

UAD-1 POWERED PLUG-INS

USER MANUAL

VERSION 3.9

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UNIVERSAL AUDIO

analog ears | digital minds

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Notice

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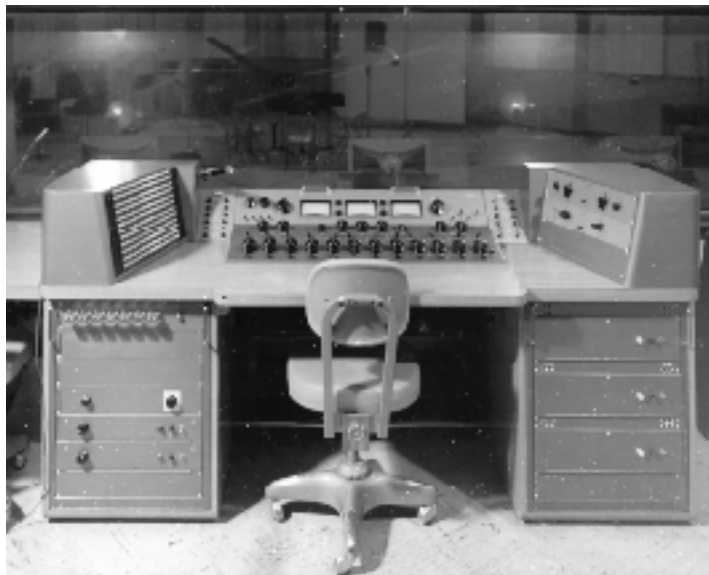
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CHAPTER 1

Introduction

Welcome

Thank you for purchasing UAD-1 Powered Plug-Ins™, the most powerful combination of digital signal processing hardware and high-quality software plugins available for host-based Windows and Macintosh digital audio workstations!

Thanks to the UAD-1™ DSP card, Powered Plug-Ins offer a new level of power and complexity not found with host-based plugins. By reducing the burden on your computer's CPU, your host application will be able to deliver more tracks, automation, and native effects.

The UAD-1 Powered Plug-Ins bundle gives the native user a fully professional suite of plugins including EQ, compression, modulation, delay, and more, and features our acclaimed RealVerb Pro™, DreamVerb™, and Plate 140 reverbs. Also included are our Vintage Compressor™ Plug-Ins, the 1176LN™ and LA-2A™. We've combined the best of our analog expertise with our digital signal processing capabilities to deliver emulations that capture every nuance of these classic compressors. Nigel™ offers the latest generation of guitar processing technology integrated into a complete multi-effects plugin solution. Nigel's Preflex™ advanced guitar amp modeling technology goes well beyond the usual pre-amp/amp/cabinet emulators. The Cambridge EQ offers precise tonal manipulation with an 'analog' sound. The Pultec EQP-1A Program Equalizer and MEQ-5 Midrange Equalizer have legendary sound that is revered by mastering engineers worldwide. The Fairchild 670 stands alone in its reputation as one of the finest compressors available at any cost and the Precision Limiter ensures your audio is production ready.

At the heart of the Powered Plug-Ins package is the revolutionary UAD-1 DSP card. Because of its high precision data path, floating point processing, high-speed memory, and hardware dithering, the UAD-1 delivers outstanding, distortion-free, high-resolution sound quality.

The UAD-1 features a ground-breaking super-DSP chip with a proprietary audio engine. Unlike other DSP cards (which juggle DSP tasks between multiple chips), the UAD-1 uses a single, unpartitioned processor, allowing for more sophisticated plug-in algorithms.

Features

- No-compromise professional audio quality
- UltraDither™ hardware algorithm provides maximum signal quality
- Artifact-free smoothing on all parameters (no zipper noise)
- All parameters can be automated
- Distortion free, high-resolution signal path due to floating point processor
- Single, unpartitioned super-computing DSP chip provides optimal performance and flexibility
- Up to 32-bit, 192kHz resolutions are supported, limited only by the host application

UAD-1 PCI DSP Card

- UltraDither™ supported in hardware for all plug-ins
- Floating point processor for maximum dynamic range
- Bus mastering DMA (direct memory access) for minimal host load and maximum sustained host-card transfer rate
- Fully PCI 2.1 Compliant, supports 66MHz operation and fast bus timing
- 7" form factor (PCI short card)
- Up to four cards supported with automatic load balancing

Precision Limiter™

- 1.5ms look-ahead brickwall Limiter with zero overshoot performance
- Totally colorless, no upsampling
- High-resolution RMS and peak metering conforms to "K System" specs
- User-adjustable or automatic release
- Unique Mode feature affects attack shape for subtle tonal variations

Precision Equalizer™

- Stereo or dual mono four-band EQ and high-pass filter designed primarily for program material mastering
- Based on industry standard filters and classic control arrangements
- Upsamples to 192kHz for utmost in performance and accuracy
- Two sets of two overlapping frequency ranges
- 18dB per octave high-pass filter aids in sub-harmonic management
- Flexible EQ configuration auditioning

Fairchild 670

- The ultimate vintage compressor/limiter
- Highly usable, incredibly musical sonic characteristics
- Precision emulation of actual circuitry and performance
- Complete absence of audible thumps often associated with other limiters
- Extremely fast attack times
- Program dependent release times available
- Lateral/Vertical (sum and difference) processing for vinyl mastering
- Sidelink Chain modification provides additional modes
- Mono or Stereo operation

1176LN Limiting Amplifier

- Modeled after 1176LN (blackface, versions D and E)
- Precision emulation of actual circuitry and performance
- Compression ratios of 4:1 8:1, 12:1, 20:1, including All Buttons mode
- Attack time: 20 microseconds to 800 microseconds
- Release time: 50 milliseconds to 1.1 second
- Mono or Stereo operation

Teletronix LA-2A Leveling Amplifier

- Precision emulation of actual circuitry and performance
- 0 to 40 dB gain limiting
- Controls: Gain, Peak reduction, Meter selector, Compress/Limit Mode
- Mono or Stereo operation

1176SE Limiting Amplifier

- “Special Edition” compressor derived from UA 1176LN
- Optimized for efficient DSP usage

Vintage Equalizers

Pultec EQP-1A Program Equalizer

- Legendary EQ revered by mastering engineers
- Highly usable, incredibly musical sonic characteristics
- Precision emulation of actual circuitry and performance
- UAD-1 DSP load remains constant even at highest sample rates
- Mono or Stereo operation

Pultec-Pro with MEQ-5

- Pultec EQP-1A and Pultec MEQ-5 together in one plugin
- Faithful modeling of vintage Pultec MEQ-5 midrange equalizer
- Precision emulation of actual circuitry and performance
- UAD-1 DSP load remains constant even at highest sample rates
- Mono or Stereo operation

Cambridge EQ

- Five bands of parametric or shelving equalization
- Additional low cut and high cut filters with seventeen filter slope types, including butterworth, bessell, and elliptic
- Complex Lattice Filters provide smooth, analog-like sound
- Graphical display of equalization curve with “bats” for adjusting the frequency, gain, and bandwidth directly on the EQ curve
- Three types of resonant shelving: a peak at the edge of the stopband, a peak at edge of the passband, or both provide smooth, Pultec-like low end
- Two channels of EQ instantly accessible within one preset for quick A/B switching between two curves
- Special “Type” modes automatically adjust Q as band gain is changed
- Proprietary algorithm avoids problems typical to digital EQs
- Filters work at high frequencies without oversampling
- Parametric section controls emulate popular high-end analog consoles
- Mono or Stereo operation

Reverbs

Plate 140

- Delivers highly prized smooth and natural plate sound
- Stunningly accurate models based on plates from The Plant Studios
- Three plates to choose from-, each with a unique sound
- Look and feel based on original 140 reverb system
- Mono or Stereo operation

DreamVerb™

- Amazing sound quality rivals high-end dedicated hardware reverbs
- Comprehensive interface for in-depth parameter editing
- 21 room shapes
- 48 room filtering materials
- Unique “Air” medium for blending with materials
- Level ramping for early reflections and late-field reverberations
- 5-band equalizer with dedicated shelving bands
- Diffusion control for late-field reverberations
- Real-time Shape and Materials blending offers dynamic sound
- Built-in preset management
- Mono or Stereo operation

RealVerb Pro

- Design custom rooms, controlling shape, size, and materials
- Adjust room sizes from 1 to 99 meters
- 15 room shapes
- 36 room materials
- Independent stereo placement of direct path, early reflections, and late-field reverberations, as well as control over the perceived source position
- Realtime morphing between presets
- Control intensity and timing of early reflections and late-field reverberation
- Diffusion control for late-field reverberations
- Blend between two different room shapes and sizes

- Blend between two different room materials and adjust relative thickness
- Mono or Stereo operation

CS-1™ Channel Strip

EX-1™ Equalizer/Compressor

- Mono or Stereo operation
- 5 band fully parametric EQ
- Switchable Hi or Low pass/shelving/peaking on bands 1, 2, 4, & 5
- Attack (0.05ms – 100ms)
- Release (30ms – 2.25 seconds)
- Either EQ or compression may be bypassed in realtime for improved processor efficiency

DM-1™ Delay Modulator

- Mono or Stereo operation
- 2400ms maximum delay per channel
- Multiple modulation waveforms with adjustable phase, including quadrature, in-phase and out of phase
- Mode selector provides all popular forms of chorus, flanging, and echo

RS-1™ Reflection Engine

- Mono or Stereo operation
- 300ms maximum pre-delay per channel
- Adjustable room size from 1–99 meters
- Wide range of delay presets including single echo, pattern echo and spatial room simulations
- Room shapes/simulations developed in conjunction with NASA scientists
- Special effects include forward and reverse gated reverb

Nigel

- Preflex advanced guitar processor with user updatable amp models
- Continuously variable morphing between any two amp types
- Gate/Compressor for noise and dynamics control
- Phasor capable of modern and classic sounds such as those produced by the Mutron Bi-Phase, Small Stone and MXR series of phasors
- Mod Filter capable of wah, auto-wah, and envelope follower effects, modeled after the Mutron III and other popular filters
- Tremolo with Classic, Shimmer™, VariTrem™, and Fade modes
- Fade-in for gorgeous swells and reverse tape effects
- Modulated Delay capable of chorus, flange and vibrato; can be synchronized to the Trem/Fade module for unprecedented new sounds
- Echo Delay with 1200ms of stereo delay time
- No-compromise professional audio quality
- All parameters are MIDI controllable with full automation
- Unlimited presets can be saved and loaded as desired
- Artifact-free smoothing on all parameters (no zipper noise)

Preflex

- Exciting guitar processing technology offers dynamic sonic possibilities
- Pre and post Lo, Mid, and High equalization controls
- Color and Bent controls modify frequency and gain characteristics in interesting and musically useful ways
- Amp type menu provides a starting point for the “classic” guitar tones
- Selectable speaker cabinet emulation for complete tonal control
- Real-time component-level morphing between any two amp types
- Threshold control for Gate
- Threshold, Ratio, Attack, and Release controls for Compressor
- Separate on/off controls for each Preflex submodule for maximum flexibility and UAD-1 DSP efficiency

Roland CE-1

- Reproduction of famous Chorus Ensemble in partnership with Roland
- Faithful to original Boss CE-1 hardware in every regard
- Chorus and vibrato modes

System Requirements

UAD-1 Powered Plug-Ins require the following hardware and software:

All Platforms:

- 60 MB of available disk space
- Available PCI slot for each UAD-1 card
- 1024 x 768 or higher resolution monitor
- CD drive or internet connection for software installation
- An AGP graphics video adapter (PCI graphics is not recommended)
- Internet connection required for registration and obtaining optional plugins
- Additional platform-specific requirements are listed below

Windows platform:

- Microsoft Windows 98SE/ME, 2000/XP, or Server 2003
- 256 MB of RAM (512 MB is strongly recommended)
- VST compatible host application software, such as Steinberg Cubase or Nuendo
- DirectX is supported under Cakewalk Sonar and Project5, and Sonic Foundry Sound Forge, Acid, and Vegas

Mac OS X platform:

- Mac OS X 10.2.x or higher
- 256 MB of RAM (512 MB is strongly recommended)
- VST version requires VST-compatible host software, such as Steinberg Cubase or Nuendo
- Audio Units version requires AU-compatible host software, such as Apple Logic Pro or MOTU Digital Performer
- Processor upgrade cards are not officially supported

Mac OS 9 platform:

- Mac OS 9.x
- 128 MB of RAM (256 MB is strongly recommended)
- VST version requires VST-compatible host software, such as Steinberg Cubase or Nuendo
- MAS version requires MOTU MAS version 2.3 or higher, and Digital Performer or AudioDesk host software
- G3 or better recommended; processor upgrade cards are not officially supported

Manual Conventions

Cross-Platform Solution

UAD-1 Powered Plug-Ins is a cross-platform solution for both Windows and Mac OS-based computers. The UAD-1 PCI card can be installed into either platform; it is the exact same hardware for both platforms. Operation of the plugins are practically identical regardless of the host system platform and application. However, certain platform-specific instructions will differ according to the host system you are using.

Headings

Instructions in this guide that are platform-specific will be indicated with a heading in red letters. Instructions that are identical regardless of platform are not differentiated.

Windows

Instructions specific to the Windows platform will use this red Windows heading.

Mac OS

Instructions specific to the Macintosh platform will use this red Mac OS heading.

Screen Shots

Screenshots in this manual may be taken from the Windows and/or Mac OS version of the software, and are used interchangeably when the content and functionality of the screenshot is the same on both platforms. Slight variations in the appearance of a screenshot between operating systems are inevitable.

When the content of and function of the software represented in a screenshot is identical on both platforms, no differentiation is made in the screenshot title. If there is a significant difference between platforms, screenshots from both platforms are included.



CHAPTER 2

Installation

Refer to the QuickStart Guide

Software installation and removal for each of the various platforms and operating systems has its own particular procedures. Please refer to the QuickStart Guide documentation that is included in the software bundle and the support pages on our website for complete instructions on how to install and remove the software on each system.

The QuickStart Guide can be found on the product CD-ROM and it is placed into the Powered Plug-Ins Tools folder during software installation. The QuickStart Guide, and other important technical information, is online at:

- <http://www.uaudio.com/support/software/UAD-1/techbulletins.html>

Install Software First

For best results, the Powered Plug-Ins software must be installed before installing the UAD-1 PCI card. See the QuickStart Guide for instructions.

Instructions for hardware installation follows in the next section.



Installing the UAD-1 Hardware

After installing the UAD-1 Powered Plug-Ins software, install the UAD-1 PCI DSP card(s). Hardware installation is the same for all platforms.

To install the UAD-1 DSP card(s):

1. Turn off your computer.
2. Open the computer case. If necessary, refer to the computer manufacturer's documentation for instructions.
3. Remove the rear slot cover and screw of the lowest-numbered available PCI expansion slot.
4. Before handling the UAD-1 card, discharge any static electricity by touching the outer casing of the power supply.
5. Remove the UAD-1 card from its protective anti-static bag. Do not touch the gold PCI edge connector contacts.
6. Hold the card gently by the top edges, and line up its PCI connector with the PCI slot inside the computer.

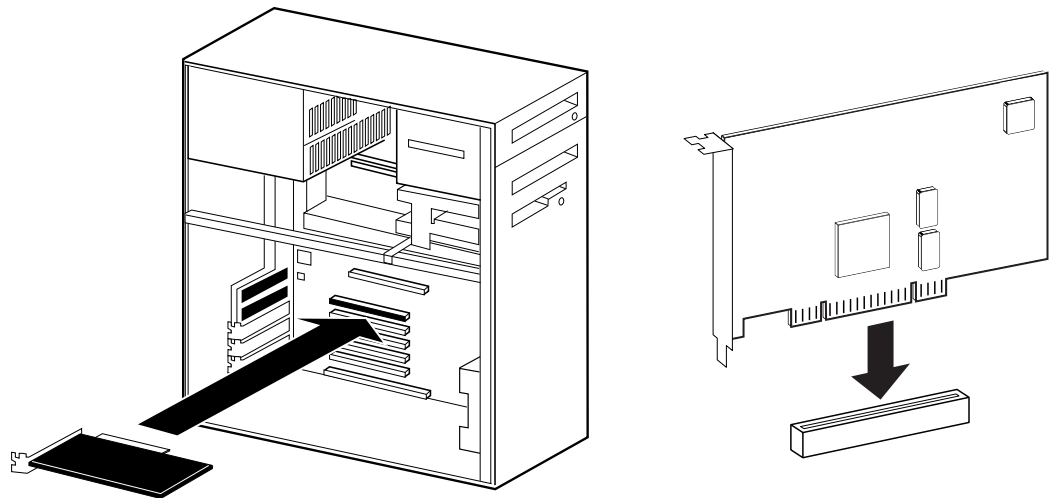


Figure 1. UAD-1 card installation

7. When the connector and slot are aligned, press the card into the slot using firm, even pressure. The card should “pop” into place. The top of the PCI slot on the motherboard should be flush and parallel with the edge of the UAD-1 card.
8. Secure the card with the previously removed screw.
9. Replace the computer case

Hardware installation is now complete.

CHAPTER 3

Using UAD-1 Powered Plug-Ins

Overview

Once the UAD-1 card and Powered Plug-Ins have been properly installed, the UAD-1 Powered Plug-Ins are accessed and used just like any host-based plugin. All UAD-1 Powered Plug-Ins can run concurrently with each other and with host-based plugins simultaneously, in any combination.

All UAD-1 Powered Plug-Ins support up to 32-bit, 192KHz operation. Resolution is limited only by the resolution of the host application. Please note that Powered Plug-Ins running at 96KHz use twice as much UAD-1 DSP resources than those used at 48KHz, and so forth.

Adjusting Parameters

The parameter settings for each of the UAD-1 Powered Plug-Ins can be adjusted to achieve a desired effect. Parameter values are easily modified by dragging sliders, rotating knobs, clicking switches and buttons, or by selecting values in a pop-up menu. The function of all parameters are detailed in later chapters.

The parameter adjustment style can be Circular, Relative Circular, or Linear. For more information, see [“User Interface Settings” on page 38](#).

Note: *To increase resolution when adjusting rotary controls in circular and relative circular modes, increase the radius of the mouse relative to the knob while dragging (i.e. move the mouse farther away from the knob while dragging).*

Text Entry

Parameter values can be modified directly with text entry. To enter a parameter value using text entry, single-click the parameter value text. The text value will highlight indicating it is ready to receive a new value. Type in a new value, then press Return, Enter, or Tab, or click outside of the text box. Press Esc if you want to revert to the prior setting without entering the new value.

Values entered via text entry are rounded to the closest significant digit. If an entered value is out of range, it will be ignored.

To enter time values, the units must be specified. m = milliseconds, and s = seconds. Examples: 400 milliseconds = .400s or 400m; 1.5 seconds = 1.5s or 1500m.

Scroll Wheel

If your mouse has a scroll wheel, it can be used to adjust knob and slider controls if the host application supports this functionality (not many do). Place the mouse cursor over any knob or slider control to increment or decrement the parameter value with the scroll wheel. This feature cannot be supported under Mac OS due to a limitation of the operating system software.

Keyboard Control (Mac OS)

If you control-click a control it selects that control for keyboard control. This is useful for when you're in circular mode, and you want to fine-adjust a control. Normally, clicking on a control in this mode makes the value jump to where you clicked. Control-clicking will select the control so that you can use the keyboard to adjust it, without making its value jump first.

Shortcuts

Table 1 lists the keyboard shortcuts that are available for modifying parameter values. When using keyboard shortcuts, the last edited control will be modified (or, on Mac OS, you can use control-click to select a different control as the target for keyboard shortcuts without changing the control's value).

Note: *Not all host applications support sending keystrokes to plugins.*

Table 1. Keyboard shortcuts

Keyboard Action:	Result:
Control + Click Parameter (Mac OS only)	Select parameter for keyboard control (without changing its value)
Shift + Drag	Fine Control
UpArrow RightArrow Shift + PageUp	Increment Fine
DownArrow LeftArrow Shift + PageDown	Decrement Fine
Shift + UpArrow Shift + RightArrow PageUp	Increment coarse
Shift + DownArrow Shift + LeftArrow PageDown	Decrement coarse
Home	Maximum
End	Minimum
Control + Click parameter (Windows) Modifier* + Click parameter (Mac OS) (*Modifier key set in Configuration Window)	Toggle initial editor setting (the value when the editor window was last opened)
Control + Shift + Click parameter (Windows) Modifier* + Shift + Click parameter (Mac OS) (*Modifier key set in Configuration Window)	Revert to initial editor setting (the value when the editor window was last opened)

Automation

Every UAD-1 Powered Plug-In parameter can be automated if this feature is supported by the host application. Each host application has its own particular methods for automation. Consult the host application documentation for specific instructions on using automation with the application.

Powered Plug-Ins reduce their UAD-1 DSP load when bypassed or disabled, but not their memory load. This feature allows for automatable load balancing of DSP power, and keeps the track delay constant to avoid on/off clicks.

Note: *If there is not enough DSP available when automating, the plugin may not turn on.*



Launching a UAD-1 Powered Plug-In

Each host application has its own particular methods for instantiating (launching) a plugin. Consult the host application documentation for specific instructions on loading and using plugins with the application.

Steinberg Cubase/Nuendo

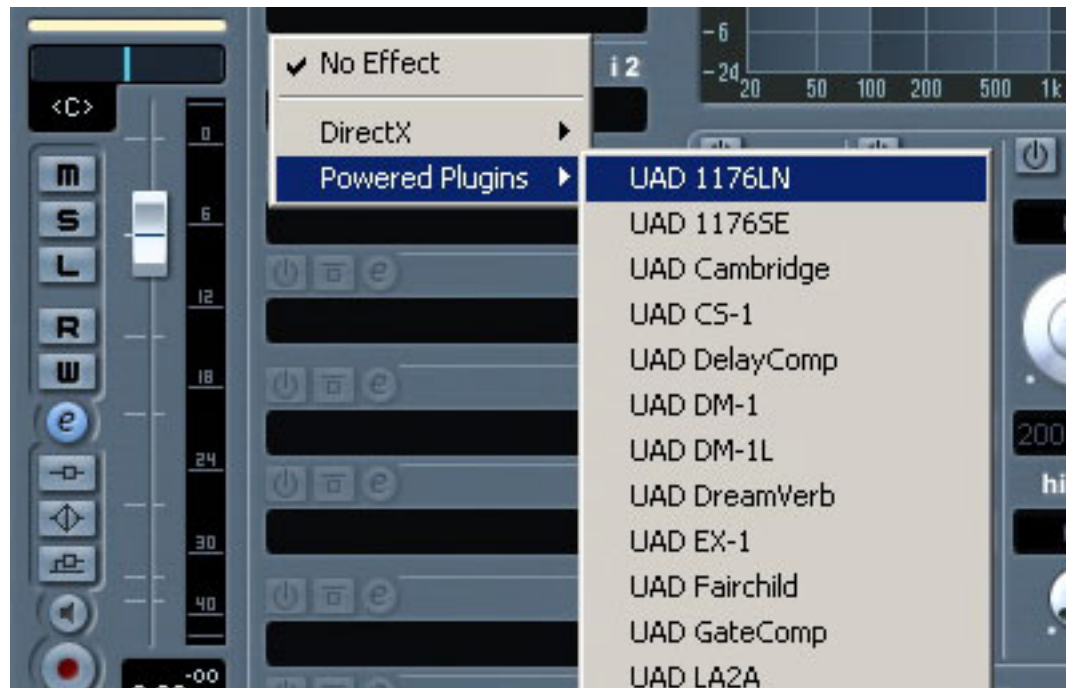


Figure 2. Launching a UAD-1 Powered Plug-In in Steinberg Cubase and Nuendo

Emagic Logic Audio



Figure 3. Launching a UAD-1 Powered Plug-In in Emagic Logic Audio

UAD-1 DSP Performance Meter Application

Overview

The UAD-1 Performance Meter is an application that displays the current CPU and memory status of the UAD-1 DSP hardware card in realtime. Its small floating window enables you to monitor the resource load of the UAD-1, while simultaneously using your host application.

It also contains system information and configuration windows that enable you to confirm the UAD-1 is functioning properly, check the version of the software drivers, and adjust the UAD-1 buffers.

If multiple UAD-1 cards are installed, the displayed CPU and memory usage is the total for all installed cards. Usage statistics of individual cards can be viewed using the System Information window (see [page 33](#)).



Figure 4. The UAD-1 Performance Meter application window (Windows)

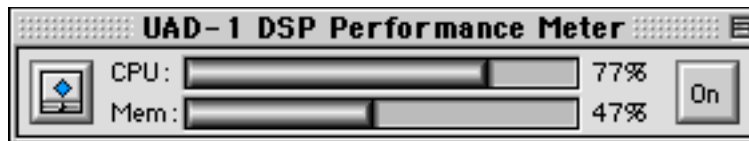


Figure 5. The UAD-1 Performance Meter application window (Mac OS 9)

Launching the Meter Windows

To launch the UAD-1 Performance Meter application in Windows:

1. Double-click the UAD-1 Meter shortcut that was placed on the Desktop during installation. OR,
2. Access the application from the Start Menu at Programs/UAD-1 Powered Plug-Ins/UAD-1 Meter. OR,
3. Double-click the executable file on the hard drive located at C:\Program Files\Universal Audio\Powered Plug-Ins\UADPerfMon.exe.

Launching the Meter Mac OS

To launch the UAD-1 Performance Meter application in Mac OS:

1. Double-click the UAD-1 Meter alias that was placed on the Desktop (OS 9) or in the Dock (OS X) during installation. OR,
2. Double-click the UAD-1 Meter application file that was copied to your hard drive inside the Powered Plug-Ins Tools folder during installation.

Accessing Meter Functions

The UAD-1 DSP Performance Meter view mode, System Information Window, and Configuration Window functions are accessed from the System menu (Windows) or the File menu (Mac OS). After clicking the System or File menu with the mouse, the available functions are listed in the menu.

Windows

Open the system menu by clicking the small icon at the upper left of the UAD-1 DSP Performance Meter window, or the alternate system menu on the right side of the Meter window, just above the Disable menu.



Figure 6. System menu for the UAD-1 Performance Meter (Windows)

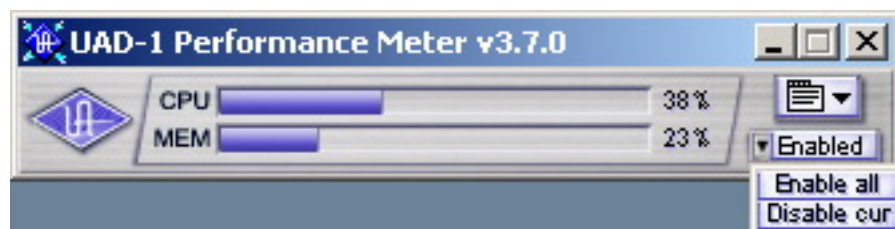


Figure 7. Alternate system menu for the Meter (Windows)

Mac OS

The File menu is available when the UAD-1 DSP Performance Meter is in the foreground. When the UAD-1 Performance Meter application is in the foreground, toggle the 'Always on top' item in the File Menu (or type the Command + T shortcut). You can easily bring the Meter to the foreground by clicking the Bring to Front button (see [Figure 9 on page 32](#)).



File	
Load Settings File...	⌘L
✓ Always On Top	⌘T
System Info...	⌘I
Configuration...	⌘;
Authorizations...	⌘A
Quit	⌘Q

Figure 8. File Menu for UAD-1 Performance Meter (Mac OS 9)

Using the Meter

The UAD-1 DSP Performance Meter can be launched or quit at any time. It does not need to be open or active to use UAD-1 Powered Plug-Ins. It is completely independent of any other applications and does not require a host application. Move the Performance Meter to a convenient location on your screen by dragging its window title bar.

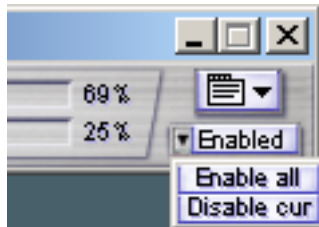
The CPU gauge indicates the percentage of UAD-1 DSP that is currently in use. It indicates the total available UAD-1 DSP statistics, regardless of the number of UAD-1 cards that are installed. When UAD plugins are disabled, DSP requirements are decreased.

The Memory gauge indicates the percentage of UAD-1 memory that is currently in use. It indicates the total available UAD-1 memory available, regardless of the number of UAD-1 cards that are installed. When UAD plugins are disabled, memory requirements are *not* decreased. In this case, memory remains loaded so that reverb tails and delay lines are not cut off when the plugin is disabled.

Always On Top

The Performance Meter window can be set to a normal or 'Always On Top' view mode. In normal mode, the window can be covered by windows of the foreground application. When in 'Always on top' mode, the Performance Meter window always floats on top of other windows, even when other applications are in the foreground, so you can always see the meter and access the Enable Menu (Windows) and On/Off button (Mac OS). This setting is saved when the meter is quit.

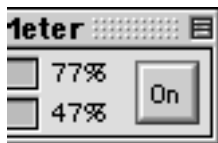
Enable Menu (Windows)



The Enable Menu allows you to disable all UAD-1 Powered Plug-Ins that are currently running. This enables you to add new plugins for offline processing if the UAD-1 is low on DSP, or easily compare the sound of the processed and unprocessed audio.

When the menu displays "Enabled" all UAD-1 plugins are active. Select "Disable current" from the menu to disable the active plugins. New UAD-1 plugins can then be added. Select "Enable all" to re-activate all UAD-1 plugins.

On/Off Button (Mac OS)



The On/Off button allows you to disable all UAD-1 Powered Plug-Ins that are currently running. This enables you to add new plugins for offline processing if the UAD-1 is low on DSP, or easily compare the sound of the processed and unprocessed audio.

When the button displays "On" all UAD-1 plugins are active. Click the button to disable the active plugins. New UAD-1 plugins can then be added. Click the button again to reactivate the plugins.

Bring to Front Button (Mac OS 9)

To access the File menu of the Performance Meter application, the application must be in the foreground. The "Bring To Front" shortcut button (see [Figure 9 on page 32](#)) makes this easy by immediately bringing the Performance Meter application to the foreground when clicked, saving a mouse trip to the Mac OS 9 application menu.

Note: The “Bring To Front” button is only present on Mac OS 9 systems when the Performance Meter is in “Always On Top” mode. When the Performance Meter is not in this mode, the application comes to the foreground whenever you click anywhere in the Performance Meter window.



Figure 9. Performance Meter Bring to Front button (Mac OS 9)



UAD-1 System Information Window

The UAD-1 System Information window ([Figure 10 on page 34](#) and [Figure 11 on page 35](#)) displays the version of the UAD-1 software drivers in use by the UAD-1 hardware and also allows you to confirm that the card is working properly. When the window displays UAD-1 Status: OK and UAD-1 DSP: OK, the card is operating properly. The number of UAD-1 plugins loaded on the card(s) is also displayed here.

If more than one UAD-1 card is installed, information for each of the cards is displayed. The card that has the lowest DSP usage will receive the next plugin load.

Important: *The version of the UAD-1 Drivers and the Powered Plug-Ins files must match. If they don't, a "driver mismatch" error will occur when attempting to process audio. If this occurs, you must reinstall the latest UAD-1 Powered Plug-Ins software. Refer to the QuickStart Guide for instructions*

Card Enabled

Individual UAD-1 cards can be disabled using the Card Enabled function. This can be useful, for example, if creating a session on a system with multiple cards that will be transferred to a system with fewer cards or to streamline the performance of the host system when multiple cards are not needed.

For additional information regarding the use of multiple cards, see ["Multiple Cards" on page 54](#).

Note: *For optimum results, quit any host applications using UAD-1 plugins before disabling/enabling cards.*

In Mac OS, the current UAD-1 plugin latency is displayed in the System Information window. In Windows, this information is displayed in the Configuration window ([page 42](#)). The latency is usually twice the hardware buffer size.

Windows

Click a card column to view more information about that card in the status area. If a card has errors, the error information for that card is automatically displayed in the status area without having to click its card column.

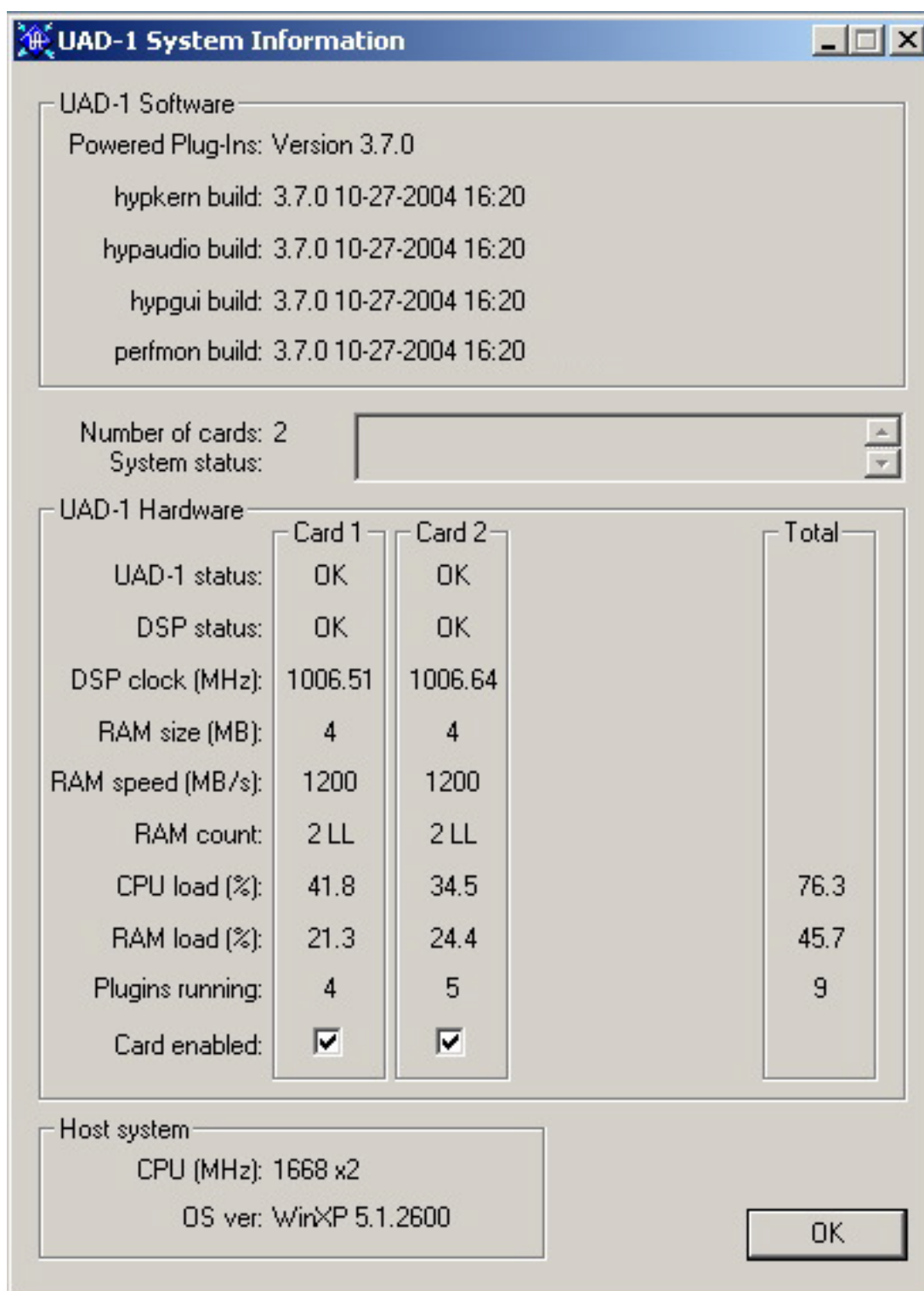


Figure 10. The UAD-1 System Information window (windows)

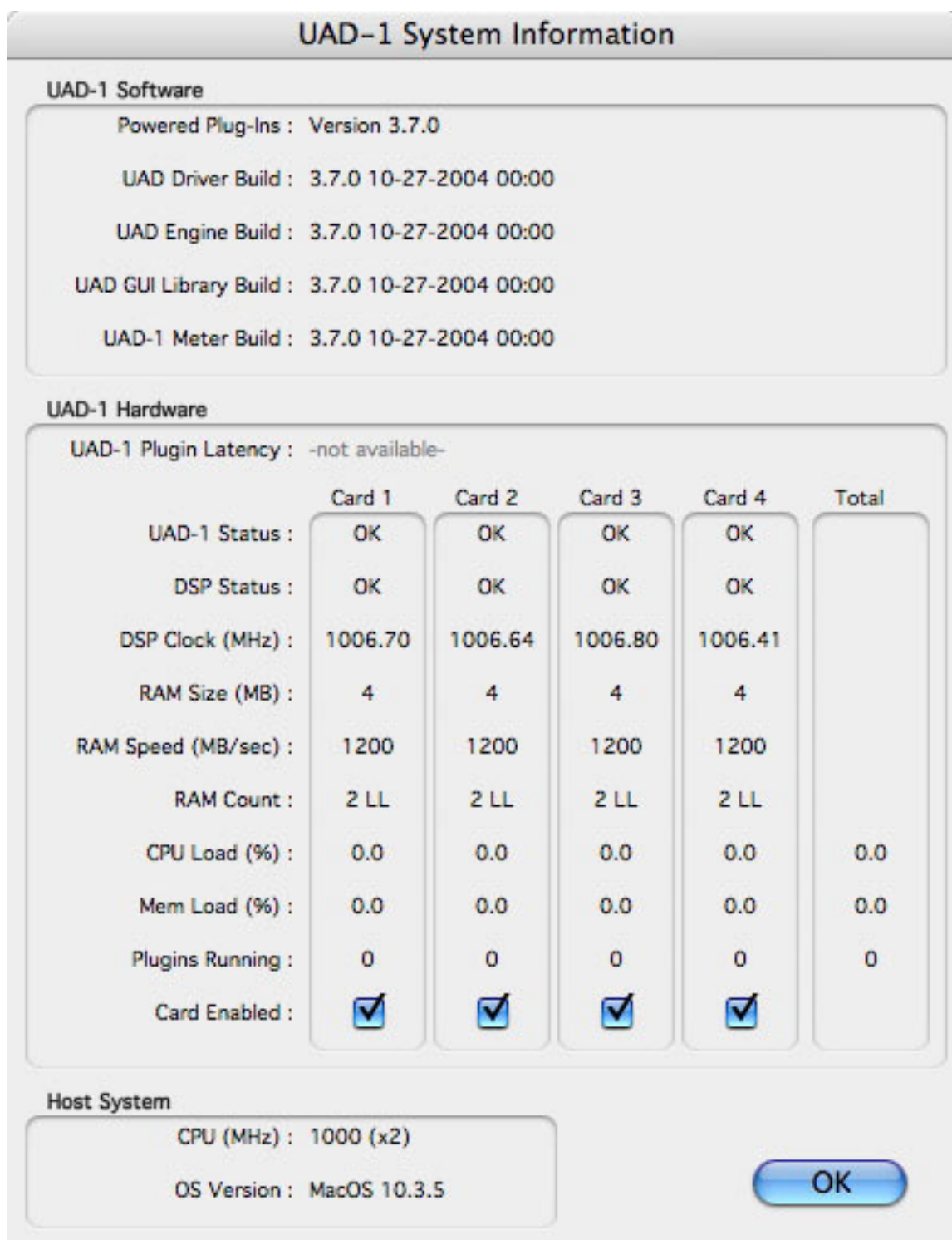


Figure 11. The UAD-1 System Information window (Mac OS)

UAD-1 Configuration Window

The UAD-1 Configuration Window (Figure 13 on page 42 and Figure 12 on page 41) displays additional information about the UAD-1 card and is also used to modify some UAD-1 settings.

Latency Calculator

The number of active UAD-1 Powered Plug-Ins, the sample rate, and the current buffer size are displayed. The window uses this information to calculate and display the resulting latency in milliseconds. In Mac OS, the latency calculator is displayed in the System Information window.

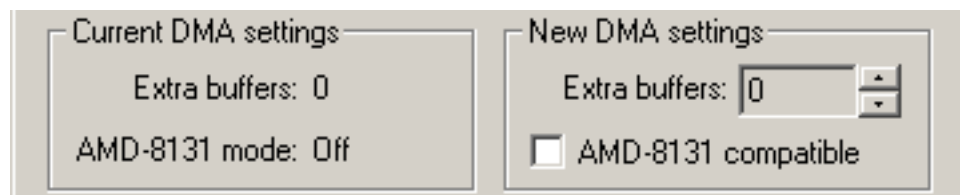
DMA Settings (Windows)

Configuring Extra Buffers

Extra Buffers are required when “Buffer Size” displayed in the UAD-1 Configuration Window is smaller than the actual ASIO buffer size selected for the active ASIO hardware device.

Note: Extra Buffers are not required for Cubase/Nuendo version 2 or higher, Logic Audio, or Mac OS.

This situation may occur when users of Steinberg Cubase and Nuendo select ASIO buffer sizes of 2048 samples or greater, or when UAD-1 Powered plug-Ins are used with a DirectX wrapper. If the situation is not corrected, the use of UAD-1 Powered Plug-Ins will introduce excess host CPU load.

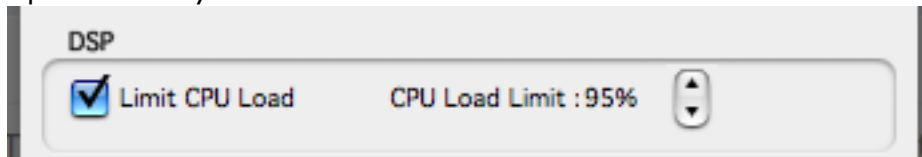


To configure the UAD-1 for Cubase and Nuendo large ASIO buffer size support:

1. Launch the UAD-1 Performance Meter.
2. Open the system menu by clicking the icon at the upper left of the Performance Meter and select the ‘Configuration’ option.
3. Increase the Extra Buffers control until “New latency” matches the current buffer size of the ASIO device.
4. Reset the ASIO device using one of the following methods:
 - Close the re-open the session
 - Stop then restart the audio engine
 - Modify or reset the audio device settings
5. The “Current latency” display should now match the “New latency” display. Configuration of Extra Buffers is complete.

DSP Settings

These controls limit the maximum UAD-1 CPU load before no more plugins will be processed by the UAD-1.



Limit CPU Load

CPU load limiting was introduced in version 3.0. The option is off by default so sessions with heavy UAD-1 loads that have been saved with UAD-1 software versions previous to 3.0 will playback without any UAD-1 plugins being disabled due to overload.

Without UAD-1 Powered Plug-Ins installed, overloading the host system with native (host based) plugins can cause dropouts and possibly system lockup. Steinberg hosts, for example, provide a switch that allows you to trade latency for stability when the system is overloaded. Similarly, the UAD-1 DSP load cannot exceed 100% without unpredictable behavior.

With the Limit CPU Load feature, the UAD-1 CPU can also be limited so the load cannot exceed 100%, thereby increasing overall system stability in high load situations. With very heavy UAD-1 loads, CPU load limiting may also improve host CPU performance.

There are many variables that affect DSP load (sample rate, bit depth, buffer size, parameter values, mono/stereo, automation, host system, etc). Although these variables are taken into account, the resulting measurement cannot be absolutely accurate. This is due to variations in system configurations, specifically PCI bus loading which is impossible to predict. Systems that are heavily loaded due to the presence of other devices or suboptimal configuration may cause additional DSP loading that cannot be predicted by the plugin load calculator. The CPU load limit should be reduced in this case.

It is possible for certain (non-typical) conditions to be met where another UAD-1 plugin can't be added, even when the UAD-1 Meter says you should have CPU available when compared to the CPU Load Limit value.

Limit CPU Load

UAD-1 CPU Load Limiting is enabled when this box is checked.

CPU Load Limit

This setting controls the maximum UAD-1 CPU when load limiting is enabled. It has no affect when CPU load limiting is off.

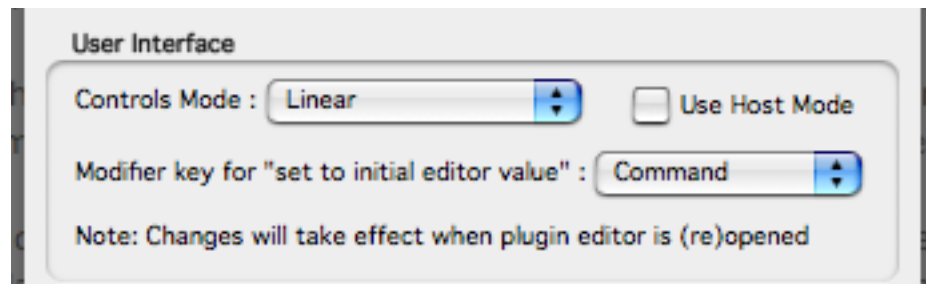
Note: When *Limit CPU Load* is enabled and the *CPU Load Limit* is exceeded when instantiating a new UAD-1 plugin, the plugin that was just added will be disabled. Even though its interface will load, it will not process audio.

User Interface Settings

Controls Mode

This setting determines how Powered Plug-In parameter knobs respond to adjustment. Three control modes are offered: Circular, Relative Circular, and Linear.

Note: To increase resolution when in adjusting rotary controls in circular and relative circular modes, increase the radius of the mouse relative to the knob while dragging (i.e. move the mouse farther away from the knob while dragging in a circular motion).



Circular (jump)

In Circular mode, the software knobs behave similar to physical knobs. Values are changed by clicking on the knob then rotating in a circular direction. When the edge of the knob is clicked, the parameter value jumps to the mouse position.

Relative Circular (grab)

Relative Circular mode operates similar to Circular mode, but the knob value does not jump to the mouse position when clicked. Instead, the knob value is modified relative to its original value.

In this mode you can click anywhere on the knob to make an adjustment originating at the original value. You don't have to click on the current knob position.

Linear (slider)

In Linear mode, the knob is adjusted by dragging horizontally or vertically instead of by rotating. This behavior is similar to moving a slider.

Use Host Mode

When Use Host Mode is checked, the control mode set within the host application preferences is used if this feature is supported by the host. This setting forces the host to override the control mode set in the UAD-1 user interface settings.

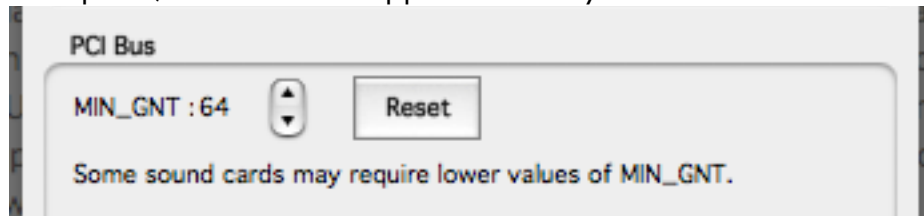
Note: When Use Host Mode is checked, the UAD-1 Meter user interface settings have no effect unless control mode is NOT supported by the host.

Modifier Key (Mac OS)

The Modifier Key drop-menu allows you to specify which modifier key will be used for the “set to last saved value” keyboard shortcut. It also affects the “select + click” modifier. This feature is not supported under Windows. For a complete list of keyboard shortcuts, see [“Shortcuts” on page 25](#).

PCI Bus Settings

MIN_GNT is a low-level system setting that affects PCI bandwidth. The default value of 64 is recommended for most systems. If you are experiencing crackles or dropouts, our technical support team may recommend a different value.



Important: System performance can be adversely affected by changing this setting. This parameter DOES NOT AFFECT AUDIO LATENCY in any way!

AMD-8131 Mode

If your computer uses the AMD-8131 PCI controller chipset, check the “AMD-8131 Compatible” box. This will improve UAD-1 performance on these systems. For the new setting to take affect, you must reset the ASIO device using one of the following methods:

- Close the re-open the session
- Stop then restart the audio engine
- Modify or reset the audio device settings

Note: Do not enable AMD-8131 Mode unless your computer uses this PCI controller chipset. AMD-8131 Compatible Mode is only required when the card is attached directly to an AMD-8131 PCI bus. If the UAD-1 is in an external PCI expansion chassis, this mode should be disabled (unchecked).

Macintosh G5 Systems

The AMD-8131 chipset is used in most Macintosh G5 systems. The UAD-1 software automatically determines when it is running on a G5 with AMD-8131 and sets the mode appropriately. If the UAD-1 is in an external PCI expansion chassis, AMD-8131 mode should be unchecked.

Misc Settings (Mac OS X)

Release all DSP resources on AudioUnit bypass

Some Audio Unit hosts dynamically bypass plugins when they are not being used during playback, for example when no audio is present at the current playback position. As of version 3.9.0, UAD-1 plugins are no longer unloaded/reloaded each time the host performs this dynamic bypass. Instead, the UAD-1 plugins stay loaded on the card, which reduces playback glitches.

Checking the “Release all DSP resources on AudioUnit bypass” option will unload UAD-1 plugins from the card during dynamic bypassing. When the option is checked, UAD-1 DSP usage may be reduced during dynamic bypassing, but the possibility of glitching during playback is increased.

Note: *This setting affects the Mac OS X Audio Units platform only.*

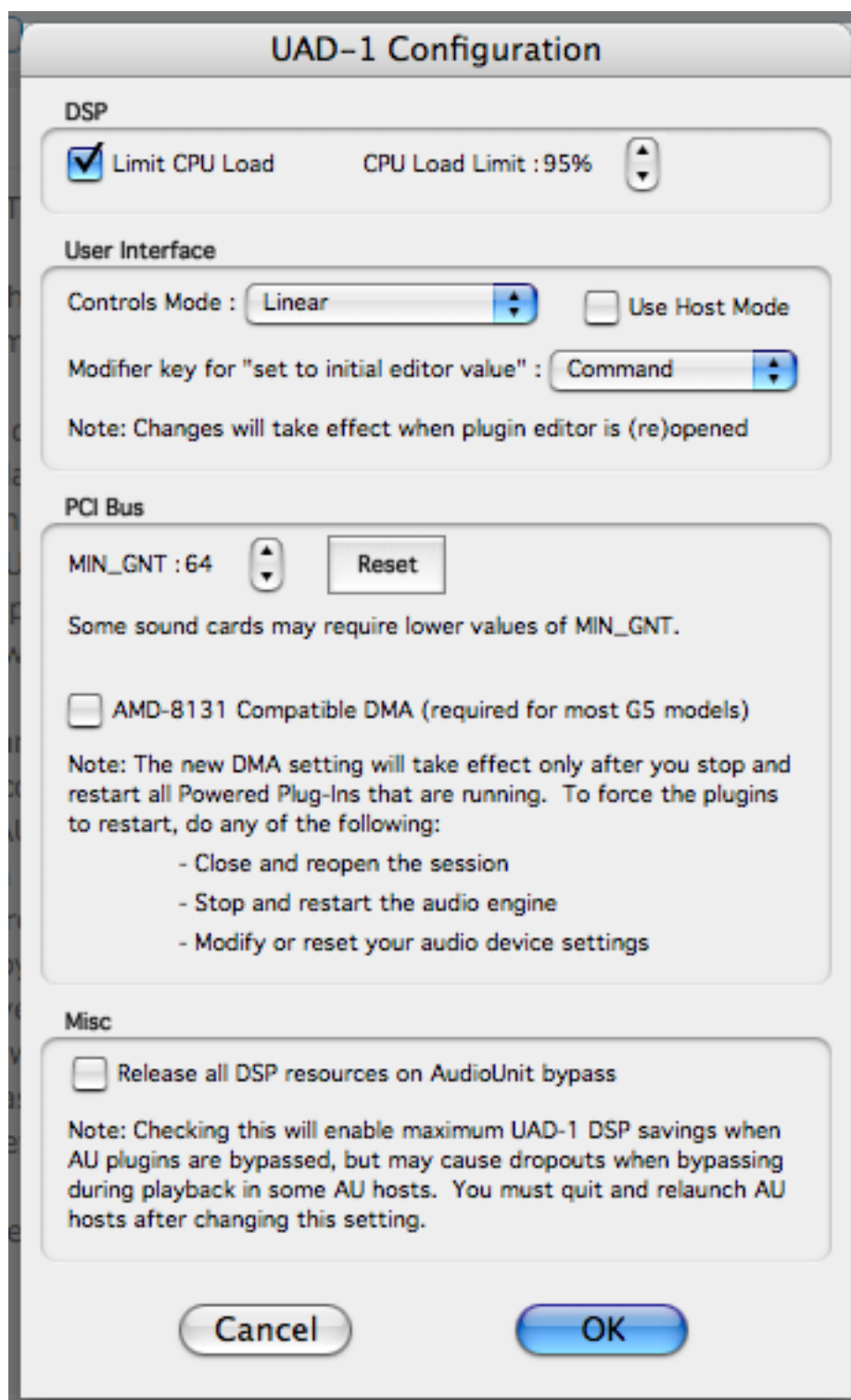


Figure 12. The UAD-1 System Configuration window (Mac OS)

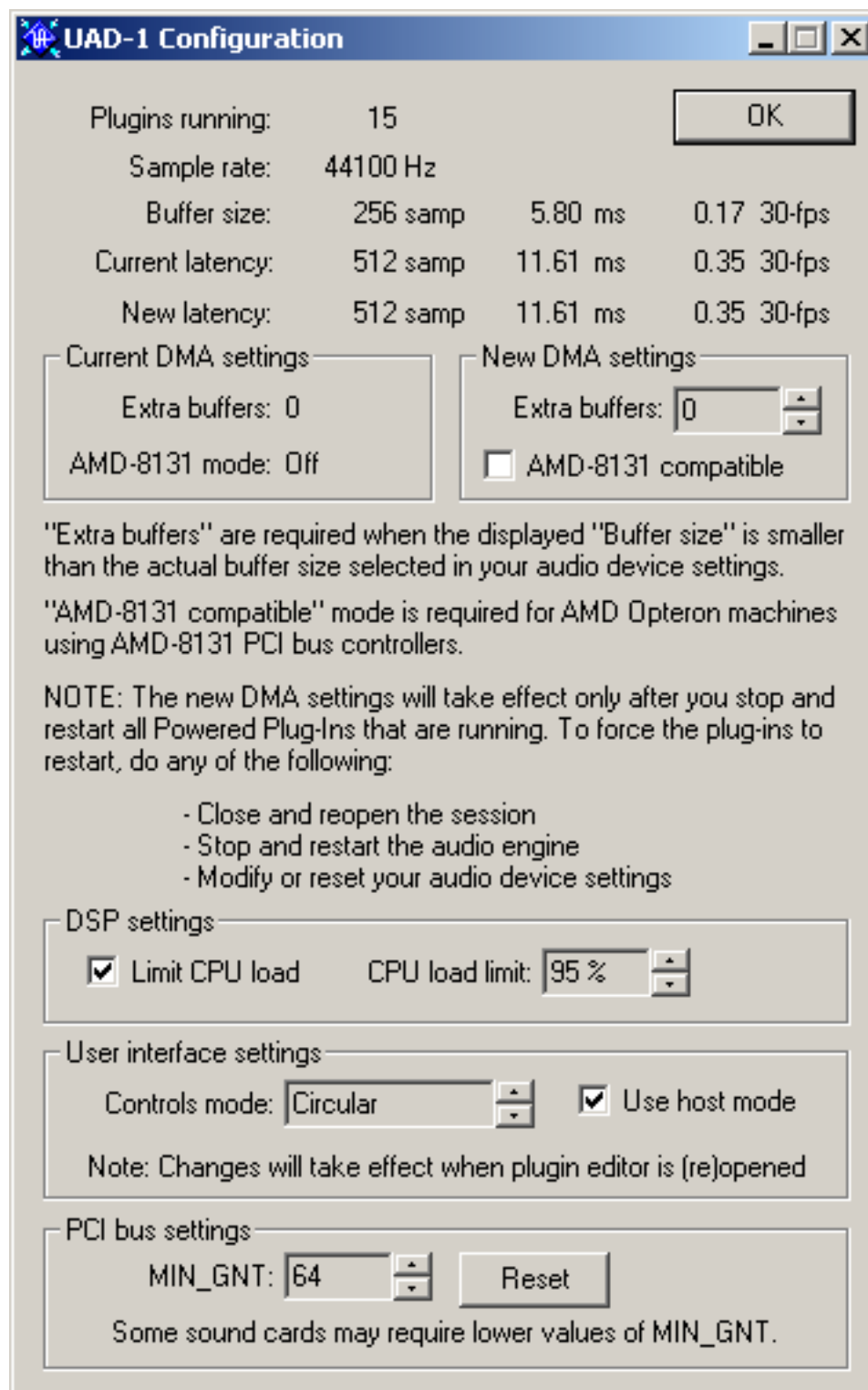


Figure 13. The UAD-1 System Configuration window (windows)

Delay Compensation

Overview

When UAD-1 Powered Plug-Ins are used, audio data to be processed by a Powered Plug-In is sent by the host application to the UAD-1 card. The audio is then processed by the UAD-1 card and sent back to the host application.

This back-and-forth shuffling of audio data produces a latency (delay) in the audio signal being processed. Latency time is determined by the sample rate, the hardware device driver (ASIO or similar) buffer setting, and the Extra Buffers (if any) in the UAD-1 Configuration window.

If this latency is not compensated, the processed audio will not be perfectly synchronized with unprocessed audio. Fortunately, most host applications automatically compensate for this latency when plugins are used on track inserts by simply turning on the “Plugin Delay Compensation” or similar Preferences setting. Some hosts even provide “Full Delay Compensation” throughout the entire signal path, including sends, groups, and busses.

However (depending on the host application implementation), the delay compensation feature may not provide automatic compensation when UAD-1 Powered Plug-Ins are inserted on sends, groups, or busses. In this situation, the solution is to use the UAD Delay Compensator plugin (“[UAD Delay Compensator plugin](#)” on page 44).

Note: *UAD DelayComp should not be used in situations where the host application provides delay compensation automatically, such as on track inserts. Some host applications provide fully automatic delay compensation throughout the entire signal path. UAD DelayComp is not needed at all in such hosts. As of this writing, hosts with fully automatic compensation include all DirectX hosts, Steinberg Nuendo 2.x/Cubase SX 2.x, Magix Samplitude 7.x, and Apple Logic Pro 7.1.*

Note: *These explanations of delay compensation apply primarily to playback only. For more information about using UAD-1 Powered Plug-Ins for live performance and during recording, “[Live Processing](#)” on page 53.*

Note: *For information about using UAD-1 Powered Plug-Ins on audio tracks while simultaneously running MIDI tracks, “[UAD Track Advance](#)” on page 50.*

Host Application Settings

For optimum results, the “Plugin Delay Compensation” option setting should be active in the host application. This will provide automatic latency compensation when UAD plugins are used on track inserts (and sends/group/busses if full compensation is supported), so the UAD DelayComp will not have to be used. This option is usually found in the audio or plugin preferences window. The specific location of the switch for this option within several popular applications is as follows:

- Cubase 5.x: Options Menu>Audio Setup>System...
- Nuendo 1.x, Cubase SX 1.x: File Menu>Preferences>VST
- Logic Audio: Audio Menu>Audio Preferences...

UAD Delay Compensator plugin

Overview

The UAD Delay Compensator (DelayComp for short) is a simple plugin which can be used to synchronize unprocessed tracks with those that are processed by UAD-1 Powered Plug-Ins. It provides a mechanism of delay compensation for situations when the host application does not implement automatic plugin latency compensation, such as on sends, groups, and busses.

The UAD Delay Compensator acts as a dummy UAD-1 Powered Plug-In, automatically introducing the necessary amount of latency for tracks which are *NOT* processed by UAD-1 Powered Plug-Ins. It requires no DSP from the host CPU or the UAD-1 card and allows you specify the number of UAD-1 Powered Plug-Ins instances you wish to compensate.

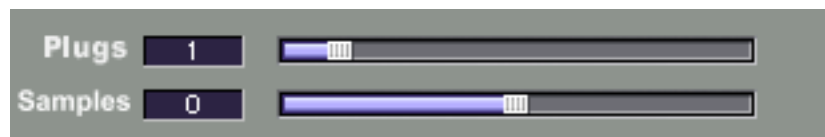


Figure 14. The UAD Delay Compensator plugin window

When to use DelayComp

UAD DelayComp should be used whenever unprocessed audio tracks are played alongside audio tracks that are assigned to a send/group/bus that is using a UAD plugin(s). In this scenario, inserting a UAD DelayComp plugin on the UNPROCESSED track(s) will automatically re-synchronize the audio.

VSTi

Cubase and Nuendo currently do not automatically compensate for latency on MIDI virtual instrument (VSTi) tracks. Therefore, UAD DelayComp should also be used on non-VSTi tracks when VSTi's are in use.

Note: *This does not apply to Nuendo 2.x and Cubase SX 2.x, which support fully automatic delay compensation throughout the entire signal path.*

Note: Check out our UAD DelayComp Examples on the UAD-1 Powered Plug-Ins CD-ROM (or download them from our website) for “real-world” examples formatted for several popular host applications.

Note: When the UAD Pultec EQ is being used in conjunction with DelayComp, the Samples parameter should be used in addition to the Plugs parameter (“Compensating for Pultec EQ” on page 47).

Plugs parameter

The DelayComp Plugs parameter value to be used on an unprocessed track or tracks is simply the number of UAD-1 Powered Plug-Ins that are being used in sequence on the send, group, or bus.

For example, if three separate sends are used and each send has one instance of UAD plugins, the Delay Compensator Plugs value for the unprocessed tracks would be one. However, if one send/group/bus is used that has three instances of UAD-1 Powered Plug-Ins stacked up, the Delay Compensator Plugs value for the dry tracks would be three.

Note: The Delay Compensator “Plugs” value matches the total of UAD-1 Powered Plug-Ins used serially (stacked one above another in series), NOT the total number of UAD-1 Powered Plug-Ins used.

Samples parameter

The Samples parameter shifts the audio with single-sample accuracy in either direction. It is provided mainly for compensation of the Pultec EQ (“Compensating for Pultec EQ” on page 47 for more information), Precision Limiter, and Precision Equalizer. However, it can be used anytime minute shifting of audio is desired. Audio can be shifted up to 128 samples in either direction.

Grouping Tracks Requiring DelayComp

The UAD DelayComp plugin is generally used on track inserts. However, when many tracks require delay compensation, instead of placing individual Delay Compensator plugins on each track you may find it easier to send the output of each unprocessed tracks to a bus or group. Then simply put one UAD Delay Compensator on that bus or group.

DelayComp Examples

Insert

Situation: You have a song with bass, drums, and guitar. You want a room simulator on the guitar so you put an RS-1 on an insert of the guitar track. Result: All tracks are perfectly aligned.

Solution: None needed. Delay compensation on track inserts is handled automatically by most host applications.

Send

Situation: You have a song with bass, drums, guitar, and 2 vocal tracks. You want a fantastic reverb on the vocals so you send both vocal tracks to the UAD RealVerb Pro via an effect send. Result: The RealVerb Pro effect return plays late in relation to the dry tracks.

Solution: Send the output of all the tracks (including the dry vocal tracks but NOT the RealVerb Pro return) to a different send/group/bus and put one UAD DelayComp with a Plugin value of 1 on this send/group/bus that contains the dry tracks. Keep the Sample value at zero.

Group/Bus

Situation: You have a song with bass, drums, guitar, and 2 vocal tracks. You want a smoother vocal blend so you put both vocal tracks on a group/bus for compression with the infamous LA2A. Result: The vocal tracks play late in relation to the instrument tracks.

Solution: Send the output of the unprocessed instrument tracks (but not the vocal tracks or LA2A return) to a different group/bus and put one DelayComp with a Plugin value of 1 on this group/bus that contains the unprocessed tracks. Keep the Sample value at zero.

Note: Some host applications provide fully automatic delay compensation throughout the entire signal path. UAD DelayComp is not needed at all in such hosts. As of this writing, hosts with fully automatic compensation include all DirectX hosts, Steinberg Nuendo 2.x/Cubase SX 2.x and higher, Magix Samplitude 7.x, and Apple Logic Pro 7.1.

Note: Check out our UAD DelayComp Examples on the UAD-1 Powered Plug-Ins CD-ROM (or download them from our website) for “real-world” examples formatted for several popular host applications.

Compensating for Pultec EQ

The UAD Pultec and UAD Pultec-Pro equalizers use an internal sample rate of 192kHz to achieve their magic quality. This upsampling results in a slightly larger latency than other UAD-1 plugins. Therefore, they require slightly more compensation to remain perfectly synchronized with other tracks. Specifically, they require an extra 13 samples of compensation when the session sample rate is below 100kHz.

Therefore, when using the DelayComp or TrackAdv plugin to manually compensate for a Pultec EQ, enter a Samples value of 13 (in addition to the appropriate "Plugs" value) for each instance of Pultec and Pultec-Pro.

Note: *Compensating for Pultec and Pultec-Pro is not required if the host application supports “full delay compensation” throughout the signal path.*

Note: *When running audio at or above 100kHz resolution, no Pultec upsampling is performed. In this case, leave the Samples value at zero.*

Note: *The keyboard shortcuts PageUp/PageDown and Shift+Arrow automatically add the 13-sample Pultec value for your convenience.*

Pultec Group/Bus Example

Situation: You have a song with bass, drums, guitar, and 2 vocal tracks. You want a fat, warm vocal blend so you put both vocal tracks on a group/bus and apply one instance of Pultec EQ to the vocal bus. Result: The vocal tracks play late in relation to the instrument tracks.

Solution: Send the output of the unprocessed instrument tracks (but not the vocal tracks or the Pultec return) to a different group/bus, and put one DelayComp with a Plugin value of 1 and a Sample value of 13 on this group/bus that contains the dry non-vocal tracks. If you put 2 Pultec EQ's stacked in series on the vocal bus, the unprocessed bus DelayComp Plugin value would be 2, and the Sample value would be 26.

Note: *UAD Pultec-Pro only requires one compensation per instance. For example, if using both MEQ-5 and EQP-1A within a single Pultec-Pro, only one instance compensation is required.*

Compensating for Precision Limiter

The Precision Limiter has a 1.5ms look-ahead window to ensure clipping does not occur. This look-ahead function results in a slightly larger latency than other UAD-1 plugins. This is not normally an issue because the Precision Limiter is designed to be used for program material on the output bus, where latency is not a consideration. However, if the Precision Limiter is used elsewhere in the signal chain and the host does not automatically compensate for latency at that point in the signal chain, it requires slightly more compensation to remain perfectly synchronized with other tracks.

Therefore, when using the DelayComp or TrackAdv plugin to manually compensate for the Precision Limiter, enter the Samples value from [Table 2 on page 48](#) (in addition to the appropriate "Plugs" value) for each instance of Precision Limiter.

Note: *Compensating for Precision Limiter is not required if the host application supports "full delay compensation" throughout the entire signal path, or when it is used only on the outputs.*

The compensation value to use depends on the session sample rate. Use the following values to compensate for Precision Limiter latency when using it on track inserts or busses.

Table 2. Precision Limiter Latency Compensation Values

Session Sample Rate	UAD DelayComp "Samples" Value
44.1kHz	64 Samples
48kHz	69 Samples
88.2kHz	129 Samples
96kHz	140 Samples
176.4kHz	259 Samples
192kHz	281 Samples

Precision Limiter Group/Bus Example

Situation: You have a song with bass, drums, guitar, and 2 horn tracks. You want to maximize the level for the horns so you put both horn tracks on a group/bus and apply one instance of Precision Limiter to the horn bus. Result: The horn tracks play late in relation to the other instrument tracks.

Solution: Send the output of the unprocessed instrument tracks (but not the horn tracks or the Precision Limiter return) to a different group/bus, and put one DelayComp with a Plugin value of 1 and a Sample value from [Table 2 on page 48](#) on this group/bus that contains the dry non-horn tracks. If you put 2 Precision Limiters stacked in series on the horn bus, the unprocessed bus DelayComp Plugin value would be 2, and the Sample value would be twice the chart value.

Compensating for Precision Equalizer

The Precision Equalizer uses an internal sample rate of 192kHz to facilitate its amazing sonic quality. This upsampling results in a slightly larger latency than other UAD-1 plugins. Therefore, it requires slightly more compensation to remain perfectly synchronized with other tracks. This is not normally an issue because the Precision Equalizer is designed to be used for program material on the output bus, where latency is not a consideration. However, if the Precision Equalizer is used elsewhere in the signal chain and the host does not automatically compensate for latency at that point in the signal chain, it requires slightly more compensation to remain perfectly synchronized with other tracks.

Therefore, when using the DelayComp or TrackAdv plugin to manually compensate for the Precision Equalizer, enter the Samples value from [Table 3 on page 49](#) (in addition to the appropriate "Plugs" value) for each instance of Precision Equalizer.

Note: *Compensating for Precision Equalizer is not required if the host application supports “full delay compensation” throughout the entire signal path, or when it is used only on the outputs.*

The compensation value to use depends on the session sample rate. Use the following values to compensate for Precision Equalizer latency when using it on track inserts or busses.

Table 3. Precision Equalizer Latency Compensation Values

Session Sample Rate	UAD DelayComp “Samples” Value
48kHz and lower	31 Samples
88.2kHz and 96kHz	13 Samples
176.4kHz and 192kHz	Zero

Precision Equalizer Group/Bus Example

Situation: You have a song with bass, drums, guitar, and 4 string tracks. You want to change the EQ for all the strings so you put the string tracks on a group/bus and apply one instance of Precision Equalizer to the string bus. Result: The string tracks play late in relation to the other instrument tracks.

Solution: Send the output of the unprocessed instrument tracks (but not the string tracks or the Precision Equalizer return) to a different group/bus, and put one DelayComp with a Plugin value of 1 and a Sample value from [Table 3 on page 49](#) on this group/bus that contains the dry non-string tracks. If you put 2 Precision Equalizers stacked in series on the string bus, the unprocessed bus DelayComp Plugin value would be 2, and the Sample value would be twice the chart value.

UAD Track Advance

Overview

The previous discussions on delay compensation (see [“Delay Compensation” on page 43](#)) apply mainly when using only audio tracks. When MIDI tracks are played simultaneously alongside audio tracks, a different (but related) synchronization issue can arise.

Let’s say you have a MIDI track and an audio track with a UAD plugin on the audio track insert. In this scenario, the host application will automatically compensate for latency and no use of the UAD DelayComp or UAD Track Advance is required.

However, if the audio track is sent to a send/group/bus and that send/group/bus has a UAD plugin on it, the audio track will be delayed in relation to the MIDI track because the host does not compensate for latency automatically on groups/busses. If the MIDI track was an audio track, you would use the UAD DelayComp on it to compensate for the latency. But you can’t put a UAD DelayComp on a MIDI track, so what to do?

Enter the UAD Track Advance plugin (TrackAdv for short). It operates just like the DelayComp plugin, but backwards. Instead of delaying unprocessed tracks, it shifts them forward. It does this by reporting to the host application that a track has a UAD plugin on it, so the host compensates for the latency. However, the track audio is not actually processed by the UAD-1 so the net result is that the audio plays early.

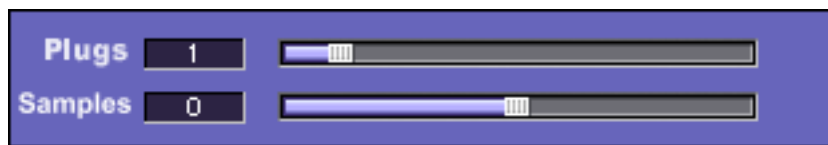


Figure 15. The UAD Track Advance plugin window

Note: *The Track Advance plugin only works in hosts that support automatic delay compensation. This is why the MAS version does not include UAD TrackAdv.*

When to use TrackAdv

UAD TrackAdv should be used whenever MIDI tracks are played alongside audio tracks that are assigned to a send/group/bus that is using UAD plugin(s).

Note: *UAD TrackAdv should not be used in host applications that provide fully automatic delay compensation throughout the entire signal path. UAD TrackAdv or DelayComp is not needed at all in such hosts. As of this writing, hosts with fully automatic compensation include all DirectX hosts, Steinberg Nuendo 2.x/Cubase SX 2.x and higher, Magix Samplitude 7.x, and Apple Logic Pro 7.1*

How to use TrackAdv

UAD TrackAdv is designed to be used on audio track inserts of tracks that are assigned to a send/group/bus that has one or more UAD plugins applied. By first advancing the audio with TrackAdv on the track insert then processing the same track on a send/group/bus that has a UAD plugin, the “net latency result” is zero and the audio will be perfectly aligned with the MIDI tracks.

Plugs parameter

The TrackAdv Plugs parameter value to be used on a track insert is simply the number of UAD plugins that are being used in sequence on the send, group, or bus that the track is assigned to.

For example, if three separate sends are used and each send return has one instance of UAD plugins, the TrackAdv Plugin value for the audio tracks insert would be one. However, if one send/group/bus is used that has three instances of UAD-1 Powered Plug-Ins stacked up, the TrackAdv Plugs value for the tracks inserts would be three.

Note: *The TrackAdv “Plugs” value on the track insert matches the total of UAD-1 Powered Plug-Ins used serially (stacked one above another in series) on the send/group/bus that the track is assigned to, NOT the total number of UAD-1 Powered Plug-Ins used.*

Samples parameter

The Samples parameter shifts the audio with single-sample accuracy in either direction. It is provided mainly for compensation of the Pultec EQ (“Compensating for Pultec EQ” on page 47 for more information). However, it can be used anytime minute shifting of audio is desired. Audio can be shifted up to 128 samples in either direction.

Note: *The keyboard shortcuts PageUp/PageDown and Shift+Arrow automatically add the 13-sample Pultec value for your convenience.*

TrackAdv Examples

Insert

Situation: You have one track with MIDI and one track with audio. You put a UAD plugin on the audio track. Result: All tracks are perfectly aligned.

Solution: None needed. Delay compensation on track inserts is handled automatically by most host applications.

Send

Situation: You have a song with drums and guitar on audio tracks, and a MIDI bass line. You want a cohesive room reverb on the audio tracks so you send them to the UAD RealVerb Pro via an effect send. Result: The RealVerb Pro effect return plays late in relation to the MIDI track.

Solution: Put a TrackAdv plugin on the track insert of the audio tracks with a Plugs value of 1. If you had an 1176LN and a RealVerb Pro on the send return, the TrackAdv Plugs value would be 2. Keep the Sample value at zero.

Group/Bus

Situation: You have a song with 2 vocals on audio tracks, and a MIDI piano. You want a smoother vocal blend so you put both vocal tracks on a group/bus for compression with the infamous LA2A. Result: The vocal tracks play late in relation to the MIDI track.

Solution: Put a TrackAdv with a Plugin value of 1 on the track inserts of the vocal tracks. Keep the Sample value at zero.

Situation: You have a song with drums, guitar, and 2 separate vocals on audio tracks, and a MIDI bass line. You want a smoother vocal blend so you put both vocal tracks on a group/bus for compression with the 1176LN. Result: The vocal tracks play late in relation to the instrument tracks.

Solution: Put a TrackAdv with a Plugin value of 1 on the track inserts of the vocal tracks. Keep the Sample value at zero. Note that the DelayComp plugin is not needed at all in this situation.

Live Processing

The previous discussion of delay compensation applies primarily to playback and mixing of existing tracks. During recording (tracking), the primary concern usually centers around getting the absolute lowest possible latency out of your hardware and software combination. The lower the latency is, the closer you can get to a realtime, “ears match the fingers” performance situation in the digital environment where some latency is unavoidable.

Minimizing realtime latency is simply a matter of setting the hardware device driver (ASIO or similar) buffer setting as low as possible before system overloads or diminished audio quality (such as distortion) occurs. The manufacturer of the sound output device in use may offer additional tips for optimizing latency on systems that use their hardware.

Note: *Keep in mind the latency for each instance of UAD-1 Powered Plug-Ins is equal to twice the current buffer size of the host system. This is because audio needs to travel to the UAD-1 card, then back again. For example, with a buffer size of 256 samples, one Powered Plug-In will introduce 512 samples of latency, and two Powered Plug-Ins in succession will introduce 1024 samples of latency.*

DSP Usage

The UAD-1 card features an on-board CPU and 4 MB of memory for processing Powered Plug-Ins. The host system memory and CPU are never used for Powered Plug-Ins processing. However, there will always be a small amount of load on the host CPU induced by PCI data transfer and user interface operations. This is unavoidable when using a DSP card.

- UAD-1 CPU usage is proportional to the host application sample rate and system PCI bus speed. Therefore, more plugins can be used simultaneously in a 44.1K session than in a 96K session, and likewise a higher speed PCI bus will use less CPU load than a slower bus.
- Bypassing individual components will conserve CPU. For example, bypassing the compressor in the EX-1 when only the EQ is in use, and/or bypassing any of the unused bands of the EX-1 EQ will use less UAD-1 CPU.
- The UAD-1 CPU resources required by each successive UAD-1 Powered Plug-In instance will slightly decrease.

Multiple Cards

When multiple UAD-1 cards are installed in the host computer, the CPU and memory load of the cards are automatically balanced dynamically in real-time. With multiple cards there is no major difference in operating procedures, except that more Powered Plug-Ins can be loaded in the session.

Note: For information about authorizing copy-protected plugins on multiple cards, see [“Authorizing Multiple Cards” on page 58](#).

Power Requirement

Up to four UAD-1 cards can be installed simultaneously in the host computer. Each UAD-1 card uses a maximum of 14 watts of 5 volt power from the PCI bus. The PCI specification provides for up to 25W per device, however some host systems don't provide (or require) this much power.

Important: If insufficient power is available to the UAD-1 cards, unpredictable behavior may result.

Multicard Use

The UAD-1 card that has the lowest resource usage will receive the next Powered Plug-In load. Note that an individual UAD-1 plugin cannot be split across two (or more) UAD-1 cards.

For example, let's say you have two UAD-1 cards installed, the UAD-1 Meter displays 90%, you load another UAD-1 plugin that requires 6% CPU, yet you get a “plugin unable to load” message. This would occur if both cards are already at 95% (the meter shows the *total* available CPU, not the per-card CPU), so a 6% plugin can't load.

System Info Window

UAD-1 CPU and memory resources used for each installed card, and the ability to enable/disable individual cards, is displayed in the System Information window (see [page 34](#)).

Disabling Cards

Individual UAD-1 cards can be disabled using the Card Enabled function (see [page 33](#)). This can be useful, for example, if creating a session on a system with multiple cards that will be transferred to a system with fewer cards.

Note: For optimum results, quit any applications using UAD-1 plugins before disabling/enabling cards.

If a Powered Plug-In is loaded on a card then that card is subsequently disabled, an error message will be displayed. This occurs because a plugin is assigned to a card when it is first instantiated. It stays assigned to the same card until it is de-instantiated (i.e. removed from the insert slot).

Host CPU

Using more than one card can cause a slight increase in host CPU requirements, so disabling unused cards can help you squeeze in a bit more host performance if you need it. Using additional devices on the PCI bus requires host resources, so running 15 UAD plugins on three cards at five plugins per card may require more host CPU than running the same 15 UAD plugins on one card.

For example, if you are trying to minimize latency during tracking by using a smaller buffer size (which will increase host CPU) and need a bit more host CPU, disabling one or more UAD-1 cards during tracking may give the extra pinch of host CPU you need. The buffer size can then be increased and the UAD-1 card(s) re-enabled for mixing.



Optional Plug-Ins

Overview

The UAD-1 Powered Plug-Ins software installation bundles always includes every component that is part of the current software version, such as drivers, plugins, UAD-1 Meter application, and documentation. Version 3.9 includes eight optional copy-protected plugins that can be authorized for an additional fee.

UAD Cambridge, UAD DreamVerb, UAD Fairchild, UAD Pultec-Pro, UAD Precision Limiter, UAD Plate 140, UAD Precision Equalizer, and UAD Roland CE-1 (also UAD 1176LN and UAD LA-2A with ProjectPAK) require authorization for unlimited use. Without authorization, these plugins can be enabled to run for 14 days without functional limitations in a timed demo mode (see “[Demo Mode](#)” on page 61).

Purchasing

To authorize the plugin for permanent unlimited use, an authorization license key is purchased on the internet via our web-based online store. To purchase authorizations, go to:

- <https://my.uaudio.com/store/>

Process

To purchase a plugin, you must have an account on our secure web server at my.uaudio.com (your account ID is your email address). You enter your information, including UAD-1 hardware identification number(s), into your account. Plugins can then be securely purchased with a valid credit card or personal check.

After payment is received, a “.reg” settings file containing the authorization key(s) can then be downloaded directly from your personalized account pages at my.uaudio.com. On the PC, this file is loaded into the host computer containing the UAD-1 card by simply double-clicking the file. On Mac OS, this file is dragged onto the UAD-1 Meter application icon (not the DSP/Memory bar graph window!) or loaded using the “Load Settings File” menu item from within the Meter.

The specific details of the authorization process is detailed later in this section (see “[Plug-In Authorization Procedure](#)” on page 62).

Ultra PAK

The UAD-1 Ultra PAK includes a license for the following paid UAD plugins: 1176LN, Cambridge, DreamVerb, Fairchild, Fairchild Mono, LA2A, Pultec Pro, Plate 140, Precision Limiter, and Precision Equalizer.

Studio PAK

The UAD-1 Studio PAK includes a license for the following paid UAD plugins: 1176LN, Cambridge, DreamVerb, Fairchild, Fairchild Mono, and LA2A.

To authorize these plugins for use with UAD-1 Ultra Pak or Studio PAK, create your account at my.uaudio.com. Ultra PAK and Studio PAK licenses are issued automatically and can be downloaded after entering your hardware identification number(s) into your my.uaudio.com account.

Project PAK

The UAD-1 Project PAK does not include paid plugins, which can be acquired after registration at my.uaudio.com. If you have previously purchased plugins for UAD-1 Studio PAK or the Mackie UAD-1, those plugins are licensed to run on up to four cards (including UAD-1 Project PAK cards which have been added to existing UAD-1 setups).

Note: *The Mackie UAD-1 card includes licenses for UAD 1176LN and UAD LA-2A, but the Project PAK card does not.*

.reg Authorization Key File

The .reg file contains the authorization key(s) that allow a plugin to run on the authorized cards, and is independent of the machine and operating system. The authorizations can be used on multiple machines if the card is moved around (for example, if a PCI expansion chassis is shared between a laptop and desktop machine). These authorizations will remain valid even if the plugin software is subsequently replaced with the same or newer version, or if any host system hardware or operating system changes are made.

The .reg files contain only the Hardware ID and authorization codes that enable a plugin to run on specific UAD-1 card(s). It does not contain other system information that restrict use of the cards or authorized plugins.

Note: *It is the UAD-1 hardware that is authorized, not the software plugin.*

If the UAD-1 card is installed into a different system, the same (already acquired) .reg file must be loaded into the different computer and the card becomes authorized to run the plugin on that system. The same UAD-1 and its associated .reg file can be loaded into an unlimited number of computer systems (including Windows and Mac systems, as the card and key file are cross-platform).

Important: *The .reg settings file contains the authorization key for the specific UAD-1 card. It is required every time the card needs to be authorized, such as if it is installed into a different computer. Back it up and keep it in a safe place!*

Mac OS

In OS 9, once a card is authorized and it is used with a different System Folder or CPU, you can copy the UAD Registry file (inside System Folder:Preferences) into the current System Folder:Preferences instead of loading the .reg file. This has the advantage of not only copying your authorizations, but all your other UAD-related preferences as well. In OS X, once a card is authorized and it is used with a different System or CPU, you can copy the com.uaudio.uad.plist file (inside Library:Preferences) into the current Library:Preferences instead of loading the .reg file.

Order Fulfillment Timing

As of version 3.4, purchasing and downloading plugin authorizations is a completely automatic process. Purchased plugin authorization keys can now be downloaded within minutes of payment verification.

Note: *We recommend you do not wait until the last day of demo period to purchase plugin authorizations, in case there is a unexpected delay in obtaining authorizations.*

Authorizing Multiple Cards

When you buy a copy-protected plugin, your purchase is valid for up to four UAD-1 cards. However, the authorization key is tied to the specific Hardware IDs entered during purchase. If you have four cards when you purchase the plugin, your .reg file will authorize all four cards. If you have one card during purchase then later acquire another card(s), you will need to acquire a new .reg file containing the authorization for the new card(s). However, there is no charge for the new .reg file containing the additional authorizations. Your purchase buys authorizations for a maximum of four cards.

Each individual card must be authorized in order to run copy-protected plugins. If one card is authorized then another card is added without obtaining a new .reg file, the copy-protected plugin will only load on the authorized card.

To obtain a new .reg key file for additional cards, visit my.uaudio.com on the internet.

If two cards are authorized at the same time then one is removed, the remaining card stays authorized. If you install an authorized card into another system or want to use it with a different OS partition on the same system, you will need to reapply any .reg files containing the card's authorizations on the new system or OS partition.

Authorizations Window

The Authorizations window (Figure 16) is accessed within the UAD-1 Meter application.

Hardware IDs	
Card 1 : 9a7a-ca07-e06f-acfa-a0ea S/N :	Copy
Card 2 : 6a7a-c1c7-eb6d-ac9a-f248 S/N :	Copy
Card 3 : aa7a-6a7a-c1c7-eb6d-ac9a S/N :	Copy
Card 4 : 4a7a-ca07-e06f-acfa-a0ea S/N :	Copy

Authorizations	
UAD Cambridge :	Authorized for all cards.
UAD DreamVerb :	Authorized for all cards.
UAD Fairchild :	Authorized for all cards.
UAD Pultec-Pro :	Authorized for all cards.
UAD 1176LN :	Authorized for all cards.
UAD LA2A :	Authorized for all cards.
UAD Precision Limiter :	Authorized for all cards.
UAD Plate 140 :	Authorized for all cards.

Figure 16. The UAD-1 Authorizations window

On Windows systems, access the Authorizations window from the System Menu or alternate system menu. On Mac OS, use the File Menu. See ["Accessing Meter Functions" on page 29](#) for specific instructions.

Hardware IDs

This section of the window is where the unique hardware identification numbers of all installed cards are displayed. These ID numbers are required to obtain the authorization key.

The Copy button can be used to copy the hardware ID text to the operating system clipboard. This text can then be pasted into the web form.

Authorizations

The status of each authorizable plugin and function is displayed in this area. A "Start demo" button is present if any UAD-1 demo timer has not been started.

When the Start demo button is clicked, it checks to see if any UAD-1 plugins are running and if they are, a message instructing you to quit the host application is displayed. The demo cannot be started if any UAD-1 plugins are running.

Demo Mode

If demo mode has never been activated and a copy-protected plugin is started, the message “Close the program and run the UAD-1 Meter to start the timed demo” appears. If you click Cancel, the plugin interface appears and can be manipulated but audio is not processed by the plugin.

Once the demo mode is activated, the plugin will run without functional limitations for 14 days. Demo mode can only be activated one time. Once the demo period has expired, demo mode cannot be activated again on the same UAD-1 card.

Important: *The 14 day demo period can only be activated once, and cannot be stopped or restarted during this period! We recommend you do not activate demo mode until you have the opportunity to thoroughly check out its sound and functionality.*

Demo Activation

To activate demo mode:

1. Ensure UAD-1 Powered Plug-Ins version 3.1 or higher is already installed and configured properly (see “[Installation](#)” on page 22).
2. Quit all open VST, DirectX, MAS, and Audio Units host applications.
3. Launch the UAD-1 Meter application ([page 29](#)).
4. When the UAD-1 Meter is in the foreground, select “Authorizations...” from the System Menu (Windows; [page 29](#)) or File Menu (Mac OS; [page 30](#)). The Authorizations window ([Figure 16 on page 59](#)) appears.
5. Click the Start Demo button. A confirmation window appears and the timed demo can be activated or demo activation can be cancelled.

The timed demo is activated for 14 days.

Important: *Demo mode relies on the computer system date and time. Manipulating the system clock can result in a decreased demo period.*

Plug-In Authorization Procedure

This section details how to acquire and load the plugin authorization file. Please read the overview (page 56) before proceeding with authorization. The procedure is slightly different under Windows and Mac OS; it is detailed separately per platform.

Note: *It is the UAD-1 hardware that is authorized, not the plugin file.*

Important: *The .reg settings file contains the authorization key for the specific UAD-1 card. It is required every time the card needs to be authorized, such as if it is installed into a different computer or OS. Back it up and keep it in a safe place!*

Authorization Procedure (Windows)

To authorize a UAD-1 card to run a copy protected plugin under Windows:

1. Ensure UAD-1 Powered Plug-Ins version 3.1 or higher is already installed and configured properly (see “Installation” on page 22). Versions before 3.1 will not generate the proper Hardware IDs.
2. Quit all open VST and DirectX host applications.
3. Launch the UAD-1 Meter application (page 28).
4. When the UAD-1 Meter is in the foreground, select “Authorizations...” from the System Menu (page 29). The Authorizations window appears.
5. Click the Copy button within the Hardware IDs window. The Hardware IDs text is copied to the system clipboard.
6. Go to the following URL on the internet:
 - <https://my.uaudio.com/store/>
7. Follow the instructions on the web pages. You can paste the Hardware ID text if it is in the clipboard. The ID text must match EXACTLY.
8. After payment is received, a link to the .reg file is provided.
9. The .reg settings file containing the authorization key for the UAD-1 is then downloaded directly from your account page at the web store.
10. Save the .reg file to disk, then double-click the resulting .reg settings file on the disk. Click “Yes” to automatically update the windows registry. The UAD-1 card is authorized to run the copy protected plugin.
11. Back up the .reg settings file containing the authorization key and keep it in a safe place for future authorizations!

The authorization procedure for Windows is complete.

Authorization Procedure (Mac OS)

To authorize a UAD-1 card to run a copy protected plugin under Mac OS:

1. Ensure UAD-1 Powered Plug-Ins version 3.1 or higher is already installed and configured properly (see “Installation” on page 22). Versions before 3.1 will not generate the proper Hardware IDs.
2. Quit all open VST, MAS, and Audio Units host applications, then launch the UAD-1 Meter application (page 29).
3. When the UAD-1 Meter is in the foreground, select “Authorizations...” from the File Menu (page 30). The Authorizations window appears.
4. Click the Copy button within the Authorizations window. The Hardware IDs text is copied to the system clipboard.
5. Go to the following URL on the internet:
 - <https://my.uaudio.com/store/>
6. Follow the instructions on the web page(s). You can paste the Hardware ID text if it is in the clipboard. The ID text must match EXACTLY.
7. After payment is received, a link to the .reg file is provided. The .reg settings file containing the authorization key for the UAD-1 is then downloaded directly from your account page at the web store.
8. Save the .reg file to disk (control-click the link to view the save menu).
9. Load the resulting .reg settings file using one of the following techniques:
 - Drag and drop the file onto the UAD-1 Meter application icon or its alias (NOT the CPU/Memory bar graph window!).
 - Select “Load Settings...” from the File Menu within the UAD-1 Meter, navigate to the location of the file on disk, then click Open.
10. The UAD-1 card is authorized to run the plugin.
11. Back up the .reg settings file containing the authorization key and keep it in a safe place for future authorizations!

The authorization procedure for Mac OS is complete.

Important: The .reg settings file contains the authorization key for the specific UAD-1 card. It is required every time the card needs to be authorized, such as if it is installed into a different computer or OS. Back it up and keep it in a safe place!

Note: If the UAD Registry file (in OS 9 it is System Folder:Preferences:UAD Registry; in OS X it is Library:Preferences:com.uaudio.uad.plist) is moved or deleted, you will lose your authorization(s) and the .reg authorization key will need to be reloaded.

CHAPTER 4

RealVerb Pro

Overview

RealVerb Pro uses complex spatial and spectral reverberation technology to accurately model an acoustic space. What that gets you is a great sounding reverb with the ability to customize a virtual room and pan within the stereo spectrum.

Room Shape and Material

RealVerb Pro provides two graphic menus each with preset Room Shapes and Materials. You blend the shapes and material composition and adjust the room size according to the demands of your mix. Controls are provided to adjust the thickness of the materials – even inverse thickness for creative effects. Through some very clever engineering, the blending of room shapes, size and materials may be performed in real-time without distortion, pops, clicks or zipper noise. Once you've created your custom room presets, you can even morph between two presets in real-time, with no distortion.

Resonance, Timing and Diffusion

RealVerb Pro also includes intuitive graphic control over equalization, timing and diffusion patterns. To maximize the impact of your recording, we put independent control over the direct path, early reflections and late-field reverberation in your hands.

Stereo Soundfield Panning

Capitalizing on the psychoacoustic technology that went into the design of RealVerb 5.1, we have incorporated some of those principals into RealVerb Pro. Our proprietary Stereo Soundfield Panning allows you to spread and control the signal between stereo speakers creating an impression of center and width. The ability to envelop your listener in a stereo recording is an entirely new approach to reverb design.

Don't rely on your old standby. Let RealVerb Pro bring new quality and space to your recordings!

RealVerb Pro Background

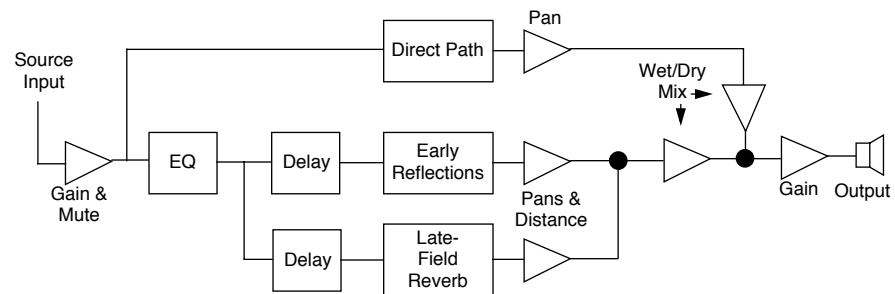


Figure 17. RealVerb Pro signal flow

Figure 17 illustrates the signal flow for RealVerb Pro. The input signal is equalized and applied to the early reflection generator and the late-field reverberation unit. The resulting direct path, early reflection, and late-field reverberation are then independently positioned in the soundfield.



Figure 18. The RealVerb Pro plugin window

The RealVerb Pro user interface is similarly organized (see [Figure 18](#)). Reflected energy equalization is controlled with the Resonance panel. The pattern of early reflections (their relative timing and amplitudes) is determined by the room shapes and sizes in the Shape panel; early reflection predelay and overall energy is specified at the top of the Timing panel. The Material panel is used to select relative late-field decay rates as a function of frequency. The overall late field decay rate is chosen along with the room diffusion, late-field predelay, and late-field level at the bottom of the Timing panel. Finally, the Positioning panel contains controls for the placement of the source, early reflections, and late-field reverberation.

Spectral Characteristics

The Shape and Material panels specify the room shape, room size, room material and thickness. These room properties affect the spectral characteristics of the room's reflections.

Shape and Size

The pattern of early reflections in a reverb is determined by the room shape and size. RealVerb Pro lets you specify two room shapes and sizes that can be blended to create a hybrid of early reflection patterns. There are 15 room shapes available, including several plates, springs, and classic rooms; room sizes can be adjusted from 1–99 meters. The two rooms can be blended from 0–100%. All parameters can be adjusted dynamically in real time without causing distortion or other artifacts in the audio.

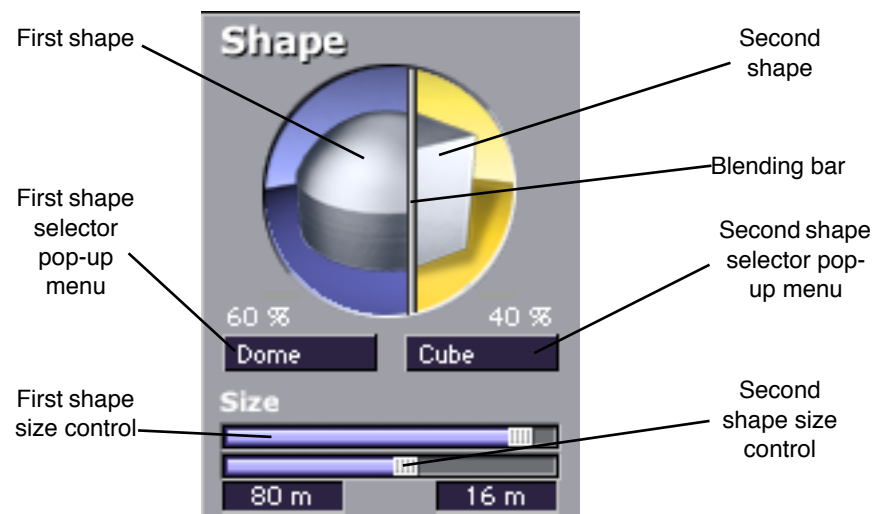


Figure 19. RealVerb Pro Shape panel

To configure the room shape and size:

1. Select a room shape from the first (left) pop-up menu. The selected shape appears in the left side of the Shape circle. Adjust the room size with the top horizontal slider.
2. Select a room shape from the second (right) pop-up menu. The selected shape appears in the right side of the Shape circle. Adjust the room size with the bottom horizontal slider.
3. Blend the early reflection patterns of the two rooms by dragging the Blending bar. The relative percentages of the two rooms appear above their pop-up menus. Drag to the right to emphasize the first room shape; drag to the left to emphasize the second room shape. To use only one room shape, drag the Blending bar so the shape is set to 100%.

The resulting early reflection pattern is displayed at the top of the Timing panel (see [Figure 22 on page 73](#)), where each reflection is represented by a yellow vertical line with a height indicating its arrival energy, and a location indicating its arrival time.

Material and Thickness

The material composition of an acoustical space affects how different frequency components decay over time. Materials are characterized by their absorption rates as a function of frequency—the more the material absorbs a certain frequency, the faster that frequency decays. RealVerb Pro lets you specify two room materials with independent thicknesses, which can be blended to create a hybrid of absorption and reflection properties. For example, to simulate a large glass house, a blend of glass and air could be used.

There are 24 real-world materials provided, including such diverse materials as brick, marble, hardwood, water surface, air, and audience. Also included are 12 artificial materials with predefined decay rates. The thickness of the materials can be adjusted to exaggerate or invert their absorption and reflection properties. For a description of the different room materials, see [“About the Materials” on page 69](#).

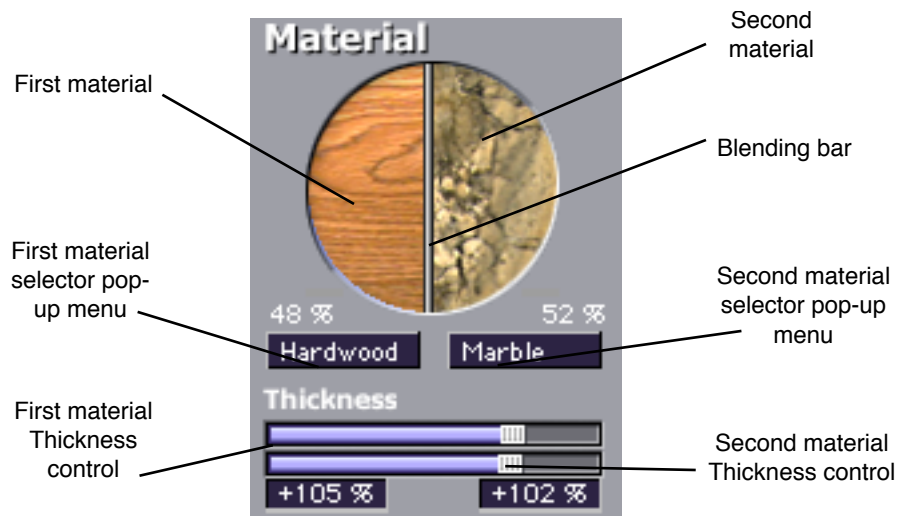


Figure 20. RealVerb Pro Material panel

Note: While materials are used to control decay rates as a function of frequency, the overall decay rate of the late-field reverberation is controlled from the Timing panel (see [Figure 22 on page 73](#)).

To configure the room material and thickness:

1. Select a room material from the first (left) pop-up menu. The selected material appears in the left side of the Material circle.
2. Adjust the thickness for the first material with the top horizontal slider:
 - A default thickness of +100% yields normal, real-world decays for the material.
 - Thicknesses beyond the default (up to +200%) exaggerate how the frequencies are absorbed and reflected.
 - Negative thicknesses invert the response of the material. If the material normally absorbs high frequencies (causing them to decay quickly) and reflects low frequencies (causing them to decay slowly), a negative thickness will instead absorb low frequencies (causing them to decay quickly) and reflect high frequencies (causing them to decay slowly).
 - A thickness of 0% yields decay rates that are not affected by the material.
3. Select a material from the second (right) pop-up menu. The selected material appears in the right side of the Material circle. Adjust the material thickness with the bottom horizontal slider.

4. Blend the absorption properties of the two materials by dragging the Blending bar. The relative amount of each material, expressed as a percentage, appears above their respective pop-up menu. Drag the Blending bar to the right to emphasize the first material, and drag it to the left to emphasize the second material. To use only one room material, drag the Blending bar so the material is set to 100%.

About the Materials

Some materials absorb high frequencies and reflect low frequencies, while other materials absorb low frequencies and reflect high frequencies. This characteristic is determined by the material surface and density.

Fiberglass, for example, absorbs high frequencies. When high frequencies strike fiberglass they bounce around inside the fibers and lose much of their energy.

At a thickness of 100%, fiberglass rolls off the high frequencies, a little bit each millisecond. After a while the high frequencies dissipate and the low frequencies linger. If we were to take fiberglass and increase its thickness to +200%, the high frequencies would roll off even faster. At +200%, this high frequency decay happens at twice its normal rate, producing a very heavy reverberant tail. At -200%, a very “sizzly” late field is created.

Some materials, such as plywood, naturally absorb low frequencies while reflecting high frequencies. Since plywood is usually very flat with little surface texture to capture high frequencies, high frequencies tend to be reflected. At +100%, the reverberation produced is very sizzly and increasingly bright. At -100%, it is very heavy.

Keeping this in mind, if you look at the graphics in the material control panel, you can get a sense of how chosen materials, material blend, and thickness will affect the decay rate as a function of frequency. Hard materials that have lots of small cavities (Brick, Gravel, Plaster on Brick) and soft materials (Carpet, Grass, Soil) tend to absorb high frequencies. Flat, somewhat flexible materials (Heavy Plate Glass, Hardwood, Seats) tend to reflect high frequencies. Marble is the one material that tends to uniformly reflect all frequencies.

You probably noticed the artificial materials the top of the Materials menu. These are materials designed to have predictable behavior and can be very handy for achieving a desired reverberation preset when you know what decay rates you desire. All these materials preferentially absorb high frequencies; they give the selected decay time at low frequencies, and a much shorter decay time at high frequencies. The frequency in each graphic is the transi-

tion frequency, the frequency at which the decay rate is halfway between the low-frequency and high-frequency values. At 100% thickness, the ratio of low-frequency to high-frequency decay times is 10:1. This means that the high frequencies will decay 10 times faster than the low frequencies. At 200% thickness, this is multiplied by two (high frequencies decay at 20x the rate of the low frequencies). At negative 100%, the sense of low frequency and high frequency is swapped —low frequencies decay 10 times faster than the high frequencies.

Many hardware and software reverbs tend to compensate for the high frequency absorption that air provides. RealVerb Pro instead provides “Air” as a material. If you do not choose to use Air as one of the materials, you can effectively compensate for the high frequency absorption properties of air with the Resonance filters. Set the right-hand Transition Frequency slider to 4.794 kHz, and bring the level down about –10 dB to –15 dB for large to huge rooms, and down about –4 dB to –9 dB for small to medium rooms.

To help you out, the following lists classify the materials under two headings: those that tend to reflect high frequencies, and those that tend to absorb them. They are listed in order of their transition frequencies, from lowest to highest.

Table 4. Materials with high-frequency absorption

Audience	Fiberglass
Cellulose	Grass
Drapery	Plaster on Brick
Plaster on Concrete Block	Water Surface
Soil	Sand
Gravel	Brick
Paint on Concrete Block	Air
Carpet	

Table 5. Materials with high-frequency reflection

Heavy Plate Glass	Seats
Plywood	Marble
Hardwood	Concrete Block
Glass Window	Linoleum
Cork	

Resonance (Equalization)

The Resonance panel has a three-band parametric equalizer that can control the overall frequency response of the reverb, affecting its perceived brilliance and warmth. By adjusting its Amplitude and Band-edge controls, the equalizer can be configured as shelf or parametric EQs, as well as hybrids between the two.

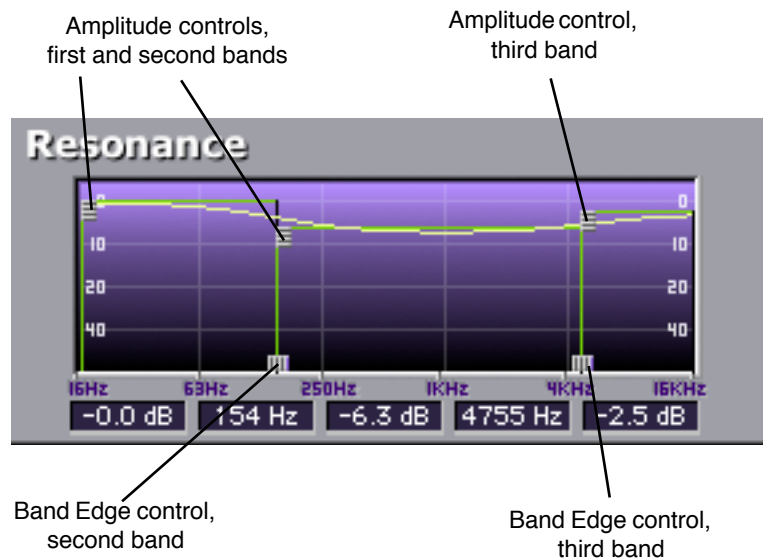


Figure 21. RealVerb Pro Resonance panel

To configure the reverb's Resonance as a parametric EQ:

1. Drag the Band Edge controls horizontally for the second and third bands to the desired frequencies. The first band is preset to 16 Hz. The frequencies for all three bands are indicated in the text fields at the bottom of the Resonance panel.
2. Adjust the amplitude of the bands (from -60 dB to 0 dB) by dragging their Amplitude controls either up or down. The amplitude values for all three bands are indicated in the text fields at the bottom of the Resonance panel. The shape of the EQ curve is displayed in the Resonance graph.

To configure the reverb's Resonance as a high-shelf EQ:

1. Drag the Amplitude control for the second EQ band all the way down.
2. Drag the Amplitude controls for the first and third bands all the way up, to equal values.

3. Adjust the Band-edge controls for the second and third bands so they are adjacent to each other. To raise the frequency for the high-shelf, drag to the right with the Band-edge control for the second band. To lower the frequency for the high-shelf, drag to the left with the Band-edge control for the third band.
4. To attenuate the frequencies above the shelf frequency, drag the Amplitude controls for the first and second bands up or down. For a true shelf EQ, make sure these amplitudes are set to equal values.

To configure the reverb's Resonance as a low-shelf EQ:

1. Drag the Amplitude control for the second EQ band all the way up.
2. Drag the Amplitude controls for the first and third bands all the way down, to equal values.
3. Adjust the Band-edge controls for the second and third bands so they are adjacent to each other. To raise the frequency for the low-shelf, drag to the right with the Band-edge control for the second band. To lower the frequency for the low-shelf, drag to the left with the Band-edge control for the third band.
4. To attenuate the frequencies below the shelf frequency, drag the Amplitude controls for the first and second bands up or down. For a true shelf EQ, make sure these amplitudes are set to equal values.

Timing

The Timing panel offers control over the timing and relative energies of the early reflections and late-field reverberations. These elements affect the reverb's perceived clarity and intimacy. The early reflections are displayed at the top of the Timing panel, with controls for Amplitude and Pre-delay. The late-field reverberations are displayed at the bottom, with controls for Amplitude, Pre-delay, and Decay Time. To illustrate the relation between both reverb components, the shape of the other is represented as an outline in both sections of the Timing panel (see [Figure 22](#)).

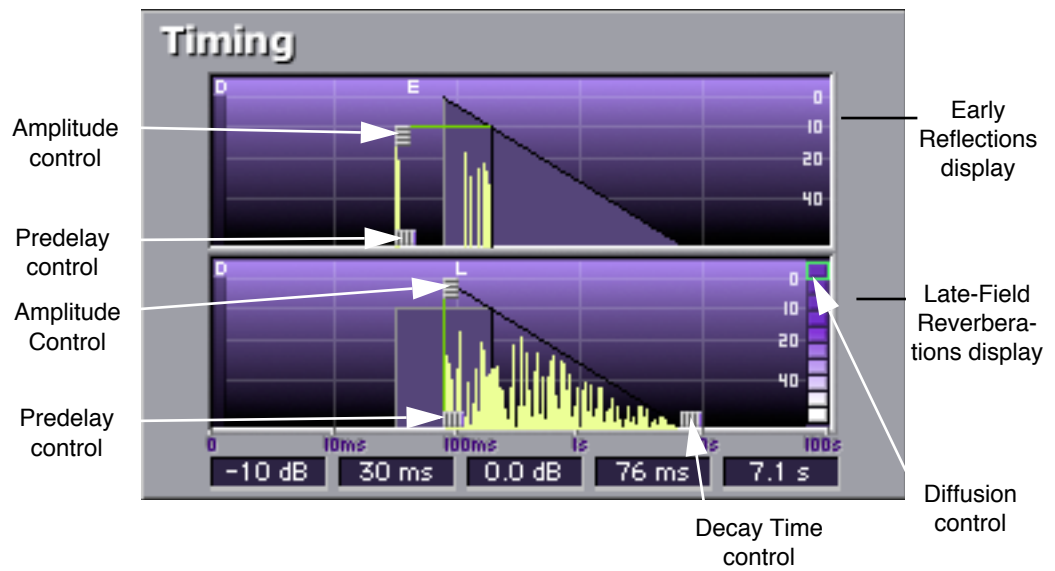


Figure 22. RealVerb Pro Timing panel

To adjust the timing of the early reflections:

1. Drag the Amplitude control for the early reflections up or down (from -80 dB to 0 db) to affect the energy of the reflections. The Amplitude value is indicated in the text field at the bottom of the Timing panel.
2. Drag the Predelay control for the early reflections left or right (from 1–300 milliseconds) to affect the delay between the dry signal and the onset of early reflections. The Pre-delay time is indicated in the text field at the bottom of the Timing panel.

Note: The length in time of the early reflections cannot be adjusted from the Timing panel, and instead is determined by the reverb's shape and size (see Figure 19).

To adjust the timing of the late-field reverberations:

1. Drag the Amplitude control for the late-field reverberations up or down (from -80 dB to 0 db) to affect the energy of the reverberations. The Amplitude value is indicated in the text field at the bottom of the Timing panel.
2. Drag the Predelay control for the late-field reverberations left or right (from 1–300 milliseconds) to affect the delay between the dry signal and the onset of late-field reverberations. The Predelay time is indicated in the text field at the bottom of the Timing panel.

3. Drag the Decay Time control for the late-field reverberations left or right (from 0.10–96.00 seconds) to affect the length of the reverb tail. The Decay Time is indicated in the text field at the bottom of the Timing panel.
4. To affect how quickly the late-field reverberations become more dense, adjust the Diffusion control at the right of Late Reflection display in the Timing panel. The higher the Diffusion value (near the top of the display), the more rapidly a dense reverb tail evolves.

Positioning

One of the unique features of RealVerb Pro is the ability to separately position the direct path, early reflections, and late-field reverberation. The Position panel (see [Figure 23](#)) provides panning controls for each of these reverb components. In addition, a proprietary Distance control adjusts perceived source distance. These controls allow realistic synthesis of acoustic spaces—for instance listening at the entrance of an alley way, where all response components arrive from the same direction, or listening in the same alley next to the source, where the early reflections and reverberation surround the listener.



Figure 23. RealVerb Pro Positioning panel

To pan the direct (dry) signal:

1. Drag the Direct slider left or right. A value of <100 pans the signal hard left; a value of 100> pans the signal hard right. A value of <0> places the signal in the center of the stereo field.

Set the positioning for the early reflection or late-field reverberation with any of the following methods:

1. Drag the left and right slider handles to adjust the stereo width. The length of the blue slider is adjusted. For a full stereo signal, drag the left handle all the way to left, and right handle all the way to the right.
2. Drag the blue center of the slider left or right to set the positioning of the signal. If you drag all the way to the left or right, the stereo width is adjusted. For a mono signal panned hard left or right, drag the slider all the way to the left or right.

Distance

RealVerb Pro allows you to control the distance of the perceived source with the Distance control in the Positioning panel (see [Figure 23](#)). In reverberant environments, sounds originating close to the listener have a different mix of direct and reflected energy than those originating further from the listener.

To adjust the distance of the source:

1. Drag the Distance slider to the desired percentage value. Larger percentages yield a source that is further away from the listener. A value of 0% places the source as close as possible to the listener.

Wet/Dry Mix

The wet and dry mix of the reverb is controlled from the Mix slider in the Positioning panel (see [Figure 23](#)). The two buttons above this slider labeled “D” and “W” represent Dry and Wet; clicking either will create a 100% dry or 100% wet mix.



Levels

The Levels panel lets you adjust the Input Gain and Output Gain for RealVerb Pro. These levels are adjusted by dragging the sliders to the desired values. You can mute the input signal by clicking the Mute button.

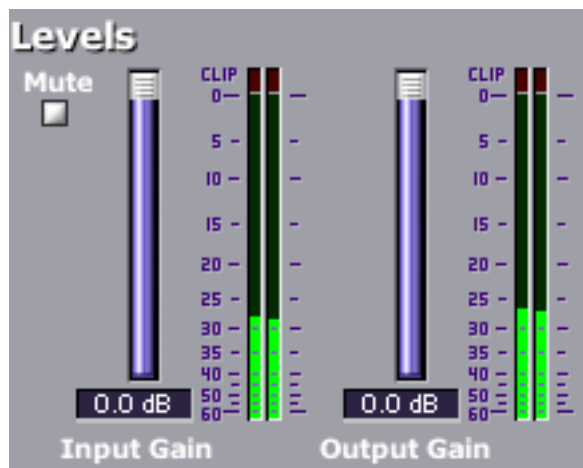


Figure 24. RealVerb Pro Levels panel

Morphing

All RealVerb Pro controls vary continuously using proprietary technology to smoothly transition between selected values. This capability enables RealVerb Pro to morph among presets by transitioning between their parameter sets. This approach is in contrast to the traditional method of morphing by cross-fading between the output of two static reverberators. The method employed by RealVerb Pro produces more faithful, physically meaningful intermediate states.

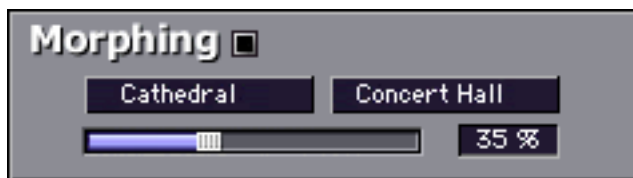


Figure 25. RealVerb Pro Morphing panel

Figure 25 depicts the Morphing Panel. Click the Morphing Mode button to enable Morphing mode. When RealVerb Pro is in morphing mode, the other RealVerb Pro spectral controls are grayed out and cannot be edited. In morphing mode, two presets are selected using the pull-down menus. Once the desired presets are selected in the pull-down menus, the morphing slider is used to morph from one preset to the other.

When in Morphing mode, non user-adjustable controls will change their appearance and will no longer be accessible. When inserted on a Send effect, the 'W' button automatically turns on (to keep the mix at 100% wet).

On an insert effect, the Mix will change back and forth between the two mix values of each preset.



Figure 26. RealVerb Pro in Morphing mode



RealVerb Pro Preset Management

Factory Presets

In the preset menu there are thirty factory presets that can be changed by the user. Any modification to a preset will be saved even if you change presets. If you want to return all the presets to their default settings, select “Reset all to Defaults” at the bottom of the presets menu.

Edits to any and all presets in the list are maintained separately within each instance of a plugin in a session.

Using Host Application Management

Most host applications include their own method of managing plugin presets.

For example, the currently selected preset is saved in Cubase/Nuendo when “Save Effect” is used. Morphing parameters and the solo/mute buttons (wet, dry, input) are not saved.

All presets and programs are saved in Cubase/Nuendo when “Save Bank” is used. They are also saved in the session file for each instance of the plugin.

Editing the name in Cubase/Nuendo modifies the current preset's name. The new name will appear in all preset select lists, and will be saved with the session, bank or effect.

RealVerb Pro Preset List

Table 6. RealVerb Pro Presets

Acoustic Guitar	Hairy Snare
Apartment Living	High Ceiling Room
Big Ambience	Jazz Club
Big Bright Hall	Large Bathroom
Big Cement Room	Large Dark Hall
Big Empty Stadium	Long Tube
Big Snare	Medium Drum Room
Big Warm Hall	Nice Vocal 1
Cathedral	Nice Vocal 2
Church	Slap Back
Dark Ambience	Small Bright Room
Drums in a Vat	Small Dark Room
Eternity	Sparkling Hall
Far Away Source	Tight Spaces
Ghost Voice	Wooden Hall

CHAPTER 5

LA-2A and 1176LN

Overview

The LA-2A and 1176LN compressor/limiters long ago achieved classic status. They're a given in almost any studio in the world - relied upon daily by engineers whose styles range from rock to rap, classical to country and everything in between. With so many newer products on the market to choose from, it's worth looking at the reasons why these classics remain a necessary part of any professional studio's outboard equipment collection.

The basic concept of a compressor/limiter, is of course, relatively simple. It's a device in which the gain of a circuit is automatically adjusted using a predetermined ratio that acts in response to the input signal level. A compressor/limiter "rides gain" like a recording engineer does by hand with the fader of a console: it keeps the volume up during softer sections and brings it down when the signal gets louder. The dynamic processing that occurs at ratios below 10 or 12 to one is generally referred to as compression; above that it's known as limiting.

Modern day compressors offer a great degree of programmability and flexibility; older devices such as the 1176LN and the LA-2A are more straightforward in their design. Perhaps it is this fact that has contributed to their appealing sound and the longevity of their popularity.

Compressor Basics

Before discussing the LA-2A and 1176LN plugins, this section will cover some compressor basics. A *compressor* automatically adjusts the gain of a signal by a predetermined ratio. In a sense, a compressor "rides" gain—much like a recording engineer does (by hand) with a fader—keeping the volume up during softer sections and bringing it down when the signal gets louder.

Figure 27 depicts the input and output characteristics of a compressor and perfect amplifier. When operated within its specified range, an amplifier provides a constant amount of gain regardless of the input signal level. In Figure 27, the signal level of a perfect amplifier is represented with a constant output gain of 10 dB. In this example, a signal with an input level of -30 dB results in an output level of -20 dB, which is an increase of 10 dB. Similarly, an input level of 0 dB results in an output level of 10 dB (the gain stays fixed at 10 dB regardless of the input level).

In contrast to an amplifier, whose function is to present a constant gain, a compressor varies its gain in response to the level of the input signal. Large input signals result in less gain, thus reducing or *compressing* the dynamic range of the signal. In Figure 27, a compressed signal with an input level of -30 dB results in an output level of -20 dB, indicating a gain of 10 dB. However, with input levels of -20 dB and -10 dB, the compressor exhibits gains of 5 dB and 0 dB (respectively), thereby illustrating that the gain decreases as the input signal increases. This increase in output level by 5 dB for every 10 dB is defined as a compression ratio of 2:1 (reduced from 10:5).

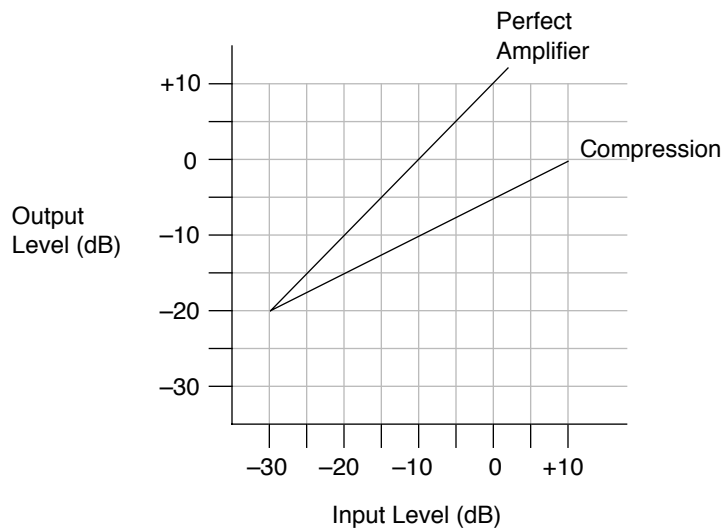


Figure 27. Input and output characteristics of a compressor and perfect amplifier

The amount of compression, or gain reduction, typically expressed in decibels (dB), is defined as the amount by which the signal level is reduced by the compressor. Graphically, this can be represented (see Figure 28) by the difference in output levels between the original signal (without compression) and the compressed signal. The LA-2A and 1176LN display this value when their VU Meters are set to Gain Reduction.

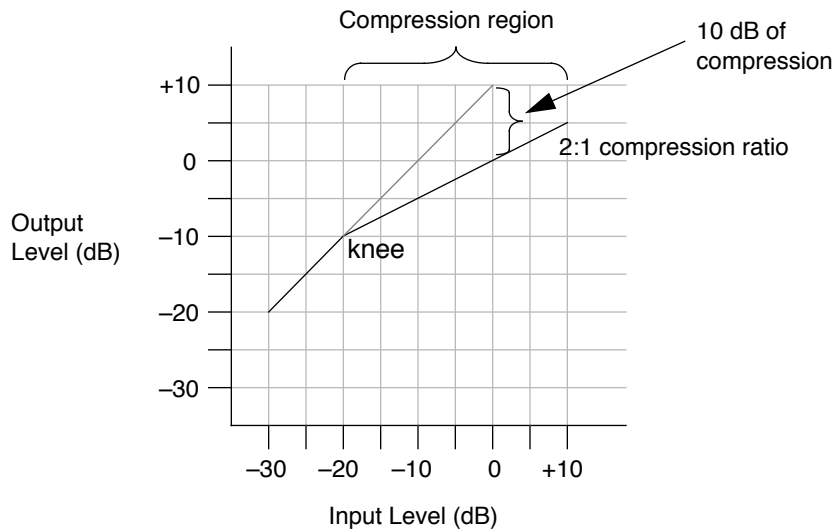


Figure 28. Input and output curve of compressor with 2:1 ratio and -20 dB threshold

As mentioned previously, the compression ratio is defined as the ratio of the increase of the level of the input signal to the increase in the level of the output signal. In [Figure 28](#), the input level is increased by 10 dB while the output level increases 5 dB. This is a compression ratio of 2:1. Lower compression ratios such as 2:1 result in mild compression. A compression ratio of 1:1 yields no compression.

Note: Compression ratios above 10:1 are commonly referred to as “limiting” or “peak-limiting,” where amplitude peaks are reduced.

Compressors often let you set a threshold, the point at which gain reduction starts to take place. When the level of an audio signal is below this threshold there is no gain reduction. As the level of the signal increases above the threshold level, gain reduction and compression occurs. The point at which a signal transitions into compression is commonly referred to as the *knee*. In practical compressors, this transition is more gentle than what is depicted in [Figure 28](#).

Most modern compressors provide a control that adjusts the threshold directly. In the case of the LA-2A, the Peak Reduction control adjusts both the threshold and the amount of gain reduction. Similarly, the 1176LN uses its Input control to adjust the threshold and amount of gain reduction.

Teletronix LA-2A Leveling Amplifier

Background

Audio professionals passionate about their compressors revere the LA-2A. The original was immediately acknowledged for its natural compression characteristics. A unique electro-optical attenuator system allows instantaneous gain reduction with no increase in harmonic distortion – an accomplishment at the time, still appreciated today.

The LA-2A is known for adding warmth (such as for vocals, guitar, or synths) and fatness (such as for drums or bass) to signals.

LA-2A Signal Flow

A functional block diagram of the LA-2A Leveling Amplifier is provided in [Figure 29](#). The input transformer provides isolation and impedance matching. After this the signal is fed into both the side-chain circuit and the gain reduction circuit. The side-chain is comprised of a voltage amplifier, a pre-emphasis filter, and a driver stage that provides the voltage necessary to drive the electro-luminescent panel. This signal controls the gain of the compressor. After the gain reduction circuit, the signal is sent through an Output Gain control and a two-stage output amplifier, followed by the output transformer.

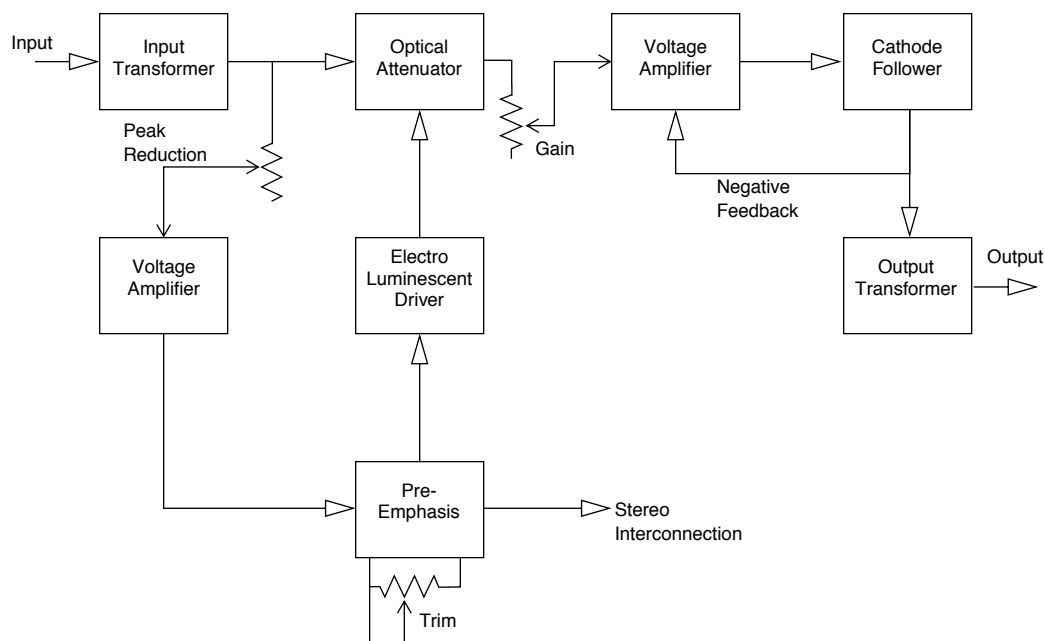


Figure 29. LA-2A signal flow

LA-2A Controls



Figure 30. The LA-2A plugin window.

- Limit/Compress** Changes the characteristics of the compressor I/O curve. When set to Compress, the curve is more gentle, and presents a low compression ratio. When set to Limit, a higher compression ratio is used.
- Gain** Adjusts the output level (by up to 40 dB). Make sure to adjust the Gain control *after* the desired amount of compression is achieved with the Peak Reduction control. The Gain control does not affect the amount of compression.
- Peak Reduction** Adjusts the amount of gain reduction, as well as the relative threshold. A Peak Reduction value of 0 yields no compression. Rotate this control clockwise until the desired amount of compression is achieved (to monitor the Peak Reduction, set the VU Meter to Gain Reduction). The Peak Reduction should be adjusted independently of the Gain control.
- Meter** This knob (in the upper right) sets the mode of the VU Meter. When set to Gain Reduction, the VU Meter indicates the Gain Reduction level in dB. When set to +10 or +4, the VU Meter indicates the output level in dB.
- On/Power Switch** Determines whether the LA-2A plugin is active. When the Power switch is in the Off position, the plugin is disabled and UAD-1 DSP usage is reduced.
- Stereo Operation** Phase-coherent stereo imaging is maintained when the LA-2A plugin is used on a stereo signal.

1176LN Solid-State Limiting Amplifier

The 1176LN is known for bringing out the presence and color of audio signals, adding brightness and clarity to vocals, and “bite” to drums and guitar.

1176LN Signal Flow

A functional block diagram of the 1176LN Limiting Amplifier is provided in [Figure 31](#). Signal limiting and compression is performed by the Gain Reduction section. Before the signal is applied to the Gain Reduction section, the audio signal is attenuated by the Input stage. The amount of attenuation is controlled by the input control potentiometer. The amount of gain reduction as well as the compressor Attack and Release times are controlled by Gain Reduction Control circuit. After Gain Reduction a pre-amp is used to increase the signal level. The Output Control potentiometer is then used to control the amount of drive that is applied to the output amplifier. The 1176LN is a feedback style compressor since the signal level is sensed after the gain reduction is applied to the signal.

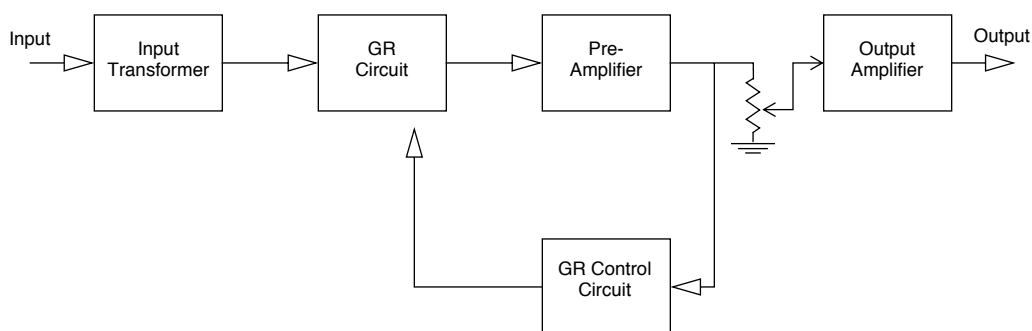


Figure 31. 1176LN signal flow



1176LN Controls



Figure 32. The 1176LN plugin window

Input

Adjusts the amount of gain reduction as well as the relative threshold. An Input value of ∞ (turned fully counterclockwise) yields no compression (and no signal level). Rotate this control clockwise to increase the amount of compression.

Output

Adjusts the output level (by up to 45 dB). Make sure to adjust the Output control *after* the desired amount of compression is achieved with the Input and Attack controls. To monitor the Output level, set the VU Meter to +8 or +4. The Output control does not affect the amount of compression.

Attack

Sets the amount of time (from 20–800 microseconds) that must elapse once the input signal reaches the Threshold level before compression is applied. Faster attack times are achieved by rotating the Attack control clockwise. The faster the Attack, the more rapidly compression is applied to signals above the threshold.

Release

Sets the amount of time (from 50–1100 msec.) it takes for compression to cease once the input signal drops below the threshold level. Faster release times are achieved by rotating the Release control clockwise. Slower release times can smooth the transition that occurs when the signal dips below the threshold, especially useful for material with frequent peaks. However, if you set too large of a Release time, compression for sections of audio with loud signals may extend to lengthy sections of audio with lower signals.

Ratio

These four pushbutton switches (to the left of the VU Meter) determine the compression ratio. Ratios of 20:1, 12:1, 8:1, and 4:1 are provided. The 20:1 and 12:1 settings are typically used when peak-limiting is desired, while the 4:1 and 8:1 settings are used for general dynamic range compression.

All Buttons mode

Just like the hardware version of the 1176LN, it is possible to depress all the Ratio buttons simultaneously, a well-known studio trick.

In this mode, the ratio is around 12:1, and the release happens faster, and the shape of the release curve changes. With lower amounts of compression, the attack is delayed slightly, as there is a slight lag before the attack attenuated the signal. That attack value remains at whatever the value is on the Attack control.

To enter All Button Mode

Shift-click any of the Ratio buttons. All of the buttons will appear depressed.

To exit All Button Mode

Click any Ratio button without the shift key modifier.

Meter

These four pushbutton switches (to the right of the VU Meter) determine the mode of the VU Meter, and whether the plugin is enabled. When set to GR, the VU Meter indicates the Gain Reduction level in dB. When set to +8 or +4, the VU Meter indicates the output level in dB; when set to +4, a meter reading of 0 corresponds to an output level of +4 dB.

In gain reduction mode with all buttons depressed, the VU meter will appear to behave strangely. This is normal behavior in the hardware 1176LN, and is faithfully recreated in the plugin.

When the Meter Off switch is selected, the 1176LN plugin is disabled and UAD-1 DSP usage is reduced.

Grit

One trick you can do with the 1176 is turning the attack and release up all the way to their fastest setting. This has the audible effect of adding distortion to the audio source, and is especially pronounced in all-buttons mode. What happens here is the attack and release are happening so fast that minute level fluctuations sound like distortion. It can add a very useful, gritty compression effect.

This effect is useful on bass, where you might need compression and distortion at the same time, and the 1176 can provide both in a unique way. This trick also sounds great on screaming lead vocals. And yes, the hardware does this too!

Stereo Operation

Phase-coherent stereo imaging is maintained when the 1176LN plugin is used on a stereo signal.

1176SE “Special Edition”



Figure 33. The 1176SE plugin window

Overview

The 1176SE is derived from the 1176LN. Its algorithm has been revised in order to provide sonic characteristics similar to the 1176LN but with significantly less DSP usage. It is provided to allow “1176LN-like sound” when DSP resources are limited.

The 1176SE behavior is practically identical to the 1176LN. Its sound is nearly identical too, but certain compromises had to be made in order to squeeze the extra DSP performance that the 1176SE provides. At nominal settings the sonic difference is negligible. At extreme (cranked) settings, nobody with “golden ears” will say it sounds exactly like the 1176LN, but it still sounds great and is very usable in most situations.

1176SE Controls

The 1176SE controls are exactly the same as the 1176LN. Please refer to the the 1176LN section for 1176SE control descriptions (see [“1176LN Controls” on page 85.](#)).



POWERED PLUG INS

CHAPTER 6

CS-1 Channel Strip

Overview

The CS-1 Channel Strip provides the EX-1 Equalizer and Compressor, DM-1 Delay Modulator, and RS-1 Reflection Engine combined into one plugin. Individual effects in the CS-1 Channel Strip can be bypassed when not in use to preserve UAD-1 CPU use.

The CS-1 effects can also be accessed individually by using the individual plugins. This is useful if you want to use the plugins in a different order, or if you want to use multiple instances of the same plugin (such as a flange routed to a ping-pong delay with the DM-1 plugin).



Figure 34. The CS-1 Channel Strip plugin window

EX-1 Equalizer and Compressor



Figure 35. The EX-1 EQ/Compressor plugin window

The EX-1 plugin consists of a five-band parametric EQ and compressor.

EX-1 Equalizer Controls

The Equalizer portion of the EX-1 is a five-band fully parametric EQ. Each band has its own set of controls. The first two bands can also be enabled to function as low-shelf or high-pass filter. Similarly, the last two bands can be enabled to function as either a high-shelf or low-pass filter.

Band Disable Button

Each band can be individually deactivated with the Band Disable button. All bands default to enabled (brighter blue). To disable any band, click the Disable button. The button is darker blue when the band is disabled.

You can use these buttons to compare the band settings to that of the original signal, or to bypass the individual band.

Gain (G) Knob	The Gain control determines the amount by which the frequency setting is boosted or attenuated. The available range is ± 18 dB.
Frequency (fc) Knob	Determines the center frequency to be boosted or attenuated by the Gain setting. The available range is 20 Hertz to 20 kiloHertz. When operating at sample rates less than 44.1 kHz, the maximum frequency will be limited.
Bandwidth (Q) Knob	<p>Sets the proportion of frequencies surrounding the center frequency to be affected. The Bandwidth range is 0.03–32; higher values yield sharper bands.</p> <p>In either of the first two bands, when the Bandwidth value is at minimum the band becomes a low-shelf filter, and at maximum the band becomes a high-pass filter.</p> <p>Similarly, in either of the last two bands, when the Bandwidth value is at minimum the band becomes a high-shelf filter, and at maximum the band becomes a low-pass filter.</p>
Enable/Bypass Switch	Globally enables or disables all bands of the Equalizer. You can use this switch to compare the EQ settings to that of the original signal or bypass the entire EQ section to reduce UAD-1 DSP load.
Output Knob	Adjusts the signal output level of the plugin. This may be necessary if the signal is dramatically boosted or reduced by the EQ and/or compressor settings.

EX-1 Compressor Controls

Attack Knob	Sets the amount of time that must elapse, once the input signal reaches the Threshold level, before compression will occur. The faster the Attack, the more rapidly compression is applied to signals above the Threshold. The range is 0.05 milliseconds to 100.00 milliseconds.
Release Knob	Sets the amount of time it takes for compression to cease once the input signal drops below the Threshold level. Slower release times can smooth the transition that occurs when the signal dips below the threshold, especially useful for material with frequent peaks. However, if you set too large of a Release time, compression for sections of audio with loud signals may extend to lengthy sections of audio with lower signals. The range is 25 milliseconds to 2500 milliseconds (2.5 seconds).

Ratio Knob	Determines the amount of gain reduction used by the compression. For example, a value of 2 (expressed as a 2:1 ratio) reduces the signal by half, with an input signal of 20 dB being reduced to 10 dB. A value of 1 yields no compression. Values beyond 10 yield a limiting effect. The range is 1 to Infinity.
Threshold Knob	<p>Sets the threshold level for the compression. Any signals that exceed this level are compressed. Signals below the level are unaffected. A Threshold of 0dB yields no compression. The range is 0dB to -60dB.</p> <p>As the Threshold control is increased and more compression occurs, output level is typically reduced. However, the EX-1 provides an auto-makeup gain function to automatically compensate for reduced levels. Adjust the Output level control if more gain is desired.</p>
Meter Pop-up Menu	Determines whether the VU Meter monitors the Input Level, Output Level, Gain Reduction, or Meter Off. Click the menu above the meter display to select a different metering function.
Enable/Bypass Switch	Enables or disables the Compressor. You can use this switch to compare the compressor settings to that of the original signal or bypass the entire compressor section to reduce UAD-1 DSP load.
Compressor Output Knob	Adjusts the relative output of the plugin.

EX-1M Overview

The EX-1M is a monophonic version of EX-1 that enables independent left and right EQ settings in master effects chains and allows Logic Audio users to conserve UAD-1 DSP resources.

EX-1M requires half the processing power compared to that of EX-1 when used on a mono audio track within Logic Audio. Therefore, EX-1M should be used on monophonic audio tracks within Logic whenever possible to conserve UAD-1 resources.

DM-1 Delay Modulator



Figure 36. The DM-1 Delay Modulator plugin window

The DM-1 Delay Modulator provides stereo effects for delay, chorus, and flange.

DM-1 Controls

L-Delay Knob

Sets the delay time between the original signal and the delayed signal for the left channel. When the Mode is set to one of the delay settings, the maximum delay is 300 msec. When the Mode is set to one of the chorus or flange settings, the maximum delay is 125 msec.

R-Delay Knob

Sets the delay time between the original signal and the delayed signal for the right channel. When the Mode is set to one of the delay settings, the maximum delay is 300 msec. When the Mode is set to one of the chorus or flange settings, the maximum delay is 125 msec.

In the Flanger modes, the L and R delay controls have a slightly different functions than in the chorus modes. The high peak of the flanger is controlled by the settings of the L and R delay controls. The low Peak of the flanger is determined by the setting of the Depth control.

When delay times longer than 300ms are desired, use the DM-1L plugin instead. DM-1L has a maximum time of 2400ms per channel.

Mode Pop-up Menu

Determines the DM-1 effect mode. The available modes are: Chorus, Chorus180, QuadChorus, Flanger1, Flanger2, Dual Delay, and Ping Pong Delay. In addition to reconfiguring the DM-1's settings, the Mode also determines the available parameter ranges for L/R Delay and Depth.

In Chorus mode, both oscillators (or modulating signals) are in phase.

In Chorus 180 mode, both oscillators (the modulating signals) are 180 degrees out of phase.

In QuadChorus mode, both oscillators (the modulating signals) are 90 degrees out of phase.

In Ping Pong delay mode, you will only get a ping-pong effect if you have a mono source feeding the DM-1 on a stereo group track or send effect. On a mono disk track, it works exactly like Dual Delay.

Rate Knob

Sets the modulation rate for the delayed signal, expressed in Hertz.

Depth Knob

Sets the modulation depth for the delayed signal, expressed as a percentage.

In Dual Delay and Ping Pong Delay modes, adjusting the Depth and Rate controls can offer some very otherworldly sounds.

LFO Type Pop-up Menu

Determines the LFO (low frequency oscillator) waveshape and phase used to modulate the delayed signal. The waveshape can be set to triangle or sine, each with a phase value of 0, 90, or 180-degrees.

Recirculation (RECIR) Knob

Sets the amount of processed signal fed back into its input. Higher values increase the number of delays and intensity of the processed signal.

Recirculation allows both positive and negative values. The polarity refers to the phase of the delays as compared to the original signal. If Recirculation displays a positive value, all the delays will be in phase with the source. If it displays a negative value, then the phase of the delays flips back and forth between in phase and out of phase.

In the flanger mode, Recir has the potential to make some very interesting sounds. Try turning RECIR fully clockwise or counter-clockwise, and set the delay to very short but different values.

The RECIR units are expressed as a percentage in all Modes except Dual Delay and Ping Pong. In these modes, RECIR values are expressed as T60 time, or the time before the signal drops 60 decibels.

Damping Knob

This low pass filter reduces the amount of high frequencies in the signal. Turn down this control to reduce the brightness. Higher values yield a brighter signal. Damping also mimics air absorption, or high frequency rolloff inherent in tape-based delay systems.

Wet/Dry Mix Knob

This control determines the balance between the delayed and original signal. Values greater than 50% emphasize the wet signal, and values less than 50% emphasize the dry signal. A value of 50% delivers equal signals. A value of 0% is just the dry signal.

Wet/Dry Mix allows both positive and negative values. The polarity refers to the phase of the delays as compared to the original signal. If a positive value is displayed, then all the delays will be in phase with the source. With a negative value, the delayed signal is flipped 180 degrees out of phase with the source.

L-Pan Knob

Sets the stereo position for the left channel, allowing you to adjust the width or balance of the stereo signal. For a mono signal, L-Pan behaves as the level control for the left delay tap.

R-Pan Knob

Sets the stereo position for the right channel, allowing you to adjust the width or balance of the stereo signal. For a mono signal, R-Pan behaves as the level control for the right delay tap.

Enable/Bypass Switch

Enables or disables the Delay Modulator. You can use this switch to compare the DM-1 settings to that of the original signal or bypass the entire DM-1 section to reduce UAD-1 DSP load.

Output Knob

Adjusts the relative output of the plugin.

DM-1L

DM-1L is identical to the DM-1 except that the maximum available delay time per channel is 2400milliseconds. DM-1L requires significantly more memory resources of the UAD-1 than the DM-1. Therefore, we recommend using the DM-1L only when very long delay times are needed.

Link Button

This button links the left and right delay knobs so that when you move one delay knob, the other follows. The ratio between the two knobs is maintained.



Figure 37. The DM-1L includes a Link button

RS-1 Reflection Engine



Figure 38. The RS-1 Reflection Engine plugin window

Overview

The RS-1 Reflection Engine simulates a wide range of room shapes, and sizes, to drastically alter the pattern of reflections. While similar to that of the RealVerb Pro plugin, the RS-1 does not offer the same breadth of features (such as room hybrids, room materials, morphing, and equalization). However, if you do not need the advanced capabilities that RealVerb Pro offers, you can use the RS-1 to achieve excellent room simulations, while also preserving DSP resources on the UAD-1 card.

The Delay control sets the time between the direct signal and the first reflection. The Size parameter controls the spacing between the reflections. The Recir control affects the amount of reflections that are fed back to the input and controls how many repeats you hear.

RS-1 Controls

Shape Pop-up Menu

Determines the shape of the reverberant space, and the resulting reflective patterns.

Table 7. Available RS-1 Shapes

Cube	Square Plate
Box	Rectangular Plate
Corr	Triangular Plate
Cylinder	Circular Plate
Dome	Echo
Horseshoe	Ping Pong
Fan	Echo 2
Reverse Fan	Fractal
A-Frame	Gate 1
Spring	Gate 2
Dual Spring	Reverse Gate

Delay Knob

Sets the delay time between the original signal and the onset of the reflections.

Size Knob

Sets the size of the reverberant space (from 1–99 meters) and defines the spacing of the reflections.

Delay/Size Settings Interaction

You may notice that when Delay is set to its maximum value (300 ms) and you move the Size control to its maximum value (99), the Delay value is decreased to 16.85. This occurs because the maximum delay time available to the plugin has been reached. The available delay time is limited and it needs to be divided among the Delay and Size values. Therefore, if the value of the Delay or Size setting is increased towards maximum when the other control is already high, its complementary setting may be reduced.

Recirculation (RECIR) Knob

Sets the amount of processed signal fed back into its input. Higher values increase the number of reverberations/delays and intensity of the processed signal.

Recirculation allows both positive and negative values. The polarity refers to the phase of the delays as compared to the original signal. If Recirculation displays a positive value, all the delays will be in phase with the source. If it displays a negative value, then the phase of the delays flips back and forth between in phase and out of phase.

Damping Knob

This low pass filter reduces the amount of high frequencies in the signal. Turn down this control to reduce the brightness. Higher values yield a brighter signal. Damping also mimics air absorption, or high frequency rolloff inherent in tape-based delay systems.

Wet/Dry Mix Knob

This control determines the balance between the delayed and original signal. Values greater than 50% emphasize the wet signal, and values less than 50% emphasize the dry signal.

Wet/Dry Mix allows both positive and negative values. The polarity refers to the phase of the delays as compared to the original signal. If a positive value is displayed, then all the delays will be in phase with the source. With a negative value, the delayed signal is flipped 180 degrees out of phase with the source.

L-Pan Knob

Sets the stereo position for the left channel, allowing you to adjust the width or balance of the stereo signal. For a mono signal, set both the L-Pan and R-Pan to the left.

R-Pan Knob

Sets the stereo position for the right channel, allowing you to adjust the width or balance of the stereo signal. For a mono signal, set both the L-Pan and R-Pan to the left.

Enable/Bypass Switch

Enables or disables the Reflection Engine. You can use this switch to compare the RS-1 settings to that of the original signal or bypass the entire RS-1 section to reduce UAD-1 DSP load.

Output Knob

Adjusts the relative output of the plugin.

CHAPTER 7

Nigel

Introducing Nigel

Nigel offers the latest generation of guitar processing technology integrated into a complete multi-effects plugin solution. Utilizing Universal Audio's exclusive component modeling technology, along with some very creative digital design, Nigel delivers a complete palette of guitar tones along with most every effect a guitar player might need, all with minimal latency and no load on your host computer's CPU.

Nigel's Preflex™ advanced guitar amp modeling technology goes well beyond the usual pre-amp/amp/cabinet emulators. In addition to delivering a wide range of highly playable classic amp tones from the "Clean & Warm" California tube sound to more metal soaked "British" tones, a bevy of original timbres simply not possible on any other guitar system can be realized. Preflex also offers variable component-level morphing between any two amp presets, truly bringing creative guitar voicing to the next level.

As with the UAD-1 Powered Plug-Ins CS-1 channel strip, the components of Nigel are also supplied as individual plug-ins for unprecedented DSP and creative efficiency. Each Nigel module includes Universal Audio's proven smoothing algorithm for zipper free automation of all parameters.

Nigel may change the way you think about guitar signal processing. Never before have such exciting, realistic, and extreme guitar sounds been heard from a software plugin. Enjoy!



POWERED PLUG INS

Nigel Screenshot



Figure 39. The Nigel plugin window

Nigel Modules

Nigel is comprised of eight modules: Gate/Compressor, Phasor, Mod Filter, Preflex, Cabinet, Trem/Fade, Mod Delay, and Echo. In order to conserve UAD-1 DSP resources when all of the modules are not required simultaneously, some of the Nigel components are also supplied as separate plugins.

The following UAD-1 Powered Plug-Ins are part of the complete Nigel package:

- Nigel (all of the modules in one plugin)
- Preflex (Gate/Compressor + Amp + Cabinet)
- GateComp (Gate/Compressor)
- Phasor
- Mod Filter
- TremFade (Tremolo/Fade)
- TremModEcho (Tremolo/Fade + Mod Delay + Echo)

Preflex Plugin

Preflex is the heart of Nigel. All of our plug-ins sound amazing but when it comes to guitar, Preflex really shines. This exciting new guitar processing technology offers truly dynamic sonic possibilities. Multiple equalizers, amp types, and cabinets use sophisticated algorithms to provide analog sound quality never before available in a digital environment.

The Color and Bent controls modify frequency and gain characteristics in interesting and musically useful ways, and realtime component-level morphing between any two amp types is possible. Additionally, the Amp models are user-updatable so your guitar sounds will never be obsolete.



Figure 40. The Preflex plugin window

Preflex Modules

The Preflex plug-in consists of three sub-modules: gate/compressor, amplifier, and cabinet simulator. Controls for each of these sub-modules is described below.

Gate/Comp Module



Figure 41. The Gate/Comp module

The Gate is the first sub-module in the Preflex signal chain. Its output is passed to the input of the Compressor. The compressor output is then passed to the input of the Amp module within Preflex.

A gate stops the input signal from passing when the signal level drops below a specified threshold value. Gates are generally used to reduce noise levels by eliminating the noise floor when the 'main' signal is not present, but they are also useful for special effects.

The Preflex Gate is optimized for use with guitars. The threshold is dynamic and the gate output has multiple soft knees and dynamic slope, providing a more natural and less choppy sound.

The Compressor reduces the dynamic range of the signal based on the threshold and ratio settings. Guitarists often use compressors to increase perceived sustain on long notes and for special effects. Refer to Chapter 5 for more details on compressor theory and operation. Note that Nigel's compressor sounds different than the CS-1/EX-1 compressor; it sounds "more vintage".

Gate Level Display

This LED-style VU meter displays the level of the signal at the input of Preflex. For minimum distortion and maximum signal-to-noise, the input level should be as high as possible. The signal is at 0dB just before the red 'LED' is illuminated.

Gate Off/On Button

Enables or disables the Gate module within Preflex. The Gate is engaged when the button indicator is bright red. Use this switch to compare the Gate settings to that of the original signal or bypass the entire Gate section to reduce UAD-1 DSP load.

Gate Fast Button

The Fast control reduces the release time of the gate. It has no effect on the attack time. When enabled, the gate will release immediately. On signals that slowly decay and/or have a wide dynamic range, a smoother (less choppy) sound may be obtained with Fast mode turned off.

Fast mode is engaged when the button indicator is bright red. The time values are 50ms when engaged and 170ms when off.

Gate Threshold Knob

Sets the threshold level for the gate. Any signals that exceed this level are passed into the module. Signals below the threshold level are increasingly attenuated. A Threshold of 0dB means the gate is always open. The range is 0dB to -96dB.

In typical use it's best to set the gate threshold value to just above the noise floor of the desired signal (so the noise doesn't pass when you are not playing), but below the desired signal input level (so the signal passes as you play).

Boost Button

The Boost button ([Figure 40 on page 100](#)) increases the overall signal level within Preflex by 20dB. It is completely independent of the Gate and Compressor On/Off controls and will provide a signal boost even with the Gate and Compressor are off.

Note: The Boost button is only available within Nigel and Preflex. The individual Gate/Comp plugin does not contain the Boost button because Boost only affects the Amp within Preflex.

Compressor Threshold Knob

Sets the threshold level for the compression. Any signals that exceed this level are compressed. Signals below the level are unaffected. A Threshold of 0dB yields no compression. The range is 0dB to -60dB.

As the Threshold control is increased and more compression occurs, output level is typically reduced. However, the compressor provides an auto-makeup gain function to automatically compensate for reduced levels. Adjust the Output level control if more gain is desired.

Compressor Ratio Knob

Determines the amount of gain reduction used by the compressor. For example, a value of 2 (expressed as a 2:1 ratio) reduces the signal by half, with an input signal of 20 dB being reduced to 10 dB. A value of 1 yields no compression. Values beyond 10 yield a limiting effect. The range is 1 to 60dB.

Compressor Attack Menu

Sets the amount of time that must elapse, once the input signal reaches the Threshold level, before compression will occur. The faster the Attack, the more rapidly compression is applied to signals above the Threshold.

Three Attack values are available: Slow (50ms), Medium (8ms), and Fast (400µs).

Compressor Release Menu

Sets the amount of time it takes for compression to cease once the input signal drops below the Threshold level. Slower release times can smooth the transition that occurs when the signal dips below the threshold, especially useful for material with frequent peaks. However, if you set too large of a Release time, compression for sections of audio with loud signals may extend to lengthy sections of audio with lower signals.

Three Release values are available: Slow (500ms), Medium (120ms), and Fast (40ms).

Compressor On/Off Button

Enables or disables the Compressor module within Preflex. The Compressor is engaged when the button indicator is bright red. You can use this switch to compare the compressor settings to that of the original signal or bypass the entire compressor section to reduce UAD-1 DSP load.



Amp Module

The Preflex Amp is where Nigel's real magic happens. Behind its deceptively simple user interface is "rocket science" in action. The input to the Amp module is received from the Compressor output. The Amp output is passed to the input of the Cabinet module.



Figure 42. The Amp module within Preflex

Amp Type and Variable Knob Functions

The function of the amp knobs vary depending on the amp type. When an amp type is selected, Preflex is internally reconfigured. Although the amp types are essentially factory programmed presets, they are not simply a set of knob values. As different amp types are selected, the actual function and range of the amp knobs assume new characteristics.

Color and Bent: Supercontrol

The Color and Bent knobs have especially powerful functionality. Each modifies several amplifier characteristics simultaneously, so they behave as "super controls" that can have a dramatic effect on your sound with just one knob turn.

These are generally the main controls you will reach for when you want to make major changes to the overall dynamic response, timbre, or distortion characteristics of Preflex.

Knob Values Are Offsets

Knob settings do not change to new values when an amp type is selected. This is because knob values are not absolute. Instead, they are an offset to the factory programmed amp type value. For example, if Post-Lo EQ displays a value of 3.0, then 3dB is added to the amp type internal (preset) value. Of course, knob settings do change when user settings are loaded.

Amp Types and Morph

The Amp submodule within Preflex is actually comprised of two independent amplifier processors, Amp-A and Amp-B. The amp types to be used are selected with the Amp Type pull-down menus. The two amp types share the amp controls.

These two amp types can then be 'morphed' to smoothly transform one amp type into another, creating new sounds never before possible. Morph accomplishes this task by interpolating between amplifier component values of the A and B Amp types as the slider is moved. Morph is NOT a blend or crossfade control.

Morph allows you to continuously shift between two completely different amp sounds in realtime with full automation. And because the Color and Bent knobs also control multiple parameters simultaneously (which is essentially a morph), amazing new dynamically shifting timbres can be realized.

Amp Controls

Amp EQ Groups

Preflex has two groups of Lo, Mid, and Hi equalizer controls. Pre-EQ is before the amplifier, and Post-EQ is after the amplifier. Both sets of EQ are available simultaneously.

The actual frequency and bandwidth of a particular EQ knob depends on the amp type setting. The EQ knob values are offsets relative to the preset amp type value; they do not display absolute values.

Amp Pre-EQ Knobs

The Pre-EQ group modifies the tone of the signal before it passes into the Amp. Note that the EQ knob values are offsets relative to the preset amp type value; they do not display absolute values.

Pre-EQ Lo Knob

Modifies the low frequency response of the signal before the Amp. This control is set to a fixed frequency, but the frequency changes with the amp type.

Pre-EQ Mid Knob

Modifies the middle frequency response of the signal before the Amp. The frequency that this knob controls is determined by the Color knob (see Color knob description for more details).

Pre-EQ Hi Knob

Modifies the high frequency response of the signal before the Amp. This knob behaves differently than the Lo and Mid knob. Rather than boosting or cutting the gain of a certain frequency, the Hi knob increases the amplifier's sensitivity to high frequencies. The Hi control is VERY interactive with the Bent control.

Amp Post-EQ Knobs

The Post-EQ group modifies the tone of the signal after it passes through the Amp but before it goes to the Cabinet. Note that the EQ knob values are offsets relative to the preset amp type value; they do not display absolute values.

Post-EQ LO Knob

Modifies the low frequency response of the signal after the Amp. This control is a set to a fixed frequency, but the frequency changes with the amp type.

Post-EQ Mid Knob

Modifies the middle frequencies response of the signal after the Amp. The frequency that this knob controls is determined by the Color knob (see Color knob description for more details).

Post-EQ Hi Knob

Modifies the high frequency response of the signal after the Amp. This control is a set to a fixed frequency, but the frequency changes with the amp type.

Amp Color Knob

The Color knob is like a super tone control. It controls several amplifier characteristics simultaneously, and its behavior is determined by the selected amp type.

Amp Bent Knob

The Bent knob is like a super gain control. It controls several amplifier characteristics simultaneously, and its behavior is determined by the selected amp type.

Amp Output Knob

Adjusts the signal output level of Preflex. This may be necessary if the signal is dramatically boosted or reduced by the Gate/Compressor or Amp settings.

Bright Button

Increases the brightness of the Amp model. Bright is on when the button glows bright red.

Amp On/Off Button

Enables or disables the Amp module within Preflex. The Amp is engaged when the button indicator is bright red. You can use this switch to compare the Amp settings to that of the original signal or bypass the entire Amp section to reduce UAD-1 DSP load.

Amp Type Menus

The Amp Type pull down menus establish the overall sound and response of Preflex and also determine the function and ranges of the Amp knobs. Two amp types (A and B) can be active simultaneously by positioning the Morph control between them.

Note: For the following descriptions of the Amp models and other references that you may find throughout this manual, please be aware that Fender, Marshall, Mesa, Matchless, Aiken, and any other manufacturer, model name, description, and designations are all trademarks of their respective owners, which are in no way associated or affiliated with Universal Audio. These trademarks and names are used solely for the purpose of describing certain timbres produced using Universal Audio's exclusive modeling technology.

Amp Type List and Descriptions

Table 8. Amp Type List and Descriptions

AMP TYPE	DESCRIPTION
Rectified	Modern super-high gain amplifiers
Marsha	Emulations from range of new and old Marshall amps
Bassmon	Fender Bassman and similar amplifiers
Boutique	Matchless, Aiken, and other high-end tube amplifiers
Custom Blues	Designed to achieve those hard-to-nail blues tones. Lower gain.
Supa Clean	Direct input into a channel strip
Super Sat	Extremely high gain amp, breaks up easily in low end
Gemini	Fender Twin and similar clean tube amplifiers
Big Beaver	Distortion pedal stomp-box emulations
Super Custom	Higher-gain and more power than Custom Blues
Big Bottom	Optimized for bass guitar
Super Tweed	Small Fender Champ and Princeton when cranked up loud

Amp-A Type Menu

Determines the amp type for the “A” section of the Amp. Selecting an Amp Type reconfigures the amplifier characteristics and the function of the other Amp parameters.

Amp-B Type Menu

Determines the amp type for the “B” section of the Amp. Selecting an Amp Type reconfigures the amplifier characteristics and the function of the other Amp parameters.

Amp Morph Slider

The Morph control is used to smoothly transform one amp type into another, creating new sounds never before possible. Morph accomplishes this task by interpolating between amplifier component values of the A and B Amp types as the slider is moved. Morph is NOT a blend or crossfade control.

Morph allows you to continuously shift between two completely different amp sounds in realtime with full automation. And because the Color and Bent knobs also control multiple parameters simultaneously (which is essentially a morph), amazing new dynamically shifting timbres can be realized.

Cabinet Module

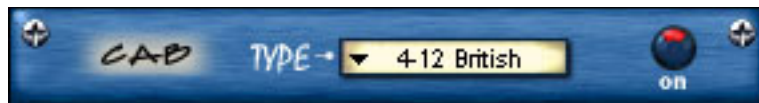


Figure 43. The Cabinet module within Preflex

The Cabinet module reproduces the sonic character of a guitar speaker and its enclosure as captured by a microphone. The Cabinet receives its input signal from the Preflex Amp output. The Cabinet output is the final Preflex signal output.

The Preflex Cabinets are emulations of actual guitar speaker enclosures that were captured by a Shure SM57 microphone then meticulously analyzed (as usual) by our team of rocket scientists. A wide variety of cabinets are included, using several speaker types, configurations, and microphone placement techniques.

Cabinet Type Menu

Each cabinet type has a unique sound and frequency response characteristic. Select the desired speaker from the Cabinet Type pull-down menu. Abbreviations used in the Cabinet Types list for the speaker, enclosure, and mic techniques are detailed in [Table 9](#). The Cabinet Types list itself is in [Table 10](#).

Note: For the following descriptions of the Cabinet models and other references that you may find throughout this manual, please be aware that Celestion, Greenback, Oxford Blue, Marshall, Fender, Line 6, Pod, SansAmp, Shure, ADA, Utah and any other manufacturer, model name, description, and designations are all trademarks of their respective owners, which are in no way associated or affiliated with Universal Audio. These trademarks and names are used solely for the purpose of describing certain timbres produced using Universal Audio's exclusive modeling technology.

Cabinet Abbreviations

Table 9. Cabinet Abbreviation Descriptions

ABBREVIATION	DESCRIPTION
1-12, 2-12, 4-12	One, two, or four twelve-inch speaker(s)
1-10, 2-10, 4-10	One, two, or four ten-inch speaker(s)
OB	Open Back cabinet
SC	Sealed Cabinet (closed back cabinet)
On Axis	Mic close and perpendicular (at 90 degrees), off-center
Off Axis	Mic close and angled, off-center
Edge	Mic close and angled at edge of speaker
Far	Mic approximately 2 feet from speaker
1-12 OB	90-watt Celestion (early 1990's)
2-12 OB	Left speaker: Oxford Blue, Right: Utah (both 60-watt, early 1960's)
2-12 SC	90-watt Celestions (early 1990's)
4-12 SC	25-watt Celestion Greenbacks (circa 1967)
British	Emulation of Marshall effects box cabinet
NoCA FXB	Emulation of ADA effects box cabinet
LA FXB	Emulation of Line 6 Pod effects box cabinet
NY FXB	Emulation of SansAmp effects box cabinet

Cabinet Type List

Table 10. List of Cabinet Types

1-12 OB Off Axis	4-12 SC Edge
2-12 OB Off Axis	2-12 SC Far
1-12 OB On Axis	4-12 SC Far
2-12 OB On Axis	4-12 British
1-12 OB Edge	1-10 NoCA FXB
2-12 OB Edge	2-10 NoCA FXB
1-12 OB Far	4-10 NoCA FXB
2-12 OB Far	1-12 LA FXB
2-12 SC Off Axis	2-12 LA FXB
4-12 SC Off Axis	4-10 LA FXB
2-12 SC On Axis	1-12 NY FXB
4-12 SC On Axis	

Cabinet On/Off Button

Enables or disables the Cabinet module within Preflex. The Cabinet is engaged when the button indicator is bright red. You can use this switch to compare the Cabinet settings to that of the original signal or bypass the entire Cabinet section to reduce UAD-1 DSP load.

Output Level Meter

This LED-style VU meter displays the level of the signal at the output of the Cabinet. Just before the red 'LED' is illuminated, the signal is at 0dB. In order to avoid overloading your host application signal path, adjust the Preflex output level so that the signal is at or below 0dB.

Phasor Module

The Phasor is a frequency-variable comb-filter with low frequency oscillator modulation. It is capable of producing dramatic sweeping and swooshing effects, including modern and classic sounds such as those produced by the Mutron Bi-Phase, Small Stone and MXR series of phasors.



Figure 44. The Phasor plugin window

Rate Knob

Sets the LFO modulation (sweep) rate of the Phasor. The available range is from 0.03Hz to 10Hz.

Sweep Knobs

The Sweep knobs determine the frequency range that will be affected by the Phasor. The low and high frequency values can be independently adjusted. This flexible arrangement allows the Phasor to affect a narrow or broad frequency range, and also enables you to tune the frequency response characteristic to match the signal if desired.

Sweep Lo Knob

Sets the lowest frequency of the Phasor. The available range is from 50Hz to 6000Hz.

Because the Sweep Lo frequency cannot be set higher than the Sweep Hi frequency, if the Lo value is increased beyond the Hi value the Hi value will increase to match the Lo value.

Sweep Hi Knob

Sets the highest frequency of the Phasor. The available range is from 50Hz to 6000Hz.

Because the Sweep Hi frequency cannot be set lower than the Sweep Lo frequency, if the Hi value is decreased below the Lo value the Lo value will decrease to match the Hi value.

Recirculation (Recir) Knob

Sets the intensity of the filtering effect. Higher values increase the intensity.

Recirculation allows both positive and negative values. The polarity refers to the phase of the feedback as compared to the original signal. If Recirculation displays a positive value, the feedback will be in phase with the source. If it displays a negative value, then the feedback will be out of phase.

Mix Knob

This control determines the balance between the processed and the original signal. Values greater than 50% emphasize the processed signal, and values less than 50% emphasize the original signal. A value of 100% delivers just the processed (wet) signal, and a value of 0% delivers just the source (dry) signal.

Mix allows both positive and negative values. The polarity refers to the phase of the processed signal as compared to the original signal. If a positive value is displayed, then the processed signal will be in phase with the source. With a negative value, the processed signal is flipped 180 degrees out of phase with the source signal.

LFO Type Menu

Determines the LFO (low frequency oscillator) waveshape and phase used to modulate the signal. The waveshape can be set to triangle or sine, each with varying duty cycles and phases.

Table 11. Phasor LFO Types and Descriptions

Sin	Pure sine wave.
Sin 2	Modified sine wave that stays high longer.
Sin 3	Modified sine wave that stays low longer.
Square	Square wave.
Square 2	Modified square wave that stays high longer.
Square 3	Modified square wave that stays low longer.
Sin 180	Sine wave 180 degrees out of phase.
Square 180	Square wave 180 degrees out of phase.

Order Menu

Determines the filter order for the Phasor filter banks. This setting affects the tonal complexity of the Phasor. Higher Order filters are more detailed than lower Order filters. Ten filter Order values are available, 3 through 12.

Phasor On/Off Button

Enables or disables the Phasor module. You can use this switch to compare the Phasor settings to that of the original signal or bypass the Phasor to reduce UAD-1 DSP load.



Mod Filter Module

The Mod Filter is an advanced filter plug-in that is capable of fixed-wah, auto-wah, envelope follower, sample/hold-driven filter, and other modulated filter effects. It has been modeled after the Mutron III and other popular filters. The filter cutoff frequency can be controlled by the signal level at the input to the module or a low frequency oscillator (LFO). This realtime dynamic response is what gives the Mod Filter its unique sound.



Figure 45. The Mod Filter plugin window
The label and function of the first knob depends upon the Mod Type menu selection.

Sens/Rate/ Wah Knob

The function and label of the first knob in the Mod Filter is determined by the Mod Type setting. When the Mod Type is an envelope, the label changes to “Sens” and determines the gain sensitivity of the Mod Filter. When the Mod Type is an LFO, the label changes to “Rate” and determines the rate of the LFO. When the Mod Type is set to Wah, the label changes to “Wah” and adjusts the wah pedal position.

Sens

When the knob is controlling Sensitivity, a higher setting will have a greater (more sensitive) response to variations in dynamic level.

Rate

When the knob is controlling Rate, a higher setting will increase the period of filter cutoff frequency modulation by the LFO. The range is from 0Hz to 8Hz.

Wah

When the knob is controlling Wah, a higher setting will have a brighter sound, just like when a real wah pedal is pressed forward.

On a real wah pedal, the wah filter is alternately enabled and disabled by rocking the pedal to the maximum forward position. Similarly, when the Wah knob is moved to the maximum position the wah effect is alternately enabled/disabled until the knob (or an external controller mapped to the knob) is moved to maximum again. This emulates real wah pedal behavior when an external MIDI control pedal is used in realtime. (Hint: add a rubber stopper to the front of your MIDI pedal to fully emulate a real wah pedal.)

Sweep Knobs

The Sweep knobs determine the frequency range of the Mod Filter. The low and high frequency values can be independently adjusted. This flexible arrangement allows the Mod Filter to affect a narrow or broad frequency range, and also enables you to tune the frequency response characteristic to match the signal if desired.

Sweep Lo Knob

Sets the lowest frequency to be affected by the Mod Filter. The available range is from 50Hz to 4000Hz.

Because the Sweep Lo frequency cannot be set higher than the Sweep Hi frequency, if the Lo value is increased beyond the Hi value the Hi value will increase to match the Lo value.

Sweep Hi Knob

Sets the highest frequency to be affected by the Mod Filter. The available range is from 50Hz to 4000Hz.

Because the Sweep Hi frequency cannot be set lower than the Sweep Lo frequency, if the Hi value is decreased below the Lo value the Lo value will decrease to match the Hi value.

Resonance (Res) Knob

Sets the amount of filter intensity for the Mod Filter. A higher value will deliver a sharper, more pronounced effect.

Output Knob

Adjusts the signal output level of the Mod Filter. This may be necessary if the signal is dramatically boosted or reduced by the other settings within the module. The range is from -20dB to 40dB.

Mod Type Menu

Determines the source of the filter modulation. There are three main Mod Types: LFO, Envelope, and Wah Pedal. Each Mod Type is described below.

LFO Mode

Three LFO modes are available: Sine, Square, and Random. The only difference between the three LFO modes is the waveshape of the Low Frequency Oscillator. Random LFO is chromatically tuned for maximum musicality.

When LFO mode is engaged, the filter cutoff frequency does not respond dynamically to changes in input signal level. Instead, the filter cutoff frequency is varied according to the Rate knob setting.

Envelope Mode

When Envelope mode is engaged, the filter cutoff frequency responds dynamically in realtime to variations in the input signal level. The amount of dynamic response is determined by the Sensitivity (Sens) knob.

In Env Up mode, a higher signal level sets the filter cutoff to a higher value. In Env Down mode, the envelope is inverted, and a higher signal level sets the filter cutoff to a lower value.

Wah Mode

When the Wah mode is engaged, the filter cutoff frequency is varied according to the Wah knob setting.

Wah Pedal Mode

Similar to Wah mode, in Wah Pedal mode the filter cutoff frequency is varied according to the Wah knob setting. However, when the knob reaches its maximum value the effect is bypassed until the knob reaches its maximum value again at which time the effect is re-engaged.

Wah Pedal mode is ideally suited to emulating a real Wah pedal by using a MIDI foot pedal controller.

Mod Menu Table

Table 12. Mod Filter: Mod Types and Descriptions

Sin	LFO mode with Sine waveshape.
Square	LFO mode with Square waveshape.
Random	LFO mode with Random waveshape.
Env Up	Normal Envelope mode. Filter cutoff frequency is dynamically increased as signal level increases.
Env Down	Inverted Envelope mode. Filter cutoff frequency is dynamically decreased as signal level increases.
Wah	Fixed Wah mode.
Wah Pedal	Fixed Wah mode with latched bypass mode.

Filter Type Menu

Determines the type of filter to be used by the Mod Filter. This parameter will affect the overall sonic character of the plugin. Four filter types are available.

Table 13. Mod Filter: Filter Types and Descriptions

Lowpass	Frequencies below the filter cutoff frequency are allowed to pass through the filter.
Bandpass	Frequencies around the filter cutoff frequency are allowed to pass through the filter. Lowest and highest frequencies are not passed.
Highpass	Frequencies above the filter cutoff frequency are allowed to pass through the filter.
Wah	Traditional wah pedal setting.

Mod Filter On/Off Button

Enables or disables the Mod Filter. You can use this switch to compare the Mod Filter settings to that of the original signal or bypass the Mod Filter to reduce the UAD-1 DSP load.

TremModEcho Plugin

The TremModEcho is loaded as one plugin but consists of three modules: Trem/Fade, Mod Delay, and Echo. Each of the module controls is described in the following pages.



Figure 46. The TremModEcho plugin contains three modules

Trem/Fade Module



Figure 47. The Trem/Fade module

Trem/Fade is a sophisticated envelope-controlled modulation processor that can produce classic tremolo, fade, and other gain modulation effects. Tremolo is achieved by modulating the amplitude (volume) of a signal with a low frequency oscillator (LFO). Trem/Fade includes some new modes such as Shimmer and VariTrem that enable the production of new volume effects.

Threshold (Thresh) Knob

Sets the threshold level for the Trem/Fade effect. When the signal level exceeds the threshold, the Trem/Fade effect is triggered. The range is 0dB to -80dB.

Note: When Trem/Fade is used within the complete Nigel plugin, the threshold detector is connected to the output of the Gate module. This allows for optimal level tracking independent of the amplifier and other effect settings.

Once a Trem/Fade effect is instigated by crossing the threshold level, the effect will continue until the signal drops below the Threshold level. For example, if a signal is faded in, the signal won't fade in again until its level drops below the Threshold value.

Trigger LED

The Trigger LED indicates when the Trem/Fade input signal is above the Threshold. It provides visual feedback for optimizing the Threshold setting. The Trigger LED glows bright red when the signal is above the Threshold value.

Fade In Knob

Determines the signal fade in time. Fade In is typically used to create automatic volume swells. The range is from None to 4000 milliseconds. When set to None, there is no fade in and only the Tremolo effect is active.

Onset Knob	Determines the time for the Tremolo effect to reach the specified depth. Onset behaves as an intensity ramp for the Tremolo effect. The range is from None to 4000 milliseconds. When set to None, the Tremolo effect is always on.
Rate knob	Sets the LFO rate (period) for the Tremolo. The range is from 0Hz to 16Hz.
Depth Knob	Sets the maximum Tremolo depth. The range is from zero to 100%.
LFO Type Menu	Determines the LFO waveshape used to modulate the signal. The waveshape can be set to sine or square.
Mode Menu	The Mode menu reconfigures the behavior of the Trem/Fade algorithms and/or the preset parameter settings. Each of the Modes is described below.

Fade Mode

In Fade mode, when the input signal level crosses the threshold value, the audio will fade in (ramp up) according to the time set with the Fade In knob. The Onset, Rate, and Depth controls are also active in Fade mode.

Two Fade modes are available. Each has a different Fade In curve and therefore a different volume envelope shape.

Note: *If the Threshold value is set too high for the source signal in Fade mode, the effect will not be triggered and the audio will never fade in.*

Shimmer Mode

In Shimmer mode, when the input signal level crosses the threshold value, the Tremolo effect will gradually increase according to the time set with the Onset knob. The Fade In knob is also active in Shimmer mode.

Three Shimmer modes are available. Each has a different Onset curve.

Note: *If the Depth value is zero and/or the Threshold value is set too high in Shimmer mode, you will not hear the Shimmer effect.*

Tremolo Mode

When Tremolo mode is selected, the Fade In and Onset controls are set to zero and the Trem/Fade module behaves as a 'normal' tremolo effect. However, the Fade In and Onset controls are still active and can be adjusted as desired.

Two Tremolo modes are available. Each has different settings but the controls behave exactly the same in both modes.

Note: *If the Depth value is zero and/or the Threshold value is set too high in Tremolo mode, you will not hear the tremolo effect.*

VariTrem Mode

In VariTrem mode, the tremolo rate is automatically increased or decreased in realtime. The rate is ramped up or down according to the value of the Onset control. For example, if VariTrem Up is selected and Onset has a value of 2 seconds, the Tremolo rate will gradually increase for 2 seconds.

Two VariTrem modes are available. Vari T Up gradually increases the Tremolo rate, and Vari T Dn gradually decreases the Tremolo rate.

Note: *If the Depth value is zero and/or the Threshold value is set too high in VariTrem mode, you will not hear the VariTrem effect.*

Trem/Fade On/Off Button

Enables or disables Trem/Fade. You can use this switch to compare the Trem/Fade settings to that of the original signal or to disable Trem/Fade amplitude processing.

UAD-1 DSP load is not reduced when Trem/Fade is disabled with the On/Off button. The Trem/Fade amplitude processor remains active even when its audio is disabled so it can be used as a modulation source when using “Trem” as the LFO Type in the Mod Delay module.



POWERED PLUG INS

Mod Delay Module



Figure 48. The Mod Delay module
The label and function of the second two knobs depend upon the Mode menu selection.

The Mod Delay is a short digital delay line that includes a low frequency oscillator. The Mod Delay produces lush chorus, flange, and vibrato effects.

Because the Trem/Fade amplitude processor can be used to control the Mod Delay, sophisticated envelope-controlled flange, chorus, and vibrato modulations can be achieved.

Rate Knob

Sets the LFO modulation rate (period) of the delayed signal. The available range is 0.01Hz to 25Hz.

If the LFO Type menu is set to one of the Trem modes, the Rate is linked to the Trem/Fade module rate. In this scenario the Rate knob value changes to "Trem", adjusting the Mod Delay Rate will have no effect, and the modulation rate is determined by the Trem/Fade module settings (even if the Trem/Fade module is disabled with the On/Off button).

Depth & Time/ Sweep Knobs

The function and label of the second and third controls in the Mod Delay module are determined by the Mode pull-down menu. When the Mod Delay Mode is set to Flanger, the second and third knobs are labeled Sweep Lo and Sweep Hi respectively. When the Mod Delay Mode is set to Chorus or Vibrato, the second and third knobs are labeled Depth and Time respectively.

Sweep Knobs

The Sweep knobs determine the frequency range that will be affected by the Mod Delay. The low and high frequency values can be independently adjusted. This flexible arrangement allows the Mod Delay to affect a narrow or broad frequency range, and also enables you to tune the frequency response characteristic to match the signal if desired.

Note: *The Sweep knobs are only visible in Flanger mode.*

Sweep Lo Knob

Sets the lowest frequency to be affected by the Mod Delay. The available range is from 100Hz to 6000Hz.

Because the Sweep Lo frequency cannot be set higher than the Sweep Hi frequency, if the Lo value is increased beyond the Hi value the Hi value will increase to match the Lo value.

Sweep Hi Knob

Sets the highest frequency to be affected by the Mod Delay. The available range is from 100Hz to 6000Hz.

Because the Sweep Hi frequency cannot be set lower than the Sweep Lo frequency, if the Hi value is decreased below the Lo value the Lo value will decrease to match the Hi value.

Depth Knob

Sets the amount of modulation to be applied to the signal. The maximum available range is 0 to 300 cents. However, the available range is dependent on the Rate setting. Less Depth range is available at slower Rate settings.

Note: *The Depth knob is only visible in Chorus and Vibrato modes.*

Time Knob

Sets the modulation delay time. The available range is from 0 to 125 milliseconds. In Vibrato mode, this setting will appear to have no effect if the Recirculation value is zero because the signal is "100% wet" in Vibrato mode.

Note: *The Time knob is only visible in Chorus and Vibrato modes.*

Recirculation (Recir) Knob

Sets the amount of processed signal fed back into its input. Higher values increase the intensity of the processed signal.

Recirculation allows both positive and negative values. The polarity refers to the phase of the delays as compared to the original signal. If Recirculation displays a positive value, all the delays will be in phase with the source. If it displays a negative value, then the phase of the delays flips back and forth between in phase and out of phase.

In the flanger mode, Recirculation has the potential to make some very interesting sounds. Try turning RECIR fully clockwise or counter-clockwise, and set the delay to very short values.

Damping Knob

This low pass filter reduces the amount of high frequencies in the signal. Turn down this control to reduce the brightness of the sound. Higher values yield a brighter signal. Damping also mimics air absorption, or high frequency rolloff inherent in tape-based delay systems.

LFO Type Menu

Determines the LFO (low frequency oscillator) source, waveshape, and phase used to modulate the Mod Delay signal.

When the LFO Type is set to one of the Trem modes, the Rate is linked to the Trem/Fade module rate. In this scenario, the Rate knob value changes to "Trem" and adjusting Rate will have no effect.

By using the Trem/Fade amplitude processor as the LFO source of the Mod Delay module, extraordinary new timbres can be realized.

Mod Delay LFO Type Table

Table 14. Mod Delay LFO Types and Descriptions

Sin 0	In-phase sine wave
Sin 90	Sine wave 90 degrees out of phase
Sin 180	Sine wave 180 degrees out of phase
Tri 0	In-phase triangle wave
Tri 90	Sine wave 90 degrees out of phase
Tri 180	Sine wave 180 degrees out of phase
Trem Up	The Trem/Fade module is used as the LFO source. On a stereo signal, both channels ascend in pitch in synchronization with the Trem/Fade amplitude ramp.
Trem Down	The Trem/Fade module is used as the LFO source. On a stereo signal, both channels descend in pitch in synchronization with the Trem/Fade amplitude ramp.
Trem U/D	The Trem/Fade module is used as the LFO source. On a stereo signal, the left channel ascends in pitch as the right channel descends in synchronization with the Trem/Fade amplitude ramp.
Trem D/U	The Trem/Fade module is used as the LFO source. On a stereo signal, the right channel descends in pitch as the left channel ascends in pitch in synchronization with the Trem/Fade amplitude ramp.

Mode Menu

The Mode menu reconfigures the settings of the Mod Delay controls. Additionally, the function and label of the second and third controls in the Mod Delay module are determined by the Mode menu.

When the Mod Delay Mode is set to Flanger, the second and third knobs are labeled Sweep Lo and Sweep Hi respectively. When the Mod Delay Mode is set to Chorus or Vibrato, the second and third knobs are labeled Depth and Time respectively.

In all modes except Flanger, the function and sound of the controls are identical; only the settings are different. Similarly, in Flanger 1 and 2 modes, the function and sound of the controls are identical; only the settings are different.

Table 15. Mod Delay Mode Menu List

Chorus 1	Flanger 1	Vibrato 2
Chorus 2	Flanger 2	Comb Filter 1
Quad Chorus	Vibrato 1	Comb Filter 2

Mod Delay On/Off Button

Enables or disables the Mod Delay. You can use this switch to compare the Mod Delay settings to that of the original signal or bypass the Mod Delay to reduce UAD-1 DSP load.

Echo Module



Figure 49. The Echo module

The Echo module is a delay line used primarily for longer echo effects. When very short delay times or modulation are desired, use the Mod Delay instead. When VERY long delay times are desired, use the UAD DM-L plugin which has up to 2400 milliseconds available delay per stereo channel.

Input Knob

The Input knob attenuates the signal coming into the Echo module. The signal already passed into the Echo module is still processed even when the Input knob is at its minimum value (maximum attenuation).

This functionality enables the Echo to continue to process its signal even when no new signal is being input. Therefore, volume swells with Echo can be automated and high Recirculation effects such as sampling and “infinite repeat” techniques can be realized.

Time Knob

Sets the delay time between the original signal and the delayed signal. The maximum available delay time is 1200 milliseconds.

Recirculation (Recir) Knob

Sets the amount of processed signal fed back into its input. Higher values increase the number of delays and intensity of the processed signal.

Recirculation allows both positive and negative values. The polarity refers to the phase of the delays as compared to the original signal. If Recirculation displays a positive value, all the delays will be in phase with the source. If it displays a negative value, then the phase of the delays flips back and forth between in phase and out of phase.

Damping Knob

This low pass filter reduces the amount of high frequencies in the processed signal. Higher values yield a brighter signal. Turn down this control for a darker sound. Damping also mimics air absorption, or high frequency rolloff inherent in tape-based delay systems.

Mix Knob

This control determines the balance between the delayed and original signal. Values greater than 50% emphasize the wet signal, and values less than 50% emphasize the dry signal. A value of 50% delivers equal signals. A value of 0% is just the dry signal.

Wet/Dry Mix allows both positive and negative values. The polarity refers to the phase of the delays as compared to the original signal. If a positive value is displayed, then all the delays will be in phase with the source. With a negative value, the delayed signal is flipped 180 degrees out of phase with the source.

Mix Knob

This control determines the balance between the processed and the original signal. Values greater than 50% emphasize the processed signal, and values less than 50% emphasize the original signal. A value of 100% delivers just the processed (wet) signal, and a value of 0% delivers just the source (dry) signal.

Mix allows both positive and negative values. The polarity refers to the phase of the processed signal as compared to the original signal. If a positive value is displayed, then the processed signal will be in phase with the source. With a negative value, the processed signal is flipped 180 degrees out of phase with the source signal.

Mode Menu

The Mode menu determines how the Echoes are processed. The Echo Mode differences can only be heard when the module is applied to a signal on an insert, buss, group, or return that has a stereo output path.

Table 16. Echo Mode Menu List

Echo 1	Ping Pong 2
Echo 2	Clang 1
Echo 3	Clang 2
Ping Pong 1	Slapback

Echo On/Off Button

Enables or disables Echo. You can use this switch to compare Echo settings to that of the original signal or bypass Echo to reduce UAD-1 DSP load.

CHAPTER 8

Pultec and Pultec-Pro

Overview

The Pultec EQP-1A Program Equalizer and Pultec MEQ-5 plugins are faithful electronic reproductions of the classic hardware equalizers. Our DSP wizards have ensured that every revered sonic nuance of these vintage processors are faithfully maintained.

UAD Pultec and UAD Pultec-Pro

The UAD Pultec plugin is the EQP-1A Program Equalizer that was introduced in version 2.2 to much acclaim. UAD Pultec-Pro was introduced in version 3.5, and includes both the EQP-1A and MEQ-5 modules. The EQP-1A is identical in UAD Pultec and UAD Pultec-Pro.

In designing the Pultec equalizer plugins, we performed detailed analyses of the signal path and equalization characteristics of selected well-maintained, in-spec Pultec equalizers used regularly in professional studios. A “golden unit” was selected, and the resulting model reproduces the measured equalization and signal path characteristics to within a fraction of a dB mean error for all knob settings.

All of the unique features of the original Pultec EQ’s are included in the plugins, including the separate boost and attenuation controls, the smooth, sweet top end, and the ability to dial in seemingly dangerous amounts of boost without getting into trouble. All front panel controls are included, and all of the knob tapers are accurately modeled. The Pultec has long been a choice of recording and mastering engineers for its ability to bring out individual frequency ranges without significantly altering other frequencies. In addition, the Pultec is one of those magical pieces of gear that makes audio program sound better just by passing through it. The sophisticated modeling technology used in the Pultec plugins captures both of these key attributes.

Note: *The Pultec and Pultec-Pro plugins always operates at a high internal sample rate for maximum accuracy. Therefore, the UAD-1 DSP load does not increase even when processing audio at the highest sample rates.*

Pultec Latency

The Pultec and Pultec-Pro plugins introduce an additional 13 samples of delay due to upsampling when the session sample rate is below 100kHz. This additional latency does not occur at sample rates above 100kHz. You may enter a value of 13 in the "Samples" parameter in DelayComp or TrackAdv to compensate. See [“Compensating for Pultec EQ” on page 47](#) for more information.

Note: *Compensating for Pultec and Pultec-Pro is not required if the host application supports “full delay compensation” throughout the signal path.*

Pultec EQP-1A Screenshot



Figure 50. The Pultec EQP-1A Program Equalizer plugin window

Pultec EQP-1A Controls

The EQP-1A can control three frequency ranges simultaneously, using three groups of interacting parameters.

The first group controls the low frequencies and has three controls: boost, attenuation, and frequency select. The second group controls the high frequencies and has three controls: boost, bandwidth, and frequency select. The third group also controls the highs and has two controls: attenuation amount and frequency select.

The placement and grouping of the sections and their related controls are shown in [Figure 51 on page 129](#).



Figure 51. Control grouping within the Pultec EQP-1A

In/Out Toggle Switch

This is a signal bypass control. It allows you to compare the processed and unprocessed signal. It does NOT reduce UAD-1 DSP load.

In the hardware EQP-1A, the audio is still slightly colored even when the switch is in the Out position. This is due to the fact that the signal is still passing through its circuitry. Because the plugin emulates the hardware in every regard, the signal will be slightly processed when this switch is in the Out position. If a true bypass is desired, use the On/Off rotary switch.

On/Off Rotary Switch

This switch enables or disables the EQP-1A altogether. You can use this switch to compare the processed settings to that of the original signal, or to bypass the plugin to reduce UAD-1 DSP load.

Low Frequency Controls

CPS Selector Switch

This switch determines the frequency of the low shelf portion of the equalizer. CPS is an acronym for Cycles Per Second (Hertz). Four frequencies are available: 20, 30, 60, and 100 Hertz.

Boost Knob

This knob determines the amount of low shelf gain to be applied to the frequency set by the CPS switch.

Attenuation Knob

This knob determines the amount of low shelf cut to be applied to the frequency set by the CPS switch.

Note: In the documentation supplied with hardware version of the EQP-1A, it is recommended that both Boost and Attenuation not be applied simultaneously because in theory, they would cancel each other out. In actual use however, the Boost control has slightly higher gain than the Attenuation has cut, and the frequencies they affect are slightly different too. The EQ curve that results when boost and attenuation are simultaneously applied to the low shelf is an additional feature.

High Frequency Controls

KCS Selector Switch

This switch determines the frequency of the high boost portion of the equalizer. KCS is an acronym for KiloCycles per Second (kiloHertz). Seven frequencies are available (all in kiloHertz): 3, 4, 5, 8, 10, 12, and 16.

Bandwidth Knob

This knob sets the proportion of frequencies surrounding the center frequency (determined by the KCS switch) to be affected by the high boost. This is a 'Q' control. Lower values yield a narrower band and effect fewer frequencies.

Boost Knob

This controls sets the amount of gain for the high frequency portion of the equalizer.

High Attenuation Controls

Attenuation Selector Switch

This switch determines the frequency of the high frequency attenuator. Three frequencies are available (all in kiloHertz): 5, 10, and 20.

Attenuation Knob

This knob determines the amount of high shelf cut to be applied to the frequency set by the Attenuation Selector switch.

Pultec MEQ-5 Screenshot



Figure 52. The Pultec-Pro MEQ-5 Midrange Equalizer plugin window

Pultec MEQ-5 Controls

The MEQ-5 can control three frequency ranges simultaneously, using three groups of interacting parameters.

The first group controls the low -mid frequencies and has two controls: frequency select and boost. The second group controls the mid frequencies and has two controls: frequency select and attenuation. The third group controls high-mids and has two controls: frequency select and boost.

The placement and grouping of the sections and their related controls are shown in [Figure 53](#).



Figure 53. Control grouping within the Pultec-Pro MEQ-5

On/Off Toggle Switch

This switch disables the MEQ-5 portion of Pultec-Pro. It allows you to compare the processed and unprocessed signal of the MEQ-5. When in the out position, the UAD-1 DSP load is reduced.

In the hardware MEQ-5, the audio is still slightly colored even when the switch is in the Out position and the peak/dip controls are at zero. This is due to the fact that the signal is still passing through its circuitry. Because the plugin emulates the hardware in every regard, the signal will be slightly processed when this switch is in the In position and the peak/dip controls are at zero. If a true bypass is desired, use the host disable switch.

Low Peak Controls

Frequency Selector Switch

This switch determines the frequency of the low-midrange portion of the equalizer. Five frequencies are available: 200Hz, 300Hz, 500Hz, 700Hz, and 1 kHz.

Boost Knob

This knob determines the amount of low-midrange “Peak” (gain) to be applied to the frequency set by the low-midrange frequency selector.

Dip Controls

Frequency Selector Switch

This switch determines the frequency of the midrange portion of the equalizer. Eleven frequencies are available: 200Hz, 300Hz, 500Hz, 700Hz, 1 kHz, 1.5kHz, 2kHz, 3kHz, 4kHz, 5kHz, and 7kHz.

Attenuation Knob

This knob determines the amount of midrange “Dip” (cut) to be applied to the frequency set by the midrange frequency selector.

High Peak Controls

Frequency Selector Switch

This switch determines the frequency of the high-midrange portion of the equalizer. Five frequencies are available: 1.5kHz, 2kHz, 3kHz, 4kHz, and 5kHz.

Boost Knob

This knob determines the amount of high-midrange “Peak” (gain) to be applied to the frequency set by the high-mid frequency selector.

MEQ-5 Response Curves

We’ve included a few frequency response plots that illustrate the response curves of the MEQ-5. All plots were taken at a sample rate of 192kHz.

Low Peak Response

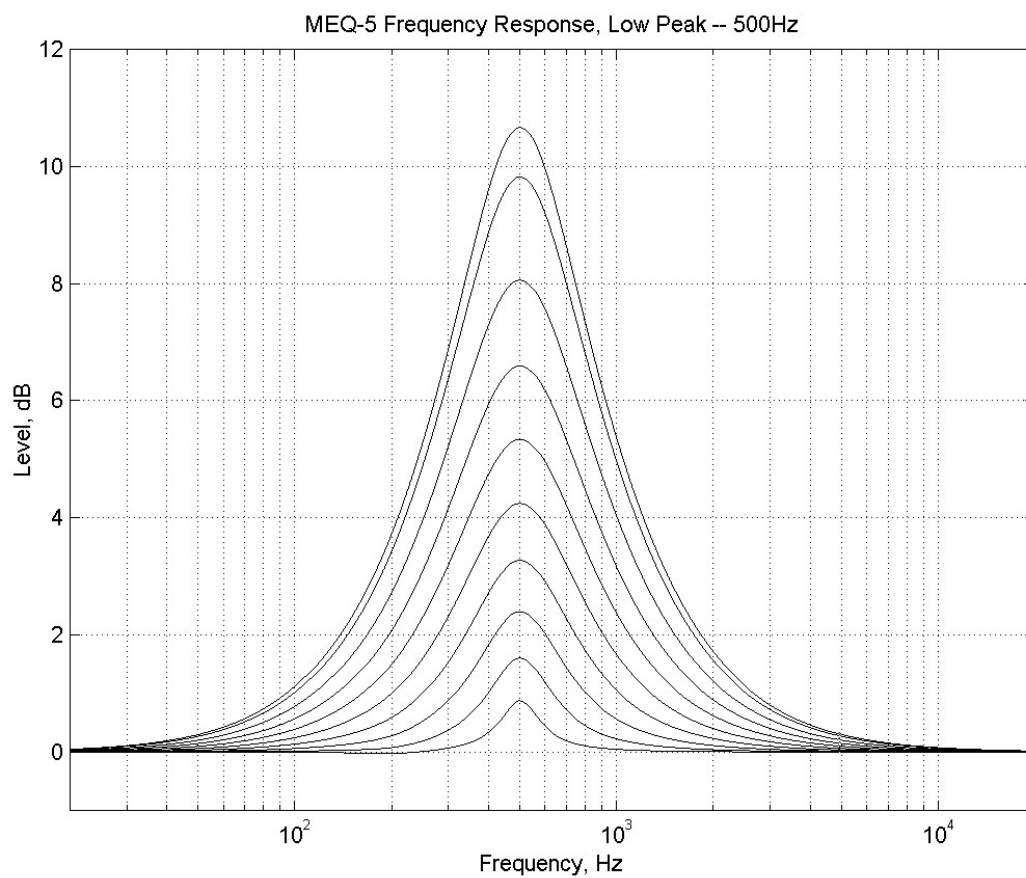


Figure 54. Pultec MEQ-5 Low Peak Response

Dip Response



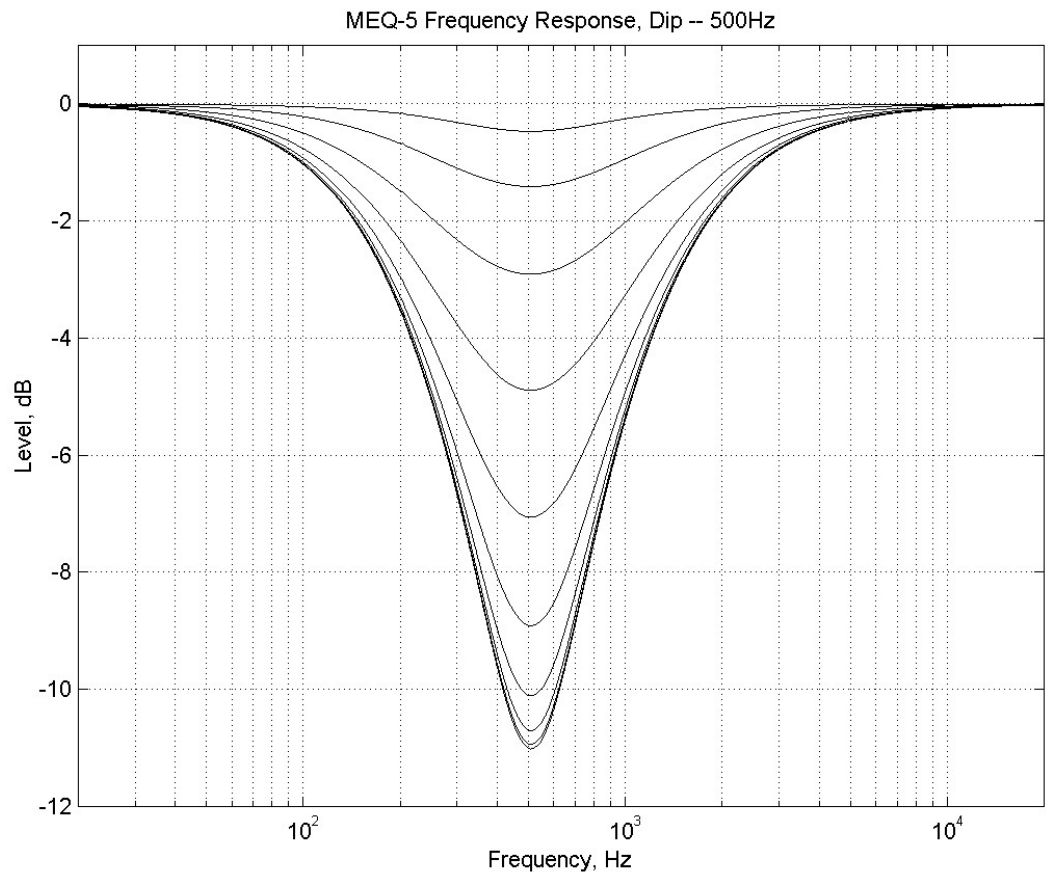


Figure 55. Pultec MEQ-5 Dip Response



High Peak Response

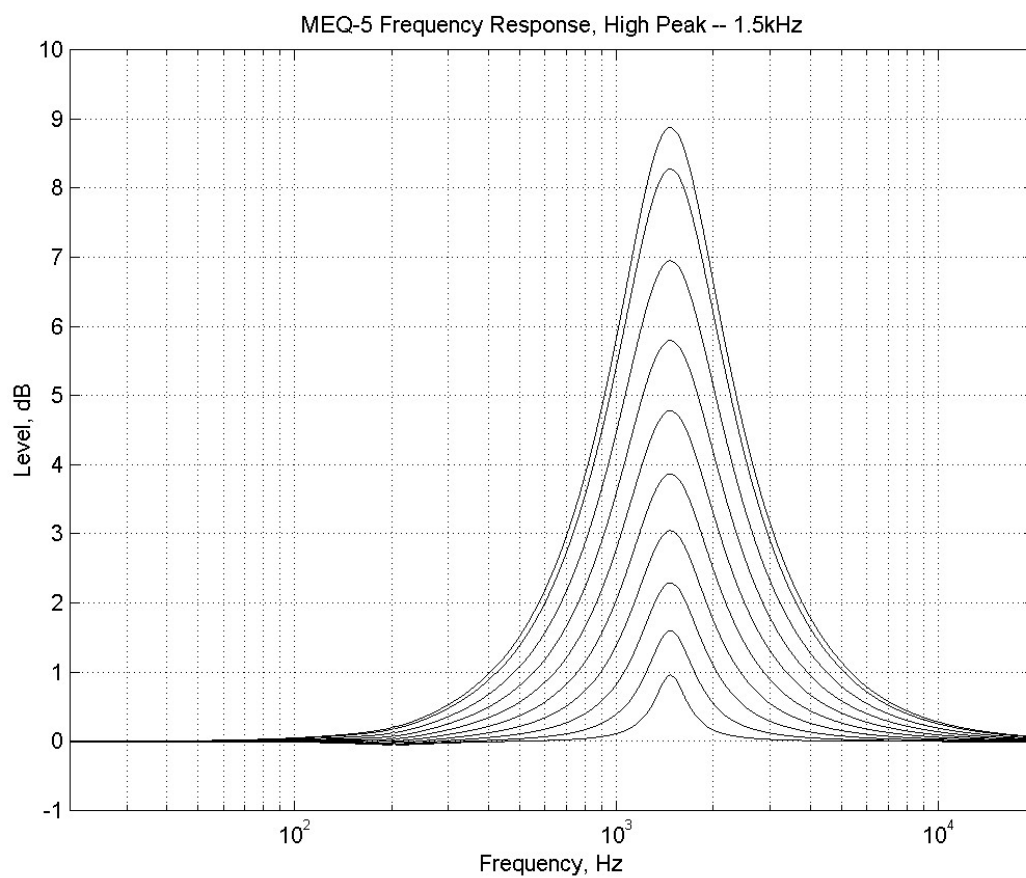


Figure 56. Pultec MEQ-5 High Peak Response



CHAPTER 9

Cambridge EQ

Overview

The UAD Cambridge EQ plugin is a mastering-quality, no-compromise equalizer that enables powerful tonal shaping of any audio source. Its algorithm was modeled from various high-end analog filters, providing a sonically rich foundation for timbral manipulation. Special attention was given to the handling of higher frequencies, resulting in a much smoother and more satisfying high-end response than is found in most digital filters.

Cambridge EQ is highly flexible, offering a broad spectrum of options facilitating surgical precision and delivering superior aural results in every application. This may be the most satisfying, full-featured equalizer in your arsenal of creative tools.

Cambridge EQ Screenshot



Figure 57. The UAD Cambridge EQ plugin window

Cambridge EQ Controls

Each feature of the Cambridge EQ interface is detailed below.

Response Curve Display

The Response Curve Display plots the frequency response of the current Cambridge EQ settings. It provides instant visual feedback of how audio is being processed by the equalizer.

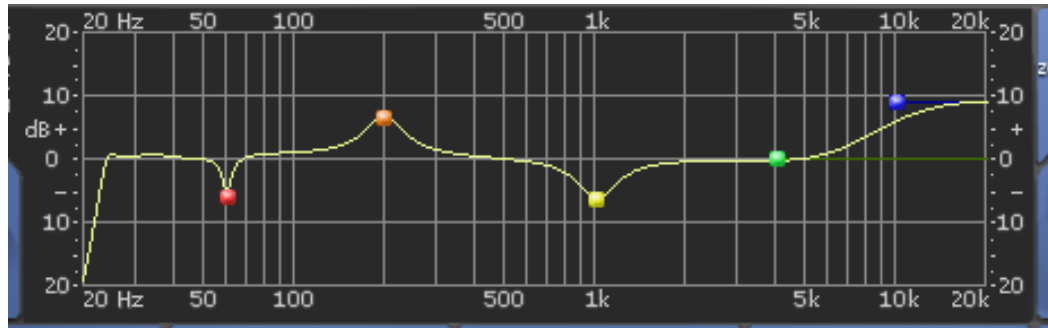


Figure 58. Cambridge EQ Response Curve display

The entire audio spectrum, from 20Hz to 20KHz, is displayed along on the horizontal axis. Gain and attenuation of frequencies (up to +/- 40dB) are displayed along the vertical axis. The vertical resolution of this display can be modified with the Zoom buttons.

Response Curve Color

The color of the response curve depends on the value of the A/B Selector control. When A is active, the curve is yellow. When B is active, the curve is green (see ["A/B Selector Button" on page 139](#)). When Cambridge EQ is disabled, the response curve is grey.

Zoom Buttons

The vertical scale of the Curve Display can be increased or reduced with the Zoom buttons. This function allows the resolution of the Curve Display to be changed for enhanced visual feedback when very small or very large amounts of boost or cut are applied. Four vertical ranges can be selected with the Zoom buttons: ± 5 , 10, 20, and 40dB.

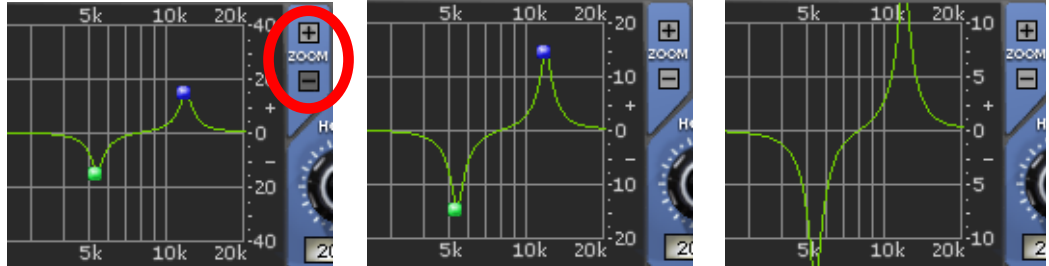


Figure 59. Vertical resolution of the Response Curve can be changed with the Zoom buttons

Curve Control Bats

There are five control “bats” on the curve display. Each bat is color coded and corresponds to each of the five EQ bands. The position of the bat on the curve display reflects the frequency and gain of its corresponding band, even if the band is disabled.

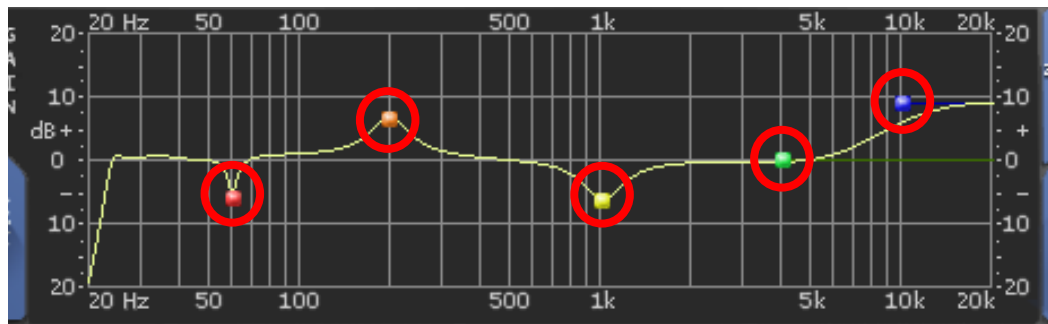


Figure 60. The Curve Control Bats can be used to control EQ band frequency, gain, and Q

The gain and frequency of an EQ band can be modified simultaneously by dragging its bat with the mouse. If a band is disabled when its bat is touched for the first time, the band is enabled.

Note: To modify the Q of a band with its bat, hold down the Control key while dragging vertically.

When a band is enabled, the EQ curve usually touches the bat. However, because the EQ curve always displays the actual frequency response of Cambridge EQ, if two bands are close together in frequency and/or at extreme gain values, the bat may not touch the curve itself.

Master Level Knob



This control adjusts the signal output level of Cambridge EQ. This may be necessary if the signal is dramatically boosted or reduced by the EQ settings. The available range is +/- 20dB.

A/B Selector Button



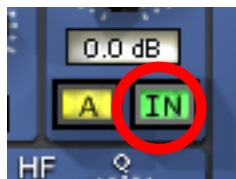
The A/B Selector switches between two separate sets of Cambridge EQ plugin values. This feature enables easy switching between two completely independent EQ curves which can be useful for comparison purposes or for automating radical timbre changes. Both the A and B curves reside within a single Cambridge EQ preset.

Click the A/B Selector button to switch between the two curves. When A is displayed, the button and the EQ response curve is yellow. When B is displayed, the button and the curve is green.

Note: To reset the A or B curve to a null (flat) response, control-click the A/B Selector button. The active curve will be nulled.

Note: To copy one curve to another, shift-click the button. The active curve will be copied to the inactive curve.

EQ Enable Button



This button enables or disables the Cambridge EQ altogether. You can use this switch to compare the processed settings to that of the original signal, or to bypass the plugin to reduce UAD-1 DSP load.

Low Cut / High Cut Filters

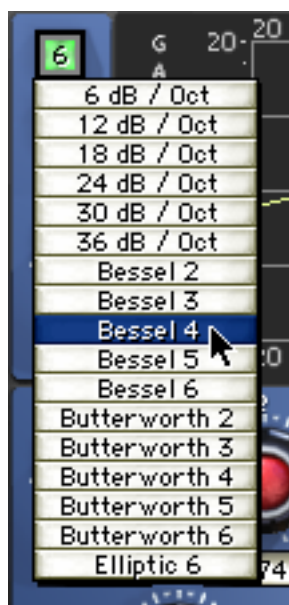


The Low Cut and High Cut filters are offered in addition to the five parametric/shelf bands. A wide range of filter types is provided to facilitate tonal creativity. Many filters that are available are represented.

Three controls are offered: Cut Type, Enable, and Frequency. Each control is detailed below.



Cut Type Menu



The Cut Type menu determines the sound of the low and high cut filters. To view the Cut Type menu, click and hold the green cut type button.

Four types of responses are provided: Coincident Pole, Bessel, Butterworth, and Elliptic. The numbers represent the filter order, i.e. Bessel 4 is a fourth-order filter. Each offers a different sound. To select a new cut response, drag to the desired response and release.

The responses are more gentle on filters with lower numbers, and get steeper and more aggressive as the numbers increase. The coincident-pole filters are first-order filters cascaded in series and offer gentle slopes. Bessel filters are popular because of their smooth phase characteristic with decent rejection. Butterworth filters offer even stronger rejection. The Elliptic setting is about as “brick wall” as you can get. Generally speaking, more phase shifting occurs as the response gets steeper.

Note: UAD-1 DSP usage does increase some as the filters get stronger.

Cut Enable Button

This button activates the cut filters. The filters are enabled when the “In” button is green. UAD-1 DSP usage is slightly reduced when the cut filters are disabled.

Cut Frequency Knob

This knob determines the cutoff frequency for the Cut filters. The available range is from 20Hz-5kHz for the low cut filter, and 20Hz-20kHz for the high cut filter.

EQ Bands

All five of the EQ bands can be used in parametric or shelf mode. Each band has identical controls, the only difference is the frequency range values.

The function of the controls is similar in both parametric and shelf modes. The two modes are described separately (see “[Parametric EQ](#)” on page 142 and “[Shelf EQ](#)” on page 145).



Figure 61. The EQ Band controls

Enable Button

Each band can be individually engaged with the Enable button. The button is green when the band is enabled. All bands default to disabled. To enable any band, click the Enable button.

You can use these buttons to compare the band settings to that of the original signal, or to bypass the individual band. UAD-1 DSP usage is slightly decreased when a band is disabled.

Frequency Knob

This parameter determines the center frequency to be boosted or attenuated by the Gain setting. The available range for each of the five bands is the same for both parametric and shelf modes. The ranges are shown in [Table 17](#).

Table 17. Available ranges for the Band Frequency parameter

Low Frequencies (LF)	20-400Hz
Low-Mid Frequencies (LMF)	30-600Hz
Mid Frequencies (MF)	100-6kHz
High-Mid Frequencies (HMF)	900-18kHz
High Frequencies (HF)	2k-20kHz

Note: When operating at sample rates less than 44.1kHz, the maximum frequency will be limited.

Gain Knob

This parameter determines the amount by which the frequency setting for the band is boosted or attenuated. The available range is ± 20 dB.

Q (Bandwidth) Knob

The behavior of the Q parameter varies depending on the band mode and the gain. For this reason Q is detailed separately in the parametric and shelf mode sections (see [“Parametric Q” on page 142](#) and [“Shelf Q” on page 145](#)).

Parametric EQ

A band is in parametric mode when shelf mode is disabled (see [“Shelf Enable Button” on page 145](#)). Three types of parametric EQ are available, as determined by the Parametric Type selector.

Parametric Type Selector



The Parametric Type selector changes the response of the band controls to reflect the behavior of various analog equalizers. It is a global control for all 5 bands, and has no effect on the low and high cut filters. Click the Parametric Type display to rotate between Types I, II, and III.

The filter algorithm is the same in all three parametric types. The difference is in the dependency between the gain and Q parameters. Each parametric type has its own response characteristics.

In Type I mode, the Q remains constant regardless of the gain setting. In Type II mode, the Q increases as gain is boosted, but remains constant as gain is attenuated. In Type III mode, the Q increases as gain is boosted and attenuated. See [Figure 62](#), [Figure 63](#), and [Figure 64](#).

Parametric Q

The Q (bandwidth) knob sets the proportion of frequencies surrounding the center frequency to be affected by the gain control. The Q range is 0.25–16; higher values yield sharper slopes.

Note that the Q numeric value in relation to its knob position is warped (i.e. not linear) and varies according to the parametric type.

Type I

When set to Type I, the bandwidth remains at a fixed Q regardless of the gain setting for the band; there is no Q/Gain interdependency. In addition, there is a finer resolution of the Q knob in the middle of its range. This makes it easier to achieve subtle bandwidth changes. Note that the Q value and knob positions do not change as the gain is modified. See [Figure 62](#).

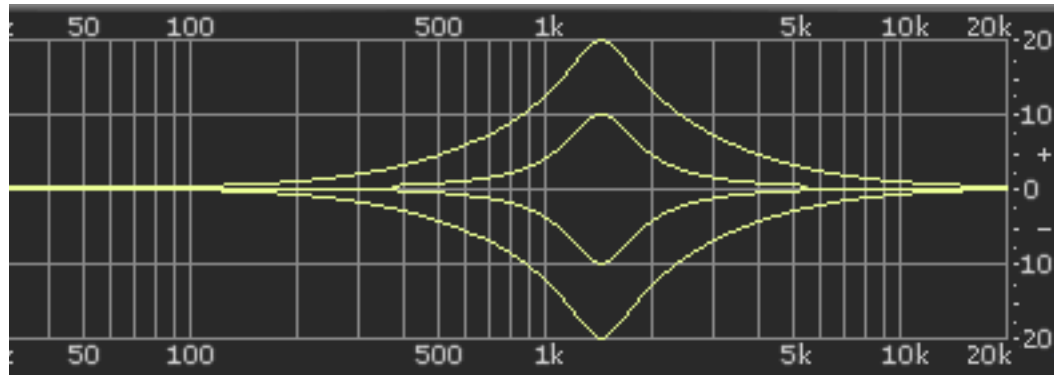


Figure 62. Parametric Type I response

Type II

When set to Type II, there is a Q/Gain dependency on boost. The bandwidth increases continuously as the gain is boosted, but not when attenuated. The Q knob position determines the maximum Q at full gain.

Filter bandwidth is broader at lower boost settings and narrower at higher boost settings. This can produce a smoother, more natural response when boosting filter gain.

Note that the Q value increases as gain is boosted but the knob position does not change. The Q value is approached as gain increases, and reaches the knob position at maximum gain. See [Figure 63](#).

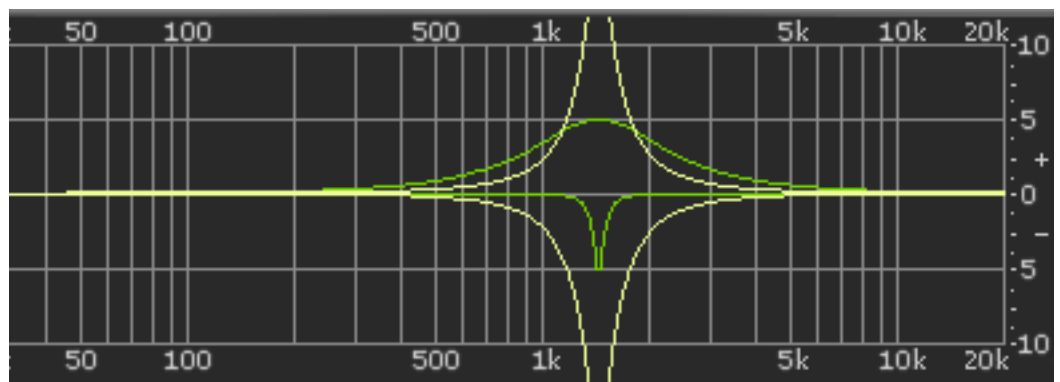


Figure 63. Parametric Type II response

Type III

When set to Type III, there is a Q/Gain dependency on boost and attenuation. The bandwidth increases continuously as the gain is boosted and attenuated. The Q knob position determines the maximum Q at full gain.

Filter bandwidth is broader at lower gain settings and narrower at higher gain settings. This can produce a smoother, more natural response when adjusting filter gain.

Note that the Q value increases as gain is increased but the knob position does not change. The Q value is approached as gain increases, and reaches the knob position at maximum gain. See [Figure 64](#).

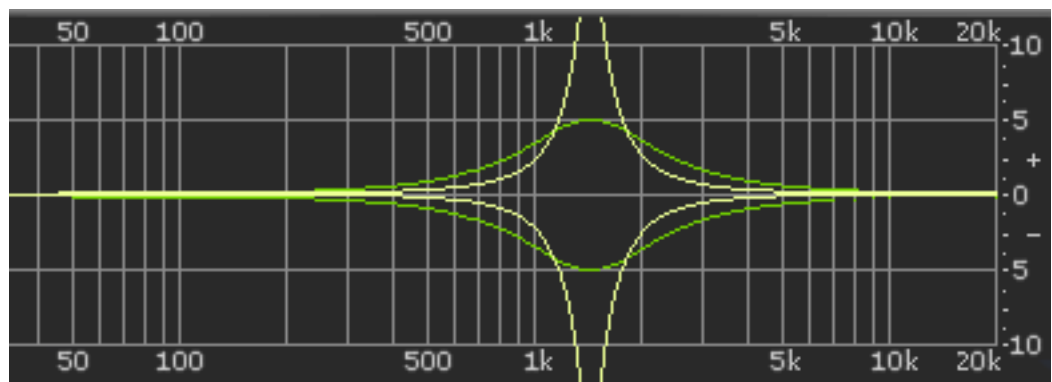


Figure 64. Parametric Type III response



Shelf EQ

Shelf Enable Button



Each band can be switched from parametric mode to shelf mode by clicking the shelf enable button. The button is off by default. To enable shelving on any band, click the shelf button.

The button is green when shelving is enabled. Additionally, the control bar associated with the band has a horizontal shelf indicator line in the response curve display (see [Figure 66 on page 146](#)) when shelf mode is active.

Shelf Type Button



When a band is in shelf mode and its Q is above the minimum value, a resonant peak occurs in the filter response. The Shelf Type button affects where this resonant peak occurs in relation to the shelf frequency.

Its purpose is to emulate the response curves of classic high-end analog mixing consoles. It's yet another tool to help you find the exact sound you are looking for.

The Shelf Type button places the resonant peak at (A) the edge of the stopband ([Figure 65 on page 146](#)), (B) the edge of the passband ([Figure 66](#)), or (C) at the edge of the stopband and the passband ([Figure 67](#)).

Shelf Q

When a band is in shelf mode, the Q knob sets the resonance of the band. The range of the Q knob is 0-100% when in shelf mode.

Note: When a band is in shelf mode, the Gain setting will affect the Q of the band.

When the Q is at its minimum value, there is no resonant peak. The resonance increases and becomes more prominent as the Q is increased. Therefore, for the shelf type to have any effect the Q must be above its minimum value.

Note: In order for this button to have any affect, the band must be in shelving mode, some gain must be applied, and the Q must be above its minimum value.

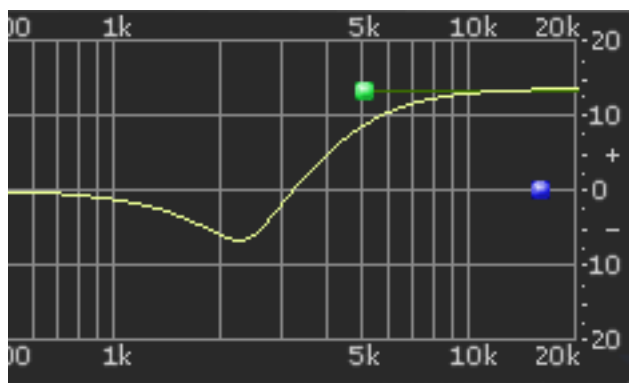


Figure 65. Shelf Type A

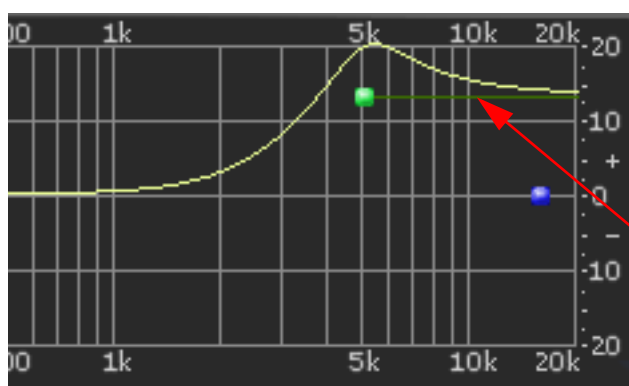


Figure 66. Shelf Type B

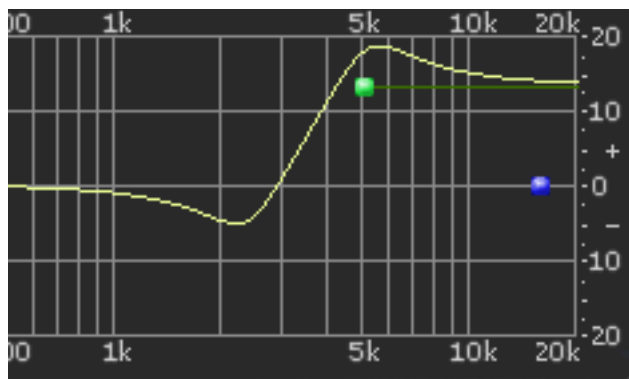


Figure 67. Shelf Type C

analog ears | digital minds

CHAPTER 10

DreamVerb

Overview

DreamVerb™, Universal Audio's flagship stereo reverb plug-in, draws on the unparalleled flexibility of RealVerb Pro. Its intuitive and powerful interface lets you create a room from a huge list of different materials and room shapes. These acoustic spaces can be customized further by blending the different room shapes and surfaces with one another, while the density of the air can be changed to simulate different ambient situations.

DreamVerb also features a flexible 5-band active EQ and unique level ramping for the early and late reflections for ultra-realistic dynamic room simulation. And with Universal Audio's proprietary smoothing algorithm, all parameters can be adjusted with automation or in real-time without distortion, pops, clicks, or zipper noise.

DreamVerb provides two graphic menus for selecting preset room shapes. The shapes can be blended according to the demands of your mix. Room materials are selected with two graphic menus containing preset Materials. A third menu specifies the air density for further spectral control. As with the room shapes, the materials and air can be blended as desired.

DreamVerb also includes intuitive graphic control over equalization, timing and diffusion patterns. To maximize the impact of your recording, we put independent control over the direct path, early reflections, and late-field reverberation in your hands.

Capitalizing on the psychoacoustic technology that went into the design of RealVerb Pro, we have incorporated some of these principles into DreamVerb. Our proprietary Stereo Soundfield Panning allows you to spread and control the signal between stereo speakers creating an impression of center and width. The ability to envelop your listener in a stereo recording is an entirely new approach to reverb design.

Screenshot

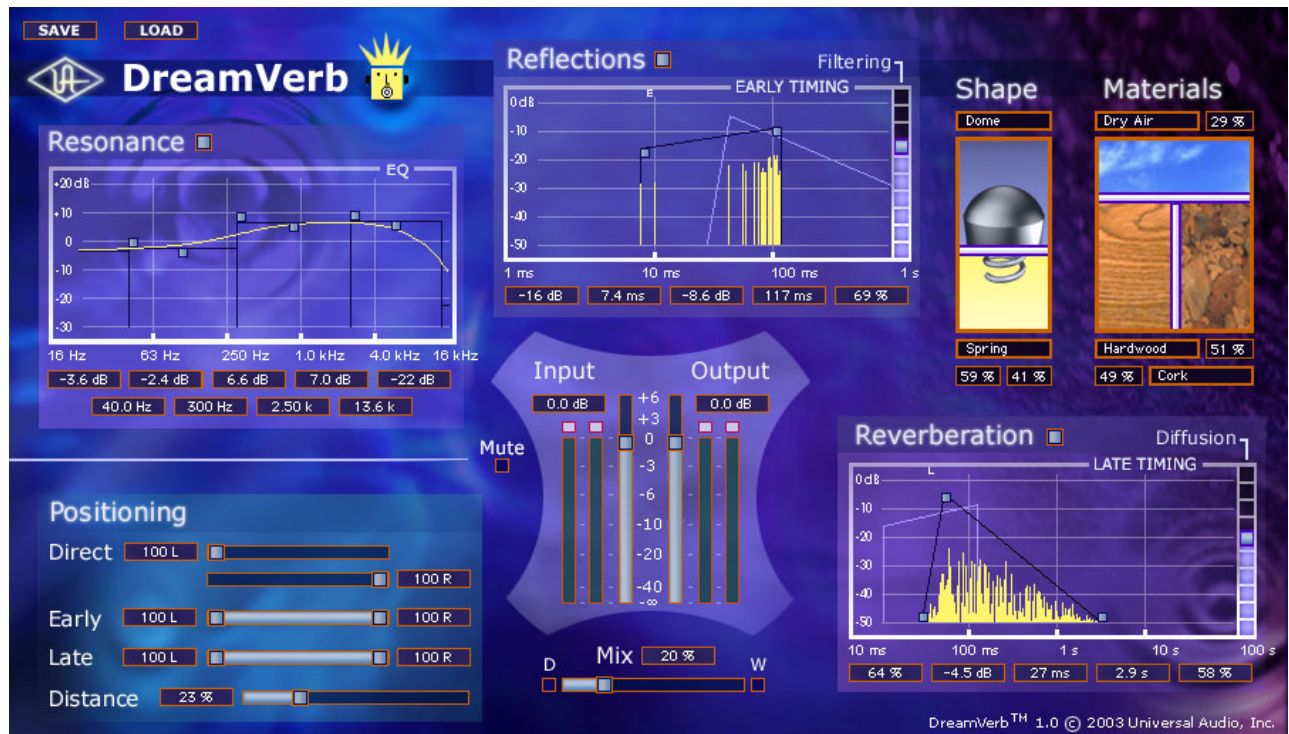


Figure 68. The DreamVerb plugin window

Signal Flow

Figure 69 illustrates the signal flow for DreamVerb. The input signal is equalized then delay lines are applied to the early reflection and late field generators. The resulting direct path, early reflection, and late-field reverberation are then independently positioned in the soundfield.

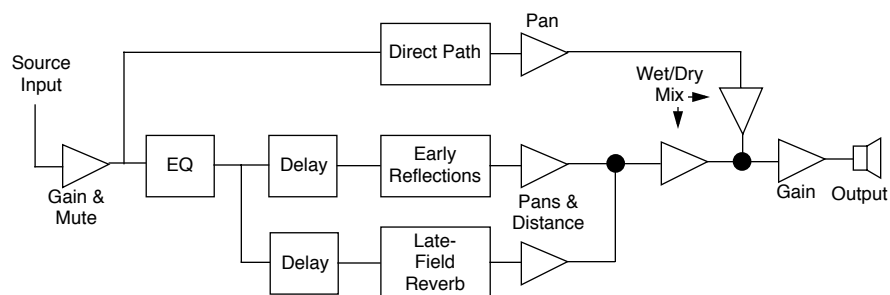


Figure 69. DreamVerb signal flow

The DreamVerb user interface (Figure 68 on page 148) is similarly organized. Reflected energy equalization is controlled with the Resonance panel. The pattern of early reflections (their relative timing and amplitudes) is determined by the room shapes in the Shape panel (Figure 72 on page 151). Early reflection pre-delay, slope, timing, and amplitude are specified in the Reflections panel (Figure 74 on page 156). The Materials panel (Figure 73 on page 153) is used to select relative late-field decay rates as a function of frequency. The late-field predelay, decay rate, room diffusion, slope, and level is specified in the Reverberation panel (Figure 75 on page 157). Finally, the Positioning panel (Figure 76 on page 159) contains controls for the placement of the source, early reflections, and late-field reverberation.

Resonance (Equalization) Panel

The Resonance panel (Figure 70 on page 150) is a five-band equalizer that can control the overall frequency response of the reverb, effecting its perceived brilliance and warmth. By adjusting its Amplitude and band Edge controls, the equalizer can be configured as shelving or parametric EQs, as well as hybrids between the two.

The EQ curve effectss the signal feeding both the early reflections and the late field reverberations, but not the direct path.

Bands 1 and 5 are configured as shelving bands. Bands 2, 3, and 4 also have an Edge control for adjusting its bandwidth.

Generally speaking, a lot of high-frequency energy results in a brilliant reverberation, whereas a good amount of low-frequency content gives a warm reverberation.

Note: *The values for the EQ parameters are displayed in the text fields at the bottom of the Resonance panel. The values can also be entered directly using the text entry method.*

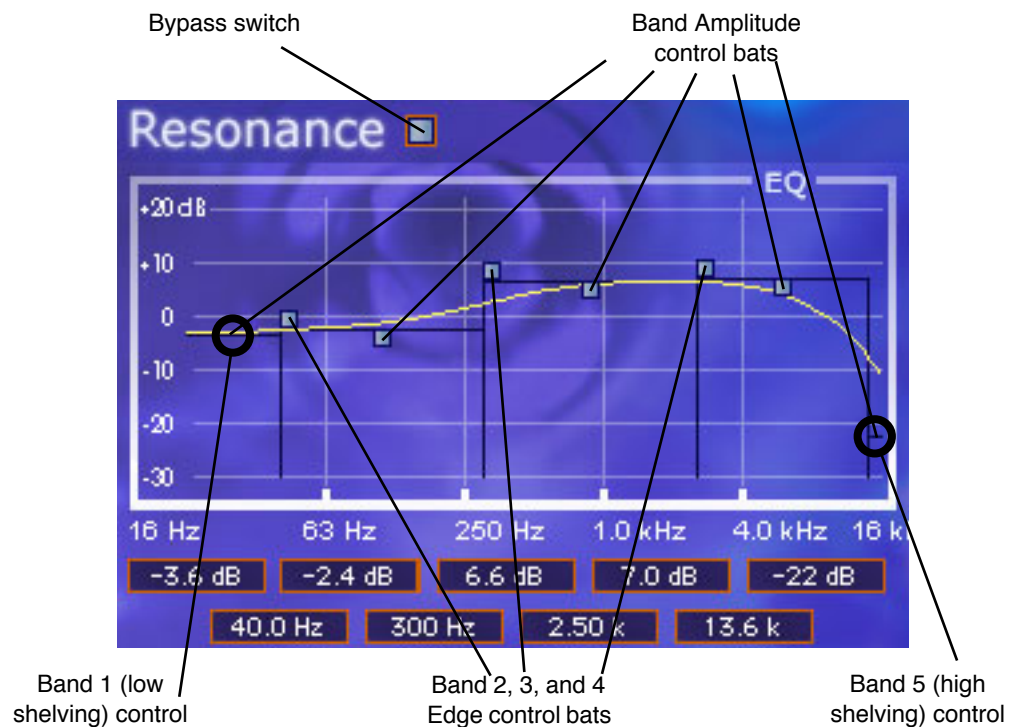


Figure 70. DreamVerb Resonance panel

Bypass

The equalizer can be disabled with this switch. When the switch is off (black instead of grey), the other resonance controls have no effect. This switch has no effect on the direct signal path.

Band Amplitude

Each of the five bands has its own amplitude (gain) control. The amplitude range of each band is -30dB to +20dB.

To adjust the amplitude of bands 2, 3, and 4, grab its control bat and drag vertically or use the direct text entry method. For bands 1 and 5, drag the horizontal line (these do not have a control bat).

Band Edge

Bands 2, 3, and 4 have an Edge control. This parameter effects the band-width of the band. To adjust the band edge, grab its control bat and drag horizontally or use the direct text entry method.

The effect of the band edge on the filter sound can depend upon the settings of the adjacent bands. For example, the sonic effect of this parameter is more pronounced if the amplitude of adjacent bands is significantly different than that of the band whose edge is being adjusted.

Shelving

The simplest (and often most practical) use of the equalizer is for low and/or high frequency shelving. This is achieved by dragging the left-most or right-most horizontal line (the ones without control bats) up or down, which boosts or cuts the energy at these frequencies.

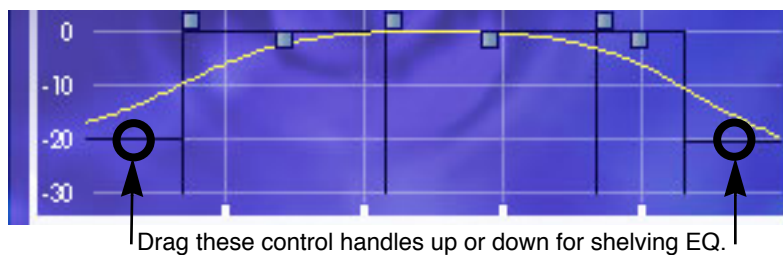


Figure 71. DreamVerb Resonance Shelving Bands

Shape Panel

The parameters in the Shape panel, in conjunction with the Materials panel (Figure 73 on page 153), effect the spatial characteristics of the reverb.

The pattern of early reflections in a reverb is determined by the room shape(s) and the ER start and end points. Two shapes can be blended from 0–100%. All parameters can be adjusted dynamically in real time without causing distortion or other artifacts in the audio. 21 shapes are available, including various plates, springs, rooms, and other acoustic spaces.

Note: The Shape parameters effect only the early reflections. They have no effect on the late field reverberation.

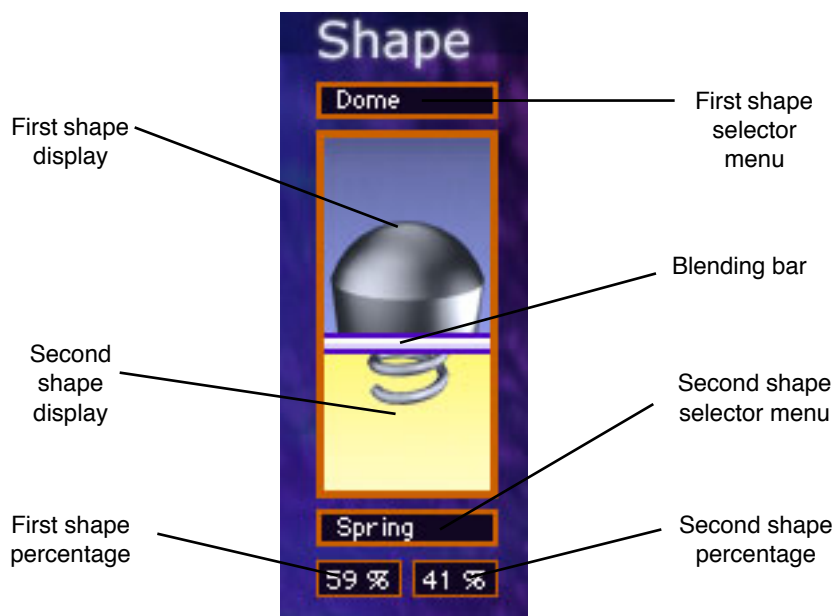
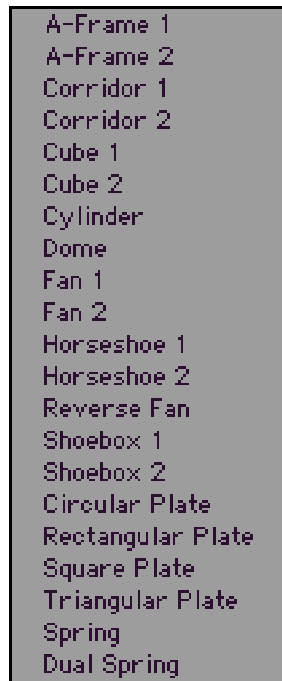


Figure 72. DreamVerb Shape panel

Shape Menus



A-Frame 1
A-Frame 2
Corridor 1
Corridor 2
Cube 1
Cube 2
Cylinder
Dome
Fan 1
Fan 2
Horseshoe 1
Horseshoe 2
Reverse Fan
Shoebox 1
Shoebox 2
Circular Plate
Rectangular Plate
Square Plate
Triangular Plate
Spring
Dual Spring

DreamVerb lets you specify two room shapes that can be blended to create a hybrid of early reflection patterns. The first and second shape each have their own menu. The available shapes are the same for each of the two shape menus.

The first shape is displayed in the upper area of the Shape panel, and the second shape is displayed in the lower area.

To select a first or second shape, click its shape pop-up selector menu to view the available shapes, then drag to the desired shape and release.

Shape Blending Bar

The Shape Blending Bar (see [Figure 72 on page 151](#)) is used to blend the two shapes together at any ratio. The two shapes are not just mixed together with this parameter; the early reflections algorithm itself is modified by blending.

Blend the early reflection patterns of the two rooms by dragging the Blending Bar. Drag the bar to the bottom to emphasize the first shape; drag to the top to emphasize the second shape.

The relative percentages of the two rooms appear at the bottom of the Shape panel. To use only one room shape, drag the Blending Bar so a shape is set to 100%.

The resulting early reflection pattern is displayed at the top of the Reflections panel ([Figure 74 on page 156](#)), where each reflection is represented by a yellow vertical line with a height indicating its arrival energy, and a location indicating its arrival time.

Materials Panel

The parameters in the Materials panel, in conjunction with the Shape panel (Figure 72 on page 151) and Reverberation panel (Figure 75 on page 157) effect the spatial characteristics of the reverb.

The material composition of an acoustical space effects how different frequency components decay over time. Materials are characterized by their absorption rates as a function of frequency—the more the material absorbs a certain frequency, the faster that frequency decays.

Note: While materials are used to control decay rates as a function of frequency, the overall decay rate of the late-field reverberation is controlled from the Reverberation panel (see Figure 75 on page 157).

24 real-world materials are provided, including such diverse materials as brick, marble, hardwood, water surface, and audience. Also included are 24 artificial materials with predefined decay rates, and seven air densities.

Note: The parameters in the Materials panel always effect the late-field reverberations. However, the materials parameters effect the early reflections ONLY if the “Filtering” parameter in the Reflections panel (Figure 74 on page 156) is set to a non-zero value.

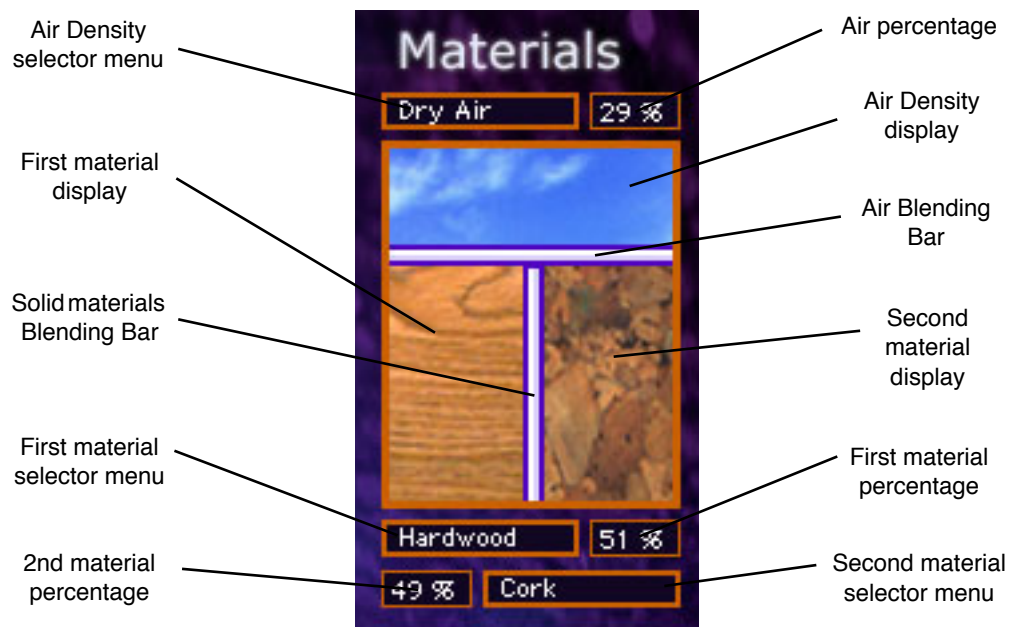


Figure 73. DreamVerb Materials panel

Materials Menus

DreamVerb lets you specify two room materials, which can be blended to create a hybrid of absorption and reflection properties. The first and second room material each has its own menu. The available materials are the same for each of the two materials menus.

The first material is displayed in the lower left area of the Materials panel, and the second material is displayed in the lower right area.

To select the first or second material, click its material pop-up selector menu to view the available materials, then drag to the desired material and release.

For a discussion of the various materials, see [“About the Materials” on page 69](#).

In addition to the “perfect” materials marked with a K, DreamVerb provides “J” materials that are not found in RealVerb Pro. These perform the inverse of the “K” materials. The materials marked with a J preferentially absorb low frequencies; they give the selected decay time at high frequencies, and a much shorter decay time at low frequencies.

Air Density Menu

DreamVerb allows you to specify the density of the air in the reverberant space with this menu, enabling another dimension of sonic control.

Ideal Gas
Dry Air
Humid Air
Smokey Club
Thick Fog
Inverse Air
Inverse Thick Fog

The more dense the air is, the more it absorbs high frequencies. At the top of the Air Density menu is Ideal Gas, where no frequencies are absorbed. The air quality increases in density with each selection as you go down the menu.

Inverse Air and Inverse Thick Fog absorb more low frequencies instead of high frequencies.

Air
Audience
Brick
Carpet
Cellulose
Concrete Block
Cork
Drapes
Fiberglass
Glass Window
Grass
Gravel
Hardwood
Linoleum
Marble
Painted Block
Plaster On Brick
Plaster On Concrete
Plate Glass
Plywood
Sand
Seats
Soil
Water Surface
K0175 Hz
K0250 Hz
K0350 Hz
K0500 Hz
K0700 Hz
K1.00 KHz
K1.40 KHz
K2.00 KHz
K2.80 KHz
K4.00 KHz
K5.60 KHz
K8.00 KHz
J0175 Hz
J0250 Hz
J0350 Hz
J0500 Hz
J0700 Hz
J1.00 KHz
J1.40 KHz
J2.00 KHz
J2.80 KHz
J4.00 KHz
J5.60 KHz
J8.00 KHz

Materials Blending Bars

The Materials Blending Bars (see [Figure 73 on page 153](#)) are used to blend the three materials together at any ratio. The materials are not just mixed together with the bars; the reverberation algorithm itself is modified by blending.

Materials Blending

Blend the two materials by dragging the vertical Blending Bar horizontally. Drag the bar to the right to emphasize the first material; drag to the left to emphasize the second material.

The relative percentages of the two materials appear next to each menu in the Materials panel. To use only one material, drag the Blending Bar so a material is set to 100%.

Air Blending

Blend the air density with the materials by dragging the horizontal Blending Bar vertically. Drag the bar to the top to emphasize the solid materials; drag to the bottom to emphasize the air.

The percentage of air used appears next to the Air Density menu. To use only solid materials, drag the horizontal Blending Bar to the top so air is set to 0%. To use only air, drag the horizontal Blending Bar to the bottom so air is set to 100%.

Reflections Panel

The Reflections panel ([Figure 74 on page 156](#)) offers control over the timing and relative energies of the reverb early reflections (ER). These parameters effect the reverb's perceived clarity and intimacy. Each early reflection is visually represented by a yellow vertical line with a height indicating its arrival energy and a location indicating its arrival time.

Unique to DreamVerb is independent control of the amplitude at the early reflection start and end points which facilitates envelope shaping of the reflections. This allows the ability to fade-in or fade-out the reflections to more accurately emulate acoustic environments or for special effects.

Note: *The values for the Start and End bars are displayed in the text fields at the bottom of the Reflections panel. These values can also be entered directly using the text entry method.*

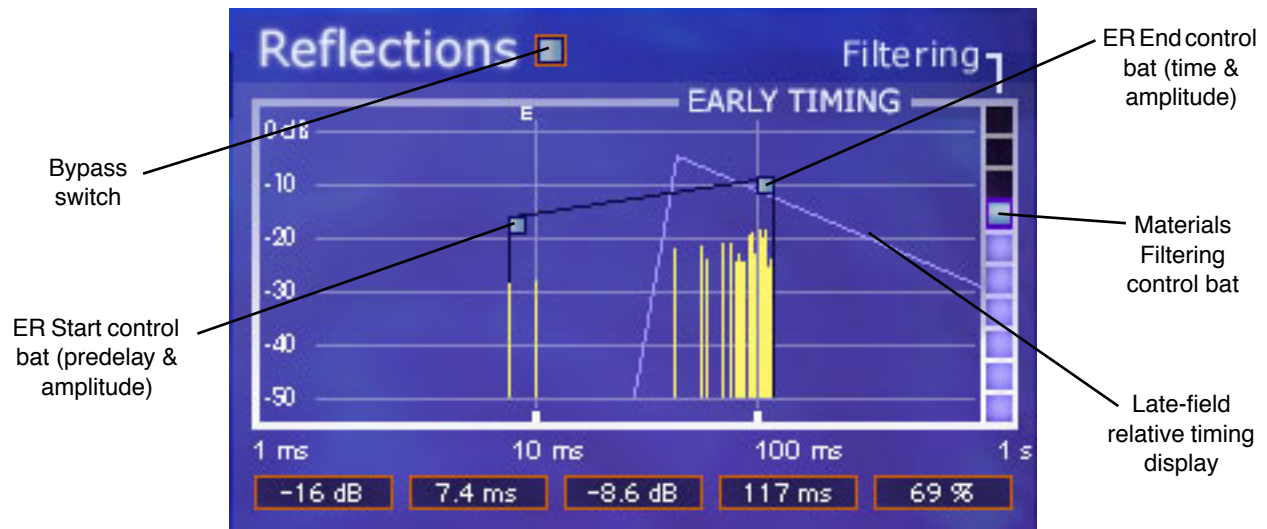


Figure 74. DreamVerb Reflections panel

- Bypass** The early reflections can be disabled with this switch. When the switch is off (black instead of grey), the other Reflections controls have no effect. This switch has no effect on the direct signal path.
- Reflections Start** This bat controls two early reflections start parameters. Dragging the bat horizontally controls the ER predelay (the delay between the dry signal and the onset of the ER). Dragging it vertically controls the amplitude of the reflections energy at the ER start time.
- Reflections End** This bat controls two ER end point parameters. Dragging the bat horizontally controls the ER end time (the time at which the ER is no longer heard). Dragging it vertically controls the amplitude of the reflections energy at the end point.
- Filtering** This parameter determines the amount of filtering from the Materials panel to be applied to the early reflections. The Materials effect upon the ER is most pronounced when Filtering is set 100%.
- Note:** The parameters in the Materials panel have no effect on the early reflections unless this parameter value is above 0%.

Late-Field Relative Timing

To highlight the relative timing relationship between the early reflections and late-field reverberation components, the shape and timing of the late-field is represented as an outline in the Reflections panel. The shape of this outline is modified by parameters in the Reverberations panel, not the Reflections panel.

Reverberation Panel

The Reverberation panel ([Figure 75](#)) contains the parameters that control the late-field (LF) reverb tail for DreamVerb.

The primary spectral characteristics of the late-field reverberation is determined by the parameters in the Materials panel ([page 153](#)) in conjunction with the Reverberation panel settings.

Note: The values for the late-field controls are displayed in the text fields at the bottom of the Reverberations panel. These values can also be entered directly using the text entry method.

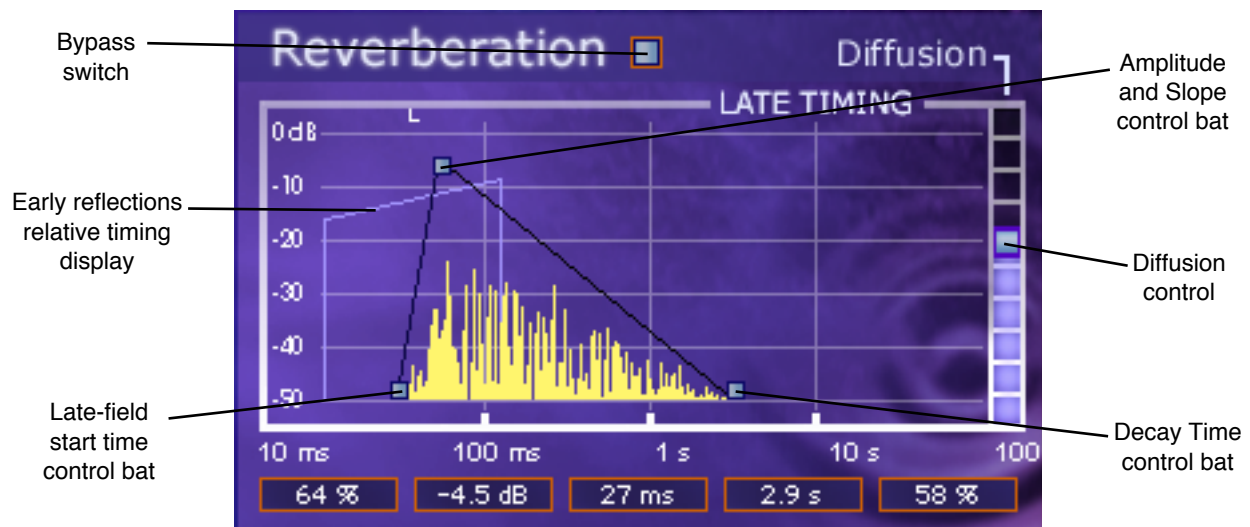


Figure 75. DreamVerb Reverberation panel

Bypass

The late-field reverberations can be disabled with this switch. When the switch is off (black instead of grey), the other Reflections controls have no effect. This switch has no effect on the direct signal path.

Late-Field Start	This parameter defines when the late-field reverb tail begins (the delay between the dry signal and the onset of the LF) in relation to the dry signal.
Amplitude & Slope	This bat controls two late-field parameters. Dragging the bat vertically controls the maximum amplitude of the LF reverb energy. Dragging it horizontally controls the LF slope (fade-in) time.
Decay Time	This control effects the length of the reverb tail. Drag the bat to the left for a short decay, or to the right for a long decay.
Diffusion	This slider effects how quickly the late-field reverberations become more dense. The higher the Diffusion value, the more rapidly a dense reverb tail evolves.
ER Relative Timing	To highlight the relative timing relationship between the early reflections and late-field reverberation components, the shape and timing of the early reflections is represented as an outline in the Reverberation panel. The shape of this outline is modified by parameters in the Reflections panel, not the Reverberation panel.

Positioning Panel

DreamVerb has the ability to separately position the direct path, early reflections, and late-field reverberation. The Positioning panel ([Figure 76 on page 159](#)) provides panning controls for each of these reverb components. In addition, a proprietary Distance control adjusts perceived source distance. These controls allow realistic synthesis of acoustic spaces—for instance listening at the entrance of an alley way, where all response components arrive from the same direction, or listening in the same alley next to the source, where the early reflections and reverberation surround the listener.

Note: When DreamVerb is used in a mono-in/mono-out configuration, all Positioning controls except Distance are unavailable for adjustment.

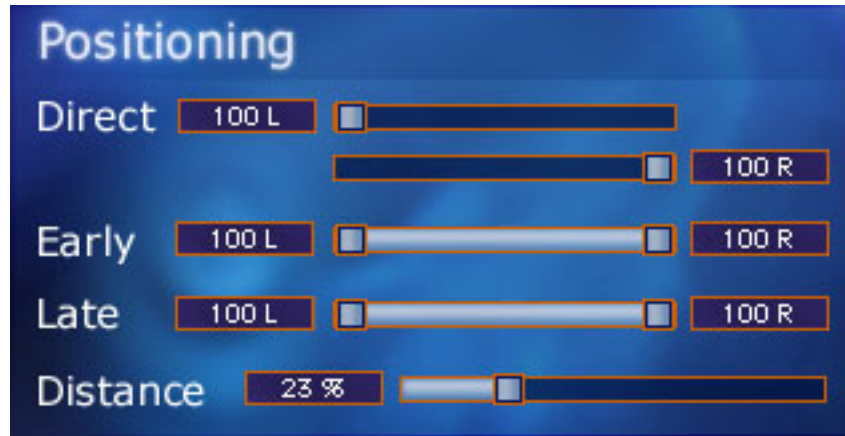


Figure 76. DreamVerb Positioning panel

Direct

These two sliders control the panning of the dry signal. The upper Direct slider controls the left audio channel, and the lower Direct slider controls the right audio channel.

A value of <100 pans the signal hard left; a value of 100> pans the signal hard right. A value of <0> places the signal in the center of the stereo field.

Note: If the DreamVerb “Mix” parameter (page 160) is set to 100% wet or the Wet button is active, these sliders have no effect.

Early

This slider, which contains two control handles, adjusts the stereo width of the early reflections.

Late

This slider, which contains two control handles, adjusts the stereo width of the late-field reverberations.

Early & Late Adjustment

The left and right slider handles are dragged to adjust the stereo width. For a full stereo spread, drag the left handle all the way to left and right handle all the way to the right. When the slider handles are not set to maximum width, the center of the slider can be dragged left or right to set the positioning of the signal.

To pan a mono signal hard left or right, drag the slider all the way to the left or right.

Distance

DreamVerb allows you to control the distance of the perceived source with this slider. In reverberant environments, sounds originating close to the listener have a different mix of direct and reflected energy than those originating further from the listener.

Larger percentages yield a source that is farther away from the listener. A value of 0% places the source as close as possible to the listener.

Levels Panel

This panel is where DreamVerb input/output levels, wet/dry mix, and reverb mute controls can be modified.

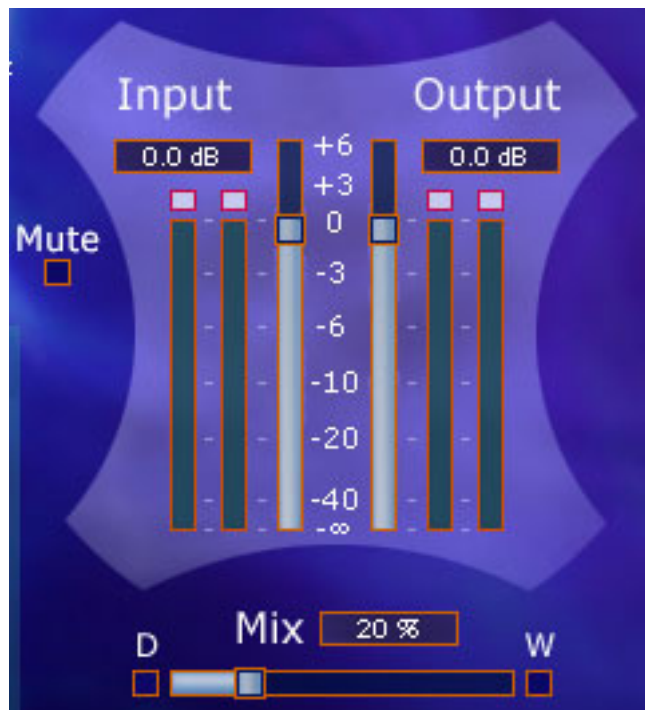


Figure 77. DreamVerb Levels panel

Input

Modifies the signal level at the input to DreamVerb. A value of zero is unity gain.

Output

Modifies the signal level at the output of DreamVerb. A value of zero is unity gain.

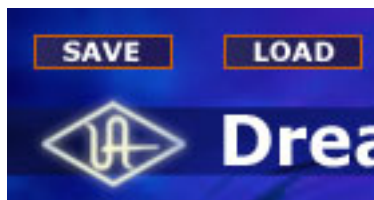
- Mute** This switch mutes the signal at the input to DreamVerb. This allows the reverb tail to play out after mute is applied, which is helpful for auditioning the sound of the reverb. Mute is on when the button is gray and off when the button is black.
- Mix** The wet and dry mix of DreamVerb is controlled with this slider. The two buttons above this slider labeled “D” and “W” represent Dry and Wet; clicking either will create a 100% dry or 100% wet mix.
- Dry** When this button (labeled “D”) is enabled, DreamVerb is 100% dry. It has the same effect as moving the Mix slider to 0%. Dry is on when the button is gray and off when the button is black.
- Wet** When this button (labeled “W”) is enabled, DreamVerb is 100% wet. It has the same effect as moving the Mix slider to 100%. Wet is on when the button is gray and off when the button is black.

DreamVerb Preset Management

- Factory Presets** In the preset menu there is a bank of 32 factory presets. Presets modified in the bank are saved when another preset within the bank is selected.
- Edits to any and all presets in the list are maintained separately within each instance of a plugin in a session.

To return to the default factory bank settings, reload the factory bank.

Save/Load



Not all plugin hosts include a method for loading and saving plugin settings. DreamVerb includes Save and Load buttons within the graphical interface itself to accommodate hosts that do not have this feature.

The Save/Load feature in DreamVerb supports presets but not banks. To save and load banks, use the host's bank management feature (if available).

Default Preset Location

When the Save and Load buttons are used within DreamVerb, the file open and save dialogs default to the same location each time.

PC

On Windows systems, the default preset location is inside the Presets directory, which is created within the directory selected during software installation. For example, if the default location was specified when running the UAD-1 Powered Plug-Ins Installer, the location would be:

C:\Program Files\Universal Audio\Powered Plugins\Presets

This default location can be changed during installation; the new location will be remembered as the default.

Mac OS

On OS 9 systems the default preset location is:

System Folder:Application Support:Universal Audio:Presets

On OS X systems the default preset location is:

Library:Application Support:Universal Audio:Presets

Using Host Application Management

Most host applications include their own method of managing plugin presets. For example, the currently selected preset is saved in Cubase/Nuendo when "Save Effect" is used. The solo/mute buttons (wet, dry, input) are not saved.

All presets and programs are saved when "Save Bank" is used in the host. They are also saved in the session file for each instance of the plugin.

Editing the name modifies the current preset's name. The new name will appear in all preset selection lists and will be saved with the session, bank, or effect.

Spatial Characteristics

Size

The apparent size of a reverberant space is dependent on many factors. Most reverbs on the market have a "size" parameter, which usually modifies several facets of the reverb algorithm at once. You may notice DreamVerb does not have a "size" parameter. Instead, the elements that control the reverberant space are available to the user.

In DreamVerb, room size is determined by the interaction between all the parameters in the Reflections and Reverberation panels. To get a larger-sounding space, increase the T60 (reverberation time), use proportionally more air, increase the pre-delays, and slightly shift the Resonance transition frequencies to lower values.

Pre-Delay

Intimacy and remoteness are largely controlled by the pre-delays. Generally speaking, use shorter pre-delays for more intimate spaces. Clear spaces have most of their energy in the first eighty milliseconds or so; muddy spaces have a lot of late arriving energy.

Space

In some sense, Shape determines the spatial characteristics of the reverberator, whereas Materials effects the spectral characteristics.

Preset Design Tips

Here are some practical tips for creating useful reverbs with DreamVerb. These are not rules of course, but techniques that can be helpful in designing the perfect sonic environment.

ER = Early Reflections	Hf = High frequency
LF = Late-field Reverberation	Lf = Low frequency

General Tips (a tour):

- Start by setting a general timing on the ER and LF graphs to give a rough reverb size. This timing ordinarily needs to be tweaked several times along the way.
- The materials and air density define the frequency decay of the LF, and also the coloration of the ER if ER filtering is used (the slider on the right of the Reflections panel).
- Typically, materials should be blended. Try blending contrasting high frequency roll-off materials with high-frequency reflecting materials or inverse materials. This tends to add nice dimension to the LF tail. Start with one useful material and experiment with blending.
- Materials can have an extreme filtering effect if no air density is used. Most presets sound better with an air blending. If you don't want the additional coloration of air, blend with "Ideal Gas" which performs no filtering.
- The room shapes define the ER pattern; they do not effect the LF. Solo the ER and choose a shape that works well for your source or environment.
- Blending shapes does not always yield desirable results. Use shape blending with discretion, or to define a more complex room.
- Start with the EQ flat, set the approximate sound with the materials, then EQ the input to cut or boost specific frequencies.

- The EQ is often most useful for a simple Lf or Hf roll-off/boost, or to notch out bothersome frequencies for particular sources. For full mix ambience/mastering presets, use the EQ to cut most of all LF input, which yields added ambience without mucking up the mix. This is a powerful EQ, so experiment!
- Try different diffusion settings for your preset (the slider on the right of the Reverb panel). Diffusion radically alters the reverberation sound and is source dependent. Higher diffusion values yield a fuller sound, good for percussive sounds; lower diffusion values yield a less dense sound, good for vocals, synths, etcetera.
- When monitoring your preset, try switching from Dry solo, Wet solo, and a useful mix. Solo the reflections and reverberation, and disable/enable EQ. Try different sources and mixes. Reach for the headphones every now and then. In general just keep things moving, as ear fatigue can be particularly deceiving with reverb sounds.
- The Positioning panel is generally only needed for automation. Ignore these settings for preset design unless going for a panning effect or monitoring real-world use.
- Often when you've got a really great preset designed, all it takes are a few subtle changes to make a number of other great presets.

Tips for designing a natural environment sound:

- Make timing proportional. As the size of the simulated environment increases, the length of the pre-delay for the EF, LF, and LF tail should increase proportionally. Typically, ER and LF pre-delay should be not too far apart, with LF starting shortly after ER.
- Place the ER timing preceding/leading into the LF
- ER amplitude naturally decays. Slope the amplitude down from left to right.
- Use ER filtering, as this improves the reverb sound in almost all situations.
- Try a gradual Lf or Hf roll-off (or boost) with the EQ section. The left and right-most EQ bands are shelf filters, which are perfect for this job. The adjacent bands can be used to shape the roll-off.
- Try natural materials and air densities before the unnatural custom or inverse materials and air densities.
- Try adding onset (slope) to the LF, as many environments naturally have an LF onset.

For additional info:

- Read [Chapter 4, "RealVerb Pro"](#) (page 64) of this manual.

CHAPTER 11

Fairchild 670

Overview

In the annals of compressor history, the products produced by Fairchild are some of the best built and most highly prized on the vintage market. The most famous Fairchild products produced were the 660 and 670 compressor/limiters, which are famous for their fantastic sound quality.

The stereo Fairchild 670 is probably the “Holy Grail” of compressors in studio gear esoterica, not only because of its price (known to fetch \$30,000 and beyond on the vintage market), but also its extreme rareness and the difficulties in maintaining such a unit. With its 14 transformers, 20 vacuum tubes, 6 rack-space encompassing girth and weighing in at 65 pounds, the Fairchild 670 is truly the heavyweight champion of compression.

Originating from the early 1950’s, the design of the 670 uses a single push-pull stage of amplification with an extremely high control voltage. The Fairchild 670 is a variable-mu tube limiter. Variable-mu limiters are unique in that they use tubes for gain reduction, and not just as amplifiers. The audio path is quite simple, and compression happens directly in the audio path, rather than sending out to a separate compression circuit.

The unit can be used as a limiter or compressor, depending upon personal taste and program material. It can go from a 2:1 ratio as a compressor to a peak limiter with a 30:1 ratio. The unit can also be adjusted to operate anywhere between these two extremes using the Threshold and DC Bias controls.

The UAD Fairchild was created by meticulously modeling (down to the component level as usual) a carefully selected hardware unit. Our “golden unit” was the Fairchild 670 (SN #505) at Ocean Way Recording in Hollywood. The Fairchild was advertised as “The World Accepted Standard for Level Control” back in the 1950’s when it was originally sold. It is still revered for its extremely smooth, artifact-free sound, and now Universal Audio has made it easily accessible to you. And you don’t have to let it warm up for 30 minutes before use!

Fairchild Screenshot



Figure 78. The Fairchild plugin window

2 Compressors, 4 Modes

There are two compressors within the Fairchild 670. They can be used as dual L/R, dual mono/stereo, or they can be linked together and used on either the L/R or mono/stereo signals.

The mode in which the compressors operate is determined by the combination of the AGC switch and the Sidechain Link switch. See [“Fairchild Modes” on page 168](#) for detailed mode information.

Controls Overview

Most of the controls are associated with one or the other of the compressors, as opposed to being strictly associated with one channel of input/output (depends on active mode). These controls include Threshold, Time Constant, Bias Current Balance, and DC Bias.

There are two sets of controls that always work on the left and right signals: input level and output level. In Lat/Vert mode, left is the mono input, and right is the stereo input.

Controls on the main panel are all original Fairchild controls, except for the meter select switch which was used to calibrate bias currents on the original hardware. The hardware does not have provision for monitoring input/output or gain reduction levels. For the plugin, the 'zero' screw-slot control has been disabled, since the meter cannot become uncalibrated on the plugin.

Controls on the lower auxiliary panel are original controls, modifications, or additions as follows:

- The DC Bias controls are original controls, but were on the back of the hardware units.
- The sidechain link control is a common modification which had been performed on the unit we modeled.
- The Controls Link is a plugin-only switch.
- The output level controls are an addition for the plugin.



Parameter Labels

Some hosts use plugin parameter names in place of the graphical interface, for example when viewing automation or using control surfaces. Some Fairchild parameters are named with "A" and "B" instead of "L" and "R" because that's how the plugin operates; depending on the mode (ie lat/vert), the channel parameters are not always associated with the left and right channels.

Fairchild Modes

Dual Left/Right

In Dual L/R mode, the Fairchild operates as two monophonic compressors with completely independent controls for the left and right channels. There is no interaction between the left and right channels.

Lateral – Vertical

In Lat/Vert mode, the 670 acts on the lateral and vertical (the sum and difference) components of the two stereo channels. This is accomplished by first bringing the two stereo channels through a matrixing network which divides them into their respective lateral and vertical components, limiting these lateral and vertical components, then recombining them through a second matrixing network into the left and right channels again.

The L+R (mono) signal is sent to one compressor, and the L-R (stereo) signal is sent to the other. The two compressors work independently of each other, and after compression the L and R signals are recovered once again by sum and difference. This mode was created for use in mastering records, where the mono and stereo components of the signal are encoded in this way. It also has application for psychoacoustic processing, since the stereo panning will change dynamically in this mode.

Lat/Vert processing provides maximum useable level and efficient use of available groove space in phonograph mastering. This results in higher volume recordings with longer playing times.

Note: *The terms lateral (side-to-side) and vertical (up-and-down) refer to the mechanical modulations in a vinyl record groove that are transduced into electrical audio signals by the phonograph stylus and cartridge.*

Stereo, coupled left/right

In this mode, the left channel is fed to one compressor, and the right channel is fed to the other. The two compressors are constrained so that they both compress the same amount at any instant. This prevents transients which appear only on one channel from shifting the image of the output. Any big transient on either channel will cause both channels to compress. The amount of compression will be similar to the amount of compression for a transient which appears on both channels at the same time. Also, the attack and release times for the two compressors will be the same, and attack and release behavior will be the average of the settings for the two channels. Mono transients should have an effective attack time of about one half the attack time for transients on only one of the two channels.

Stereo, coupled mono/stereo

This mode, like stereo couple left/right, causes the two compressors to be linked together so that they always compress the same amount. But here, the inputs to the two compressors are fed with the mono and stereo components of the signal. This means that in general a transient which occurs in both channels will cause a bit more compression than a transient which only appears on left or right. The attack and release behavior is determined by the average of the settings for the two channels.

Modes Table

The mode in which the compressors operate is determined by the combination of the AGC switch and the Sidechain Link switch. The switch positions required for each mode is shown in [Table 18](#) below.

Table 18. Fairchild Operating Modes

AGC Switch	Sidechain Link	Operating Mode
Left - Right	Unlinked	Dual Left -Right (Dual Mono)
Lat - Vert	Unlinked	Lateral - Vertical (Dual mono - stereo)
Left - Right	Linked	Stereo, couple left - right
Lat - Vert	Linked	Stereo, couple mono - stereo

Gratuitous Question

Is there any reason I would want to link the two sidechains together and still have the settings for the two channels different?

Yes. Linking the sidechains simply prevents left-right image shifting. Threshold and input gains can be set independently to cause the compressor to be more sensitive to instruments which are panned to one side or the other. Output controls can be set separately in order to correct an overall image shift at the output.

Controls

Power Switch

This switch determines whether the plugin is active. When the Power switch is in the Off position, plugin processing is disabled and UAD-1 DSP usage is reduced.

Metering

VU Meters

There are two calibrated VU meters, one for each channel. What the Meter displays is determined by the Meter Switch.

Meter Select Switch

This switch determines what is displayed on the VU meters. If GR is selected, the meter will show gain reduction in dB for the corresponding compressor channel (which is not necessarily left or right; depends on the active mode).

If the AGC switch has been set to left/right, the GR shown will be for the left or right channel. If the AGC switch has been set to lat/vert, the GR shown will be for the mono or stereo channel. In GR mode, the upper labels show gain reduction in dB.

If the meter select switch is set to IN or OUT, then that meter will reflect the level of the right or left input or output signal (however, the meters are not calibrated).

Zero

On the hardware unit this screw adjusted the meter pointer to compensate voltage fluctuation and tube wear. Because the meter in the plugin cannot go out of calibration, this control is permanently disabled.

AGC Mode

This control determines whether the two compression channels will receive L/R or mono/stereo as the inputs. When used in conjunction with the Sidechain Link switch, the operating mode of the compressor can be modified.

See the “[Fairchild Modes](#)” on page 168 and [Table 18](#) on page 169 for detailed mode descriptions.

Left – Right

If Left – Right is selected and Sidechain Link is off, the compressor is in dual mono mode. If Sidechain Link is on, the mode is stereo, trigger left/right.

Lateral – Vertical

If Lat/Vert is selected and Sidechain Link is off, the compressor is in lateral/vertical mode and will receive mono/stereo as the inputs. If Sidechain Link is on, the mode is stereo, trigger mono/stereo.

Threshold

This continuously variable control determines the amount of compression to be applied. Turn clockwise for more compression. When fully counter-clockwise, the unit behaves as a simple unity gain line amplifier.

Time Constant

This 6-position switch provides fixed and variable time constants (attack and release times) to accommodate various types of program material. Positions 1-4 provide successively slower behavior, and 5 and 6 provide program dependent response. The values published by Fairchild for each position are in [Table 19](#) below. The actual measured times are a bit different, but the overall trend is the same.

Table 19. Fairchild Time Constants

Time Constant	Attack Time	Release Time
Position 1	200 microseconds	300 milliseconds
Position 2	200 microseconds	800 milliseconds
Position 3	400 microseconds	2 seconds
Position 4	800 microseconds	5 seconds
Position 5	200 microseconds	Program dependent: 2 seconds for transients 10 seconds for multiple peaks
Position 6	400 microseconds	Program dependent: 300 milliseconds for transients 10 seconds for multiple peaks 25 seconds for consistently high program level

Sidechain Link

When this control is set to Link, it causes the two channels of the compressor to compress equal amounts. This does not mean that the compressor will be equally sensitive to either channel however; that depends on the settings of the other controls. It simply means that the instantaneous amount of compression for the two channels will always be the same.

Balance

Balance controls the bias current balance, and always goes with one channel of the compressor, regardless of what the nearby 'metering' switch is set to. The point of perfectly calibrated bias currents is achieved when the "dot" in the screw slot is at 12 o'clock. At this setting, the amount of additive signal deflection ("thud") which happens due to an attack is minimized. Setting this control counter-clockwise from this position results in a thud of one polarity on transients, and going clockwise produces a thud of opposite sign.

DC Bias

DC Bias controls the ratio of compression as well as the knee width. As the knob is turned clockwise, the ratio gets lower and the knee gets broader. The threshold also gets lower as the knob is turned clockwise. The 'factory cal' tick mark should be aligned with the screw slot "dot" for factory specification.

It would probably be more technically accurate to say that this control simply changes the knee width, since no matter where it's set the ratio always approaches true limiting eventually. However, the knee becomes so broad that it becomes more practical to speak of the ratio changing, because for reasonable (<25 dB) amounts of compression, this is the case.

Channel Input Gain

This is a stepped attenuation control which always applies to left or right input, regardless of the AGC control setting. The steps are approximately 1 dB apart, with approximate unity gain coming at a value of 18, where the gain is -0.33 dB.

In Lat/Vert mode, left is the mono input, and right is the stereo input.

Output Gain

These controls always apply to the L and R channels, even when in Lat/Vert mode. The labels around the knobs are in dB and the controls are stepped, 49-position controls, with each step being separated by 0.5 dB.

Controls Link

This allows the two sets of controls for the interface to be linked. If the controls are given an offset while unlinked and the controls are subsequently linked, the offset is preserved up to the range of travel of the linked controls.



CHAPTER 12

Precision Limiter

Overview

The Universal Audio Precision Limiter™ is a single-band, look-ahead, brick-wall limiter designed primarily for mastering with program material. The easy-to-use Limiter achieves 100% attack within a 1.5ms look-ahead window, which prevents clipping and guarantees zero overshoot performance. Both the attack and release curves are optimized for mastering, which minimizes aliasing.

Since the Precision Limiter is a colorless, transparent mastering limiter—no up-sampling is used, nor does the UA Precision Limiter pass audio through any filters—audio remains untouched unless the compressor is working, in which case only gain is affected.

To really be considered a professional limiter, the metering needs to be superb. The Precision Limiter features comprehensive, high-resolution metering and conforms to the Bob Katz “K-System” metering specifications. This metering allows the user to see what is happening to audio with a great deal of accuracy, with simultaneous RMS and Peak metering and adjustable Peak Hold. And since we know how valuable good metering is, the plug-in can also be bypassed and used strictly as a high-resolution meter!

Key features include user-adjustable Release or intelligent Auto Release, which allows for fast recovery—minimizing distortion and pumping—and a unique selectable Mode switch, which allows you to delicately tailor the attack shape and control the “presentation” for different material. Mode A is the default shape, suitable for most material, while Mode B can be particularly useful on minimal and/or acoustic program material, yielding a more subtle touch.

UAD Precision Limiter is yet another indispensable UAD-1 tool for your audio arsenal.

Precision Limiter Screenshot



Figure 79. The Precision Limiter plugin window

Controls Overview

Control knobs for the Precision Limiter behave the same way as all UAD plugins. Input, Output, and Release values can be modified with text entry. See [“Text Entry” on page 24](#) for more information.

The Precision Limiter introduces a new control style for UAD plugins. For the Mode, Meter, Scale, and Clear parameters, click the parameter label, the value text, or the LED to toggle between available values.

Precision Limiter Controls

Input Knob

The Input knob controls the signal level that is input into the limiter. Increasing the input will result in more limiting as the input signal exceeds 0dB.

The default value is 0dB. The available range is -6dB to 24dB.

Output Knob

The Output knob determines the maximum level at the output of the plugin. This control does not affect the actual limiting.

The Precision Limiter always limits the signal to 0dB internally, and the actual output is set by attenuating this internal level. Likewise, the input control can drive the signal over 0dB to get more limiting.

If the Precision Limiter is the last processor in the signal path when mixing down to disk (bouncing), the Output value will be the level of the highest peak in the resultant audio file.

The default value is -0.10dB. The available range is from -12dB to 0dB. Non-zero values are always negative, therefore during text entry operations positive or negative values may be entered and the result will be negative.

Release Knob

The Release knob sets the value of the limiter release time. The default value is Auto. The available range is from 1 second to 0.01 milliseconds.

Auto Mode

When the Release knob is fully clockwise, Automatic mode is active. In Auto mode, release time is program-dependent. Isolated peaks will have a fast release time, while program material will have a slower release.

Note: You can type "A" or "a" to enter Auto mode during text entry.

Mode Switch

The Mode toggle switch affects the attack shape of the limiter. Subtle tonal variations are possible by switching the Mode between A and B.

Mode A is the default shape, suitable for most material, while Mode B can be particularly useful on minimal and/or acoustic program material, yielding a more subtle touch.

Power Switch

The Power Switch determines whether the plugin is active. When the Power switch is in the Off position, plugin processing is disabled and UAD-1 DSP usage is reduced. When the plugin is bypassed with this switch (but not by the host bypass), the VU meter displays the unprocessed input signal level.

Precision Limiter Meters Overview

The UAD Precision Limiter has precise, calibrated stereo metering. The main stereo Input/Output meter actually displays three meters simultaneously: The RMS and instantaneous peak levels, which follow the signal, and the "peak-hold" (also known as global peak) level.

The peak-hold level is the maximum instantaneous peak within the interval set by the Hold button, and is also displayed as text to the right of the meters. To reset the peak hold levels, press the Clear button.

This is the "K-System" metering specification as conceived by Bob Katz of Digital Domain. Detailed information can be found on the internet at:

- http://digido.com/portal/pmodule_id=11/pmdmode=fullscreen/pageadder_page_id=59
- (Click here to go directly to the page)

Precision Limiter metering is also active when plugin processing is deactivated with the Precision Limiter Power switch. Metering is disabled when the plugin is bypassed by the host application.

- Input Meter** When the Meter I/O switch is in Input mode, the main level meters display the signal level at the input of the plugin (and is not affected by the Input knob).
- Output Meter** When the Meter I/O switch is in Output mode, the main level meters display the level at the output of the plugin. When the Limiter is enabled, the Output and Input knobs will affect this display.
- Gain Reduction Meter** The Gain Reduction meter displays the amount of limiter gain reduction. More green bars moving to the left indicate more gain reduction is occurring.
- Gain reduction only occurs when the input signal level exceeds 0dB. Therefore, increasing the Input knob usually results in more gain reduction.
- Meter I/O Switch** The Meter I/O toggle switch specifies the signal source for the main stereo meter, either input or output.
- Meter Scale Switch** The Meter Scale switch increases the resolution of the main stereo level meter (See [Figure 80 on page 176](#)). Activating Zoom mode changes the low end of the scale from -60dB to -18dB.
- The main level meters in Normal mode, and the gain reduction meter in both Normal and Zoom modes, are linear (level differences between LED segments is the same). In Zoom mode, the main level meters use two different linear ranges for increased accuracy: -18 to -6dB (0.2dB per segment), and -6 to 0dB (0.1dB per segment).



Figure 80. Precision Limiter meter scale in Zoom mode

- Meter Hold Time Switch** The Meter Hold Time switch determines how much time will pass before the peak values for the main meter and the gain reduction meter are reset. It affects both the peak LED's and the peak text display.

Values of 3 seconds, 10 seconds, or Infinite (indicated by the lazy-8 symbol) can be selected.

Meter Peak Clear Switch

The Meter Peak Clear switch clears the meter peak value display. It affects both the peak LED's and the peak text display.

Precision Limiter Latency

The Precision Limiter has a 1.5ms look-ahead window to ensure clipping does not occur. This look-ahead function results in a slightly larger latency than other UAD-1 plugins. You may enter a value in the "Samples" parameter in DelayComp or TrackAdv to compensate. See ["Compensating for Precision Limiter" on page 48](#) for more information.

Note: *Compensating for Precision Limiter is not required if the host application supports "full delay compensation" throughout the signal path.*



CHAPTER 13

Plate 140

Overview

Universal Audio “steps up to the plate”, rendering yet another classic tool for the DAW that no mixer should be without: Introducing the Plate 140 Plug-In.

German company EMT made a breakthrough in 1957 with the release of the EMT 140, which utilized a resonating metal plate to create ambience. Nothing is quite like the wonderfully smooth sound of plate reverb that still endures as part of the fabric of modern music. However, plate reverb systems are expensive, bulky, need to be isolated from vibration and maintained regularly—therefore plates are usually found only in major studios.

Universal Audio faithfully recreates that unmistakable sound with the Plate 140 Plug-In. The Plate 140 replicates the sonic signature of three uniquely different EMT 140s found at The Plant Studios in Sausalito. That’s nearly two thousand pounds of sound in one plug-in! We thoughtfully combined the look of various elements from the EMT 140 system into one convenient panel, replicating the original damper controls for decay, and adding additional controls for the convenience of the modern DAW user.

Plate 140 Screenshot



Figure 81. The UAD Plate 140 plugin window

Plate 140 Controls

The Plate 140 interface is an amalgam of controls found at the plate amplifier itself and the remote damper controls, plus a few DAW-friendly controls that we added for your convenience. The GUI incorporates the original look and feel of those controls, and utilizes that look for the DAW-only controls.

Note: When adjusting parameters, keyboard shortcuts are available for fine, coarse, and other control methods. See [“Shortcuts”](#) on page 25.

Reverb



Plate reverb systems are extremely simple: A remote damper setting, and a high pass or shelf filter found at the plate itself. Additional manipulation is often used, including reverb return equalization, which is typically achieved at the console. Predelay is/was often achieved when necessary with tape delay, sending the return to a tape deck. Different tape speeds allowed different pre-delay amount.

The original damper controls are remote control devices, usually found somewhere near the control room for quick access. Our hybrid panel combines three remotes into the panel, with a switch to select each of the three available systems.

Note: The reverb controls (select and time) are completely independent from the other plugin controls (EQ, Predelay, Width, etcetera).

Plate Select Switch



Three plate models (algorithms) are available for reverb processing. This switch specifies which plate will be active.

Each setting is a model of a completely separate and unique plate system. Three 140's for the price of one!

Note: You can also switch the active plate by clicking the A, B, or C letters above the Plate Select switch and the Reverb Time meters.

Reverb Time Meters



The Reverb Time Meters display the reverb time of plates A, B, and C in seconds. The meter for the active plate model (as specified by the Plate Select switch) is illuminated.

Note: The meter value can be changed by dragging its “needle” in addition to its corresponding Damper controls.

Damper Controls (Reverb Time)



The Damper Controls change the reverb time for each plate. The range is from 0.5 to 5.5 seconds, in intervals of 0.1 sec.

Click the buttons to increment or decrement the reverb time.

EQ



This group of parameters contain the controls for Plate 140’s onboard utility equalizer. It is a two band (low and high) shelving EQ that uses analog-sounding algorithms for great tonal shaping options.

The EQ section is independent from the reverb algorithms and the low cut filter on the modeled plate systems. See [“Cut Filter” on page 184](#).

The frequency parameters specify the center of the transition band, which is defined as the frequency at which the level in dB is the midpoint between DC and the band edge level.

Note: There is one EQ per plugin instance. Each plate model (A, B, C) within a preset cannot have unique EQ values.

EQ Enable Switch

The Plate 140 equalizer can be disabled with the EQ Enable switch. UAD-1 DSP usage is not increased when EQ is enabled.

Low Frequency Knob

This parameter specifies the low shelving band transition frequency to be boosted or attenuated by the low band Gain setting. The range is 20Hz to 2kHz.

Because this is a shelving EQ, all frequencies below this setting will be affected by the low band Gain value.

Low Gain Knob

This parameter determines the amount by which the transition frequency setting for the low band is boosted or attenuated. The available range is ± 12 dB, in increments of 0.5dB (fine control) or 1.0dB (coarse control).

High Frequency Knob

This parameter determines the high shelving band transition frequency to be boosted or attenuated by the high band Gain setting. The range is 200Hz to 20kHz.

Because this is a shelving EQ, all frequencies above this setting will be affected by the high band Gain value.

High Gain Knob

This parameter determines the amount by which the frequency setting for the high band is boosted or attenuated. The available range is ± 12 dB, in increments of 0.5dB (fine control) or 1.0dB (coarse control).

Output VU Meter



The vintage-style VU Meter represents the plugin output level. It is active when the Power switch is on, and slowly returns to zero when Power is switched off.

Predelay Knob



The amount of time between the dry signal and the onset of the reverb is controlled with this knob. The range is 0.0 to 250 milliseconds.

This control uses a logarithmic scale to provide increased resolution when selecting lower values. When the knob is in the 12 o'clock position, the value is 50 milliseconds.

Width Knob



This control narrows the stereo image of Plate 140. The range is from 0 – 100%. At a value of zero, Plate 140 returns a monophonic reverb. At 100%, the stereo reverb field is as wide as possible.

Mix Knob



The Mix control determines the balance between the original and the processed signal. The range is from Dry (0%, unprocessed) to Wet (100%, processed signal only).

This control uses a logarithmic scale to provide increased resolution when selecting lower values. When the knob is in the 12 o'clock position, the value is 15%.

Note: *If Wet Solo is active, adjusting this knob will have no affect.*

Wet Solo Button



The Wet Solo button puts Plate 140 into "100% Wet" mode. When Wet Solo is on, it is the equivalent of setting the Mix knob value to 100% wet.

Wet Solo defaults to On, which is optimal when using Plate 140 in the "classic" reverb configuration (placed on an effect group/bus that is configured for use with channel sends). When Plate 140 is used on a channel insert, this control should be deactivated.

Note: *Wet Solo is a global (per Plate 140 plugin instance) control. Its value is not saved within presets.*

Power Switch



This toggle switch enables or disables Plate 140. You can use it to compare the processed settings to that of the original signal, or to bypass the plugin which reduces (but not eliminates) the UAD-1 DSP load.

Power Lamp

The red power indicator glows brighter when the plugin is enabled by the Power switch.



Hidden Controls

Several controls exist that are not available within the Graphical User Interface. They can only be viewed or modified in automation mode or “controls” mode.



Figure 82. Plate 140 in Controls mode

Note: *Not all host applications support automation and/or controls mode.*

Accessing

Each host application has its own particular operating methods. Consult the host application documentation for specific instructions on accessing automation parameters and controls mode.

If the host application does not support automation or controls mode, use the factory presets as starting points for your own custom presets.

Cut Filter

The Cut Filter is an equalizer that is used to reduce low frequency content in the reverb. On hardware plate systems, the cut filter setting is rarely modified because it is found at the plate amplifier unit itself and is not easily accessed from the control room.

Note: *There is one Cut Filter per plugin instance. Each plate model (A, B, C) within a preset cannot have a unique Cut Filter value.*

Plate 140 contains two types of cut filters: original EMT electronics, and Martech electronics which was/is a common plate system retrofit.

In the modeled source units at The Plant, plates A and B use the EMT electronics while Plate C utilizes the Martech electronics. In Plate 140, you can use either cut type with any of the three available plates.

Off
M 90Hz
M 125Hz
M 180Hz
M 250Hz
M 270Hz
M 360Hz
E -4dB
E -10dB
E -16dB

The values prefaced with an “E” designate the original electronics model. This is a cut filter centered at 80Hz, with three available levels of attenuation: -4dB, -10dB, and -16dB.

The values prefaced with an “M” designate the Martech electronics model. This is a shelf filter (all frequencies below the frequency are reduced). Six shelving frequencies are available: 90Hz, 125Hz, 180Hz, 250Hz, 270Hz, and 360Hz.

Balance Control

This stereo control balances the level between the left and right channels of the reverb return. Rotating the knob to the left attenuates the right channel, and vice versa (it is not a mono pan control).

Modulation

The Plate 140 reverb time can be modulated by a low frequency oscillator using rate and depth controls. The effect is subtle but it can increase dispersion and reduce ringing on some source material, such as loud signals with sudden endings and percussive content.

Mod Rate

Mod Rate controls the rate of reverb time modulation. The available range is from 0.01Hz to 1.0Hz.

Mod Depth

This parameter controls the amount of reverb time modulation. The available range is from 0 – 10 cents.

Note: *The Plate 140 distills 1800+ pounds of sound into a single plugin. Exercise caution when lifting.*

CHAPTER 14

Precision Equalizer

Overview

The Universal Audio Precision Equalizer™ is a stereo or dual-mono four band EQ and high-pass filter designed primarily for mastering program material. The Precision Equalizer may also be used in recording and mixing where the utmost in EQ quality is required. The Precision Equalizer is based on industry standard analog mastering filters, and uses the classic parametric controls arrangement. The Precision Equalizer utilizes the best from those designs while incorporating features convenient to digital mastering. To preserve the greatest sonic detail and ensure a minimum of artifacts in the upper frequency range, the Precision Equalizer is upsampled to 192khz.

Precision Equalizer Screenshot



Figure 83. The UAD Precision Equalizer plugin window

Precision Equalizer Controls

The easy to use Precision Equalizer features stepped controls throughout for easy recall. Both the left and right channels feature four bands of EQ, grouped in two overlapping pairs. There are two bands for low frequencies (L1 and L2), and two for highs (H1 and H2). There is also a shelving or peak/notch filter available for each band, along with five peak/notch (Q) responses per band. The high-pass filter is a far-reaching 18dB per octave, which enables precise filtering of power-robbing sub-harmonic content, or other creative uses.

The Precision Equalizer also features flexibility in auditioning. There are three separate EQ configurations, allowing selection of two complete sets of stereo parameters or the Dual mode when disparate channel adjustments are necessary. In addition, parameter values can be easily transferred between parameter groups using the Copy buttons.

Control Grouping



The L and R equalizer sections are independent groups of parameters, each controlling one side (left or right) of the stereo source signal.

The L and R controls are linked except when in Dual mode. In Dual mode, control groups L and R can be independently adjusted.

Modes



The Mode switches define the operating mode of Precision Equalizer. The currently active mode is indicated by a blue light. Each mode is detailed below.

Stereo Mode

In Stereo mode, the L and R equalizer sections both control one side of the stereo source signal. The L and R controls are linked in stereo mode.

In stereo mode there are two sets of EQ settings (referred to as A and B), with each set containing the full set of L and R parameter values (the high-pass filter value is global per preset). This feature enables easy switching between two EQ settings for comparison purposes. Both the A and B parameter sets are contained within a single Precision Equalizer preset.

Dual Mode

In Dual mode (dual-mono mode), the left and right parameters can be independently adjusted so that each side of the stereo signal can have different EQ settings. Note that this mode is infrequently used during mastering because phase, imaging, and level inconsistencies may be induced in the resulting stereo signal.

Mode Selection

Any of the below methods may be used to modify the Mode value:

- Click the Stereo button to cycle through modes A and B
- Click the Dual button to activate dual-mono mode
- Click the indicator light above each mode
- Click+hold+drag the indicator light above each mode.

Parameter Copy Buttons



The Parameter Copy buttons provide an easy method for copying parameter values. The behavior of the buttons is determined by the current operating mode of Precision Equalizer.

Note: *The values that existed at the destination before copying are lost.*

Stereo Mode

When in Stereo mode (see “[Stereo Mode](#)” on page 186), clicking A > B copies the left AND right parameter values from parameter set A to parameter set B, and clicking the A < B button copies all the values from parameter set B to parameter set A.

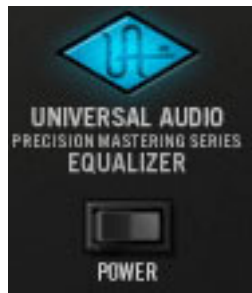
This feature is useful when you want to make an EQ change to a stereo signal while maintaining the original values so the two settings can be easily compared.

Note: *The high-pass filter parameter is global per preset and is not affected by this control.*

Parameter Copy in Dual Mode

When in Dual mode, the A and B buttons behave as left and right channel copy buttons. Clicking A > B copies all the values from the left channel parameters to the right channel parameters, and clicking A < B copies all the values from the right channel parameters to the left channel parameters.

Power Switch



The Power Switch determines whether the plugin is active. This is useful for comparing the processed settings to that of the original signal, or to bypass the plugin to reduce the UAD-1 DSP load.

Click the rocker switch to change the Power state. Alternatively, you can click the blue UA logo to toggle the Power state.

Band Controls

Each control set (L and R) has four EQ bands. Two bands are overlapping low frequency bands labeled L1 and L2, and two bands are overlapping high frequency bands labeled H1 and H2.

Each of the four bands has a control for bandwidth, enable, frequency, and gain. All four of the EQ bands can be used in parametric or shelf mode. The controls are exactly the same for each band; only the available frequency values differ.

Bandwidth Knob

The Bandwidth (Q) knob defines the proportion of frequencies surrounding the band center frequency to be affected by the band gain control.

The numbers represent the filter slope in dB per octave. The available selections are 4, 6, 9, 14, 20, and Shelf.

When set to Shelf on the L1 and L2 bands, the band becomes a low shelving filter. When set to Shelf on the H1 and H2 bands, the band becomes a high shelving filter.

Band Enable Button



Each band can be individually engaged with the Enable button. All bands default to disabled. When a band is enabled, the button glows blue. To enable a band, click the Enable button or move the band Gain knob.

You can use these buttons to compare the band settings to that of the original signal, or to bypass the individual band. UAD-1 DSP usage is slightly decreased when a band is disabled.

Frequency Knob

The Frequency knob determines the center frequency of the filter band to be boosted or attenuated by the band Gain setting.

This knob is stepped with 41 values for easy reproducibility during mastering. To double the resolution of the available knob values (for fine control), press the shift key on the computer keyboard while adjusting the knob. This increased 2x frequency resolution (within the available range) can also be specified using text entry, parameter automation, or 'controls' mode. The available values for each of the four bands is the same in both parametric and shelf modes, and are listed in [Table 20](#) below.

Note: *Not all host applications support automation and/or controls mode.*

Table 20. Precision Equalizer Band Frequency Ranges

Low Frequencies (L1 and L2)	19 – 572 Hertz
High Frequencies (H1 and H2)	617 – 27k Hertz

Gain Knob

The Gain knob determines the amount by which the frequency setting for the band is boosted or attenuated. The available Gain values are listed in [Table 21](#) below.

Table 21. Precision Equalizer band gain values

0.0dB	±2.0dB	±5.0dB
±0.5dB	±2.5dB	±6.0dB
±1.0dB	±3.0dB	±8.0dB
±1.5dB	±4.0dB	

High-Pass Filter



The high-pass filter is useful for reducing low frequency content. It is a global filter; it always affects both left and right channels, regardless of the active mode. See [Table 22](#) below for available settings.

Table 22. Precision Equalizer high-pass filter frequencies

Off (disabled)	40Hz
10Hz	60Hz
20Hz	80Hz
30Hz	100Hz

Precision Equalizer Latency

The Precision Equalizer uses an internal sample rate of 192kHz to facilitate its amazing sonic quality. This upsampling results in a slightly larger latency than other UAD-1 plugins. You may enter a value in the "Samples" parameter in DelayComp or TrackAdv to compensate. See ["Compensating for Precision Equalizer" on page 49](#) for more information.

Note: *Compensating for Precision Equalizer is not required if the host application supports "full delay compensation" throughout the signal path, or when it is used only on the outputs.*



CHAPTER 15

Roland CE-1

Overview

The Roland CE-1 Chorus Ensemble is another classic effect faithfully reproduced by our ace modeling engineers. The CE-1 is considered by many to be the definitive chorus effect, renowned for its rich and unique timbres.

Roland CE-1 Screenshot



Figure 84. The UAD Roland CE-1 plugin window

Roland CE-1 Controls

The Roland CE-1 has two operating modes, chorus and vibrato. Only one mode can be active at a time. The operating mode is set using the Vibrato/Chorus switch.

Clip LED



The red Clip LED illuminates when signal peaks in the plugin occur.

Normal/Effect Switch



This is an effect bypass switch. Click to enable/disable the chorus or vibrato effect. The effect that will be heard is determined by the Vibrato/Chorus switch.

The active state is black text. The inactive state has gray text. The default state is effect.

This is not a plugin bypass switch. The hardware CE-1 has a slight affect on the sound even when the effect is “bypassed” in normal mode. We have modeled the plugin faithfully and like the hardware unit, when the effect is bypassed with this switch, audio is still processed to sound like the CE-1 in “normal” mode. To disable audio processing, use the CE-1 Power Switch.

Rate LED



The yellow Rate LED blinks according to the current low-frequency oscillator (LFO) rate. When CE-1 is in Vibrato mode, the LFO rate is determined by the vibrato rate knob. When in Chorus mode, this LED is affected by the Intensity knob.

Note: In Chorus mode, the fastest LFO rate is slower than the slowest LFO rate in Vibrato mode.

Vibrato/Chorus Switch



This switch determines the operating mode of the plugin. Click to switch between chorus and vibrato modes.

The active mode is black text. The inactive mode has gray text. The default mode is chorus.

Stereo Mode Switch



The Stereo Mode switch determines the operating mode of CE-1 when the plugin is used in a configuration with stereo input, such as a stereo audio track insert or stereo effects bus.

The hardware CE-1 has only a monophonic input. Its output can be mono (wet and dry signal mixed at one output jack) or stereo (dry signal in one output jack, wet signal in other output jack). We've adapted the model for the modern era, enabling a true stereo input.

Note: This switch has no affect in a mono-in/mono-out or mono-in/stereo-out configuration.

When CE-1 is used in a stereo input configuration, the Stereo Mode switch affects the output as follows:

Dual Mode

In Dual mode the CE-1 behaves as a dual-mono device, functioning as two independent CE-1's, each running in mono mode on one side of the stereo signal.

The left output contains a mix of the dry left input signal and the processed left channel signal, while the right output contains a mix of the dry right input signal and the processed right channel signal. Additionally, the LFO's of the dual CE-1 channels are 90 degrees out of phase (quadrature) for maximum effect.

Classic Mode

In Classic mode, the CE-1 behavior is similar to that of a mono-in/stereo out configuration. The left and right channel inputs are mixed to mono, and the dry signal (mixed left and right channels) appear at the left output, and the wet effect signal appears at the right output.

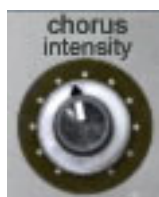
Output Level Knob



This knob determines the signal level at the output of the plugin. The range is 0 – 100%.

Note: This is not a wet/dry mix control.

Chorus Intensity Knob



When CE-1 is in chorus mode, the amount of chorusing effect is determined by this knob.

Note: When in vibrato mode, chorus intensity has no affect.

Vibrato Controls



These two knobs control rate and depth of the vibrato effect when CE-1 is in vibrato mode.

Depth Knob

The depth knob controls the intensity of the vibrato effect.

Rate Knob

The rate knob controls the rate of the vibrato LFO. The rate is indicated by the the Rate LED indicator.

Note: *When in chorus mode, the vibrato controls have no affect.*

Power Switch



This switch determines whether the plugin is active. This is useful for comparing the processed settings to that of the original signal, or to bypass the plugin to reduce the UAD-1 DSP load.

Click the rocker switch to change the Power state.



CHAPTER 16

History

LA-2A

The LA-2A leveling amplifier, a tube unit with hand wired components and three simple controls, was introduced in the mid-1960s. It utilized a system of electro-luminescent optical gain control that was quite revolutionary. Gain reduction was controlled by applying the audio voltage to a luminescent driver amplifier, with a second matched photoconductive cell used to control the metering section. With its 0 to 40 dB of gain limiting, a balanced stereo interconnection, flat frequency response of 0.1 dB from 30-15,000Hz and a low noise level (better than 70 dB below plus 10 dBm output), the LA-2A quickly became a studio standard. Originally patented by Jim Lawrence, it was produced by Teletronix in Pasadena, California, which became a division of Babcock Electronics Corporation. in 1965. In 1967 Babcock's broadcast division was acquired by the legendary Bill Putnam's company, Studio Electronics Corporation shortly before he changed the company's name to UREI®. Three different versions of the LA-2A were produced under the auspices of these different companies before production was discontinued around 1969.

1176LN

It was Bill Putnam himself who, in 1966, was responsible for the initial design of the 1176. Its circuit was rooted in the 1108 preamplifier which was also designed by Putnam. As is evident from entries and schematics in his design notebook, he experimented with the recently developed Field Effect Transistor (F.E.T.) in various configurations to control the gain reduction in the circuit. He began using F.E.T.s as voltage variable resistors, in which the resistance between the drain and the source terminals is controlled by a voltage applied to the gate. His greatest challenge was to ensure that distortion was minimized by operating the F.E.T.s within a linear region of operation.

After several unsuccessful attempts at using F.E.T.s in gain reduction circuits, Putnam settled upon the straightforward approach of using the F.E.T. as the bottom leg in a voltage divider circuit, which is placed ahead of a preamp stage.

The output stage of the 1176 is a carefully crafted class A line level amplifier, designed to work with the (then) standard load of 600 ohms. The heart of this stage is the output transformer, whose design and performance is critical. Its primary function is to convert the unbalanced nature of the 1176 circuit to a balanced line output, and to provide the proper impedance matching to drive the line impedance of 600 ohms. These two jobs are accomplished by the primary and secondary windings whose turns' ratio defines the impedance ratio.

This transformer is critical due to the fact that it uses several additional sets of windings to provide feedback, which makes it an integral component in the operation of the output amplifier. Putnam spent a great deal of time perfecting the design of this tricky transformer and carefully qualified the few vendors capable of producing it.

The first major modification to the 1176 circuit was designed by Brad Plunkett in an effort to reduce noise—hence the birth of the 1176LN, whose LN stands for low noise. Numerous design improvements followed, resulting in at least 13 revisions of the 1176. Legend has it that the D and E blackface revisions sound the most “authentic”.

The original Universal Audio 1176LN designed by Bill Putnam was a major breakthrough in limiter technology – the first true peak limiter with all transistor circuitry offering superior performance and a signature sound. Evolved from the popular Universal Audio 175 and 176 vacuum tube limiters, the 1176LN retained the proven qualities of these industry leaders, and set the standard for all limiters to follow.

analog ears | digital minds

Thank You

We would like to thank you again for becoming a Universal Audio customer. We urge you to fill out your registration card and send it back to us as soon as possible so we can keep you informed about new Powered Plug-In products that we will be releasing in the months to come.

We always like to hear from our customers and welcome your comments and suggestions. If you have any questions you can email us at:

info@uaudio.com

In case your audio toolbox needs might include hardware such our UA Classics series please be sure to have a look at our web site for more information about the entire UA family of products at:

- <http://www.uaudio.com>

The Universal Audio Team



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