Hyperprism-VST Manual

[Table of Contents](#)

Arboretum SYSTEMS

Hyperprism-VST for MacOS VST Plug-in Compatible Applications

Real Time Digital Audio Signal Processing Software by Arboretum Systems Inc.

This documentation current for Hyperprism-VST version 1.5.5, last updated 11/10/98.

What's New in Hyperprism-VST v1.5.5:

- New Harmonic Exciter, Bass Maximizer and Tube/Tape Saturation effects
- Frequency Shifter, HyperVerb, Limiter and M-S Matrix now included in Hyperprism-VST
- Improved Low Pass, High Pass Filters; no more distortion!
- Improved Compressor and Noise Gate
- New simplified Installation procedure
- Now compatible with Emagic Logic Audio 3.5.3, Opcode Vision DSP and Cakewalk Metro 4.5.1, as well as Cubase VST
- Fix to "long plug-in name" bug

New In Hyperprism-VST v1.04:

- New Vocoder and Doppler effects

Contents

About This Manual

The documentation may be viewed or printed from any browser, such as Netscape Navigator or Microsoft Internet Explorer. If the text is too small or you wish to change the background color, go into your browser's Preferences and set them as you wish. Internet Explorer users, just hit the Larger button on your browser tool bar.

Important: For better scrolling comfort use the scroll bar, or the page up/page down keys, rather than the up and down arrows.

It may also be viewed using the excellent HTML Viewer By Sassafras Software, a shareware program supplied on this CD-ROM. If the text is too small, replace the *HTML Viewer Prefs* file in your Systems' Preference folder by the one on this CD. You can find it by doing a "Get Info" of your HTML Viewer alias followed by a "Find Original" command.

Manual Conventions

Throughout this manual, you will see several different symbols in sections describing step-by-step operations.

Text following a right pointing arrow ▋▶ invites you to execute a command with the mouse or keyboard.

Text following a down pointing arrow ↘ describes the result produced by executing a command.

A pointing hand ☞ indicates an important note!

If your Browser is connected to the Internet,
visit us by clicking the button below!
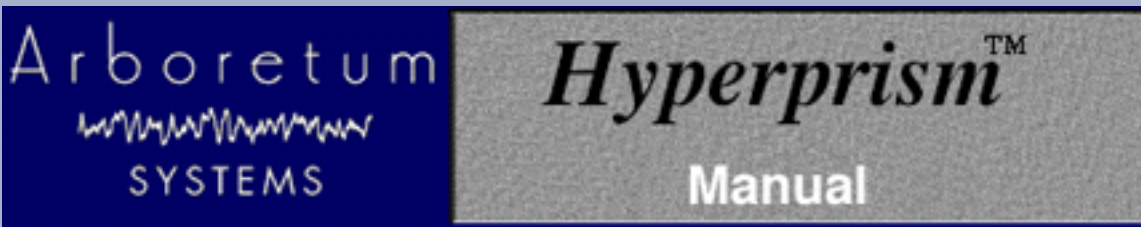
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Hyperprism-VST for MacOS VST Plug-in Compatible Applications

Welcome to Hyperprism-VST

Hyperprism-VST is a set of high-quality real-time audio plug-in effects. More than thirty different sound processing modules, tailored for the needs of electronic music production, film/video post and multimedia audio production. Each can be used from within any MacOS audio editing application that supports VST format plug-ins.

Hyperprism-VST is a component of the Hyperprism Plug-in Pack. Plug-in Pack also includes editions of Hyperprism for the MacOS Premiere and AudioSuite plug-in formats, as well as the Windows DirectX format.

The VST audio plug-in format is a specification for adding real-time audio DSP processors to sound editing software. The core advantages of the VST format are twofold: VST effects run in real-time, processing audio "on the fly," and VST effects require no additional audio DSP hardware, running on the computer's CPU instead of a dedicated sound card. The VST format was developed by Steinberg, their Cubase VST was the first program to implement the spec. In the Spring of 1998 Steinberg began licensing VST to third-party developers; Steinberg's competitors have been quick to adopt this framework.

Applications currently accepting VST audio plug-ins include Opcode Vision DSP, Cakewalk Metro, Emagic Logic Audio Platinum, Emu/Ensoniq Paris and Cubase VST. Support for the VST audio plug-in format is growing within the MacOS developer community; other programs which accept VST plug-ins may already exist, and more are sure to come.

Implementations of the VST format vary between various "VST-compatible" products. Many aspects of accessing and using plug-ins appear far different in Vision compared to Cubase, for example. The depth of VST support also varies; we'll make note of core discrepancies between the implementations in the various programs throughout the course of this manual.

Cubase-VST, by virtue of being the first program to offer the VST plug-in spec, serves as the general "model" for the purposes of this manual. We'll outline the basics of getting VST plug-ins going in each of the four products we were able to test during this development cycle: Cubase VST/24 version 4.0, Metro version 4.5.1, Logic Audio Platinum version 3.5.3 and Vision DSP version 4.1.1b3.

As of press time (11/15/98) we were still resolving technical issues surrounding Paris support; our engineering team is in direct contact with the Paris group and Hyperprism-VST compatibility with the Paris system may well be shipping by the time you read this. Paris-specific documentation will follow in a future version of this manual.

Metro 4.5.1 operations are documented in this guide, but at press time we were still working with the Cakewalk Metro team to solve some lingering problems. We anticipate that the Metro support and features discussed herein will be fully functioning by the time you read this material.

VST plug-in support is a new thing in these programs (besides Cubase.) As you'll see in the following pages, there are still some issues to be resolved. We've attempted to note in the pages of this manual all of the known compatibility problems and bugs as of the time of this writing. We're in regular communication with all our development partners regarding these type of issues, we're a diligent bunch of small companies, and chances are good that many of the anomalies we've noted here are already fixed.

Refer to your sound editing programs's manual for instructions on installing and opening audio processing plug-ins such as

Hyperprism-VST. Also be sure to read on to the next chapter in this document, [Installation](#), for step-by-step setup instructions, and tips on getting running.

Hyperprism-VST List of Effects

Hyperprism-VST includes 34 different effect algorithms:

[Auto Pan](#)
[Band Pass Filter](#)
[Band Reject Filter](#)
New! [Bass Maximizer](#)
[Chorus](#)
[Compressor](#)
[Doppler](#)
[Echo](#)
[EchoTranz](#)
[Flanger](#)
New! [Frequency Shifter](#)
[Hall Reverb](#)
NEW! [Harmonic Exciter](#)
[High Pass Filter](#)
NEW! [HyperVerb (TM)](#)
New! [Limiter](#)
[Low Pass Filter](#)
[M-S Matrix](#)
[More Stereo](#)
[Multi Delay](#)
[Noise Gate](#)
[Pan](#)
[Phaser](#)
[Pitch Changer](#)
[Quasi Stereo](#)
[Ring Modulator](#)
[Room Reverb](#)
[Single Delay](#)
[Sonic Decimator](#)
[Stereo Dynamics](#)
[Tremolo](#)
NEW! [Tube/Tape Saturation](#)
[Vibrato](#)
[Vocoder](#)

System Requirements & Compatibility

Hyperprism-VST requires a Power Macintosh computer, 80 MHz or faster with MacOS 7.6 or later and one or more audio editing programs which support the MacOS VST plug-in format. Applications which have been proven compatible with Hyperprism-VST include Cubase VST/24 version 4.0, Logic Audio Platinum version 3.5.3 and Vision DSP version 4.1.1b3. Metro version 4.5.1 and Paris compatibility is pending as of 11/15/98; compatibility with any other MacOS "VST-compliant" applications has not yet been demonstrated.

About Hyperprism-VST

Hyperprism can be thought of as a "rack" of studio-quality effects processors. Each effect in the package can be selected from a menu or insert point in the host application, previewed and adjusted in real-time and applied individually to any track, region or file.

One advantage of the VST plug-in format is Hyperprism and other plug-ins can work in real-time, processing "live" on an audio track. Compare this to other plug-in formats such as Premiere or AudioSuite which force you to render each process directly to file; the VST format means you can keep making changes to your effects as you build an arrangement, without "committing" to any process or sound until the final mix.

The key drawback of the VST plug-in format is that each active effect taxes the computer's CPU, and there's little to prevent you from adding more effects to a session than your CPU can process in real-time. Some effects are more processor-intensive than others; the HyperVerb and Vocoder are particularly hungry for clock cycles. If you're running out of processor headroom you'll likely have degraded previewing, sluggish controls, error messages or even freezes and crashes. All real-time audio applications, including Hyperprism-VST, will simply run better on faster Macintoshes.

Each plug-in instantiation takes up additional RAM, about 500k for each Hyperprism effect. Running short of RAM, especially during preview or playback can have severe performance implications.

Naturally, Hyperprism-VST effects can process directly to file, outside of real-time, which frees up CPU bandwidth otherwise dedicated to keeping the effect active in-line. It takes a moment or two for the process to be calculated once you've decided to render it to file. This also means that you will be consuming additional hard disk space.

Any effect parameter can be made to follow the movement of the mouse in the Blue Window. Once you have assigned effect parameters to mouse movements, simply click and drag the mouse within the Blue Window. As you do so, you'll hear and see the results; as your sound is being previewed, your gestures will change the effect settings. When you find the "spot" you like, you may process your sound with those particular effect settings. Note that most effect plug-ins only allow you to adjust one knob or slider at a time; Hyperprism gives you access to the entire set of controls, with a single mouse move.

☞ You may also use your host application's automation Write and Read capabilities to record dynamic effect changes in any Hyperprism-VST effect. These will be played back in time with your music, allowing a much more spontaneous and creative use of effects than the tradition "set-it-and-forget-it" behavior of typical signal processors. Hyperprism-VST can be played much like a musical instrument, changing your effects right along with the music...and it remembers and recreates your every move! VST implementations vary between the "compatible" sound editing programs; although Cubase-VST and Logic Audio Platinum have support for parameter automation, Vision DSP 4.1.1 and Metro 4.5.1 omit this feature; other applications may have varying degrees of automation control. Consult your sound editing program for complete information on parameter automation recording, editing and playback.

# Hyperprism-VST for MacOS VST Plug-in Compatible Applications

Installing Hyperprism-VST

- [Before You Install](#)
- [Easy Install](#)
- [Custom Install](#)
- [Configuring the Host Program](#)
- [Installation Support Notes](#)

Please read this section once through entirely and then follow these steps exactly!

Before you Install

▐▶ If you haven't already, install your sound editing program (specifically Cubase VST/24 version 4.0, Metro version 4.5.1, Logic Audio Platinum version 3.5.3 or Vision DSP version 4.1.1b3 or later) as per the instructions in the manufacturer's user's guide.

▐▶ You will need 12 MB of free hard disk space for each Hyperprism-VST installation you do. Installing in multiple applications takes additional increments of 12 MB of disk space per program.

▐▶ Place the Arboretum CD in your CD-ROM drive and double-click on the Hyperprism Plug-in Pack Installer.

▐▶ Follow all on-screen directions. You'll be asked to Accept the License Agreement, then you'll see the Read Me. Be sure to peruse the Read Me for important last-minute instructions which may not have made it into this manual.

➴ After the Read Me, the following installer dialog appears:
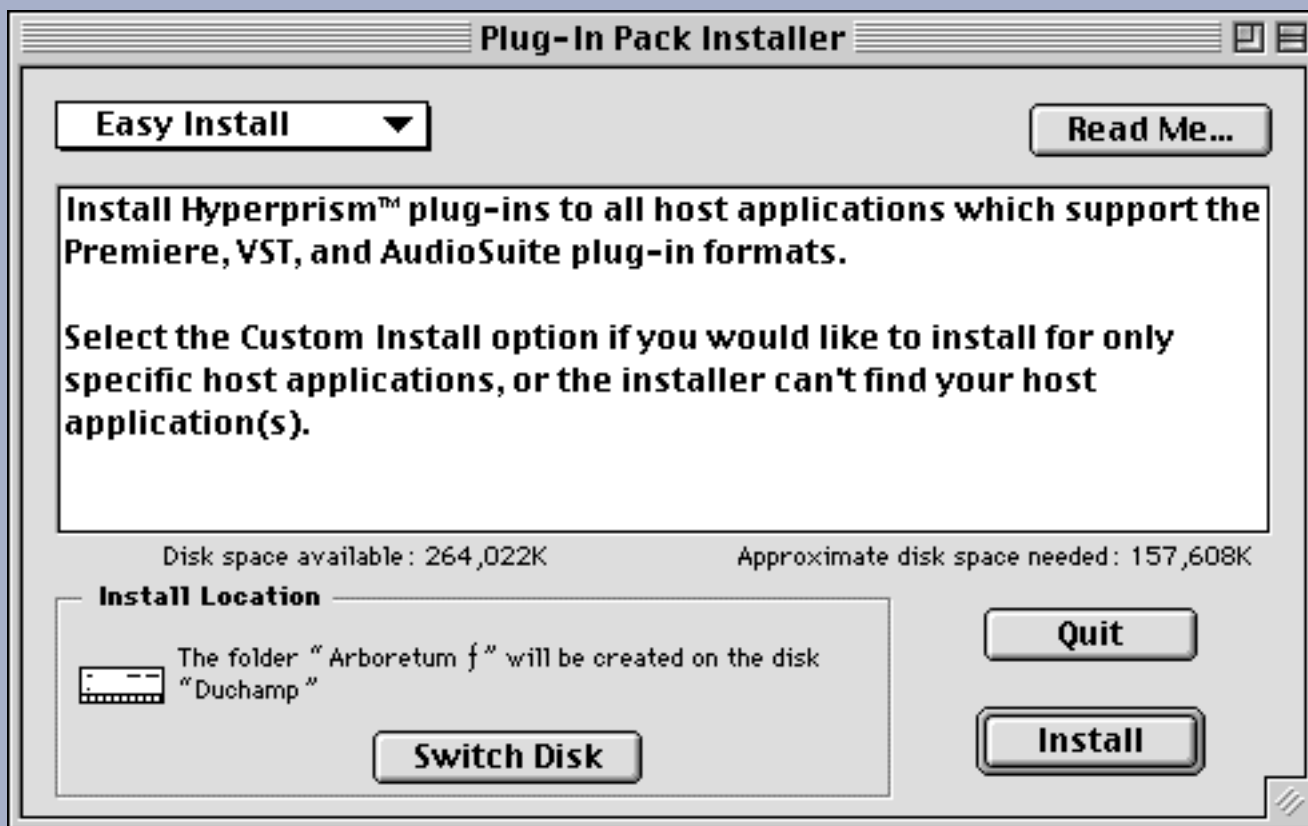
This is the Easy Install dialog. Note the pop-up menu at the upper left hand corner of this display; mouse-click on this menu if you wish to switch to Custom Install.

Performing an Easy Install:

▮▶ Check the Install Location display in the lower part of this Install dialog. Install Hyperprism onto the same hard drive where your audio editing applications reside. Click on Switch Disk if you need to specify another drive for installation.

▮▶ Click on the Install button.

⬥ Next you'll be asked for your name (mandatory), company (optional) and Hyperprism Plug-in Pack or Hyperprism-VST serial number. We've gone ahead and entered the first few numerals for you.

▐▶ Make sure you enter your serial number accurately. Double check your entry. When all fields are complete the OK button will highlight. Click on OK to proceed.

➤ If you have more than one copy of a particular audio program installed, or different versions of a certain program, you'll see the following dialog:
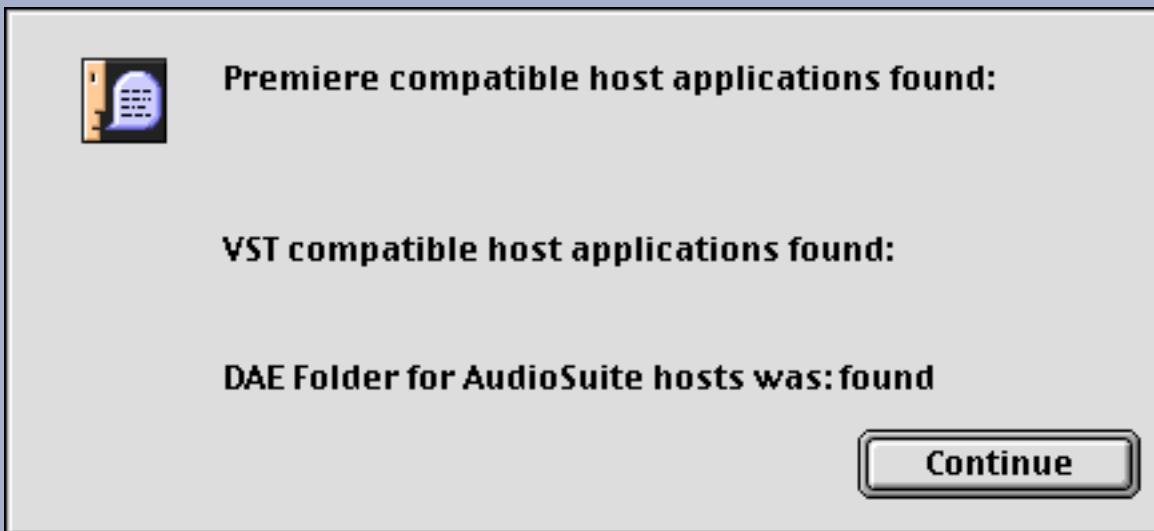

▐▶ Select the program/version you wish to install Hyperprism into by clicking on the program name in the top part of the dialog, then click on the OK button to proceed.

➤ After the Installer has found each of the audio programs on your hard drive, and copied Hyperprism to those programs' plug-ins folders, you'll receive a prompt much like the following, which gives you a report of the Installer's
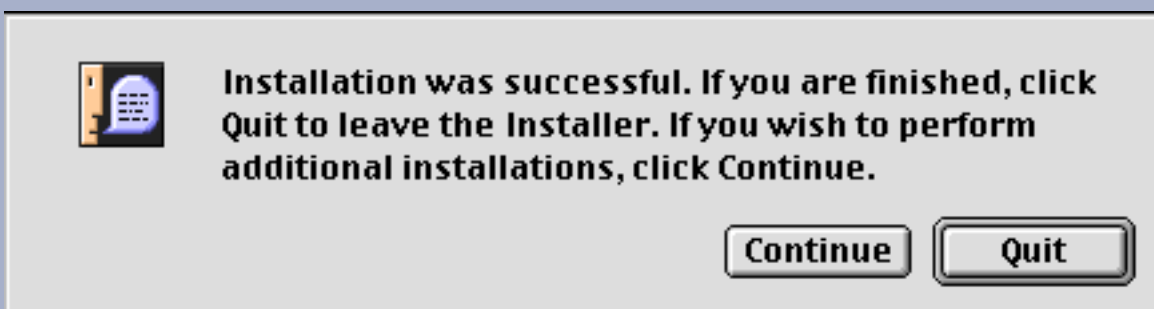
activity:

Premiere compatible host applications found:
Peak   Studio Vision   Logic Audio
Digital Performer   Adobe Premiere   Deck II

VST compatible host applications found:
Cubase VST   Logic Audio 3.5   Vision DSP   Metro

DAE Folder for AudioSuite hosts was: found

[Continue]

If the installer is unable to locate any Premiere or VST compatible audio programs, you'll get the following prompt:

Premiere compatible host applications found:

VST compatible host applications found:

DAE Folder for AudioSuite hosts was: found

[Continue]

If you're getting this message, with no program names listed after the Premiere and VST compatible entries, it means that no Premiere or VST compatible host applications could be found. Check that you're installing Hyperprism onto the same hard drive where your audio programs reside. Another reason for install failure could be that you've either changed the name of an audio program, or changed the name of the default folder where that program resides.

Click on the Continue button to proceed with the installation.

Assuming all the previous steps were completed successfully, you should next see the following prompt:

Installation was successful. If you are finished, click Quit to leave the Installer. If you wish to perform additional installations, click Continue.

[Continue] [Quit]

▐▶ Since you're doing an Easy Install, there's probably little need to perform additional installation. Click on the Quit button to proceed to the final stage of the installation.

↘ Now the Arboretum Hyperprism Serializers spring into action. The Serializers automatically apply your earlier serial number input to each of the Hyperprism plug-ins, so as to unlock all the plug-ins prior to first usage. After each batch of plug-ins has been serialized, you'll see the following prompt:
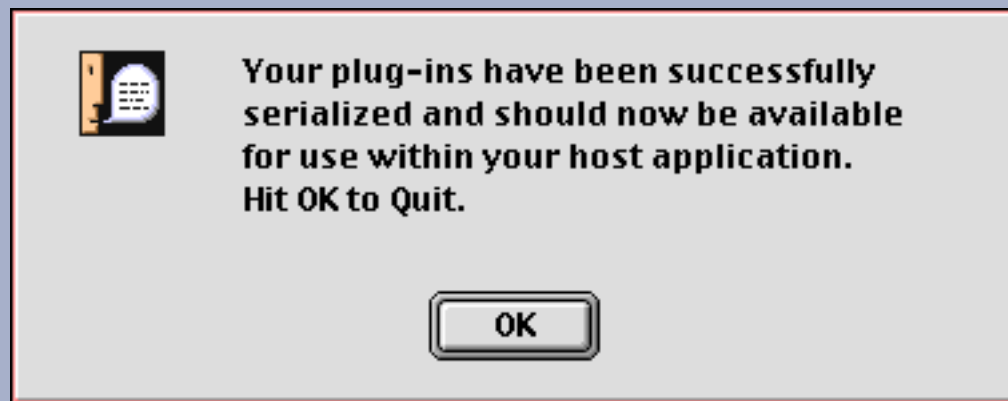
▐▶ Click on the OK button to proceed or conclude. If you're installing Hyperprism into multiple audio programs, this prompt will appear several times; click on OK to get through all of them.

↘ Your Hyperprism-VST installation is now finished!

Your Hyperprism-VST plug-ins are installed into the VST plug-ins folder for your host application(s). This plug-ins folder generally resides at the same directory level as your audio program.
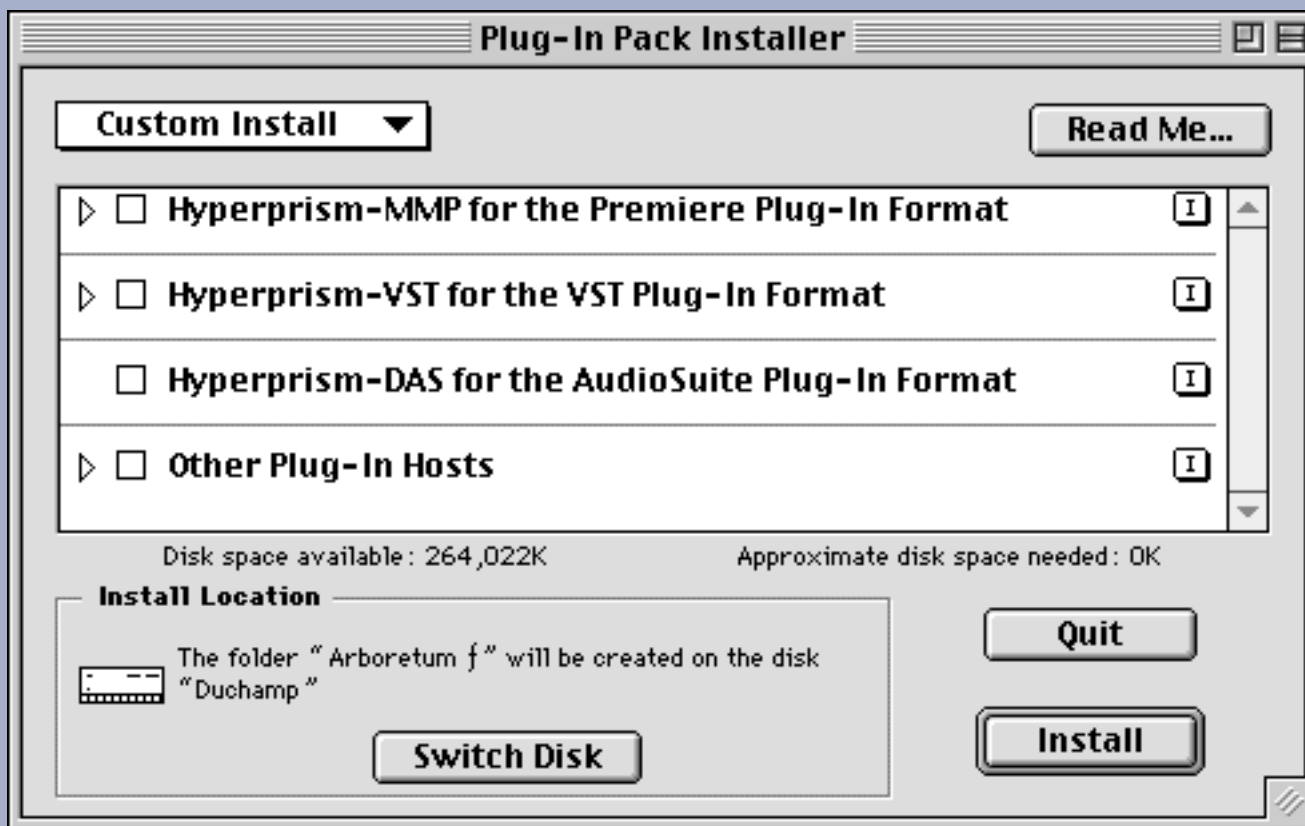

VstPlugIns

In addition, you will have a new folder of Hyperprism-VST Presets. Presets are usually stored directly alongside or inside the Plug-ins folder.

Hyperprism VST Presets

☞ Problems? If any difficulty arises during the course of installation, check out the installation tips at the end of this chapter, or the complete Troubleshooting section of this manual.

Performing a Custom Install:

This is the Custom Install dialog. Note the pop-up menu at the upper left hand corner of this display; mouse-click on this menu if you wish to switch to Easy Install.

In this window the various components of Hyperprism Plug-in Pack are segregated by plug-in format. Hyperprism-VST is the second group of selections in the dialog's window.

Click on the arrow next to the Hyperprism-VST for the VST Plug-in Format entry to reveal the detailed list of recognized host programs.
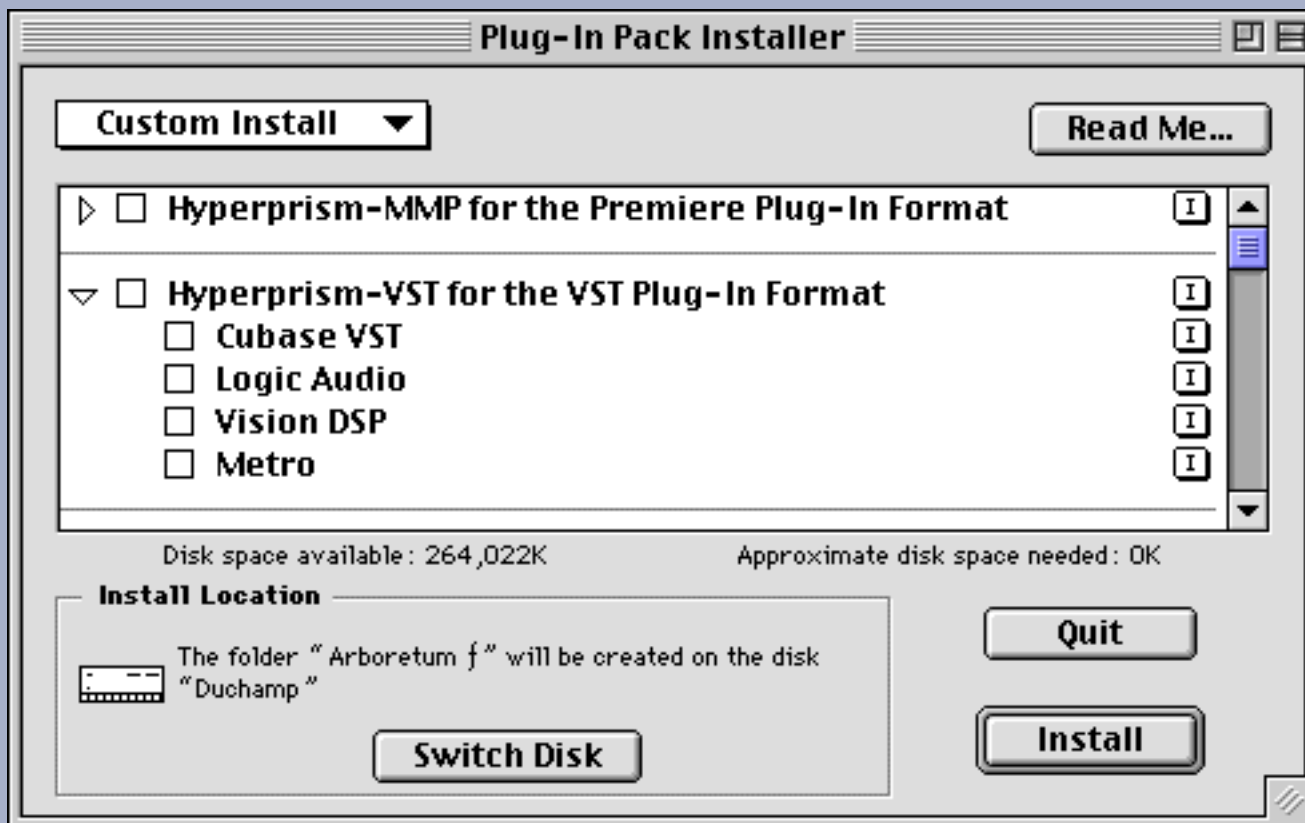
➘ Note the list of each "known" VST-compatible audio editing program (Cubase, Logic Audio, Vision DSP, Metro.)

⮚ Click on the boxes next to the programs in which you wish to use Hyperprism. For example, if you own Cubase and Logic, just select those two applications. Alternately, you can "select all" by clicking on the top item ("Hyperprism-VST") and the installer will search out each of the programs in the list, installing Hyperprism into those it can find.
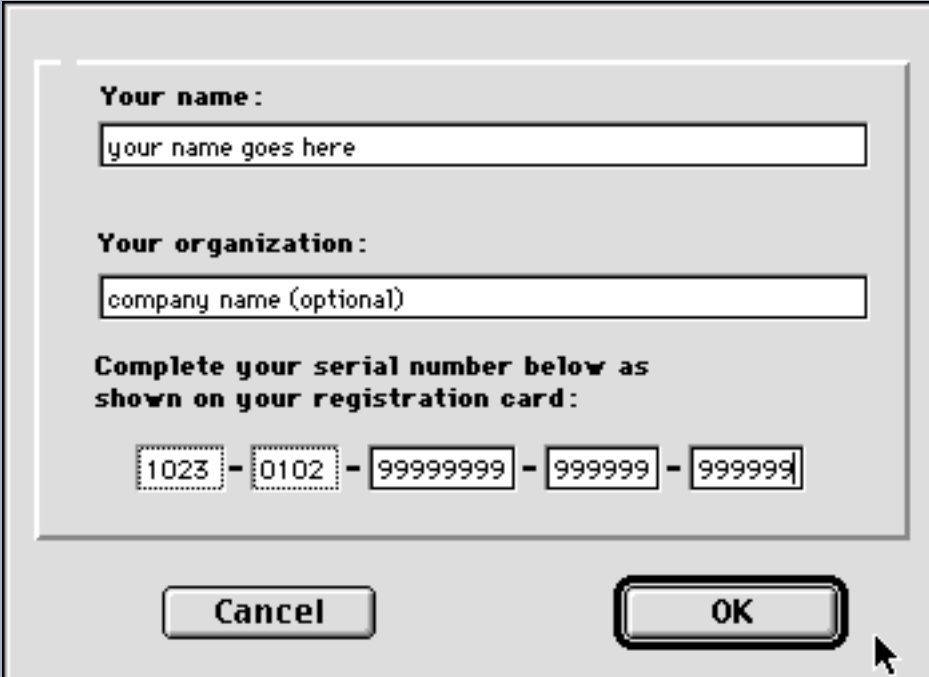
⮚ If the audio program you own is not on the list, or if you're encountering some other difficulty with installation, you can select the option "Other Plug-in Hosts" which will create a new folder on your hard drive called "Arboretum f" and will then install and serialize a set of Hyperprism-VST (etc.) plug-ins in that folder. If you're selecting this option you'll eventually have to drag your Hyperprism plug-ins into your audio program's plug-ins folder **by hand** once you've completed the installation.

☞ We suggest that you only use the "Other Plug-in Hosts" options if: 1) You have a Premiere or VST compatible audio program which is not included in the Custom Installer's list; 2) You've changed the name of your audio editing program or its default folder and the Hyperprism Installer can not locate them; or 3) You're instructed to do so by Arboretum technical support. Installing to Arboretum f unnecessarily wastes hard disk space.

⮚ Check the Install Location display in the lower part of this Install dialog. Install Hyperprism onto the same hard drive where your audio editing applications reside. Click on Switch Disk if you need to specify another drive for installation.
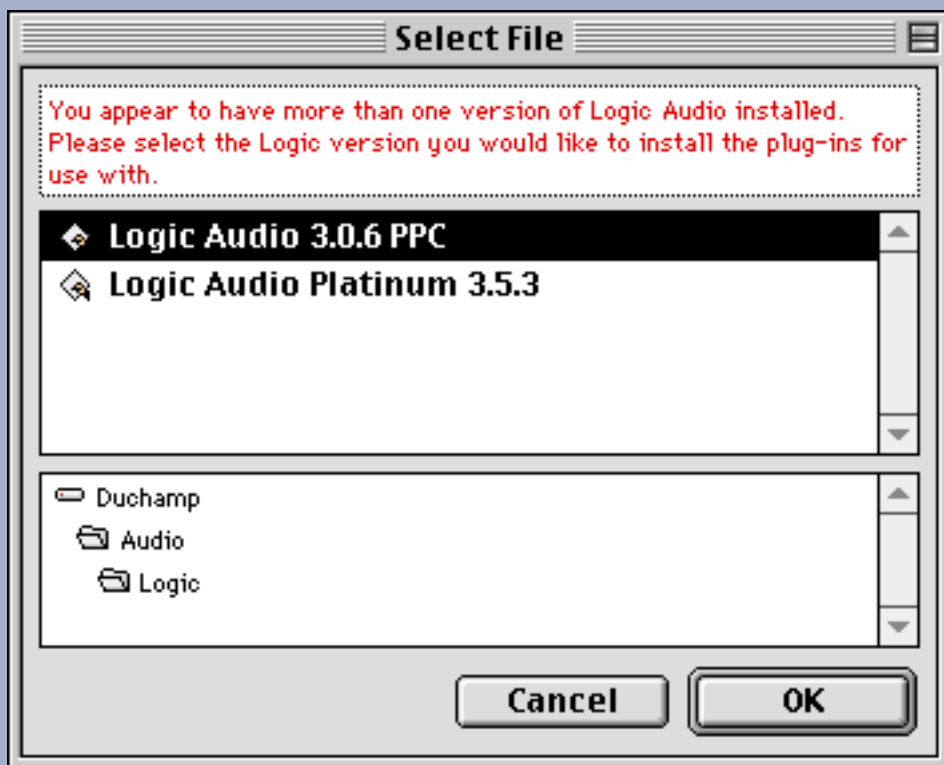
⮚ Click on the Install button.

➘ Next you'll be asked for your name (mandatory), company (optional) and Hyperprism Plug-in Pack or Hyperprism-VST serial number. We've gone ahead and entered the first few numerals for you.

```
Your name:
┌─────────────────────────────────────┐
│your name goes here                  │
└─────────────────────────────────────┘


Your organization:
┌─────────────────────────────────────┐
│company name (optional)              │
└─────────────────────────────────────┘

Complete your serial number below as
shown on your registration card:

  ┌──────┐   ┌──────┐   ┌──────────┐   ┌────────┐   ┌────────┐
  │1023  │ - │0102  │ - │99999999  │ - │999999  │ - │999999  │
  └──────┘   └──────┘   └──────────┘   └────────┘   └────────┘


   ┌───────────┐          ╔═══════════════╗
   │  Cancel   │          ║      OK       ║
   └───────────┘          ╚═══════════════╝
                                            ▶
```

⮚ Make sure you enter your serial number accurately. Double check your entry. When all fields are complete the OK button will highlight. Click on OK to proceed.

➘ If you have more than one copy of a particular audio program installed, or different versions of a certain program, you'll see the following dialog:

▐▶ Select the program/version you wish to install Hyperprism into by clicking on the program name in the top part of the dialog, then click on the OK button to proceed.

◣ Your installation is now in process. The installer will locate your selected audio editing programs and copy the Hyperprism-VST plug-ins into their respective Plug-Ins folders.

◣ Assuming all the previous steps were completed successfully, you should next see the following prompt:
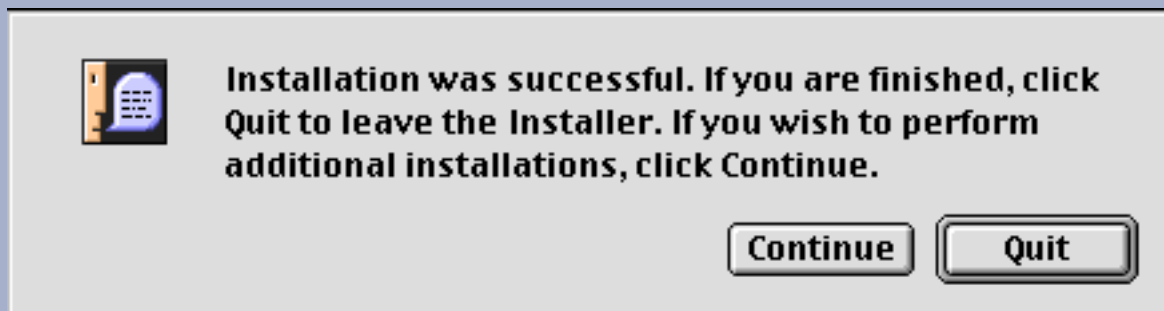
▐▶ Click on the Quit button to proceed to the final stage of the installation.

◣ Now the Arboretum Hyperprism Serializers spring into action. The Serializers automatically apply your earlier serial number input to each of the Hyperprism plug-ins, so as to unlock all the plug-ins prior to first usage. After each batch of plug-ins has been serialized, you'll see the following prompt:

Your plug-ins have been successfully serialized and should now be available for use within your host application.
Hit OK to Quit.

OK

▐▶ Click on the OK button to proceed and conclude. If you're installing Hyperprism into multiple audio programs, this prompt will appear several times; click on OK to get through all of them.

➤ Your Hyperprism-VST installation is now finished!

Your Hyperprism-VST plug-ins are installed into the VST plug-ins folder for your host application(s). This plug-ins folder generally resides at the same directory level as your audio program.

VstPlugIns

In addition, you will have a new folder of Hyperprism-VST Presets. Presets are usually stored directly inside the Plug-ins folder.

Hyperprism VST Presets

Configuring the Host Program

Each Hyperprism effect that you load requires RAM in which to operate. Allocate additional RAM to your sound editing program as necessary, we suggest increasing the Preferred setting by as much as 500 k for each Hyperprism-VST plug-in you wish to load concurrently.

▐▶ Go to the Mac Desktop and locate the host audio application on your hard disk.

▐▶ Click once on the program's icon so as to select it (being careful not to double-click and launch the application).

▐▶ Do a Finder "get-info" (press the keyboard shortcut Command-I).

▐▶ You must increase the default Preferred Size setting of the VST host program to allow memory space for each Hyperprism plug-in which will be loaded simultaneously. Allocate as much as 500k of RAM for each plug-in and more for delay plug-ins. If you have the memory to spare, allocate up to 2 MB for each concurrent plug-in instantiation.
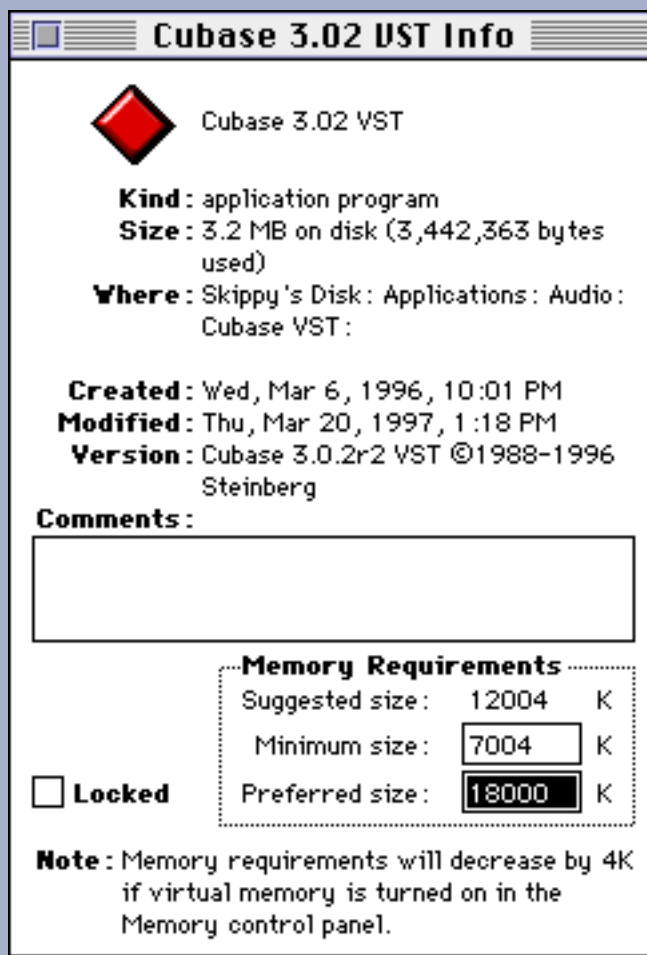
Cubase Get Info dialog (note Preferred size field is highlighted)

▐► Close the Get Info Dialog and launch your audio editing program.

Installation Support Notes

☞ The Arboretum installer supports all third-party VST and Premiere-compatible audio editing software packages that are known to us to be compatible as of November 1998. **If the sound editing program you own is not on our list** (such is the case with the forthcoming Bias Peak 2.0) or if you're encountering some other difficulty with installation, you can select the option "Other Plug-in Hosts" which will create a new folder on your hard drive called "Arboretum f" and will then install and serialize a set of Hyperprism plug-ins in that folder. If you're selecting this option you'll eventually have to drag your Hyperprism plug-ins into your audio program's plug-ins folder **by hand** once you've completed the installation.

☞ **Problems?** If any unexpected error messages come up during the course of installation, or if the Installer fails to place the Hyperprism plug-ins into your target program's Plug-ins folder, immediately quit the Installer and Restart your computer while holding the shift key down, so as to temporarily disable all Extensions and eliminate any possible system conflicts. (If your CD ROM drive is not an Apple built-in model, you may have to eschew turning off Extensions.)

Once restarted, go into the System folder, open the Preferences folder and drag the Hyperprism Preferences file into the Trash. Deleting the Preferences files will clear out the results of a faulty serialization procedure. Then run the Installer again and follow the preceding installation steps as directed. Also make sure you're using the correct Arboretum product serial number. Hyperprism 2 (for HyperEngine), Ionizer, Ray Gun, Hyperprism-TDM and Hyperprism-Plug-in Pack are each sold separately, and each require their own unique serial numbers.

When installing Hyperprism an Arboretum Serializer utility will also be installed in the Plug-ins folder. This utility will batch-serialize any Hyperprism plug-ins that are within the folder. When installing future updates to Hyperprism you'll be able to simply double-click the Serializer to authenticate any new plug-ins.

☞ If Trouble Persists: Link over to the [Troubleshooting](#) chapter of this manual. Also take a look at the Support frame up on the Arboretum web site <http://www.arboretum.com> for the latest technical notes and troubleshooting tips. If all else fails either email support@arboretum.com or phone our service line during our regular weekday business hours, 10 am to 6 pm Pacific: (650) 738-4750
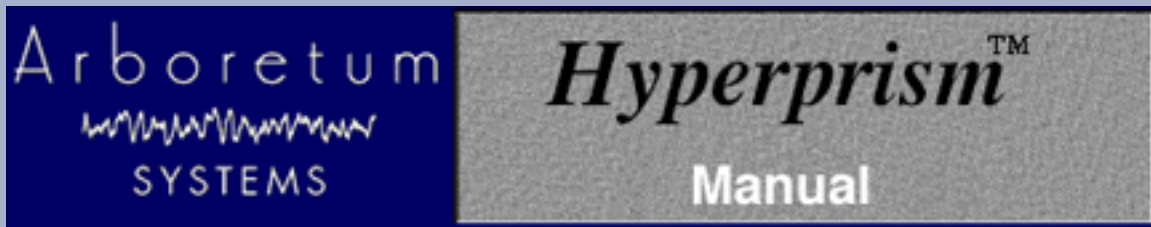
Hyperprism-VST for MacOS VST Plug-in Compatible Applications

Using Hyperprism-VST

- [Controlling Hyperprism](#)
- [Hyperprism-VST and Cubase-VST](#)
- [Hyperprism-VST and Logic Audio](#)
- [Hyperprism-VST and Metro](#)
- [Hyperprism-VST and Vision DSP](#)

Controlling Hyperprism

When you launch a Hyperprism-VST effect you'll see the Hyperprism display, which looks much like the Room Reverb effect below. Note the effect parameter sliders in the top section of the screen, and the Blue Window control area. Logic and Vision DSP add special plug-in buttons to the effect window; their functions are documented later in this chapter. Refer to the effects reference section of the Hyperprism documentation for specifics on using each different effect.

Arboretum
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| | 0 | Feedback (%): | | 100 | 85.306 |
	0	Mix (%):		100	100.0
	0	Diffusion (%):		140	140.0
	0	Brightness (%):		100	11.429

The Blue Window

The main part of Hyperprism-VST's edit screen is the Blue Window. By clicking and dragging your mouse in the Blue Window's "Play Zone," you may adjust two or more effect parameters simultaneously. Some parameters may be manipulated by horizontal mouse motions, others by vertical ones. Change these assignments to suit your working style and the effects you are trying to achieve. The sliders at the top of the screen determine the effect parameters that are controlled by the Blue Window.

The Sliders

The Sliders are used to set effect parameters. They can be individually adjusted directly with the mouse, or they can be set to respond to horizontal or vertical mouse movements in the Blue Window. But how?
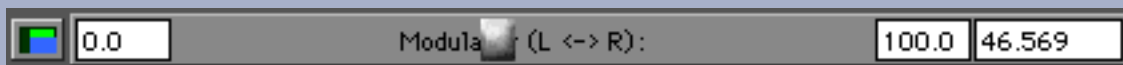
| | 0.0 | Modulate (L <-> R): | 100.0 | 46.569 |

Hyperprism Slider. From left: Axis button, minimum value field, slider throw area and handle, maximum value field and current value field.

☞ Note the small blue button to the left of each parameter slider. Clicking this Axis button will cycle through three possible states, shown by a small green bar. The three choices are vertical control, horizontal control or off.

When the green bar appears horizontally across the top of the Axis button, that slider will be controlled by horizontal mouse movements in the Blue Window.

Clicking the Axis button a second time will cause the green bar to appear vertically at the left side of the blue button and the corresponding slider will be controlled by vertical mouse movements in the Blue Window.

Finally, clicking the Axis button once more will cause the little green bar to disappear. In this case, this slider will be unaffected by movements in the Blue Window.

How might this feature be used? For example, you may wish to control a Reverb by assigning Diffusion and Brightness to horizontal mouse movements while Feedback is assigned to vertical mouse movements. Use this gestural control over multiple parameters, combined with Hyperprism's parameter automation capability, to create effects that change and move in real-time. Design filters which "dance" with the music or effects that morph and change in unusual ways.

Changing Slider Ranges

Sometimes, the default range of values (from one end of a sliders travel to the other) may be larger or smaller than you'd like. If so, you may alter the parameter range by changing the minimum and maximum value numeric fields at either end of each slider. You can change the parameter's current setting by altering the current value numeric field to the far right of the slider.
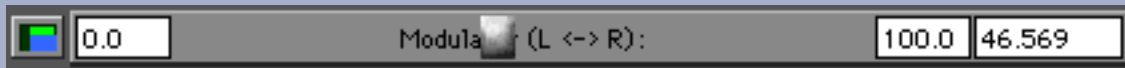
Hyperprism Slider. From left: Axis button, minimum value field, slider throw area and handle, maximum value field and current value field.

Changing the minimum, maximum and current values is easy: click in the numeric field and drag the mouse up to increase the value or down to decrease it. The further away you drag the mouse, the faster the value will change. Hold down the Option (alt) key while mouse dragging to scroll by units of 100.

Stereo File Support

Hyperprism-VST includes a number of effects which require a stereo file for input. Make sure you have your stereo tracks panned 100% left and right, otherwise you will get little or no effect with the Doppler, M-S Matrix, Vocoder and other stereo processes.

Hyperprism-VST and Cubase-VST

Steinberg's Cubase VST was the first program to implement the VST audio plug-in format. In the Spring of 1998 Steinberg opened up the VST spec to third-party developers; Steinberg's competitors have been quick to adopt this framework. The next section describes the basics of getting Hyperprism-VST effects going in Cubase VST 3.5 and Cubase VST/24 4.0.

Effects in Cubase are applied to individual channels, or to the overall stereo mix. Access them from the Effects and Master Effects racks, which can be opened from the Audio menu.

To Add effects to Individual Channels:

Open the Cubase Monitor window and click the "FX" button in the top portion of any channel.

The Channel window will open.

Make sure the effect ON button is illuminated and that the effect send is turned up.

☞ Have your stereo tracks panned 100% left and right or you will get little or no effect with certain stereo processes.

II▶ Choose the Effects rack from the Audio menu. Select any effect from the effects list pop-up menu, on the right-hand side of the display.

II▶ Click the POWER button on and turn the Effects Master knob up part way, as shown here:
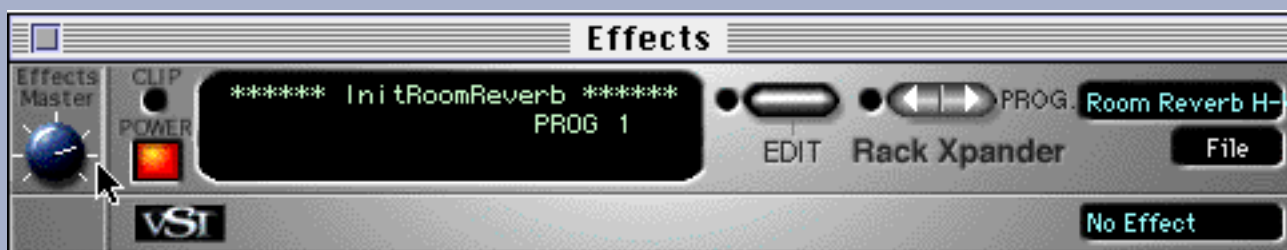
II▶ To change the effect settings (and view the Hyperprism window), click the EDIT button on the effects window.

↘ You should now see the Hyperprism Blue Window (not shown here.)

II▶ Adjust your effect settings as desired while listening to the preview playback.

☞ Even when you close the Hyperprism effect window, each active effect in the rack will remain processing in-line in the background, until the channel is muted or effect is powered off.

Hyperprism-VST and Logic Audio

Logic Audio allows VST effects to be placed on any track insert or the insert of a bus object. Track inserts allow 100% of the signal to go through the effect. Bus inserts allow varying amount of signal to be routed to the effect. Emagic suggests that computation-intensive effects like reverb be used on a bus rather than a track, as well as effects which are to be used on more than one track.

Logic Audio plug-in operations (and everything else involving Logic) have been documented in painstaking detail in the Emagic Logic Audio user's guide. Here's a quick look at how to get Hyperprism-VST started in Logic Audio 3.5:

‖▶ Open the Logic Audio Track Mixer window.

‖▶ Click on the Inserts button of any audio track or bus (below, left.)

‖▶ Select the desired Hyperprism effect (below, center.)

�‣ You may notice that Hyperprism-VST mono effects are showing up in the m/s (mono-to-stereo) group. This is OK, it's simply a characteristic of Logic's VST implementation. The mono Hyperprism effects will process as expected.

‖▶ Release the mouse button to select the effect for insert; the effect name will now show in the Inserts button (below, right.)

Audio 1

EQ
Thru
Inserts

Sends

I/O
Input 1-2

Track 1
0
90
M S
REC
Audio 1
1

EQ
Thru
Inserts

Sends

I/O
Input 1-2

Track 2
0
90
M S
REC
Audio 2
2

H-Auto Pan   (s/s)
H-Band Pass   (s/s)
H-Band Rej   (s/s)
H-Bass Max   (s/s)
H-Chorus   (s/s)
H-Compressor   (s/s)
H-Doppler   (s/s)
H-Echo   (s/s)
H-EchoTranz   (s/s)
H-Flanger   (s/s)
H-FreqShift   (s/s)
H-Hall Reverb   (s/s)
H-Exciter   (s/s)
H-High Pass   (s/s)
H-HyperVerb   (s/s)
H-Limiter   (s/s)
H-Low Pass   (s/s)
H-MSMatrix   (s/s)
H-MoreStereo   (s/s)
H-MultiDel   (s/s)
H-NoiseGate   (s/s)
H-Pan   (s/s)
H-Phaser   (s/s)
H-PitchChanger   (s/s)
H-QuasiStereo   (s/s)
H-RingMod   (s/s)
H-Room Reverb   (s/s)
H-Single Del   (s/s)
H-Sonic Dec   (s/s)
H-Stereo Dyn   (s/s)
H-Tremolo   (s/s)
H-Saturator   (s/s)
H-Vibrato   (s/s)
H-Vocoder   (s/s)
DQualificater   (s/s)

Audio 1

EQ
Thru
Inserts
H-HyperV

Sends

I/O
Input 1-2

Track 1
0
90
M S
REC
Audio 1
1

EQ
Thru
Inserts

Sends

I/O
Input 1-2

Track 2
0
90
M S
REC
Audio 2
2

From left to right: In the Logic Track Mixer, click-and-hold on any insert point (left) to select from the VST plug-ins list (center.) Selected effect name is then visible on the Insert Button (right.)

▌▌➤ Double-click on the Inserts button to edit the current effect.

➘ You should now see the Hyperprism Blue Window (below):

Untitled:Plug-In

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Room Size:
Large Room

Early Reflection Spread:
70 ms

| | | Pre-Delay (ms) | 500 | 100 |
	0	Diffusion (%)	100	33
	2000	Brightness (Hz)	20000	20000
	0	Reverb Time (s)	50	1.0
	0	Early /Late Mix (%)	100	40
	0	Master Mix (%)	100	80

Bypass ▼ Track 2 | H-HyperVe (s/s)

Hyperprism-VST effect window, as it appears in Logic Audio.

▌▶ Adjust your effect settings as desired while listening to the preview playback.

☞ Even when you close the Hyperprism effect window, each Inserted effect will remain active, processing in-line in the background, until the channel is muted or effect is de-selected.

Logic Plug-in Controls

Every Logic Plug-in window includes a set of switches which function the same in all plug-ins. These buttons are described in greater detail in the Logic manual; here's a quick summary of their functions (from left to right:)

Bypass ▼ Track 2 | H-HyperVe (s/s)
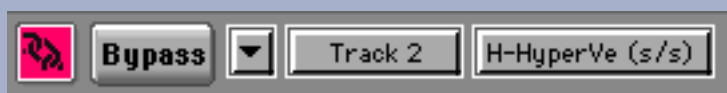
Link: Switch off to open several plug-in windows simultaneously. Switch on to display all double-clicked plug-ins.

Bypass: Temporarily removes the effect's process from the signal chain.

Arrow: Used to access plug-in presets.

Audio Object: Use this switch to swap between tracks so as to compare or copy settings

Insert Slot Selection: Pull up other effects applied to the same audio object.

☞ **No Presets:** Logic Audio 3.5.3 is unable to recognize Hyperprism-VST effect presets. It also will not create or save presets. This appears to be a shortcoming of Emagic's implementation of the VST plug-in format. We're communicating with the Logic team about this issue, and are hopeful that a fix may be available shortly.

Hyperprism-VST and Metro

VST plug-ins, as implemented in Cakewalk Metro 4, are fairly straightforward to access and use. Note that Hyperprism-VST and Ray Gun are only compatible with Metro version 4.5.1 or later.

▐▐▶ Before accessing VST plug-ins you must make sure you have at least one audio file loaded in your current Metro session; the Effects window will be disabled if no audio is present.
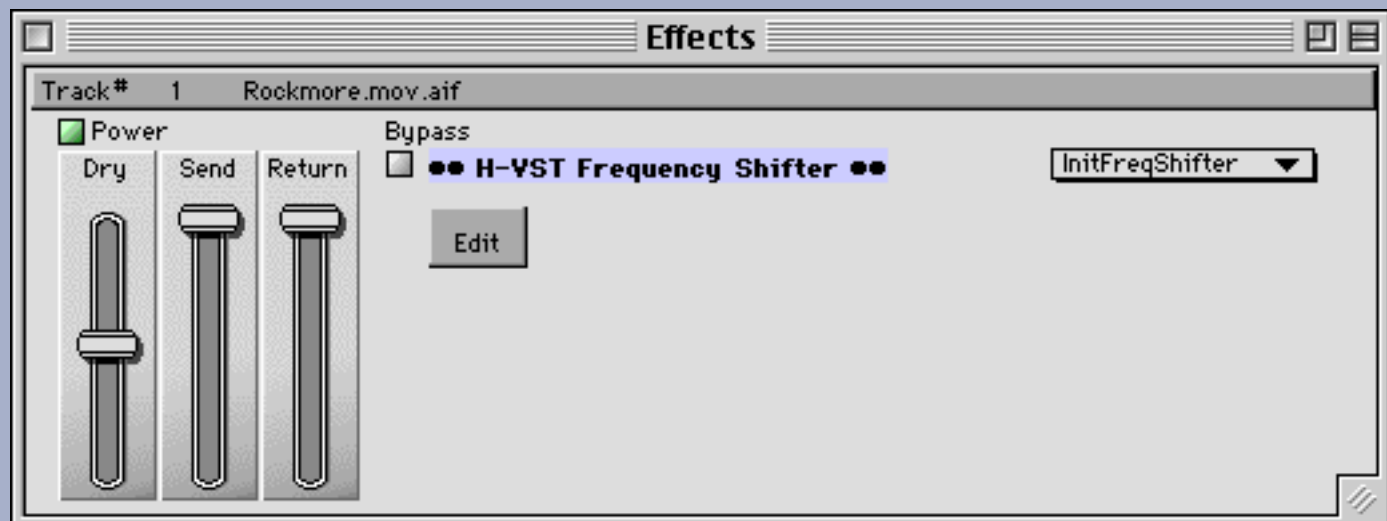
▐▐▶ Open the Plug-ins Window by selecting it from the Windows Menu (keyboard shortcut is Command-9)

▐▐▶ You'll see the list of available real-time effects. Scroll down the window to view the entire list.

▐▐▶ Select the desired process by double-clicking on the effect name.

➤ The Metro Effects window appears on screen:

Metro Effects Window Controls

The Metro Effects window includes a number of controls which function the same in all plug-ins. These features are described in greater detail in the Metro manual; here's a quick summary of their functions:

Track Assignment: Displays/selects the track to be processed by the current effect chain.

Power: Turns on/off the effect. Power off effects when not auditioning them to conserve CPU headroom.

Dry: Adjusts the amount of dry (un-processed signal) to be sent to the audio track's pan and volume controls (and corresponding output bus.)

Send: Determines the amount of dry signal to be sent to the first plug-in. Think of this setting as "effect input volume."

Return: Controls the amount of wet (processed) signal to be sent to the audio track's pan and volume controls (and corresponding output bus.)

Bypass: Temporarily removes the current effect from the signal chain.

Effect Name: Displays the name of the selected plug-in. Click and drag the name field to change the order of effects in the signal chain; select and clear to remove an effect from the path.

Preset Menu: Save and Recall Hyperprism-VST effect settings. Be sure to see the note on Preset File incompatibility, below.

Edit: Click on the Edit button to display and adjust the Hyperprism-VST Blue Window and sliders.

▌▌▶ When you click on the Edit button, Hyperprism-VST's effect display will open, as shown below:
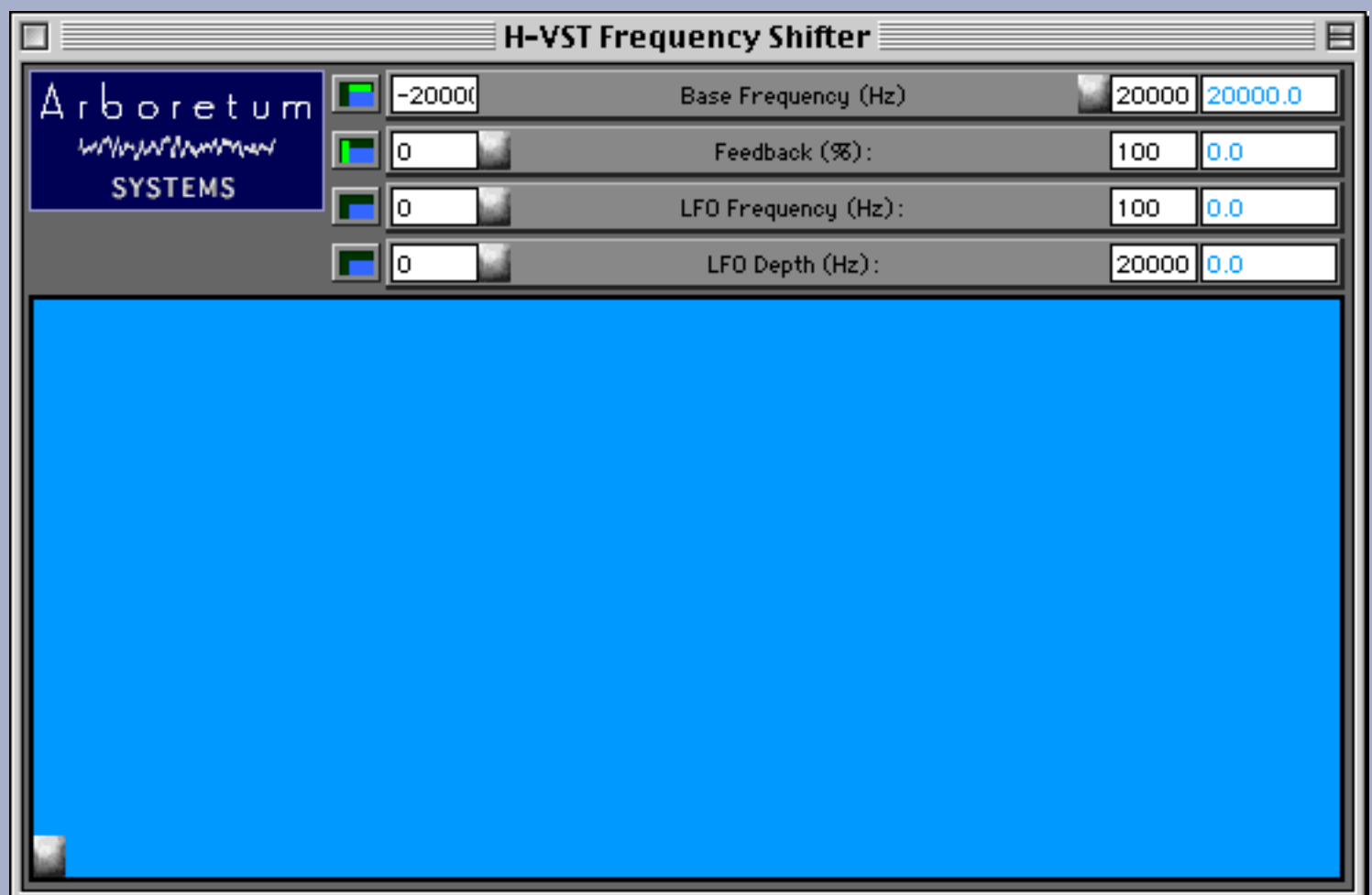
⏸▶ Adjust the Hyperprism settings while listening to the preview playback until you've created an effect which is to your liking.
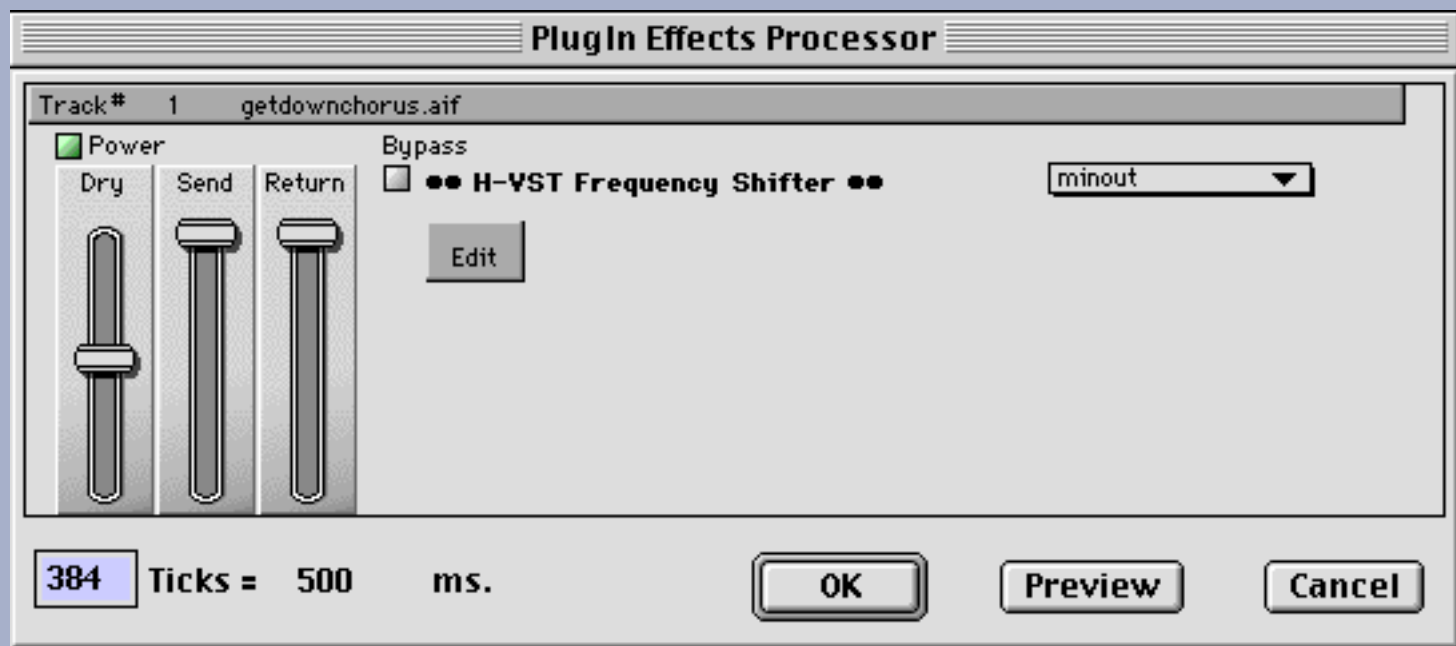
⏸▶ Close the Hyperprism effect display; effect processing is still active in the background as long as the effect is enabled in the Metro Effects window.

One of the core shortcomings of the VST plug-in format is the fact that each active effect taxes the CPU. If you're comfortable with your effect settings and sound design decisions on a particular sound file or track, you can process the effect to file. This will create a new audio file which incorporates the Hyperprism processing. Note that if you change your mind later, you can always go back and restore the original sound file. To process an effect to file:

⏸▶ Select the audio file or region to be processed in the Metro waveform display.

⏸▶ Select the effect name from the Edit menu's Audio submenu.

🔍 A Metro Effects window will be displayed. This window is very similar to the one used for real-time processing, with the addition of controls for previewing, processing and PPQN setting (not used here).

⏸▶ Click on Preview to audition the sound.

⏸▶ Click on the Edit button to display and adjust the Hyperprism effect display.

⏸▶ When you're happy with the sound click on OK to process.

🔍 The Cancel button allows you to exit the Effects dialog without calculating the effect and processing to file.

👉 **Incompatible Presets:** Metro 4.5.1 is unable to recognize the original Hyperprism-VST effect presets which were created for Cubase VST. Metro has its own unique preset file format, which is incompatible with Cubase. We've gone ahead and re-created the "factory" presets for your use in Metro. The only consideration here is that you will not be able to use Hyperprism-VST presets which were created in Cubase (or Vision DSP) when running Metro (and vice-versa); you'll have to re-make by hand any custom settings you may have developed on one platform for the alternate system.

☞ **No Parameter Automation:** Unlike Cubase and Logic, Metro 4.5.1 has no provision for recording, replaying or editing Hyperprism parameter changes and Blue Window control gestures. We're in close contact with Metro's creator about this topic and although no Metro support for parameter automation has been announced yet, we feel confident that a solution will eventually follow.


Hyperprism-VST and Vision DSP

Opcode debuted VST plug-in support in the new Vision DSP package. Opcode's earlier Vision, Studio Vision and Studio Vision Pro packages accepted Premiere (and TDM) format plug-ins. During the development of Hyperprism-VST 1.5.5 we tested with beta releases of Vision DSP 4.1.1.

Opcode's implementation of VST plug-ins requires a few extra steps to get things going; we'll outline just the basic procedure here and refer all in-depth inquiries regarding Vision DSP features and tools to the Opcode Vision DSP user's guide.

▐▶ Open your audio session, or import a piece of audio to be processed into the current document.

▐▶ Under the Windows menu, select a Console from the Consoles submenu. Here we've selected Console 1:
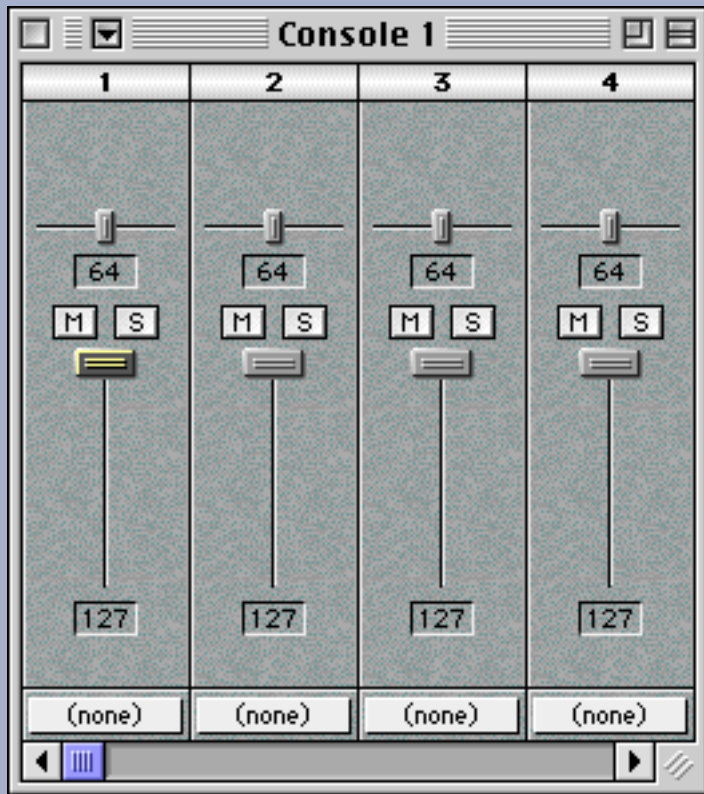
Vision DSP Console window

➤ Notice the little down-arrow button, located in the left-hand part of the Console window title bar, next to the "Close" tool:

Console title bar, note down-arrow button (left side)

▐▶ Mouse click on the down arrow button to reveal the Console pop-up menu (not shown here.)
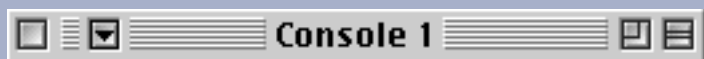
▐▶ While holding down the mouse button, drag to the Plug-ins option and select the number of plug-ins you wish to allow per track, up to a maximum of four.

➘ Now that you've set up the Console to accept plug-ins, direct your attention to the text field at the bottom of each Console fader. This field is labeled as (none) below:
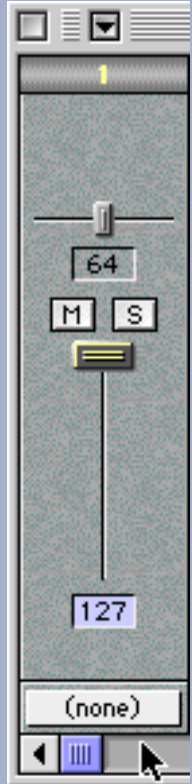
Channel input/output selector (bottom)

▐▶ Mouse clicking on the (none) text field reveals a pop-up menu where you select channel input and output.

▐▶ Click and drag in this menu (not shown) to assign Audio (input) and Output choices. For the sake of this example we're assigning the fader to Audio 1, and directing output to Sound Manager.

➘ Once this patching is complete, a new channel insert control appears at the top of the fader strip (see below):
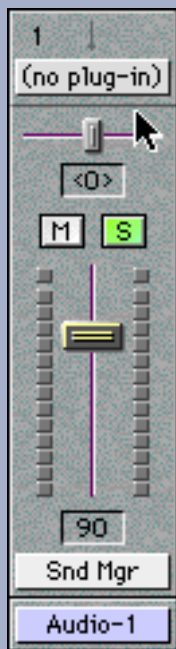
Input and Output selected, fader is now ready for plug-ins; note insert point, labeled (no plug-in).

▌▌► Click on the channel insert button (labeled "no plug-in" in the above picture) and hold down the mouse button to view your available VST plug-ins.

➘ If you've just installed all 30+ Hyperprism-VST effects, it'll be a pretty massive list (not shown here.)

▌▌► Let go of the mouse button to select any effect.

➘ The pop up menu will disappear and the chosen effect's name will now be visible in the insert display.
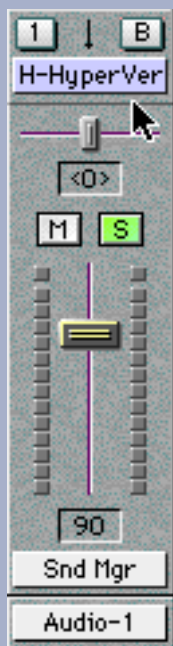
Effect is now inserted in channel, note name of effect is displayed (and truncated).

➘ You should now be able to hear your effect; the Hyperprism-VST processor is now active in-line.

⏸▶ To bypass the effect, click on the "B" button above the insert point.

↘ Note the down arrow, located top and center on the fader strip. This arrow indicates the type of channel processing (mono to mono, mono to stereo or stereo to stereo.)

⏸▶ To adjust the Hyperprism settings, click on the "1" button above the insert display.

↘ You will now be viewing the Hyperprism-VST window.
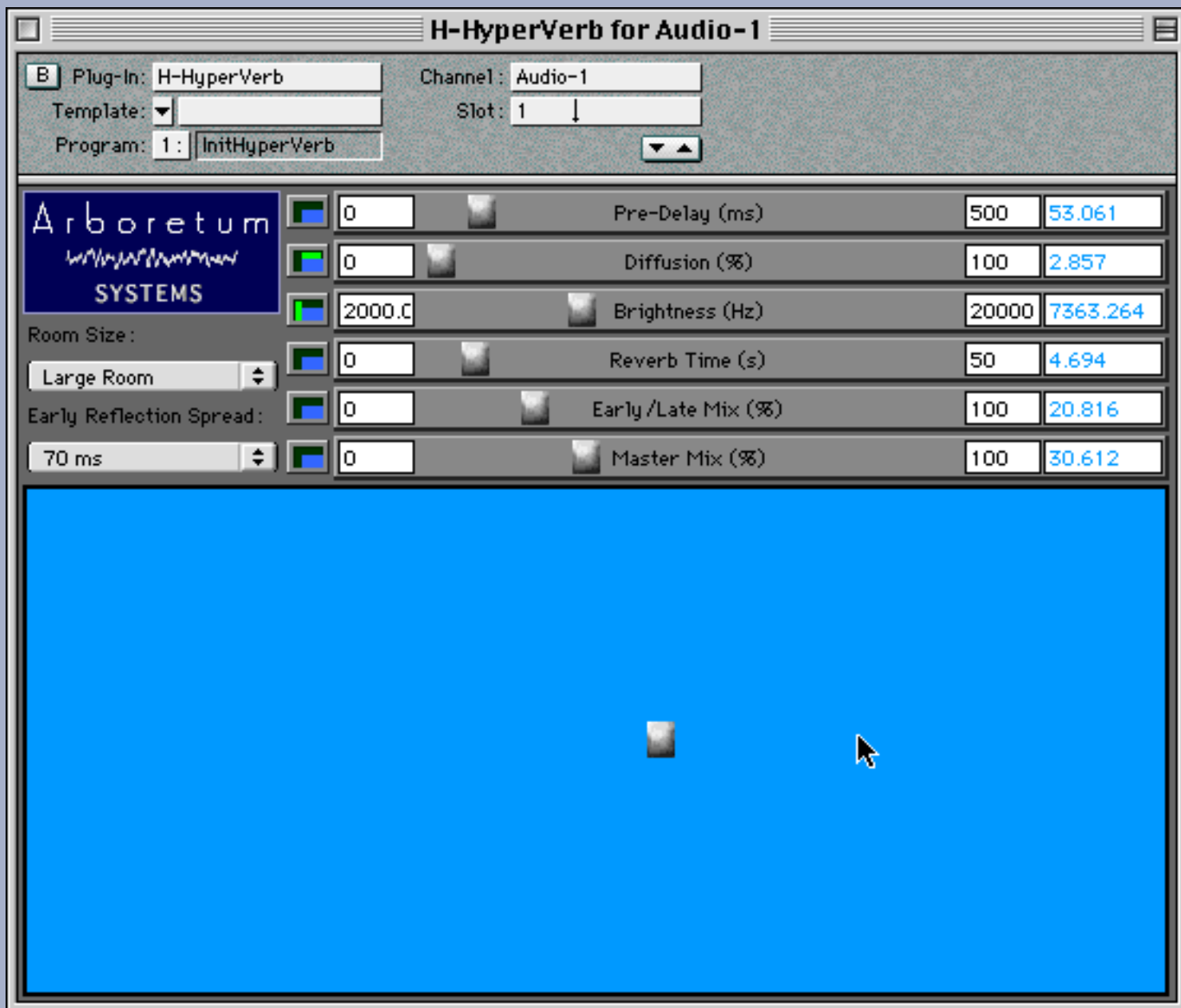
Hyperprism-VST HyperVerb window, as it appears in Vision DSP

⏸▶ Adjust the Hyperprism settings until you find a combination you like, then simply close the window.

↘ The effect will remain active in the background, processing your audio in-line, in real time.

Vision DSP Plug-in Window Controls

Vision DSP plug-in window tool bar

The Vision DSP Plug-in window includes a number of controls which function the same in all plug-ins. These features are described in greater detail in the Vision DSP manual; here's a quick summary of their functions:

Bypass (B): Temporarily removes the plug-in from the signal chain, so you can compare before-and-after results.

Plug-In: Select from any of your available VST plug-ins from this pop up menu, which duplicates the function of the insert display and menu in the Console.
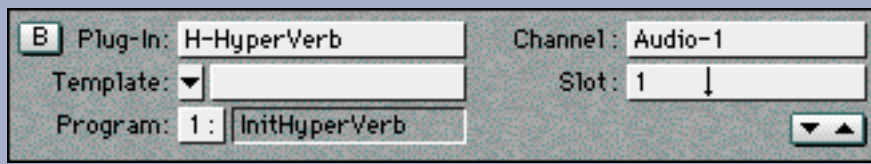
Template: Use this pop-up menu to locate, load and save Hyperprism-VST Presets and Preset Banks.

Program: Select any Preset from the current Bank.

Channel: Select input and output assignments for the current plug-in and console channel. Duplicates the function of the channel input/output selector in the Console display.

Slot: If you're inserting multiple effects into a single track, the Slot display allows you to view their relative position in the signal chain, and also indicates channel processing status (mono to mono, mono to stereo, stereo to stereo).

Tool Bar Locator (up and down arrow button): Allows you to specify whether this set of plug-in window controls appear at the top or bottom of the screen. Clicking this button and moving the tool bar will seemingly "zero out" your current effect settings, however all is not lost! Simply click anywhere in the Blue Window to restore your current settings after a tool bar location change.

Mono to Stereo Effects: Certain Hyperprism effects which are intended for mono to stereo processing (such as the Auto Pan or Quasi Stereo) show up in the insert effects pop-up menu labeled under the mono to mono section. This display anomaly is a quirk of Vision DSP. Selecting a mono to stereo effect from the mono to mono grouping may result in faulty processing or a performance interruption. To do mono to stereo work be sure to pick the desired effect from the mono to stereo list.

No Parameter Automation: Unlike Cubase and Logic, Vision DSP 4.1.1 appears to have no provision for recording, replaying or editing Hyperprism parameter changes and Blue Window control gestures. We're in direct contact with Opcode Systems about this topic and although there has been no announcement of parameter automation support at the time of this writing (11/11/98) we remain hopeful.

This concludes our coverage of basic Hyperprism-VST operations. In the following chapters we'll examine Hyperprism-VST presets and parameter automation control.

[Last Chapter] [Table of Contents] [Next Chapter]

Hyperprism-VST for MacOS VST Plug-in Compatible Applications

Hyperprism-VST Presets

Presets let you store all the parameter settings of the current effect so you can restore them later or apply them to other files. Over time you'll create a library of custom effects settings that you can reuse at will.

Hyperprism-VST includes a number of factory presets. The Hyperprism-VST Installer creates a folder named Hyperprism VST Presets, and places these factory patches in the folder location deemed most appropriate by the manufacturer of your audio editing software. This will generally be inside the program's main folder, either at the same directory level as the host application itself or within the VST Plug-ins folder. The Presets are in separate sub-folders, grouped by Hyperprism effect type.

In the VST plug-in format effect presets are handled by the "host" application. Each program which supports Hyperprism-VST does presets a little differently. Preset implementations generally allow one to **same** and **name** the current settings, **recall** a previously-saved setting or factory patch, **group** the presets into libraries or Banks, and **delete** any preset from a bank. These functions may be named and organized slightly differently among the various programs; fortunately it's all fairly self-evident.

The VST plug-in spec does not put any limitation on where preset files may reside; presets can be stored in folders or volumes far removed from the host application. When loading a preset or bank you may have to do some navigating and to locate your presets. It's a smart practice to always keep them in one place, have them clearly named and keep them grouped according to the type of effect they serve.

There's also nothing to prevent you from loading a preset file into the wrong effect type. One could carelessly open a Vocoder setting in a HyperVerb effect and unfortunately the result is not good. An error message or other performance interruption will probably result. Therefore, organize your presets and navigate carefully through the preset "open" and "save" dialogs.

☞ **Logic Audio Platinum version 3.5.3** doesn't appear to save effect presets, and does not recognize the original Cubase-VST preset files. We're in contact with the Logic team about this issue and hope to reach an expedient resolution.

☞ **Metro 4.5.1** is unable to recognize the original Hyperprism-VST effect presets which were created for Cubase VST. Metro has its own unique preset file format, which is incompatible with Cubase. We've gone ahead and re-created the "factory" presets for your use in Metro. The only consideration here is that you will not be able to use Hyperprism-VST presets which were created in Cubase (or Vision DSP) when running Metro (and vice-versa); you'll have to re-make by hand any custom settings you may have developed on one platform for the alternate system.

Using effect presets is naturally described in some detail in the documentation that accompanied your audio editing software. Here's how to get started with effect presets in **Cubase-VST**:

▐▶ To load a preset, click the File area at the lower right of the Effects window.

▐▶ Choose **Load Effect** from the pop-up menu.

↘ A file-open dialog box will appear

▐▶ Navigate to the folder containing the Hyperprism-VST presets.

☞ IMPORTANT: Make sure you open the effect presets folder who's name matches the effect type you currently have open! For example, if you have the Chorus effect active, and try to open a Reverb preset, you'll get an error message.

▐▶ Choose the effect you want to open and click OK. The effects settings will be loaded.

Hyperprism-VST for MacOS VST Plug-in Compatible Applications

Hyperprism-VST Parameter Automation

- [Recording Blue Window gestures in Cubase-VST](#)
- [Recording Blue Window gestures in Logic Audio](#)

Cubase-VST includes a mix automation feature that makes it possible to record your Blue Window gestures in any Hyperprism-VST effect. Use this feature to create effects changes that are synchronized with elements of your music. Create filters that dance to the beat or unusual effects which change and morph over time. Hyperprism was the first sound design software to offer real-time parameter change automation, way back in the Mac II / Sound Tools days, and the Blue Window is still going strong in Hyperprism-VST.

As with many other aspects of VST plug-in implementation, each VST-compliant sound editing program deals with effect parameter automation in a different way. In this chapter we'll cover the steps necessary to create Hyperprism parameter automation in Cubase-VST, and take a look at the corresponding features in Logic Audio.

Metro 4.5.1 and Vision DSP 4.1.1b3 have not implemented parameter change automation as of 11/11/98. We're presently consulting with the relevant parties and anticipate that eventually all professional sound editing programs will support parameter change automation.

Recording Blue Window gestures in Cubase-VST:

Set up the effect you want to automate following the directions in the [Using Hyperprism-VST](#) section of this documentation.

Open the Monitor window using the Audio menu.

▸ Click the Write button as shown:

▸ Open the Blue Window of the effect you want to automate

▶ Begin playing your song.

▶ As the song plays, use the Blue Window or individual effect parameter sliders to alter the effect settings in time with the music.

▶ When you are done, stop playback.

▶ Return to the Monitor window, and click the Read button as shown:

▶ Play your song again.

↘ Notice that the effect parameter moves you made during the "write" pass are dynamically recreated during playback.

☞ Refer to your Cubase-VST documentation for further information on effects and mix down automation.
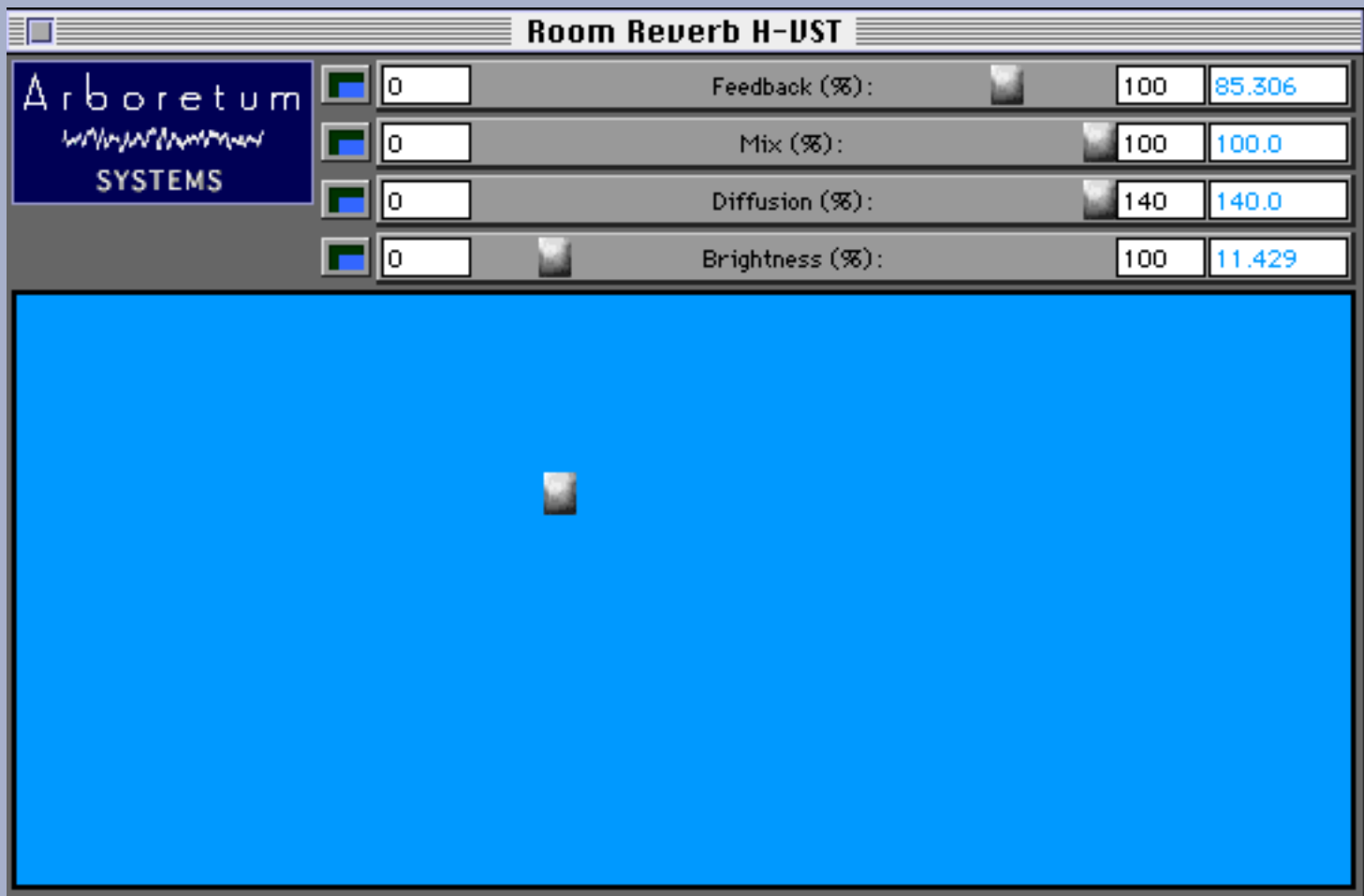

Recording Blue Window gestures in Logic Audio:

▐▐▶ Set up the effect you want to automate following the directions in the [Using Hyperprism-VST](#) section of this documentation.

▐▐▶ In the Logic Audio Arrange window click on the A - Playback track, as shown below:
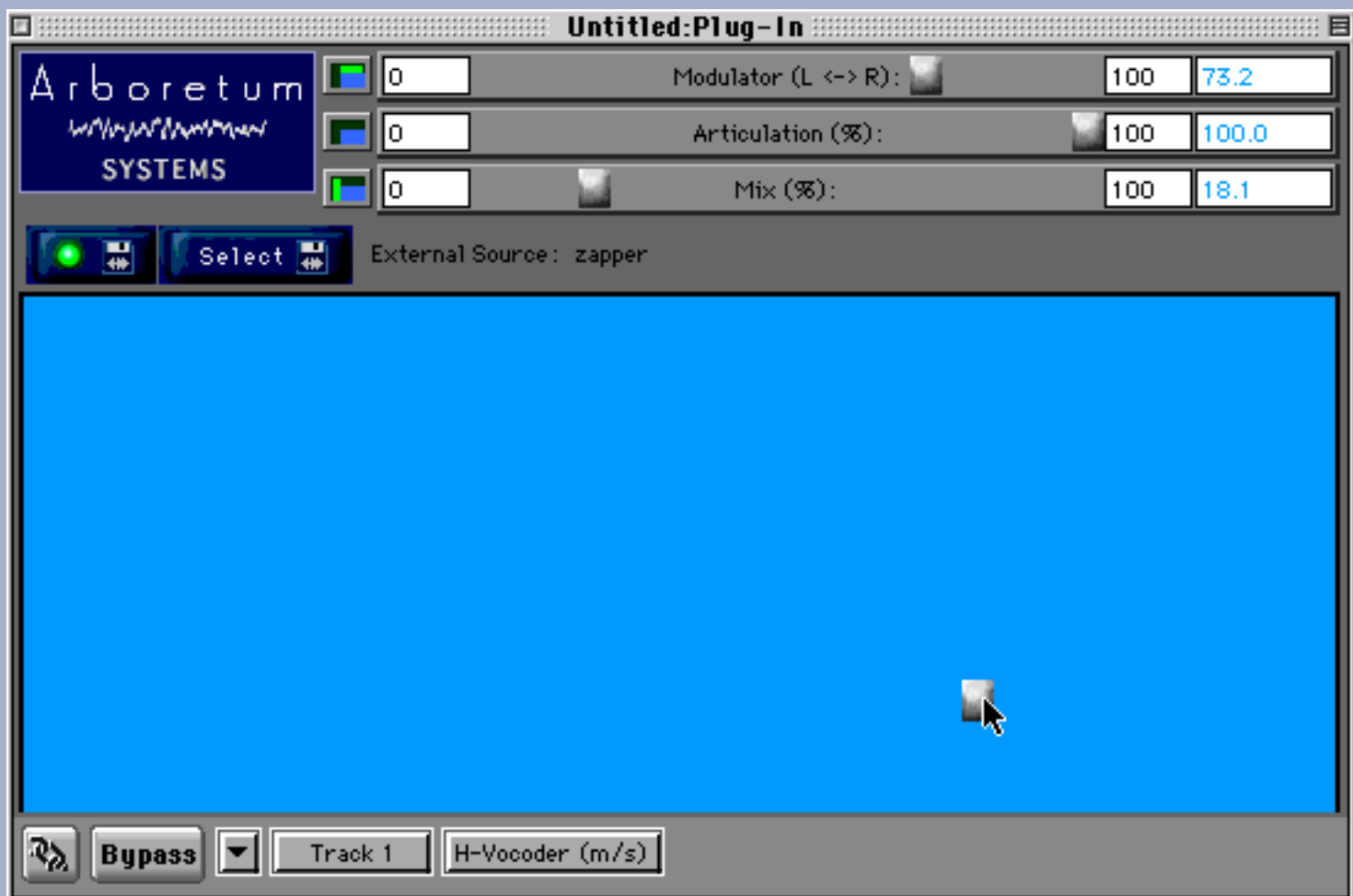

▐▐▶ Record-enable the transport:

▐▐▶ Open the Logic Audio Track Mixer window.

▐▐▶ Double-click on the effect name at the track insert point to open the Hyperprism display, as shown below:

▌▶ Click off Pause to begin recording:

▌▶ Move the cursor around the Blue Window, or move sliders individually, to create dynamic effect changes, settings which morph and evolve over time.

Your parameter automation gestures are being recorded. These slider and Blue Window movements are saved by Logic Audio as MIDI commands.

▐▐▶ When finished, click on stop or pause to halt the automation recording.

▐▐▶ Rewind and play to hear/view the results

↘ Notice that the effect parameter moves you made during the "write" pass are dynamically recreated during playback.

↘ Automation events are recorded to the A-Playback track in the Logic Audio Arrange window.
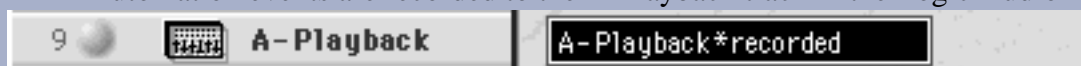
☞ See the Emagic Logic Audio manual for complete information on automation editing, MIDI editing in general and the Logic Hyperdraw features.

Conclusion

The next sections of this manual consist of the process reference chapters, troubleshooting guide and appendices. Each Hyperprism-VST effect is described in detail in the following pages, and the coverage also extends to the other MacOS Hyperprism versions in the Hyperprism Plug-in Pack, as well as the Hyperprism 2 "stand alone" edition for HyperEngine.

In leaving the VST specific sections of this guide, it's worth reflecting on the fact that industry-wide VST support is a very recent and somewhat unusual development. Although some of the rough edges on the third-party implementations are still

being honed down, it's a remarkable thing to see such a disparate, competitive and strong-minded group as Opcode, Emagic, Cakewalk and Emu/Ensoniq all agreeing on a single format, and a format which their esteemed competitor Steinberg controls, at that. Although there may yet be some die-hard hold outs among the remaining key players, it's evident that MacOS audio developers, by and large, are willing to work together (to a degree) and can agree on some "open" formats. As we saw a decade ago with the MIDI spec, which helped get this whole computer audio thing started. Anyway, making real-time native-processing DSP effects more widely available at a lower cost is a fine thing in our humble opinion. We hope you agree, and we encourage your comments, questions and suggestions.

Special Arboretum thanks to Ray Lenini, Paris product manager at Emu/Ensoniq, Jeremy Sagan, the legendary creator of Metro, Mark Gavini, music and games evangelist at Apple and the Opcode QA team.

Hyperprism Processes

This chapter contains a detailed description of every Hyperprism effect. Depending on which hardware platform and/or host application you use, you may not have every effect listed in this section. This is often because certain host-programs' plug-in architectures may not currently support certain types of processing, for example, effects that change the duration of a sound.

A series of small icons displayed in each effect description indicates which version of Hyperprism that effect is included in.

H 2 Effect included in Hyperprism-2

VST Effect included in Hyperprism-VST (For the MacOS VST plug-in format)

MMP G Effect included in Hyperprism-MMP (Adobe Premiere-compatible plug-in format)

DAS Effect included in Hyperprism-DAS (Digidesign Audio Suite plug-in format)

Hyperprism-TDM and Hyperprism-DX (Windows 95) processes are documented elsewhere, in the special manuals for those specific editions of Hyperprism.

Hyperprism Filter Processes

- Low Pass
- High Pass
- Band Pass
- Band Reject
- NEW! Low Shelf
- NEW! Hi Shelf
- NEW! Parametric EQ

Hyperprism Modulation Processes

- NEW! Vocoder
- NEW! Frequency Shifter
- NEW! Z-Morph
- Phaser
- Flanger
- Chorus
- Ring Modulator

- Tremolo
- Vibrato

Hyperprism Delay and Reverb Processes

- Single Delay
- Multi Delay
- Echo
- EchoTranz
- NEW! HyperVerb (TM)
- Room Reverb
- Hall Reverb

Hyperprism Stereo Processes

- Pan
- Auto Pan
- Quasi Stereo
- Stereo Dynamics
- More Stereo
- M-S Matrix

Hyperprism Miscellaneous Processes

- NEW! Harmonic Exciter
- NEW! Bass Maximizer
- NEW! Tube/Tape Saturation
- Pitch Time Changer
- Pitch Changer
- NEW! Vari-Speed
- Noise Gate
- Compressor
- NEW! Limiter
- Sonic Decimator

Go to Effects: [Filters] [Modulators] [Delay and Reverb] [Stereo] [Misc Effects]
Go to Table of Contents: [Hyperprism-VST]

[Troubleshooting] [About Digital Audio]

Hyperprism Filter Processes

- Low Shelf
- High Shelf
- Parametric EQ
- Low Pass
- High Pass
- Band Pass
- Band Reject

Low Shelf

`H 2`

The Low Shelf filter boosts or cuts frequencies below the cutoff, and passes frequencies above the shelf cutoff with no change made to their gain. Use this effect to enhance or diminish any amount of low frequency material in a sound.
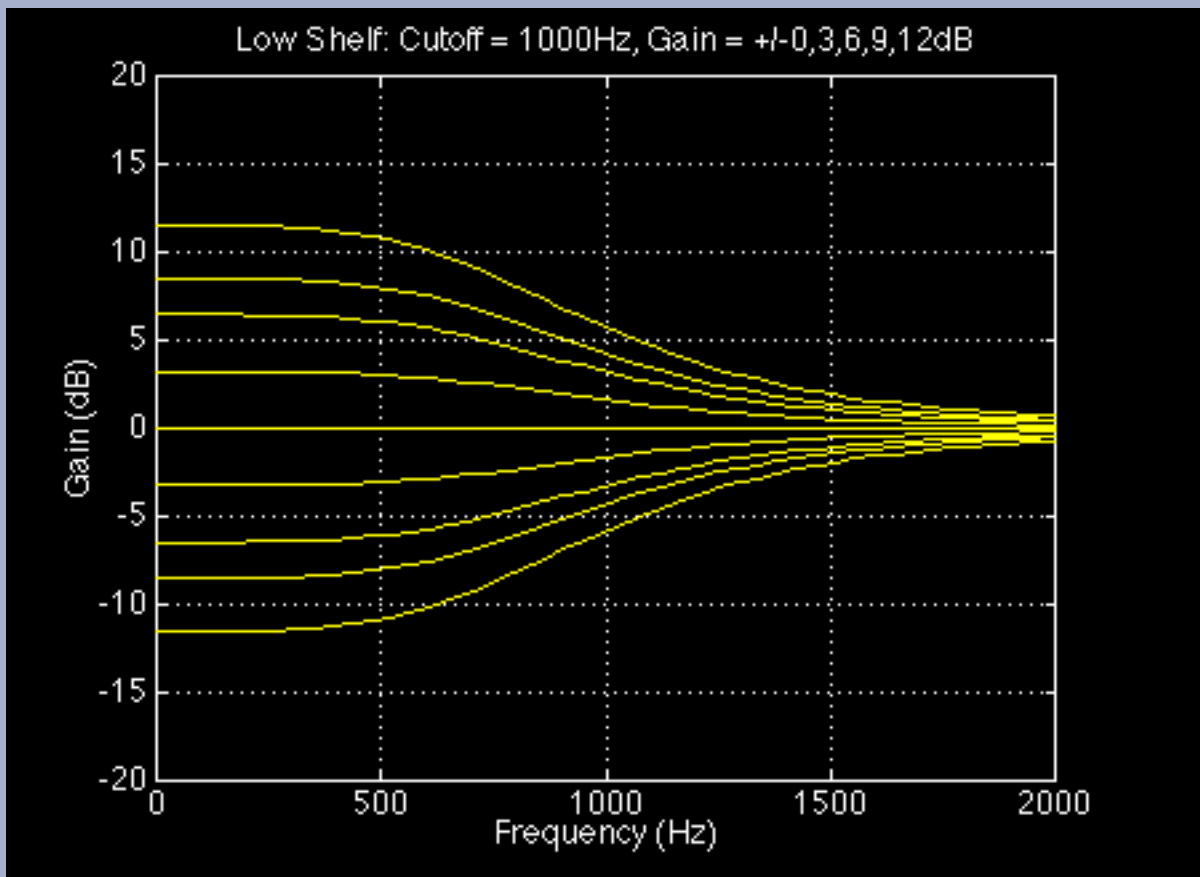
Low Shelf filter response characteristics at various gain levels.

Parameters

Cutoff Frequency

Ranges from 13-20,000 Hz. Indicates the midpoint of the shelf curve. The 3 dB points are roughly at twice and half the midpoint.

Gain

Ranges from -36 to +36 dB. Determines the gain applied to the shelved frequencies.

High Shelf

`H 2`

The high shelf filter boosts or cuts frequencies above the cutoff, and passes frequencies below the shelf cutoff with no change made to their gain. Use this effect to enhance or diminish any adjustable amount of high frequency material in a sound.
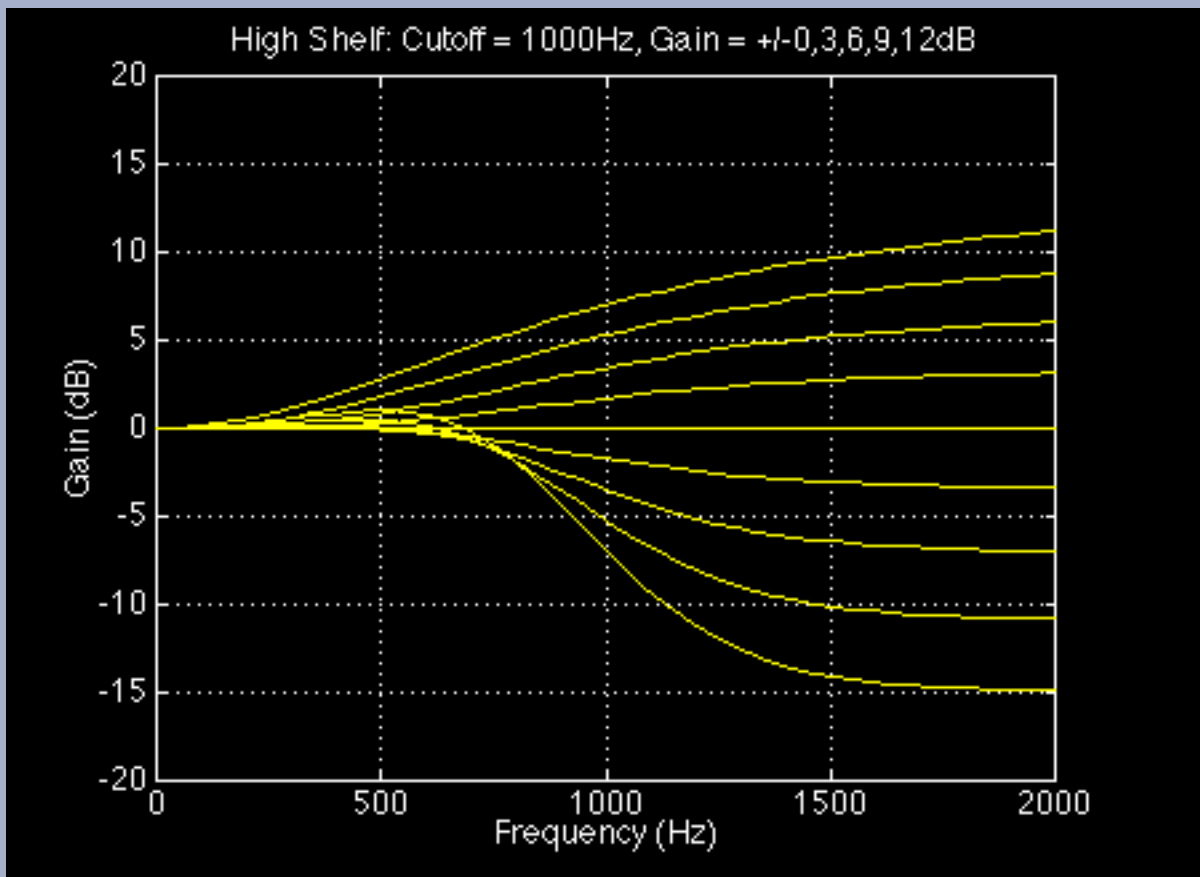
High Shelf filter response characteristics at various gain levels.

Parameters

Cutoff

Ranges from 13-20,000 Hz. Indicates the midpoint of the shelf curve. The +/- 3 dB points are roughly at twice and half the midpoint.

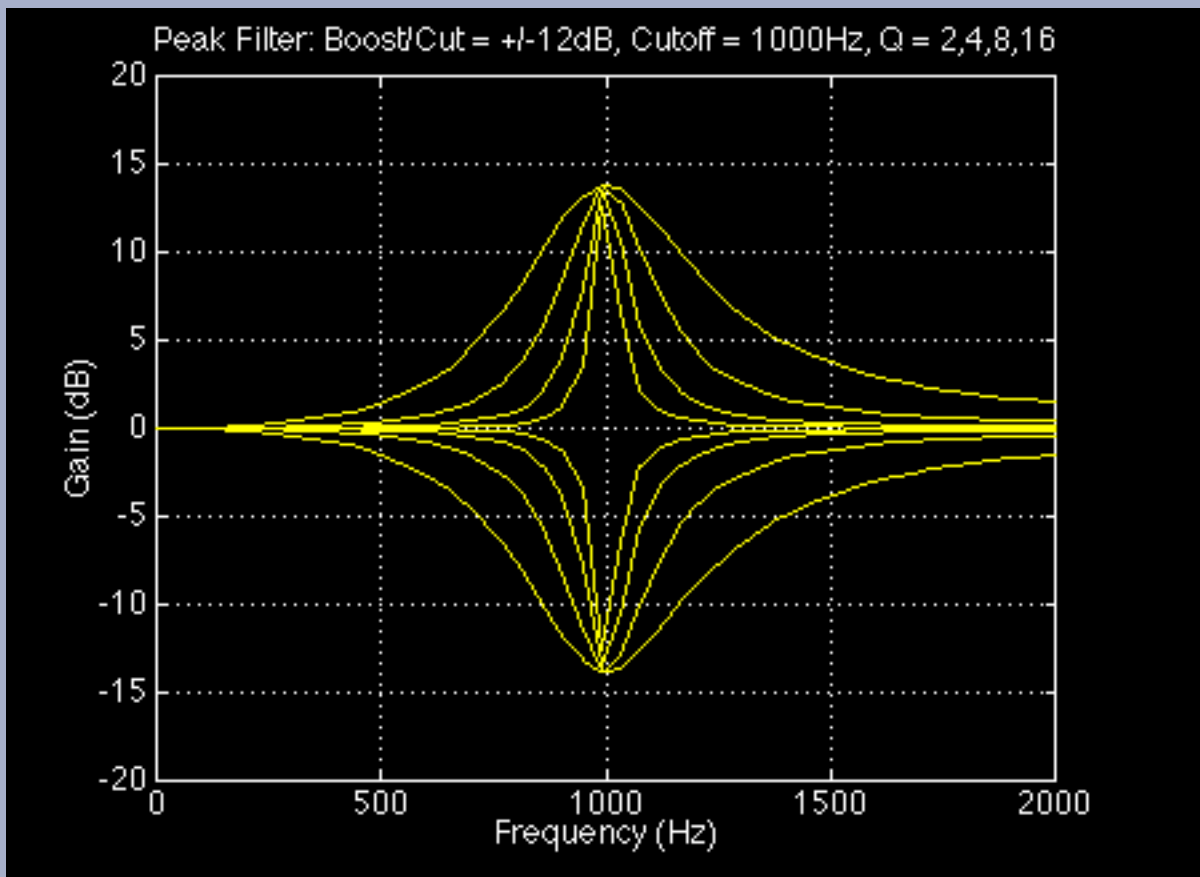**Gain**

Ranges from -36 to +36 dB. Determines the gain applied to the shelved frequencies.

Parametric EQ

H 2

This process is a one band parametric equalizer. It's a very flexible filter, capable of creating notches or peaks without influencing nearby frequencies. Select a center frequency to process, the amount of gain change and the bandwidth of the filter.

Parametric EQ response characteristics at various gain levels.

Parameters

Center Frequency

Sets the frequency to be boosted or cut, from 13-20,000 Hz.

Q Factor

Ranges from 1-100. Determines the width of the peak/notch. Low Q's result in wide peaks/notches, and high Q's result in sharp, resonant peaks/notches.

Boost/Cut

Controls the amount of gain applied to the selected frequencies, ranging from -36 to +36 dB.

Low Pass

H 2   VST   MMP G   DAS

A low pass filter lets you hear only low frequencies; it blocks higher frequencies. An important property of a low pass filter is its cutoff frequency. This is the point in the frequency range at which the filter reduces the signal to -3 dB of its original power. (In visual terms, a -3 dB reduction in a waveform displayed on your monitor will result in a signal that appears to be half the peak-to-peak height of the original.)

Spectral components that are attenuated below the cutoff of a filter are said to be in the stop band of a filter. Those above the half-power point are said to be in the pass band of the filter.

In an ideally sharp filter, the cutoff frequency is a kind of brick wall: anything outside it is maximally attenuated, dividing the frequency response neatly into a pass band and a stop band. In actual filters, the slope of the filter leading up to the cutoff frequency is not linear (there is a slight ripple in the frequency response), and the area between the pass band and the stop band is called the transition band.

The steepness of a filter's slope is usually specified in terms of decibels of attenuation or boost per octave, abbreviated "dB/octave." For example, a 6 dB/octave slope on a low pass filter makes a smooth attenuation (or roll-off), while a 90 dB/octave slope makes a sharp cutoff. The use of a smooth or sharp slope depends on the musical situation. For example, a sharp band reject filter might be needed to completely eliminate a tone centered at a particular frequency, while a gentle low pass filter could be the most unobtrusive way of attenuating background noise in the high frequency range.

☞ One of the shortcomings of early Hyperprism versions has been the ease with which you could overdrive the filters with extreme cutoff frequency settings. Among the many improvements we've made in Hyperprism is that we're now normalizing the output of our filter effects so the output stays smooth and even (and listenable) across the entire range of cutoff frequency settings.

Parameters

Frequency

Frequency scale, from 13 Hz to 20,000 Hz. Select the cutoff frequency within this band. This parameter is called Cutoff Frequency in Hyperprism 2 and Hyperprism-VST.

Q Factor

The Q sets the sharpness of the response of the filter, determining its +/- 3 dB band width and degree of resonance. The greater the Q, the sharper the filter. When the Q is high, the frequency response is sharply focused around a peak (resonant) frequency. A high Q setting will tend to reduce the total signal level coming out of the filter. Ranges from 1 to 100 in Hyperprism 2, Hyperprism-DAS and Hyperprism-VST, from 0.001 to 10 in Hyperprism-MMP
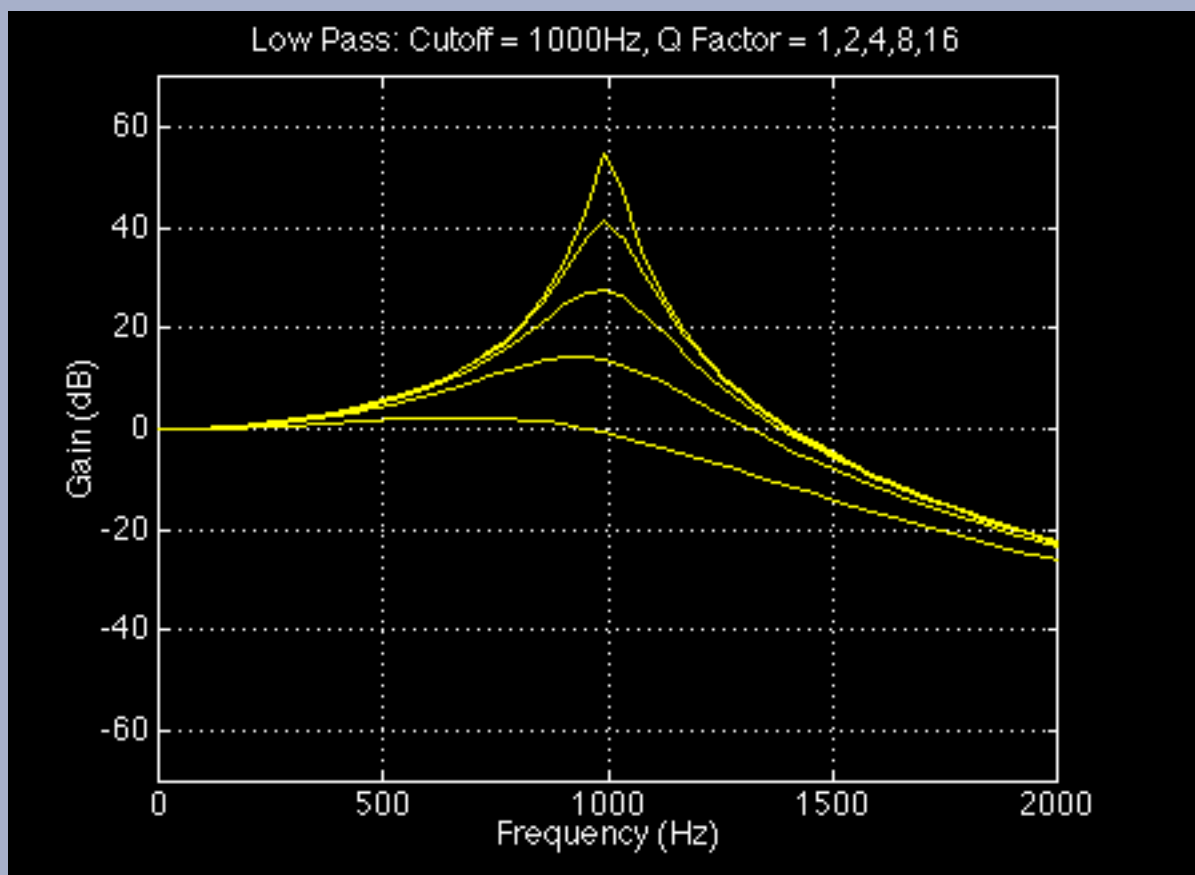
Low pass filter response curves for different values of Q. A low Q filter has a smooth roll off. A high Q filter has a resonance peak before the cutoff frequency.

Example

▐▌▶ Load the example sound file Provence Ambience and Play it. You should hear crickets while a propeller plane passes in the sky.

▐▌▶ Select Loop playback (if necessary).

▐▌▶ Select the Low Pass filter from the Plug-ins menu.

➘ A Blue Window appears.

▐▌▶ Change the Cutoff Frequency maximum setting to 2000 Hz in order to get a more accurate definition.

▐▌▶ Set Cutoff Frequency to 450 Hz with Q factor of 2.

➘ You should be hearing the plane only. Note that if you raise the cutoff frequency, you'll hear the crickets coming in.

☞ Note that the overall level may have decreased. You can use the Faders to amplify the signal.

High Pass

| H 2 | VST | MMP G | DAS |

A High Pass filter only lets you hear high frequencies. It attenuates frequencies below the specified cutoff frequency. (For more on general properties of filters, see the description of the Low Pass filter, above.)

☞ One of the shortcomings of early Hyperprism versions has been the ease with which you could overdrive the filters with extreme cutoff frequency settings. Among the many improvements we've made in Hyperprism 2 is that we're now normalizing the output of our filter effects so the output stays smooth and even (and listenable) across the entire range of cutoff frequency settings.

Parameters

Frequency

Frequency scale from 13 to 20,000 Hz. Select the cutoff frequency within this spectrum. This parameter is called Cutoff Frequency in Hyperprism 2.

Q Factor

(See the previous explanation of Q for the Low Pass filter.) Ranges from 1 to 100 in Hyperprism 2, Hyperprism-DAS and Hyperprism-VST, from 0.001 to 10 in Hyperprism-MMP.
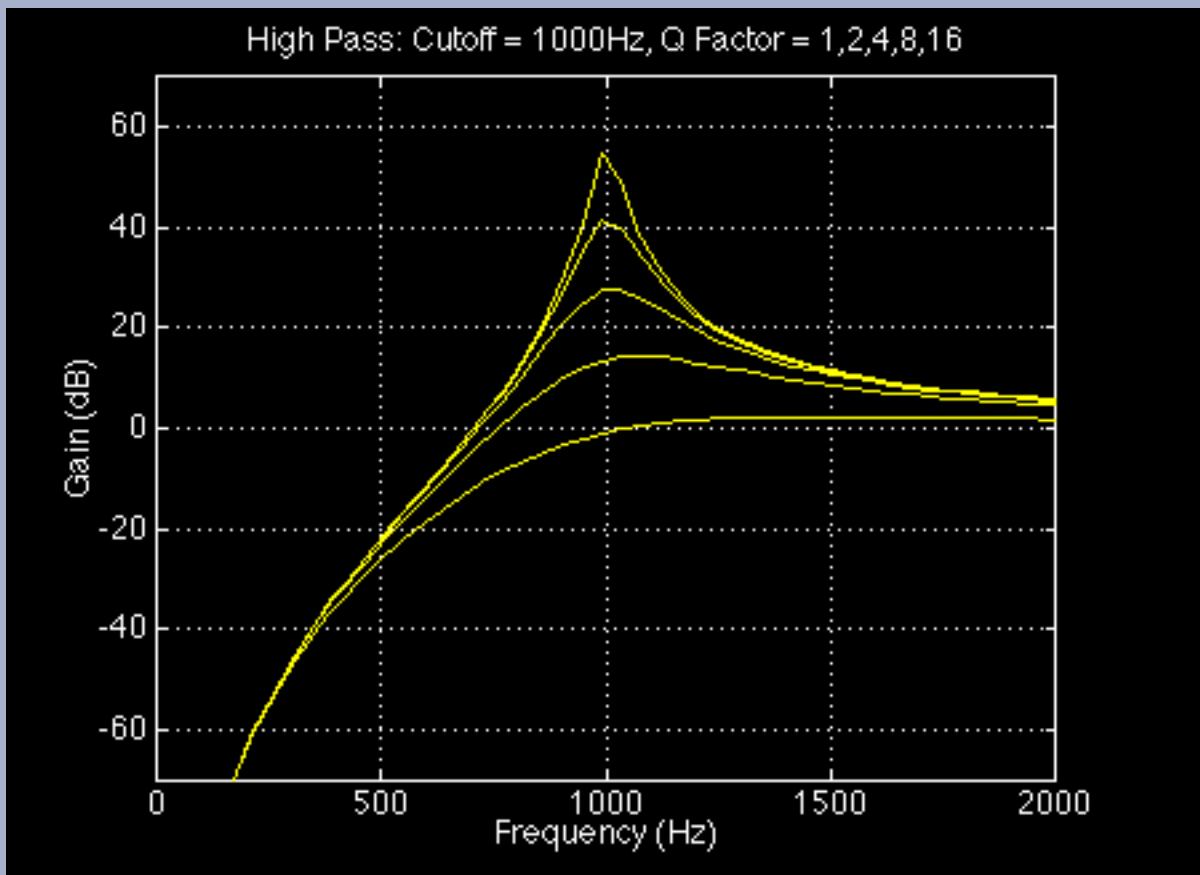
High pass filter response curves for different values of Q. A low Q filter has a smooth roll off. A high Q filter has a resonance peak before the cutoff frequency.

Example

▐▶ Load the example sound file Provence Ambience and Play it.

➤ You should hear crickets while a propeller plane passes in the sky.

▐▶ Select Loop playback (if necessary).

▐▶ Select the High Pass filter from the Plug-ins menu.

➤ A Blue Window appears.

▐▶ Change the maximum value for cutoff frequency to 5000 in order to get a more accurate definition.

▐▶ Make sure you have the Q and Cutoff sliders mapped to the Blue Window axes, then click and drag the mouse in the Blue Window.

➤ As you move the cutoff frequency from low to high, the low frequencies are attenuated more and more until you stop hearing the sound of the plane. Notice also how different Q settings effect the sound, high Q settings creating a noticeable resonance, low Q settings allowing some attenuated low frequency signal to pass through.

☞ Note that the overall level may have gone down. You can use the Faders to amplify the signal.

Band Pass

`H 2` `VST` `MMP G` `DAS`

A band pass filter lets you hear a certain band of frequencies within an upper and lower range. Frequencies above and below this band are attenuated. The distance between the higher and lower cutoff frequencies in a band pass filter is called the bandwidth of the filter. The center frequency of a band pass filter is the maximum point of amplitude.

Parameters

Center Frequency

Also called "Frequency" in some Hyperprism versions. Use this parameter to select the center frequency, which is the maximum point of amplitude. This parameter ranges from 13 Hz to 20 kHz in Hyperprism 2, Hyperprism-MMP and Hyperprism-VST, and ranges from 0-20 kHz in Hyperprism-DAS.

Q Factor

Ranges from 1 to 100 in Hyperprism 2 and Hyperprism-VST, from 0.001 to 100 in Hyperprism-MMP, from 0 to 10 in Hyperprism-DAS. It determines the sharpness of the filter, and is akin to the resonance factor. Q can be defined precisely for a band pass filter as the ratio of the center frequency to the spread of its -3 dB point (cutoff point) bandwidth.

$$Q = \frac{f_{center}}{f_{highcutoff} - f_{lowcutoff}}$$

where fcenter is the filter's center frequency, fhighcutoff is the upper -3 dB point, and flowcutoff is the lower -3 dB point. When the center frequency is constant, adjusting the Q is the same as adjusting the bandwidth.
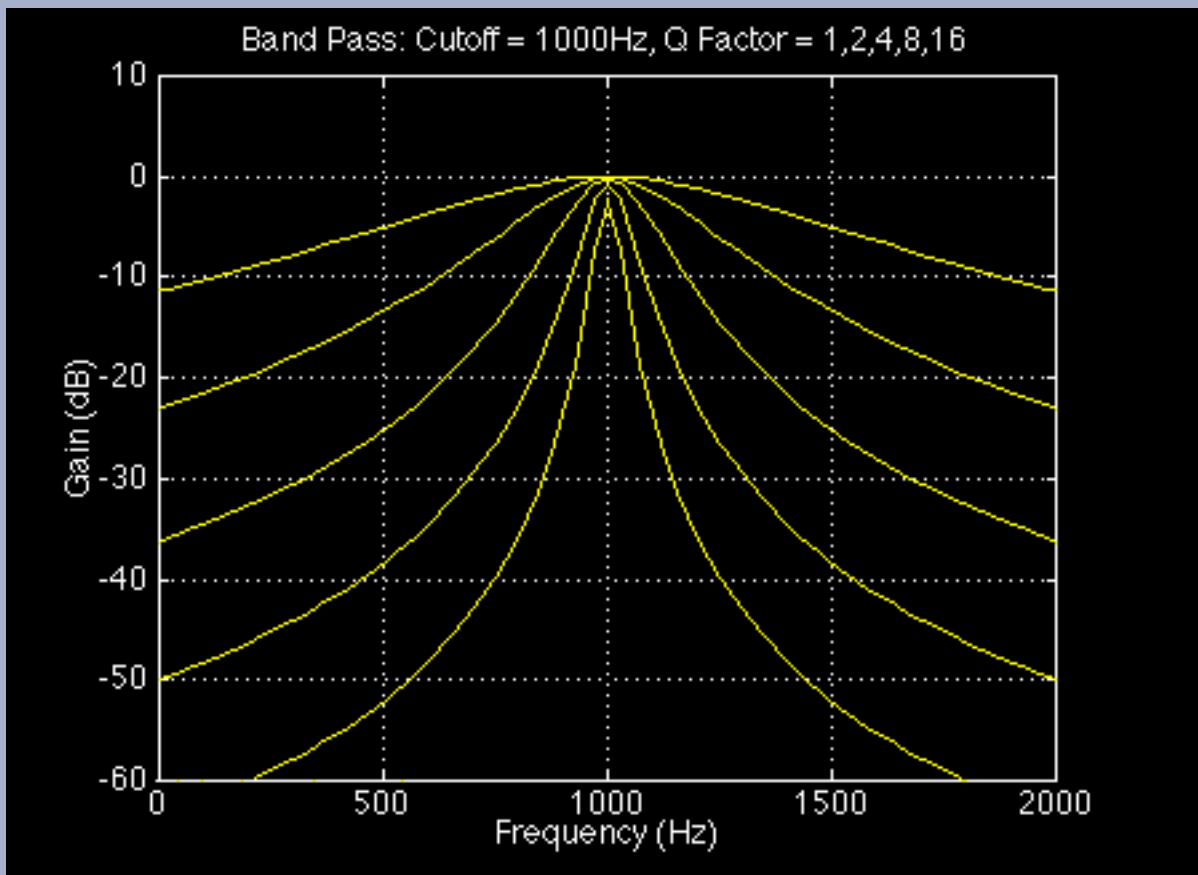
Band pass filter response curve for different Q values

☞ When the Q is high, the filter will begin to resonate or ring at the center frequency. However, a high Q setting will also tend to reduce the overall signal level coming out of the filter. The Q factor can be increased to a very high value, in which case it generates a sinusoidal tone, amplitude of which varies according to the input signal. If this signal is loud enough to cause distortion, reduce the level of Hyperprism-PPC's Faders.

Example

▮▶ Load the example sound file Airport+1050 Hz and Play it.

↘ You should hear an announcement recorded in the Paris Orly airport that contains an additional signal at 1050 Hz.

The Band Pass filter lets one isolate this frequency and eliminate everything else. In order to do this:

▮▶ Click on Loop playback (if necessary)

▮▶ Select the Band Pass filter from the Plug-ins menu.

↘ A Blue Window appears.

▮▶ Modify the frequency scale to a bandwidth between 500 and 2000 Hz, and the Q factor from 10 to 200.

▮▶ Select a center frequency of 1050 Hz and a Q factor of 200.

You should now be hearing just the 1050 Hz tone, with the announcement filtered out.

Band Reject

`H 2` `VST` `MMP G` `DAS`

A band reject filter cuts out sound within a bandwidth centered around a selected frequency. It passes frequencies above and below this bandwidth. In this sense, it is the inverse of a band pass filter, since the center frequency of a band reject filter is the minimum point of amplitude.
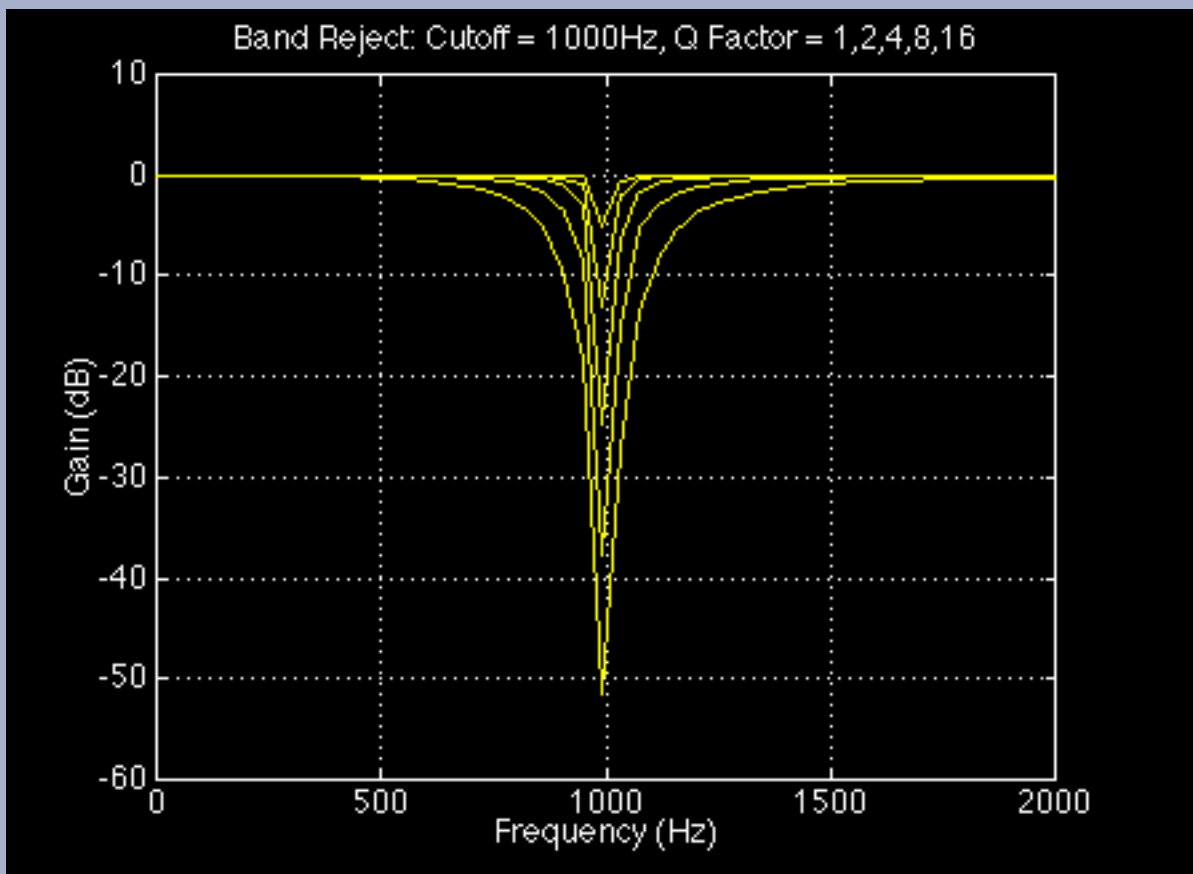
Approximate response curves for the Band Reject filter at a fixed center frequency, but at various Q values. At a Q of 1, the dip in the center is smooth.

Parameters

Center Frequency

Frequency scale selects the center frequency of the rejected band, from 13 Hz to 20 kHz. This parameter is simply called "Frequency" in Hyperprism-MMP and Hyperprism-DAS.

Q Factor

Ranges from 1 to 1000 in Hyperprism 2, Hyperprism-VST and Hyperprism-DAS, from 0.001 to 1000 in Hyperprism-MMP. Determines the sharpness of the cutoff of the filter. The higher the Q, the sharper the cutoff. See the description of Q for the Band Pass filter (above).

Example

▌▶ Load the example sound file Airport+1050 Hz and Play it.

➘ You should hear an announcement recorded in the Paris Orly airport that contains an additional signal at 1050 Hz.

The Band Reject filter lets one eliminate this frequency. In order to do this:

▌▶ Click on Loop playback (if necessary)

▌▶ Select the Band Reject filter from the Processes menu.

➘ A Blue Window appears.

▌▶ Attach the Center Frequency parameter to the horizontal axis, and Q to the vertical axis .

▌▶ Change the Center Frequency mix/max values to 1000 and 1100 Hz in order to get a more accurate definition on the horizontal axis.

▌▶ Click in the middle of the Blue Window. Drag the mouse toward the bottom to decrease the Q factor until the 1050 Hz tone is not heard anymore. Your Q setting will be very close to zero when the tone disappears.

Go to Effects: [Filters] [Modulators] [Delay and Reverb] [Stereo] [Misc Effects]
Go to Table of Contents: [Hyperprism-VST]

[Troubleshooting] [About Digital Audio]

Hyperprism Modulation Processes

- [Vocoder](#)
- [Frequency Shifter](#)
- [Z-Morph](#)
- [Phaser](#)
- [Flanger](#)
- [Chorus](#)
- [Ring Modulator](#)
- [Tremolo](#)
- [Vibrato](#)

Vocoder

Ah, the sweet sounds of the Vocoder. First heard as a vocal effect on Wendy Carlos and Kraftwerk recordings, still in widespread use on the latest films and hit records. Use it to create talking guitars or merge synth pads with drum samples for intense new loops. Special care was taken to give the Hyperprism Vocoder a real, warm, analog vocoding sound.

If you're using Hyperprism 2, Hyperprism VST, or Hyperprism-MMP you can vocode mono sources, by selecting any other sound file to be the modulator or source for vocoding. In Hyperprism 2 (Power Mac) you can modulate any live signal with a sound file, or even vocode two live sources against each other by modulating the left channel against the right channel. In the AudioSuite and DirectX editions you'll vocode the left channel against the right channel.

Vocoding is the process of using one signal to create a set of filters which are applied to a second signal. The signal from which the filters are set is called the modulator; the signal that gets filtered (and is heard) is called the carrier. The Hyperprism Vocoder is a 26-band vocoder; compare that to other low-res software Vocoders which may only use a handful of bands. The Vocoder looks at the modulator signal and adjusts the bands to match the spectrum of the incoming signal. It then applies the 26-band filter to the carrier signal. Remember that the center frequency of each of the 26 filters is fixed; the modulating signal is setting the gain for each of the filters. And either signal can be the modulator or carrier, to varying degrees.

 Hyperprism-VST users: Make sure you have your stereo tracks panned 100% left and right or you will get no effect with this process.

Internal/External Signal

(If you're not using Hyperprism 2 (with HyperEngine), Hyperprism VST, or Hyperprism-MMP for Premiere, you can skip

the next bit and continue below.) The Hyperprism Vocoder for Power Mac introduces the concept of the internal signal, and the external signal. The internal signal is sound #1, the sound you have selected for processing in the host application. This is generally a sound file which is loaded in your editing program, although if you're working with a Play-Thru document, your internal signal can actually be live audio played in through your sound card or mic input.

The external signal is an optional 2nd sound file which you can load to vocode with the internal signal. If you have no external signal loaded, or have switched External Off, the Vocoder will process the left channel of your internal audio signal (file or live) against the right channel. If both sides of your stereo image are similar (or if you have a mono file as your internal sound) you won't hear any effect. But if the left and right channel carry different program material you can get some radical results.

▐▐► Clicking on the Select External button results in a dialog where you select a sound file to be the external signal. Note that the External signal must be a sound file; you can't use live play-thru as the "external" source, play-thru is always an "internal" signal.
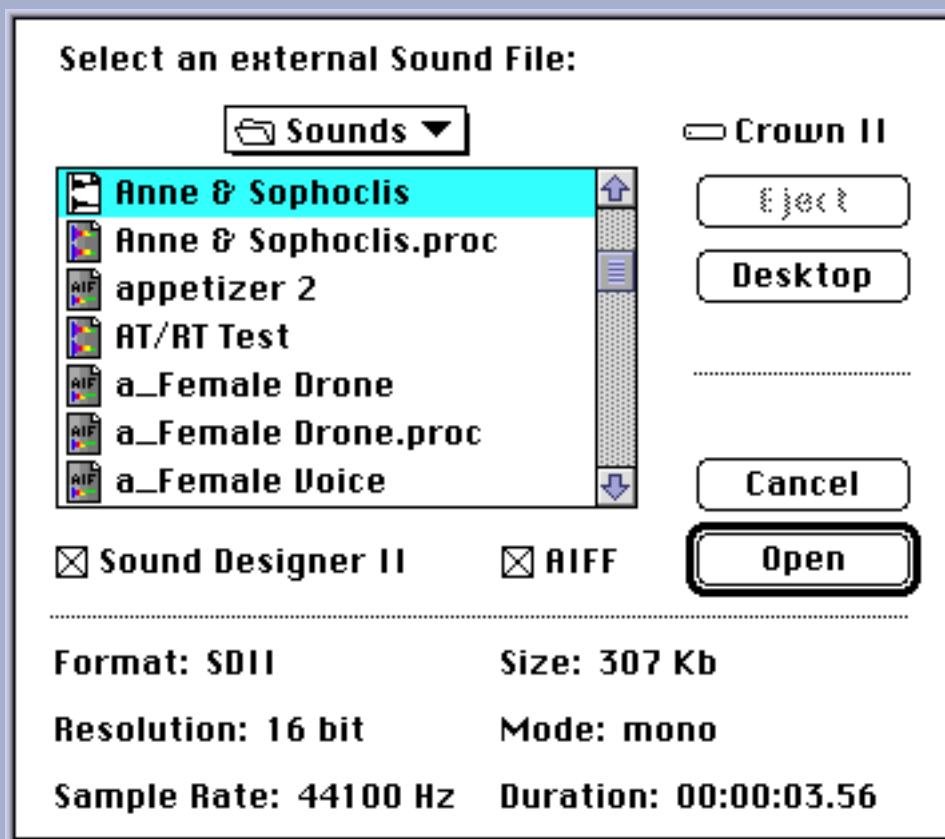
Select External sound file dialog

☞ The Vocoder's external file function will not read 24-bit files at this time. Please check www.arboretum.com for updates.

▐▐► Once you've selected a mono or stereo file as the External source, switch the External On/Off button to the on position, which enables modulation with the file you've just selected.
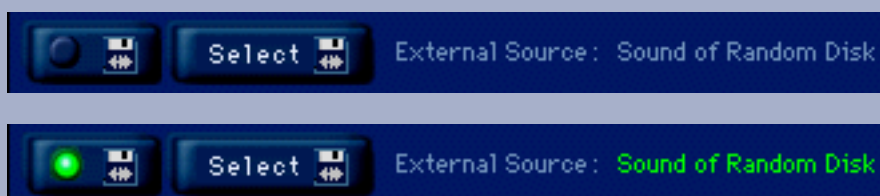
l-r: External On/Off and Select External buttons, plus External Source text display (Off and On states)

▌▶ Experiment with moving the modulation setting from one end of the scale to another; you're determining which signal (internal or external) is the modulator, and which is the carrier. Try a heavy modulation setting, plus heavy articulation, using a fat "analog" synth tone and a vocal part as inputs, to experiment with the popular "robot voice" effect.

If you have a mono source as the internal and you load a stereo external file, just the left channel of the external file will modulated with the internal signal. If you have a stereo sound as an internal source and you load a mono external file, the mono file will be modulated with both channels. If both internal and external are stereo, the left channels of each will be modulated with each other, as will the right channels.

☞ If you get a CPU Processor Overload error dialog when using this plug-in, try turning off the external file, playing the internal sound for a moment, then turn the external file back on again. This should clear the problem.

⚠ **Error: Processor Overload**
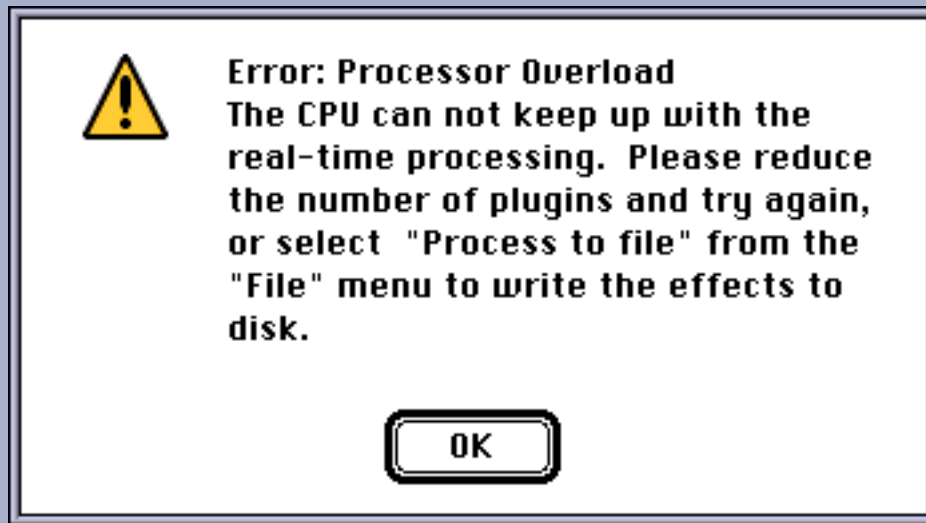The CPU can not keep up with the real-time processing. Please reduce the number of plugins and try again, or select "Process to file" from the "File" menu to write the effects to disk.

[OK]

Try temporarily turning off the External file, or closing unused plug-ins, to remedy this state.

Parameters

Modulator

0-100% Determines which sound is the source, and which is the modulator. O% means the left channel modulates the right, or if using an external sound file, that the internal left channel modulates the external source. When set to 100%, the right channel modulates the left, or the external sound modulates the internal source.

Articulation

0-100% Controls how quickly changes in the modulation sound are applied to the source sound. When set at 100%, fast changes are applied, when at 0% no changes are applied. Turn the articulation down to smooth out the vocoded sound, or turn it up to make the sound sharper, and speech more intelligible.

Mix

0-100% Mix of the original sound with the vocoded sound. Note that you can use this to monitor an external sound source by setting the Modulator to 0% and the Mix to 0%.

External On/Off

Turn on and off the currently selected external source. If no external file is selected, turning this On will prompt you to select a file. This function is in Hyperprism 2, Hyperprism VST, and Hyperprism-MMP only.

Select External

Opens a dialog where you pick a new sound file as your external sound. This function is in Hyperprism 2, Hyperprism VST, and Hyperprism-MMP only.

Example (All Versions)

▐▐▶ Load one of the example sounds from CD into your editor, select the "Freakazoid.aiff" file.

▐▐▶ Once you've loaded the file, select the Vocoder. You should be in the Hyperprism Blue Window.

▐▐▶ Turn the Mix slider to 100%, you want to hear all vocoded signal, no original.

▐▐▶ Assign the Modulator and Articulation parameters to the X and Y axis of the Blue Window by using the tri-state Axis switches.

select vertical or horizontal axis by using the three-way toggles

▐▐▶ Press the spacebar to begin previewing playback, then move the mouse cursor around the Blue Window to hear how different combinations of Modulator sources and Articulations sound.

▐▐▶ Set the Modulator parameter to 0 (all the way left) and the Articulation to 100.

➘ You should be hearing that classic vocoding effect.

▐▐▶ Now try moving the Modulator to 100 (R) for an interesting switch.

➘ With the Modulator back down at 0 (L) reduce the Articulation from 100% down to 10% to hear the smoothing of the vocode effect.

Example (Hyperprism 2, Hyperprism VST, and Hyperprism-MMP Only)

▐▐▶ Load the CD example sound file called "Rhythmic" into your sound editor.

▐▐▶ Once you've loaded the file, select the Vocoder. You should be in the Hyperprism Blue Window.

▐▐▶ Now click on the Select External button; in the resultant dialog select the example sound file "Surf Lounge."

▐▐▶ Set Modulator to 0, Articulation to 100 and Mix to 100.

➘ You should be hearing the guitar (from the left channel of the external signal) being mapped onto the drums.

▐▐▶ Now try setting Modulator to 100 and hear the drums mapped into the guitar.

Frequency Shifter

The Frequency Shifter is a great electronic-sounding process which instantly transforms any sound, adding lots of fur and movement. This new effect shifts all frequencies of your sound up or down the spectrum by an equal amount.

It's related to the Ring Modulator, but unlike that process the number and the relative positions of the frequency components are left unchanged. That leaves the sound far more intelligible while still imparting a metallic flavor to the signal. The Frequency Shifter maintains the harmonic series as it moves along the spectrum, meaning the lowest frequency in the original sound still corresponds to the lowest frequency in the new sound. No new low sidebands are created, as in the Ring Modulator.

The movement of the spectrum is controlled by a base frequency and an additional LFO frequency. Try using the Frequency Shifter to create robotic voices or pulsing drums.

Parameters

Base Frequency

Ranges from -20,000 to 20,000 Hz. Set the modulation frequency which your sample will be shifted to. Ranges from -2,000 to 2,000 Hz in Hyperprism-MMP.

Feedback

Ranges from 0-100%. Adds resonant regeneration of the modulated sound.

LFO Frequency

Ranges from 0-100 Hz. Use to add movement to the modulation frequency.

LFO Depth

Ranges from 0-1000%. Determines the amount of the LFO applied to the base frequency. At the time of this writing (11/1/98) this parameter range was mislabeled in Hyperprism-DAS, where it is represented as ranging from 0 to 20,000 Hz. We expect this to be corrected in the next edition of H-DAS.

Example

⏸▶ Load the example sound file "Morgan.aiff" into your sound editing software.

⏸▶ Select the Frequency Shifter plug-in; you should see the Blue Window.

⏸▶ Set the Base Frequency to -200 Hz, set all other parameters to 0.

➘ This shifts Morgan's voice down 200 Hz.

Z-Morph

`H 2`

This interesting effect models an input sound as variable number of resonant filters, and then applies that model to another sound, morphing the "heard" sound (the carrier) into the modeled sound (modulator).

The Z-Morph operates somewhat like the Vocoder except frequency bands are not fixed. This allows the algorithm to best fit the incoming signal. Frequency bands move depending on the incoming frequency.

Internal/External Signal

The Z-Morph introduces the concept of the internal signal, and the external signal. The internal signal is sound #1, the sound you have selected for processing in the host application. This is generally a sound file which is loaded in your editing program, although if you're working with a Play-Thru document, your internal signal can actually be live audio played in through your sound card or mic input.

The external signal is an optional 2nd sound file which you can load to modulate with the internal signal. If you have no external signal loaded, or have switched External Off, the Z-Morph will process the left channel of your internal audio signal (file or live) against the right channel. If both sides of your stereo image are similar (or if you have a mono file as your internal sound) you may not hear much effect. But if the left and right channel carry different program material you can get some radical results, as with the example above.

▌▶ Clicking on the Select External button results in a dialog where you select a sound file to be the external signal. Note that the External signal must be a sound file; you can't use live play-thru as the "external" source, play-thru is always an "internal" signal.
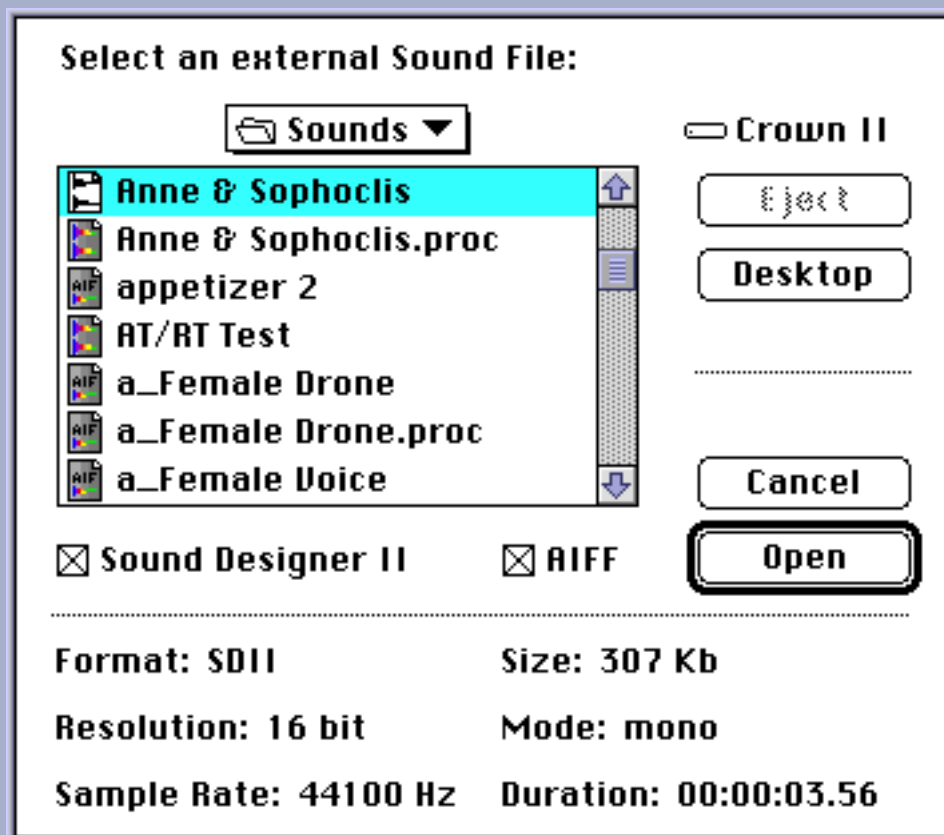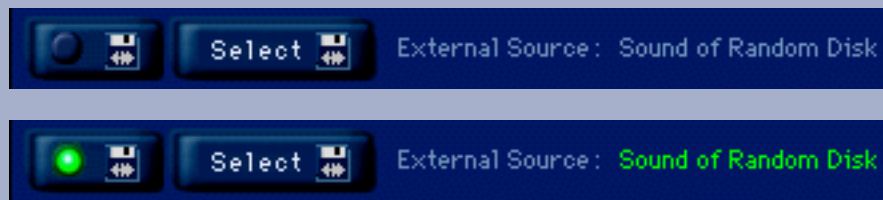

Select External sound file dialog

☞ The Z-Morph's external file function will not read 24-bit files at this time. Please check www.arboretum.com for

updates.

▮▶ Once you've selected a mono or stereo file as the External source, switch the External On/Off button to the on position, which enables modulation with the file you've just selected.

l-r: External On/Off and Select External buttons, plus External Source text display (Off and On states)

If you have a mono source as the internal and you load a stereo external file, just the left channel of the external file will modulated with the internal signal. If you have a stereo sound as an internal source and you load a mono external file, the mono file will be modulated with both channels. If both internal and external are stereo, the left channels of each will be modulated with each other, as will the right channels.

▮☞ If you get a CPU power error dialog when using this plug-in, try turning off the external file, playing the internal sound for a moment, then turn the external file back on again. This should clear the problem.
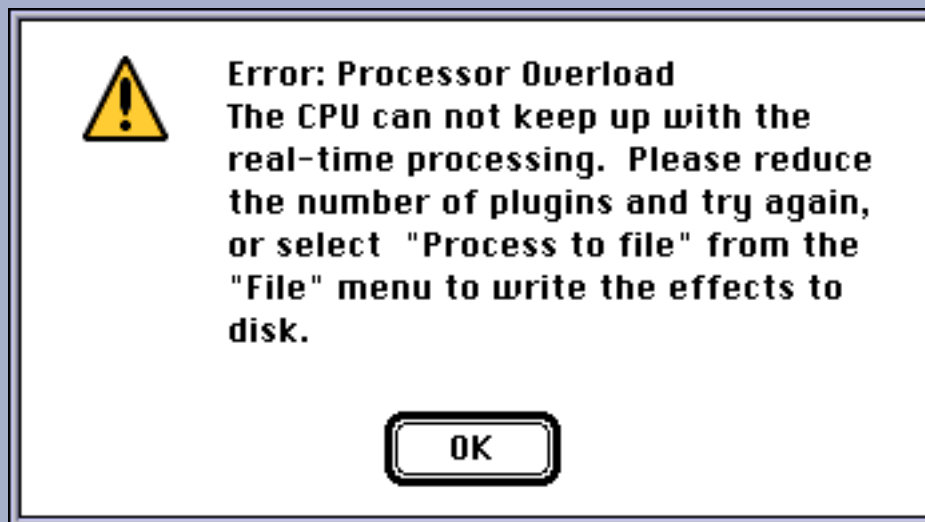
Try temporarily turning off the External file, or closing unused plug-ins, to remedy this state.

Parameters

Morph

Ranges from 0-100%. Sets which way the morph is applied, and the degree of morphing. 0% means using the right channel or external file to fully morph the left channel or internal sound, 100% means the left channel or internal file is morphed to the right channel/external file fully.

Poles

Controls how many resonators (poles in engineering terms) are used to model the source sound. Possible settings range from 0-128. Using one pole, the Z-Morph will pick one frequency which dominates the input sound to be the resonator, and this will lead to a more bassy sound. The more resonators used, the more accurate the model will be, the more high frequencies will be modeled, and therefore the more convincing the morphing sound will be. However, using more that a few poles can quickly slow down real-time processing. If you want to use a lot of poles you may need to Process to New File in order to hear such higher order modeling; live preview processing can easily crap out, even on the fastest

computers, if you use dozens of poles. The amount of processing also depends on the material being used, and therefore the number of poles usable for real-time processing will always vary. On a 100 MHz Power Mac 4 poles is the real-time limit on average.

Filter Source

0-100%. Use to determine which sound will be modeled (the filter source) and which sound will be then be morphed to the modeled sound. 0% means that the internal sound or left channel is the filter source, and 100% means that the right channel or external file is the source.

External On/Off

Turn on and off the currently selected external source. If no external file is selected, turning this On will prompt you to select a file.

Select External

Opens a dialog where you pick a new sound file as your external sound.

Examples

▌▶ Create a new File reference document (command-n) and open the example sound file "Surf Lounge."

↘ This file has guitar playing in the left channel and vocals in the right.

▌▶ Now open the Z-Morph plug in, and begin previewing playback.

▌▶ Set Morph to 5, Poles to 1 and Filter Source to 100%.

↘ This gives a filtering effect similar to a wah-wah pedal.

▌▶ Leave Morph a 5, Set poles between 10 and 13, and Filter Source to 100%.

↘ This gives more of a talk box effect.

▌▶ Now close this file document and create another new File reference document, opening the example sound file "DrumLp 120 bpm."

▌▶ Click on the Select External button and in the resultant dialog select the example sound file "1930 Recording."

▌▶ Set Morph to 50, Poles to 4 and Filter Source to 96.

▌▶ Feel free to adjust the settings as you wish, and experiment!

Phaser

`H 2` `VST` `MMP G` `DAS`

The Phaser produces an effect reminiscent of old phase shifter boxes from the late 70's. Here, an all-pass filter introduces a frequency dependent delay, resulting in phase distortion. A wide range of results can be achieved by adjusting the amount

of this distortion and using feedback to increase the intensity of the effect.

Parameters

Frequency

Frequency is the number of delay cycles per second, from 0 to 10 Hz in Hyperprism-MMP, from 0 to 4 kHz in Hyperprism-DAS and Hyperprism-VST, 0 to 1000 Hz in Hyperprism 2.

Depth

Depth controls the intensity of the phase shift effect, values range from 0 to 10 samples.

Feedback

The Feedback slider introduces regeneration, in which the output of the Phaser is mixed back at its input. A setting of 0% is no feedback, 100% will lead to run-away regeneration, as the full output of the effect is constantly being routed to it's own input.

Base Frequency

The Base Frequency slider sets the lowest frequency of the original sound that will be processed by the Phaser algorithm. Higher settings will leave lower frequency components of the sound relatively un-effected. The range of this setting is from 0 to 10,000 Hz.

Flanger

Flanger applies a time-varying delay to the input signal and then mixes the original signal with the delayed signal, creating a time-varying comb filter effect. The time-variance of the delay changes according to a low-frequency sine wave generated by Hyperprism.

Hyperprism 2 adds mono to stereo capability to this effect.

Parameters

Frequency

The number of delay cycles per second. Use low settings for classic flange effects; high settings produce bizarre "klang" tones. Goes to 1 kHz in Hyperprism 2, to 2 kHz in Hyperprism-DAS and Hyperprism-VST, to 10 Hz in Hyperprism-MMP.

Depth

The intensity of the flange effect, expressed in number of samples. The Flanger delay oscillates (with a sine wave shape) between 0 and this value. Higher settings result in a more pronounced Flanging effect. The maximum setting is 100%, which equals 1024 samples.

Feedback

The Feedback slider introduces regeneration, in which the output of the Flanger is mixed back at its input. A setting of 0% is no feedback, 100% will lead to run-away regeneration, as the full output of the effect is constantly being routed to it's own input.

Mix

Controls the volume balance between original signal and flanged signal. Set it to 100% to hear only flanged sound, 0% gives you only "dry" signal.

Chorus

H 2   VST   MMP G   DAS

A chorus effect can be heard whenever several voices sing the same note in unison. Because of slight differences in the timing, pitch, vibrato and spectrum of the different voices, we hear an ensemble effect. Hyperprism can apply a chorus effect to any input sound.

Pro Tools/Avid users: Although there's only one H-DAS Chorus plug-in in your DAE folder, the Chorus effect shows up twice in the AudioSuite menu as separate Mono Chorus and Stereo Chorus selections. Choose according to the number of audio channels to be processed. Note that there is no direct mono-to-stereo processing under the AudioSuite format. But you can easily work around this limitation by copying any mono selection to an adjoining track, then selecting both regions for stereo processing. The Stereo Chorus requires two-channel input or a DAE error message -7450 will result when you attempt to process. Also note that Hyperprism stereo AudioSuite effects require the following settings: Use in Playlist (not Region) and be in Multi-Input Mode.

Parameters

Frequency

Vibrations per second, which cause an undulating pitch variation in the delayed signal. Goes to 2 kHz in Hyperprism 2, to 1000 Hz in Hyperprism-DAS and Hyperprism-VST, to 10 Hz in Hyperprism-MMP.

Depth

The intensity of the chorus effect, expressed as number of samples. The Chorus delay oscillates (with a sine wave shape) between 0 and this value. Higher settings result in a more pronounced chorusing and detuning effect. The maximum setting is 100%, which equals 1024 samples.

Mix

Controls the volume balance between original and processed signals. Set Mix to 100% to hear only processed sound, 0% gives you only "dry" signal.

Delay

Allows you to control the base time difference between the original signal and the additional "chorused" signal. Ranges from 0 to 100 ms.

Contour

A high pass filter allowing you to control the "color" of the sound. Cut off frequency ranges from 0 to 20 kHz in Hyperprism 2, goes to 10 kHz in Hyperprism-DAS, Hyperprism-VST and Hyperprism-MMP. Any processed signal below the cut off will be filtered out.

Channel Switch
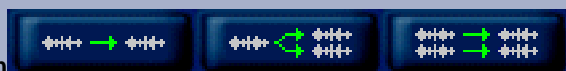
If you're processing a mono source you have the option of making the output either mono or stereo. Simply click on the Channel Switch to toggle between the two settings. If you're processing a stereo source your output will be stereo by default; the button will default to the stereo/stereo display and be deactivated from toggling. Hyperprism 2 (Power Mac) only.

Ring Modulator

`H 2`   `VST`   `MMP G`   `DAS`

The Ring modulator is capable of generating a wide variety of metallic and bell-like tones.

The input sound is used as a carrier signal. A second signal called a modulator frequency, is used to generate two frequencies that are the sum and the difference of the carrier and the modulator frequencies, each at half the amplitude value of the carrier. Thus, for an input frequency at 400 Hz and a modulator frequency at 25 Hz, the output will be a mixture at 375 Hz and 425 Hz. It is the blending of non-harmonically related sounds that results in the effect's metallic, sometimes harsh textures.

Parameters

Frequency

The modulator frequency in cycles per second. Goes from 0 to 20 kHz in Hyperprism 2, Hyperprism-VST and Hyperprism-MMP, to 10 kHz in Hyperprism-DAS where's it's called "Base Frequency."

Depth (or "Mix")

Depth controls the direct/effect mix between the original sound and the Ring modulator's output, from 0% to 100%. A setting of 100% lets you hear only the processed output signal, without any of the original source. A setting of 0% lets you hear only the original signal, with no effect. This parameter is called "Mix" in Hyperprism-VST, Hyperprism-MMP and Hyperprism-DAS.

LFO Frequency

A low frequency oscillator can be applied to the modulator, allowing you to create even richer sonic effects. This slider controls the LFO frequency and ranges from 0 to 1000 Hz in Hyperprism 2, Hyperprism-VST and Hyperprism-DAS, 0 to 40 Hz in Hyperprism-MMP.

LFO Depth

Controls the amount of LFO applied to the modulator tone. Ranges from 0 to 100%.

Examples

▌▶ Load the example file **Bell**, Play the sound, and then select the Ring Modulator process.

▌▶ Click at different mouse locations and the Blue Window and hear the changes in the sound.

☞ The Ring Modulator creates a wide range of interesting sounds.

Tremolo

H 2 **VST** **MMP G** **DAS**

Tremolo varies the amplitude (volume) of the input sound smoothly according to a sinusoidal function. For many sustained sounds (such as violins) a tremolo between 4 and 7 cycles per second adds expressiveness, particularly when applied in an increasing manner over the course of the sound. (See the following figure.) Very fast oscillations (i.e. 5,000 Hz) result in amplitude modulation effects.

Parameters

Frequency

Number of amplitude variations per second. Goes from 0 to 2 kHz in Hyperprism 2, to 1 kHz in Hyperprism-DAS and Hyperprism-VST, to 30 Hz in Hyperprism-MMP.

Depth

Degree of amplitude variation as a percentage of the total output, from 0 to 200%. Goes to 100% in Hyperprism-MMP.

Examples

▐▶ Create a New HyperEngine document for the file "Train," Play it and select a Tremolo process.

▐▶ Once the Tremolo effect opens, connect the Depth parameter to the vertical axis and the Frequency parameter to the horizontal axis.

▐▶ In Loop mode, click in the Blue Window at the bottom left corner and drag the mouse toward the upper left corner, then staying at the top of the window, move slowly to the right.

➘ Listen as your modern Train changes into an old-time steam engine!

☞ Tremolo works well with Vibrato in multi-pass processing or as part of a HyperEngine effect chain, particularly on melodic instrumental sounds.
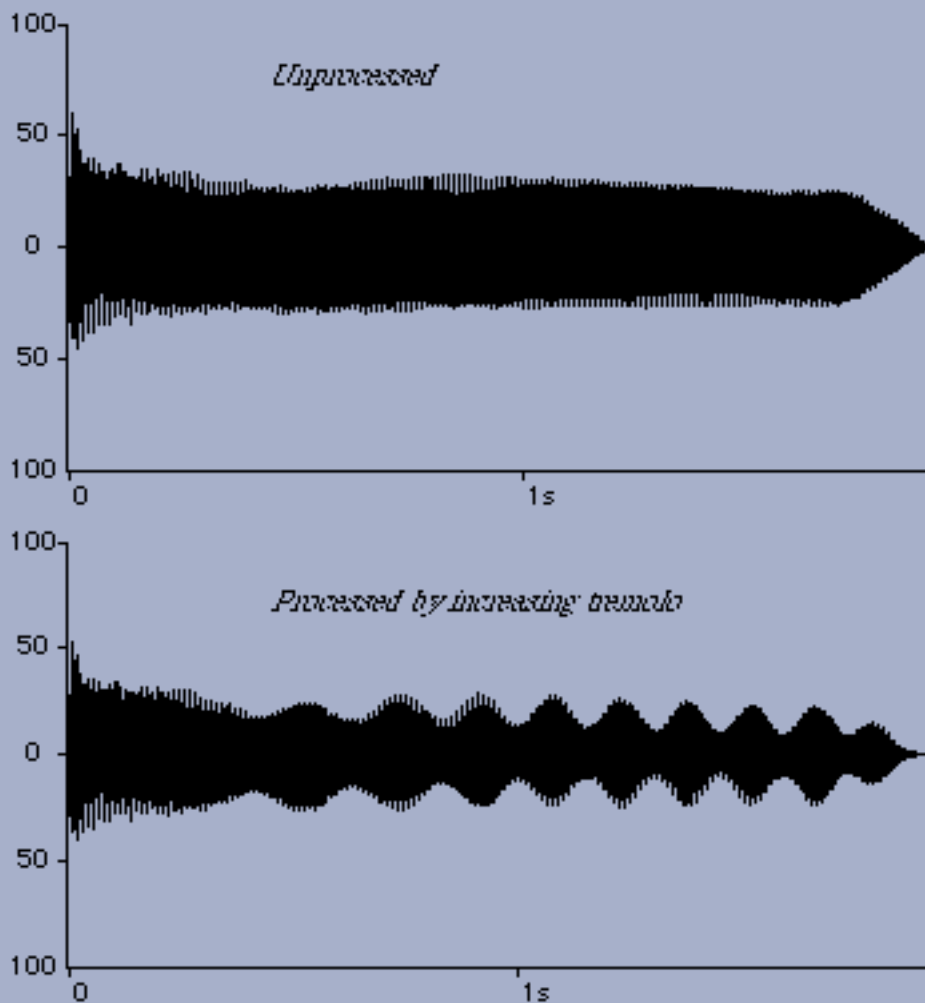
Effect of an increasing tremolo effect on a guitar pluck. The tracer went from 0 Hz to 7 Hz on the horizontal scale, and 0 to 55% on the vertical scale.

Vibrato

H 2   VST   MMP G   DAS

Vibrato imposes a time-varying pitch change or modulation on the input signal. The modulation varies sinusoidally at a rate controlled by the user. For many sustained sounds, a vibrato between 4 and 7 cycles per second adds expressiveness, particularly when applied in an increasing manner over the course of the sound. Extreme settings (i.e. maximum depth, plus frequency rates of 20 to 200 Hz result in frequency modulation-like effects.

Parameters

**Frequency**

The number of pitch variations per second. Goes to from 0 to 2 kHz in Hyperprism 2, to 500 kHz in Hyperprism-VST, to 30 Hz in Hyperprism-MMP, to 1000 Hz in Hyperprism-DAS.

Depth

Degree of pitch variation, from 0 to 100%.

Example

Try applying Vibrato to the Bell sound. Process the results to a new file, and then process again using the Tremolo effect. This gives very interesting results.

Go to Effects: [Filters] [Modulators] [Delay and Reverb] [Stereo] [Misc Effects]
Go to Table of Contents: [Hyperprism-VST]

[Troubleshooting] [About Digital Audio]

Hyperprism Delay & Reverb Processes

- [Single Delay](#)
- [Multi Delay](#)
- [Echo](#)
- [EchoTranz](#)
- [HyperVerb (TM)](#)
- [Room Reverb](#)
- [Hall Reverb](#)

Single Delay

`H 2` `VST` `MMP G` `DAS`

Single Delay lets you vary the length of time between the original sound and a delayed copy of that sound. It gives you a single "echo" of your signal, which you presumably mix in with the original sound.

Parameters

Delay

Delay in milliseconds. By default, from 0 to 2000 ms (two seconds). In Hyperprism 2 (Power Mac) you can create delays up to three seconds.

Mix

Mix between the original and delayed signal as a percentage, by default from 0 to 100%.

Multi Delay

`H 2` `VST` `MMP G` `DAS`

Multi Delay delays the input signal three times. The user sets the first two and the program calculates the third Delay time by adding the times of Delays 1 and 2.

Parameters

Delay 1

Delay 1 in milliseconds, by default 0 to 1000. Goes up to 2 k in Hyperprism 2.

Delay 2

Delay 2 in milliseconds, by default 0 to 1000. Goes up to 2 k in Hyperprism 2.

Delay 3

The Multi-Delay's D3 delay time is calculated automatically by adding D1 and D2 delay times.

Mix
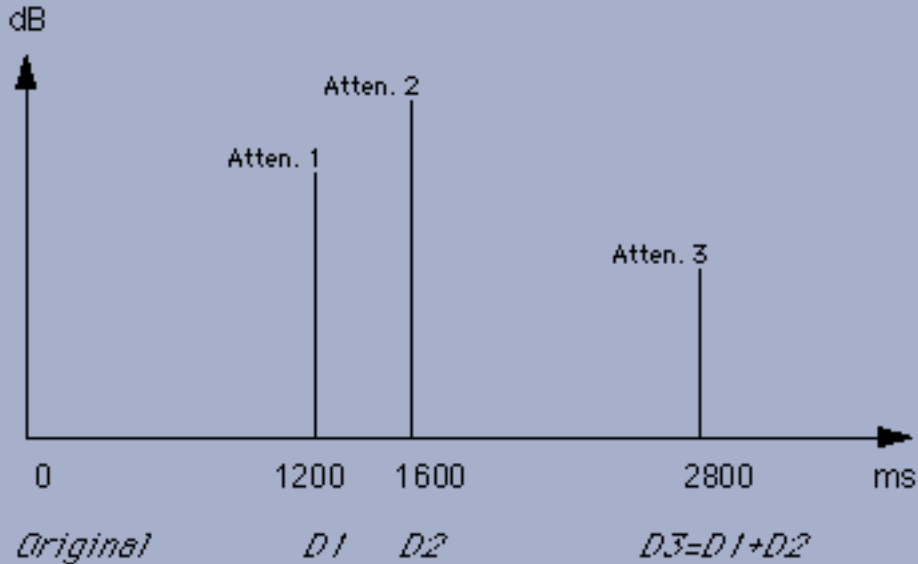
The Mix slider sets the balance between the original sound's volume and the overall volume of the repeats, from 0 to 100%. The remaining sliders provide volume control over the three individual repeats.

Delay 1 Attenuation

The Delay 1 Attenuation slider cuts the volume of Delay 1. The range of volume control is from 0 dB to -36 dB attenuation.

Delay 2 Attenuation

The Delay 2 Attenuation slider cuts the volume of Delay 2. The range of volume control is from 0 dB to -36 dB attenuation.

Delay 3 Attenuation

The Delay 3 Attenuation slider cuts the volume of Delay 3, the delay length of which is set by sum of the delay times of Delay 1 and Delay 2. The range of volume control is from 0 dB to -36 dB attenuation.

Example

▐▌▶ Load the example sound file Gunshot and select a Multi Delay process. Play it and you will hear a single shot fired by a hunting rifle.

▐▌▶ Select Loop playback.

▐▌▶ (Hyperprism 2: Click on the Audition button before moving tracer.) Click and move the mouse slowly around in the

Blue Window and you'll hear a Western-style shoot-out.

Limitations

Depending on the input signal, you may hear a clicking sound as you move the mouse in the Blue Window. To eliminate this artifact of rapidly-changing delay time, decrease the value of the Parameter Speed Change. (Parameter omitted from Hyperprism 2.)

Echo

H 2 VST MMP G DAS

The Echo effect lets you vary delay time, feedback and/or mix with gestures in the Blue Window. A feature of this algorithm is the ability to specify a negative feedback. Negative feedback settings cause the echoes to build in intensity, rather than die out. In Hyperprism 2 the Feedback range is 0% to 200%, 101% to 200% create negative feedback values.

Parameters

Delay

Delay in milliseconds, from 0 to 2000. Hyperprism 2 allows delay times up to three seconds.

Feedback

Feedback of the repeated signal, from 0 to 100%. Zero percent feedback will result in only a single repeat, while increased feedback settings will generate numerous repeats. Entering negative values (-1 to -100%) causes the echoes to build in intensity rather than die out. In Hyperprism 2 the negative feedback range is 101% to 200%. Negative feedback values are not allowed in Hyperprism-MMP and Hyperprism-DAS.

Mix

The Mix slider sets the balance between the original sound's volume level and the volume level of the echoes.

EchoTranz (TM)

H 2 VST

What's especially interesting about the EchoTranz effect are the dramatic, undulating pitch transpositions that result from rapid changes in delay time.

Each echo is attenuated according to the vertical position of the mouse. If the feedback is 100%, for example, the echoes will repeat indefinitely. A feature of this algorithm is the ability to specify a negative feedback, typically in the range of -10 to -30. Negative feedback settings (or settings over 100% in Hyperprism 2) cause the echoes to build in intensity, rather than die out. Using a combination of varying time delays and positive and negative attenuation gestures, you can create effects reminiscent of the classic tape-delay feedback sounds used in electronic music studios of the 1960's.

Parameters

Delay

Delay in milliseconds, by default from 0 to 2000. Goes up to three seconds in Hyperprism 2.

☞ Sweeping through Delay values creates nasty pitch effects; this is the best part of EchoTranz. If you're using Hyperprism 2 make sure you have a healthy Delay Rate Change setting (see below) so you can fully hear the effect.

Feedback

Feedback of the repeated signal, by default from 0 to 100%. Zero percent feedback will result in only a single repeat, while increased feedback settings will generate numerous repeats. Negative feedback values are also accepted in most versions; Hyperprism 2 allows you to enter feedback values above 100% instead. In both cases, these latter type of values will result in echoes which built in intensity. It can get pretty scary.

Parameter Speed Change

When you move the mouse using this effect, this results in varying the reflection time slightly, as if the wall reflecting the source was moving. This movement is not natural, and can introduce clicks that affect the audio quality of the processing. The role of Parameter Speed Change is to interpolate between mouse movements in order to lessen the unwanted side-effects.

A value of 100% makes all delay values coming from the Blue Window take effect instantaneously. This will often result in soft clicking when rapid mouse movements are made. To avoid this, use lower values. A setting of 1% corresponds to the slowest change possible.

Parameter Speed Change can be used as an effect in itself. Notice how the delay progresses depending on different settings in the Parameter Speed Change parameter. High Parameter Speed Change settings can result in extreme tape-shuttling effects.

Mix

The Mix slider sets the volume balance between the original sound and the overall volume of the repeats.

Delay Rate Change

Hyperprism 2 only. Controls how quickly the Delay time parameter moves. At a 0% setting delay rate will not change. At low settings pitch changes will occur more slowly. At higher settings faster pitch changes will be heard when adjusting the Delay settings.

HyperVerb (TM)

`H 2` `VST` `MMP G` `DAS`

HyperVerb offers a significant improvement in sound quality over previous software reverbs. It simply sounds more realistic than any previous attempt. The two most important improvements are significantly increased echo density and virtually no coloration in the late response, all while running in real-time on a modest processor (e.g. 166 MHz Pentium or 120 MHz Power Mac). One or both of these deficiencies has plagued most digital reverberators since their initial development more than 30 years ago.

HyperVerb features super smooth impulse response and sounds great on percussive tracks. HyperVerb won't color your sound unnaturally, gives you lots of parameter control and dense reflections without ringing or metallic sounding artifacts.

☞ Want to know more about why HyperVerb sounds better, and view hard evidence that proves it? Click here for our report on how HyperVerb really stacks up against its competitors.

Naturally there's full Blue Window gestural control over all effect parameters, but if you're after a naturalistic effect it's

usually better to just dial up the perfect setting, save it as a preset, then treat your file with all the parameter settings staying constant.

☞ **Pro Tools/Avid users:** Although there's only one H-DAS HyperVerb plug-in in your DAE folder, the HyperVerb effect shows up twice in the AudioSuite menu as separate Mono HyperVerb and Stereo HyperVerb selections. Choose according to the number of audio channels to be processed. Note that there is no direct mono-to-stereo processing under the AudioSuite format. But you can easily work around this limitation by copying any mono selection to an adjoining track, then selecting both regions for stereo processing. The H-DAS Stereo HyperVerb requires two-channel input or a DAE error message -7450 will result when you attempt to process. Also note that Hyperprism stereo AudioSuite effects require the following settings: Use in Playlist (not Region) and be in Multi-Input Mode.

Parameters:

PreDelay

Ranges from 0-500 ms. Sets the delay time of the first echo, after which reverberation begins.

Diffusion

Ranges from 0-100%. Controls how smeared out individual echoes are. High diffusion will wash them all together creating lush sounds, and 0 diffusion gives a bouncy, tight sound.

Brightness

Adjustable from 2000-20000 Hz. Applies low pass filtering to the reverberations to create mellow or shimmering soundscapes. At the time of this writing (11/1/98) this parameter ranges from 0 to 20 kHz in Hyperprism-DAS; this value may be altered in a forthcoming version as settings below 2000 Hz aren't particularly useful.

Reverb Time

From 0-50 seconds. Use to set the time it takes for reverberations to fall below -60 dB. Shorter reverb times correspond to smaller rooms. For reference, large cathedrals have reverb times of about 7 seconds, and small rooms have reverb times under half a second.

Early/Late Reflection Mix

Ranges from 0-100%. This controls the mix of early reflection to late reflections. Late reflections contain more of the washed reverberant sound, early reflections have more of the bouncy initial echoes. This control is called Early/Late in the DirectX (PC/Windows) version of Hyperprism.

Master Mix

Ranges from 0-100%. Sets mix of original signal with reverberated signal.

Room Size
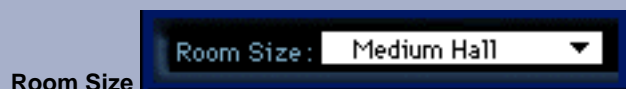
From small closet to large cathedral. Sets the perceived size of the reverberation space.

Early Reflection Spread

From 30-120 ms. Controls the distance between early reflection times, clustering them within the time set.

Channel Switch

If you're processing a mono source HyperVerb gives you the option of having the output in either mono or stereo. Simply click on the Channel Switch to toggle between the two settings. If you're processing a stereo source your output will be stereo by default; the button will default to the stereo/stereo display and be deactivated from toggling. The channel switch is not included in Hyperprism-MMP and Hyperprism-DAS, as the Premiere and AudioSuite plug-in formats do not allow direct mono-to-stereo processing. In multitrack programs such as Deck or Pro Tools you can easily work around this limitation by copying any mono selection to an adjoining track, then selecting both regions for stereo processing.

Room Reverb

Although there's a lot of excitement about the HyperVerb, we're also still offering our classic reverb algorithms, Hall and Room. The Room has an intrinsically shorter decay time than the Hall Reverb. Both create a diffuse sound field with variable brightness, feedback, wet/dry mix and diffusion, while utilizing a minimum of CPU cycles.

Parameters

Mix

Wet/Dry mix, in percentages from 0% (original signal only) to 100% (reverberant signal only) by default.

Feedback

Feedback, from 0% to 100%. Higher feedback settings lengthen the time it takes a give sound's reverb to decay to silence. A setting of 0% will result in the shortest possible decay for the algorithm, 100% feedback will linger almost indefinitely.

☞ You will typically need to increase your Post Roll value to hear the full decay created by Hyperprism's reverb algorithms when processing a sound to a new file.

Brightness

The brightness slider simulates the loss of high frequencies due to room reflections and atmospheric absorption. In practice, you can consider it a low pass filter applied continuously to the ongoing reverb decay.

A setting of 0% will result in very dark sounding reverbs, with very little high-frequency energy. A setting of 100% will be very bright; all the high frequencies from the source sound will persist in the reverberant decay.

Diffusion

Diffusion controls the phase smearing of the reflected sound. The range of values is from 0 to 140% in Hyperprism 2, Hyperprism-MMP and Hyperprism-DAS.

Low diffusion settings result in more discrete delayed repeats, while adding diffusion smears out the attacks of each repeat. Higher diffusion results in a smooth wash of sound, without identifiable reflected attacks.

☞ Diffusion settings over 100% will induce feedback.

☞ **Pro Tools/Avid users:** Although there's only one H-DAS Room Reverb plug-in in your DAE folder, the Room Reverb effect shows up twice in the AudioSuite menu as separate Mono Room Reverb and Stereo Room Reverb

selections. Choose according to the number of audio channels to be processed. Note that there is no direct mono-to-stereo processing under the AudioSuite format. But you can easily work around this limitation by copying any mono selection to an adjoining track, then selecting both regions for stereo processing. The H-DAS Stereo Room Reverb requires two-channel input or a DAE error message -7450 will result when you attempt to process. Also note that Hyperprism stereo AudioSuite effects require the following settings: Use in Playlist (not Region) and be in Multi-Input Mode.

Hall Reverb

H 2 **VST** **MMP G** **DAS**

Although there's a lot of excitement about the [HyperVerb](), we're also still offering our classic reverb algorithms, Hall and Room. The Hall Reverb has an intrinsically longer decay time than the Room Reverb. Both create a diffuse sound field with variable brightness, feedback, wet/dry mix and diffusion, while utilizing a minimum of CPU cycles.

Parameters

Mix

Wet/Dry mix, in percentages from 0% (original signal only) to 100% (reverberant signal only) by default.

Feedback

Feedback, from 0% to 100% by default. Higher feedback settings lengthen the time it takes a give sound's reverb to decay to silence. A setting of 0% will result in the shortest possible decay for the algorithm, 100% feedback will linger almost indefinitely.

☞ You will typically need to increase your Post Roll value to hear the full decay created by Hyperprism's reverb algorithms when processing a sound to a new file.

Brightness

The brightness slider simulates the loss of high frequencies due to room reflections and atmospheric absorption. In practice, you can consider it a low pass filter applied continuously to the ongoing reverb decay.

A setting of 0% will result in very dark sounding reverbs, with very little high-frequency energy. A setting of 100% will be very bright-all the high frequencies from the source sound will persist in the reverberant decay.

Diffusion

Diffusion controls the phase smearing of the reflected sound. The range of values is from 0 to 140%.

Low diffusion settings result in more discrete delayed repeats, while adding diffusion smears out the attacks of each repeat. Higher diffusion results in a smooth wash of sound, without identifiable reflected attacks.

☞ Diffusion settings over 100% will induce feedback.

☞ **Pro Tools/Avid users:** Although there's only one H-DAS Hall Reverb plug-in in your DAE folder, the Hall Reverb effect shows up twice in the AudioSuite menu as separate Mono Hall Reverb and Stereo Hall Reverb selections. Choose according to the number of audio channels to be processed. Note that there is no direct mono-to-stereo processing under the AudioSuite format. But you can easily work around this limitation by copying any mono selection to an adjoining track, then selecting both regions for stereo processing. The H-DAS Stereo Hall Reverb requires two-channel input or a DAE error message -7450 will result when you attempt to process. Also note that Hyperprism stereo AudioSuite effects require the following settings: Use in Playlist (not Region) and be in Multi-Input Mode.
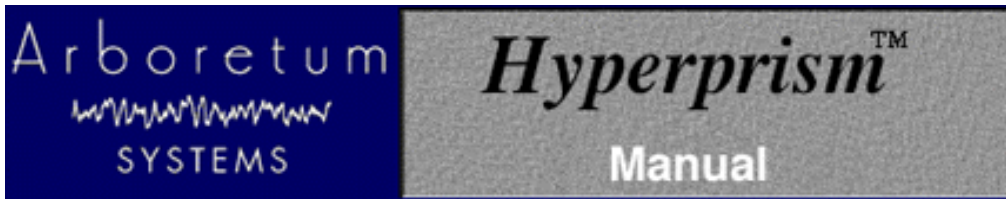
Digital Reverberator Face Off

During the development of the HyperVerb digital reverberator we spent some time analyzing common digital reverberators, as well as natural reverberation. This analysis, in addition to years of research, has helped us design a digital reverberator that, in certain respects, offers a significant improvement in sound quality over previous technologies. The two most important improvements are significantly increased echo density, and virtually no coloration in the late response, all while running in real-time on a modest processor (e.g. 150MHz Pentium or 66MHz 56002). One or both of these deficiencies has plagued most digital reverberators since their initial development 30 plus years ago.

The properties of high quality reverberation are very complicated, but for the purposes of this report we will limit the analysis to aspects that can be easily and objectively measured and that are commonly problematic with most digital reverbs. Of course the most important test is simply listening to the reverberator, though other tests can also be useful. For this reason you can also download impulse response recordings for a number of commercial reverberators (ftp.arboretum.com/pub/demos/reverb/) and listen to them as is or convolve them with actual sounds using a product like Sonic Foundry's Acoustic Modeler. Many of these files have large amounts of hiss and/or quantization noise. This is generally due to the impulse recording process and was not necessarily a deficiency of the reverberator. Since this only is a demonstration, such additional noise should not be too much of a problem. Also, since most reverberators can be made to sound "bad" if you tweak the parameters in a certain way, we tried use a generic hall factory preset for each reverberator in an attempt to make things as fair as possible. But since we are still biased with regard to this whole subject, we welcome you to create your own impulse responses so you can verify our results. If you do so please send the impulse responses our way!

All of the tests performed here are based on examination of each reverberator's impulse response. An impulse response was obtained by sending a one sample wide "click" into the reverberator and recording the output. Frequency domain and time domain plots can then be made from the impulse response. From these plots, it is then very easy to make qualitative and even quantitative comparisons based on echo density, coloration, amplitude envelope and other properties.

Echo density, often referred to as density or diffusion, can be defined as the number of "echoes" occurring per second. For example, an impulse response where the energy is concentrated at a few samples (i.e.. most samples are zero) will have a low echo density, whereas an impulse response with energy spread out in time (i.e. few samples are zero) will have a high echo density. To compare the echo density of various reverberators, plots have been made of the first 200 milliseconds of the of the impulse response. To examine the impulse responses in greater detail they can be downloaded and viewed with an audio editing program.

Coloration defines how the reverberator changes the frequency response of a sound. Frequency domain plots were made using Cool Edit Pro, using a 4096 point FFT and a Blackman-Harris window. Successive FFTs were made starting at roughly 500ms to 1000ms and then averaged to get the overall frequency response of the late reflections. Late reflections are those which occur after roughly the first 100ms. In the case of a "well liked" concert hall or room the frequency response of the late reflections will be relatively smooth and uncolored. In other words, there won't be any sharp peaks or valleys in the frequency response. In the case of a poor sounding digital reverberator, there will likely be lots of coloration. For example the common problem of a "ringy" sound corresponds to a few distinct peaks in the frequency response, where as a resonant or metallic sound quality generally corresponds to a very "rough" frequency response.

For the purposes of this report we will assume that a good digital reverberator can generate reverberation that is perceptually indistinguishable from actual reverberation of "well liked" acoustical space (e.g. a concert hall). For a good point of reference, we will first examine naturally occurring reverberation. Of course other "unnatural" types of reverberation may be useful in some cases. Here is the first 200ms of impulse responses from a chapel and a large hall.

RecordedImpulse Response of a Chapel
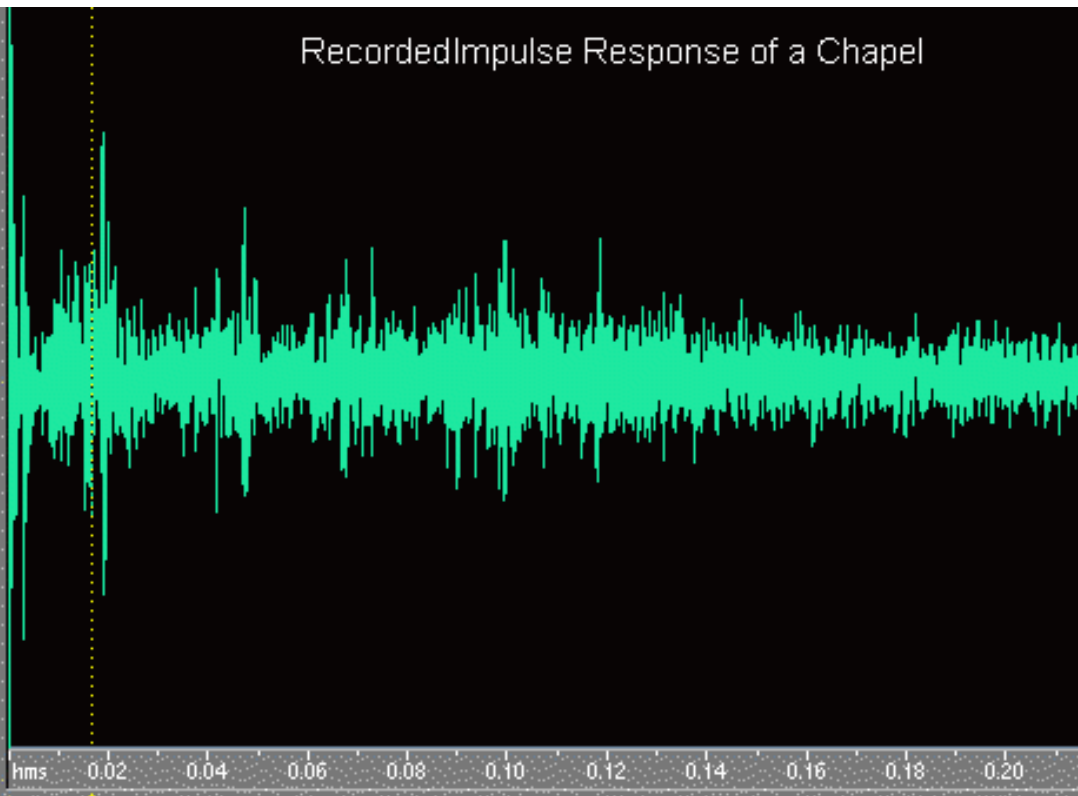
Recorded impulse of a large hall

FFT Plot of a large hall
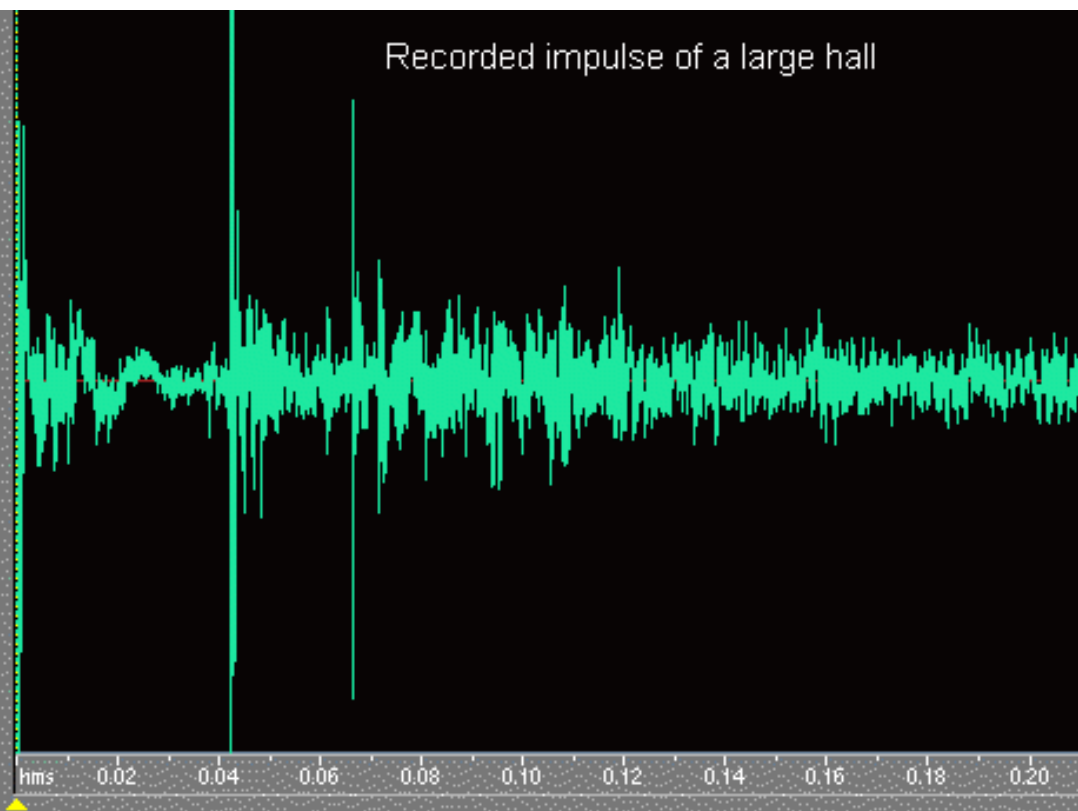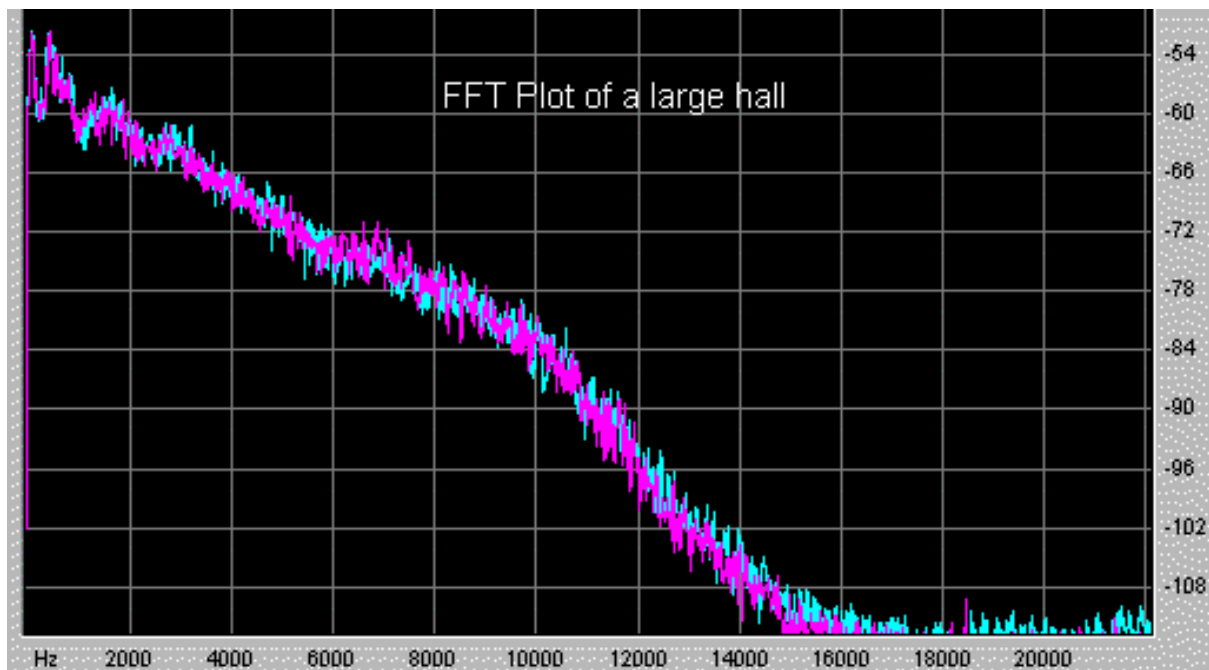
(NOTE: Cyan = Left Channel and Magenta = Right Channel)

Looking at these plots we can observe a number of distinguishing features. Some of these features include:

- Smooth decay of late reflections (i.e. little flutter)
- Relatively uncolored frequency response of the late reflections (i.e. no ringing)
- Sharp attack (usually)
- Extremely high echo density

While these features are generalizations, most naturally occurring reverberation has these properties, especially for rooms and halls with "good acoustics." Now lets examine some common digital reverberators, including a TC Electronic M2000 and a Lexicon 480L.

Sound Forge Reverb:

Sound Forge - Rich Hall
Impulse Response
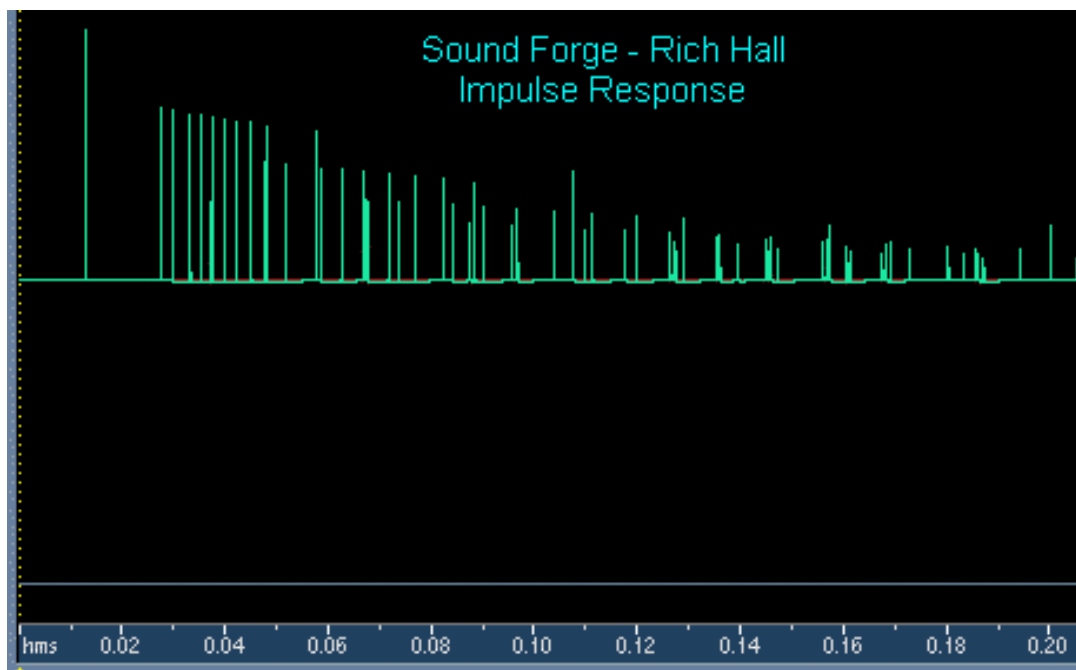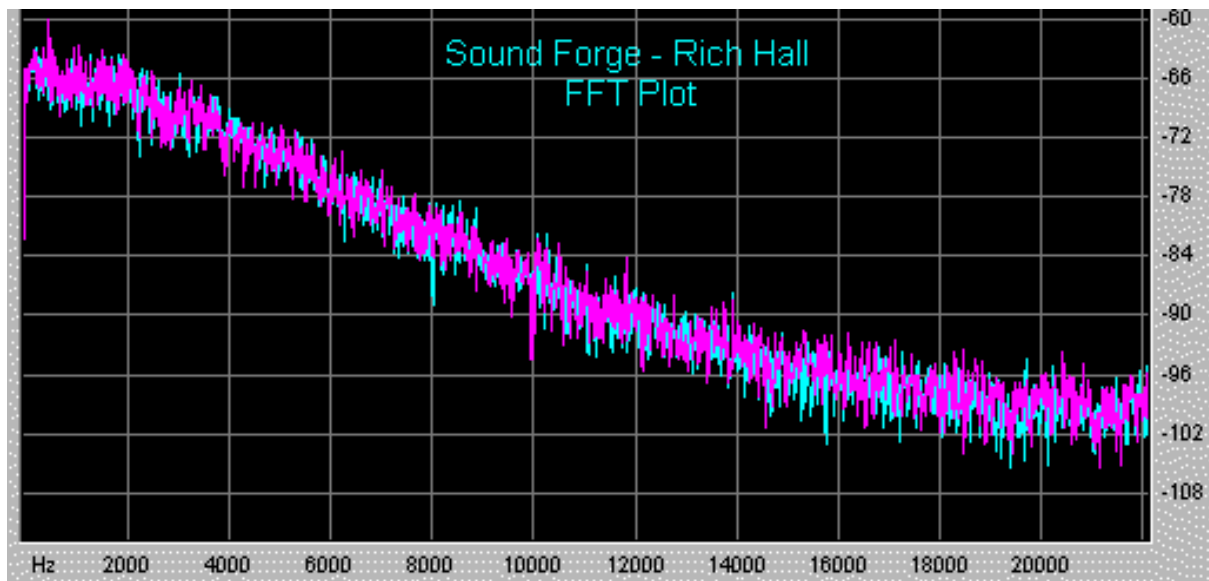
Pros:

- Somewhat colorless late decay

Cons:

- Severe lack of echo density
- Fluttery late decay
- Echo density does not increase with time

Digitech GigaVerb:
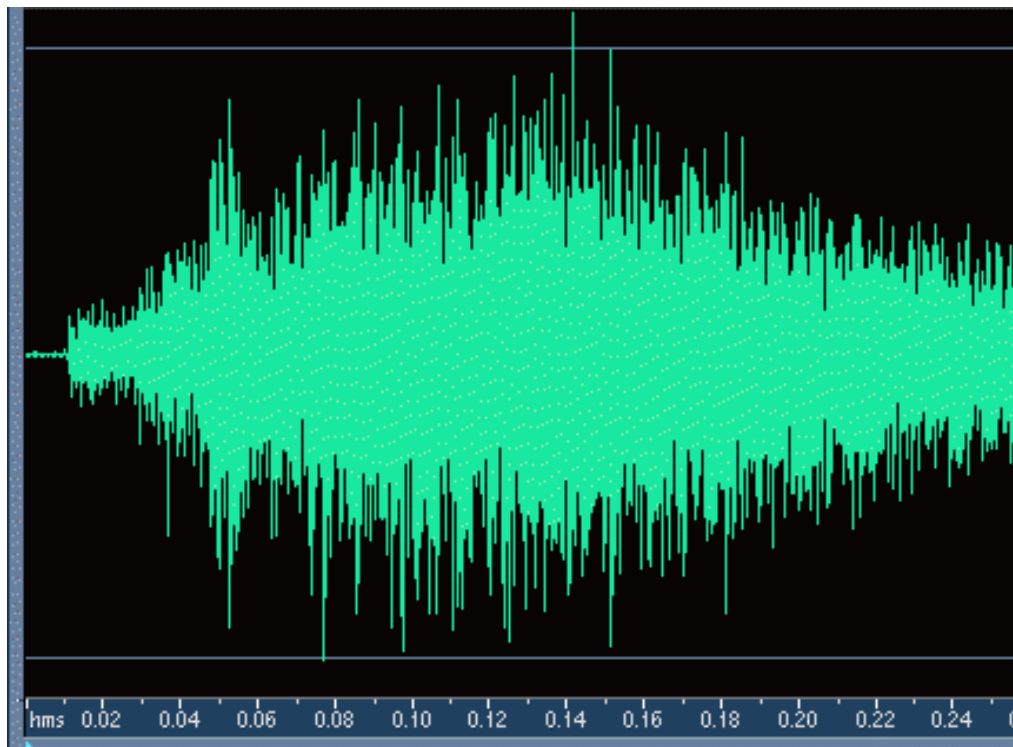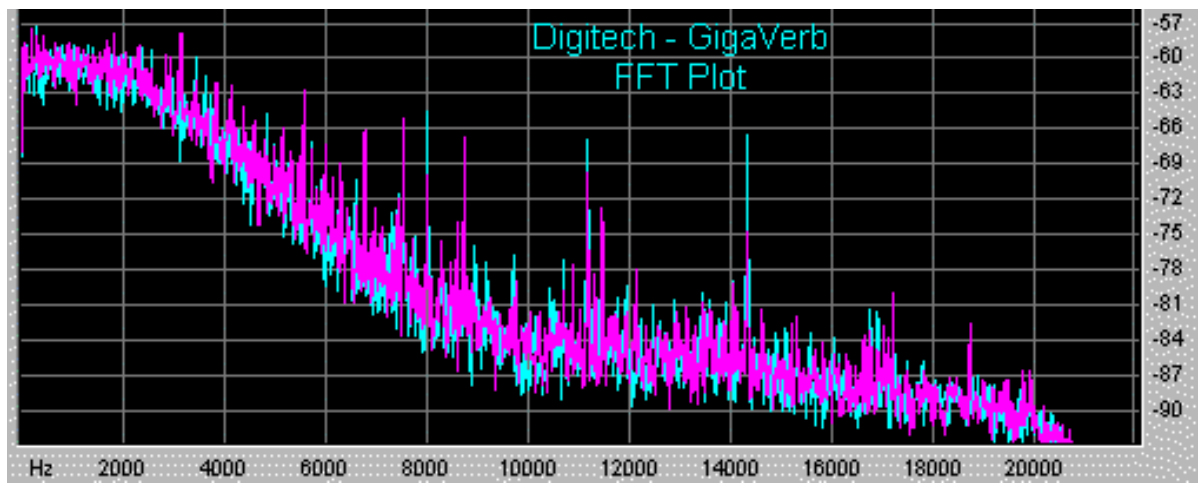
Pros:

- High echo density

Cons:

- Very long attack time (it takes over 100ms to reach maximum amplitude)
- Highly colored late reflections (i.e. resonant and ringy)

Power Technology DSP-FX Reverb:
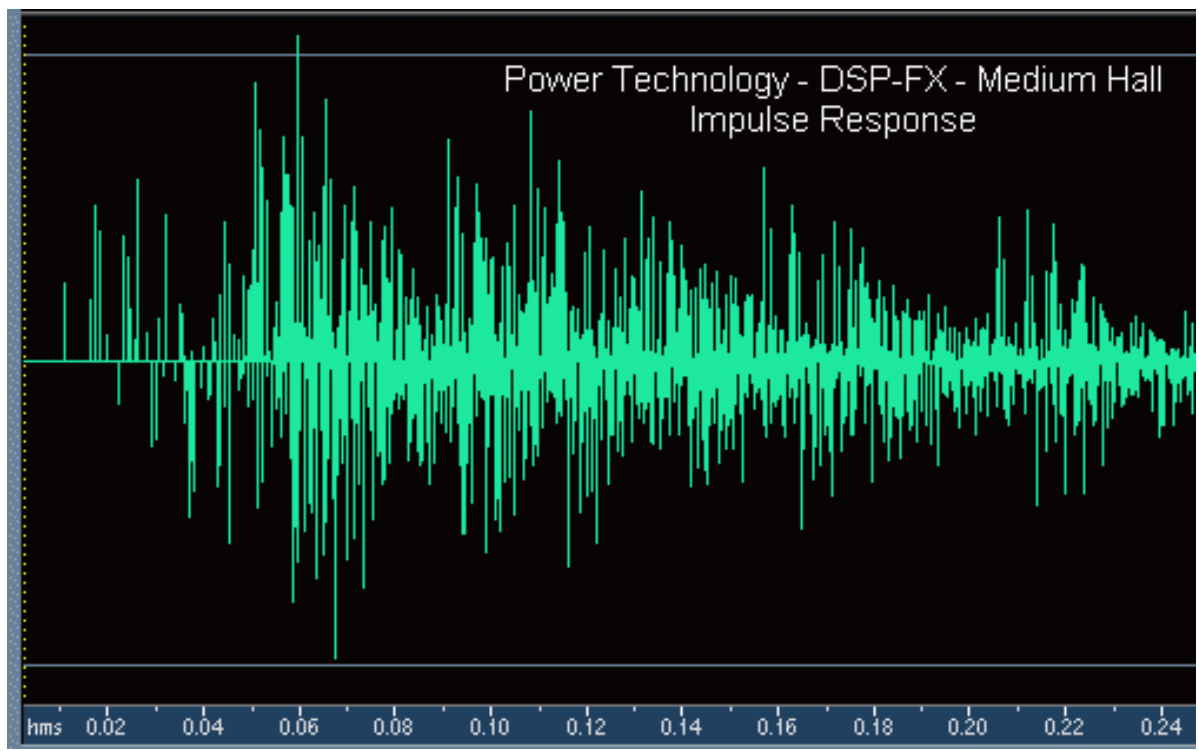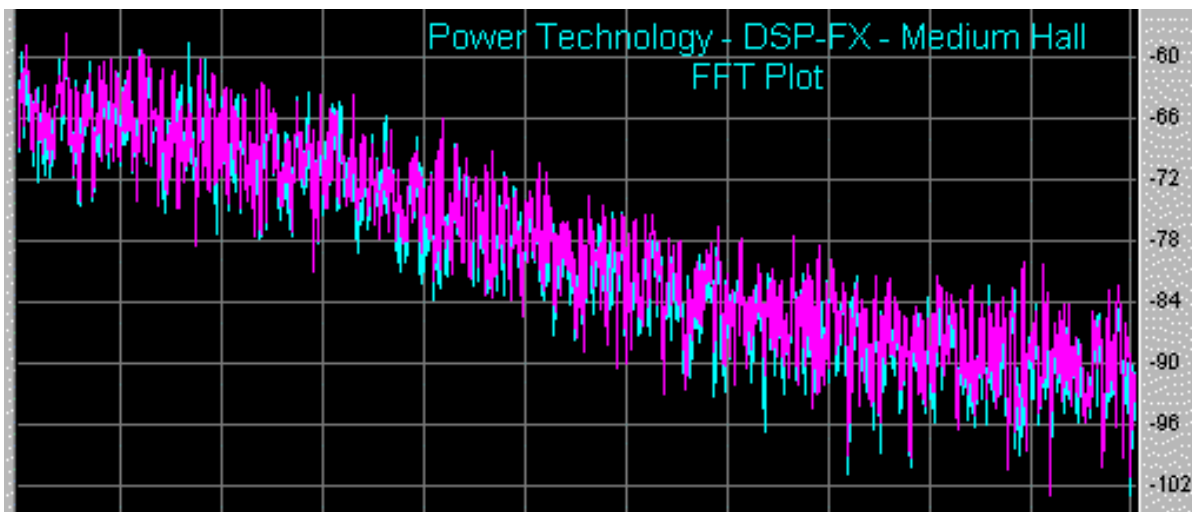
Pros:

- Moderate echo density (though not always enough)

Cons:

- Highly colored late reflections (i.e. metallic and resonant)
- Lack of echo density in the early reflections (causes "chatter" for some sounds)

TC Electronic Native Reverb:

TC Electronic - Native Reverb
FFT Plot
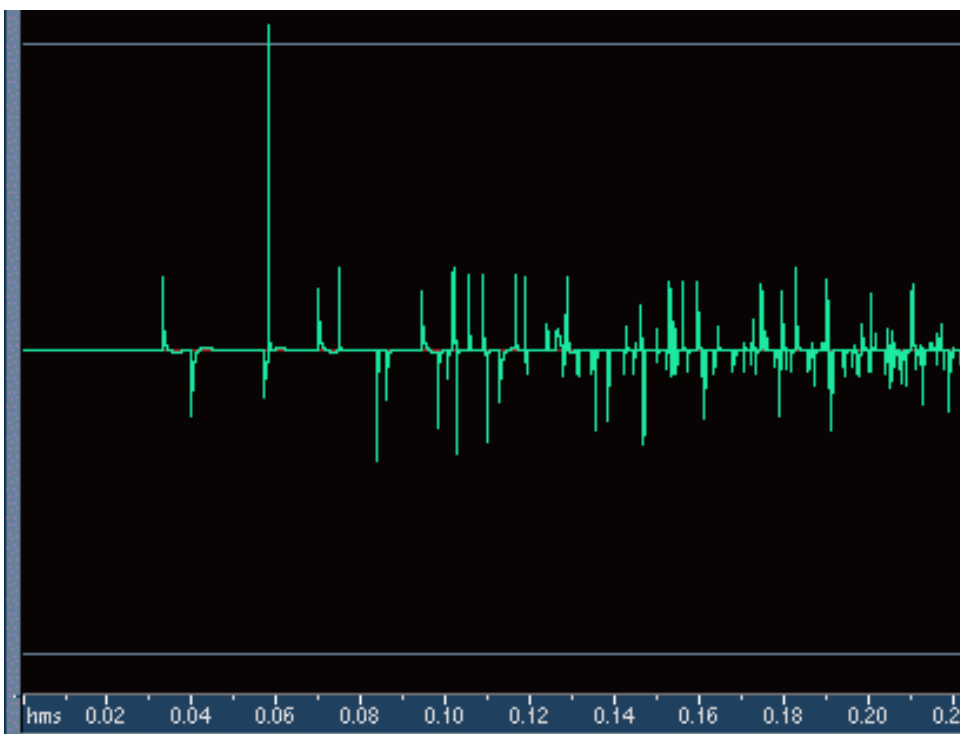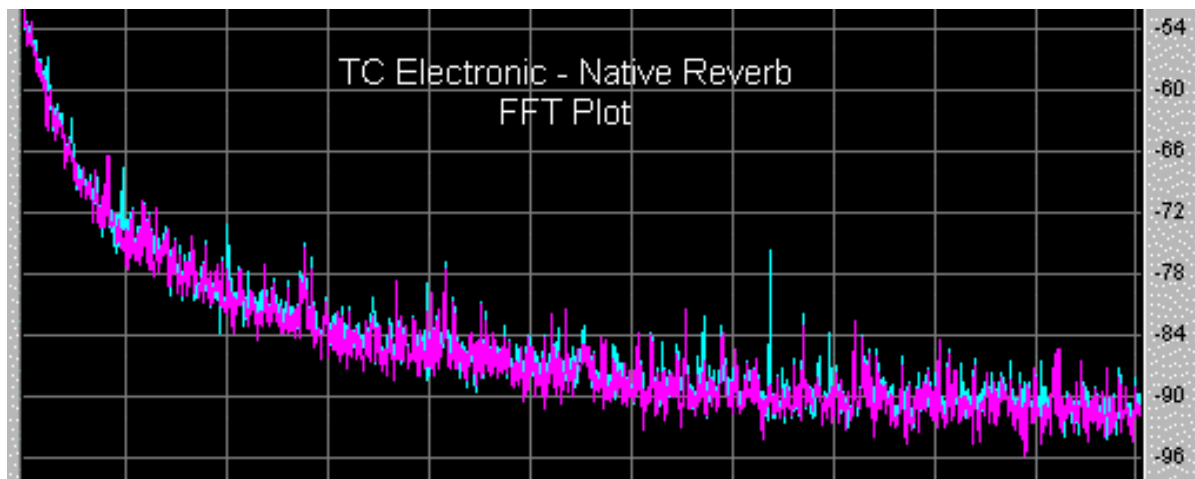
Pros:

- Deficiencies are not a pronounced as some reverbs

Cons:

- Echo density does not increase fast enough
- Fluttery late decay
- Colored late reflections (i.e. resonant and ringy)

Waves TrueVerb:

Waves - TrueVerb
Impulse Response
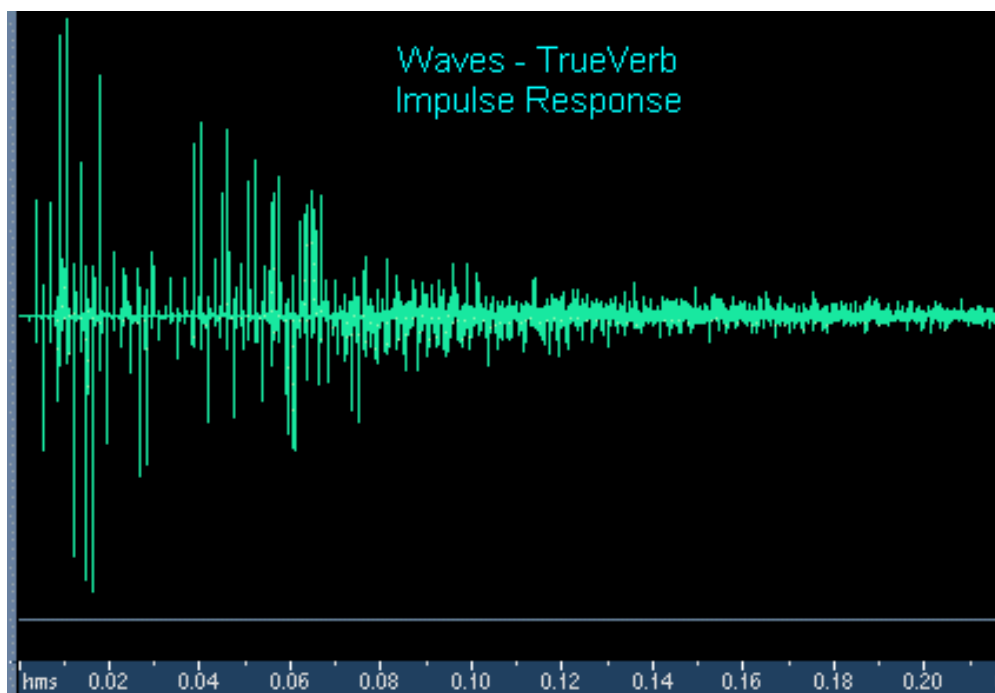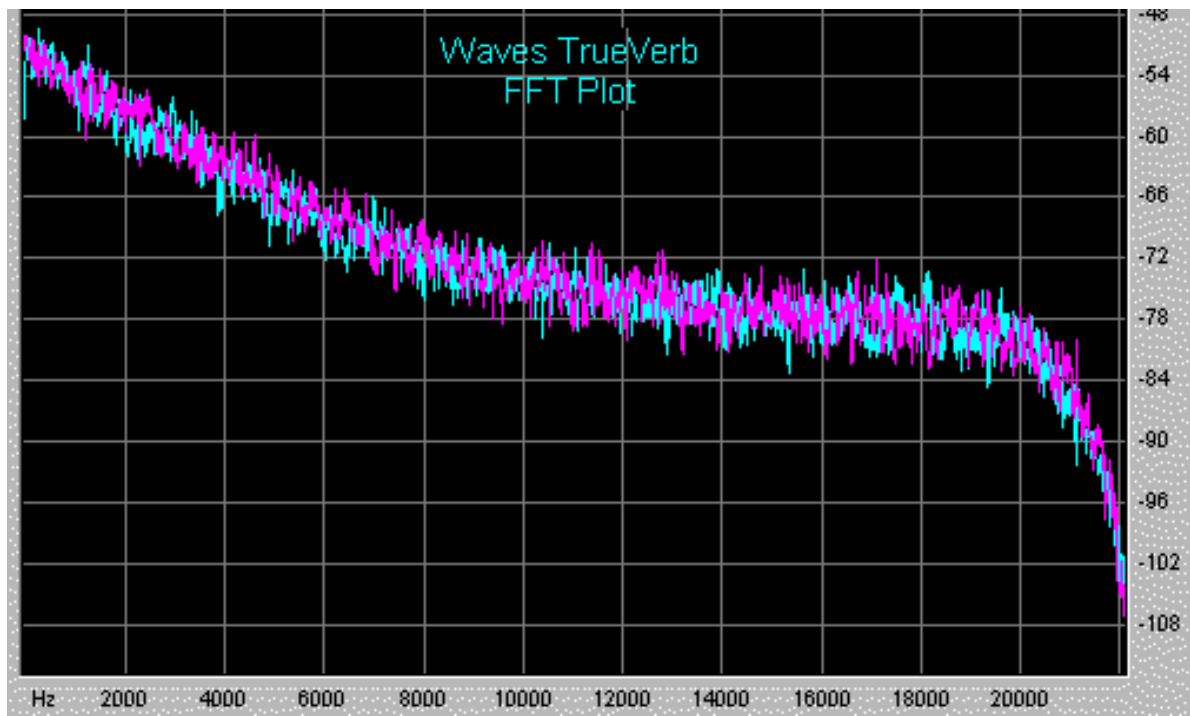
Pros:

- Moderately smooth decay of late reflections (i.e. minimal flutter)
- Late reflections are fairly colorless

Cons:

- Lack of echo density in the early reflections (causes "chatter" for some sounds)

TC Electronic M2000:

M2000 - Medium Hall
FFT Plot
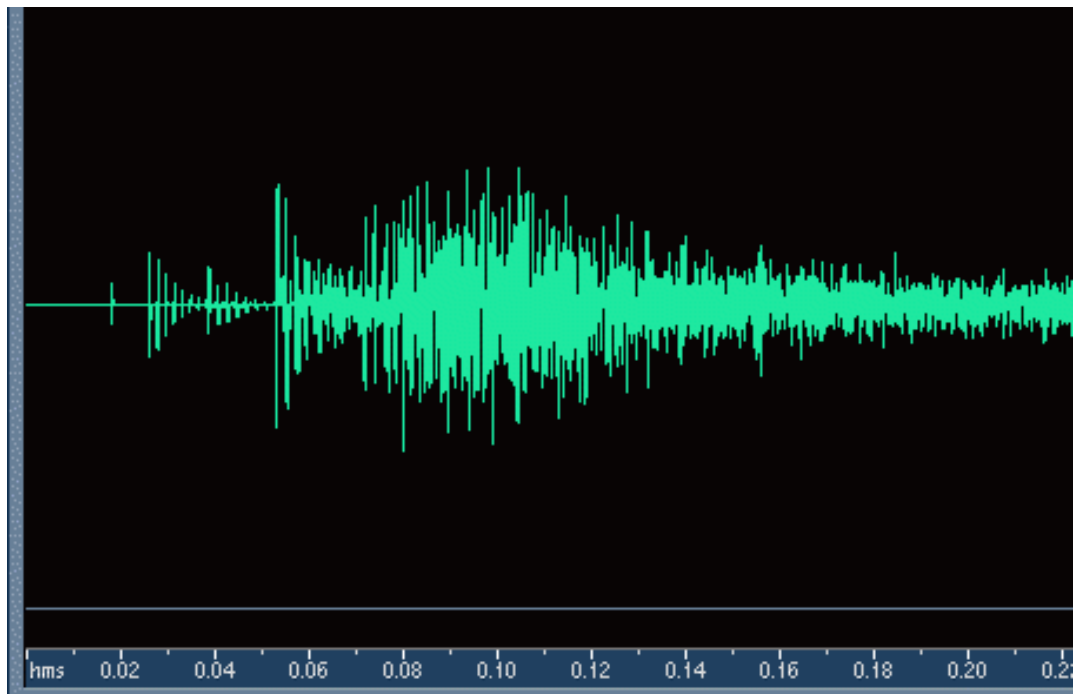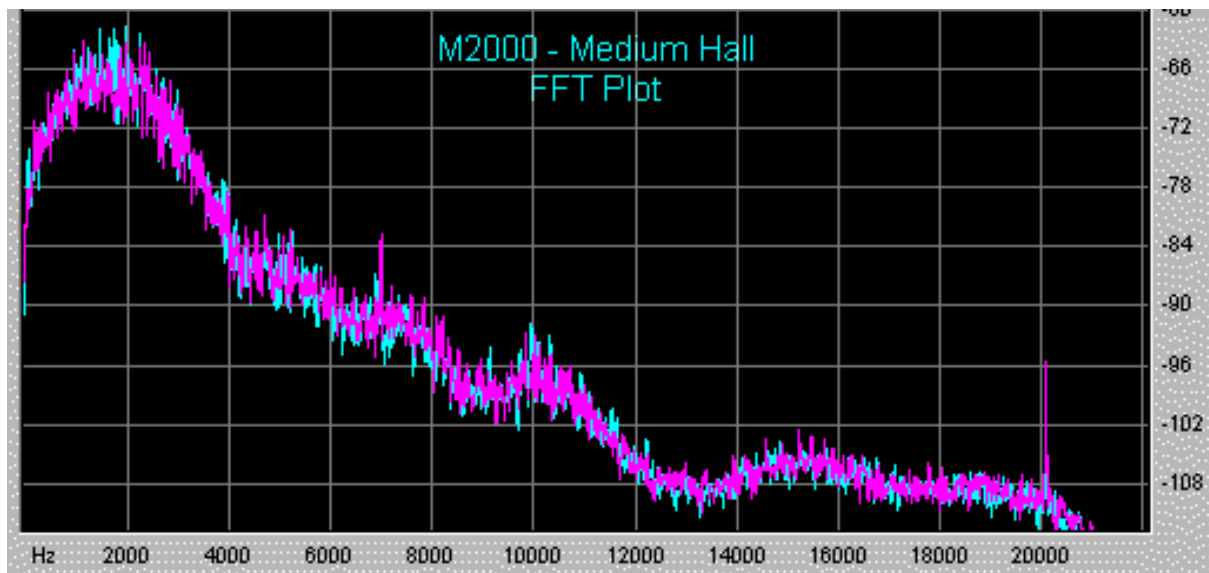
Pros:

- Very high density and diffusion
- Moderately smooth decay of late reflections (i.e.. minimal flutter)

Cons:

- Slightly colored late reflections (i.e.. ringy)

Lexicon 480L:

Lexicon - 480L - Small Hall
Impulse Response

Lexicon - 480L - Small Hall
FFT Plot
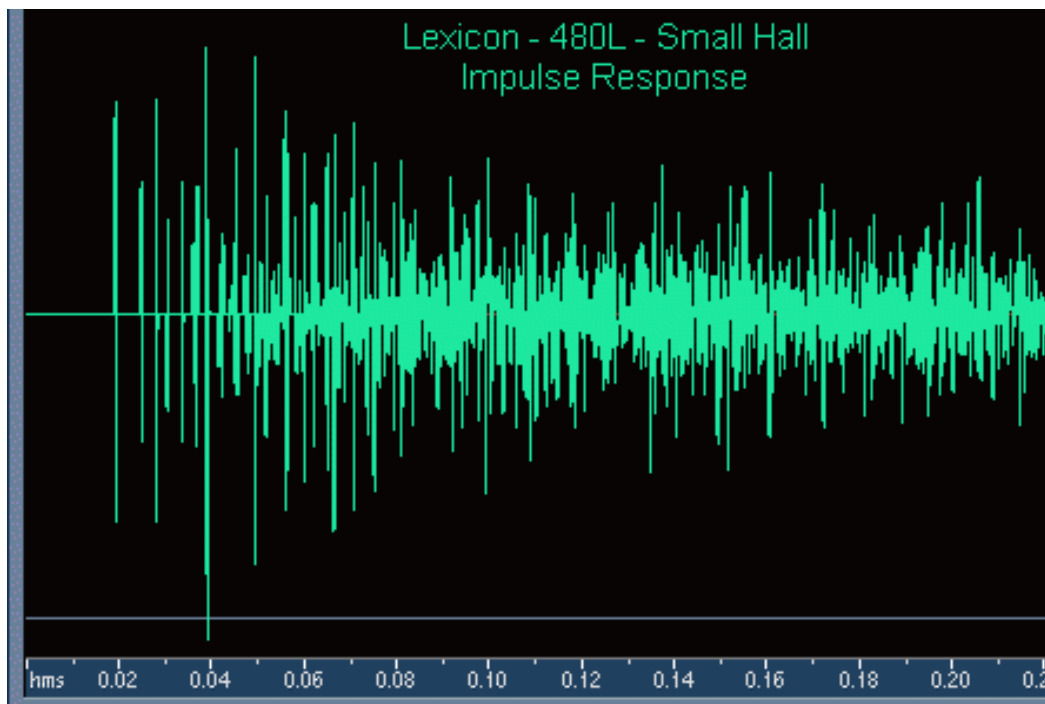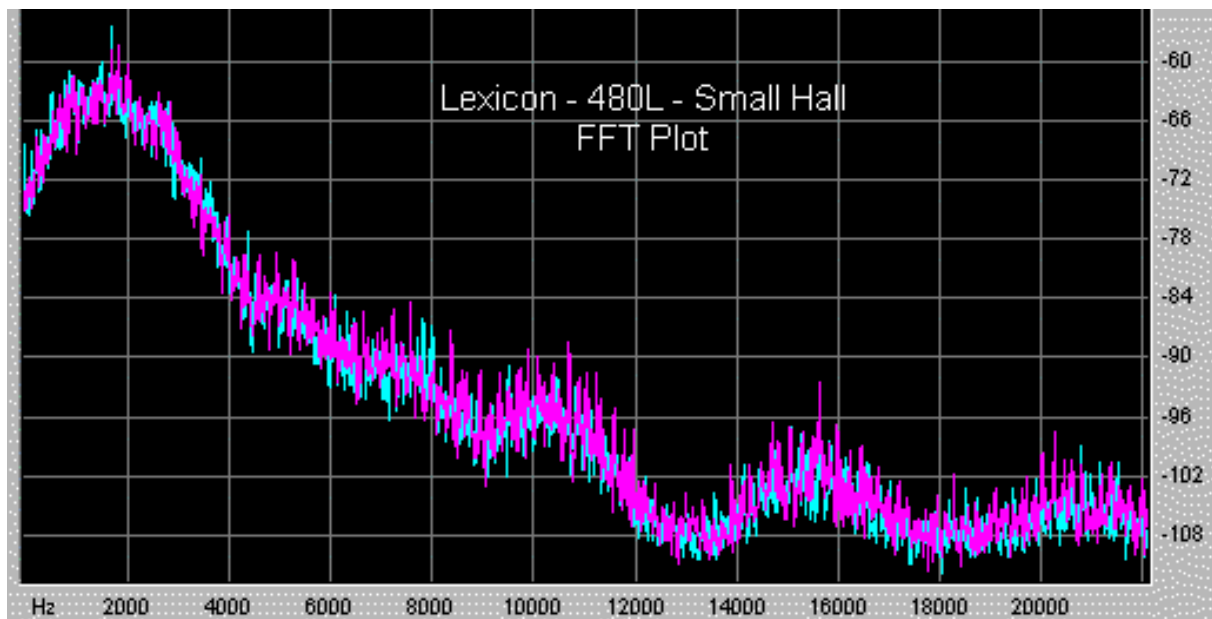
Pros:

- Drastic improvement in sound quality compared to most reverbs
- Sharp attack
- Smooth decay of late reflections (i.e. minimal flutter)

Cons:

- Slightly colored late reflections (i.e. resonant and ringy)
- Echo density could be higher

Arboretum Systems HyperVerb:

Arboretum Systems - HyperVerb
FFT Plot
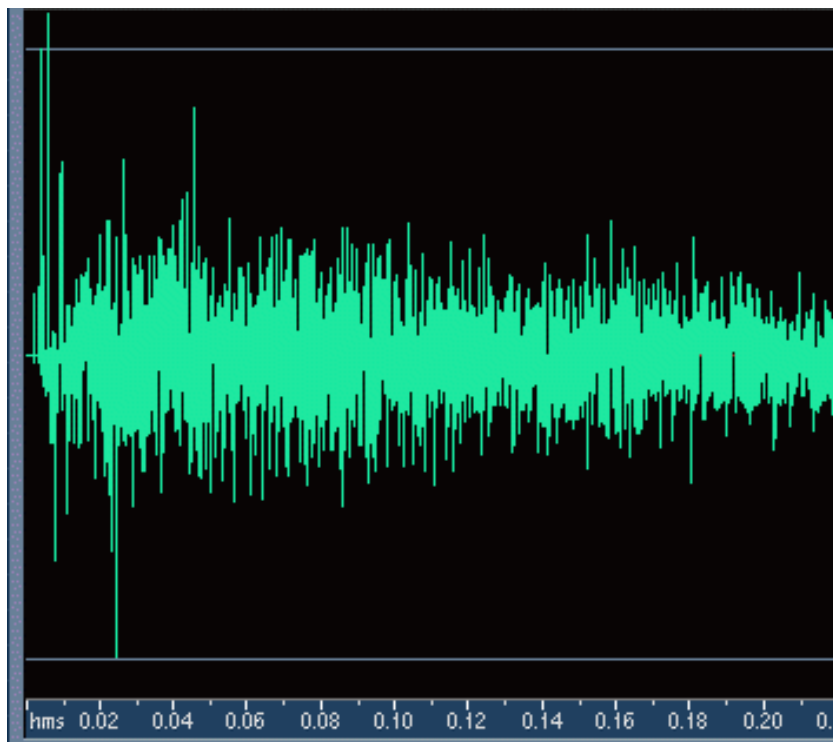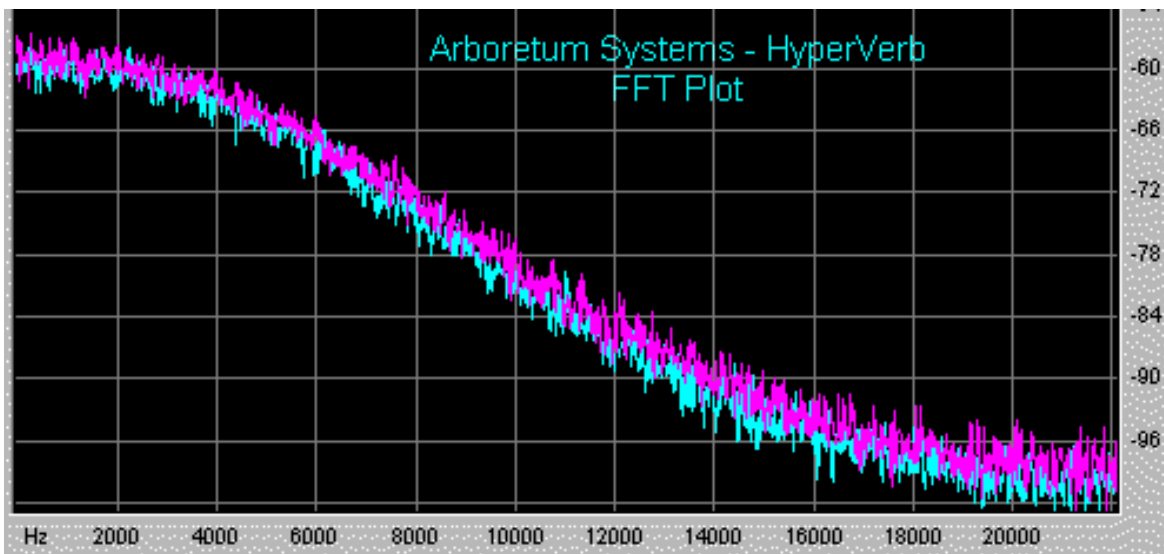
Pros:

- Very high echo density
- Echo density increases with time at a sufficiently high rate
- Smooth decay of late reflections (i.e. minimal flutter)
- Highly uncolored frequency response in the late reflections
- Sharp attack if desired

Cons:

- Could benefit from higher echo density for very large room sizes (easy to fix at the expense of somewhat higher processing requirements)

Of course none of this "scientific" data guarantees that the HyperVerb actually sounds better than any of these reverberators, so we welcome you to give it a serious listen. Although we don't claim that the HyperVerb is the ultimate digital reverberator, we think you will find the level of quality very high. If you have any questions or comments regarding the HyperVerb please contact Chris Weare at cweare@arboretum.com.

Impulse Response Files:

| File Name	Manufacturer	Product/Type	Preset
CoolEdit-ConcertHall.wav	Syntrillium	Cool Edit Pro	Concert Hall Lite
Digitech-GigaVerb.wav	DigiTech	GigaVerb	Studio Strat (GSP2101)
DSP-FX MediumHall.wav	Power Technology	DSP/FX	Medium Hall
DSP-FX MediumRoom.wav	Power Technology	DSP/FX	Medium Room
HyperVerb-MediumHall.wav	Arboretum Systems	HyperVerb	Medium Hall
HyperVerb-MediumHall2.wav	Arboretum Systems	HyperVerb	Medium Hall 2
HyperVerb-MediumHall3.wav	Arboretum Systems	HyperVerb	Medium Hall 3
HyperVerb-BrightLargeHall.wav	Arboretum Systems	HyperVerb	none
HyperVerb-StateCapitol.wav	Arboretum Systems	HyperVerb	State Capitol
Lexicon 480L-SmallStage2.wav	Lexicon	480L	Small + Stage (???)
Sound Forge-RichHall.wav	Sonic Foundry	Sound Forge	Rich Hall
TCWorks-NativeReverbDemo.wav	TC Electronic	Native Reverb	(fixed demo preset)
TrueVerb-MediumConcert.wav	Waves	TrueVerb	Medium Concert
WaveLab-HallReverb.wav	Steinberg	WaveLab	(???)

These impulse response files are available from ftp.arboretum.com/pub/demos/reverb/. we have not included the impulse response for the M2000 Hall since it is copyrighted by SEK'D. Lexicon 480L small stage impulse response file is also copyrighted by SEK'D, but it is

available as a demo file free of charge on the SEK'D web site at http://www.sekd.com/Download/irdemo.zip, although for your convenience we have provided it here.

Hyperprism Stereo Manipulation Processes

- [Pan](#)
- [Auto Pan](#)
- [Doppler](#)
- [Quasi Stereo](#)
- [Stereo Dynamics](#)
- [More Stereo](#)
- [M-S Matrix](#)

Pan

Pan is similar to the "balance" control on a home stereo, determining which channel the sound comes through. It moves the input sound from the left speaker to the right speaker, following the mouse position, and combines this with attenuation, or gain, if desired.

Hyperprism-VST users: Make sure you have your stereo tracks panned 100% left and right or you will get no effect with this process.

Pro Tools/Avid users: Note that there is no direct mono-to-stereo processing under the AudioSuite format. But you can easily work around this limitation by copying any mono selection to an adjoining track, then selecting both regions for stereo processing. The Hyperprism-DAS Pan effect requires two-channel input or a DAE error message -7450 will result when you attempt to process. Also note that Hyperprism stereo AudioSuite effects require the following settings: Use in Playlist (not Region) and be in Multi-Input Mode.

Parameters

Position (L <-> R)

Called LR Balance (%) in Hyperprism 2, this parameter controls the position of the sound in the plane from left to right. This parameter ranges from -100 to +100% where 0 corresponds to the center position in Hyperprism 2. In Hyperprism-VST, Hyperprism-MMP and Hyperprism-DAS this parameter ranges from 0 to 100%, with 50 being the center position.

Gain (dB)

This is a volume setting. Adjust attenuation or gain of the sound from -96 dB (silence) to 0 dB. The range is measured in percentage, with 100% equaling 0 dB.

Application Diagram

The following example assumes you're running Hyperprism 2 or Hyperprism-VST, which support parameter automation (otherwise you just hear the pan motion during preview in Hyperprism-MMP or Hyperprism-DAS). For a dramatic panning motion, start with the signal attenuated in one of the channels; as you pan, put in a crescendo as the sound passes through the center. The opposite effect, a decrescendo pan, may also be effective, since it creates a "hole in the middle" effect. Crescendo and Decrescendo pans
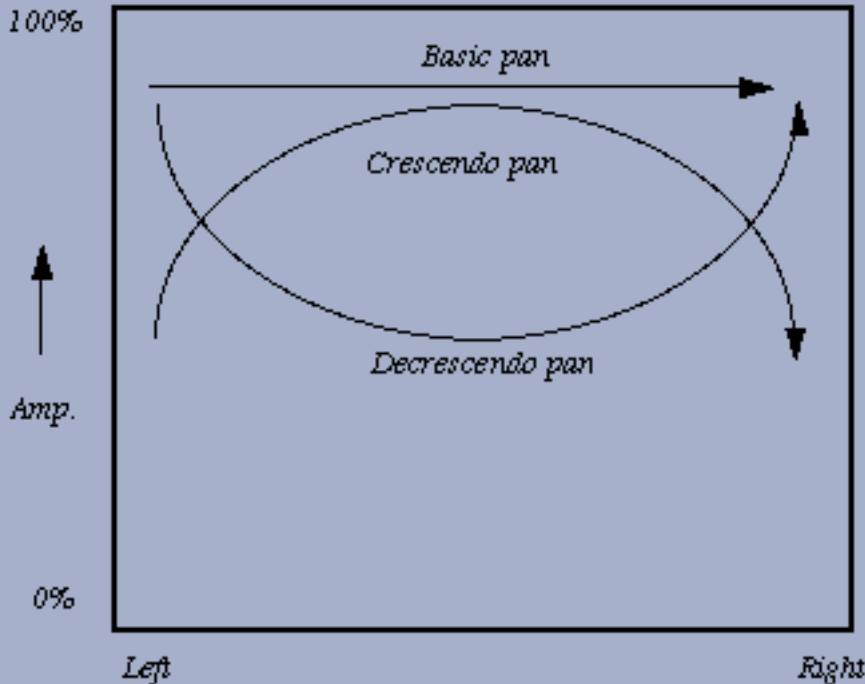
☞ This effect transforms a monaural input signal into a stereo output signal. Using the Hyperprism 2 Process to New File... command creates a stereo sound file as the output.

☞ The ability to apply gain to the original sound can be a very useful feature, since you can use your mouse to "ride the gain," bringing out quiet parts of a sound. However, be aware that whenever your gain is above zero dB, a loud passage in the sound you are processing could cause digital distortion. If this occurs, simply re-draw your path and add less gain at that point.

If you want to use this effect to change gain, but not left/right balance, set the horizontal balance axis to "0" at both ends, so you don't have to keep the mouse perfectly centered while drawing your gain adjustments. Alternately, you can hold down the shift key when drawing your paths. This constrains the motion of the Tracer to vertical or horizontal movements.

Auto Pan

| H 2 | VST | MMP G | DAS |

The Auto Pan effect moves a mono or stereo signal back and forth between the left and right channel. At lower speeds, the effect is one of a gradual pan between speakers. At higher speeds, the effect is similar to a rotating speaker sound. At extremely high speeds, the effect introduces interesting stereo amplitude modulation effects.

☞ **Hyperprism-VST users:** Make sure you have your stereo tracks panned 100% left and right or you will get no effect with this process.

Pro Tools/Avid users: Note that there is no direct mono-to-stereo processing under the AudioSuite format. But you can easily work around this limitation by copying any mono selection to an adjoining track, then selecting both regions for stereo processing. The Hyperprism-DAS Auto Pan effect requires two-channel input or a DAE error message -7450 will result when you attempt to process. Also note that Hyperprism stereo AudioSuite effects require the following settings: Use in Playlist (not Region) and be in Multi-Input Mode.

Parameters

LFO Frequency

This controls the speed at which the sound is panned back and forth between channels, from 0 to 20 Hz.

LFO Depth

The Depth setting controls how pronounced the effect is, by default from 0 to 100%. At 0%, there is no effect. At 100%, when the sound is being panned to the left channel, the right channel will be completely silent. At intermediate settings, the sound will not be completely muted in the "quiet" channel even at the instant is at it maximum level in the opposite channel.

Position

From 0% to 100%. In Hyperprism-DAS this parameter is called "Pan." Position determines the amount of time the panned signal will spend in the left or right channels. Technically speaking, this parameter varies the "duty cycle" of the low frequency oscillator that drives the Auto Pan effect. At a setting of 50%, the Auto Pan swings the sound equally between left and right channels in a sinusoidal pattern, much like a pendulum swings back and forth evenly about it's center.

However, Position settings below 50% will cause the sound to spend correspondingly more time in the left channel, thus appearing to alter it's "position" relative to the center of the stereo field. Position settings above 50% will cause the sound to spend correspondingly more time in the right channel. At extreme Position settings (near 0 or 100%), the sound appears only as a momentary "blip" in one channel or the other. This can sound especially interesting at high Depth and Frequency settings.

Doppler

The Doppler effect simulates a sound source passing by a listener. The source may be mono or stereo, both with stereo output resulting. The listener is always positioned at the center of the Blue Window, (X = 0, Y = 0). See the Doppler field diagram below. As a sound source moves towards the user, the sound source's perceived amplitude and pitch increase. As a sound source moves away from the user, the amplitude and pitch will decrease.

If you move a source away from the user, the pitch of the sound will slowly return to 0 pitch shift so that you can begin a new sweep back towards the listener. A standard doppler shift can be created by setting the Y Axis distance to about +10 meters (in front of the listener) and sweeping the X Axis parameter. Note that this may require high value of smoothing and a steady hand with the mouse to mimic the intended velocity of the source sound. Additional gain increase/decrease can be done with the input gain sliders, eg. to completely eliminate the sound source after a "fly-by." Remember that all parameter changes can be automated and saved to create repeatable, layerable effects.
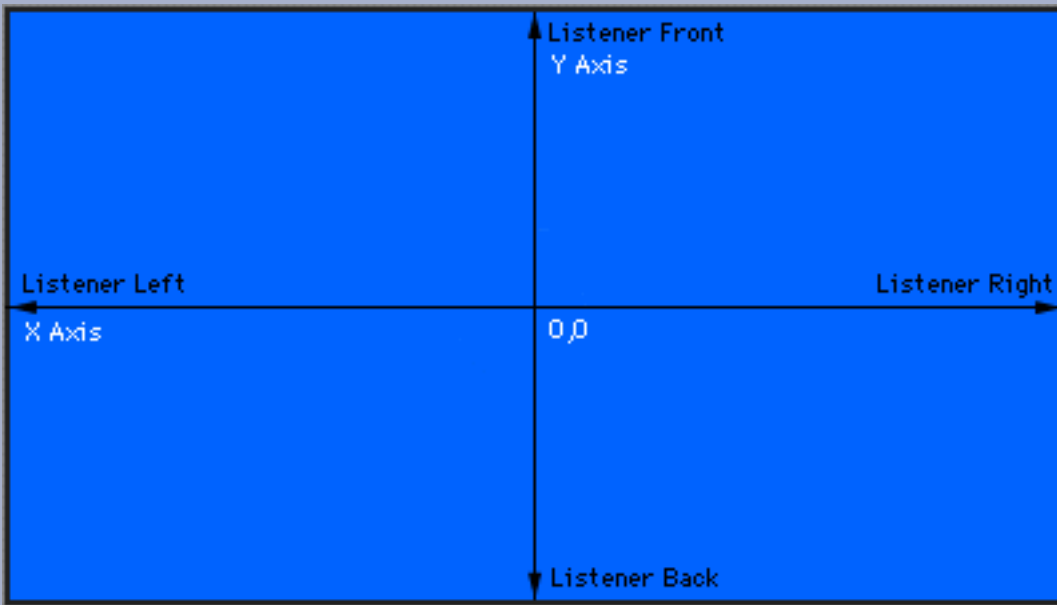
Hyperprism-VST users: Make sure you have your stereo tracks panned 100% left and right or you will get no effect with this process.

Parameters

X Axis

The X axis parameter controls the left/right distance from the listener. Negative distances are defined to be left of the user and positive distances to the right. This parameter ranges from -200 to 200 meters.

Y Axis

The Y axis parameter controls the front/back distance from the listener,. Negative distances are defined to be behind the user and positive distances to be in front of the listener. This parameter ranges from -200 to +200 meters.

Quasi Stereo

Quasi Stereo accepts a monaural input signal and produces a pseudo stereo two-channel output. It divides the mono signal path and applies complementary comb filters to each channel. These filters disperse their outputs to the left and right channels in such a way as to create the impression of a stereo image.
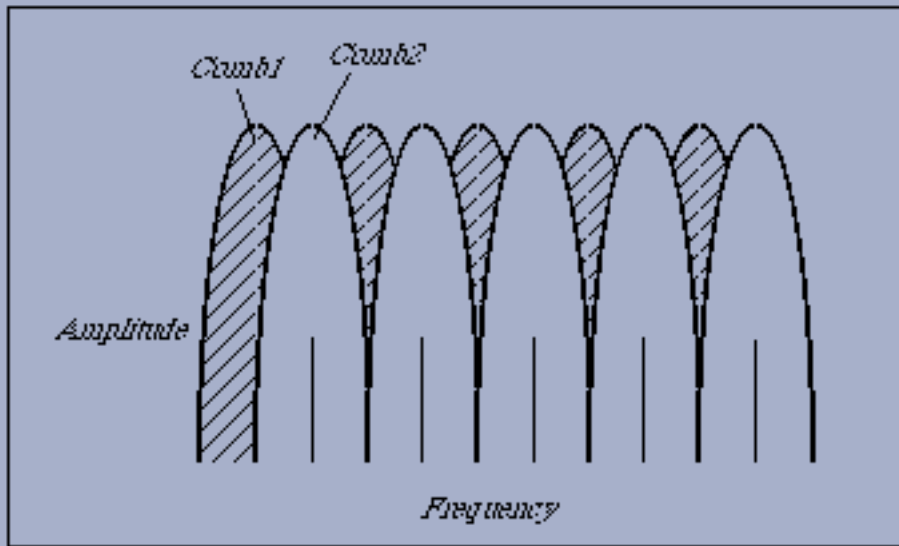
Complementary comb filters in Quasi Stereo

👉 **Hyperprism-VST users:** Make sure you have your stereo tracks panned 100% left and right or you will get no effect with this process.

👉 **Pro Tools/Avid users:** Note that there is no direct mono-to-stereo processing under the AudioSuite format. But you can easily work around this limitation by copying any mono selection to an adjoining track, then selecting both regions for stereo processing. The Hyperprism-DAS Quasi Stereo effect requires two-channel input or a DAE error message -7450 will result when you attempt to process. Also note that Hyperprism stereo AudioSuite effects require the following settings: Use in Playlist (not Region) and be in Multi-Input Mode.

Parameters

Depth

Width of the stereo image, where 0 means mono (the input is routed to both outputs with no change), +100 means 100% pseudo stereo.

Low Cut

Plug-in Pack users please note: This parameter is **mis-labeled** "High Cut" in Hyperprism-VST, Hyperprism-MMP and Hyperprism-DAS, it's **actually controlling Low Cut**. It sets the cutoff frequency for a 12 dB per octave high pass filter. At a value of 0 the effect is switched off. As you raise the value, a high pass filter attenuates low frequencies in the range up to 6000 Hz (up to 10 kHz in Hyperprism-DAS) before they are injected into the Quasi Stereo algorithm. These frequencies will still be present in the final signal. By using the filter, you can reduce low-frequency artifacts at the expense of a slightly narrower stereo image.

👉 For an interesting spatial effect, apply Quasi Stereo to a two-channel stereo input signal. The effect acts on only the mono (common) part of the signal. The discrete stereo channel information is added to the processed pseudo stereo signals to form the output.

Delay Time

The comb filtering used to create the pseudo-stereo image uses a short delay. The Delay Time slider lets you adjust the amount of the delay. A setting of 0 will cancel the effect, while higher settings make for progressively wider results. This control goes from 0 to 100 ms.

Stereo Dynamics

`H 2` `VST` `MMP G` `DAS`

Stereo Dynamics spatializes the input sound, positioning it in a virtual space situated in front of the listener. It moves the input sound from front to back and from left to right, depending on the position of the tracer in the Blue Window.

☞ This effect transforms a monaural input signal into a stereo output signal. Using the Hyperprism 2 Process to New File... command creates a stereo sound file as the output.

☞ **Pro Tools/Avid users:** Note that there is no direct mono-to-stereo processing under the AudioSuite format. But you can easily work around this limitation by copying any mono selection to an adjoining track, then selecting both regions for stereo processing. The Hyperprism-DAS Stereo Dynamics effect requires two-channel input or a DAE error message -7450 will result when you attempt to process. Also note that Hyperprism stereo AudioSuite effects require the following settings: Use in Playlist (not Region) and be in Multi-Input Mode.

☞ If you're using the VST, MMP (Premiere format) or DAS (AudioSuite format) versions of Hyperprism, the following parameter list is for you. If you're using **Hyperprism 2** for Power Mac, skip down to the special parameter list for your version; some of the ranges are different, giving you even more control over this effect...

☞ **Hyperprism-VST users:** Make sure you have your stereo tracks panned 100% left and right or you will get no effect with this process.

Parameters

Right-Left Gain

This is "right-minus-left gain," which controls the proportion of one channel relative to the other. For example, moving the tracer to the right will increase the proportion of the right channel in the output. The range is 0 to 100.

Right+Left Gain

This is "right-plus-left gain," controlling the overall volume level of both channels. The range is 0 to 100.

Application Diagrams

The figure below shows the layout of the play zone for the Stereo Dynamics effect, assuming R-L Gain is on the horizontal axis and R+L Gain is on the vertical axis.
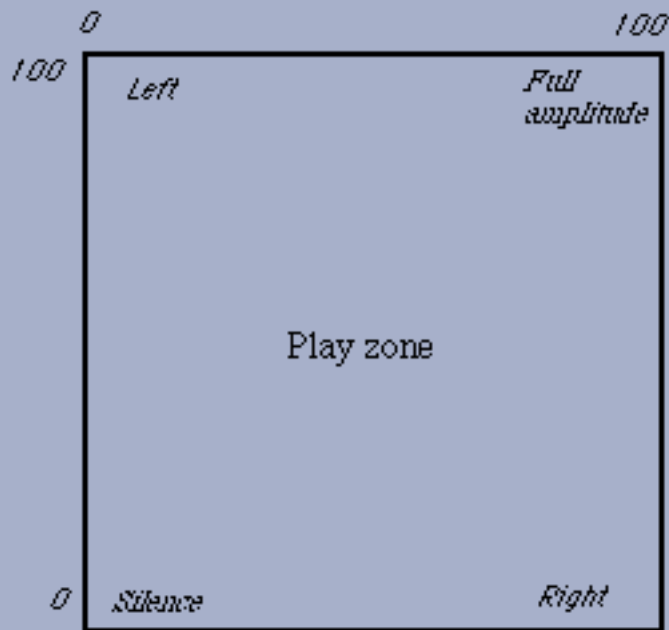
Default layout of the play zone for Stereo Dynamics in Hyperprism-VST, Hyperprism-MMP and Hyperprism-DAS

Notice in the figure below, that as you move the tracer from top left to bottom right, you pan the sound.
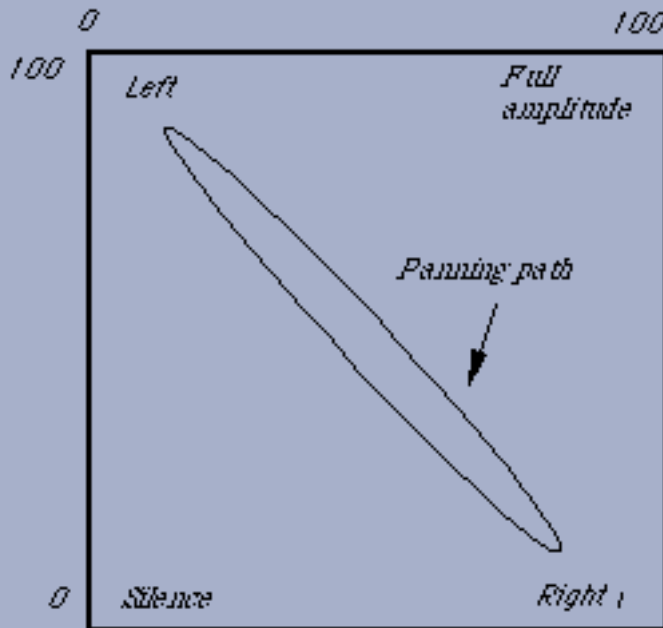
Tracer path for a panning motion

As you move from top right to bottom left, the signal is attenuated. So if you combine the two motions, as the following figure shows, the sound appears to spin in a circle in front of the listener situated precisely between two loudspeakers. This is also an excellent effect on headphones.
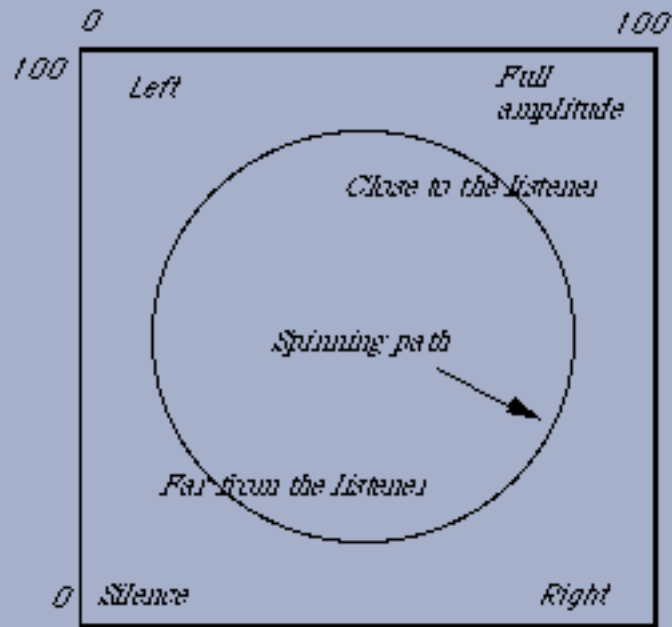
Tracer path for a spinning motion

Hyperprism 2 Stereo Dynamics Parameters

Right-Left Gain

This is "right-minus-left gain," which controls the proportion of one channel relative to the other. For example, moving the tracer to the right will increase the proportion of the right channel in the output. The range of values is -100 to 100.

Right+Left Gain

This is "right-plus-left gain," controlling the overall volume level of both channels. The range of values is -100 to 100.

Application Diagrams

The figure below shows the layout of the play zone for the Hyperprism 2 Stereo Dynamics effect, assuming R-L Gain is on the horizontal axis and R+L Gain is on the vertical axis.

	-100 %	0 %	100 %
100 %	Full Phased	Left	Full
0 %	Right Phased	Silence	Right
-100 %	Full	Left Phased	Full Phased
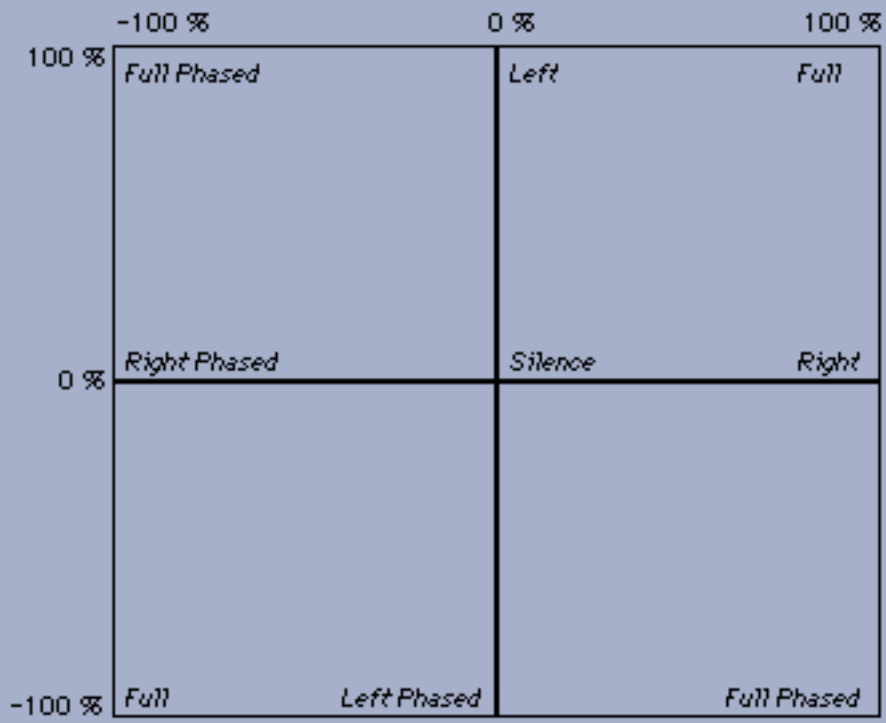
Default layout of the play zone for Stereo Dynamics in Hyperprism 2

Notice in the figure below, that as you move the tracer from top left to bottom right, you pan the sound.
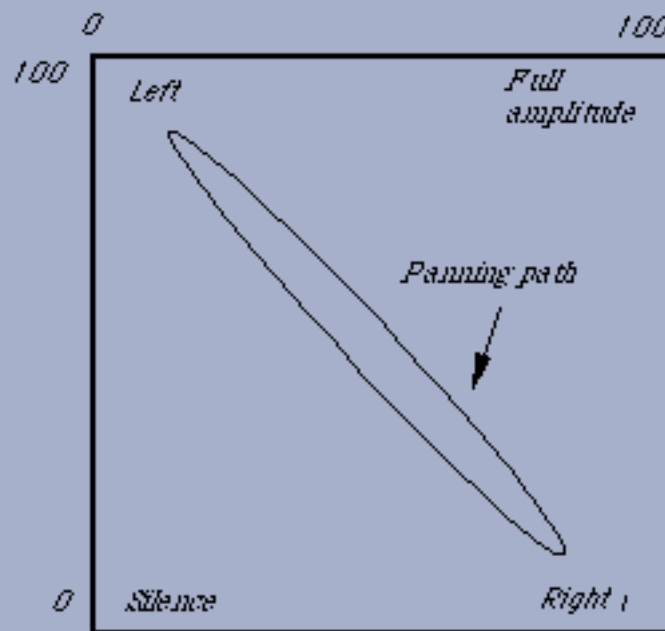
Tracer path for a panning motion

As you move from top right to bottom left, the signal is attenuated. So if you combine the two motions, as the following figure shows, the sound appears to spin in a circle in front of the listener situated precisely between two loudspeakers. This is also an excellent effect on headphones.
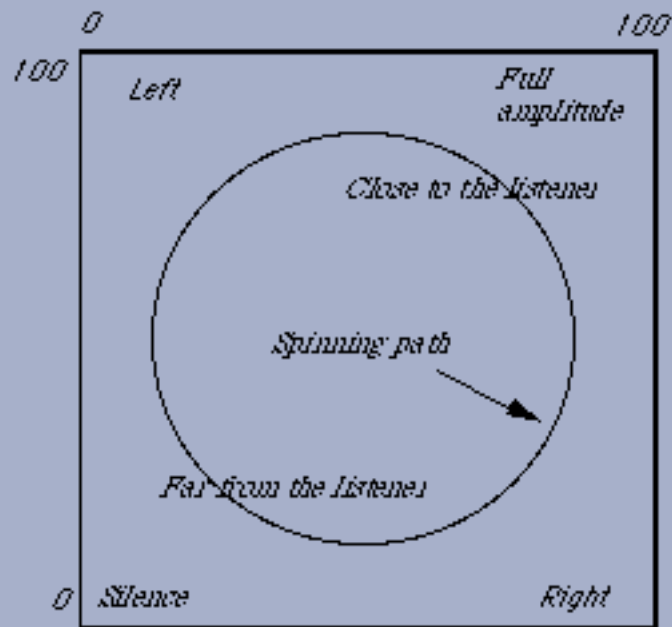
Tracer path for a spinning motion

More Stereo

`H 2` `VST` `MMP G` `DAS`

More Stereo alters the stereo effect by widening or narrowing the spatial image, without changing the overall volume. It optionally applies a variable-cutoff high pass filter to signal, reducing potential phase-cancellation artifacts in the low-frequency content of the original sound file. When using a file with very different left channel and right channel material, a third image is created in the center of the stereo field. When using a sound with similar material in both channels the stereo spread is enhanced.

☞ This effect requires a stereo file as input.

☞ **Hyperprism-VST users:** Make sure you have your stereo tracks panned 100% left and right or you will get no effect with this process.

☞ **Pro Tools/Avid users:** This effect requires two-channel input or a DAE error message -7450 will result when you attempt to process. Also note that Hyperprism stereo AudioSuite effects require the following settings: Use in Playlist (not Region) and be in Multi-Input Mode.

Parameters

Stereo Level

Width of the stereo image in percentage, where 0 means mono, 100 means normal stereo (no change to the input). 200 doubles the proportion of the discrete channel information in the output by attenuating the common or mono part of the signal, and 300 triples it. A horizontal value greater than 100 has the effect of widening the stereo image, pushing the sound to the extreme lateral poles. This parameter goes up to 1000 in Hyperprism 2, Hyperprism-VST and Hyperprism-DAS, up to 300 in Hyperprism-MMP.

Low Cut

This parameter is called "Cutoff Frequency" in Hyperprism-DAS and Hyperprism-VST. It controls the cutoff frequency for a 12 dB per octave high pass filter. At the bottom of the Blue Window (a value of 0) the effect is switched off. As you raise the position of the tracer, a high pass filter attenuates low frequencies in the range up to 6000 Hz before they are injected into the widening algorithm. These frequencies will still be present in the final signal. By using the filter, you can control low-frequency artifacts at the expense of a slightly narrower stereo image. Ranges from 13 to 22,050 Hz in Hyperprism 2 and Hyperprism-VST, 0 to 20 kHz in Hyperprism-DAS.

Example

▶ Load any stereo file from your library of sounds. Play it and then select More Stereo in the Processes menu.

▶ Click on Loop playback if necessary.

▶ Move the Stereo Level slider from left (0) to right (300). At the left-most position, the output is monaural, with the contributions from both channels mixed and sent to both outputs. In the center position (100) the stereo sound file is played back unchanged. In the extreme right position (300) the stereo image widens to the left and right, with less sound in the middle.

Limitations

More Stereo has no effect on mono input signals. It treats only the discrete stereo part of the signal (that part of the signal that is not common to both channels).

M-S Matrix

H 2   MMP G   DAS

M-S refers to either Mid-Side or Mono-Stereo. This classic stereo recording technique has the advantage that one can change the apparent width of the stereo image after the recording session, using a device called an M-S decoder. In the past, an M-S decoder took the form of an external analog circuit. The M-S Matrix effect replaces this circuit, making experimentation with the M-S technique as easy as tracing a line in the Blue Window. The description below tells you how to use Hyperprism's M-S decoding facilities.
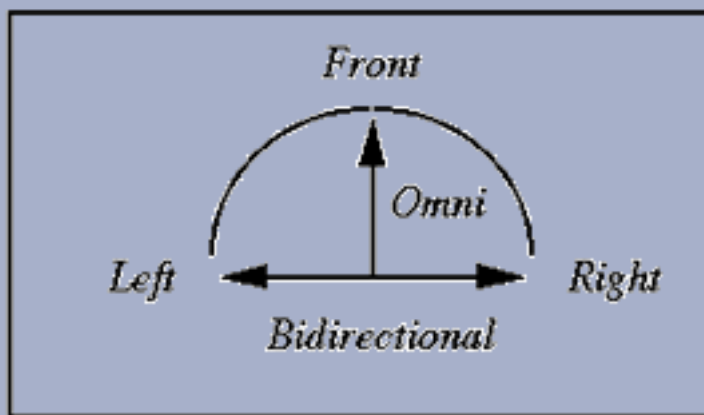
M-S microphone configuration

Theory of M-S Recording

In order to perform M-S decoding you first need to make M-S encoded recordings. This requires two microphones. One of these microphones faces front, toward the audio source. Its polar pattern is usually cardioid. (A variation on the M-S technique uses an omnidirectional microphone in the center.) The other microphone is bidirectional (also called

figure-eight). The cardioid microphone captures the middle or mono part of the stereo image. The bidirectional microphone picks up the sides or discrete stereo part of the image. You record the output of these microphones on a two-channel recorder, with one channel called M and the other S, as opposed to the usual left (L) and right (R).

Assuming you have made an M-S recording, connect the M output of your recorder to Channel 1 of your Digidesign system, and connect the S to Channel 2.

 Pro Tools/Avid users: The Hyperprism-DAS M-S Matrix effect requires two-channel input or a DAE error message -7450 will result when you attempt to process. Also note that Hyperprism stereo AudioSuite effects require the following settings: Use in Playlist (not Region) and be in Multi-Input Mode.

Parameters

Stereo Position

Displacement of the stereo image in degrees [-90, +90], where the center position 0° means no change. This can be useful in processing sound tracks recorded on location where the dialog is off-center, and one wants to restore it to the center position. This parameter is labeled Center Position in Hyperprism 2, Hyperprism-MMP and Hyperprism-DAS.

 This parameter maintains constant power at any position, so that at extreme left or right positions the level is boosted. If this causes distortion you can compensate by lowering the input level by 3 dB in the Settings window.

Side Volume

Gives you control over the level of the stereo part of the signal. In M-S processing, the higher the stereo level, the wider the apparent stereo image. The default range is 0% to 150%, which corresponds to a range from -96 dB to +6 dB. This parameter's range is from 0 to 100% in Hyperprism 2, Hyperprism-DAS and Hyperprism-MMP.

Input

Middle Left and Side Left. By default M-S Matrix assumes that the signal going into the "Left" input jack is the Mid part of the M-S signal. By selecting Left = Side, the program assumes that the left input is the Side part of the M-S signal. Click on this button to switch input settings.

Output

This button can correct for an out-of-phase microphone configuration. It determines the phase of the M source. M+S in Left is the normal case; the M input sums positively. Click on this button to switch output channels.

Example

 Load the example sound file Clifden Beach, Ireland. This is an M-S encoded file. Enable Loop playback

 Select M-S Matrix from the Plug-ins menu.

According to the sound engineer who made this recording, the normal stereo effect is rendered when the center position is 0 degrees and the Side Volume is 30%. Do this and listen to the sound for a while. Now decrease the Side Volume. As you drag toward the bottom of the range you'll hear the sound gradually becoming mono.

▐▐▶ As you boost Side Volume above 30 you'll hear the sound image becoming wider and wider.

▐▐▶ With the Side Volume up high, move the Center Position control. The apparent "center" of the stereo image shifts to the right and left, just as if you were readjusting the position of the M microphone during a recording session.

Spatial Patterns Obtainable with M-S Decoding

The polar diagrams below show the range of spatial patterns obtainable from a single M-S recording using M-S Matrix and Hyper Stereo (if included in your Hyperprism package). The captions list the X and Y positions of the tracers followed by a figure in decibels that indicates the level of the S part of the signal relative to the M part.
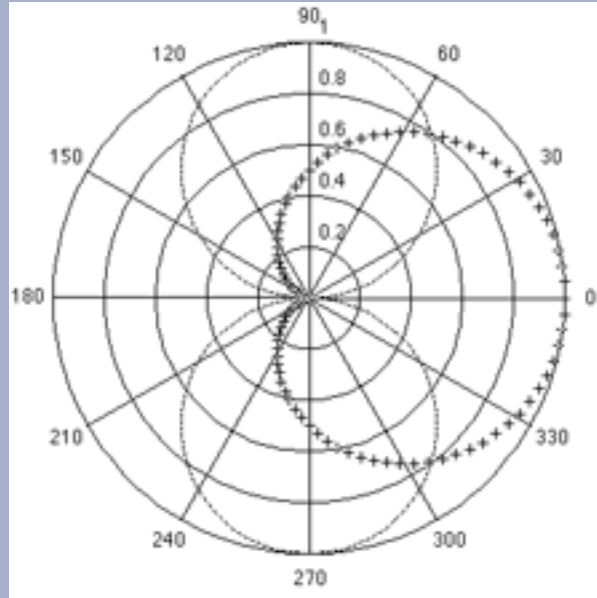
Directivity patterns for a bi-directional (--) and cardioid (++) microphone in M-S recording configuration. This reflects the physical configuration of the microphones in the original recording session.
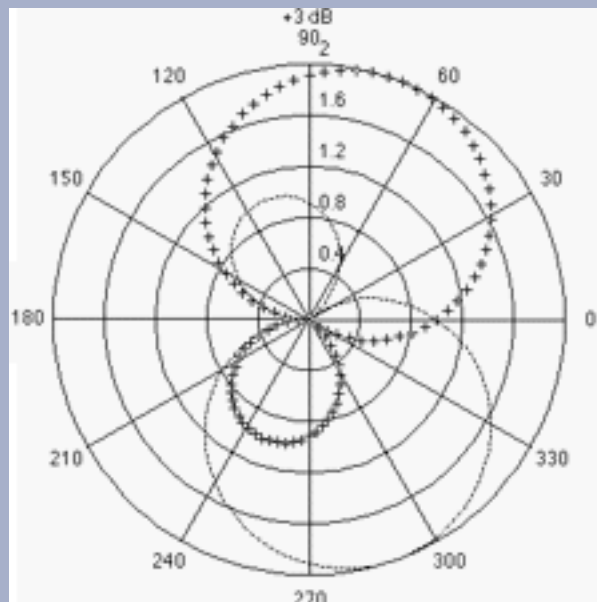
Hyper Stereo: Stereo +100%, Mono 50%,M-S Matrix: pattern at +3 dB
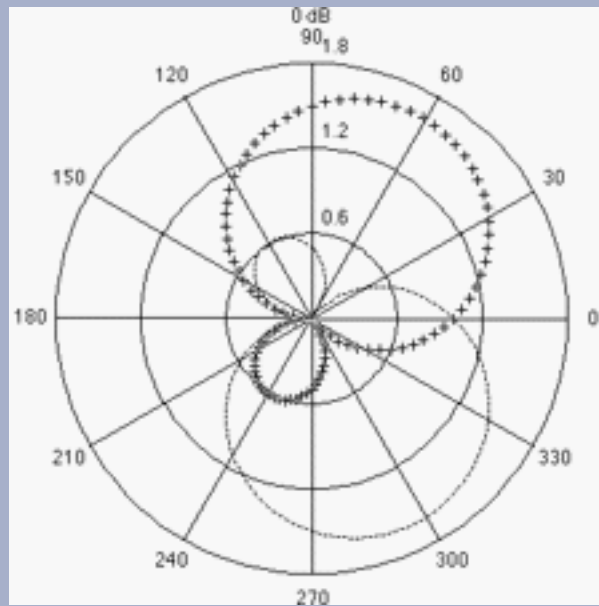
Hyper Stereo: Stereo +100%, Mono +100%, M-S Matrix: pattern at O dB (equal level)
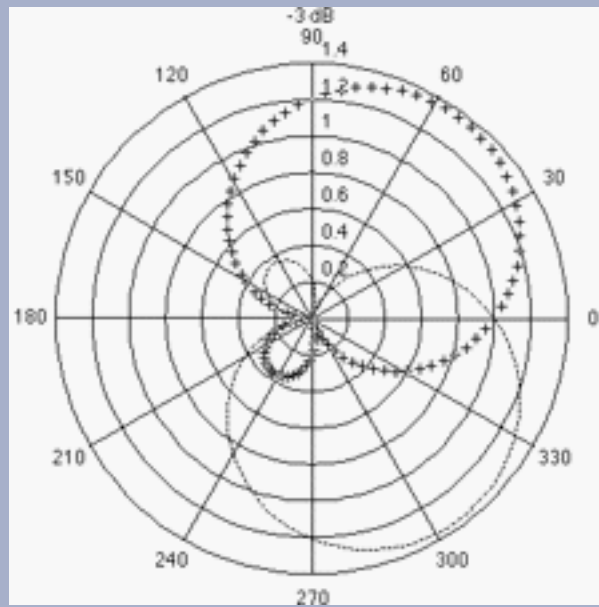
Hyper Stereo: Stereo +71%, Mono +100%, M-S Matrix: pattern at -3 dB
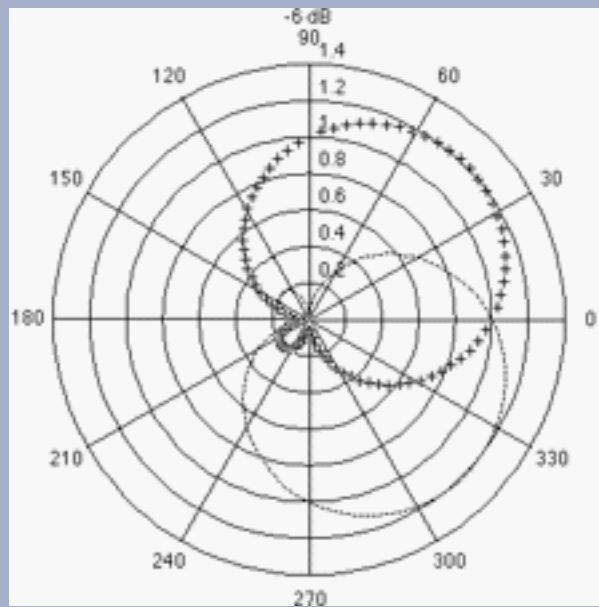
Hyper Stereo: Stereo +50%, Mono +100%, M-S Matrix: pattern at -6 dB
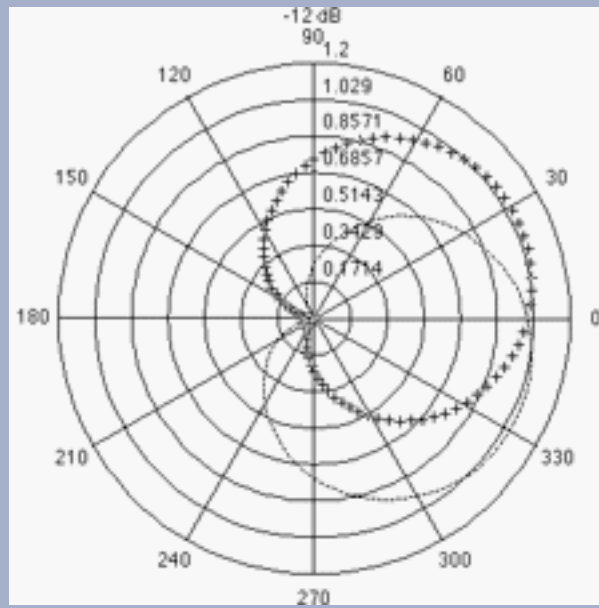
Hyper Stereo: Stereo +25%, Mono +100%,M-S Matrix: pattern at - 12 dB
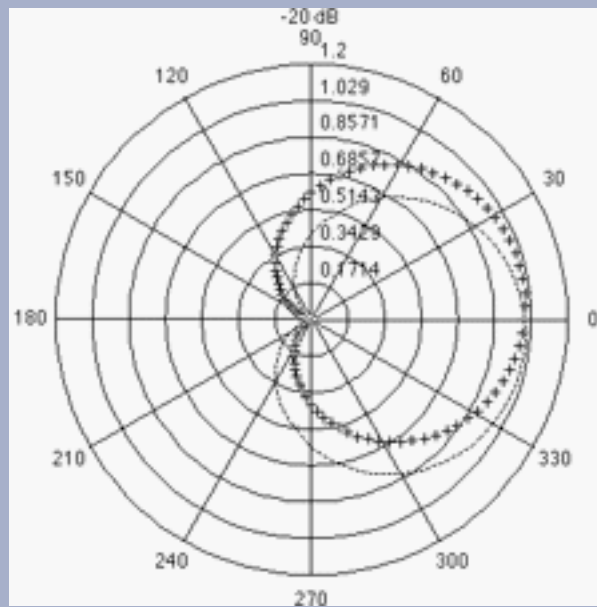
Hyper Stereo: Stereo +10%, Mono +100%, M-S Matrix: pattern at -20 dB

☞ M-S Matrix has no effect on mono input signals but interesting effects can be obtained with rich frequency spectrum stereo signals.

Go to Effects: [Filters] [Modulators] [Delay and Reverb] [Stereo] [Misc Effects]
Go to Table of Contents: [Hyperprism-VST]

[Troubleshooting] [About Digital Audio]

Hyperprism Miscellaneous Processes

- [Harmonic Exciter](#)
- [Bass Maximizer](#)
- [Tube/Tape Saturation](#)
- [Pitch Time Changer](#)
- [Pitch Changer](#)
- [Vari-Speed](#)
- [Noise Gate](#)
- [Compressor](#)
- [Limiter](#)
- [Sonic Decimator](#)

Harmonic Exciter

Have you ever had an audio track that sounded dull and boring, which no amount of EQ could fix? Add a bit of excitement to your audio with Arboretum's Harmonic Exciter. Instead of simply boosting frequencies already present, as occurs with EQ, the Harmonic Exciter generates new high frequency harmonics, thereby extending the bandwidth and increasing the liveliness of the original audio.

A problem with many exciter algorithms is that the harmonic generator either does not add a significantly high level of audible harmonics or adds very unpleasant artifacts, such as harsh and gritty distortion to the signal path. With Arboretum's Harmonic Exciter great care has been taken to maximize the amount of harmonic generation, yet greatly minimize the amount of artifacts. Also, the Harmonic Exciter's phase compensated side chain avoids any phasing and frequency notching problems commonly associated with most exciter algorithms. In addition the Exciter algorithm avoids most of the aliasing artifacts that are problematic with most digitally based exciters. This is done by substantially raising the internal sampling rate of the plug-in.

Effect Usage:

The most important thing to remember when using the Harmonic Exciter is that the quality of the results are highly signal dependent. For example, certain settings may sound great on a drum track, but harsh and distorted on an acoustic guitar track. Often a fair amount of experimentation is required to achieve good results, especially because the parameters interact considerably.

A general guideline to setting the effect, is to adjust the Dynamics to around 70% and the Crossover between 2000 to 4000 Hz. For the Harmonics Type set it to "Odd and Even" for a warmer sound or "Odd" for a brighter sound. Then raise the Harmonics slider towards the maximum level, but if any unpleasant distortion is heard, then stop and back down the slider just a bit. Then adjust the Spectral Mix so the high and low frequencies are well balanced. Generally the more harmonics

added the lower the Spectral Mix should be set. Then readjust the Dynamics and Crossover sliders as desired.

For something with a delicate harmonic structure such as an acoustic guitar, it may be a good idea to set the harmonics fairly low (e.g. less than 50%) and the crossover fairly high (e.g. 3000Hz). On the other hand, for audio such as a rock drum kit, it maybe more important to maximize the harmonic generation, by setting the Harmonics very high (e.g. 200%) and the Crossover fairly low (e.g. 1500Hz). Also, for speech lowering the Dynamics slider may help minimize the exaggeration of sibilance.

The Quality Level control within the Harmonic Exciter allows the user to adjust the amount of internal sample rate increase. Settings of "Good", "Better" and "Best" correspond to successively higher internal sample rates and therefore higher levels of quality. The difference in sound quality between "Good" and "Best" range from barely noticeable to fairly obvious depending upon the sound being processed. Sounds with little high frequency content may have little problem with aliasing, whereas sounds with lots of high frequency content could have lots of aliasing artifacts. Since increasing the Quality Level also greatly increases the CPU usage, it is not worth setting the Quality Level higher than needed.

For a more natural sounding effect we recommend you use the Harmonic Exciter on a relatively dry signal, and then add reverb or other effects afterwards. Although if natural is not what you're looking for, go ahead and try adding lots of reverb or other effects first. We also recommend you process each track of a mix separately, instead of processing the entire mix. The advantage of processing each track individually is that the settings can be tailored to each specific sound and inter-modulation distortion and other artifacts will generally be minimized. Certainly good results can be achieved by applying the Harmonic Exciter to an entire mix, but much more care should be taken not to over do the effect.

Parameters

Harmonics

Controls the amount of harmonics generated, with 0 to 100 percent being the nominal range. For extreme harmonic generation the harmonics slider can be set as high as 200 percent. Adjusting the Harmonics will affect the balance of the low and high frequency, so it may be necessary to adjust the Spectral Mix to compensate.

Dynamics

This novel feature adjusts the dynamics of the harmonic generation algorithm. For example, higher settings will cause the Exciter to mainly add harmonics to high amplitude transient signals only; lower settings will cause the harmonics to be applied more evenly to both low and high amplitude signals which will generally increase the overall harmonic content. The range is from 0 to 100%. Adjusting the Dynamics will sometimes affect the balance of the low and high frequency, so it may be necessary to adjust the Spectral Mix to compensate.

Crossover

Ranges from 500 to 10000Hz. Adjusts the frequency at which the harmonic generation occurs. For example, if the crossover is set to 2000Hz, then all frequencies above 2000Hz will have harmonics added.

Spectral Mix

Ranges from -24.0 to 12.0dB. Sets the gain of the frequency components which are above the crossover point.

Harmonic Type

Choices are "Odd and Even (Warm)" and "Odd (Bright)."

Quality Level

Choices are Good, Better and Best. Adjusts the accuracy of the exciter effect, by increasing the internal sample rate at which processing occurs. On the down side, raising the quality level will significantly increase processor utilization.

Bass Maximizer

```
VST   MMP G   DAS
```

The Bass Maximizer plug-in offers a wide assortment of bass enhancement and maximization effects, ranging from subtle bass enhancement to extreme bass generation. The harmonic generation feature increases the perceived amount of bass, even on low end systems with poor bass frequency response. It does so by adding harmonics which fool our brains into hearing lower frequencies than may exist in the original sound. The compression feature allows even more bass to be added without the risk of clipping or system overload.

Effect Usage:

The usage of the Bass Maximizer is relatively straightforward. A good starting point is to set the Crossover to be between 50 and 100Hz, the Low Cut to less than 20Hz, and the Resonance to less than 1.0. Then increase the Harmonics to the desired level, and add some compression. Finally adjust the Mix to balance out the low and high frequencies as desired. If deep bass is required set the Resonance to around 3.0, increase the Low Cut parameter to about 40Hz, and set the compression to around 10-20. The Bass Maximizer can be used effectively to process individual tracks, or an entire mix.

Parameters

Compression

Ranges from 0.0 to 20.0dB. Controls the amount of compression applied to the bass frequencies.

Harmonics

Controls the amount of harmonics generated, with 0 to 100 percent being the nominal range. For extreme harmonic generation the harmonics slider can be set as high as 200 percent.

Crossover

Ranges from 40 to 400Hz. Adjusts the frequency below which the harmonic generation and compression occur. For example, if the crossover is set to 100Hz, then all frequencies below 100Hz will have harmonics and compression added.

Low Cut

Ranges from 5.0 to 200.0Hz. The Low Cut feature rolls off frequency content below the specified point. This control is useful when the Bass Maximizer generates so much very low frequency content that the intended reproduction system becomes overloaded. For example, if the intended system is a typical multimedia computer which can't accurately reproduce frequencies below 70Hz, then set the Low Cut to around this frequency.

Resonance

Ranges from 0.5 to 5.0. Controls the amount of resonance applied to the low cut filter. This feature effectively adds an extra boost at the Low Cut frequency. For typical material that does not require resonance a setting of roughly 0.7 is a good starting point. For bass heavy musical styles, a setting of 3.0 to 5.0 may work best.

Mix

Ranges from 0.0 to 100.0%. Sets the mix between the frequency components which are above and below the crossover point.

Harmonic Type

Choices are "Odd and Even" and "Odd". Odd and Even gives smoother bass enhancement, while Odd harmonics will give a tighter bass sound.

Tube/Tape Saturation

[VST] [MMP G] [DAS]

Overview

The Tube/Tape Saturation plug-in has been designed to accurately model the non-linear saturation characteristics of a wide range of high-end tube amplifiers and pre-amplifiers, as well as analog reel-to-reel tape recorders. This plug-in does not model various other aspects related to tape recorders or tube amplifiers, such as high frequency roll-off or tape hiss. The Tube/Tape Saturation plug-in gives you warm, smooth tube and tape characteristics, all while staying within the digital domain.

The Tube/Tape Saturation plug-in also increases the internal processing sample rate so that aliasing artifacts are greatly reduced, something that many digitally based distortion effects neglect to do. The process of distortion adds harmonics to the signal. What happens if a harmonic is generated which has a frequency greater than the highest frequency that can be sampled (i.e. 1/2 the sample rate)? The answer is, the harmonic is folded back into audible frequency range, creating a very unpleasant type of distortion called aliasing distortion.

Effect Usage:

To produce subtle tube or tape saturation, generally adjust the Gain to between 3.0dB and 10.0dB and set the Warmth to around 80 to 90 percent. It should be noted that even a Gain of 0dB or less can often produce a considerable amount of distortion, so don't over do it. It is also generally recommended to perform the saturation before any other processing such as reverb or limiting, although certain effects such as EQ, wah wah filters, or subtle compression may work well when applied pre-saturation. But given that these are just general guidelines, certainly don't be afraid to experiment.

The Quality Level control allows the user to adjust the level of internal sample rate increase. Settings of "Good", "Better" and "Best" correspond to successively higher internal sample rates and therefore higher levels of quality. The difference in sound quality between "Good" and "Best" range from barely noticeable to fairly obvious, depending upon the sound being processed. Sounds with little high frequency content will have little problem with aliasing, whereas sounds with lots of high frequency content will have lots of aliasing artifacts. Since increasing the Quality Level also greatly increases the CPU usage, it is not worth setting the Quality Level higher than needed.

This plug-in also offers the following 4 saturation types: Tube Type1, Tube Type2, Tape Type1 and Tape Type2. (Try saying that out loud a few times...)

The tube emulations offer a significantly different harmonic structure than the tape emulations, with the tape emulation generating more higher order harmonics than the tube emulation. Tube Type1 and Tape Type1 emulate slightly more compressed tube and tape distortions, respectively, whereas Tube Type2 and Tape Type2 emulate slightly warmer, more dynamic distortions.

Another very important thing to remember when using the Tube/Tape Saturation plug-in is that because it is a non-linear effect, different results will occur by processing each track of a mix separately or processing the entire mix together. It is generally recommended that each track be processed individually. This allows the saturation settings to be optimized for each track, and generally reduces intermodulation distortion and emphasizes harmonic distortion. Certainly good results can be achieved by applying the Tube/Tape Saturation plug-in to an entire mix, but much more care should be taken not to over do the effect. Also, when applying Tube/Tape Saturation to an entire mix, it is generally best to make sure the mix has not already been limited. Not doing so will often cause ugly artifacts when the Tube/Tape Saturation is applied.

Another use for the Tube/Tape Saturation plug-in is in place of a limiter or level maximizer. Limiters or level maximizers,

while generally being able to reduce the maximum amplitude with minimal distortion, often have the unpleasant artifact of squishing sound until it is dull and lifeless. On the other hand, the Tube/Tape Saturation plug-in can reduce peak level yet retain, and even add to, the punchiness of the original audio track, while often sounding warm and not overly distorted. The Tube/Tape Saturation will not work well as a level maximizer for some sounds, such as a string section, but for other sounds, such as a rock drum kit, it can be quite effective. Because Tube Type1 and Tape Type1 saturation types have higher compression characteristics, it is recommended that they be used instead of Tube Type2 and Tape Type2, when trying to perform level maximization.

Parameters

Warmth

Ranges from 0.0 to 100.0%. Controls the amount of harshness or smoothness of the saturation effect. With 0.0% tending towards the harsh side and 100.0% being smooth and warm, like a high-end tube distortion.

Gain

Ranges from -6.0 to 50.0dB. Adjusts the amount of saturation produced by the effect. For subtle tube or tape saturation stay within the 3.0 to 10.0dB range. For outrageous distortion effects try a gain of 50.0dB.

Saturation Type

Choices are Tube Type1, Tube Type2, Tape Type1 and Tape Type2.

Quality Level

Choices are Good, Better and Best. Adjusts the accuracy of the saturation effect, by increasing the internal sample rate at which processing occurs. On the down side, raising the quality level will significantly increase processor utilization.

Pitch Time Changer

`H 2`

The Pitch Time Changer lets you vary the playback speed (duration) and the pitch of the input signal independently.

For pitch-shifting, the algorithm works by loading memory with an incoming signal sampled at rate A and reading out the samples at rate B. The ratio A/B determines the pitch change. To maintain a continuous output signal, samples must be repeated (for upward pitch shifts) or skipped (for downward pitch shifts). Because the output address pointer repeatedly overtakes the input address pointer (for pitch increases) or is overtaken by the recirculating input address pointer (for pitch decreases), the output address must occasionally jump to a new point in the memory. To stretch the time base of the input signal, the algorithm repeats small "grains" (segments) of the input signal, while for time shrinking, it deletes intermediate grains.The sound quality of the Pitch Time Changer is based on the nature of the input signal and on the ratio of pitch change it is asked to perform. Small pitch and time changes tend to generate less-audible side effects.

There are no "magic" settings. Only by experimenting with a given input sound can you explore the range of sound transformation possibilities this effect can realize.

👉 **Hyperprism 2 users:** Pitch Time Changer's time calculations, and the resultant sound quality, are influenced by the Processing Preferences settings found in HyperEngine's Preference menu. Try boosting the filter and interpolation settings if your time slow-downs are sounding too grainy.

👉 Do not be afraid to experiment with what might be considered "absurd" settings. For example, extremely slow playback rates can lead to very interesting and un-recognizable sounds.

☞ At some extreme settings of the Pitch Time Changer, especially when very high feedback settings are in use, the algorithm may mute its output when certain processing limits are exceeded. In this case, simply move the mouse back to a slightly less "extreme" setting and sound playback will resume.

☞ **Hyperprism 2 users:** Pitch Time Changer won't do time changes on live signal when used with a HyperEngine Play-Thru document. This process is meant more for use on disk files, which are processed in HyperEngine File documents.

Parameters

Speed

Speed variation, where 1% results in an unrecognizably-elongated version of the sound and 200 corresponds to a double-speed rendition of the input signal. Settings up to 400% (4x normal speed) are permitted.

Pitch

Pitch variation from -200% to +200%, where 0% represents a transposition towards extreme low frequencies and 200 transposes the input sound up an octave.

☞ Note that negative values of pitch shifting cause the pitch of the source signal to be shifted the same amount as positive percentage values, but the playback direction of the individual sound fragments, or "grains" are reversed. This can be especially interesting on speech or rhythmic sounds. For example, a pitch shift of -100% results in playing the sound at it's original pitch, but broken into small, backwards chunks.

⏸▶ Mouse click in the numeric field to the left of the Pitch slider, and drag the low-range value down to -200%. Now you can experiment with negative pitch shifts.

Grain Size

To stretch or shrink the time base of the original sound signal, the algorithm repeats (or skips) small segments, or "grains" of the original sound. The length of these grains can be varied, resulting in a great difference in sound. The effect can range from echoing to "granulation" to obliteration of the identity of the input signal.

Small grain settings can lead to buzzing sounds, while long ones may be perceived as a doubling, or echo effect. The actual results depend heavily on the character of the original sound. Hyperprism works with a maximum grain size of 1024 samples; this parameter is controlled by a percentage value, ranging from 0 to 100%.

X-Fade Size

The X-Fade slider sets the crossfade time, from 0% (no crossfade) to 100% (where the crossfade lasts the entire length of the grain. Short settings tend to produce audible artifacts at the boundaries set by the Grain size.

X-Fades of zero will often click. Longer crossfades will be smoother, but can lead to more chorusing or doubling sounds.

Feedback

The Feedback slider introduces regeneration, where the output of the pitch time changer is mixed back at it's input. A setting of 0% is no feedback, 100% will lead to run-away regeneration, as the full output of the effect is constantly being routed to it's own input.

Different feedback settings can result in anything from flanging or resonant effects to cascading showers of pitch-bending echoes as the mouse is moved.

Pitch Change Mix

Ranges from 0-100%. Sets the mix of time stretched sound and pitch shifted sound. 0% settings allow you to hear only time stretched signal; 100% means all pitch shifted signal.

Pitch Changer

| H 2 | VST | MMP G | DAS |

This effect is called "Pitch Shift" in Hyperprism-MMP. The Pitch Changer lets you vary the pitch of the input signal.The algorithm works by loading memory with an incoming signal sampled at rate A and reading out the samples at rate B. The ratio A/B determines the pitch change. To maintain a continuous output signal, samples must be repeated (for upward pitch shifts) or skipped (for downward pitch shifts). Because the output address pointer repeatedly overtakes the input address pointer (for pitch increases) or is overtaken by the recirculating input address pointer (for pitch decreases), the output address must occasionally jump to a new point in the memory.

There are no "magic" settings. Only by experimenting with a given input sound can you explore the range of sound transformation possibilities this effect can realize.

☞ At some extreme settings of the Pitch Time Changer, especially when very high feedback settings are in use, the algorithm may mute its output when certain processing limits are exceeded. In this case, simply move the mouse back to a slightly less "extreme" setting and sound playback will resume.

Parameters

Pitch

Pitch variation, from -200% to +200%, where 0% represents a transposition towards extreme low frequencies and 200 transposes the input sound up an octave.

☞ Note that negative values of pitch shifting cause the pitch of the source signal to be shifted the same amount as positive percentage values, but the playback direction of the individual sound fragments, or "grains" are reversed. This can be especially interesting on speech or rhythmic sounds. For example, a pitch shift of -100% results in playing the sound at it's original pitch, but broken into small, backwards chunks.

▶ Mouse click in the numeric field to the left of the Pitch slider, and drag the low-range value down to -200%. Now you can experiment with negative pitch shifts.

Grain Size

To stretch or shrink the time base of the original sound signal, the algorithm repeats (or skips) small segments, or "grains" of the original sound. The length of these grains can be varied, resulting in a great difference in sound. The effect can range from echoing to "granulation" to obliteration of the identity of the input signal.

Small grain settings can lead to buzzing sounds, while long ones may be perceived as a doubling, or echo effect. The actual results depend heavily on the character of the original sound. Hyperprism works with a maximum grain size of 1024 samples; this parameter is controlled by a percentage value, ranging from 0 to 100%.

X-Fade Size

The X-Fade slider sets the crossfade time, from 0% (no crossfade) to 100% (where the crossfade lasts the entire length of the grain. Short settings tend to produce audible artifacts at the boundaries set by the Grain size.

X-Fades of zero will often click. Longer crossfades will be smoother, but can lead to more chorusing or doubling sounds.

Feedback

The Feedback slider introduces regeneration, where the output of the pitch time changer is mixed back at it's input. A setting of 0% is no feedback, 100% will lead to run-away regeneration, as the full output of the effect is constantly being routed to it's own input.

Different feedback settings can result in anything from flanging or resonant effects to cascading showers of pitch-bending echoes as the mouse is moved.

Mix

The Mix parameter varies the balance between the level of the original signal and the new, pitch-shifted signal.

Vari-Speed

`H 2`

Emulates the behavior of analog recording devices; as you slow down the playback speed, the pitch is also lowered. Speed up the playback and the pitch is also raised. With adjustments ranging from twice as fast to 100 times slower, you can create sub-bass growls or rapid-fire speech.

☞ **Hyperprism 2 users:** Vari-Speed's time calculations, and the resultant sound quality, are influenced by the [Processing Preferences](#) settings found in HyperEngine's Preference menu. Try boosting the filter and interpolation settings if your time slow-downs are sounding too grainy.

☞ Vari-Speed won't do time changes on live signal when used with a HyperEngine Play-Thru document. This process is meant more for use on disk files, which are processed in HyperEngine File documents.

Parameters

Playback Speed

Control the playback speed of the input sound, from 1/100th of original to 400% of original speed.

LFO Frequency

Modulates the playback speed to create subtle shifts or huge swings in speed, from 0 to 20 Hz. Note that when previewing in real-time, frequencies above 5-10 Hz may sound similar depending on the speed of your machine. Try processing faster LFO frequencies to a new file if you have this problem.

LFO Depth

Controls the degree of shift in playback speed induced by the LFO. Note that with 100% depth the playback speed may shift to very slow depending on the ranges you have set for the Playback Speed parameter.

Noise Gate

`H 2` `VST` `MMP G` `DAS`

The Noise Gate provides a way to remove unwanted quiet portions of a sound. The most important parameter of the noise gate is the threshold. When the level of your audio is softer than the volume set by the threshold, the Noise Gate effect will

turn the audio off, leaving silence. However, when the signal level exceeds the threshold, the audio will be automatically turned back on. Setting the threshold to "0" will result in nothing but silence, since the audio can never get above that point. Similarly, setting the threshold to a very low number (the minimum is -96 dB) will result in a gate that never "closes," in other words, the sound will never be shut off.

The Noise Gate is "stereo coupled" when processing a stereo file. Left channel and right channel signal levels are calculated jointly and one gain is applied to both channels. This prevents weird shifts in the stereo balance.

Parameters

Attack

This parameter is called "Open Time" in Hyperprism-VST. It controls attack time from 0 milliseconds to one second. When the gate is closed (meaning the Noise Gate is shutting off your sound), it is constantly watching for the signal to exceed the level of the threshold. When this happens, the gate will open and begin passing sound again. However, the gate can be programmed to open immediately, or to "ramp" open at a particular rate. This rate is set by the attack time parameter. For example, if you are using the noise gate on a percussion track, set the attack time very short, unless you want to create a "fade in" effect on each drum transient. Feel free to experiment!

Threshold

Threshold, from -96 to 0 dB. The threshold sets the level at which the Noise Gate opens and closes. Audio is passed through the Noise Gate when it's signal level exceeds the Threshold setting. When the level of the signal drops below Threshold, it is muted. The rate at which the gate opens and closes is set by the Attack and Release times.

Release Time

This parameter is called "Close Time" in Hyperprism-VST. The Release Time is a complement to the Attack time parameter (described above). When the signal falls below the Threshold, the gate closes. This can happen instantaneously, or can be programmed to fade out more slowly. The Release time control sets the duration of this fade out. If you want to try and preserve a sense of room ambience when the gate closes, set the release time to a longer value. Or, if you are trying to create a very abrupt truncation of a sound, use a fast Release time. This parameter ranges from 0 to 1000 ms.

Metering lights

Red light indicates signal is below the threshold, so gating is applied. Green light means signal exceeds threshold, no gating is applied and signal passes unchanged. Metering lights are not included in Hyperprism-DAS for AudioSuite.

Compressor

The Compressor effect lets you manipulate the dynamic range of an audio signal. It does this by reducing the peak levels of the signal and then bringing up the overall level by a corresponding amount. The result is that "soft" sounds seem "louder," and the overall signal's dynamic extremes are smoothed out.

The Compressor is "stereo coupled," left channel and right channel signal levels are calculated jointly and one gain is applied to both channels. This prevents abnormal shifts in the stereo balance.

The Hyperprism Compressor features new discrete Ratio and Threshold settings, new Hard Knee/Soft Knee option and smoother, more transparent sound.

Parameters

Volume (or "Output Gain")

This control allows you to vary the overall output volume of your signal after it has been compressed. This parameter is called Output Gain in Hyperprism 2, Hyperprism-MMP and Hyperprism-DAS and ranges from -20 to 20 dB in the Hyperprism 2 and Hyperprism-MMP editions. Volume ranges from -36 dB to +36 dB in Hyperprism-VST and Hyperprism-DAS.

Attack

Attack time from 0 ms to 1000 ms. As the compressor operates, it is constantly watching the level of your audio signal and adjusting it's dynamic range. The Attack parameter determines how quickly the compressor will respond to the signal's changing dynamics. Slower Attack times will let individual transients, such as drum hits, through without turning them down. Faster attack times will "catch" these transient events.

Release

Release time from 0 ms to 1000 ms. As the compressor operates, it is constantly watching the level of your audio signal and adjusting it's dynamic range. The Release parameter determines how quickly the compressor will respond to the signal's changing dynamics. Slower release times will generally result in less noticeable artifacts in your signal's low-level sounds, such as room ambience. Fast release times may result in the room sound fluctuating in level, perhaps in a distracting way. On the other hand, this might be just the effect you are looking for. Experiment!

Ratio

As in any traditional hardware compressor, Hyperprism offers a Ratio setting to control the amount of gain reduction. A ratio of 1:1 means no reduction, and a ratio of 10:1 means a 10 dB increase in the input signal results in only 1 dB increase in the output signal. Possible settings range from 1:1 to 10000:1. In Hyperprism-MMP this value is called "Compression Ratio."

Threshold

This control allow you to set the volume level above which compression begins. Possible values range from -60 to +20 dB. Use the metering LED's to note when your input signal is moving above and below the threshold. In Hyperprism-MMP this value ranges from -60 to 0dB.

Soft/Hard Knee

When in hard knee mode, gain is applied as soon as the input signal crosses above the threshold (after attack and release times, if any, are considered). The soft knee gradually applies the compression gain to the input when the signal is from -6 dB below the threshold to 6 dB above the threshold. Above 6 dB over the threshold, full compression gain is applied. Using the soft knee allows for more compression with smaller attack and release times.
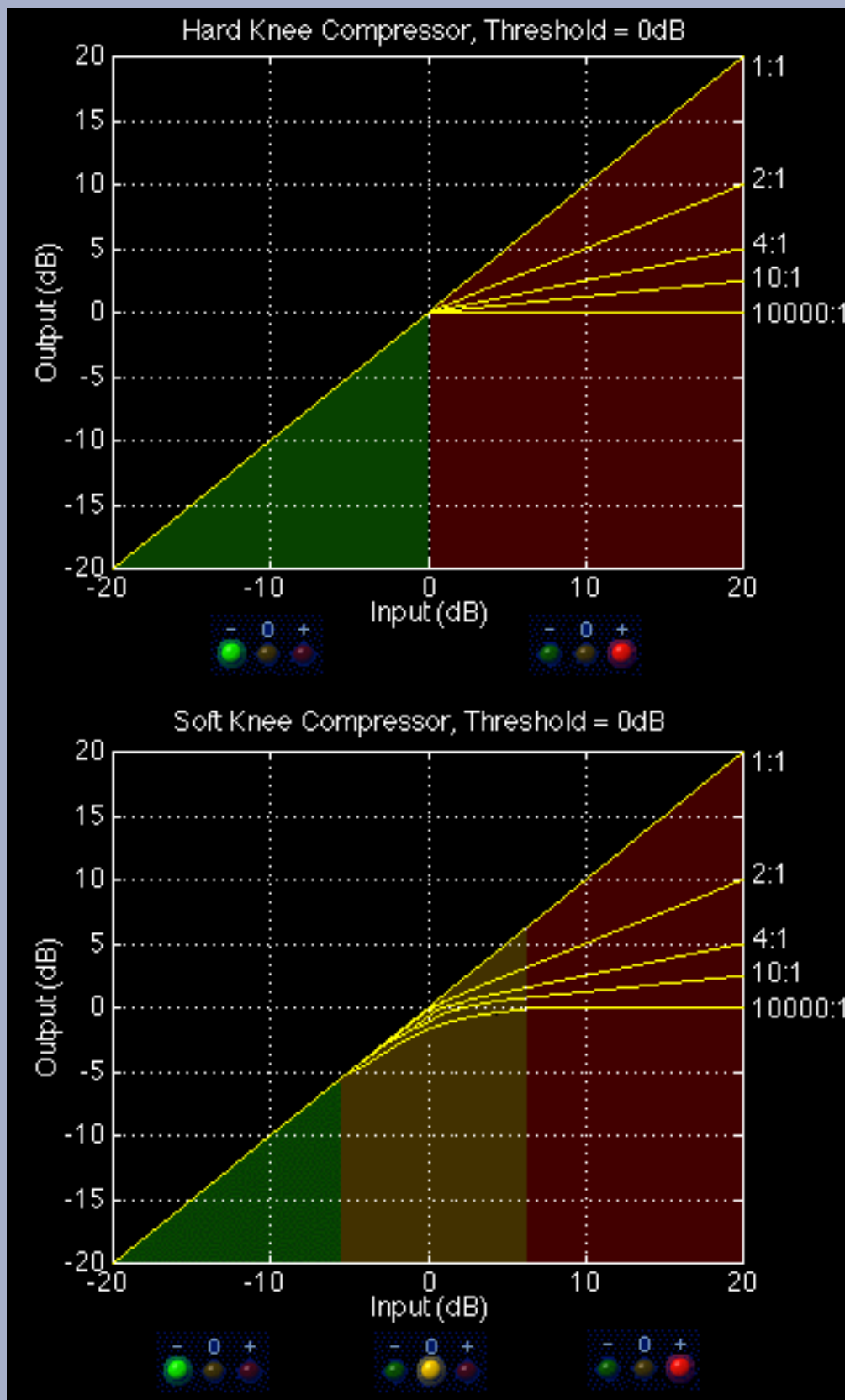
Output characteristic charts showing the difference between Hard Knee and Soft Knee compression

Metering lights

Green light indicates signal is below the threshold, no processing is occurring. Orange light denotes signal in the soft knee region and incremental compression is occurring. Red light means signal above the threshold, full compression value is

being applied. These are set taking into account the attack and release times, so that they light up only when the gain is actually being applied. Metering lights are not included in Hyperprism-DAS for AudioSuite.

Limiter:

| H 2 | VST | MMP G | DAS |

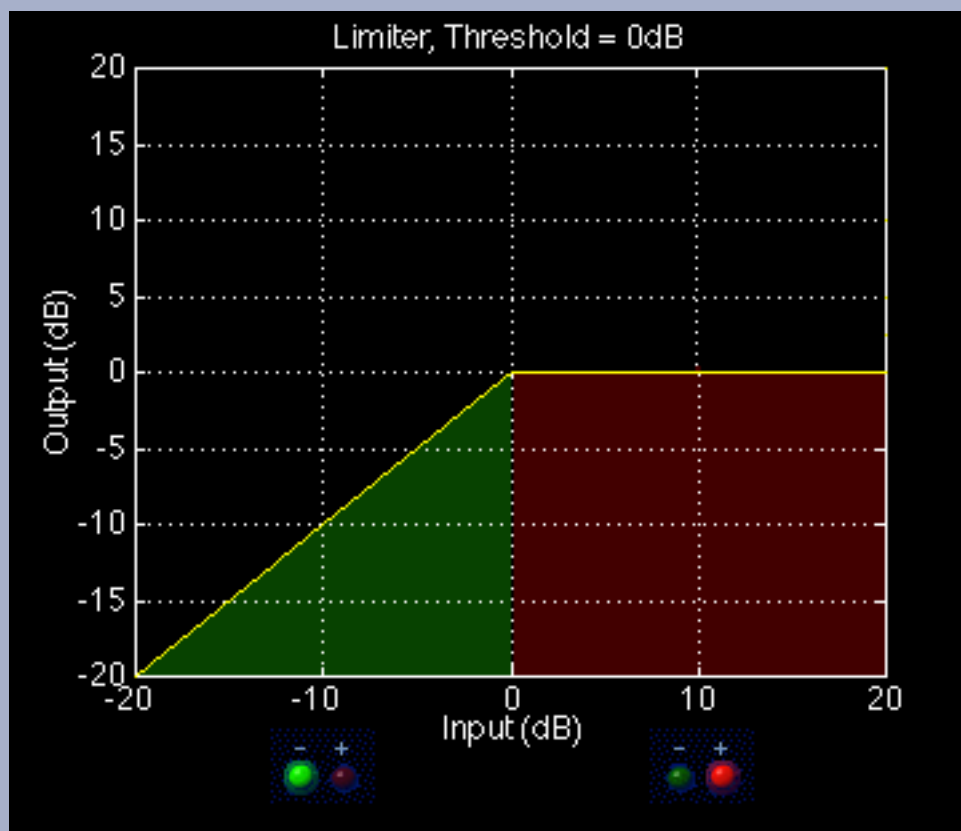Like the Compressor, the Limiter effect lets you manipulate the dynamic range of an audio signal. It does this by sensing the peak levels of the signal and attenuating those peaks by adding an adjustable amount of gain. Reducing the peaks allows you to bring up the softer parts, making your file sound louder.

As signal rises toward zero dB, no limiting is applied. Any signal that exceeds zero dB will be limited to that setting (when Threshold is set to zero dB).

Set the threshold for the level at which you want the limiter to kick in. As long as your signal is close to, or slightly above the threshold, the output level of the signal will match your gain setting. When you put the threshold below the level of incoming signal, you'll be attenuating signal which exceeds the threshold, resulting in a perceived increase in the loudness of the signal. If the threshold is set above the level of incoming signal, no processing is applied.

The limiter is "stereo coupled," meaning the signal level is calculated by considering the left and right channels at the same time; one gain setting is applied to both channels. This prevents undesirable shifts in the stereo balance.

You may notice that the Hyperprism Limiter has a Release time, but no Attack time setting; we've set attack time parameter permanently at 0, so it responds immediately to increases in the input signal.

Ceiling

Threshold setting, from -96 to 0 dB. Input signal exceeding threshold will have the volume setting applied to it.

Volume

The amount of gain applied to the signal, ranging from -36 to 36 dB.

Release

An adjustable time lag, you determine how long it takes for limiting to cease once signal falls below threshold. Measured in milliseconds from 0 to 1000 (one second).

Metering lights

The green light means signal is below threshold and no limiting is occurring. The red light indicates that input has exceed the threshold and limiting is engaged. Metering lights are not included in Hyperprism-DAS for AudioSuite.

Sonic Decimator (TM)

`H 2` `DAS` `VST` `MMP G`

The Sonic Decimator allows you to reduce the apparent sample rate or bit-depth of your sound file. This is an excellent way to re-create the sound of the first generation of digital sampling devices. At moderate settings, the can create a "low-fi" effect, as though the sound was being processed by a low-budget sampler. At extreme settings, large amounts of hiss and distortion, as well as intense aliasing effects are added to the sound.

Parameters

Sample Rate

This control reduces the apparent sample rate of your file. This has the effect of reducing the sound's high frequency content and can introduce intense aliasing at very low settings. Default range is 1% to 100%.
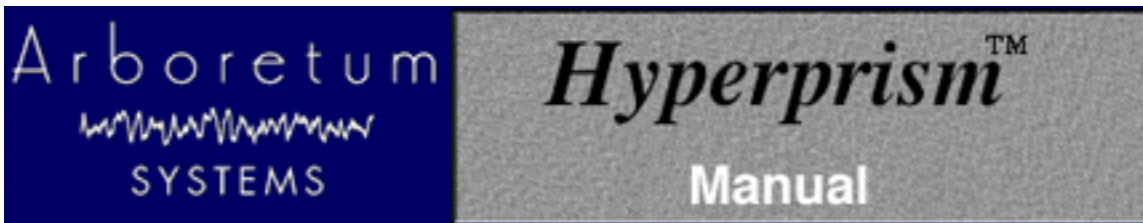
Bit Depth

This parameter reduces the files equivalent-bit-depth, reducing the overall dynamic range, increasing the noise floor and adding digital resolution distortion. Default range is 1 bit to 16 bits.

Appendix A-Troubleshooting

For up to date trouble shooting tips and general questions, please check our Web site at http://www.arboretum.com

In This Chapter:

- Installation Failure
- Eliminating System Conflicts
- Optimizing Performance
- Recording or Play-Thru Sounds Distorted
- Effect Process Sounds Distorted
- Noise Reduction Alters Original Signal
- Effect Changes Seem Coarse or "Jumpy"
- Current 3rd-Party Support Issues

Installation Failure

First, make sure you're using the correct serial number. Arboretum's Harmony, Hyperprism 2, Hyperprism-TDM, Hyperprism Plug-in Pack, Ray Gun and Ionizer products are each sold separately, and each requires a unique serial number. Also note that Ionizer 1.3, Hyperprism Plug-in Pack 1.5 (and the component 1.5 editions of Hyperprism-MMP for Premiere, Hyperprism-DAS for AudioSuite and Hyperprism-VST) are all paid upgrades and will require purchase of a new serial number for installation.

The most common source of installation failure is system conflict with extensions or control panels, especially virus detection software. If you're not installing or serializing successfully quit the Installer and restart your computer while holding the Shift key down. This will disable all extensions and allow you to get around virus checking software or other conflicting system resources.

The second most common source of failure is simply entering the serial number incorrectly. If you've entered a wrong number into our Serialization dialog three times, the Serializer will quit and your plug-ins will refuse to serialize (even if you come up with the right serial number on try number four). If this has occurred to you, quit the current application and go to the Finder. Open the System Folder and locate the Preferences folder. Within the Preferences folder you should find Hyperprism, Ionizer and/or Ray Gun preferences files. Select the applicable preference file and drag it to the Trash, taking care not to delete any associated Presets files. Then re-launch the application, call the plug-in and input the correct serial number when the Serialization dialog appears.

If the preceding fails, the next step is to deinstall and re-install the plug-ins. Go into the Preferences folder and Trash your preferences as described in the preceding paragraph. Then throw away any Arboretum plug-ins which were unsuccessfully installed. Next, run the Custom Install routine to copy a fresh set of plug-ins to your drive, and proceed to serialize using the correct serial number. Make sure to check your typing before clicking on OK, because accuracy counts.

If your host application does not have a specific installer on the Arboretum CD-ROM or if your host app does not have a plug-ins folder (c.f. TurboMorph) then install your plug-ins to another applications plug-ins folder and move them to the desired destination once the installation routine is successfully completed. The Installer allows you to install plug-ins into the Arboretum f folder for just this purpose.

If you're doing everything correctly but you still can't get the software to serialize or instantiate successfully, feel free to email support@arboretum.com or call (650) 738-4750 for further support.

Eliminating System Conflicts

Strange behavior may sometimes be caused by certain System Extensions and Control Panels. With real-time audio programs in general, it's a good idea to remove Control Panels or Extensions that can cause interrupts in the background; we suggest disabling your **fax software** and turning OFF **AppleTalk** and **File Sharing**, to ensure uninterrupted audio recording and playback. Connectix **RAM Doubler** is a KNOWN CONFLICT, always restart with RAM Doubler turned OFF before using any audio program. You should also turn OFF **Virtual Memory** in the Memory Control Panel. If you're still having unstable operation, disable all unnecessary Control Panels and Inits, either by using a utility such as Extension Manager, or simply by dragging Extensions and Control Panels to another folder (labeled "Disabled," for example) and then restarting your Macintosh.You can then retry the configuration by re-installing the extensions one by one.

For best performance, run a "Lean" system with as few Control Panels, Extensions and other programs running in the background as feasible.

HyperEngine editions prior to v2.4 required the Open Transport extension set; this is no longer the case with HyperEngine 2.4 and later. In fact, under MacOS 8.0 and later, no System Extensions whatsoever are required to launch HyperEngine 2.4 and the associated Ray Gun, Ionizer and Hyperprism 2 HyperEngine plug-ins. HyperEngine 2.4 requires QuickTime 3 for its advanced import/export services. Running QuickTime 2.5 is OK but it will automatically truncate the number of channels, bit depth and sample rate of any audio imported to mono 8-bit 22050.

Although most audio programs which accept plug-ins do require that Virtual Memory be disabled, the HyperEngine is a possible exception. Members of our development team routinely run HyperEngine and associated Hyperprism 2, Ionizer and Ray Gun "-HE" plug-ins with Virtual Memory on and report few problems. That said, if you're experiencing frequent freezes, crashes or unexplainable error messages we continue to recommend disabling Virtual Memory.

Optimizing Performance

Audio programs perform best on faster machines, with plenty of RAM. Make sure your hard drives are not too fragmented. A fragmented disk makes it more difficult to read a sound file since it is scattered in small fragments. We suggest the use of a disk optimization program, and we specifically recommend Norton Speed Disk for this purpose.

The other ways to improve performance are listed below:
- Do not use Virtual Memory
- Do not use RAM Doubler
- Use few System extensions
- Do not use AppleTalk
- MacOS 8 or higher users select "system-wide platinum appearance" in the Appearance control panel
- Close the Time Display and/or Peak Meters palettes

Recording or Play-Thru Sounds Distorted

First, check the Meters and make sure your levels are OK. If you're constantly clipping (meters in the red) then use the Faders to cut the input level, or reduce the output of your external signal source.

If you still are experiencing problems, make sure that the settings in your MacOS Sound control panel for bit depth, sample rate and number of channels (stereo/mono) match the corresponding HyperEngine settings in the Configure Audio

System dialog (or the corresponding settings in your third-party audio editing program). Having a mis-match between the Configure Audio System and Sound control panel settings for bit depth or sample rate can result in massive amounts of digital noise on recorded or play-through signal.

If play-thru or recorded audio sounds like its skipping, cutting out or stuttering, adjust the Buffer setting in HyperEngine's Configure Audio System dialog (or similar input buffer setting in your third-party sound editing software). Larger input buffers will smooth this out, but remember that the buffer takes additional free RAM, beyond the normal Preferred RAM setting.

Effect Process Sounds Distorted

Although some Hyperprism effects are meant to destroy your sound (such as the Sonic Decimator) other effects (such as the Filters) can create unintended distortion when adverse parameter settings are applied, or when too much gain is applied at the HyperEngine Faders. If a Filter is distorting, adjust the Q and Cut-off settings. Rapid effect parameter changes may also create clicking or distortion in some cases; stop moving parameter sliders for a moment and clicking/stuttering should disappear.

Distorted processing is also a characteristic of running these products on a machine which is not fast enough to handle it. Arboretum's Hyperprism 2, Ionizer, Ray Gun and HyperEngine software require a Power Macintosh which runs at 120 MHz or faster. The Hyperprism Plug-in Pack (and its Hyperprism-MMP for Premiere, Hyperprism-DAS for AudioSuite and Hyperprism-VST components) need a Mac with a clock speed of 80 MHz or faster.

Noise Reduction Alters Original Signal

Ray Gun and **Ionizer** users can run into problems when applying too much attenuation to a noise reduction process. Try using more moderate settings if you've lost all the brightness or introduced new artifacts into your signal.

For Ray Gun's Noise Reduction processes, try putting your Threshold at plus or minus 3 dB, with -3 dB Attenuation, as a starting point. Ray Gun Pop/Click removal settings in the 50%-75% range seem to work best in most cases, settings around 100% tend to stutter or cut into target signal.

Ionizer users should use the Fit command threshold settings as a starting point, then customize the threshold and Gain Curve to find the best settings for your particular sound file. If chirping or flanging artifacts are introduced, increase the Correlation, Attack and Release settings. Try 30% Correlation, 10% Attack and 30% Release as a starting point, then customize your settings from there.

☞ Finding the best "compromise" settings, where you've taken out the most noise with the least impact on your original signal, is the real trick in noise reduction work. Experiment, and let your ears be your guide.

Effect Changes Seem Coarse or "Jumpy"

If your effect changes don't sound very smooth, here's a few things to try:
- Alter the effects parameter ranges, varying a parameter over a fairly small value range often gets the best results.
- Increase the size of the Blue Window by dragging its lower right hand corner down and to the right.

Current 3rd-Party Support Issues 11/15/98:

G3 Upgrade Cards Not Supported: We are now able to independently verify what we long suspected, which is that G3 processor upgrade cards for older Power Macs (from manufacturers including Newer Technologies and others) exhibit a number of incompatibilities which well may interfere with audio editing programs. The upgrade card manufacturers themselves are beginning to acknowledge the problem; this ends the mystery surrounding a number of heretofore unresolvable technical issues pertaining to G3 upgraded Macs. Our advisory on this topic is therefore reiterated and intensified: We will not be able to guarantee performance on systems with CPU upgrades. Don't use an upgrade card to extend the life of your old Mac. We suggest you consider trading up to a new factory-built G3 instead.

Hyperprism-TDM and Studio Vision Pro, Digital Performer, Logic or Cubase: Hyperprism-TDM is not yet fully functional in these 3rd party TDM "host" applications, due to inconsistencies in plug-in implementations and the demands of cross-platform support. As of this writing (11/9/98) Hyperprism-TDM requires and claims only to support Digidesign Pro Tools software version 4.1 or later.

Logic Audio Users: Ionizer AudioSuite edition keyboard shortcuts are disabled in Logic Audio. However, the Ionizer Premiere version key commands work in Logic as documented. And on the VST plug-in side of Logic Audio, please note that mono effects show up at Mono-to-Stereo stereo inserts, stereo effects show up on the Stereo-to-Stereo inserts.

Cakewalk Metro 4 Users: The VST version of Ray Gun, and certain new Hyperprism-VST effects require Metro 4.5.1 or later. The Premiere format edition of Ray Gun and Hyperprism-MMP for Premiere are both compatible with Metro 4 and 4.5.

Peak Users: If you are using Hyperprism-MMP, Ray Gun or Ionizer in Bias Peak, make sure you upgrade to Peak 1.6.1 or later.

Pro Tools Can't Open Stereo Sound File: Pro Tools may not be able to open some stereo files generated by HyperEngine if their titles becomes too long when Pro Tools adds the suffix left or right to them (not a problem, therefore, with mono files). This is typical when you generate a file with a default title such as Jazz Guitar-FX-Untitled Balance for which only one more character is enough to go beyond the maximum number of characters (31) allowed by your Macintosh System. To remedy this situation, just shorten the names of the files you generate with Hyperprism. This may be fixed in a future version of Pro Tools.

Pro Tools users (who have Hyperprism-TDM and/or Hyperprism-DAS) need to be current and upgrade to PT version 4.1 (in addition to WaveShell 2.3.1 - see below).

Pro Tools users who are running Hyperprism-DAS, Ray Gun or Ionizer for AudioSuite will encounter DAE error -7450 when attempting to use a mono-to-stereo process (such as Quasi Stereo) on a mono track/sound file. The AudioSuite format does not allow plug-ins to do mono-to-stereo processing. To simulate mono to stereo processing simply make a new audio track and insert a second copy of the mono file there, then process them as a stereo pair. You will also need to configure your plug-in for Multi-Input Mode any time you do stereo processing. Ionizer/Ray Gun users, just select either the mono or stereo versions of these effects from the AudioSuite menu before you start processing.

Waves users: WaveShell 2.3.1 fixes earlier conflicts between Waves and Arboretum plug-ins for TDM and AudioSuite. Download WaveShell 2.3.1 from Waves' web site.

Hyperprism-DX and Cubase VST (Windows): Make sure you're running Cubase v3.502 or higher. It fixes a number of problems and can be downloaded from: http://www.steinberg.de

DECK II Users: Because OSC/Macromedia DECK II doesn't support interleaved stereo files (offering only mono and split-stereo support as of version 2.6) you will not be able to use any Hyperprism effects which require two-channel input. Therefore **you will not get** the Auto Pan, M-S Matrix, More Stereo, Pan, Quasi Stereo, Stereo Dynamics or Vocoder effects when you do the Hyperprism-MMP installation for Deck. These effects won't work in Deck, so we don't bother installing them.

This all stems from an inherent limitation of Deck's mono track architecture, and for a while there it looked like no changes would ever be forthcoming. However, in light of the recent acquisition of Deck by the good folks at Bias, it now appears that Deck indeed has a future and that some enhancements to plug-in implementations may be made at some later point.

Adobe Premiere Users: Audio plug-ins including Ionizer, Ray Gun or Hyperprism-MMP should be installed in a folder called "Adobe Premiere Plug-ins" in Premiere version 4.x. Adobe changed the name of this folder in conjunction with the Premiere 5 upgrade, in Adobe Premiere 5.x install audio plug-ins into a folder called simply "Plug-ins".

Hyperprism-MMP (Premiere): Due to the limitations of the Premiere plug-in format, the H-MMP Quasi Stereo and Stereo Dynamics effects won't accept a mono (one-channel) input for conversion to stereo. The work-around is to mix your target mono file into a stereo file, then process. Because this problem stems from Adobe's Premiere plug-in architecture, we're unsure whether any remedy will be provided.

Hyperprism-MMP (Premiere) Stereo File Support: Hyperprism-MMP includes a number of effects (such as the M-S Matrix) which require a stereo file for input. If your audio editing program supports interleaved stereo files (as does Bias Peak) then you should be able to use all the stereo effects successfully. If your audio editing program only supports mono or split stereo (.L/.R) files, which is the case in Macromedia Deck II, you simply won't be able to use these stereo effects successfully, and Hyperprism will give you a message saying "this effect requires a stereo file."

Premiere Format/Preview is Too Short: Note that the Premiere-format Preview function can only play as much of your audio selection as can be loaded in RAM. RAM-based preview is one of the fundamental shortcomings of the Premiere plug-in format, and is a behavior you'll see with all Premiere compatible plug-in effects, including Hyperprism-MMP, Ray Gun and Ionizer. If you're not hearing your entire file or selection when you Preview, don't worry. The full piece of audio will be treated when you hit the Process button.

Many sound editing programs allow the user to adjust the size of the preview buffer, thereby increasing or decreasing the total amount of possible preview time. When increasing the buffer remember that this will take up more RAM, in addition to the amount you've already allocated to the program. So if you've allocated 20 MB to the program, and are running with a 6 MB buffer, you'll need 26 MB in which to operate. This setting resides in various places; check your audio editing program's manual details.

[Next Chapter]

Go to Table of Contents: [Hyperprism-VST]

Appendix B: About Digital Audio Files

- Memory Requirements
- Sample Rates
- Bit-Depth & Dither
- Mono/Stereo
- File Formats

Memory Requirements

Working with computer-based digital audio requires large amounts of hard disk space. If you are planning on creating new audio files on disk, you'll need enough hard drive space to contain them.

A good rule of thumb is:

Each minute of 16-bit stereo sound at 44.1 kHz requires about 10 Mbytes of disk space.

Thus, with an empty 200 Mbyte hard disk you can record a little less than 20 minutes of CD-quality stereo sound (precisely, 19 minutes, 20 seconds).

Sample Rates

Sample Rate values are indicated in Hertz (Hz), or "cycles per second." For professional audio work always use the 44,100 Hz (44.1 kHz) setting, which is the CD-quality sample rate. 22,050 Hz is considered acceptable quality for many interactive multimedia applications. 11025 Hz is a low-resolution, "voice quality" sample rate.

Most audio editing programs support several different sampling rates:
- 48 kHz
- 44.1 kHz
- 22.050 kHz
- 22255 kHz *
- 11.025 kHz
- 11127 kHz *

☞ The 48 kHz sample rate is not directly supported by the Macintosh Sound Manager in most Apple Computer models; you'll need a 3rd-party sound card to do 48 kHz work.

* These two rates were the original Macintosh "close enough" sample rates. While they are supported, they should probably be avoided, unless your sounds will primarily be played on older Macs.

Using a lower sample rate file will proportionally reduce your disk storage requirements, i.e. a 22 kHz file will only take half the space of the same sound stored at the audio-CD rate of 44.1 kHz. However, using a lower sample rate will reduce

the high frequency response of your sounds.

To avoid aliasing (a metallic-sounding type of digital audio distortion), don't use high-frequency filter settings above about 10,000 Hz when using 22 kHz sample rates, or settings above 5,000 kHz for 11 kHz sounds.

Bit-Depth & Dither

In addition to multiple sample rates, you may be working with either 24-, 16- or 8-bit files.

Lower resolution formats reduce storage space requirements. Unfortunately, lower bit depth and sample rate settings can compromise the audio quality of your sound files. Lower sample rates lose high frequency response, and 8-bit storage causes a reduction of your sound's dynamic range, resulting in noisier, "grainy-sounding" audio, especially during softer passages.

If you are creating 8-bit audio (for example, for multimedia or Internet distribution), you will get best results if you do all your signal processing at 16-bits and 44.1 or 48 kHz, and then create an 8-bit file at the end of the process.

The distortion of low-level signals can be minimized by normalizing your 16-bit file before 8-bit conversion, or better still, by using dynamic range compression or manual gain-riding to make soft parts of your sounds louder before reducing their resolution.

The use of dither can improve the sound quality of 8-bit audio, although at the expense of a slightly noisier signal. In HyperEngine select the Dither option in the Process to New File dialog. Other programs have dither located in the Save As, Mix or Export dialogs. Dither adds a small bit of noise to the signal to mask the audible artifacts that occur during bit reduction.

Adding dither is most appropriate when the resulting sound won't be undergoing any further editing, gain scaling, EQ, etc. Additional processing can result in an unwelcome increase in perceived dither noise. Therefore, if you do expect the sounds you create with Hyperprism to be edited further, you may wish to leave dither off.

Mono/Stereo

Audio files come in three flavors: Mono (one-channel), interleaved stereo (two channels in a single file) and split-stereo (two channels in two separate files). Some programs will open and process only mono or split stereo files, some support only mono and interleaved, some handle all three. Most programs at least have some provision for importing and exporting all three types.

File Formats

The most common MacOS audio formats are the Sound Designer II (SDII) and AIFF file formats. Most PC/Windows products process Wave (.wav) audio files. The sound quality of these three formats is identical.

If you're recording audio for eventual use on a CD, we recommend that you always create 16-bit stereo .WAV or SDII files, sampled at 44,100 Hz. This is the file specification you'll need to create "Red Book" audio CDs, and is what programs such as Adaptec Toast or Easy CD Creator 3.5 accept for burning audio discs. Sticking with these settings will prevent file-conversion headaches down the road. Note also that most digital audio/sequencing programs have native support for "CD-quality" SDII or .WAV files (according to whether they run under the MacOS or Windows), and allow you to open them directly so you won't have to chew up additional hard drive space making format-converted copies. Unfortunately, a few multitrackers (Deck, for example) won't accept interleaved stereo files without a format conversion to split-stereo, which does use additional disk space.

Many different sound file formats exist, including .SND, .AU, and others. While some programs cannot read these files directly, there are a number of sound file format conversion utilities available. After conversion to Wave, SD II or AIFF, any sound can be used by Hyperprism/HyperEngine/Ray Gun/Ionizer, as long as it has been converted to a

compatible sampling rate and bit-resolution.