

UAD-1 POWERED PLUG-INS

USER MANUAL

VERSION 2.1

Manual Version 020408D



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Notice

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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™ reverb. 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. Last, but certainly not least, is Nigel™ which 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.

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
- 24-bit, 96kHz support

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 zero 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)

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

Vintage Compressors

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

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 in one plug-in

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)
- Full 96kHz support

Preflex

- Exciting new 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
- User updatable amp models



System Requirements

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

Windows™

- Windows computer with available PCI slot
- Windows 98, ME, 2000, or XP
- 128 MB of RAM
- 50 MB of available disk space
- CD drive for software installation
- Steinberg VST-compatible host software, such as Cubase, Nuendo, or Emagic Logic Audio
- 1024 x 768 or higher resolution monitor
- An AGP graphics video adapter is recommended for optimum performance

Macintosh™

- Mac OS PowerPC computer with available PCI slot (G3 or better processor recommended)
- Mac OS 9.x
- 128 MB of RAM (256 MB recommended)
- 45 MB of available disk space
- CD drive for software installation
- Steinberg VST-compatible host software, such as Cubase, Nuendo, or Emagic Logic Audio
- 1024 x 768 or higher resolution monitor
- An AGP graphics video adapter is recommended for optimum performance

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.

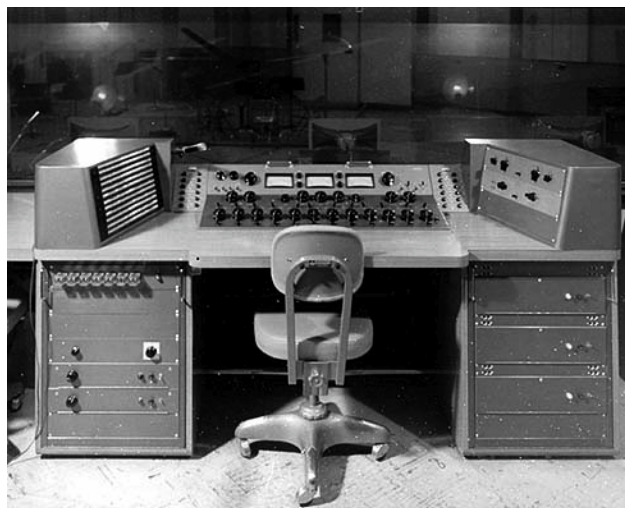
Mac OS

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.

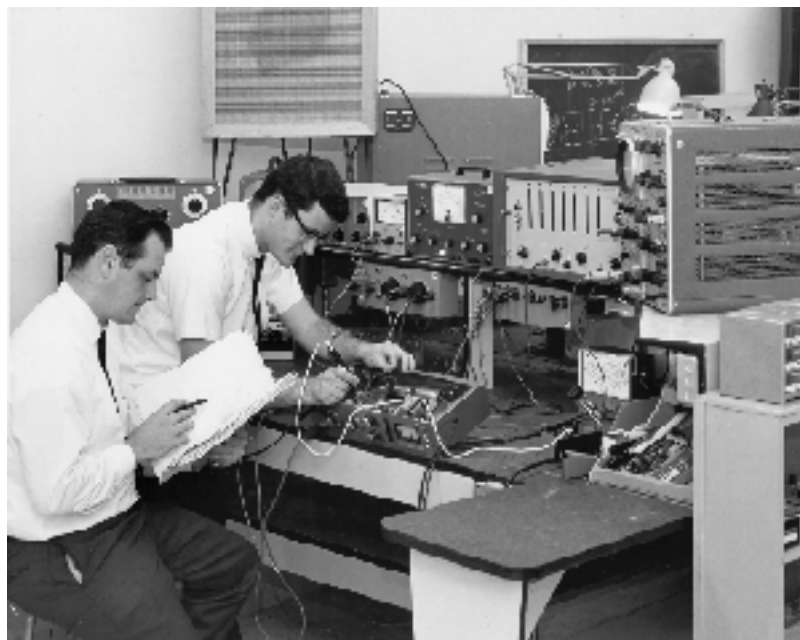
Please refer to the printed QuickStart documentation included in the product box for complete instructions on how to install and remove the software on each system.

The QuickStart guide is also included in Adobe's Portable Document Format ("pdf") in the downloadable version of the software bundle.

Install Software First

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

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. Hardware installation is the same for all platforms.

To install the UAD-1 DSP card:

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.

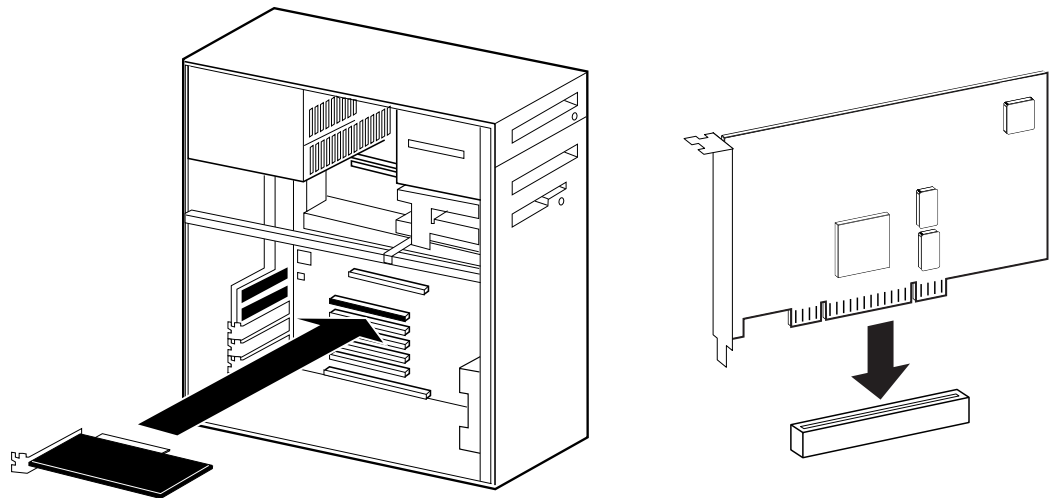


Figure 1. UAD-1 card installation

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 24-bit, 96KHz operation. Please note that Powered Plug-Ins running at 96KHz use twice as much UAD-1 resources than those used at 48KHz.

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.

Fine Control

A higher degree of knob resolution can be achieved by pressing the shift key while changing the knob value.

Null Settings

To return a parameter to its null (default) value, press the Control (Windows) or Command (Mac OS) key on the computer keyboard while clicking the parameter with the mouse.

Automation

Every UAD-1 Powered Plug-In parameter can be automated if this feature is supported by the VST 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.

Launching a UAD-1 Powered Plug-In

Each host application has its own particular methods for launching (instantiating) 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.

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 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.



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



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

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 icon at the upper left of the UAD-1 DSP Performance Meter window.

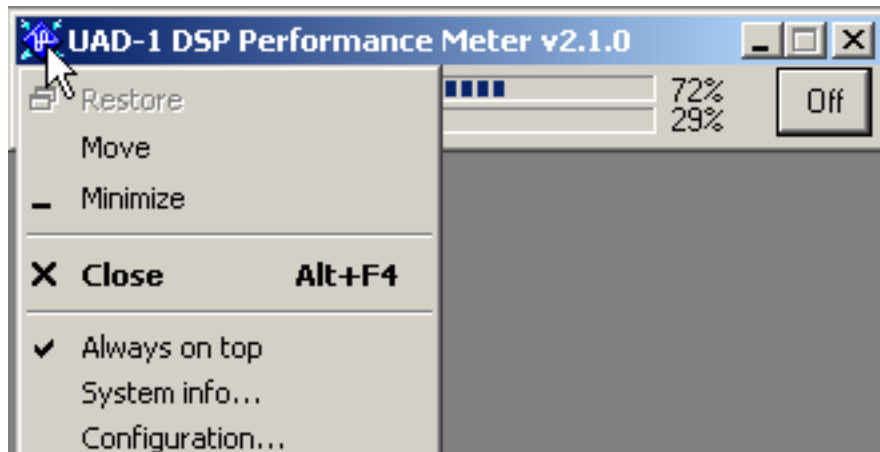


Figure 6. System menu for UAD-1 Performance Meter (Windows).

Mac OS

The File menu is available when the UAD-1 DSP Performance Meter is in the foreground. You can easily bring the Meter to the foreground by clicking the Bring to Front button (see [Figure 8 on page 21](#).)

When the UAD-1 Performance Meter application is in the foreground, toggle the 'Always on top' File Menu item (or type the command-T shortcut).

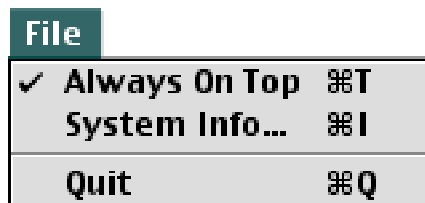


Figure 7. File Menu for UAD-1 Performance Meter (Mac OS).

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.

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 On/Off button.

On/Off Button

The On/Off button disables all UAD-1 Powered Plug-Ins that are currently running. This allows you to easily compare the sound of the processed and unprocessed audio.

Bring to Front Button (Mac OS)

To access the File menu of the Performance Meter application, the application must be in the foreground. The "Bring To Front" shortcut button makes this easy by immediately bringing the Performance Meter application to the foreground when clicked, saving a mouse trip to the Mac OS application menu.

Note: The "Bring To Front" button is only present 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.

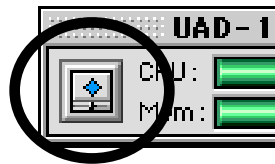


Figure 8. Performance Meter Bring to Front button (Mac OS).

UAD-1 System Information Window

The UAD-1 System Information window 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 Status: OK and UAD-1: OK, the card is operating properly.

Note: 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.

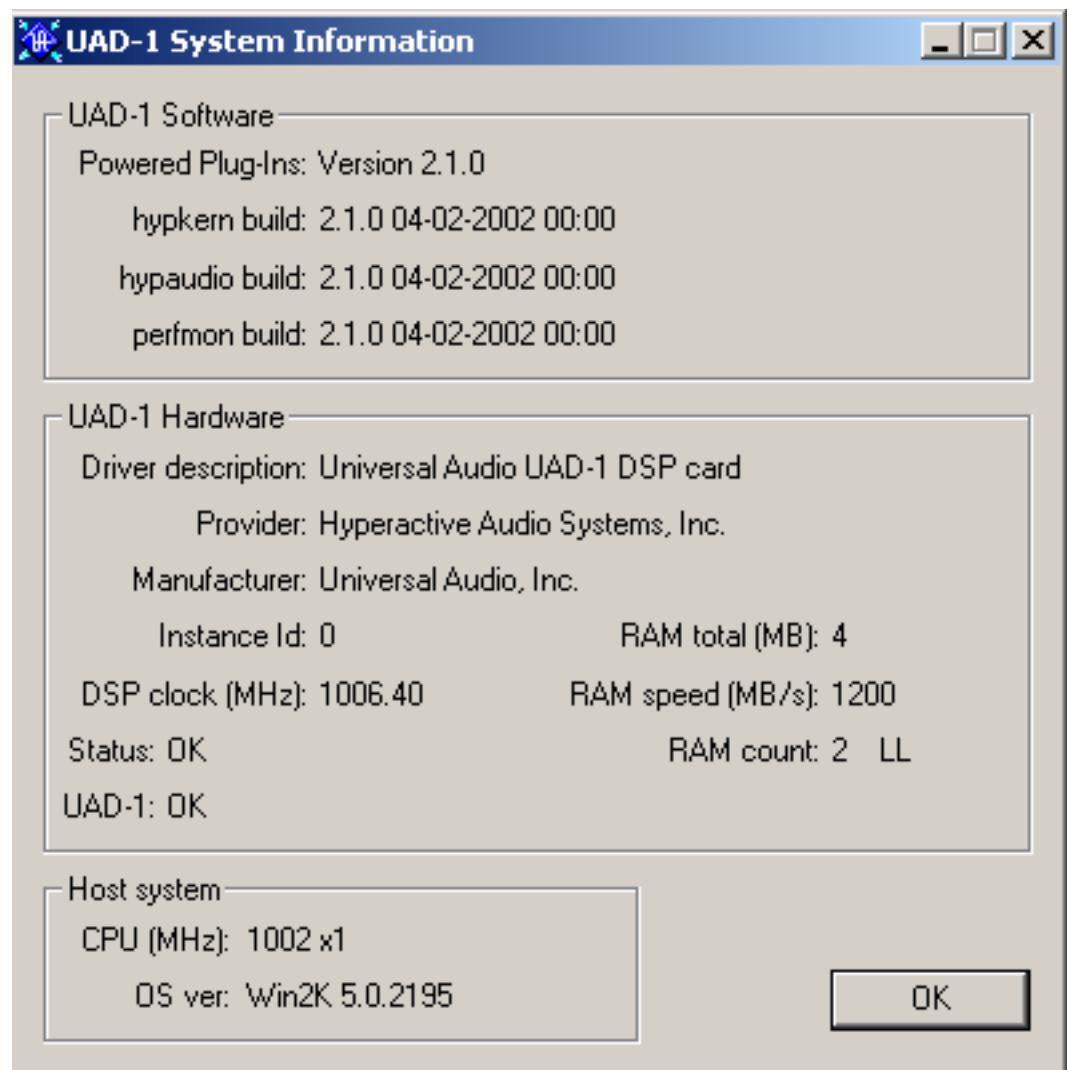


Figure 9. The UAD-1 System Information window.

UAD-1 Configuration Window

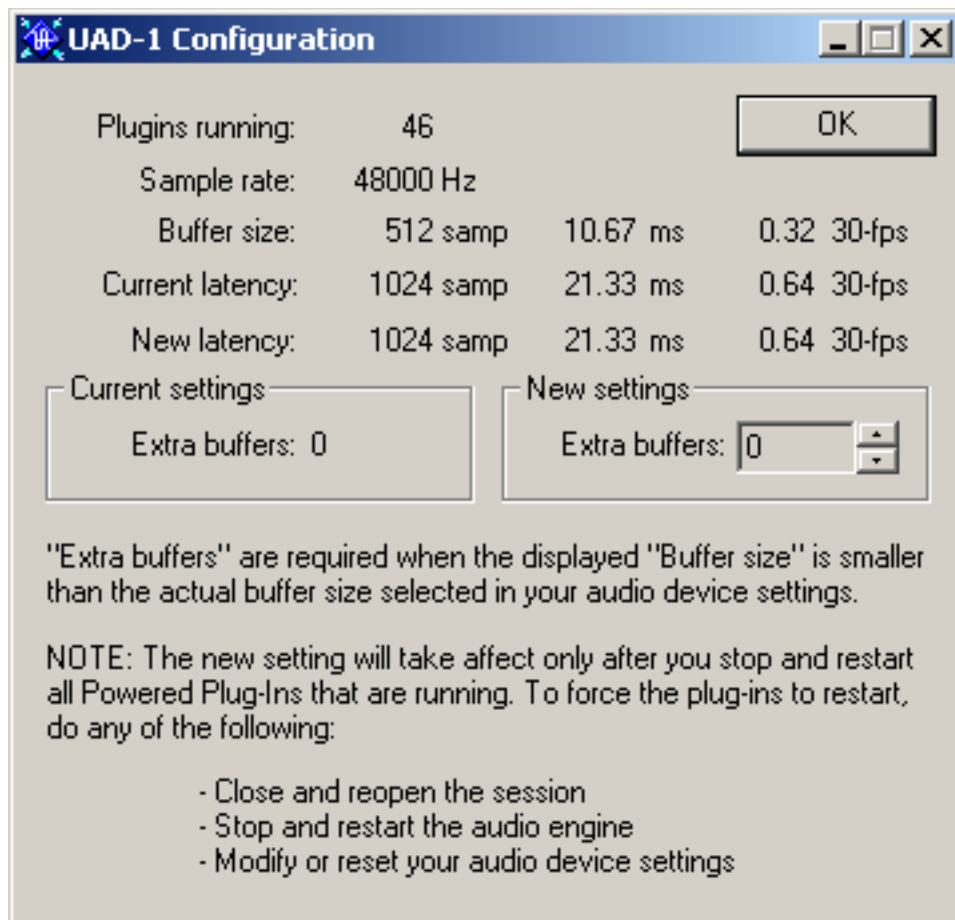


Figure 10. The UAD-1 System Configuration window.

The UAD-1 Configuration Window displays additional information about the UAD-1 card and is also used to insert extra processing buffers for the UAD-1 card when required (see "Configuring Extra Buffers" on page 24).

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 and frames per second.

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.

This situation occurs when users of Steinberg Cubase and Nuendo select ASIO buffer sizes of 2048 samples or greater. If the situation is not corrected, the use of UAD-1 Powered Plug-Ins will introduce excess host CPU load. Extra Buffers are not required for Logic Audio users.

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 now complete.

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.

Host Application Settings

For optimum results, the “Plugin Delay Compensation” option setting should be active in the host application. 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: Options Menu>Audio Setup>System...
- Nuendo: File Menu>Preferences>VST
- Logic Audio: Audio Menu>Audio Preferences...

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 (see “UAD Delay Compensator plugin” on page 26).

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, see “Live Processing” on page 28.*

UAD Delay Compensator plugin

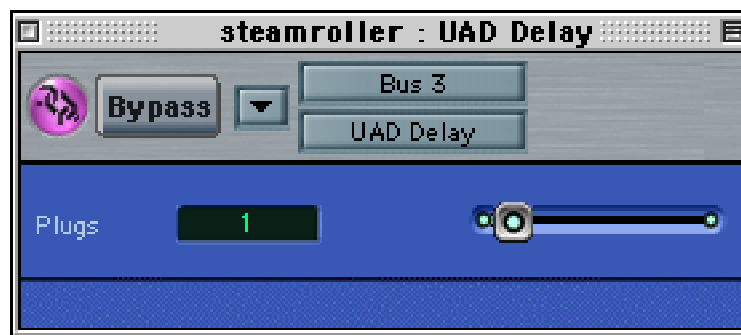
Overview

The UAD Delay Compensator (DelayComp for short) is a simple plugin which is 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 on sends, groups, or 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. It has one parameter with which you specify the number of UAD-1 Powered Plug-Ins instances you wish to compensate.



Cubase/
Nuendo



Logic Audio

Figure 11. The UAD Delay Compensator plugin window.

When to use DelayComp

When UAD-1 Powered Plug-Ins are used on the insert of a send, group, or bus, a latency may be induced on the tracks that are using that send, group, or bus. In this scenario, inserting a UAD DelayComp plugin on the UNPROCESSED track(s) will automatically re-synchronize the audio.

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 Plugs parameter

The UAD DelayComp plugin has one parameter: Plugs. The DelayComp Plugs 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.

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

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

DelayComp Examples

Sends

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 main tracks.

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

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 instrument tracks (but not the vocal tracks or LA2A return) to a group/bus and put one DelayComp with a value of 1 on this group/bus that contains the dry tracks.

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.*

UAD-1 Powered Plug-Ins DSP Usage

The UAD-1 card features an onboard 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 operations. This is unavoidable when using a DSP card.

- DSP usage is proportional to the host application sample rate. Therefore, more plugins can be used simultaneously in a 44.1K session than in a 96K session.
- The DSP resources required by each successive UAD-1 Powered Plug-In instance will slightly decrease.
- Bypassing individual components will conserve DSP. 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.

Note: These measurements are approximate and may vary slightly from system to system. All measurements are on the UAD-1 DSP Performance Meter with audio running at a 48K sample rate. Load units is a percentage of total available resources.

Table 1. UAD-1 Powered Plug-Ins Resource Usage.

Powered Plug-In	CPU LOAD	MEMORY LOAD
UAD 1176LN	13	0
UAD CS-1	8	8
UAD DelayComp	0	0
UAD DM-1	3	4
UAD DM-1L	3	36
UAD EX-1/EX-1M (EQ only)	2	0
UAD EX-1/EX-1M (Compressor only)	3	0
UAD EX-1/EX-1M (EQ & Compressor)	4	0
UAD Gate/Comp	5	0
UAD LA2A	7	0
UAD Mod Filter	2	0
UAD Nigel	33	23
Phasor	4	0
Preflex (Cab Only)	6	0
Preflex (No Cab)	20	0
Preflex (All)	23	0
UAD RealVerb Pro	12	10
UAD RS-1	5	5
UAD TremModEcho	6	23
UAD Tremolo	3	0

analog ears | digital minds

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

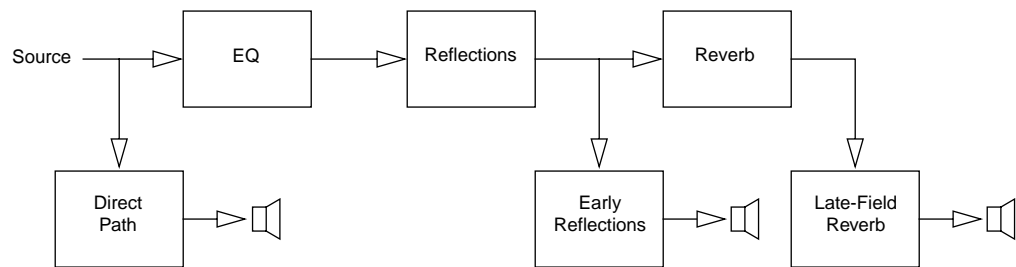


Figure 12. RealVerb Pro signal flow.

Figure 12 illustrates the signal flow for RealVerb Pro. The input signal is equalized and applied to an early reflection generator, which in turn drives a late-field reverberation unit. The resulting direct path, early reflection, and late-field reverberation are then independently positioned in the soundfield.



Figure 13. The RealVerb Pro plugin window.

The RealVerb Pro user interface is similarly organized (see [Figure 13](#)). 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.

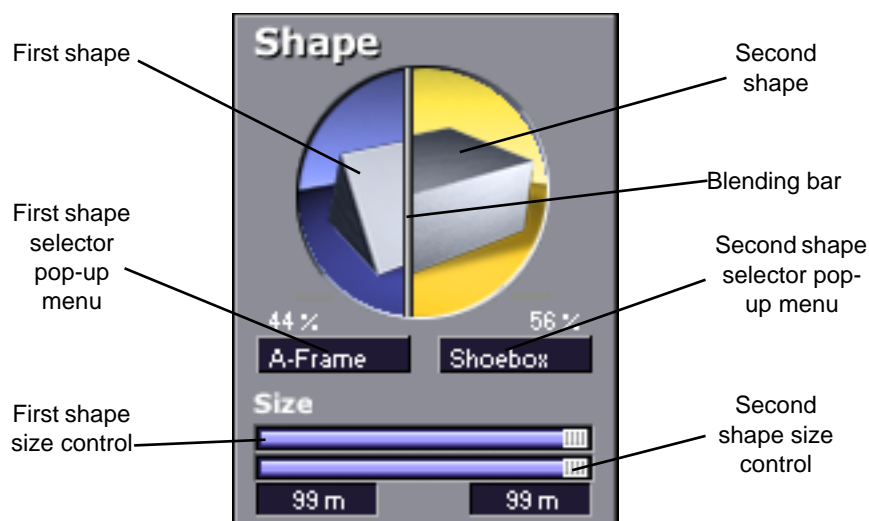


Figure 14. RealVerb Pro Shape panel.

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 17 on page 39](#)), 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 35](#).

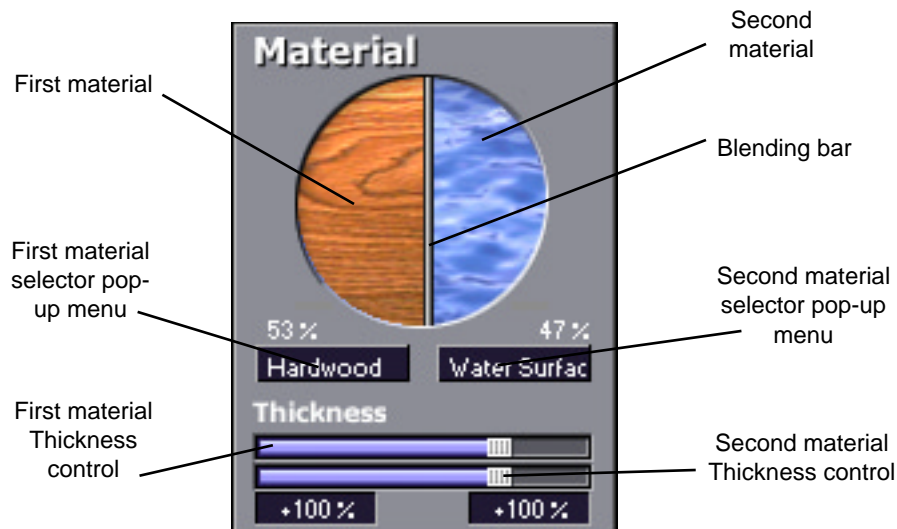


Figure 15. RealVerb Pro Material panel.

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 17 on page 39).

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 2. 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 3. 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.

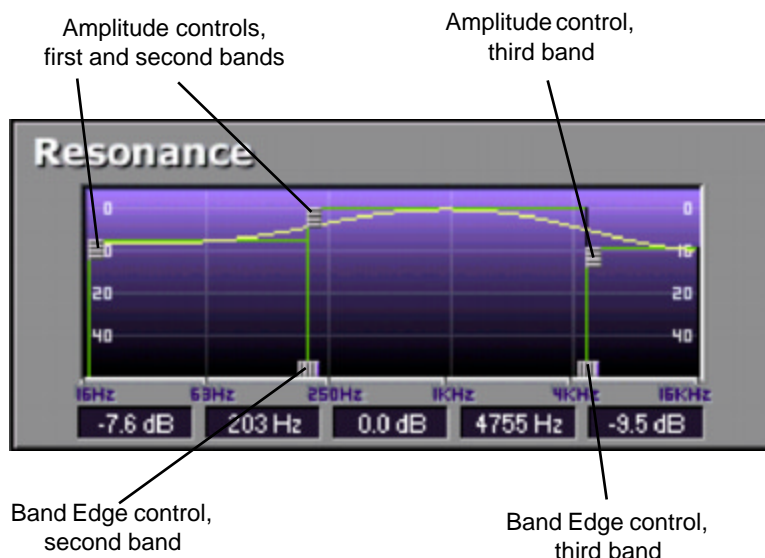


Figure 16. RealVerb Pro Resonance panel.

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 17](#)).

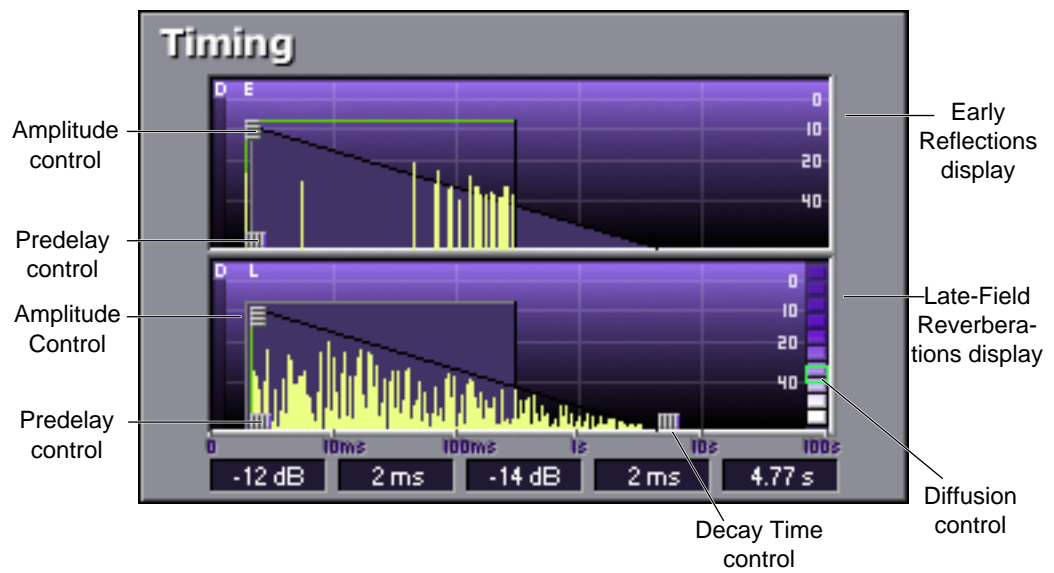


Figure 17. RealVerb Pro Timing panel.

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 14).

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 18](#)) 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.

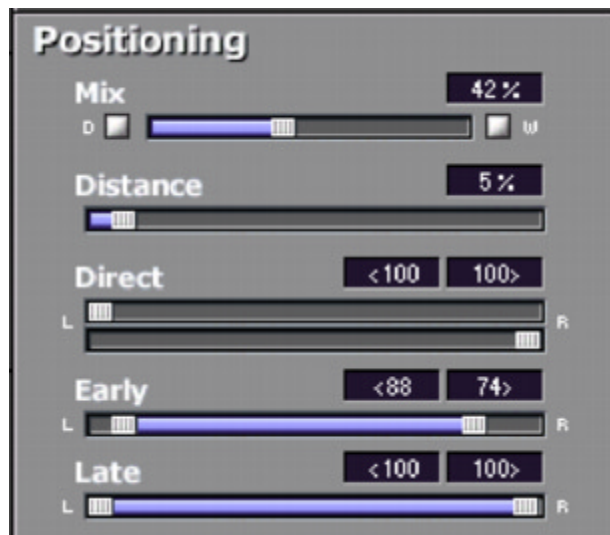


Figure 18. RealVerb Pro Positioning panel.

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 18](#)). 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 18](#)). 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.

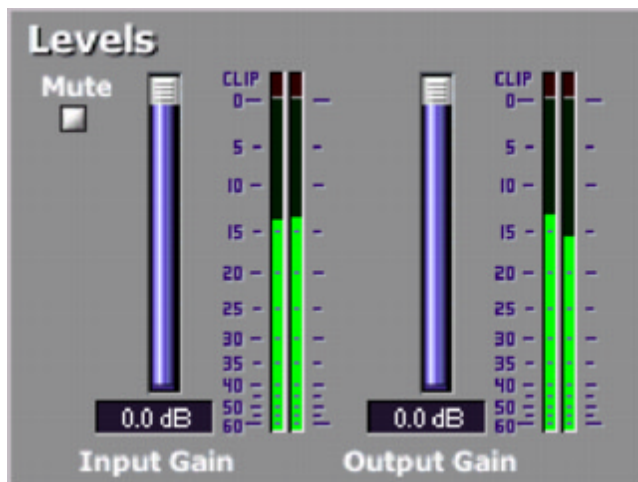


Figure 19. RealVerb Pro Levels panel.

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.

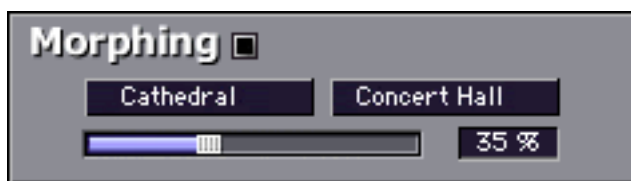


Figure 20. RealVerb Pro Morphing panel.

Figure 20 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 21. 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 VST host applications include their own method of managing VST 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 4. 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 22 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 22, 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 22, 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).

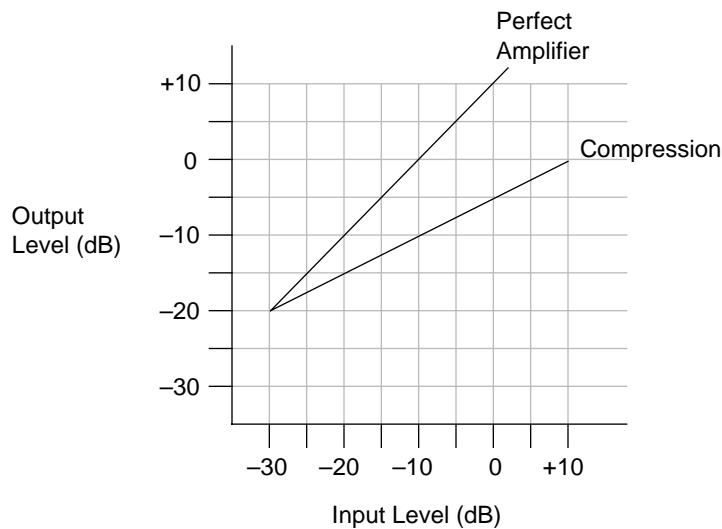


Figure 22. Input and output characteristics of a compressor and perfect amplifier.

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 23) 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.

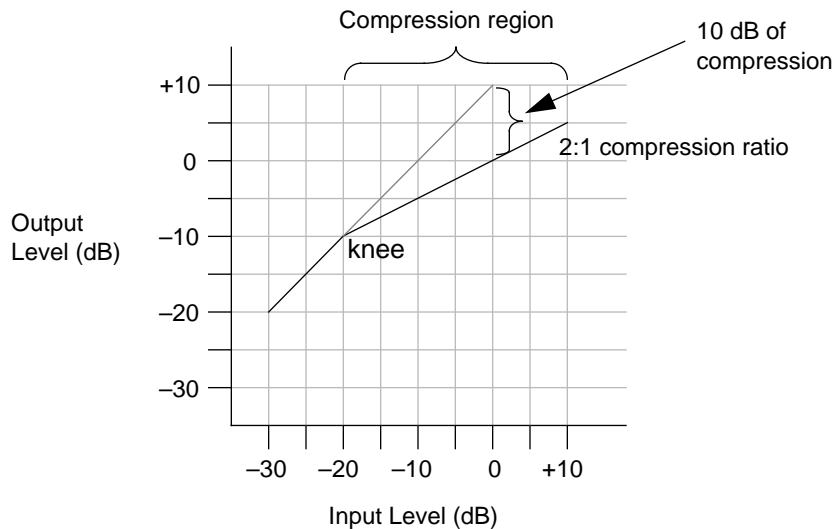


Figure 23. Input and output curve of compressor with 2:1 ratio and -20 dB threshold.

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 23](#), 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 23](#).

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 24](#). 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.

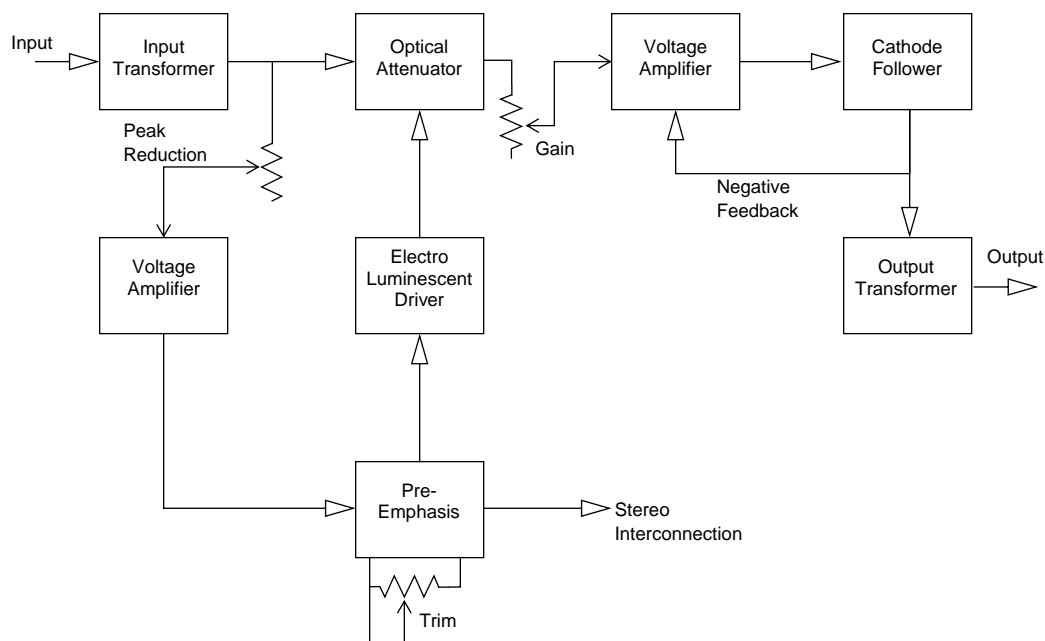


Figure 24. LA-2A signal flow.

LA-2A Controls



Figure 25. 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 26](#). 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.

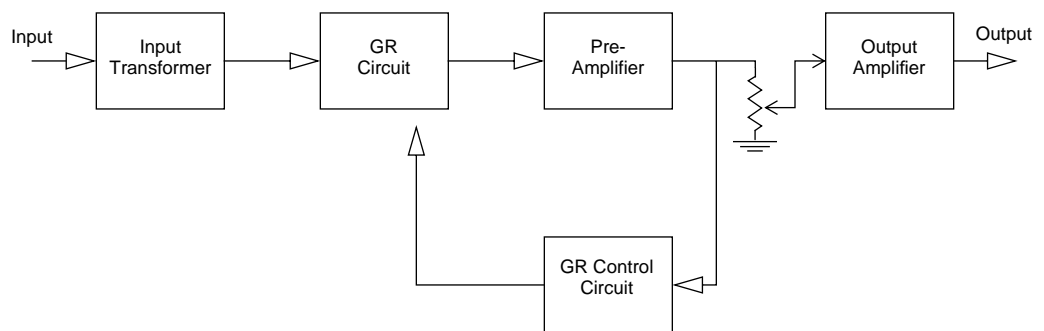


Figure 26. 1176LN signal flow.

1176LN Controls



Figure 27. 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.

Stereo Operation

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



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 28. The CS-1 Channel Strip plugin window.

EX-1 Equalizer and Compressor



Figure 29. 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 30. 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 31. The DM-1L includes a Link button.

RS-1 Reflection Engine



Figure 32. 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 5. 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!

Nigel Screenshot



Figure 33. 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 34. 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 35. 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.

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 34 on page 65](#)) 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 36. 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.

Amp Types can be easily updated by the user. Check our web site at <http://www.poweredplugins.com> for the latest Amp Type downloads.

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 6. 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 37. 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 7](#). The Cabinet Types list itself is in [Table 8](#).

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 7. 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 8. 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 38. 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 9. 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 39. 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 Pedal Mode

When the Wah Pedal mode is engaged, the filter cutoff frequency is varied according to the Wah knob setting. This mode is ideally suited to emulating a real Wah pedal by using a MIDI foot pedal controller.

Mod Menu Table

Table 10. 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 Pedal	Fixed Wah mode for use with MIDI foot pedal control.

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 11. 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 40. The TremModEcho plugin contains three modules.

Trem/Fade Module



Figure 41. 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.

Mod Delay Module



Figure 42. 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 12. 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 13. 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 43. 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 14. 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

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.

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:

ppifedback@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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