DigiRack™ Plug-Ins

Version 7.0

contents

chapter 1

Introduction

Plug-ins are special-purpose software components that provide additional signal processing functionality to Pro Tools.

The DigiRack plug-ins included with Pro Tools provide a comprehensive set of digital signal processing effects that include EQ, dynamics, delay, and other essential functions.

Plug-In Formats

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

(Pro Tools HD Only)

TDM (Time Division Multiplexing) plug-ins function as track inserts, are applied to audio during playback, and process audio non-destructively in real time. TDM plug-ins are designed for use with Pro Tools HD software on Pro Tools|HD 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 DSP cards (such as HD Core, HD Accel, or HD Process cards) in your computer. This power-on-demand aspect is a significant advantage of Pro Tools|HD 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 it will have on other aspects of your system's performance, such as maximum track count, number of available voices, the density of edits possible, and latency in automation and recording.

RTAS plug-ins can be used on both with Pro Tools HD, LE, and M-Powered.

AudioSuite Plug-Ins

AudioSuite plug-ins are used to process and modify audio files on disk, rather than non-destructively in real time. Depending on how you configure a non-real-time AudioSuite plug-in, it will either create an entirely new audio file, or alter the original source audio file. Audio-Suite plug-ins can be used on all Pro Tools systems and Avid software.

System Requirements

To use DigiRack plug-ins you need one of the following:

- A Digidesign-qualified Pro Tools system running Pro Tool
- A Digidesign-qualified Pro Tools system and a third-party software application that supports the Digidesign TDM, RTAS, or AudioSuite plug-in standard
- A qualified Avid Xpress, Avid Xpress DV or Avid DNA system

For complete system requirements visit the compatibility page of the Digidesign Web site (www.Digidesign.com/compato).

Compatibility Information

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 the Digidesign Web site (www.digidesign.com/compato).

Conventions Used in This Guide

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

Convention	Action
File > Save	Choose Save 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.

About www.digidesign.com

The Digidesign Web site (www.digidesign.com) is your best online source for information to help you get the most out of your Pro Tools system. The following are just a few of the services and features available.

Registration Register your purchase online. See the registration form included with your system for instructions.

Support Contact Digidesign Technical Support or Customer Service; download software updates and the latest online manuals; browse the Compatibility documents for system requirements; search the online Answerbase or join the worldwide Pro Tools community on the Digidesign User Conference.

Training and Education Study on your own using courses available online or find out how you can learn in a classroom setting at a certified Pro Tools training center.

Products and Developers Learn about Digidesign products; download demo software or learn about our Development Partners and their plug-ins, applications, and hardware.

News and Events Get the latest news from Digidesign or sign up for a Pro Tools demo.

To learn more about these and other resources available from Digidesign, visit our Web site (www.digidesign.com).

chapter 2

Installation

Installing Plug-Ins

The Digidesign DigiRack plug-ins are installed when you install Pro Tools. For more information about installing Pro Tools, see the *Getting Started Guide* that came with your system.

If installing multiple packages of plug-ins, do so in the following order:

1 DigiRack plug-ins (installed with Pro Tools).

2 Packs and other bundles included with your Digidesign system.

3 Any additional plug-ins. Check with the manufacturer of your plug-ins for compatibility information and support.

chapter 3

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 Pro Tools|HD systems.

RTAS Plug-Ins Rely on the processing power of your computer. RTAS plug-ins run on Pro Tools|HD 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 (Window > System Usage) shows how much DSP is available on your system, and gives an approximation of how it is currently being used.

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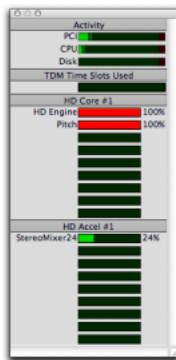
System Usage window

📄 *See Appendix A, "DSP Requirements for TDM Plug-Ins" for relative numbers of instances of each DigiRack TDM plug-in that can be powered by a single DSP chip. DSP capacity differs with 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.

To take advantage of this capability, plug-ins must be MultiShell II compatible. All DigiRack TDM plug-ins with the exception of Direct Connect and Pitch are MultiShell II compatible.

Some third-party plug-ins may not be MultiShell II compatible. To check compatibility, contact the developer.

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 (Window > 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 monitoring latency and using 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 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 Instrument or MIDI tracks.

To change the Hardware Buffer Size:

1 Choose Setup > 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 such as RTAS plug-in performance and screen redraws.

◆ 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 Setup > 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.)

3 Click OK.

📄 *For more information on the Hardware Buffer Size and CPU Usage Limit, refer to the Pro Tools Reference Guide.*

RTAS Plug-Ins on Auxiliary Inputs or Master Faders

With Pro Tools HD, RTAS plug-ins can be inserted on Auxiliary Input and Master Fader tracks, or after TDM plug-ins on any kind of track.

This can affect voice usage and total latency, and limitations on inserting or removing plug-ins during playback. For more information, see "RTAS Plug-Ins on Auxiliary Input and Master Fader Tracks" on page 29.

Delay in Signal Processing

DSP and host-based processing in digital audio systems incurs signal delay of varying amounts. Such delays can vary from as short as 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 cancellation problems.

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

Compensating for Delay

Use the following methods, as available, to compensate for processing delay.

Delay Compensation

Use Delay Compensation to automatically calculate and compensate for processing delay. For more information on Delay Compensation, see the *Pro Tools Reference Guide*.

💡 *Delay Compensation is automatically enabled on Pro Tools LE systems and cannot be turned off.*

TimeAdjuster

You can compensate for TDM or RTAS 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 "TimeAdjuster" on page 73 for more information about the TimeAdjuster plug-in.

📖 *For information about TimeAdjuster in Delay Compensation sessions, see the Pro Tools Reference Guide.*

Manual Delay Compensation

You can manually compensate for processing delay by first calculating the amount of delay on each track, and then nudging other track's playlists later or earlier in time. This method is useful when Delay Compensation is unavailable, or when you want to conserve resources.

💡 *You can also compensate for offsets incurred by processing delays by using the same plug-ins on all tracks.*

Low-Latency Recording with Instrument Plug-Ins

When an Instrument track containing an instrument plug-in (or a MIDI track routing MIDI data to an instrument plug-in) is record enabled, Pro Tools automatically suspends delay compensation through the main outputs of the au-

dio track, Instrument track, or Auxiliary Input on which the instrument plug-in is *inserted*. This allows for latency-free monitoring of the instrument plug-in during recording.

⚠️ *Delay Compensation for instrument plug-ins works only when all MIDI and audio connections take place inside of Pro Tools. For example, Pro Tools does not suspend delay compensation when you are using Re-Wire to connect software synthesizers and samplers.*

MIDI and Audio Processing Plug-Ins

Some plug-ins, such as Bruno and Reso, process audio while allowing MIDI data to control processing parameters. When you record enable an Instrument or MIDI track that is controlling an Audio processing plug-in, the track the plug-in is inserted on will go into low-latency mode, effectively making the processed audio play early.

To keep audio time-aligned when using a MIDI controlled plug-in on an audio track:

■ Start-Control-click (Windows) or Command-Control-click (Macintosh) the Track Compensation indicator for the audio track to apply delay compensation.

To keep audio time-aligned when using a MIDI controlled plug-in on an Auxiliary Input:

1 Start-Control-click (Windows) or Command-Control-click (Macintosh) the Track Compensation indicator for the Auxiliary Input to bypass delay compensation.

2 Enter the total system delay into the User Offset field.

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.

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.

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.
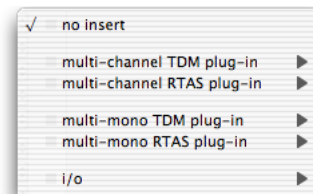
Plug-in formats

Mono Plug-Ins Used on mono tracks. Some mono plug-ins (such as Mod Delay) can generate stereo or greater-than-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 can 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 tan-

dem. You can unlink controls for independent adjustment using the Master Link button. See "Linking and Unlinking Controls on Multi-Mono Plug-ins" on page 19.

⚠ *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.

Organizing Plug-Ins

You can customize how plug-in lists (plug-in menus) are organized in the Insert selector and Plug-In selector.
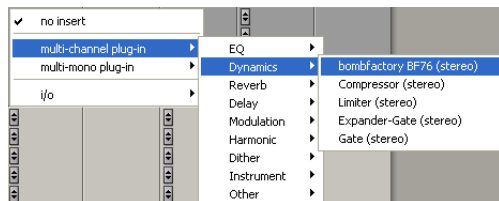
Plug-ins organized by categories

Plug-ins organized by manufacturer

Use the "Organize Plug-In Menus By" pop-up menu in the Pro Tools Display Preferences page to organize plug-in menus in a Flat List (with plug-ins in alphabetical order), Category, Manufacturer, or Category *and* Manufacturer.

"Organize Plug-In Menus By" Option

Flat List Organizes plug-ins in a single list, in alphabetical order.

Category Organizes plug-ins by process category (such as EQ, Dynamics, Effects, and Delay), with individual plug-ins listed in the category submenus. Plug-Ins that do not have a Category defined will appear in the Other Category folder. Plug-ins can appear in more than one category.

Plug-In categories include:

- EQ
- Dynamics
- Pitch Shift
- Reverb
- Delay
- Modulation
- Harmonic
- Noise Reduction
- Dither
- Sound Field
- Hardware
- Instrument
- Other
- Wrapped Plug-Ins
- Effects

Manufacturer Organizes plug-ins by their manufacturer (such as Digidesign, Eventide, Line 6, McDSP), with individual plug-ins listed in the manufacturer submenus. Plug-Ins that do not have a Manufacturer defined will appear in the "Other" manufacturer folder.

Most Digidesign Development Partner plug-ins will be sorted as Digidesign when view by manufacturer is enabled.

Category and Manufacturer Organizes plug-ins in two levels of menus. The top menus display plug-ins by process category (such as EQ, Dynamics, and Delay), with individual plug-ins listed in the category submenus. The bottom menus display plug-ins by their manufacturer (such as Digidesign, Eventide, Line 6, McDSP), with individual plug-ins listed in the manufacturer submenus.

To select how plug-ins are organized:

1 Open or create a session.

2 Choose Setup > Preferences, and click the Display tab.

3 Click the "Organize Plug-In Menus By" pop-up menu, and select one of the view options.
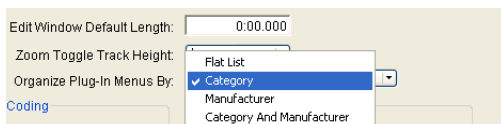
Choosing Category view for plug-in menus

4 Click Done.

The current Organize Plug-In Menus By setting is saved with Pro Tools preferences (it is not saved with the Pro Tools session file).

Plug-In Favorites

To find a plug-in faster, a plug-in can be designated as a *favorite*. Favorite plug-ins are shown at the top of the plug-in menu.

When a plug-in becomes a favorite, it is still shown in plug-in category and/or manufacturer sub-menus (if enabled) and its name appears in bold.
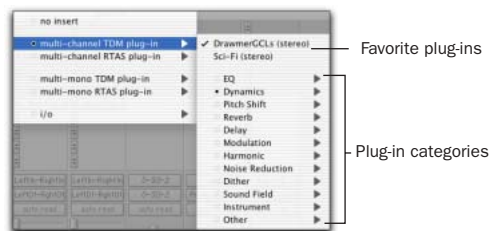
Menu display of favorite plug-Ins (Category view shown)

To designate a plug-in as a favorite:

■ Control-click (Windows) or Command-click (Macintosh) on a plug-in Insert button, and select a plug-in from the menu (or sub-menus, if enabled) to designate it as a favorite.

To change the status of a plug-in favorite:

1 Control-click (Windows) or Command-click (macintosh) on a plug-in Insert button, a nd select a plug-in favorite that you no longer want to designate as a favorite.

2 When selecting the plug-in favorite to change, you can select the plug-in from the list of favorites at the top of the plug-in menu, or in its plug-in category/manufacturer sub-menu (where the plug-in appears in bold).

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 or Edit window.

To show inserts in the Mix window:

■ Choose View > Mix Window > Inserts.

To show inserts in the Edit window:

■ Choose View > Edit Window > Inserts.

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.
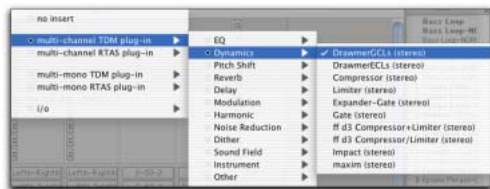
Inserting a plug-in with Plug-Ins by Category enabled
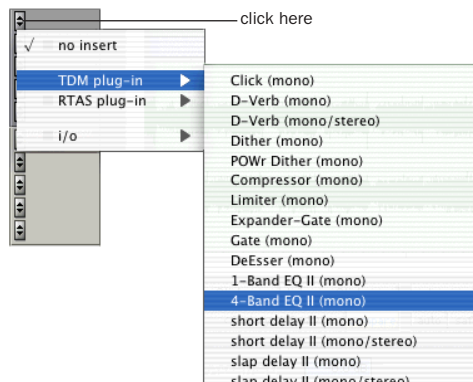
Inserting a plug-in (standard view)

Removing an Insert from a Track

To remove an insert from a track:

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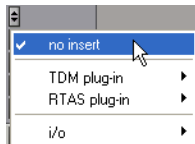
Removing a plug-in

Configuring Plug-Ins During Playback

Playback must be stopped before performing any of the following operations:

◆ Inserting or removing a plug-in while recording

◆ Dragging a plug-in to a different location

◆ Inserting or removing a plug-in that changes a track's format

◆ Removing a plug-in that contains automation data

◆ Enabling plug-in controls for automation

◆ Creating an external key side-chain input

◆ Inserting an RTAS plug-in on or removing an RTAS plug-in from an Auxiliary Input or Master Fader track

◆ Inserting an RTAS plug-in after a TDM plug-in on an audio track

◆ Removing an RTAS plug-in from an audio track if it comes after a TDM plug-in

◆ Inserting TDM plug-ins before an RTAS plug-in on an audio track

◆ Removing a TDM plug-in from an audio track if it comes before an RTAS plug-in

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.
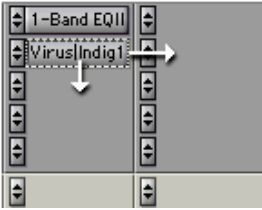
Moving a plug-in

💡 *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

You can 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.

Clip Indicators

Many plug-ins provide meters that let you see when the processed signal is clipping.

In addition to clipping displays that are a part of a plug-in's individual interface, some plug-ins also report the clipping in the plug-in header. The insert button in the Mix and Edit window also turns red if a plug-in has clipped.

Plug-In Clipping indicator in the plug-In header
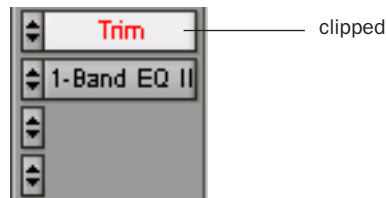
Plug-In Clipping indication in the Mix window

To clear plug-in clipping:

■ Click the clip LED in the plug-in header

To clear a clip indicator:

■ Click the clip LED in the plug-in header

You can also clear plug-in, send, and, and track window clip indicators simultaneously.

To clear all clip indicators, do one of the following:

■ Alt-click (Windows) or Option-click (Macintosh) any meter.

■ Press Alt-C (Windows) or Option-C (Macintosh).

■ Choose Track > Clear All Clip Indicators.

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.
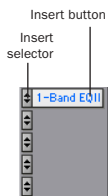
Insert button and selector on track
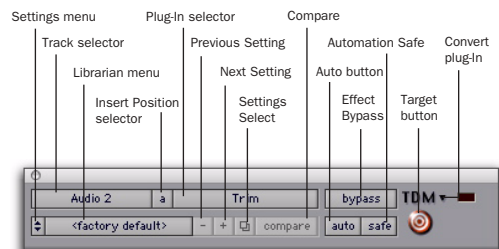
Plug-In window (mono Trim shown)

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

Insert Position Selector Accesses any insert on the current track.

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).

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

Convert Plug-In Selector 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.

**Clip LED** Lights red to indicate if any meter in a plug-in has clipped (not available on all plug-ins). The Plug-In Clip LED follows Pro Tools settings for clip indication (see the *Pro Tools Reference Guide* for more information).

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

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 24.

Next (+) and Previous (–) Setting Buttons Let you select the next or previous plug-in setting from the Librarian menu.

⚠ *You will lose the current settings if they are not saved before you use the Next and Previous Setting buttons. Always save your settings to the Librarian menu.*

Plug-Ins Settings Select Button Accesses the Plug-In Settings dialog, which lists the settings files for the current plug-in. From this list, you can select a new setting, or audition a series of settings.

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

Auto Lets you enable individual plug-in controls for automation recording. See "Automating Plug-Ins" on page 22.

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

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

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.

MIDI Node Display Shows MIDI node information for MIDI-enabled plug-ins. MIDI nodes are virtual connections from Pro Tools to software instruments and other MIDI-enabled plug-ins. MIDI nodes are useful for routing multiple MIDI tracks to different channels of a single virtual MIDI device. MIDI node numbers are also displayed in the plug-in header and in a track's MIDI Output selector.
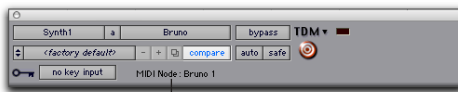
Midi Node display in a plug-in header (Bruno shown)

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.

Plug-In window (multi-mono 1-Band EQ shown)

Link Enable Buttons Let you selectively link or unlink 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 19.

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.

Phase Invert Button Inverts the phase polarity of the input signal on some plug-ins.
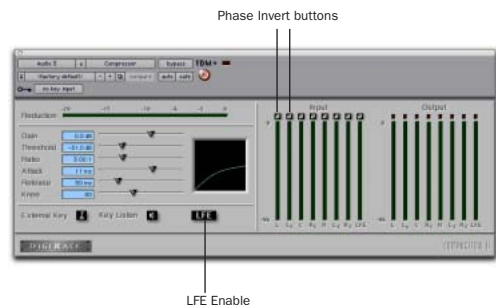
Plug-In window (multichannel Compressor shown)

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 on some plug-ins. To disable LFE processing, deselect this button. For more information on the LFE channel, refer to the *Pro Tools Reference Guide*.

Tempo Sync Enables compatible plug-ins to automatically synchronize to the session tempo, for rhythmic delay, echo, and similar effects. See "Tempo Sync" on page 21 for more information.

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:

■ 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 plug-in-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 (or Category sub-menu).
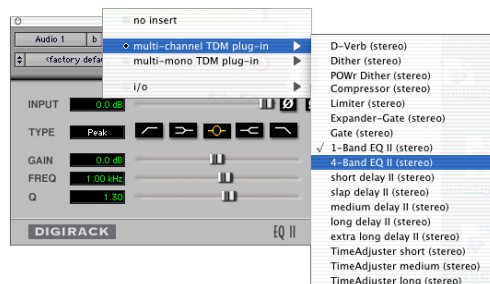
Choosing a plug-in from the Plug-In window

To choose a different track:

■ Click the Track selector and choose a track from the pop-up menu.
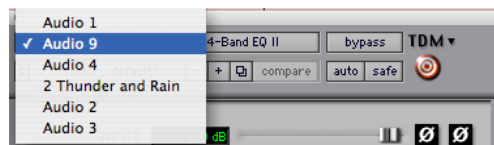
Choosing a track from the Plug-In window

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.*

Some channels bypassed (purple)

Bypassed (blue)   Unbypassed (gray)

Bypass states indicated by color

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.
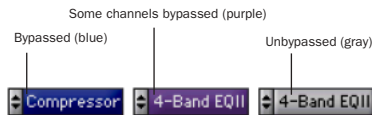
Master Link button
Link Enable buttons
Channel selector
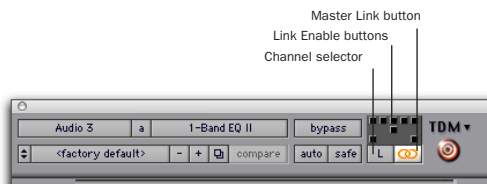
Channel selector and Link controls

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 Plug-In Parameters

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. Additonally, some plug-ins have switches that can be enabled by clicking on them.

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. Refer to "Editing Parameters Using a Mouse" on page 19 and "Editing Parameters Using a Computer Keyboard" on page 20.

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

Editing Parameters Using a Mouse

You can adjust rotary controls by dragging horizontally or vertically. Parameter values increase as you drag upward or to the right, and decrease as you drag downward or to the left.

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

Some controls have text boxes that display the current value of the parameter. You can edits the numeric value of a parameter with your computer keyboard.

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

To change control values with a computer keyboard:

1 Click on the text box corresponding to the control that you want to adjust.

2 Change the value.

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

– or –

- Type the desired value.

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.

3 Do one of the following:

- 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.

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

Enabling Switches

To enable a switch, click on the switch.

Using the Side-Chain Input

The *side-chain* input is the split-off signal used by a plug-in's detector to trigger dynamics processing, and is generally drawn internally from the input signal. However, some plug-ins let you switch between internal and external side-chain processing.

With external side-chain processing, a plug-in's detector is triggered by an external signal (such as a separate reference track or audio source) known as the *key input*.

A typical use for external side-chain processing 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.

Side-Chain Filters

Some plug-ins feature key high-pass and low-pass filters. These controls let you define a specific frequency range in the side-chain 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 frequency sounds (such as a hi-hat) or low frequency ones (such as a tom or a kick) trigger the effect.

Using a Key Input for External Side-Chain Processing

To use a key input for external side-chain processing:

1 Click the plug-in's Key Input selector and select the input or bus carrying the audio you want to use to trigger the plug-in.
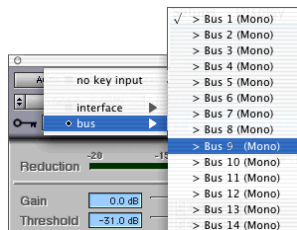
Selecting a key input

2 Click External Key to activate external side-chain processing.

3 Press the Spacebar to begin playback. The plug-in uses the input or bus that you chose as a key input to trigger its effect.

4 During playback, do any of the following to fine-tune side-chain triggering:

- To hear the audio source you have selected to control the side-chain, enable Side-Chain Listen or Key Listen (depending on the plug-in). When finished, disable Side-Chain Listen/Key Listen to resume normal plug-in monitoring.

- To filter the side-chain so that only specific frequencies trigger the plug-in, use the filtering controls (if available) to select the desired frequency range.

5 Adjust the plug-in's Threshold control (if available) and other controls as needed.

Tempo Sync

Pro Tools provides *Tempo Sync* to enhance MIDI Beat Clock support and overall tempo capabilities of plug-ins (such as ReWire client applications) that utilize MIDI Beat Clock.

Tempo Sync provides a direct connection between session tempo and plug-in parameters that support MIDI Beat Clock. This direct connection lets plug-in parameters such as delay, auto-pan, and other time-domain effects automatically synchronize to and follow changes in session tempo.

Tempo Sync simplifies MIDI Beat Clock configuration by making session tempo available to relevant plug-in parameters directly from within the plug-in window. For plug-ins that do not support Tempo Sync, the original MIDI Beat Clock window (Setup > MIDI > MIDI Beat Clock) remains available in Pro Tools.

Compatibility and Settings

When opening older sessions that included plug-ins that subscribe to MIDI Beat Clock, Tempo Sync will be automatically enabled. Any plug-ins that used automation for tempo change should have that automation suspended or deleted to avoid conflict with Tempo Sync.

⚠ *Tempo Sync is not available in plug-ins with which it would limit functionality. For example, for the Access Music Virus Indigo and IndigoV40 plug-in, use the Pro Tools MIDI Beat Clock window (Setup > MIDI > MIDI Beat Clock). See the Pro Tools Reference Guide for more information.*

To utilize Tempo Sync:

1 Insert a plug-in that supports Tempo Sync, such as the DigiRack Mod Delay II.

2 Click the Tempo Sync icon. The tempo shown will change to match the current session tempo.

Tempo Sync icon (Mod Delay II shown)

3 To set a rhythmic delay, click a note value (whole, half, quarter, eighth, or sixteenth), then do any of the following to further adjust the rhythm:

- To enable triplet delay timing, click the 3 button so that it is lit.
- To set a dotted delay value, click to enable the dotted icon.
- To apply some swing to the delay value, adjust the Groove slider.

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 to build up complex automation in stages.

💡 *To create automation for a stereo plug-in with separate controls for each channel, (such as Mod Delay), record the automation for one channel, then copy and paste it to the other channel.*

📖 *For complete instructions on Pro Tools automation, refer to the Pro Tools Reference Guide.*

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 Enable button in the Plug-In window.
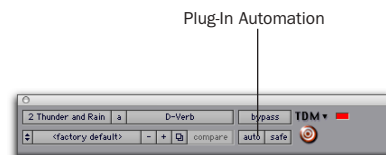
Accessing the Plug-In Automation dialog

– or –

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

💡 *You can also Control-Alt-Start-Right-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.*

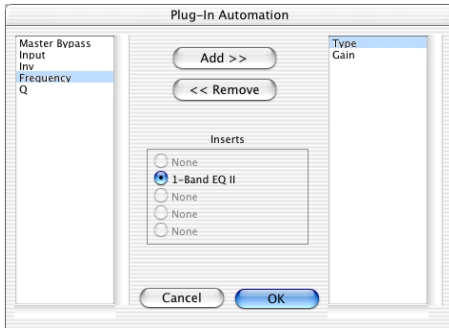**3** Choose the controls to automate and click Add. If there are multiple plug-ins on the same track, you can select from among these by clicking their buttons in the Inserts section of this dialog.

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.*
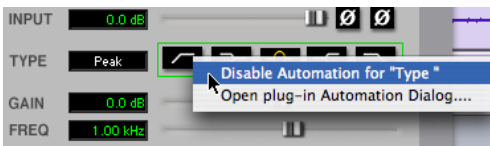
Shortcut for enabling plug-in automation

To automate a plug-in:

1 In the Automation Enable window, make sure plug-in automation is enabled.

Plug–In Automation Enable button
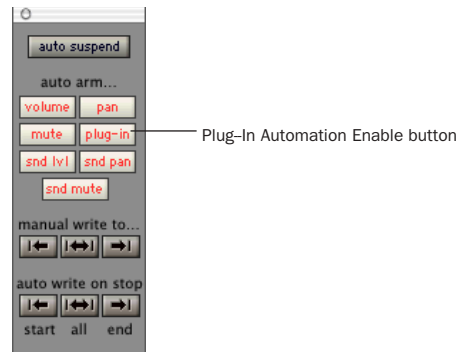
Automation Enable window

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.

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.
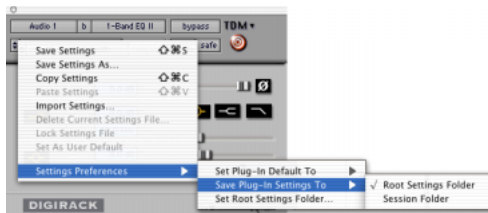

Librarian menu

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 (and in the Plug-In Settings dialog).

You can also use the Plug-In Settings dialog to select settings or audition them. See "Plug-In Settings Dialog" on page 27.

The Settings Menu

Use the Settings menu commands to 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 you specify with the Set As User Default command).

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 (see below).

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.

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 settings to the default Root 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 27.*

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 source plug-in.

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

3 Open the destination plug-in.

4 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 Settings Preferences > Set Plug-In Default To > User Setting.

Creating Subfolders for Settings

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

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.

Changing Settings with the Next (+) and Previous (–) Settings Buttons

The Next (+) and Previous (–) Settings buttons let you select the next or previous setting of the Librarian menu.

To change plug-in settings using the Next (+) and Previous (–) Settings buttons:

■ Press the Plus (+) or Minus (–) buttons to select the next or previous plug-in settings file. The next (or previous) settings file are enabled, and the Librarian menu changes to reflect the new 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.
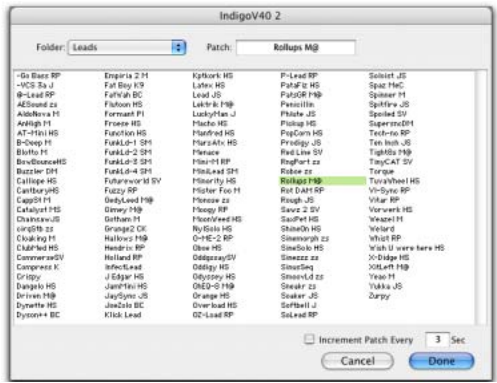
Plug-In Settings Dialog

The Plug-In Settings dialog lists the settings files for the current plug-in type. From this list, you can select a new setting, or audition a series of settings. Auditioning plug-in settings is especially useful for selecting patch names for virtual MIDI instrument plug-ins or trying out different effects types applied to your audio.

The Plug-In Settings dialog includes the following:

Folder This pop-up menu lets you switch between session settings located in root or session folders (and any available subfolders)

Patch Field This field displays the name of the active (highlighted) setting.

Increment Patch Every X Sec This option lets you audition contiguous plug-in settings on-the-fly by automatically scrolling through them for a specified number of seconds.

Plug-In Settings dialog

To change plug-in settings using the Plug-In Settings dialog:

1 Click the Settings Select button.
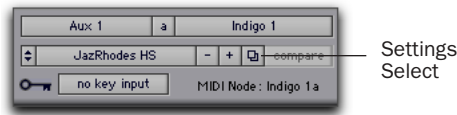
Plug-in header settings

▣ *To switch between settings folders, press Control (Windows) or Command (Macintosh) and use the Up/Down Arrow keys.*

2 From the dialog list, select the desired plug-in setting.

– or –

Use the Arrow keys to scroll through the available settings.

3 Click Done.

Auditioning Plug-in Settings

When the Plug-in Settings dialog is open, you can have Pro Tools automatically scroll through and audition the settings files saved in the plug-in's root settings folder or in the current session's Settings folder.

To audition plug-in settings:

1 Click the Settings Select button.

2 From the Settings dialog, click a setting—the starting point from which you will scroll through the settings files.

3 Enter a value for the number of seconds that will elapse between each program change, and press Enter.

4 Select the option for Increment Patch.

After the specified number of seconds, Pro Tools selects the next setting.

5 When you are finished auditioning settings, do one of the following:

• Click Done to stop incrementing patches, and to close the Plug-In Settings dialog.

– or –

• Select the Increment Patch option to stop incrementing patches and keep the Plug-In Settings dialog open.

Multiple Outputs for Instrument Plug-Ins

You can route the output of an instrument plug-in to the main track outputs and to individual track inputs on other tracks. This lets you route multiple samples or synth sounds through separate outputs for individual processing.

📖 *For specific information about using multiple plug-in outputs with third-party plug-ins, consult the documentation from the plug-in manufacturer.*

Main and Auxiliary Plug-In Outputs

When a plug-in that has both main and auxiliary outputs is inserted on a track, the plug-in's main outputs are routed through the track that the plug-in is inserted on, and the plug-in's auxiliary mono or stereo outputs are available in the track input pop-up menu on other tracks.

💡 *If none of the inserted plug-ins has auxiliary outputs, the plug-in category in the track input pop-up menu is grayed out.*

HTDM Plug-In Conversion to RTAS

When you use Pro Tools 7.0 or higher to open a session that was last saved in a version of Pro Tools prior to 7.0, most HTDM plug-ins will be automatically converted to their RTAS counterparts.

However, in rare cases as with some third-party plug-ins, these plug-ins will not automatically convert to RTAS and display as inactive. This means that the plug-in either has not been configured for automatic conversion or that no RTAS version is available.

In such cases, you must manually convert each individual HTDM plug-in to its RTAS counterpart by selecting it's RTAS version from the Insert selector or the Convert Plug-In pop-up menu. If an RTAS version is not available, the HTDM version will remain inactive.

⚠ *When HTDM plug-ins are converted to RTAS, any RTAS plug-ins that precede TDM plug-ins on audio tracks will be bypassed if you enable Record Enable or TrackInput monitoring on that track. See "Ordering RTAS and TDM Plug-Ins on an Audio Track" on page 30 for details.*

RTAS Plug-Ins on Auxiliary Input and Master Fader Tracks

(Pro Tools HD Only)

In Pro Tools HD, RTAS plug-ins can be inserted on Auxiliary Input and Master Fader tracks, or after TDM plug-ins on any kind of track.

💡 *While HTDM plug-ins have a fixed buffer size, latency caused by RTAS plug-ins is adjustable via the HW Buffer Size setting in the Playback Engine dialog. For more information, see the Pro Tools Reference Guide.*

⚠ *When you use a previous version of Pro Tools to open a session originally created in Pro Tools 7.0, any RTAS plug-ins inserted on Auxiliary Input or Master Fader tracks—or after TDM plug-ins on any kind of track—will become inactive.*

This also affects the following RTAS-related behavior in Pro Tools:

- Voice usage and total latency for RTAS plug-ins
- Limitations on inserting or removing plug-ins during playback

Voice Usage and Total Latency for RTAS Plug-Ins

With Pro Tools HD, the initial insert of an RTAS plug-in may cause additional latency, and will take up two additional voices per channel (1 voice for input and 1 voice for output) in the following conditions:

- When inserted on an Auxiliary Input or Master Fader track
- When inserted on an Instrument track that does not contain an instrument plug-in
- When inserted after a TDM plug-in on any kind of track

For example, the initial insert of an RTAS plug-in on a mono Auxiliary Input track uses two voices (one channel with two voices), while the initial insert of that plug-in on a *stereo* Auxiliary Input track uses four voices (two channels with two voices each).

Subsequent RTAS plug-ins on the same track do not take up additional voices unless a TDM plug-in is inserted between other RTAS plug-ins.

⚠ *Avoid inserting TDM plug-ins between RTAS plug-ins on any kind of track—this causes unnecessary voice usage and may cause additional latency.*

Furthermore, one additional voice is used for each occurrence of any of the following conditions when using voices for RTAS plug-ins on a track:

- When you use an external side-chain for an RTAS plug-in on that track
- When you select multiple outputs for that track (one voice used for each output)
- When you select an AFL/PFL Path output in the Output tab of the I/O Setup dialog (one voice used for each channel)

Ordering RTAS and TDM Plug-Ins on an Audio Track

With Pro Tools HD, when RTAS and TDM plug-ins are combined on an audio track, the order in which they are inserted has different results:

RTAS Plug-Ins Grouped Before TDM Plug-Ins No additional voices will be used, and no processing latency will occur. RTAS plug-ins will be bypassed when Record Enable or TrackInput monitoring is enabled for that track.

TDM Plug-Ins Grouped Before RTAS Plug-Ins

Each initial insert of an RTAS plug-in after a TDM plug-in will cause processing latency, and will use voices as described in "Voice Usage and Total Latency for RTAS Plug-Ins" on page 29. RTAS plug-ins will stay active while Record Enable or TrackInput monitoring is enabled for that track.

⚠ *As a rule, always group plug-ins of the same type together, especially if inserting RTAS after TDM plug-ins. Otherwise, additional voices may be unnecessarily used, and unnecessary latency may result.*

Ordering Recommendations

When combining RTAS and TDM plug-ins on an audio track, use one of the following two strategies based on your recording needs:

• To ensure that RTAS plug-ins stay active when you record enable a track or use TrackInput monitoring, group all TDM plug-ins before RTAS plug-ins.

– or –

• To conserve voices and minimize processing latency, group all RTAS plug-ins before TDM plug-ins.

Limitations for Inserting or Removing Plug-Ins During Playback

When playing back tracks containing RTAS plug-ins, the following limitations on inserting or removing plug-ins apply:

• RTAS and TDM plug-ins cannot be inserted on or removed from Auxiliary Input, Master Fader, or Instrument tracks.

• An RTAS plug-in cannot be inserted after a TDM plug-in on an audio track, or removed from an audio track if it comes after a TDM plug-in.

• A TDM plug-in cannot be inserted before an RTAS plug-in on an audio track, or removed from an audio track if it comes before an RTAS plug-in.

chapter 4

Working with AudioSuite Plug-Ins

AudioSuite plug-ins differ from TDM and RTAS plug-ins in that they are used to process and modify audio files on disk rather than non-destructively in real time. Depending on how you configure a non-real-time AudioSuite plug-in, it will either alter the original source audio file or create an entirely new audio source file.

The AudioSuite Menu

AudioSuite plug-ins are accessed through the AudioSuite menu.

Plug-ins can be organized in the AudioSuite menu in Pro Tools by Flat List (with plug-ins in alphabetical order), Category, Manufacturer, or Category *and* Manufacturer.

📖 *For more information, see "Organizing Plug-Ins" on page 12.*

AudioSuite Plug-Ins

1-Band EQ III Processes audio with selectable EQ filters in a 1-band module.

2–4 Band EQ III Processes audio with two, three, or four selectable EQ filters.

7-Band EQ III Processes audio with up to seven selectable EQ filters.

Click Creates an audio click as a tempo reference for performing and recording.

Compressor Processes audio with compression.

Chorus Processes audio with chorusing.

Flanger Processes audio with flanging.

Limiter Processes audio with limiting.

Gate Processes audio with gating.

Expander Gate Processes audio with expander-gating.

De-esser Processes audio with de-essing to re-move sibilance.

Duplicate Creates a new, continuous source audio file (and region) from the selection.

Delay Adds up to 10.9 seconds of delay.

Multi-Tap Delay Adds up to four independently-controllable delays to the original audio signal.

Normalize Uniformly adjusts all levels in a region or regions to a user-definable level, using the loudest peak in the audio file as the reference.

Ping-Pong Delay Adds a stereo, ping-pong delay to the audio signal by feeding back the original signal to its opposite channel.

Gain Adjusts gain (volume) change, in decibels or percentage before clip.

Invert Inverts phase (polarity).

Pitch Shift Changes an audio file's pitch with or without changing its duration.

**Reverse** Rewrites selected audio in reverse.

DC Offset Removal Recognizes and removes DC offset from audio files.

Signal Generator Generates audio tones for equipment calibration purposes.

Time Compression/Expansion Changes an audio file's duration with or without changing its pitch.

The AudioSuite Window

When you choose a plug-in from the AudioSuite menu, the AudioSuite window appears. From this floating window you can access and edit the controls for any non-real-time AudioSuite plug-in.
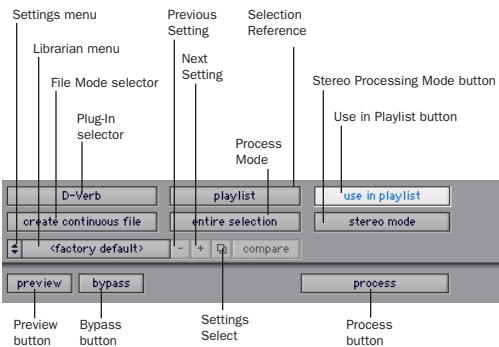
AudioSuite window

Plug-In Selector

From this menu you can select any AudioSuite plug-in installed in your Plug-Ins folder. (The Plug-In selector follows the Pro Tools preference for Organize Plug-Ins Menu By.)
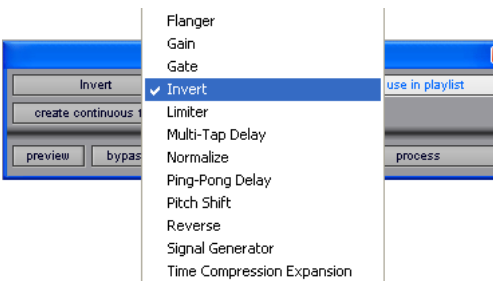

Plug-In selector menu (flat list, non-category view)

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.

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.
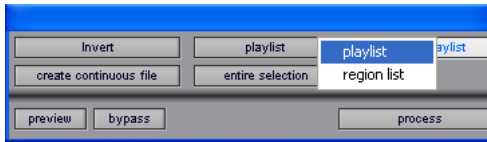

Selection Reference pop-up menu (Invert plug-in shown)

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.

⚠ *If the Use In Playlist option is enabled in addition to Region List being selected in the Selection Reference pop-up menu, the processing will be applied to the selected region in both the playlist and the Regions List.*

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.
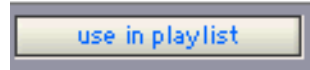
Use in Playlist button

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.
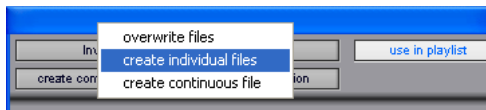
File Mode pop-up menu (Invert plug-in shown)

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 earlier.)

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.
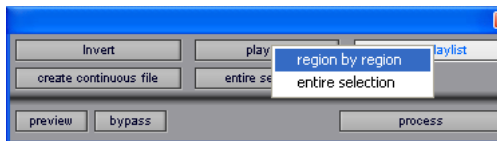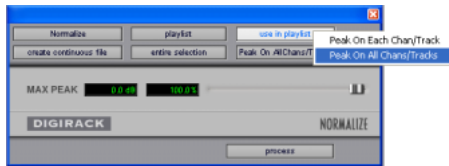

Process Mode selector (Invert plug-in shown)

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 (Normalize plug-in shown)

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.

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 menu 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 "AudioSuite Processing Preferences" on page 37.

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 36.

Multichannel Processing

Most DigiRack AudioSuite plug-ins can process up to 48 channels of audio simultaneously.

⚠ *When processing multiple regions of different formats (mono and multichannel surround, for example), all channels will be processed according to the channel format of the AudioSuite plug-in. Mono AudioSuite plug-ins can process stereo regions and stereo AudioSuite plug-ins can process mono regions.*

Undoing AudioSuite Processing

If you have processed an audio selection nondestructively, 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 configured for destructive editing since the process has already overwritten the source audio 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 standard Pro Tools file and region numbering.

◆ If a plug-in's File Mode pop-up is set to Overwrite, 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 appended 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 Settings and Librarian pop-up menus that appear in some AudioSuite plug-ins provide a means of saving, loading, copying, pasting and organizing custom plug-in settings files. See "Using the Librarian" on page 24.

Analyze Button

This button analyzes a selection without actually processing it. On the DigiRack Gain plug-in, you can use the Analyze feature to determine the maximum peak level on a track at a specific gain value before you process the audio.

Key Input Selector

Selects a track/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. Side-chain input is monophonic. See "Using the Side-Chain Input" on page 20.

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

Channel Mode Button

The Channel Mode button lets you process contiguous channels within a selected region as though they were mono or stereo tracks (and in some cases greater-than-stereo tracks), regardless of the type of tracks they occupy.

Mono Mode Processes each contiguous channel within a selected region as a separate mono track. For example, a region comprised of six mono tracks would be processed as six mono tracks.

Stereo Mode Processes each contiguous pair of channels within a selected region as a stereo track. When applied to an odd number of channels, the plug-in processes the last odd, unpaired channel 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.

For example, a region comprised of two mono tracks would be processed as a stereo track, with the first mono track processed as the left channel and the second mono track processed as the right channel.

Multi Mode Processes each set of three or more contiguous channels within a selected region as a multichannel greater-than-stereo track, with the number of selected channels determining the type of track. For example, five contiguously-selected channels would be processed as a 5.0 track, while seven contiguously-selected channels will be processed as a 7.1 track.

⚠ *Multi mode is available only for certain plug-ins, such as Dynamics III.*

AudioSuite Processing Preferences

Before you begin using AudioSuite plug-ins, use the Preferences dialog (Setup > 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.
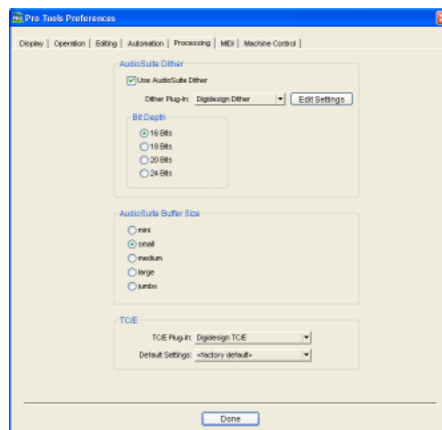
Processing Preferences page

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 menu. 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. With the Dither Plug-In pop-up menu you can choose the dither algorithm used for dithering higher-bit source files during AudioSuite processing.

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

📄 *See "Noise Shaping" on page 57 for more information.*

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.

📄 *For an explanation of dither, See "Dither" on page 56.*

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 Setup > 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 Selects the Time Compression & Expansion plug-in and default settings used when editing with the Time Trimmer tool. If you own a compatible third-party time compression and expansion plug-in, you have the option of using it for Pro Tools time compression and expansion processes by selecting it here.

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

Using AudioSuite Plug-Ins

AudioSuite plug-ins can be applied to whole re-
gions, 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 in-
cludes partial regions, the regions will automat-
ically be split into two or more regions when
they are processed. Processing will occur only
on the selection, leaving other regions un-
changed.

⚠ *Processing a locked region will cause the re-
gion to become unlocked, and processing a
muted region will cause it to become un-
muted. For more information on locked and
muted regions, refer to the Pro Tools Refer-
ence 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 (unless it is a Unity
network).

Selecting Tracks for AudioSuite Processing

Because AudioSuite processes are performed on
the specific regions that you select, it is impor-
tant 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 di-
alog. Refer to 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, it is important that you make
a selection that is *longer* than the original source
material so that the plug-in can write it to the
audio file. If you select only the original mate-
rial, without leaving additional space at the end,
any 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 se-
lect more blank space than you need, you can
remove it using the Trimmer 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.

For more information, see "Channel Mode Button" on page 37.

DigiRack Real-Time TDM and RTAS Plug-Ins

EQ III

The EQ III Plug-In provides a high-quality 7 Band, 2–4 Band, or 1 Band EQ for adjusting the frequency spectrum of audio material.

EQ III is available in the following formats:

- 7 Band: TDM, RTAS, AudioSuite
- 2–4 Band: TDM and RTAS only
- 1 Band: TDM, RTAS, AudioSuite

EQ III supports all Pro Tools session sample rates: 192 kHz, 176.4 kHz, 96 kHz, 88.2 kHz, 48 kHz, and 44.1 kHz. EQ III operates as a mono or multi-mono plug-in only. Stereo and multi-channel tracks are supported through multi-mono operation.

EQ III can be operated from the following control surfaces:

- Digidesign D-Control
- Digidesign ProControl
- Digidesign Control|24
- Digidesign 002
- Digidesign Command|8
- Mackie HUI-compatible controllers

EQ III has a Frequency Graph display that shows the response curve for the current EQ settings on a two-dimensional graph of frequency and gain. The frequency graph display also lets you modify frequency, gain and Q settings for individual EQ bands by dragging their corresponding points in the graph.

By choosing from the 7 Band, 2–4 Band, or 1 Band versions of the EQ III plug-in, you can use only the number of EQ bands you need for each track, conserving DSP capacity on Pro Tools|HD systems.

EQ III Configurations

The EQ III Plug-In appears as three separate choices in the plug-in insert pop-up menu and in the AudioSuite menu:

7 Band ("7-Band EQ 3")

2–4 Band ("4-Band EQ 3")

1 Band ("1-Band EQ 3")

7 Band EQ and 2–4 Band EQ

The 7 Band EQ is available in TDM, RTAS, and AudioSuite formats. The 2–4 Band EQ is available in TDM and RTAS formats only.

The 7 Band EQ and the 2–4 Band EQ share the same window and identical controls, but with the 2–4 Band EQ, a limited number of the seven available bands can be active at the same time.
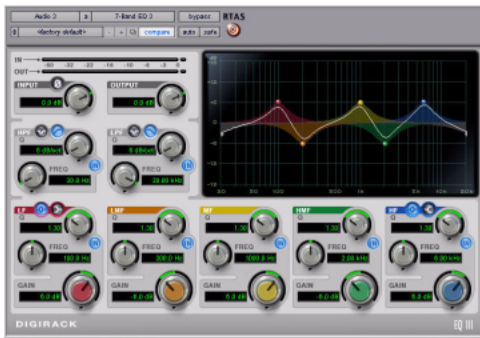
7 Band EQ and 2–4 Band EQ window

1 Band EQ

The 1 Band EQ is available in TDM, RTAS, and AudioSuite formats.

The 1 Band EQ has its own window, with six selectable filter types.

1 Band EQ window

EQ III Controls

Adjusting EQ III Controls

You can adjust the EQ III plug-in controls by any of the following methods:

Dragging Plug-In Controls

The rotary controls on the EQ III Plug-In can be adjusted by dragging over them horizontally or vertically. Dragging up or to the right increments the control. Dragging down or to the left decrements the control.

Dragging a plug-in control

Typing Control Values

You can enter control values directly by clicking in the corresponding text box, typing a value, and pressing Enter (Windows) or Return (Macintosh).

Typing a control value

Inverting Filter Gain

(Peak EQ Bands Only)

Gain values can be inverted on any Peak EQ band by Shift-clicking its control dot in the Frequency Graph display, or its Gain knob in the plug-in window. This changes a gain boost to a cut (+9 to –9) or a gain cut to a boost (–9 to +9). Gain values cannot be inverted on Notch, High-Pass, Low-Pass, or shelving bands.

Dragging in the Frequency Graph Display

You can adjust the following by dragging the control points directly in the Frequency Graph display:

Frequency Dragging a control point to the right increases the Frequency setting. Dragging a control point to the left decreases the Frequency setting.

Gain Dragging a control point up increases the Gain setting. Dragging a control point down decreases the Gain setting.

Q Start-dragging (Windows) or Control-dragging (Macintosh) a control point up increases the Q setting. Start-dragging (Windows) or Control-dragging (Macintosh) a control point down decreases the Q setting.
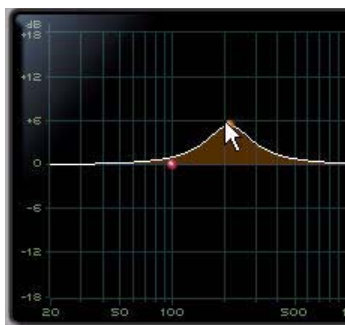
Dragging a control point in the Frequency Graph display

Adjusting Controls with Fine Resolution

Controls and control points can be adjusted with fine resolution by holding the Control key (Windows) or the Command key (Macintosh) while adjusting the control.

Resetting Controls to Default Values

You can reset any on-screen control to its default value by Alt-clicking (Windows) or Option-clicking (Mac OS) directly on the control or on its corresponding text box.

Using Band-Pass Mode

You can temporarily set any EQ III control to Band-Pass monitoring mode. Band-Pass mode cuts monitoring frequencies above and below the Frequency setting, leaving a narrow band of mid-range frequencies. It is especially useful for adjusting limited bandwidth in order to solo and fine-tune each individual filter before reverting the control to notch filter or peaking filter type operations.

💡 *Band-Pass mode does not affect EQ III Gain controls.*

To switch an EQ III control to Band-Pass mode:

• Hold Start+Shift (Windows) or Control+Shift (Macintosh), and drag any rotary control or control point horizontally or vertically.
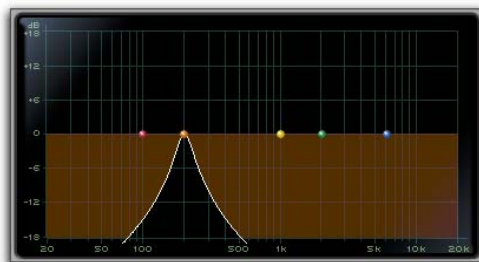
EQ III interactive graph displaying Band-Pass mode

When monitoring in Band-Pass mode, the Frequency and Q controls function differently.

Frequency Sets the frequency above and below which other frequencies are cut off, leaving a narrow band of mid-range frequencies.

Q Sets the width of the narrow band of mid-range frequencies centered around the Frequency setting.

**To switch an EQ III control out of Band-Pass mode:**

• Release Start+Shift (Windows) or Control+Shift (Macintosh).

Using a Control Surface

EQ III can be controlled from any supported control surface, including Digidesign's D-Control, D-Command, ProControl, Control|24, Digi 002, or Command|8. Refer to the guide that came with the control surface for details.

I/O Controls

The following Input and Output controls are found on all EQ III configurations, except where noted otherwise.
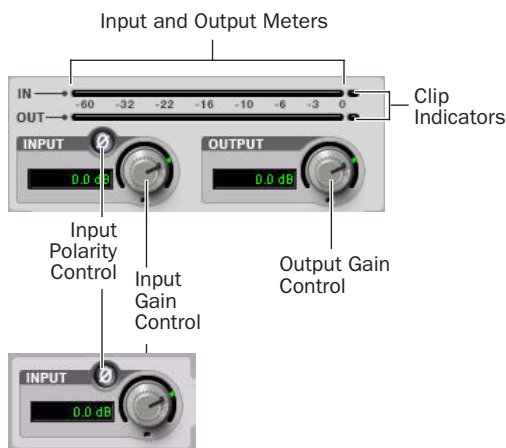
I/O controls and meters for 7 Band EQ and 2–4 Band EQ (top) and 1 Band EQ (bottom)

Input Gain Control

The Input Gain control sets the input gain of the plug-in before EQ processing, letting you make up gain or prevent clipping at the plug-in input stage.

Output Gain Control

(7 Band EQ and 2–4 Band EQ Only)

The Output Gain control sets the output gain after EQ processing, letting you make up gain or prevent clipping on the channel where the plug-in is being used.

Input Polarity Control

The Input Polarity button inverts the polarity of the input signal, to help compensate for phase anomalies occurring in multi-microphone environments, or because of mis-wired balanced connections.

Input and Output Meters

(7 Band EQ and 2–4 Band EQ Only)

The plasma-style Input and Output meters show peak signal levels before and after EQ processing, and indicate them as follows:

Green Indicates nominal levels

Yellow Indicates pre-clipping levels, starting at –6 dB below full scale

Red Indicates full scale levels (clipping)

The clip indicators at the far right of each meter indicate clipping at the input or output stage of the plug-in. Clip indicators can be cleared by clicking the indicator.

EQ Band Controls

The individual EQ bands on each EQ III configuration have some combination of the following controls, as noted below.
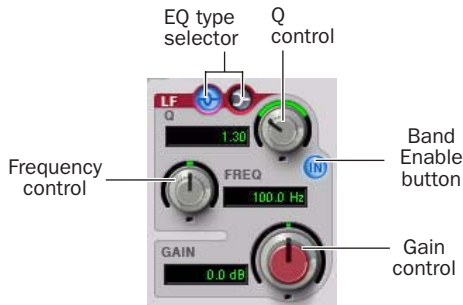
EQ Band controls (7 Band EQ Low-Mid band shown)

EQ Type Selector

On the 1 Band EQ, the EQ Type selector lets you choose any one of six available filter types: High-Pass, Notch, High-Shelf, Low-Shelf, Peak, and Low-Pass.

On the 7 Band EQ and the 2–4 Band EQ, the HPF, LPF, LF, and HF sections have EQ Type selectors to toggle between the two available filter types in each section.

Band Enable Button

(7 Band EQ and 2–4 Band EQ Only)

The Band Enable button on each EQ band toggles the corresponding band in and out of circuit. When a Band Enable button is highlighted, the band is in circuit. When a Band Enable button is dark gray, the band is bypassed and available for activation. On the 2–4 Band EQ, when a Band Enable button is light gray, the band is bypassed and unavailable.

Band Gain Control

Each Peak and Shelf EQ band has a Gain control for boosting or cutting the corresponding frequencies. Gain controls are not used on High-Pass, Low-Pass, or Notch filters.

Frequency Control

Each EQ band has a Frequency control that sets the center frequency (Peak, Shelf and Notch EQs) or the cutoff frequency (High-Pass and Low-Pass filters) for that band.

Q Control

Peak and Notch On Peak and Notch bands, the Q control changes the width of the EQ band. Higher Q values represent narrower bandwidths. Lower Q values represent wider bandwidths.

Shelf On Shelf bands, the Q control changes the Q of the shelving filter. Higher Q values represent steeper shelving curves. Lower Q values represent broader shelving curves.

Band Pass On High-Pass and Low-Pass bands, the Q control lets you select from any of the following Slope values: 6 dB, 12 dB, 18 dB, or 24 dB per octave.

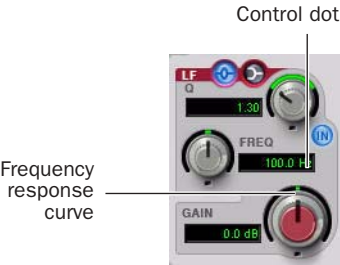## Frequency Graph Display

(7 Band EQ and 2–4 Band EQ Only)

The Frequency Graph display in the 7 Band EQ and the 2–4 Band EQ shows a color-coded control dot that corresponds to the color of the Gain control for each band. The filter shape of each band is similarly color-coded. The white frequency response curve shows the contribution of each of the enabled filters to the overall EQ curve.
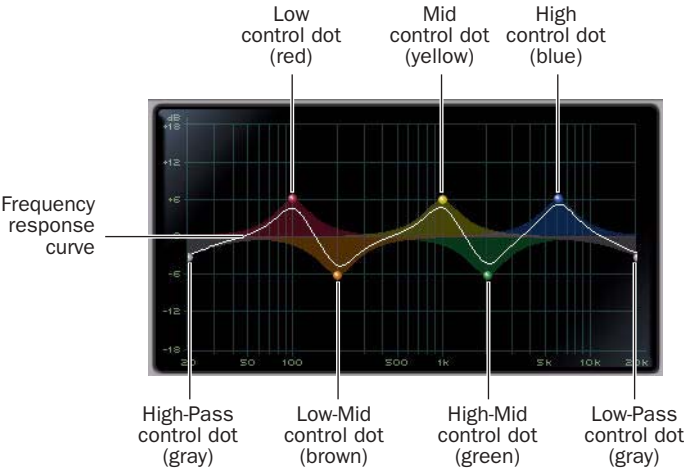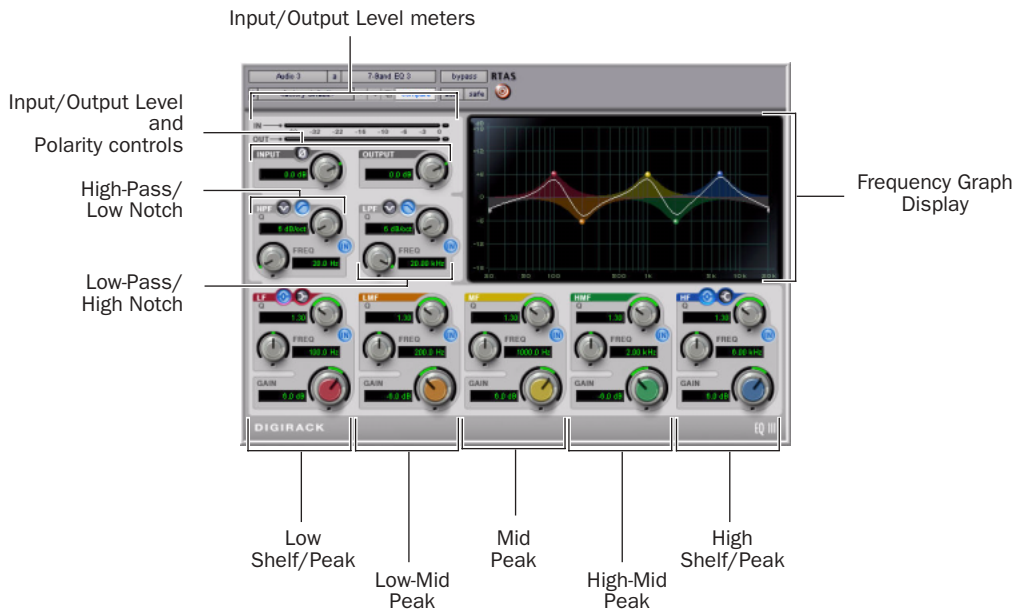

Frequency Graph display for the 7 Band EQ

1 Band EQ

The Frequency Graph display in the 1 Band EQ shows a control dot that indicates the center frequency (Peak, Shelf and Notch Filters) or the cutoff frequency (High-Pass and Low-Pass filters) for the currently selected filter type.

7 Band EQ

Input/Output Level meters

Input/Output Level and Polarity controls

High-Pass/ Low Notch

Low-Pass/ High Notch

Frequency Graph Display

Low Shelf/Peak

Low-Mid Peak

Mid Peak

High-Mid Peak

High Shelf/Peak

7 Band EQ and 2–4 Band EQ window

The 7 Band EQ has the following available bands: High-Pass/Low Notch, Low-Pass/High Notch, Low Shelf/Low Peak, Low-Mid Peak, Mid Peak, High-Mid Peak, and High Shelf/High Peak.

All seven bands are available for simultaneous use. In the factory default setting, the High-Pass/Low Notch and Low-Pass/High Notch bands are out of circuit, the Low Shelf and High Shelf bands are selected and in circuit, and the Low-Mid Peak, Mid Peak, High-Mid Peak bands are in circuit.

High-Pass/Low Notch

The High-Pass/Notch band is switchable between High-Pass Filter and Notch EQ functions. By default, this band is set to High-Pass Filter.

High-Pass Filter Attenuates all frequencies below the Frequency setting at the selected slope while letting all frequencies above pass through.

Low-Notch EQ Attenuates a narrow band of frequencies centered around the Frequency setting. The width of the attenuated band is determined by the Q setting.
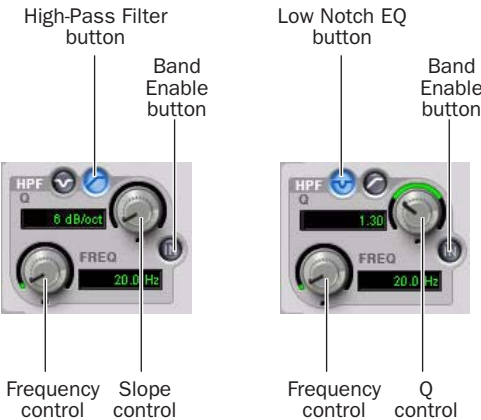
High-Pass filter (left) and Low Notch EQ (right)

The High Pass and Low Notch controls and their corresponding graph elements are displayed on-screen in gray.

High-Pass Filter and Low Notch EQ control values

Control	Value
Frequency Range	20 Hz to 8 kHz
Frequency Default	20 Hz
HPF Slope Values	6, 12, 18, or 24 dB/oct
Low Notch Q Range	0.1 to 10.0
Low Notch Q Default	1.0

Low-Pass/High Notch

The Low-Pass/Notch band is switchable between Low-Pass Filter and Notch EQ functions. By default, this band is set to Low-Pass Filter.

Low-Pass Filter Attenuates all frequencies above the Frequency setting at the selected slope while letting all frequencies below pass through.

High-Notch EQ Attenuates a narrow band of frequencies centered around the Frequency setting. The width of the attenuated band is determined by the Q setting.
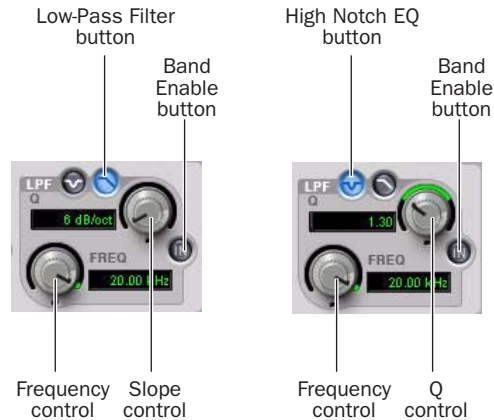
Low-Pass filter (left) and High Notch EQ (right)

The Low Pass and High Notch controls and their corresponding graph elements are displayed on-screen in gray.

Low-Pass Filter and High Notch EQ control values

Control	Value
Frequency Range	120 Hz to 20 kHz
Frequency Default	20 kHz
LPF Slope Values	6, 12, 18, or 24 dB/oct
High Notch Q Range	0.1 to 10.0
High Notch Q Default	1.0

Low Shelf/Low Peak

The Low Shelf/Peak band is switchable between Low Shelf EQ and Low Peak EQ functions. By default, this band is set to Low Shelf.
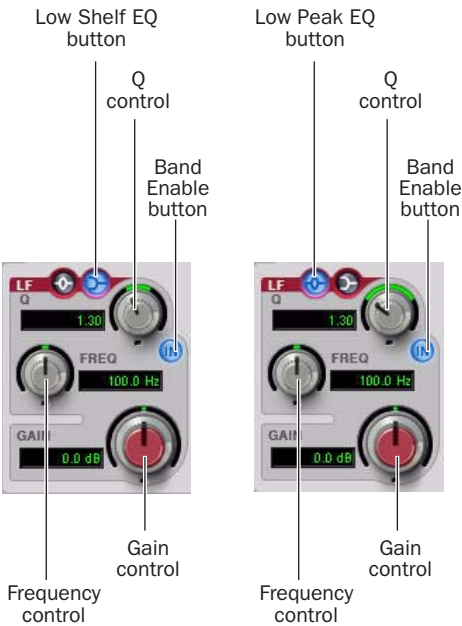
Low-Shelf EQ Boosts or cuts frequencies at and below the Frequency setting. The amount of boost or cut is determined by the Gain setting. The Q setting determines the shape of the shelving curve.

Low Peak EQ Boosts or cuts a band of frequencies centered around the Frequency setting. The width of the affected band is determined by the Q setting.

The Low Shelf and Low Peak Gain controls and their corresponding graph elements are displayed on-screen in red.

Low Shelf EQ and Low Peak EQ control values

Control	Value
Frequency Range	20 Hz to 500 Hz
Frequency Default	100 Hz
Low Shelf Q Range	0.1 to 2.0
Low Peak Q Range	0.1 to 10.0
Q Default	1.0
Low Shelf Gain Range	–12 dB to +12 dB
Low Peak Gain Range	–18 dB to +18 dB

Low Shelf EQ (left) and Low Peak EQ (right)

Low-Mid Peak

The Low-Mid Peak band boosts or cuts frequencies centered around the Frequency setting. The width of the band is determined by the Q setting.
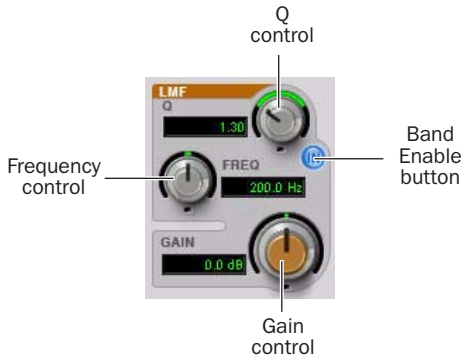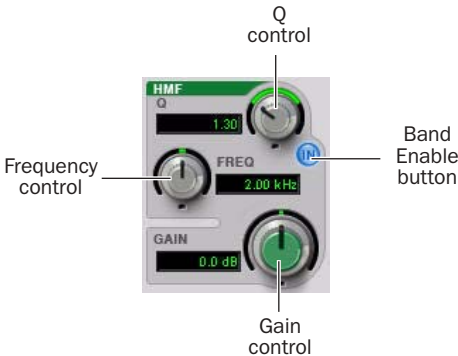

Low-Mid Peak EQ

The Low-Mid Gain control and its corresponding graph elements are displayed on-screen in brown.
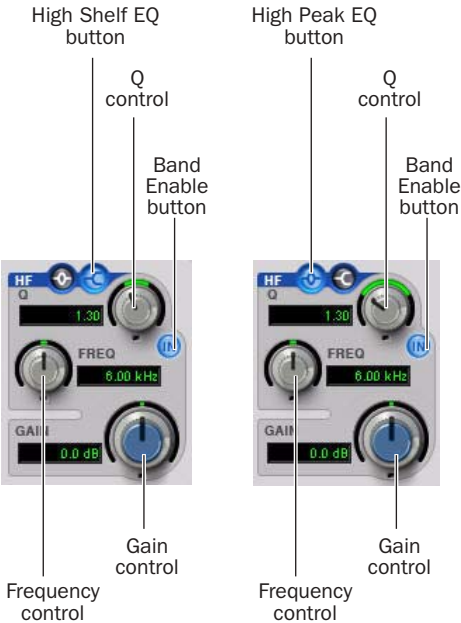
Low-Mid Peak EQ control values

Control	Value
Frequency Range	40 Hz to 1 kHz
Frequency Default	200 Hz
Low-Mid Peak Q Range	0.1 to 10.0
Low-Mid Peak Q Default	1.0
Low-Mid Peak Gain Range	–18 dB to +18 dB

Mid Peak

The Mid Peak band boosts or cuts frequencies centered around the Frequency setting. The width of the band is determined by the Q setting.
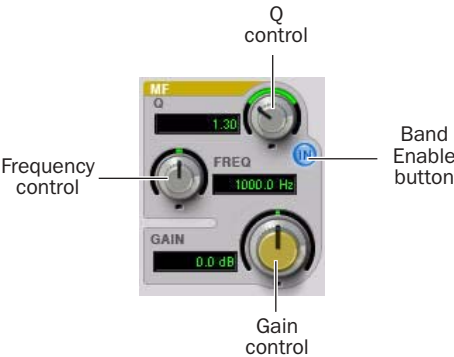
Mid Peak EQ

The Mid Gain control and its corresponding graph elements are displayed on-screen in yellow.

Mid Peak EQ control values

Control	Value
Frequency Range	125 Hz to 8 kHz
Frequency Default	1 kHz
Mid Peak Q Range	0.1 to 10.0
Mid Peak Q Default	1.0
Mid Peak Gain Range	–18 dB to +18 dB

High-Mid Peak

The High-Mid Peak band boosts or cuts frequencies centered around the Frequency setting. The width of the band is determined by the Q setting.

High-Mid Peak EQ

The High-Mid Gain control and its corresponding graph elements are displayed on-screen in green.

High-Mid Peak EQ control values

Control	Value
Frequency Range	200 Hz to 18 kHz
Frequency Default	2 kHz
Mid Peak Q Range	0.1 to 10.0
Mid Peak Q Default	1.0
Mid Peak Gain Range	−18 dB to +18 dB

High Shelf/High Peak

The High Shelf/Peak band is switchable between High Shelf EQ and High Peak EQ functions. By default, this band is set to High Shelf.

High-Shelf EQ Boosts or cuts frequencies at and above the Frequency setting. The amount of boost or cut is determined by the Gain setting. The Q setting determines the shape of the shelving curve.

High Peak EQ Boosts or cuts a band of frequencies centered around the Frequency setting. The width of the affected band is determined by the Q setting.

High Shelf EQ (left) and High Peak EQ (right)

The High Shelf and High Peak Gain controls and their corresponding graph elements are displayed on-screen in blue.

High Shelf EQ and High Peak EQ control values

Control	Value
Frequency Range	1.8 kHz to 20 kHz
Frequency Default	6 kHz
High Shelf Q Range	0.1 to 2.0
High Peak Q Range	0.1 to 10.0
Q Default	1.0
High Shelf Gain Range	−12 dB to +12 dB
High Peak Gain Range	−18 dB to +18 dB

2–4 Band EQ

The 2–4 Band EQ uses the same plug-in window as the 7 Band EQ, but on the 2–4 Band EQ, but a limited number of the seven available bands can be active at the same time.

In the factory default setting, the High-Pass/Low Notch, Low-Pass/High Notch and Mid Peak bands are out of circuit, the Low Shelf and High Shelf bands are selected and in circuit, and the Low-Mid Peak and High-Mid Peak bands are in circuit.

💡 *For Pro Tools HD, using a 2–4 Band EQ instead of a 7 Band EQ saves DSP resources.*

Changing from 2–4 Band to 7 Band

After switching from a 2–4 band EQ to a 7 Band EQ, or importing settings from a 2–4 Band EQ, all control settings from the 2–4 Band EQ are preserved, and the bands in the 7 Band EQ inherit their enabled or bypassed state from the 2–4 Band plug-in.

Additional EQ bands can then be enabled to add them to the settings inherited from the 2–4 Band plug-in.

Filter Usage with 2–4 Band EQs

With a 2–4 Band EQ, a maximum of four filters may be active simultaneously, with each of the five Peak bands (Low Shelf/Peak, Low-Mid Peak, Mid-Peak, High-Mid Peak and High Shelf/Peak) counting as one filter. Each of the Band-pass and Notch filters (High-Pass, Low Notch, Low-Pass and High-Notch) counts as *two* filters.

When any combination of these filter types uses the four-filter maximum on the 2–4 Band EQ, the remaining bands become unavailable. This is indicated by the Band Enable buttons turning light gray. When filters become available again, the Band Enable button on inactive bands turns dark gray.

### Switching Between the 2–4 Band EQ and 7 Band EQ

When you switch an existing EQ III plug-in between the 2–4 Band and 7 Band versions, or when you import settings between versions, the change is subject to the following conditions:

Changing from 7 Band to 2–4 Band

After switching from a 7 band EQ to a 2–4 Band EQ, or importing settings from a 7 Band EQ, all control settings from the 7 Band EQ are preserved in the 2–4 Band EQ, but *all* bands are placed in a bypassed state.

Bands can then be enabled manually, up to the 2–4 Band EQ four-filter limit.

1 Band EQ
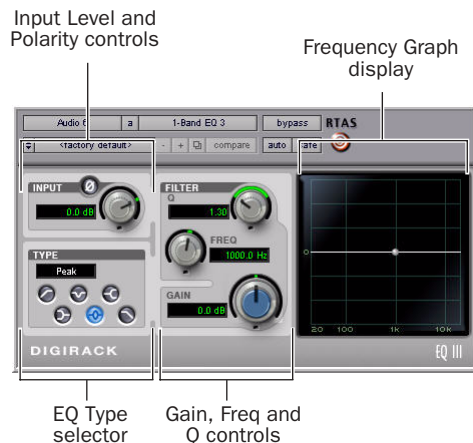
1 Band EQ window

The 1 Band EQ may be set to any one of six EQ types: High-Pass, Notch, High-Shelf, Low-Shelf, Peak, and Low-Pass, by clicking the corresponding icon in the EQ Type selector.

Band Controls

The individual EQ types have some combination of the following controls, as noted below.

1 Band EQ control values

Control	Value
Frequency Range (All)	20 Hz to 20 kHz
Frequency Default (All)	1 kHz
Q Range (Low/High Shelf)	0.1 to 2.0
Q Range (Peak/Notch)	0.1 to 10.0
Q Default (All)	1.0
Gain Range (Low/High Shelf)	–12 dB to +12 dB
High Peak Gain Range	–18 dB to +18 dB

EQ Types

High-Pass Filter

The High-Pass filter attenuates all frequencies below the Frequency setting at the selected rate (6 dB, 12 dB, 18 dB, or 24 dB per octave) while letting all frequencies above pass through. No gain control is available for this filter type.

1 Band EQ set to High-Pass Filter

Notch Filter

The Notch Filter attenuates a narrow band of frequencies centered around the Frequency setting. No gain control is available for this EQ type. The width of the attenuated band is determined by the Q setting.
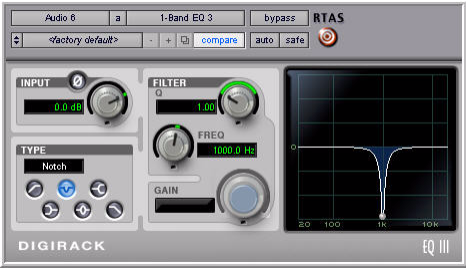
1 Band EQ set to Notch Filter

High-Shelf EQ

The High-Shelf EQ boosts or cuts frequencies at and above the Frequency setting. The amount of boost or cut is determined by the Gain setting. The Q setting determines the shape of the shelving curve.
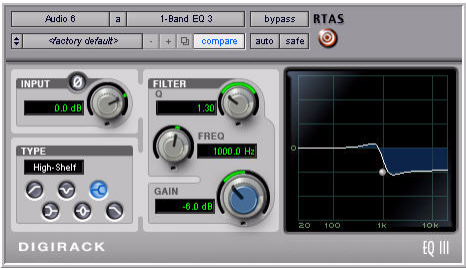
1 Band EQ set to High-Shelf EQ

Low-Shelf EQ

The Low-Shelf EQ boosts or cuts frequencies at and below the Frequency setting. The amount of boost or cut is determined by the Gain setting. The Q setting determines the shape of the shelving curve.

1 Band EQ set to Low-Shelf EQ

Peak EQ

The Peak EQ boosts or cuts a band of frequencies centered around the Frequency setting. The width of the affected band is determined by the Q setting.

1 Band EQ set to Peak EQ

Low-Pass Filter

The Low-Pass filter attenuates all frequencies above the cutoff frequency setting at the selected rate (6 dB, 12 dB, 18 dB, or 24 dB per octave) while letting all frequencies below pass through. No gain control is available for this filter type.

1 Band EQ set to Low-Pass Filter

Automating EQ III Controls

All EQ III plug-in controls can be automated in Pro Tools.

To automate a control directly from the plug-in window:

■ Control-Alt-Start-click (Windows) or Command-Option-Control-click (Macintosh) the control, and choose Enable Automation.

Enabling automation for a control

To automate multiple controls:

■ Control-Alt-Start-click (Windows) or Command-Option-Control-click (Macintosh) the control, choose Open Plug-In Automation Dialog, and choose the controls you want to automate.

The automation-enabled status of EQ III plug-in controls is indicated by the colored LED beneath each on-screen knob and colored outlines around each on-screen button.

Automation-enabled buttons

Automation-enabled knobs
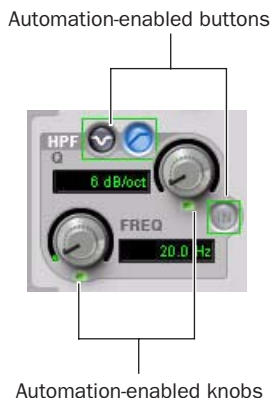
Automation-enabled controls

Even if an EQ III band is bypassed, for example, when the filter capacity of a 2–4 Band EQ is surpassed, its controls are still available for automation. This lets you continue to update settings in a session in preparation for the next time the bypassed band is enabled.

📄 *For complete instructions on automating plug-ins in Pro Tools, see the Pro Tools Reference Guide.*

EQ II

For information, see the electronic PDF guide for the EQ II plug-ins.

Click

The Click plug-in creates an audio click during session playback that you can use as a tempo reference when performing and recording. The Click plug-in receives its tempo and meter data from the Pro Tools application, enabling it to follow any changes in tempo and meter in a session. The Click plug-in is a mono-only plug-in. Several click sound presets are included.
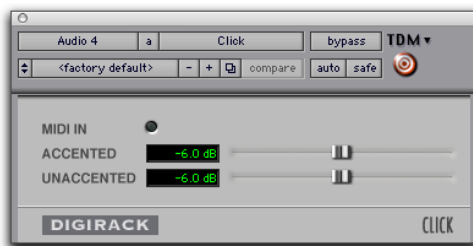
Click plug-in

Click Controls

MIDI In LED Illuminates each time the Click plug-in receives a click message from the Pro Tools application, indicating the click tempo.

**Accented** Controls the output level of the accent beat (beat 1 of each bar) of the audio click.

Unaccented Controls the output level of the unaccented beats of the audio click.

To use the Click plug-in:

1 Choose Options > Click to enable the Click option (or enable the Click button in the Transport window).

2 Create a mono Auxiliary Input and insert the Click plug-in.

3 Select a click sound preset.

4 Choose Setup > Click and set the Click and Countoff options as desired.

⚠ *The Note, Velocity, Duration, and Output options in this dialog are for use with MIDI instrument-based clicks and do not affect the Click plug-in.*
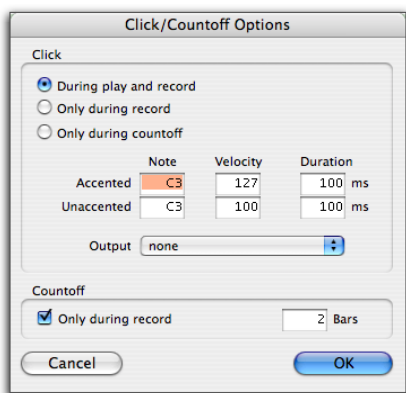

Click Options dialog

📄 *Refer to the Pro Tools Reference Guide for more information on configuring Click options.*

5 Begin playback. A click is generated according to the tempo and meter of the current session and the settings in the Click/Countoff Options dialog.

Dither

The Dither plug-in is designed for reducing quantization noise when mixing or fading low-level audio signals during word size reduction to 16-, 18-, or 20-bits.

💡 *For more advanced dithering, use the Digi-Rack POW-r Dither plug-in. See "POW-r Dither" on page 58.*
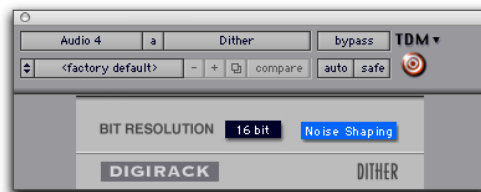
Dither plug-in

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.

The introduction of dithering can reduce these quantization artifacts with very low-level noise, minimizing artifacts as audio reaches low level. With dithering there is a trade-off between signal-to-noise performance and less-apparent artifacts. Proper use of dithering squeezes better subjective performance out of 16-bits (or whatever the destination bit-depth).

The most common application of dithering is to use it on a master output mix as the last processor in the signal path when preparing a 24-bit session for CD mastering. In this case, you

would insert the Dither plug-in on a Master Fader to reduce session bit-depth from 24-bits to 16-bits. The inserts on a Master Fader track are always post-fader.

⚠ *Insert the Dither plug-in (or any dithering plug-in, such as Dither, POW-r Dither, or Maxim) on a Master Fader track when using Bounce to Disk to create a 16-bit file (or any bit depth lower than 24-bit) from a 24-bit session; otherwise the resulting 16-bit file will be truncated at the destination bit depth. For more information on Bounce to Disk, refer to the Pro Tools Reference Guide.*

The Dither plug-in has user-selectable bit resolution and a noise shaping on/off option.

💡 *If outputting 24-bit audio to an analog destination with a 96 I/O, or 192 I/O, you do not need to use Dither. This allows maximum output fidelity from the high-performance 24-bit digital-to-analog convertors of the interfaces.*

Noise Shaping

The Dither plug-in features a technique known as noise shaping to further improve audio performance and reduce perceived noise in low-level signals. Noise shaping utilizes filtering to reduce noise that falls in the middle of the audio spectrum (specifically, around 4 kHz). This is the range where human hearing is most sensitive.

In reality, since the noise plays an important role in reducing quantization artifacts, the noise is not reduced, rather it is shifted into a frequency range where it is harder to hear. Essentially, noise shaping lessens our perception of the noise inherent in dithering schemes by shifting audible noise components into a less audible range.

Dither and Output Bit Resolution

Dither has two user-selectable settings for optimizing its operation:

Bit Resolution

Use this pop-up menu to choose one of three possible resolutions for the Dither processing. As a general rule, set this control to the maximum bit resolution of your destination.

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

18-bit Recommended for output to analog devices if you are using an 18-bit audio interface, such as the 888 I/O or 882 I/O audio interface; this is the maximum resolution available from the 18-bit digital-to-analog convertors of these devices.

20-bit Recommended for output to digital devices that support a full 20-bit recording data path, such the Sony PCM-9000 optical mastering recorder, or the Alesis ADAT XT 20. Use this setting for output to analog devices if you are using a 20-bit audio interface, such as the 882|20 I/O audio interface. The 20-bit setting can also be used for output to 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.

If you want, you can choose not to use the Dither plug-in and instead utilize the full 24-bit resolution of Pro Tools' digital output, depending on your destination device. If you are using a 24-bit audio interface (such as the 192 I/O, 96 I/O, or 888|24 I/O) and are outputting audio to an analog destination, we recommend that you do not use dither.

⚠ *The DigiRack Dither plug-in only provides eight channels of uncorrelated dithering noise. If DigiRack Dither is used on more than eight tracks, the dithering noise begins to repeat and dither performance is impaired. For example, if two Quad DigiRack Dithers are used, both Quad instances of Dither will have all of their dither noise uncorrelated. However, any additional instances of the Dither plug-in will begin to repeat the dithering noise.*

Noise Shaping

This button engages or disengages Noise Shaping. Noise shaping is on when the button is highlighted in blue. See "Noise Shaping" on page 57 for more information.

📄 *Refer to the Pro Tools Reference Guide for details on using the Dither plug-in during mixdown.*

POW-r Dither

(Pro Tools HD and Pro Tools LE Only)

The POW-r Dither plug-in is an advanced type of dither that provides optimized word length reduction. It is designed for final-stage critical mixdown and mastering tasks where the highest possible fidelity is desired when reducing bit depth. For more information on dithering, see "Dither" on page 56.

POW-r Dither plug-in

⚠ *The POW-r Dither plug-in does not run on third party applications that use DAE.*

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

POW-r Dither Controls

Bit Resolution

Use this pop-up menu to choose either 16- or 20-bit resolutions for POW-r Dither processing. Set this control to the maximum bit resolution of your destination.

16-bit Recommended for output to devices with a maximum resolution of 16-bits, such as DAT and CD recorders.

20-bit Recommended for output to devices that support a full 20-bit recording data path.

Noise Shaping

Noise shaping improves audio performance and lessens our perception of the noise inherent in dithering schemes by shifting audible noise components into a less audible range.

The POW-r Dither plug-in is not appropriate for truncation stages that are likely to be further processed. It is recommended that POW-r Dither be used only as the last insert in the signal chain (especially when using Type 1 Noise Shaping).

The POW-r Dither plug-in provides three types of noise shaping, each with its own characteristics. Try each noise shaping type and choose the one that adds the least amount of coloration to the audio being processed.

Type 1 Has the flattest frequency spectrum in the audible range of frequencies, modulating and accumulating the dither noise just below the Nyquist frequency. Recommended for less stereophonically complex material such as solo instrument recordings.

Type 2 Has a psychoacoustically optimized low order noise shaping curve. Recommended for material of greater stereophonic complexity.

Type 3 Has a psychoacoustically optimized high order noise shaping curve. Recommended for full-spectrum, wide-stereo field material.

Refer to the Pro Tools Reference Guide for details on using the Dither plug-in.

D-Verb

D-Verb is a studio-quality reverb provided in TDM, RTAS, and AudioSuite formats.

The TDM version of the D-Verb plug-in is not supported at 192 kHz; use the RTAS version instead.

D-Verb Controls

D-Verb plug-in

Output Meter

The Output Meter indicates the output level of the processed signal. With the stereo version of D-verb, it represents the summed stereo output. It is important to note that this meter indicates the output level of the signal—not the input level. If this meter clips, it is possible that the signal clipped on input before it reached D-Verb. Monitor your send or insert signal levels closely to help prevent this from happening.

Clip Indicator

The Clip Indicator shows if clipping has occurred. It is a clip-hold indicator. If clipping occurs at any time during audio playback, the clip lights remain on. To clear the clip indicator, click it. With longer reverb times there is a greater likelihood of clipping occurring as the feedback element of the reverb builds up and approaches a high output level. See also "Clip Indicators" on page 15.

Input Level

The Input Level slider adjusts the input volume of the reverb to prevent the possibility of clipping and/or increase the level of the processed signal.

Mix

The Mix slider adjusts the balance between the dry signal and the effected signal, giving you control over the depth of the effect. This control is adjustable from 100% to 0%.

Algorithm

This control selects one of seven reverb algorithms: Hall, Church, Plate, Room 1, Room 2, Ambience, or Nonlinear. Selecting an algorithm changes the preset provided for it. Switching the Size setting changes characteristics of the algorithm that are not altered by adjusting the decay time and other user-adjustable controls. Each of the seven algorithms has a distinctly different character:

Hall A good general purpose concert hall with a natural character. It is useful over a large range of size and decay times and with a wide range of program material. Setting Decay to its maximum value will produce infinite reverberation.

Church A dense, diffuse space simulating a church or cathedral with a long decay time, high diffusion, and some pre-delay.

Plate Simulates the acoustic character of a metal plate-based reverb. This type of reverb typically has high initial diffusion and a relatively bright sound, making it particularly good for certain percussive signals and vocal processing. Plate reverb has the general effect of thickening the initial sound itself.

Room 1 A medium-sized, natural, rich-sounding room that can be effectively varied in size between very small and large, with good results.

Room 2 A smaller, brighter reverberant characteristic than Room 1, with a useful adjustment range that extends to "very small."

Ambient A transparent response that is useful for adding a sense of space without adding a lot of depth or density. Extreme settings can create interesting results.

Nonlinear Produces a reverberation with a natural buildup and an abrupt cutoff similar to a gate. This unnatural decay characteristic is particularly useful on percussion, since it can add an aggressive characteristic to sounds with strong attacks.

Size

The Size control, in conjunction with the Algorithm control, adjusts the overall size of the reverberant space. There are three sizes: Small, Medium, and Large. The character of the reverberation changes with each of these settings (as does the relative value of the Decay setting). The Size buttons can be used to vary the range of a reverb from large to small. Generally, you should select an algorithm first, and then choose the size that approximates the size of the acoustic space that you are trying to create.

Diffusion

Diffusion sets the degree to which initial echo density increases over time. High settings result in high initial build-up of echo density. Low settings cause low initial buildup. This control interacts with the Size and Decay controls to affect the overall reverb density. High settings of diffusion can be used to enhance percussion. Use low or moderate settings for clearer and more natural-sounding vocals and mixes.

Decay

Decay controls the rate at which the reverb decays after the original direct signal stops. The value of the Decay setting is affected by the Size and Algorithm controls. This control can be set to infinity on most algorithms for infinite reverb times.

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.

Hi Frequency Cut

Hi Frequency Cut controls the decay characteristic of the high frequency components of the reverb. It acts in conjunction with the Low-Pass Filter control to create the overall high frequency contour of the reverb. When set relatively low, high frequencies decay more quickly than low frequencies, simulating the effect of air absorption in a hall. The maximum value of this control is Off (which effectively means bypass).

Low-Pass Filter

Low-Pass Filter controls the overall high frequency content of the reverb by setting the frequency above which a 6 dB per octave filter attenuates the processed signal. The maximum value of this control is Off (which effectively means bypass).

Dynamics II

There are five types of DigiRack Dynamics processors: Compressor, Limiter, Gate, Expander/Gate and DeEsser.

Compressor II
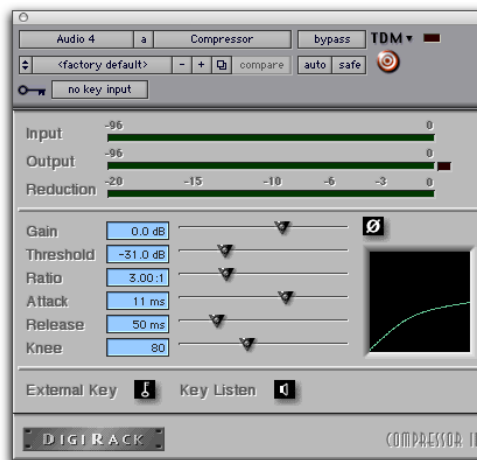
Compressor II plug-in (mono shown)

The Compressor reduces the dynamic range of signals that exceed a selected threshold by a specific amount. The increase of input signal needed to cause a 1 dB increase in the output signal of the compressor is called the compression ratio. For example, with a ratio of 4:1, an 8 dB increase of input produces a 2 dB increase in the output.

Audio material often varies in loudness, and can be above the threshold at one moment and below it the next. The Attack slider sets the compressor's response time, or attack. The Release slider sets the amount of time that it takes for the compressor's gain to return to its original level.

Using Compression Effectively

To use compression most effectively, the attack time should be set so that signals exceed the threshold level long enough to cause an increase in the average level. This helps ensure that gain reduction doesn't decrease the overall volume.

Release times should be set long enough that if signal levels repeatedly rise above the threshold, they cause gain reduction only once. If the release time is too long, a loud section of the audio material could cause gain reduction that persists through a soft section. Of course, compression has many creative uses that break these rules.

The Compressor has built-in metering that shows the amount of gain reduction taking place. The gain reduction meter usually remains at 0 level when the input signal is below the threshold and falls to the left to show the amount of gain reduction in decibels when the input signal exceeds the threshold.

Compressor Controls

Phase Invert Inverts the phase (polarity) of the input signal to change frequency response characteristics between multi-miked sources or to correct for miswired microphone cables.

Phase Invert

Gain Provides overall output gain adjustment for compensating for heavily compressed signals.

Input Meter Indicates the level of the unprocessed input signal to the Compressor.

Output Meter Indicates the output level of the Compressor, including any gain compensation added with the Gain control.

Reduction Indicates the amount of gain reduction in dB.

Threshold Sets the threshold level. Signals that exceed this level will be compressed. Signals that are below it will be unaffected. A level setting of 0 dB is equivalent to no compression.

Unlike scales on analog compressors, metering scales on a digital device reflect a 0 dB value that indicates full scale (fs)—the full-code signal level. There is no headroom above 0 dB.

Ratio Sets the compression ratio. The range is based on decibels above the threshold. For example, if this is set to 2:1, it will compress changes in signals above the threshold by one half.

Attack Sets the Compressor's attack time. The smaller the value, the faster the attack. The faster the attack, the more rapidly the Compressor applies attenuation to the signal. If you use fast attack times and heavy limiting, you should generally use a proportionally longer release time, particularly with material that contains many peaks in close proximity.

Release Controls how long it takes for the Compressor to be fully deactivated after the input signal drops below the threshold level. In general, if you use heavy compression, you should use proportionally longer release times in order to avoid pumping that may occur when the Compressor is forced to jump back and forth between compressed and uncompressed signal levels. Lengthening the release time will help smooth these changes in level by introducing a lag in the ramp-up/down time of attenuation. Use shorter release times on material with fewer peaks that do not occur in close proximity to each other.

Knee Sets the rate at which the compressor reaches full compression once the threshold has been exceeded. This control ranges from 0 (hardest response) to 200 (softest response).

Graph Displays the response curve set by the Compressor's Threshold, Ratio, and Knee settings. As you adjust these controls, refer to the graph to see how the shape of this curve change.

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 on some plug-ins. To disable LFE processing, deselect this button. For more information on the LFE channel, refer to the *Pro Tools Reference Guide*.

External Key Enables and disables side-chain processing. Side-chain processing uses the varying amplitude of a separate reference track to trigger compression. See "Using the Side-Chain Input" on page 20.

Key Listen When enabled, this lets you listen to the reference track controlling the side-chain input. This is useful for fine tuning the Compressor's settings to the Key Input. Key Listen auditions the right audio channel on RTAS and AudioSuite versions of the Compressor.

⚠ *Key Listen is not saved with the other plug-in settings.*

Limiter II
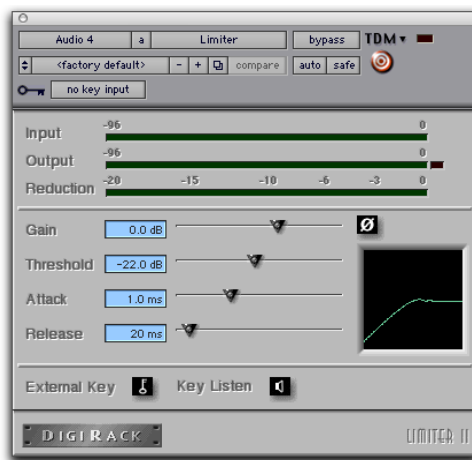
Limiter II plug-in

The Limiter is used to prevent signal peaks from ever exceeding a chosen level so that they don't overload amplifiers or recording devices. Most limiters have ratios of 10:1 or 20:1, although some provide ratios of up to 100:1. Large ratios effectively limit the dynamic range of the signal to a specific value by setting an absolute ceiling for the dynamic range.

The Limiter's ratio is internally set to 100:1, and the Attack time defaults to 1 millisecond. The Limiter is similar to heavy compression. It can be useful for reducing pops and clicks, or for hard-limiting dynamic range for broadcast or band-limited mediums such as cassette.

Limiting is used to prevent short-term peaks from reaching their full amplitude. Used judiciously, limiting produces higher average levels, while avoiding overload (clipping or distortion), by limiting only some short-term transients in the source audio. To prevent the ear from hearing the gain changes, extremely short attack and release times are used.

Limiting is used to remove only occasional peaks because gain reduction on successive peaks would be noticeable. If audio material contains many peaks, the threshold should be raised and the gain manually reduced so that only occasional, extreme peaks are limited.

Limiter Controls

Phase Invert Inverts the phase (polarity) of the input signal to change frequency response characteristics between multi-miked sources or to correct for miswired microphone cables.

Gain Provides overall output gain adjustment.

Input Meter Indicates the level of the unprocessed input signal to the Limiter.

Output Meter Indicates the output level of the Limiter, including any gain compensation added with the Gain control.

Reduction Indicates the amount by which the signal is being attenuated.

Threshold Sets the threshold level. Signals that exceed this level will be limited. Signals that are below it will be unaffected.

Attack Sets the Limiter's attack time. The smaller the value, the faster the attack. The faster the attack, the more rapidly the Limiter applies attenuation to the signal. If you use fast attack times and heavy limiting, you should generally use a proportionally longer release time, particularly with material that contains many peaks in close proximity.

Release Controls how long it takes for the Limiter to be fully deactivated after the input signal drops below the threshold level. In general, if you use heavy limiting, you should use proportionally longer release times in order to avoid pumping that may occur when the Limiter is forced to jump back and forth between limited and unlimited signal levels. Lengthening the release time will help smooth these changes in level by introducing a lag in the ramp-up/down time of attenuation. Use shorter release times on material with fewer peaks that do not occur in close proximity to each other.

**Graph** Displays the response curve set by the Limiter's Threshold setting. As you adjust these controls, refer to the graph to see how the shape of this curve changes.

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

External Key Enables and disables side-chain processing. Side-chain processing uses the varying amplitude of a separate reference track to trigger limiting. See "Using the Side-Chain Input" on page 20.

Key Listen When enabled, this lets you listen to the reference track controlling the side-chain input. This is often useful for fine tuning the Limiter's settings to the Key Input. Key Listen auditions the right audio channel on RTAS and AudioSuite versions of the Limiter.

⚠ *Key Listen settings are not saved with other plug-in settings.*

Gate II
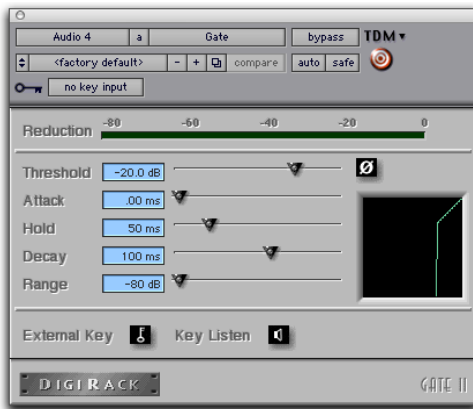
Gate II plug-in

The Gate (also known as a noise gate) lets a signal above the selected threshold pass through to the output at unity gain, without dynamic processing. Once the input signal falls below the threshold level, the gating shuts down the signal by fully or partially attenuating the output. In this way, the desired signal is allowed to pass, but unwanted signal is not. The Gate is useful for noise removal on individual tracks, and can be used for special effects such as cutting off a reverb tail.

Gate Controls

Phase Invert Inverts the phase (polarity) of the input signal to change the frequency response characteristics between multi-miked sources or to correct for miswired microphone cables.

Gating Indicates the amount of reduction in dB.

Threshold Sets the threshold level. Signals that exceed this level will pass through. Signals that are below it will be gated, depending on the settings of the Attack, Hold, Decay, and Range controls (explained below).

Attack Sets the attack time of the Gate.

Hold Specifies a duration (in seconds or milliseconds) that the Gate will stay open after the initial attack cycle. This can be used as a one-shot function to keep the Gate open for longer periods of time with a single crossing of the threshold. It can also be used to prevent gate chatter that may occur if varying input levels near the threshold cause the Gate to open and close very rapidly.

Decay Controls how long it takes for the Gate to close after the signal falls below the threshold level.

Range Sets the depth of the Gate when closed. It has a maximum depth of –80 dB. Setting the gate to higher range levels allows more of the gated audio that falls below the threshold to peek through the gate at all times. This is useful for things such as drum leakage, where you may want to suppress the overall drum kit sound by a specific amount, while emphasizing the gated instrument such as a snare.

Graph Displays the response curve set by the Gate's Threshold and Range settings. As you adjust these controls, refer to the graph to see how the shape of this curve changes.

**External Key** Enables and disables side-chain processing. Side-chain processing uses the varying amplitude of a separate reference track to trigger limiting. See "Using the Side-Chain Input" on page 20.

Key Listen When enabled, this lets you listen to the reference track controlling the side-chain input. This is often useful for fine tuning the Gate's settings to the Key Input.

⚠ *Key Listen is not saved with the other plug-in settings.*

Expander/Gate II
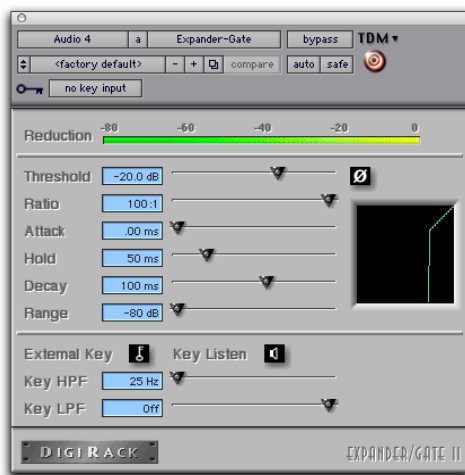
Expander/Gate II plug-in

The Expander/Gate reduces noise by decreasing the gain of signals that fall below a user-selectable threshold. Expanders are particularly useful for reducing noise or signal leakage that creeps into recorded material as its level falls, as often occurs in the case of headphone leakage.

Expanders can be thought of as soft noise gates since they provide a gentler way of cutting off noisy low-level signals than the typically abrupt cutoff of a gate. If you want, however, you can actually use this plug-in as a gate, simply by setting the Ratio to its maximum value and using short attack, decay, and release settings.

Expander/Gate Controls

Phase Invert Inverts the phase (polarity) of the input signal to change the frequency response characteristics between multi-miked sources or to correct for miswired microphone cables.

Reduction Meter Indicates the amount of signal reduction in dB.

Threshold Sets the threshold level. Signals that fall below the threshold will be reduced in gain. Signals that are above it will be unaffected. (When adjusting the Threshold slider be sure that audio material is playing through the Expander/Gate to see changes reflected in the Reduction meter.)

Ratio Sets the amount of expansion. For example, if this is set to 2:1, it will lower signals below the threshold by one half. At higher ratio levels (such as 30:1 or 40:1) the Expander/Gate functions like a gate by reducing lower level signals much more dramatically. As you adjust the ratio control, refer to the built-in graph to see how the shape of the expansion curve changes.

Attack Sets the Expander's attack time. This determines how quickly a signal's level is reduced once it falls below the threshold. Use this along with the Ratio setting to control how soft the Expander's gain reduction curve is.

Hold Specifies a duration (in seconds or milliseconds) that the Expander/Gate will stay open after the initial attack cycle. This can be used as a one-shot function to keep the Expander/Gate open for longer periods of time with a single crossing of the threshold. It can also be used to prevent gate chatter that may occur if varying input levels near the threshold cause the gate to open and close very rapidly.

Decay Controls how long it takes for the gate to close after the input signal falls below the threshold level and the hold time has passed.

Range Sets the depth of the gate when closed. It has a maximum depth of –80 dB. Setting the gate to higher range levels allows more and more of the gated audio that falls below the threshold to peek through the gate at all times.

Key HPF This is a high-pass filter that removes low frequencies in the Key Input signal. Use this to define a specific frequency range of the Key Input to trigger the Expander/Gate. A typical use for this feature is to filter a drum track used as a Key Input so that only specific high frequencies (such as a snare drum) will open the Gate.

Key LPF This is a low-pass filter that removes high frequencies in the Key Input signal. Use this to define a specific frequency range of the Key Input to trigger the Expander/Gate. A typical use for this feature is to filter a drum track used as a Key Input so that only specific low frequencies (such as a kick drum or tom) trigger the Expander/Gate. The maximum value for Key LPF is Off. This lets the Key Input to pass through without limiting its high-frequency bandwidth.

Graph Displays the response curve set by the Expander/Gate's Threshold, Ratio, and Range settings. As you adjust these controls, refer to the graph to see how the shape of this curve changes.

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

External Key Enables and disables side-chain processing. Side-chain processing allows the Expander/Gate to listen to the varying amplitude of a separate reference track and use it to trigger expansion. See "Using the Side-Chain Input" on page 20.

Key Listen When enabled, this allows you to listen to the reference track controlling the side-chain input. This is often useful for fine tuning the Expander/Gate's settings to the Key Input.

⚠ *Key Listen is not saved with the other plug-in settings.*

DeEsser
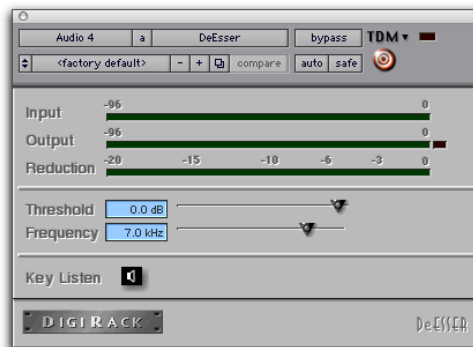
DeEsser plug-in

The DeEsser reduces sibilants ("s," "sh," and "t" sounds) and other high frequency noises that can occur in vocals, voice-overs, and wind instruments such as flutes. These sounds can cause peaks in an audio signal and lead to distortion.

The DeEsser reduces these unwanted sounds using fast-acting compression. A Threshold control sets the level above which compression starts, and a Frequency control sets the frequency band in which the DeEsser operates.

⚠ *The DeEsser is a monophonic-only plug-in.*

Using De-essing Effectively

To use de-essing most effectively, insert the DeEsser before compressor or limiter plug-ins.

Because too much de-essing can make audio material sound lifeless, it is best used on individual tracks rather than entire mixes.

The Frequency slider should be set to remove sibilants (typically the 4–10 kHz range) and not other parts of the signal. This helps prevent de-essing from changing the original character of the audio material.

Similarly, Threshold should be set high enough that de-essing is triggered only by sibilants. If the Threshold is set too low, a loud, non-sibilant section of audio material could cause unwanted gain reduction or cause sibilants to be over attenuated.

To improve de-essing of material that has both very loud and very soft passages, automate the Threshold control so that it is lower on soft sections.

DeEsser Controls

Input Meter Indicates the level of the unprocessed input signal to the DeEsser.

Output Meter Indicates the output level of the DeEsser.

Reduction Indicates the amount of gain reduction in dB. It remains at 0 dB level when the input signal is below the threshold.

Threshold Sets the threshold level. Signals that exceed this level will be compressed. Signals that are below it will be unaffected. A setting of 0 dB is equivalent to no de-essing.

Frequency Sets the frequency band in which the DeEsser operates. Frequencies in the specified range will be gain reduced. To find the optimum Frequency setting, sweep this control back and forth during audio playback.

Key Listen Monitors the sibilant peaks used by the DeEsser as a Key Input to trigger compression. This is useful for listening only to the sibilance and fine tuning settings to remove them.

Mod Delay II

The DigiRack Mod Delay II plug-ins provide time delay-based effects such as echo, slap echo, doubling, chorusing, and flanging. Mod Delay II supports sample rates up to 192 kHz.

⚠ *Pro Tools 7.0 supports the original Mod Delay plug-in, but it is recommended that you only use Mod Delay II for all new sessions. Use Mod Delay (located in the Additional Files folder) only for compatibility with existing sessions. If you do not need to use Mod Delay, move it into the Plug-Ins (Unused) folder. Mod Delay settings can be imported into Mod Delay II.*

Mod Delay II

Mod Delay II plug-in (Long Delay shown)

There are five different Mod Delay II plug-ins, capable of different maximum delay times:

- The Short Delay provides 43 ms of delay at all sample rates.

- The Slap Delay provides 171 ms of delay at all sample rates.

⚠ *Short Delay and Slap Delay do not have Tempo, Meter, Duration, and Groove controls.*

- The Medium Delay provides 341 ms of delay at all sample rates.

- The Long Delay provides 683 ms of delay at all sample rates.

- The Extra Long Delay provides 2.73 seconds of delay at all sample rates.

⚠ *The TDM versions of the Extra Long Delay mono-to-stereo and stereo plug-in are not supported at 96 kHz. All TDM versions of the Extra Long Delay plug-in are not supported at 192 kHz. RTAS versions of the Extra Long Delay plug-in are fully supported at all sample rates.*

Mod Delay II Controls

Input Controls the input volume of the delay to prevent clipping.

Mix Controls the balance between the delayed signal (wet) and the original signal (dry). If you are using a delay for flanging or chorusing, you can control the depth of the effect somewhat with the Mix setting.

LPF (Low-Pass Filter) Controls the cutoff frequency of the Low-Pass Filter. Use the LPF setting to attenuate the high frequency content of the feedback signal. The lower the setting, the

more high frequencies are attenuated. The maximum value for LPF is Off. This lets the signal pass through without limiting the bandwidth of the plug-in.

Delay Sets the delay time between the original signal and the delayed signal.

Depth Controls the depth of the modulation applied to the delayed signal.

Rate Controls the rate of modulation of the delayed signal.

Feedback Controls the amount of feedback applied from the output of the delay back into its input. It also controls the number of repetitions of the delayed signal. Negative feedback settings give a more intense "tunnel-like" sound to flanging effects.

Tempo Sync Tempo Sync provides a direct connection between session tempo and plug-in parameters that support MIDI Beat Clock. This direct connection lets plug-in parameters such as delay, auto-pan, and other time-domain effects automatically synchronize to, and follow changes in, session tempo. For plug-ins that do not support Tempo Sync, the original MIDI Beat Clock window (Setup > MIDI > MIDI Beat Clock) remains available in Pro Tools.

⚠ *Tempo Sync is not available in plug-ins with which it would limit functionality. For example, for the Virus Indigo and IndigoV40 plug-in, use the Pro Tools MIDI Beat Clock window (Setup > MIDI > MIDI Beat Clock). See the Pro Tools Reference Guide for more information.*

For more information, see "Tempo Sync" on page 21.

Tempo Available in some plug-ins that do not support Tempo Sync, Tempo selects the desired tempo in beats per minute (bpm). This setting is independent of Pro Tools' tempo. When a specific Duration is selected (see "Duration" below), moving this control will affect the Delay setting. Likewise, the range of both controls will be limited to the maximum available delay with the currently selected Duration. To enter very short or long delays it may be necessary to deselect all Duration buttons.

☿ *Use Event > Identify Beat or Event > Beat Detective to determine the tempo. For more information about Identify Beat and Beat Detective refer to the Pro Tools Reference Guide.*

Meter Use this to enter either simple or compound time signatures. The Meter control defaults to a 4/4 time signature.

Duration Specifies a desired delay from a musical perspective. Enter the desired delay by selecting appropriate note value (whole note, half note, quarter note, eight note, or sixteenth note). Select the Dot or Triplet modifier buttons to dot the selected note value or make it a triplet. For example, selecting a quarter note and then selecting the dot indicates a dotted quarter note, and selecting an eighth note and then selecting the triplet indicates a triplet eight note.
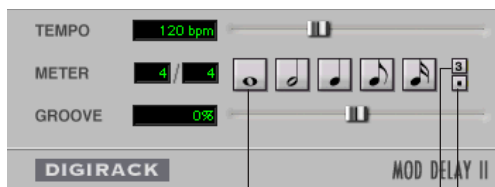
Tempo, Meter, Duration, and Groove controls

Groove Provides fine adjustment of the delay in percentages of a 1:4 subdivision of the beat. It can be used to add "swing" by slightly offsetting the delay from the precise beat of the track.

⚠ *It is not possible to exceed the maximum delay length for a particular version of Mod Delay II. Consequently, when adjusting any of the tempo controls (Tempo, Meter, Duration, and Groove) you may not be able to adjust the control across its full range. If you encounter this behavior, switch to a version of Mod Delay II that has a longer delay time (for example, switch from Medium Delay to Long Delay).*

Multichannel Mod Delay II

The Tempo and Meter controls are linked on multichannel versions of Mod Delay II. Each channel has its own Duration and Groove controls, but the Tempo and Meter controls are global.

Tempo, Meter, Duration, and Groove controls for a stereo instance of Mod Delay II

Signal Generator

The Signal Generator plug-in produces audio test tones in a variety of frequencies, waveforms, and amplitudes. It is particularly useful for generating reference signals with which to calibrate Pro Tools|HD interfaces (such as the 96 I/O or 192 I/O) and other elements of your studio.

📖 *Refer to the Pro Tools|HD Getting Started Guide for instructions on using Signal Generator to calibrate the 96 I/O, or 192 I/O audio interfaces, as well as older audio interfaces.*

Signal Generator plug-in

💡 *The TDM Signal Generator produces a tone as soon as it is inserted on a track. To mute the Signal Generator, use the Bypass button. When using the RTAS version of Signal Generator, start playback to generate.*

Signal Generator Controls

Frequency Sets the frequency of the signal in hertz. Values range from a low of 20 Hz to a high of 20 kHz in a 44.1 kHz session. The upper limit of the frequency range for this setting will increase to match the *Nyquist frequency* (half the sample rate) in 96 kHz and 192 kHz sessions (HD-series systems only).

Level Sets the amplitude of the signal in decibels. Values range from a low of –95 dB to a high of 0.0 dB.

Signal These buttons select the waveform. Choices are sine, square, sawtooth, triangle, white noise, and pink noise.

⚠ *The Signal Generator plug-in is not intended for rigorous test purposes; it is a simple level calibration tool.*

Peak Generates signal at the maximum possible level without clipping.

RMS Generates signal at levels consistent with the RMS (Root-Mean-Square) value, or the effective average level of the signal.

Pitch

(Pro Tools HD Only)

The Pitch plug-in is designed for a variety of audio production applications ranging from pitch correction of musical material to sound design.
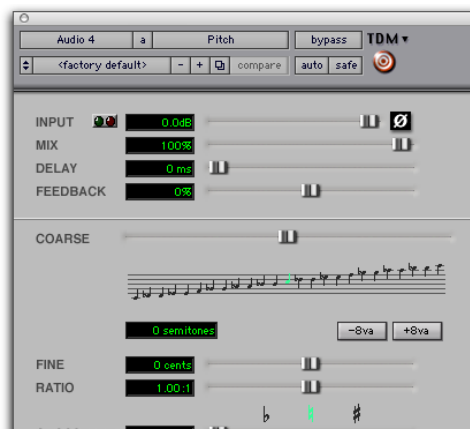
Pitch plug-in

Pitch processing uses the technique of varying sample playback rate to achieve pitch transposition. Because changing audio sample playback rate results in the digital equivalent of vari-speeding with tape, this is an unsatisfactory method since it changes the overall duration of the material.

Pitch transposition with the Pitch plug-in involves a much more complex technique: digitally adding or subtracting portions of the audio waveform itself, while using de-glitching crossfades to minimize undesirable artifacts. The result is a processed signal that is transposed in pitch, but still retains the same overall length as the original, unprocessed signal.

💡 *The Pitch plug-in was formerly called DPP-1. It is fully compatible with all settings and presets created for DPP-1.*

Pitch Controls

Input Level Attenuates the input level of the Pitch plug-in to help prevent internal clipping.

Signal Present Indicator LED Indicates the presence of an input signal.

Clip Indicator Indicates whether clipping has occurred on output. It is a Clip-Hold Indicator. If clipping occurs at any time, the clip light will remain on. To clear the Clip Indicator, click it. Long delay times and high feedback times increase the likelihood of clipping.

Mix Adjusts the ratio of dry signal to effected signal in the output. In general, this control should be set to 100% wet, unless you are using the Pitch plug-in in-line on an Insert for an individual track or element in a mix. This control can be adjusted over its entire range with little or no change in output level.

Delay Sets the delay time between the original signal and the pitch-shifted signal. It has a maximum setting of 125 milliseconds. You can use the Delay control in conjunction with the Feedback control to generate a single pitch-shifted echo, or a series of echoes that climb in pitch.

Feedback Controls the amount and type of feedback (positive or negative) applied from the output of the delay portion of the Pitch plug-in back into its input. It also controls the number of repetitions of the delayed signal. You can use it to produce effects that spiral up or down in pitch, with each successive echo shifted in pitch.

Coarse Adjusts the pitch of a signal in semitones over a two octave range. Pitch changes are indicated both in the Semitones field and in the Musical Staff Relative section below this slider. Using the –8va and +8va buttons in conjunction with the Coarse slider provides a full 4-octave range of adjustment.

–8va and +8va Buttons Clicking the –8va button adjusts pitch down one octave from the current setting of the coarse and fine pitch controls. Clicking the +8va button adjusts pitch up one octave from the current setting of the coarse and fine pitch controls.

Relative Pitch Entry (Musical Staff)

Clicking on any note on this musical staff selects a relative pitch transposition value that will be applied to an audio signal. If Middle C is illuminated, it indicates that no pitch transposition has been selected. If a pitch transposition is selected, the note interval corresponding to the selected transposition value is indicated in yellow. Alt-clicking (Windows) or Option-clicking (Macintosh) on the staff will set the coarse pitch change value to zero.
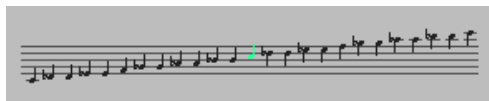
Relative pitch entry

Fine Controls the pitch of a signal in *cents* (hundredths of a semitone) over a 100 cent range. The range of this slider is –49 to +50 cents. Precise Pitch change values are indicated in the Cents field. The flat, natural, and sharp signs below this slider indicate deviation from the nearest semitone.

Ratio Indicates the ratio of transposition between the original pitch and the selected transposition value.

Crossfade Adjusts crossfade length in milliseconds to optimize performance of the Pitch plug-in according to the type of audio material you are processing. The Pitch plug-in performs pitch transposition by replicating or subtracting portions of audio material and very quickly crossfading between these alterations in the waveform of the audio material.

Crossfade length affects the amount of smoothing performed on audio material to prevent audio artifacts such as clicks from occurring as the audio is looped to generate the pitch shift.

In general, small, narrow-range pitch shifts require longer crossfades and large shifts require smaller ones. The disadvantage of a long crossfade time is that it will smooth the signal, including any transients. While this is sometimes desirable for audio material such as vocals, it is not appropriate for material with sharp transients such as drums or percussion.

The default setting for this control is Auto. At this setting, crossfade times are set automatically, according to the settings of the Coarse and Fine pitch controls. The Auto setting is appropriate for most applications. However, you can manually adjust and optimize crossfade times using the Crossfade slider if necessary. For audio material with sharper attack transients, use shorter crossfade times. For audio material with softer attack transients, use longer crossfade times.

Minimum Pitch Sets the minimum fundamental pitch that the Pitch plug-in will recognize when performing pitch transposition. Use this to optimize the Pitch plug-in's performance by adjusting this control based on the lowest fundamental pitch of the audio material that you want to process.

On audio material with a low fundamental pitch frequency content (such as an electric bass guitar) setting this control to a lower frequency such as 30 Hz will improve the Pitch plug-in's performance. The most important thing to remember when using this control is that the fundamental frequency of audio material you want to process must be above the frequency you set here.

The range of this slider is from 15 Hz to 1000 Hz. The default setting is 60 Hz. Adjustment is tied to the current setting of the Maximum Pitch control so that the minimum range is never less than one octave, and the maximum range never more than five octaves.

Maximum Pitch Adjusts the maximum fundamental pitch that the Pitch plug-in will recognize when performing pitch transposition. To optimize the Pitch plug-in's performance, adjust this setting (and the Minimum Pitch setting) based on the highest fundamental pitch of the audio material that you want to process. The range of this slider is from 30 Hz to 4000 Hz. The default setting is 240 Hz.

TimeAdjuster

The TimeAdjuster plug-in provides three functions:

- Compensation for delays due to TDM- or RTAS-based routing
- Gain compensation (+/– 24 dB)
- Phase inversion for correcting out-of-phase signals
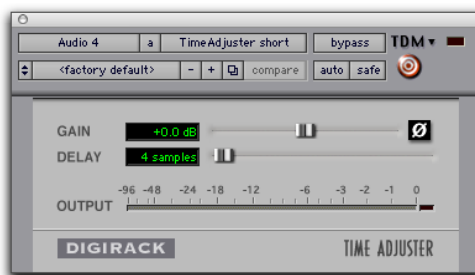
TimeAdjuster plug-in

☼ *For special information on Delay Compensation and Time Adjuster, see the Pro Tools Reference Guide.*

The TimeAdjuster plug-in is an efficient way to compensate for DSP or host-based processing delays in your Pro Tools system.

There are three versions of the TimeAdjuster plug-in, each of which supports different sample delay ranges:

Short Supports a maximum delay of 256 samples at all sample rates.

Medium Supports a maximum delay of 2048 samples at all sample rates.

Long Supports a maximum delay of 8192 samples at all sample rates.

TimeAdjuster Controls

Phase Invert Inverts the phase (polarity) of the input signal. While most Digidesign plug-ins supply a phase invert button of their own, some third-party plug-ins may not. Phase inversion is also useful for performing delay compensation by tuning unknown delay factors by ear (see "Using TimeAdjuster for Manual Delay Compensation" on page 74).

Gain Provides up to 24 dB of positive or negative gain adjustment. This control is useful for altering the gain of a signal by a large amount in real time. For example, when you are working with audio signals that are extremely low level, you may want to adjust the channel gain to a reasonable working range so that a fader is positioned at its optimum travel position. Use the Gain control to make a wide range of gain adjustment in real time without having to permanently process the audio files, as you would with an AudioSuite plug-in.

Delay Provides up to 8192 samples of delay compensation adjustment, or general adjustment of phase relationships of audio recorded with multiple microphones, depending on which version of TimeAdjuster is used. It defaults to a minimum delay of four samples, which is the delay created by use of the TimeAdjuster plug-in itself.

While phase inversion controls have been used for many years by engineers as creative tools for adjustment of frequency response between multiple microphones, sample-level delay adjustments provide far more control. Creative use of this control can provide a powerful tool for adjusting frequency response and timing relationships between audio signals recorded with multiple microphones.

Using TimeAdjuster for Manual Delay Compensation

DSP and host-based processing in all digital systems incurs delay of varying amounts. You can use the TimeAdjuster TDM plug-in to apply an exact number of samples of delay to the signal path of a Pro Tools track to compensate for delay incurred by specific plug-ins. TimeAdjuster provides presets for common delay-compensation scenarios.

To compensate for several plug-ins in-line, use the delay times from each settings file as references, and add them together to derive the total delay time.

⚠ *Some plug-ins (such as Digidesign's Maxim and DINR BNR) have different delays at different sample rates. Refer to the Digidesign Plug-Ins Guide for more information about these plug-ins.*

Alternatively, look up the delay in samples for the plug-ins you want to compensate for, then apply the appropriate amount of delay.

A delay table for DigiRack plug-ins appears in Appendix B, "DSP Delays Incurred by TDM Plug-Ins."

To manually compensate for DSP-induced delays, try one of the following methods:

Phase Inversion

If you are working with phase-coherent track pairs, or tracks recorded with multiple microphones, you can invert the phase to negate the delay. If you don't hear any audio when you invert a signal's phase, you have precisely adjusted and compensated for the delay. This is because when you monitor duplicate signals and invert the polarity (phase) of one of them, the signals will be of opposite polarity and cancel each other out. This technique is convenient for finding the exact delay setting for any plug-in.

To determine the delay of a plug-in by inverting its signal phase:

1 Place duplicate audio regions on two different audio tracks and pan them to the center (mono).

2 Apply the plug-in whose delay you want to calculate to the first track, and a Time Adjuster plug-in to the second track.

3 With TimeAdjuster, invert the phase.

4 Control-drag (Windows) or Command-drag (Macintosh) to fine-tune delay in one sample increments, or use the up/down arrow keys to change the delay one sample at a time until the audio signal disappears.

5 Change the polarity back to normal.

6 Save the TimeAdjuster setting for later use.

Comb-Filter Effect Cancellation

Adjust the delay with the signal in phase until any comb-filter effects cancel out.

Viewing Channel Delay

Because plug-ins display their delay values in the channel delay indicators, this can be used as another method for determining delay compensation.

To view time delay values and use TimeAdjuster to compensate for the delay:

1 Control-click (Windows) or Command-click (Macintosh) the Track Level Indicator to toggle between level (that appears on the display as "vol"), headroom ("pk"), and channel delay ("dly") indications. Delay values are shown in samples.

Determining the DSP delay of track inserts (Mix window shown)

2 Apply the TimeAdjuster plug-in to the track whose delay you want to increase, and Control-click (Windows) or Command-click (Macintosh) its Track Level indicator until the channel delay value is displayed for that track.

3 Change the delay time in TimeAdjuster by moving the Delay slider or entering a value in the Delay field, until the channel delay value matches that of the first track.

4 Test the delay values by duplicating an audio track and reversing its phase while compensating for delay.

When to Compensate for Delays

If you want to compensate for delays across your entire system with Time Adjuster, you will want to calculate the maximum delay incurred on any channel, and apply the delays necessary to each channel to match this channel.

However, this may not always be necessary. You may only really need to compensate for delays between tracks where phase coherency must be maintained (as with instruments recorded with multiple microphones or stereo pairs). If you are working with mono signals, and the accumulated delays are small (just a few samples, for example), you probably needn't worry about delay compensation.

📄 *For more information about delays and mixing with Pro Tools HD, see the Pro Tools Reference Guide.*

Trim

The Trim plug-in can be used to attenuate an audio signal from –∞ (Infinity) dB to +6 dB or –∞ (Infinity) dB to +12 dB. For example, using a multi-mono Trim plug-in on a multi-channel track provides simple, DSP-efficient muting control over the individual channels of the track. This capability is useful, since Track Mute buttons mute *all* channels of a multi-channel track and do not allow muting of individual channels within the track.
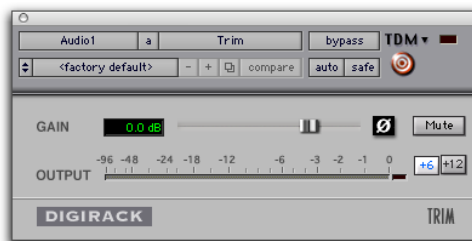
Trim plug-in

💡 *Alt-click (Windows) or Option-click (Macintosh) the Trim selector to open a Plug-In window for each channel of a multi-channel track.*

Trim Controls

Phase Invert Inverts the phase (polarity) of the input signal to change the frequency response characteristics between multi-miked sources or to correct for miswired microphone cables.

Gain Provides –∞ dB to +6 dB or +12 dB of gain adjustment, depending whether the Gain toggle is set to +6 or +12.

+6/+12 Gain toggle Switches the maximum level of attenuation between –∞ dB to +6 dB and –∞ dB to +12 dB.

⚠ *Automation data adjusts to reflect the current Gain setting. When working with automation data from an older version of the Trim plug-in, ensure the Gain setting is set at +6 dB.*

Output Meter Indicates the output level, including any gain compensation added using the Gain control.

Mute Mutes the signal output.

📄 *See also "Linking and Unlinking Controls on Multi-Mono Plug-ins" on page 19.*

DigiRack AudioSuite Plug-Ins

EQ III

The EQ III Plug-In provides a high-quality 7 Band, 2–4 Band, or 1 Band EQ for adjusting the frequency spectrum of audio material.

The EQ III Plug-In appears as two separate choices in the plug-in the AudioSuite menu: 7 Band, or 1 Band. (2–4 Band is not available in AudioSuite format; use the 7 Band instead).

7 Band EQ and 2–4 Band EQ

The 7 Band EQ and the 2–4 Band EQ have the same window and identical controls, but on the 2–4 Band EQ, a limited number of the seven available bands can be active at the same time.

7 Band EQ and 2–4 Band EQ window

1 Band EQ

The 1 Band EQ has its own window, with six selectable filter types.

1 Band EQ window

The features of these plug-ins are non-real-time, but otherwise identical to their real-time counterparts.

📄 *For information on the individual controls for these plug-ins, see "EQ III" on page 41*

EQ II

For information, see the electronic PDF guide for the EQ II plug-ins.

Dynamics II

There are five different AudioSuite dynamics plug-ins. They include:

- Compressor II
- Limiter II
- Gate II
- Expander/Gate II
- DeEsser

The features of these plug-ins are non-real-time, but otherwise identical to their real-time counterparts.

📄 *For information on the individual controls for these plug-ins, see "Dynamics II" on page 61.*
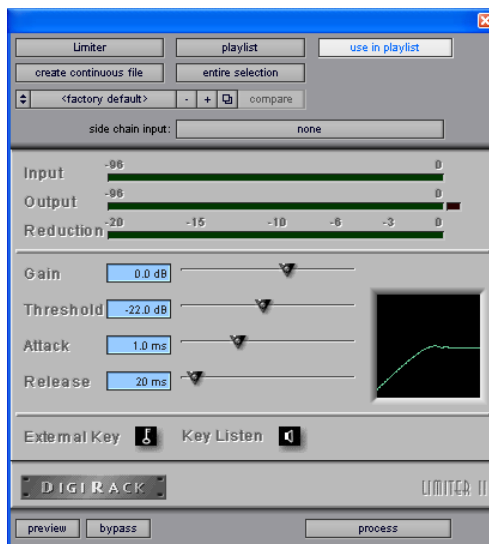
Limiter II AudioSuite plug-in
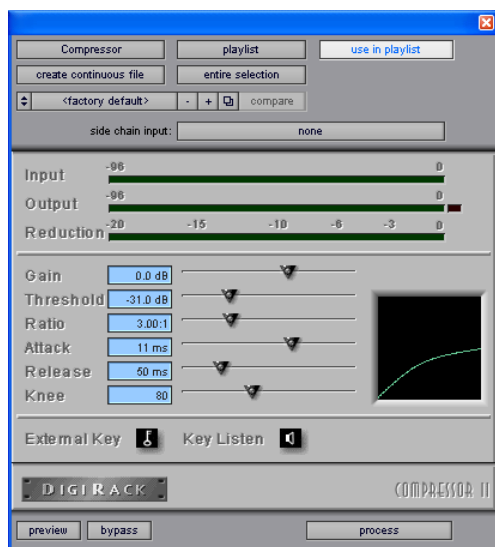
Compressor II AudioSuite plug-in
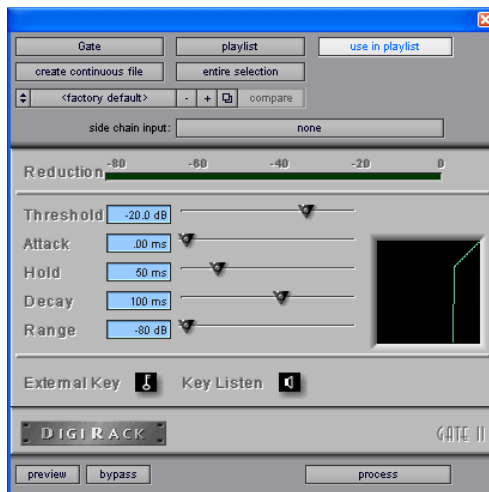
Gate II AudioSuite plug-in
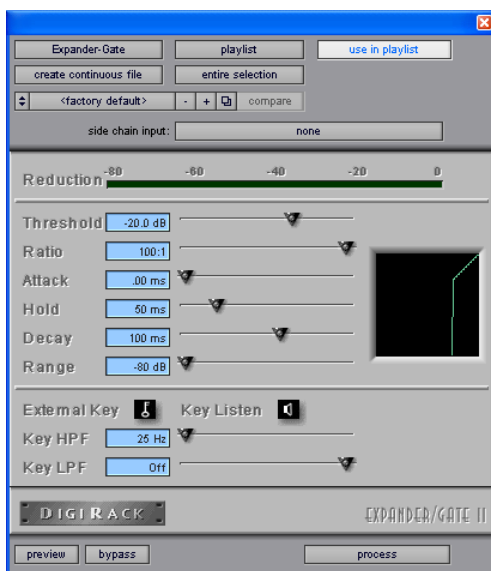
Expander/Gate II AudioSuite plug-in
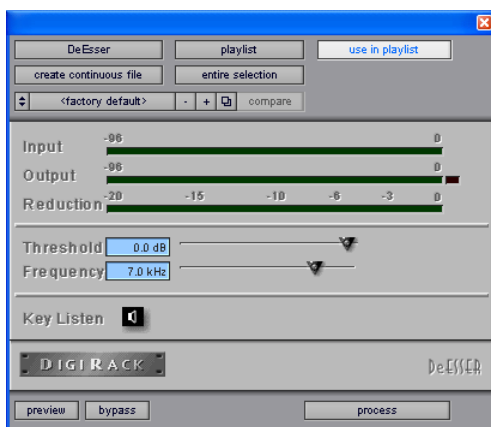
DeEsser AudioSuite plug-in

Chorus

Chorus adds a shimmering quality to audio material by combining a time-delayed, pitch-shifted copy of an audio signal with itself.

💡 *The Chorus plug-in was formerly called D-fx Chorus. It is fully compatible with all settings and presets created for D-fx Chorus.*

Chorus plug-in

Gain Adjusts the input volume of the chorus to prevent clipping or increase the level of the processed signal. This slider is set to a default of +3 dB. If your source audio has been recorded very close to peak level, this +3 dB default setting could cause clipping. Use this control to reduce the input level.

Selecting the Sum Input button sums the dry input signals (mono or stereo) before processing them. The dry signal then appears in the center of the stereo field and the wet, effected signal will be output in stereo.

When the Sum Inputs button is selected, the LFO waveform on the right channel is automatically phase inverted to enhance the mono-stereo effect.

Sum Inputs button

Mix Adjusts the balance between the effected signal and the original signal and controls the depth of the effect. Mix is adjustable from 0% to 100%.

Low-Pass Filter Controls the cutoff frequency of the Low-Pass Filter. Use this to attenuate the high frequency content of the feedback signal. The lower the setting, the more high frequencies are removed from the feedback signal.

The range of the Low-Pass Filter is 20 Hz to 19.86 kHz, with a maximum value of Off (which effectively means bypass).

Delay Sets the delay time between the original signal and the chorused signal. The higher the setting, the longer the delay and the wider the chorusing effect. Delay is adjustable from 0–20 milliseconds.

LFO Rate Adjusts the rate of the LFO (low frequency oscillator) applied to the delayed signal as modulation. The higher the setting, the more rapid the modulation. You can select either a sine wave or a triangle wave as a modulation source, using the LFO Waveform selector.

LFO Width Adjusts the intensity of the LFO applied to the delayed signal as modulation. The higher the setting, the more intense the modulation. Use the LFO Waveform selector to select a sine or a triangle wave as a modulation source.

Feedback Controls the amount of feedback applied from the output of the delayed signal back into its input. Negative settings provide a more intense effect.

LFO Waveform Selects a sine wave or triangle wave for the LFO. This affects the character of the modulation. The sine wave has a gentler ramp and peak than the triangle wave.

Flanger

The Flanger animates and adds a swirling, moving quality to audio material by combing a time-delayed copy of an audio signal with itself.

The Flanger uses a *through-zero* flanging algorithm that results in a tape-like flanging effect. This technique delays the original dry signal by 256 samples, then modulates the delayed signal back and forth in time in relation to the dry signal, passing through its zero point on the way.

💡 *The Flanger plug-in was formerly called D-fx Flanger. It is fully compatible with all settings and presets created for D-fx Flanger.*
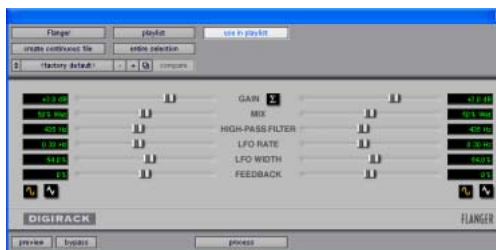
Flanger plug-in

Gain Adjusts the input volume of the flanger to prevent clipping or increase the level of the processed signal. This slider is set to a default of +3 dB. If your source audio has been recorded very close to peak level, this +3 dB default setting could cause clipping. Use this control to reduce the input level.

Selecting the Sum Input button sums the dry input signals (mono and stereo) before processing them. The dry signal then appears in the center of the stereo field and the wet, effected signal will be output in stereo.

When the Sum Inputs button is selected, the LFO waveform on the right channel is phase inverted to enhance the mono-stereo effect.

Mix Adjusts the balance between the effected signal and the original signal and controls the depth of the effect. Mix is adjustable from 0% to 100%.

High-Pass Filter Controls the cutoff frequency of the High-Pass Filter. Use this to attenuate the frequency content of the feedback signal and the frequency response of the flanging. The higher the setting, the more low frequencies are removed from the feedback signal.

LFO Rate Adjusts the rate of the LFO (low frequency oscillator) applied to the delayed signal as modulation. The higher the setting, the more rapid the modulation. You can select either a sine wave or a triangle wave as a modulation source, using the LFO Waveform selector.

LFO Width Adjusts the intensity of the LFO applied to the delayed signal as modulation. The higher the setting, the more intense the modulation.

Feedback Controls the amount of feedback applied from the output of the delayed signal back into its input. Negative settings provide a more intense effect.

LFO Waveform Selects a sine wave or triangle wave for the LFO. This affects the character of the modulation. The sine wave has a gentler ramp and peak than the triangle wave.

Multi-Tap Delay

The Multi-Tap Delay adds up to four independently-controllable delays or *taps* to the original audio signal. Use the Multi-tap delay to add spatialization or complex rhythmic echo effects to audio material. You can individually control the delay time and number of repetitions of each of the four taps.

☼ *The Multi-Tap Delay plug-in was formerly called D-fx Multi-Tap Delay. It is fully compatible with all settings and presets created for D-fx Multi-Tap Delay.*
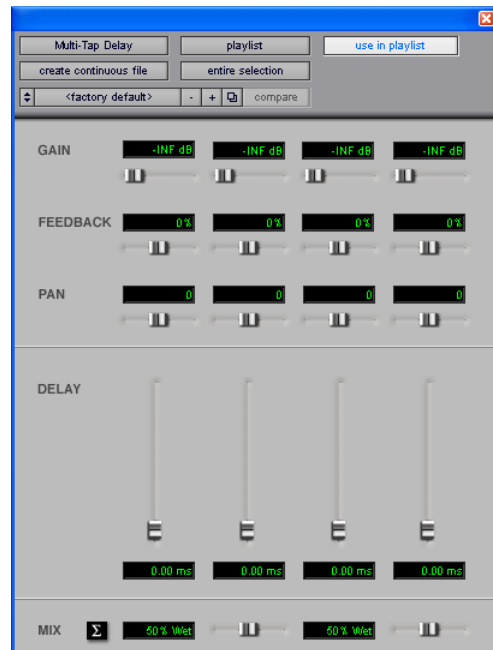
Multi-Tap Delay plug-in

Gain The Gain slider for each of the Multi-Tap delay's four delay lines controls the input level of that individual delay tap. Adjust Gain to prevent clipping or increase the level of the processed signal.

Selecting the Sum Input button sums the dry input signals (mono or stereo) before processing them. The dry signal then appears in the center of the stereo field and the wet, effected signal will be output in stereo.

Feedback Controls the amount of feedback applied from the output of the delay into its input. It also controls the number of repetitions of the delayed signal. For the feedback feature to function, the Gain slider must be raised above its lowest setting.

Pan The pan sliders for each of the four taps control their apparent location in the stereo field.

Delay Sets the delay time between the original signal and the delayed signal. The higher the setting, the longer the delay. This control is adjustable from 0–1500 milliseconds (1.5 seconds).

Mix Adjusts the balance between the effected signal and the original signal and controls the depth of the effect. Mix is adjustable from 0% to 100%.

Selecting Audio for Processing with AudioSuite Delay

Because Multi-Tap Delay adds additional material to the end of selected audio (a delay tap), make a selection that is longer than the original source material so AudioSuite Delay can write it into the audio file.

Select only the original material, without leaving additional space at the end, will produce a delay that occurs after the end of the region will be cut off.

To accommodate 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 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.

Ping-Pong Delay

The Ping-Pong Delay plug-in adds a controllable delay to the original audio signal. Use the Ping-Pong delay to add spatialization, and panned echo to audio material. This plug-in feeds back delayed signals to their opposite channels, creating a characteristic ping-pong echo effect.

The Ping-Pong Delay plug-in was formerly called D-fx Ping-Pong Delay. It is fully compatible with all settings and presets created for D-fx Ping-Pong Delay.

Ping-Pong Delay plug-in

Gain Adjusts the input volume of the Ping-Pong Delay to prevent clipping or increase the level of the processed signal.

Mix Adjusts the balance between the effected signal and the original signal and controls the depth of the effect. Mix is adjustable from 0% to 100%.

Delay Sets the delay time between the original signal and the delayed signal. The higher the setting, the longer the delay. This control is adjustable from 0–1500 milliseconds (1.5 seconds).

Low-Pass Filter Controls the cutoff frequency of the Low-Pass Filter. Use this to attenuate the high frequency content of the feedback signal. The lower the setting, the more high frequencies are removed from the feedback signal.

The range of the Low-Pass Filter is 20 Hz to 19.86 kHz, with a maximum value of Off (which effectively means bypass).

Feedback Controls the amount of feedback applied from the output of the delay into its input. It also controls the number of repetitions of the delayed signal.

Cross-Feedback Cross-Feedback feeds the delayed signals to their opposite channel: The left channel delay is fed to the right channel input and vice-versa. The result is a stereo echo that ping-pongs back and forth between the right and left channels.

Selecting Audio for Delay Processing

Because Ping-Pong Delay adds additional material (a delay tap) to the end of selected audio, make a selection that is longer than the original source material to allow the delay tap to be written into the end of the audio file.

Selecting only the original material, without leaving additional space at the end, will cause delay taps that occur after the end of the region to be cut off.

Invert

The Invert plug-in reverses the polarity of selected audio. Positive sample amplitude values are made negative, and all negative amplitudes are made positive.

This process is useful for altering the phase or *polarity* relationship of tracks. The Invert plug-in is useful during mixing for modifying frequency response between source tracks recorded with multiple microphones. You can also use it to correct audio recorded out of phase with an incorrectly wired cable.
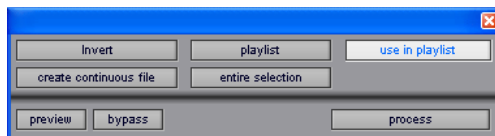
Invert plug-in

Duplicate

The Duplicate plug-in duplicates the selected audio in place. Depending on how its controls are configured, the new region will appear in either the Regions List or playlist. You can use this to flatten or consolidate an entire track consisting of multiple regions into one continuous audio file that resides in the same place as the original individual regions.
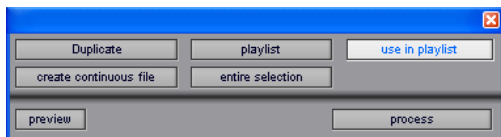
Duplicate plug-in

The audio is unaffected by Pro Tools volume or pan automation, or by any real-time plug-ins that may be in use on the track as inserts. The original audio file regions are merely rewritten in place to a single duplicate file.

The Duplicate plug-in works nondestructively. You cannot choose to overwrite files.

Delay

The Delay plug-in provides time delay-based effects, with up to 10.9 seconds of delay, at any session sample rate. You can also use it to produce multiple delay-based effects, including slap echo, doubling, chorusing, and flanging.

Delay AudioSuite plug-in

The controls of the AudioSuite Delay plug-in operate in non-real-time, but are otherwise identical to the real-time Mod Delay II plug-in.

📖 *For information on the individual controls for the AudioSuite Delay plug-in, see "Mod Delay II" on page 68.*

If you have the Mod Delay Plug-In installed, Short Delay, Slap Delay, Medium Delay, and Long Delay will be available in the AudioSuite menu in addition to Delay (Mod Delay II). The features of these plug-ins are identical to their real-time counterparts. The only difference is that the AudioSuite versions operate in non-real time.

📖 *For information on the individual controls for the AudioSuite Short Delay, Slap Delay, Medium Delay, and Long Delay plug-ins, see "Mod Delay II" on page 68.*

Selecting Audio Delay Processing

Because AudioSuite Delay adds additional material (a delay tap) to the end of selected audio, make a selection that is longer than the original source material to allow the delay tap to be written into the end of the audio file.

Selecting only the original material, without leaving additional space at the end, will cause delay taps that occur after the end of the region to be cut off.

Normalize

The Normalize plug-in optimizes the volume level of an audio selection. Use it on material recorded with too little amplitude, or on material whose volume levels are inconsistent (as in a poorly recorded narration).

Unlike compression and limiting, which modify the dynamics of audio material, normalization preserves dynamics by uniformly increasing (or decreasing) amplitude.

💡 *To prevent clipping during sample rate conversion, Normalize to no greater than the range between –2 dB to –0.5 dB. Optimum settings will vary with your program material and your Conversion Quality setting (in the Editing tab of the Preferences dialog).*
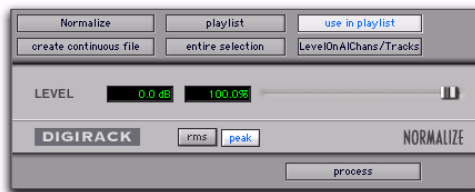
Normalize plug-in

Max Peak At Specifies how close to maximum level (clipping threshold) the peak level of a selection is boosted. Set this value by adjusting the Max Peak At slider, by entering a numeric decibel value below the clipping threshold, or by entering a percentage of the clipping threshold.

You can normalize stereo pairs together so that two sides of a stereo signal are processed relative to each other.

RMS/Peak Toggle Switches the calibration of normalizing between Peak or RMS modes. Peak normalizes the input signal at the maximum possible level without clipping. RMS normalizes the input signal at a level consistent with the RMS (Root-Mean-Square) value, or the effective average level of the selected region.

Normalizing Multiple Regions Across Tracks

When multiple regions are selected across multiple tracks, the Normalize plug-in can search for peaks in two different modes:

Peak On Each Chan/Track Searches for the peak level on a channel-by-channel or track-by-track basis.

Peak On All Chans/Tracks Searches for the peak level of the entire selection. If ten tracks are selected, for example, the Normalize function will find the peak value from all ten.

Gain

The Gain plug-in boosts or lowers a selected region's amplitude by a specific amount. Use it to smooth out undesired peaks and other dynamic inconsistencies in audio material.
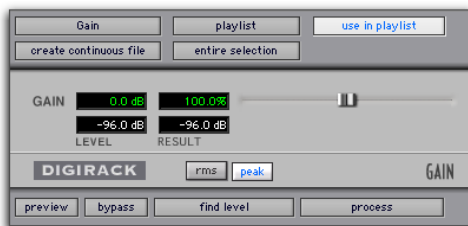
Gain plug-in

Gain Specifies the desired gain level. Set this value by manually adjusting the Gain slider, or by entering a numeric decibel value, or by entering a percentage.

Find Level When clicked, displays the peak amplitude value of the current selection.

RMS/Peak Toggle Switches the calibration of gain adjustment between Peak or RMS modes. Peak adjusts the gain of the input signal to the maximum possible level without clipping. RMS adjusts the input signal to a level consistent with the RMS (Root-Mean-Square) value, or the effective average level of the selected region.

Reverse

The Reverse plug-in replaces the audio with a reversed version of the selection.
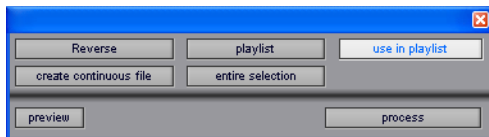
Reverse plug-in

Signal Generator

The Signal Generator plug-in produces audio test tones in a variety of frequencies, waveforms, and amplitudes.

Signal Generator plug-in

The features of this plug-in are non-real-time, but otherwise identical to its real-time counterpart.

☼ *You can use the AudioSuite Signal Generator plug-in for musical purposes as well as for testing purposes. For example, you might want to add a little color to a kick drum track by doubling it with a 50 Hz tone, using the kick track as the key input signal gating the tone track.*

To create an audio region using the Signal Generator plug-in:

1 Make a selection in the Tracks window.

2 Choose AudioSuite > Signal Generator.

3 Enter values for the Frequency, Level, and Signal controls.

4 Click Process in the Signal Generator plug-in.

💡 *Select the Create Continuous File option for greater flexibility in making audio selections for use with the Signal Generator plug-in.*

📖 *For information on the individual controls for the Signal Generator plug-in, see "Signal Generator" on page 70.*

DC Offset Removal

The DC Offset Removal plug-in removes DC offset from audio files. DC offset is a type of audio artifact (typically caused by miscalibrated analog-to-digital convertors) that can cause pops and clicks in edited material.

To check for DC offset, find a silent section in the audio material. If DC offset is present, a near-vertical fade-in with a constant or steady-state offset from zero will appear in the waveform. Use the DC Offset Removal plug-in to remove it.
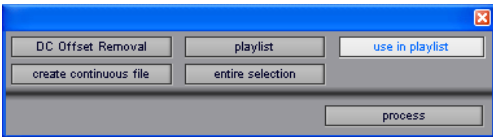
DC Offset Removal plug-in

Time Compression/ Expansion

The Time Compression/Expansion plug-in adjusts the duration of selected regions, increasing or decreasing their length without changing pitch.
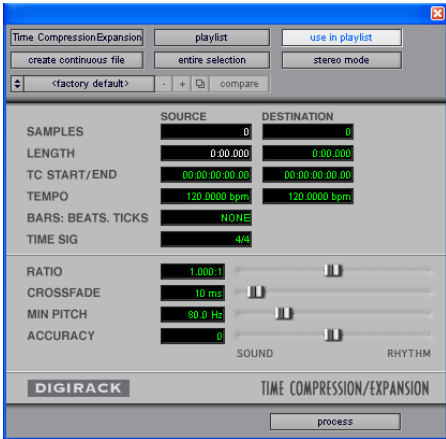
Time Compression/Expansion plug-in

It is especially useful in audio post-production for adjusting audio to specific time or SMPTE durations for synchronization purposes. Time Compression/Expansion is nondestructive.

💡 *Normalizing a selection before using Time Compressing/Expansion may produce better results.*

Source and Destination The Source fields display the length of the current selection before processing in each of the listed formats. All fields are always active; a change made to one value is immediately reflected in the others.

The Destination fields both display and control the final length of the selection after processing. Enter the length of the Destination file by double-clicking the appropriate field in the Destination column.

Use the Ratio, Crossfade, Min Pitch, and Accuracy controls to fine-tune the Time Compression/Expansion process.

Ratio Sets the destination length in relation to the source length. Moving the slider to the right increases the length of the destination file, while moving the slider to the left decreases its length.

Crossfade Adjusts the crossfade length in milliseconds, optimizing performance of the Time Compression/Expansion according to the type of audio material being processed. (This plug-in achieves length modification by replicating or subtracting very small portions of audio material and very quickly crossfading between these alterations in the waveform of the audio material.)

Crossfade length affects the amount of smoothing performed on audio material. This prevents audio artifacts such as clicks from occurring. Long crossfade times may over-smooth a signal and its transients. This may not be desirable on drums and other material with sharp transients.

Use the Crossfade slider to manually adjust and optimize crossfade times if necessary. For audio material with sharper attack transients, use smaller crossfade times. For audio material with softer attack transients, use longer crossfade times.

Min Pitch Sets the minimum, or lowest, pitch that will be used in the plug-in's calculations during the Time Compression/Expansion process. The control has a range of 40 Hz to 1000 Hz.

This control should be set lower when processing bass guitar or audio material with a low frequency range. Set this control higher when processing higher frequency range audio material.

Accuracy Prioritizes the processing resources allocated to audio quality (Sound) or timing (Rhythm). Moving the slider towards "Sound" generally results in better sonic quality and fewer audio artifacts. Moving the slider towards "Rhythm" puts the emphasis on keeping the tempo consistent.

When you are working with audio loops, listen carefully and adjust the Accuracy slider until you find a setting that keeps timing solid within the region. If you don't, start and end times may be precise, but the beats in rhythmic material may appear to be shuffled if too little priority is given to Rhythm.

⚠ *Time Compression & Expansion settings created in version 4.x and later of Pro Tools for Windows are not compatible with later versions.*

Pitch Shift

The Pitch Shift plug-in adjusts the pitch of any source audio file with or without a change in its duration. This is a very powerful function that transposes audio a full octave up or down in pitch with or without altering playback speed.
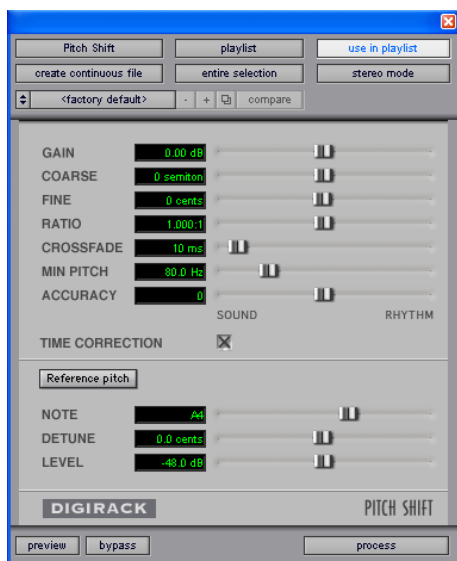
Pitch Shift plug-in

Gain Adjusts input level, in 10ths of a dB. Dragging the slider to the right increases gain, dragging to the left decreases gain.

Coarse and Fine Adjusts amount of pitch shift. The Coarse slider transposes in semitones (half steps). The Fine slider transposes in cents (hundredths of a semitone).

Time Correction Disabling this option has the effect of "permanently varispeeding" your audio file. The file's duration will be compressed or extended according to the settings of the Coarse and Fine pitch controls. When Time Correction is enabled, fidelity can be affected. For example, time expansion as a result of Time Correction when lowering pitch can cause the audio to sound granulated.

Ratio Adjusts the amount of transposition (pitch change). Moving the slider to the right raises the pitch of the processed file, while moving the slider to the left decreases its pitch.

Crossfade Use this to manually adjust crossfade length in milliseconds to optimize performance of the Pitch Shift plug-in according to the type of audio material you are processing. This plug-in achieves pitch transposition by processing very small portions of the selected audio material and very quickly crossfading between these alterations in the waveform of the audio material.

Crossfade length affects the amount of smoothing performed on audio material. This prevents audio artifacts such as clicks from occurring. In general, smaller pitch transpositions require longer crossfades; wider pitch transpositions require smaller crossfades. Long crossfade times may over-smooth a signal and its transients. This is may not be desirable on drums and other material with sharp transients.

Use the Crossfade slider to adjust and optimize crossfade times. For audio material with sharper attack transients, use smaller crossfade times. For audio material with softer attack transients, use longer crossfade times.

Min Pitch Sets the lowest pitch used in the plug-in's Pitch Shift processing. The control has a range of 40 Hz to 1000 Hz. Use it to focus the Pitch Shift process according to the audio's spectral shape.

Use lower values when processing lower frequency audio material. Use higher values when processing higher frequency audio material.

Accuracy Sets the processing resources allocated to audio quality (Sound) or timing (Rhythm). Set the slider toward Sound for better audio quality and fewer audio artifacts. Set the slider toward Rhythm for a more consistent tempo.

Reference Pitch Activates a sine wave-based pitch generator that you can use as an audible reference when pitch-shifting audio material.

Note Adjusts the frequency of the reference tone in semitones (half steps).

Detune Provides finer adjustment of the frequency of the reference tone in cents (100ths of a semitone).

Gain Adjusts the volume of the reference tone in dB.

To use the Reference Pitch:

1 Select the audio material you want to use as a pitch reference. Click the preview button to begin playback of the selected audio.

2 Click the Reference Pitch button to activate the reference sine wave tone.

3 Adjust the Note and Detune settings to match the reference tone to the pitch of the audio playback. Adjust the Gain setting to change the relative volume of the reference tone. It may also be helpful to toggle the Reference Pitch on and off to compare pitch.

4 Select the audio material to be pitch shifted.

5 Adjust the Coarse and Fine Pitch Shift controls to match the pitch of the audio playback to the reference pitch.

6 Click Process to apply pitch shift to the selection.

chapter 7

Using Digi ReWire

Pro Tools supports ReWire version 2.0 technology developed by Propellerheads Software. ReWire is available as a DigiRack RTAS plug-in.

ReWire provides real-time audio and MIDI streaming between applications, with sample-accurate synchronization and common transport functionality.

ReWire RTAS plug-in

Using ReWire, Pro Tools can send MIDI to a ReWire client application, such as software synthesizer and receive audio back from the ReWire client.

⚠ *Pro Tools does not support receiving MIDI from ReWire client applications.*

Compatible ReWire client applications are automatically detected by Pro Tools and are available in the RTAS Plug-Ins Insert menus in Pro Tools. Selecting a ReWire client application within Pro Tools will automatically launch that application (if the client application supports this feature). Any corresponding MIDI nodes for that application will be available in the MIDI Track Output selector.

Once the outputs of your software synthesizers and samplers are routed to Pro Tools, you can:

- Process incoming audio signals with plug-ins
- Automate volume, pan, and plug-in controls
- Bounce To Disk
- Take advantage of the audio outputs of your Digidesign audio interfaces

⚠ *Exchange of additional metadata such as controller and note names between Pro Tools and ReWire clients is not supported.*

ReWire Requirements

To use the ReWire plug-in, you will need:

- A Digidesign-qualified Windows or Macintosh computer running Pro Tools HD or Pro Tools LE software version 7.0 or higher

- ReWire-compatible client software, such as:
  - *Reason* from Propellerheads Software
  - *Live* from Ableton

⚠️ *Client software must support the same sample rate as the session using ReWire. For example, third-party client software that does not support sample rates beyond 48 kHz cannot be used in 96 kHz Pro Tools sessions.*

ReWire support is also under development for other third-party companies. For availability, check with the manufacturer or the Digidesign Web site (www.digidesign.com).

ReWire and Track Count with Pro Tools HD

With Pro Tools HD, the ReWire RTAS plug-in can be inserted on any kind of track. Each channel of audio transmitted through ReWire then uses the same amount of resources as the audio track on which it is inserted.

Consequently, you can only use a total combination of audio tracks and ReWire audio streams that does not exceed the maximum number of possible voices for your system. For example, if you are playing 40 audio tracks in a 48 kHz/24-bit session on a 128-voice Pro Tools|HD 2 core system, that will leave 88 channels of audio (88 mono, or 44 stereo) that can be used with ReWire. (ReWire supports a maximum of 64 audio streams per application.)

Using ReWire at higher sample rates will increase the load on the CPU. For example, CPU load at 96 kHz is double the load at 48 kHz. You can monitor Pro Tools CPU usage in the System Usage window, making sure to not overtax your system.

💡 *With Pro Tools HD, the standard Hardware Buffer size of 512 samples is strongly recommended for using ReWire in sessions with sample rates above 48 kHz.*

ReWire and Track Count on Pro Tools LE Systems

With Pro Tools LE, performance is determined by several factors, including host CPU speed, available memory, and buffer settings. Digidesign cannot guarantee 64 simultaneous audio channel outputs with ReWire on all computer configurations.

For the latest information on recommended CPUs and system configurations, check the Digidesign Web site (www.digidesign.com/compato).

Using ReWire

The ReWire plug-in is installed when you install Pro Tools. All inter-application communications between Pro Tools and ReWire client software is handled automatically.

To use a ReWire client application with Pro Tools:

1 Make sure that the ReWire client application is installed properly and that you have restarted your computer.

2 In Pro Tools, choose Track > New and specify one Instrument track (or audio or Auxiliary Input track), and click Create.

3 In the Mix window, click the Insert selector on the track and assign the ReWire RTAS client plug-in to the track insert.

The ReWire client application will launch automatically in the background (if the client applications supports auto-launch). If the client application does not support auto-launch, launch it manually.

4 Configure the ReWire client application to play the sounds you want.

5 In Pro Tools, set the output of the client application in the ReWire plug-in window.

⚠ *Some ReWire client applications, such as Reason, support a single stereo output path (Reason's remaining 62 outputs are mono only). If you want to use multiple stereo outputs with Reason you will need to insert multiple multi-mono ReWire plug-ins, then unlink and assign left and right outputs separately using the Link Enable and Channel selector buttons in the Plug-In window.*

6 In the Mix window, click the track's MIDI Output selector a and select the ReWire client application. Some ReWire clients (such as Reason) may list multiple devices. If so, choose the device that you want.

7 Choose Options > MIDI Thru and record enable the MIDI track. Play some notes on your MIDI controller to trigger the client application.

8 If your ReWire client application is a sequencer and you want to begin synchronized playback with Pro Tools, press the Spacebar or click the Play button on the Pro Tools Transport.

⚠ *If you experience system performance problems while using Pro Tools with ReWire client applications, you may need to increase the Pro Tools CPU Usage Limit. See "CPU Usage Limit" on page 9 for instructions.*

Quitting ReWire Client Applications

When quitting Pro Tools sessions that integrate ReWire client applications, quit the client application first, then quit Pro Tools.

⚠ *If you quit Pro Tools before quitting ReWire client applications, a warning dialog box may appear stating that "one or more ReWire applications did not terminate." To avoid this, quit all ReWire client applications before quitting Pro Tools.*

Tempo and Meter Changes

Pro Tools transmits both Tempo and Meter data to ReWire client applications, allowing ReWire-compatible sequencers to follow any tempo and meter changes in a Pro Tools session.

With the Pro Tools Conductor button selected, Pro Tools always acts as the Tempo master, using the tempo map defined in its Tempo Ruler.

With the Pro Tools Conductor button deselected, the ReWire client acts as the Tempo master. In both cases, playback can be started or stopped in either application.

⚠ *Pro Tools supports tempo values from 30–300 bpm. When slaved to a ReWire client application, Pro Tools playback will be restricted to this range even if the client application's tempo is outside this range. Additionally, some ReWire client applications (such as Reason) may misinterpret Pro Tools meter changes, resulting in mismatched locate points and other unexpected behavior. To prevent this, avoid using meter changes in Pro Tools when using Reason as a ReWire client.*

Looping Playback

Because Pro Tools does not offer separate loop markers as found in other third party applications such as Reason, if you want to loop playback, do one of the following:

To loop playback in Pro Tools:

1 In the Pro Tools Time Line, select the time range that you want to loop.

2 Begin playback by pressing the Spacebar or clicking the Transport's Play button.

To loop playback within a ReWire client sequencer

■ With playback stopped, specify the loop within the ReWire client application and begin playback.

⚠ *If you create a playback loop by making a selection in the Pro Tools Time Line, once playback is started, any changes made to loop or playback markers within the ReWire client application will deselect the Pro Tools Time Line selection and remove the loop.*

Automating ReWire

ReWire supports automation for switching inputs during playback.

To automate switching inputs during playback:

1 Set the track's automation to write.

2 Change the input link pop-up menu manually.

– or –

Draw the automation in the Edit window.

📄 *For more information on drawing automation, see the Pro Tools Reference Guide.*

appendix a

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 is not limited by CPU processing power.

The DSP tables on the following pages show the theoretical total number of instances of each DigiRack TDM plug-in that can be powered by a single DSP chip on Pro Tools|HD-series cards. DSP usage differs according to card type and DSP chip availability.

⚠ *DSP tables show the theoretical maximum performance when no other plug-ins or system tasks (such as I/O) are sharing available DSP resources. You will typically use more than one type of plug-in simultaneously. The data in these tables are provided as guidelines to help you gauge the relative efficiency of different plug-ins on your system. They are not guaranteed performance counts that you should expect to see in typical real-world sessions and usage.*

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).

Not all plug-ins are supported on all types of chips. The following tables indicate the number of compatible chips per card.

DSP Requirements

DSP requirements for DigiRack TDM plug-ins are provided on the following pages.

Mono and Stereo DigiRack Plug-Ins on HD Cards

The following table compares the maximum number of instances of each DigiRack TDM plug-in that can be powered by a single DSP chip on an HD Accel card to that of HD Core and HD Process cards.

Maximum instances of real-time TDM plug-ins per DSP chip for Pro Tools|HD cards at 44.1 and 48 kHz

Plug-In	HD Accel		HD Core and HD Process	
	mono/stereo		mono/stereo	
EQ III (1-band)	66	33	28	14
EQ III (2–4 band)	36	18	15	7
EQ III (7-band)	23	11	10	5
Compressor II	30	20	26	17
Click	18	n/a	3	n/a
De-esser	46	n/a	20	n/a
Dither	46	43	34	18
D-Verb	6	6	4	4
Limiter II	60	40	16	17
Gate II	106	51	59	20
Expander/Gate II	54	46	24	20
Mod Delay II (short)	74	36	43	21
Mod Delay II (slap)	73	36	32	16
Mod Delay II (medium)	36	18	16	8
Mod Delay II (long)	17	7	8	4
Mod Delay II (extra long)	4	2	2	1
Pitch	1	1	1	1
POW-r Dither	22	10	9	4
Time Adjuster (short)	146	72	117	58
Time Adjuster (medium)	32	16	114	56
Time Adjuster (long)	72	36	32	16

Maximum instances of real-time TDM plug-ins per DSP chip for Pro Tools|HD cards at 44.1 and 48 kHz

Plug-In	HD Accel		HD Core and HD Process	
	mono/stereo		mono/stereo	
Trim	145	n/a	119	n/a
Signal Generator	48	n/a	25	n/a

Maximum instances are approximately half at 88.2 kHz and 96 kHz, and approximately a quarter at 176.4 and 192 kHz. Actual results will likely be less due to system and session variables.

Multi-Mono and Multi-Channel DigiRack Plug-Ins

Plug-ins used in multi-mono format on greater-than-stereo tracks require one mono instance per channel of the multi-channel audio format. A multi-mono EQ used on a 5.1 format track, for example, requires six mono instances since there are six audio channels in the 5.1 format. Because multi-channel versions of the Compressor II, Limiter II, and Expander/Gate II plug-ins are available for use on greater-than-stereo tracks, separate instance counts are given for those configurations.

Table 2. Theoretical maximum instances of real-time TDM plug-ins per DSP chip for a Pro Tools HD Core and HD Process card at 48 kHz**

Plug-In	LCR	Quad & LCRS	5.0	5.1 & 6.0	6.1 & 7.0	7.1	DSP chips per HD card
Compressor II	11	9	8	6	6	6	9
De-esser*	6	5	4	3	2	2	9
DigiRack Pitch	partial	partial	partial	partial	partial	partial	9
D-Verb	partial	partial	partial	partial	partial	partial	9
Limiter II	11	9	8	6	6	6	9
Gate II	19	14	11	9	8	7	9
Expander/Gate II	15	13	12	11	11	10	9
Mod Delay I (short)	14	11	8	7	6	5	9
Mod Delay I (slap)	10	8	6	5	4	4	9
Mod Delay I (medium)	5	4	3	2	2	2	9
Mod Delay I (long)	partial	partial	partial	partial	partial	partial	9

Table 2. (Continued) Theoretical maximum instances of real-time TDM plug-ins per DSP chip for a Pro Tools HD Core and HD Process card at 48 kHz**

Plug-In	LCR	Quad & LCRS	5.0	5.1 & 6.0	6.1 & 7.0	7.1	DSP chips per HD card
Mod Delay II (short)	14	11	8	7	6	5	9
Mod Delay II (slap)	10	8	6	5	4	4	9
Mod Delay II (medium)	5	4	3	2	2	2	9
Mod Delay II (long)	2	2	1	1	1	1	9
Mod Delay II (extra long)	partial	partial	partial	partial	partial	partial	9
Dither	12	9	7	6	5	4	9
POW-r Dither	3	2	1	1	2	1	9
Time Adjuster (short)	30	22	18	15	13	11	9
Time Adjuster (medium)	30	22	18	15	13	11	9
Time Adjuster (long)	10	8	6	5	4	4	9
Trim*	39	29	23	19	16	14	9
Signal Generator*	8	6	5	4	3	3	9

"Partial" indicates that a single instance of the multi-mono plug-in is sharing more than 1 DSP chip.

* Instance counts are provided for multi-mono versions of plug-ins that have no available multichannel version.

** Maximum instances are approximately half at 88.2 kHz and 96 kHz, and approximately a quarter at 176.4 and 192 kHz. Actual results will likely be less due to system and session variables.

Monitoring DSP Usage

The System Usage window (Window > System Usage) shows approximately how much DSP is available in your system and how it is being used in the current Pro Tools session.

For more information about DSP usage and allocation, see the Pro Tools Reference Guide.

appendix b

DSP Delays Incurred by TDM Plug-Ins

Virtually all TDM plug-ins incur some amount of signal delay.

If you are working with mono tracks, or are processing all channels with the same plug-in, the signal delays are not long enough to be significant and should not be a concern.

This signal delay is significant only if you use a plug-in on one channel of a stereo or multi-channel signal but not the others, since this can cause the channels to be slightly out of phase.

Pro Tools|HD systems provide automatic Delay Compensation to compensate for signal processing delays. For detailed information, see the Pro Tools Reference Guide.

Table 3 on page 100 shows the delays inherent in each type of Digidesign TDM plug-in.

Table 3. Samples of delay incurred by each real-time DigiRack TDM plug-in on HD cards

Plug-In	Samples of delay on HD cards
EQ III (1-band)	4
EQ III (2–4 band)	4
EQ III (7-band)	4
Compressor II	4
De-esser	4
DigiRack Pitch	3
D-Verb	3
Limiter II	4
Gate II	4
Expander/Gate II	4
Mod Delay I/II (short)	4
Mod Delay I/II (slap)	4
Mod Delay I/II (medium)	4
Mod Delay I/II (long)	4
Mod Delay II (extra long)	4
Dither	4
POW-r Dither	4
Time Adjuster	4
Signal Generator	4

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