PCM NATIVE | REVERB BUNDLE

OWNER'S MANUAL

The Lexicon® Legacy continues...

...with the PCM Native Reverb Bundle

For over 40 years Lexicon® has been recognized as the golden standard of digital reverb and effects processing and has continuously introduced leading edge technology for the audio industry. Lexicon has again rocked the audio industry with a complete collection of the finest reverb plug-ins available. The PCM Native Reverb Plug-In Bundle is the ultimate bundle for creating professional, inspirational mixes within popular DAWs like Pro Tools®, Logic®, and any other VST®, Audio Unit™, or RTAS® compatible platform.

With all the flexibility you would expect from native plug-ins, this powerhouse bundle delivers 7 legendary Lexicon reverb plug-ins with hundreds of the most versatile and finely crafted studio presets. Designed to bring the highest level of sonic quality and function to all your audio applications, the PCM Native Reverb Bundle will take center stage in your DAW.

Congratulations and thank you

for purchasing the PCM Native Reverb Plug-in Bundle, an artful blend of seven celebrated Lexicon® reverb plug-ins. With decades of legacy products to pull from, the PCM Native Reverb Bundle includes the finest collection of Factory Presets available. Designed to bring the highest level of sonic quality and function to all of your audio applications, the PCM Native Reverb Bundle will take center stage in your DAW.

Quick Start

- Choose one of the seven Lexicon plug-ins. Each plug-in contains a different algorithm:

Chamber	(LexChamber)
Hall	(LexHall)
Random Hall	(LexRandomHall)
Plate	(LexPlate)
Vintage Plate	(LexVintagePlate)
Concert Hall	(LexConcertHall)
Room	(LexRoom)

- In the plug-in's window, select a category
- Select a preset
- Adjust parameters (optional)
- Save the preset (optional)

It can be as simple or as in-depth as you'd like. The hundreds of included presets work well for most situations, but you can easily adjust any parameter and save any preset. See the "Factory Presets" section for more information on editing, loading and saving presets.

Included Items

- PCM Native Reverb Bundle software CD
- iLok license code
- This manual

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Installation

Install the Plug-in Bundle

Insert the CD into your CD-ROM drive. The Installer should start automatically.

If the Installer doesn't start automatically, you can start it manually by opening the appropriate file on the CD:

Mac®:	PCM Native Reverb Plug-in.mpkg
Windows®:	setup.exe

Once the Installer is open, follow the on-screen instructions to install the software you wish to use.

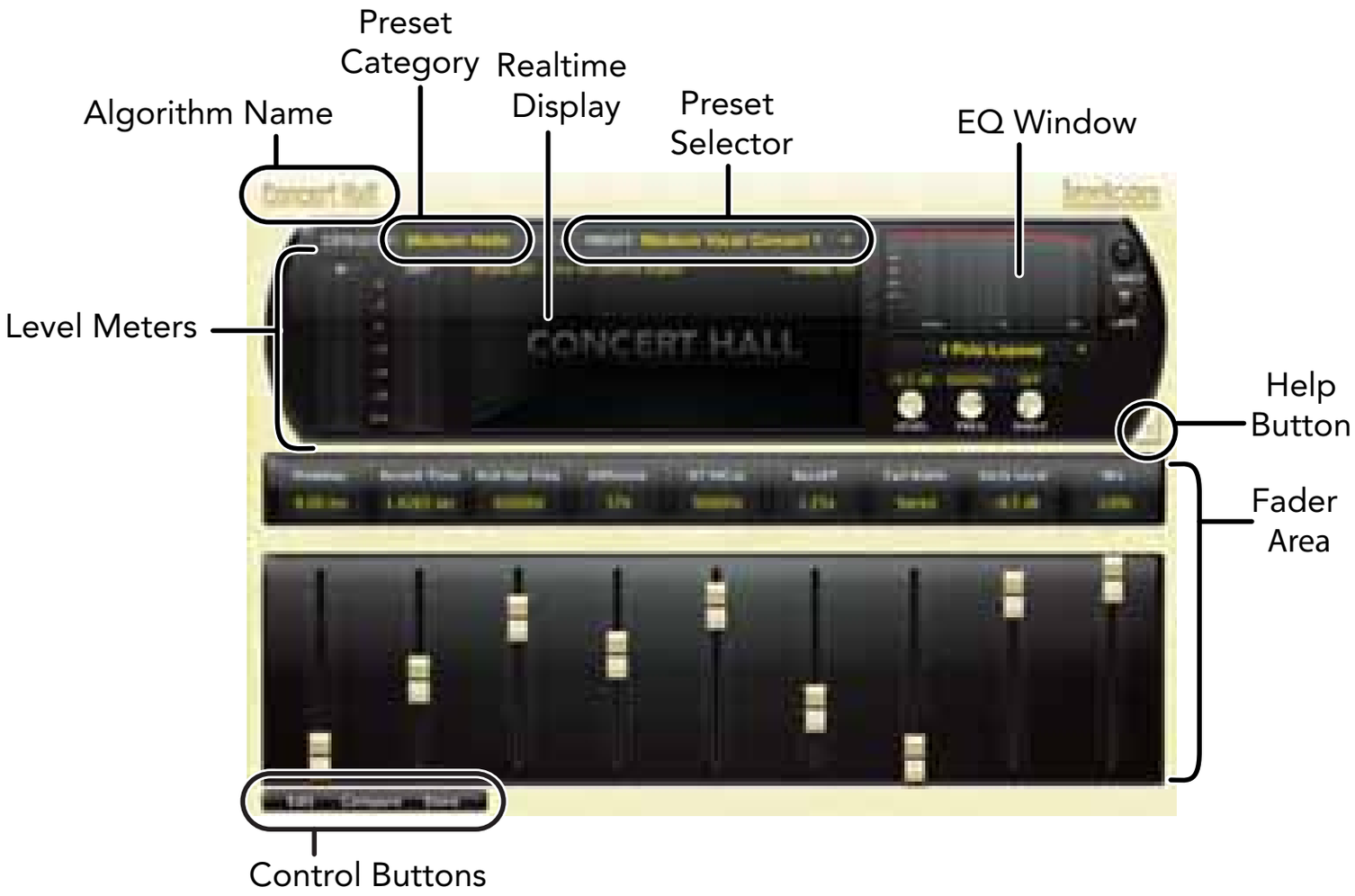
iLok License

You must download a license to your iLok USB smart key before running any program that might use these plug-ins. If a valid iLok license is not present, the initial plug-in validation scan will fail and you may not be able to run the plug-ins without digging into your system. For more information go to www.iLok.com.

Important! Your PCM Native Reverb Bundle will not work without an authorized iLok USB smart key inserted in your computer's USB port!

First Look - The User Interface

When the plug-in is instantiated, you will see a window that looks something like this:



Algorithm Name
Displays the name of the currently selected algorithm.

Preset Category
Presets for the algorithm are grouped into categories to make them easier to find. For example, the Hall algorithm is broken into Small Halls, Medium Halls and so on. Clicking in this area will show the list of categories. Selecting a category will load the presets for that category in the Preset Selector.

Preset Selector
Displays the list of presets within the selected category. Selecting a preset from the list will cause the preset to be loaded into the plug-in.

Realtime Display
Three unique visualizations help you to see inside the reverb. This is explained in greater depth in the "Realtime Display" section of this manual.

EQ Window
Allows you to see the type of EQ filters that are applied to the early and late reverb signals. This is explained in greater depth in the "EQ Window" section of this manual.

Level Meters
Show the input and output levels.

Help Button
Click on the Help button to enable tooltip help. Then hover your mouse cursor over the button or parameter you want to know about. An explanation will appear onscreen.

Fader Area
Parameter values are shown and controlled in this area. This is explained in more depth in the "Fader Area" section of this manual.

Control Buttons
These let you access all of a preset's parameters, manage presets, and compare changes you've made to the original settings. This is explained in more detail in the "Control Buttons" section of this manual.

The Realtime Display

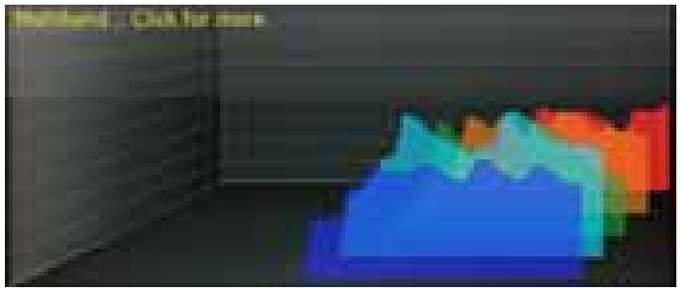
The realtime display window allows you to visualize the reverberation in three ways. While your ears will always be the primary tool in choosing and adjusting presets, the displays provide a direct way to see what's going on. Click anywhere in the realtime display window to change the view.

Off

This mode shows only the algorithm name. It is the default view for the plug-in and requires the least amount of CPU cycles.

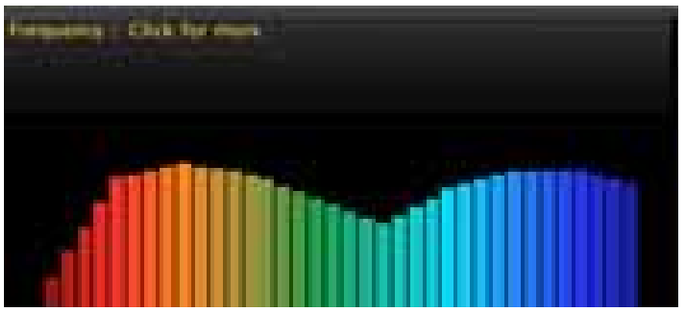
Multiband

This display shows the reverb in five frequency bands, with the lowest frequency in the rear. The image moves from right to left as it ages.



Frequency

This display shows the reverb as a more traditional RTA image, with lower frequencies on the left.

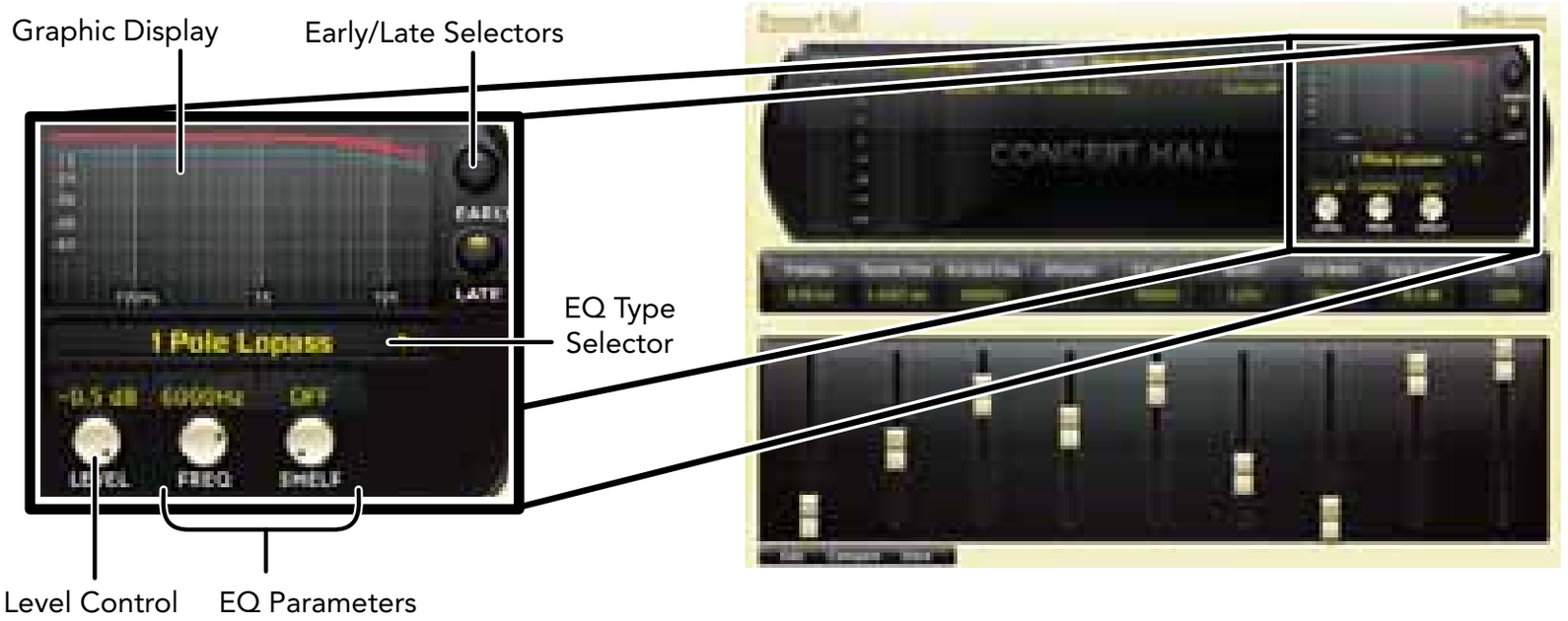


Impulse

This display shows the signal as a single impulse response, The image moves to the left as it ages.

The EQ Window

This window provides control of the output equalization:



Graphic Display
Shows the EQ in a graphic form. The active section (Early or Late) is in the forefront.

Early/Late Selectors
These buttons select which EQ section is currently being edited. 'Early' includes reflections, direct echo outputs and room patterns. 'Late' includes the reverb tail.

EQ Type Selector
Allows any of six types of EQ to be applied to the active section. Those types are:

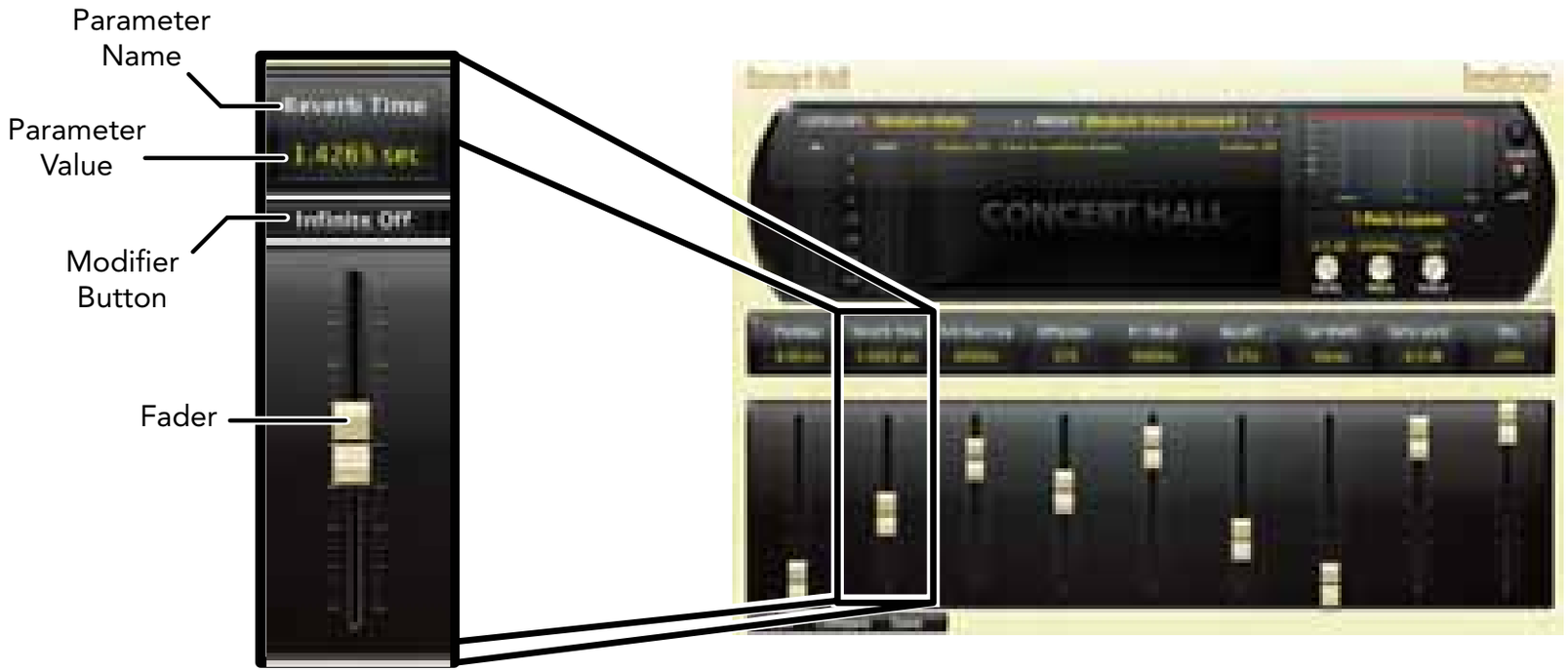
- Single-pole lowpass (6db / Octave)
- Single-pole highpass (6db / Octave)
- Double-pole lowpass (12db /Octave)
- Double-pole highpass (12db / Octave)
- Bandpass
- Notch

Level Control
Overall output level of the active section. These may be edited by holding the mouse down and moving up and down over the knob. Alternatively, the value field may be directly edited.

EQ Parameters
Frequency, shelf and bandwidth parameters of the active section. To edit these, click and hold on the knob, then move the mouse up or down. Alternatively, the value field may be directly edited. The bandwidth knob is only available when the filter type is Bandpass or Notch.

The Fader Area

The fader area may be populated by up to nine faders. This illustration shows what may be found for each fader.



Parameter Name

Shows the parameter being adjusted.

Parameter Value

Shows the current value of the parameter. This field may be directly edited.

Modifier Button

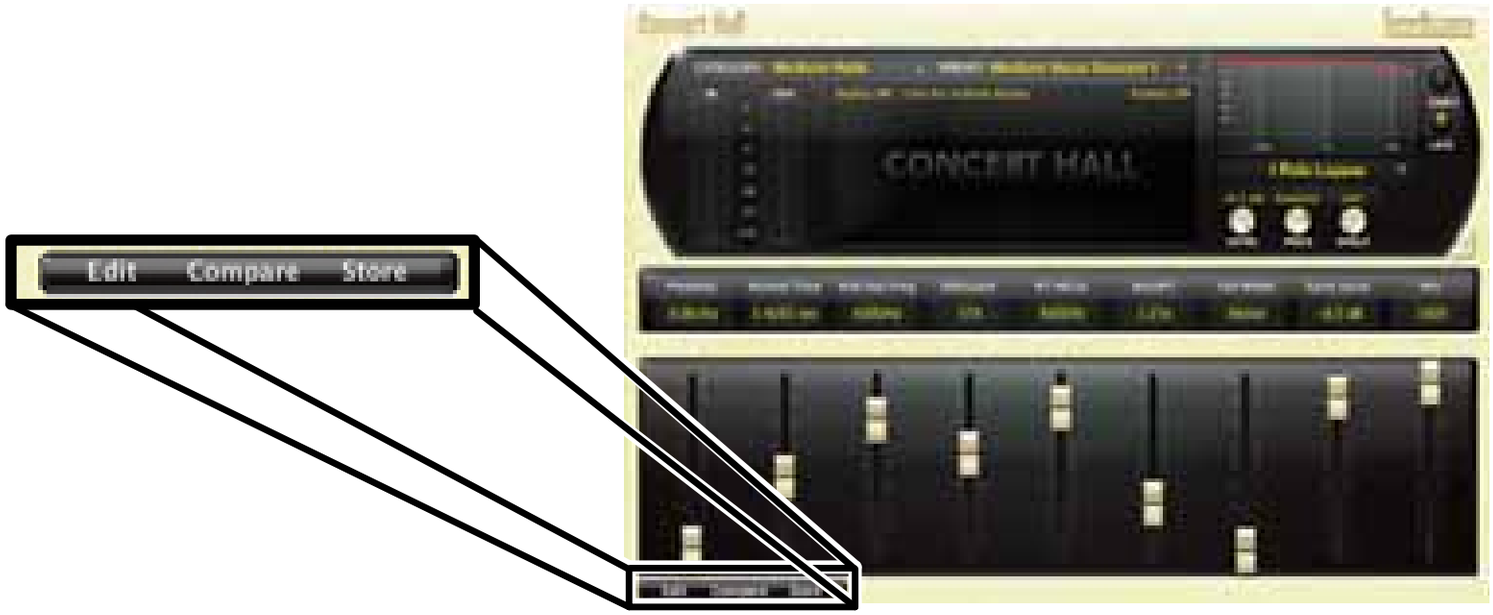
This button only appears for certain types of parameters. It controls aspects of the parameter. In this illustration, it serves as a button to send the reverb into infinite reverberation.

Fader

Changes the value of the parameter. May be dragged, control-dragged (for higher resolution) or may be controlled by the mouse wheel.

The Control Buttons

These buttons let you access all of a preset's parameters, manage presets, and compare changes you've made to the original settings.



Edit

The Edit button makes the Navigation buttons visible. These buttons allow all of the algorithm's parameters to be edited. See page 10 for more detailed information.

Compare

The Compare button temporarily restores the plug-in to the selected preset. It gives you a chance to see what was changed. While in compare mode, the plug-in is not editable. Click Compare again to restore your edits.

Store

This allows the you to create and manage your own presets. See page 18 for more information.

Edit Navigation and Full Edit Mode

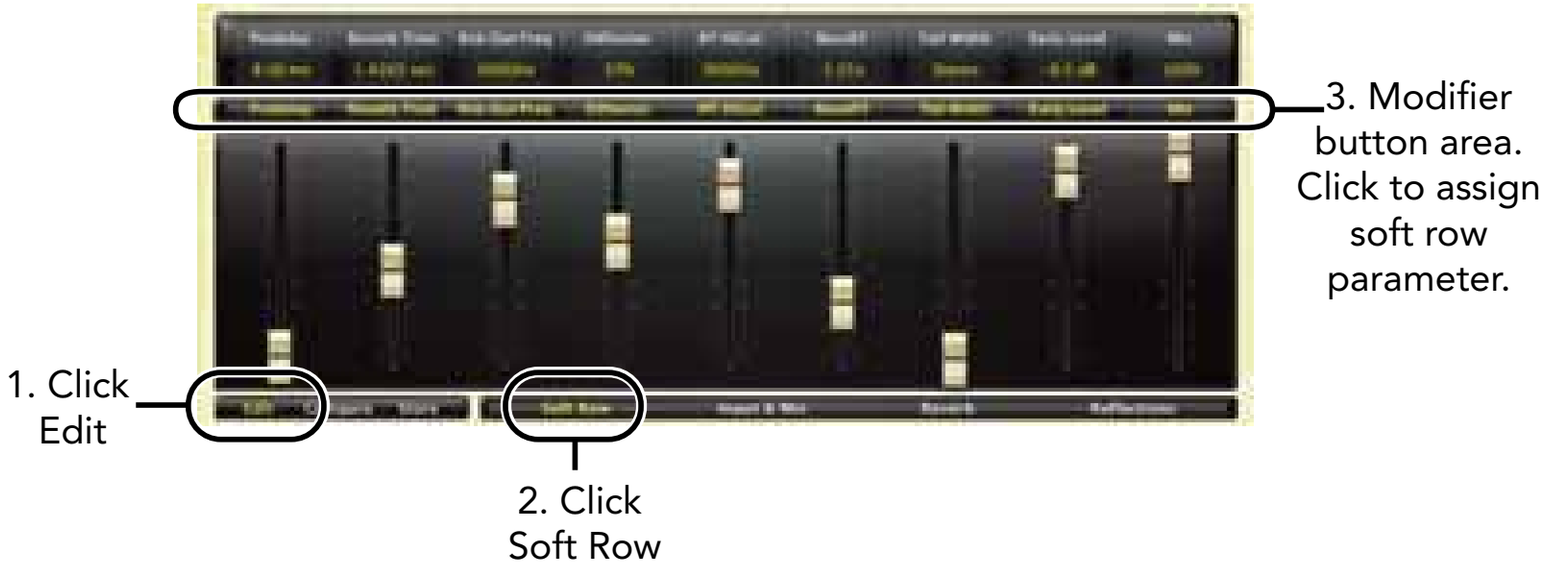
When the "Edit" button is clicked, an additional row of buttons appear along the lower right portion of the plug-in. The names and number of buttons vary among algorithms. Clicking one of the buttons will bring up an edit page holding parameters related to the button name. Those parameters may be edited until the desired sound is achieved.

The Soft Row

When the plug-in is first instantiated, a number of parameters appear. This page of parameters is called the "Soft Row". These parameters were chosen as being the most useful parameter for a particular preset. In most cases, we've tried to make a consistent set of choices, but many presets call for different parameters. For most people, the soft row is all that's needed.

Customizing the Soft Row

The Soft Row assignments are part of the preset. You can change the Soft Row assignments and store the preset as a User preset. To do this, press the "Edit" key to go into edit mode and press the button for "Soft Row". The plug-in will look something like this:



In the modifier button area, just above the fader, you will see the name of the parameter that is assigned to that position in the soft row. Clicking on any of these parameter buttons reveals a list of all the algorithm's parameters. Select a parameter (or "no selection") from the list, and that parameter is then assigned to the soft row. The assigned parameter is still available on its original edit page, but now it appears in the soft row as well.

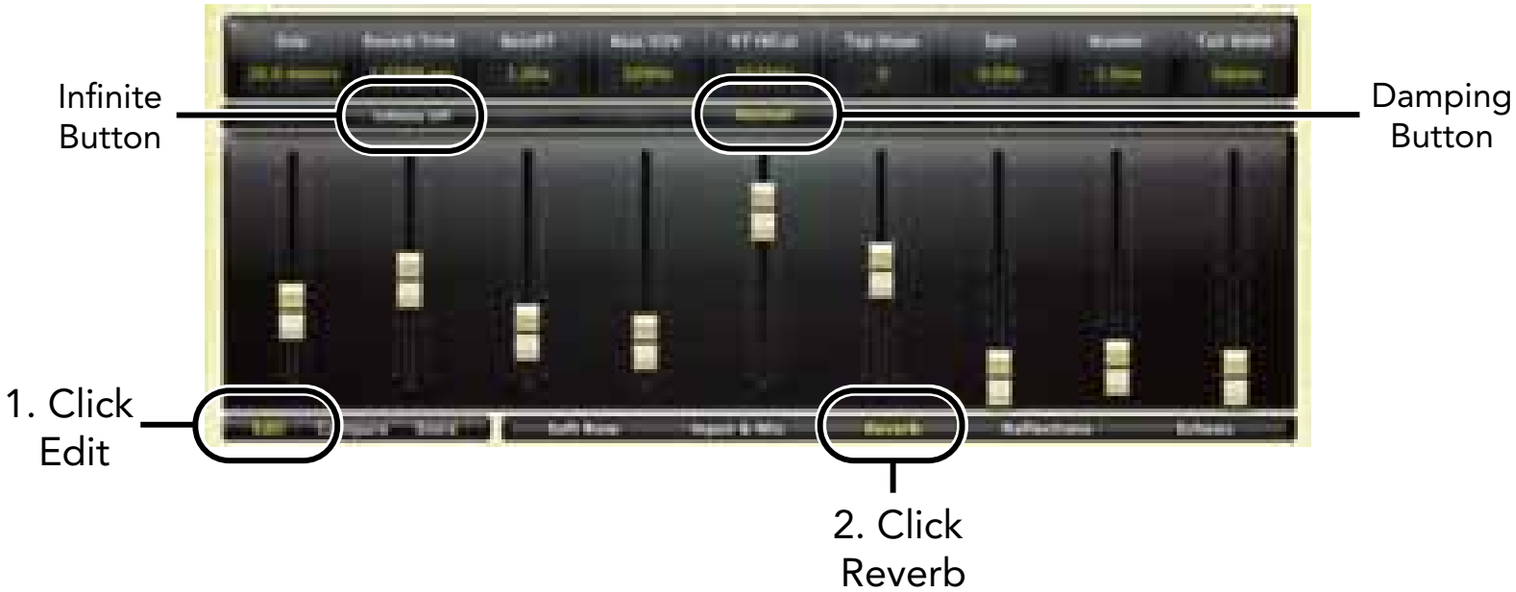
You can assign any parameter, whether it's on an edit page or in the EQ window. You can even assign the same parameter more than once. When you're done, store it as a User Preset and you'll always be able to recall the preset with your customized soft row.

Input and Mix

Clicking the Input Mix button reveals a straightforward group of parameters, all of which are described in the "Parameters" section starting on page 38. The Predelay parameter includes an Absolute/Tempo button. In Absolute Mode, a delay is shown in milliseconds. In Tempo Mode, a rhythmic value is shown and the delay time responds to changes in the tempo using this plug-in. If tempo slows down, delays get longer.

Reverb

The Reverb button opens a selection of parameters that are pertinent to the selected algorithm. Each parameter is described in the "Parameters" section starting on page 38. This edit page also includes an Infinite button (which lets the reverb echoes continue endlessly) and a Damping button, which controls the high frequency damping of the signal during the reverb tail. It may be considered analogous to air absorption.

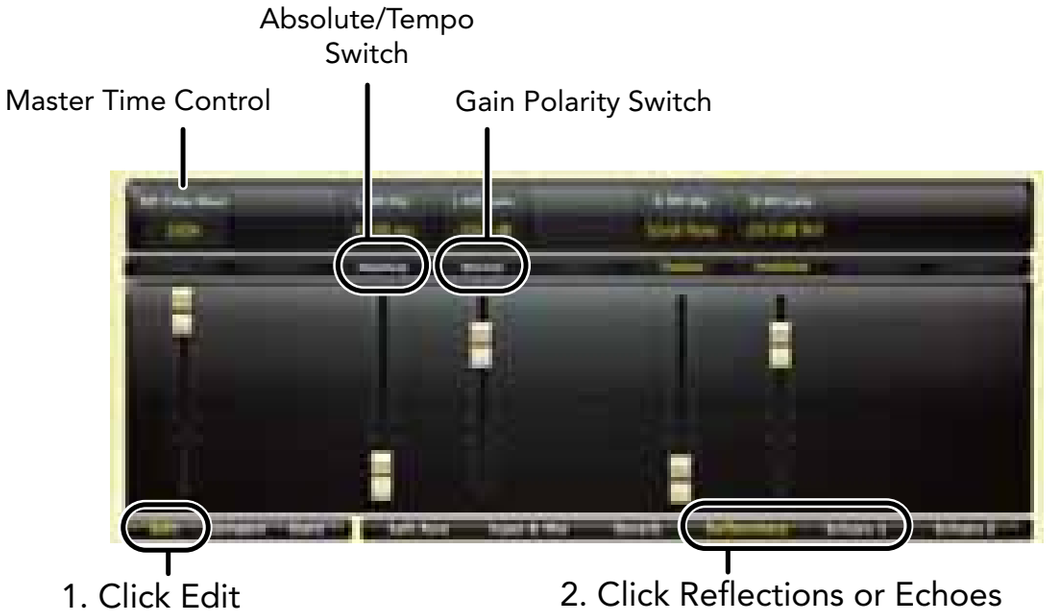


Reflections and Echoes

There are two types of delay voices in the Lexicon® plug-ins:

Reflections are simple delay voices leading from one of the input channels to one of the output channels. Their delay time may be modified, as well as the output level and polarity.
Echoes are similar to reflections but they have an additional control that allows them to feed a delayed signal back into the input.

Both types of voices pass through the input diffusors.



Master Time Control

Scales all reflection voices by a percentage value. Once individual voices are set, this is a handy method for quickly changing the effect of the reverb.

Absolute/Tempo Switch

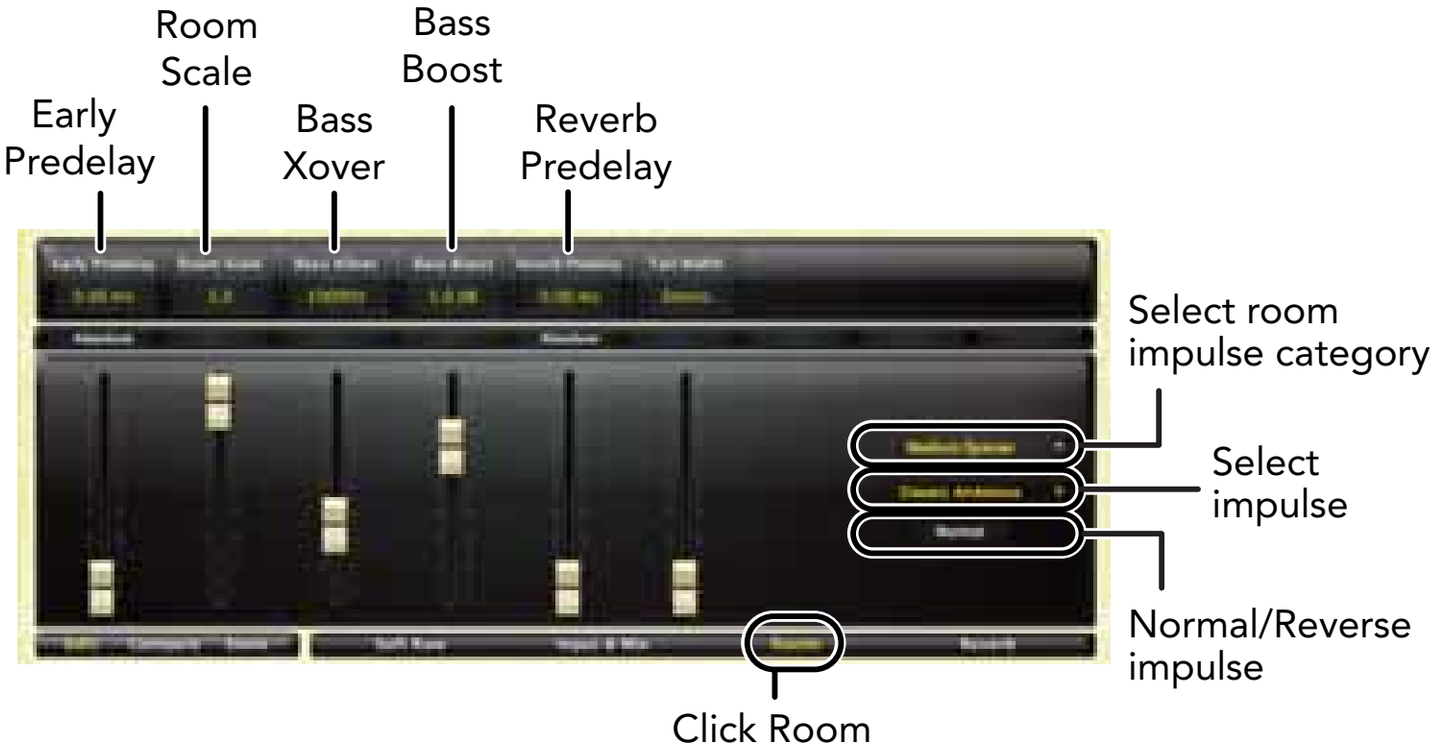
Nearly all delays (including predelays) have this switch. In Absolute Mode, a delay is shown in milliseconds. In Tempo Mode, a rhythmic value is shown and the delay time responds to changes in the tempo using this plug-in. If tempo slows down, delays get longer.

Gain/Polarity Switch

Changes the gain from normal to inverted.

Room Control

Most of the Room algorithm's parameters are self-explanatory, and you can find their descriptions starting on page 38. Here are a few parameters unique to this algorithm:



Predelay

You will notice that there are two predelay parameters:

- **Early Predelay**
  Is an overall predelay that delays all of the signal. As with a more typical reverb, it's often used to create some separation between the wet and dry portions of the signal.

- **Reverb Predelay**
  Is an additional delay that comes between the room impulse and the reverberation section. It can have a large effect on the perception of size and/or liveness. A small value means that reverb will begin while the room impulse is still being output. A larger value will delay the reverb until nearer the end of the room impulse. An even larger value will create actual separation and may be perceived as a slapback.

NOTE: Though the term "impulse" is used in this manual, these algorithms are NOT convolution reverbs!

Room Scale
Is used to change the size of a room impulse.

Bass Xover
Controls the frequency of a filter that is applied to the room impulse. It works in conjunction with Bass Boost.

Bass Boost
Can boost or cut the signal below the Bass Xover value.

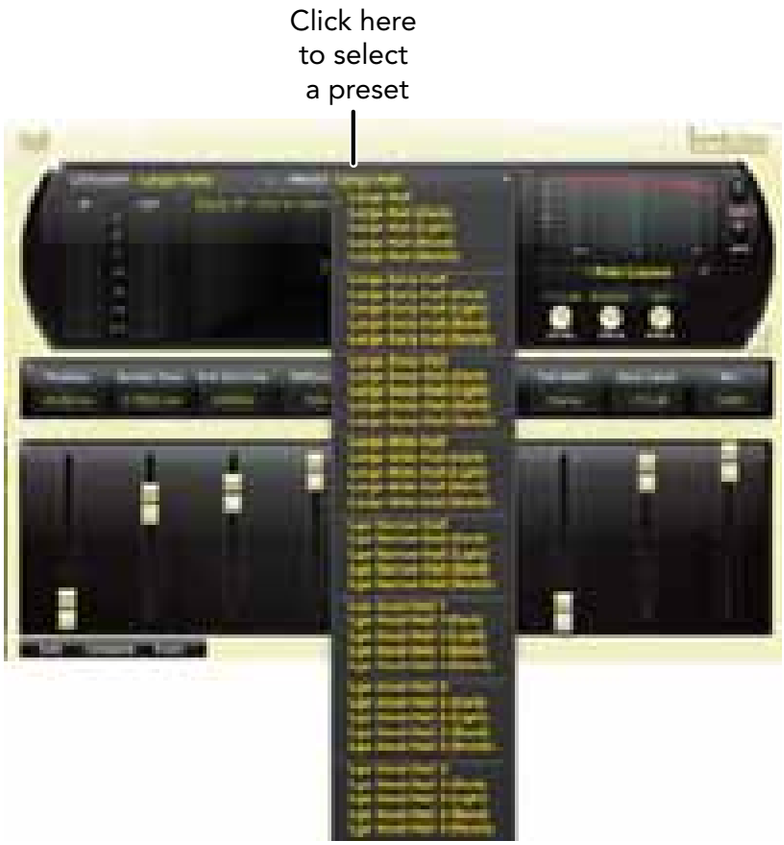
On the right side of the fader field, you will see a pair of drop-down menus and a button. The upper menu allows you to select a category of room impulses. The lower menu allows you to select an impulse from the active category. The bottom button is used to switch the impulse from Normal to Reversed, a rather significant effect.

Factory Presets

Each plug-in comes with a large complement of Factory Presets. These presets are grouped in categories (Medium Halls, Large Halls, etc) that make it easier for you to find the sound you need. Most presets also come with several variants that may save you a lot of editing time. For example, most of the reverb presets have dark, light, bandpass and notch variants. These variants share the same basic characteristics, but are EQ'ed differently.

Loading a Preset

The preset category is chosen from the drop-down menu in the upper-left part of the edit window. When you select a category, the preset menu is updated to show availability of the new presets within the selected category. An initial preset will be loaded from the category. It will be the first preset in the list unless you have previously visited the category. In that case, it will be the last preset you loaded from that category.



Storing and Managing Presets

When you load a Lexicon® plug-in, the overall appearance may be something like this:

You will notice that the Lexicon plug-in (with the gold border) is embedded within a larger window provided by the DAW. Appearance will differ from host to host (this example is from Logic®). Oftentimes, the DAW itself provides the ability to save presets. The presets saved by the DAW can only be used with the DAW that saved them. If you work with multiple DAWs, this could be a problem.

Portable Presets

Lexicon® has provided a solution by providing portable User Presets. When you store presets with the Lexicon "Store" button (shown below), your presets are available to all DAWs on your machine. With only a little effort, you can also share them with other machines, even when going between Mac and PC.

If you've made some edits you'd like to save as a plug-in, press the "Store" button on the lower left corner of the plug-in editor. You'll see a screen that looks like this:

1. Click Store

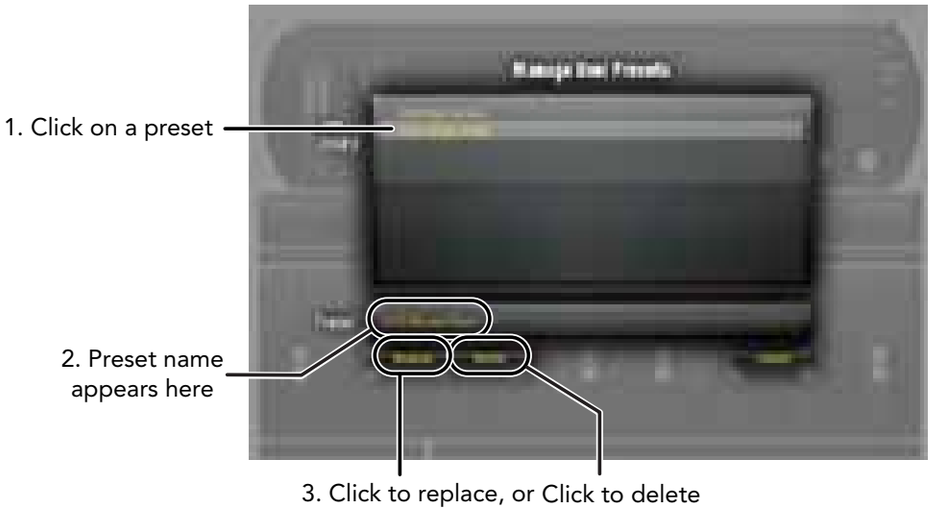
2. Click to edit name

3. Click Store to save

The upper field contains the list of User Presets that have already been created for this algorithm. Presets appear in alphabetical order. Click the name field to edit the name of the preset. Once done, click the Store button.

The lower field contains the name of the preset that was loaded before you began editing. You can easily click in this field and change the name to anything you like. Once you have the appropriate name, click the "Store" button and the preset will be saved. If you decide not to save it, then the "Cancel" button will return you to the editor.

If you select one of the existing presets in the library, you can replace or delete that preset.



Where are presets stored?

On the Mac, User Presets are stored in:
/Home/Library/Application Support/LexiconPro/Native/User Presets/AlgorithmName

Home is the name of the account you're logged in under. AlgorithmName is the name of the plug-in.

On Windows Vista/Seven, presets are stored in:
Users/"username"/appdata/"usertype"/Lexicon PCM Native/User Presets/AlgorithmName

On Windows XP it's:
Documents and Settings/"username"/Application Data/Lexicon PCM Native/User Presets/AlgorithmName

Getting the most out of your computer

These plug-ins are written to be efficient, and most computers will run more copies than you may need. But if you find yourself needing to squeeze a little bit more out of your CPU, you can take the following simple steps:

- Turn off the active realtime display. The active display window can represent your effect in real time in several ways. However, this display consumes some of your computer's processing power. If you click until you reach the static display, your computer will have more CPU power available.

- Close the plug-in editor. Unless you're actively controlling your mix in realtime, you don't need the editor after you've made the appropriate settings. All those knobs, faders and displays require processing power. You can always reopen the editor if you need to edit.

Reverb Tail Behavior

Each host treats plug-ins a little bit differently. You'll notice this especially when you stop, loop or relocate. You may notice that the tail stops immediately on one platform and continues to ring on another. You may also notice that the behavior changes when the plug-in is inserted on an audio track, an instrument track or an aux track. We'd like to behave as consistently as we can, but some things are beyond our control.

Tempo Mode Presets

In some of the plug-ins, you will find presets with '(T)' in the preset name. These are tempo mode presets. These presets have some characteristic (predelay time, delay time, etc) that responds to the tempo of the DAW. They are best used when the tempo of the project actually reflects the tempo of the musical material (MIDI sequences and such). In those cases, the tempo component of the preset will reinforce the tempo in some way.

The Lexicon® Heritage

Lexicon® occupies a unique position as a leading innovator in both professional and consumer audio industries. Since the release of the first digital delay, Lexicon has stood at the forefront of digital audio with a reputation as a manufacturer of exceptional professional audio and home theater products and an inventor of new technologies. Years of research, development, and learning allow us to continue expanding the boundaries of the listening experience.

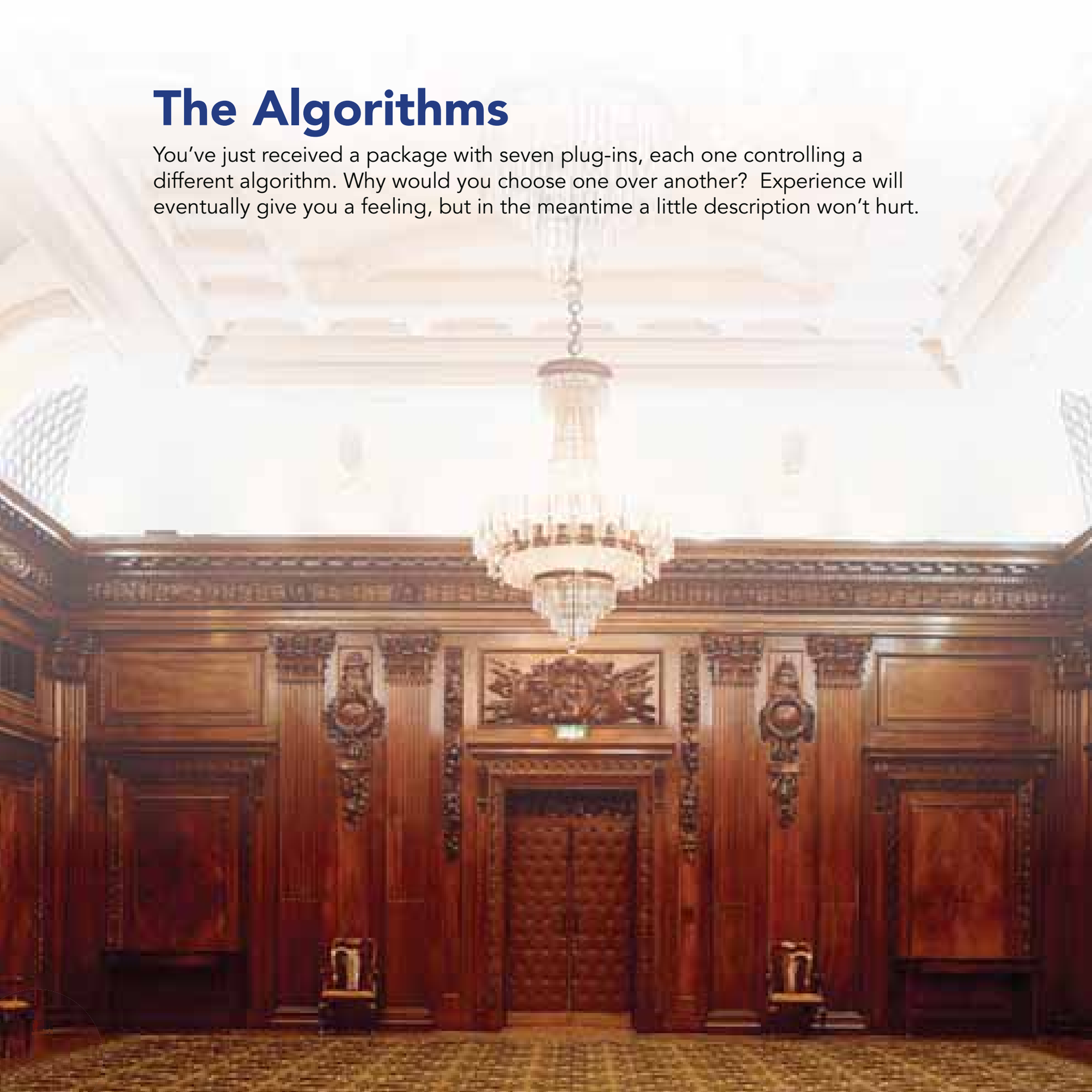
Our professional products are prominent in the creation of worldwide music, television and film productions. These products have won numerous awards, including an Emmy® and numerous TEC awards, including a TEC Hall of Fame award for the Lexicon Delta T-101, the world's first digital delay. Lexicon processors have been embraced as the standard in professional signal processing since the introduction of the 480L Digital Effects Processor, which has retained tremendous popularity for the past 22 years. It has since been replaced as the standard in professional signal processing by the 960L Multi-channel Digital Effects System and the PCM96 series Reverb/Effects Processors, which have themselves garnered an impressive following of producers, artists, and engineers.

Growing demand for proprietary Lexicon technologies has led to its appearance in numerous applications – with dramatic results. Our processing is relied upon to enhance the sound of prestigious live halls and venues.

Knowingly or unknowingly – you experience Lexicon products and technologies on a daily basis. Chances are that Lexicon processing was involved in the television program you watch at home, the film you see at the cinema, or the song you listen to on the radio. From the initial tracks to your listening room or automobile, Lexicon is part of the process that brings these recordings to life. Our commitment to the audio professional and content delivery ensures an unbroken chain between the artist and the audience. Now hear this…

The Algorithms

You've just received a package with seven plug-ins, each one controlling a different algorithm. Why would you choose one over another? Experience will eventually give you a feeling, but in the meantime a little description won't hurt.



Chamber

The echo chamber was the original high-end reverberator. It was a small-to medium-sized space with highly reflective and irregularly-shaped walls. Inside this space would be a microphone or two and a speaker or two. The input signal was sent to the speakers and picked up by the microphones. With careful design and setup, a highly satisfying reverb could be generated. There are still a number of first-rate echo chambers in operation around the world.

The principal characteristics of the chamber algorithm include a rapid high density and avoidance of noticeable wall slaps. This complex miniature-space effect resembles an echo chamber at its smaller settings and, at its larger ones, a small performance space with a more rapid build-up of reflection density than a hall. Reverberant tails are randomized.

Hall

The highly acclaimed Lexicon® Hall, Concert Hall, and Random Hall reverb algorithms have regularly been used by live sound and recording engineers because of their exceptional ability to reproduce the musical ambience of large, wide, panoramically wonderful spaces.

A hall is the principal venue for classical ensembles, but has proven to be useful for all types of music. A hall is comparatively large, with wall-to-wall distances that are typically several tens of meters. Smaller halls may be used for smaller ensembles. The characteristic sound of a hall includes very low initial reflection density, with little reflection energy before 60-100 milliseconds. Density buildup is more gradual, because of the larger distances between reflecting surfaces. Reverberation time is somewhat longer as well. Finally, in most halls lower frequencies reverberate longer than higher frequencies.

This new hall algorithm shares these basic characteristics with Random Hall and Concert Hall, but is smoother with a more even decay. Its initial density is also lower than Random Hall so it may be a better choice when the reverberation needs to be unobtrusive.

Random Hall

The highly acclaimed Lexicon® Hall, Concert Hall, and Random Hall reverb algorithms have regularly been used by live sound and recording engineers because of their exceptional ability to reproduce the musical ambience of large, wide, panoramically wonderful spaces.

Random Hall is similar to Hall, with gradual build-up, well suited to complex sounds like orchestral music. Its reverberators change over time in controlled random ways to avoid the buildup of tinny, grainy, metallic, or other colorations. The modulation can be noticeable and is often a desirable effect. This is one of the classic Lexicon® sounds.

The early reflections are user adjustable in amplitude and delay. Some skill is needed to set useful reflection patterns. The pattern can be expanded or contracted in time using the "Delay Master" control, and the overall level of the pattern can be set with the "Early Level" control.

A hall is the principal venue for classical ensembles, but has proven to be useful for all types of music. A hall is comparatively large, with wall-to-wall distances that are typically several tens of meters. Smaller halls may be used for smaller ensembles. The characteristic sound of a hall includes very low initial reflection density, with little reflection energy before 60-100 milliseconds. Density buildup is more gradual, because of the larger distances between reflecting surfaces. Reverberation time is somewhat longer as well. Finally, in most halls lower frequencies reverberate longer than higher frequencies.

One of its charms is a bit of irregularity in the decay. In some cases (very small rooms with precisely-pitched instruments), this modulation may not be the best choice. But in general, this is a time-tested reverberator.

Plate

A Plate reverb is a large, thin sheet of metal suspended upright under tension on springs. Transducers attached to the plate transmit a signal that makes the plate vibrate, causing sounds to appear to be occurring in a large, open space.

The Plate plug-in mimics the sound of plate reverberators, with high initial diffusion and a relatively bright, colored sound. For this reason, they are good choices for percussion. They are designed to be heard as part of the music, mellowing and thickening the initial sound itself. The Plate sound is what many people associate with the word reverb, and it is useful for all popular music.

Vintage Plate

This version of the Plate algorithm has the new equalization touches from the PCM96 plate, but also has characteristics of older plate implementations. It "speaks" differently, with different input diffusion and it also has an additional pair of echo voices.

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Concert Hall

The highly acclaimed Lexicon® Hall, Concert Hall, and Random Hall reverb algorithms have regularly been used by live sound and recording engineers because of their exceptional ability to reproduce the musical ambience of large, wide, panoramically wonderful spaces.

This reverb is an updated version of one of Lexicon's oldest algorithms. It was an essential part of many of the mixes of the late seventies and eighties. It is a less-dense reverb, allowing it to add lushness to a mix without stepping on the dry source material. It also has quite noticeable modulation, causing strong pitch effects at higher settings. The reverb tail has a life of its own, desirable in pop music, less so in jazz or classical applications.

A hall is the principal venue for classical ensembles, but has proven to be useful for all types of music. A hall is comparatively large, with wall-to-wall distances that are typically several tens of meters. Smaller halls may be used for smaller ensembles. The characteristic sound of a hall includes very low initial reflection density, with little reflection energy before 60-100 milliseconds. Density buildup is more gradual, because of the larger distances between reflecting surfaces. Reverberation time is somewhat longer as well. Finally, in most halls lower frequencies reverberate longer than higher frequencies.

Room

The Room algorithm was introduced in the PCM96 and is one of the most flexible algorithms we've ever invented. Reflection patterns can be easily selected, scaled and equalized all while simultaneously passing audio, and the room size can be instantly changed or reversed. It is similar to other Lexicon® reverbs in that it is used to create the illusion of space. However it differs in important ways. A room is comprised of a selectable early impulse taken from actual room measurements. There are several categories of responses, including small rooms, large rooms and odd rooms (impulses from unconventional sources). There are parameters to allow adjustments to this response, including scaling and reversing the response. In many cases this is sufficient.

There are three primary applications for the Room algorithm:

- **Post Production** - Lexicon® reverbs are renowned for smooth tails and controlled frequency response. However, in many cases the earlier components are more important. This may be true for some forms of popular music and is even more true for post-production. In these cases, specific types of spaces need to be invoked for convincing dialog and effects.

- **Reverbs** – This algorithm can also be used to generate large reverberant spaces that have a very different character from the more traditional Lexicon reverbs. These reverbs provide a high-quality counterpoint to the original "Lexicon Sound". There are a number of presets which take advantage of this.

- **Effects** – Because room patterns can be reversed, there are many additional sounds available, such as backwards reverb and gated sounds.

The Room algorithm provides the inherent realism of the actual space with all the precise control and manipulation that is only possible with a reverb synthesizer.

NOTE: Though the term "impulse" is used in this manual, these algorithms are NOT convolution reverbs!

The Parameters

The seven plug-in reverbs contain hundreds of presets, covering just about every possible need. But you can refine and customize any preset by adjusting its parameters. Parameters are the building blocks within each preset that determine how it sounds and behaves.

Each algorithm contains a set of parameters, and a variety of those parameters are combined to create a preset.

Below are all the algorithm parameters you can edit in the PCM Native Plug-ins. Note that sometimes parameter names are abbreviated when they appear onscreen, and they may have prefixes indicating what channel they control (for example, Level may appear as RightInLvl).

Bandwidth

This parameter lets you set the bandwidth of a multimode filter. Bandwidth is specified in octaves or fractions thereof. This parameter is available only when it has an effect (bandpass and notch modes). In all other cases, it is hidden.

Bass Boost (Room)

This parameter is closely tied to the Bass Crossover parameter. It controls the boost (or cut) of signal below that crossover.

Depending on the type of impulse response selected, the listener may become aware of an excess or insufficiency of low frequency output. This parameter may be used to correct the frequency response.

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BassRT

This parameter controls bass reverb time. It is closely associated with the Bass Crossover and Reverb Time parameters. BassRT is a multiple of Reverb Time that applies to signal below the frequency described by Bass Crossover. If BassRT is less than 1.0, then the low frequency part of the reverb tail will be shorter than the midrange part. If BassRT is greater than 1.0, then the low frequency part of the tail is longer.

Bass XOV (Bass Crossover)

The Bass XOV parameter is closely tied to the BassRT parameter. It represents the frequency below which BassRT has an effect. Note that there are two of these parameters in the Room algorithm. One is tied to BassRT, the other is found in the Pattern settings menu and affects the BassBoost.

Category (Room)

This parameter lets you select a specific category from which a room response may be chosen. Changes here have a direct effect on the Pattern Selector parameter.

Chorus

This parameter (in the Concert Hall algorithm) controls the rate at which the reverb chorus is run. Low values may cause a barely-noticeable undulation. Higher values will cause noticeable wobble in fixed-pitch instruments such as piano. It is closely tied to the Reverb Chorus Depth parameter.

Chorus Depth

This parameter (in the Concert Hall algorithm) controls the amount of randomization of the chorus tap. Higher values are generally preferred in order to minimize reverb coloration. Pitch effects may result and are closely tied to the Chorus parameter.

Definition

Definition controls the density of the Concert Hall reverb. Higher values result in lower density.

Delay Feedback Master

Controls all delays in the algorithm. Each voice has its nominal feedback gain adjusted by this percentage.

Delay Level Master

Controls all delays in the algorithm. Each voice has its nominal gain value adjusted by this percentage.

Delay Time Master

Controls all delays in the algorithm. Each voice has its nominal delay time adjusted by this percentage.

Diffusion

Input diffusion is the first part of processing for any signal entering a reverb or delay. It can be described as a smearing or softening of the signal and is typically used to lessen the impact of strong transients.

Bass Crossover (Bass XOver) (Room)

This parameter is closely tied to the Early Bass Boost parameter. It represents the frequency below which early bass boost has an effect.

Echo Parameters

- **Eko Delay**
  Controls the delay time of an echo that comes from the indicated input (L or R) and goes to the same output. It may also recirculate back into the same input. This can be toggled to Tempo Mode, in which case its delay time will be related to tempo.

- **Eko FBack**
  Controls the feedback of an echo that comes from the indicated input (L or R) and goes back into the same input. The sign of the gain coefficient by be optionally inverted.

- **Eko Gain**
  Controls the gain of an echo that comes from the indicated input (L or R) and goes to the same output. The sign of the gain coefficient by be optionally inverted.

- **Eko Time Mast**
  Controls all echo voices in the algorithm. Each echo delay has its nominal time adjusted by this percentage.

- **Eko Fbck Mast**
  Controls all echo voices in the algorithm. Each echo feedback has its nominal gain adjusted by this percentage.

Feedback

This controls the amount of echo output that is fed back to the input of the echo buffer. The actual feedback level is modified by the Master Echo Feedback parameter, if present. The master value is a percentage (0-100%) that is applied to the Echo Feedback level.

Frequency
This parameter lets you set the cutoff frequency of the multimode filter. The audible effect of this is determined by the Type parameter.

Front Early Level
This parameter acts as a master control for any early signals going to the output channels. Early signals include echoes, reflections, and the room reflection patterns.

Infinite
This parameter captures the reverb tail as an infinite loop. This may be useful in music, where a note or chord can be extended. It is also useful in post-production for creating ambience backgrounds.

Mix (Wet Dry Mix)
Mix is the proportion of wet (processed) signal to dry (unprocessed) signal. This must be used with care. If the plug-in is used as an insert on a single track, then it is probably appropriate to use the mix control directly in the plug-in. However, reverbs are often used on Aux tracks and may be sourced over internal busses in the DAW. In this case, it may be better to control the level on the Aux track rather than wet/dry mix. Not all DAWs feature delay compensation, so it's important not to have a dry signal in more than one path (cancellation being a possible outcome).

The mix parameter is "sticky". Like other parameters, it is saved and restored by the DAW. But after the plug-in is loaded, the mix value will stay where you put it--even if you load another preset in the same plug-in. This makes it easier to preview presets and choose the one you want, since the mix won't spring back to 100% every time you load another preset.

Output Level
Output Level controls the amount of post-mix signal emitted onto a channel of output.

Pattern (Room)
Selects a space available in the selected Category (see Category parameter).

Predelay

This is used to add a small amount of delay to a signal before it enters the reverberator. This is used to create a little separation between the wet and dry signals, in turn creating greater clarity in the mix. This delay can also be toggled into Tempo Mode, causing the delay value to be related to the tempo of the music.

Reflection Parameters

- **Rfl Delay**
  Controls the delay time of a reflection. This can be toggled to Tempo Mode, in which case its delay time will be related to tempo. Note that this parameter can have different routing configurations, indicated in its name. For example, if its name appears as "L Rfl Delay" or "L-L Rfl Delay," the signal comes in from the left and goes out the left. If the name appears as "L-R Rfl Delay," the signal comes in from the left and goes out the right.

- **Rfl Gain**
  Controls the gain of a reflection. The sign of the gain coefficient by be optionally inverted. Note that this parameter can have different routing configurations, indicated in its name. For example, if its name appears as "L Rfl Gain" or "L-L Rfl Gain," the signal comes in from the left and goes out the left. If the name appears as "L-R Rfl Gain," the signal comes in from the left and goes out the right.

- **Rfl Time Mast**
  This parameter is used to scale all reflection times at once.

Reverb Time

Reverb Time is the mid frequency reverb time (in the frequency above Bass XOV and below RT Hicut). As such, it is one of the primary controls affecting the length of the reverb tail. It may be considered to model the reflectivity of the walls in the reverberant space.
The parameter most closely associated with Reverb Time is Size. A large room size with a small Reverb Time can easily have the same decay time as a small room with a large Reverb Time. Even though the two solutions may result in the same reverb time, they will sound quite different from one another. The value that is actually displayed is an approximate reverb time based on both of those values. The actual time is also highly-dependent on other parameters as well as the audio material.

Reverb Wander

This parameter (also known as just Wander), along with Reverb Spin, is used to control the reverb randomizer. It controls the maximum size of randomizer steps, and is expressed in a time value (typically milliseconds).

RT Hi Cut

This parameter, also known as Hicut or RTHC is a low-pass filter in the recirculating part of the reverb. It represents a frequency above which the tail dies away more quickly. In some ways, it represents the opposite end of the scale from BassRT, and may be considered an analog of air absorption. It is also closely tied to the RTHC Damping parameter.

RtHC Damping

The damping parameter is closely tied to the RTHC parameter. It controls the strength of the hi frequency absorption and has three values: Light, Normal, and Heavy. The normal value gives filter response identical to previous Lexicon® reverbs. The other values should be self-descriptive.

Scale (Room)

This parameter is used to modify the overall time of the selected early impulse. It is a multiplier that goes from 0.5x to 2.0x. The actual time of the responses varies from response to response. The Early Scale parameter simply scales that amount of time. It's fair to describe this as "rubber-banding" the impulse response.

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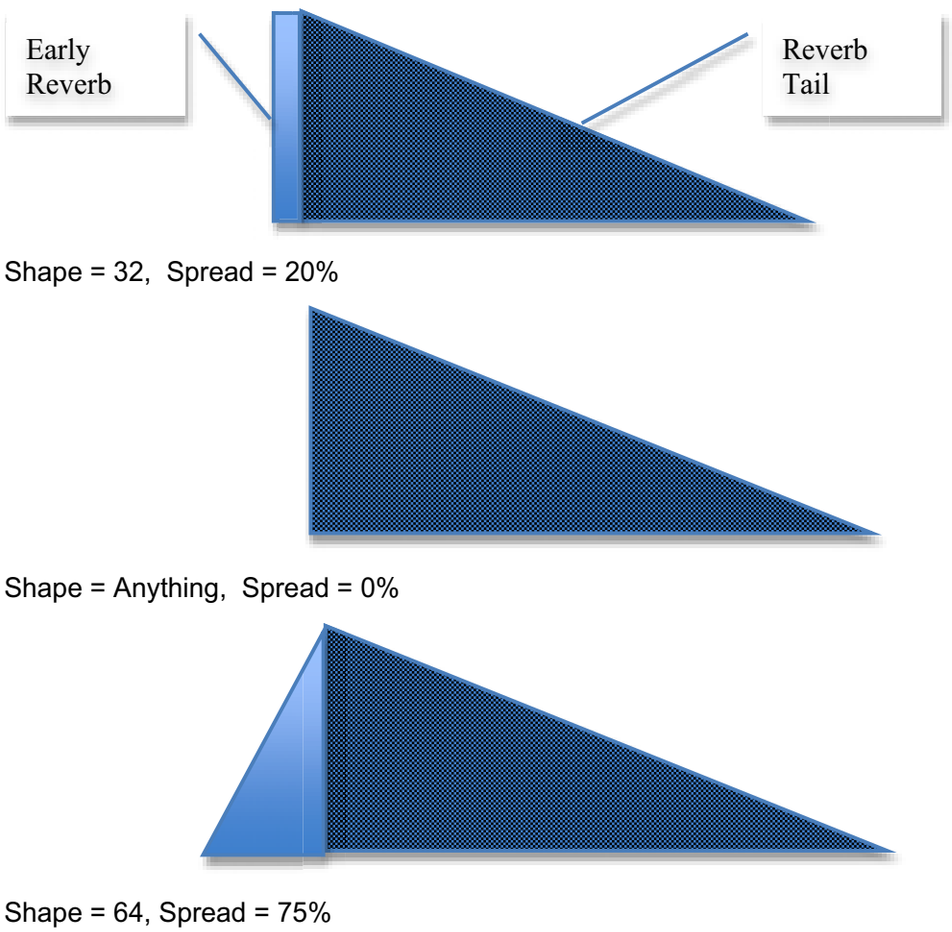
Shape and Spread

The Shape parameter is closely tied to the Spread parameter. Shape controls how energy is injected into the reverberator. A low value means that sound enters the reverb at the beginning of the spread window. A high value means that most sound moves into the reverb at the end of the spread window. A value somewhere in the middle means that sound enters the reverb evenly across the spread window.

Tip: Shape will not have this effect if spread is at its minimum value. But even then, it can make a difference. In this case, it affects reverb timbre and density. Higher values of shape may be both darker and denser, although the effect is subtle.

The Spread parameter is closely tied to the Shape parameter. Spread is a window of time during which a signal is injected into the reverb. Shape controls just how the signal is injected during this window. Together, the two parameters create an envelope for the early portion of the reverb, as shown in this illustration:



Shelf

This parameter effects all filter types. The shelf specifies the audio level below which the filter has no effect. For example, let's imagine we're using a one-pole lowpass with shelf set at -12 dB. The lowpass filter affects only the strongest 12 dB of the signal. Below the shelf, the signal is unfiltered. This is easy to understand by viewing the EQ window.

Size (Reverb Size)

Size corresponds roughly to the length of the longest wall of a rectangular room. In a more general sense, it corresponds to the overall dimension of some mythical space. This space has a geometry that causes sound to bounce around. When the room size is small, the "walls" of this space are closer together and the resultant reflection density increases. When the room size is large, that density decreases. The most natural reverbs use room sizes that vary from about 24 meters to 45 meters or so, but there are many useful reverbs that are outside of this range. There is a relationship between this parameter and the Reverb Time parameter. Please see the Reverb Time parameter for clarification.

Spin and Wander

These two parameter control randomization effects within the reverb. Randomization is used to control the spectral purity of the reverb. It can also—at some settings—create a more noticeable modulation effect. Artificial reverbs sometimes create audible artifacts with some types of signals. For example, a voiceover in a very small room might sound a little metallic. Randomization works to minimize these effects and remove the sense of ringiness.

Spin controls the rate of randomization. In most cases, values between 1 - 2 Hz are appropriate. Wander controls the amount by which various delay values may be modulated. Typically, these values should be smaller for plates and chambers or for very small rooms. But it's always best to listen with the material you're using.

Tail Width

This parameter is present in all stereo reverbs and rooms. The reverb tail (all components of the reverb except for early reflections, etc.) is passed through a simple 2x2 matrix. This provides an encoding of the tail that dramatically changes its spatial characteristics. The tail can be changed to feel narrower (even down to mono) or wider than normal stereo. There are values for the parameter to encode the tail in such a way that it decodes into surround channels.

The parameter operates in the range of 0 to 360 degrees, with an incremental change of 1 degree. The matrix uses Sine/Cosine rules so that power distribution remains constant. This may be acceptable—even highly desirable—but the mix engineer must fully understand the process and its implications. It is possible to compromise the mono compatibility of a mix by using this parameter carelessly.

This is a powerful tool for audio whose release format is two-channel. It is not useful in any other format. Formats include compact disc and radio/television broadcast. Some of these effects are clear and noticeable without any sort of decoder. Many are even more dramatic when a decoder is in place, such as in a home theater.
It is strongly recommended that you monitor in all possible formats (stereo, mono, matrix surround) when using this parameter.

Tap Slope

This is an unusual parameter which affects the relative gain of impulses coming from the reverb. While this parameter does not (as it might appear) make the reverb sound 'bumpy', it may affect timbre or the overall sense of spaciousness.

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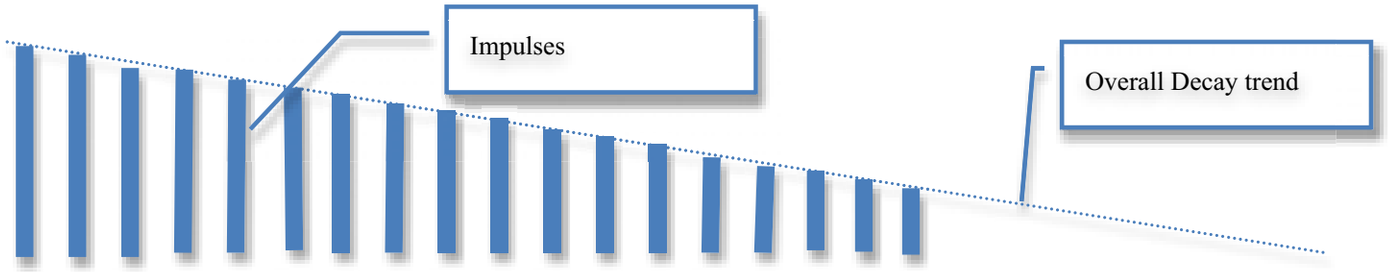
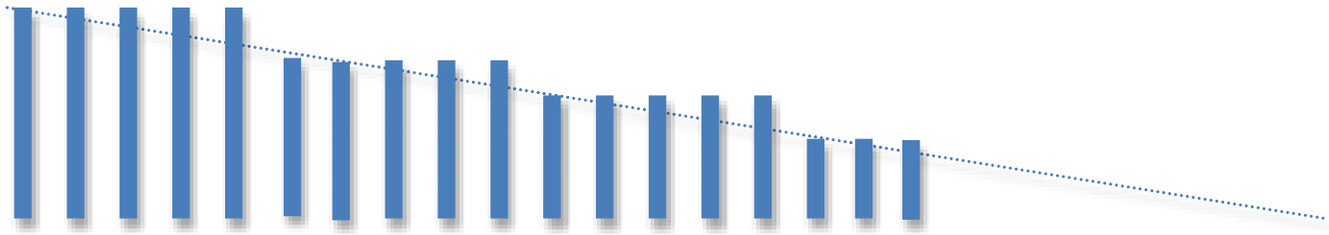
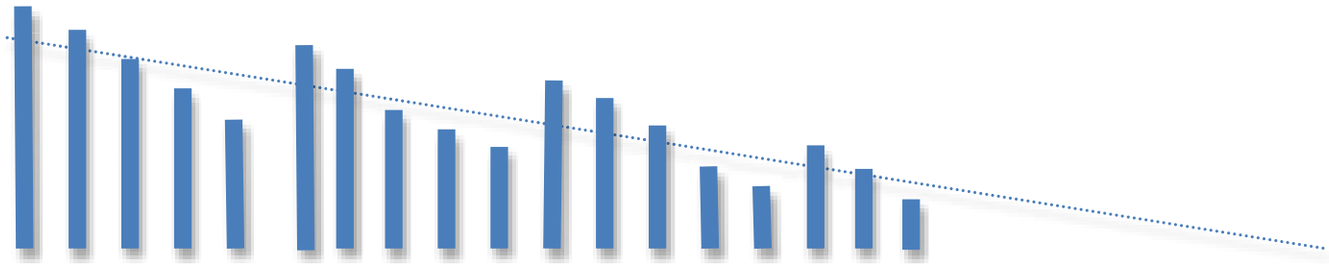


Tap Slope = 0



Positive Tap Slope



Negative Tap Slope

It can also be used to create inverse and gated effects. In this case, Reverb Time should be at 0. Spread should be fairly high. Shape should be fairly high. Size should be fairly high. TapSlope should be in the range of (5 to 10). Play with spread, shape and size to vary the effect.

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