Exponential AUDIO

PhoenixVerb Surround/ R2 Surround

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Welcome to *PhoenixVerb Surround* and *R2 Surround*, two new plugins from Exponential Audio. *PhoenixVerb Surround* is an easy-to-use multichannel reverb with the flexibility to fit into many sorts of mixes--music, foley, FX, dialog--and it comes with a large assortment of built-in presets to get you right to work. *R2 Surround* has all of the features of *PhoenixVerb Surround* as well as chorus and gate features. They both feature a novel keyword architecture that makes it easy to find just what you're looking for. And of course, they both sound great. *PhoenixVerb Surround* is extremely pure and natural, while *R2 Surround* has the more active attributes that make it popular with users of 'vintage' reverbs. They're both based on the popular stereo versions, but have been extended in many ways.

PhoenixVerb Surround and *R2 Surround* can work in any of the following formats (depending on workstation):

•Mono,
•Stereo,
•LCR and LCR.1
•LRS and LRS.1
•LCRS and LCRS.1
•Quad and Quad.1
•5.0 and 5.1
•6.0 and 6.1 (Cinema, w/center and center-surround)
•6.0 and 6.1 (Music, no center and 2 additional sides
•7.0 and 7.1
•SDDS and SDDS.1
•Auro3D ® High (6-channel with top center channel passed as LFE)

The "dot 1" (.1) form indicates added LFE channel. As indicated above, not every form is available in every workstation.

In addition to the 'pure' forms of each format (7-in->7-out, for example) there are also mono-in and stereo-in versions of every format. In those formats with a center channel, there is also the ability to specify hard or phantom center with the "Surround Format" popup menu in the user interface.

With Exponential Audio's new *3D Link* technology, surround formats can be extended to include height channels. Any two reverbs—one on a track going to low channels and another going to high channels—can be linked to share both audio and control. Reverbs with track counts up to 14.1 can be supported.

Most operations of the two plugins are identical (even though the sound is different). The *PhoenixVerb Surround* graphics will be used for illustration, except for those features that are unique to *R2 Surround*.

Note: illustrations in this guide may differ slightly from the product itself.

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1. System Requirements

1.1. Mac

These plugins require an Intel-based Mac running OSX 10.6 (Snow Leopard) or later. It does not run and will not be supported on older version of OSX. It has not been tested on non-Apple hardware (hackintoshes). While some users may have success on other system variants, those systems will not be tested or supported by Exponential Audio.

1.2. Windows

These plugins may be operated on Windows 7 or 8 as 32 or 64-bit plugins. Windows Vista is not recommended for audio applications and will not be tested or supported by Exponential Audio.

2. General Requirements

It is recommended that you use a multicore CPU with a clock speed in excess of 2 GHz. For processing of higher sample rates (especially 192K), something closer to 3GHz is a good idea.
It is recommended that you have *at least* 2GB of RAM in your system. The more RAM, the better.
It is recommended that your monitor have greater than 1024 x 768 pixels.

2.1. iLok

A second-generation iLok (iLok2) is required to operate these plugin. iLok is a product of Pace and may be purchased directly from www.ilok.com or from any music retailer.

3. Supported Plugin Formats

3.1. Mac

- Audio Units 32/64-bit
- VST 32/64-bit
- VST3 32/64-bit
- RTAS 32-bit under ProTools 9 or ProTools10. ProTools 7 is not supported and ProTools8 has not been tested.
- AAX 32/64-bit

3.2. Windows

- VST 32/64-bit
- VST3 32/64-bit
- RTAS - 32-bit
- AAX - 32/64-bit

The core features of the plugins are available in every format. Some extended features (such as ProTools GUI automation) may only be available in certain formats.

4. Installation and Removal

4.1. Install the iLok License Manager

The iLok License Manager may be downloaded from www.ilok.com. Even if you already use the iLok, it's always a good idea to verify that your driver is up-to-date. Some systems may have an older iLok driver. This must be updated to the License Manager in order to use these plugins. Please be sure to install or update your driver before installing *PhoenixVerb Surround*. But before doing so, please check any *read me* files in your installer package.

4.2. Make sure you have your license authorization

When you purchased these plugins (or decided to test a demo version) you will have received a license key (a long sequence of digits). That key must be entered into the License Manager and dragged to the appropriate iLok. Alternatively you may have received a message that your authorization is already waiting for you at iLok. In that case, simply drag the license to the appropriate iLok using the License Manager.

4.3. Run the Installer

You'll need administrator privileges to install, but no reboot is needed.
Windows users will need to set their DAWs to scan the plugin folders so the plugins can be loaded. Those locations are shown in the Where things go on Windows section.

4.4. To Uninstall

On Windows, *PhoenixVerb Surround* and *R2 Surround* can be removed just like any other program. Launch the control panel for uninstalling programs, find the plugin, and remove it. On Mac, find the uninstaller script on the original installer disk image and run it. Your user presets will not be removed (just in case). See the following section to learn how to find those files if you wish to remove them.

4.5. Where things go on the Mac

Apple provides a very formalized set of locations for plugins and support files. You can find factory presets and other support files in:

/Library/Application Support/ExponentialAudio

The plugins go in specific areas for each plugin format:

- AU are in /Library/Audio/Plug-Ins/Components
- VST are in /Library/Audio/Plugin-Ins/VST
- RTAS are in /Library/Application Support/Digidesign/Plug-Ins/ExponentialAudio
- AAX are in /Library/Application Support/Avid/Audio/Plug-Ins/ExponentialAudio

Your user presets and favorites are stored in ~/Library/Application Support/ExponentialAudio/

4.6. Where things go on Windows

On 32-bit systems:

Shared components of the plugins are stored in C:\ProgramData\ExponentialAudio\
The DLL of PhoenixVerb is stored by default in C:\ProgramData\Vstplugins\
This may be changed during installation if the user wishes.

On 64-bit Systems:

Shared components of 32-bit plugins are stored in C:\ProgramData (x86)\ExponentialAudio\
Shared components of 64-bit plugins are stored in C:\ProgramData\ExponentialAudio\
The 32-bit VST DLLs of the plugins are stored by default in C:\ProgramData (X86)\Vstplugins\
This may be changed during installation if the user wishes.
The 64-bit Vst DLLs of the plugins are stored by default in C:\ProgramData\Vstplugins\
This may be changed during installation if the user wishes.

Your user presets and favorites are stored in YourName\AppData\Roaming\ExponentialAudio\

4.7. Other Installation Notes

There is also a logfile which may be helpful in diagnosing problems. Its location is displayed on the info window.
Do not install or uninstall by hand. Use the provided installers.

5. Walkthrough

The following walkthrough will show you how to begin using your plugins.

5.1. The Plugin Windows

Note: The plugin window will be embedded in a window provided by your workstation program. That is not shown here.

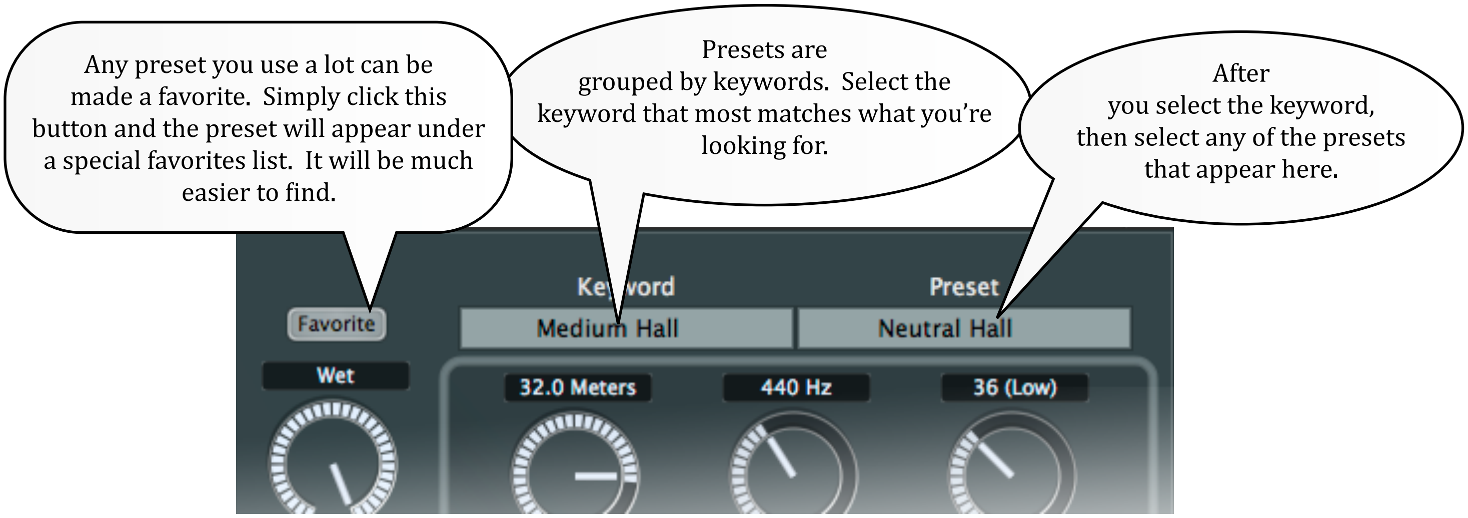
PhoenixVerb Surround

R2 Surround

5.2. Loading Presets

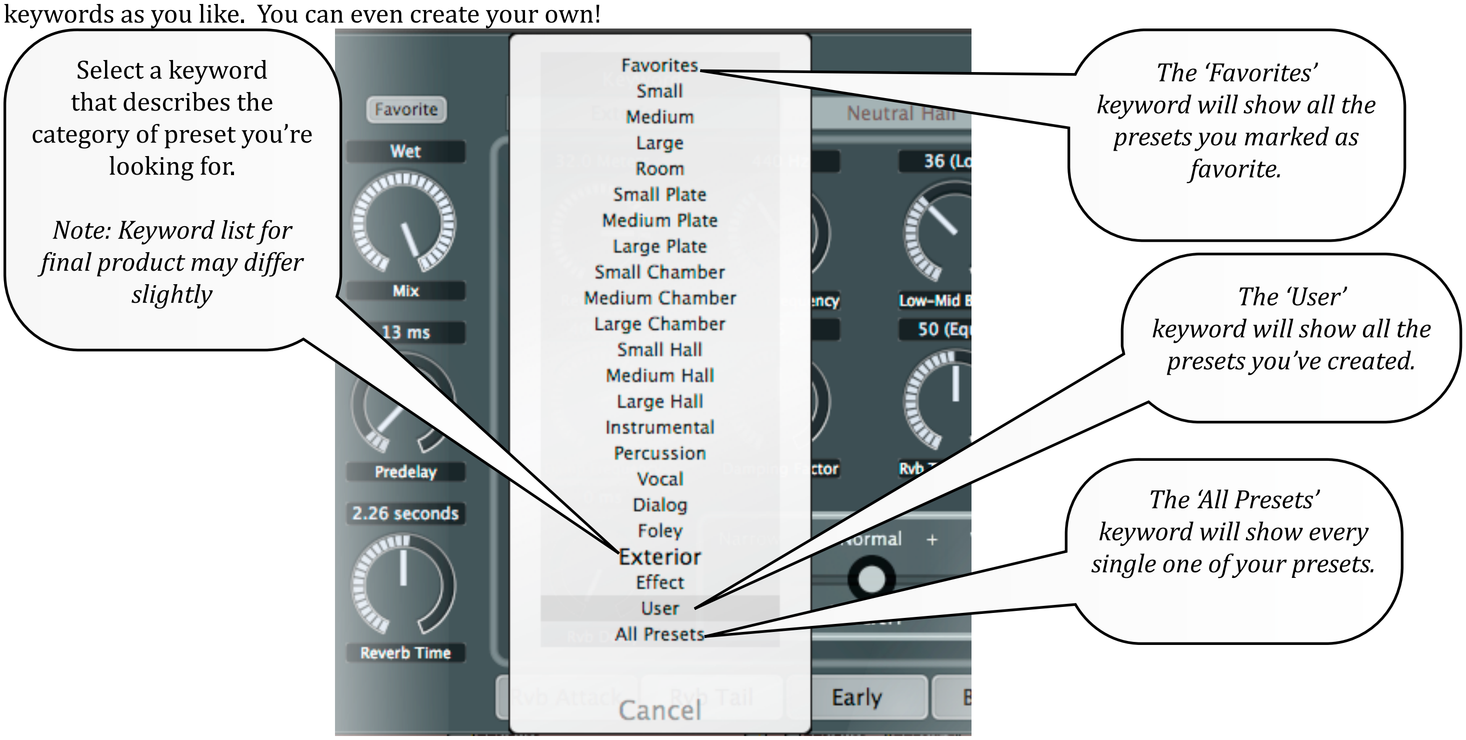
The first thing you'll want to do is to listen to the many presets that come with *PhoenixVerb Surround* and *R2 Surround*. On the upper right corner of the plugin window, you'll see a pair of combo boxes (popup menus). The left box shows Keywords, which are something like the old idea of banks (but more powerful). The right box shows Presets (you might know them as patches) which are the actual sounds you can load.

Tip: In just a few more pages there are some tricks about changing keywords and presets rapidly.



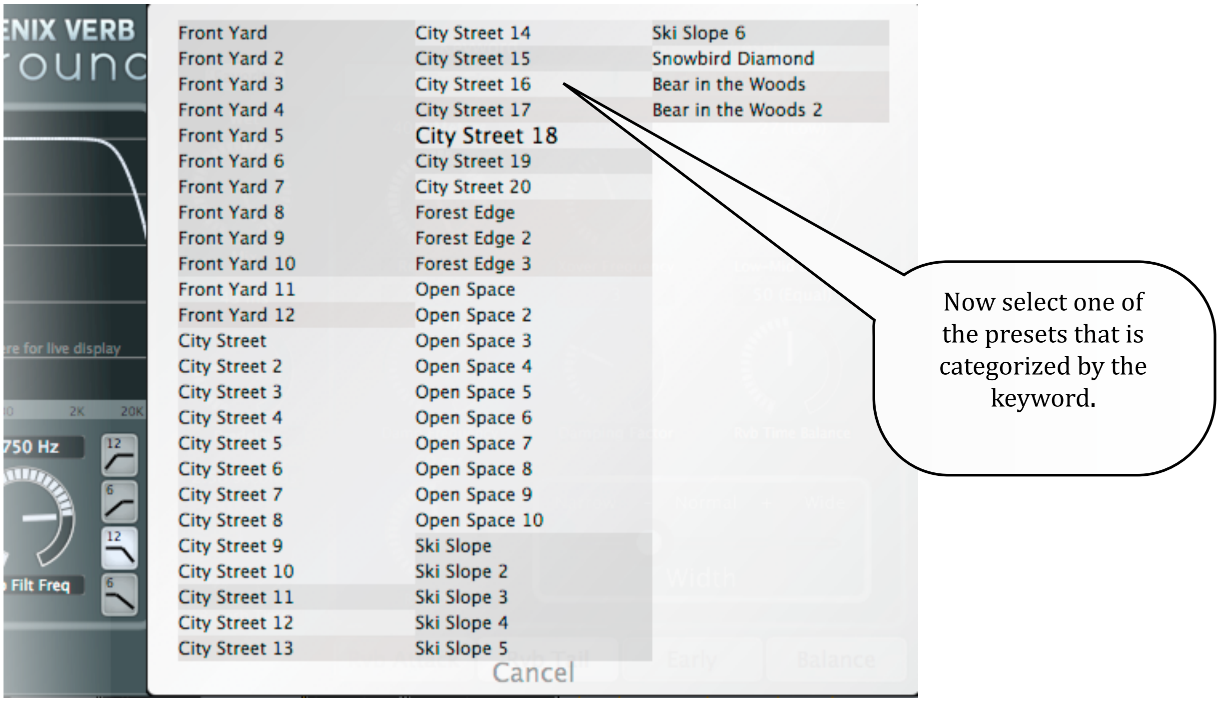
5.2.1. Keywords

Most mix engineers search for a preset that fits a specific need. Perhaps a small room for dialog. Perhaps something wide exterior shot. Search for a keyword that describes the application. Remember that a preset might appear under several different keywords if the preset might be used in that application. When you create your own preset, you can use as many keywords as you like. You can even create your own!



5.2.2. Presets

Once a keyword has been selected, several presets will appear when you click the preset popup. Audition them until you find the right one. Don't be surprised to see the same preset showing up under several keywords--most presets have more than one application. This is the power of keyword organization. You can add keywords to any preset and save it for later use. If you don't like the choice of keywords, we'll show you how to create your own.



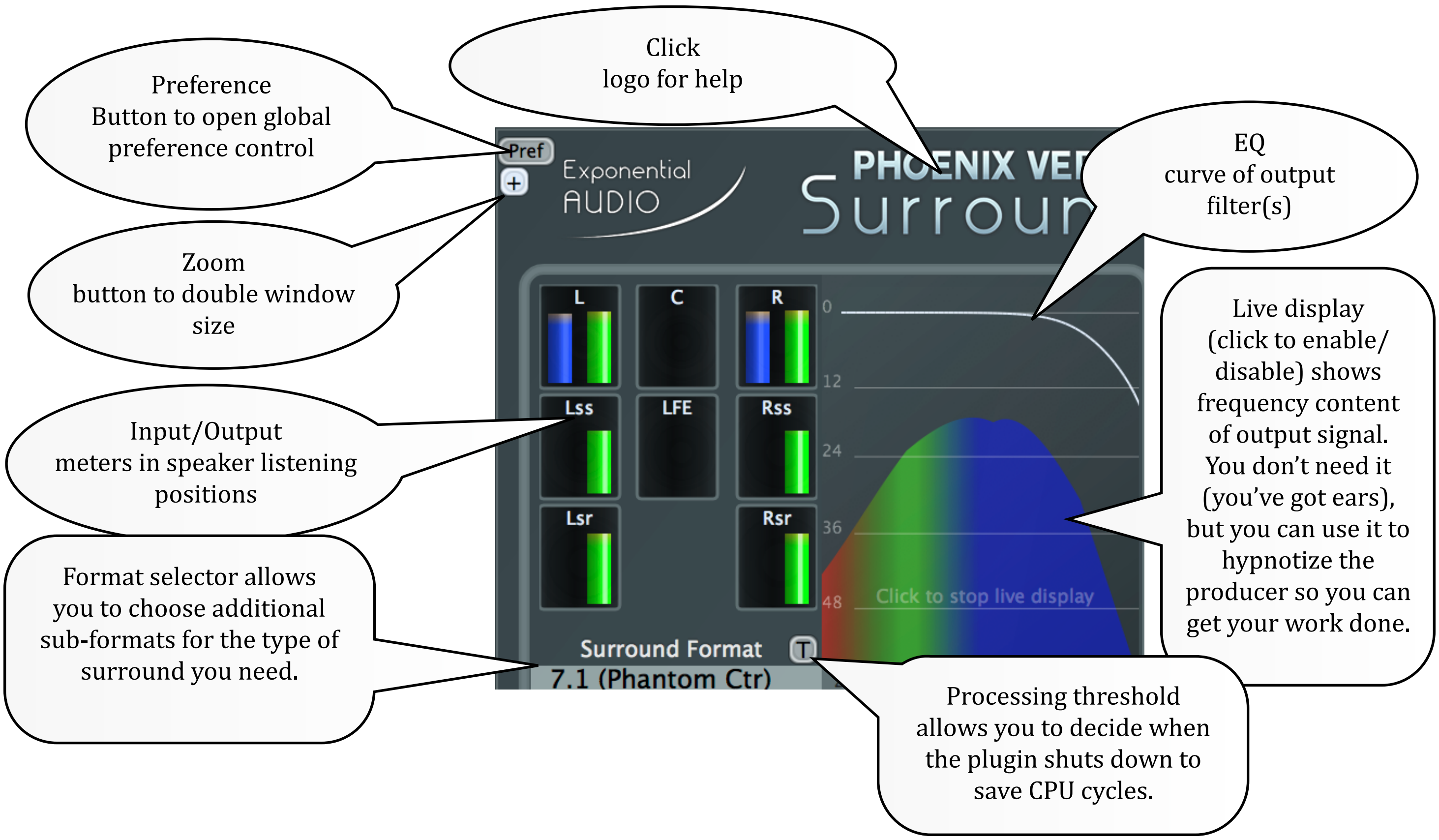
5.2.3. Tip for quick auditioning:

You actually don't even have to click on the Keyword or Preset menu to make changes. On most workstations, a simple up-arrow or down-arrow will advance the keyword. A left-arrow or right-arrow will select the next or previous preset. If that doesn't work on your DAW, try a combination like Command-arrow or Alt-arrow. This will let you focus on listening instead of operating the GUI.

You should also notice that presets are grouped. In the example on the previous page you'll see there are several variants of "City Street" for example. If you'd like an even quicker audition to see of the preset group is appropriate, use the "Page Up" and "Page Down" buttons on your keyboard. That will move from group to group instead of preset to preset.

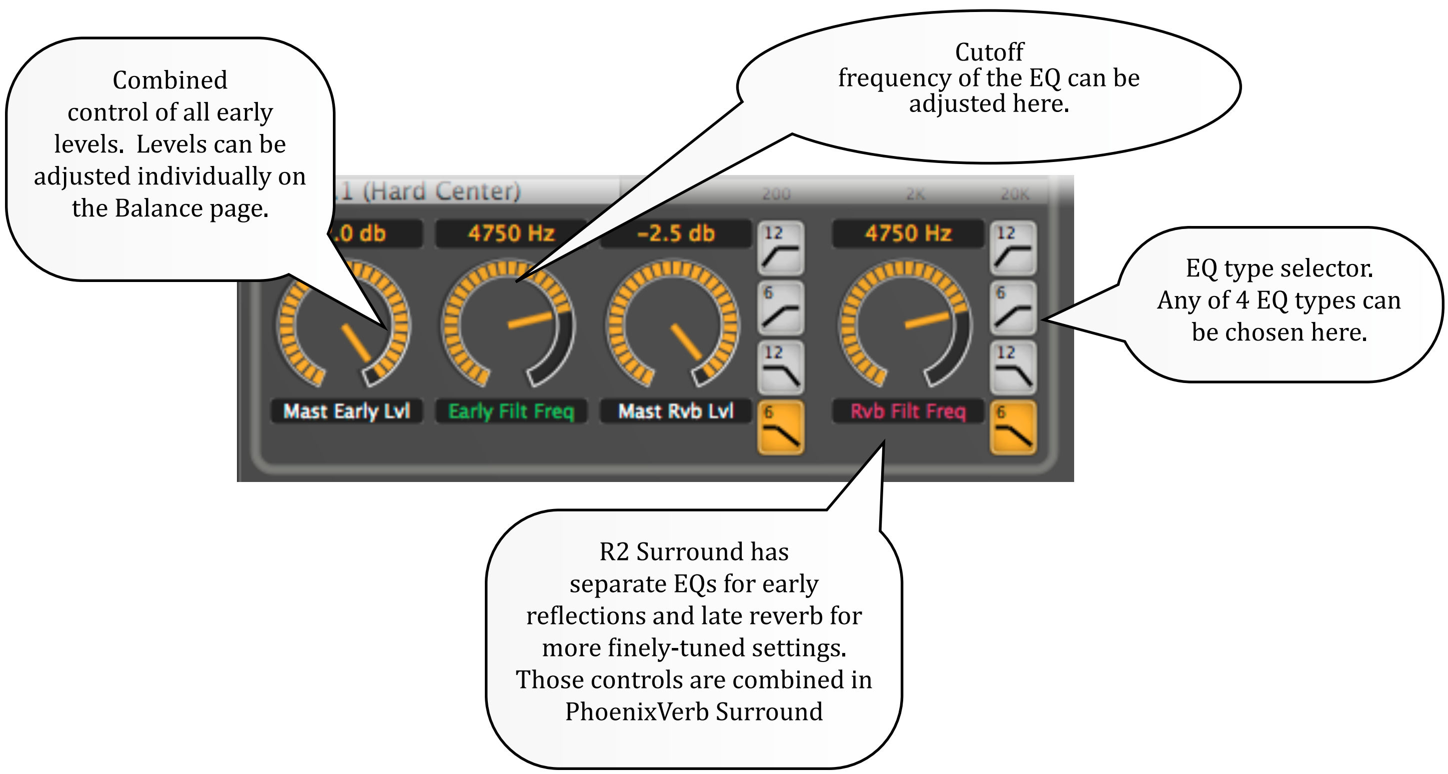
5.3. The Meter Area

The live meter area at the upper left portion of the plugin window provides feedback on the signals entering and leaving the plugin. It also provides access to many features of the plugin



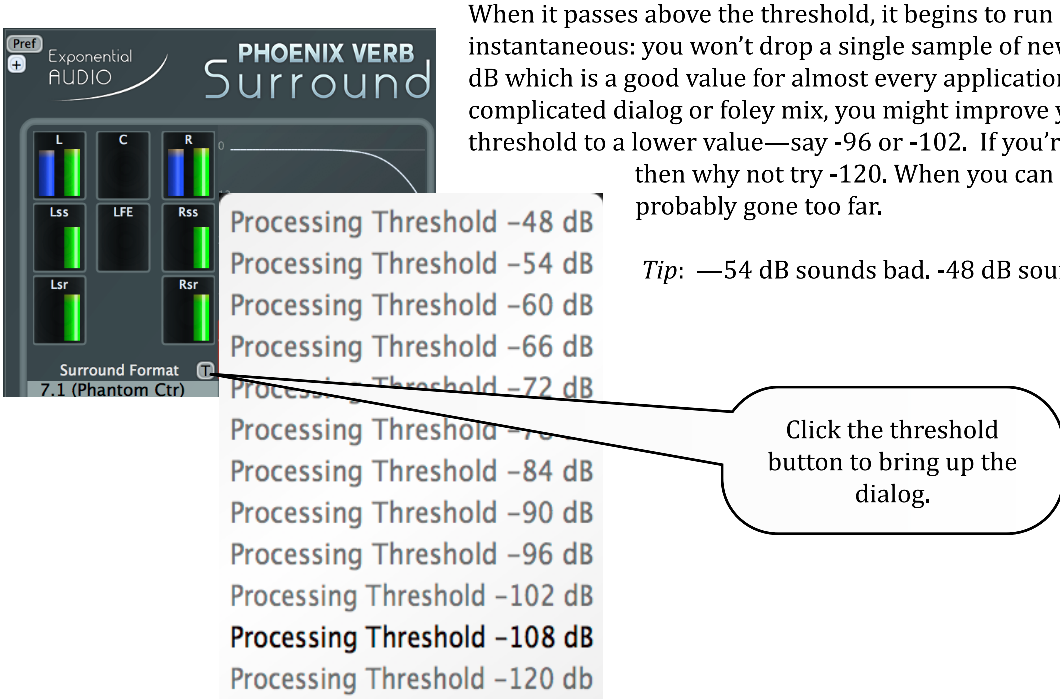
5.4. Output Controls

The output controls at the lower-left area allow the early reflections and late reverb levels to be balanced and equalized.



5.5. Processor Threshold

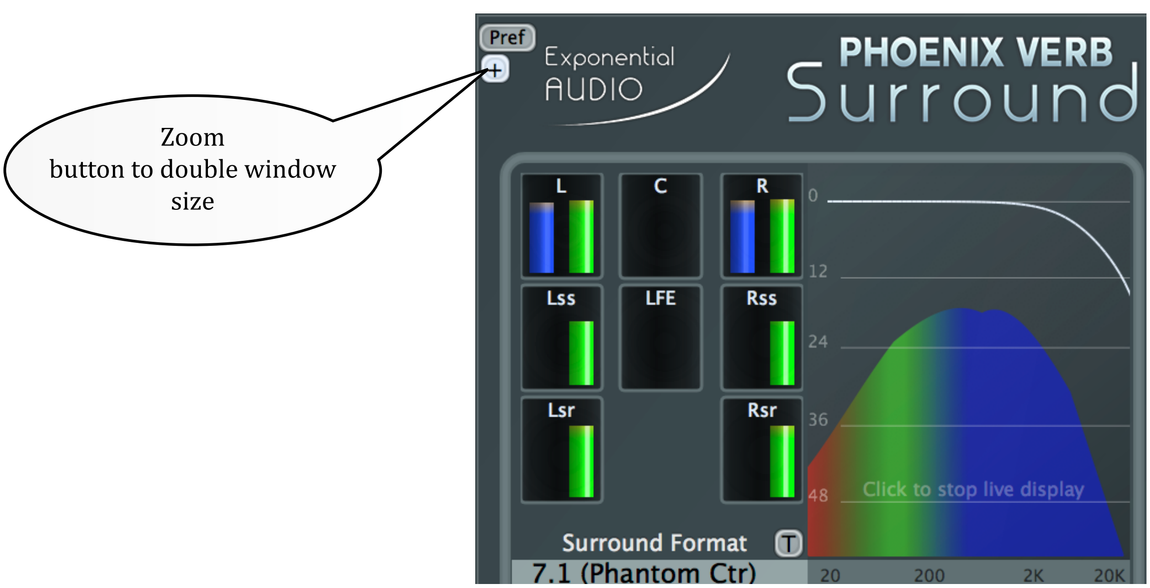
Conserving processor power is always important, especially in cases where the mix is made of small segments of audio. There's no reason for a reverb to run if there's not something in it. The threshold button allows you to determine just where the reverb stops and starts processing. When the signal falls below the threshold, reverb stops. When it passes above the threshold, it begins to run again. Response is instantaneous: you won't drop a single sample of new input. The default is set at -108 dB which is a good value for almost every application. If you're working on a complicated dialog or foley mix, you might improve your performance if you set the threshold to a lower value—say -96 or -102. If you're on a high-end classical mix, then why not try -120. When you can hear the reverb shut off you've probably gone too far.



Tip: —54 dB sounds bad. -48 dB sounds *really* bad.

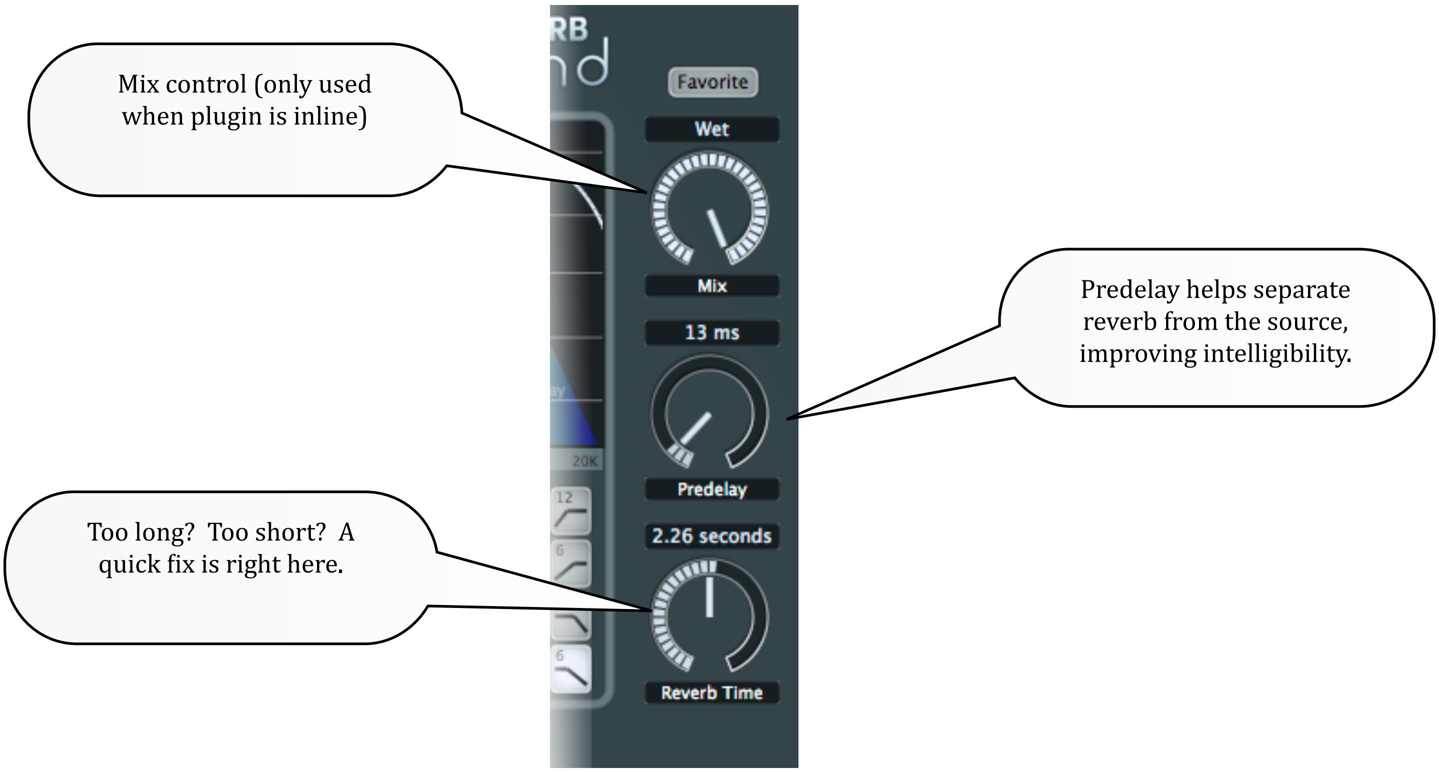
5.6. Scaleable Display

PhoenixVerb Surround and *R2 Surround* will analyze your display size at startup and may give you options for its window size. These options are shown as a Zoom button in the upper left corner of the plugin window. If your display is large enough to support a double-size image, the **+** button will appear. Pressing this button will double the size of the plugin window (the button will also change to "-"). If you save plugins with your project, window size and position will be recalled. When loading a new plugin instantiation, *PhoenixVerb Surround* will default to the smallest window size. If your monitor resolution is too low, no zoom button will appear.



5.7. Basic Controls

In addition to the output EQ, mix engineers like to have three more controls front and center. Those are controls for Wet/Dry Mix, Predelay and Reverb time. In *PhoenixVerb Surround* and *R2 Surround*, those controls are always available.



5.8. Edit Subpages

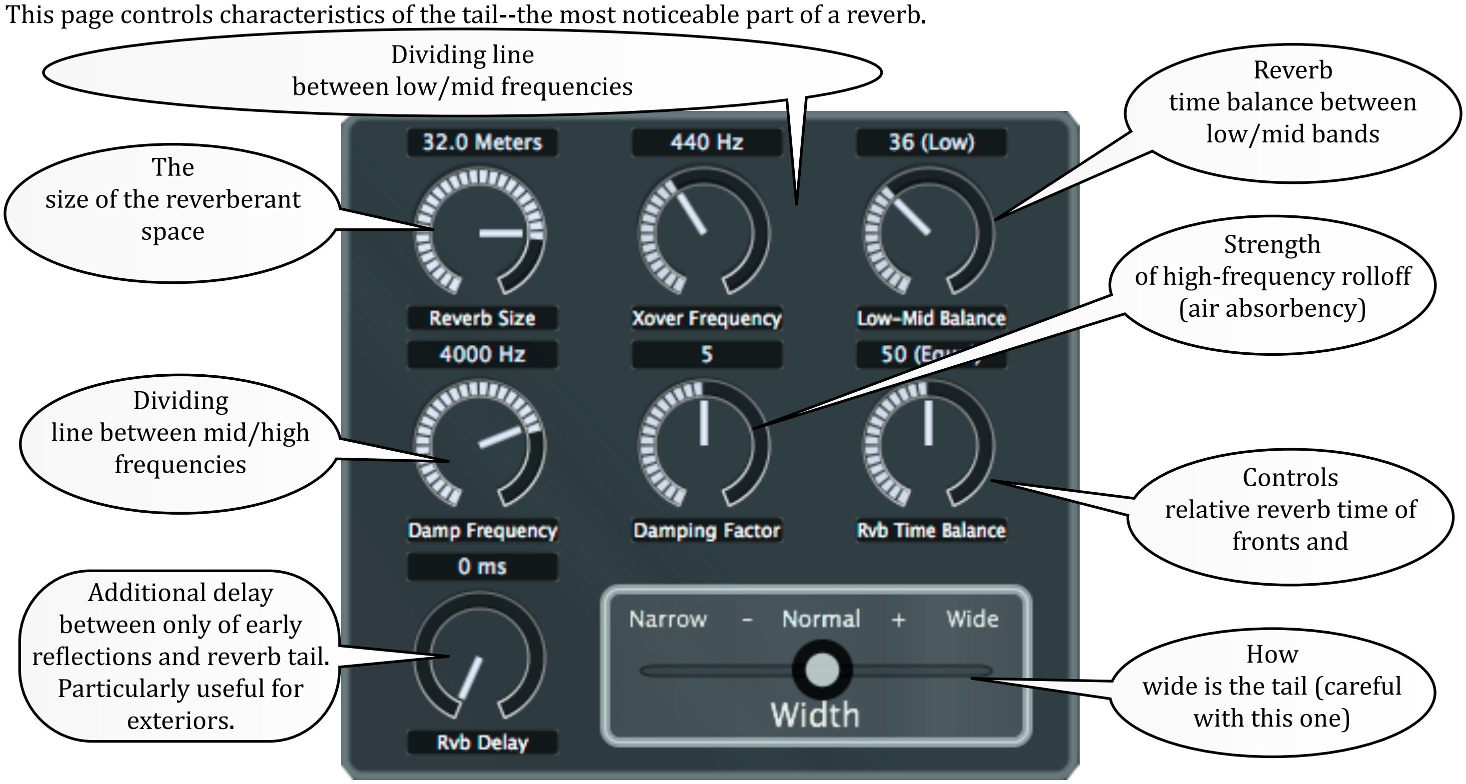
While *PhoenixVerb Surround* and *R2 Surround* are designed for ease of use, a curious mix engineer may wish to get deeper into the plugin. The buttons at the lower right-hand portion of the window allow any of the remaining groups of parameters to be accessed.

5.8.1. Reverb Attack

This page controls the way that audio energy enters the reverb tail.

5.8.2. Reverb Tail

This page controls characteristics of the tail--the most noticeable part of a reverb.

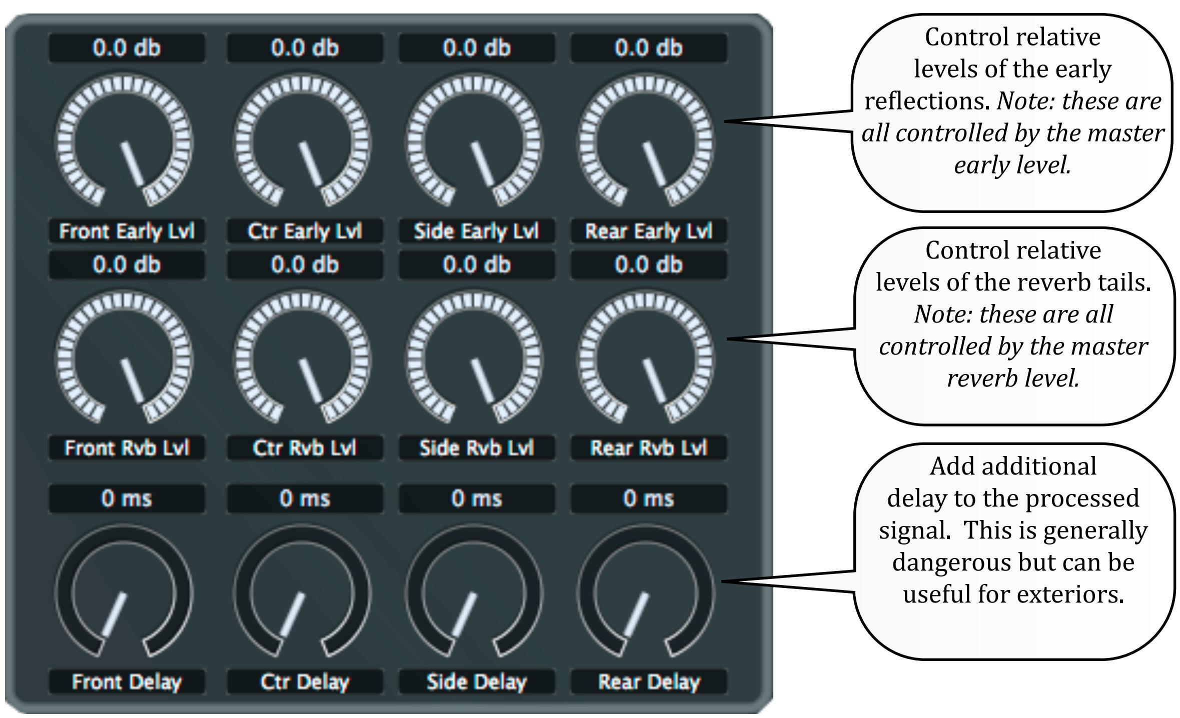


5.8.3. Early Reflection

Early reflections affect our sense of audio placement--distance and environment.

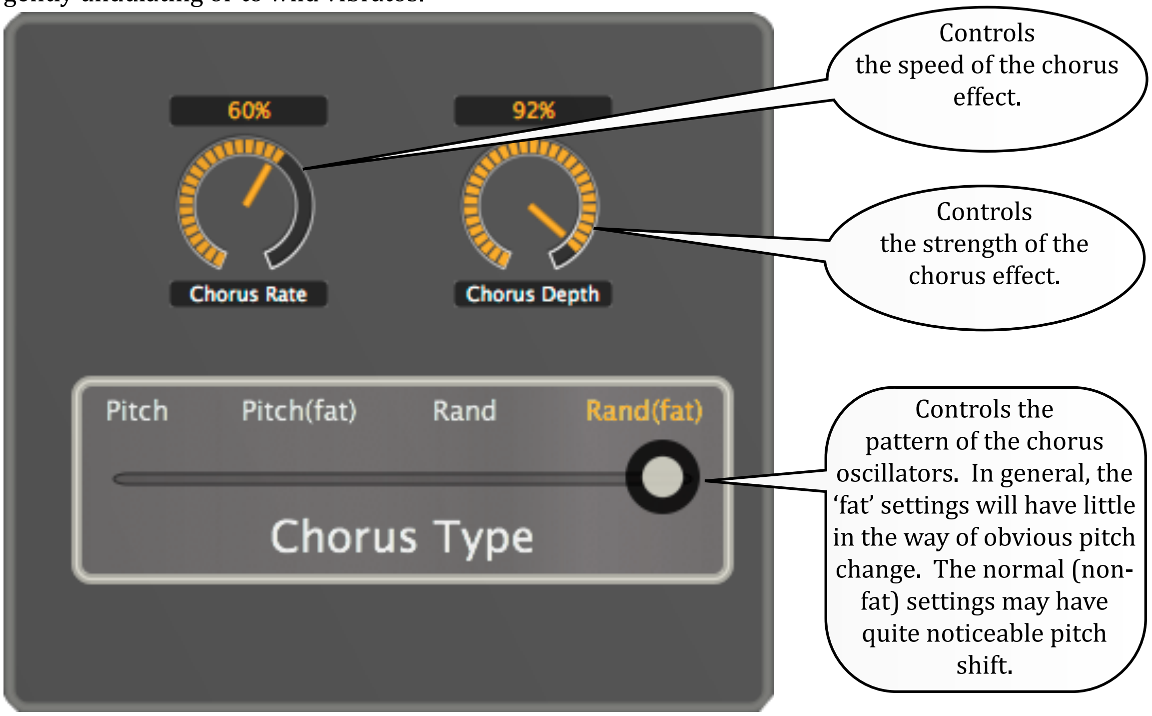
5.8.4. Balance

The Balance parameters are available for ultimate tweaking. Most users won't need to spend much time on this page.



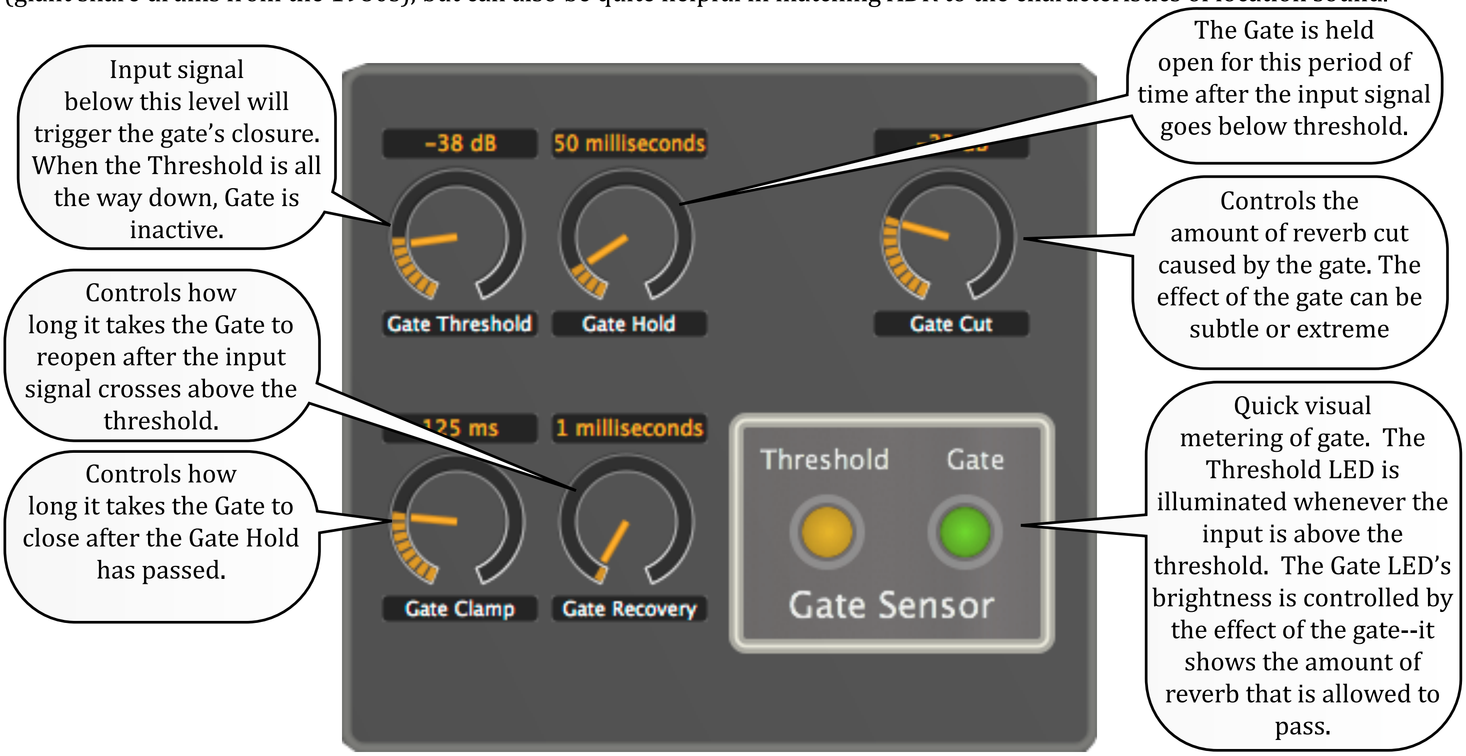
5.8.5. Chorus (R2 Surround only)

These control features of the chorus. This adds character to the overall reverb. Chorus effects can go from unnoticeable to gently undulating or to wild vibratos.



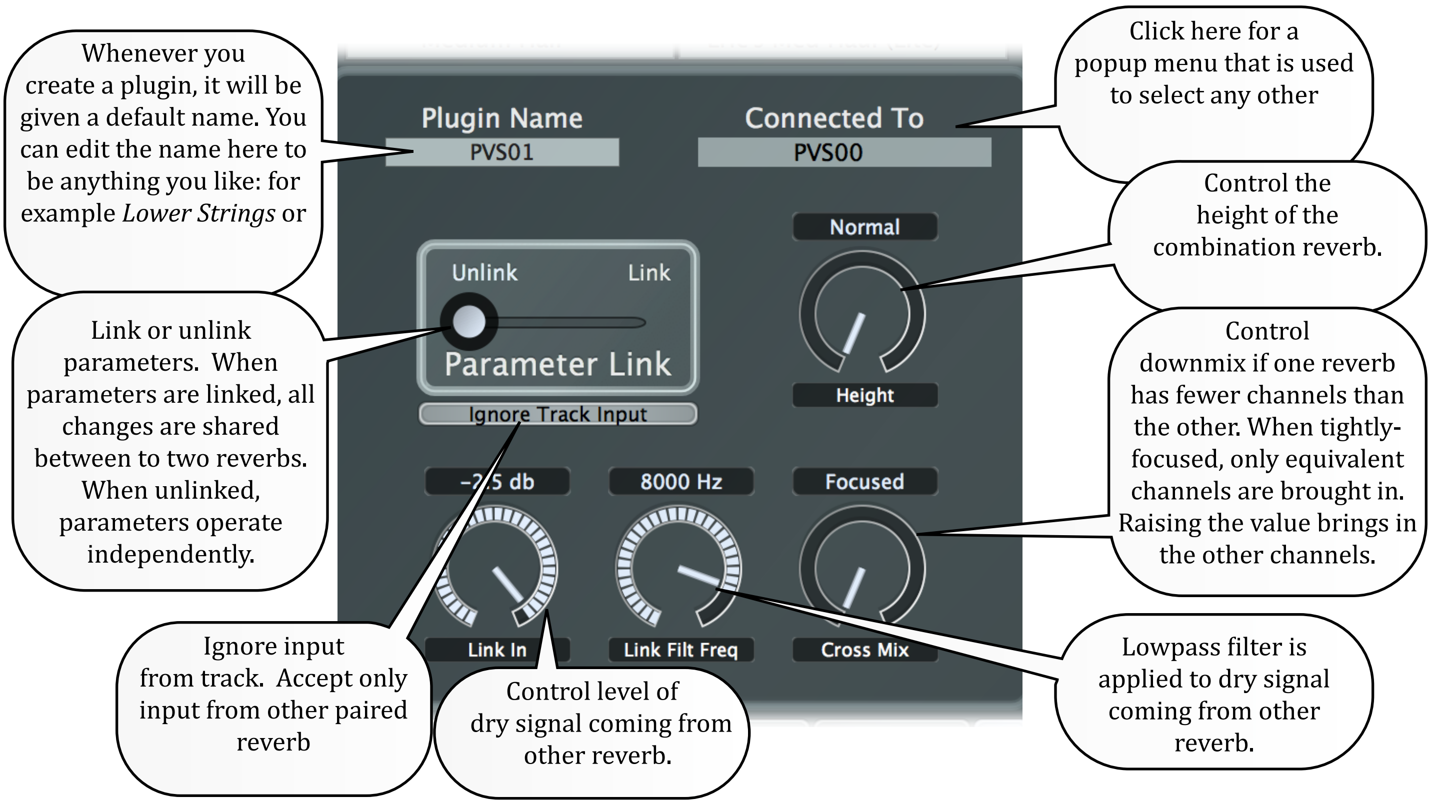
5.8.6. Gate (R2 Surround only)

The Gate allows the reverb levels to be controlled by the characteristics of the input signal. This can be used for special effects (giant snare drums from the 1980s), but can also be quite helpful in matching ADR to the characteristics of location sound.



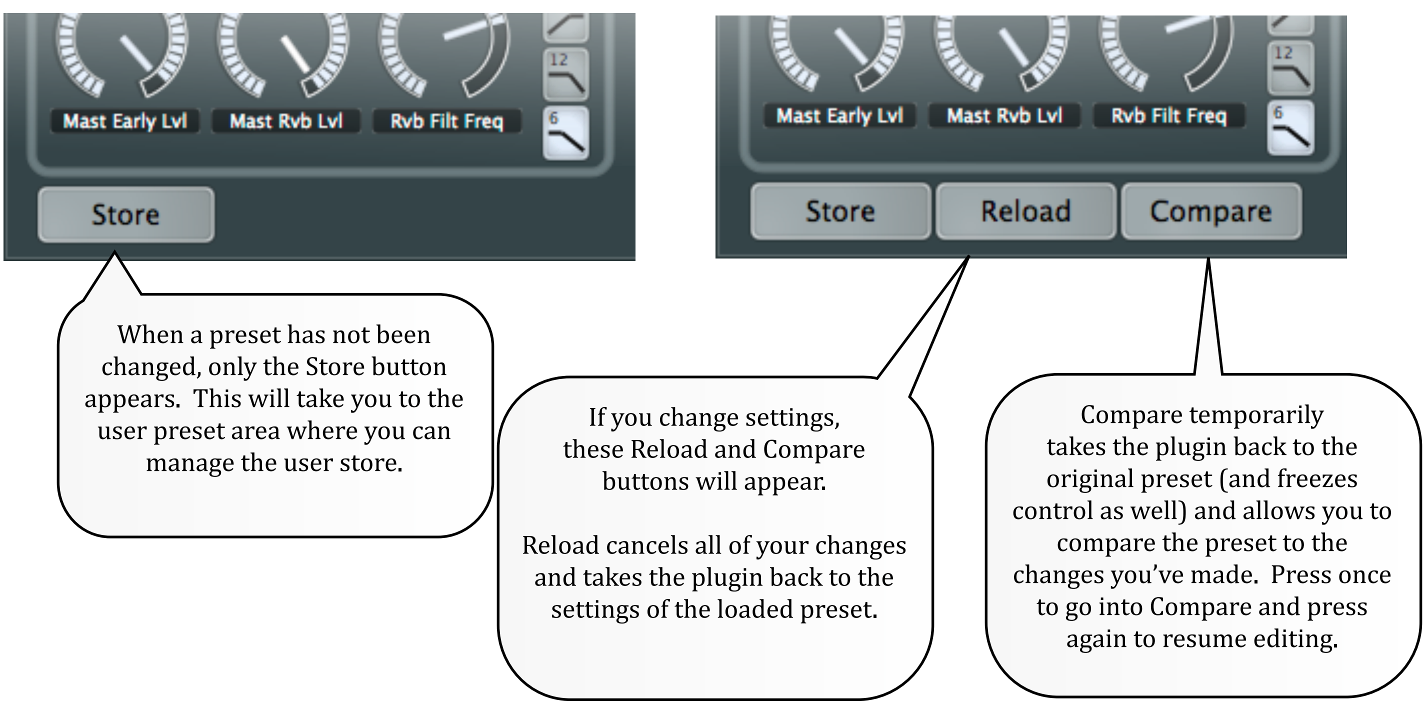
5.8.7.3D

The 3D page allows any two reverbs to be linked in order to create a single 3D reverb. This reverb may be used in projects using Atmos®, Auro 3D®, or any other format using height channels.



5.9.Compare and Reload

At the bottom left of the plugin, you'll find ways to access the library as well as ways to compare and undo changes you've made to current settings.

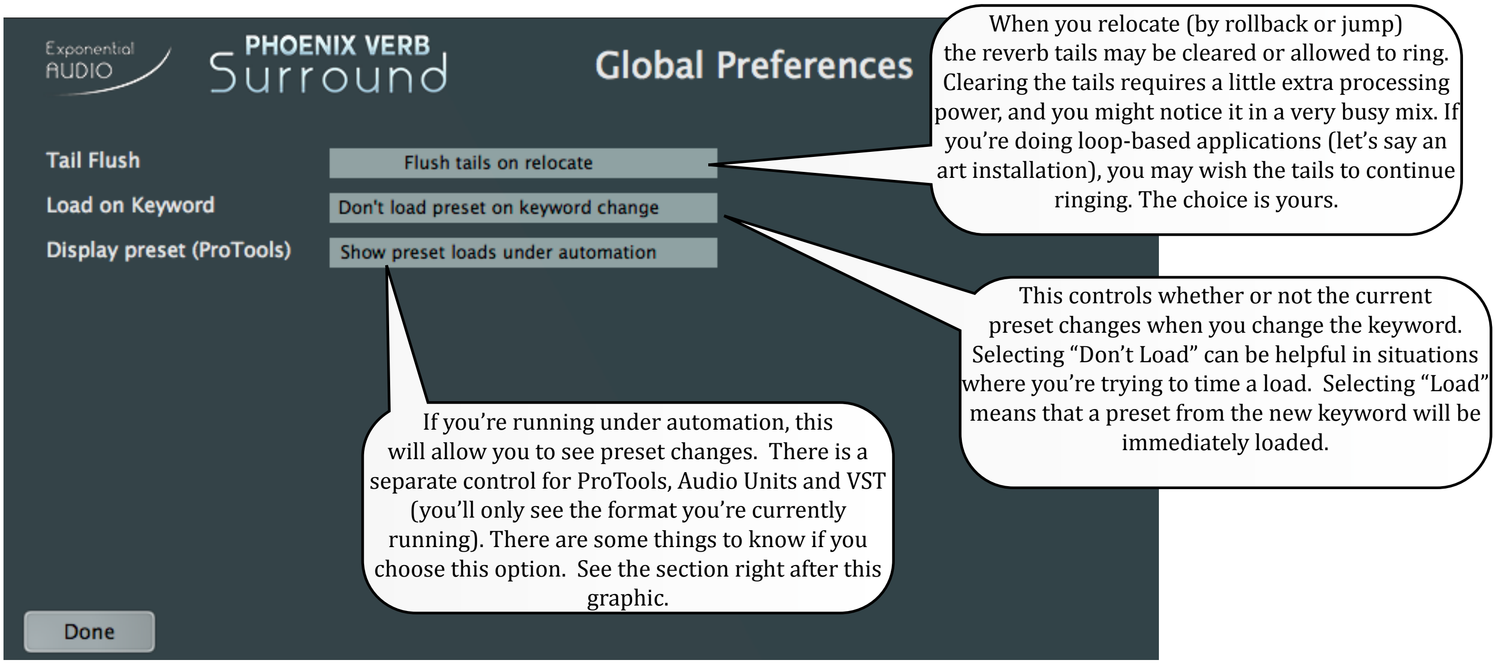


5.10. Settings stored with job.

When you save a job, preset names are stored for each plugin, along with any adjustments you've made after loading the preset. Reloading that job will restore your *PhoenixVerb Surround* and *R2 Surround* plugins back to the exact state they were when you saved the job. This is true even if the presets no longer exist on a system. Let's say you copy a job from one computer to another. The second computer does not have the user presets that exist on the first computer. The preset names will still appear in the plugin, except they will be dimmed and in parentheses. You can still adjust and save settings, but this is your cue that the source preset is not on your system. You may wish to save those settings locally as a user preset. The preset will then be available to any other instantiations of the plugin.

5.11. Global Parameters

In some cases, you may wish to change the behavior of all copies of the plugin, wherever they are. For this, you can control global parameters through the Preferences Page. To launch the page, simply click the "Pref" button in the extreme upper-left corner of the plugin.



5.11.1. More about Global Parameters

The *Display Preset* option is a little complicated. It works by comparing all of the plugin's current parameters to the values stored for Factory and User presets. If there's a match, the matched preset name will appear in the preset field of the GUI. If there's no match, the preset name will not change. There are two basic rules to keep in mind:

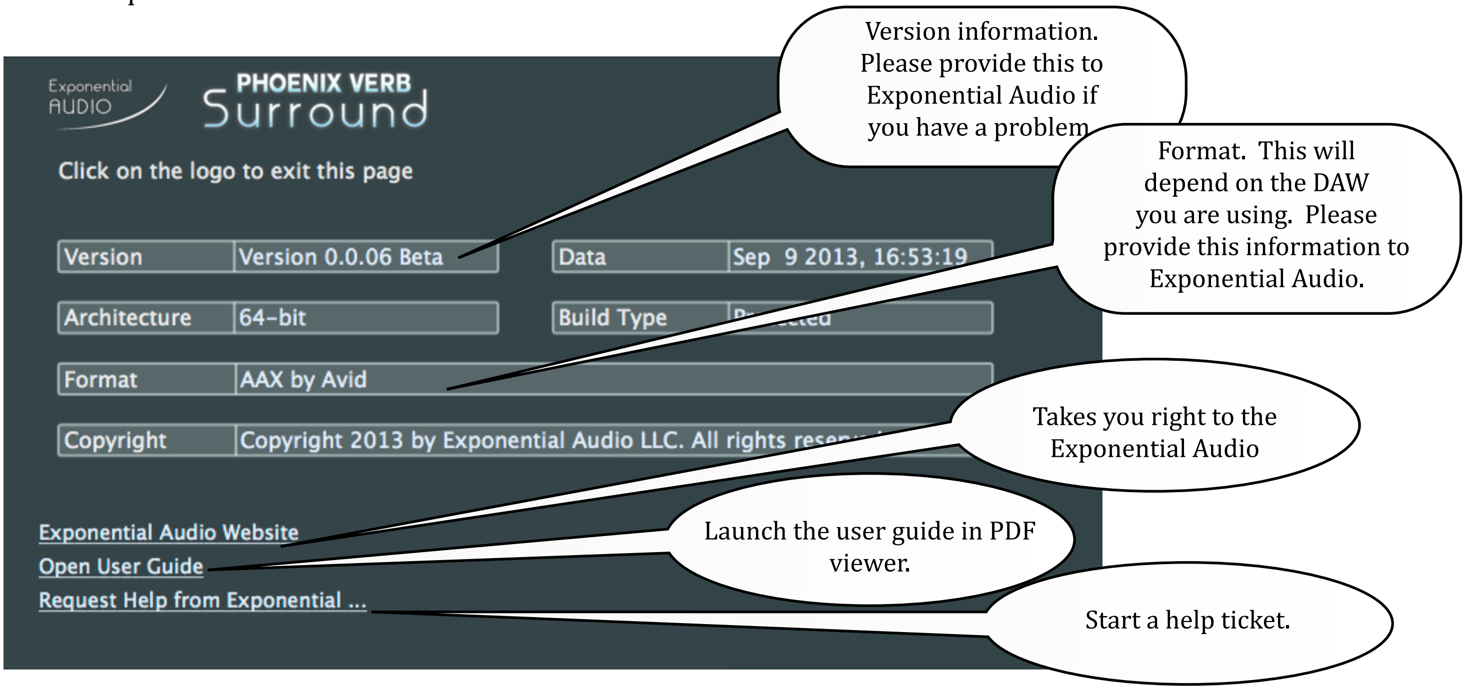
- You must automate all of the preset parameters (you can exclude bypass if you wish).
- Your workstation program (DAW) must not glide parameters.

Two DAWs work pretty will with this option: ProTools and Cubase. There are probably others. Some DAWs can't seem to turn off gliding (even with an option). Those include Logic and Digital Performer, but there are surely others.

So if you'd like to try this out, turn on the option, make sure all your parameters are automated and give it a try. If it doesn't work, turn it off.

5.12. Getting Version Information and help

If you need version information for a support or upgrade issue, simply click on the Exponential Audio logo in the upper left corner of the plugin. You'll see a page with version information as well as links for this user guide and online help.



6. Editing, Saving, Importing and Exporting

6.1. Editing

There are many ways to edit parameters. Here's a quick look.

6.1.1. Editing by Knobs

Most parameters are edited by knobs. Simply click on the knob (you'll know you have it when the color changes) and drag the mouse up or down.

6.1.2. Editing by Typing Values

Parameters with knobs also have display areas. Sometimes it's simpler simply to type in the value you want. *PhoenixVerb Surround* will do its best to make sense of what you've typed.

6.1.3. Editing by Switches

A few parameters (such as *Diffuser Type*) use a multi-position switch. Just click where you want the switch to go.

6.1.4. Editing by Buttons

Some parameters--EQ types--use graphic buttons. Just click the button.

6.1.5. Editing by External Controller

Exponential Audio supports EUCON controllers as well as most recent Avid/Euphonix control surfaces. The quality of non-Avid EUCON implementations is spotty. The Cubase EUCON translator crashes pretty dependably. AudioUnits implementations display parameter values in the range of 0-1.

6.1.6. Special treatment of Mix parameter

Nearly all parameters are saved with presets (built-in or user-created), but is one place where this rule is not followed. Although the mix parameter is saved when a project is saved, you may notice that it's not changed when you load new presets into a plugin instance. This is to help you in auditioning presets. Any wet/dry balance will be preserved as you try out different presets.Saving User Presets

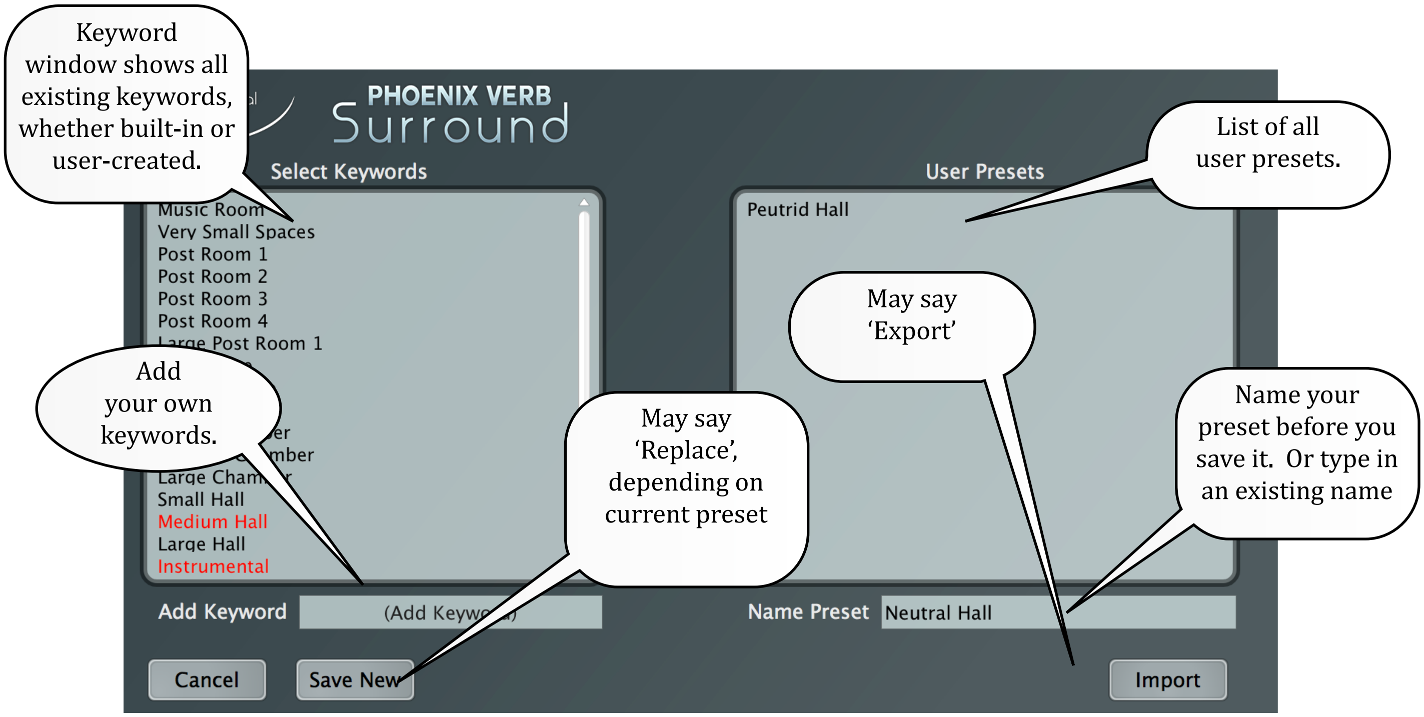
6.1.7. A word about preset format

You may notice that there are two ways to save presets. Your workstation program (Logic, ProTools, etc) will probably provide a way to store and recall user presets. That will appear at the top of your plugin window, in the wrapper area. Exponential Audio also provides a method that appears in the main body of the plugin window, and is accessed by the **Store** button. Why are there two methods and which should you use?

Most modern-day mix engineers use more than one workstation program. This often means that presets you created in one program are not available to another. This is especially true if the workstation program uses different plugin formats (AU, VST, etc). The workstation-specific method does not create portable presets. For this reason, Exponential Audio does not support the workstation-specific method. It might work and it might not. Instead, Exponential Audio creates truly portable presets. Any preset you create in one DAW is easily available in any other.

6.1.8. Entering Store

If you've edited the plugin in such a way that you'd like to use those settings again, it's time to create a preset. First press the Store button at the lower left corner of the plugin window. You'll now see this window:



The buttons you see on the Store page are context-sensitive and will appear depend on the state of your system. The 'Save New' button appears when the name of the preset you're going to save is different from anything in the current user preset folder. If the name matches a user preset you already have, then the button will change to 'Replace'. If you've selected any presets in the User Preset List, then the 'Delete' button will appear, giving you the chance to delete user presets.

If you haven't selected any presets in the User Preset List, then the 'Import' button will appear, giving the chance to import user presets you have received from elsewhere (like a collaborator). If you *have* selected presets, then this button will change to 'Export', allowing you to export those user presets to share with a friend.

6.2. Saving Presets in Store

6.2.1. Selecting Keywords

You can now select whatever keywords you'd like to apply to your new preset. Click any keyword (selected keywords will show up in red). To select multiple keywords hold down *Command* as you click on keywords. All selected keywords will be applied to your new preset. Note: A hidden keyword (User) is always applied to user-created presets.

6.2.2. Creating New Keywords

It's possible that you'll need a keyword that isn't there. Let's say you often record a certain singer. You'd like to have a special set of presets for that voice. Simply type your keyword in the New Keyword field, save the preset, and it's done. The new keyword, let's call it "Special Sauce" has now been applied to the preset. It will also show up in the keyword list so that it's available for other presets. Now when you're loading presets, the keyword list will now show 'Special Sauce' as a keyword.

6.2.3. Saving the preset

Once you've edited parameters, entered Store mode and selected keywords, you're ready to save your preset. The preset name field will already have the name of the preset you loaded. But you've made changes so chances are you wish to rename it. Type your new name into the field and store it, using the "Save New" button.

6.2.4. Overwriting an Existing User Preset

If you've loaded a user preset and made more changes, you might do either of two things. You might decide to keep both the original user preset and your new edits. If that's the case, be sure to type a new name into the name field. But if you decide to replace the old user preset with the new one, the "Save New" button should have changed to "Replace". Simply press the button and you're done.

Don't worry about the factory presets. They're always there and you can't overwrite them.

6.3. Deleting Presets in Store

If you have user presets you no longer need, enter Store mode at any time. Select any preset you wish to delete (you can select multiple presets). You should now see a "Delete" button. Press the button and the old presets are gone. If you happen to delete the preset you're currently using, the plugin will revert to a factory default preset.

6.4. Exporting Presets

You may wish to share your user presets with someone else. Perhaps it's a friend, perhaps a collaborator. Simply select the presets you wish to share (you can make multiple selections) and press the 'Export' button. You will enter a dialog that allows you to overwrite any shared presets (if you already have users presets of the same name in the target folder), or to keep the old ones. Once you've made your choice, you'll be able to select a target folder. Your presets will be exported (don't worry—you still have the originals). You can then send the exported folder to anyone you like.

6.5. Importing Presets

If a friend or collaborator has sent you a folder of presets, you can import those into your own user folder. Once you're in Store, make sure that none of your own presets are selected. You should see the Import button. Click the button and you'll be taken to a dialog where you can choose whether imported plugins overwrite your own plugins of the same name or if your old plugins stay in place. Once you choose, you'll enter a Eile selection dialog. You can select any plugins you'd like to import.

6.6. Canceling

If for any reason you decide not to store, replace or delete a preset, just press the "Cancel" button and you'll return to the normal plugin window.

7. DAW Automation

DAW Automation uses the built-in abilities of the workstation program. Ever DAW is different, but they're all somewhat similar. You have the ability to record parameter changes as your project plays. You might record small changes in reverb time on one pass, and then record filter changes on another. These will always play back in exactly the same way. This is how many mix engineers fine-tune a complicated mix. Those changes will play back faultlessly every time you play the project. It's important to keep two things in mind:

- While you can change a preset during automation, you may not see the actual preset name change as you play back the mix. All parameters should change and the sounds should be fine. It's simply that you may not see the name change. See the earlier section on global parameters. You might be able to see the preset changes.
- You cannot load or unload a plugin during automation. That's just not the way it works. If you think you need to do something like that, it's best to insert all the plugins you'll need on a track and then automate bypass or wet/dry mix.

8. More about 3D Link

Many projects now require the use of height channels. Anyone who's heard Auro 3D® or Atmos® in a large theater will know how much is added by the use of height channels. At the time of this writing, no workstation program directly supports this sort of channel count. 3D formats—at a minimum—require a 'bed' with some number of low and high channels. The DAW will be set up with busses and tracks to cover those channels. Some busses are mapped to low channels and some to high channels. Unfortunately there is currently no such thing as a true 3D bus, so workarounds must be used. Panners are available which allow a source to be moved around this 3D space.

Because there are no true 3D busses in most workstations, it's currently not possible to create a single 3D reverb. But Exponential Audio has invented a technology called *3D Link* that allows two reverbs—one for low channels and one for high channels—to be linked so that they respond to inputs in the way a true 3D reverb would respond. No matter where an input is panned, the reverb responds in a natural way. When a signal enters the low reverb, it propagates through the entire 3D space, low and high. If the signal enters the high reverb it propagates through high and low space, with appropriate delay and EQ. The user can link parameters of the two reverbs so that they respond as a single unified space. Or the parameters may be unlinked so that high and low have different characteristics (but still share natural signal propagation).

8.1. What can you link?

You can link any *two* reverbs of the same basic type, as long as they're not currently linked. You can not link a *PhoenixVerb Surround* to an *R2 Surround*. You can only link two reverbs in a pair, but you can have as many pairs as your CPU will support. You can link reverbs of any channel count. For example, you can have a 7.1 on the bottom and a 2.0 on top (Atmos® bed), or a 5.1 on the bottom and a 4.0 on top (Auro® bed). If you're doing dialog you might prefer LCR + LCR. For score or complicated FX, why not go 7.1 + 7.1? If you have a way of getting all the channels to a final mix, you can combine the reverbs however you like.

8.2. Restrictions of 3D Link

This technology must use techniques that are not provided by the workstation, meaning the DAW doesn't always know what's going on in a channel. It's entirely possible that the DAW may not know a signal is active in a track. For example, there may only be input to a "low speaker" reverb. The DAW isn't aware that there is signal being distributed to the "high speaker" reverb, so it may actually shut that track down. You may need to experiment with your setup to make sure both paired reverbs are allowed to run. Here are a few rules of thumb:

- In some workstations, you may need to make sure the channel with the plugin has an input, even if the input uses a dummy bus. Some DAWs will not run a track if it has no input. This is not a problem in ProTools.
- If you are running a dummy bus so that a track will run, you might find it useful to click the *Ignore Track Input* button. This will shut off the actual track input while still allowing the track to run.
- Do not link two reverbs on the same track. This serves no purpose and will probably cause terrible feedback.
- If Parameter Link is on, you should only automate one of the reverbs. The other reverb will follow.

8.3. About the Auro3D ® High Surround Formats

These are special formats that appears in the Surround Format popup menu when the plugin is created in a 5.1 stem. In this format, the top center (VOG) channel is passed in the LFE channel by the Auro panner. This is necessary to correctly create the high space in an 11.1 or 13.1 configuration. There are two versions of this format:

- Auro3D ® High - in this form, early reflections and reverberation are output appropriately in the Top and Top Center channels (as well as the other high channels). While this is the closest match to reality, it has the possibility to be problematic in fold-down. If the Top channel is included at a high gain, it may actually reduce the sense of spaciousness because it's duplicated on all channels of the folded-down result. Careful monitoring is essential.
- Auro3D ® High (Phantom) - in the Phantom form, input at the Top Center and Top positions is still passed into the reverbs. But no early or reverberant energy is output on those channels. Instead, it's transmitted naturally to the four high corner channels. In an actual mix, there may be little audible difference as compared to the non-Phantom version. But the folded-down result may have a greater sense of spaciousness.

8.4. Do linked pairs consume more CPU?

There is a small additional load for each linked pair.

8.5. Can I link only a few parameters in a pair?

No. It's all or none.

8.6. Are LFE channels shared in a link?

Absolutely not. The inherent delay in the link process would cause cancellation problems in LFE.

9. The Algorithms and their Parameters

9.1. Reverb for Newbies

If this is your first experience with reverb, you might be confused by some of the terms. The two most important terms are Early Reflections and Tail. The early reflections represent the first few bounces of sound--off the stage floor, off the sidewalls. After a hundred milliseconds or so, the number of reflections grows so much that you perceive only that pleasant effect of the sound gradually dying away. This second part is called the tail (although old hands might call it 'echo' as well).

PhoenixVerb Surround and *R2 Surround* generate both of those components in ways that are both powerful and subtle. Experience is always the best teacher in learning how to get the most out of it. Experiment with the presets. See how they differ from each other and learn how parameter changes can affect the sound of the reverb.

Don't be afraid to use different presets on individual tracks or subgroups. Reverb can help place sounds into three dimensions and make each component sit in the mix more nicely. Reverb is almost always most effective when it's subtle. Many a mix has been ruined by the too-liberal application of reverb. Dial in what you think you need and then back off a notch.

9.2. Description of the Algorithms and their applications

PhoenixVerb Surround and *R2 Surround* are designed to all-purpose reverbs--but with a few very special twists.

9.2.1. PhoenixVerb Surround

If you're looking for a natural transparent sound, you've come to the right place. If the intent of your mix is to feel like it's in a real space, then *PhoenixVerb Surround* is the one. It can gracefully move the sound from original through early reflections into reverberant tail. It never sounds like an added-on reverb: it simply becomes part of the source Whether the source comes from a studio or from spot mics onstage, *PhoenixVerb Surround* helps you to move it into an absolutely convincing world.

9.2.2. R2 Surround

Sometimes reality just isn't enough. You're looking for that reverb that floats all around you. Maybe it's that guitar solo or maybe it's a choir. The chorusing features of R2 Surround make a savory part of the mix. And there's more. If your a dialog editor trying to drop ADR into the middle of location dialog, the gate feature gives you that little halo of reverb that matches up naturally with room tone.

So which one should you use? Exponential Audio obviously hopes you'll choose both. But either *PhoenixVerb Surround* or *R2 Surround* can cover your needs in ways no other reverb can.

9.3. The Format Selector

There is a popup selector on the left side of the plugin window, about halfway down the page. Depending on the DAW and the track's channel configuration, this may offer additional choices for surround format. In most cases, it will provide a "phantom center" variant of the primary format. This format will accept audio on the center channel, but all reverb will be routed to left and right, keeping the center clear in cases where that's necessary.

In certain DAWs, the format selector is needed in order to select the actual format. Not all DAWs provide sufficient information for the plugins to determine the desired format. In that case, the format selector must be used. There are notes for particular DAWs elsewhere in this user guide.

9.4. Parameter Descriptions

9.4.1. Mix

Mix controls the ratio between wet (processed) signal and dry (unprocessed) signal. It should only be used when the plugin acts as an insert. There are many cases when a reverb is placed on a send path, shared by several channels. In that case, the mix should stay at 100% and reverb level should be controlled by changing the level of the channel strip holding the reverb. The reason is simple: there should only be one path of a signal to the output. If a plugin is on a send channel with a mix of less than 100%, there's the chance of dry signal reaching the output from both the reverb channel strip and the original signal channel. While DAWs are very good at delay compensation, there's always the chance of cancellation. Don't do it.

9.4.2. Predelay

Predelay is a delay added to the entire effect. In a general sense, it represents the difference in time between the direct audio signal and the first reflections reaching the ear of the listener. Practically, it adds a little (or a lot of) separation to the dry signal and the reverb. This can work wonders in increasing the clarity of the signal while still adding warm and enveloping reverb.

9.4.3. Rvb Delay

This is a delay used to separate the reverb tail from the early reflection. The amount of delay specified by this parameter also has predelay time added to it. In most cases, it should simply be left at zero. But for certain special cases (slapback, exteriors, etc), this parameter can be used to great advantage.

9.4.4. Reverb Time

Reverb Time works in conjunction with the Reverb Size parameter. Reverb Time may be seen as the reflectivity of the space you're modeling. More reflectivity (harder surfaces)—no matter the size of the room--will cause the reverberant energy to last longer. A small Reverb Size with a long reverb time will produce a long reverb with dense reflections and perhaps some coloration. A large reverb size with a shorter reverb time may also create a longer length with lower density and a more natural sound. It's best to test these parameters with impulses such as clicks or snare drum strikes.

9.4.5. Rvb Time Balance

In a natural space, reverb time is equal in all directions. But sometimes that's not what you want. This parameter allows you to adjust the reverb time (overall time controlled by the Reverb Time parameter) so that time in the front or rears is longer. Lower values (knob to the left) make time longer in front. Higher values (knob to right) make reverb time longer in the rears.

9.4.6. Master Early Level

The early signal may be thought of as the energy that's only been reflected off one or two surfaces. It can give the listener a sense of position relative to the sound source (near/far) and also give some sense of the area immediately around the source (boxy, open, etc). The Master Early Level parameter gives the mix engineer control over the level of this component. If the studio is blessed with a good recording space, it might make sense to reduce this level. If a sample library is in use, or if a vocal or drum booth has been used, a little more early signal can help to place the audio into a real space.

As indicated by its name, this is a master control. There are individual levels for front, center, side and rear that are controlled by this master.

9.4.7. Front Early Level

This is the early level for the front channels (left and right). It is controlled by the Master Early Level.

9.4.8. Center Early Level

This is the early level for the center channel. It is controlled by the Master Early Level.

9.4.9. Side Early Level

This is the early level for the side channels. It will only appear in 7.0 and 7.1 formats. It is controlled by the Master Early Level

9.4.10. Rear Early Level

This is the early level for the rear channels. It is controlled by the Master Early Level.

9.4.11. Master Reverb Level

Reverb Level is used to control the amount of reverb 'tail' in the signal. The tail is the most noticeable part of the reverb--the energy that dies away slowly and gives the sense of a small or large space. Balancing the Reverb Level and Early Level can give a good sense of microphone placement. For example, a low level with a long reverb tail might indicate close micing in a large space. If the reverb tail is higher and early level is lower, that might give a sense of more distant micing.

As indicated by its name, this is a master control. There are individual levels for front, center, side and rear that are controlled by this master.

9.4.12. Front Reverb Level

This is the reverb level for the front channels (left and right). It is controlled by the Master Reverb Level.

9.4.13. Center Reverb Level

This is the early level for the center channel. It is controlled by the Master Early Level.

9.4.14. Side Reverb Level

This is the early level for the side channels. It will only appear in 7.0 and 7.1 formats. It is controlled by the Master Reverb Level

9.4.15. Rear Reverb Level

This is the early level for the rear channels. It is controlled by the Master Reverb Level.

9.4.16. Out Frequency

This controls the cut-off frequency of the output filter. This value is frequently adjusted. This may be for reasons of material--getting the most natural sound with the source audio--or to slot into a busy mix.

9.4.17. Front Delay

This is a delay that's added to all processed sound (early reflections and reverb) for the left and right channels before it is sent to the output. There's a certain danger in the use of this parameter: the overall reverb image will not be as natural. But sometimes you need it anyway. Perhaps it's an exterior or special effect you're looking for.

9.4.18. Ctr Delay

This is similar to the Front Delay, except it specifies a delay to the center channel. The same cautions apply.

9.4.19. Side Delay

This is similar to the Front Delay, except it specifies a delay to the side channels. It will only appear in 7.0 or 7.1 formats. The same cautions apply.

9.4.20. Rear Delay

This is similar to the Front Delay, except it specifies a delay to the rear channels. It will not appear in stereo, mono, or LCR formats. The same cautions apply.

9.4.21. Output Filter Type

While reverbs may sound most natural with a gentle lowpass filter on the output, there are situations that can change that. For example, live performance venues may be somewhat 'boomy' and need some cut on low frequencies. A 'vintage' style mix may need a steeper cutoff to mimic the characteristics of low-bandwidth processors. *PhoenixVerb Surround* give you the choice of four filter types:

- 1-pole (6dB per octave) lowpass
- 2-pole (12dB per octave) lowpass
- 1-pole highpass
- 2-pole highpass

9.4.22. Reverb Type

This control adjusts many of the internal characteristics of the plugin. In general, plates are the most dense, with a little potential coloration. Chambers are also quite dense, but without coloration. Halls are the least dense, with a little more obvious back-wall effects.

9.4.23. Diffuser Size

Diffuser Size controls the feature size of the imaginary material that covers the wall of our space. Feature size is one way to describe what might be lined up along the wall. Your shelves full of Grammys would be small features. A row of life-sized Greek statues would be larger. In most cases, the 'linked' choice is best. Diffuser Size will be linked to Reverb Size. But the diffuser size can be controlled independently as well. In most cases, it's best to test with percussion and short reverb times.

9.4.24. Diffusion

When a sharp transient hits a wall, the way it reflects is driven by the shape of the wall, that row of Grammys, and the material that makes up the wall. There may be a single hard reflection, or there may be many smaller reflections with tiny time delays between them. This is diffusion. The diffusion control, unsurprisingly, controls the overall amount of diffusion. Once the basic Reverb Type and Diffuser Size parameters have been adjusted, this is used to make final adjustments. As a rule of thumb, sharper transients will benefit from more diffusion. But rules are made to be broken. Feel free to experiment.

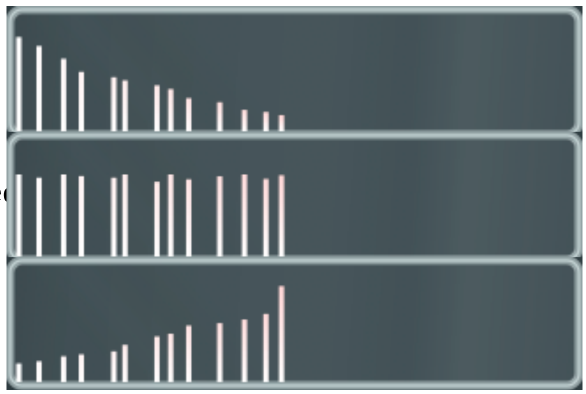
9.4.25. Envelope Attack

The envelope parameters are among the harder parameters to understand. They control the way that the audio signal enters the reverb. In some cases they may affect your sense of microphone placement. In others they may affect your sense of *listener* placement. The user interface gives a strong sense of what's going on with these parameters. The narrow vertical bars indicate reflections (the number of bars and relational placement are only approximations for the purpose of illustration). *Note: it's easier to hear the effect of the envelope parameters by turning early level off and using a short reverb time.*

<u>Low Attack Value</u>
Early audio is stronger

<u>Medium Attack Value</u>
Audio evenly distributed

<u>High Attack Value</u>
Late audio is stronger

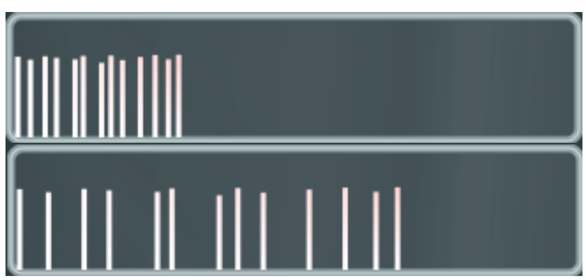


9.4.26. Envelope Time

The time parameter adjusts the overall time of the reverb envelope. This can have a great effect on the sense of reverb distance and depth.

<u>Short Envelope Time</u>
Signal injection in a
short time window

<u>Long Envelope Time</u>
Signal injection in a
longer time window

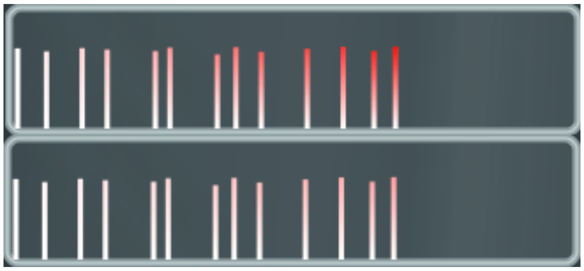


9.4.27. Envelope Slope

The reverb envelope has a lowpass filter for each delayed signal. Those filters are adjusted by this parameter. The lower the slope value, the more filtering on later signals. In many ways this is a model of air absorption. The red coloration on the delay bars helps to illustrate the effect.

Low Slope
Later energy is highly filtered

High Slope
Later energy is slightly filtered



9.4.28. Reverb Size

Reverb Size works hand-in-hand with the Reverb Time parameter. The size parameter gives you a general sense of the overall size of the space you're modeling (expressed in meters) and the time parameter controls the reflectivity of the walls. A larger size will lower the density of reflections and is generally more natural-sounding. But there's nothing like experimentation.

9.4.29. Xover Frequency

Natural reverberation in a large space typically lasts longest at the lowest frequencies. In very small reverberent spaces (locker rooms, for example). the lower frequencies may die away sooner. The reverb passes through a crossover filter, which is typically set to divide the low range from the mid range. This parameter controls that frequency. The Low-Mid Balance parameter controls how the reverb time is affected.

9.4.30. Low-Mid Balance

This parameter controls the way the reverb operates below and above the Xover Frequency. In the center position, low and midrange reverb time stays approximately the same (reverb time is always affected by the sort of audio material you use). Lower values of the parameter favor the low frequencies, meaning the midrange dies away sooner. Higher values favor the midrange and the lower frequencies die sooner.

9.4.31. Damp Frequency

In the real world, the highest frequencies die away sooner than midrange and low frequencies. This has many causes, including air absorbency and room treatment. Air absorbency is a function of basic humidity as well as humidity cause by a room full of breathing people. Room treatment typically means carpeting, absorbers on the walls, ceiling tile and so on. *PhoenixVerb Surround* gives you the ability to control the way these highest frequencies die away. The Damp Frequency parameter allows you to set the frequency above which this damping takes place.

9.4.32. Damping Factor

This parameter controls how quickly frequencies above Damp Frequency actually die away. The middle range approximates normal damping (*your* normal may differ). Lower values mean that the sound is darker and higher values mean it is lighter. Lower damping values may be used to simulate band-limited vintage equipment.

9.4.33. Width

The reverb tail in *PhoenixVerb Surround* is naturally wide and enveloping. Depending on your application, you may need to vary this. A wider tail will open up the space. High widths may cause some cancellation upon fold-down. A narrower tail might be useful in focusing the source more tightly, especially when centering dialog. It's important to note that the width control applies only to the tail: Early reflections are not affected.

9.4.34. Early Pattern

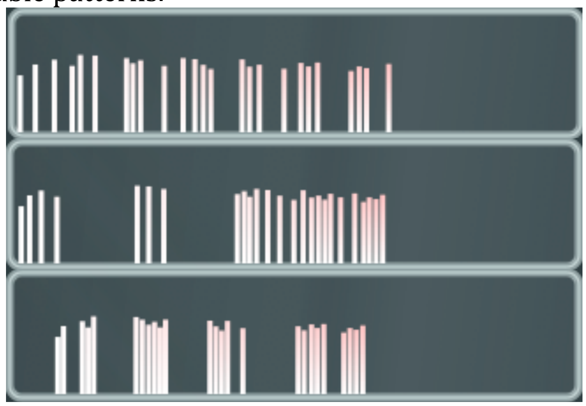
The early parameters work in much the same way as the envelope parameter, but with greater flexibility. They control the nature of early reflection patterns and are very important in generating the desired sense of space and directionality. In many reverbs, these reflections must be adjusted individually. In some ways that's a powerful feature, but in most cases it's time-consuming and fussy. In *PhoenixVerb Surround* and *R2 Surround* there are several sets of reflections that can be rubber-banded. The Early Pattern parameter allows any of several basic patterns to be chosen. The patterns are named according to their best use, but you don't have to follow those guidelines. If you're building a recital hall and an exterior pattern works, then use it! *Note: it's easier to hear the effect of the early parameters by turning reverb level off and turning early level up.*

The illustration shows just a few of the available patterns.

Smooth Pattern
Even natural distribution

Exterior 2 Pattern
Gap before last cluster

Slap 2
Good for problematic spaces



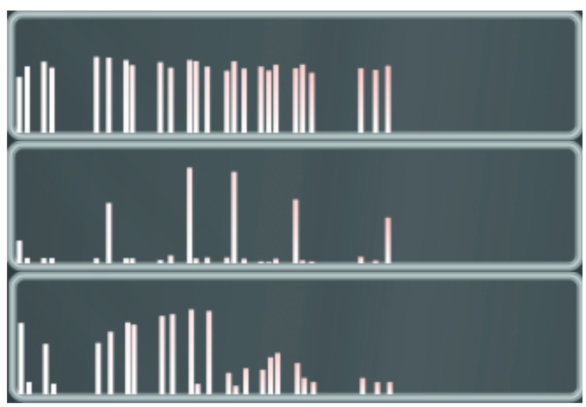
9.4.35. Early Density

The early density parameter controls the relative strength of reflections in the chosen pattern. It can have a large effect on the perception of that pattern. This illustration shows just a few of the available densities. The pattern is the same for all examples (the Irregular pattern)

Even Density
Levels of adjacent reflections are very close

Sparse 2 Density
A few reflections are very prominent

Choppy 1 Density
Irregular energy distribution between taps



9.4.36. Early Time

The time parameter adjusts the overall time of the early reflections. This can have a great effect on the sense of soundstage. It can also be used to smooth the harshness of close mics. This parameter works in the same manner as Envelope Time

9.4.37. Early Slope

The reflection group has a lowpass filter for each delayed signal. This parameter affects the early reflections in the same manner as the reverb Attack Slope parameter.

9.4.38. Early Distribution

This parameter allows you to take control of the placement of your audio source. In all cases, the reflections are related to the input channel where the audio enters the plugin (input channel has the earliest reflections, most distant output channel has the latest reflections). But for many applications--for example foley in a dense mix--the purely natural approach just won't do. Here is a list of the settings and their effect:

- Mono - all reflections are output only on the channel where there were input. This causes a very tight focus for foley or dialog. It can also compound the coloration of certain early patterns, such as box or slap patterns.
- Wide Mono - all reflections are output on the same channel as input and on the adjacent channels in the surround field (for example, in a 7.1 mix with input on Left Front, the reflections go to Left Front, Left Side and Center (unless phantom center is selected, in which case that last signal goes to right). This still provides a strong locational focus, but does blend into a surround mix a little more naturally.
- Left-Right - all reflections remain in the left-right plane of the input channel. For example, an input in Left Rear only reflects in Left Rear and Right Rear. An input in Right Front reflections only in Right Front, Center and Left Front. This allows the mixer to generate separate front and back planes in the mix.
- Front-Back - All reflections remain on the side of the input channel where the signal entered. For example, a signal coming in Left-Front reflects only in Left Front, Left Rear and Left Side (if the mix is a 7-channel mix). This allows the mixer to keep separate left and right planes in the mix.
- Surround - reflections are distributed evenly and naturally around the entire surround field.

9.4.39. Parameter Link

This switch only appears when two copies of the plugin are linked in a 3D configuration. Turning the switch on connects the plugins into a single unified plugin. Parameter changes and preset changed on one copy are reflected on the other.

9.4.40. Link In

When two reverbs are linked with *3D Link*, they share input. Input from the *other* reverb is treated (diffused, delayed, filtered) and passed to the input of *this* reverb. Link In controls the level of that additional input.

9.4.41. Link Filt Freq

In addition to the input level treatment in *3D Link*, the signal from the other reverb may also be filtered before being combined with the input of *this* reverb.

9.4.42. Ignore Track Input

As mentioned elsewhere in this document, we must sometimes trick a DAW into running a track. If the DAW has no input (for example, you're creating a 3D reverb while only bringing dry signal into the lower area), you'll usually have to connect a send or some sort of signal to both reverbs in the pair. For the reverb connected to the dummy bus, you can click the *Ignore Track Input* button. This will trick the DAW into running the track (as long as there's some signal on the bus), but will not admit the track input into the reverb. This means that—in a pinch when you're out of busses—you can connect *any* bus to the track input while keeping that signal out of the reverb.

9.4.43. Link Cross Mix

When linking a pair of reverbs with different channel count (for example, link a 7-channel reverb to a 2 channel reverb), there is the question of how to handle the channels that are not shared by both reverbs. In this example, it would be the center, rear and side channels. The Cross Mix parameter is a way to manage this.

At its lowest setting, only equivalent channels are shared between the two reverbs. The 7-channel reverb would receive the inputs from the 2-channel in its own left and right channels. The 2-channel reverb would receive left and right front from the 7-channel. As you increase the value of Cross-Mix, the other 5 channels from the 7-channel will be downmixed into the 2-channel reverb.

Your use of this parameter will depend very much on your intentions in the mix. It is expected that this will be a useful parameter to automate.

9.4.44. Link Height

There is a small delay in the input that's transmitted from one reverb to its paired reverb. The Height parameter allows this to be increased for a more dramatic expansion of the space.

9.4.45. Chorus Type (R2 Surround only)

This parameter lets you choose from four basic chorus types:

- Pitch - a regular modulation of pitch. This was a key feature of many reverbs in the late '70s and '80s.
- Pitch(fat) - similar to Pitch, except that pitch moves in two directions simultaneously. The effect is more a fattening of the sound than a chorus.
- Rand - pitch modulation that happens on a more random basis. If you wish to avoid the regularity of the 'Pitch' setting, then try this.
- Rand(fat) - Similar to Rand, except that pitch moves in two directions simultaneously. The effect is more a fattening of the sound than a chorus. Chorus Rate (R2 Surround only)

Adjusts the rate of the chorus effect.

9.4.46. Chorus Depth (R2 Surround only)

Adjusts the depth of the chorus effect.

9.4.47. Gate Threshold (R2 Surround only)

The level of signal that must be coming into the reverb in order for the gate to stay open. When signal falls below this level, the gate will start to close over the time specified by Gate Clamp. If the threshold is off, then then gate will remain open and the reverb will be unaffected. Threshold is a sensitive parameter. The value you select will be determined by the material you are processing.

9.4.48. Gate Hold (R2 Surround only)

When the input signal falls below the gate threshold level, the Gate Hold parameter comes into effect. This is the amount of time that must pass before the gate begins to close. If new signal comes in before this time passes, the gate will remain open. This parameter is useful in shaping the gated reverb. It is also useful with choppy material and will keep the gate from thrashing open and closed.

9.4.49. Gate Cut (R2 Surround only)

This controls the total effect of the gate. If its value is "Off", then the gate will shut off all signal until the threshold is again crossed. If it's at -3dB, the effect of the gate will be very low and the signal will barely be cut.

9.4.50. Gate Clamp (R2 Surround only)

This controls how rapidly the gate will close after the Gate Hold period has passed. Short value cause the classic gate effect. Longer values are more natural in effect and may simply help to clear out the mix.

9.4.51. Gate Recovery (R2 Surround only)

This controls how quickly the gate will reopen when new signal enters the system. Generally short values are best, but longer values can create an interesting pumping effect.

10. Notes for Specific Workstations

Not every format is available in every workstation. Those formats should not be presented in those cases, but it's possible they might. You may find that they work imperfectly in those situations. As more DAWs support surround, Exponential Audio will review and update these reverbs as needed. Here's what's tested so far:

10.1. Pro Tools

All formats available, with the exception of Windows RTAS. In that case, 7.0, 7.1 and SDDS are not available. 3D link is fully supported.

Pro Tools gives you the ability to de-activate or re-activate tracks. This capability may be used to save CPU cycles or reduce visual clutter. If you are deactivating tracks with linked plugins, you must deactivate and reactivate those tracks together. Otherwise the link will be lost.

10.2. Digital Performer

Quad, LCRS, 5.1, 6.1, SDDS. Note: Digital Performer 8 does not currently provide a true 7.1 (L,C,R, Ls, Rs, Lr, Rr, LFE) format. It provides only SDDS. 3D link works in most situations.

10.3. Logic 9 and Logix X

Quad, LCRS, 5.1, 6.1, 7.1, SDDS. 3D link not supported because Logic supports only a single surround out.

10.4. Cubase 7 and 7,5

All formats support by Cubase are supported. 3D link supported.

10.5. Nuendo

Full support. 3D link supported

10.6. Wavelab

Not yet tested. Assumed to be the same as Cubase.

10.7. Pyramix

5.1

10.8. Fairlight Xynergi

There has been a favorable report from a customer mixing in 5.0/5.1. This has not been tested in other formats yet.!

10.9. Samplitude

Mono, stereo and 5.1. There are serious issues with Samplitude. Please read the section on Sequoia.

10.10. Sequoia

There is insufficient support in the Samp/Sequioa environment to provide all plugin formats or the provide them as conveniently as with other DAWs. At the present time, mono, stereo, LCR, 5.1 and 6.0 are supported. Whenever a plugin is instantiated, it will appear as mono. The proper format must be selected from the Format Selector popup window in the center-left portion of the plugin window. *It is not recommended that you purchase the Exponential Audio surround plugins if Samplitude/Sequoia are your primary DAWs*. An effort is being made to establish contact with Magix engineering to resolve the situation.

10.11. Ableton Live

Ableton currently provides support only for stereo. *PhoenixVerb Surround* and *R2 Surround* load as stereo reverbs in the Ableton environment.

10.12. Bitwig Studio

Bitwig currently provides support only for stereo. *PhoenixVerb Surround* and *R2 Surround* load as stereo reverbs in the Bitwig environment.

10.13. Sonar

Sonar uses a different method to support surround plugins than other DAWs. At the present time, that is not compatible with Exponential Audio.. *PhoenixVerb Surround* and *R2 Surround* load only as stereo reverbs in the Sonar environment.

10.14. Reaper

There is insufficient support in the Reaper environment to provide all plugin formats or to provide them as conveniently as with other DAWs. Only the formats shown in the table below are supported.. Whenever a plugin is instantiated, it will appear as stereo. The proper format must be selected from the Format Selector popup window in the center-left portion of the plugin window.

In addition, there is a fixed signal routing that *must* be used in the Reaper environment. It is up to the user to configure routing to and from the track holding the plugin. It is unlikely that this routing can always be achieved inline, so use of an aux track is recommended. This routing is listed here (L=Left, R=Right, C=Center, LFE=LFE, LR=Left Rear, RR=Right Rear, CR=Center Rear, LS=Left Side, RS=Right Side):

Format	Order (1 to n, consecutively)
Stereo	L, R
Quad	L, R, LR, RR
LCRS	L,R,C,S
5.1 and 5.1 Phantom	L, R, LR, RR, C, LFE
6.0 and 6.0 Phantom(cinema)	L, R, LR, RR, C, CR
6.0 (music)	L, R, LR, RR, LS, RS
7.1 and 7.1 Phantom	L, R, LR, RR, C, LFE, LS, RS

10.15. Blue Cat Patchwork

If you're using Blue Cat Patchwork, you'll have to be sure to select the right format from the plugin format popup window in the left-center of the plugin window. Although the plugins are aware of the number of channels in your stem, they need a little help to know whether 6 channels means 5.1 or 6.0. Your choice will be saved with the session.

Blue Cat provides the capability of creating presets of multiple effects. Such an effects chain can be created in one DAW and used in another. Or it might be opened in a stem with a different channel count. In most cases, Exponential Audio plugins will open correctly in the new format. But it's strongly recommended that you check the Surround Format popup window to make sure the correct format has been chosen. You may find that you need to change the value. Once that's done, your selection will save and restore properly with any session.

11. Getting Help

11.1. Exponential Audio Website

If you're having difficulty with the plugin, the first place to look is www.exponentialaudio.com. If you encounter what you believe to be a bug, then please report it by going to the info page of the plugin (click on the logo in the upper left corner) then clicking the "Request Help from Exponential Audio" link. This will prepare an email with important system information and a log that may include your problem. In the email, please describe what you were doing when you encountered the bug, and the best ways to reproduce the problem. Then send it along.

If *PhoenixVerb Surround* or *R2 Surround* have difficulty connecting with your email program, it will place the log file on your desktop. Please send this file to support@ExponentialAudio.com, along with a description of your problem.

11.2. iLok Website

If you're having problem with licensing or with your iLok, then be sure to visit www.ilok.com.

11.3. Public Forums

Exponential Audio maintains a presence on several popular forums and blogs.

12. Known Problems

Check the surround FAQ on the Exponential Audio website.

13. Updates

Be sure to check www.exponentialaudio.com periodically for bug fix updates to *PhoenixVerb Surround* and *R2 Surround*. While you're there, be sure to check out new products coming available.

14. Tech Notes

Most modern DAW programs handle plugin delay compensation automatically. But if you need to know, the delay of a dry signal through PhoenixVerb varies depending on the sample rate:

- 44.1/48K - 32 samples
- 88.2/96K - 64 samples
- 176.4/192K 128 samples
- Anything above - 256 samples

If you are loading down your DAW (and who doesn't), be sure to put away the GUI when you no longer need it. It does take processor cycles to run the user interface, and there's no need to burn the cycles if you don't need to control the plugin. The live frequency display also takes cycles, so most of the time it's best to leave it off: its default position. Turn the display on to hypnotize the producer so he'll let you get your work done!

15. About Exponential Audio

Exponential Audio is a new company with a long tradition. It was founded by a lifetime audio professional whose designs and algorithms have long been part of films, television programs and recordings of all sorts. It's a company dedicated to the established and new professional alike. Exponential Audio makes tools that sound good and are easy to use. This is a personal and professional commitment to you.

16. Acknowledgements

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