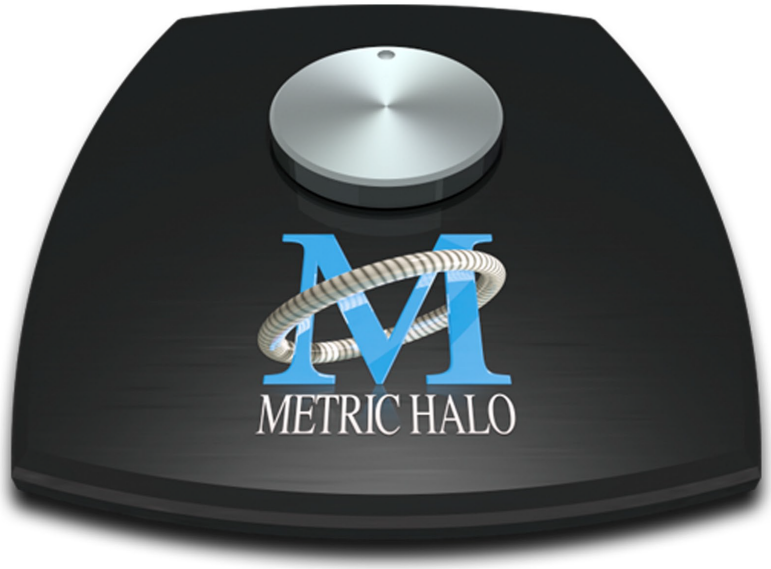

Metric Halo 3d Users Guide

Metric Halo 3d Operations Manual

Metric Halo

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MIOConsole, MIOConsole3d, MH Production Bundle, MH ChannelStrip, MH Character, MH Dirty Delay, MH HaloVerb, MH Multiband Dynamics, MH Multiband Expander, MH Precision DeEsser, MH Sonic EQ, MH SuperGate and MH TransientControl are our trademarks.

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1. Metric Halo 3d Start Here

Documentation

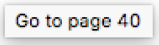
This PDF format Operations Manual covers all the basic functions of the MIOConsole3d application environment.

Links to related information appear throughout the manual as blue text.

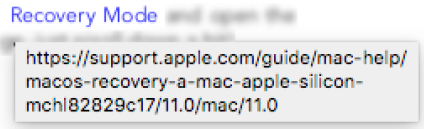
Clicking a thin blue text link will take you to a different location within the manual.

Clicking a bold blue text link will take you to an internet site via your default web browser.

The destination of any link can be previewed by hovering your mouse cursor over the link for a few seconds before clicking on it:



Page Link Tooltip



Web Site Link Tooltip

Modifier Key and Menu Item Conventions

Keyboard commands aid navigation within the manual and are used extensively as command shortcuts throughout all areas of MIOConsole3d. Apple keyboard modifier keys are represented throughout the documentation as follows:

⇧ = Shift	^ = Control	⌥ = Option	⌘ = Command

Key commands are shown in bold type, depicting the modifier and keyboard keys which must be pressed simultaneously to execute the described function:

- When viewing the manual in Apple Preview, the key command **⌘ [** takes you back to your previously viewed page, and **⌘]** to go forward again.
- In Adobe Reader, use **⌘ ←** to return to the previously viewed page, and **⌘→** to go forward again.

MIOConsole3d application menu items are represented italicized in the format *header bar menu* > *sub-menu* > *sub-submenu*:

- In the MIOConsole3d application, use the *Help* > *Open MIO Manual…* command from the MIOConsole3d menu bar to download and open the most recent version of the manual.

The fastest way to navigate the manual is to expose the Table of Contents in your PDF viewer sidebar (using the "View" menu in Apple Preview and/or the "Bookmarks" selection in Adobe). Each entry in the Table of Contents is a link to that section of the manual.

A black triangular icon to the left of a TOC entry indicates subsections within each entry. **Option click** a chapter arrow to expose all the subsections of that chapter at once to more quickly find the topic you are looking for.

Metric Halo 3d Overview

What It Is

Metric Halo **3d** is a blanket term for our third generation digital audio interface platform.

3d is an affordable, user-installable upgrade for every Metric Halo device ever made, and is the common foundation for all Metric Halo hardware moving forward.

Multi-core ARM SoC: similar to the one in your new tablet computer boosted with gobs of internal memory and a massive FPGA, but with an operating system painstakingly optimized specifically for audio. The platform is built to be self-configuring and future-proof, built on breakthrough technologies with much, much more to come...

- **MHLink**: High-speed audio data backplane utilizing Gigabit Ethernet to chain multiple ULN-8, LIO-8, 2882 and ULN-2 interfaces into a single sample-aligned audio über-interface, supporting up to 100 meter hops between connected devices using qualified cables
  - Self-configuring: attach a 3d box to a live MHLink chain, turn it on, and it's immediately online with all I/O ports ready for routing
  - Dynamically add or remove devices at the end of the chain without interrupting host DAW operations
  - 128 channels in + 128 channels out with 32 bit integer resolution at 192kHz sample rate
  - No added latency no matter how many boxes are connected or how far apart they are
  - Extremely low latency userspace audio device driver, with realtime data flow controls to optimize driver performance to your match your computer hardware and workflow demands
  - Hardware synchronized clock and system control
  - automatic DSP load sharing between MHLinked devices
  - Offline operations: Entire chains of MHLinked 3d devices can be pre-configured with all clocking, routing and processing saved as a preset within the hardware and recallable on-site without a host computer.

    Additionally, the MIOConsole3d application can open and modify saved state files without the boxes attached, applying the changes transparently the next time the boxes are connected.

- **MIOConsole3d**: control all 3d units from a single intuitive user interface
  - Integrates all 3d system management, from software and firmware updates to routing of all host computer, analog, digital, USB, MHLink and SMPTE I/O, busing, summing, Monitor and Cue mix control
  - No-nonsense routing engine:
    - Any input to any output on any box, direct, processed, bussed, mixed or all of the above at once
    - Time-saving routing tools such as 'selection strip' groups and cascaded channel assignment streamline the creation and modification of high channel count layouts
    - All routing selection menus include filtering by I/O port type, per 3d unit and via text-entry search; similar to the plug-in selection menus in most professional DAWs
  - Mackie Control Protocol (MCP) and EuCon remote control surface support
  - External MIDI Control Mapping throughout the Console environment, for all the stuff MCP and EuCon don't really handle
  - Metric Halo Production Bundle, Sonic EQ3d, Sontec MES-432D9D and MakeBelieve Studios processor family are embedded in the 3d hardware and share preset libraries with their AU, VST 2/3 and AAX plug-in counterparts

- **Comprehensive Monitor Controller, Cue and Talkback/Listenback facilities**
  - Archival-grade d/a conversion with digitally controlled analog gain stages direct to the speaker amps eliminate the need for expensive outboard monitor controllers while providing vastly improved sound quality and flexibility
  - Hardware relay mute output protection on analog line out channels (standard in mkIV)
  - Fully customizable Monitor and Cue outputs support mono to 9.2.6 channel widths
  - All monitor and cue output channels have their own processing Graph for building custom crossovers, speaker delay and eq, headphone HRTF correction, and much more...
  - External Monitor Controller Source inputs for direct monitoring of external inputs at channel widths up to 9.2.6 (Atmos™ and other immersive audio renderers a specialty)
  - Independent save and recall of all monitor output configurations to any MIOConsole3d mixer layout

- **Satellite Computer Port**: more than one DAW to connect? No problem.
  - Every 3d device includes a USB UAC2 Class Audio port on the rear panel which functions as an additional computer audio interface.
  - When 3d devices are MHLinked, the USB-C port on each box can route up to 48 channels to and from any DAW, VI rig or NLE running on its own computer
  - Cross platform: capable of interfacing Mac, Windows, Linux, iOS and Android devices.
  - No software driver required. The USB UAC2 driver is embedded in the 3d main board firmware.
  - Perfect for external immersive audio renderer returns

- **EdgeBus™ Expansion Port**: yet more extensibility
  - EdgeCard™ expansion modules offer additional AES, SPDIF, MADI, ADAT and MIDI input and output ports to every 3d device.
  - Per-port autosensing of input sample rate and audio data format (including S/MUX).
  - MADI frame rate and channel output formats are independently configurable per MADI port, maximizing compatibility with all current and legacy MADI spec hardware.

3d: Need To Know Terms

The MIOConsole3d application manages all aspects of the unified 3d audio interface system.

MIOConsole is designed to run alongside your DAW, and is extremely lightweight in cpu requirements. All routing and signal processing in MIOConsole mixer strips is running on the 3d boxes, not in your computer.

Audio processes can be freely placed in the signal path before your DAW inputs, after your DAW outputs, as external process loops within the DAW, or all at once, including Cue mixes and multichannel Monitor Control.

The multi-box nature and nearly unlimited routing available with 3d and MHLink obviously adds crazy amounts of power and flexibility, and it is MIOConsoles' job to keep the crazy under control. So, even though many devices are operating as one, each 3d device with it's own set of analog, digital, USB and EdgeCard audio ports must be clearly addressable both as a unit and as part of the whole system.

To that end, MIOConsole3d uses a few organizational terms to be aware of:

Domain

The *Domain* is any single box or chain of MHLinked boxes. Domains .
Each domain is recognized by your host computer as an independent physical audio device with its own clocking, I/O, Mixer, Session record/playback engine, Monitor and Cue Controls setups and routing maps. The host computer Sound preferences, Audio MIDI Setup and MIOConsole3d each domain by the connection type (MHLink or USB) and the Root box serial number.
Note: While MIOConsole3d does support management of multiple domains, there aren't many scenarios where it provides an advantage over a single extended MHLink chain. The feature is included primarily because we couldn't think of a good reason to turn it off, and Metric Halo users tend to push boundaries in fascinating ways. Feel free to experiment and let us know what you find via the MIOConsole3d *Help > New Feature Request...* application menu.

Host

...as in *Host computer*. The Host is the computer the 3d boxes are connected to and is running MIOConsole3d. This is also the computer that is running your primary DAW.
The MIOConsole audio routing menus use the term *Host* for routing audio specifically to and from the main host computer and your primary DAW.
The term *SCP* (*Satellite Computer Port*) is used in the routing menus to address secondary computers connected to the USB ports on each MHLinked 3d box.

Think "Host = Domain" and "SCP USB = per box". The host computer sees only the Domain, whereas MIOConsole3d sees all the ports in each 3d device that makes up the Domain.

Root

The Root box is the 3d device directly connected to the host computer.
In a multibox MHLink domain, the root box manages all audio and system control communications between the host computer and all MHLinked boxes connected to it. Domains are identified by the model and serial number of their root box.
Note: USB Root boxes (3d boxes connected to your computer via USB) operate as a single box only. 3d devices connected to the MHLink ports of a USB Root may be shown in the Status Pane for identification purposes, but will not be available for routing audio.

System Status Pane

The System Status Pane is an information display and control interface at the left edge of the main MIOConsole3d window (shown on the next page).
At the system level, the Status Pane displays the current version of MIOConsole3d and the MHLink driver, the firmware version of each connected 3d device, and alerts you when new versions become available.
Sample rate, clock master and external clock source controls are found in the Status Pane Domain header.

MHLinked units are listed in their connection order from the host computer; the Root being unit #1 at the top of the list, the next in the chain is #2, and so on.
Unit Status Displays identify each box's model, serial number and firmware version, ethernet and USB port status, USB I/O channel count, and EdgeCard type (if one is installed).
The Unit Status **Digital...** button opens the Digital Audio I/O port controls and sync status window for all physical internal and EdgeCard digital I/O for the unit.
^ **click** a Unit Status display to open a contextual menu with the following commands:

- *Identify Box* makes the front panel lights of that unit dance for a few seconds
- *Set Number of Channels for USB...* opens the USB I/O Configuration window that the unit
- *Delete Offline Unit* is active only when the selected unit is offline. The command is useful when replacing a physical unit or otherwise reconfiguring the boxes within a domain

Routing Model

It is important to fully understand that, between each LIO and ULN-8 supporting not just 8 channels of analog and AES I/O, but up to an additional 48 channels in and out via USB *and* the EdgeBus expansion slot supporting up to 256 channels of I/O (with the MADI Copper 4x EdgeCard). That's per box, with each box supporting a different EdgeCard.

With MHLink interfacing 128 channels to and from the host computer and MIOConsole3d aggregating a dozen boxes or more into a single unit, including full support to directly route any host or hardware input to any output on any attached device (and vice versa) *plus* the ability to dynamically change out devices on the fly, a static routing grid is simply not going to work, no matter how many tabs there are. There would have to be thousands of possible patch points represented at each axis, and that is not even accounting for mixer strip Group and Aux/Cue bus routes, Pre- and Post-Insert Direct Outs and I/O Inserts, and Monitor Control I/O.

So, rather than artificially limit what the system can do, we decided to make the computer environment actually work *for us* for a change.

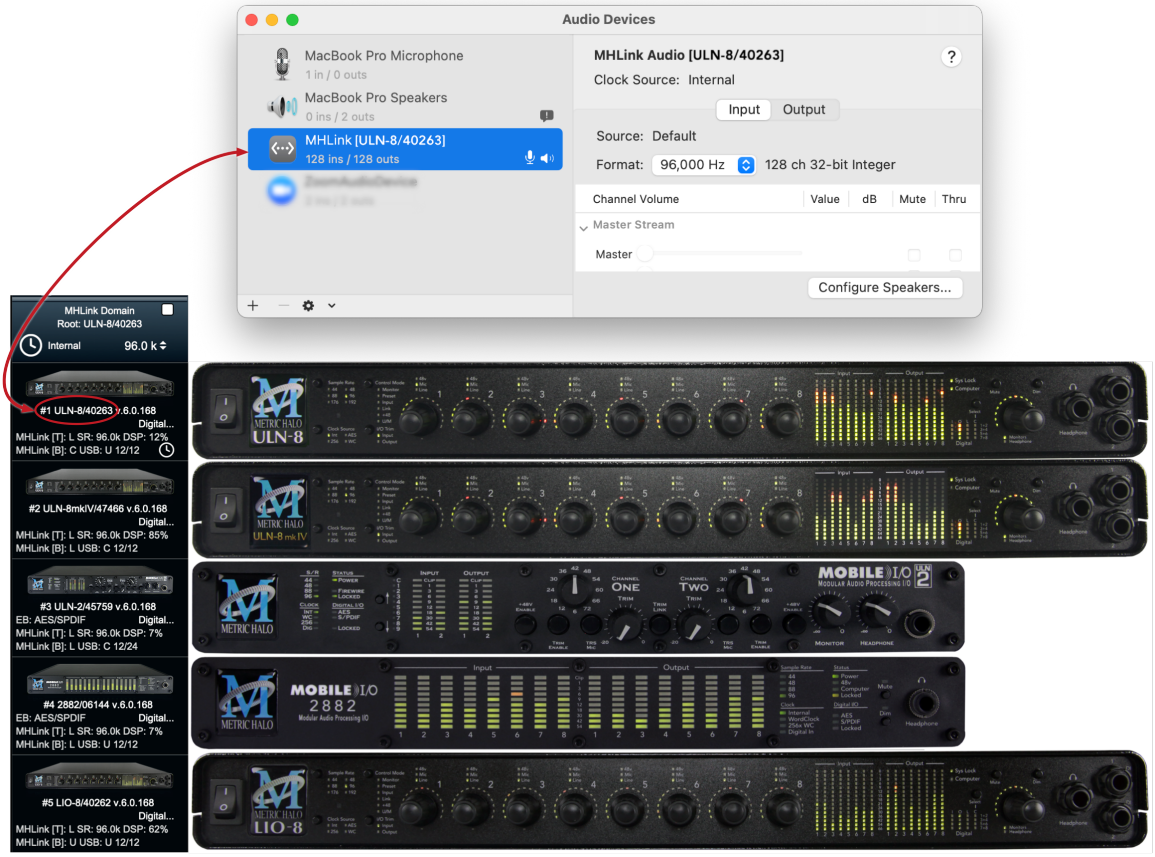
Latency Management

MHLink: How it works

In practice, MHLink is extemely simple: The computer is connected to the **Root** box - the first box in the MHLink chain. All other 3d boxes are connected, as links in a chain, to the Root box.

The Root box manages all communications with the host computer and acts as the master sync and system coordinator for all downstream MHLinked boxes.

Any 3d device can serve as a root box... no configuration necessary.



When connected, the host computer OS registers the Root box, including its serial number (shown at the top) and makes it available as an audio device.

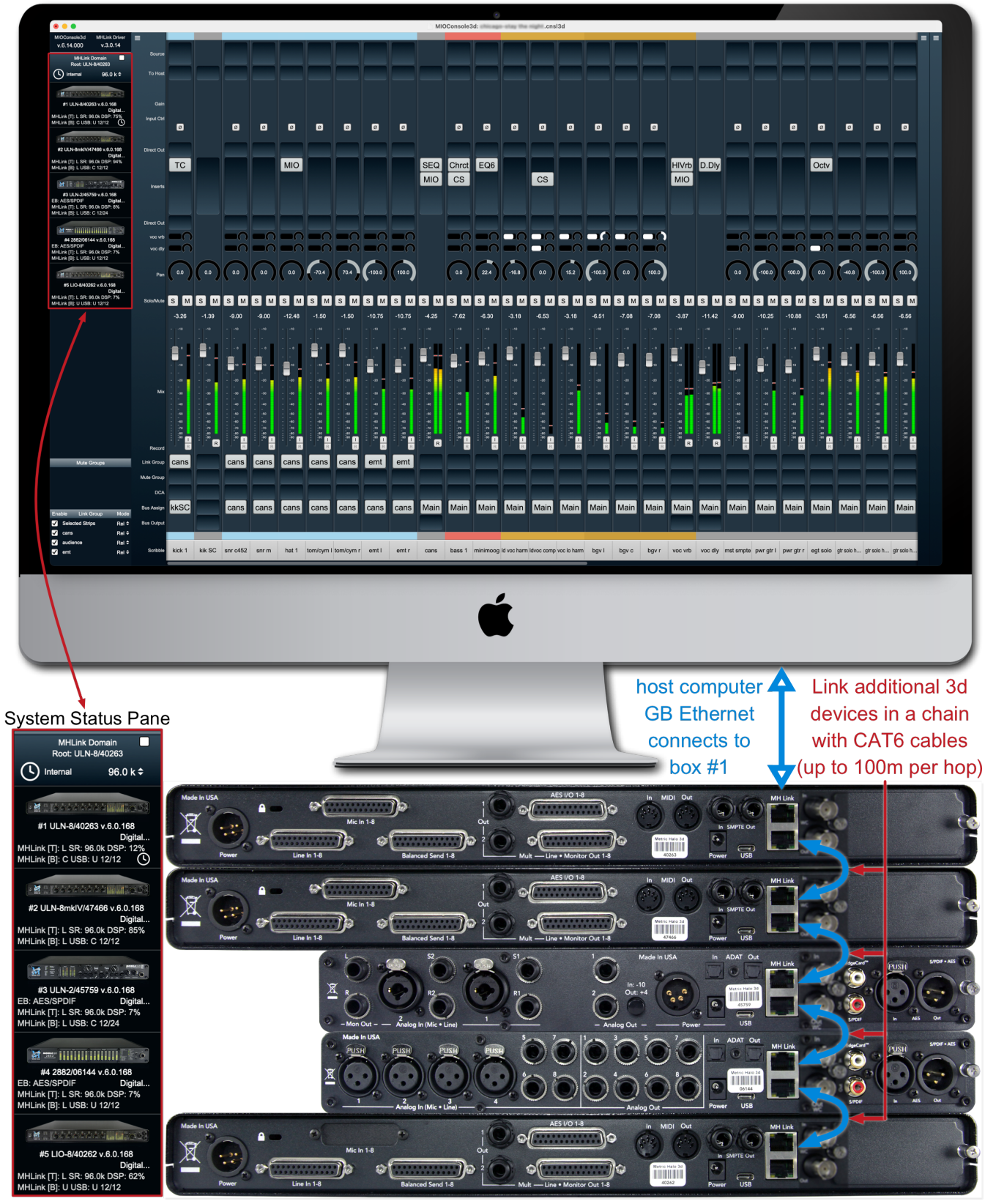
The MHLink System Status Pane from the MIOConsole3d application reveals all the MHLinked boxes, with the Root box at the top of the chain and all of the connected in order below.

Since the host DAW is only communicating with the Root box, you can add or remove 3d boxes at the end of the MHLink chain live without interrupting any DAWs' communications with the Root box. Boxes removed (or turned off) show as -*Offline*- until you reconnect them, and boxes added to the chain sync up and add their DSP horsepower and I/O ports to the system automatically within a few seconds.

Adding boxes does not add system latency; ten or twenty boxes act and respond the same as one.

Thanks to the Gigabit Ethernet backplane connecting everything, boxes in the MHLink chain can be up to 100 meters apart with no effect on performance. So think like three boxes at front of house, another six on stage, a couple on the wings and another at the broadcast truck. Or maybe two in the control room, four in the live room, one in each booth... it all still acts like a single box on your desktop.

Here is how the system looks with MIOConsole3d:



System Requirements

Host Computer Compatibility

The MIOConsole3d control application and the MHLink Driver are fully native on Apple Intel and Apple Silicon Macs running macOS 10.9 and later. The MHLink Driver runs in the macOS user space, eliminating issues concerning kernel driver compatibility and T2 chip security.

The USB-C port on the rear panel of every 3d device is a USB UAC2 Class Audio peripheral port, and is compatible with most modern Windows and Linux PCs, Macs, iOS and Android devices. See the **3d USB UAC2 Class Audio Port Compatibility** section below for more specific details.

DAW / Software Compatibility

All Core Audio compliant software is compatible with both MHLink and USB audio, including:

Cubase , Digital Performer , GarageBand , iTunes , Live , Logic , Nuendo , Pro Tools , Studio One , Reaper , MixBus / Ardour , and hundreds more…

Some older streaming / conferencing programs have been known to experience difficulty addressing high channel count audio devices. For best compatibility in these cases, set the MHLink Core Audio channel count in the MIOConsole3d **I/O** menu to 2 channels, then relaunch your application.

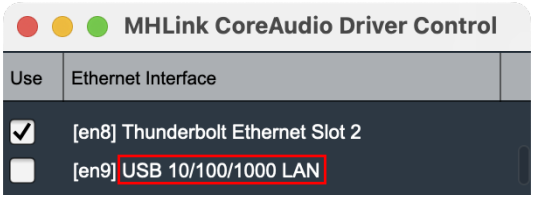
MHLink Gigabit Ethernet Audio Backplane

- All Apple Silicon Macs, and any Intel Mac running Mac macOS 10.11.5 or newer
- a dedicated internal or peripheral Gigabit Ethernet NIC (MHLink connection through IP switches and routers is not supported)

  Gigabit Ethernet Adaptors supported:
  - Apple Thunderbolt Gigabit Ethernet Adapter. Use the Apple TB 3/4 (USB-C) to TB2 Adapter connected to the Apple TB to GB Ethernet Adapter.
  - PCIe Gigabit Ethernet Adapter
  - Third-party Thunderbolt Gigabit Ethernet Adapter (e.g. as part of a dock)
- Cat 5e or higher interconnect cable. Cat 6 or higher recommended for MHLink runs over 75 meters.

USB-based Gigabit Ethernet Adapters are not supported for MHLink.

To verify whether a USBC-connected dock is actually using Thunderbolt for its Ethernet transport, go to the MIOConsole3d I/O menu and select Show MHLink CoreAudio Driver Control Window.



All the available ethernet ports on your computer are listed at the top. If the ethernet NIC you were hoping to use for MHLink is reported as USB device (as shown higlighted in red above), uncheck the selector box. The MHLink Driver will ignore that port and you can use it for regular public IP networking.

Many users have reported issues with USBC cables and expansion docks marketed as “Thunderbolt compatible” when in fact the data transfer is all USB-based.

USB UAC2 Class Audio Port Compatibility

The rear panel USB-C port on your 3d box is classified as a USB peripheral port that will only be recognized by a computer, tablet or smartphone with an operating system capable of identifying and transporting USB UAC2 audio streams.

3d USB-C ports can not recognize other USB peripherals, and do not supply power.

Sharing a USB port with other peripherals is not supported. In order to ensure stability of USB audio connections, always use a dedicated USB port for connection to the 3d device.
Dedicated USB ports connected via Thunderbolt (as in a true Thunderbolt dock) are supported.
However, so-called "Thunderbolt-compatible" peripherals are USB-based, and are not supported.

- macOS version 10.10.x and up and iOS devices version 11 and later perform consistently well with 3d 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.

  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 even crash.

  Reducing the USB port I/O count to 8 channels or less from the MIOConsole3d device Status Pane 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.
- UAC2 driver code for the Linux kernel has been available since 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, but may not be supported by all products. Consult your Android device operations documentation for information on UAC2 Class Audio support.

> **Note**: When changing the 3d device sample rate or I/O channel count often requires that the USB port of the host computer be reset before the operating system and applications running on the host will recognize the change.
>
> Usually just quitting the DAW software, making the change to the SR and channel count, and then relaunching the DAW is the

Remote Control Surface Hardware and software

MIOConsole3d features user-defined ad-hoc mapping of external MIDI controllers to virtually every aspect of the the system, complemented by built-in mixer, monitor control and Session transport control with EuCon™ and Mackie Control Protocol-compatible hardware control surfaces.

- **External MIDI Controller Mapping** is supported from any MIDI control device available in AudioMIDI Setup MIDI Studio and/or Network MIDI on the host computer. See External Controller Support for details.
- **EuCon™** hardware controllers require the Avid EuControl management software to be running with a separate USB or Ethernet interface to the host computer. Avid Control is supported via host computer WiFi. See EuCon in MIOConsole3d for details.
- **Mackie Control Protocol** control surfaces are supported via a separate USB or Ethernet interface to the host computer. MCP-compatible wireless applications are likewise supported via host computer WiFi. See Mackie Control Protocol in MIOConsole3d for details.

Installation & Registration

First time installation - MIOConsole3d and MHLink Driver

All software, driver and firmware updates are installed from within the MIOConsole3d application.

Before launching MIOConsole3d for the first time, please review the system requirements on the preceding pages for 3d MHLink and USB audio connections.

Installation walkthrough:

- Connect your 3d unit to either a free USB port, the internal gigabit ethernet port or a Thunderbolt gigabit ethernet adapter.

  Power up the box.

- Download the MIOConsole3d control application from the MIOConsole downloads page.

- Unzip the MIOConsole3d application, drag it to your Applications folder and launch.

  If MIOConsole opens with visible mixer strips, that means you have a 3d unit connected at a USB port. If you wish to use MHLink, disconnect any USB connections to 3d devices before installing the MHLink Driver.

  If MIOConsole appears with blank windows, it means you need to install the MHLink Driver so MIO-Console and your host computer can communicate with the 3d boxes via gigabit ethernet.

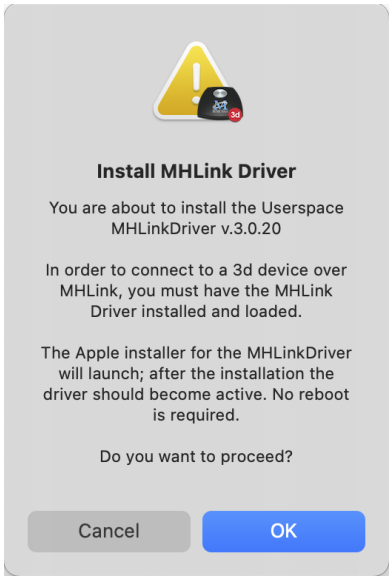
- Install the MHLinkDriver:

  The Software Version header in the very upper left of the MIOConsole3d main window will show the current version number of the MIOConsole3d application, and the term " <Not Loaded> " in place of the MHLink Driver version number.



MIOConsole3d - First launch status (prior to MHLink Driver installation)

Click the orange " <Not Loaded> " text to launch the driver installation process:



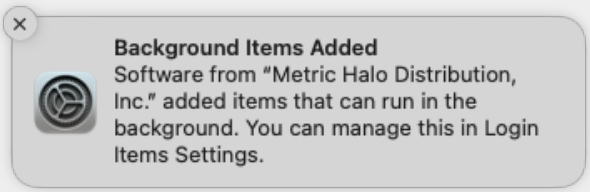
Click OK and follow the installer instructions.

The MHLink Driver version number will turn from orange to white after a few seconds, confirming that the new driver is installed and running. Any connected 3d devices will now appear as a domain in the MHLink Unit Status fields below the version header.

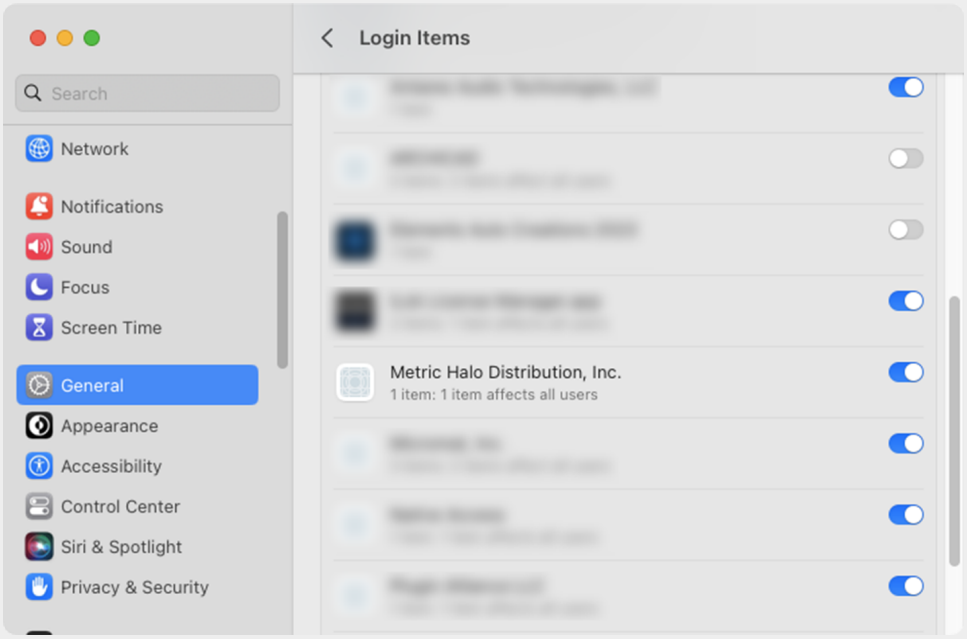
The 3d box directly connected to your computer will be #1 in the System Status Pane Unit list, identified as the Root in the Domain Header. 3d devices connected to the root will be listed as #2, #3, and so on below, by their place in the MHLink chain.

If you haven't already, connect and power up any 3d boxes you wish to add to the MHLink chain. They will immediately connect and appear in the MIOConsole3d window, ready to play.

macOS Ventura note: During the above procedure, macOS Ventura users will see a notification pop up announcing that the MHLink Driver has been installed with permission granted to run and connect your computer audio services to MHLink.



No further action is required on your part, but clicking on this notification will open the System Preferences > General > Login Items pane, where you can confirm that, indeed, the driver is installed and enabled as shown below.



Firmware Installation

New versions of MIOConsole3d may also include updates to the 3d hardware devices themselves.

Firmware updates will be indicated as orange version number text in the Unit Status display of any connected 3d box which requires the update.



MIOConsole3d - a new firmware update is available!

The firmware update will be applied to all boxes in the selected domain. An activity window will appear with progress bars showing the firmware installation for each device being updated.

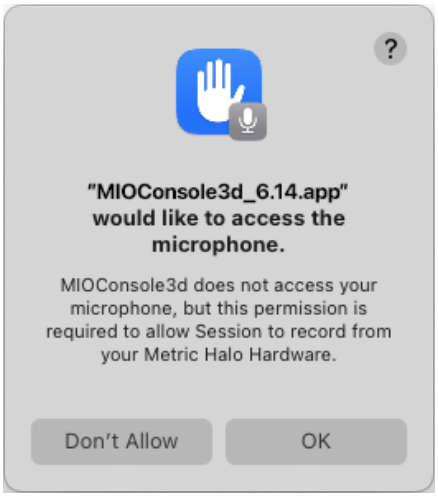
When the installation process window closes, all boxes must be powered down and rebooted. The boxes will automatically re-appear in the System Status Pane when they come back online.

Tip: If some of your 3d boxes are located across the room, in other rooms, or are otherwise just a pain to get to, you can reboot all the boxes in your domain from MIOConsole3d *Utilities > Reboot Attached Units* menu command.

macOS: Permission to Use the “Microphone”

When you first start running audio into MIOConsole3d (or any Mac audio application), macOS puts up a dialog asking you if you want to let the application access the “Microphone”.

Click OK.



definitely click “OK”

If you do not allow access, your audio applications will still work fine, all meters properly active and appearing to play and record as usual - but incoming audio will be stripped of data before in can be recorded. In other words, you have audio files all filled with silence.

This permission was added as a security measure in macOS 10.14 (Mojave) to help prevent unauthorized data into the computer environment posing as audio without the users knowledge.

MIOConsole3d manages audio passing both to and from your host computer and also includes a built-in recording engine, so this security measure applies.

As a troubleshooting measure, you can verify your “Microphone” permissions in the *System Preferences > Security + Privacy*, *Privacy* section and select *Microphone* from the list. Select the checkboxes for MIOConsole, your DAWs, streaming apps and any other audio applications you need to run audio through.

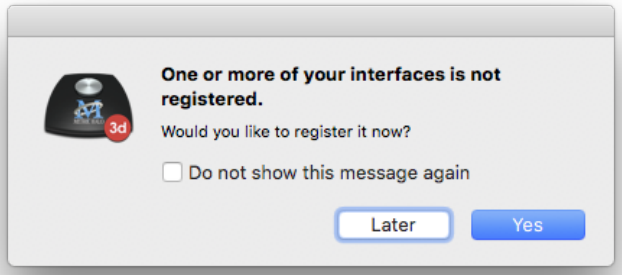
Note: “Microphone” permissions in macOS Ventura and later are located at *System Preferences > Privacy + Security, Privacy > Microphone* pane.

Registration

Registering your interface

> **Internet connection required**
> Your computer must be online with a live internet connection to register MH 3d hardware.

When you launch MIOConsole3d with an unregistered interface attached, you'll see the following dialog:



Registration Message

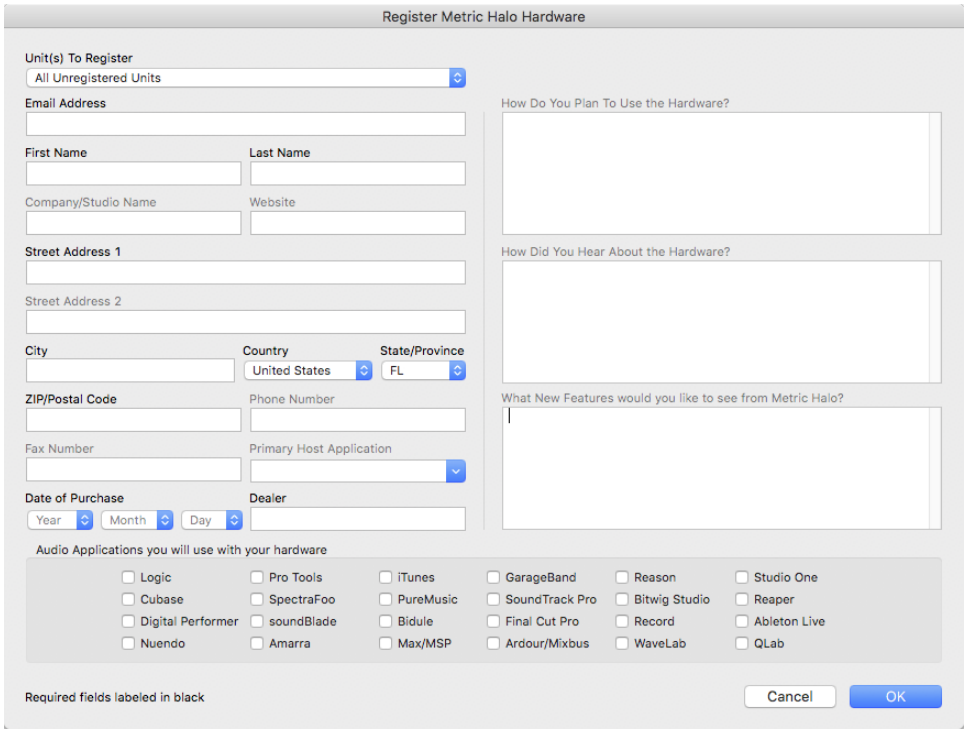
Your options are:

- *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**.



Preferences: Discovery: *Check for registered units* selector

Click OK, and the registration window will open:



Registration Window

In the upper left corner is a menu to select which interface to register (if more than one unregistered unit is connected):

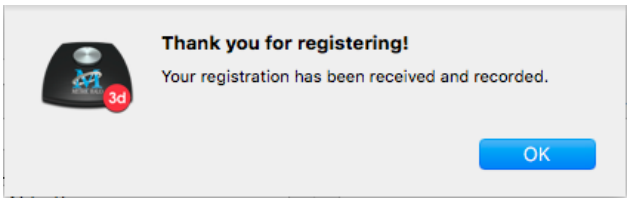


Registration Selection

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:



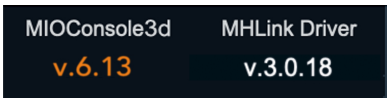
Registration Successful

You will receive an email from confirming your registration. This email will also include information on support, help resources and other useful links.

Software Updates

MIOConsole3d automatically checks for version updates at launch.

When a new version is found, the version header in the top left of the MIOConsole3d main window will show the Console application version number as **orange text**.



MIOConsole Version Header: a newer Console is available and no MHLink Driver is installed

Click on the orange text. A "Release Notes" window will open with detailing the new features and bugfixes of the new version, including new MHLink Driver and 3d device firmware info, if any.



MIOConsole Installer: update release notes

Clicking the blue "Download Update" button at the bottom of the window will download and unzip the new version, then prompt you to quit MIOConsole.

When updating, once you have downloaded the newest MIOConsole version, quit MIOConsole3d, replace the old version in your Applications folder and launch as usual.

New versions of the MIOConsole3d application often contain further updates to the MHLink Driver and 3d device firmware.

First Listen: Default I/O Routing

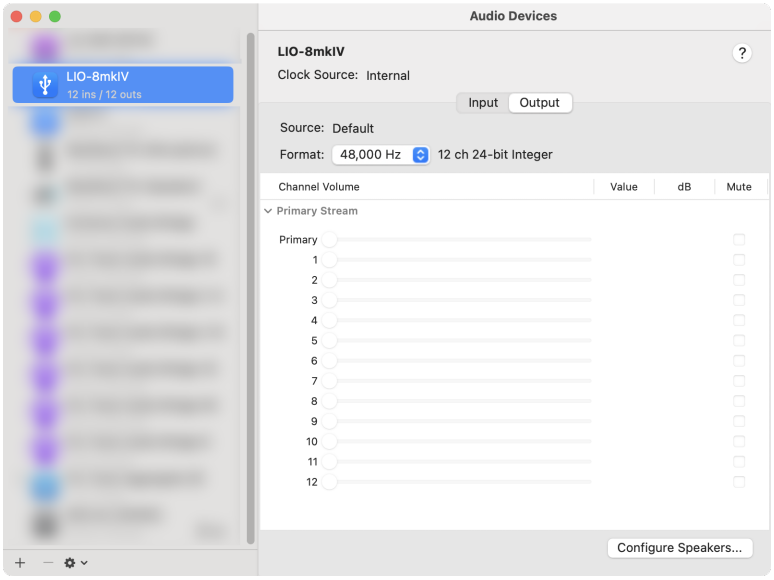
All 3d devices are pre-configured to get audio to and from your computer immediately upon first power up without having to launch MIOConsole3d.

USB (instant gratification)

Connecting your new 3d box via USB is the fastest way to get audio through the box and into your DAW. Literally connect it and power it up.

As with any audio device using USB, dedicating a USB port on your computer motherboard is the best way to ensure weirdness-free operation.

Plugging in a 3d box via USB and turning it on, you will see it immediately appear in the Sound / Audio MIDI Setup preferences windows.



default USB connection (LIO-8mkIV shown)

- The default sample rate is 48kHz, adjustable from you DAW or OS sound preferences.
- The default I/O channel count is 12 channels in and 12 out. 12 channels in & out is the maximum for operating at all sample rates from 44.1kHz to 192kHz
- The default sample rate is 48kHz.

- **Playback from computer:**

  Host computer output channels 1 and 2 will play audio from both the headphone monitor jack on the front panel and the analog Line Out channels 1 and 2 on the rear panel.

  You can use your host computer Volume Up and Volume Down controls to set the line out volume. Additionally, you can use the Volume knob on the front panel. Pressing the volume knob toggles the Volume, Mute and Dim controls to operate the monitor line outs or the headphone output.

  As with any newly connected audio device, it is safest to turn the volume control down before starting playback, then bring it up slowly to the desired listening level to get a feel for the control.

- **Inputs to computer:**

  Analog inputs set to *Line In +4* mode are routed to host computer input for immediate recording into a DAW. In other words, Line inputs 1-8 on the box will automatically show up on inputs 1-8 in your DAW.

MIOConsole Overview / QuickStart

When launched, the MIOConsole3d application scans and registers all available 3d devices connected to your host computer and presents you a clean, intuitive interface with total control of the combined I/O and DSP processing power of every box.

At first launch, MIOConsole3d opens with three windows visible. The largest is the main MIOConsole3d Mixer window. To the right are the Monitor Controller and the Cue Controls.



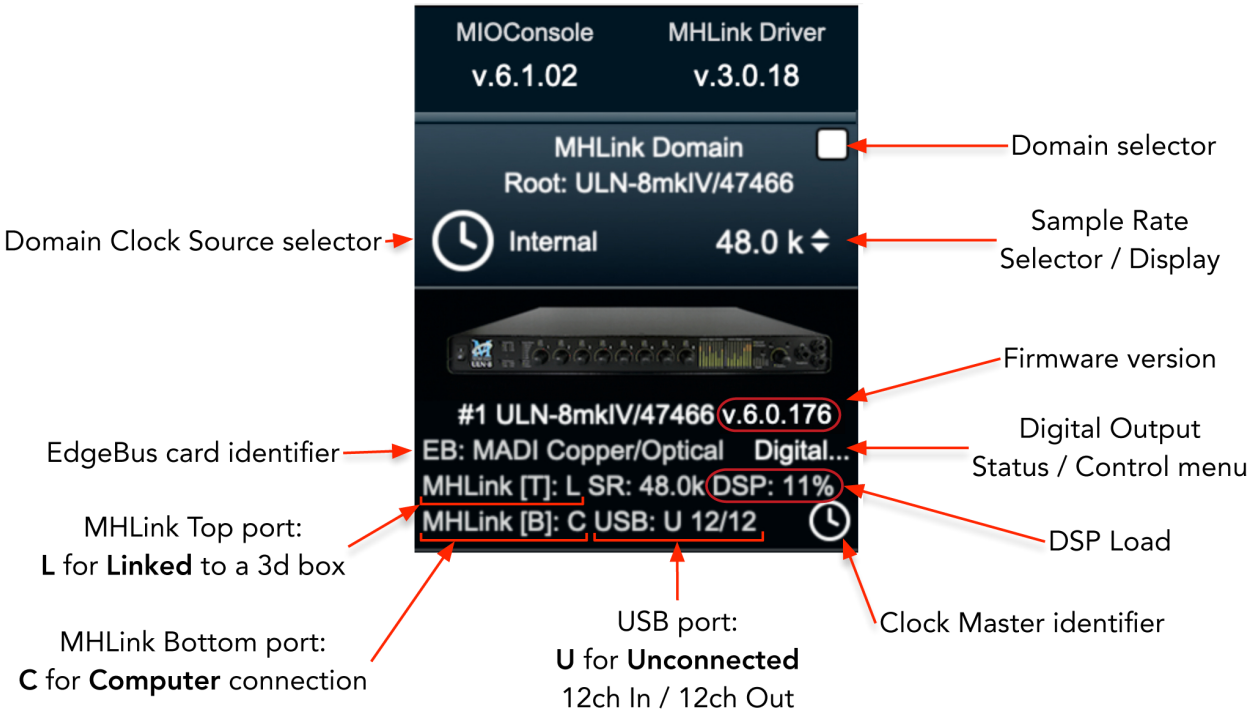
1. The **System Status Pane** is the system setup and control interface for all domain and box-level digital I/O operations.
2. **Mute Groups** houses trigger buttons for Mute Groups assigned to Input and bus strips in the mixer.
3. **Link Group** provides enable/disable and link behavior modes for parameter linked mixer strip groups, including the *Selected Strips* semi-automatic link group.
4. The MIOConsole **Mix Desk** provides a familiar and logical interface for all I/O routing. The ability to dynamically combine boxes into a single operational unit, each with its own expansion port very
   • Input strips (mixer strips with **Source** input routing at the top of the strip) are assigned for all incoming audio. Input strips include semi-automated direct routes to the host computer, with a full complement of direct and bus outputs for direct passthrough alongside realtime inline Aux/Stem/Cue mixing. Multiple Input strips can source the same audio input as desired for maximum flexibility.
   • Aux mixer return strips, Group Bus strips and the Main mixer output bus are fed from Input strips or other bus strips. Bus strips provide their own semi-automated direct routes to the host and multiple direct out options.
   • All bus strips are routed as the Monitor Controller / Cue Control sources behind the scenes by default. See the Configure Mixer section for more details.

Input and Bus strips are independently configurable as mono, stereo or multichannel up to 8 channels wide.

5. The **Monitor Controller** is an independent entity which taps the bus outputs from the mixing engine behind the scenes, simplifying mixer routing.
   • Additional Monitor Controller sources may be assigned from any input or group of inputs, bypassing the 3d mixer entirely. This feature has been tailored specifically for monitoring immersive audio renderer returns in all channel formats (currently up to 9.2.6 Atmos™).
   • All Monitor outputs include their own independent signal processing stages with all the tools required for time alignment, bass management, crossovers, and room acoustics compensation for each speaker channel.
6.

You will find the System Status Pane at the left of the main MIOConsole3d window.

The System Status Pane



System and Unit Status display (ULN-8mkIV shown)

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 (a ULN-8mkIV in this case) 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-8mkIV icon is the box ID label and current firmware version.



System and Unit Status display (ULN-8mkIV shown)

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.

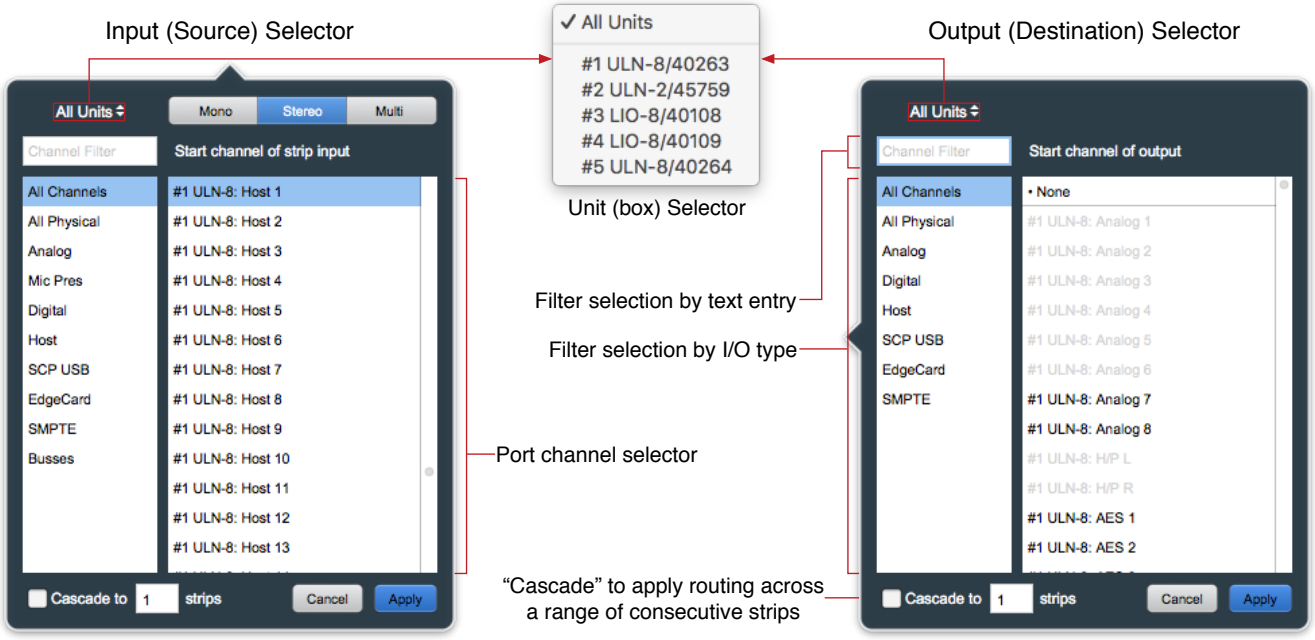
You'll see a pop-up with that device's type and serial number, with Clock Input Lock status for each of the built-in rear panel AES channel pairs.

This window will also identify any digital audio ports present on an EdgeBus™ (abbreviated "EB") I/O card installed in the rear panel expansion slot, with Output Mode settings appropriate to each digital port format. See the Digital I/O Status / Control Menu section for details.

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.

Routing



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-8mkIV 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.

Intro to SCP USB

Every 3d device includes a single USB-C port on the rear panel as a secondary, multi-platform computer interface.

This port supports the UAC2 class audio driver built into all USB2-compliant devices. Most modern macOS, Windows 10, Linux, iOS and Android devices are capable of recognizing and hosting 3d boxes as USB peripherals.

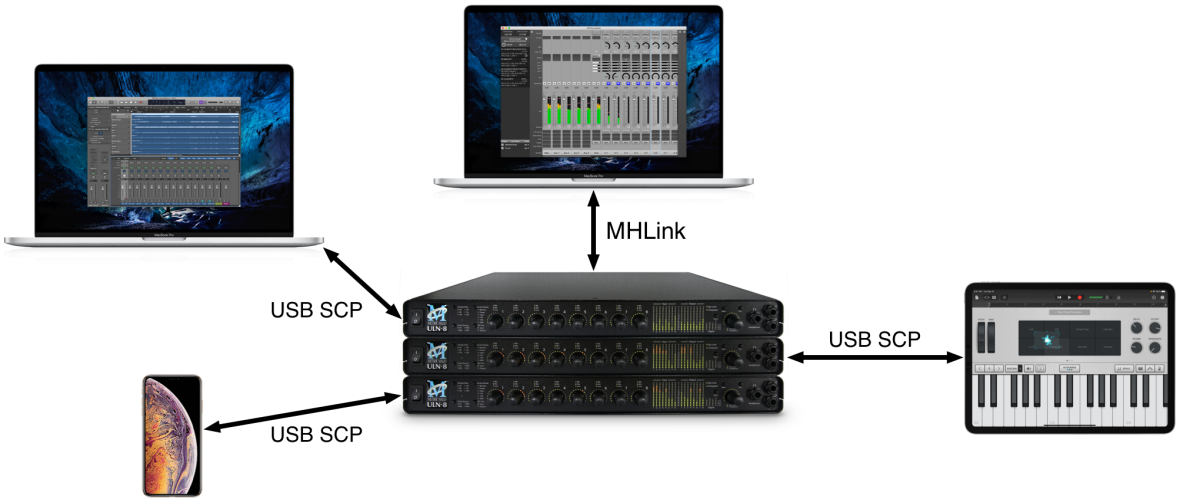
However...

What becomes of those USB ports when your 3d boxes are all members of MHLink chain? Well...

Have you ever wanted to add an iPad to your main studio system (like, just plug it in)? How about dedicating an older computer to running a VI or a sampler? What about when a client or collaborator comes in with their laptop or tablet and wants to plug their DAW into your system?

With MHLinked boxes and SCP, you can connect any of these devices directly into your MHLink domain without having to add another interface just for that purpose. One cable, no driver, no hassle.

3d supports explicit routing to and from the USB connector of every box in the MHLink chain. When used in this way the USB ports are referred to as Satellite Computer Ports (hence SCP because, acronyms).



SCP connections

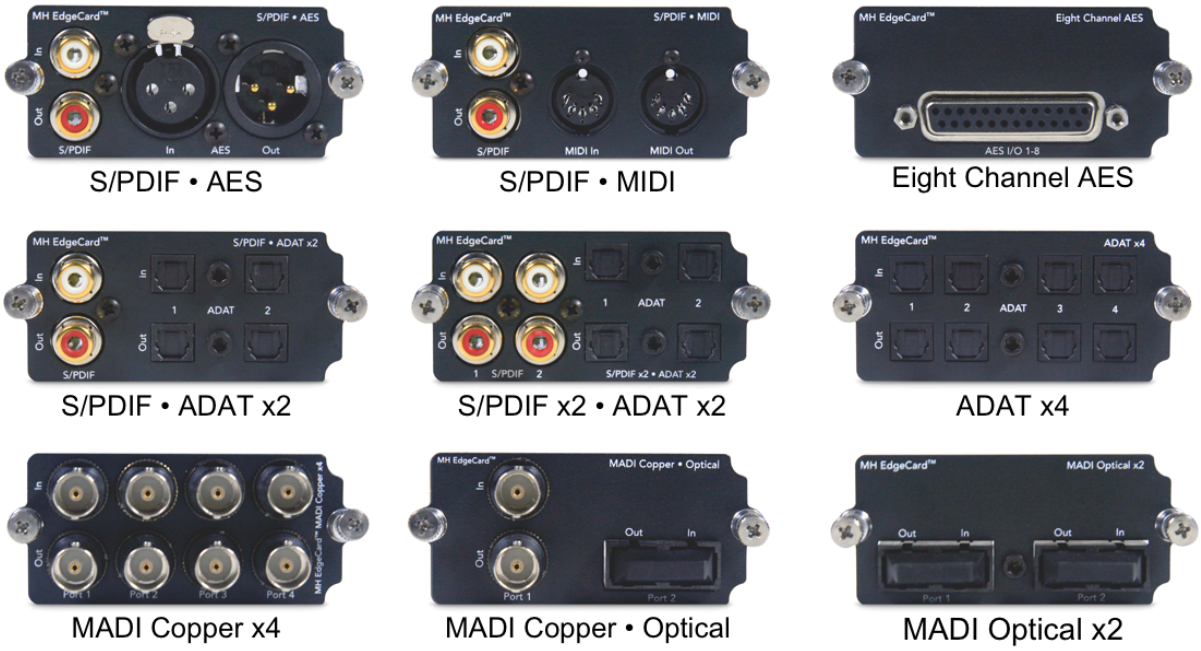
You can use the SCP functionality to integrate external DAWs running on their own computers into your MHLink mixing engine. All USB source and destination routes for every MHLinked box are available in the MIOConsole3d routing selectors.

You can transport audio and MIDI to and from mobile devices (and embedded devices) that don't allow the installation of custom drivers, as long as the devices provide built-in audio and/or MIDI class support. At present iOS devices (including iPhone and iPad), Android devices (phones and tablets) and AKAI MPC have the appropriate class drivers included.

EdgeBus™ expansion port

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™.



EdgeCard™ expansion options

The current crop of EdgeCards (shown above) can provide anywhere from 4 to 256 additional channels of digital I/O per card. If you have more than one 3d unit in your MHLink domain, every unit can host its own EdgeCard module.

Any channel 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.

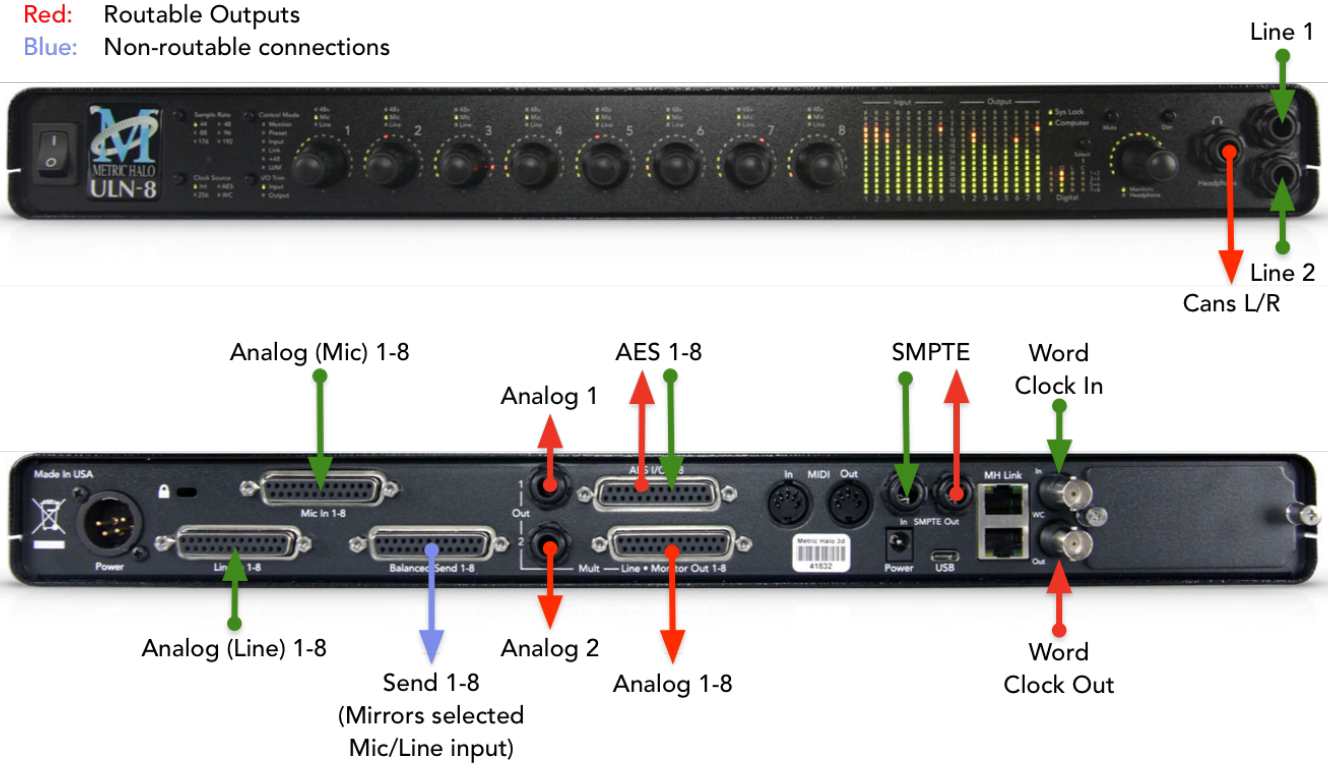
Input data format for each physical digital format is automatically sensed and adapted.

All sample rates up to 192kHz are supported. ADAT optical S/MUX, and MADI native and S/MUX at both 56 and 64 channel frame formats are fully supported.

Physical Connections

ULN-8 and LIO-8 (all models)

Green: Routable Inputs
Red: Routable Outputs
Blue: Non-routable connections



ULN-8 Physical Audio I/O

ULN-8 and LIO-8 legacy and mkIV models are all physically identical aside from the absence of mic preamps in the LIOs. A LIO-8 can be upgraded to a ULN-8 by installing the ULN-R mic preamp package, and any legacy unit can upgraded with the mkIV upgrade package. For simplicity this section uses the term "ULN-8" to apply to all of the above models.

Analog Audio Connections

The analog I/O connections on the ULN-8 have been engineered to support both balanced and unbalanced connections with a wide range of operating levels and circuit impedances. The microphone inputs have an impedance of 3.3kΩ, and the line inputs are 10kΩ.

All multichannel audio connections are made using industry standard DB25 connectors, which are pin-compatible with cables using the Tascam/Digidesign DB25 Pinout wiring standard.

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 issues. If you cannot utilize balanced interconnects, use the connection schemes shown in the link above to maximize performance.

ULN-8 DB25 cables should be wired pin 2 or Tip hot (for TS/TRS). All 1/4" connectors on the front and rear panels are wired Tip hot. The headphone connector is wired Tip/Left, Ring/Right, Sleeve/Ground.

On input, at line level, it is sufficient to simply use standard unbalanced (TS) connections. .

Front Panel Direct Inputs (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 mixer strip analog headamp 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 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 AES-3 digital audio over copper-based connections. 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 units 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. Feeding a legitimate word clock source to the WC Input BNC.
2. Supplying a legitimate AES clock source into any AES digital input channel pair.
3. Sending a legitimate digital audio source signal to an EdgeCard digital audio input port.

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 separate SMPTE input and output ports dedicated for linear timecode use. These are straight audio I/O, set up to isolate SMPTE square waves from the primary audio channels. 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 lock to or generate SMPTE LTC at this time. The SMPTE ports are for reading, recording and playing/passthrough of SMPTE streams alongside multitrack audio.

Power

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.

One of the ULN-8s 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 rear panel are a 2.1mm (center positive) coaxial power connector, and a 4-pin XLR connector (Pin 4 Hot, Pin 1 ground). 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.

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, to avoid the spark just connect the power connector to ULN-8 before connecting the power source to the wall.

2. 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.

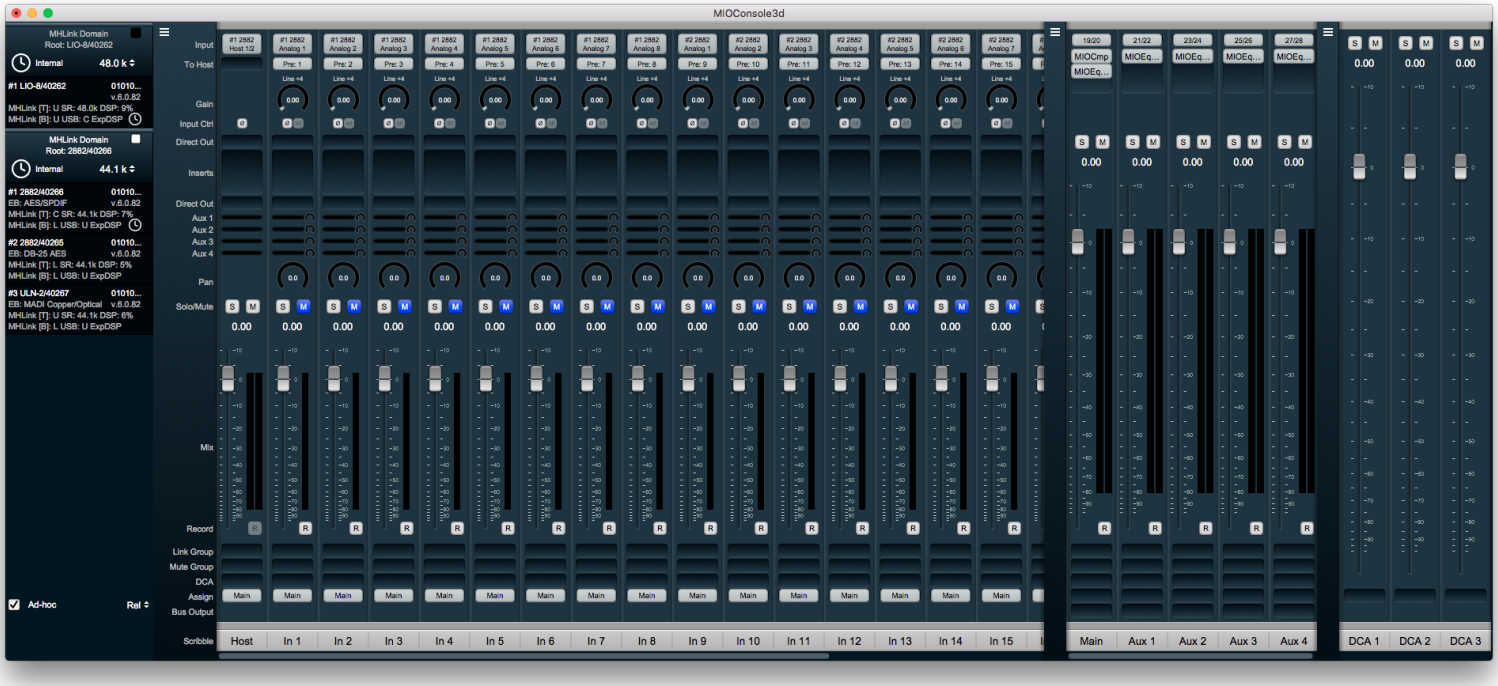
The I/O Routing interface includes a search engine and lets you narrow the selection view by box and port type, making it easy to target just the ports you want, even in large installations with hundreds of available computer, analog and digital I/O ports. Cascade and Group assignment features are included 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.



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 9.2.6. 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.



MIOConsole3d Monitor Controller and Cue Controls windows

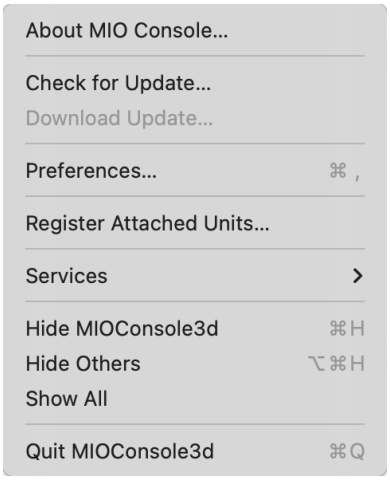
The MIOConsole3d Menu Bar

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.



MIOConsole3d Menu Bar

MIOConsole3d Application Menu



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.

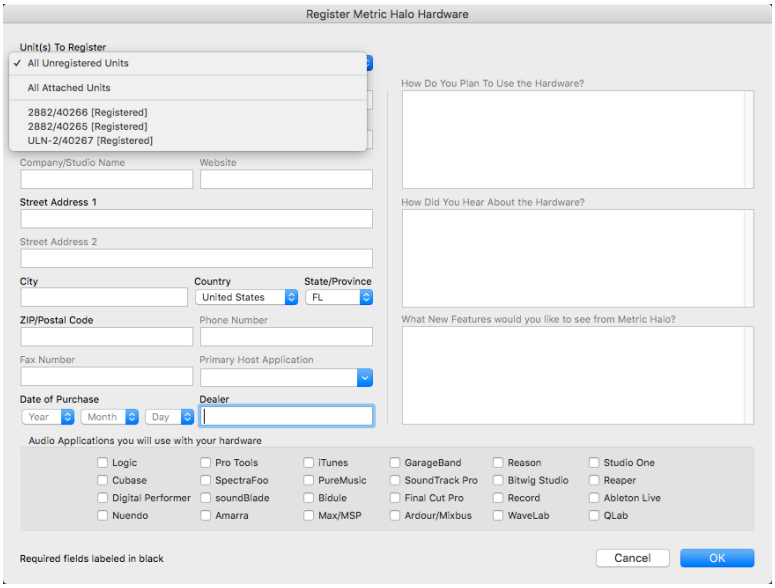


About MIOConsole3d

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.



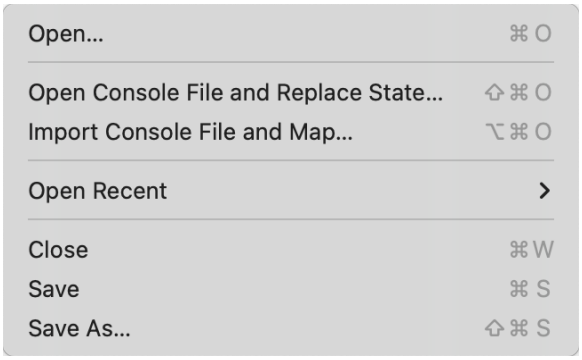
MIOConsole3d Registration Form

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.



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**

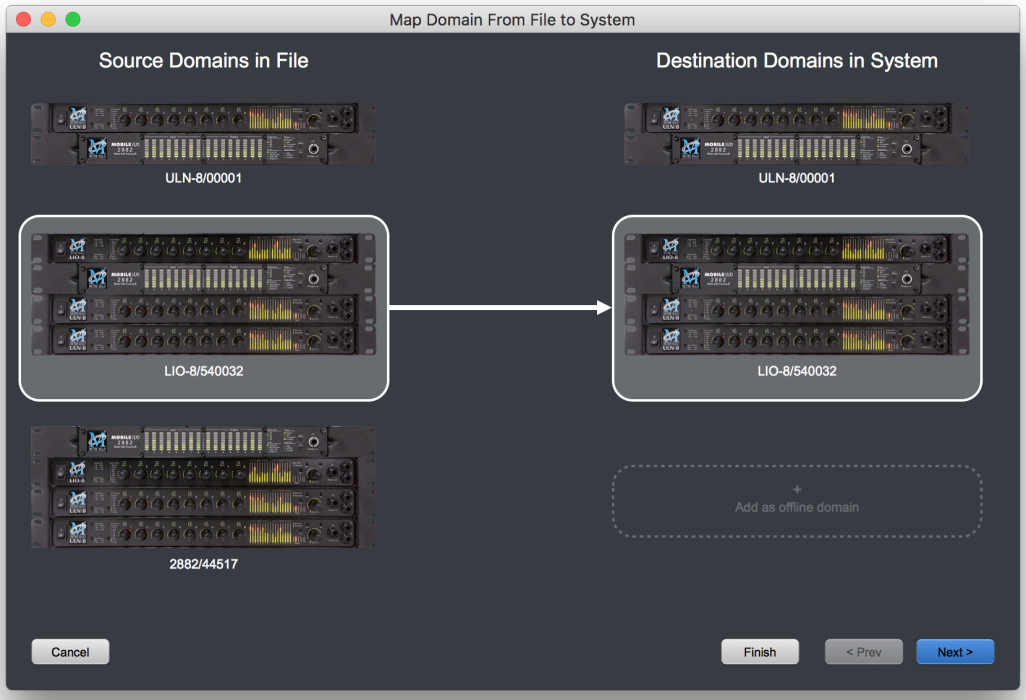
The Box Mapping interface walks you through the process of translating all box settings, routing assignments and mixer parameters from a saved console mixer file to your local 3d system. There are three stages to the process: first, selecting the Domain to import, then matching which source boxes will 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:



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

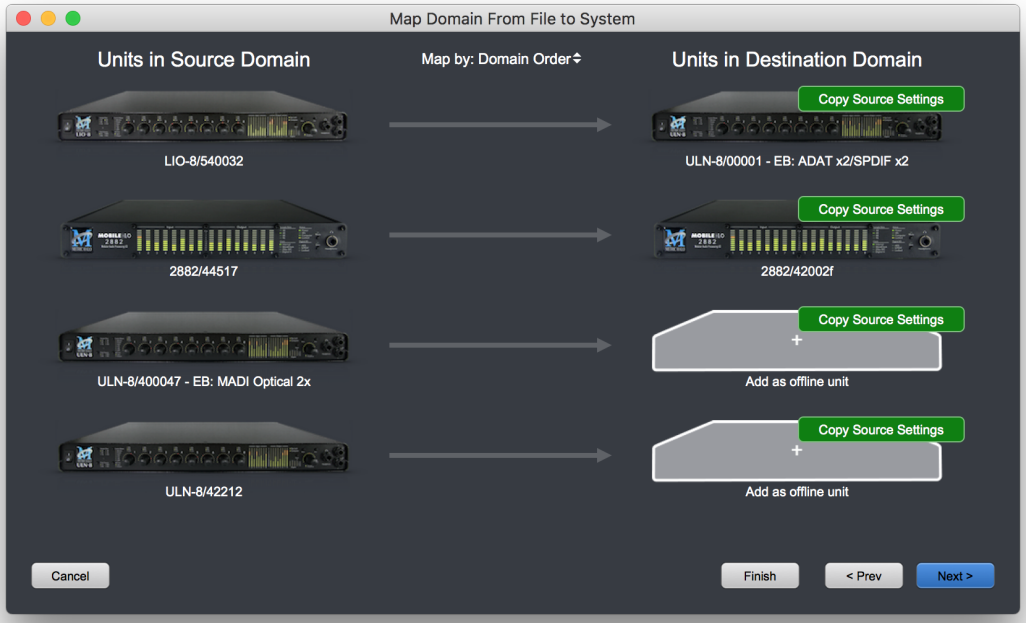
current root box to exist as both "online" and "offline" at the same time, so "Add as offline domain" will not be available as a target when the Source domain includes your current root box as its root (as in the example above)).

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:



Map Domain: Source Boxes To System Boxes

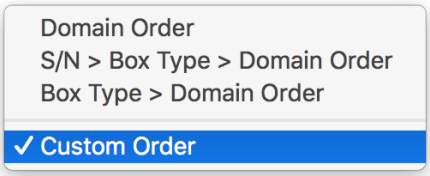
The **Box Mapping** pane allows you to choose how the saved boxes in the file are mapped to the actual boxes you have in your system. You can click and drag the boxes in the right hand column (drag them up and down) to set the mapping from a given source box to a destination box.

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.

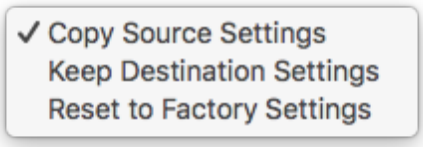
By default, the boxes will be ordered by the order they appear in the domain. By dragging the boxes around you can set the mapping to whatever custom order meets your needs. That being said, there are a few automatic mappings you might want to use. You can select from them via the "Map By" popup menu:



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:

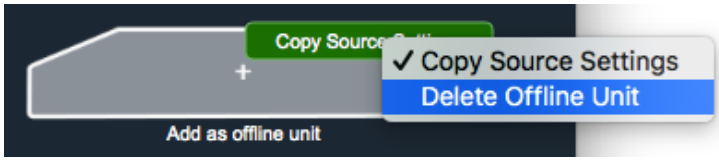


Map Boxes: Destination Box mapping mode

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.



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**:

Map Domain From File to System

Channel Map

#	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

Cancel < Prev Finish

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:

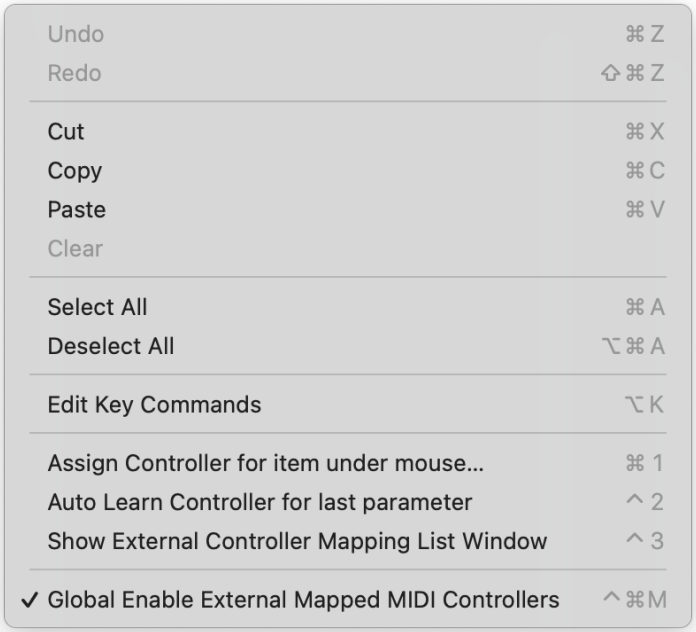
- **Index** lists the number of the strip in the Mixer desk layout starting at the left (#1) and moving to the right.
- **Type** Indicates the channel map routing object type:
  - **Mixer Strip:** A channel route point on a mixer input strip
  - **Bus Strip:** A channel route point on a mixer bus master 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).



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.

The last four commands:

- **Assign Controller for item under mouse...** (⌘1),
- **Auto Learn Controller for last parameter** (^2),
- **Show External Controller Mapping List Window** (^3),
- **Global Enable External Mapped MIDI Controllers** (⌘^M),

... are all related to External MIDI Controller mapping features built into MIOConsole3d. Hit the link above for more details.

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.



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.

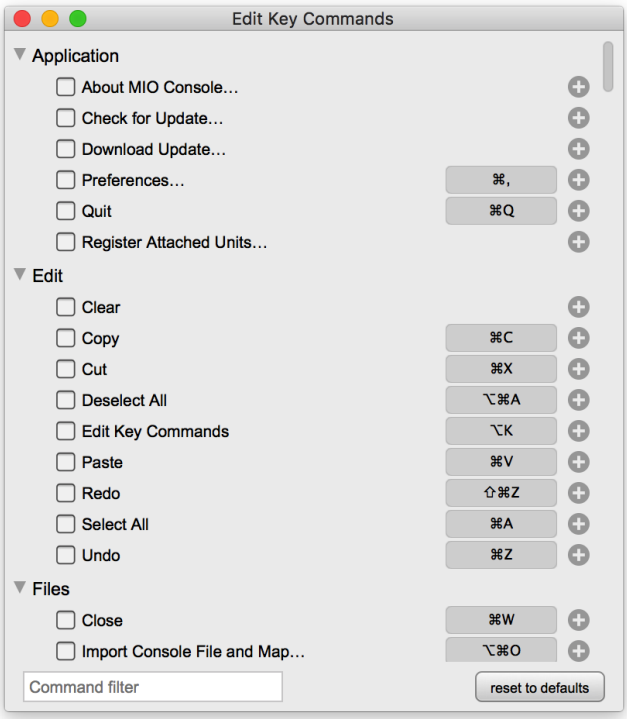
Some key commands are specifically geared towards controlling Mixer and Monitor functions from external control surfaces, and do not have equivalent control buttons in the MIOConsole3d UI.

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.

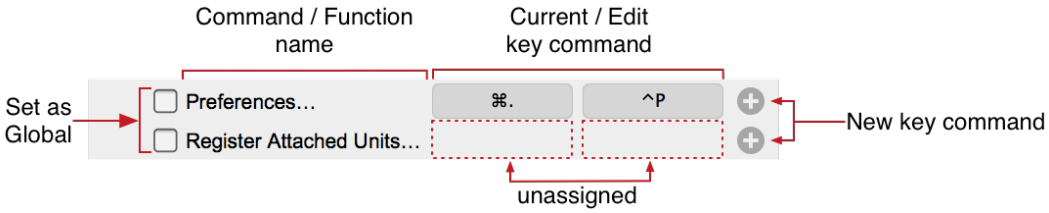
At the bottom right-hand corner is the **"reset to defaults"** button, should you wish to reset all key commands to the original factory default settings.

Below is the default view when you open Edit Key Commands…



Default view: Edit Key Commands default window

…and here is the Key Commands controls layout.



Edit Key Command UI

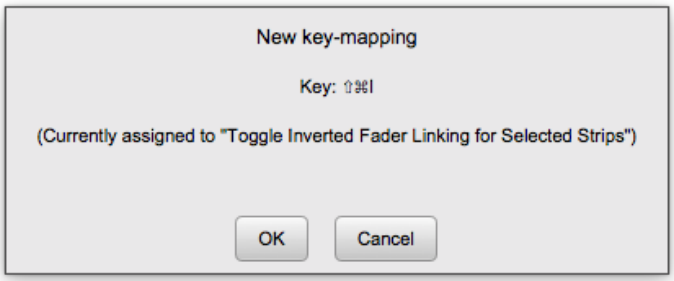
At the left are the Global Key Commands selection boxes, detailed later in this chapter. Global key commands will trigger MIOConsole3d functions even when the MIOConsole3d application is not in the foreground.

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.

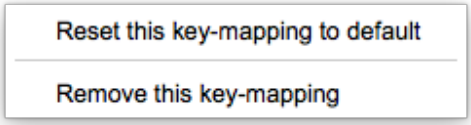
Clicking the plus sign “add” icon of a function that already has a key command will let you add a second or third key command for that function. After adding a third keystroke, the “add” icon disappears.

To modify an existing keystroke, select the current command key button, and enter your new key command for that function.



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.



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.

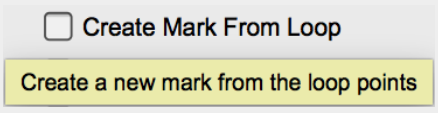
There are three extra Key Commands categories; **Monitor Controller**, **Parameter Controls** and **Talkback Controller** provide additional keyboard controls not available through the menu bar items or their submenus.

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.



Edit Key Commands menu tooltip example

Shown here is the Session timeline command, "Create Mark from Loop".

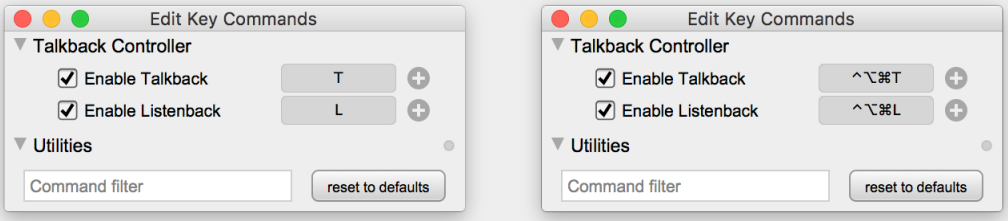
Global Key Commands

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.

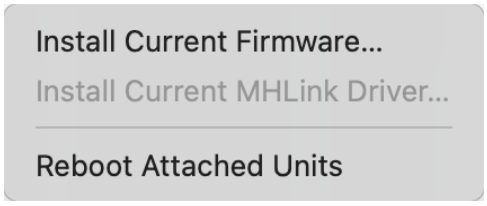


Global Key Commands example

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.



Utilities Menu

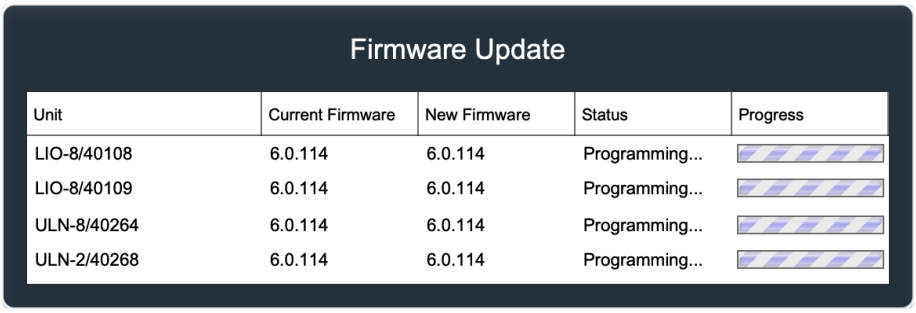
Install Current Firmware…

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.



Firmware Update dialog

Clicking "OK" in the “Install Firmware Update” dialog box will immediately launch the update process window, shown below.



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 on-line 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

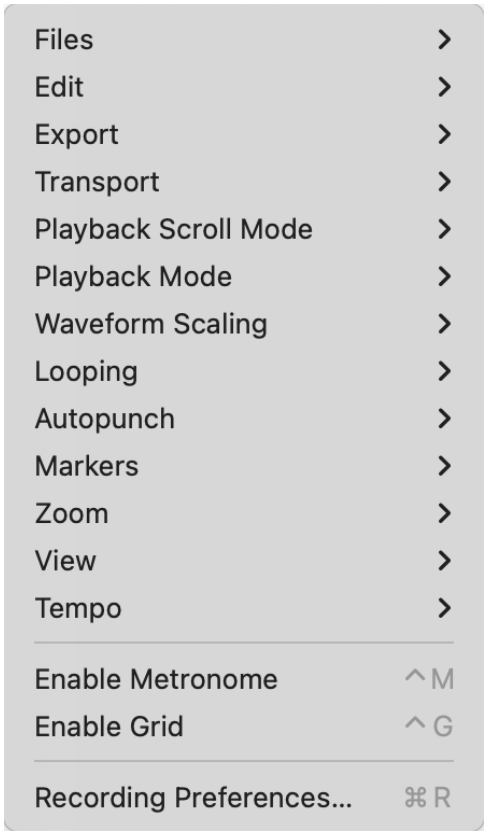
The Session panel has been designed with an eye towards hardware-based multitrack studio workflows, with the MIOConsole mix desk as a hardware mix desk and patchbay (which it actually is), and the Session panel as the multitrack tape deck.

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.



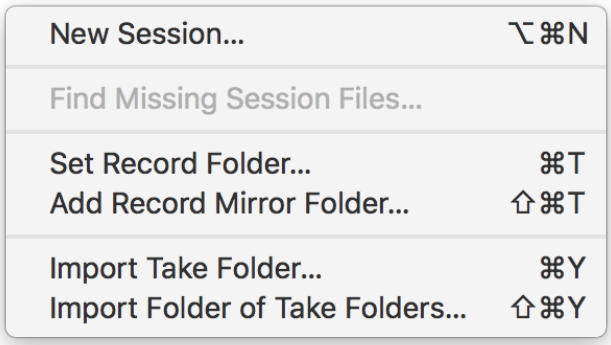
Session Menu (top level)

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

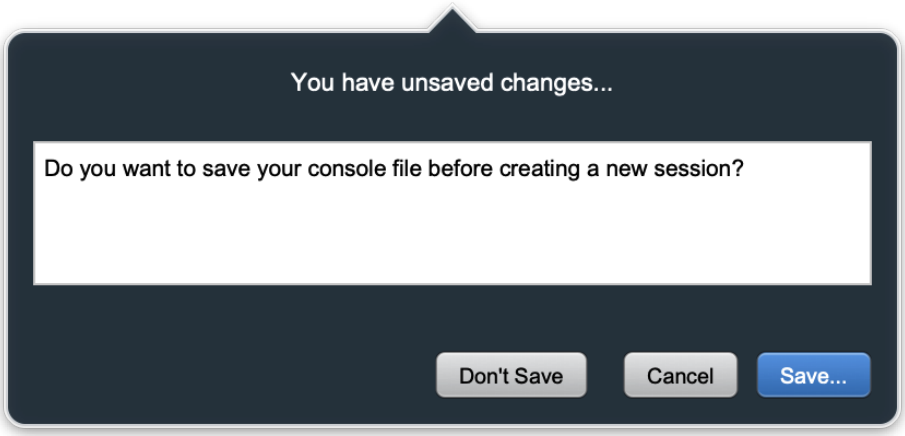
The "Files" menu lets you select folders for recording, importing source audio for playback, and basic file-related session functions.



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.

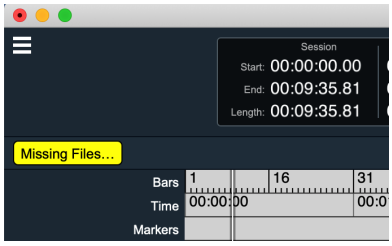


Session menu: Files: New Session "You have unsaved changes…" dialog

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.

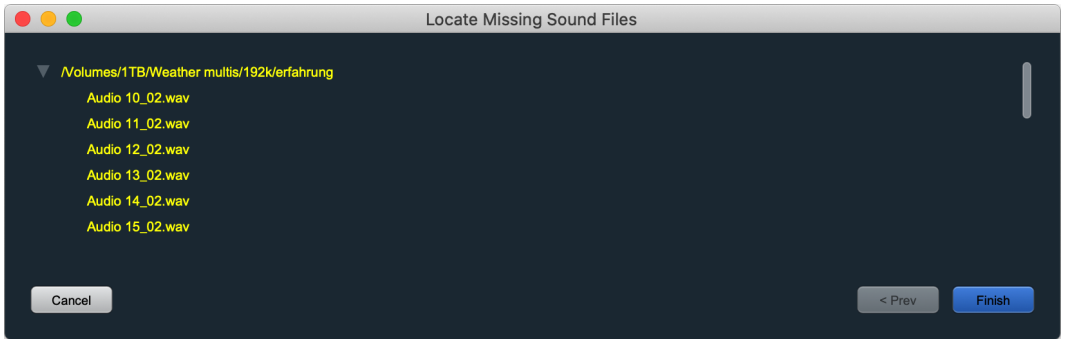
Upon naming and saving your console file, all existing audio files will be cleared from the Session tracks, ready for you to import new source audio.

- 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.



"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.

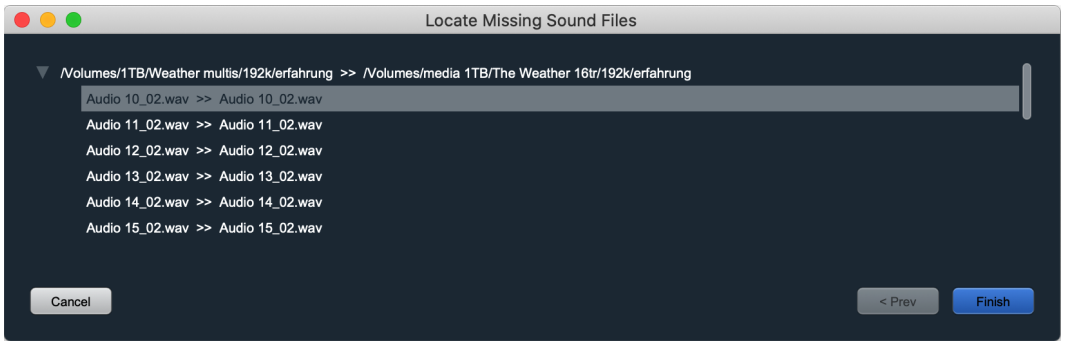


"Locate Missing Sound Files"

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.



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.

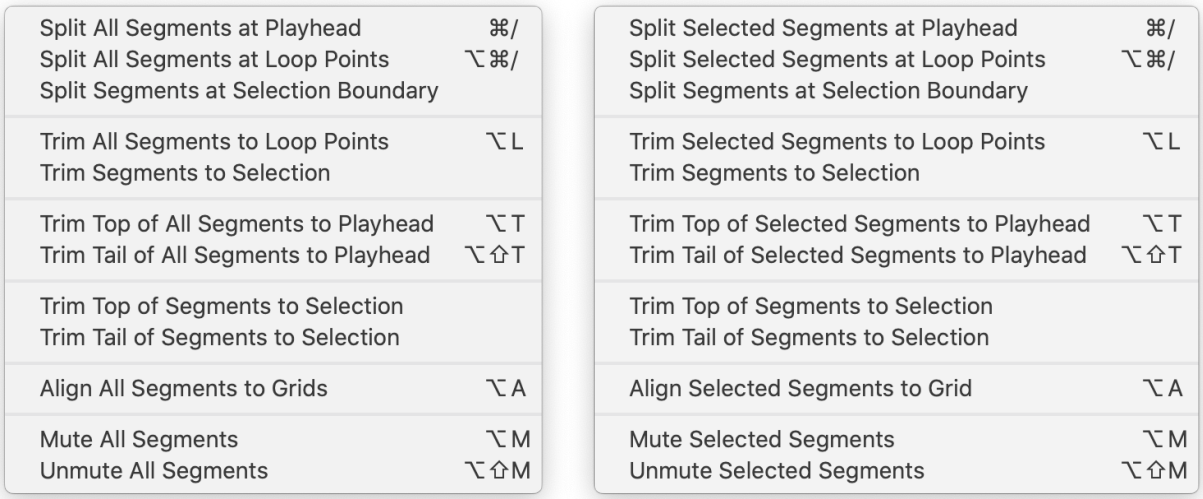
  Between the extreme efficiency of the MIOConsole3d Record Panel and the speed of todays SSD drives on USB3 and Thunderbolt connections, it is possible to assign multiple record mirror safety destinations for every session.

  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



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.

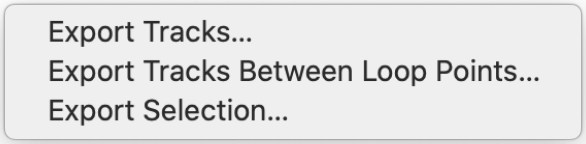
- **Split Selected Segments at Playhead** (default: ⌘/) places an edit splice in all selected segments at the current playhead location.
- **Split Selected Segments at Loop Points** (default: ⌥⌘/) uses the current Loop Start and End points as edit in/out points, and places edit splices at those points 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.



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.

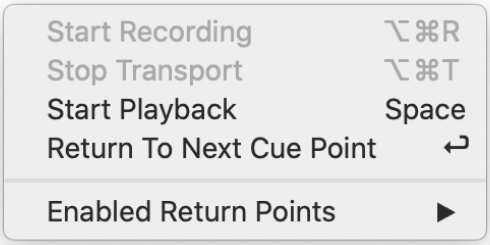
- **Export Selection...** is similar to "Export Tracks", but exports the audio only from within the current Timeline Range selection. It is best to export these files to a folder named with a description of the selected audio, with a reference to the Session from which it was derived.

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



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.

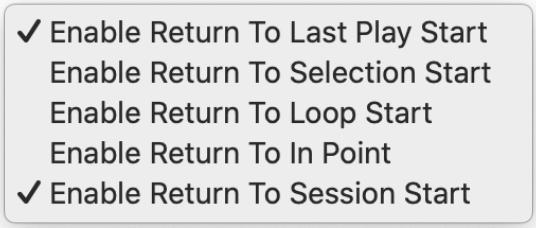
  Hitting **Start Recording** while a take is already in progress will immediately start a new take, including any manual break overlap assigned in the Recording Preferences. (see the Preferences: Recording section for further details)

- **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.

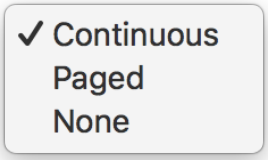
- **Return To Next Cue Point** (default: **Return key**) will toggle the playhead through the cue points selected in the "Enabled Return Points" submenu (immediately below). Each time you hit "Return" will move the playhead to the next cue point.

- The **Enabled Return Points** submenu lets you select the cue points to be included in the "Return To Next Cue Point" toggle rotation.



Session: Transport: Enabled Return Points selection menu

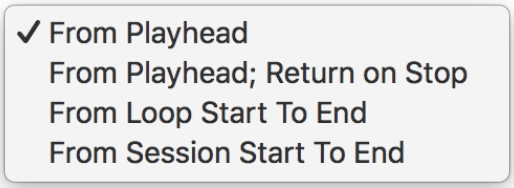
Session: Playback Scroll Mode menu



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

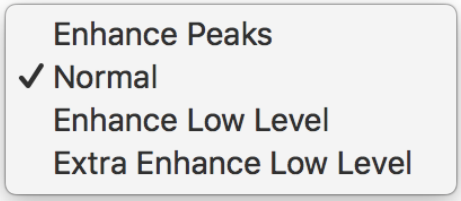


Session: Playback Mode menu

"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



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



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.

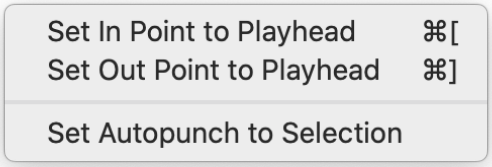
  Looped audio will play continuously unit stopped.

- **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

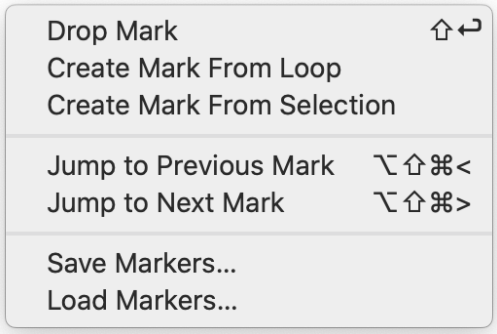


Session: Autopunch menu

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



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.

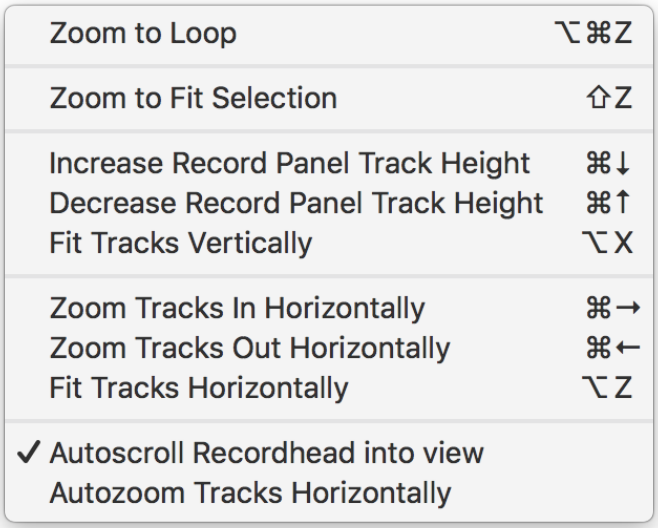
Note: "Start", "End" and "duration" fields display location and duration information as defined from the timeline and are not editable in the Marker List.

- **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 meta-data 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.

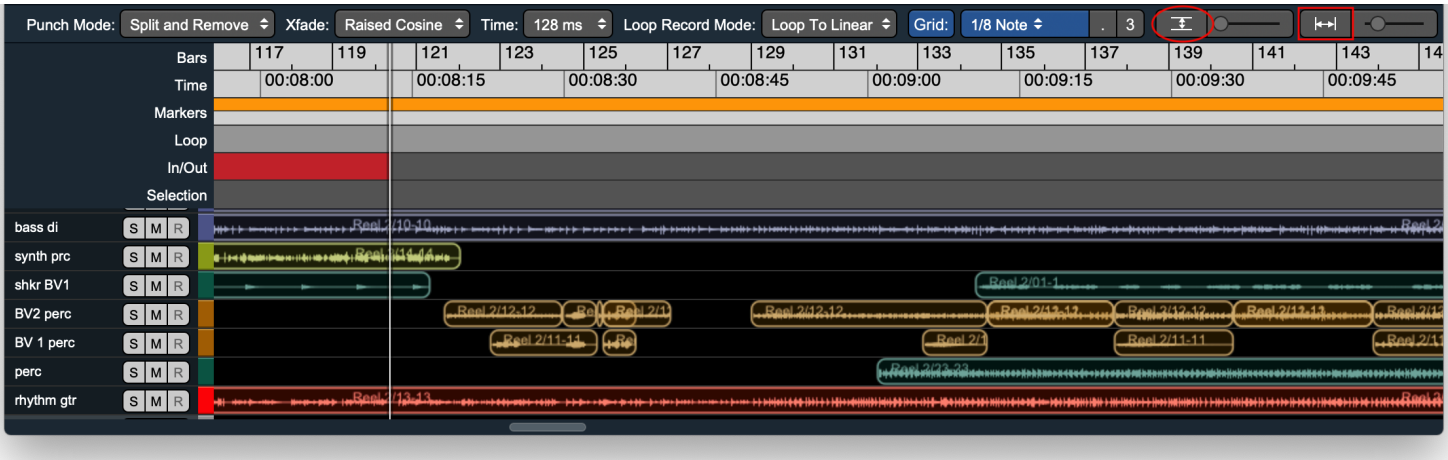
Session: Zoom menu



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.



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.

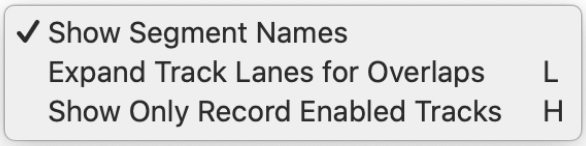
- **Fit Tracks Horizontally** is another toggle control which, when enabled, zooms out to show the entire length of all sound files in the current Session timeline.

  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



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.

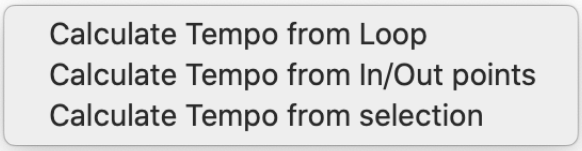
  Track *Lanes* are employed to visually manage overlapping Takes and/or segments within a track. New Track lanes are created to accomodate any new audio segments, whether a new recorded take or an audio segment copied or moved from another track.

  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.

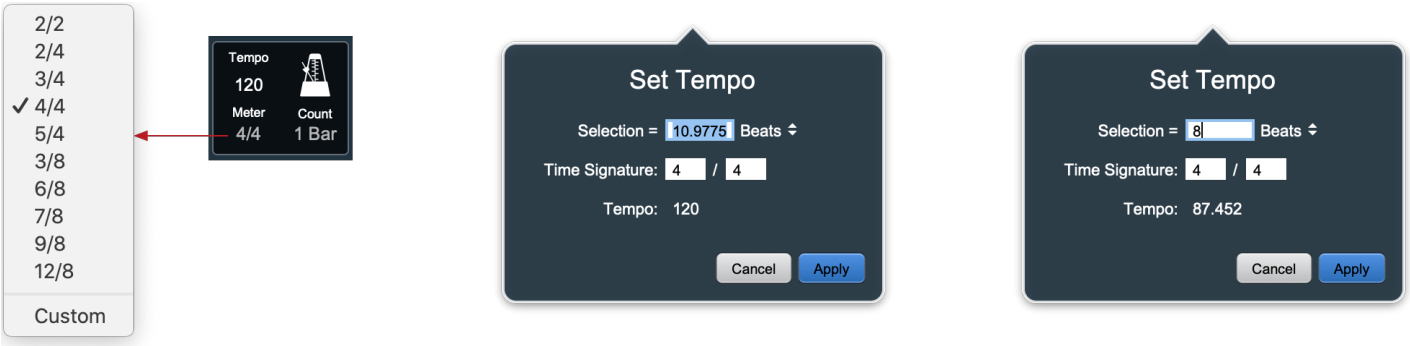
Session: Tempo menu

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.



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

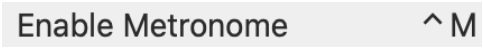
tempo is immediately calculated

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



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



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

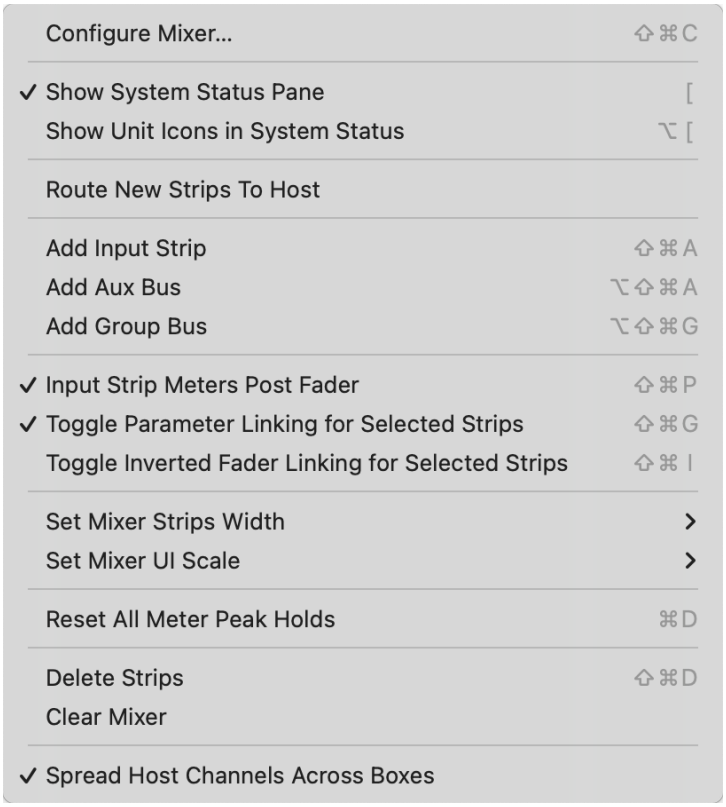


Session: 'Recording Preferences' menu item

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.



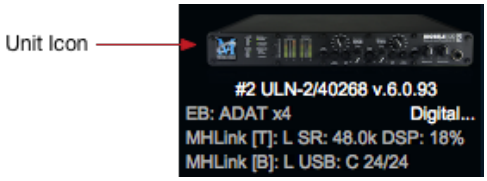
Mixer Menu

- **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.



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

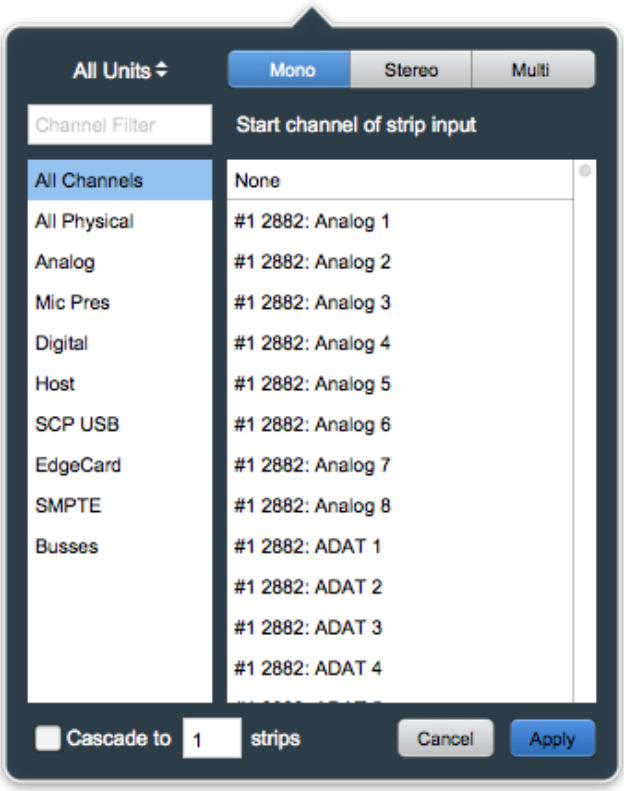
…is for adding new Input strips to your mixer. If you have a strip selected in the mixer and invoke **Add Input Strip**, a new empty strip will be added to the right of your selected strip. with no strip selected, the new strip is placed after the the last strip of your mix desk, all the way to the right. 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.



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.

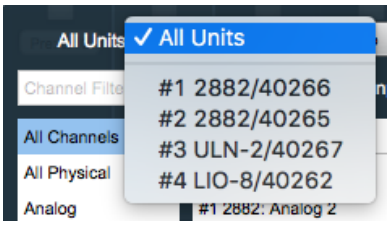


Source selection window

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.



"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
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 2.1. 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 key-stroke 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).

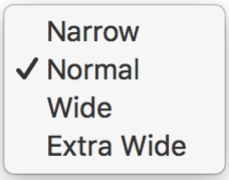
  See "Configure Mixer: Group Buses" to customize your Group bus configuration.

  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'.

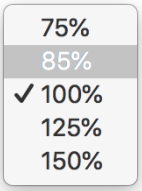
  Hint: Having easy-to-remember key commands for the two above toggles is especially handy for performance-oriented mixing

- **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.



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.



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:

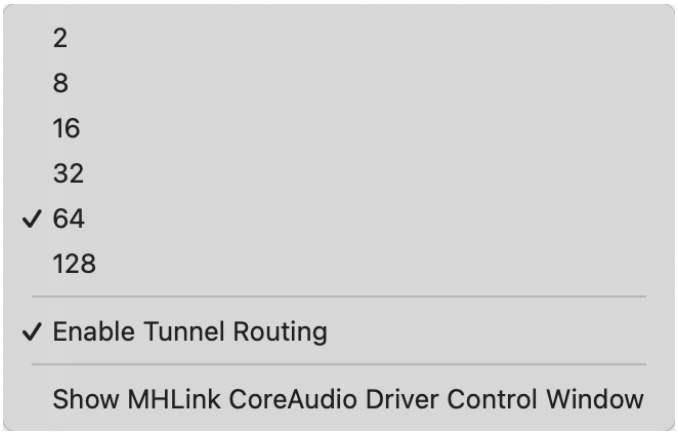
  1. DSP processes assigned within a single Mixer strip will be assigned to the same box, so if you have a ton of insert processes running on one particular strip, that process load will not be shared across boxes.

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.



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.

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. 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 2 channel I/O option is available for compatibility with a few popular mono/stereo-centric 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.

Tunnel Routing

Tunnel Routing is an internal mechanism which allows MHLink to route audio between boxes without introducing additional latency. It is enabled by default and in most circumstances should stay enabled.

Disabling Tunnel Routing is only necessary if you will be using more than 6 boxes in an MHLink chain at 4x sample rates. This is due to the bandwidth reservation required to make Tunnel Routing work over Gigabit Ethernet.

In a future release of the software, MIOConsole3d will manage this automatically and this menu item will be removed.

Show MHLink CoreAudio Driver Control Window

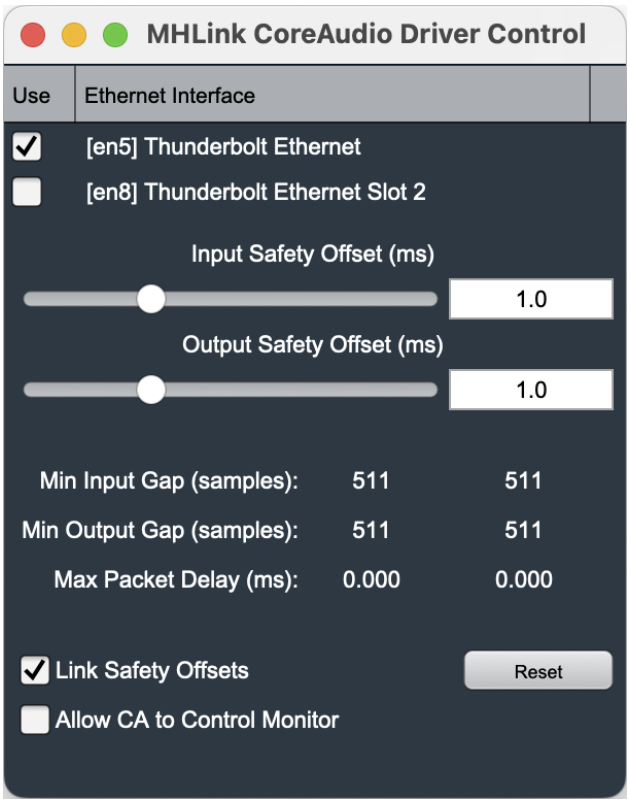
The MHLink CoreAudio Driver Control Window provides a set of controls to optimize MHLink driver performance and provide information when troubleshooting audio transport in general.

The MHLink user space driver allows a far greater degree of control than the system extension drivers of the past. This gives us the ability to optimize driver performance to any given scenario as needed.

With MHLink, a single computer/DAW workstation can be set up for absolute minimal latency, or for maximum headroom when running at high cpu loads.

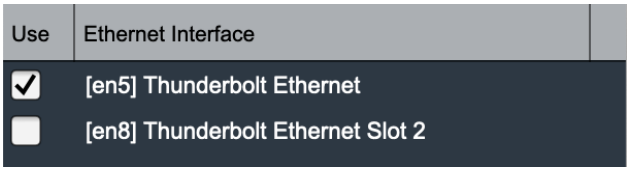
The MHLink CoreAudio Driver Control Window provides you with a UI to control the drivers' configurable parameters. These parameters include:

- Which Ethernet ports in the system will be used for MHLink
- The input and output safety offset used by the driver
- Whether or not the driver allows CoreAudio apps (including macOS) to change the monitor level



MHLink CoreAudio Driver Control

Ethernet Interface

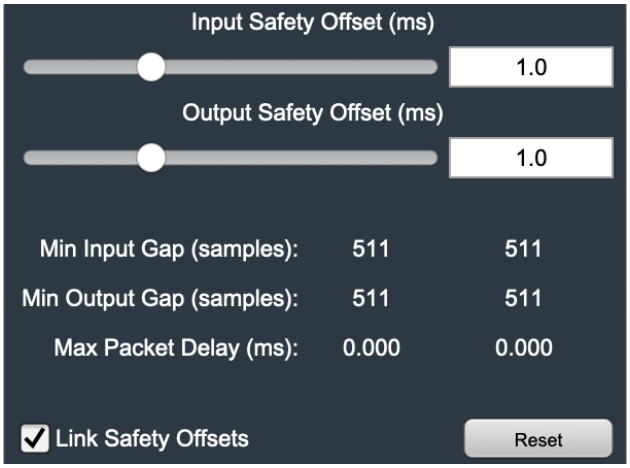


Some users have reported issues with other devices (like network routers) that receive MHLink discovery packets. Since we recommend dedicating a network port to MHLink, the other ports in your system do not need to be serviced by the MHLink driver. The Ethernet Interface table at the top of the MHLink CA Driver window allows you to choose which Ethernet ports in your system will be serviced by MHLink.

Uncheck the *Use* button for your regular IP network ports. The MHLink driver will detach from the port and will not reattach in the future unless you re-enable it. The Ethernet Interface name displayed here matches the name shown in Network pane of host OS System Preferences/Settings.

> *Note to early adopters*: In the past, with the KEXT driver, it was possible to disable IPv4 to reduce spurious traffic on a port that was dedicated to MHLink. Due to a quirk in macOS, the userspace driver cannot connect to an ethernet port if IPv4 is disabled. Any such ports will be be listed in red, with the message "[Unavailable; Enable IPv4]". If you wish to use such a port for MHLink, go to the Network pane in System Preferences/Settings and set "Configure IPv4" to DHCP.
>
> To be clear, this does not mean that MHLink is suddenly using IP... it isn't.

Safety Offsets, Input/Output Gap and Max Packet Delay



The Safety Offset controls are pretty straightforward: higher safety offsets offer greater stability at the cost of adding driver latency. Set the controls while monitoring audio such that the Min Input Gap and Min Output Gap readouts are stable and not turning Yellow or Red.

That said, knowing more about the information displayed will go a long way towards milking more performance out of your computer and DAW.

The good news is most modern computers can handle DAW sessions with driver safety offsets of less than 1ms, and even with very heavy cpu loads a safety offset of 5ms or less offers plenty of headroom.

That said, life seems to enjoy putting audio engineers in unreasonable situations, especially on the road. For those special cases we increased the amount of safety offset you can set using the MHLink CoreAudio Driver Control Window to 100ms. (See the Setup Guidelines on the next page for further details.)

Keep in mind this control only displays and affects the MHLink hardware driver. The majority of system audio latency is still happening in your DAW. By isolating driver offset control, this interface provides a handy confirmation of where exactly your audio path latency is happening.

MHLink Driver Control Setup In Detail

The window shows how the OS's scheduling is affecting performance reliability at the current safety offset. The performance statistics area of the window shows the gap in samples between the driver and Core Audio on behalf of host apps. These readouts (especially the Output Gap) can be unreliable when a host initializes its I/O buffers to start transporting audio, so you should reset the the statistics after you start audio streaming.

Simply put...
If the Input Gap goes below 0 samples, then there will be a dropout on the input to the hardware.
If the Output Gap goes below 0 samples then there will be a dropout on the output from the hardware.
If the Max Packet Delay goes below 0 then there will likely be a dropout in both directions.

Troubleshooting Note: If the Max Packet Delay and Input Gap are both greater than 0 samples but the Output Gap is below 0, then one or more of the host apps have had an internal overload and have delivered audio to the system late. This is not a problem caused by the driver, so adjusting the safety offsets here will not fix these kinds of issues.

Setup Guidelines

We have found that with the userspace driver, setting the safety offset to 1.0ms in both directions is consistently reliable for most modern Macs, and so we have made that the default setting. As a result, all things being equal, this new driver will reduce the Round Trip Latency (RTL) over MHLink by 6ms and the output latency by 5.5ms versus the old 3d KEXT driver.

For Apple Silicon systems, we have found that you can further reduce the safety offsets; for instance we have found 0.5 or 0.8ms to be reliable, which would further reduce the round trip latency by as much as 1ms.

For older systems (for example older Mac Pro towers), on older versions of the OS, or for Ethernet NICs that are bursty, you may find that the Max Packet Delay exceeds the safety offset resulting in audible clicks. For these situations you should increase the safety offset to about 25-30% larger than the Max Packet Delay that you get while streaming with your usual CPU load. On older Mac Pro towers, that will probably mean increasing the safety offset to 2-2.5ms. This is still a substantial improvement over the old KEXT driver.

The extreme case: Recording high channel counts at high sample rates in MIOConsole Session

For people recording very large channel counts at very high sample rates, especially on older computers, the scheduling jitter in the OS can be high enough to swamp the previous maximum safety offset of 10ms. For people doing large live recordings, Round Trip latency is not an issue (as the computer is not used for monitoring for talent) whereas absolute safety is. So for this use case, you can now increase the safety offset up to 100ms. This is not required (or even desirable) for most users, but in case you need it the capability is there.

You can adjust the safety offsets with the sliders or by typing a number in one of the entry fields (the number is in milliseconds). While it is possible to change the safety offsets independently, in virtually all cases they should be set to the same value.

Note: Changing the Safety Offset while audio is running through the driver will cause the driver to start and stop audio transport while the offset is being updated. This may confuse some hosts. In addition, while the new offset is immediately reported to CoreAudio hosts, some hosts may not take note of the changed value. We recommend that you first quit any audio applications you are running, then change the Safety Offset, then relaunch your audio apps in order to ensure your audio systems are all properly communicating.

Allow CA to Control Monitor

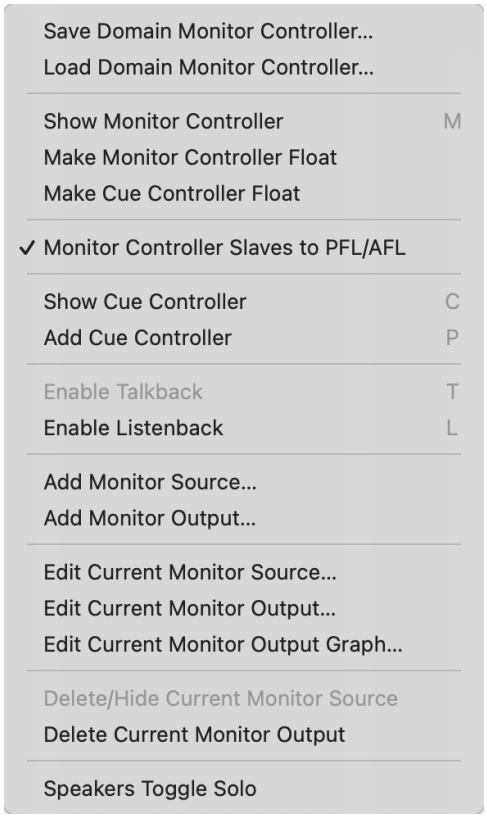


The kernel driver has always provided a CoreAudio control to change the Monitor Level via CoreAudio. The system volume control (e.g. keyboard volume keys) use this control when the MHLink CoreAudio device is selected as the System output. Some host apps will also use this control (e.g. Cubase and Nuendo). While this functionality is convenient, sometimes it can result in unexpected changes in the monitoring level.

If you don't want CoreAudio to be able to control the monitor level, de-select this checkbox, and the control will be removed from CoreAudio.

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.



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.

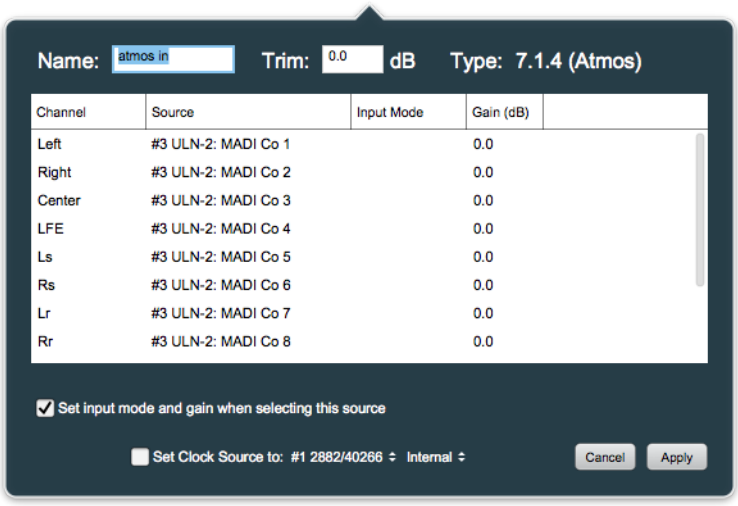
  See also Interaction Preferences for details regarding Monitor Controller loading options.

- **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 sourced from any external device, bypassing the MIOConsole3d mixer environment.

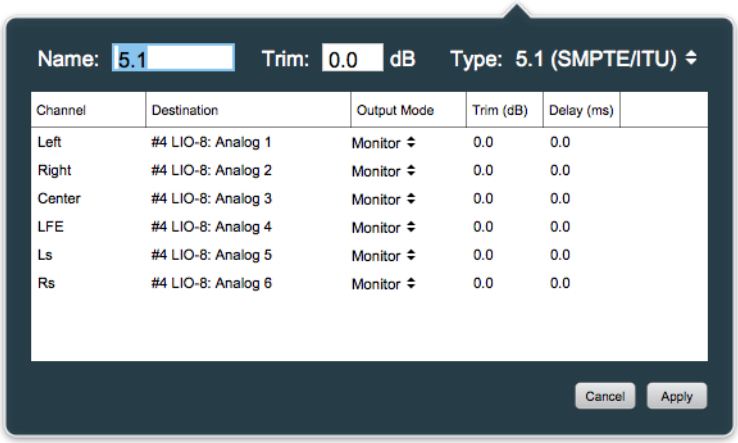


Monitor Source routing window

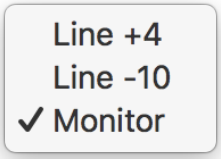
The checkbox "Set input mode and gain when selecting this source" allows you to configure the physical input when you select this input. For instance you could use this to, say, switch between the 'Line +4' and the 'Mic S/R' inputs on a ULN8 depending on your MC Source selection.

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 its 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.



Add Monitor Output window



MC Output Mode selector

Output Mode sets the analog gain stage for the selected output channel between +4, -10 and Monitor. The Monitor reduces pads the output signal by 30dB for directly feeding high-gain amplifiers. Generally amplifiers are designed for optimal performance at full volume, with gain staging handled by a high quality pre-amp (the MH Monitor Controller).

Trim (dB) sets a gain offset for each channel. This simplifies running mixed amp/speaker setups and allows fine volume-tuning from the listening position.

Delay(ms) allows you to delay the output of speakers which are physically closer to the listening position such that they are time-aligned with speakers that are further away. Delay is accurate to 0.001ms (less than a millimeter) and the maximum delay available is 100ms (well over 30 meters).

- **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. Window menu items above "Minimize" will vary depending on your display setup and macOS version.



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**, **Show/Hide Cue Controller** and **Show/Hide External Controller Mapping List Window** 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.

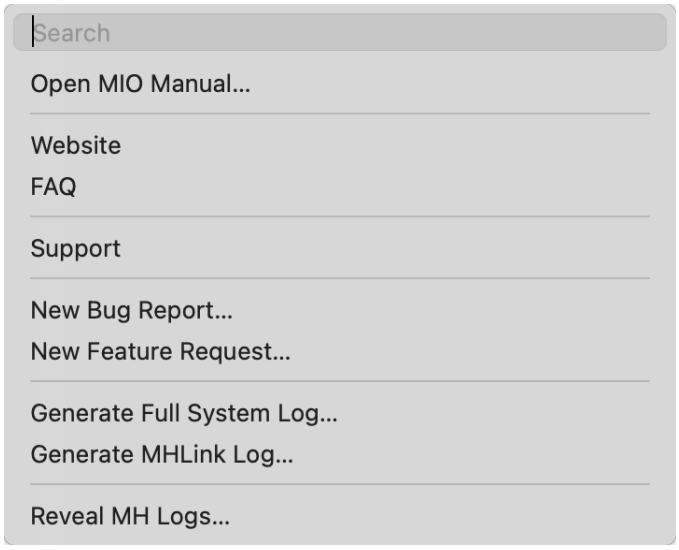
  Note in the Window menu screenshot at the top of this section, a check mark '✓' indicates the window currently in focus, in this case, the Cue Controls.

  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.



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.

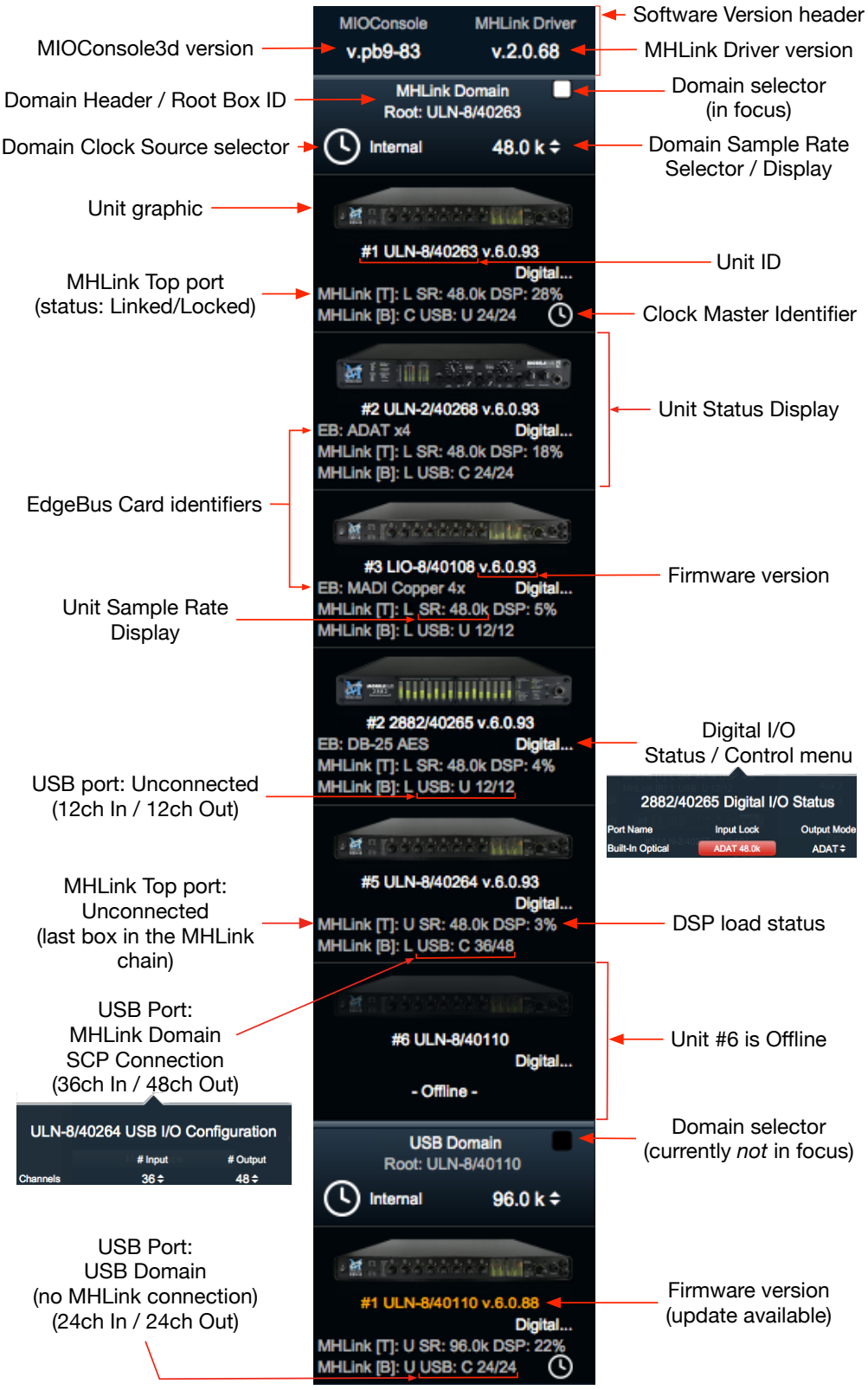
  If you ever need to use the Bug Reporter, please make sure to include your return email address so we can follow up with you and make sure we get the issue resolved as quickly as possible.

  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



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 left bracket **[** 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

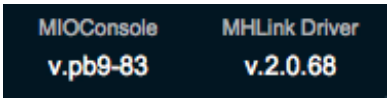
There are five sections graphically represented within the Status Pane, but only the first three concern us here. The other two sections, Mute Groups and Link Groups are covered in their own sections of the "Mixer Strip Controls Breakdown" chapter.

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 Downloads folder.

See the Software Installation section for full installation details, including important macOS security considerations when installing audio software in general. (Short version: launch the new Console and click the orange text…)



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.



System Status Pane: 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.



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 MIOConsole3d interface.

Domain Header UI

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.

Note: Domains can be a single unit connected via USB

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).

Primary Domain Header Controls



MHLink Domain Header details

Domain Selector

The Domain Selection status box at the upper right of the Domain Header turns 'white' to indicate the Domain currently controlled by the MIOConsole3d UI.

In most cases, there will be only a single domain (one domain header, which is naturally always active).

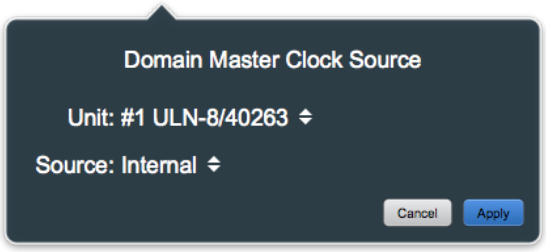
When a Domain is not in focus, its status box will be dark.

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 MIOConsole3d software 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 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.



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.

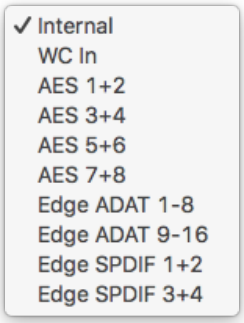


Master Clock Unit selector

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.



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.



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.



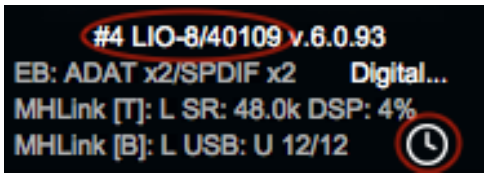
Domain Clock Source and Sample Rate Selector/Display examples

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.



Clock Master Identifier icon

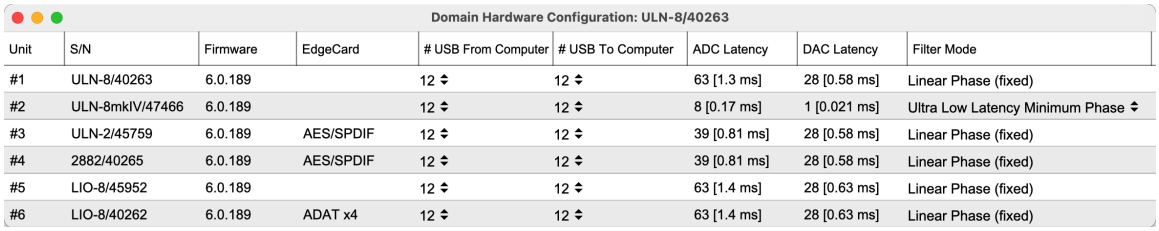
The **Clock Master Identifier** icon will always be attached to the Root box when the Domain is set to Internal clock, even if you set the Unit selector to a different box.

Secondary Domain Header Controls

Right- / control-clicking the Domain Header of an opens a pop-up menu of operations at the domain level, affecting either the domain itself or every device in the domain.

Domain Configuration Window

The top entry in the Domain Header Controls submenu opens the **Domain Hardware Configuration** window, a consolidated information and controls interface for every box in the domain.



Unit	S/N	Firmware	EdgeCard	# USB From Computer	# USB To Computer	ADC Latency	DAC Latency	Filter Mode
#1	ULN-8/40263	6.0.189		12	12	63 [1.3 ms]	28 [0.58 ms]	Linear Phase (fixed)
#2	ULN-8mkIV/47466	6.0.189		12	12	8 [0.17 ms]	1 [0.021 ms]	Ultra Low Latency Minimum Phase
#3	ULN-2/45759	6.0.189	AES/SPDIF	12	12	39 [0.81 ms]	28 [0.58 ms]	Linear Phase (fixed)
#4	2882/40265	6.0.189	AES/SPDIF	12	12	39 [0.81 ms]	28 [0.58 ms]	Linear Phase (fixed)
#5	LIO-8/45952	6.0.189		12	12	63 [1.4 ms]	28 [0.63 ms]	Linear Phase (fixed)
#6	LIO-8/40262	6.0.189	ADAT x4	12	12	63 [1.4 ms]	28 [0.63 ms]	Linear Phase (fixed)

Domain Hardware Configuration window

The window is laid out as follows:

- **Unit**: The Unit number of the box, starting with #1 (the Root box) connected to the host computer through the MHLink chain.
- **S/N**: lists the model and serial number of the device.
- **Firmware**: shows the currently installed firmware version.
- **EdgeCard**: lists the installed EdgeCard (if any).
- **# USB From Computer**: sets the number of output channels from the computer to the box over USB/ USBSCP.
- **# USB To Computer**: sets the number of input channels to the computer from the box over USB/ USBSCP.
- **ADC Latency**: lists the box ADC latency in samples and milliseconds.
- **DAC Latency**: lists the box DAC latency in samples and milliseconds.
- **Filter Mode**: lists the ADC/DAC filter mode. Filter mode is MkIV boxes only - see below).

The USB Channel configuration and Converter Filter Mode is persistently stored on the unit hardware and will remain as set (even across reboots) until you change it.

If you select multiple units in the the list, then any changes you make to a popup in a selected unit will be applied to all the selected units. This allows you to set all of the units to the same configuration in a minimum of clicks. You can also set each unit's configuration independently, which may be appropriate for USB configs depending on the hosts connected via USB, and may also be appropriate if you want to compare the different filter modes between two or more boxes in real-time.

MkIV A/D - D/A Converter Filter Modes

The converters on the mkIV units support multiple filter types. Since mkIV shipped, it has been using the **Low Latency Minimum Phase** filter option that provides both low converter latency and what we found to be the best, most natural and musical sounding conversion.

The different filters make different engineering tradeoffs (basically low latency vs. high accuracy), and users can now examine and audition those tradeoffs and select the filter mode which best suits their current workflow and personal preferences.

Note: A/D converters only change latency between Linear vs. Minimum Phase filters, whereas the D/A latency changes with each filter setting.

The 4 filter modes available are as follows (in order of lowest to highest latency):

- **Ultra Low Latency Minimum Phase**: shaves 3 samples off the trip through the DAC at the expense of more in-band phase shift and less out-of-band attenuation. This filter is available, but we found no particular advantage to it even in live monitoring through the box.
- **Low Latency Minimum Phase** (default): provides low latency conversion with a minumum phase filter with no pre-ringing. The minimum phase filter does exhibit in-band phase shift, and so creates a larger signal difference for D/A/D loops than the higher-latency linear phase options. We found this filter to sound the most "natural" in double blind testing and the least fatiguing over long listening sessions (which is why it is the default).
- **Linear Phase**: does not introduce any in-band phase shift from the antialiasing/antiimaging filters, and so provides decreased signal difference for D/A/D loops at the expense of higher latency and the introduction of pre-ringing which many people perceive as a "hardening" of the sound.
- **High Attenuation Linear Phase**: is also linear phase, however this filter improves the amount of out-of-band attenuation in the DACs anti-imaging filter. This filter is the closest to the filters in the mkIII converters.

For additional latency control, see the MHLink Core Audio Driver Control section.

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 currently connected to your Host computer. You will note that all of the hardware mixer settings, layout, routing and processing appear as normal, but routing UIs within the mixer all show as yellow and all routing menus show routes as **[Offline]**, meaning those routes are not available.

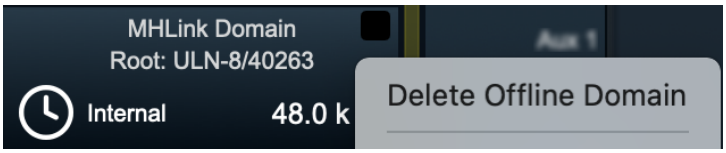
This is a feature requested by touring and location engineers which allows them to examine and edit all MIOConsole3d routings, settings and parameters without being connected to the boxes. This editing can be done on a laptop on a bus or flight between tour stops.

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 and colored yellow since the physical boxes are unavailable.

Opening and editing saved MIOConsole3d domains while offline: Use the MIOConsole3d File > Open... command to open up and examine any .cnsl3d file as an offline domain.

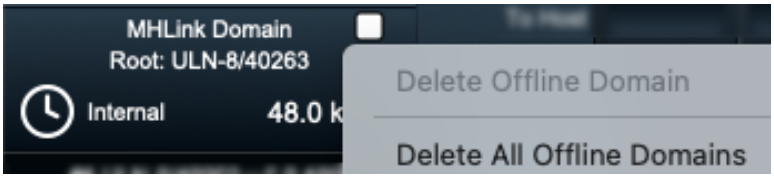
Removing unwanted offline domains: You can remove an offline Domain by right- or control-clicking the offline Domain Header. This will open a pop-up menu which will remove the offline Domain from the System Status Pane.



Delete Offline Domain pop-up

Select Delete Offline Domain and the offline domain will be removed.

Alternatively, if you click the header of an online domain, the Delete Offline Domain menu item will be grayed out (since the domain you've selected is actually on line), but you can remove any offline domains by selecting Delete All Offline Domains.



Delete All Offline Domains 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 a computer running MIOConsole3d. They are appropriate for the following use cases:

- Preconfigured routing and/or processing to use with a system that doesn't have MIOConsole3d available (e.g. Windows, Linux, iOS, Android)
- Preconfigured routing and processing for standalone fixed installation or conversion (e.g. as a sidecar for an analog console, a front end for a venue FOH + recording system, a set multi-user cue mix/ monitoring array, or all of the above at once
- Preconfigured fixed routing to allow the use of the HW with a Mac or PC without having to launch MIOConsole3d (e.g. to configure the MIO domain as a straight-up, super-low latency, multi-headed "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 stored in the hardware boot-up memory space to be loaded automatically when powered up. 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:

- Master Clock Source
- Master Sample Rate
- Computer Routing including Host and SCP
- Interbox Routing and DSP load sharing (MHLink)
- Physical I/O routing, including Headamp Input Type, +48 Phantom Power state, Polarity and Gain Trims
- All Mixer Strip Gain, Pan, Mute, Solo, Aux and Group bus send and return trims
- All Mixer DSP processing parameters
- All Monitor and Cue control assignments and gains

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, all 3d boxes must be connected in exactly the same configuration as 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 I/O routes 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, preferably via prewired cable interfaces. 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 first look for a computer connection with the MIOConsole3d software running.

If no MIOConsole3d is found, it checks to see if there is a valid boot-state snapshot stored. 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.

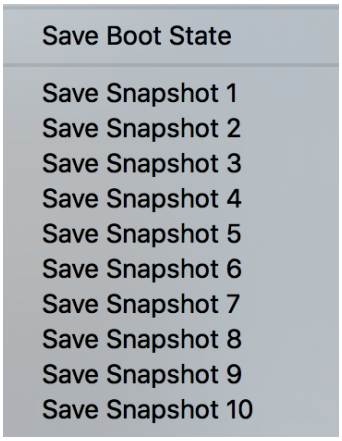
Creating Boot States and Snapshots for offline operation

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:



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.

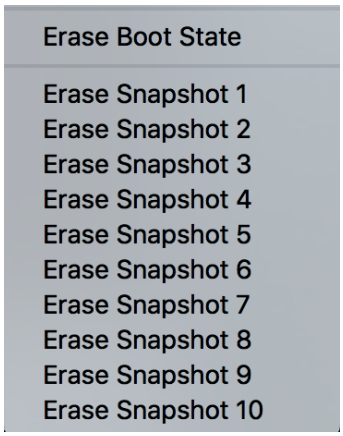
Pre-configuring Snapshots for USB

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. In other words, the 3d box is smart enough to simply use the assigned USB routes even when MHLink is not connected.

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.



Domain Header Contextual Menu: "Erase Boot State / Snapshot"

- "Erase Boot State" will delete the current boot state of each connected box in the domain. This completely clears the boot memory register, returning each box to its factory default condition.
- To erase the snapshot to one of the other snapshot slots, choose the appropriate "Erase Snapshot x" item (where x is the appropriate slot number).

Ordinarily it is fine to overwrite a Snapshot in a given slot with a new one, but the Erase commands are available to completely clear the memory registers for a fresh start.

Front Panel Controls: 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 preconfigured Boot State.



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.



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.

Front Panel Control: 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) 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.

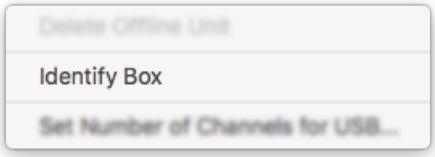
Unit Status Display Overview

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.



"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.



Unit ID and Firmware version

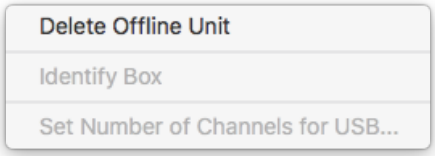
The 'Unit ID and Firmware' text will turn orange when a new version of the firmware becomes available. Click on the orange firmware version number to initiate the firmware update process.

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- 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.

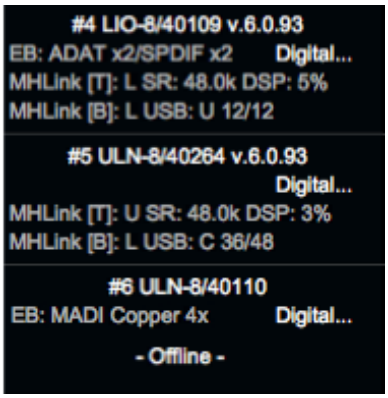


"Delete Offline Unit" pop-up

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 cable from the computer to the Root box 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.



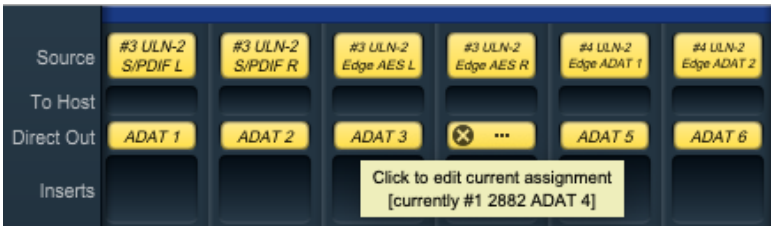
Offline Unit detail

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.



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:

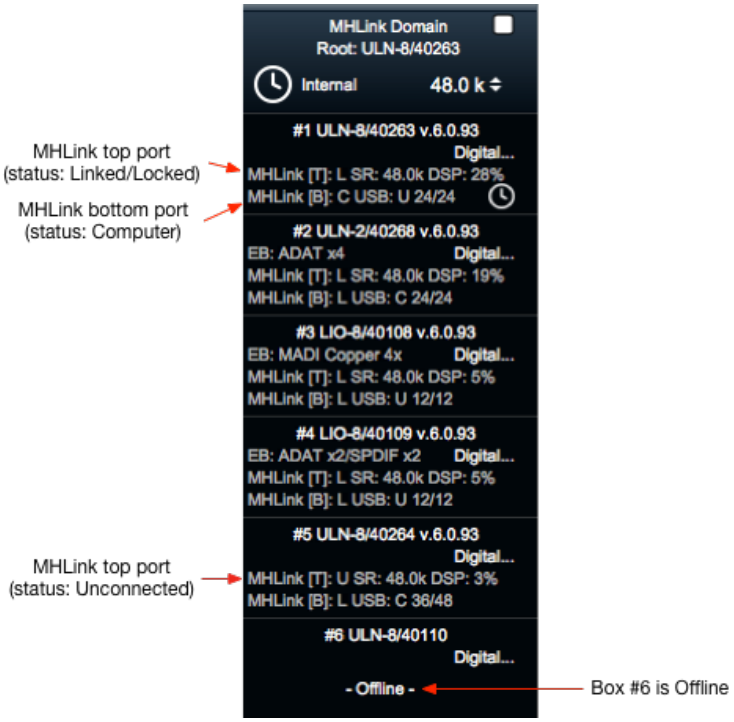
1. When you re-attach the missing 3d box(es), all I/O routes and DSP processors are restored as if they were never removed.

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.



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.

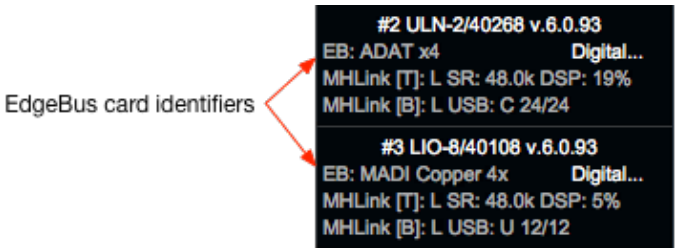
- "MHLink [B]: C" indicates the Host **C***omputer* connection, in this case to the 'Bottom' MHLink port. Only box #1, the Root box, will have this designation. If by chance you ever see a "C" on any MHLink unit port other than the Root, you want to pull that plug. You can also think of it as the "**C**onsole" 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.



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.



Digital I/O Status and output controls

This window provides details regarding every physical digital audio I/O port in the selected 3d unit.



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™.

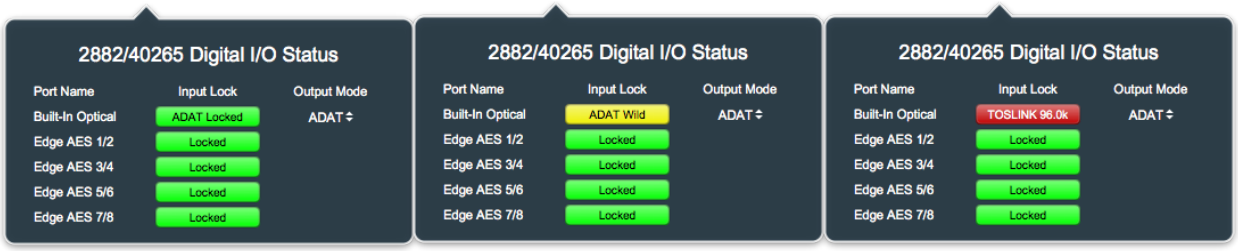
Input Lock tells you *if* there is a digital audio signal present on that port, and (on ports which support multiple data formats) tells you what format it is. All 3d digital audio inputs auto-sense the sample rate

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:



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.



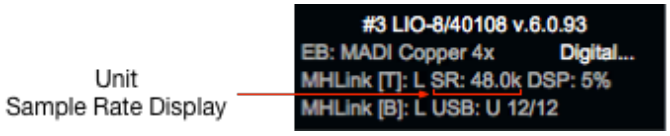
Test session: first-generation ULN-2 with 3d MADI clocked and locked to Sony 3348HR



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.



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.



DSP Load Status

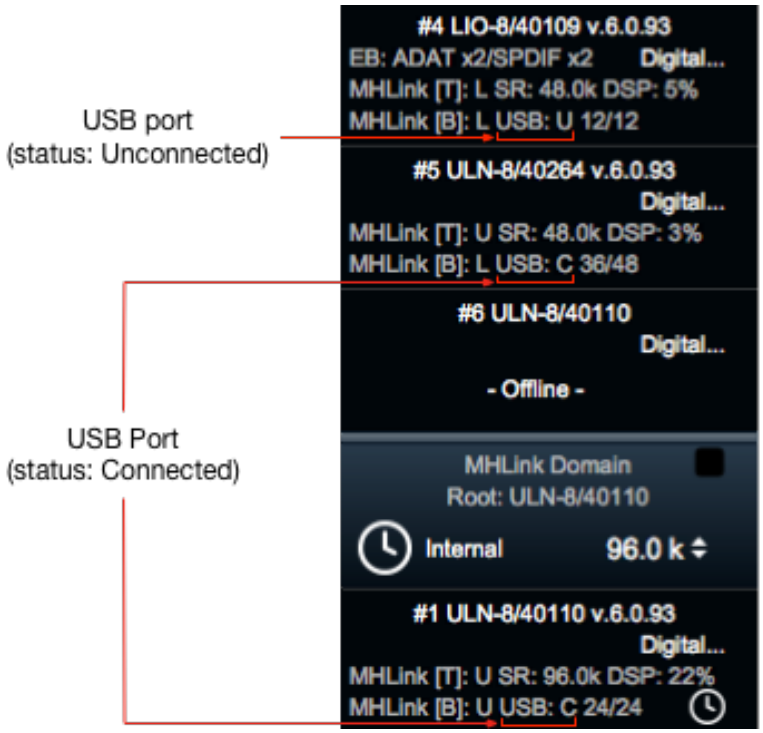
USB Port Status

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

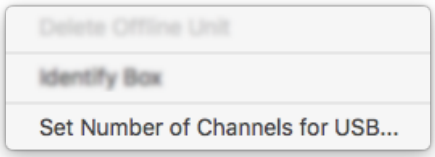


USB Port Status

- **"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

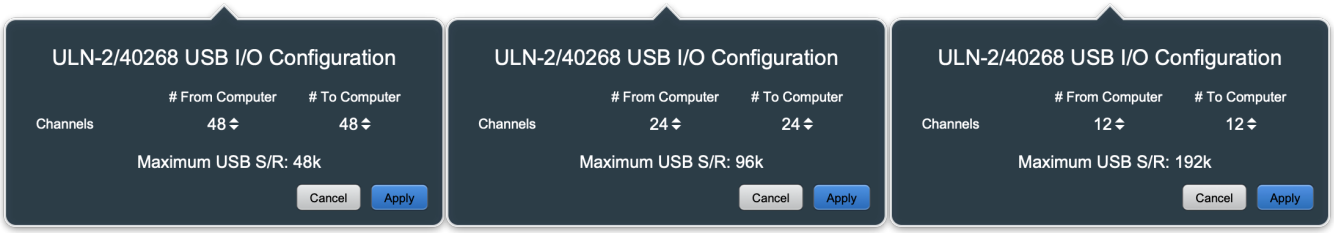
You can configure the USB port by "right-clicking" or "control-clicking" the "Unit Status Display" box. Select "Set Number of Channels for USB…" as shown below.



"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:

- 2-12 channels supports all sample rates through 176.4 and 192khz
- 2-24 channels supports 44.1khz through 96kHz sample rates
- 2-48 channels supports 44.1kHz and 48kHz



"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 Adapter' or its 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

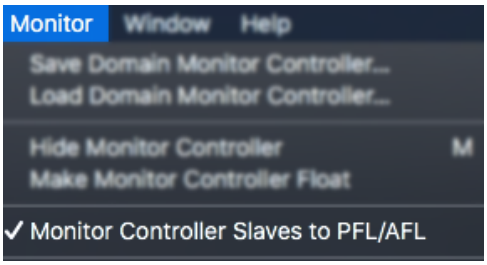
When you connect a new 3d box to your computer, MIOConsole3d syncs and registers the new box by model and serial number, checks the installed firmware, and sets up a default configuration for that box in the Mixer. The default configuration includes Input routes from the Host computer and the analog 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



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

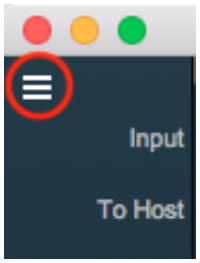
If the new box is an addition to an existing Domain, a mono analog Input strip will be created for each analog input channel on that box (eight strips for a 2882, LIO-8 or ULN-8, two strips for a ULN-2). New strips created by adding a new box will always appear at the far right of the existing Mixer 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.

MIOConsole3d 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.



Mixer Panes and the Hamburger Menus

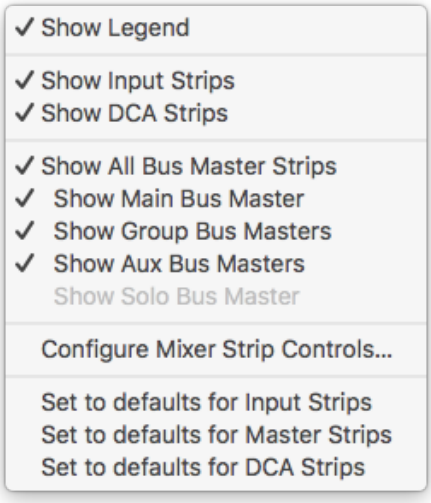
There are two more hamburger menu icons across the top of the mixer desk in the upper right corner. Each of these three hamburger menus are used to configure one of the three Mixer Panes provided to help organize your mixer. In the default view, only Mix Pane 1 is set up and visible - Mix Panes 2 and 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.



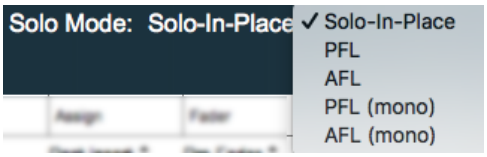
Mixer Pane Hamburger Menu

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.



"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.

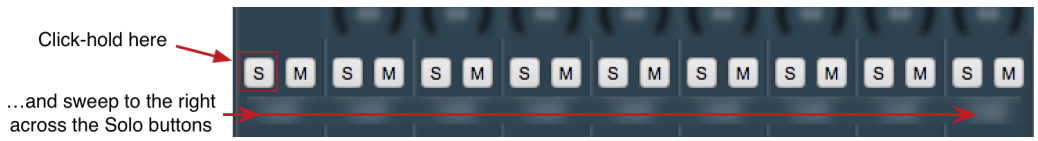
  By default, moving the scrollwheel while over the mixer desk surface will scroll the fader strips to the left or right. However, when holding the **Opt** (on Mac) or **Alt** (on PC) modifier key, the scrollwheel will move a fader or other control element.

  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.



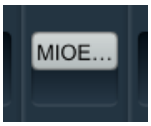
"Sweep" to toggle multiple buttons in one gesture

  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:



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).

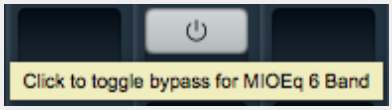
When you move the mouse over an inserted plug-in, the Insert label will change to show three control icons. The tooltips for each of these controls have been exposed in the example graphic below.



Inserted plug-in controls

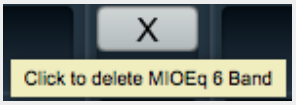
- The "On / Off" switch icon on the left is the plug-in Bypass. When Bypassed, the Insert button will turn yellow.
- Clicking the "…" icon in the middle opens the inserted plug-in editor UI. When the plug-in editor is open/visible, the Insert button will turn blue.
- Clicking the "up/down" arrows icon 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



"⌘-click" / <Command>-click to Bypass Insert

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



<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.

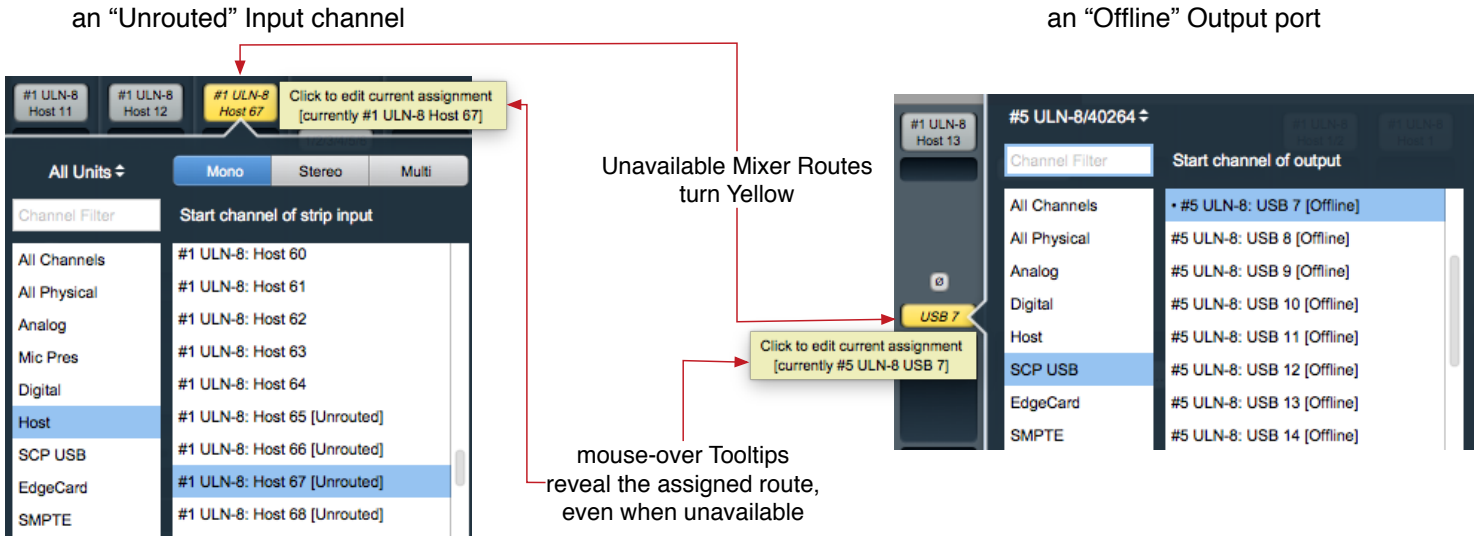
  NOTE: Due to the hardware-routing nature of I/O Inserts, copying routed I/O Inserts is not supported. To avoid potential routing conflicts, in all cases I/O Inserts are instantiated as unrouted.

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 MIOConsole3d '.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.



[Unrouted]:

Designates a computer port route assigned to a channel which is currently outside the I/O configuration of that port.
The port itself is active, but the assigned channel is currently unavailable.

[Offline]:

Designates a route to a physically unavailable port - usually either a 3d box that is disconnected or turned off, or an EdgeCard that has been removed or relocated

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 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.

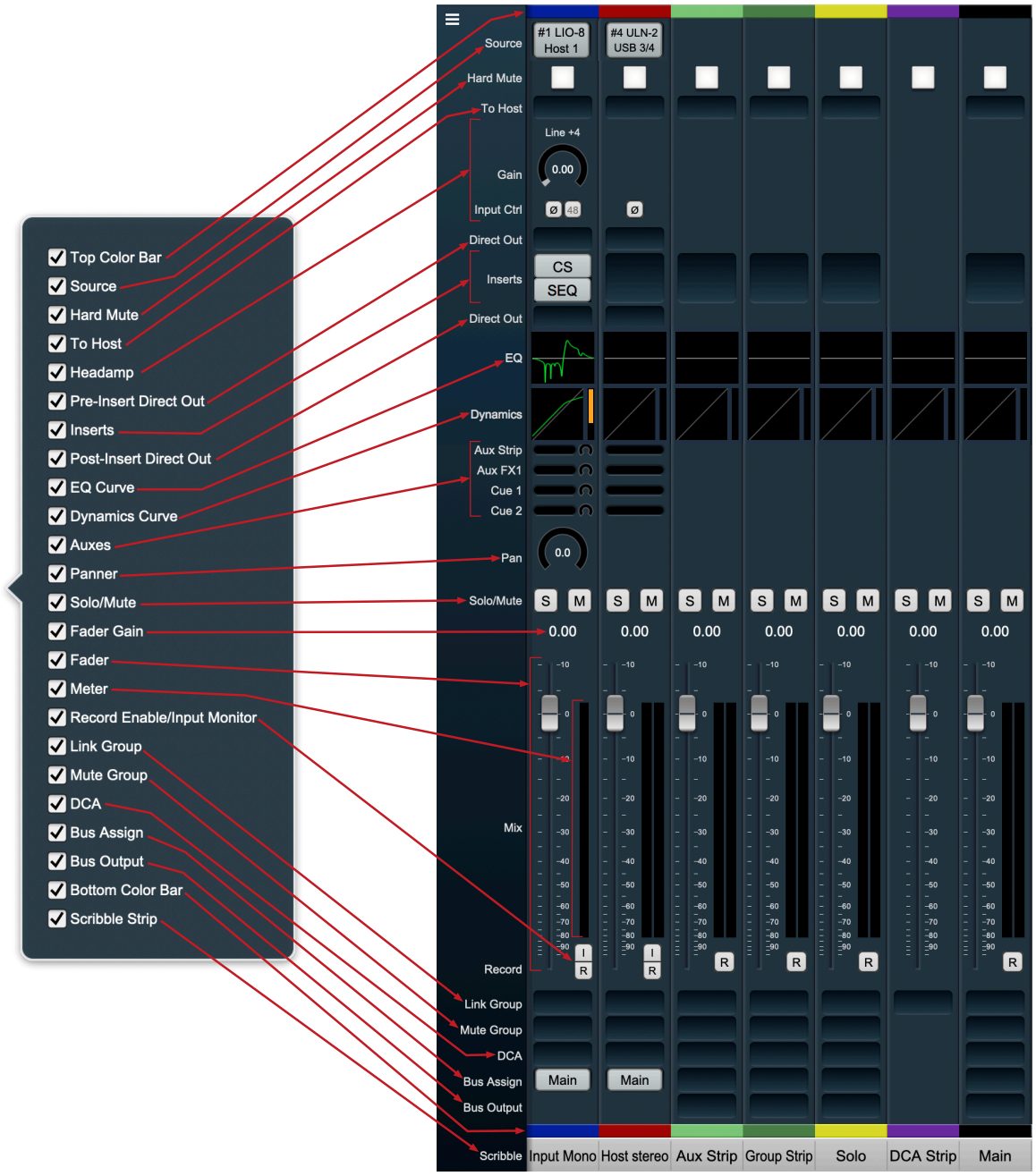
Launching MIOConsole3d with some hardware offline will generally initiate the Box Remapping page, where, if you wish, you can re-assign any existing routes to an appropriate I/O port in your currently-attached domain. This new Console configuration will have the string "[mapped]" added to the Console 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 selector 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.

The graphic on the next page shows the Configure Mixer Strips Controls menu on the left, with all Mixer strip control elements visible with arrows to their corresponding controls on the 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.



Mixer Strip: Color Bars, 'Source' selector, 'Hard Mute', 'To Host' returns, 'Headamp'

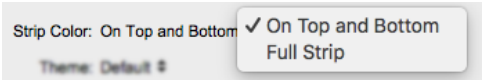
Now that we have a map, let's get down to it. From top to bottom, the Grand Tour begins…



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.



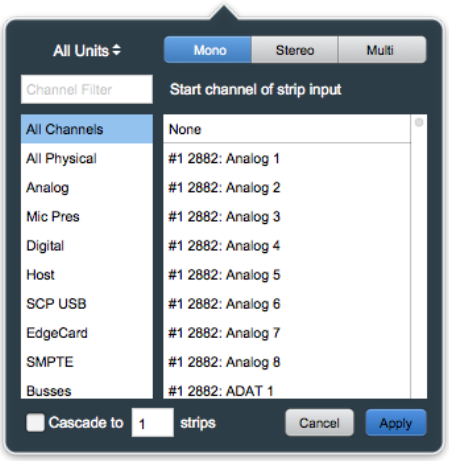
Console3d Prefs: Appearance: Strip Color configurator

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

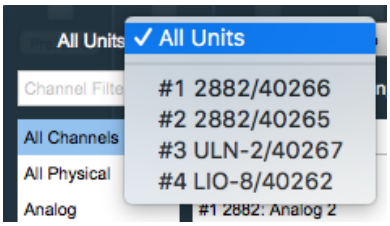
This control is available only on Input strips. When you select **Source**, it opens the input source routing menu (shown below). Input strips can source audio streams from practically anywhere in the Mixer - all analog and digital hardware inputs, bus outputs, computer connections over MHLink ethernet 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.



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.



"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 2.2. 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.



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.

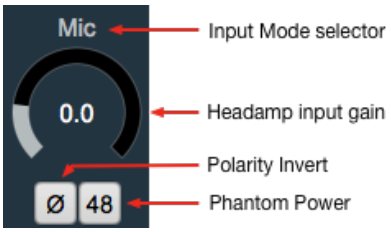
For sending specific channel routes back to the Host computer, use a Direct Out from an Input Strip, or "Bus Output" from a Bus Strip

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

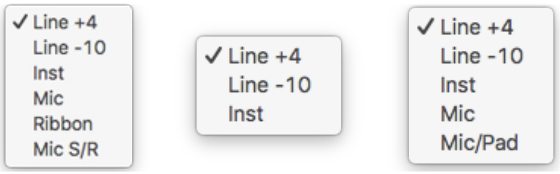
The **Headamp** section adapts to the type of audio signal selected in the Input strip "Source". For digital sources such as Host, USB and hardware digital audio streams, the headamp will only show the "Polarity Invert" button.

The analog Headamp has controls for signal Input Mode (level standard, impedance, etc.), gain/trim, phantom power toggling and polarity inversion.



Analog Headamp controls

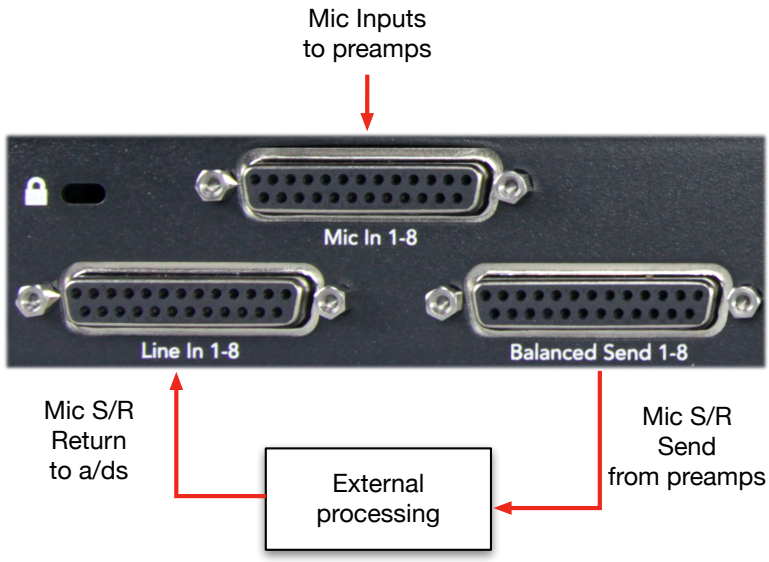
When an analog input source is selected, the headamp section expands to include controls specific to the particular MIO model routed by the Input strip.



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.



ULN-8 'Mic S/R' external process loop

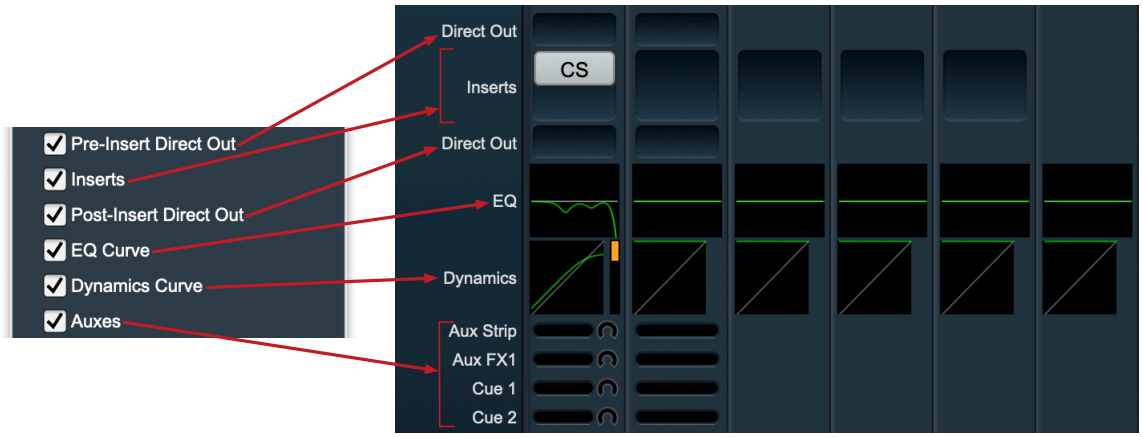
This graphic shows the signal path for the Mic S/R external processing loop. Note that the 'Balanced Send 1-8' mirrors whatever analog input you select in the Mixer Input strips, Mic, Line or any combination thereof. This allows you to use the sends as a splitter to feed a mixing console, recorder or 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, 'EQ Curve' display, 'Dynamics Curve' display, 'Aux Sends'

This section is concerned primarily with routing options from the Mixer Strip and processing within the Mixer Strip. an EQ curve showing the total applied EQ of all in-line equalizers, and a Dynamics Curve for a quick visual reference of the current dynamics processing.



Pre-Insert Direct Out, Inserts, Post-Insert Direct Out, EQ Curve, Dynamics Curve 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.

Note! Use Direct Outs to send to specific Host DAW input channels in any case where the automatic channel selection provided by the "To Host" routes does not suit your workflow.

The **Pre-Insert Direct Out** sends signal from the mixer strip prior to Insert processing. Post-Insert Direct Outs are found below the Insert processor section.

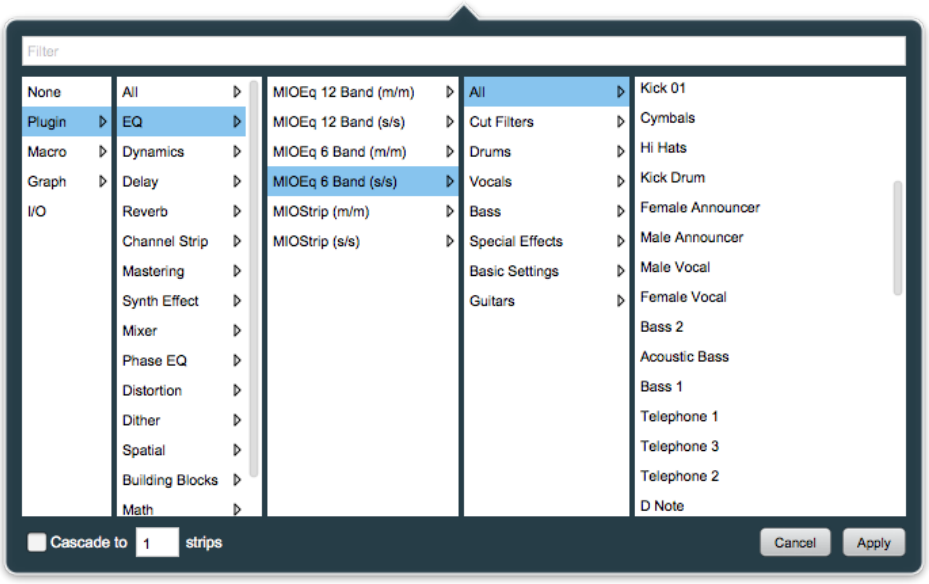
Inserts

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.



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 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 with signal paths possible only within the 3d DSP engine.

- **I/O**: Not so much a plug-in, but 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 mixer strip processing chain, send to any available hardware or computer output port and/or return from any hardware or computer input port.

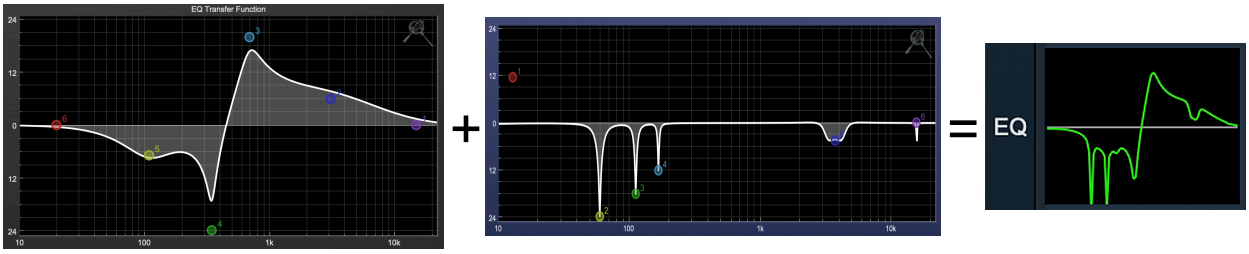
  Note: Since an I/O send/return insert sends audio to external systems outside the 3d Mixer engine and back, I/O loops can not be automatically latency-compensated.

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.

EQ Curve display

EQ Curve shows the EQ curve of all 3d DSP EQ processors active in the mixer strip.

If multiple EQ instances are active, the **EQ Curve** displays the sum total of all equalizer processors. This applies to any MIOConsole3d-native EQs including MHChannelStrip, MBEq, Sontec MES-432D9D, MHSonic EQ, etc.



ChannelStrip EQ plus Sonic EQ in EQ Curve display

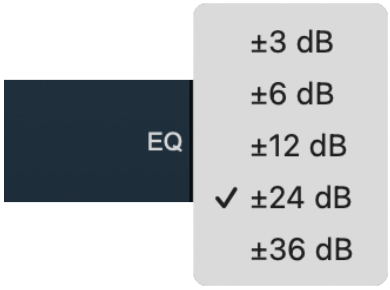
When the EQ Curve is empty (showing a grey flat line), **click** it to open your preferred EQ processor in the first available insert slot. MHChannelStrip is selected by default. You may change your preferred default EQ in the MIOConsole3d Interaction Preferences pane.

When there are EQs already present, click the EQ Curve display to open the UI of the topmost EQ inserted in the strip. Click again to open the second EQ, and so on.

Command-click the EQ Curve display to bypass all EQ processing in the strip. Repeat the keystroke to un-bypass.

Note that EQ processors running in DSP Graph inserts are not shown in the mixer strip EQ Curve display and can not be bypassed by the above key command.

Right-click the EQ label in the mixer strip legend to the left to change the EQ Curve graph gain range. This setting is global for all EQ Curve displays.



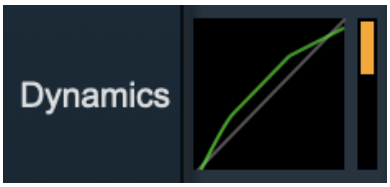
EQ Curve display range selector

The mixer strip EQ Curve display scale default is ±24dB.

Dynamics Curve display

Dynamics Curve adds a visual Dynamics Graph and Gain Reduction meter to each mixer strip.

If multiple dynamics processors are active in the strip, the Dynamics Curve displays the combined effect of those processors. This applies to any MIOConsole3d-native dynamics including MHChannelStrip, MHSuperGate, MIOComp, MIOLimit, etc.



MIO Mixer Dynamics Curve and Gain Reduction display

When the Dynamics Curve is empty (showing a grey gain line), **click** it to open your preferred dynamics processor in the first available insert slot. MHChannelStrip is selected by default. You may change your preferred default comp/limiter in the MIOConsole3d Interaction Preferences pane.

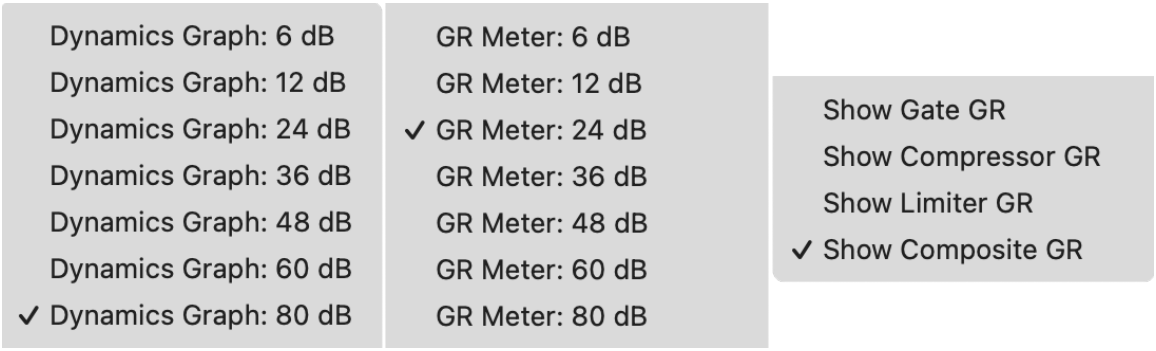
The green dynamics curve displays the combined gain control curve set by gate/expander, compressor, limiter and makeup gain stages within the strip.

The orange gain reduction meter to the right of the gain curve displays the combined gain reduction for all dynamics processors in the chain.

Command-click the Dynamics Curve display to bypass all dynamics processing in the strip. Repeat the keystroke to un-bypass.

Note that Dynamics processors running in DSP Graph inserts are not shown in the mixer strip Dynamics Curve display and can not be bypassed by the above key command.

Right-click the Dynamics label in the mixer strip legend to the left to show the Dynamics Graph and Gain Reduction display gain range selector menus.



Dynamics Curve display controls (default setting shown)

By default, the Dynamics Curve and Gain Reduction displays show the Composite results of all the processors in the chain. In cases of multiple dynamics processors, use the selections at the bottom of the menu to 'solo' the GR meter to show just the gate, compressor or limiter stages.

The above settings are global for all Dynamics Curve displays.

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.

It's worth repeating so... please **Note!** Use Direct Outs to send to specific Host DAW input channels in any case where the automatic channel selection provided by the "To Host" routes does not suit your workflow.

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.

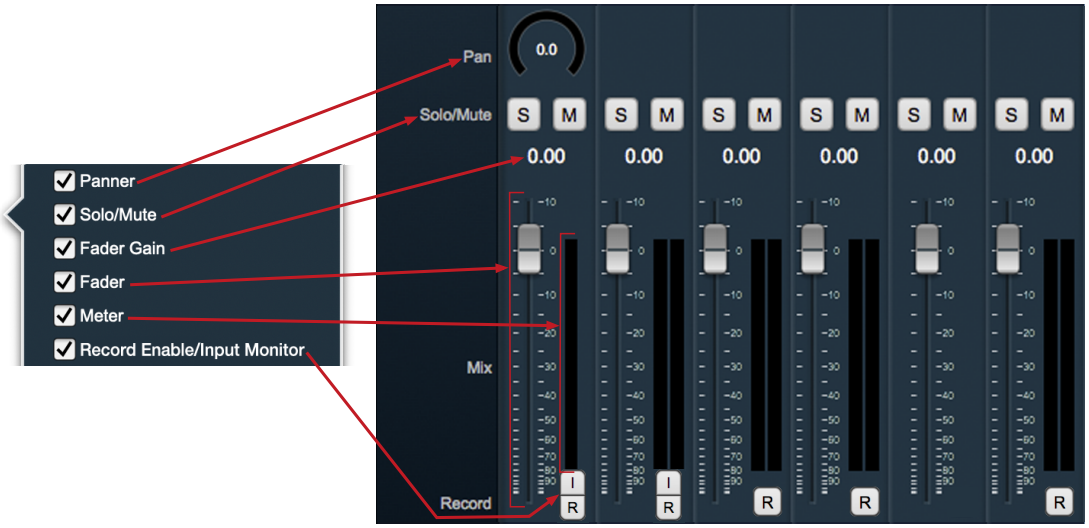
Aux mix buses can be named either in "Configure Mixer", or by double-clicking on their bus master scribble strip at the bottom of the Aux bus strip.

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', 'Input Monitor', '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.



Panner, Solo/Mute, Fader Gain, Fader, Meter, Input Monitor 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.

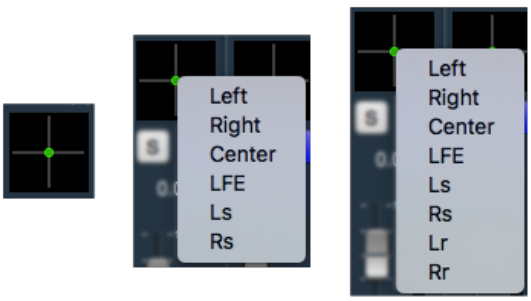
When the Main mix bus is set to stereo or LCR, you will get a stereo panner:



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:



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.

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.

When disengaged, mixer strip playback will follow the behaviors set in the Input Monitor Preferences pane.

Note: The Input control is shown/hidden in tandem with Record Enable.

Record Enable

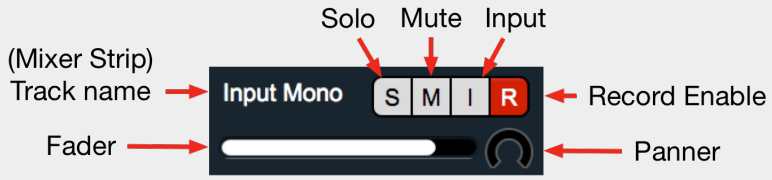
Use **Record Enable** ("R") to arm the strip for recording in Session. The Record Enable button will become solid red, and audio from this strip will now be captured when Record Start is triggered.

Record Arming a strip automatically creates a "Pre-Insert To Host" route from the 3d mix environment back to the computer to provide paths for the audio to be recorded in Session. Automatic routing uses the next available To Host channel, which is immediately visible in the To Host control at the top of the mixer strip.

Disarm Record Enable by clicking the "R" button again and the To Host route will be automatically released (unrouted).

Note: "To Host" routing/unrouting by the Record Enable control will remain automatic unless the "To Host" routing is manually altered, such as by changing the 'Pre Insert' route to 'Post Insert'. The MIOConsole3d mixer respects the change and disables automatic To Host routing for that mixer strip so as not to overwrite the users' setting.

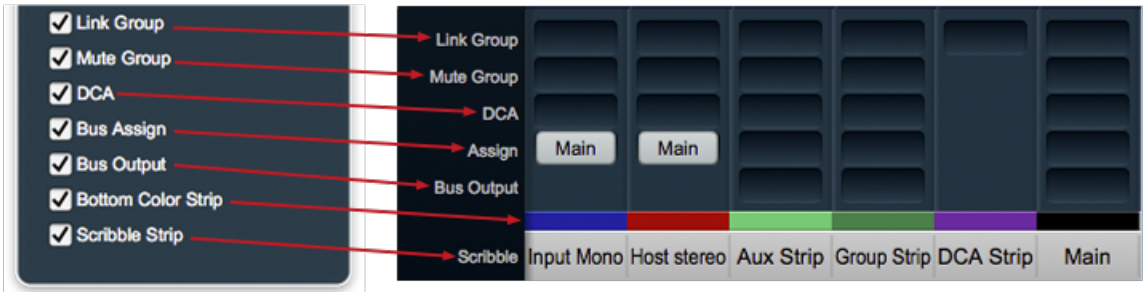
Note that in MIOConsole3d, many Fader section controls are also available in the Session Track headers, making it possible to control the key aspects of a capture-on-the-fly mix without having to swing back to the main Mixer desk:



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.

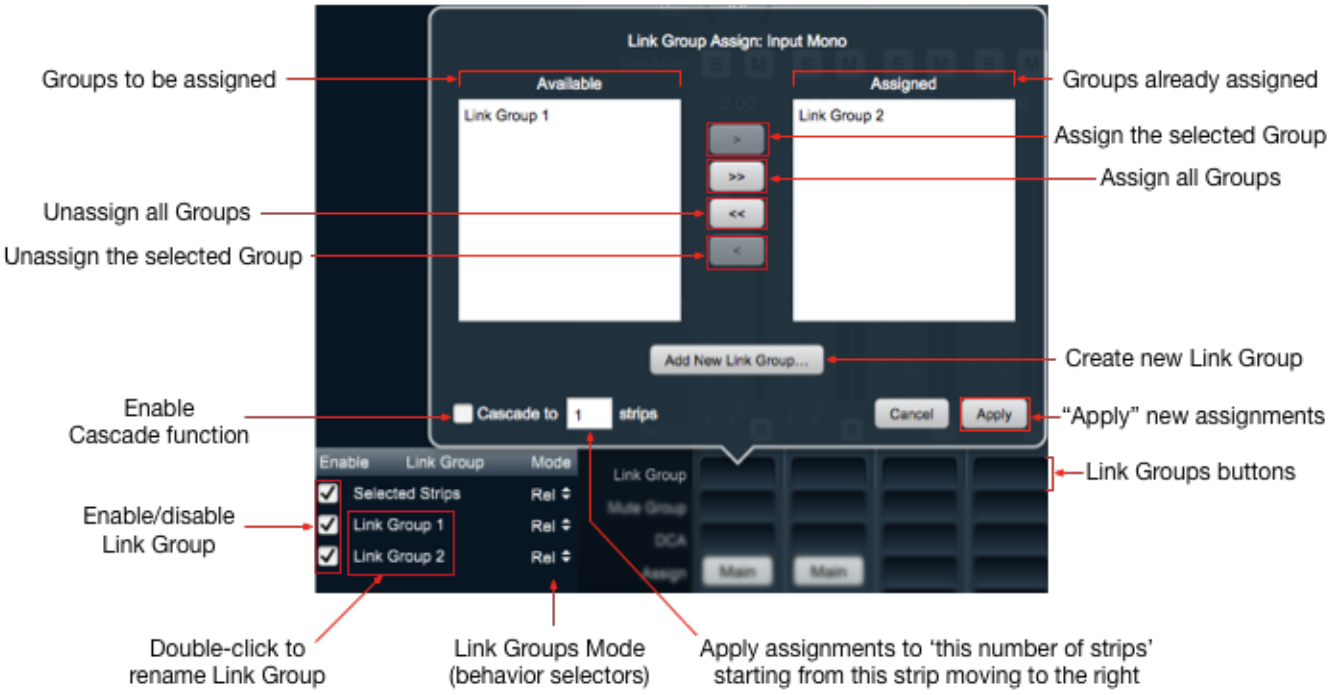


Link and Mute Group Assigns, DCA assigns, Bus Assign, Bus Output, Color bar and Scribble Strip

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.



'Link Group Assign' window with related Mixer Desk features overview

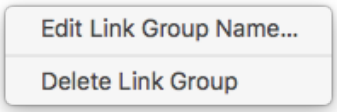
You start working with Link Groups by hitting the 'Link Groups' button at the bottom of any Input or Bus strip. This opens the 'Link Group Assign' window. The window is titled at the top with the window function and the name of the Mixer Strip from which you opened the window.

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:



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.

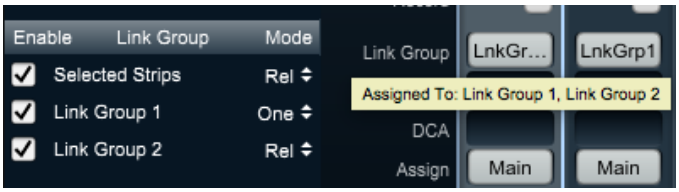
- The **Single Arrow** buttons in the 'Link Groups Assign' window are for moving Link Groups between the 'Available' and 'Assigned' lists in the direction the arrow points. Make a selection in the 'Available' list and click the "**>**" button to move your selection to 'Assigned'. Use the standard macOS "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'**:



'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:



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 2.3. '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 2.4. 'Selected Strips' Parameter Link map:

- **Hold the "Shift" key to momentarily defeat Link Groups and operate a fader or control independently.**

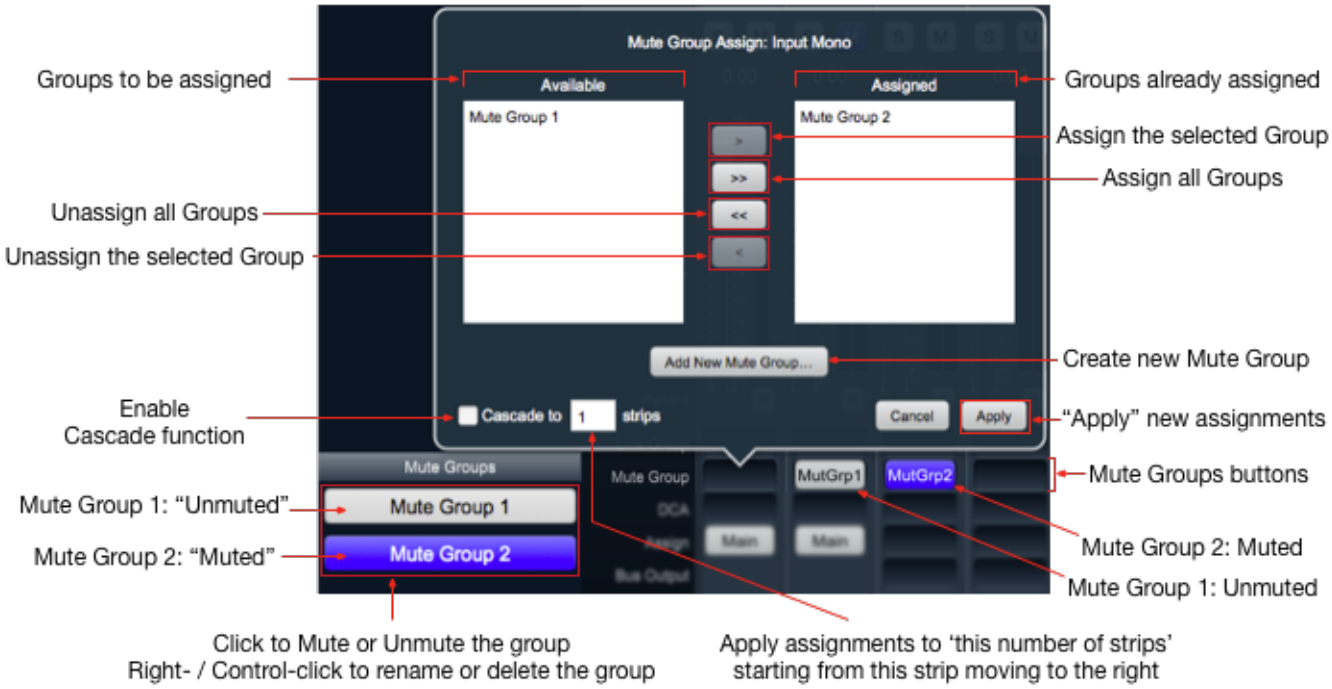
  Once you release the "Shift" key, the fader will re-link with the group in its new position.

  Hold Shift+Option to reset a single parameter within a Link Group to its default setting.

Mute Group

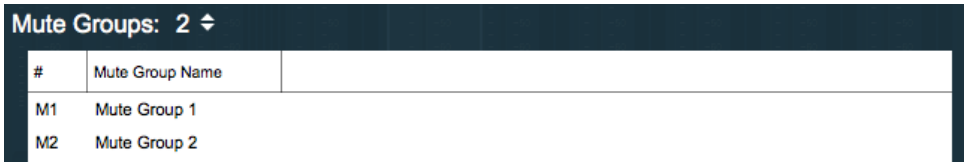
Clicking the 'Mute Group' button will open the 'Mute Group Assign' window. Here you can select one or multiple Mute Groups, create a new Mute Group, and apply your Mute Group selection across multiple strips in one shot.

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.



'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):



"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:



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.

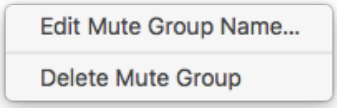
Mute Groups are engaged by hitting the Mute Group button in the Mute Groups list in the Status pane to the left of the Mixer desk. The advantage here is pretty obvious - those buttons can always be visible and accessible regardless of your Mixer strip layout.

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:



Status Pane: Mute Groups List: 'Edit Mute Group Name…', 'Delete Mute Group' submenu

DCAs

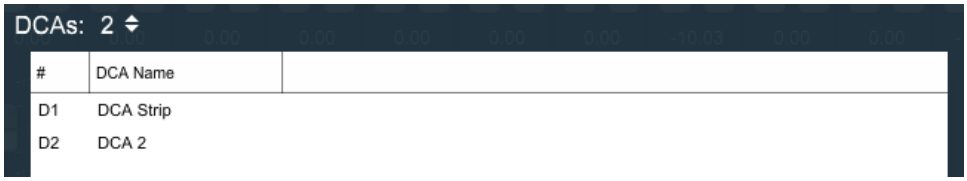
Digitally Controlled Amplifier strips in MIOConsole3d work the same as DCAs on a hardware digital console. No audio signal passes through a DCA strip, the DCA fader simply tells faders under its control to move up or down by the amount the DCA fader knob moves. This gain change is not reflected by the 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):

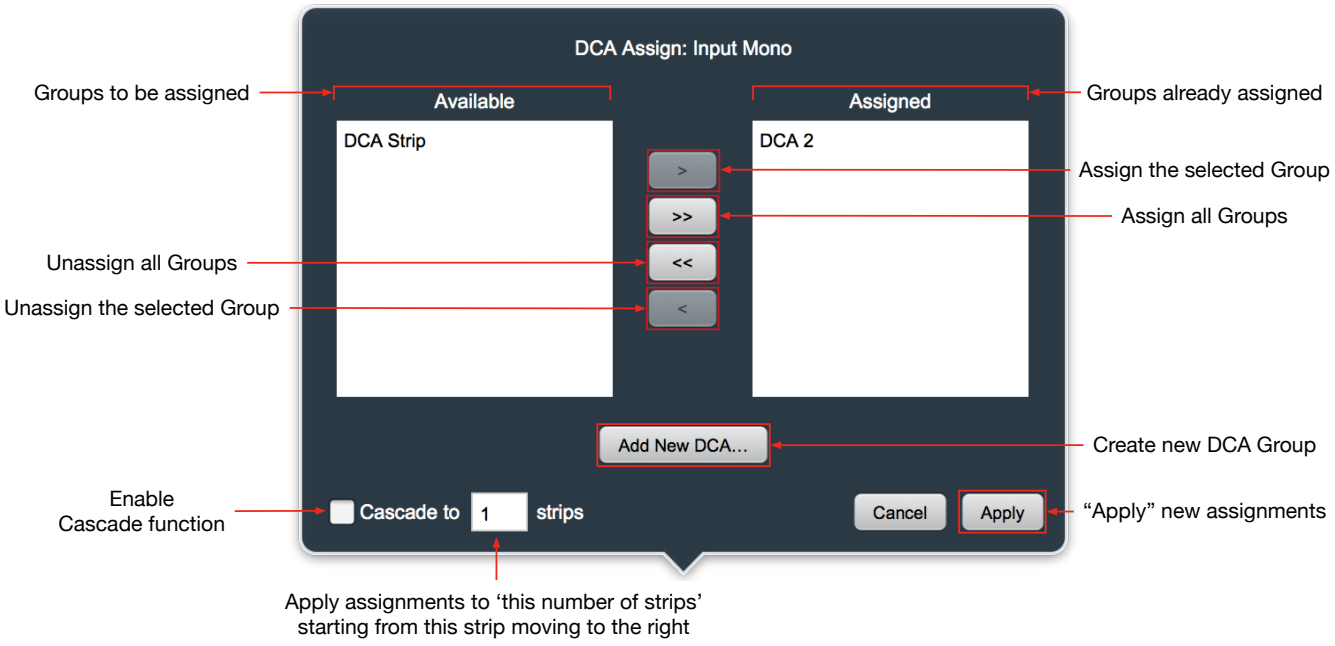


"Configure Mixer" DCAs Assign window

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.



'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.

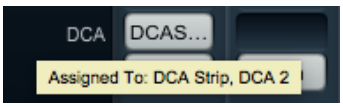
- Double-click a DCA to move it immediately to the opposite list.

- 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:

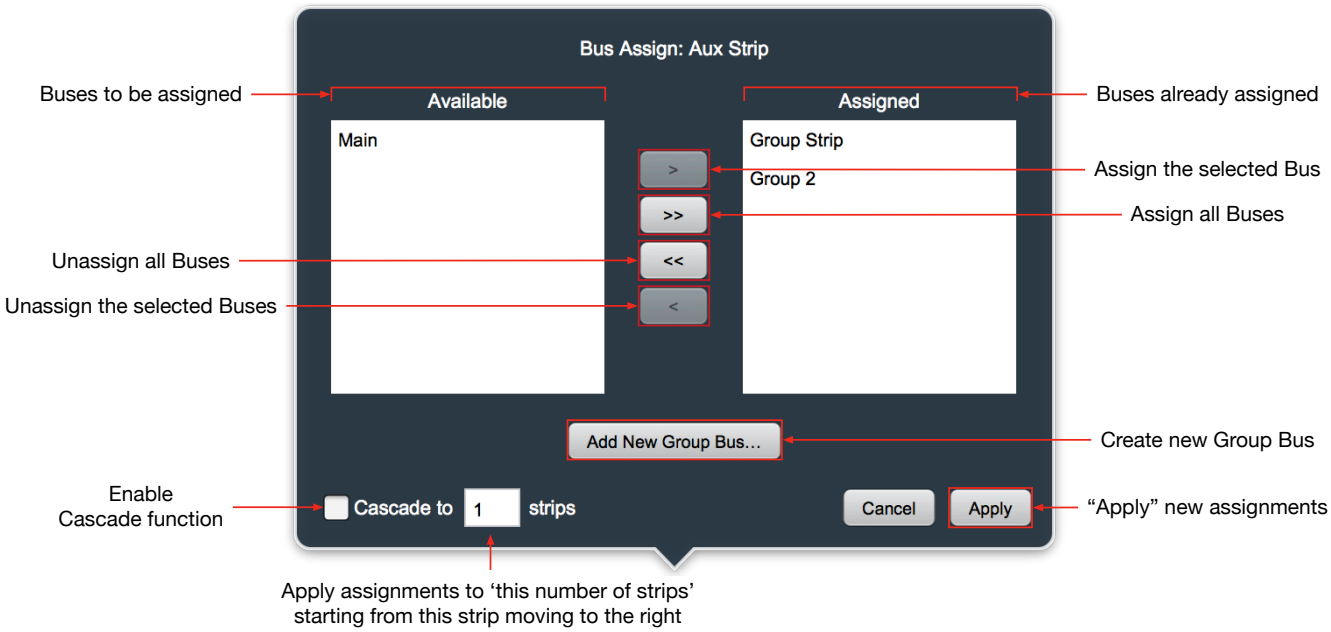


Tooltip revealing multiple DCA assignments

- 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.



'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.

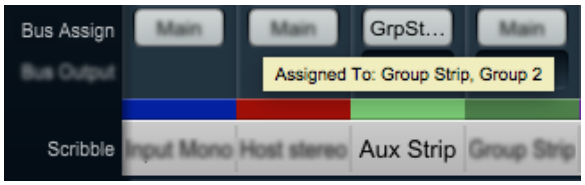
- Double-click a Bus to move it immediately to the opposite list.

- 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:



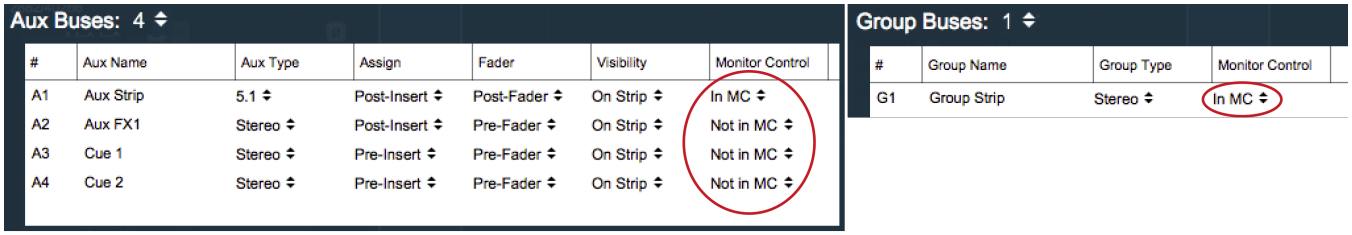
Tooltip revealing multiple Group Bus assignments

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.

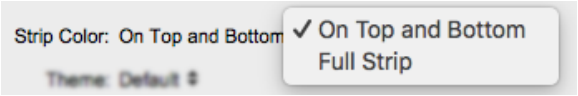


"Configure Mixer": Bus Strip routing to Monitor/Cue Controllers

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.

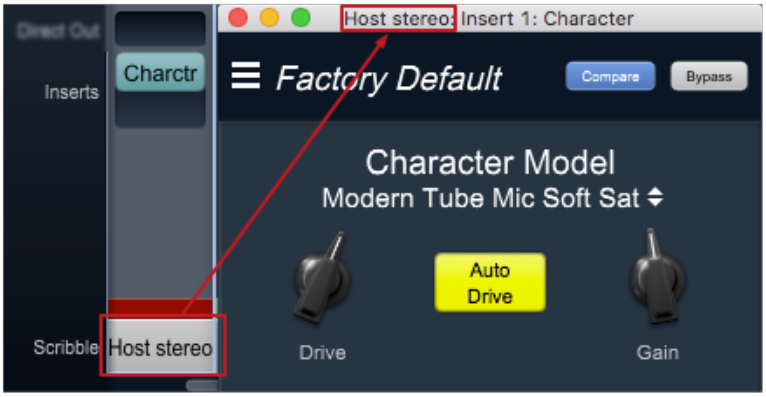


Console3d Prefs: Appearance: Strip Color configurator

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

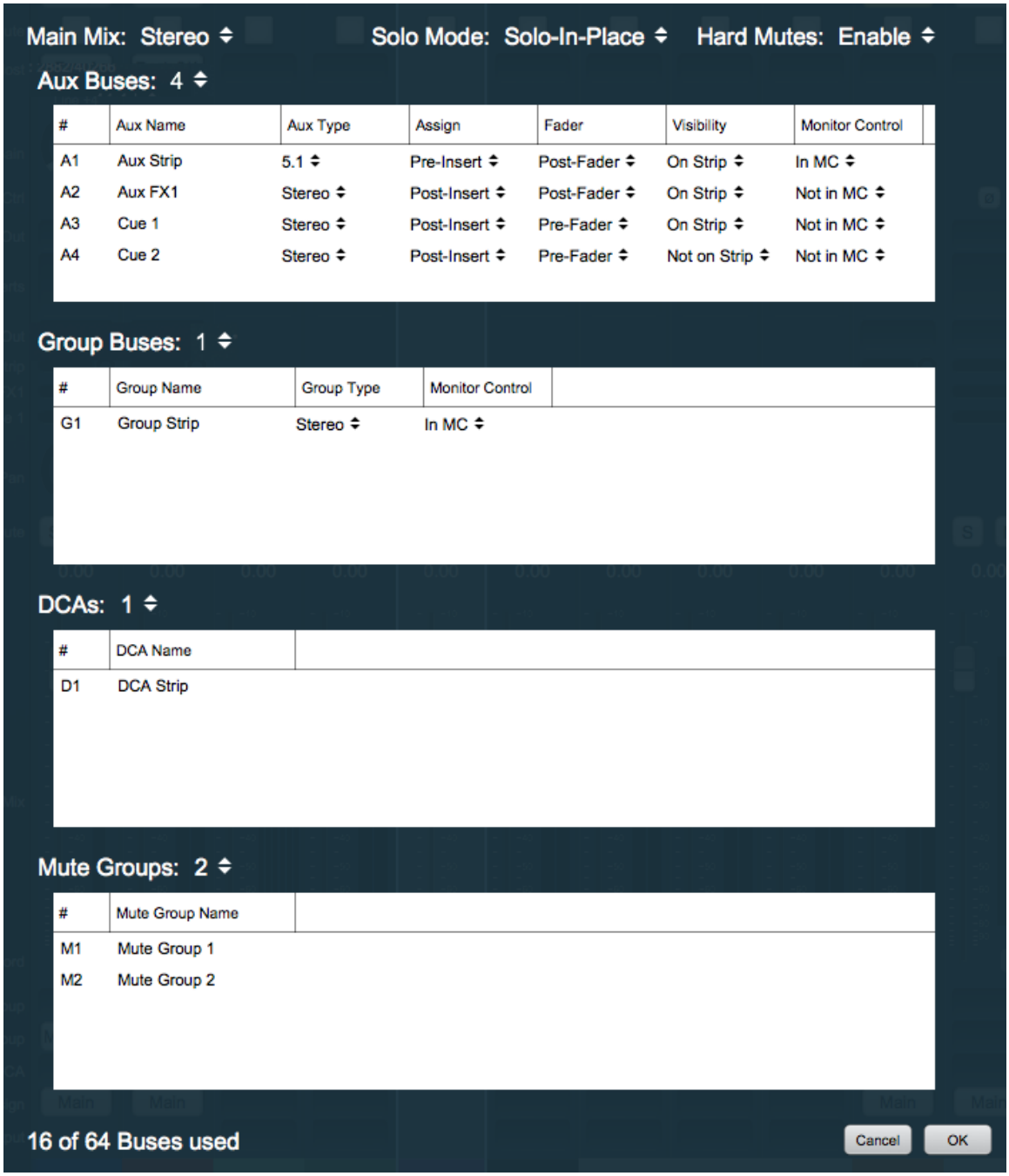
Double-click to name any Mixer strip. The name you enter in the Scribble Strip will be propagated through-out the 3d user interface wherever a Mixer Strip name is found, including plug-in headers, Link Group, Mute Group, DCA and Bus Assign selection windows.



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



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.



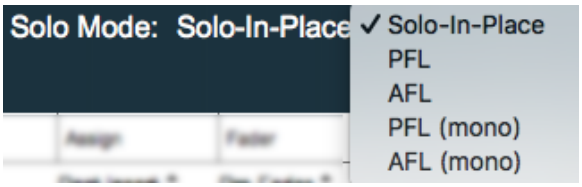
Main Mix configuration menu

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.



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

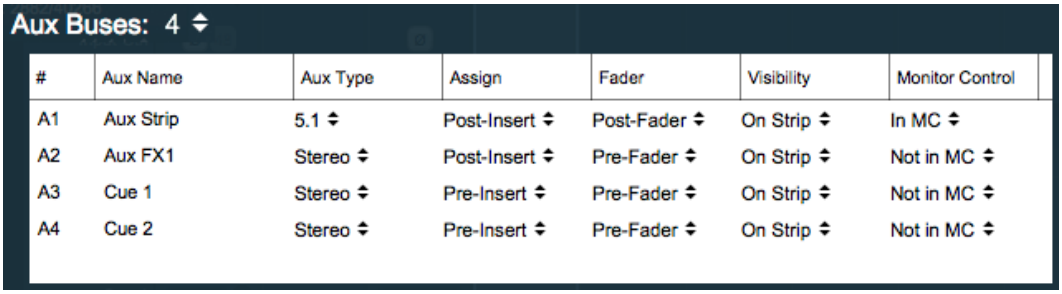
'Hard Mute' controls are Disabled and hidden from Mixer strips by default. Enabling Hard Mutes places a big white square button at the top of every Input and Bus strip in the mixer, just below the Input routing selector. 'Hard Mutes', when enabled, will kill *all* audio routes through the strip, including 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:" 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 2.5. 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 "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 2.6. 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 2.7. 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.

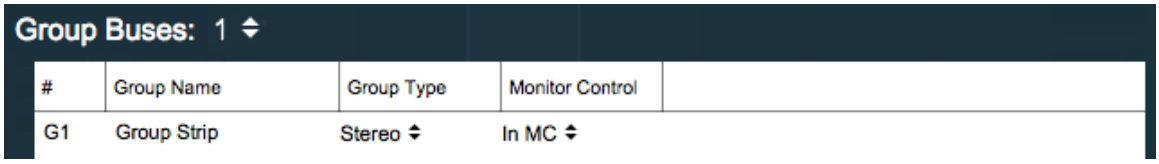
That said, it is the way things like Pro Tools work, so it is also important to not lose the ability to support pre-mute, pre-fader sends to aux buses.

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.



"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.

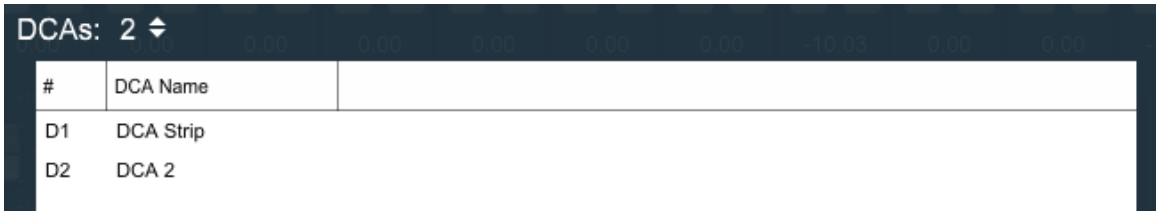
Note that switching the 'Solo Mode' to PFL/AFL will add the stereo PFL/AFL Solo bus strip to the mixer, using another two bus channels.



'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.



"DCAs:" configuration menu

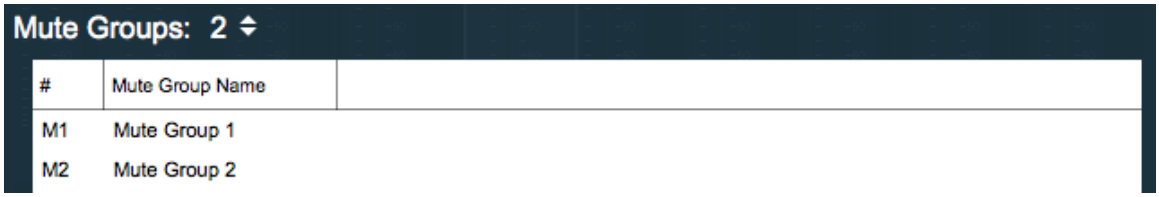
Click in the **DCA Name** field to enter a name for the DCA.

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.



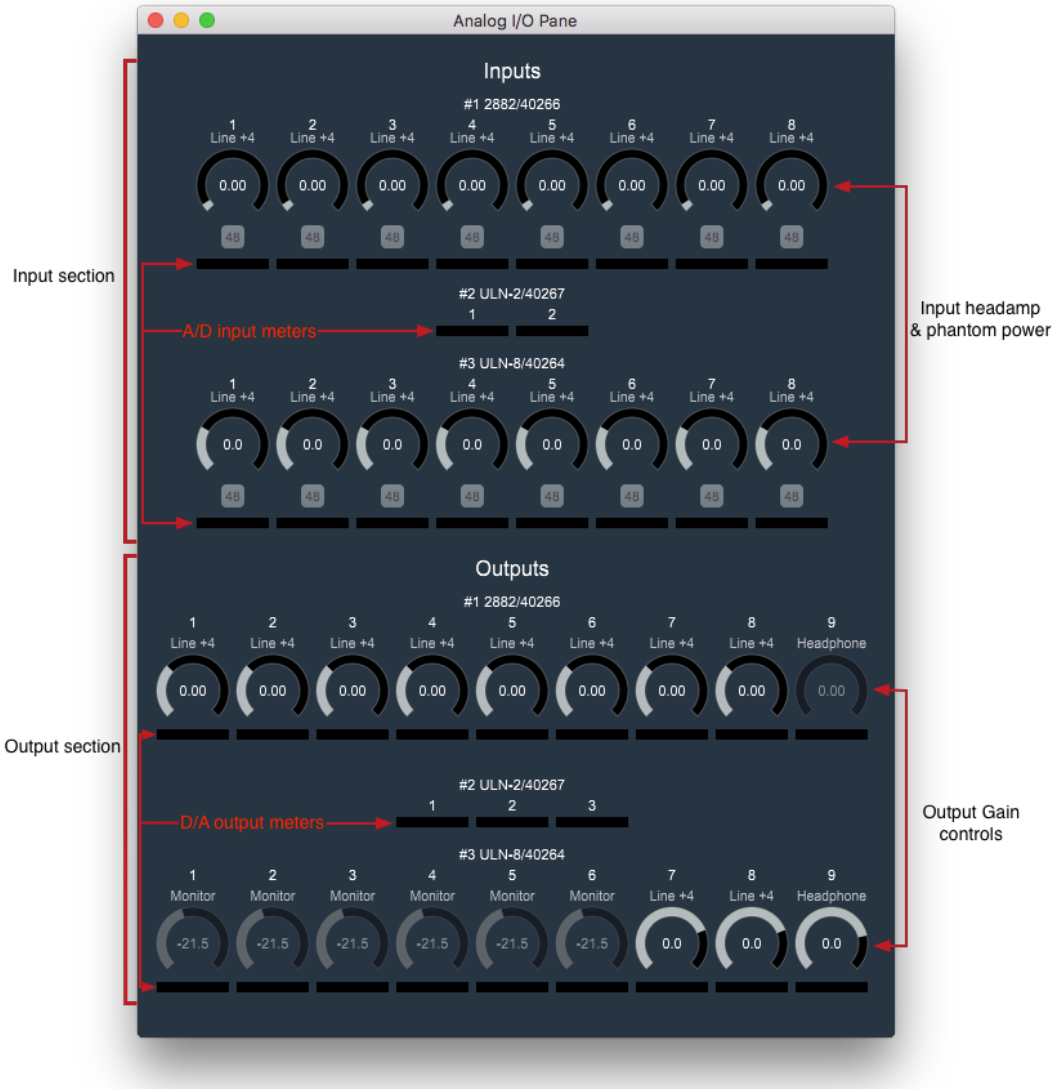
"Mute Groups:" configuration menu

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.



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.

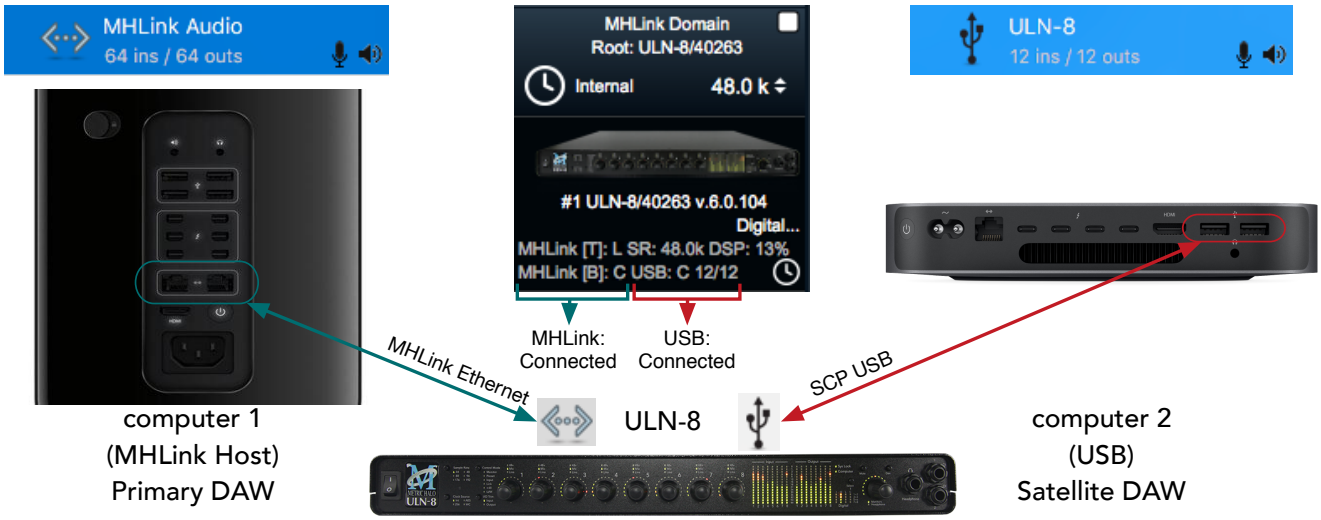
Since the ULN-2 has discrete analog controls on its front panel, there are no software gain controls.

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:



MHLink Host with Satellite computer

- On the left is the primary MHLink host computer (a Mac Pro 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.

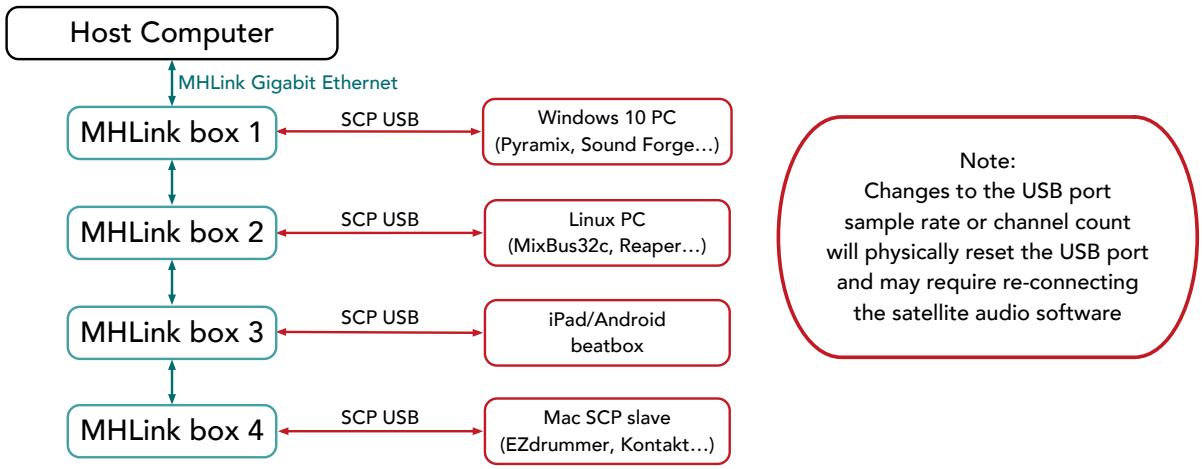
A secondary audio application on Computer 2 connect to the USB "ULN-8" port with 12 channels of I/O.

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.

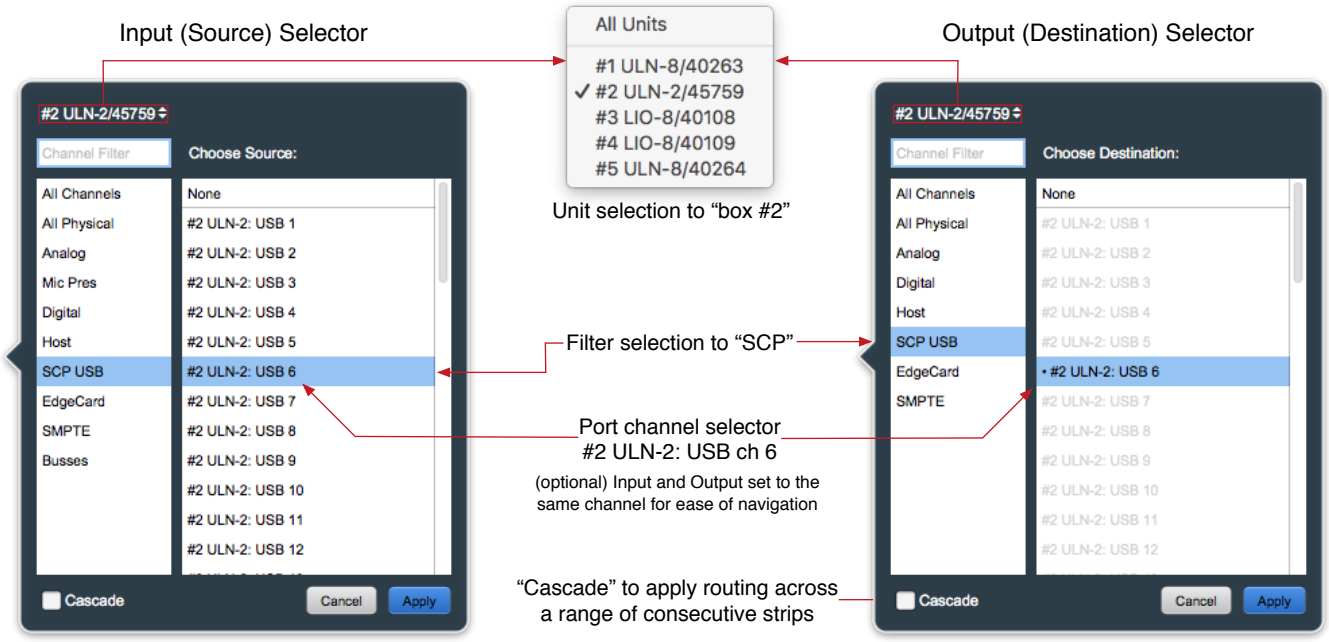
The 3d Domain *is* after all a fully-functional hardware digital mixing/routing/processing environment - the only difference is you are controlling the hardware mixer with a computer UI (MIOConsole3d, with maybe a control surface or two).

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:



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.



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.

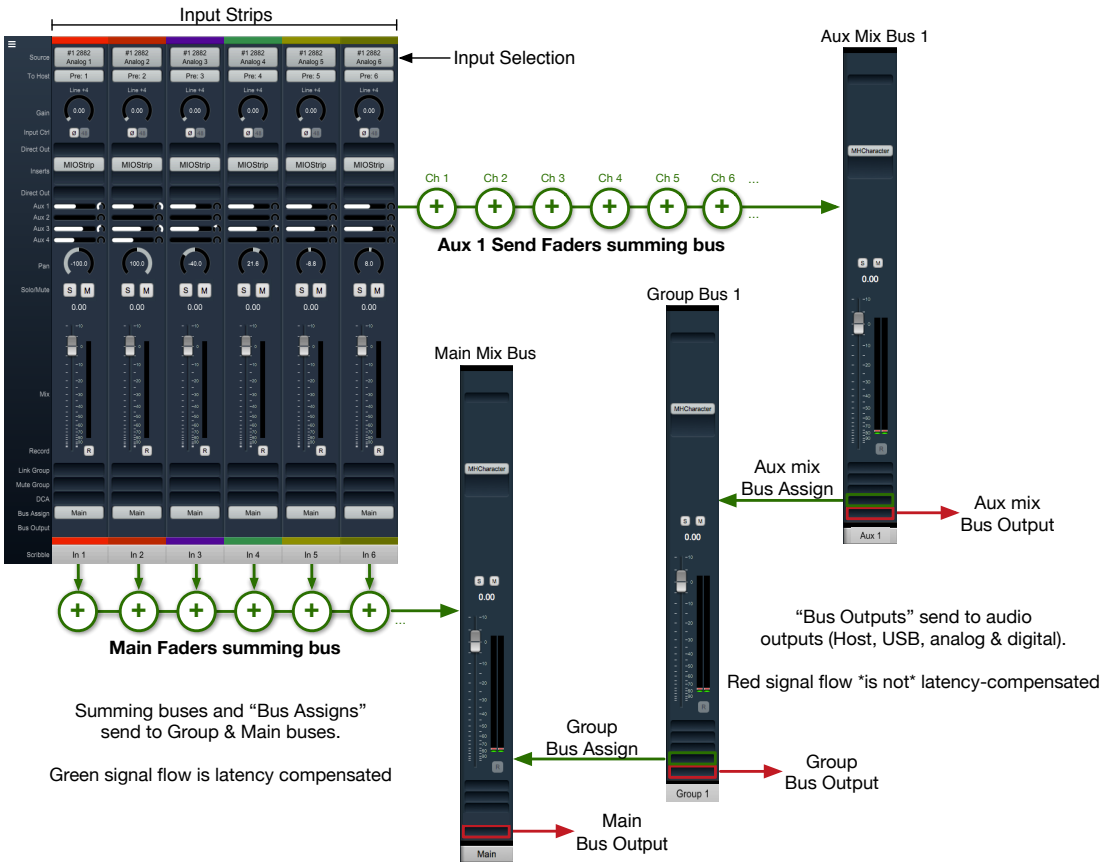
3. 3d Mixer: Bus Structure, Plug-ins, and Processing

Signal Flow: Input Strips, Bus Strips and the Main Bus

Every Input strip, Aux Bus, Group Bus and the Main Bus strip supports in-line plug-in Insert 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: Input strip -> Aux Bus strip -> Group strip -> Main Bus. Looping through multiple Group buses on the way to the Main Bus is also latency compensated, even with Inserts on each bus.



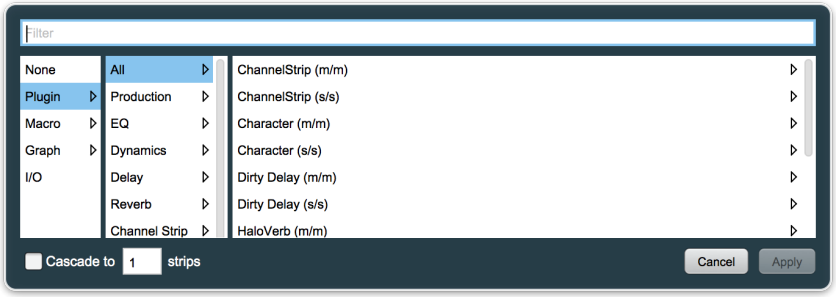
Basic 3d Mixer signal flow

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 which leaves the mixer and returns via an Input strip (shown in red) can not be automatically latency compensated. This includes the I/O routing inserts, and any other route that loops out to external devices via analog, digital, MHLink or USB.

Inserts

Click on any empty stereo insert slot, and you will see the 3d Insert selector menu:



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.

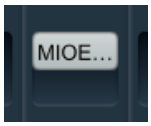
If you are working with a multi-channel strip, only plug-ins that make sense for the number of channels of the strip will also be available. For example, with a stereo strip, you will see both the mono and stereo versions of the MIOComp and MIOLimit dynamics processors. You can select the version that 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:



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.

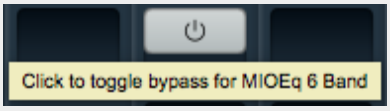


Inserted plug-in controls

- 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.

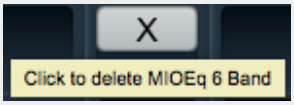
  Note: Normally opening a new plug-in window will replace any currently-open plug-in UI, such that only one is open at a time. <Shift>-click to open a new plug-in editor window while keeping existing plug-in UIs open.
- Clicking the "up/down" arrows icon at the right opens the Insert selector window, 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



"⌘-click" / <Command>-click to Bypass Insert

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



<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.



"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.

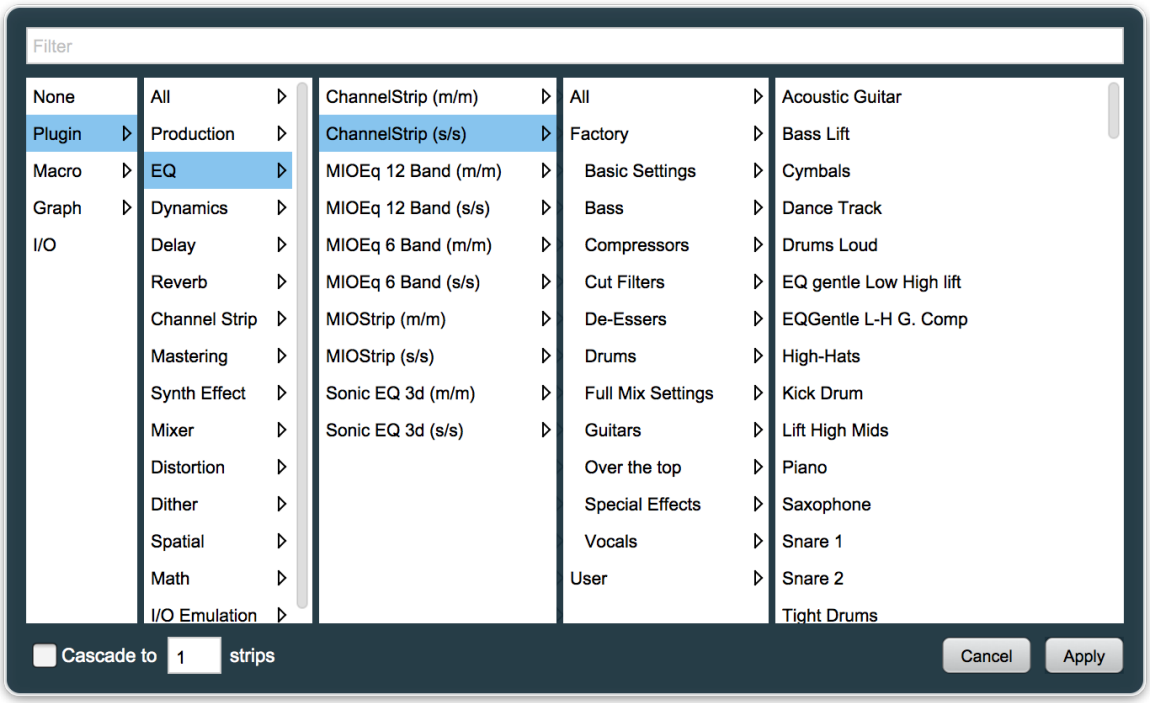
The move works with Polarity Invert, Solo, Mute and Record Enable buttons.

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



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.

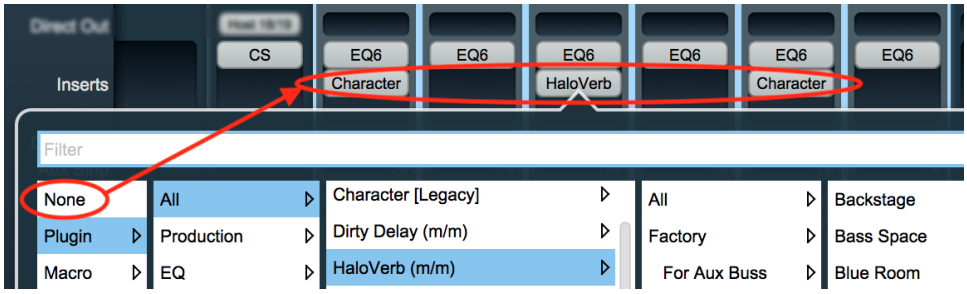
Insert Types

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.



"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 Bundle, third party special projects 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.

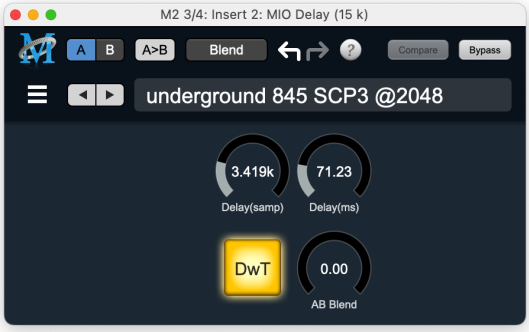
'I/O' insert "Send" routes are active from the moment they are routed. "Bypassing" an 'I/O' insert defeats the "Return" path, but the "Send" is always sending. This lets you use 'I/O' inserts as extra Direct Outs, with an optional switchable alternate input path in the strip.

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:



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.



MIOConsole3d Channelstrip UI

Plug-in Headers

All Metric Halo and Make Believe family plug-ins display the MH header bar at the top of the plug-in window. This header bar allows you to organize and access all your presets across all supported plug-in formats on Mac, Windows and Metric Halo hardware DSP via MIOConsole3d.

It is especially useful in that, regardless of platform, it provides a straightforward, powerful and consistent processing workflow wherever you might be working.

With the constant evolution of computer capabilities opening the door to new production techniques and music delivery formats, the differences between DAW software workflows have become ever more diverse. Many of the major DAWs provide their own plug-in headers within every plug-in instance window, providing their own feature set catering specifically to their internal workflow.

Conversely, other equally popular DAWs provide no added feature support for plug-ins (such as plug-in parameter Undo/Redo), opting instead to insert plugs as a straight processing block.

The MH family plug-in header bar bridges that gap by offering the most asked for plug-in functions in a simple GUI, making all of our plug-ins functions and their presets available to every user on every platform.

The plug-in header bar has two rows of controls, with the Metric Halo logo icon at the top row left.

The top row handles plug-in operations, including update notification and download, access to MH online resources, GUI preferences, tooltip help, A/B parameter snapshots, snapshot Blend, plug-in Undo/Redo, Compare and soft Bypass.

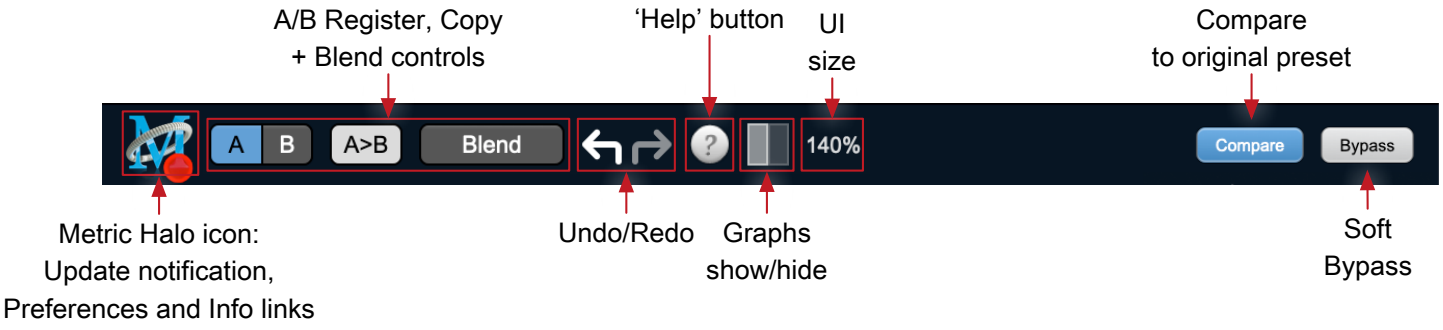


MH ChannelStrip Plug-in header (default setting: eq and dynamics graph display closed)

The lower row (with the 'hamburger' menu icon at the left) is all about preset management.

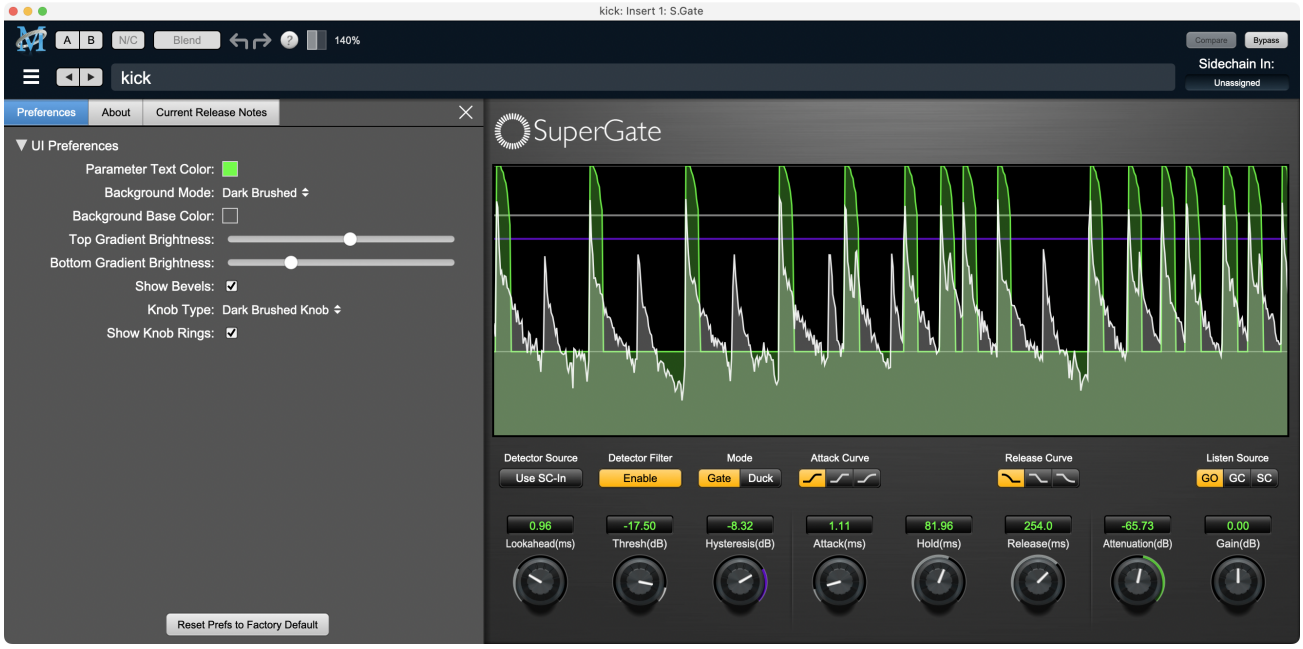
The menu includes organizational and preset import/export features for sharing saved preset files with the outside world, and restoring presets from early days MIO and MH plug-in versions.

Plug-In Header: Top Row



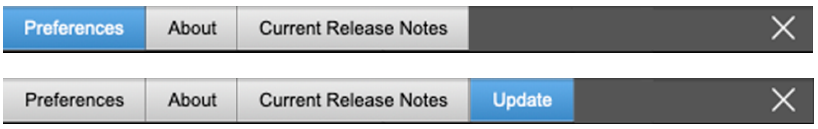
Metric Halo Header Icon

Clicking the MH icon expands the entire plug-in window to the right and opens a multi-function control sidebar with the Preferences tab in focus:



Preferences tab (example shown is SuperGate with sidechain filter graph closed)

This sidebar has context-sensitive tabs across the top for GUI **Preferences** (shown above), **About**, and **Current Release Notes**.

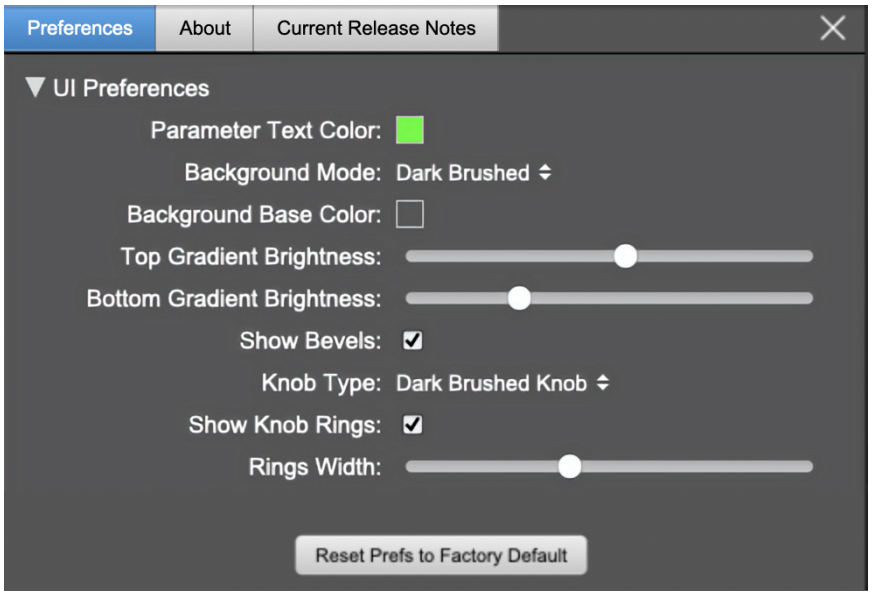


Plug-in control pane tabs

When updates are available for download from Metric Halo, the MH icon will sport a blatant red dot (shown in the header map at the top of this page) and an **Update** tab is added to the sidebar. These tabs are dynamic by design, and additional tabs may appear as new content becomes available.

To close the sidebar, click the MH icon again, or the "**X**" at the right edge of the tab bar.

Preferences tab



standard plug-in UI Preferences

The Preferences tab provides controls to modify the color scheme of text and graphical elements.

Initially added in response to requests from users with various visual impairments, in practice the ability to match the color of the plug-in to the mixer strip where it resides makes for a nice improvement in workflow.

Shown above are the basic parameter controls used by the majority of Production Bundle plug-ins.

More complex processors, such as ChannelStrip with multiband EQ and dynamics graph grids, will have additional preferences to further customize those UI elements. Processors with simpler graphics will have fewer UI preference controls.

UI parameter changes are live, so you can see changes to the plug-in graphics as you make them. This includes colors, gradients, graphic element show/hide, knob types and bevels.

Clicking color selection swatches opens the standard OS color pickers window with the eyedropper tool for sampling any color visible on your display.

> **Note**: The color scheme you set will be saved to the computer and automatically applied to the next instance of that plug-in type of whatever native format, even on another DAW.
>
> You can always revert to the factory default settings by clicking the **Reset Prefs to Factory Default** button at the bottom of the Preferences pane.

Complete Plug-in Preferences List

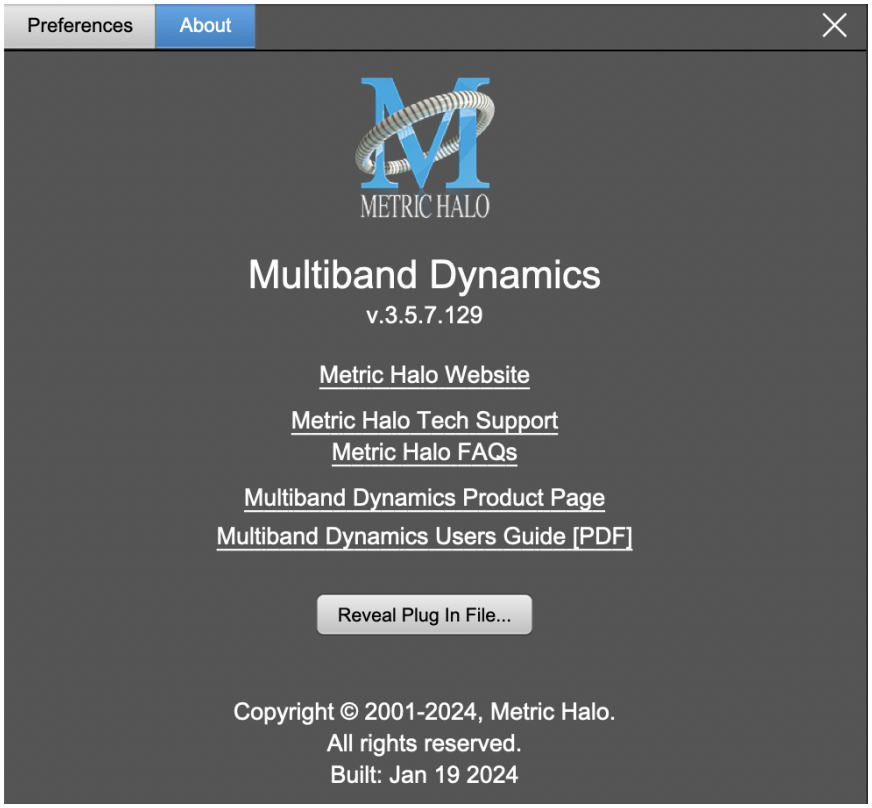
The following is a breakdown of all the editable plug-in display prefs currently in play.

Two items to note:

1. **Background Base Color** affects only *Flat* and *Gradient* **Background Mode**s.
2. *Top* and *Bottom* **Gradient Brightness** controls affect only the two *Gradient* **Background Mode**s.

- **UI Preferences**
  - **Parameter Text Color** sets the color of all click-to-enter numeric readouts.
  - **Background Mode** selects the background type and texture:
    1. **Light Brushed**: a light brushed metallic facing with mild gradient lighting (the "vintage" Production Bundle faceplate)
    2. **Dark Brushed**: a dark neutral brushed metal facing with a bit stronger gradient and lighting
    3. **Flat Color**: select the background color with the **Background Base Color** picker.
    4. **Gradient Color**: select the background color with the **Background Base Color** picker and adjust Top Gradient Brightness and Bottom Gradient Brightness to taste
    5. **Grad w/Hits**: same as **Gradient Color** with two mild rack light sources from above and one centered below
  - **Background Base Color** when in color background modes, sets the background color
  - **Top Gradient Brightness** when in gradient background modes, sets the top gradient brightness
  - **Bottom Gradient Brightness** when in gradient background modes, sets the bottom gradient brightness
  - **Show Bevels** show/hide shaded edge and processor section separator bevels
  - **Knob Type** select from six "physical" knob types, or no knob at all
  - **Show Knob Rings** show/hide the colored rings surrounding parameter knobs
  - **Show Band Swatches** show/hide the color swatches above each EQ band on/off switch (EQ Band Color controls listed below work with swatches on or off)
  - **Show Graph Grids** show/hide the numeric grids in dynamics and EQ graph displays
  - **Graph Grid Color** set the color of the numeric grids in dynamics and EQ graph displays
  - **Graphs Background Color** set the background color of the dynamics and EQ graph displays
  - **Fill EQ Curves** shades the area of deviation of each eq band from 0dB reference (makes it easier to see the effect of each filter band amongst multiple active filters)
- **EQ Colors**
  - **Overall EQ Curve Color** sets the color of the aggregate EQ energy curve (the total sum effect of all active bands)
  - **Band 1 Color** sets the color of filter band 1
  - **Band 2 Color** sets the color of filter band 2
  - **Band 3 Color** sets the color of filter band 3
  - **Band 4 Color** sets the color of filter band 4
  - **Band 5 Color** sets the color of filter band 5
  - **Band 6 Color** sets the color of filter band 6
- **Sidechain EQ Colors**
  - **Gate EQ Color** sets the color of Gate Sidechain filter band
  - **Comp EQ Color** sets the color of Comp Sidechain filter band

About tab



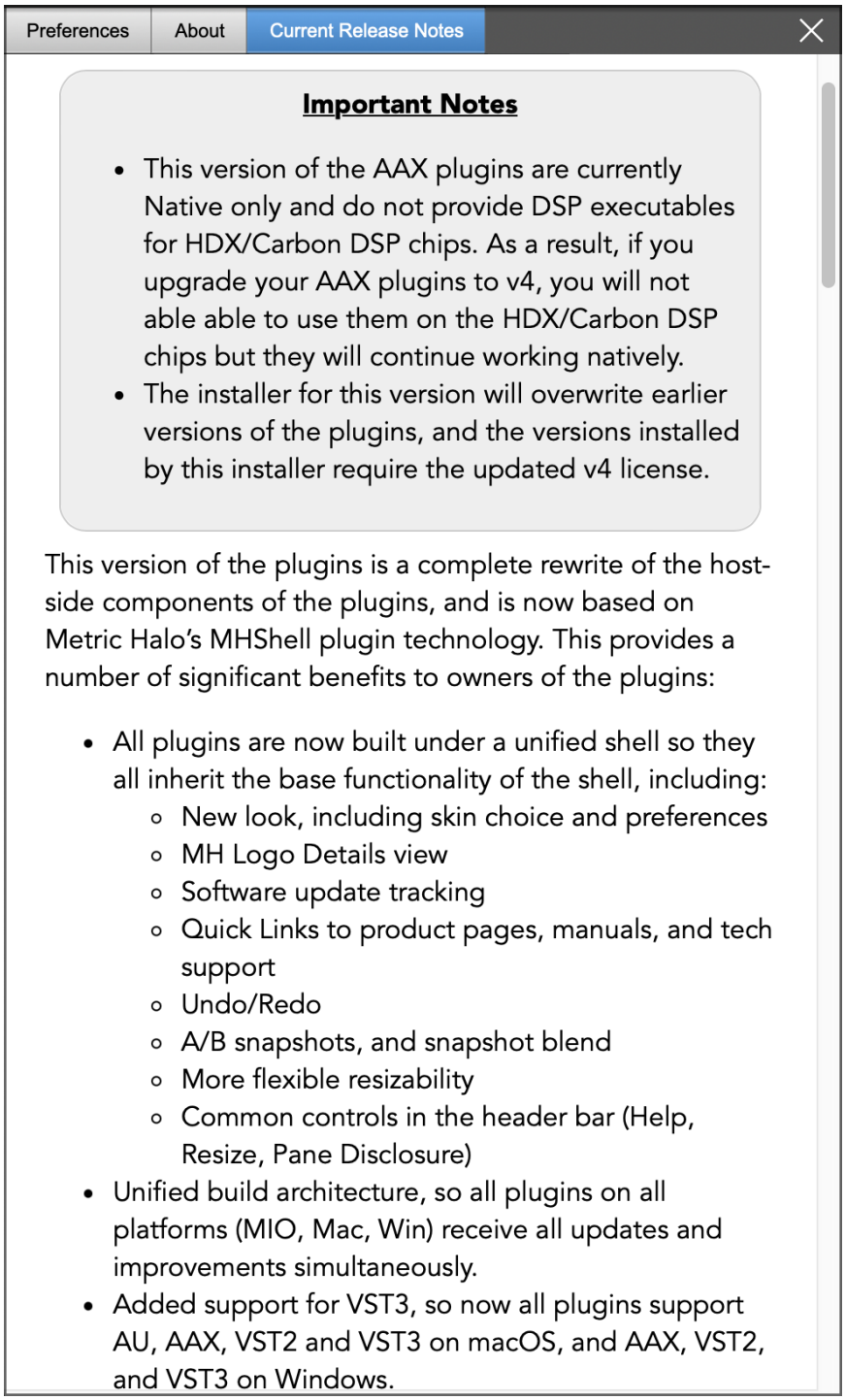
About tab (Multiband Dynamics shown)

The **About** tab reveals the current plug-in version information and provides convenient web links to product info, support pages and the current manual on the Metric Halo main website.

Clicking the "...Users Guide [PDF]" link will open and display the plug-in manual version in your default web browser.

Reveal Plug In File... will open the folder containing the current plugin file, with the plug-in file itself selected. Very handy for troubleshooting on the fly.

Current Release Notes



Current Release Notes example

Current Release Notes lists a synopsis of major changes (a snippet of which is shown above), with feature and bugfix revisions for the most recent software releases listed below.

Plugin Snapshot Registers: A/B



Plug-in Header: Snapshot Registers: A/B

The A and B buttons control the A and B state registers. The A/B registers are used to store modified parameter snapshots in addition to the original saved preset called up by **Compare**.

The Blend function can be used to smoothly morph between the parameters set in the A and B registers, and Blend is a mappable parameter so it can be operated with external MIDI control. Details of the A/B Snapshot Blend feature follow on the next page.

For each of the A and B buttons the visual display tells you the state of the register:

- Light Grey means the register is empty
- Dark Grey means it has settings, but is unselected
- Blue means it has settings and is selected

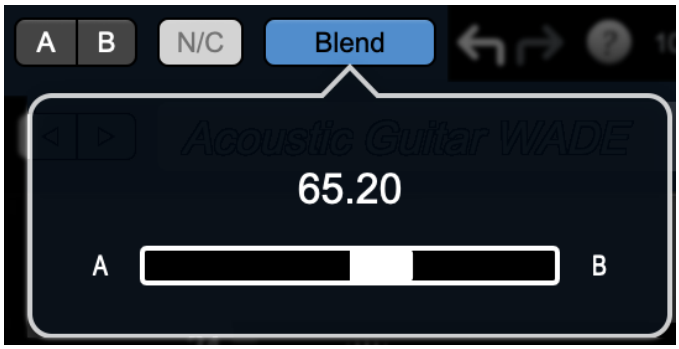
You can perform the following actions:

- Clicking on an empty register takes a snapshot of the current settings and saves them to the register.
- Clicking on an unselected filled register copies the register parameters to the current active plug-in settings.
- Clicking on a **selected** register toggles to the other register; this lets you toggle between the register settings without having to move the mouse.
- The Copy button will show **N/C** and remain inactive until one or both of the registers is in use. The Copy button will alternate between **A>B** (copy A to B) or **B>A** (copy B to A) depending on which register is selected. Clicking the Copy button then copies the settings from the selected register to the target register, overwriting the prior contents (if any).
- <Option> - clicking on a register snapshots the current settings and saves them to the register, overwriting the prior contents (if any).
- Changing settings when a register is selected will update the settings in the register to reflect the change.

Snapshot Blend

The Blend button allows you to interpolate (or morph) between the parameter snapshots stored in the A and B registers. It becomes active when both A and B have a parameter set stored.

Blend is a MIDI-mappable parameter so it can be operated in realtime with external MIDI control and/or automated in the DAW. This allows you to automate a transition from the A → B register, the B → A register or any setting between the two.



Plug-in Header: Snapshot Blend

The Blend button's visual display tells you the state of the register:

- Light Grey means it is empty
- Dark Grey means it has settings, but is unselected
- Blue means it has settings and is selected

Click on Blend to popup the blend control. Slide all the way to the left to apply the settings in the A register. Slide all the way to the right to apply the settings in the B register. Intermediate settings for blend will give you intermediate settings for any parameter that is different in register A and B. The blend control does not change the state of Bypass.

Note that the Blend is not a parallel processing mode where two instances of the processor are running the A and B settings and the output is a parallel blend of the two settings. Rather, Blend interpolates the parameter settings of the two registers to one instance of the processor. You can see the parameter controls move between A and B settings as you slide the Blend control.

The A/B and Blend settings are stored and recalled as part of the plugin state, but are not saved as individual presets in the preset bar.

While you can use the blend with arbitrary A and B settings we find it works best when you craft the settings in the two registers in such a way as they are related to each other. Specifically, if an indexed (stepped) parameter is different between the two settings, the interpolated value will snap to one of the indexes between the two settings, which can be jarring.

It is best if the parameters that you blend are smooth parameters (e.g. gains, frequencies) and make sure the indexed parameters (enables, modes, band types) are set the same for both registers.

The easiest way to do this is to load the same setting into both registers and then tweak the settings of one of the registers.

This works especially well if you make one of the registers be the basic settings with all the gains or thresholds flattened out so that you can smoothly interpolate between a setting and effectively bypassed - we have found that this allows you to zero in a perfect configuration between too much and too little.

Plugin Undo/Redo

All the plugins provide support for undo/redo from the plugin header bar.



Plug-in Header: Undo/Redo

The left and right curved arrows represent Undo (Left) and Redo (Right). These arrows are grey when there is nothing to Undo or Redo.

The arrows are white when it is possible to Undo (Left) or Redo (Right). Clicking the left arrow when it is white will undo the last action you made in the plugin. When you undo something that change is placed on the redo stack, and the Redo button will turn white.

Clicking the Redo button (when it is white) will restore the state that the last Undo changed.

If the Redo button is white and you make a change in the plugin, the Redo button will go grey as the redo buffer will be cleared.

Help Button



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.

Graph Visible Selector



Graphs Closed



Graphs Open

This button toggles visibility of various response graphs. This button allows you to maximize screen real-estate while still providing details on the processing when they are needed. Not all MH family plug-ins will have this button, and on some plug-ins the button will have three panes. Clicking the button will step through the available views, with hidden panes shown darker in the button graphic.

UI Size Selector

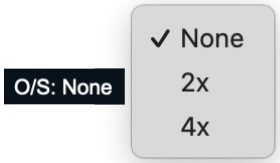


UI Size Selector

This pull-down menu lets you set the plug-in UI size to taste.

The last selected setting for a plug-in processor will be applied to the next new instance you create of that same plug-in. For example, if you open an instance of ChannelStrip and set it to 80%, the next instance of ChannelStrip you create will also open at 80%.

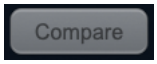
Oversampling Selector



Oversampling Multiplier Selector

Select MIOConsole3d processors including ChannelStrip and TransientControl now support 2x and 4x DSP oversampling. The default Oversampling setting is None.

Compare Button



Compare Button (inactive)



Compare Button (active)

To use the compare button, a preset must first be loaded. The compare button will be lit up when the current settings differ from the selected preset. If you click this button while it is lit, the preset settings will be restored, but you can still return to the changes you made by clicking on the button again. It is important to note that any changes you make to activate the compare light are always for comparison to the last loaded preset.

Soft Bypass



Soft Bypass Button (not bypassed - plugin is processing)



Soft Bypass Button (bypassed - plugin is *not* processing)

When glowing yellow, this button will maintain the time delay through the channel and will continue to show metering, but will cleanly disable the processing.

For MH 3d hardware users: A note regarding plug-in Sidechains

In host-based DAWs, the mechanism for routing signals to plug-in sidechain inputs is handled within the DAW itself in a separate routing UI from the plug.

In MIOConsole3d, plug-ins which support external sidechain inputs are assigned from within the plug-in UI (as shown below).

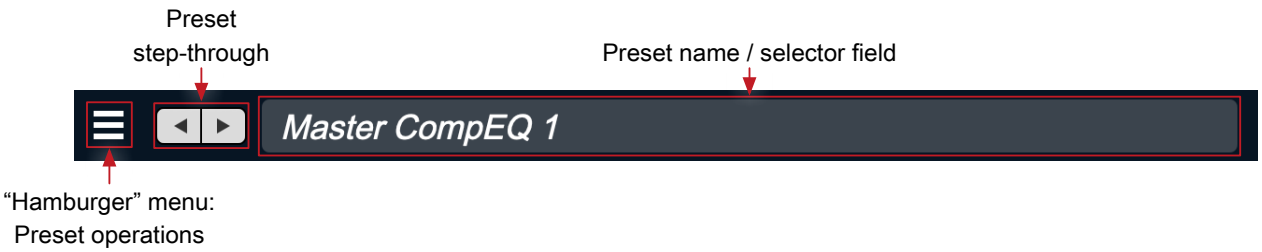


MIOConsole3d ChannelStrip header

In MIOConsole3d, sidechain inputs may be routed from any input or bus available in the system, except for the Main bus. These routes are naturally going to be different from what would be assigned in a DAW session.

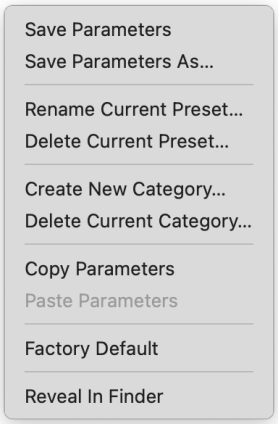
For this reason, external sidechain assignments are the one parameter which can not translate between a host-based DAW preset and the same preset opened within MIOConsole3d.

Plug-In Header: Preset Row



Plug-in Hamburger menu

The preset and parameter functions within the hamburger menu break down as follows:



- **Save Parameters** saves the current plug-in parameters to the current preset.
- **Save Parameters As...** opens a dialog box where you can name and choose a category to save 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 paste 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.

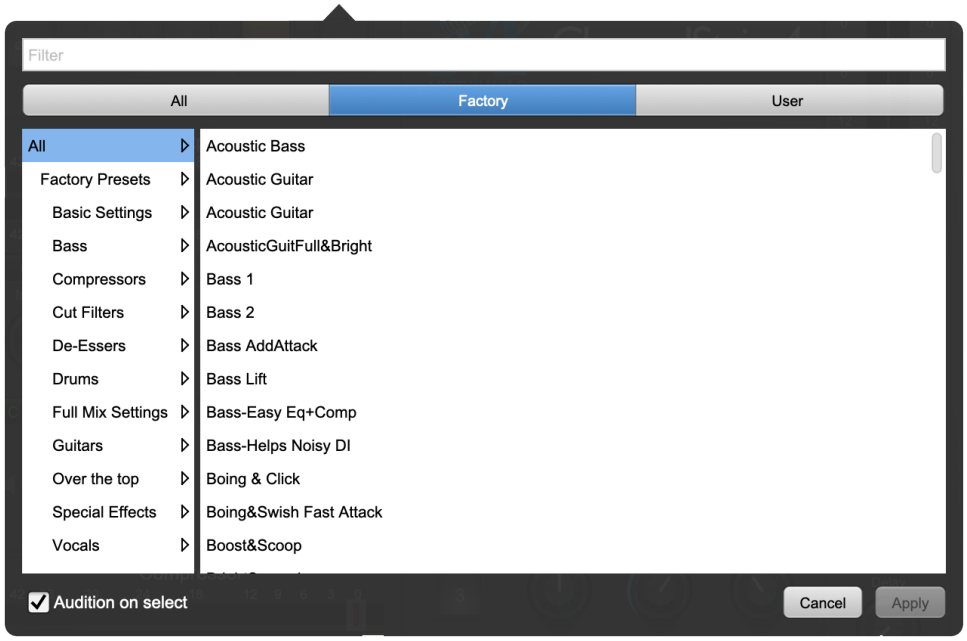
Preset Step-Through Buttons



Preset Step-Through Buttons

These buttons step through Factory and User Presets in succession, as they are listed in the Preset Selector window. The left arrow chooses the previous preset. The right arrow chooses the next preset.

Preset Name/selector menu:



Plug-in Header: Preset selector menu (ChannelStrip shown)

The Preset selector will open to show all the available preset categories, and the presets within those categories.



Preset selector menu: Audition on select

With "**Audition on select**" enabled at the bottom of the window, selecting a preset will temporarily load those parameters so you can hear the effect on the audio you are playing, without actually committing to the preset.

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.

Multichannel / Multi-mono Plug-in processing

All MH family production plug-ins within MIOConsole3d 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 only on stereo tracks.

On a multichannel (LCR to 7.1) input or bus, the mono instance is used to allow "multi-mono" operation. 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 strip, or...
- Insert a Graph into the multichannel source or bus strip, and instantiate multiple plug-ins there...

  The beauty of this approach is that it lets you create your own custom multichannel processor, with channel processing routed, grouped and linked to your specific needs and tastes. Within the Graph, you can route and process compressor sidechains logically to match your dynamics control requirements, and assign external MIDI controls across the various plug-ins parameters. Once it is working the way you want, save your new multichannel processing Graph with all the MIDI assigns intact for future use.

Multichannel vs. Multi-Mono considerations
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. In some stereo dynamics processors this is called a 'linked detector sidechain'.

Graphs

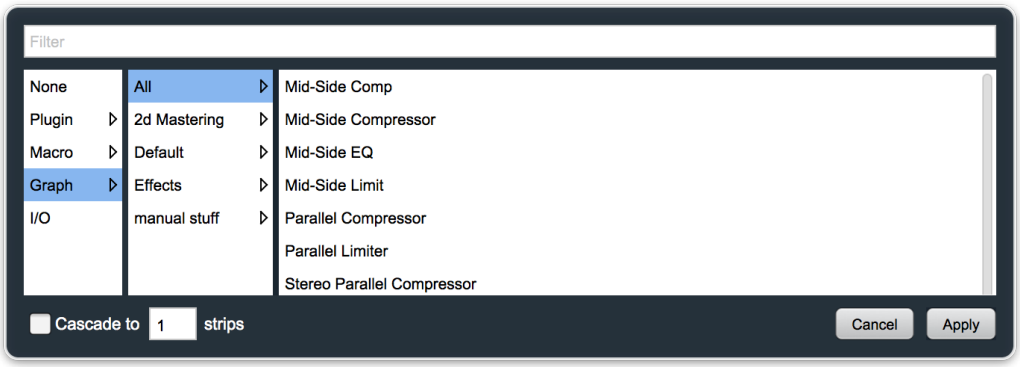
A graph is a freeform area in which you can create your own DSP processing chains.

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.

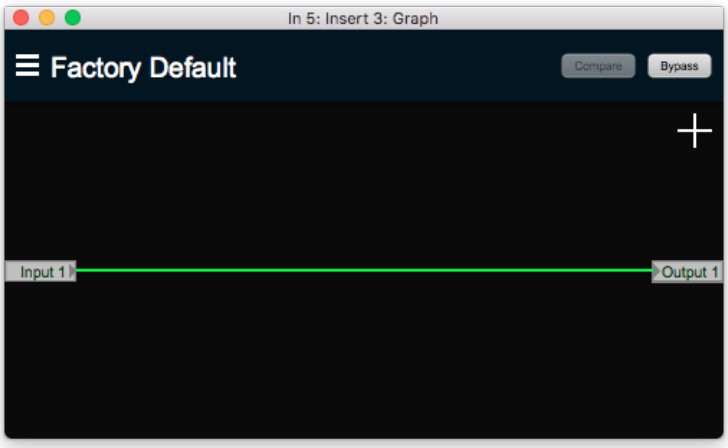
When you open the graph UI for the insert, you can insert any set of plug-ins into the graph that is shown in the graph UI window. These plug-ins can be connected by virtual cables, their UIs opened, and parameters set. The graph I/O connections will automatically be routed to the appropriate points 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:



Selecting the Graph

and you will see this window:



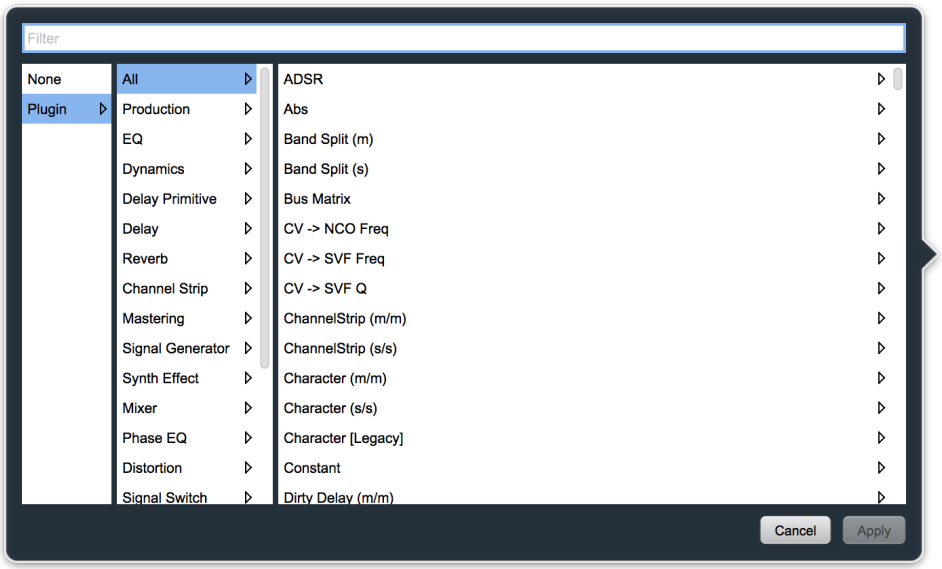
A default mono Graph window

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.

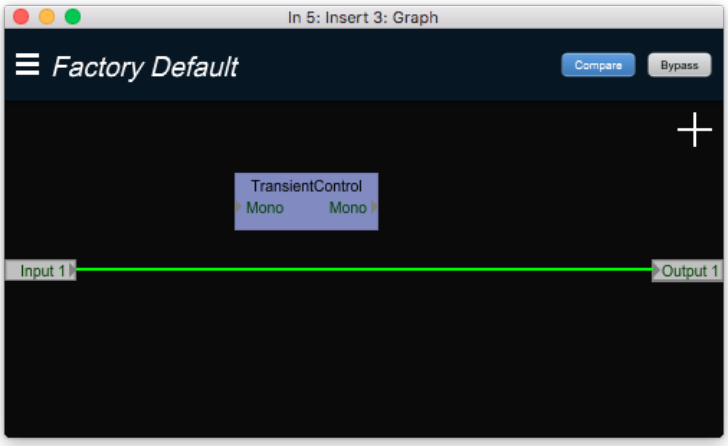
As you can see, the Graph has the same plug-in window header, hamburger menu and preset menu as every other plug-in. In this regard, Graph preset categories and save/recall is no different than any other plug-in.

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 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.



Selecting a new plug-in instance from the Plug-in Menu

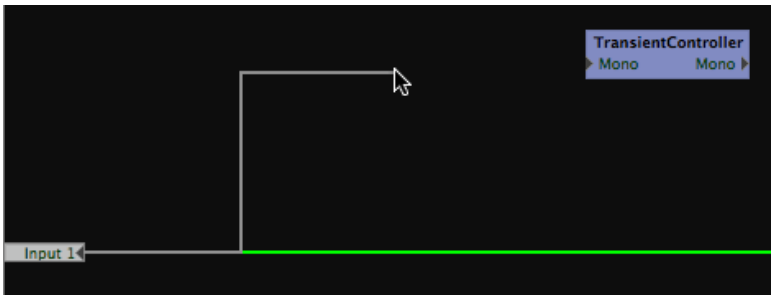
Note that, yes indeed, you can insert full-blown processors like Channelstrip, TransientControl, Sonic EQ 3d, MIOStrip and Haloverb in the Graph without restriction, right alongside CV control and mathematical function instruments. It truly is a playground.



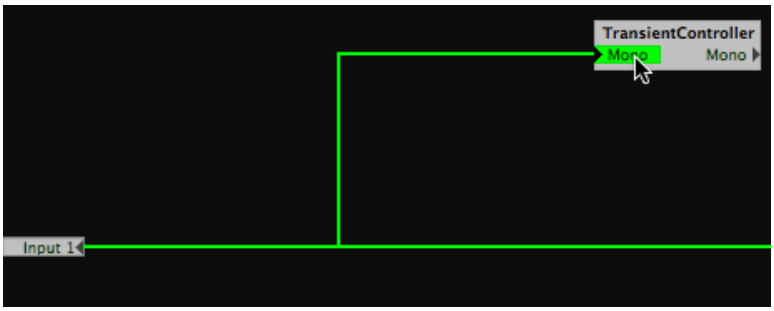
Positioning the new plug-in instance in the Graph

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

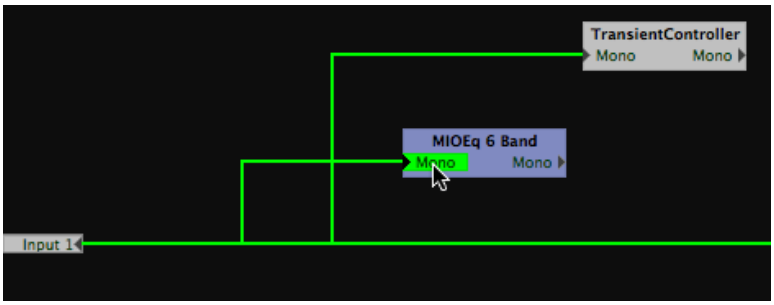
target. When you have made a valid connection, the connection line will switch from Gray to Green. You can make as many mults as you like of a signal source but only one connection can be made to a processor input or process bus port. If you make a new connection to an input that already has a connection, 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.



Starting a Connection

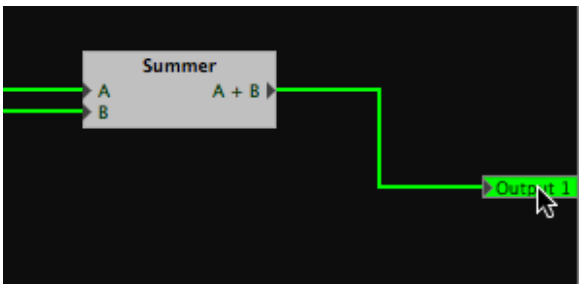


Completing the Connection



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:



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



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.

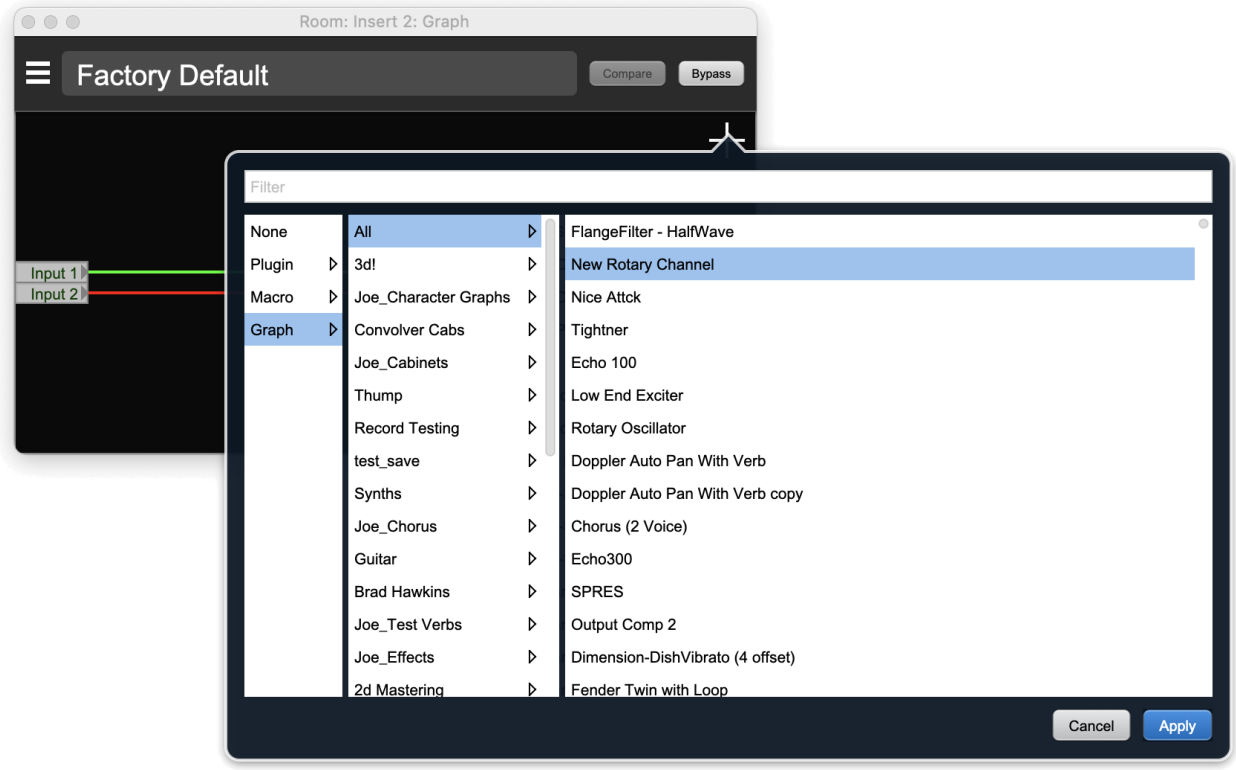
Sub-Graph Nodes

Once you get into working within the Graph, you will find yourself building custom processing devices of your own, saving them as presets and using them to define your own sound. Like learning how to express yourself with a musical instrument, it just happens.

Some of these devices will be simple to start with, but you will refine and modify them as you go along. Soon you will build a gadget and realize that you want to incorporate a Graph processor preset you've already created into your new device - a graph within the graph. This is where you use a subgraph.

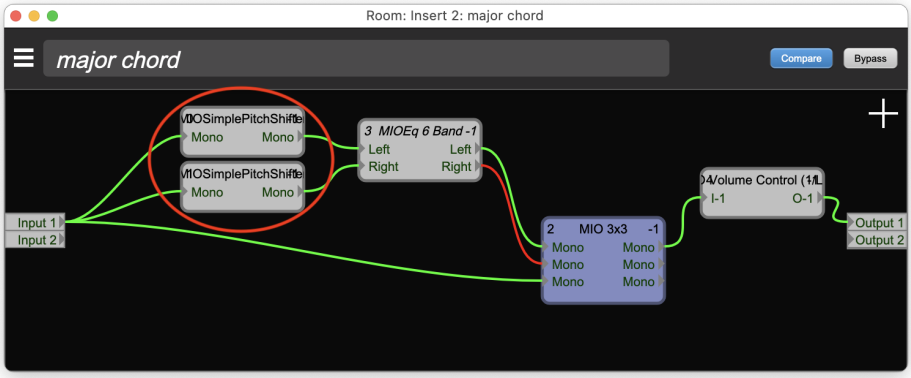
We have added support to the +DSP graph to allow creating or inserting graphs as nodes within the graph. This works any place you can insert a graph (including insert slots, monitor and cue output graphs, and within subgraphs).

In the upper right corner of the graph UI is a white **+** icon. Click to open a plugin insert selector.



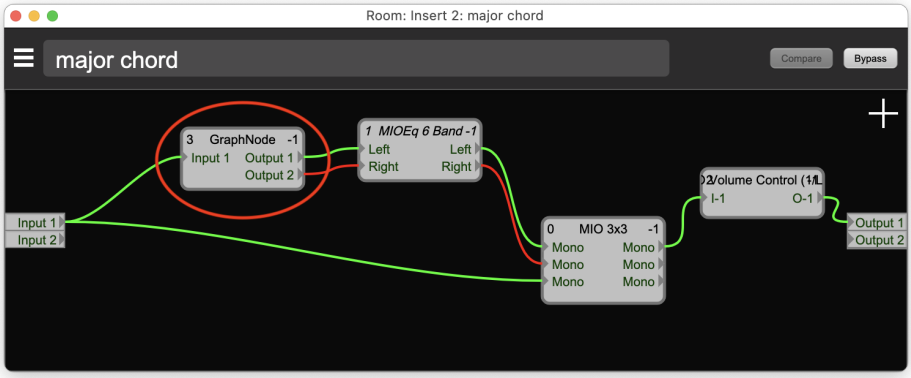
To insert a graph inside a graph, just select "Graph" or "Macro" from the insert popup selector. If you insert an empty graph, you'll get a stereo->stereo graph node. If you insert a Graph preset or Macro, you'll get a node that has the I/O config of the preset.

To edit the contents of a Graph node, you double-click it, just like a plugin node. That will open a graph window representing the subgraph. As we mentioned above, you can also insert graphs within subgraphs - this can be done recursively.



You can also create a subgraph in-place from a group of selected Nodes within a graph. To do this, select the nodes you would like to push down into the new subgraph node, and then press the "n" key.

This will move the selected nodes as well as all the connections that are internal to the selected group of node into the new subgraph; it will create an input port for any connection that enters the group from outside the selected set, and it will create an output port for any connection that exits the selected set. The connections entering and exiting the selected set will be connected to the created ports, and the created ports will connect to existing connection points within the containing graph.



The net effect is that the signal graph will run exactly the same way as before you extracted the subgraph, but the selected nodes will now be within the newly created subgraph.

At this point your new subgraph node can be saved as a preset and recalled just like any other graph.

This allows you to simplify graphs that have gotten too complicated, and also allows you to build up complicated graphs from a library of simpler subgraphs.

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
Create a subgraph Node of the selected nodes within a graph	N

MH 3d DSP Production Plug-ins



The Metric Halo Production Bundle

3d DSP = MH Production Bundle (and more)

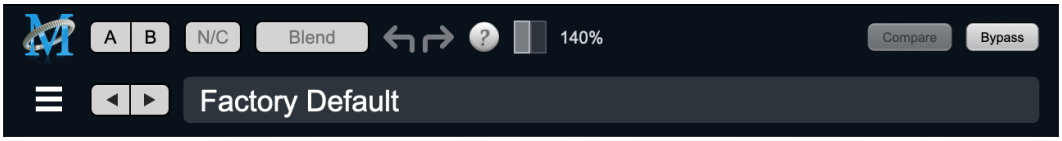
Naturally, all of Metric Halo's MIO 2d DSP processors have been updated to run natively in the 3d DSP engine, including MIOComp, MIOLimit, MIOEQ 6 & MIOEQ 12. All 2d Graph building blocks have also carried over to the new platform, ensuring even your most complex custom processing creations will smoothly transition over to 3d.

Equally naturally, Metric Halo's world-renowned AU/VST/VST3/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/VST3/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/VST3/AAX software versions, so any parameter presets you save in either environment are accessible to the other.

Sharing presets between Metric Halo family AU/VST/VST3/AAX and 3d DSP plug-ins is automatic as long as you use the Metric Halo header bar Plugin Presets Selector popup menu (shown below).



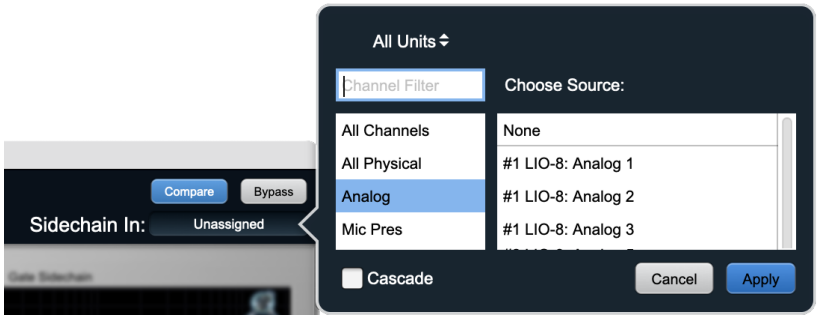
AU/VST/VST3/AAX and 3d DSP plug-in header

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.

Sidechain Input selector



Sidechain In selector window (ChannelStrip in Light Brushed mode shown)

ChannelStrip and SuperGate currently support external detector sidechain input, and in 3d DSP 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.

Character

Plug-in categories: **Production**, **Distortion** and **I/O Emulation**

Introduction

Character is an oversampled analog modeling plug-in for digital audio workstations. 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.

Character has always employed 8x oversampling, and with v4 more oversampling options have been added for even greater flexibility.



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.

Operation

The Character user interface uses a few different control elements to control its processing. These elements are:

Character Model selector



The selection grid selects Character's model. Click, click-drag or use the arrow keys to instantly change models. There are over 20 models available, which are detailed in the Processing chapter.

Control Knob



Control Knobs are used to control the Drive and Output Gain parameters. 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 Mac ⌘ (Command) key or Windows **Control** key when you click, you will be able to adjust the value with finer precision. If you hold the Mac ⌥ (Option) key or Windows **Alt** key when you click, the knob will reset to its default value. You may also double-click a knob to reset it.

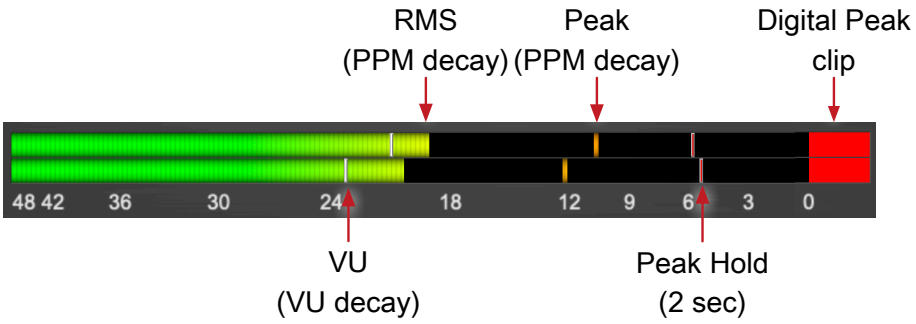
Right-click the knob to display a text entry field that allows you to type in a number directly. The pop-up will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, Mac ⌘. (Command + .), Windows **Alt** key or **ESC** keys. Hit **return** or **enter** 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 Mac ⌘. (Command + .), Windows **Control.** (Control + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.

Toggle Button



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



For the main output stage of Character we have provided meters driven with SpectraFoo™ metering technology. These meters show digital Peak, Peak Hold, RMS level and VU level.

Both the Peak and RMS level are represented with fast PPM ballistics.

The instantaneous Peak level is represented by the floating colored pip. Peak Hold is 2 seconds, indicated as a red floating marker pip.

The RMS level is the solid colored bar which transitions from green at "safe" high-headroom levels to yellow at -18dBFS, then to orange and finally red approaching -6dBFS.

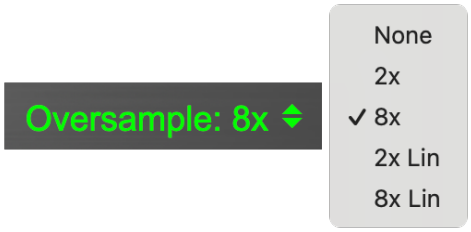
The VU meter is the white floating marker pip showing IEEE standard 300 ms RMS average level.

When Character is on a mono insert there will be a single meter.

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.

> **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.

Oversampling selector



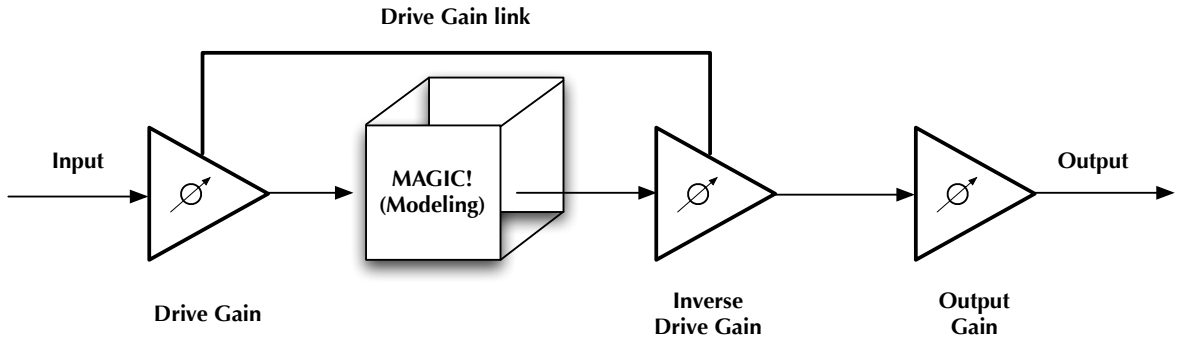
Character has always operated at 8x oversampling to ensure coverage of all potential harmonics generated for all character models. Character v4 has expanded on the oversampling options and made them selectable via the **Oversampling** menu above the Character Model selector.

The oversampling options are detailed in the Processing chapter.

Processing

A Detailed Description

In this section 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.



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.
- Soft Sat Gain: Soft Sat, with a set gain compensation for peak level loss due to the saturation effect. Gain compensation is referenced at Drive = 0.
- 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.

Oversampling

The Oversampling selector provides five oversampling settings:

- **None**: No oversampling... for that vintage digital grit. 0 latency.
- **2x**: IIR minimum phase filters. Provides good aliasing suppression with low cpu cost. 2 samples latency. A great fit for less complex characters in a heavy mix.
- **8x**: Default setting. IIR minimum phase filters. Near-perfect aliasing suppression (second only to the 8x Lin) with somewhat higher cpu cost. 4 samples latency. Best compromise for complex characters in a busy mix environment.
- **2x Lin**: FIR linear phase filters. Lower distortion, better depth and detail with a higher cpu tax. Suggested for cpu-conscious immersive sessions and high sample rate production. 15 samples latency.
- **8x Lin**: FIR linear phase filters. Best all around performance (lowest distortion, greatest detail, least time smear) with a relatively higher cpu cost. Suggested for sub-mix buses/stems and exposed tracks featuring harmonically-rich character. 15 samples latency.

Note that the 'None' and 'Lin' settings are perfectly time-compensated by the host and will properly null against source audio (if you are into that kind of thing). The minimum phase settings by definition do incur a phase shift and will not return a perfect null no matter what.

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.

Using Character in Your Projects

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…

  Use MIDI automation to control the Drive parameter of multiple Character instances as a way to push track groups forward or back in the mix in a more natural and colorful way than standard fader control.
- Inserted on the master bus: Put a Soft Sat model on the master 2 bus for the analog "glue" to finish off your mix.
- All of the above!

ChannelStrip

Plug-in categories: **Production**, **EQ**, **Dynamics** and **Channel Strip**

Overview

ChannelStrip provides the essential channel strip processing of a modern mixing console.

Processing functions include:

- Input level control
- Phase Invert
- Expander/Gate with filtered sidechain
- Compressor with filtered sidechain
- 6 band Parametric EQ
- Channel delay
- Limiter

ChannelStrip 4 has several notable new features:

- MH Logo Details view
- New look, including skin choice and preferences
- Software update tracking
- Quick Links to product pages, manuals, and tech support
- Undo/Redo
- A/B snapshots, and snapshot Blend
- More flexible resizability
- Common controls in the header bar (Help, Resize, Pane Disclosure)
- Enhanced multi-function Spectrafoo dynamics processor and output level metering



Operation

ChannelStrip 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:



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.



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.



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.



Limiter Knob

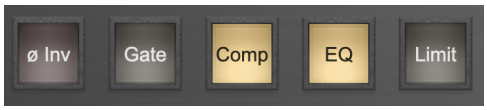
The ring around this encoder displays the amount of gain reduction from the limiter. The display sweeps around the encoder from right to left, with a fully left display indicating 12dB of gain reduction.

Control tips

You can change the value of each knob in a number of different ways.

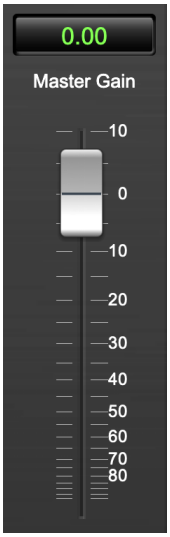
- Click and drag the knob to change the value continuously.
- Dragging up or to the right will increase the value, while dragging down or to the left will decrease the value. Hold down the ⌘ (Command) key when you click to adjust the value with finer precision.
- ⌥ (Option)-click to reset any knob to 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



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 processor sections, select keying sources and to switch the order of processing within ChannelStrip.

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 can be set to one of six filter shapes:
Peaking/Parametric
Low Shelf
High Shelf
High Cut
Low Cut
Bandpass

The details of each filter type can be found in the EQ Processing: Filter Type section.

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 Mac ^ (Control) key when you click or right-click (Mac/Win).

Sidechain Routing Switch



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



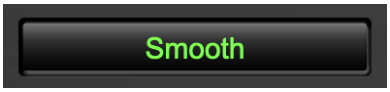
Sidechain Listen
(Disabled)



Sidechain Listen
(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

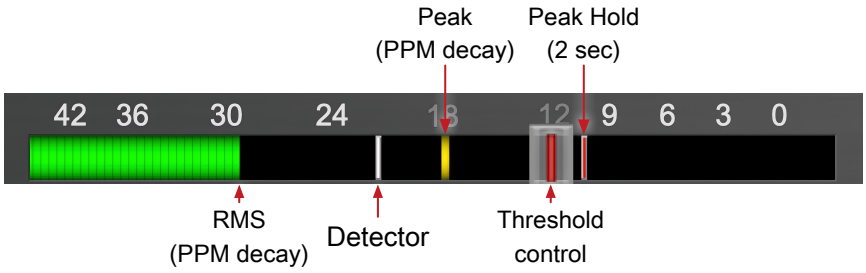


Compressor Character Switch

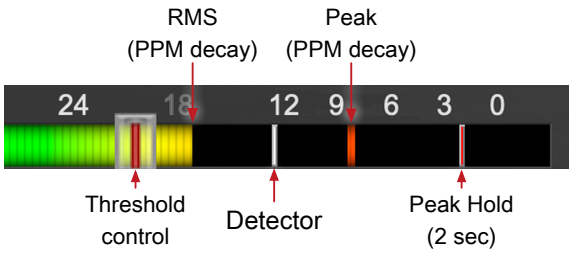
The compressor character controls the time constants of the compressor section. It functions identically to the Filter Type control, except there are only four choices: Smooth, Warm, Fast and MIO. See the section on the compressor for more information.

ChannelStrip also uses a number of standard visual representations to give you feedback about what is happening within the processor. These elements include:

Input Meters



Gate/Expander Peak Meter



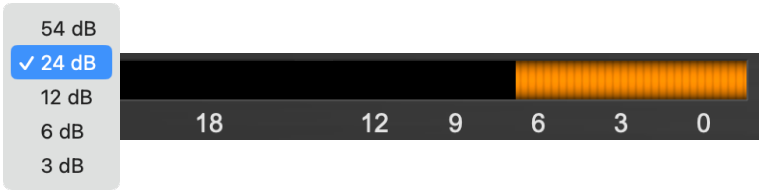
Compressor Peak Meter

SpectraFoo™ multi-function metering is provided at the input stage of ChannelStrip's Gate/Expander and Compressor processing blocks.

- RMS level is indicated by the colored solid bar, which transitions from green at "safe" high-headroom levels to yellow at -18dBFS, then to orange and finally red approaching -6dBFS. The RMS meter uses the fast PPM standard for decay time (0.9 seconds per 20 dB) and the digital PPM standard legend for calibration.
- Instantaneous Peak is displayed by the floating colored pip which follows the RMS level bar color and decay standard as above.
- Temporary Peak Hold is show as a floating red pip with a hold time of 2 seconds.
- On the dynamics sections (gate and compressor), the white floating pip indicates the current detector level. Detector display is after the Sidechain filter in the signal path, and the detector behavior itself varies depending on the mode of the dynamics processor:
  - The Gate/Expander uses a fast peak detector, with Attack and Release applied at the gain reduction stage.
  - With the Compressor in *MIO* mode, the attack and release are applied to the envelope detector, and the gain reduction is driven directly from the detector. This is why, in *MIO* mode, the detector pip response visually slows down following the Attack and Release controls.
  - The *Smooth*, *Warm* and *Fast* compressor modes employ the fast peak detector, with gain reduction controlled by Attack and Release as applied by each individual compression mode characteristic.
- 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 and tracks along with changes made by the Threshold knob.
- 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**/**Alt**-click any meter to reset the clip lights on all of the meters in ChannelStrip.

When ChannelStrip is running in stereo mode, both the Gate/Expander and Compressor input meters show the higher of the two input levels and will detect digital clipping on either input channel.

Gain Reduction Meter

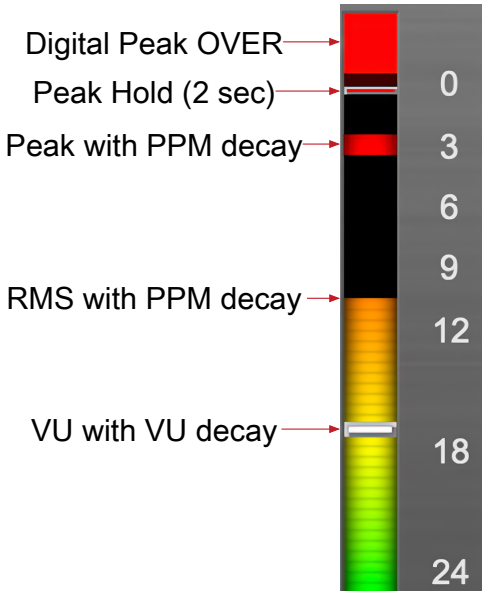


Gain Reduction Meter

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. If you right-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 with Peak Hold



Output Meter (mono / enlarged)

For the main output stage of ChannelStrip we have provided meters driven with SpectraFoo™ metering technology. These meters show digital Peak, Peak Hold, RMS level and VU level.

Both the Peak and RMS level are represented with fast PPM ballistics.

The instantaneous Peak level is represented by the floating colored pip. Peak Hold is 2 seconds, indicated as a red floating marker pip.

The RMS level is the solid colored bar which transitions from green at "safe" high-headroom levels to yellow at -18dBFS, then to orange and finally red approaching -6dBFS.

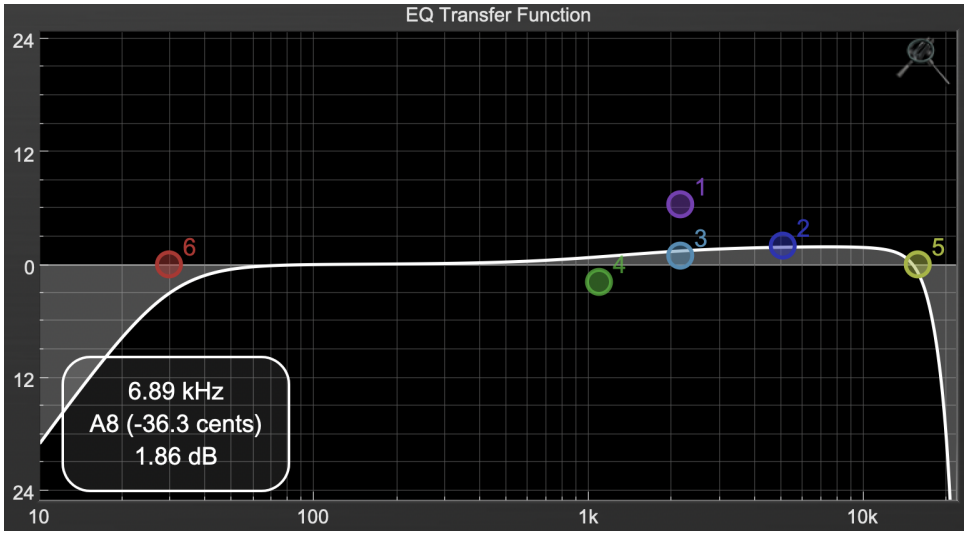
The VU meter is the white floating marker pip showing IEEE standard 300 ms RMS average level.

When ChannelStrip is on a mono insert there will be a single meter.

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**/**Alt**-click any meter to reset the clip lights on all of 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 the risk of actually clipping the input of another processor or D/A convertor.

EQ Transfer Function



EQ Transfer Function

The following information on the EQ Transfer Function and Spectragraph Analyzer apply to the EQ processing block as well as the sidechain filter displays of the gate and compressor.

The horizontal axis provides frequency calibration in Hertz (Hz). The vertical axis provides level calibration in decibels (dBr). The heavy line (white by default) 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 dot represents the center frequency or knee of the filter band, and acts as a control handle for the band within the transfer function field.

The band that is currently being edited will be highlighted and 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 band handle dot and dragging will allow you to adjust the frequency and gain of the associated band.

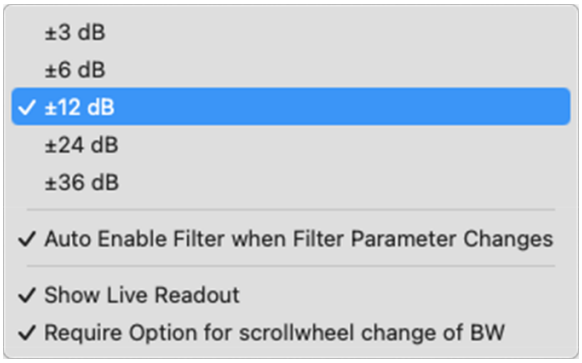
- ⌘ (Command)–click or double-click the dot to toggle the band enable (single-click to disable a band when Auto Enable Bands is turned on).
- Click and drag the smaller dots associated with a larger dot to adjust the filter Q. You can also ⌥ (Option)–click the dot to adjust the Q (dragging left increases the Q, right decreases the Q).
- ⌘⌥ (Command + Option)–click the dot to switch the band filter type.

In the transfer function field, multiple filter bands may be temporarily grouped by shift-click or click-drag selection of the filter control handles.

To dismiss the filter curve, click anywhere in the black area of the transfer function. This will deselect the filter, and the only trace displayed will be the master eq curve.

Transfer Function options

Right-clicking the EQ transfer function background brings up an EQ functions control menu.

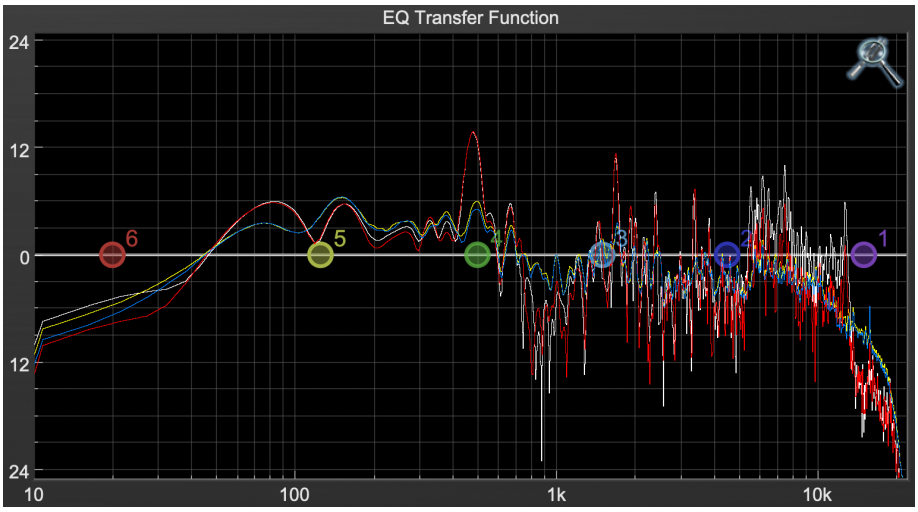


- EQ display scale: Sets the EQ curve dB scale from ±3 to ±36 dB.
- **Auto Enable Bands when band parameter changes**: Automatically enables a filter band when one of its parameters is adjusted within the EQ Transfer Function. This preference will apply to all instances of Sonic EQ.

  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 or the Special filters.
- **Show Live Readout** (shown on the previous page): Enables an overlay in the lower left corner of the transfer function which displays the relative amplitude of the master EQ curve at the frequency indicated by your mouse cursor. Moving the cursor off the EQ display hides the readout. Live Readout control is enabled per instance.
- **Require Option for scrollwheel change of BW**: Enabled by default, this control requires the Option modifier key to change the Q parameter using a mouse scrollwheel. Scrollwheel filter width control works for filter band knobs (one at a time) and selected filters in the transfer function display (multiple selections supported).

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 overlaid on the Transfer Function EQ curve:



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 filters.

To disable the spectragraph entirely, click the active Spectrafoo icon.

If you right-click or ^ (Control) click on the Spectrafoo icon, you will see a menu to set options for the spectragraph:



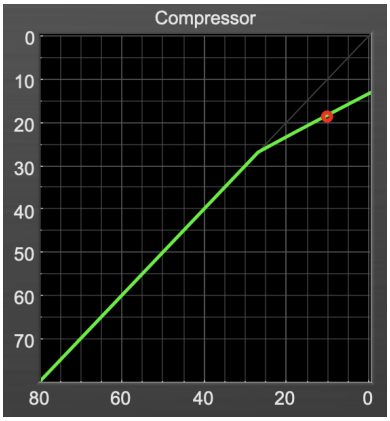
- Show Instantaneous Trace: Toggles the instant response traces for both channels.
- Show Average Trace: Toggles the averaged response traces for both channels.
- Show Left Channel: Toggles the left channel spectragraph display.
- Show Right Channel: Toggles the right channel spectragraph display.

These settings are stored independently for each instance of ChannelStrip.

Note that just the Left channel display is shown by default.

Controls for Left and Right channel will only be available in stereo instances of ChannelStrip.

Dynamics Knee



Dynamics Knee

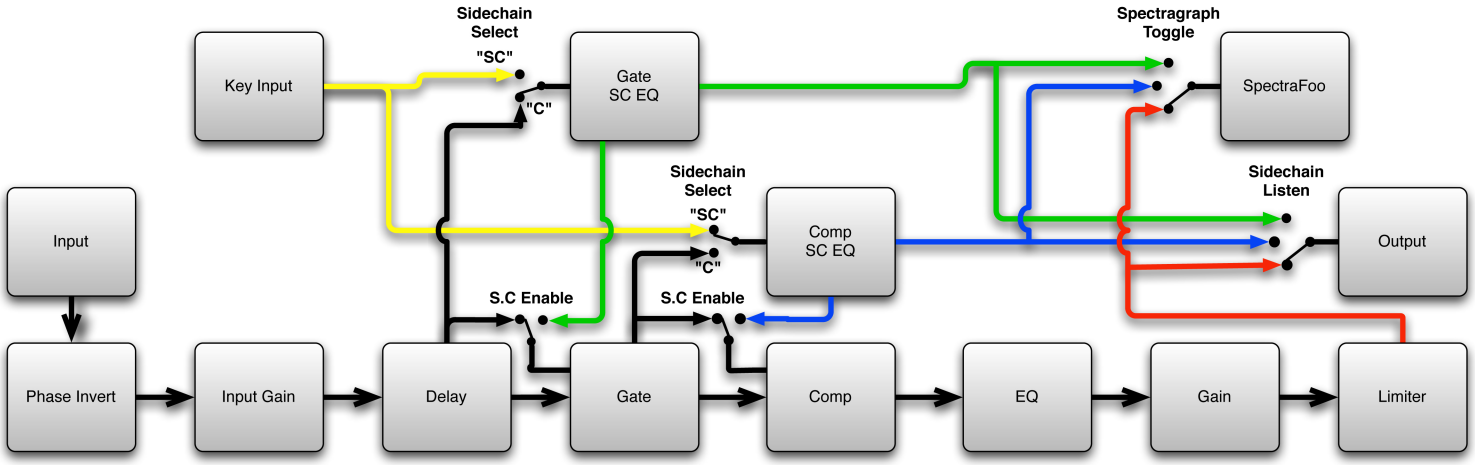
ChannelStrip contains a Dynamics Knee diagram for the gate and compressor processing sections. The diagram provides feedback on the response of the associated dynamics processor. Both the horizontal and vertical axes are calibrated in dBFS. The horizontal axis corresponds to the input level and the vertical 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.

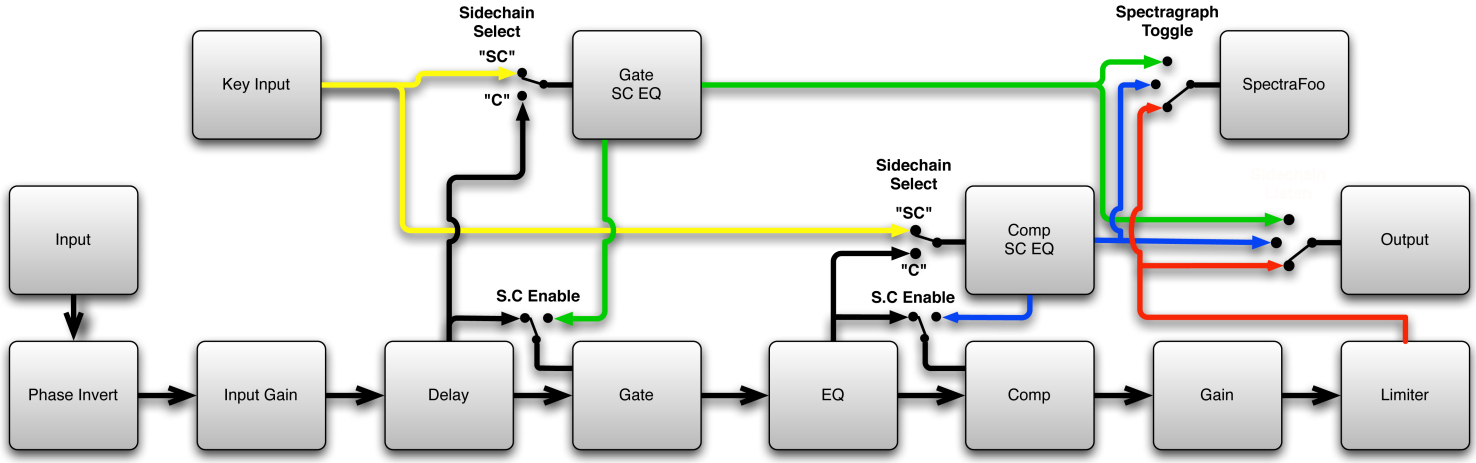
Processing

A Detailed Description

In this chapter we'll discuss what each processing block does and how the controls work.



Signal Flow: Compressor is Pre-EQ



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.



Gate/Expander

Theory Of Operation

Based upon your setting for the sidechain routing switch, either the Pro Tools 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).



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.

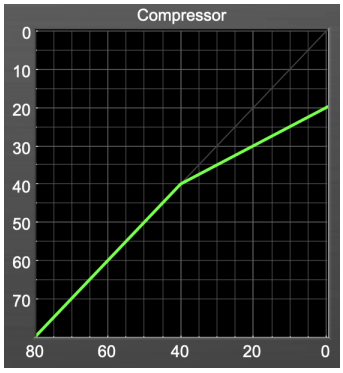
Auto Gain

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

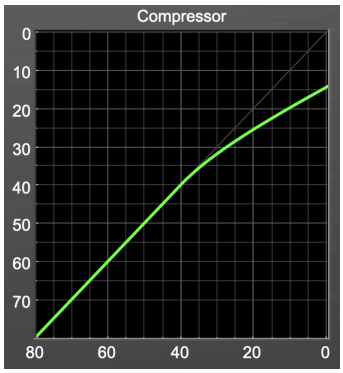
The "Knee" knob allows you to adjust shape of the compressor transfer function when the Compressor Character is set to MIO (the Knee control has no effect for the other compressor character algorithms and will not be visible in those modes).

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.



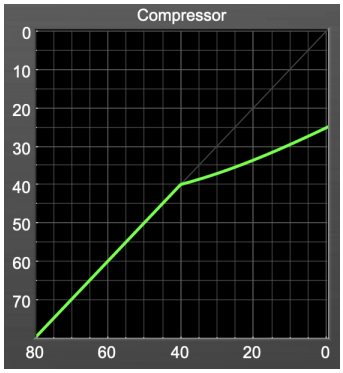
Knee Control at 0

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.



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.



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 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.
- **MIO** – provides the compressor characteristics from the MIOStrip compressor that runs on our interfaces. 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.



EQ Section

Theory Of Operation

The equalizer in ChannelStrip work just like every other EQ under the sun with the exceptions that it 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:

- **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.



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.



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.



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.



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.



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.



Bandpass

Filter Band Color Swatch

Clicking the color swatch below the Filter Type button will open the OS color picker where you can change the color elements of that band. The swatch duplicates the *EQ Colors* controls found in the Plug-in Preferences pane.

De-selecting "Show Band Swatches" in the Plug-in Preferences will hide them from the EQ UI.

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



Delay

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



Limiter

The final processing block in ChannelStrip 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 plug-in for digital audio workstations 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



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

The Link Controls button at the upper left corner of the Dirty Delay UI toggles stereo linking of the left and right channel parameters.



Link Off (default state) & Link Controls On

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.



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 the Mac Option key or Window Alt 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, or right-click or control-click on the knob itself to display a text entry field that allows you to type in a number directly (shown above). The pop-up will remain active until you dismiss it by clicking somewhere else or hitting Return, Enter, Tab, Mac Command +., Windows Alt key 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 and entry field for the next control. Shift + Tab will display the entry field for the previous control.

Hit the Mac Command + ., Windows Control + . or ESC (Escape) to dismiss the pop-up and cancel the change.

Toggle Button



Toggle Button (Off & On)

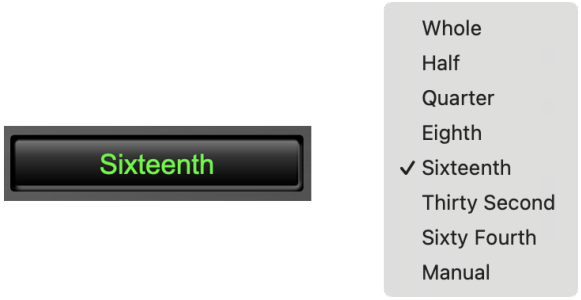
Toggle buttons are simple on/off switches. They light up when they are on and are dark when they are off. Toggle the state of the button by clicking on it. The button shown is used to enable Tempo Sync within Dirty Delay.

Tempo Tap



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



Note Length Selector and Popup Menu

The Note Length selector (above left) opens the pop-up menu (above right) to select the note value of the delay with respect to the tempo.

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. Manual Delay is a full manual mode which overrides all Tempo control of the delay, including Sync and Tap.

Radio Button

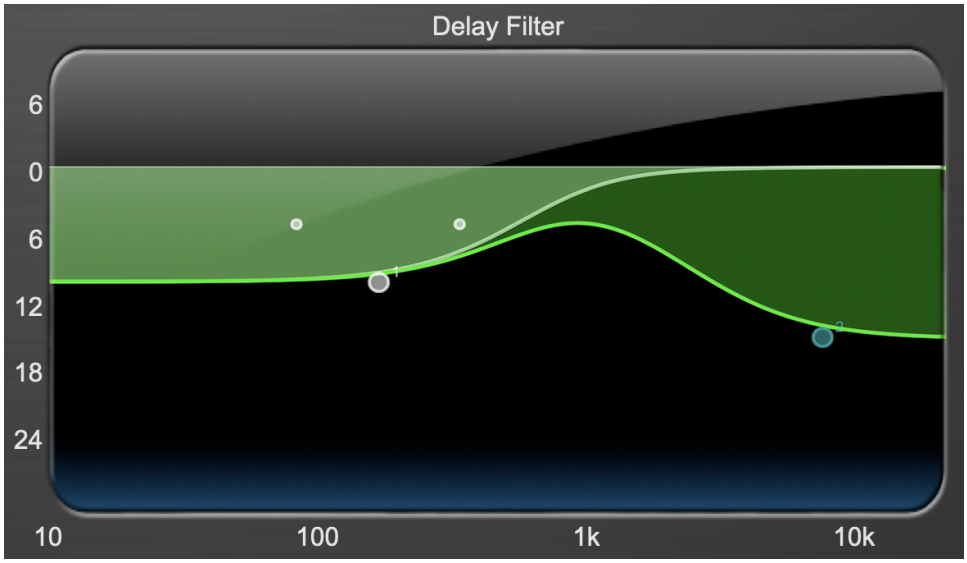


Triplet & Dotted Note length modifiers

The buttons used to modify the delay note values with triplet or dotted note time are “radio buttons”.

Radio buttons are like toggle buttons, except that they simultaneously disable another parameter when enabled. In the case of triplet and dotted note length modifiers, you can select triplet, dotted or neither, but not both.

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-click 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. 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.



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.



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.



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.



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.



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.



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 Mac ^ (Control) key when you click or right-click (Mac/Win).

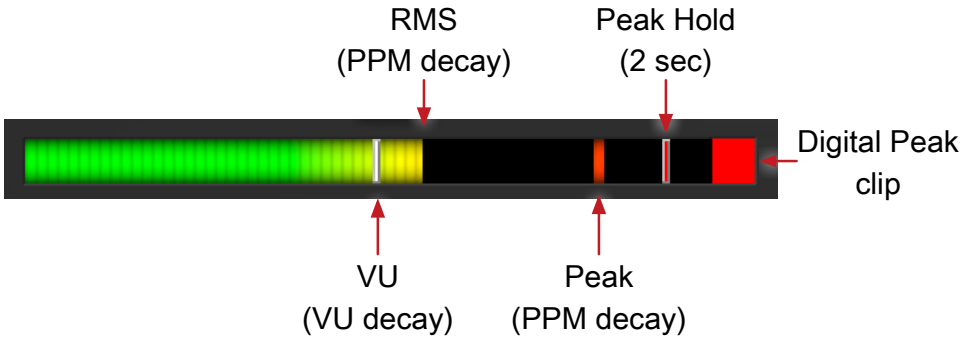
Character Model Menu



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



Dirty Delay Output Level Meter

For the main output stage of Dirty Delay we have provided meters driven with SpectraFoo™ metering technology. These meters show digital Peak, Peak Hold, RMS level and VU level.

Both the Peak and RMS level are represented with fast PPM ballistics.

The instantaneous Peak level is represented by the floating colored pip. Peak Hold is 2 seconds, indicated as a red floating marker pip.

The RMS level is the solid colored bar which transitions from green at "safe" high-headroom levels to yellow at -18dBFS, then to orange and finally red approaching -6dBFS.

The VU meter is the white floating marker pip showing IEEE standard 300 ms RMS average level.

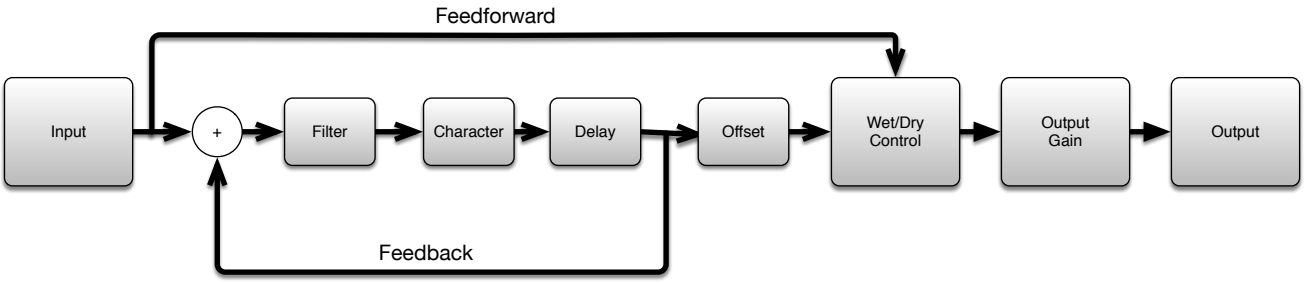
The output section clip lights activate if there is an over in the output stage or in any of the processing section input stages. It is reset by clicking on the meter.

> **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.

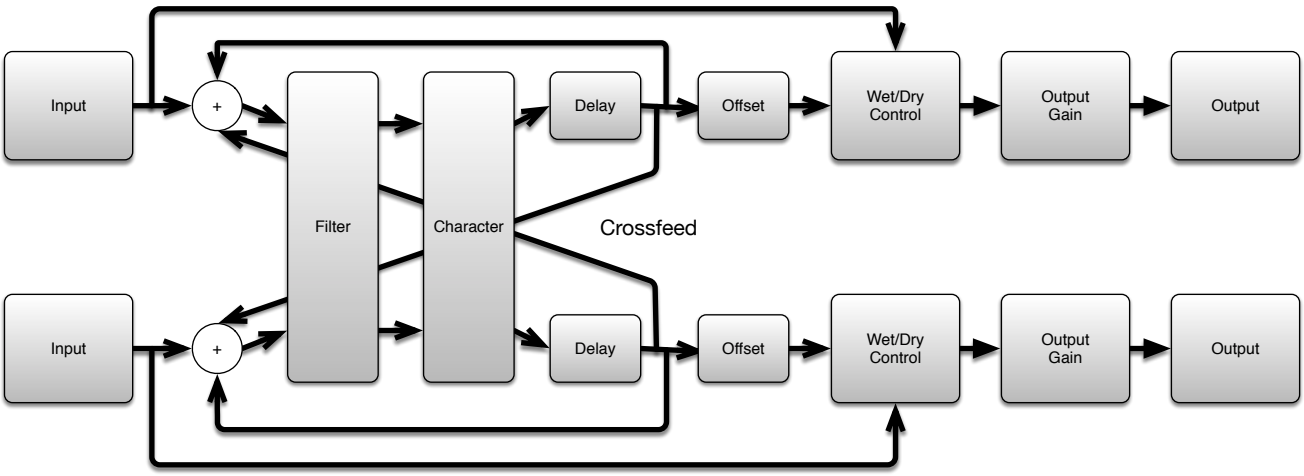
Processing

A Detailed Description

In this chapter we'll discuss what each processing block does and how the controls work.



Mono



Stereo

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

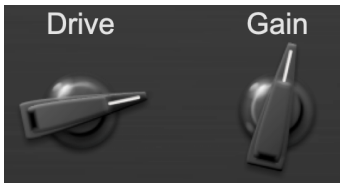
The filter block provides frequency shaping to a copy of the input signal prior to being delayed. The available controls allow you to set the frequency and level of the variable pre-filters of the delay. Using these controls allows you to tailor desired frequencies with up to +6 dB of boost to -24 dB 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.



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

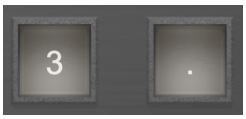
This block generates the reference tempo for metrical time delays. Depending on the setting of the controls in the UI, this may generate the tempo from data provided by the host, by the rate of taps on the tap button or directly from the Tempo knob if the tempo is set manually.

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.



Triplet and Dotted Note Length modifier buttons

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.



Phase Inversion Button - On (at left) & Off (at right)

> 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 plug-in for digital audio workstations which provides a flexible algorithm-based reverb to add depth and ambience to your recordings.

Features include:

- Comprehensive controls
- Graphic display of the Low and High Frequency Pre-Filters
- Graphic display of the Reverb Impulse



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:



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.



Spread Knob

The ring around this encoder starts at 12 o'clock and spreads to both sides equally as the Stereo Width is increased.



Plus/Minus Knob

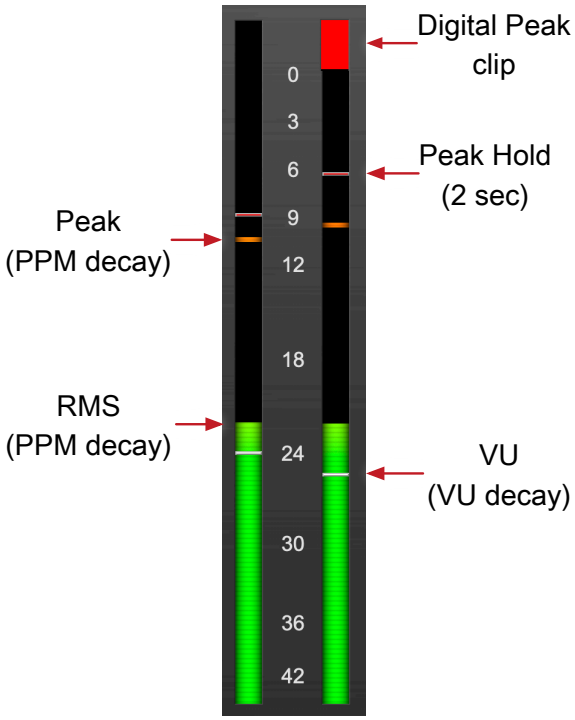
The rings around these encoders start at 12 o'clock and sweep to either side. These knobs are used for Wet/Dry and Gain controls.

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 Mac ⌘ (Command) key or Windows **Control** key when you click, you will be able to adjust the value with finer precision. If you hold the Mac ⌥ (Option) key or Windows **Alt** 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**, Mac ⌘**.** (Command + .), Windows **Alt** key 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 Mac ⌘**.** (Command + .), Windows **Control.** (Control + .) 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



HaloVerb Output Level Meters

For the main output stage of HaloVerb we have provided meters driven with SpectraFoo™ metering technology. These meters show digital Peak, Peak Hold, RMS level and VU level.

Both the Peak and RMS level are represented with fast PPM ballistics.

The instantaneous Peak level is represented by the floating colored pip. Peak Hold is 2 seconds, indicated as a red floating marker pip.

The RMS level is the solid colored bar which transitions from green at "safe" high-headroom levels to yellow at -18dBFS, then to orange and finally red approaching -6dBFS.

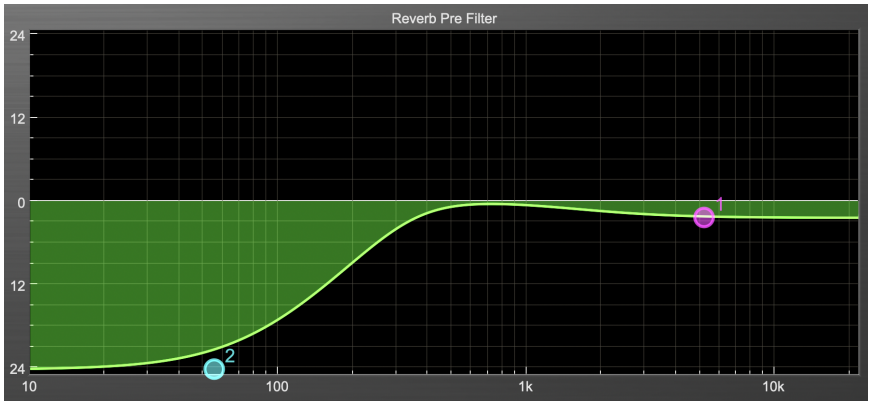
The VU meter is the white floating marker pip showing IEEE standard 300 ms RMS average level.

When HaloVerb is on a mono insert there will be a single meter.

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.

> **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.

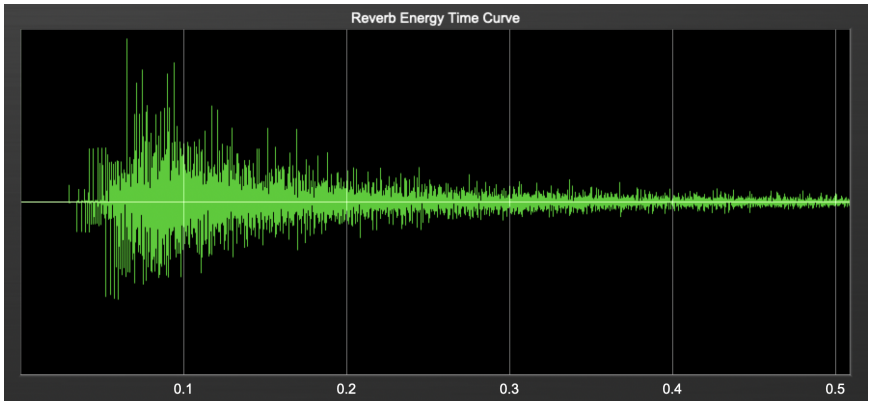
Low and High Frequency Pre-filter



Low and High Frequency Pre-filter

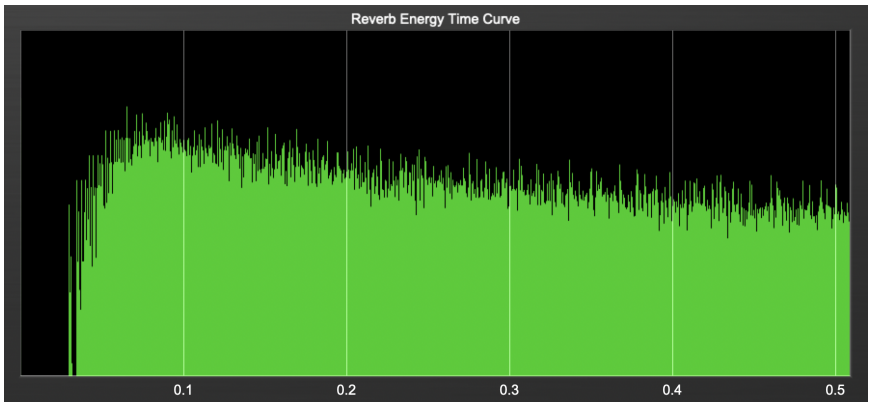
The pre-filter graph is a visual representation of the filter shape as set by the Cutoff Hz and Cutoff dB controls for the Low and High filters. As these controls are changed, the dots shows the knee freqencies and levels. You may also click and drag the dots to change the filter placements.

Reverb Impulse Display



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.

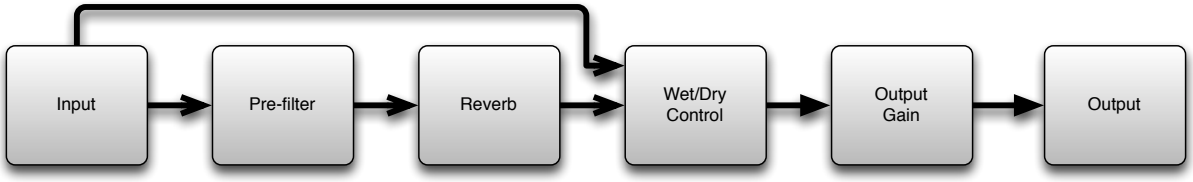


Energy Time Curve

Processing

A Detailed Description

In this chapter we'll discuss how the controls work.



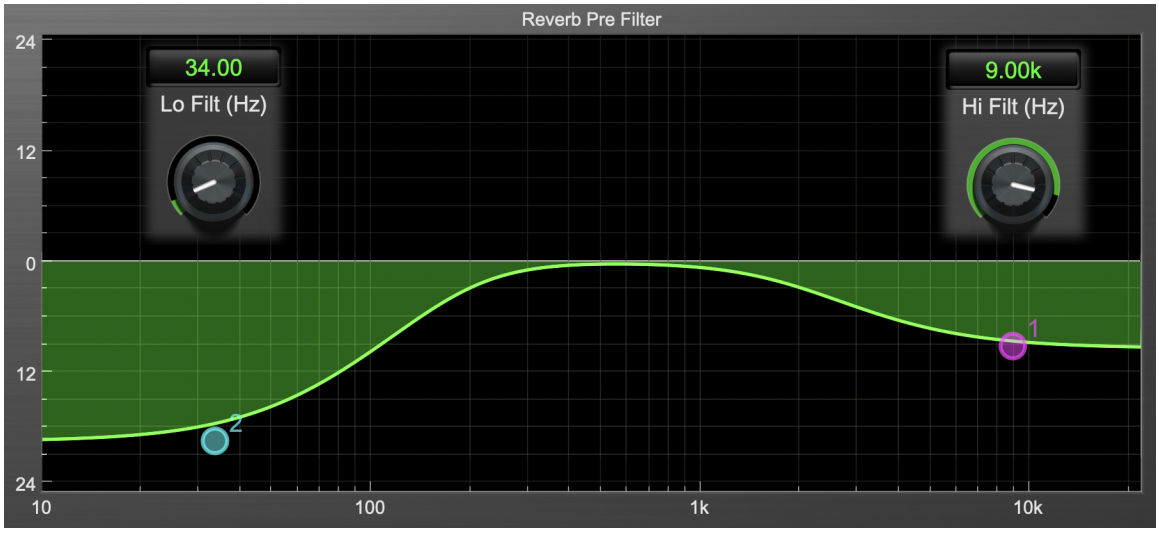
Signal Flow

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.



Pre-filters at 34 Hz and 9kHz (controls shown graphically overlaid for clarity)

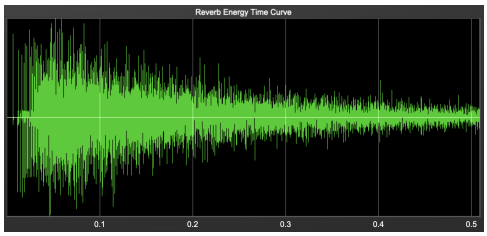
Reverb

Let's take a look at what HaloVerbs 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.

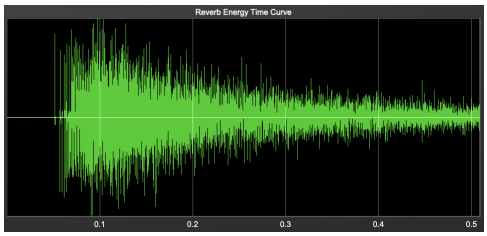
Please note that some combinations of settings will look similar in the impulse display, but will sound very different.

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:



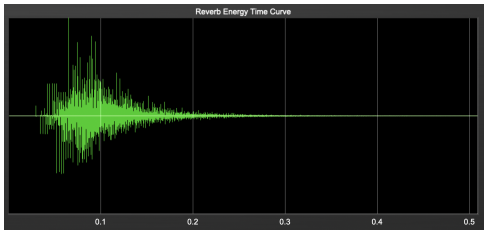
Predelay of 0 ms



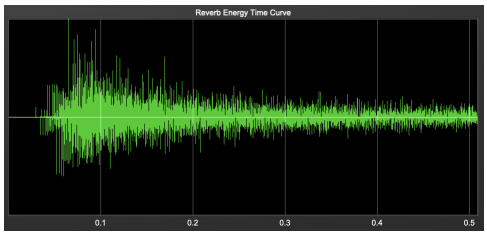
Predelay of 45 ms

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:



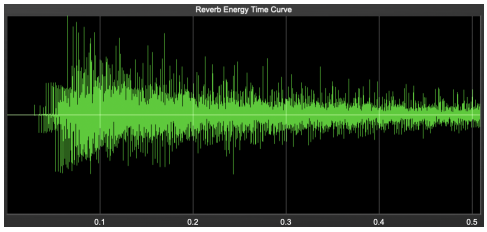
RT60 of .1 Seconds



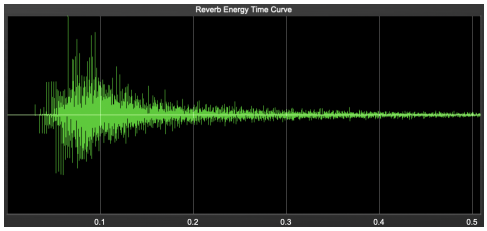
RT60 of 7 Seconds

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.



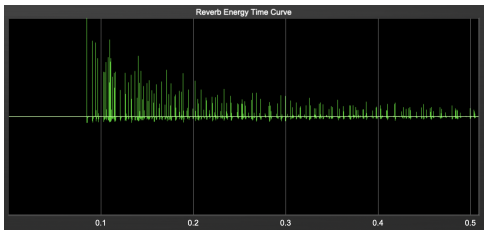
Damping of 0%



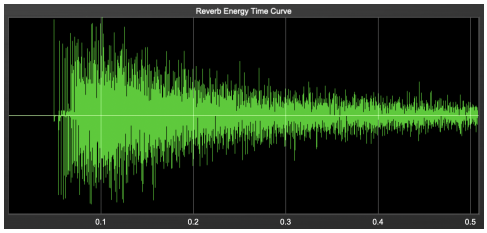
Damping of 100%

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.



Diffusion of 0%



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 plug-in for digital audio workstations which provides frequency based dynamic processing.

Features include:

- Three-way crossover
- Three independent channels of compression
- Limiter
- Disclosable graphic displays
- Dynamic transfer function display
- SpectraFoo™ spectrum analysis
- SpectraFoo™ multi-function output level metering



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:



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.



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.

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 Mac ⌘ (Command) key or Windows **Control** key when you click, you will be able to adjust the value with finer precision. If you hold the Mac ⌥ (Option) key or Windows **Alt** 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**, Mac ⌘**.** (Command + .), Windows **Alt** key 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 Mac ⌘**.** (Command + .), Windows **Control.** (Control + .) 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



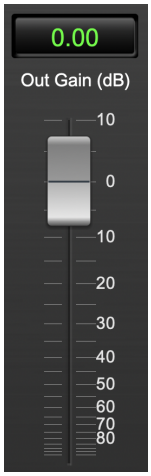
Toggle Button (Off)



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



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



Band Solo Button
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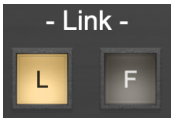
The band solo button allows you to listen to an individual crossover band or combination of them. This allows you to hear the effect of each compressor section independently.

Master Link Control



Leader:
Followers will follow changes made to this bands' parameters



Follower:
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Not Linked:
does it's own thing

Lead and Follow Link Buttons

Each compressor band has a set of Lead/Follow parameter linking buttons.

The **L** button sets which band will be the link control Leader.

The **F** button sets which bands will *Follow* the link control Leader.

Changes made in the Leader link band to Threshold, Gain, Ratio, Knee, Attack and Release will change the same parameters in any Following bands by the same amount.

Click to toggle the link control for each band on or off.

Only one band can be set as the Leader. Clicking the master link control for a different band automatically de-selects the previous Leader band.

Gain Reduction Meter

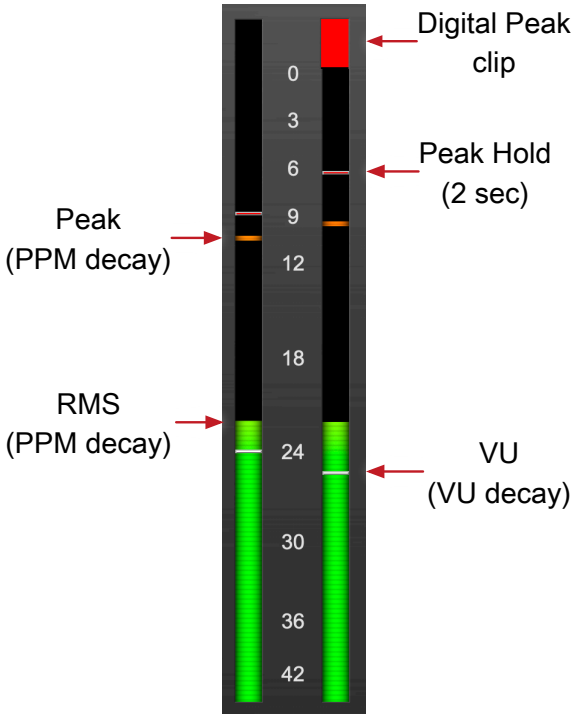


Gain Reduction Meter

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



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For the main output stage of Multiband Dynamics we have provided meters driven with SpectraFoo™ metering technology. These meters show digital Peak, Peak Hold, RMS level and VU level.

Both the Peak and RMS level are represented with fast PPM ballistics.

The instantaneous Peak level is represented by the floating colored pip. Peak Hold is 2 seconds, indicated as a red floating marker pip.

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When Multiband Dynamics is on a mono insert there will be a single meter.

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.

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



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 (Mac/Win) or Mac ^ (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
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- ±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:



Spectragraph Display

The traces are:

- White: Left channel instantaneous display
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- 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.

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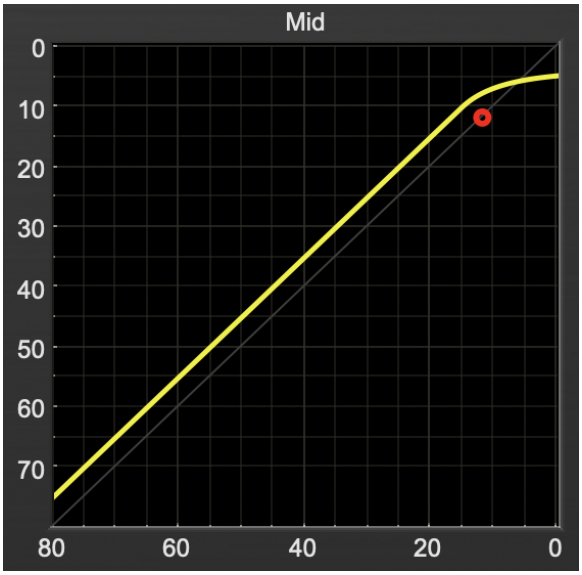
Dynamics Knee



Dynamics Knee

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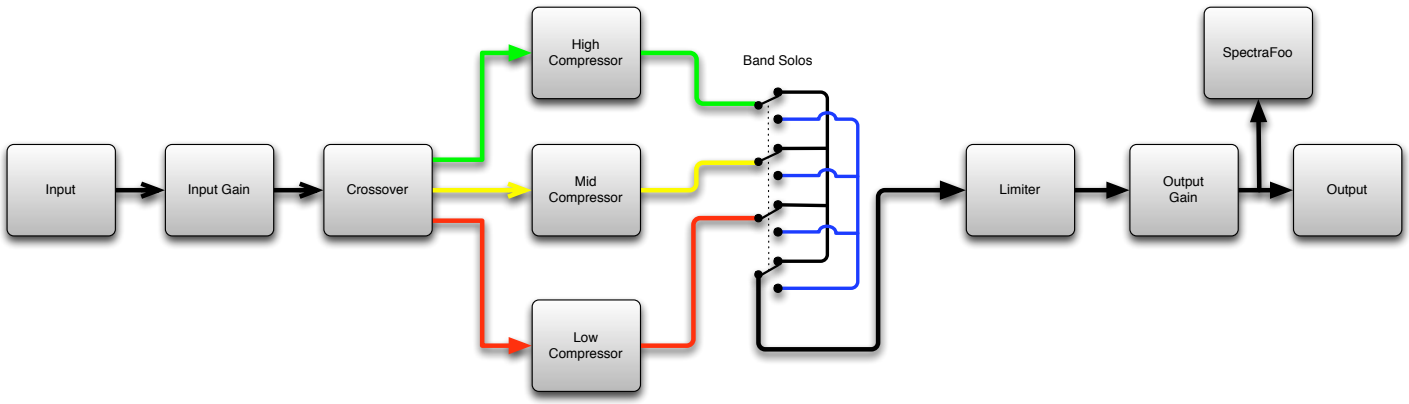
The horizontal axis corresponds to the input level and the vertical axis represents the output level. The heavy line shows the dynamic of the associated processing block. This means that if you sent in a sine wave at a given input level, the output level would be equal to the level shown on the graph. 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.



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

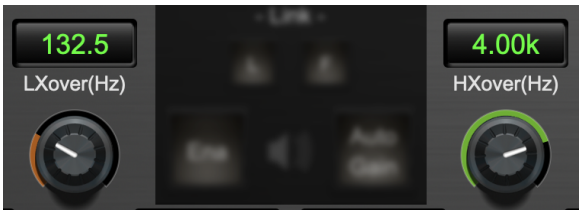
After the signal is routed to Multiband Dynamics it runs through an input gain block that provides input gain of up to +30 dB. You can use this gain to condition signals that are low in level.

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:



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:



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



Compressor (Low Frequency Shown)

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

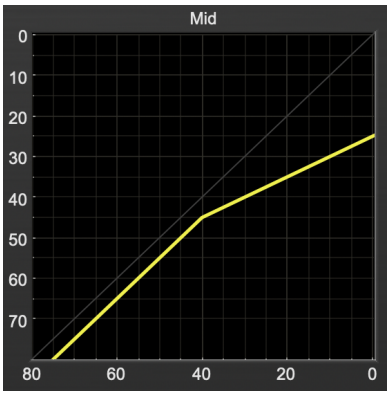
The "Gain" knob allows you to manually adjust the makeup gain applied to the signal after the gain reduction applied by the compressor by ±30 db. If the Auto Gain switch is off, this is the amount of makeup gain applied. If the Auto Gain switch is on, then this parameter is a trim added to the internally 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.

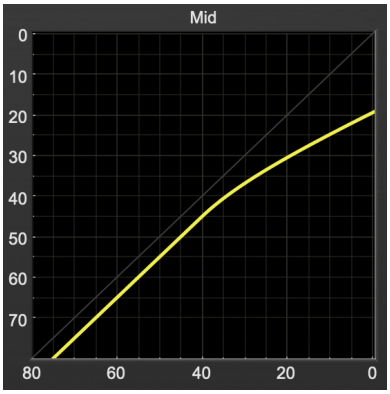
Compressor Knee Control

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.



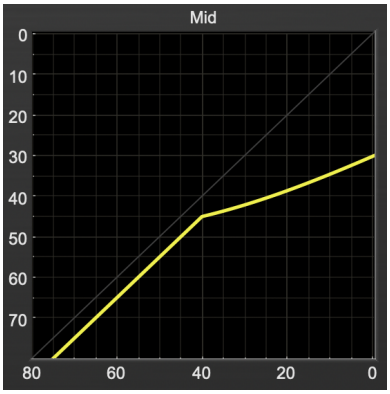
Knee Control at 0

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.



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.



Knee Control at -0.5

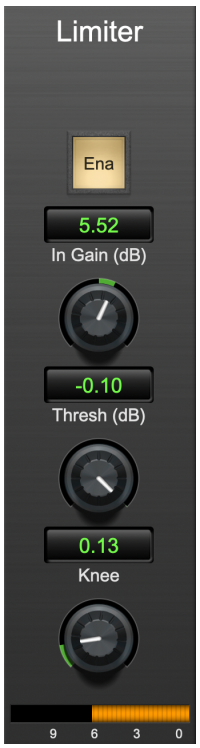
Attack Control

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



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.

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Multiband Expander

Plug-in categories: **Production**, **Dynamics** and **Mastering**

Introduction

Multiband Expander's features include:

- Input level control
- Three-way crossover
- Three independent channels of expansion
- Disclosable graphic displays
- Dynamic transfer function display
- SpectraFoo™ spectrum analysis
- SpectraFoo™ multi-function output level metering



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:



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.



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.

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 Mac ⌘ (Command) key or Windows **Control** key when you click, you will be able to adjust the value with finer precision. If you hold the Mac ⌥ (Option) key or Windows **Alt** 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**, Mac ⌘**.** (Command + .), Windows **Alt** key 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 previous control). Hit the Mac ⌘**.** (Command + .), Windows **Control.** (Control + .) 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.

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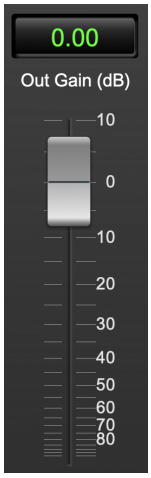

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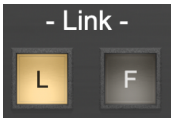

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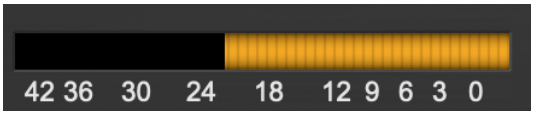

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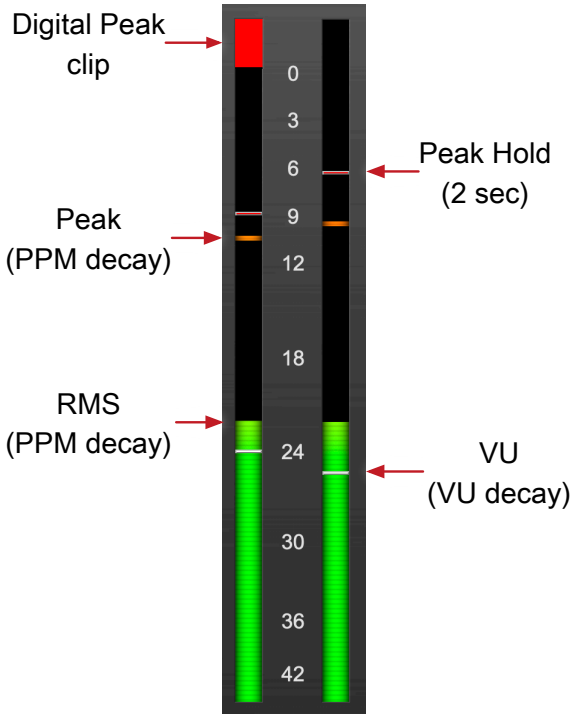

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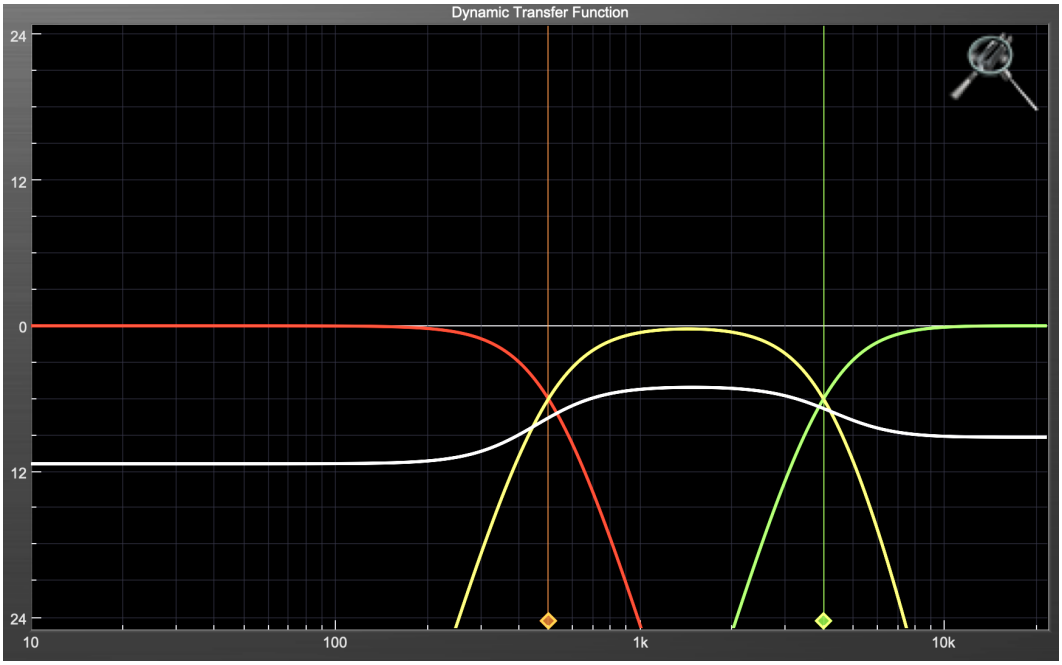

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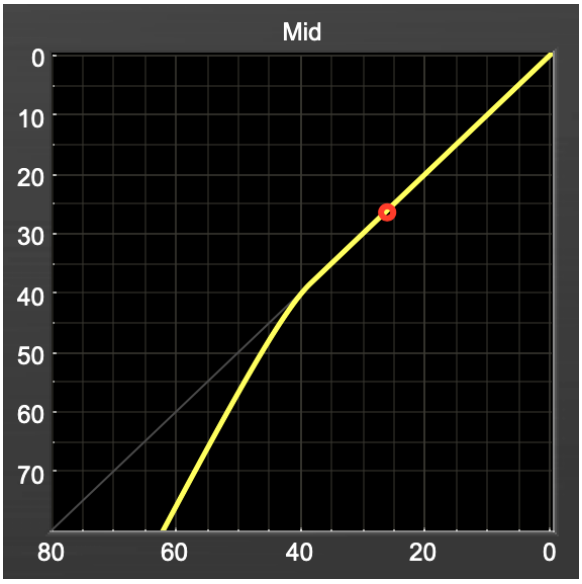

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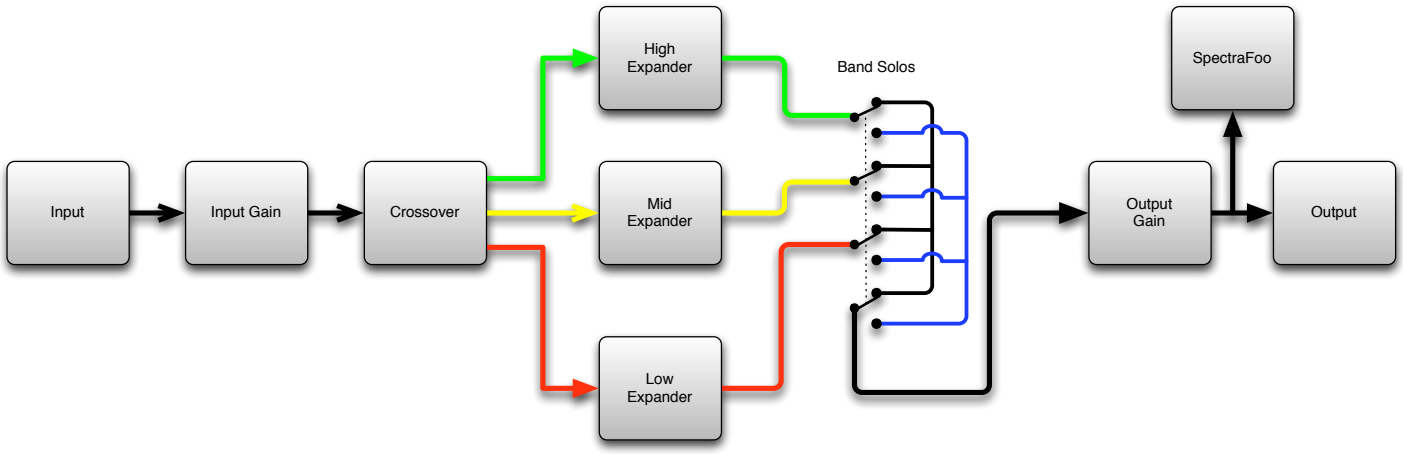

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Input Conditioning

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 separates 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:



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:



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



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 plug-in for digital audio workstations 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



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:



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.



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.

If you hold down the Mac ⌘ (Command) key or Windows **Control** key when you click, you will be able to adjust the value with finer precision. If you hold the Mac ⌥ (Option) key or Windows **Alt** key when you click, the knob will reset to its default value. You may also double-click a knob to reset it.

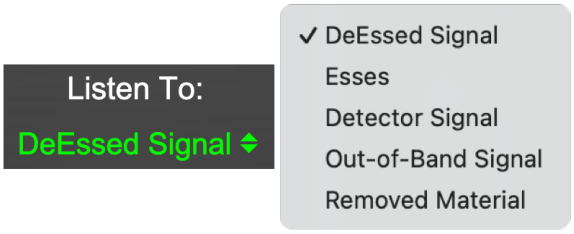
Click on the number (readout) of the knob to display a text entry field that allows you to type in a number directly. The pop-up will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, Mac ⌘**.** (Command + .), Windows **Alt** key or **ESC** keys.

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 Mac ⌘**.** (Command + .), Windows **Control.** (Control + .) 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.

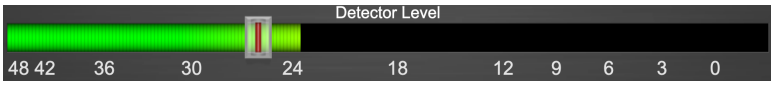
Listen Control



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



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**/**Alt**-click the meter to reset the clip light.

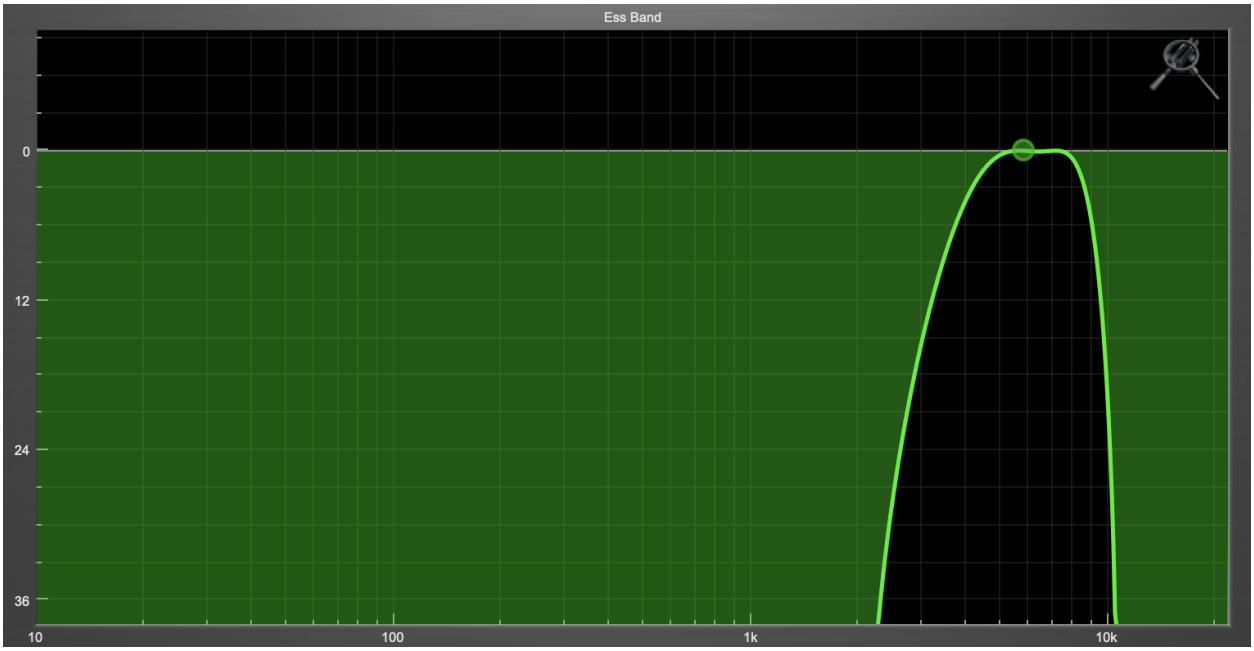
Gain Reduction Meter



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 (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

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.

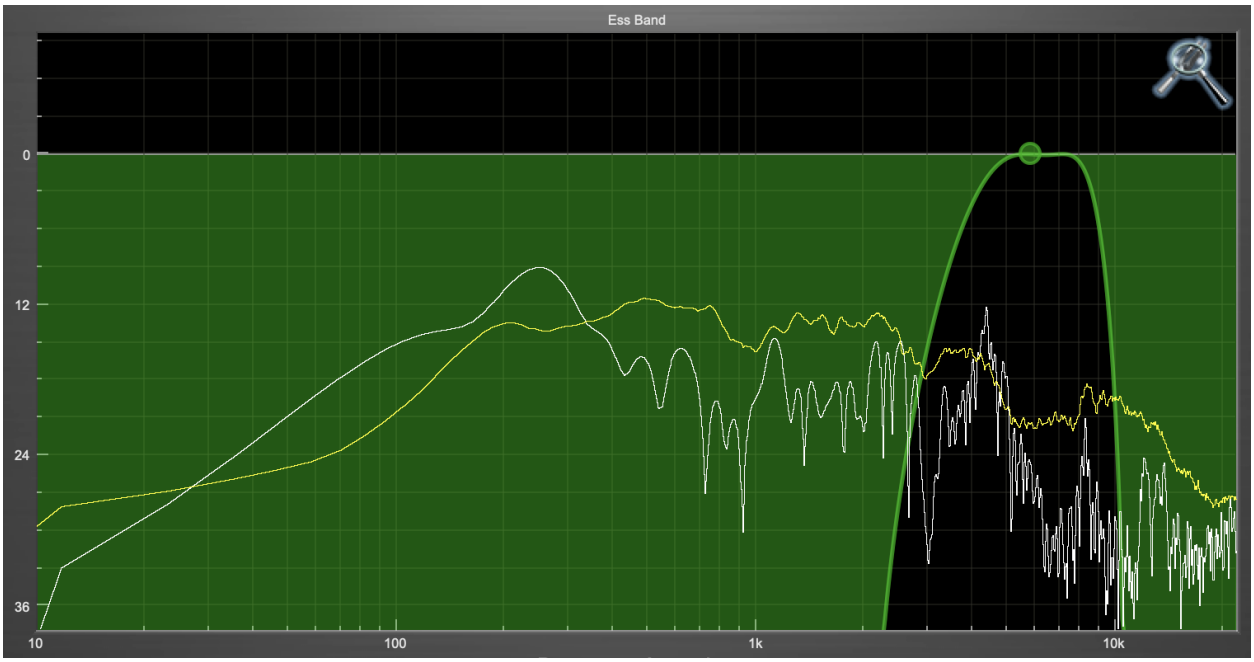
Mac ⌥ (Option)–click or Windows **Alt** the dot to adjust the bandwidth (dragging right increases the bandwidth, left decreases the bandwidth).

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.

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:



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 (Mac/Win) or Mac ^ (Control) click on the transfer function, you will see a menu to set options for the spectragraph:

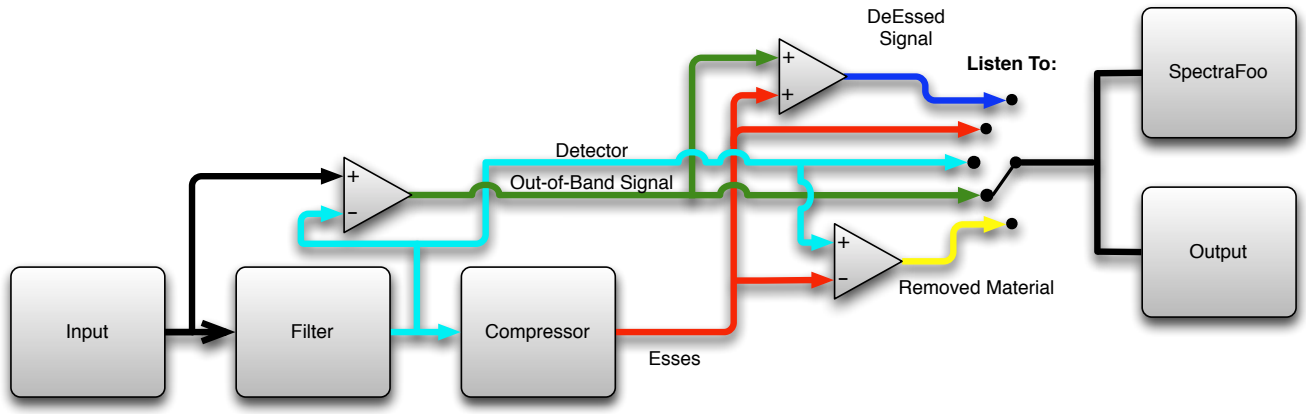
- Show Instantaneous Trace: Toggles whether the spectragraph shows the instant response of your audio.
- Show Average Trace: Toggles whether the spectragraph shows the averaged response of your audio.

These settings are stored for each instance of the Precision DeEsser.

Processing

A Detailed Description

In this section we'll discuss how the controls work.



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.



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.



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.

Sonic Eq

Plug-in categories: **Production**, **EQ**, and **Mastering**

MH Sonic EQ Overview

Metric Halo Sonic EQ has six independent minimum-phase filter bands plus 5 function-specific equalization topologies. Each band offers your choice of 8 filter types and four filter orders, with in-band and out-of-band Ripple control for the higher orders.

Processing functions include:

- Original Sonic Solutions SonicSystem™ filter coefficients, personally blessed by the creator, Dr. James A. Moorer
- Per-band adjustable filter slope order
- Process Oversampling up to 16x
- Passband Ripple and Stopband Ripple resonant filter controls (2nd through 4th order)
- Updated and improved SpectraFoo™ spectrum analysis
- EQ Transfer Function *Live Readout* displays the precise filter curve gain at the mouse cursor frequency
- multifunction Spectrafoo™ level meters display realtime Peak, RMS and VU of source input and post-EQ output levels



History

MH Sonic EQ is an modern version of the original Sonic EQ developed by Dr. James A. Moorer for Lucasfilm's SoundDroid project, and then expanded for the Sonic Solutions SonicSystem mastering DAWs in the '90's and early 2000's.

In 2007 Sonic EQ was re-created and enhanced for the Metric Halo-designed Sonic Studio 30x Interfaces. The Sonic Studio 30x version of Sonic EQ went back to first principles, updating the UI and engine.

MH Sonic EQ today is a 6-band fully parametric equalizer in which every band in the EQ supports 2nd to 8th order filters using an updated, low-noise 64-bit state-space EQ structure with the core Sonic EQ coefficients.

In every generation of Sonic EQ, three primary characteristics have remained consistent:

1. the *SOUND* - a full, sweet yet powerful musicality which brings a presence and clarity to anything you pass through it,

2. its *precision* - with high-order slopes and a high degree of control over filter ripple, both in-band and out, Sonic EQ makes it easy to manage very specific frequency content with minimal effect on surrounding program,

3. and its unrivaled *versatility*. Whether working in forensic restoration, sound design, high-end mastering, post-production, recording, re-recording, multi-tracking or live sound, the power and simplicity of this EQ 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 that look completely broken, yet still sound perfectly clean and natural (refer to the screenshot on the preceding page).

Sonic EQ also provides a special set of laboratory-grade filters for vinyl (RIAA encode and decode) and early generation digital formats transfer (DC removal and digital de-emphasis).

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.

This version of Sonic EQ, originally incorporated exclusively in the hardware DSP of Sonic Studio branded Model 30x audio interfaces is now reborn as an AAX, VST and AU plug-in for Mac and Windows DAW users with a cleaner layout, SpectraFoo™ spectrum analysis and level metering and an even higher order of musicality, precision and power.

Operation

Sonic EQ's control interface employs the same "virtual knob" encoder elements seen throughout the MH Production Bundle 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 Sonic EQ:



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).



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.



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.

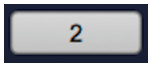
  Hint: If you hold down the ⌘ (Command) key when you click, you will be able to adjust the value with finer precision.
- ⌥ (Option)-click to reset any knob to its default value. You may also double-click a knob to reset it.
- **Rapid parameter entry**
  - Double-click the numeric readout of the knob to 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 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 ghosted (visible, but mostly transparent and unresponsive to touch).
>
> The De-emphasis, DC and RIAA filters are dedicated-purpose with no adjustable parameters.

Toggle Button



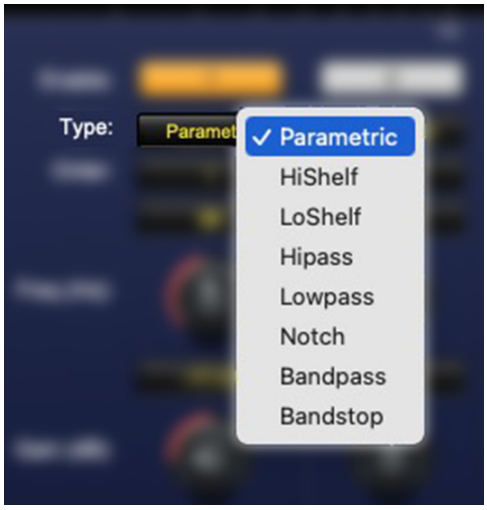
Filter Enable (On)



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 and engage "Compare" and "Bypass" in the plug-in header.

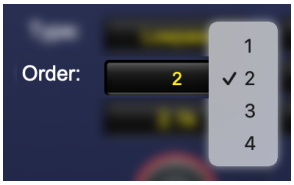
Filter Type selector



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.

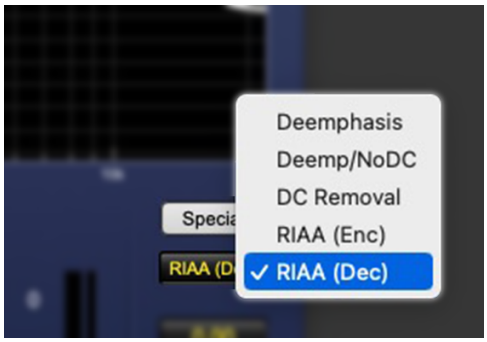
Filter Order selector



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.

Special Filters selector



Special Filters selector (with submenu)

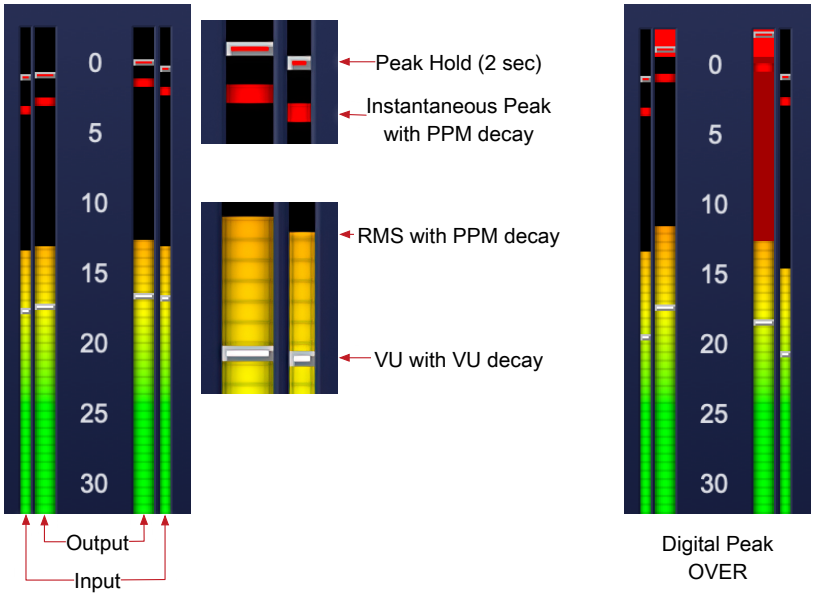
An additional set of equalization tools is available to the far right, above the master output fader. Special Filters includes RIAA encode/decode, DC Removal and digital Deemphasis.

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.

Input/Output Meters



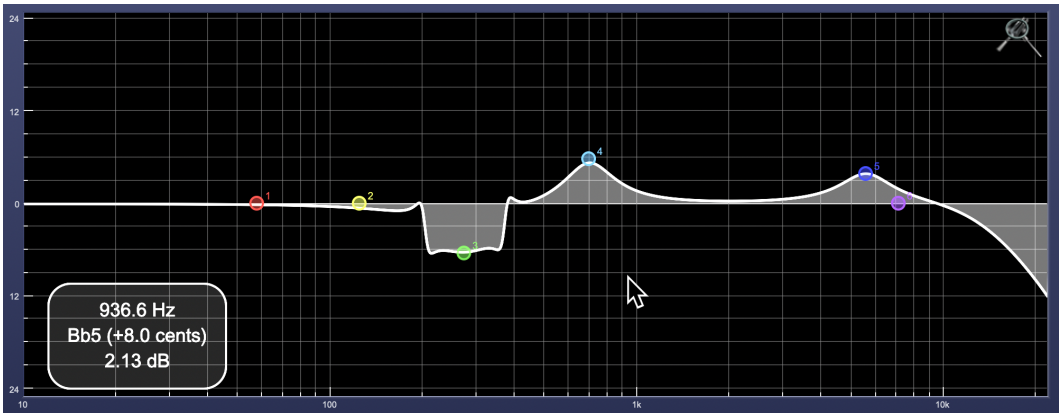
Sonic EQ's multifunction level meters simultaneously display instantaneous Peak, Peak Hold, RMS and VU levels for both the pre-EQ source input and post-EQ output levels.

Source Input levels are displayed on the thinner meters adjacent outside of the larger left and right Output meters, for a more immediate and accurate I/O gain reference.

Digital OVER is indicated when the meter above the RMS level bar flashes red with a peak hold pip above 0dBFS. In the graphic above right, the right output channel clipped a bit harder than the left, so the OVER indication on the right output meter is taking longer to fade.

Note that 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. You should lower the output level to ensure you do not actually clip the input of another processor or D/A convertor.

EQ Transfer Function



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 line (white by default) 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 dot represents the center frequency or knee of the filter band, and acts as a control handle for the band within the transfer function field.

The band that is currently being edited will be highlighted and 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 band handle dot and dragging will allow you to adjust the frequency and gain of the associated band.

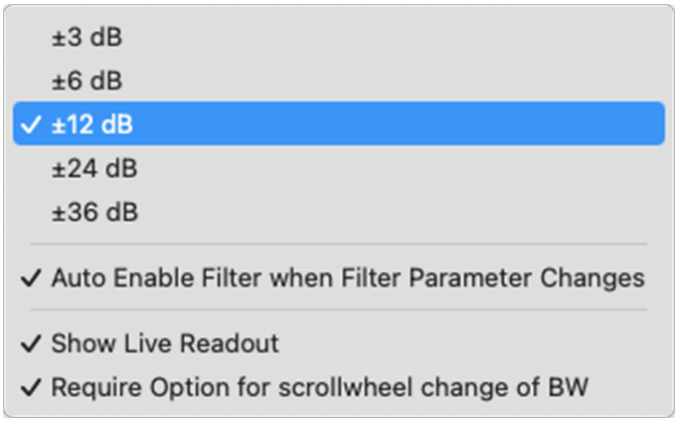
- ⌘ (Command)–click or double-click the dot to toggle the band enable (single-click to disable a band when Auto Enable Bands is turned on).
- Click and drag the smaller dots associated with a larger dot to adjust the filter Q. You can also ⌥ (Option)–click the dot to adjust the Q (dragging left increases the Q, right decreases the Q).
- ⌘⌥ (Command + Option)–click the dot to switch the band filter type.

In the transfer function field, multiple filter bands may be temporarily grouped by shift-click or click-drag selection of the filter control handles.

To dismiss the filter curve, click anywhere in the black area of the transfer function. This will deselect the filter, and the only trace displayed will be the master eq curve.

Transfer Function options

Right-clicking the EQ transfer function background brings up an EQ functions control menu.

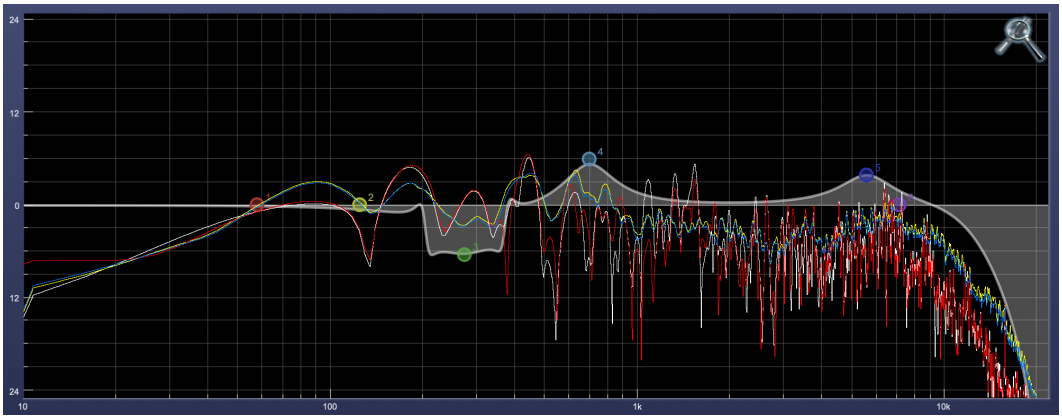


- EQ display scale: Sets the EQ curve dB scale from ±3 to ±36 dB.
- **Auto Enable Bands when band parameter changes**: Automatically enables a filter band when one of its parameters is adjusted within the EQ Transfer Function. This preference will apply to all instances of Sonic EQ.

  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 or the Special filters.
- **Show Live Readout** (shown above): Enables an overlay in the lower left corner of the transfer function which displays the relative amplitude of the master EQ curve at the frequency indicated by your mouse cursor. Moving the cursor off the EQ display hides the readout. Live Readout control is enabled per instance.
- **Require Option for scrollwheel change of BW**: Enabled by default, this control requires the Option modifier key to change the Q parameter using a mouse scrollwheel. Scrollwheel filter width control works for filter band knobs (one at a time) and selected filters in the transfer function display (multiple selections supported).

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 overlaid on the Transfer Function EQ curve:



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 filters.

To disable the spectragraph entirely, click the active Spectrafoo icon.

If you right-click or ^ (Control) click on the Spectrafoo icon, 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.

These settings are stored independently for each instance of Sonic EQ.

Note that just the Left channel display is shown by default.

Controls for Left and Right channel will only be available in stereo instances of Sonic EQ.

Processing

Sonic EQ 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.

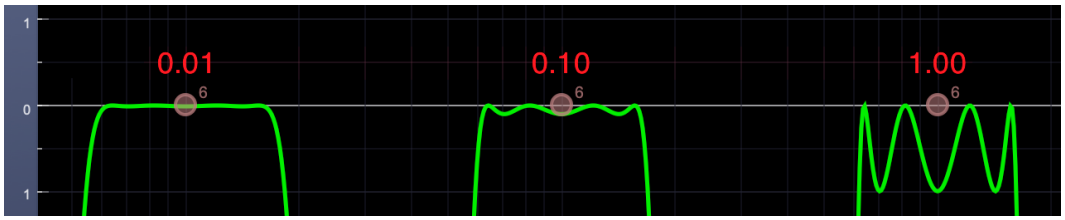
Expressing the width of a filter as a 'Quality Factor', rather than bandwidth, provides a more intuitive sense of the filter's subjective "sound", since the same value of Q will produce different bandwidths at different frequencies. The higher the frequency, the wider the bandwidth will be for a given 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 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 Gain range is ±24dB.
- **Q** factor is a numerical value that describes the filter 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 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 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.



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 (EQ curve colored green in the UI Prefs for visibility).

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 design/effects territory for audio filters.

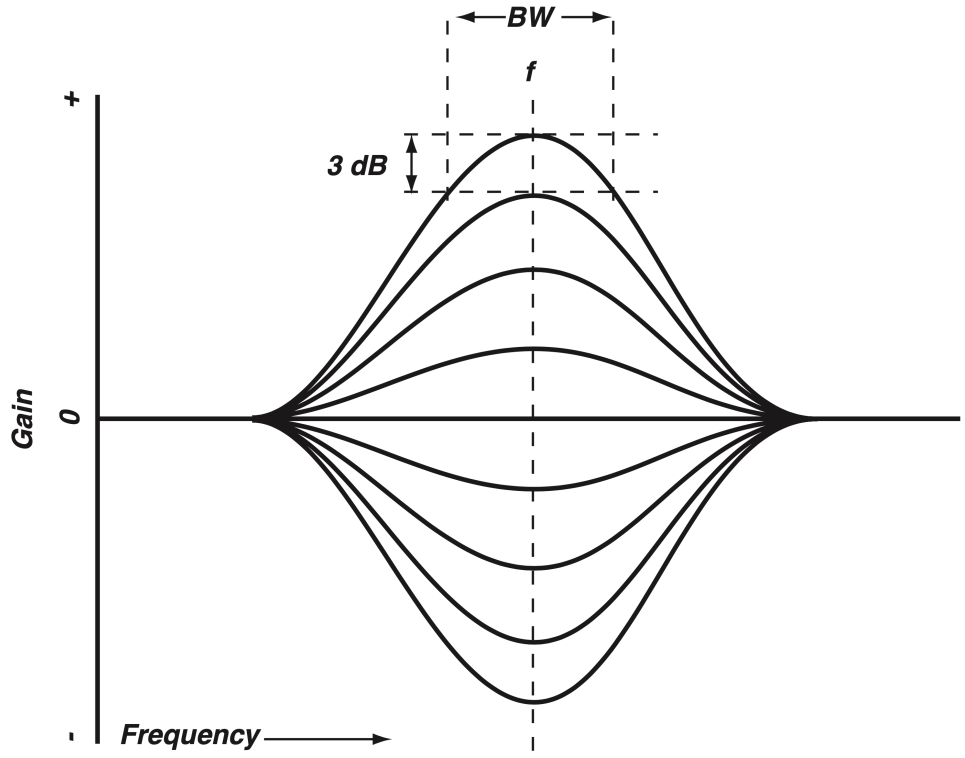
- **Stop Ripple** (short for Stopband Ripple) describes the amount of amplitude variation or ripple in a filter's out of band response. Indirectly, it describes two more important parameters. One is out of band suppression or, how much "leakage" of unwanted signal you receive, and the other is phase 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



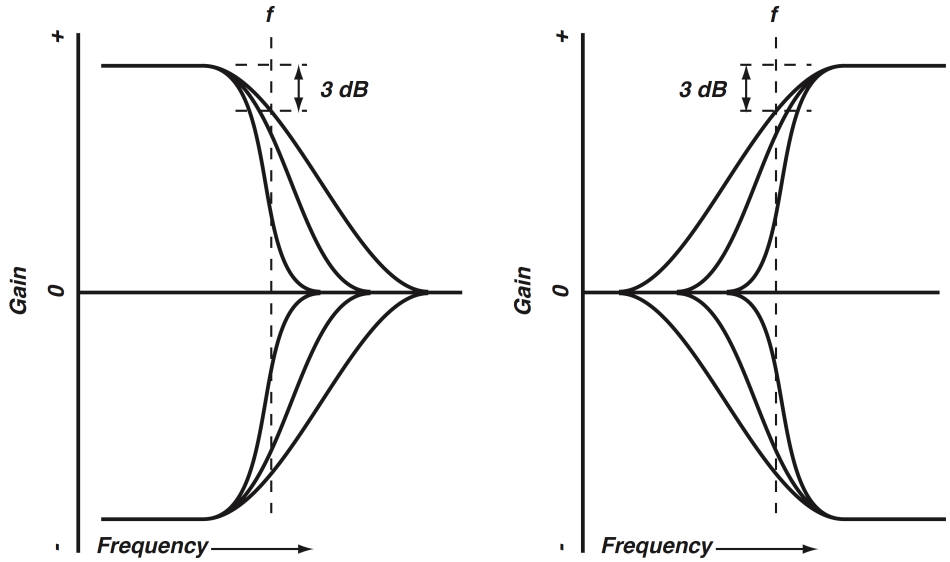
Parametric Filter Response Curves

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).



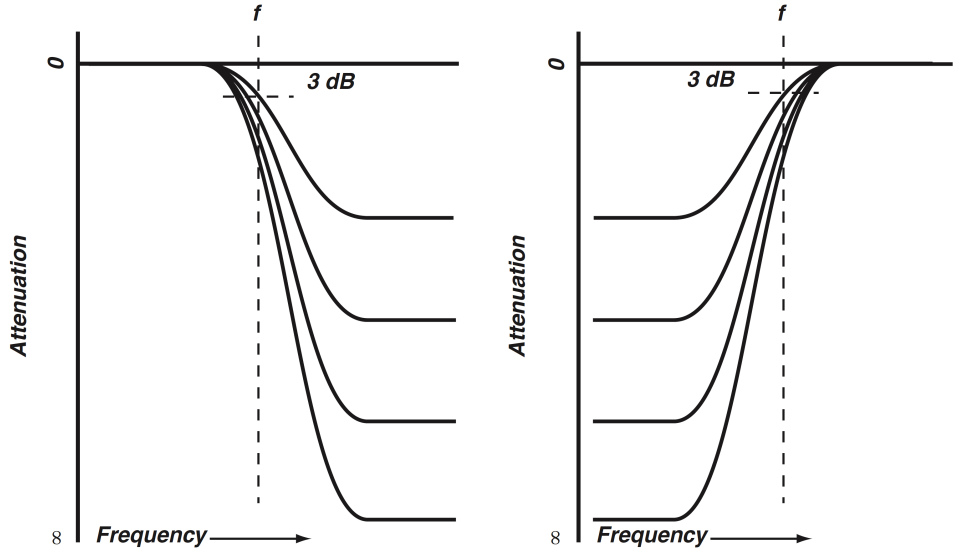
LoShelf (left) and HiShelf (right) Filter Response Curves

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.



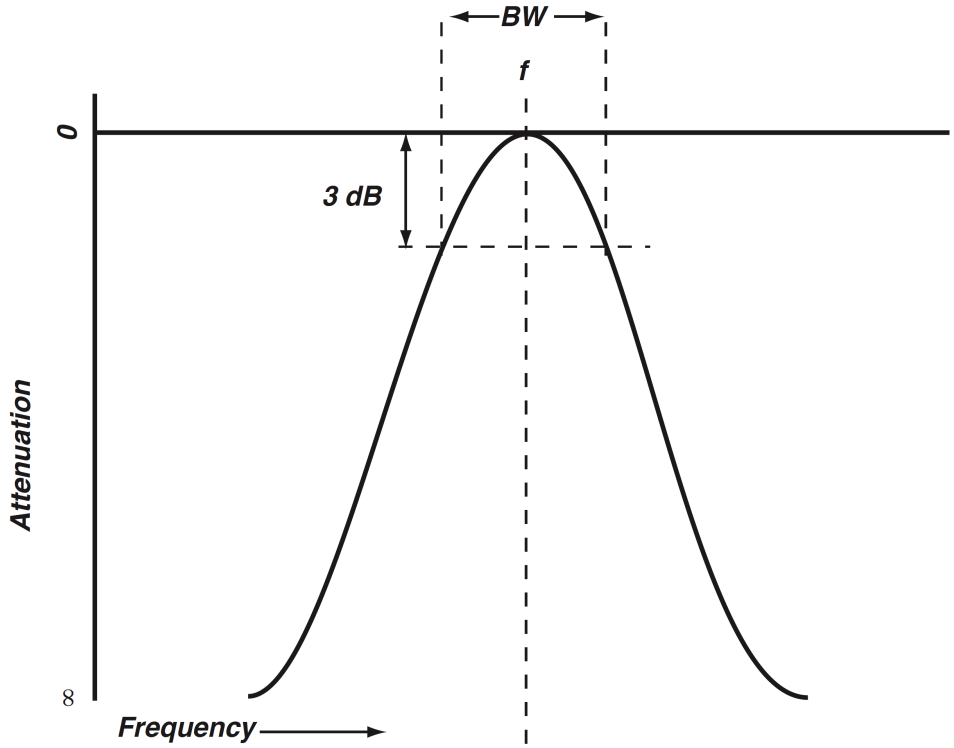
LowPass (left) and HiPass (right) Filter Response Curves

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.



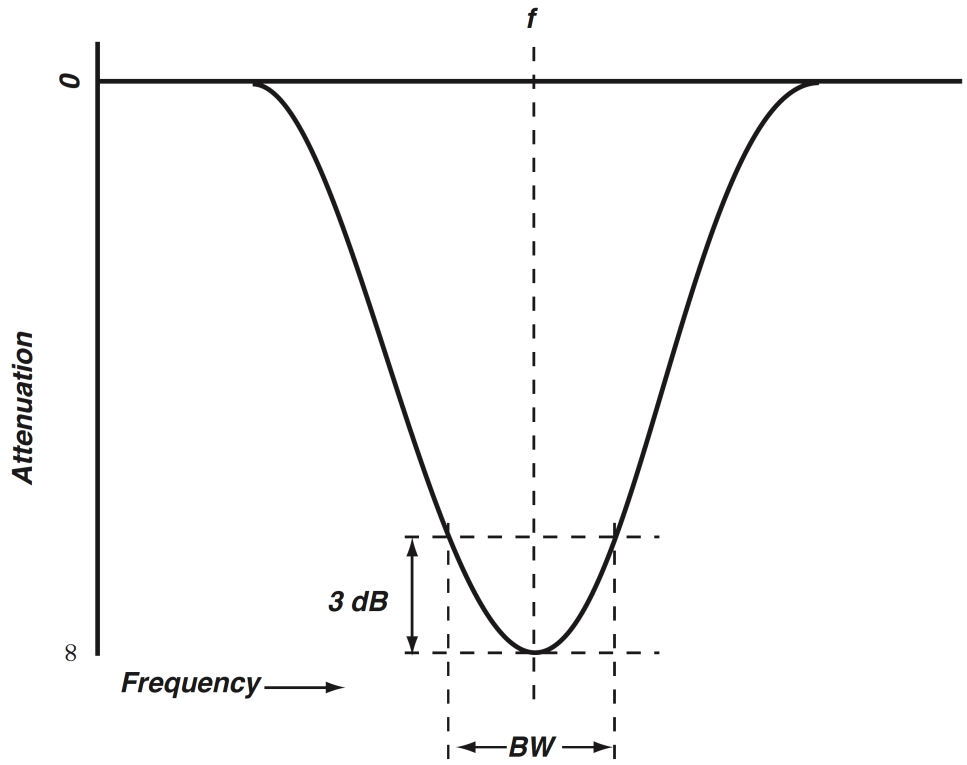
Bandpass Filter Response Curve

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.



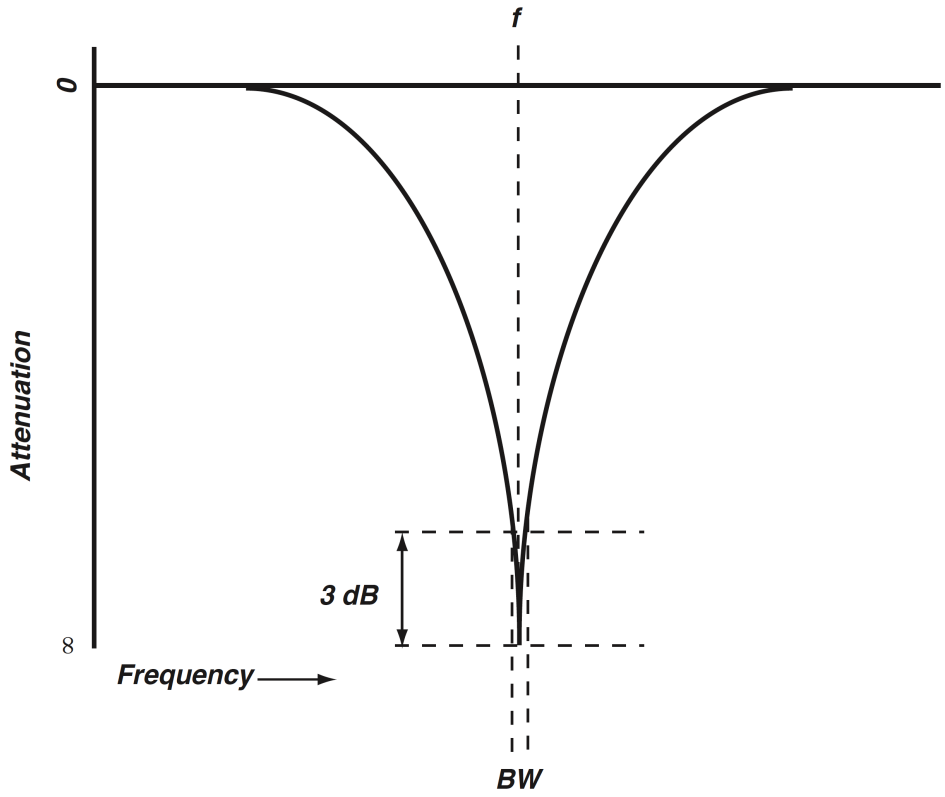
Bandstop Filter Response Curve

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.



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.

Special Filters

Above the master output fader is a button labeled *Special*. This section houses an additional filter stage containing specialized filters included in the original Sonic Solutions SonicSystem filter suite for vinyl production and early digital formats transfers.

There are no adjustable parameters for the filters within the Specials section, rather they are applied as an additional filter band to the master Sonic EQ filter curve to be modified to taste by the six primary filter bands.

The Special filters section is inactive by default, with DC Removal as the default setting.

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 designed to transparently 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 at the default display scale, but you can see it by zooming in.

In most cases this filter will be inaudible, but engaging and disengaging the filter while playing program containing significant DC offset is likely to produce a digital click. This is normal, especially considering you may be correcting a digital offset of several dB.

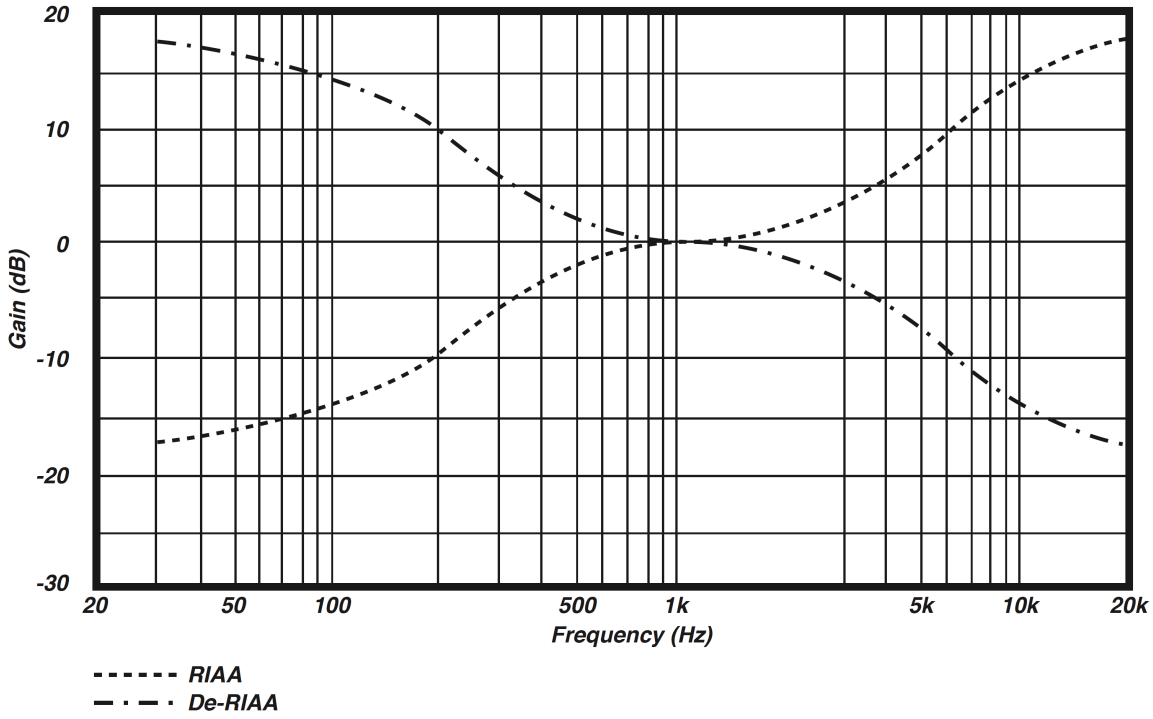
This filter is an extremely handy tool for cases where DC offset reduces the digital headroom of either the positive or negative peaks, increasing the chances of clipping distortion during processing.

Note this is not intended to correct for naturally positive-centric recordings of brass and similar instruments, but in any other case where the waveforms look otherwise symmetrical but centered above or below the 0.0dB gain line, try it.

RIAA Filters

The RIAA filter set is a pre-emphasis/de-emphasis standard developed for cutting and playback of vinyl records. RIAA filters are used to decrease low frequency levels in order to control physical groove width and add high-frequency emphasis at lacquer cutting, while an inverse RIAA filter in the phonograph cartridge preamplifier restores flat frequency response when the record is played on a turntable.

In Sonic EQ, the emphasis (mastering/cutting) filter is named RIAA (Enc); the de-emphasis (playback) filter is named RIAA (Dec).



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).

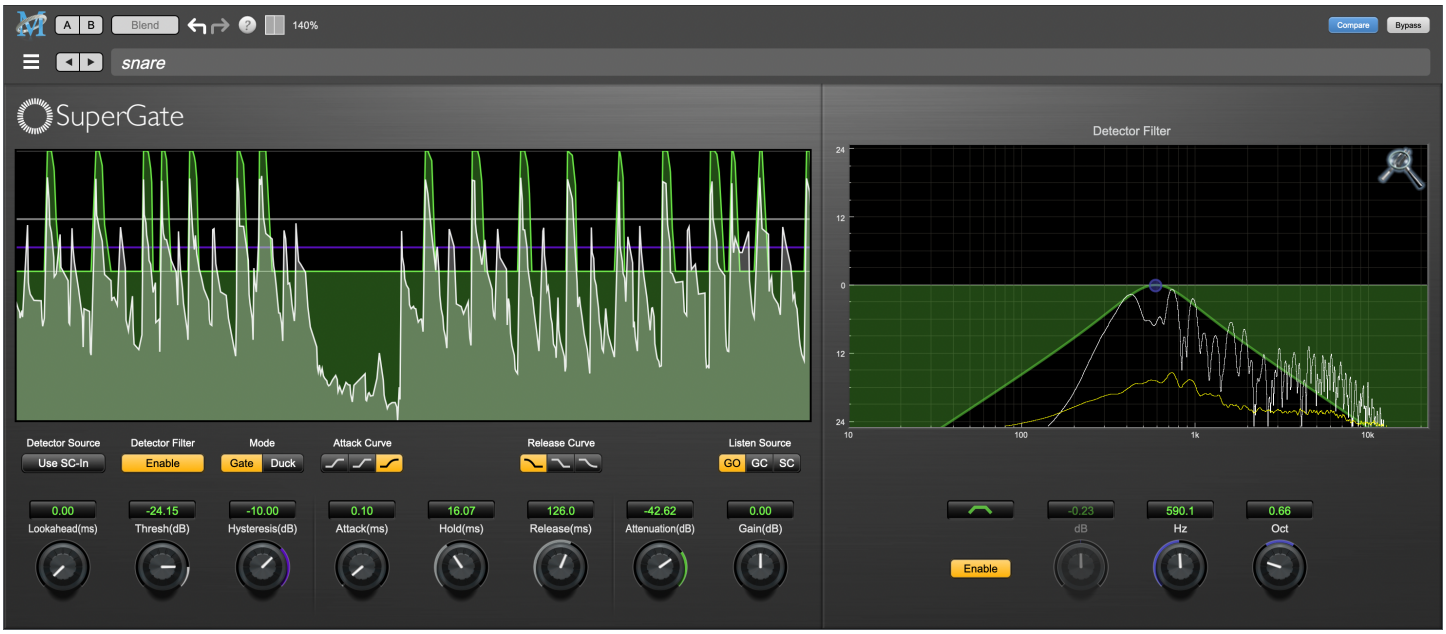
> WARNING: The RIAA filters should never be engaged while monitoring audio.
>
> The RIAA (Dec) filter is +20dB at 20Hz, and the RIAA (Enc) is +20dB at 20kHz.

SuperGate

Plug-in categories: **Production** and **Dynamics**

MH SuperGate Overview

SuperGate is a next-generation dynamics plug-in specializing in isolating (or removing) audio events based on the program dynamic signature.



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

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:



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.



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.



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



***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

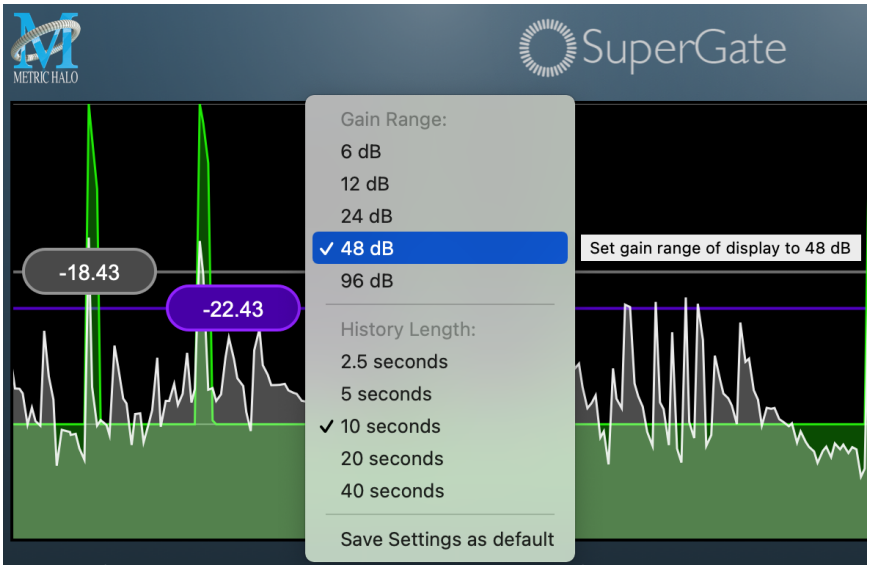


Radio Buttons (to choose one of each set)

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



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

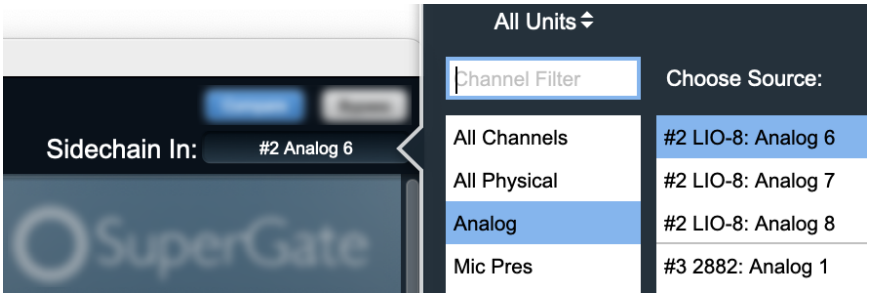
History Length is the period of time shown scrolling across the window. Events happening "Now" take place at the right edge of the screen, with recent events scrolling down the path of history to the left. The default History Length shows 20 seconds of audio events in the display. There are five settings 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.

Sidechain Input selector



"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.

Processing

SuperGate Theory of Operation

Yes, it's *that* much better. This is why…

Processing terms

- *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, functionally no different than a gain fader, 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, like any other gate, you have two jobs: find and isolate the target sound, and determine when to trigger the gate. Unlike other gates, SuperGate lets you customize the gate attack and release gain curves precisely to the sound you are isolating, as you listen, while visually referencing the Detector/Gain History display.

Further, the gate gain curves you build are not a set of dynamic expander parameters, but a programmed set of sample-by-sample instructions defining the gain structure and duration of the Attack-Hold-Release gate envelope itself. When the trigger point is reached during playback, this set of instructions modifies the samples describing the target sound, thereby applying the gate.

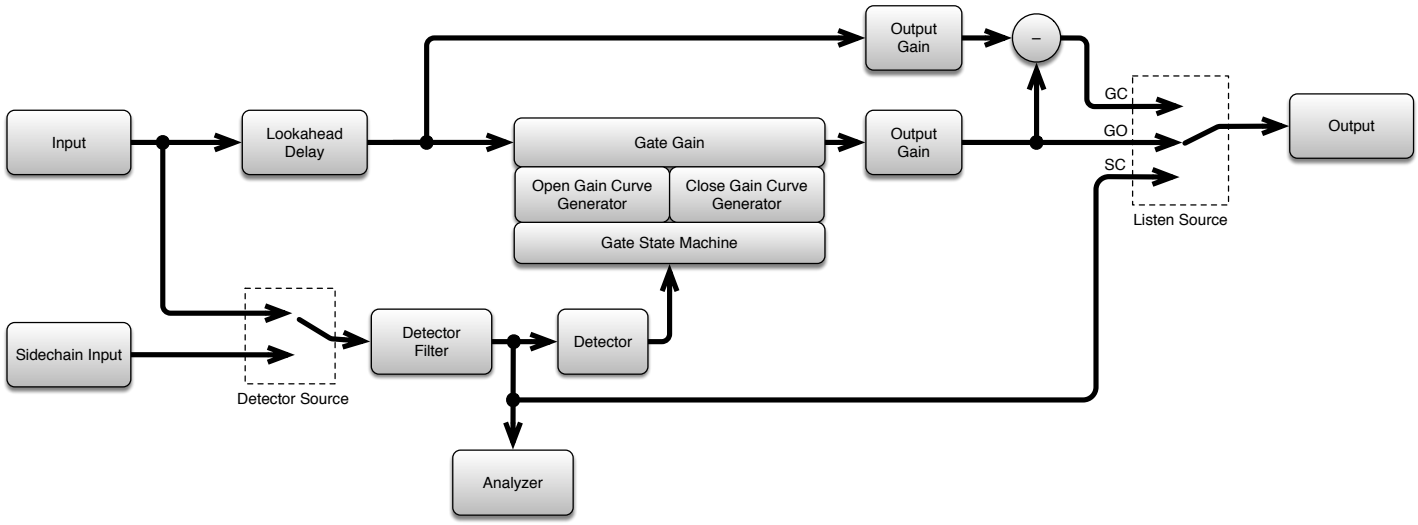
Applying the gain curves in this way completely avoids aliasing artifacts, the interpolation stages required to correct for them, and the modulation and smearing effects that degrade the gain transition.

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?

SuperGate Process Overview

The block diagram below illustrates the overall structure of the SuperGate.



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.

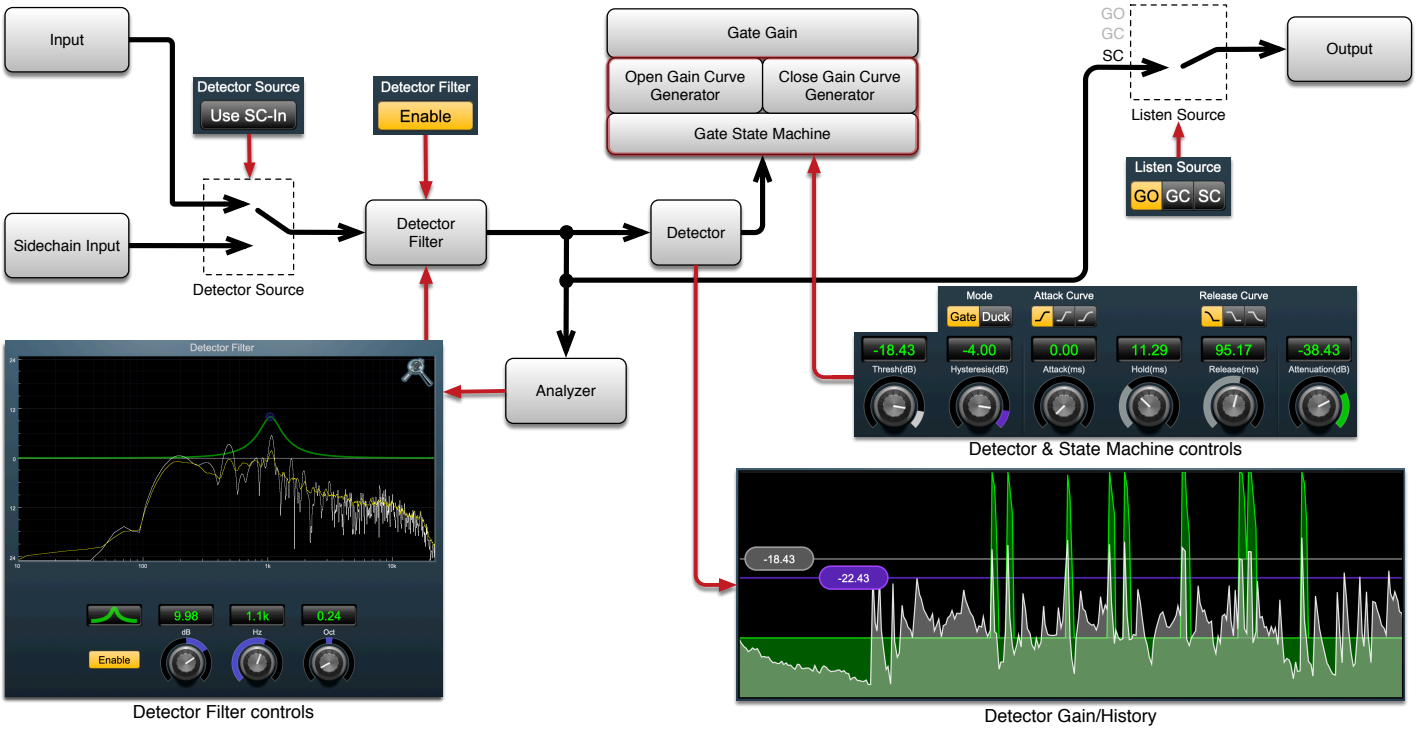
The detector itself doesn't actually react to any of the parameter controls; it is the state machine and the gain curve generators that do.

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.



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.

Detector Source



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.

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.

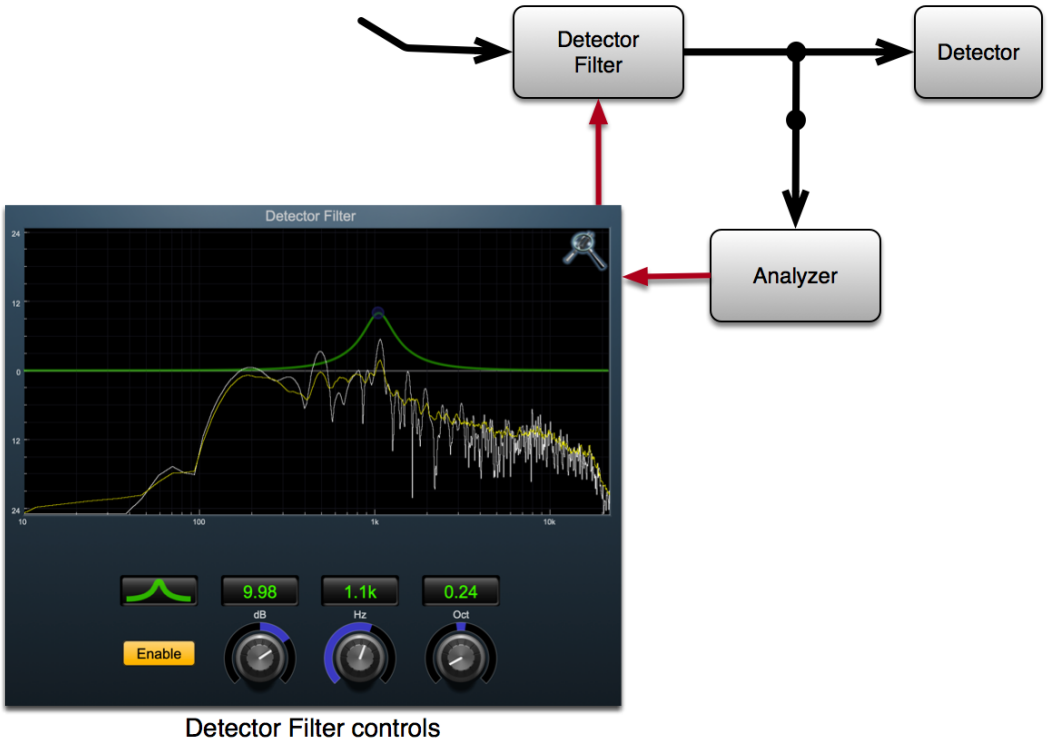
The Detector Filter

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.



Detector Filter (flow chart focus)

Detector Filter Enable

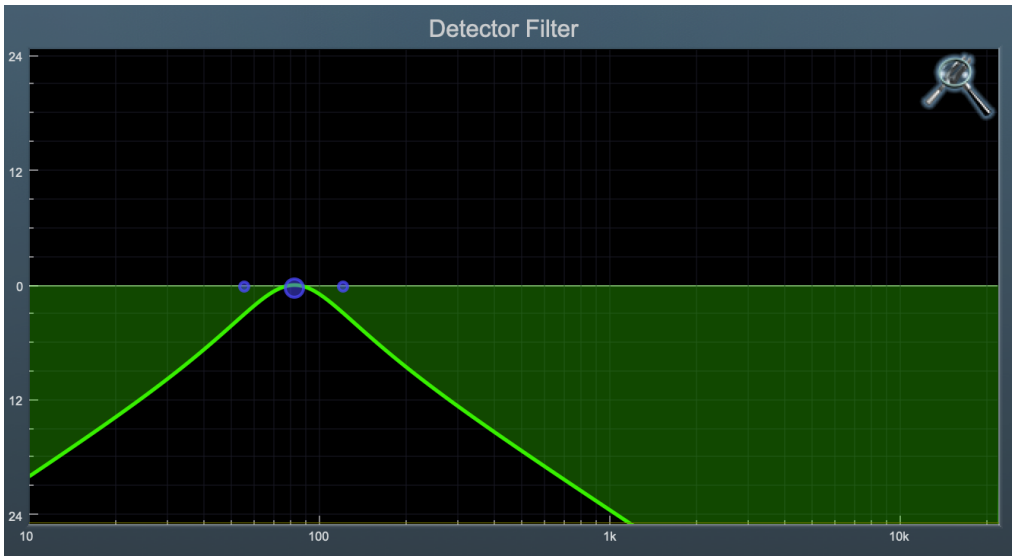
The *Detector Filter Enable* switch is used to insert the "Detector Filter" into the sidechain feed to the Detector. When dark, the filter is bypassed. When "Enable" is lit, the filter is in-line and processing.



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



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.

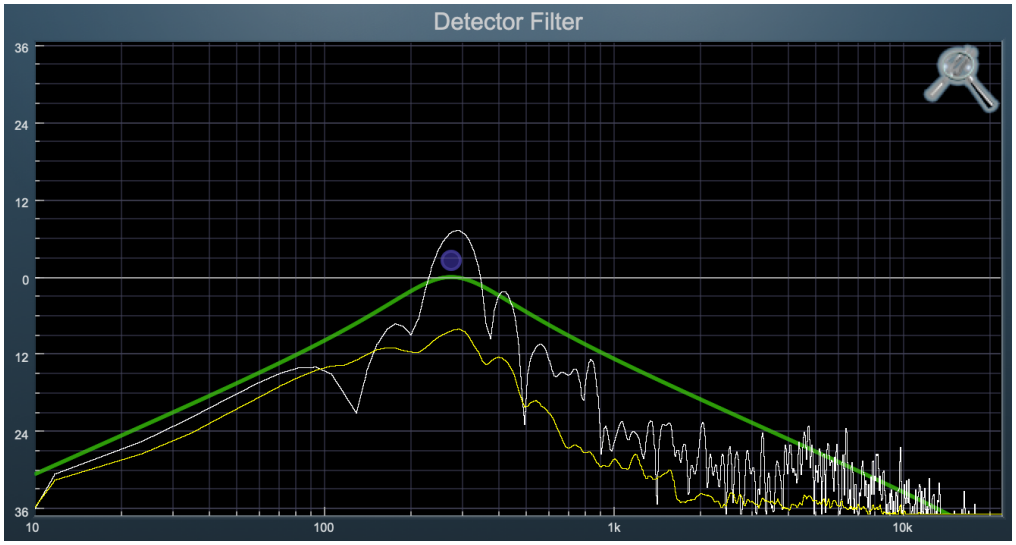
The values are:

- ±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:



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.



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.



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.



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.



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.



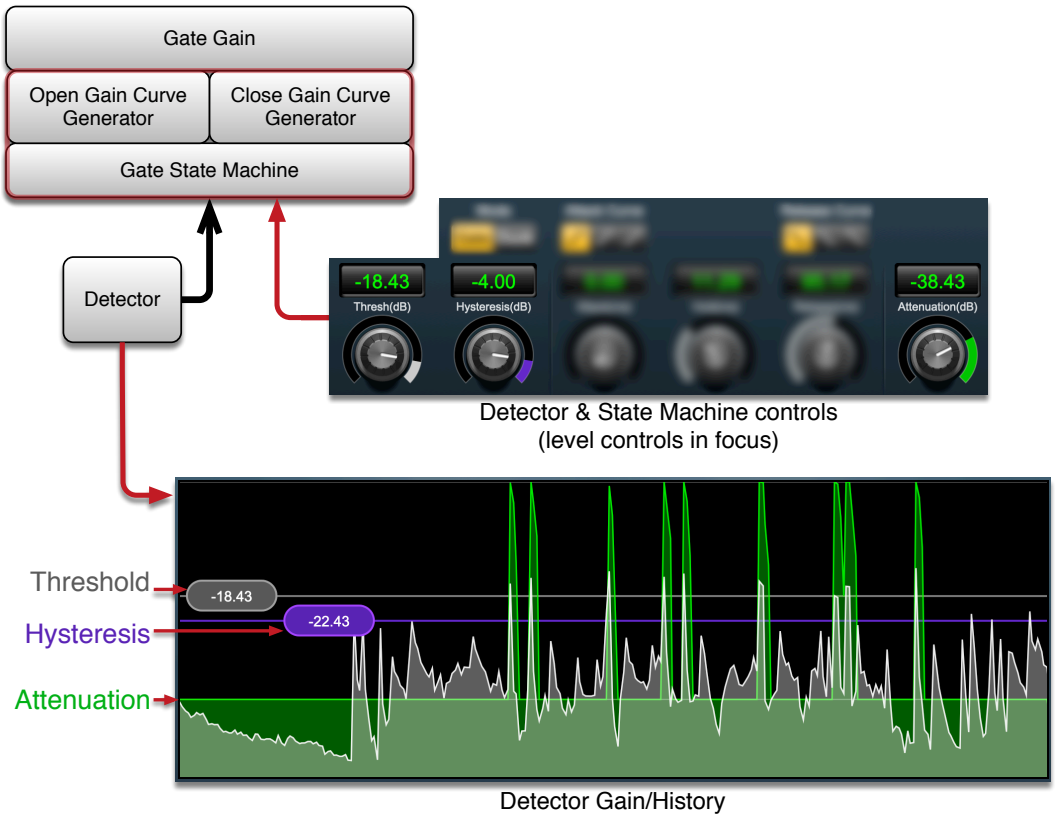
Low Cut

- **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 gain boost/cut control for the bandpass filter.



Bandpass

Gate State Machine: Levels controls



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"



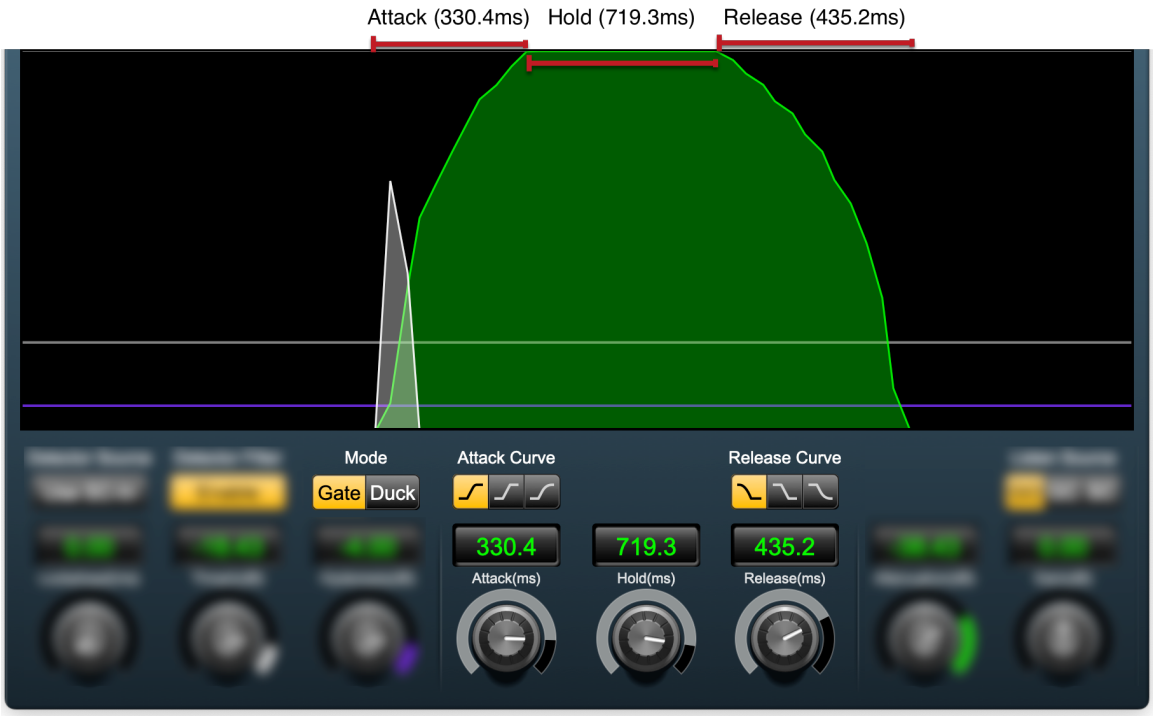
Threshold and Hysteresis controls
shown within the display range (left) and below the display range (right)

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.
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



Gate State Machine: Gain Curve shape, Attack, Hold and Release controls
(Attack, Hold & Release greatly extended for visibility)

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 ADSR programming or inline fade curves 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.

(Note the slow Attack in the graphic above shows the trick of triggering off one event in order to isolate quieter audio behind it.)

Hold(ms): Gain Curve Hold time

The *Hold* control determines how long the gate stays fully open at 0dB of attenuation.

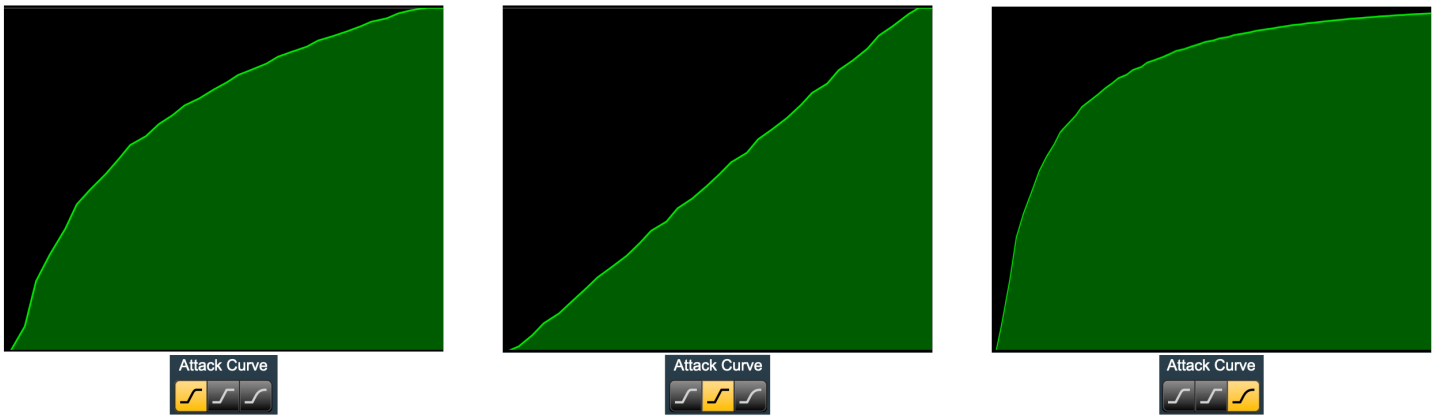
Gate 'Hold' duration may be from 0ms to 2000ms, starting from the moment the detector hits the Hysteresis de-trigger level.

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

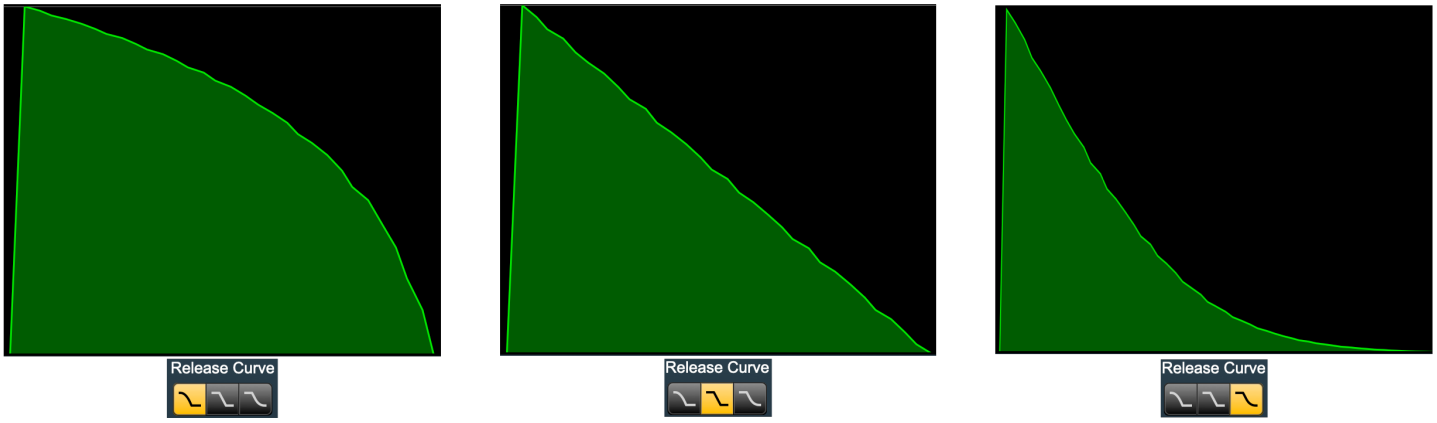


Attack Curve shapes

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



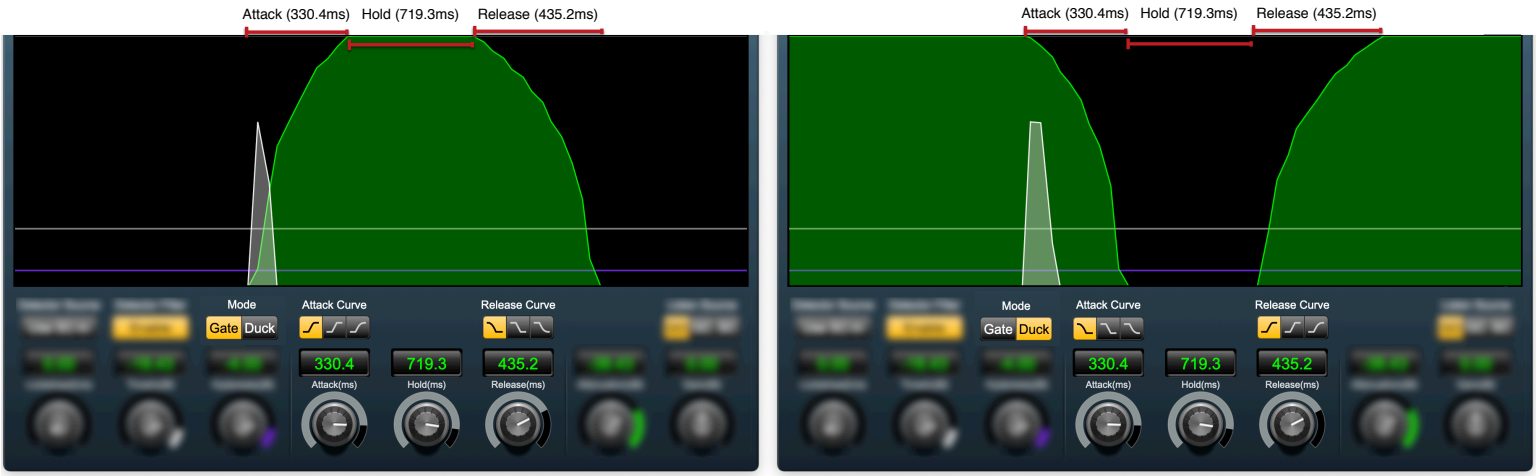
Release Curve shapes

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



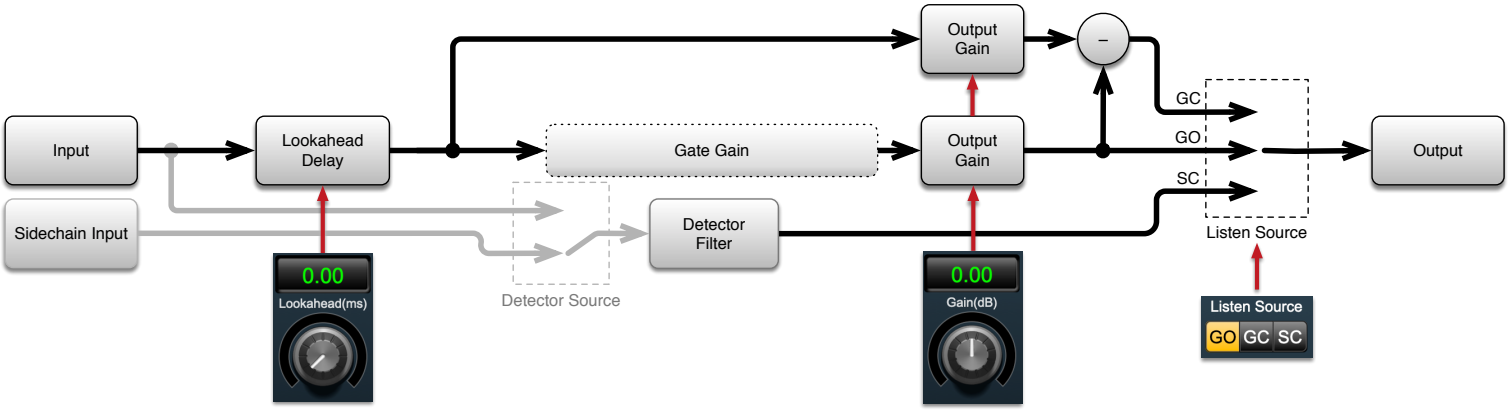
Gate vs. Duck Mode

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



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 **GO** = Gate Open), the signal being rejected by the gate (**GC** = Gate Closed), or the filtered detector sidechain input signal (**SC** = SideChain).

As you can see on the signal flowchart, there are two parallel output gain stages above and controlled by 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 **(GO)** *Gate Open* Listen Source feed.
- This Gate Open output 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, which is available at the Listen Source **(GC)** *Gate Closed* selector.

- The filtered detector sidechain audio can be monitored at the **(SC)** SideChain Listen Source selector. Auditioning the isolated sidechain feed is crucial when dialing in the Detector Filter. Oftentimes you may find a better trigger point for a given event at a different frequency than you expect. A real world example would be when a fat body snare has a deep reach into kick drum territory, and you end up finding a faster and more consistent trigger point at the higher frequencies of the kick beater.

It should be noted here that removing an instrument or event 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 to smooth the gate transitions, and 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.

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/VST3/AAX version for your DAW is avaliable as a free download from the Metric Halo website.



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.



Swept Knob

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 Mac ⌘ (Command) key or Windows **Control** key when you click, you will be able to adjust the value with finer precision. If you hold the Mac ⌥ (Option) key or Windows **Alt** 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**, Mac ⌘**.** (Command + .), Windows **Alt** key 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 Mac ⌘**.** (Command + .), Windows **Control.** (Control + .) 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



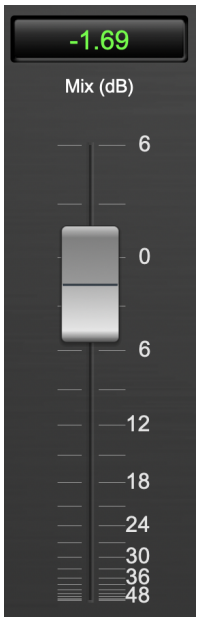
Toggle Button (Off)



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.

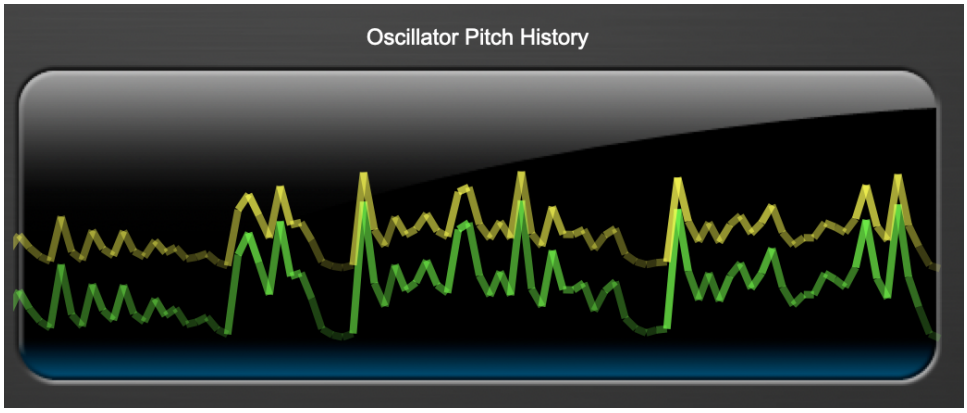
Fader



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.

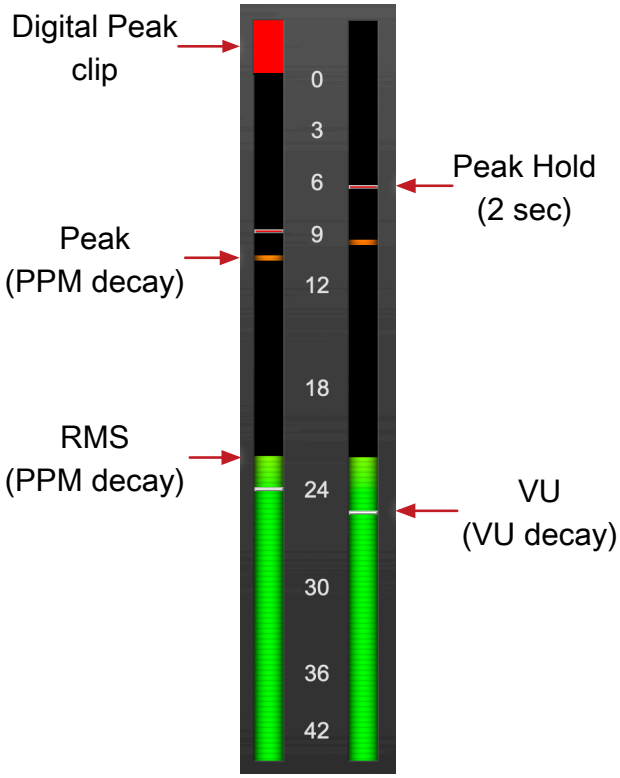
Oscillator Pitch History



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



Output Meter

For the main output stage of Thump we have provided meters driven with SpectraFoo™ metering technology. These meters show digital Peak, Peak Hold, RMS level and VU level.

Both the Peak and RMS level are represented with fast PPM ballistics.

The instantaneous Peak level is represented by the floating colored pip. Peak Hold is 2 seconds, indicated as a red floating marker pip.

The RMS level is the solid colored bar which transitions from green at "safe" high-headroom levels to yellow at -18dBFS, then to orange and finally red approaching -6dBFS.

The VU meter is the white floating marker pip showing IEEE standard 300 ms RMS average level.

When Thump is on a mono insert there will be a single meter.

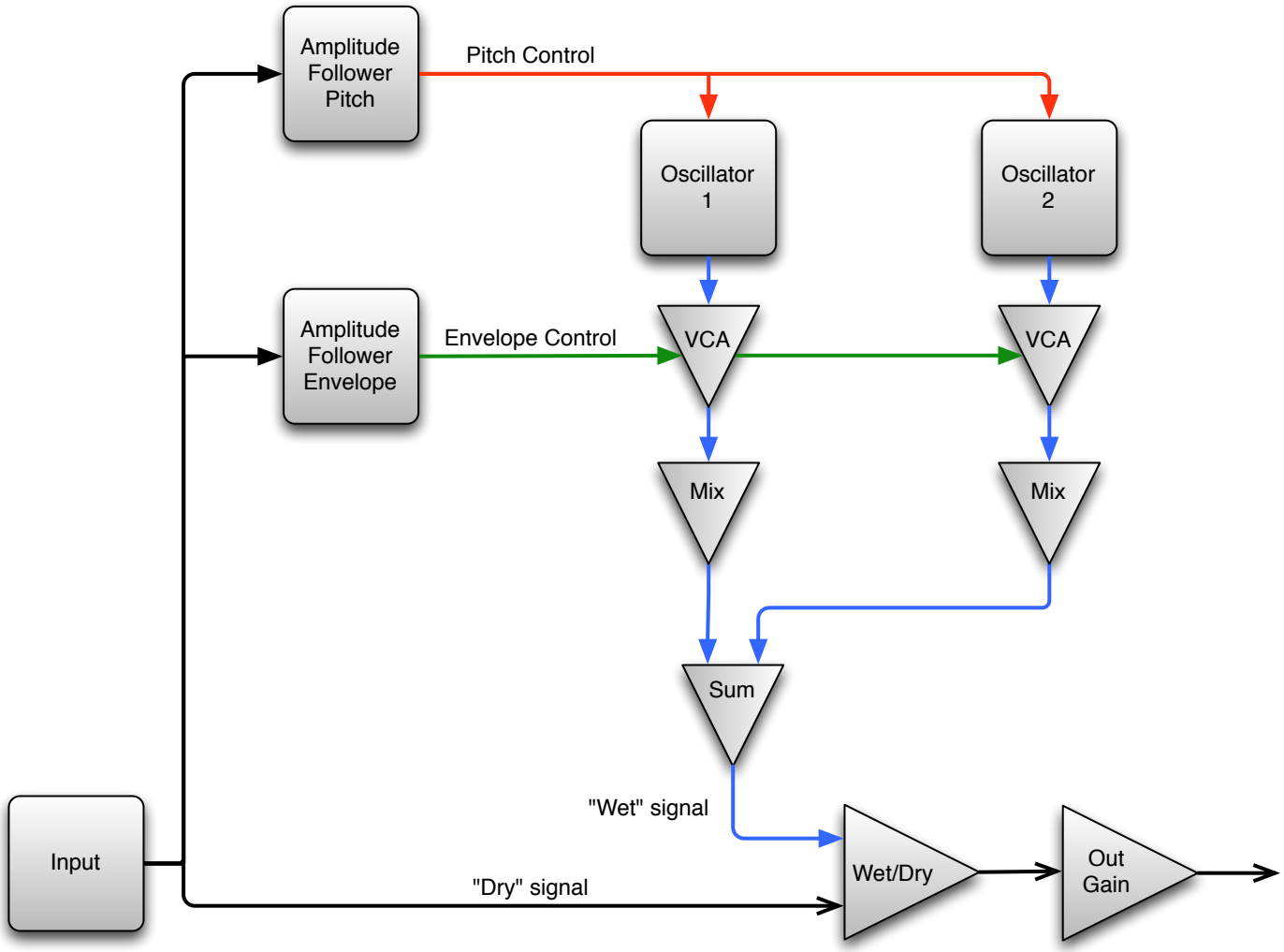
The output section clip lights activate if there is an over in the output stage or in any of the processing section input stages. It is reset by clicking on the meter.

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 section we'll discuss what each processing block does and how the controls work.



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 lets 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 it's 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 plug-in for digital audio workstations which provides a unique dynamic shaping tool.



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

Control Knobs are used to control the value of various continuous parameters of TransientControl:

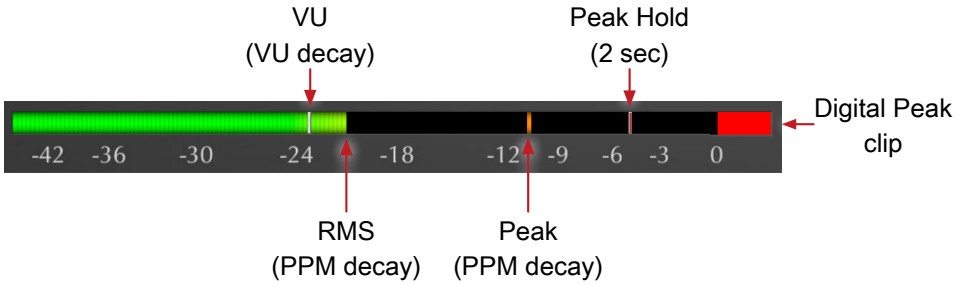
Control Knob



Examples of these types of parameters include: Transient, Sustain, Gain, etc. You can 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 Mac ⌘ (Command) key or Windows **Control** key when you click, you will be able to adjust the value with finer precision. If you hold the Mac ⌥ (Option) key or Windows **Alt** 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**, Mac ⌘**.** (Command + .), Windows **Alt** key 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 Mac ⌘**.** (Command + .), Windows **Control.** (Control + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.

Output Meter



The main output stage has SpectraFoo™ level metering showing digital Peak, Peak Hold, RMS level and VU level.

- Both the Peak and RMS level are represented with fast PPM ballistics.
- The instantaneous Peak level is represented by the floating colored pip. Peak Hold is 2 seconds, indicated as a red floating marker pip.
- The RMS level is the solid colored bar which transitions from green at “safe” high-headroom levels to yellow at -18dBFS, then to orange and finally red approaching -6dBFS.
- The VU meter is the white floating marker pip showing IEEE standard 300 ms RMS average 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.

When TransientControl is on a mono insert there will be a single meter.

UI Mode Button



UI Mode Button

The UI Mode button in the top plug-in header bar switches TransientControl's user interface mode. There are three modes available:



Basic: Provides the Minimum Controls and Metering

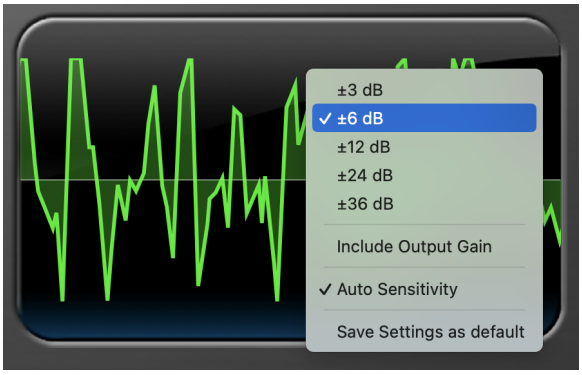


Basic with Process Metering: Adds Process Metering



Advanced with Process Metering: Adds Access to the Advanced Envelope Controls

Process Meter



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. The scale of this display can be adjusted by clicking the process meter. Selecting "Choose Output Gain" in this menu will let you configure whether the gain adjustment from the Gain control is factored into the meter. 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. Lastly, "Save Settings as default" allows you to save your metering preferences as default.

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

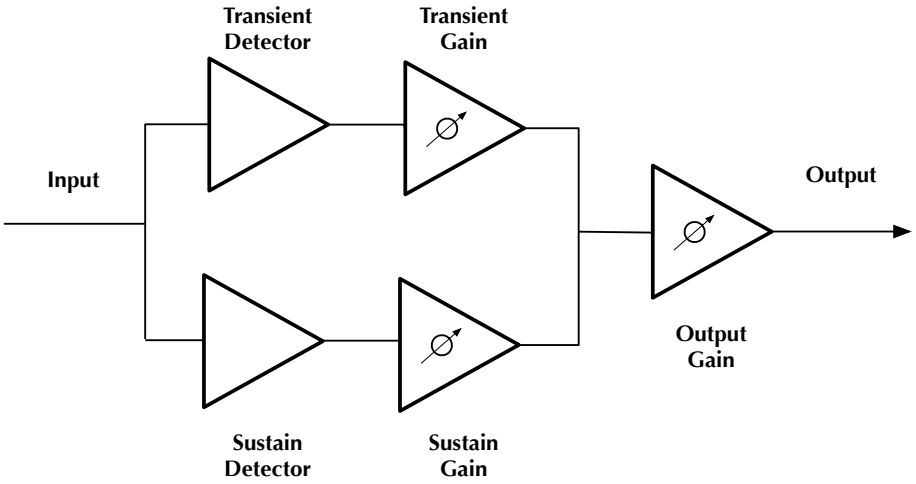
The Advanced UI mode gives you access to the controls used to fine tune the transient and sustain envelope detectors.

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.



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.

You generally will not need to change these controls from the default values.

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 and Friends

Special Projects and Collaborations

Metric Halo has collaborated with other companies for custom solutions (and just plain great ideas) behind the scenes many times over the decades, and that tradition continues...

Some of these friends and colleagues have graciously allowed us to include the results of our collaborations within the 3d DSP environment, and for that we are both honored and grateful.

Third party 3d processors can be found listed in the mixer Plug-in: *Production* menu category, and by each processors' functional subcategory (EQ, Dynamics, etc.).

This area will list third party processors as they are included in the 3d environment, with a brief introduction and links to their home pages for documentation and demo downloads of the AU/VST/VST3/AAX versions.

Make Believe Studios

Founded by producer/engineer Rick Carson, Make Believe Studios is a full service recording facility / plug-in development company based in Omaha, Nebraska, and is staffed by some of the most talented and knowledgable maniacs you'll find anywhere.

The Make Believe Bundle is a set of special tools curated from the production processes Rick has developed over his many successful years in the industry - fine-tuned and packaged into a set of beautifully straight-forward one-trick pony plug-ins and quite possibly the worlds most modern vintage EQ available.

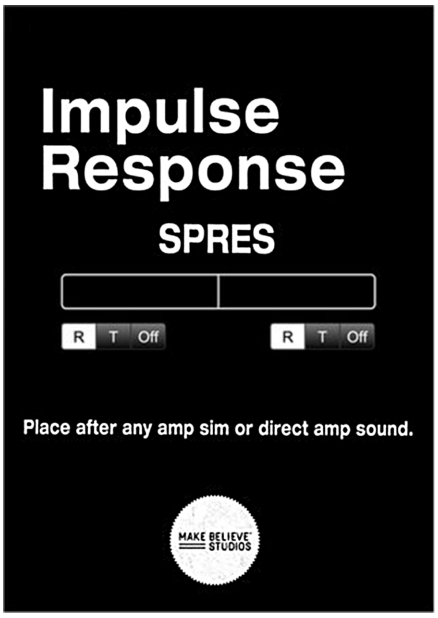
...And of course, the lovely Sontec MES-432D9D: The only Burgess Macneal certified and blessed digital re-creation of the legendary disc mastering equalizer.

Good Math



Over the years Rick found that using multiple gain stages operating in small increments sounded better (more open, less veiled) than one gain stage doing all the work. Is he insane? Yes, but that doesn't make him wrong. Even if you can't quite bring yourself to buy into the gain-staging concept, it's a uniquely sweet vintage British clipping stage when you click the golden A-hole.

SPRES



(to quote the MB web page) "SPRES (also known as Catharsis) has continued to live on. Not just in our hearts but on hundreds of records and a few different plug-ins. Now you can officially get your hands on the Impulse that changed heavy guitars forever. This plug-in also includes a new impulse designed by Ryan Harvey and Toni D'aquino to compliment the original in the best of ways! It also has some eq and some other shit going on. It's rad and makes guitars sound better."
Not to disagree, but it also does pretty cool things to vocals, snares, bass, brass sections... all sorts of things. Savory as opposed to sweet. Adds meat to the matter.

Kevin's Limiter



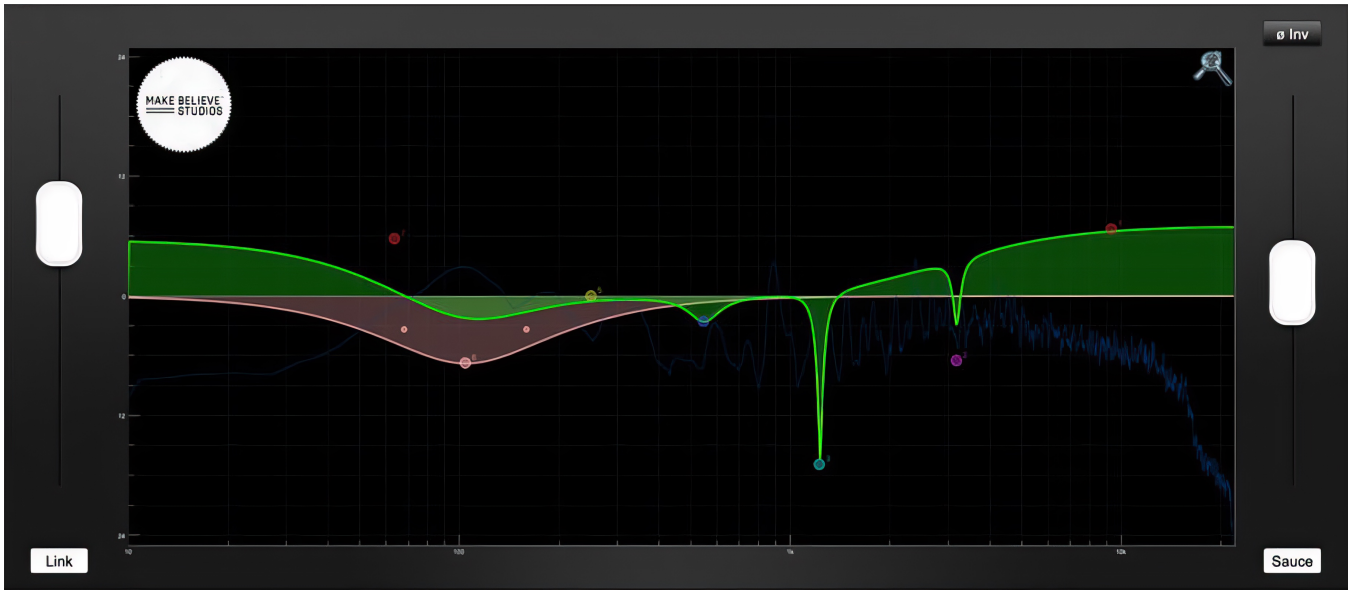
Just a great sounding limiter. Drop it in, job done. Applause all 'round.

The Parallel Processors



Rick Carson's secret parallel processor techniques, distilled and finely packaged.

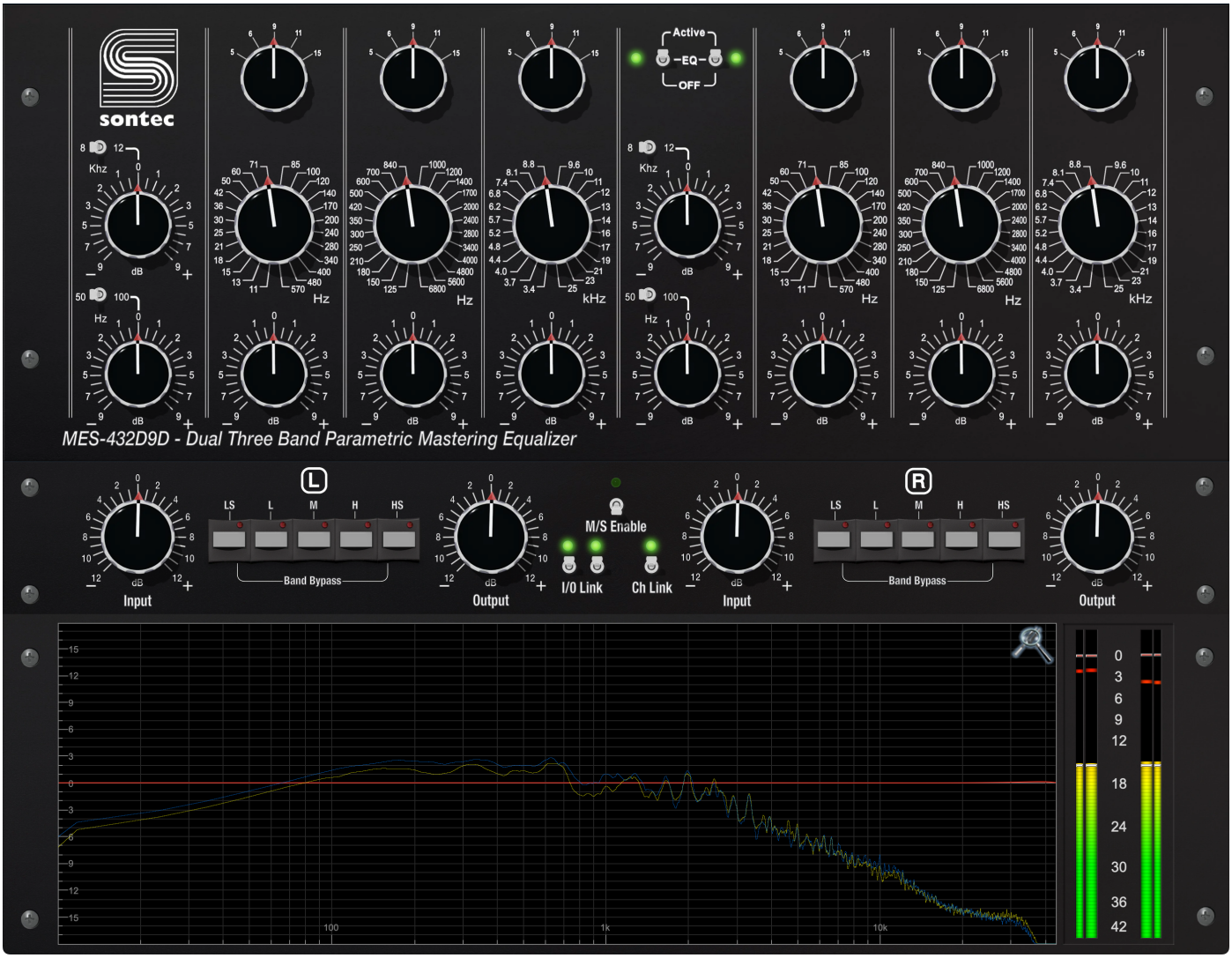
MBEQ



If MH ChannelStrip EQ were re-designed as a modern virtual instrument with the soul of a 1960's vintage BBC console.

Sontec MES-432D9D

Burgess Macneal himself provided the MES-432D9 on which this plug-in is based. Using Metric Halo's revolutionary new State Space Model Extraction Process and in concert with Mr. Macneal and Make Believe, this gorgeous equalizer has finally become available digitally in a way that is true to Burgess's vision.



The latest release of Sontec MES-432D9D adds Oversampling modes and an Options panel, providing some of the most requested features in a friendly and easy to use layout.

New features of the Options panel include Input and Output gain control, the ability to bypass bands individually, Mid/Side mode and I/O and Channel Link features that allow you to fine tune the equalizer to your particular workflow.

When in M/S mode, L/R input becomes mid/side respectively within the processor, and reverts back to Left/Right stereo at the output.

MixHead

Introduction from Rick Carson

For years and years I've been staring at photos of classic and modern sessions, always trying to learn what I can about my favorite producers, mixers, and engineers. I've probably spent hundreds of hours staring at photos, studying the racks and searching for unique pieces of gear. I eventually stumbled across a quote from Michael Brauer, about about a certain someone being 100% "in the box" except for a single piece of hardware, which led me to the discovery that I am proud to present to you. The MixHead.



There are few who wax nostalgic about the early days of digital recording, but in the transitional years of the late '90s and early '00s, when digital audio processing was still young, there were several hardware devices that crunched the ones and zeros themselves, saving the CPU for the complicated workloads required just to run the DAWs of the time. Some famous examples of these "digital in/digital out" devices include expensive equalizers and limiters that remain in use by professional engineers today. And, with modern computing power being vastly superior, many have been ported to plugins as well. The processor presented here was one of these devices, and it did something that at the time was quite unique: it aimed to add warmth and increased perceived loudness to digital recordings with the characteristic sound of analog tape saturation, but without the harshness associated with digital clipping.

This was done using a dedicated 1U rack unit with internal digital processing and AES inputs and outputs. This "plugin in a box" was designed to give digital recordings the glue that engineers felt they missed, without the hassle, wow and flutter, noise, and maintenance of actual tape machines or the sonic degradation associated in that era with yet another a/d-d/a conversion pass.

With the complex modeling available today and a market flooded with so many digital recreations, emulations, and mutations, looking back on the time of the original device provides an interesting retrospective: It was created with a very specific goal in mind, and this goal was achieved with a lot of theory (and assumptions) about what tape really did. Oddly, the results of all that effort created a processor that may well be even more effective and relevant now than it was in the years it was actually in production.

Make Believe Studios MixHead captures the lightning in a bottle that this unique box offered. Knowing what we do now, and with myriad tape plugins at our disposal, some of the design approaches taken with the original unit may seem backwards or counter-intuitive today, but they are approaches that still work, and are quite fun to experiment with. We even added a few requested features like tenth of a decibel HF Adjust and a new 3.75ips lo-fi mode.

In practice, you may find that the 30ips setting sounds like it has more distortion than the 15ips setting. That's fine. You may hear some interesting phase interplay at in the higher frequencies that you wouldn't normally associate with tape. That's also fine. You will also find that the input and drive controls are highly interactive, and that the HF Adjust is great for adjusting the top end in a way that sounds natural.

MixHead wasn't created to do what your other tape plugins do. It resurrects and modernizes an outboard hardware DSP box previously relegated to the dreaded digital graveyard.

MixHead is an essential plugin that can be used in any stage of the recording process, with an efficiency in the code that allows multiple instances without worrying about overtaxing your CPU.

It was created to sound good, and now, it sounds fantastic.

Plug-In Header Bars

All Metric Halo and Make Believe family plug-ins display the MH header bar at the top of the plug-in window. This header bar allows you to organize and access all your presets across all supported plug-in formats on Mac, Windows and Metric Halo hardware DSP via MIOConsole3d.

It is especially useful in that, regardless of platform, it provides a straightforward, powerful and consistent processing workflow wherever you might be working.

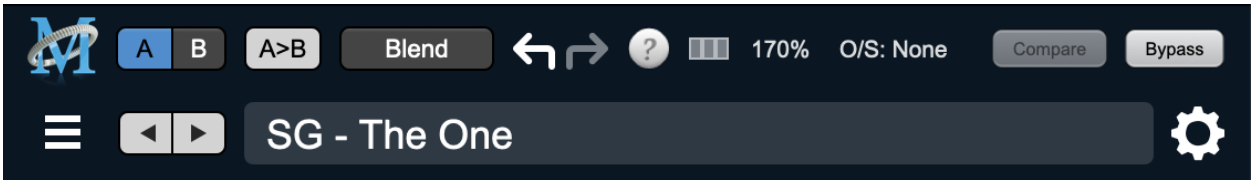
With the constant evolution of computer capabilities opening the door to new production techniques and music delivery formats, the differences between DAW software workflows have become ever more diverse. Many of the major DAWs provide their own plug-in headers within every plug-in instance window, providing their own feature set catering specifically to their internal workflow.

Conversely, other equally popular DAWs provide no added feature support for plug-ins (such as plug-in parameter Undo/Redo), opting instead to insert plugs as a straight processing block.

The plug-in header bar bridges that gap by offering the most asked for plug-in functions in a simple GUI, making all of our plug-ins functions and their presets available to every user on every platform.

The plug-in header bar has two rows of controls, with the Metric Halo logo icon at the top row left.

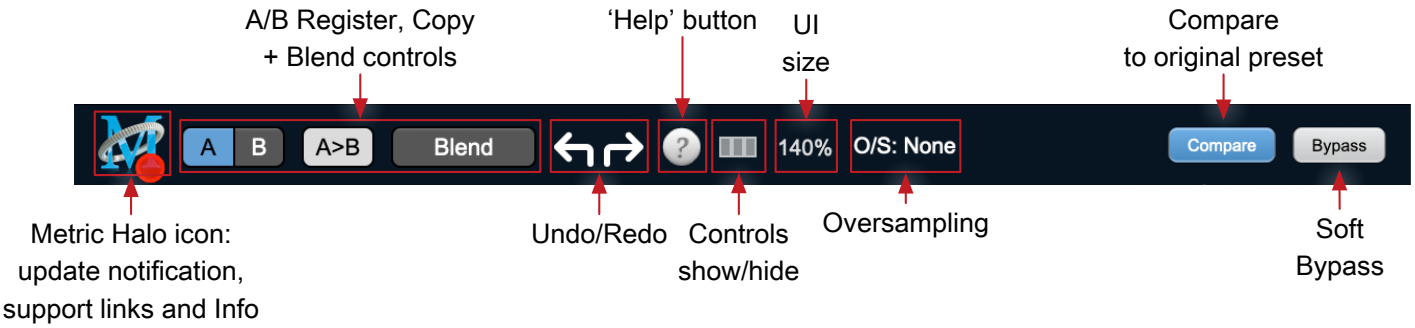
The top row handles plug-in operations, including update notification and download, access to MH online resources, GUI preferences, tooltip help, A/B parameter snapshots, snapshot Blend, plug-in Undo/Redo, Oversampling modes, Compare and soft Bypass.



MixHead Plug-in header

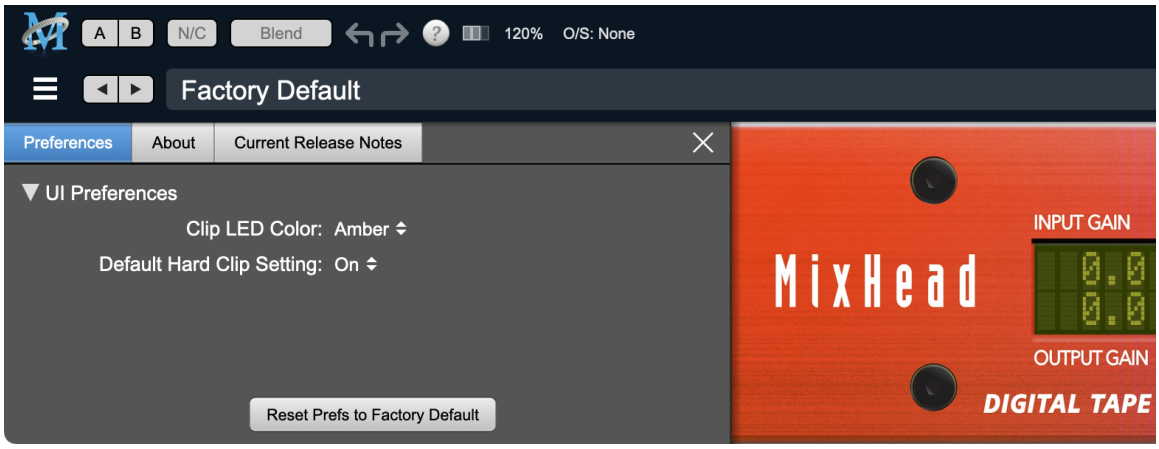
The lower row (with the ‘hamburger’ menu icon at the left) is all about preset management.

Plug-In Header: Top Row

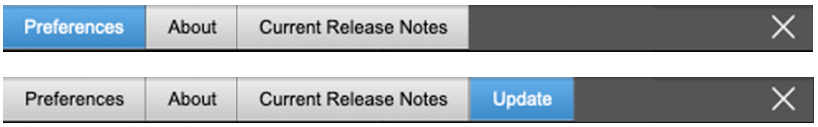


Metric Halo Header Icon

Clicking the MH icon expands the entire plug-in window to the right and opens a multi-function control sidebar with the Preferences tab in focus:



This sidebar has context-sensitive tabs across the top for GUI **Preferences** (shown above), **About**, and **Current Release Notes**.

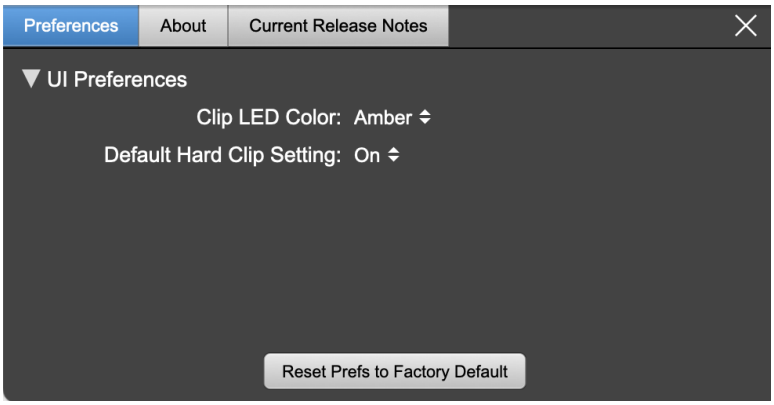


Plug-in control pane tabs

When updates are available for download from Metric Halo, the MH icon will sport a blatant red dot (shown in the header map at the top of this page) and an **Update** tab is added to the sidebar. These tabs are dynamic by design, and additional tabs may appear as new content becomes available.

To close the sidebar, click the MH icon again, or the "**X**" at the right edge of the tab bar.

Preferences tab



MixHead Preferences

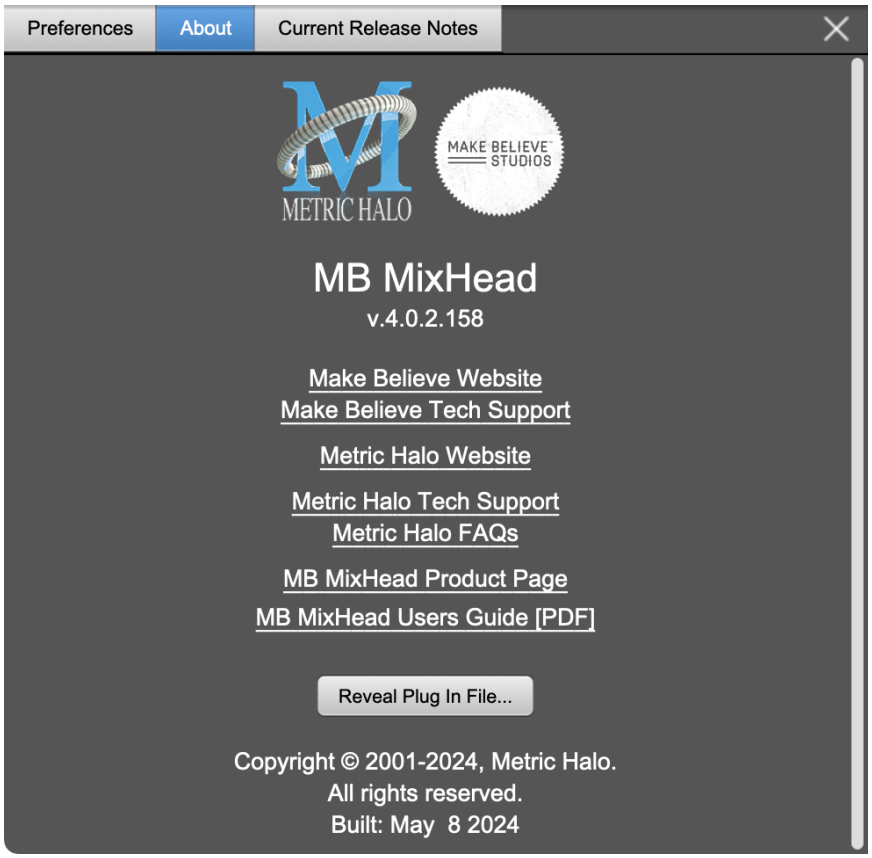
Note: UI preferences will be saved to the computer and automatically applied to the next instance of MixHead of whatever native format, even on another DAW or MIOConsole3d.

You can always revert to the factory default settings by clicking the **Reset Prefs to Factory Default** button at the bottom of the Preferences pane.

- **Clip LED Color**: set to *Amber* as the default (like the original hardware), this control allows you to set the topmost LED in the meter stacks to show *Red* when the signal clips.
- **Default Hard Clip Setting**: set to *On* to match the behavior of the original hardware device which used 24-bit AES interfacing and as such had to clip outputs to 0dBFS. Disabling Hard Clip in the Advanced Parameters UI allows processing that results in output levels over 0dBFS to pass on to the next stage in the signal path in floating point mixers.

  To disable Hard Clip on all new instances of Mixhead opened on your host computer, set Default Hard Clip Setting to *Off*.

About tab



About tab

The **About** tab reveals the current plug-in version information and provides convenient web links to product info, support pages and the current manual on the Metric Halo main website.

Clicking the "MixHead Users Guide [PDF]" link will open and display the latest Mixhead manual in your default web browser.

Reveal Plug-In File... will open the folder containing the current plug-in file, with the plug-in file itself selected. Very handy for troubleshooting on the fly.

Update notification tab



Update tab (only appears when an update is available)

The **Update** tab will contain a link to download the new installer package in the header at the top of the pane.

Below the download link header will be release notes detailing the major changes included in the update, with bugfix revisions for the most recent software releases listed further below. Windows users will see a link to view the release notes using your current default web browser.

Click the Installer link to download, unzip and run the installer manually, preferably when your audio software is inactive so it can properly scan the new versions at launch.

Plug-in Snapshot Registers: A/B



Plug-in Header: Snapshot Registers: A/B

The A and B buttons control the A and B state registers. The A/B registers are used to store modified parameter snapshots in addition to the original saved preset called up by **Compare**.

The Blend function can be used to smoothly morph between the parameters set in the A and B registers, and Blend is a mappable parameter so it can be operated with external MIDI control. Details of the A/B Snapshot Blend feature follow on the next page.

For each of the A and B buttons the visual display tells you the state of the register:

- Light Grey means the register is empty
- Dark Grey means it has settings, but is unselected
- Blue means it has settings and is selected

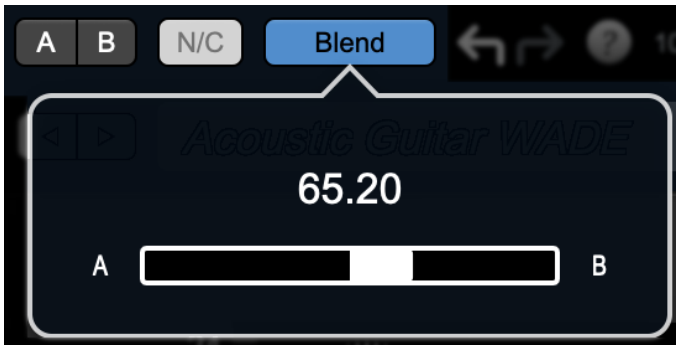
You can perform the following actions:

- Clicking on an empty register takes a snapshot of the current settings and saves them to the register.
- Clicking on an unselected filled register copies the register parameters to the current active plug-in settings.
- Clicking on a **selected** register toggles to the other register; this lets you toggle between the register settings without having to move the mouse.
- The Copy button will show **N/C** and remain inactive until one or both of the registers is in use. The Copy button will alternate between **A>B** (copy A to B) or **B>A** (copy B to A) depending on which register is selected. Clicking the Copy button then copies the settings from the selected register to the target register, overwriting the prior contents (if any).
- <Option> - clicking on a register snapshots the current settings and saves them to the register, overwriting the prior contents (if any).
- Changing settings when a register is selected will update the settings in the register to reflect the change.

Snapshot Blend

The Blend button allows you to interpolate (or morph) between the parameter snapshots stored in the A and B registers. It becomes active when both A and B have a parameter set stored.

Blend is a MIDI-mappable parameter so it can be operated in realtime with external MIDI control and/or automated in the DAW. This allows you to automate a transition from the A → B register, the B → A register or any setting between the two.



Plug-in Header: Snapshot Blend

The Blend button's visual display tells you the state of the register:

- Light Grey means it is empty
- Dark Grey means it has settings, but is unselected
- Blue means it has settings and is selected

Click on Blend to popup the blend control. Slide all the way to the left to apply the settings in the A register. Slide all the way to the right to apply the settings in the B register. Intermediate settings for blend will give you intermediate settings for any parameter that is different in register A and B. The blend control does not change the state of Bypass.

Note that the Blend is not a parallel processing mode where two instances of the processor are running the A and B settings and the output is a parallel blend of the two settings. Rather, Blend interpolates the parameter settings of the two registers to one instance of the processor. You can see the parameter controls move between A and B settings as you slide the Blend control.

The A/B and Blend settings are stored and recalled as part of the plug-in state, but are not saved as individual presets in the preset bar.

While you can use the blend with arbitrary A and B settings we find it works best when you craft the settings in the two registers in such a way as they are related to each other. Specifically, if an indexed (stepped) parameter is different between the two settings, the interpolated value will snap to one of the indexes between the two settings, which can be jarring.

It is best if the parameters that you blend are smooth parameters (e.g. gains, frequencies) and make sure the indexed parameters (enables, modes, band types) are set the same for both registers.

The easiest way to do this is to load the same setting into both registers and then tweak the settings of one of the registers.

This works especially well if you make one of the registers be the basic settings with all the gains or thresholds flattened out so that you can smoothly interpolate between a setting and effectively bypassed - we have found that this allows you to zero in a perfect configuration between too much and too little.

Plug-in Undo/Redo

All the plug-ins provide support for undo/redo from the plug-in header bar.



Plug-in Header: Undo/Redo

The left and right curved arrows represent Undo (Left) and Redo (Right). These arrows are grey when there is nothing to Undo or Redo.

The arrows are white when it is possible to Undo (Left) or Redo (Right). Clicking the left arrow when it is white will undo the last action you made in the plug-in. When you undo something that change is placed on the redo stack, and the Redo button will turn white.

Clicking the Redo button (when it is white) will restore the state that the last Undo changed.

If the Redo button is white and you make a change in the plug-in, the Redo button will go grey as the redo buffer will be cleared.

Help Button



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 Elements Selector



This button toggles visibility of various UI graphical elements. This button allows you to maximize screen real-estate. Clicking the button will step through the available views, with elements shown darker in the button graphic.



Right-Click selection menu

Right-clicking this button will open a menu to select your desired view directly.

- Full UI: All elements
- Mini UI: LCD panel, Active bypass switch and Meters
- Micro UI: LCD Panel and Active bypass only

UI Size Selector



UI Size Selector

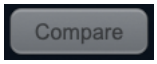
This pull-down menu lets you set the plug-in UI size to taste.

The size is remembered and applied the next time you insert a MixHead plug-in.

Oversampling Selector

2x and 4x oversampling is available for for MixHead in MIO DSP. The default Oversampling setting is **None**.

Compare Button



Compare Button (inactive)



Compare Button (active)

To use the compare button, a preset must first be loaded. The compare button will be lit up when the current settings differ from the selected preset. If you click this button while it is lit, the preset settings will be restored, but you can still return to the changes you made by clicking on the button again. It is important to note that any changes you make to activate the compare light are always for comparison to the last loaded preset.

Soft Bypass



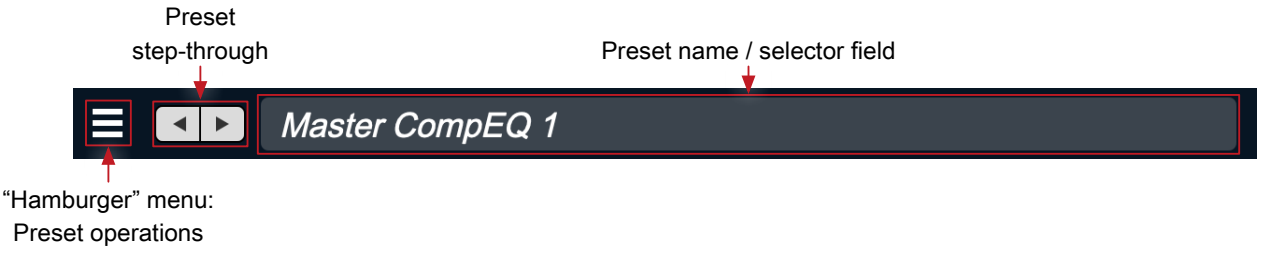
Soft Bypass Button (not bypassed - plug-in is processing)



Soft Bypass Button (bypassed - plug-in is *not* processing)

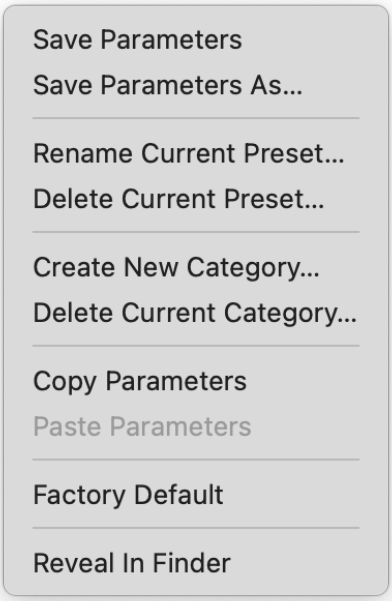
When glowing yellow, this button will maintain the time delay through the channel and will continue to show metering, but will cleanly disable the processing.

Plug-In Header: Preset Row



Plug-in Hamburger menu

The preset and parameter functions within the hamburger menu break down as follows:



- **Save Parameters** saves the current plug-in parameters to the current preset.
- **Save Parameters As...** opens a dialog box where you can name and choose a category to save 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 paste 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.

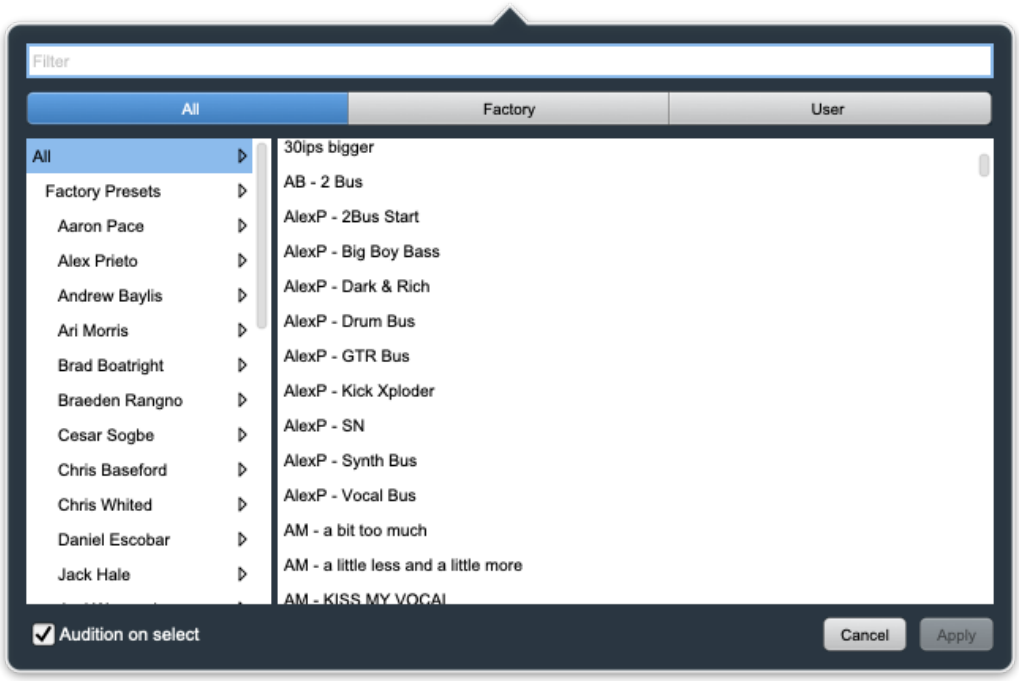
Preset Step-Through Buttons



Preset Step-Through Buttons

These buttons step through Factory and User Presets in succession, as they are listed in the Preset Selector window. The left arrow chooses the previous preset. The right arrow chooses the next preset.

Preset Name/selector menu:



Plug-in Header: Preset selector menu (ChannelStrip shown)

The Preset selector will open to show all the available preset categories, and the presets within those categories.



Preset selector menu: Audition on select

With "**Audition on select**" enabled at the bottom of the window, selecting a preset will temporarily load those parameters so you can hear the effect on the audio you are playing, without actually committing to the preset.

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.

Advanced Parameters



Advanced Parameters button

The gear icon at the far right of the Presets bar opens a window with Advanced Parameter settings for MixHead.

Operation

The MixHead user interface is straightforward and intuitive with an active LCD-style display, four control knobs, three switches and three sets of stereo meters.



A brief overview of the controls:

- The LCD display at the left shows the current setting of all MixHead parameters.

  The individual LCD parameter displays can also be used for parameter entry.

- The four knobs are standard rotary encoders for parameter entry.

- The **Tape Speed** switch toggles through the tape speed saturation algorithms.

- The **Active** switch enables MixHead processing: when lit, MixHead processing is active, when dark processing is bypassed.

  The plug-in header bar Bypass button reflects the state of the Active switch, and vice versa.

Please Note: The *Power* button at the far right and the *Presets* segment of the LCD panel are currently inactive. They are placeholder controls for features to be added in a future release.

Tooltip mouse-over pop-ups are available for all UI elements by clicking the circular **?** Help icon in the plug-in header bar.

The individual parameters and key commands for each control are detailed in the following pages. All key commands and gestures operate equally for both the parameter knobs and switches and their corresponding LCD readout controls.

UI display options

The MixHead UI operates in three modes, selected by toggling the disclosure icon in the top row of the header bar.

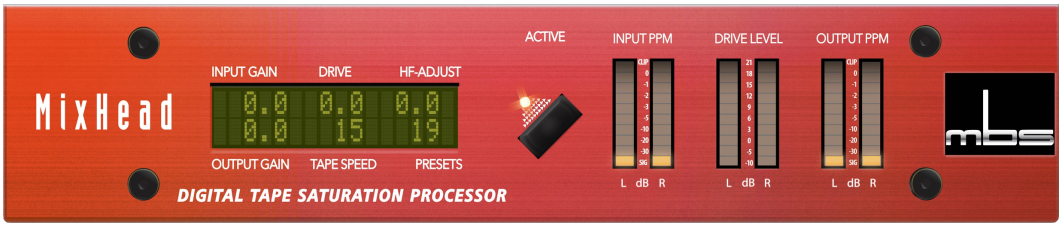


Right-clicking the disclosure icon reveals a menu allowing direct selection of the UI display mode:

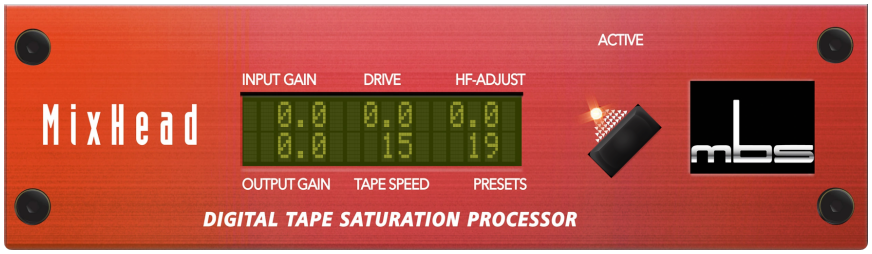




Full UI (All control and display elements shown)



Mini UI (LCD control/display, Active bypass and Meters shown)



Micro UI (LCD control/display and Active bypass only)

Advanced Parameters UI



The **Advanced Parameters** button to the right of the Presets bar replaces the MBS logo plate with the Hard Clip Disable control button.

Hard Clip Disable



Allows you to disable the 0dBFS hard clip. This hard clip was part of the original device. You may disable it to allow > 0dBFS signals to be passed to other processors in the chain. The 0dBFS clipper is active when this button is illuminated.

Hard Clip is enabled (button is lit) by default to match the behavior of the original hardware device which used 24-bit AES interfacing and as such had to clip outputs to 0dBFS. Disabling hard clipping allows processing that results in output levels over 0dBFS to pass on to the next stage in the signal path in floating point mixers.
Note that this setting is per-instance and will not automatically persist to new instances of MixHead. To disable Hard Clip as the default for all new instances, go to the UI Preferences Tab by clicking the Metric Halo header icon, and set **Default Hard Clip Setting** to *Off*.

Primary Controls

There a five primary parameters to control the MixHead processor.

Input Gain

Input Gain adjusts the pre-process input level, not unlike the line in calibration on a tape deck. Input Gain adjusts input level in 0.1dB steps, with values spanning from -12.0dB to +12.0dB.



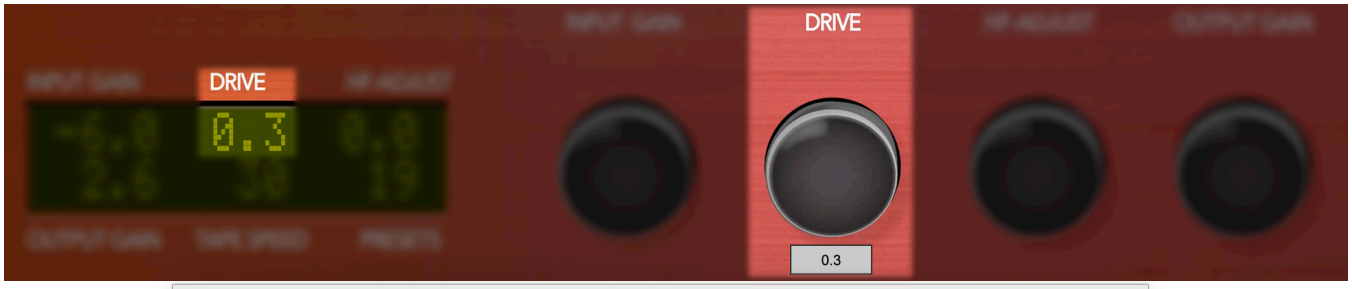
Input Gain (dB) Knob. Click-Drag to change. Control-Shift-drag to apply inverse change to Output Gain. Alt-click to reset. Control-Alt-click to set to minimum. Control-click (or Right Click) for text entry.

Input Gain interacts greatly with the Drive stage and Output level.

Tip: **Control-Shift-drag** Input Gain to inverse link with the Output Gain control, to maintain your current gain structure through the processor while shifting the reference point at which the Drive circuit operates.

Drive

Drive adjusts the amount of harmonic distortion and saturation effect. The saturation effect is non-linear with regard to gain, and often requires some degree of Output level compensation.



Drive Gain (dB) Knob. Click-Drag to change. Alt-click to reset. Control-Alt-click to set to minimum. Control-click (or Right Click) for text entry.

The Drive parameter has a range of -7 to +14. The default Drive setting of 0.0 is intended to emulate a record level of +7dB to tape, so the saturation effect is already fairly well engaged at 0.0 Drive. More subtle tape emulation characteristics with higher headroom and less dynamic saturation are available at Drive levels below 0, where the Input and Output levels tend to have more relative effect.

Tip: High Drive settings can result in serious signal breakup, especially with modern day deep low frequency program. That said, MixHead works brilliantly as a high-passed parallel processor even on deeply bass-heavy master tracks, bringing massive saturation crunch to the mids and a silky sheen to the highs without stepping on the bass foundation.

HF-Adjust

HF-Adjust allows you to control the amount of high frequency cut or boost independent of the Drive setting. The control range is -6 (max. cut) to +6 (max boost).



High-Frequency Adjust Gain (dB) Knob. Click-Drag to change. Alt-click to reset. Control-Alt-click to set to minimum. Control-click (or Right Click) for text entry.

With real tape, the higher the saturation the more high frequency damping takes place. MixHead lets you break that rule by running high frequency energy levels contrary to what you would otherwise expect from tape. The effect can be used to simulate differences in tape formulations or playback electronics or different bias/EQ setups.

Tip: Unlike pretty much anything with a real tape deck, HF-Adjust is automatable. It is especially well suited to pushing instruments back in the mix or bringing them dramatically forward as the song demands. At low saturation levels a bit of HF-Adjust boost acts very much like a vintage exciter.

Output

Output Gain adjusts output level post-processing. Output Gain adjusts output level in 0.1dB steps, with values spanning from -12.0dB to +12.0dB.

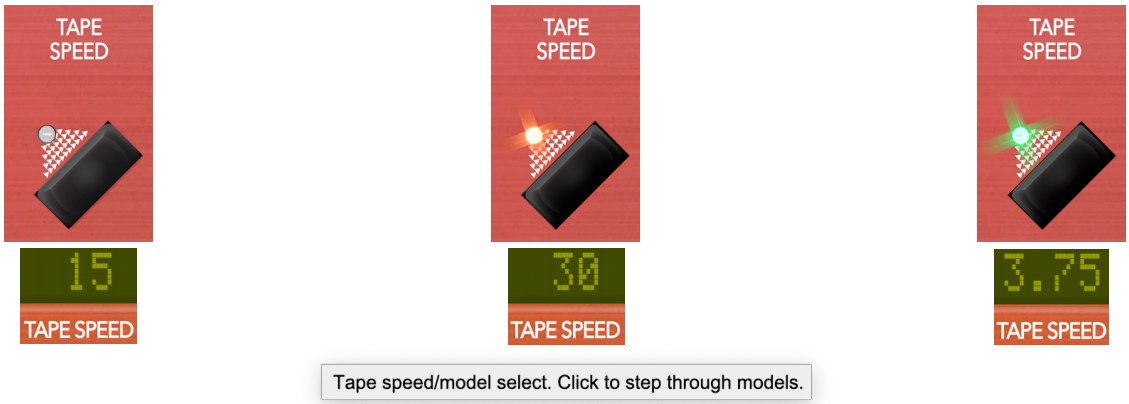


Output Gain (dB) Knob. Click-Drag to change. Control-Shift-drag to apply inverse change to Input Gain. Alt-click to reset. Control-Alt-click to set to minimum. Control-click (or Right Click) for text entry.

Generally Output Gain is used as a final compensation for the changes in perceived loudness from the Drive stage saturation effect, and to ensure that the output of MixHead does not clip the input of the next process in the signal chain.

Tape Speed

Tape Speed toggles between MixHeads three tape speed emulation algorithms.



- **15ips** (the default): emulates the general distortion and saturation characteristics associated with tape recording at 15ips, with an unexpected subtle stereo widening effect.

  In 15ips mode, the Tape Speed switch light is dark.

- **30ips**: emulates the generally lower distortion and higher saturation headroom curve associated with tape recording at 30ips.

  Note that the 40Hz to 70Hz bass reduction induced when recording to analog tape at 30ips is not reproduced.

  In 30ips mode, the Tape Speed switch light is lit amber.

- **3.75ips**: A new addition to MixHead not available on the original hardware, 3.75 was inspired by the modeling of a 1950's Webcor tape machine. The main characteristics that were borrowed for the 3.75 machine are its headbump curve and the amount of distortion it can generate. In true MixHead fashion we utilized the Mixheads' unique approach to compression, limiting and hf adjust as the other models.

  In 3.75ips mode, the Tape Speed switch light is lit green.

Meters

There are three sets of stereo meters, displaying signal levels after the Input and Output gain stages respectively, and the "virtual tape machine record level" of the Drive stage.

Note: in mono plug-in instances, both channels of the meter pairs will show signal activity.



- The **Input PPM** meters show the peak level after the Input Gain control. They are PPM response from -30dBFS to 0dBFS, with a 1 second peak hold.
- The **Drive Level** meter indicates the amount of harmonic saturation being applied at the Drive stage - basically how hard it's working. The segments display a range from -10db to +21dB.
- The **Output PPM** meters show the peak level after the Output Gain control. They are PPM response from -30dBFS to 0dBFS with a 1 second peak hold.

Important Note
The topmost segment of each meter set indicates consecutive samples into digital clipping. By default, it does not change color as a warning (matching the behavior of the original hardware).

To set the topmost LED to flash *Red* at clipping as the default for all new instances of MixHead, go to the UI Preferences Tab by clicking the Metric Halo header icon, and set **Clip LED Color** to *Red*.

The second segment from the top indicates 0dBFS, so for safety in 24bit integer operation the top two segments of the Output PPM meter should stay dark. Note, however, that the hard clip behavior of the original hardware can be disabled in MixHeads Advanced Parameters UI, allowing greater-than-0dBFS data to pass unaltered to the next stage in the signal chain.

On both the Input and Output meters, the bottom segment on the hardware to indicate a clock-synced and legal AES audio input stream.

MBSI

Introduction

I want to tell you about one of the best parties I ever attended in Los Angeles.

Sometime, pre-pandemic, I had moved in for a couple of months, with a guy named Nolan Neal, and we became close friends. Nolan was famous for being a contestant on America's Got Talent and The Voice, where he performed an Elton John song that people loved. We remained in touch after I moved out, and one day he invited me to a party at his house. After a couple of hours, Nolan got up on a chair and announced to the whole room that he was moving to Nashville, and leaving Los Angeles. Among the boos of the crowd, someone asked him when he was leaving. "Tonight", he said, "and you're all going to help me move." Outside of his home, there were a couple of portable storage pods, and he asked that everyone grab something on their way out. Nothing was even packed, so as people left the party, they did as he asked, tossing everything into the modular storage units.

As this was occurring around me, Nolan ran up, tapped me on the shoulder, and said "Rick, this is Michael. Michael, this is Rick. You guys should know each other." Holding out my hand, I said "Rick Carson, nice to meet you", to which he replied, in turn, "Michael Szymczyk, nice to meet you." I looked at him and said "Szymczyk… that's interesting. Are you related to a guy named Bill?"

"Yeah, my dad's name is Bill."

"No shit… did your dad make Hotel California?"

His eyes got really big, and he looked at me, excitedly telling me that in his years in Los Angeles, no one had ever made that connection. Yes, his father did indeed make Hotel California.

We spent the night talking about Bill, and the work he had done. Being an engineer from Michigan, like myself, his father had always been a huge inspiration. We talked, at length, about Bill's moves to New York his work with B.B. King, Los Angeles, to the Rocky Mountain Way of Colorado to work with Joe Walsh, and, ultimately, his time in Miami working with The Eagles.

It was one hell of a party; Michael and I exchanged numbers, and remained friends after that meeting. Sadly, Nolan passed away in 2022, prompting a phone call with Michael to catch up and see how the other was doing, as it had been a while since we spoke. As we were ending the call, I casually mentioned that we were now making software, and that if there was anything his dad encountered in his career that he'd like to use again, "in the box", to let me know.

About a half hour later, Michael texted me a photo of a console, telling me that it was the console that Bill had used to make Hotel California for roughly the better part of a year during the lockout session he had at Criteria Recording Studios in Miami. Surprisingly, after that stint at Criteria, Bill never saw the console again.

My journey had begun, leading me to the wonderful Lij Shaw in Nashville, who owned the console. I've made so many wonderful friends on my journey to bring this console to life, from Michael, Bill, Lij and Mark and I am incredibly excited for people to hear it.

The console was the third custom console originally built for Criteria, and was quite unique in its design. While this one of a kind console might seem very primitive now, at the time it was incredibly progressive, with hot-swappable inductor based EQ modules, a dynamics section on every tape return, and touch-sensitive routing to the multi-track outputs (Bill would use these extensively in his pioneering of multi-track comping). We can now see that many of the features on this console would evolve into the MCI JH-416, and would inspire many more designs over the decades. Completely discrete, with Pantion resistor ladder faders (rather than Penny & Giles faders, or the API faders used on early 416s), this new MCI would start its life as a 16 track console in a 16 track room. While there were 20 input modules (the same input modules that exist on the desk today), the return side had only 16 modules. To accommodate Bill for his lockout of the room at Criteria, however, he required a 24 track facility, and the desk was subsequently modified to meet his requirements. Photos can even be seen of the desk before and after this modification.

The console lived its life in Studio C for over a decade, where it was utilized on many hit records of the time: records by the Bee Gees, Eric Clapton, Grand Funk Railroad, and Elvin Bishop (Fooled Around And Fell In Love is another wonderful Bill Szymczyk Production). Producers and engineers loved the studio and it seems this desk played a big part in that!! I can understand the appeal, The console is incredibly musical and forces you to make decisions you may not come to without the pushing the technical limits of the time. It is no surprise to me that this console left a lasting impression on Bill and all the wonderful music created with it!!

Thank you so much Bill for all of the help bringing this one of a kind piece of history into the box for everyone to enjoy!!!

Rick Carson

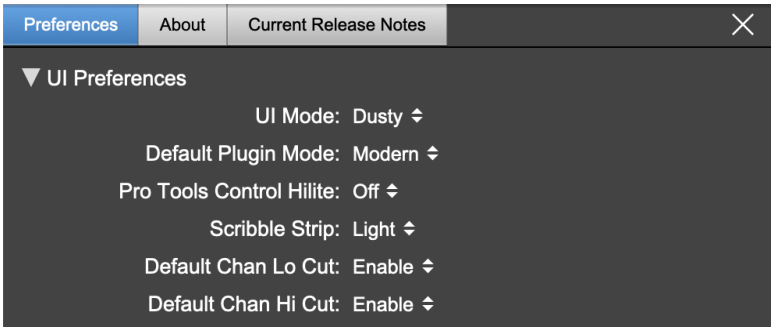




MBSI: Special Header Functions

In addition to all the standard controls provided in every Metric Halo plug-in header bar, MBSI includes several special features.

Preferences tab



MBSI UI Preference

Note: UI preferences will be saved to the computer and automatically applied to the next instance of MBSI of whatever native format, even on another DAW or MIOConsole3d.

You can always revert to the factory default settings by clicking the **Reset Prefs to Factory Default** button at the bottom of the Preferences pane.

- **UI Mode**: sets the MBSI console surface graphic to be either *Dusty* or *Clean*.

  It is interesting to note that very few people so far are on the fence about which they prefer. Rather the reverse. Let's all get along, shall we?

- **Default Plugin Mode**: sets all new instances of MBSI to open in *Modern* or *Classic mode*. Default is Modern.

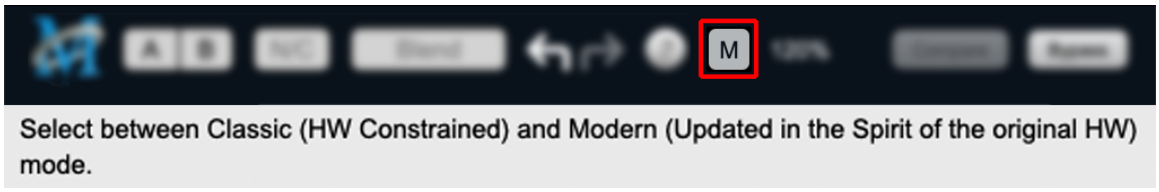
- **Pro Tools Control Hilite**: default is *Off*. Set to *On* enables the garish purple parameter UI highlight whenever EuCon or Pro Tools automation addresses an MBSI parameter to be applied to all new instances of MBSI.

- **Scribble Strip**: sets the Scribble Strip UI to either *Light* (the default) or *Dark*.

- **Default Chan Lo Cut**: sets the channel box tone low cut filter on all new instances of MBSI to open either *Enabled* (the default) or *Disabled*.

- **Default Chan Hi Cut**: sets the channel box tone high cut filter on all new instances of MBSI to open either *Enabled* (the default) or *Disabled*.

Modern / Classic switch





"C" for Classic, "M" for Modern

This button toggles between the **Classic** and **Modern** modes of the MBSI processor.

See the Operational Modes section for details.

Advanced Parameters



Advanced Parameters button

The gear icon at the far right of the Presets bar opens a window with Advanced Parameter settings for MBSI including Compressor Dual-Mono mode enable/disable, and Parameter Variation and Randomization controls.

See the Advanced Parameters section for details.

Operation

First, a word of warning.

MBSI is a relentlessly faithful reproduction of the beloved 1960's MCI mixing console custom-built for Studio C of the legendary Criteria Recording Studios in Miami, Florida.

"Relentlessly", in this case meaning: if you have a digital background, this plug-in is going to seem completely broken and wrong, *but* if you grew up doing music in studios fifty-plus years ago, listening and working with MBSI you may shed a tear of fond remembrance of how things worked back then. It was an era where "broken and wrong" equated to an opportunity for even greater creativity and beauty. Screw convention, screw what everyone else is doing, just feel it and float in the moment.

Tech heads and digital perfectionists are going to analyze the output of MBSI and swear it's broken. We did too at first to be perfectly honest, but when the engineers that used the console back in the day confirm that this is actually how it worked and most importantly how it sounded, and comparing our measurements with a 192kHz/24bit flac of Hotel California confirmed the oddities, like, *exactly*, we're feeling pretty good about it and are ecstatic we can bring MBSI to all of you.

Be aware, though, that since this is a hand-built old school analog console, built over 50 years ago having lived a life making history all that time, by the time we captured its soul it was very much showing its age. Thankfully, the major controls were all stepped resistor circuits which (mostly) responded well to electrical tender loving care, but when you see values in this plug-in referring to steps of 1dB or 2dB, like any mixing console of this vintage, that's an approximation rather than a mathematically assured value. MBSI's controls accurately reflect the actual values output by the best-performing processor stages in the console exactly as the console circuitry delivered it.

That said, we need to honor the fact that no channel was identical, so we've included a **Parameter Variation and Randomization** feature that allows you to tweak some of the parameter values for the various stages based on the vast library of measurements we made of the console. And to make it more productive, we've devised a way to direct the tweaks rather than having them totally random... so less searching for just the right variant.

Box Tone / Hardware Input Circuit

Hardware Signal Flow: Mic Pre → Line In → Suppressor → EQ → Insert (Compressor) → Fader → Bus Out

As you first pass audio through MBSI, it is immediately apparent that the board has a *sound*, even with EQ and dynamics sections turned off. This is a characteristic of the input circuit design, and will be present until you bypass the plug-in. This box tone model is present in both Classic and Modern modes.

Beyond the expected rich analog distortion signature and the overall linear frequency response, the board output is filtered at the extreme high and low extents of the frequency range. These are filters are mildly resonant, and add a distinct flavor to the console.

- At the low end, there is about a 1dB resonant dip at 35Hz and little bump at 20Hz before rolling off from there to the noise floor.

  In MBSI, this response is reproduced for all operating sample rates.

- Up top, the board response again has a bit of resonance with a little dip at 20k and bump at 26k, and rolls off to -3 at around 36kHz and down into noise around 85k to 90kHz. This filter response is reproduced exactly to the hardware in MBSI running at sample rates of 192kHz and above.

  At lower sample rates the resonant characteristics of the high frequency response curve are maintained, but the knee frequency is pulled back to fit in the available audio band, so at 2x sample rates the high frequency rolloff -3dB point is about 30kHz, and at 1x rates it's -3dB at 21kHz or so.

Operational modes: Classic vs. Modern

Being a child of the 60's as it is, it's not really surprising to find that some aspects of the original console design don't translate well to modern production methods and needs.

For example, the original console in question did not have a compressor (not unusual for consoles of that vintage), but it did have a Suppressor stage in each channel of the tape returns section, most likely as a kind of semi-automatic tape noise reduction. The Suppressor stage is included in MBSI for the sake of completeness, but the fact remains that modern channel strips are really expected to include some more sophisticated form of dynamics processing.

For this reason, MBSI offers **Classic** and **Modern** modes of operation:

Classic mode ***IS*** the original console hardware path, funky and weird, warts and all, purely analog in all it's glory.

Modern mode retains all the sonic characteristics of Classic Mode, with a few loving tweaks in consultation with Bill Szymczyk and others to bring it into the DAW / NLE age and make it completely indispensable.

In the MBSI header bar, next to the UI size selector is a button marked with with a **C** or an **M**. Clicking this button toggles between Classic and Modern modes.



"C" for Classic, "M" for Modern

The primary differences between Classic and Modern modes include:

1. **Suppressor (Classic) vs. Compressor (Modern)**: The Suppressor is essentially a dynamic expander with Threshold, Release and Depth controls. It is interesting and worth experimenting with because it is quite unique, but is more of a special-case tool rather than a modern-day music production workhorse.

   In Modern mode, the Suppressor is replaced by a newly created Compressor stage, designed specially by Make Believe and Metric Halo for MBSI, informed by deep analysis of the signal path characteristics and after many long conversations with those most familiar with the console in it's heyday.

   The Compressor incorporates the distortion characteristics of the Suppressor and support circuits, re-purposes the Threshold, Release and Depth controls, and adds a selection of Fast, Medium and Slow attack/release modes for the compressor response rates.

   The compressor attack/release mode switch is located between the Depth and Release knobs, and serves as a visual indicator that you are in Modern mode. If you do not see this switch, you are in Classic mode.

2. **EQ knob controls**: In Classic mode, the EQ section gain controls operate in 2dB steps, and the Mid Frequency selector is stepped values as well. The hardware controls were all fixed resistor rotary switches.

   In Modern mode, these controls are all continuous, interpolated at 0.01dB resolution from the individual steps as captured from the hardware circuits. Hold the Shift key while in Modern mode to use Classic mode stepped control values.

3. **CUT switch**: In Classic mode, the Mid band gain control is Boost only, and increments in 2dB steps.

   In Modern mode, a CUT switch has been added to the right of the Mid-Boost knob. When the switch is up, the Mid-Boost boosts gain as usual. When the switch is down, the same control cuts the gain value. Mid-Boost is continuous gain in Modern mode. Hold the Shift key while in Modern mode to use Classic mode stepped control values.

   The Cut switch serves as another visual indicator that you are in Modern mode. If you do not see this switch, you are in Classic mode.

4. **BOTH switch for 40Hz/10kHz Cut Filters**: In Classic mode, only the 40Hz low cut or the 10kHz high cut filter can be engaged at a time.

   In Modern mode, a BOTH switch has been added to the right of the 40Hz/10kHz Cut Filter switch. When the switch is up, selecting either cut filter will engage both the 40Hz and the 10kHz cut filters.

   Again, the BOTH switch serves as another visual indicator that you are in Modern mode. If you do not see this switch, you are in Classic mode.

5. **Phase Invert**: In Classic mode, there is no channel phase flip switch.

   In Modern mode, a Phase Inversion button is added above the main Fader control.

6. **Fader**: In Classic mode, the output Fader operates in 1dB steps. As with the EQ gain controls, the Fader was a stepped resistor circuit, switching between fixed resistor circuit paths as the fader was moved.

   In Modern mode, the Fader control is continuous, interpolated at 0.01dB resolution from the individual steps as captured from the hardware circuits.

7. **Box Tone Filter controls**: In Classic mode, the characteristic hardware circuit box tone is always engaged.

   In Modern mode, Advanced Parameters switches allow you to disable the low end box tone cutoff, the high end box tone cutoff, or both. See Channel EQ Enable for details.

MBSI Controls

The primary processing controls are the same between Classic and Modern mode. Both mode functions will be covered here, including the secondary switches added in Modern mode.

> **Please Note**: The MBSI UI reproduces the appearance and function of the original hardware EQ, dynamics, gain controls and switches as much as possible, down to the '70s vintage Dymo tape label for the Modern mode Cut switch.
>
> The scribble strip beneath the fader is currently just for vibe and to host easter-eggy type stuff (check out Bill Szymczyk's factory presets).

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.
- **Rapid parameter entry**
  - **Right** 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. **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.

Classic / Modern mode selector

Click the **Classic / Modern** button to toggle MBSI between Classic and Modern modes.



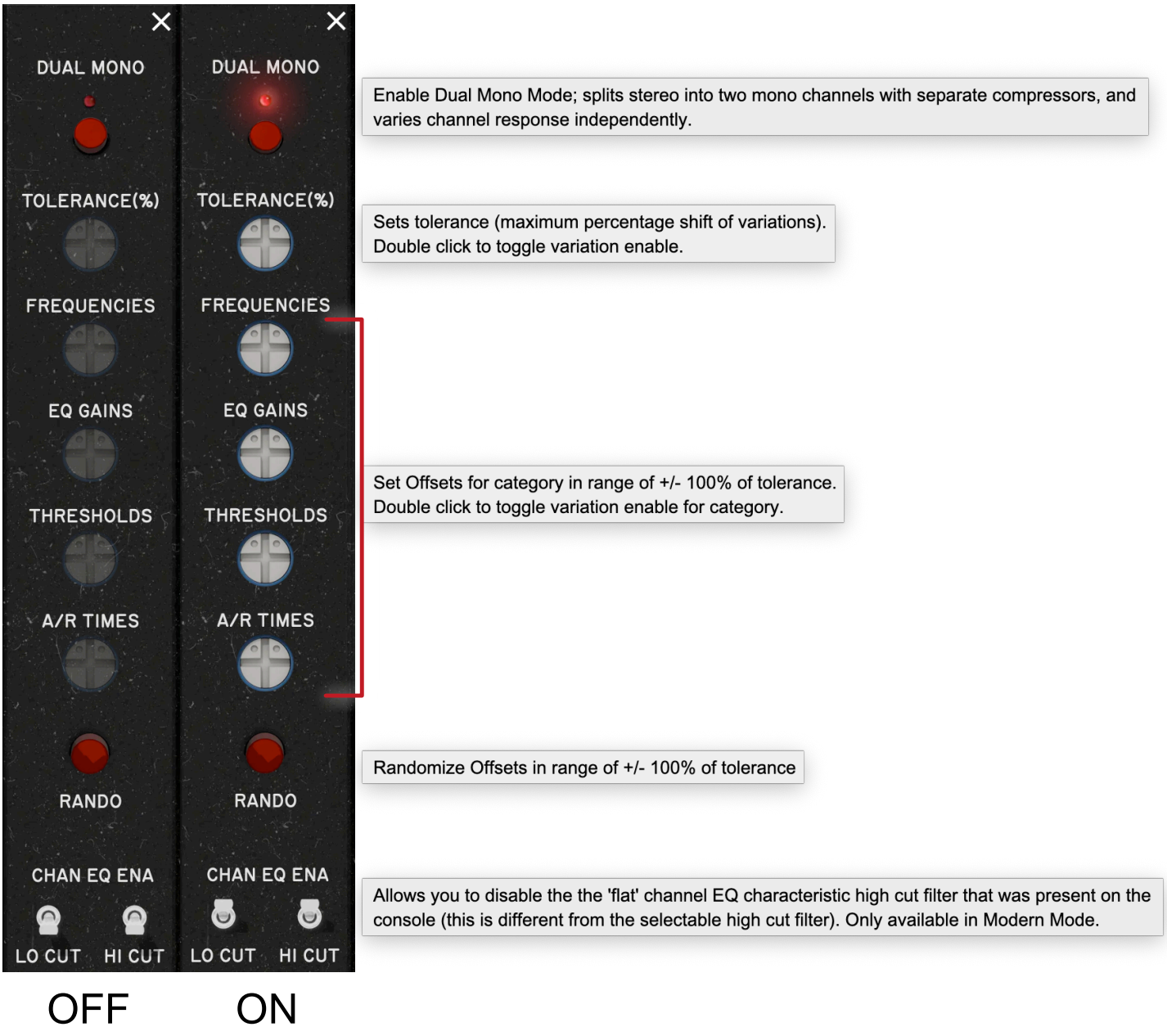
Select between Classic (HW Constrained) and Modern (Updated in the Spirit of the original HW) mode.

Parameter settings are not affected when switching between modes. Modern mode functions will engage and disengage cleanly when switching modes.
The mode setting is stored with saved presets.

Advanced Parameters UI

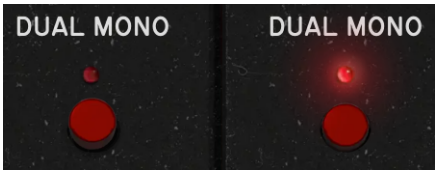


The **Advanced Parameters** button to the right of the Presets bar replaces the fader section with a control panel for operating two important features of MBSI: **Compressor Dual-Mono mode** and **Parameter Variation and Randomization**:



Compressor Dual-Mono Mode

In a stereo compressor instance, the **Dual Mono Mode** push button will become available in the Advanced Parameters UI. The *Up* position (LED OFF) is standard stereo mode and pushed *Down* (LED ON) engages dual-mono. You can see the compressor switch change shape as you press it.



In stereo mode, the detector sidechain is shared between the two compressor processor channels, with the loudest amplitude of the left and right inputs triggering both comp stages. This is the standard mode for stereo compression, maintaining a solid center image and consistent left/right balance with most stereo program across a wide range of compression settings.

In dual-mono mode, the left and right detectors are split, allowing the left and right channel comp stages to react independently to the dynamics of each channel. On well-balanced stereo program this can result in a more open and wider soundstage, taking advantage of minute variations between the two channels to expose details that might otherwise be masked in standard stereo compression mode.

Parameter Variation and Randomization

No analog console channel strip performs exactly like the one next to it. There is always some variance in noise, frequency, dynamics and distortion response for any given process settings even if it is not always obvious. The personality of a mixing console is due every bit as much to the differences between channels as it is the sound character of each channel component. This section of MBSI has been added to allow controlled emulation of that behavior in a way that is predictable and useful to every music production scenario, based on the real-world performance of the hardware.
Parameter Variation takes your current MBSI parameter settings as a starting point, and tweaks them a bit, per your instructions. Double-click a control to turn variance off for any parameters you want to keep unaltered.

In dual-mono mode, EQ and Compressor variations are applied differently to the two channels. So even if the Comp is turned off, if the EQ is engaged and gain applied you will get divergence between the two channels in a stereo strip.

Note: Core hardware console characteristics such as box tone and the distortion signatures of each stage are not altered. Only the user-controlled parameters listed below are subject to variation.

The variations controls below the dual-mono button operate as follows:

- **Enable Tolerance**: Disabled by default. Double-click to enable parameter variance between stereo channels.



The Tolerance control determines the amount of variation allowed for each of the parameter categories below, from 0% to 20%. 20% is a lot of swing, depending how the controls below are set up. For a ±10dB gain control, that's up to ±2dB variance, so, non-trivial.

- **Enable Frequencies**:



Sets the amount of variation for the 8kHz/10kHz Shelves and Mid-band Frequency. Values can be positive (for variation trending towards higher frequencies) or negative (trending towards lower frequencies).

- **Enable EQ Gains**:



Sets the amount of variation for EQ gain controls, including the Hi and Lo Shelf gains and Mid-Boost/Cut. Values can be positive (for variation trending towards increased gain level) or negative (trending towards decreased gain).

- **Enable Thresholds**:



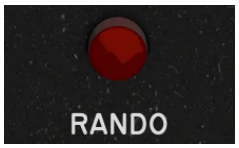
Sets the amount of variation for Compressor threshold level. Values can be positive or negative to trend the variation towards higher or lower threshold levels.

- **Enable A/R Times**:



Sets the amount of variation of Compressor attack and release times. Compressor attack times will vary from the preset values. Values can be positive or negative to trend the variation towards slower (positive adding attack/release time) or faster (negative subtracting attack/release time).

- **Rando**:



  Clicking **Rando** randomly scrambles active Frequencies, EQ Gains, Thresholds, and A/R Times parameters to values within the set Tolerance range.

- **Channel EQ Enable (Modern mode only)**:



  The switches labeled **CHAN EQ ENA** let you disable MBSI's characteristic resonant box tone low and high cut filters, allowing operation with just the circuit characteristics of the dynamics and EQ sections engaged. These filters are enabled (switch handle in the *UP* position) by default to represent the full sonic character of the original hardware console.

  Tracks with deeply extended digital bass energy can sometimes benefit from disabling the low cut filter.

Note: Parameter values on the processor controls do not change with Parameter Variation and Randomization. These are under-the-hood variances just like the behavior of the hardware console. The only functional difference is in MBSI you have creative control of the variances.

Saved MBSI user presets will include the Dual Mono Mode and Parameter Variation/Randomization settings.

> **Important Note: Blend**
> Advanced Parameter Dual-Mono mode and Parameter Variation features are considered non-automatable functions per the AU, VST3 and AAX specifications, and since the Blend functions as an automation tool, Blend can not affect compressor Dual-Mono mode and Parameter Variations settings.
>
> Advanced Parameters variances are properly maintained in saved presets and A/B registers, but can not (at least for the present) be Blended like the primary parameters of MBSI.
>
> The MIOConsole3d version of MBSI follows the same rules for the sake of consistency.

The primary MBSI processor controls are detailed in the pages to follow, listed per section from left top to bottom right, including the tooltips for each control as shown in MBSI when the Help button in the header bar is enabled.

Input section

The Input section on MBSI consists of the **Trim** and **Input Selector** controls at the upper left of the MBSI UI. Input section controls are the same for both Classic and Modern modes.

Trim knob



Input Trim Gain (-12 to +12dB).

Trim is a straight non-stepped linear gain control placed prior to the MBSI Line and Mic Input circuits. Gain range is from -12dB to +12dB.

Use the Trim to reduce the level going into MBSI before it hits the Input circuit distortion models.

Input Selector switch



Selects between Line, Mic Pre, and O/D (Mic Pre Overdrive) models. This is a zoned control. Click Left, Center, or Right to select the mode:
Line is the path from Line In through the strip.
Mic cascades the Mic Pre Module with the Line path.
O/D is the Mic Pre module with overdriven input, cascaded with the line input path.
NOTE: O/D is distorted; you will need to adjust the input trim (or the signal level before the plugin) to hit the sweet spot.

- **LINE**: selects the Line input model for the least colored and most linear response.
- **MIC**: Selects the Mic Pre input model. This is a Mic Pre analog stage feeding a Line input stage, as wired on the original hardware. The Mic Pre adds a richer transformer distortion element with slight high frequency bump up to about 12kHz.
- **O/D**: OverDrive. Two hard-driven mic pre circuits in series feeding a Line input stage. By OverDrive, we mean it. You should probably (definitely) turn your monitors down before engaging...

Inputs are all gain-compensated such that the MIC and O/D input selection outputs the same level as the LINE, so you can switch between input distortion models without the actual gain structure changing.

Dynamics section

The Dynamics section is below the Input section at the lower left of MBSI. The four primary controls are the Threshold, Release and Depth knobs and the Enable push-button, which also acts as a gain reduction indicator based on the brightness of the button lamp. These controls do not change when switching between modes.

Both Modern and Classic modes for each control are listed below. Modern mode controls are listed first since that is the MBSI default mode.

Threshold



Modern Mode: Threshold of Compressor. Classic Mode: Threshold of Suppressor.

- **Modern: Compressor mode**: Controls the level at which the compressor begins to reduce the gain applied to the signal. Compressor response characteristics are determined by Release, Depth and Compressor Attack/Release Mode controls.
- **Classic: Suppressor mode**: Controls the level below which the suppressor reduces the signal gain. Think of the suppressor as a very slow gate or a level-triggered autofade.

Release



Modern Mode: Release Time of Compressor (range controlled by Attack/Release mode switch.
Classic Mode: Release time of Supressor (SL = 3 seconds to FA = 1.25 seconds).

Important Note! The Release control operates backwards: turn *clockwise for faster* release rates and *counter-clockwise for slower* release rates. The knob is labeled **SL** for Slow on the left, and **FA** for Fast on the right, but it still feels weird.

- **Modern: Compressor mode**:Controls the release rate of the compressor gain stage. Release time range is set by the Compressor Attack/Release Mode control.
- **Classic: Suppressor mode**: Sets the release time of the Suppressor. **SL** (slow) = 3 seconds, **FA** (fast) = 1.25 seconds.

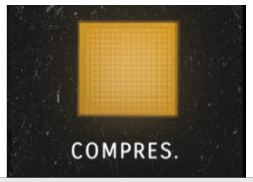
Depth



Modern Mode: Gain Reduction Limit of Compressor.
Classic Mode: Gain Reduction Limit of Suppressor.
0 dB effectively disables Dynamics for both modes.

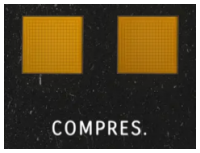
In both Compressor and Suppressor modes, **Depth** sets the maximum gain reduction to be applied by the processor.

Dynamics processor Enable / Gain Reduction Indicator



Enables the Dynamics block (Compressor in Modern Mode / Suppressor in Classic mode), and indicates Gain Reduction based on brightness.

In both Compressor and Suppressor modes, this push-button lamp switch enables/disables dynamics processing. When enabled, gain reduction is displayed by lamp brightness; the brighter the lamp, the more gain reduction is being applied.



In Dual-Mono mode, the Compressor Enable/Gain Reduction switch splits into two, with the two lamps showing independent left/right gain reduction. Clicking toggles the comp stage in/out as usual.

Compressor Attack / Release mode (Modern mode only)



Selects attack/release mode for the compressor. This is a zoned control. Click Left, Center, or Right to select the mode:
F = Fast (0 Attack, Release 50-120ms)
M = Medium (400us Attack, Release 50-120ms)
S = Slow (33ms Attack, Release 17-300ms)

Sets the Compressor Attack/Release mode. Attack times of each mode are fixed. Release times are set by the Release control.

- **F**ast: 0 Attack, 50 - 120ms Release range
- **M**edium: 400µs Attack, 50-120ms Release range
- **S**low: 33ms Attack, 17-300ms Release range

EQ / Filter section

The EQ section of MBSI is a familiar vintage EQ layout consisting of a high frequency shelf, a sweepable midrange band, a low shelf, and cut filters at 40Hz and 10kHz.

In Classic mode, all knob controls are stepped, including the mid-band frequency sweep.

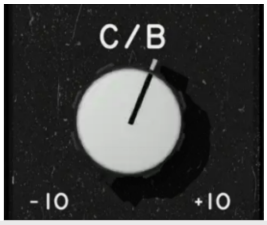
In Modern mode, aside from knob controls becoming continuous, the EQ functions are essentially the same with two important modifications:

1. a "Cut" toggle switch is added to turn Mid-Boost control into a Mid-Cut control, and
2. a "Both" switch allowing the 40Hz Lo-Cut and 10kHz Hi-Cut filters to be engaged at the same time

Keep in mind, aside from interacting with each other, each EQ section filter circuit also interacts with the inherent hardware box tone characteristic, and with the HF energy bump of the **Mic** input model.

It *will* get crunchy when pushed, just like the hardware console. If at the edge of too much crunch, try toggling the box tone Channel EQ filters off and on to see what effect it has.

Hi Shelf Gain



Selects the gain of the Hi Shelf. Stepped in Classic Mode. Hold <shift> to use stepped values in Modern Mode. Actual gain varies from calibration and readout, matching behavior of the hardware.

±10dB Hi Shelf gain control. Very wide slope. Frequency knee selected by the **Hi Shelf 8kHz or 10kHz switch** below.

Mid-band Boost



Selects the gain of the Mid Filter. Stepped in Classic Mode. Hold <shift> to use stepped values in Modern Mode. Actual gain varies from calibration and readout, matching behavior of the hardware.

- **Classic: Boost**: +14dB gain boost in 2dB steps for the mid-band bell filter.
- **Modern: Boost or Cut**:



Switches Mid Filter to 'Cut' Mode, gain is applied as cut rather than boost. Only available in Modern Mode.

Cut switch up = continuous gain boost up to +14dB for the mid-band bell filter.

Cut switch down = continuous gain reduction down to -14dB for the mid-band bell filter.

Mid-Band Frequency Select



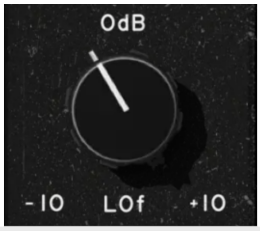
Selects the center frequency for Mid Filter. Stepped in Classic Mode. Hold <shift> to use stepped values in Modern Mode. Filter frequency readout matches processing, but consistent with the hardware does not match the legend on the panel.

OK, bear with us through this, please.
The hardware mid-band filter is inductor based, so as you increase the selected frequency approaching 7-8kHz (the resonance of the inductor) the band Q gets narrower and gains amplitude, then opens up again as you go higher. At the highest frequency setting it interacts with the boards' inherent response characteristic, softening extreme boosts in that range.

Note: The silkscreen frequency labels on the console UI are just plain wrong, and apparently always have been. It's frankly a mystery. The actual frequency set by the control as measured on the hardware is shown by the numeric readout as you click and drag the control.

In reality, the lowest frequency setting is 140Hz, and the highest is 9.87kHz.

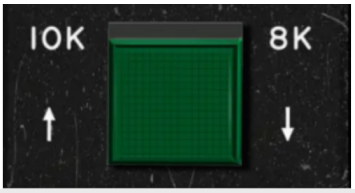
Low Shelf Gain



Selects the gain of the LF Shelf. Stepped in Classic Mode. Hold <shift> to use stepped values in Modern Mode. Actual gain varies from calibration and readout, matching behavior of the hardware.

±10dB Lo Shelf gain control. Very wide slope. Frequency knee selected by the **Lo Shelf 60Hz or 100Hz switch** below.

Hi Shelf 8kHz or 10kHz switch



Selects between 8k (On) and 10k (Off) for the HF Shelf.

Selects the knee frequency for the Hi Shelf. Lamp on = 8kHz knee. Lamp off = 10kHz knee.

Mid-band Filter Q Width switch



Selects between Narrow (On) and Wide (Off) Q for the Sweepable Mid Filter.

Sets the width (Q) of the Mid-band EQ stage. Lamp on = Narrow Q. Lamp off = Wide Q.
Note: Nearing 7-8kHz (the resonance of the mid-band inductor), narrow Q will have higher apparent amplitude at the center frequency than wide Q.

Low Shelf 60 Hz or 100Hz switch



Selects between 60 Hz (On) and 100 Hz (Off) for the LF Shelf.

Selects the knee frequency for the Lo Shelf. Lamp on = 60Hz. Lamp off = 100Hz.

EQ / Filter stage In/Out switch



Enables EQ when On.

Enables / disables the EQ section. Lamp on = EQ enabled. Lamp off = EQ disabled.

Low Cut and High Cut Filters switches



Selects between Low Cut, Off and High Cut Filter. This is a zoned control. Click Left, Center, or Right to select the mode.

- **Classic: Filters OFF, 40Hz Cut or 10kHz Cut**:
  OFF: No cut filter.
  40Hz: Engages sharp cut filter below 40Hz.
  10kHz: Engages sharp cut filter above 10kHz.
- **Modern: Filters OFF, 40Hz Cut, 10kHz Cut or BOTH**:



Enables both Low Cut and High Cut Filters if either is enabled.

Select either 40Hz or 10kHz and set **BOTH** switch *UP* to engage both 40Hz and 10kHz cut filters.

Output Section

Polarity Invert



Inverts the polarity of the signal when active. Only available in Modern Mode.

Enables / disables polarity inversion. Lamp off = No polarity inversion. Lamp on = polarity inverted.

Output Fader



Selects the output gain of the strip. Stepped in Classic Mode. Hold <shift> to use stepped values in Modern Mode. The 9 mark on the fader corresponds to 0dB / unity gain.

The channel output fader on the original desk was a stepped resistor ladder attenuator with no active gain amplification circuit. At **0** on the fader it was unity: no gain change. Moving the fader down reduced the output level in 1dB steps: from 1dB to 30dB of attenuation.

The final click at the very bottom of the attenuator engaged a hardware relay which muted the signal entirely. The small light to the left of the fader at that final click glowed white to indicate the mute relay was fully engaged.

In console of the day, ladder attenuators were often preferred as cheaper, more reliable and sounding more transparent than active circuitry. In modern digital workflows, though, one dB steps on everything might be a bit awkward.

For **MBSI**, *fader operation has been modified a bit.*

1. **Fader Calibration**: The fader UI remains true to the original hardware and has not been altered, but DAW channelstrips generally have some make-up gain available at the output. This is especially necessary considering the Modern mode compressor stage.

   In MBSI, unity gain is at the **9** mark on the fader. **9** is the default fader location when instantiating or resetting the plug-in.

   This allows the fader to provide up to +9dB of output gain when set to **0**, and attenuation of -21dB when set to **30** on the fader control.

   The true gain values are reported in the numeric pop-up below the fader as you move the control.

2. **Fader transition from -21dB to Mute**: In Classic mode, rather than a hard relay switch type mute, the transition from 30 to Mute stop is interpolated to better hide the transition from -21 to digital zero.

   In Modern mode, the fader attenuation range below **30** is interpolated and tapered down to about -60dB before engaging Mute. Hold the **Command** key while operating the Fader in this range for greater control.

   The relay mute light will glow white when the signal is fully muted, just like the hardware.

MB Philly DI

Intro & Operation

"My name is George Hajioannou and I was lucky to not only get to know Joe Tarsia but to become a colleague and a friend. He was a true audio visionary and his role at Sigma Sound Studios put Philadelphia on the musical map.

One of the promises I made to Joe Tarsia was to perpetuate the Sigma Sound Studios name in a positive light. This plug in is an exact copy of the Sigma Sound Studios Direct Box that Joe and I created. There were 50 made and were all signed by Joe.

It took two years to create this Direct Box and bring it to market. Every unit was quickly sold so this is a truly rare and special piece of gear." **- George Hajioannou**



The Philly DI is a plug-in based on the rare custom-built DI created by Joe Tarsia, founder and owner of Sigma Sound Studios (Philadelphia/New York).

"In 1967, before stock consoles and turn key studios, I and some believers, wire by wire, component by component, built Sigma's first studio. Primitive by todays' standards the console had just 12 inputs that fed a then state-of-the-art 8 track tape machine. It was then that I built the first Sigma DI. We wanted a general purpose direct box that would not only direct inject electric keyboards, but more importantly would get the best out of the electric bass and guitar. The unexpected results were astonishing: the Sigma DI quickly became the favorite of the bass and guitar players.

Always looking for the better mousetrap, as new DIs came to market we put our box to the test. In every head to head comparison with both active and passive store bought DIs, the Sigma box came out on top.

You know the sound; you hear it in the recordings of the Jays, Lou Rawls, Teddy Pendergrass, the Stylistics, the Spinners, MFSB, Grover Washington Jr and Patti LaBelle. It's 'The Sound of Philadelphia'.

The day Sigma opened its doors and the 35 years that followed, it was the only DI I used for recording musicians like Ronnie Baker, Gerald Veasley, Steve Beskrone, Bob Babbitt, Anthony Jackson and many others.

It was only at the urging of a musician friend, who wanted a Sigma DI of his own that I considered building a few. Unfortunately I found that the heart of the Sigma DI, a high quality audio transformer, had been out of production for years and is now just about impossible to find. We spent months searching supply houses, the web and electronic equipment manufacturers to find just 50 of the rare transformers." **- Joe Tarsia**

The **MB Philly DI** plug-in is a lovingly crafted digital emulation of a pristine original hardware unit provided by George Hajioannou of Studio Logic Sound - with a couple of twists...

In this plug-in, the standard signal path as it was used for tracking in the studio is emulated in full: instrument -> DI -> leveling/flavor compressor stage -> console line in.

In consultation with many engineers familiar with the Sigma setup, we've captured not only the sound of the orignal DI, but also provide a one-knob compressor stage which effectively reproduces the action and character of that leveling stage.

As a result, the controls in MB Philly DI are super straightforward:
Drive for anger management, and **Comp** to set the degree of dynamics control.

- **Drive**



  The transformer in the DI has so much headroom it actually requires quite a bit of drive to engage the transformer saturation. We added a volume compensated drive control to allow you to put the signal into the sweet spot of the transformer.

  The Drive setting has a range of ±24dB, with the default in the middle at 0.00.

- **Comp**



  Since the DI was almost always used with a leveling amp, we include a one knob compressor; simply turn the Comp knob to engage the compressor - it is gain compensated and has a fixed attack and release.

  The Comp setting spans from 0 to a maximum setting of 40.

Use the MB Philly DI to recreate that classic Philadelphia sound on lead vocals, bass, guitar, keyboards and all sorts of other sources you might ordinarily not think of for a DI plug-in.

MB CGII

Introduction

"In 2023 I was at NAMM with my son Tommy. We were walking around and we ran into one of his friends Ryan Harvey who was walking around with a laptop and had the Sontec MES-432 open on it. We struck up a conversation because I was so interested in that plug-in as I've used an actual Sontec for decades in the studio. I was later introduced to Rick Carson who said we should talk about making a plug-in.

Fast-forward two years later, when Rick asked me what I *really* wanted to release. Did I want to release a whole mastering chain? The answer was no... I wanted to release something that would help engineers master in todays modern environment. So with that being said, we've created my plug-in (CG-II) which helps you gain level to loud tracks and add some character (which seems to be what's going on in the industry today).

Our intention going forward is to create tools that I have always wanted to make amazing sounding masters - tools that are necessary for todays music." **- Chris Gehringer**



CGII is a mastering dynamics finishing toolkit.

CGII is comprised of two uniquely complementary dynamics enhancement sections, a new True Peak Limiting stage, DC blocking and dither with LUFS/true peak metering via tube filament displays, all operating at up to 16x oversampling.

The design prioritizes musical integrity and ease of use, with the ability to add effective loudness, weight and definition without dimishing spatial depth and visceral impact the way traditional dynamics processors do.

Operation



CGII features a level metering display above a streamlined UI with two control knobs and a small selector switch array.

- The Nixie tube numeric readout across the top displays EBU R128 specified Integrated, Momentary or Short Term LUFS or True Peak values at the output of the processor.
- density brings up previously masked small-signal detail without affecting existing peak dynamics.
- filter engages a 1Hz DC Block filter.
- true peak switch engages our brand new oversampled and look-ahead True Peak Limiter as the final process stage.
- The **0dB, -0.1dB** and **-1dB** switches set the limiter output ceiling.
- **bypass** duplicates the Bypass button in the plug-in header bar.
- The loudness control operates a uniquely tuned soft saturation stage before the peak limiter.
- The dither adds dither and outputs the specified audio word length from 24 bits down to 16 bits.

Tooltip mouse-over pop-ups are available for all UI elements by clicking the circular **?** Help icon in the plug-in header bar.

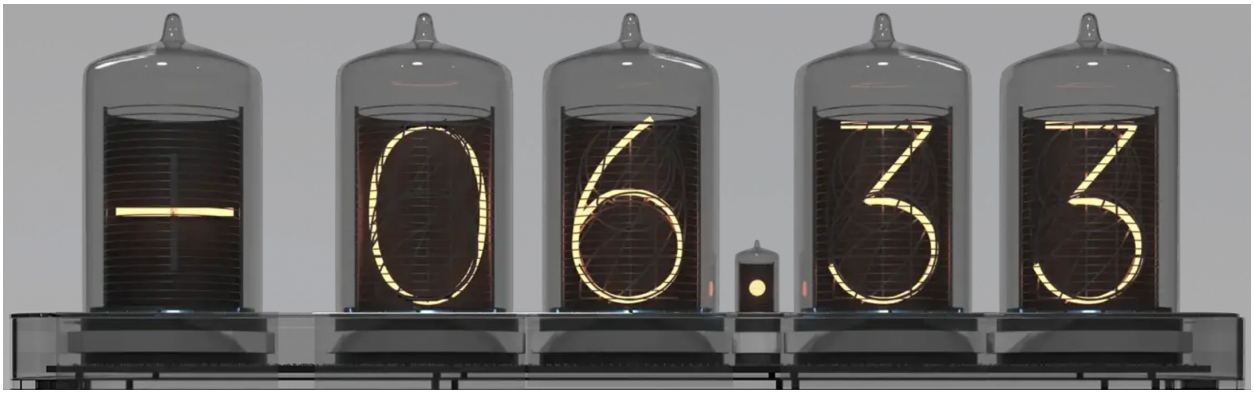
Knob parameter values are displayed via tooltip overlay as you adjust the knob.

Right-click a knob to enter a value manually.

Features and controls are individually detailed in the following pages.

EBU R128 LUFS / True Peak Meter display

The Nixie cold cathode tube display shows the current LUFS or True Peak levels taken at the output of the processor as per the EBU R128 specification.



Click the decimal point to select the display type:



CGII numeric display selector

Click anywhere in the display area to release/reset the meter.

Density control

The **density** control knob progressively promotes lower level program detail without affecting the peak volume range. The control range is from "0" (no effect) to "30" (maximum effect). The actual degree of perceived level change is dependent on the source program.

The higher you turn the knob, the more small-signal detail is brought up.



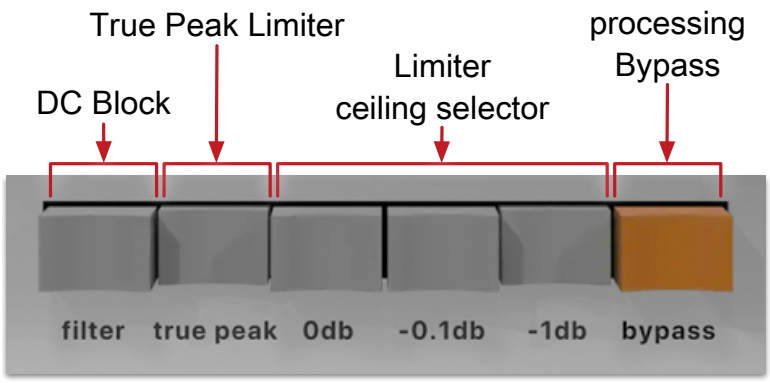
The algorithm increases perceived program level by pushing up previously masked detail without affecting higher level peak dynamics. In louder, more modern mixes this often adds weight and heft (hence the parameter name).

In more open audio with lots of dynamic range, **density** will bring up critical program information that might end up masked or unnaturally degraded by codec and/or dynamic range post-processing for streaming.

In mixing, the **density** control is specially useful as a performance leveler for vocal and acoustic instrument tracks where louder phrases sit perfectly but more consistency and presence is needed in quieter sections of a given track. It offers a lighter touch than a traditional leveling comp/limiter stage, but because of its purity and transparency, less processing is needed for desired outcomes.

Button bar controls

The push switch array controls are (from left to right):



DC Block filter

filter inserts a DC block at the input stage. The filter is a 2nd Order high-pass with the corner frequency at 1Hz, down to -∞ dB at DC.

True Peak Limiter

The **true peak** switch engages the CGII True Peak Limiter after the Loudness soft saturator stage.

The function of the **0db, -0.1dB, 1.0dB** Limiter Ceiling Selector is dependent on the state of the True Peak Limiter:

- **Limiter OFF**

  When the True Peak Limiter is OFF, the **0db, -0.1dB, 1.0dB** threshold value is used as a gain offset to the output of the Loudness stage. It will set the output ceiling to the level specified, but there is no protection against higher true peaks slipping through.

- **True Peak Limiter ON**

  With the True Peak Limiter ON, **0db, -0.1dB, 1.0dB** sets the True Peak threshold/ceiling, beyond which there will be no intersample peaks.

To confirm, the True Peak Limiter stage is by nature look-ahead and internally oversampled. When the Limiter is OFF, CGII switches to a delayed parallel path bypassing the processor stages entirely to ensure that plug-in latency remains constant and as transparent as possible at all times.

Note: The **0db, -0.1dB, 1.0dB** threshold values are radio-style buttons, so only one can be selected at a time.

Bypass

Pressing the **bypass** button cleanly disables all processing. When engaged, CGII will maintain the time delay through the channel and will continue to show metering. This bypass is a mirror of the Bypass control in the plug-in header bar.

Loudness control

The Loudness knob sets the soft saturator stage threshold in dB (inverted). If the knob is at 6, the threshold is -6dB.



The saturator has been designed to work in concert with the **density** stage to stabilize and drive higher levels of perceived loudness without the usual sacrifices of soundstage width, depth and detail which compromise the impact and vibe of the music.

Dither control

Toggling the Advanced Settings icon at the right of the plug-in header bar shows / hides the CGII **Dither** control.



CGII Dither (active when yellow, as shown below) engages TPDF dither at the output stage.



The **Dither Bits** control sets the appropriate amount of dither applied for final destination audio word length reduction (such as delivering to 16 bit CD format).

The Dither Bits range is from a maximum of 24 bits down to 16 bits.

MELD

Introduction

When I started doing Atmos mixes, my business partner Serban Ghenea and I sat and had a conversation:

What direction did we want to go in Atmos?

Working for 20+ years with Serban, I know how much goes into the small details of every mix. These mixes are tweaked to perfection with the artists and producers. Every detail has been dialed in against every other detail.

So starting in Atmos, we decided that we wanted to first respect those mix decisions, and among the most critical of those mix decisions are volume, loudness, compression, limiting; the "glue" that holds a mix together. The seeming obsession over the "master buss" chain that we use cemented in my mind that this was a critical component of the mix to be retained when doing Immersive mixes.

However, when the mix is printed out to stems, we are losing some of that glue.

When a stereo mix "master buss" is split out into many different instances for beds and objects, we are losing some of that glue.

And when a stereo mix is mastered, adding layers of compression, limiting, loudness, and EQ; the Immersive mix should follow or that glue is also lost.

For many years I tried to compensate for these losses by very carefully tailoring the volumes of objects and beds, adding limiting to some tracks or some objects, and side-chain compression to my beds and objects.

But what I really needed was a tool where all instances follow one instance and where the "side-chain" doesn't key off of one feed, but all feeds combined.

I needed it to be light on processing power so that it could be added to the end of an already filled set of mix buss plugins. I needed it to be reliable so that older versions would not be broken by newer versions. I needed it to be simple to use, so that I could quickly set up, meter, and monitor the results.

With the relationship built with Make Believe Studio and Metric Halo working on the MixHead plugin, I knew who to turn to. I asked them to build me this plugin and MELD was born.



This re-created the buss processing, the mastering processing, and the missing glue so that Immersive mixes can have the same feel as the stereo mixes; because from this basis, when it sounds right, you can make those panning decisions into Immersive space with the confidence that you are not losing the soul of the mix, but are immersing yourself into that highly detailed, dialed in production.

We added features such as the ability to use multiple instances; Not only can you recreate the glue on the mix buss, but do it on the drums buss, or BGV subgroups as well. Or you might have a Music, Dialog, Effects setup; A, B, C instances ready to go!

This opened up MELD to not only be a tool for Immersive mixing, but a really cool general mixing tool, because it can replace complex audio routing tasks with better results. Creative routing results without the complex routing.

Replace your aux buss, your folder, your sub-master with MELD. Now your sends from channels feeding the Aux or folder will follow the channel's EQ and dynamics processes, and your routing is simplified so your stems will print more closely to the mix.

I know of one well known engineer and producer who does not like to use any processing on his stereo master buss because, working with film projects, his stems need to match the mix exactly. With MELD he can have the best of both worlds.

So what started as a need for dynamics and EQ control for Immersive mixing has become a really interesting creative mixing tool as well.

- John Hanes

What exactly is MELD, though?

MELD is a channelstrip plug-in... in a sense it is *the* ChannelStrip plug-in, since it houses some of the main tools used at Mixstar Studios: Metric Halo ChannelStrip and Make Believe Studios MixHead.

MELD is designed to be inserted across any and all tracks in a host DAW session which would be bussed and processed in aux or group submix stems. All the instances of MELD communicate with each other, and the MELD UI lets you group and link the processing parameters to apply identical processing to every instance in the group, without any bus routing, prior to any panning stage. In the case of transitioning a completed stereo mix session to immersive, you just insert MELD across all the tracks that were sent to submix buses, and create MELD groups for each set of tracks per submix. In essence, each MELD group replaces each stereo submix bus, and since the grouped tracks are now untethered from the submix bus, they can be panned throughout an immersive soundfield without restriction.

MELD takes the EQ, Compressor, and Limiter from MH ChannelStrip, adds Make Believe MixHead, and includes a new Loudness processor optimized for dynamic presence control at the Atmos LUFS target range.

The EQ, Comp and MixHead processor blocks support recalling factory and custom presets created in the standalone MH ChannelStrip and MixHead plugins. As with parameter changes, preset recalled to a grouped instance will be applied to all instances in the group.

The MELD Overview tab displays quick-access control interfaces for the EQ, Comp, Loudness and Limiter modules, and the module tabs can be dragged to re-arrange the order of the processes.

With MELD, huge track folders of harmonized backing vocals, massed guitar overlays and FX channel arrays routed to dozens of buses and object channels can be dialed in quickly and easily in a single UI, saving time and hassle while improving the quality of the result all at once. Naturally, MELD groups can be automated in the same fashion as submix processors - automating a single instance within the MELD group applies that automation to every track in the group.

One of the greatest advantages of a MELD workflow is, MELD grouped tracks are processed as individual signal paths with the exact same processors set to the same settings, and because each process path is discrete, the distinctive performance characteristics of each track come through fully articulated. This results in a richer soundfield presentation with more detail and localization of each track than processing through a group bus or multichannel stem.

Of course, sometimes you want the extra glue of bus-processed tracks, and MELD provides the Advanced Parameter **Shared Sidechain Support** to accomplish precisely that effect by allowing full control of the detector sidechain linking of each groups' compressor and/or limiter blocks.

Important Note for MELD in MIO DSP: The Shared Sidechain feature requires that all the instances have access to the same memory buffers in order to get access to the audio data from the other instances in the sidechain, which is not possible with the MIO 3d multibox architecture. As a result, we cannot support that feature set within MIOConsole on the hardware DSP.

Links to videos

John Hanes making an Atmos mix from stems

John Hanes making an Atmos mix from mix session

Mike Monseur showing how he uses MELD in an Atmos session

Ari Morris using MELD for ATMOS mixing

Rick Carson MELD video walkthrough with an advanced sidechaining tools overview with Metric Halos BJ Buchalter

Plug-In Header Bars

All Metric Halo and Make Believe family plug-ins display the MH header bar at the top of the plug-in window. This header bar allows you to organize and access all your presets across all supported plug-in formats on Mac, Windows and Metric Halo hardware DSP via MIOConsole3d.

It is especially useful in that, regardless of platform, it provides a straightforward, powerful and consistent processing workflow wherever you might be working.

With the constant evolution of computer capabilities opening the door to new production techniques and music delivery formats, the differences between DAW software workflows have become ever more diverse. Many of the major DAWs provide their own plug-in headers within every plug-in instance window, providing their own feature set catering specifically to their internal workflow.

Conversely, other equally popular DAWs provide no added feature support for plug-ins (such as plug-in parameter Undo/Redo), opting instead to insert plugs as a straight processing block.

The plug-in header bar bridges that gap by offering the most asked for plug-in functions in a simple GUI, making all of our plug-ins functions and their presets available to every user on every platform.

The plug-in header bar has two rows of controls, with the Metric Halo logo icon at the top row left.

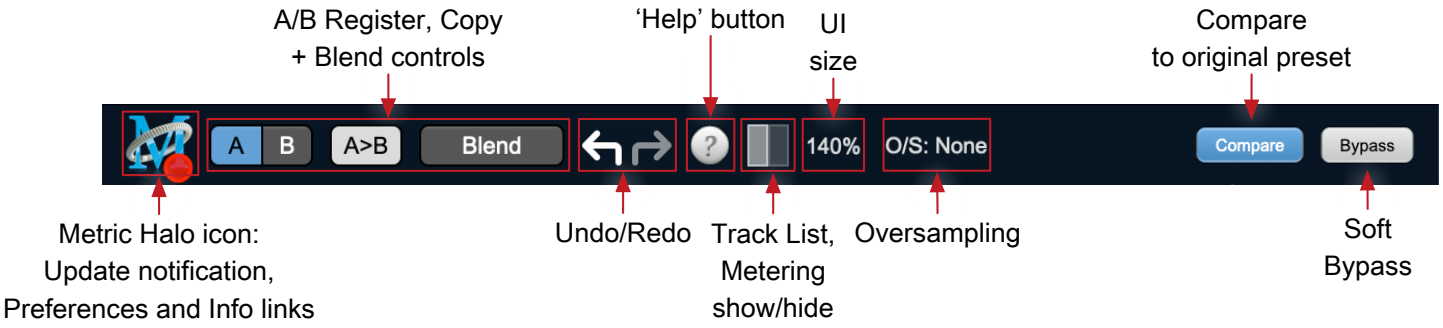
The top row handles plug-in operations, including update notification and download, access to MH online resources, GUI preferences, tooltip help, A/B parameter snapshots, snapshot Blend, plug-in Undo/Redo, Oversampling modes, Compare and soft Bypass.



MELD Plug-in header

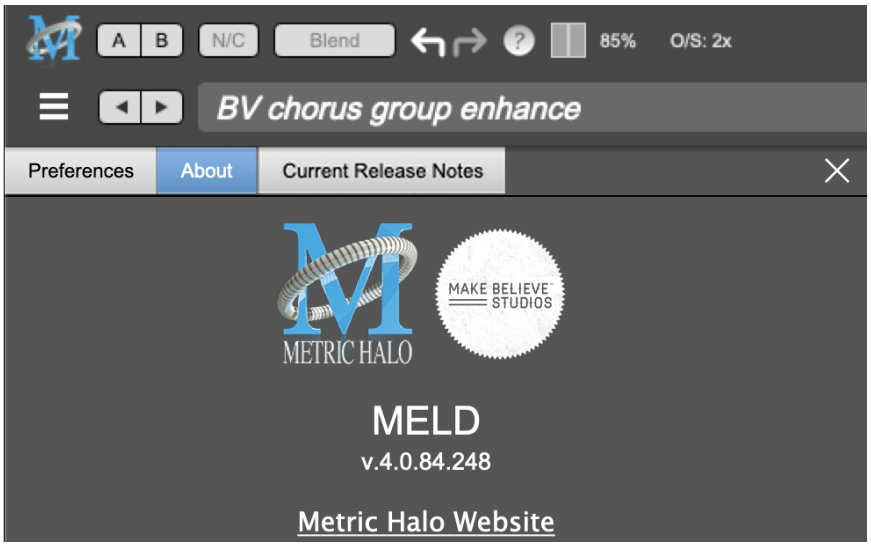
The lower row (with the ‘hamburger’ menu icon at the left) is all about preset management.

Plug-In Header: Top Row

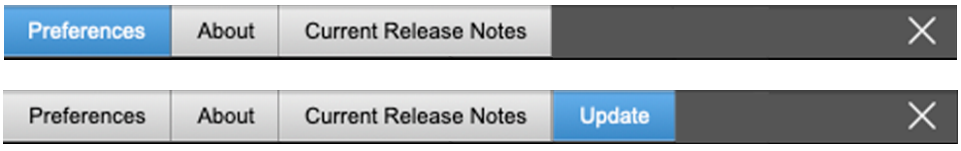


Metric Halo Header Icon

Clicking the MH icon expands the entire plug-in window to the right and opens a multi-function control sidebar with the Preferences tab in focus:



This sidebar has context-sensitive tabs across the top for GUI **Preferences** (shown above), **About**, and **Current Release Notes**.



Plug-in control pane tabs

When updates are available for download from Metric Halo, the MH icon will sport a blatant red dot (shown in the header map at the top of this page) and an **Update** tab is added to the sidebar. These tabs are dynamic by design, and additional tabs may appear as new content becomes available.

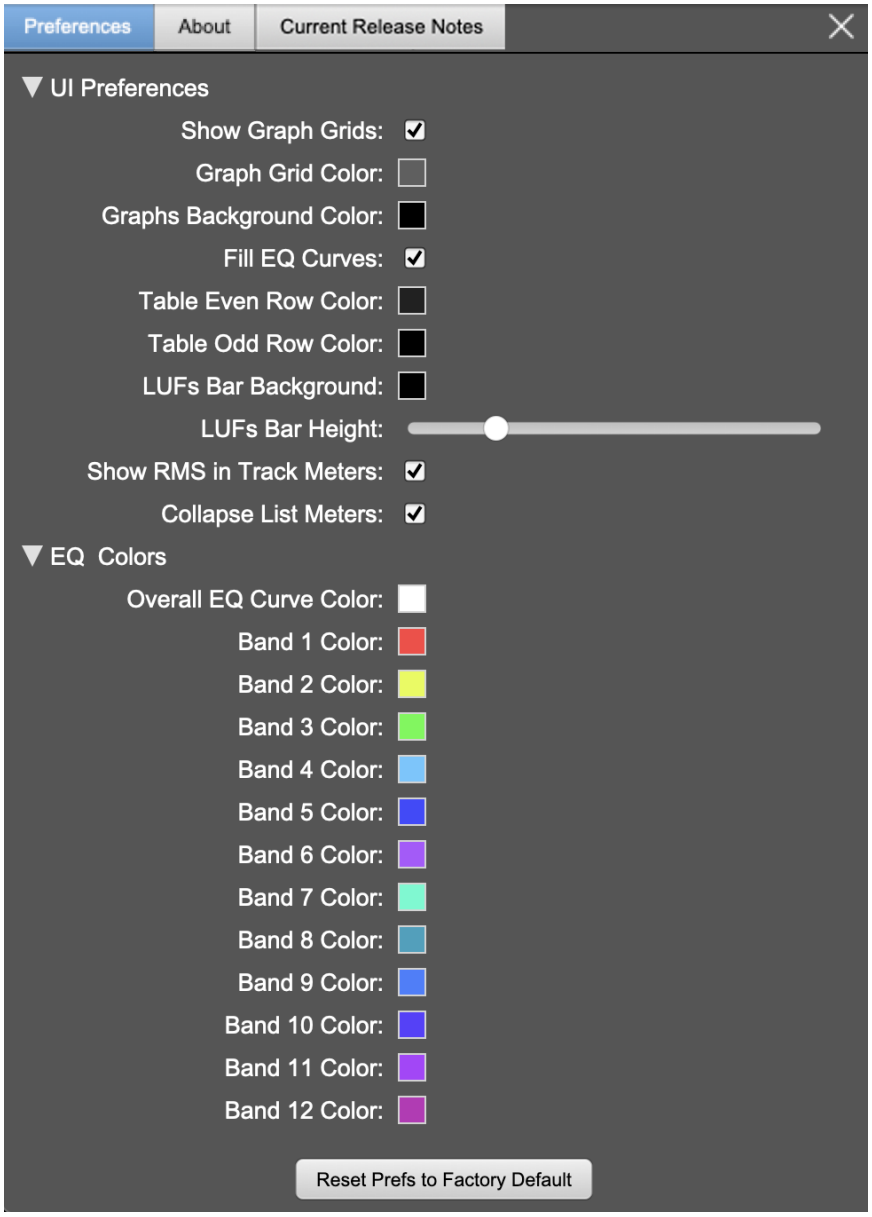
To close the sidebar, click the MH icon again, or the "**X**" at the right edge of the tab bar.

Preferences tab

The Preferences tab provides controls to modify the color scheme of various graphical elements within the MELD UI.

UI parameter changes are live, so you can see changes to the plug-in graphics as you make them. This includes colors, gradients, graphic element show/hide, knob types and bevels.

Clicking color selection swatches opens the standard OS color pickers window with the eyedropper tool for sampling any color visible on your display.



UI Preferences

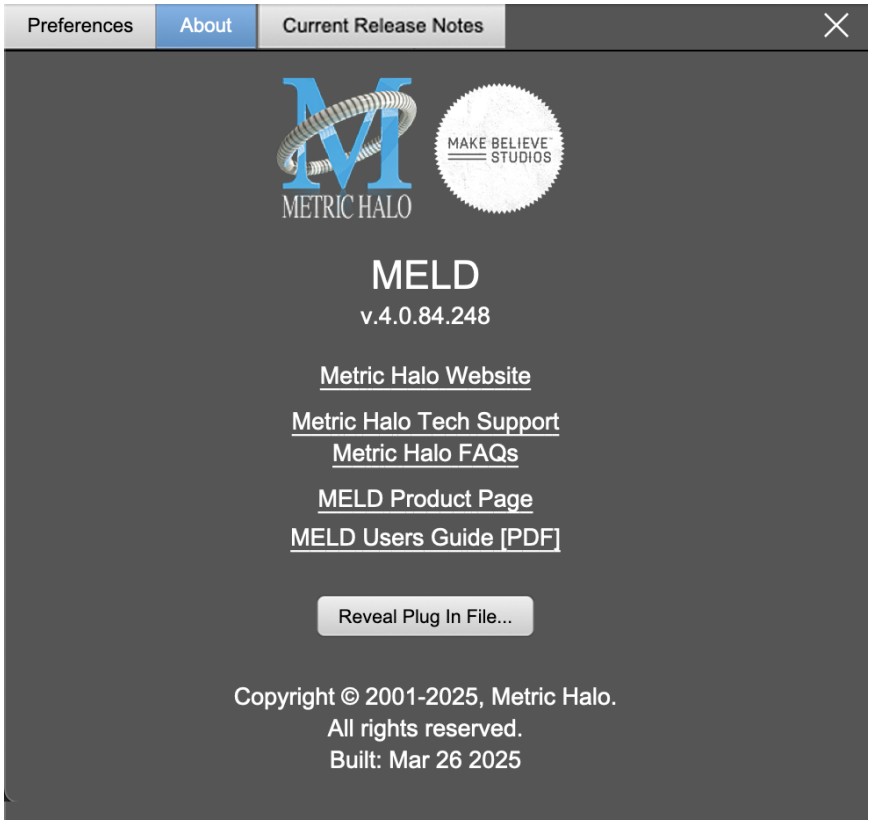
- **Show Graph Grids** toggles the display of transfer function graph reference grids.
- **Graph Grid Color** sets the transfer function grid lines color. The default color is a medium grey.
- **Graphs Background Color** sets the transfer function background color.
- **Table Even Row Color** sets the background color of the even-numbered Track List rows.
- **Table Odd Row Color** sets the background color of the odd-numbered Track List rows.
- **LUFS Bar Background** opens the macOS color picker to set the background color behind the Short and Long Terms LUFS and True Peak numeric readouts below the Track List.
- **LUFS Bar Height** sets the height of the Short and Long Terms LUFS and True Peak numeric readout display.
- **Show RMS in Track Meters** sets the Spectrafoo Level Meters in the Track List to show RMS along with PPM and Peak. With RMS enabled, the solid bar of the level meter indicates RMS, the colored pip shows PPM Peak, and the red pip a 3 second Peak Hold.

  With RMS disabled, the solid bar fills in to display PPM Peak with the Peak Hold pip.
- **Collapse List Meters**, when checked, flattens stereo and multichannel PPM meters to a single highest-level meter, and locks Track List row heights to their minimum size.
- **EQ Colors**
  - **Overall EQ Curve Color**: sets the color of the aggregate EQ energy curve (the total sum effect of all active bands)
  - **Band 1 Color**: sets the color of filter band 1 UI elements
  - **Band 2 Color**: sets the color of filter band 2
  - **Band 3 Color**: sets the color of filter band 3
  - **Band 4 Color**: sets the color of filter band 4
  - **Band 5 Color**: sets the color of filter band 5
  - **Band 6 Color**: sets the color of filter band 6
  - **Band 7 Color**: sets the color of filter band 7
  - **Band 8 Color**: sets the color of filter band 8
  - **Band 9 Color**: sets the color of filter band 9
  - **Band 10 Color**: sets the color of filter band 10
  - **Band 11 Color**: sets the color of filter band 11
  - **Band 12 Color**: sets the color of filter band 12

Note: All UI preferences listed above will be saved to the computer and automatically applied to the next instance of MELD of whatever format, even on another DAW or MIOConsole3d.

You can always revert to the factory default settings by clicking the **Reset Prefs to Factory Default** button at the bottom of the Preferences pane.

About tab

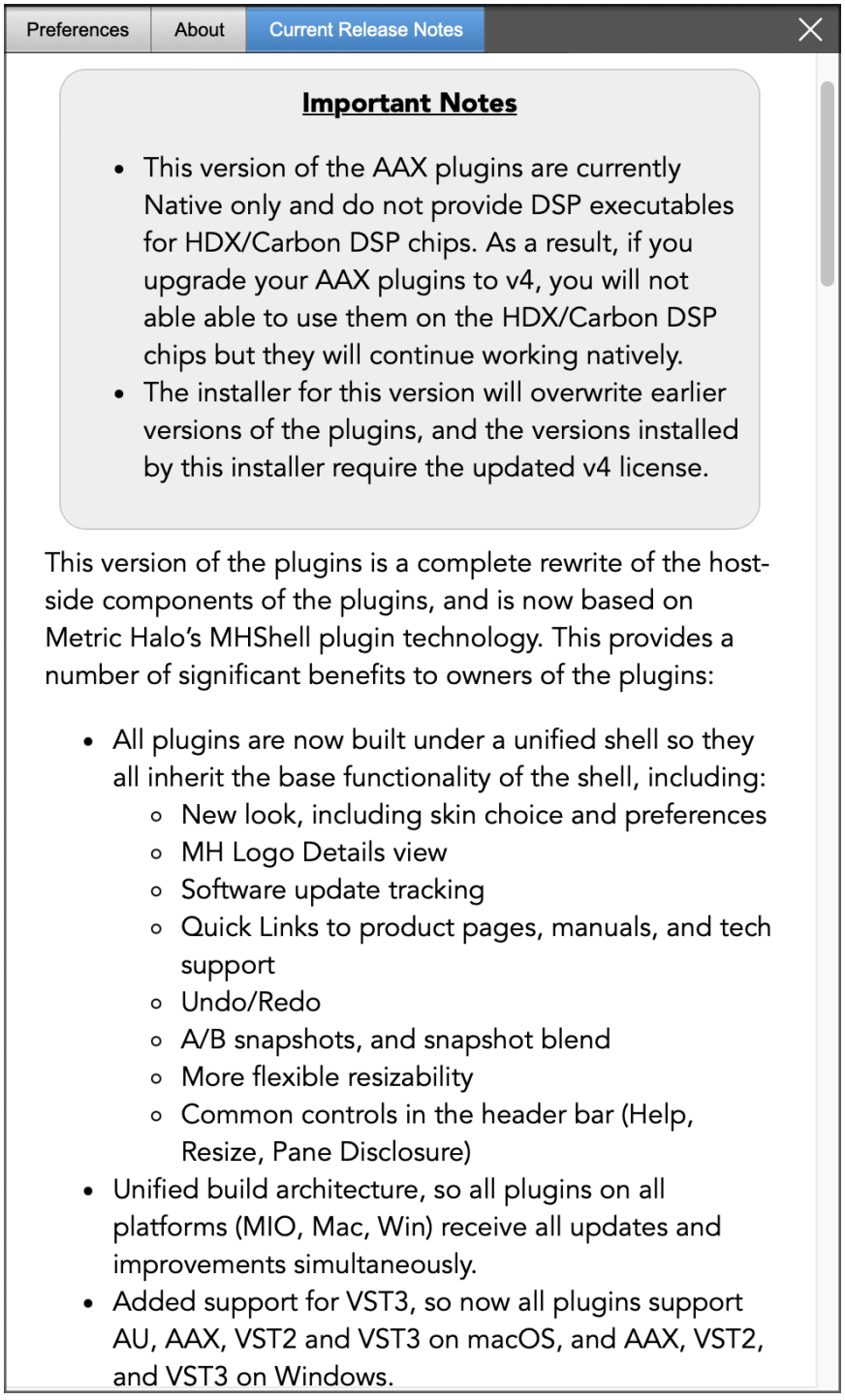


The **About** tab reveals the current plug-in version information and provides convenient web links to product info, support pages and the current manual on the Metric Halo main website.

Clicking the "MELD Users Guide [PDF]" link will open and display the latest Mixhead manual in your default web browser.

Reveal Plug-In File... will open the folder containing the current plug-in file, with the plug-in file itself selected. Very handy for troubleshooting on the fly.

Current Release Notes



Current Release Notes example

Current Release Notes lists a synopsis of major changes (a snippet of which is shown above), with feature and bugfix revisions for the most recent software releases listed below.

Please note that on Windows, the release notes will not be displayed in the pane, and a link to the release notes will be displayed instead. You can click the link to view the release notes using your current default web browser.

Update notification tab



Update tab (only appears when an update is available)

The **Update** tab will contain a link to download the new installer package in the header at the top of the pane.

Below the download link header will be release notes detailing the major changes included in the update, with bugfix revisions for the most recent software releases listed further below. Windows users will see a link to view the release notes using your current default web browser.

Click the Installer link to download, unzip and run the installer manually, preferably when your audio software is inactive so it can properly scan the new versions at launch.

Plug-in Snapshot Registers: A/B



Plug-in Header: Snapshot Registers: A/B

The A and B buttons control the A and B state registers. The A/B registers are used to store modified parameter snapshots in addition to the original saved preset called up by **Compare**.

The Blend function can be used to smoothly morph between the parameters set in the A and B registers, and Blend is a mappable parameter so it can be operated with external MIDI control. Details of the A/B Snapshot Blend feature follow on the next page.

For each of the A and B buttons the visual display tells you the state of the register:

- Light Grey means the register is empty
- Dark Grey means it has settings, but is unselected
- Blue means it has settings and is selected

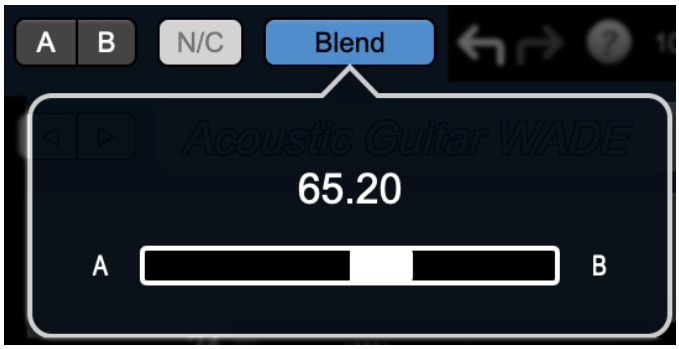
You can perform the following actions:

- Clicking on an empty register takes a snapshot of the current settings and saves them to the register.
- Clicking on an unselected filled register copies the register parameters to the current active plug-in settings.
- Clicking on a **selected** register toggles to the other register; this lets you toggle between the register settings without having to move the mouse.
- The Copy button will show **N/C** and remain inactive until one or both of the registers is in use. The Copy button will alternate between **A>B** (copy A to B) or **B>A** (copy B to A) depending on which register is selected. Clicking the Copy button then copies the settings from the selected register to the target register, overwriting the prior contents (if any).
- <Option> - clicking on a register snapshots the current settings and saves them to the register, overwriting the prior contents (if any).
- Changing settings when a register is selected will update the settings in the register to reflect the change.

Snapshot Blend

The Blend button allows you to interpolate (or morph) between the parameter snapshots stored in the A and B registers. It becomes active when both A and B have a parameter set stored.

Blend is a MIDI-mappable parameter so it can be operated in realtime with external MIDI control and/or automated in the DAW. This allows you to automate a transition from the A → B register, the B → A register or any setting between the two.



Plug-in Header: Snapshot Blend

The Blend button's visual display tells you the state of the register:

- Light Grey means it is empty
- Dark Grey means it has settings, but is unselected
- Blue means it has settings and is selected

Click on Blend to popup the blend control. Slide all the way to the left to apply the settings in the A register. Slide all the way to the right to apply the settings in the B register. Intermediate settings for blend will give you intermediate settings for any parameter that is different in register A and B. The blend control does not change the state of Bypass.

Note that the Blend is not a parallel processing mode where two instances of the processor are running the A and B settings and the output is a parallel blend of the two settings. Rather, Blend interpolates the parameter settings of the two registers to one instance of the processor. You can see the parameter controls move between A and B settings as you slide the Blend control.

The A/B and Blend settings are stored and recalled as part of the plug-in state, but are not saved as individual presets in the preset bar.

While you can use the blend with arbitrary A and B settings we find it works best when you craft the settings in the two registers in such a way as they are related to each other. Specifically, if an indexed (stepped) parameter is different between the two settings, the interpolated value will snap to one of the indexes between the two settings, which can be jarring.

It is best if the parameters that you blend are smooth parameters (e.g. gains, frequencies) and make sure the indexed parameters (enables, modes, band types) are set the same for both registers.

The easiest way to do this is to load the same setting into both registers and then tweak the settings of one of the registers.

This works especially well if you make one of the registers be the basic settings with all the gains or thresholds flattened out so that you can smoothly interpolate between a setting and effectively bypassed - we have found that this allows you to zero in a perfect configuration between too much and too little.

Plug-in Undo/Redo

All the plug-ins provide support for undo/redo from the plug-in header bar.



Plug-in Header: Undo/Redo

The left and right curved arrows represent Undo (Left) and Redo (Right). These arrows are grey when there is nothing to Undo or Redo.

The arrows are white when it is possible to Undo (Left) or Redo (Right). Clicking the left arrow when it is white will undo the last action you made in the plug-in. When you undo something that change is placed on the redo stack, and the Redo button will turn white.

Clicking the Redo button (when it is white) will restore the state that the last Undo changed.

If the Redo button is white and you make a change in the plug-in, the Redo button will go grey as the redo buffer will be cleared.

Help Button



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.

Track List / Metering Selector



This button toggles visibility of the Track List and Metering section of the MELD interface. This button allows you to maximize screen real estate while keeping processing and Track selection controls available.

Above left shows the button with the right pane dark, meaning the Track List / Metering section is hidden. Above right shows both MELD UI sections visible.

UI Size Selector

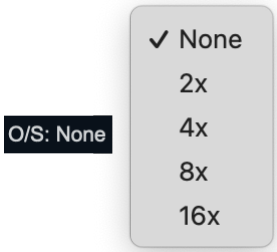


UI Size Selector

This pull-down menu lets you set the plug-in UI size to taste.

The size is remembered and applied the next time you insert or open MELD.

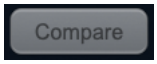
Oversampling Selector



Oversampling Multiplier Selector

Oversampling is applied per MELD Group, with O/S settings from 2x to 16x. The default Oversampling setting is None.

Compare Button



Compare Button (inactive)



Compare Button (active)

To use the compare button, a preset must first be loaded. The compare button will light up when the current settings differ from the selected preset. If you click this button while it is lit, the preset settings will be restored, but you can still return to the changes you made by clicking on the button again. It is important to note that any changes you make to activate the compare light are always for comparison to the last loaded preset.

Soft Bypass



Soft Bypass Button (not bypassed - plug-in is processing)

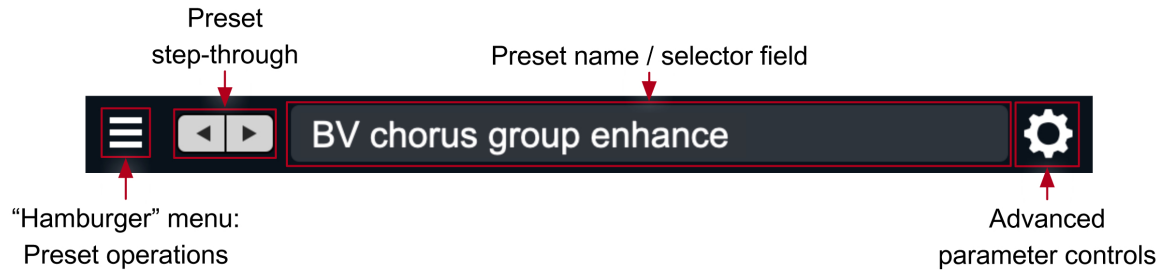


Soft Bypass Button (bypassed - plug-in is *not* processing)

When glowing yellow, this button will maintain the time delay through the channel and will continue to show metering, but will cleanly disable the processing.

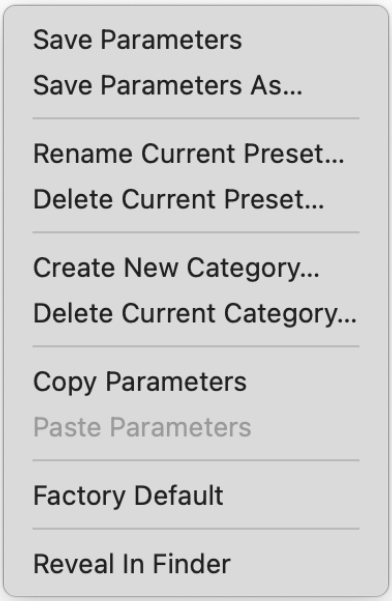
Plug-In Header: Preset Row

Important Note: In MELD, presets are recalled to all instances in the currently selected MELD Group. Otherwise, MELD preset operations are just like any other Metric Halo plug-in.



Plug-in Hamburger menu

The preset and parameter functions within the hamburger menu break down as follows:



- **Save Parameters** saves the current plug-in parameters to the current preset.
- **Save Parameters As...** opens a dialog box where you can name and choose a category to save 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 paste 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.

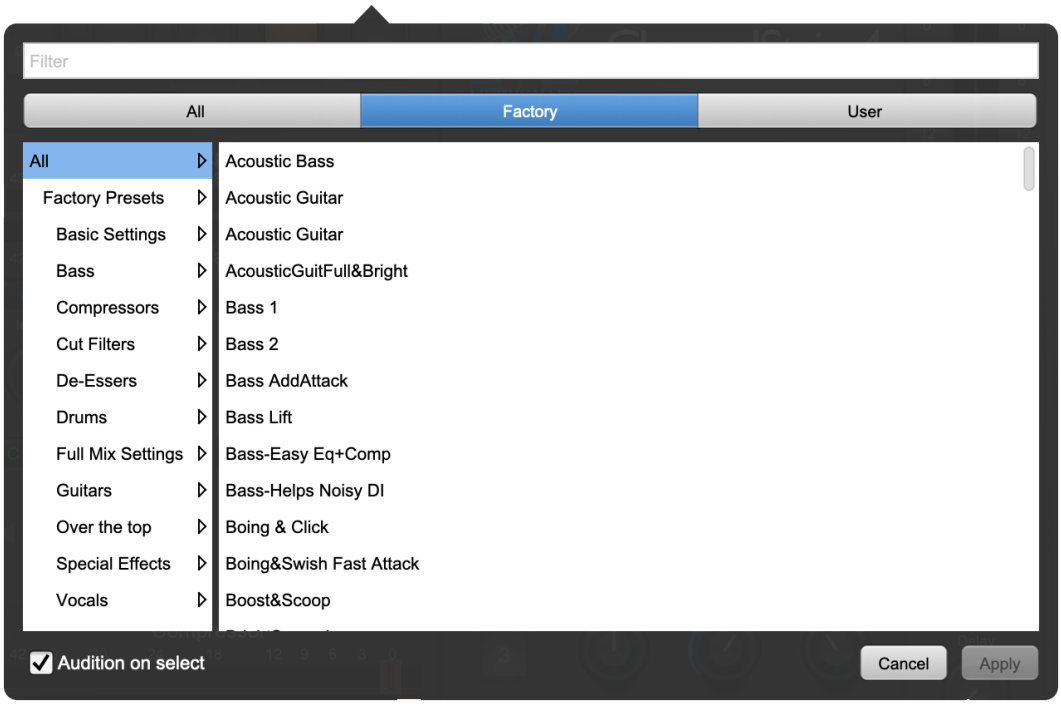
Preset Step-Through Buttons



Preset Step-Through Buttons

These buttons step through Factory and User Presets in succession, as they are listed in the Preset Selector window. The left arrow chooses the previous preset. The right arrow chooses the next preset.

Preset Name/selector menu:



Plug-in Header: Preset selector menu (ChannelStrip shown)

The Preset selector will open to show all the available preset categories, and the presets within those categories.



Preset selector menu: Audition on select

With "**Audition on select**" enabled at the bottom of the window, selecting a preset will temporarily load those parameters so you can hear the effect on the audio you are playing, without actually committing to the preset.

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.

Advanced Parameters



Advanced Parameters button

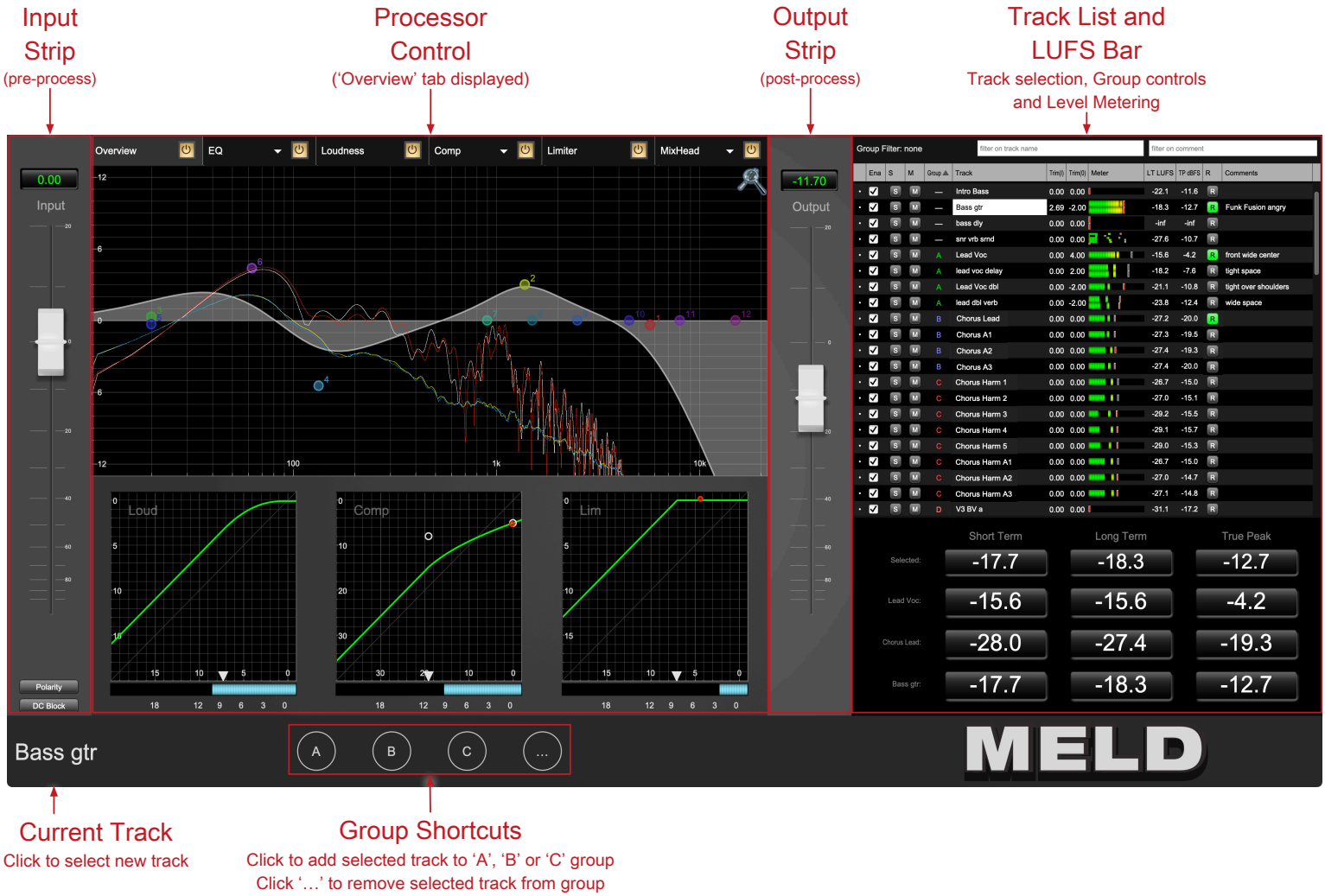
The gear icon at the far right of the Presets bar opens the MELD Advanced Parameter settings UI, containing the control to **Enable Shared Sidechain Support**.

See the Advanced Parameters section for details.

Operation

MELD User Interface

The primary MELD interface opens to the Overview display tab, with basic graphic views and controls for the primary EQ and dynamics processors.



The overall plug-in UI is arranged thusly (with details sections linked):

- Input Strip: Sets the incoming signal level before processing.
  - The Input Fader sets the input level for all Tracks in the group.
  - Input Polarity and DC Block filter switches are at the bottom of the strip.

- Processor Control: The main processing control area. The Tab Bar across the top serves several functions:
  - Click a tab to view the individual processor blocks or the Overview.
  - Each tab has enable/disable toggles for that block. The MixHead, EQ and Comp tabs also include preset loading from your MixHead and MH ChannelStrip preset libraries, respectively.
  - Click-drag the processor block tabs to re-order the signal processing chain.
- Output Strip: Sets the post-processed output level.
  - The Output Fader sets the output level for the group.
- The Track List displays operational controls and metering for all MELD-enabled DAW mixer tracks and buses, using the track names from the DAW mixer strips (for DAW hosts that support this function).

  The Track List is where you create, modify and navigate MELD groups. Currently you can assign up to 26 groups, labeled **A** through **Z**.

  There are several columns of information presented in the Track List, which can be auto-sized, or hidden by right-clicking in the column header bar and selecting from the pull-down menu.

  - The first column (with the blank header) is the Track Selector column, for selecting tracks for group assignment.
  - Ena enables/disables MELD processing per group (basically a bypass).
  - S and M operate MELD Group Solo and Mute functions.
  - The Track List Trim(I) and Trim(O) columns provide un-linked Input and Output Trim controls so you can easily view, modify and automate individual Track/instance trims independently from the rest of the group.
  - Each Track includes a Comments field for text notes.
  - Signal output Meters for each Track includes: PPM level with RMS, Long Term LUFS and True Peak dBFS. PPM meters show each channel within a Track (a 7.1.4 instance would show 12 active meter displays), while LUFS and TP dBFS numeric meters show the highest level of all channels within the track.
  - Reference tracks, selected in the **R** column, add separate Short and Long Term LUFS and True Peak readouts in the LUFS Bar, below the main Track List UI.
  - The Track List view can be filtered per Group, and search by text string is available for both Track Name and Comments content. The search filters are found above the Track List table.

    The Track List view can be sorted in ascending or descending order for most columns by clicking column headers.
- Current Track in the lower left corner of the plug-in window, shows the track name currently in view. Click on this track name (revealing the full list of all MELD tracks) to directly go to a new track. The main Track List will scroll to bring your newly selected track into view there as well.
- The **A, B, C, ...** Group Shortcut buttons below the processor block are for more efficient group management in Music, Dialog and Effects workflows.

Reminder: any DAW mixer strips which do not have MELD inserted can not be controlled by MELD. MELD instances can be added any time and (for most DAW hosts) in any insert slot - new instances will immediately appear in the Track List and MELD group processing settings will be applied automatically to a any instance added to a group.

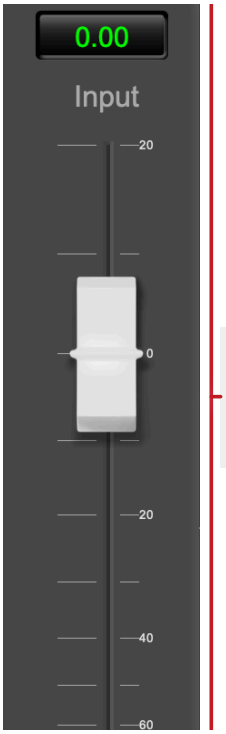
MELD Advanced Parameters

Important Note for MELD in MIO DSP: The Shared Sidechain Advanced Parameters feature requires that all the instances have access to the same memory buffers in order to get access to the audio data from the other instances in the sidechain, which is not possible with the MIO 3d multibox architecture. As a result, we cannot support that feature set within MIOConsole on the current hardware DSP.

Input Strip

Input Fader

The **Input Fader** is a group control, setting the input level to the processing block across the group.



Sets the input gain (in dB) applied to the signal before the gate. This gain is always applied unless the master bypass has been enabled. Click and drag to change. Option-Click or double-click to reset to default. Command-click and drag for fine-adjust mode. You may also use the scroll-wheel or scroll-gesture to adjust.

The Input Fader has a gain range of +20dB to -∞.

Polarity

Polarity is a group control, inverting the input signal polarity for all tracks in the group.



Enables polarity invert



polarity invert enabled

Polarity is inverted when the Polarity button glows yellow.

DC Block

The **DC Block** is a group control, applying a 1Hz cut filter reaching -∞ at 0Hz.



Enables DC Block filter (1Hz corner frequency)



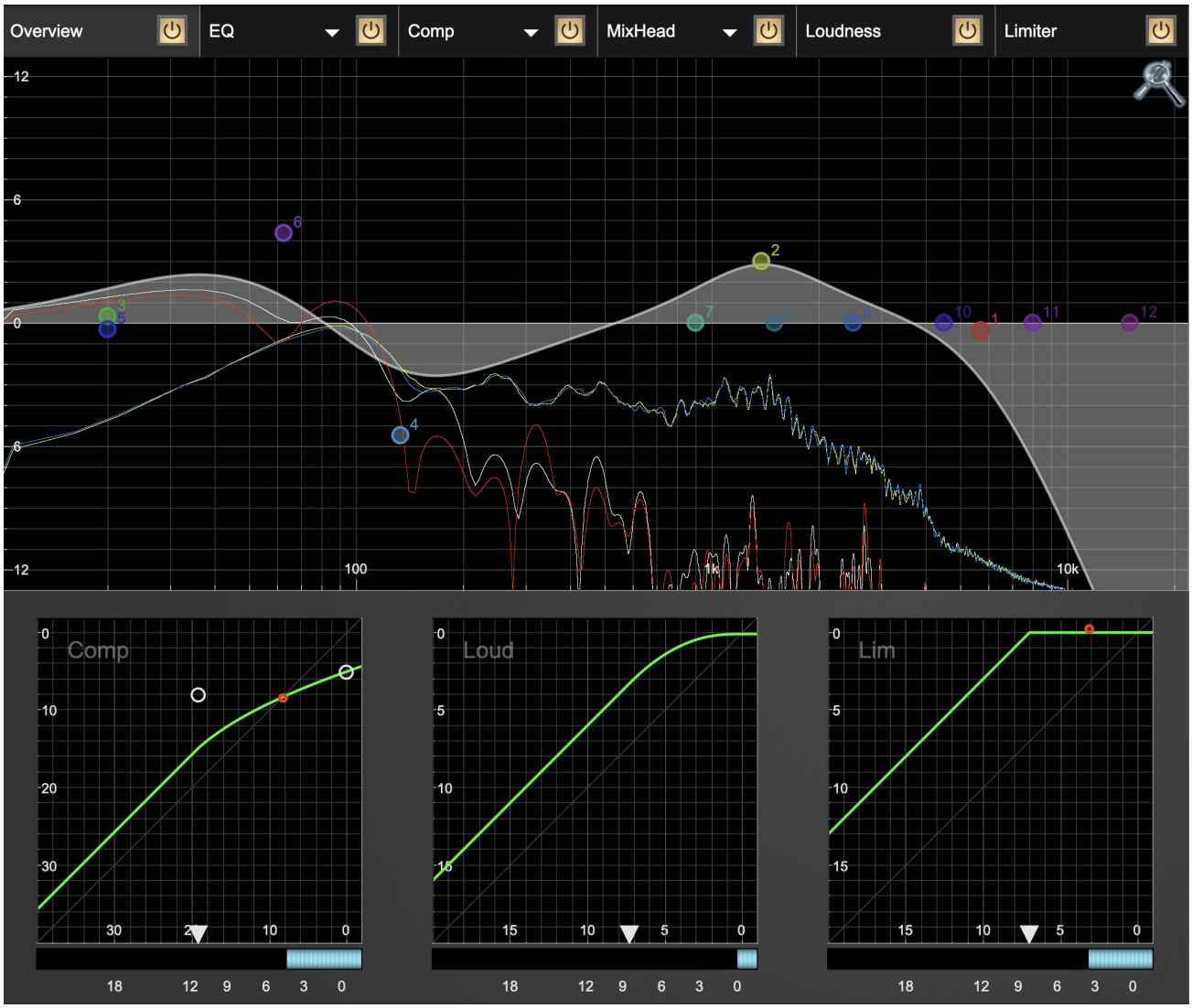
DC Block filter enabled

DC is blocked when the DC Block button glows yellow, and is applied after the Input gain/polarity stage.

Processor Control section

MELD is built around five world class signal processors: the EQ, Compressor and Limiter blocks from Metric Halo ChannelStrip, Make Believe Studios MixHead, and a new process designed for MELD called "Loudness".

Note: All parameter settings in this section are applied to all Tracks within the selected MELD Group.



Processor Block Tabs

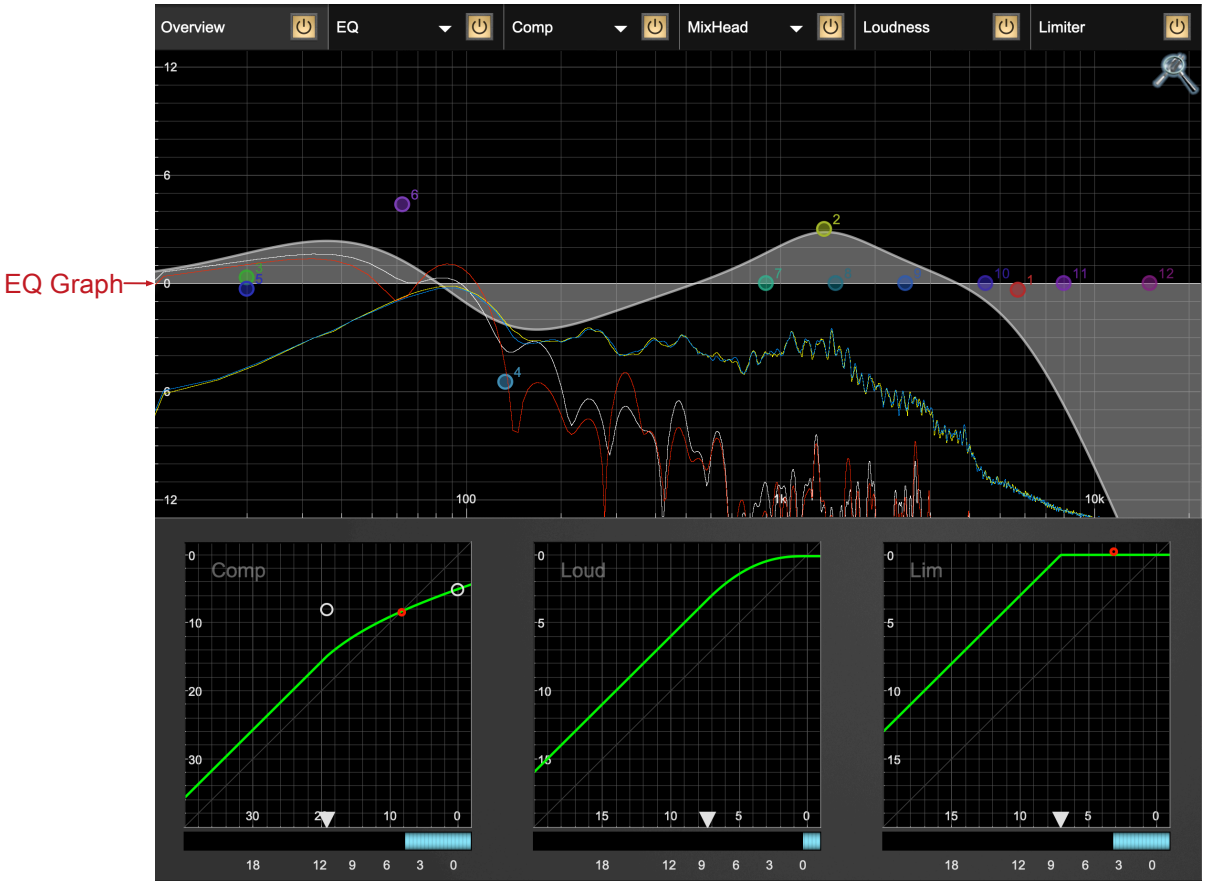


The six tabs of the Tab Bar are used to select the processor block view, engage / disengage each processor, and re-sequence the five processing blocks in the signal chain.

Click a tab to focus the view on the individual blocks.

Click-drag tabs to move them earlier or later in the signal process chain.

Processing “Overview” tab



The Overview tab is selected by default, providing core parameter controls for the four main process blocks - EQ, Comp, Loudness and Limiter.

The Overview is designed to be MELDs primary work space, allowing you to stay “in the zone” with as little switching between processor UIs as possible.

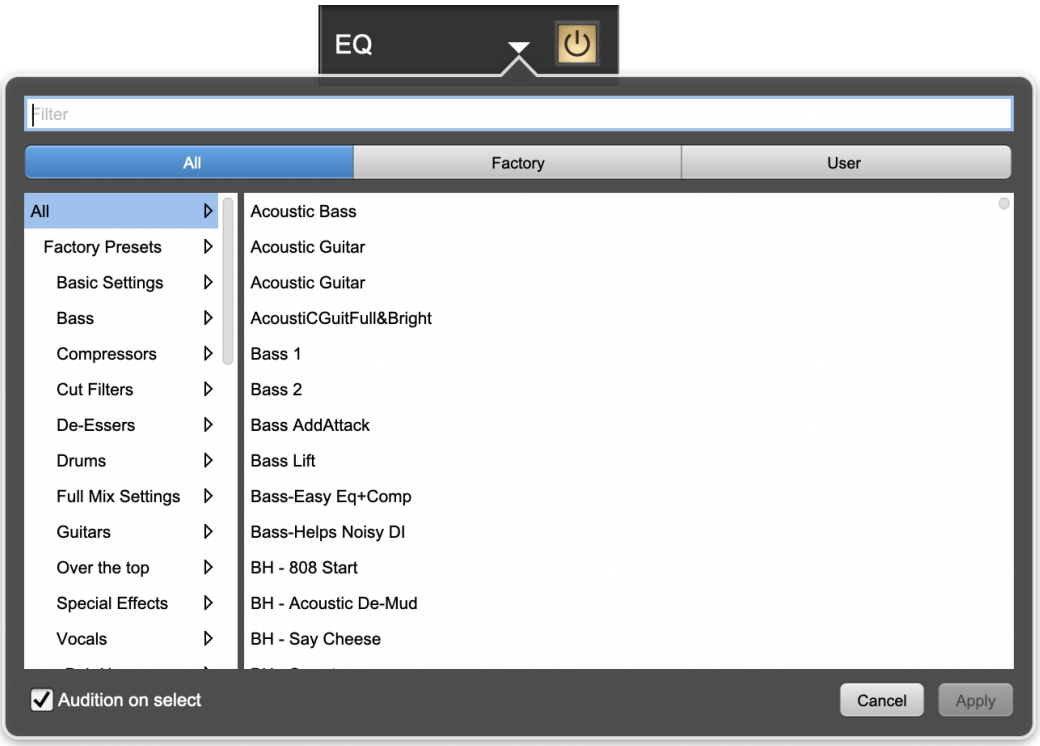
- The block Enable / Disable switch in the Overview tab toggles all MELD processing including the Trims. It is a double of the MELD plug-in header Bypass control.
  When lit, MELD is processing... when dark, MELD is bypassed. The other tab block Enable switches toggle the individual blocks.
- The Overview EQ graph includes all functions of the full EQ block view. See the EQ Block page for the full EQ control run-down.
- The Overview Compressor graph has controls for Threshold and Ratio (and Knee when in **MIO** compressor mode). See the Comp Block page for full Compressor details.
- The Loudness block has only a Threshold control. See the Loudness page for MELD Loudness processor details.
- The Limiter also has just a Threshold control. See the Limiter page for MELD Limiter details.
- In MELD dynamics graphs, the white ∇ is Threshold. O controls Ratio (and Knee) in the Comp.

EQ

EQ Block Tab

Select the EQ Tab to bring the EQ processor block UI into focus.
The EQ Preset and EQ Block Enable controls are available whether or not the EQ Tab is selected.

EQ Tab Preset menu



The MELD EQ Block is compatible with MH ChannelStrip EQ presets, loading the EQ band parameters into the first six bands of the 12-band MELD EQ.
The disclosure triangle in the EQ tab opens the Preset Selector UI, where you can select from any MH ChannelStrip Factory or User EQ preset installed on your system.

Loading a ChannelStrip preset from the EQ tab loads only the EQ settings from the preset.

See MELD Preset Menu section for detailed information on the preset menu.

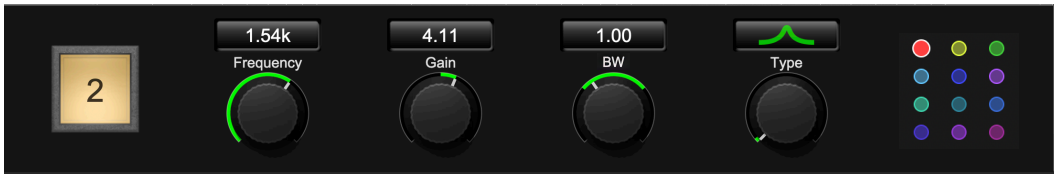
EQ Block Enable button



The MELD EQ block can be enabled or bypassed by toggling the EQ Block Enable button. The EQ is disabled when the button is dark (left), and is enabled when the button is lit (right).

EQ Filter Band controls

The EQ Filter Band control strip appears below the EQ Transfer Function graph.



EQ Band Enable Button

Use this toggle button to enable the filter band. When the filter band is turned off (dark grey) the signal will pass through the filter unchanged.

Filter Band Frequency

Use this knob 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 Band Boost/Cut Control

Use this knob (labeled "Gain" 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 Bandwidth

Use this knob (labeled **BW** above) 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).

Filter Type Button

This button (indicating a peaking/parametric filter in the illustration above) is used to select the filter type of the EQ band. You may choose from 6 different types of filters:

- 

  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.

- 

  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.

- 

  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.

- 

  High Cut – a 12 dB/octave high cut filter with a -3dB point that is continuously adjustable between 20 Hz and 20 kHz.

- 

  Low Cut – a 12 dB/octave low cut filter with a -3dB point that is continuously adjustable between 20 Hz and 20 kHz.

- 

  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.

EQ Band Selector

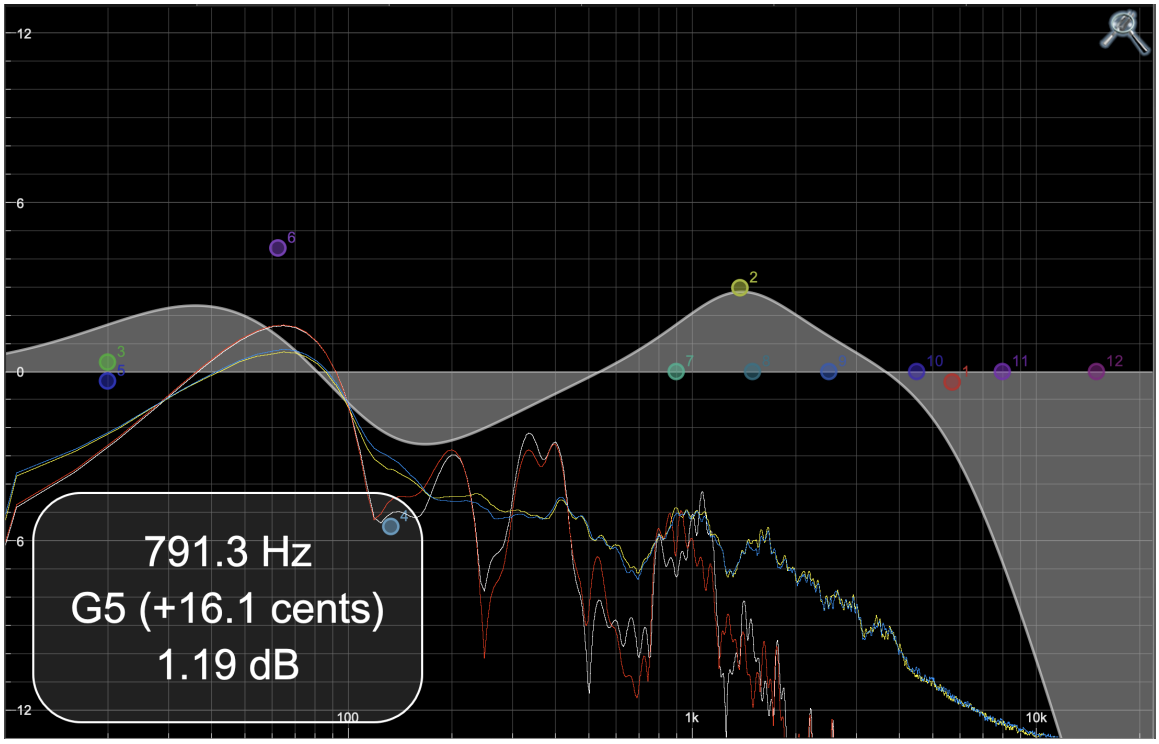


The twelve colored buttons at the right of the EQ control bar correspond to the EQ fiter band colors in the transfer function graph. Click a color to bring that filter into focus of the EQ cinbtrol knobs.

The selected EQ band will highlight in the EQ Transfer Function graph above, showing the effect of that band within the composite EQ curve.

EQ band colors can be edited in the MELD UI Preferences tab opened by clicking the Metric Halo logo in the upper left corner of the MELD header.

EQ Transfer Function



EQ Transfer Function

The horizontal axis provides frequency calibration in Hertz (Hz). The vertical axis provides level calibration in decibels (dBr). The heavy line (white by default) 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 graph.

The dot represents the center frequency or knee of the filter band, and acts as a control handle for the band within the transfer function field.

The band that is currently being edited will be highlighted and displayed along with the overall response curve, making it easy to see how each filter is affecting the overall 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 bandwidth of the filter.

Clicking on a band handle 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.

In the transfer function field, multiple filter bands may be temporarily grouped by shift-click or click-drag selection of the filter control handles.

To dismiss the filter curve, click anywhere in the black area of the transfer function. This will deselect the filter, and the only trace displayed will be the master EQ curve.

Reminder: clicking the **?** "Help" icon in the MELD plug-in header will activate tooltips for the various UI elements. The tooltip for the EQ transfer function is especially handy:

Displays the transfer function of the 6-band EQ. Click the SpectraFoo icon to enable spectrum analysis of the output of MELD.

Click and drag any of the control handles to change the associated EQ band's frequency and gain.

Click and drag the smaller control handles of a selected band to adjust the band's bandwidth.

Command-click a control handle to toggle the band's enable.

Option-click and drag a control handle to adjust the band's bandwidth.

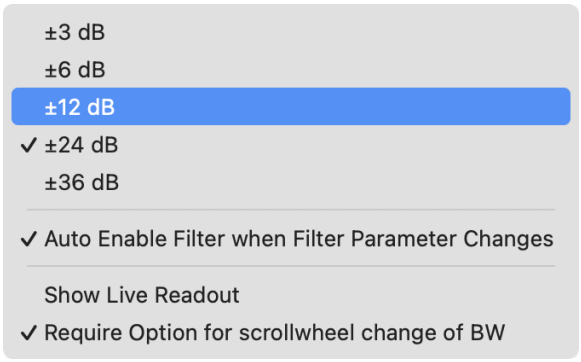
Command-Option-click a control handle to cycle through the possible band types.

Control-click or Right-click in the transfer function area to popup a menu to select the vertical range of the transfer function display.

EQ Transfer Funciton tooltip

Transfer Function options

Right-clicking the EQ transfer function background brings up an EQ functions control menu.

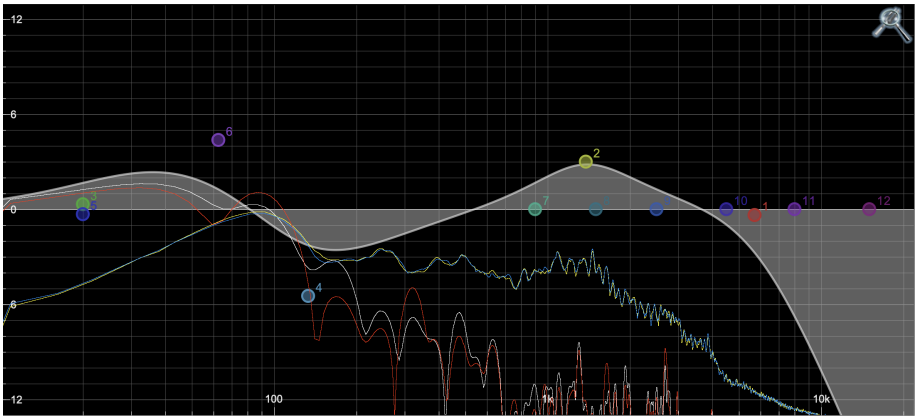


- EQ display scale: Sets the EQ curve dB scale from ±3 to ±36 dB.
- **Auto Enable Bands when band parameter changes**: Automatically enables a filter band when one of its parameters is adjusted within the EQ Transfer Function. This preference will apply to all instances of MELD.

  Note that the Auto-Enable applies only to the filter band enables, and does not affect the master plug-in Bypass state.
- **Show Live Readout** (shown on the previous page): Enables an overlay in the lower left corner of the transfer function which displays the relative amplitude of the master EQ curve at the frequency indicated by your mouse cursor. Moving the cursor off the EQ display hides the readout. Live Readout control is enabled per MELD instance.
- **Require Option for scrollwheel change of BW**: Enabled by default, this control requires the Option modifier key to change the BW parameter using a mouse scrollwheel. Scrollwheel filter width control works for filter band knobs (one at a time) and selected filters in the transfer function display (multiple selections supported).

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 overlaid on the Transfer Function EQ curve:



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 analyzes the signal post-filter, allowing you to see the effect of your EQ filters.

To disable the spectragraph entirely, click the active Spectrafoo icon.

If you right-click or control-click on the Spectrafoo icon, 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.

These settings are stored independently for each instance of MELD.

Controls for Left and Right channel will only be available in stereo instances of MELD.

Default view shows the Left channel only.

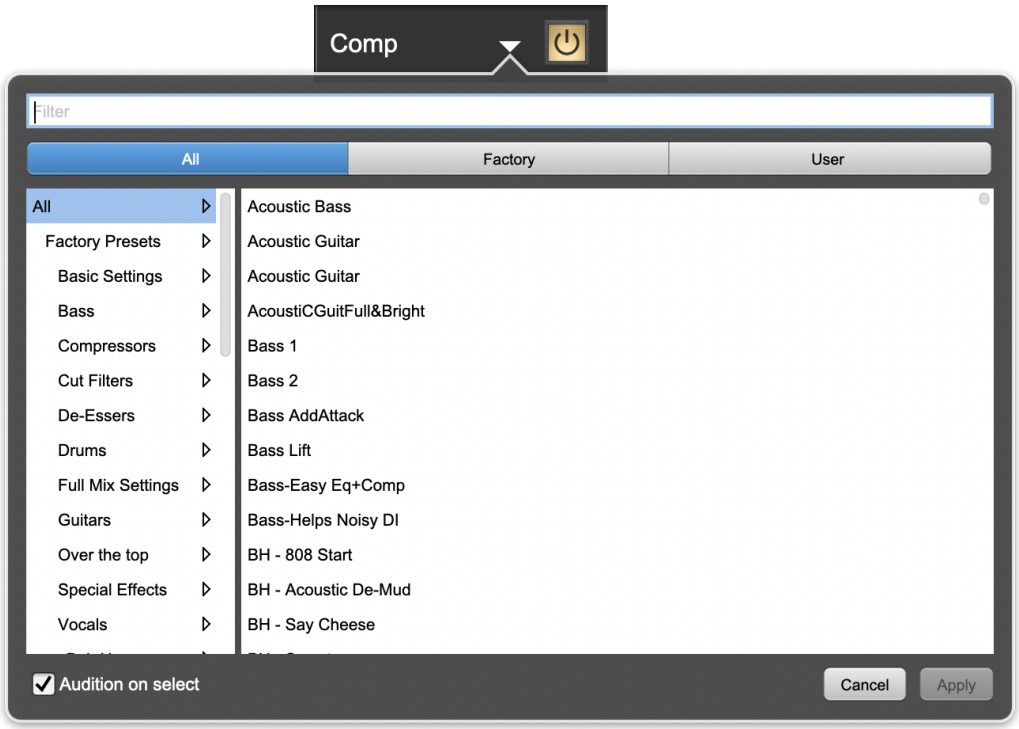
Compressor

Compressor Block Tab

Selecting the **Comp** tab brings the MELD Compressor block into focus.

Note that the Comp Preset and Comp Block Enable controls are available whether or not the Comp Tab is selected.

Comp Tab Preset menu



The MELD Compressor Block is compatible with MH ChannelStrip compressor presets.
The disclosure triangle in the Comp tab opens the Preset Selector UI, where you can select from any MH ChannelStrip Factory or User preset installed on your system.

Loading a ChannelStrip preset from the Comp tab loads only the compressor settings from the preset.

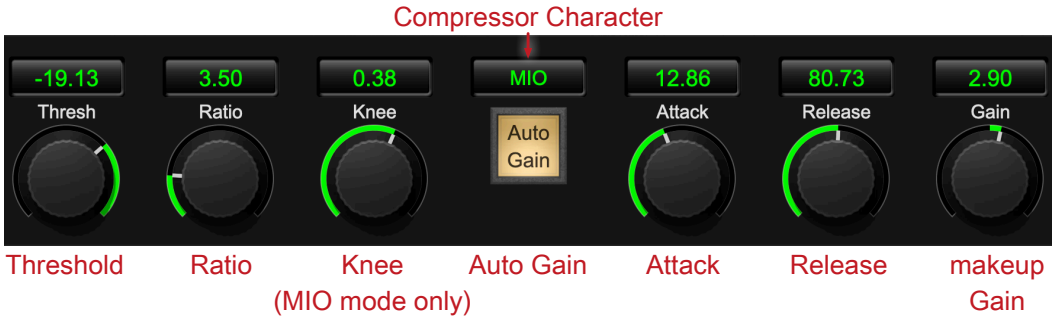
See MELD Preset Menu section for detailed information on the preset menu.

Comp Block Enable button



The MELD Compresssor block can be enabled or bypassed by toggling the Comp Block Enable button. The Compressor is disabled when the button is dark (left), and is enabled when the button is lit (right).

Compressor Controls



Threshold Control

The **Threshold** 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.

Note: In the Compressor Graph, Threshold is controlled by moving the white ∇ handle across the bottom of the graph.

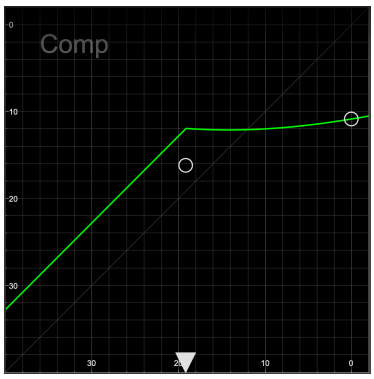
Ratio Control

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 tube limiter/compressor.

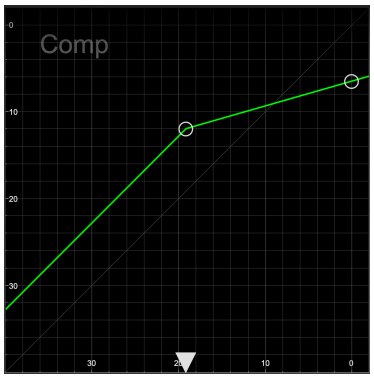
Note: In the Compressor Graph, Ratio is controlled by moving the white ○ handle at the right edge of the graph down from the top.

Knee Control (MIO mode only)

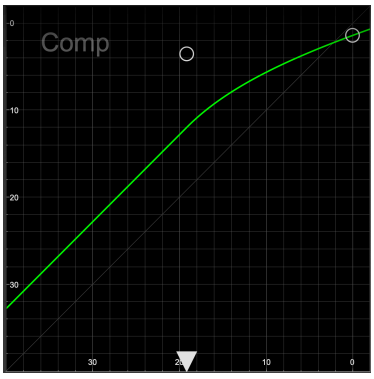
The **Knee** knob allows you to adjust shape of the compressor when the Compressor Character is set to **MIO**. The Knee control has no effect for the other compressor character algorithms and will not be available in those modes.



Knee = -0.5



Knee = 0



Knee = 1

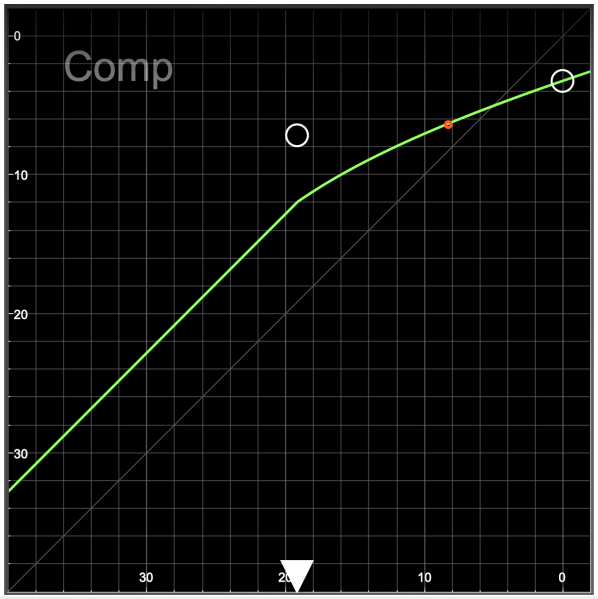
When the Knee control is set to the default 0, the transfer function of the compressor is a classic "hard-knee" in which the compressor applies no gain reduction when the detector level is below the threshold, and the gain is reduced by the ratio when the detector level is above the threshold.

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 by the time you get to 1.

You can also adjust the Knee parameter to negative values to a max of -0.5, which has the effect introducing a "kink" in the compressor transfer function at the threshold. This is definitely worth experimenting with, and often yields useful results on percussive material and FX tracks.

Note: In the Compressor Graph, the Knee is controlled by moving the white ○ handle which is always directly above the Threshold ∇ handle.

Compressor Graph



Shows the Compressor transfer function knee graph. The red dot indicates the instantaneous output level of the compressor as a function of the input level.

MELD provides a Dynamics Knee graph for the compressor processing block. 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 green line shows the static gain response curve set by the Threshold, Ratio and Knee controls. This curve is generally 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 exactly match that set gain response curve. In order to represent this, we have included a "bouncing ball" meter for the compressor. This *Dynamic Knee Indicator* is shown as a red circle that is overlaid on the graph. 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.

Gain Reduction Meter



Compressor Gain Reduction Meter. Shows how much gain reduction is being applied by the Compressor. Control-click or Right-click to popup a menu to set the display range of the Meter.

The gain reduction meter grows down from 0dB, showing the amount of attenuation being applied by the compressor. If you right-click on the meter, you may set the scale of the meter to ranges of 0dB to 54dB, 24dB, 12dB, 6dB, or 3dB

Compressor Character



Compressor Character Switch options

Use the compressor character button to determine the overall dynamic characteristics of the compressor. There are three "classic" ChannelStrip 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** – is the most versatile basic compressor mode. 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.
- **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 which can be used to adjust the compressor response from hard to soft, or into special effect territory.

  With the Compressor in MIO mode, the attack and release are applied to the envelope detector, and the gain reduction is driven directly from the detector. This is why, in MIO mode, the detector Dynamic Knee Indicator response visually slows down following the Attack and Release controls.

  In general, the MIO algorithm is more flexible and controllable than the other algorithms, and as a result, we have made it the default compressor option for MELD.

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 output **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 output 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.

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.

Manual MakeUp Gain

The **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 becomes a trim added to the internally computed makeup gain.

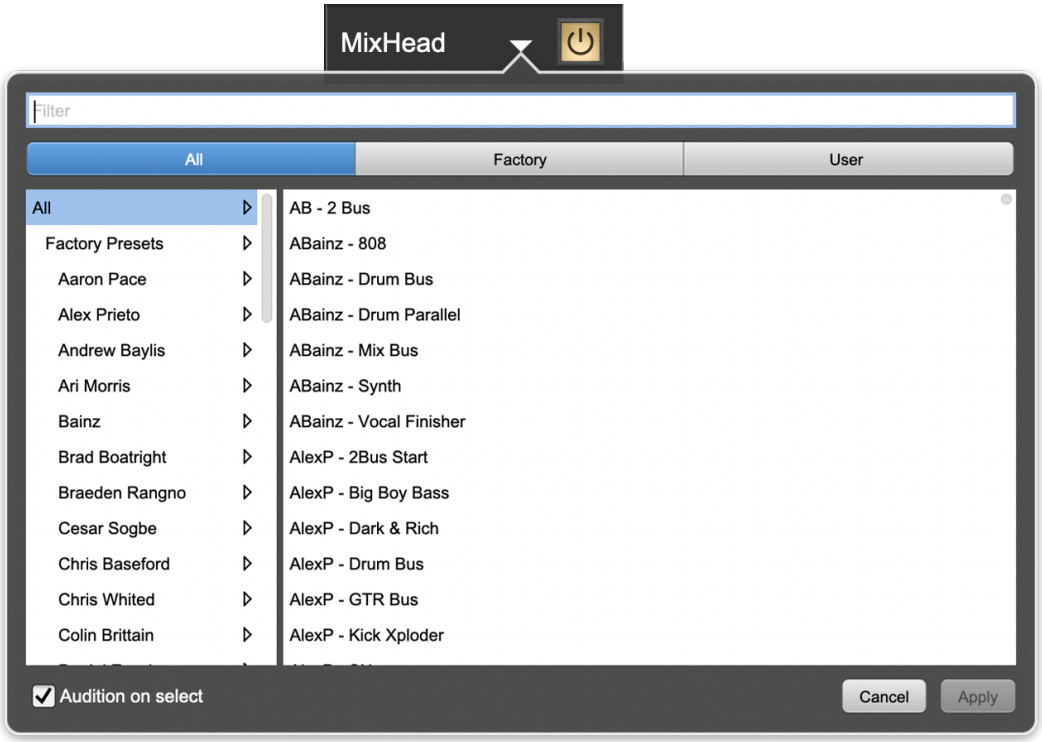
MixHead

MixHead Block Tab

Selecting the **MixHead** tab brings the MELD MixHead block into focus.

Note that the MixHead Preset and MixHead Block Enable controls are available whether or not the MixHead Tab is selected.

MixHead Tab Preset menu



The MELD MixHead Block is compatible with standard MB MixHead plug-in presets.
The disclosure triangle in the MixHead tab opens the Preset Selector UI, where you can select from any MB MixHead Factory or User preset installed on your system.

See MELD Preset Menu section for detailed information on the preset menu.

MixHead Block Enable button



MixHead can be enabled or bypassed by toggling the MixHead Block Enable button. MixHead is disabled when the button is dark (left), and is enabled when the button is lit (right).

MixHead Controls

There a five primary parameters to control the MixHead processor.

Input Gain

Input Gain adjusts the pre-process input level, not unlike the line in calibration on a tape deck. Input Gain adjusts input level in 0.1dB steps, with values spanning from -12.0dB to +12.0dB.



Input Gain (dB) Knob. Click-Drag to change. Control-Shift-drag to apply inverse change to Output Gain. Alt-click to reset. Control-Alt-click to set to minimum. Control-click (or Right Click) for text entry.

Input Gain interacts greatly with the Drive stage and Output level.

Tip: **Control-Shift-drag** Input Gain to inverse link with the Output Gain control, to maintain your current gain structure through the processor while shifting the reference point at which the Drive circuit operates.

Drive

Drive adjusts the amount of harmonic distortion and saturation effect. The saturation effect is non-linear with regard to gain, and often requires some degree of Output level compensation.



Drive Gain (dB) Knob. Click-Drag to change. Alt-click to reset. Control-Alt-click to set to minimum. Control-click (or Right Click) for text entry.

The Drive parameter has a range of -7 to +14. The default Drive setting of 0.0 is intended to emulate a record level of +7dB to tape, so the saturation effect is already fairly well engaged at 0.0 Drive. More subtle tape emulation characteristics with higher headroom and less dynamic saturation are available at Drive levels below 0, where the Input and Output levels tend to have more relative effect.

Tip: High Drive settings can result in serious signal breakup, especially with modern day deep low frequency program. That said, MixHead works brilliantly as a high-passed parallel processor even on deeply bass-heavy master tracks, bringing massive saturation crunch to the mids and a silky sheen to the highs without stepping on the bass foundation.

HF-Adjust

HF-Adjust allows you to control the amount of high frequency cut or boost independent of the Drive setting. The control range is -6 (max. cut) to +6 (max boost).



High-Frequency Adjust Gain (dB) Knob. Click-Drag to change. Alt-click to reset. Control-Alt-click to set to minimum. Control-click (or Right Click) for text entry.

With real tape, the higher the saturation the more high frequency damping takes place. MixHead lets you break that rule by running high frequency energy levels contrary to what you would otherwise expect from tape. The effect can be used to simulate differences in tape formulations or playback electronics or different bias/EQ setups.

Tip: Unlike pretty much anything with a real tape deck, HF-Adjust is automatable. It is especially well suited to pushing instruments back in the mix or bringing them dramatically forward as the song demands. At low saturation levels a bit of HF-Adjust boost acts very much like a vintage exciter.

Output

Output Gain adjusts output level post-processing. Output Gain adjusts output level in 0.1dB steps, with values spanning from -12.0dB to +12.0dB.

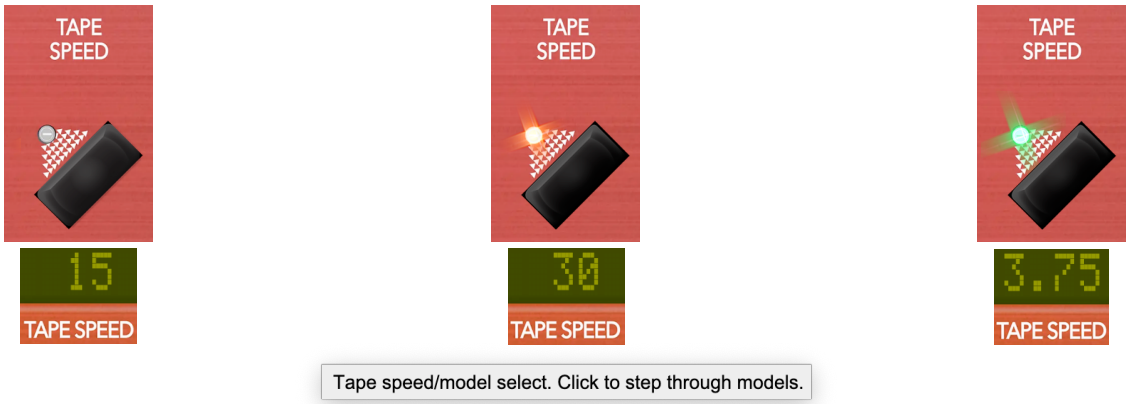


Output Gain (dB) Knob. Click-Drag to change. Control-Shift-drag to apply inverse change to Input Gain. Alt-click to reset. Control-Alt-click to set to minimum. Control-click (or Right Click) for text entry.

Generally Output Gain is used as a final compensation for the changes in perceived loudness from the Drive stage saturation effect, and to ensure that the output of MixHead does not clip the input of the next process in the signal chain.

Tape Speed

Tape Speed toggles between MixHeads three tape speed emulation algorithms.



- **15ips** (the default): emulates the general distortion and saturation characteristics associated with tape recording at 15ips, with an unexpected subtle stereo widening effect.

  In 15ips mode, the Tape Speed switch light is dark.

- **30ips**: emulates the generally lower distortion and higher saturation headroom curve associated with tape recording at 30ips.

  Note that the 40Hz to 70Hz bass reduction induced when recording to analog tape at 30ips is not reproduced.

  In 30ips mode, the Tape Speed switch light is lit amber.

- **3.75ips**: A new addition to MixHead not available on the original hardware, 3.75 was inspired by the modeling of a 1950's Webcor tape machine. The main characteristics that were borrowed for the 3.75 machine are its headbump curve and the amount of distortion it can generate. In true MixHead fashion we utilized the Mixheads' unique approach to compression, limiting and hf adjust as the other models.

  In 3.75ips mode, the Tape Speed switch light is lit green.

Meters

There are three sets of stereo meters, displaying signal levels after the Input and Output gain stages respectively, and the "virtual tape machine record level" of the Drive stage.

Note: in mono plug-in instances, both channels of the meter pairs will show signal activity.



- The **Input PPM** meters show the peak level after the Input Gain control. They are PPM response from -30dBFS to 0dBFS, with a 1 second peak hold.
- The **Drive Level** meter indicates the amount of harmonic saturation being applied at the Drive stage - basically how hard it's working. The segments display a range from -10db to +21dB.
- The **Output PPM** meters show the peak level after the Output Gain control. They are PPM response from -30dBFS to 0dBFS with a 1 second peak hold.

Important Notes

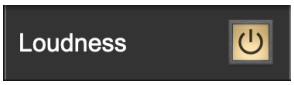
Since MELD MixHead resides as a process block within MELD, the Advanced Parameters of the standalone MixHead which were designed to improve compatibility across various workflows are not included. MELD MixHead operates as follows:

- The topmost segment of each meter set indicates consecutive samples into digital clipping. In MELD MixHead, it does not change color as a warning.
- Since it is always feeding another high-resolution processor stage, MELD MixHead does not hard clip its output, allowing greater-than-0dBFS data to pass unaltered to the next stage in the signal chain.

And finally, as on the standalone version, the bottom segments of both the Input and Output meters are always lit to emulate the original hardware indication of a clock-synced and legal AES audio input stream.

Loudness

Loudness Block Tab

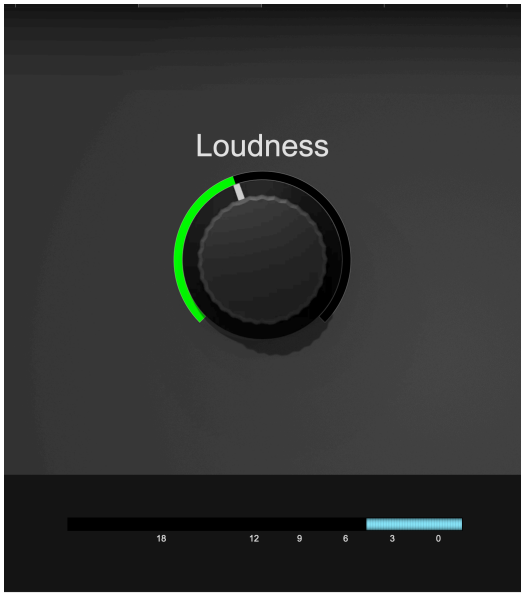


Selecting the **Loudness** tab brings the MELD Loudness block into focus.



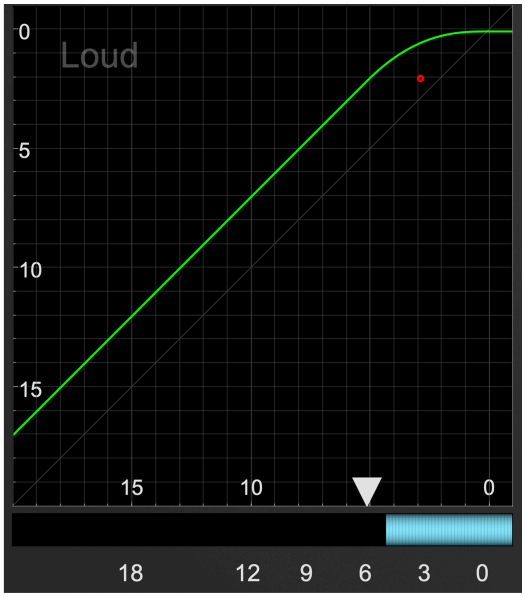
The MELD Loudness block can be enabled or bypassed by toggling the Loudness Block Enable button. Loudness is disabled when the button is dark (left), and is enabled when the button is lit (right).

Loudness Control and Meter



Sets the threshold of the Loudness processor, in dB, from 0dB to -12dB. Higher settings increase the perceived loudness of the program.

Click and drag to change. Option-click or double-click to reset to default. Command-click and drag for fine-adjust mode.
You may also use the scroll wheel or scroll wheel gesture to adjust.



Loudness transfer function knee graph.

The red dot indicates the instantaneous output level of the compressor as a function of the input level.
Threshold control is the white triangle at the bottom, and uses the same control gestures as the knob UI shown to the left.

MELD Loudness increases the average signal level (and therefore the apparent overall loudness) while leaving the peak program alone.
The Threshold control determines how far below peak dynamic range Loudness reaches for program material to reinforce.

The result is naturally very dependent on the source material, but it is especially effective in smoothing out vocal and acoustic instrument performances by bringing up quieter moments in a performance to match the more energetic passages. Things like a singer momentarily turning away from a mic, or differences in performance intensity between overdub takes are much less apparent with Loudness in the chain.

Loudness is tuned to be most effective within the LUFS target range for mixing immersive music, broadcast and post production program.

Limiter

Limiter Block Tab

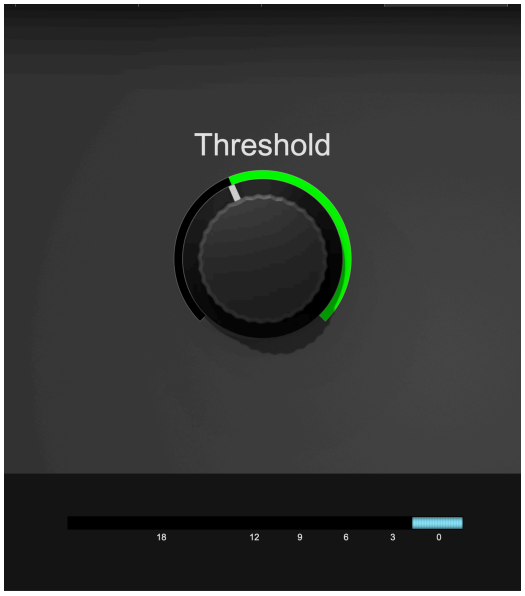


Selecting the **Loudness** tab brings the MELD Limiter block into focus.



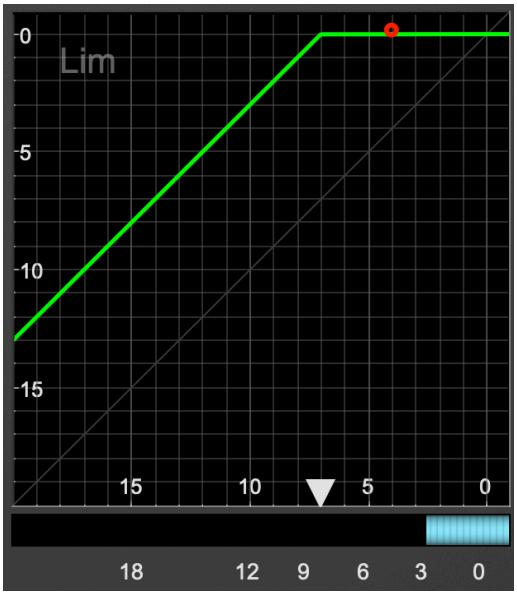
The MELD Limiter block can be enabled or bypassed by toggling the Limiter Block Enable button. The Limiter is disabled when the button is dark (left), and is enabled when the button is lit (right).

Limiter Control and Meter



Sets the threshold level where the limiter will begin to limit the output signal. The limiter will automatically apply makeup gain so the maximum output level is at 0dB if the input level goes above the threshold.

Click-drag to change. Option-click to reset to default. Command-click and drag for fine-adjust mode. You may also use the scroll wheel or scroll-wheel gesture to adjust



Limiter transfer function knee graph.

The red dot indicates the instantaneous output level of the limiter as a function of the input level.
Threshold control is the white triangle at the bottom, and uses the same control gestures as the knob UI shown to the left.

The Limiter threshold may be set between 0 and -12 dB using the Limiter Threshold knob.

Gain reduction is displayed in the gain reduction meter below, sweeping from right to left.

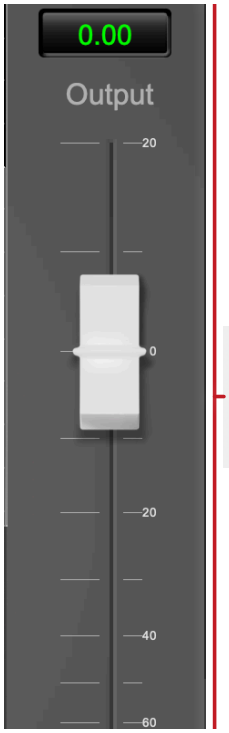
The limiter applies autogain to the audio when engaged; as the threshold is lowered, the output gain is increased a complementary amount.

The MELD Limiter action is identical to MH ChannelStrip, but the MELD Limiter display includes dedicated gain reduction meters rather than using the knob ring, and has a dynamics knee graph available in Overview mode which displays the dynamics curve and the instantaneous signal output level.

Output Strip

Output Fader

The **Output Fader** sets the processed MELD group output level.



Sets the gain (in dB) applied to the signal after the last process block. This gain is always applied unless the master bypass has been enabled. Click and drag to change. Option-Click or double-click to reset to default. Command-click and drag for fine-adjust mode. You may also use the scroll-wheel or scroll-gesture to adjust.

The Output Fader has a gain range of +20dB to -∞.

Track List / Groups control

The Track List displays every MELD instance in the host DAW session, with the Track Name of each instance provided by the host DAW per the AAX, AU or VST specifications.

MELD Track List name fields are read-only to ensure accurate location of each MELD instance in the host DAW mixer. Track name edits in the host DAW are immediately reflected in the Track List.
Note: If the Track name field remains empty after instantiating MELD, that just means your particular DAW does not publish the track name to the plug-ins. Please file a support ticket with your DAW publisher requesting the feature.

MELD can be instantiated on mono, stereo or multichannel DAW tracks. Multichannel and multi-mono instances of MELD are displayed as a single track with multichannel level meters. For example, the top track in the graphic below is L-C-R-Ls-Rs-LFE 5.1.

The Track List includes Long Term LUFS and True Peak hold metering for every track in your session with a MELD instance.

Note: MELD instances can be placed in any available track insert slot, allowing full freedom to apply additional plug-ins before or after MELD. There is also no restriction on placing multiple instances of MELD in a single track. Most host DAWs provide the track name to the plug-ins, but do not provide either the slot number. As a result, if you insert more than one MELD plugin instance on the same track, you will get multiple instances with the same track name showing up in the Track List. If you go this route (which John Hanes mentioned in his introduction), we recommend you use the Comment field to identify which instance is which.



Track List example

The sections that follow will detail the functions of each column in the MELD Track List.

Please Note: Unless otherwise stated, all Track List group functions operate the same for single unassigned tracks as they do for track groups.

Track List interface

The Track List provides enough information and control to run the majority of a mix without ever leaving the MELD interface. The vertical columns of the list provide an array of control and metering options for both groups and individual tracks.

Track List Search Filters

One of the most impactful aspects of MELD is its ability to focus on and directly compare any set of tracks' I/O trim settings and output levels.

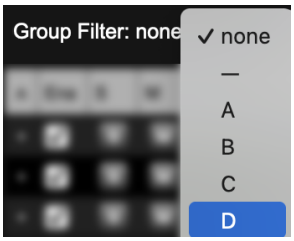
The Track List interface provides a variety of search and sorting functions to make creating and navigating high track count sessions straightforward and efficient.

Across the top of the List UI is the Search section.



Group Filter

The Group Filter sets the list view to show all groups, a single group, or all the tracks not assigned to any group. Click the Group Filter button to open the pull-down menu selector.



The filter selections are:

- **none** (no filter): shows all MELD tracks
- **-** (no group): shows only tracks not assigned to a group
- **A through Z**: shows the tracks of the selected group (A - D shown)

The check mark in the menu shows the current filter setting. Items will highlight blue as you move through the list. Click an item to select.

Track Name Filter

Type a set of characters in the **filter on track name** field to show only tracks with that text string in the track name.

Comment Filter

Type a set of characters in the **filter on comment** field to show only tracks with that text string in the Comments field.

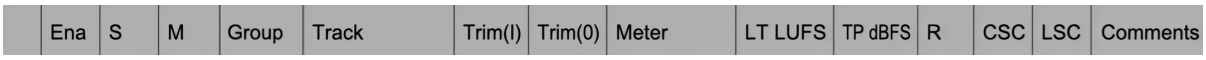
Note: Both Track Name and Comment filters are dynamic and will sort as you type.
Note also: Both search filters work even when the Track or Comment columns are hidden.

Column arrangement

The order of the columns can be changed freely by dragging any column name in the header to the left or right. The new column arrangement will apply to all MELD instances, and will be stored and recalled with the saved session.

Sorting

Click a column header to sort the tracks by that columns contents. A sorting arrow will appear to show the column is selected and is sorting the tracks below. Click again to toggle the sort order between ▲ *ascending* (lowest value at the top) or ▼ *descending* (highest value at the top).



Sorting the Track List by the various header criteria is a powerful time-saver for navigating MELD.

- The **{blank header}** column at the far left is the track selector: Click to sort tracks by the host DAW mixer track order at the time the session was launched.

  Note: host DAWs do not transmit track mixer position to plug-ins, so MELD can only derive track position by reading the order each individual MELD instance is loaded when opening a saved session. Generally, DAWs instantiate tracks starting at 1 and progressing to 2, 3 and so on, however...

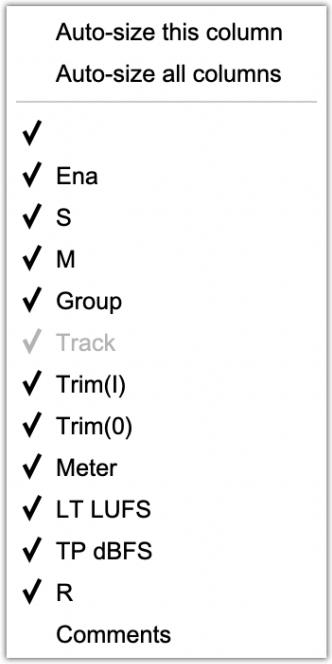
  Note also that some DAWs do not launch plug-ins in a linear per-track sequence (Logic is one...). Additionally, using track folders to organize the host mixer (almost a given these days) may also affect the track order. In these cases, the only way to correctly sort the DAW track sequence in MELD is to name the DAW mixer tracks alphanumerically for their position in the mixer, such as 01-, 02-, 03-, and so on, then sort by Track Name.

- **Ena** Group Enable: Click to sort between enabled and disabled tracks.
- **S**: Click to sort between Solo'd and non-Solo'd tracks.
- **M**: Click to sort between Muted and non-Muted tracks.
- **Group**: Click to sort alphabetically by Group letter. Ungrouped tracks are listed with the **-** symbol before group A (like **0** (zero) is numerically listed before **1**).
- **Track**: Click to sort alphabetically by Track name.
- **LT LUFS**: Click to sort by Long Term LUFS level.
- **TP dBFS**: Click to sort by True Peak dBFS level.
- **Comments**: Click to sort by Comments in alphabetical order.

Sorting is not available for **Trim**, **Meter**, **R**eference, or the advanced mode **CSC** and **LSC** columns.

Column Auto-size and Visibility

Right-click any column header other than **Meter** to open the columns control menu.



- **Column Auto-size**: Click to auto-size all columns, or just your selected column.
- **Show/Hide**: Checked items are visible, unchecked items are hidden. This control affects visiblility only, the column functions are still active when hidden.

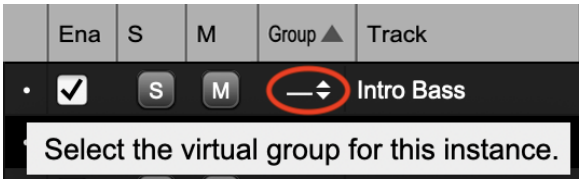
  In the graphic above, the **Comments** column is unchecked and hidden.

  Track is dimmed above because the it is the currently selected column (i.e. the Show/Hide menu was opened by right-clicking the **Track** header, and you can't hide the object that is displaying the current menu).

Creating MELD virtual groups

MELD is pretty intuitive just by clicking around and seeing what happens, but let's get straight to the most important thing - making virtual track groups.

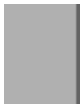
To assign a single instance to a virtual group, click in the Group column of that track to reveal the Group selector menu, then click a group from the list A through Z.



Group selector tooltip shown on an unassigned track

Selecting multiple tracks for group assignment

The column with the **{blank header}** (farthest left by default) is for selecting multiple tracks to be assigned to MELD groups:



Track Selector Header

Clicking on track selector boxes readys tracks for assignment to a group - green means selected.

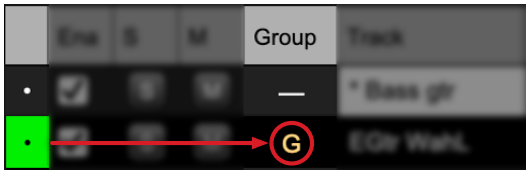
Selecting a group in the Group column will assign all selected tracks (including any not in the current window view) to that group.



Track not selected. Click to select. Shift-click or click-drag to select a range. Command-click or command-drag to select non-contiguous tracks.

Track selected and ready to be grouped. Command-click to de-select. Option-click anywhere within the column to clear all selections.

To assign selected tracks to a Group:



Click the Group Assignment button for any selected Track, and select the desired Group from the pop-up menu. All selected Tracks will be assigned to the new group.

Tracks can be removed from groups and/or re-assigned to other groups at will.

When a track is added to a group, the group processor parameters are immediately applied the track.

Keep in mind that MELD virtual groups have no relation to any type of group in the host DAW mixer. MELD instances can only communicate with other MELD instances within a session. As far as the host DAW is concerned, MELD is just another plug-in instance on a track: all of the parameter links, metering and sidechain linking happen between MELD instances directly.

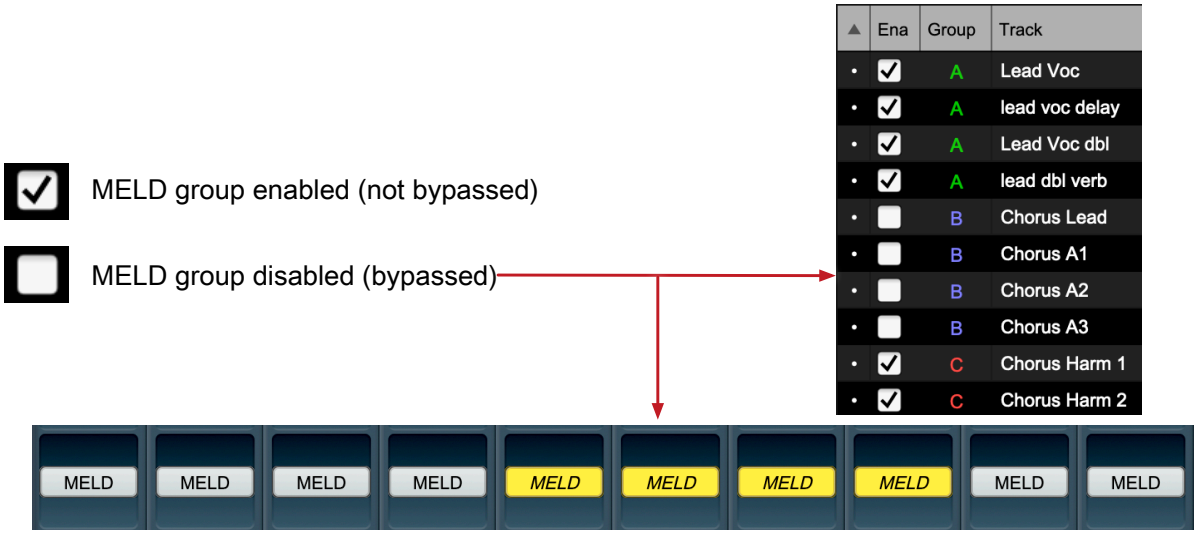
Track Enable / Bypass



Enable column header

The **Ena**ble checkbox column enables MELD processing per group. Unchecked instances are bypassed.

Click the checkbox for any track in the group to MELD processing in or out for the entire group.



Disabled MELD tracks become bypassed MELD instances in the plug-in host mixer.

Solo



Solo column header

Click the **S** button on any track in the group to solo the group.



Group not soloed. Click any solo button in a group to toggle the group solo state. Shift-click to solo-safe the group.



Group is soloed. Click to un-solo the group. Option-click anywhere within the Solo column to clear all MELD solos **except solo-safed instances**.



Shift-click to solo-safe. Click to “un-safe” but remain soloed or shift-click again to un-solo. Tracks must be manually “un-safed” before they can be un-soloed.

Notes:

- MELD Solo can only apply to tracks which contain an instance of MELD. Any tracks in the host DAW mixer which do not have MELD inserted will not be affected by MELD Solo.
- Host DAW Solo functions always operate independent of MELD.

  Click the Solo header to ▼ *descending* sort order to quickly find solo’d MELD instances.

Mute



Mute column header

Click the **M** button on any track in the group to mute the group.



Group not muted. Click any mute button in a group to toggle the group mute state.



Group is muted. Click to un-mute the group.
Option-click anywhere within the Mute column to clear all MELD mutes.

Notes:

- MELD Mute can only affect MELD instances. Any tracks in the host DAW mixer which do not have MELD inserted will not be affected by MELD Mute.
- Host DAW Mute functions always operate independent of MELD.

  Click the Mute header to ▼ *descending* sort order to quickly find muted MELD instances.

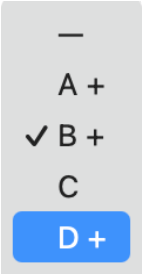
Group



Group column header

The **Group** column shows the current virtual group assignment for each track. Click the Group button of any track to open the list of groups to assign the track to a new group.

There are 26 virtual groups available in MELD, A through Z. Below is a segment of the group selection list illustrating the



— indicates tracks that are not assigned to any virtual group

\+ sign means group A is an active group (clicking A+ will **add** the track to group A)

✓ means the selector menu was opened from a track belonging to group B

group C is inactive: an open group with no existing members

The blue highlight shows the mouse cursor position. Click to assign to this group.

Notes:

- Assigning tracks to an active existing group will immediately apply all group processor settings to the new tracks, overwriting all existing settings.
- The Undo arrow in the MELD header bar can be used to reverse an errant group assignment (restoring the original processor settings) for only one track at a time, so if an entire group is mistakenly reassigned, the tracks must be selected individually and Undo applied to each track independently. This is an constraint of MELD operating within the AAX/AU/VST3 plug-in environments.

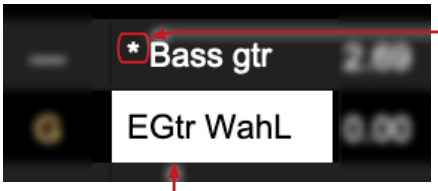
Track



Track column header

The **Track** field lists the name of the host DAW mixer track in which this instance of MELD resides.

Track names are published by the host DAW as part of the AAX, AU and VST3 specifications and are not editable from within MELD. However, editing the track name within the host DAW is immediately reflected in the MELD UI.



An asterisk (*) in the Track name indicates the original instance in the host DAW from which the MELD plug-in UI was opened, in this case, the “Bass gtr” track

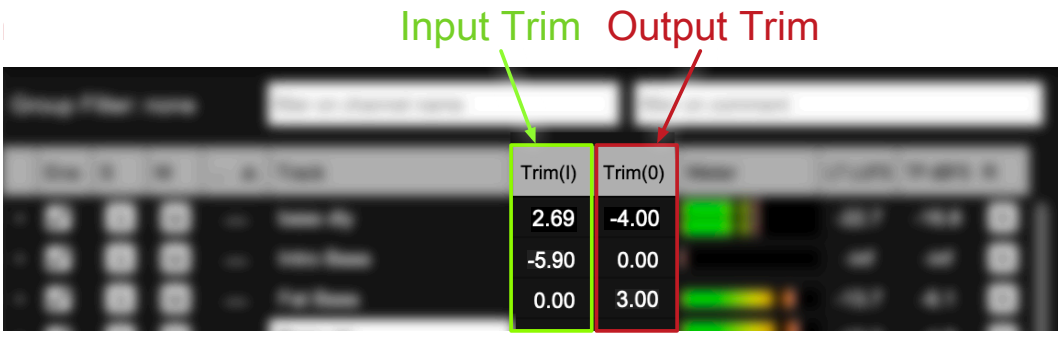
The white highlight reflects the track currently selected and visible in the MELD UI

Input / Output Trims

Input and Output Trim controls are available for each track. Trims are unlinked from group operations, allowing fully independent level adjustment for each track within a group.

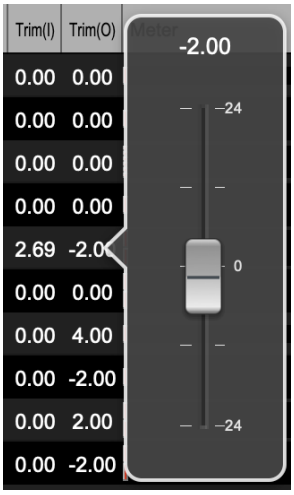
The Input and Output Trims are applied at the same gain stage as the Input Strip and Output Strip faders respectively. The faders are linked to the group setting, but the trims are not linked, operating as independent offsets to the fader gain settings.

Trims are essential for maintaining balances within the group, where some tracks may naturally drift in relative level.



Control-click the desired trim numeric field to directly enter a new trim gain.

Clicking a trim numeric field opens a fader control to ride the trim gain manually and to record automation moves.



Metering in MELD

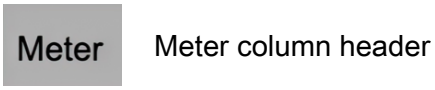
MELD provides two primary level metering locations: per-instance in each track row in the Track List, plus user-assignable LUFS Bar numeric readouts below the Track List.

The Track List displays three types of level meters per instance: Spectrafoo Digital PPM level with RMS, Long Term LUFS and True Peak dBFS.

A second always-visible **LUFS Bar** below the Track List shows Short Term and Long Term LUFS plus True Peak hold for the currently selected instance, and any tracks selected as **Reference** tracks.

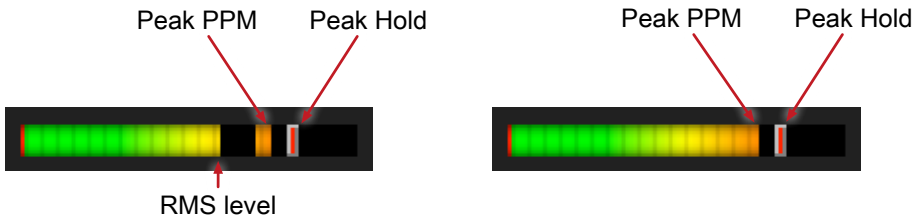
All MELD meters are reset by clicking on any numeric readout in the LUFS Bar meter array below the Track List.

Spectrafoo™ PPM meters



SpectraFoo™ multi-function level metering is provided for every channel of every MELD instance.

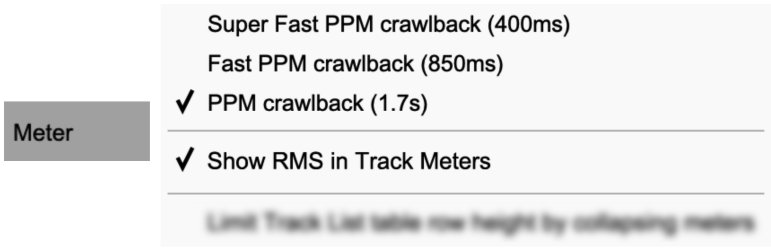
By default, the meters display PPM Peak, with RMS and Peak Hold.



Track List PPM meters: Default setting (with RMS) on left, RMS disabled to the right

- RMS level is indicated by the colored solid bar, which transitions from green at "safe" high-headroom levels to yellow at -18dBFS, then to orange and finally red approaching -6dBFS. The RMS meter uses the PPM standard for decay time (1.7seconds per 20dB) and the digital PPM standard legend for calibration.
- Instantaneous Peak is displayed by the floating colored pip which follows the bar color and decay standard as above.
- Temporary Peak Hold is show as a floating red pip with a hold time of 2 seconds.

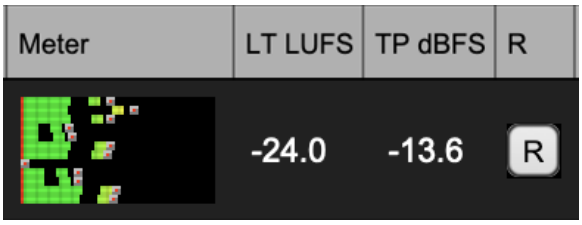
To customize the PPM meters, right-click the Track List Meters column header to open the Meters settings menu:



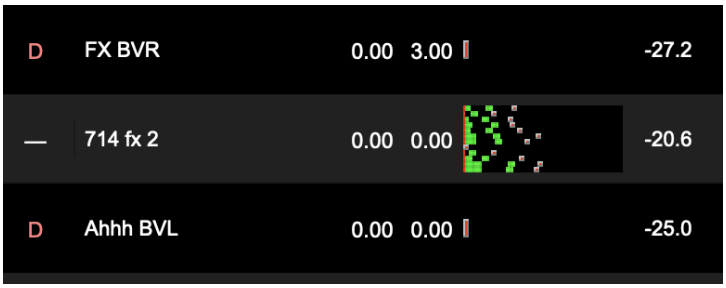
- **Super Fast PPM crawlback (400ms)**, **Fast PPM crawlback (850ms)** and the default **PPM crawlback (1.7s)** set the decay speed for all PPM level meters in the Track List.
- **Show RMS in Track Meters** is checked by default. When unchecked, the RMS readout is removed and the level bar becomes a standard PPM level meter. (See 'Track List PPM meters' graphic on the previous page).

Multichannel metering

Multichannel / multi-mono MELD instances are limited only by the number of channels supported by the host DAW. The PPM digital meters show each channel within a Track, e.g. a 7.1.4 track would show 12 active meters.

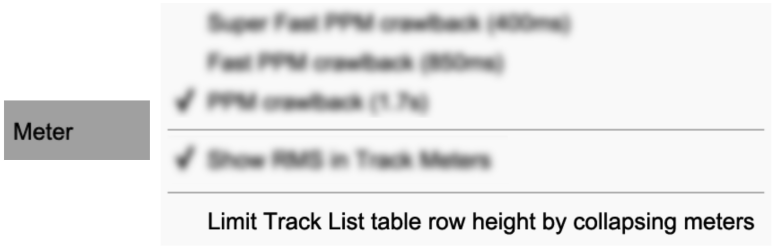


Track heights example with >8 channel instance present

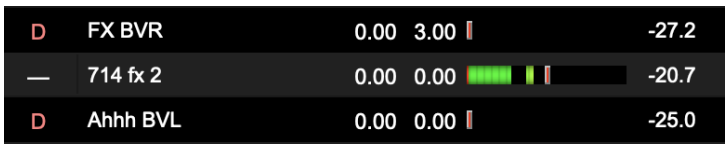


Track List meters showing 7.1.4 PPM default view
(RMS view enabled, "Limit Track List table row height by collapsing meters" preference disabled)

When displaying multichannel tracks greater than 8 channels, the Track List row height must become larger to accomodate. For cases where the greater row height is inconvenient, the "Limit Track List table row height by collapsing meters" option is available in the Meters column header preference menu.



- **Limit Track List table row height by collapsing meters** is a visual aid which simplifies the meter displays for easier navigation of large sessions. When checked, stereo and multichannel PPM meters are flattened to a single "mono" meter reflecting the highest level of all channels within the track, and track row heights are locked to their minimum size.



Note: **Show RMS in Track Meters** and **Limit Track List table row height by collapsing meters** are duplicates of the MELD UI preferences *Show RMS in Track Meters* and *Collapse List Meters*. These settings are saved to the host computer and automatically applied to the next instance of MELD of whatever format, even on another DAW or MIOConsole3d.

LT LUFS

LT (Long Term) LUFS meters display the average loudness of each track over the duration of playback. The LT LUFS reading is held while play is stopped, and is amended when play resumes.

LT LUFS is duplicated for the selected track in the top LUFS Bar readout below the Track List.

TP dBFS

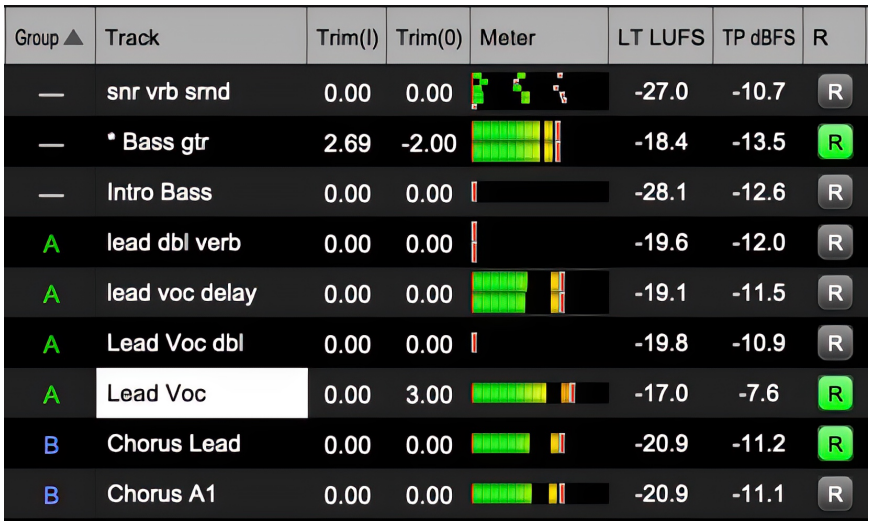
True Peak dBFS meters throughout MELD display 4x oversampled and interpolated intersample peak level, even when MELD processing is not in an oversampled mode.

TP dBFS is duplicated for the selected track in the top LUFS Bar readout below the Track List.

R (Reference LUFS Bar selector)

In pretty much every mix session there will be certain key tracks that serve as guides or references for other support tracks and groups. Equally, some tracks are just plain troublesome and need keeping an eye on. Unfortunately, especially in larger sessions, those tracks may end up scrolled out of view.

In such cases, click the **R** button on a track to set it as a "Reference" track.



Reference tracks gain an additional, always visible meter readout in the LUFS Bar directly below the main Track List.

> **Group parameter automation**: Since parameter changes to any instance within a MELD group are applied to all group instances, it

LUFS Bar (Reference meter array)

The **LUFS Bar** is an additional meter grid which is always visible regardless of the Track List focus.

The LUFS Bar provides a larger, easier to read display of Long Term LUFS and True Peak dBFS for the currently selected MELD track. Secondly, the LUFS Bar adds Short Term LUFS for even more is useful for displaying representative instances from each major virtual group and/or potential problem tracks, up front and always visible as a guide throughout the mixing process.



Important reminder: All track PPM level, LUFS and True Peak metering displayed within the MELD Track List and LUFS Bar UIs represent the output level of the MELD plug-in instance return to the host DAW.

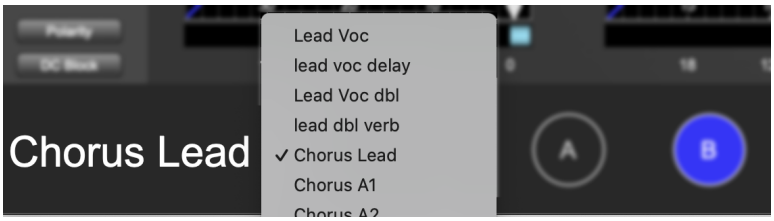
Comments

The Comments field can be used for any kind of text notes. Comments can be anything from descriptions of track content, mix notes, or keywords for text search filters.

In a pinch, the Comments field can be used in lieu of Track Name column in cases where the host DAW does not publish track names to MELD.

Instance / Track Selector

The currently in-focus instance name is shown at the bottom left of the main MELD window. Not only does this always show the current MELD instance in focus, but it allows you to instantly re-focus the MELD UI to any track in the session without having to scroll through the Track List.



Click to reveal a selection menu of all MELD instances for instant navigation without having to change the Track List view. This list is always in the order of host session track 1 at the top.

Post-Production Workflow Shortcuts

The **A, B, C, ...** Group Shortcut buttons below the processor block are to enhance workflows based on three primary groups - such as Music, Dialog and Effects.



The **A, B and C** buttons will highlight when a track belonging to one of those groups is selected.

Clicking **A, B or C** when highlighted will remove the currently selected track from the group.

Clicking **A, B or C** when *not* highlighted will add the currently selected track to the group.

The **...** button will highlight when the selected track is assigned to a group other than A, B or C.



Clicking the **...** button when it is lit will remove the selected track from its group.

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.

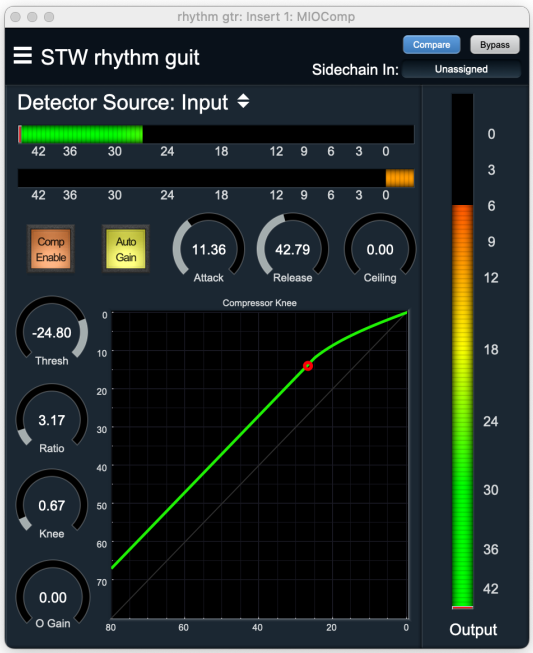
The Core plugs are extremely useful as standalone elements where you just need one job done quickly and efficiently. Along with basic EQ and dynamics, the Core plugs category includes M/S modules, sample-rate-aware delay lines and dithered word-length reduction blocks.

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**



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}

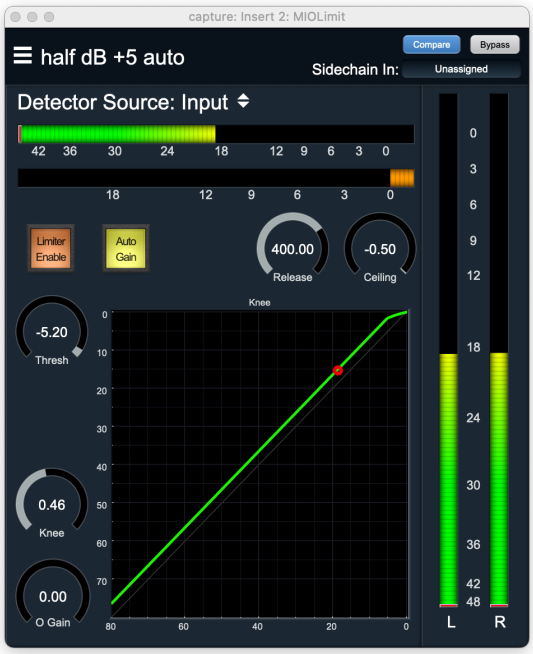
When enabled, fully bypasses the plugin.

See also:

MIOLimit

MIOLimit

Plug-in categories: **Production**, **Dynamics** and **Mastering**



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**



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 Ena`ble) 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**



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**



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



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



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] °

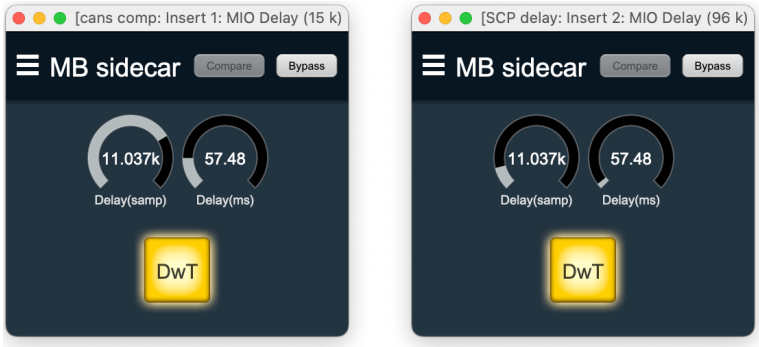
Rotates the stereo field by the specified angle.

`Master Bypass` {On, Off}

Bypasses all of the processing.

MIO Delays (15k and 96k)

Plug-in category: Delay



MIODelay 15k MIODelay 96k

SR = 192kHz

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 3.1. 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.

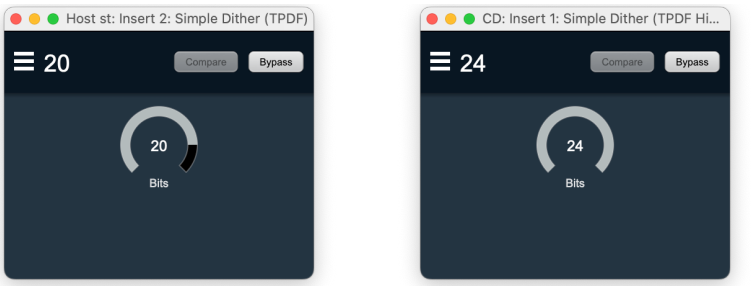
When `DwT` is disengaged (button is dark/uncolored) the `Delay(samp)` parameter will remain unchanged, and the `Delay(ms)` time will vary based on the system sample rate.

`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

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:

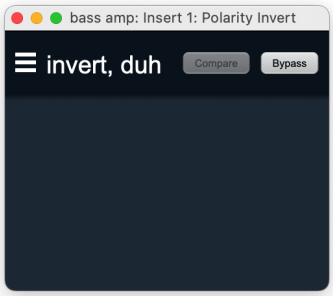
`Bits` [1, 24] bits

Used to specify the final the bit depth of the dithered and truncated signal.

`Master Bypass` {On, Off}

Polarity Invert

Plug-in category: Math



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.
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- Modern Tube Soft Sat: Mastering quality EQ with saturation.
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- 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:

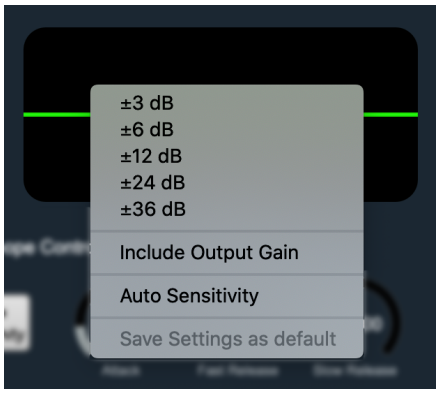
TransientControl is a dynamics shaping processor. By using the `Transient` and `Sustain` controls, it is possible to accentuate the transient or sustained components of a signal; for example, by increasing the Transient and decreasing the Sustain of a snare drum, you can accentuate the hit and decrease 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.



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.



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

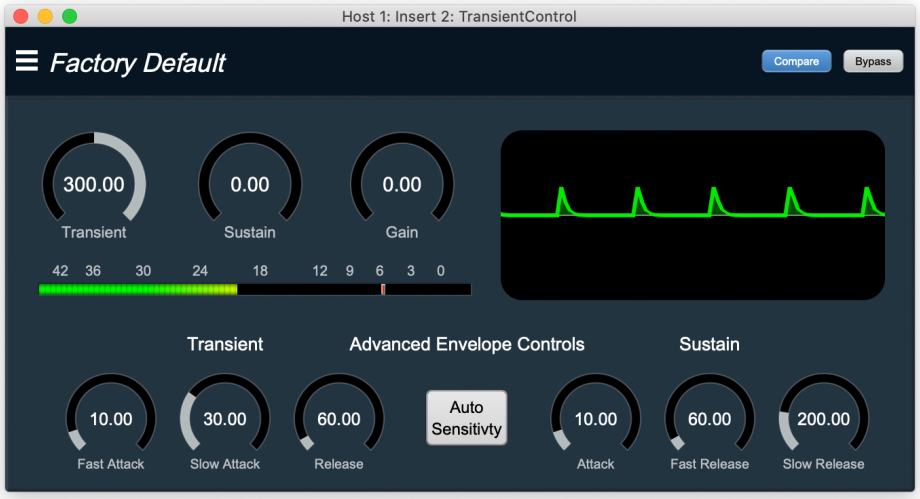
Transient detector

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:



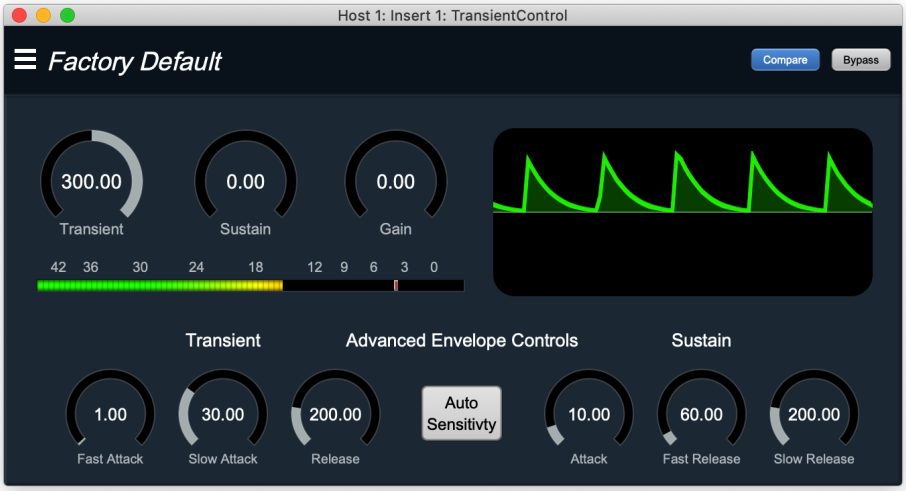
Small Detector Window

When the difference between the Fast and Slow Attacks are larger (1 and 60), more audio is processed:



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:



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.

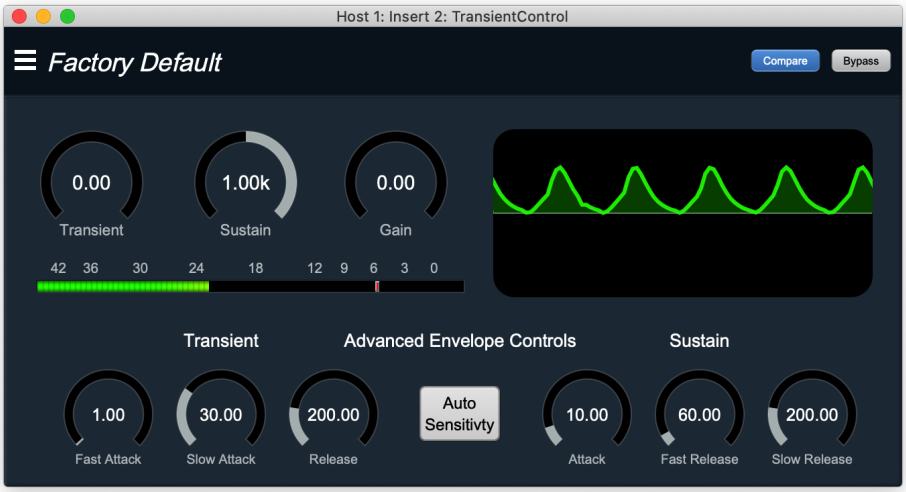
If you make the Release control too short, you may find the applied processing sounds choppy.

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.

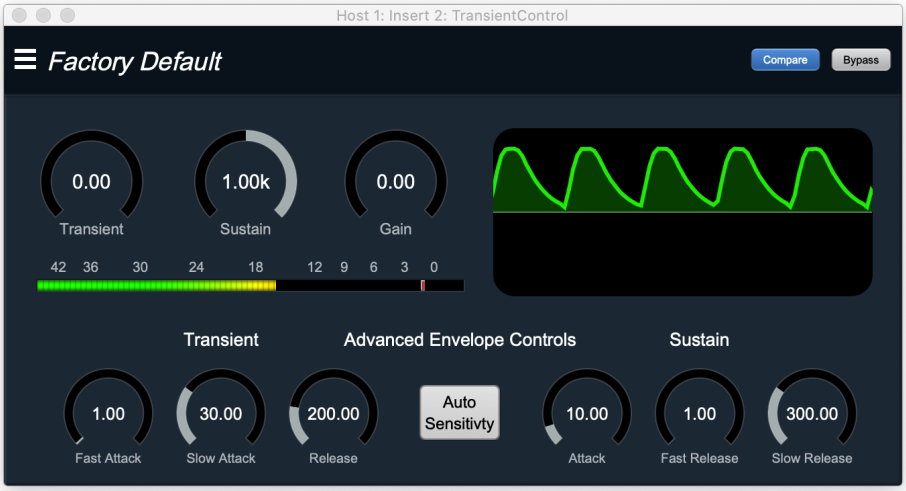
If you make either of the Release controls too short, you may find the applied processing sounds choppy.

For example, when the difference between the Fast and slow parameters is small (here they are 60 and 200), less material is processed:



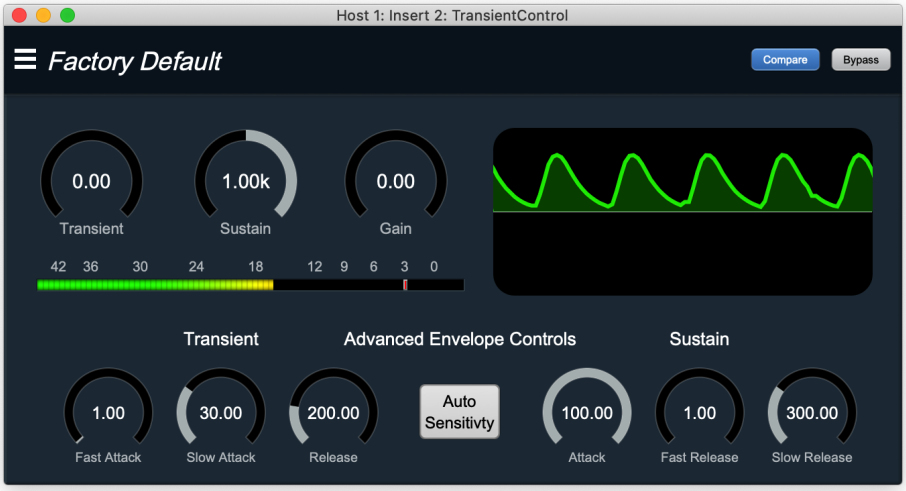
Small Detector Window

When the difference between the Fast and Slow Attacks are larger (1 and 300), more audio is processed:



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:



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.

4. 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 independently of the Monitor Output stages.

- 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 MC window is resizable to accomodate.
- The Monitor Controller Sources and Outputs can be any channel width from mono to 9.2.6. Multiple channel configurations for both Source and Output are fully supported.
- Monitor Controller functions can be mapped and operated via key commands, External Controller (MIDI), 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 9.2.6.
- Every Monitor and Cue output includes per channel +4, -10, Monitor/Headphone output gain modes, level trim and delay (to tenths of a millisecond).
- 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 binaural downmixes for IEMs. Monitor and Cue output Graph processors are placed in the signal path just prior to the final output gain stage and are completely independent from the mixer.
- 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/Headphones 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 selistitemte 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 listitemmeters and custom room tuning graphs will be engaged. 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.



Monitor Controller controls

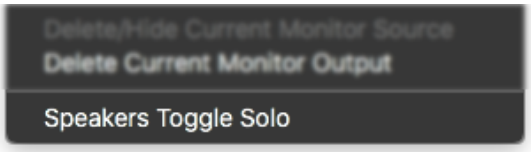
- **Source** Selector: Click on the Source you wish to monitor. MC Sources are created by default for each bus (or as assigned in Configure Mixer).

  The Main mix bus will always be the first (leftmost) radio button on the Monitor Controller Source selector at the top of the MC window. This is a permanent connection from the 3d mixer.

  MC sources from Group and Aux buses and any hardware direct inputs will be listed to the right of the Main in the order they are created. See Monitor menu > Add Monitor Source... for more on adding hardware MC source inputs.

- **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:



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**: Double-click on the numeric readout to enter a volume level. MC Gain is controlled by mouse click-drag or scrollwheel/trackpad gestures, EuCon, MCP or MIDI external controls.

  When addressing a LIO or ULN-8, the MC Gain control automatically syncs with the front panel Monitor control of that unit. In the case of multiple MHLinked units feeding a multichannel speaker array, the MC Gain control will be linked to the front panel control of the box feeding channel 1 of the MC output array. In the case of addressing a Headphone analog output, the MC Gain control will link to the HP control of the target box rather than the Monitor.

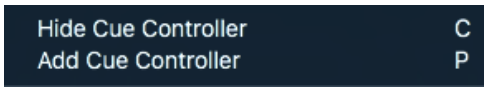
  See Monitor Controller "0.0" reference for details about the MC volume control and output headroom.

- **"-"** 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.

  See Monitor menu > Add Monitor Output... for more on adding hardware MC outputs.

The Cue Controller Interface

There are two important Cue Controller commands which appear in the menu bar Monitor pulldown menu:



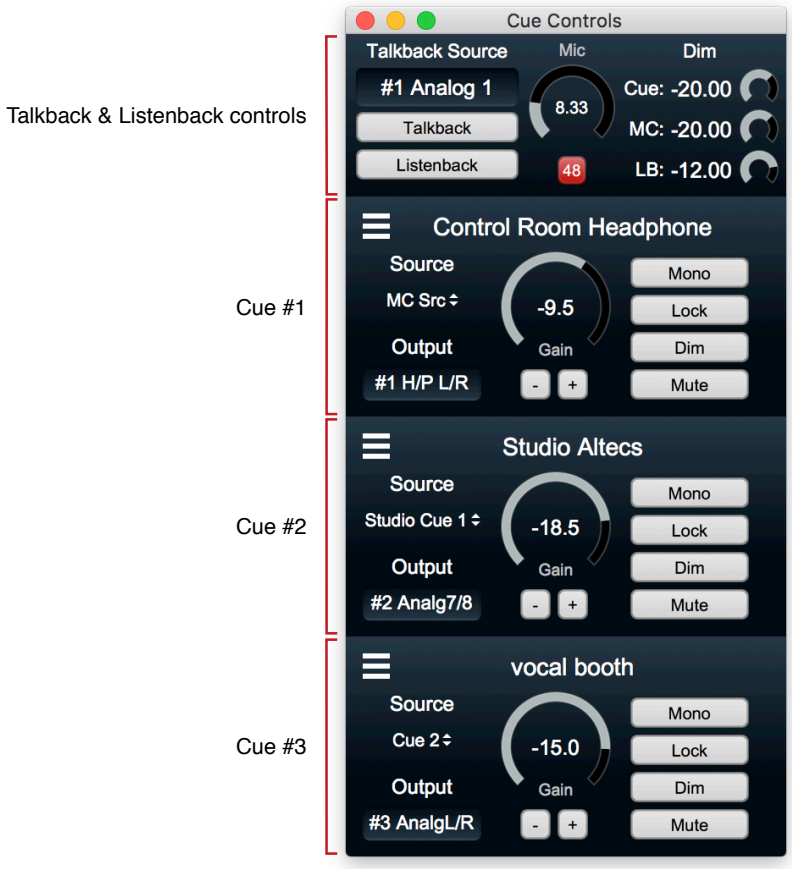
Monitor menu Cue commands

- **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.

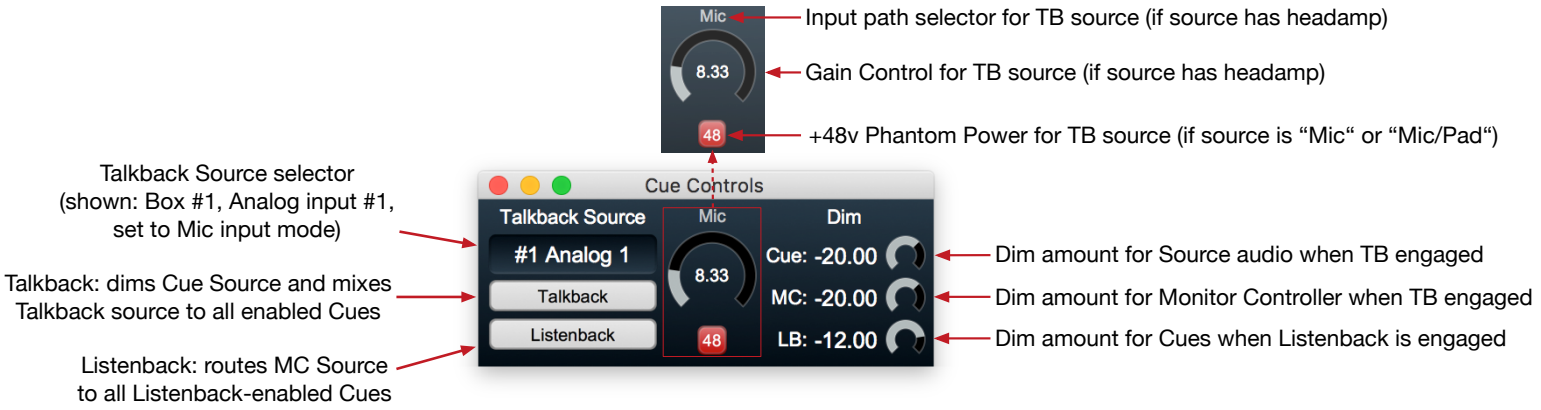
The top section is for setup and control of the Talkback source and Listenback which will feed all Cue sends, plus Dim level settings (shown here at their default values).



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



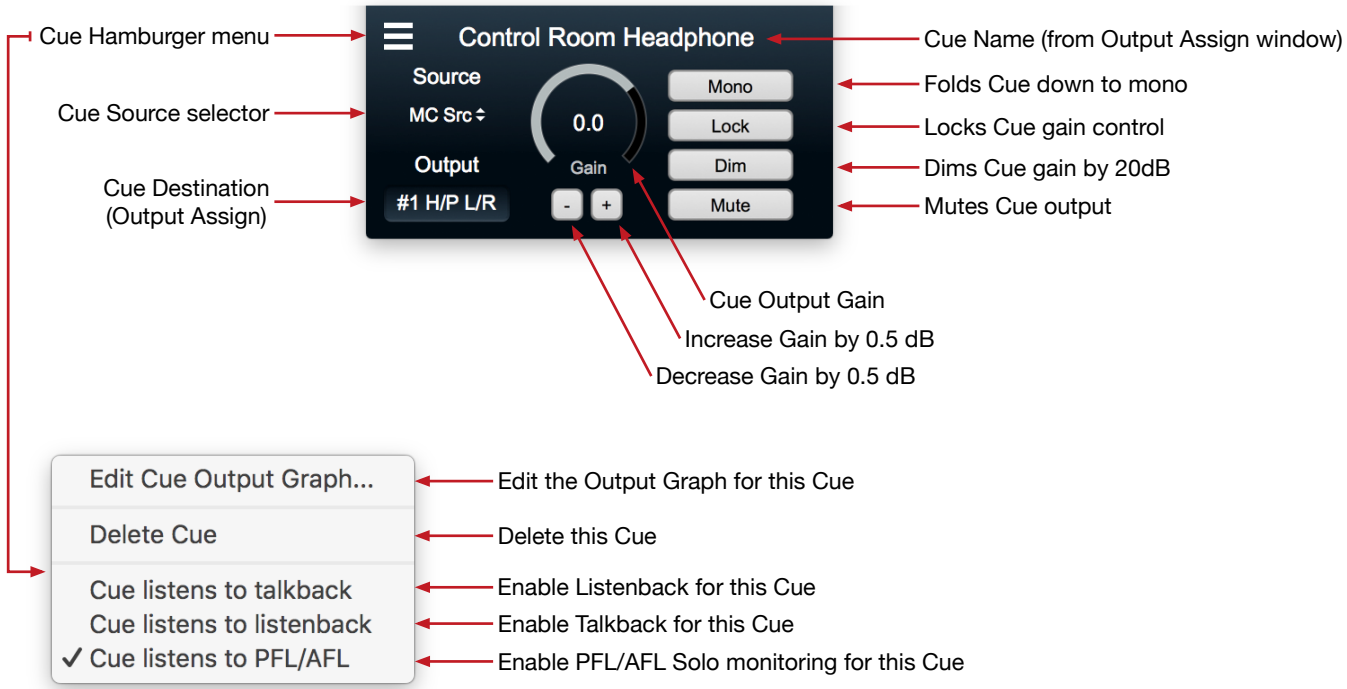
Cue Talkback controls

- 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

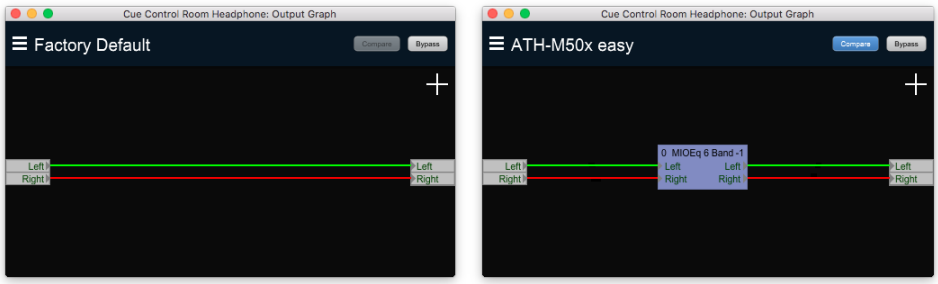


Cue Controls

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.



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 bus source as the feed for this Cue. External hardware MC sources are not currently available as Cue sources.

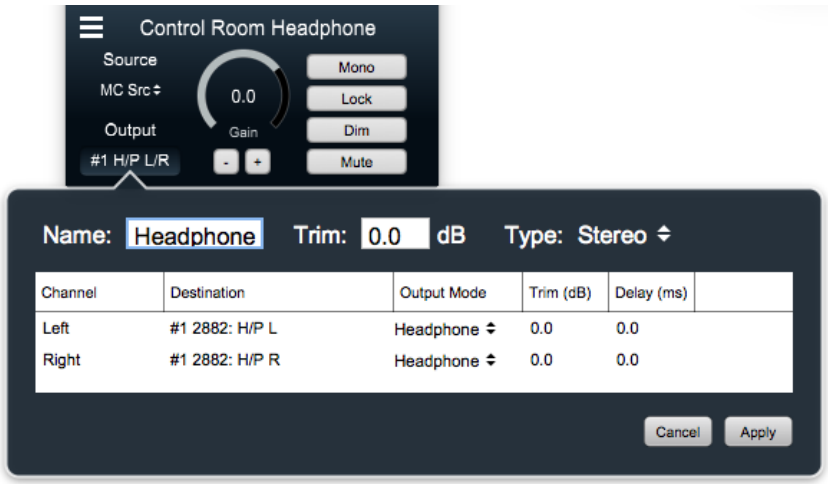


Cue Source selector

MC Src follows the current main Monitor Controller Source selection.

All other selections are static, e.g. 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 Name text field, and assign the outputs for the Cue in the routing section.



Cue Output assign

This example shows the default settings for Cue #1: a stereo cue assigned to the headphone jack of box #1.

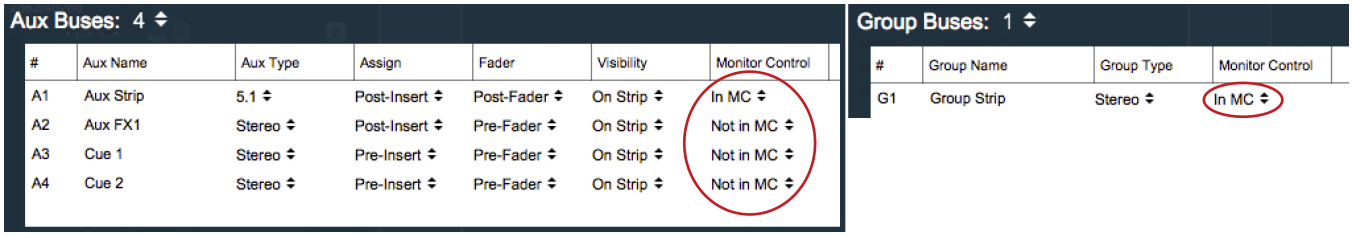
- **Cue Name**: Shows the Cue name you entered in the Cue Output assign window. As shown in the example, the name may be longer than is shown in the name entry field.
- **Mono**: Folds a stereo or multichannel Cue feed down to mono.

- **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. Cue Gain is controlled by mouse click-drag or scroll-wheel/trackpad gestures, EuCon or MIDI external controls.

  When addressing a LIO or ULN-8, the Cue Gain control automatically syncs with the front panel Monitor control of that unit. In the case of multiple MHLinked units feeding a multichannel speaker array, the Cue Gain control will be linked to the front panel control of the box feeding channel 1 of the Cue output array. In the case of addressing a Headphone analog output, the Cue Gain control will link to the HP control of the target box rather than the Monitor.
- **"-"** and **"+"** (Decrement and Increment Gain): Clicking these buttons will subtract 0.5dB from, or add 0.5dB to the current volume level, respectively.

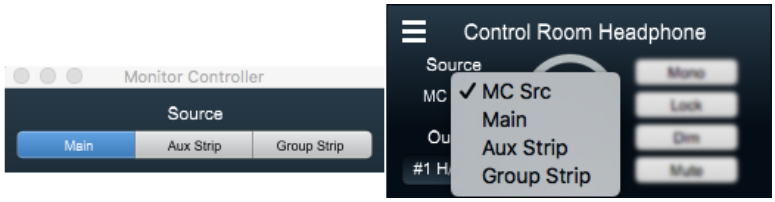
Monitor/Cue source management

- Group and Aux buses are assigned as Monitor Controller sources by default. The connection of each bus to the Monitor Controller is managed from the menu bar Mixer > Configure Mixer window.



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 are not routed to the Monitor Controller source selector.



Monitor and Cue Controller sources as assigned

Note: Cue Control inputs are provided by the Monitor Controller source selector, so if you want to use an Aux as a Cue send, it will need to be set as *In MC*.

Atmos™ multichannel monitoring

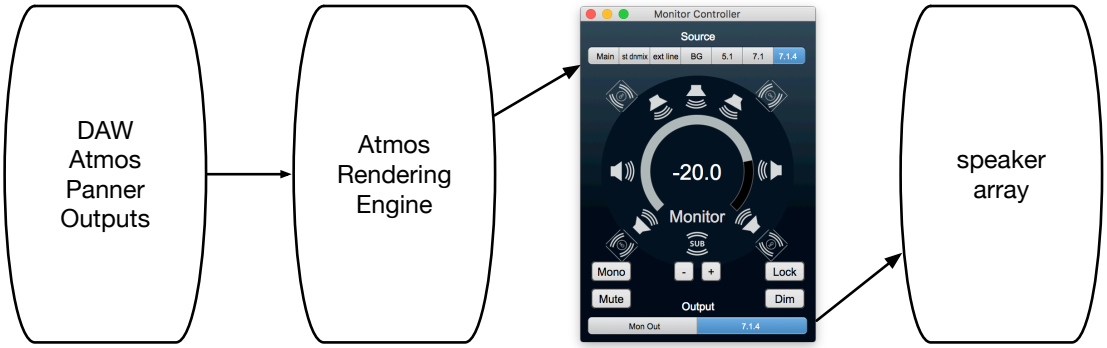
This article assumes the use of multiple 3d ULN-8 and/or LIO-8 devices for multichannel monitoring. The conventions that follow apply to all supported speaker configurations (currently up to 9.2.6) and will apply equally to new formats as they are added.

7.1.4 (Atmos) is used here as an example.

In the case of Atmos 5.1.4 and higher surround configurations, the number of speaker feeds obviousaly exceeds the eight digitally-controlled analog output channels available on a single 3d box.

With MHLink, all connected boxes are time-aligned to the sample and synchronized to operate as a single physical unit. This means you can assign the outputs of multiple 3d boxes to feed a single multichannel immersive speaker array, all synced to a single volume/mute/dim control on the root box. This allows the boxes to be placed near the speaker amps, wherever they may be (out to 100 meters per hop).

The signal flow for Atmos and other object-oriented multichannel surround formats requires both the bed and object channels from your DAWs' surround panners to be routed directly to an external rendering engine. The rendering engine then encodes and outputs your multichannel mix as a 10 or 12 channel (5.1.4 or 7.1.4) audio stream.



Signal Flow for Immersive Multichannel Monitoring (7.1.4 as an example)

See the instructions included with your DAW and/or Atmos rendering package for details on routing audio from your DAW and the renderer inputs, and from the renderer outs to external monitoring.

The MIOConsole3d Monitor Controller fits between the Atmos Rendering Engine and your speaker array (as shown above). So first, create a signal path from your renderer output to the Monitor Controller:

1. **Assign your Atmos monitor controller Source inputs**

   Go to the menu bar and select **Monitor** > **Add Monitor Source...**.

   In the source setup window, enter a name for the MC Source in the Name field to the left, then select **7.1.4 (Atmos)** from the **Type** pull-down menu to the right.

   You will see many more speaker positions listed in the speaker channel routing section below. Now, next to the **Left** channel (under the Source column), click **None** to select the first channel output from your rendering engine.

   The Atmos renderer output channel 1 will always be the Left channel followed by Right, Center, LFE and so on through the Atmos channel layout. Check **Cascade** to auto-assign all twelve Atmos channels to their appropriate channel inputs at once.

   Hit **Enter** or click **Apply** to complete the Atmos MC Source routing and creation of the 7.1.4 MC Source.

2. **Assign your Atmos monitor controller speaker Outputs**

   Go to the menu bar and select **Monitor** > **Add Monitor Output...**.

   In the MC Output setup window, enter an output name in the Name field to the left, then select **7.1.4 (Atmos)** from the **Type** pull-down menu to the right.

   As with the Source setup, click **None** next to the **Left** channel (under the Destination column) to select the first channel output from your rendering engine.

   In the Channel Filter field to the upper left, type in the first few letters of the kind of connection you will use to feed your speakers. Usually this will be "an" for "analog" outputs (without listing the headphone outputs), "ma" for "MADI", etc.

   As above, use **Cascade** to auto-assign all Atmos channels to their appropriate channel output.

Cascade will always assign inputs and outputs in sequence starting from Left in the Atmos channel order; Atmos 7.1.4 being 1=Left, 2=Right, 3=Center, 4=LFE, 5=Ls, 6=Rs, 7=Lr, 8=Rr, 9=LFO, 10=RFO, 11=LBO, 12=RBO. In cases where there may be channels already in use within the assigned range, to avoid conflicts, Cascade will jump over those channels and continue with the next available ports.

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17
Mono	L																
Stereo	L	R															
LCR	L	C	R														
2.1	L	R	LFE														
LCRS	L	R	C	S													
Quad	L	R	Ls	Rs													
5.0	L	R	C	Ls	Rs												
5.1	L	R	C	Ls	Rs	LFE											
5.1 (Film)	L	C	R	Ls	Rs	LFE											
5.1 (SMPTE/ITU)	L	R	C	LFE	Ls	Rs											
5.1 (dts)	L	R	Ls	Rs	C	LFE											
7.0	L	Lc	C	Rc	R	Ls	Rs										
7.1	L	Lc	C	Rc	R	Ls	Rs	LFE									
7.1 (ITU)	L	R	C	LFE	Ls	Rs	Lr	Rr									
5.1.4 (Atmos)	L	R	C	LFE	Ls	Rs	LFO	RFO	LBO	RBO							
7.1.4 (Atmos)	L	R	C	LFE	Ls	Rs	Lr	Rr	LFO	RFO	LBO	RBO					
5.1.6 (Atmos)	L	R	C	LFE	Ls	Rs	LFO	RFO	LMO	RMO	LBO	RBO					
9.1.4 (Atmos)	L	R	C	LFE	Lw	Rw	Ls	Rs	Lr	Rr	LFO	RFO	LBO	RBO			
7.1.6 (Atmos)	L	R	C	LFE	Ls	Rs	Lr	Rr	LFO	RFO	LMO	RMO	LBO	RBO			
9.1.6 (Atmos)	L	R	C	LFE	Lw	Rw	Ls	Rs	Lr	Rr	LFO	RFO	LMO	RMO	LBO	RBO	
2.2	L	R	Sub l	Sub r													
5.2.4 (Atmos)	L	R	C	Sub l	Sub r	Ls	Rs	LFO	RFO	LBO	RBO						
7.2.4 (Atmos)	L	R	C	Sub l	Sub r	Ls	Rs	Lr	Rr	LFO	RFO	LBO	RBO				
5.2.6 (Atmos)	L	R	C	Sub l	Sub r	Ls	Rs	LFO	RFO	LMO	RMO	LBO	RBO				
9.2.4 (Atmos)	L	R	C	Sub l	Sub r	Lw	Rw	Ls	Rs	Lr	Rr	LFO	RFO	LBO	RBO		
7.2.6 (Atmos)	L	R	C	Sub l	Sub r	Ls	Rs	Lr	Rr	LFO	RFO	LMO	RMO	LBO	RBO		
9.2.6 (Atmos)	L	R	C	Sub l	Sub r	Lw	Rw	Ls	Rs	Lr	Rr	LFO	RFO	LMO	RMO	LBO	RBO

Monitor Output speaker configurations

Shown here are the speaker layouts currently available in the MIOConsole3d Monitor Controller.

The **.2** layouts are provided at the request of multi-purpose immersive mastering and production facilities to allow their subwoofers to be employed both as stereo Subs for the main stereo pair and as dual LFE channels when in immersive array mode. The MIOConsole MC lets the same speakers easily perform multiple jobs just by setting up a Monitor Out with a different processing path.

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 an input is 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**

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.



Default Monitor Controller and Cue Control windows

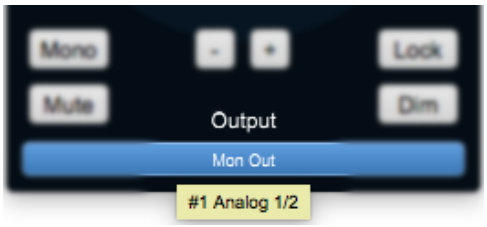
By default, MIOConsole3d will create a Monitor Controller for your domain. If you don't want to use the monitor controller, you can just delete the Monitor Output to return Analog outputs 1+2 to manual control.

> **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 available to feed your Cue sends. This lets you use keystroke commands and outboard EuCon or MCP control surfaces to select Cue sources even when the Monitor Output section is not being used.

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 your Monitor Output channel assignments: Box #1, analog outputs 1 and 2.

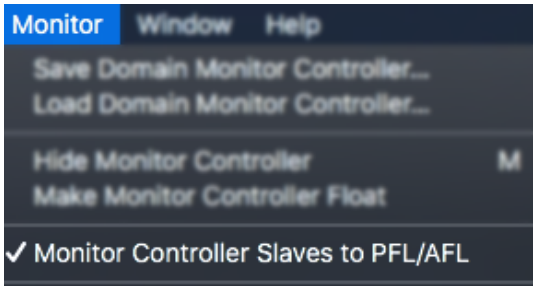


Monitor Controller Output tooltip

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, assigned to box #1 analog output channels 1/2
  - Monitor Output mode: Monitor, (with output Trim and Delay at 0.0 for LIO and ULN models)
  - Solo Mode: Solo-In-Place

    Monitor Controller Slaves to PFL/AFL: ON (ready to engage upon selecting PFL or AFL)



Menu bar Monitor menu: Monitor Controller Slaves to PFL/AFL control

The Solo mode itself is set in Mixer > Configure Mixer panel. See the Configure Mixer Solo Mode section for more detail.

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.

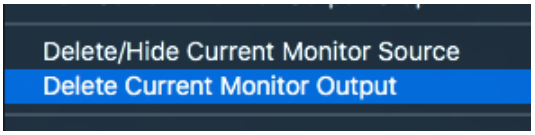
Summary of Cue Controller default settings

- Cue Controller is visible
- Cue Controller Source: stereo, MC Source (Cue source follows Monitor Controller source selection)
- Talkback Source: unassigned (Talkback becomes active when an input is assigned)
- Cue Controller Graph: passthrough
- Cue Controller Output: stereo, assigned to box #1 headphone output

  Note for ULN-2 users: ULN-2s have dedicated Monitor route to both the front panel headphone jack and stereo TRS monitor outs on the back panel, so only a default Monitor Controller output is established. The default Cue Controller for a ULN-2 is unconfigured.

Freeing up Monitor Controller outputs

If your workflow does not require the best-sounding and most powerful Monitor Controller on the planet, highlight the Monitor output you want to remove, go to the menu bar Monitor menu and select *Delete Current Monitor Output*.



Disabling Monitor Controller Outputs

This will immediately remove the MC output and make the outputs available in the output routing dialogs.

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.

5. MIOConsole3d Session

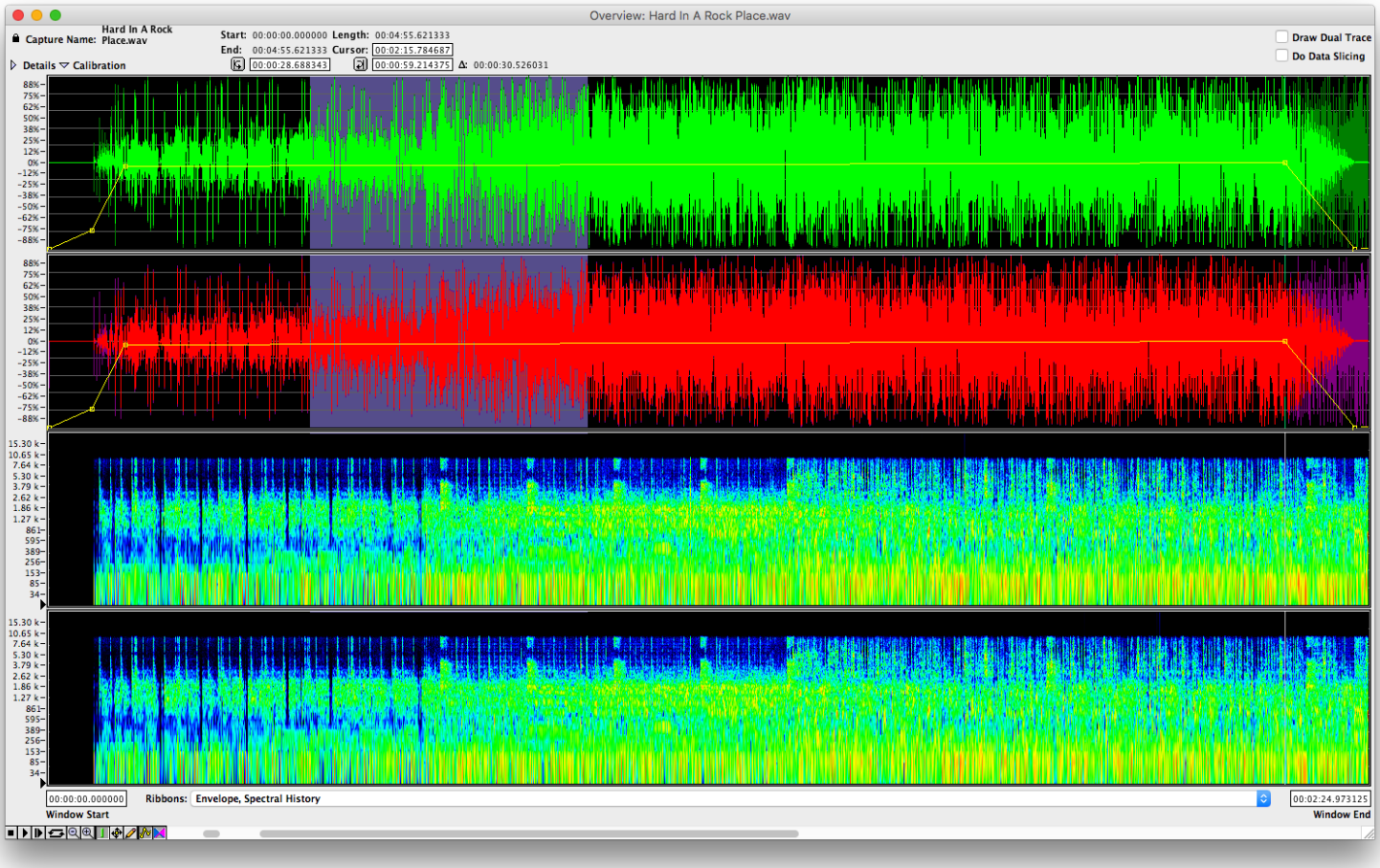
Preliminary Documentation



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.



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



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 5.1. 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 record-

ings organized and making sure all multichannel tracks stay grouped together in the correct channel layout.

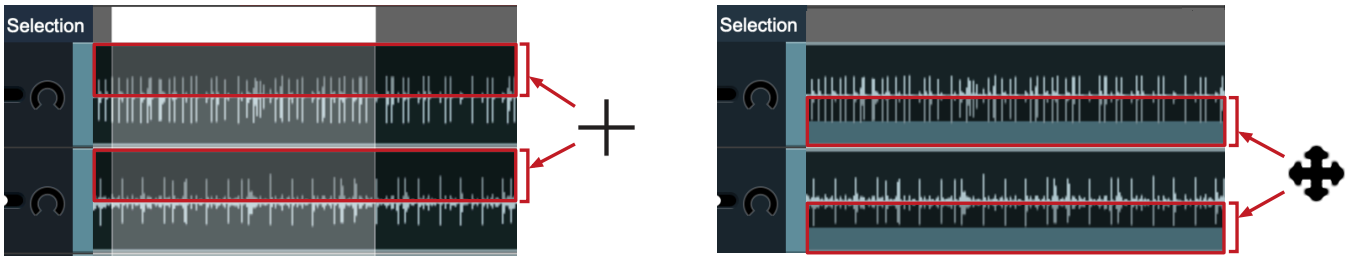
- **Track Lanes:** Each Track can contain multiple lanes. Track lanes allow multiple takes to be recorded into the same track, edited, crossfaded, comped, and mixed within the track itself, without taking up extra mixer strips.

  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.



Time Range Selection (left) - Segment Selection (right)

Session Capabilities

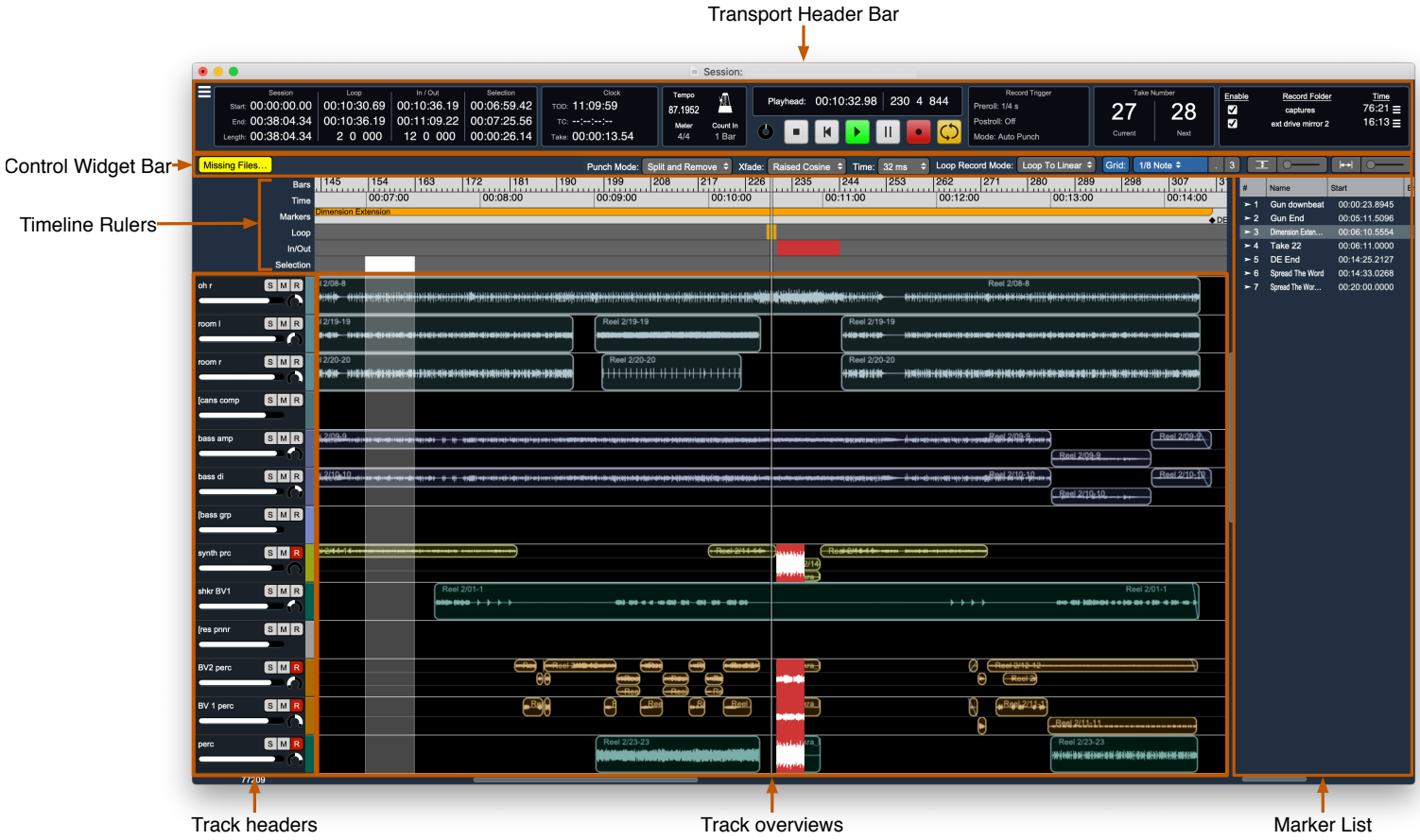
For the merely curious, here is a brief overview of the current **Session** feature set.

- 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.

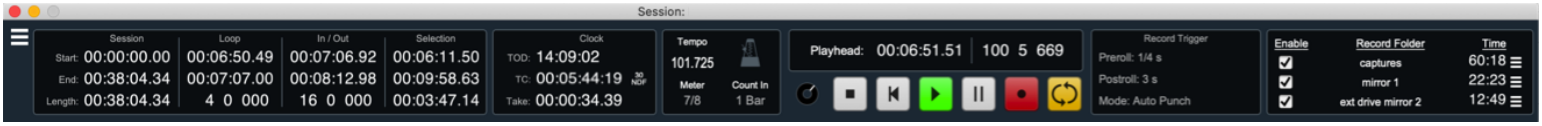
Session UI: Quick Reference



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:



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:



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		

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.

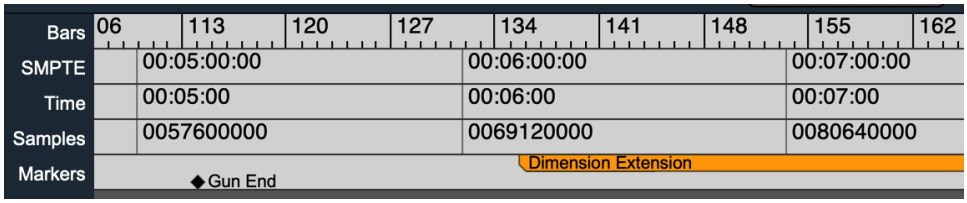
<control>-clicking the marker index will popup a context menu with the following commands:

- 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:



Timeline rulers

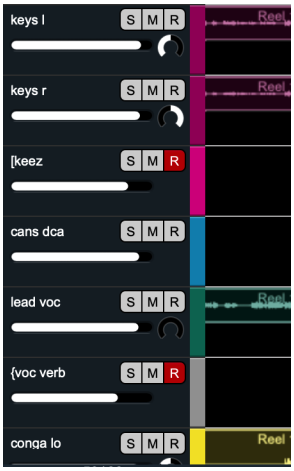
- Bars and Beats
- SMPTE
- Time
- Samples
- Markers

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.



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.



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.

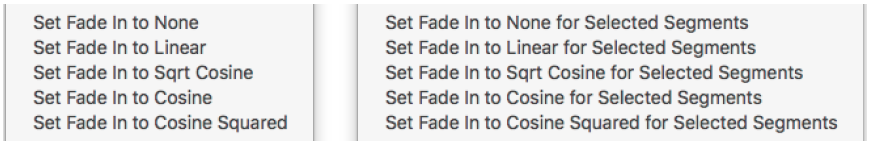
Click and drag fade line to adjust the length of the fade.

> 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.



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).

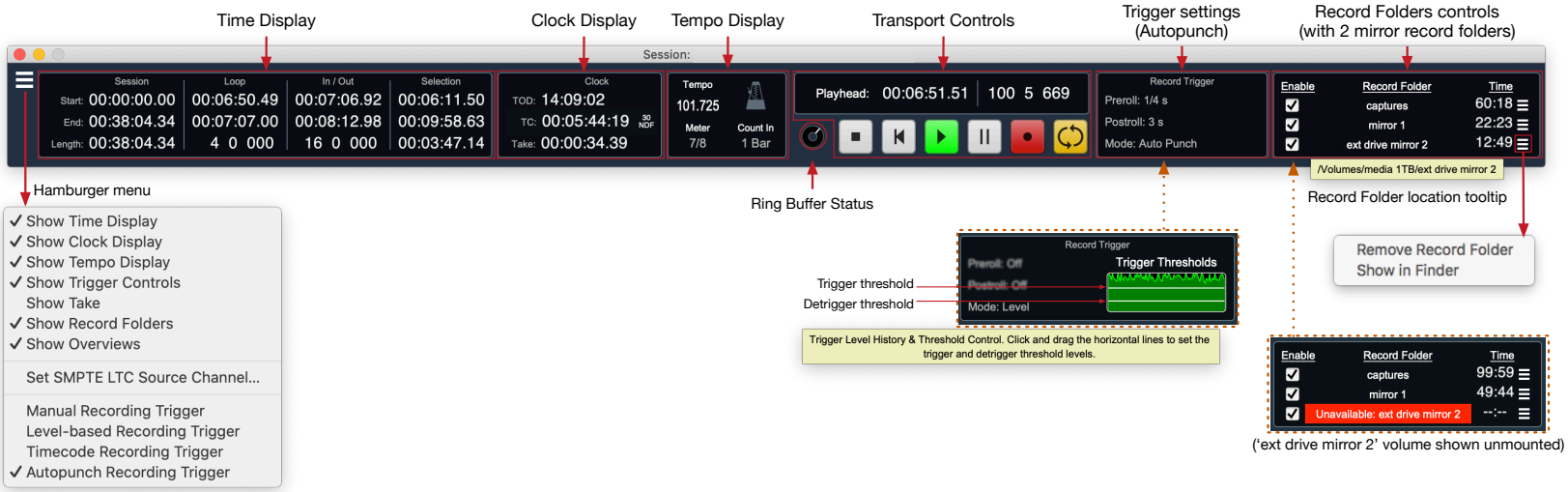
The Session menu has replaced the old **Recording** menu, enhanced with additional commands to support the new **Session** implementation:

- 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

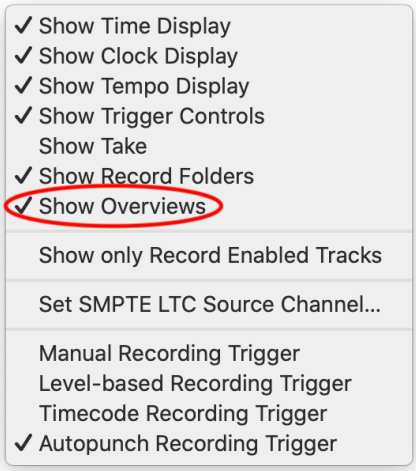
The Session Transport Header bar is comprised of seven primary sections: Time Display block, Clock Display, Tempo Controls, Transport Controls, the “Take” Number Display, Record Trigger settings, and the Record Folders section.



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.



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.

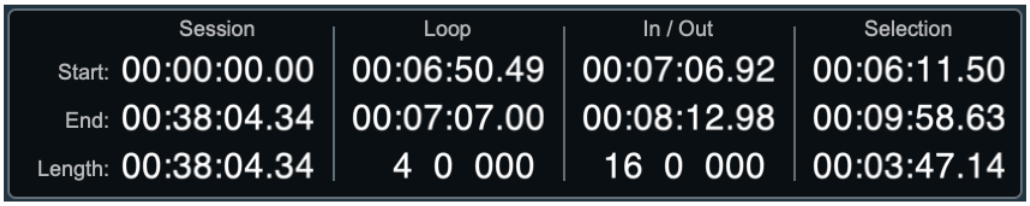
The Session window can be configured all the way down to show only the show Transport Controls block, after which, well, you just minimize or close the window itself. The Transport Controls block can not be independently hidden.

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

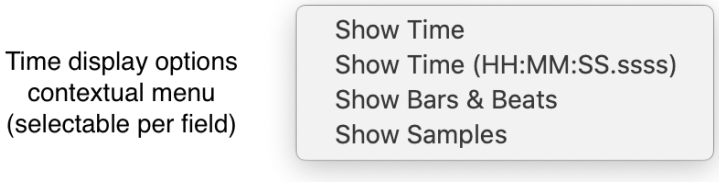
The Time block shows the current Session timeline region selection status.



Transport Header bar: Session Time display

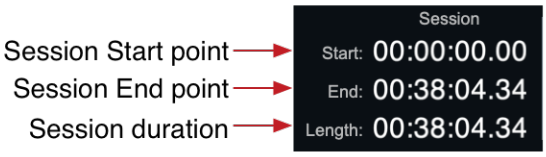
All fields in this display are read-only. Each field may be independently set to display in regular HH:MM:SS time, time with fractional seconds (HH:MM:SS.ssss), Bars & Beats or samples.

Control-click any Time display field to open a contextual menu and select a time format for that field.



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.



Transport Header bar: Session timeline display

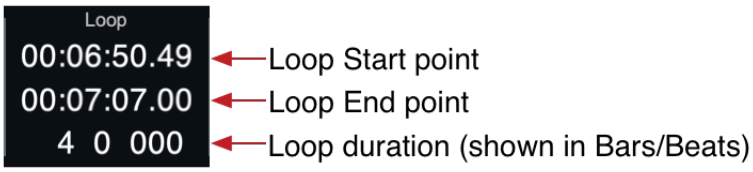
Session 'Start' is the timeline start point (usually 00:00:00.00). Support for timeline Start point offset has not as yet been implemented.

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'.

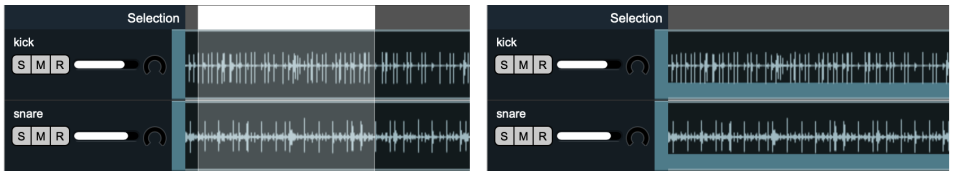
The Time block **Loop** section displays the Start, End and Length of the currently selected timeline Loop.

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.



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.)



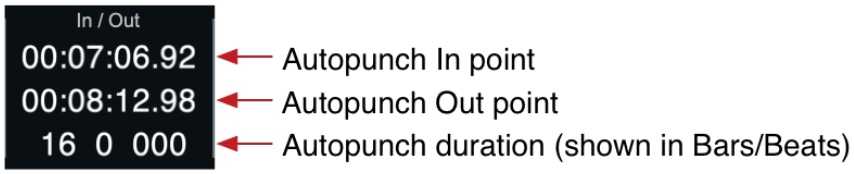
Transport Header bar: Timeline Loop, In/Out and Selection ruler and selection modes

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.

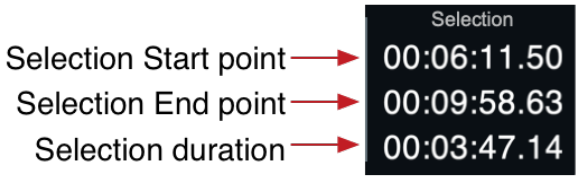


Transport Header bar: Autopunch In/Out time display

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.



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.



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.



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.

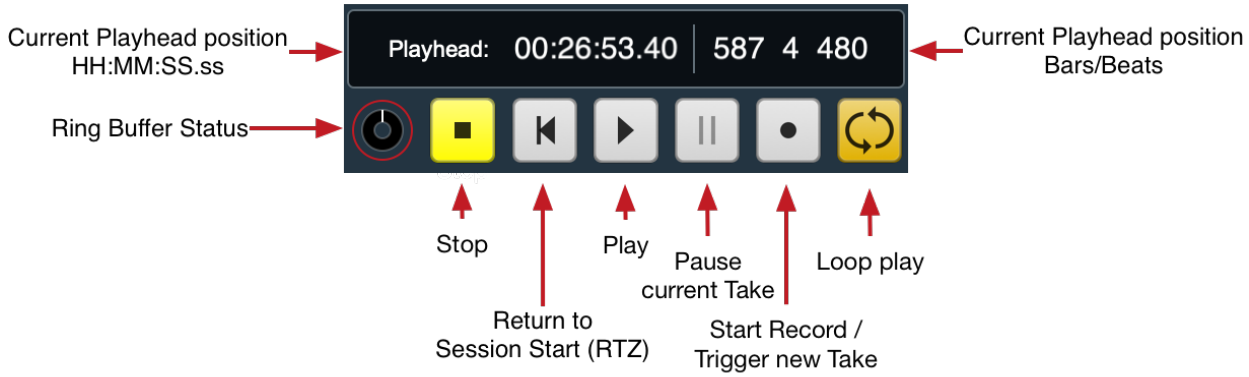
  - **Count In: None** will start both playback and recording at the current playhead position, Loop Start mark (when Loop play mode is engaged), or the Autopunch In point (when Autopunch trigger mode is engaged).

  - **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.



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).



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).



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.



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.



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.



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).



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.

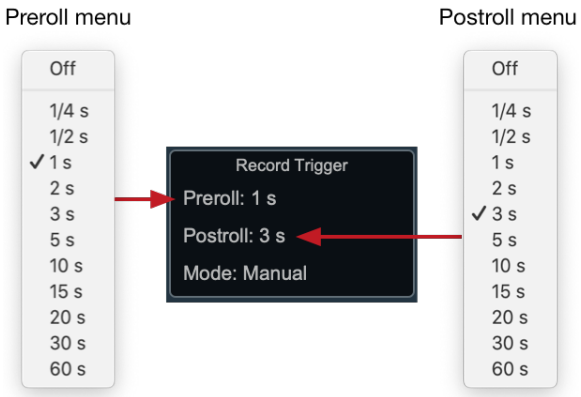
Remember, if you set Preroll to 10 seconds, it requires 10 seconds of pre-loading from the moment you record-enable your tracks to fill that preroll memory allocation. If you start recording a take befoer that buffer is full, it will work fine, but the amount of preroll will be a bit short of the full 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.



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.



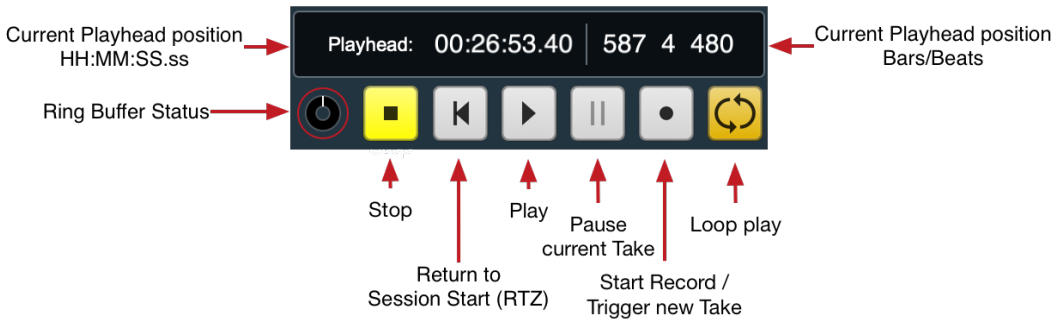
Manual | Level-based | Timecode | Punch In/Out

Session Record Trigger modes

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.



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.

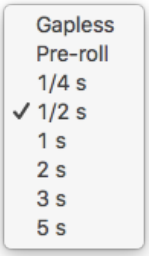
Record Pause: While recording, you can hit ‘Pause’ to temporarily suspend recording while remaining within the current Take. When you click ‘Pause’ the Pause button turns blue, waveform drawing in Overview stops and audio stops being written to your drives. The Record button will remain red, confirming 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.



MIOConsole Preferences: Recording pane: 'Manual Break Overlap' selector

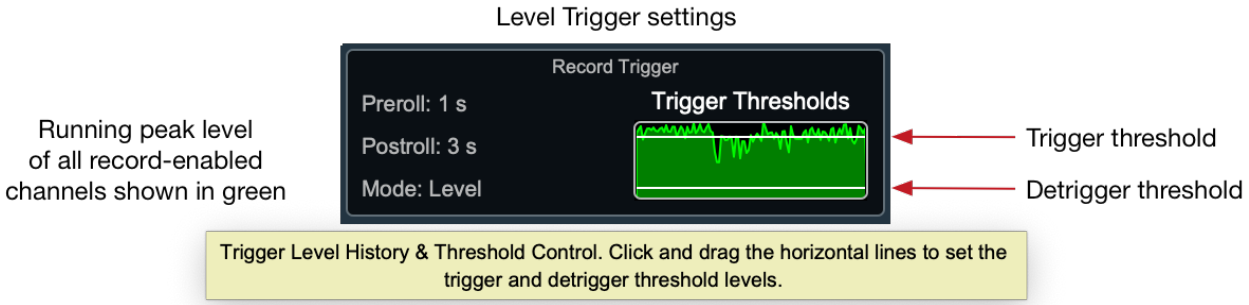
The **Manual Break Overlap** preference lets you set the amount of overlap between the two takes in case you want to edit them back together later. "Gapless" places zero-duration fades at Take file transitions, meaning the last sample of the previous Take is immediately followed by the first sample of 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.



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.

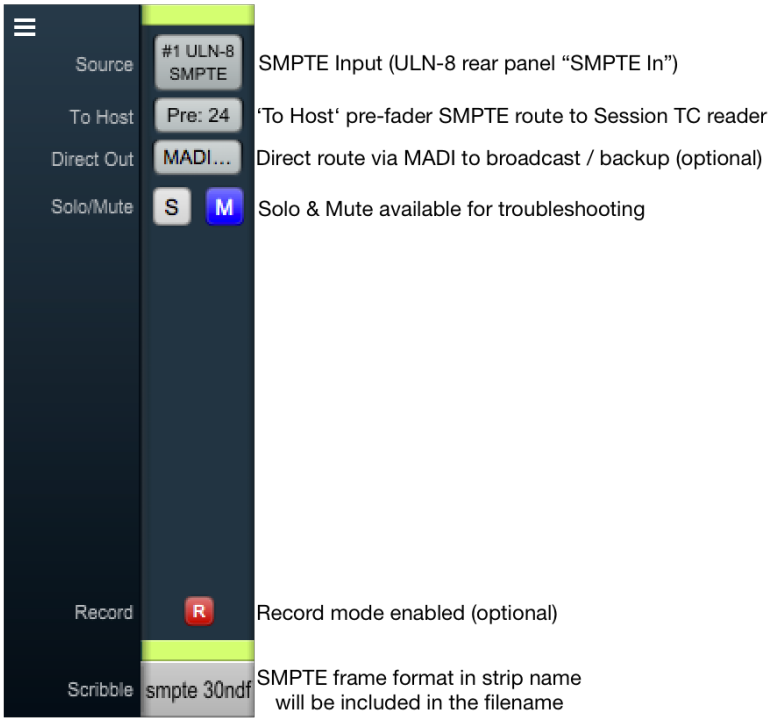
The *other* best thing about level trigger mode is, in your home studio you can leave Record engaged all the time... Thus in the middle of the night (or whenever inspiration strikes) you can stagger into the studio and just start playing - Session will record your ideas automagically (including a lovely 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:



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

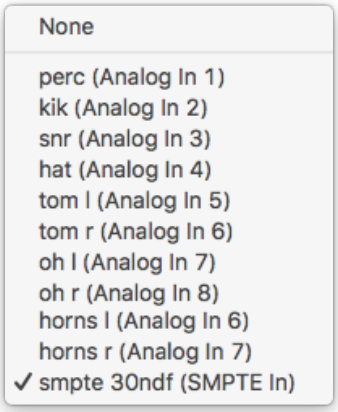
Once you have your SMPTE input strip set up, go to the Session and click the Time Display "Timecode" readout to reveal the "SMPTE Source" selector.

You can also open this selector from the Session Hamburger Menu "Set SMPTE LTC Source Channel..." item.



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.



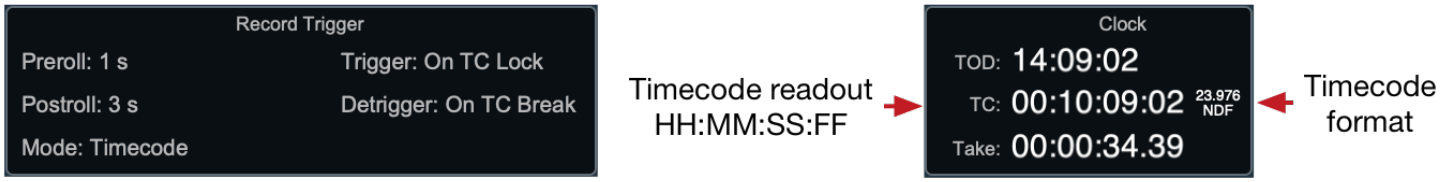
Session: "Set SMPTE LTC source channel..." example selection list

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.



Timecode record trigger mode

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.

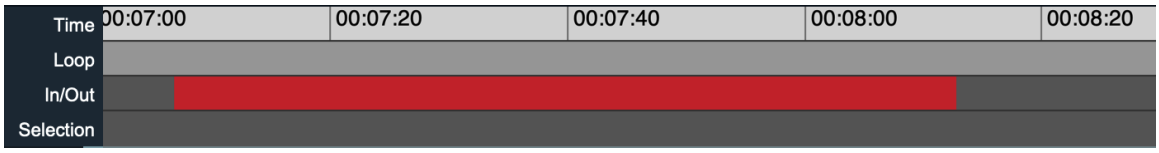
Autopunch Trigger Mode

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).



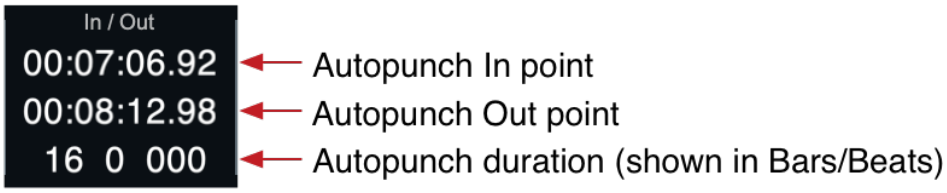
Session Transport: Auto Punch Trigger Mode

Autopunch In/Out points are specified in the In/Out timeline. The selected Autopunch range is highlighted red when active



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:



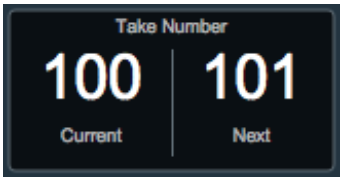
Session Transport Time block: Autopunch In/Out display

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

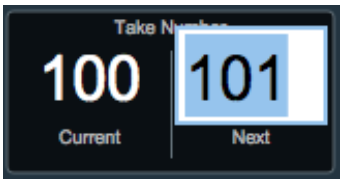
The **“Take”** block appears to the right of the Transport Control section.

- **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.



Session: “Take” display

- The **Next Take** field shows the number of the next Take. Click on this field to edit the ‘Next Take’ number.



“Next Take” field: click to edit



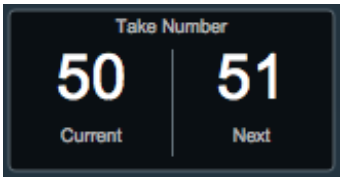
“Next Take” field: with new Next Take number

- Hitting the ‘Return’ or ‘Tab’ key commits the change:



“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:



“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

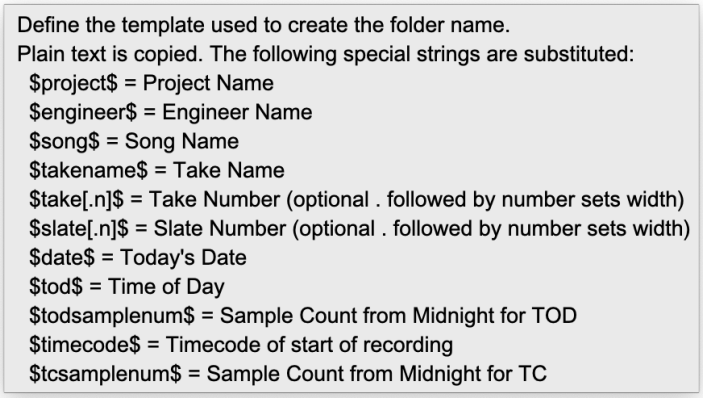
The Take management engine described above ensures that the Session record engine will not inadvertently over-write any previously recorded files, but the generic Take number is hardly sufficient to keep your recordings organized.

The 'MIOConsole3d Preferences: Recording' panel provides a scripting engine to help make Take management more livable.



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:



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".

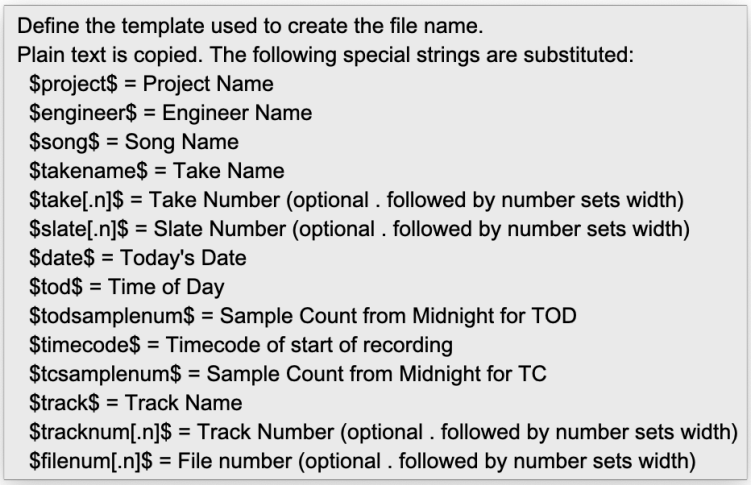
tod says: paste the current Time Of Day into the folder name. The 'time' format is hours-minutes-seconds using a 24-hour clock, so 4:28 and 17 seconds in the afternoon would appear in the folder name as "16-28-17".

$todsamplenum$ 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:



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.

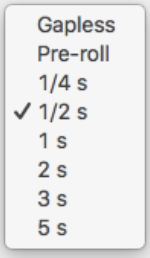
The Take count is automatically incremented at each record start.

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.



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.



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.

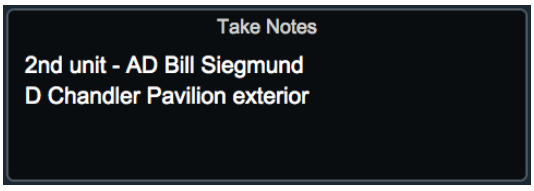


Slate field: click to edit

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.



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.

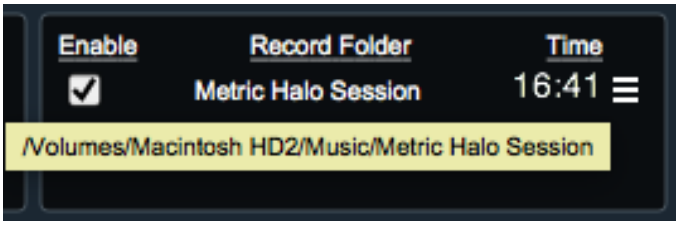
Record Folder block

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.

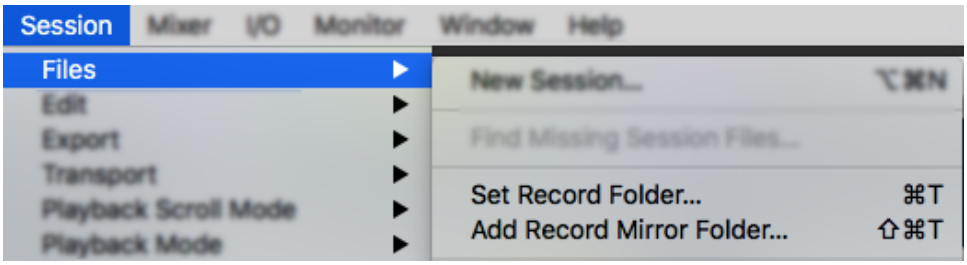
  Holding the mouse cursor over the folder name will pop up a tooltip revealing the full file system path to that Record Folder:



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:



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.



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

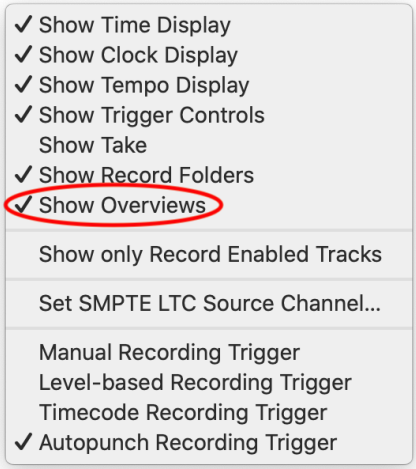
Every Take captured by Session is accompanied by a "Recording.log" file. The Recording.log is a generic text file which can be opened in any text reader, and includes timestamps, file path and file type for every file in the Take, Number of Tracks in the Take, Number of Channels in the Take, Number of 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.



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.



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.

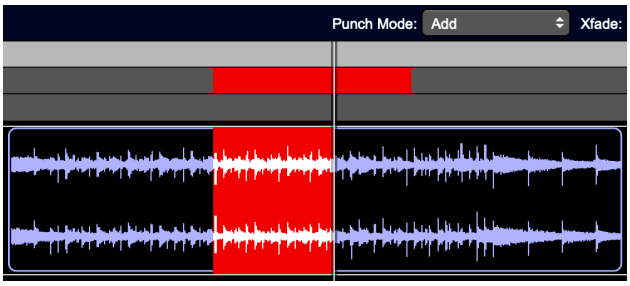
Recognized folder and filenames will change from yellow text to white. Hit **Return**, **Enter** or the "Finish" button to finish, and remember to **⌘ S** (Command + S) to "Save" the Session with the new audio locations, or **⌘⌥** (Command + Option + S) to "Save As…" with a new name.

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.



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.



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.

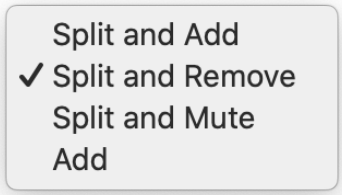
Time sets the duration of new segment and edit fades.

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:



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.



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.



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.

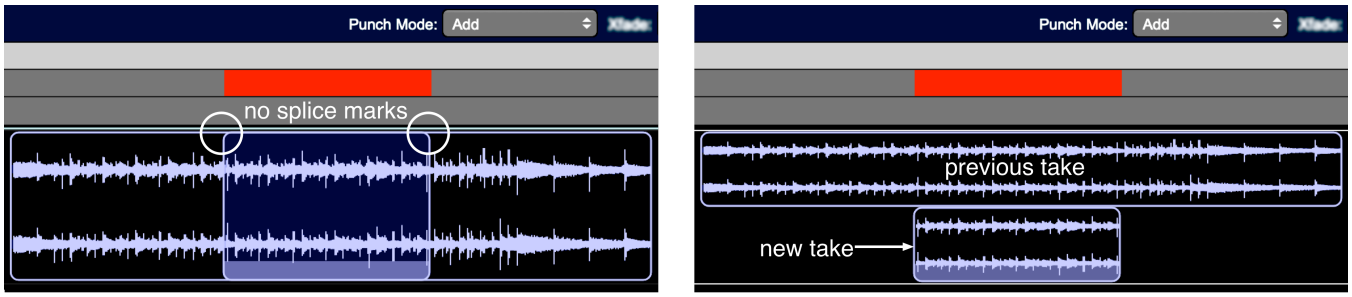


Punch Mode: Split and Mute

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.



Punch Mode: Add

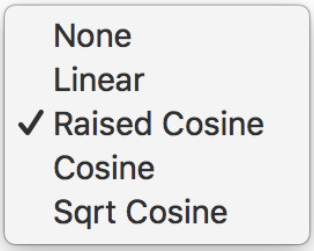
The sole difference between Add and Split and Add modes is the convenience of the perfectly-placed splices Split and Add inserts on the prior audio takes.

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

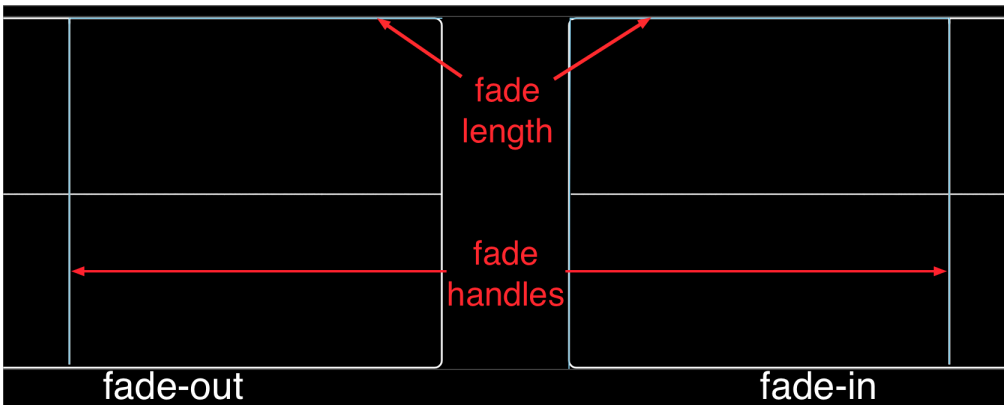
The **Xfade** popup lets you set the slope type of fade or crossfade to be applied to new segments.



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

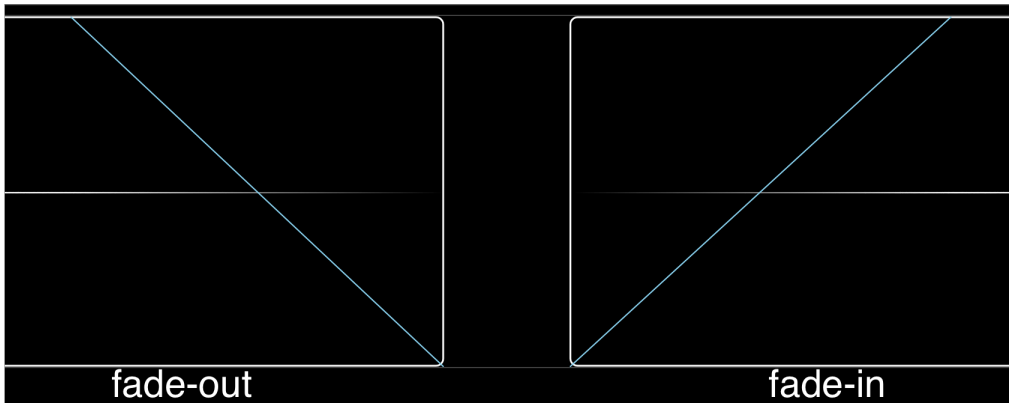


***None* (zero-duration crossfade - a butt-splice)**

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.

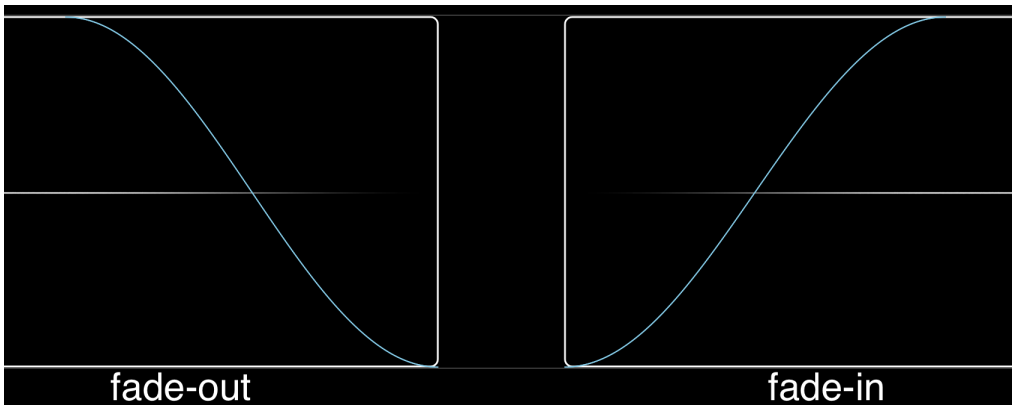
Xfade: Linear



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.

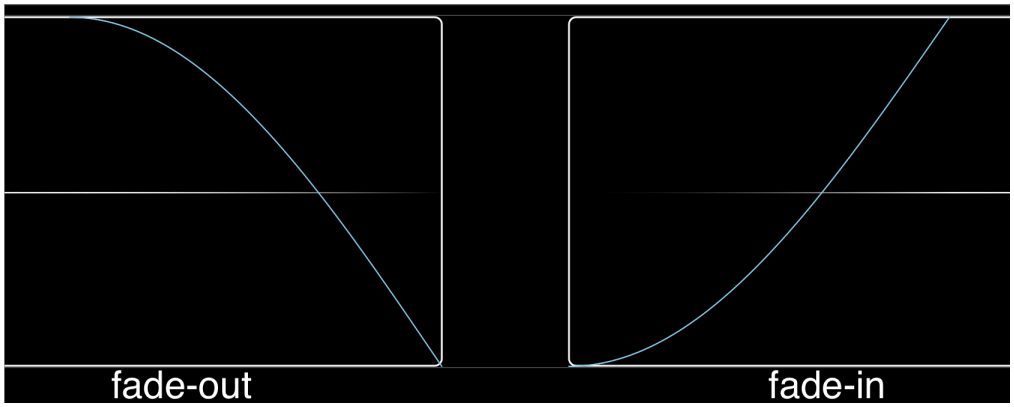
Xfade: Raised Cosine



***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.

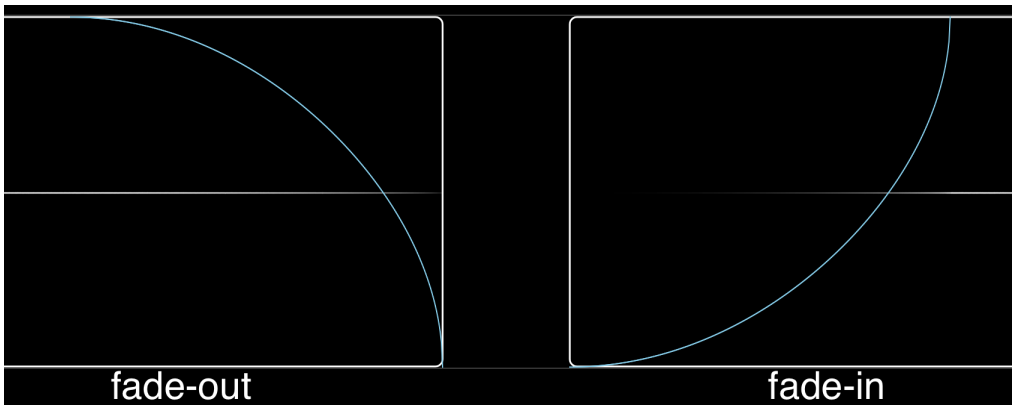
Xfade: Cosine



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.

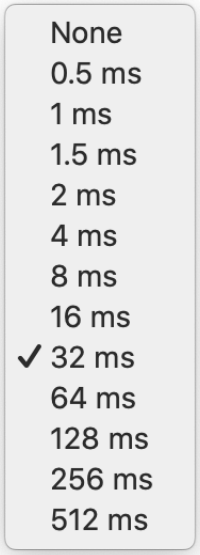
Xfade: Sqrt Cosine



***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



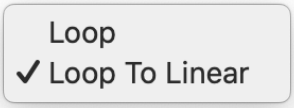
fade/crossfade length menu

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.



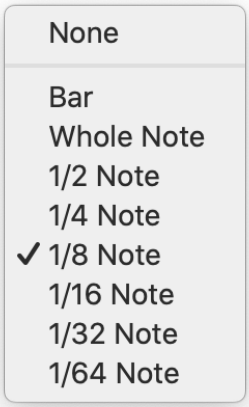
loop record mode menu

- **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



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.

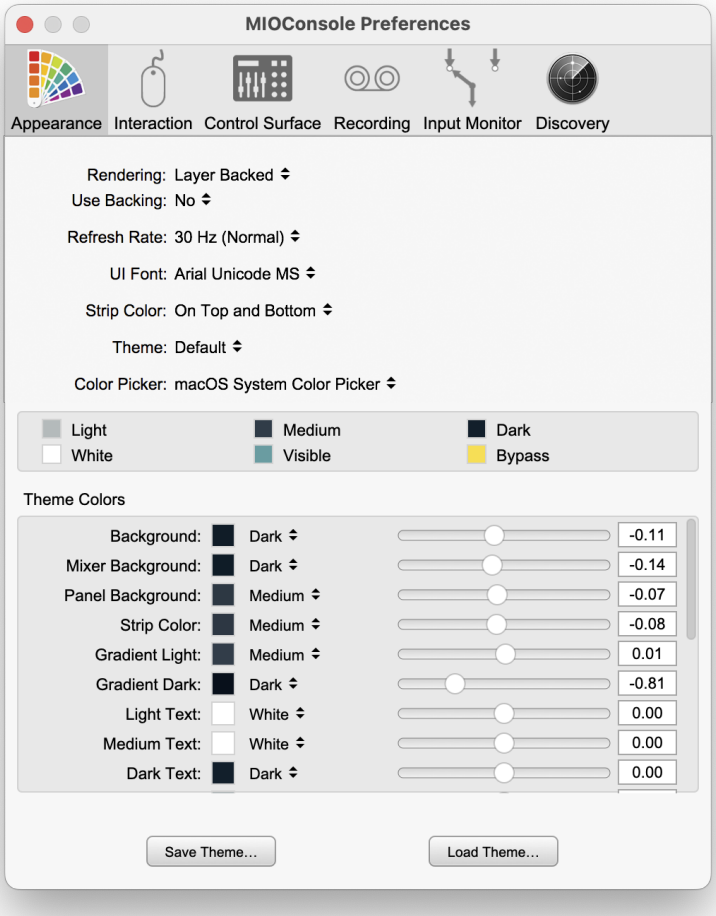
Command	Key Sequence
Zoom to Fit Selection	⇧Z
Increase Session Track Height	⌘↓
Decrease Session Track Height	⌘↑
Scroll Up / Down	scrollwheel
Fit Tracks Vertically	⌥X
Zoom Tracks In Horizontally	⌘→
Zoom Tracks Out Horizontally	⌘←
Scroll Left / Right	⇧ scrollwheel
Fit Tracks Horizontally	⌥Z

Table 5.2. Session navigation: default key commands

6. 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:



MIOConsole3d Preferences (Appearance preference pane shown)

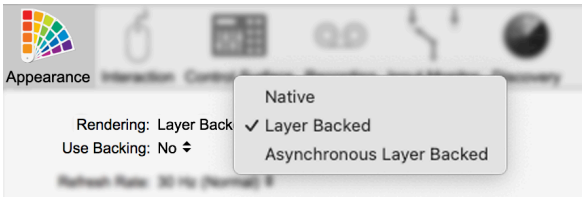
Across the top of the window are six selector tabs: Appearance, Interaction, Control Surface, Recording, Input Monitor and Discovery.

- **Appearance** is all about the look and feel of the MIOConsole UI, from graphics engine tweaks to the mixer font and color scheme.
- **Interaction** dictates global MIOConsole3d behaviors.
- **Control Surface** is where you set up EuCon and MCP remote control surfaces.
- **Recording** handles record engine behaviors, file type, filenaming and iXML metadata.
- **Input Monitor** controls Mixer modes and interaction with Session playback (tape monitor behavior...)
- **Discovery** provides important settings for scanning and registering 3d devices when launching the MIOConsole application.

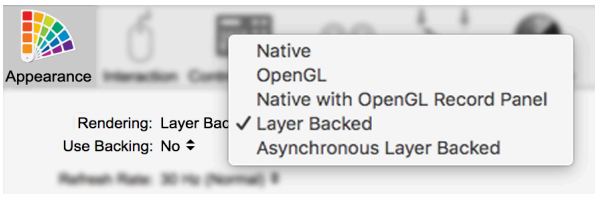
Appearance Preferences

- **Rendering**

  Macs are equipped with Native graphics rendering which uses the main CPU, plus GPU-based Layer Backed and Asynchronous Layer Backed rendering engines. In an ideal world, software would run optimally with just one of them, but in testing we see variations in graphics 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.



macOS 10.14 (Mojave) and later



macOS 10.13 (High Sierra) and older

Rendering submenu

Macs running OSX 10.13 and earlier may also select the Open GL rendering engine (shown above right). Support for the older OpenGL renderer ended with macOS 10.14 (Mojave).

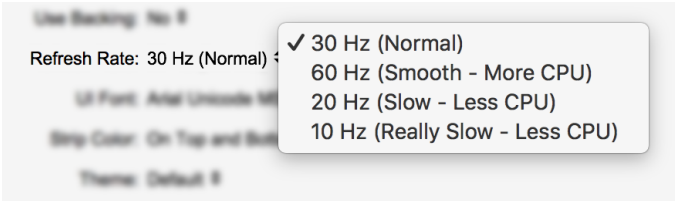
Layer Backed is the default for all machines, but if OpenGL is available to you, give it a look. Native with OpenGL Record Panel might also be worth a shot, traditionally offering a nice compromise between the smoother text and outline rendering of the Native engine, with the more complex scrolling waveforms in Session drawn more smoothly and efficiently by the OpenGL rendering engine.

Use Backing: No (the default) will be more CPU/GPU-efficient on modern machines. The control is available as an optimization option for older machines/OSs.

Switching between Native, Layer Backed and Asynchronous Layer Backed modes should be immediate, as is switching Backing. However, switching to and from OpenGL modes requires quitting and re-launching MIOConsole3d.

Note: The **Rendering** and **Refresh Rate** settings are global preferences and are not saved as part of a Theme.

- **Refresh Rate**



Refresh Rate submenu

Refresh Rate controls how quickly MIOConsole3d will attempt to update the meters and Playhead graphics. 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.

- **UI Font**

  UI Font allows you to change the primary text font throughout the MIOConsole user interface. The default font Arial Unicode MS looks pretty good, is nicely readable and being Unicode it supports non-English character sets.

  But, maybe you feel like injecting a little more personality into your mixer… Popular alternatives include Futura, Roboto and DIN Alternate. Comic Sans is always good for a '90s vibe, and if you favor lengthy prose from Hobbiton you might like Luminari.

  Known good fonts which work well throughout MIOConsole are grouped at the top of the list. Fonts which do not scale properly, are missing characters or a just plain illegible have been filtered from the available selections as a time saving measure.

  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.
  Note: UI Font is a global preference and is not saved as part of a Theme.

- **Strip Color**

  MIOConsole mixer strips have small color bars at the very top and bottom (just above the Scribble Strip). Clicking on these color bars opens a Color Picker 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.
  - **On Top and Bottom** colors only the color bars. The body of the strip follows the *Strip Color* Theme Colors setting.
  - **Full Strip** applies the color to the entire strip.

  Note: Strip Color is a global behavior setting and is not saved as part of a Theme or within a saved Console. Importing a Console file from another system will show up with your saved Theme and Strip Color settings.

  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.

- **Theme**
  MIOConsole3d supports customizing the color scheme of nearly all visible elements of the UI. Customized UIs can be saved, shared and loaded as Themes.

  Theme opens a selection list which includes the Default theme plus 24 factory presets. 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.

  Unsaved edited themes are shown as *Customized*. Switching to another theme preset will clear any unsaved changes.

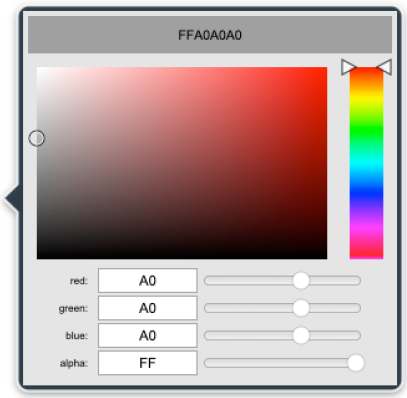
  MIOConsole3d theme presets are saved with a '.3dTheme' file extension in the User > Library/Preferences/MIOConsole3d/Themes folder.

- **Color Picker**

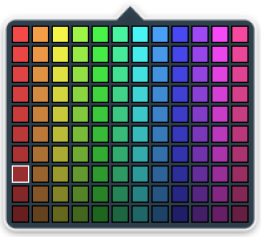
  The Color Picker allows you to choose which color picker you would like to use to modify mixer strip and theme colors.



macOS System Color Picker



Popup HSB Color Picker

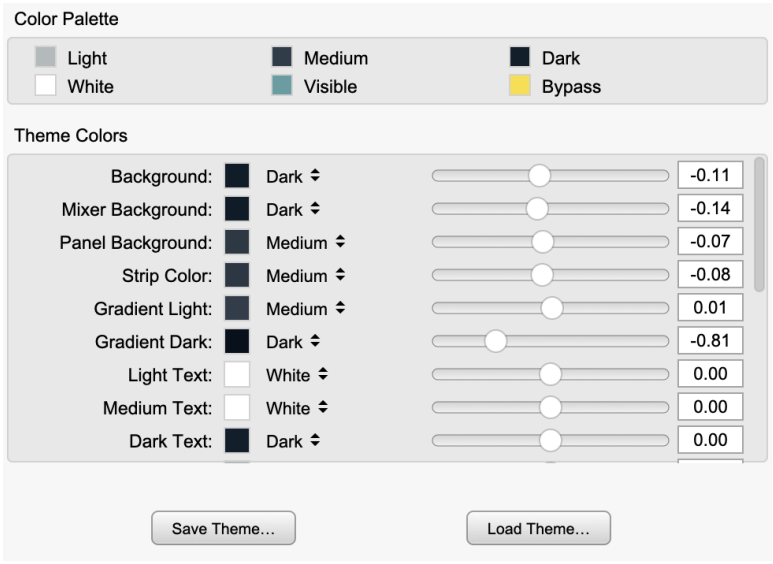


Popup Swatch Color Picker

  - The **macOS System Color Picker** opens as a floating window, with the standard Color Wheel, Slider Control, Preset System Palettes, Spectrum and Crayon Box tools all available across the top.
  - **Popup HSB Color Picker** is a popup window that allows you to choose via Hue, Saturation and Brightness controls or directly set the color components numerically.
  - **Popup Swatch Color Picker** is a popup window that allows you to choose via a fixed color palette.

- **Color Palette: Controls and Customization**

  In addition to mixer strip colors and user-selectable fonts, the color scheme of MIOConsole3d is completely user-tweakable.



Color Palette controls,

There are two control types for color palette manipulation:

Theme Colors affect the underlying color structure of the various UI elements across all MIOConsole3d windows directly. The six Color Palettes are pre-set groups of Theme Colors. 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 selector pull-down. The slider with each Theme Color is a basic brightness control.

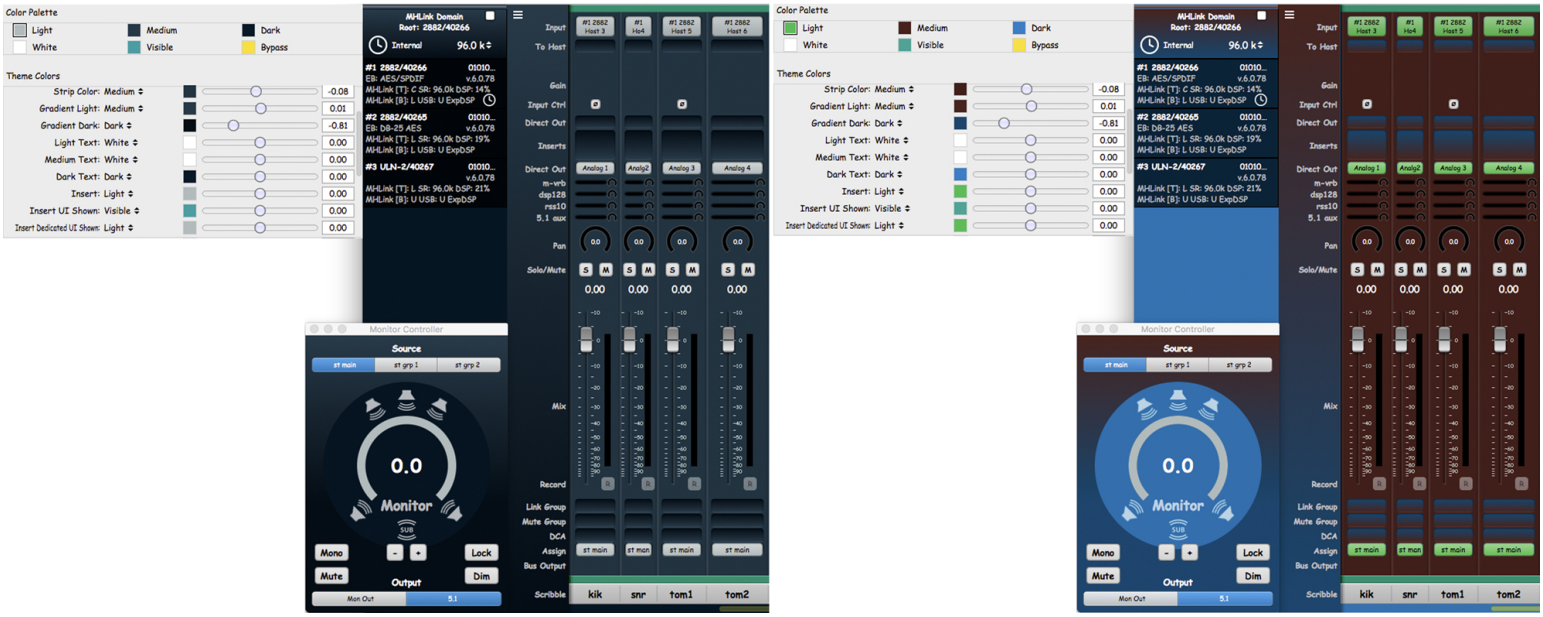
At the same time, you can make individual adjustments to any of the Theme Colors independently of their Palette group, adding a little variety and differentiation between similar UI elements while still keeping them visually related.

Clicking a square color button brings up color selector specified as your color picker (Color Picker details on the next page). Opacity controls within the color selectors are respected, so you can create color blending effects for added depth.

- **Save and Load Theme**

  The controls for loading and saving themes are placed at the bottom of the Appearance preferences pane.
  - **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 ~/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).

- **Color customization hints/suggestions**



Just three little changes...

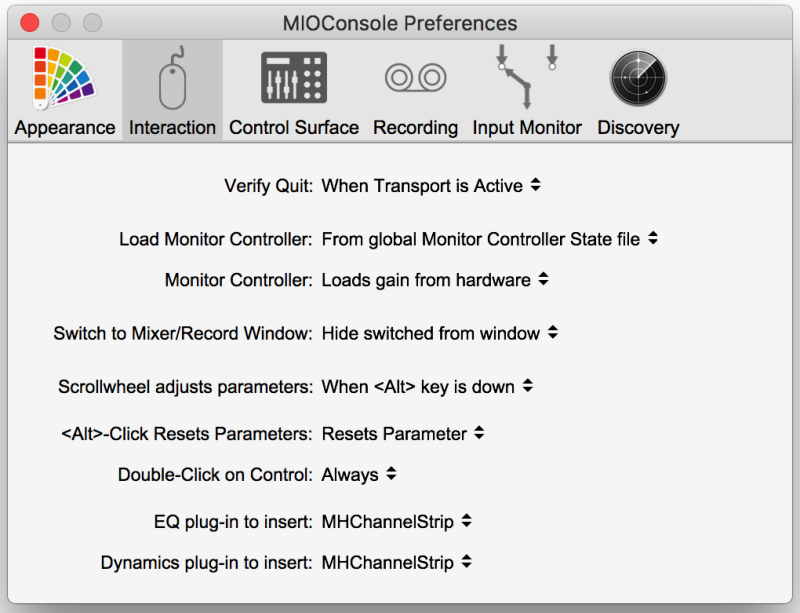
The above (apologetically unattractive) example illustrates how much trouble you can get into with just thirty seconds of downtime between takes. Starting with the default color scheme, quick color wheel tweaks were made to the Light, Medium and Dark Color Palette group controls. You can see the changes 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.

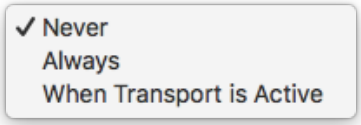
Interaction Preferences



Interaction preference pane

- **Verify Quit**
  The "Verify Quit" preference instructs MIOConsole3d to show a warning dialog to help prevent accidentally quitting MIOConsole.

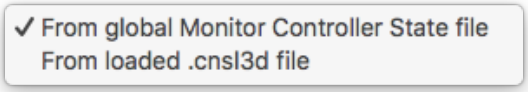
  There are three options:



'Verify Quit' behavior selector

  - **Never** is the default. When selected, quitting MIOConsole will quit without a warning.
  - **Always** will always show a warning dialog before quitting MIOConsole.
  - **When Transport is Active** will show a warning prompt only when Session transport is actively recording or playing back. No warning will be given if quitting when the Session transport is stopped.

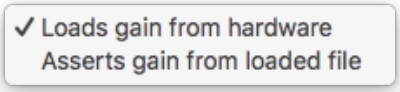
- **Load Monitor Controller**
  MIOConsole3d maintains a 'last unsaved state' file in your Preferences folder - essentially a snapshot taken of every Console parameter including Mix desk, Cues and Monitor Controller at the time you quit the MIOConsole3d application. This is so the next time you launch 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.



'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.

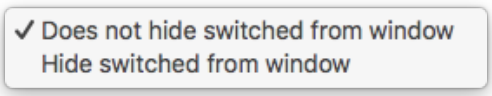


'Load Monitor Controller: Apply Gain From...' behavior selector

The default selection "Loads gain from hardware" maintains your current monitor levels even when loading a new monitor configuration from a saved console file.

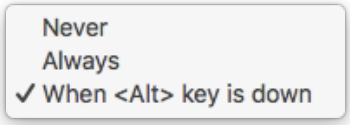
"Asserts gain from loaded file" applies the monitor levels from the loaded file to your monitoring chain. When using this option, it is best to double-check the levels it loads to your mains and Cue feeds to head off any unintended surprises.

- **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.



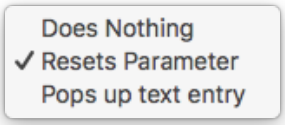
...when switching between *Mixer* and *Session* windows...

- **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.



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".



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.

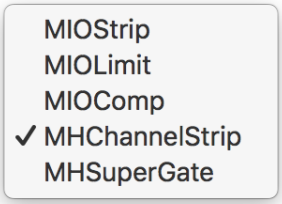
- **EQ plug-in to insert**
  Set to "MHChannelStrip" by default, open the drop-down menu to select the MIOConsole EQ processor you would like to open when clicking an empty mixer strip EQ Curve display.



EQ Curve default selector menu

This list will grow as new EQs are added to MIOConsole3d.

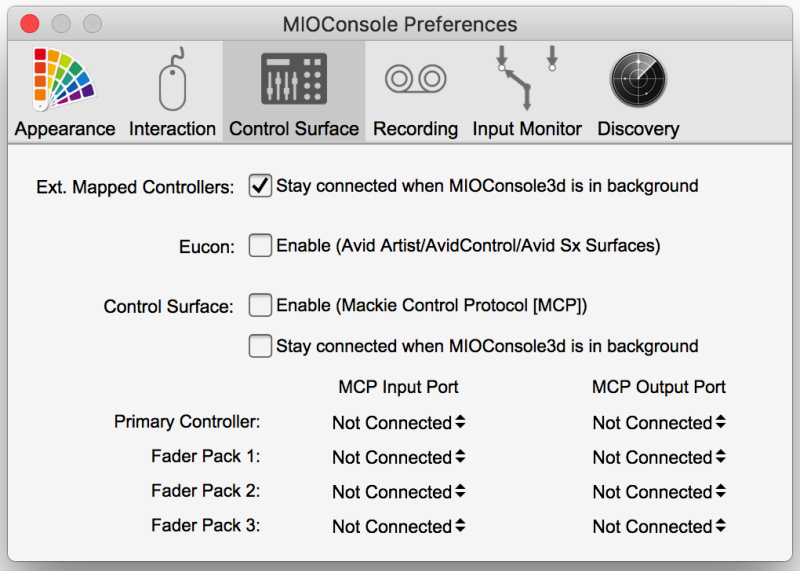
- **Dynamics plug-in to insert**
  Set to "MHChannelStrip" by default, open the drop-down menu to select the MIOConsole dynamics processor you would like to open when clicking an empty mixer strip Dynamics Curve display.



Dynamics Curve default selector menu

This list will grow as new dynamics processors are added to MIOConsole3d.

Control Surface Preferences



Control Surface preference pane

Detailed information on External MIDI, EuCon and Mackie Control Protocol [MCP/MCU] Control Surface operations is available in the Control Surface Support chapter.

Note that the MIOConsole3d application must be running on the host computer for external control of any type to be relayed to the 3d hardware.

- **Ext. Mapped Controllers** • **Stay connected when MIOConsole3d is in background**

  MIOConsole3d monitors all available MIDI ports on the host computer for Note On/Off, CC(7-bit), Aftertouch, Program Change, Channel Pressure, Pitch Wheel, and SysEx commands.

  When MIDI Learn is invoked on a supported UI element, MIOConsole3d automatically assigns the first MIDI command it receives to that element. When 'Set', that parameter is mapped to that external MIDI control and responds to MIDI control when MIOConsole3d is in the foreground.

  Stay connected when MIOConsole3d is in background is unchecked by default.

  When checked, MIOConsole3d will also respond to external MIDI control when MIOConsole3d is hidden or in the background. Uncheck this if you will use the same MIDI control ports in another host program too.

  This preference does not affect the operation of EuCon or MCP control surfaces.

- **EuCon** • **Enable (Avid Artist/Avid Control/Avid Sx Surfaces)**

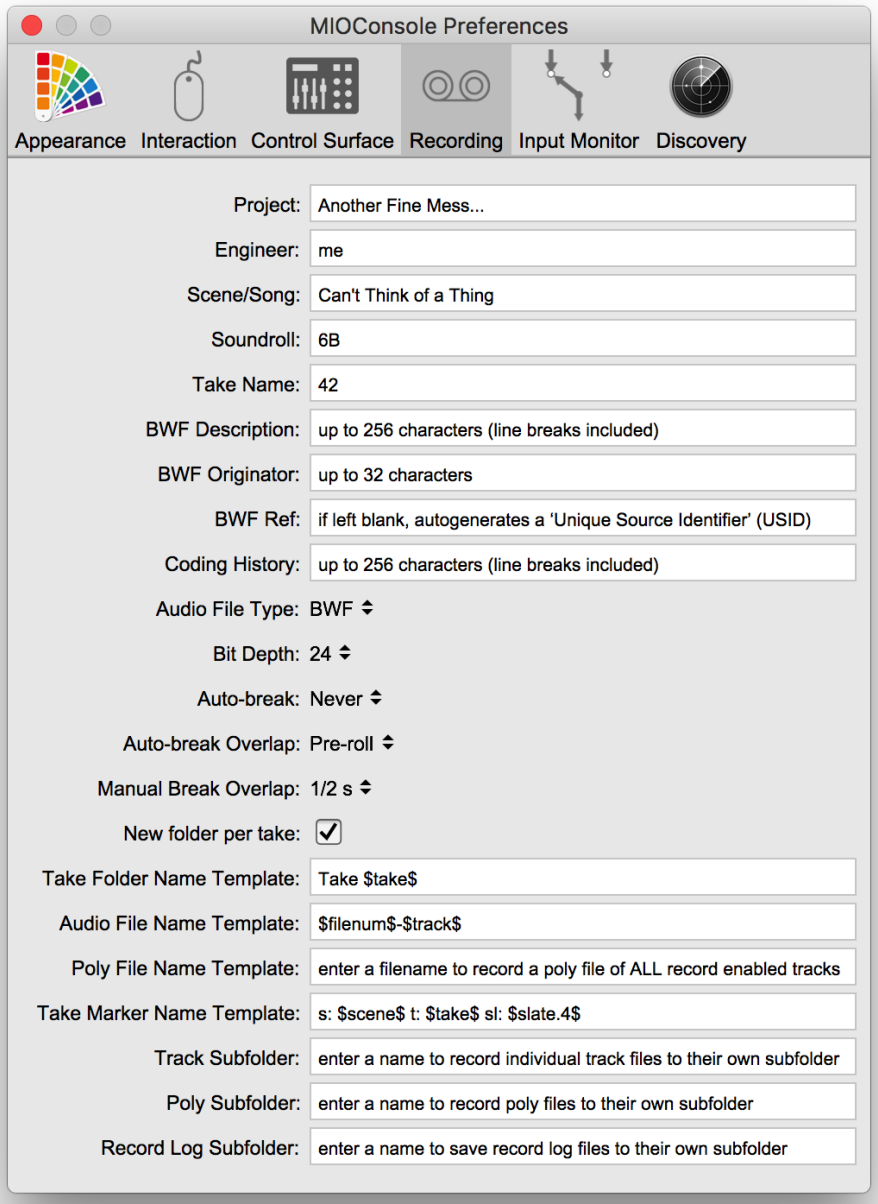
  When checked, MIOConsole3d will automatically connect to EuControl services running on your host computer. This box is not checked by default. If you have a EuCon control surface, you must check this box in order to use it with MIOConsole3d. If you do not wish to use your EuCon control surface with MIOConsole3d, uncheck this box and MIOConsole3d will disconnect from EuCon services.

- **Control Surface** • Enable (Mackie Control Protocol [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.

  - **Stay connected when MIOConsole3d is 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.

Recording Preferences



Recording preference pane: Filenaming, Filetype and Metadata 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.

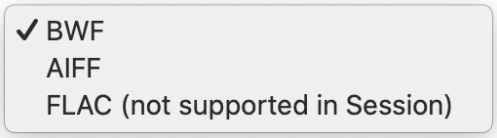
MIOConsole3d Session propagates all project, Session Metadata and Take Notes information to iXML file header chunks for all BWF and poly soundfiles.

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.



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.

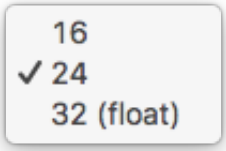
  "32 (float)" audio bit depth is not available for the AIFF file format.

- **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.



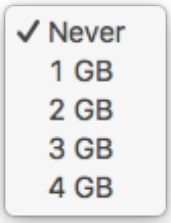
Audio Bit Depth selector

- **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.

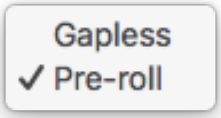
  There are a number of reasons you might consider breaking up recorded audio files into smaller segments:

  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.



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.

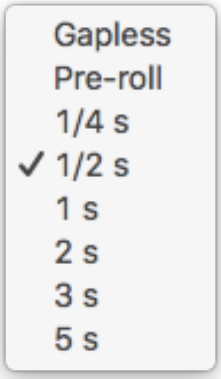


Auto-break Overlap selector

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.



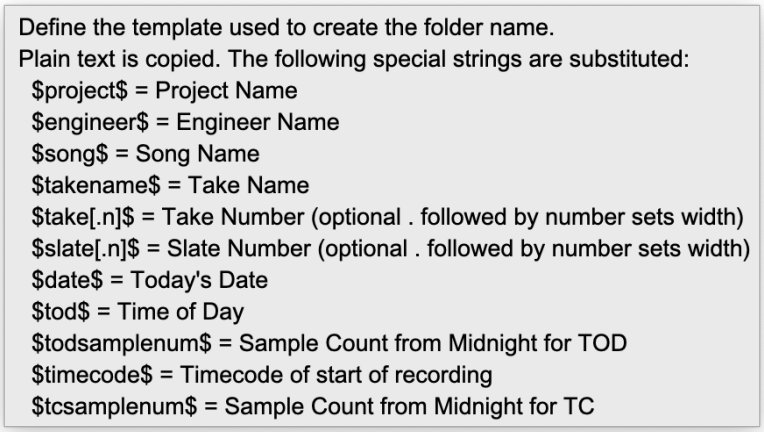
Manual Break Overlap selector

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:



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".

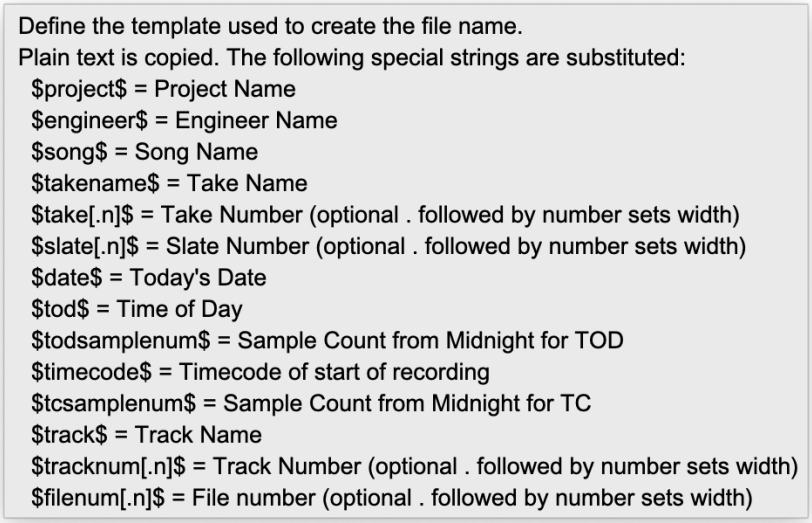
tod says: paste the current Time Of Day into the folder name. The 'time' format is hours-minutes-seconds using a 24-hour clock, so 4:28 and 17 seconds in the afternoon would appear in the folder name as "16-28-17".

$todsamplenum$ 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:



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.

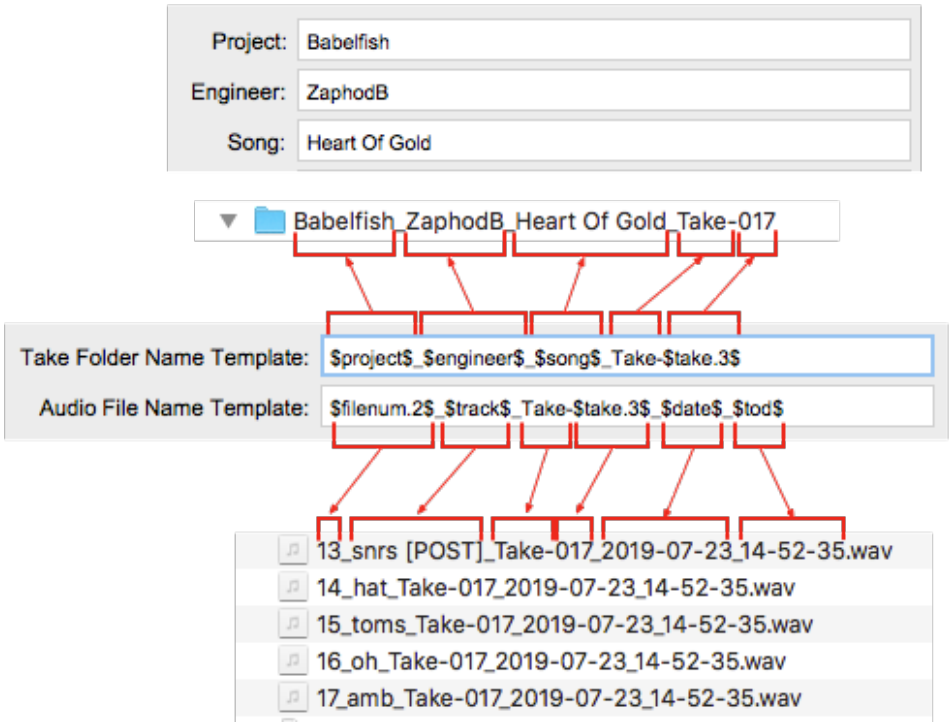
$tracknum[.n]$ numbers all tracks shown in the Session Track Overview, whether they are Record-Armed or not. In cases where not all tracks are armed for any given take, the recorded track numbers will have gaps in their sequence, but they will always match the same source audio and **$track$** name from 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.



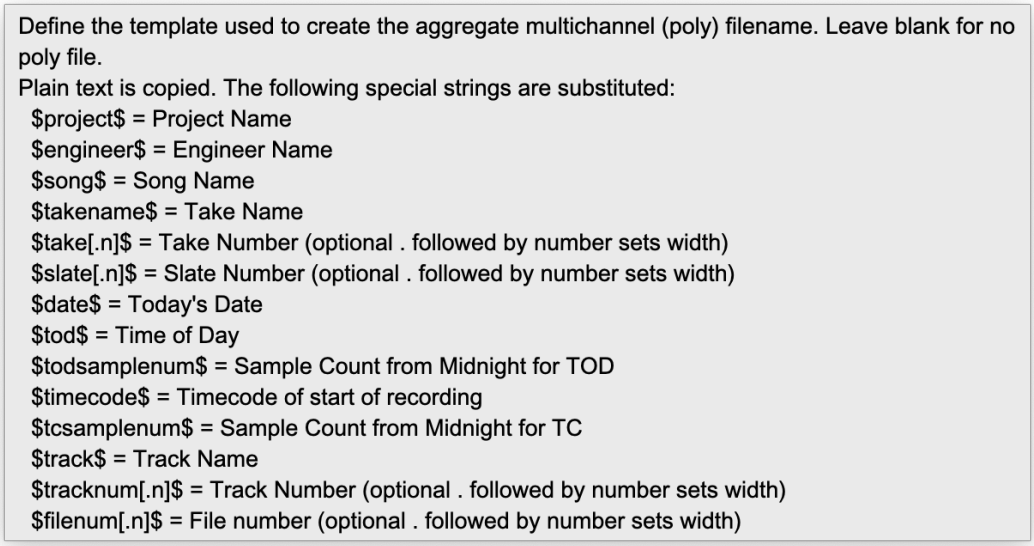
Name Templates script example

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.

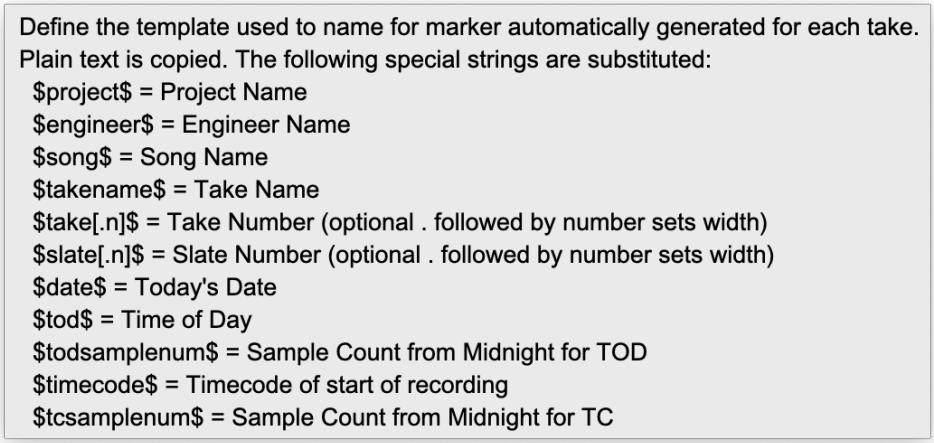
  The Poly File Name Template follows the same script logic as the Audio File Name Template. Here is the tooltip pop-up menu:



Poly File Name Template **tooltip**

- Use the **Take Marker Name Template** field to markers generated at the head of each take.

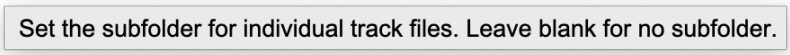
  The Take Marker Name Template follows the same script logic as the Take Folder Name Template. Here is the tooltip pop-up:



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:



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:



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:



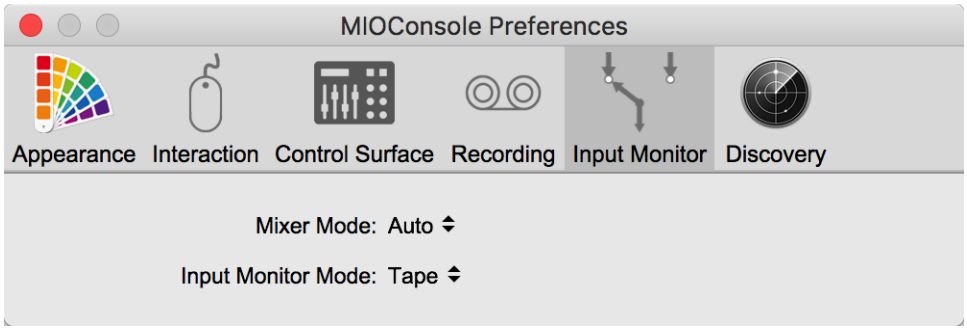
Record Log Subfolder **tooltip**

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.

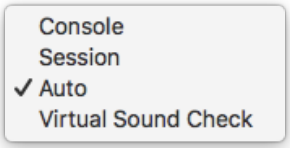


Input Monitor preference pane

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.



- **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.
- **Virtual Sound Check** is a special mode to support the playback of large channel count sessions recorded with MIOConsole Session when using an external console for mixing. A common use case would be where an external console is used for main FOH mixing while sending 128 channels of MADI into MIOConsole for recording and MADI back out to the external console for monitoring playback. This is useful when recording shows live, and then using the recording of the previous performance to provide a "real life" sound check of the FOH/Monitor system in-situ. While this removes the line-

check (e.g. it doesn't include checking whether or not all signals are making it from the mics/sources to the console), it allows for checking the mix and drive-rack settings in the hall with representative material from the previous show.

In Virtual Sound Check mode, all MIOConsole input channels should use "To Host" in pre-insert mode to route audio streams to the host computer for recording, and the "Pre-Insert Direct Outs" should be routed back to the the external console Tape Return inputs for playback monitoring.

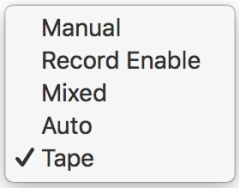
When Session is in record, or the transport is not engaged, the strip inputs will be routed (normalled) to the "Pre-Insert Direct Outs", so you can run the show through the MIO without having to re-patch the external console (e.g. you simply insert the MIO in the send/return path for all channels on the external console).

When Session is in playback, the playback from the Session tracks will be routed to the "Pre-Insert Direct Outs", replacing the normalled input signals. This will route the discrete channel Session playback to the external console's channel returns, thus allowing you to run the full show from the recorded session.

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.



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.

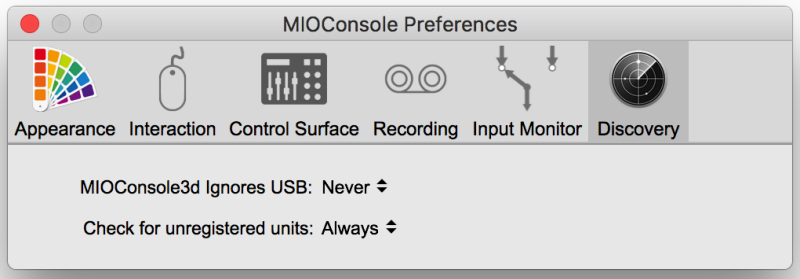
Input Monitor Mode	Record Enable	Transport: Stop	Transport: Play	Transport: Record
Manual	OFF	Muted	Playback	Playback
Record Enable	OFF	Muted	Playback	Playback
Mixed	OFF	Muted	Playback	Playback
Auto	OFF	Muted	Playback	Playback
Tape	OFF	Muted	Playback	Playback
Manual	ON	Muted	Playback	Playback
Record Enable	ON	Input	Input	Input
Mixed	ON	Input	Input + Playback	Input
Auto	ON	Muted	Playback	Input
Tape	ON	Input	Playback	Input

Table 6.1. 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 6.2. Mixer Mode: Auto

Discovery Preferences

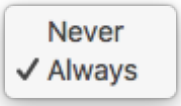


Discovery preference pane

- **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 *Always ignore* the computer USB ports 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.



Discovery: Parameter selection options

- **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.

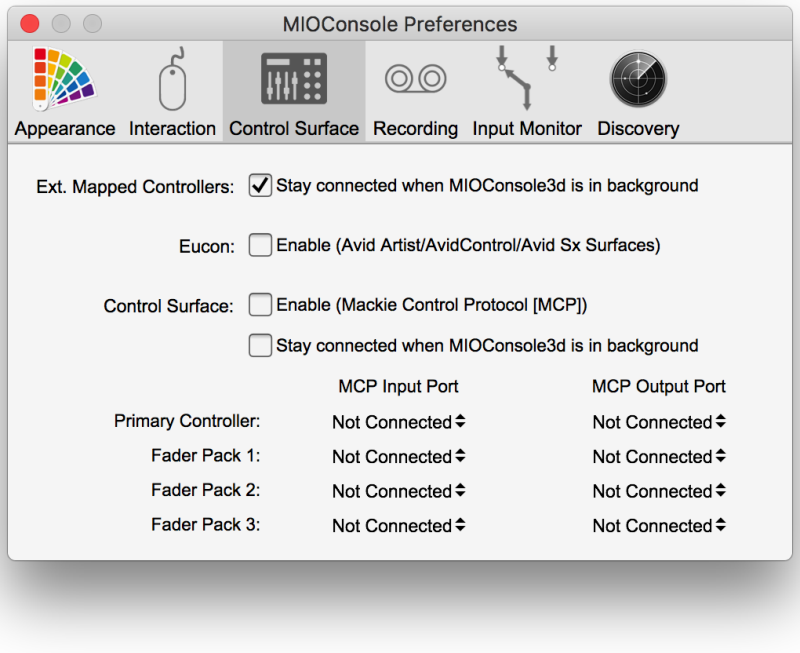
  Note that you can always register hardware or update your registration information manually using the menu bar: "MIOConsole3d: Register Attached Units..." command.

7. Control Surface Support

Control Surface Preference Pane

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:



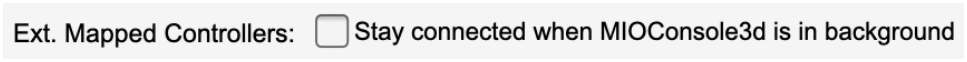
MIOConsole3d Preferences: Control Surface parameters

The following entries offer operational info supplementary to the MIOConsole3d Control Surface Preferences pane:

- **Ext. Mapped Controllers** • Stay connected when MIOConsole3d is in background

  MIOConsole3d monitors all available MIDI ports on the host computer for Note On/Off, CC(7-bit), Aftertouch, Program Change, Channel Pressure, Pitch Wheel, and SysEx commands.

  When MIDI Learn is invoked on a supported UI element, MIOConsole3d automatically assigns the first MIDI command it receives to that element. When 'Set', that parameter is mapped to that external MIDI control and responds to MIDI control when MIOConsole3d is in the foreground.

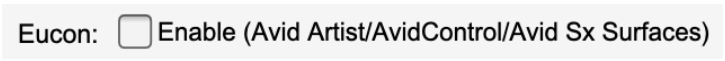


  Stay connected when MIOConsole3d is in background is unchecked by default.

  When checked, MIOConsole3d will also respond to external MIDI control when MIOConsole3d is hidden or in the background. Uncheck this if you will use the same MIDI control ports in another host program too.

  This preference does not affect the operation of EuCon or MCP control surfaces.

- **EuCon** • Enable (Avid Artist/Avid Control/Avid Sx Surfaces)



  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.

- **Control Surface** • Enable (Mackie Control Protocol [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.

- **Stay connected when MIOConsole3d is 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.

External Controller Support in MIOConsole3d

MIOConsole3d incorporates a suite of features added to provide flexible support for user-defined ad-hoc mapping of external controllers to virtually every aspect of the system.

We have added extensive instrumentation throughout to support binding of external controllers to specific controllable elements of the system. The mapping system has been architected to support multiple external controller types (such as MIDI, OSC, Front Panel User Mode).

This initial release provides support for mapping external MIDI controllers.

Mapping

Mapping is a software connection between the messages sent by a physical control on an external controler device and changes to a specific virtual control in the system (*this* physical knob/fader controls *that* console parameter).



Mappings are represented in MIOConsole as a set of filters that restrict the response to messages of the specific type emitted by the external controls.

Mappings can be made between external controls and the parameters, commands and various UI elements in the software.

Control Surfaces vs. External Controller Mapping

This mapping system differs from MIOConsole3d's existing support for EuCon and Mackie Control Protocol based control surfaces (also known as Control Surface Support). Both the new mapping system and the existing Control Surface Support run in parallel with each other, and any or all of the external control systems can be enabled and active at the same time.

The Control Surface support dynamically assigns physical controls from surfaces with well known layouts to the various supported virtual controls in the system (with support for paging and nudging, and context sensitive control modes).

The mapping system aims to provide a different capability.

Rather than having a dynamic mapping based on the control surface layout and Metric Halo's mapping choices, the the new mapping system provides the user with the capability of making fixed mappings between arbitrary external controllers and specific parameters and/or commands in MIOConsole3d on the fly.

While you use the GUI to easily select what parameter or command you want to map to, the bindings you create are not dependent on the UI, so they remain active even when the UI is not on screen.

For global elements of the system (eg. Monitor Controller, Cue Controllers, Transport, Key Commands), this fixed mapping is quite natural. For more dynamic elements of the system (eg. strips, busses, plugins) the fixed mapping may feel limiting or surprising.

Think of it like a mapping for performing with a specific setup (performing in the generic sense, as this could just as easily apply to mastering as it does to performing on stage). You associate a fixed control on your external surface with a fixed element in the system.

The most important difference to keep in mind between EuCon and MCP Control Surface operations and free-form External Controller Mapping is a physical one:

Control Surface support is bi-directional, in that MIOConsole3d sends live data back to the control surface to sync meter displays on the control surface hardware and ensure that changes made in MIOConsole3d (such as touch selection, fader position and mix desk fader strip layout) are reflected in real time back on the control surface hardware.

External Controller Mapping is one direction only - the external control tells MIOConsole what to do, period. The upside is, it allows any gizmo that outputs supported MIDI messages to become a remote controller for nearly any parameter within the MIOConsole3d mixer and processing environment - something EuCon or MCP is unable to manage.

Check the boneyard

Since the system utilizes virtually any MIDI message for control, any of a variety of MIDI control devices you may have collected over the years can now be given a new life running MIOConsole3d.

No controllers - no problem...

If you don't have any MIDI controllers available in your personal inventory of gear, there are a huge number of different MIDI controllers readily available from different vendors, many of them are very inexpensive with direct self-powered USB connections. Pretty much whatever generic MIDI controller(s) you choose, it'll be plug and play.

Another big upside is, your controller MIDI can be easily recorded, edited and played in your DAW as realtime automation control for everything in the MIOConsole3d mixer environment.

MIDI Controller Mapping in Practice

Mappings can be established in one of two ways:

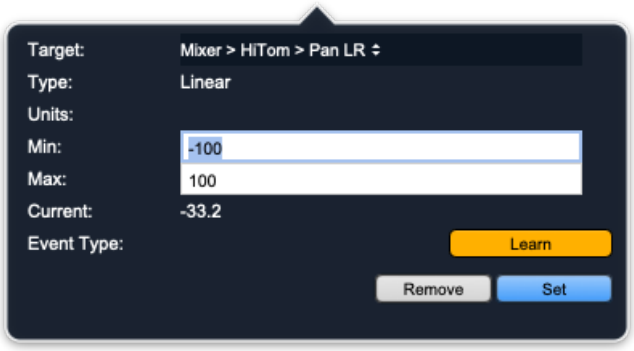
1. Per-control assignment
2. Auto-assignment mode

Per-control Assignment

Per-control assignment is initiated by typing **⌘1** while hovering the mouse over the control that you want to map to.

The control you are hovering over can be any on the list below, or the name of a command in the command-keys window.

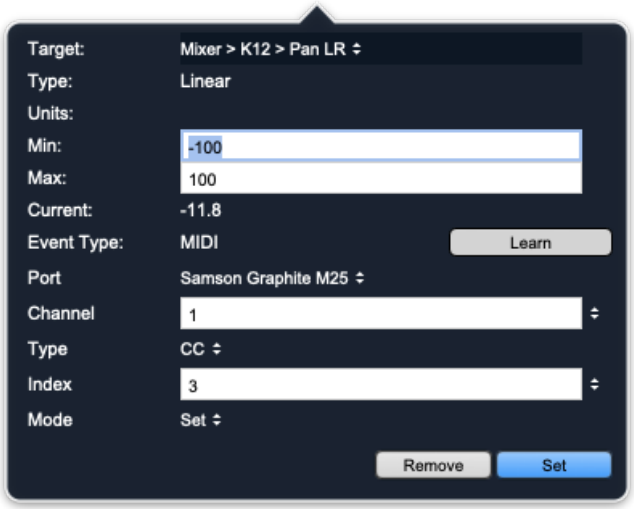
When you hit **⌘1**, an assignment window will popup pointing at the control.



If the control doesn't have a mapping, the assignment window will automatically be in Learn mode (e.g. the Learn Button will be lit up). While in Learn mode, you can trigger a MIDI message by touching a button, key knob, expression controller or the like on the external MIDI controller, and the assign window will capture the message and set up the mapping for you. You can finalize the mapping by clicking on the "Set" button (or hitting the return/enter key), or you can further edit the mapping parameters using the controls in the assignent window before finalizing the mapping.

If you want to cancel the mapping rather than finalizing it, you can click the "Remove" button. This can also be used to delete an existing mapping.

If the control already has a mapping, the Learn Button will be light grey and the message parameters will be listed at the bottom of the assignment window like this:



If there is a mapping present, you can either edit it via the controls in the assignment window, or you can click the "Learn" button to enter Learn mode (as described above).

If the UI item you are hovering over when you hit **⌘1** has multiple parameter targets associated with it (for example a surround panner, or an aux send control), then all the available parameters will be in the "Target:" popup menu. You can click on that popup to switch between the available targets



If you select a new target, the current controller filter will be saved for the target you are switching from (as if you had clicked the "Set" button); this allows you to hit **⌘1**, touch an external control to assign the first target of the UI control, then use the popup menu to switch to the next target and repeat the process, clicking the "Set" button once you have set the controllers for all the targets you wish to assign.

The MIDI mapping engine supports the following MIDI messages:

- Note On/Off
- CC (7-bit)
- Aftertouch
- Program Change
- Channel Pressure
- Pitch wheel
- SysEx

By default, the learned mappings filter MIDI events to the specific message from the specific port that the learned message was received from. So if you, for example, learn CC#0, on channel 2 of Port "Sampson Graphite M25", even if you send a CC#0 from a different port, or even the same port on a different channel the filter will treat it as a different controller.

However, you can if you choose set the various filters to “Any”, and if a given filter is set to “Any”, then the mapping will trigger as long as the rest of the filters match. So you could set the port to “Any”, and then any CC#0 on channel 2 would trigger this mapping.

For each message, the following processing modes are available (although some messages and/or targets only support a subset of the processing modes):

- Set Value: Set the value of the underlying parameter based on the value in the message
- Toggle: Toggle parameter on/off each time the message is received (works best with Note/Program Change/SysEx)
- Momentary: Turn parameter on/off on press/release (works best with note messages)
- Trigger: Fire associated command (Note/Program Change/Sysex)
- V-Pot: Only for CC; CC must use Mackie Control Protocol V-Pot encoding

The system will attempt to select the correct processing mode automatically, but you can set it via the "Mode" popup menu in the Assignment window.

Each mapping also includes a user-configurable range for mapping the external controller value to the underlying parameter's value when the Mode is “Set”.

The Min setting will be the value used when the external controller is set to its minumum value. The Max setting will be the value used when the external controller is set to its maximum value, and in-between values for the external controller will interpolate between the user specified min and max (using the parameter’s built-in taper) for the underlying parameter.

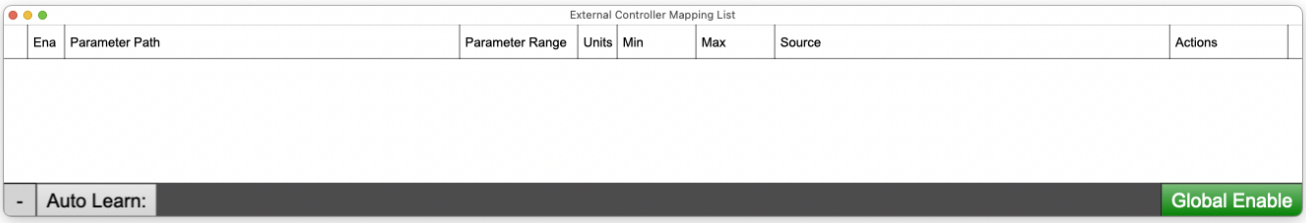
Note: You can set the Min to be larger than the Max, and this will have the effect of inverting the sense of the parameter, such that turning a knob to the left-to-right will decrease the number value rather than increase it.

This user-defined range mapping can be used to good effect to do things like:

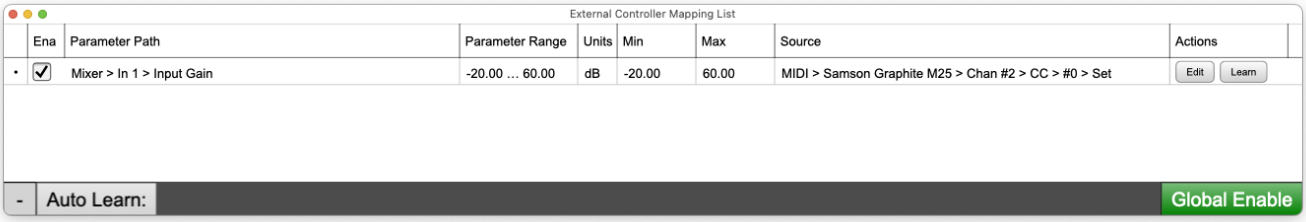
- Limit the range of a control - for example, map a subset of MC gains to a MIDI CC
- Invert a control (e.g. turn a bypass into an enable or vice versa)
- Stagger or offset ganged controls

The External Control Mapping List window

We have added a new window (External Control Mapping List) that shows all of the defined mappings in the system; when there are no mappings defined, it looks like this:



After you define a mapping (e.g. via **⌘1** or the **Auto-assignment mode** shown below), the mappings will be added to the list:



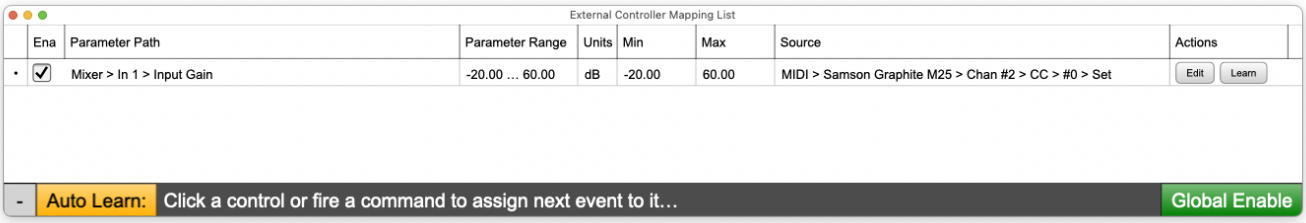
You can sort the columns by clicking on them, (Ena, Parameter Path, Source). You can edit a mapping directly from the list by clicking the associated mapping button, or you can relearn the mapping directly via the learn button. The Enable checkbox allows you to disable (or re-enable) the mapping without removing it. The Min and Max columns in the table are editable directly from the table without having to invoke the mapping editor.

Finally, you can delete a mapping in one of the following three ways:

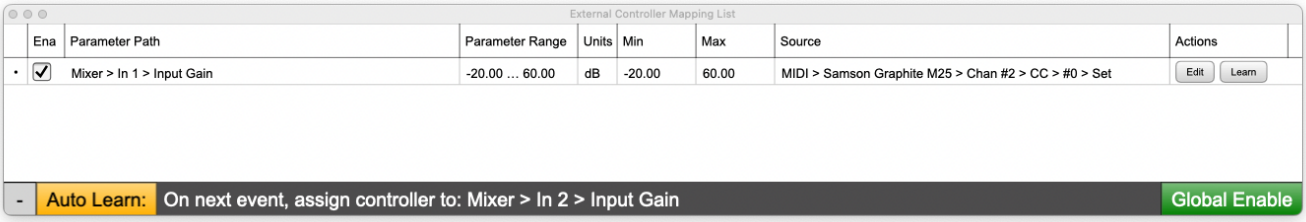
- **⌘1** or click the edit button, and click the Remove button
- Right-click the dot in the left most column and select "Delete Row"
- Click and then command- or shift-click the dots in the leftmost column to select multiple rows, and then click the **–** button in the very bottom left hand corner of the window (next to **Auto Learn:**). Keyboard "Delete" does not function in this window.

Auto-assignment mode

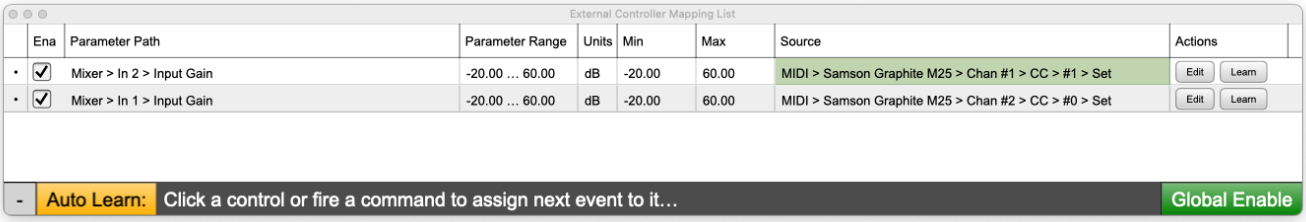
This is initiated by typing **^2**. The External Controller mapping list window will be shown if it was not already visible and the "Auto Learn:" button in that window will turn yellow.



As you touch controls or fire commands in the App, the text next to the button will be updated to reflect the current command/parameter that will be mapped on the next MIDI message.



When the next MIDI message is received, it will be mapped to the current parameter (as if you used the Per-control assignment mode, and immediately hit the set button after the MIDI message is received).



You can then touch another control, and send a MIDI message to map it. Lather, Rinse, Repeat. When you are done mapping controllers, you can click the "Auto Learn:" button or hit ^2 again to toggle off the auto-learn mode.

Note: When an external controller's message matches the filter assigned to a target the associated "Source" field in the External Control Mapping List will flash green to let you see that the messages are being received and matched.

The green **Global Enable** button in the lower right corner of the **External Control Mapping List** window is connected (and synchronized) with the Global Enable External Mapped MIDI Controllers menu command. When the button is green, the mappings are active and will respond to messages. When the button is light grey, the mappings are globally disabled. You can use this to turn off all the mappings without having to delete them or disable them 1 by 1.

Multiple Mapping

Each parameter or command can only have one external controller mapped to it, but any external controller can be mapped to multiple parameters, so you can control more than one parameter with one external controller.

In addition, each mapping can have an independent processing mode, and can also have a different mapping range. One useful thing you can do is set on/off controls to have opposite mappings with one external source. This allows for arranging multiple bypasses or enables to be operated in opposition to each other with one CC or note controller.

External Controller Menu Commands

There are 4 new menu commands in the MIOConsole3d application **Edit** menu that relate to external controllers:

- **Assign Controller for item under mouse...** - Default key command ⌘1
  Displays the Controller assignment window for the control that is under the mouse pointer when the command-key is pressed; not very useful as a menu command, but it is in the menu to make it easier to see the associated command key.
- **Auto Learn Controller for last parameter** - Default key command ^2
  Toggles Auto Learn mode on and off.
- **Show/Hide External Controller Mapping List Window** - Default key command ^3
  Toggles the visibility of the External Controller Mapping List Window.
- **Global Enable External Mapped MIDI Controllers** - Default key command ⌘^M
  Enables (when checked) or Disables (when unchecked) responding to MIDI mappings. You can use this to temporarily disable all MIDI mappings without having to delete them or disable them individually. This item is checked by default, and is stored as an App wide preference (and not as parameter of the .cnsl3d file).

Externally Controllable UI elements

- All mixer strip parameters for Input, Bus, and DCA strips including:
  - Fader Gain
  - Pan
  - Mute
  - Solo
  - Record Enable
  - Input Enable
  - Preamp Gain
  - Polarity Invert
  - Phantom Enable
  - Insert Bypasses
  - Hard Mute
  - Aux Send Gain
  - Aux Send Pan
  - Aux Send Mute
  - Aux Send Solo
- Mute Group Enables
- All Plugin Parameters
- Plugin Bypasses
- I/O Insert Bypasses
- Graph Insert Bypasses
- Sub-Graph Bypasses
- Monitor Parameters including:

    - Monitor Gain
    - Input Source selector
    - Output Destination
    - Monitor Graph Bypasses
    - Speaker Mutes
    - Mono
    - Mute
    - Dim
- Cue Parameters including:
    - Cue Gain
    - Cue Source selector
    - Cue Graph Bypasses
    - Mono
    - Mute
    - Dim
- Talkback Enable
- Listenback Enable
- Session Related:
    - Transport Controls
        - Play
        - Stop
        - Rewind
        - Record
        - Pause
        - Loop Play Enable
    - Metronome Enable Toggle
    - Grid Enable Toggle
    - Fit Tracks Vertical
    - Fit Tracks Horizontal
    - Vertical Zoom
    - Horizontal Zoom
- All Commands that can have a keystroke assigned to them

Elements not yet controllable are:

- Input and Output gains in the Analog I/O window
- Strip Selection
- Headamp input mode
- Internal Sample Rate
- Clock Source
- Link Group Enable/Disable
- Strip Input Source
- Strip Direct Out Destination
- Mix bus output assigns
- Strip to Bus assigns
- Cue controller dim levels
- Talkback Input gain
- Playhead position
- Recording mode controls
- Digital I/O Output mode controls
- Preferences

EuCon in MIOConsole3d

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.

MIOConsole3d 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 9.2.6)
    - 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

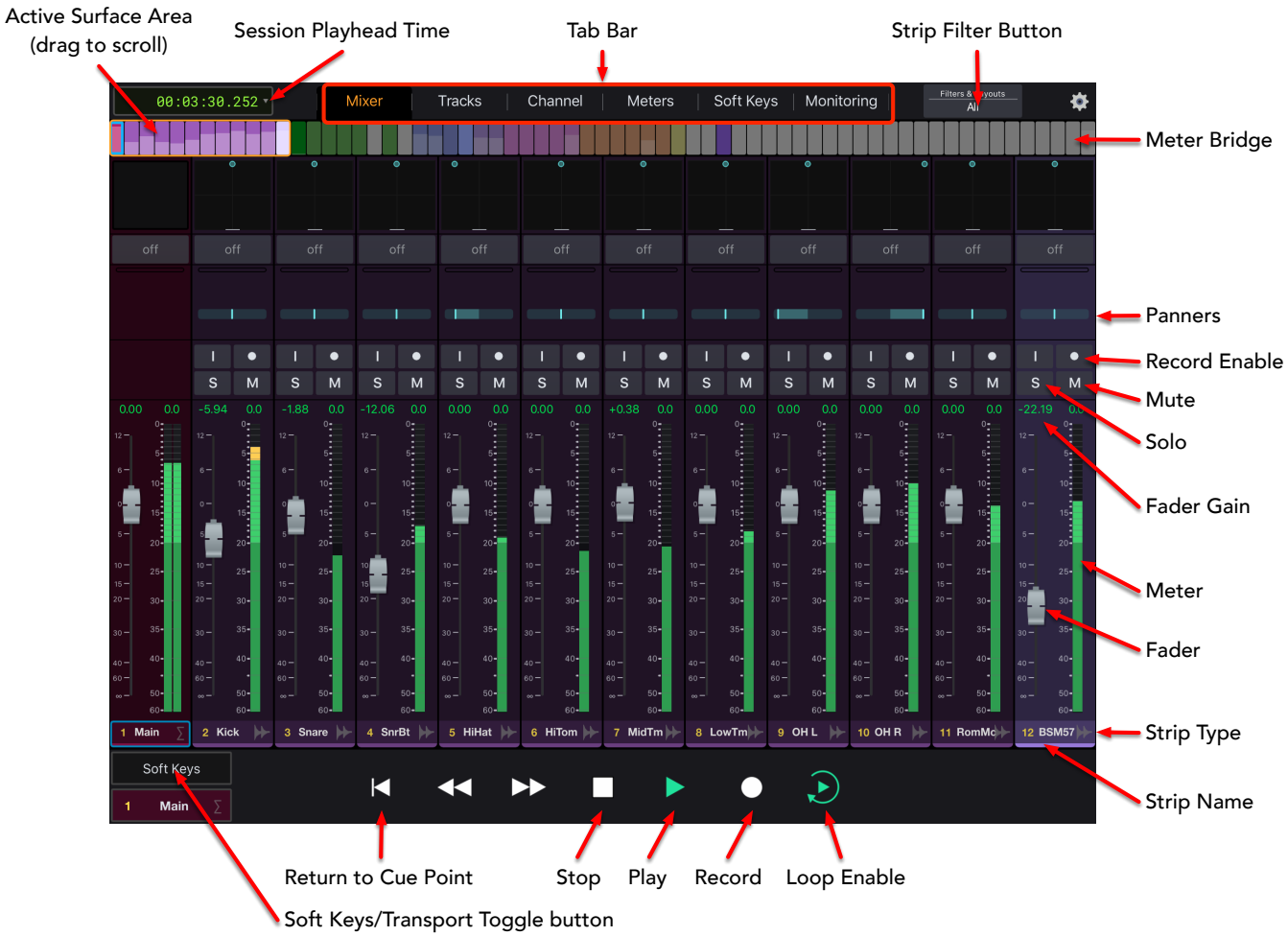
You can find out how to download Avid Control and the EuCon software from Avid's Avid Control page.

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.



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.

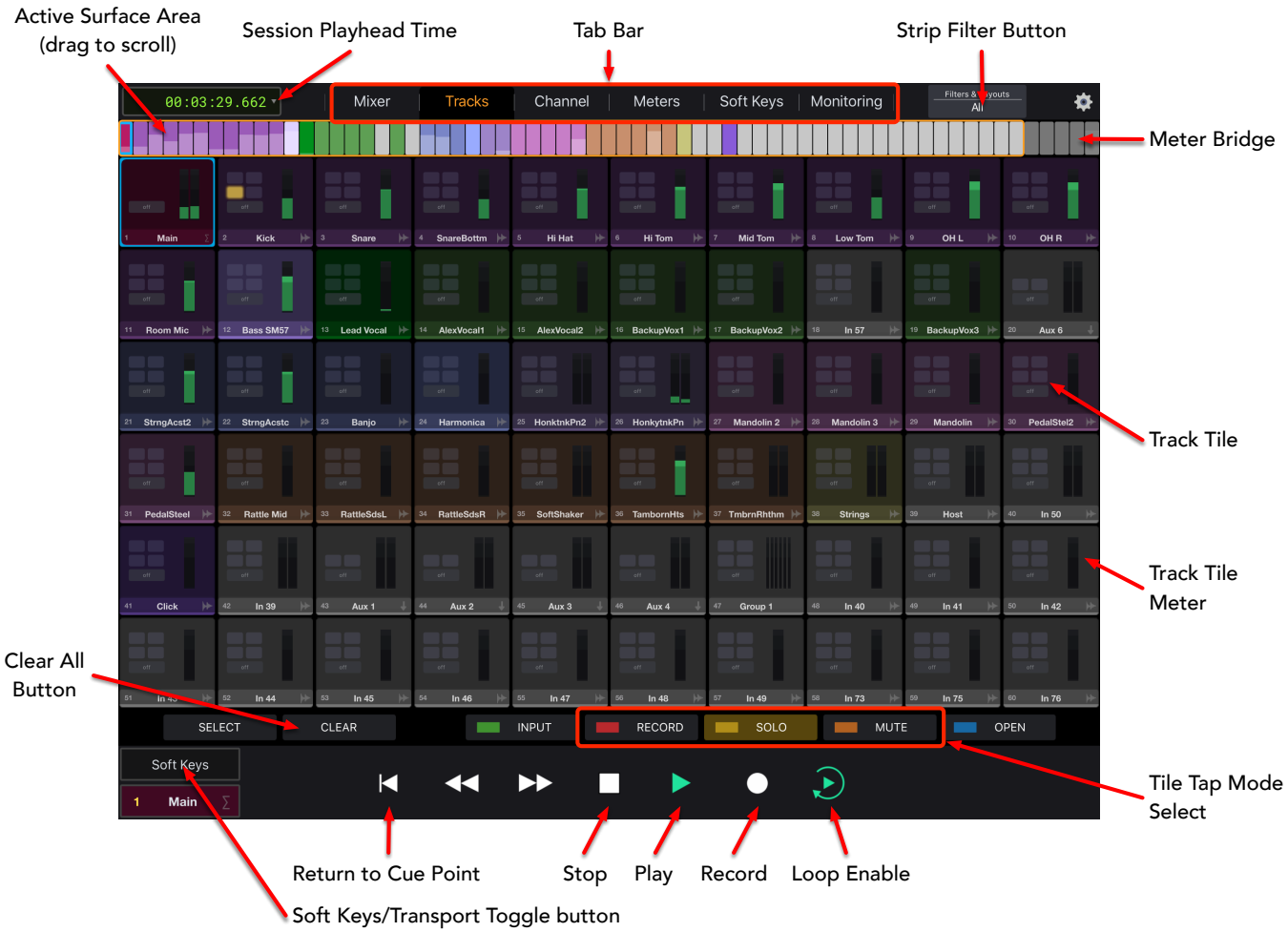
When the Soft Keys controls are shown you can use the preconfigured soft keys to switch between the active bus shown on the faders. Unfortunately, EuCon does not provide a mechanism for MIOConsole3d to indicate which bus is currently active, so you will have to affirmatively set the current bus if you 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.

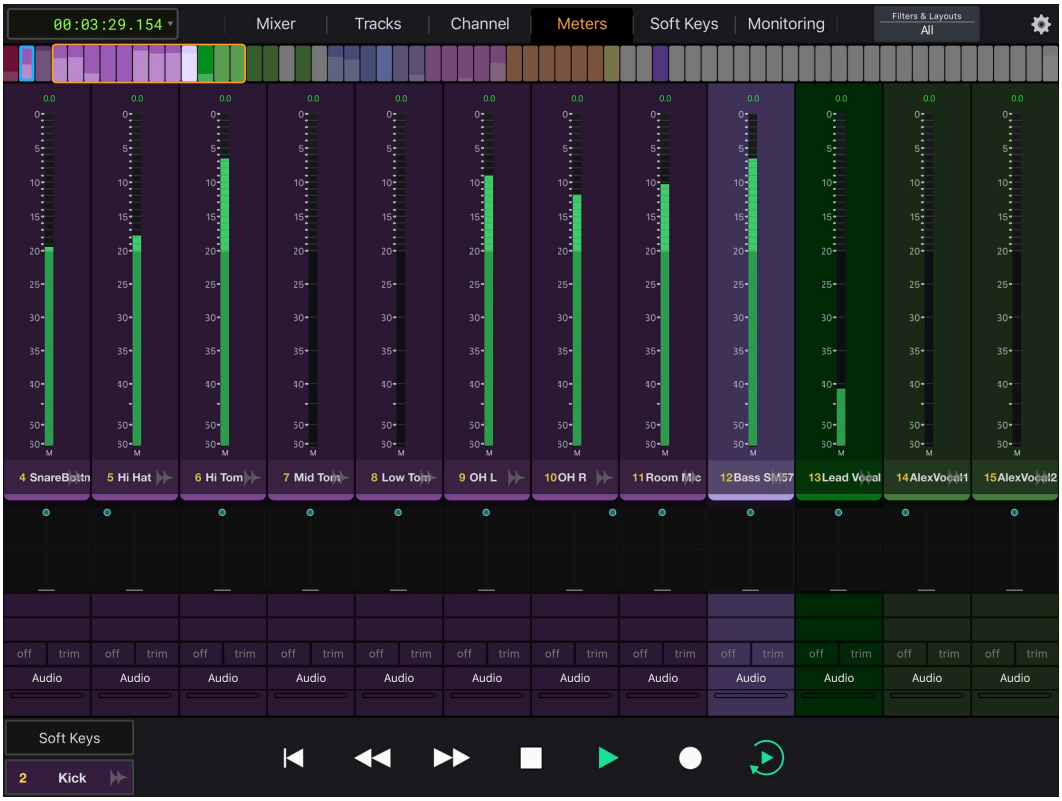
All the other elements of this page are the same as what you find on the Mixer pane.



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.



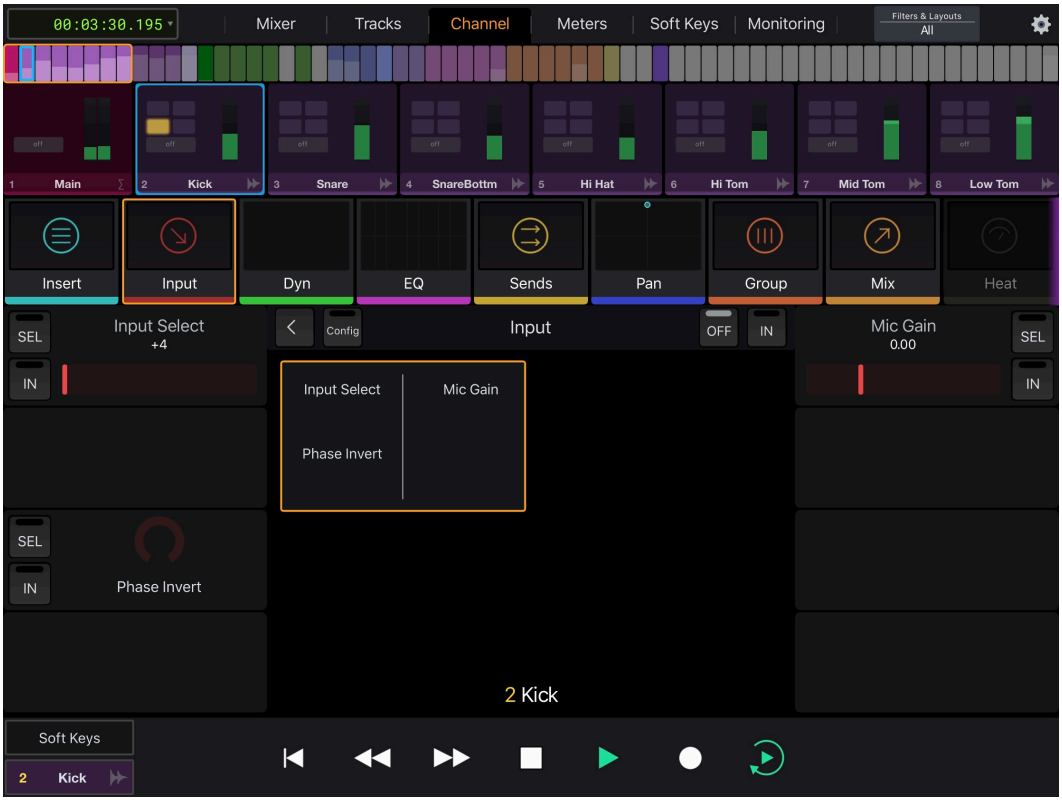
MIOConsole3d Meters View in Avid Control

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.



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.

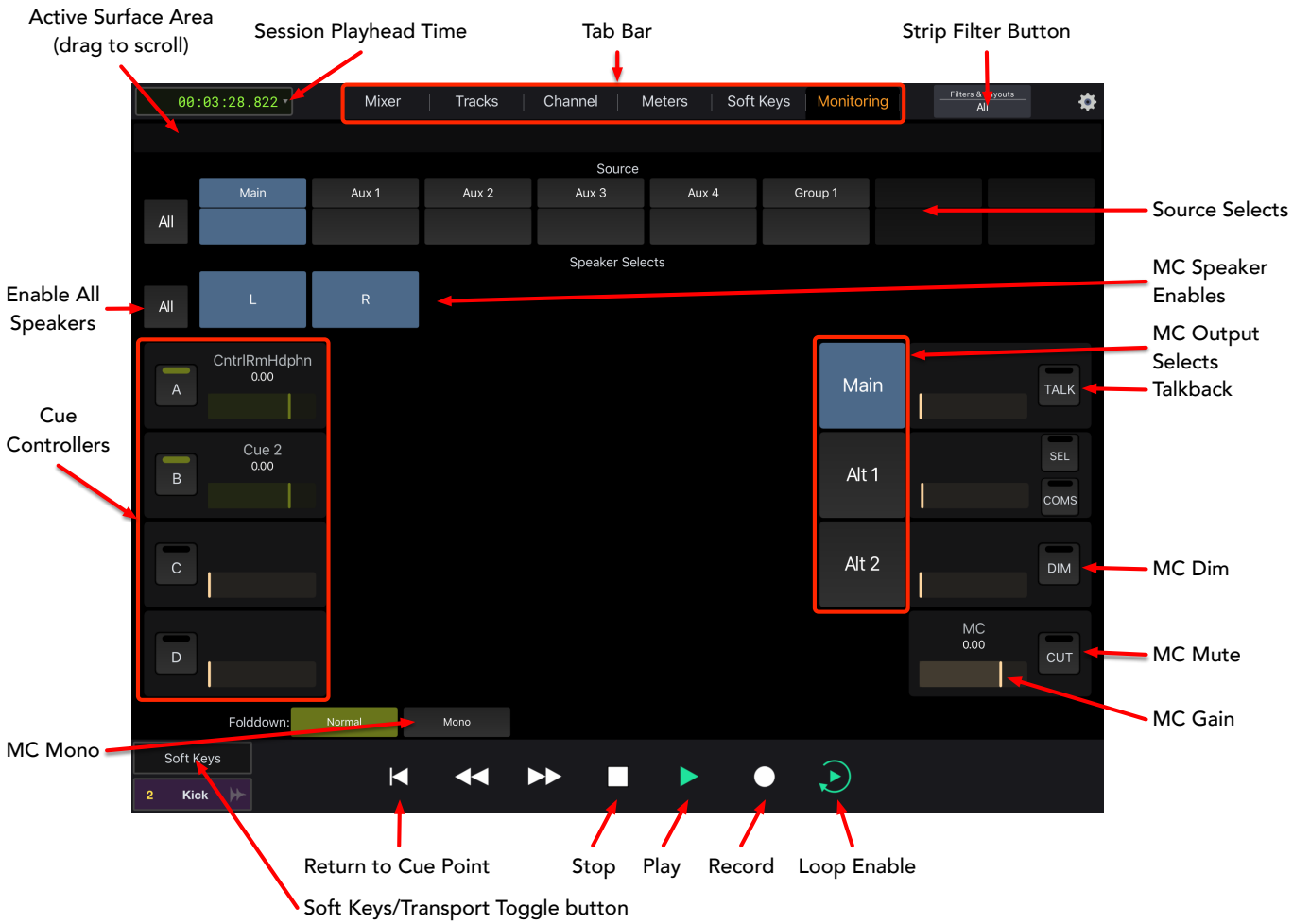
The next row down are the speaker enables; this area will automatically update based on the selected output path and the speakers in the path. When a button is illuminated, the speaker is enabled (not muted). Tap a button to mute it. Tap again to unmute. Tap "All" to enable all the speakers. This area 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.

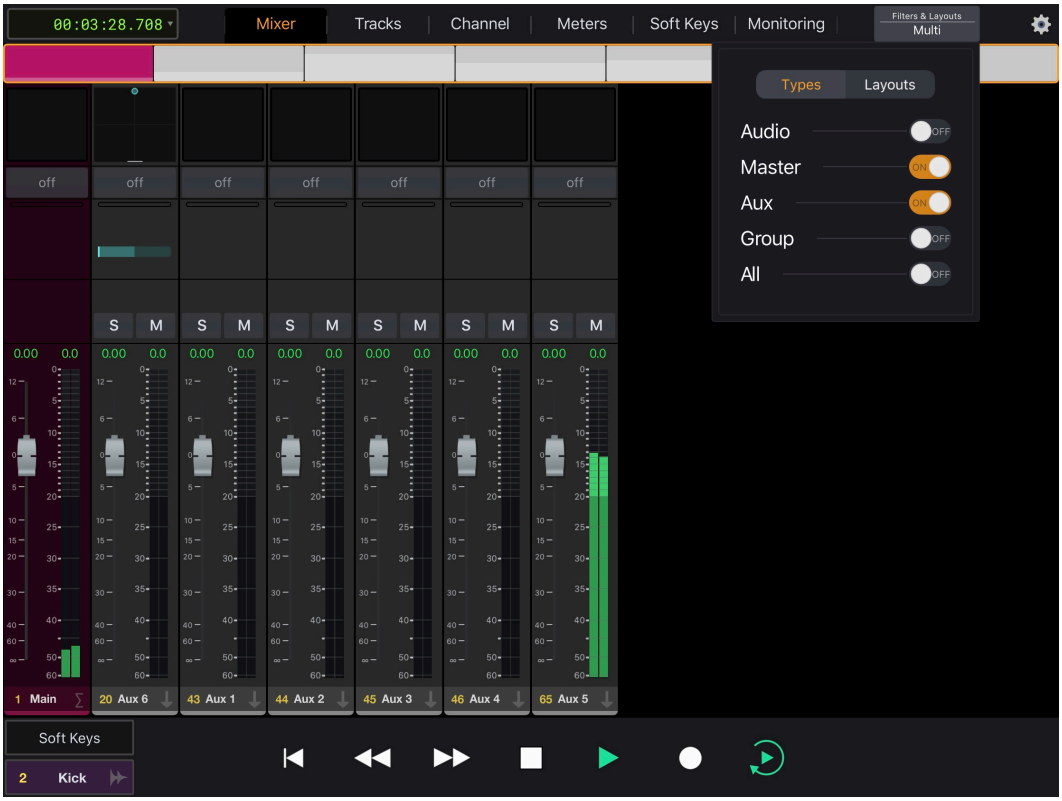
Finally on the left hand side, you find the Mute and Gain controls for the first 4 Cue Controllers you instantiate in MIOConsole3d.



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.

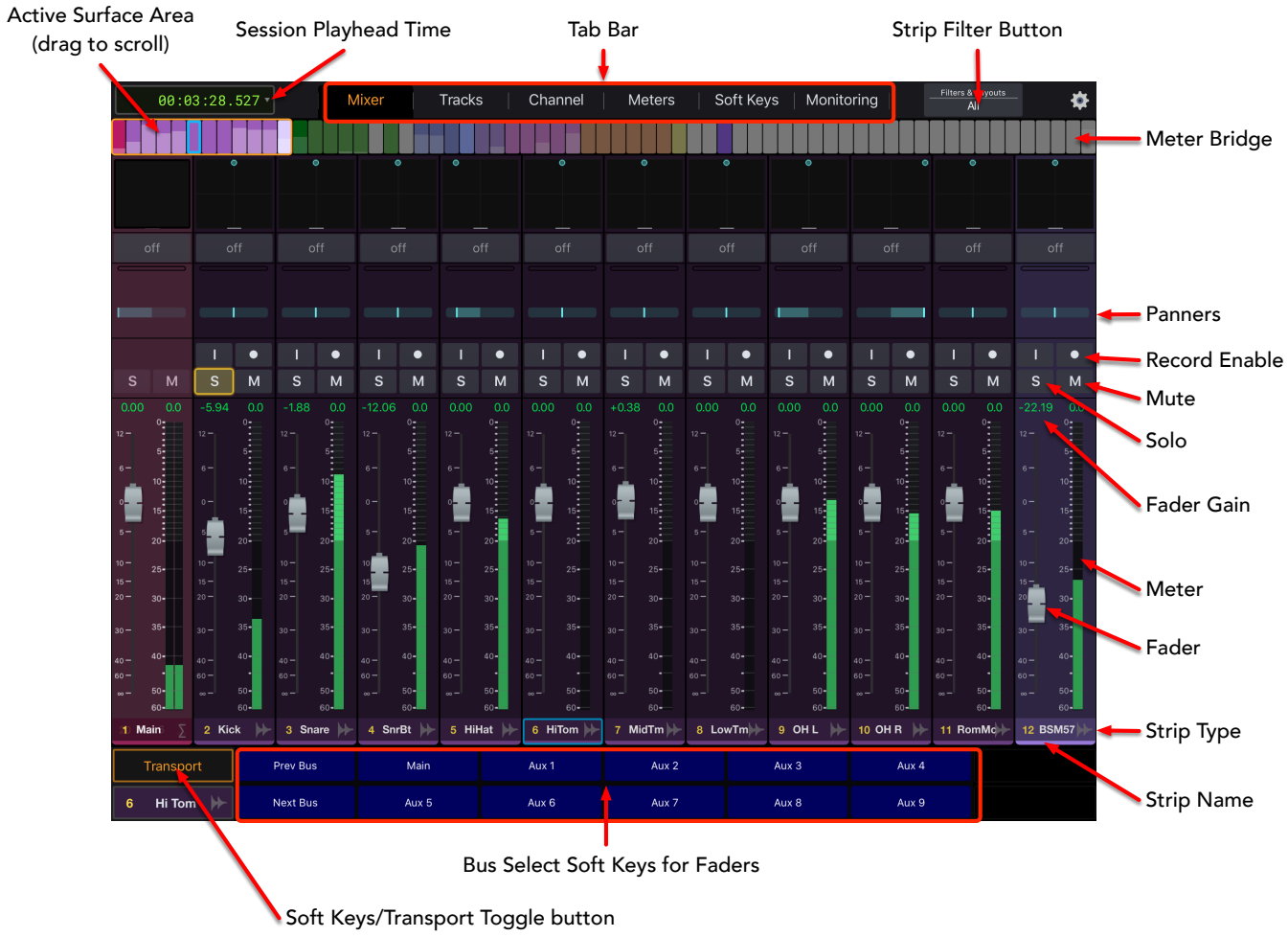


MIOConsole3d Mixer View Strip Filter in Avid Control

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.



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.)

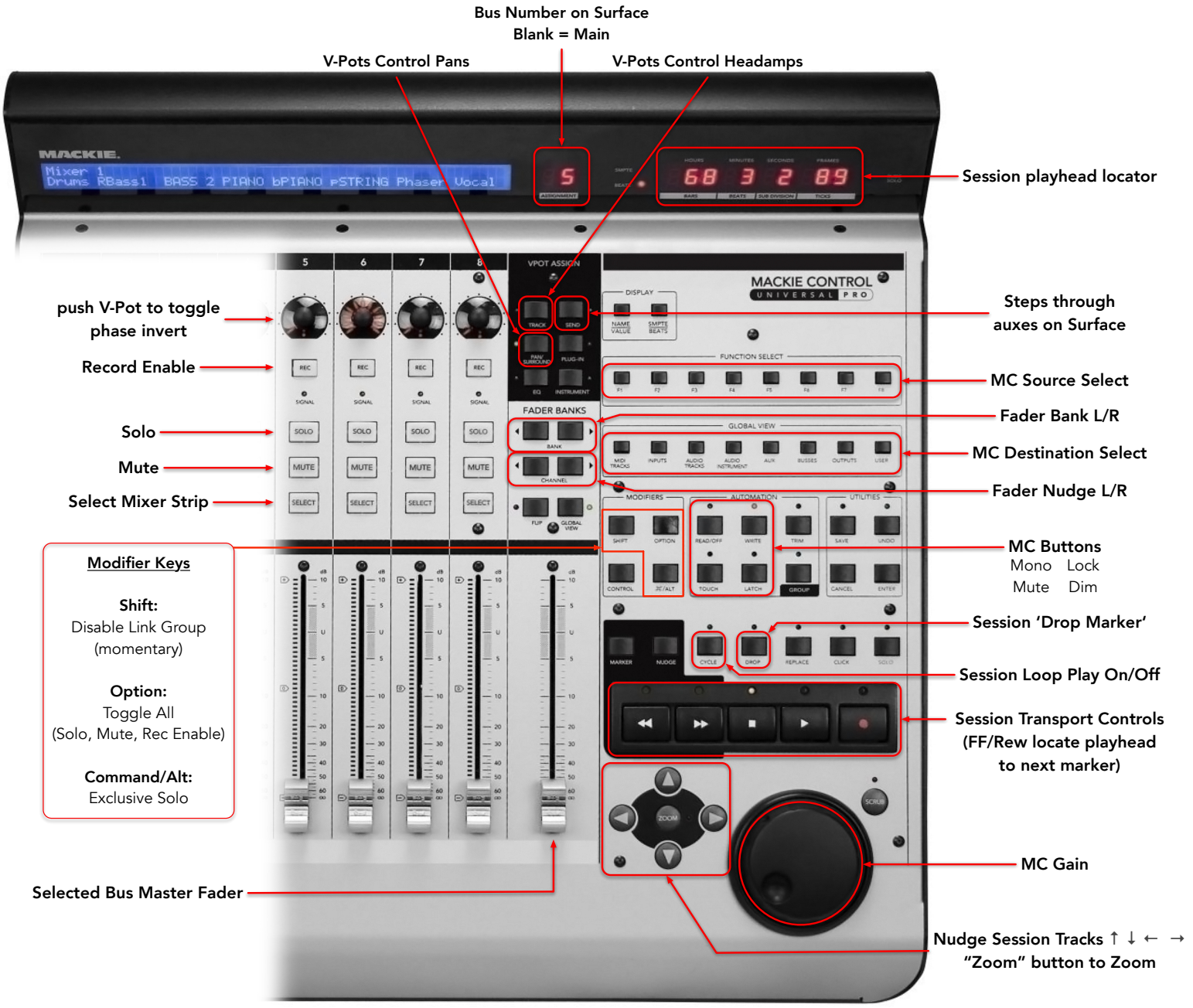
In the EuCon environment, the channel controls are mapped onto so-called 'knob-sets'.

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) in MIOConsole3d

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.



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 7.1. 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.

8. 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.



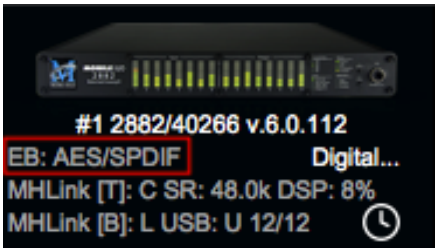
EdgeBus expansion port (2882/ULN-2 base configuration SPDIF•AES EdgeCard shown at left)

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.

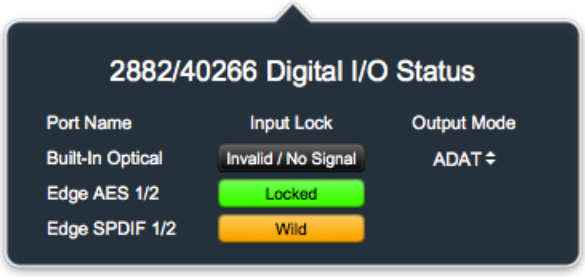
Detailed instructions for installation and removal of EdgeCards are provided at the end of this chapter.

When an EdgeCard is installed, it will be identified in the Unit Status Display of the Status Pane in MIOConsole3d:



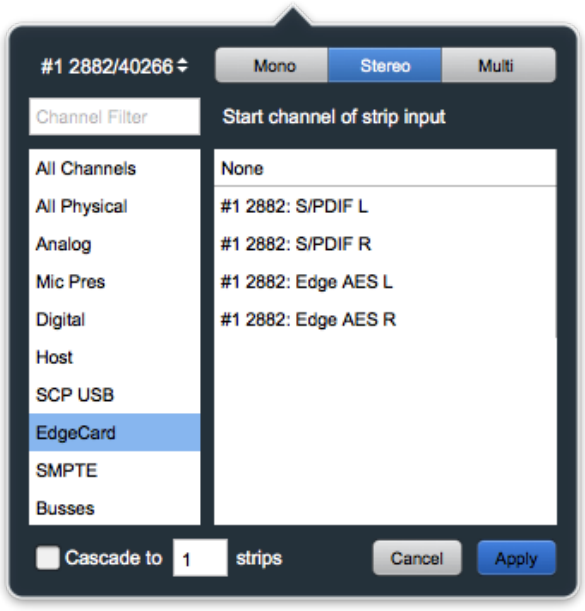
Status Pane: Unit Status Display: EdgeBus identifier (outlined in red)

Controls for the EdgeCard ports will become available in the Units' Digital I/O Status/Control Menu:



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:



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



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



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



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



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**



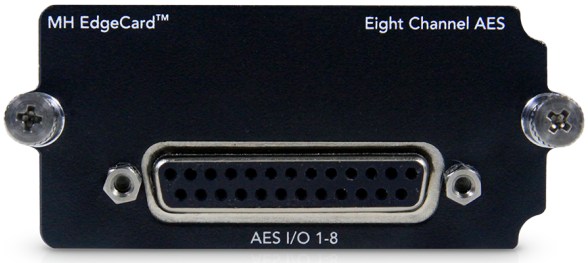
MH EdgeCard: SPDIF • MIDI

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**



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



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



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



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



#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



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.



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.



Insert EdgeCard into Slot

5. Turn the two screws to tighten the board in place.



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.



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.

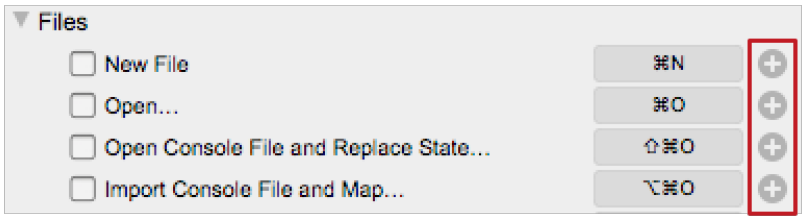
If you must store an EdgeCard module externally, be sure to keep it securely in an anti-static bag within a well-padded box or envelope at all times when not in use. Larger boxes and envelopes are also harder to misplace, so it's a win-win: more protection and easier to find in a pinch.

A. MIOConsole3d Key Commands

Assigning / Modifying Key Commands

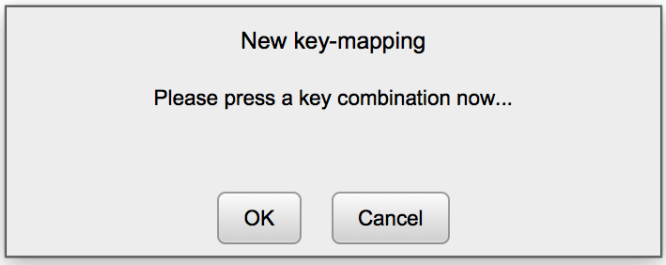
MIOConsole3d supports fully user-editable modifier-key and keystroke commands to improve efficiency and ease mousing around the Console. In many cases these key commands are supplied with an eye to compatibility with third-party HID (Human Interface Device) controller device (for example, a Contour Shuttle 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



Edit Key Commands window: create new Key-Command buttons

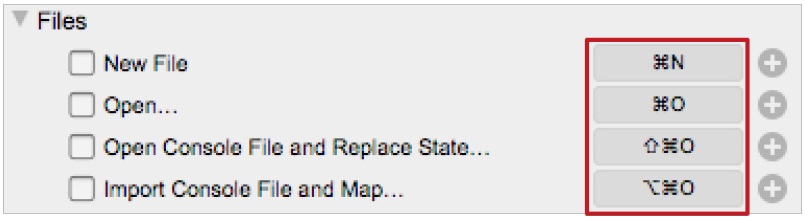
- To enter a new key command to an unassigned function, simply click the "**+**" button for the command in the list and the "New Key-mapping" window will appear:



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.

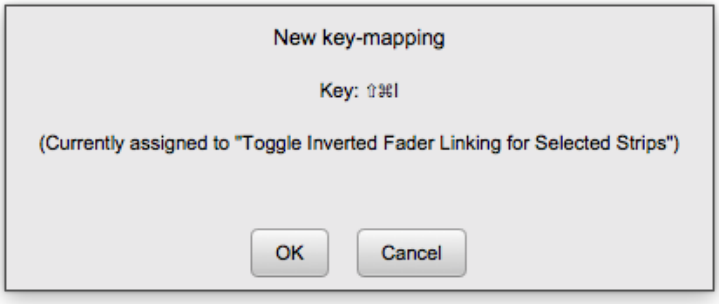
To modify an existing keystroke, click the "**+**" button or the current command keystroke button



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.



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.

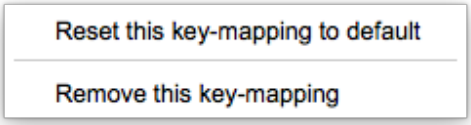
Click 'OK' to proceed.



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.



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.



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
Assign Controller for item under mouse…	⌘1
Auto Learn Controller for last parameter	^2
Show External Controller Mapping List Window	^3
Global Enable External Mapped MIDI Controllers	⌘^M

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

Quad Modulation LFO



Quad Modulation LFO

- Synopsis: Multi-waveform LFO Generator
- Category: Signal Generator, Building Blocks
- Channels: 0

Description:

MH Quad Modulation LFO plugin provides an analog-synth style multi-waveform LFO generator.

This signal generator creates a low-frequency waveform (0 - 20 Hz). The two outputs of the plugin are in quadrature (e.g. one is 90° phase shifted from the other).

The signal generator can create the following waveforms (selectable from the popup menu). These waveforms are bandlimited only when the filter is engaged.

Parameters:

`Rate` [0, 20] Hz

The frequency of the generated signal is controlled by the Rate parameter.

`Filt Cutoff` [0, 20] Hz

Filt Cutoff controls a Low Pass Filter that can be used to shape the waveform and smooth sharp transitions. The Filt Cutoff parameter sets the cutoff frequency of the filter in Hz.

If you set the Filt Cutoff parameter to 0 Hz, the plugin will automatically make the filter cutoff frequency track the LFO frequency.

`Filt Enable` [0, 1] Toggle

The Filt Ena parameter controls whether the filter is on or off.

`Depth %` [0, 100] Percentage

The Depth % parameter controls the magnitude of the LFO waveform.

`Waveform Type` [Triangle, Sawtooth, Sine, Square] Selector

The signal generator can create the following waveforms (selectable from the popup menu).
- Triangle
- Sawtooth
- Sine
- Square

These waveforms are not bandlimited when the filter is not engaged.

`Master Bypass` {On, Off}

When bypassed, MH Quad Modulation LFO will output no signal.

See also:

MIOQuadLFO MIOQuadOsc, MIOQuadNCO - Numerically Controlled Oscillator, MIOQuad-NCO-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, MIOQuadNCO-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.

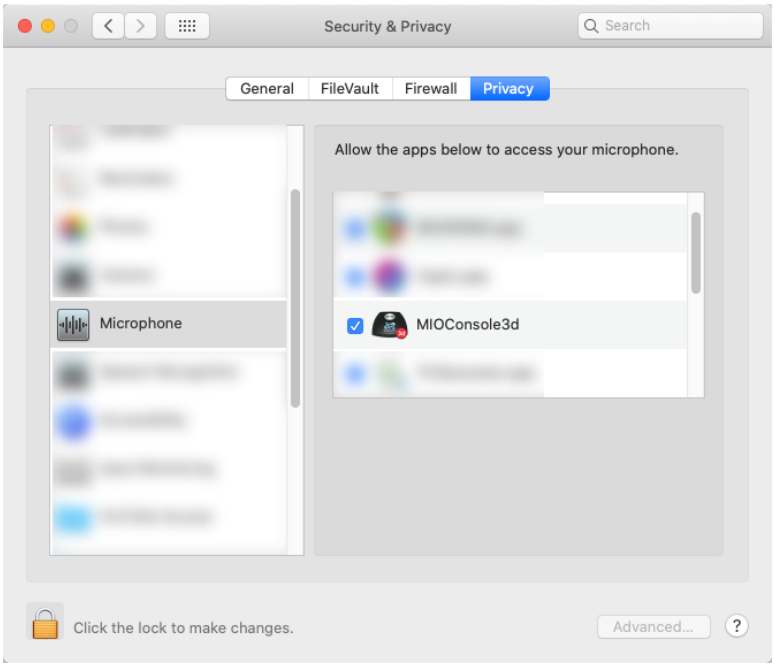
If the 3d device is powered up and booted properly, the Power, Sample Rate, and Locked front panel indicators will be illuminated. If these indicators are not illuminated, the 3d device is not powered properly or the units firmware has been corrupted. If you determine that you are powering 3d device 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.



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.

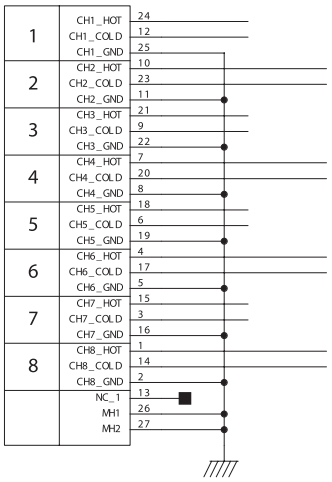
It is possible for firmware updates to "not take". This appears to be related to DSP loading issues in the 3d device, other devices on the USB bus (if connected via USB), and the state of the USB system software on the Mac. If you have problems with updating the firmware, try the following procedure:

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

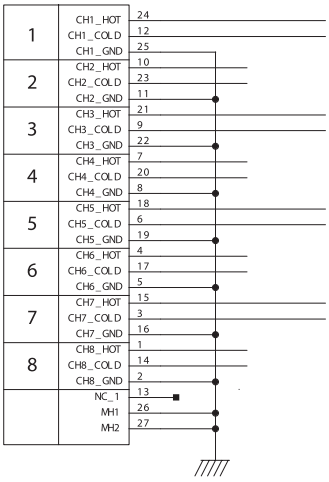
In cases where a firmware update to an MHLinked device does not seem to have registered, just run the firmware update procedure again, making sure to fully power down every box on the MHLink chain for a good 20 seconds before bringing them all back on-line.

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



Analog Input DB-25



Analog Output DB-25

Analog DB25 pinouts

Common wiring pinouts:

XLR:

- HOT: Pin 2
- COLD: Pin 3
- GND: Pin 1

TRS:

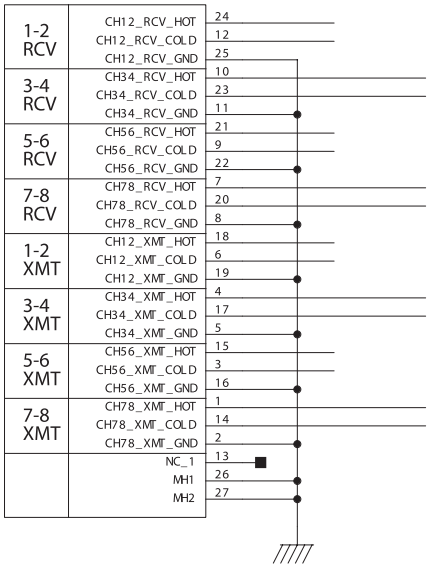
- HOT: Tip
- COLD: Ring
- GND: Shield

TS (unbalanced 1/4"):

- HOT: Tip
- COLD: Do not connect
- GND: Shield

RCA:

- HOT: Pin
- COLD: Do not connect
- GND: Barrel



AES/EBU DB-25

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

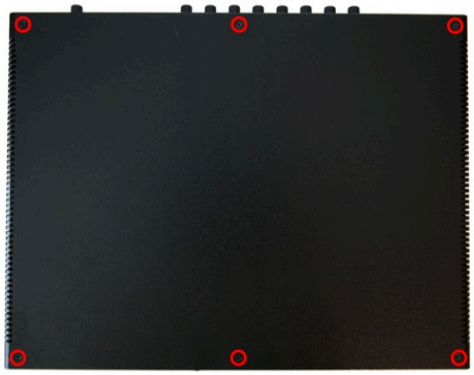
F. ULN-8/LIO-8 Jumper Settings

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:



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.*



Side Screw Placement, with Rack Ear

If the rack ears are not fitted, there will be three screws per side.

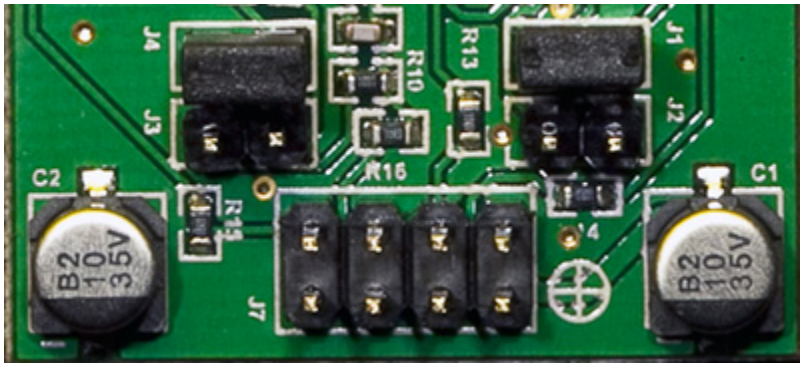


Side Screw Placement, No Rack Ear

Remove the top cover, and rotate the unit so that the rear panel is facing you.

Reassembly is the reverse.

D.I. Board



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.

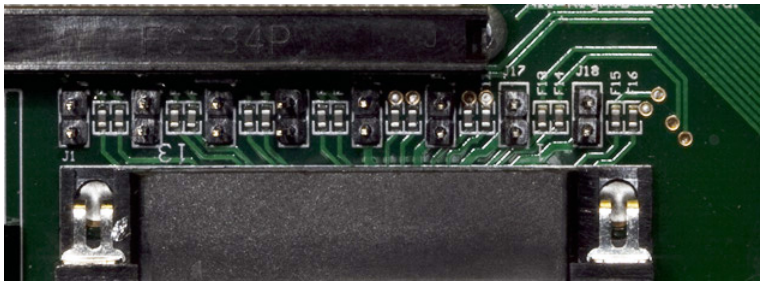
When installing a jumper, it should be within the white box next to the jumper label, not connecting adjacent jumpers.

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.



Shield Lift Jumper Sites

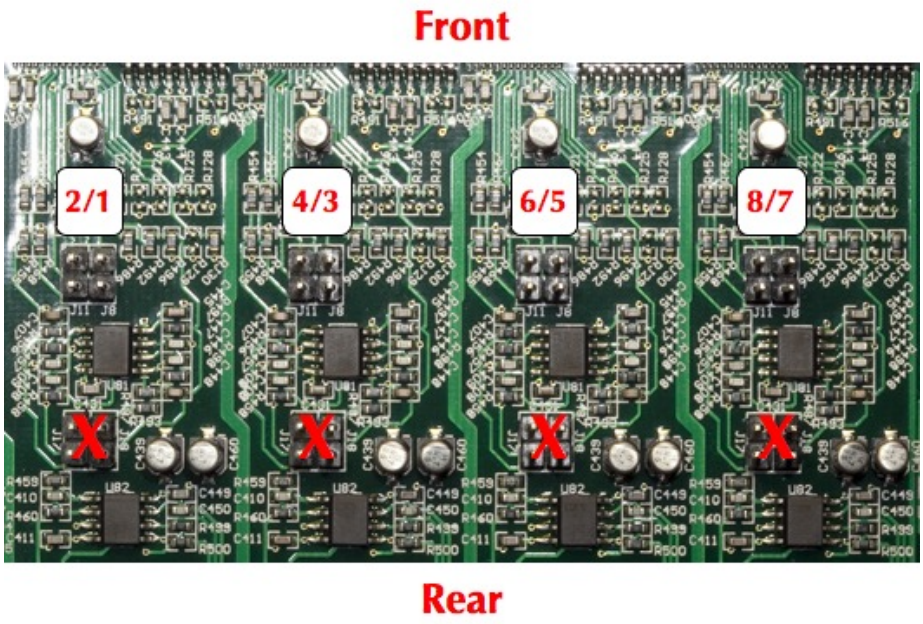
By removing the jumpers, you disconnect the shield from the connected input. Having the jumper installed connects the shield to the ground of the interface.

> **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.



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.

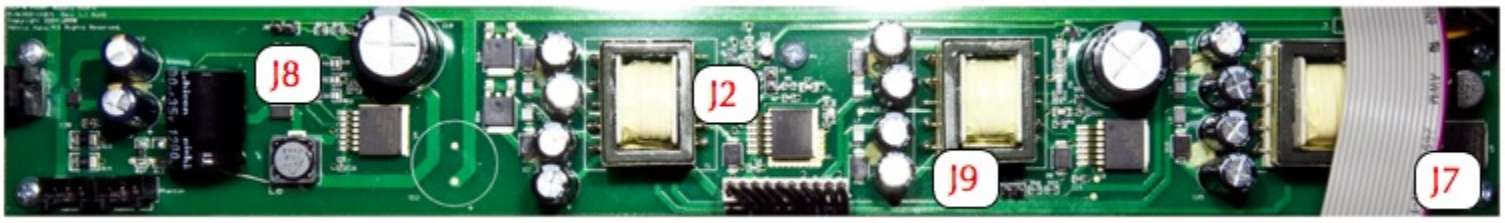
Power supply

There are four 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.



PSU Jumper Sites

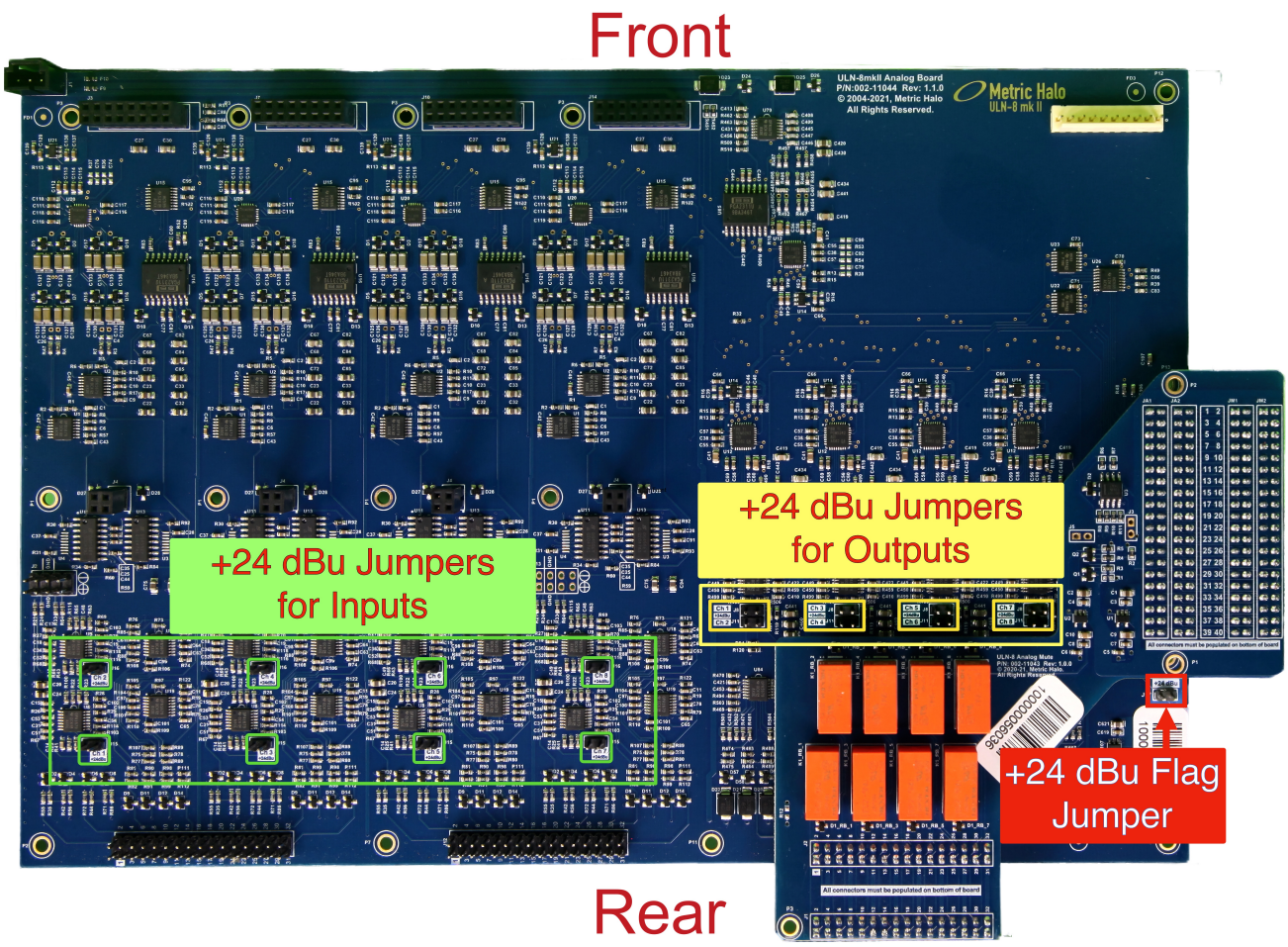
mkIV Analog Board Jumpers Addendum

For LIO-8 and ULN-8 mkIV users (whether new or a mkIV upgrade), only the analog board jumpers are different from earlier generations. Here is an overview and a map of the mkIV jumper locations:

- The D.I. board and line input grounding jumpers are unchanged, still functioning as covered in the previous pages.
- The J7, J8 and J9 jumpers for the power supply remain the same. However on mkIV, the high power J2 jumper is no longer required for +24dBu operation.

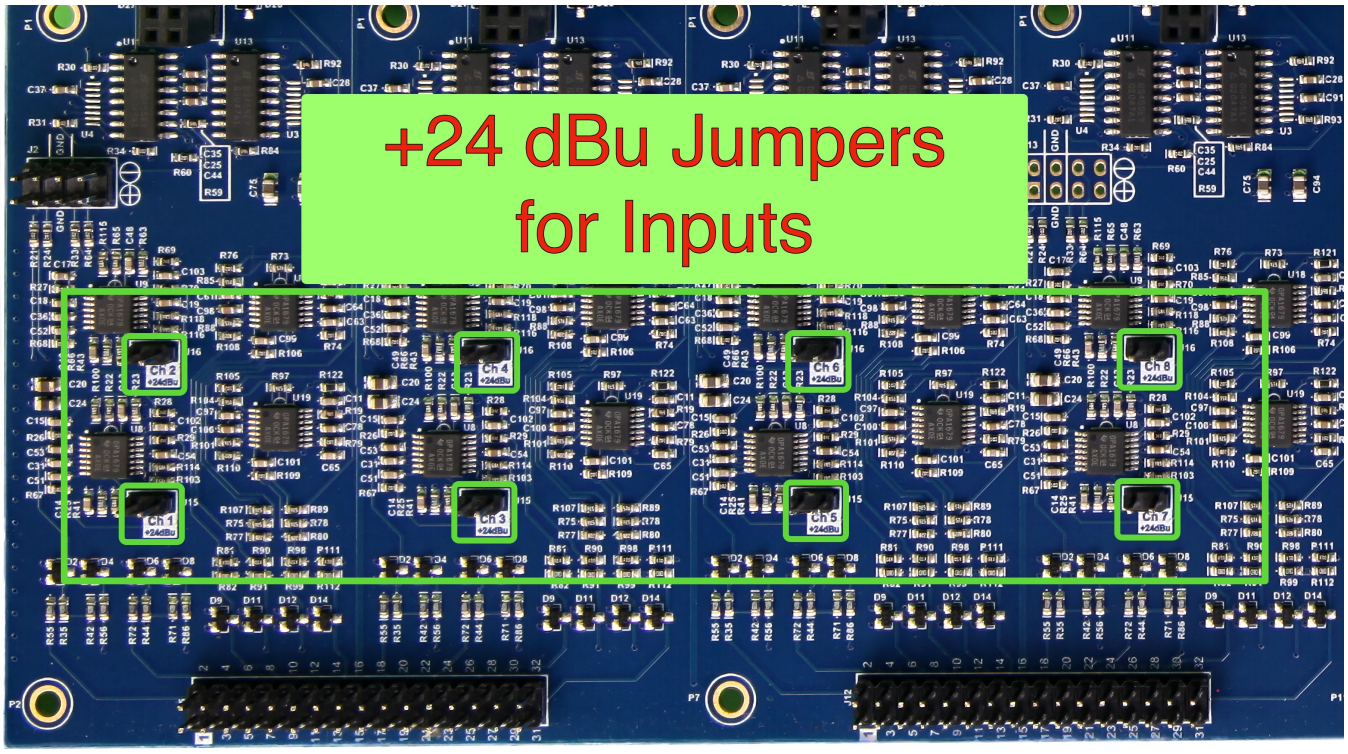
  This is intentional. One of the primary design targets for the mkIV was to improve power efficiency and reduce the heat generated by LIO-8 and ULN-8 interfaces.

  As a result, all of our tests indicate that with mkIV there is no longer any technical or audible advantage to installing the PSU J2 high power jumper; it only serves to make the box use more power and run a little hotter.
- The mkIV design now includes +24dBu operating level jumpers for each analog line input channel.
- Analog output +24dBu jumpers are in a different location on the new boards.
- A "+24dBu Flag Jumper" is provided to tell the MIOConsole software that the unit is fully jumped for +24dBu operation. This should be used only when all the other +24dBu jumpers on the board are installed. **Note**: this feature is part of a larger scheduled functional upgrade to MIOConsole3d and is not currently implemented.
- All mkIV analog board jumpers are clearly labeled on the circuit board. (see following pages)



mkIV analog board jumper locations

Analog Line Input Level



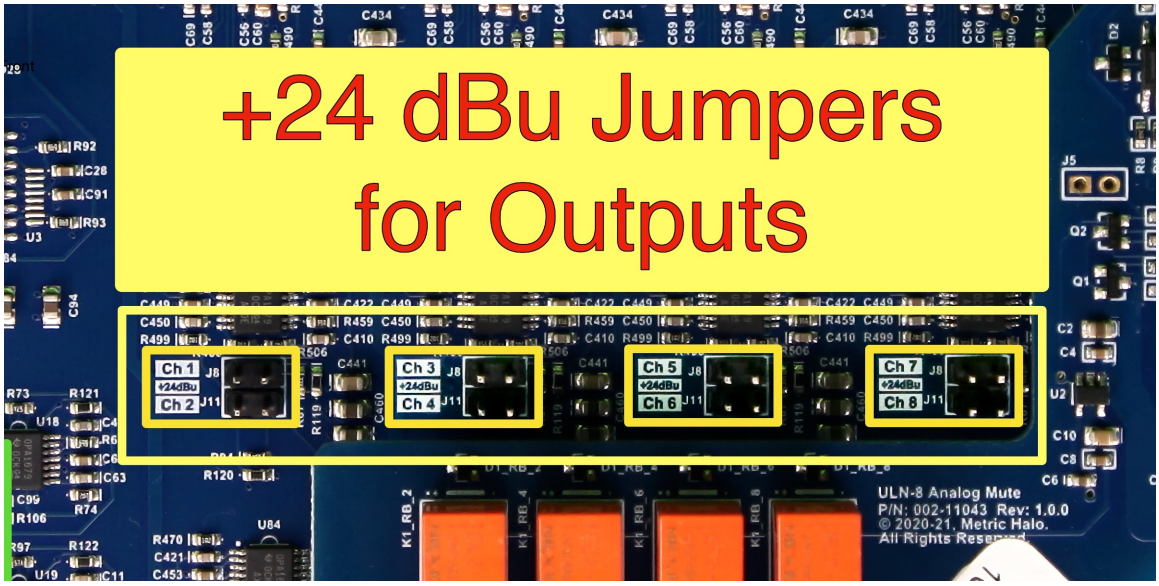
Viewed with the rear panel connectors closest to you, the analog input +24dBu jumpers are grouped in the lower left corner, just above the rear panel multipin connectors.

The jumpers are arranged with channel 1 at the lowest left position, channel 2 above it, channel 3 to the immediate right of channel 1, channel 4 above channel 3, and so on.

Jumpers can be placed on any or all input channels to change the line input operating level from +18dBu to +24dBU.

At +24dBu operating level, 0 VU = -20dBFS.

Analog Line Output Level



Viewed with the rear panel connectors closest to you, the analog output +24dBu jumpers are grouped just above the Analog Mute Board hardware relays. If the Mute Board is not installed, they are the four jumper pairs in a horizontal line about 2.5in / 7cm above the right-most multipin connector at the bottom edge of the mkIV analog board.

The jumpers are arranged with channel 1 at the left position top, channel 2 below it, channel 3 to the immediate right of channel 1, channel 4 below channel 3, and so on.

Jumpers can be placed on any or all output channels to change the line output operating level from +18dBu to +24dBU.

At +24dBu operating level, 0 VU = -20dBFS.

+24dBu Flag Jumper



Viewed with the rear panel connectors closest to you, +24dBu flag jumper is located ner the lower right corner of the mkIV analog board, just above the board bar code serial number sticker. It is visible and accessible just below the Analog Mute Board multipin connectors to the 3d digital board.

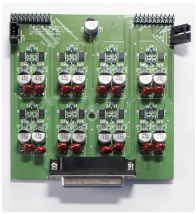
As noted earlier, the sole function of the +24dBu flag jumper is to inform the MIOConsole software that all analog line inputs and outputs have been jumpered to operate at +24dBu. This feature is currently not implemented, but will be enabled in a future update to the MIOConsole3d software.

G. ULN-R Installation Guide

ULN-R Parts and Tools

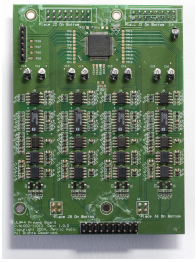
The ULN-R Mic Pre (Ch. 1-4) kit includes the following parts:

- 1) DB25 connector board



DB25 Connector Board

- 1) 4 channel ULN-R mic pre board



4 Channel ULN-R Mic Pre Board

- 1) 3 pin phantom power cable



3 Pin Phantom Power Cable

- 1) 20 Pin Ribbon Cable Jumper



20 Pin Ribbon Cable Jumper

- 2) 7/16" standoffs



7/16" Standoffs

- 3) Phillips head screws



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!

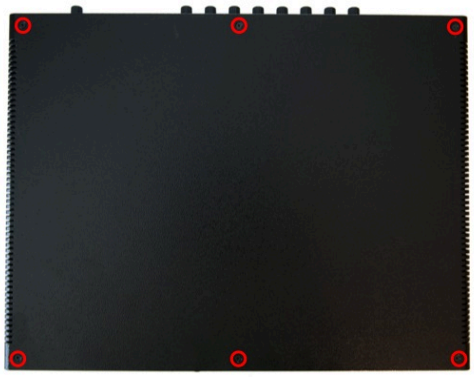
To install the ULN-R mic pre option in the LIO-8 you will need:

- #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:



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.*



Side Screw Placement, with Rack Ear

If the rack ears are not fitted, there will be three screws per side.



Side Screw Placement, No Rack Ear

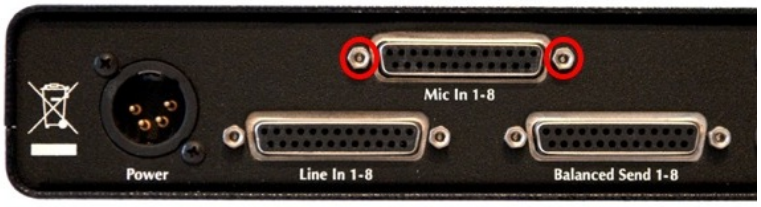
Remove the top cover, and rotate the unit so that the rear panel is facing you. If you are installing the Ch. 5-8 kit only, jump to step 9.

3. Remove the two screws holding the Mic In cover plate, then remove the plate from inside the LIO-8.



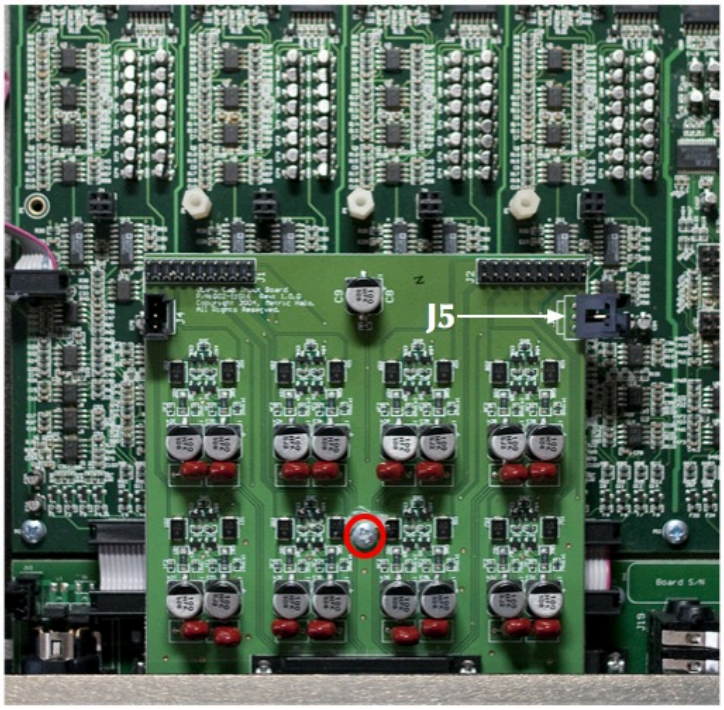
Mic In Blank Plate

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.



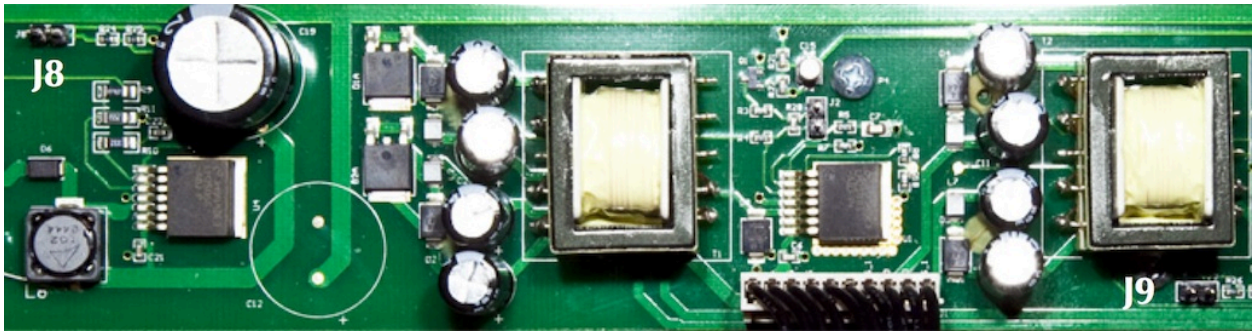
DB25 Connector Board Fitted

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.



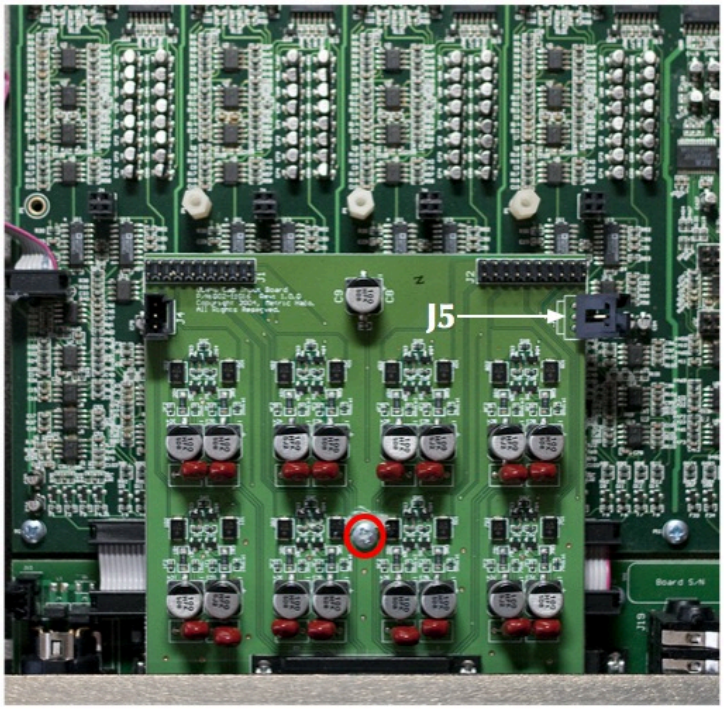
DB25 Connector Board (Interior)

6. Remove the jumpers from J8 and J9 on the power supply board at the front of the LIO:



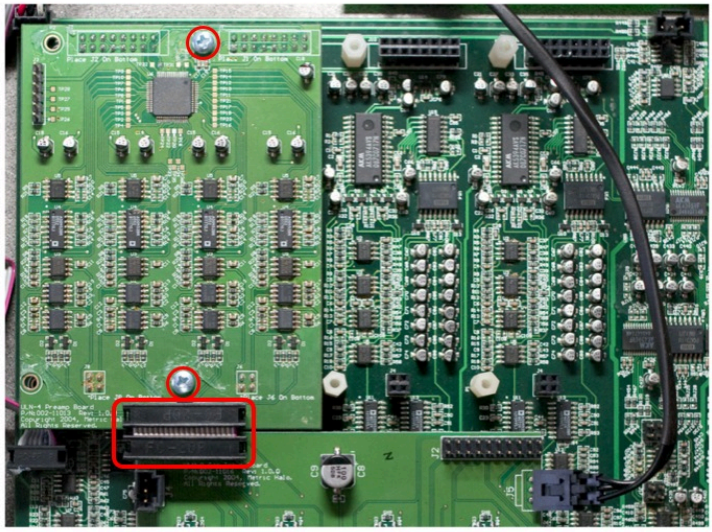
Phantom Power Jumpers on the PSU Board

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:



Phantom Supply Cable and Ch. 1-4 Connectors

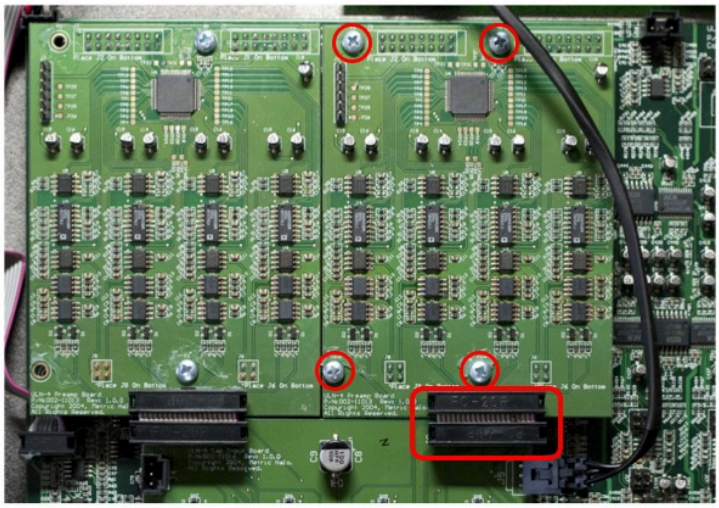
8. Install the mic pre board, making sure that the pins on the bottom of the board line up with the sockets highlighted in the picture above. You must install the channel 1-4 board in this position. Use two Phillips head screws to secure the mic pre board and dab them with caulk. Fit the ribbon cable jumper between the mic pre and DB25 connector board, making sure that the pins are lined up.



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.



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.

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