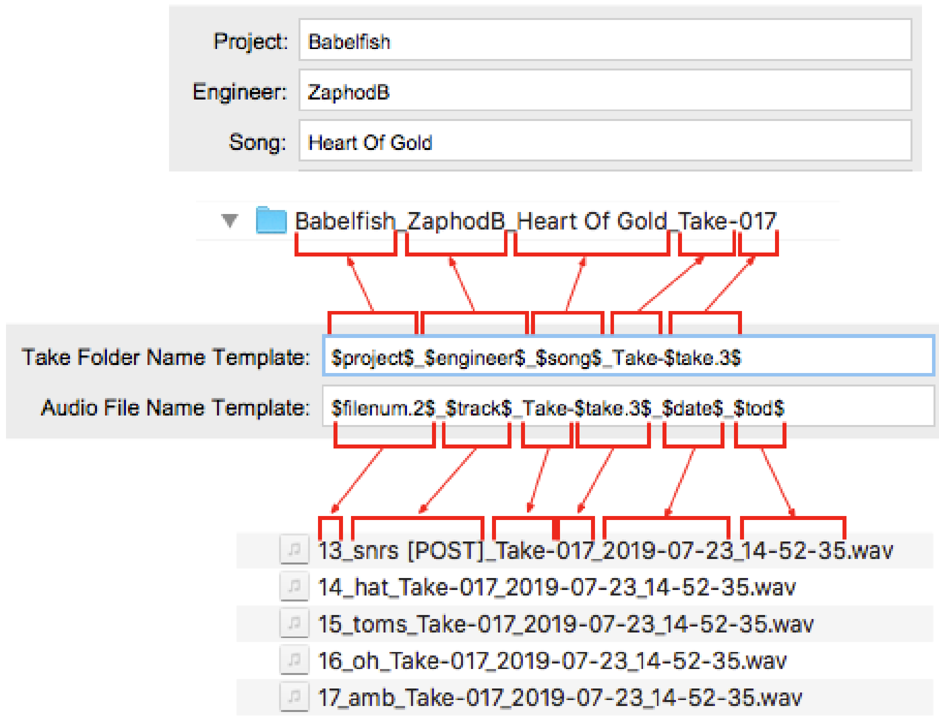


Figure 14.22: Name Templates script example

Above is an example of the **Take Folder Name Template** and **'Audio File Name Template'** scripts in action. You can see the text from the 'Project', 'Engineer' and 'Song' fields were referenced by the $project$, $engineer$ and $song$ tokens in the **Take Folder Name Template** and have been copied to the folder name. The underscores between the tokens and the word "Take-" (with the hyphen) were manually typed in.

In this example, the number of digits to represent take numbers was set to 3: **$take.3$** so there is a leading 0 before the take number 17. The use of track names, date and time of day are shown in the file names. In all cases, the $slate$ token script logic is identical to $take$.

- Entering a name in the **Poly File Name Template** field will tell the record engine to write polyphonic audio files in addition to the usual Tracks audio files.

  The Poly File Name Template follows the same script logic as the Audio File Name Template. Here is the tooltip pop-up menu:

Define the template used to create the aggregate multichannel (poly) filename. Leave blank for no poly file.
Plain text is copied. The following special strings are substituted:
$project$ = Project Name
$engineer$ = Engineer Name
$song$ = Song Name
$takename$ = Take Name
$take[.n]$ = Take Number (optional . followed by number sets width)
$slate[.n]$ = Slate Number (optional . followed by number sets width)
$date$ = Today's Date
tod = Time of Day
$todsamplenum$ = Sample Count from Midnight for TOD
$timecode$ = Timecode of start of recording
$tcsamplenum$ = Sample Count from Midnight for TC
$track$ = Track Name
$tracknum[.n]$ = Track Number (optional . followed by number sets width)
$filenum[.n]$ = File number (optional . followed by number sets width)

Figure 14.23: Poly File Name Template **tooltip**

- Use the **Take Marker Name Template** field to markers generated at the head of each take.

  The Take Marker Name Template follows the same script logic as the Take Folder Name Template. Here is the tooltip pop-up:

Define the template used to name for marker automatically generated for each take.
Plain text is copied. The following special strings are substituted:
$project$ = Project Name
$engineer$ = Engineer Name
$song$ = Song Name
$takename$ = Take Name
$take[.n]$ = Take Number (optional . followed by number sets width)
$slate[.n]$ = Slate Number (optional . followed by number sets width)
$date$ = Today's Date
tod = Time of Day
$todsamplenum$ = Sample Count from Midnight for TOD
$timecode$ = Timecode of start of recording
$tcsamplenum$ = Sample Count from Midnight for TC

Figure 14.24: Take Marker Name Template **tooltip**

- Enter a **Track Subfolder** name to create a subfolder within the Record Folder specifically for discrete track audio files. This would be like the "Audio" or "Audio Files" subfolders found in most DAW project folder hierarchies.

  Leave this field blank to keep track audio files in the main Record Folder. Here is the tooltip pop-up for Track Subfolder:

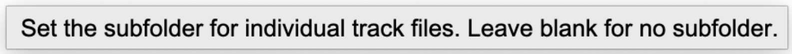


Figure 14.25: Track Subfolder **tooltip**

- Enter a **Poly Subfolder** name to create a subfolder within the Record Folder to contain polyphonic audio files.

  Leave this field blank to keep poly files in the main Record Folder. Here is the Poly Subfolder tooltip:

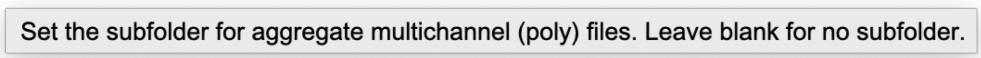


Figure 14.26: Poly Subfolder **tooltip**

- Enter a **Record Log Subfolder** name to create a subfolder within the Record Folder to contain the record log files generated with each recorder file set.

  Leave this field blank to keep record logs in the main Record Folder. ...and the tooltip:

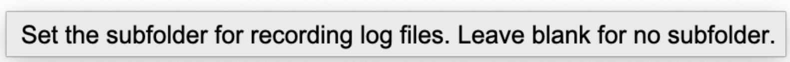


Figure 14.27: Record Log Subfolder **tooltip**

Input Monitor Preferences

The Input Monitor Preferences define the overall MIOConsole3d operating model as it relates to Session playback, and sets mixer strip monitoring behaviors for location recording, multitracking and overdubbing workflows.

Mixer strip "input" refers specifically to the Source input at the top of each input mixer strip. In an analog console this would the Mic/Line input. Of course, in MIOConsole3d the input strip Source can also be a Host computer input, SCP USB or a mixer bus return, so "Input".

Playback from Session tracks (e.g.: tape returns) is automatically routed to each tracks' input strip when Play is engaged, based on the rules set in these preferences.

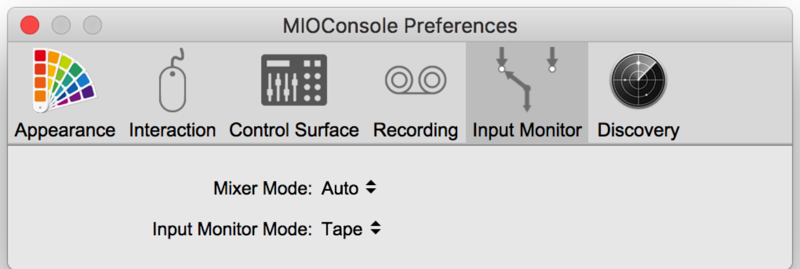


Figure 14.28: Input Monitor: Parameter Controls

Please note that in all cases, engaging the Input control below the fader on each input strip will override Session playback on that strip.

Mixer Mode

Mixer Mode sets the overall Session playback switching behavior into the MIOConsole mixer.

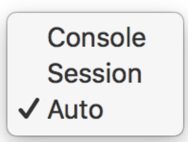


Figure 14.29: Mixer Mode selector

- **Console** disables Session playback routes to the mixer (similar to engaging all mixer strip Input buttons). Recording is still available as usual, but this mode ensures that accidentally engaging Session playback (such as with remote EuCon or MCP controller) will not affect your mixer input source audio. This is essentially the old Record Panel modus operandi.
- Conversely, **Session** mode disables all mixer strip source Inputs, unless the strip Input is manually engaged. Select this mode when editing and mixing with Session as your DAW.
- The default **Auto** mode switches between Console and Session modes depending on the Session transport state. Engaging Session playback will engage 'Session' mode until play transport is stopped and you return to Console mode.

Input Monitor Mode

Input Monitor Modes apply to Session and Auto Mixer Modes.

When using Session as your recording, editing and overdubbing platform, the Input Monitor modes allow you to tailor the mixer strip input source monitoring behavior to your recording process and personal preferences.

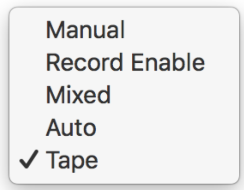


Figure 14.30: Input Monitor Mode selector

- **Manual** provides input monitoring only when the mixer strip Input control is on (yellow); when the Input control for the strip is off, the strip will monitor the playback signal from Session (or be muted when there is no playback).
- **Record Enable** sets the strip to monitor the input when the strip Record Enable is engaged (red); otherwise the strip monitors the playback signal from Session.
- **Mixed** mode mixes the Input signal and Session playback when the strip Record Enable is engaged unless the transport is in record. Use this mode for cases where the talent wants to hear and play against the pre-recorded track before punching into an overdub. Once you punch in, the strip will only monitor the live input.
- **Auto** plays the Session track, switches to Input at punch-in, and back to Session when punched out. When the strip is Record Enabled, Input audio is muted when Session playback is stopped, and is audible only in Record.
- **Tape** emulates ye olde Tape Machine Mode: Input is monitored when playback is stopped or when the track is punched into Record; otherwise the Session audio is monitored.

The tables on the next page break down the audio monitored through a mixer strip for each of the five Input Monitor Modes with Record Enable off and on, and with the Session transport stopped, playing and recording.

The first column lists the Input Monitor Preferences > Input Monitor Mode setting. Each mode is listed twice: first with mixer strip Record Enable off, then with Record Enable engaged. The Record Enable state is listed in the second column.

The three Transport columns show the audio source monitored through the mixer strip for each of the Session Transport play/record modes at each setting.

Console Mixer Mode always monitors source Input, ignoring Session transport altogether. No table required.

Session mode will mute the source Input when Session transport is stopped, unless:

A) Record Enable is engaged, *and*
B) Input Monitor Mode is set to either Record Enable, Mixed or Tape.

Auto mode, however *plays* the source Input when Session transport is stopped. Otherwise Auto is the same as Session mode.

Note: In all cases, the **Input Monitor** button on the strip overrides the automatic monitor switching, and monitors the strip's input when it is engaged.

Input Monitor Mode	Record Enable	Transport: Stop	Transport: Play	Transport: Record
Manual	OFF	Muted	Playback	Playback
Record Enable	OFF	Muted	Playback	Playback
Mixed	OFF	Muted	Playback	Playback
Auto	OFF	Muted	Playback	Playback
Tape	OFF	Muted	Playback	Playback
Manual	ON	Muted	Playback	Playback
Record Enable	ON	Input	Input	Input
Mixed	ON	Input	Input + Playback	Input
Auto	ON	Muted	Playback	Input
Tape	ON	Input	Playback	Input

Table 14.1. Mixer Mode: Session

Input Monitor Mode	Record Enable	Transport: Stop	Transport: Play	Transport: Record
Manual	OFF	Input	Playback	Playback
Record Enable	OFF	Input	Playback	Playback
Mixed	OFF	Input	Playback	Playback
Auto	OFF	Input	Playback	Playback
Tape	OFF	Input	Playback	Playback
Manual	ON	Input	Playback	Playback
Record Enable	ON	Input	Input	Input
Mixed	ON	Input	Input + Playback	Input
Auto	ON	Input	Playback	Input
Tape	ON	Input	Playback	Input

Table 14.2. Mixer Mode: Auto

Discovery Preferences

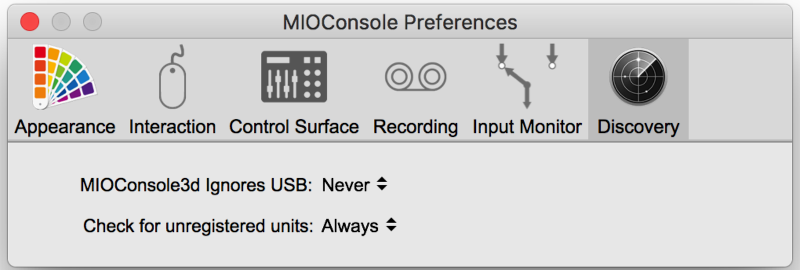


Figure 14.31: Discovery: Parameter Controls

- **MIOConsole3d Ignores USB:**
  As detailed in the USB and SCP sections of the manual, there are use cases where you may wish to connect the USB port of a second MHLink Host computer to the USB port of an MHLinked 3d box. While this does not link the two Domains, it does allow you to route audio applications residing on one computer into the MHLink Domain hosted and operating on the other computer.

  Set “MIOConsole3d Ignores USB” to “Always”, quit and re-launch MIOConsole3d before making the USB connection. This will ensure the MIOConsole3d application will *ignore* the computer USB port at launch - when it scans for boxes to control. This avoids the conflict where MIOConsole3d on each computer would be trying to control the same box at the same time.

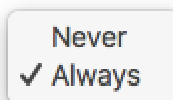


Figure 14.32: Discovery: Parameter selection options (for both controls)

> Remember...
>
> This control can *not* take effect until the next launch of the MIOConsole3d application.

- **Check for unregistered units:**
  When set to “Always” (the default setting) MIOConsole3d will scan for any visible unregistered 3d devices. If an unregistered box is found, the hardware registration form will open automatically for you.

  Set this preference to “Never” to disable the scan, for times when connecting rental 3d units or adding friends' 3d boxes to the domain.

  Note that you can always register hardware or update your registration information manually using the menu bar: “MIOConsole3d: Register Attached Units...” command.

15. Control Surface Support

Control Surface Preferences

MIOConsole3d supports both Avid EuCon and Mackie Control Protocol. Both protocols are supported simultaneously, so you can use both types of controllers at the same time. If both controllers are controlling the same elements of MIOConsole3d, MIOConsole3d will keep both protocols synchronized. More interesting, however, is that the two different control surface systems can be used to control different parts of MIOConsole3d at the same time, allowing you to operate main and aux/cue mix buses simultaneously and independently with multiple control surface units.

The first step in using the Control Surface support is enabling and configuring your Control Surfaces. This is done from the MIOConsole3d Preferences panel. These preferences are accessed via the *MIOConsole3d > Preferences…* command (or via the ⌘, (Command + comma key sequence). When you select the Preferences command, the Preferences sheet is shown on the MIOConsole3d window:

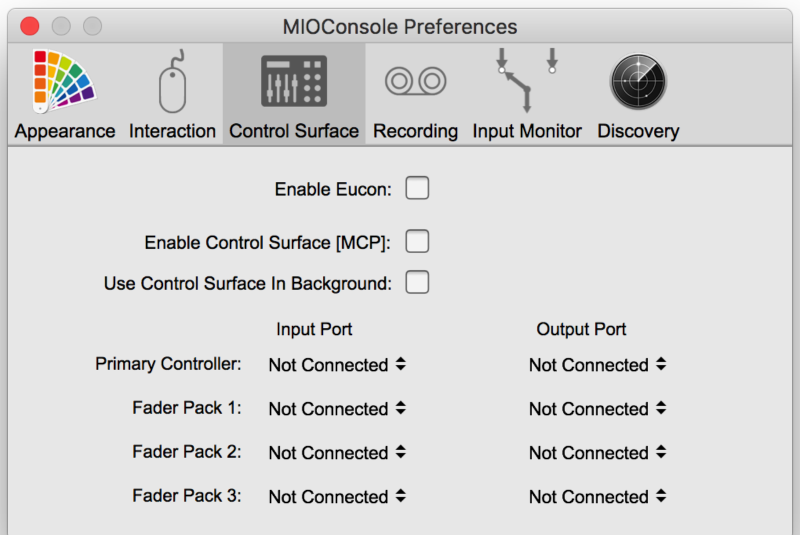


Figure 15.1: MIOConsole3d Preferences: Control Surface parameters

The following entries offer operational info supplementary to the MIOConsole3d Control Surface Preferences pane:

- **Enable EuCon Support**

  When checked, MIOConsole3d will automatically connect to EuControl services running on your host computer. This box is not checked by default.

  The scan for EuCon services can significantly slow down MIOConsole3d application launch on older computers, especially if EuControl is not running. MIOConsole fires up pretty fast even with a full Session and a bunch of boxes on-line, so if MIOConsole is acting sluggish, check EuCon Enable.

  EuCon supports both many different devices as both wired and wireless network clients so the scan for devices needs time to be thorough. Unfortunately, if you are not expecting it that lag tends to make one feel like there's a problem, so, well… just something to be aware of.

- **Enable Control Surface [MCP]**

  When checked, MIOConsole3d will listen to **Mackie Control Protocol** messages on the specified MIDI Ports. Use the controller pop-ups to select the MIDI I/O for your control surface. The default is for this to be unchecked.

  Toggle this box off and on to rescan/reset the MCP connection. This will register MCP devices connected or powered up after MIOConsole3d had been launched, and will re-sync the control surface to your mixer.

  - **Use Control Surface in Background** — When checked, MCP control surfaces stay connected to MIOConsole3d even when it's hidden or in the background. When unchecked, your control surface will connect to whatever MCP-enabled DAW is brought into focus. This preference affects MCP only.
  - **Primary Controller** — Select the primary MCP control surface unit (the one with the master section, if any). MIOConsole3d will treat this as the right-most controller in terms of fader layout.
  - **Fader Pack 1** — Select the MIDI ports for the Mackie Control Protocol first fader pack used in the system. MIOConsole3d will place this fader pack to the left of the primary in terms of fader layout.
  - **Fader Pack 2** — Select the MIDI ports for the second MCP fader pack used in the system. MIOConsole3d will place this fader pack to the left of Fader Pack 1.
  - **Fader Pack 3** — Select the MIDI ports for the third MCP fader pack. MIOConsole3d will place this fader pack to the left of Fader Pack 2.

Once you have enabled support for the type of surface you want to use, and, in the case of Mackie Control surfaces, have selected the relevant communication ports, MIOConsole3d will connect to the surfaces and begin communication, including updating of the surface faders, scribble strip read-outs, and metering.

Details of EuCon Control Surface Support

The EuCon protocol supported by MIOConsole3d has been tested with the most common Avid control surfaces including the Artist series (MC Control and MC Mix), Avid S1, Avid S3, Avid Dock, and Avid Control.

The EuCon protocol works by having the EuCon client application (MIOConsole3d in this case) build a model to represent the controllable items in the client and their structure and organization. This model is used by the EuCon system to map the controls over the available control surface hardware. Since the mapping is done by EuCon, much of the details of interacting with the control surface are actually implemented in EuCon and are consistent from application to application.

EuCon and EuControl provide facilities for locking strips to faders and locking the Control Room (Monitor Controller) section and Transport Control sections to specific applications. All of these facilities can be useful when working with MIOConsole3d, especially the Control Room locking if you are using the Monitor Controller to control your source switching and monitor level. Please consult your EuCon documentation and/or Avid's updated collection of EuCon support documents for details on how to use these facilites of EuCon.

Avid has released a free EuCon control surface application for the iPad, iPhone and Android devices called *Avid Control*. This application forms the display portion of the Avid Dock control surface and may be used by itself or in concert with other EuCon-enabled surfaces.

Avid Control implements many of the more advanced features of the EuCon control surfaces in an easy-to-use application, and provides a great way to wirelessly control MIOConsole3d. In order to maximize the value to our users, MIOConsole3d greatly enhances EuCon support to allow you to utilize a significant fraction of the features of EuCon and the *Avid Control* application.

Please note: Even with these enhancements, MIOConsole3d control surface support does not yet include mappings for plugins; that will come in a future release.

MIOConsoled3d supports the following EuCon features across the full set of EuCon control surfaces:

- Meter Bridge on Avid Control
- Mixer View on Avid Control (and on surfaces with faders)
  - Fader
  - Mute
  - Solo
  - Record Enable
  - Stereo Panning
  - Mono through 7.1 Metering
  - Strip Type Filtering on Avid Control
- Track View on Avid Control
  - Record
  - Solo
  - Mute
  - Clear (items with selected mode)
- Meter View on Avid Control
- Channel View on Avid Control (and knob sets on surfaces)
  - Input Knob set for Input Strips
    - Input Mode Select
    - Polarity Invert
    - Headamp Gain
    - Phantom Power
  - Pan Knob set for Channels with Pan control
- Monitoring View on Avid Control (and surfaces with Monitor Control sections)
  - MC Source Select
  - MC Output Path Select (first 3 paths)
  - MC Speaker Mutes/enables (up to 7.1.4)
  - MC Output Gain
  - MC Dim
  - MC Mute
  - MC Mono Folddown
  - Cue 1-4 Gain + Mute
  - Talkback Enable
- Transport Control
  - Session Time Readout
  - RTZ
  - Stop
  - Play
  - Record
  - Loop Enable
- Display Soft keys preconfigured to switch Aux busses onto Faders

You can find out how to download Avid Control and the EuCon software from Avid's Avid Control page.

Avid Control Details

The free *Avid Control* app provides a deep and flexible way to wirelessly control the bulk of MIOConsole3d's features. It also works in concert with other physical EuCon control surfaces. This section details the operation of MIOConsole with the Avid Control application. *Note:* only one Avid Control app instance can be connected to MIOConsole3d at one time; this is a EuCon limitation.

Avid Control Mixer Pane

The Mixer view in Avid Control provides the primary mixer control surface for controlling the MIOConsole3d mixer.

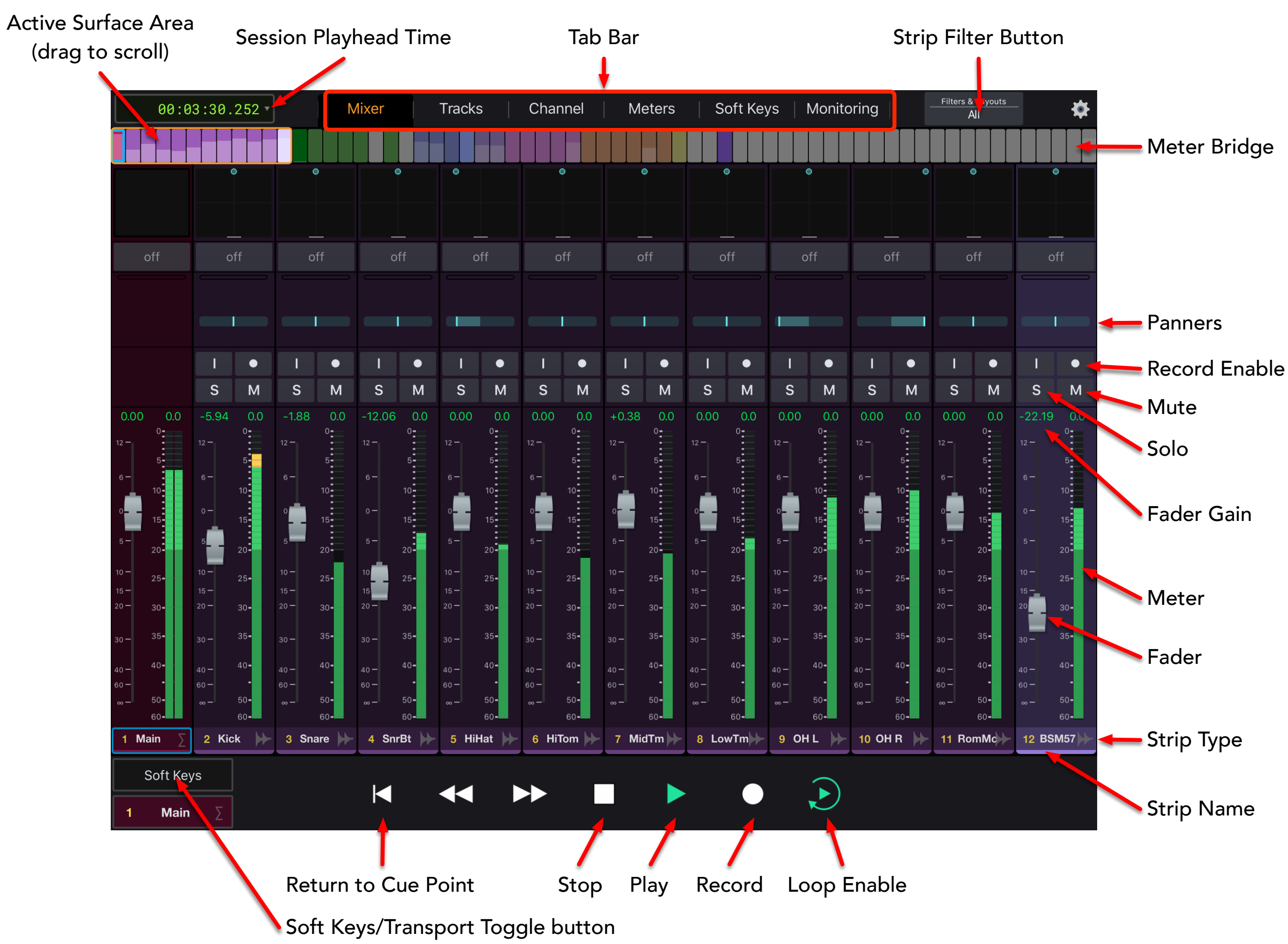


Figure 15.2: MIOConsole3d Mixer View in Avid Control

The top of the screen includes a display of the current Session playhead time, a set of view selector tabs and the strip filter button. Tap in the view selector to choose which view is active in the app. Tap the strip filter button to control how the strips are filtered on the surface (see below for details).

Right below the top of the screen is a Meter Bridge that shows all the unfiltered strip meters in the order that they appear in the mixer colored by the associated strip colors. There is an active area indicator that shows which strips are visible on the surface. You can touch and drag the active area indicator to scroll the visible strips. You can also touch and drag in the strip area to scroll strips into view.

Next down are the strips of the mixer; see the labeled controls below to see which controls are active with MIOConsole3d. Note that the strips are tagged with their strip type (e.g. Audio Input, Master, Aux and Group) and include the strip name at the bottom. The meters per strip will be automatically adjusted to reflect the width of the strip or bus. Hint: you can double-tap the fader cap to set it back to 0dB. Tap the panner on strip to pop-up a panner to allow adjusting the pan for that strip.

At the bottom left of the screen there is a “Soft Keys/Transport” button; tap this button to switch the bottom strip between the soft keys and transport controls.

When the Transport controls are shown, the transport is used to control the transport of Session; RTZ triggers a “Return to Cue event”. Stop stops the transport, Play starts playback, Record triggers a record start event, and the Loop Enable button toggles (and shows the state of) loop playback in Session.

When the Soft Keys controls are shown you can use the pre-configured soft keys to switch between the active bus shown on the faders. Unfortunately, EuCon does not provide a mechanism for MIOConsole3d to indicate which bus is currently active, so you will have to affirmatively set the current bus if you have lost track of which bus is active on the surface.

Avid Control Tracks Pane

The Tracks view in Avid Control provides a quick overview for showing a large number of strips on a single page, plus quick access to the Mute, Solo and Record Enables for those tracks.

Each track tile shows the Strip name and type, the meter associated with the strip, and the status of the strip's Record, Solo and Mute. Based on the current button mode, tapping a tile will toggle the state of the associated strip's Record, Solo or Mute. The current mode is selected by taping the associated button at the bottom of the screen in the "Tile Tap Mode Select" area. The "Clear" button to the left of the "Tile Tap Mode Select" area will clear the state of the current mode on all the tracks, so you can use it to clear all solo's or record enables (for example). While you can also clear all the mutes, that seems less useful.

All the other elements of this page are the same as what you find on the Mixer pane.

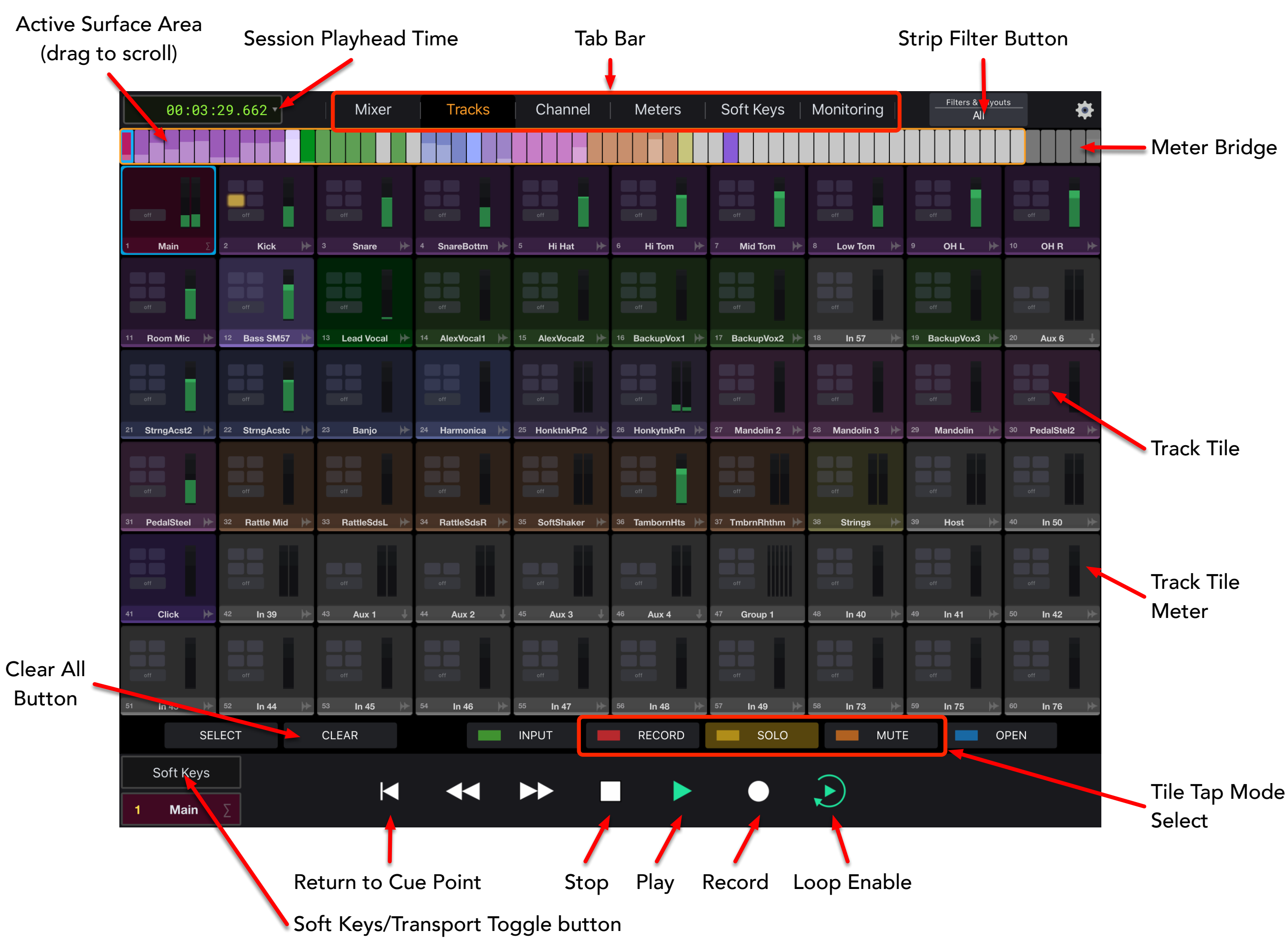


Figure 15.3: MIOConsole3d Tracks View in Avid Control

Avid Control Meters Pane

The Meters view in Avid Control provides a larger Meter Bridge view for the mixer strips in the system. There is not much more to say about this view, other than it can be filtered like the other views.

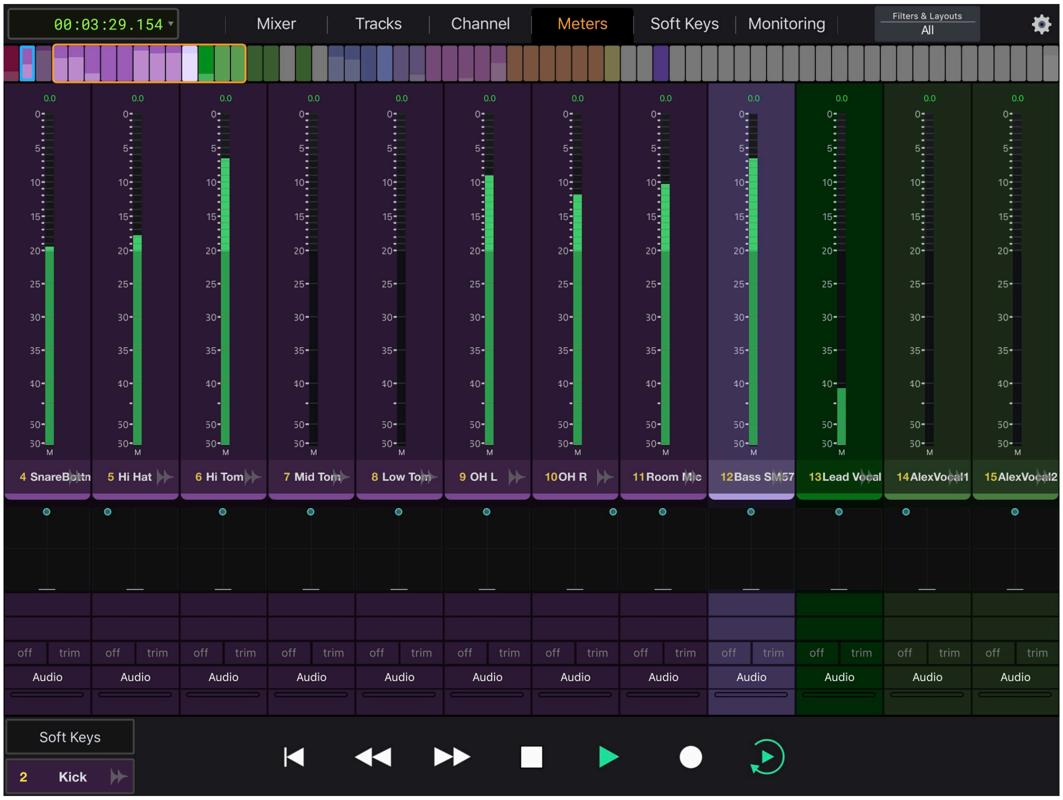


Figure 15.4: MIOConsole3d Meters View in Avid Control

Avid Control Channel Pane

The Channel view in Avid Control provides access to the EuCon knob sets. MIOConsole3d only implements the Input and Pan knob sets. The Pan knob set is better addressed via the mixer page, but it can be accessed here as well.

The Input knob set provides control over the selected strip's headamp and input controls. You can use it to control the Input mode for channels that have this control (e.g. Line +4, Line -10, Mic, etc.). You can also control the +48v Phantom for channels with preamps, the phase invert for all input channels, and the headamp gain for channels with digitally controlled headamp gain.

Even though there other knob sets are not implemented in 3d, there is no visual indication that they are not available, other than the fact that pressing the associated button does not select anything. This appears to be a bug in EuCon or Avid Control.

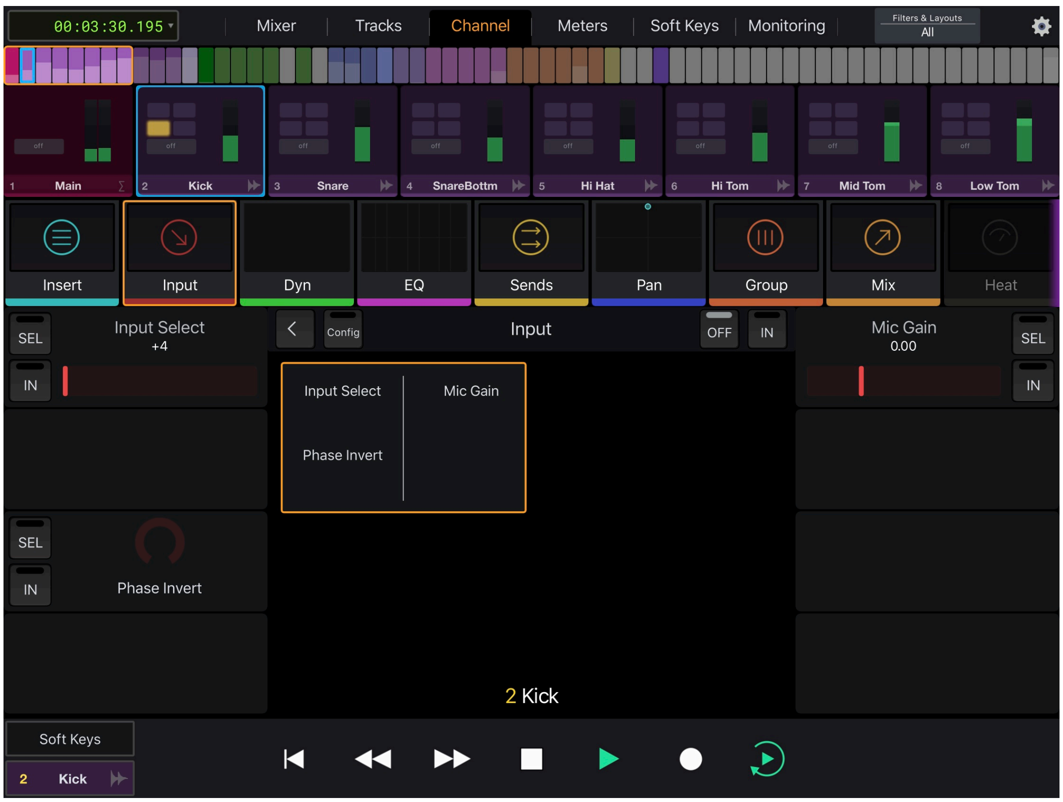


Figure 15.5: MIOConsole3d Channel View Input Knob set in Avid Control

Avid Control Monitor Pane

MIOConsole3d maps the Monitor, Cue and Talkback controllers onto the EuCon Monitor section. This means that the monitor controls are available on all EuCon surfaces that support the Monitor Control section. The Avid Control Monitor is especially flexible as it is implemented on the touch screen.

The various controls on the Monitor Pane are labelled below.

The top row allows you to choose the MC source; the mapping between MC sources and EuCon sources is not fixed - you have to choose which MC sources you want to use within the EuCon surface. If you look closely, you will see that the buttons are split between top and bottom. Tapping the top of the button will show a selector list in the bottom section of the window that lists all of the sources in the MIO MC; you choose from that list to assign it to the button you tapped in the Source selector area. Tapping the bottom of one of the source buttons will select that source in the MIO Monitor Controller.

The next row down are the speaker enables; this area will automatically update based on the selected output path and the speakers in the path. When a button is illuminated, the speaker is enabled (not muted). Tap a button to mute it. Tap again to unmute. Tap "All" to enable all the speakers. This area supports from Mono all the way up to 7.1.4 Atmos.

On the right hand side towards the center, you find the controls for MC output selection; there are only three of these, and they map to the first three MC output paths. Unfortunately, the button list is not dynamic, and the names of the buttons do not update to reflect the names you set in MIOConsole; this is a EuCon limitation. When you change the output path, the speaker enables will update to reflect the output path configuration.

On the right hand side, you find the rest of the controls for the MC output selection; including Talkback Enable, MC Dim, MC Mute, and MC Gain.

Above the bottom of the screen towards the center-left, you find the folddown controls which map onto the MC Mono button.

Finally on the left hand side, you find the Mute and Gain controls for the first 4 Cue Controllers you instantiate in MIOConsole3d.

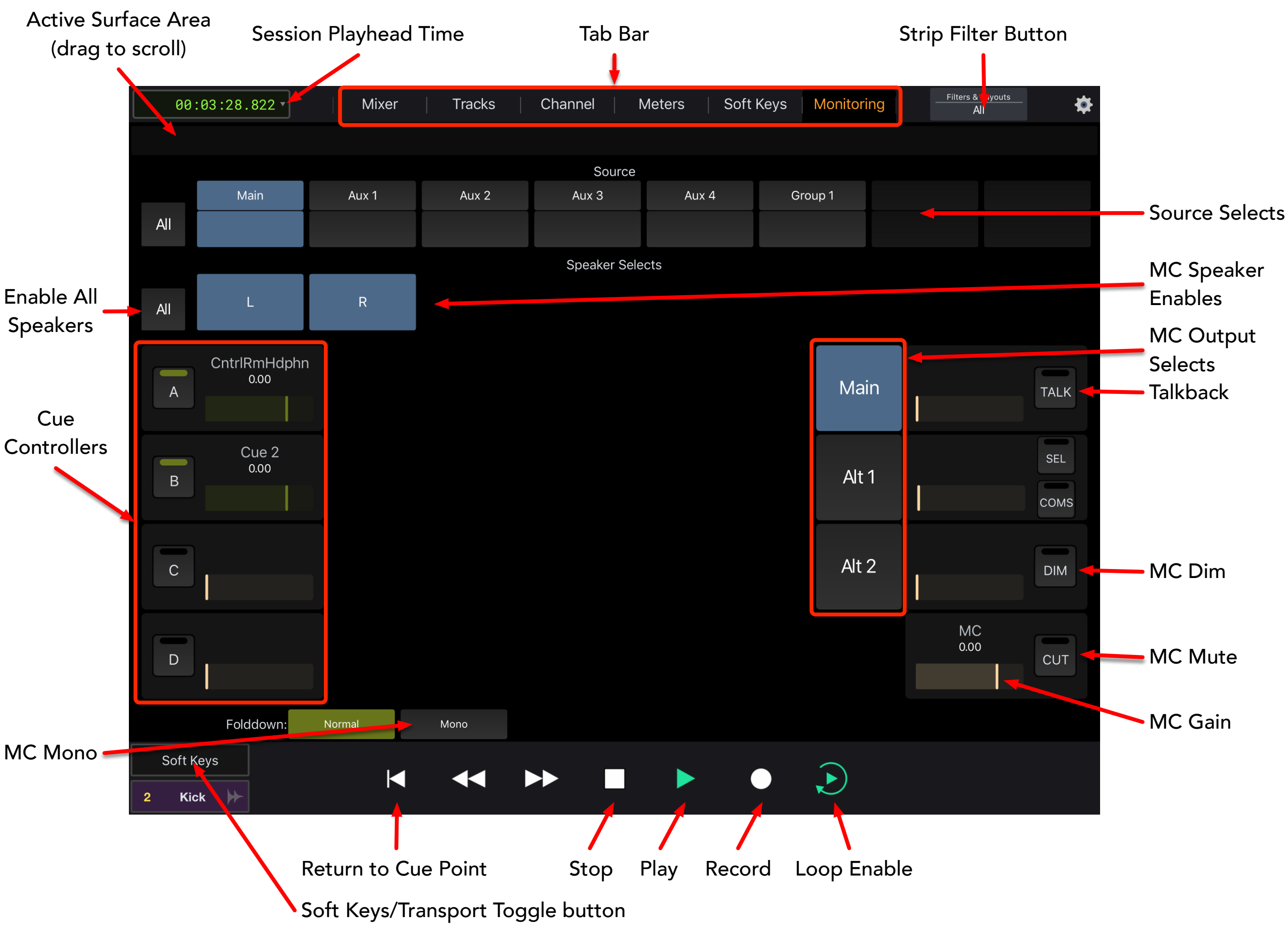


Figure 15.6: MIOConsole3d Monitor View in Avid Control

Avid Control Filter Control

As mentioned above, each of the strips is tagged with its strip type. When you tap on the Strip Filter button at the top-right of the screen you will see the Filter panel that lists the different types of strips. Normally all strips will be shown, but you can tap the on/off button for a specific type to set a filter that only shows the strip types that are on. This makes it easy to find the master fader, or aux masters, for example.

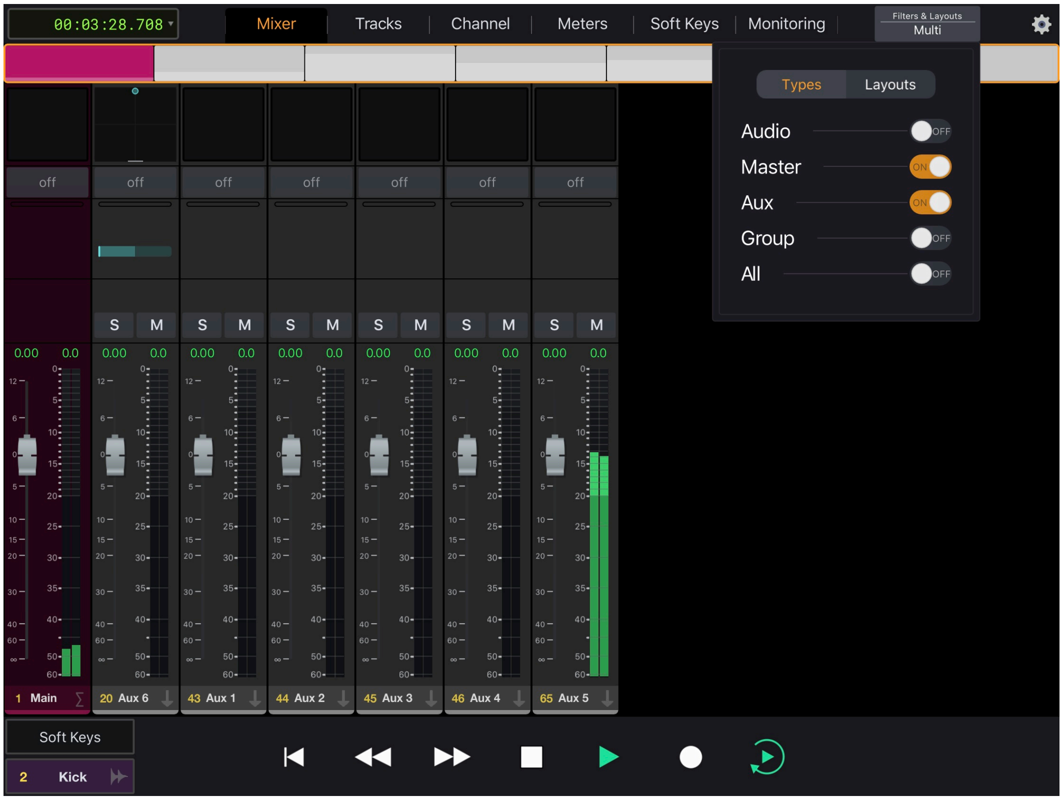


Figure 15.7: MIOConsole3d Mixer View Strip Filter in Avid Control

Avid Control Bus Soft Keys

Finally, you can switch the bottom section of the screen to show a cluster of soft keys. MIOConsole3d pre-configures these keys to select a specific bus to place on the faders. The soft keys allow you to directly select the Main bus, and Aux 1-9. You can also step forward and backwards through the bus list.

Unfortunately, the soft keys in EuCon are not dynamic, so the names on the buttons will not be updated to reflect the names you set in MIOConsole3d, nor will they indicate which specific bus is currently on the faders.

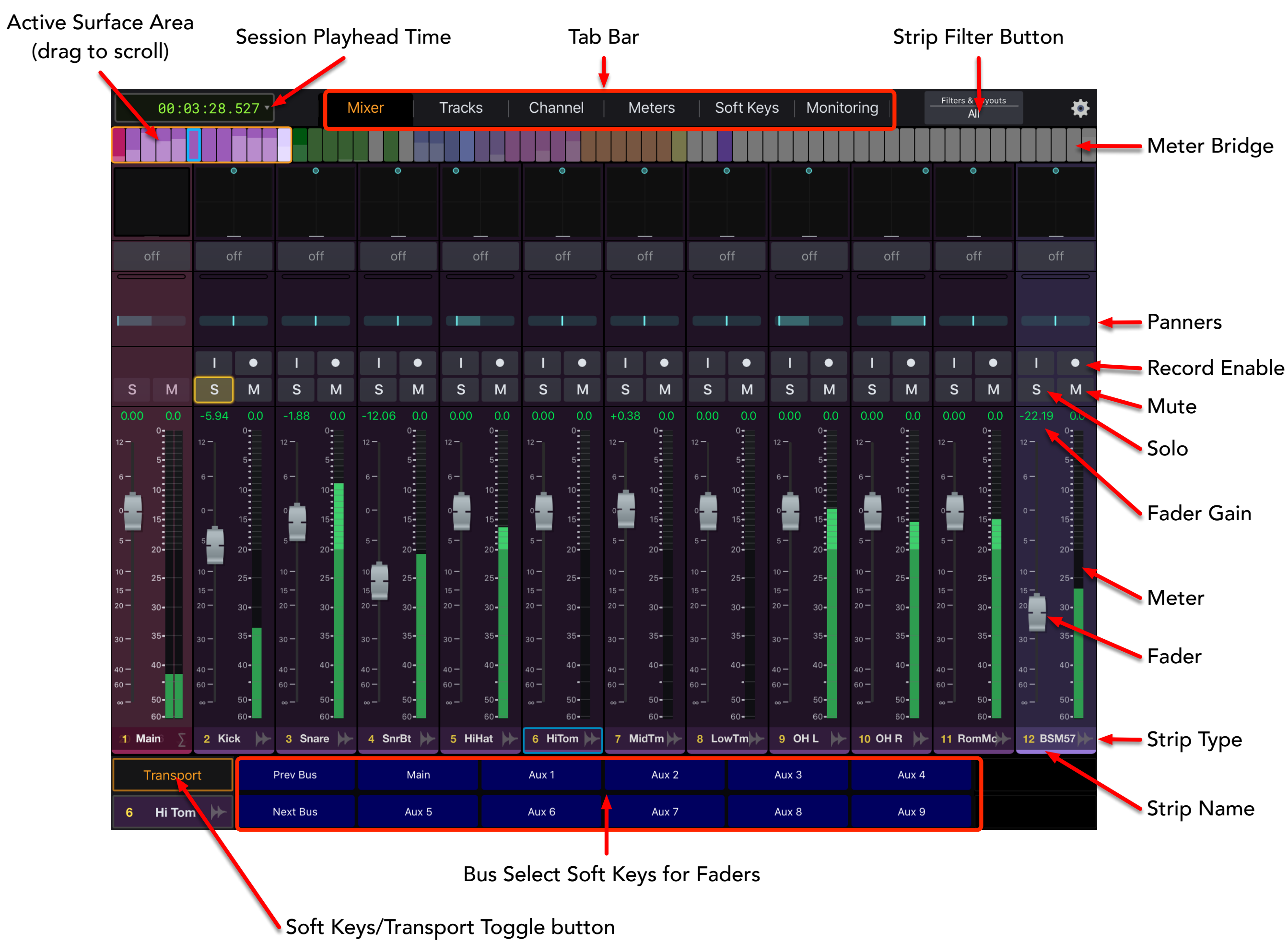


Figure 15.8: MIOConsole3d Mixer View Main Bus in Avid Control

EuCon Control Room Support

As described above, MIOConsole3d maps the Monitor Controller onto the EuCon monitor section. While this is well supported by Avid Control, the physical control surfaces have varying degrees of support for the monitor section. Please refer to the Avid documentation for your specific control surface to see what Monitor Controller features are supported.

Mixer Model

MIOConsole3d maps the mixer strips onto the EuCon surface in the order that they appear in the mixer window. If you want to change the order of the strips on the surface, simply change the order of the strips in the mixer window. As you re-order the strips, MIOConsole3d will update the control surface.

Since each Aux bus in the 3d Mixer has its own strip corresponding to the send, MIOConsole3d supports switching the EuCon model to map Aux Bus mixers onto the surface. You use the provided soft keys to select the Aux bus or Main bus to be placed on the faders. The associated pan, solo and mute controls are also placed onto the surface.

> EuCon controllers operate mix buses independently of the bus in focus on the MIOConsole3d desk, such that two engineers may control separate mixes on the same console simultaneously.
>
> Aux buses that are hidden from the MIOConsole3d mix desk (i.e. Configure Mixer: Aux Buses > 'Not on Strip') may still be operated from the control surface.

MIOConsole3d will automatically configure the EuCon strips to match the configuration of the underlying mixer strip (including channel name, panner, number of channel meters, input controls, etc.)

In the EuCon environment, the channel controls are mapped onto so-called 'knob-sets'.

On the MC Control, you access the knob set by pressing the soft-knob labelled with the knob set name. The knob sets that are supported by MIOConsole3d are the "Pan" and "Input" knob sets.

On the MC Mix, the knob sets are selected via the buttons on the left hand side of the surface. Again, the operative knob sets are "Pan" and "Input" knob sets (as well as the "Aux" and "Mix" knob sets for switching amongst the send busses - but these need to be avoided due to the bug described above).

Mackie Control Protocol (MCP) Details

The Mackie Control Protocol is supported by the Mackie Control Universal (with integrated USB interface) and Mackie Control Universal XT Fader pack products. It is also supported by the older Mackie Control products. In addition to the Mackie products, the Mackie Control Protocol is emulated by a number of third-party USB and MIDI controllers.

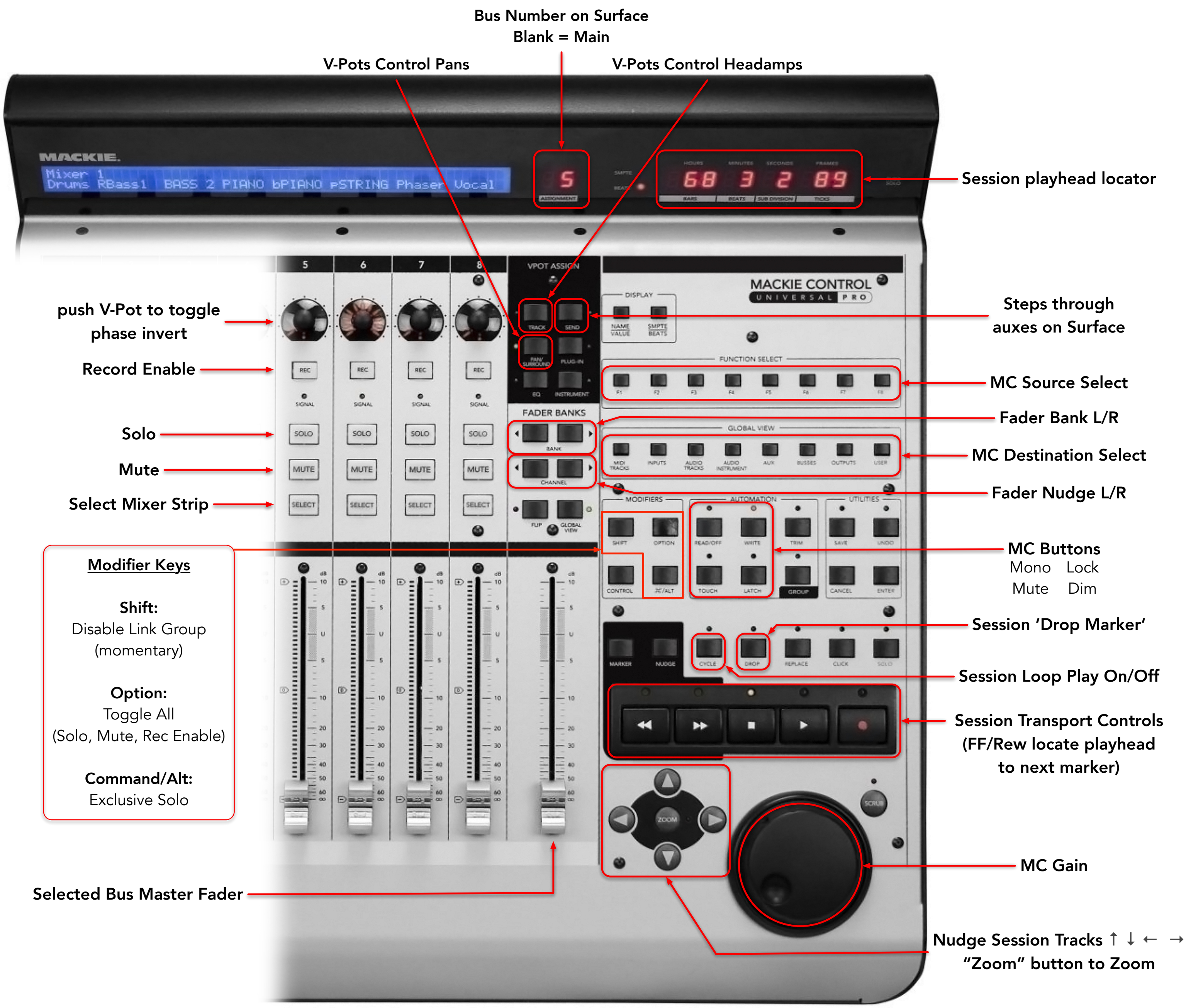


Figure 15.9: 3d Mackie Control UI map

With the Mackie protocol, only the master section provides surface navigation buttons for nudging and banking as well as input, pan and send selection, so with most control surface implementations you need at least one master section to be able to access these features.

With MIOConsole3d, we have duplicated some of these control surface commands as mappable keyboard commands. This allows you access these functions from the keyboard of your computer, or even better through a mappable HID like the ShuttleXpress or ShuttlePro. Since you can access these commands without having the master section, it is possible to configure a control surface by just using fader packs, which can provide a cost savings and a density improvement for certain applications.

Mackie Control Default Key Commands

Command	**Key Sequence**
Select Main Mix Bus on Control Surface	^⌥0
Assign V-Pots to Panner	^⌥1
Assign V-Pots to Input Gain	^⌥2
Select Next Aux Bus on Control Surface	^⌥3
Fader Bank Up (to the right)	^⌥4
Fader Bank Down (to the left)	^⌥5
Fader Channel Up (to the right)	^⌥6
Fader Channel Down (to the left)	^⌥7

Table 15.1. Mackie Control Default Key Commands

Control Room Support

The MCP control surface Jog wheel is mapped to the master volume on the Monitor Controller, and may be used to control your main monitor level.

Monitor Controller Source selection is currently mapped to Function Select buttons F1 - F8.

Monitor Controller Output selection is currently mapped to the “View by Type” row of eight buttons (vertically adjacent to F1 - F8), labeled (in order, left to right): MIDI Tracks, Inputs, Audio Tracks, Audio Instrument, Aux, Buses, Outputs, and User.

Monitor Controller Mono, Mute, Lock and Dim are currently mapped to the four automation buttons on the MCU: Mono -> Read/Off, Mute -> Touch, Lock -> Write, and Dim -> Latch

Mixer Model

MIOConsole3d maps the mixer strips onto Mackie Control surfaces in the order that they appear in the mixer window, left to right onto the units in the following order: Fader Pack 3, Fader Pack 2, Fader Pack 1, Primary Controller. The current implementation of MCP in MIOConsole3d assumes that the master control module should occupy the Primary slot in the MIOConsole Control Surface Preferences Input Port and Output Port selectors.

- Changing the order of mixer strips in the MIOConsole3d mix desk will be immediately update the control surface to match.
- The **Fader Bank** and **Fader Nudge** buttons allow you to move the faders on the surface across the mixer in banks of eight or one at a time, respectively. MIOConsole3d will automatically update the scribble strips, metering, fader levels, VPot readouts and button states as you move across the mixer.
- The **Send** button steps controller focus through the Main and Aux mixer fader arrays. When the control surface is running an Aux bus mixer, the controller master fader operates the master bus return fader from that Aux.
  The Assignment display on the control surface indicates which mixer bus is in focus. The display will be blank when the controller faders are focused on the Main mix bus, and will read ‘01’ when controlling Aux 1, ‘02’ for Aux 2, and so on. Note that the control surface does not have a way to display the name of your Aux - it can only follow the order of the Aux bus as shown in the Configure Mixer: Aux Buses list.

  The MCP controller operates mix buses independently of the bus in focus on the MIOConsole3d desk, such that two engineers may control separate mixes on the same console simultaneously.

  Aux buses that are hidden from the MIOConsole3d mix desk (i.e. Configure Mixer: Aux Buses > ‘Not on Strip’) may still be operated from the control surface.
- Press the V-Pot knob switch to toggle **Polarity Invert** on the associated strip.
- **Record Enable** toggles the record enabled state for that mixer strip.
- **Solo** toggles solo mode on/off for that mixer strip.
  Use the control surface **<⌘/ALT>** modifier key plus the solo button to Exclusive Solo. Exclusive Solo of a linked mixer strip overrides the link group.
- **Mute** toggles the Mute state of each strip.
- The **Select** button on each strip will toggle the select state of the associated mixer channel strip. You can use this with the mixers “Selected Strips” linking feature to make ad-hoc multi-channel adjustments to the mixer.
  Hold the control surface **<Option>** modifier key to toggle Record Enable, Solo and Mute across all mixer strips.
  Hold the control surface **<Shift>** modifier key to operate only a single channel within any active link group.
- **Play**, **Record** and **Stop** operate Session transport and also reflect the state of the transport
  << and **>>** keys will immediately relocate the playhead to the previous marker or next marker location, respectively. Marker relocation works the same with transport stopped or playing. New markers dropped during transport play will not be registered until play is stopped.
- The Timecode readout displays the current playhead location in HHH.MM.SS.sss format.

All other master section controls are currently unassigned.

Part IV. EdgeCard™s

Table of Contents

16. EdgeBus™ expansion port

Introduction to EdgeBus

Every Metric Halo 3d motherboard includes a high-speed peripheral I/O interface; the EdgeBus slot.

A variety of digital audio expansion modules are available for the EdgeBus expansion slot. The modules themselves are called EdgeCards™.

The current crop of EdgeCards can provide anywhere from 4 to 256 extra channels of digital I/O per card. Naturally, if you have more than one 3d unit in your MHLink domain, any port in any EdgeCard may be routed directly to or from any other physical or computer port available in the domain, the same as any built-in port.

Figure 16.1: EdgeBus expansion port (2882/ULN-2 base configuration SPDIF•AES EdgeCard shown at left)

The graphic above shows the EdgeBus port of a 3d ULN-2 populated with the base configuration SPDIF • AES EdgeCard on the left, and at the right an empty LIO-8 EdgeBus port covered with a blank panel.

EdgeCards are designed to be convenient, field-installable and bulletproof.

Detailed instructions for installation and removal of EdgeCards are provided at the end of this chapter.

When an EdgeCard is installed, it will be identified in the Unit Status Display of the Status Pane in MIOConsole3d:

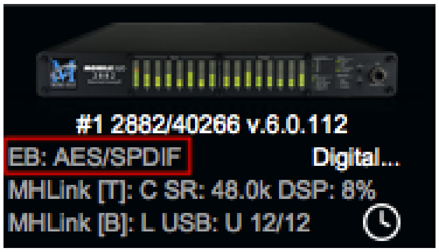


Figure 16.2: Status Pane: Unit Status Display: EdgeBus identifier (outlined in red)

Controls for the EdgeCard ports will become available in the Units' Digital I/O Status/Control Menu:

Figure 16.3: Status Pane: Unit Status Display: Digital I/O Status/Control Menu with EdgeCard ports

And its I/O port channels appear in the MIOConsole3d routing UI categorized as "EdgeCard" digital audio routes:

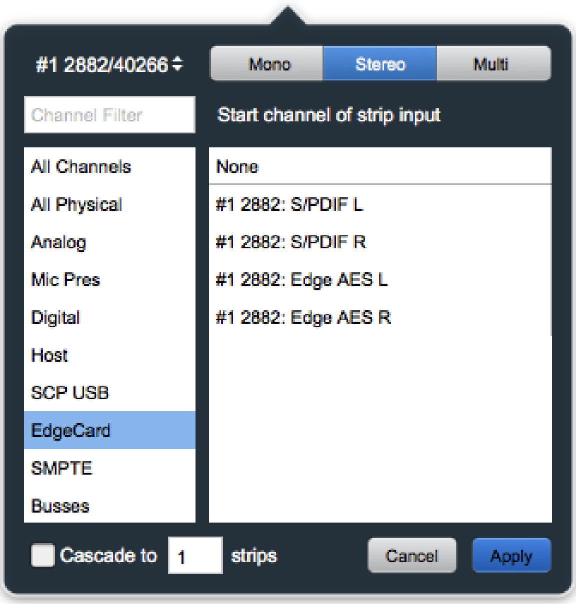


Figure 16.4: The "EdgeCard" category in the MIOConsole3d Routing UI

See Digital I/O Status/Control for all digital formats port configuration and clocking details.

The EdgeCard family

- SPDIF • AES

Figure 16.5: MH EdgeCard: SPDIF • AES

The single SPDIF • AES EdgeCard provides independent RCA unbalanced AES-3 standard copper and XLR balanced AES-3 standard digital I/O ports, each supporting stereo operation at 44.1-192kHz.

This card is included pre-installed in the base configuration of both 3d ULN-2 and 2882 models as shown above.

- SPDIF • ADAT x2

Figure 16.6: MH EdgeCard: SPDIF • ADAT x2

The SPDIF • ADAT x2 EdgeCard provides one RCA copper SPDIF I/O port plus two independent ADAT/TOSLINK optical digital I/O ports.

EdgeCard optical input ports auto-sense and auto-configure to the incoming digital audio stream, whether TOSLINK stereo or ADAT S/MUX.

The output format for each optical port is manually selectable, and is independent of the input format of the selected port.

- SPDIF x2 • ADAT x2

Figure 16.7: MH EdgeCard: SPDIF x2 • ADAT x2

The SPDIF x2 • ADAT x2 EdgeCard provides two independent RCA copper SPDIF I/O ports plus two independent ADAT/TOSLINK optical digital I/O ports.

EdgeCard optical input ports auto-sense and auto-configure to the incoming digital audio stream, whether TOSLINK stereo or ADAT S/MUX.

The output format for each optical port is manually selectable, and is independent of the input format of the selected port.

> Due to a physical interference in both the 2882 and ULN-2 chassis, it is not possible to insert the SPDIF x2 • ADAT x2 EdgeCard into the EdgeCard™ Slot on these models. As a result, this card is not compatible with either the 2882 or ULN-2.
>
> The SPDIF • ADAT x2 and all other EdgeCards fit perfectly in the 2882 and ULN-2 (and all other 3d hardware units).

- ADAT x4

Figure 16.8: MH EdgeCard: ADAT x4

The ADAT x4 EdgeCard provides four independent ADAT/TOSLINK optical digital I/O ports.

As with the built-in ports on the 2882 and ULN-2, each optical EdgeCard input port auto-senses and auto-configures to the incoming digital audio stream, whether TOSLINK stereo or ADAT S/MUX.

The output format for each port is manually selectable, and is independent of the input format of the selected port.

- TOSLINK mode supports stereo audio at all sample rates from 44.1k to 192kHz.
- ADAT mode supports standard ADAT and ADAT S/MUX formats, providing eight channels of I/O at 1x sample rates, four channels at 2x, and two channels at 4x sample rates.

- **SPDIF • MIDI**

Figure 16.9: MH EdgeCard: SPDIF • MIDI

The SPDIF • MIDI EdgeCard provides an RCA copper SPDIF I/O port and a standard 5-pin MIDI IN / MIDI Out port.

The SPDIF RCA I/O supports stereo operation at 44.1-192kHz.

The EdgeCard MIDI I/O is currently supported on the 3d USB connection from its installed 3d device to your computer. When you connect a 3d device via USB, the device will automatically show up as a MIDI device in AMS/CoreMIDI. Port 1 is currently reserved for the internal MIDI ports on the ULN-8 and LIO-8. The EdgeCard MIDI will be Port 2.

- **Eight Channel AES**

Figure 16.10: MH EdgeCard: Eight Channel AES

The Eight Channel AES EdgeCard operates in single-wire mode, providing 8 channels of digital audio at all sample rates from 44.1 to 192kHz. The ULN-8's digital I/O connections are made using industry standard Tascam/Digidesign pinout DB25 cables.

Wiring details for this connector can be found in the DB25 Pinouts appendix.

- MADI Optical x2

Figure 16.11: MH EdgeCard: MADI Optical x2

The MADI Optical x2 EdgeCard provides two independent full-duplex SC glass fibre-optic MADI I/O ports.

EdgeCard MADI input ports auto-sense and auto-configure to the incoming MADI stream, fully supporting both 56 and 64-channel frame formats and S/MUX data configurations at all sample rates from 44.1 to 192kHz.

The output format for each optical port is manually selectable, and is independent of the input format of the selected port. This allows the MADI EdgeCard to operate as a realtime re-clocking formats converter between legacy and modern digital audio transmission standards.

- MADI Copper • Optical

Figure 16.12: MH EdgeCard: MADI Copper • Optical

The MADI Copper • Optical EdgeCard provides one full-duplex BNC copper MADI I/O, and one full-duplex SC glass fibre-optic MADI I/O port.

Both EdgeCard MADI input ports auto-sense and auto-configure to the incoming MADI stream, fully supporting both 56 and 64-channel frame formats and S/MUX data configurations at all sample rates from 44.1 to 192kHz.

The output format for each port is manually selectable, and is independent of the input format of the selected port. This allows the MADI EdgeCard to operate as a realtime re-clocking formats converter between legacy and modern digital audio transmission standards.

- MADI Copper x4

Figure 16.13: MH EdgeCard: MADI Copper x4

The MADI Copper x4 EdgeCard provides four independent full-duplex BNC copper MADI I/O ports.

EdgeCard MADI input ports auto-sense and auto-configure to the incoming MADI stream, fully supporting both 56 and 64-channel frame formats and S/MUX data configurations at all sample rates from 44.1 to 192kHz.

As with the optical MADI ports, the output format for each copper port is manually selectable, and is independent of the input format of the selected port. This allows the MADI EdgeCard to operate as a realtime re-clocking formats converter between legacy and modern digital audio transmission standards.

EdgeCard Installation Guide

Introduction

A video version of this guide is available here.

You will be working with electronic equipment so we strongly advise you to take the necessary steps to ground yourself and to work in an environment with minimal static to avoid the potential for electrostatic discharge.

Please familiarize yourself with the parts and instructions before proceeding with installation.

Required Tools

You will need the following tools to do the installation:

- #2 Phillips Head Screwdriver

Figure 16.14: #2 Phillips Head Screwdriver

Installation Components

A variety of EdgeCards are available for expanding the capabilities of your interface with optional I/O. They come with a metal cover plate over the various I/O ports, two retentive screws, and rubber feet on the bottom.

- EdgeCard

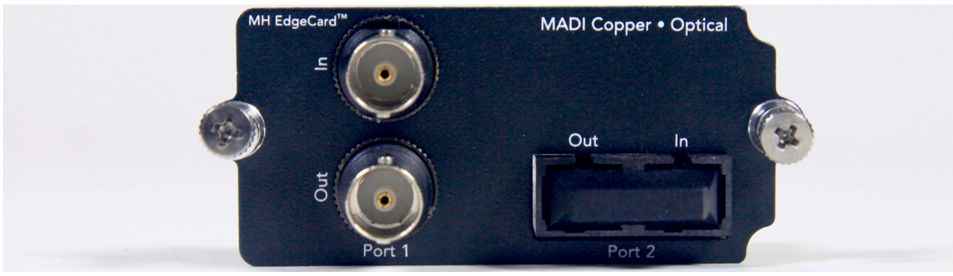


Figure 16.15: Copper/Optical MADI EdgeCard

> Please familiarize yourself with the parts and instructions before opening your interface.

> Be sure to discharge any static energy on your body before touching the interior of the interface.

Installation

1. Fully power off, disconnect power supply and all other connections to the unit.
2. Touch a metal object in your work area other than the interface or EdgeCard to discharge static.
3. Remove the cover plate from the back your interface covering the Edge Bus using a screwdriver.

Figure 16.16: Remove Cover Plate

4. Remove the EdgeCard from the static bag, guide the card inside the hole in the back with the metal EdgeCard plate centered and aligned with the receiver hole. Let the rubber feet track along the bottom of the unit.

   You will be able to feel when the EdgeCard finds its position in the receiver slot.

   Push into the EdgeBus receiver slot on the 3d card inside the unit so it slots into place as it becomes flush with the back panel.

   There is *never* a need to force the EdgeCard into the receiver slot.

Figure 16.17: Insert EdgeCard into Slot

5. Turn the two screws to tighten the board in place.

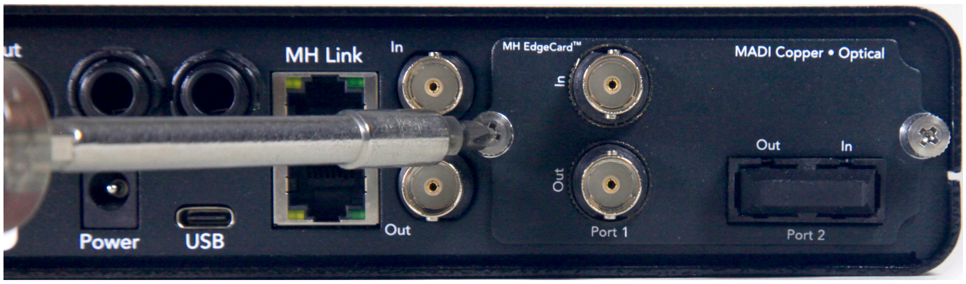


Figure 16.18: EdgeCard Installed Available for Routing in MIOConsole3d

6. The next time you boot the interface, the new I/O from the EdgeCard will be discoverable in software and made available to your system.

Figure 16.19: EdgeCard Available for Routing in MIO Console 3d

The installation is finished!

If you have any questions about any steps in the preceding instructions or run into any problems, please reach out by emailing support@mhsecure.com with the subject "EdgeCard Installation".

Removal

Removal of an EdgeCard is really just a reversal of the installation process:

1. Disconnect all audio connections from the EdgeCard,
2. Power down and disconnect power from the 3d unit,
3. Ground yourself and the box chassis to eliminate any static charge,
4. Unscrew the thumbscrews fully (see below),
5. Carefully pull the EdgeCard out, and
6. Store the module in an anti-static bag inside a protective container, or better yet in another 3d unit.

In practice, there are two additional things to keep in mind when removing an EdgeCard (especially when you're in a hurry and the unit is buried in a rack).

First, the EdgeCard thumbscrews were designed for use (and likely abuse) of life on the road. So, although the thumbscrews are locked into the EdgeCard faceplate, for the mechanical security of the module these screws are longer than you might at first expect. When removing an EdgeCard, keep unscrewing until the thumbscrews are sprung all the way out and are wiggling freely before pulling the card from the EdgeBus connector.

Second, if at all possible, cover the open EdgeBus port by securely screwing in a blank port cover. Like any piece of electronic equipment, protection of the internal bits is key to keeping it working.

EdgeCard module storage

Generally, the safest place to keep any EdgeCard module is installed in a 3d box, even if you aren't using it at the moment. It shows up in the MIOConsole Status Pane, so you always know where it is, and it is protected both physically and electromagnetically.

If you must store an EdgeCard module externally, be sure to keep it securely in an anti-static bag within a well-padded box or envelope at all times when not in use. Larger boxes and envelopes are also harder to misplace, so it's a win-win: more protection and easier to find in a pinch.

Part V. Appendices

Table of Contents

A. MIOConsole3d Key Commands

Assigning / Modifying Key Commands

MIOConsole3d supports fully user-editable modifier-key and keystroke commands to improve efficiency and ease mousing around the Console. In many cases these key commands are supplied with an eye to compatibility with third-party HID (Human Interface Device) controller device (for example, a Contour Shuttle Pro), or programmable gaming controllers.

To view Key Commands editing window, you can select the **Edit > Edit Key Commands...** menu item, or type the **⌥K** (Option + K) key command

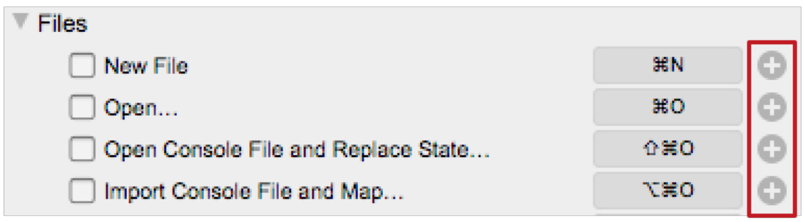


Figure A.1: Edit Key Commands window: create new Key-Command buttons

- To enter a new key command to an unassigned function, simply click the "**+**" button for the command in the list and the "New Key-mapping" window will appear:

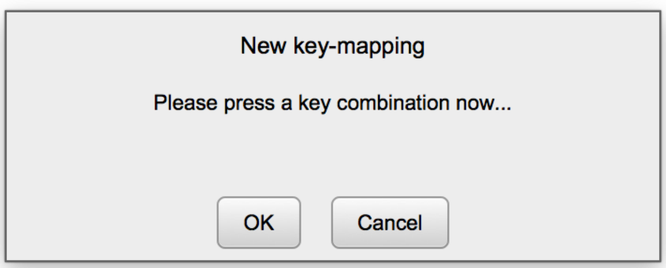


Figure A.2: Edit Key Commands window: Enter new key-command

Type the new key sequence you would like to use and click the OK button to complete the re-assignment.

To modify an existing keystroke, click the "**+**" button or the current command keystroke button

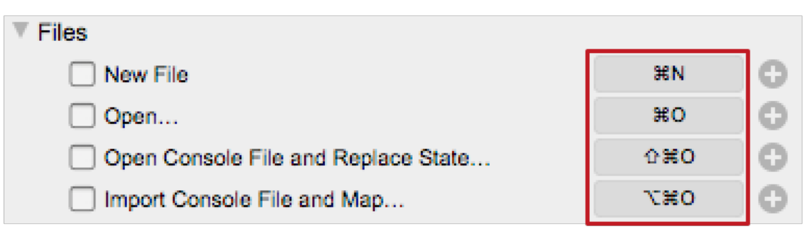


Figure A.3: Edit Key Commands window: Current key-command assignment

and enter your new key command for that function.

- A dialog box will pop up to indicate any key map conflicts.

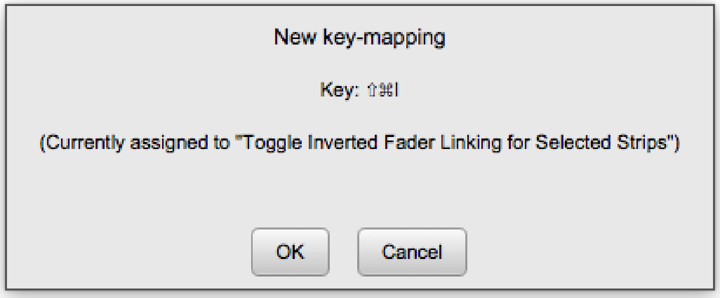


Figure A.4: Edit Key Commands window: Conflict box

This dialog box reveals the keystroke in question as well as the command that keystroke is currently assigned to.

Click 'OK' to proceed.

Figure A.5: Edit Key Commands window: Over-write warning window

The above warning window will appear, giving you one last chance to avoid over-writing the existing keystroke.

- Right-clicking an existing keymap entry will provide options to 'Reset' the current key-mapped command to it's default setting, or 'Remove' the key command entirely.

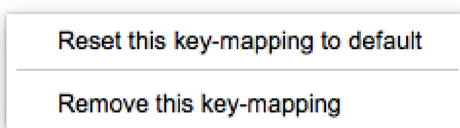


Figure A.6: Reset / Remove Key-map Entry box

- The easiest way to locate a command within *Edit Key Commands* is to use the *Command Filter* search field at the bottom of the window.

  Use the *reset to defaults* button to reset all key commands to their factory default assignments.

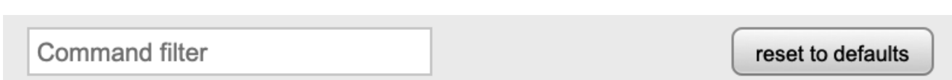


Figure A.7: Command Filter search & full key command Reset

Key Commands: MIOConsole3d Menu Bar menu items

> Please Note: Based on user feedback, the categories and commands in the *Edit Key Commands* window are listed alphabetically.
>
> In order to maintain logical function grouping and continuity with the MIOConsole menu structure, the Key Command listings in this chapter mirror the menu layout as much as possible.

The following tables list all of the default key commands that can be edited.

Menu Key Command table headers (blue text) are links to the respective MIOConsole3d Menu Bar menu item description pages.

⇧ = Shift	^ = Control	⌥ = Option	⌘ = Command

Parameter "Nudge"

Any numeric control element in the MIOConsole3d user interface (faders, panners, EQ frequency, etc.) can be "Nudged" to increment or decrement the parameter value. This command is available throughout the MIOConsole3d environment.

The amount of nudge varies by the type of parameter. For example, Fader Gain 'coarse' nudges are 5dB steps per nudge and 'fine' nudges are 0.5dB.

Pressing and holding the key command will repeat the nudge command per the Keyboard: "Key Repeat" settings in your computer operating system preferences.

Command	Key Sequence
Nudge Up (coarse)	^⌥Z
Nudge Down (coarse)	^⌥X
Nudge Up (fine)	⌥Z
Nudge Down (fine)	⌥X

> See also Mix Desk Command & Control for information regarding scrollwheel and gesture control options within the MIOConsole3d environment.

Application menu key commands (the *MIOConsole3d* menu)

Note: Application menu entries *Hide MIOConsole3d*, *Hide Others* and *Show All* are global OS commands, and are not included in MIOConsole3d **Edit Key Commands**.

Command	Key Sequence
About MIO Console…	unassigned
Check for Update…	unassigned
Download Update…	unassigned
Preferences…	⌘,
Register Attached Units…	unassigned
Quit	⌘Q

⇧ = Shift	^ = Control	⌥ = Option	⌘ = Command

File menu key commands

Note: File menu entry *Open Recent* is an OS function, and is not included in MIOConsole3d **Edit Key Commands**.

Command	**Key Sequence**
Open…	⌘O
Open Console File and Replace State…	⇧⌘O
Import Console File and Map…	⌥⌘O
Open Template…	^⌥⌘O
Close	⌘W
Save…	⌘S
Save As…	⇧⌘B

Edit menu key commands

Command	**Key Sequence**
Undo	⌘Z
Redo	⇧⌘Z
Copy	⌘C
Cut	⌘X
Paste	⌘V
Clear	unassigned
Select All	⌥⌘A
Deselect All	⌘V
Edit Key Commands	⌥K

Utilities menu key commands

Command	**Key Sequence**
Install Current Firmware…	unassigned
Reboot Attached Units	unassigned
Install Current MHLink Driver…	unassigned

⇧ = Shift	^ = Control	⌥ = Option	⌘ = Command

Session menu key commands

(listed by Session submenu)

Session Files submenu

Command	Key Sequence
New Session…	⌥⌘N
Find Missing Session Files…	unassigned
Set Record Folder…	⌘T
Add Record Mirror Folder…	⇧⌘T
Import Take Folder…	⌘Y
Import folder of Take Folders…	⇧⌘Y

Session Edit submenu

Command	Key Sequence
Split All Segments at Playhead	⌘/
Split All Segments at Loop Points	⌥⌘/
Split All Segments at Selection Boundary	unassigned
Trim All Segments to Loop Points	⌥L
Trim Segments to Selection	unassigned
Trim Top of All Segments to Playhead	⌥T
Trim Tail of All Segments to Playhead	⌥⇧T
Trim Top of Segments to Selection	unassigned
Trim Tail of Segments to Selection	unassigned
Align All Segments to Grids	⌥A
Mute All Segments	⌥M
Unmute All Segments	⌥⇧M

Session Export submenu

Command	Key Sequence
Export Tracks…	unassigned
Export Tracks Between Loop Points…	unassigned
Export Selection…	unassigned

⇧ = Shift	^ = Control	⌥ = Option	⌘ = Command

Session Transport submenu

Command	Key Sequence
Start Recording…	⌥⌘R
Stop Transport…	⌥⌘T
Start Playback…	Spacebar
Return To Next Cue Point	↩ (Return)
Enabled Return Points (selection submenu)	*key command toggles selections listed below:*
—> Enable Return To Last Play Start	unassigned
—> Enable Return To Selection Start	unassigned
—> Enable Return To Loop Start	unassigned
—> Enable Return To In Point	unassigned
—> Enable Return To Session Start	unassigned

Session Playback Scroll Mode submenu

Command	Key Sequence
Continuous	unassigned
Paged	unassigned
None	unassigned

Session Playback Mode submenu

Command	Key Sequence
From Playhead	unassigned
From Playhead; Return on Stop	unassigned
From Loop Start to End	unassigned
From Session Start to End	unassigned

Session Waveform Scaling submenu

Command	Key Sequence
Enhance Peaks	unassigned
Normal	unassigned
Enhance Low Level	unassigned
Extra Enhance Low Level	unassigned

⇧ = Shift	^ = Control	⌥ = Option	⌘ = Command

Session Looping submenu

Command	Key Sequence
Loop Playback	⌘L
Set Loop when Cueing to Mark	⌥⌘L
Set Loop Start to Playhead	⇧⌘<
Set Loop End to Playhead	⇧⌘>
Set Loop to Selection	unassigned

Session Autopunch submenu

Command	Key Sequence
Set In Point to Playhead	⌘[
Set Out Point to Playhead	⌘]
Set Autopunch to Selection	unassigned

Session Markers submenu

Command	Key Sequence
Drop Mark	⌘↩ *("Take" Record Mode only)*
Create Mark From Loop	unassigned
Create Mark From Selection	unassigned
Jump to Previous Mark	⌥⇧⌘<
Jump to Next Mark	⌥⇧⌘>
Save Markers…	unassigned
Load Markers…	unassigned

Session Zoom submenu

Command	Key Sequence
Zoom To Loop	⌥⌘Z
Zoom to Fit Selection	⇧Z
Increase Session Track Height	⌘↓
Decrease Session Track Height	⌘↑
Fit Tracks Vertically	⌥X
Zoom Tracks In Horizontally	⌘→
Zoom Tracks Out Horizontally	⌘←
Fit Tracks Horizontally	⌥Z
Autoscroll Recordhead Into View	unassigned
Autozoom Tracks Horizontally	unassigned

⇧ = Shift	^ = Control	⌥ = Option	⌘ = Command

Session View submenu

Command	Key Sequence
Show Segment Names	unassigned
Expand Track Lanes for Overlaps	L
Show Only Record Enabled Tracks	H

Session Tempo submenu

Command	Key Sequence
Calculate Tempo from Loop	unassigned
Calculate Tempo from In/Out Points	unassigned
Calculate Tempo from Selection	unassigned

Command	Key Sequence
Enable Metronome	^M
Enable Grid	^G
Recording Preferences	⌘R

⇧ = Shift	^ = Control	⌥ = Option	⌘ = Command

Mixer menu key commands

Command	Key Sequence
Configure Mixer…	⇧⌘C
Show System Status Pane	[
Show Unit Icons in System Status	⌥[
Route New Strips To Host	unassigned
Add Input Strip	⇧⌘A
Add Aux Bus	⇧⌥⌘A
Add Group Bus	⇧⌥⌘G
Input Strip Meters: Post-Fader	⇧⌘P
Toggle Parameter Linking for Selected Strips	⇧⌘G
Toggle Inverted Fader Linking for Selected Strips	⇧⌘I
Reset All Meter Peak Holds	⌘D
Delete Strips	⇧⌘D
Clear Mixer	unassigned
Spread Host Channels Across Boxes	unassigned
Select Pans on Control Surface	^⌥1
Select Input Gain on Control Surface	^⌥2
Select Next Bus on Control Surface	^⌥3
Select Main Bus on Control Surface	^⌥0
Shift Bank Up	^⌥4
Shift Bank Down	^⌥5
Shift Channel Up	^⌥4
Shift Channel Down	^⌥5
Set Mixer Strips Width (selection submenu)	*key command toggles selections listed below:*
—>Strip Width: Narrow	unassigned
—>Strip Width: Normal	unassigned
—>Strip Width: Wide	unassigned
—>Strip Width: Extra Wide	unassigned
Set Mixer UI Scale (selection submenu)	*key command toggles selections listed below:*
—>Mixer UI Scale: 75%	unassigned
—>Mixer UI Scale: 85%	unassigned
—>Mixer UI Scale: 100%	unassigned
—>Mixer UI Scale: 125%	unassigned
—>Mixer UI Scale: 150%	unassigned

⇧ = Shift	^ = Control	⌥ = Option	⌘ = Command

I/O menu key commands

Command	Key Sequence
2	unassigned
8	unassigned
16	unassigned
32	unassigned
64	unassigned
128	unassigned
Enable Tunnel Routing	unassigned

Monitor menu key commands

Command	Key Sequence
Save Domain Monitor Controller…	unassigned
Load Domain Monitor Controller…	unassigned
Show Monitor Controller	M
Make Monitor Controller Float	unassigned
Monitor Controller Slaves to PFL/AFL	unassigned
Show Cue Controller	C
Add Cue Controller	P
Add Monitor Source…	unassigned
Add Monitor Output…	unassigned
Edit Current Monitor Source…	unassigned
Edit Monitor Output…	unassigned
Edit Current Monitor Output Graph…	unassigned
Delete/Hide Current Monitor Source	unassigned
Delete Current Monitor Output	unassigned
Speakers Toggle Solo	unassigned

⇧ = Shift	^ = Control	⌥ = Option	⌘ = Command

Window menu key commands

Command	Key Sequence
Minimize	⌘M
Zoom	unassigned
Arrange In Front	unassigned
Switch Mixer/Record Window	⌘=
Show Mixer Window	X
Show Analog I/O Window	A
Show Record Window	R
Show Session Window	S

Help menu key commands

Command	Key Sequence
Open MIO Manual…	unassigned
Website	⌘⇧D
FAQ	⌘⇧P
Support	⌘B
New Bug Report…	unassigned
New Feature Request…	⌘⇧D
Generate MHLink Log…	⌘⇧P
Generate Full System Log…	⌘B
Reveal MH Logs…	unassigned

Key Commands: Non-Menu Bar functions

The following tables list all of the default key commands listed in *Edit Key Commands* which are not duplicated in a Menu Bar menu.

Menu Key Command table headers (blue text) are links to the main manual pages which describe the respective functions in detail.

⇧ = Shift	^ = Control	⌥ = Option	⌘ = Command

Monitor Controller key commands (non-menu)

Note: All Monitor Controller Key Commands default as "Global"

Command	Key Sequence
Toggle Dim	^⌥⌘D
Toggle Mute	^⌥⌘M
Toggle Mono	^⌥⌘N
Volume Down 0.5dB	^⌥⌘↓
Volume Up 0.5dB	^⌥⌘↑
Select Monitor Source 1	^⌥⌘1
Select Monitor Source 2	^⌥⌘2
Select Monitor Source 3	^⌥⌘3
Select Monitor Source 4	^⌥⌘4
Select Monitor Source 5	^⌥⌘5
Select Monitor Source 6	^⌥⌘6
Select Monitor Source 7	^⌥⌘7
Select Monitor Source 8	^⌥⌘8
Select Monitor Output 1	⌥⌘1
Select Monitor Output 2	⌥⌘2
Select Monitor Output 3	⌥⌘3
Select Monitor Output 4	⌥⌘4
Select Monitor Output 5	⌥⌘5
Select Monitor Output 6	⌥⌘6
Select Monitor Output 7	⌥⌘7
Select Monitor Output 8	⌥⌘8

Talkback Controller key commands (non-menu)

Command	Key Sequence
Enable Talkback	T
Enable Listenback	L

⇧ = Shift	^ = Control	⌥ = Option	⌘ = Command

Trigger key commands (non-menu)

…as in **Record Trigger mode**

Command	**Key Sequence**
Set Trigger: Auto Punch	unassigned
Set Trigger: Level	unassigned
Set Trigger: Manual	unassigned
Set Trigger: Timecode	unassigned

Control Surface key commands (non-menu)

Note: …key commands to remote control your Mackie Control Protocol remote controller by remote, found in *Edit Key Commands: Mixer* section.

Command	**Key Sequence**
Select Main Bus on Control Surface	^⌥0
Select Pans on Control Surface	^⌥1
Select Input Gain on Control Surface	^⌥2
Select Next Bus on Control Surface	^⌥3
Shift Bank Up	^⌥4
Shift Bank Down	^⌥5
Shift Channel Down	^⌥6
Shift Channel Up	^⌥7

B. DSP Graph Processor Documentation

Everything you ever wanted to know about the processing blocks available in the 3d Graph.

MIO Volume Control (Linear)

- Synopsis: MIO Volume Control - Linear Interpolation
- Category: Building Blocks
- Channels: 1, 2, 3, 4, 5, 6, 7, 8

Description:

The output of this plugin is the input with the specified gain applied to each input channel. The gain is specified by the `Volume` parameter in dB. The `Glide` parameter controls how quickly the applied gain will change when you change the `Volume` parameter. This version of the plugin changes the gain linearly between the old value and the new value. To change the gain logarithmically, use the MIO Volume Control (LPF) plug-in.

Parameters:

`Volume` [-120, 20] dB

Specifies the gain applied to the input signal.

`Glide` [0, 1k]

Specifies the rate at which the gain will change. If the Glide value is close to 0 (especially between .1 and 1), the gain will change slowly. If Glide is close to 1000, the gain will change quickly.

`Master Bypass` {On, Off}

When bypassed, the gain will be set to 0dB, passing the input signal unchanged.

MIO Volume Control (LPF)

- Synopsis: MIO Volume Control - Low Pass Filtered Interpolation
- Category: Building Blocks
- Channels: 1, 2, 3, 4, 5, 6, 7, 8

Description:

The output of this plugin is the input with the specified gain applied to each input channel. The gain is specifed by the `Volume` parameter in dB. The `Glide` parameter controls how quickly the applied gain will change when you change the `Volume` parameter. This version of the plugin uses a single-pole low pass filter (LPF) to change the gain between the old value and the new value logarithmically. To change the gain linearly, use the MIO Volume Control (Linear) plug-in.

Parameters:

`Volume` [-120, 20] dB

Specifies the gain applied to the input signal.

`Glide` [0, 1k]

Specifies the rate at which the gain will change. If the Glide value is close to 0 (especially between .1 and 1), the gain will change slowly. If Glide is close to 1000, the gain will change quickly.

`Master Bypass` {On, Off}

When bypassed, the gain will be set to 0dB, passing the input signal unchanged.

MIO Static Matrix

- Synopsis: Static Matrix Mixer
- Category: Building Blocks
- Channels: 2x2 through 16x16

Description:

MIO Static Matrix implements a matrix mixer with a 'static' crosspoint matrix, which means that gain changes are not interpolated. As a result, changing the gain of a crosspoint that has signal running through it will result in an audible click at the output of the associated bus.

Each crosspoint gain connects an input to an output bus. The gains multiply the input signal, and all the signals in each bus are summed to form the output of the bus. The gains in this implementation are specified as linear multipliers. This means that rather than being specified in dB, the gain specified in the UI is the actual multiplier coefficent used by the mixer. This means that the input signal is multipled by the number specified for the crosspoint gain. The gain that you specify for each crosspoint can range from -1.0 to 1.0.

When the gain is set to 1.0, this is equivalent to having a fader set to 0dB, which means that the signal will be pass through unchanged. This is easy to understand becaus anything multiplied by 1 is unchanged. As the coefficient you set is decreased, the gain is decreased as well, until the coefficient is set to 0. For example, if you use a coefficient of 0.5, the signal will be at half the original level, or a gain of -6.02 dB. When coefficent is 0, the signal is muted, and the effective gain is -∞.

When coefficient is less than 0, the crosspoint has the effect of both applying gain AND inverting the signal. The gain applied is the same as would occur with a positive coefficient, but the signal will be inverted.

Parameters:

`1->1` [-1, 1]

Sets the gain of input 1 mixed into output 1.

`2->1` [-1, 1]

Sets the gain of input 2 mixed into output 1.

`1->2` [-1, 1]

Sets the gain of input 1 mixed into output 2.

`2->2` [-1, 1]

Sets the gain of input 2 mixed into output 2.

`… and so on for the rest of the available channels.`

`Master Bypass` {On, Off}

Puts the mixer into direct route mode (e.g. all inputs are directly routed to the corresponding outputs, and no mixing occurs).

MIO Channel Summer

- Synopsis: Channel Summer
- Category: Building Blocks
- Channels: 2

Description:

Summer takes its inputs and forms the sum (A+B) of the two signals.

Parameters:

`Master Bypass` {On, Off}

See also:

MIO Channel Difference, MIO Channel Sum/Difference, MIO Channel Multiplier

MIO Channel Difference

- Synopsis: Channel Difference
- Category: Building Blocks
- Channels: 2

Description:

Difference takes its inputs and forms the difference (A-B) of the two signals.

Parameters:

`Master Bypass` {On, Off}

See also:

MIO Channel Summer, MIO Channel Sum/Difference, MIO Channel Multiplier

MIO Channel Sum/Difference

- Synopsis: Channel Sum/Difference

- Category: Building Blocks
- Channels: 2

Description:

Sum/Difference takes its inputs and forms the sum (A+B) and difference (A-B) of the two signals.

Parameters:

`Master Bypass` {On, Off}

See also:

MIO Channel Summer, MIO Channel Difference, MIO Channel Multiplier

MIO Channel Multiplier

- Synopsis: Channel Multiplier
- Category: Building Blocks
- Channels: 2

Description:

Multiplier takes its inputs and forms the product (A × B) of the two signals.

Parameters:

`Master Bypass` {On, Off}

See also:

MIO Channel Summer, MIO Channel Difference, MIO Channel Sum/Difference

MIODelay

- Synopsis: Adjustable Sample Delay - 255 samples max
- Category: Delay, Building Blocks
- Channels: 1, 2

Description:

Provides an integer sample delay from the input to the output. Changes in the delay time are not interpolated, which means that changing the delay time will result in a glitch. As a result, this delay is most suited for use as a fixed delay (as part of an algorithm or for a fixed compensation).

This version of the delay provides up to 255 samples of delay (5.3 ms memory in the DSP to provide the highest possible DSP performance.

Parameters:

`Delay(samp)` [0, 255] samps

Specifies the delay throught the process block in samples.

`Master Bypass` {On, Off}

When engaged sets the delay through the block to 0.

See also:

MIOModDelay, MIO Delay (1k), MIO Delay (24k), MIO Delay (96k), MIO Delay (1k IM), MIO MultiTap Delay

MIOModDelay

- Synopsis: Control Signal Modulated Interpolated Delay
- Category: Delay, Building Blocks
- Channels: 1, 2

Description:

MIOModDelay is a dynamic, interpolating delay that uses a audio control signal to set the delay time through the process. The input control signal is clipped to the range of 0.0…1.0, and then is multipled with the maximum delay specified by `Delay(samp)`. The resulting fractional delay is applied to the input signal to form the output signal.

Since the delay is variable on a sample by sample basis, and it supports fractional delays, the MIOModDelay can be used for automatic modulation effects like vibrato and chorus.

Parameters:

`Delay(samp)` [0, 1.023k] samps

Sets the maximum delay, in samples, applied to the input signal.

`Master Bypass` {On, Off}

When bypasssed, the delay through the process is 0.

See also:

MIODelay, MIO Delay (1k), MIO Delay (24k), MIO Delay (96k), MIO Delay (1k IM), MIO MultiTap Delay

MIO Delay (1k)

- Synopsis: Short (1024 sample) Delay
- Category: Delay, Building Blocks
- Channels: 1, 2

Description:

Provides an integer sample delay from the input to the output. Changes in the delay time are interpolated, allowing you to change the delay time with no glitches. For large changes in the delay time, the interpolation has the effect of shifting the pitch of the delayed signal as the delay time is changing.

This version of the delay utilizes the external memory block for the delay buffer. If you get a message that MIOConsole3d can't instantiate the plug-in, try using the IM version.

Parameters:

`Delay(samp)` [0, 1.023k] samps

Specifies the delay throught the process block in samples.

`Delay(ms)` [0, 23.1973] ms

Specifies the delay throught the process block in milliseconds.

`Master Bypass` {On, Off}

When engaged sets the delay through the block to 0.

MIO Delay (24k)

- Synopsis: Medium (500ms) Delay
- Category: Delay, Building Blocks
- Channels: 1, 2

Description:

Provides an integer sample delay from the input to the output. Changes in the delay time are interpolated, allowing you to change the delay time with no glitches. For large changes in the delay time, the interpolation has the effect of shifting the pitch of the delayed signal as the delay time is changing.

This version of the delay utilizes the external memory block for the delay buffer. If you get a message that MIOConsole3d can't instantiate the plug-in, you will need to use fewer memory-intensive plug-ins such as delays and reverbs.

Parameters:

`Delay(samp)` [0, 23.999k] samps

Specifies the delay throught the process block in samples.

`Delay(ms)` [0, 544.195] ms

Specifies the delay throught the process block in milliseconds.

`Master Bypass` {On, Off}

When engaged sets the delay through the block to 0.

MIO Delay (96k)

- Synopsis: Long (2 sec) Delay
- Category: Delay, Building Blocks
- Channels: 1, 2

Description:

Provides an integer sample delay from the input to the output. Changes in the delay time are interpolated, allowing you to change the delay time with no glitches. For large changes in the delay time, the interpolation has the effect of shifting the pitch of the delayed signal as the delay time is changing.

This version of the delay utilizes the external memory block for the delay buffer. If you get a message that MIOConsole3d can't instantiate the plug-in, you will need to use fewer memory-intensive plug-ins such as delays and reverbs

Parameters:

`Delay(samp)` [0, 95.999k] samps

Specifies the delay throught the process block in samples.

`Delay(ms)` [0, 2.17685k] ms

Specifies the delay throught the process block in milliseconds.

`Master Bypass` {On, Off}

When engaged sets the delay through the block to 0.

MIO Delay (1k IM)

- Synopsis: Short (1024 sample/Internal Memory) Delay
- Category: Delay, Building Blocks
- Channels: 1, 2

Description:

Provides an integer sample delay from the input to the output. Changes in the delay time are interpolated, allowing you to change the delay time with no glitches. For large changes in the delay time, the interpolation has the effect of shifting the pitch of the delayed signal as the delay time is changing.

This version of the delay utilizes the Internal Memory block for the delay buffer. If you get a message that MIOConsole3d can't instantiate the plug-in, try using the non-IM version.

Parameters:

`Delay(samp)` [0, 1.023k] samps

Specifies the delay throught the process block in samples.

`Delay(ms)` [0, 23.1973] ms

Specifies the delay throught the process block in milliseconds.

`Master Bypass` {On, Off}

When engaged sets the delay through the block to 0.

MIO Delay (2k-15k IM)

- Synopsis: Internal Memory Delays
- Category: Delay, Building Blocks
- Channels: 1

Description:

Provides an integer sample delay from the input to the output. Changes in the delay time are interpolated, allowing you to change the delay time with no glitches. For large changes in the delay time, the interpolation has the effect of shifting the pitch of the delayed signal as the delay time is changing.

This version of the delay utilizes the Internal Memory block for the delay buffer. If you get a message that MIOConsole3d can't instantiate the plug-in, try using the non-IM version.

Parameters:

`Delay(samp)` [0, See table] samps

Specifies the delay throught the process block in samples.

`Delay(ms)` [0, See table] ms

Specifies the delay throught the process block in milliseconds.

`Master Bypass` {On, Off}

When engaged sets the delay through the block to 0.

Plug-in version	**Delay (samples)**	**Delay (ms)**
2k	2048	46.4172
3k	3072	69.6372
4k	4096	92.8571
5k	5120	116.077
6k	6144	46.4172
7k	7165	162.517
8k	8192	185.737
9k	9216	208.957
10k	10240	232.177
15k	15360	348.277

Table B.1. Internal memory delay times

MIO Delay (2k-15k)

- Synopsis: External Memory Delay
- Category: Delay, Building Blocks
- Channels: 1

Description:

Provides an integer sample delay from the input to the output. Changes in the delay time are interpolated, allowing you to change the delay time with no glitches. For large changes in the delay time, the interpolation has the effect of shifting the pitch of the delayed signal as the delay time is changing.

This version of the delay utilizes the external memory block for the delay buffer. If you get a message that MIOConsole3d can't instantiate the plug-in, try using the IM version.

Parameters:

`Delay(samp)` [0, See table] samps

Specifies the delay throught the process block in samples.

`Delay(ms)` [0, See table] ms

Specifies the delay throught the process block in milliseconds.

`Master Bypass` {On, Off}

When engaged sets the delay through the block to 0.

Plug-in version	**Delay (samples)**	**Delay (ms)**
2k	2048	46.4172
3k	3072	69.6372
4k	4096	92.8571
5k	5120	116.077
6k	6144	46.4172
7k	7165	162.517
8k	8192	185.737
9k	9216	208.957
10k	10240	232.177
15k	15360	348.277

Table B.2. External memory delay times

MIO MultiTap Delay

- Synopsis: MultiTap Delay
- Category: Delay, Building Blocks
- Channels: 1

Description:

MIO MultiTap Delay implements a multi-tapped delay line. The total delay line length in this implementation varies; see the table below. Each delay tap is independent from the other taps, and provides controls to set the tap delay, gain level and polartity invert. Each tap can be enabled and disabled independently as well.

The delayed signal for each tap is available on dedicated outputs for further processing. The contributions of each tap are also summed together and the summed signal is available on the last output of the processing block.

The MIO MultiTap Delay is appropriate for creating the initial delays for a reverb algorithm. It can also be used to create multiple delayed copies of a signal. This second application can be used in a musical context or could be used to create multiple time-aligned feeds for a distributed sound reproduction system (e.g. a PA with multiple fill zones).

Parameters:

`Enable` {Off, On}

Enable the tap. When enabled, the tap generates output from on its dedicated output port and contributes to the summed output.

`Delay(samp)` [0, 1.023k] samps

Sets the delay, in samples, between the input and output of this tap.

`Gain` [-160, 10] dB

Sets the gain of the delayed signal.

`Invert` {Off, On}

When enabled, inverts the delayed signal relative to the input polarity.

`Enable` {Off, On}

Enable the tap. When enabled, the tap generates output from on its dedicated output port and contributes to the summed output.

`Delay(samp)` [0, 1.023k] samps

Sets the delay, in samples, between the input and output of this tap.

`Gain` [-160, 10] dB

Sets the gain of the delayed signal.

`Invert` {Off, On}

When enabled, inverts the delayed signal relative to the input polarity.

`Enable` {Off, On}

Enable the tap. When enabled, the tap generates output from on its dedicated output port and contributes to the summed output.

`Delay(samp)` [0, 1.023k] samps

Sets the delay, in samples, between the input and output of this tap.

`Gain` [-160, 10] dB

Sets the gain of the delayed signal.

`Invert` {Off, On}

When enabled, inverts the delayed signal relative to the input polarity.

`Enable` {Off, On}

Enable the tap. When enabled, the tap generates output from on its dedicated output port and contributes to the summed output.

`Delay(samp)` [0, 1.023k] samps

Sets the delay, in samples, between the input and output of this tap.

`Gain` [-160, 10] dB

Sets the gain of the delayed signal.

`Invert` {Off, On}

When enabled, inverts the delayed signal relative to the input polarity.

`Enable` {Off, On}

Enable the tap. When enabled, the tap generates output from on its dedicated output port and contributes to the summed output.

`Delay(samp)` [0, 1.023k] samps

Sets the delay, in samples, between the input and output of this tap.

`Gain` [-160, 10] dB

Sets the gain of the delayed signal.

`Invert` {Off, On}

When enabled, inverts the delayed signal relative to the input polarity.

`Enable` {Off, On}

Enable the tap. When enabled, the tap generates output from on its dedicated output port and contributes to the summed output.

`Delay(samp)` [0, 1.023k] samps

Sets the delay, in samples, between the input and output of this tap.

`Gain` [-160, 10] dB

Sets the gain of the delayed signal.

`Invert` {Off, On}

When enabled, inverts the delayed signal relative to the input polarity.

`Enable` {Off, On}

Enable the tap. When enabled, the tap generates output from on its dedicated output port and contributes to the summed output.

`Delay(samp)` [0, 1.023k] samps

Sets the delay, in samples, between the input and output of this tap.

`Gain` [-160, 10] dB

Sets the gain of the delayed signal.

`Invert` {Off, On}

When enabled, inverts the delayed signal relative to the input polarity.

`Enable` {Off, On}

Enable the tap. When enabled, the tap generates output from on its dedicated output port and contributes to the summed output.

`Delay(samp)` [0, 1.023k] samps

Sets the delay, in samples, between the input and output of this tap.

`Gain` [-160, 10] dB

Sets the gain of the delayed signal.

`Invert` {Off, On}

When enabled, inverts the delayed signal relative to the input polarity.

`Master Bypass` {On, Off}

Plug-in version	**Delay (samples)**	**Delay (ms)**
Short	1024	21
Medium	24000	500
Long	96000	2000

Table B.3. Multitap delay times

MIOAllpass

- Synopsis: First Order Allpass Filter
- Category: Phase EQ, Building Blocks
- Channels: 1

Description:

MIOAllpass provides an adjustable high-order all-pass filter. When the `Coeff` is 0, the filter acts as a pure single sample delay. As the `Coeff` is adjusted away from 0, the phase shift becomes more and more non-linear as a function of phase. The MIOAllpass can be used to implement a phase equalizer and is also useful as a reverb building block.

Parameters:

`Coeff` [-1, 1] s

Sets the coefficient of the of the all-pass filter.

See also:

MIOAllpassVD

MIOAllpassVD

- Synopsis: Variable Delay First Order Allpass Filter
- Category: Phase EQ, Building Blocks
- Channels: 1

Description:

MIOAllpassVD provides an adjustable high-order all-pass filter. When the `Coeff` is 0, the filter acts as a pure delay, with the delay time set by `Allpass(samp)`. As the `Coeff` is adjusted away from 0, the phase shift becomes more and more non-linear as a function of phase. The MIOAllpassVD can be used to implement a phase equalizer and is also useful as a reverb building block.

Parameters:

`Coeff` [-1, 1] s

Sets the coefficient of the of the all-pass filter.

`Allpass(samp)` [0, 255] samps

Sets the delay time of the pure-delay portion of the allpass filter. This parameter is in samples.

See also:

MIOAllpass

MIOHardClip

- Synopsis: Adjustable Threshold Hard Clipper
- Category: Distortion, Building Blocks
- Channels: 1

Description:

Applies a hard clipping threshold to the input signal. The `Threshold` specifies the absolute level at which the signal will be clipped. If the absolute value of the signal (on a sample by sample basis) is greater than the level specified by `Threshold` the output will be set to ± `Threshold` (with the sign determined by the sign of the input sample).

This is equivalent to the hard clipping that occurs when a fixed point signal hits full-scale, with the addition that you can adjust the level of the clipping point. This is not musically useful by itself, but it is very useful in combination with other basic processing blocks.

Parameters:

`Threshold` [0, 4] samps

Sets the linear sample value at which the input will be clipped.

`Master Bypass` {On, Off}

Disables the clipper and passes the signal through untouched.

See also:

MIOSoftClip Type 1, MIOSoftClip Type 2, MIOSoftClip Type 3

MIOSoftClip Type 1

- Synopsis: Type 1 Soft Saturation Clipper
- Category: Distortion, Building Blocks
- Channels: 1

Description:

When the the absolute value of the input signal level reaches the specified `Threshold` (on a sample by sample basis), Metric Halo type 1 saturation is applied to the signal. Type 1 saturation is very gentle, low order distortion applied to the signal symmetrically.

The overall gain through the process block is automatically adjusted so that a full-scale input signal creates a full-scale output. When the input signal is below the specified `Threshold`, the output signal is just the input signal with the over-all gain applied, and no additional non-linearity.

MIOSoftClip(1) is useful for applying soft saturation in signal processing loops (for example, inside a feedback delay line). It is also useful for saturating the output of other processors like the MIO Compressor and MIO Limiter.

Parameters:

`Threshold` [0, 4] samps

Threshold at which the saturation begins to be applied.

`Master Bypass` {On, Off}

Disables the saturation and passes the signal through untouched.

See also:

MIOHardClip, MIOSoftClip Type 2, MIOSoftClip Type 3

MIOSoftClip Type 2

- Synopsis: Type 2 Soft Saturation Clipper
- Category: Distortion, Building Blocks
- Channels: 1

Description:

When the the absolute value of the input signal level reaches the specified `Threshold` (on a sample by sample basis), Metric Halo Type 2 saturation is applied to the signal. Type 2 saturation is more aggressive than Type 1 saturation, but still provides relatively gentle, low order distortion that is applied to the signal symmetrically.

The overall gain through the process block is automatically adjusted so that a full-scale input signal creates a full-scale output. When the input signal is below the specified `Threshold`, the output signal is just the input signal with the over-all gain applied, and no additional non-linearity.

MIOSoftClip(2) is useful for applying soft saturation in signal processing loops (for example, inside a feedback delay line). It is also useful for saturating the output of other processors like the MIO Compressor and MIO Limiter.

Parameters:

`Threshold` [0, 4] samps

Threshold at which the saturation begins to be applied.

`Master Bypass` {On, Off}

Disables the saturation and passes the signal through untouched.

See also:

MIOHardClip, MIOSoftClip Type 1, MIOSoftClip Type 3

MIOSoftClip Type 3

- Synopsis: Type 3 Soft Saturation Clipper
- Category: Distortion, Building Blocks
- Channels: 1

Description:

When the the absolute value of the input signal level reaches the specified `Threshold` (on a sample by sample basis), Metric Halo Type 3 saturation is applied to the signal. Type 3 saturation is more aggressive than Type 2 saturation, and when driven sufficently hard, can cause audible distortion, but is very useful in punching up percussive signals, and in softening rarely clipping signals.

The overall gain through the process block is automatically adjusted so that a full-scale input signal creates a full-scale output. When the input signal is below the specified `Threshold`, the output signal is just the input signal with the over-all gain applied, and no additional non-linearity.

MIOSoftClip(3) is useful for applying soft saturation in signal processing loops (for example, inside a feedback delay line). It is also useful for saturating the output of other processors like the MIO Compressor and MIO Limiter.

Parameters:

`Threshold` [0, 4] samps

Threshold at which the saturation begins to be applied.

`Master Bypass` {On, Off}

Disables the saturation and passes the signal through untouched.

See also:

MIOHardClip, MIOSoftClip Type 1, MIOSoftClip Type 2

MIOSoftDistortion Type 1

- Synopsis: Type 1 Soft Distortion Generator
- Category: Distortion, Building Blocks

- Channels: 1

Description:

When the the absolute value of the input signal level reaches the specified `Threshold` (on a sample by sample basis), Metric Halo Type 1 distortion is generated and sent to the output. This signal may be added to the input to generate the same effect as the Type 1 soft clipper. The point of this signal processing module is to allow you to further process the distortion signal before adding it back into the original signal.

Parameters:

`Threshold` [0, 4] samps

Threshold at which the saturation begins to be applied.

`Master Bypass` {On, Off}

Disables the distortion generator and causes the output to be zero.

See also:

MIOSoftDistortion Type 1, MIOSoftDistortion Type 2

MIOSoftDistortion Type 2

- Synopsis: Type 2 Soft Distortion Generator
- Category: Distortion, Building Blocks
- Channels: 1

Description:

When the the absolute value of the input signal level reaches the specified `Threshold` (on a sample by sample basis), Metric Halo type 2 distortion is generated and sent to the output. This signal may be added to the input to generate the same effect as the Type 2 soft clipper. The point of this signal processing module is to allow you to further process the distortion signal before adding it back into the original signal.

Parameters:

`Threshold` [0, 4] samps

Threshold at which the saturation begins to be applied.

`Master Bypass` {On, Off}

Disables the distortion generator and causes the output to be zero.

See also:

MIOSoftDistortion Type 1, MIOSoftDistortion Type 2

MIOSoftDistortion Type 3

- Synopsis: Type 3 Soft Distortion Generator
- Category: Distortion, Building Blocks

- Channels: 1

Description:

When the the absolute value of the input signal level reaches the specified `Threshold` (on a sample by sample basis), Metric Halo Type 3 distortion is generated and sent to the output. This signal may be added to the input to generate the same effect as the Type 3 soft clipper. The point of this signal processing module is to allow you to further process the distortion signal before adding it back into the original signal.

Parameters:

`Threshold` [0, 4] samps

Threshold at which the saturation begins to be applied.

`Master Bypass` {On, Off}

Disables the distortion generator and causes the output to be zero.

See also:

MIOSoftDistortion Type 1, MIOSoftDistortion Type 2

MIOSlew

- Synopsis: Output Slew Rate Limiter
- Category: Distortion, Building Blocks
- Channels: 1

Description:

MIOSlew limits the slew rate of the output to the maximum slew rate specified by the `Slew Rate` parameter. This block simulates the effect of slew rate limiting on the output of a signal driver. The slew rate limitation has the effect of a slow-rolloff nonlinear filter, and is an interesting distortion effect.

Parameters:

`Slew Rate` [0, 100] samps

Sets the maximum change in the output on a sample-by-sample basis.

`Master Bypass` {On, Off}

Causes the maximum slew rate to be ∞, and provides a hard-bypass to the signal processor.

See also:

MIOInSlew

MIOInSlew

- Synopsis: Input Slew Rate Limiter
- Category: Distortion, Building Blocks
- Channels: 1

Description:

MIOInSlew limits the slew rate of the input to the maximum slew rate specified by the `Slew Rate` parameter. This block simulates the effect of slew rate limiting on the output of a signal driver. The slew rate limitation has the effect of a slow-rolloff nonlinear filter, and is an interesting distortion effect.

Parameters:

`Slew Rate` [0, 100] samps

Sets the maximum change in the input on a sample-by-sample basis.

`Master Bypass` {On, Off}

Causes the maximum slew rate to be ∞, and provides a hard-bypass to the signal processor.

See also:

MIOSlew

MIO 4th Order Nonlinear Map

- Synopsis: 4th Order Nonlinear Map
- Category: Distortion, Building Blocks
- Channels: 1

Description:

MIO NL(4) Map applies the nonlinear map specified by the parameters to the input signal to form the output signal. The quadratic term creates second-harmonic distortion. The cubic term creates third-harmonic distortion. The quartic term creates both second-harmonic and fourth-harmonic distortion.

Theres non-linearities can be applied to generate a wide variety of wave-shapers and distortion effects.

Parameters:

`x^1` [-4, 4]

Sets the coefficient of the linear term of the map.

`x^2` [-4, 4]

Sets the coefficient of the quadratic term of the map.

`x^3` [-4, 4]

Sets the coefficient of the cubic term of the map.

`x^4` [-4, 4]

Sets the coefficient of the quartic term of the map.

`Master Gain` [-4, 4]

Sets the overall gain of the output post mapping.

`Master Bypass` {On, Off}

When enabled, bypasses the process so that the output is the same as the input.

MIO 4th Order Symmetrical Nonlinear Map

- Synopsis: 4th Order Symmetrical Nonlinear Map
- Category: Distortion, Building Blocks
- Channels: 1

Description:

MIO SNL(4) Map applies a symmetrized version of the nonlinear map specified by the parameters. This means that the applied non-linearity has the same relative polarity to the input signal polarity. You can use this to apply a non-linearity that always reduces the amplitude of the input signal, whether the input is positive or negative.

The non-linearity generated by this symmetrized map generates high-order odd harmonics for the even order terms, and low-order odd harmonics for the odd order terms.

Parameters:

`x^1` [-4, 4]

Sets the coefficient of the linear term of the map.

`x^2` [-4, 4]

Sets the coefficient of the quadratic term of the map.

`x^3` [-4, 4]

Sets the coefficient of the cubic term of the map.

`x^4` [-4, 4]

Sets the coefficient of the quartic term of the map.

`Master Gain` [-4, 4]

Sets the overall gain of the output post mapping.

`Master Bypass` {On, Off}

When enabled, bypasses the process so that the output is the same as the input.

MIO 4th Order [dB] Nonlinear Map

- Synopsis: 4th Order [dB] Nonlinear Map
- Category: Distortion, Building Blocks
- Channels: 1

Description:

MIO NL(4) [dB] Map applies the nonlinear map specified by the parameters to the input signal to form the output signal. The quadratic term creates second-harmonic distortion. The cubic term creates third-harmonic distortion. The quartic term creates both second-harmonic and fourth-harmonic distortion.

These non-linearities can be applied to generate a wide variety of wave-shapers and distortion effects.

Parameters:

`x^1` [-300, 12] dB

Sets the coefficient (in dB) of the linear term of the map.

`x^2` [-300, 12] dB

Sets the coefficient (in dB) of the quadratic term of the map.

`x^3` [-300, 12] dB

Sets the coefficient (in dB) of the cubic term of the map.

`x^4` [-300, 12] dB

Sets the coefficient (in dB) of the quartic term of the map.

`Inv` {Off, On}

Sets the sign of the linear coefficient (the sign is negative when enabled).

`Inv` {Off, On}

Sets the sign of the quadratic coefficient (the sign is negative when enabled).

`Inv` {Off, On}

Sets the sign of the cubic coefficient (the sign is negative when enabled).

`Inv` {Off, On}

Sets the sign of the quartic coefficient (the sign is negative when enabled).

`Master Gain` [-300, 12] dB

Sets the overall gain of the output post mapping.

`Master Bypass` {On, Off}

When enabled, bypasses the process so that the output is the same as the input.

MIO 4th Order [dB] Symmetrical Nonlinear Map

- Synopsis: 4th Order [dB] Symmetrical Nonlinear Map
- Category: Distortion, Building Blocks
- Channels: 1

Description:

MIO SNL(4) [dB] Map applies a symmetrized version of the nonlinear map specified by the parameters. This means that the applied non-linearity has the same relative polarity to the input signal polarity. You can use this to apply a non-linearity that always reduces the amplitude of the input signal, whether the input is positive or negative.

The non-linearity generated by this symmetrized map generates high-order odd harmonics for the even order terms, and low-order odd harmonics for the odd order terms.

Parameters:

`x^1` [-300, 12] dB

Sets the coefficient (in dB) of the linear term of the map.

`x^2` [-300, 12] dB

Sets the coefficient (in dB) of the quadratic term of the map.

`x^3` [-300, 12] dB

Sets the coefficient (in dB) of the cubic term of the map.

`x^4` [-300, 12] dB

Sets the coefficient (in dB) of the quartic term of the map.

`Inv` {Off, On}

Sets the sign of the linear coefficient (the sign is negative when enabled).

`Inv` {Off, On}

Sets the sign of the quadratic coefficient (the sign is negative when enabled).

`Inv` {Off, On}

Sets the sign of the cubic coefficient (the sign is negative when enabled).

`Inv` {Off, On}

Sets the sign of the quartic coefficient (the sign is negative when enabled).

`Master Gain` [-300, 12] dB

Sets the overall gain of the output post mapping.

`Master Bypass` {On, Off}

When enabled, bypasses the process so that the output is the same as the input.

MIO A/B Switch (Linear)

- Synopsis: A/B Switch - Linear Interpolation
- Category: Signal Switch, Building Blocks
- Channels: 2->1, 4->2, 6->3, 8->4, 10->5, 12->6, 14->7, 16->8, 18->9, 20->10, 22->11, 24->12, 26->13, 28->14, 30->15, 32->16

Description:

The MIO A/B (Linear) is a multichannel process block that allows you to crossfade between multichannel input ports. The MIO A/B (Linear) implements a mono channel input port selector switch. Since the switch implements cross-fading between its inputs, it can also be used as a constant amplitude mixer. The `Glide` parameter controls how quickly the gain coefficient changes. This implementation of the switch interpolates the gain linearly. This means that the gain changes at a constant rate per sample as the internal gain approaches the selected gain.

Parameters:

`A Blend` [0, 1]

Sets the portion of the output that comes from the first input port when the `A/B` switch is turned on. The portion of the output that comes from the second input port is the complement of this value.

`B Blend` [0, 1]

Sets the portion of the output that comes from the first input port when the `A/B` switch is turned off. The portion of the output that comes from the second input port is the complement of this value.

`Glide` [0, 1]

Sets the rate that the process uses to change the internal values of its cross-fade parameter. Large values lead to faster changes. A value of 1 will cause the gain to change instantly, but could lead to clicks. Very small values will cause the gain change to take place very slowly, which can be used to provide a triggerable, automatic crossfade.

`A/B` {On, Off}

Chooses the Port 1 gain from the `A Blend` (on) or `B Blend` (off) parameters.

`Master Bypass` {On, Off}

When bypassed, MIO A/B (1/Linear) will pass the signal from the A port(s) untouched.

See also:

MIO A/B Switch (LPF)

MIO A/B Switch (LPF)

- Synopsis: A/B Switch - Low Pass Filtered Interpolation
- Category: Signal Switch, Building Blocks
- Channels: 2->1, 4->2, 6->3, 8->4, 10->5, 12->6, 14->7, 16->8, 18->9, 20->10, 22->11, 24->12, 26->13, 28->14, 30->15, 32->16

Description:

The MIO A/B (LPF) is a multichannel process block that allows you to crossfade between multichannel input ports. The MIO A/B (LPF) implements a mono channel input port selector switch. Since the switch implements cross-fading between its inputs, it can also be used as a constant amplitude mixer. The `Glide` parameter controls how quickly the gain coefficient changes. This implementation of the switch uses a single-pole Low Pass Filter to interpolate the gain. This means that the gain will change more quickly when it initially changes and the change per sample will decrease as the internal gain approaches the selected gain.

Parameters:

`A Blend` [0, 1]

Sets the portion of the output that comes from the first input port when the `A/B` switch is turned on. The portion of the output that comes from the second input port is the complement of this value.

`B Blend` [0, 1]

Sets the portion of the output that comes from the first input port when the `A/B` switch is turned off. The portion of the output that comes from the second input port is the complement of this value.

`Glide` [0, 1]

Sets the rate that the process uses to change the internal values of its cross-fade parameter. Large values lead to faster changes. A value of 1 will cause the gain to change instantly, but could lead to clicks. Very small values will cause the gain change to take place very slowly, which can be used to provide a triggerable, automatic crossfade.

`A/B` {On, Off}

Chooses the Port 1 gain from the `A Blend` (on) or `B Blend` (off) parameters.

`Master Bypass` {On, Off}

When bypassed, MIO A/B (1/LPF) will pass the signal from the A port(s) untouched.

See also:

MIO A/B Switch (Linear)

MIO Blend (Linear)

- Synopsis: Mixer - Linear Interpolation
- Category: Mixer, Building Blocks, Internal
- Channels: 2->1, 4->2, 6->3, 8->4, 10->5, 12->6, 14->7, 16->8, 18->9, 20->10, 22->11, 24->12, 26->13, 28->14, 30->15, 32->16

Description:

The MIO Blend (Linear) provides two multichannel input ports and sums the two ports together with an adjustable gain for each port. These are used internally by the mixer to manage the trim gains and crossfades for the inputs and tape returns on input strips. The MIO Blend (Linear) implements a mono channel input port summer. Since the summer supports cross-fading between its inputs, it can also be used as a constant amplitude mixer. The `Glide` parameter controls how quickly the gain coefficient changes. This implementation of the switch interpolates the gain linearly. This means that the gain changes at a constant rate per sample as the internal gain approaches the selected gain.

Parameters:

`A Gain` [-160, +12]

Sets the gain of the output that comes from the first input port.

`B Gain` [-160, +12]

Sets the gain of the output that comes from the second input port.

`Glide` [0, 1]

Sets the rate that the process uses to change the internal values between the A and B Gain parameters. Large values lead to faster changes. A value of 1 will cause the gain to change instantly, but could lead to clicks. Very small values will cause the gain change to take place very slowly, which can be used to provide a triggerable, automatic crossfade.

`Master Bypass` {On, Off}

When bypassed, MIO Blend (Linear) will pass the signal from the A port(s) untouched.

See also:

MIO Blend (LPF)

MIO Blend (LPF)

- Synopsis: A/B Switch - Low Pass Filtered Interpolation
- Category: Signal Switch, Building Blocks
- Channels: 2->1, 4->2, 6->3, 8->4, 10->5, 12->6, 14->7, 16->8, 18->9, 20->10, 22->11, 24->12, 26->13, 28->14, 30->15, 32->16

Description:

The MIO Blend (LPF) provides two multichannel input ports and sums the two ports together with an adjustable gain for each port. These are used internally by the mixer to manage the trim gains and crossfades for the inputs and tape returns on input strips. The MIO Blend (LPF) implements a mono channel input port summer. Since the summer supports cross-fading between its inputs, it can also be used as a constant amplitude mixer. The `Glide` parameter controls how quickly the gain coefficient changes. This implementation of the switch uses a single-pole Low Pass Filter to interpolate the gain. This means that the gain will change more quickly when it initially changes and the change per sample will decrease as the internal gain approaches the selected gain.

Parameters:

`A Gain` [-160, +12]

Sets the gain of the output that comes from the first input port.

`B Gain` [-160, +12]

Sets the gain of the output that comes from the second input port.

`Glide` [0, 1]

Sets the rate that the process uses to change the internal values of its cross-fade parameter. Large values lead to faster changes. A value of 1 will cause the gain to change instantly, but could lead to clicks. Very small values will cause the gain change to take place very slowly, which can be used to provide a triggerable, automatic crossfade.

`Master Bypass` {On, Off}

When bypassed, MIO Blend (LPF) will pass the signal from the A port(s) untouched.

See also:

MIO Blend (Linear)

MIOQuadOsc

- Synopsis: Quadrature Sine Oscillator
- Category: Signal Generator, Building Blocks
- Channels: 0

Description:

MIOQuadOsc is a signal generator which functions as a Quadrature Oscillator whose oscillation frequency is precisely controlled by the `Frequency` parameter. The Quadrature oscillator generates a very pure, low-distortion sinusoidal waveform at one output and the quadrature (90 degree phase shifted) waveform at the other output.

Parameters:

`Frequency` [1m, 22k] Hz

Precisely sets the frequnecy of the oscillator.

See also:

MIOQuadLFO, MIOQuadNCO - Numerically Controlled Oscillator, MIOQuadNCO-Glide, Noise

MIOQuadLFO

- Synopsis: Quadrature Sine Low Frequency Oscillator
- Category: Signal Generator, Building Blocks
- Channels: 0

Description:

MIOQuadLFO is a signal generator which functions as a Low Frequency Quadrature Oscillator whose oscillation frequency is precisely controlled by the `Frequency` parameter. The Quadrature oscillator generates a very pure, low-distortion sinusoidal waveform at one output and the quadrature (90 degree phase shifted) waveform at the other output.

Parameters:

`Frequency` [1m, 20] Hz

Precisely sets the frequnecy of the oscillator.

See also:

MIOQuadOsc, MIOQuadNCO - Numerically Controlled Oscillator, MIOQuadNCO-Glide, Noise

MIOQuadNCO

- Synopsis: Numerically Controlled Oscillator
- Category: Signal Generator, Building Blocks
- Channels: 1

Description:

MIOQuadNCO provides a Quadrature Oscillator whose oscillation frequency is controlled by the audio signal at the control input. The oscillation frequency changes approximately linearly with the input signal.

The frequency of oscillation is 0 Hz when the control input is 1, and is Fs/2 when the control input is -1.

See also:

MIOQuadOsc, MIOQuadLFO, MIOQuadNCO-Glide, Noise

MIOQuadNCO-Glide

- Synopsis: Interpolated Numerically Controlled Quadrature Sine Oscillator
- Category: Signal Generator, Building Blocks
- Channels: 1

Description:

MIOQuadNCO-Glide provides a Quadrature Oscillator whose oscillation frequency is controlled by the audio signal at the control input. The oscillation frequency changes approximately linearly with the input signal.

The frequency of oscillation is 0 Hz when the control input is 1, and is Fs/2 when the control input is -1.

MIOQuadNCO-Glide filters the control input signal using the `Glide` parameter to control how quickly the oscillator frequency follows the control input.

Parameters:

`Glide` [0.1, 1k]

Sets the glide rate of the change of the oscillator frequency to changes in the control signal input. When the Glide is set to 0, the oscillator will not change in response to changes in the control signal. When the glide is set to 1, the oscillator will change instantaneously (like the normal QuadNCO). In between values cause the oscillator to interpolate changes in the control signal at a faster or slower rate.

See also:

MIOQuadOsc, MIOQuadLFO, MIOQuadNCO - Numerically Controlled Oscillator, Noise

Noise

- Synopsis: Full Scale White Noise Generator
- Category: Signal Generator, Building Blocks
- Channels: 1

Description:

Noise is a signal generator that creates uniform randomly distributed full-scale white noise.

See also:

MIOQuadOsc, MIOQuadLFO, MIOQuadNCO - Numerically Controlled Oscillator, MIOQuad-NCO-Glide

Simple Dither (TPDF)

- Synopsis: Triangular Flat Dither
- Category: Dither, Building Blocks
- Channels: 1

Description:

Simple Dither (TPDF) applies white power spectrum dither to the input signal at the bit-depth specified by the `Bits` parameter, and truncates the dithered signal to the specified bit depth. The applied dither is created with 2 LSB pk-pk triangular probability distribution function white noise.

Parameters:

`Bits` [1, 24] bits

Used to specify the final the bit depth of the dithered and truncated signal.

`Master Bypass` {On, Off}

See also:

Simple Dither (TPDF Hipass)

Simple Dither (TPDF Hipass)

- Synopsis: Triangular Hipass Dither
- Category: Dither, Building Blocks
- Channels: 1

Description:

Simple Dither (TPDF Hipass) applies high-pass filtered dither to the input signal at the bit-depth specified by the `Bits` parameter, and truncates the dithered signal to the specified bit depth. The applied dither is created with 2 LSB pk-pk triangular probability distribution function white noise filtered with a high-pass characteristic.

Parameters:

`Bits` [1, 24] bits

Used to specify the final the bit depth of the dithered and truncated signal.

`Master Bypass` {On, Off}

See also:

Simple Dither (TPDF)

Scale/Offset

- Synopsis: Scale and Offset Signal

- Category: Building Blocks
- Channels: 1

Description:

Scale/Offset is used to apply an affine transformation to the input signal on a sample by sample basis. This can be used to suitably scale and offset control signals used as audio-rate controls for blocks that accept such signals, or it can be used to set scale and bias on signals before they are driven into non-linear elements. The added DC bias can be used to condition the waveshaping applied by the nonlinear signal processing blocks. The remaining DC offset can be removed by another Scale/Offset instance, or by a DC-removal filter.

Parameters:

`Scale` [-5, 5]

Gain applied to input signal.

`Offset` [-1, 1]

DC-Offset added to the scaled input signal.

`Master Bypass` {On, Off}

When engaged, the process block is hard-bypassed, and the output signal is the same as the input signal.

SVF Control

- Synopsis: Interpolated Multimode Filter
- Category: EQ, Synth Effect, Building Blocks
- Channels: 1

Description:

SVF Control is a monophonic process block that simultaneously applies multiple filters (high pass, low pass, band pass and notch) to the input signal. Each filter is available on a different output of the process block.

The filter parameters are fully interpolated, so you can modify them without causing any glitches in the audio being processed. In addition, the SVF Control provides a `Glide` control parameter that allows you to control the interpolation rate. This allows you to configure the filter so that it reacts instantly to parameter changes, or has a slower gliding behavior.

The `F` parameter allows you to set the center/cutoff frequency of the various filters. The `Q` parameter allows you to control the resonance at the cutoff frequency. Larger Q's will increase the peaking at the cutoff frequency.

Parameters:

`F` [0, 5.5k] Hz

Sets the center frequency/corner frequency of the filter.

`Q` [0.5, 1k]

Sets the quality factor of the filter, and controls the resonance at the cutoff frequency.

`Glide` [1m, 1]

Sets the glide rate of changes to the other parameters. When the Glide is set to 0, the filter will not change in response to parameter changes. When the glide is set to 1, the filter will change instantaneously. In between values cause the filter to interpolate changes in the parameters at a faster or slower rate.

`Gain` [-100, 27] dB

Sets the output gain of the filter.

NC SVF

- Synopsis: Numerically Controlled State Variable Filter
- Category: EQ, Synth Effect, Building Blocks
- Channels: 1

Description:

NC SVF is a monophonic process block that simultaneously applies multiple filters (high pass, low pass, band pass and notch) to the input signal. Each filter is available on a different output of the process block.

The filter parameters are driven by the 2 control inputs (Cutoff and Resonance) and are fully interpolated, so you can modify them at audio rates without causing any glitches in the audio being processed.

The Cutoff input allows you to set the center/cutoff frequency of the various filters. The Resonance input allows you to control the resonance at the cutoff frequency. The Resonance control signal is the inverse of the Q factor of the filter (1/Q). Larger Q's will increase the peaking at the cutoff frequency.

MIOSimplePitchShifter

- Synopsis: Simple Pitch Shifter
- Category: Synth Effect, Building Blocks
- Channels: 1

Description:

The MIOSimplePitchShifter is a monophonic process block that shifts the pitch of the input signal by the specified number of half-steps (12 half-steps per octave).

The MIOSimplePitchShifter provides one parameter (`Steps`), which is the amount of pitch shift to apply to the signal. The shift parameter has a range of -24 half-steps (-2 Octaves) to +24 half-steps (+2 Octaves), and may take on any fractional value in the range.

There is a small amount of effective delay on the pitch shifted signal, but it is generally not noticable when the shifted signal is summed back with the original signal. Depending upon the frequency of the input signal there may be a small amount of AM modulation in the output signal. This will appear as apparent tremolo on the output.

Parameters:

`Steps` [-24, 24]

Sets the amount of pitch-shift applied to the input in half-steps. This parameter supports micro-tonal shifts.

EnvelopeDetector

- Synopsis: Envelope control signal extractor
- Category: Building Blocks
- Channels: 1

Description:

EnvelopeDetector is a mono process that generates the signal amplitude envelope from its input. The output of EnvelopeDetector may be used to apply the envelope of one signal to another or may be used as the control signal for any plug-in that supports control input. For example, it can be used to control the corner frequency of an NC SVF filter or the oscillation frequency of a MIOQuadNCO - Numerically Controlled Oscillator. For these sorts of control inputs, you will generally need to map the range of the output to the range expected by the control input. There are a variety of plug-ins in the 3d DSP that can be used for this purpose, including Map Range and CV -> NCO Freq (m/m).

EnvelopeDetector provides two parameters for controlling the envelope extraction process. `Attack` controls how quickly the envelope follows the attack of the input signal and `Release` controls how quickly the envelope follows the signal as the level drops below the current envelope level.

Parameters:

`Attack` [0, 100] ms

Controls how quickly the envelope follows the attack of the input signal.

`Release` [0, 1k] ms

Controls how quickly the envelope follows the signal as the level drops below the current envelope level.

`Master Bypass` {On, Off}

Has no effect on this plug-in.

CV -> NCO Freq

- Synopsis: Convert Linear Control Signal to Control Signal for NCO
- Category: Building Blocks
- Channels: 1

Description:

> CV -> NCO Freq converts a control signal in the range [0, 1] to a control signal for the MIOQuad-NCO - Numerically Controlled Oscillator. You can specify the frequency that will be synthesized when the input control signal is 0 using the `Lo Freq` parameter. Conversely, when the input control signal is 1, the synthesized frequency is determined by the `High Freq` parameter. The frequency generated is linearly interpolated between these the frequencies specified by the parameters for input signals between 0 and 1.If you set the `Lo Freq` to be higher than the `High Freq`, the frequency synthesized will decrease as the input control signal increases.

Parameters:

`Lo Freq` [1m, 22k] Hz

sets the frequency generated by a connected NCO when the input control signal is 0.

`High Freq` [1m, 22k] Hz

sets the frequency generated by a connected NCO when the input control signal is 1.

Abs

- Synopsis: Generates the absolute value of its input
- Category: Math
- Channels: 1

Description:

> The Abs process functions as a full-wave rectifier, generating the absolute value of its input. This means that the output of Abs is always positive. Postive input signal samples are passed through with no change; negative input signal samples are negated so thet positive equivalent sample value is passed to the output.

Max

- Synopsis: Selects the maximum value of its two inputs
- Category: Math
- Channels: 2

Description:

> Max passes the input that has the largest value on a sample by sample basis. The output of Max is the greater of its inputs.

Min

- Synopsis: Selects the minimum value of its two inputs

- Category: Math
- Channels: 2

Description:

Min passes the input that has the smallest value on a sample by sample basis. The output of Min is the smaller of its inputs.

Select

- Synopsis: Selects the value of its inputs based upon the value of the control input
- Category: Math
- Channels: 2

Description:

Select passes one of its inputs (x or y) based upon the value of the signal at its Cntl input. When Cntl is ≥ 0, the signal at input x is passed to the output. When Cntl < 0, the signal at input y is passed to the output. This determination is made on a sample by sample basis, so the Select plug-in can be used to modulate or select between two different signals based upon the value of a third signal, which could be a control signal or an audio signal.

Map Range

- Synopsis: Linearly maps input signal based on parameter selections
- Category: Math
- Channels: 1

Description:

Map Range linearly maps the signal at the input to its output based upon the control parameters specified. Map Range allows you to map an arbitrary input range to an arbitrary output range, optionally clipping the output signal to the specified output range. The output signal is determined by the equation out = in * (`Out Hi` - `Out Low`)÷(`In Hi` - `In Low`) + `Out Low`. If `Clip Out` is enabled, the output signal will be clipped to the range [`Out Low`, `Out Hi`].

Parameters:

`In Low` [-1, 1]

Low End of input range.

`In Hi` [-1, 1]

High End of input range.

`Out Low` [-1, 1]

Low End of output range.

`Out Hi` [-1, 1]

High End of output range.

`Clip Out` {Off, On}

If Enabled, the output signal is clipped to the range specifed by [`Out Low`, `Out Hi`].

Constant

- Synopsis: Output a constant DC signal
- Category: Math
- Channels: 0

Description:

Constant generates a constant DC signal with the value specified by `Constant`. This is useful for generating a constant signal to be used as a control signal to be used as an input for other processing blocks such as MIO Channel Multiplier, NC SVF, Max, Min and Select.

Parameters:

`Constant` [-100, 100]

The value of the output signal.

Divide

- Synopsis: Math operation to divide one signal by another
- Category: Math
- Channels: 2

Description:

Divide generates the signal (x÷y) from the input signals x and y. Mathematically, this would be ∞ when y=0, but in general that will cause numerical stability problems in the processing graph, so the output is limited to a very large, but not infinite value in the case that y=0. In general, this will be most useful when y is a control signal with a non-zero value.

Please note that Divide is a relatively expensive operation, and should be avoided if possible (for example, if you want to scale input signal X by a constant value, you are much better off multiplying by the reciprocal of the scale factor).

Square Root

- Synopsis: Computes the Square Root of the input signal
- Category: Math
- Channels: 1

Description:

Square Root generates an output signal that is the Square Root of the input signal on a sample by sample basis. Since the 3d DSP does not support processing imaginary numbers, the output of Square Root is flushed to zero if the input value is negative.

Reciprocal Square Root

- Synopsis: Computes the 1/(Square Root of the input signal)
- Category: Building Blocks
- Channels: 1

Description:

Reciprocal Square Root generates an output signal that is the inverse of the Square Root of the input signal on a sample by sample basis. Since the 3d DSP does not support processing imaginary numbers, the output of Reciprocal Square Root is flushed to a large, but not infinite number if the input value is zero or negative.

ADSR

- Synopsis: Generates an envelope and frequency from a MIDI note message
- Category: Math
- Channels: 0

Description:

ADSR generates a note envelope and NCO oscillator control signal from an input MIDI note message. The ADSR plugin listens for MIDI notes from every MIDI device in the system, and when it detects a note on message, it generates an envelope control signal based upon the `Attack`, `Decay`, `Sustain`, and `Release` parameters and the MIDI note on velocity. It also generates a NCO oscillator control signal based upon the note value from the incoming MIDI message.

The envelope generated ramps linearly between off, the volume set by the note-on velocity, the sustain level and off again; the linear ramp can be used for a variety of special effects, but the exponential ramp is much more natural sounding than a linear ramp for most applications.

To utilize this plugin, you would connect the Envelope output to one side of a Channel Multiplier block, and the Frequency output to the input of a NCO plugin, and then output of the NCO to the other side of the Channel Multiplier block. You can add additional processing to both the Envelope and Frequency outputs to make much more interesting synth tones.

ADSR is a monophonic envelope processor and a new note on message will steal the voice from a currently active note-on message.

Parameters:

`Attack` [0, 1k] ms

The amount of time for the volume envelope to reach the peak volume after the detection of the note on message.

`Decay` [0, 5k] ms

The amount of time for the volume envelope to decay to the sustain volume after the detection reaching the peak level.

`Sustain` [-60, 0] dB

The sustain level of the signal in dB below the peak volume.

`Release` [0, 5k] ms

The amount of time for the volume envelope to decay back to muted after the plugin detects a corresponding note-off message.

Exponential ADSR

- Synopsis: Generates an envelope and frequency from a MIDI note message
- Category: Math
- Channels: 0

Description:

Exponential ADSR generates a note envelope and NCO oscillator control signal from an input MIDI note message. The Exponential ADSR plugin listens for MIDI notes from every MIDI device in the system, and when it detects a note on message, it generates an envelope control signal based upon the `Attack`, `Decay`, `Sustain`, and `Release` parameters and the MIDI note on velocity. It also generates a NCO oscillator control signal based upon the note value from the incoming MIDI message.

The envelope generated ramps exponentially between off, the volume set by the note-on velocity, the sustain level and off again; the exponential ramp is much more natural sounding than a linear ramp, although the linear ramp can be used for a variety of special effects.

To utilize this plugin, you would connect the Envelope output to one side of a Channel Multiplier block, and the Frequency output to the input of a NCO plugin, and then output of the NCO to the other side of the Channel Multiplier block. You can add additional processing to both the Envelope and Frequency outputs to make much more interesting synth tones.

Exponential ADSR is a monophonic envelope processor and a new note on message will steal the voice from a currently active note-on message.

Parameters:

`Attack` [0, 1k] ms

The amount of time for the volume envelope to reach the peak volume after the detection of the note on message.

`Decay` [0, 5k] ms

The amount of time for the volume envelope to decay to the sustain volume after the detection reaching the peak level.

`Sustain` [-60, 0] dB

The sustain level of the signal in dB below the peak volume.

`Release` [0, 5k] ms

The amount of time for the volume envelope to decay back to muted after the plugin detects a corresponding note-off message.

Band Split (m)

- Synopsis: BandSplit (m)

- Category: Math
- Channels: 1 In, 2 Out

Description:

Band Split is a mono in, two out crossover module. The input is processed by a 24dB/octave Linkwitz-Riley filter, with frequencies below F output from the "Low" output and those above F output from the "High" output.

Parameters:

F [10, 20k]

Sets the crossover frequency (in Hz) of the bandsplit.

Band Split (s)

- Synopsis: BandSplit – Crossover
- Category: Math
- Channels: 2 In, 4 Out

Description:

Band Split (s) is a stereo in, four out stereo crossover module. The input is processed by a 24dB/octave Linkwitz-Riley filter, with frequencies below F output from the "Low" output and those above F output from the "High" output.

Parameters:

F [10, 20k]

Sets the crossover frequency (in Hz) of the bandsplit.

C. MIOConsole3d Ontology

What was that called again…?

Terms, Conventions and Translations

It seems every manufacturer of audio production tools has to create their own special terminology for all the hardware, software and UI elements that we humans interact with to perform our respective audio-centric tasks. Naturally, the logical way to approach these terms is to try to adopt as many as possible from earlier (generally more analog) forms of the devices and/or operations being emulated.

Unfortunately, as is the nature of progess and innovation, the tools that have become available to us for making music and manipulating audio have evolved such that terms from digital graphics editors and computer networking are actually more appropriate to describe many of the newer functions than anything having to do with actual audio.

In the case of computer-based digital audio workstations, the difference in the specialized workflows offered by the various DAWs (plus the marketing departments desire to differentiate ones' own product from the competition) has led to a proliferation of user interface terms specific to each platform.

This would ordinarily be fine, but over time many of these terms have come to either outright duplicate each other (in name but not in function), or are similar in nature but have significantly different behaviors (which is frankly even more confusing). Since many if not most of us use more than one DAW in our ever-more-complex production workflows, this disparity in terms is frankly becoming a real-world pain in the nethers.

SO… in keeping with Metric Halos renowned obsession with fixing problems that get in the way of making music, we present the MIOConsole3d Ontology: a glossary of terms and conventions applied in the 3d audio production environment, along with translations to the nearest functional terms of each of the most popular digital audio workstations.

Some items mentioned in the Introduction: Decoding 3d and Operations sections are repeated here within the Ontology context. Links are included between item entries wherever terms are closely related.

The hope is to make this document available on-line in "wiki" form such that it can feed this section of the MIOConsole3d manual and be updated periodically with user input from the on-line wiki:

Structural Terms

These terms are basic fundamental definitions which provide context and foundational base for all the terms to follow.

Channel

- **Short definition:** A single monophonic audio stream
- **Details:** A mono mic feed carries one channel. A stereo recording is two channels, a 5.1 submix bus is six channels, etc. The Tracks Overview shows the audio waveform of every channel within each Track as it is being recorded. Audio source channels are organized, routed and processed in the Console3d Mix desk as *Mixer Strips*, and are recorded, played back and edited in Session *Tracks*.

Audio file

- AKA: sound file
- **Short definition:** A file that contains one or more audio channels
- **Details:** Audio files are represented in the Track Overview timeline as Segments, with the audio content shown as amplitude waveforms. Each channel in the audio file will be represented with its own waveform.

'Split' or 'Multichannel' Audio Files

- AKA: split stereo, split multichannel, multiple mono, multi-mono...
- **Short definition:** A group of monophonic audio files which, when played together, represent a single multichannel audio stream.
- **Details:** Multi-mono audio files are often related to each other with a specific file name convention, such as with a tag like **.L** and **.R** to represent the Left and Right channels of a split stereo audio fileset.

  Some DAWs require that multichannel interleaved audio files be broken out to mono files before they can be edited.

Interleaved Audio File

- AKA: multichannel file
- **Short definition:** A single audio file that contains more than one channel.
- **Details:** In Session, a multichannel Strip feeds a multichannel Track which records the audio channels woven (i.e., "interleaved") together within the recorded audio file.

  Writing to interleaves is hugely helpful in keeping multichannel Session recordings organized and making sure all multichannel tracks stay grouped together, properly time-aligned and in the correct channel layout. Writing and reading interleaved audio files is also more efficient for the computer than multiple mono files.

  Session supports and edits mono and interleaved files equally.

  For those of us coming from an "*all recorded files are mono*" background: when you look in the Take folder, please do not freak out when you see the file sizes for multichannel tracks are vastly larger than mono tracks of the same length.

Mixer Strip

- AKA: Channel Strip
- **Short definition:** a processing and routing element in the hardware that manages one or more audio channels. Mixer Strips route and process audio channels and Tracks record, edit and play those audio channels
- **Details:** Mixer Strips are the vertically-oriented audio routing and processing control interfaces which reside in the Mixer desk. Each strip can contain up to eight channels. Mixer Strips can be Input strips, Aux bus strips, Group bus strips, the Main bus strip (and the Solo bus, if you're in PFL/AFL Solo mode). All Mixer Strips are eligible to be recorded, but only audio recorded on an Input strip will have an automatic playback route. All mixer strips automatically have an associated track in the Session, but only Input mixer strips have an input routing point to play back from Session Tracks.

Track

- AKA: EDL Panel
- **Short definition:** A list of one or more potentially overlapping segments arranged in time (basically the horizontal Session strip which contains waveforms). There is a one-to-one relationship between tracks and mixer strips.

  A *Track* in MIOConsole3d does not mean (and should not be confused with) a *Track* referring to a song in a CD, LP or streaming playlist. When using Session for album mastering and/or assembly, a playlist-type *Track* would be more closely represented by using a 'Marker With Duration' to denote the start and end of a complete song within the Session Marker List.
- **Details:** *Tracks* directly correspond to, and are named by the Mixer Strip sending audio to and playing back from its Track. The relationship between Mixer Strip and Session Track is analogous to a channel strip on a hardware recording console and the tracks on a 2" 24-track recording deck… the console strips all do the processing and routing to and from the multitrack, and the deck records to tracks and plays them back. That said, there is a small difference in that Tracks in Session can be multichannel. When you record to a track, the audio file will be recorded with the width of the associated mixer strip (e.g. mono strips record mono files, stereo strips record stereo files, etc.).

  Each Track contains a list of segments positioned in time. You can mix segments that refer to audio files with different channel widths within a single track by dragging in files of a different channel width, and the Session playback engine will do its best to accommodate. Stereo segments dragged into a mono Track will be summed to mono, and stereo segments dragged to a 5.1 track will play as channels L and R. Mono segments added to wider Tracks will send the audio to all the channels of the strip.

Session UI conventions

Session in many ways approximates the functions of a typical digital audio workstation, in that Session can record, edit and play audio files

Tracks Overview

- AKA:
- **Short definition:** A visual representation of Session Tracks and their audio segments
- **Details:** Within the Session window is a Track Overview for every track in the session. The track overview shows the position, length and contents of the segments in each track. It provides a user interface for you to move, trim, split, cut, copy and paste segments, as well as manipulate the segment metadata.

Tracks Overview: Mixer Input Strip

- AKA:
- **Short definition:** Tracks associated with Mixer Input Strips are the primary recording, playback and editing tracks
- **Details:** Only Tracks associated with Mixer Input Strips are able to both record and play back audio automatically, since only Input Strips include a hardware Source input route back to the Mixer.

Tracks Overview: Mixer Bus Strips

- **Short definition:** Bus Mixer Strips do have associated Tracks in the Tracks Overview primarily to maintain parity between Mixer Strip and Tracks layouts (critical for EuCon and MCP control surfaces operation)
- **Details:** Tracks associated with Mixer Bus Strips may be recorded to, and may contain audio, but since Bus Strips operate entirely within the Mixer engine with no connection to external Input routes, there is no direct route for playback from the Session Track.

Tracks Overview: DCAs

- AKA:
- **Short definition:** Like Bus tracks, DCAs also have associated Tracks in the Tracks Overview to maintain parity between Mixer Strip and Tracks layouts.
- **Details:** DCAs are control-only and can not record or play audio. DCA tracks may be used as a temporary "holding tank" within the Session for audio segments to be stored and used later, without fear of them accidentally interfering in the mix.

Track Headers

- AKA:
- **Short definition:**The control area for each Track, located at the left edge of the Session Tracks Overview
- **Details:** The Track Header displays the name of its' associated Mixer Strip, and provides remote control of the Mixer from the Session window, including the color, solo, mute and record enable buttons, panner (where applicable), and fader gain.

Timeline

- AKA: Timeline Rulers
- **Short definition:** A generic term referring to the visual representation and measurement of audio and musical events with regard to time in the Session Tracks Overview.
- **Details:** The *Bars*, *SMPTE*, *Time* (HH:MM:SS:ss), and *Samples* are ways of measuring and representing events in time. These timelines are also referred to as *Timeline Rulers*.

  The *Markers*, *Loop*, *In/Out* and *Selection* timelines are interactive tools used to define, display and operate Markers (location and duration), Loop Start and End, Autopunch In and Out points, and timeline Selection location and duration respectively.

  The *Length* component of a timeline event or operation is often referred to as its "duration" or "range".

Segment

- AKA: clip, soundbite, region
- **Short definition:** A reference to a contiguous section of an audio file. Includes metadata like fades, name, color, gain, mute.
- **Details:** Multiple segments may refer to the same section of an audio file. Segments referring to different audio files may also reside in the same Track.

Selection (verb)

- AKA:
- **Short definition:** The process of selecting a set of segments or a range of time in the session.
- **Details:** The methods for selection of Segments and Time Ranges are detailed in the immediately ensuing entries…

Selection

- AKA:
- **Short definition:** *(Segment Selection) verb*: The visual representation of said audio segments, once they are selected.
- **Details:** Segment selections apply to the entirety of one or more segments in the session. Segments are selected by clicking in the bottom half of the desired segment. Cmd-click to choose multiple segments. Multi-segment selections may be non-contiguous anywhere within the current timeline (including offscreen).

  Selected segments are indicated by a color bar across the bottom half of the segment.

Selection

- AKA: timeline selection
- **Short definition:** *(Time Range Selection) noun*: a region of time that applies either to:
  1. all tracks in the session or…
  2. to a contiguous set of tracks.
- **Details:** Time range selection may be applied across all Tracks in the Session (by click-dragging a range the "Selection" timeline) or by click-dragging a selection box across an area of the Tracks Overview UI. Time range selection gestures are initiated by clicking in the top half of a Track or Lane and dragging in any direction.

  All existing edits within Time Range selection boundaries are respected and reproduced exactly as selected.

Track Lane

- AKA:
- **Short definition:** A visual aid to reveal and allow selection and editing of overlapping Segments within a Track.
- **Details:** Session Track Lanes support unlimited segment overlaps, unlimited fade overlaps, and independent segment gain for all segments. All Track Lane summing, gain and fade operations occur within the playback engine prior to being sent to the 3d Mixer, and are completely independent of Mixer Strip processing and gain.

Transport

- AKA: play engine, record engine
- **Short definition:** The "engine" of Session which actively plays and/or records audio files in real time, and continuously positions the playhead and recordhead on the timeline with respect to the audio files represented in the Tracks Overview.
- **Details:** On a hardware reel-to-reel recorder, the *Transport* consists of the reel and capstan motors, the record and play heads, and the user controls [Stop, Play, Record, Fast-Forward, Rewind, Pause and auto-locate] which command the motion and re-location of the tape, and the record/play modes of the head stacks and audio circuits - basically the engine which actively moves the tape so real-time audio can be *played back* and *recorded*.

  In Session, the Transport serves the same purpose: moving the playhead and recordhead such that real-time audio is played or recorded. With Session Transport, however, the Playhead and Recordhead can be operating independently at different places on the timeline. Fast Forward or Rewind are of course replaced with instant timeline location cueing.

Playhead

- AKA: DP: "Wiper"
- **Short definition:** A white vertical line which spans all Session Timelines and Tracks. In Play *Continuous* and *Paged* Scroll Modes, the Playhead indicates the current Timeline position of the audio being played.
- **Details:** The Playhead line also serves as a cursor for placing edit split/trim points, Markers, Autopunch In/Out points, Loop Start and End, and for playback start/resume/relocate.

Recordhead

- AKA:
- **Short definition:** The point in the timeline at which the record engine is currently capturing audio to the Session track. Graphically, the Recordhead is the rightmost edge of the white-on-red audio waveforms being recorded.
- **Details:** Similar in nature to the Playhead (which marks the point in the Session timeline currently playing back), the Recordhead marks the point in the timeline currently being *recorded*. Usually the Play and Recordheads will be coincident, but Session *Take* and *Loop-to-Linear* modes especially allow for playing and recording to operate independently.

"Take"

- AKA: capture (as a noun)
- **Short definition:** A Session 'Take' can be defined as any set of audio files recorded simultaneously to the same location with a common naming convention and identical start and end file timestamps. A 'Take' is initiated every time *Record Mode* is engaged.
- **Details:** See the MIOConsole3d Session: Transport Bar: "Take" block section, Organizing and Identifying Takes: Take Folder and Audio File naming tools and Recording "Takes": Manual and Automatic features for further information.

Transport Full Stop

- AKA: "Stop" mode
- **Short definition:** The state in which both the playhead and recordhead are stationary (not moving) and *Record Pause* is not engaged. and the current "Take" is ended. Transport Stop button shows *yellow*.
- **Details:**

Record Pause

- AKA: record standby
- **Short definition:** A temporary state in which Record is engaged, but the Recordhead is stationary. Session *Pause* does not affect Play transport in any mode.
- **Details:** In Session, Transport *Pause* is a *record-mode-only* control which serves a specific purpose for live or field location recording, where the goal is to record contiguous soundfiles across extended periods of time, spanning hours or days with the possibility of extended breaks between recording events. It is not designed for use in tracking or mixing.

  Specifically, *Pause* is used to temporarily suspend recording while remaining in Record mode and within the current Take. When you click *Pause* the Pause button turns blue, waveform drawing in Tracks Overview stops and audio stops being written to your drives. The Record button will remain red, confirming that you are still in Record mode, waiting to continue recording the current Take.

Record Enable

- AKA:
- **Short definition:** *noun*: The control element on Mixer Strip and Track Header UIs which routes per-insert audio from that strip to the recording engine, such that when recording is engaged, the audio will be captured in realtime as an audio file.
- **Details:** It's that little button with the capital "R" located below the mixer strip level meter (and in the Track header) that turns red to indicate *Record Enabled* state.

  Engaging *Record Enable* anywhere in the MIOConsole 3d application immediately engages the record engine Ring Buffer in preparation for capturing a "Take".

Record Enable

- AKA: Record Arm, Track Arm
- **Short definition:** *verb*: The act of engaging (or arming) the *Record Enable* state.
- **Details:**

Record Ready

- AKA: Record Armed, Track Armed
- **Short definition:** The state in which a strips' audio is routed to the record engine (e.g. *Record Enabled*) but recording is not yet engaged.
- **Details:**

Record Mode

- AKA: Recording (as a verb), Record engaged
- **Short definition:** The state in which the Record transport (i.e., the Recordhead) is running and the record engine is actively capturing audio streams to audio files. Audio being recorded is drawn in its Track as white waveforms on a bright red background, progressing forward from left-to-right on the Session timeline.
- **Details:**

Record Standby

- AKA:
- **Short definition:** The state in which Record Mode is engaged, but the Recordhead transport is awaiting a level or timecode trigger, or an auto punch-in *Start* command to start the Recordhead moving and fully engage active recording mode
- **Details:** *Record Standby* is indicated in the Transport Controls by a flashing red Record button.

Preroll

- AKA:
- **Short definition:** A record engine feature which inserts a set duration of audio at the head of each audio file *before* the start of a Take.
- **Details:**

Ring Buffer

- AKA:
- **Short definition:** In the Session recording context, the *Ring Buffer* is a cyclic FIFO memory buffer which continually preloads 60 seconds of audio from all record-enabled inputs.
- **Details:** Audio stored in the ring buffer is used for data integrity insurance and for insertion as preroll audio before the beginning of a Take.

Postroll

- AKA:
- **Short definition:** A timer that delays disengaging record mode for a set time after the transport receives an automated *End Take* command
- **Details:** *Postroll* is most often used to capture long ringouts which might otherwise fall below Level De-trigger settings, and/or to act as a pseudo-freewheel over broken or unstable timecode when in Timecode trigger record mode.

Record Trigger mode

- AKA:
- **Short definition:** The four methods in Session which can engage Record Mode, specifically, *Manual*, *Level*, *Timecode* and *Autopunch*.
- **Details:** follow below…

Manual (Record Trigger mode)

- AKA:
- **Short definition:** Record Mode can only be engaged by a manual command, i.e., a button click, key command or control surface command.
- **Details:**

Level Trigger

- AKA:
- **Short definition:** In Level Trigger mode, the amplitude setting (in dBFS) at which Record Mode will be engaged.
- **Details:** When the input level at at Record Enabled Input reaches the Level Trigger setting, the recordhead starts moving and a new Take is recorded.

Level De-trigger

- AKA:
- **Short definition:** In Level Trigger mode, the amplitude setting (in dBFS) at which Record Mode will be disengaged.
- **Details:** when the input level at at Record Enabled Input falls below the Level De-trigger setting, the Postroll timer is engaged. If the input level reaches the Level Trigger setting before the Postroll times out, the Postroll timer is reset and the Take continues uninterrupted. Otherwise when Postroll times out, Session enters *Record Standby* mode, awaiting a new Level Trigger.

Timecode Trigger

- AKA:
- **Short definition:** In *Timecode Trigger Mode*, the presence of stable SMPTE time code at the selected SMPTE input port triggers Record Start (starting a new Take).
- **Details:** Note: Session Transport does not chase SMPTE timecode.

Timecode De-trigger

- AKA:
- **Short definition:** In *Timecode Trigger Mode*, the absence of stable SMPTE at the selected SMPTE input port *de-triggers* the current recording in progress (in other words initiates a Transport Record Stop command, stopping the current Take).
- **Details:** when SMPTE input stops (or becomes too weak or unstable to lock), the Postroll timer is engaged. If the timecode stream recovers before the Postroll times out, the Postroll timer is reset and the Take continues uninterrupted. Otherwise when Postroll times out, Session enters *Record Standby* mode, awaiting a new SMPTE stream.

Autopunch

- AKA:
- **Short definition:** A record engine function that automatically enters recording at a user-specified time (the In point) and exits record mode at a later user-specified time (the Out point).
- **Details:** Autopunch In/Out points are specified in the In/Out timeline. The selected Autopunch range is highlighted red when active.

In Point (Autopunch)

- AKA:
- **Short definition:** The point set in the In/Out timeline at which Autopunch engages record mode.
- **Details:** "In Point" in the Autopunch context refers to a point on the Session In/Out timeline and should not be confused with "In Point" in the 3 or 4-point audio editing context. Currently, Session does not support the 3 or 4-point edit model.

Out Point (Autopunch)

- AKA:
- **Short definition:** The point set in the In/Out timeline at which Autopunch exits record mode.
- **Details:** *Out Point* in the Autopunch context refers to a point on the Session In/Out timeline and should not be confused with *Out Point* in the 3 or 4-point audio editing context. Currently, Session does not support the 3 or 4-point edit model.

Loop

- AKA:
- **Short definition:** A playback function that seamlessly repeats a user-defined section of audio.
- **Details:** Loop Start and Loop End points are set in the Loop timeline. The selected Loop range is highlighted yellow when active. Loops may be saved as 'Markers with duration' to the Marker List.

Loop Start

- AKA:
- **Short definition:** The point set in the Loop timeline at which the Playhead start playing and returns to when it reaches *Loop End*.
- **Details:** Session Loop play is seamless regardless of how many audio channels are being looped, or at what sample rate.

Loop End

- AKA:
- **Short definition:** The point set in the Loop timeline at which the playhead returns to the "Loop Start" point and continues playback.
- **Details:**

Fade (Session)

- AKA:
- **Short definition:** An editing tool by which a segment smoothly transitions from silence to full amplitude or full amplitude to silence as a fixed function of the segment over time. Some duration of fade transition is required to safeguard against digital artifacts occurring at the beginning and end of a segment.
- **Details:**

Fade (Mixer)

- AKA: Fader move, Fade Out
- **Short definition:** A manual or automated Fader gain change over time, often but not always to or from silence.
- **Details:** Common use: referring to the slow 'fade to silence' at the end of a song.

In-Fade (Session)

- AKA:
- **Short definition:** *In-Fade* refers specifically to the fade tool placed at the beginning of every Session segment to transition from -∞ dB (silence) to full amplitude.
- **Details:** All Session segments have an In-Fade at the beginning of the segment, even if they are set to zero duration.

Out-Fade (Session)

- AKA:
- **Short definition:** *Out-Fade* refers specifically to the fade tool placed at the end of every Session segment to transition from full amplitude down to -∞ db (silence).
- **Details:** All Session segments have an Out-Fade at the end of the segment, even if they are set to zero duration.

Cross-Fade (Session)

- AKA:
- **Short definition:** The overlapping of one segments Out-Fade and the In-Fade of another segment, to smoothly and seamlessly transition from one segment directly to the next.
- **Details:** Crossfades are best visualized and edited in *Track Lane* view.

Import (Session)

- AKA:
- **Short definition:** A method to import new audio filesets. Session *Import* creates new Tracks in the Session without changing Mixer routing or processing.
- **Details:** Adding imported files as new tracks allows the user to audition each files' audio and then decide whether to place that audio into an existing Session Track, or to incorporate the new Track into the existing mix as-is.

Import Take Folder…

- AKA:
- **Short definition:** The Session: File menu command which places the audio files within a selected folder to a new set of Tracks within a Session.
- **Details:** *Import Take Folder…* functions assume that the audio files within selected folder were all recorded with the exactly the same start time, and are of the same sample rate.

Import Folder of Take Folders…

- AKA:
- **Short definition:** The Session: File menu command which places the audio files from folders nested within the selected enclosing folder to a new set of Tracks with in a Session, such that the contents of each folder are placed sequentially in the Session timeline.
- **Details:** In this way, multiple takes from the same live recording, or songs from the same multitrack recording session using the same basic instrument-to-track layout may inhabit the same Mix desk.

Import (manual)

- AKA: 'drag and drop' import
- **Short definition:** .wav and .aif audio files may be manually dragged and dropped from the file system into any Session Track
- **Details:** As with *Import Take…*, Session allows the manual import of any legitimate .wav or .aif file without regard to sample rate (note: FLAC audio files are not supported for Session playback).

Session Export (general function)

- AKA:
- **Short definition:** Session Exports are executed in the Session export engine at the file level, and create ".wav" files for maximum compatibility. All Session Export and Bounce commands follow the same basic ruleset:
- **Details:** All exports are at the native sample rate, bit depth and channel layout of the source audio file, regardless of the current hardware sample rate.

  Empty spaces in any Track are padded with digital zeroes such that all relative time relationships across all tracks is maintained, and all exported files are of identical length to the sample.

  All fades, crossfades and track lane summing are executed in the export engine during export. Since this is a file-level operation rather than playback through the mix desk, Input and bus strip gain and Insert processing is not applied.

Export Tracks (Session: Export menu)

- AKA:
- **Short definition:** A high-speed file-level command which converts all tracks in the Session as separate contiguous soundfiles, from timeline 00:00:00.00 through the end of the last segment Out-Fade, such that all exported soundfiles maintain their relative timing and are of identical duration.
- **Details:** Empty spaces in any Track are padded with digital zero such that all exported files are of identical length to the sample. All fades, crossfades and track lane summing are executed in the export engine during export. Since Session Export functions are file-level operations and do not pass through the mix desk, Input and Bus strip processing is not applied.

Export Tracks Between Loop Points… (Session: Export menu)

- AKA:
- **Short definition:** A high-speed file-level command which converts audio from all tracks in the Session as separate contiguous soundfiles, beginning at the selected Loop Start and ending at the Loop End point.
- **Details:**

Export Segment To File (Session contextual menu)

- AKA:
- **Short definition:** A file-level contextual menu command which exports the selected segment as its own .wav file
- **Details:** Note: *Export Segment To File* is currently a single-segment operation. To export multiple segments while maintaining their time relationship, use *Export Selection*.

Export Selection (Session: Export menu)

- AKA:
- **Short definition:** A file-level Export of the audio within timeline selection boundaries as individual sound files per track
- **Details:** As mentioned above, empty space between the segments and the selection boundaries will be padded with digital zeroes such that all exported files will be the same length with all audio time relationships maintained.

Bounce Track To File… (Session contextual menu)

- AKA:
- **Short definition:** A file-level contextual menu command which exports the selected track as its own .wav file.
- **Details:** Similar to *Export Segment To File (Session contextual menu)*, *Bounce* always exports a single entire track, regardless of segment selection.

Marker

- AKA:
- **Short definition:** A saved location on the *Markers* timeline. Each marker contains metadata including the name and duration of the marker, plus two comments fields. Markers are listed in the *Marker List* at the right side of the Session Overview.
- **Details:** Markers without duration (i.e., duration = zero) are used as playhead cue and event locators within the timeline.
  Markers with duration can define any section of audio, and are used to define, save and play back Loops. Markers with duration can be cued to both the Marker Start and End points.

Grid

- AKA:
- **Short definition:** The Grid is a quantization field for playhead and edit cursor placement within the Session track overview, making all Session timeline cursor placement snap to a Bars/Beats boundary.
- **Details:** The Grid popup sets the granularity of the grid in terms of musical units from bars down to 1/64 notes. The *dot* (.) and *triplet* (3) buttons modify the selected grid time to turn it into dotted (3/2) or triplet (2/3) duration respectively.

Cue (Cue & Monitor Controller)

- AKA:
- **Short definition:** In the context of the MIOConsole Mixer, Talkback and Cue Controller signal routing, a *Cue* is an independent monitoring path from the Control Room monitors, with Talkback features and often used to provide custom mixes for talent or production staff.
- **Details:** A *Cue* send generally refers to audio routed to an Aux mix bus being used to create a custom mix to studio talent headphone or speaker feeds. This Aux bus mix is itself called a *Cue mix*.

  You can think of Cues as named sends to physical outputs; they provide advanced routing controls to allow you feed them with the Monitor Controller source, dedicated buses (for cue mixes) and with a talkback or listenback signal.

Cue (Session)

- AKA: Cue Point, Return Point, Autolocate point
- **Short definition:** In the Session context, a *Cue Point* is any location in the timeline from which to start playback… usually a significant musical event which you will revisit often during recording or mixing.
- **Details:** All Marker start points can be used as Session Cue points. Cue to a Marker start by selecting the Marker number in the Marker List.

  Return Points (as in "a place the Playhead can **return** to") are more flexible Cue points within Session which move with you as you work through a recording or mix. Eligible events for Cue *Return Points* include: the Session start point (timeline 00:00:00.00), the current Selection start point, the current *Loop Start* point and the current Autopunch *In* Point. These *Return Points* may be enabled individually in the "Session: Transport: Enabled Return Points…" menu, and 'returned to' by hitting the *Return* key.

Mute (Mixer Strip)

- AKA:
- **Short definition:** The "M" button on any strip or track silences audio streams controlled by that strip or track (including strips controlled by Link Groups, Mute Groups, or DCAs)
- **Details:**

Segment Mute (Session contextual menu)

- AKA:
- **Short definition:** The Segment Mute command silences the selected segment.
- **Details:** When muted, the segment waveforms show dimmed, and the *Segment Mute* command becomes *Unmute Segment* in the contextual menu.

"Hard Mute" (Mixer Strip)

- AKA:
- **Short definition:** "Hard Mute" kills all input to the mixer strip, just like pulling out the source input cable. When enabled, "Hard Mute" is shown as a large square button at the top of each mixer strip. The "Hard Mute" button turns blue when engaged.
- **Details:** "Hard Mute" is disabled by default: this feature must be enabled at the top right of the "Configure Mixer" pane (**⇧⌘C**) before becoming available in "Configure Mixer Strips Controls".

D. Troubleshooting Guide

Computer does not see 3d device

If you attach a 3d device to your computer, and the computer is unable to communicate with the 3d device hardware there are five basic possibilities for the source of the problem:

1. The 3d device is not powered up
2. If connected with USB, the USB cable is not connected or bad
3. If connected with Ethernet, the Ethernet Adapter (NIC) is not Gigabit capable
4. If connected with Ethernet, the Ethernet cable is not connected or bad
5. If connected with Ethernet, the MHLinkDriver is not installed properly, or is being blocked by the operating system
6. The connection hardware has been damaged

3d device is not powered up

The first thing to check is that the 3d device is, in fact, powered up.

If the 3d device is powered up and booted properly, the Power, Sample Rate, and Locked front panel indicators will be illuminated. If these indicators are not illuminated, the 3d device is not powered properly or the unit's firmware has been corrupted. If you determine that you are powering 3d device properly and the indicators are not illuminated, you will need to contact Metric Halo support.

If the 3d device is properly powered, then check the next possibility.

Software is not installed properly

When connected to the computer via Ethernet, in order for the computer to properly communicate with the 3d device the driver software needs to be installed correctly. If the driver is not installed correctly the computer will fail to connect or discover the 3d device. If the MHLinkDriver.kext is not properly installed in the /Library/Extensions folder of your computer, and is not properly authorized to load by the user, you will not be able to use your 3d device for audio and you will not be able to control the device.

- The symptom of this is that the Front Panel FireWire/Computer indicator is not illuminated, and the Mobile I/O does not appear as a Sound Output device in the Sound panel of the "System Preferences" application, it does not appear in the Apple Audio/MIDI Setup Application, and it does not appear in MIOConsole3d.
- To correct this condition, make sure the MHLinkDriver.kext file is installed correctly, and reboot and then reconnect the 3d device to the computer. You must ensure that you have granted permission for the driver to load if you are running on High Sierra macOS 10.13 or newer (see our FAQ for more details: Driver Installation on High Sierra). If you have verified that the software is installed properly, check the next possibility.
- Another common problem was introduced with macOS Mojave (10.14.x) and Catalina (10.15.x). It involves yet another security measure, where you must grant permission for the MIOConsole3d application to access the "Microphone". Without this permission, everything appears to be set up and working properly but Core Audio passes digital zeroes instead of actual audio.

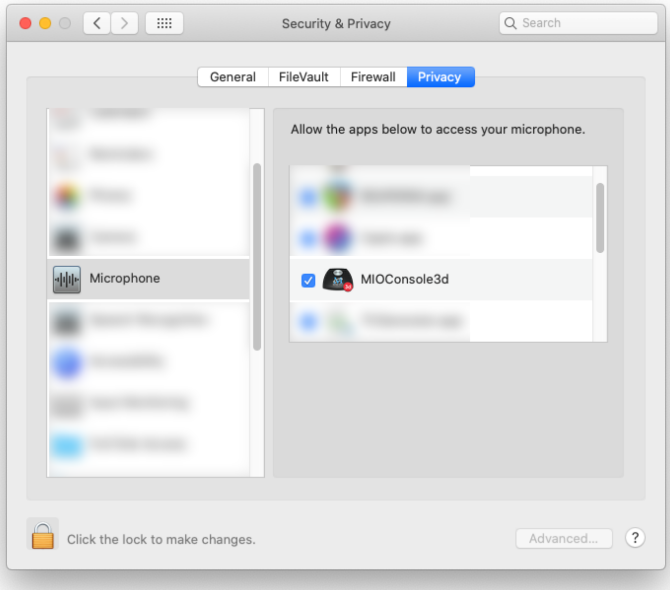


Figure D.1: Monitor Menu

To double-check this little loophole, go to the Apple menu: System Preferences.../Security & Privacy pane, select "Microphone" in the column to the left, and make sure that MIOConsole3d is checked.

The USB cable is not connected or bad

Check the simple thing first; is the unit connected? If the FireWire/Computer status LED on the front panel is not on and/or there is no information listed in the Unit Info pane in MIOConsole3d, confirm that the USB cable is plugged in firmly at both ends.

Metric Halo provides a high-quality USB cable for use with your 3d device and we recommend you use it. For various reasons you may decide to use other cables than the ones provided by Metric Halo. Under ideal circumstances all USB cables will provide years of service. However, cables will and do go bad. Cable failures can be difficult to track down. If you are experiencing problems with connecting your 3d device you should try swapping the cable with another known-good cable.

If the USB cable is not the source of the problem, check the next possibility.

The Ethernet cable is not connected or bad

Check the simple thing first; is the unit connected? If the FireWire/Computer status LED on the front panel is not on and/or there is no information listed in the Unit Info pane in MIOConsole3d, confirm that the Ethernet cable is plugged in firmly at both ends.

Metric Halo provides a high-quality Ethernet cables for use with your 3d device and we recommend you use them. For various reasons you may decide to use other cables than the ones provided by Metric Halo. Make sure that you use a proper Gigabit capable Cat 5 cable with all 8 conductors wired to the connectors. Your Ethernet cable may be up to 100 meters long. Longer may work, but it is possible that as you go past 100m connectivity becomes unreliable.

Under ideal circumstances all Ethernet cables will provide years of service. However, cables will and do go bad. Cable failures can be difficult to track down. If you are experiencing problems with connecting your 3d device you should try swapping the cable with another known-good cable.

If the Ethernet cable is not the source of the problem, check the next possibility.

Ethernet Adapter (NIC) is not Gigabit capable

MHLink relies on Gigabit connectivity; it will not work with a NIC that is running at 100Mbit/s. Make sure that the NIC you are using is Gigabit capable and is running in Gigabit mode. USB2 adapters do not support Gigabit Ethernet and must be avoided. Using a 4 conductor Ethernet cable will force the NIC to drop back to 100Mbit mode and will not connect to the MHLink device.

If the NIC is connecting at Gigabit speeds, check the next possibility.

The connection hardware has been damaged

If all else fails, it may be that the connectivity hardware on either the 3d device or the computer has been damaged. While this is an exceptionally rare occurrence, it is a possibility.

If the connectivity hardware on the computer has been damaged, it will not communicate with any other devices connected via the port. Be sure that you are not checking this case with a bad cable, as a bad cable can make it seem like the connectivity hardware has failed since it will consistently keep devices from connecting properly to the computer. If the computer is damaged, you will need to contact the manufacturer for a repair or, as a stopgap measure, you can use a third-party adapter card.

If the connectivity hardware on the 3d device has been damaged the device will not communicate with any other devices. In this case, please contact Metric Halo support for help in getting your Metric Halo hardware repaired.

Digital distortion

If your interface passes audio but there is distortion or other audible artifacts, check the DSP Usage in the Box Info area of the Console window. If this value is at 99-100%, you are overloading the DSP. Try removing some plug-ins, graphs, or busses to lower the DSP load.

Interface has crashed

If your interface becomes non-responsive or stops passing audio, you can try to reboot one or all attached interfaces by using the "Reboot Attached Units" commands in the Utilities menu. If this does not work, you will have to cycle power on the unit.

Corrupted console state

If your interface does not function properly when connected to a computer and MIOConsole3d is *running*, you may have a corrupted console state file.

Removing a corrupted state file
Make sure MIOConsole3d is not running, then open a Finder window and go to

`~/Library/Preferences/MIOConsole3d`

in your user folder (you can get there by doing the key command **⌘⇧G** in the Finder and pasting in that path). Delete the file “MIOConsoleState.cnsl3d” in the name. You may also want to delete any files that begin with “MHMonitorControlCfg”. This will clear the stored information from MIOConsole3d.

Ground Loops

> The **MHLink Gigabit Ethernet backplane** is immune to ground loops between 3d boxes plugged in at remote locations. This is because all Ethernet connections are transformer-coupled at both ends, providing galvanic isolation from the grounding issues normally caused by connecting devices running on different AC power circuits.
>
> In turn, the "twist" of the Ethernet cable Unshielded Twisted Pair (UTP) wiring provides rejection of RF noise interference over long (and short) cable runs.
>
> Please remember however, that local ground loops are still possible any time you connect analog or digital hardware devices.

Audio systems, in general, are susceptible to ground loop problems. Digital Audio Interfaces for computers are even more susceptible to grounding issues since they must interface with the computer's system ground, which tends to be much more dirty than the ground used by audio gear. By taking care when you connect the various components of your audio system you can avoid the hums, buzzes, and noises that characterize ground loops and other grounding problems.

First of all, most grounding issues go away if you utilize balanced interconnects between your audio gear. Balanced interconnects inherently reject ground differentials and common mode interference introduced by grounding problems. Balanced connections are not much more expensive than unbalanced connections and solve so many problems that if both ends of the connection support balanced interconnect, you should not even consider using unbalanced cables.

You may get the idea that we hate unbalanced connections. You're right. We do. You should too.

If you have to use unbalanced connections, or if any ground-related problems remain, you will find that the key to the issue is ensuring that you have a common hard ground between all the gear that you are interfacing. This is commonly referred to as a technical ground. A technical ground is characterized by a consistent low impedance path between each device and a common reference ground, ideally connected directly to earth ground. The above is sometimes difficult due to electrical wiring problems in the house, studio, or stage you are using. In the extreme case, you may need to hire a qualified electrician to untangle and correct electrical service problems in your working environment.

Unbalanced connections are a fact of life when interfacing with guitar amps, and, paradoxically, guitar amps are extremely sensitive to grounding issues since they use so much gain to achieve the effect of a "Guitar Amp". If you will be interfacing with guitars and guitar amps, you need to be very careful about grounding.

Common electrical wiring approaches to residential installations, and sub-par studio and stage installations use daisy-chained grounds for ease of installation and economy. Unfortunately, daisy-chained grounds can introduce significant ground differentials between sockets, and these differentials can vary depending on other loads (like refrigerators, TV's and other household appliances) on the circuit.

Other problems with electrical service installations are improper wiring of power phases to the three-phase service and improper connections between the safety ground and hot legs of the three-phase service. These types of problems tend to be characterized by loud 60Hz hums in the audio system. Unfortunately, these types of problems extend well beyond noise in your audio system to genuine safety hazards. If you determine that your electrical wiring has problems beyond a simple daisy-chained ground, you should consult a licensed electrician immediately, as ignoring these problems can damage either you or your gear.

If you do not have a well implemented technical ground, you will want to ensure that all of the devices in your audio system are plugged into the same phase and same ground. You can generally accomplish this by running all your gear off of the same socket (using a power strip or power conditioner) if your gear uses less power than is supplied by a single circuit from your premises' wiring (generally 10-15 amps in residential installations and 20 amps in commercial installations).

It is usually a bad idea to put some devices in your system on a power conditioner and other devices on a separate strip, socket or conditioner, unless you have a technical ground. The power conditioner can introduce a ground differential.

The power supply provided with Mac laptops does not have a hard ground. This means that if the laptop is plugged in, it will dump high frequency buzz into the ground. That ground is shared with the 3d USB cable. If a 3d device will be connected unbalanced to other audio gear, the ground buzz can contaminate the signal if the 3d device is not hard-grounded to the same ground as your other audio gear. To hard ground the 3d device you will need to use a 3-pin power cable on the 3d device power supply and power the 3d device with the power supply. Plug the 3-pin IEC power cable into the same circuit and same ground as your other gear.

Note that unlike USB, the Ethernet connection is galvanically isolated (e.g. transformer coupled) and disconnects the ground between the devices at the two ends of cable. As a result any connection made via MHLink cannot introduce a ground loop, and in the case of connecting to the computer, fully isolates the audio ground from the computer ground.

On the other hand, if you are encountering ground loop problems while operating with the 3d device's power supply, you may find that lifting the device's ground resolves the problem. This can be accomplished by using a 2-Pin IEC cable (without the third ground pin), or by using a ground lift block (generally available in hardware stores, also known as a 3 pin to 2 pin converter). In general, it is better to resolve the fundamental grounding problems in your system, but this is a quick fix that may help. There are no hard and fast rules for solving this type of problem other than fixing the fundamental grounding issues, so if you go this route, you will have to experiment with lifting various grounds in your system until you find the magic combination. Or switch to balanced interconnects.

Firmware update problems

Details on updating the firmware of 3d Devices are available in the Firmware Installation section.

It is possible for firmware updates to "not take". This appears to be related to DSP loading issues in the 3d device, other devices on the USB bus (if connected via USB), and the state of the USB system software on the Mac. If you have problems with updating the firmware, try the following procedure:

1. Remove all devices from the USB bus
2. Power down the 3d device
3. Reboot your computer
4. Power up the 3d device
5. Run the firmware update

In cases where a firmware update to an MHLinked device does not seem to have registered, just run the firmware update procedure again, making sure to fully power down every box on the MHLink chain for a good 20 seconds before bringing them all back on-line.

Since 3d devices implement safe boot and safe firmware update protection in the hardware, you should always be able to use this procedure to update the firmware, even if something goes horribly wrong (like losing power during an update).

E. DB25 Pinouts

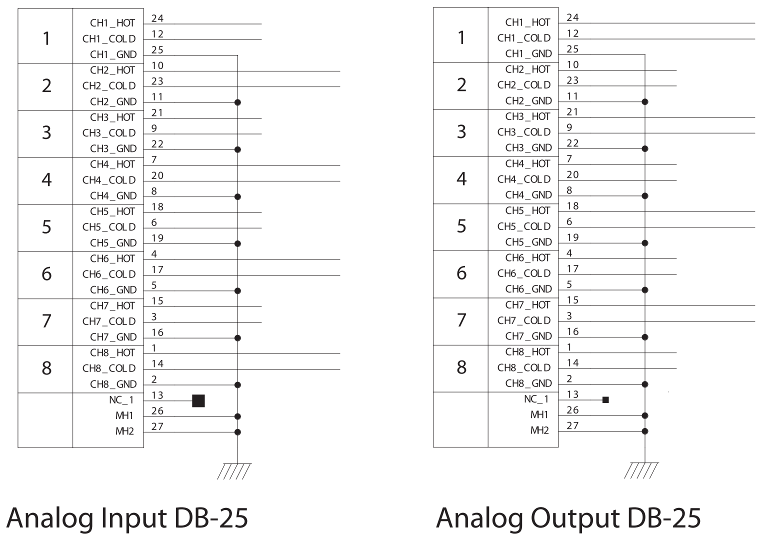


Figure E.1: Analog DB25 pinouts

Common wiring pinouts:

XLR:

- HOT: Pin 2
- COLD: Pin 3
- GND: Pin 1

TRS:

- HOT: Tip
- COLD: Ring
- GND: Shield

TS (unbalanced 1/4"):

- HOT: Tip
- COLD: Do not connect
- GND: Shield

RCA:

- HOT: Pin
- COLD: Do not connect
- GND: Barrel

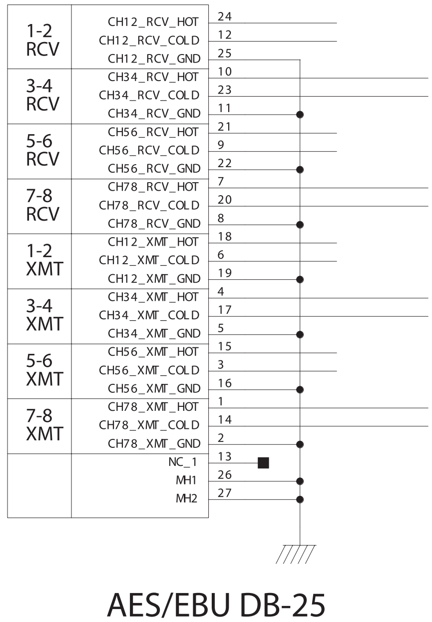


Figure E.2: AES I/O DB25 pinouts

Wiring pinout for XLR:

- HOT: Pin 2
- COLD: Pin 3
- GND: Pin 1

DB25 #1	DB25 #2
24	18
12	6
25	19
10	4
23	17
11	5
21	15
9	3
22	16
7	1
20	14
8	2

Table E.1. AES crossover cable pinout

Click here for a larger version of all pinouts

F. ULN-8/LIO-8 Jumper Settings

Overview

There are a number of settings that can be configured in the ULN-8 and LIO-8 using jumpers. We'll detail them here.

To access the ULN-8/LIO-8 configuration jumpers:

1. Remove the six screws from the top of the case:

Figure F.1: Top Screw Placement

2. Remove the screws from the left and right sides of the case. If the rack ears are fitted, there will be five screws per side. *Please note that the screws on the rack ears are longer than the others. Be sure to put the longer screws back in the rack ears when you reassemble the LIO-8.*

Figure F.2: Side Screw Placement, with Rack Ear

If the rack ears are not fitted, there will be three screws per side.

Figure F.3: Side Screw Placement, No Rack Ear

Remove the top cover, and rotate the unit so that the rear panel is facing you.

Reassembly is the reverse.

D.I. Board

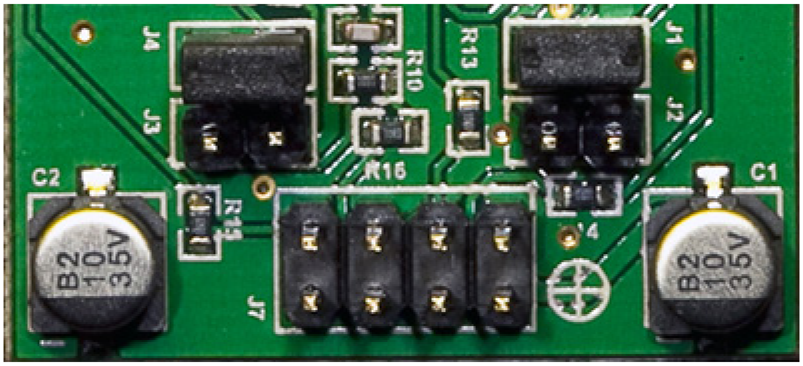


Figure F.4: D.I. Board Jumper Sites

There are 4 pairs of pin headers on the DI board mounted to the front panel, two pair per channel. The pair of pins closest to the TRS connectors (for each channel) is the low gain set. The pair closest to the ribbon cable is the high gain set.

When installing a jumper, it should be within the white box next to the jumper label, not connecting adjacent jumpers.

Gain	Channel 1	Channel 2
0.0 dB	None	None
10 dB	J4	J1
17.85 dB	J3	J2
20 dB	J3 and J4	J1 and J2

Table F.1. D.I. board gain settings

Line input grounding

The shield on the line inputs can be lifted on a per-channel basis. These jumpers can be accessed directly on the LIO-8; to access them on the ULN-8 (or LIO-8 with mic pres) you must remove the mic pre(s) and connector board. These jumpers are located on the connector board at the rear of the interface, directly behind the Line input connector. The channels start with input 1 on J1 and go the the right, ending with channel 8 on J18.

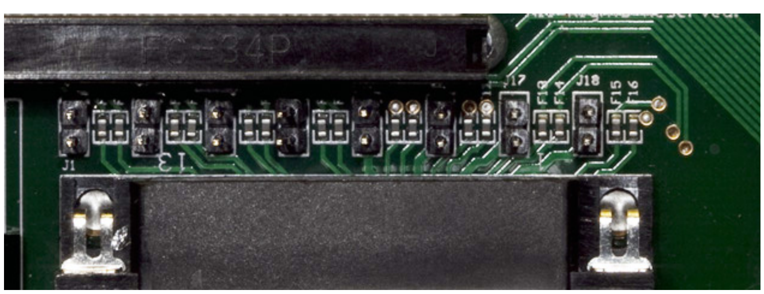


Figure F.5: Shield Lift Jumper Sites

By removing the jumpers, you disconnect the shield from the connected input. Having the jumper installed connects the shield to the ground of the interface.

> **Connecting unbalanced sources**
> It has been our experience that it is best to have the ground jumpers *installed* when connecting unbalanced sources to the line inputs.

Output levels

Jumpers can be installed to raise the operating output level of the analog outputs. There are a few reasons why they are not installed by default:

- The factory configuration of the ULN-8 Line outputs is +18 dBu, which is a very common level for interfacing with a large majority of existing gear.
- The noise performance is better optimized when the outputs are set to to Monitor mode when interfacing with powered monitors.
- The box runs cooler and uses less power.
- The jumpers can always be applied to raise the outputs to +24 dBu, for a chain that is optimized at this operating level (i.e. chains at many mastering studios). At this level, 0 VU = -20dBFS.

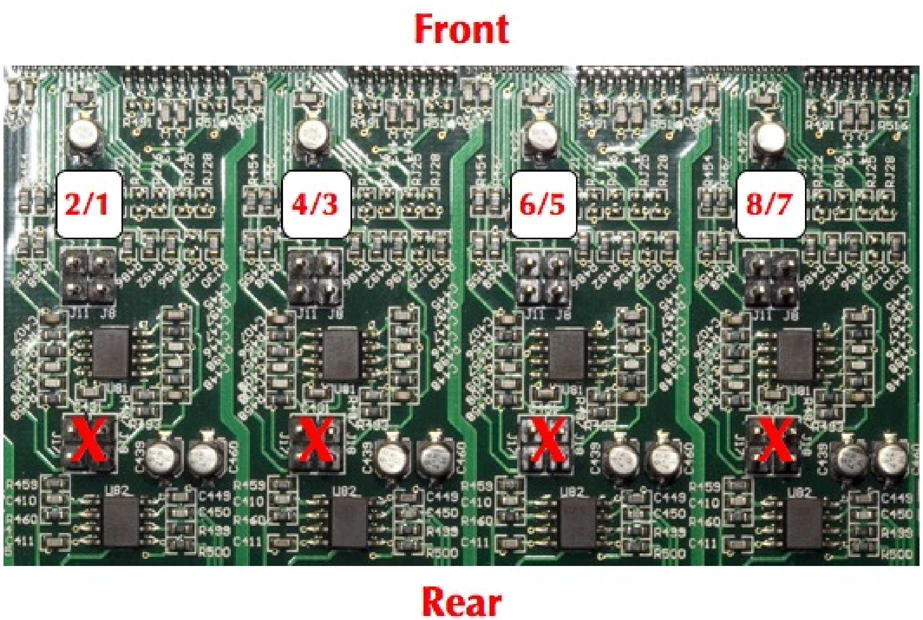


Figure F.6: Output Level Jumper Sites

These jumpers are located on the largest circuit board, in line with the Line/Monitor output connector. In the diagram above, the channels are in the following order from left to right: 2 1 4 3 6 5 8 7. In each case, J11 corresponds to the even (right) channel of the pair, and J8 corresponds to the odd (left) channel of the pair.

Setting the output level jumpers

To increase the output levels, install the jumpers on the sites toward the front of the unit.

Do not install jumpers to the sites closest toward the rear panel.

You must also install jumper J2 on the power supply board.

The jumpers *must* run from front to rear, not side to side. Unless you have a *very* specific reason all jumpers should either be removed or present, not intermixed. The difference in output level could lead to confusion.

Power supply

There are three jumpers on the power supply board:

- J9 disables the power supply for the preamps
- J8 disables the phantom power supply

These should *both* be installed if no preamps are installed. They must be removed if any preamps are installed.

- J7 bypasses the front panel power switch. Installing a jumper at this site will provide constant power from the 4 pin XLR power jack.
- J2 enables high power mode. It must be installed if any of the output jumpers are installed.

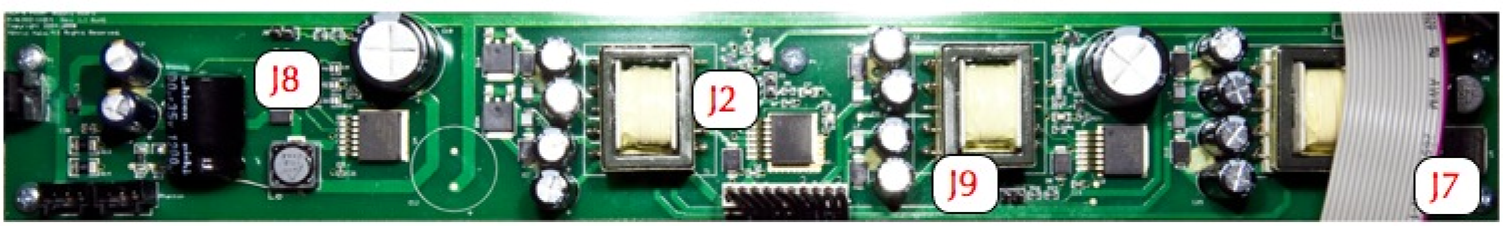


Figure F.7: PSU Jumper Sites

G. ULN-R Installation Guide

ULN-R Parts and Tools

The ULN-R Mic Pre (Ch. 1-4) kit includes the following parts:

- 1) DB25 connector board

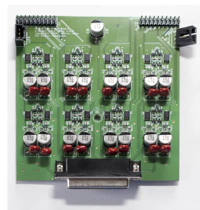


Figure G.1: DB25 Connector Board

- 1) 4 channel ULN-R mic pre board

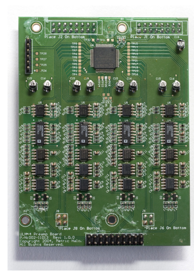


Figure G.2: 4 Channel ULN-R Mic Pre Board

- 1) 3 pin phantom power cable

Figure G.3: 3 Pin Phantom Power Cable

- 1) 20 Pin Ribbon Cable Jumper

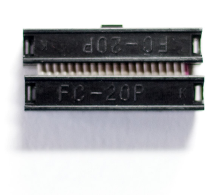


Figure G.4: 20 Pin Ribbon Cable Jumper

- 2) 7/16” standoffs

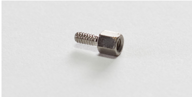


Figure G.5: 7/16" Standoffs

- 3) Phillips head screws

Figure G.6: Phillips Head Screw

The ULN-R Mic Pre (Ch. 5-8) kit includes the following parts:

- 1) 4 channel ULN-R mic pre board
- 1) 20 pin ribbon cable jumper
- 4) Phillips head screws

The Ch. 1-4 kit must be installed for the Ch 5-8 kit to function!

To install the ULN-R mic pre option in the LIO-8 you will need:

- #1 Phillips screwdriver
- 7/16" nutdriver (Ch. 1-4 kit only)
- silicone caulk

> Please familiarize yourself with the parts and instructions before opening the LIO-8.
>
> Be sure to discharge any static energy on your body before touching the interior of the LIO-8.

Installing the mic pres:

1. Remove the six screws from the top of the case:

Figure G.7: Top Screw Placement

2. Remove the screws from the left and right sides of the case. If the rack ears are fitted, there will be five screws per side. *Please note that the screws on the rack ears are longer than the others. Be sure to put the longer screws back in the rack ears when you reassemble the LIO-8.*

Figure G.8: Side Screw Placement, with Rack Ear

If the rack ears are not fitted, there will be three screws per side.

Figure G.9: Side Screw Placement, No Rack Ear

Remove the top cover, and rotate the unit so that the rear panel is facing you. If you are installing the Ch. 5-8 kit only, jump to step 9.

3. Remove the two screws holding the Mic In cover plate, then remove the plate from inside the LIO-8.

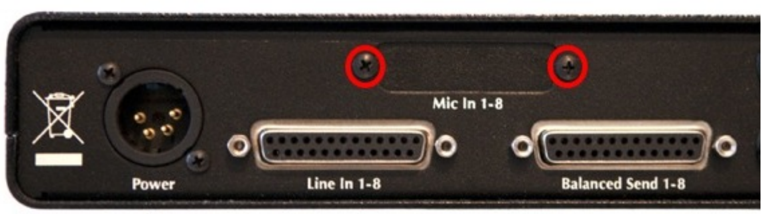


Figure G.10: Mic In Blank Plate

4. Put the DB25 connector board into the LIO-8 so that the connector comes through the hole that you just uncovered. Put the two 7/16" standoffs through the rear panel and screw them into the DB25 connector.

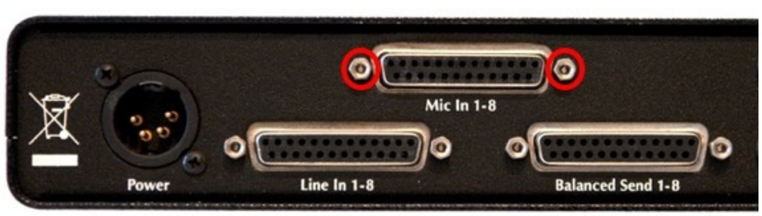


Figure G.11: DB25 Connector Board Fitted

5. Use a Phillips head screw to secure the connector board to the standoff below it and put a dab of silicone caulk on the screw to hold it in place. Next, make sure that connector J5 is bent out at a slight angle; this is to ensure that the phantom power cable will clear the top cover.

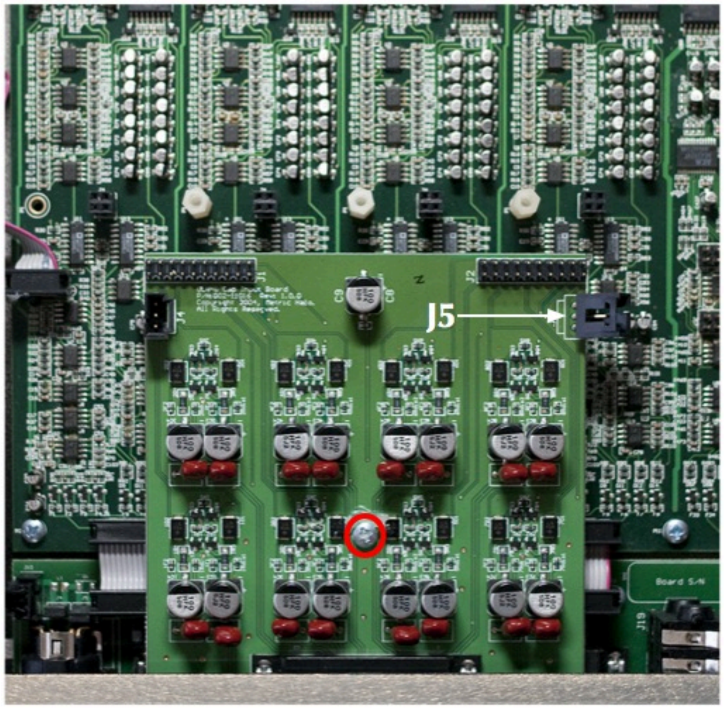


Figure G.12: DB25 Connector Board (Interior)

6. Remove the jumpers from J8 and J9 on the power supply board at the front of the LIO:

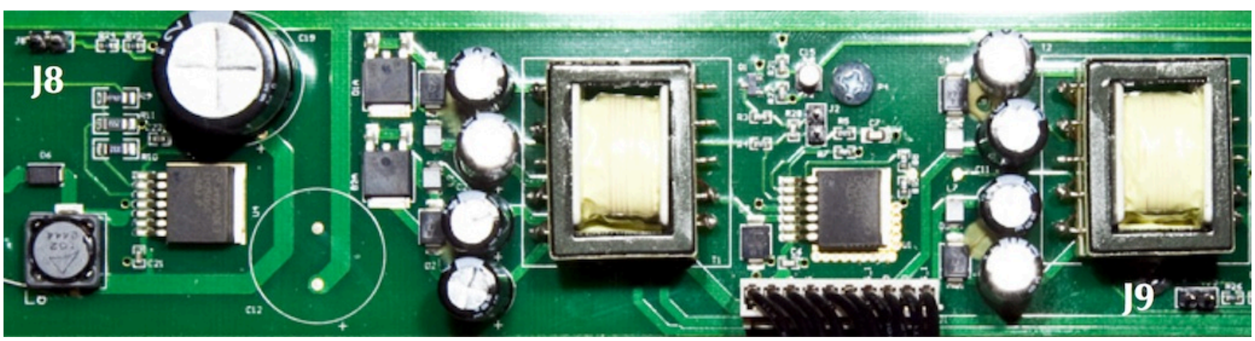


Figure G.13: Phantom Power Jumpers on the PSU Board

7. Connect the phantom power cable between connector J5 on the DB25 connector board to connector J5 on the power supply board at the front of the LIO:

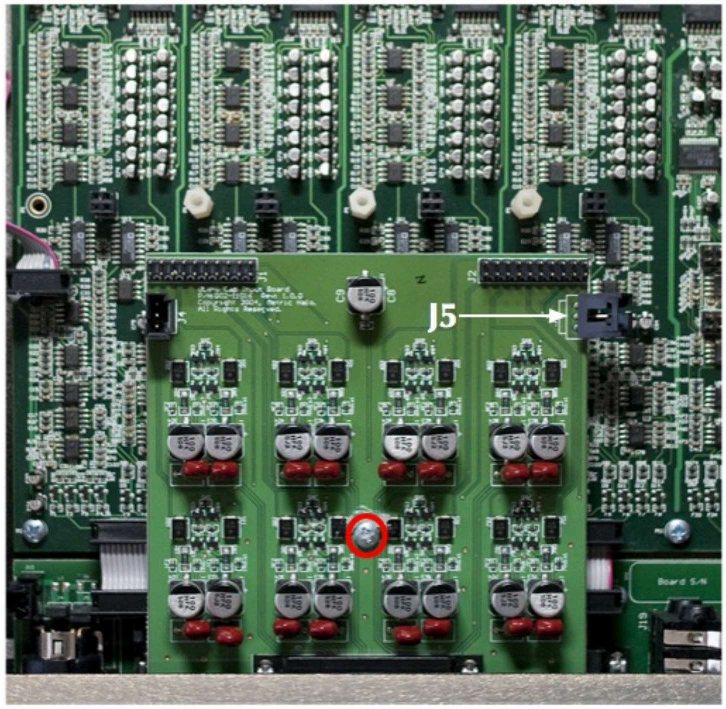


Figure G.14: Phantom Supply Cable and Ch. 1-4 Connectors

8. Install the mic pre board, making sure that the pins on the bottom of the board line up with the sockets highlighted in the picture above. You must install the channel 1-4 board in this position. Use two Phillips head screws to secure the mic pre board and dab them with caulk. Fit the ribbon cable jumper between the mic pre and DB25 connector board, making sure that the pins are lined up.

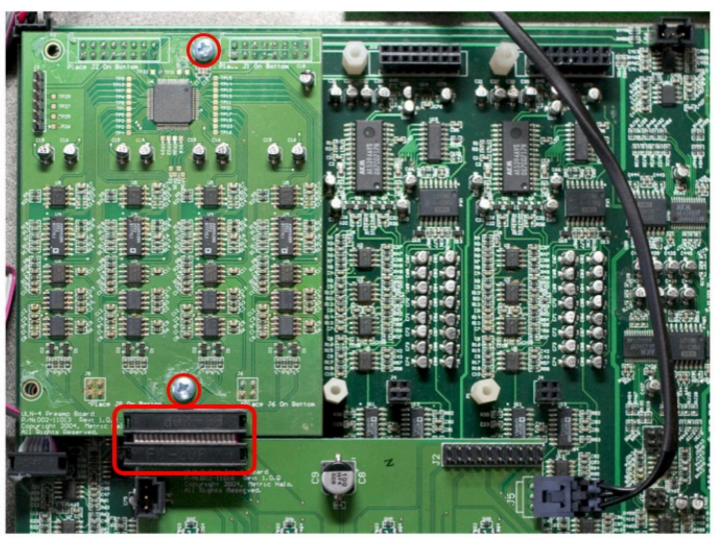


Figure G.15: Installing the Ch. 1-4 Mic Pre Board

9. If you are installing the channel 5-8 mic pre kit, install it in the sockets next to the channel 1-4 board, making sure the pins are properly seated. Use four Phillips head screws to secure the mic pre board

and dab them with caulk. Fit the ribbon cable jumper between the mic pre and DB25 connector board, making sure that the pins are lined up.

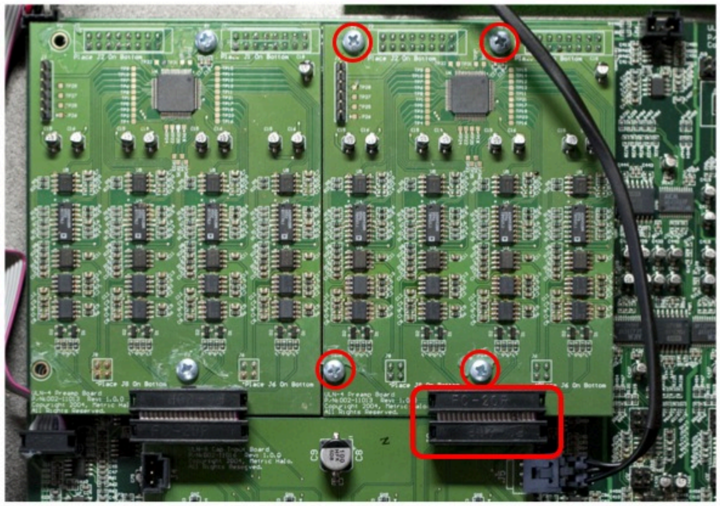


Figure G.16: Installing the Ch. 5-8 Mic Pre Board

The installation is finished! Replace the top cover, and replace the six crews on the top of the case. Then replace the screws on the left and right side, remembering to use the longer screws on the rack ears. It sometimes helps to squeeze the top and bottom of the case together when replacing the side screws, to ensure that the outer holes line up with the threaded inserts.

Once you have reassembled the unit, connect it to your computer and power it up. When looking at an Analog input in MIOConsole3d, you will now have the "Mic" and "Mic S/R" selections as input choices, as well as phantom power control.

If you have any questions about the installation process, contact support@mhlabs.com.

H. Support Resources

Metric Halo has several resources to help you; if you have questions that aren't answered in this document, we have further materials online:

- Our FAQ: https://www.mhsecure.com/faq
- Our technote and tutorial library: https://mhsecure.com/metric_halo/support/tutorials.html

You may also consider joining our user email list, where you may ask questions of other users. You can subscribe to the list at https://mail.music.vt.edu/mailman/listinfo/mobileio.

If you still have questions or are experiencing problems with your interface, you can open a support ticket online at https://www.mhsecure.com/support.

Glossary

Symbols

DSP — A collection of plug-ins that run on the processors inside our interfaces. There are over 100 plug-ins included in the Metric Halo 3d mix environment.

2d — Metric Halo’s second generation DSP card.

3d — Metric Halo’s third generation audio platform, comprised of the MH-Link Gigabit Ethernet audio backplane, automatic multi-box aggregation, EdgeBus expansion card slots, with driverless USB UAC2 computer connectivity. The 3d platform is part of every new interface, and is available as an upgrade for every previous Metric Halo model.

Ω — See *Ohm*

A

ADAT — In the context of Metric Halo interfaces, a standard for digitally connecting two devices together. ADAT sends and receives 8 channels of 24 bit/48k sample rate audio over one TOSLINK optical cable.

ADC — Analog to Digital Converter. The integrated circuit that receives an analog signal at its input and outputs a digital representation of it.

AES — In the context of Metric Halo interfaces, a standard for digitally connecting two devices together. AES sends and receives two channels of audio per cable, using 110 Ω cable. A single cable is usually terminated with XLR (or sometimes BNC) connectors. Multichannel AES devices (such as the LIO-8 and ULN-8) often will use multichannel cables terminated with DB25 connectors.

Attenuate — To turn down. Commonly refers to a tool used to lower the output of a device to make its operating level compatible with another device's input, i.e. an inline attenuator placed between the output of a 2882 and the input of a power amplifier.

Aux — See *Bus*

B

Balanced connection — A method of connecting audio equipment that is very resistant to electrical noise. A balanced connector usually has three pins per channel. At the output the connections are:

- ”Hot” or “Plus”: The unmodified signal
- ”Cold” or “Minus”: The signal 180° out of phase relative to the original

- Ground

At the input, the "Cold" signal is rotated in phase 180° again and summed with the "Hot" signal. Any noise picked up in the cable between the devices is now out of phase in the "Cold" signal, and when the "Hot" and "Cold" are summed the noise will cancel and only the original signal will remain.

BNC
: A locking connector used for connecting RF and digital interfacing signals over a coaxial cable. Commonly used in audio for wordclock connections, as well as digital interfacing.

Boost
: To increase gain.

Bus
: A bus is used to mix signals together. To hear all of your inputs, they need to be assigned to a bus, and the bus must be assigned to an output (such as monitors or headphones). There are several types of buses:

- Main/Master summing mix bus: this is where all the signals are finally mixed together.

  In MIOConsole3d, the master bus outputs to any Group Bus, MH-Link or USB Host channel, SCP and/or physical output. The main bus is always routed to the "Main" Monitor Controller source input.

- Group Bus: Also called sub-mix buses, Group buses share the same fader, mute, and pan settings with the Main mix bus. Group buses can be routed to hardware ports out to the world, to other Group buses, and/or the Main mix.

- Aux bus: Used for "auxiliary" mixes, such as cue mixes and reverb sends. Aux buses include their own fader, pan and mute for each input strip independently of the Main mix bus. In MIOConsole3d, an Aux bus outputs to a Group Bus, MHLink or USB Host channel, SCP and/or physical output and/or the Main/Master bus. Example uses would be for reverb sends or stem mixing.

C

Cans
: Another term for headphones.

Coaxial
: Within the scope of this manual, coaxial refers to a type of cable that has a central conductor (used for the signal) surrounded by insulation which is then surrounded by a woven metal shield (used for ground) and finally an outer jacket.

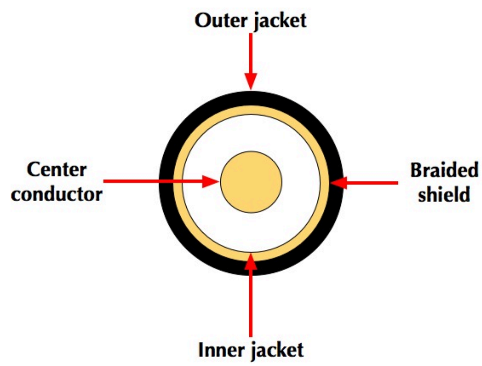


Figure 723: Cross-section of a Coaxial Cable

Cut — To decrease gain.

D

DAC — Digital to Analog Converter. The integrated circuit that receives digital data at its input and outputs an analog representation of it.

DB25 — A D-sub connector with 25 pins. These are becoming more common in audio devices, since you can connect eight balanced channels of audio with one DB25. Two devices with DB25 connectors can be directly patched together with a DB25 to DB25 cable, or you may use a DB25 fan-out to interface with other types of connectors, such as XLR or TRS.

dB — Decibel; a logarithmic unit of measurement used to describe sound levels relative to a reference level, both acoustically and electrically. There are several suffixes added to dB to indicate the reference quantity or weighting function used:

- dBFS: Decibels relative to full scale; 0 dBFS is the maximum level in a digital audio system, since the level cannot go over 0.
- dBu: Decibels relative to 0.775 volts, across a 600 Ω load.
- dBV: Decibels relative to 1 volt, with no impedance specification.

DI — Direct Inject. Used to connect a high impedance, low level unbalanced signal to a low impedance balanced input. The DI input provides a high-impedance to the source keeping the input from loading down or filtering the source. DI inputs are used to connect guitar, bass and keyboards to the interface. Sometimes also called Direct Input.

Direct out — An output that allows you to route a signal without going through a bus. In MIOConsole3d's Mixer, input strips have both pre-insert and post-insert direct outs that allow you to send audio to the Host computer, USB or physical analog/digital outputs.

Dither — The process of intentionally adding low-level noise to a signal to remove quantization distortion. Dither is most commonly used when preparing 24 bit material for CD delivery (which is 16 bit) and should be the last step in the process.

Dry

An uneffected audio signal. If you were to record a signal before and after running it through a reverb, the pre-reverb signal would be dry and the post-reverb signal would be wet. Many processors have controls to let you set the wet/dry ratio.

DSP

Digital Signal Processing. This can mean:

- A physical integrated circuit that processes audio- “All Metric Halo 3d interfaces have DSP chips.”
- The program that makes a DSP chip do useful things, usually referred to as plug-ins- “The MIOStrip DSP plug-in gives me gating, compression and EQ right in the interface!”

F

Fader

A form of level control; inputs in the Mixer have faders that set the level at which they send audio to buses, and buses have faders that set their output level.

fs

Stands for sampling rate. For example, fs=44.1k means that the sample rate is 44,100 samples per second.

G

Gain

The measure of a device’s ability to increase the level of an audio signal. In audio, gain is usually referred to in dB; for example, the mic preamps in the ULN-8 have 92.5 dB of gain. This means that they can take an incoming signal and increase its level by 92.5 dB.

Graph

A free-form area for creating signal processors. Inputs are on the left and outputs are on the right. DSP building blocks are placed in the Graph and connected by dragging virtual wires between the inputs and outputs of the blocks and the Graph itself. Graphs can be saved for repeated use. More in-depth information is available in the Graphs section of the DSP Implementation Guide.

Group

See *Stem*

H

Head amp

Head Amplifier - another term for microphone preamplifier.

Headroom

The amount that an audio signal can go above a nominal point before distortion. The maximum level in a digital system is 0 dBFS, so if we say the nominal point is -18 dB, the system has 18 dB of headroom. Exceeding the headroom in a system or device will cause distortion.

I

Impedance

The opposition of a circuit to alternating current. In audio, it is important to match the impedance characteristics of inputs and outputs:

- High impedance sources (guitar, bass, keyboards) should be plugged into a DI. These are commonly called “Hi-Z”.
- Low impedance sources (microphones, line-level devices) should be plugged into mic preamps or line-level inputs. These are commonly called “Lo-Z”.

Insert

There are two kinds of inserts used by Metric Halo:

- Analog: The ULN-2 and ULN-8 offer insert points that allow external processors to be patched between the onboard mic pres and A/D converters. The output is called the send, and the input back into the interface is the return.
- DSP: MIOConsole3d has 10 insert slots per strip that allow plug-ins, Graphs, and sends to be placed in the signal path.

L

Latency

The delay between when an audio signal enters and exits a device, process or sound system. A system with high latency sounds "behind" the performers, because there is an audible delay between when they make a sound and it is heard from the audio gear.

Line level

Describes the electrical signal level used to connect audio devices together. Common nominal line levels are +4 dBu and -10 dBV.

Linear fade

A fade up or down which has a constant gain change over time. In the example below, the linear fade causes the straight lines in the fade in and out.

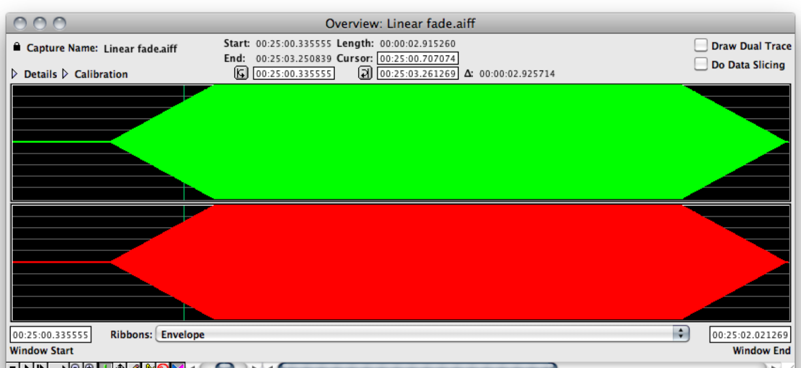


Figure 724: Linear Fade

Logarithmic fade

A fade up or down which changes gain faster at the end of a fade up and faster at the beginning of a fade out. In the example below, the Logarithmic fade causes the curved lines in the fade in and out. A logarithmic fade tends to sound more natural on audio material.

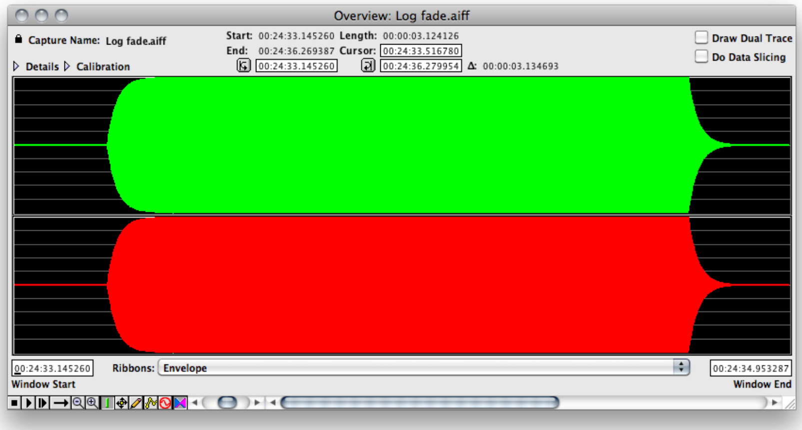


Figure 725: Logarithmic Fade

M

Mic Pre

Microphone Preamplifier. A device (or part of a larger device such as a mixing console or audio interface) that takes the low output signal of a microphone (referred to as mic level) and amplifies it to a higher level (referred to as line level).

This must be done in order to make the microphone's output compatible with the working level of mixers and other devices, giving it the name "pre" amplifier.

M/S

Stands for Mid/Side, and most commonly refers to a technique of using two microphones-

- Mid, (cardioid, omni or figure-8) facing the event to be recorded
- Side, (always figure-8) turned 90° in relation to the Mid. Should be placed as close as possible to the Mid’s diaphragm.

The two signals are matrixed to stereo using a M/S decoder. The advantage of M/S recording is that the stereo width is variable from true mono through stereo and can be modified after recording.

The M/S Processor included with +DSP adds the ability to *encode* stereo to M/S, allowing you to separately processes the center and sides of a pre-recorded stereo signal.

Mult

To send one signal to “multiple” places at once. For example, you could mult your stereo mix to Analog 1-2 and Digital 1-2 to send them to different devices simultaneously.

Mult can also refer to a multichannel cable; for example, a DB25 cable is used as an eight channel mult with the LIO-8 and ULN-8.

N

Nominal level

The operating level at which a device is designed to operate. For example, the nominal level of a professional audio device is usually +4dBu, while the nominal level of a microphone is typically -60 dBv. To connect these devices together we need a third device to make their nominal levels compatible, in this case a mic pre.

O

Ohm (Ω)

Unit used in the measurement of resistance and impedance. Often used with prefixes to indicate large values, such as "k" (kilo) which equals 1000; for example, 5 kΩ means five thousand ohms (or 5 kilo ohms) . Similarly, "M" (mega) is used to indicate 1,000,000; so 10 MΩ means 10 million ohms (or 10 megaohms).

Optical SPDIF

A standard for digitally connecting two devices together. Optical SPDIF sends and receives two channels of audio, using a TOSLINK optical cable.

Over

Occurs when a signal exceeds the headroom of a system, potentially causing unpleasant distortion.

P

Panner

Allows you to determine where a sound is heard in the soundfield. The panners in MIOConsole3d are context sensitive, meaning they change depending on how many channels are in the bus you are sending to:

- Mono bus: No panner
- Stereo or LCR bus: Pan knob
- Quad to 7.1 bus: "X-Y" surround panner

Phantom power

A DC voltage applied to a microphone input used to power a condenser microphone or DI box. Phantom power is most commonly 48 volts, and should not be turned on if a line level device is connected to the input.

Phase

Describes the position of one sound wave relative to another, or in relation to time. Let's look at phase between two signals. In the first example, the two signals have the same phase:

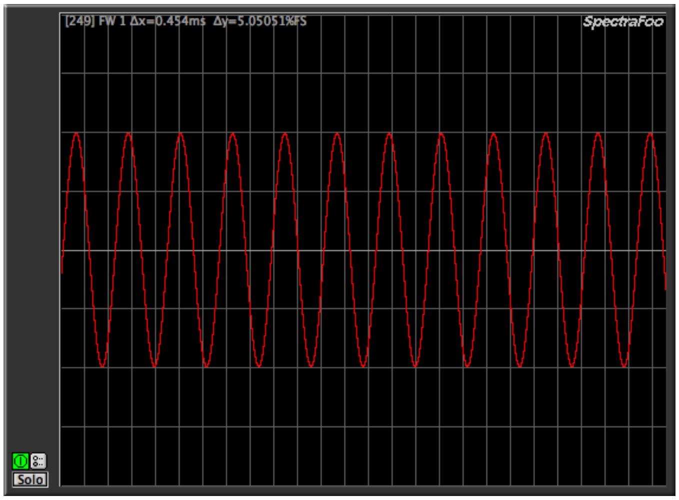


Figure 726: 0° Phase Offset

The two signals are drawn perfectly on top of each other. In the next example, the right signal is 180° out of phase with the left signal. When the audio on the left is at its highest, the right is at its lowest and vice versa. A signal that is 180° out of phase is also called "polarity inverted" or "phase inverted"

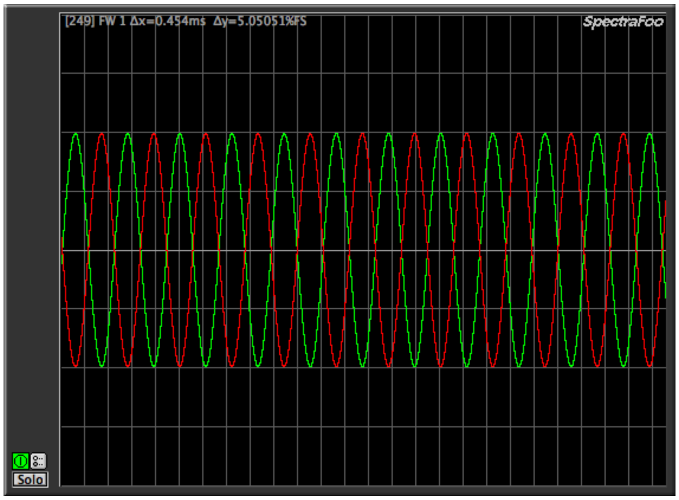


Figure 727: 180° Phase Offset

If we sum the signals together, we get nothing; the high of the left is cancelled out by the low of the right. This is called phase cancellation

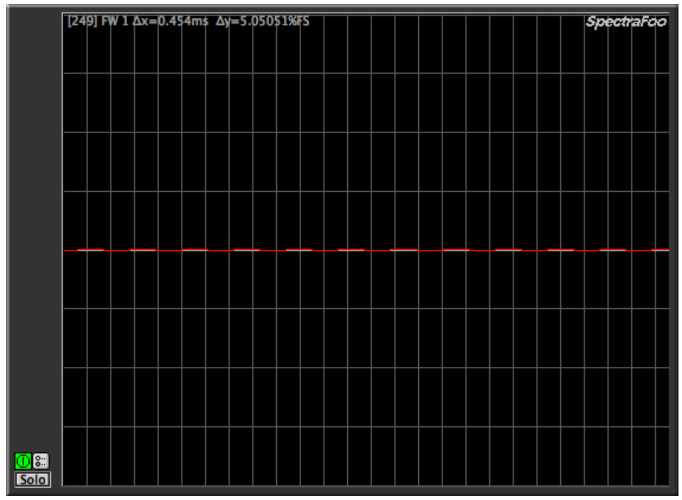


Figure 728: Phase Cancellation

Post

Means "after". Common uses:

- Post-fader: After the fader. A post-fader meter would show the level of an audio signal including the boost or cut of the fader.
- Post-insert: After the insert. A post-insert direct out would send wet audio.

PPM meter

Peak Program Meter. This type of audio meter features a fast response, allowing you to see peak signal level and catch potential overs quickly.

Pre

Means "before". Common uses:

- Pre-fader: Before the fader. A pre-fader meter would show the level of the audio before the fader, so the meter's display would not show boost or cut from the fader.
- Pre-insert: Before the insert. A pre-insert direct out would send dry audio.

See also *Mic Pre*

R

RCA

A two conductor connector, sometimes called a phono connector. RCA connectors use a two conductor (often coaxial) cable to carry unbalanced signals, and are wired with signal to the center pin and ground to the ring. Inputs and outputs are both on female (jacks) and interconnect cables are male to male (plugs). Extensions are male to female.

S

Scene

A "picture" of your MIOConsole3d setup. All your routing, fader levels, DSP plug-ins, etc. are saved so that they can be quickly recalled. This can be done by saving from MIOConsole3d using "File/Save" and "File/Open" or from your DAW using MIOConsoleConnect.

SCP

Satellite Computer Port: When 3d boxes are connected and controlled by the Host computer via MHLink Gigabit Ethernet, the USB port on each 3d device becomes a computer audio I/O port.

Any compatible computer or mobile device connected to the USB port becomes a "Satellite Computer" to the MHLink domain and audio master clock. Upon successful connection to the Satellite Computer, the USB audio I/O channels automatically become available to the 3d mixer under the "SCP" category in the MIOConsole3d routing interface.

Send

There are two kinds of sends used by Metric Halo:

- Analog: In the ULN-2 and ULN-8, the send is the physical output of the insert.
- Digital: In MIOConsole3d, a send is used to route an channel to buses other than the main bus. Sends are accessed via the inserts.

SNR

Signal to Noise Ratio; a measurement used to indicate how much a signal is corrupted by noise. The higher this number is in dB, the less noise will be added to your signal by the equipment. For example, a 120 dB SNR means that the gear's self noise is 120 dB below the audio.

SPDIF

Sony/Philips Digital InterFace, a standard for digitally connecting two devices together. SPDIF sends and receives two channels of audio, using a 75 Ω cable terminated with RCA connectors.

Stem

Also called a group or subgroup. A stem is a part of a mix that contains similar inputs. For example, a drum stem contains only the drums tracks in a song; this allows for easy mixing and processing of the drums. MIOConsole3d allows you to create stems, as well as mix the stems together for an overall final mix.

Subgroup

See *Stem*

Sum

To add or combine signals. For example, a summing bus combines the signals from several inputs into a single mix.

T

TOSLINK

An optical interface and cable designed by Toshiba to carry digital audio between devices. TOSLINK uses a plastic fiber to carry the digital information, and is either terminated in a semi-square plug (used on the 2882 and ULN-2) or a round "Mini-TOSLINK" found on some laptops.

TRS

Tip Ring Sleeve. A type of connector found on musical instruments, headphones, patchbays, and other audio electronics. TRS connectors are commonly 1/8" or 1/4" in diameter. Wiring schemes for TRS connectors are:

- Mono balanced: "Hot" or "Plus" on Tip, "Cold" or "Minus" on Ring and Ground on Sleeve
- Stereo unbalanced (headphones, etc.): Left on Tip, Right on Ring, Ground on Sleeve

Inputs and outputs are both on female (jacks) and interconnect cables are male to male (plugs). Extensions are male to female.

U

Unbalanced connection	A method of connecting audio equipment. A balanced connector usually has two pins per channel, signal and ground. Unbalanced connections are susceptible to noise from external sources. An unbalanced connection can also occur when a balanced connection loses one leg due to a faulty connector or cable; the connection is now susceptible to noise and will drop in level.

W

Wet	If a signal has been mixed with an effect, it is said to be wet. If there is no uneffected signal present, the signal is 100% wet. Many processors have controls to let you set the wet/dry ratio; a processor used on a send or aux is usually set to 100% wet.
Word clock (WC)	Used to synchronize multiple digital devices. Commonly used to connect a master clock to all devices in a studio to ensure that they are in sample accurate sync. Word clocks are connected via 75 Ω BNC cables, and must be terminated for proper operation. All Metric Halo interfaces feature self-terminating WC connectors.

X

XLR	A professional, locking connector that (usually) has three conductors and carries a balanced signal; Ground on pin 1, "Hot" or "Plus" on pin 2 and "Cold" or "Minus" on pin 3. Male XLR connectors are used for outputs and females for inputs.

Index

Symbols

A

B

C

D

E

F

G

H

I

J

W