Hyperprism 2 Manual

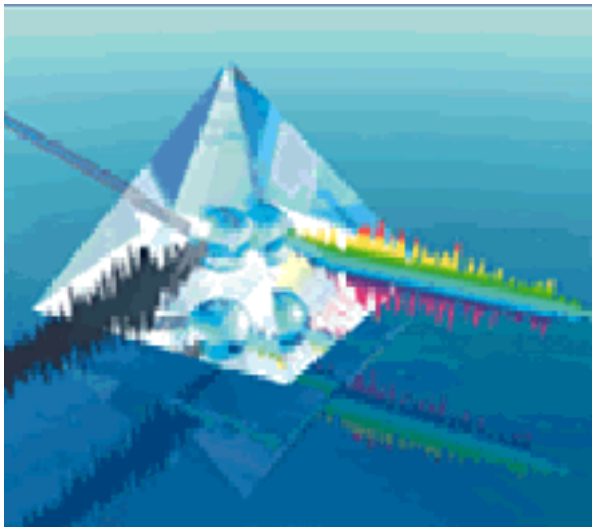
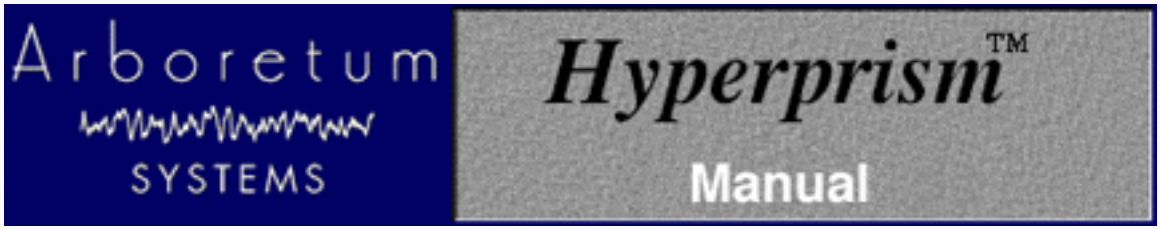


Table of Contents

Arboretum SYSTEMS



Hyperprism 2

Real Time Dynamic Audio Effects Processing Software

Version 2.1

New in Hyperprism 2:

- New Vocoder, Frequency Shifter, HyperVerb, Doppler, Z-Morph and Vari-Speed effects.
- Now HyperEngine provides the stand-alone framework for Hyperprism, offering "on-the-fly" live audio processing, sound file editing and more.
- Improved Compressor includes new soft knee setting plus clearer, smoother compression
- Filters now sound even better, work easier...plus new Lo/Hi Shelf and Parametric EQ.
- Combine multiple effects for complex multi-processing chains, all in real-time.
- New Blue Window automation editing, along with easy geometric path creation.
- Select sound file range for processing/automation directly from the Blue Window.
- User-definable Blue Window axis assignments.
- New real-time Play-Thru processing, plus processing during recording.
- Plus a cool new look, direct sound card support and complete documentation detailing every feature and function in HyperEngine and Hyperprism 2.

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 toolbar.

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!

Underlined Text indicates a Hyper link; mouse-click on the link to access more information on the designated topic.

Documentation current for Hyperprism version 2.0, last revised 4/5/98. Manual written and edited by Todd Souvignier and Aram Lindahl with contributions from Chris Townsend, plus legacy material by Georges Jaroslaw, Chris Weare, Curtis Roads and David (Rudy) Trubitt. New graphics by Aram Lindahl. Hyperprism 2 and HyperEngine programming by Aram Lindahl and Chris Weare, Ph.D.

---

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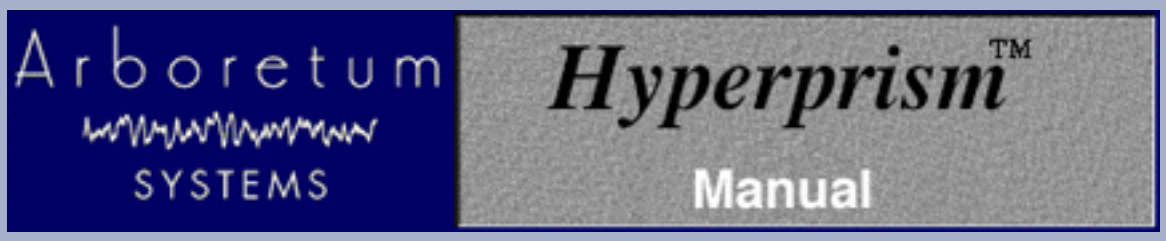
If your Browser is connected to the Internet, you can visit us by clicking the button below to find more information as well as the latest available software and documentation updates.

[go to First Chapter]



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You acknowledge that you have read this Agreement, understand it, and agree to be bound by its terms.

Arboretum SYSTEMS

Hyperprism™ Manual

Hyperprism 2

Introduction to Hyperprism 2

"Les formes musicales possibles sont aussi
inombrables que les formes extérieures des cristaux."

The universe of possible musical forms
is as unlimited as the countless shapes of crystals.

--Edgard Varèse, speaking about his composition Hyperprism (1923)

Welcome to Hyperprism 2 - the real-time effects processors that you play like a musical instrument. Built for exploring and controlling sound transformation, Hyperprism 2 combines the ease of the Macintosh interface with the signal processing power of the PowerPC microprocessor. Its primary features are the following:

- 36 different high-quality effect algorithms, including filtering, delay, modulation, pitch and time control and stereo manipulation effects.
- Interactive gestural control of effect parameters in real time, using the mouse.
- Mouse control movements are recorded and can be played back by Hyperprism 2 in synchronization with the original sound file.
- Effect parameter automation can be edited on screen.
- 32-bit, floating-point calculations are used internally during all effects processing for maximum fidelity.
- Apply any number of effects to a single sound file, all in real-time.
- Non-destructive document-based processing means that your original sound files are always safe and that disk storage requirements are minimized.
- Live play-through processing allows you to use Hyperprism like a performance instrument, on stage or in the studio.
- HyperEngine (included with Hyperprism 2) provides sound input/output services to Hyperprism effects and offers simple file editing capabilities.

For instructions on installing and configuring Hyperprism 2, please click here.

Be sure to peruse the Read Me files on your CD for any important additional information that we were not able to include in this manual.

Hyperprism Processes

Hyperprism 2 implements 36 musical effects:

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

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

Hyperprism Miscellaneous Processes

- Pitch Time Changer
- Pitch Changer
- NEW! Vari-Speed
- Noise Gate
- Compressor
- NEW! Limiter
- Sonic Decimator

Each effect can operate over a wide range of settings, all under your interactive real-time control. Each effect can be combined with other effects to create complex multi-processing chains. And all processing is non-destructive. Hyperprism 2 effects are plug-ins, which work ONLY inside Arboretum's HyperEngine program, included free with Hyperprism 2. If you want to use Hyperprism effects in other Mac/PC sound editing programs, check out Arboretum's Hyperprism Plug-in Pack (available separately) which is the complete assortment of Hyperprism plug-in editions for use with popular 3rd party audio software.

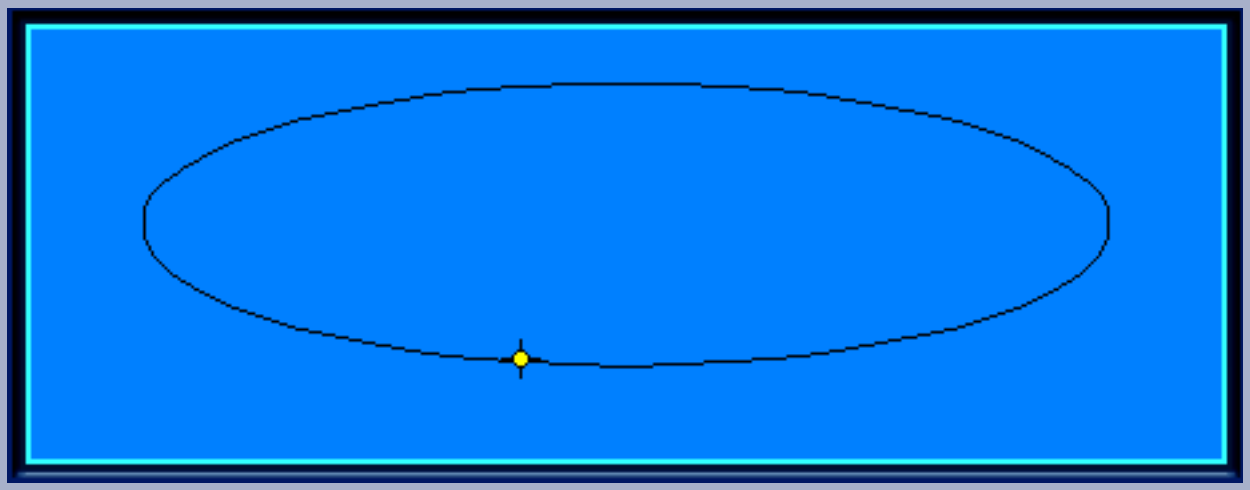
Getting Sound Into Hyperprism

How do you get your sounds into Hyperprism? Hyperprism works with live audio files being played into your Mac sound input or your professional sound card (such as Digidesign Audiomedia II and III or Korg 1212 I/O). Hyperprism also works with sounds files stored on disk. These sound files can be created by other audio applications, or you can use Hyperprism to record sounds connected to your Mac by recording in a HyperEngine Play-Thru document.

Hyperprism 2 will create, open and process sound files in the two popular Mac audio formats: Sound Designer II (SDII) and Audio Interchange File Format (AIFF). Hyperprism 2 works with mono and interleaved stereo files at any supported sample rate, and supports both 8 and 16 bit files.

Processing Sounds: The Blue Window

One of Hyperprism's most distinctive features is its "Blue Window," which is allows you to control signal processing changes simply by moving the mouse.



Hyperprism-PPC's Blue Window is used to dynamically control effect parameter variations.

Most modern effects hardware and signal processing software require you to pick the settings of an effect before you use it. Hyperprism is different. Instead of a fixed set of effects parameters, Hyperprism lets you change the parameters of an effect while the sound is playing. The result is a very interactive method of effects control, one that had nearly disappeared in the industry's transition from analog to digital signal processing.

Hyperprism tracks the horizontal and vertical movements of your mouse and uses this information to instantly update an effect's parameters. For example, moving the mouse left or right could change the delay time between the repeats of an echo effect, while up and down movements adjusted the number of individual repeats.

Parameter changes are controlled in Hyperprism's Blue Window. 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 processed, your gestures will trace a path in the Blue Window.

The paths you draw can be recorded, so you may hear the results of your gestures after you make them. If you don't like the results, just draw another path. Or edit the path breakpoints until you've created the parameter change you want. You can also use the effect control sliders in the Blue Window for additional discrete control over each effect algorithm.

If you're worried about disk space, relax. When you're creating your effect combinations and parameter automations, you're building a HyperEngine Reference Document, which stores all the information about the processes your designing, plus any file edits you may do. HyperEngine Reference Documents are comparatively small, a few Kbytes each, so you can make as many new documents and store as many variations as you wish.

Designing effects in the Blue Window doesn't change your original sound file. Hyperprism 2 processing is "non-destructive," because the original samples in your sound file are never altered. The HyperEngine document is linked to the sound file you are processing, so that when you open a previously-saved HyperEngine document, the correct sound is loaded automatically.

Getting Sound Out of Hyperprism

Whether you are working on a music CD, sound-for-picture, multimedia or other project, you'll soon want to create new, processed sound files from Hyperprism to incorporate in your finished work. HyperEngine only creates a new disk file when you tell it to, which is a nice space-saving consideration.

Since HyperEngine is capable of storing many different effect variations for the same sound, you must first choose the variation(s) you want to "export."

Then, you'll use HyperEngine Process to New File command. This command runs your original audio file through the settings and edits in the current File document to create a brand new processed sound file. Once this file is created, you can import it into virtually any other audio or multimedia software application, or use a program such as Toast or MasterList CD to burn it to an audio CD.

Another way to get your processed sounds out of the program would simply be to record HyperEngine's playback directly from the audio output of your Macintosh into an external DAT recorder, sampler or other recording device. This option would also be attractive if you were working with limited hard disk space, and didn't have the room to create a processed file on your hard disk.

And don't forget HyperEngine's new Play-Thru processing features, which allow you to run live signal into Hyperprism - from a tape deck, instrument or microphone - process it on-the-fly, and output it to a mixer, PA or recording device. Use Hyperprism like a musical instrument; it will remember your every move and respond instantly to each touch and gesture. All with a single mouse drag.

HyperEngine has new direct driver support for Digidesign Audiomedia II and III and Korg 1212 I/O cards, allowing you to use these professional sound cards without installing 3rd party drivers.

Arboretum SYSTEMS

Hyperprism™ Manual

Hyperprism 2

Installation and Configuration

This Chapter describes the procedure for installing and configuring Hyperprism 2 on your Apple Macintosh computer

Hardware and Software Requirements

Before using Hyperprism, please make sure that you have the following:

 You should possess a basic knowledge of the Macintosh operating system. This includes knowing how to use the mouse, windows, menus, etc. If these terms are strange to you, please consult your Macintosh documentation.

You'll need a Power Macintosh with a clock speed of 120 MHz or faster in order to user Hyperprism 2. For processor-intensive effects such as the HyperVerb or Vocoder, or for real-time play-thru processing we recommend a clock speed of 200 MHz or faster. Your Power Mac's operating system must be System 7.6 or later, System 8 suggested. You must have at least 16 Mbytes of random-access memory (RAM) available for the Hyperprism 2 application. If you want to create long delay effects you will need to allocate more memory to Hyperprism. This is because Hyperprism 2 uses available Macintosh memory for its delay lines. You can check the amount of memory allocated by selecting the Hyperprism 2 icon in the Finder and typing Command-I (for "Get Info"). Hyperprism 2 runs only on Macintoshes with PowerPC microprocessors. Your Macintosh must be correctly connected to the rest of your audio equipment, and the Mac's Sound control panel and Sound Manager system extension must be installed and configured correctly. Finally, if you are using a third party audio card, you must have the card correctly installed and the appropriate drivers either placed in your Extensions folder or accessible through the HyperEngine IO Drivers folder. Click here for more information about Sound Manager & 3rd party sound card installation.

Before Beginning

When first opening your Hyperprism 2 package you will find the following ingredients:

- One registration card
- One CD-ROM
- One Master Key Disk
- One Serial Number Card

Please fill out your registration card and mail or fax it to Arboretum Systems as soon as possible in order to:

- receive FREE technical support
- keep up-to-date on new developments and upgrades
- help shape new Hyperprism features

Note that in order to receive ANY telephone support or technical assistance from Arboretum Systems, you **MUST** be a registered user.

The Master Key Disk is very precious. It contains the program to authorize Hyperprism 2 on a hard disk and also the copy protection codes. Therefore,

You Should Never Lose, Erase or Reformat Your Master Key Disk!

If you do, you will void Arboretum's commitment to the terms of the License agreement.

This Master Key Disk allows you to authorize (two installations) or deauthorize Hyperprism 2 on a hard disk and/or launch Hyperprism 2 from your hard disk without authorizing it by inserting the Master Key Disk when requested.

Installing Hyperprism

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

To install **Hyperprism 2.0** and HyperEngine:

- You will need 17 megs of free hard disk space to install Hyperprism 2 and HyperEngine.

- Place the Arboretum CD in your CD-ROM drive, and double-click on the Hyperprism 2.1 for HyperEngine Installer.

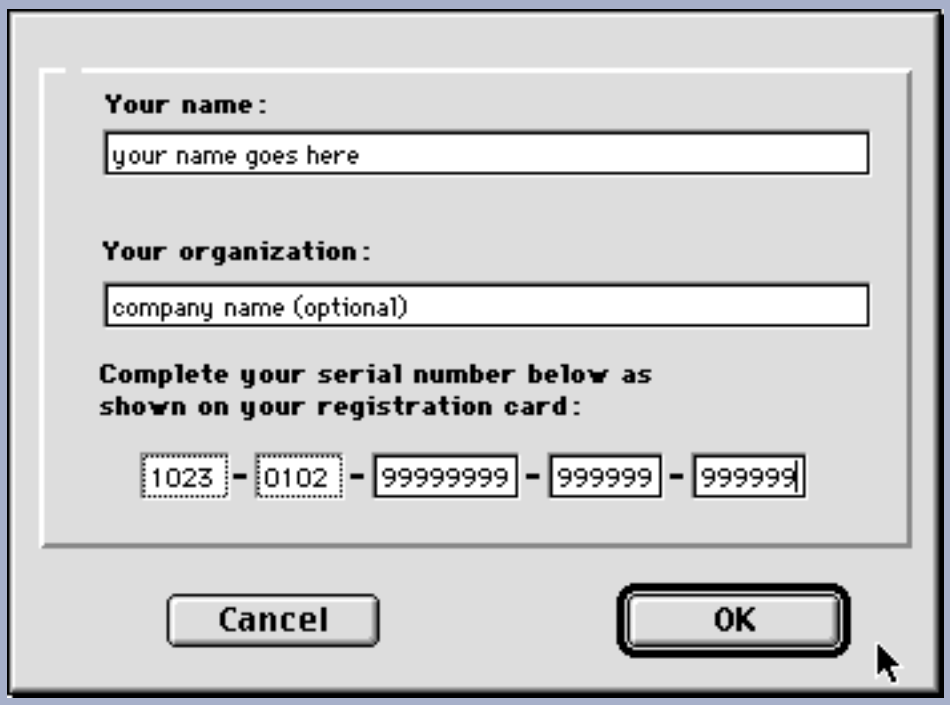
- Follow all on-screen directions. You'll be asked to Accept the License Agreement, then you'll see the Read Me. Be sure to peruse the Read Me for important last-minute instructions which may not have made it into this manual.

- After the Read Me, the following installer dialog appears:

- 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 2 serial number. We've gone ahead and entered the first four 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.

The installer will commence with copying the Hyperprism 2 plug-ins and HyperEngine application to your hard drive. The software will be installed into a new folder called "Arboretum f." Assuming all the previous steps were completed successfully, you should next see the following prompt:

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 Serializers spring into action. The Serializers automatically apply your earlier serial number input to each Hyperprism plug-in, so as to unlock the plug-ins prior to first usage. You'll see the following prompt:

Click on the OK button to proceed and conclude.

Your Hyperprism installation is now finished! The software will be installed into a new folder called "Arboretum f."

Hyperprism 2 and the Macintosh Sound Manager

At this point, your Hyperprism 2 software is ready to run. However, you may need to properly configure your Macintosh Sound Control Panel, or your sound card drivers in order to hear what's going on. There's a special document about that topic right over here...

Otherwise, you can proceed to the next chapter, a quick tutorial that will get you up and running with Hyperprism 2 and show you the basics of HyperEngine, or you can link over to the HyperEngine manual and get the full skinny on the framework which surrounds Hyperprism 2.

Problems? If any unexpected error messages come up during the course of installation, or if the Installer fails to place the HyperEngine application, its IO Drivers and the Hyperprism 2 plug-ins into your target drive's "Arboretum f" 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.

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

Arboretum SYSTEMS

Hyperprism™ Manual

Hyperprism 2

Quick Start Tutorial

Once you have installed and configured the program you are ready to trek through the world of Hyperprism. You needn't follow every step of this tour; if you like, you can skip around.

☞ If you have an optional sound card, you need to configure your Mac's Sound Manager before using it.

☞ If you experience any problems in the course of this tour, refer to the Troubleshooting section in Appendix A.

Caution-Watch Your Levels!

When using Hyperprism and the HyperEngine, please use caution when setting your listening levels. Some of Hyperprism's effects can add large amounts of gain to your original sound. Therefore, it is very important that you are aware of your overall monitor levels to protect your speakers, and most importantly, your hearing, from unexpected loud sounds.

Using high-Q filters, echo effects with large amounts of feedback or adjusting the Pan effect to provide gain are all ways that louder-than-the-original sounds can be created.

You may use HyperEngine's level fader to adjust the overall output volume of the program. However, different effect variations may produce sound at very different levels, so if you are using effects that result in high-gain output, be careful.

Starting Hyperprism

After having installed HyperEngine and Hyperprism 2 and copied the example files onto your hard disk as explained in Chapter 2, start Hyperprism 2 as follows:

➠ Double-click on the HyperEngine icon.

HyperEngine Application icon

➘ HyperEngine launches and displays its playback window.

HyperEngine playback window

These faders and transport controls may be moved to any screen location you find convenient. They may also be hidden and shown by using the Windows menu.

Opening a File reference document

Choose Open Document in the File menu or type its keyboard equivalent command-n.

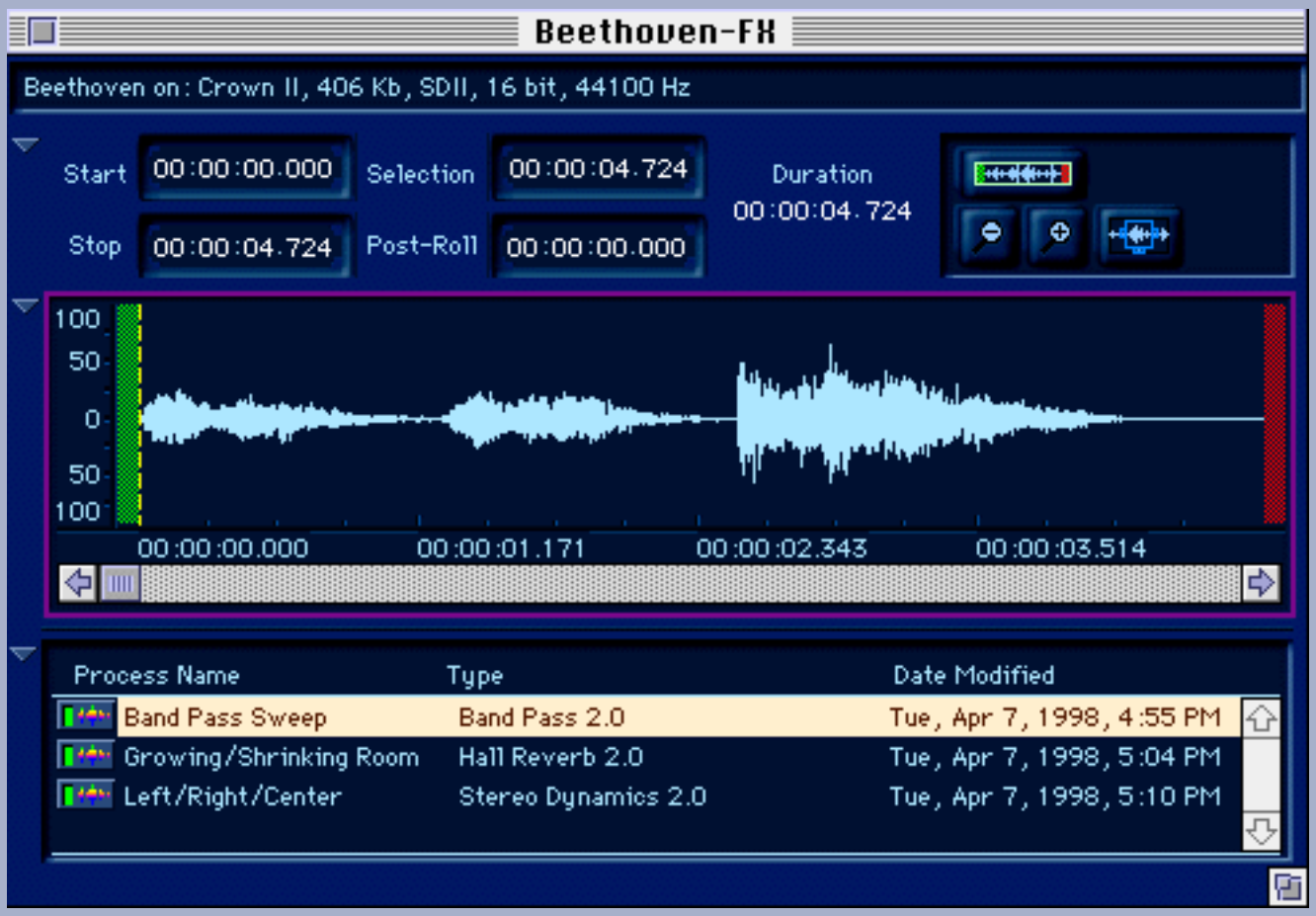
The File menu's Open and Save commands refer to HyperEngine documents, not digital audio sound files. Use the New File (command-n) function to open a sound file. In this example, however, we're opening a File reference document, so choose Open (command-o) instead.

A document selection window appears, showing the contents of your current disk.

Navigate to the folder which contains the Hyperprism examples. (You can't open reference documents from the Arboretum CD, make sure you've copied the Sound Examples folder to your hard drive prior to use.)

Double-click on the File Reference Document named Beethoven-FX (or click it once and select Open).

The screen displays the document reference window for the file Beethoven-FX. Notice that there are three lines near the bottom of the window. Each of these represents a different effect that can been applied to the Beethoven sound.



This is the File Document Window. It contains information about the master sound file and the effects applied to it.

Choosing an Existing Effect Variation

Note the Effect Bypass switch at the left hand side of each process entry. That's the little button with the green light next preceding the process name. This allows you to turn on/off each process discretely. When the green light is lit, the process is engaged. If the Effect Bypass is toggled off, the process will not be heard (but all settings are still remembered).

Click on the the Bypass switches so only the first effect, "Band Pass Sweep" is active, and the others are bypassed.

Double click on the first line, "Band Pass Sweep."

The Blue Window for the previously created example effect "Band Pass Sweep" opens.

Double clicking an effect in the Document Reference Window opens its corresponding Blue Window.

Hearing an Effect

Click Play or hit the space bar.

Clicking the Play button (or hitting the space bar) will play the effect described by the currently selected Blue Window

You should hear the sound play as the Tracer follows the previously defined path. You will hear an orchestral recording excerpt, processed through a sweeping resonant band-pass filter.

Notice that the sound changes in sync with the movement of the tracer.

The meters indicate the sound level, and you may adjust the sound level by clicking and dragging on the Faders.

If you don't hear anything, check the trouble-shooting section in Appendix A.

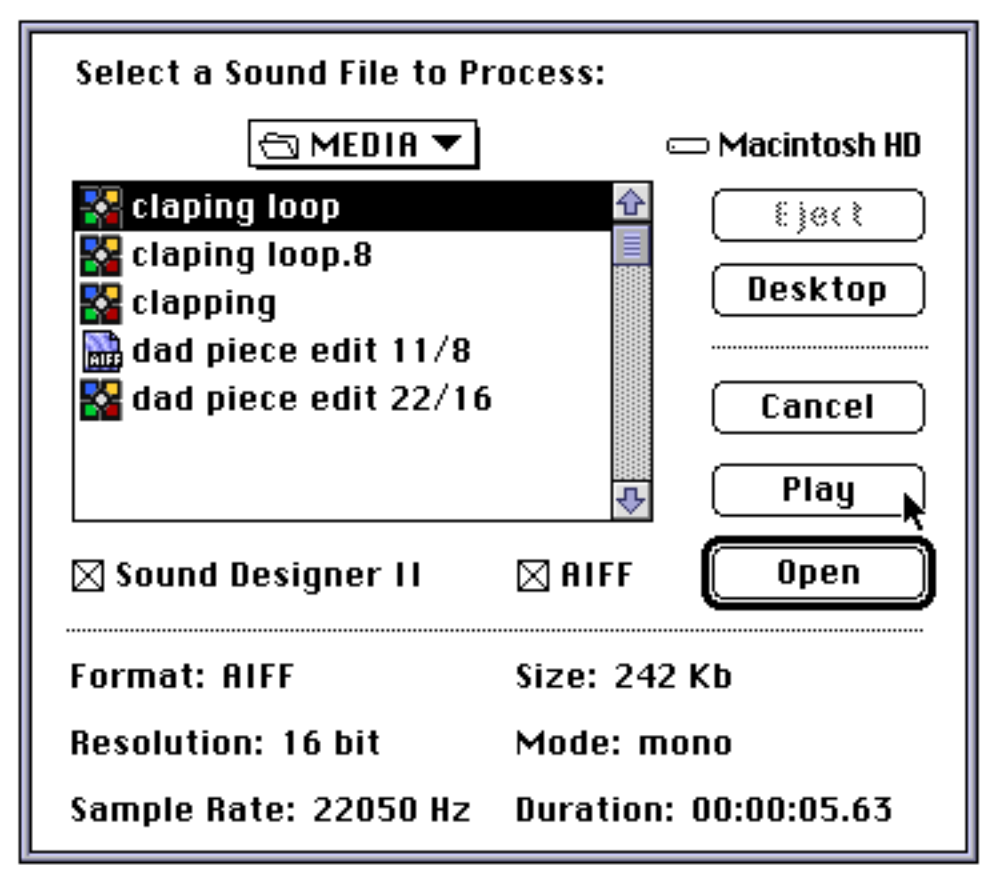
The Document Reference Window contains two additional effect variations, titled "Growing/Shrinking Room" and "Left/Right/Center." Repeat the steps you just followed to hear these examples.

Creating New Effect Variations

This example will show you how to create a new HyperEngine document, load a sound file into it and create a new effect variation for that sound.

Select New File Document (command-n) from the File menu.

A file open dialog box will appear.



This dialog is used to select the sound file you want to add effects to

Navigate to the folder where the Hyperprism example file were stored and select the file "Rhythmic."

A new File Document window appears, displaying the sound's waveform.

Choosing an Effect

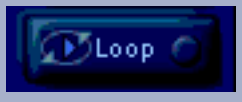
Select Pitch Time Changer from the Plug-ins menu.

A Blue Window for the Pitch Time Changer effect appears.

Choosing a new effect from the Plug-ins menu opens the effect's Blue Window.

Looping Your Sound

Since this is a drum loop, click the Loop button so the sound will repeat automatically while you experiment.



Click here to make your sounds repeat automatically

Clicking the Loop button on the Playback palette causes the sound file to continuously repeat during playback

A/B the Effect

In the Hyperprism Blue window click on the Bypass button (labeled In or Out).

Effect Bypass button, In and Out states

Note that this button duplicates the function of the Bypass switches in the document Process List, but is conveniently located in the effect display. Use this button to compare between original and processed sound.

Creating an Effect Gesture

Notice the tri-state switches to the left of each parameter slider. Click once on any switch to assign a parameter to the horizontal axis of the Blue Window, click twice to attach the parameter to the vertical axis. Click a third time to disconnect the parameter from the Blue Window.

Click on the Draw button, to engage Hyperprism's Blue Window automation recording.

Hyperprism Draw button engages parameter automation

Hit the spacebar, mouse click on the Play button, or just mouse-click in the Blue Window to start playback and automation recording.

Click and drag around the Blue Window to create parameter changes. Listen to the effects parameters change as the mouse moves

As you draw with the tracer, you'll hear the result of varying the parameters of the effect in real time.

When you're finished, hit the Return key on your keypad or click on the Stop button to end playback and disengage tracer recording...Or wait until you've reached the end of file, if you're not in Loop mode.

Click here to end recording/playback. The Return key is a shortcut for this function.

Now, click on the Play button to play back the effect you created.

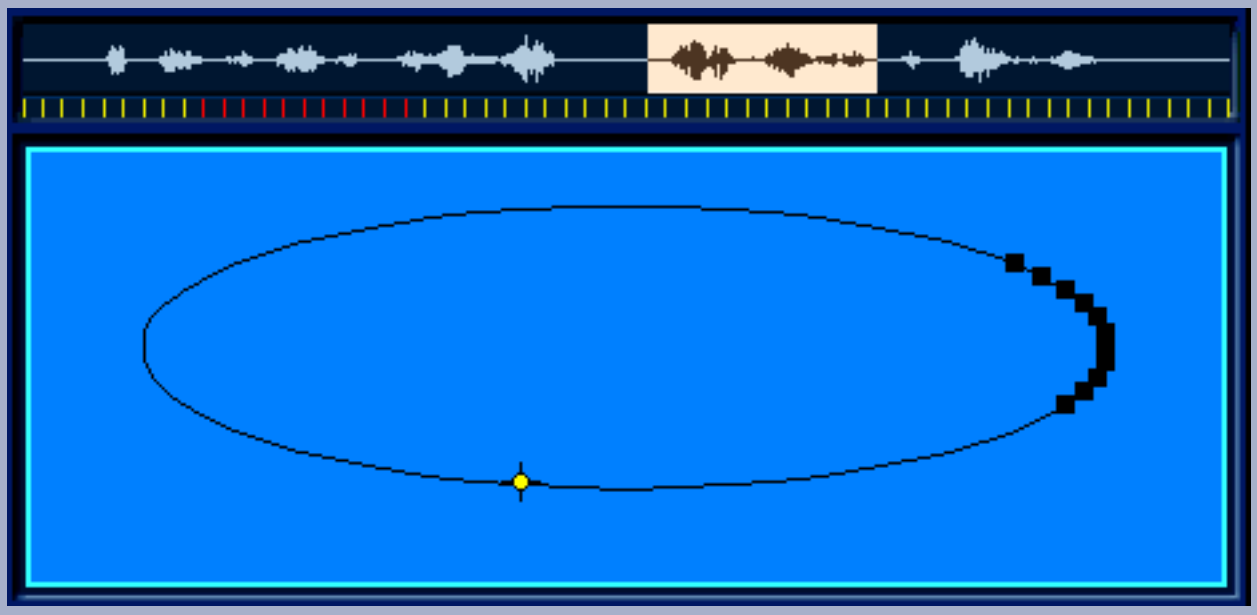
Hyperprism 2 remembers your movements and retraces them automatically.

Editing A Tracer Path

If you liked your last tracer automation, but you want to make a few minor changes to it, click on the Automation Edit button.

Click here to edit path automation breakpoints

Now mouse drag through the Blue Window so as to select a portion of the automation path you've just created. The selected breakpoints will be highlighted.



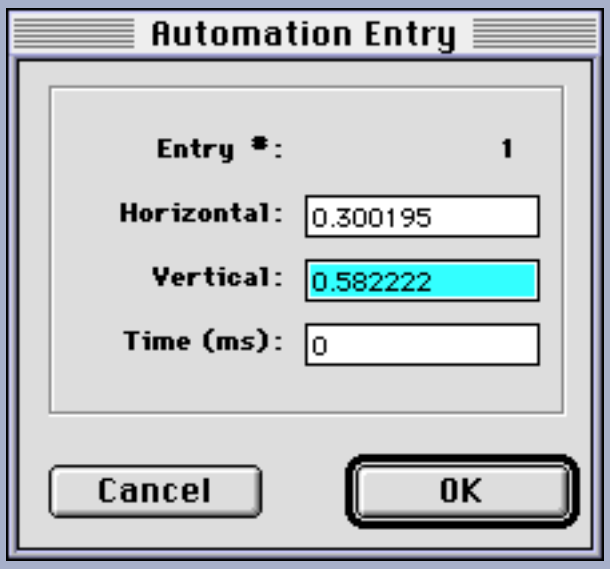
Note automation breakpoints selected for editing, screen right.

Now you can see each of the selected breakpoints, which represent every automation event recorded along the path of the tracer movement you just made.

To alter any breakpoint location, simply mouse click on the point (the mouse cursor will turn into a hand when it is positioned for grabbing) then drag the point to a new location in the Blue Window.

Play back your sound to hear the result of your new edit.

If you want to change the time coordinate for any given breakpoint, just double-click on that point and a time edit dialog will appear:

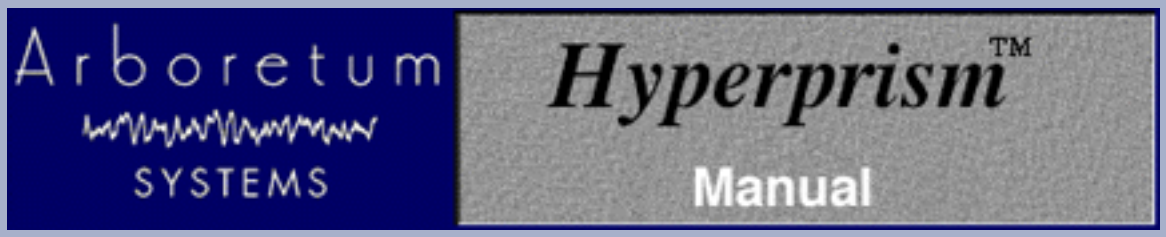


Enter the time coordinate and physical location for specific automation breakpoints with this dialog

Enter the new time coordinate for the specified automation event. The setting is in milliseconds, and is measured from the beginning of the audio file or current selection.

If you like, try other effects by selecting them from the Plug-ins menu. There's more information on these functions, and additional step-by-step instructions in the documents that follow....Read on!

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



Hyperprism 2

Hyperprism 2 and HyperEngine

This Chapter will describe in brief how to prepare to process your own sounds. We'll cover the basic information you'll need to get your sounds into Hyperprism, and explain how HyperEngine keeps track of your sound files and information about the effects you apply to it.

A lot of the work you'll do with Hyperprism 2 will actually take place in the accompanying HyperEngine, which provides input/output, editing and management services to the Hyperprism plug-ins (as well as to other Arboretum products such as Ray Gun and Ionizer). HyperEngine's core functions and all plug-in effects for HyperEngine can be accessed through HyperEngine's menus. Click here to see the chapter detailing every function and command. In addition to the menu commands, there are a number of time-saving keyboard shortcuts for all the main features.


The material in this section is merely a brief summary; for complete details on any function or process, click on the appropriate hyper link and you'll be taken to the appropriate document.

Getting Started

The Arboretum CD-ROM you received with your Hyperprism purchase contains several example audio files for demonstration purposes. These files are used in the tutorial sections of this manual; you should probably copy them over to your hard drive if you plan to go through the step-by-step sections of this manual. However, you'll quickly want to be processing your own sounds.

Hyperprism 2 works with digital audio, which can be either a live play-through stream or stored as disk files (in Sound Designer II or AIFF format). In order to add effects to your own material, you must first get those sounds into your computer.

There are several ways you could do this. If you are working with other digital audio applications, such as waveform editors or multi track recording systems, you can open the sounds you've recorded using those programs, as long as they

are in a compatible file format.

HyperEngine's Record command provides another way to get external audio into hard disk sound files, directly from live sources through your sound card or Macintosh audio inputs. You can even record and process audio at the same time, allowing you to do things like add reverb to a singer or filter hum from a guitar track, while it's being recorded.

If you have a suitable CD-ROM drive, you may even be able to extract audio from regular compact discs, either via direct SCSI data-transfer, or by recording it's analog output. If your computer has a built-in Apple CD-ROM drive you can use the AppleCD Audio Player desk accessory to play audio CDs right into Sound Manager for recording or live processing.

Before we begin creating sound files, we must first consider the file formats Hyperprism 2 is able to work with.

File Formats: SDII or AIFF?

Hyperprism 2 supports mono and stereo files in the Sound Designer II (SD II) and Audio Interchange File Format (AIFF). These files can be either 8- or 16-bit resolution, with sampling rates of 11.025, 22.050, 44.1 and 48 kHz. (If these terms are unfamiliar to you, click here.)

There is no sound quality difference between using the Sound Designer II format instead of the AIFF format, or vice-versa, and Hyperprism 2 is equally comfortable with both. You may wish to choose one over the other if you will be opening your Hyperprism 2-created sound files in other applications for further processing, editing or mixing. If this is the case, determine which format is more convenient for the other applications you're using.

 If you're recording audio for eventual use on a CD, we recommend that you always create 16-bit stereo 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 Toast and MasterList CD accept for burning audio CD-Rs.

Recording Sounds

Before you record, you'll need to connect the external signal to your Mac's "sound input device." Hook up the hardware, then configure the Sound Control Panel. Note that your sound source could be the mic jack, a sound card or even an audio CD played from your Apple CD-ROM drive. See our chapter on the Sound Manager and choosing sound in and sound out devices.

By using the Record function, you'll be able to record external sounds into your own hard disk files. Once captured, they can be processed through Hyperprism 2 effects. You can even processing while recording! Recording is accomplished by opening a HyperEngine Play-Thru document, from the File menu. For the section on HyperEngine Record function, with step-by-step instructions, click here.

HyperEngine Documents and Sound Files

HyperEngine uses documents to keep track of your original sound file and the different effect variations you apply to it. ("Document" is a generic Macintosh term referring to any file created by an application.)

Many audio editing programs simply modify your original sound file. Because HyperEngine always processes files non-destructively, new users may not be completely clear on how HyperEngine manages sound files and effects documents.

When you process a sound with Hyperprism 2, you are working with two files. One is the sound file itself, the other is a separate reference document HyperEngine creates to store your paths, edits and other effect settings. Hyperprism File reference documents, by default, are denoted by the "-FX" file name suffix, as seen in the example below.

Sound Designer II sound file (left) and Hyperprism File reference document (-FX) icons.

There's a second type of of reference document used by HyperEngine: The Play-Thru document, which is utilized for real-time processing and recording live signals. These documents have no default file name extension, and can be saved and opened for later use with any live play through signal.

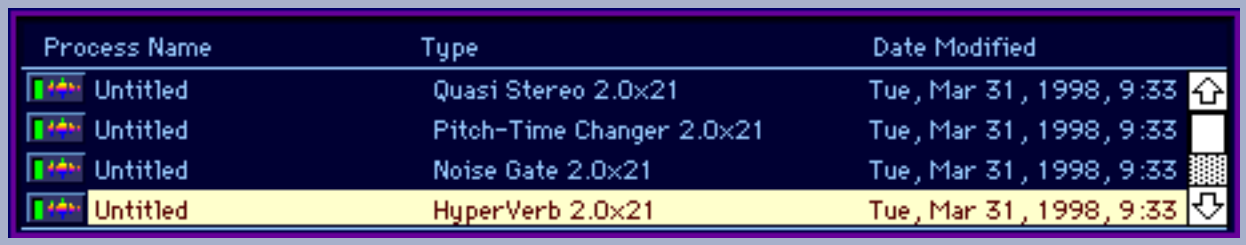
This document architecture makes it possible to try out many different effect variations immediately, rather than destructively processing a file and altering your original media. Once you have found effect settings you like, you can use the Process to New File command to create a new sound file processed with your effect settings.

Both the File Reference and Play-Thru documents store references to any effects you have chosen, as well as any parameter automation associated with the effect. File Reference documents additionally contain a reference to the sound file's location on disk and a list of any edits which have been applied to the file.

Whenever you want to open a new sound file from disk for processing, you need to create a New File Reference Document; try the command-n keyboard shortcut. The Open command (command-o) opens an existing File or Play-Through Document, NOT a sound file. This is a little different from how many other audio programs work; remember to use command-n any time you want to start editing a new sound file.

HyperEngine Process List

Each HyperEngine Reference Document maintains a list of effects which have been added to the document, this list is called the Process List.



HyperEngine Process List

Each time you call an effect from the HyperEngine Plug-ins menu, that effect is added to the Process List. Effect processing happens in the sequence shown by the Process List; in the above example the Quasi Stereo effect is applied to the signal first, followed by the Pitch-Time Changer, Noise Gate and HyperVerb. Click on any Process List entry to highlight the selection (in white), once highlighted an effect can be mouse-dragged to a new position in the signal chain. Double-click on any Process List entry to view the effect's Blue Window display. To Bypass any effect, so its processing is temporarily disengaged, mouse click on the Bypass switch at the far left side of the Process List entry. Effects can be removed from the Process List by highlighting the list entry, then pressing the Delete key.

Any effects in the Process List (which are not Bypassed) will be applied to your live or file-based sound.

Remember, even if an effect's Blue Window has been closed so it is no longer visible on screen, that effect and its associated parameter automation will still be processing your audio so long as the effect is in the Reference Document's Process List and is not Bypassed.

Exporting Effects: Process to New File

Once you've designed an effect you like, you'll probably want to make a new "treated" copy of your sound file for export

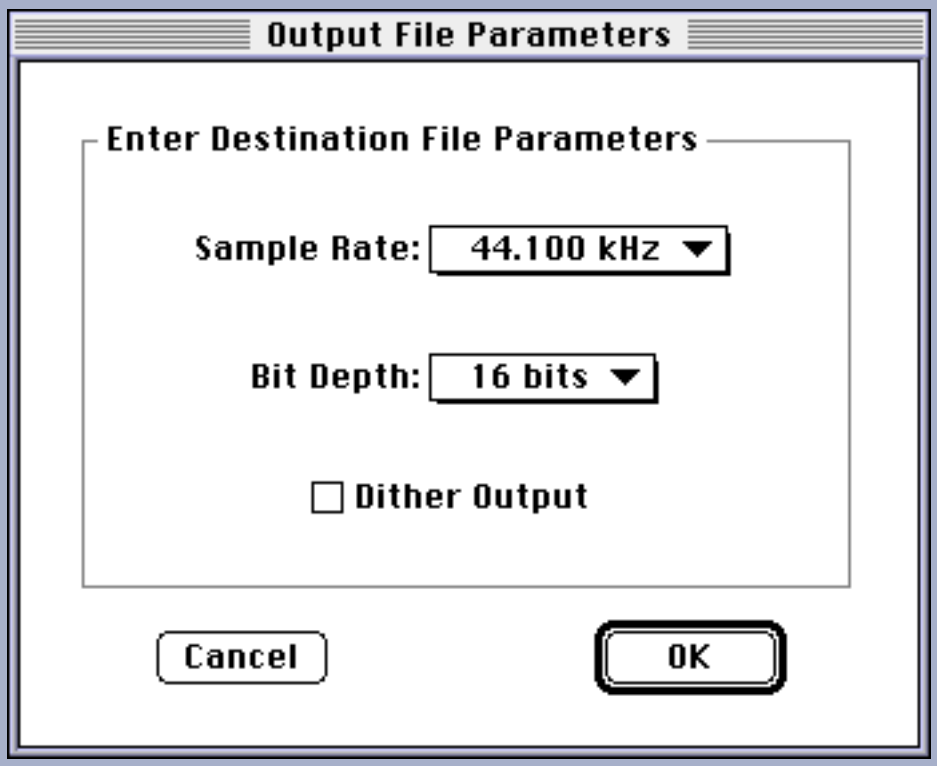
into other programs. This rendering process is accessed through the Process to New File command in the HyperEngine File menu.

When generating a new sound file you'll specify its file format (SDII or AIFF), sample rate and bit-resolution.To create a new file processed with an effect:

You must open the File Reference Document with the sound file and effect variation(s) you wish to use for the new processed file. Simply click on the Reference Document's header to make sure it is the front-most of any open Blue Windows.

Choose the Process to New File command from the File menu.

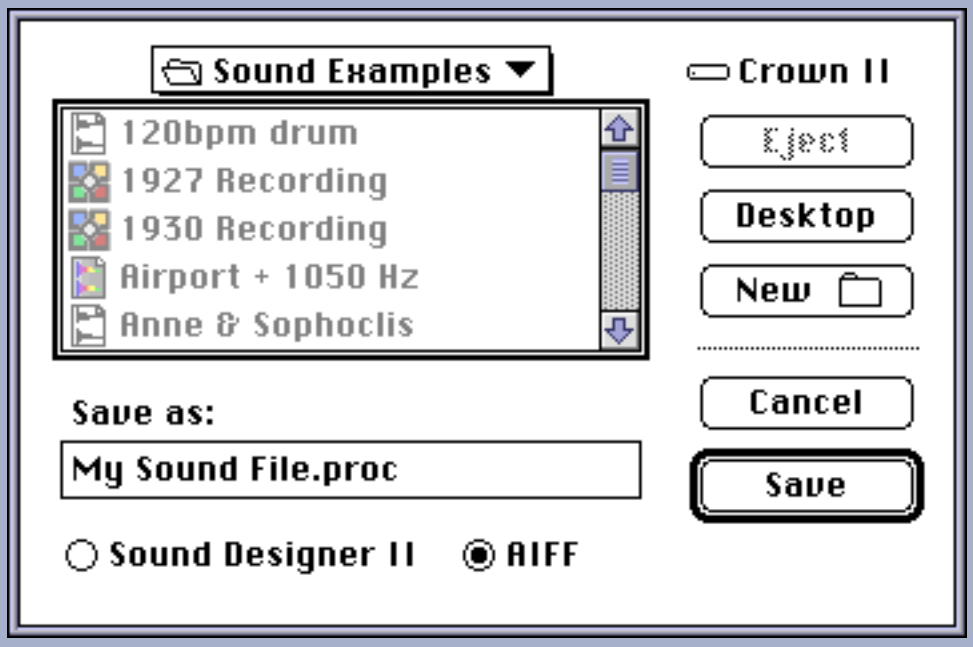
The Output File Parameters dialog will be displayed:



When using the Process to New File command, you can change sample rate and bit-resolution of the processed file.

Choose the desired sample rate, bit-resolution and dither option and click OK. (More information on these topics is found in Appendix B.)

A Save dialog will appear:

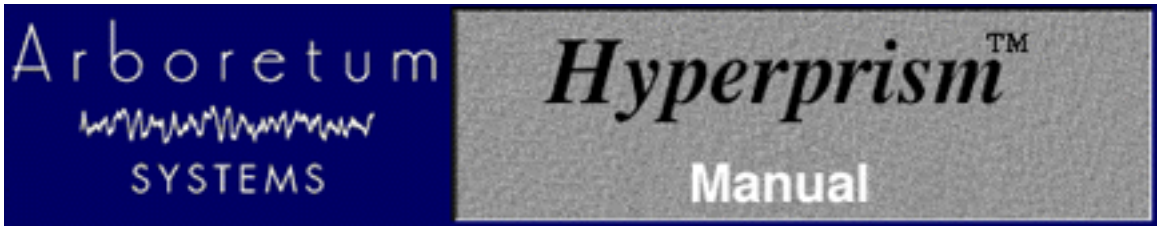


Choose the desired name and location for the new processed file and click OK. You may select the sound file format for the new file (Sound Designer II or AIFF).

HyperEngine will proceed to calculate a new processed version of your audio file, which can then be exported to any other sound editing program, or re-opened in HyperEngine for further editing and processing.

When generating a lengthy new file, you may put Hyperprism 2 "in the background" simply by clicking the Desktop or another application in the Finder. Hyperprism 2 will continue to render the new file as you work on something else.

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



Hyperprism 2

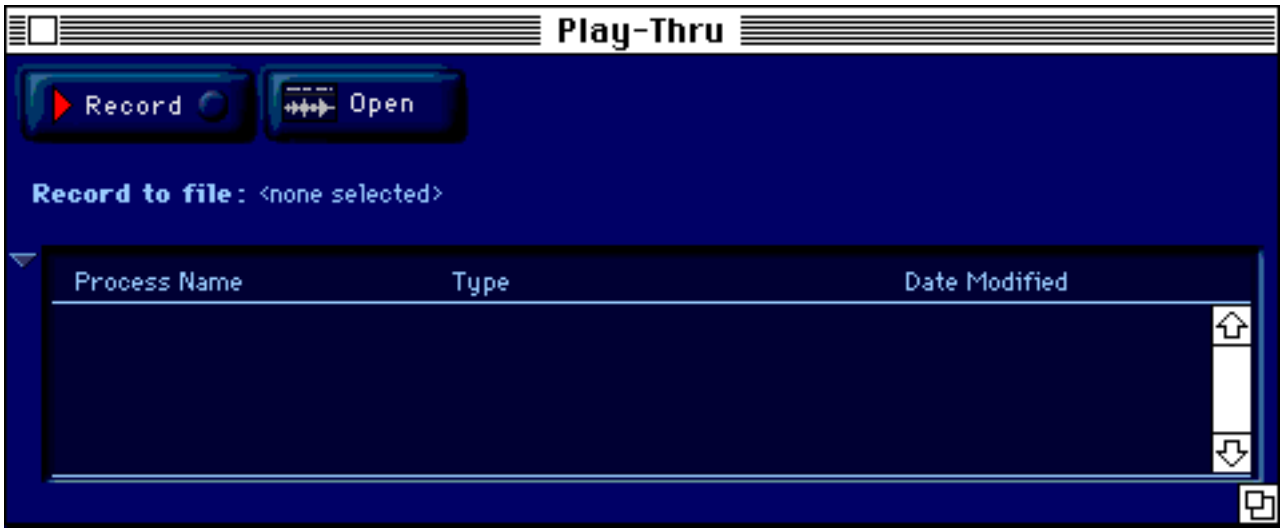
Hyperprism Controls

To use Hyperprism 2 you'll use three different groups of controls: monitoring/playback controls and Document Reference windows in HyperEngine, and the Hyperprism Blue Windows. It's important to understand all these features, because they are used by all of Hyperprism's effects.

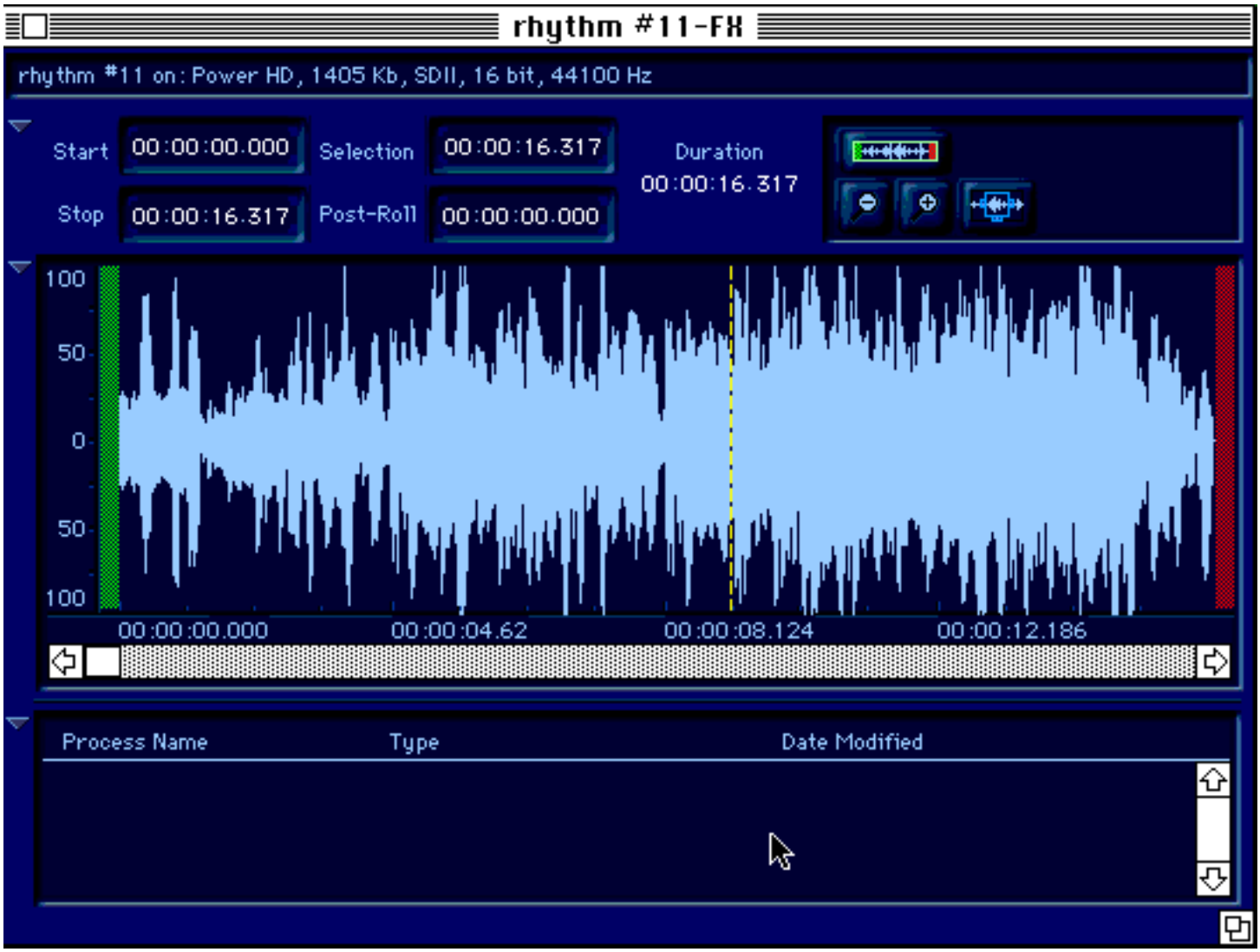
The Playback Window appear on the left of your screen when HyperEngine starts up. It controls playback, monitoring and metering when using Hyperprism.

HyperEngine Playback Window

The Reference Document's purpose is to keep track of all the edits and effect you apply to your original sound. There's two types of Reference Documents: The Play-Thru document for recording audio or doing live, on-the-fly signal processing, and the File Reference Document for working with sound files on already on disk. Here's examples of how they appear:



HyperEngine Play-Thru document, for working with live audio signals.

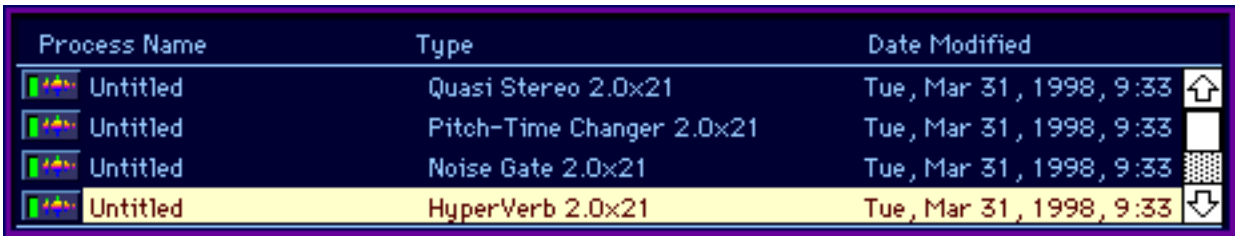


HyperEngine File Reference Document, for working with sound on disk.

Hyperprism's distinctive Blue Window is used to alter the settings of each individual effect parameter.

Hyperprism's Blue Window interface is used to dynamically control effect parameters.

One or more Blue Windows can be stored and accessed in the Process List of any Reference Document.



Process List, as it appears in HyperEngine Play-Thru and File Reference Documents

The Blue Windows' ability to dynamically control effect parameters--graphically recording and playing back "patch" changes in real-time--is a key feature of Hyperprism, setting it apart from all other audio software products. As you use Hyperprism 2, the interaction between the program's three control elements (monitor/playback controls, Reference Documents and Blue Windows) will become obvious. However, on first using the package, the relationships may not be clear until you have familiarized yourself with all of the control groups.

The remainder of this chapter focuses on controls found within Hyperprism's Blue Window interface. Only elements found within the effect display will be covered here. The detailed documentation on the HyperEngine Playback Window and Reference Documents is provided in the HyperEngine manual.

The Blue Window

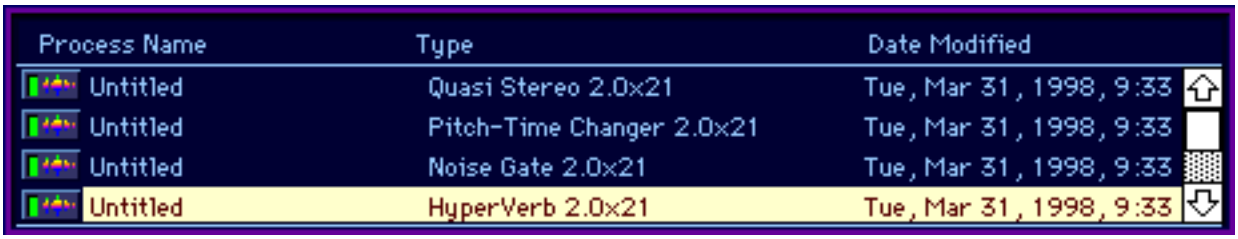
Hyperprism's Blue Window is used to design audio effects. It was created with the goal of giving you gestural control over signal processing, allowing you to "play" your processors as if they were musical instruments.

If you were changing settings manually on a hardware signal processor, its likely that you'd want to adjust two or more knobs simultaneously. But when you're using a mouse as your input device, you're limited to clicking on one point at a time. Hence the X/Y axis graphical control metaphor embodied by the Blue Window. Hyperprism allows you to connect any effect parameter to the Blue Window's horizontal (X) axis, or the vertical (Y) axis, so that you can change two or more parameters at once, by just moving the mouse around.

Use the Blue Window Axis Switches to map each parameter to the X or Y axes.

And here's the clencher: Not only can you listen to these parameter changes as they're processing your signal in real-time, you can also record and edit any Blue Window mouse movement, then ultimately apply your dynamically-changing effects to a new sound file (or to live signal processed with a Play-Thru document).

Select any effect from HyperEngine's Plug-ins menu. The effect's Blue Window will open on screen, and the effect will be added to the current HyperEngine Reference Document. You can add many Hyperprism effect to a single Reference Document. Each effect appears as an entry in the Reference Document's Process List.



Process List, as it appears in HyperEngine Play-Thru and File Reference Documents

The Process list allows you to manage Blue Windows. Here you can Bypass any effect (using the toggle switch at the left hand side of each list entry). You can delete any effect from the list by clicking on the list entry to highlight the effect, then hitting the Delete key. Re-arrange your processing chain by clicking and dragging list entries around in the Process List display, putting one effect before/after any other. To open and view an effect simply double-click on any Process List item and the effect's Blue Window will appear.

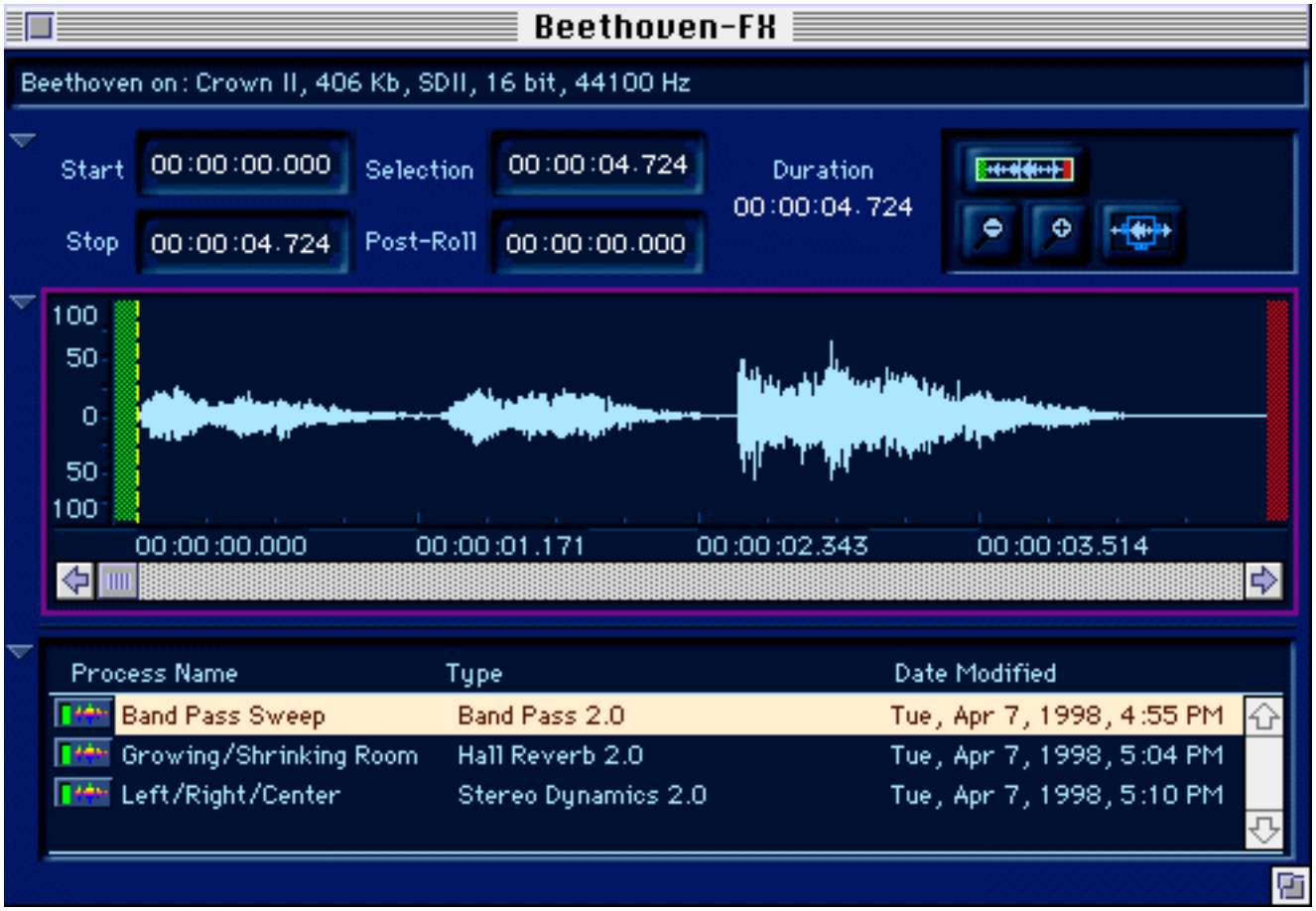
 You can add many different effects to a Process List, however there are processor-speed limitations on how many simultaneous effects (and associated parameter automation) can be calculated in real-time on any Mac. If your multi-processing chain is so long that real-time processing becomes intermittent, try turning off a few effects, increase the Buffer size in HyperEngine's Configure Audio System dialog, or just use the Process to New File command to render your new sound to a disk file.

There's more on the Process List, including step-by-step operating instructions, over here.

Now let's begin to explore the world of sound transformation with Hyperprism. This example uses the same file mentioned in an earlier tutorial section of this manual.

Opening a Blue Window

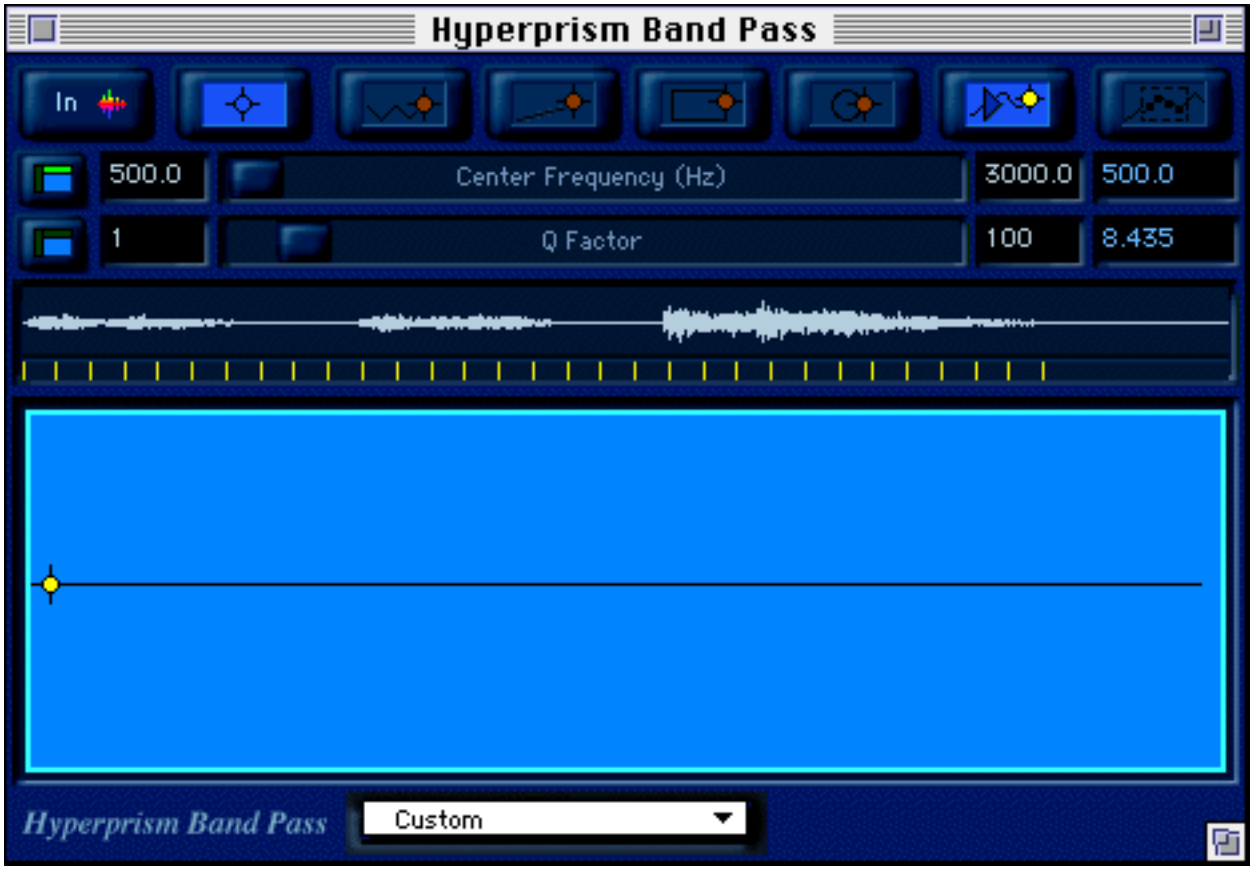
 Open (command-o) the example HyperEngine File Reference document Beethoven-FX (which should be included in the Sound Examples folder on your CD).



File Reference document Beethoven-FX

Click in the Plug-ins menu and choose Band Pass.

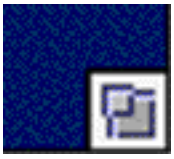
A new Blue Window appears.



The Blue Window interface

Re-sizing the Blue Window

The Blue Window is re-sizable via the resize tool at the extreme bottom right of the window.



The resize tool

Click and hold on the resize tool, then drag the mouse pointer to enlarge or shrink the window. This feature is useful when you want more precision in manipulating the parameters. With a large Blue Window you can realize more precise control over effect parameters if required. Small windows can be useful for changing values over a wide range with only a small gesture. The pixel-to-parameter relation can be fine-tuned to your needs using the grow box and/or re-scaling the minimum and maximum parameter values. Note that resizing the Blue Window does not change the range of parameter control (which is done with the minimum/maximum value fields) it merely allows you to adjust the screen resolution applied to that range.

A Blue Window has seven parts:

Title bar

Contains the name of the Reference Document (by default the name of the original sound file followed by -FX) plus the name of the chosen effect (which in the case below is Untitled), i.e. "120bpm drum-FX::Untitled."

Hyperprism title bar

Switches

Blue Window switches

These buttons allow you to control Hyperprism's Blue Window parameter automation features.

Bypass 

Engages/disengages effects processing. Use this switch to "A/B" your sound with, and without, the current effect settings. Duplicates the functionality of the HyperEngine Process List Bypass switches.

Audition 

Allows you to make mouse movements in the Blue Window without creating any parameter automation. Use this mode if you're simply trying out settings or dialing up a new effect. When Audition is on, playback will begin as soon as you mouse-click in the Blue Window. Hold down the Shift key in Audition mode while moving the mouse (without holding the mouse button) to adjust settings without starting playback.

Draw 

Turns on Blue Window automation, which will record and re-create any mouse movements made in the Blue Window, so that you can automate parameter changes. When the Draw tool is selected, playback will begin as soon as you mouse-click in the Blue Window. You can Pause playback (spacebar) then click in the Play Zone again to resume automation recording. Hold down the Shift key in Draw mode while moving the mouse (without holding the mouse button) to adjust settings without starting playback.

Line 

Draws a straight line between any two successive mouse click and release events in the Blue Window. Assists in creating linear parameter automations. Hold down the Option key when drawing to bring up the Automation Shape Parameters dialog, which lets you define the characteristics of the automation. This tool is disabled during playback.

Rectangle 

Draws a square or rectangle between any two successive mouse click and release events in the Blue Window. Assists in creating geometric parameter automations. Hold down the Option key when drawing to bring up the Automation Shape Parameters dialog, which lets you define the characteristics of the automation. This tool is disabled during playback.

Oval 

Draws a circle or oval between any two successive mouse click and release events in the Blue Window. Assists in creating geometric parameter automations. Hold down the Option key when drawing to bring up the Automation Shape Parameters dialog, which lets you define the characteristics of the automation. This tool is disabled during playback.

Automation Play/Bypass 

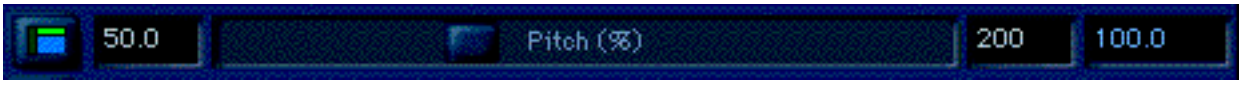
Allows you to toggle on/off the playback of Blue Window parameter automation. Use this to temporarily dis-engage the effect from the automation paths you're creating.

Automation Edit 

Allows you to select automation breakpoints for editing. Click on this button to begin breakpoint editing. Once you're in edit mode click and drag through any section of Blue Window tracer path to highlight automation breakpoints, which can then be moved to new locations or assigned new time coordinates. Use the Select All keyboard shortcut (command-a) to highlight all breakpoints for edit access.

Parameter Sliders

The central part of the Blue Window is a number of slider controls. These are used to control the parameters of each effect. The Pitch-Time Changer shown here has six sliders. Other effects may have as few as one. Each slider is labeled with the name of the parameter it controls. You can mouse drag on the slider's knob to alter the parameter's setting. You'll notice that when a parameter is connected to the Blue Window, any parameter moves will be reflected by the slider knob.



Effect parameter slider

Axis Switches

Hyperprism allows you to connect any effect parameter to the Blue Window's horizontal (X) axis, or the vertical (Y) axis, so that you can change two or more parameters at once, by just moving the mouse around. The Blue Window Axis Switches, which are located to the extreme left of each parameter slider, allow you to map the parameter to the horizontal (X) axis, or the vertical (Y) axis.

Blue Window Axis Switches, note the three possible states. Use the Axis Switches to map each parameter to the X or Y axes.

Click once on any Axis switch for horizontal control, twice for vertical control, or a third time to disconnect the parameter from the Blue Window.

Minimum/Maximum/Current value fields

Directly to the left and right of the parameter slider are its minimum and maximum value fields. These numeric fields

allow you to define the range of control for each parameter. At the far right is the Current Value field, which reflects the parameter's current setting.

Minimum value (highlighted)

Maximum value (highlighted)

Current value (highlighted)

Double-click in any value field to highlight the field for direct numeric entry of a new value from the keypad. You can also click and drag from any value field to increase or decrease its setting. Holding down the Option key while mouse dragging increases the speed of numeric change, allowing you to scroll by hundreds of units. Hold down both the Option and Command keys while mouse dragging to increment/decrement values by 1/100th of a unit.

The Current Value field changes color depending on its parameter's automation state. When the Blue Window is "at rest," when the parameter is disengaged from Blue Window control, or when in Audition mode, the Current Value field is light blue (as above). When recording path automation using the Draw tool, Current Value field turns red. The Current Value field switches to yellow when playing back parameter automation.

Current Value when recording automation (field turns red)

Current Value when playing automation (field turns yellow)

Waveform Display

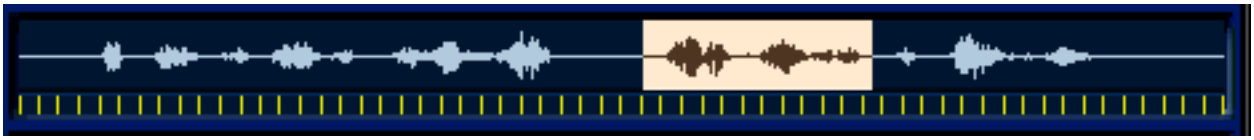
Blue Window Waveform Display

Located below the parameter sliders is the new Blue Window Waveform Display. This display is a miniature version of the display found in HyperEngine's Waveform Editor, and is situated inside the Hyperprism effect display for your convenience, saving you from having to constantly bounce between windows.

Click and drag across any portion of the Waveform Display to select a range of your audio file for playback, processing or parameter automation. The selected region will be highlighted in white. Hold down the mouse button while dragging to extend or contract the current selection. If you want to alter a selection (after you've released the mouse button) simply hold down the Shift key, then click and drag again near the beginning or end of your current selection to lengthen or constrain your selection.

Only the region selected in the Waveform Display will be played back, processed, or have Blue Window parameter automation applied. If no region is selected then the entire file is active.

Automation Ticks



Parameter Automation Ticks (yellow lines along bottom of Waveform Display)

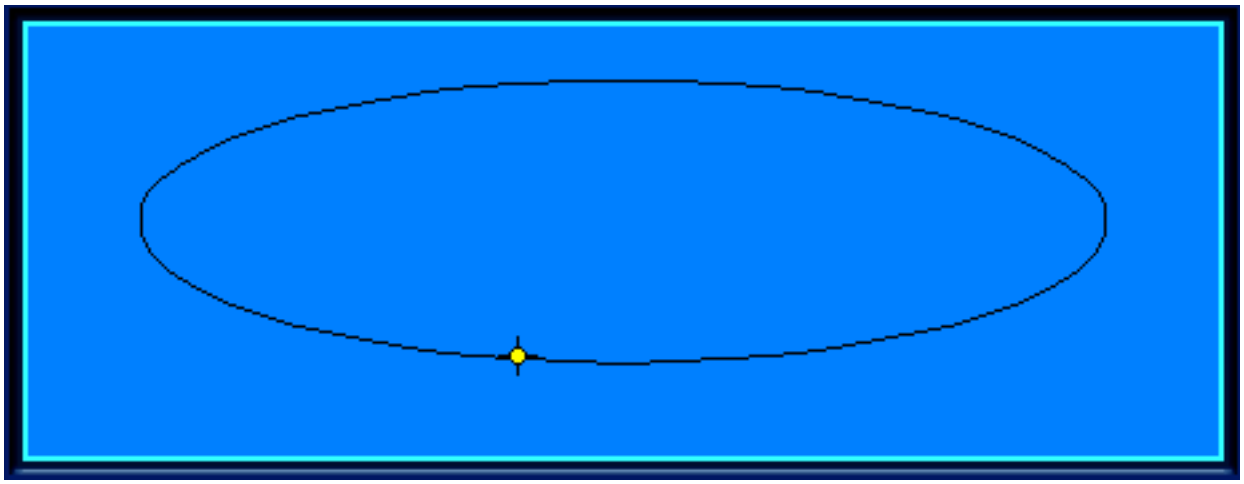
Automation Ticks appear directly below the Waveform Display when parameter automation is associated with an effect. Each Tick (yellow line) represents one automation breakpoint along the tracer path. Ticks are created automatically whenever Blue Window automation is engaged.

Click and drag across the Automation Ticks display to select any section of automation. Hyperprism will automatically toggle into Automation Edit mode. Selected Ticks will highlight in red, and the corresponding breakpoints along the tracer path will likewise be highlighted, and can be repositioned to new times and locations.If you want to alter a selection (after you've released the mouse button) simply hold down the Shift key, then click and drag again near the beginning or end of your current selection to lengthen or constrain your selection.

The Waveform and Automation Ticks displays allow you to extend a selection from one display into the other, simply by dragging into the neighboring display. And note that any time you select breakpoints along a tracer path, the corresponding Automation Ticks will highlight

Blue Window Play Zone

The Play Zone is the blue part of the window. When the Blue Window is open, either the Play Zone or one of the numeric fields controlling effects parameters will be active. Clicking on the Play Zone, or hitting the Tab key until the Play Zone highlights will activate it. The Play Zone's border is highlighted by a light blue color, indicating "active" status. Clicking in the Play Zone when the Audition button is engaged will begin audio playback so you can preview different combinations of settings. Clicking in the Play Zone when the Draw tool is engaged starts audio playback and activates parameter automation recording.



The sound begins to play when you press the mouse button and drag around the Play Zone. As you press the mouse button and drag the tracer across the surface, you change the parameters of the effect algorithm.

 There's more on parameter automation, coming up in the next chapter...

Presets Pop-up

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.

You can save and recall any combination of effect parameter settings by using the Presets pop-up menu. Presets store the minimum, maximum and current values for each effect parameter, but they do not save Blue Window parameter automation paths (which are saved in HyperEngine Reference Documents).

Whenever you alter a parameter value, the Presets pop-up will display the word "Custom," indicating that you're customizing the settings and anticipating that you may wish to save a preset. When you've dialed up a combination you like, select Save from the pop-up menu; a dialog will ensue in which you name your new preset. Once saved, a preset will be made available to all instances of an effect type, so that you may access favorite presets within any Reference Document. User Presets can be Renamed or Deleted by calling up the appropriate menu option from the Presets pop-up. Factory Presets may not be deleted.

Presets are saved in a file named (by default) Hyperprism Presets, which is located (by default) in your System's Preferences folder. Hyperprism Preset files can have any name, and reside at any location on disk. To create a new collection of Presets, select New Preset File; you'll be prompted to specify a name and location for your new collection. Select Load Preset File to recall any collection of Presets, or Load Default Preset File to restore all of the original factory presets. Selecting the Default option resets the effect to its factory settings without changing the currently loaded Preset file.

Editing the Presets File

The individual Presets are stored as resources in a file named (by default) Hyperprism Presets. Advanced users may use ResEdit or a similar Macintosh resource editing utility to copy or delete individual Presets between Hyperprism Presets files. For example, this would allow you to exchange presets with other users or archive rarely used Preset files.

 Always make a back-up copy of your Presets file before opening it with ResEdit or similar utilities!

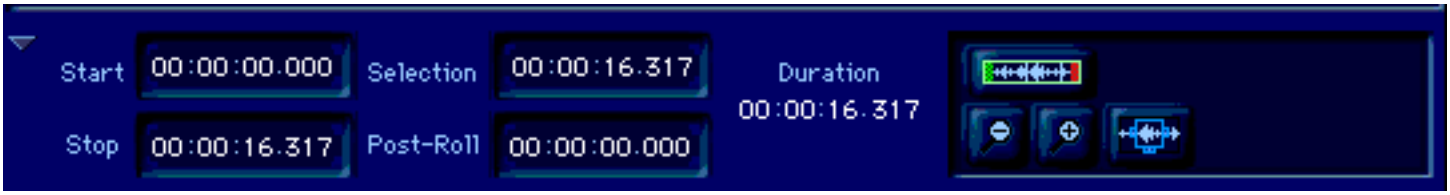
Post-Roll

Some Hyperprism effects can extend the length of your original sound. For example, adding reverb to a sound will almost certainly increase its length.

However, by default, HyperEngine will stop playback when it reaches the end of the original sound's duration. In this case, reverb tails, lingering echoes and similar effects can have their final decay unpleasantly truncated.

By increasing the Post-Roll setting, the lingering parts of an effect can be preserved.

The Post-Roll setting is displayed in the Time and Tools section of each HyperEngine File Reference Document. The value shown will be applied to the selection when previewing in Blue Window, and when using the Process To New File command to create a new audio file.



Post-Roll display (lower/center)

Changing Post-Roll

To Change the Post-Roll setting:

- Double-click in the Post-Roll numeric field with the mouse.
- Type in a new value. It will be taken into account when the file starts playing.

Post-Roll settings are ignored when Loop playback is enabled.

HyperEngine 2.3 Keyboard Shortcut Summary

Most HyperEngine keyboard shortcuts duplicate functions associated with HyperEngine menu commands. However, there are also a number of "hidden" shortcuts, including spacebar and return to control playback functions, control-drag to create new View Frame, option-drag to paste a copy, shift-drag to resize selection and the arrow keys which control zoom functions.

Note that these key commands & shortcuts apply only to HyperEngine, many of these commands are de-activated when you have a plug-in such as Ionizer or Hyperprism at the top of your desktop.

Learning key commands is the secret to becoming a "Power User," and will result in increased convenience and speed. Click on any of the links below for a more complete description of each key command's function.

Play Functions:

- spacebar - Play/Pause
- return - Stop Playback and Return to Start

File/Document Functions:

- command-n - Create New File Document (open sound file)
- command-o - Open existing File Reference Document
- command-w - Close window
- command-s - Save Reference Document
- command-i - Open Process Info dialog ("Get Info")
- command-q - Quit HyperEngine

Editing Functions:

- command-z - Undo/Redo last edit
- command-x - Cut current selection from waveform display
- delete - Cut current selection from waveform display
- command-c - Copy current selection to clipboard
- command-v - Paste clipboard contents to current selection or cursor location
- command-a - Select all of waveform overview
- option-mouse drag - Copy selection to new location, leaving original selection behind

View Frame (Zoom) Functions:

- control-mouse drag - Define New View Frame (zoom area)
- command-/ (forward slash) - Switch waveform display to Zoom View/Over-view
- command-= (equals) - Zoom in (contract View Frame)
- up arrow - Zoom in (contract View Frame)
- command-- (minus or hyphen) - Zoom out (expand View Frame)
- down arrow - Zoom out (expand View Frame)
- left arrow - Move View Frame (zoom area) to the left
- right arrow - Move View Frame (zoom area) to the right

Selection Range Functions:

- shift-mouse drag - Adjust Selection Start/End
- shift-mouse click - Extend Selection Start/End
- command-; (semicolon) - Move Selection Range Left
- command-' (apostrophe) - Move Selection Range Right
- command-[(left bracket) - Go To Selection Start
- command-] (right bracket) - Go To Selection End
- command-\ (backslash) - Fit Selection to Window
- command-r - Repeat Selection (multiple paste)
- mouse click - De-select

Hyperprism 2 Key Commands

While in a numeric value field:

- drag +/- 1.0
- option-drag +/- 100
- command-option-drag +/- 0.01
- up/down arrow +/- 1.0
- option-up/down arrow +/- 100
- command-option-up/down arrow +/- 0.01
- left/right arrow - move cursor position
- tab - move from left to right between text boxes and to next row of text boxes, also into Blue Window and back to top.

*** Note:** When you tab into or click on a numeric value box, all text is selected. This means you may immediately type over the existing text, and if you hit delete the text box will be empty. If you want to move the cursor position, use the arrow keys. After typing you must tab, click on another text box or click on the Blue Window to apply the new value.

With Audition Tool in Blue Window:

- shift-drag - moves sliders (if any are connected), use to set parameter values without initiating playback.

With Draw Tool in Blue Window:

- shift-drag - moves sliders without writing any automation, use to set parameter values without initiating playback or to jump between settings accurately during automation recording.

With Geometric Tools in Blue Window:

- shift-drag - moves sliders (if any are connected)
- option-click - opens Automation Shape Parameters dialog after shape is created

With Automation Edit Tools in Blue Window:

- double-click when on over selected automation (finger cursor) - opens Automation Entry dialog
- shift-click-drag when over selected automation (finger cursor) - moves all selected points
- option-click when over selected automation (eraser cursor) - deletes that automation entry

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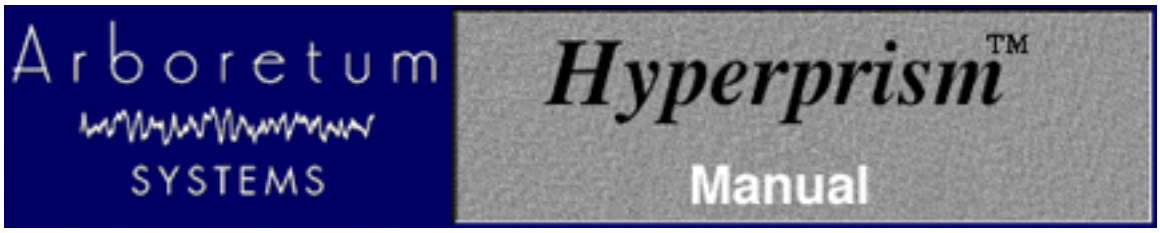


Hyperprism 2

Creating and Editing Blue Window Automation

In this chapter we'll dive into the real heart of Hyperprism, its gestural control functions associated with the Blue Window Play Zone. We'll learn how to configure the Blue Window to remember and recreate your parameter changes, and we'll

explore the new automation editing functions.

We're going to go step-by-step through many of the new features of Hyperprism 2. The material in this chapter assumes that you're already familiar with the basic HyperEngine functions as well as the elements of the Hyperprism Blue Window. Even if you're a long-time Hyperprism user this chapter will still be of interest, because it explains a lot of the new stuff.

Getting Started

Before creating new effects processes in Hyperprism, you'll need to first create a new HyperEngine Play-Thru Document (if you'll be processing/recording a live signal) or a new HyperEngine File Reference Document (if you're processing/editing a sound file already on disk).

Select the appropriate command (New Play-Thru Document or New File Document) from HyperEngine's File menu to start your new document.

If you're creating a new File Reference document you'll direct the program to your target audio file on disk using the File Selection dialog.

If you're creating a Play-Thru document, make sure your external audio source is connected to your Mac's sound input device.

Hit the HyperEngine Play button (or spacebar) to begin monitoring Play-Thru.

Once you've established that your sound is playing through properly and your monitoring levels are OK, you can proceed to the next steps in auditioning and automating parameter changes.

Mouse click on the Stop button (or hit the Return key) to stop sound playback.

HyperEngine will not give you access to the Plug-ins menu as long as sound is playing back; you always need to Stop playback in order to add effects to your signal chain.

Now select any Hyperprism effect from the HyperEngine Plug-ins menu. If the menu is ghosted and won't give you access, first make sure playback is stopped (not Paused) then click once in the waveform display. Either of these actions should activate the Plug-ins menu, whereupon you can proceed to select an effect.

The effect's Blue Window will appear on screen and the chosen effect will be added to your Reference Document's Process List.

Mapping Parameters to the Blue Window

Hyperprism allows you to connect any effect parameter to the Blue Window's horizontal (X) axis, or the vertical (Y) axis, so that you can control two or more parameters at once, with a single mouse movement. The Blue Window Axis Switches, which are located to the extreme left of each parameter slider, allow you to map the parameter to the horizontal (X) axis, or the vertical (Y) axis.

Blue Window Axis Switches, note the three possible states. Use the Axis Switches to map each parameter to the X or Y axes.

Click once on any Axis switch to map its parameter slider to the horizontal axis, click twice for vertical mapping, or a third time to disconnect the parameter from the Blue Window.

Connect any or all of the effect's sliders to the Blue Window Axes to engage their parameters for gestural control.

Note that sometimes you may not want to map a particular Parameter to the Blue Window. Just connect up the parameters which you want to have dynamic control over, and leave disconnected any parameters that are better left static and unchanging.

Altering the Parameter Range

Directly to the left and right of the parameter slider are its minimum and maximum value fields. These numeric fields allow you to define the range of control for each parameter. At the far right is the Current Value field, which reflects the slider's current setting.

Minimum value (highlighted)

Maximum value (highlighted)

Current value (highlighted)

Click in any value field to highlight the field for direct numeric entry of a new value from the keypad.

You can also click and drag from any value field to increase or decrease its setting by 1.0 increments.

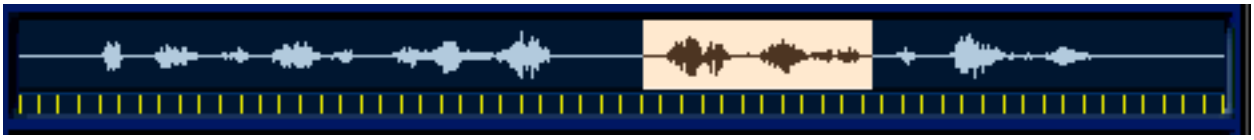
Holding down the Option key while mouse dragging increments/decrements the value fields by 100.0 units.

Hold down the Command and Option keys while mouse dragging increments/decrements the value fields by 0.01 units.

Hyperprism's factory default settings are generally configured for the widest range of control. But in many cases the most aesthetically pleasing Parameter settings may be within a narrow range of values. Rescaling the minimum/maximum Parameter values allows you to zero-in on any range so as to exercise a finer degree of control.

Selecting a Region

Hyperprism allows you to process your entire sound file, or any sections thereof. To help you manage this aspect we've included a special waveform display within the effect window, so that you can select portions of the file right within Hyperprism (freeing you from always bouncing out to HyperEngine every time you want to alter a selection).



Hyperprism waveform display. Current selection is highlighted in white, note existing automation ticks along bottom of display.

The waveform display will not appear in an effect assigned to a Play-Thru document, since HyperEngine has no way of pre-calculating a live signal's overview.

Create a New File Document (from the File menu) and point that document to a sound file on disk.

Select any Hyperprism effect from the HyperEngine Plug-ins menu.

The effect's Blue Window will open on screen.

Click and drag across any portion of the waveform display.

The selected range will highlight in white.

Click on the Play button or press the spacebar.

Playback will begin at the start of the new selection.

Click on the HyperEngine's Loop button, then hit Play or spacebar again.

Playback will repeat over and over, looping through just the selected region.

As you begin creating and editing parameter automation, bear in mind that automation will only occur within the time coordinates represented by the currently-selected region of audio. If there is no selection active, then Hyperprism will work with the entire length of the file by default.

Auditioning Settings

Before creating a parameter automation path, you'll probably want to play around with various settings, auditioning the results before recording automation. Using Audition mode will allow you to freely explore different settings with the Record Automation function turned off.

From this point on we'll assume you're working in a File Document, with a Hyperprism effect window open. But remember that all the following techniques can be used in HyperEngine Play-Thru documents on live signal as well as in File Documents with disk files.

Mouse click on the Audition button to enter Audition mode. The button will highlight and all other automation switches will be de-activated.

Hyperprism Audition switch

Hold down the shift key and move the mouse around the Play Zone.

Shift key allows you to adjust your parameter settings without starting playback.

Now release the Shift key, and mouse click and drag around the Blue Window Play Zone.

Audio playback will begin as soon as you start dragging across the Play Zone. Note the variance in your sound, as X/Y mouse movements alter the current values for each mapped effect Parameter.

Drawing Tracer Paths

Once you've gotten acquainted with Audition mode, and tried out a few parameter change gestures in the Play Zone, you'll be ready to start recording parameter change automation. Hyperprism 2 includes several new tools for creating parameter change tracer paths: The Draw tool, for freehand path creation, and the Line, Rectangle and Oval tools which assist in making geometric automation paths.

Make sure that the HyperEngine Loop button is switched off.

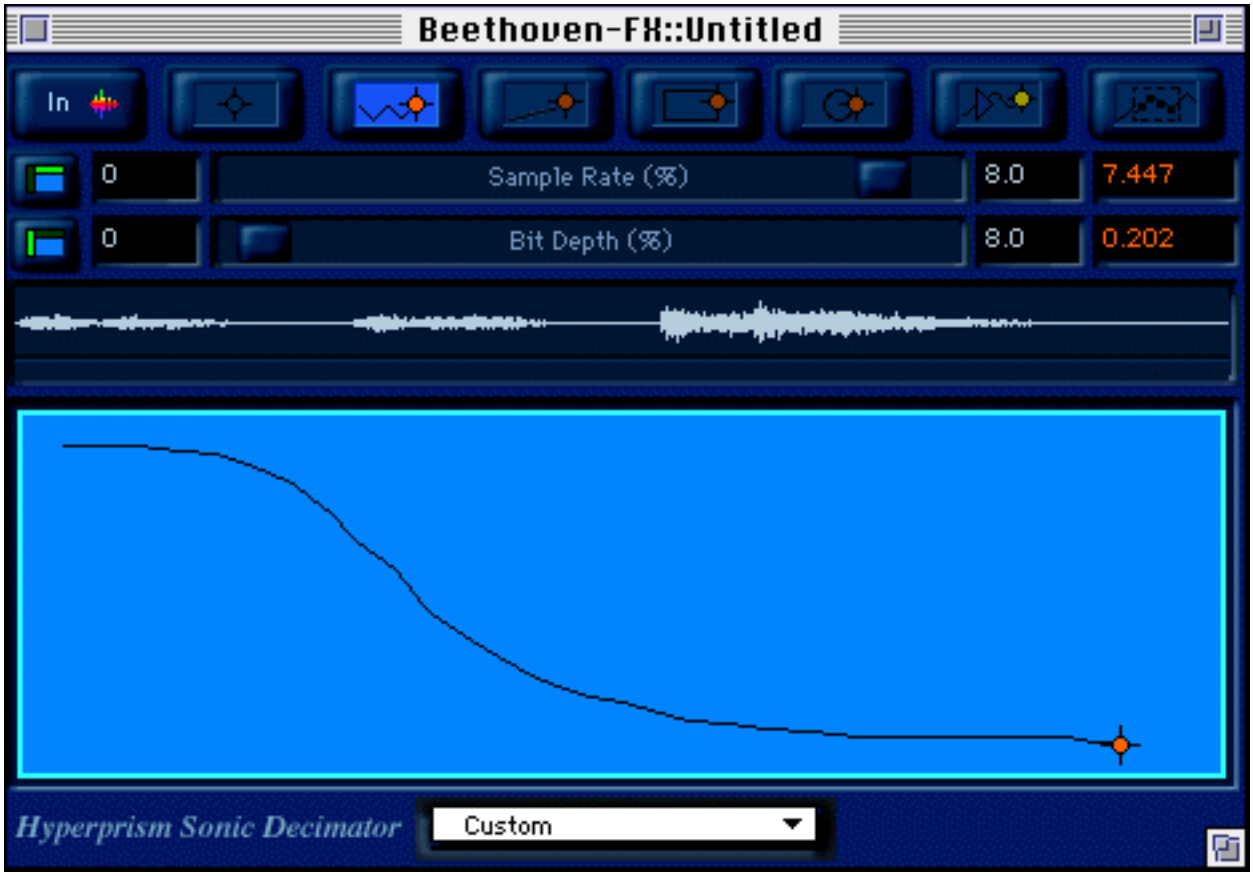
Mouse click on the Draw tool to enter freehand mode.

Hyperprism Draw tool

Hold down the shift key and move the mouse around the Play Zone.

Shift key allows you to adjust your parameter settings without starting playback.

Now release the Shift key and mouse click anywhere in the Play Zone, and with the mouse button held down, drag the cursor around to different locations within the window.



Drawing a parameter automation tracer path in the Blue Window Play Zone

Audio playback will begin as soon as you start dragging across the Play Zone. Note the variance in your sound, as X/Y mouse movements alter the current values for each mapped effect Parameter.

As you record automation you'll notice that the Current Value fields for parameters under mouse control have turned to red, indicating their "record" status.

When you reach the end of the file audio playback and automation recording will come to a stop. Or you can just press the Return key to end the pass.

When you finish recording automation, new yellow Automation Ticks will be displayed underneath the waveform display, representing the timeline position of each automation breakpoint. These Automation Ticks can be used as a breakpoint selection shortcut, and also provide a continuous visual reference for each automation event.

About The Tracer

When the mouse cursor appears over the blue "play zone," it changes into the Tracer icon. As you click the mouse button and drag, it takes control of your parameter settings and Hyperprism begins processing your sound with the current effect. As you drag around the play area, you'll hear the parameters of the effect change in relation to the Tracer's position in the Blue Window's Play Zone.

The tracer icon changes color, depending on the current automation state:

Tracer states: Record (orange), Audition (transparent) and Play (yellow)

Orange: (or black if you have a monochrome monitor) The Tracer turns orange when you are Drawing a path to record parameter automation moves.

Transparent: When in Audition mode, the tracer is transparent; only the cross hairs are visible. When in Draw mode, if the mouse is moved over the Play Zone but the mouse button is not pressed, the Tracer cross hairs will have an orange ring within them. You can move around without affecting the sound, positioning the mouse at a precise point before recording a path. Hold down the Shift key in Audition or Draw mode to adjust effect parameters without beginning playback and automation recording.

Yellow: (or white if you have a monochrome monitor) The Tracer is yellow when stopped and during playback.

Playing Back/Bypassing Automation

Now you'll want to hear the results of the automation you've just created.

Press the spacebar, or click on the Play button, to replay the selected audio with the new Parameter change automation.

You'll see the Blue Window Tracer follow the automation path, and notice the parameter sliders and current value fields updating to reflect their changing settings.

Current Value fields change to a yellow color when they're being updated by automation playback.

One of the new Blue Window tools is the Bypass Automation switch, which allows you to select whether automation will be played back, or suspended, while keeping the current tracer path on screen. This button is switched On automatically when using the Draw tool, and can be switched to Bypass any time you want to suspend tracer automation.

Bypass Automation switch

Mouse click on the Bypass Automation switch so it goes to its darker "Bypass" state.

Press the spacebar, or click on the Play button, to replay the selected audio with the Parameter change automation suspended.

You'll notice that the Play Zone and sliders stay frozen at their last setting.

Mouse click on the Bypass Automation switch again so it goes to its highlighted "On" state.

Press the spacebar, or click on the Play button, to replay the selected audio with the Parameter change automation re-engaged.

Tracer movement along the automation path resumes.

Drawing Automation in Loop mode

HyperEngine's Loop button causes the audio file (or current selection) to play back repeatedly, returning instantly to the beginning once the end is reached. Using the Loop mode is especially helpful when first auditioning and configuring an effect setting, saving you from constantly hitting the Play button or spacebar. You can also exploit the Loop mode when creating a path automation.

HyperEngine Loop button

Click on the HyperEngine Loop button to enter Loop playback mode.

Mouse-drag through a section of the Hyperprism waveform display to select a region of your audio file for processing.

Click on the Hyperprism Draw tool so that you can create a new automation tracer path.

Click and drag around the Blue Window's Play Zone, creating a freehand automation path, and keep drawing as the HyperEngine cycles through a few iterations of your looped audio selection.

After four or five times through the loop, let up on the mouse button to discontinue automation recording.

Notice that only the automation created during the last loop iteration is preserved.

Hyperprism automation events are placed on a linear timeline which corresponds to the length of the audio file in your File Reference Document, as represented by the Automation Ticks display beneath the waveform view. On each successive "record" automation pass the previously-recorded automation events are over-written by the new current pass information.

If you want to create a single long automation gesture that extends across multiple iterations of a loop, you'll need to copy-and-paste the loop in the HyperEngine Waveform Editor display, so that your multiple copies of the loop are laid end-to-end along the timeline. The Repeat Selection command really comes in handy for this purpose, reducing the cut-and-paste to a single selection and dialog entry.

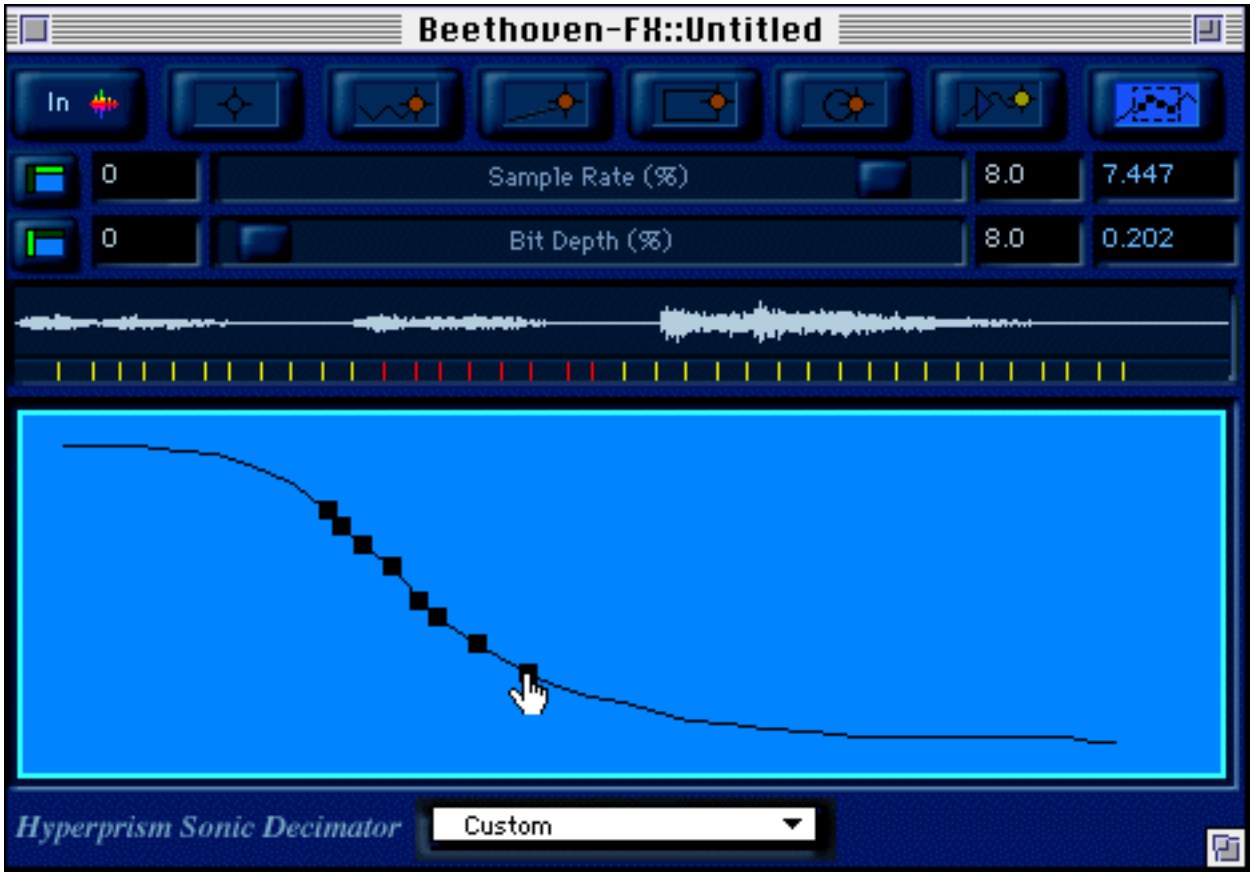
Editing Automation

Now that you've created and played back parameter change automation, lets learn how to edit the automation tracer path.

Mouse drag through the yellow Automation Ticks display (below the waveform view) to select a section of your automation for editing.

The selected Automation Ticks will highlight in red.

Notice that the corresponding automation path breakpoints are likewise highlighted in the Play Zone, showing up as black squares along the tracer path.



A section of path automation selected for editing. Note red highlights in the Ticks display, black highlighted breakpoints in the Play Zone, and the cursor changing to a hand tool when positioned above a selected breakpoint.

Note also that the Automation Edit button has toggled automatically to its active state, indicating that you're now in Edit mode. Other automation tools will toggle off.

Automation Edit switch

Position the cursor above any of the highlighted breakpoints in the Play Zone.

The mouse pointer turns into a hand when situated above a selected breakpoint.

Click on any breakpoint and mouse-drag it to a new location within the play zone.

Click on the Bypass Automation switch so that it is highlighted, indicating that automation playback is engaged.

Bypass Automation button, active state

Press the spacebar or click on the Play button to start playback

Your automation will now play, reflecting the new breakpoint position.

Note that in the preceding sequence of steps we've altered the grid location of an automation event, but we've kept its time location. The breakpoint's relative position along the timeline has stayed constant.

Now lets alter a breakpoint's position in time.

If you're not still in edit mode, click on the Automation Edit switch to begin editing the tracer path.

Click and drag through any section of the tracer path within the Blue Window Play Zone.

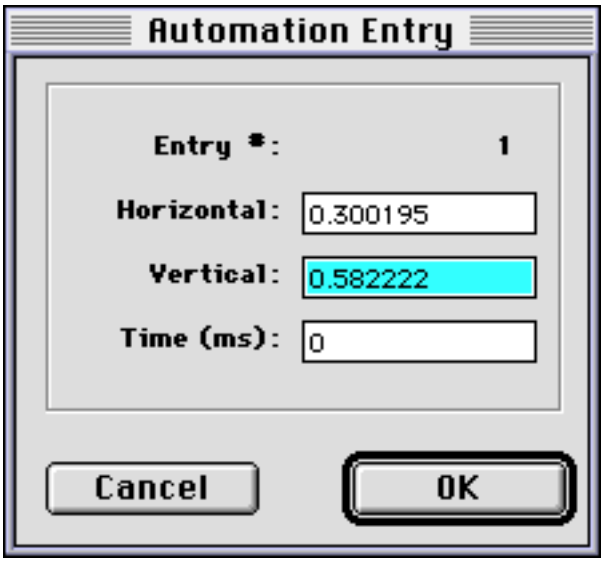
When selecting a range in Automation edit mode, you'll notice that the mouse cursor turns into a marquee selection tool, allowing you to select any group or area within the Play Zone for editing.

Let the mouse button up, your selected tracer path breakpoints will be highlighted in black.

Notice also that the Automation Ticks display highlights in red, showing the timeline position of each selected breakpoint.

In the Play Zone, double-click on any highlighted breakpoint.

The Automation Entry dialog will appear:



Hyperprism 2 Automation Entry dialog

This dialog allows you to specify any time and grid coordinate for the selected breakpoint.

The Horizontal and Vertical fields accept entries between 0 and 1, where 0/0 represents the lower left hand corner and 1/1 represents the upper right hand corner of the Blue Window Play Zone. This is a tweaky way to enter geometric coordinates, compared to simply moving the points around, but it's there if you need it. Note that grid coordinates can be applied to many different parameters, each with its own measurement units and unique value range.

The Time field is what we'll focus on here. The breakpoint timeline position is shown in milliseconds, as calculated from the beginning of the sound file.

Double-click in the Time field to highlight it for numeric entry.

Enter a new time value for the breakpoint, in milliseconds (1/1,000th of a second).

Click on the OK button to exit the dialog and apply the new coordinates.

If you've specified a time value for the breakpoint that takes it out of its current path order, you'll notice the tracer path will be redrawn to reflect the new time position of this breakpoint.

Deleting Automation

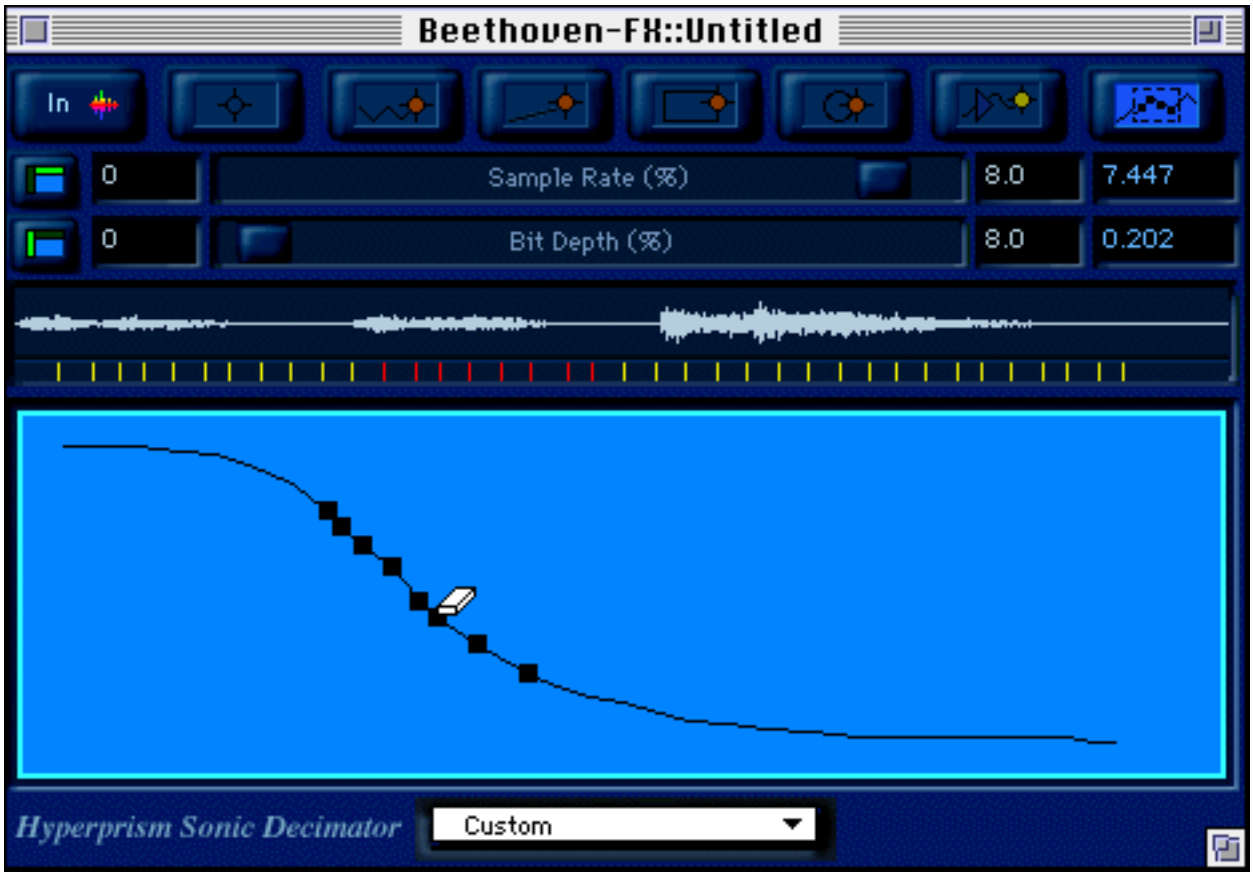
While we're in Edit mode, let's delete a few breakpoints. There's several methods:

Mouse drag through the Automation Ticks display to select a group of breakpoints for editing.

The selected area will highlight in red, and the breakpoints will show up as black spots on the Play Zone.

Hold down the Option (alt) key while moving the mouse cursor down to the Play Zone.

As you keep the Option key pressed, you'll notice that the mouse pointer changes into an Eraser tool when it is positioned above a selected breakpoint.



Automation breakpoint deletion (holding down Option key)

Mouse click with the eraser on any breakpoint to remove it from the tracer path.

Using this Option key trick is crucial if you're trying to edit out discrete points. Next, we'll try a more wholesale approach:

While still in Automation Edit mode, click and drag across any section of the automation tracer path to select a

group of breakpoints.

Selected breakpoints will be highlighted in black; their corresponding Automation Ticks will highlight in red.

Now press the Delete or Backspace key on your keypad.

All selected breakpoints will be removed from the Automation path. The path will be redrawn to reflect its new configuration.

Finally, use the keyboard shortcut command-a to Select All breakpoints.

All the remaining breakpoints along the tracer path will be highlighted.

Press the Delete or Backspace key on your keypad.

Your entire tracer automation path will be erased.

Now select the Undo command from HyperEngine's Edit menu, or use the keyboard shortcut command-z.

Your deleted path will be restored.

Remember that the HyperEngine Undo/Redo commands, found in the Edit menu, are applicable to any edit or change you make to Hyperprism Parameter automation. This allows you to do A/B comparisons of any edit or change; remember to use the command-z shortcut for speed and ease of use.

Inserting Automation

Now that you're comfortable with creating and editing automation tracer paths, lets learn how to do some insert editing. First we'll create an automation gesture, then we'll insert some new movements into a selected part of the automation path.

Click once anywhere along the waveform display to remove any selection.

Any highlighted ranges of your sound file will be de-selected and the waveform view will be returned to its normal blue color.

Click on the Draw tool to engage automation recording.

Mouse drag through the Play Zone to begin audio playback, and continue dragging around to create an automation gesture which lasts the entire duration of your sound file.

When you let up on the mouse Hyperprism will stop recording automation events.

Press the Return key, or click on the Stop button to cease audio playback.

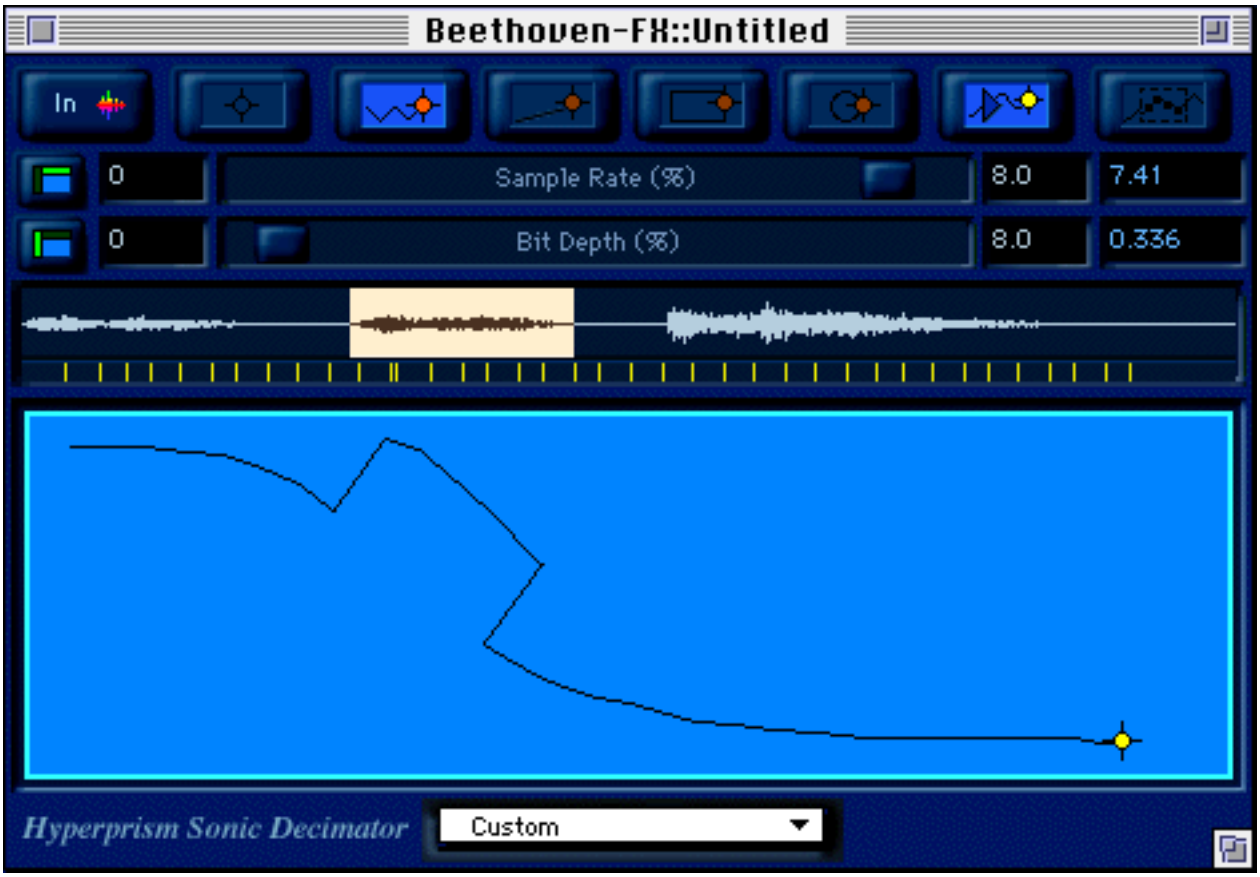
Next, we're going to select a section of the sound file, and insert new automation events in that particular range.

Mouse drag through a portion of the Hyperprism waveform display to select a range within your audio file.

If the Draw tool is no longer active, click on it again to engage automation recording.

Mouse drag through the Play Zone to initiate playback of your selected range, and continue dragging to create an automation gesture for your selected audio.

Once you've stopped, your display should be similar to the one below, showing your initial automation path, which deviates to the new inserted automation events, then returns to the original path beyond the selection range.



The result of an automation insertion (note new path in selected range)

This technique can be utilized any time you want to re-do a certain part of an automation path, without re-recording the entire gesture. Using the waveform selection and the insert automation methodology in a sequential fashion also allows you to build up a series of smaller automation events along a much larger sound file, saving you from having to perform the entire automation movement in a single pass.

Creating Geometric Paths

Hyperprism 2 includes a number of new geometric drawing aids. These are the Line tool, the Rectangle Tool and the Oval Tool. These tools by default create 20 Automation Ticks (breakpoints) per second of selected audio.

Hyperprism 2 Line, Rectangle and Oval tools

To Create a linear automation path:

Mouse click on the Line tool to activate line drawing.

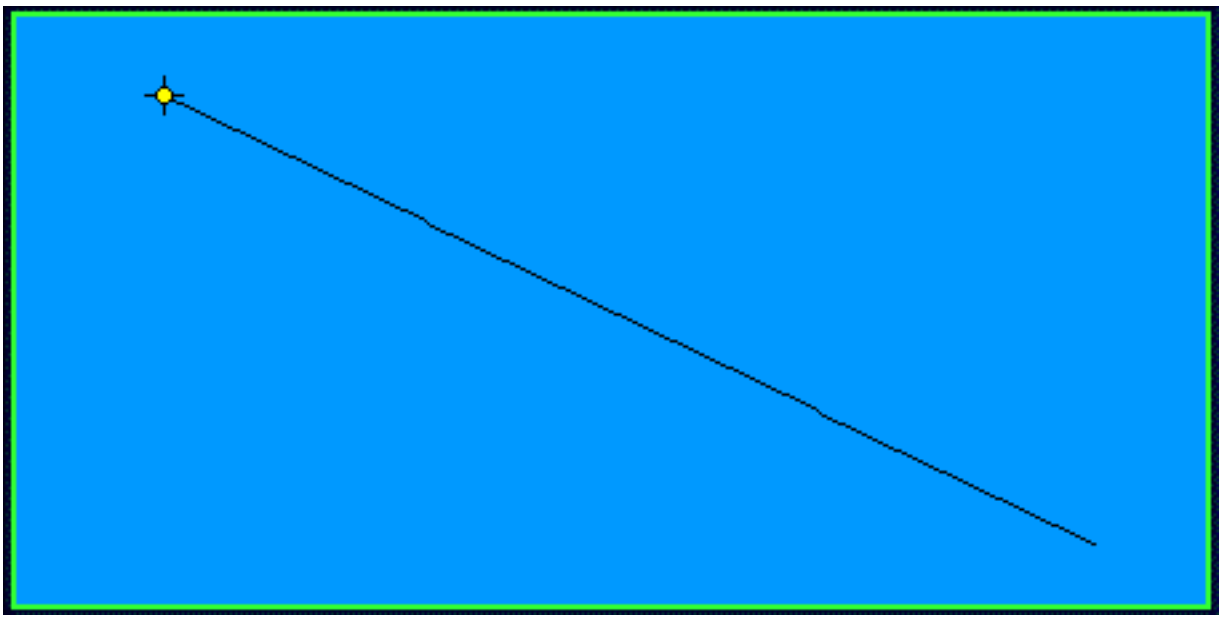
Click the mouse pointer anywhere in the Blue Window Play Zone.

With the mouse button held down, position the end point of the line you're creating.

Notice that the line follows your mouse movements as you position the end point.

Release the mouse button to set the line path.

Your automation should look something like this:



Linear automation path created with the Line Tool

Now press the spacebar or click on the Play Button.

Your new automation movement will be played, and you'll see the cursor go from beginning to end point, and hear the associated parameter changes.

A geometric automation path, by default, will last the length of the entire file (or current selection, if any) and will go through its course one time. We'll learn how to alter those defaults in just a moment.

Next, lets create oval and rectangular automation paths:

Mouse click on the Rectangle tool to activate line drawing.

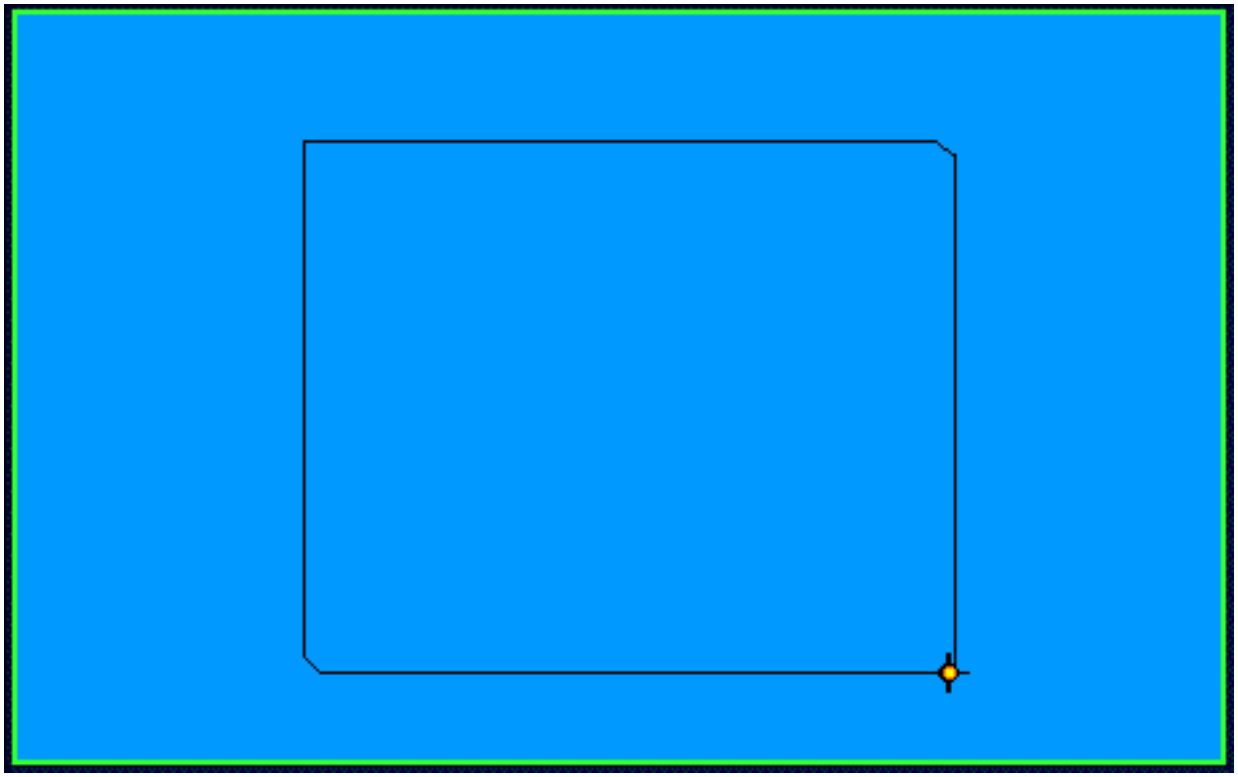
Click the mouse pointer anywhere in the Blue Window Play Zone.

With the mouse button held down, position the boundary of the rectangle you're creating.

Notice that the line follows your mouse movements as you position the boundary.

Release the mouse button to set the rectangle path.

Your automation should look something like this:



Geometric path created with the Rectangle tool

Now press the spacebar or click on the Play Button.

Your new automation movement will be played, and you'll see the cursor travel the course of the rectangular path, and hear the associated parameter changes.

A geometric automation path, by default, will last the length of the entire file (or current selection, if any) and will go through its course one time. We'll learn how to alter those defaults in just a moment.

Now mouse click on the Oval tool to activate line drawing.

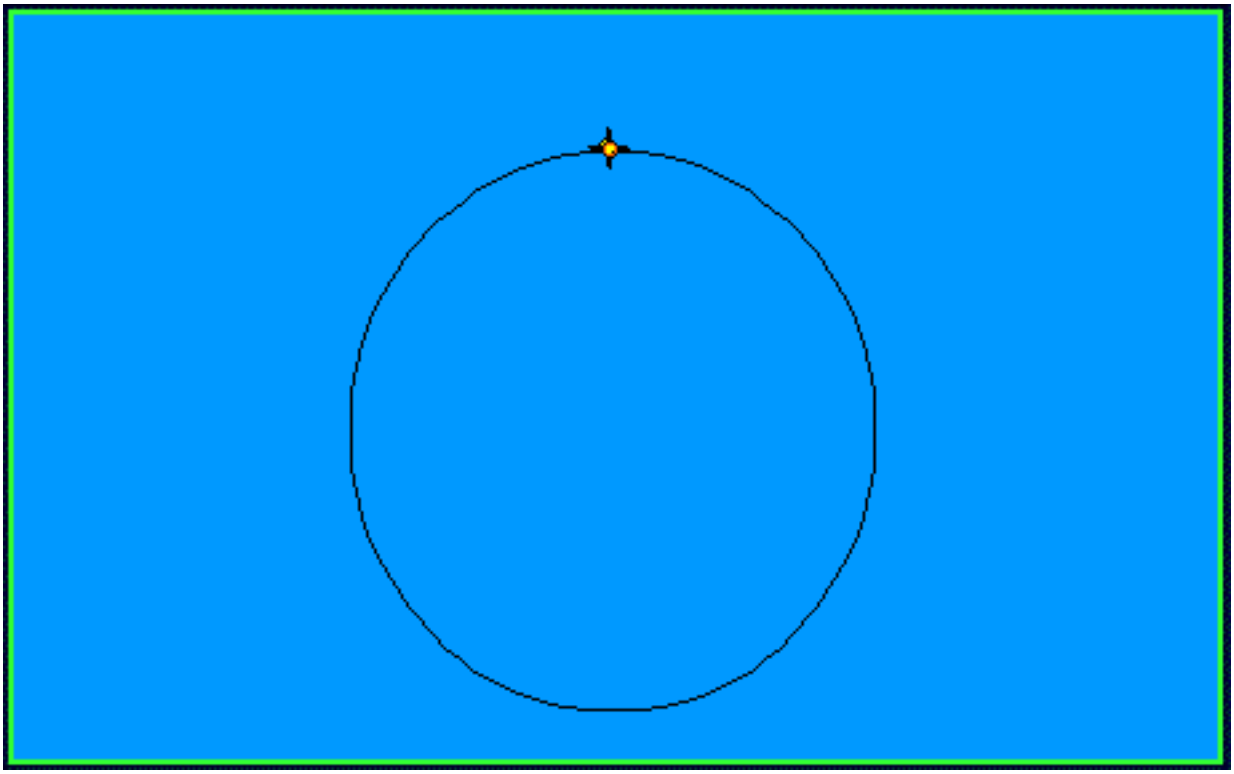
Click the mouse pointer anywhere in the Blue Window Play Zone.

With the mouse button held down, position the boundary of the oval you're creating.

Notice that the line follows your mouse movements as you position the boundary.

Release the mouse button to set the oval path.

Your automation should look something like this:



Geometric path created with the Oval tool

Now press the spacebar or click on the Play Button.

Your new automation movement will be played, and you'll see the cursor travel the course of the oval path, and hear the associated parameter changes.

A geometric automation path, by default, will last the length of the entire file (or current selection, if any) and will go through its course one time. We'll learn how to alter those defaults next, using the Automation Shape Parameters.

Automation Shape Parameters

The Automation Shape Parameters dialog allows you to control the behavior of Blue Window geometric Parameter automations. This dialog is accessed when drawing a geometric path, through a simple key command.

To create a geometric automation with customized Shape Parameters:

Mouse click on the Rectangle tool to activate line drawing.

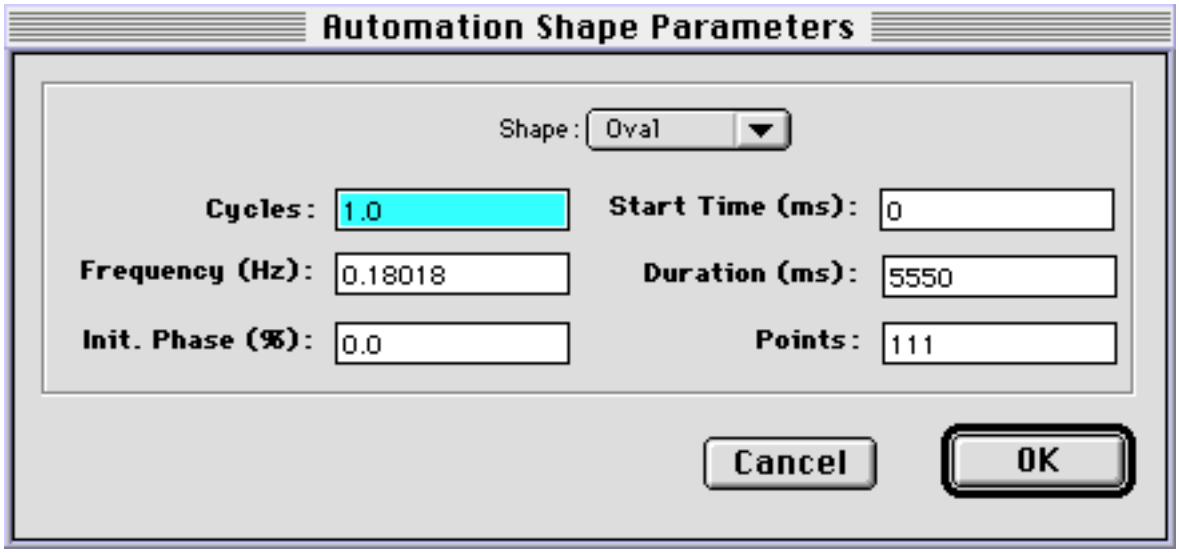
Click the mouse pointer anywhere in the Blue Window Play Zone.

Now press the Option (alt) key on your keypad, and while holding both the Option key and the mouse button down, position the boundary of the rectangle you're creating.

Notice that the line follows your mouse movements as you position the boundary.

Release the mouse button and Option key to set the rectangle path.

You'll be presented with the Automation Shape Parameters dialog:



Automation Shape Parameters dialog is accessed by holding down the Option key while drawing a geometric automation.

The Automation Shape Parameters dialog shows you the following fields:

Shape

Line, Rectangle or Oval. This is set according to the drawing tool you're using, and is not adjustable in this dialog.

Cycles

The number of times the tracer will cycle through the path during the course of the file or selection. This parameter is linked to the Frequency Parameter. Cycles defaults to 1.0 on Oval and Rectangle paths, and to 0.5 on Line paths. Negative values result in counter-clockwise motion, positive values result in clockwise motion.

Frequency

The number of times per second that the tracer will cycle through the path. This Parameter is linked to the Cycles Parameters.

Initial Phase

This Parameter sets the starting point for the tracer, allowing you to position it anywhere along the path. 0% equals beginning, 99% equals end. Defaults to 0.

Start Time

Specify the time coordinate at which the automation movement will begin. This Parameter defaults to 0, meaning the start of the file or selection.

Duration

Specify the length of the automation movement. By default this value equals the end time of the file or current selection.

Points

The number of automation breakpoints created in the geometric automation path. The program will auto-calculate a default value based on the length of the selection. Geometric Draw tools by default create 20 Automation Ticks (breakpoints) per second of selected audio. Use this field to create any number of breakpoints along a geometric automation path.

Next, lets explore some of the interesting tracer patterns which can be created by manipulating the Cycles setting.

Mouse click on the Rectangle tool to activate line drawing.

Click the mouse pointer anywhere in the Blue Window Play Zone.

Now press the Option (alt) key on your keypad, and while holding both the Option key and the mouse button down, position the boundary of the rectangle you're creating.

Release the mouse button and Option key to set the rectangle path.

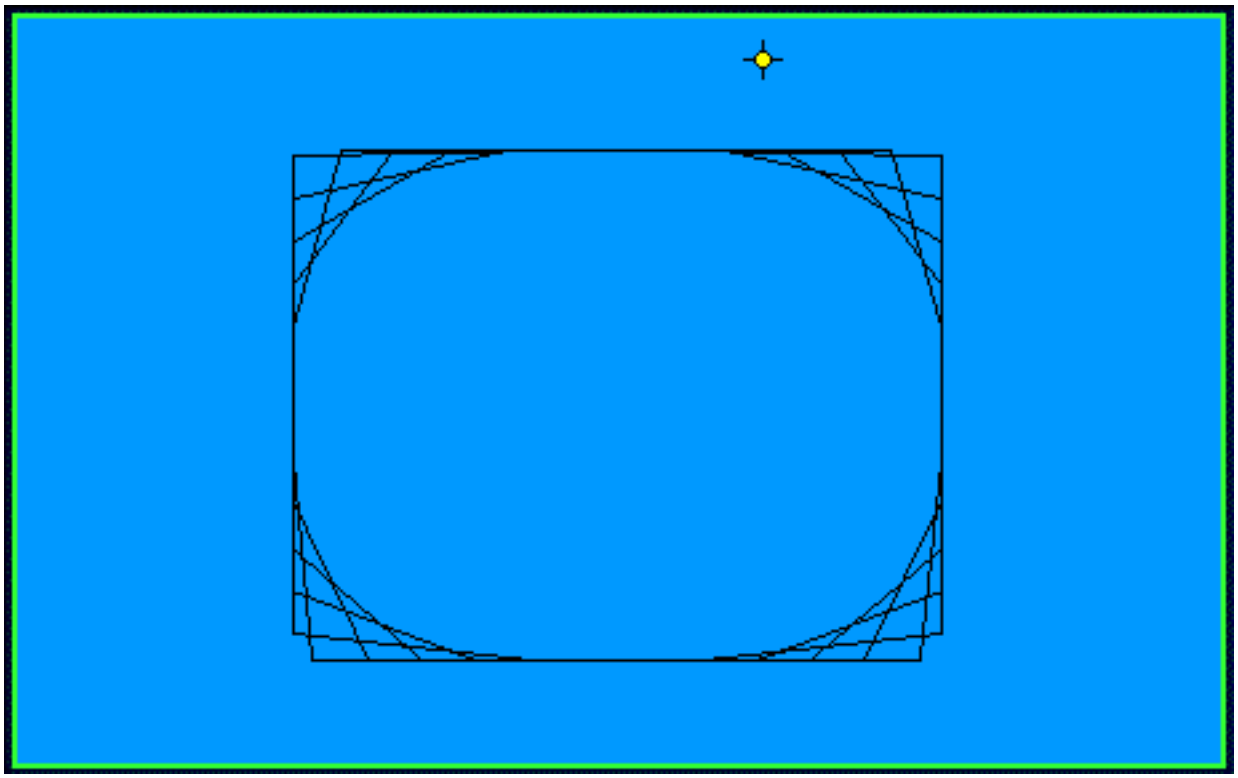
You'll be presented with the Automation Shape Parameters dialog.

Change the value in the Cycles field from 1 to 10. You can do this by mouse dragging through the field to scroll to a new value, or simply click in the box, backspace over the earlier values and type in the new value.

Notice that the Frequency setting automatically re-adjusts to follow your new Cycles value.

Press the Return key or click on the OK button to exit the dialog.

Your new tracer path should look something like this:



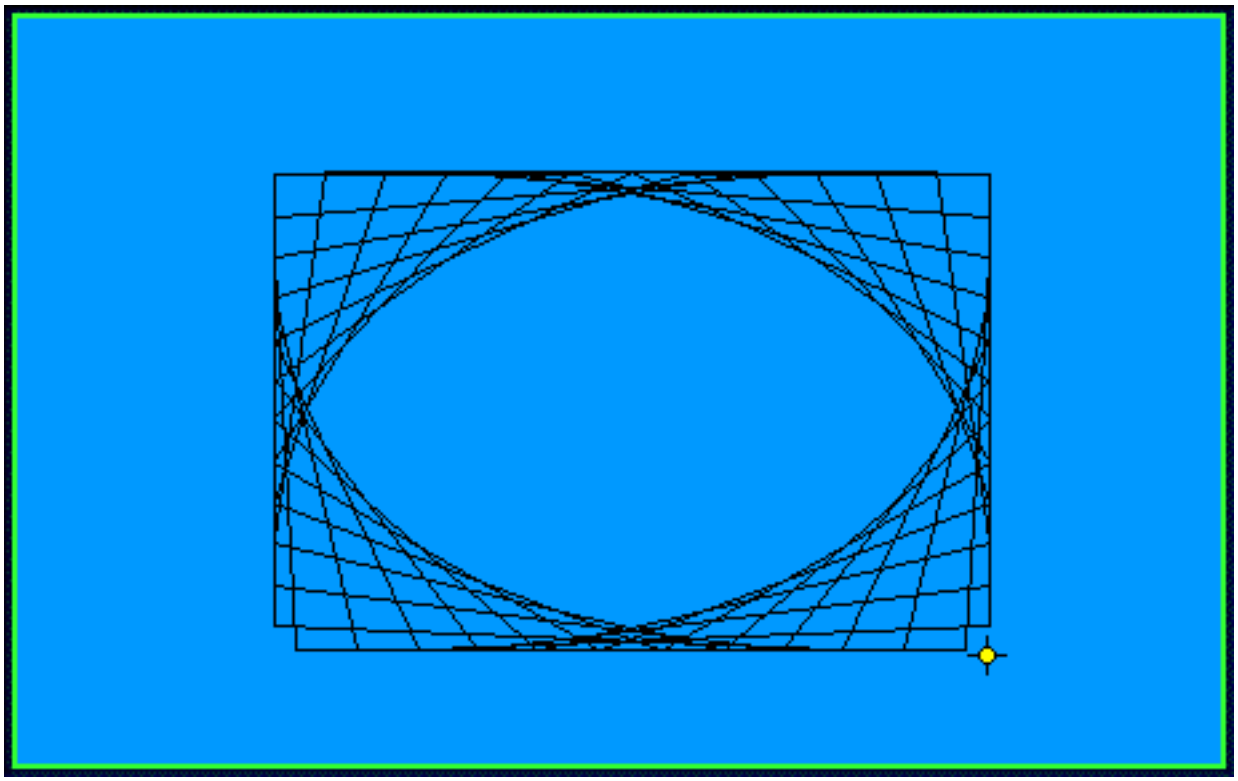
Rectangle automation path with a ten Cycles setting

Since breakpoints can't be re-used Hyperprism takes the number of available automation points (as determined by the Points setting) and divides them among the number of cycles when creating a multi-cycle automation path. Break point placement is limited to the boundaries of the geometric path you've drawn, but as you increase the number of cycles while holding the point number constant Hyperprism will start to run out of points to define the path shape, resulting in the "shortcuts" or "rounding" you're seeing here.

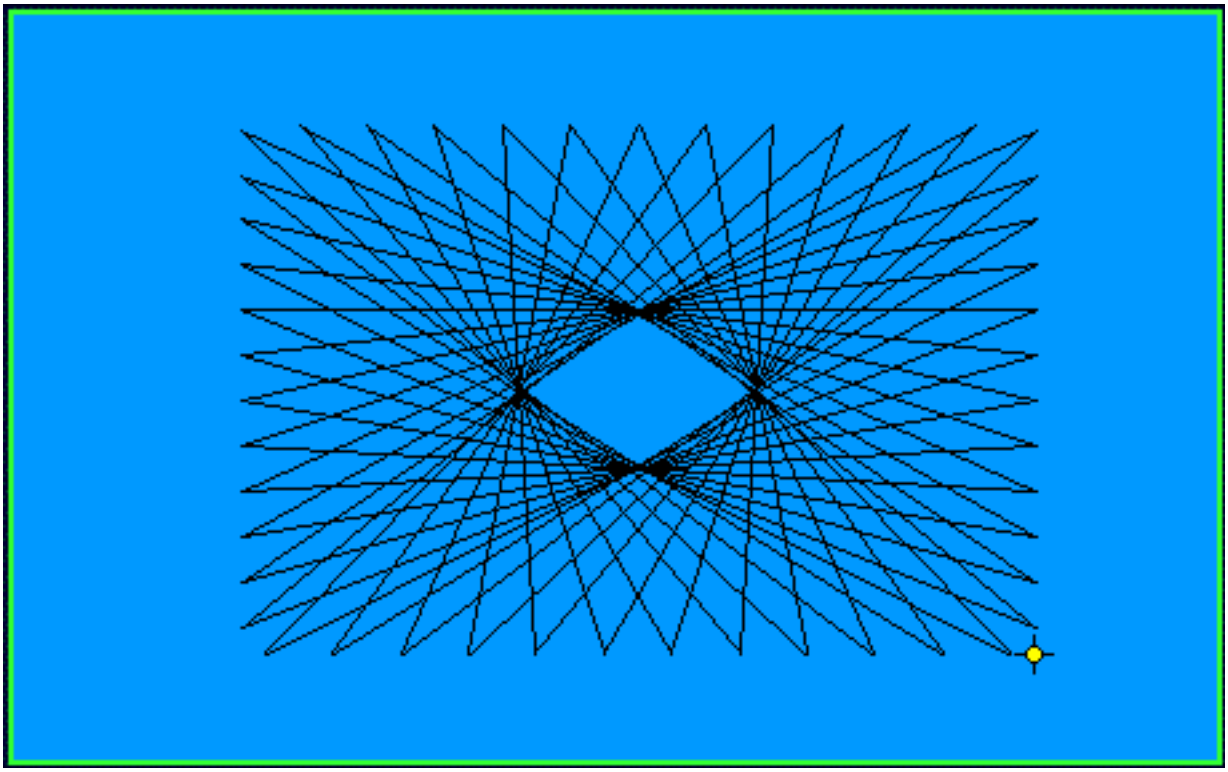
By controlling the number of Points and the number of Cycles you can create numerous basic or complicated parameter automation tracer paths. For example:

- Using the Oval tool to create an oval that has three points (and one Cycle) will result in a triangle automation path.
- An oval with five points and one Cycle will be a pentagon.
- An oval with five points that goes for two Cycles will be a five pointed star.

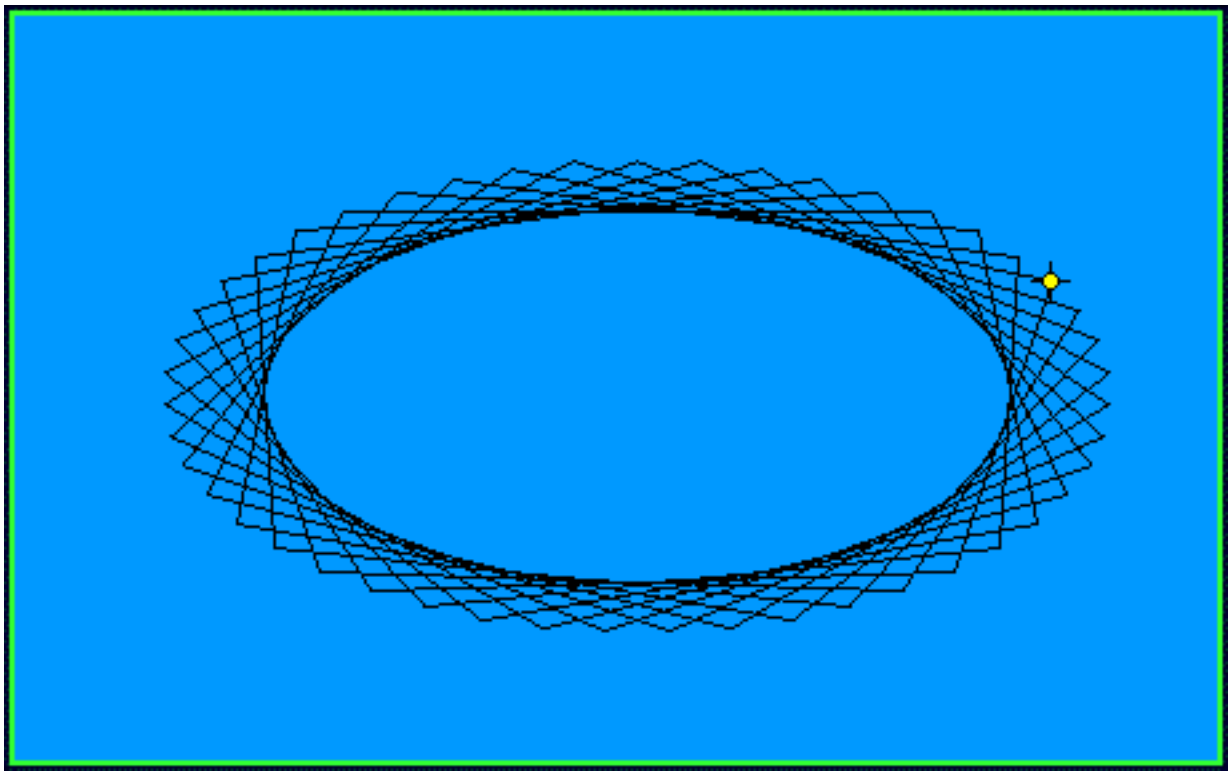
Now lets look a the results of even higher Cycles settings. Again, hold down the option key while drawing a Geometric shape, then enter a new Cycles setting while holding the number of Points constant (in these examples, 90 points) to get results like we see below:



Rectangle automation path with a twenty Cycles setting



Rectangle automation path with a forty Cycles setting



Oval automation path with a twenty cycles setting

You'll also notice as you play these multi-cycle automations that they can result in extremely fast Parameter changes.

Inserting Geometric Paths

Now that you're comfortable with creating geometric automation tracer paths, lets do some insert editing with them. First

we'll create an automation gesture, then we'll insert some new geometric movements into a selected part of the automation path.

Click once anywhere along the waveform display to remove any selection.

Any highlighted ranges of your sound file will be de-selected and the waveform view will be returned to its normal blue color.

Click on the Draw tool to engage automation recording.

Mouse drag through the Play Zone to begin audio playback, and continue dragging around to create an automation gesture which lasts the entire duration of your sound file.

When you let up on the mouse Hyperprism will stop recording automation events.

Press the Return key, or click on the Stop button to cease audio playback.

Next, we're going to select a section of the sound file, and insert new automation events in that particular range.

Mouse drag through a portion of the Hyperprism waveform display to select a range within your audio file.

Click on the Oval tool to engage geometric automation drawing.

Click and mouse drag though the Blue Window play zone to create your new circular automation gesture

Once you've stopped, your display should be similar to the one below, showing your initial automation path, which deviates to the new inserted automation events, then returns to the original path beyond the selection range.

Note Oval automation movement inserted in selected range

Copying, Pasting and Cutting Paths

Paths can be Cut, Copied and Pasted between the Blue Window Play Zones of any Hyperprism effects. Check out the Edit menu, these are the same commands you're seen in nearly every other Macintosh application. Using the keyboard shortcuts--command-X for Cut, command-C for Copy and command-V for Copy--will save you lots of time.

When attempting to Paste a path, make sure the active area of the Blue Window is highlighted, not one of the numeric fields along the axes of the window.

Use the Tab key to move the highlight between the numeric fields and the Blue Window

Deleting a Path

To delete a path drawn in the Blue Window, first type the keyboard shortcut command-a (for Select All) then select Clear in the Edit menu or press Backspace or Delete on the keyboard.

Existing paths will also be deleted the instant you begin recording a new path.

You can also delete a path (and its effect) directly from the Reference Document's Process List , by selecting its corresponding list entry and pressing Delete. If the effect's Blue Window is open, it will disappear.

Parameter Resolution Trade-offs

The screen of your Macintosh has fundamental limits in the number of pixels it displays, which limits the resolution of the values you can specify using a mouse. To control effects with maximum resolution, expand the size of the Blue Window to the largest size your monitor permits (by dragging the resize tool at the Blue Window's lower right corner). If this doesn't provide fine enough control, adjust the parameter min/max ranges.

For example, your play zone is 1000 pixels wide and the chosen effect is a filter with a frequency range from 20 to 20,000 Hz, notice that you have only 1000 pixels of addressing into a space that is 19980 values wide. This means that if you nudge the tracer by one pixel, the value jumps by 19 Hz. To increase the resolution of the tracer you can either increase the size of the play zone (large screens help here) or reduce the range of the parameter limits. If you reduce the frequency bandwidth of the above window to 1000 Hz, for example, you attain approximately 1 Hz resolution per pixel on the screen.

Working With Multiple Paths

As mentioned earlier, each Blue Window and its Tracer path is stored as a separate variation of the effect applied to the input sound file. These variations are listed in the Reference Document's Process List. Closing a Blue Window does not delete its entry from the Process List. All you have to do to bring it into view again is:

- Double-click on the effect's entry in the Reference Document's Process List.

↘ The Blue Window containing the variation reappears.

Removing Effects

If you are not pleased with the effect you've created, you don't have to keep it. To remove an effect:

- Click on the effect's entry in the Reference Document's Process List and press the Delete key.

The entry in the document reference window disappears, as does the Blue Window associated with that variation.

Note that there is no "Undo" when removing effects from the Process List. Once you've deleted an effect from the Reference Document's Process List, it's gone.

Saving Effects and Automations

If you like the parameter automation path you've created, you can save it as part of your HyperEngine Reference Document.

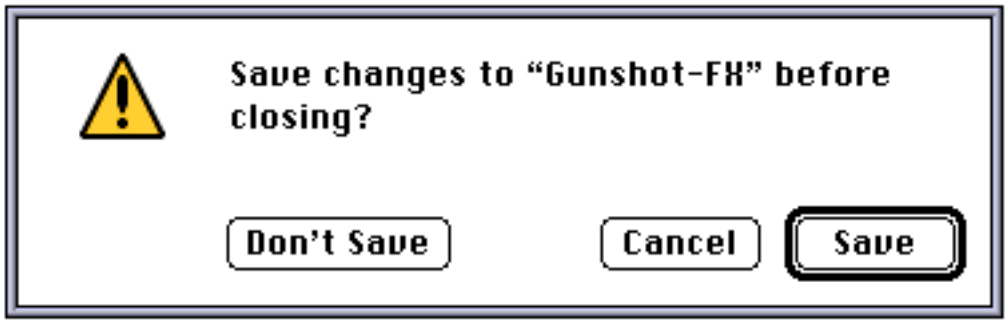
Click on the Close button on the Reference Document window, and the system will ask you whether you want to save the document or not, or...

Select Save in the File menu.

Saving a Hyperprism 2 file saves only the effect and automation parameters. This is generally a very small disk file, and the original sound file is left unchanged. Note that you must keep the original audio file on your disk to be able to hear the effects in your Hyperprism document.

Reference Document icon

If you Quit the application without having Saved a reference document containing changes, a save option will be automatically suggested before the program quits.



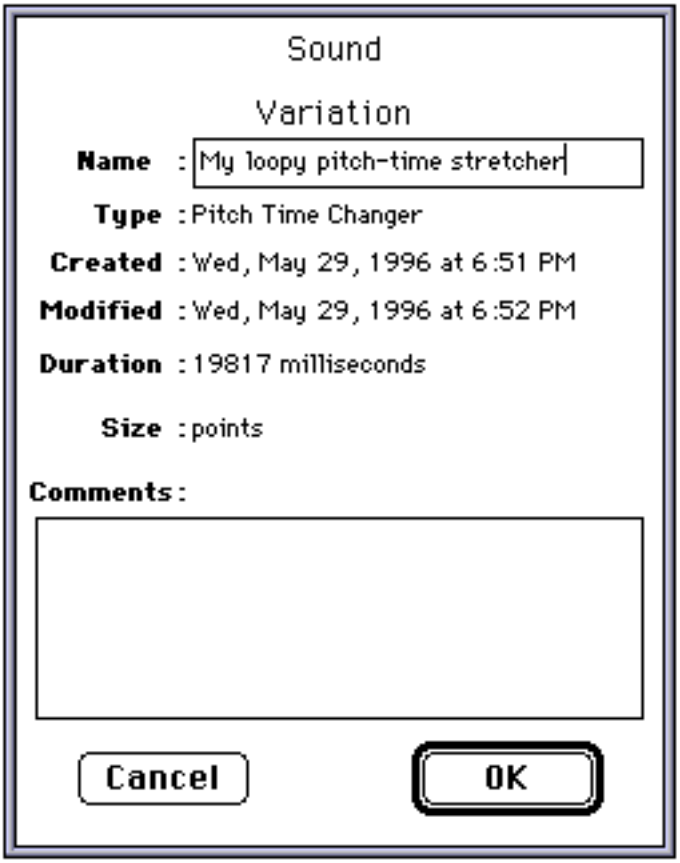
If you have un-saved changes, HyperEngine will suggest saving them before closing the document

Naming Your Effect Variations

Renaming each effect variation is a good way to keep track of the differences between effects with different Parameter configurations and automation paths. To change the process name, you must have the corresponding Blue Window open in front or its line in the document reference window selected.

Select Get Info from the File Menu or type command-I.

A new window appears, describing the variation. Type the new name.



The Get Info box for an effect variation.

You can also include comments. These comments pertain to a specific variation, not to the entire file, so when you look in the Get Info box for the entire file, you won't see these comments. They appear only in the Get Info box of a particular variation.

When you are done typing comments, click OK.

The Get Info window disappears and the Reference Document Process List will display the new effect name

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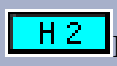
[Previous Chapter] [Table of Contents] [Next Chapter]

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.



Effect included in Hyperprism-2

Effect included in Hyperprism-VST (For Cubase-VST plug-in format)

Effect included in Hyperprism-MMP (Premiere plug-in format)

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
- Doppler
- Stereo Dynamics
- More Stereo
- M-S Matrix

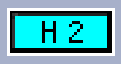
Hyperprism Miscellaneous Processes

- Pitch Time Changer
- Pitch Changer
- NEW! Vari-Speed
- Noise Gate
- Compressor
- NEW! Limiter
- Sonic Decimator

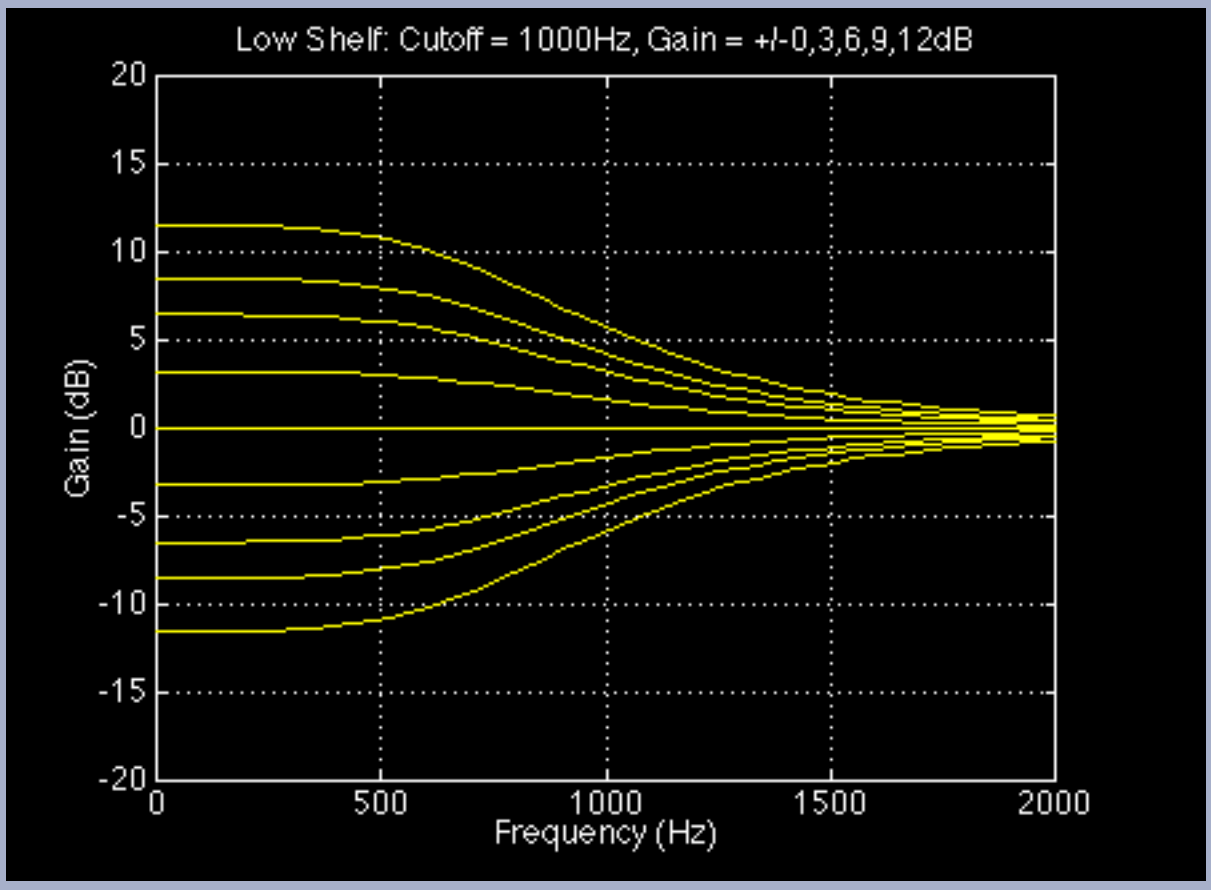
Go to Effects: [Filters] [Modulators] [Delay and Reverb] [Stereo] [Misc Effects]
Go to Table of Contents: [Hyperprism 2] [Appendices]

Hyperprism Filter Processes

Low Shelf



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.

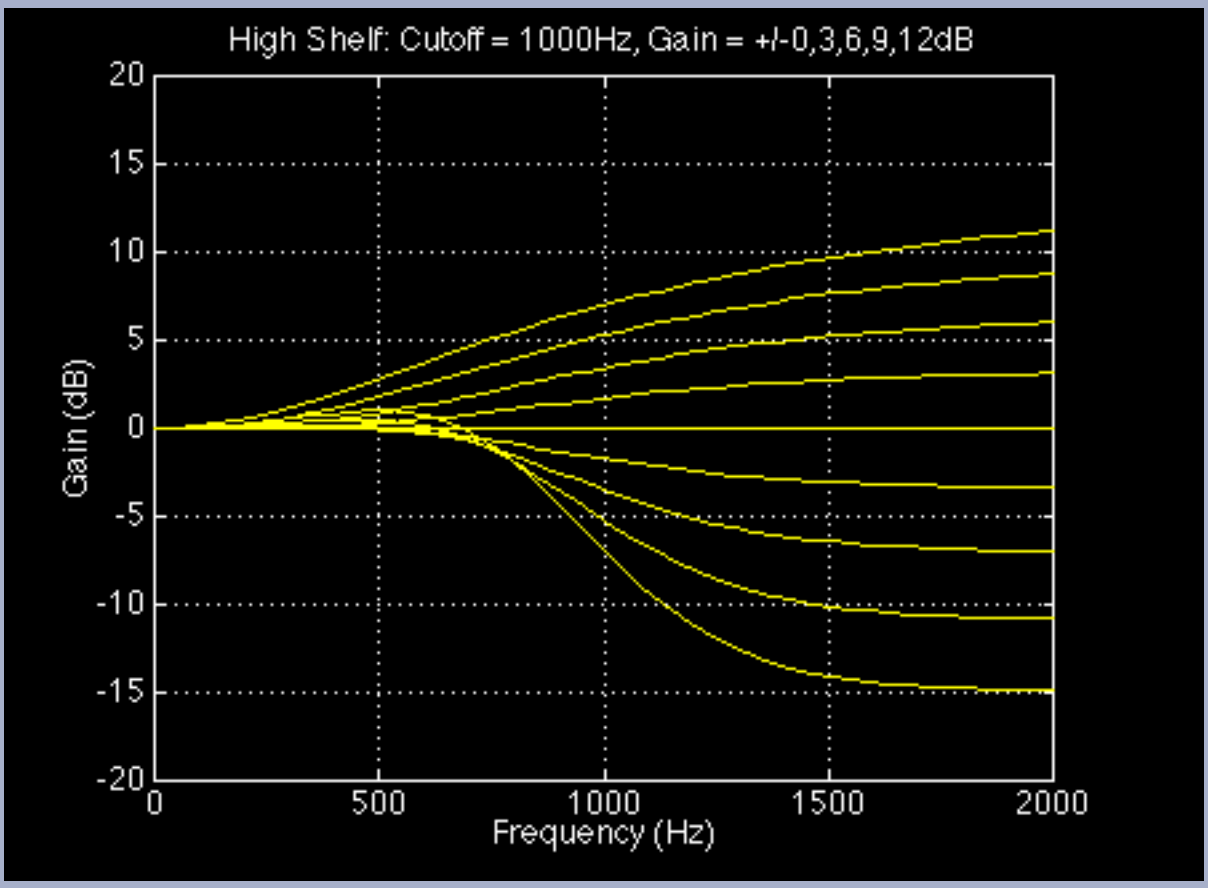
Gain

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

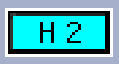
Cutoff

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.

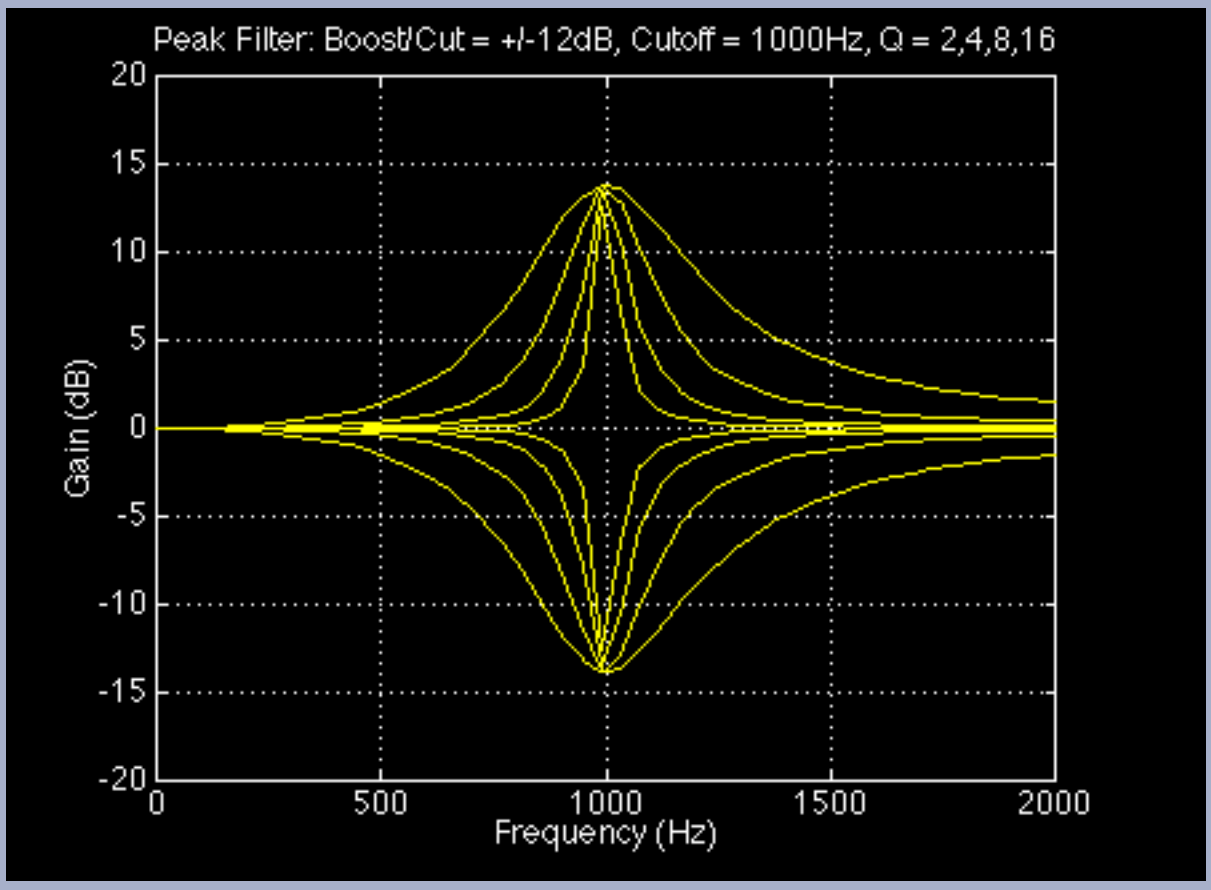
Gain

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

Parametric EQ



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 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.

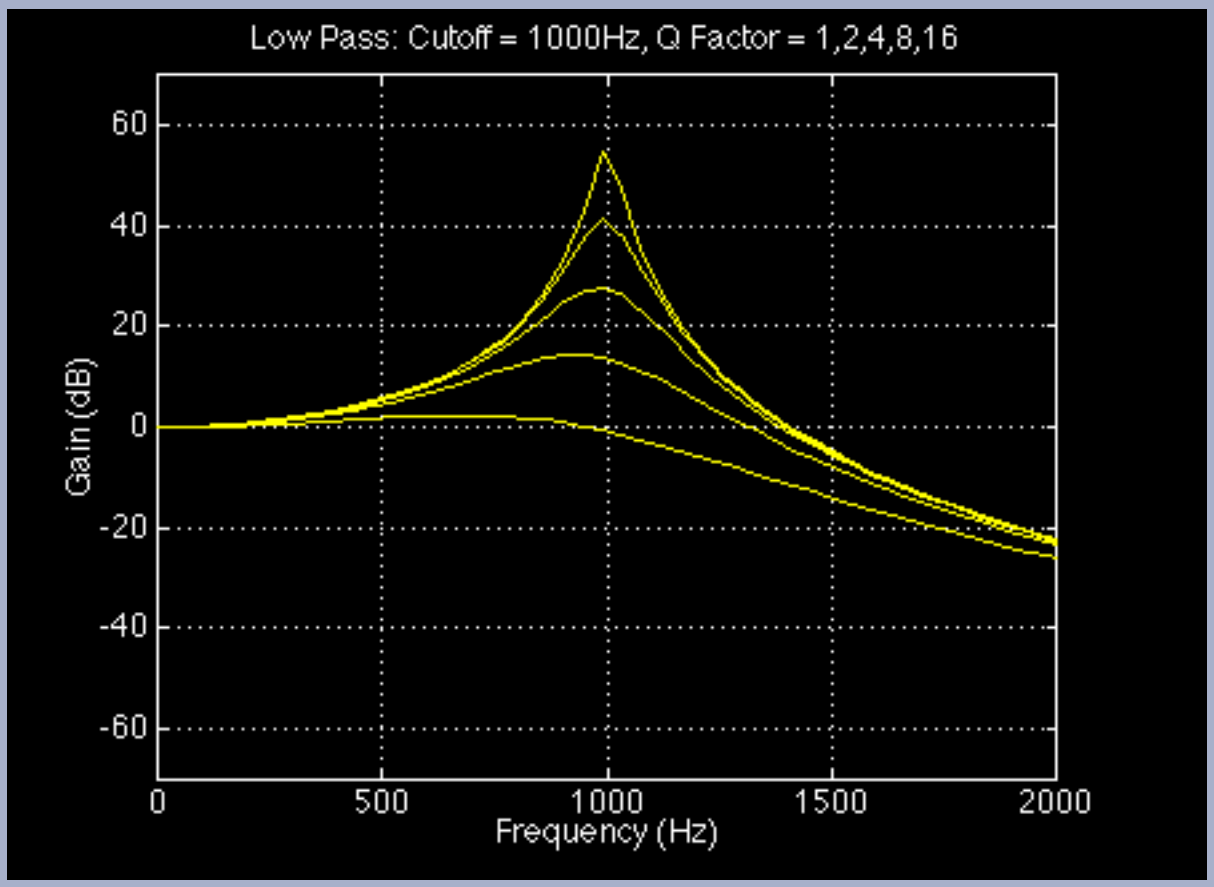
Parameters

Frequency Scale

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.

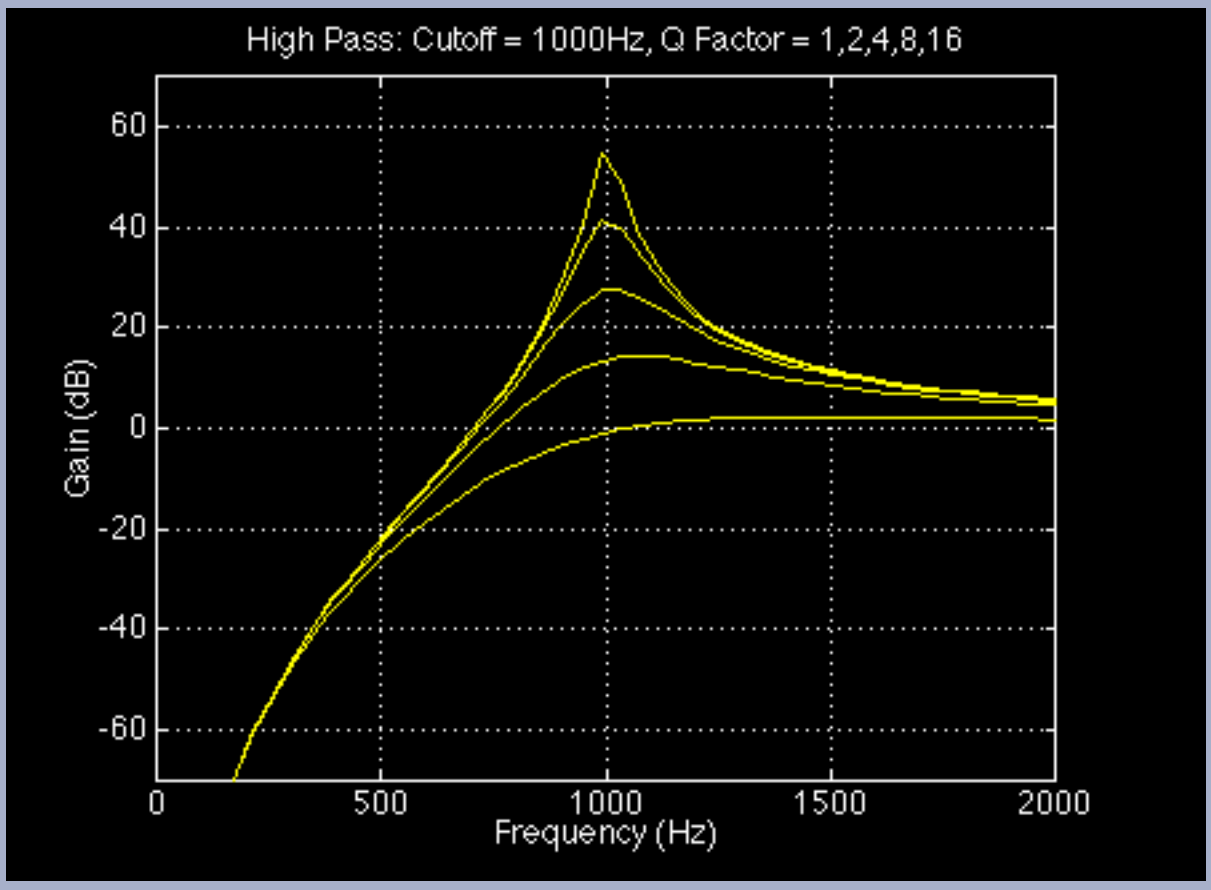
Parameters

Frequency Scale

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

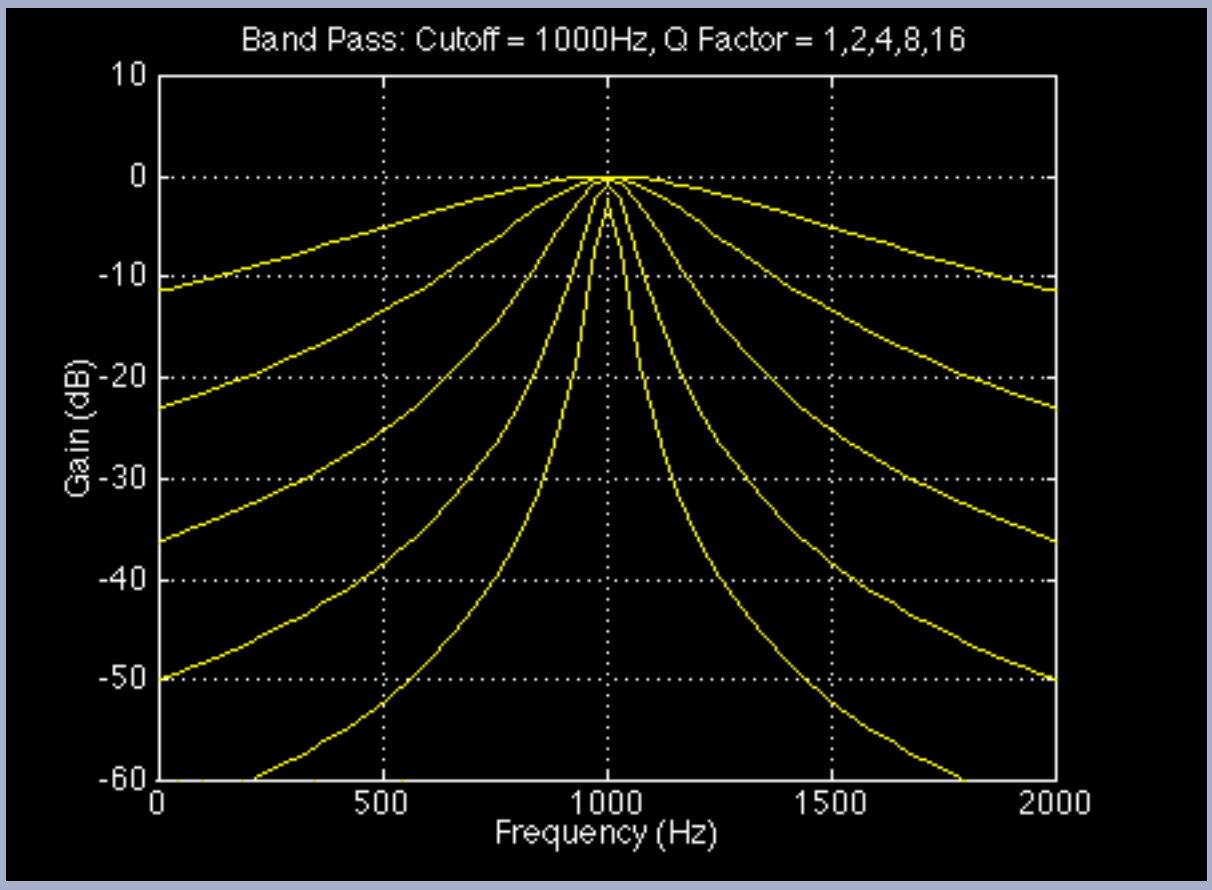
Also called "Frequency" in some Hyperprism versions. Use this parameter to select the center frequency, which is the maximum point of amplitude. This parameter ranges from 13 Hz to 20 kHz in Hyperprism 2, Hyperprism-MMP and Hyperprism-VST, and ranges from 0-20 kHz in Hyperprism-DAS.

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.

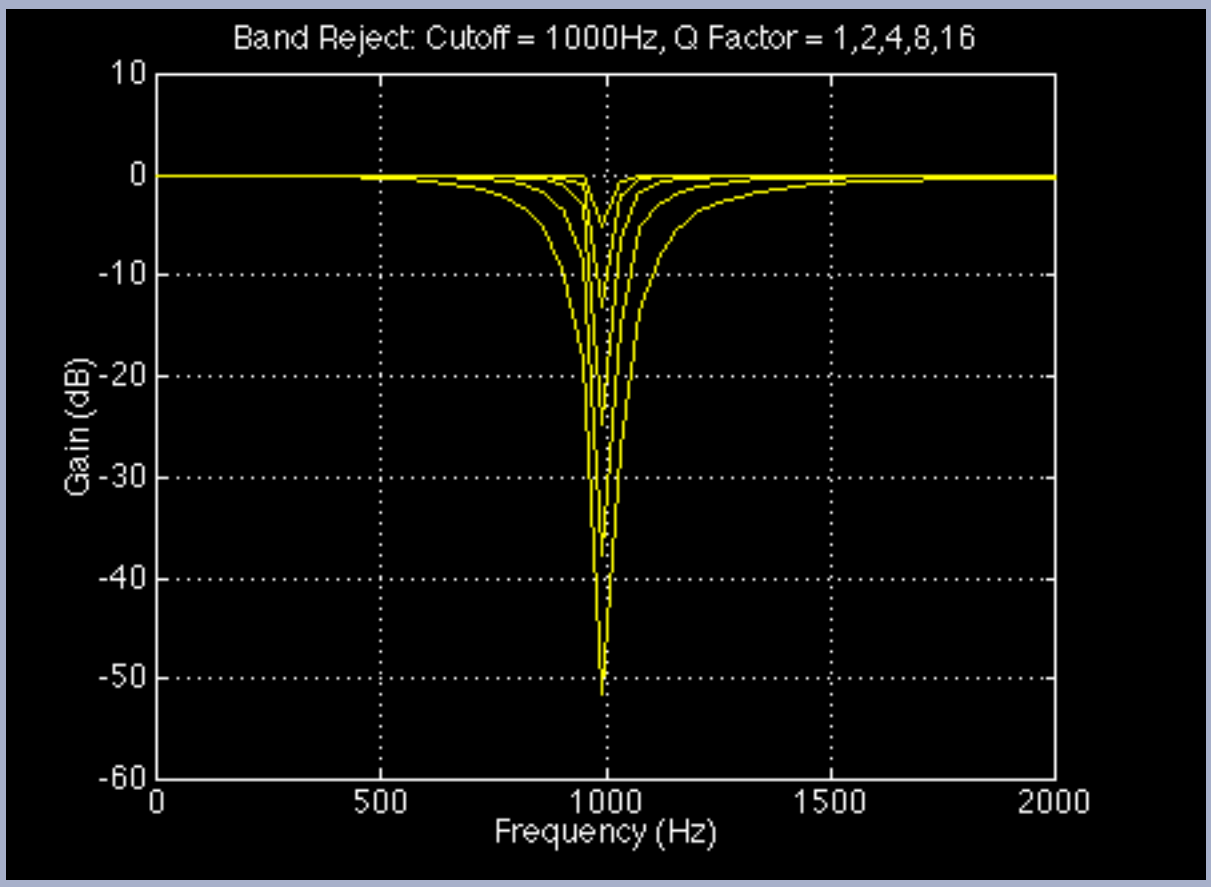
- Select a center frequency of 1050 Hz and a Q factor of 200.

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 2] [Appendices]

Arboretum SYSTEMS

Hyperprism™ Manual

Hyperprism Modulation Processes

Vocoder

H2 VST MMP G DAS

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 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 DirectX edition 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.

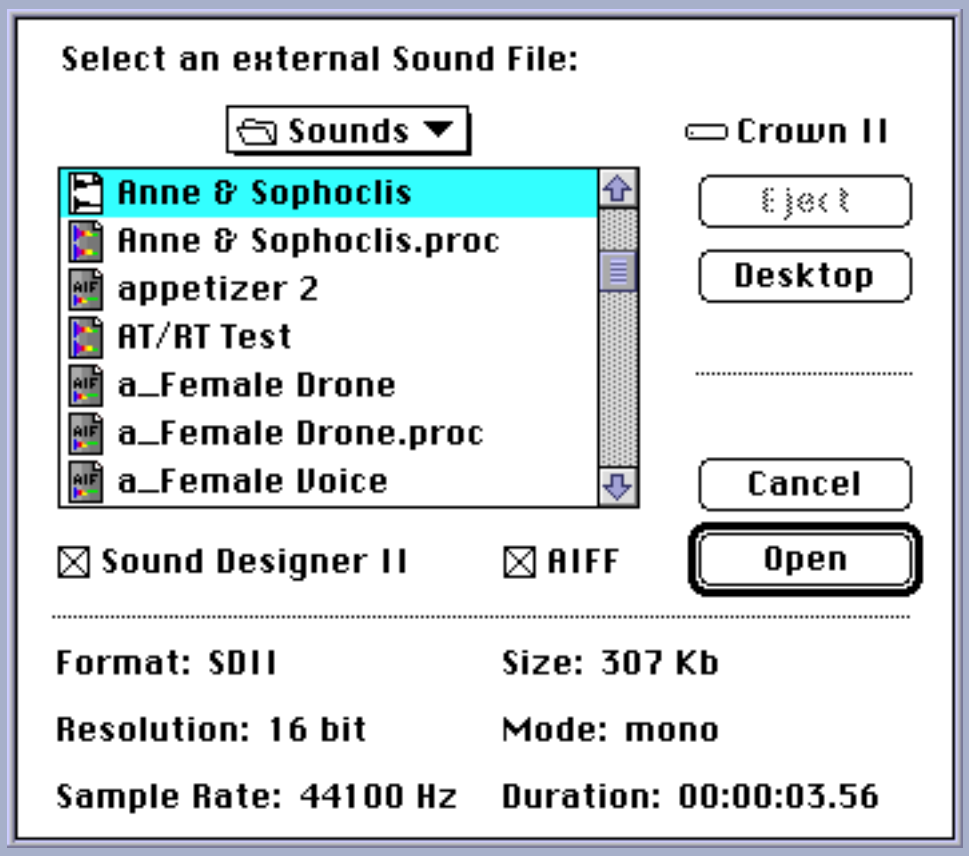
Internal/External Signal

(If you're not using Hyperprism 2 with HyperEngine, Hyperprism VST, or Hyperprism MMP, you can skip the next bit

and continue below.) The Hyperprism 2 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.

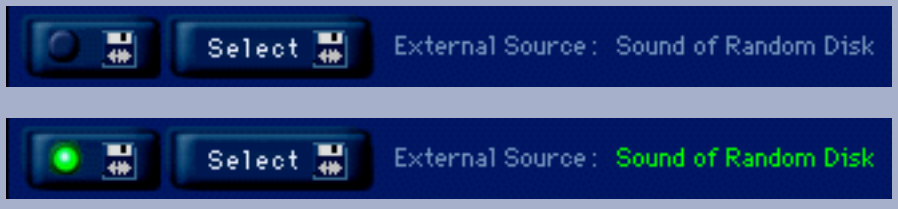
Clicking on the Select External button results in a dialog where you select a sound file to be the external signal. Note that the External signal must be a sound file; you can't use live play-thru as the "external" source, play-thru is always an "internal" signal.



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.

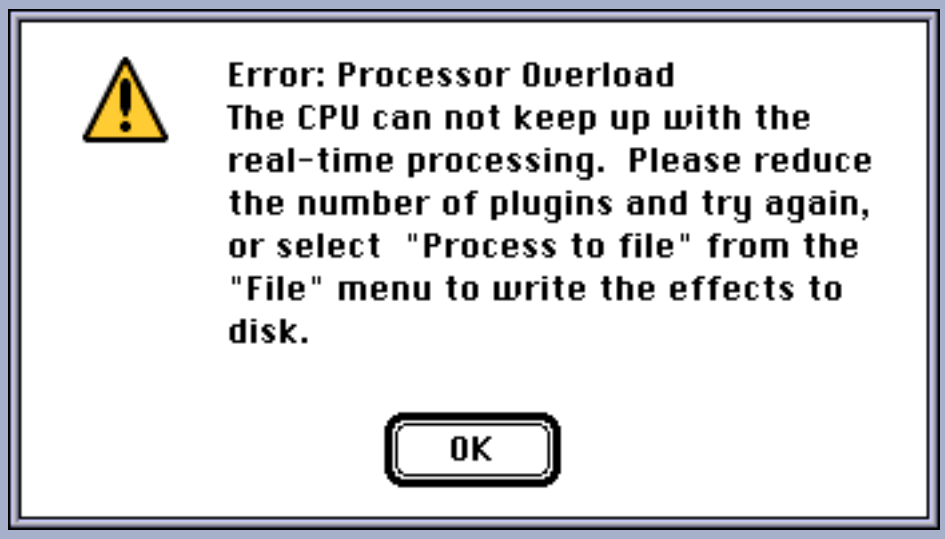


l-r: External On/Off and Select External buttons, plus External Source text display (Off and On states)

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" signal 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.



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 MMPonly.

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.

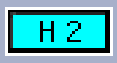
select vertical or horizontal axis by using the three-way toggles

- 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.

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.

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).

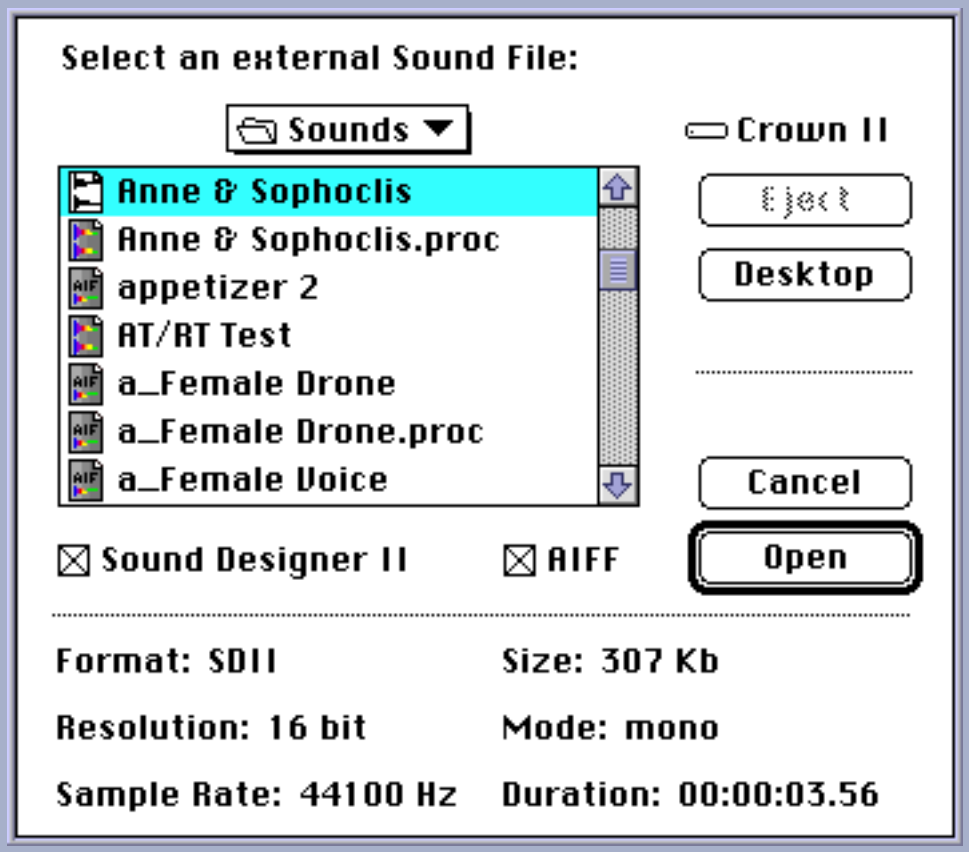
The Z-Morph operates somewhat like the Vocoder except frequency bands are not fixed. This allows the algorithm to best fit the incoming signal. Frequency bands move depending on the incoming frequency.

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.

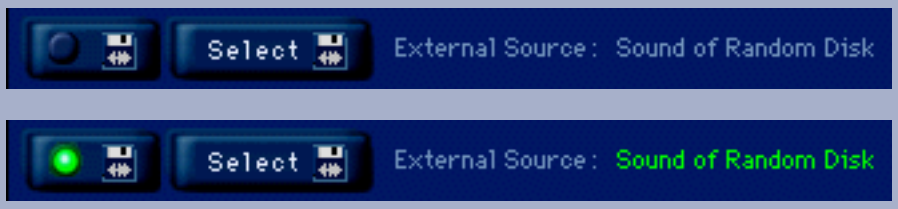


Select External sound file dialog

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

www.arboretum.com for updates.

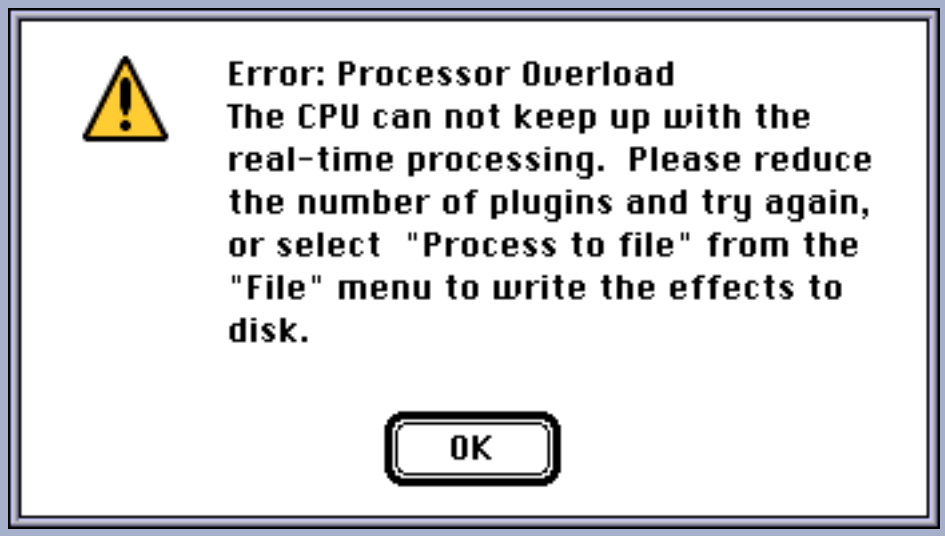
Once you've selected a mono or stereo file as the External source, switch the External On/Off button to the on position, which enables modulation with the file you've just selected.



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

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 kHz 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.

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

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 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

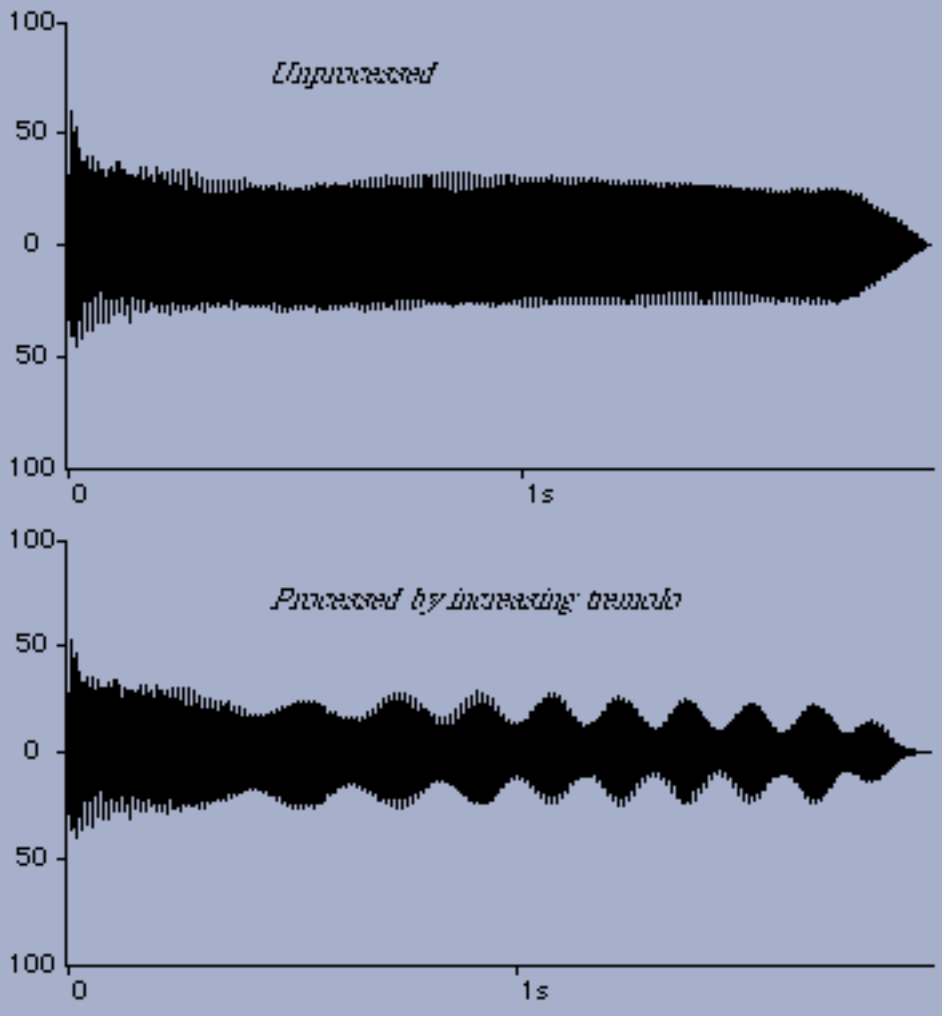
Create a New HyperEngine document for the file "Train," Play it and select a Tremolo process.

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

H2 VST MMPG 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 2] [Appendices]

Arboretum SYSTEMS

Hyperprism™ Manual

Hyperprism Delay & Reverb Processes

Single Delay

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

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

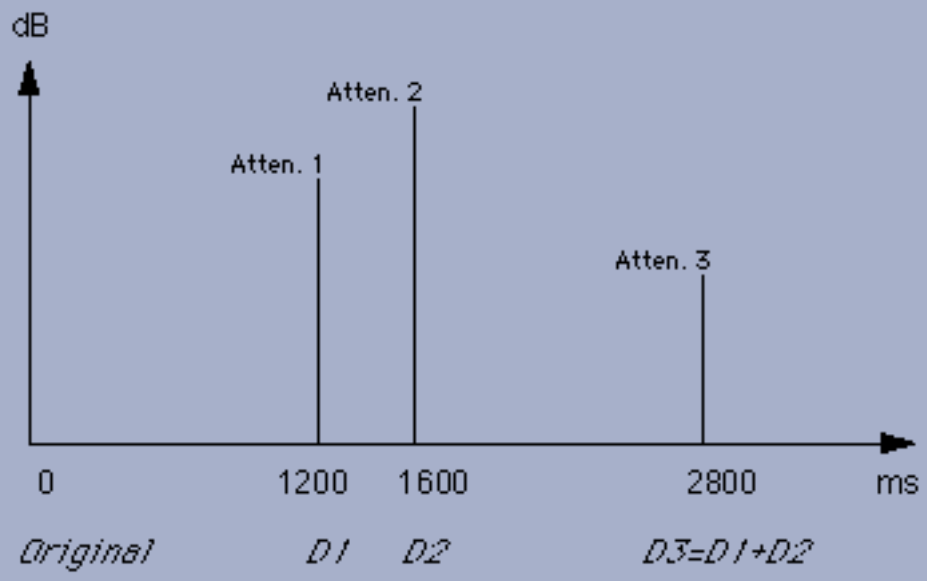
Delay 1 in milliseconds, by default 0 to 1000. Goes up to 2 k in Hyperprism 2.

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%.

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.

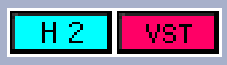
Mix

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)



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.

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.

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.

Room Reverb

H2 VST MMPG DAS

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.

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.

Hall Reverb

H2 VST MMPG 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.

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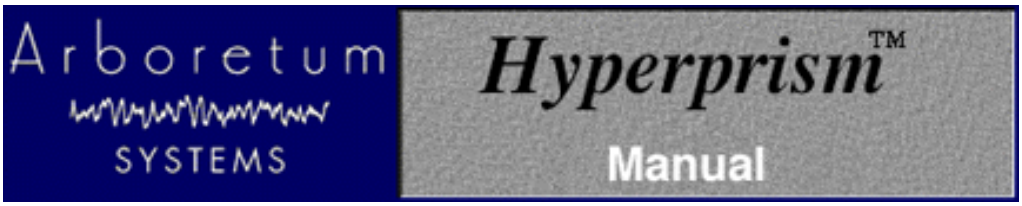
Go to Effects: [Filters] [Modulators] [Delay and Reverb] [Stereo] [Misc Effects]
Go to Table of Contents: [Hyperprism 2] [Appendices]



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.

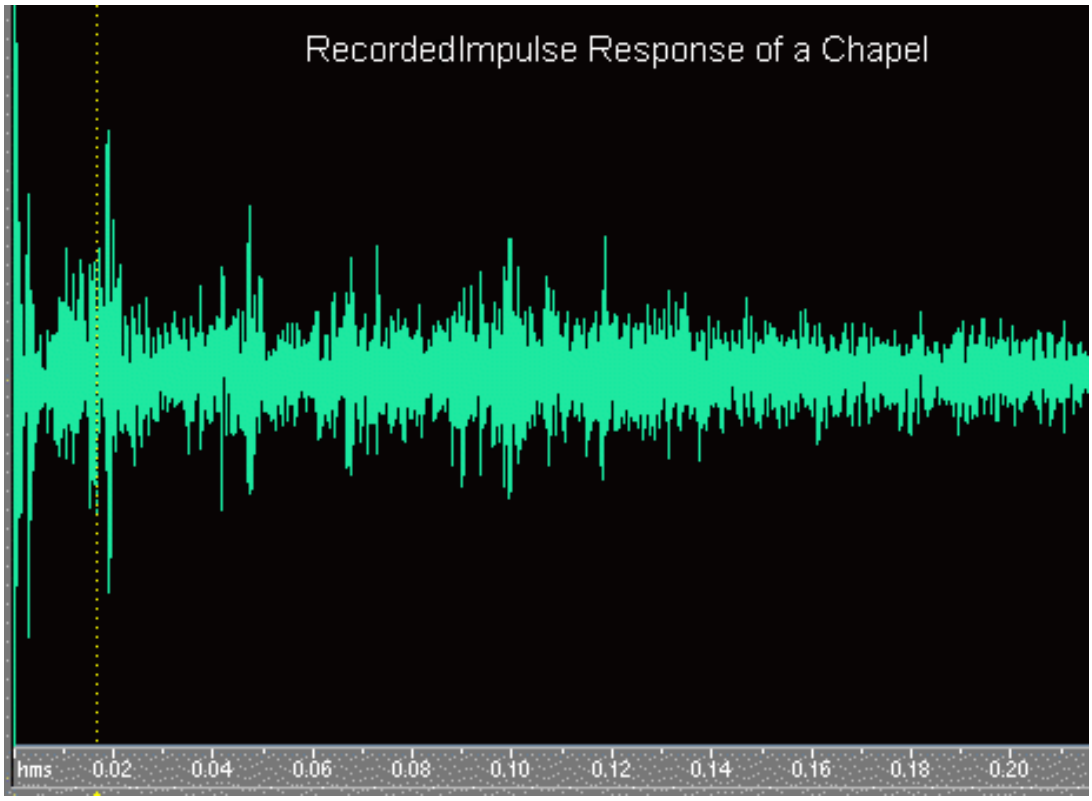
The properties of high quality reverberation are very complicated, but for the purposes of this report we will limit the analysis to aspects that can be easily and objectively measured and that are commonly problematic with most digital reverbs. Of course the most important test is simply listening to 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!

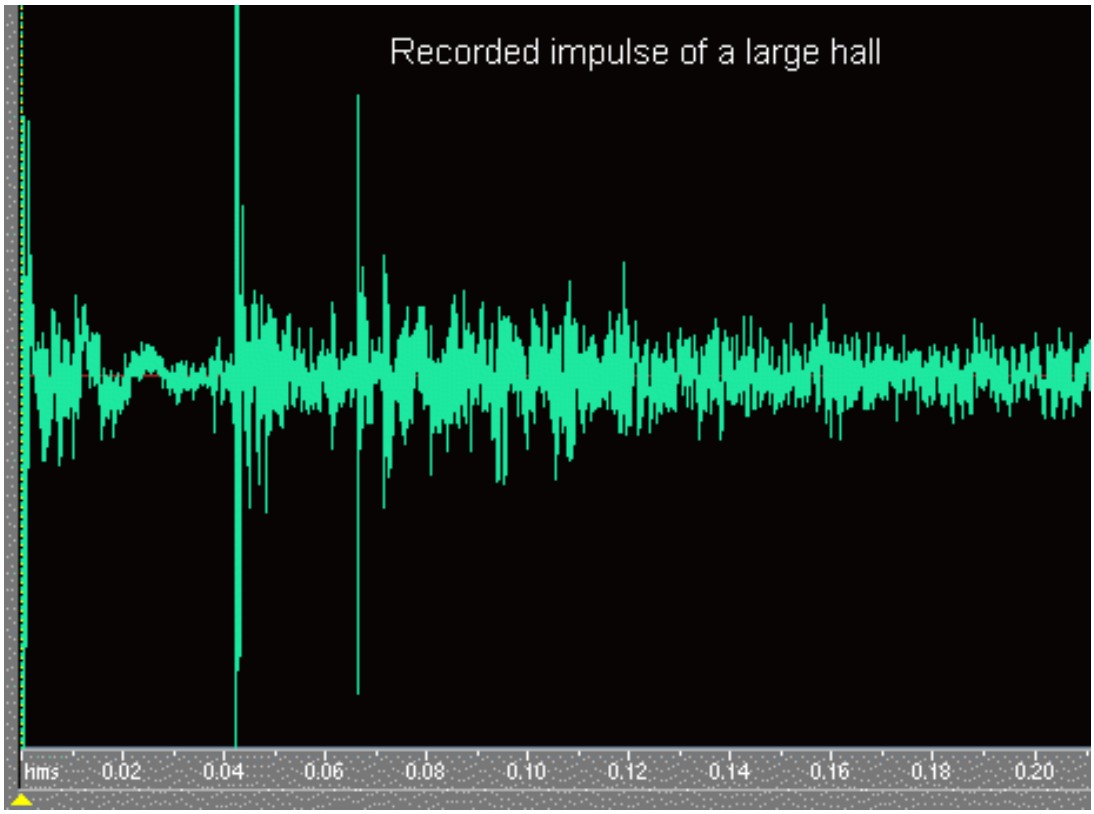
All of the tests performed here are based on examination of each reverberator's impulse response. An impulse response was obtained by sending a one sample wide "click" into the reverberator and recording the output. Frequency domain and time domain plots can then be made from the impulse response. From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

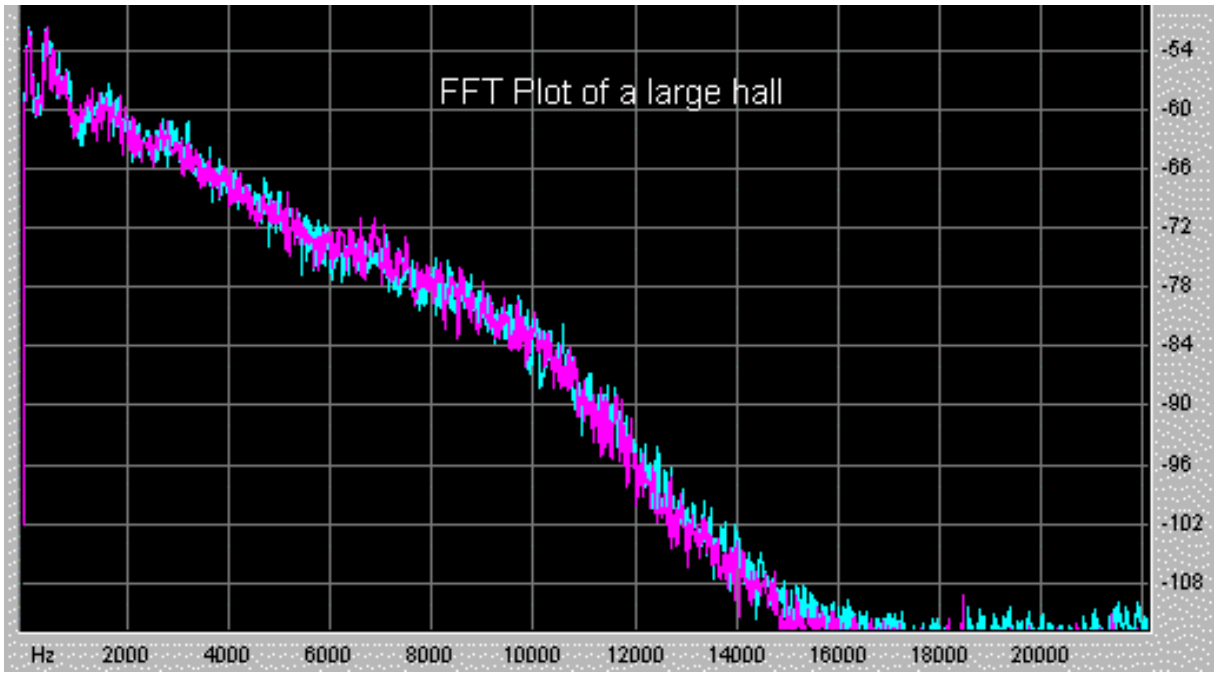
Echo density, often referred to as density or diffusion, can be defined as the number of "echoes" occurring per second. For example, an impulse response where the energy is concentrated at a few samples (i.e.. most samples are zero) will have a low echo density, whereas an impulse response with energy 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
hms
0.02
0.04
0.06
0.08
0.10
0.12
0.14
0.16
0.18
0.20


Recorded impulse of a large hall
hms
0.02
0.04
0.06
0.08
0.10
0.12
0.14
0.16
0.18
0.20



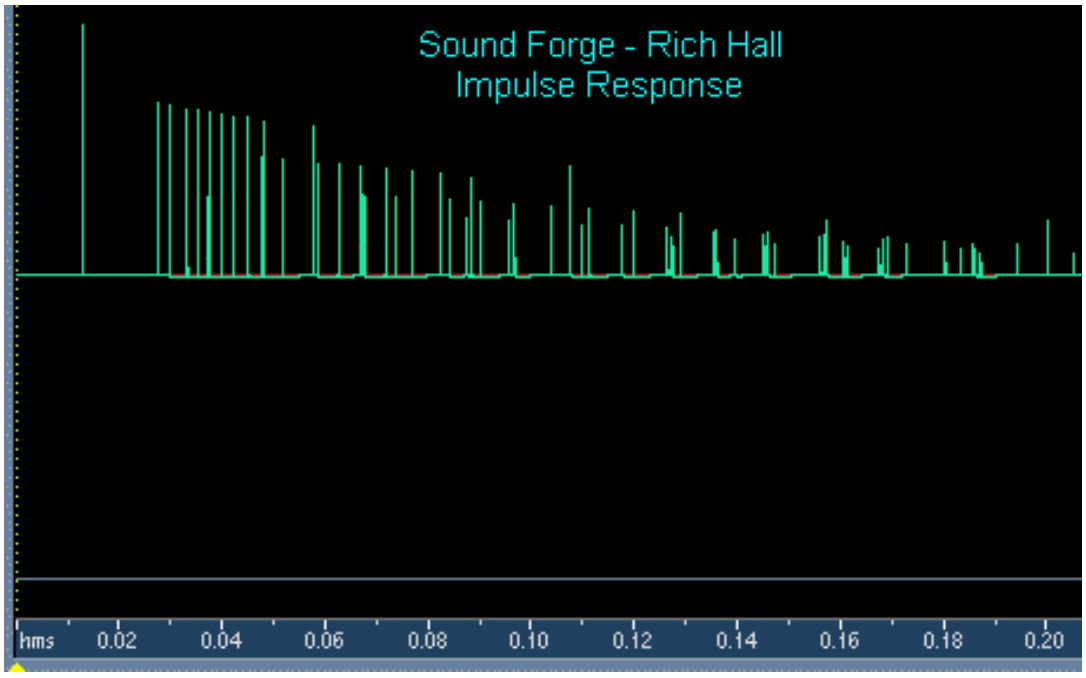
(NOTE: Cyan = Left Channel and Magenta = Right Channel)

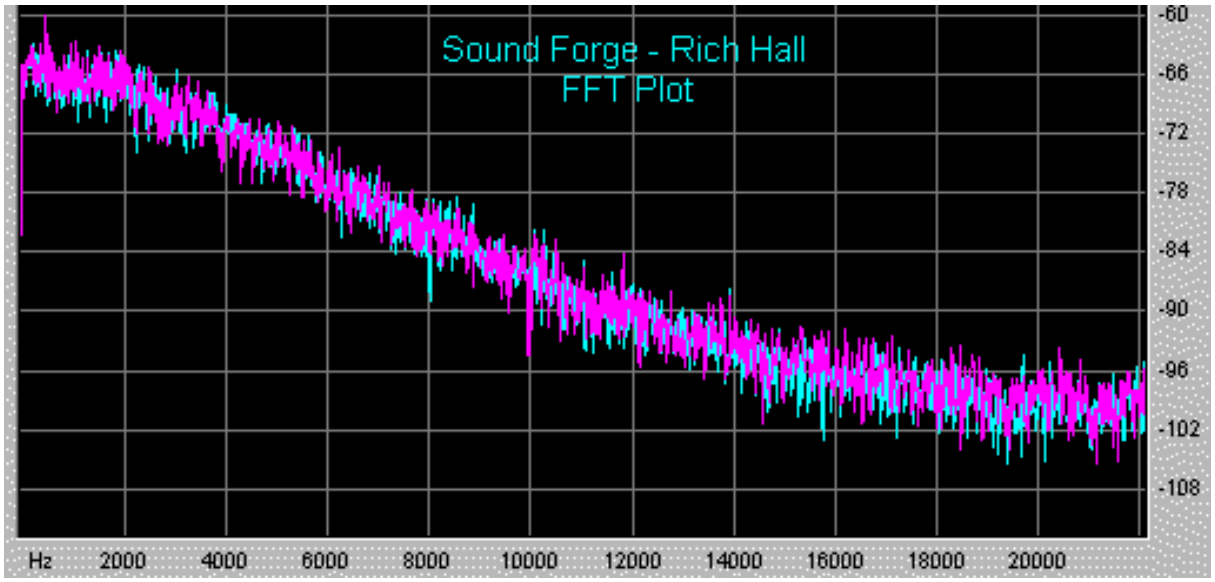
Looking at these plots we can observe a number of distinguishing features. Some of these features include:

- 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:





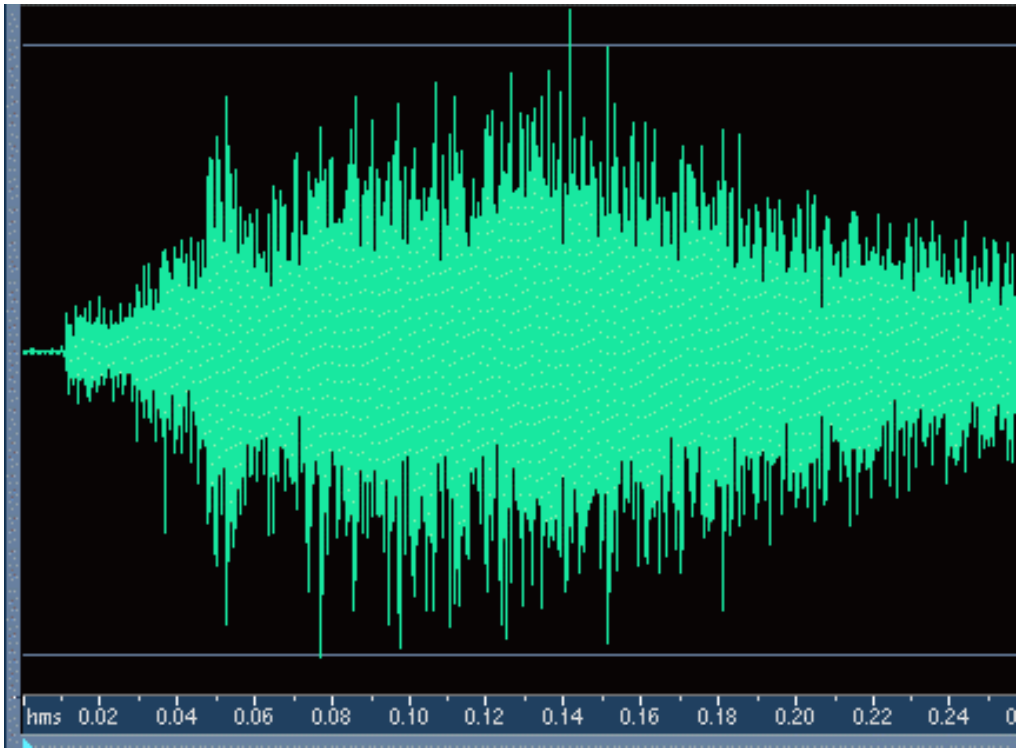
Pros:

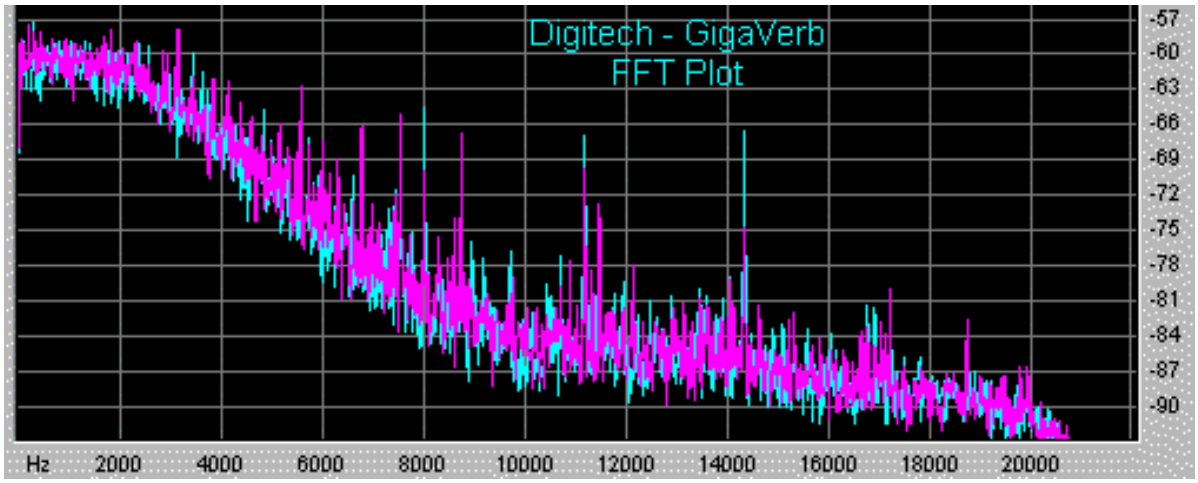
- Somewhat colorless late decay

Cons:

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

Digitech GigaVerb:





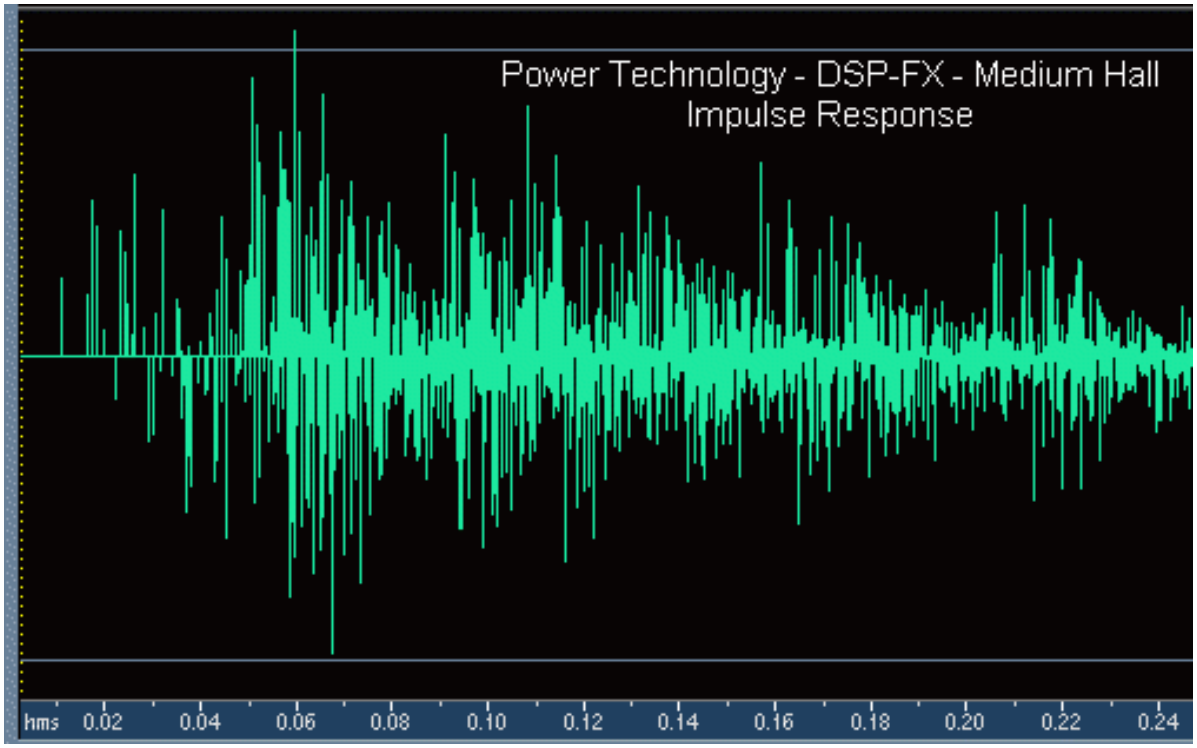
Pros:

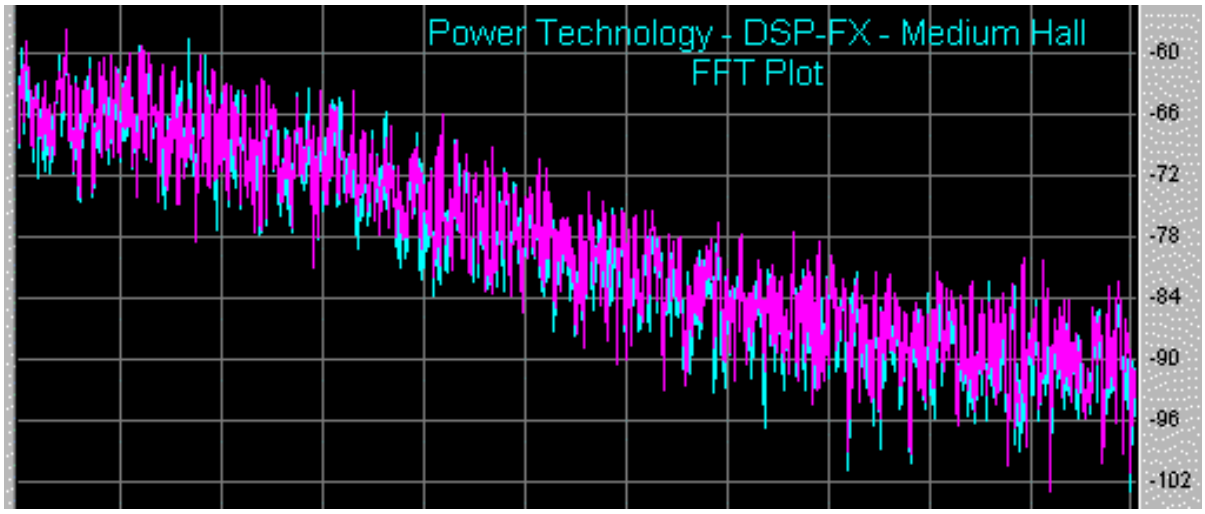
- High echo density

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:





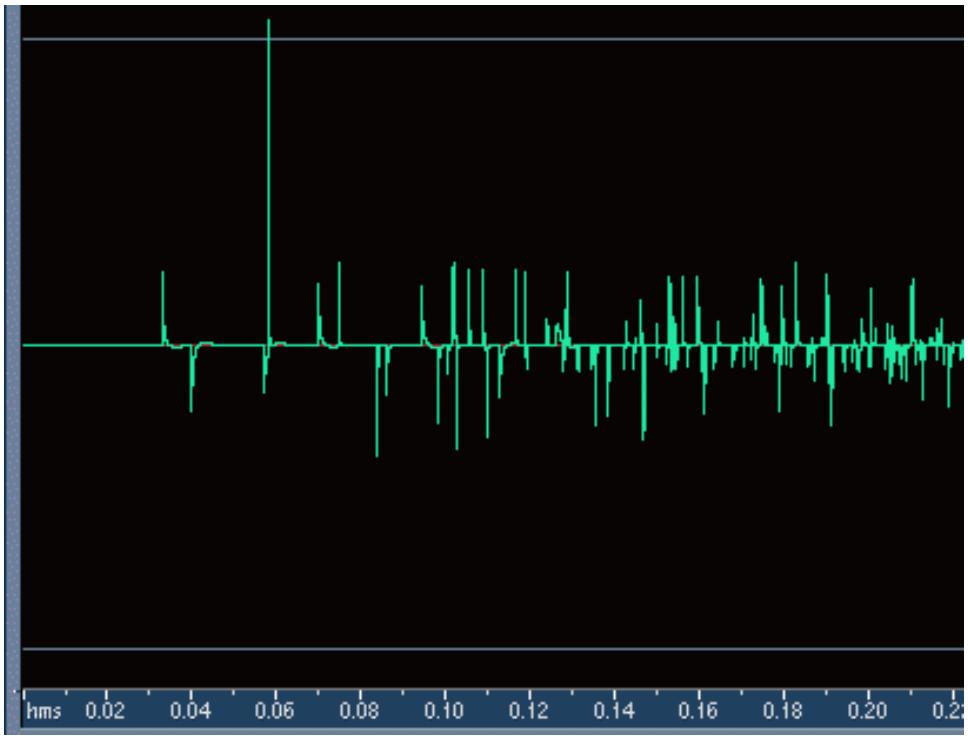
Pros:

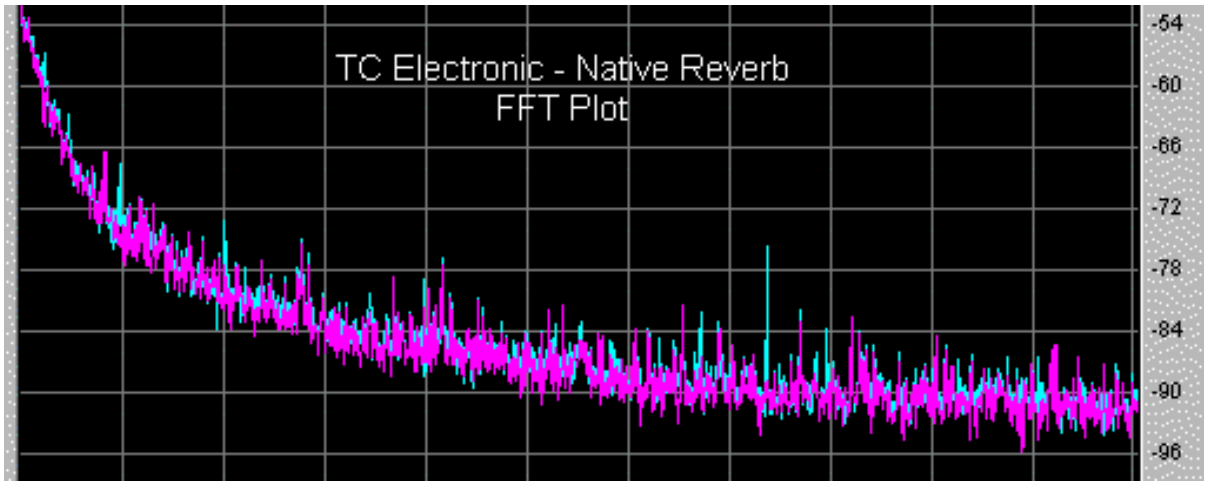
- Moderate echo density (though not always enough)

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:





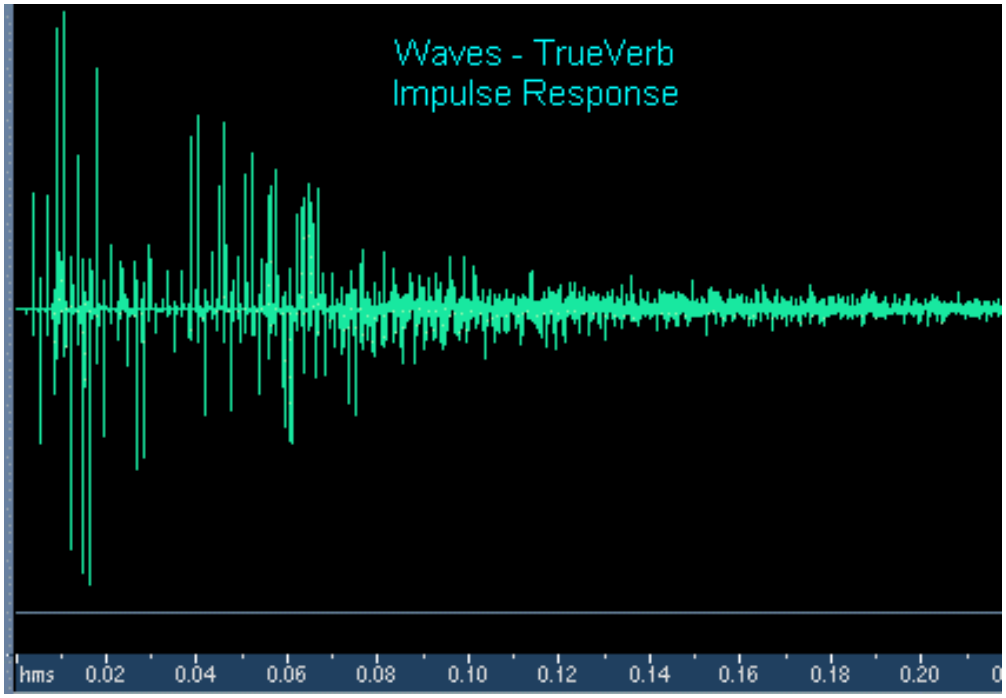
Pros:

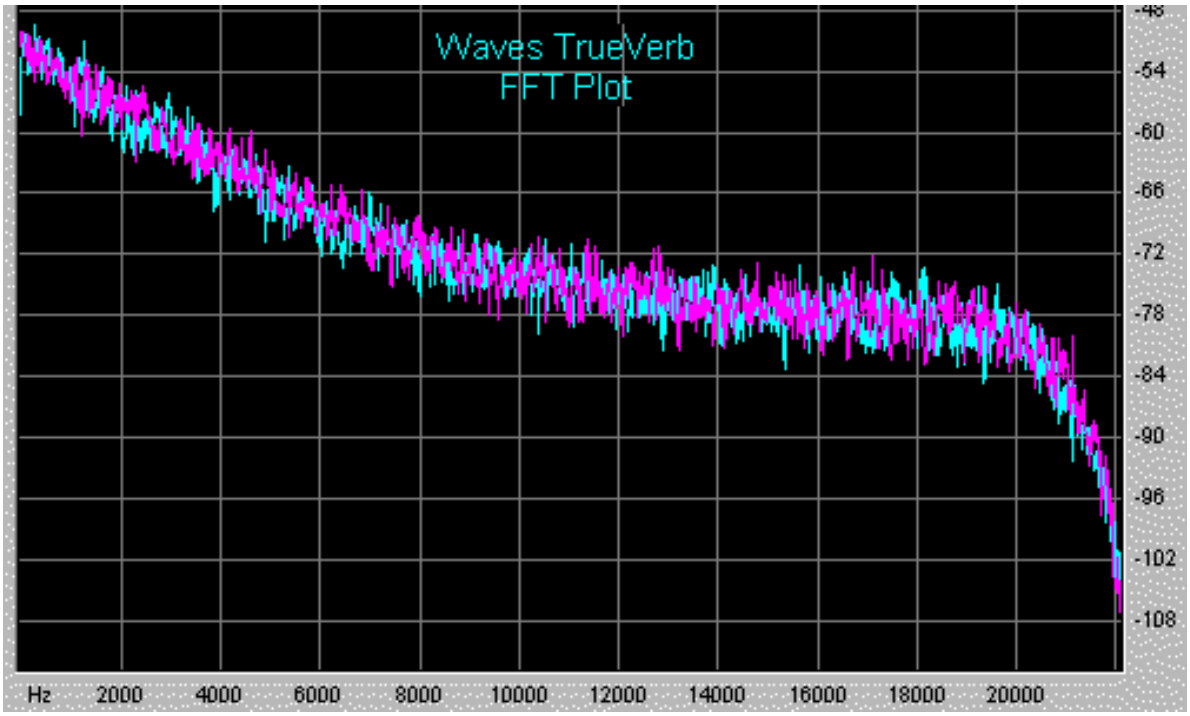
- Deficiencies are not a pronounced as some reverbs

Cons:

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

Waves TrueVerb:





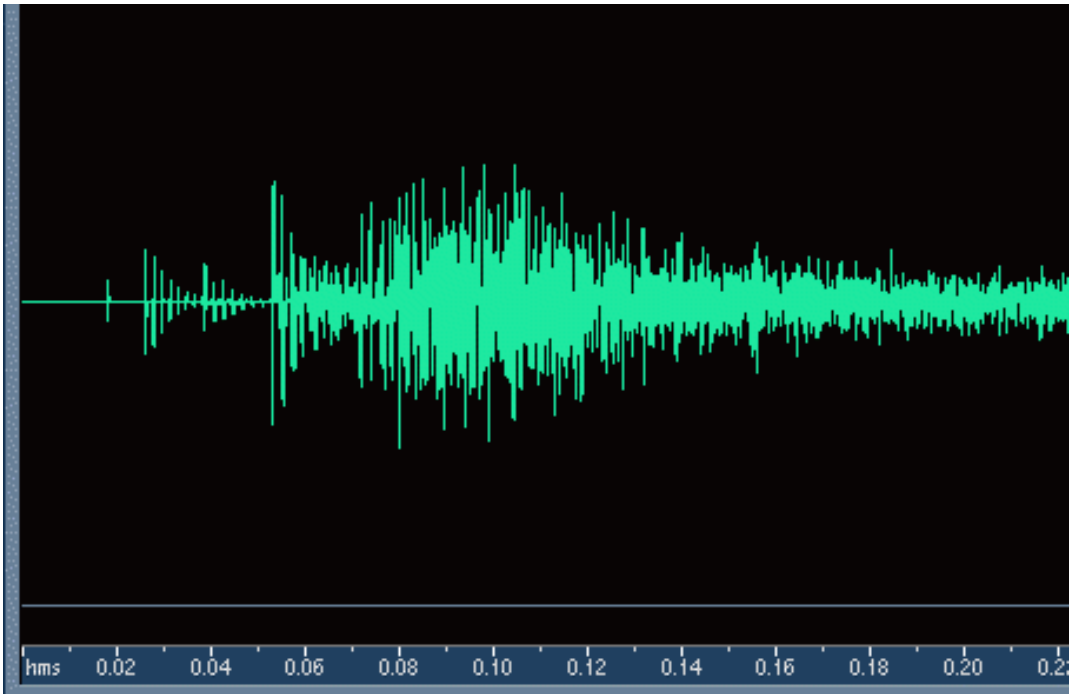
Pros:

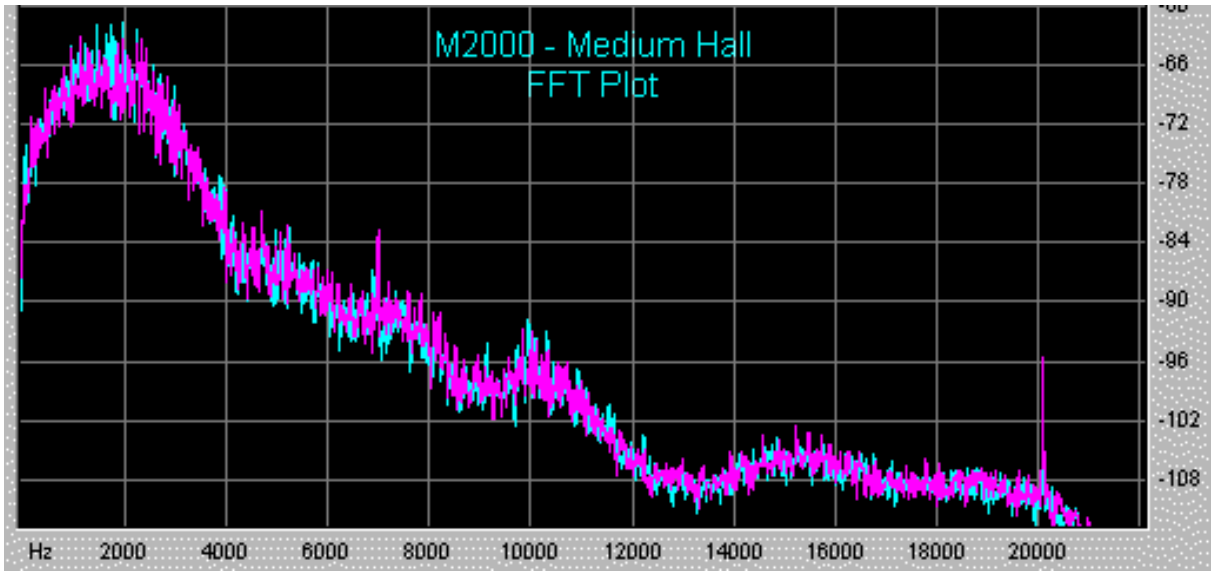
- Moderately smooth decay of late reflections (i.e. minimal flutter)
- Late reflections are fairly colorless

Cons:

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

TC Electronic M2000:





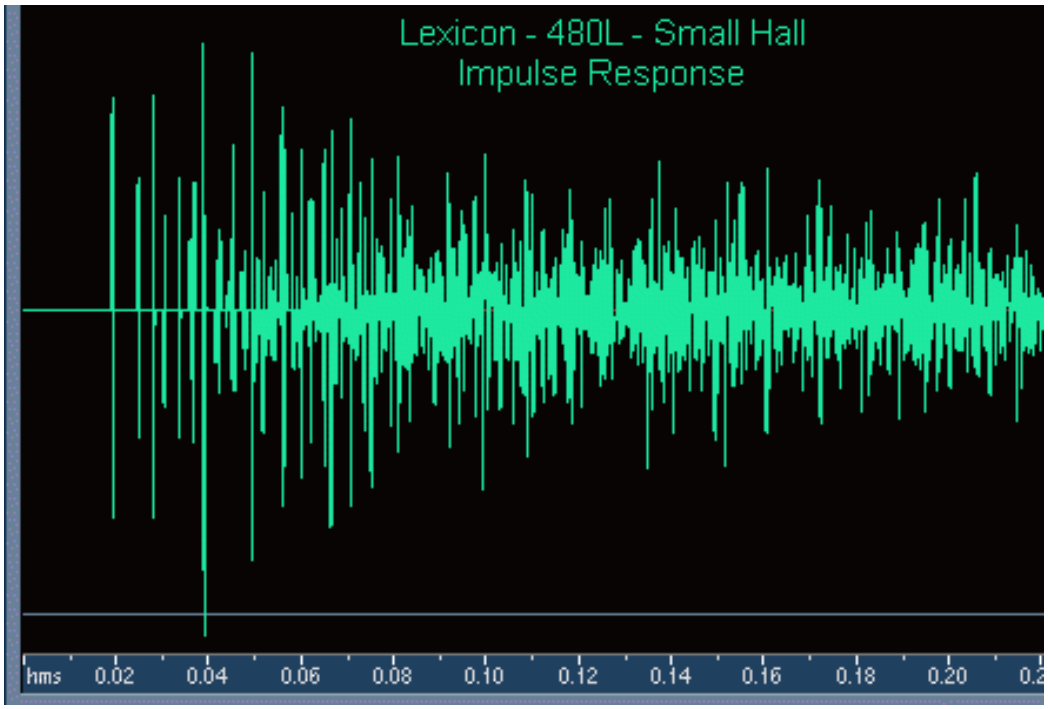
Pros:

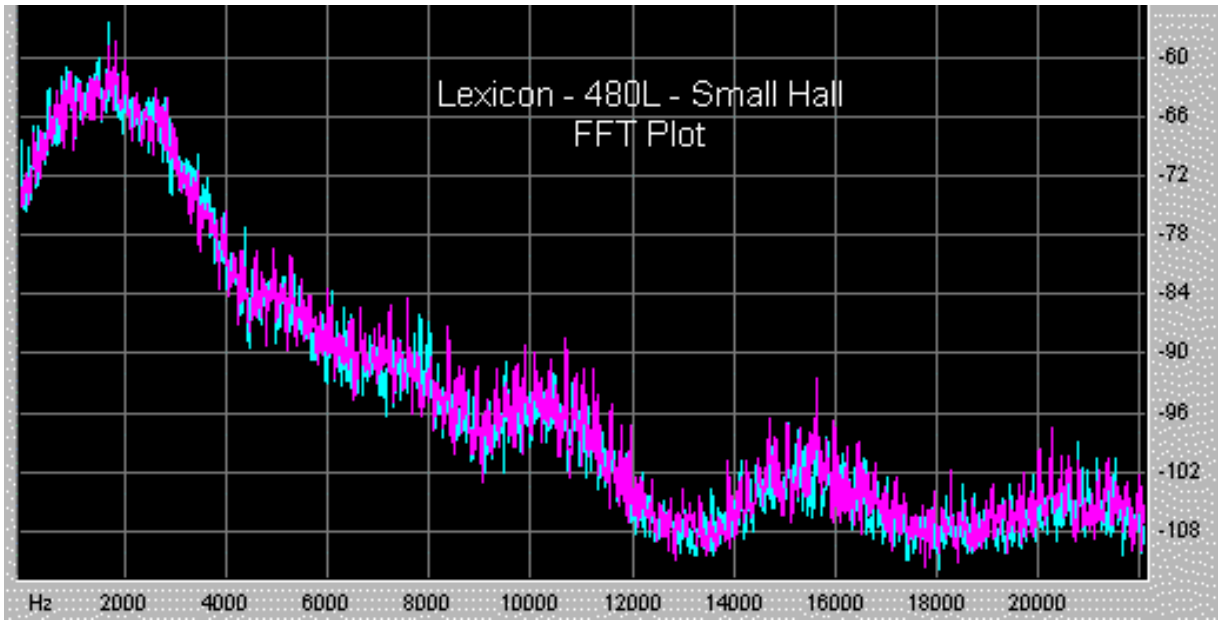
- Very high density and diffusion
- Moderately smooth decay of late reflections (i.e.. minimal flutter)

Cons:

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

Lexicon 480L:





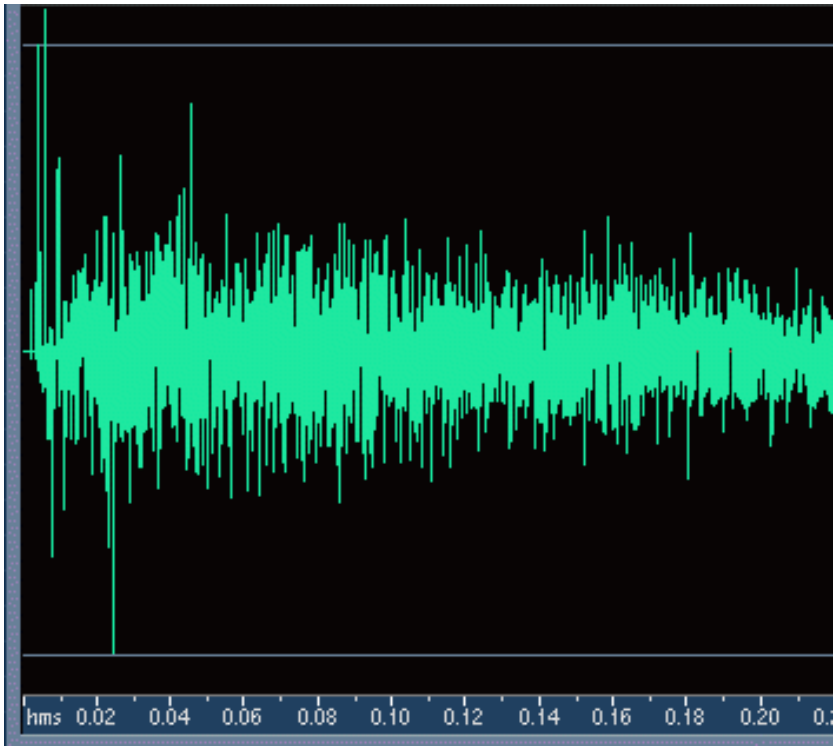
Pros:

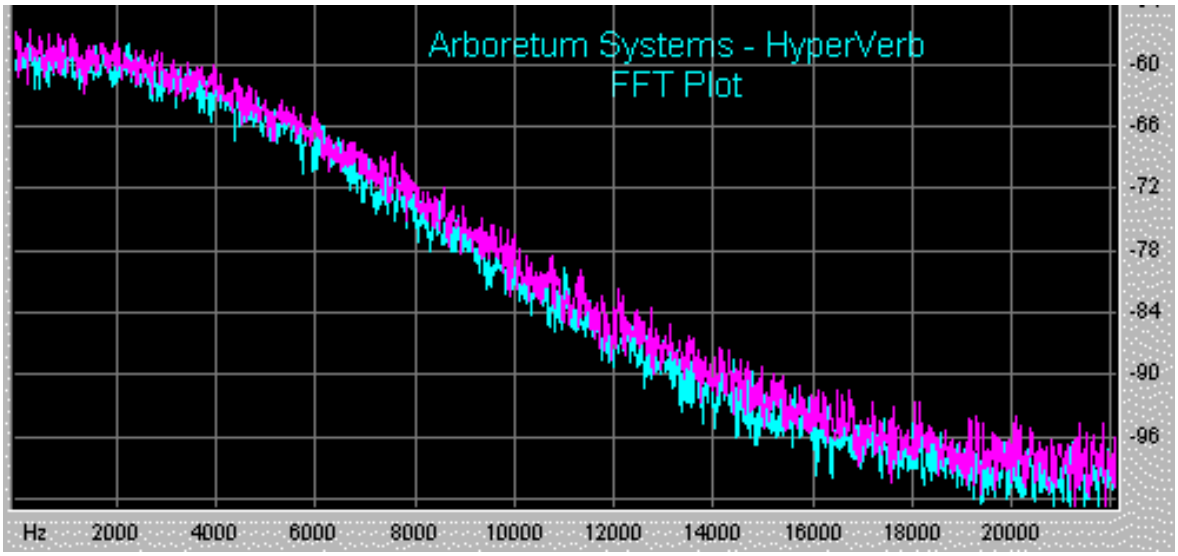
- Drastic improvement in sound quality compared to most reverbs
- Sharp attack
- Smooth decay of late reflections (i.e. minimal flutter)

Cons:

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

Arboretum Systems HyperVerb:





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.

Arboretum SYSTEMS

Hyperprism™ Manual

Hyperprism Stereo Manipulation Processes

Pan

VST MMP G DAS

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.

Parameters

Position (L <-> R)

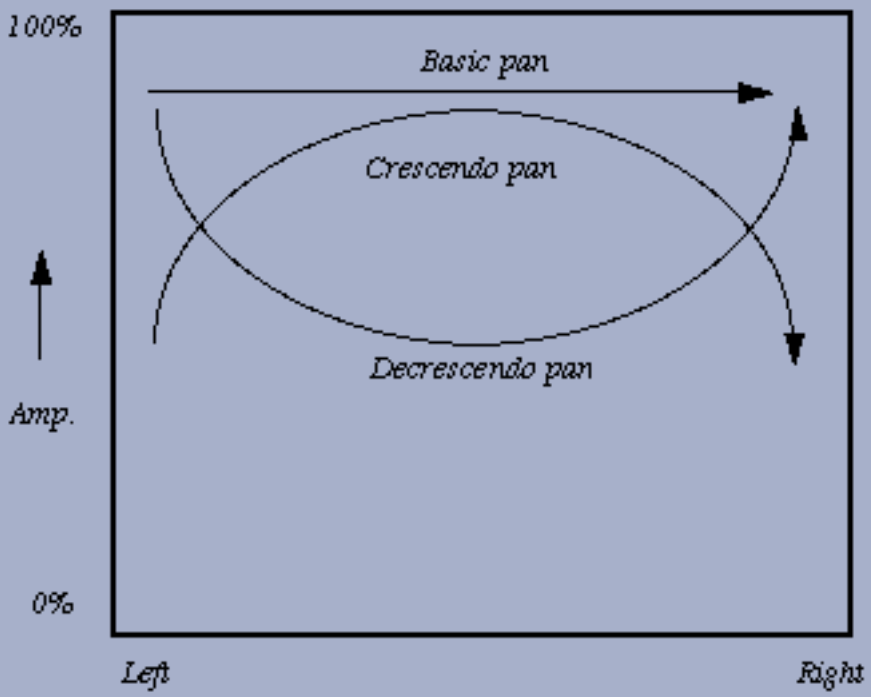
Called LR Balance (%) in Hyperprism 2, this parameter controls the position of the sound in the plane from left to right. This parameter ranges from -100 to +100% where 0 corresponds to the center position in Hyperprism 2. In Hyperprism-VST, Hyperprism-MMP and Hyperprism-DAS this parameter ranges from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

For a more 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 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. By definition, this effect transforms a mono signal into a stereo one. 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.

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.

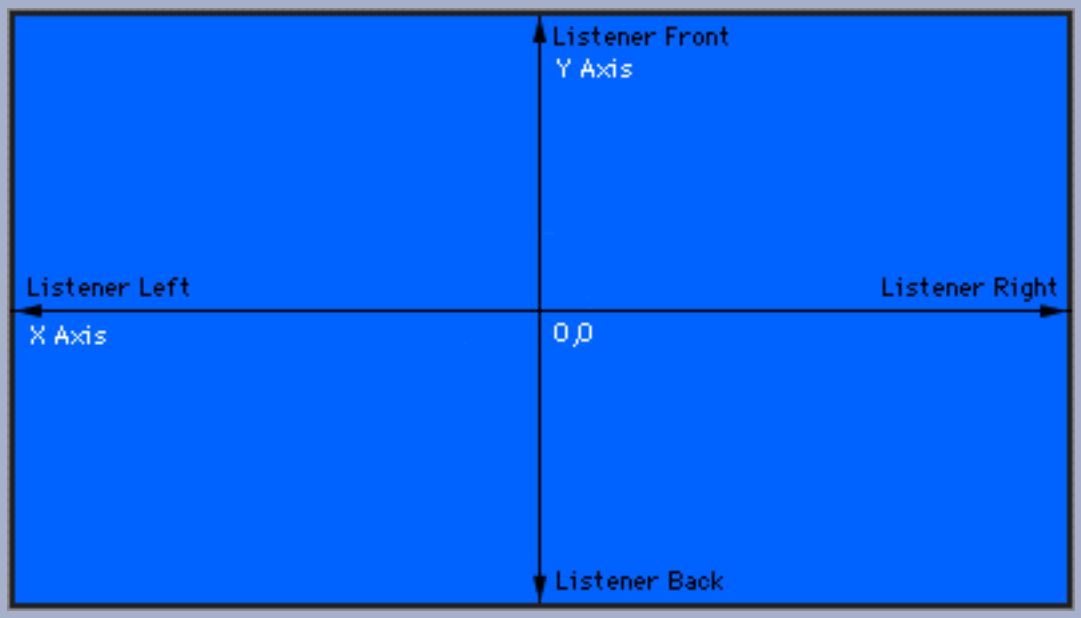
However, Position settings below 50% will cause the sound to spend correspondingly more time in the left channel, thus appearing to alter it's "position" relative to the center of the stereo field. Position settings above 50% will cause the sound to spend correspondingly more time in the right channel. 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

H 2 VST

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.

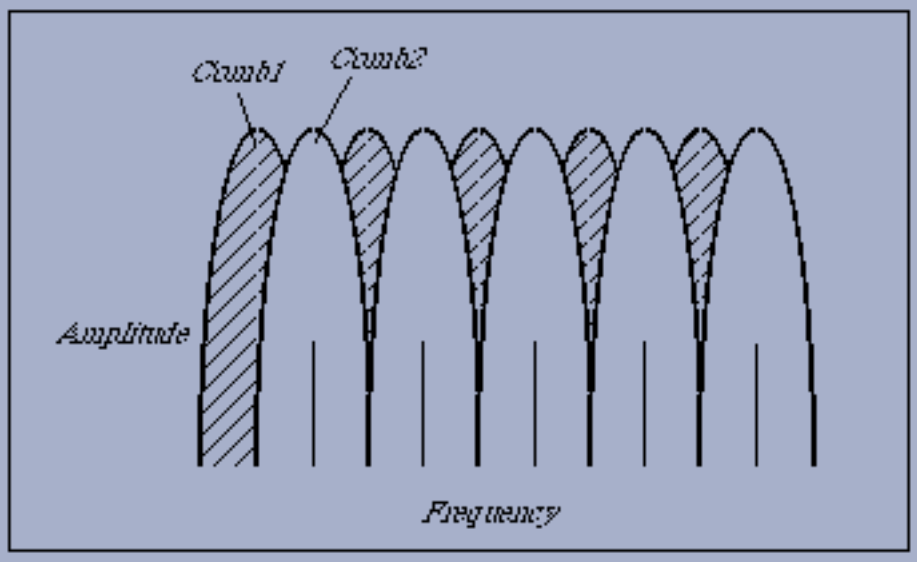
Y Axis

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

H2 VST MMPG DAS

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.

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

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 Process to New File... command creates a stereo sound file as the output.

☞ 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

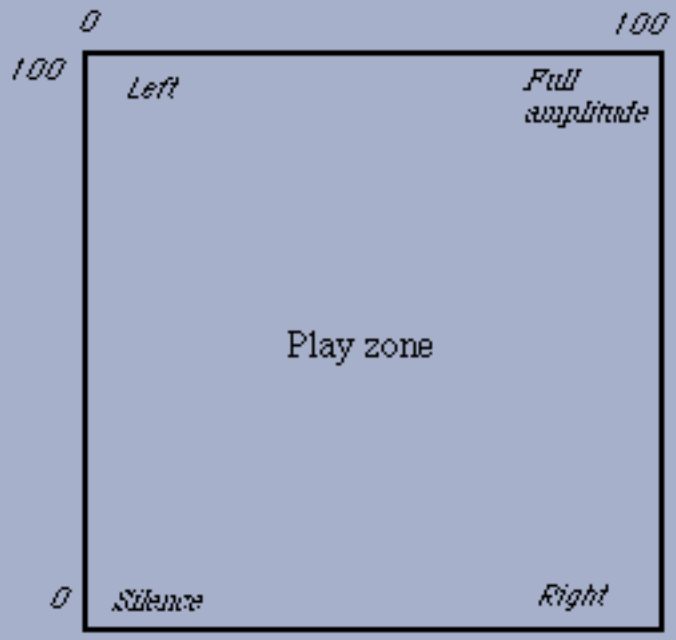
This is "right-minus-left gain," which controls the proportion of one channel relative to the other. For example, moving the tracer to the right will increase the proportion of the right channel in the output. The range is 0 to 100.

Right+Left Gain

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

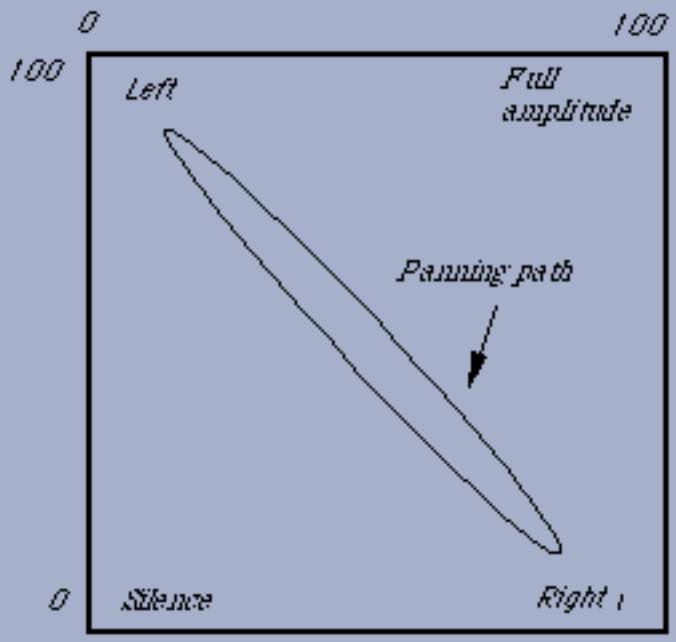
Application Diagrams

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.



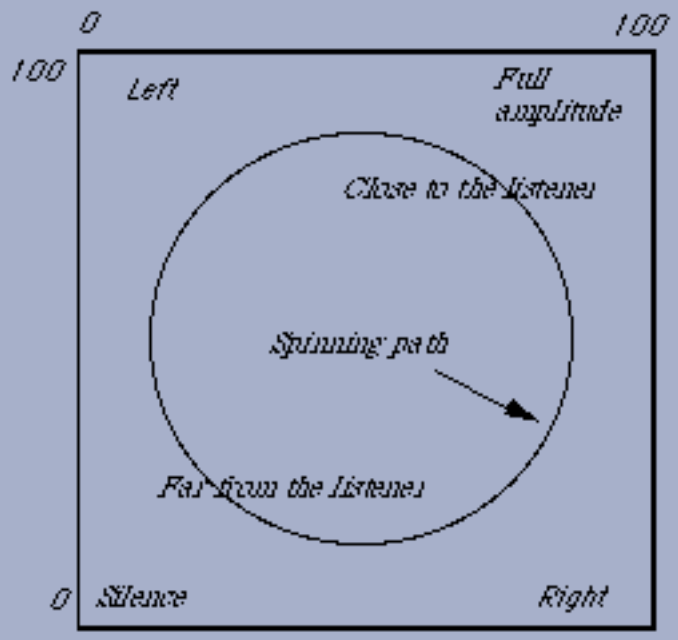
Default layout of the play zone for Stereo Dynamics in Hyperprism-VST, Hyperprism-MMP and Hyperprism-DAS

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

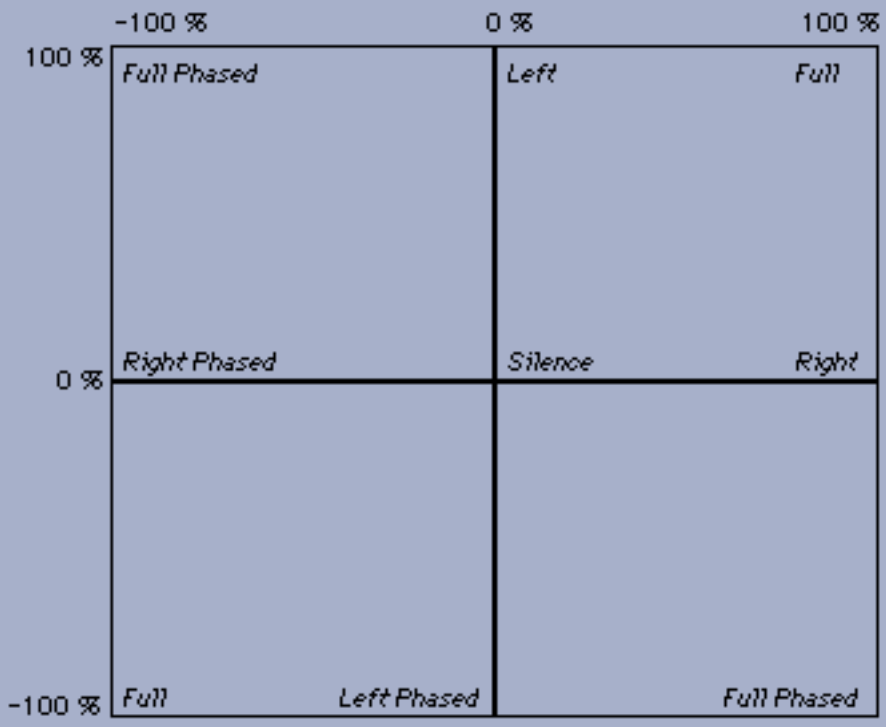
This is "right-minus-left gain," which controls the proportion of one channel relative to the other. For example, moving the tracer to the right will increase the proportion of the right channel in the output. The range of values is -100 to 100.

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.

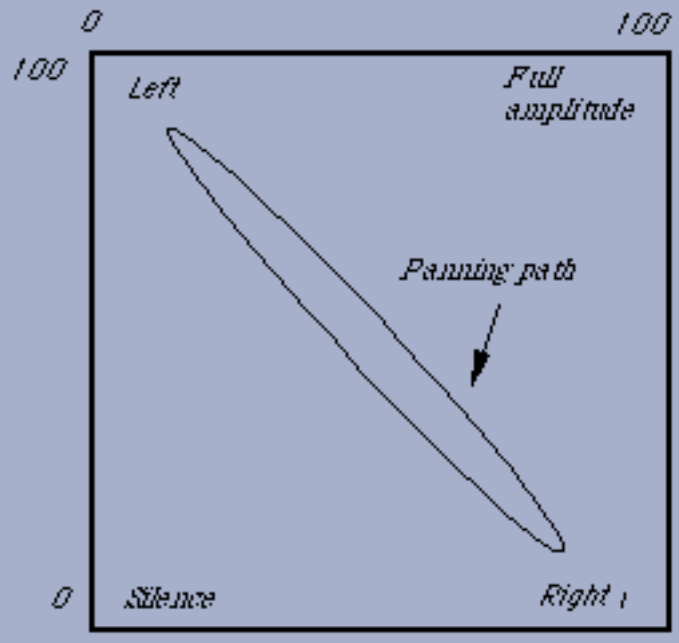
Application Diagrams

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.



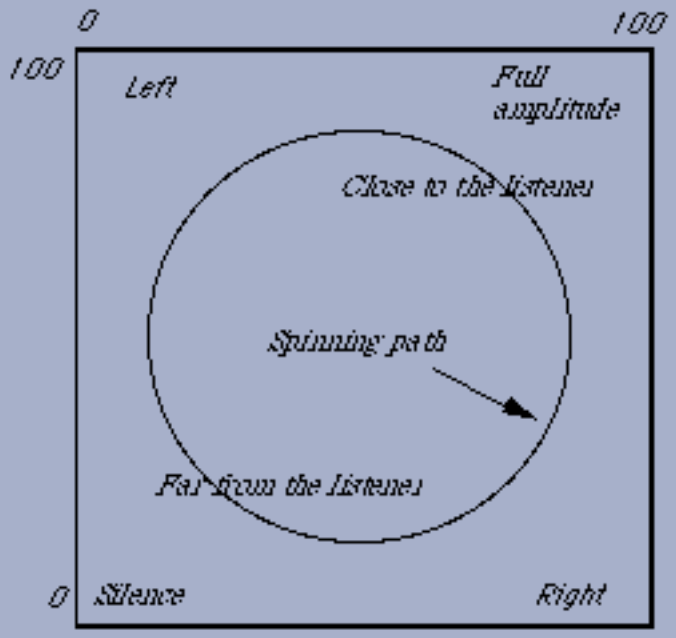
Default layout of the play zone for Stereo Dynamics in Hyperprism 2

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

H2 VST MMPG 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.

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.

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.

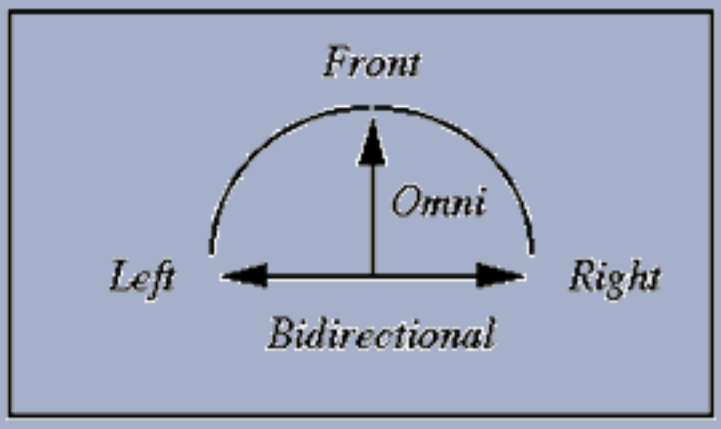
Limitations

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 | 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.

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

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 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.

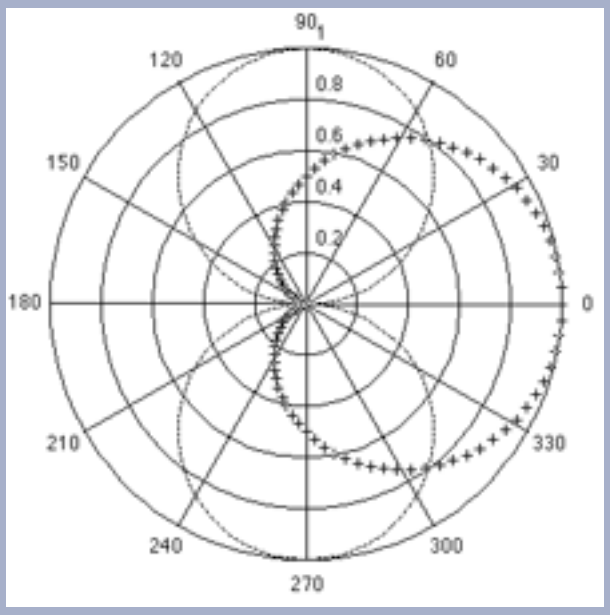
According to the sound engineer who made this recording, the normal stereo effect is rendered when the center position is 0 degrees and the Side Volume is 30%. Do this and listen to the sound for a while. Now decrease the Side Volume. As you drag toward the bottom of the range you'll hear the sound gradually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

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

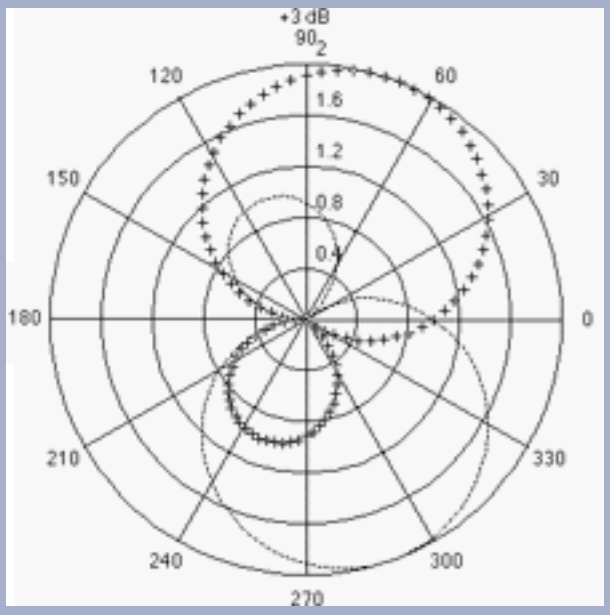
With the Side Volume up high, move the Center Position control. The apparent "center" of the stereo image shifts to the right and left, just as if you were readjusting the position of the M microphone during a recording session.

Spatial Patterns Obtainable with M-S Decoding

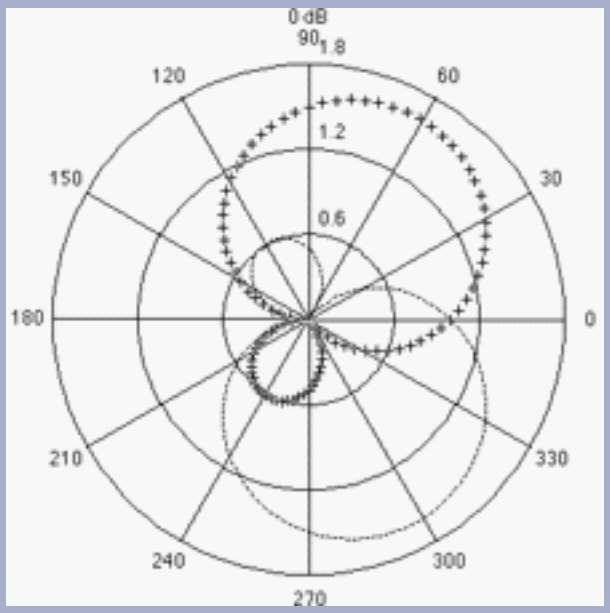
The polar diagrams below show the range of spatial patterns obtainable from a single M-S recording using M-S Matrix and Hyper Stereo (if included in your Hyperprism package). The captions list the X and Y positions of the tracers followed by a figure in decibels that indicates the level of the S part of the signal relative to the M part.



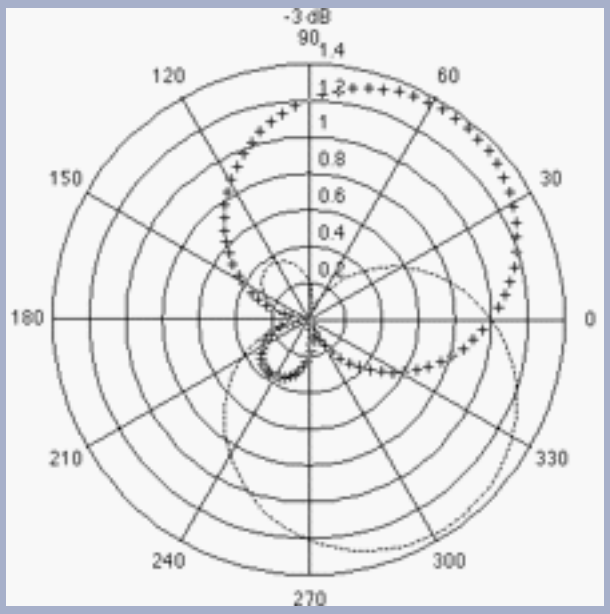
Directivity patterns for a bi-directional (--) and cardioid (++) microphone in M-S recording configuration. This reflects the physical configuration of the microphones in the original recording session.



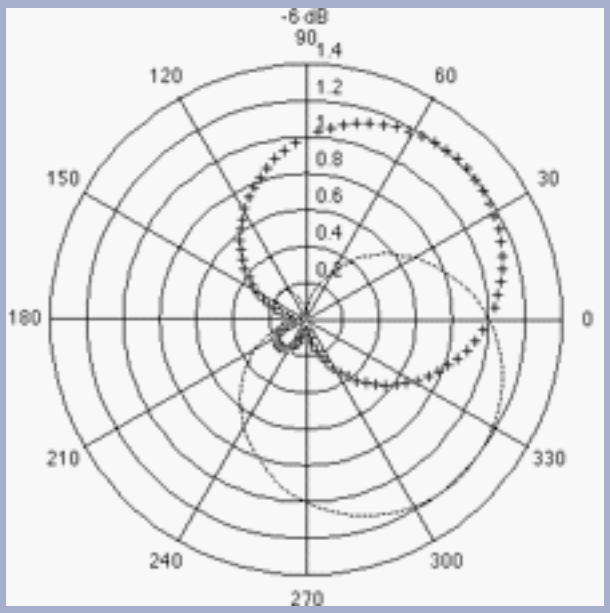
Hyper Stereo: Stereo +100%, Mono 50%,M-S Matrix: pattern at +3 dB



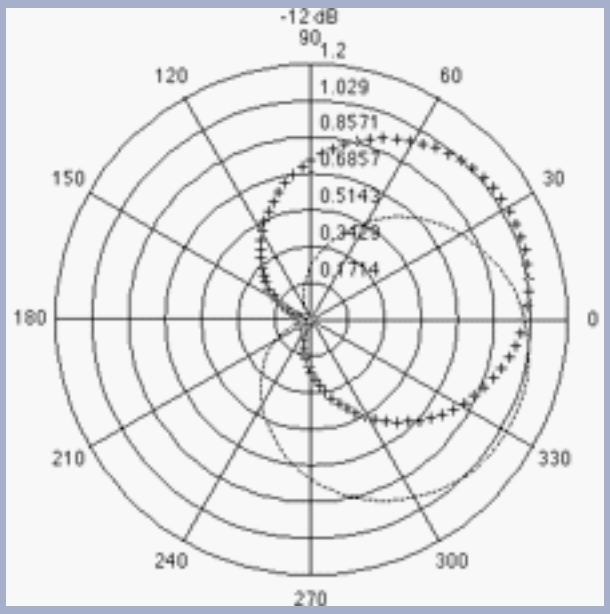
Hyper Stereo: Stereo +100%, Mono +100%, M-S Matrix: pattern at O dB (equal level)



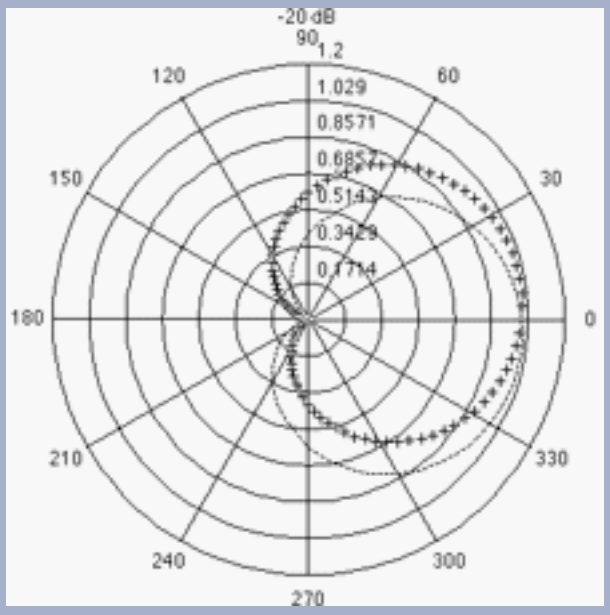
Hyper Stereo: Stereo +71%, Mono +100%, M-S Matrix: pattern at -3 dB



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 2] [Appendices]

Arboretum SYSTEMS

Hyperprism™ Manual

Hyperprism Miscellaneous Processes

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, including the Aphex Aural Exciter(TM), 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

The Bass Maximizer plug-in offers a wide assortment of bass enhancement and maximization effects, ranging from subtle bass frequency harmonic generation to extreme compression. The harmonic generation feature increases the perceived amount of bass, even on low end systems with poor bass frequency response, by adding harmonics which our brains use to help recreate the missing fundamental frequency. 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 if need be. Finally adjust the Mix to balance out the low and high frequencies as desired. If pumping Rap style bass is required set the Resonance to around 3.0, and increase the Low Cut parameter to about 40Hz. The Bass Maximizer is generally very effective at processing 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 4000Hz. 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. In the case of Rap music, 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".

Tube/Tape Saturation

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. In the majority of instances, tape hiss and high frequency roll-off are undesirable artifacts of the recording and amplification process, yet it is the smooth and warm saturation characteristics of these devices that are often highly desirable. In this sense, the Tube/Tape Saturation plug-in gives you all of these good tube and tape characteristics with none of the bad, 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. Don't be fooled by others, that claim they have analog sounding distortion within the digital domain.

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. Subtlety is key! 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. Certainly the Tube/Tape Saturation will not work well as a level maximizer for many 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

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.

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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

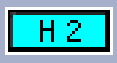
The Feedback slider introduces regeneration, where the output of the pitch time changer is mixed back at it's input. A setting of 0% is no feedback, 100% will lead to run-away regeneration, as the full output of the effect is constantly being routed to it's own input.

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



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

H 2 VST MMP G DAS

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

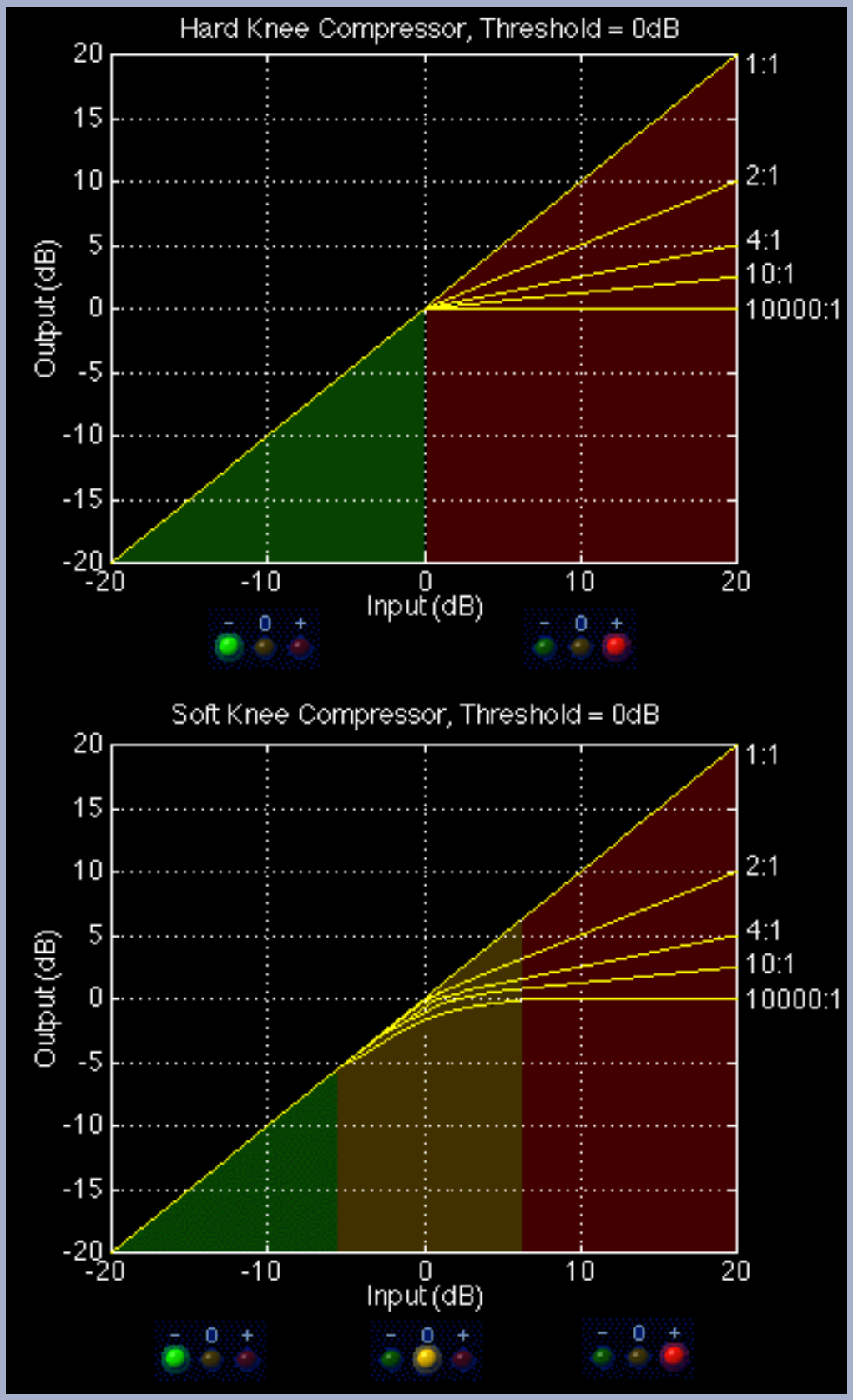
As in any traditional hardware compressor, Hyperprism offers a Ratio setting to control the amount of gain reduction. A ratio of 1:1 means no reduction, and a ratio of 10:1 means a 10 dB increase in the input signal results in only 1 dB increase in the output signal. Possible settings range from 1:1 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 

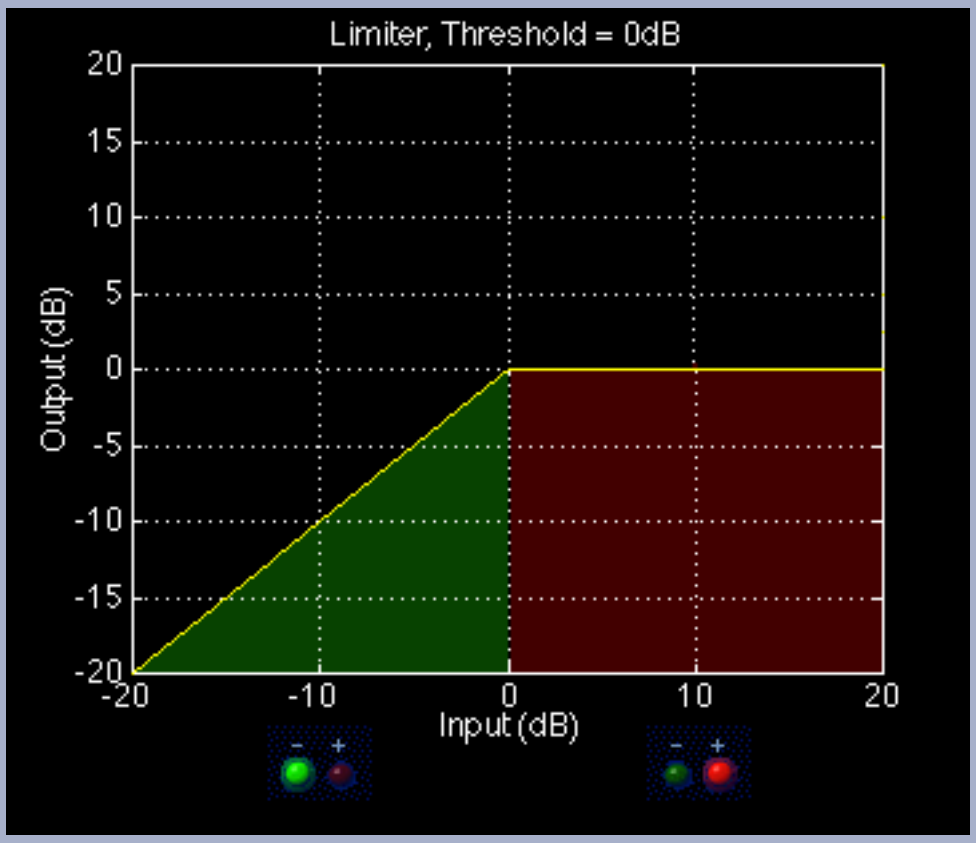
Green light indicates signal is below the threshold, no processing is occurring. Orange light denotes signal in the soft knee region and incremental compression is occurring. Red light means signal above the threshold, full compression value is

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)

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.

Go to Effects: [Filters] [Modulators] [Delay and Reverb] [Stereo] [Misc Effects]
Go to Table of Contents: [Hyperprism 2] [HyperEngine]

[Troubleshooting] [About Digital Audio]

Arboretum SYSTEMS

HyperEngine™ Manual

HyperEngine

Real-Time Signal Processing Platform

What's New In HyperEngine 2.4.4:

- 24 bit file support
- Fix to "long file" bugs

New Features in HyperEngine 2.4:

- Import Split Stereo (.L/.R) files
- Import Foreign Files (Wave, QuickTime and more)
- Import CD audio tracks with supported Apple CD-ROM drives
- New file export features
- Open Transport no longer required. In fact, no system Extensions whatsoever are required under MacOS 8.0 and later!
- Overall improved stability
- More Troubleshooting and Configuration tips

New Features in HyperEngine 2.3:

- Drag-and-Drop Waveform Editing · Freely copy audio segments from one file to another, or within the same file. One-touch drag-and-drop editing makes HyperEngine the fast and easy way to cut sound.
- Live Play-Through Processing · Process input from a DAT player or other source, in real-time, without creating and reading hard disk files. HyperEngine lets you use Ionizer, Ray Gun and Hyperprism like regular hardware effects!
- Record and Process at the same time! Capture those old vinyl recording to digital, and clean them up at the same time with Ray Gun.
- Flexible Selection Range and Viewing Tools · Simple key commands and mouse clicks allow you to navigate, select and edit audio with speed and accuracy.
- Direct Sound Card Support · HyperEngine includes special custom drivers for Apple Sound Manager, Korg 1212 I/O, Digidesign Audiomedia II and Audiomedia III cards. Take advantage of HyperEngine's 32-bit internal processing, allowing you to keep your digital signals at the highest possible resolution throughout the processing chain.

Note that the new chapter on HyperEngine Reference Documents contains a lot of material, and is a real slow-loader. The HyperEngine Command Reference chapter contains synopses of each menu function, and is heavily hyper-linked to the detailed explanations contained in the chapter on Reference Documents. You may wish to start your searches in the Command Reference chapter, then access the Reference Documents chapter when you need detailed step-by-step instructions.

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!

Underlined Text indicates a Hyper link; mouse-click on the link to access more information on the designated topic.

Documentation current for HyperEngine version 2.4.1 last revised 11/9/98. Manual written and edited by Todd Souvignier, with past contributions from Georges Jaroslaw, Chris Weare, Curtis Roads and David (Rudy) Trubitt.

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[First Chapter]

Arboretum SYSTEMS

HyperEngine™ Manual

Introduction

This chapter describes:

About HyperEngine

HyperEngine is a program built for real-time signal processing and sound design which also provides a framework that lets Arboretum's plug-in effects operate independently from any third-party software. HyperEngine allows Ray Gun, Ionizer, Hyperprism, and Harmony to run as if they were self-contained programs.

In addition to providing real-time and disk-based audio input/output services to our plug-ins, HyperEngine offers some significant advantages over other audio host programs, so your Arboretum plug-ins can work at their maximum power:

Using HyperEngine, you can open an existing sound file from disk, record a new sound file to disk, or process real time audio connected to the input of your Mac without creating or reading a disk file, much like any hardware effects device. Regardless of whether the sound is disk-based or real-time, you can use HyperEngine to process the sound using as many simultaneous effects plug-ins as your computer's speed allows.

HyperEngine processes sound non-destructively, which means that your original sound file is never changed. Instead, a new copy of the processed file can be created. However, rather than force you to make multiple copies of your file to hear the effects of different edits or effects, HyperEngine creates small Reference Documents that store effect parameter settings and waveform edits which can be applied to your original sound file. The HyperEngine "Reference" document "remembers" or more accurately, is linked to the sound file you are processing, so that when you open a previously saved Reference document, the correct sound is loaded automatically.You may also create HyperEngine Reference documents for storing effect settings used to process real-time audio connected to your Mac.


Once you find the effect settings you are happy with, create a processed copy of your original file using the Process to New File command, found in the File menu.

This documentation explains how to use HyperEngine, but does not describe the functions of the plug-ins that can be used within HyperEngine. For details on the plug-ins themselves, see the specific documentation related to each effect.

Hardware and Software Requirements

HyperEngine requires a 120 MHz or faster Power Macintosh (or compatible) with a minimum of 16 Megabytes of free RAM, running MacOS 7.6 or later and the Sound control panel version 8.0 or later. If you're using a MacOS version earlier than v8.0 the Sound Manager system extension version 3.2.1 or later will also be required.

In order to really utilize HyperEngine (for more than simple recording, playback and editing) you'll need one or more HyperEngine-compatible plug-in effects, such as Ionizer, Ray Gun, Harmony, or Hyperprism 2 from Arboretum Systems. Note that HyperEngine is included **free** with these products. For more information on hardware/software requirements, check out the compatibility issues section and the troubleshooting chapter.

Getting Sound Into HyperEngine

HyperEngine offers two modes of operation: File-based, where an audio file on your disk is used as HyperEngine's audio input, and real-time "play-through" processing, where external audio devices can be connected to your Mac's audio input and output for real-time effects processing.

HyperEngine with File-based processing

HyperEngine can open sounds created by other audio programs, or you can use HyperEngine to record new audio files, from any audio source connected to your Mac.

HyperEngine will open and save sound files in both of the popular Mac sound formats: Sound Designer II (SDII) and Audio Interchange File Format (AIFF). HyperEngine accepts 8-bit, 16-bit and 24-bit audio files at the 44.1 and 48 kHz sample rates, as well as a number of lower sample rates (based on multiples of 44.1 and 48).

New in HyperEngine version 2.4 is special support for QuickTime 3, which allows HyperEngine to import and export "foreign" file formats such as .Wav, QuickTime and .SND. Installing QuickTime 3 also allows HyperEngine to directly import CD audio tracks. The new "Import Split Stereo" feature creates a new interleaved stereo file out of any .L/.R file pair, for use in HyperEngine.

HyperEngine and Play-through processing

HyperEngine also works with "live" audio connected to the Sound Input of your Mac or audio card. This means you could connect the output of a DAT recorder, microphone or other audio source to your sound input, play the audio through HyperEngine for processing, and send the processed audio to your speakers or an external recording device, all in real-time. With this method, hard disk space is never an issue, because digital audio is never stored on your drive.

By means of our custom sound drivers you can also use professional audio cards to route your signal directly in and out of HyperEngine. We offer direct driver support for the Digidesign Audiomedia II and III cards, the Korg 1212 I/O card as well as the Apple Sound Manager.

Getting Sound Out of HyperEngine

HyperEngine lets you store many different edits and processing settings for the same sound. Since these variations contain only the edit decisions and effect settings, they take up very little disk space. But whether you are working on a music CD, sound-for-picture, multimedia or other project, you'll eventually want to create new, processed sound files from HyperEngine to incorporate in your finished work.

To do so, first open the HyperEngine Reference document you want to "export." Make sure the audio you wish to export is selected in the waveform display and the effects you wish to hear are turned on in the HyperEngine Process List.

Then, select HyperEngine's Process to New File command (located in the File menu). This command creates a brand new sound file based on the settings in the Reference document. Once this file is created, you can import it virtually any other audio or multimedia software application.

Depending on the speed of your Power Mac, you will probably be able to run several effects processes within HyperEngine at once. And, even when you have reached the real-time performance limit of your computer, you can still add additional effects to a HyperEngine Reference document. While you won't be able to preview all these effects simultaneously, they all can be applied when exporting the audio with the Process to New File command.

Alternately, you could apply a large number of effects in two or more passes, exporting the intermediate stages to disc and then applying further processing to these new sound files.

This concludes the Introduction chapter; next we'll learn how to install and configure HyperEngine, then take a look at each of its features.

[Table of Contents] [Next Chapter]

Arboretum SYSTEMS

HyperEngine™ Manual

Installing HyperEngine

Installing HyperEngine and HyperEngine plug-ins

HyperEngine requires no special installation and is not copy protected. HyperEngine plug-ins (Ray Gun, Ionizer, Harmony, Hyperprism 2) however **are** copy protected.

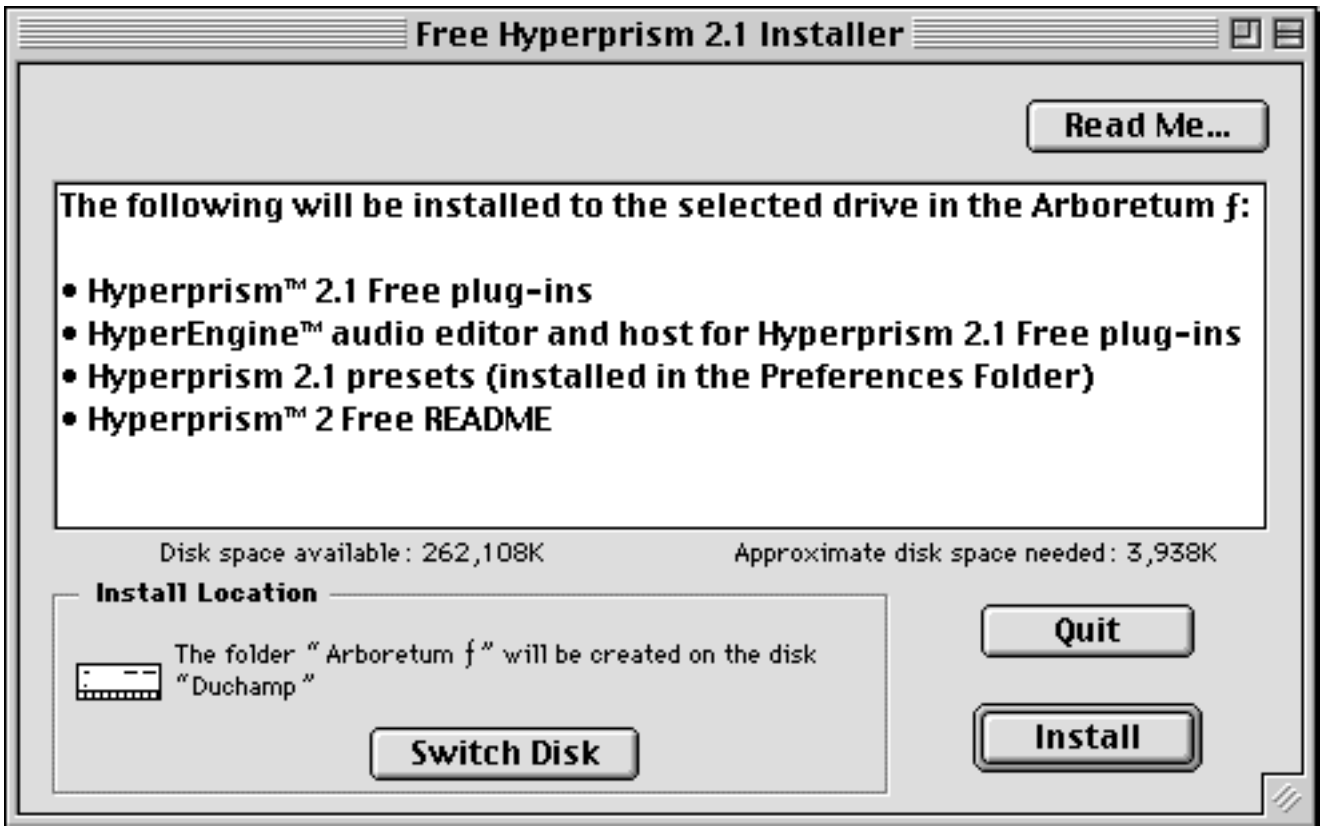
Please read this section once through entirely and then follow these steps exactly!

If you are an owner of Hyperprism 2, Ray Gun, Harmony or Ionizer, see the special installation instructions for those programs. For your convenience HyperEngine is automatically installed when doing the "Easy" installation for any of these products.

HyperEngine is also given away freely as "shareware," and the giveaway includes a few free Hyperprism 2 effects (as a little taste of what you'll get in Hyperprism 2). The following directions document installation of the Shareware HyperEngine.

- Insert the Arboretum CD in your CD-ROM Drive.
- Double click on the folder labeled "Free Software."
- Once inside the "Free Software" folder, double-click on the "Free Hyperprism 2.1" Installer.
- Mouse-click past the Arboretum Logo display, click on Accept to accept the terms of the software license agreement, then after reading/printing the subsequent ReadMe display, click on OK to proceed to the installation.

You're now at the main installation screen:



The Hyperprism 2.1 Free installer offers only the "Easy Install" option.

Check the Install Location display in the lower part of this Install dialog. Install HyperEngine and the free Hyperprism 2.1 plug-ins onto any local hard drive. Click on Switch Disk if you need to specify another drive for installation.

Click on the Install button.

Installation will begin. Click OK through any on-screen prompts to continue or conclude.

Your HyperEngine installation is now finished!

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.

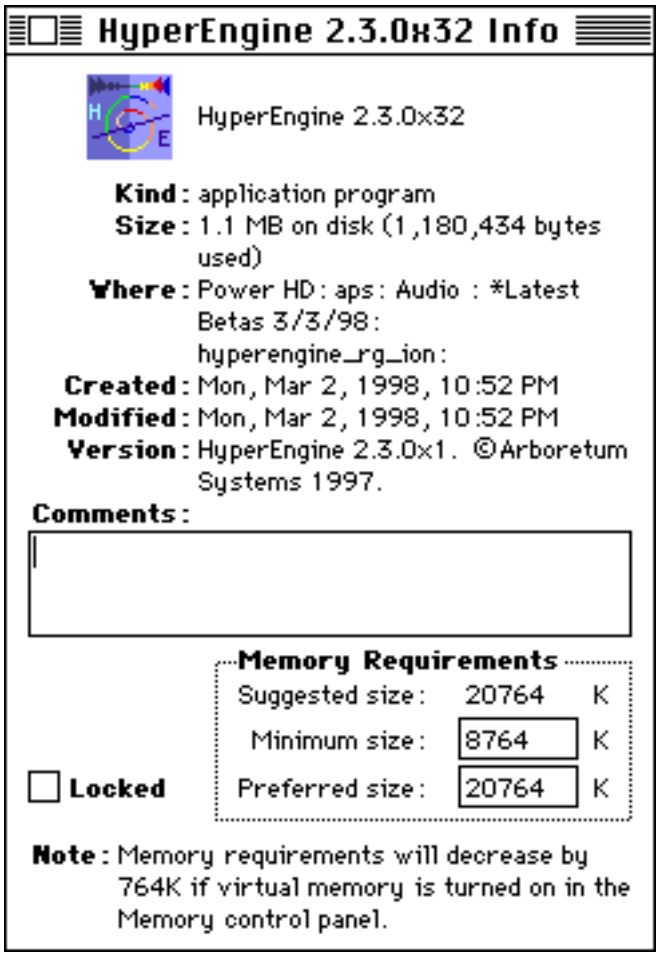
Allocating Memory to HyperEngine

As with any real-time digital audio application, HyperEngine will perform best with adequate memory. The processing speed of your Macintosh will determine how many effects HyperEngine will be able to run at once. To adjust the amount of memory (RAM) you are allocating to the HyperEngine, locate the HyperEngine application's icon.

The HyperEngine application's icon

Mouse-click once on this icon, so that it is highlighted (don't double click, or you'll launch the program.)

Next, type command-i or select Get Info from the MacOS File menu.

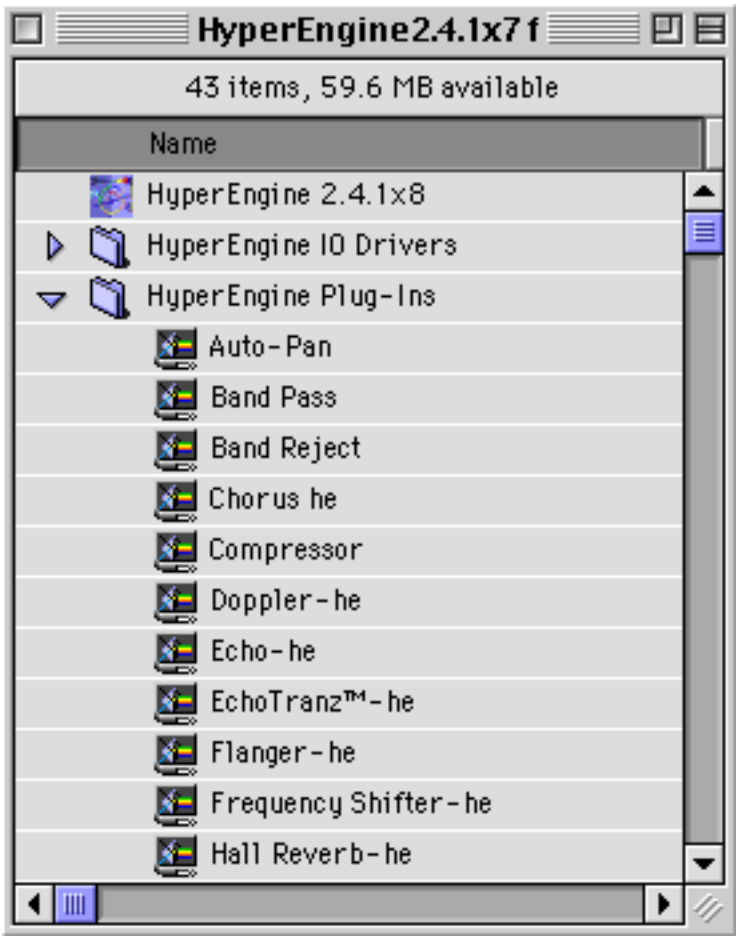


The HyperEngine's Get Info dialog. Note Memory Requirements settings, lower R.H. corner

The Get Info dialog allows you to set the minimum and preferred memory amounts for the program. If you have a limited amount of memory, you may wish to decrease the preferred setting. If you have plenty of RAM, feel free to allocate more to the HyperEngine. Ensuring that you have plenty of memory is key to smooth performance. If you're encountering repeated crashes and freezes, the most likely cause is simply running out of program memory.

Locating HyperEngine Effects

HyperEngine places its effects plug-ins in a folder called "HyperEngine Plug-Ins." This folder must be in the same folder as the HyperEngine application, as illustrated below:



The HyperEngine Plug-in folder and the HyperEngine IO Drivers folder must reside at the same directory level (within the same folder) as the HyperEngine application. Hyperprism, Ionizer and Ray Gun -HE plug-ins are automatically installed into the HyperEngine Plug-in folder . IO Drivers simply remain in the IO Drivers folder, as installed.

This is the default configuration upon installation, and assuming you haven't moved these components around, you won't need to worry about it further. HyperEngine also needs to see its IO Drivers folder, read the next section for details. See the documentation on your individual plug-ins for more information on their installation and hard-disk authorization.

Installing HyperEngine Drivers

HyperEngine includes special custom drivers for the Digidesign Audiomedia II and Audiomedia III cards, plus the Korg 1212I/O card and the Apple Sound Manager. HyperEngine looks for its drivers in a folder called "HyperEngine IO Drivers." This folder must be in the same folder, at the same directory level, as the HyperEngine application (as illustrated above.) This is the default configuration upon installation, and assuming you haven't moved these components around, you won't need to worry about it further.

If you have a supported sound card, these custom drivers eliminate the need for using special Sound Manager drivers, such as the Digidesign Sound Drivers. Select the desired driver and sound hardware in HyperEngine's Preference menu. Use the Configure Audio System dialog to set up the hardware.

System Compatibility

HyperEngine requires a 120 MHz or faster Power Macintosh (or compatible) with a minimum of 16 Megabytes of free RAM, running MacOS 7.6 or later and the Sound control panel version 8.0 or later. If you're using a MacOS version earlier than v8.0 the Sound Manager system extension version 3.2.1 or later will also be required.

To exploit new CD audio import and foreign file import/export features QuickTime 3 is required. This can be downloaded free of charge from Apple's web site.

HyperEngine is officially incompatible with RAM Doubler and Virtual Memory, but several thrill-seekers we know routinely run with Virtual Memory enabled and report few problems.

HyperEngine is compatible with most other known system extensions ("Inits") and control panels.

Certain versions of HyperEngine (2.3 and earlier) required the Open Transport extension set for operation; this is no longer required in HE 2.4. In fact, no system Extensions whatsoever are required by HyperEngine 2.4 under MacOS 8.0 and later.

Be sure to check out Appendix A-Troubleshooting for more information on optimizing performance and eliminating system conflicts. Also take a look at the chapter on HyperEngine and the Sound Manager for information about hardware compatibility.

Installation Support Notes

HyperEngine requires a minimum of 16 Megabytes of free RAM. If you're running out of memory you'll likely see a "Not Enough Memory" or type -2 error message. HyperEngine has a default preferred setting of 16 MB, which should be adequate for opening multiple audio files and several effect windows simultaneously.

The Arboretum installer will create a new folder on your hard drive called "Arboretum f" and will then install the HyperEngine, its drivers and it's Plug-ins folder within that folder. If you're installing new Arboretum plug-ins, such as updates you've downloaded from our web site, you'll have to drag your new plug-in into HyperEngine's plug-ins folder **by hand**.

Problems? If any unexpected error messages come up during the course of installation, or if the Installer fails to place the HyperEngine and associated files into your Arboretum f 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/Ray Gun/Ionizer/HyperEngine Preferences file(s) 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

Starting HyperEngine

To begin using HyperEngine, go into the new Arboretum f folder on your hard drive, open the folder and double-click on the HyperEngine icon:

The HyperEngine application's icon

HyperEngine will launch. After the splash screen you'll see the HyperEngine play back window appear on screen.

To load an audio file for processing, type Command-n (or select New File Document from the HyperEngine File menu).

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

Monitoring and Playback Controls

When HyperEngine is launched, you will see HyperEngine's Play Back window appear at the left of your screen. It's used to control HyperEngine's monitoring, metering and playback.

HyperEngine Play Back window

☞The Play Back window may be closed by clicking the small window close box in it's upper left corner. If you close it, and then want to re-open it later, choose the Play Back item from the HyperEngine "Windows" menu.

Output Toggles

The top two buttons route HyperEngine's audio output to your computer speakers or monitoring system. Click on either button to turn on/off the right and/or left channels.

Note that in the case of mono (one-channel) source sound files, the setting of the left channel will apply to both left and right channel playback. Note also that mono sounds are "panned center" on playback, so they're heard equally through both the left and right speakers.

Transport Controls

The four main buttons in the Play Back window let you control the playback of your sound file, much like a traditional tape transport.

Play

Clicking on the Play button begins sound file playback from the beginning of the area selected in the waveform overview when working with a File Document. The Play button is also used to enable sound throughput when working with a HyperEngine Play-Thru Document. The space bar is a keyboard shortcut for Play. Note that the space bar is also the Pause shortcut.

Playback normally begins at the start of the current selection; however, if you reposition the cursor within the selection, playback will begin at the new cursor location.

Stop

The Stop button stops HyperEngine playback and recording. When the Stop button is pressed the playback cursor returns to the beginning of the current selection, or to the beginning of the file if there is no selection currently made.The Return key is a shortcut for the Stop button (think of it as "Stop and Return to beginning.")

Pause

The Pause button suspends playback, and allows playback to resume from the current cursor position in the waveform overview. The space bar is a keyboard shortcut for the Pause button.

Loop

The Loop button causes HyperEngine to play back repeatedly, returning instantly to the start of the file (or current selection) once the end is reached. If no selection is made, HyperEngine will Loop playback of the entire file.

Meters and Faders

HyperEngine's Play Back window is dominated by the meter and fader assembly. HyperEngine's meters show you the current signal level (or "volume.") HyperEngine's faders allow you to control the volume levels for recording and playback.

The left and right sides of the meters reflect energy in the left and right channels. Mono sounds are panned center by HyperEngine, so both meters are active when playing mono files.

☞ The meters reflect output volume when HyperEngine is playing back or processing a disk file. The meters reflect input volume when HyperEngine is recording, or when HyperEngine is processing real-time audio with a Play-Thru document.

At the top of the scale, red "LED" clip lights illuminate when one or more samples have been clipped.

Clip lights and Clip button

Note that there is no consensus on how may consecutively clipped samples constitute "a problem;" many devices won't show clipping of a single sample. However, HyperEngine indicates a single clipped sample. The red clip lights remain lit even if playback is interrupted by clicking on Pause or Stop. To turn the clip lights off, click on the Clip button at the top of the meters (between the clip lights).

☞ Because some effects can increase the level of your original sounds, it is possible to introduce unintended digital distortion during processing. If this happens, you can use the faders to reduce the overall level. There's more tips on reducing distortion in the Troubleshooting chapter.

The faders are graphic potentiometers that control the gain (volume level) of input and output signals. They are located in between the meters. When HyperEngine is playing back or processing a disk file, the meters control HyperEngine's output. When HyperEngine is recording audio or processing "live" sound with a Play-Thru document, the meters control input, the level of signal being routed into HyperEngine.

The faders consist of two discrete sliders (left and right) graduated from -(infinite) dB to +18 dB. These can be used independently or grouped together. The faders are aligned to 0 dB by default. Mouse-click and drag on either fader knob to adjust input or output volume. To move both faders simultaneously, click the knob in the middle and drag the mouse up or

down. To bring left and right channels to the same level, simply click the center fader control and the left and right faders will both jump to that level.

Click on the Lock button to tie the left and right faders together,allowing you to move them jointly while preserving their relative settings. Click on the 0 dB button to automatically reset both faders to the default 0 dB setting.

The 0 dB and Lock buttons

You can change the fader levels during recording and playback. If you want to change the gain, be prudent when increasing the level, as you might overload your ears and loudspeakers. If you saturate the digital-to-analog converter hardware, the red clip lights above the peak meters light up. They remain lit until you explicitly turn them off with the Clip button at the top of the meters.

Time Display

The Time Display shows run time as a sound is recorded or played back. It displays hours, minutes, seconds and hundredths of a second. The Time Display may not show an accurate reading when processing with a time stretch or Vari-speed effect.

The Time Display Window

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

HyperEngine Reference Documents

This chapter describes:

Introduction: HyperEngine File Documents and Play-Thru Documents

HyperEngine is a non-destructive sound editor, meaning edits and effects made to your audio never alter the original files. In order to provide this type of functionality, HyperEngine has a "document-based" architecture. All edits, effects, parameter automations and other actions are saved in a HyperEngine Reference Document. The reference document is separate and distinct from your sound files on disk.

HyperEngine uses two types of Reference Documents: the Play-Thru Document and the File Document. Play-Thru

Documents are used when processing live audio in real-time, and when recording new sound files to disk from a live source. File Documents are used when processing or editing audio files which already exist on disk.

It's important to understand the distinction between the Reference Document and the sound file. A sound file contains the actual audio. A Reference Document points to a sound file (referencing its original location, on disk or in playthrough) and contains editing and processing data related to the sound file, but no sound samples. Sound files can be huge, hundreds of Megabytes for longer recordings. Reference Documents are comparatively small, typically below one hundred Kbytes.

You can create many different Reference Documents for a particular sound file (or play-through track) so as to try out different variations and combinations. You can also reassign any File Reference Document to any sound file you wish, so as to apply one set of effects to several different files.

Reference Documents never change the original sound file, nor do they automatically create additional audio files. In order to make a new sound file, you either use the Process to New File command (when working with a File Document) or the Record function (when working with a Play-Thru Document). Naturally, any active effects (or file edits) will be rendered to the new file you're creating.

When you open a new sound file, the Reference Document you've created is given a default name, consisting of the name of the current audio file plus the suffix "-FX." Don't confuse this "-FX" File Reference Document with the actual sound file, which retains its original name and data under all circumstances.

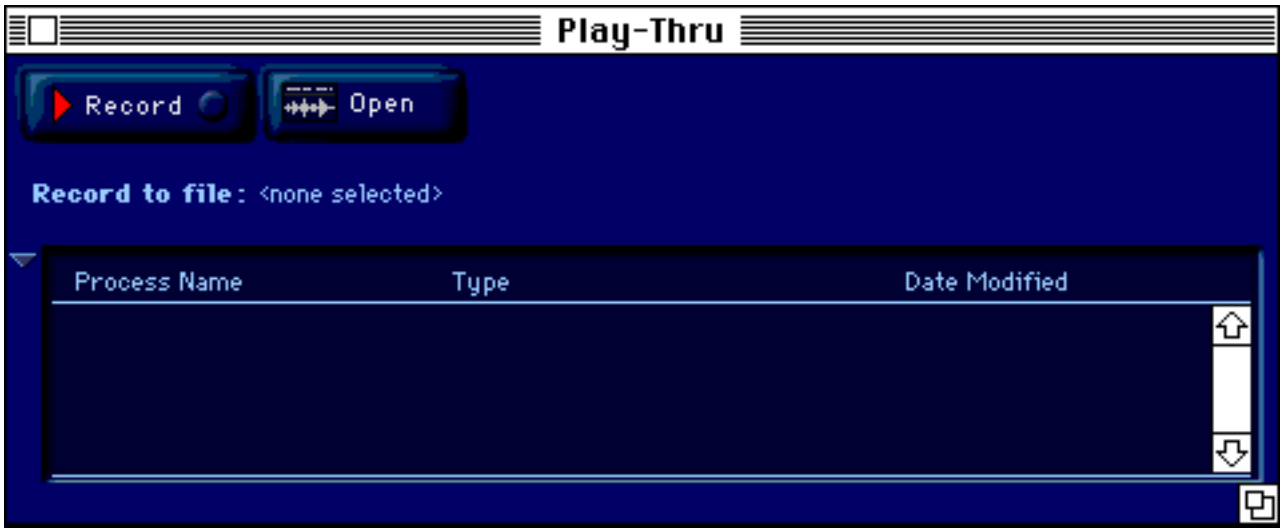
The remainder of this chapter will teach you to process live signals using a Play-Thru Document, how to record a new audio file, and how to work with File Documents in order to manipulate and process sound on disk.

Processing Real-Time Audio: The Play-Thru Document

HyperEngine can process audio appearing at your Mac's Sound input device in real-time. This will allow you to use HyperEngine much like a hardware processor, and frees you from reading and creating new sound files on disk. This functionality makes it possible to use HyperEngine in a live performance setting, on stage or in the studio. Use the Mac's microphone port, the inputs on your sound card, or the AppleCD Audio Player desk accessory to get live signal into HyperEngine.

Connect the audio inputs and outputs of your Mac to the rest of your audio system. Check your Sound Control Panel input options (from the Control Panel submenu in the Apple Menu) if necessary.

Chose New Play-Thru Document from the File menu. A HyperEngine Play-Thru document will appear on-screen:



a HyperEngine Play-Thru Document, with no active processes

HyperEngine Play-Thru Documents allow sound to be played through the HyperEngine from any external source, allow you to insert effects into the signal stream, and let you record incoming sound.

Begin playing audio from your DAT machine, tape deck, CD player, mic or other external sound source into your Mac's input device.

Hit the Play button or press the spacebar to begin HyperEngine play-thru

You should now hear your live sound being played through. If not, check your connections.

Adjust input volume using the HyperEngine's Faders. When your levels are OK, click on the Stop button (or press the Return key) to suspend play-thru and give you access to the Plug-ins menu.

Now select an effect from the Plug-ins menu.

The selected effect's interface will appear on-screen. (If you get the Serial Number dialog, you need to serialize this plug-in. Review the Installation chapter for tips on serializing and setup.)

HyperEngine Play-Thru document with three effects added to the Process List

Effects selected for a Play-Thru document are added to the Process List, in the lower half of the reference document window. This list displays the process name (untitled by default) along with process type and date. Note also the Bypass toggle switches along the left hand side of this display, effects are toggled to "active" by default when first added to a document. For more information on using and managing the Process List, see the section below.

Hit the Play button (or spacebar) again to resume play-thru and begin processing your sound in real-time. Adjust Ray Gun, Ionizer or Hyperprism settings and you'll hear the changes made instantly to your live signal.

If you're hearing an undesirable delayed (echo) effect when doing HyperEngine play-thru, it's the result of having Sound Manager playthrough enabled. To solve this, go into the HyperEngine Preference Menu and select Configure Audio System from under the Hardware submenu. In this dialog, de-select (check box off) the Playthrough option. Configuring your system this way will allow you to hear HyperEngine play-thru without the additional unwanted Sound Manager playthrough.

Bear in mind that there's a built-in degree of latency (time lag) to all computer audio systems, and doing live effects

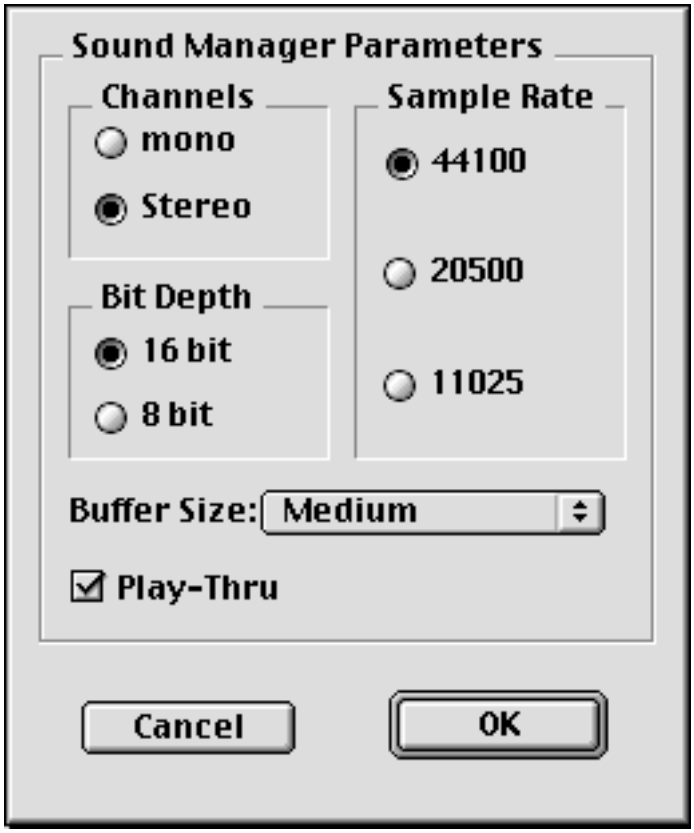
processing is one application where you can really notice the inherent latency of your audio card or the Sound Manager. Depending on the type of audio you're processing this latency may or may not present a problem in live performance situations. Going with a smaller buffer size in the Configure Audio System dialog can make a big difference, but there's no way to completely eliminate latency.

Recording a New Audio File

Once you have created a Play-Thru Document (see section above for details) you can record incoming live audio to a disk file for subsequent processing, editing or archiving.

Before you begin recording new files, you should specify the type of file you want to acquire. This is done from the Preference menu.

Go to the Preference menu, mouse drag into the Hardware submenu and select Configure Audio System. You'll be presented with the settings dialog:



Configure Audio System (settings) dialog as it appears for the Sound Manager drivers; AMII, AMIII and 1212I/O dialogs vary somewhat.

Select the bit depth, sample rate and number of channels (mono=1/stereo=2) for the file you're about to record. When you're happy with your selections, click on OK to exit this dialog.

Don't understand these values? Check out the chapter on digital audio files.

You don't need to configure the settings each time you record a new file, as HyperEngine will remember your current settings, but it's a good idea to check the settings before beginning a session, or any time you want to change your file type.

Make sure your Sound control panel settings for bit depth, sample rate and mono/stereo match your HyperEngine settings; a mismatch will usually create massive digital noise during playthrough. If you only have the Monitors and Sound control panel in your system, we suggest adding the Sound control panel so that you get direct control over these settings.

If you're recording audio for eventual use on a CD, we recommend that you always create interleaved stereo 16-bit Sound Designer II 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 Toast and MasterList CD accept for burning audio CD-Rs. Sticking with these settings will prevent file-conversion headaches down the road.

Now that your audio system is configured to acquire the right type of audio file, you can go ahead and set up HyperEngine for recording.

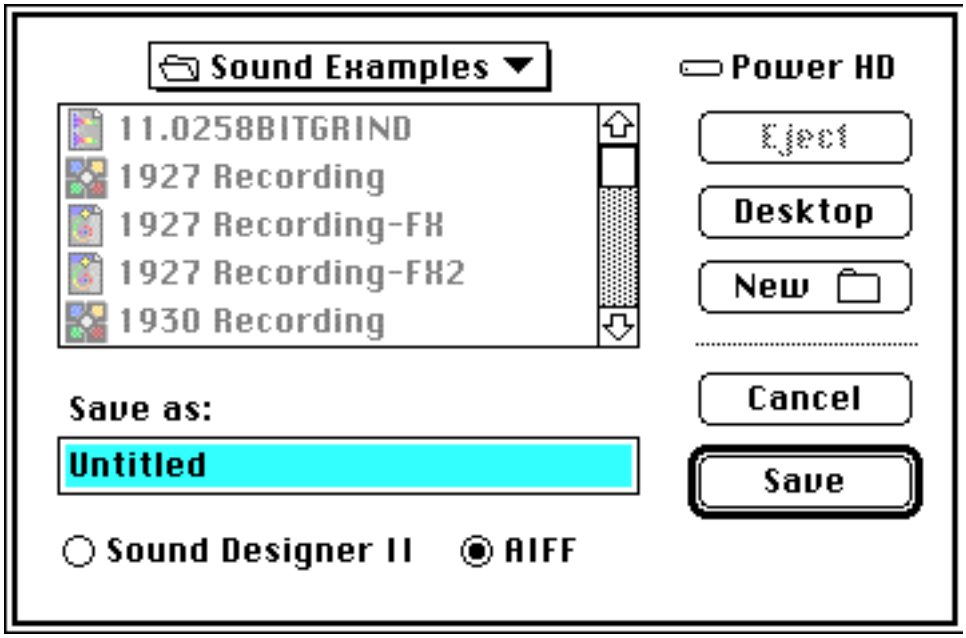
- Create a new HyperEngine Play-Thru Document from the File menu, as described above.

- Begin playing audio from your DAT machine, tape deck, CD player, mic or other external sound source into your Mac's sound input device (your sound card's inputs or your Mac's mic input).

- Hit the HyperEngine Play button; you should now hear your live sound being played through. If not, check your connections.

- Adjust input volume using the HyperEngine's Faders.

- When you have good levels (strong signal with no clipping) stop and rewind your external source to the beginning of the material you wish to record. Then click on the Play-Thru Document's Record button. You'll now see the file capture dialog:



HyperEngine file capture dialog

- Specify the audio file format (SDII or AIFF), a unique name and target location on hard disk for the sound file you're going to create. When you're done, click on the Save button.

- Exiting the capture dialog returns you to the HyperEngine; note that the Record button's red light is now "on," your new file name is listed in the "Record to file:" display and the reference document indicates the total number of minutes of audio that you can record to the target volume.

Recording enabled: The red Record light is on, available recording time (free disk space) is displayed and the target file name is shown

☞ If you don't have enough empty hard disk space to record all of your source audio, either backup and delete any unused files from your hard drive to free up additional space or use a target volume that has adequate room. Remember that stereo 16-bit audio sampled at 44.1 kHz takes up about 10 MB per track minute. That means you'll need about 200 MB of free disk space to record a CD-quality 20-minute stereo file (the typical length of an album side).

➠ Click on the HyperEngine's Play button to begin recording, and start playback "from the top" at your external source.

➘ You are now creating a sound file on hard disk, a digital recording of the audio being played in to your Mac's sound input device.

☞ If you adjust the HyperEngine faders during recording, you'll alter the level of the recording being made. Remember that the faders control input level during recording and playthrough, and output level during playback.

➠ When you've reached the end of your recording, click on the Stop button to discontinue file acquisition.

➠ If you wish to add additional audio to the current disk file, simply click on the Play button again. Recording will start again, and the new audio will be appended to the earlier recording.

➠ If you want to create another disk file, simply close the current Play-Thru document, then select the New Play-Thru Document option from the File menu, and repeat the recording steps listed above.

☞ Since you can only play one audio stream at a time through your Mac, it's a good idea to only have one Play-Thru document open at a time. Avoid having multiple Play-Thru documents open simultaneously; it'll help keep both you and HyperEngine from getting confused.

➠ If you want to begin editing and adding effects to the audio file you've just created, click on the Open button and a File reference document will appear, showing the waveform overview of your new file. Read on for more information on working with File documents.

☞ If your recorded file sounds like it's stuttering or kind of cuts out, try increasing the Buffer size in the Configure Audio System dialog. If your recording contains nothing but massive digital noise, make sure your Sound control panel settings for bit depth, sample rate and mono/stereo match your HyperEngine settings. Other tips on recording can be found over in the Troubleshooting section.

Processing While Recording

One of the most powerful features of HyperEngine is its ability to do signal processing in real-time, while a file is being recorded. This time-saving feature enables you to do things like pitch shift a singer or de-click and denoise an old

recording on-the-fly, creating a new processed file simultaneously. To do this, simply add plug-ins such as Ray Gun, Ionizer or Hyperprism effects from the Plug-ins menu to the Play-Thru document's process list.

Open a new HyperEngine Play-Thru Document from the File menu, as described above.

Now select an effect from the Plug-ins menu.

The selected effect's interface will appear on-screen. (If you get the Serial Number dialog, you need to serialize this plug-in. Make sure you have retained your Arboretum Serial Number card. Review the Installation chapter for tips on serializing and setup.)

Effects selected for a Play-Thru document are added to the Process List, in the lower half of the reference document window:

HyperEngine Play-Thru document with three effects active in the Process List

The Process List displays the process name (untitled by default) along with process type and date. Note also the Bypass toggle switches on the left hand side of this display, effects are toggled to "active" by default when first added to a document. For more information on using and managing the Process List, see the section below.

Begin playing signal from your external audio source (tape deck, mic, whatever) and hit the HyperEngine Play button; you should now hear your live sound being played through. If not, check your connections.

Alter your Ray Gun, Ionizer or Hyperprism effect parameters while listening to the signal being processed, until you have satisfactory settings.

Adjust input volume if necessary, using the HyperEngine's Faders.

When you have good levels (strong signal with no clipping) and you like the sound of your effect settings, stop and rewind your external source to the beginning of the material you wish to record. Then click on the Play-Thru Document's Record button.

Use the file capture dialog to define file type, name and location (as detailed in the previous section). Click on the Save button to exit the dialog and engage the recording function.

Click on the HyperEngine's Play button to begin recording, and start playback "from the top" at your external source.

You are now creating a disk file, a digital recording of the audio being played in to your Mac's sound input device, as it's being treated by your current effect selections.

If you adjust Ray Gun, Ionizer or Hyperprism effect parameters while the signal is being recorded, your effect changes will be heard in the resultant file.

When you've reached the end of your recording, click on the Stop button to discontinue file acquisition.

If your recorded file sounds like it's stuttering or kind of cuts out, try increasing the Buffer size in the Configure Audio System dialog. If your recording contains nothing but massive digital noise, make sure your Sound control panel settings for bit depth, sample rate and mono/stereo match your HyperEngine settings. Other tips on recording can be found over in the Troubleshooting section.

Now that you understand how to process and record audio in real-time, let's examine how HyperEngine deals with audio files which are on disk. The next section discusses the HyperEngine file document.

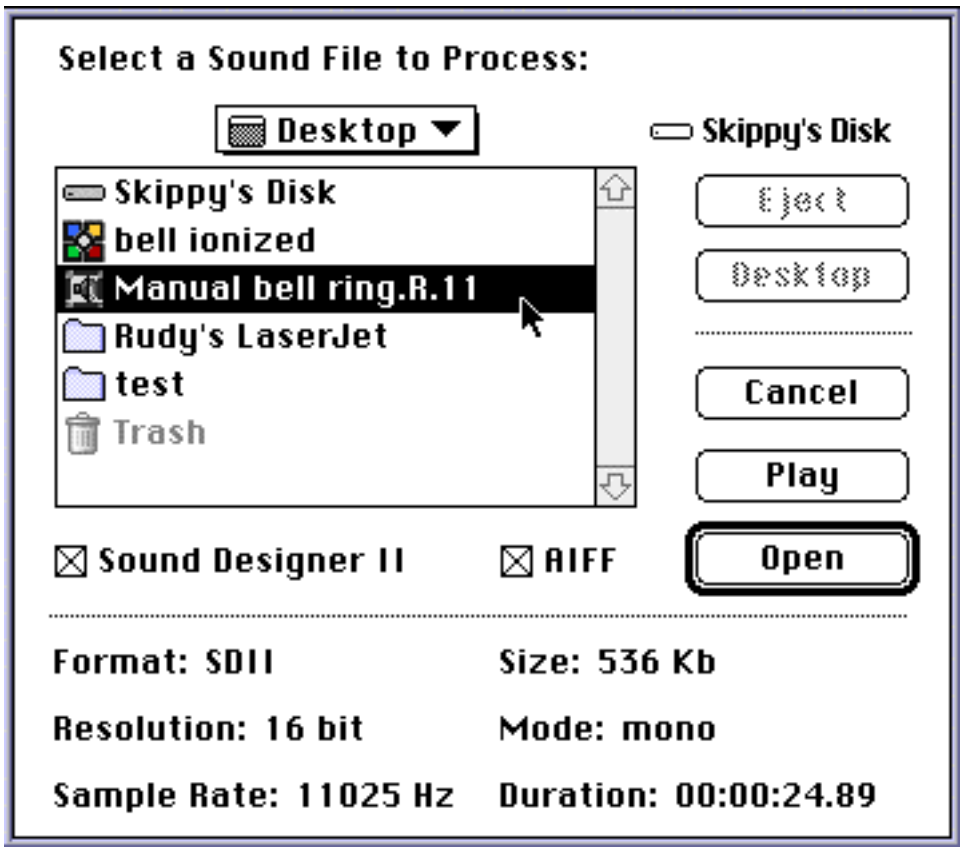
Processing Disk Files: The File Document

Processing disk files with HyperEngine requires the creation of a File document. This example shows how to create a File document to process an existing sound file stored on your hard drive.

Go to the File Menu

Choose New File Document, or simply use the keyboard shortcut, command-n.

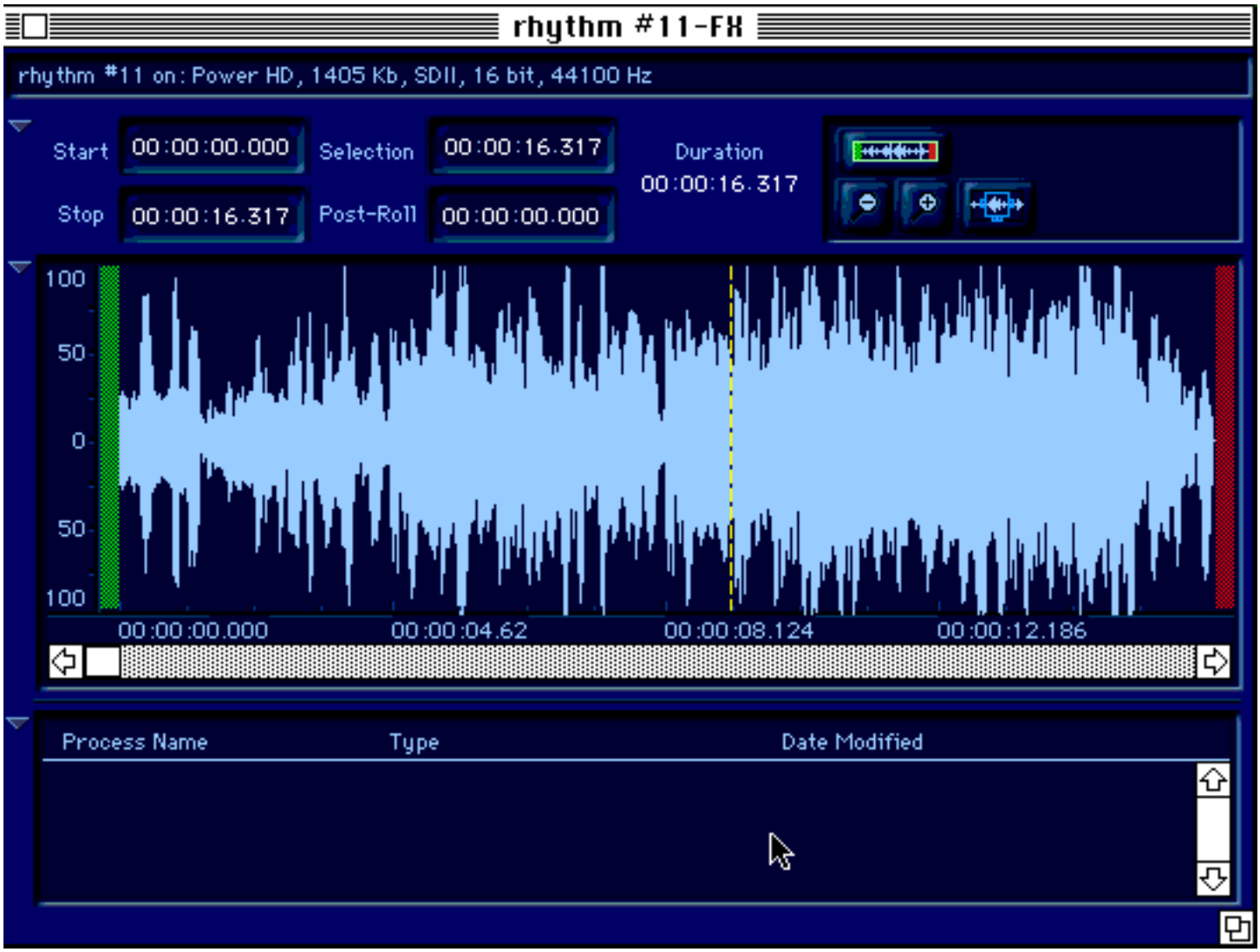
A Dialog box will appear, asking you to select a sound file to work with:



select a sound file from your hard drive for processing

Choose a sound file and click Open (or press either the Return or Enter keys, which are the shortcuts for clicking Open).

A New File Document reference window will appear, along with a graphic waveform overview showing the audio in the sound file you just selected. Depending on the length of the sound file, the overview may take a few moments to calculate.

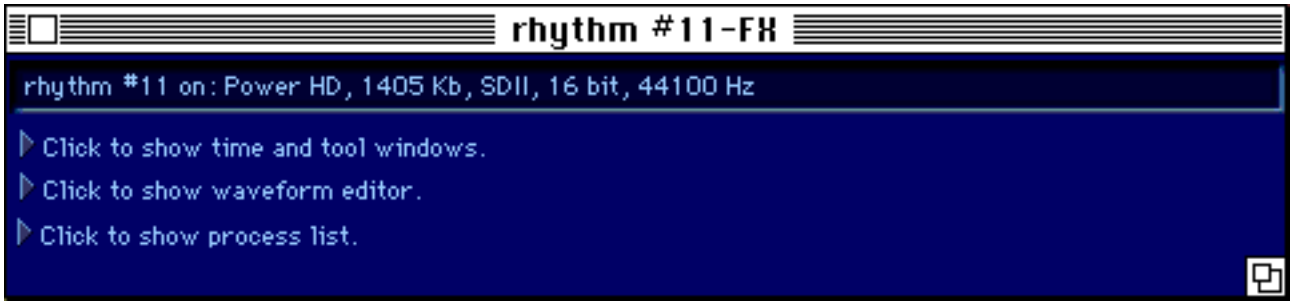


the File Document reference window, with waveform overview displayed

You have just created a new File Document. Note that by default, it bears the name of your sound file, plus the filename extension "-FX" which helps differentiate it from your original sound file.

Note that you can only create one File Document per sound file at a time. Opening simultaneous multiple File Documents for a single sound file will result in unpredictable performance.

Try clicking on the triangle show/hide buttons located along the left-hand margin of the File Document window. These allow you to hide or show the main elements of the document.



File Document window, collapsed view. Click on the triangle buttons to show or hide elements of the display.

Naturally, all elements of the File Document, such as effect processes and selections, remain active even when you've hidden them from view. Also note the "grow tool" in the lower right hand corner of the File Document window. This resizing tab allows you to make the HyperEngine File Document display as large or as small as you need, to match your particular editing needs and screen size.

Elements of a HyperEngine File Document

The File Document has four main sections, which are detailed below: The Header Information, the Time and Tool windows, the Waveform Editor and the Process List.

File Document Header Information

The top of the File document window is a title bar which by default specifies the name of the original input sound file appended by the suffix "-FX." This helps you differentiate between a HyperEngine File (-FX) document and the original sound file. Below the title bar is a text field containing the following general information about the sound file:

Sound file: Name of the sound file linked to this HyperEngine document, followed by:

on: Name of the disk on which the file resides

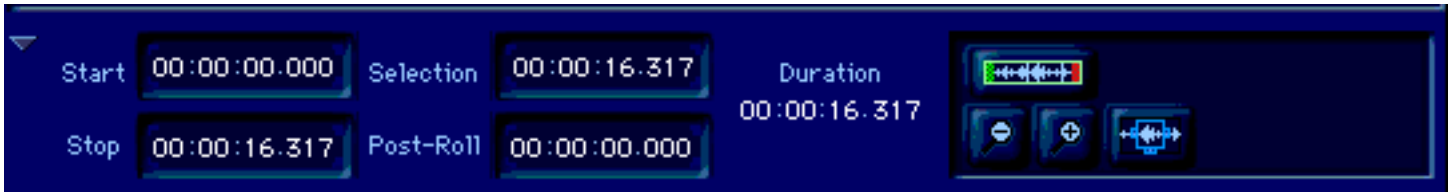
Size on disk: in Kbytes

File Format: AIFF or SDII (Other "foreign" file formats are automatically converted to AIFF upon importing them to HyperEngine 2.4 when QuickTime 3 is installed.)

Resolution: Number of bits per sample (8, 16 or 24)

Sample Rate: 11025, 11127, 22050, 22255, 44100 or 48000 Hz

Time and Tool Windows



Beneath the document header are the Time and Tool Windows, which provide information about the current file or selection, and allow you access to the viewing and navigation tools. The Time and Tool display is collapsible; use the triangle button on the left hand side to hide or show this portion of the File Document.

Time Fields

Start: The start time of the file or the current selection, in hours, minutes, seconds and milliseconds.

Stop: The end time of the file or the current selection, in hours, minutes, seconds and milliseconds.

Selection: The time duration of the file or the current selection, in hours, minutes, seconds and milliseconds.

Post-Roll: This user-definable field adds a set amount of time to the end of any processed file or processed selection. Adding Post-Roll is useful when adding reverb to a file, which creates a "tail" that extends beyond the original end point of the file or selection. Also consider using post roll with any effect that has a feedback parameter, which would tend to make a processed sound longer than it originally was.

Duration: The time duration of the file or the current selection, plus any applied Post-Roll time, in hours, minutes, seconds and milliseconds.

Start, Stop and Selection values can be adjusted with a simple mouse gesture, by clicking and dragging through the waveform overview to make a new selection. Alternately, you can hand-enter numeric values into these fields, and the Post-Roll field, by clicking on any numeric field and typing in a new value from the keypad. You can also adjust Stop, Start, Selection and Post-Roll values by mouse dragging up or down from the fields, to scroll to a new setting.

The Duration value, because it is calculated from the sum of the Selection and Post-Roll values, can not be directly altered except by changing Selection and/or Post-Roll times.

Tool Window

The Tool window contains implements which can assist you with viewing, navigating and editing audio.

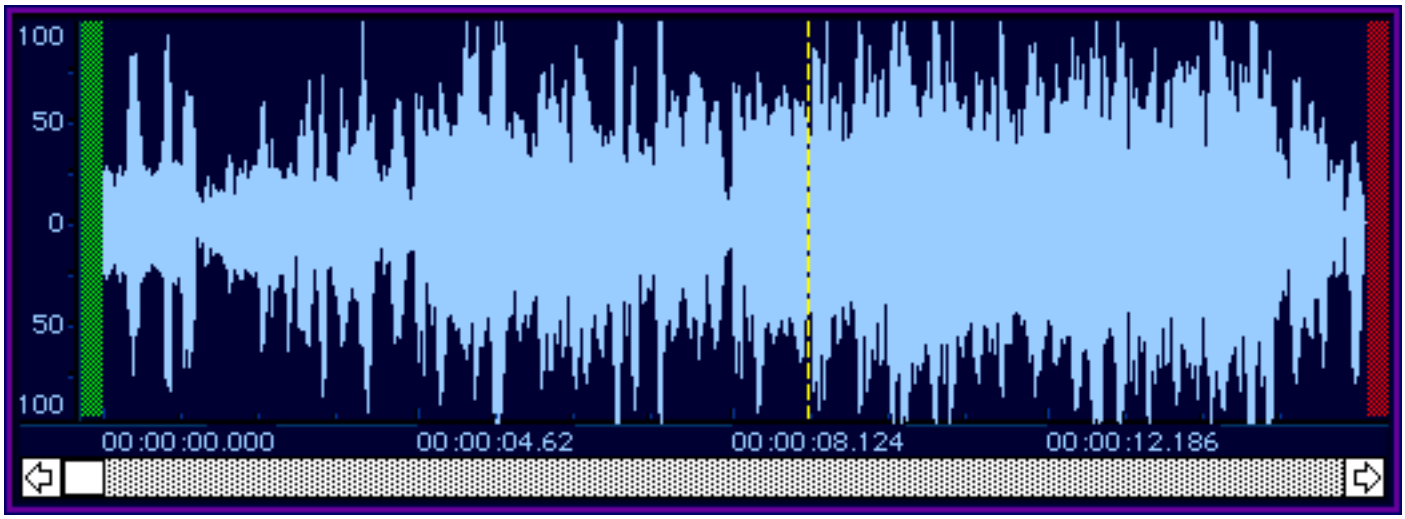
 **View Toggle:** This button lets you switch between the "full view" and the zoomed-in view. The zoomed view is represented and controlled by the View Frame, detailed in the next section. This button is duplicated by the Switch to Zoom-View/Switch to Over-View command in HyperEngine's Wave menu. Also make sure to try out the keyboard alternative, command-/ (forward-slash).

 **Power User Tip:** Leave the View Toggle "On" all the time, so you're always in "zoomed" view, then just navigate and zoom around the waveform display using the arrow keys.

 **Zoom In/Zoom Out:** Gives you incremental control over the resolution of the zoomed view (and the size of the View Frame). The functions of these buttons are duplicated by the hitting the up arrow and down arrows on your keypad, as well as by using the command-- (hyphen/minus) and command-= (equals) keyboard alternatives. There's also Zoom In/Out options in the HyperEngine Wave menu.

 **Hide/Show View Frame:** The light blue View Frame, found in the Waveform Editor display is a navigating and editing tool which allows you to control the area of the zoomed view. The Hide/Show View Frame button toggles the View Frame on and off in the Editor display. If you're currently inside a zoomed view (looking inside the frame, as it were) this button is disabled. For more on the View Frame, read on.

The Waveform Editor



HyperEngine's Waveform Editor display is dominated by the waveform overview. The waveform overview is a graphic representation of your audio file, which is used for viewing, processing and editing audio files. Let's take a close look at the elements of the Waveform Editor display, then see how it acts when playing pieces of audio.

Elements of the Waveform Editor

When first opening a File Document, the Waveform Editor display shows the entire file (zoomed out all the way) as the initial default display.

In this display you can navigate (zoom) through the file, from the lowest level of detail (seeing entire file) to the highest level of resolution (viewing each sample). Navigation is assisted by the View Frame, detailed in the following section. When you're zoomed into an audio file, the scroll bar at the bottom of the editor display becomes active, and allows you to quickly navigate through a zoomed view. For fastest, easiest navigation just use your Mac keyboard's arrow keys.

At the far left hand side of the Waveform Editor display is a numeric scale, ranging from 0% to 100% indicating waveform amplitude as a percentage of total system headroom.

The green band located between the amplitude scale and the beginning of the waveform overview is a marker indicating the beginning of file. The red band located at the far right hand end of the waveform overview is another marker indicating the end of file. The green and red bands are there for your visual reference, so you'll always know when you're at the top, or tail end, of a particular file. They also come into play as assistants to the selection and View Frame tools, as we'll examine later.

Along the bottom edge of the Waveform Editor is a time display indicating file time in hours, minutes, seconds and milliseconds. Note the "tick mark" directly above the "hours" numerals, indicating the exact position of the particular time reading.

Waveform Editor: Playing Sound Files

When you play a sound file, a dotted yellow playback cursor scrolls across the waveform overview to show you the current file position. The playback cursor is not only a visual aid, it can be used to specify where you want playback to begin (assuming you've made no selection in the waveform overview).

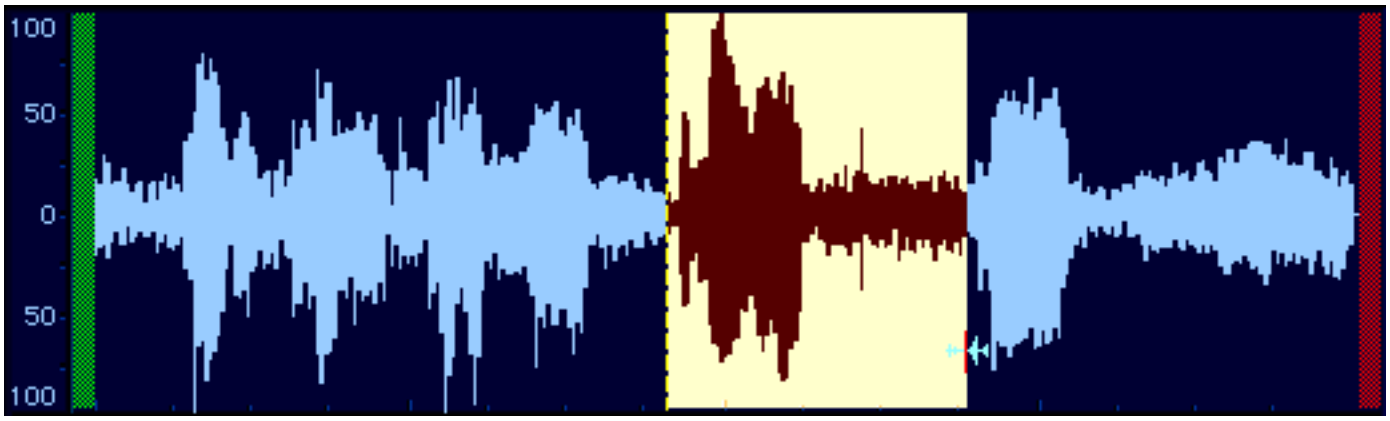
- Click anywhere in the waveform overview to reset the playback cursor to a new position.
- Now click on the Play button (or press the space bar) to begin waveform playback from the playback cursor

location.

Click on the Stop button (or press the Return key) to end playback and return the playback cursor to the beginning of the file.

Note that the Pause (spacebar) command affects the playback cursor differently than the Stop (Return) command. Pause simply pauses playback at the current time coordinate. When you resume playback from a paused state, the playback cursor continues from the point at which it was paused. Clicking in the waveform to reset playback cursor when in a paused state will have no effect, playback will always resume from the point at which you paused.

To select any portion of the waveform overview for processing or editing, simply click and drag with the mouse across the display. Selected areas will be highlighted.



Range of audio file selected in Waveform Editor

Lets examine how playback works when you have an active selection:

One-Shot Playback

Click and drag across a section of the waveform overview to create a selection.

Now click on the Play button (or press the space bar) to begin selection playback from the beginning of the selection.

Click on the Stop button (or press the Return key) to end playback and return the playback cursor to the beginning of the selection.

Click on the Play button again to resume playback from the beginning of the selection.

Note that here again, the Stop button (Return key) always returns the playback cursor to the beginning, except that in this case its to the beginning of the active selection. Think of the Stop button as "stop and return."

If you play the file to its end the cursor will likewise return to the start of file.

Looped Playback

Click on the Loop Button button to make HyperEngine play the section in a loop (repeatedly).

Now click on the Play Button to begin selection playback. Note that the current selection plays over and over.

Now click on the Pause button (or hit the spacebar) to pause playback.

Click on the spacebar (or press Play) to resume playback from the paused location.

Remember that when playing back audio, the Pause (spacebar) simply pauses playback at the current time coordinate. When you resume playback from a paused state, the playback cursor continues from the point at which it was paused.

Click anywhere in the waveform display to de-select, removing the active selection and allowing you to create another selection.

Power User Tip: You can alter the waveform view or selection range while play back is going. Try zooming way in and shift-dragging the selection start/end times to new locations while listening to the looped playback to fine-tune loops by ear. Use the Command-[and Command-] shortcuts to jump right to the selection start/end points.

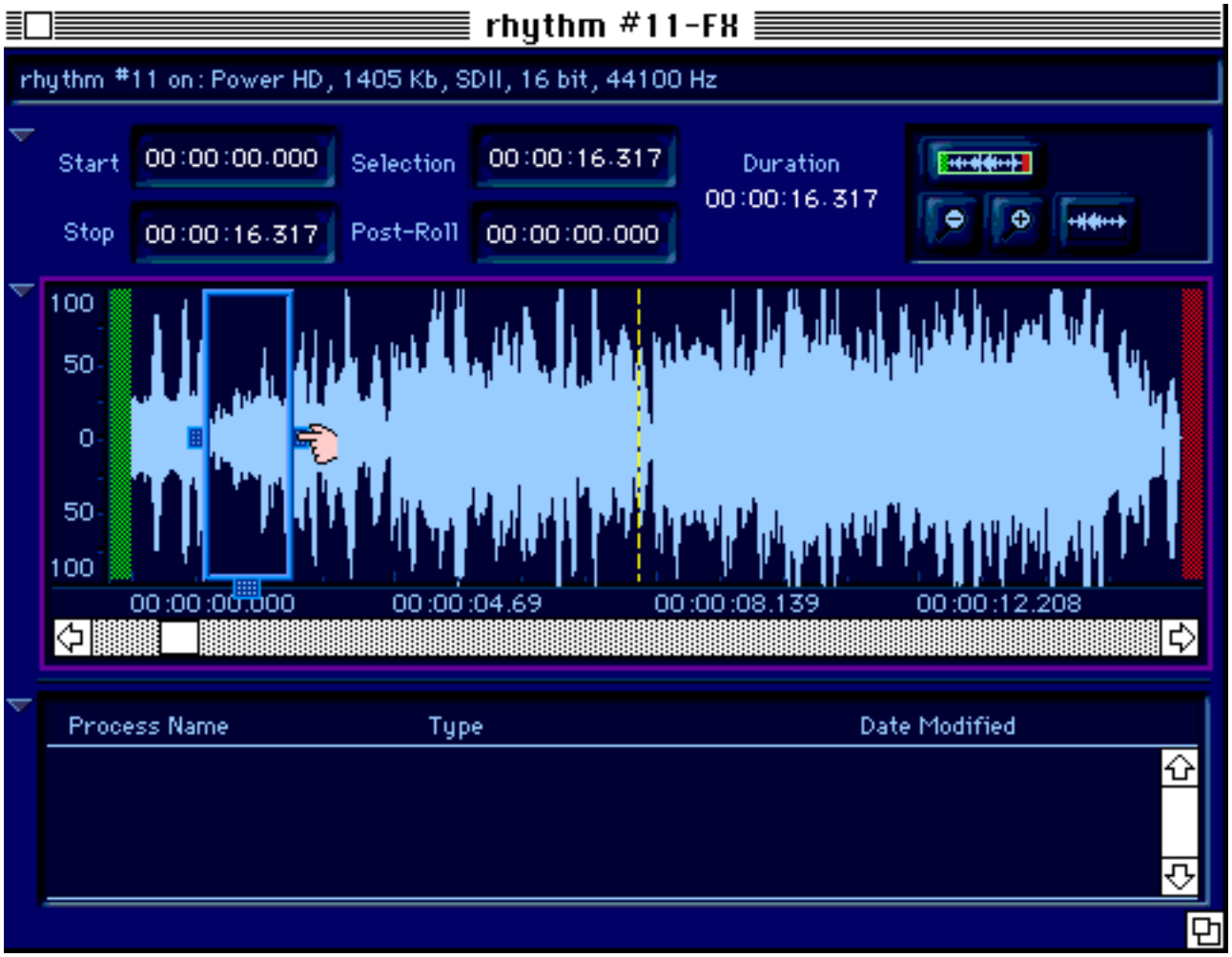
Now that you understand the basic elements of the Waveform Editor and how to use the playback cursor and selection range, let's next take a close look at the viewing, selection and editing tools included in HyperEngine.

Navigating Sound Files with the View Frame

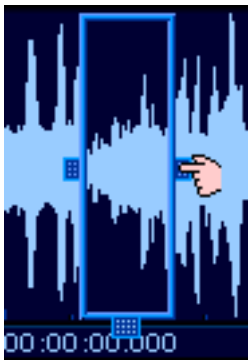
The View Frame is a new tool (with version 2.3) which makes it easy to view and edit sections of your sound file in detail. All "zoom" functions are linked to the View Frame. The View Frame defines the area being zoomed into; when you're looking at a zoomed view, you're seeing the section of the file that's currently in the View Frame.

Manipulating the View Frame

Click on the Hide/Show View Frame button to see the View Frame in the editor display, if it's not already visible.



File Document with View Frame visible, note "hand" cursor for View Frame resizing is active.



View Frame (detail)

Note that the mouse cursor changes into a "hand" when it is over the rectangular View Frame tabs, located along the left, right and bottom edges of the View Frame.

The View Frame tabs allow you to alter the size and location of the View Frame with a simple mouse move.

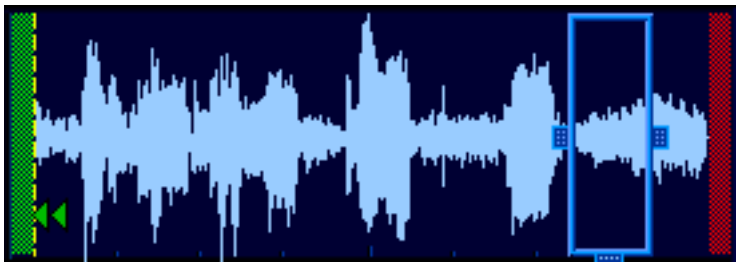
- Click and drag on the left View Frame tab to readjust the beginning of your zoom view area.
- Click and drag on the right View Frame tab to readjust the end point of your zoom view area.
- Click and drag on the bottom View Frame tab to move the current zoom range to a different location in the file.

Next, let's see how the green and blue zones (file header and ending markers) can assist you with View Frame Location.

Adjust the view frame so it's open to just a small portion of your file, and move the frame to the center of the Waveform Editor display.

Next, move your mouse cursor above the green zone, denoting the beginning of your audio file.

Notice that the mouse cursor changes to a green double-arrow when located over the green zone, as shown below:



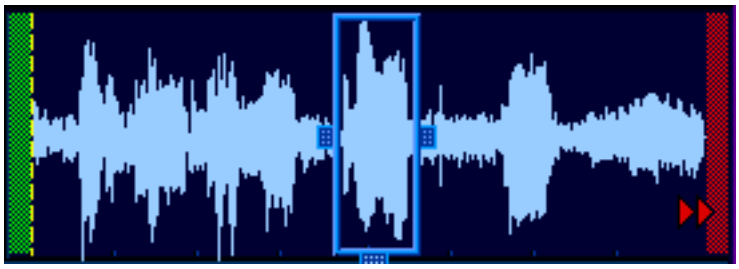
Note green double-arrow cursor, lower left-hand side

Now mouse-click in the green zone.

Notice that the View Frame has relocated to the beginning of the file.

Next, move your mouse cursor above the red zone, denoting the end of your audio file.

Notice that the mouse cursor changes to a red double-arrow when located over the red zone, as shown below:



Note red double-arrow cursor, lower right-hand side

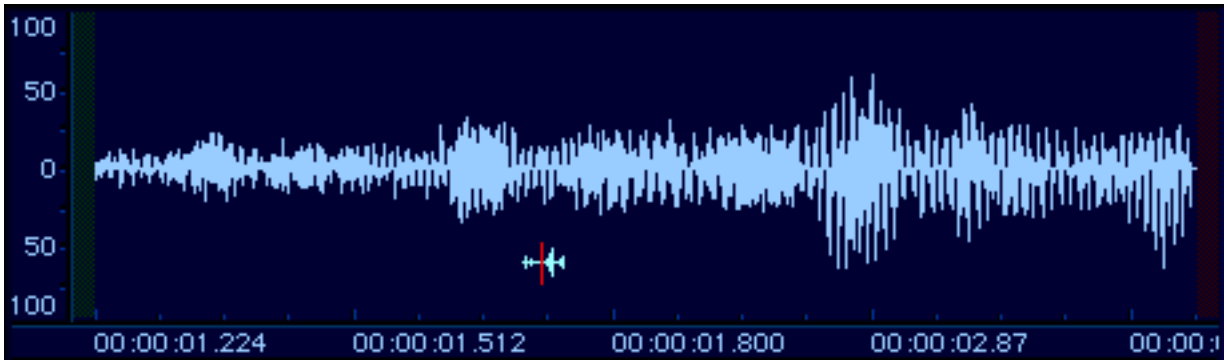
Now mouse-click in the red zone.

Notice that the View Frame has relocated to the end of the file.

Now that we've experimented with resizing and relocating the View Frame, let's jump into the frame and look at the zoom view.

Navigating Within the View Frame

Click on the View Toggle button (Or try the keyboard shortcut, command-/ [forward slash]) to switch over to the zoom view, so as to examine the range selected by the current View Frame.



Zoom View, seen when toggling into View Frame; note that "selection" cursor is active

Note that when you're looking at a zoom view, the View Frame is no longer visible (since you're now "inside" the frame).

Adjust the scroll bar along the bottom of the Waveform Editor display to navigate through your file, while still in the zoom view. The scroll bar is moving the View Frame along the duration of the sound file.

Click on the Zoom In/Zoom Out buttons to incrementally alter the size of the View Frame while inside the zoom view.

Note that the Zoom In/Out buttons are **not** the only way to alter the size of your View Frame. We've built in four other alternate ways of zooming, for your convenience:

While still in the zoom view (inside the View Frame) go to HyperEngine's Wave menu and select the Zoom In or Zoom Out menu items. View Frame will resize upon selection.

Next, try the keyboard alternates for those menu commands: Command-dash (a.k.a. hyphen or minus) to zoom out, command-= (equal/plus sign key) to zoom in.

If you have an enhanced keypad (which includes the numeric 10-key layout) try using the equals and minus keys there, while holding down the command key, which will also zoom your display.

Now, for fast-and-easy one-touch zooming, use the Mac arrow keys. Press the up arrow to zoom in, the down arrow to zoom out. Keep the up or down arrow key pressed to quickly zoom through multiple resolutions.

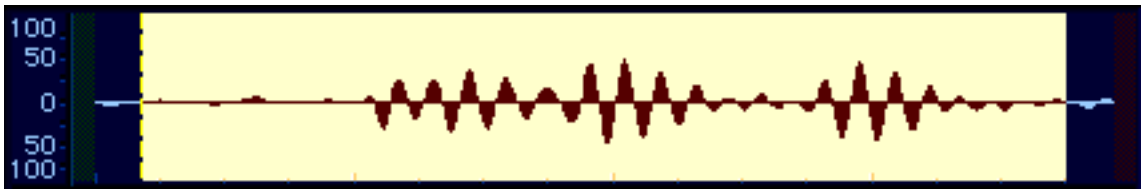
While we're here at the arrow keys, try pressing the left arrow and then the right arrow. You'll notice that these keys move the View Frame along the timeline, to the left and the right. Keep the left or right arrow keys depressed to scroll smoothly through the display. Left and right arrow keys duplicate the functionality of the scroll bar, which is located along the bottom of the Waveform Editor display.

Tips for Fast Zooming and Navigation

Power User Tip #1: Arrow Keys Once you get going with the arrow keys, using them to control the size and location of your zoom views (as described in the preceding paragraphs), you can almost completely forget about the View Frame. Just keep the View Toggle in "zoom" mode all the time, then do all your zooming and navigating with the arrow keys for maximum speed and convenience. Even if you learn only one keyboard alternative in HyperEngine, make sure it's the arrow keys. They're total time (and wrist) savers.

Power User Tip #2: Fit Selection When you're involved in sound file editing, you may frequently wish to select a

range of your file, and view that area in detail. We've included a special function to make that happen instantly. Simply click-and-drag across a section of the waveform overview to select a range, then use the keyboard shortcut command-\ (backslash) to fit the current selection to the zoomed view, as defined by the View Frame. This will create a new Frame view which surrounds your active selection range.You can also use the Wave menu command "Fit Selection to Window" to accomplish the same thing. This is probably the second most useful shortcut in HyperEngine, after the arrow keys described previously.

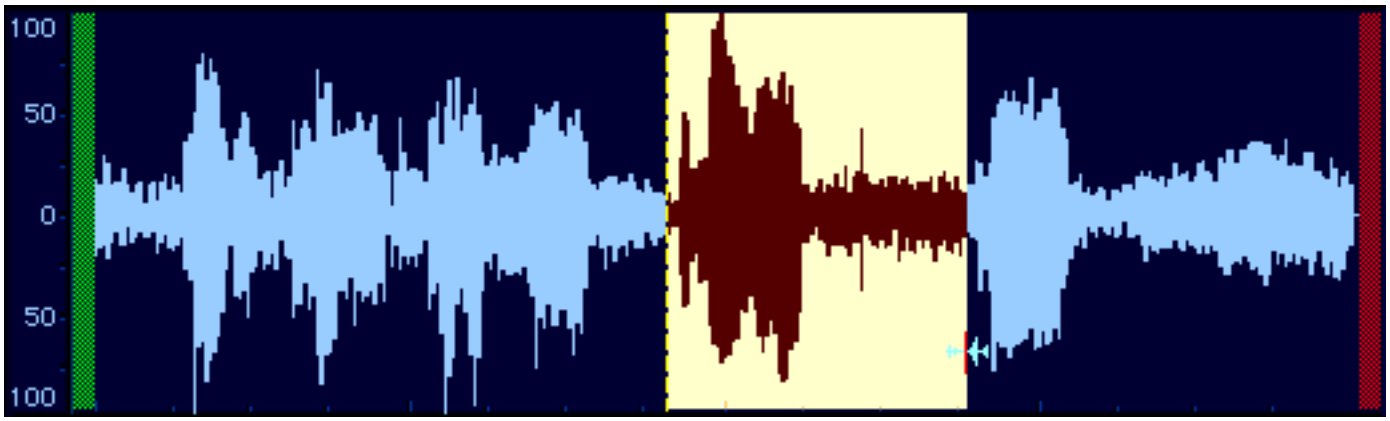


The result of a Fit Selection to Window (command-\) operation

Power User Tip #3: Control-Drag This variant on tip #2 allows you to quickly set a View Frame when no selection is active. Simply hold down the Control key while mouse-dragging across the waveform display. You'll notice that the mouse cursor turns into a magnifying glass, a "ghosted" View Frame appears, and this frame is resized as you mouse drag. Once you let up the mouse button you'll jump to the new view you've just defined. This feature works the same regardless of whether you're currently looking inside or outside the View Frame.

Waveform Editor Selection Tools

When the mouse pointer is located over the Waveform Editor window, the mouse cursor turns into a little blue waveform, with a red line bisecting it, indicating that the cursor is active for selecting. To select any portion of the waveform overview for processing or editing, simply click and drag with the mouse across the display. Selected areas will be highlighted.



Range of audio file selected in Waveform Editor, note "selection " active

The selection range defines the area of the waveform overview which is active for playback, editing and processing functions. Use the command-a keyboard shortcut, or the Select All option from HyperEngine's Edit menu, to select the entire audio file. Mouse-click anywhere in the waveform display to de-select.

The selection range can be manipulated in a variety of ways:

Resize Selection

- First, click and drag across a portion of the waveform overview to create a new selection.
- Now, while holding down the Shift key, mouse click again anywhere in the waveform overview to adjust the

beginning/ending of the selection.

↘ Remember that Shift-clicking when a selection is active lets you grow or constrain the current selection, operating much like any graphics program.

➠ Next, mouse-click anywhere in the waveform display to de-select, eliminating the selection range.

➠ Hit the Return key or press the Stop button to return the playback cursor to the beginning of the file.

➠ Now, hold down the shift key and click anywhere along the waveform display.

↘ You'll see that a new selection is created "from the top," extending from the beginning of the audio file to the cursor location.

Resize Selection on-the-fly

➠ While you still have an active selection, hold down the Shift key. Then mouse-click, and while holding down the mouse button, drag the cursor along the waveform overview.

↘ Note that holding down the Shift key while mouse dragging allows you to smoothly reset the beginning and end points of the current selection.

➠ Now press the Loop and Play buttons, so as to begin looped playback of the selection range.

➠ While your selection is playing back, hold down the Shift key and mouse-drag in the waveform display to readjust the start/end point of the selection.

↘ You'll notice that HyperEngine will readjust the playback range on-the-fly while you're resizing the selection. This allows you to audition a selection while you're simultaneously creating and adjusting the selection range, in real-time.

☞ **Power User Tip:**Using Loop playback and the shift key to audition selections on-the-fly is another major time saving feature, freeing you from the trial-and-error process of selecting, auditioning, reselecting and re-auditioning.

Move Selection Left/Right

Before we go on to the editing functions, there are a few other selection manipulations we should examine. Much as HyperEngine includes shortcuts to the zooming/View Frame functions, we've also built in some selection range helpers:

➠ Select a small portion of the audio file, something in the middle of the display.

➠ Now, from HyperEngine's Wave menu, try selecting the Move Selection Right or Move Selection Left functions.

↘ Move Selection Left/Right will move the active selection range earlier, or later, in the waveform display. The amount of movement is equal to the size of the selection range.

➠ Now, try the keyboard shortcuts for Move Selection: Press command-; (semicolon) and command-' (apostrophe) to shift the selection range to the left or right by the size of the range.

☞ **Power User Tip:**The Move Selection shortcut is especially handy when cutting loops out of a long instrumental performance, like a drum track, for example. It enables you to select the length of your loop (such as a measure or bar)

then quickly cruise down the entirety of your file, auditioning each measure until you find the perfect iteration, while always maintaining the original size of the selection range. This is an enormous time-saver, compared with re-selecting each measure separately, and allows you to maintain a defined "block" size while auditioning and cutting loops.

↘ Note that the playback cursor does not automatically relocate to the start of the selection range when using the Move Selection command. Hit the Stop button or Return key after doing a Move Selection to reset playback to the new selection beginning.

Go to Selection Start/End

One of the problems that is frequently encountered when working at high zoom resolutions is that it's easy to "misplace" the beginning or ending of your current selection, resulting in a lot of excessive scrolling through the display, trying to find where your start/end point was. To solve this, we've included a couple of selection locating aids:

➠ Select a small portion of the audio file, something in the middle of the display.

➠ Zoom deep into the display, by repeatedly pressing the up arrow key.

➠ Type the command-[(left bracket) key combination.

↘ Note that the beginning point of the current selection is now center-screen.

➠ Next, use the command-] (right bracket) shortcut.

↘ Now you'll see the end point of the current selection center-screen.

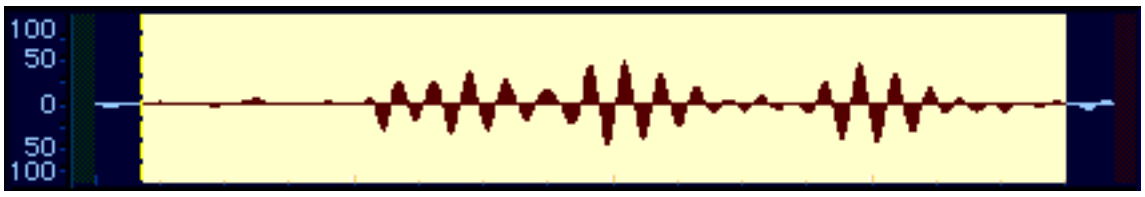
☞ International users please note: Some non-English Mac keypads (such as those made for the German market) do not include bracket keys. If this is the case with your system, just utilize the corresponding menu commands found in HyperEngine's Wave menu for the Go To Selection Start/End functions.

Fit Selection to Window

When you're involved in sound file editing, you may frequently wish to select a range of your file, and view that area in detail. We've included a special function to make that happen instantly.

➠ Use the keyboard shortcut command-\ (backslash) to fit the current selection to the zoomed view. You can also use the Wave menu command "Fit Selection to Window" to accomplish the same thing.

↘ This will create a new Frame view which surrounds your active selection range.



The result of a Fit Selection to Window (command-\) operation

Now that you understand how to make, move, view and manipulate a waveform selection range, let's use those tools for some file chopping and explore HyperEngine's new editing functions.

File Editing: Basic Cut/Copy/Paste

A new feature in HyperEngine 2.3 is the ability to do basic audio file editing, involving cut, copy and paste operations. Not only can you cut/paste within a single File Document, you also have the ability to copy material between two different File Documents.

 When pasting material between different file documents, note the following restrictions: You can only copy/paste from one mono file to another mono file, or from one stereo file to a second stereo file. You are prohibited from pasting a mono region into a stereo file, or from copying a stereo selection into a mono file. Note also that you'll only get predictable results when copying between files that are at the same sample rate. For instance, if you paste a 44.1k selection into a file that is at 22.05k, that new 44.1 region will play back at "half speed" in the 22.05k File Document.

The Cut/Copy/Paste commands are located in HyperEngine's Edit menu, and are duplicated by the familiar command-x, command-c and command-v keyboard alternatives. Here's how to use them:

Cut and Undo/Redo

 Mouse-drag through a section of the waveform overview to create an active selection.

 Select the Cut command from HyperEngine's Edit menu, use the command-x keyboard shortcut or simply press the delete key on your keypad.

 Your selection will be cut from the waveform display, and the remaining audio will be butt-spliced so as to cover the gap.

 Select the Undo (command-z) option from the Edit menu to undo your cut and restore the audio to its original position.

 Select the Redo (command-z) option from the Edit menu to restore your edit and take the selection back out of your file.

Copy and Paste

 Mouse-drag through a section of the waveform overview to create an active selection.

 Select the Copy command from HyperEngine's Edit menu, or use the command-c keyboard shortcut.

 This command copies the current selection to your Mac clipboard memory.

 Click anywhere on the waveform display to select an edit insertion point

 Select the Paste command from HyperEngine's Edit menu, or use the command-v shortcut.

 Clipboard data is now pasted at the location you've specified.

 Select any portion of the waveform overview by mouse-dragging through the display

 Select the Paste command from HyperEngine's Edit menu, or use the command-v shortcut.

 Clipboard data is now pasted at the location you've specified, replacing the current selection.

Remember, you can copy and paste waveform data between two separate File Documents, as long as both documents have the same number of channels and sample rate.

Clear

The Clear command is not currently implemented by HyperEngine, and will remain ghosted in the Edit menu.

File Editing: Drag-and-Drop

Now that you're familiar with HyperEngine's basic editing commands, let's look at a faster way to edit: The Drag-and-Drop functions. Drag-and-Drop editing allows you to "tear off" any portion of the waveform display and move that selection to another location in the file, or even to a separate file.

Mouse-drag through a section of the waveform overview to create an active selection.

Mouse-click within the selection range and hold down the mouse button to "grab" the selected range.

Notice that the mouse cursor changes to a "fist" denoting that the selection has been grabbed.

While keeping the mouse button depressed, move the pointer to another area in the wave display.

An outline view of the selection range is seen as you drag the selection to the new location. Notice also that the playback cursor can be seen as you drag the selection, indicating your new "in" point.

While still keeping the mouse button depressed, position the playback cursor to your desired insertion point.

Release the mouse button and your selection will be re-located to the new insert point.

The procedure above works the same way whether editing with a single File Document, or moving audio ranges between two different File Documents.

Drag a Copy

A variant on the Drag-and-Drop procedure is "drag a copy" which is accomplished by holding down the option (alt) key while editing. Here's how it works:

Mouse-drag through a section of the waveform overview to create an active selection.

While holding down the option (alt) key, mouse-click within the selection range and hold down the mouse button to "grab" the selected range.

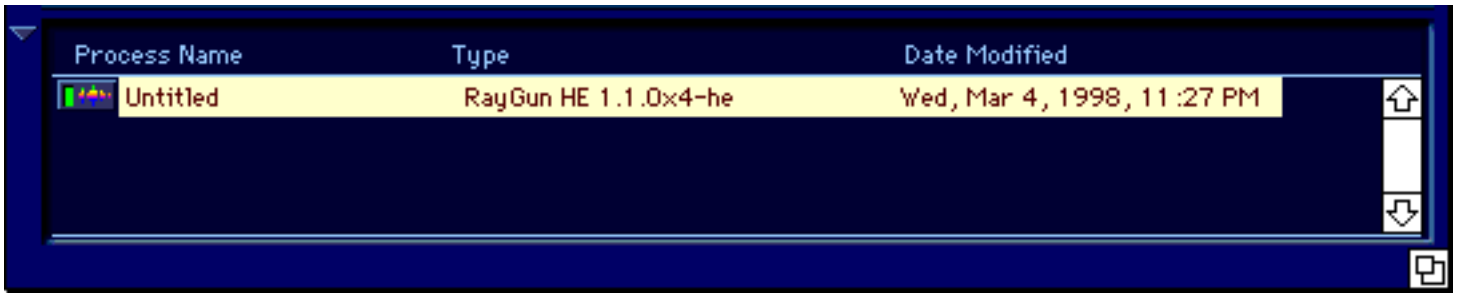
While keeping the option key and mouse button depressed, move the selection copy to a new position in the waveform display. When you've positioned the playback cursor to your desired insertion point, release the mouse button.

Your selection copy is now in a new position, and the original selection area remains untouched, as it was before this edit.

Remember, you can Drag-and-Drop waveform data between two separate File Documents, as long as both documents have the same number of channels and sample rate.

Now that you're comfortable with all of HyperEngine's Waveform Editor features, lets examine in detail the last element of every Reference Document, the Process List.

The Process List



Now, let's examine the Process List. This list appears, and behaves, exactly the same in both the Play-Thru Document and File Document reference windows. When a new Reference Document is created, this list is empty. You may add Hyperprism 2, Ionizer or Ray Gun effects by selecting them from the Plug-Ins menu. Access the effects in your document by clicking on their entries in the List. You also engage/bypass the processing in the Process List by clicking the toggle button to the left of each list entry.

The Process List allows you to add multiple processes to a single reference document, so you can create complex multi-effects. And note that even when you close a Hyperprism, Ionizer or Ray Gun window, if the effect is still in the process list, and still toggled to "engaged" (not bypassed) then the effect processing, and any associated parameter automation, will be heard in the playback and in any processed new file.

Using the Process List

Once you have created a reference document (either a Play-Thru document or a File document) you can add one or more HyperEngine-compatible effects to it. In this example, we will add a Ray Gun effect.

- Go to the File Menu

- Choose New File Document, or simply use the keyboard shortcut, command-n.

- In the resultant Select Sound File dialog, choose a sound file from disk and click Open (or press either the Return or Enter keys, which are the dialog shortcuts for clicking Open).

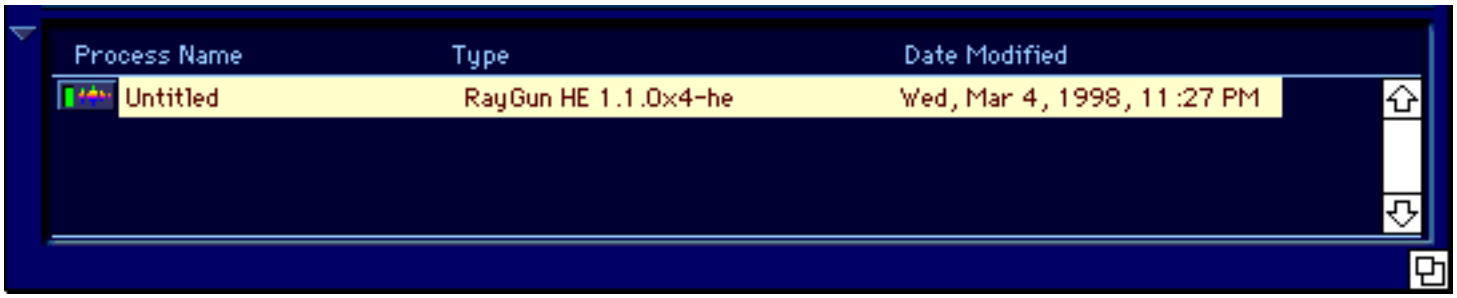
- A New File Document reference window will appear, along with a graphic waveform overview showing the audio in the sound file you just selected. Depending on the length of the sound file, the overview may take a few moments to calculate.

- Press the spacebar or hit the Play button to audition your sound file. When you're done auditioning, press Stop (Return key) or Pause (spacebar) to suspend playback.

- From the Plug-Ins menu, choose Ray Gun (or other HyperEngine plug-in effect).

If the Plug-Ins menu is ghosted, simply Stop or Pause playback and HyperEngine will allow you access to the Plug-Ins menu. You are **not** allowed to add plug-ins to a document while playback is underway, HyperEngine will resist giving up control over the Sound Manager until you've suspended playback. Stopping/pausing, then adding the effect and resuming playback is the required procedure.

The Ray Gun interface will appear on screen and Ray Gun will be added to the Process List in the document window. The process name is "untitled" by default (as displayed below):



Use the Ray Gun (or other effect) interface to adjust your process settings, while listening to HyperEngine playback. Enable the Loop function if you're working with a short file or selection, so playback will loop continuously as you dial in your settings.

You may now begin experimenting with effect variations. Adjust effect settings as described in the Ionizer, Ray Gun or Hyperprism 2 documentation supplied on the Arboretum CD.

Once you're satisfied with your effect settings you can close the Ray Gun (or other effect's) control window.

You'll still hear the effect processing "in the background" as long as the effect remains in the Process List, and is toggled to "engaged" (not bypassed).

If you want to add additional effect to the signal chain, simply select another entry from the Plug-Ins menu.

Your new effect will be added to the bottom of the Process List, and to the tail end of the signal chain.

Removing an Effect from the Reference Document

Select the effect you want to delete by clicking on the Process List. The entry will be highlighted.

Hit the Delete key on your keypad

The effect will be deleted from the Process List and removed from the signal chain.

Note that there is no "Undo" after this function!

Naming An Effect Variation

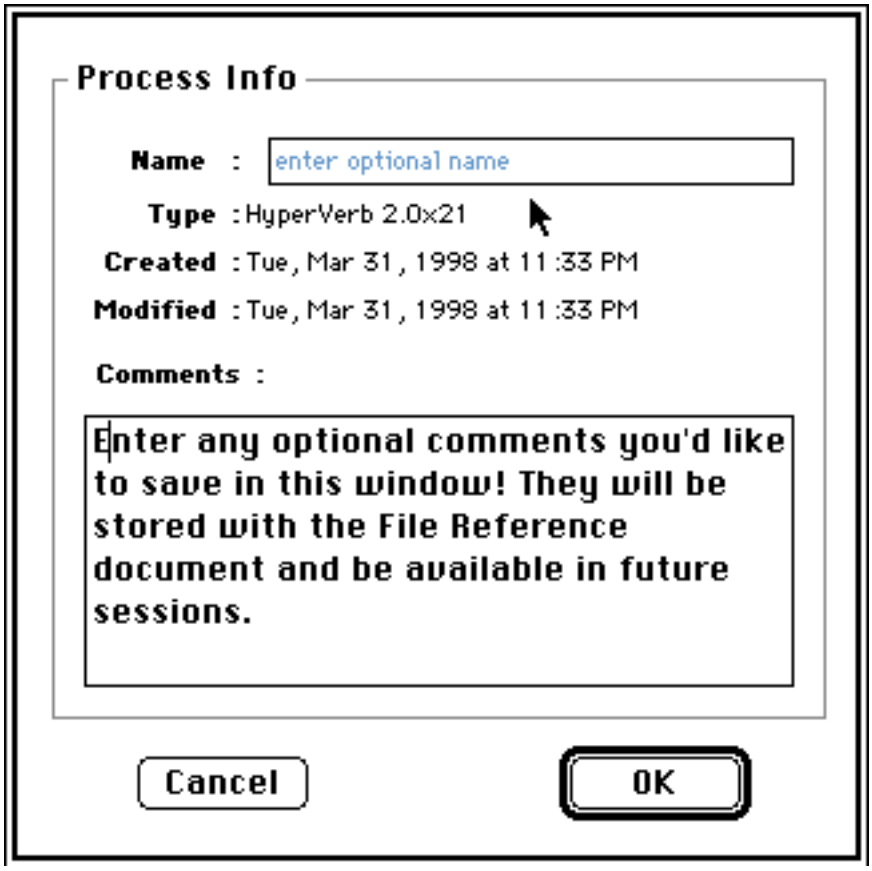
Since you can add more than one effect variation to a reference document, you may want to use unique names to help tell them apart.

To name an effect in the reference document process list:

Select the effect you want to name by clicking on the Process List.

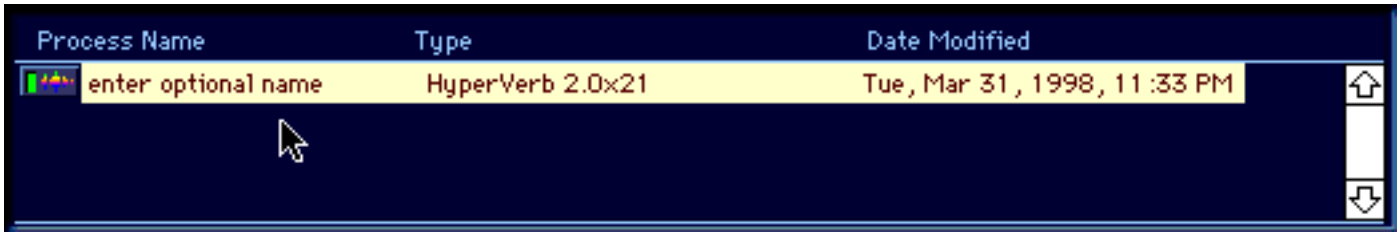
Choose Get Info from the File menu.

The Process Info dialog will appear.



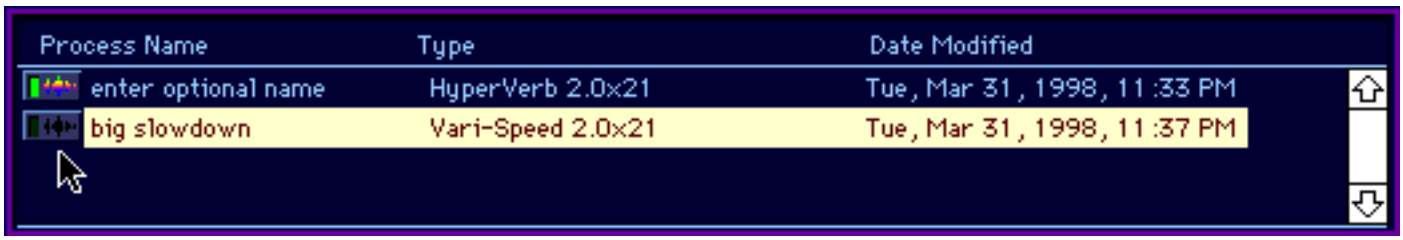
Enter the new name and any optional comments and click OK.

The effect will appear with your new name in the document window's Process List.



Bypassing One Effect

It is possible to have multiple simultaneous effects active in the HyperEngine window. Each effect may be engaged or bypassed individually. Every effect in the HyperEngine document window includes its own Bypass button. Simply click the button to bypass or engage that particular effect. When the button is illuminated, the effect process is engaged. When an effect's process is engaged, you'll hear its effect on the sound, even though the particular effect's screen (HyperVerb, in this example) is closed.



The Hyperprism Vari-Speed effect named "big slowdown" is bypassed, while the HyperVerb effect called "enter optional name" is still engaged.

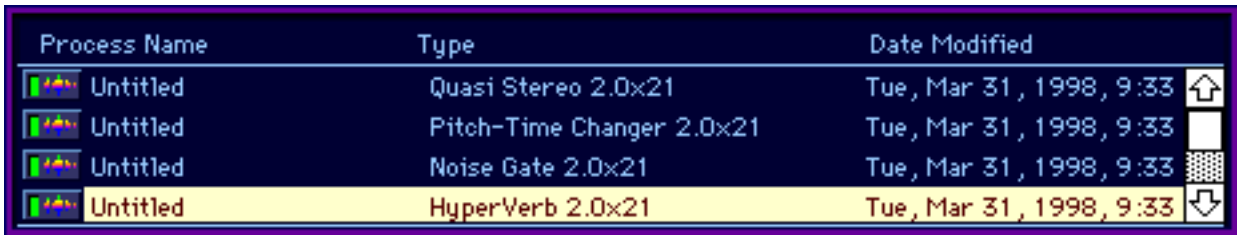
Changing the Order of Effects

HyperEngine processes run in series, from top to bottom, as displayed in the reference document's Process List. As you add additional processes to a reference document, each new effect is added to the bottom of the list (and is thus inserted at the end of the signal processing chain). When creating a "multi-effect" incorporating numerous processes, you may wish to rearrange the processing chain. Rearranging the Process List is a simple drag-and-drop operation:

- Click on the Process List entry for the effect you wish to move. The effect's list entry will be highlighted.

- While holding down the mouse button "drag" the highlighted effect to its new position in the list.

Resequencing the Process List can have a tremendous impact on the way your processes sound. Common applications can include placing a compressor at the "head" of the chain, followed by filters and reverb. Or perhaps you'd use a Quasi Stereo (mono-to-stereo) process as the first stage, followed by other spatial effects.



An example of a Hyperprism multi-effect chain

Process to New File

Once you have edited your file and adjusted your effect settings to your liking, use the Process to New File command to create a new sound file using the current File Document settings.

To use the Process to New File command:

- Select the range of audio you wish to process by click-dragging in the waveform overview of the File Document window. Use the command-a shortcut if you want to select the entire file for processing.

- Select a Post-Roll value if the processing should alter the length of the file, such as in instances where a reverb effect should add a "tail" to a sound.

- Choose the Process to New File command from the File menu.

- Choose the output file specifications (sample rate and bit-depth).

- If exporting stereo audio select either interleaved or split stereo (.L/.R) file output.

- If you're saving the sound as an 8-bit file, try the Dither and Noise Shaping options, which add a small bit of noise to the signal to mask the audible effects of the bit reduction. These can also be used when doing straight 16-bit to 16-bit processing; the effect is less noticeable in those situations.

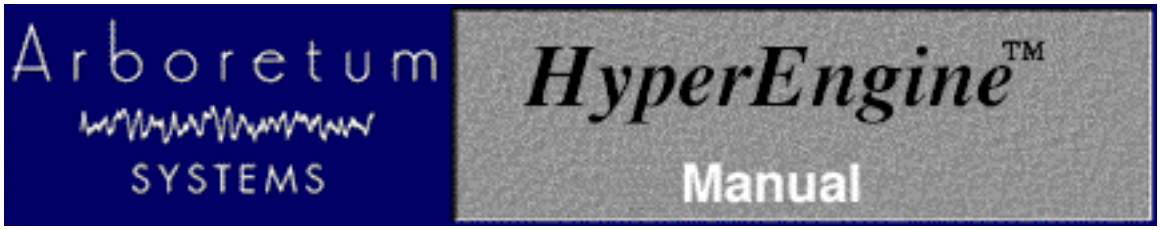
If you're recording audio for eventual use on a CD, we recommend that you always create 16-bit interleaved stereo Sound Designer II files, sampled at 44,100 Hz. This is the file specification you'll need to create "Red Book" audio CDs, and is what programs such as Toast and MasterList CD accept for burning audio CDs. Sticking with these settings will prevent file-conversion headaches down the road.

 The next dialog will ask you to specify file format (AIFF or SDII), a name for the file and its target location on disk. Click "Save" (or hit the Return key) to begin the process. The selected range of audio will be written to disk.

 Your original sound file will be unchanged!

This concludes our chapter on the HyperEngine Reference Documents. Read on through the next chapter for more information on HyperEngine menu commands and key command shortcuts.

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



HyperEngine Command Reference

This document describes the function of each HyperEngine menu command, as well as key commands and shortcuts. Click on any command to learn about it.

File Menu

- New File Document
- New Play-Thru Document
- Open Document
- Reassign
- Import Split Stereo
- Import Foreign File
- Close
- Save
- Save As
- Process to New file
- Export Current File
- Get Info
- Quit

Edit Menu

- Undo/Redo
- Cut
- Copy
- Paste
- Clear
- Select All

Plug-ins Menu

Wave Menu

- Switch to Zoom View/Overview
- Zoom in

- Zoom out
- Move Selection Left
- Move Selection Right
- Go To Selection Start
- Go To Selection End
- Fit Selection to Window
- Repeat Selection

Windows Menu

Preference Menu

- Configure Audio System
- Driver Selection
- Processing Preferences

HyperEngine Keyboard Shortcut Summary

File Menu

The File Menu contains commands which allow you to open, save and manipulate HyperEngine Reference Documents.

New File Document

This creates a new HyperEngine File Document. File Documents allow you to edit and process sound files on disk, without ever altering your original media. When you choose this command, you will be prompted to select any existing AIFF or SDII mono or interleaved stereo sound file to open within the HyperEngine File Document. This file will not be changed; HyperEngine always writes the modified audio to a new sound file, using the Process to New File command. Command-n is the New File Document keyboard shortcut. For more information on understanding and using HyperEngine File Documents, peruse the material in the previous chapter, beginning with the section called "Processing Disk Files: The File Document." For background information on file formats, sample rates and other aspects of digital audio, click here.

New Play-Thru document

This creates a HyperEngine document that does not reference a sound file on disk. Instead, it references the live audio signal connected to your Mac's Sound Input device. Play-Thru documents are used when recording new audio files from a live source, and when processing real-time audio in a performance situation. For more information on understanding and using HyperEngine Play-Thru Documents, peruse the material in the previous chapter, beginning with the section on "Processing Real-Time Audio." For more about HyperEngine, sound cards and the Apple Sound Manager, click here.

Open Document

The Open command opens any existing HyperEngine File Document or Play-Thru Document, which you would have created by the New Document commands described above. Command-o is the keyboard alternative.


This command will NOT open a sound file, it opens HyperEngine Reference Documents. To work with sound files, create a New File Document (use the command-n keyboard shortcut) which will prompt you for the AIFF or Sound Designer II file you want to work with.

For complete details on opening and using HyperEngine Reference Documents, peruse the material in the previous

chapter.

Reassign

Every HyperEngine File Document is linked to one, and only one, sound file. However, using the Reassign command, you can link any File Document to a new sound file. For example, you might do this if you wanted to apply a particular set of effects to more than one sound file.

Import Split Stereo

HyperEngine has direct support for mono and interleaved stereo audio files. However, some audio editors create split stereo files, in which the left and right channels are two discrete files. Split stereo files typically share the same file name, with the addition of a .L or .R suffix to denote their channel position.

Selecting the Import Split Stereo option results in two dialogs in which you locate and select the .L and .R files which you wish to import. The first dialog will ask you to locate the left channel file, the second dialog requests the right channel.

Next you'll see a dialog where you determine the name and location for the new audio file to be created. Upon pressing OK HyperEngine will combine the .L and .R files into a new interleaved stereo file, and open that resultant file in a new File Reference Document .

Bear in mind that the Import Split Stereo function is creating new audio files on disk which may consume substantial disk space.

One interesting aspect of Import Split Stereo is that HyperEngine does NOT require that file names have .L/.R suffixes, nor does it require that they have the same names at all. Moreover, you can import any two mono files, of any length, and combine them in a new stereo file by using this function. If the mono sound files are of different lengths then the new stereo file will have the length of the shorter file.

Import Foreign File

Import Foreign File takes advantages of the file services offered by Apple's QuickTime 3 technology. With HyperEngine 2.4 and later you may import any Wave (.WAV), QuickTime (.MOV), .System 7 Sound (.SND) or uLaw audio file. Better still, you can import tracks from audio CDs directly into HyperEngine, if you're using a supported Apple CD-ROM drive.

Using the Import Foreign File features requires that QuickTime be loaded in your Extensions folder. QuickTime 3 includes a number of different System extensions; for the purposes of HyperEngine we're using the QuickTime and QuickTime Power Plug.

Selecting Import Foreign File results in a dialog where you locate the file you wish to import. While in the selection dialog you may preview any prospective files. Once you've selected a file or CD track for import, you'll be asked to specify the name and location for the new AIFF file that will be created. Upon Okaying these choices the import will proceed, a new AIFF file containing the audio will be created, and the sound file will open up inside a new HyperEngine File Reference Document.

Bear in mind that the Import Foreign File function is creating new audio files on disk which may consume substantial disk space.

Using Import Foreign File while running QuickTime 2.5 is OK but it will automatically truncate the number of channels, bit depth and sample rate of any CD audio imported to mono 8-bit 22050. Get QuickTime 3 by free download from http://www.apple.com

Close

This command closes the current HyperEngine Reference Document. HyperEngine will always ask you if you want to save all changes made to the document, prior to executing the Close command. Command-w is the keyboard shortcut.

Save

This command saves the current HyperEngine Reference Document. When you're working with a File Document, HyperEngine by default will name your File Document with the name of the sound file it references, followed by the extension "-FX." If you're working with a Play-Thru Document, HyperEngine just names the document "Play-Thru" by default. Naturally, you can rename any Reference Document with the custom name of your choice. Command-s is the keyboard shortcut.

 The Save command writes only the Reference Document to disk, it does not duplicate the sound file. Reference Documents take up very little disk space, so don't worry about saving or copying them as often as you like.

Save as

Saves the current HyperEngine Reference Document to a new name.

Process to New File

Creates a new sound file based on the audio file referenced by the current File Document, as modified by the current settings of all active effects in the Process List plus any edits you've performed in the Waveform Editor display. When selecting Process to New File you'll see the Output File Parameters dialog. Here you specify your new file's sample rate and bit-depth, choose between split setereo (.L/.R) and interleaved stereo, and you may also select Dithering and Noise Shaping options.

Dithering and Noise Shaping are ways of masking audio artifacts which can result from converting sound from a higher bit depth to a lower one. Dithering and Noise shaping can make a noticeable difference on audio being converted from 16 to 8 bits, and as HyperEngine does all its work at 32-bit floating point you can also dither and noise shape when rendering 24 or 16-bit files.

After the Output File Parameters dialog you will be prompted to set the file format (AIFF or SDII,) name and location of the new file. Once you've OK'd these selections the new file will be processed and all edits and effects which are active in the File Reference Document will be incorporated into the new audio file.

Process to New File creates only AIFF or SDII files; once you have created a processed file you can export it to other formats such as Wave or QuickTime using the Export Current File command.

 Executing this command will create a new file similar in size to the original sound file you are working with. This could require a lot of free hard disk space! Allow 5 MB for each minute of a 16-bit mono file sampled at 44.1k, or 10 MB for each stereo minute.

For further step-by-step instructions on processing new files, please review the related material in the previous chapter.

Export Current File

Export Current File takes advantages of the file services offered by Apple's QuickTime 3 technology. With HyperEngine 2.4 and later you may export any audio as a Wave (.WAV), QuickTime (.MOV), .System 7 Sound (.SND) or uLaw audio file.

Using the Export Current File features requires that QuickTime be loaded in your Extensions folder. QuickTime 3 includes a number of different System extensions; for the purposes of HyperEngine we're using the QuickTime and QuickTime Power Plug.

Selecting Export Current File results in a dialog where you specify the name, location and file format for the new sound file. Upon Okaying these choices the export will proceed and a new file containing the audio will be created on disk.

 Export Current File, as its name implies, exports only the current audio file. This function ignores all effects, settings, edits and selections in the File Reference Document and merely exports the raw audio file, in its entirety. To create new files which incorporate Hyperprism/Ionizer/Ray Gun processing, use the Process to New File command.

Bear in mind that the Export Current File function is creating new audio files on disk which may consume substantial disk space.

Note that using Export Current File with QuickTime 2.5 is OK, but you'll only be able to export .SND or .MOV files. QuickTime 3 is strongly recommended. Get QuickTime 3 by free download from http://www.apple.com

Get Info

The Get Info command brings up information on the currently-selected process from your File Document. Use this function to change the names of settings or enter comments. Command-i is the shortcut. More information on Get Info applications can be found in the previous chapter.

Quit

Exits the HyperEngine program. Command-q is the keyboard alternative.

Edit Menu

The Edit Menu contains commands which pertain to HyperEngine's waveform editing functions. Edit Menu commands are only active when working with a HyperEngine File Document since Play-Thru documents have no provision for waveform editing.

Undo/Redo

The Undo command allows you to reverse the last editing operation. When you execute an Undo, this menu option changes into Redo, allowing you to reverse the previous Undo and restore your last editing change. Command-z is the Undo/Redo keyboard shortcut. Check out the File Editing section in the previous chapter for step-by-step instructions on using Undo/Redo.

 Certain File Document functions, such as removing an effect from the Process List, can not be undone or redone.

Cut

The Cut command allows you to remove your current selection from the File Document's waveform display and copy it to clipboard memory; remaining audio, if any, will be butt-spliced so as to close the gap created by a Cut operation. Command-x is a keyboard shortcut for Cut. You can also simply use the Delete key for one-touch removal of any selected waveform range, but remember that Delete does not save the selection to memory. Check out the File Editing section in the previous chapter for step-by-step instructions on using the Cut command.

Copy

The Copy command copies the current selection from the File Document's Waveform Editor display to the Mac's clipboard memory. Command-c is the keyboard shortcut for the Copy function.

Copy functions are generally used in conjunction with the Paste command, keep reading for more details on Paste. Check out the File Editing section in the previous chapter for step-by-step instructions on using the Copy command.

Remember, you can copy and paste waveform data between two separate File Documents, as long as both documents have the same number of channels and sample rate.

If you're running out of memory, HyperEngine may not be able to execute a Copy command, especially if a long piece of audio is selected in a low-RAM situation. If you encounter "out of memory" messages, or if Copy fails to function properly, quit any other open applications or try allocating more system RAM to HyperEngine.

Paste

The Paste command copies the current contents of the Mac's clipboard memory to any insertion point or selection range within the File Document's Waveform Editor Display. If you're pasting into a selection range, the Paste command will replace the selection's contents with the clipboard memory's contents.Command-v is the keyboard shortcut for the Paste function.

Mouse-click in the Waveform Editor display if you want to paste again at a new insert point (otherwise it just pastes into the selection, right over the previous paste). If you want to paste multiple copies of a selection in a series, first perform the Paste command, then select the Repeat Selection command from the Wave menu.

Paste is generally used after executing a Cut or Copy command. Check out the File Editing section in the previous chapter for step-by-step instructions on using the Paste command.

Remember, you can copy and paste waveform data between two separate File Documents, as long as both documents have the same number of channels and sample rate.

Clear

The Clear command is not currently implemented by HyperEngine and will remain ghosted in the Edit menu.

Select All

The Select All command allows you to select the entire duration of the audio file displayed in the File Document's Waveform Editor window. This command serves as a quick alternative to mouse-dragging through the waveform overview in situations where you wish to audition, edit or process the entire sound file. Command-a is the keyboard shortcut for the Select All function.

Plug-Ins Menu

Any HyperEngine-compatible effect plug-ins that are placed in the HyperEngine Plug-ins Folder will appear in this menu. This includes the Ray Gun and Ionizer for HyperEngine, plus the Hyperprism 2 effects set.

Once you've opened a HyperEngine File or Play-Thru document, select any plug-ins you want to work with from this menu. Each new effect will be added to the bottom of the Process List in the current Reference Document.

If the Plug-Ins menu is ghosted, simply Stop or Pause playback and HyperEngine will allow you access to the Plug-Ins menu. You are **not** allowed to add plug-ins to a document while playback is underway, HyperEngine will resist giving up control over the Sound Manager until you've suspended playback. Stopping/pausing, then adding the effect and resuming playback is the required procedure.

In order for HyperEngine to use effects plug-ins, these plug-ins must be "HE" compatible and must be installed in a folder named "HyperEngine Plug-ins" which will reside in the same folder, on the same directory level, as the HyperEngine application. Also remember that Hyperprism effects should be batch Serialized prior to launching

HyperEngine, in order to save you inputting a LOT of serial numbers. Refer back to the HyperEngine installation chapter, as well as the set-up and installation chapters in the documentation associated with the particular plug-ins you're using.

Wave Menu

Wave menu commands are used by the File Document's Waveform Editor display to assist you with viewing, navigating and selecting portions of the waveform overview. Wave menu commands allow you to manipulate the View Frame (associated with "zoom" functions) and the waveform selection range.

Switch to Zoom-View/Switch to Over-View

This command duplicates the function of the File Document's View Toggle button, allowing you to switch the Waveform Editor display between the overview (of the entire sound file) and the zoom view, as defined by the View Frame. Command-/ (forward slash) is the keyboard shortcut for this function. For complete step-by-step instructions on using and manipulating Views, please peruse the View Frame coverage in the previous chapter.

Zoom In

Zoom In decreases the size of the Waveform Editor display's View Frame, allowing you to examine sections of your waveform overview in greater detail. Command-=(equals) is a keyboard shortcut. For fast one-touch zoom-ins just press and hold the "up arrow" key. Zoom commands can only be used when the View Toggle is engaged (when you're looking "inside" the View Frame).

☞ This command has the same effect as the "+" magnifying glass button located above the Waveform Editor display.

For step-by -step instructions on zooming, plus some Power User Tips, see this manual's section on Navigating Sound Files.

Zoom out

Zoom Out increases the size of the Waveform Editor display's View Frame, allowing you to examine sections of your waveform overview in lesser detail. Command--(minus or hyphen) is a keyboard shortcut. For fast one-touch zoom-outs just press and hold the "down arrow" key. Zoom commands can only be used when the View Toggle is engaged (when you're looking "inside" the View Frame).

☞ This command has the same effect as the "-" magnifying glass button located above the Waveform Editor display.

For step-by -step instructions on zooming, plus some Power User Tips, see this manual's section on Navigating Sound Files.

Move Selection Left

Move Selection Left will shift the active selection range to an earlier position in the waveform display. The amount of movement is equal to the size of the selection range. Command-; (semicolon) is the keyboard shortcut for Move Selection Left. For step-by-step instructions on using this command please peruse the material on Selection Tools in the previous chapter.

Move Selection Right

Move Selection Right will move the active selection range to a later position in the waveform display. The amount of movement is equal to the size of the selection range. Command-' (apostrophe) is the keyboard shortcut for Move Selection

Right. For step-by-step instructions on using this command please peruse the material on Selection Tools in the previous chapter.

 The Move Selection Left/Right shortcuts are especially handy when cutting loops out of a long instrumental performance, like a drum track, for example. They enable you to select the length of your loop (such as a measure or bar) then quickly cruise down the entirety of your file, auditioning each measure until you find the perfect iteration, while always maintaining the original size of the selection range. This is an enormous time-saver, compared with re-selecting each measure separately, and allows you to maintain a defined "block" size while auditioning and cutting loops.

Go To Selection Start

Go To Selection Start will re-locate the Waveform Editor display's View Frame so that the beginning of your current active selection is seen in the middle of the waveform overview display. Command-[(left bracket) is the keyboard shortcut for this function. The Go To Selection Start/End commands can save you a lot of scrolling-and-searching when you're editing at a high zoom resolution. For step-by-step instructions on using this command please peruse the material on Selection Tools in the previous chapter.

Go To Selection End

Go To Selection End will re-locate the Waveform Editor display's View Frame so that the ending of your current active selection is seen in the middle of the waveform overview display. Command-] (right bracket) is the keyboard shortcut for this function. The Go To Selection Start/End commands can save you a lot of scrolling-and-searching when you're editing at a high zoom resolution. For step-by-step instructions on using this command please peruse the material on Selection Tools in the previous chapter.

 International users please note: Some non-English Mac keypads (such as those made for the German market) do not include bracket keys. If this is the case with your system, just utilize the corresponding menu commands found in HyperEngine's Wave menu for the Go To Selection Start/End functions.

Fit Selection to Window

Fit Selection to Window will re-size and re-locate the Waveform Editor display's View Frame so as to match the current active selection, allowing you to view your selection in detail. This saves you from having to fiddle with the View Frame every time you wish to view a selection at a higher resolution. Command-\ (backslash) is the keyboard shortcut for this function. For step-by-step instructions on using this command please peruse the material on Selection Tools in the previous chapter.

Repeat Selection

Repeat Selection is a "multiple-paste-after" command which allows you to append any number of copies of the current selection to the end of the selection range. Use this command to create a repeating "loop" in your sound file. Simply select (highlight) any portion of the waveform overview in the File Document's editor display, then call the Repeat Selection command or use the command-r shortcut.

Repeat Selection is also helpful after a copy/paste operation; once you've pasted in data to any location in the waveform display, call the Repeat Selection command. As your pasted data is still highlighted after the Paste command, there is no need to re-select the range.

Repeat Selection will show you a simple dialog, asking you to enter the desired number of repetitions. Enter the number of additional iterations you desire, click on the OK button, and the process will be executed. There's more information on Copy and Paste commands as well as the basics of file editing elsewhere in this guide.

Windows Menu

The Windows menu is a screen management tool. If you've closed the HyperEngine Play Back window, you can bring it back on-screen by selecting it from this menu. Any Reference Document windows you have open are also listed in this menu. Select any listed Reference Document window to bring that screen into view.

Preference Menu

The Preference Menu is where you can access various settings pertaining to audio input/output and hardware.

Hardware Submenu

The Hardware Submenu contains the Configure Audio System dialog and the sound card driver selections.

Driver Selection

Use this submenu to select your sound input/output device. Sound Manager is the default, which should be selected if you're planning on using the Macintosh Mic input and speaker/headphone output as your audio hardware. If you have loaded HyperEngine IO drivers for any other sound cards, they should show up under this submenu as well.

HyperEngine 2.3 currently ships with drivers for the Apple Sound Manager, the Korg 1212 I/O card and the Digidesign Audiomedia II and Audiomedia III cards. HyperEngine sound card drivers reside in a folder named "HyperEngine IO Drivers" which must be located within the same folder, on the same directory level, as the HyperEngine application.

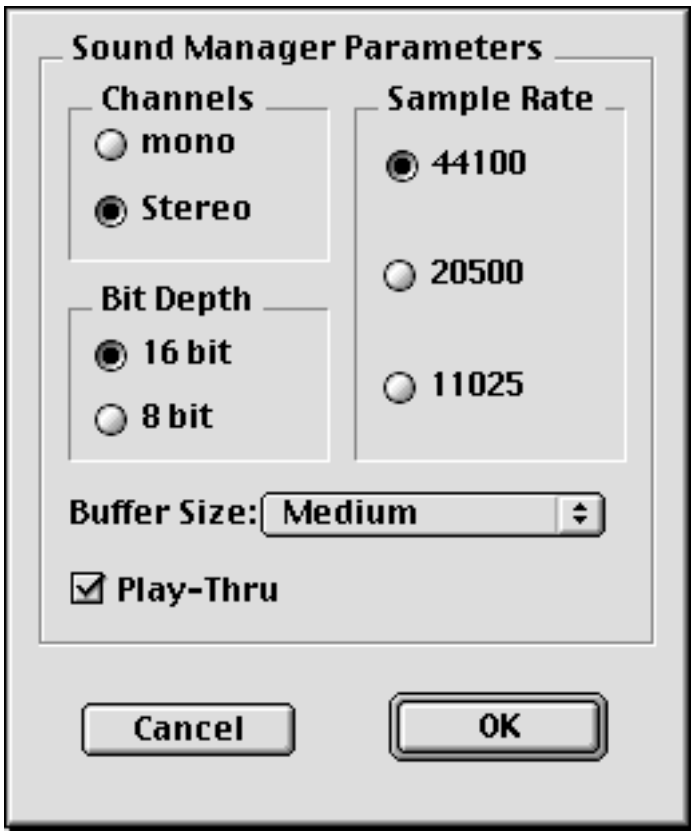
Once a driver has been selected, you can set up the hardware using the Configure Audio System dialog, described next:

Configure Audio System

This option brings up a dialog where you control a number of attributes pertaining to the type of audio HyperEngine will play, and the type of sound files that will be created when you Process to New File. Settings in this dialog will over-ride the settings in your MacOS Sound (or Monitors and Sound) control panel, while HyperEngine is running. Quitting HyperEngine will allow the MacOS to reset the Sound control panel to its previous state.

Configure Audio System dialog attributes vary depending on the driver and hardware currently selected. The HyperEngine currently has direct driver support for the Apple Sound Manager, the Korg 1212 I/O card and the Digidesign Audiomedia II and Audiomedia III cards. Configure Audio System dialog parameters are detailed below; for further information on the Configure Audio System dialog, see the section on Recording a New Audio File.

Sound Manager IO Driver settings:



Channels: Select either Mono (one-channel) or Stereo (two channel) audio.

Bit Depth: Select the number of bits used to describe each sample. For professional audio work, always use the 16-bit setting. If you're exporting files for web or CD-ROM applications where bandwidth is an issue, choose the lower-quality 8-bit setting.

Sample Rate: Select the number of samples that are taken of your audio signal each second. 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.

 For more information on digital audio specifications such as number of channels, bit depth and sample rate, please peruse Appendix B: About Digital Audio Files

Playback Buffer Size: The Playback Buffer is a piece of RAM memory that holds the audio chunk currently being processed by HyperEngine. The Playback Buffer Size pop-up menu allows you to set the size of that buffer, thereby controlling how large or small a piece of audio HyperEngine will be working with during processing and playback. Set the number of samples you wish the buffer to contain by selecting a value from the The Playback Buffer Size pop-up menu.

Large Playback Buffer Size settings introduce more latency (a longer time lag) in audio playback and processing and cause a decrease in the resolution of any Hyperprism tracer automation paths you may record.

Small Playback Buffer Size settings reduce the amount of latency in playback/processing and allow for higher resolution values when recording Hyperprism tracer automation.

If your playthrough or recorded audio sounds like its skipping, stuttering or cutting out, try increasing Buffer Size.

The Playback Buffer uses additional system RAM, beyond the amount allocated in HyperEngine's "Preferred" setting. If you're increasing the buffer size be sure you don't gobble up all the remaining free RAM or unstable operation could result.

Playthrough: The Configure Audio System dialog's Playthrough control allows you to decide whether you want to hear unprocessed Sound Manager playthrough (in conjunction with your processed audio) or only audio that is being processed

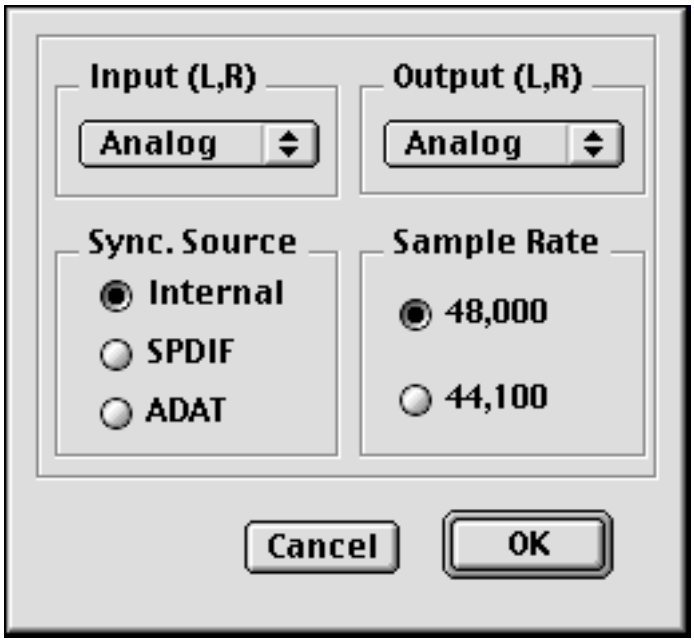
by HyperEngine. Checking this option allows audio to be routed directly from your sound input device to the sound output device. Disabling this option makes it so that you only hear audio input being played in (or recorded into) an active HyperEngine Play-Thru Document.

☞ Don't confuse this option with the functions associated with HyperEngine Play-Thru Documents. The Configure Audio System dialog's Playthrough control is simply creating a direct pipeline between your sound input and output devices. HyperEngine Play-Thru Documents are used whenever you are processing &/or recording live audio with HyperEngine.

☞ In most situations, we suggest leaving the Configure Audio System dialog's Playthrough control disabled (not checked).

☞ If you're hearing an undesirable delayed (echo) effect when doing HyperEngine play-thru or monitoring a recording pass, it's the result of having the Configure Audio System dialog's Playthrough control enabled. To solve this, de-select (check box off) the Playthrough option in the Configure Audio System dialog.

Korg 1212 I/O Driver settings:



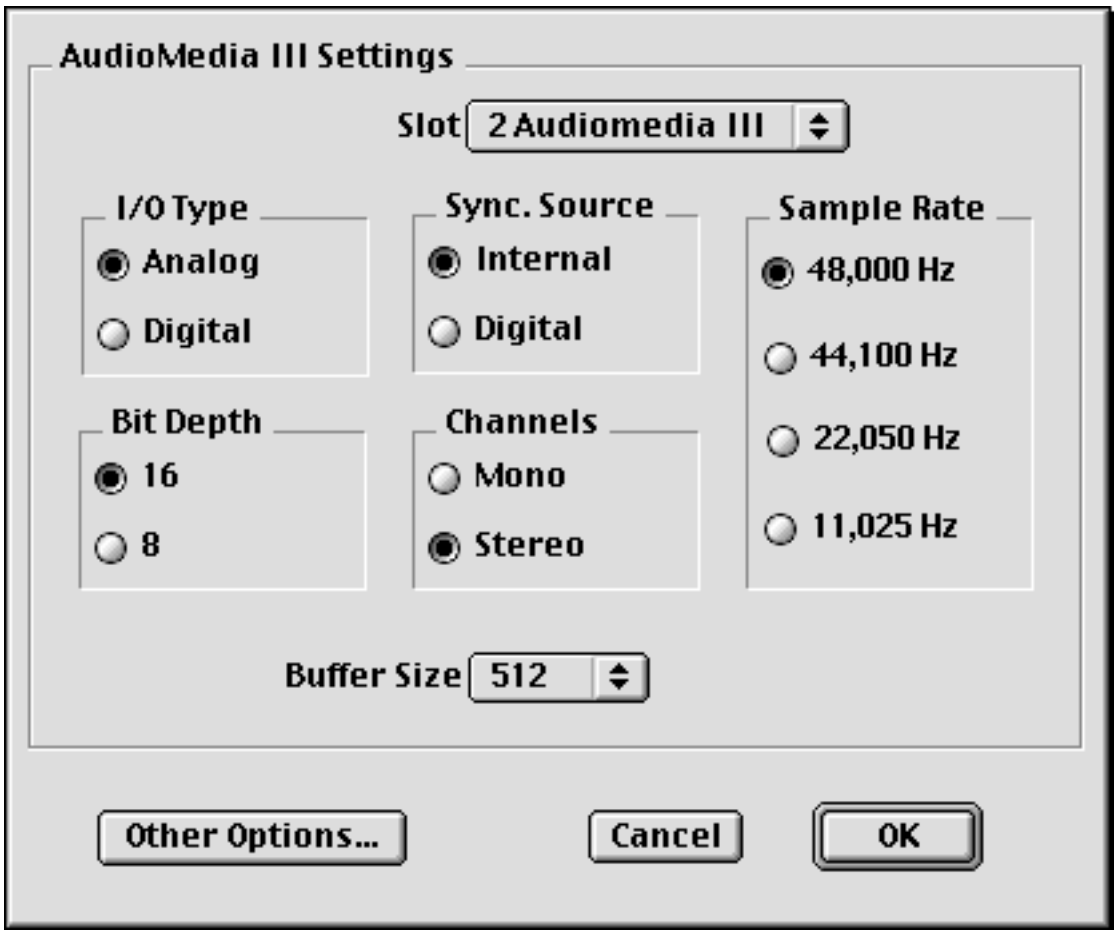
Input (L,R): Select the 1212 I/O card's audio input channels. Choose between the stereo analog channels, the eight ADAT optical channels (any of four stereo pairs), and the SPDIF channels.

Output (L,R): Select the 1212 I/O card's audio output channels. Select from the stereo analog channels, the eight ADAT optical channels (any of four stereo pairs), and the SPDIF channels.

Sync Source: If you're locking your computer to an external time code source, such as an ADAT or other digital recorder, you can select the external clock here. Default setting is Internal; the computer's own clock is the default sync source.

Sample Rate: Choose the desired rate for recording. Korg 1212 I/O offers 48 and 44.1 kHz sample rates.

Digidesign Audiomedia II and Audiomedia III Drivers settings:



Slot: Select the computer slot in which the Audiomedia card you're using resides

I/O Type: Choose between Audiomedia's analog and digital inputs/output channels

Bit Depth: Select either 8- or 16-bit audio

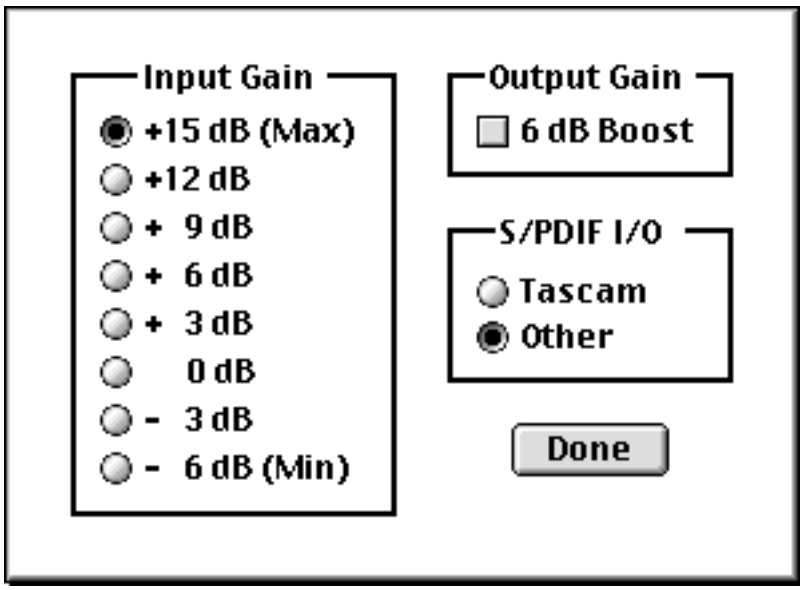
Sync. Source: Allows you to choose between internal sync (using the Mac's own clock) or an external digital word clock.

Channels: Select mono (one channel) or stereo (two-channel) input/output.

Sample Rate: Choose any of the sample rates which are supported by your card: 48, 44.1, 22 or 11 kHz

Buffer Size: Adjust the size of the Audiomedia input buffer.

Other Options: The other options button opens a new dialog, seen below:



Input Gain: Select the input preamp level, range of options is from - 6 dB to + 15 dB

Output Gain: This on/off checkbox allows you to add +6 dB of gain to the output signal, please make sure your board, amp or monitors are calibrated to accept +6 signal before selecting this option.

S/PDIF I/O: Other is the default selection, select the Tascam option only if you're acquiring audio from, or laying off to, a Tascam brand digital recorder, some models of which use an idiosyncratic implementaion of the S/PDIF protocol.

Processing Preferences

New in HyperEngine 2.4 is the Processing Preferences dialog. The processing preferences come into play when HyperEngine is calculating time stretch processes, such as the Pitch-Time Changer or Vari-Speed effect in Hyperprism 2. Here you can select the degree of quality (and resulting speed of calculation) for HyperEngine's internal filter and interpolation routines. If you're doing a time slow-down, having a higher interpolation setting will smooth out the characteristic grainy sound. Higher settings tax the CPU more, which could be a performance consideration when previewing time effects on slower systems. Medium settings are the default; click on the pop-up menus to choose higher or lower settings.

HyperEngine Keyboard Shortcut Summary

Most HyperEngine keyboard shortcuts duplicate functions associated with HyperEngine menu commands. However, there are also a number of "hidden" shortcuts, including spacebar and return to control playback functions, control-drag to create new View Frame, option-drag to paste a copy, shift-drag to resize selection and the arrow keys which control zoom functions.

Note that these key commands & shortcuts apply only to HyperEngine, many of these commands are de-activated when you have a plug-in such as Ionizer or Hyperprism at the top of your desktop.

Learning key commands is the secret to becoming a "Power User," and will result in increased convenience and speed. Click on any of the links below for a more complete description of each key command's function.

Play Functions:

- spacebar ? Play/Pause
- return ? Stop Playback and Return to Start

File/Document Functions:

- command-n ? Create New File Document (open sound file)
- command-o ? Open existing File Reference Document
- command-w ? Close window
- command-s ? Save Reference Document
- command-i ? Open Process Info dialog ("Get Info")
- command-q ? Quit HyperEngine

Editing Functions:

- command-z ? Undo/Redo last edit
- command-x ? Cut current selection from waveform display
- delete ? Erase current selection from waveform display
- command-c ? Copy current selection to clipboard
- command-v ? Paste clipboard contents to current selection or cursor location
- command-a ? Select all of waveform overview
- option-mouse drag ? Copy current selection to new location, leaving original behind

View Frame (Zoom) Functions:

- control-mouse drag ? Define New View Frame (zoom area)
- command-/ (forward slash) ? Switch waveform display to Zoom View/Over-view
- command-= (equals) ? Zoom in (contract View Frame)
- up arrow ? Zoom in (contract View Frame)
- command-- (minus or hyphen) ? Zoom out (expand View Frame)
- down arrow ? Zoom out (expand View Frame)
- left arrow ? Move View Frame (zoom area) to the left
- right arrow ? Move View Frame (zoom area) to the right

Selection Range Functions:

- shift-mouse drag ? Adjust Selection Start/End
- shift-mouse click ? Extend Selection Start/End
- command-; (semicolon) ? Move Selection Range Left
- command-' (apostrophe) ? Move Selection Range Right
- command-[(left bracket) ? Go To Selection Start
- command-] (right bracket) ? Go To Selection End
- command-\ (backslash) ? Fit Selection to Window
- command-r ? Repeat Selection (multiple paste)
- mouse click ? De-select

HyperEngine and the Macintosh Sound Manager

This document describes:

Macintosh Built-in sound

While HyperEngine supports optional high-quality audio cards, they are not required. This is because Apple's PowerMac computers provide good-quality analog audio input and output built into each Macintosh. The input and output ports are typically 1/8" mini stereo jacks, labeled with "mic" and "speaker" icons, although some Macintosh models have separate left and right channel RCA jacks as well. You will have to decide if your Mac's built-in audio provides suitable fidelity for your project.

To control this built-in hardware, Apple provides various system resources. In all likelihood, this software is already installed on your Mac.

Sound Manager extension

For System 8.0 and later the Sound Manager Extension is no longer required by HyperEngine (version 2.4 or later). In System 7.x please use the Sound Manager 3.2.1 or later in your Extensions folder.

Sound Out

By default, HyperEngine will play out of whatever speaker your Mac is using for system beeps and other sounds. The sound out device is set in the **Sound control panel**, by selecting the Sound Out page from the Sound control panel's pop-up menu.

 Can't find your Sound control panel? Try the Monitors and Sound control panel...Later versions of the MacOS have incorporated the sound features with the monitor controls, and omitted direct control over playthrough, bit depth and number of channels. Using either control panel (or both) is fine. We usually have both Sound and Monitors & Sound control panels loaded, and control all audio settings through the Sound one.

If you have any third party audio hardware, such as a Korg or Digidesign card, it should appear in addition to the "Built-in" option (more on this in a moment). You will also find options for setting the number of channels, sample rate and bit-resolution here.

If your Mac is not making any sound (including system alert beeps or startup sounds) double check the sound output jack

on your Mac's back panel. When an audio plug is inserted in this jack, the Mac's internal speaker is automatically disconnected. If you have connected a cable to your Mac's audio output, make sure your external audio system is functioning properly.

Sound In

The Sound control panel also sets the source for sound coming into your Mac. Select the Sound In page from the pop-up menu in the Sound control panel. HyperEngine will use the Sound In device for play-through mode, as well as when recording files to disk for sound-file based processing.

As is the case with the Sound Out setting, if you have any third party audio hardware, such as a Digidesign card, it should appear in addition to the "Built-in" option (more on this in a moment).

 Clicking the Options... button in the Sound Control Panel (Sound Input pop-up in Monitors & Sound control panel) lets you choose between your Mac's internal CD player (if present) and the external audio input (Microphone) connector on the back of your Mac. If you have a built-in Apple CD-ROM drive, try using the AppleCD Audio Player control panel to play audio CDs into HyperEngine for recording or Play-Thru processing.

Using Third-Party Audio Cards

If you have an audio card installed in your Mac, you can select that card for your Mac's audio input and output. Doing so will give you higher fidelity than your Mac's built-in audio hardware, and may also allow you to import and export digital audio from an external source, such as a DAT recorder.

To use an optional audio card with HyperEngine:

- Install the audio card in your Macintosh following the manufacturer's directions.

- Install the manufacturer-provided software. This should place an extension in your System folder to allow Sound Manager to access the new audio hardware. For Digidesign users, this is the Digidesign Sound Drivers extension.

 Note that if you're using Pro Tools we suggest using Digidesign Sound Drivers v.1.4.1 **only**, both earlier and later versions have exhibited stability problems. If you're using an Audiomedia II or III card, you may omit the Digidesign Sound Drivers completely and simply use HyperEngine's own Audiomedia II/III IO Drivers.

- Open the Sound control panel, and go to the Sound Out page by selecting Sound Out from the pop-up menu.

- Click on the Digidesign icon (or other manufacturer's icon, depending on the hardware you have installed).

- Set the sample rate and bit-resolution controls as desired.

- If you will be recording audio into the card, go to the Sound In page of the Sound Control Panel and set the Sound In device to match your audio card.

 Your audio card is now configured to work with HyperEngine and any other audio programs that use the Sound Manager input and output device. Be sure the appropriate cables are connected to your external audio system, or you won't be able to hear anything.

Special Direct Sound Card Support

HyperEngine now includes special custom drivers for the Digidesign Audiomedia II and Audiomedia III cards, plus the Korg 1212I/O card. Ray Gun looks for its drivers in a folder called "HyperEngine IO Drivers." This folder should be in the

same folder, at the same directory level, as the HyperEngine application. This is how they are installed by default, so you generally will never need to worry about their location. If you have a supported sound card, these custom drivers eliminate the need for using special Sound Manager drivers, such as the Digidesign Sound Drivers.

A Note About Latency

 Bear in mind that there's a built-in degree of latency (time lag) to all computer audio systems, and doing live effects processing is one application where you can really notice the inherent latency of your audio card or the Sound Manager. Depending on the type of audio you're processing this latency may or may not present a problem in live performance situations. Going with a smaller buffer size in the Configure Audio System dialog can make a big difference, but there's no way to completely eliminate latency.

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

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

First, make sure you're using the correct serial number. Arboretum's 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, Harmony, 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: [HyperEngine]

Appendix B: About Digital Audio Files

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.

HyperEngine supports several different sampling rates:

- 48 kHz (playback and export only, not Capture)
- 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 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. Hyperprism 2 offers dither, which can be selected when the Process to New File command is selected. Dither adds a small bit of noise to the signal to mask the audible artifacts that occur during bit reduction.

HyperEngine's dither algorithm can also be used for 24 or 16-bit sound file processing. Here's why: HyperEngine does all its effect processing using 32-bit floating point calculations. This has much greater resolution than 24 or 16-bit linear digital audio sound files provide. However, the output of the effect must be converted to a 24 or 16-bit sound file. The use of dither can make an improvement in low-level detail, just as is the case when working with 8-bit sounds. However, the difference between a dithered and non-dithered 24 and 16-bit file is far more subtle than when working with 8-bit audio. Use your ears to determine what sounds best to you.

HyperEngine also offers a variant on dither, which is called Noise Shaping. Noise shaping uses the math errors created in the bit depth truncation as the source for the noise which is then dithered into the signal. Noise Shaping has a different audible quality from regular dither, and provides superior results on musical signals.

 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). HyperEngine supports all three types. HyperEngine 2.4 will allow you to import split-stereo (.L/.R) files into the same File Reference document but it requires pre-mixing the two files into a new interleaved file prior to opening. This import process therefore will eat up significant amounts of additional hard disk space.

File Formats

HyperEngine offers direct support for the Sound Designer II (SDII) and AIFF file formats. Arboretum's PC/Windows products (Hyperprism-DX and Ray Gun) process Wave (.WAV) audio files. The sound quality of these three formats is identical.

With HyperEngine 2.4 we have implemented support for QuickTime 3 features, allowing import of .WAV, .SND,

QuickTime and audio CD tracks. Installation of QuickTime 3 required. These foreign formats are converted to AIFF upon import.

 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 Toast and MasterList CD accept for burning audio CD-Rs. 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.

 HyperEngine does not support the old Sound Designer I mono file format.

[Previous Chapter]
Go to Table of Contents: [HyperEngine]