Production Bundle
User's Guide

Production Bundle Users Guide

Metric Halo

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1. Introduction

The Metric Halo Production Bundle is a collection of ten essential and gorgeous-sounding plug-ins, with Spectrafoo™ multi-standard input and output metering, completely revamped graphics and new plug-in headers featuring vastly improved user control and functionality:

- 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)
- Spectrafoo dynamics processor metering with sidechain detector tracking
- Multi-function Spectrafoo output level meters with VU, RMS, PPM Peak and Peak Hold

The Production Bundle includes:

- **MH Character**: Classic modeled distortion with improved metering and oversampling, increasing the range of applications in both directions at once.

- **MH ChannelStrip**: The original and still the champion channel strip plug-in. Double-precision since its introduction in 1998, updated and improved once again while retaining the qualities that made it the most universally loved go-to signal processor since music production arrived on computers.

- **MH Dirty Delay**: Imagine a rich, gooey tube delay, with the precision feedback and decay control, tempo sync and note division tracking of a modern computer delay line. Here you go...

- **MH Haloverb**: In a world of "flavor" reverbs eating half your CPU to emulate the great halls and electronic reverbs of the past, most of the time what you really want is a nice clean unobtrusive space creator that you can run a few dozen at a time. Meet HaloVerb.

- **MH Multiband Dynamics**: Does an SSL 9000J come with a 3-band compressor? No? Oh, well if it did, this would be that.

- **MH Multiband Expander**: The ultimate solution to dense noisy tracks and stems that need to fade naturally and cleanly, but don't using conventional methods. Also brillant at cleaning up live mic feeds for guitar, bass, vocals, pianos, horns, halls, .

- **MH Precision DeEsser**: Not just another de-esser, it's a precision high frequency dynamics miracle-worker. Suppressing "sss" and "sshhh" better than anything else is one thing, but it can also isolate high frequency events allowing a vast array of creative signal enhancement and special effect possibilities.

- **MH Sonic EQ**: The literal ultimate high-end audiophile purist EQ - designed in the mid '80's before the rules that govern modern EQ design were invented. All at once Sonic EQ is supremely musical, supremely precise, supremely powerful and supremely easy to use. It . does . everything . better.

- **MH SuperGate**: The ultimate high-end audiophile purist geek gate... yeah I know, sorry, but it is. SuperGate is by light-years the most precise and distortion-free gate in this or any other world. (Disclaimer: It's not actual alien technology.)

- **MH TransientControl**: Enhance or suppress transients, enhance or suppress what comes after the transients, or both at once for a different kind of leveler control. A different approach to dynamics manipulation is always a valuable asset, yes?

2. System Requirements

Hosts:

- *Pro Tools™ (Mac)*: Pro Tools 11 or higher running on a Macintosh computer. The v4 software currently supports Native AAX operation only.

- *Pro Tools™ (Windows)*: Pro Tools 10 or higher running on a Windows computer. The v4 software currently supports Native AAX operation only.

- *Native (Mac)*: Any Intel or Apple Silicon-native Mac DAW (64-bit) that supports AU, VST2, VST3 or AAX plug-ins.

- *Native (Windows)*: Any Intel Windows (64-bit) DAW that supports VST2, VST3 or AAX plug-ins.

Please Note: All v4 plug-ins are 64-bit (as were v3). If you have a need to work with an old 32-bit host, please contact MH Support and ask for the 32-bit Production Bundle v2 installer.

Operating System:

- *Mac*: Any Apple Silicon (ARM) or Intel-based Mac running Mac OS X 10.9 or newer

- *Windows*: Any Intel-based Windows PC running Windows 10 or newer.

Licensing:

- A PACE iLok.com account. You can authorize your v4 license to your computer, iLok Cloud or any 2nd or 3rd generation iLok USB key.

  The first generation blue-green iLok USB keys are no longer supported by PACE for new product authorizations.

  Please note that prior v2 and v3 licenses are separate and remain valid: you do not have to trade in your old iLok license. Production Bundle v3 and earlier plug-ins will continue to serve on older systems in addition to the v4 installations on newer platforms, with full preset compatibility between v3 and v4.

- One license authorizes the software on any platform.

- The most recent iLok License Manager installer can be found here: iLok License Manager application and driver installers.

  Older operating systems may require a specific version of the iLok driver, which can be found here: Legacy iLok application and driver installers.

3. Installation

For both Mac and Windows, there is a single standard installer for the Production Bundle that allows you to decide which plug-ins to install to your computer, depending on what individual plug-in licenses you own and what formats you use.

IMPORTANT: Please make sure you have your license installed and available *before* you first launch your host after running the installer. Otherwise, you will receive a message on launch that the software is not licensed, asking you to "Buy" or "Quit". There are some hosts that will not rescan for your plug-ins on the next launch after the license is installed, and some hosts do not provide a user menu preference to rescan.

Mac

Please note– The following graphics show installation on a macOS 12 system; the process may be slightly different in other versions of the OS, but the basic concepts are the same. Small details such as file sizes shown may vary with subsequent releases.

• Double-click the "MH Production Bundle.pkg" application
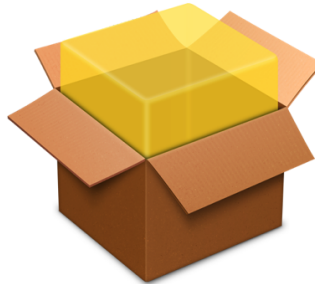
MHProductionBundleInstaller_
v4.0.2.157.pkg

- The first window requests permission for the installer program to scan for the presence of earlier versions of Metric Halo plug-ins. Clicking "Cancel" will quit the installer. Click "Allow" to proceed:
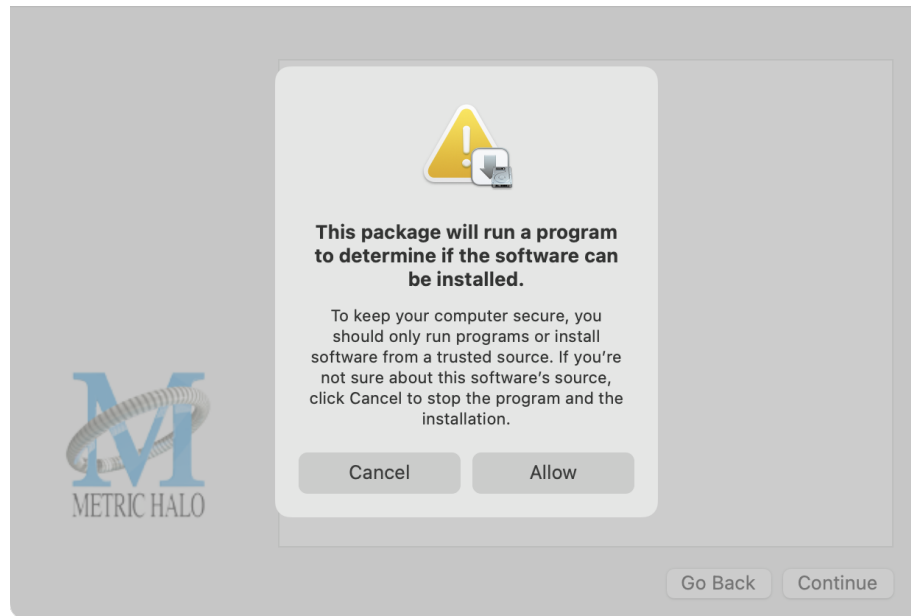
Click "Allow" to proceed…

- The installer dialog will appear, detailing the plug-in formats and system requirements to be installed:
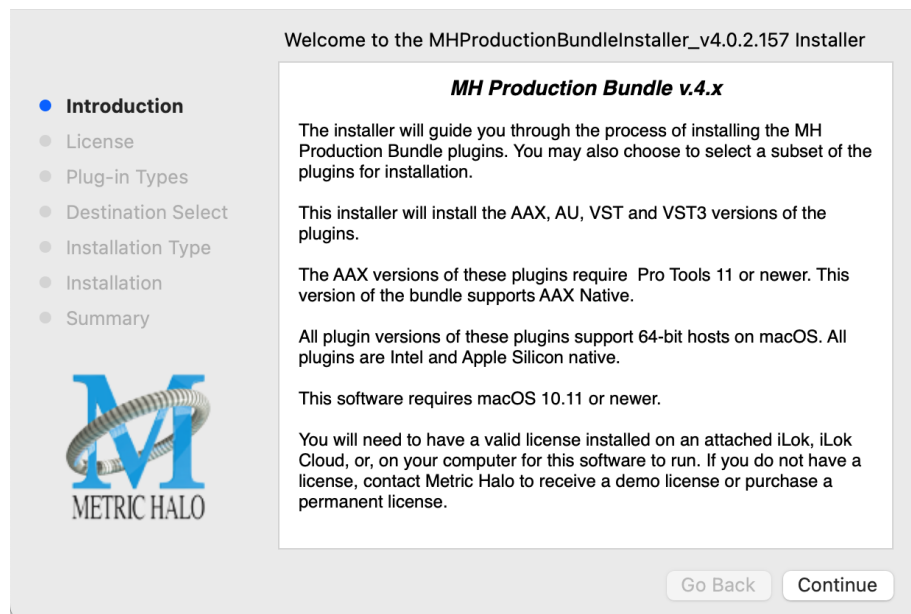
Click "Continue"…

• Now you will see the Metric Halo License Agreement:
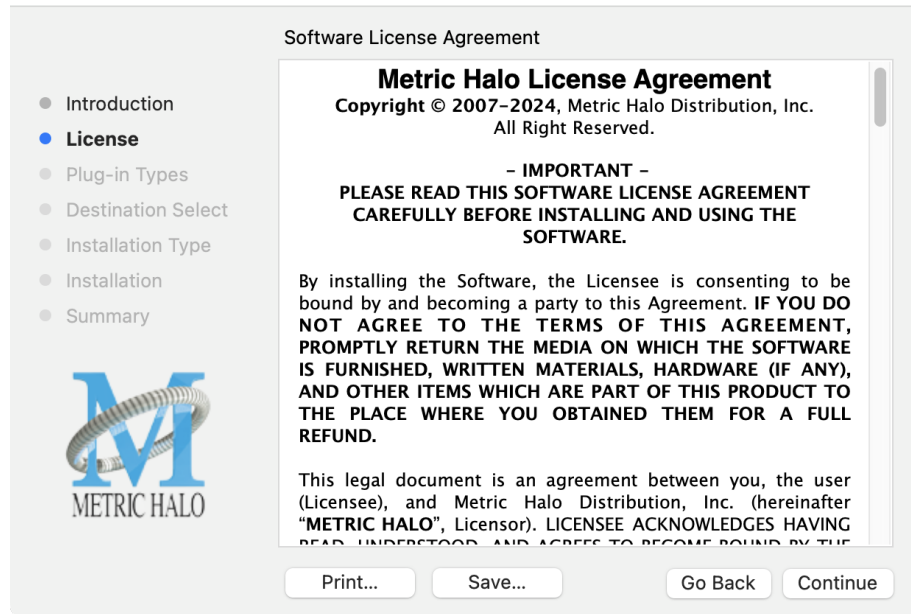
After you have read it, click "Continue"…

• Next, click "Agree" to accept the License Agreement:
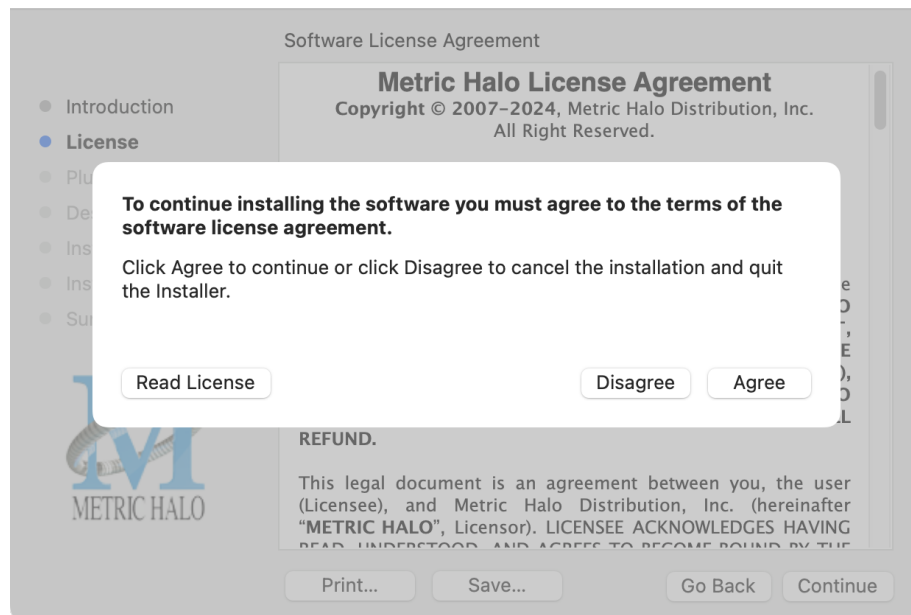
- The Choose Plug-in Types selector window defaults to no plug-ins selected, so that you may choose just the plug-in formats you wish to install. Checking *All Plug-In Types* will install Audio Unit, VST2, VST3 and AAX plug-ins to their respective folders in the root Library directory:
  - AU to **/Library/Audio/Plug-Ins/Components**
  - VST 2 to **/Library/Audio/Plug-Ins/VST**
  - VST 3 to **/Library/Audio/Plug-Ins/VST3**
  - AAX to **/Library/Application Support/Avid/Audio/Plug-Ins**

  Selecting any one or more specific plug-in types will install or upgrade only those formats, leaving older plug-ins in unselected format types untouched.

  Your plug-in format selection will be saved as a preference and pre-set automatically for future Metric Halo family plug-in installations on this computer. Of course you may change your selections at that time.
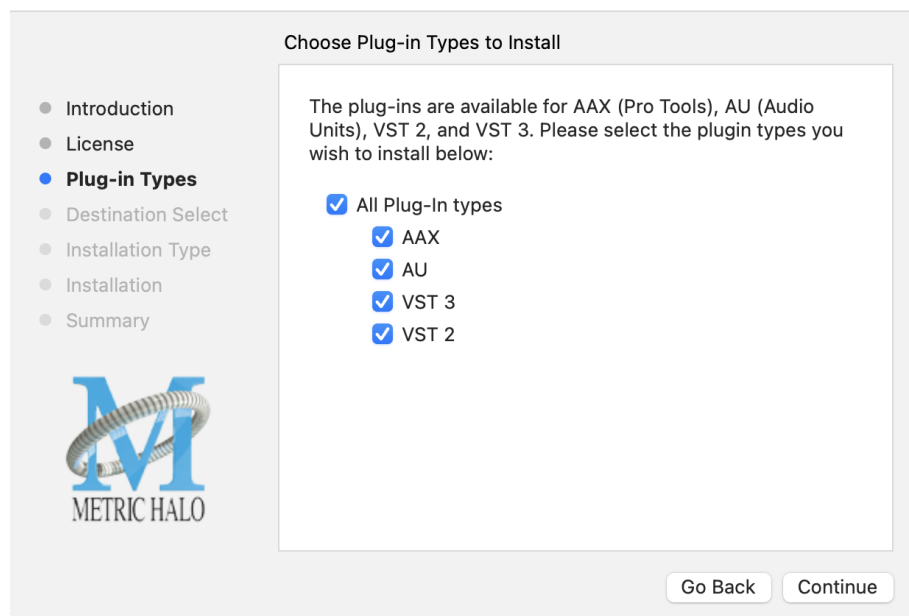
- **Important note for Pro Tools users**

  Version 4 of the AAX plugins will install alongside v3, so both will remain available in your system. There are two reasons for this:

  1. The v4 plug-ins are currently Native only and do not provide DSP executables for HDX/Carbon DSP chips.
  2. At the request of many Pro Tools users, we have made changes to the AAX v4 plug-ins to provide finer and more consistent behavior of EQ parameters when controlled by EuCon control surfaces and Pro Tools automation.

     In some specific workflow cases, these changes have the potential to affect automation playback of older sessions, so the safe route would be to keep older automated sessions running on v3, and using v4 for new sessions.

     For specifics on these changes, please see the latest release notes within the v4 plug-in UI by clicking on the Metric Halo logo in the upper left corner of the plug-in window.

  Existing sessions will recall with the v3 instances for both Native/DSP, and you can continue to add new instances of the v3 plugins (for example, if you want to be able to use DSP versions for tracking).

  You can add v4 instances as you wish to take advantage of the new features and new plug-ins.

  You can switch v3 instances to v4, and your settings will be preserved.

  If you load sessions with v3 instances, but only have v4 installed, Pro Tools will automatically migrate the instances to v4 Native with the parameter settings intact.

- "Select a Destination" lets you install the plugs for *all users* (if you are logged in as an admin), *only your user*, or to another volume entirely.
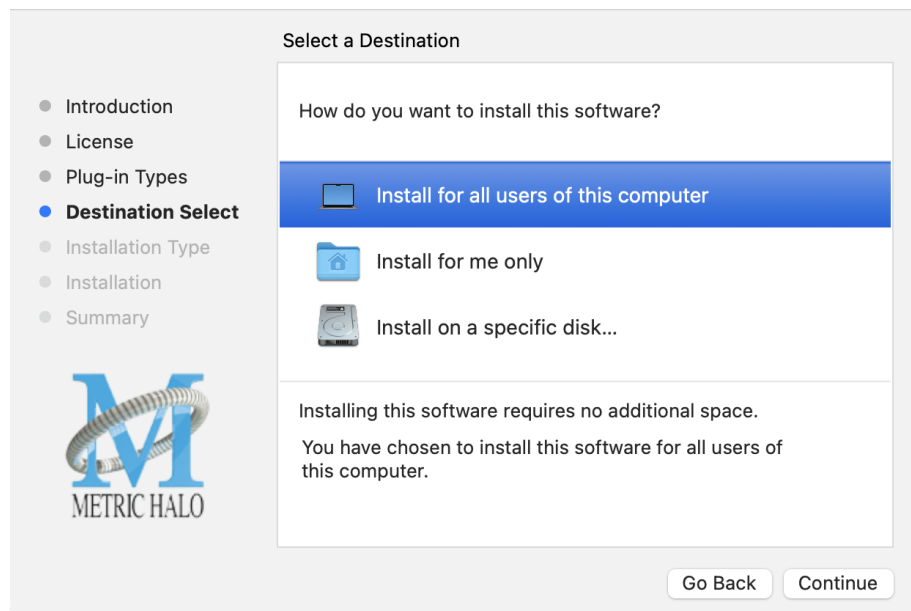
We recommend that you use the standard installation location unless you have a specific reason not to.

- The "Custom Install" selector window defaults to no plug-ins selected, so that you may choose exactly the plug-ins you wish to install.

Custom Install on "Macintosh HD"

Package Name	Action	Size
> ☐ All Plug-Ins		237 MB

- Introduction
- License
- Plug-in Types
- Destination Select
- **Installation Type**
- Installation
- Summary

Space Required: Zero KB   Remaining: 342.38 GB

Go Back   Continue

Custom Install on "Macintosh HD"

Package Name	Action	Size
∨ ✓ All Plug-Ins		237 MB
✓ MHChannelStrip	Install	24.6 MB
✓ MHCharacter	Install	23.8 MB
✓ MHDirtyDelay	Install	24.2 MB
✓ MHHaloVerb	Install	24.2 MB
✓ MHMultibandExpander	Install	24 MB
✓ MHMultibandDynamics	Install	24.1 MB
✓ MHPrecisionDeesser	Install	24 MB
✓ MHTransientControl	Install	20 MB
✓ MHSonicEQ	Install	24.2 MB
✓ MHSuperGate	Install	24 MB

- Introduction
- License
- Plug-in Types
- Destination Select
- **Installation Type**
- Installation
- Summary

Space Required: 237 MB   Remaining: 342.14 GB

Install all flavors of all Plug-Ins in the package.

Go Back   Continue

Clicking the disclosure triangle opens the selection list of the individual plug-ins. Make your selections and click "Continue" to proceed.

- The final confirmation window displays the total size of the selected installation. Hit "Install" to proceed.
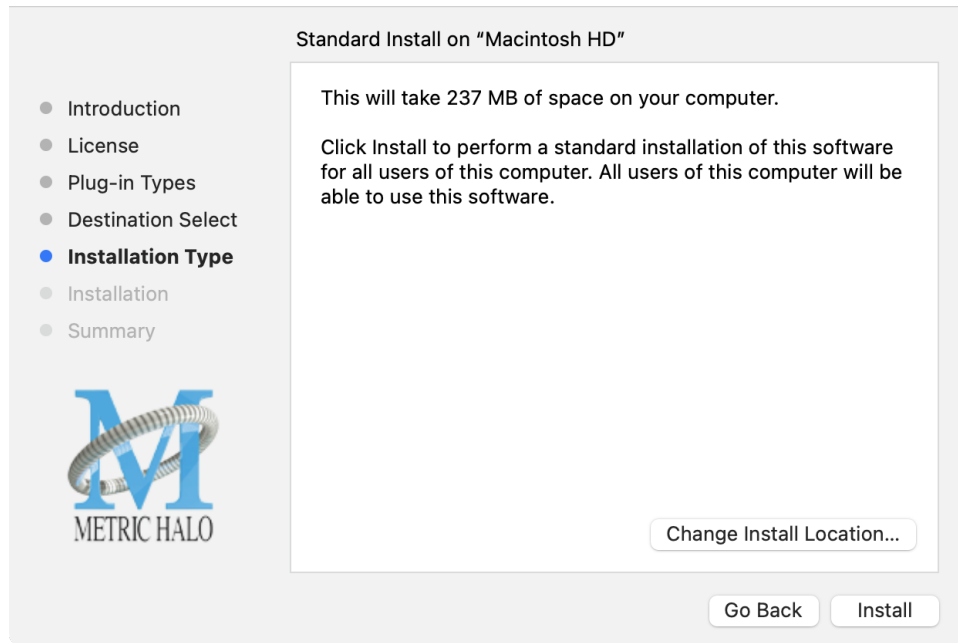
- If present, Touch ID/Face ID will execute the installation once it recognizes your biometrics:

Otherwise, enter your login password as usual and click "Install Software".

- Once the installer has finished, you'll see this dialog:
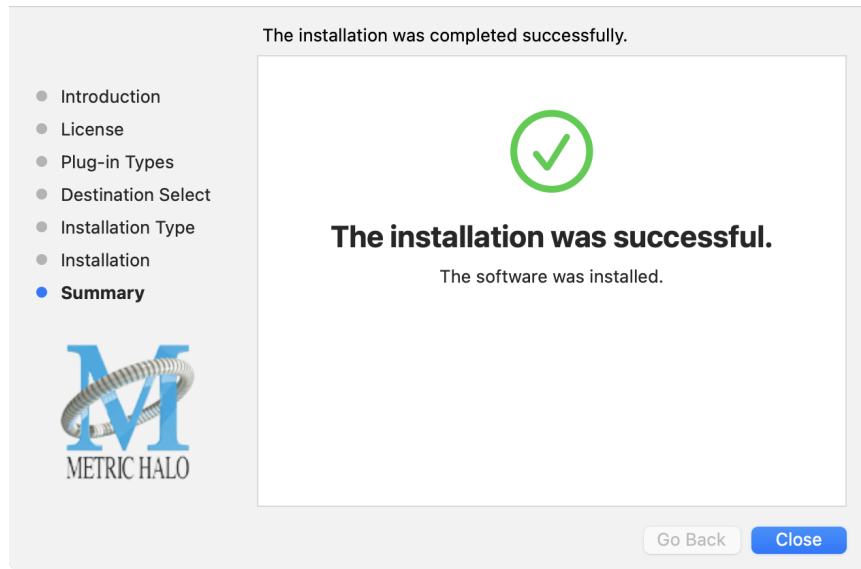
If you do *not* see the "Installation Successful" message, contact MH Support.

- The final step is to go to https://www.ilok.com/ and ensure you are up to date with the latest version of the iLok License Manager. Use the iLok License Manager to install the software license to your iLok key or directly to your computer.

That's it! Enjoy using Metric Halo plug-ins!

Windows

Please note – The following graphics show installation on an Windows 10 system; the process may be slightly different in other versions of the OS, but the basic concepts are the same. Small details such as file sizes shown may vary with subsequent releases.

- Double-click the "MHProductionBundleInstaller_v4.x.x.xxx" application

MH Production Bundle Installer

- The installer dialog will appear:
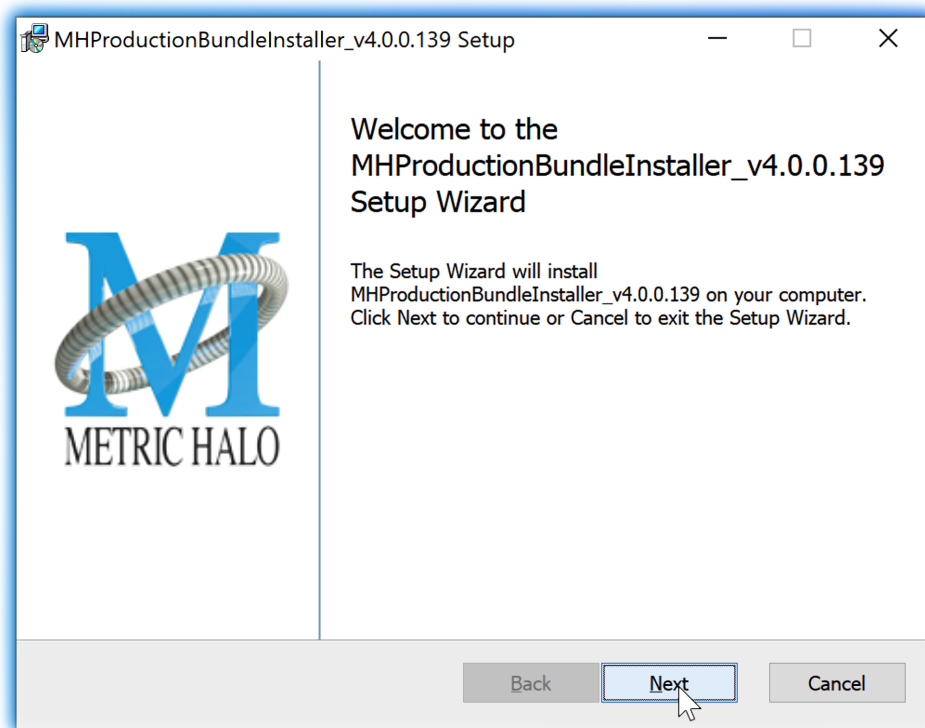
Welcome Dialog

Click "Next" to proceed.

- Read the Metric Halo License Agreement:
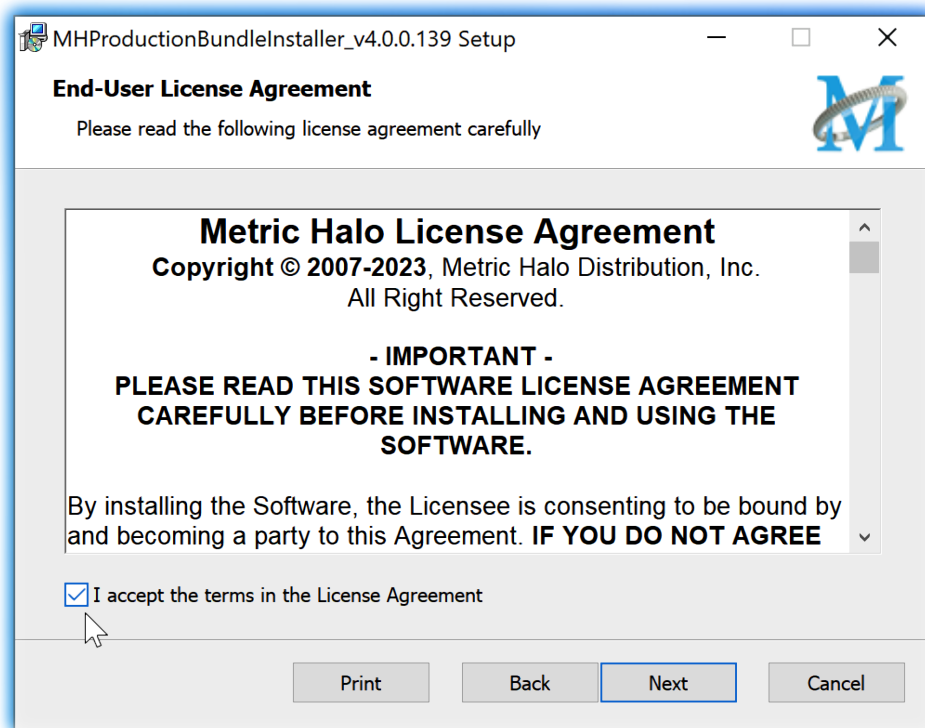
License Agreement

After you have read it, click next to "I accept the terms of the License Agreement" and click "Next".

• Custom Setup Options
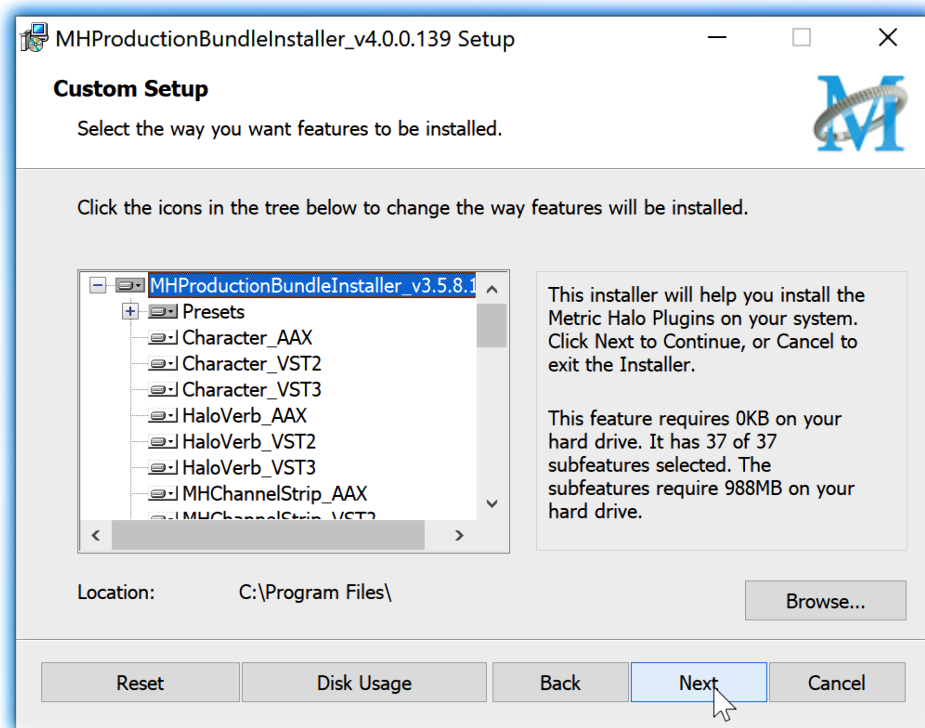
Custom Setup

The Windows installer **Custom Setup** page allows you to refine the features to be installed and their location.

By default, the full bundle with VST2, VST3 and AAX will be selected for installation to the C:\Program Files\ folder. Click **Browse** to change the installation target folder.

We recommend that you use the file path that will allow your host software to recognize the plug-ins. See your host software documentation.

When you have made your selections, click "Next" to continue.

- The next page is a confirmation to continue, offering a last chance to go back and review your selections from the previous window:
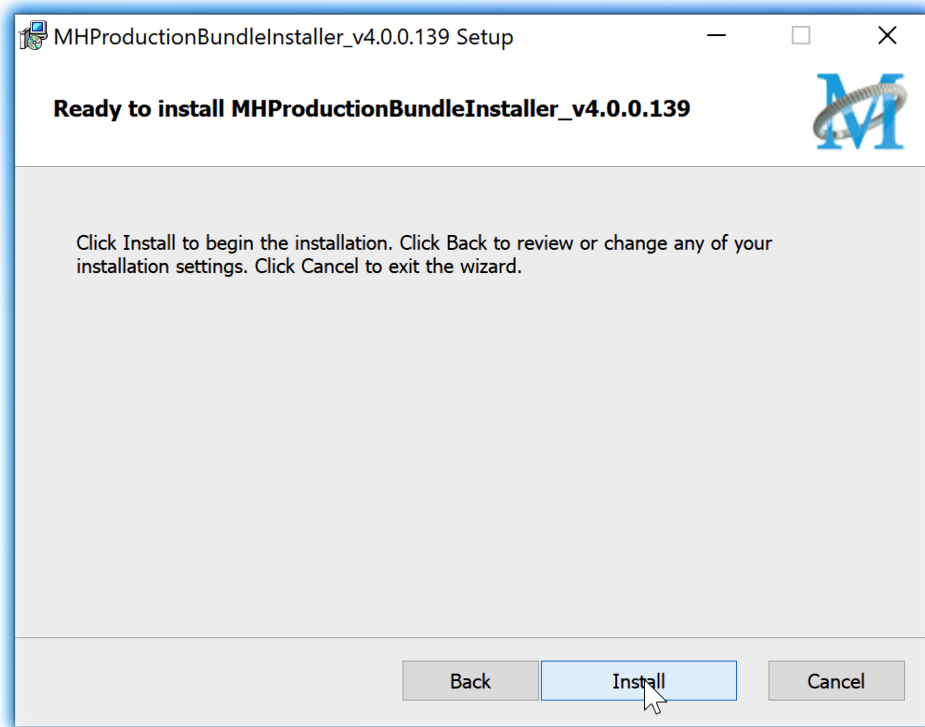
Ready to Install

Click "Back" to return to the Setup page, "Cancel" to cancel the installation, or click "Next" to continue.

- A final dialog before executing the installation process is a security confirmation, providing a verification of the location of the installer executable and information regarding the publisher of the installer package and its contents:
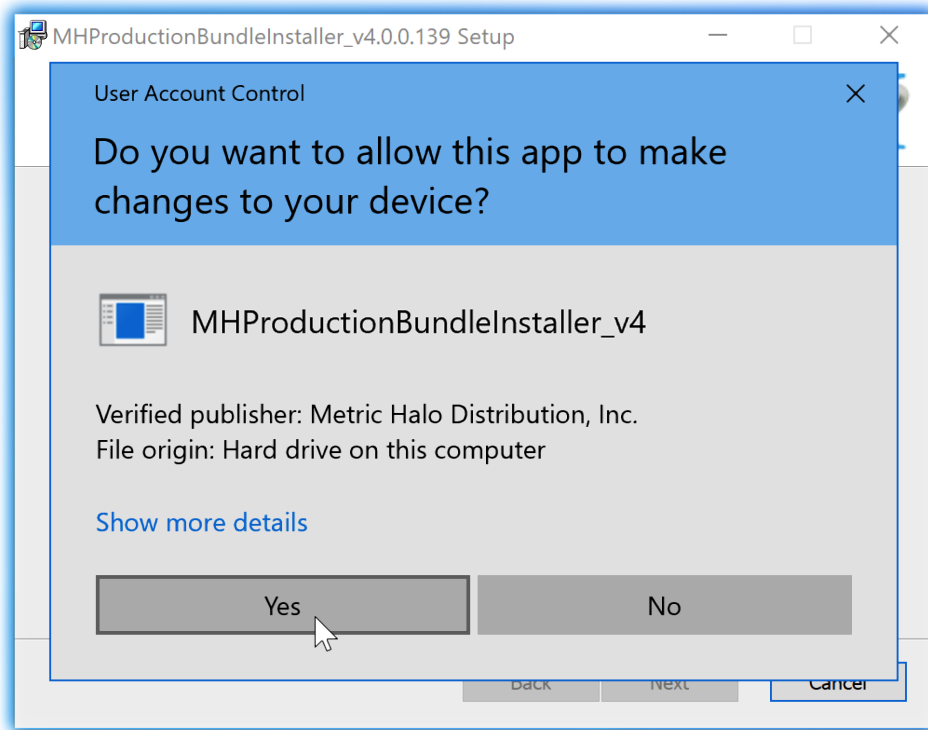
Final installer verification and permission to execute

Clicking "Show Details" and digging through the ensuing submenus will reveal signing credentials and other data for confirming the legitimacy of the installer executable.

Click "Yes" to proceed with the installation.

- The Setup Wizard will install and register the files:
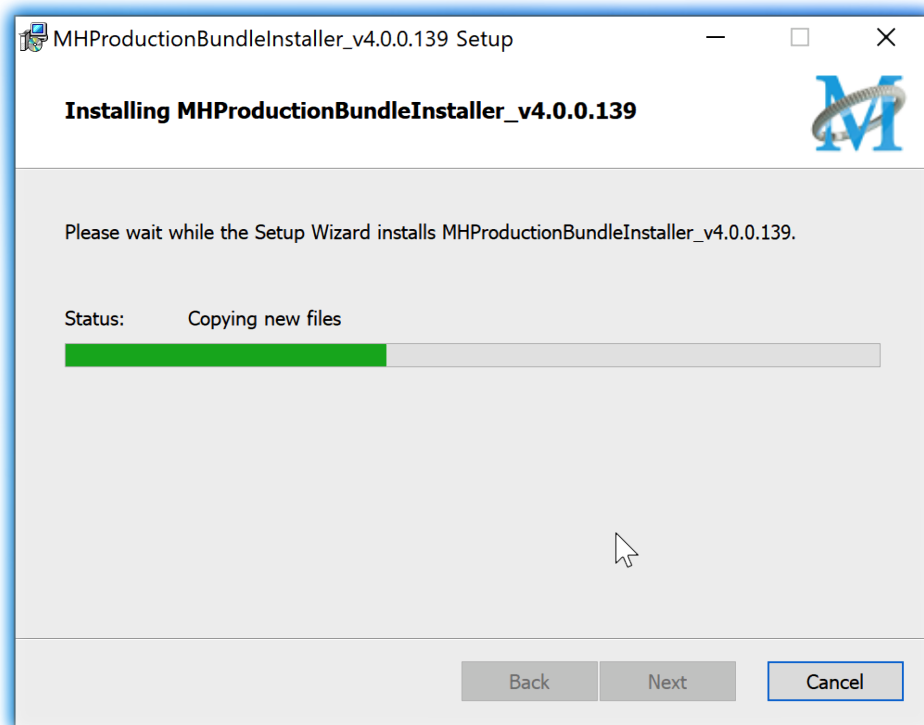
Installation in Progress

- Once the installer has finished, you will see this dialog:
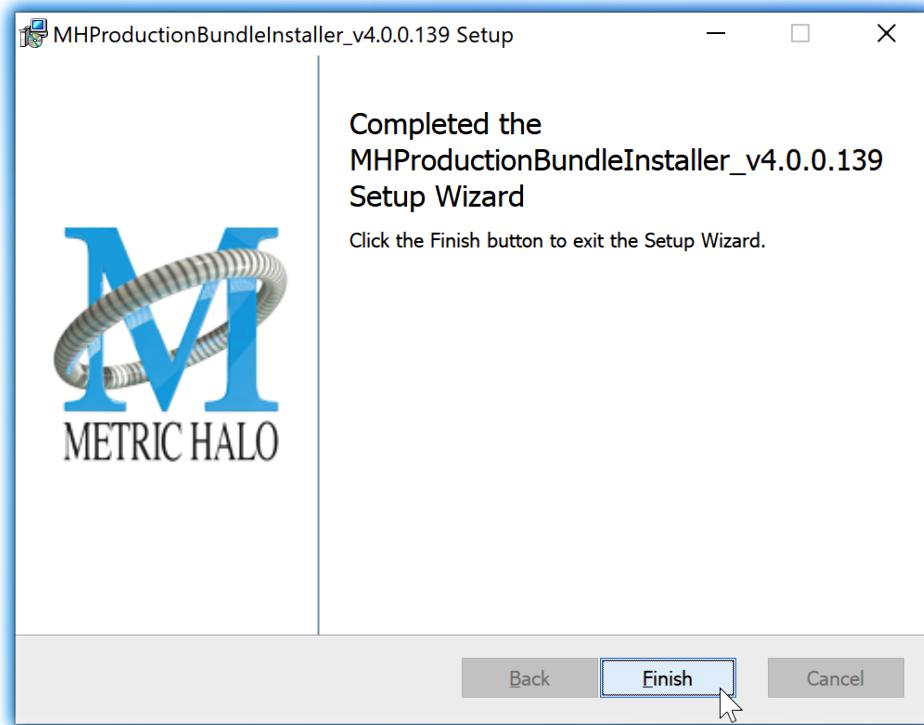
Installation Complete

If you do *not* see this installation "Completed" dialog, contact MH Support.

- The final step is to go to https://www.ilok.com/ and ensure you are up to date with the latest version of the iLok License Manager. Use the iLok License Manager to install the software license to your iLok key or directly to your computer.

That's it! Enjoy using the Metric Halo Production Bundle!

Update Notification (all platforms)

All Metric Halo Production Bundle plug-ins will automatically check for newer version availability (if your computer is connected to the internet).

Plug-in Update Alert

If a new version is found, the Metric Halo icon in the plug-in header bar will sport a lovely red dot. Click on the dotted icon and check the Update Notification tab for release notes and download instructions.

4. Plug-In Header Bars

All Metric Halo family plug-ins display the MH control bar directly above the processor control UI. This control bar is very helpful for organizing your MH presets, as it allows you to organize and access all your presets across all supported plug-in formats on Mac, Windows and our 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.
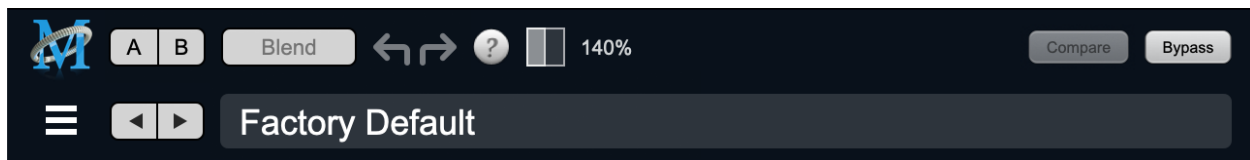
MH Plug-in header (default setting: eq and dynamics graph display closed)

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
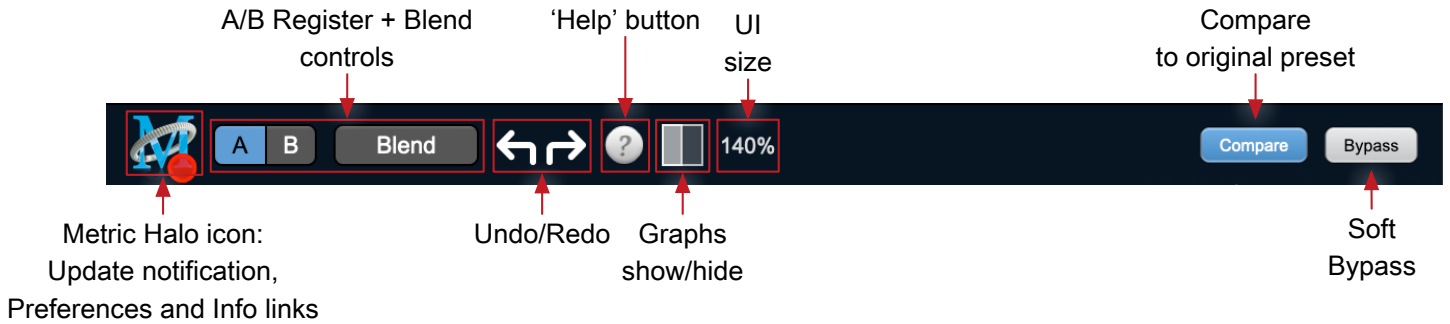
Metric Halo Header Icon

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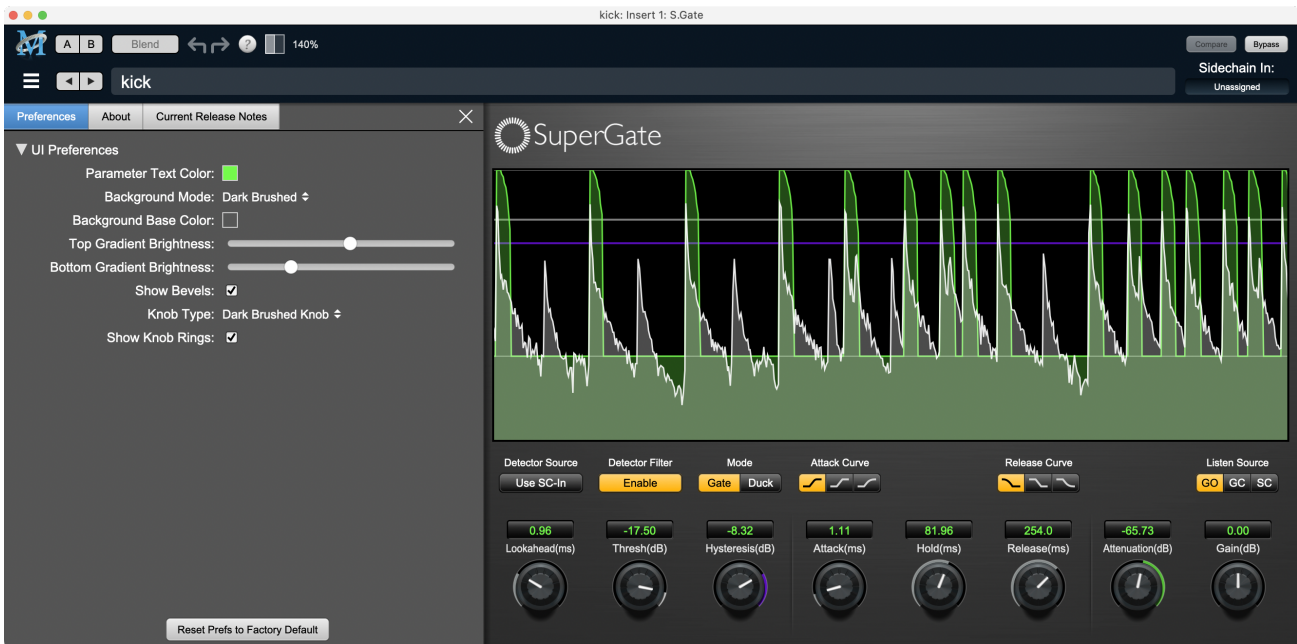
Preferences tab (example shown is SuperGate with sidechain filter graph closed)

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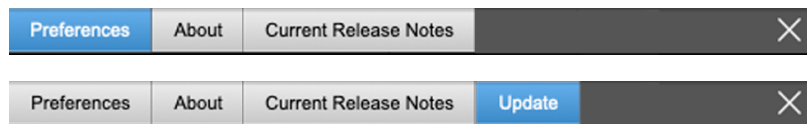
Plug-in control pane tabs

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
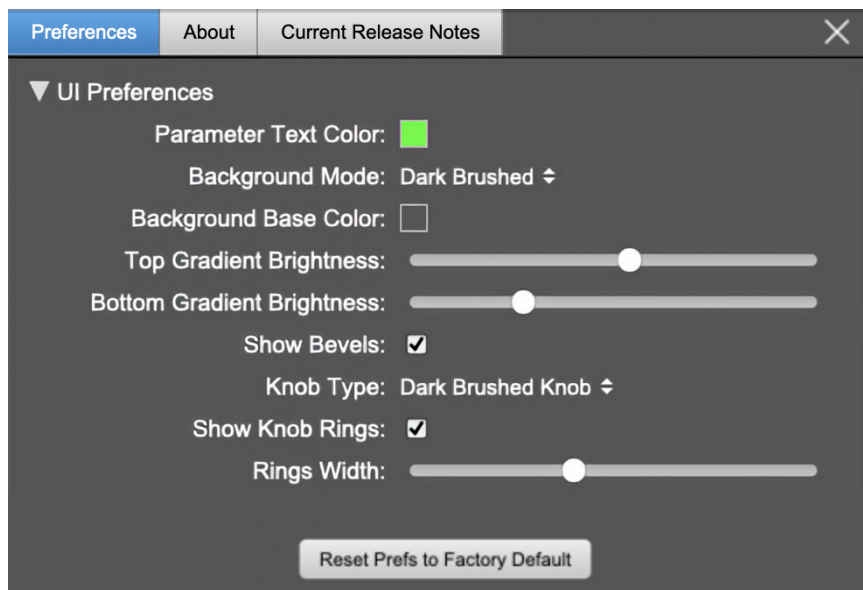
standard plug-in UI Preferences

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.

UI Preferences functions are listed on the next page.

> **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 list of all the preferences available across all of the plug-ins. They don't all apply to every plug-in, but you can refer here to find the documentation for the prefs that that are displayed in the actual plug-in.

- **UI Preferences**
  - **Parameter Text Color**: sets the text color of all click-to-enter numeric readouts.
  - **Background Mode**: selects the background type and texture.

    Brushed background modes place an uncolored textured metal graphic behind the control elements. Color and Gradient controls will not affect the Light and Dark Brushed backgrounds.
    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 stronger gradient and lighting effect
    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
  - **Rings Width**: widen/narrow the colored rings surrounding parameter knobs
  - **Show Band Swatches**: show/hide the color swatches above each EQ band enable switch (used to set the color of each EQ band)
  - **Swatch Alpha**: opacity control for the band color swatch buttons (does not affect Band Color selection)
  - **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 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

- **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
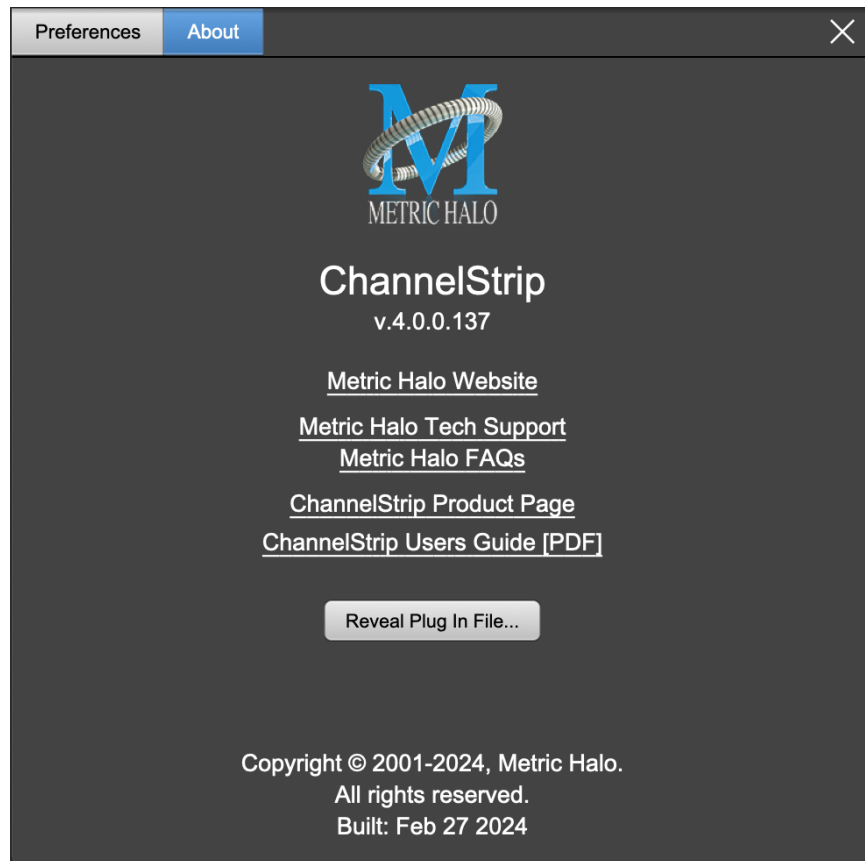
About tab (ChannelStrip shown)

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 plug-in file, with the plug-in file itself selected. Very handy for troubleshooting on the fly.

Current Release Notes
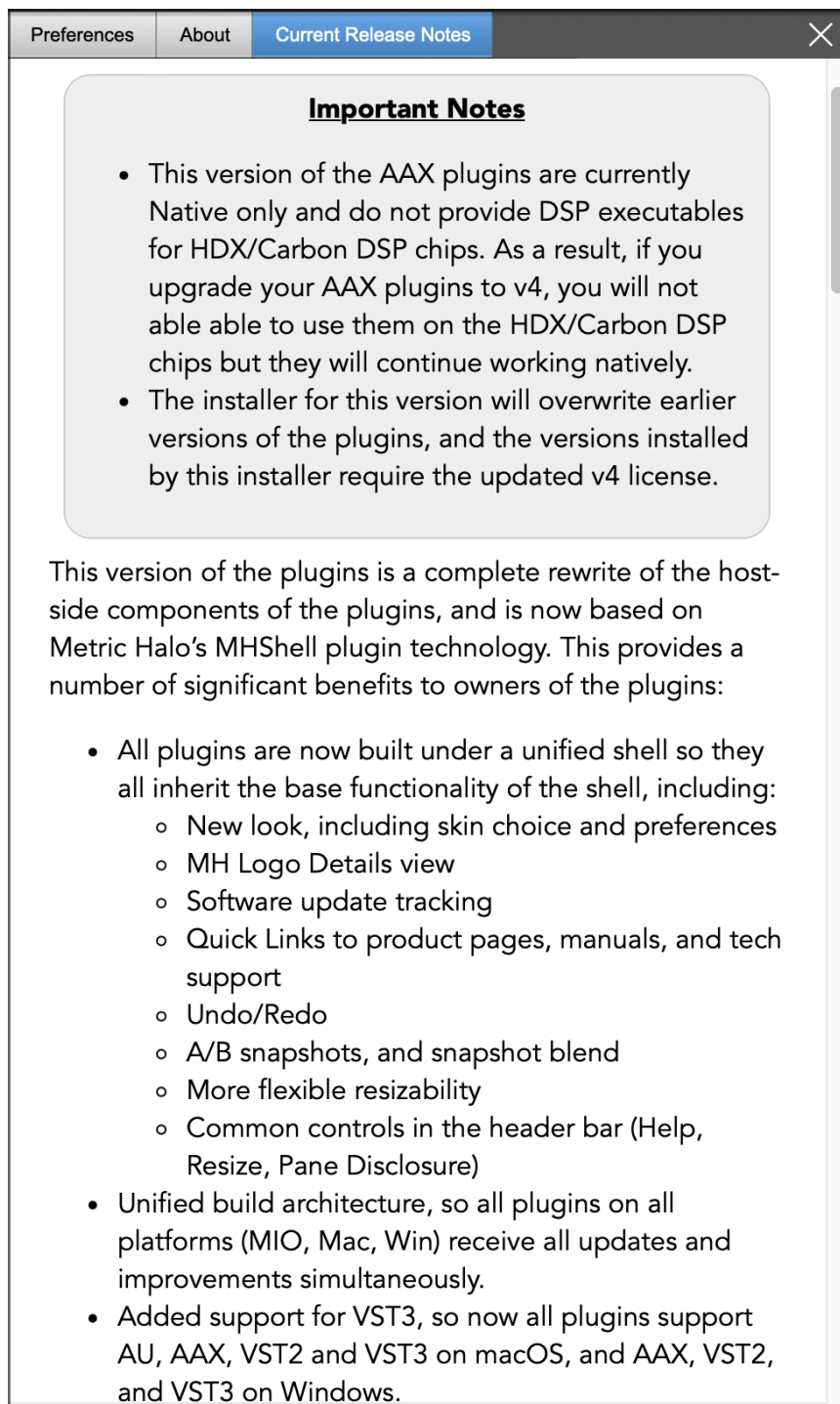
Current Release Notes example

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
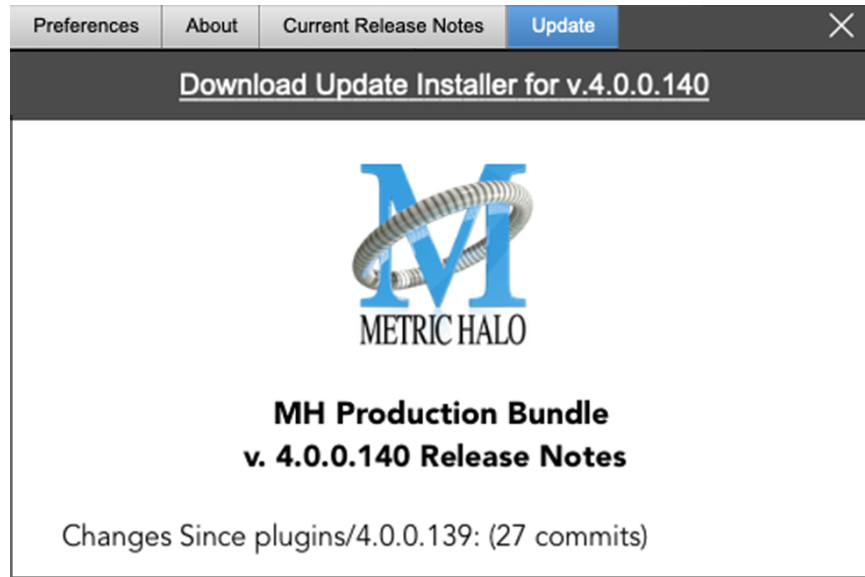
Update tab with new version release notes header
(text from update release notes omitted from above for brevity)

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, you will see new version information with a synposis of the of major changes included in the update, and bugfix revisions for the most recent software releases listed further below. As with the Current Release Notes tab, 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

- <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.
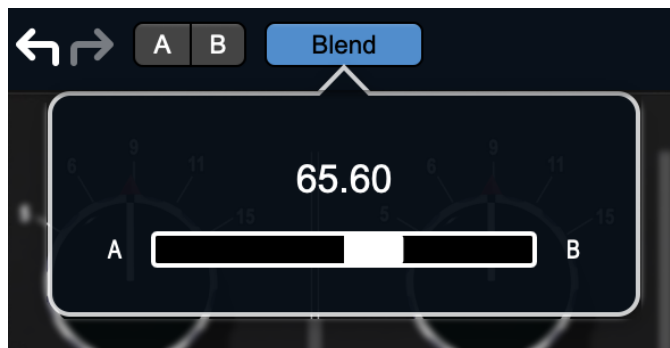
Plug-in Header: Snapshot Blend

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.

Graph Visible Selector

Graphs Closed

Graphs Open

This button toggles visibility of various response graphs. This button allows you to maximize screen re-al-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%.

Compare Button
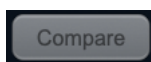
Compare Button (inactive)

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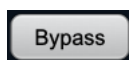
Soft Bypass Button (not bypassed - plug-in is processing)
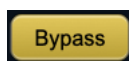
Soft Bypass Button (bypassed - plug-in is *not* processing)

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.

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.

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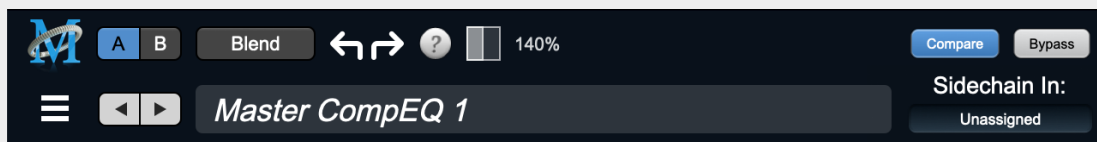
MIOConsole3d ChannelStrip header

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

Preset
step-through

Preset name / selector field

Master CompEQ 1
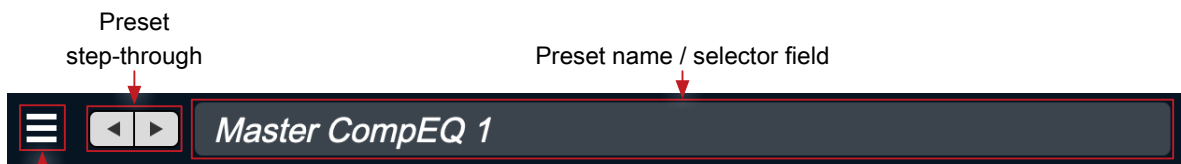
"Hamburger" menu:
Preset operations

Plug-in Hamburger menu

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

Save Parameters
Save Parameters As…

Rename Current Preset…
Delete Current Preset…

Create New Category…
Delete Current Category…

Copy Parameters
Paste Parameters

Factory Default

Reveal In Finder
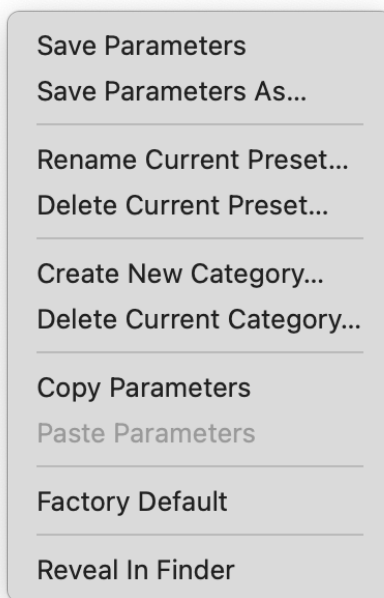
- **Save Parameters** saves the current plug-in parameters to the current preset.
- **Save Parameters As…** opens a dialog box where you can name and choose a category to save your current plug-in settings.
- **Rename Current Preset…** lets you rename the current preset.
- **Delete Current Preset…** 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:
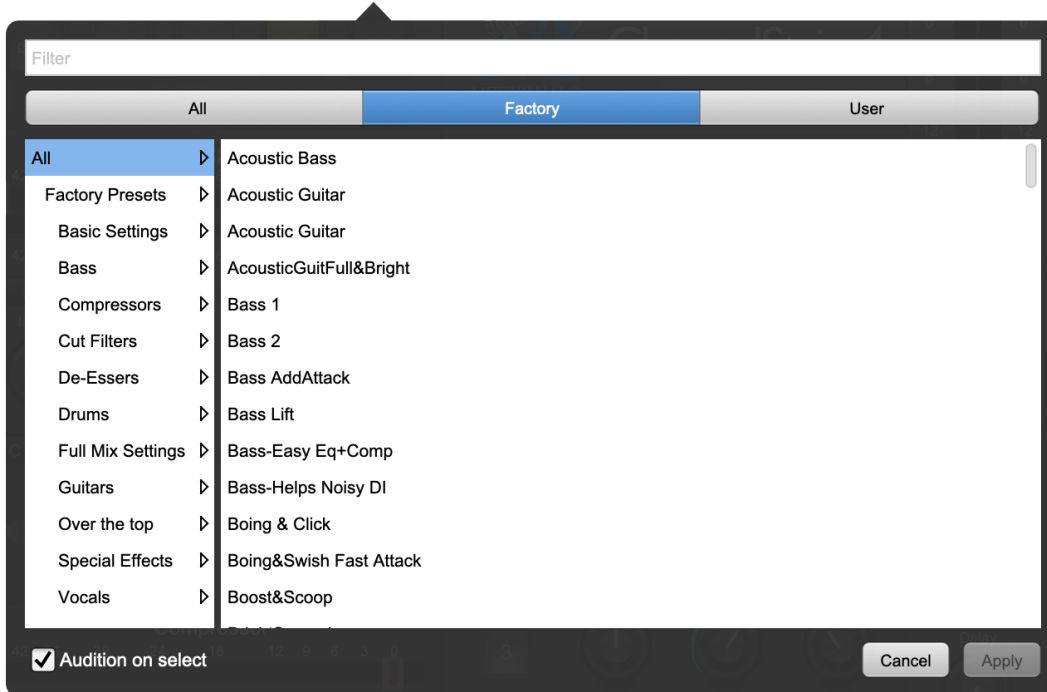
Plug-in Header: Preset selector menu (ChannelStrip shown)

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.

Part I. MH Character

Table of Contents

5. 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.

6. Operation

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

Character Model selector

None	Transformer	Valve	FET
SoftSat	SoftSat Gain	Boutique Tube	American Transformer 1
American Transformer 2	California Tube Mic	California Tube Line	Modern Tube DI
Modern Tube EQ	Modern Tube Soft Sat	Modern Tube Mic LG	Modern Tube Mic MG
Modern Tube Mic HG	Modern Tube Sym	Modern Tube Mic Soft Sat	Classic British Mic Pre
American Solid State	California Vocal Box	California Vocal Box Drive	British Mic Pre Clone

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
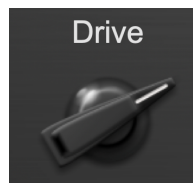
Control Knobs are used to control the Drive and Output Gain parameters. You can change the value of each knob in a number of different ways. Click and drag the knob to change the value continuously. Dragging up or to the right will increase the value, while dragging down or to the left will decrease 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
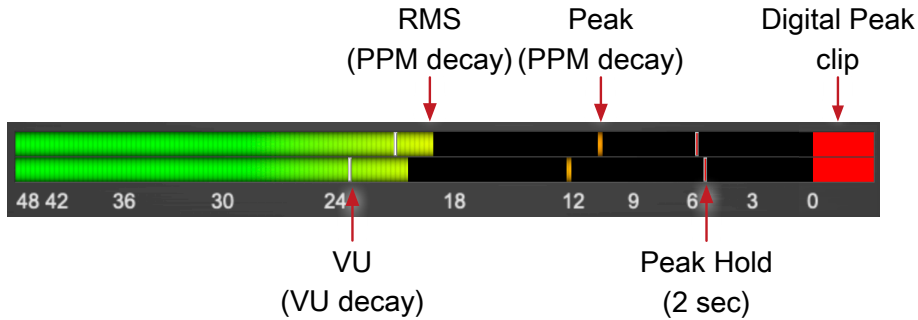
For the main output stage of Character we have provided meters driven with SpectraFoo™ metering technology. These meters show digital Peak, Peak Hold, RMS level and VU level.

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.

7. Processing

A Detailed Description

In this chapter we discuss what each parameter does and how the controls work. While Character has a fixed number of model types, there are an infinite number of combinations in conjunction with the Drive control.
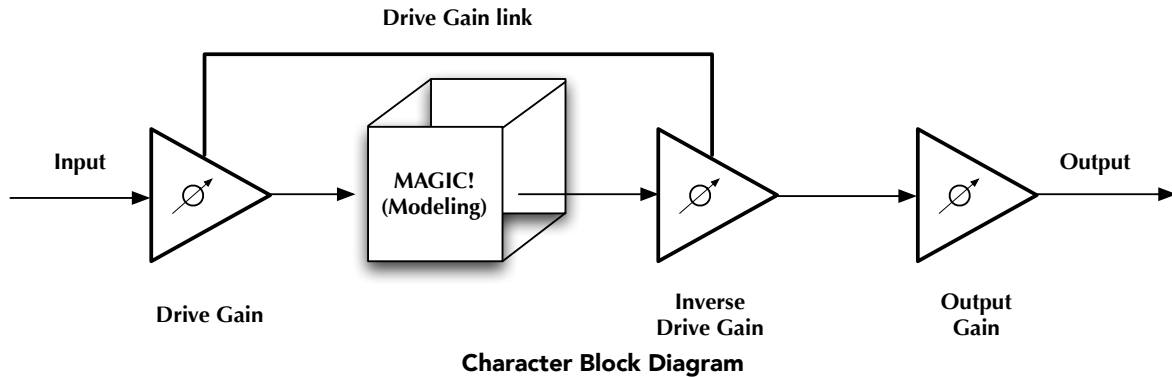
Drive Gain link

Input → ∅ → **MAGIC! (Modeling)** → ∅ → ∅ → **Output**

Drive Gain   **Inverse Drive Gain**   **Output Gain**

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!

Part II. MH ChannelStrip

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8. ChannelStrip Overview

ChannelStrip is a plug-in for digital audio workstations which provides the essential basic channel processing found in the channel strip of a modern mixing console.

Processing functions include:

- Input level control
- Phase Invert
- Expander/Gate with filtered sidechain
- Compressor with filtered sidechain
- 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

9. Operation

As with most channel strips, ChannelStrip provides many copies of controls that are all operated in a similar manner. The ChannelStrip user interface uses a few different control elements to control all of the processing. These elements are:

Control Knob

Control Knobs are used to control the value of various continuous parameters of a process. Examples of these types of parameters include: Attack time, Release Time, Threshold, etc. There are four styles of encoders:

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.

    Hint: If you hold down the ⌘ (Command) key when you click, you will be able to adjust the value with finer precision.
- ⌥ (Option)-click to reset any knob to its default value. You may also double-click a knob to reset it.
- **Rapid parameter entry**
    - Double-click the numeric readout of the knob to type in a number directly. The text-entry field will remain active until you dismiss it by clicking somewhere else or hitting the **return**, **enter**, **tab**, or ⌘. (Command + .) keys.
    - Hit **return** or **enter** to confirm the value and dismiss the pop-up.
    - Hit the **tab** key to confirm the value and immediately activate the entry field for the next control. ⇧-**tab** (Shift + tab) will display the entry field for the previous control).
    - Hit ⌘. (Command + .) or **ESC** (Escape) to dismiss the pop-up and cancel the change.
    - When you enter a number into the pop-up entry, you can use a couple of abbreviations: "k" multiplies the number by 1000 and "m" divides the number by 1000. So if you want to enter 16,500 Hz you can just type 16.5k.

Toggle Button

Toggle Buttons

Toggle buttons are simple on/off switches. They light up when they are on and are dark when they are off. You toggle the state of the button by clicking on it. These buttons are used to enable processor sections, select keying sources and to switch the order of processing within ChannelStrip.

Fader
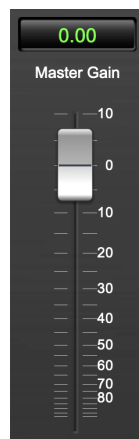
Master Fader

The fader is unique in that only one fader is used in the interface for ChannelStrip. It works in much the same fashion as the control knobs. Instead of dragging up/right or down/left to change the value, you

directly drag the fader knob. The other "tricks" described for the knobs also work with the fader. The fader is used to control the master output gain of the plug-in before the limiter stage.

Filter Type

Each filter band in the strip (6 EQ bands and 2 Side-chain bands) has a filter type control that allows you to choose the shape of the filter applied by that band. Each band provides 6 different types of filter shapes:

- **Peaking/Parametric** – a second order bell-shaped parametric boost/cut filter. The Gain control has a boost/cut range of ± 24 dB. When the boost is greater than +15 dB the filter gains a resonant quality. The center frequency of the filter can be any frequency between 20 Hz and 20 kHz. The bandwidth of the filter is continuously variable between 0.1 octaves and 2.5 octaves.

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).

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
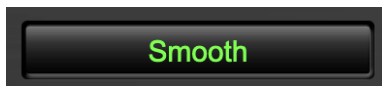
Compressor Character Switch

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 are:

Input Meters

Peak
(PPM decay)

Peak Hold
(2 sec)

| 42 | 36 | 30 | 24 | 18 | 12 | 9 | 6 | 3 | 0 |

RMS
(PPM decay)

Detector

Threshold
control

Gate/Expander Peak Meter

RMS
(PPM decay)

Peak
(PPM decay)

| 24 | 18 | 12 | 9 | 6 | 3 | 0 |

Threshold
control

Detector

Peak Hold
(2 sec)

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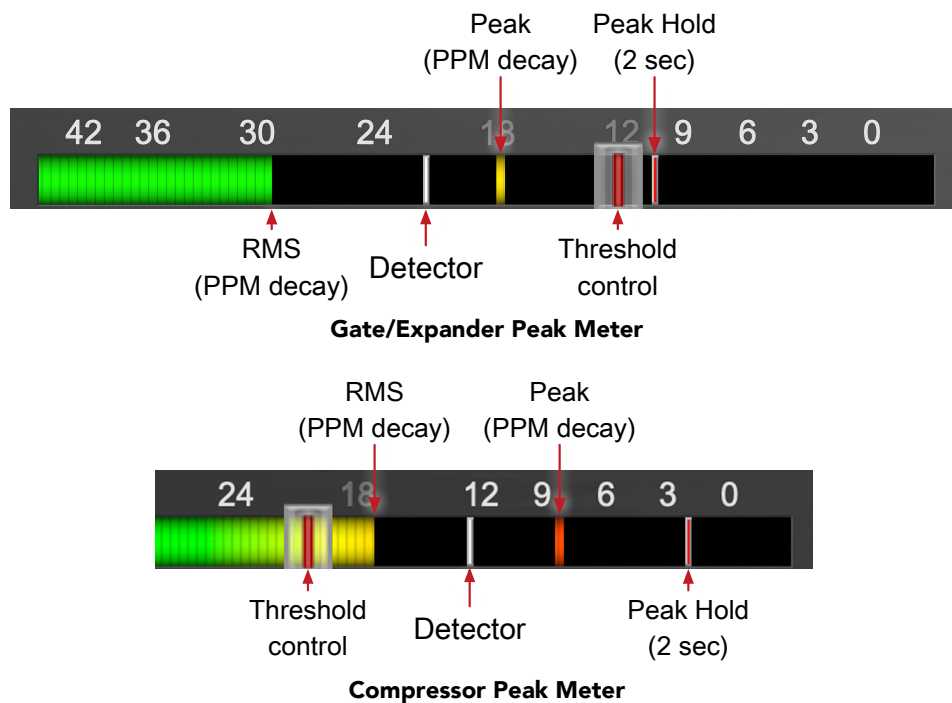
SpectraFoo™ multi-function metering is provided at the input stage of ChannelStrip's Gate/Expander and Compressor processing blocks.

- 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
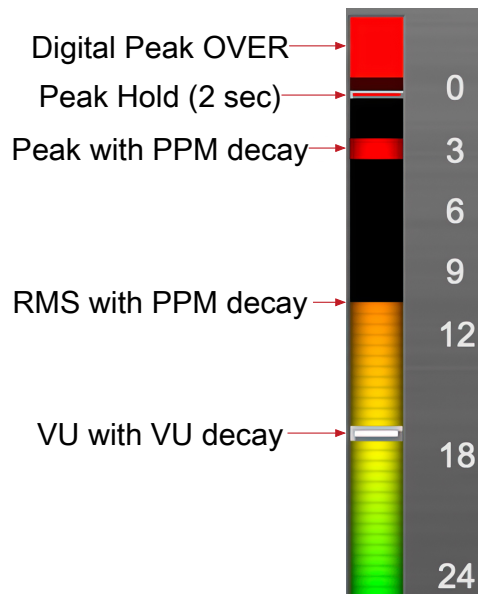
Output Meter (mono / enlarged)

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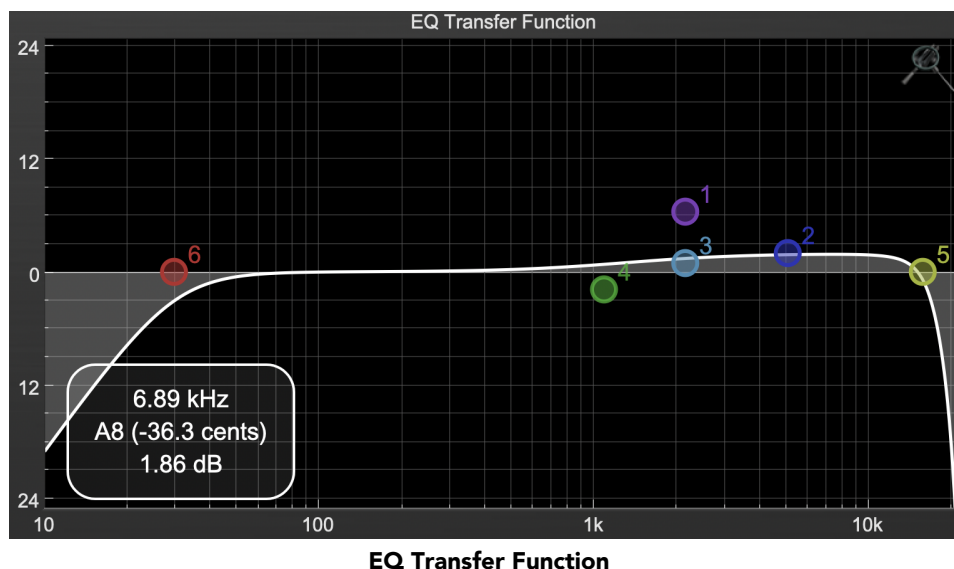
The following information on the EQ Transfer Function and Spectragraph Analyzer apply to the EQ processing block as well as the sidechain filter displays of the gate and compressor.

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.

```
      ±3 dB
      ±6 dB
 ✓ ±12 dB
     ±24 dB
     ±36 dB

 ✓ Auto Enable Filter when Filter Parameter Changes

 ✓ Show Live Readout
 ✓ Require Option for scrollwheel change of BW
```

- 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
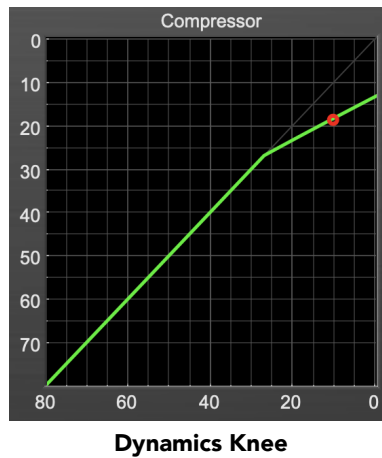
Dynamics Knee

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.

10. Processing

A Detailed Description

In this chapter we'll discuss what each processing block does and how the controls work.
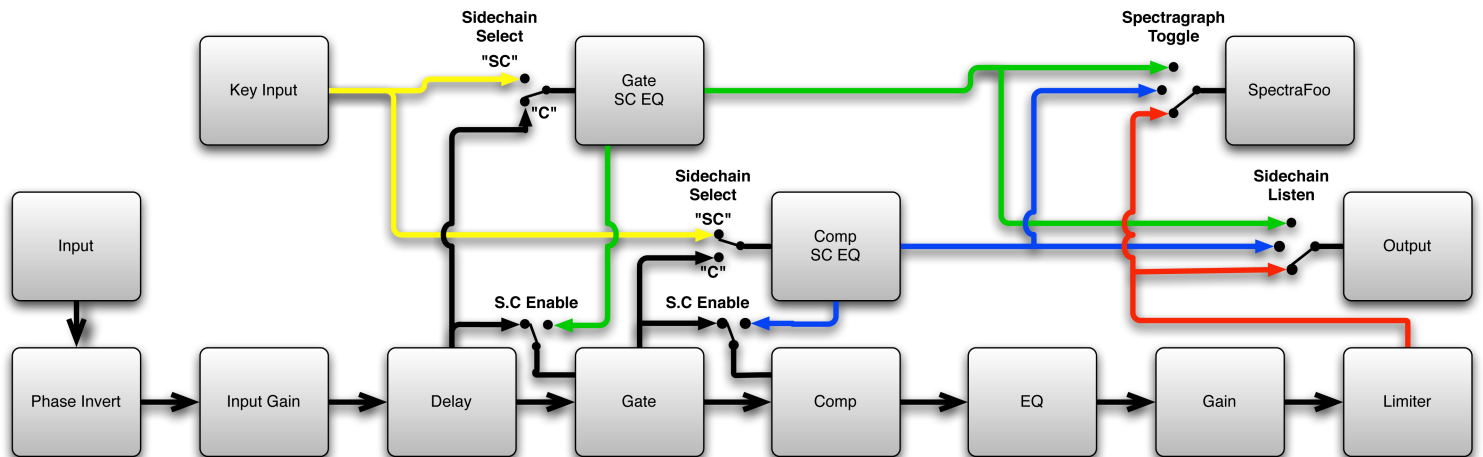
Signal Flow: Compressor is Pre-EQ
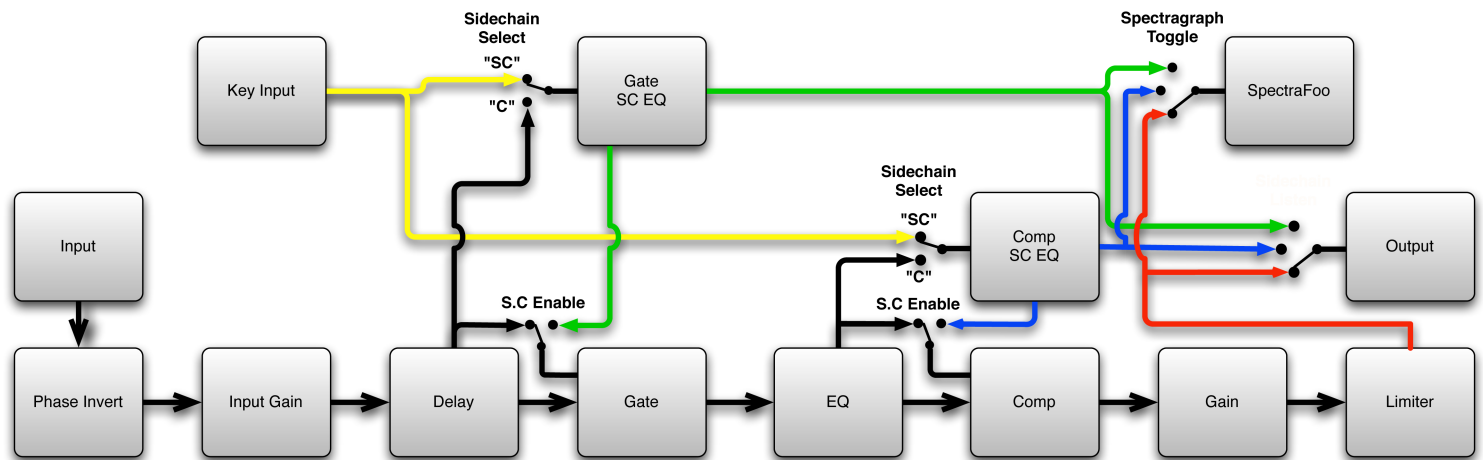
Signal Flow: Compressor is Post-EQ

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.

Part III. MH Dirty Delay

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11. 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.

12. 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
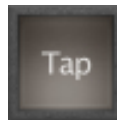
The Tempo Tap Pad allows you to set the delay time in BPM with successive mouse clicks. Tap in quarter-note beats to set the Tempo.

Note Length
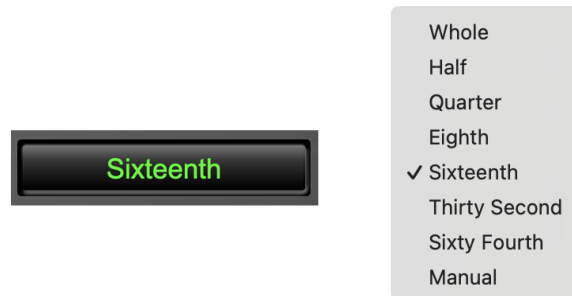
Note Length Selector and Popup Menu

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
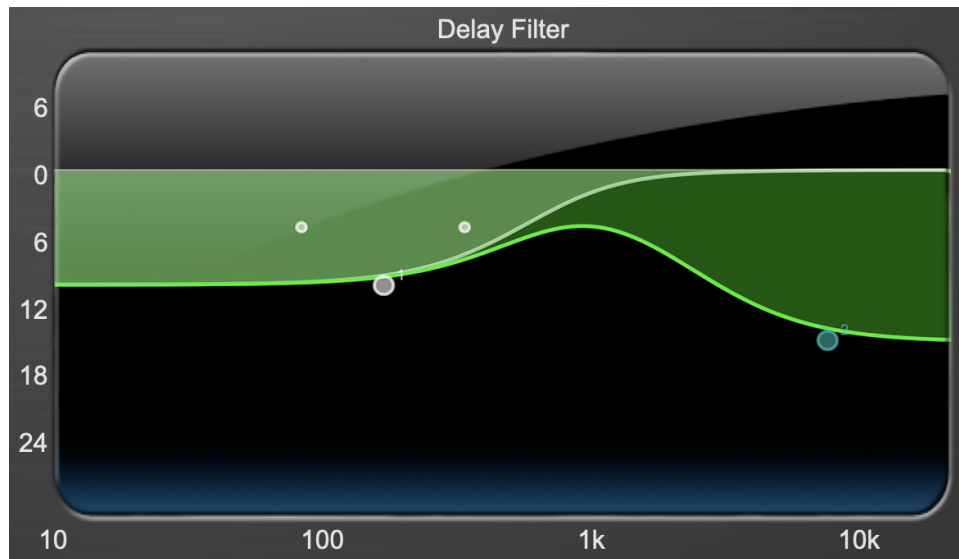
This section allows detailed control over the 2-band EQ applied to the delayed signal.

- 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

Peaking/Parametric – a second order bell-shaped parametric boost/cut filter. Boost/cut has a range of ± 24 dB. When the boost is greater than +15 dB the filter gains a resonant quality. The center frequency of the filter can be any frequency between 20 Hz and 20 kHz. The bandwidth of the filter is continuously variable between 0.1 octaves and 2.5 octaves.

Low Cut

Low Cut – a 12 dB/octave low cut filter with a -3dB point that is continuously adjustable between 20 Hz and 20 kHz.

High Cut

High Cut – a 12 dB/octave high cut filter with a -3dB point that is continuously adjustable between 20 Hz and 20 kHz.

Low Shelf

Low Shelf – a shelving filter that applies boost/cut to low frequencies. Boost/cut is limited to +12 dB/–24dB. The bandwidth controls the dip/peak that is added at the end of the transition band.

High Shelf

High Shelf – a shelving filter that applies boost/cut to high frequencies. Boost/cut is limited to +12 dB/–24dB. The bandwidth controls the dip/peak that is added at the end of the transition band.

Bandpass

Bandpass – a bandpass filter with 6dB per octave skirt on the high and low ends of the pass band. The width of the pass band can be adjusted between 0.1 octaves and 2.5 octaves and the center of the pass band is continuously adjustable between 20 Hz and 20 kHz.

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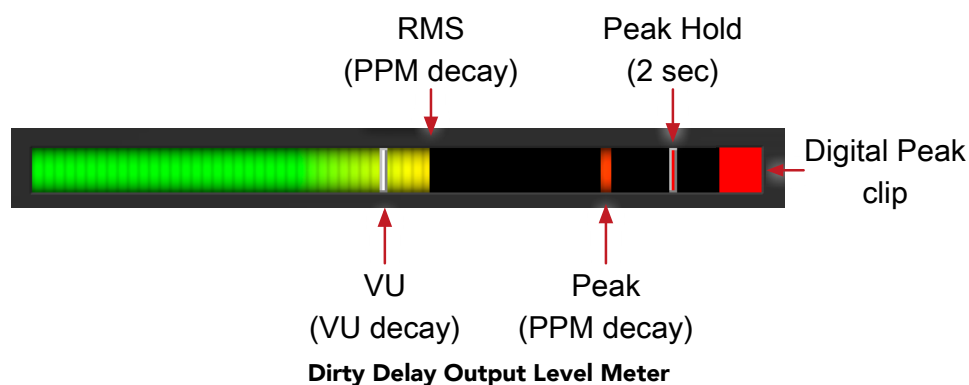
For the main output stage of Dirty Delay we have provided meters driven with SpectraFoo™ metering technology. These meters show digital Peak, Peak Hold, RMS level and VU level.

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.

13. Processing

A Detailed Description

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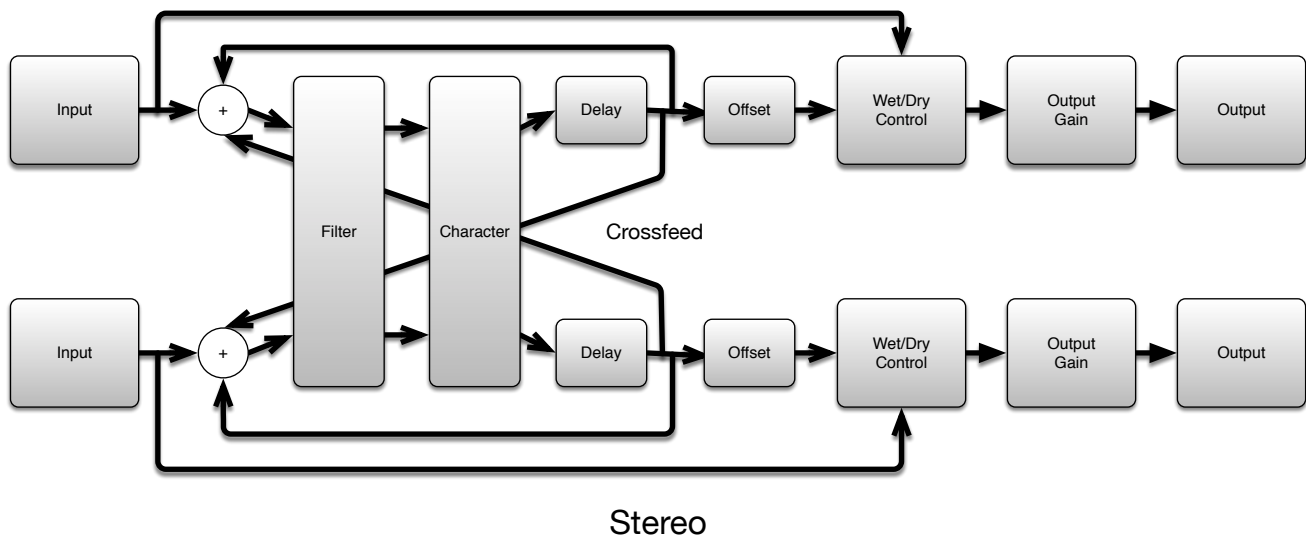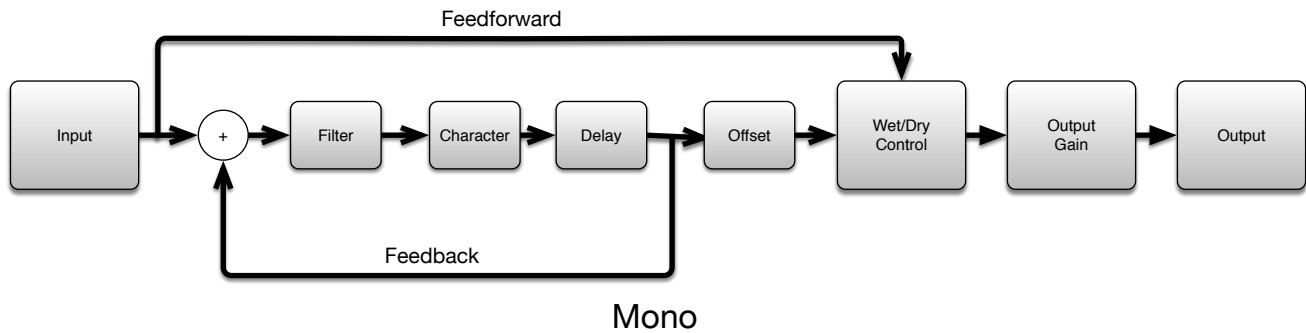
Mono

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.
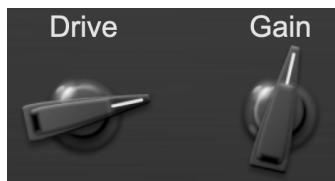
Character Drive and Gain controls

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.

Part IV. MH HaloVerb

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14. 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

15. 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:
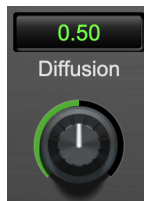
Swept Knob

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

Digital Peak clip

Peak Hold (2 sec)

Peak (PPM decay)

RMS (PPM decay)

VU (VU decay)
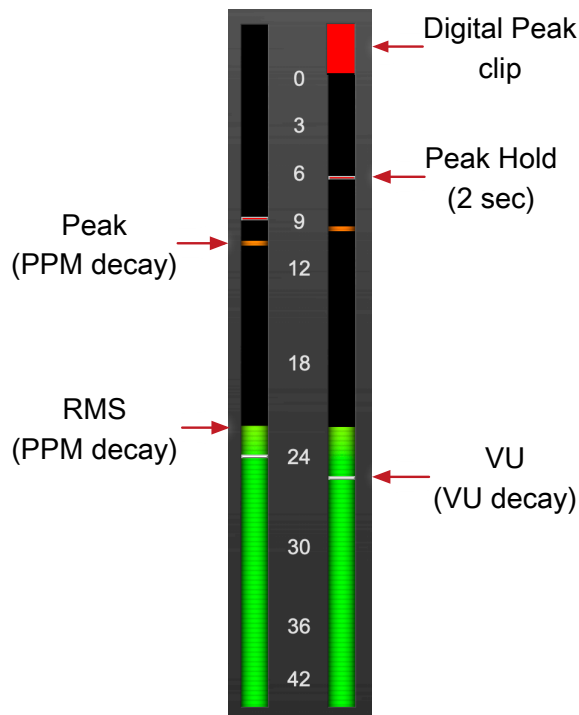
HaloVerb Output Level Meters

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
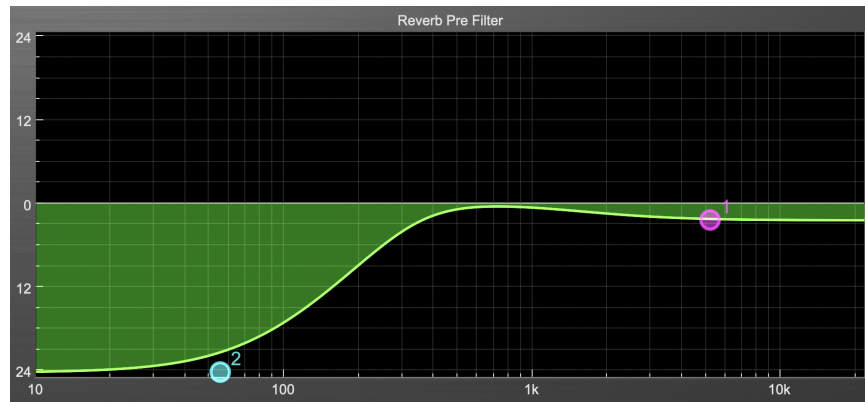
Low and High Frequency Pre-filter

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
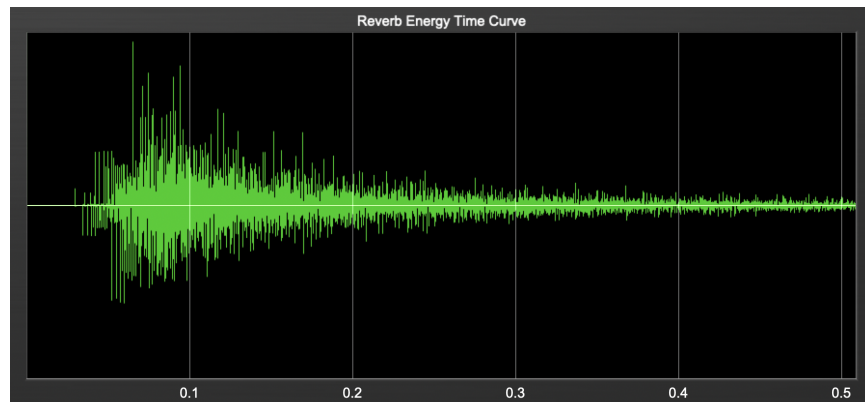
Reverb Impulse Display

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.
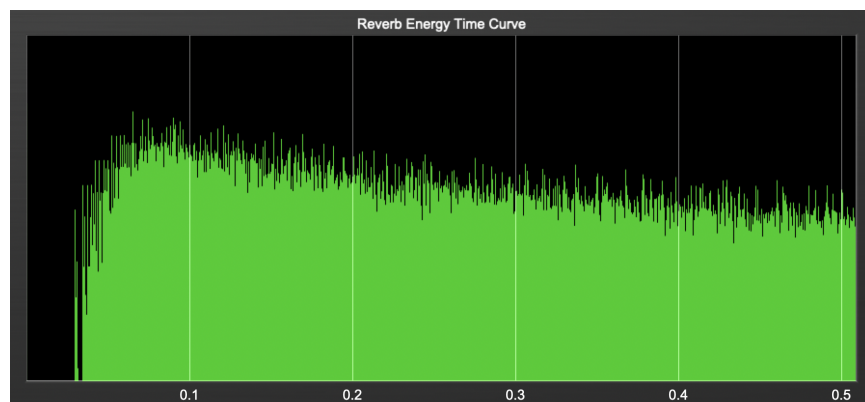
Energy Time Curve

16. Processing

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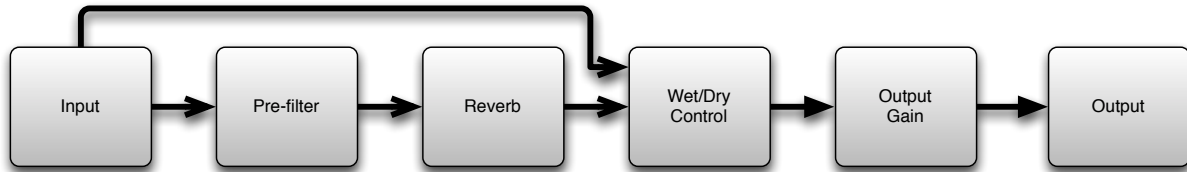
In this chapter we'll discuss how the controls work.

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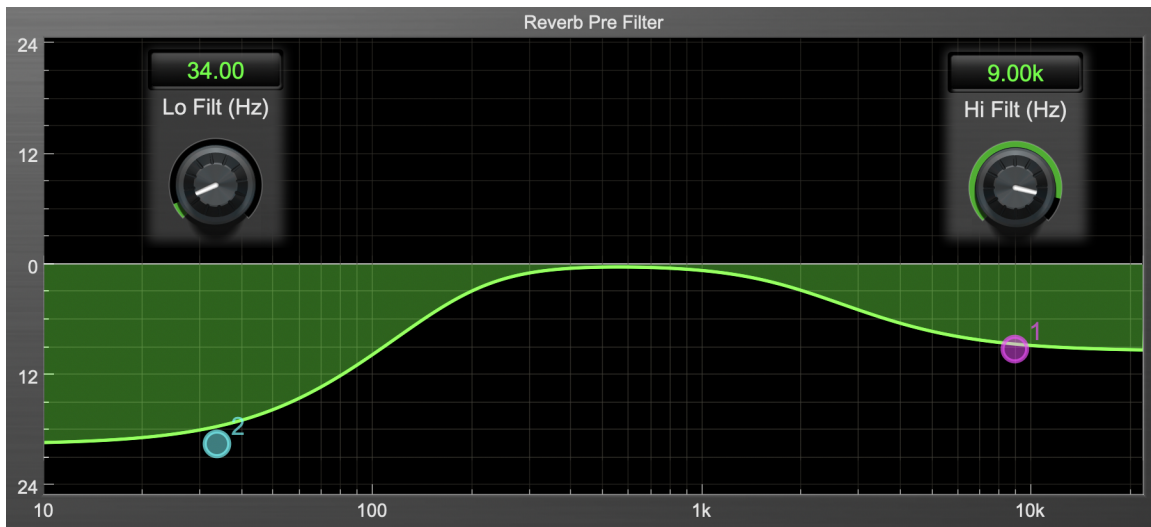
Pre-filters at 34 Hz and 9kHz (controls shown graphically overlaid for clarity)

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:
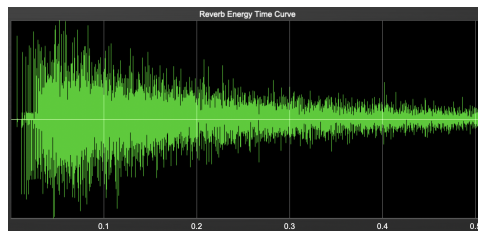
Predelay of 0 ms
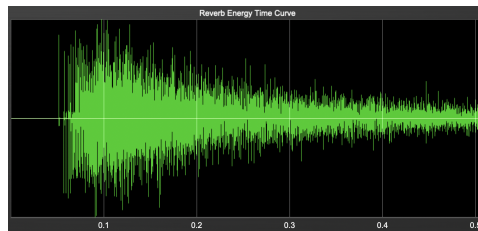
Predelay of 45 ms

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:
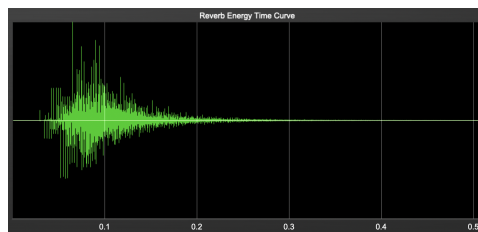
RT60 of .1 Seconds
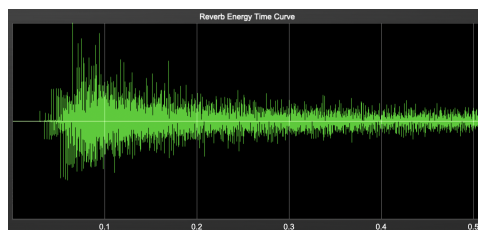
RT60 of 7 Seconds

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.
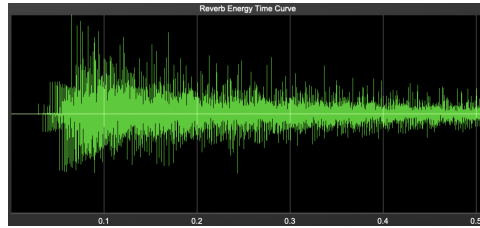
Damping of 0%
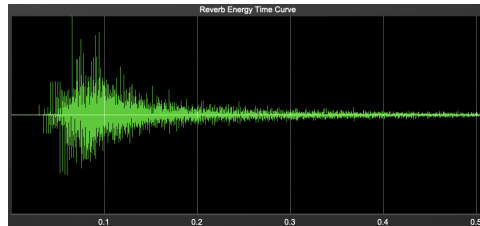
Damping of 100%

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.
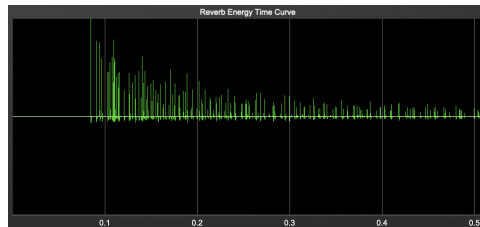
Diffusion of 0%
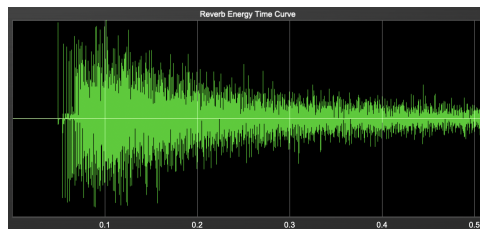
Diffusion of 75%

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.

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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

Part V. MH Multiband Dynamics

Table of Contents

17. 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
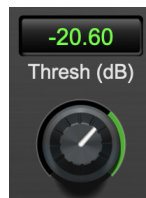
Multiband Dynamics's User Interface

18. 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

Ena

Ena

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

0.00

Out Gain (dB)

— 10

— 0

— 10

— 20

— 30

— 40

— 50
— 60
— 70
— 80

Output Gain
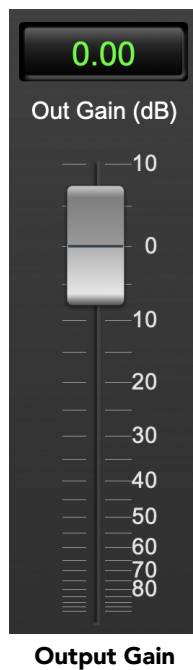
The fader is unique in that only one fader is used in the interface for Multiband Dynamics. It works in much the same fashion as the control knobs. Instead of dragging up/right or down/left to change the value, you directly drag the fader knob. The other "tricks" described for the knobs also work with the fader. The fader is used to control the master output gain of the plug-in.

Band Solo Button

Band Solo Button   Band Solo Button
(Disabled)   (Enabled)

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

Master Link Control

| Leader:
Followers will follow changes
made to this bands' parameters | Follower:
changes made to the Leaders'
parameters will also affect this band | 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

Output Meter

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.

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 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
- ±24 dB
- ±36 dB

Spectragraph Analyzer

Clicking the SpectraFoo™ logo in the upper right hand corner of the transfer function will activate the spectragraph, showing the realtime frequency analysis of your signal:

Spectragraph Display

The traces are:

- White: Left channel instantaneous display

- Red: Right channel instantaneous display

- Yellow: Left channel average display

- Blue: Right channel average display

The *instantaneous* trace updates in real-time, allowing you to see the immediate peak level of your audio. The *average* trace displays the level as averaged over a short period, giving you a more general view.

The spectragraph analyses the signal post-output gain, allowing you to see the effects of the processors. To disable the spectragraph entirely, click the active 'Foo icon.

If you right-click (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.

- Show Left Channel: Toggles the left channel spectragraph display.

- Show Right Channel: Toggles the right channel spectragraph display.

These settings are stored for each transfer function window separately, and for each instance of Multiband Dynamics.

Dynamics Knee
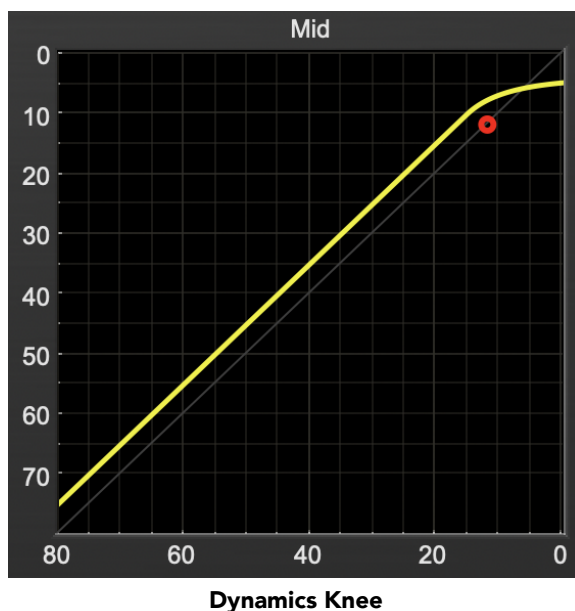
Dynamics Knee

Multiband Dynamics contains a Dynamics Knee diagram for each compressor processing section. The diagram provides feedback on the response of the associated dynamics processor. Both the horizontal and vertical axes are calibrated in dBFS.

The horizontal axis corresponds to the input level and the vertical axis represents the output level. The heavy line shows the dynamic of the associated processing block. This means that if you sent in a sine wave at a given input level, the output level would be equal to the level shown on the graph. When the processor is working with real dynamic signals, the graph is a good approximation of the response when the attack is fast and the release is slow.

In most cases, however, the dynamic response of the processor will not match its static response. In order to represent this, we have included a "bouncing ball" meter for both the gate and the compressor. This metering is shown as a red circle that is overlaid on the knee diagram. The red circle is placed so its horizontal position is equal to the instantaneous input level and its vertical level is equal to the instantaneous output level. Examining this meter while you are adjusting the dynamics controls will provide you with a great deal of information about how the processor is operating and how the controls interact.

19. Processing

A Detailed Description

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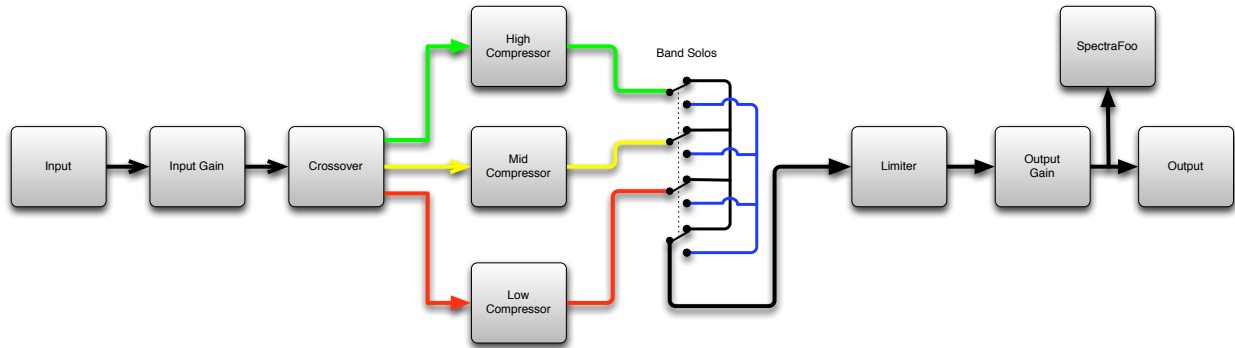
Multiband Dynamics Signal Flow

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
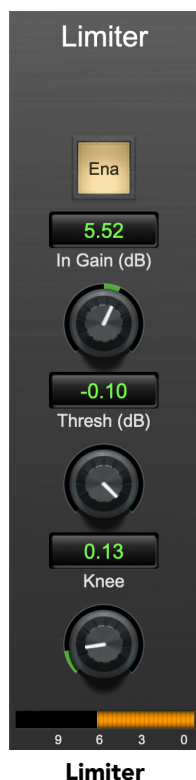
Limiter

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.

Limiter Knee Control

The "Knee" knob allows you to adjust shape of the limiter gain reduction. When the Knee control is set to 0, the transfer function of the limiter is a classic "hard-knee" in which the compressor applies no gain reduction when the detector is below the threshold, and the gain is reduced by the ratio when the detector level is at the threshold.

Part VI. MH Multiband Expander

Table of Contents

20. 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

21. 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.

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 expander sections.

Fader

Output Gain
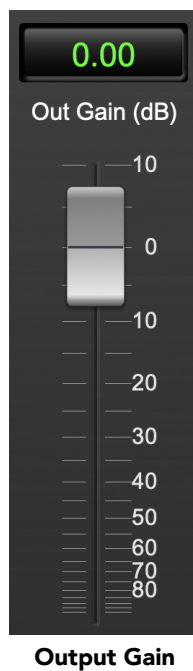
The fader is unique in that only one fader is used in the interface for Multiband Expander. It works in much the same fashion as the control knobs. Instead of dragging up/right or down/left to change the value, you directly drag the fader knob. The other "tricks" described for the knobs also work with the fader. The fader is used to control the master output gain of the plug-in.

Band Solo Button

Band Solo Button   Band Solo Button
(Disabled)   (Enabled)

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

Master Link Control

Leader:	Follower:	Not Linked:
Followers will follow changes	changes made to the Leaders'	does it's own thing
made to this bands' parameters	parameters will also affect this band	

Lead and Follow Link Buttons

Each expander 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, 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

Digital Peak
clip

Peak Hold
(2 sec)

Peak
(PPM decay)

RMS
(PPM decay)

VU
(VU decay)

0
3
6
9
12
18
24
30
36
42

Output Meters

For the main output stage of Multiband Expander 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 Multiband Expander 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
- ±24 dB
- ±36 dB

Spectragraph Analyzer

Clicking the SpectraFoo™ logo in the upper right hand corner of the transfer function will activate the spectragraph, showing the realtime frequency analysis of your signal:

Spectragraph Display

The traces are:

- White: Left channel instantaneous display

- Red: Right channel instantaneous display

- Yellow: Left channel average display

- Blue: Right channel average display

The *instantaneous* trace updates in real-time, allowing you to see the immediate peak level of your audio. The *average* trace displays the level as averaged over a short period, giving you a more general view.

The spectragraph analyses the signal post-output gain, allowing you to see the effects of the processors. To disable the spectragraph entirely, click the active 'Foo icon.

If you right-click (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.

- Show Left Channel: Toggles the left channel spectragraph display.

- Show Right Channel: Toggles the right channel spectragraph display.

These settings are stored for each transfer function window separately, and for each instance of Multiband Expander.

Dynamics Knee
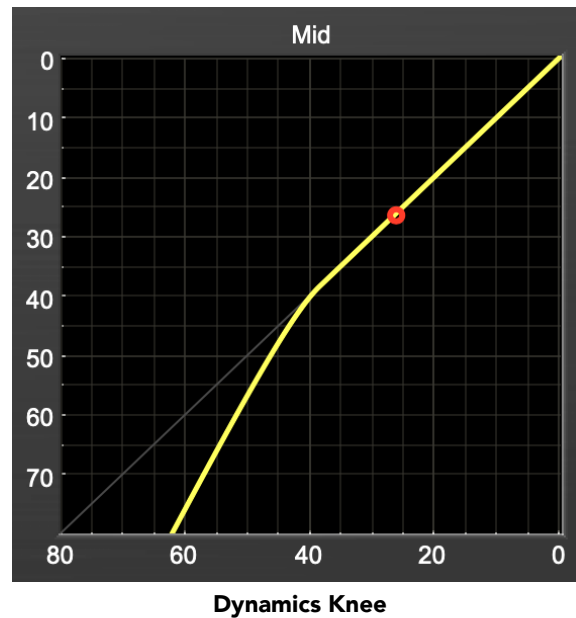
Dynamics Knee

Multiband Expander contains a Dynamics Knee diagram for each expander section. The diagram provides feedback on the response of the associated dynamics processor. Both the horizontal and vertical axes are calibrated in dBFS.

The horizontal axis corresponds to the input level and the vertical axis represents the output level. The heavy line shows the dynamic of the associated processing block. This means that if you sent in a sine wave at a given input level, the output level would be equal to the level shown on the graph. When the processor is working with real dynamic signals, the graph is a good approximation of the response when the attack is fast and the release is slow.

In most cases, however, the dynamic response of the processor will not match its static response. In order to represent this, we have included a "bouncing ball" meter. 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.

22. Processing

A Detailed Description

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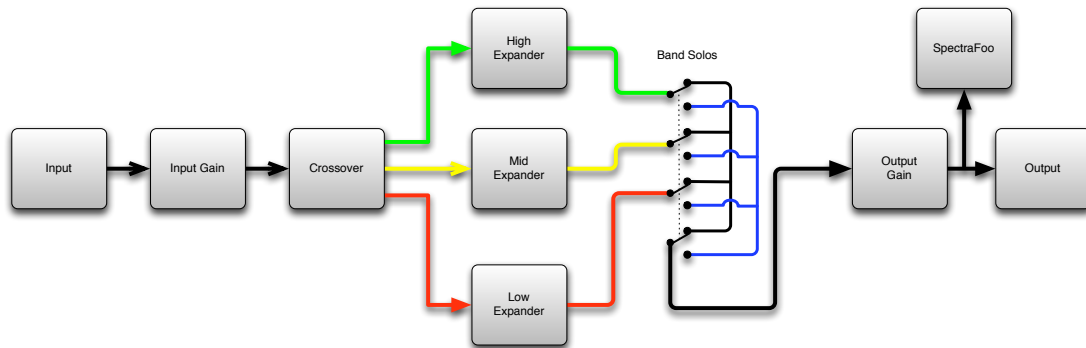
Multiband Expander Signal Flow

The block diagram above illustrates the overall structure of the processing system provided by Multiband Expander. 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 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
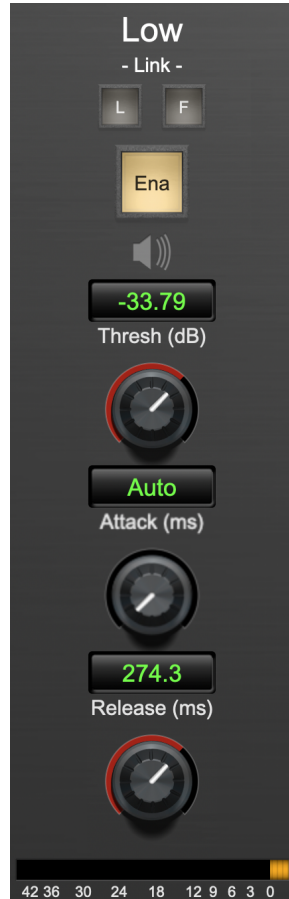
Expander (Low Frequency Shown)

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.

Part VII. MH Precision DeEsser

Table of Contents

23. 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
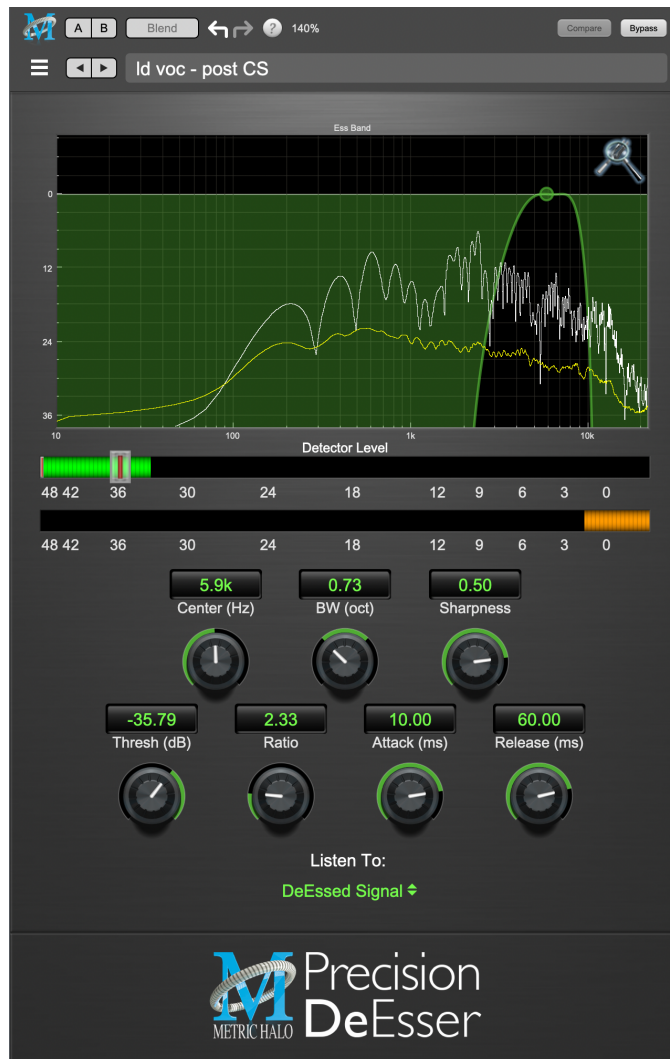
Precision DeEsser's User Interface

24. 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
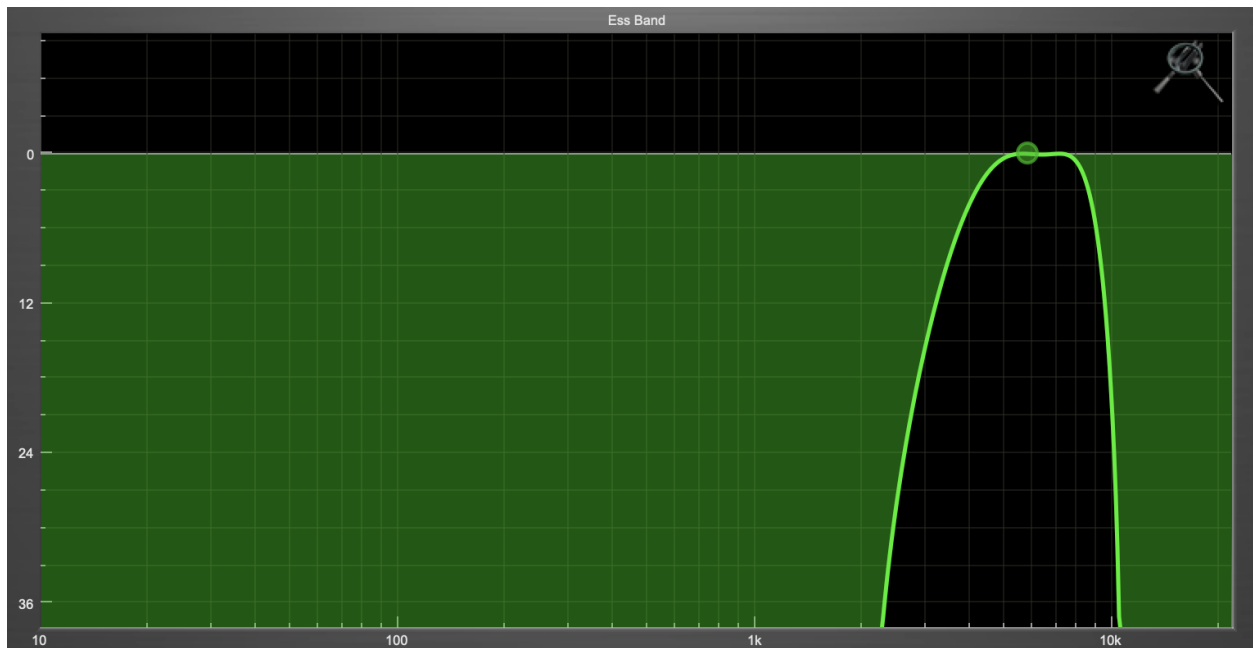
The ess band transfer function is a combination of a visual representation of how the de-esser is processing the signal and an intuitive controller for the associated filter. This display is sometimes called a "Cartesian Graph" by other manufacturers.

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:
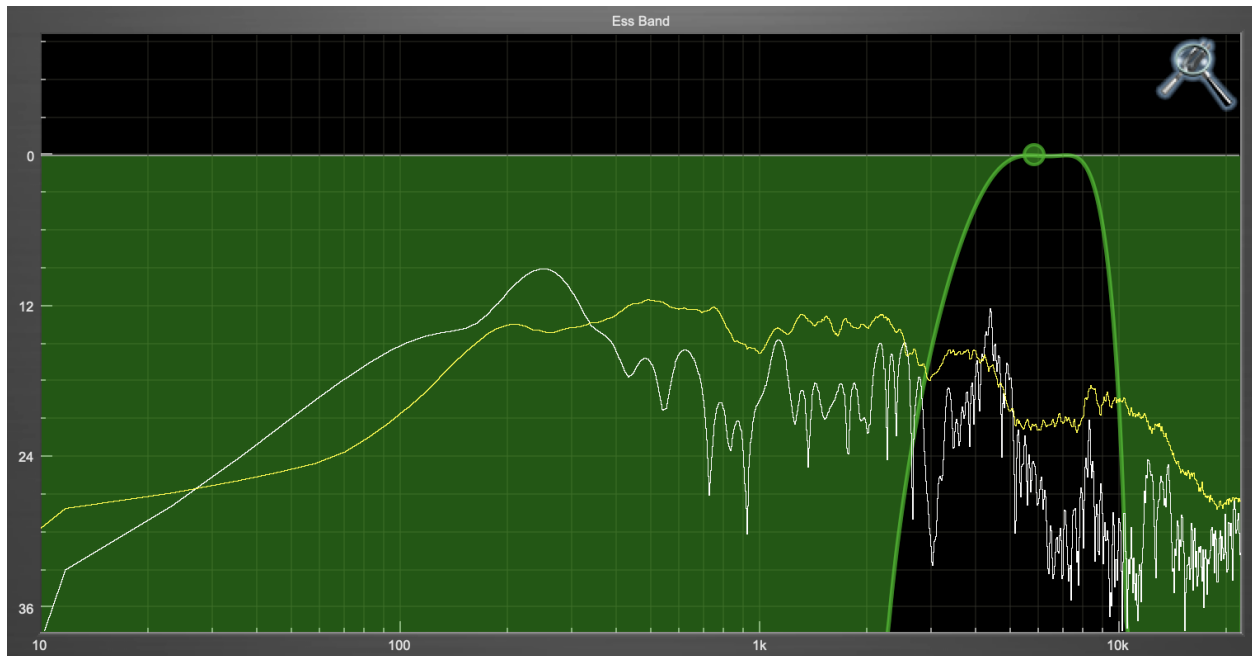
The traces are:

- 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.

25. Processing

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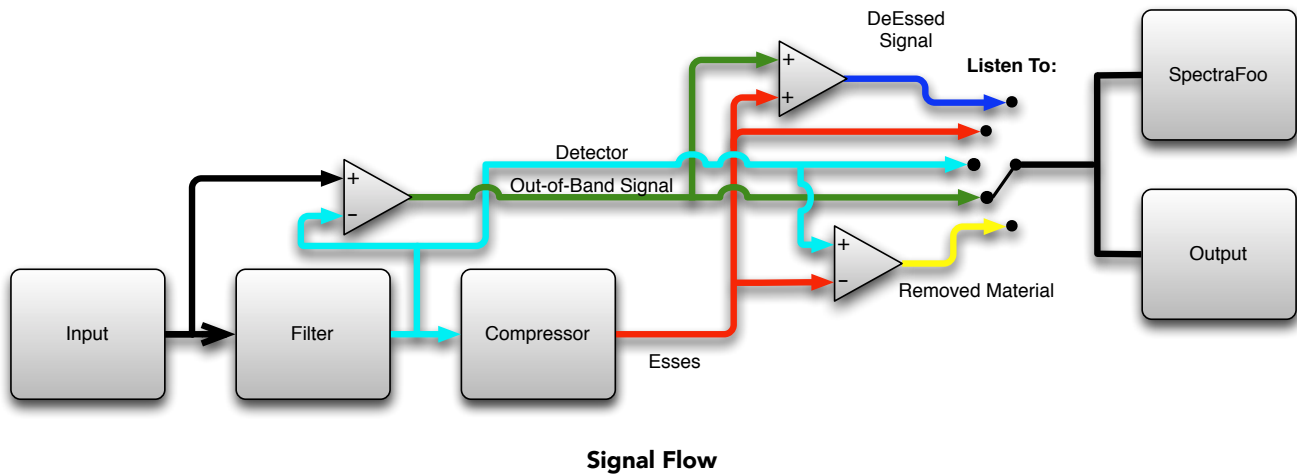
In this chapter we'll discuss how the controls work.

Signal Flow

The block diagram above illustrates the overall structure of the processing system provided by 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.

Part VIII. MH Sonic EQ

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26. 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
- 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 principals, 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.

27. 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).
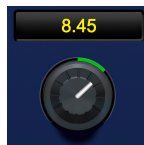
Plus/Minus Knob

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
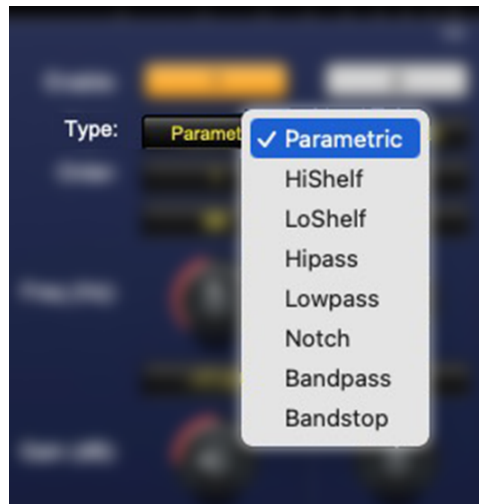
Filter Type selector (with submenu)

Below the band on/off toggle is the filter selector. Click this control to bring up the filter selection submenu.

Filter Order selector
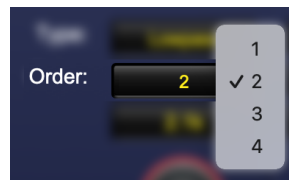
Filter Order selector (with submenu)

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
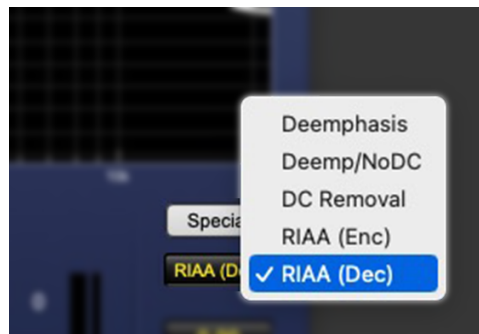
Special Filters selector (with submenu)

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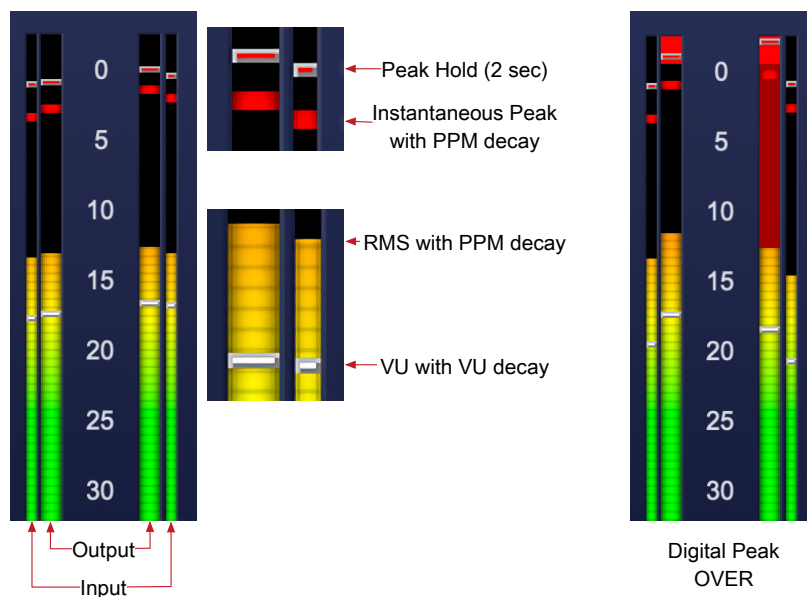
Sonic EQ's multifunction level meters simultaneously display instantaneous Peak, Peak Hold, RMS and VU levels for both the pre-EQ source input and post-EQ output levels.

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
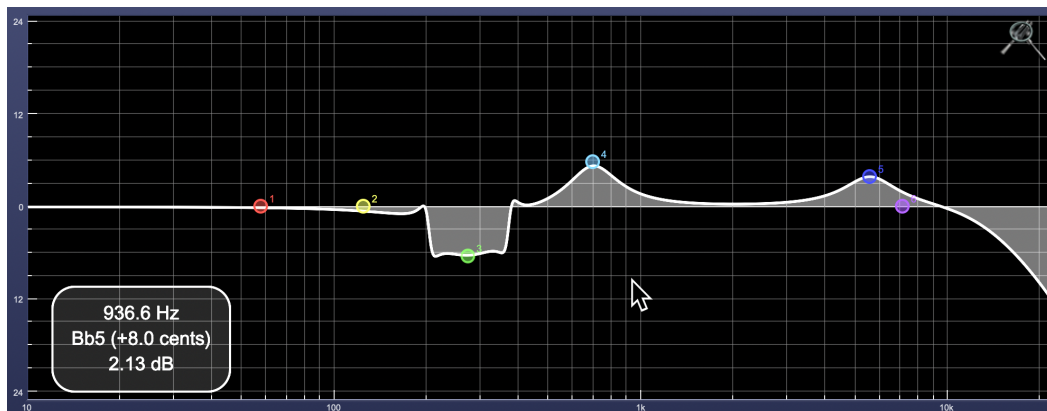
EQ Transfer Function

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.

```
±3 dB
±6 dB
✓ ±12 dB
±24 dB
±36 dB
─────────────────────────────────────
✓ Auto Enable Filter when Filter Parameter Changes

✓ Show Live Readout
✓ Require Option for scrollwheel change of BW
```

- 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.

28. 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
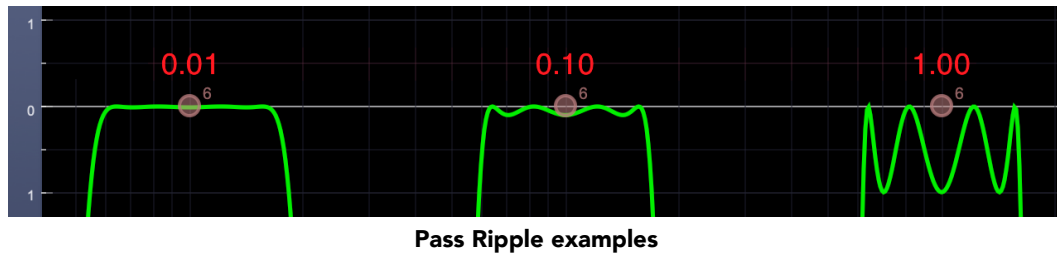
This figure shows the passband ripple of a Bandpass filter at the minimum 0.01dB setting, 0.10db (the default) and 1.00dB (EQ curve colored green in the UI Prefs for visibility).

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
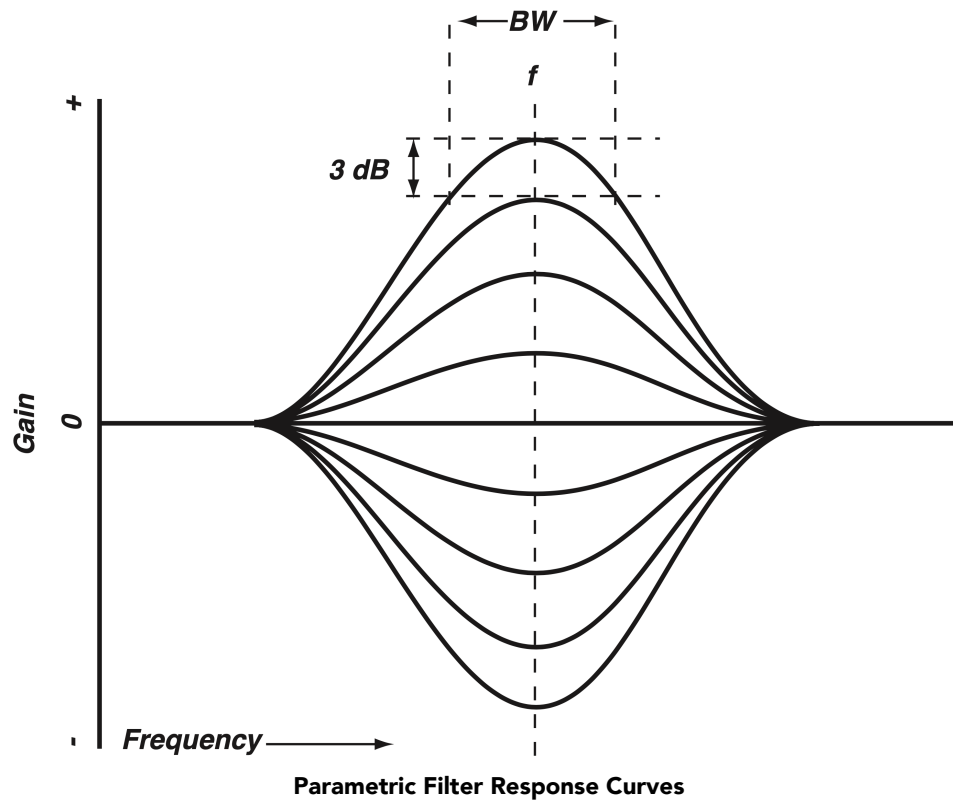
A parametric filter affects a defined region of the audio spectrum.

- **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).
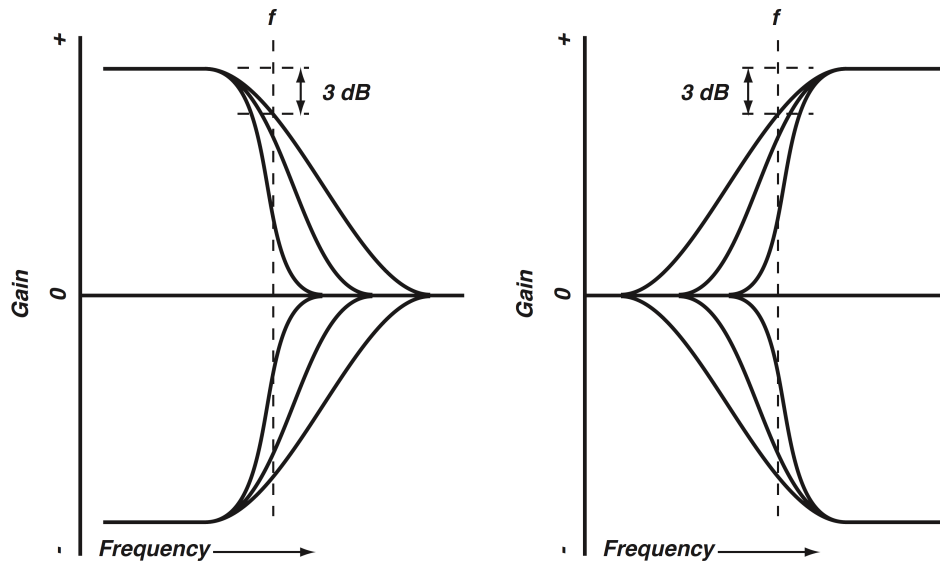
LoShelf (left) and HiShelf (right) Filter Response Curves

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
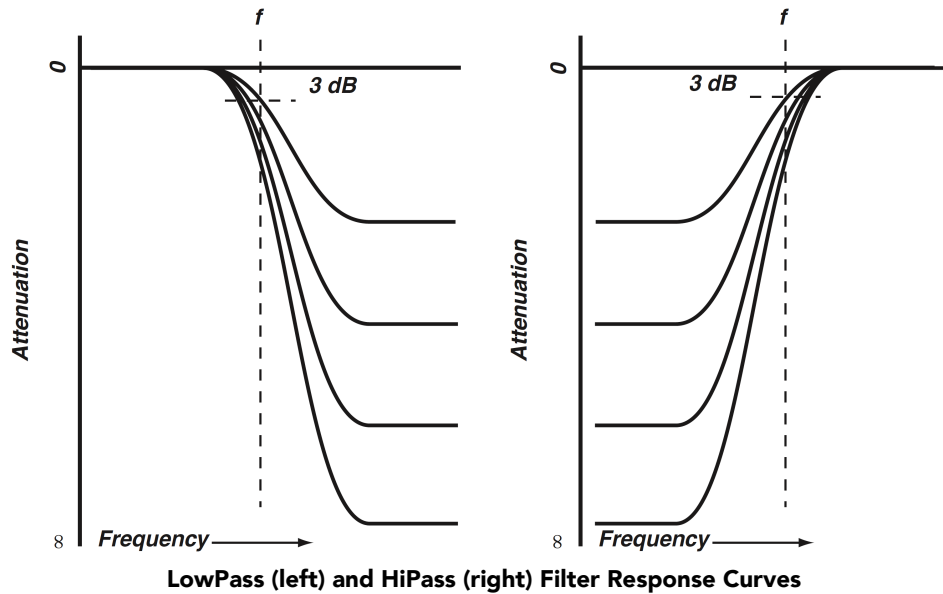
The graphic shows low and high pass filter responses for various settings of cut. The effect of the Order parameter on these filters is similar to that shown for shelving filters. Up to four parameters define the characteristic of a Lowpass or Hipass filter:

- **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
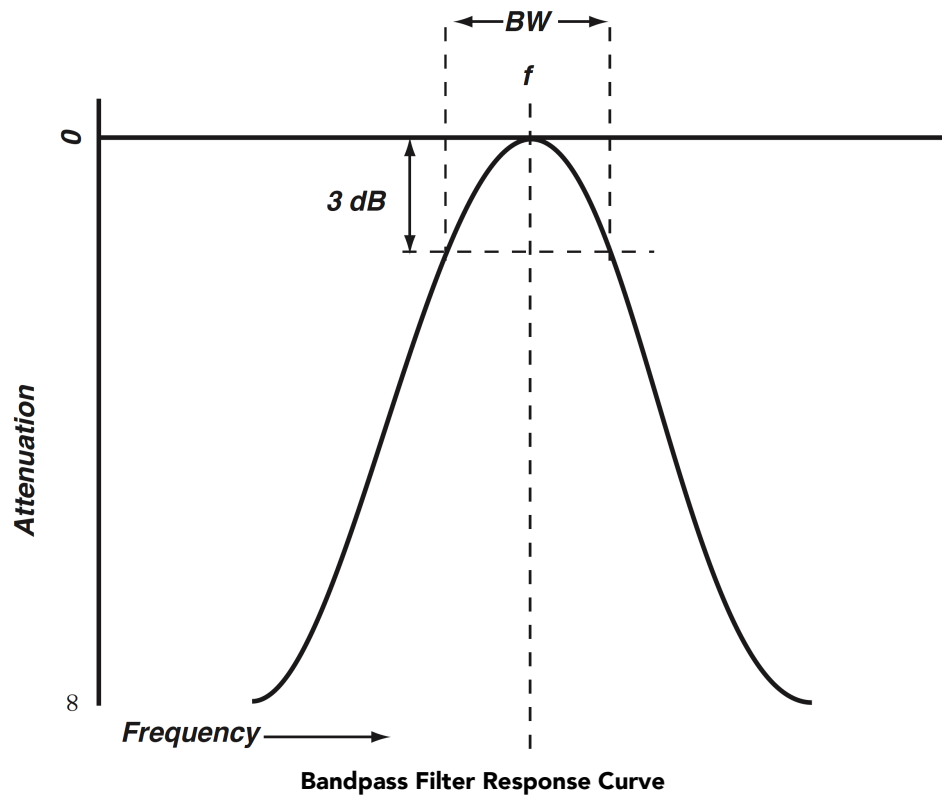
The graphic shows a typical bandpass filter response curve. The dotted lines indicate the -3 dB frequencies of the filter, which define its bandwidth. Up to five parameters define the characteristic of a Bandpass filter:

- **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.
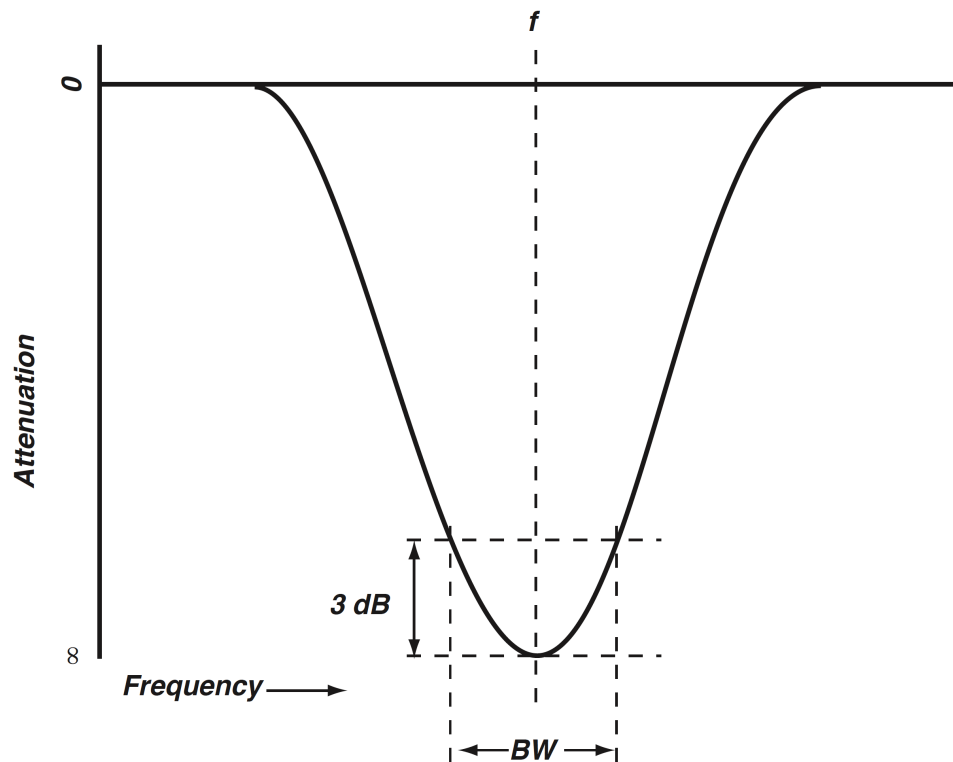
Bandstop Filter Response Curve

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.
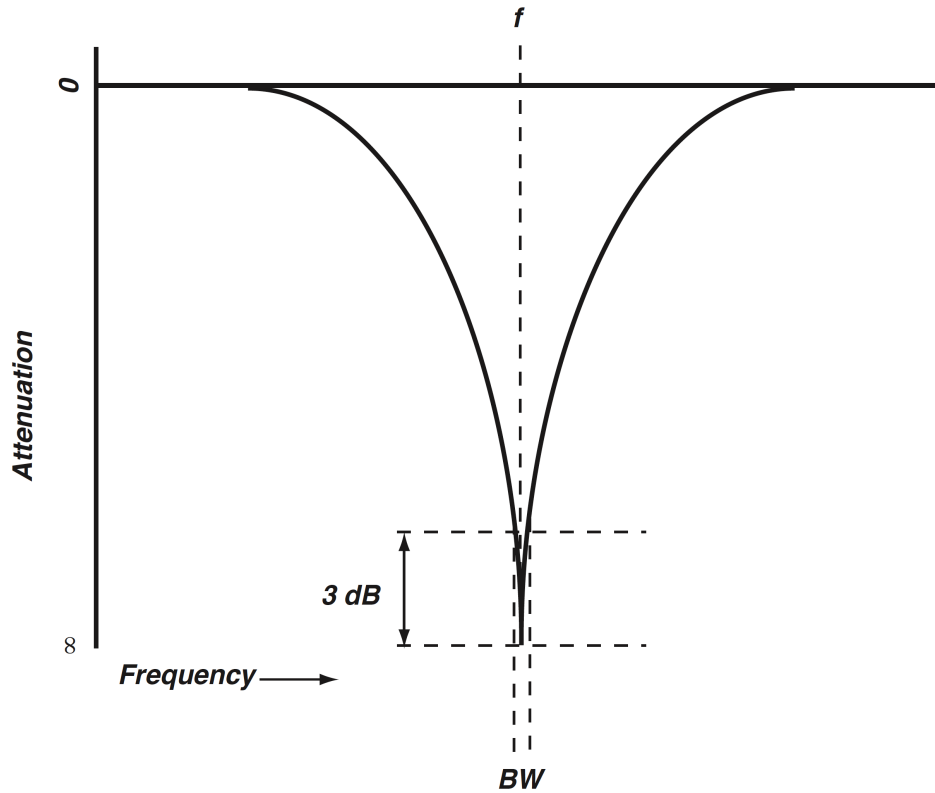
Notch Filter Response Curve

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).
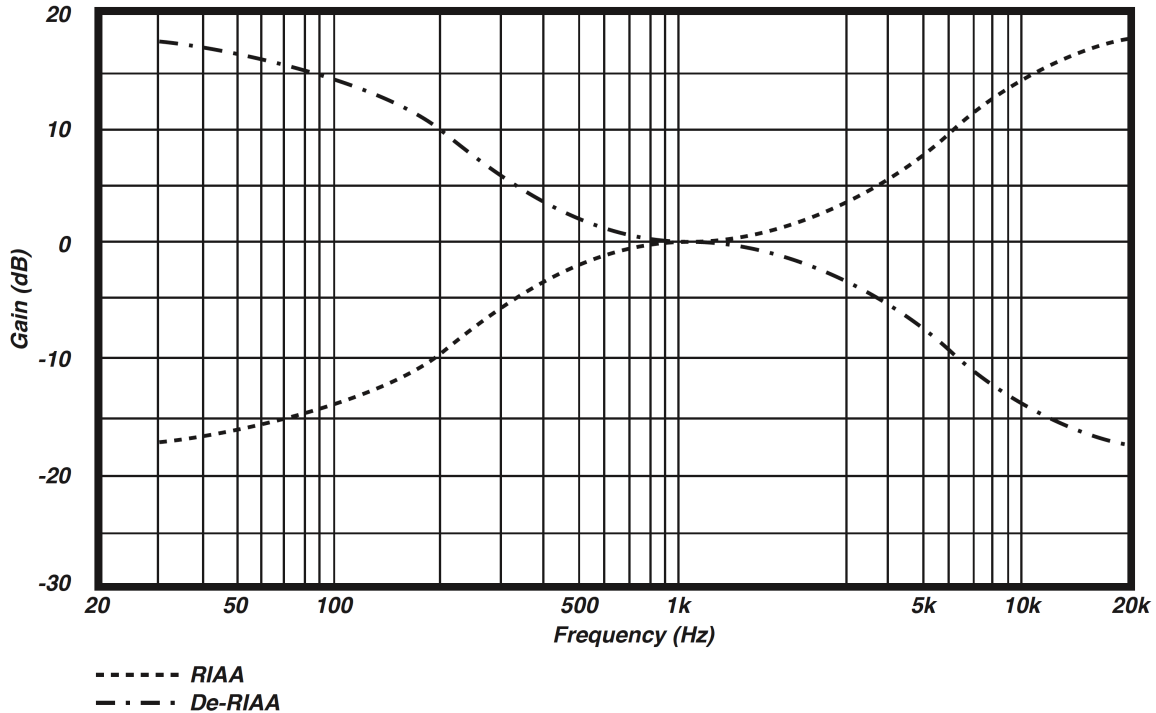
Notch Filter Response Curve

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.

Part IX. MH SuperGate

Table of Contents

29. 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.

30. 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
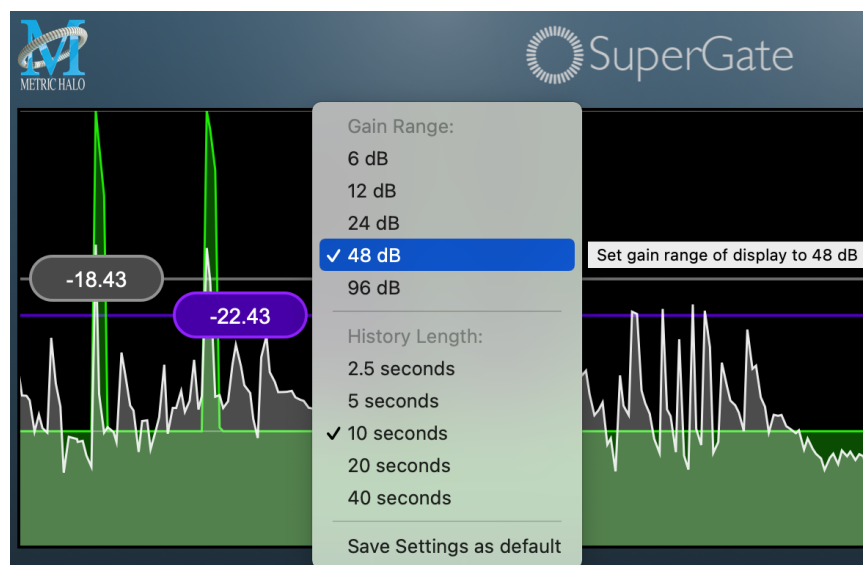
Detector/Gain History with Settings Menu

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.

31. 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 target audio through unaffected and only attenuate the unwanted signal, right?

SuperGate Process Overview

The block diagram below illustrates the overall structure of the SuperGate.
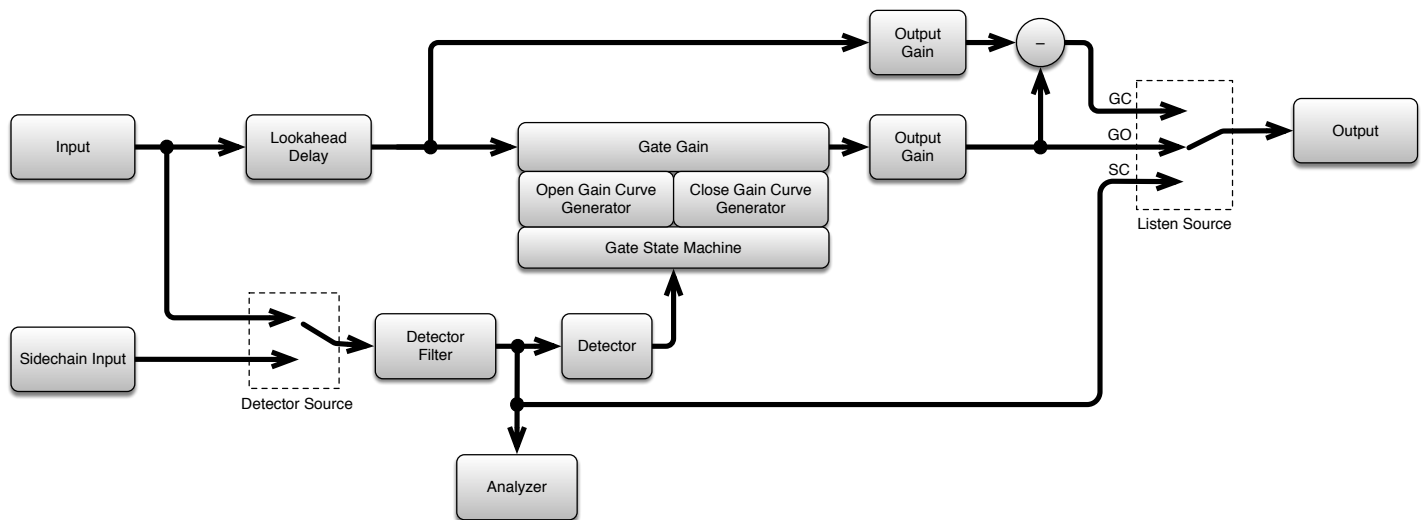
If you follow the horizontal line drawn from the Input to the Output, that is your primary audio signal path. There are actually only two controls in the SuperGate UI directly that directly affect the audio path - the *Lookahead* delay and the *Output Gain* control.

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.
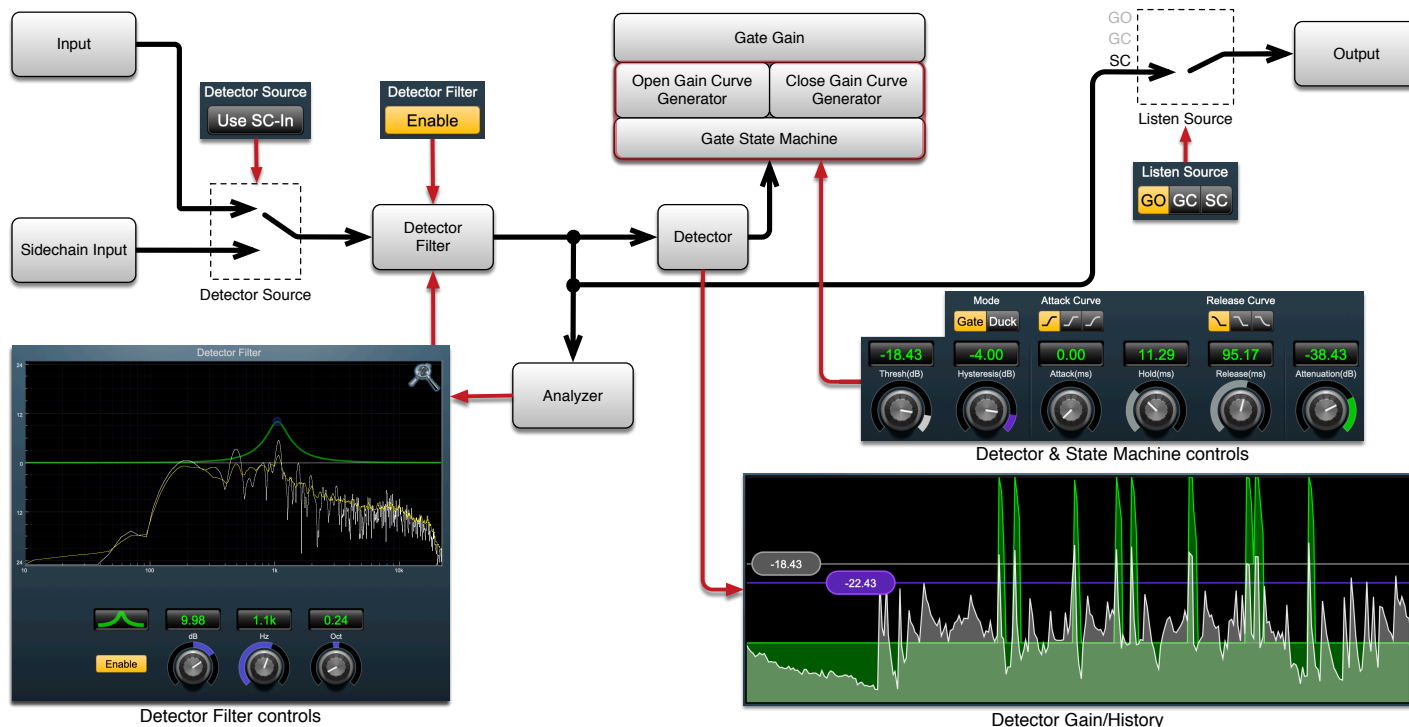
SuperGate Detector and Gate State Machine control map

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.
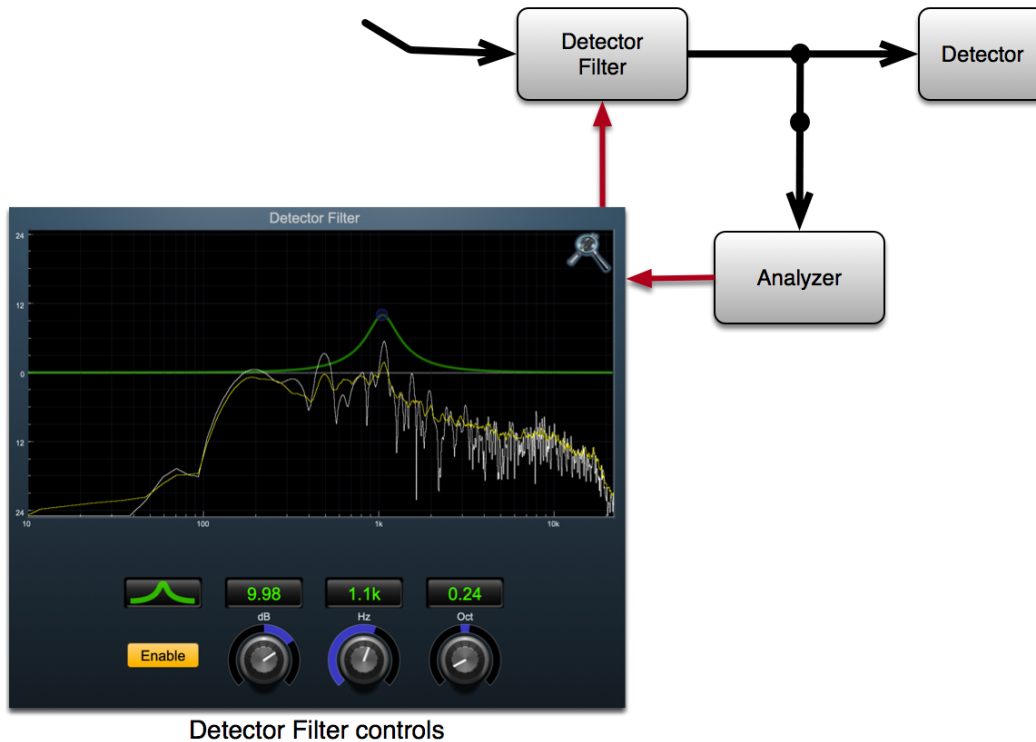
Detector Filter (flow chart focus)

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
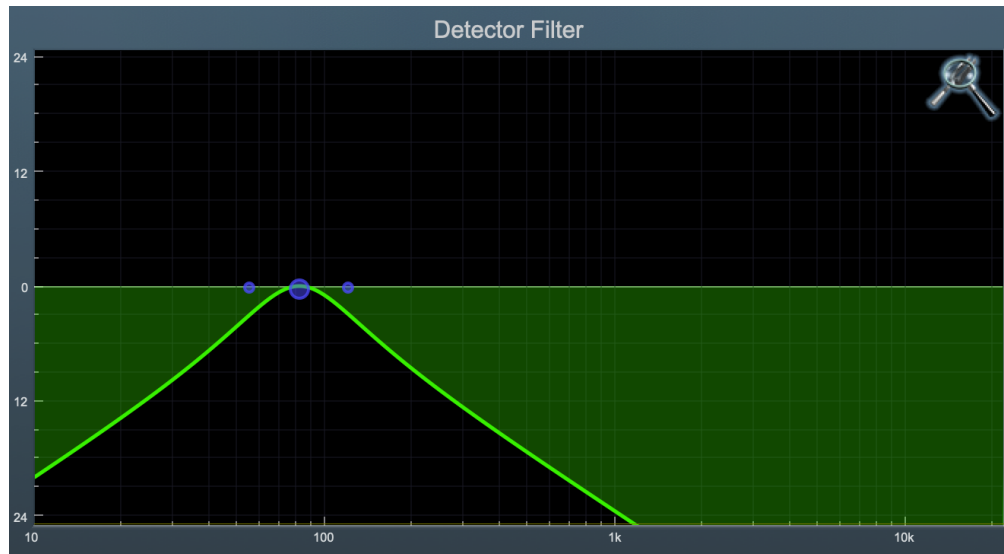
Detector Filter Transfer Function (bandpass filter shown)

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

Detector & State Machine controls
(level controls in focus)

Detector Gain/History

Gate State Machine: Level controls
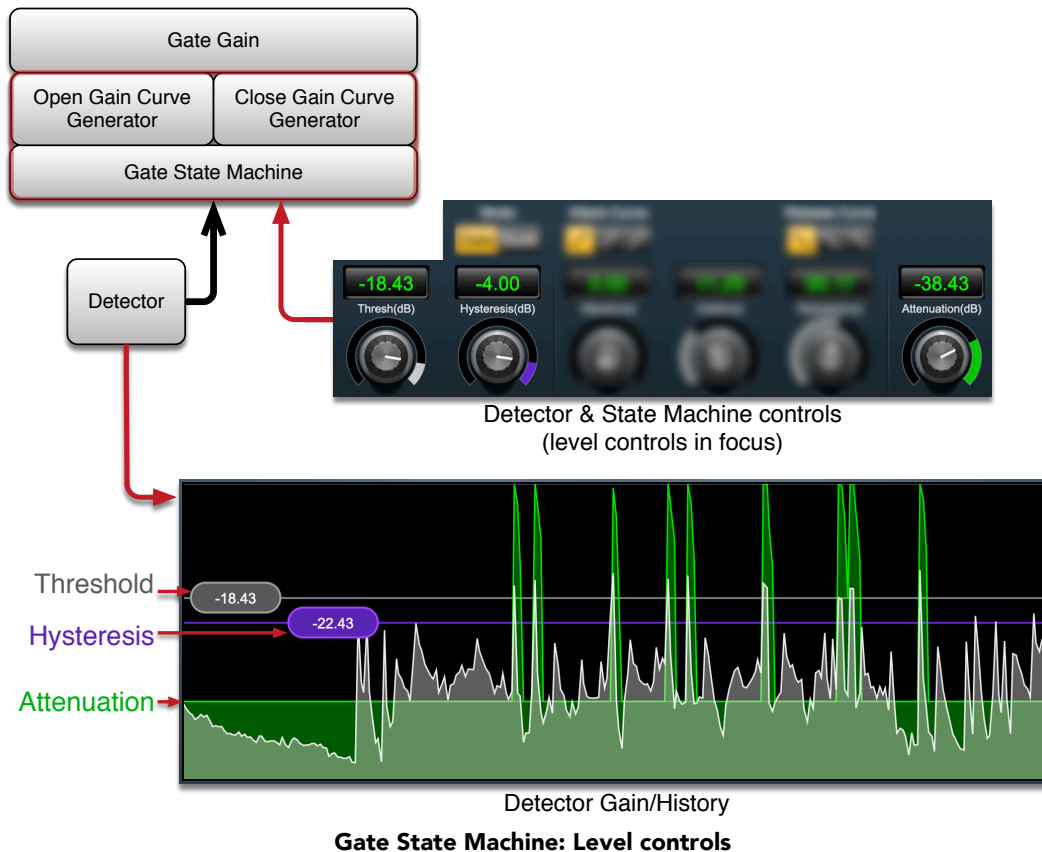
Attenuation(dB): Attenuation level

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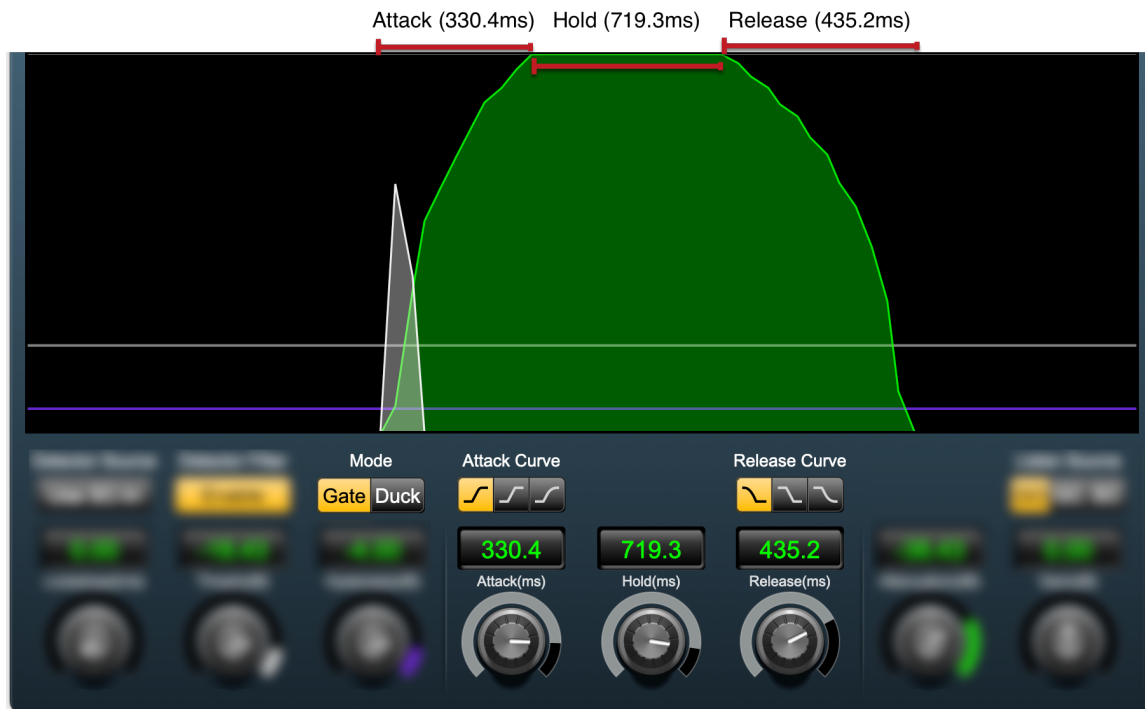
**Gate State Machine: Gain Curve shape, Attack, Hold and Release controls
(Attack, Hold & Release greatly extended for visibility)**

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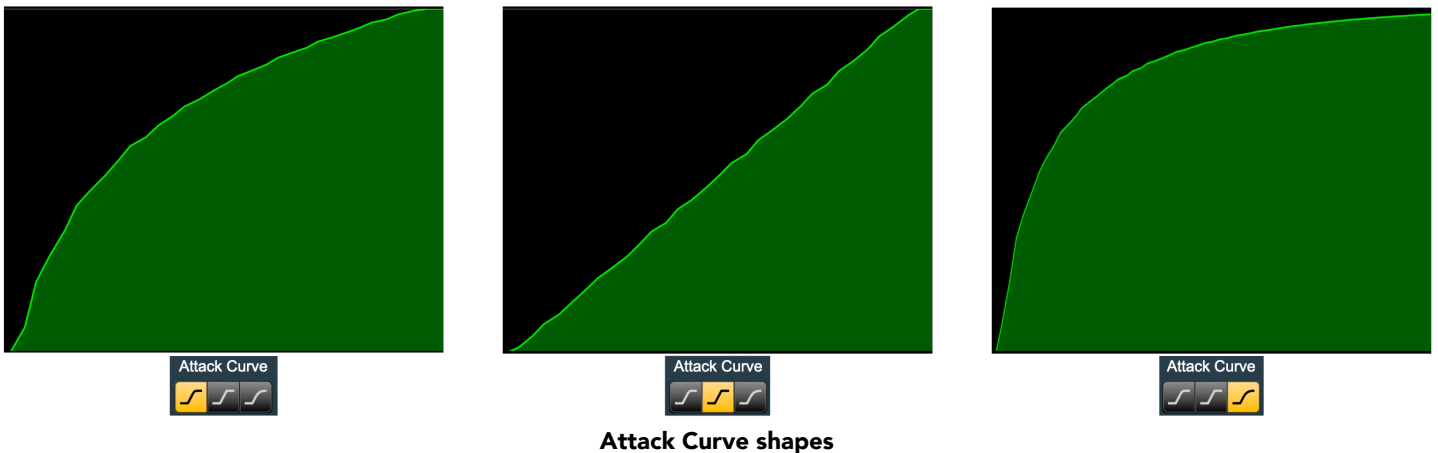
Gate 'Hold' duration may be from 0ms to 2000ms, starting from the moment the detector hits the Hysteresis de-trigger level.

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
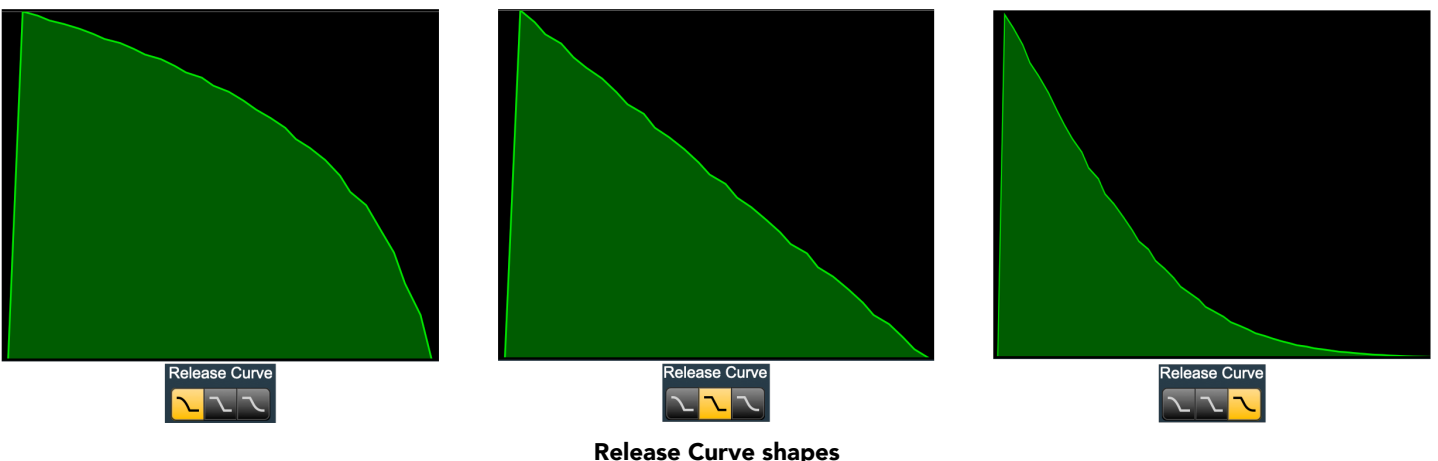
Release Curve shapes

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
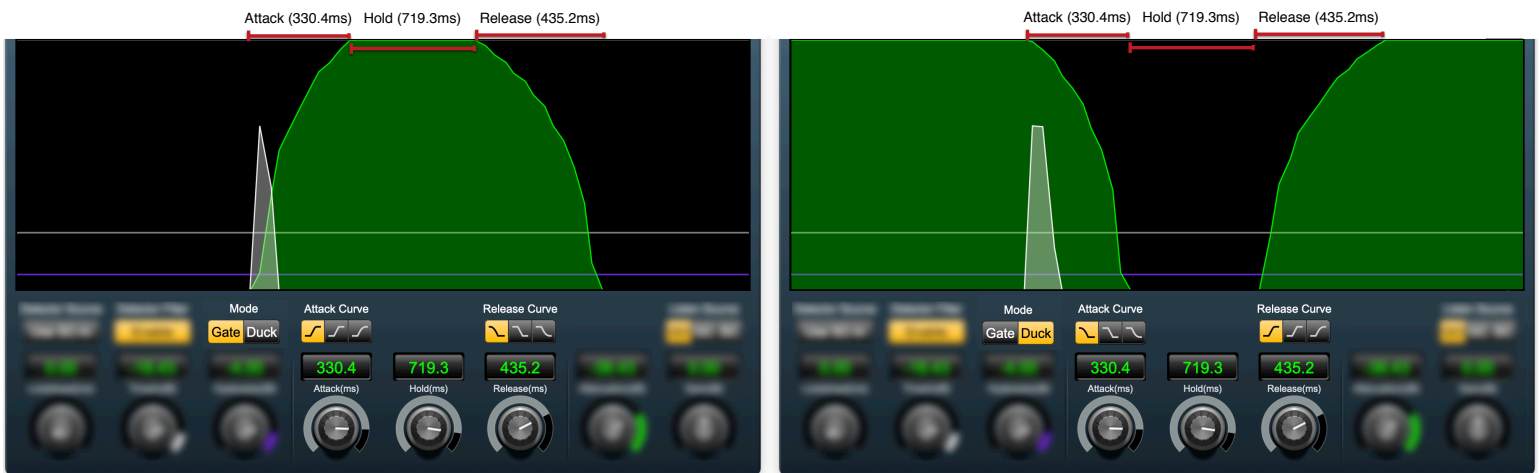
Gate vs. Duck Mode

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
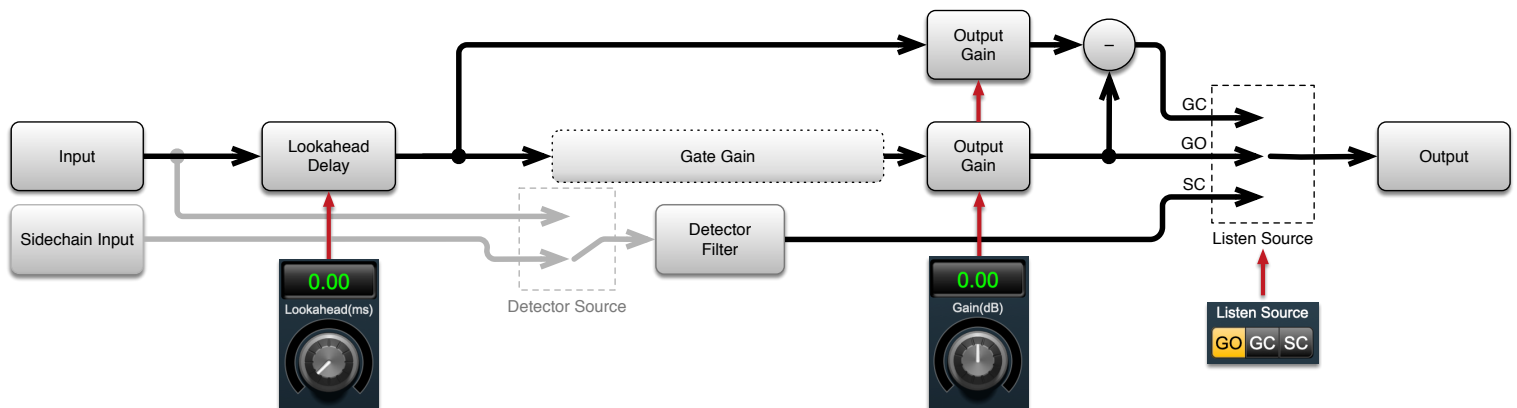
SuperGate primary audio path controls

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.

Part X. MH TransientControl

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32. 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.

33. Operation

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

Control Knob

Control Knob

Control Knobs are used to control the value of various continuous parameters of a process. Examples of these types of parameters include: Transient, Sustain, Gain, etc. You can change the value of each knob in a number of different ways. Click and drag the knob to change the value continuously. Dragging up or to the right will increase the value, while dragging down or to the left will decrease the value. 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
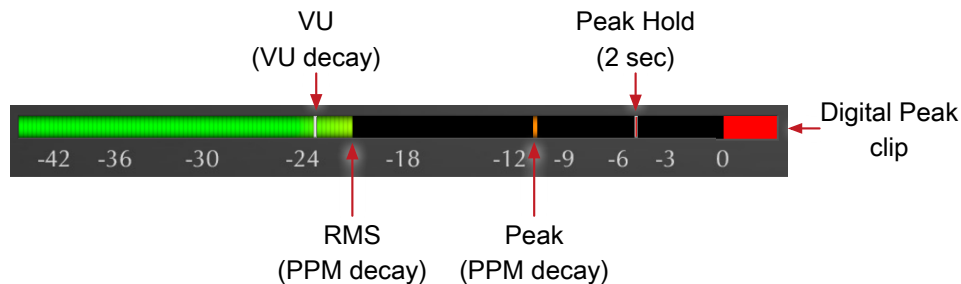
Output Meter

For the main output stage of TransientControl 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 TransientControl 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.

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:

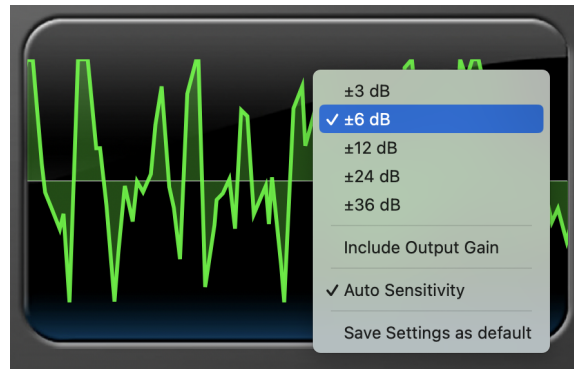**Basic: Provides the Minimum Controls and Metering**

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.

34. Processing

A Detailed Description

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

TransientControl Block Diagram
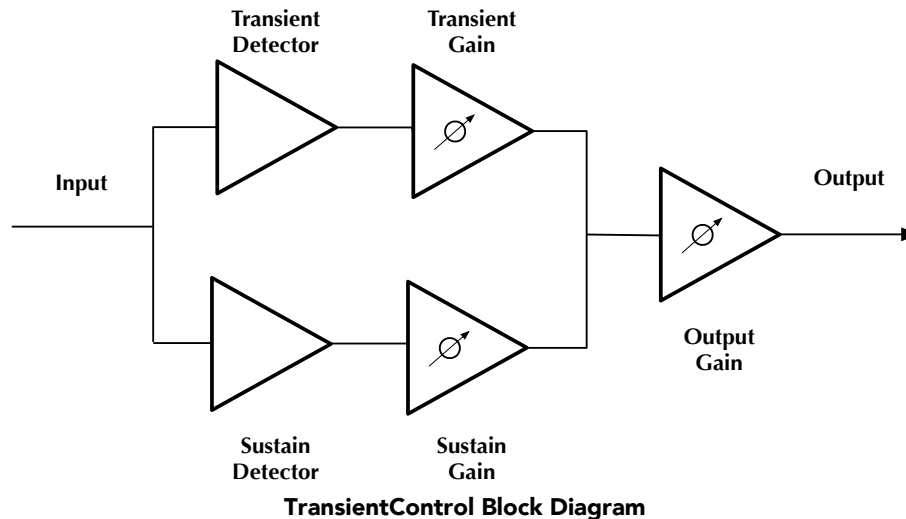
The block diagram above illustrates the overall structure of the processing system provided by Transient-Control. This diagram does not indicate the various metering blocks.

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.

Part XI. Appendices

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A. Key Commands

There are several key commands used as modifiers combined with mouse actions:

Command	Mac Key Sequence	Windows Key Sequence
Show tooltips	Hold **?** down	Hold **?** down

Table A.1. Tooltip Control

Command	Mac Key Sequence	Windows Key Sequence
Fine control	⌘ (Command)-click and drag	**Control**-click and drag
Reset to default value	⌥ (Option)-click or double-click	**Alt**-click or double-click

Table A.2. Control Knob Modifiers

Command	Mac Key Sequence	Windows Key Sequence
Confirm & dismiss numeric pop-up	**return**, **enter**	**return**, **enter**
Confirm & move to next entry	**tab**	**tab**
Confirm & move to previous entry	⇧-**tab** (Shift + tab)	⇧-**tab** (Shift + tab)
Dismiss numeric pop-up & cancel change	⌘**.** (Command + .), **ESC**	**Control.** (Control + .), **ESC**

Table A.3. Numeric Field Modifiers

Command	Mac Key Sequence	Windows Key Sequence
Reset Clip	⌥ (Option)–click the meter	**Alt**-click the meter

Table A.4. Meters

Command	Mac Key Sequence	Windows Key Sequence
Toggle band enable	⌘ (Command)–click or double-click frequency dot	**Control**-click or double-click frequency dot
Adjust bandwidth (click then drag)	⌥ (Option)–click frequency dot	**Alt**-click frequency dot
Change filter type	⌘⌥ (Command + Option)–click frequency dot	**Control**+**Alt**-click frequency dot
Access EQ TF settings	⌃ (Control) or right-click graph	Right-click graph

Table A.5. EQ Transfer Functions

Command	Mac Key Sequence	Windows Key Sequence
Popup Parameter Menu	⌃⌥⌘ (Control + Option + Command)–click	**Alt** + **Control**+**Windows**–click
Show Automation Breakpoint	⌃⌘ (Control + Command)–click	**Control** + **Windows**–click
Set Parameter to Default Value	⌥ (Option)–click or double–click	**Alt**-click or double-click
Set Parameter to Minimum Value	⌃⌥ (Control + Option)–click	**Control** + **Alt**-click

Table A.6. Pro Tools Key Commands

B. Service and Support

Metric Halo takes great pride in the reputation for customer service and support that we have built. If you have any problems, questions, or suggestions please get in touch with us at:

- https://mhsecure.com/support

- support@mhsecure.com

- (727) 725-9555

Please keep us informed about your successes and projects. We love to hear from you!