Digidesign
Plug-Ins Guide

Version 6.2 for HD Systems on Windows or Macintosh
Version 6.1 for TDM or LE Systems on Windows or Macintosh
Version 5.1.1 to 5.3.x for TDM or LE Systems on Macintosh

digidesign

contents

chapter 1

Getting Started With Plug-Ins

Digidesign plug-ins provide a comprehensive set of digital signal processing tools for professional audio production.

This guide explains the use of each of the plug-ins currently available from Digidesign.

These plug-ins include:

◆ Bruno™ & Reso™ cross-synthesis plug-ins

◆ D-Fi™ creative sound design plug-ins

◆ DINR™ intelligent noise reduction

◆ Maxim™ peak limiter/sound maximizer

◆ Reverb One™

◆ SoundReplacer™ drum and sound replacement plug-in

◆ SurroundScope™

▤ *For more information on using plug-ins during mixdown with Pro Tools, see the Pro Tools Reference Guide.*

System Requirements

To use Digidesign plug-ins you need:

◆ A Digidesign-qualified Pro Tools system

– or –

◆ A third-party software application that supports the Digidesign TDM, RTAS, or AudioSuite plug-in standard

Digidesign can only assure compatibility and provide support for hardware and software it has tested and approved. For a list of Digidesign-qualified computers, operating systems, hard drives, and third-party devices, refer to the latest compatibility information on our Web site (www.digidesign.com/compato).

Plug-In Formats

Plug-ins are special-purpose software components that provide additional signal processing functionality to Pro Tools. There are three formats of plug-ins:

- TDM plug-ins (real-time, DSP-based)
- RTAS plug-ins (real-time, host-based)
- AudioSuite plug-ins (non-real-time, file-based processing)

TDM Plug-Ins

(TDM Systems Only)

TDM (Time Division Multiplexing) plug-ins function as track inserts, are applied to audio during playback, and process audio nondestructively in real time. TDM plug-ins are designed for use on Pro Tools TDM systems, and rely on the processing power of Digidesign DSP cards.

The number and variety of TDM plug-ins that you can use simultaneously in a session are limited only by the amount of DSP available. You can increase available DSP by installing additional HD Accel, HD Process, MIX Farm, or DSP Farm cards in your computer. This power-on-demand aspect is a significant advantage of TDM-based systems.

⚠ *MIX Farm and DSP Farm cards are not supported on Pro Tools|HD-series systems. HD Accel and HD Process cards are not supported on Pro Tools|24 MIX-series systems.*

RTAS Plug-Ins

RTAS (Real-Time AudioSuite) plug-ins provide features and functionality similar to their TDM counterparts, but unlike TDM plug-ins, they rely on and are limited by the processing power of your computer. The more powerful your computer, the greater the number and variety of RTAS plug-ins that you can use simultaneously.

Because of this dependence on the CPU or *host processing*, the more RTAS plug-ins you use concurrently in a session, the greater the impact on other aspects of system performance, such as available track count, edit density, and automation latency.

RTAS plug-ins can be used on both Pro Tools TDM and LE systems.

AudioSuite Plug-Ins

Non-real-time AudioSuite plug-ins are not used during playback, but are instead used to process audio files on disk, creating new, rewritten audio files with the effect permanently applied. AudioSuite plug-ins can be used on all Pro Tools systems.

Installing Plug-Ins

The Digidesign Plug-In Installer CD-ROM lets you install and use not only the plug-ins which you have purchased, but also a demo version of other Digidesign plug-ins. Demo versions of plug-ins can be used for the length of their demo period, after which they will expire. Once the demo version of a plug-in has expired, purchase the plug-in and authorize it, or remove it (see "Removing Expired Plug-Ins" on page 4).

Demo Plug-Ins and Authorization

Some plug-ins require an iLok USB Smart Key present in order to run in demo mode. Demo authorization for such plug-ins can only be installed over the internet from Digidesign's Web site (www.digidesign.com) while you launch Pro Tools. Make sure that your Pro Tools-equipped computer is internet-ready and that your iLok USB Smart Key is inserted in an available USB port before you launch Pro Tools to complete this automated process. This is a one-time-only procedure for plug-ins that require it.

Updating Older Plug-Ins

Because the Plug-In Installer CD-ROM contains the latest versions of all Digidesign plug-ins, you can use it to update any plug-ins you already own by installing the desired plug-in from the CD-ROM. If the installed iLok is already authorized, or there is already a valid authorization key on the destination hard disk, you will not be prompted for a new one.

⚠ *Digidesign plug-ins for Pro Tools 5.3 and lower are not supported in Pro Tools 6.0 and higher.*

Installation

To install a plug-in:

1 Insert the Plug-Ins Installer CD-ROM in your CD-ROM drive.

2 Open the Digidesign Plug-Ins directory.

3 Locate and double-click the Install file on the Installer CD-ROM for the plug-in you want to install.

4 Follow the on-screen instructions.

5 When the installation is complete, click Finish (Windows) or Quit (Macintosh).

6 Repeat steps 3 through 5 for additional plug-ins that you want to install.

You will be prompted to authorize the plug-ins when you launch Pro Tools.

Digidesign Registration

Make sure to complete and return the registration card included with your Digidesign plug-in. Registered users will receive periodic software update and upgrade notices. Please refer to the registration card for information on technical support.

Authorizing Plug-Ins

Digidesign plug-ins are authorized using the iLok USB Smart Key from PACE Anti-Piracy. The iLok is similar to a dongle, but unlike a dongle, it is designed to securely authorize multiple software applications from a variety of software developers.

One iLok USB Smart Key is included with Pro Tools|HD core systems. This key can hold over 100 authorizations for all your iLok-enabled software. Once a software authorization is added to an iLok USB Smart Key, you can use the iLok to authorize that software on any computer. (For more information, see the *iLok Usage Guide*.)

Authorizations are added to an iLok using License Cards that have a small punch-out GSM plastic chip. License Cards are specific to each plug-in. You will receive the appropriate License Cards for the plug-ins that you purchase.

To add an authorization to an iLok:

1 Insert the iLok into an available USB port on your computer.

2 Launch Pro Tools. You will be prompted to authorize any installed unauthorized plug-ins.

💡 *If you are already using a demo version of the plug-in, launch Pro Tools before you insert the iLok, then insert the iLok into any available USB port when prompted by Pro Tools.*

3 Follow the on-screen instructions until you are prompted to insert the License Card into the iLok.

4 Separate the License Card—the smaller GSM cutout—from the larger protective card by pulling the cutout up and out with your thumb.

5 Insert the License Card into the iLok. Visually verify that the License Card makes contact with the iLok's metal card reader.

iLok with License Card

6 Follow the on-screen instructions to complete the authorization process.

7 After authorization is complete, remove the License Card from the iLok. (If you have to remove the iLok to remove the License Card, be sure to re-insert the iLok in any available USB port on your computer.)

Removing Expired Plug-Ins

If you let a demo version of a plug-in expire, you should remove it from your system. Otherwise, each time you open Pro Tools you will be prompted with a message that the plug-in has expired.

Windows

To remove an expired plug-in:

1 From the Start menu, choose Settings > Control Panel and double-click Add or Remove Programs.

2 Select the expired plug-in from the list of installed applications and click the Change/Remove button.

3 When removal is complete, click OK to close the window.

Mac OS X

To remove an expired plug-in:

1 Locate the Library/Application Support/Digidesign folder on your Startup drive.

2 Open the Plug-Ins folder or Plug-Ins (Unused) folder and drag the expired plug-in to the Trash.

3 Empty the Trash.

Mac OS 9

To remove an expired plug-in:

1 Open the DAE folder inside your System Folder.

2 Open the Plug-Ins folder or Plug-Ins (Unused) folder and drag the expired plug-in to the Trash.

3 Empty the Trash.

Allocating Additional Memory to Pro Tools

(Mac OS 9 Only)

If you plan to use a large number of TDM plug-ins in addition to the DigiRack TDM plug-ins included with Pro Tools, allocating additional memory to Pro Tools will help ensure reliable system performance.

If enough RAM is available in your computer, allocate 1–2 megabytes of additional RAM to Pro Tools for each non-DigiRack plug-in installed on your system.

To allocate additional memory to Pro Tools:

1 Start Pro Tools so that it can calculate its basic memory allocation.

2 Go to the Finder and choose About This Computer from the Apple menu.

3 If you have 3 MB or more of memory available (as indicated in the Largest Unused Block portion of this window), go to step 4. If you have less than 3 MB of free memory, stop here: Do not allocate additional memory to Pro Tools unless you install additional RAM in your computer.

4 Quit Pro Tools.

5 Open the Pro Tools folder, select the Pro Tools application, and choose Get Info from the Finder's File menu.

6 Choose Memory from the Show menu.

7 Enter the desired amount of memory *above the minimum requirement* in the Preferred Size field. For example, if the Preferred Size field currently says "30410"k and you want to allocate an additional 3 MB of memory (1 megabyte equals 1,024 kilobytes), enter "33482"k into the Preferred Size field.

8 Close the Get Info dialog.

The next time you start Pro Tools, it will use this new memory allocation.

Conventions Used in This Guide

All Digidesign guides use the following conventions to indicate menu choices and key commands:

Convention	Action
File > Save Session	Choose Save Session from the File menu
Control+N	Hold down the Control key and press the N key
Control-click	Hold down the Control key and click the mouse button
Right-click (Windows)	Click with the right mouse button

The following symbols are used to highlight important information:

User Tips are helpful hints for getting the most from your Pro Tools system.

Important Notices include information that could affect your Pro Tools session data or the performance of your Pro Tools system.

Shortcuts show you useful keyboard or mouse shortcuts.

Cross References point to related sections in the Pro Tools Guides.

chapter 2

Working with Real-Time Plug-Ins

Real-time plug-ins process audio nondestructively in real time. They do not alter the original source audio, but only apply their effect during playback.

There are two formats of real-time plug-ins:

TDM Plug-Ins Rely on the processing power of Digidesign DSP cards. TDM plug-ins run only on TDM-equipped Pro Tools systems.

RTAS Plug-Ins Rely on the processing power of your computer. RTAS plug-ins run on Pro Tools TDM and LE systems.

Processing Power Requirements of TDM and RTAS Plug-Ins

TDM and RTAS plug-ins differ in their processing power requirements.

TDM Plug-Ins

Each real-time TDM plug-in that is inserted in a Pro Tools session uses a portion of your system's total available DSP resources. Since these DSP resources reside on the cards that make up your particular Pro Tools hardware configuration, the amount of DSP available depends entirely on the number and type of DSP cards in your system.

You can add more mixing and processing power to your system by installing additional DSP cards, provided you have unused PCI expansion slots in your computer or use a Digidesign-approved Expansion Chassis.

The System Usage window (Windows > Show System Usage) shows how much DSP is available on your system and approximately how it is currently being used.

📄 *For more information on the System Usage window, see the Pro Tools Reference Guide.*
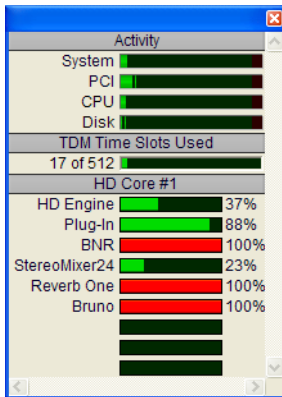
System Usage window

📄 *See Appendix A, "DSP Requirements for TDM Plug-Ins" for the number of instances of each Digidesign TDM plug-in that can be powered by a single DSP chip. DSP usage differs according to card type.*

DSP Sharing Between TDM Plug-Ins

With Digidesign's MultiShell II technology, different types of TDM plug-ins can share the same DSP chip at the same time. This lets you simultaneously use a greater variety of plug-ins by efficiently managing the DSP available on each chip in your system.

In order to take advantage of this capability, plug-ins must be MultiShell II compatible.

The following Digidesign plug-ins are Multishell:

- Maxim
- D-Fi
- SurroundScope

SoundReplacer is an AudioSuite only plug-in.

RTAS Plug-Ins

RTAS plug-ins rely on and are limited by the processing power of your computer's CPU. The more RTAS plug-ins you use concurrently in a session, the greater the impact it will have on other aspects of your system's performance, such as maximum track count, the density of edits possible, and latency in automation and recording.

The CPU meter in the System Usage window (Windows > Show System Usage) shows how much of your computer's processing power is currently being used by Pro Tools, and how much is still available.

📄 *For more information on the System Usage window, see the Pro Tools Reference Guide.*

Improving RTAS Plug-In Performance

You can increase the number of RTAS plug-ins your system can use concurrently by increasing the Hardware Buffer Size and CPU Usage Limit.

Hardware Buffer Size

The Hardware Buffer Size (H/W Buffer Size) controls the size of the hardware cache used to handle host processing tasks such as Real-Time AudioSuite (RTAS) plug-ins.

◆ Lower Hardware Buffer Size settings reduce monitoring latency, and are useful when you are recording live input.

◆ Higher Hardware Buffer Size settings allow for more audio processing and effects, and are useful when you are mixing and using more RTAS plug-ins.

⚠ *In addition to causing slower screen response and monitoring latency, higher Hardware Buffer Size settings can affect the accuracy of plug-in automation, mute data, and timing for MIDI tracks.*

To change the Hardware Buffer Size:

1 Choose Setups > Playback Engine.

2 From the H/W Buffer Size pop-up menu, select the audio buffer size, in samples.

3 Click OK.

CPU Usage Limit

The CPU Usage Limit controls the percentage of CPU resources allocated to Pro Tools host processing tasks.

◆ Lower CPU Usage Limit settings limit the effect of Pro Tools processing on other CPU-intensive tasks, such as screen redraws, and are useful when you are experiencing slow system response, or when running other applications at the same time as Pro Tools.

◆ Higher CPU Usage Limit settings allocate more processing power to Pro Tools, and are useful for playing back large sessions or using more real-time plug-ins.

⚠ *Increasing the CPU Usage Limit may slow down screen response on slower computers.*

To change the CPU Usage Limit:

1 Choose Setups > Playback Engine.

2 From the CPU Usage Limit pop-up menu, select the percentage of CPU processing you want to allocate to Pro Tools. (On dual-processor Macintosh computers, this setting controls the allocation of a single processor.)
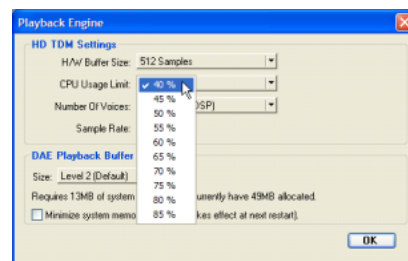
Choosing the CPU Usage Limit

3 Click OK.

⚠ *For more information on Hardware Buffer Size and CPU Usage Limit, see the Pro Tools Reference Guide.*

Delay in Digital Signal Processing

DSP processing in digital audio systems incurs signal delay of varying amounts. Such delays can vary from as short as a few samples to as long as several hundred samples, depending on the type of processing applied.

If you have recorded an instrument on multiple tracks using multiple microphones (a drum kit for example) and process the different tracks with different plug-ins, the tracks may go out of phase. You will then need to compensate for these delays to avoid phase correlation problems.

See Appendix B, "DSP Delays Incurred by TDM Plug-Ins" for information on delays inherent in specific Digidesign TDM plug-ins. See the Pro Tools Reference Guide for a guide to calculating DSP-induced delays.

Compensating for Delay with TimeAdjuster

(TDM Systems Only)

You can compensate for TDM plug-in-induced delays by using the TimeAdjuster plug-in. This plug-in lets you apply a specific number of samples of delay to the signal path of a Pro Tools track. TimeAdjuster provides settings files that apply the correct compensation time in samples for delay introduced by one or more plug-ins. See the *DigiRack Guide* for information about the Time Adjuster plug-in.

Plug-Ins as Inserts

Real-time plug-ins are available as in-line inserts on audio tracks, Auxiliary Inputs, and Master Faders. A maximum of 5 real-time plug-ins can be used per track.

When more than one insert is used on a track, they process the audio in series, each effect being added to the previous one, from top to bottom in the Mix window.

⚠ *If you are using both TDM and RTAS plug-ins on the same track, RTAS plug-ins must occur first in the signal chain, followed by TDM plug-ins. You cannot place TDM plug-ins before RTAS plug-ins.*

Inserts can be used in two ways:

On Single Tracks An insert can be applied to an individual audio track or Auxiliary Input using the Insert Selector on that track.

With in-line inserts, you control the level of effect by adjusting the controls of the plug-in.

As Shared Resources An insert can be used as a shared resource in a send-and-return arrangement by bussing signals from several tracks to an Auxiliary Input, and applying the insert to the Auxiliary Input track. With such an arrangement, you can control the send level for each track and the overall level of the effect can be controlled from the Auxiliary Input track.

Shared arrangements let you make more efficient use of your system's processing power.

⚠ *On TDM systems, RTAS plug-ins can be inserted on audio tracks only. For Auxiliary Inputs or Master Faders, use TDM plug-ins.*

Pre-Fader Operation

Real-time plug-ins function as *pre-fader* inserts (except on Master Fader tracks, where inserts are post-fader), meaning that their input levels are not affected by a track's volume fader.

Real-time plug-ins are pre-fader, but post-disk. This means that if you record to disk with a plug-in inserted on the record track, you will hear the effect of the plug-in, but the effect will not be recorded to disk.

To record with a plug-in effect, create an Auxiliary Input, insert the desired effect on the Auxiliary Input track, then route the Auxiliary Input to the audio track to which you want to record. Alternatively, bounce the audio track with the plug-in after recording in order to write the effected audio to disk.

Mono, Multi-Mono, and Multichannel Plug-Ins

Plug-ins can be used in mono, multi-mono, or multichannel formats, depending on the type of plug-in and whether the destination is a mono or multichannel track.

In general, when working with stereo and greater-than-stereo tracks, use multichannel plug-ins. If a multichannel version of a plug-in is not available, use a multi-mono version.

Plug-in formats

Mono Plug-Ins Used on mono tracks. Some mono plug-ins let you generate a stereo output from a mono channel. Plug-ins that occur on a track after a stereo plug-in are used in stereo as well.

Multi-Mono Plug-Ins Used on stereo or greater-than-stereo multichannel tracks when a multichannel version of the plug-in is not available. Multi-mono plug-ins will analyze and process each channel independently without regard to each other. Controls for all channels are linked by default so that you can adjust them in tandem. You can unlink controls for independent adjustment using the Master Link button. See "Linking and Unlinking Controls on Multi-Mono Plug-ins" on page 16.

⚠ *Multi-mono plug-ins, such as dynamics-based or reverb plug-ins, may not function as you expect. Use the multichannel version of a multi-mono plug-in when available.*

Multichannel Plug-Ins Used on stereo and greater-than-stereo multichannel tracks. On greater-than-stereo multichannel tracks, the controls for all channels are always linked together.

Inserting Plug-Ins on Tracks

To use a real-time plug-in in a Pro Tools session, insert it on a track. Before doing so, make sure the Inserts View is shown in the Mix window.

To show inserts in the Mix window:

■ Choose Display > Mix Window Shows > Inserts View.

To insert a plug-in on a track:

■ Click the Insert Selector on the track and select the plug-in that you want to use.
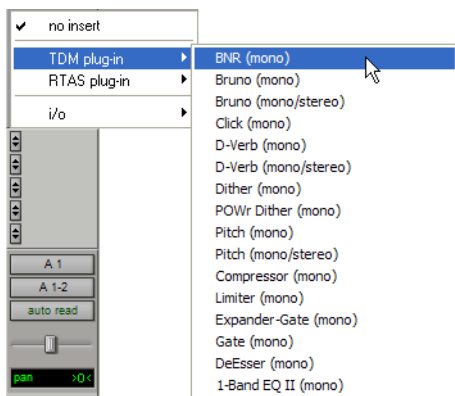
Inserting a plug-in

To remove an insert from a track:

■ Click the Insert Selector and select No Insert.
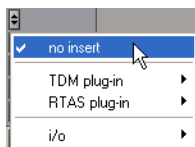
Removing a plug-in

Inserting Plug-Ins During Playback

On Pro Tools|HD and Pro Tools|24 MIX systems (Pro Tools 6.x only) and Pro Tools LE systems, plug-ins can be inserted or removed during playback, with the following restrictions:

◆ Plug-ins cannot be inserted or removed during recording (Pro Tools 6.x only).

◆ A plug-in cannot be dragged to a different insert location during playback or recording. Stop playback to do this.

◆ Plug-ins that change a track's format (a mono-to-stereo plug-in, for example) cannot be inserted or removed during playback. Stop playback to do this.

◆ Plug-ins that contain automation cannot be removed during playback. Stop playback to do this.

◆ Playback must be stopped when enabling plug-in controls for automation.

◆ Side-chain inputs cannot be created during playback. Stop playback to do this.

Moving and Duplicating Inserts

You can move or duplicate an insert by dragging it to a different position on the same track or a different track. Inserts that are moved or duplicated retain their original settings and automation.

To move an insert:

■ Drag the insert to a new insert location.
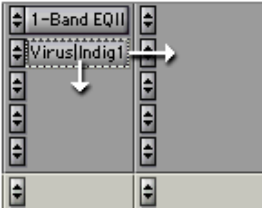
Moving a plug-in

⚠ *Dragging an insert on top of an existing insert will replace it.*

To duplicate an insert:

■ Alt-drag (Windows) or Option-drag (Macintosh) the insert to a new insert location. The duplicated plug-in retains its original settings and automation.

Making Plug-Ins Inactive

Pro Tools lets you set plug-ins as inactive in order to free up DSP resources for other plug-ins and processing. When a plug-in is inactive it retains its assignment, position, and related automation playlists. However, it will not pass audio and does not consume any DSP or TDM resources.

To toggle a plug-in active or inactive:

■ Control-Start-click (Windows) or Command-Control-click (Macintosh) the Insert button.

– or –

Make the track inactive.

📖 *For more information about making tracks inactive, refer to the Pro Tools Reference Guide.*

To toggle plug-ins in the same insert position on all tracks active or inactive:

■ Control-Start-Alt click (Windows) or Command-Control-Option click (Macintosh) an Insert button in the position you want to toggle.

To toggle plug-ins in the same insert position on all selected tracks active or inactive:

■ Control-Start-Alt-Shift-click (Windows) or Command-Control-Option-Shift-click (Macintosh) an Insert button in the position you want to toggle.

The Plug-In Window

The Plug-In window appears whenever you click a plug-in Insert button on a track. This floating window lets you set the controls of any real-time plug-in insert in use on a track.
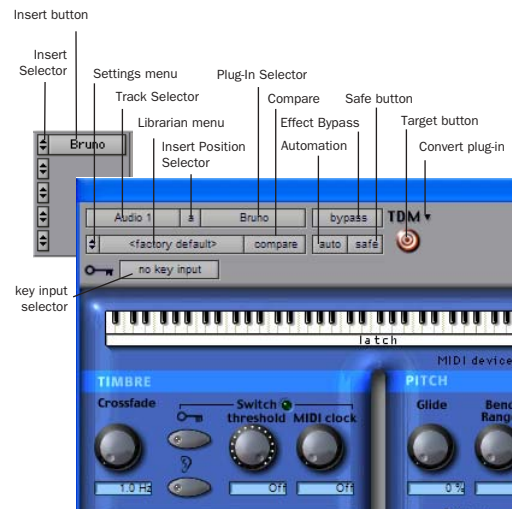
Plug-In window (mono Bruno shown)

Link Enable buttons   Master Link button

Channel Selector

Plug-In window (multi-mono Bruno shown)

Settings Menu Lets you copy, paste, save, and import plug-in settings.

Track Selector Accesses any non-MIDI track in a session.

Librarian Menu Recalls settings files saved in the plug-in's root settings folder or in the current session's Settings folder. See "Using the Librarian" on page 20.

Insert Position Selector Accesses any insert on the current track.

Key Input Selector Lets you select audio on a particular input or bus and route it to trigger the plug-in. This menu only appears on plug-ins that feature side-chain processing. Key inputs are monophonic.

Plug-In Selector Lets you select any real-time plug-in installed in the Plug-Ins folder (that is created when you install Pro Tools).

Compare Toggles between the original saved plug-in setting and any changes you have made to it so you can compare them.

Effect Bypass Disables the currently displayed plug-in. This lets you compare the track with and without the effect.

Auto Lets you enable individual plug-in controls for automation recording. See "Automating Plug-Ins" on page 19.

Safe When enabled, prevents existing plug-in automation from being overwritten.

Convert Plug-In Lets you convert the insert from a TDM plug-in to an RTAS plug-in of the same type (or vice-versa). This feature can only be used on plug-ins that are available in both TDM and RTAS formats.

Target Button When multiple Plug-In windows are open, clicking this button selects that plug-in as the target for any computer keyboard commands.

Phase Invert Button Inverts the phase polarity of the input signal.

Channel Selector Accesses a specific channel within a multichannel track for plug-in control editing. This menu appears only on multi-mono plug-ins inserted on tracks with more than one channel. Shift-clicking this selector opens a separate Plug-In window for each channel of the multichannel track on which the plug-in is inserted.

Master Link Button When enabled, links the controls on all channels of a multi-mono plug-in so that they can be adjusted in tandem.

Link Enable Buttons Let you selectively link the controls of specific channels of a multi-mono plug-in. Each square represents a speaker channel. The Master Link button must be disabled to use the Link Enable buttons. See "Linking and Unlinking Controls on Multi-Mono Plug-ins" on page 16.

LFE Enable Enables plug-in processing of the LFE (low-frequency effects) channel on a multi-channel track formatted for 5.1, 6.1, or 7.1 surround formats. To disable LFE processing, deselect this button. For more information on the LFE channel, refer to the *Pro Tools Reference Guide*.

Opening Plug-In Windows

To open a Plug-In window:

■ Click the plug-in button in the Mix or Edit window channel strip.

By default, each plug-in you open will appear in the same location as a currently open plug-in, replacing it in the same window location.

Opening Multiple Plug-In Windows

Pro Tools normally displays a single Plug-In window from which you can adjust the controls of any plug-in in a session. You can also open additional Plug-In windows for specific plug-ins.

Once you begin working with multiple Plug-In windows, you will need to click the Target button on the plug-in whose controls you want to adjust using keyboard commands.

To open an additional Plug-In window:

■ In the Mix window, Shift-click the Insert button for the additional plug-in.

To open Plug-In windows for each channel of a multi-mono plug-in:

■ Alt-click (Windows) or Option-click (Macintosh) the Channel Selector in the Plug-In window of the multi-mono plug-in.

To close all currently open Plug-In windows:

■ Alt-click (Windows) or Option-click (Macintosh) the close box of any currently open Plug-In window.

Plug-In Window Controls

All plug-ins provide standard Pro Tools controls for track and insert selection, bypass, and other controls, in addition to processor-specific controls.

To select a different plug-in on the same track:

■ Click the Insert Selector and select a plug-in from the pop-up menu.

Choosing a plug-in from the Plug-In window

To choose a different track:

■ Click the Track Selector and select a track from the pop-up menu.

Choosing a track from the Plug-In window

Bypassing Plug-Ins

To bypass a plug-in:

■ Click the Plug-In window's Bypass button.

– or –

■ Control-click (Windows) or Command-click (Macintosh) the plug-in's Insert button in the Mix or Edit window.

💡 *When a plug-in insert is bypassed, the Insert Selector in the Mix window changes its color to blue for easy visual reference. If some, but not all channels of an unlinked multi-mono plug-in are bypassed, the Insert Selector appears purple (Pro Tools 6.x) or half blue and half black (Pro Tools 5.x).*
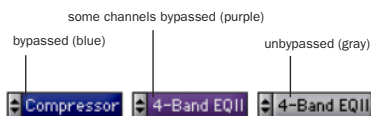
Bypass states indicated by color (Pro Tools 6.x)
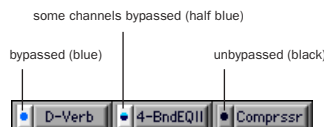
Bypass states indicated by color (Pro Tools 5.3)

Linking and Unlinking Controls on Multi-Mono Plug-ins

When a multi-mono plug-in is used on a multi-channel track, the controls are normally linked. Adjusting the Gain control on one channel, for example, will adjust it for all channels.

If necessary, you can unlink plug-in controls on specific channels of a track and edit them independently. You can also selectively link the controls of specific channels.
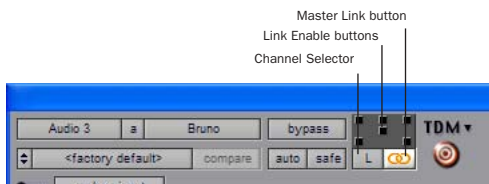
Channel Selector and Link controls

To unlink controls on a multi-mono plug-in:

■ Deselect the Master Link button.

To access controls for a specific channel:

■ Select the channel from the Channel Selector.

To link the controls of specific channels:

1 Deselect the Master Link button if it is not already deselected.

2 Click the Link Enable buttons for the channels whose controls you want to link.

Adjusting Parameters

Editing Parameters Using a Mouse

You can adjust plug-in controls by dragging the control's slider or knob, or by typing a value into the control's text box.

To adjust a plug-in control:

1 Begin audio playback so that you can hear the control changes in real time.

2 Adjust the controls of the plug-in for the effect you want.

3 Closing the plug-in will save the most recent changes.

Keyboard Shortcuts

◆ For finer adjustments, Control-drag (Windows) or Command-drag (Macintosh) the control.

◆ To return a control to its default value, Alt-click (Windows) or Option-click (Macintosh) the control.

Editing Parameters Using a Computer Keyboard

You can use your computer keyboard to edit plug-in controls.

If multiple Plug-In windows are open, Tab and keyboard entry remain focused on the plug-in that is the target window.

Most controls have a corresponding parameter text field directly below it. This displays the current value of the parameter. You can edit the numeric value of a parameter with your computer keyboard.

To type a parameter value:

1 Click on the parameter text that you want to edit.

2 Change the value.

- Type the desired value.

– or –

- To increase a value, press the Up Arrow on your keyboard. To decrease a value, press the Down Arrow on your keyboard.

- In fields that support values in kilohertz, typing "k" after a number value will multiply the value by 1,000. For example, type "8k" to enter a value of 8,000.

- To move forward through the different control fields, press the Tab key. To move backward, press Shift+Tab.

3 Press Enter on the numeric keyboard to input the value and remain in keyboard editing mode.

– or –

Press Enter on the alpha keyboard (Windows) or Return (Macintosh) to enter the value and leave keyboard editing mode.

Using a Key Input for Side-Chain Processing

Some plug-ins feature side-chain processing capabilities. With side-chain processing you can trigger a plug-in from a separate reference track or external audio source. The source used for triggering is referred to as the *Key Input*. Key inputs are monophonic.

A typical use for this feature is to control the dynamics of one audio signal using the dynamics of another signal (the Key Input). For example, a kick drum track could be used to trigger gating of a bass track to tighten it up, or a rhythm guitar track could be used to gate a keyboard pad.

⚠ *RTAS plug-ins do not provide side-chain processing when used on TDM-based systems. If you want to use side-chain processing, use the TDM versions of plug-ins on TDM-based systems.*

Key Input Filters

Some plug-ins feature key high-pass and low-pass filters. These controls let you define a specific frequency range in the Key Input signal with which to trigger the plug-in effect. A common production technique is to use these controls to filter a drum track so that only specific high frequencies (a hi-hat, for example) or low frequencies (a tom or a kick, for example) trigger the effect.

To use a Key Input for side-chain processing:

1 From the Key Input menu, choose the input or bus carrying the audio you want to use to trigger the plug-in.
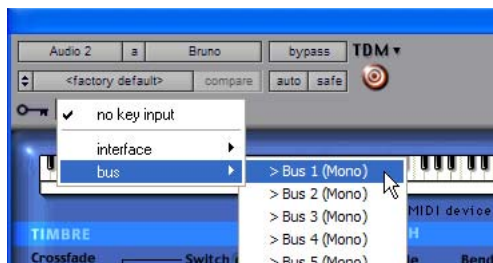
Choosing a Key Input

2 Click External Key to activate side-chain processing.

3 To hear the audio source you have selected to control side-chain input, click Key Listen.

4 To filter the Key Input so that only specific frequencies trigger the plug-in, use the Key HPF and Key LPF controls (if available) to select the frequency range.

5 Begin playback. The plug-in uses the input or bus that you chose as a Key Input to trigger its effect.

6 Adjust the plug-in's Threshold control (if available) to fine-tune Key Input triggering.

7 Adjust other controls to achieve the desired effect.

Automating Plug-Ins

You can automate changes to plug-in controls. Because Pro Tools creates a separate playlist for each plug-in control that you automate, you can later edit and modify each automated control individually. This lets you build up complex automation in stages.

☼ *To create automation for a stereo plug-in with separate controls for each channel, record the automation for one channel then copy and paste it to the other channel.*

To enable plug-in controls for automation:

1 Open the Plug-In window for the plug-in you want to automate.

2 Click the Automation button in the Plug-In window.
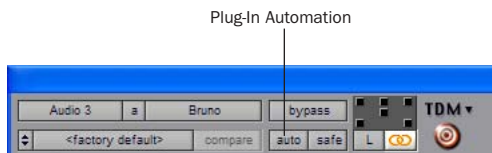
Plug-In Automation

Accessing the Plug-In Automation dialog

– or –

Control-Alt-Start-click (Windows) or Command-Option-Control-click the Track View Selector in the Edit window.

☼ *You can also use this keyboard shortcut to open the Plug-In Automation dialog. Control-Alt-Start-click (Windows) or Command-Option-Control-click (Macintosh) any plug-in control in the Plug-In window, then choose Open Automation Dialog from the pop-up menu.*

3 Choose the controls to automate and click Add. If there are multiple plug-ins on the same track, you can select from among these by clicking their buttons in the Inserts section of this dialog.
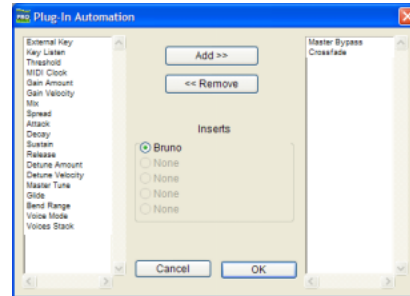
Plug-In automation dialog

4 Click OK to close the Plug-In Automation dialog.

☼ *As an alternative to using the Plug-In Automation dialog, you can enable individual plug-in controls directly from the Plug-In window by Control-Alt-Start-clicking (Windows) or Command-Option-Control-clicking (Macintosh) the control.*
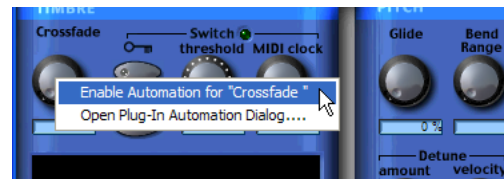
Shortcut for enabling plug-in automation

To automate a plug-in:

1 In the Automation Enable window, make sure plug-in automation is enabled (Windows > Show Automation Enable).

Plug–In Automation Enable button
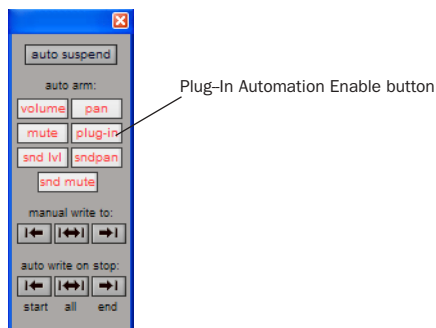
Automation Enable window

2 Select an Automation mode for each track containing plug-ins you want to automate. For the initial automation pass, choose Auto Write.

3 Click Play to begin writing automation, and move the controls you want to automate.

4 When you have finished, click Stop.

After the initial automation pass, you can write additional automation to the track without completely erasing the previous pass by choosing Auto Touch mode or Auto Latch mode. These modes add new automation only when you actually move the control.

Record Safing Plug-In Automation

You can protect plug-in automation from being overwritten by using Automation Safe mode.

To enable plug-in safe mode:

1 Open a plug-in.

2 Click the Safe button so that it is highlighted.

Safe enabled

Automation Safe enabled plug-in

To disable plug-in safe mode:

■ In the Plug-In window, click the Safe button so that it is unhighlighted.

📖 *For more information on creating and editing automation, see the Pro Tools Reference Guide.*

Using the Librarian

The Settings Librarian makes it easy to create your own library of plug-in settings. Using the Librarian and Settings pop-up menus, you can copy, paste, save, and import these settings from plug-in to plug-in, or from session to session.

Librarian menu

Once you create and save settings files to disk (and tell Pro Tools where to find them by assigning their *root* folder) they will appear in the Librarian menu.

The Settings Menu

The Settings menu let you save, copy, paste, and manage plug-in settings files.

Settings menu

Settings menu commands include:

Save Settings Saves the current settings. This command overwrites any previous version of the settings. The setting then appears in the Librarian menu.

Save Settings As Saves the current settings under a different name.

Copy Settings Copies the current settings. You can then apply these settings to the same type of plug-in on a different track by choosing the track in the Plug-In window and pasting the settings with the Paste Settings command.

Paste Settings Pastes settings copied with the Copy Settings command.

Import Settings Imports a settings file from a location other than the Root Settings folder or Session folder.

Delete Current Settings File Permanently deletes the current settings file from disk.

Lock Settings File Prevents the current settings from being overwritten by the Save command. If you attempt to save control settings, you will be required to save them using a different name or disk location.

Set As User Default Defines the current settings as the User Default for a specific plug-in.

Settings Preferences

The Settings Preferences sub-menus are for choosing preferences for saving and importing plug-in settings. They include:

Set Plug-In Default To Sets the default setting to either Factory Default (the standard default setting for that plug-in) or User Default (your custom setting).

Save Plug-In Settings To Selects the folder where plug-in settings are saved. If you choose Session Folder, the settings are saved in a folder named "Plug-In Settings" within the current Session folder. If you choose Root Settings Folder, the settings are saved in the folder you specified with the Set Root Settings Folder command.

Set Root Settings Folder Selects a location or *root* folder for saving and importing settings. This root settings folder must be named "Plug-In Settings." Use this command only if you want to choose a location other than the default Plug-In Settings folder.

In Pro Tools 6.1 and higher, the default location for the Root Plug-In Settings folder is as follows:

Windows Program Files\Common Files\Digidesign\DAE\Plug-In Settings

Macintosh Library\Application Support\Digidesign\Plug-In Settings

⚠ *To save plug-in settings to any location other than the default Plug-In Settings folder, you must first create a folder named "Plug-In Settings" in the new location and then save to that folder.*

Choosing a Destination for Settings

Before you save plug-in settings, select their destination folder.

To select a destination folder:

■ From the Settings pop-up menu, choose Settings Preferences > Save Plug-In Settings To, and choose Session Folder or Root Settings Folder.

If you choose Root Settings folder, Pro Tools saves to the default Plug-In Settings folder unless you have specified a different location for the Root Settings folder.

To select a different Root Settings folder:

1 From the Settings pop-up menu, choose Settings Preferences > Set Root Settings Folder.

2 Select the folder you want to use as your Root folder and click Select.

Managing Settings

Use the Settings pop-up menu to manage settings.

⚠ *Unlinked multi-mono plug-ins have specific rules for settings. See "Editing Settings on Unlinked Multi-Mono Plug-ins" on page 23.*

To save a setting:

1 Choose Save Settings from the Settings pop-up menu.

2 Type a name and click OK. The setting appears in the Librarian menu.

🔲 *Press Control+Shift+S (Windows) or Command+Shift+S (Macintosh) to save plug-in settings.*

To load a previously saved setting:

■ Choose the setting from the Librarian pop-up menu.

To import a setting:

1 Choose Import Settings from the Settings pop-up menu.

2 Locate the settings file you want to import and click Open. Pro Tools loads the setting and copies it to the root destination folder.

To copy a setting:

■ Choose Copy Settings from the Settings pop-up menu.

🔲 *Press Control+Shift+C (Windows) or Command+Shift+C (Macintosh) to copy plug-in settings.*

To paste a setting:

1 Open the destination plug-in.

2 Choose Paste Settings from the Settings pop-up menu.

🔲 *Press Control+Shift+V (Windows) or Command+Shift+V (Macintosh) to paste plug-in settings.*

To create a custom User Default setting:

1 Create and save a setting.

2 Choose Set As User Default from the Settings pop-up menu.

To make a plug-in default to your custom setting:

■ From the Settings pop-up menu, choose Set Plug-In Default To > User Setting.

Editing Settings on Unlinked Multi-Mono Plug-ins

When a multi-mono plug-in is unlinked, importing, copying, pasting, or bypassing settings affects only the currently selected channel.

To apply an operation to all channels of an unlinked plug-in:

■ Hold down the Alt key (Windows) or Option key (Macintosh) while performing the command.

Creating Subfolders for Settings

To make it easier to find specific types of settings, you can subdivide settings files by creating subfolders for them.

If you are using Reverb One, for example, you may want to organize settings by room size. This lets you keep your library well organized with hierarchical menus for settings.

Librarian menu with subfolders for settings

To create a settings subfolder:

1 From the Settings pop-up menu, choose Save Settings.

2 Click the New Folder button and type a name for the subfolder.

3 Name the setting and click Save. The setting is saved within the subfolder.

Working with AudioSuite Plug-Ins

AudioSuite plug-ins differ from TDM and RTAS plug-ins in that they are not used nondestructively in real time, but are instead used to process and modify audio files on disk. Depending on how you configure a non-real-time Audio-Suite plug-in, it will either alter the original source audio file or create an entirely new audio source file.

In Pro Tools, AudioSuite plug-ins are accessed through the AudioSuite menu.

AudioSuite menu

The AudioSuite Window

When you choose a plug-in from the AudioSuite menu, the AudioSuite window appears. This floating window lets you access and edit the controls for any AudioSuite plug-in.
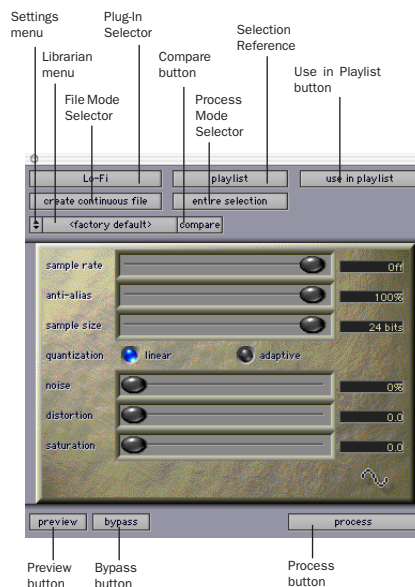
AudioSuite controls

Plug-In Selector

This menu lets you select any AudioSuite plug-in installed in your Plug-Ins folder.
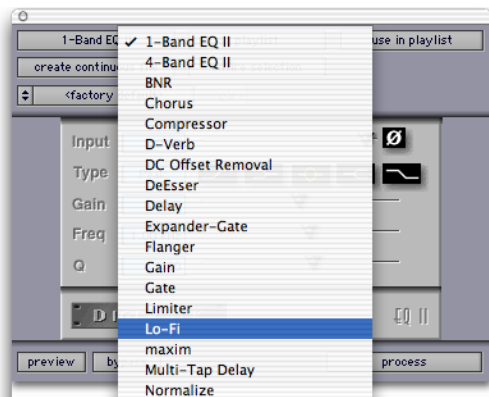
Plug-In Selector pop-up menu

Selection Reference

This menu aims the selected process at either regions selected in an audio track or playlist, or at regions selected in the Audio Regions List.

When you select a region on-screen (in either a track/playlist or the Region List) Pro Tools normally selects both occurrences of the region. Since you may not want to process both, this menu provides a convenient means of limiting the AudioSuite process to regions selected in one or the other.
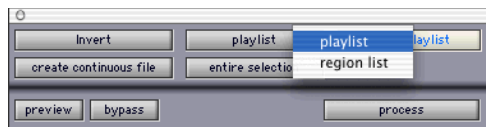
Selection Reference pop-up menu

Playlist Applies AudioSuite processing only to regions currently selected in tracks/playlists in the Edit window. Processing will not be applied to regions in the Audio Regions List.

Region List Applies AudioSuite processing only to regions currently selected in the Audio Regions List. Processing will not be applied to regions located in tracks and playlists in the Edit window.

Use In Playlist Button

This button determines whether the AudioSuite plug-in will replace *all instances* of the selected regions everywhere in the session, or *only* the occurrences of the regions that are currently selected.

Use in Playlist button

Use In Playlist Off When Use in Playlist is disabled, a new, processed version of the selected region will be added to the Regions List. No original regions will be replaced or overwritten anywhere in the session.

Use In Playlist On, Reference = Region List When Use in Playlist is enabled and the Selection Reference pop-up menu is set to Regions List, all copies of the selected region will be replaced everywhere in the session.

Use In Playlist On, Reference = Playlist When Use in Playlist is enabled and the Selection Reference menu is set to Playlist, only regions selected in tracks in the Edit window will be replaced. If the session includes other copies of the region used in other playlists, those copies of the original, unprocessed region will not be replaced.

⚠ *If the Selection Reference pop-up menu is set to Region List, the Use in Playlist button will automatically be disabled to ensure that you do not accidentally replace every occurrence of the region in a session.*

File Mode Selector

This menu lets you select whether or not the AudioSuite plug-in will process the selected audio destructively or nondestructively, and how the original files will be modified.
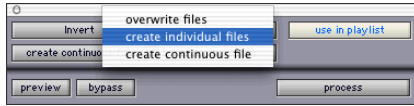
File Mode pop-up menu

There are three options:

Overwrite Files Processes the selected regions destructively, overwriting the original audio. Not all AudioSuite plug-ins can be used destructively.

Create Individual Files Processes the selected regions nondestructively, creating a new audio file for each region. The new audio files are added to the session, leaving the original source audio files in the Regions List unchanged. (Whether the processed audio is added to the current playlist or only to the Regions List, is determined by the Selection Reference setting, as explained above.)

Create Continuous File Processes the selected regions nondestructively and creates a new audio file consisting of the selected regions consolidated into a single, unbroken region. This mode is particularly useful if you are assembling a composite track from multiple takes. This mode is not available when the Selection Reference is set to Region List.

☞ *The Create Continuous File option is not available with some time domain plug-ins. To achieve a similar result, use the Duplicate plug-in to consolidate regions processed with these plug-ins.*

Process Mode Selector

If you have made a selection that includes multiple regions, this pop-up menu lets you specify whether AudioSuite processing is performed on a region-by-region, or entire-selection basis.
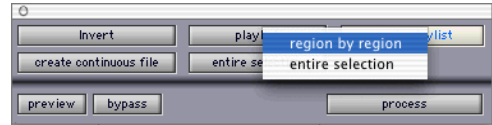
Process Mode Selector

Region by Region Analyzes each region in a selection individually, rather than over the entire multi-region selection as a whole.

Entire Selection Uses the entire selection for analysis. All regions will be analyzed and processed relative to the entire selection.

Chan/Track Process Mode Selector

If you have made a selection that includes regions from multiple tracks, the Normalize AudioSuite plug-in (as well as some third-party AudioSuite plug-ins) lets you choose whether you want to perform processing on a channel-by-channel/track-by-track, or an all-channel/all-tracks basis using the Chan/Track Process Mode Selector.
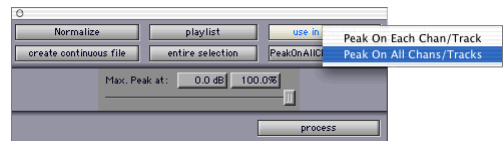
Chan/Track Process Mode Selector pop-up menu

Peak On Each Chan/Track Analyzes and processes each selected track individually. If you apply the Normalize plug-in to multiple tracks or a multichannel track in Peak On Each Chan/Track mode, each channel or track will be normalized independently, without regard to the other selected tracks.

Peak On All Chans/Tracks Uses all currently selected channels and tracks for analysis. If you apply the Normalize plug-in to a multichannel track or multiple tracks in Peak On All Chans/Tracks mode, the tracks will be analyzed as a single entity and regions will be normalized relative to the averaged peak level within all selected channels and tracks.

⚠ *See the DigiRack Plug-Ins Guide for more information on the Normalize AudioSuite plug-in.*

Preview

The Preview button lets you audition the effect of a plug-in before you process the audio. By adjusting the plug-in controls while you listen to this audio preview, you can fine-tune the effect. Not all AudioSuite plug-ins support this feature.

The Preview function routes audio to the specific outputs you have chosen with the Audition pop-up in the Outputs page of the I/O Setup dialog. Make sure you have configured this option correctly for your system or you may not be able to hear previewed audio.

📄 *For more information on using the I/O Setup dialog to configure your system's audio outputs, refer to the Pro Tools Reference Guide.*

Before you use Preview, be aware that:

◆ The performance of the Preview function depends on the speed of your CPU. Faster computers preview AudioSuite effects better than slower computers.

◆ Regardless of how many tracks and regions are currently selected, the Preview button will audition only the first selected stereo track or the first selected pair of mono tracks.

◆ If you are using Region-by-Region processing mode, the Preview function will preview only the first region within a multi-region selection. To hear all selected regions, temporarily select Create Continuous File from the File Mode menu before previewing.

◆ The Preview function is affected by the AudioSuite Buffer Size option (on the Processing page of the Pro Tools Preferences dialog). See "The AudioSuite Processing Preferences Dialog" on page 30.

Bypass

When Bypass is enabled, the selected audio is auditioned without AudioSuite processing. The Bypass button applies only to previewing. It does not affect actual AudioSuite processing.

Process

Clicking this button begins AudioSuite processing of the selected audio. Processing can occur during playback (though it may take slightly longer). Processed files are auto-named with the region or audio file's name plus an acronym for the chosen AudioSuite process.

New files are written to the hard disk specified for that track in the Disk Allocation dialog, or to the same drive as the original file if the region is not currently on a track. See "Auto-File Naming of AudioSuite-Processed Audio" on page 29.

Multichannel Processing

Many Digidesign AudioSuite plug-ins allow up to 48 channels of audio to be processed simultaneously.

Undoing AudioSuite Processing

If you have processed an audio selection nonde-structively, the Undo and Redo commands let you undo the selected AudioSuite process. You can undo or redo an AudioSuite process during audio playback.

⚠ *Undo is not available when a plug-in is con-figured for destructive editing since the pro-cess has already overwritten the source au-dio file.*

Auto-File Naming of AudioSuite-Processed Audio

When new audio files are created as a result of AudioSuite processing, Pro Tools will auto-name these files according to the type of plug-in used. The name of the region determines the prefix, while the type of AudioSuite plug-in determines the suffix.

Auto file-naming follows these rules:

◆ New regions are named beginning with the region name, followed by an abbreviation of the current AudioSuite process, followed by stan-dard Pro Tools file and region numbering.

◆ If a plug-in's File Mode pop-up is set to Over-write, the original region's name will not be changed.

◆ If a plug-in's File Mode pop-up is set to Create Individual Files, the resulting regions will have an abbreviated version of the plug-in name ap-pended to them.

Other AudioSuite Controls

In addition to the standard AudioSuite controls, there are a number of special-purpose controls found on certain plug-ins:

Plug-In Librarian and Settings Menus The Set-tings and Librarian pop-up menus that appear in some AudioSuite plug-ins provide a means of saving, loading, copying, pasting and organiz-ing custom plug-in settings files. See "Using the Librarian" on page 20.

Analyze Button Lets you analyze a selection without actually processing it. "For example, on the DigiRack Gain plug-in, the Analyze feature lets you determine the maximum peak level on a track at a specific gain value before you process the audio."

Side-Chain Input Selector Lets you select a track or bus to be used to trigger processing. In order to use this feature, the key input source audio must occur at the same time as the target audio. See "Using a Key Input for Side-Chain Process-ing" on page 18.

⚠ *Side-chain inputs for plug-ins have no effect on AudioSuite processes when the Selection Reference is set to Region List.*

The AudioSuite Processing Preferences Dialog

Before you begin using AudioSuite plug-ins, use the Preferences dialog (Setups > Preferences > Processing) to configure default AudioSuite options according to your needs. These options include the default dither setting, a dither on/off checkbox, and the AudioSuite buffer size option.
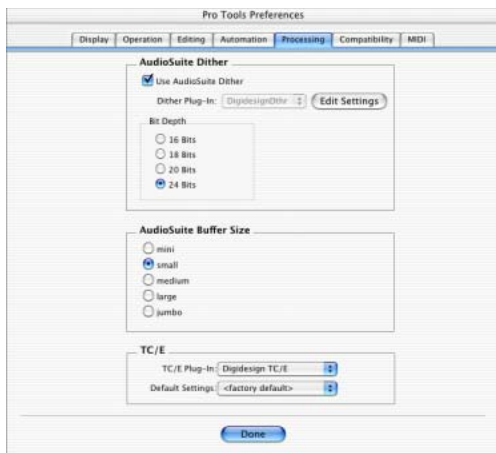
Processing Preferences page

AudioSuite Dither When selected, dither is applied during certain AudioSuite processes (such as Normalize and Gain), using the dither algorithm selected in the Dither Plug-In pop-up. Dither is most useful when working with low-level files that contain a fade in or fade out.

Dither Plug-in AudioSuite plug-ins always create files that inherit the session's bit depth. The Dither Plug-In pop-up menu lets you choose the dither algorithm used for dithering higher-bit source files during AudioSuite processing.

Edit Settings Accesses options for the presently selected Default Dither plug-in. In the case of the Digidesign Dither plug-in, this lets you turn noise shaping on or off.

Refer to the DigiRack Plug-Ins Guide for more information about noise shaping.

Bit Depth Selects the default bit depth used by the dither function during AudioSuite processing. Bit depth can be set to 16-, 18-, 20-, or 24-bit.

Refer to the DigiRack Plug-Ins Guide for more information about dither.

AudioSuite Buffer Size Selects the size of Pro Tools memory buffer for processing and previewing AudioSuite plug-ins. Choices are Mini, Small, Medium, Large, or Jumbo. In general, choosing a smaller buffer speeds up AudioSuite previewing functions. Choosing a larger buffer speeds up AudioSuite processing of audio files.

To configure the AudioSuite Buffer Size:

1 Choose Setups > Preferences.

2 Click Processing.

3 Select an AudioSuite Buffer Size.

Though the default setting will work well for most situations, you may want to set the buffer according to your current task: Before you audition an AudioSuite plug-in, set this buffer to Mini or Small. Then, when you are ready to process a file, reset it to Large or Jumbo.

4 Click Done.

TC/E Lets you select the Time Compression Expansion plug-in and default settings used when editing with the Time Trimmer.

Refer to the Pro Tools Reference Guide for more information about the Time Trimmer.

Using AudioSuite Plug-Ins

AudioSuite plug-ins can be applied to whole regions, partial regions, or selections that are made up of whole and partial regions across one or more tracks.

When audio selected in the Edit window includes partial regions, the regions will automatically be split into two or more regions when they are processed. Processing will occur only on the selection, leaving other regions unchanged.

⚠ *Processing a locked region will cause the region to become unlocked, and processing a muted region will cause it to become unmuted. For more information on locked and muted regions, refer to the Pro Tools Reference Guide.*

Only audio files on locally connected hard drives can be processed with AudioSuite plug-ins. You cannot process audio files on remote hard drives over a network.

Selecting Tracks for AudioSuite Processing

Because AudioSuite processes are performed on the specific regions that you select, it is important that you select only those regions, or parts of regions, you actually want to process.

Selecting a region in the Audio Regions List will automatically select it in a track (if it currently resides on one). This is the default behavior for Pro Tools. However, you can change this using the Region Selection Follows Track Selection, or

Track Selection Follows Regions List Selection options in the Editing page of the Preferences dialog. See the *Pro Tools Reference Guide* for an explanation of these Preferences.

⚠ *When processing a single audio channel with an AudioSuite plug-in, make sure that the plug-in is set to Mono mode (if available).*

Selecting Tracks for Delay or Reverb Processing

Because some AudioSuite effects such as delay and reverb add additional material to the end of the selected audio (a reverb tail or a delay tap), it is very important that you make a selection that is *longer* than the original source material so that the plug-in can write it into the audio file.

If you select only the original material, without leaving additional space at the end, a reverb decay or delay that occurs after the end of the region will be cut off.

To compensate for this, place the region in a track and select the desired audio *plus* an amount of blank space at the end of the region equal to the amount of delay or reverb decay that you have added in the plug-in. The plug-in will then have space at the end of the region in which to write the final delay or decay.

If you select more blank space than you need, you can remove it using the Trim tool after applying AudioSuite processing.

To process audio with an AudioSuite plug-in:

1 Select regions to process in one or more target tracks and/or in the Audio Regions List. Shift-click to select multiple regions. Only regions that are selected will be processed.

2 Choose the desired AudioSuite plug-in from the AudioSuite menu.

3 Click the Preview button to begin playback of the selected material.

4 Adjust the plug-in controls to achieve the effect that you want. These settings will determine how the file is processed and what effect the processing will have on the original regions. Follow these guidelines:

- To process the selected region only in the track in which it appears, select Playlist from the Selection Reference pop-up. Alternatively, if you want to process the selected region in the Audio Regions list only, choose Region List from this pop-up.

- If you want to process and update every occurrence of the selected region throughout your session, enable the Use In Playlist button (and also select Region List from the Selection Reference pop-up). Alternatively, if you do not want to update every occurrence of the selected region, deselect the Use In Playlist button.

- To configure the plug-in for destructive processing, select Overwrite Files from the File Mode pop-up menu. This will overwrite and permanently modify the original source audio files.

- Alternatively, to configure the plug-in for nondestructive processing, select Create Individual Files from the File Mode pop-up menu. This will create new audio files that have been processed with the AudioSuite Plug-in and leave the original source audio files untouched.

- If you have selected multiple regions for processing and want to create a new file that connects and consolidates all of these regions together, select Create Continuous File from the File Mode pop-up menu.

5 Finally, when you are ready, click the Process button.

The selected audio is processed according to the settings you have specified. Pro Tools appends an acronym to the region's name indicating the AudioSuite process that has been applied. The new audio files then appear in your session.

Using AudioSuite Plug-Ins in Stereo

Some AudioSuite plug-ins can be used in either mono or stereo. If you plan to use them in stereo, be aware of the following:

◆ To process a mono track and obtain a stereo result, select the desired track or region plus an empty track or region. Then set the plug-in to Stereo mode and select the Sum Inputs button (if present) to center the dry signal. When you process the audio, the result will be two tracks or regions that represent the right and left channels of the processed audio. You should then pan these tracks hard right and hard left in your mix.

◆ If you set a plug-in to Stereo mode, then select an odd number of Pro Tools tracks for processing (as opposed to an even number), the plug-in will process the selected tracks in pairs, in stereo. However, the last odd, unpaired track will be processed as mono, using the left channel settings of the stereo plug-in. If you want the last track to be processed in stereo, you must select an additional track to pair it with—an empty one if necessary.

D-Fi

D-Fi consists of four separate plug-ins for TDM, RTAS, and AudioSuite. D-Fi plug-ins form a unique sound design tool kit for processing and deconstructing audio in several retro and synthesis-oriented ways.

Lo-Fi

Lo-Fi provides retro and down-processing effects, including:

- Bit-rate reduction
- Sample rate reduction
- Soft clipping distortion and saturation
- Anti-aliasing filter
- Variable amplitude noise generator

Lo-Fi can be used as either a real-time TDM or RTAS plug-in or as a non-real-time AudioSuite plug-in.

⚠ *The multichannel TDM version of the Lo-Fi plug-in is not supported at 192 kHz, use the multi-mono TDM or RTAS version instead.*

Sci-Fi

Sci-Fi provides analog synthesizer-type effects, including:

- Ring modulation
- Frequency modulation

- Variable-frequency, positive and negative resonator
- Modulation control by LFO, envelope follower, sample-and-hold, or trigger-and-hold

Sci-Fi can be used as either a real-time TDM or RTAS plug-in or as a non-real-time AudioSuite plug-in.

⚠ *The multichannel TDM version of the Sci-Fi plug-in is not supported at 192 kHz. Use the multi-mono TDM or RTAS version instead.*

Recti-Fi

Recti-Fi provides additive harmonic processing effects through waveform rectification, and includes:

- Subharmonic synthesizer
- Full wave rectifier
- Pre-filter for adjusting effect frequency
- Post-filter for smoothing generated waveforms

Recti-Fi can be used as either a real-time TDM or RTAS plug-in or as a non-real-time AudioSuite plug-in.

Vari-Fi

Vari-Fi provides a pitch-change effect similar to a tape deck or record turntable speeding up from or slowing down to a complete stop. Features include:

- Speed up from a complete stop to normal speed

- Slow down to a complete stop from normal speed

Vari-Fi is an AudioSuite-only plug-in.

Purposely Degrading Audio

Contemporary music styles, especially hip-hop, make extensive use of retro instruments and processors such as vintage drum machines, samplers, and analog synthesizers. The low bit-rate resolutions and analog "grunge" of these devices are an essential and much-desired part of their sonic signatures. That is why Digidesign created D-Fi.

The D-Fi suite of plug-ins combines the best of these instruments of the past with the flexibility and reliability of the Pro Tools audio production system. The result is a set of sound design tools that let you create these retro sounds without the trouble and expense of resampling audio through 8-bit samplers or processing it through analog synthesizers.

Lo-Fi

Lo-Fi down-processes audio by reducing its sample rate and bit resolution. It is ideal for emulating the grungy quality of 8-bit samplers.

Lo-Fi

Sample Rate

The Sample Rate slider adjusts an audio file's playback sample rate in fixed intervals from 700 Hz to 33 kHz in sessions with sample rates of 44.1 kHz, 88.2 kHz, or 176.4 kHz; and from 731 Hz to 36 kHz in sessions with sample rates of 48 kHz, 96 kHz, or 192 kHz. Reducing the sample rate of an audio file has the effect of degrading its audio quality. The lower the sample rate, the grungier the audio quality.

The maximum value of the Sample Rate control is Off (which effectively means bypass).

⚠ *The range of the Sample Rate control is slightly different at different session sample rates because Lo-Fi's subsampling is calculated by integer ratios of the session sample rate.*

Anti-Alias Filter

The Anti-Alias filter works in conjunction with the Sample Rate control. As you reduce the sample rate, aliasing artifacts are produced in the audio. These produce a characteristically dirty sound. Lo-Fi's anti-alias filter has a default setting of 100%, automatically removing all aliasing artifacts as the sample rate is lowered.

This control is adjustable from 0% to 100%, letting you add precisely the amount of aliasing you want back into the mix. This slider only has an effect if you have reduced the sample rate with the Sample Rate control.

Sample Size

The Sample Size slider controls the bit resolution of the audio. Like sample rate, bit resolution affects audio quality and clarity. The lower the bit resolution, the grungier the quality. The range of this control is from 24 bits to 2 bits.

Quantization

Lo-Fi applies quantization to impose the selected bit size on the target audio signal. The type of quantization performed can also affect the character of an audio signal. Lo-Fi provides you with a choice of *linear* or *adaptive* quantization.

Linear Linear quantization abruptly cuts off sample data bits in an effort to fit the audio into the selected bit resolution. This imparts a characteristically raunchy sound to the audio that becomes more pronounced as the sample size is reduced. At extreme low bit-resolution settings, linear quantization will actually cause abrupt cut-offs in the signal itself, similar to gating. Thus, linear resolution can be used creatively to add random percussive, rhythmic effects to the audio signal when it falls to lower levels, and a grungy quality as the audio reaches mid-levels.

Adaptive Adaptive quantization reduces bit depth by adapting to changes in level by tracking and shifting the amplitude range of the signal. This shifting causes the signal to fit into the lower bit range. The result is a higher apparent bit resolution with a raunchiness that differs from the harsher quantization scheme used in linear resolution.

Noise Generator

The Noise slider mixes a percentage of pseudo-white noise into the audio signal. Noise is useful for adding grit into a signal, especially when you are processing percussive sounds. This noise is shaped by the envelope of the input signal. The range of this control is from 0 to 100%. When noise is set to 100%, the original signal and the noise are equal in level.

Distortion/Saturation

The Distortion and Saturation sliders provide signal clipping control. The Distortion slider determines the amount of gain applied and lets clipping occur in a smooth, rounded manner.

The Saturation slider determines the amount of saturation added to the signal. This simulates the effect of tube saturation with a roll-off of high frequencies.

Fans of *Spinal Tap* will be pleased to know that the Distortion and Saturation controls can be set to eleven for maximum effect.

Output Meter

The Output Meter indicates the output level of the processed signal. Note that this meter indicates the output level of the signal—not the input level. If this meter clips, the signal may have clipped on input before it reached Lo-Fi. Monitor your send or insert signal levels closely to prevent this from happening.

Sci-Fi

Sci-Fi is designed to mock-synthesize audio by adding effects such as ring modulation, resonation, and sample & hold, that are typically found on older, modular analog synthesizers. Sci-Fi is ideal for adding a synth edge to a track.
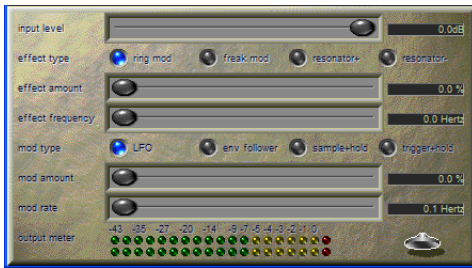
Sci-Fi

Input Trim

Input Trim attenuates signal input level to the Sci-Fi processor. Since some of Sci-Fi's controls (such as the Resonator) can cause extreme changes in signal level, the Input Trim is particularly useful for achieving unity gain with the original signal level. The range of this control is from –12 dB to 0 dB.

Effect Type

Sci-Fi provides four different types of effects:

Ring Mod The Ring Modulator modulates the signal amplitude with a carrier frequency, producing harmonic sidebands that are the sum and difference of the frequencies of the two signals. The carrier frequency is supplied by Sci-Fi itself. The modulation frequency is determined by the Effect Frequency control. Ring modulation adds a characteristic hard-edged, metallic sound to audio.

Freak Mod Freak Mod is a frequency modulation processor that modulates the signal frequency with a carrier frequency, producing harmonic sidebands that are the sum and difference of the input signal frequency and whole number multiples of the carrier frequency. Frequency modulation produces many more sideband frequencies than ring modulation and an even wilder metallic characteristic. The Effect Frequency determines the modulation frequency of the Freak Mod effect.

Resonator+ and Resonator– Resonator+ and Resonator– add a resonant frequency tone to the audio signal. This frequency is determined by the Effect Frequency. The difference between these two modules is that Resonator– reverses the phase (polarity) of the effect, producing a hollower sound than Resonator+. The Resonator can be used to produce metallic and flanging effects that emulate the sound of classic analog flangers.

Effect Amount

Effect Amount controls the mix of the processed sound with the original signal. The range of this control is from 0–100%.

Effect Frequency

Effect Frequency controls the modulation frequency of the ring modulator and resonators. The frequency range is dependent on the effect type. For the Ring Modulator, the frequency range of this control is from 0 Hz to 22.05 kHz. For Freak Mod, the frequency range is from 0 Hz to 22.05 kHz. For Resonator+, the frequency range is from 344 to 11.025 kHz. For Resonator–, the frequency range is from 172 Hz to 5.5 kHz.

You can also enter a frequency value using keyboard note entry.

To use keyboard note entry:

1 Windows-click (Windows) or Option-click (Macintosh) the Effect Frequency slider to display the pop-up keyboard.

2 Select the note on the keyboard that you want for the Effect Frequency.

Sci-Fi Keyboard Note Entry

Modulation Type

Modulation Type determines the type of modulation applied to the frequency of the selected effect. Depending on the type of modulation you select here, the sliders below it will change to provide the appropriate type of modulation controls. If the Mod Amount is set to 0%, no dynamic modulation is applied to the audio signal. The Effect Frequency slider then becomes the primary control for modifying the sound.

LFO Produces a low-frequency triangle wave as a modulation source. The rate and amplitude of the triangle wave are determined by the Mod Rate and Mod Amount controls, respectively.

Envelope Follower Causes the selected effect to dynamically track the input signal by varying with the amplitude envelope of the audio signal. As the signal gets louder, more modulation occurs. This can be used to produce a very good automatic wah-wah-type effect. When you select the Envelope Follower, the Mod Amount slider changes to a Mod Slewing control. Slewing provides you with the ability to smooth out extreme dynamic changes in your modulation

source. This provides a smoother, more continuous modulation effect. The more slewing you add, the more gradual the changes in modulation will be.

Sample+Hold Periodically samples a random pseudo-noise signal and applies it to the effect frequency. Sample and hold modulation produces a characteristic random stair-step modulation. The sampling rate and the amplitude are determined by the Mod Rate and Mod Amount controls, respectively.

Trigger+Hold Trigger and Hold modulation is similar to Sample and Hold modulation, with one significant difference: If the input signal falls below the threshold set with the Mod Threshold control, modulation will not occur. This provides interesting rhythmic effects, where modulation occurs primarily on signal peaks. Modulation will occur in a periodic, yet random way that varies directly with peaks in the audio material. Think of this type of modulation as having the best elements of both Sample and Hold and the Envelope Follower.

Mod Amount and Mod Rate

These two sliders control the amplitude and frequency of the modulating signal. The modulation amount ranges from 0% to 100%. The modulation rate, when LFO or Sample and Hold are selected, ranges from 0.1 Hz to 20 Hz.

If you select Trigger and Hold as a modulation type, the Mod Rate slider changes to a Mod Threshold slider, which is adjustable from –95 dB to 0 dB. It determines the level above which modulation occurs with the Trigger and Hold function.

If you select Envelope Follower as a modulation type, the Mod Rate slider changes to a Mod Slewing slider, which is adjustable from 0% to 100%.

Output Meter

The Output Meter indicates the output level of the processed signal. Note that this meter indicates the output level of the signal—not the input level. If this meter clips, the signal may have clipped on input before it reached Sci-Fi. Monitor your send or insert signal levels closely to prevent this from happening.

Recti-Fi

Recti-Fi provides additive synthesis effects through waveform rectification. Recti-Fi multiplies the harmonic content of an audio track and adds subharmonic or superharmonic tones.
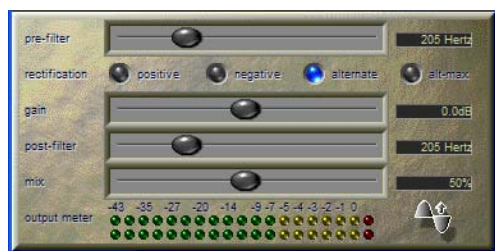
Recti-Fi

Recti-Fi Controls

Pre-Filter

The Pre-Filter filters out high frequencies in an audio signal prior to rectification. This is desirable because the rectification process can cause instability in waveform output—particularly in the case of high-frequency audio signals. Filtering out these higher frequencies prior to rectification can improve waveform stability and the quality of the rectification effect. If you wish to create classic subharmonic synthesis effects, set the Pre-Filter and Post-Filter to a relatively low frequency, such as 250 Hz.

The range of the Pre-Filter is from 43 Hz to 21 kHz, with a maximum value of Thru (which effectively means bypass).
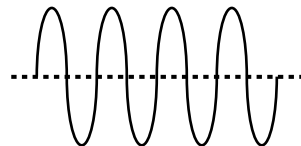
Normal waveform

Rectification

Positive Rectification

This rectifies the waveform so that its phase is 100% positive. The audible effect is a doubling of the audio signal's frequency.

Positive rectification

Negative Rectification

This rectifies the waveform so that its phase is 100% negative. The audible effect is a doubling of the audio signal's frequency.

Negative rectification

Alternating Rectification

This alternates between rectifying the phase of the first negative waveform excursion to positive, then the next positive excursion to negative, and so on, throughout the waveform. The audible effect is a halving of the audio signal's frequency, creating a subharmonic tone.
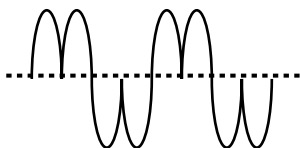
Alternating rectification

Alt-Max Rectification

This alternates between holding the maximum value of the first positive excursion through the negative excursion period, switching to rectify the next positive excursion, and holding its peak negative value until the next zero crossing. The audible effect is a halving of the audio signal's frequency, and creating a subharmonic tone with a hollow, square wave-like timbre.
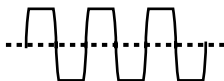
Alt-Max rectification

Gain

Gain lets you adjust signal level before the audio reaches the Post-Filter. This is particularly useful for restoring unity gain if you have used the Pre-Filter to cut off high frequencies prior to rectification. The range of this control is from –18dB to +18dB.

Post-Filter

Waveform rectification, particularly alternating rectification, typically produces a great number of harmonics. The Post Filter lets you remove harmonics above the cutoff frequency and smooth out the sound. This Post-Filter is useful for filtering audio that contains subharmonics. To create classic subharmonic synthesis effects, set the Pre-Filter and Post-Filter to a relatively low frequency.

The range of the Post-Filter is 43 Hz to 21 kHz, with a maximum value of Thru (which effectively means bypass).

Mix

Mix adjusts the mix of the rectified waveform with the original, unprocessed waveform.

Output Meter

The Output Meter indicates the output level of the processed signal. Note that this meter indicates the output level of the signal—not the input level. If this meter clips, the signal may have clipped on input before it reached Recti-Fi. Monitor your send or insert signal levels closely to prevent this from happening.

Vari-Fi

Vari-Fi is an AudioSuite-only plug-in that provides a pitch-change effect similar to a tape deck or record turntable speeding up from or slowing down to a complete stop. Vari-Fi preserves the original duration of the audio selection.

Vari-Fi

Vari-Fi Controls

Speed Up

Speed Up applies a pitch-change effect to the selected audio, similar to a tape recorder or record turntable speeding up from a complete stop. The effect doesn't change the duration of the audio selection.

Slow Down

Slow Down applies a pitch-change effect to the selected audio, similar to a tape recorder or record turntable slowing down to a complete stop. The effect doesn't change the duration of the audio selection.

D-Fi Demo Session

D-Fi includes a demo session that illustrates some of the effects you can produce with Lo-Fi, Sci-Fi, and Recti-Fi.

The D-Fi demo session contains drum, bass, and guitar loops. Memory locations let you quickly locate a particular loop and apply different D-Fi effects.

Before you begin:

1 Open the demo session.

2 Choose Windows > Show Memory Locations.

Memory Locations window

Sci-Fi Examples

The following examples demonstrate Sci-Fi. Follow the instructions in each section below to hear useful applications for this plug-in.
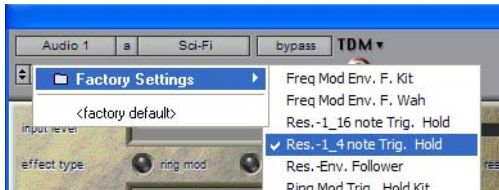
Choosing a Sci-Fi setting

Hi-Hat Loop

1 Click memory location #1, "Hat Loop."

2 Click the Sci-Fi insert on the Master Fader to display Sci-Fi.

3 Press the Spacebar to audition the Hi-Hat loop. Since the Bypass button is enabled, you will hear the loop without Sci-Fi processing.

4 Press the Spacebar to stop the Hi-Hat loop.

5 Choose "Res-1/4 note Trig. & Hold."

6 Deselect the Bypass button to hear the effect.

7 Press the Spacebar to audition the Hi-Hat loop.

8 Listen to the effect. Note how Trigger and Hold is used to cause modulation to follow the amplitude. This provides a much more interesting type of modulation than standard envelope following.

9 Adjust the Mod Threshold to vary the modulation on 1/4 note accents.

10 Choose "Res. –16 note Trig & Hold." This setting demonstrates a similar type of modulation that occurs on 16th notes.

11 Choose "Wah Res-LFO Faux Flange." This setting demonstrates a basic flanging-type effect. Try changing the Rate control and switching to the Resonator+. Experiment with the Mod Type for interesting effects.

Drum Kit Loop

1 Click memory location #2, "Drum Kit Loop."

2 Select Bypass to hear the Drum Kit loop without Sci-Fi processing.

3 Choose "Ring Mod Trig & Hold Kit."

4 Deselect Bypass to hear the effect.

5 Press the Spacebar to audition the Drum Kit loop. This setting uses ring modulation, and trigger and hold for modulation that changes only on audio peaks.

6 Choose "Res-Env. Follower." This setting demonstrates the use of the Envelope Follower to create resonant flanging that modulates and matches the dynamics of the source audio.

7 Choose "Freq. Mod Env. F. Kit." This setting demonstrates frequency modulation.

8 Experiment with the other settings.

9 Finally, click memory location #4, "Bass/Drums Loop." Try each of the Sci-Fi settings with this loop.

Wah Guitar Loop

1 Click memory location #3, "Wah Guitar Loop."

2 Select Bypass to hear this loop without Sci-Fi processing.

3 Choose "Freq Mod Env. Follower Wah."

4 Deselect Bypass to hear the effect.

5 Press the Spacebar to audition this loop.

6 Try each of the Sci-Fi settings with this loop.

Lo-Fi Examples

The examples that follow demonstrate Lo-Fi. Follow the instructions in each section below to hear useful applications for this plug-in.

Before you begin:

1 Open the demo session.

2 Choose Windows > Show Memory Locations.

3 Select Bypass in the Sci-Fi Plug-In window to take it out of the mix.

4 Click the Lo-Fi insert on the master fader to display Lo-Fi.
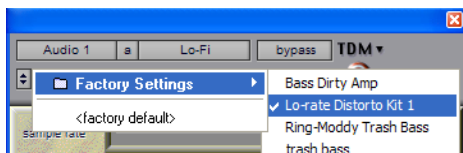
Choosing a Lo-Fi setting

Slam Kit Loop

1 Open the Lo-Fi Plug-In window.

2 Click memory location #7, "Slam Kit Loop."

3 Select Bypass to hear the loop without Lo-Fi processing.

4 Press the Spacebar to audition the loop.

5 Deselect Bypass to hear the effect.

6 Try each Lo-Fi setting with this loop.

The loop has a hip-hop feel, and demonstrates how Lo-Fi can be used to create textures with hard percussive elements.

Drum Kit Loop

1 Click memory location #2, "Drum Kit Loop."

2 Choose "Lo-rate Distorto Kit."

3 Experiment with the Sample Rate, Saturation, and Distortion controls to vary the results.

This loop demonstrates how Lo-Fi can be used to create grungy drums.

Bass Only

1 Click memory location #6, "Bass Only."

2 Choose "Bass Dirty Amp."

3 Use the Bypass button to compare the sound of the processed and unprocessed bass.

This setting simulates a gritty bass amp with limited high-end. Adjust the Saturation and Distortion controls to experiment with the distortion effect.

4 Choose "Trash Bass."

This setting demonstrates an unusual distortion effect. Experiment with bit depth to hear how it affects audio quality.

5 Choose "Ring Moddy Bass."

This setting demonstrates extreme Lo-Fi processing.

Recti-Fi Examples

The examples that follow demonstrate Recti-Fi. Follow the instructions in each section below to hear useful applications for this plug-in.

Before you begin:

1 Open the demo session.

2 Choose Windows > Show Memory Locations.

3 Select Bypass in the Sci-Fi Plug-In window to take it out of the mix.

4 In the Mix window, select Recti-Fi in the place of Lo-Fi on the Master Fader.
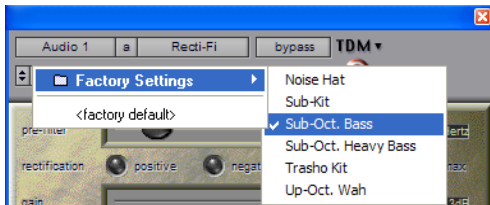
Choosing a Recti-Fi setting

Sub Octave Bass

1 Click memory location #6, "Bass Only."

2 Choose "Sub Octave Bass."

In this setting, the Pre-Filter and Post-Filter are optimized for octave-doubling beneath the bass.

Sub-Oct. Heavy Bass

■ Choose "Sub-Oct. Heavy Bass."

This setting uses Alt-Max rectification to provide more bottom end. Try experimenting with the Mix control and other controls.

Drum Kit Loop

1 Click memory location #2, "Drum Kit Loop."

2 Choose "Sub Kit."

3 Compare the sound of the processed and unprocessed audio using the Bypass button.

This setting demonstrates how to use sub-octave rectification to enhance low frequencies.

Hat Loop

1 Click memory location #1, "Hat Loop."

2 Choose "Noise Hat."

3 Compare the sound of the processed and unprocessed audio using the Bypass button.

This setting demonstrates how Recti-Fi can produce a periodic noise version of the hi-hat signal that varies with the original audio.

4 Adjust the Mix control to hear the signal fully wet.

5 Adjust the Pre-Filter during playback and listen to the results. Automating changes in the Pre-Filter frequency can produce useful effects.

Wah Guitar

1 Click memory location #3, "Wah Guitar."

2 Choose "Up Octave Wah."

3 Play the audio.

This setting produces a signal that is an octave higher than the original and adds some interesting audio artifacts.

4 Experiment with the Mix control, Pre-Filter, and Post-Filter then listen to the results.

Slam Kit Loop

1 Click memory location #7, "Slam Kit Loop."

2 Choose "Trasho Kit."

3 Play the audio.

This setting illustrates the use of Recti-Fi as a basic sound modifier for percussive sounds.

DINR

Digidesign Intelligent Noise Reduction (DINR) consists of a pair of DSP plug-ins (TDM and AudioSuite only) that provide two types of noise removal:

Broadband Noise Reduction (BNR) Provides broadband and narrowband noise reduction for suppressing such unwanted elements as tape hiss, air conditioner rumble, and microphone preamp noise. BNR is available as a real-time TDM and as an AudioSuite plug-in.

⚠ *The TDM version of Broadband Noise Reduction is not supported at sample rates above 96 kHz. The AudioSuite version of Broadband Noise Reduction supports 192 kHz.*

Hum Removal Removes pitch-based noise such as guitar pickup buzz, AC line noise, and fluorescent hum. Hum Removal is available as a real-time TDM plug-in only.

⚠ *Hum Removal is a Macintosh-only plug-in. It is not compatible with Mac OS X, Pro Tools version 6.0 and higher, Pro Tools|HD or Pro Tools|24 MIX cards. It requires a DSP Farm to operate.*

Broadband Noise Reduction

The Broadband Noise Reduction module (BNR) removes many types of broadband and narrowband noise from audio material. It is best suited to reducing noise whose overall character doesn't change very much: tape hiss, air conditioner rumble, and microphone preamp noise. In cases where recorded material contains several types of noise, the audio can be processed repeatedly according to the specific types of noise.

For noise that has more of a pitched quality, the Hum Removal module may yield more satisfactory results. See "Hum Removal" on page 56.
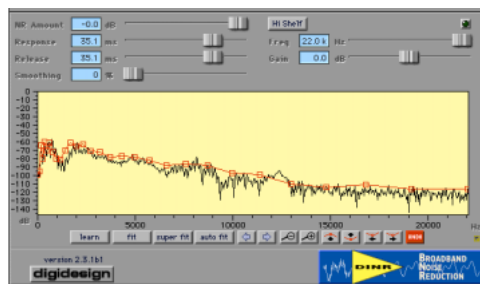
BNR

How Broadband Noise Reduction Works

The Broadband Noise Reduction module uses a proprietary technique called *Dynamic Audio Signal Modeling*™ to intelligently subtract the noise from the digital audio file. Noise is removed with multiple downward expanders that linearly decrease the gain of a signal as its level falls.

Creating a Noise Signature

The first step in performing broadband noise reduction is to create what is called a *noise signature* by selecting and analyzing an example of the noise within the source material. Using this noise signature, a *noise contour line* is created which is used to define the thresholds for the downward expanders that will perform the broadband noise reduction. The noise contour represents an editable division between the noise and non-noise audio signals.

At the same time, DINR also creates a model of what the non-noise audio signal looks like. DINR then attempts to pull apart these two models, separating the bad from the good—the noise from the desired audio. The noise portion can then be reduced or eliminated.

The noise reduction itself is achieved through the use of multiple *downward expanders*. The threshold of these expanders is set so that the noise signal will fall below them and be decreased while the desired audio signal will remain above them, untouched.

The Contour Line

Once the signal level has fallen below the specified Contour Line (which represents BNR's threshold), the downward expanders are activated and decrease the gain of the signal as its level falls. Over five hundred individual downward expanders are used linearly across the audio spectrum to reduce the effects of unwanted noise.

Psychoacoustic Effects of Noise Reduction

One of the psychoacoustic effects associated with broadband noise reduction is that listeners often perceive the loss of noise as a loss of high frequencies. This occurs because the noise in the higher frequency ranges fools the ear into thinking the original signal has a great deal of energy in that range. Consequently, when the noise is removed it feels as if there has been a loss of high-frequency signal. DINR's High-Shelf EQ is useful for compensating for this effect. See "High-Shelf EQ" on page 49.

Limitations of Noise Reduction

It is important to understand that there is a certain amount of trade-off inherent in any type of noise reduction system. Implementing noise reduction means that you have to choose the best balance between the following three things:

◆ The amount of *noise* removed from the signal

◆ The amount of *signal* removed from the signal

◆ The number of *artifacts* added to the signal

DINR gives you a considerable amount of control over the above three elements, and lets you maximize noise reduction while minimizing signal loss and artifact generation. However, as powerful as it is, DINR does have limitations. In particular, there are two instances in which DINR may not yield significant results:

◆ Cases in which the noise components of the audio are so prominent that they obscure the actual signal components of the audio.

◆ Cases in which the noise amplitude of a 24-bit file is less than –96 dB. DINR is not designed to recognize noise that is lower than this level.

Broadband Noise Reduction Controls

The following section describes the Broadband Noise Reduction controls and their use.
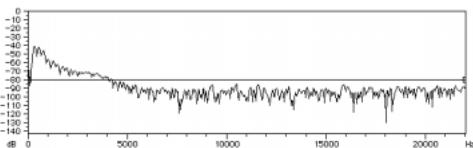
Spectral Graph

The Spectral Graph Displays the noise signature and the editable noise Contour Line. The Spectral Graph's horizontal axis shows frequency, which is displayed in Hertz, from 0 Hz to one-half the current audio file's sample rate. The Spectral Graph's vertical axis shows amplitude, which is displayed in dB, from 0 dB to –144 dB (below full-scale output of the audio).

The Noise Signature The jagged line is a graph of noise. This is called a noise signature. It is created when you use the Learn button in the Broadband Noise Reduction window. Once you have the noise signature of an audio file, you will be able to begin removing the noise by generating and editing a threshold or Contour Line (covered next) between the noise and the desired audio signal.
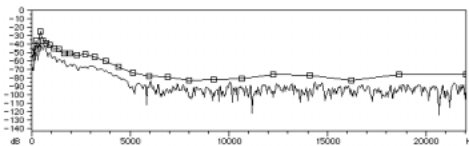
Contour Line

The Contour Line The line with a series of square breakpoints is called the *noise contour line*. The Contour Line is an editable envelope which represents the division between the noise and the non-noise signal in the current audio file. The Contour Line is created by clicking the Fit or AutoFit button in the Broadband Noise Reduction window after you have learned a section of noise. By moving this envelope up or down, or by moving the individual breakpoints, you can modify which signals are removed and which remain.

The noise modeling process treats audio below the line as mostly noise, and audio above the line as mostly signal. Therefore, the higher you move the Contour Line upwards, the more audio is removed. To maximize noise reduction and minimize signal loss, the Contour Line should be above any noise components, but below any signal components.

To fine-tune the broadband noise reduction, move breakpoints at different locations along this line to find out which segments remove the noise most efficiently. Editing the Contour Line

to follow the noise signature as closely as possible will also help maximize noise reduction and minimize signal loss. See "Editing the Contour Line" on page 53.

NR Amount, Response, Release, and Smoothing

Noise Reduction Amount Controls how much the noise signal is reduced. It is calibrated in decibels. A setting of 0 dB specifies no noise reduction. Increasing negative amounts specify more noise reduction. The default value is 0 dB.

In many cases, as much as 20–30 dB of noise reduction can be used to good effect. However, because higher amounts of noise reduction can generate unwanted audio artifacts, you may want to avoid setting the NR Amount slider to its maximum value.

Response Adjusts how quickly the downward expanders and noise reduction process responds to the overall changes in the noise in milliseconds. Depending on the character of the noise, different settings of this control will produce varying amounts of artifacts in the signal, as the modeling process attempts to track the noise signal faster or slower.

The Response speed ranges from 0 ms to 116 ms. A setting of 116 ms (slow) specifies that the modeling process should not attempt to track very fast changes in the noise character. A setting of 0 ms (fast) specifies that the modeling process should attempt to follow every change in the noise character very closely.

A faster setting can yield more noise removal, but it may generate more artifacts. This is similar to how a noise gate produces chatter when attempting to track highly dynamic material. A slower setting will allow slightly less noise removal, but will generate much fewer artifacts.

Release Use in conjunction with the Response slider. It controls how quickly DINR reduces the amount of noise reduction when the amount of noise present in the audio diminishes. Release times range from 0 ms to 116 ms. Like the Response control, a faster setting can yield more noise removal, but it may also generate artifacts. You may want to avoid setting this control to its slowest position, since it will cause the noise tracking to slow to the point that the other controls seem to have no effect.

Smoothing Controls the rate at which noise reduction occurs once the threshold is crossed. It lets you reduce the audibility of any artifacts generated in the modeling process, at the expense of noise reduction accuracy. This is done by limiting the rate of change of the Response and Release controls to the specified Smoothing setting. As soon as the frequency threshold is reached, the full NR amount value is immediately applied according to Response and Release settings. When the frequency threshold is reached, DINR will ramp to the NR Amount level. Settings range from 0 to 100%. A setting of 0% specifies no smoothing. A setting of 100% specifies maximum smoothing.

High-Shelf EQ

The High-Shelf EQ (Hi Shelf) is a noiseless filter that can be applied after noise reduction has been performed in order to compensate for a perceived loss of high-frequency content. It is unique because it operates only on the signal, not on any remaining noise. The Freq slider controls the center frequency of the filter. Values range from 20 Hz to 22 kHz.
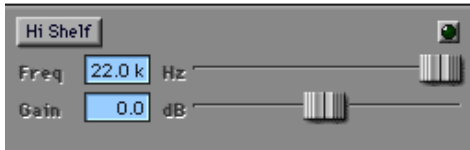
High-Shelf EQ

The Gain slider controls the gain of the filter. Values range from –12 dB to +6 dB. The High-Shelf EQ can be enabled and disabled by clicking the Enable button.

You can also use the High-Shelf EQ to reduce the amount of high frequencies in a signal. This is particularly useful if you are working with older recordings that are band-limited, since the high-frequency content in these is probably made up of noise and not signal.

Learn

Learn

Clicking the Learn button creates a noise signature based on the audio segment currently selected on screen. There are two Learn modes: Learn First Audio mode and Learn Last Audio mode.

Learn First Audio Mode Learn First Audio mode is the default Learn mode. It is designed for use with audio that has an identifiable noise-only section that you can locate and pre-select. To use this mode, locate and select the noise-only portion of the audio, click the Learn button, start playback, and BNR will build a noise signature based on the first 16 milliseconds of audio playback. First Audio Learn mode can be thought of as a trigger-learn mode, since noise capturing is triggered by the first audio that DINR receives.

Learn Last Audio Mode Learn Last Audio mode is designed to let you locate and identify a segment of noise on-the-fly as you listen to audio playback. In this mode, you first Alt-click (Windows) or Option-click (Macintosh) the Learn button, then initiate audio playback. When you hear the portion of audio that contains the noise you want to identify and remove, click the Learn button a second time. BNR will build a noise signature based on the last 16 milliseconds of audio playback. The Spectral Graph displays data in real-time in Learn Last Audio mode.

Fit

Fit

The Fit button computes a noise Contour Line with approximately 30 breakpoints to fit the shape of the current noise signature. The Contour Line can then be edited to more closely fit the noise signature or to reduce specific frequency bands by dragging, adding or deleting breakpoints.

Pressing the Up Arrow or Down Arrow keys on your computer keyboard lets you raise or lower the entire Contour Line, or a selected portion of the Contour Line. The Left/Right arrows lets you move a selection left or right. To select a portion of the Contour Line with multiple breakpoints, Control-drag (Windows) or Command-drag (Macintosh) to highlight the desired area.

After you use the Fit function, BNR will automatically boost the entire Contour Line 6 dB above the noise signature so that all noise components of the audio file are below the Contour Line. You may want to adjust the Contour Line downwards as needed to modify the character of the noise reduction.

Super Fit

Super Fit

The Super Fit button creates a noise Contour Line consisting of over five hundred breakpoints in order to follow the shape of the noise signature more precisely.

Auto Fit

Auto Fit

The Auto Fit function is designed to generate a noise curve for audio that lacks a noise-only portion for DINR to learn. Clicking Auto Fit computes this generic noise curve based on the points contained within the currently selected audio, then fits the Contour Line to it. To use the Auto Fit function, you must first make a selection in the Spectral Graph by Control-dragging (Windows) or Command-dragging (Macintosh).

If the selected audio has both noise and desired sound components, you can generate an approximate noise-only Contour Line by selecting a frequency range that appears to be mostly noise, then pressing the auto fit button. You can then edit the resulting noise Contour Line to optimize the noise reduction.

Scroll Left/Right

Scroll Left/Right

These buttons scroll the Spectral Graph to the left or right, respectively.

To scroll the Spectral Graph (Macintosh only), use Control-Option-Left Arrow or Control-Option-Right Arrow.

Zoom Out/In

Zoom Out/In

Clicking on these buttons zooms in or out of the Spectral Graph. This lets you view and edit the noise contour with greater precision. If you have selected a breakpoint or breakpoints, press Alt+Start+Plus (Windows) or Control+Option+Plus (Macintosh) to zoom the beginning of the selection to the center of the screen. Press Alt+Start+Minus (Windows) or Control+Option+Minus (Macintosh) to zoom back out.

Move Breakpoints Up/Down/Left/Right

Move Breakpoints Up/Down/Left/Right

These arrows behave differently depending on whether or not there is a selection of points along the Contour Line.

No Selection: When there is no selection, the Up and Down arrows move the entire Contour Line up or down by 1 dB, respectively, and the Left and Right arrows scroll the display left and right.

With a Selection: Clicking these buttons moves a selected breakpoint or breakpoints up, down, right, or left. If there is currently a selection in the Spectral Graph, clicking the left and right arrow buttons will move the selected breakpoints left or right. The Up and Down arrows will move the selected breakpoints up or down, respectively. Alt-Start key-clicking (Windows) or Control-Option-clicking (Macintosh) the Arrow keys on your computer keyboard performs the same function.

Undo

UNDO

Undo

Clicking the Undo button undoes the last edit to the Spectral Graph Display. The Undo button does not undo changes made to slider positions.

Using Broadband Noise Reduction

Before you start using BNR, take a moment to think about the nature of the noise in your session and where it's located: Is it on a single track, or several tracks? Is it a single type of noise, or several different types? The answers to these questions will affect how you use BNR.

If there is a single type of broadband noise on a single track, insert the BNR plug-in onto the track. Solo the track to make it easier hear as you remove the noise. If a single track contains different types of noise, you may need to use more than one DINR insert to remove the other types of noise. If multiple tracks contain the same noise, you may want to bus them all to an Auxiliary Input so you can use a single DINR plug-in insert. This will minimize the amount of DSP you use.

To use Broadband Noise Reduction:

1 From the Insert pop-up on the track with the noise, select BNR. The Broadband Noise Reduction window appears.

2 In the Edit window, select the noisiest portion of the track—ideally, a segment with as little of the desired signal as possible. This will make it easier for BNR to accurately model the noise. If the track contains a segment comprised of noise only, select that portion.

3 Start audio playback, and in the Broadband Noise Reduction window, click the Learn button. BNR samples the first 16 milliseconds of the selected audio and creates its noise signature.

– or –

Locate and identify noise on the fly, during playback, using BNR's Learn Last Audio mode. To do this, Alt-click (Windows) or Option-click (Macintosh) the Learn button. Begin playback, and when you hear the segment that you want DINR to sample as noise, click Learn a second time. BNR will build a noise signature based on the 16 milliseconds of audio immediately preceding the second click.

4 Click Fit. BNR will fit a Contour Line to the noise signature just created. If you want to create a Contour Line that follows the noise signature even more precisely, click the Super Fit button. A Contour Line with five hundred breakpoints is created.

5 To audition the effects of the noise reduction interactively, in the Edit window, select a portion of audio containing the noise. Then select Loop Playback from Pro Tools' Options menu and press the Spacebar to begin looped audio playback.

6 Adjust the NR amount slider to reduce the noise by the desired amount. To compare the audio with and without noise reduction, click the Bypass button.

7 To fine-tune the effects of the noise reduction, adjust the Response, Release, and Smoothing sliders to achieve optimal results.

8 To further increase noise reduction, edit the Contour Line. The quickest way to do this is to move the entire Contour Line upwards. In the Spectral Graph, Control-drag (Windows) or Command-drag (Macintosh) to select the entire waveform range. Then click the Move Breakpoint Up button. The higher you move the Contour Line above the noise signature, the more noise is removed. See "Editing the Contour Line" on page 53.

9 If you feel that some of high end frequencies of the audio have been lost due to the noise reduction process, try using the High-Shelf EQ to compensate. To do this, click BNR's Hi Shelf button and adjust the frequency and gain sliders until you are satisfied with the results.

If you are happy with the results of the noise reduction, use the Settings and Librarian menus to save the settings so that you can use them again in similar sessions.

To enable Learn Last Audio mode, Alt-click (Windows) or Option-click (Macintosh) the Learn button. This button flashes red when armed for Learn Last Audio mode. When you hear the target noise, click Learn a second time.

Performing Noise Reduction on Audio that Lacks a Noise-Only Portion

Ideally, audio that you want to perform noise reduction on will have a noise-only portion at the beginning or end of the recording that DINR can analyze and learn. Unfortunately this is not always the case, and in many recordings some amount of signal is always mixed with the noise. Obviously, analyzing such audio will produce a noise signature that is based partially on signal. Luckily, DINR has provisions for cases such as this, and this is where the Auto Fit feature comes in.

If your audio file lacks a noise-only portion for DINR to analyze, you can still obtain reasonable results by selecting and learning a segment of audio that has a relatively low amount of signal and a high amount of noise (as in a quiet passage). By then selecting a frequency range of the noise signature and using the Auto Fit function to generate a generic noise curve, you can recompute the Contour Line based on this selection.

Some editing of the newly generated Contour Line will probably be necessary to yield optimum results, since it is not based entirely on noise from your audio file. See "Editing the Contour Line" on page 53.

To generate a Contour Line for audio that lacks a noise-only portion:

1 In the Edit window, select a segment of audio with a relatively low amount of signal and a high amount of noise.

2 Click the Inserts pop-up on the track with the noise and select BNR. The Broadband Noise Reduction window appears.

**3** Click the Learn button to create a preliminary noise signature.

4 Click the Fit button to fit a Contour Line to it.

5 In BNR's Spectral Graph, Control-drag (Windows) or Command-drag (Macintosh) to make a selection. Select points where the high-frequency noise components are most evident. In general, the flatter areas of the Spectral Graph, are better, since they represent quieter areas where there is probably less signal and more noise.

6 Click the Auto Fit button. DINR computes a generic noise curve and corresponding Contour Line based on your selection. If you want to remove the selection in the Spectral Graph Display, Control-click (Windows) or Command-click (Macintosh) once.

7 Follow the steps given in the previous section removing the noise using the NR Amount slider and other controls.

8 Since the Contour Line is not based entirely on noise from your audio file, you may also want to edit its envelope in order to fine-tune the noise reduction. See "Editing the Contour Line" on page 53.
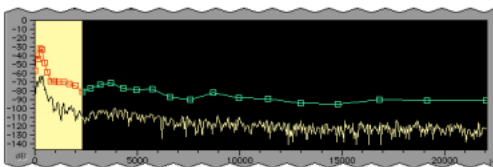
Noise components on the Spectral Graph

Editing the Contour Line

One of the most effective ways to fine-tune the effects of broadband noise reduction is to edit the Contour Line. The Contour Line treats audio below the line as mostly noise, and audio above the line as mostly signal. Therefore, the higher your move the Contour Line upwards, the more audio is removed.

To maximize noise reduction and minimize signal loss, the Contour Line should be above any noise components, but below any signal components. To fine-tune the broadband noise reduction, try moving individual breakpoints at different locations along this line to find out which segments remove the noise most efficiently. For more dramatic results, try moving the entire Contour Line upwards. One drawback of the latter technique is that it will typically remove a considerable amount of signal along with the noise.

Remember that high-frequency noise components are typically more evident in the flatter, lower amplitude areas of the Spectral Graph. Try editing the Contour Line in these areas first.

To hear the changes you make to the Contour Line in real time:

1 Select the target audio in Pro Tools' Edit window. Make sure the selection is at least a second or two in length. If the selection is too short, you won't be able to loop playback.

2 Select Options > Loop Playback.

3 Begin playback.

To edit the Contour Line:

1 To move a breakpoint, click directly on it and drag it to the desired position. Moving a breakpoint higher increases noise reduction at that range. Moving a breakpoint lower decreases noise reduction at that range.
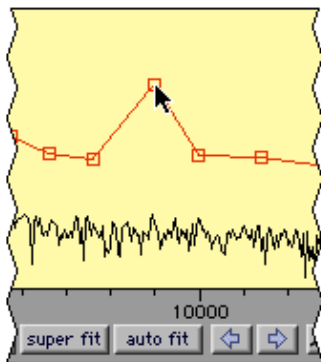
Dragging a breakpoint

2 To move multiple breakpoints, Control-drag (Windows) or Command-drag (Macintosh) to select the desired breakpoints. Click the appropriate Move Breakpoint button (below the Spectral Graph) to move the selected breakpoints in 1 dB increments. Control-Shift-drag (Windows) or Command-Shift-drag (Macintosh) to extend your selection.

Moving selected breakpoints

3 To move the entire Contour Line, Control-drag (Windows) or Command-drag (Macintosh) to select the entire range. Click the appropriate Move Breakpoint button (below the Spectral Graph) to move the selected breakpoints in 1 dB increments. The higher you move the Contour Line above the noise signature, the more noise is removed.

4 To create a new breakpoint, click on the Contour Line.

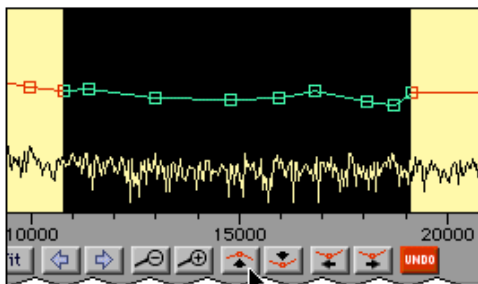**5** To delete a breakpoint, Alt-click (Windows) or Option-click (Macintosh) the breakpoint. As long as you click and hold the mouse, you will delete all breakpoints that the cursor passes over.

Using BNR AudioSuite

BNR AudioSuite is identical to the real-time version of BNR, with the addition of two features to enhance the noise reduction process. These features are:

Audition Lets you listen specifically to the noise portion being removed from the target material. This makes it easier to fine-tune noise reduction settings to maximize noise reduction and minimize signal loss.

Post-processing Applies post-processing to the audio file to help remove undesirable artifacts that are a result of noise reduction.

To enable either of these features, click the corresponding button. To disable them, click again.
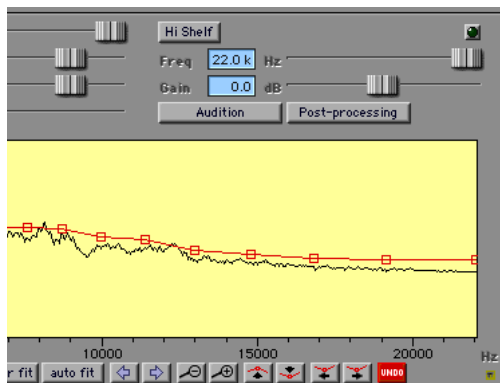
BNR AudioSuite

To process a region with the BNR AudioSuite plug-in:

1 Select the desired regions in the target tracks or the Audio Regions List. Only tracks and regions that are selected will be processed.

2 From the Pro Tools AudioSuite menu, choose BNR.

3 Click Learn to capture the noise signature of the selected material. If you have selected more than one track or region, BNR will build the noise signature based on the first selected track or region when used in Mono mode, or the first two selected track or region when used in Stereo mode.

4 Click Fit or Super Fit to create a Contour Line that matches the noise signature.

5 Click Preview to begin playback of the selected material.

6 Adjust BNR controls and fine-tune the noise reduction using the techniques explained above (See "Using Broadband Noise Reduction" on page 51.)

7 To hear the noise components that are being removed, click Audition. Adjusting BNR's controls while toggling this on and off will let you fine-tune the noise reduction. It also lets you hear exactly how much signal is being removed with the noise, and adjust your controls accordingly.

8 If unwanted artifacts are generated by the noise reduction process, click Post-processing. For best results, set the Response and Release controls to zero.

To begin AudioSuite processing:

1 Adjust the AudioSuite File controls. These settings will determine how the file is processed and what effect the processing will have on the original regions. Here are some guidelines:

◆ To process the selected region only in the track in which it appears, choose Playlist from the Selection Reference pop-up.

– or –

To process the selected region in the Audio Regions List only, choose Region List from this pop-up.

◆ To process and update every occurrence of the selected region throughout your session, enable the Use In Playlist button (and also choose Region List from the Selection Reference pop-up).

– or –

If you do not want to update every occurrence of the selected region, disable the Use In Playlist button.

◆ If you have selected multiple regions for processing and want to create a new file that connects and consolidates all of these regions together, choose Create Continuous File from the File mode pop-up menu.

⚠ *BNR AudioSuite does not allow destructive processing, so the Overwrite Files option is not available in the File mode pop-up menu.*

2 From the Destination Track pop-up, choose the destination for the replacement audio.

3 Click Process.

Hum Removal

(Mac OS 9 Only)

DINR's Hum Removal module is designed to remove many types of harmonic noise from audio material. It is especially effective on noise such as AC line hum, light dimmer buzz, generator noise, guitar pickup noise, fluorescent hum, passing car drones, and noise from computer monitor emissions. The Hum Removal plug-in does this by utilizing a 200-notch comb filter to remove the individual frequency components—fundamental and associated harmonics—from the audio material.

⚠ *Hum Removal is a Macintosh-only plug-in. It is not compatible with Mac OS X, Pro Tools version 6.0 and higher, Pro Tools|HD or Pro Tools|24 MIX cards. It requires a DSP Farm to operate.*

How Hum Removal Works

The Hum Removal module provides harmonic filters specifically designed to remove harmonic pitched noise. Harmonic noise is noise that not only contains a fundamental frequency component, as in the 60 Hz component of AC line noise, but also contains significant components at the odd or even harmonics of the fundamental frequency. If you were to remove only the 60 Hz component of AC line noise, you would still hear significant amounts of noise at 120 Hz (the second harmonic), 180 Hz (the third harmonic), and so on.

Harmonic filters will remove not only the fundamental frequency they are set to, but also all odd and even harmonics of that frequency. The advantage of this type of filter is that it effectively creates hundreds of digital notch filters at exactly the frequencies needed to remove certain types of noise.

However, while a harmonic filter is extremely efficient, it also has drawbacks. The disadvantage of this type of filter is that it can alter the sound by imparting a ringing or phase canceling quality to the sound. The harmonic filters used in the Hum Removal module give you the ability to choose the best balance between noise reduction and the amount of undesired sound alteration that occurs.

Hum Removal Controls

The following section describes the Hum Removal controls and their use.

Hum Removal

Center Frequency

Center Frequency Sets the fundamental frequency of the Harmonic filter. Though the Learn function automatically sets the center frequency to the fundamental frequency of the learned noise, this setting can also be manually edited. The other notches in the harmonic filter are automatically set to the related harmonic frequencies.

Cut Level Determines the depth of the notches within the Harmonic Filter. This lets you specify the amount of hum removal that takes place when the signal falls below the threshold. Higher dB settings result in deeper notch filters and more noise reduction. Lower dB settings produce fewer undesired alterations of the audio. A cut level of 0 dB applies no harmonic filtering. A cut level of –∞ applies maximum hum removal.

Threshold Sets the signal level at which the harmonic filtering occurs. If the input signal level is below the threshold, the harmonic filter is activated and hum is reduced. When the signal level rises above the threshold, the harmonic filter is bypassed, since loud signals psychoacoustically mask the audibility of noise components.

Setting the threshold to 0 dB (its highest level) will cause the harmonic filter to always be active. For optimum hum removal and a minimum of artifact generation, adjust the Threshold so that it is just above the noise level.

Attack Controls how long it takes the Harmonic Filter to reach its full effect once the signal has fallen below the threshold you have set. The default attack value is 15 ms.

Release Controls how long it takes the Harmonic Filter to switch off once the signal has increased above the threshold (determined by your Threshold setting). The default release value is 15 ms.

Learn Analyzes the selected audio and automatically sets the center frequencies of the filters for maximum hum removal. The Learn function saves you the trouble of manually sweeping through the frequency controls in order to find the appropriate fundamental center frequency for hum removal. There are two Learn modes: Learn First Audio mode and Learn Last Audio mode.

Learn First Audio Mode Designed for use with audio that has an identifiable noise-only section that you can locate and pre-select. This is the default Learn mode. To use this mode, locate and select the noise-only portion of the audio, click the Learn button, start Pro Tools playback, and the Hum Removal plug-in will set an appropriate fundamental center frequency for hum removal based on the first 16 milliseconds of au-

dio playback. First Audio Learn mode can be thought of as a trigger-learn mode, since noise capturing is triggered by the first audio that DINR receives.

Learn Last Audio Mode Activated by Alt-clicking (Windows) or Option-clicking (Macintosh) the Learn button. It is designed to let you locate and identify a segment of noise on-the-fly as you listen to audio playback. In this mode, you first Alt-click (Windows) or Option-click (Macintosh) the Learn button, then initiate audio playback. When you hear the portion of audio that contains the noise you want to identify and remove, click the Learn button a second time. The Hum Removal plug-in will set an appropriate fundamental center frequency for hum removal based on the last 16 milliseconds of audio playback.

In order for Hum Removal to learn the audio signal from live inputs, Pro Tools must be in playback mode.

Signal Present LED Lit when an audio signal is received.

Using Hum Removal

As in the BNR module, you must have Hum Removal learn the target noise in an audio track. To learn the noise, the Hum Removal module sweeps through the material's frequency spectrum, scanning the noise to find the most prominent frequency characteristic of the audio signal. When this is completed, it automatically configures itself for maximum hum removal by setting its center frequency control to the fundamental frequency of the most prominent hum.

Before You Begin

Before you start using Hum Removal, think about the nature of the noise and where it is located. Is it on a single track, or several tracks? Is it a single type of noise, or several different types? The answers to these questions will affect how you use Hum Removal.

If there is a single type of hum on a track, insert the Hum Removal plug-in onto the track. Solo the track to make it easier hear as you remove the noise. If a single track contains different types of noise, you may need to use more than one DINR insert to remove the other types of noise. If multiple tracks contain the same noise, you may want to bus them all to an auxiliary input so you can use a single DINR plug-in insert. This will minimize the amount of DSP you use.

To use Hum Removal:

1 Click the Inserts pop-up on the track with the noise and select Hum Removal. The Hum Removal window appears.

2 In the Edit window, select the portion of the track where the hum is strongest—ideally, a segment with as little of the desired signal as possible. This will make it easier for the Hum Removal module to accurately analyze the hum. If the track contains a segment comprised of hum only, select that portion.

3 Begin audio playback, and in the Hum Removal window, click the Learn button. The plug-in samples the first 50 milliseconds of the selected audio and searches for the most prominent hum frequency.

– or –

Locate and identify hum on the fly, during playback, using the Hum Removal module's Learn Last Audio mode. To do this, Alt-click (Windows) or Option-click (Macintosh) the Learn button. Begin playback, and when you hear the

hum that you want to identify, click Learn a second time. The plug-in will identify the hum based on the 50 milliseconds of audio immediately preceding the second click.

4 Adjust the Cut Level slider to increase or decrease the degree to which the hum frequency is cut.

Although the Hum Removal module automatically sets the Cut Level to maximum hum removal, you may still want to adjust this control to minimize any undesired side-effects. To compare the audio with and without noise reduction, click the Bypass button.

5 Fine-tune the hum removal by adjusting the Threshold, Attack, and Release sliders.

chapter 6

Maxim

Maxim is a unique and powerful peak-limiting and sound maximizing plug-in provided in TDM, RTAS, and AudioSuite formats. Maxim is ideal for critical mastering applications, as well as standard peak-limiting tasks.

Maxim offers several critical advantages over traditional hardware-based limiters. Most significantly, Maxim takes full advantage of the random-access nature of disk-based recording to anticipate peaks in audio material and preserve their attack transients when performing reduction.

This makes Maxim more transparent than conventional limiters, since it preserves the character of the original audio signal without clipping peaks or introducing distortion.

⚠ *The multichannel TDM version of Maxim is not supported at 192 kHz. Use the multi-mono TDM or RTAS version instead.*

Maxim features include:

◆ "Perfect attack-limiting" through look-ahead analysis accurately preserves transient attacks and the character of original program material.

◆ A full-color histogram plots input dB history during playback and provides visual feedback for setting threshold level.

◆ A user-adjustable ceiling lets material be level-optimized for recording.

◆ Dither for noise shaping during the final mix-down.

• Online Help (accessed by clicking a control name) provides descriptions of each control.
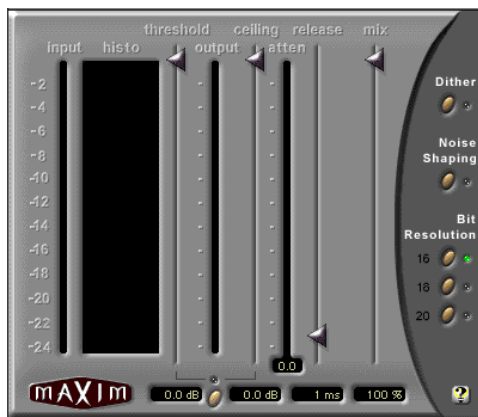
Maxim

About Peak Limiting

Peak limiting is an important element of audio production. It is the process of preventing signal peaks in audio material from clipping by limiting their dynamic range to an absolute, user-selectable ceiling and not letting them exceed this ceiling.

Limiters let you select a threshold in decibels. If an audio signal peak exceeds this threshold, gain reduction is applied, and the audio is attenuated by a user-selectable amount.

Limiting has two main uses in the audio production cycle:

• Adjusting the dynamic range of an entire final mixdown for premastering purposes

• Adjusting the dynamic range of individual instruments for creative purposes

Limiting a Mixdown

The purpose of applying limiting during final mixdown is to flatten any large peaks remaining in the audio material to have a higher average signal level in the final mix. By flattening peaks that would otherwise clip, it is possible to increase the overall level of the rest of the mix. This results in higher average audio levels, potentially better signal to noise ratio, and a smoother mix.

Limiting Individual Instruments

The primary purpose of applying limiting to individual instruments is to alter their dynamic range in subtle or not-so-subtle ways. A common application of this type of limiting is to modify the character of drums. Many engineers do this by applying heavy limiting to flatten the snap of the attack portion of a drum hit. By adjusting the release time of the limiter it is possible to bring up room tone contained in the decay portion of the drum sound.

In some cases, this type of limiting can actually change a drum's character from a very dry sound to a relatively wet sound if there is enough room tone present. This method is not without its drawbacks, however, since it can also bring noise levels up in the source audio if present.

How Maxim Differs From Conventional Limiters

Maxim is superior to conventional limiters in several ways. Unlike traditional limiters, Maxim has the ability to anticipate signal peaks and respond instantaneously with a true zero attack time.

Maxim does this by buffering audio with a 1024-sample delay while looking ahead and analyzing audio material on disk before applying limiting. Maxim can then instantly apply limiting before a peak builds up. The result is extremely transparent limiting that faithfully preserves the attack transients and retains the overall character of the original unprocessed signal.

In addition, Maxim provides a histogram, that displays the distribution of waveform peaks in the audio signal. This provides a convenient visual reference for comparing the density of waveform peaks at different decibel levels and choosing how much limiting to apply to the material.

⚠ *The TDM version of Maxim introduces 1028 samples of delay at 48 kHz into any processed signal. The RTAS version of Maxim introduces 1024 samples of delay. These delays will increase proportionally at higher sample rates. To preserve phase synchronicity between multiple audio sources when Maxim is only applied to one of these sources, use the DigiRack Time Adjuster plug-in to compensate.*

Maxim Controls

Maxim features the following controls and indicators:

Input Level

This meter displays the amplitude of input signals prior to limiting. Unlike conventional meters, Maxim's input meter displays the top 24 dB of dynamic range of audio signals, which is where limiting is typically performed. This provides you with much greater metering resolution within this range so that you can work with greater precision.
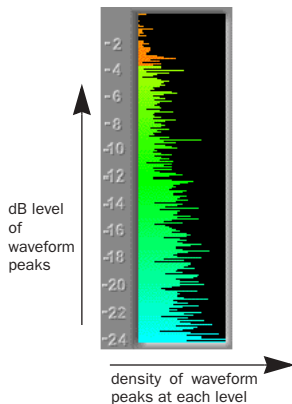
Histogram

The histogram displays the distribution of waveform peaks in the audio signal. This graph is based on audio playback. If you select and play a short loop, the histogram is based on that data. If you select and play a longer section, the histogram is based on that. Maxim holds peak data until you click the histogram to clear it.

The histogram provides a visual reference for comparing the density of waveform peaks at different decibel levels. You can then base limiting decisions on this data.

The X axis of the histogram shows the number of waveform peaks occurring at specific dB levels. The Y axis shows the specific dB level at which these peaks occur. The more waveform peaks that occur at a specific dB level, the longer the X-axis line. If there appears to be a pronounced spike at a certain dB level (4 dB for example), it means that there are a relatively large number of waveform peaks occurring at that level. You can then use this information to decide how much limiting to apply to the signal.

By dragging the Threshold slider downwards, you can visually adjust the level at which limiting will occur. Maxim displays the affected range in orange.

dB level of waveform peaks

density of waveform peaks at each level

Histogram

Threshold

This slider sets the threshold level for limiting. Signals that exceed this level will be limited. Signals below it will be unaffected. Limited signal peaks are attenuated to match the threshold level, so the value that you set here will determine the amount of reduction applied.

Output

This meter displays the amplitude of the output signal. The value that appears here represents the processed signal after the threshold, ceiling, and mixing settings have been applied.

Ceiling

This slider determines the maximum output level. After limiting is performed you can use this slider to adjust the final output gain. The value that you set here will be the absolute ceiling level for limited peaks.

Attenuation

This meter displays the amount of gain reduction being applied over the course of playback, with the maximum peak displayed in the numeric readout at the bottom of the meter. For example, if the numerical display at the bottom of the Attenuation meter displays a value of 4 dB, it means that 4 dB of limiting has occurred. Since this is a peak-hold readout, you can temporarily walk away from a session during playback and still know the maximum gain reduction value when you come back. To clear the numeric readout, click it with the mouse.

Release

This slider sets how long it takes for Maxim to ease off of its attenuation after the input signal drops below the threshold level. Because Maxim has an attack time of zero milliseconds, the release slider has a very noticeable effect on the character of limiting. In general, if you are using heavy limiting, you should use proportionally longer release times in order to avoid pumping that may occur when Maxim is forced to jump back and forth between limited and unlimited signal levels. Lengthening the release time has the effect of smoothing out these changes in level by introducing a lag in the ramp-up or ramp-down time of attenuation. Use short release times on material with peaks that are relatively few in number and that do not occur in close proximity to each other. The Release control has a default value of 1 millisecond.

Mix

This slider sets the ratio of dry signal to limited signal. In general, if you are applying Maxim to a main output mix, you will probably want to set this control to 100% wet. If you are applying heavy limiting to an individual track or element in a mix to modify its character, this control is particularly useful since it lets you add precisely the desired amount of the processed effect to the original signal.

Link Button

When depressed, this button (located between the Threshold and Ceiling numeric readouts) links the Threshold and Ceiling controls. These two sliders will then move proportionally together. As you lower the Threshold control, the Ceiling control is lowered as well. When these controls are linked you can conveniently compare the effect of limiting at unity gain by clicking the Bypass button.

Link button

Link button

Dither

When selected, this applies dither. Dither is a form of randomized noise used to minimize quantization artifacts in digital audio systems. Quantization artifacts are most audible when the audio signal is near the low end of its dynamic range, such as during a quiet passage or fade-out.

Applying dither helps reduce quantization noise that can occur when you are mixing from a 24-bit TDM environment to a 16-bit destination, such as CD-R or DAT. If you are using Maxim on a Master Fader during mixdown, Maxim's built-in dither function saves you the trouble and DSP resources of having to use a separate Dither plug-in.

If Dither is disabled, the Noise Shaping and Bit Resolution controls will have no effect.

Noise Shaping

When selected, this applies noise-shaped dither. Noise shaping biases the dither noise to less audible high frequencies so that it is not as readily perceived by the ear. Dither must be enabled in order to use Noise Shaping.

Bit Resolution

These select dither bit resolution. In general, set this control to the maximum bit resolution of your destination media.

◆ 16-bit is recommended for output to digital devices such as DAT recorders and CD recorders since they have a maximum resolution of 16-bits.

◆ 18-bit is recommended for output to analog devices if you are using an 888 I/O or 882 I/O Audio Interface since the 18-bit setting lets you obtain the maximum quality available from the 18-bit digital-to-analog converters of these devices.

◆ 20-bit is recommended for output to digital devices that support a full 20-bit recording data path. Use this setting for output to analog devices using an 882|20 I/O Audio Interface. It is also recommended for use with digital effects devices that support 20-bit input and output, since it provides for a lower noise floor and greater dynamic range when mixing 20-bit signals directly into the TDM environment.

Using Maxim

Following are suggestions for using Maxim most effectively.

To use Maxim:

1 Insert Maxim on the desired track.

2 Select the portion of the track containing the most prominent audio peaks.

3 Loop playback and look at the data displayed by the histogram and attenuator meter.

4 Select the Link button to link the Threshold and Ceiling controls. You can then adjust these controls together proportionally and, using the Bypass button, compare the audio with and without limiting.

5 Adjust the Threshold downwards until you hear and see limiting occur, then bring the Threshold back up slightly until you have roughly the amount of limiting you want.

6 Periodically click and clear the attenuation meter to check attenuation. In general, applying 2 dB to 4 dB of attenuation to occasional peaks in pop-oriented material is appropriate.

7 Use the Bypass button to compare the processed and unprocessed sound and to check if the results are acceptable.

8 Avoid pumping effects with heavier limiting by setting the Release slider to longer values.

9 When you get the effect you want, deselect the Link button and raise the output level with the Ceiling slider to maximize signal levels without clipping.

In general, a value of 0.5 dB or so is a good maximum ceiling. Don't set the ceiling to zero, since the digital-to-analog convertors on some DATs and CD players will clip at or slightly below zero.

If you are using Maxim on an output mix that will be faded out, enable the dithering options you want to improve the signal performance of the material as it fades to lower amplitudes.

Maxim and Mastering

If you intend to deliver audio material as a 24-bit audio file on disk for professional mastering, be aware that many mastering engineers prefer material delivered without dither or level optimization.

Mastering engineers typically want to receive audio material as undisturbed as possible in order to have leeway to adjust the level of the material relative to other material on a CD. In such cases, it is advisable to apply only the limiting that you find creatively appropriate—adding a little punch to certain instruments in the mix, for example.

However, if you intend to output the material to DAT or CD-R, use appropriate limiting and add dither. Doing so will optimize the dynamic range and preserve the activity of the lower, or least significant bits in the audio signal, smoothly dithering them into the 16-bit output.

Bruno and Reso

Bruno and Reso are a pair of TDM plug-ins that process audio using a sound generation technique known as *cross-synthesis*.

Cross-synthesis generates complex sound textures by using an audio track as a tone source then applying a variety of synthesizer-type effects to that tone source.

Bruno and Reso each use a different sound generation method:

◆ Bruno uses *time-slicing*, a technique whereby timbres are extracted from the source audio during playback and crossfaded together. This crossfading between signals can create a rhythmic pulse in the sound as the timbre changes.

◆ Reso uses a *resonator,* which adds harmonic overtones to the source audio through a short signal delay line with a feedback loop.

In both cases, the processed sound can then be played in real time or sequenced using the MIDI recording and playback capabilities of Pro Tools.

Maximum Voices with HD Accel Card

Bruno and Reso on Pro Tools|HD systems equipped with an HD Accel card offer up to 62 voices of polyphony at the maximum voice setting (at 44.1 kHz and 48 kHz).

Bruno features include:

• Time-slice tone generation with adjustable crossfade

• Polyphony: Up to 62 voices of polyphony (on Pro Tools|HD Accel systems)

• Multi-voice detuning

• Editable ADSR envelope generator

• Portamento

• Velocity-sensitive gain and detuning

• Time-slice switching using envelope triggering or MIDI beat clock

• Voice-stacking

• Side-chain input for control using an external audio source

• Supports sample rates up to 192 kHz

• Online help

Reso features include:

- Harmonic resonance generation
- Up to 62 voices of polyphony (on Pro Tools|HD Accel systems)
- Multi-voice detuning
- Resonant low-pass filter
- Editable ADSR envelope generator
- Portamento
- Velocity-sensitive resonance, damping, gain, and detuning
- Harmonic switching using envelope triggering or MIDI beat clock
- Voice-stacking
- Side-chain input for control using an external audio source
- Supports sample rates up to 192 kHz
- Online help

DSP Requirements

Bruno and Reso each require one full DSP chip on a Pro Tools|HD, MIX, or DSP Farm card.

DSP and Voice Polyphony

The maximum number of Bruno/Reso voices available per DSP chip depends on the sample rate of the session and the type of DSP cards in your system.

HD Accel On Pro Tools|HD systems equipped with an HD Accel card, Bruno and Reso provide up to 62 voices at their maximum setting. The 62-voice versions of Bruno and Reso require one entire DSP chip on an HD Accel card. Polyphony is reduced by half for sessions at 88.2 kHz and 96 kHz.

HD Core and HD Process On Pro Tools|HD systems not equipped with an HD Accel card, Bruno and Reso provide a maximum of 24 voices of polyphony. Polyphony is reduced by half for sessions at 88.2 kHz and 96 kHz (up to 14 voices).

MIX Cards On Pro Tools|24 MIX systems, a maximum of 24 voices of polyphony are available per chip on MIX cards. MIX Farm cards are equipped with six DSP chips, three of which are the type required by Bruno/Reso. For this reason, a maximum of three Bruno/Reso plug-ins can be used per MIX Farm card.

DSP Farm Cards A maximum of eight voices of polyphony are available per chip on DSP Farm cards. All four DSP chips on a DSP Farm card can be used by Bruno/Reso (if available).

Inserting Bruno/Reso onto an Audio Track

To use Bruno/Reso in a Pro Tools session, you must add it to a track as an insert. Once Bruno/Reso is inserted on the track, you can adjust its controls to get the effect that you want, then play the plug-in using the on-screen keyboard, an external MIDI controller, or a MIDI track.

To add Bruno/Reso as a track Insert:

1 Click the Insert Selector on the desired rack and select Bruno or Reso.

2 Click Play on the Pro Tools Transport to start audio playback.

3 Play Bruno/Reso with the on-screen keyboard or by MIDI control. See "Playing Bruno/Reso" on page 69.

4 Adjust Bruno/Reso controls to get the effect you want.

Playing Bruno/Reso

To generate sound, Bruno/Reso must be played during audio playback. You can play Bruno/Reso in two ways:

◆ In real time, using either the on-screen keyboard or an external MIDI controller.

◆ Using a MIDI track.

Using the On-Screen Keyboard

The simplest way to play Bruno/Reso is to use its on-screen keyboard. You can click one note at a time or use keyboard latch to hold multiple notes.

Notes played with the on-screen keyboard are triggered at a MIDI velocity of 92.

To play Bruno/Reso with the on-screen keyboard:

1 Open the plug-in window for Bruno/Reso.

2 Click Play on the Pro Tools Transport to start audio playback.

3 Click the on-screen keyboard. Bruno/Reso will only produce sound while audio plays on the source track.

To latch keys on the on-screen keyboard:

1 Click the Latch bar, then click multiple keys. Chords can be played in this way.

2 To turn off a latched key, click it a second time.

3 To turn off key latching entirely, click the Latch bar a second time.

Saving a Bruno or Reso setting while keys are latched also saves the latched keys.

Using MIDI

You can play Bruno/Reso live using a MIDI keyboard controller. You can also use the MIDI keyboard controller to record your performance on a MIDI track and route it to Bruno/Reso for playback.

⚠ *On Mac OS 9, Pro Tools requires OMS for MIDI communication with Bruno/Reso. If you are running Mac OS 9 and OMS is not currently installed and configured on your system, refer to your Pro Tools Getting Started Guide for instructions.*

To configure Bruno/Reso for MIDI input:

1 Insert Bruno/Reso on an audio track.

2 Choose File > New Track and specify 1 new MIDI track, then click Create. Create a separate MIDI track for each Bruno/Reso plug-in you use.

3 Click the MIDI track's MIDI Device/Channel Selector and choose Bruno or Reso.

If you are using multiple Bruno/Reso plug-ins, they will all appear in this pop-up. Route the MIDI track to the correct one.

4 Record-enable the MIDI track.

5 Test your MIDI connection by playing notes on your MIDI keyboard. The corresponding notes should highlight on Bruno/Reso's on-screen keyboard.

To play Bruno/Reso with a MIDI controller:

1 Start audio playback.

2 Play your MIDI keyboard while audio plays.

Bruno/Reso only produces sound during audio playback on the source track.

Using MIDI Playback

You can also play Bruno/Reso using a Pro Tools MIDI track. Use a separate MIDI track for each Bruno/Reso plug-in.

To play Bruno/Reso using a MIDI track:

1 Insert Bruno or Reso on an audio track.

2 Click the MIDI track's MIDI Device/Channel Selector and choose Bruno or Reso. If you are using multiple Bruno or Reso plug-ins, they will all appear in this pop-up. Route the MIDI track to the correct one.

3 Start Pro Tools playback.

Using a Key Input for Side-chain Processing

Bruno and Reso feature side-chain processing capabilities. Side-chain processing lets you trigger certain controls from a separate reference track or external audio source. The source used for triggering is referred to as the *key input*.

You can use this capability to control the rate at which Bruno performs sample switching or Reso toggles its harmonics back and forth using the dynamics of another signal (the key input).

Typically, a rhythm track such as a drum kit is used to trigger these controls and create rhythmic timbral changes that match the groove of the key input.

To use a key input for side-chain processing:

1 Click the Side-Chain Input Selector and choose the input or bus with the audio you want to use to trigger the plug-in.
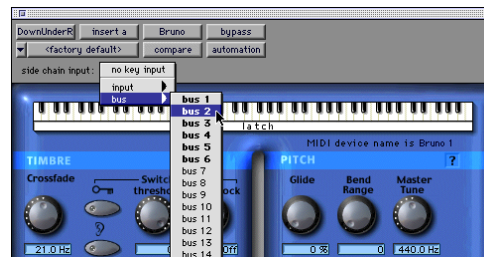
Selecting a Key Input

2 Click the Key Input button (the button with the key icon above it) to activate side-chain processing.

3 To hear the audio source you have selected to control side-chain input, click the Key Listen button (the button below the Ear icon).

4 Begin playback. The plug-in uses the input or bus that you chose as a side-chain input to trigger the effect.

5 Adjust other controls to create the desired effect.

Bruno Controls

Bruno uses time-slicing for tone generation, extracting timbres from the audio track during playback and cross-fading them together at a user-selectable rate.

Bruno

This crossfading can create a rhythmic pulse in the sound as the timbre changes. This makes Bruno ideal for creating tonal effects with a continuously shifting timbre—similar to the wave sequencing found on synthesizers such as the PPG, Prophet VS, Korg Wavestation, and Waldorf XT.

By carefully choosing the type of source audio, the crossfade length, and the type of switching, you can create unique and complex sound textures.

On-Screen Keyboard

The on-screen keyboard

The simplest way to play Bruno is to use its on-screen keyboard. You can click one note at a time or use keyboard latch to hold multiple notes.

Notes played with the on-screen keyboard are triggered at a MIDI velocity of 92.

Timbre Controls

Timbre controls

Crossfade

Crossfade sets the rate at which Bruno extracts timbres from the source audio and crossfades from one time slice to the next. The range of this control is from 2 to 40 Hz (cycles per second) in a 44.1 kHz or 48 kHz session, and from 4 to 40 Hz in a 96 kHz session.

The higher the crossfade frequency, the smaller the time slice, and the faster Bruno moves between slices. A higher frequency crossfade would retain more characteristics of the original audio source and would have a pulsed or wave-sequenced feel.

The lower the crossfade frequency, the larger the time slice, and the slower Bruno moves between slices. A lower frequency crossfade would have fewer characteristics of the original source and a more rounded or gradually evolving sound.

Switch

Switch causes Bruno to switch directly between time-sliced samples without crossfading them. This adds a distinct rhythmic pulse to the timbral changes.

Switching can be controlled by *triggering* (using the dynamics of the source audio or an external key input) or by MIDI clock.

External Key Enables switching from a separate reference track or external audio source. The source used for triggering is referred to as the *key input* and is selected using the Side-chain Input pop-up. You can assign either an audio input channel or a TDM bus channel.

Typically, a drum track is used as a key input so that switching occurs according to a definite rhythmic pattern.

Key Listen When enabled, Key Listen monitors the source of the key input. It is often useful to do this in order to fine tune Bruno's settings to the key input. See "Using a Key Input for Side-chain Processing" on page 70.

Threshold Sets the level in decibels above which switching occurs. When the audio input level rises above the Threshold level, Bruno will switch directly to a new time-slice. The range of this control is from a low of –48 dB (maximum switching) to a high of 0.0 dB (no switching). If no key input is used, the dynamics of the source audio will trigger switching. If a key input is used, the dynamics of the key input signal will trigger switching. Threshold-based switching can be used at the same time as Key Input-based switching.

MIDI Clock Triggers switching in sync with a MIDI Beat Clock signal. This creates a very regular, highly rhythmic wave sequencing effect that is ideal for sessions arranged around MIDI beat clock. This control can be set to quarter, eighth, or sixteenth notes, or dotted triplet values of the same.

☼ *For quick numeric entry of MIDI beat clock values, type "4," "8," or "16" for quarter notes, eight notes, or sixteenth notes. Add "t" for triplets, or "d" for dotted note values. Typing "4t" for example, enters a quarter note triplet value. Typing "16d" enters a dotted sixteenth note value.*

Timbrometer
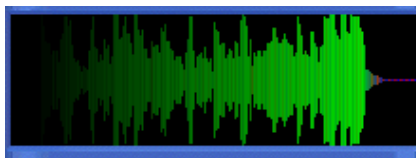
Timbrometer

This multicolor waveform display shows the amplitude and duration of the audio signal generated by Bruno as well as the frequency of timbral changes and whether they are crossfaded or switched.

Red and blue waveform segments indicate timbral changes that are crossfaded. Green waveform segments indicate timbral changes that are hard switched.

Amplitude Controls

Amplitude controls

Gain Amount

Gain Amount attenuates output level gain. Since some of Bruno's controls can cause extreme changes in signal level, this is particularly useful for preventing clipping and achieving unity gain with the original signal level. This control is adjustable from a low of –96 dB (no gain) to a high of 0.0 dB (maximum gain).

Gain Velocity

Gain Velocity sets the velocity sensitivity of the Gain Amount control. This gives you touch-sensitive control over Bruno's volume using a MIDI keyboard.

This control is adjustable from a low of –24 dB (maximum velocity sensitivity) to a high of 0.0 dB (no velocity sensitivity).

If you set Gain Velocity to –24 dB, a soft strike on a key will reduce gain up to –24 dB. A hard strike will have a maximum output level equal to the current dB setting of the Gain Amount control.

Conversely, if Gain Velocity is set to 0.0 dB, Bruno's volume will not change no matter how hard or soft you strike a key on your MIDI controller.

⚠️ *Gain Velocity only has an effect when you play Bruno with a velocity-sensitive MIDI controller.*

Mix

Mix adjusts the mix of the processed audio with the original, unprocessed audio.

Spread

When Bruno is used in stereo, the Spread control can be used to pan multiple voices within the stereo field. This control is adjustable from 0% (no stereo spread) to 100% (maximum stereo spread).

Voice stacking has a direct effect on stereo Spread. For example, setting Voice Stack to 1 and Spread to 100% will randomly pan each note played. Setting Voice Stack to 4 and Spread to 100%, will pan two of the voices hard left, and two voices hard right.

ADSR Envelope Generator

The ADSR (attack, decay, sustain, release) Envelope Generator controls Bruno's amplitude envelope. This amplitude envelope is applied to a sound each time a note is struck.

The four envelope elements can be adjusted by dragging the appropriate breakpoint, or by typing in a numeric value.

Attack Controls the amount of time in milliseconds that the sound takes to rise from zero amplitude to its full level. The longer the attack, the more time it takes for the sound to reach maximum volume after the a note is struck. This control is adjustable from 0.0 to 5000 milliseconds.

Decay Controls the amount of time in milliseconds that the sound takes to fall from its peak Attack level to the Sustain level. This control is adjustable from 0.0 ms to 5000 ms.

Sustain Level Controls the amplitude level in dB that is reached after the decay time has elapsed. The amplitude level stays constant as long as a MIDI note remains depressed. This control is adjustable from –96 dB (no sustain) to 0.0 dB (maximum sustain).

Release Controls the amount of time in milliseconds that the sound takes to fall from the Sustain level to zero amplitude after a note is released. This control is adjustable from 0.0 ms to 5000 ms.

Pitch Controls

Pitch controls

Glide

Glide, also known as *portamento*, determines the amount of time it takes for a pitch to glide from the current note to the next note played. This effect is commonly found on synthesizers.

Glide is adjustable from a low of 0.0% (no glide) to a high of 100% (maximum glide). A setting of 100% will take the longest time to travel from the current note to the next note played. The effect is also dependent on the interval (distance of pitch) between the two notes: The larger the interval, the more noticeable the effect.

Bend Range

Bend Range sets the maximum interval of pitch bend that can be applied to Bruno with a MIDI controller's pitch bend wheel. This control is adjustable from 0 semitones (no bend) to 12 semitones (1 octave).

Master Tune

Master Tune can be used to tune the pitch of Bruno's output to another instrument. By default, this control is set to 440.0 Hz It can be adjusted from a low of 430.0 Hz to a high of 450.0 Hz.

Detune Amount

Detuning is a common sound-thickening technique used on synthesizers and many effects devices. Bruno's Detune Amount control sets the maximum amount of pitch detuning that occurs when multiple voices are stacked together using Voice Stacking. Using a combination of voice stacking and detuning, you can create timbres that are exceptionally fat.

Voices can be detuned up to 50.0 cents. (One cent is equal to 1/100th of a semitone.)

Detune Velocity

Detune Velocity controls how MIDI key velocity affects voice detuning. This gives you velocity-sensitive control over voice detuning when you play Bruno with a MIDI keyboard.

This control is adjustable from a low of 0.0 cents (no velocity-sensitive detuning) to a high of 50.0 cents (maximum velocity-sensitive detuning).

If Detune Velocity is set to 0.0 cents, detuning will not change no matter how hard you strike a key on your MIDI controller. Conversely, if you set Detune Velocity to 50.0 cents, a hard strike will detune voices a maximum of 50.0 cents (in addition to the detuning specified with the Detune Amount control).

⚠ *Detune Velocity has an effect only when you play Bruno with a velocity-sensitive MIDI controller.*

Voice Controls

Voice controls

These controls set Bruno's voice polyphony and allocation.

Mode

Mono (Monophonic)

In this mode, Bruno responds monophonically, producing a single note even if more than one is played simultaneously (though multiple voices can be stacked on the same note using the Voice Stacking control). Monophonic mode gives voice priority to the most recently played note.

Poly (Polyphonic)

In this mode, Bruno responds polyphonically, producing as many notes as are played simultaneously (up to 62 on Pro Tools|HD Accel systems). The number of notes that can be played simultaneously depends on the Voice Stacking setting chosen. A voice stack setting of 1, for example, allows up to 62 individual notes simultaneously. A voice stack setting of All allows only one note at a time, but will stack all 62 voices on that note, producing an extremely fat sound.

Voice Stack

Voice Stack selects the number of voices that are used, or *stacked* when you play a single note. The number of voices that you choose to stack will directly affect polyphony. Selecting a larger number of stacked voices will reduce the number of notes that you can play simultaneously.

Voice Stack

The sample rate of your session also affects polyphony. For example, in a 96 kHz session, Bruno can simultaneously play up to:

◆ 32 notes in a 1-voice stack

◆ 16 notes in a 2-voice stack

◆ 4 notes in a 4-voice stack

◆ 2 notes in an 8-voice stack

◆ 1 note in an 12-voice (All) stack

💡 *The 62-voice Bruno requires an HD Accel card.*

In a 44.1 kHz or 48 kHz session on a Pro Tools|HD system not equipped with an HD Accel card, Bruno can simultaneously play up to:

- 24 notes in a 1-voice stack
- 12 notes in a 2-voice stack
- 6 notes in a 4-voice stack
- 3 notes in an 8-voice stack
- 1 note in a 24-voice (All) stack

Voice counts for Bruno for 44.1 kHz and 48 kHz sessions are the same on Pro Tools|HD-series systems not equipped with an HD Accel card, and Pro Tools|24 MIX-series systems.

If all available voices are being used, playing an additional note will replace the first note played in the chord.

⚠ *A maximum of eight voices of polyphony are available per Bruno/Reso plug-in on DSP Farm cards.*

Online Help

Online help

To use online help, click the name of any control or parameter and an explanation will appear. Clicking the Online Help button itself provides more details on using this feature.

Reso Controls

Reso synthesizes new harmonic overtones from the source audio signal, creating harmonically rich timbres with a metallic, synthesizer-like character.

Reso

On-Screen Keyboard

On-screen keyboard

The simplest way to play Reso is to use its on-screen keyboard. You can click one note at a time or use keyboard latch to hold multiple notes.

Notes played with the on-screen keyboard are triggered at a MIDI velocity of 92.

Timbre Controls

Timbre controls

Resonance Amount

Resonance Amount controls the intensity of harmonic overtones produced by the Resonator. Increasing the Resonance Amount will increase the overall harmonic content of the sound while increasing the sustained portions of the generated harmonics.

The frequency content of the input signal largely determines what harmonics are generated by the resonator. For this reason, the character of the resonance will change according to the type of audio that you process.

Resonance Velocity

Resonance Velocity increases or decreases resonance according to how hard a MIDI key is struck and how much resonance is initially specified with the Resonance Amount control.

Resonance Velocity is adjustable from a low of –10 to a high of +10. With positive values, the harder the key is struck, the more resonance is applied. With negative values, the harder the key is struck, the less resonance is applied.

The effectiveness of this control depends on the Resonance Amount setting. For example, if Resonance Amount is set to 0, setting the Resonance Velocity to a negative value will have no effect, since there is no resonance to remove. Similarly, if the Resonance Amount control is set to 10, setting Resonance Velocity to +10 will have no effect since the resonance is already at its maximum.

For optimum effect, set the Resonance Amount to a middle value, then set Resonance Velocity accordingly for the desired effect.

⚠️ *Resonance Velocity has an effect only when you play Reso with a velocity-sensitive MIDI controller.*

Damping Amount

Damping causes the high-frequency harmonics of a sound to decay more rapidly than the low frequency harmonics. It lets you control the brightness of the signal generated by Reso's Resonator and is particularly useful for creating harp or plucked string-like textures.

The range of this control is from 0 (no damping) to 10 (maximum damping). The greater the amount of damping, the faster the high-frequency harmonics in the audio will decay and the duller it will sound.

Damping Velocity

Damping Velocity increases or decreases damping according to how hard a MIDI key is struck and how much damping is initially specified with the Damping Amount control.

Damping Velocity is adjustable from a low of –10 to a high of +10. With positive values, the harder the key is struck, the more damping is applied. With negative values, the harder the key is struck, the less damping is applied (which simulates the behavior of many real instruments).

The effectiveness of this control depends on the Damping Amount setting. For example, if Damping Amount is set to zero, setting the Damping Velocity to a negative value will have no effect, since there is no damping to remove. Similarly, if the Damping Amount control is set to 10, setting Damping Velocity to +10 will have no effect since damping is already at its maximum.

For optimum effect, set the Damping Amount to a middle value, then set Damping Velocity accordingly for the desired effect.

⚠ *Damping Velocity only has an effect when you play Reso with a velocity-sensitive MIDI keyboard controller.*

Harmonics

The resonator adds harmonic overtones to the source audio signal that are integer multiples of the fundamental frequency of the signal. The Harmonics control selects between all of these harmonics, or just the odd-numbered intervals. Your choice will affect the timbre of the sound.

All Adds all of the harmonic overtones generated by the resonator. In synthesizer parlance, this produces a somewhat buzzier, sawtooth wave-like timbre.

Odd Adds only the odd-numbered harmonic overtones generated by the resonator. In synthesizer parlance, this produces a somewhat more hollow, square wave-like timbre.

Toggle

Reso can automatically toggle between the All and Odd harmonics settings, producing a rhythmic pulse in the timbre.

Harmonic toggling can be controlled either by triggering (using the dynamics of the source audio itself, or those of an external key input) or by MIDI Beat Clock.

External Key Toggles the harmonics from a separate reference track or an external audio source. The source used for toggling is referred to as the *key input* and is selected using the Side-chain Input pop-up. You can assign either an audio input channel or a TDM bus channel.

Typically, a drum track is used as a key input so that toggling occurs according to a definite rhythmic pattern.

Key Listen When enabled, monitors the source of the key input. It is useful to do this to fine tune Reso's settings to the key input.

📄 *See "Using a Key Input for Side-chain Processing" on page 70.*

Threshold Sets the level in decibels above which toggling occurs. When the audio input level rises above the Threshold level, Reso will toggle its harmonics setting. The range of this control is from a low of –48 dB (maximum toggling) to a high of 0.0 dB (no toggling). If no key input is used, the dynamics of the source audio will trigger toggling. If a key input is used, the dynamics of the key input signal will trigger toggling. Threshold-based switching can be used at the same time as Key Input-based switching.

MIDI Clock Triggers toggling in sync with a MIDI Beat Clock signal. This creates a very regular, highly rhythmic wave sequencing effect that is ideal for sessions arranged around MIDI beat clock. This control can be set to quarter, eighth, or sixteenth notes, or dotted triplet values of the same.

💡 *For quick numeric entry of MIDI beat clock values, type "4," "8," or "16" for quarter notes, eight notes, or sixteenth notes. Add "t" for triplets, or "d" for dotted note values. Typing "4t" for example, enters a quarter note triplet value. Typing "16d" enters a dotted sixteenth note value.*

Amplitude Controls

Amplitude controls

Gain Amount

Gain Amount attenuates output level gain. Since resonation can cause extreme changes in signal level, this is particularly useful for preventing clipping and achieving unity gain with the original signal level. This control is adjustable from a low of –96 dB (no gain) to a high of 0.0 dB (maximum gain).

Gain Velocity

Gain Velocity sets the velocity sensitivity of the Gain Amount control. This gives you touch-sensitive control over Reso's volume using a MIDI keyboard.

This control is adjustable from a low of –24 dB (maximum velocity sensitivity) to a high of 0.0 dB (no velocity sensitivity).

If you set Gain Velocity to –24 dB, a soft strike on a key will reduce gain up to –24 dB. A hard strike will have a maximum output level equal to the current dB setting of the Gain Amount control.

Conversely, if Gain Velocity is set to 0.0 dB, Reso's volume will not change no matter how hard or soft you strike a key on your MIDI controller).

⚠ *Gain Velocity only has an effect when you play Reso with a velocity-sensitive MIDI keyboard controller.*

Mix

Mix adjusts the mix of the processed audio with the original, unprocessed audio.

Spread

When Reso is used in stereo, the Spread control can be used to pan multiple Reso voices within the stereo field. This control is adjustable from 0% (no stereo spread) to 100% (maximum stereo spread).

Voice stacking affects stereo Spread. For example, setting Voice Stack to 1 and Spread to 100% will alternately pan each note played right and left. Setting Voice Stack to 4 and Spread to 100%, will pan two of the five voices hard left, and two voices hard right.

ADSR Envelope Generator

The ADSR (attack, decay, sustain, release) Envelope Generator controls Bruno's amplitude envelope. This amplitude envelope is applied to a sound each time a note is struck.

The four envelope elements can be adjusted by dragging the appropriate breakpoint, or by typing in a numeric value.

Attack Controls the amount of time in milliseconds that the sound takes to rise from zero amplitude to its full level. The longer the attack, the more time it takes for the sound to reach maximum volume after the a note is struck. This control is adjustable from 0.0 to 5000 milliseconds.

Decay Controls the amount of time in milliseconds that the sound takes to fall from its peak Attack level to the Sustain level. This control is adjustable from 0.0 ms to 5000 ms.

Sustain Level Controls the amplitude level in dB that is reached after the decay time has elapsed. The amplitude level stays constant as long as a MIDI note remains depressed. This control is adjustable from –96 dB (no sustain) to 0.0 dB (maximum sustain).

Release Controls the amount of time in milliseconds that the sound takes to fall from the Sustain level to zero amplitude after a note is released. This control is adjustable from 0.0 ms to 5000 ms.

Pitch Controls

Pitch controls

Glide

Glide, also known as *portamento*, determines the amount of time it takes for a pitch to glide from the current note to the next note played. This effect is commonly used on synthesizers.

Glide is adjustable from a low of 0.0% (no glide) to a high of 100% (maximum glide). A setting of 100% will take the longest time to travel from the current note to the next note played. The effect is also dependant on the interval (distance of pitch) between the two notes: The larger the interval, the more noticeable the effect.

Bend Range

Bend Range sets the maximum interval of pitch bend that can be applied to Reso with a MIDI controller's pitch bend wheel. This control is adjustable from 0 semitones (no bend) to 12 semitones (1 octave).

Master Tune

Master Tune can be used to tune the pitch of Reso's output to another instrument. By default, this control is set to 440.0 Hz It can be adjusted from a low of 430.0 Hz to a high of 450.0 Hz.

Detune Amount

Detuning is a common sound-thickening technique used on synthesizers and many effects devices. Reso's Detune Amount control lets you set the maximum amount of pitch detuning that occurs when multiple voices are stacked together using Voice Stacking. Using a combination of voice stacking and detuning, you can create timbres that are exceptionally fat.

Voices can be detuned up to 50.0 cents. (One cent is equal to 1/100th of a semitone.)

Detune Velocity

Detune Velocity controls how MIDI key velocity affects voice detuning. This gives you touch-sensitive control over voice detuning when you play Reso with a MIDI keyboard.

This control is adjustable from a low of 0.0 cents (no velocity-sensitive detuning) to a high of 50.0 cents (maximum velocity-sensitive detuning).

If Detune Velocity is set to 0.0 cents, detuning will not change no matter how hard or soft you strike a key on your MIDI controller. Conversely, if you set Detune Velocity to 50.0 cents, a hard strike will detune voices a maximum of 50.0 cents.

⚠ *Detune Velocity only has an effect when you play Reso with a velocity-sensitive MIDI keyboard controller.*

LPF/Voice Controls

LPF and Voice controls

LPF (Low-Pass Filter)

Reso's Low-Pass Filter is a single resonant filter that is applied to all of Reso's voices.

Frequency

The Frequency control sets the cutoff frequency of the Low-Pass Filter in Hertz. All frequencies above the selected cutoff frequency will be attenuated.

The range of this control is from 20 Hz to 20 kHz.

Q

Sometimes referred to as *resonance* on synthesizers, Q adjusts the height of the resonant peak that occurs at the filter's cutoff frequency.

Increasing the Q increases the volume of frequencies near the filter's cutoff frequency (suppressing the more remote frequencies) and adds a nasal quality to the audio. High Q settings let you create wah-wah type effects, particularly when the filter is swept with the Follower.

The range of this control is from 0 to 10.

Follower

The Follower is an envelope follower that lets the filter cutoff frequency dynamically follow the amplitude of the source audio signal.

The range of this control is from a low of –10 to a high of +10. With positive values, the louder the source audio, the higher the cutoff frequency and the wider the filter will open for a brighter sound. With negative values, the louder the source audio, the lower the cutoff frequency and the more the filter will close for a duller sound.

The effectiveness of the Follower depends on the filter's Frequency setting. For example, setting the Follower to +10 and selecting a low Frequency setting will sweep the filter wide on loud passages. However, if the cutoff frequency is at its maximum, setting the Follower to +10 will not sweep the filter at all since it is already completely open.

When used with high Q settings and a relatively low cutoff frequency, the Follower can be used to produce an automatic wah-wah-type effect.

Mono (Monophonic)

In this mode, Reso responds monophonically, producing a single note even if more than one is played simultaneously (though multiple voices can be stacked on the same note using the Voice Stacking control). Monophonic mode gives voice priority to the most recently played note.

Poly (Polyphonic)

In this mode, Reso responds polyphonically, producing as many notes as are played simultaneously (up to 62 on Pro Tools|HD Accel systems). The number of notes that can be played simultaneously depends on the Voice Stacking setting chosen. A voice stack setting of 1, for ex-

ample, allows up to 62 individual notes simultaneously. A voice stack setting of All allows only one note at a time, but will stack all 62 voices on that note, producing an extremely fat sound.

⚠ *Polyphony will be reduced by half at 96 kHz.*

Voice Stack

Voice Stack selects the number of voices that are used, or *stacked* when you play a single note. The number of voices that you choose to stack will directly affect polyphony. Selecting a larger number of stacked voices will reduce the number of notes that you can play simultaneously. The sample rate of your session will also affect polyphony. Pro Tools|HD-series cards provide better polyphony than Pro Tools|24 MIX-series cards.

Voice Stack

In a 96 kHz session, Reso on Pro Tools|HD Accel systems can simultaneously play up to:

- ◆ 32 notes in a 1-voice stack
- ◆ 16 notes in a 2-voice stack
- ◆ 4 notes in a 4-voice stack
- ◆ 2 notes in an 8-voice stack
- ◆ 1 note in an 14-voice (All) stack

In a 44.1 kHz or 48 kHz session on Pro Tools|HD systems not equipped with an HD Accel card, the standard Reso module can simultaneously play up to:

◆ 28 notes in a 1-voice stack

◆ 14 notes in a 2-voice stack

◆ 7 notes in a 4-voice stack

◆ 3 notes in an 8-voice stack

◆ 1 note in a 28-voice (All) stack

In a 44.1 kHz or 48 kHz session on MIX or DSP Farm cards, you can simultaneously play up to:

◆ 24 notes in a 1-voice stack

◆ 12 notes in a 2-voice stack

◆ 6 notes in a 4-voice stack

◆ 3 notes in an 8-voice stack

◆ 1 note in a 24-voice (All) stack

If all available voices are being used, playing an additional note will replace the first note played in the chord.

⚠ *A maximum of eight voices of polyphony are available per Bruno/Reso plug-in on DSP Farm cards.*

Online Help

Online help

To use online help, click the name of any control or parameter and an explanation will appear. Clicking the Online Help button itself provides more details on using this feature.

chapter 8

Reverb One

Reverb One is a world-class reverb processing TDM plug-in. It provides a level of sonic quality and reverb-shaping control previously found only on the most advanced hardware reverberation units.

A set of unique, easy-to-use audio shaping tools lets you customize reverb character and ambience to create natural-sounding halls, vintage plates, or virtually any type of reverberant space you can imagine.

Reverb One features include:

- Editable Reverb EQ graph
- Editable Reverb Color graph
- Reverb Contour graph
- Dynamic control of reverb decay
- Chorusing
- Early reflection presets
- Extensive library of reverb presets
- Supports 44.1 khz, 48 kHz, 88.2 kHz, and 96 kHz processing.

⚠ *For sessions with a sample rate greater than 96 kHz, Reverb One will downsample and upsample accordingly.*

System Requirements

To use Reverb One you need a Digidesign-approved Pro Tools|HD-series or Pro Tools|24 MIX-series system with version 5.3 software or higher.

For the latest compatibility information, contact your local Digidesign dealer or visit Digidesign's Web site.

DSP Requirements

Each Reverb One plug-in used in a session requires a portion of the total DSP resources available. The amount of DSP available depends on the number and type of DSP cards (HD Accel, HD Process, Mix Farm, or DSP Farm) in your Pro Tools system.

Reverb One DSP requirements:

• One entire Pro Tools|HD DSP chip per mono or stereo instance

Reverb One requires an SRAM DSP chip to function. There are nine of these particular DSP chips on HD Core, HD Accel, and HD Process cards, six on a Mix Farm card, and four on a DSP Farm card. HD Accel cards may be required to use Reverb One at sample rates above 96 kHz, depending on other session factors such as number of tracks, number of plug-ins, and total number of Pro Tools|HD cards in your system.

A Reverb Overview

Digital reverberation processing can simulate the complex natural reflections and echoes that occur after a sound has been produced, imparting a sense of space and depth—the signature of an acoustic environment. When you use a reverberation plug-in such as Reverb One, you are artificially creating a sound space with a specific acoustic character.

This character can be melded with audio material, with the end result being an adjustable mix of the original dry source and the reverberant wet signal. Reverberation can take relatively lifeless mono source material and create a stereo acoustic environment that gives the source a perceived weight and depth in a mix.

Creating Unique Sounds

In addition, digital signal processing can be used creatively to produce reverberation characteristics that do not exist in nature. There are no rules that need to be followed to produce interesting treatments. Experimentation can often produce striking new sounds.

Acoustic Environments

When you hear live sound in an acoustic environment, you generally hear much more than just the direct sound from the source. In fact, sound in an anechoic chamber, devoid of an acoustic space's character, can sound harsh and unnatural.

Each real-world acoustical environment, from a closet to a cathedral, has its own unique acoustical character or sonic signature. When the reflections and reverberation produced by a space combine with the source sound, we say that the space is *excited* by the source. Depending on the acoustic environment, this could produce the warm sonic characteristics we associate with reverberation, or it could produce echoes or other unusual sonic characteristics.

Reverb Character

The character of a reverberation depends on a number of things. These include proximity to the sound source, the shape of the space, the absorptivity of the construction material, and the position of the listener.

Reflected Sound

In a typical concert hall, sound reaches the listener shortly after it is produced. The original direct sound is followed by reflections from the ceiling or walls. Reflections that arrive within 50 to 80 milliseconds of the direct sound are called *early reflections*. Subsequent reflections are called *late reverberation*. Early reflections provide a sense of depth and strengthen the perception of loudness and clarity. The delay time between the arrival of the direct sound and the beginning of early reflections is called the *pre-delay*.

The loudness of later reflections combined with a large pre-delay can contribute to the perception of largeness of an acoustical space. Early re-

flections are followed by reverberation and re-
petitive reflections and attenuation of the
original sound reflected from walls, ceilings,
floors, and other objects. This sound provides a
sense of depth or size.

Reverb One provides control over these rever-
beration elements so that extremely natural-
sounding reverb effects can be created and ap-
plied in the Pro Tools mix environment.

Reverb One Controls

Reverb One has a variety of controls for produc-
ing a wide range of reverb effects. Controls can
be adjusted by dragging their sliders or typing
values directly in their text boxes.
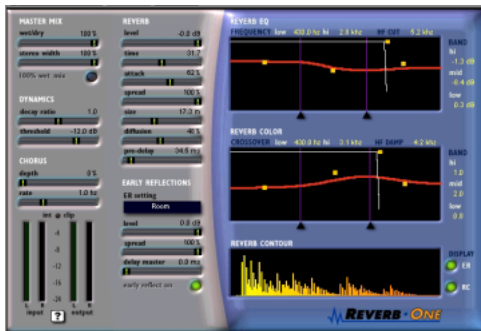
Reverb One

Editing Graph Values

In addition to the standard slider controls, the
Reverb EQ and Reverb Color graph settings can
be adjusted by dragging elements of the graph
display.

To cut or boost a particular band:

■ Drag a yellow breakpoint up or down.

To adjust frequency or crossover:

■ Drag a triangular slider right or left.

To adjust high-frequency cut or damp:

■ Drag the yellow dot right or left.
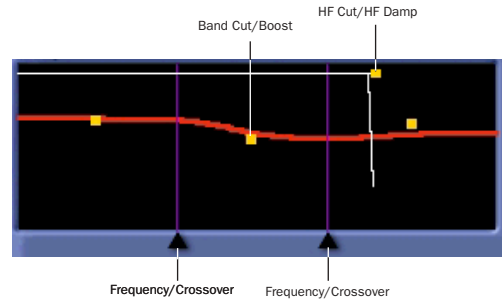
Adjusting graph controls

Master Mix Controls

The Master Mix section has controls for adjust-
ing the relative levels of the source signal and
the reverb effect, and also the width of the re-
verb effect in the stereo field.

Master Mix section

Wet/Dry Adjusts the mix between the dry, un-
processed signal and the reverb effect.

Stereo Width Controls the width of the reverb in
the stereo field. A setting of 0% produces a
mono reverb. A setting of 100% produces maxi-
mum spread in the stereo field.

100% Wet Toggles the Wet/Dry control between
100% wet and the current setting.

Dynamics Controls

The Dynamics section has controls for adjusting Reverb One's response to changes in input signal level.

Dynamics can be used to modify a reverb's decay character, making it sound more natural, or conversely, more unnatural, depending on the desired effect.

Typically, dynamics are used to give a reverb a shorter decay time when the input signal is above the threshold, and a longer decay time when the input level drops below the threshold.

This produces a longer, more lush reverb tail and greater ambience between pauses in the source audio, and a shorter, clearer reverb tail in sections without pauses.

For example, on a vocal track, use Dynamics to make the reverb effect tight, clear, and intelligible during busy sections of the vocal (where the signal is above the Threshold setting), and then "bloom" or lengthen at the end of a phrase (where the signal falls below the threshold).

Similarly, Dynamics can be used on drum tracks to mimic classic gated reverb effects by causing the decay time to cut off quickly when the input level is below the threshold.

☼ *To hear examples of decay dynamics, load one of the Dynamics presets with the Librarian.*
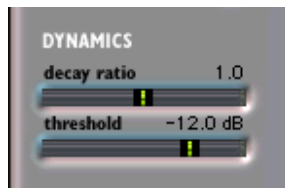
Dynamics section

Decay Ratio Controls the ratio by which reverb time is increased when a signal is above or below the Threshold level. Dynamics behavior differs when the Decay Ratio is set above or below 1. A ratio setting of greater than 1 increases reverb time when the signal is *above* the threshold. A ratio setting of less than 1 increases a reverb's time when the signal is *below* the threshold.

For example, if Decay Ratio is set to 4, the reverb time is increased by a factor of 4 when the signal is above the threshold level. If the ratio is 0.25, reverb time is increased by a factor of 4 when the signal is below the Threshold level.

Threshold Sets the input level above or below which reverb decay time will be modified.

Chorus Controls

The Chorus section has controls for setting the depth and rate of chorusing applied to a reverb tail. Chorusing thickens and animates sounds by adding a delayed, pitch-modulated copy of an audio signal to itself.

Chorusing produces a more ethereal or spacey reverb character. It is often used for creative effect rather than to simulate a realistic acoustic environment.

☼ *To hear examples of reverb tail chorusing, load one of the Chorus presets with the Librarian.*
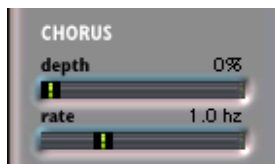
Chorus section

Depth Controls the amplitude of the sine wave generated by the LFO (low frequency oscillator) and the intensity of the chorusing. The higher the setting, the more intense the modulation.

Rate Controls pitch modulation frequency. The higher the setting, the more rapid the chorusing. Setting the Rate above 20 Hz can cause frequency modulation to occur. This will add sideband harmonics and change the reverb's tone color, producing some very interesting special effects.

Reverb Controls

The Reverb section has controls for the various reverb tail elements, including level, time, attack, spread, size, diffusion, and pre-delay. These determine the overall character of the reverb.
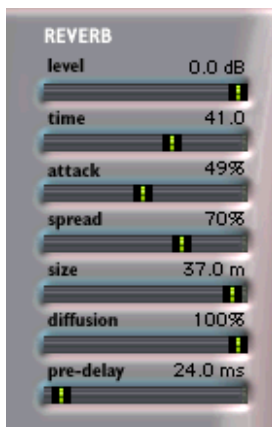
Reverb section

Level Controls the output level of the reverb tail. When set to 0%, the reverb effect consists entirely of the early reflections (if enabled).

Time Controls the rate at which the reverberation decays after the original direct signal stops. The value of the Time setting is affected by the Size setting. You should adjust the reverb Size

setting before adjusting the Time setting. If you set Time to its maximum value, infinite reverberation is produced. The HF Damping and Reverb Color controls also affect reverb Time.

Attack Attack determines the contour of the reverberation envelope. At low Attack settings, reverberation builds explosively, and decays quickly. As Attack value is increased, reverberation builds up more slowly and sustains for the length of time determined by the Spread setting.

When Attack is set to 50%, the reverberation envelope emulates a large concert hall (provided the Spread and Size controls are set high enough).

Spread Controls the rate at which reverberation builds up. Spread works in conjunctions with the Attack control to determine the initial contour and overall ambience of the reverberation envelope.

Low Spread settings result in a rapid onset of reverberation at the beginning of the envelope. Higher settings lengthen both the attack and buildup stages of the initial reverb contour.

Size Determines the rate of diffusion buildup and acts as a master control for Time and Spread within the reverberant space.

Size values are given in meters and can be used to approximate the size of the acoustic space you want to simulate. When considering size, keep in mind that the size of a reverberant space in meters is roughly equal to its longest dimension.

Diffusion Controls the degree to which initial echo density increases over time. High Diffusion settings result in high initial buildup of echo density. Low Diffusion settings cause low initial buildup.

After the initial echo buildup, Diffusion continues to change by interacting with the Size control and affecting the overall reverb density. Use high Diffusion settings to enhance percussion. Use low or moderate settings for clearer, more natural-sounding vocals and mixes.

Pre-Delay Determines the amount of time that elapses between the original audio event and the onset of reverberation. Under natural conditions, the amount of Pre-delay depends on the size and construction of the acoustic space, and the relative position of the sound source and the listener. Pre-delay attempts to duplicate this phenomenon and is used to create a sense of distance and volume within an acoustic space. Long Pre-Delay settings place the reverberant field behind rather than on top of the original audio signal.

For an interesting musical effect, set the Pre-Delay time to a beat interval such as 1/8, 1/16, or 1/32 notes.

Early Reflection Controls

The Early Reflections section has controls for the various early reflection elements, including ER setting, level, spread, and delay.

Calculating Early Reflections

A particular reflection within a reverberant field is usually categorized as an early reflection. Early reflections are usually calculated by measuring the reflection paths from source to listener. Early reflections typically reach the listener within 80 milliseconds of the initial audio event, depending on the proximity of reflecting surfaces.

Simulating Early Reflections

Different physical environments have different early reflection signatures that our ears and brain use to pinpoint location information. These reflections influence our perception of the size of a space and where an audio source sits within it. Changing early reflection characteristics changes the perceived location of the reflecting surfaces surrounding the audio source.

This is commonly accomplished in digital reverberation simulations by using multiple delay taps at different levels that occur in different positions in the stereo spectrum (through panning). Long reverberation generally occurs after early reflections dissipate.

Reverb One provides a variety of early reflections models. These let you quickly choose a basic acoustic environment, then tailor other reverb characteristics to meet your precise needs.

Early Reflections section

ER Settings Selects an early reflection preset. These range from realistic rooms to unusual reflective effects. The last five presets (Plate, Build, Spread, Slapback and Echo) feature a nonlinear response.

Early reflection presets include:

- Room: Simulates the center of a small room without many reflections.
- Club: Simulates a small, clear, natural-sounding club ambience.
- Stage: Simulates a stage in a medium-sized hall.
- Theater: Simulates a bright, medium- sized hall.
- Garage: Simulates an underground parking garage.
- Studio: Simulates a large, live, empty room.
- Hall: Places the sound in the middle of a hall with reflective, hard, bright walls.
- Soft: Simulates the space and ambience of a large concert hall.
- Church: Simulates a medium-sized space with natural, clear-sounding reflections.
- Cathedral: Simulates a large space with long, smooth reflections.
- Arena: Simulates a big, natural-sounding empty space.
- Plate: Simulates a hard, bright reflection. Use the Spread control to adjust plate size.
- Build: A nonlinear series of reflections
- Spread: Simulates a wide indoor space with highly reflective walls.
- Slapback: Simulates a large space with a long-delayed reflection.
- Echo: Simulates a large space with hard, un-natural echoes. Good for dense reverb.

Level Controls the output level of the early reflections. Turning the Early Reflections Level slider completely off produces a reverb made entirely of reverb tail.

Spread Globally adjusts the delay characteristics of the early reflections, moving them closer together or farther apart. Use Spread to vary the size and character of an early reflection preset. Setting the Plate preset to a Spread value of 50%, for example, will change the reverb from a large, smooth plate to a small, tight plate.

Delay Master Determines the amount of time that elapses between the original audio event and the onset of early reflections.

Early Reflect On Toggles early reflections on or off. When early reflections are off, the reverb consists entirely of reverb tail.

Reverb Graphs

The reverb graphs display information about the tonal spectrum and envelope contour of the reverb. The Reverb EQ and Reverb Color graphs provide graphic editing tools for shaping the harmonic spectrum of the reverb.

Reverb EQ

You can use this 3-band equalizer to shape the tonal spectrum of the reverb. The EQ is post-reverb and affects both the reverb tail and the early reflections.
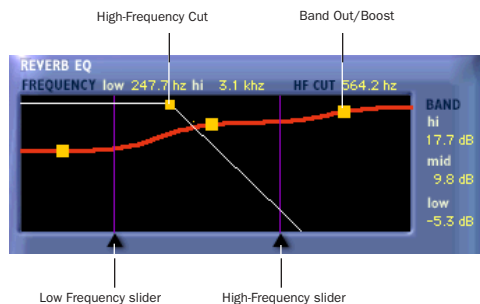
Reverb EQ graph

Frequency Sliders Sets the frequency boundaries between the low, mid, and high band ranges of the EQ.

The low frequency slider (60.0 Hz–22.5 kHz) sets the frequency boundary between low and mid cut/boost points in the EQ.

The high-frequency slider (64.0 Hz–24.0 kHz) sets the frequency boundary between the mid and high cut/boost points in the EQ.

Band Breakpoints Control cut and boost values for the low, mid, and high frequencies of the EQ. To cut a frequency band, drag a breakpoint downward. To boost, drag upward. The adjustable range is from –24.0 dB to 12.0 dB.

HF Cut Breakpoint Sets the frequency above which a 6 dB/octave low pass filter attenuates the processed signal. It removes both early reflections and reverb tails, affecting the overall high-frequency content of the reverb. Use the HF Cut control to roll off high frequencies and create more natural-sounding reverberation. The adjustable range is from 120.0 Hz to 24.0 kHz.

Reverb Color

You can use the Reverb Color graph to shape the tonal spectrum of the reverb by controlling the decay times of the different frequency bands. Low and high crossover points define the cut and boost points of three frequency ranges.

For best results, set crossover points at least two octaves higher than the frequency you want to boost or cut. For example, to boost a signal at 100 Hz, set the crossover to 400 Hz.

Set the crossover to 500 Hz to boost low frequencies most effectively. Set it to 1.5 kHz to cut low frequencies most effectively.
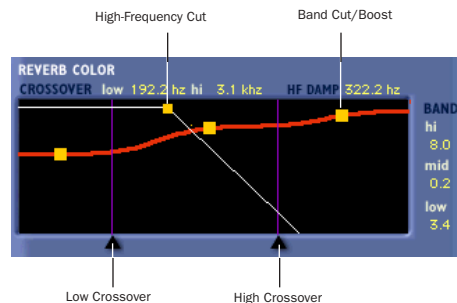
Reverb Color graph

Crossover Sliders Sets the frequency boundaries between the low, mid, and high frequency ranges of the reverberation filter.

The low-frequency slider sets the crossover frequency between low and mid frequencies in the reverberation filter. The adjustable range is from 60.0 Hz to 22.5 kHz.

The high-frequency slider sets the crossover frequency between mid and high frequencies in the reverberation filter. The adjustable range is from 64.0 Hz to 24.0 kHz.

Band Breakpoints Controls cut and boost ratios for the decay times of the low, mid, and high-frequency bands of the reverberation filter. To cut a frequency band, drag a breakpoint downward. To boost, drag upward. The adjustable range is from 0.125 to 8.0.

HF Damp Breakpoint Sets the frequency above which sounds decay at a progressively faster rate. This determines the decay characteristic of the high-frequency components of the reverb.

HF Damp works in conjunction with HF Cut to shape the overall high -frequency contour of the reverb. HF Damp filters the entire reverb with the exception of the early reflections. At low set-

tings, high frequencies decay more quickly than low frequencies, simulating the effect of air absorption in a hall. The adjustable range is from 120.0 Hz to 24.0 kHz.

Reverb Contour

The Reverb Contour graph displays the envelope of the reverb, as determined by the early reflections and reverb tail.
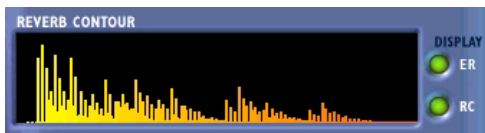
Reverb Contour graph

ER and RC Buttons Toggles the display mode. Selecting ER (early reflections) displays early reflections data in the graph. Selecting RC (reverb contour) displays the initial reverberation envelope in the graph. Early Reflections and Reverb Contour can be displayed simultaneously.

Other Controls

In addition to its reverb-shaping controls, Reverb One also features level metering and online help.

Online Help

To use online help, click the name of any control or parameter and an explanation will appear. Clicking the Online Help button itself provides further details on using this feature.

Online help

Input Level Meters

Input meters indicate the input levels of the dry audio source signal. Output meters indicate the output levels of the processed signal.

An internal clipping LED will light if the reverb is overloaded. This can occur even when the input levels are relatively low if there is excessive feedback in the delay portion of the reverb. To clear the Clip LED, click it.
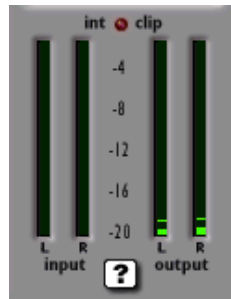
Reverb One meters

chapter 9

SoundReplacer

SoundReplacer is an AudioSuite plug-in designed to replace audio elements such as drums, percussion, and sound effects in Pro Tools tracks with alternate sounds. SoundReplacer can quickly and intelligently match the timing and dynamics of original performance material, making it ideal for both music and audio post production.

SoundReplacer features:

- Sound replacement with phase-accurate peak alignment
- Intelligent tracking of source audio dynamics for matching the feel of the original performance
- Three separate amplitude zones per audio event for triggering different replacement samples according to performance dynamics
- Zoomable waveform display for precision threshold/amplitude zone adjustment
- Crossfading or hard-switching of replacement audio in different amplitude zones for optimum realism and flexibility
- Online help

Audio Replacement Techniques

Replacing audio elements during the course of a recording session is a fairly common scenario. In music production it is often done in order to replace or augment an element that lacks punch. In film or video post-production it is typically done to improve or vary a specific sound cue or effect.

In the past, engineers and producers had to rely on sampling audio delay lines or MIDI triggered audio samplers—methods that had distinct disadvantages. Delay lines, for example, support only a single replacement sample, and while they can track the amplitude of the source events, the replacement sample itself remains the same at different amplitude levels.

The result is static and unnatural. In addition to these drawbacks, sample triggers are notoriously difficult to set up for accurate timing.

Similarly, with MIDI triggered samplers, MIDI timing and event triggering are inconsistent, resulting in problems with phase and frequency response when the original audio is mixed with the triggered replacement sounds.

The SoundReplacer Solution

SoundReplacer solves these timing problems by matching the original timing and dynamics of the source audio while providing three separate amplitude zones per audio event. This lets you trigger different replacement samples according to performance dynamics.

Each replacement sample is assigned its own adjustable amplitude zone. Variations in amplitude within the performance determine which sample is triggered at a specific time. For example, you could assign a soft snare hit to a low trigger threshold, a standard snare to a medium trigger threshold, and a rim shot snare to trigger only at the highest trigger threshold.

Replacement samples that are triggered in rapid succession or in close proximity to each other will overlap naturally—avoiding the abrupt sound truncation that occurs on many samplers.

In addition to its usefulness in music projects, SoundReplacer is also an extremely powerful tool for sound design and post production. Morphing gun shots, changing door slams, or adding a Doppler effect can now be accomplished in seconds rather than minutes—with sample-level precision.

Replacement audio events can be written to a new audio track, or mixed and re-written to the source audio track. Sample thresholds can be amplitude-switched between the replacement samples, or amplitude crossfaded for seamless transitions.

SoundReplacer Controls

SoundReplacer

Waveform Display
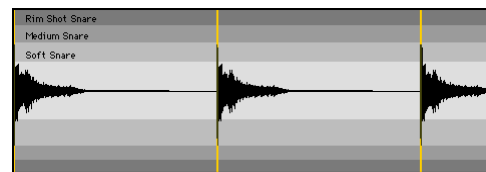
Waveform display with trigger markers shown

The waveform display shows the audio that you have selected for replacement. When you select audio on the source track, then open Sound-Replacer, the audio waveform will automatically be displayed here.

Once the audio selection is displayed, you can load the desired replacement samples and adjust their trigger thresholds while viewing the waveform peaks. Trigger markers then appear in the waveform, indicating the points at which the samples will be triggered. The color of each marker indicates which threshold/replacement sample will be triggered. The Zoomer lets you increase or decrease waveform magnification here to help accurately set trigger thresholds.

If you change the audio selection on the source track, click Update to update the waveform display. If Auto Update is selected, SoundReplacer automatically updates the waveform display each time you make a new selection or begin playback.

💡 *If you frequently change selections or start and stop playback, turn off Auto Update to prevent too-frequent redraws.*

Trigger Threshold

Threshold controls

The color-coded Trigger Threshold sliders set a total of three amplitude zones (one for each replacement audio file) for triggering replacement samples:

- The yellow slider represents amplitude zone 1, the lowest-level trigger.
- The red slider represents amplitude zone 2, the middle-level trigger.
- The blue slider represents amplitude zone 3, the highest-level trigger.

With a replacement sample loaded, drag the Threshold slider to the desired amplitude level. Color-coded trigger markers will appear in the Waveform at points where the source audio signal exceeds the threshold set for that amplitude zone. The replacement sample will be triggered at these points.

The color of the Trigger markers correspond to the matching Threshold slider. This lets you see at a glance which replacement samples will be triggered and where they will be triggered.

💡 *If you zoom the waveform display below a specific Trigger Threshold slider's amplitude zone, the slider will be temporarily unavailable. To access the slider again, zoom back out to an appropriate magnification level.*

Load/Unload Sound

Load/Unload Sound

Clicking the Load/Unload Sound icons loads or unloads replacement samples for each of the three trigger threshold amplitude zones. Clicking the Floppy Disk icon loads a new sample (or replaces the current sample). Clicking the Trash Can icon unloads the current sample.

⚠ *SoundReplacer does not perform a sample rate conversion before loading replacement samples if they are at a different sample rate from the session. Replacement samples should be at the same sample rate as the session, otherwise they will playback at the wrong speed and pitch.*

To audition a replacement sample before loading it into SoundReplacer, use the Import Audio command in Pro Tools. Once you have located and previewed the desired audio file, you can then load it into SoundReplacer using the Load/Unload Sound icons.

⚠ *SoundReplacer does not load regions that are part of larger audio files. To use a region as a replacement sample, you must first save it as an individual audio file.*

The Zoomer

Zoomer

The Zoomer increases or decreases magnification of the waveform data currently visible in the center of the waveform display so that you can more accurately set sample trigger thresholds.

• To zoom in on amplitude, click the Up Arrow.

• To zoom out on amplitude, click the Down Arrow.

• To zoom in on time, click the Right Arrow.

• To zoom out on time, click the Left Arrow.

⚠ *If you zoom the waveform display below a specific Threshold slider's amplitude zone, the slider will be temporarily unavailable. To access the slider again, zoom back out to an appropriate magnification level.*

Crossfade

When Crossfade is selected, SoundReplacer crossfades between replacement audio files in different amplitude zones. This helps smooth the transition between them.

When Crossfade is deselected, SoundReplacer hard switches between replacement audio files in different amplitude zones.

Crossfading is particularly useful for adding a sense of realism to drum replacement. Crossfading between a straight snare hit and a rim shot, for example, results in a much more "live" feel than simply hard switching between the two samples.

Peak Align

When Peak Align is on, SoundReplacer aligns the peak of the replacement file with the peak of the source file in a way that best maintains phase coherency. When Peak Align is off, SoundReplacer aligns the beginning of the replacement file with the trigger threshold point.

Depending on the characteristics of your source and replacement audio files, using Peak Align can significantly affect the timing of audio events in the replacement file. It is essential that you choose the option most appropriate to the material that you are replacing.

📖 *For more information on using Peak Align, see "Getting Optimum Results With SoundReplacer" on page 100.*

Update

When you click Update, the waveform display is redrawn, based on the audio currently selected on the source track. Each time you make a new selection on a source track, you must click Update for SoundReplacer to draw the waveform of the selection.

Auto Update

When Auto Update is selected, SoundReplacer automatically updates the waveform display each time you make a new selection on a source track. If you frequently change selections or start and stop playback, you may want to deselect Auto Update to prevent frequent redraws.

Mix

Mix adjusts the mix of the replacement audio file with the original source file. Higher percentage values weight the mix toward the replacement audio. Lower percentage values weight the mix toward the original source audio.

The Mix button toggles the Mix control on and off. When Mix is toggled off, the balance is instantly set to 100% replacement audio.

💡 *Setting Mix to 50% and clicking Preview lets you audition source audio and replacement audio together to check the accuracy of replacement triggering timing.*

Dynamics

Dynamics controls how closely the audio events in the replacement file track the dynamics of the source file:

◆ Setting the ratio to 1.00 matches the dynamics of the source file.

◆ Increasing the ratio above 1.00 expands the dynamic range so that softer hits are softer, and louder hits are louder. This is useful if the source material lacks variation in its dynamic range.

◆ Decreasing the ratio below 1.00 compresses the dynamic range so that there is less variation between loud and soft hits. This is useful if the dynamics of the source material are too extreme.

The Dynamics button provides a quick means of toggling on and off the Dynamics control. When Dynamics is toggled off, SoundReplacer will not track changes in the source audio file's dynamics. Audio events in the resulting replacement audio file will uniformly be at the amplitude of the replacement samples themselves, with no variation in dynamics.

Online Help

Online help

To use online help, click the name of any control or parameter and an explanation will appear. Clicking the Online Help button provides further details on using this feature.

Using SoundReplacer

Following are basic guidelines for using SoundReplacer effectively. Also see "Getting Optimum Results With SoundReplacer" on page 100.

To use SoundReplacer:

1 On the source track, select the audio you want to replace. Only selected audio will be replaced.

2 Choose SoundReplacer from the AudioSuite menu.

3 Click the Load Sound icon (the icon beneath the yellow slider) to import the replacement sound for amplitude zone 1.

4 Locate the desired audio file and click Open.

5 Adjust the amplitude zone slider.

6 Repeat steps 3–5 to load replacement sounds into amplitude zones 2 and 3.

💡 *If you use only a single replacement sample, you should still set all three amplitude zones for optimum results. This will ensure accurate triggering. For details, See "Mapping The Same Sample Into Multiple Amplitude Zones" on page 101.*

7 To align the amplitude peak in the replacement file(s) to threshold trigger markers in the source audio, enable Peak Align.

8 Click Preview to audition the replacement audio.

9 Adjust the Threshold sliders to fine tune audio replacement triggering.

10 Adjust the Dynamics slider to fine tune how SoundReplacer tracks and matches changes in the source audio's dynamics.

11 Adjust the Mix slider to get the desired balance between replacement audio and source audio.

To begin AudioSuite processing:

1 Adjust the AudioSuite file controls. These settings will determine how the file is processed and what effect the processing will have on the original regions. Here are some guidelines:

◆ To process the selected region only in the track in which it appears, select Playlist from the Selection Reference pop-up.

– or –

To process the selected region in the Audio Regions list only, select Region List from the Selection Reference pop-up.

◆ To process and update every occurrence of the selected region throughout your session, enable the Use In Playlist button (and also select Region List from the Selection Reference pop-up).

– or –

If you do not want to update every occurrence of the selected region, disable the Use In Playlist button.

◆ If you have selected multiple regions for processing and want to create a new file that connects and consolidates all of these regions together, select Create Continuous File from the File Mode pop-up menu.

⚠ *Because SoundReplacer does not allow destructive processing, the AudioSuite Overwrite Files option is not available.*

2 From the Destination Track pop-up, choose the destination for the replacement audio.

3 Click the Process button.

Getting Optimum Results With SoundReplacer

Getting optimum results with SoundReplacer generally means making sure that the audio events in the replacement audio file have accurate timing in relation to the source audio. The techniques given here help ensure this.

Using Peak Align

Proper use of the Peak Align feature can significantly improve the results of sound replacement. Since turning Peak Align on or off controls how SoundReplacer aligns the replacement audio with the source audio, it will significantly affect the timing of audio events in the replacement file.

In general:

◆ Turn on Peak Align if you are replacing drum or percussion sounds whose peak level occurs at the initial attack.

◆ Turn off Peak Align if you are replacing sounds whose peak level occurs somewhere *after* the initial attack. Peak Align should also be turned off if the sounds you are replacing are *not* drum or percussion sounds.

To illustrate why Peak Align makes a difference, look at the following illustrations:
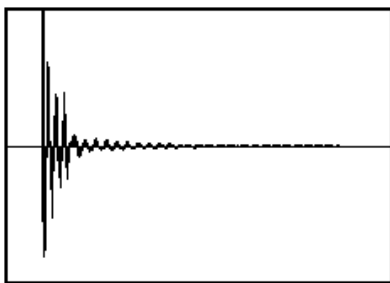
Figure 1. A fast-peaking kick drum
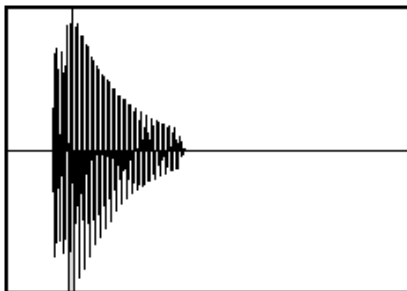
Figure 2. A slower-peaking kick drum

Figure 1 shows a kick drum whose peak level occurs at its initial attack.

Figure 2 shows a kick drum whose peak level occurs *after* its initial attack.

If you turn on Peak Align and attempt to replace the fast-peaking kick with the slow-peaking kick (or vice-versa), SoundReplacer will align their peaks—which occur at different points in the sound. The audible result would be that the replacement audio file (slow-peaking kick) would trigger too early.

Mapping The Same Sample Into Multiple Amplitude Zones

If you are performing drum replacement and intend to use just a single replacement sample, mapping it into multiple amplitude zones will ensure more accurate triggering. Here is why:

Imagine that you are replacing a kick drum part. If you look at the waveform of a kick drum, you will often see a "pre-hit" portion of the sound that occurs as soon as the ball of the kick pedal hits the drum. This is rapidly followed by the denser attack portion of the sound, where most of sound's weight is.
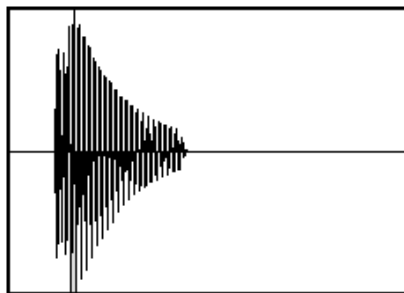
A kick drum with a pre-hit preceding a denser attack

With a sound like this, using a single amplitude threshold presents a problem because typically, in pop music, kick drum parts consist of loud accent hits and softer off-beat hits that are often 6 dB or more lower in level.

If you use a single amplitude threshold to trigger the replacement sample, you have to set the threshold low enough to trigger at the soft hits. The problem occurs at the loud hits: The threshold is now set so low that the pre-hit portion of the loud hits can exceed the threshold—triggering the replacement sample too early. This results in a replacement track with faulty timing.
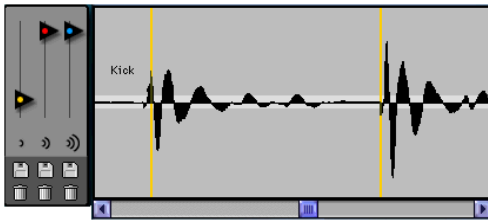
A single low threshold causes the second, louder kick to trigger too early, as evidenced by the trigger marker at the very start of the waveform.

The best way to avoid this problem is to set multiple threshold zones for the same sample using a higher threshold for the louder hit. Soft hits will trigger threshold 1 and louder hits will trigger threshold 2.
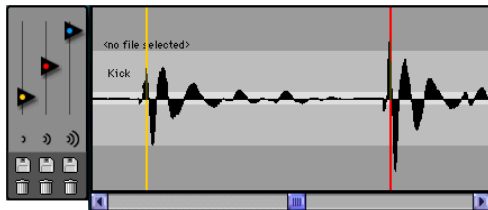
Using a second, higher threshold for the louder kick will make it trigger properly, as shown by the now properly-aligned trigger marker.

To set the precise threshold for louder hits, you may need to zoom in carefully to examine the waveform for trigger points (indicated by color-coded trigger markers) and then Command-drag the Threshold slider for more precise adjustment.

If there is a great deal of variation in the dynamics of the source audio, you may need to use all three Trigger Thresholds/Amplitude Zones for optimum results.

If only one replacement sample is loaded into SoundReplacer and it is loaded into Trigger threshold/amplitude zone 1 (yellow), SoundReplacer will let you use the red and blue Trigger Threshold sliders to set Amplitude Zones 2 and 3—without having to load the same sample again.

Using the Audio Files Folder for Frequently Used Replacement Files

If you often use the same settings and replacement sounds in different sessions, Sound-Replacer provides a convenient way to keep the replacement audio files and settings linked together.

When you choose a setting from the Librarian menu, SoundReplacer looks for the replacement audio files associated with the setting. Sound-Replacer first looks in the audio file's original hard disk location (at the time you saved the setting).

If it is not there, SoundReplacer looks in a folder named Audio Files within SoundReplacer's Root Plug-In Settings folder (Plug-In Settings/SoundReplacer/Audio Files).

If SoundReplacer finds the replacement audio file there, the Settings file will load with the associated audio.

By always putting replacement audio files in this special folder, you can freely exchange Sound-Replacer settings—and the audio files associated with them—with other users.

⚠ *Do not create subfolders within Sound-Replacer's Audio Files folder. Files located within subfolders are not recognized.*

SoundReplacer Demo Session

SoundReplacer includes a Demo session that illustrates useful applications of this plug-in.

Before you start:

1 Open the Demo session.

2 Choose SoundReplacer from the AudioSuite menu.

3 Position SoundReplacer on the bottom-left hand corner of the screen, so that you can clearly see the top audio track, as well as the Memory Locations window.

4 Begin playback of the Demo session. A voice-over will guide you through the basic use of SoundReplacer.

Example 1

This example illustrates basic sound replacement on a kick drum track.

In the Demo session:

1 Click memory Location #5, Kick Replace 1. A kick drum source track is selected and centered on screen.

2 Choose Kick Double Trigger Zone from the Librarian menu.

3 Click Update.

4 Use SoundReplacer's Zoomer to zoom out to see several of the kick drum hits.

5 Click Preview to hear the replacement sound triggered with the source kick track.

6 Adjust the wet/dry mix with the Mix slider. Turning off the Mix button lets you eliminate the dry sound entirely from the wet/dry mix.

7 Click Process to write the replacement file to disk on the same track.

– or –

Write the replacement file to a different track by choosing a destination from the Destination Track pop-up menu. An empty audio track has been provided in the Demo session for this purpose.

Example 2

This example demonstrates the use of multiple trigger zones using the same replacement sound. This allows a phase-coherent sound replacement without unwanted results such as double-attack flams, comb filter-like effects, or mistriggered samples.

Optimum results are achieved by carefully adjusting the two threshold zones so that softer kick drum hits (the yellow zone) and louder hits (the red zone) trigger at exactly the right points.

Loading a replacement sound into the yellow amplitude zone automatically loads it into the red and blue amplitude zones as well. This allows more subtle adjustment and control of triggering.

The example also illustrates why it is useful to turn off the Peak Align option in cases where the signal peak in a sound occurs after the initial attack.

In the Demo session:

1 With SoundReplacer's Zoomer, zoom in closely on one of the yellow trigger markers in the waveform display. These vertical lines show exactly where triggers occur.

2 Adjust the yellow Trigger Threshold slider and observe how it affects the placement of these trigger points and markers.

3 Click the Compare button to return to the original settings.

4 Click Memory Location #5.

5 From the Librarian menu, choose 808 Trigger No Peak Align. This kick drum sound has a strong "boom" component that occurs after the initial attack transient, making it louder than the actual attack itself.

6 Click Preview, then toggle Peak Align on and off to compare the results. (Give the preview a few moments to update each time you do this.)

You should clearly hear a difference in the timing of the replacement sound. When Peak Align turned on, the replacement sound plays out of sync because the initial attack of the sound is not the actual signal peak—the subsequent "boom" is.

Example 3

This example demonstrates sound replacement for special effects. The replacement audio file consists of a clave hit with backwards reverb preceding it and a normal reverb decay following it.

It illustrates how Peak Align can be used to back time a replacement sound to match the attack of the source file.

In this example, Peak Align is turned on. This forces SoundReplacer to find the peak in the replacement file (the clave hit), and align it with the kick drum hits on the source track.

In the Demo session:

1 From the Librarian menu, choose Reverse Sound Example.

2 Click Update.

3 Click Preview.

The replacement sound should accurately align with the original hit points.

Example 4

This example demonstrates two different snare sounds triggered from two amplitude zones.

In the Demo session:

1 Click Memory Location #6, Snare Replace 1.

2 Click Update.

3 From the Librarian menu, choose Snare dbl. Trigger zone.

4 Click Preview.

The replacement sound should switch from a medium snare hit to a rim shot at the appropriate points.

Example 5

This example demonstrates how the Crossfade option can be used to smooth the transition between replacement sounds triggered in succession.

In this example, there are five hits that vary in volume from very soft to very loud. All three amplitude zones are used, with different replacement files for each zone.

In the Demo session:

1 Zoom the waveform display out and compare the yellow trigger points with the red trigger points.

2 Click Memory Location #7, Gradual Snare. A sample snare drum source track containing several hits that vary in volume from very soft to very loud is selected and centered on-screen.

3 Click Update and zoom out to show all five drum hits in the waveform display.

4 From the Librarian menu, choose Snare Triple Trigger Zone.

5 Click Preview.

6 Preview the result.

With Crossfade selected, SoundReplacer smoothly crossfades between the softest sound and the loudest sound. If Crossfade is deselected, SoundReplacer will hard switch on each zone.

SurroundScope

SurroundScope is a TDM and RTAS plug-in that assists with multichannel mixing by visually representing the phase, individual input levels, and position of multichannel signals in the surround field.

SurroundScope supports all industry-standard, multichannel formats, including LCR, Quad, LCRS, and 5.0, 5.1, 6.0, 6.1, 7.0, and 7.1.

SurroundScope features include:

- 360° Surround Display indicates signal position within the surround field.

- Lissajous meter ensures perfect stereo images by indicating amplitude and phase.

- High-resolution meters accurately display individual input channel levels as low as –60 dB.

- Stereo Phase meter indicates signal phase.

- SurroundScope operates in both TDM and RTAS plug-in formats.

📖 *For information on using SurroundScope during mixdown with Pro Tools, see the Pro Tools Reference Guide.*

DSP Requirements

Each SurroundScope plug-in used in a session requires a portion of the total DSP resources available. The amount of DSP available depends on the number of DSP cards in your Pro Tools system. See Appendix A, "DSP Requirements for TDM Plug-Ins."

Using SurroundScope

Since it has no user-adjustable controls, SurroundScope is extremely easy to use. Once inserted on a multichannel track or Master Fader, you can use it to accurately monitor phase, position, and input level metering of multichannel signals.

📖 *For more information about multichannel signals and multichannel mixing, refer to the Pro Tools Reference Guide.*

To use SurroundScope:

1 Click the Insert Selector on the desired multichannel track and choose SurroundScope from the plug-in pop-up menu.

2 Begin audio playback.

3 Watch the Surround Display, Phase meter, and Input Level meters to visually monitor the multichannel signal.

SurroundScope Displays

Each of SurroundScope's interactive meters displays a different type of information about a multichannel audio signal. Following is an explanation of these meters and their behavior.
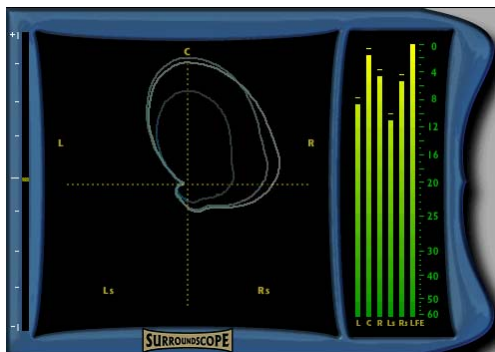
SurroundScope

Surround Display

The Surround Display shows signal position within the surround field. When SurroundScope is inserted on a track, it detects the multichannel format of the track and displays each speaker channel in the signal in a circle around the Surround Display's center.

Each speaker channel is indicated by letter and is equidistant from the center of the 360° surround field.
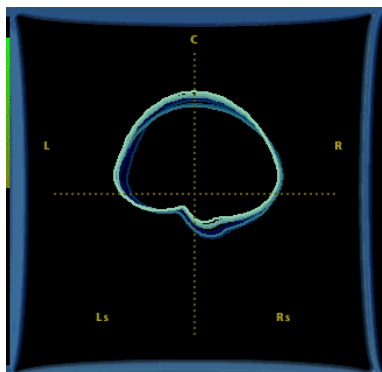
Surround Display

During audio playback, the Surround Display generates a circular metering image whose shape and position change according to the amplitude energy emanating from the various speaker channels in the signal.

To understand the precise location of an audio signal within the sound field, observe how the circular sound pattern moves:

◆ A perfect circle in the center of the Surround Display denotes a perfectly centered surround signal—one that is panned equally to all channels.

◆ A signal that balloons toward the outer edges of the display indicates which channels emanate the strongest signal.

◆ A signal that is completely panned to one speaker channel appears teardrop-shaped and points toward the channel emanating the signal.

Lissajous Meter

When SurroundScope is inserted on a stereo track, the standard Surround Display is replaced by the Lissajous Meter.
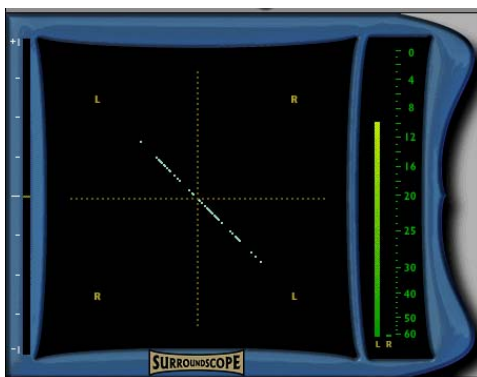
Lissajous Meter with audio panned predominantly left

The Lissajous Meter shows the relationship between the amplitude and phase of a stereo signal, enabling you to monitor and ensure perfect stereo imaging.

The Lissajous Meter display is divided into four quadrants, with left and right channels arranged diagonally. When audio is panned predominantly to a particular speaker channel, a diagonal line appears, indicating the channel.

The Lissajous Meter displays in-phase material as a vertical line and out-of-phase material as a horizontal line.

Phase Meter

The Phase Meter indicates the phase coherency of any two channels of a multichannel signal. This enables you to see at a glance if a signal is in or out of phase.

Phase Meter

The Phase Meter is green when the signal is positively out of phase (values from 0 to +1) and red when the signal is negatively out of phase (values from 0 to –1).

At center or zero position, the signal is a perfect stereo image. At the +1 position, the signal is a perfect mono image. At the –1 position, the signal is 100% out of phase.

By default, the meter displays the phase of the left and right channels. However, you can select any two channels for comparison by clicking them.

To choose the channels you want to compare on the Phase meter:

1 With SurroundScope inserted on a track, click in the Surround display on the letter corresponding to the desired speaker channel.

2 Click the letter corresponding to the second desired speaker channel.
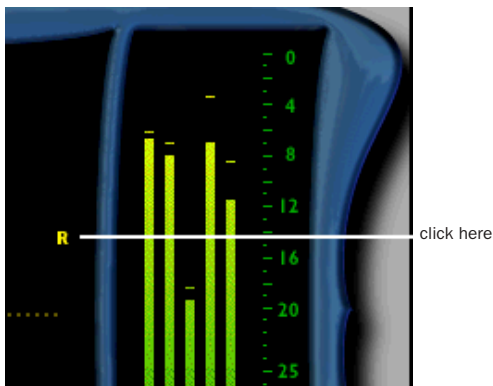
Selecting a channel for the Phase Meter

Once selected, the speaker labels are highlighted for easy visual reference.

3 Begin audio playback.

4 Watch the Phase Meter to monitor the signal.

Input Level Meters

The Input Level meters provide decibel level readings for each channel of a multichannel signal, including readings for an LFE channel if present. Surround Scope's high resolution Level meter is an RMS-style level meter, not a peak meter like those located on Pro Tools disk tracks. Meter range is from 0 to –60dB. They have a a long range with full resolution through –30dB, and then a tapered scale through –60dB.
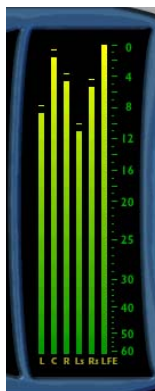
Input Level Meters

☼ *The Input Level meters always appear in standard Film layout (L, C, R, Ls, Rs, LFE) for 5.1 format mixes.*

DSP Requirements for TDM Plug-Ins

The number of TDM plug-ins you can use at one time depends on how much DSP power is available in your system. Since the TDM hardware on Pro Tools cards provide dedicated DSP for plug-ins, plug-in performance isn't limited by CPU processing power.

The tables on the following pages show the total number of instances of each plug-in that can be powered by a single DSP chip on Pro Tools|HD and Pro Tools|24 MIX cards. DSP usage differs according to card type.

There are a total of nine DSP chips on a Pro Tools|HD card (HD Core, HD Process, and HD Accel). HD Core and HD Process cards provide identical chip sets. HD Accel cards provide newer, more powerful DSP chips (making the HD Accel card ideal for DSP-intensive plug-ins, and for high sample rate sessions). There are six DSP chips on a Pro Tools|24 MIX card.

Not all plug-ins are supported on all types of chips. The following tables indicate the number of compatible chips per card.

⚠ *The tables show theoretical maximum performance when no other plug-ins are sharing available DSP resources. You will typically use more than one type of plug-in simultaneously.*

Plug-ins used in multi-mono format on greater-than-stereo tracks require one mono instance per channel of the multi-channel audio format. For example, a multi-mono plug-in used on a 5.1 format track, requires six mono instances since there are six audio channels in the 5.1 format.

DSP Requirements

Digidesign plug-ins have the following DSP requirements:

HD Accel Card

Mono and Stereo

Table 2. Maximum instances of real-time TDM plug-ins per DSP chip for an HD Accel card, at different sample rates (mono and stereo).

Sample Rate:	44.1/48 kHz		88.2/96 kHz		174.6/192 kHz		DSP Chips per HD Accel Card
Plug-In	Mono	Stereo	Mono	Stereo	Mono	Stereo	
D-Fi (Lo-Fi)	16	8	8	4	3	n/a	9
D-Fi (Recti-Fi)	28	14	13	6	6	3	9
D-Fi (Sci-Fi)	18	9	7	4	4	n/a	9
DINR (BNR)	1	1	1	1	n/a	n/a	9
DINR (Hum Removal)	n/a	n/a	n/a	n/a	n/a	n/a	9
Maxim	20	10	9	4	4	n/a	9
Bruno/Reso	1	1	1	1	n/a	n/a	6
Reverb One	2	2	1	1	1	1	9
SurroundScope	n/a	53	n/a	25	n/a	11	9

Multichannel

Table 3. Maximum instances of real-time TDM plug-ins per DSP chip for an HD Accel card at 48 kHz*

Plug-In	Quad & LCRS	5.1 & 6.0	DSP Chips per HD Accel Card
D-Fi (Lo-Fi)	4	n/a	9
D-Fi (Recti-Fi)	7	n/a	9
D-Fi (Sci-Fi)	4	n/a	9
DINR (BNR)	partial	n/a	9
DINR (Hum Removal)	n/a	n/a	9
Maxim	2	1	9
Bruno/Reso	partial	partial	6
Reverb One	partial	partial	9
SurroundScope	36	24	9

"partial" indicates that a single instance of the plug-in is sharing more than 1 DSP chip.

HD Core and HD Process

Mono and Stereo

Table 4. Maximum instances of real-time TDM plug-ins per DSP chip for an HD Core or HD Process card, at different sample rates (mono and stereo).

Sample Rate:	44.1/48 kHz		88.2/96 kHz		174.6/192 kHz		DSP Chips per HD Accel Card
Plug-In	Mono	Stereo	Mono	Stereo	Mono	Stereo	
D-Fi (Lo-Fi)	7	3	3	1	1	n/a	9
D-Fi (Recti-Fi)	12	6	5	2	2	1	9
D-Fi (Sci-Fi)	7	3	3	1	1	n/a	9
DINR (BNR)	1	1	1	1	n/a	n/a	9
DINR (Hum Removal)	n/a	n/a	n/a	n/a	n/a	n/a	9
Maxim	8	4	4	2	1	n/a	9
Bruno/Reso	1	1	1	1	n/a	n/a	6
Reverb One	1	1	1	1	1	1	9
SurroundScope	n/a	22	n/a	10	n/a	4	9

"partial" indicates that a single instance of the plug-in is sharing more than 1 DSP chip.

For maximum instance counts for plug-ins that are compatible with the most powerful chips on HD Accel, see "HD Accel Card" on page 112.

Multichannel

Maximum instances of real-time TDM plug-ins per DSP chip for an HD Core or HD Process card at 48 kHz*

Plug-In	Quad & LCRS	5.1 & 6.0	DSP Chips per HD Core or HD Process Card
D-Fi (Lo-Fi)	1	n/a	9
D-Fi (Recti-Fi)	3	n/a	9
D-Fi (Sci-Fi)	1	n/a	9
DINR (BNR)	partial	partial	9
DINR (Hum Removal)	n/a	n/a	9
Maxim	2	1	9
Bruno/Reso	partial	partial	6
Reverb One	partial	partial	9
SurroundScope	20	15	9

"partial" indicates that a single instance of the plug-in is sharing more than 1 DSP chip.

MIX Card

Table 5. Maximum instances of real-time TDM plug-ins per DSP chip for a Pro Tools|24 MIX card

Plug-In	Mono	Stereo	DSP Chips per MIX Card
D-Fi (Lo-Fi)	5	2	6
D-Fi (Recti-Fi)	8	4	6
D-Fi (Sci-Fi)	6	3	6
DINR (BNR)	1	1	3
DINR (Hum Removal)	n/a	n/a	n/a
Maxim	8	4	3
Bruno/Reso	1	1	3

DSP Farm Card

Table 6. Maximum instances of real-time TDM plug-ins per DSP chip for a DSP Farm card

Plug-In	Mono	Stereo	DSP Chips per DSP Farm Card
D-Fi (Lo-Fi)	3	1	4
D-Fi (Recti-Fi)	4	2	4
D-Fi (Sci-Fi)	3	1	4
DINR (BNR)	1	1	4
DINR (Hum Removal)	4	2	4
Maxim	4	2	4
Bruno/Reso	1	1	4

appendix b

DSP Delays Incurred by TDM Plug-Ins

Table 3 on page 118 shows the delays inherent in each type of Digidesign TDM plug-in. Delays matter only if you use a real-time TDM plug-in on one channel of a stereo or multichannel signal but not the others. This can cause the channels to be out of phase.

You can use the DigiRack TimeAdjuster plug-in to compensate for these delays. See the *DigiRack Plug-Ins Guide* for more information.

📄 *See the Pro Tools Reference Guide for details on DSP-induced delays.*

Table 3. Samples of delay incurred by each TDM plug-in on HD, MIX, and DSP Farm cards

Plug-In	Samples of Delay on Pro Tools|HD cards	Samples of Delay on Pro Tools|24 MIX cards	Samples of Delay on DSP Farm Cards
Bruno	3	3	3
Lo-Fi	4	4	4
Recti-Fi	4	4	4
Sci-Fi	4	4	4
DINR (BNR)	1538/3074/not supported*	1538	1538
DINR (Hum Removal)	n/a	n/a	3
Maxim (TDM and RTAS)	1028/2052/4100*	1027	1027
Reso	3	3	3

BNR and Maxim have different delays at different sample rates: 48 kHz/96 kHz/192 kHz.

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