RELAB DEVELOPMENT

LX480 Reverb

User Manual

LX480 Complete

LX480 RHall

English

Legal

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LX480 REVERB MANUAL

Version 1.4

1.0 Introduction

The quality of its algorithms and the integrity and flexibility of its signal path has led to it being used as a staple in almost every high–end recording and post–production studio.

First released in 1986, the Lexicon 480L Digital Effects System* has for most of its life been the standard by which all other signal processors are measured. It remains a popular choice among top producers for the most demanding tasks nearly 30 years later. It is widely considered to be one of the best–sounding reverbs ever built. The quality of its algorithms and the integrity and flexibility of its signal path has led to it being used as a staple in almost every high–end recording and post–production studio.

The original unit (and other iconic reverbs, like Lexicon 224 and 224XL) was created by under the guidance of David Griesinger, who was the main reverb developer at Lexicon for many years.

The main reason for creating the LX480 plugin was to preserve this sonic quality and to allow failing hardware units to be replaced with a perfect recreation of the firmware version v4.10 in plugin format. The algorithms in the emulation are sample–accurate compared to the original unit and the plugin is therefore the most authentic hardware emulation available. Additionally, the LX480 Reverb plugin introduces extra functions, including parameters for better sound shaping, a high–density algorithm and exhaustive modelled input and output stages – everything is modelled, from internal clipping to quirks in the feedback loops.

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1.1 This Manual

This manual covers the essential information you need to install and operate your LX480, but also delves into subsidiary subjects, with general discussion and advice to help you get the best out of the plugin:

Reverberation: What is reverb? How do reverbs work, and what features do different reverbs possess?

Operation: Description of the LX480, including an explanation of the interface and details of the controls.

The Algorithms: Description of the algorithms, their character and any extra features available in the LX480 when compared to the Lexicon 480L.

The Parameters: Detailed listing of the editable parameters and their effect.

1.2
Installation

Download

You can download the latest build of LX480 Reverb from the Relab website, at https://www.relabdevelopment.com
The file you download is a simple ZIP file containing the setup file you need.

Installation

You will need an iLok user account to use LX480 Complete. You can register your LX480 as a computer based license or onto an iLok dongle (version 2 or above). If you have not done so already please install the iLok License Manager software and activate the LX480 license.

iLok: Demo activation

1: Log into your iLok account.

2: Click on "Redeem key".

3: Enter the following key: 3169–6261–4561–9483–6462–9351–4340–47.

4: Place the new key into your iLok or machine.

5: Download and install the LX480 plugin by double clicking on the setup file.

2.0
Reverberation

While reverb may have begun life as a means of simulating the sound reflections of an actual acoustic space, it quickly gained a broader brief as a creative tool. Why? Because its effects are wide–ranging:

- » **Blending:** To various degrees, enabling different sound sources to sit well together (and possibly to be placed in specific spatial relationships with each other).
- » **Creating space:** Changing or adding ambient space to whatever the original recordings were made with.
- » **Spread:** In most cases, spreading instruments across the mix, and possibly widening as well.
- » **Sustain:** Filling gaps to increase sustain, but also adjusting the tonal properties of the reverberation (and the sustain) to artistic effect.

This section covers some general principles of reverb, along with a little bit of the history, to place in context the use to which reverb – and the LX480 in particular – can be put.

2.1 Natural ambience

Humans are very adept at deducing information about the surroundings of a sound source by listening to its reverberation, and the qualities of reverberation (or reverb as it is more commonly known) form an intrinsic part of our appreciation of sound and music.

Natural reverberation is present everywhere there is sound, with the very particular exception of an anechoic chamber. Reverberation is formed of the decaying echoes of sound. The sound that reaches a listener's ears – say, the sound of a handclap – will be a combination of the sound that travels direct to them, along with echoes from nearby or far–away surfaces, echoes of those echoes, and so on until their energy has decayed to zero. The character of these echoes will depend upon the positioning and reflective qualities of these surfaces, as well as on the frequency content of the original sound.

The medium through which the sound travels – most commonly air – also has an effect on the nature of the sound reaching the listener (air will absorb high–frequency noise). More particularly, reverberation is the term for the "halo" of echoes that are perceived as a unified whole by humans, rather than as distinct, individual echoes.

Humans are very adept at deducing information about the surroundings of a sound source by listening to its reverberation, and the qualities of reverberation (or reverb as it is more commonly known) form an intrinsic part of our appreciation of sound and music. Over time humanity has developed its design of interior spaces partly in response to the reverberative qualities of shapes, surfaces and building materials, from the Neolithic chambered cairn of Maes Howe to the most modern concert hall design.

2.2 Artificial ambience

The arrival of sound recording brought with it a specific difficulty regarding reverb. Not only were early microphones, recording media and loudspeakers limited in their frequency response, the whole process also telescoped down the natural reverb cues, leading to a distinctive "recorded" quality that was very far from the natural "live" sound. As recording techniques evolved to include multi–tracking, isolation booths and entirely electronic sound sources, a great need arose for techniques that would enable the sense of acoustic space to be added to a recording in which it was now entirely or partially missing.

Chambers

Initial efforts to capture reverb focused on providing good–sounding spaces in which to record music, and these techniques have developed and are still in use today:

» re–recording a signal played back through loudspeakers in a different space to which it was recorded

» recording instruments using multiple mic positions to allow ambience to be mixed in using various methods

» using tailor–made recording spaces, from drum rooms to concert halls to Hollywood recording stages, combined with multiple mic positions.

Clearly, such techniques require access to suitable spaces to record in, and lack flexibility.

Springs

The first electromechanical solution to the reverb problem was conceived in the 1930s, developed in the 1940s and became a staple of guitar and organ amps in the 1960s and 70s. It employed transducers and springs to simulate the bouncing of reflected sound between two facing surfaces. Although such devices were theoretically "one–dimensional", the inherent complexities of the materials involved led to a distinctive, complex sound, though not a particularly natural reverb, especially when fed with transient–rich material.

Plates

Plate reverbs were first produced in the 1950s by the German manufacturer, EMT. Plate reverb units also used transducers to physically alter sound, in this case by passing it through a metal plate suspended in a box, rather than through a set of springs. The "two–dimensional" behaviour of a plate produced a more realistic result than spring reverb, with a dense, smooth tone that was more musically useful, and became particularly widely used on vocals and drums. However, plate reverbs, though less expensive than purpose–building a recording space, were still large, cumbersome items, and the amount of adjustment to the sound that was possible was limited to adjusting the dampening on the plate.

2.3
Digital reverb simulations

Convolution can be a good solution for certain areas of post–production, but its inflexibility can make it less helpful in a mixing situation

The first attempts to use digital signal processing to provide a reverb system date back to the 1960s, but it took till 1976 for the first digital reverb to reach the market, partly because of the complexity of the theory that needed to be solved, but largely because of the limitations, in terms of both performance and cost, of early DSP chips.

Algorithmic

The earliest units, such as the EMT 250, employed fairly uncomplicated algorithms with a limited number of parameters for sculpting the effect, such as pre–delay and delay time, though they betrayed their more all–purpose nature by offering other effects besides reverb. As processing power increased, the level of complexity of the algorithms employed increased greatly, as did the number of editable parameters, and digital reverb units became capable of enormously flexible and subtle audio manipulation, with the Lexicon 480L* as one of the pinnacles of this evolution.

Convolution

Another method of creating digital reverberation was also being considered in the 1970s, though at that time the processing power was not available to put into action: convolution. This involves taking a "snapshot" of an acoustic space by recording a carefully defined sound being played in it. This snapshot can then be analyzed, removing the original "impulse" sound to produce an impulse response (IR). The IR is then used to process an audio signal by a process known as convolution. The result is an often remarkably natural result, albeit one that can often be altered very little compared to an algorithmic reverb. Convolution can be a good solution for certain areas of post–production, but its inflexibility can make it less helpful in a mixing situation.

2.4
Qualities of reverb

Another characteristic which is often overlooked is the ability to reverberate low frequencies in the range 20–100Hz and have a consistent overall frequency distribution.

What makes a good reverb algorithm? When reverbs are discussed, words like "smooth", "rich", "clean" and "spacious" are often used, though there is a good deal of subjectivity in these terms. However, one characteristic that is often overlooked even though it is in fact crucial is the 3D–quality or "envelopment" of the reverb.

The primary goal for synthetic reverbs is to create a sense of space around the sound source – sometimes a realistic space, at other times just something that sounds good but still represents a believable space.

David Griesinger, creator of the Lexicon 224 (XL) and 480L, has described envelopment (also known as "spaciousness" and "spatial impression") in the following terms:

- » The Holy Grail of acoustics.
- » Draws the listener into the music or scene.
- » Takes training to reliably perceive: most music listeners (and critics) perceive only loudness, balance, intelligibility, and localisation.
- » The effect is unconscious but powerful: in a recent blind test there was a consistent bias for high envelopment.

Many reverb plugins lack this sense of spaciousness – to the extent that the narrow spread of the sound field sounds monaural when compared to high–end hardware. The LX480 plugin follows its model in providing this sense of spaciousness.

Another characteristic which is often overlooked is the ability to reverberate low frequencies in the range 20–100Hz and have a consistent overall frequency distribution. Many reverb designs have been through a lengthy process of eliminating obvious resonances (metallic sound) in the tail which result from longer reverberation time for specific frequencies (the resonances) compared to the overall reverberation time, but forget the effect of cancellation of frequencies in the reverb tail. Without careful tuning of the algorithms you can experience a lack of energy in the low frequency spectrum which has to be compensated, but never correctly eliminated, with post processing.

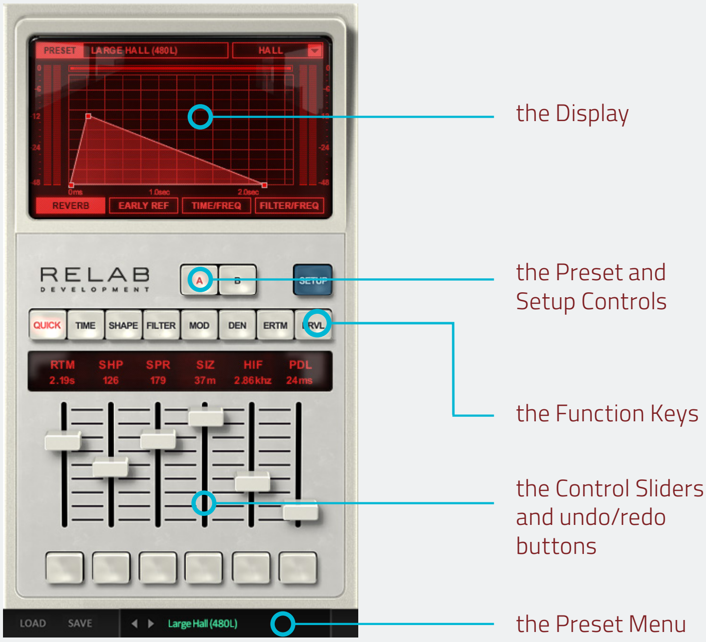
3.0
Operation

The user interface for the 480LX plugin is based closely on the LARC (Lexicon Alphanumeric Remote Control) from the original unit. If you're familiar with the LARC, you'll find operating the 480LX very comfortable, with the benefit of extra visual feedback and control and editing capabilities. The GUI is divided into:

- » the Display
- » the Setup Controls
- » the Function Keys
- » the Control Sliders and undo/redo buttons.
- » the Preset Menu

If you're familiar with the original LARC, the main differences you'll notice are that the numeric keypad is gone, replaced by more patch navigation and management buttons and a Setup button that gives access to a series of screen for adjusting preferences and "under the hood" settings. The Function keys now work to give direct access to different pages of parameters, which remain consistent between algorithms. The slider display keys at the bottom now function as A/B or undo/redo buttons.

More details of the operation of these controls is given in the specific sections below, but if you're looking for details of the effect of each control, refer to the Parameters section.

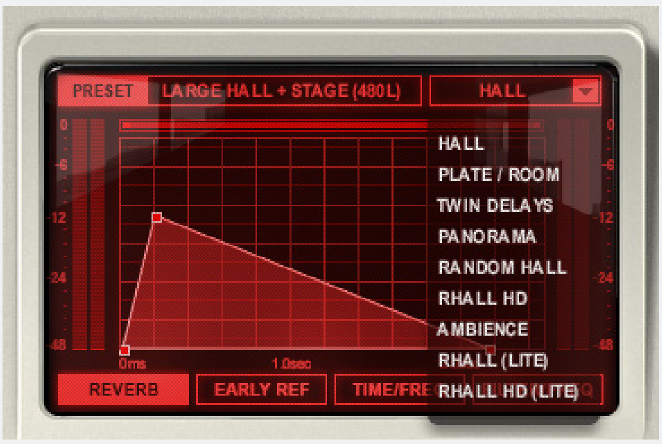


3.1
Display

The display section gives visual feedback on edits made with the control sliders, as well as allowing direct editing of parameters using the mouse.

The top bar shows the current preset (though this may of course have been tweaked). On the right–hand side is a drop–down menu allowing you to choose the algorithm:

- » Hall
- » Plate/Room
- » Twin Delays
- » Panorama
- » Random Hall
- » RHall HD
- » Ambience
- » RHall (Lite)
- » RHall HD (Lite)



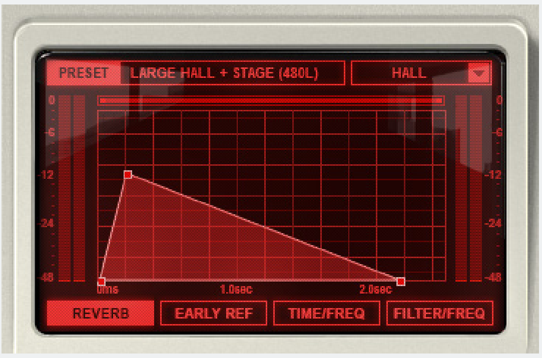
3.1
Display (continued)

You can find a fuller explanations of these in the Algorithms section. The parameters shown and available to be edited in the display section are common to most of the algorithms (though not all – details of which have no effect are given in the individual display sections below).

The sides of the display area show two level meters. The left–hand meter shows in the input signal level, the right–hand meter shows the output signal (with the selected **Wet/Dry Mix** (**MIX**)).

At the bottom of the display are four page/tab labels. Clicking on each will highlight it and bring that section onto the display for editing.

Reverb

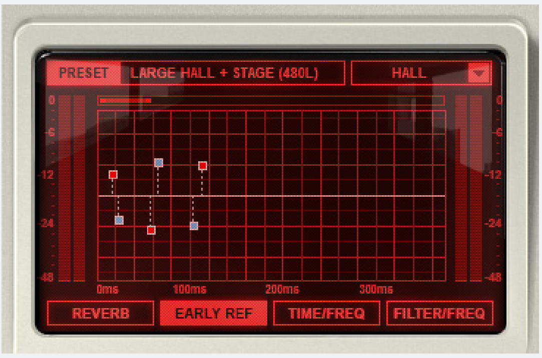


The REVERB tab gives a broad representation of the development of the reverb over time. There are three square handles on the contour, which can be dragged to change a particular parameter value:

- The left–hand handle can be dragged from left to right to change the **Predelay** (**PDL**) parameter.
- The middle handle can be dragged up and down, affecting the overall **Reverb level** parameter (**LEV**). Note that the middle handle cannot be dragged from left to right, as there are two parameters, **SHP** and **SPR**, governing this dimension.
- The right–hand handle can be dragged left and right to change the overall **Reverb Time** (**RTM**) parameter. If you drag it off to the right, the time display (y–axis) will rescale to allow the greater timescale (the maximum varies depending on the algorithm). Likewise, dragging it to the left will cause the display to rescale so that the contour occupies a similar amount of the display. This occurs when you release the mouse button.

The time axis display can also be rescaled independently using the scroll bar above the graph. Grabbing one of the scroll markers at either end will let you stretch the display in that direction, so you can focus on a particular area of the reverb contour.

Early Reflections



The EARLY REF tab shows the volume, pan and timing of up to six early reflections (sometimes known as pre–echoes). These are not clean, individual delay taps, but are actually more diffused clusters of echoes; the DIF (Diffusion) control (found under the DEN function key) governs exactly how diffuse they are.

There is no **DIF** control available when using the Twin Delays algorithm, because the algorithm functions differently to the others. However, the delay markers work in the same way as for other algorithms, although only the first four delays are functional.

NB: The Ambience algorithm has no specific early reflections, so settings made in this tab will have no effect on the sound. However, you can still add early reflections to the Ambience model by using the Stereo Split mode and combining two engine with different algorithms.

Comparing the location of each delay marker against the scale at the bottom shows when each pre–echo occurs, and the time axis can be rescaled using the scroll bar above the graph in the same way as on the Reverb tab.

3.1
Display (continued)

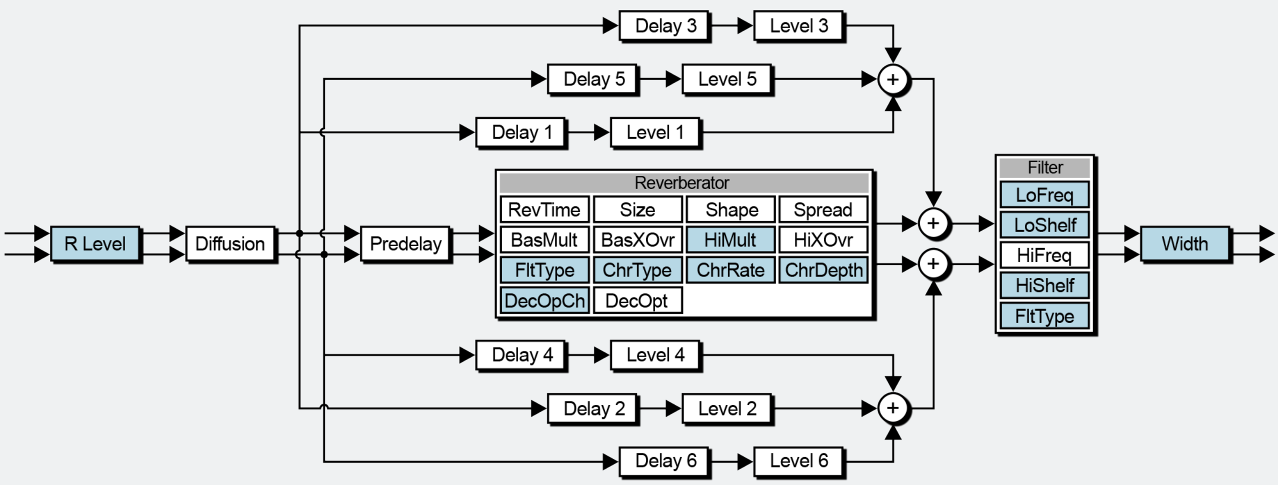
You can drag the small squares to freely adjust delay time and signal level. Vertical distance from the centre line represents signal level. If you have the ERTM or ERVL function pages open, you will see that the changes you make are reflected on those sliders.

Some presets have no early reflections set, some have only two. If you want to specify more, simply drag the square at 0 time on the centre line and place it where you want it. If none are set, marker 6 will be the first to be placed. You can always return a square to 0 on the centre line if you want to remove it.

You will notice that some markers (1, 3 and 5) are constrained to positions above the line, while the others (2, 4 and 6) are constrained to positions below the line. Also, markers 1, 2 and 3 are coloured (assuming you haven't set your display colour to grey – see Display Setup Page) while markers 4, 5 and 6 are grey. These indications relate to pan information:

DL1: L>L DL2: L>R DL3: L>L DL4: R>R DL5: R>L DL6: R>R

The first letter refers to the channel of the input (L or R), the second letter refers to the output channel, so there are three taps for each input channel (1 to 3, for L input, 4 to 6 for R input), one of which (Delay 2 for L, Delay 5 for R) is cross–panned and mixed with the taps from the opposite side. This diagram of the Hall algorithm clarifies this:

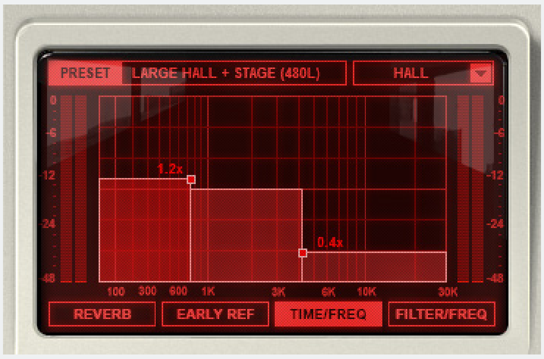


Hall algorithm diagram

3.1
Display (continued)

Time/Frequency

The TIME/FREQ tab shows a graphic representation of the adjustments made to reverb time for higher and lower frequencies, and allows these parameters to be controlled.



NB: These controls have no effect for the Twin Delays algorithm, and only the High Frequency controls have an effect for the Ambience algorithm.

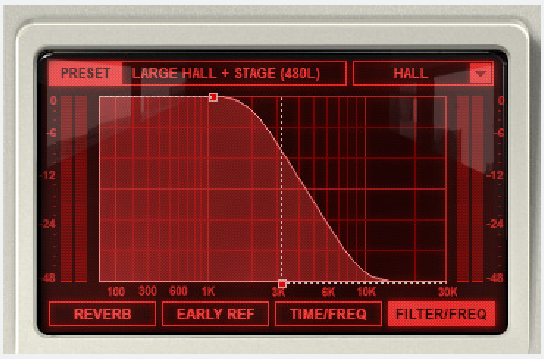
The display shows three frequency blocks. It is possible for the left and right blocks to overlap and thus obscure the central block completely. The central block represents the **Reverb Time** for the middle frequencies and always shows a level on the central line (i.e. a multiple of 1.0x). The left–hand block represents lower frequencies, and can be dragged above the central line (for multiples up to 4.0x) or below the central line (for multiples down to 0.2x). The right–hand block shows higher frequencies, and covers ranges from the central line (1.0x) down to 0.2x.

By dragging the coloured squares left and right the crossover point for low and high frequencies can be set (corresponding to the **BXO** and **HXO** parameters). Dragging vertically adjusts the reverb time for lower (**BAS**) and higher (**HIG**) frequencies, shown as a multiple of the reverb time for the mid frequencies (**RTM**). **RTM** is not adjusted directly on the graph.

NB: If the boxes are dragged so that they are overlapping, the reverb times for the overlapped frequencies are multiplied by the product of the BAS and HIG. More details of the practical use of these controls can be found in the detailed descriptions of the parameters.

Filter/Frequency

The FILTER/FREQ tab shows a graphic representation of high– and low–shelf filters that are applied to the processed signal (both reverb and pre–echoes).



Dragging the two squares allows direct adjustment of each filter as regards both frequency (horizontal plane, across the frequency scale as shown, corresponding to **LOF** and **HIF**) and amount of attenuation (vertical plane, with 0db of attenuation at the top of the scale, corresponding to **LOS** and **HOS**).

3.2
Preset and Setup controls

Preset Menu

Load

This button opens an Explorer/Finder window showing the last directory used to select a preset.

Save

This button opens an Explorer/Finder window at the previous directory used and allows one to save a new preset.

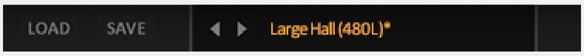
Prev/Next

By clicking the Prev/Next icons to the left preset name loads the adjacent preset.

Preset Name

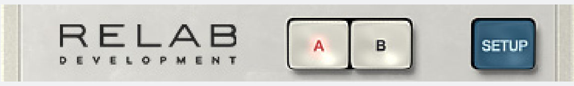
Clicking on the preset name shows the list of presets, to load a preset click on the one you wish to load.

Note: To signify that a parameter has changed from the presets saved state, the text changes from green to orange and an asterisk is appended to the preset name.



Setup Controls

A and B

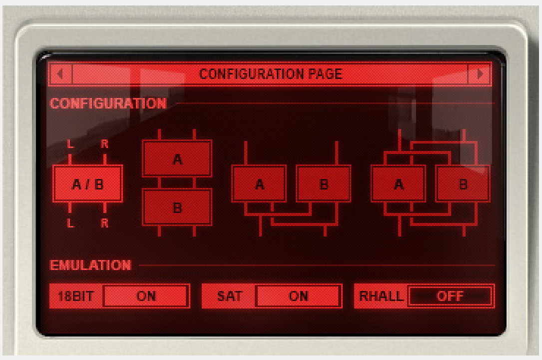


The A and B buttons control which of two engines is currently being edited. The setting chosen in Setup Page 1 determines how these two engines work together. When Single mode (the default) has been chosen, and only one engine is actively being used at any one time, the A and B buttons allow quick comparison between two presets. The gentle red highlight shows which engine is currently active, A or B. After loading a preset initially, click on B and use the LOAD button to load another preset. These two slots can now be alternated between using the A and B buttons. In Cascade, Mono split or Stereo split mode, the two A and B engines act on the stereo input in different ways. See Setup Page 1 below for more details of how these work. Loading presets in these modes works in exactly the same way as it does in Single mode.

Setup

The Setup button gives access to various setup, preference and information pages, which are displayed in the upper display pane. To switch between the setup pages, use the left and right arrows in the title bar. Detailed information about these pages can be found below.

Configuration page



Routing: Select one of the four graphics shown here to select a routing configuration. From left to right, they are:

» **Single:** This is the default mode. Only one engine is active at any one time, and it acts on the stereo input. To choose which engine is active, use the A and B buttons (see above for details of how to do use this to quickly compare two presets).
» **Cascade:** This mode simply passes the output of Engine A into Engine B. This routing mode will be more useful in the next update, which includes a new algorithm: Panorama, which makes a signal wider. Cascade mode will thus let you enhance the wideness of the reverberation of Engine A if Engine B is loaded with the Panorama algorithm.
» **Mono split:** This routing can be used to treat L and R inputs independently. The left signal input will be processed by Engine A and the right by Engine B. The resulting signals are summed for output. This is a useful technique to create a very wide stereo field.
» **Stereo split:** This routing passes the stereo signal to both engines, and combines the output from both. This capability adds a massive amount of flexibility to LX480 Reverb. For instance, one could combine distinctive aspects of two different algorithms, using Ambience on Engine A to take advantage of its early reflection engine and combine that it with Random Hall on Engine B for the reverberation.

EMULATION:
18Bit: Switches between the original hardware bit-depth of 18bit int to 32bit float. When 18bit mode is turned off, it doesn't get truncated like on the hardware.
Sat: Turns on or off a hard-clipping saturation. This saturation occurs due to the integer precision of the original hardware, producing distortion inside the algorithm, creating a more present reverb sound.
RHall: Turns on or off a bug found in the RHall algorithm of the original hardware. This bug is exhibited when Spin (SPN) and Wander (WAN) are set to zero and size is changed, resulting in a metallic sound.

3.2
Setup controls

Input / Output Page - **Analogue and noise emulations:**

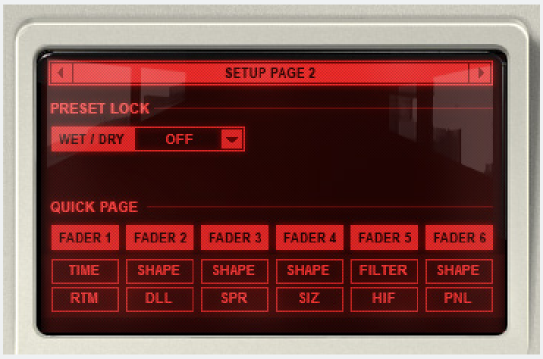
The hardware unit that LX480 emulates uses a combination of 6-bit and 16/18-bit processing, which inevitably introduces "grit", "dirt" and "artefacts" into the calculation of the reverberation (most noticeable with heavy modulation). Far from being a problem, this is a very large part of the 480L sound, and works particularly well on rock/pop productions etc.

However, such artefacts are considered undesirable in classical productions, for instance, and so LX480 Reverb offers the option to remove the inherent noise by choosing a mode without "Noise". LX480 Reverb also emulates the effects caused by the input/output stage of the hardware unit, which may be either digital in/out or analog/aux in/out. There are two "flavours" of analog available, giving the following six options in the Emulation drop–down menu (*"noise" represents the normal operating mode of the original*). - *DIGITAL (NOISE), DIGITAL, MAIN (NOISE), MAIN, AUX (NOISE), AUX*

Setup Page 2

» **Preset Lock:** Wet/Dry: Changing this setting to ON means that the MIX setting will remain unchanged as presets are changed, allowing a better comparison between presets, or at least allowing the required wet/dry level to be maintained. Note that different MIX settings can be locked for the A and B presets.

» **Quick Page:** This section allows one to specify which six parameters are chosen to appear on the QUICK page for each preset. Beneath the label for each fader are two boxes; clicking each gives access to a drop–down menu. The top one selects which function button section is selected (TIME/SHAPE/FILTER/MOD/DEN/ER TIME/ER LVL), the bottom one shows a drop–down list of the available parameters for that function button section for that algorithm.



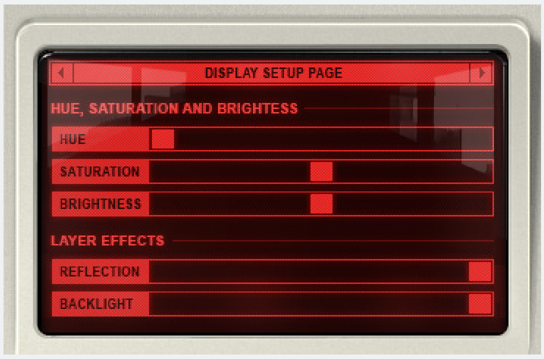
Display Setup Page

You can use the controls on this page to change the way LX480 Reverb looks.

» **Hue, Saturation and Brightness:** Use these controls to change the basic colour and brightness of the interface.

» **Layer Effects:** The Reflection slider fades in a reflection (of a room behind the user) to the main display pane. The Backlight slider adjusts the apparent brightness of the display pane's backlight; a dimmer backlight makes the reflection appear more prominent.

The settings you make here will be remembered every time you use this plugin.



Graphic Page

Graphics: These four options allow you to specify the background graphics used for the LX480 Reverb GUI:

» **Normal:** LARC–style

» **Dirty:** LARC–style with extra coffee drips and general wear and tear

» **Dark:** a dark version of the LARC style

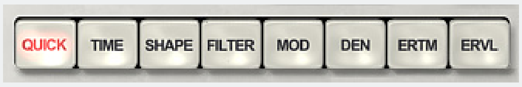
» **Custom:** this option allows for a custom skin to be used.

Info Page

The credits for the plugin. The graphics act as links to the respective websites. You can also find here exactly what version of the software you are using.

3.3
Function keys

The eight function keys, QUICK, TIME, SHAPE, FILTER, MOD, DEN, ERTL and ERVL, give access to different ranges of parameters that can be tweaked using the sliders below.

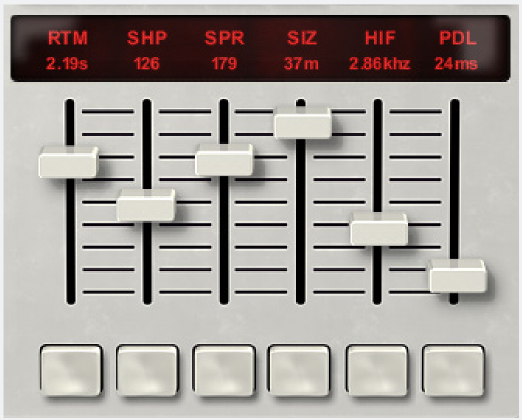


The parameters shown for each key differ depending on the algorithm being used. The Hall and Plate/Room algorithms use the same parameters, and the Random Hall is very similar to these, but several of the Ambience parameters are different, and the Twin Delays algorithm uses different parameters yet again.

The parameters that appear in the QUICK section also appear under other function key sections. They have been selected to be useful for making quick adjustments to presets, but you can alter which parameters appear here, on a per–preset basis, on Setup Page 2.

Full details of the effects of each parameter can be found in the Parameters section.

Control Sliders and Undo/Redo Buttons



The control sliders area comprises three separate sections:

- » the information bar
- » the control sliders
- » the undo/redo buttons.

The Information Bar

This areas shows which parameter the slider below will adjust. It shows the three–letter abbreviation (which will be familiar to LARC users) along with a value. Clicking anywhere on a slider track will show the full parameter name in the information area above.

The Control Sliders

The control sliders allow direct adjustment of the parameters. You can click anywhere on the slider track and drag the slider up and down. For fine control, hold down the Ctrl/Control key or the Shift key and drag.

The Undo/Redo Buttons

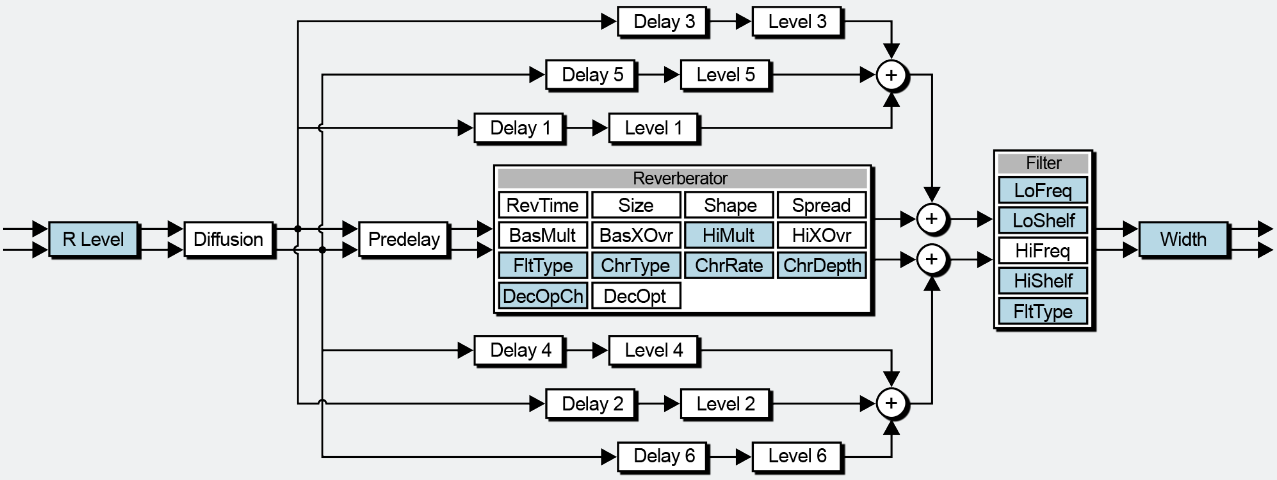
The buttons underneath the control sliders remember the last value for this slider that was selected for this parameter. This provides a simple way to undo your last change or, by pressing the button several times, to A/B two settings. Note that the button will remember the last edited value for that parameter, not for the slider (and the parameter for this slider may have changed if, for example, you have changed the algorithm).

4.0 The Algorithms

The algorithms in LX480 Reverb provide emulations of the reverb, ambience and delay algorithms found in the 480L. The 480L provided several other non–reverb features which are not emulated here. LX480 Reverb also provides an extra version of the Random Hall algorithm, Random Hall HD. See below for a description of each algorithm – what they do, what they're useful for, and any extra features that are available in LX480 Reverb compared to the original hardware.

4.1 Algorithm – Hall

This algorithm is designed to emulate the effect of reverberation in real concert halls. Unsurprisingly, this makes it particularly suited to acoustically recorded material, though it is also ideal for any sort of multitracked music, to provide a common sense of space. The algorithm comprises two distinct elements: early reflections and reverberation.



NB: The boxes coloured blue on the algorithm diagram represent additional parameters/functions not found in the original hardware.

Parameter Layout

TIME	**RTM** Reverb Time Mid	**BAS** Bass Multiply	**BXO** Bass Freq Crossover	**HIG** High Multiply	**HXO** High Freq Crossover	**TFT** Reverb Filter Type
SHAPE	**SHP** Shape	**SPR** Spread	**SIZ** Size	**PDL** Predelay		**WID** Width
FILTER	**LOF** Low Frequency	**LOS** Low Shelf	**HIF** High Frequency	**HIS** High Shelf		**FLT** Filter Type
MOD		**CHT** Chorus Type	**CHR** Chorus Rate	**CHD** Chorus Depth		
DEN	**DIF** Diffusion		**DCC** Optimization Ch.	**DCO** Decay Optimization	**LEV** Reverb Level	**MIX** Dry/Wet Mix
ERTM	**DL1** Echo Delay 1 L>L	**DL2** Echo Delay 2 L>R	**DL3** Echo Delay 3 L>L	**DL4** Echo Delay 4 R>R	**DL5** Echo Delay 5 R>L	**DL6** Echo Delay 6 R>R
ERVL	**LV1** Echo Level 1 L>L	**LV2** Echo Level 2 L>R	**LV3** Echo Level 3 L>L	**LV4** Echo Level4 R>R	**LV5** Echo Level 5 R>L	**LV6** Echo Level 6 R>R

4.1
Algorithm – Hall (continued)

Software vs Hardware

The original hardware only had a **Reverb Level** (**LEV**) parameter in the Random Hall algorithm. The reverb level was therefore effectively hardcoded to 160 for the other algorithms. We have decided to include this parameter (**LEV**) in all the other algorithms, including reproducing its behaviour. It's worth noting that this parameter is not a straightforward volume controller: with a setting between 0 and 160 it functions as a normal volume control, but settings above 160 will introduce saturation with various intensity in the different reverb modules (the diffusion and the reverberator). This can create an upfront sound compared to its no–saturation behaviour.

Moreover, the original algorithm doesn't have modulation as such, but rather uses a decay optimization function that modulates internal parameters according to changes in input level. This subtle change in the reverb tail is very different from the sound of the modulation found in earlier Lexicon hardware reverbs. The chorus controls added in LX480 Complete add the option to make the reverberation sound less metallic (reduce the build–up of resonance) by randomising the delay times in the algorithm. A side–effect of this is the introduction of pitch variations into the reverberation tail. Thus sound sources with very little pitch wobble/variation (such as guitar and piano) will need low chorus values or for the chorus to be completely off.

The **Chorus Type** (**CHT**) parameter selects between the various different chorusing modes used. With a value of zero the chorusing modulation is completely off and the resulting reverberation is identical to the hardware (which has no chorusing). Settings 1–3 increase the number of internal delay lines that are affected by the chorusing. Settings 4–7 are identical to settings 0–3, but with added chorusing to the outputs of the algorithm. Settings with a negative number have the various delay lines randomized so that they are completely independent of each other, which results in a more chaotic pitch variation.

Additional functionality has been introduced to the reverb filters and output filters, offering more flexibility in sound shaping, including the **Width** (**WID**) parameter found on newer versions of Lexicon hardware.

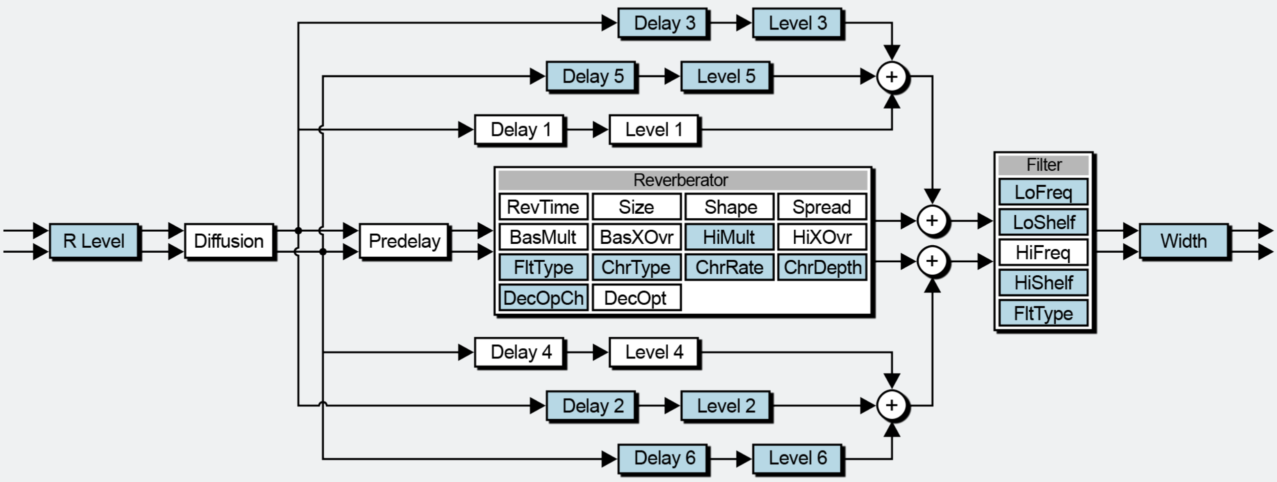
PAGE 1	**RTM** Reverb Time Mid	**SHP** Shape	**SPR** Spread	**SIZ** Size	**HFC** High Freq Cutoff	**PDL** Predelay
PAGE 2	**BAS** Bass Multiply	**XOV** Bass Crossover	**RTC** High Freq Cutoff	**DIF** Diffusion	**DCO** Decay Optimization	**MIX** Dry/Wet Mix
PAGE 3	**LV1** Echo Level 1 L>L	**LV2** Echo Level 2 R>R	**LV3** Echo Level 3 R>L	**LV4** Echo Level4 L>R	**LV6** Echo Level 1 L>L	**LV6** Echo Level 2 R>R
PAGE 4	**DL1** Echo Delay 1 L>L	**DL2** Echo Delay 2 R>R	**DL3** Echo Delay 3 R>L	**DL4** Echo Delay 4 L>R	**DL5** Echo Delay 1 L>L	**DL6** Echo Delay 2 R>R

Parameter placement on the hardware LARC

4.2
Algorithm – Plate/Room

The Room algorithm emulates smaller spaces than the hall algorithm. It is a sound used in all areas of music recording, as well as in video post–production. However, you may find the Random Ambience algorithm better suited to situating dry dialogue in a realistic space.

The Plate algorithm mimics the output of classic plate reverb units. With high initial diffusion and a bright, coloured tail in comparison with other algorithms, these are much used with percussive noises. Plate reverb will often act to become part of the music itself, having a mellowing and thickening effect on the original recording. The Plate reverb algorithm is very familiar from many decades of use in popular music.



NB: The boxes coloured blue on the algorithm diagram represent additional parameters/functions not found in the original hardware.

Parameter Layout

TIME	**RTM** Reverb Time Mid	**BAS** Bass Multiply	**BXO** Bass Freq Crossover	**HIG** High Multiply	**HXO** High Freq Crossover	**TFT** Reverb Filter Type
SHAPE	**SHP** Shape	**SPR** Spread	**SIZ** Size	**PDL** Predelay		**WID** Width
FILTER	**LOF** Low Frequency	**LOS** Low Shelf	**HIF** High Frequency	**HIS** High Shelf		**FLT** Filter Type
MOD		**CHT** Chorus Type	**CHR** Chorus Rate	**CHD** Chorus Depth		
DEN	**DIF** Diffusion		**DCC** Optimization Ch.	**DCO** Decay Optimization	**LEV** Reverb Level	**MIX** Dry/Wet Mix
ERTM	**DL1** Echo Delay 1 L>L	**DL2** Echo Delay 2 L>R	**DL3** Echo Delay 3 L>L	**DL4** Echo Delay 4 R>R	**DL5** Echo Delay 5 R>L	**DL6** Echo Delay 6 R>R
ERVL	**LV1** Echo Level 1 L>L	**LV2** Echo Level 2 L>R	**LV3** Echo Level 3 L>L	**LV4** Echo Level4 R>R	**LV5** Echo Level 5 R>L	**LV6** Echo Level 6 R>R

4.2
Algorithm – Plate/Room (continued)

Software vs Hardware

The same differences as regards **Reverb Level**, modulation and filters apply as for the Hall algorithm.

In addition, there are some minor differences in early reflections. The original hardware had limited DSP power, and a result the reverb designer had to remove some of the early taps of the Plate/Room algorithm to ensure there was enough power for the diffusion module found in the Plate/Room algorithm (which is denser than the diffusion module found in the Hall algorithm).

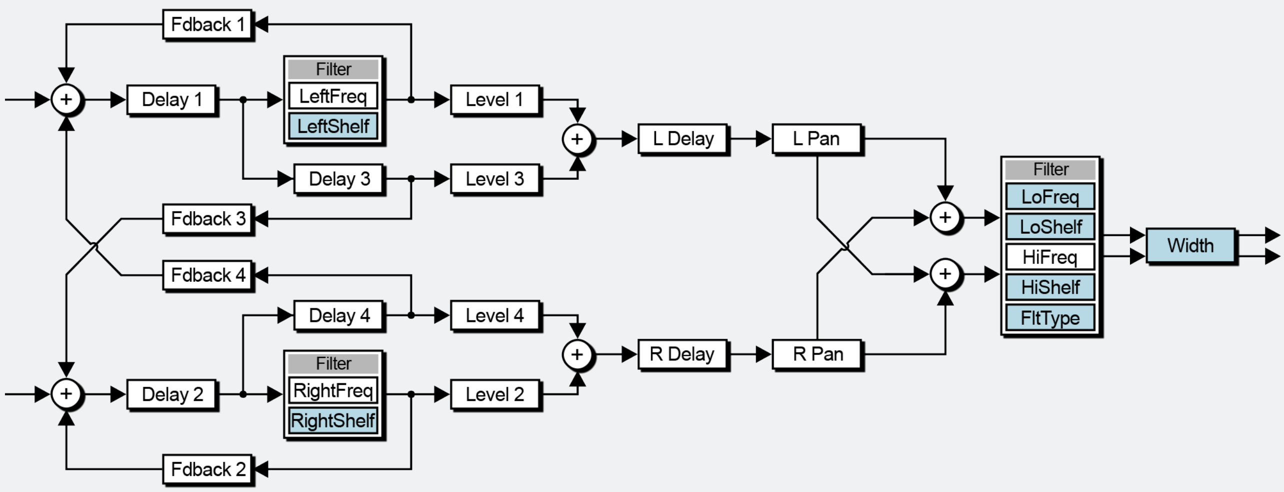
PAGE 1	**RTM** Reverb Time Mid	**SHP** Shape	**SPR** Spread	**SIZ** Size	**HFC** High Freq Cutoff	**PDL** Predelay
PAGE 2	**BAS** Bass Multiply	**XOV** Bass Crossover	**RTC** High Freq Cutoff	**DIF** Diffusion	**DCO** Decay Optimization	**MIX** Dry/Wet Mix
PAGE 3	**LV1** Echo Level 1 L>L	**LV2** Echo Level 2 R>R				
PAGE 4	**DL1** Echo Delay 1 L>L	**DL2** Echo Delay 2 R>R				

Parameter placement on the hardware LARC

4.3
Algorithm – Twin Delays

The Twin Delays algorithm offers four delay lines, each with independent level, feedback and delay time. Feedback can be positive or negative. The feedback for Delays 1 (L) and 2 (R) is preceded by a low pass filter on each line. The feedback for Delays 3 (L) and 4 (R) is cross–panned. Finally, there is a pan control for both Delays 1 and 3 (L) and Delays 2 and 4 (R) (both post feedback and filter). The delays go up to 1.5 seconds, and the filters work from full bandwidth down to 120Hz.

The feedback and panning controls and the filters allow this algorithm to be used for anything from classic doubling effects and ping–pong delays to cavernous echoes.



NB: The boxes coloured blue on the algorithm diagram represent additional parameters/functions not found in the original hardware.

Parameter Layout

TIME	**MST** Delay Multiplier	**ROL** L Del 1 Rolloff	**RSL** L Del 1 Shelf	**ROR** R Del 2 Rolloff	**RSR** R Del 2 Shelf	**TFT** Filter Type
SHAPE	**DLL** L Ch Dry Level	**DLR** R Ch Dry Level		**PNL** Left Ch Pan	**PNR** Right Ch Pan	**WID** Width
FILTER	**LOF** Low Frequency	**LOS** Low Shelf	**HIF** High Frequency	**HIS** High Shelf		**FLT** Filter Type
MOD	**FNL** L Fine Delay	**FNR** R Fine Delay				
DEN	**FB1** L Del 1 Feedback	**FB2** R Del 2 Feedback	**FB3** L-R Del 3 Feedback	**FB4** R-L Del 4 Feedback	**LEV** Level	**MIX** Dry/Wet Mix
ERTM	**DL1** Left Delay 1	**DL2** Right Delay 2	**DL3** Left Delay 3	**DL4** Right Delay 4		
ERVL	**LV1** Left Level 1	**LV2** Right Level 2	**LV3** Left Level 3	**LV4** Right Level 4		

Software vs Hardware

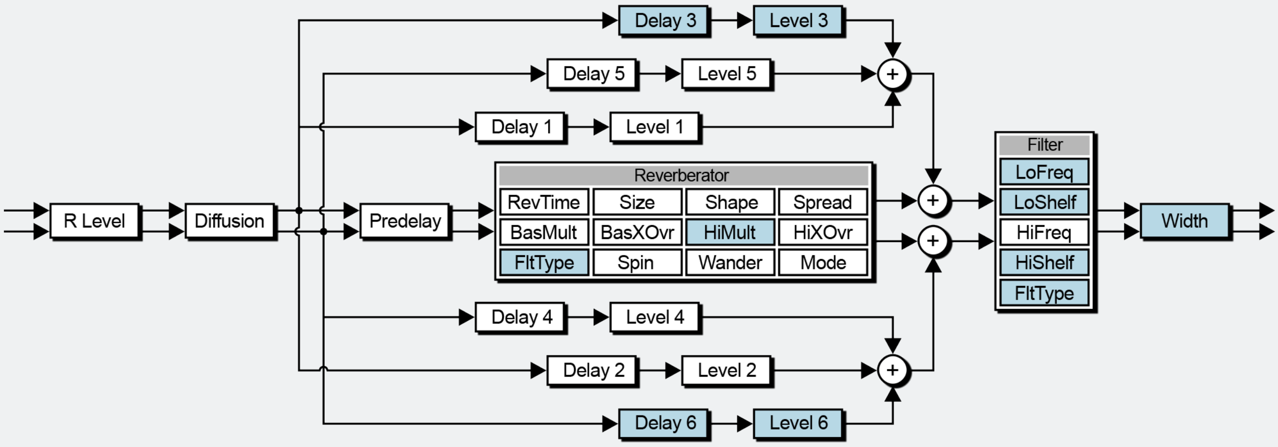
The same differences as regards filters apply as for the Hall algorithm.

PAGE 1	**DRY** L Ch Dry Level	**DRY** R Ch Dry Level	**ROL** L DLY1 Rolloff	**ROL** R DLY2 Rolloff	**PAN** L Channel Pan	**PAN** R Channel Pan
PAGE 2	**DL1** L DLY1 Value	**LV1** L DLY1 Level	**FB1** L DLY1 Feedback	**DL2** R DLY2 Value	**LV2** R DLY2 Level	**FB2** R DLY2 Feedback
PAGE 3	**DL3** L DLY3 Value	**LV3** L DLY3 Level	**FB3** L DLY3 Feedback	**DL4** R DLY4 Value	**LV4** R DLY4 Level	**FB4** R DLY4 Feedback
PAGE 4	**FIN** L Fine Delay	**FIN** R Fine Delay	**MST** Delay Multiplier			

Parameter placement on the hardware LARC

4.4
Algorithm – Random Hall

The Random Hall algorithm provides a smoother reverberant characteristic, and is better suited for material which requires large space emulation, or longer reverb time. It is similar in structure to the Hall algorithm, but with added random decay elements. The reverberation has less metallic ringing and less colouration.



NB: The boxes coloured blue on the algorithm diagram represent additional parameters/functions not found in the original hardware.

Parameter Layout

TIME	**RTM** Reverb Time Mid	**BAS** Bass Multiply	**BXO** Bass Freq Crossover	**HIG** High Multiply	**HXO** High Freq Crossover	**TFT** Reverb Filter Type
SHAPE	**SHP** Shape	**SPR** Spread	**SIZ** Size	**PDL** Predelay		**WID** Width
FILTER	**LOF** Low Frequency	**LOS** Low Shelf	**HIF** High Frequency	**HIS** High Shelf		**FLT** Filter Type
MOD			**SPN** Spin	**WAN** Wander		
DEN	**DIF** Diffusion			**MOD** Reverb Mode	**LEV** Reverb Level	**MIX** Dry/Wet Mix
ERTM	**DL1** Echo Delay 1 L>L	**DL2** Echo Delay 2 L>R	**DL3** Echo Delay 3 L>L	**DL4** Echo Delay 4 R>R	**DL5** Echo Delay 5 R>L	**DL6** Echo Delay 6 R>R
ERVL	**LV1** Echo Level 1 L>L	**LV2** Echo Level 2 L>R	**LV3** Echo Level 3 L>L	**LV4** Echo Level4 R>R	**LV5** Echo Level 5 R>L	**LV6** Echo Level 6 R>R

Software vs Hardware

The same differences as regards filters apply as for the Hall algorithm.

In addition, there are some minor differences in early reflections. The original hardware had limited DSP power, and a result the reverb designer had to remove some of the early taps of the Random Hall algorithm to ensure there was enough power for the advanced modulation found in this algorithm.

PAGE 1	**RTM** Reverb Time Mid	**SHP** Shape	**SPR** Spread	**SIZ** Size	**HFC** High Freq Cutoff	**PDL** Predelay
PAGE 2	**BAS** Bass Multiply	**XOV** Bass Crossover	**RTC** High Freq Cutoff	**DIF** Diffusion	**MOD** Reverb Mode	**MIX** Dry/Wet Mix
PAGE 3	**DL1** Echo Delay 1 L>L	**DL2** Echo Delay 2 R>R	**DL3** Echo Delay 3 R>L	**DL4** Echo Delay 4 L>R	**SPN** Spin	**WAN** Wander
PAGE 4	**LV1** Echo Level 1 L>L	**LV2** Echo Level 2 R>R	**LV3** Echo Level 3 R>L	**LV4** Echo Level4 L>R	**SHL** Shelf	**LEV** Reverb Level

Parameter placement on the hardware LARC

4.5
Algorithm – RHall HD

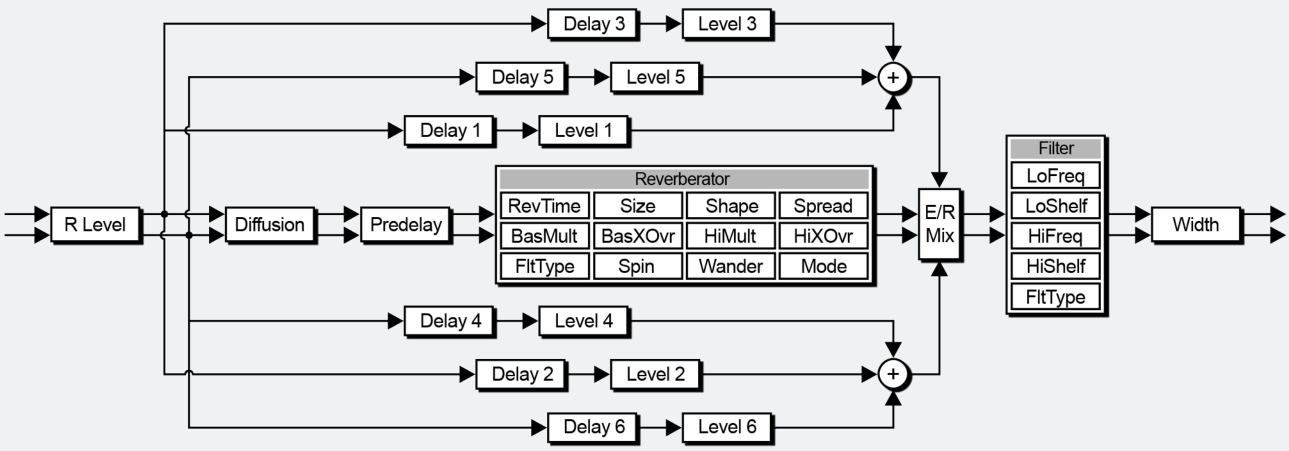
DSP performance was limited in 1984 compared to what is available to modern hardware units. This enhanced version of the original Random Hall algorithm expands and improves upon the original Random Hall algorithm. Because current CPU processing power is way ahead of the processing available in 1984, it has been possible to make some improvements:

» **Greater density,** reaching similar density to that found in convolution reverbs. This is clearly observable with a short transient test sound.

» **Highly advanced echo system:** normally, echoes and early reflections in reverbs are based on a single echo tap or a group of taps, which will introduce (unwanted) colouration of the sound. This new system emulates clusters of 1000s of reflections based on real–world behaviour including advanced randomization. This allows the recreation of the first 100ms of a real hall or a convolution reverb, but without forcing a specific coloration onto the sound.

» **Better low frequency reverberation** compared to the original (it can reverberate lower frequencies).

» **More stable imaging** with shorter reverb time and modulation. If you set **Shape** and **Spread** to the value 0 and have no or very short reverb time, then the imaging of the reverberation changes from left to right and back again constantly – it sounds as if the position of the sound source is moving around the simulated space at high speed – this is eliminated in this algorithm.

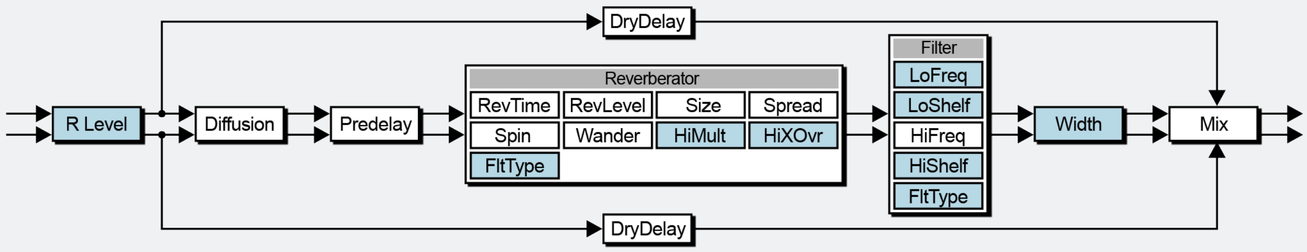


Parameter Layout

TIME	**RTM** Reverb Time Mid	**BAS** Bass Multiply	**BXO** Bass Freq Crossover	**HIG** High Multiply	**HXO** High Freq Crossover	**TFT** Reverb Filter Type
SHAPE	**SHP** Shape	**SPR** Spread	**SIZ** Size	**PDL** Predelay		**WID** Width
FILTER	**LOF** Low Frequency	**LOS** Low Shelf	**HIF** High Frequency	**HIS** High Shelf		**FLT** Filter Type
MOD			**SPN** Spin	**WAN** Wander		
DEN	**DIF** Diffusion		**ERM** ER / Reverb Mix	**MOD** Reverb Mode	**LEV** Reverb Level	**MIX** Dry/Wet Mix
ERTM	**DL1** Echo Delay 1 L>L	**DL2** Echo Delay 2 L>R	**DL3** Echo Delay 3 L>L	**DL4** Echo Delay 4 R>R	**DL5** Echo Delay 5 R>L	**DL6** Echo Delay 6 R>R
ERVL	**LV1** Echo Level 1 L>L	**LV2** Echo Level 2 L>R	**LV3** Echo Level 3 L>L	**LV4** Echo Level4 R>R	**LV5** Echo Level 5 R>L	**LV6** Echo Level 6 R>R

4.6
Algorithm – Ambience

Unlike the Hall, Room and Random hall algorithms, the Ambience algorithm is designed to become part of the input signal, rather than adding a cushion of reverberance that sits behind a clear direct sound. The Ambience algorithm is ideal for adding a room sound to recorded music or speech, and is often used in post–production to add a realistic room ambience to close–miked or studio–recorded dialog. It can also be used to add blend and distance to close–miked instruments.



NB: The boxes coloured blue on the algorithm diagram represent additional parameters/functions not found in the original hardware.

Parameter Layout

TIME	**RTM** Reverb Time Mid	**RTL** Reverb Tail Level		**HIG** High Multiply	**HXO** High Freq Crossover	**TFT** Reverb Filter Type
SHAPE			**AMS** Size	**PDL** Predelay	**IND** Dry Delay	**WID** Width
FILTER	**LOF** Low Frequency	**LOS** Low Shelf	**HIF** High Frequency	**HIS** High Shelf		**FLT** Filter Type
MOD			**SPN** Spin	**WAN** Wander		
DEN	**DIF** Diffusion				**LEV** Reverb Level	**MIX** Dry/Wet Mix
ERTM						
ERVL						

Software vs Hardware

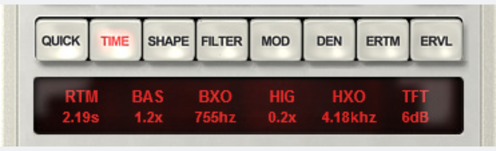
The same differences as regards **Reverb Level** and filters apply as for the Hall algorithm.

Moreover, the original algorithm doesn't have the possibility to change the rolloff frequency within the reverb network. The frequency is hardcoded to 4.62kHz in the original hardware.

PAGE 1	**RTM** Reverb Time Mid	**RTL** Reverb Level	**SIZ** Size	**ROL** Rolloff	**DIF** Diffusion	**MIX** Dry/Wet Mix
PAGE 2	**SPN** Spin	**WAN** Wander	**PDL** Predelay	**IND** Dry Delay		**MIX** Dry/Wet Mix

Parameter placement on the hardware LARC

5.0
Parameters – TIME Page



Hall, Plate/Room, Random Hall and Random Hall HD algorithms

Reverb Time Mid – RTM

Note: This is a continuous parameter on the plugin, which interpolates between the discrete values found on the original.

Range: 0s to 66s (depending on the parameter Size (SIZ))

Sets the decay time of the reverberation algorithm at mid frequencies. The **Bass Crossover** (**BXO**) parameter determines where these frequencies lie. **Reverb Time Mid** acts as a master control for the reverb time, since the low frequency and high frequency reverb time parameters are a multiplier of the **Reverb Time Mid** parameter. When the **Reverb Mode** (**MOD**) parameter is in Effect mode, the interaction between **Reverb Time Mid** and **Size** (**SIZ**) is deactivated. When the **Reverb Mode** (**MOD**) parameter is in Reverb mode, the interaction is activated and the actual value set for **Reverb Time Mid** varies with the setting of **Size** (**SIZ**). **Size** (**SIZ**) should be adjusted before **Reverb Time Mid** when a specific decay time is necessary.

Bass Multiply – BAS

Range: 0.2x to 4.0x

Controls the decay time for frequencies below the **Bass Crossover** (**BXO**) frequency in relation to **Reverb Time Mid** (**RTM**). The **Bass Multiply** parameter is a multiplier of the **Reverb Time Mid** (**RTM**) parameter. If **Bass Multiply** is set at 1.5x and the **Reverb Time Mid** (**RTM**) is set to 2.0 seconds, the low frequency reverb time will be 3.0 seconds. To emulate a natural sounding concert hall, we recommend values around 1.2x to 1.3x.

Bass Crossover – BXO (XOV – Lexicon 480L)

Note: This is a continuous parameter on the plugin, which interpolates between the discrete values found on the original.

Range: 120Hz to Full

Sets the frequency where the transition between low frequency and mid frequency takes place. Typical values for real halls are below 400Hz.

High Multiply – HIG

Note: This parameter is not available on the original. Set the value to 0.2x for true emulation.

Range: 120Hz to Full

Controls the decay time for frequencies above the **High Crossover** (**HXO**) frequency in relation to **Reverb Time Mid** (**RTM**). The **High Multiply** parameter is a multiplier of the **Reverb Time Mid** (**RTM**) parameter. It controls the amount of air absorption in the reverberation.

High Crossover – HXO (RTC – Lexicon 480L)

Note: This is a continuous parameter on the plugin, which interpolates between the discrete values found on the original. Set the value to 4.62kHz when using the Ambience algorithm for true emulation.

Range: 493Hz to 21.18Hz

Sets the frequency where the transition between mid frequency and high frequency takes place. The parameter (in combination with **High Multiply** (**HIG**)) simulates the effect of air absorption in real environments and reduces the reverb time at high frequencies – the amount of reduction is controlled by **High Multiply** (**HIG**). With a relatively low setting, the reverberation will have a darker tone. A typical setting for simulating real halls is 4.6kHz.

5.0
Parameters – TIME Page (continued)

Reverb Filter Type – TFT
Note: This parameter is not available on the original. Set the value to 6dB for true emulation.

Range: 6dB or 12dB

Controls the steepness of the internal reverberation filters. Switch between 6dB per octave and 12dB per octave.

Twin Delay algorithm

Delay Multiplier – MST
Range: 0.50x to 2.00x

Specific to the Twin Delays algorithm, this parameter is a master multiplier that allows you to adjust the delay times for all four delay lines (**DL1**, **DL2**, **DL3** and **DL4**) simultaneously.

Left Delay Rolloff Freq – ROL
Note: This is a continuous parameter on the plugin, which interpolates between the discrete values found on the original.

Range: 120Hz to Full

Specific to the Twin Delays algorithm, this parameter controls the rolloff frequency for the low pass filter on delay line 1 (left channel).

Left Delay Rolloff Shelf – RSL
Note: This parameter is not available on the original. Set the value to Cut for true emulation.

Range: Cut to 0db

Specific to the Twin Delays algorithm, this parameter controls the amount of reduction above the rolloff frequency (**ROL**) for the low pass filter on delay line 1 (left channel).

Right Delay Rolloff Freq – ROR (ROL – Lexicon 480L)
Note: This is a continuous parameter on the plugin, which interpolates between the discrete values found on the original.

Range: 120Hz to Full

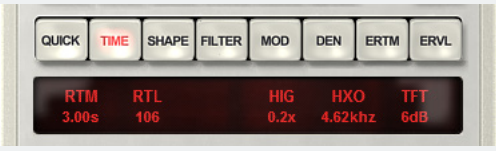
Specific to the Twin Delays algorithm, this parameter controls the rolloff frequency for the low pass filter on delay line 2 (right channel).

Right Delay Rolloff Shelf – RSR
Note: This parameter is not available on the original. Set the value to Cut for true emulation.

Range: Cut to 0db

Specific to the Twin Delays algorithm, this parameter controls the amount of reduction above the rolloff frequency (**ROR**) for the low pass filter on delay line 2 (right channel).

5.0
Parameters – TIME Page (continued)



Ambience algorithm

Reverb Tail Level – RTL

Range: 0 to 127

Specific to the Ambience algorithm, this parameter controls the level of the reverberant part of the ambient decay. At 0, only a subset of the early reflections will be present in the sound, and there is an abrupt end to the sound when these later early reflections are gone. A setting of about 70 results in a natural blend of early and late reflections.

5.1
Parameters – SHAPE Page

Hall, Plate/Room, Random Hall and Random Hall HD algorithms

Shape – SHP

Range: 0 to 255

The two parameters **Shape** and **Spread** (**SPR**) work together to control the contour of the initial envelope of the reverberation sound. **Shape** controls the actual contour of the reverberation envelope. Many small and large spaces have an uneven build–up of initial reverberation, and it is this behaviour that **Shape** and **Spread** (**SPR**) control. At low settings the reverberation builds up extremely quickly, with a bright tone, and then starts to decay immediately. As **Shape** is increased, the reverberation builds up more slowly and the overall sound becomes more filtered.

Spread – SPR

Range: 0 to 255 (depending on other parameters: Size (SIZ) and Rev Mode (MOD) / Decay Optimization (DCO))

Controls the time stretch of the initial decay and works in combination with the **Shape** (**SHP**) parameter to form the overall contour of the early part of the reverberation. Setting the parameter to a larger value results in a larger apparent size of the emulated space and in some situations is actually more effective than adjusting the **Size** (**SIZ**) parameter. A low **Spread** setting results in a rapid onset of the reverberation with low or no sustain of the contour.

When the **Reverb Mode** (**MOD**) or the **Decay Optimization** (**DCO**) parameter is in Effect mode, the **Spread** and **Size** (**SIZ**) parameters are unlinked. Normally the **Spread** parameter is linked to the **Size** (**SIZ**) parameter and the actual maximum value for **Spread** depends on selected **Size** (**SIZ**).

5.1
Parameters – SHAPE Page (continued)

Size – SIZ

Range: 1m to 39m (depending on the algorithm)

Sets the apparent size (roughly equal to the longest dimension in meters) and the diffusion build–up rate of the simulated acoustic space. It also functions as a master control for **Reverb Time Mid** (**RTM**), **Spread** (**SPR**) and the modulation engine. The **Size** control can modify the sound of the reverb from very small with high density to very large with lower density. The overall sense of size is actually a combination of the parameters **Size**, **Shape** (**SHP**) and **Spread** (**SPR**).

Predelay – PDL

Range: 0ms to 1500ms

Adjust the time delay between the reverberation and the input signal. Contrary to popular beliefs, it's not recommended to use the **Predelay** parameter for emulating the time delay in natural environments. Natural spaces are best emulated by setting **Shape** (**SHP**) and **Spread** (**SPR**) to the desired values. **Predelay** is best for special effects.

Width – WID

Note: This parameter is not available on the original. Set the value to 45 (Stereo) for true emulation.

Range: –360 to 360

Controls the width of both the reverberation and pre echoes. The parameter is able to perform phase inversion, Left/Right inversion (channel swapping) and surround–compatible L–R/R–L operation.

Value	Display	Description
360	Mono	Mono
315	R, L	R, L stereo (swapped)
270	R-L, L-R	R-L, L-R surround (disappears in Mono)
225	Stereo Inv	Stereo w/ phase inverted
180	Mono Inv	Mono w/ phase inverted
135	R, L Inv	R, L stereo (swapped) w/ phase inverted
90	L-R, R-L	L-R, R-L surround
45	Stereo	Stereo
0	Mono	Mono
-45	R, L	R, L stereo (swapped)
-90	R-L, L-R	R-L, L-R surround (disappears in Mono)
-135	Stereo Inv	Stereo w/ phase inverted
-180	Mono Inv	Mono w/ phase inverted
-225	R, L Inv	R, L stereo (swapped) w/ phase inverted
-270	L-R, R-L	L-R, R-L surround
-315	Stereo	Stereo
-360	Mono	Mono

5.1
Parameters – SHAPE Page (continued)

Twin Delays algorithm

Left Channel Dry Level – DLL (DRY – Lexicon 480L)

Range: Off to 0dB

Specific to the Twin Delays algorithm, this parameter sets the amount of dry signal that passes through unaffected from the left input to the left output.

Right Channel Dry Level – DLR (DRY – Lexicon 480L)

Range: Off to 0dB

Specific to the Twin Delays algorithm, this parameter sets the amount of dry signal that passes through unaffected from the right input to the right output.

Left Channel Pan – PNL (PAN – Lexicon 480L)

Range: Left to Right

Specific to the Twin Delays algorithm, this parameter sets the panning of the left **DL1**/**DL3** (and **FB1**/**FB4**) channel between left and right, before the final filter stage.

Right Channel Pan – PNR (PAN – Lexicon 480L)

Range: Left to Right

Specific to the Twin Delays algorithm, this parameter sets the panning of the right **DL2**/**DL4** (and **FB2**/**FB3**) channel between right and left, before the final filter stage.

Ambience algorithm

Ambience Size – AMS (SIZ – Lexicon 480L)

Range: 1m to 40m

Specific to the Ambience algorithm, this parameter varies the apparent size of the ambient space. Getting this parameter in a promising area is often a good first step to achieving the desired space, before moving on to adjust **Reverb Time Mid** (**RTM**) and **Reverb Tail Level** (**RTL**).

Dry delay – IND

Range: 0ms to 1500ms

Specific to the Ambience algorithm, this parameter sets the amount of delay applied to the dry signal that is mixed in by the **Wet/Dry Mix** (**MIX**) control. Usually this will be set to 0, but it can be invaluable when needed, for instance in sound reinforcement.

5.2
Parameters – FILTER Page

Low Frequency – LOF

Note: This parameter is not available on the original.

Range: 120Hz to Full

Sets the frequency below which a high–pass filter attenuates both the reverberation and the early reflections. The overall high–pass filtering is actually a combination of the parameters **Low Frequency** and **Low Shelf** (**LOS**).

Low Shelf – LOS

Note: This parameter is not available on the original. Set the value to 0dB for true emulation.

Range: Cut to 0db

Modifies the high–pass filter characteristic of **Low Frequency** (**LOF**) into a shelving filter. Setting the value to the minimum of Cut will modify the shelving filter into a normal high–pass filter. The normal application for the parameter is to reduce "boomy" resonances.

High Frequency – HIF (HFC – Lexicon 480L)

Note: This is a continuous parameter on the plugin, which interpolates between the discrete values found on the original.

Range: 120Hz to Full

Sets the frequency above which a low–pass filter attenuates both the reverberation and the early reflections. The overall low–pass filtering is actually a combination of the parameters **High Frequency** and **High Shelf** (**HIS**). This parameter is important for creating a realistic and natural reverberation. Typical values are 2kHz to 4kHz.

High Shelf – HIS

Note: This parameter is not available on the original. Set the value to Cut for true emulation.

Range: Cut to 0db

Modifies the low–pass filter characteristic of **High Frequency** (**HIF**) into a shelving filter. A value of 0dB will deactivate the filter.

Filter Type – FLT

Note: This parameter is not available on the original. Set the value to 6dB for true emulation.

Range: 6dB or 12dB

Controls the steepness of the two output filters. Switch between 6dB per octave and 12dB per octave.

5.3
Parameters – MOD Page

Hall and Plate/Room algorithms

Chorus Type – CHT

Note: This parameter is not available on the original. Set the value to 0 for true emulation.

Range: –7 to 7

Selects between the various different chorusing modes. With a value of 0 the chorusing modulation is completely off. Settings 1–3 increase the number of internal delay lines. Settings 4–7 are identical to settings 0–3, but with added chorusing to the outputs. Settings with a negative number have the various delay lines randomized so that they are completely independent of each other, which results in a more chaotic pitch variation.

Chorus Rate – CHR

Note: This parameter is not available on the original. Set the value to 0 for true emulation.

Range: 0 to 48

Controls the rate of the chorusing modulation.

Chorus Depth – CHD

Note: This parameter is not available on the original. Set the value to 0 for true emulation.

Range: 0 to 64

Controls the amount of the chorusing effect on the reverberation. Because this parameter (together with **CHT** and **CHR**) introduces pitch variations, sound sources with very little pitch wobble, such as piano, should have a low **Chorus Depth** value.

Twin Delays algorithm

Left Fine Delay – FNL (FIN – Lexicon 480L)

Range: 0 smp to 49 smp

Specific to the Twin Delays algorithm, this parameter controls an extra, sample–level delay for the left **DL1**/**DL3** (and **FB1**/**FB4**) channel.

Right Fine Delay – FNR (FIN – Lexicon 480L)

Range: 0 smp to 49 smp

Specific to the Twin Delays algorithm, this parameter controls an extra, sample–level delay for the right **DL2**/**DL4** (and **FB2**/**FB3**) channel.

5.3
Parameters – MOD Page (continued)

Random Hall, Random Hall HD and Ambience algorithms

Spin – SPN

Range: 0 to 48

Controls the rate of the randomization in the reverberation engine. **Spin** and **Wander** (**WAN**) continuously change the overall timbre of the reverberation to reduce coloration and make the sound more natural by altering the many delay taps in the algorithm. The modulation engine is designed to reduce the chorusing associated with the varying time delay found in older reverberation algorithms to handle some instruments, like piano, better, without pitch–shifting.

Wander – WAN

Range: 0 to 25ms (depending on the parameter Size (SIZ))

Sets the amount of time the many delay taps in the algorithms will move. For large environments use a setting of around 10ms.

5.4
Parameters – DEN Page

Hall and Plate/Room algorithms

Diffusion – DIF

Range: 0 to 100%

Controls the initial echo density of the reverb. Low or moderate settings of **Diffusion** can result in an immediate and clear sound, with distinct reflections if **Size** (**SIZ**) is large (and therefore with lower density). This is preferable when you want clearer and more natural vocals, mixes and piano music. Higher settings of **Diffusion** will enhance the sound of percussion or other sounds with sharp transients. This parameter affects both reverberation and early reflections.

Decay Optimization Channel – DCC

Note: This parameter is not available on the original. Set the value to Left for true emulation.

Range: LEFT or RIGHT

Controls which channel receives the decay optimization specified by the **DCO** parameter. In rare circumstances (less than 5% of all power up) the original hardware swaps the decay optimization modulation from the left channel to the right channel.

5.4
Parameters – DEN Page (continued)

Decay Optimization – DCO

Range: 0 RVB to 9 RVB, 0 EFX to 9 EFX

This parameter alters the overall characteristics of the algorithm with respect to its response to input levels, which can have an effect on the realism of the reverberation. The default setting is RVB 7. The nine EFX settings are similar to the RVB settings, except that they delink the **Size** (**SIZ**) and **Spread** (**SPR**) parameters, allowing the creation of unnatural, experimental sounds, such as those with a high **Spread** (**SPR**) but low **Size** (**SIZ**).

Reverb Level – LEV

Note: Set the value to 160 when using the Hall, Plate/Room and Ambience algorithms for true emulation.

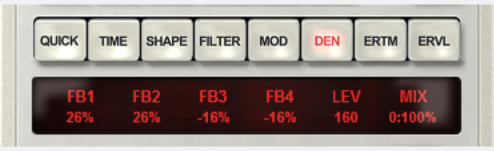
Range: 0 to 254

Controls the input gain of the reverberation engine. Settings above 160 will result in saturation of the internal calculations. This parameter affects both reverberation and early reflections.

Wet/Dry Mix – MIX

Range: 100:0% to 0:100%

Sets the ratio between processed signal and dry signal in the output. When using the plugin on a send path, the **Wet/Dry Mix** (**MIX**) parameter should almost always be set to 0:100%.



Twin Delays algorithm

Left Delay 1 FeedBack – FB1

Range: –100 to 100%

Specific to the Twin Delays algorithm, this parameter sets the feedback, either positive or negative (phase inverted), around the left delay **DL1**.

Right Delay 2 FeedBack – FB2

Range: –100 to 100%

Specific to the Twin Delays algorithm, this parameter sets the feedback, either positive or negative (phase inverted), around the right delay **DL2**.

L–R Delay 3 FeedBack – FB3

Range: –100 to 100%

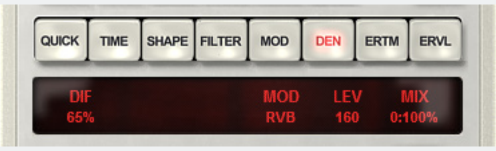
Specific to the Twin Delays algorithm, this parameter sets the cross–panned feedback, either positive or negative (phase inverted), from the left delay **DL3** into the right delay **DL2**.

R–L Delay 4 FeedBack – FB4

Range: –100 to 100%

Specific to the Twin Delays algorithm, this parameter sets the cross–panned feedback, either positive or negative (phase inverted), from the right delay **DL4** into the left delay **DL1**.

5.4
Parameters – DEN Page (continued)

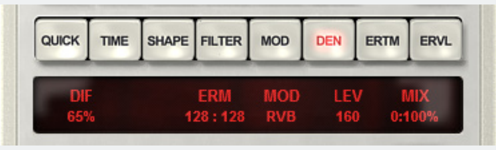


Random Hall algorithm

Reverb Mode – MOD

Range: RVB or EFX

Specific to the Random Hall and Random Hall HD algorithms, this parameter switches between RVB Mode and EFX Mode. The RVB Mode has linked operation for **Reverb Time Mid** (**RTM**), **Spread** (**SPR**) and **Size** (**SIZ**). The EFX Mode permits independent operation of the above–mentioned parameters. The RVB Mode results in the most natural and optimal operation.



Random Hall HD algorithm

ER/Reverb Mix – ERM

Range: 128:0 to 0:128

Specific to the Random Hall HD algorithm, this parameter controls the ratio between reverberation signal and early reflection signal in the output.

5.5
Parameters – ERTM Page

These parameters (and those on the ERLV page) differ slightly from all others as they have different meanings for different algorithms.

Hall, Plate/Room, Random Hall and Random Hall HD algorithms

DL1 – Echo Delay 1 L > L

Range: 0ms to 2500ms

This sets the delay time for the early reflection delay line **DL1** (L input channel, L output channel).

DL2 – Echo Delay 2 L > R

Range: 0ms to 2500ms

This sets the delay time for the early reflection delay line **DL2** (L input channel, R output channel).

DL3 – Echo Delay 3 L > L

Range: 0ms to 2500ms

This sets the delay time for the early reflection delay line **DL3** (L input channel, L output channel).

DL4 – Echo Delay 4 R > R

Range: 0ms to 2500ms

This sets the delay time for the early reflection delay line **DL4** (R input channel, R output channel).

DL5 – Echo Delay 5 R > L

Range: 0ms to 2500ms

This sets the delay time for the early reflection delay line **DL5** (R input channel, L output channel).

DL6 – Echo Delay 6 R > R

Range: 0ms to 2500ms

This sets the delay time for the early reflection delay line **DL6** (R input channel, R output channel).

Twin Delays algorithm

DL1 – Left Delay 1 Value

Range: 0ms to 1491ms

This sets the delay time for the delay line **DL1**.

5.5
Parameters – ERTM Page (continued)

DL2 –Right Delay 2 Value

Range: 0ms to 1491ms

This sets the delay time for the delay line **DL2**.

DL3 – Left Delay 3 Value

Range: 0ms to 1491ms

This sets the delay time for the delay line **DL3**.

DL4 – Right Delay 4 Value

Range: 0ms to 1491ms

This sets the delay time for the delay line **DL4**.

5.6
Parameters – ERVL Page

As with the ERTM page, these parameters have different meanings for different algorithms.

Hall, Plate/Room, Random Hall and Random Hall HD algorithms

LV1 – Echo Level 1 L > L

Range: Off to 0dB

This sets the level for the early reflection delay line **DL1** (L input channel, L output channel).

LV2 – Echo Level 2 L > R

Range: Off to 0dB

This sets the level for the early reflection delay line **DL2** (L input channel, R output channel).

LV3 – Echo Level 3 L > L

Range: Off to 0dB

This sets the level for the early reflection delay line **DL3** (L input channel, L output channel).

LV4 – Echo Level 4 R > R

Range: Off to 0dB

This sets the level for the early reflection delay line **DL4** (R input channel, R output channel).

5.6
Parameters – ERVL Page (continued)

LV5 – Echo Level 5 R > L

Range: Off to 0dB

This sets the level for the early reflection delay line **DL5** (R input channel, L output channel).

LV6 – Echo Level 6 R > R

Range: Off to 0dB

This sets the level for the early reflection delay line **DL6** (R input channel, R output channel).

Twin Delays algorithm

LV1 – Left Delay 1 Level

Range: Off to 0dB

This sets the level for the delay line **DL1**.

LV2 – Right Delay 2 Level

Range: Off to 0dB

This sets the level for the delay line **DL2**.

LV3 – Left Delay 3 Level

Range: Off to 0dB

This sets the level for the delay line **DL3**.

LV4 – Right Delay 4 Level

Range: Off to 0dB

This sets the level for the delay line **DL4**.

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